TRAP [TRue Array Principle] Design

Integrating Arrayable Systems with Mathematically Correct Topologies

A Renkus-Heinz Engineering White Paper

by Ralph D. Heinz, Vice-President, R&D

TRAP (True Array Principle) Design

Integrating Arrayable Systems with Mathematically Correct Topologies

Why Array?

For the purposes of this discussion we can define a loudspeaker array as "a group of two or more full-range loudspeaker systems, arranged so their enclosures are in contact." System designers use arrays of multiple enclosures when a single enclosure cannot produce adequate sound pressure levels, when a single enclosure cannot cover the entire listening area, or both. These problems can also be dealt with by distributing single loudspeaker systems around the listening area, but most designers prefer to use arrays whenever possible because it is easier to maintain intelligibility using a sound source that approximates a point source than by using many widely separated sources.

Array Problems and Partial Solutions: A Condensed History

First-generation portable sound systems designed for music used a very primitive form of array: they simply piled up lots of rectangular full range speaker systems together, with all sources aimed in the same direction, in order to produce the desired SPL. This type of array produced substantial interference, because each listener heard the output of several speakers, each at a different distance. The difference in arrival times produced peaks and nulls in the acoustic pressure wave at each location, and these reinforcements and cancellations varied in frequency depending on the distances involved. So although the system produced the desired SPL, the frequency response was very inconsistent across the coverage area. Even where adequate high frequency energy was available, intelligibility was compromised by multiple arrivals at each listening location.

Second-generation systems incorporated compression drivers and horn-loading techniques derived from cinema sound reinforcement and large-scale speech-only systems (the original meaning of "public address"). When these horns were incorporated in a single enclosure with trapezoidal sides that splayed the horns away from each other, the first "arrayable systems" were introduced to the marketplace. These products promised to eliminate lobing and dead spots (peaks and nulls) and to drastically reduce comb filtering (interference). They did improve performance over the stack of rectangular enclosures loaded mainly with direct radiating cones. But frequency response across the coverage area remained inconsistent. In addition to the midrange and high frequency variations across the coverage area of the array, low frequency output varied from the front to the rear and side to side. Low frequency energy was focused along the longitudinal axis of the array and close to it, producing a "power alley" that gave the seats with the best views the worst sound.

Even when a single enclosure is designed to resemble a point source, multiple enclosures will always interfere with each other when connected to a coherent audio signal.

TRAP (True Array Principle) Design

Integrating Arrayable Systems with Mathematically Correct Topologies

Conventional Array Shortcomings: Pictorial Analysis

As we said in the first paragraph, the performance advantages of the array (whether horizontal or vertical) derive from its ability to approximate a perfect acoustical point source. But even the smallest arrays typically include three or more loudspeaker enclosures, each with two or three separate acoustic centers of its own. It's easy to appreciate that getting all those discrete sources to behave like a theoretically ideal point source is difficult in practice. Signal processing solutions attempt to compensate for the difference between theory and reality by sacrificing the coherency of the electronic signal. They apply frequency shading

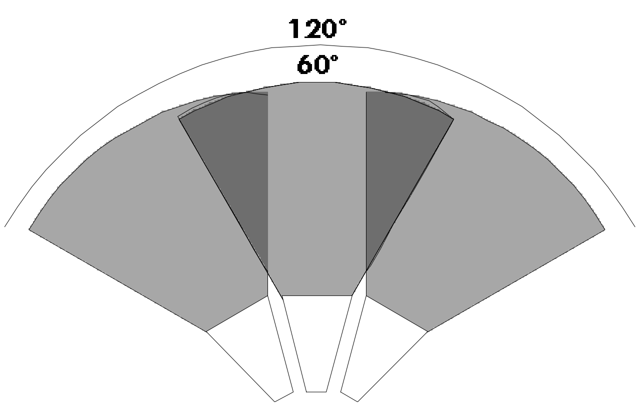
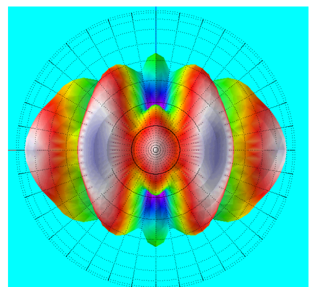
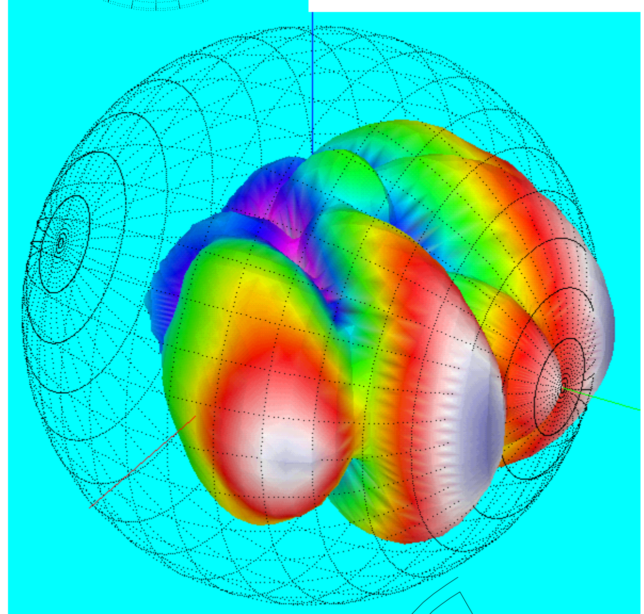


Fig. 1
A very common array uses three 60° x 40° horns in enclosures with 15° trapezoidal sides: tight-packed, this array produces substantial overlap and interference between adjacent horns.

and/or micro-delays to the signals sent to different enclosures, in order to ameliorate the acoustic problems. These approaches are costly, complicated and often meet with limited success. CoEntrant topologies (US patent #5,526,456) reduce the complexity of the problem by integrating midrange and high frequency transducers into a single acoustic source.

A rigorous analysis of the acoustical physics of the multi-enclosure array can point the way toward a practical, physical solution. First, consider what is probably the most common arrayable system in use today: 60° x 40° horns in enclosures with 15° trapezoidal sides (Fig. 1). Tight-packing three of these systems with their 15° sides touching produces a 30° splay between the horns, for a total included angle of 120°. At first glance, this seems like an ideal alignment. But the EASE interference predictions in Fig .2 show the





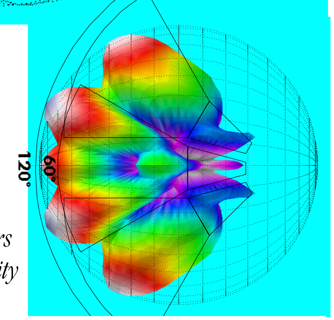


Fig. 2
The interference patterns shown above were produced by tight-packing three "arrayable" speakers using 60° x 40° constant directivity horns in enclosures with 15° trapezoidal sides. While this is an improvement over a pile of direct radiating transducers, it is far from the ideal point source array.

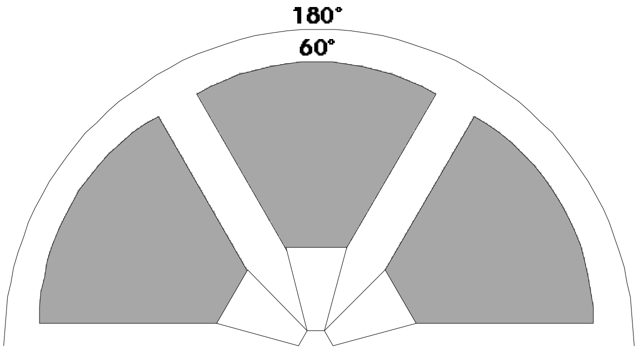


Fig.3
Widening the splay between horns reduces interference and widens the coverage angle to 180°, but reduces forward gain. As always, energy is conserved.

familiar and clearly audible problems with this configuration: significant interference above 1 kHz, with variations of 8 to 9 dB depending on the angle. On axis, there is about 10 dB of gain at frequencies below 1 kHz. Where maximum SPL is the main consideration, this type of array will deliver acceptable performance. When the front-of-house mix position can be located on the axis of left and right arrays, they can usually be "tweaked" to deliver acceptable reproduction in this limited area. Other areas of the house, including the "high roller seats" up front, will suffer.

The interference patterns displayed in Fig. 2 can be reduced by widening the splay between cabinets to 30°, as illustrated in Fig. 3. This array will not look as pretty as the first, but it does have much more even response across the coverage area (Fig. 4). At 2 kHz and 4 kHz, the individual horns are clearly discernible in the ALS-1 predictions. Also note that the "seams" between the horns become deeper with increasing frequency.

Fig. 5 shows why there will always be interference with conventional horn arrays (whether they are enclosed in "arrayable" cabinets with trapezoidal sides or mounted in free air). As the wavefronts radiate from points of origin that are separated in space, they will always create some interference at the coverage boundaries.

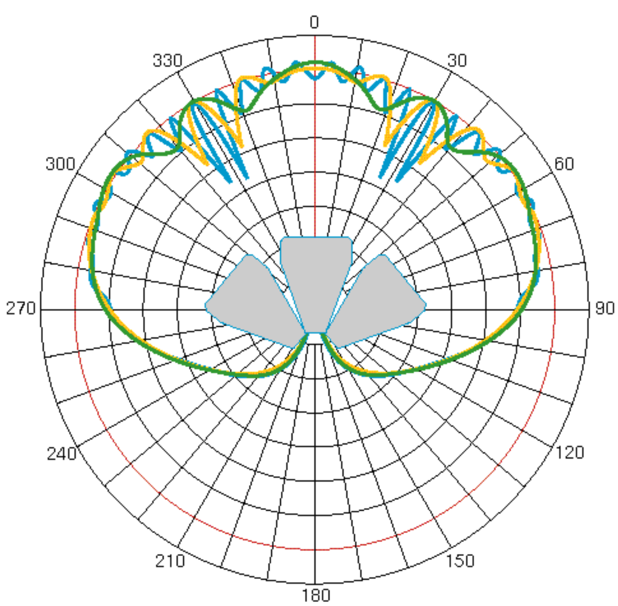


Fig. 4
ALS-1 interference predictions for a wider splay show reduced interference, but the three horns are clearly apparent at higher frequencies.

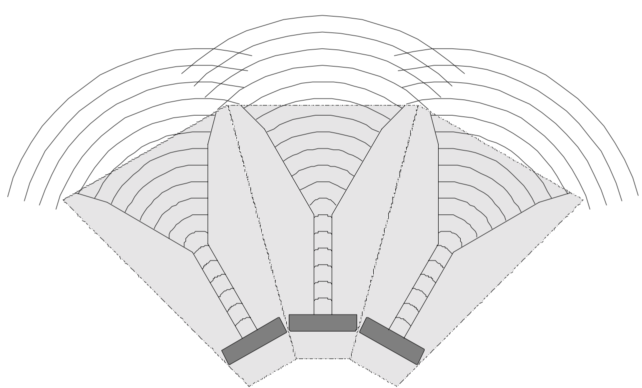


Fig. 5
The acoustic pressure wave expands as a sphere, and multiple spherical sections will always overlap unless they originate from a common center.

TRAP (True Array Principle) Design

Integrating Arrayable Systems with Mathematically Correct Topologies

Conventional Array Shortcomings: Mathematical Analysis

For an array in far field, dependence on angle is

$\mathrm{SPL}(\ ) = 10\log P_0^{\,2}\,\mathrm{dB}$

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

$P(\ ) = A(\ )\ ^{i(\quad kSi)}$

where *P* is the sound pressure,   is the angular frequency, and $A_i(\ )$ 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 i^{th} source shown in Fig. A, assuming identical sources, the pressure contribution is given by

$P_i = A_i(\ )\ ^{i(\quad kSi)}$

where *k*   c [is the wavelength,   is the frequency and *c* is the speed of sound]. S_i is the distance by which the path length from the i^{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

$P(\ ) = \ \ _i\ {}^nA_i\ ^{i(\quad kSi)} = \ \ ^{i}\ \ _i\ {}^nA_i(\ )\ ^{jkSi}$

The square of the pressure amplitude is given by

$P_0^{\,2}(\ ) = [\ \ _i\ {}^nA_i(\ )\cos(k\,S_i)]^2 + [\ \ _i\ {}^nA_i(\ )\sin(k\,S_i)]^2$

Where $A_i(\ ) = A_i(\qquad _i)$.

For a circular arc array, the additional path length S_i as shown in Fig. A, for the i^{th} source at radius *R* and angle   is given by

$S_i(\ ) = R_i\cos(\qquad _i)$

Therefore, the smaller R_i is, the smaller the S_i differences, and the less the interference between sources. Ideally, $R = 0$ for all sources. As *R* approaches 0, the interference will become less audible and frequency response across the array's intended coverage area will become more uniform.

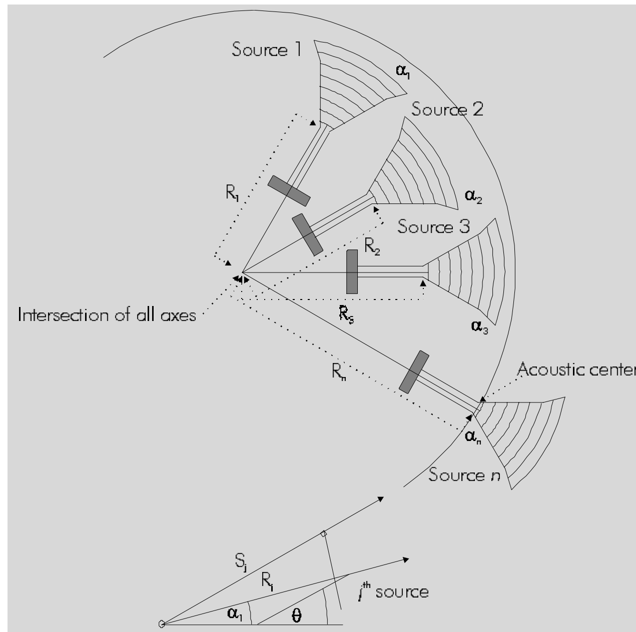


Fig. A
For a circular arc array, the additional path length Sj is as shown.

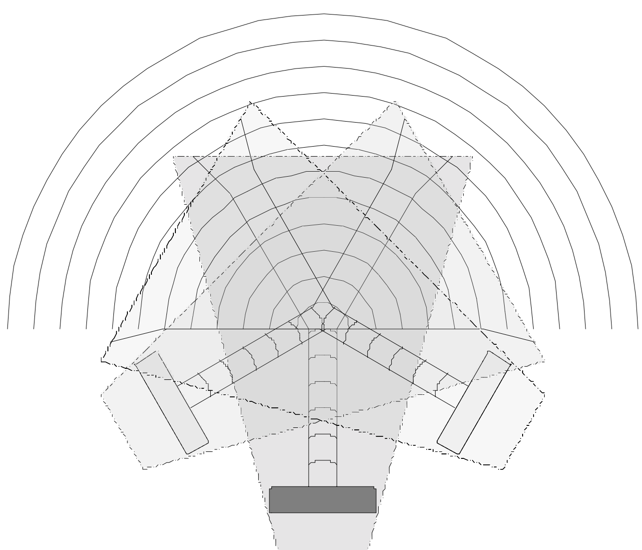


Fig. 6
The acoustic ideal - co-locating the acoustic centers of all horns - is not a practical possibility.

TRAP (True Array Principle) Design

Integrating Arrayable Systems with Mathematically Correct Topologies

Coincident Acoustical Centers: The key to true arrayability

Clearly, the ideal solution is to colocate all the acoustic points of origin, as shown in Fig. 6. We could achieve this by stacking the horns vertically, but this would solve the problem in the horizontal plane by creating a worse situation in the vertical (front to back of the listening area) direction. Fig. 7 shows a more realistic approximation that takes into account the physical constraints of loudspeaker design (the dimensions of the transducers, horns, enclosure walls, etc.). Because the acoustic sources are real physical objects, we cannot reduce R_i to 0. But we can get close enough to make measurable, audible improvements in the performance of the multi-enclosure array.

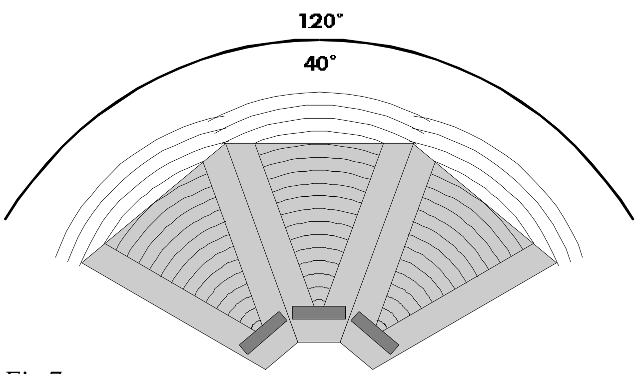


Fig.7
Because drivers and enclosures are physical objects, the acoustic centers of TRAP horns are not perfectly coincident - but they are close enough to achieve measurable and audible reductions in interference.

TRAP horns: a new approach

Fig. 7 implies that the way to minimize R_i – and the resultant interference - is to move the acoustic centers as far to the rear of the enclosure as possible. We can attempt to minimize the size of the drivers, for instance by using high-output magnetic materials such as neodymium. But the biggest obstacle to coincident acoustic centers is the horn itself. This is because typical constant directivity horns exhibit “astigmatism:” their apparent points of origin are different in the horizontal and vertical planes. In order to create a wider coverage pattern in the horizontal plane, the apparent apex is moved forward, while the vertical apex is farther to the rear because its coverage pattern is usually narrower. This is certainly the case with the most popular horn patterns in use today: 60° x 40° and 90° x 40°. One approach to a solution, then, is to use the vertical apex for the horizontal plane as well. This is the basic innovation behind TRue Array Principle designs: moving the acoustic origin as far the rear of the cabinet as possible, first by using the vertical apex as the horizontal apex instead of locating the horizontal origin far forward within the enclosure. Subsequent refinements to the horn flare itself have been awarded US Patent #5,750,943. This “Arrayguide”

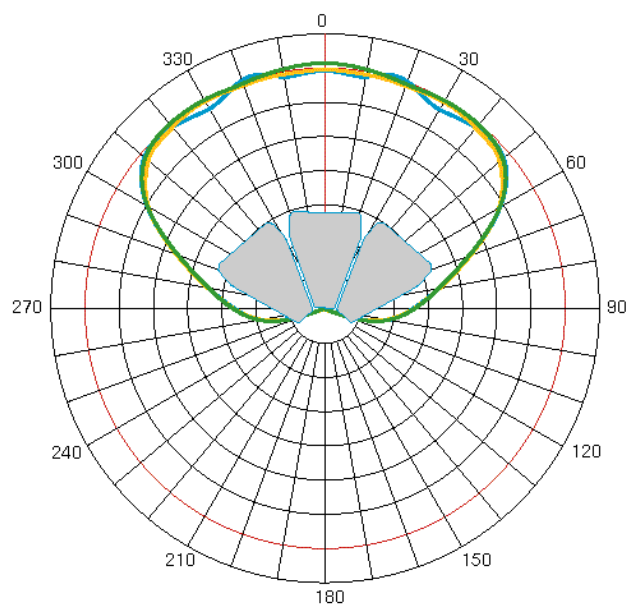


Fig. 8
TRAP design produces truly arrayable systems with minimal destructive interference in the horns’ passband.

topology goes even farther in locating the apparent acoustic origin toward the rear of the enclosure. To repeat, moving the acoustic centers to the rear

TRAP (True Array Principle) Design

Integrating Arrayable Systems with Mathematically Correct Topologies

minimizes R, the distance between acoustic points of origin within the array, and the resulting interference between array elements.

Fig. 8 shows the ALS-1 predictions for the first generation of TRAP horns. It is clear that interference has almost disappeared.

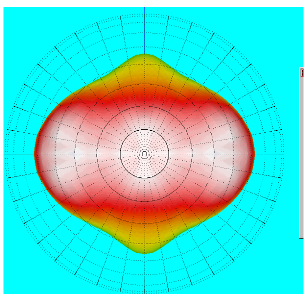
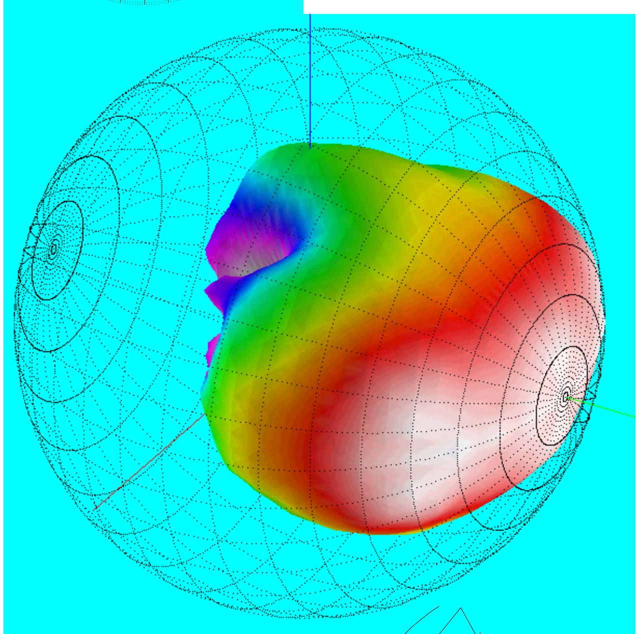


Fig. 9 shows measured EASE data for a three-wide array of TRAP40 enclosures. Frequency response is consistent in both vertical and horizontal planes within



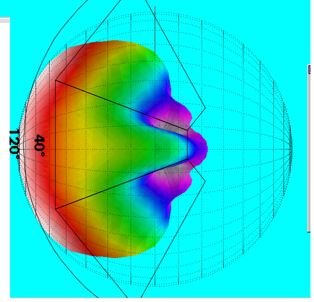


Fig.9
The TRAP array produces almost no measurable interference from a tight-packed three-wide cluster. This is because the three spherical wavefronts produced by the three horns originate from a common acoustical center. Therefore they behave as a single acoustic unit, without overlap or interference.

Fig. 10
TRAP arrays can be quite small: however, the size of the horns will determine the lower frequency limit at which the TRue Array Principle ceases to operate.

±4 dB. This is an "out of the box" array, using no frequency shading or micro-delay to improve performance. Measured results don't track the predictions 100% because the actual pattern of the horns varies somewhat with frequency: first generation TRAP horns maintain nominal coverage ±10° from 1 kHz to 4 kHz.

TRAP Performance

TRAP is a method for optimizing the mutual coupling between adjacent horns in an array. As such, the TRue Array Principle operates over the pattern bandwidth of the horns (the frequency range over which their coverage varies less than a defined amount, for instance ±10°). CoEntrant topology extends the pattern bandwidth of the horns (and therefore the effectiveness of TRAP design) in two ways: by integrating midrange and high frequency transducers so that the horn is loading a broadband acoustical source, and by permitting the designer to use a single large horn instead of two smaller ones.

TRAP (True Array Principle) Design

Integrating Arrayable Systems with Mathematically Correct Topologies

TRAP systems are designed so that the enclosures provide optimum splay angles of 40° between the horns: the trapezoidal sides are therefore steeper than many other designs at 20°. The combination of symmetrical horns and steeper sidewall angles maintains coincident acoustic centers for all the elements in the array.

Note that moving the horizontal apex to the same location as the vertical results in a symmetrical 40° x 40° coverage pattern. This in turn requires the use of four enclosures to cover 160° with almost no variation in frequency response in the horizontal (side to side) plane. With 60° x 40° cabinets we could deliver sound to 180° of coverage, albeit with some quite audible variations.

The Reference Point Array

Because TRAP horns work so well together without extensive electronic processing, it has been possible to apply signal processing techniques to other problematic areas of array performance. The TriPolar technique for low frequency pattern control provides greater consistency in the vertical plane (from the front to the rear of the listening area) at frequencies where wavelengths are too long to make horns a practical technique for pattern control. In the horizontal plane, frequency shading between adjacent woofer sections can produce flat response across the coverage area. Renkus-Heinz engineers have applied these techniques to produce Reference Point Arrays designed for several coverage angles and output levels. The Reference Point Array concept extends from line-level signal processing, through power amplification, cabling and mounting/flying hardware. It builds on the foundation of TRAP design to provide a "plug 'n play" array that functions as an integral acoustic source. The time-consuming measurement and adjustment process has been done in the Renkus-Heinz R&D facilities. Because TRAP design is inherent in the enclosures and horns, the performance achieved under controlled lab conditions is easily repeated in the real world, leaving system designers and operators free to focus on optimizing the aesthetic result rather than chasing technical problems with array performance.

For more detail on the Reference Point Array concept and its execution, please refer to Renkus-Heinz white paper #2, which covers this topic in greater depth.

Fig. 11
Reference Point Array using four 40° x 40° mid-high enclosures and six low frequency modules in Tri-Polar configuration for vertical pattern control, along with appropriate small full range systems for downfill.