# Taking the "virtual" out of arrays

**Rebuilding your loudspeakers may prevent overlapping of sound at lower frequencies.**

Arraying multiple full-range loudspeaker systems to produce increased sound-pressure levels, wider coverage or both is not a straightforward proposition. Whereas a single loudspeaker can be designed to resemble a point source, several such sources connected to a coherent audio signal will always interfere with each other. From a listener's perspective, the signals interfere with each other because they are arriving from every one of these sources and are traveling from different distances and directions, making for anything but uniform coverage. For

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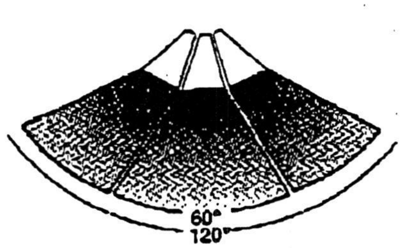


**Figure 1. Typical overlap areas of a conventional array in the horizontal plane at frequencies from 2 kHz to 20 kHz.**

instance, at a specific frequency, a signal from one source could totally cancel the signal arriving from another source that happens to be a half wavelength farther away or closer. This total cancellation is an extreme case, but partial cancellations and reinforcements at different frequencies to a larger or smaller degree occur throughout the audience area. The audible effects are often pronounced changes in frequency response as a function of listening positions within the intended coverage area. Unfortunately, our hearing system is quite sensitive to changes in frequency response.

Convex circular arrays have become the standard solution for minimizing these interference problems. In these arrays, the loudspeakers are equipped with acoustical horns that aim the higher frequencies into a limited conical space, and then individual loudspeakers are splayed to point these coverage cones away from each other.

At least at these specially controlled frequencies, usually above 1 kHz to 2 kHz, large areas within the coverage field get a greater amount of direct sound from only one horn. This minimizes interference effects from all other array sources.

However, frequencies below 1 kHz are not easily directed and therefore usually have more interferences in any array. Also, because high- and mid-frequency horns cut off gradually, there will always be areas of noticeable overlap. Interferences at frequencies above 1 kHz in the overlap areas can be just as pronounced as they are at lower frequencies throughout the entire audience area.

Although acoustical horns can be aimed enough away from each other to keep the sound-pressure levels at mid- and high-frequencies roughly constant throughout the coverage area, lower frequencies overlap and reinforce each other more as frequency decreases. This effect gives larger arrays an undesirable boost at low frequencies, and anywhere outside the direct mid to high-frequency horn coverage, low frequencies prevail audibly. Low-frequency overlap effects and bias can be controlled to some degree, but neither can be eliminated. No wonder nobody claims perfect performance, even for circular arrays.

## Is there a way around interferences in arrays?

A new method of arraying comes much closer to ideal array coverage than has previously been achieved even with the best available conventional systems. This new array construction is incorporated into two Renkus-Heinz TRAP series. The larger TRAP 40 series, with control above 1 kHz, has modified Renkus-Heinz Coentrant horns and a single 15 inch (381 mm) woofer. The smaller TRAP Jr. series, with control above 2 kHz, uses Renkus-Heinz Complex-Conic horns and a single 12 inch (305 mm) woofer.

**Figure 2. ALS-1 predictions of interference patterns at three different frequencies. Forward gain of 10 dB and quite severe interference occurs at higher frequencies.**

## Quantifying array interferences

In 1985, Renkus-Heinz commissioned Rex Sinclair of Sinclair Consultants to write a computer program to calculate and graphically display general array performance. This simple lobing program, ALS-1, has been very useful for custom line array designs and predict-

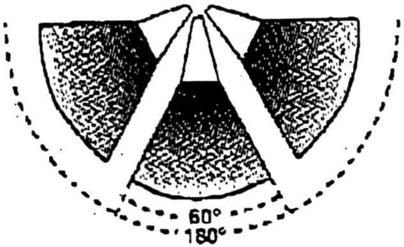


Figure 3. Doubling the splay between the cabinets to 60° reduces interference areas and improves even coverage. The coverage angle is increased to 180°. This wider and more uniform coverage comes at the expense of 10 dB of forward gain.

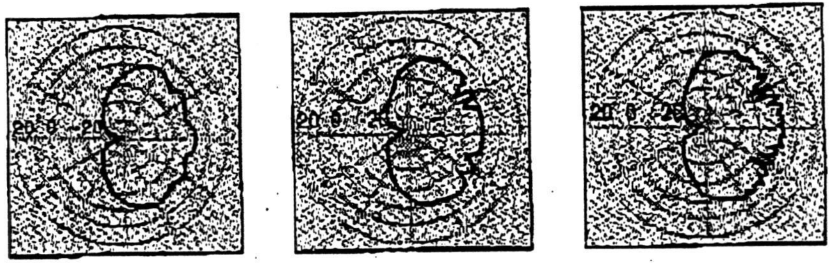

Figure 4. ALS-1 results at 1 kHz, 2 kHz and 4 kHz. At higher frequencies individual horns are now clearly discernible. Note that remaining interferences deepen with increasing frequencies.

ing directional characteristics of low-frequency line arrays.

In a typical conventional convex circular array, each high-frequency horn is designed for 60° horizontal coverage with a cutoff frequency of about 1 kHz, and the loudspeaker cabinets are made with a 15° horizontal array angle. The three cabinets are close-angle. The three cabinets are close-coupled, meaning their 15° sides touch. This setup results in a splay of 30° between the cabinets. Total intended -6 dB coverage is 120°.

One would assume that this 30° splay with 60° horns is ideal. Indeed, the commonly used 15° cabinets strongly suggest that for optimum array performance, the loudspeaker cabinets should be arrayed close-coupled. However, this is not the case at all.

Figure 1 shows the typical overlap areas of a conventional array in the horizontal plane at frequencies from 2 kHz to 20 kHz. Figure 2 shows ALS-1 predictions of interference patterns at three different frequencies. Forward gain of 10 dB and quite severe interference occurs at higher frequencies. Therefore, this is not the best arrangement for even coverage. However, if forward gain is important, then this array configuration will work.

It is possible to improve by experimenting with splay angles. Figure 2 suggests a splay angle larger than 30° might show an improvement. For minimum interference areas and most even coverage, doubling the splay between the cabinets to 60°, as shown in Figure 3, yields much better results. The coverage angle is increased to 180°. This wider and more uniform coverage comes at the expense of 10 dB of forward gain. (See Figure 1.)

Figure 4 shows the ALS-1 results at 1 kHz, 2 kHz and 4 kHz. At higher frequencies individual horns are now clearly discernible. Note that remaining interferences deepen with increasing frequencies. Array performance as shown is typically the best that can be expected. A larger splay angle will cause lack of coverage in between

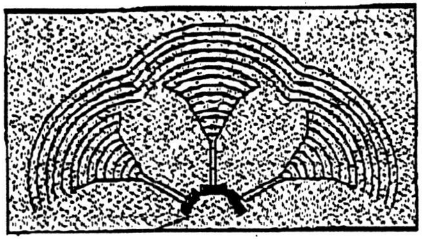

Figure 5. The wavefronts as they radiate from the points of origin. Typically, these points are separated in space, and the illustration clearly shows why interferences will happen at the coverage boundaries.

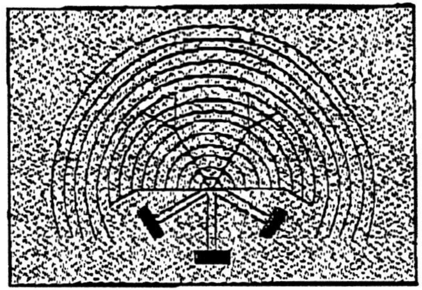

Figure 6. When the points of origin for the three wavefronts in Figure 4 coincide, this solves interference problems.

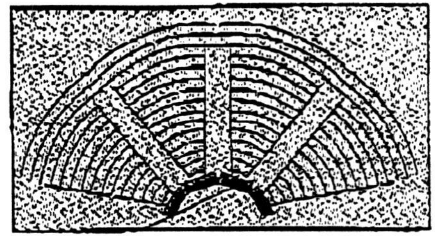

Figure 7. Locating the apparent apex of the horns almost to the rear of the cabinet improves array performance. Note that for the same nominal coverage of 120°, four cabinets are now required.

horns; smaller splay angles increase the areas of interference.

Figure 5 illustrates the wavefronts as they radiate from the points of origin: the point in each of the three horns from which the sound waves seem to originate. Typically, these points are separated in space, and Figure 5 clearly shows why interferences will happen at the coverage boundaries.

The placement and location of these points of origin in arrays turns out to be the all important key to any improvement in array performance. What would happen to the three wavefronts in Figure 4 if their points of origin could be made to coincide? Figure 6 shows this theoretical case, which would solve the interference problems.

Separated vertically, horns have been splayed as shown in Figure 6. However, the vertical displacement means that in the vertical direction one would again have a physical separation of the points of origin, which again equates with interference patterns.

## Have all points of origin coincide in one point in space

Of course, this is easier said than done. Loudspeaker system design imposes physical restraints, making this ideal case unattainable. However, once the principle is understood, a practical solution can come close enough to the theoretical one to significantly improve conventional designs.

By inspecting Figures 6 and 7 it becomes clear that loudspeaker cabinets with an acoustic origin moved very close to the rear of the cabinets should make for better array performance. To understand how this might be done, let's take a closer look at conventional constant coverage horns.

Most constant directivity horns exhibit astigmatism — their apparent points of origin are different in the horizontal and vertical planes. Typically the apparent apex in the wider coverage plane is farther forward toward the mouth of the horn, and the apparent apex for the narrower coverage plane is back toward the

# Minimizing interference in circular arc arrays

For an array far field dependence on angle is

$$SPL(\theta) = 10 \log P_0^2 \text{ dB}$$

For a distance to the listening area very much larger than the array dimensions, let the sound pressure $P$ be the real part of

$$P(\theta) = A(\theta)e^{j\omega t}$$

where $P$ is the sound pressure, $\omega$ is the angular frequency and $A(\theta)$ is a function of the angle between the array longitudinal axis and the direction of the distant listening point. It gives the ratio of the sound pressure due to the source as a ratio of its on-axis value at the same distance.

For the $n$th source shown in Figure A, assuming identical sources, the pressure contribution is given by

$$P_n = A_n(\theta)e^{j(\omega t - kS_n)}$$

where $k = 2\pi/\lambda = 2\pi f/c$, $\lambda$ is the wavelength, $f$ is the frequency, and $c$ is the speed of sound. $S_n$ is the distance by which the path length from the $n$th source to the distant point exceeds the distance from the origin to that point.

For an array of $n$ sources, the total pressure $P$ is given by the following equation:

$$P(\theta) = \sum_{n=1}^{N} A_n(\theta)e^{j(\omega t - kS_n)} = e^{j\omega t}\sum_{n=1}^{N} A_n(\theta)e^{-jkS_n}$$

The square of the pressure amplitude is given by

$$P_0^2(\theta) = [\sum_{n=1}^{N} A_n(\theta)\cos(kS_n)]^2 + [\sum_{n=1}^{N} A_n(\theta)\sin(kS_n)]^2$$

where $A_n(\theta) = A(\theta - \alpha_n)$.

For a circular arc array, the additional path length $S_n$ is as shown in Figure A. For the $n$th source at radius $R$ and

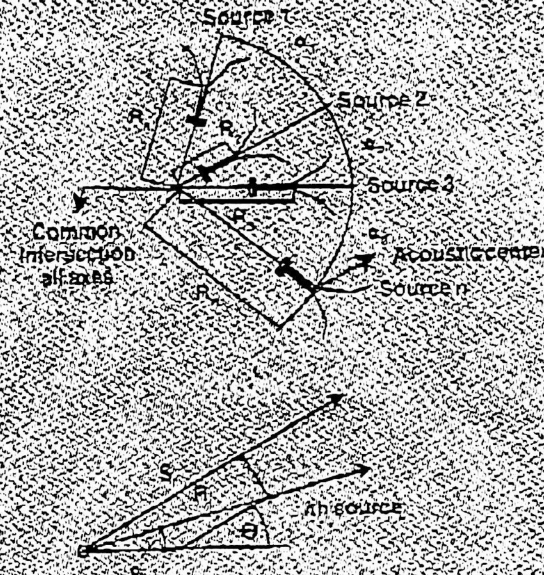


*Figure A. For a circular arc array, the additional path length $S_n$ is as shown.*

angle is given by

$$S_n(\theta) = R\cos(\theta - \alpha_n)$$

Therefore, the smaller the $R$ is, the smaller the $S_n$ differences. This means less interference. Ideally, $R=0$.

throat. Most, if not all, typical array loudspeakers are designed with the popular 60° (horizontal) × 40° (vertical) horns, which put the apparent apex quite forward. With this information in mind, Renkus-Heinz realized a better array could be made if one used the location of the vertical apex for the horizontal plane.

Figure 7 shows new horns with the apparent apex located almost to the rear of the cabinet. This is how we improved on array performance. Note that for the same nominal coverage of 120°, four cabinets are now required. No pain, no gain!

Figure 8 shows the ALS-1 simulation of this new TRAP construction. A comparison with Figure 2 shows significant improvements in interference at all frequencies.

The horn design of the TRAP 40 series cabinets ensures broadband pattern control down to the frequency at which mutual coupling between adjacent cabinets ceases. Its cabinet design provides optimum splay angles and maintains coincident acoustic centers for adjacent cabinets. A TRAP cluster of three behaves as a constant directivity point source with even frequency response throughout its coverage. Figure 9 shows measured horizontal and vertical polar responses.

With new designs, considerable improvement in array performance is possible. Frequency response variations within the intended coverage area can be held to ±4 dB. This performance is possible out of the box without applying micro-delay or frequency shading techniques. S&VC

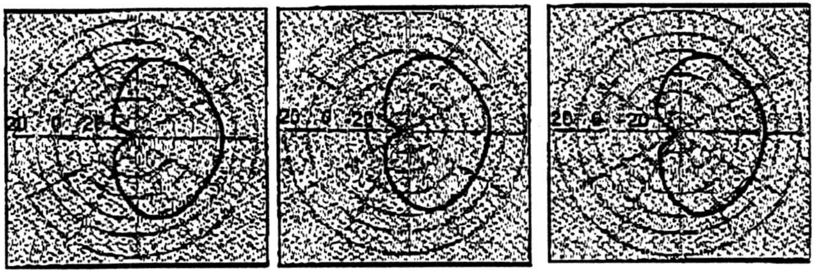

Figure 8. ALS-1 simulation of the TRAP construction.

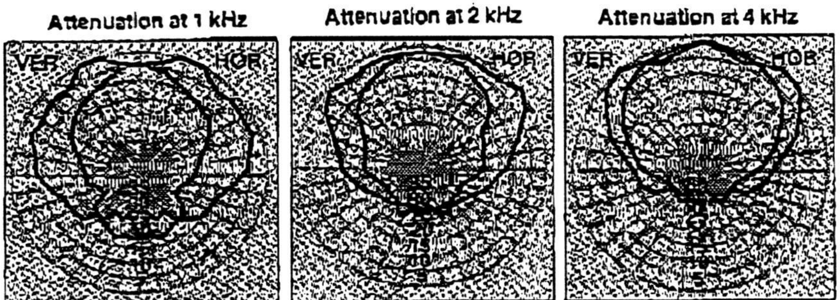


Figure 9. A TRAP cluster of three behaves as a constant directivity point source with even frequency response throughout its coverage.

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