Line-Arrays – The Hype Goes On

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Line-Array-Technology still remains a topic that greatly polarises opinion – on the one hand prejudiced scepticism and on the other a euphoric faith in technology. The following is an attempt to develop an objective perspective through technical understanding.

Why is the use of line-arrays becoming ever more popular nowadays? To explore this we first begin by considering the alternative: the cluster consisting of conventional horn loudspeakers.
As a rule, the only reason to create a cluster of conventional loudspeakers is to achieve a higher sound pressure level than is possible with one loudspeaker on its own.[1] Amplification of the sound pressure level is thus achieved in two quite distinct ways:

1. **Coupling of the radiated energy in the bass range**

2. **Directivity and separation of the radiated energy in the treble range**

Let us first take a look at coupling the individual systems in the bass range. When two sources radiate one and the same signal of the same frequency, they generate interference in the room. Wherever the signal is received with the same phase length, the amplitudes are aggregated and result in a constructive interference. The same applies to points where a differential delay occurs of one wavelength or a multiple thereof. At points where the differential delay is half a wave length or a multiple thereof, it results in a destructive interference; in this case the amplitude is theoretically zero.

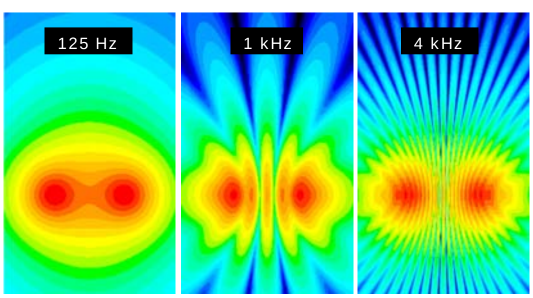

Figure 1 **Interference pattern of two point sources 1 m apart**

In the bass range of our loudspeaker cluster we now have the particular case that the wavelengths are long compared to the distance between the loudspeakers. This implies that the individual bass loudspeakers are so close together with respect to the corresponding frequencies that no destructive interference occurs at all. As a consequence, there is only constructive interference, which is expressed through an increase in the level of bass.

As already indicated, in the cluster formation a treble range increase in sound intensity occurs in a different way. In this case, due to the short wave length, bringing the individual transducers so close together that they radiate coherently – i.e. without destructive interference – does not work. The solution here is to increase the sound intensity by giving the individual treble horns a greater directivity.

You can visualise this using the analogy of a spotlight, whose light emission you more tightly focus so as to achieve a greater light intensity, even though the luminous capacity of the light source remains constant.

Of course, stronger directivity of the sound likewise causes a reduction in the opening angle of the treble horn. Accordingly, an additional treble horn is now required to again achieve a wide radiation range.

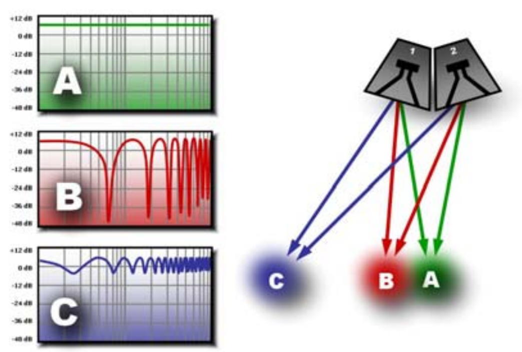

Figure 2 **Principle of cluster-formation**

Consequently, the purpose of a cluster in the treble range is for *one* treble horn to always project sound to only *one* particular part of the room, whilst minimising the unavoidable overlap areas, since it is here that destructive interference occurs due to large differences in wavelength.

Incidentally, the overlap of the radiated areas should be selected in such a way that it occurs exactly at the so-called nominal angle of radiation. By definition this lies on the polar diagram between the two points at which the sound intensity has fallen by 6 dB compared to the maximum. This ensures that the constructive interference in the overlap area is never greater than the sound level on the loudspeaker's main axes of radiation.

[1] Certainly, another reason can be to extend the sound range, in case the opening angle of a 90° horn is no longer sufficient. However, this is ignored in this case.

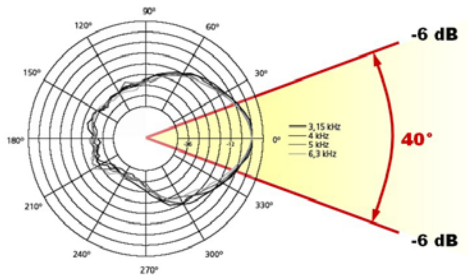

Figure 3 **Nominal angle of radiation**

Nevertheless, it is not possible to stop destructive interference occurring in the overlap area, whereby it intrinsically compromises the treble radiation in the conventional cluster.

In fact, even the bass radiation is not exempt from disturbance. Commonly, in typical cluster arrangements the loudspeakers are positioned horizontally adjacent to each other. Yet, this results in a narrowing of the bass radiation in precisely this plane – namely the horizontal, whereas in the treble range this causes the angle of the loudspeakers to widen.

Aside from these two deficiencies, you can easily appreciate that it is likewise impossible to have a seamless crossover between the two modes of operation for the bass and the treble range and it inevitably results in parasitic effects.

And so – in a way the well-intentioned reader would scarcely expect – this takes us straight to the modern line-array. In fact, the method is the same in terms of increasing the sound pressure level by coupling low frequencies and increasing the directivity of higher frequencies. However, it eliminates to a great extent the afore-mentioned deficiencies of the conventional cluster formation, wherein the vertical directional characteristic of the individual line-arrays plays a decisive part.

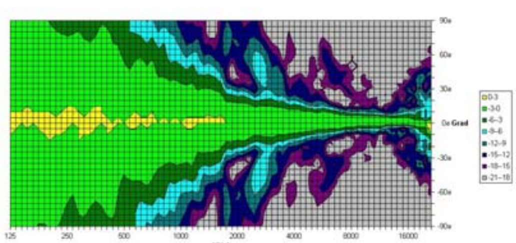

Figure 4 **Vertical isobars of a line-array**

Above all, this is evidenced by a continuous reduction in the opening angle towards higher frequencies. As a consequence, while a conventional cluster loudspeaker uses a constant directivity horn that has the same opening angle over as wide a frequency range as possible, the line array is aimed at generating an increased directivity at higher frequencies. For a line-array element this results in an increase in 'sensitivity' to higher frequencies, which is expressed tangibly in a high level of treble sound. Of course, this automatically implies that each individual line-array element has a non-linear frequency response. All the same, at this point we shall not go into why we want each individual line-array element to behave in precisely this manner – i.e. a continuous increase in sensitivity and narrowing of the opening angle towards high frequencies – and return to this issue later instead.

Next, let us look again at the way the line-array increases the sound level in the bass range. In this case the individual transducers are again so close together that they couple in the same way as in a conventional cluster. They operate to some extent as a single unit, resulting in an increase in the sound pressure level.

Up until this point there is no major difference from the conventional cluster – so, why a line-array? In the case of the conventional cluster formation we determined three disadvantages:

1. **The directivity in the bass range occurs contrary to the widening of the radiation in the treble range.**
2. **In the overlap areas of the treble radiation there is destructive interference.**
3. **The crossover from the mode of operation of the bass range to the mode of operation of the treble range is indeterminate.**

It is possible to effectively counteract point one by reverting to the old virtues of sound system technology such as "stacking and splaying". This involves lining up the transducers one on top of each other vertically and attempting, through the directivity of this arrangement, adjusting the radiation in the bass range to match the radiation in the treble range.

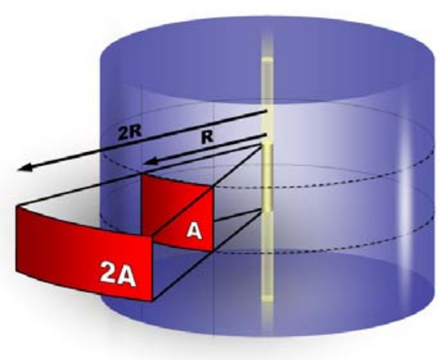

Figure 5 **Section taken from the radiation behaviour of a line source**

In the bass range this creates a quasi interlinking and infinitely extendable area of radiation. Under certain circumstances its size increases to such an extent that it can no longer be disregarded and it results in the emergence of a so-called near-field. This near-field is characterised by the fact that each time the distance is doubled, the sound level decreases on average by 3 dB. This behaviour is comparable to that of a linear transducer, where the waves spread out in form of a cylinder and a doubling of the radius results in a doubling of the surface area.

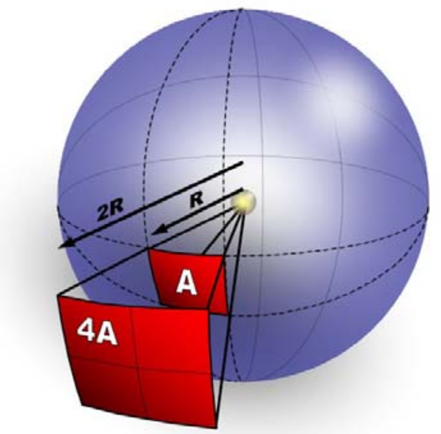

Figure 6 **Section taken from the radiation behaviour of a point source**

Beyond this near-field it is again necessary to disregard the size of the transducer arrangement. Once again, the behaviour can be described in terms of the character of a point source, where doubling the radius cause the sound level to decrease by 6 dB.

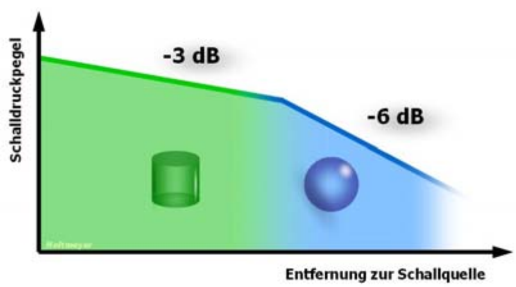

Figure 7 **Sound intensity gradient in near-field and far-field**

A transition occurs, therefore, from a near-field with the character of a line source to a far-field with the character of a point source. This introduces two important aspects that later play another important role.

However, beforehand, we should mention the solution to the other two problems of the conventional cluster. In a line-array the small vertical opening angle is a key factor in suppressing destructive interference in the treble range. As already described, this is generated not by CD horns but by so-called wave-guides, that bring about the continual narrowing of the opening angle.

A small opening angle automatically implies a relatively planar wave-front at the exit of the waveguide. If the curve of the wave-front is smaller than a quarter of the wave length, the individual wave-fronts of the wave-guides, which are arranged one on top of another, form a common coherent wave-front. They operate quasi as a single source.

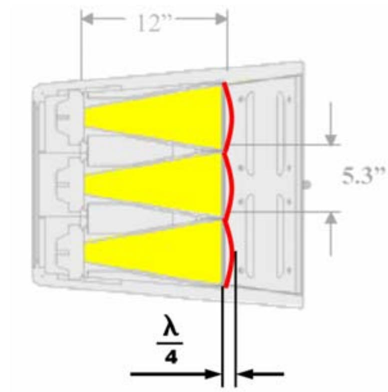

Figure 8 **Curving of the wave-fronts**

No doubt, the most frugal method of generating such a lightly curved wave-front is to use a simple conically-formed horn, whose output height is small in relation to its length. All the same, many other intricate systems of sound propagation have also been developed to obtain as narrow an opening angle as possible.

Contrary to current opinion, it is not necessary for the wave-front to be completely planar. In fact, the express requirement is for a minimal opening angle (and, therefore, not a planar wave-front!), since this enables the individual wave-guides of the respective line-array elements to be set slightly at an angle to each other, without the wave-front disintegrating. As a result, the choice of the appropriate opening angle always necessitates a compromise. On the one hand, the preference is for a small opening angle to enable a radiation that is as coherent as possible, even when the elements are only slightly angled towards each other; on the other hand, it would be beneficial to have a wide vertical radiation even with only a few elements, which necessitates, if anything, a large opening angle.

Even when the wave-front is perfectly straight, the wave-guides, due to their mechanical separation, must always be considered as separate sources, whereby at some distance away, due to the far-field conditions, an opening angle emerges again. For example, in the case of a flat wave-front with a height of 0.20 m at 8 kHz the near-field would cover no more than half a metre and a -6 dB opening angle of around 15° would emerge.

To form a coherent wave-font, it is sufficient, therefore, simply to line up an adequate number of sources, each with a corresponding minimal opening angle.

Likewise, a conical loudspeaker can be considered as a specifically dimensioned source. In this case, whereas no consideration needs to be given to the frequency dependent expansion of the near-field, it is necessary, however, to take account of the directivity behaviour. At higher frequencies a conical loudspeaker likewise tends to directionally focus the sound ever more tightly.

In a modern line-array the trick is to use sources with the appropriate opening angle for each respective frequency range and to configure the transitions in the frequency response as optimally as possible. The idea, therefore, is to create the optimum opening angle for each frequency. This results in a radiation pattern, as demonstrated in figure 4.

Consequently, the coherent overlay effect, which in the case of the conventional cluster only occurs in the bass range, is extended in a modern line array into the mid and treble range. You could also say that range B in figure 2 tends towards zero. As a result, it eliminates the conventional cluster's other two deficiencies (2 and 3), as described on page 2.

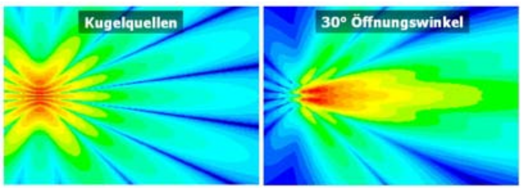

Figure 9 **Two spherical sources and two sources with an opening angle of 30° each 30 cm apart at 4 kHz**

Let us return to the near-field and far-field behaviour of a linear transducer arrangement. Were the transducer arrangement to be infinitely long, the sound field would only widen in one plane and the consequence would be a cylinder wave only. Since the arrangement is not infinitely long there is a transition from a near-field with the character of a line source to a far-field with character of a point source. By approximation the transition from near-field to far-field can be calculated using the following formula:

$$d_{Border} = \frac{h^2 f}{2c}$$

where h is the size of the transducer arrangement in m, f is the frequency in Hz and c the sound velocity in m/s.

As you can see, the spatial extension of the near-field is heavily dependent on the size of the transducer arrangement. The greater this is the more extensive is the near-field. Likewise, it is frequency dependent. The near-field extends much further for higher rather than for lower frequencies, which implies that at greater distances higher frequencies result in a tonal displacement.

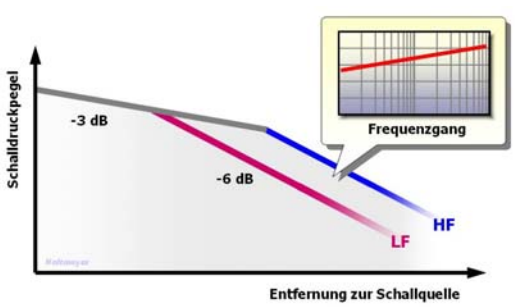

Figure 10 **Transition from near-field to far-field for a straight linear transducer at different frequencies**

The directivity in the direction of the spatial extension of the transducer arrangement (vertical) is likewise dependent on the size and the frequency. The vertical -6 dB opening angle can be determined according to:

$$D_v = 2\sin^{-1}\left(\frac{0{,}6}{3hf}\right)$$

where h is again the size of the transducer arrangement in m and f is the frequency in kHz.

It follows that the vertical opening angle is inversely proportional to the size of the transducer arrangement and the frequency. The larger the size of the transducer arrangement, the smaller is the opening angle and, likewise, as the frequency increases, the opening angle narrows.

This demonstrates that a tonal displacement exists at low frequencies in the vertical outside the main axis of radiation.

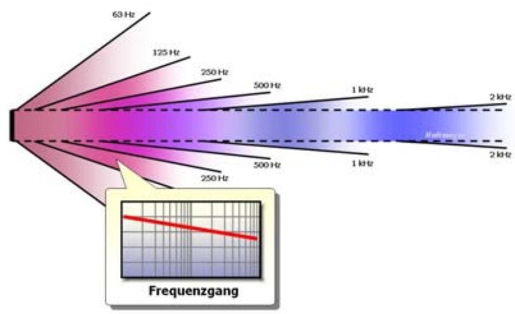

Figure 11 **Vertical opening angle for a straight linear transducer**

Therefore, we can determine two properties for the linear transducer arrangement that are, in fact, inappropriate for sound projection tasks:

1. **The frequency response on axis becomes increasing heavy on the treble the greater the distance you are away.**

2. **Outside the symmetry axis the frequency response becomes increasingly heavy on the bass as the angle widens.**

The two problem issues are addressed by what is called 'curving'. This involves the location dependent operation of the linear transducer arrangement for different frequency ranges, through a kind of length adjustment. Or in other words: alignment of the individual sound sources enables optimisation of the sound intensity picked up across all points on the listening area.

By way of clarification, two examples:

1. Distant listening location
Given a linear and uniform alignment of all sources, excess treble energy would be expected due to the near field extending further in the treble range.

By changing the angle of the lower sources less treble energy reaches the distant listener due to the strong directivity in the treble range. However, in the bass range radiation is approximately sphere-shaped, while the directional change of the lower sources is negligible. This balances the frequency response for the distant listener.

2. Close-by listening location
Given a linear and uniform alignment of all sources, excess bass energy would be expected due to increasing directivity at higher frequencies. To some extent the treble energy would go over our listener's head and he would only be affected by the minimal directivity in the bass range.

The alignment of these lower sources on this listening area increases the amount of treble energy supplied to this area and balances the frequency response.

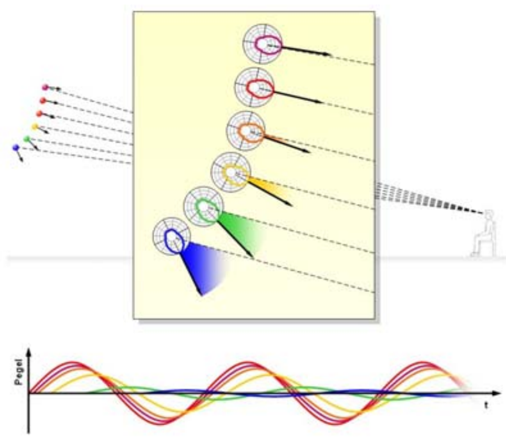

Figure 12 **Polar diagrams of the individual sources of a line-array and temporal sound intensity gradient of the individual sources at the listening location**

Due to the way the angle between the individual elements of a line-array increases progressively from top to bottom – so-called progressive curving – it is possible to create a homogenous sound level distribution with a balanced frequency response.

Further advantages of line-arrays

Up until now we have described how the modern line array eliminates the negative effects of the conventional cluster formation. This alone already offers considerable advantages that are even apparent in small line-arrays with a few elements. However, as the length increases, two other fundamental benefits emerge:

1. **The near-field extends out further.**
2. **The vertical radiation in the bass range becomes increasingly more defined.**

As already described, the transition from near-field to far-field is both frequency dependent and dependent on the length of an array. Curving quasi enables the near-field to be shortened to such an extent in the treble range that it equates to the near-field in the bass range. From this it is easy to see that the coverage of a short array is much less than that of a long line-array. The longer the array, the greater the coverage.

As a result, the mistake is often made of deriving a 'lower frequency limit' of line-arrays. However, this value is misleading, since the effect here is continuous. The correct approach would be to state that, under the desired conditions of a homogenous sound distribution, the homogeneity would begin to decrease from this frequency onward.

Clearly, the near-field as such causes other problems of understanding. It is not possible to experience it acoustically, when, for example, you move towards a line-array, unless you are directly involved in features such as 'in-your-face-sound' or 'headphone effect', which are often associated with line-arrays. In fact, you need to imagine the near-field as an intensive medium of energy transmission over the distance between loudspeaker and listener. In this context, it is important to understand that, if the line-array is correctly aligned, every listener sits in the far-field.

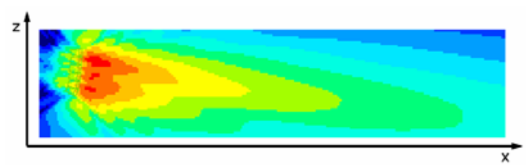

Figure 13 **Vertical sound intensity distribution of a line array with adjusted near-field for different distances to the listening area**

The other additional advantage of a long line-array is the increased directivity properties in the bass range. In order to directionally focus low frequencies, it needs to be dimensioned to equate roughly to the wavelength. This implies that the directivity properties towards low frequencies improve as the length of the array increases. This enables the radiation behaviour in the bass range to be accurately adjusted to that in the treble range. It makes it possible to create radiation behaviour that projects a lot of energy towards the back of the auditorium and, in accordance with the listening area, corresponding less energy into the front section. A further big benefit is that the radiated energy is focussed on the public and so in many areas is almost completely absorbed. As a consequence, where necessary, relatively little energy ends up in the room and the diffused sound is correspondingly low.

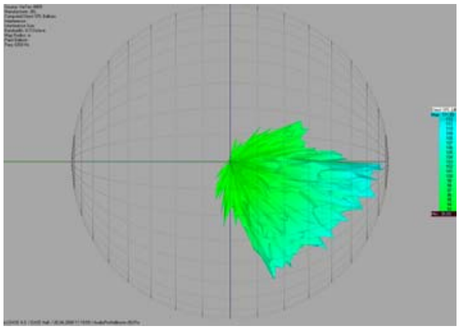

Figure 14 **Vertical radiation behaviour of a long line-array**

On the other hand, the horizontal radiation of a line-array is relatively insignificant. It is determined by the horizontal opening angle of each individual element of the line-array and is for the most part unaffected by the length of the array. That is to say, it is not dependent on how many elements are suspended under each other. The nominal horizontal radiation angles of standard line-arrays are mostly in the region of around 90°.

Compared with conventional cluster loudspeakers this appears at first quite wide and sometimes leads to the suspicion of being unsuitable for certain applications, where, for example, it is feared there might be reflections from side walls. However, you just need to think from the perspective of the loudspeaker and look into the room to see that this concern is unjustified. Basically, what we see is the ceiling, the floor and the back wall. In our field of view the surface area taken up by the side walls is relatively small (a full audience means that the floor as well as parts of the back wall and the side walls are excluded, yet this only changes the ratios marginally). As a consequence, it is much more important to have an accurate directivity in the vertical rather than in the horizontal plane.

Added to this is the effect that the parts of sound that reach the listener via side wall reflections are, for the most part, temporally very close to direct sound. In this context the acoustician talks of so-called 'beneficial reflections' (approx. 10 – 30 ms or 3 – 10 m wavelength difference). These increase the impression of loudness as well as the intelligibility. Only reflections with a greater delay appear as disruptive. However, you would expect these to come from the back wall first of all, due to the longer delay time. Consequently, a wide horizontal radiation is considered a good compromise. In narrow venues its effect is not in any way as negative as is commonly believed.

However, as has become clear, the vertical radiation of a modern line-array is very flexible and dependent on different parameters. Basically, for a standard array these are:

1. **Angle of the individual elements to each other**

2. **Vertical alignment of the whole array**

3. **Height of the array**

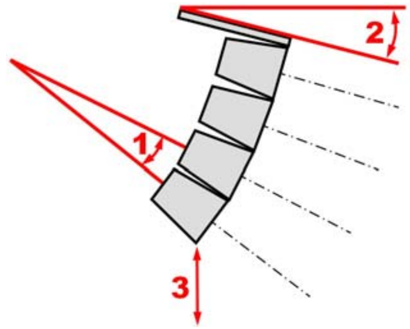

Figure 15 **The three most important parameters for the configuration of a line-array**

Most manufacturers offer special soft-ware solutions for their line-arrays with which it is possible to calculate the distribution of the sound level in the listening area along the array's main axis of radiation. Amongst others, these provide the three values above. In this process it is essential to simulate the sound level distribution at different frequencies, since the radiation of the array can very often turn out to be very unequal.

A method proven in practice is to always optimise the sound distribution in the treble range (8 kHz), before regulating the sound distribution at other frequencies. Particularly in the treble range, minor variations in angle can produce clearly audible changes that are, therefore, the easiest to assess.

The basic aim is to achieve a progressive, spiral shaped curving, which implies aiming for small angles between the upper elements and larger angles between the lower elements. As a result, the angles progressively increase as you go down. Whereby, we do not mean the actual angles between the mostly trapezium shaped housings, but the angles, as it were, at which the elements' main radiation axes drift apart. For example, a progressive curving emerges automatically for a flat listening area, when the points of impact of the main radiation axes of the individual elements are equidistant on the listening area. Some programmes also use this approach in order to provide an automatic array configuration.

When optimising the sound level gradient, thought should be given to whether the sound system will operate in the open air or in an enclosed room. The limited length of the line-array means that in the bass range (to a great extent independent of curving) it is not possible to achieve a homogenous distribution of direct sound. This leads to a perceptible decrease in sound level over distance. As a consequence, in open air situations it is appropriate under certain conditions to adjust the sound level in the mid and treble range to the gradient in the bass range, even when it would certainly be possible to achieve a homogenous sound intensity gradient in the treble range. This implies that while the overall sound intensity at the back of the auditorium is indeed lower than at the front, the spectrum is nevertheless more balanced – i.e. you achieve a more homogenous frequency response for all listener locations.

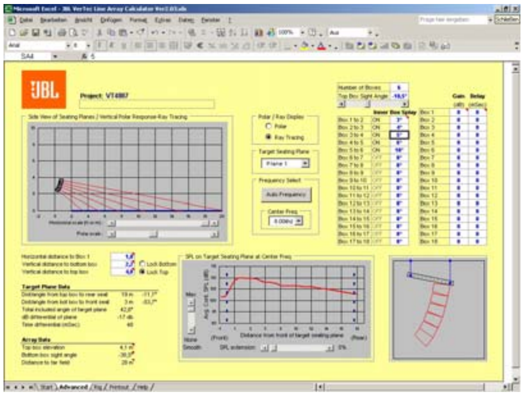

Figure 16 **Prediction software with progressively angled line array and gradually declining sound level gradient**

The difference of using a sound system in a hall etc. is simply the fact that the room acoustics plays a part. First of all, this implies adding the sound intensity of a so-called diffusion or reverberation field to that of the direct sound. From a statistical point of view, the sound level of the diffusion field in a room is uniform in level throughout. Even when a loudspeaker array has a very high directivity and the room a comparatively short reverberation time, it only takes a few metres for it to be outweighed by the sound level of the reverberation field. By implication, the reverberation field does have a big influence on tonal perception. Since generally the absorption is greater in the treble range, the reverberation field spectrum is, as a rule, heavy on bass. As a consequence, the addition of the reverberation field results primarily in a reinforcement of the bass part. For the back section of the auditorium the effect of this is to 'top up' the bass range to some extent. This means that it is certainly permissible to aim for a homogenous sound level gradient in the mid and treble range over distance, even when the calculation programme indicates that the sound level tails off in the bass range.

Besides, experience has shown that it certainly does not always make sense to suspend the array as high as possible. The higher the array is hung, the wider you have to set the vertical opening angle. When only a few elements are used, you are restricted by the maximum possible angle between the elements. For example, with four elements and an angle between each of 10° it is indeed possible to achieve an opening angle of 40°. However, the advantage of free scalability comes into play, whereby more energy can be projected into the back of the auditorium using smaller angles in the top section of the array.

Furthermore, in the case of long arrays a strong directivity emerges in the mid range. If such an array is 'flown' very high and, consequently, is set at a relatively steep angle, it is possible that the narrow vertical radiation of the mid range is not sufficient to uniformly cover the whole intended area of acoustic radiation. As a consequence, it often makes greater sense to hang the array lower down and to curve less intensely. In so doing you project out uniformly over the auditorium with the narrow mid range radiation and, additionally, have more possibilities of varying the angle in the upper section of the array.

We finish by taking another look at the necessary signal processing of a modern line-array. The basic rule is that a temporal adjustment is only required in exceptional cases or to achieve a particular result. That is to say, you do not need a delay for a standard set-up. The same applies to the sound level adjustment to the respective transducers of individual elements

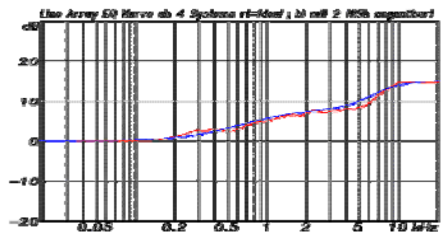

Figure 17 **Typical filter to compensate for the coupling in the bass range**

Equalisation is a different altogether. It is totally essential for the operation of a line array. Due to the stronger coupling at lower frequencies the level must be raised towards higher frequencies. This is mostly achieved using one or two high shelf filters. The longer the array, the greater the necessity to increase the level of the high shelf filter

Conclusion

A modern line-array works in very much the same way as a conventional cluster. However, it tackles head on the three integral deficiencies of the conventional cluster formation and eliminates them.

Furthermore, two further benefits emerge for long line-arrays: the formation of an extended near-field enables large distances to be bridged and, thanks to the scalability of the line-array, the vertical radiation characteristics can be definitively adapted to the intended area of acoustic radiation.