An Integrated Three-Way Constant Directivity Speaker Array

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Abstract

The drawbacks to building speaker clusters from most currently available devices include less than ideal array performance with cluster dispersion varying greatly with frequency. A new method has been developed that provides a systems approach to building coherent point source clusters with frequency invariant coverage from below 100 Hz to 18 kHz in both the horizontal and vertical planes.

Introduction

It is a well known fact that arraying multiple full range speaker systems in order to produce either increased sound pressure levels, wider coverage or both is not a straightforward proposition. Where a single speaker can be designed to resemble a point source, several such sources, connected to a coherent audio signal, will always interfere with each other. That is because from a listener's point of view, the signals arriving from every one of these sources are traveling from different distances and directions. For instance, a signal from one source could very well (at a specific frequency) totally cancel the signal arriving from another source which happens to be 1/2 wavelength further away or closer. This total cancellation is an extreme case, but partial cancellations and reinforcements occur throughout the audience area at different frequencies. This makes for anything but uniform coverage. The audible effects are often pronounced changes in frequency response as a function of listening positions within the intended coverage area.

Convex circular arrays have become the standard solution to minimize these interference problems. In these arrays the speakers are equipped with acoustical horns which aim the higher frequencies into a limited conical space, and individual speakers are then splayed so as to point these coverage cones away from each other. At these controlled frequencies (usually above 1-2 kHz) large areas within the coverage field get most of direct sound energy from only one horn. This in turn minimizes interference effects from all the other array sources. The convex array does not, however, reduce interference effects to inaudible levels.

While acoustical horns can be aimed sufficiently away from each other to keep the sound pressure levels at mid and high frequencies roughly constant throughout the intended coverage area, frequencies below 500 Hz overlap and therefore reinforce each other increasingly with decreasing frequency. This effect gives larger arrays an undesirable boost and forward directivity gain at low frequencies. When full range cabinets are arrayed in simple horizontal convex arrays, the sound pressure levels for frequencies below 500 Hz can be typically 6 to 12 dB greater directly on the array axis. At the same time they can also be 6 to 12 dB lower in level, relative to the mid and high frequencies, at the outside edges of the array coverage.

Additionally, the vertical dispersion of a simple horizontal convex arrays vary with frequency as well. Above a certain cut-off frequency, dependent on the mid and high frequency horn types and sizes, dispersion can be fairly constant. However, at frequencies typically below 500 Hz, dispersion increases proportionally with increasing wavelength. While poorly implemented horizontal convex arrays have the potential to give every audience member a different tonal balance, poor directivity control at the lower frequencies can have equally negative effects. From the audience's perspective, the effect would be a boomy bass and muffled highs in the front rows. Also gain before feed back from the stage microphones decreases at these lower frequencies, this often results in excessive equalization to compensate. The result is less than natural sound reproduction.

A new method of arraying is presented in this paper that comes much closer to ideal horizontal array coverage. A novel solution for achieving low frequency pattern control in the vertical plane within this array is also presented.

Quantifying Array Interferences

A simple array program is used to quantify the interference effects that occur with multiple arrayed devices [1].

For an array in far field dependence on angle is

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

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

$$P(\theta) = A(\varepsilon)^{j(\omega\tau - kS_i)}$$

where P is the sound pressure, ω is the angular frequency, $A_i(\theta)$ 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 i th source shown in Figure 1, assuming identical sources, the pressure contribution is given by

$$P_i = A_i(\theta)\varepsilon^{j(\omega\tau - kS_i)}$$

where $k = 2\pi / \lambda = 2\pi fc$, λ is the wavelength, f 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

$$P(\theta) = \sum_{i=1}^{n} A_i(\theta)\varepsilon^{j(\omega\tau - kS_i)} = \varepsilon^{j\omega\tau} \sum_{i=1} A_i(\theta)\varepsilon^{j\omega kS_i)}$$

The square of the pressure amplitude is given by

$$P_0^2(\theta) = [\sum_{i=1}^{n} A_i(\theta)\cos(kS_i)]^2 + A_i(\theta)\sin(kS_i)]^2$$

where $A_i(\theta) = A_i(\theta - \alpha_i)$

For a circular arc array, the additional path length S_i as shown in Figure 1. for the i th source at radius R and angle is given by

$$-S_i(\theta) = R_i \cos(\theta - \alpha_i)$$

Therefore, the smaller R is, the smaller the S_i differences or less interference. Ideally R=0.

In order to minimize interaction between speaker devices, the approach of trying to minimize the value of R was made.

Array Design Approach

The design approach for this array was broken down into two separate frequency bandpasses. First, a method for minimizing the acoustic center offset utilizing horn loaded mid/high frequency devices with specially developed speaker cabinet geometry was used for the frequencies of 400Hz and above. At frequencies below 400 Hz, horn loading of devices becomes impractical due to the longer wavelengths and the requirement to maintain a horn mouth opening the approximate dimension of the wavelength to maintain directivity control. At these wavelengths, a modular line array of three low frequency devices was developed.

The design goal for these devices and the system as a whole was to provide a modular, easy to configure and install speaker array that would provide the designer with a systems approach to array design. Speakers should be easy to array with mechanical rigging methods that will minimize physical offset errors that can cause signal misalignment. The system should also integrate all full frequency components into the array instead of requiring separate mid/high and low frequency arrays.

Array Principles for Mid and High Frequencies

The basic criteria for achieving meaningful arrayability are known. Wideband (500 to 20,000 Hz) constant directivity is just one of the prerequisites.

Figure 2 shows a typical conventional convex circular array. Each high frequency horn typically is designed for constant 60 degree horizontal coverage, (with a directivity cutoff frequency of about 1 kHz) and the speaker cabinets are constructed with a 15 degree horizontal array angle. The three cabinets are "close coupled," meaning their 15 degree sides touch. This results in a splay of 30 degree between the cabinets. Total intended -6 dB coverage is 120 degrees for the array.

One would assume, that this 30 degree splay with 60 degree would be optimum. Indeed, these commonly used 15 degree cabinets suggest very strongly, that for optimum array performance the speaker cabinets should be arrayed "close-coupled" as shown in Figure 2. However, as we will see below, this is not the case at all.

Figure 2 also shows the typical overlap areas of the above illustrated array in the horizontal plane, at frequencies from 2 kHz to-20 kHz, and Figure 3 shows predictions of interference patterns at three different frequencies. There is a forward gain of 10 dB and quite severe interference at higher frequencies. This then is not the best arrangement for even coverage. However, if forward gain is more important than minimizing interference, this array configuration is useable.

Another prerequisite for achieving meaningful arrayability is to splay adjacent constant directivity devices at their respective -6dB points. Indeed for minimum interference areas and best possible even coverage, doubling the splay between the illustrated cabinets to 60 degrees as shown in Figure 4 yields much better results. The coverage angle is increased to 180 degrees. This wider and more uniform coverage comes at the expense of 10 dB of forward gain. (See Figure 2)

Figure 5 shows the results at 1, 2 and 4 kHz. At higher frequencies individual horns are now clearly discernible. Note that remaining interferences deepen with increasing frequencies, the array performance with conventional horns varies with frequency. Array performance as shown above is typically the best that can be expected. A larger splay angle will cause lack of coverage between horns, smaller splay angles increase the areas of interference.

Figure 6 illustrates the wave fronts as they radiate from the points of origin, i. e. the point in each of the three horns where the sound waves seem to originate from. Typically, these points are separated in space, and the illustration 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 wave fronts in Figure 6 if their points of origin could be made to coincide? Figure 7 shows this theoretical case, which indeed would solve the interference problems.

Unfortunately, it is physically impossible to array horns like this on a single horizontal plane. Separated vertically, horns have been splayed as shown in Figure 7 to maintain coincident acoustic centers. 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. Best to have all points of origin coincide in one point in space.

Of course, this is easier said than done. Speaker system design imposes physical restraints that make this ideal case unattainable. However, once the principle is understood, it turns out that a solution is feasible that comes close enough to the theoretical ideal to result in a significant improvement over conventional designs.

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

It is well known that most constant directivity horns exhibit "astigmatism," that is to say, their apparent points of origin are different in the horizontal and vertical plane. Typically the apparent apex in the wider coverage plane is further forward toward the mouth of the horn, the apparent apex for the narrower coverage plane is much further back toward the throat. Most, if not all typical array speaker are designed with the popular 60 degree (Horizontal) x 40 degree (Vertical) horns, which put the apparent apex quite forward. With the above information in mind, it is apparent that a better array would result if one could use the location of the vertical apex for the horizontal plane.

Figure 8 shows new horns with apparent apex located quite far toward the rear of the cabinet. This indeed is how improved array performance is achieved. Note that for 160 degree coverage, there are now four cabinets required.

Figure 9 shows the simulation of this new system construction. Comparison with Figure 3 above shows significant improvements at all frequencies. Theoretical array performance from horns designed for frequency invariant coverage with coincident apparent apexes approaches that of a point source.

Figure 10 shows one of these cabinets. It incorporates all of the principles discussed above to provide superior performance in an array.

The horn design ensures broad band pattern control down to the frequency at which mutual coupling between adjacent cabinets ceases. By doing this, a less efficient (yet cost effective) 15" woofer can be combined with an efficient mid/high section. The frequency response for a single cabinet is now shelved below 500 Hz at a rate of roughly 3dB per octave. However, as Figure 13 shows, the frequency response of a three wide array will be reasonably flat with flat power response as well.

Its cabinet design provides optimum splay angles and maintains coincident acoustic centers for adjacent cabinets. By maximizing coverage and minimizing interference, an array based on these speakers can maintain frequency response variations within the intended coverage area to +- 4 dB. This performance is possible "out of the box" without applying micro-delay and/or frequency shading techniques.

Figures 11 and 12 shows a close coupled array of three cabinets. Figure 13 shows the horizontal and vertical polars for a three wide array as measured EASE data. Although the measured data for these "real world" devices do not demonstrate completely frequency invariant coverage, the results do support the claim for maintaining frequency response variations within +/- 4 dB. The reason the measured results don't track the predictions 100%, is because the simulations are based on perfectly behaved 40 degree horizontal coverage at each frequency. The horns used in the these speaker systems maintain a nominal 40 degree coverage with a variance of +/- 10 degrees within the frequency range of 1 kHz to 4 kHz.

Array Principles for Low Frequencies

For full range music applications additional low frequency devices would be added to extend the useable range of the cluster down to 40 Hz while maintaining closely matched directivity characteristics of the mid/high array.

In order to maintain some amount of simplicity in the system, it was decided to concentrate on matching low frequency directivity of the array in the vertical plane and not worry so much about the horizontal plane. As long as the horizontal and vertical coverage of low frequencies is similar to the mid/highs within the coverage angles, our design goals will be achieved. Again the goal as a systems designer is to provide consistent frequency response at all listener locations. This is most often not met due to the inability of speaker arrays to control the directivity of low frequencies, with the vertical plane being the most difficult. The most common result is a great deal of low frequency directly beneath a speaker system, with decreasing low frequency sound energy as you move away from the system. As you move further out in the coverage field of the system where mid/high coverage is better directed, there is generally a substantial low frequency rolloff.

The low frequency array utilizes the principle of a combination of doublet sources to derive signal cancellation off axis from the array as described by Olson [2]. Three 15" drivers are vertically arrayed, two above the mid/high modules and one below as illustrated in Figure 14. Each of these drivers are supplied with discrete program source signal that has been processed with bandpass and parametric filtering. The challenge is to provide signal to three combinations of the drivers so that two of the drivers are used at any given frequency within the low frequency bandpass that are one half wavelength apart. This approach will provide some directivity control in the vertical plane where before there was none.

Specifically, within the low frequency bandpass, the two drivers above the mid/high module (directly adjacent to each other) work together within the upper portion of the bandpass, the drivers directly above and below the mid/high module work together within the middle portion of the bandpass, and the top and bottom driver work together in the lower portion of the bandpass.

Practical implementation of this approach was investigated using several modeling programs. It was found that theoretically 90 degree off axis rejection could range from 8 dB at 50 Hz to as much as 25 dB at 160 Hz. The bandpass upper limit that was used for low frequency investigations was 400 Hz. At frequencies above this point, the spacing of the adjacent low frequency drivers creates excessive off axis lobing problems.

One of the greatest limitations with this development work was the lack of a definitive modeling program that would take into account all physical parameters and provide all the information necessary to better define testing procedures. No single prediction software program took into account the individual device directivity, changes in amplitude and phase due to filtering, and allowed investigation at any particular frequency of choice, all at the same time. Interpolation was required to input each individual device information for any given one third octave band. Results of those predictions are shown in Figures 15a through 15j.

Figure 16 illustrates the frequency response and phase plots for the signal processing of the system used during measurements. Processing included parametric filtering, crossover filtering, and all pass filtering for improvements in the phase relationship of the individual drivers. Three channels of processing were used for low frequencies, with the fourth used for passive mid/highs. Additional level matching at the amplifiers (or at the processor output) was made to compensate for variations in output levels due to processing.

Tests were made on a full size system and vertical polar measurements were made using MLSSA in a mostly free field (outdoors on a lift) environment. Compensation was made to each of the processor outputs to match the individual device measured response to that of the desired prediction. Individual low frequency line array device measurements are shown in Figure 17. Low frequency line array test results showed some asymmetrical lobing characteristics directed into the above-axis vertical pattern (shown in Figure 18). This asymmetrical lobing in the hemisphere above the array is generally directed towards the ceiling in a room and should minimally impact the system performance in all but the most highly reflective acoustic spaces. Because of this asymmetry, it is important to have the double 15" cabinet above the mid/high module with the single 15" driver below.

Figure 19 shows measurements in the vertical plane below the array. Performance 90 degree off axis is at least -6 dB for all frequencies down to 90 Hz, -5 dB down to 70 Hz making for an effective low frequency array.

Limitations

One of the main limitations of this system approach is the increased signal processing requirement for low frequencies. As described here, three discrete low frequency signals are required to provide adequate processing for the low frequencies. To compensate for this high processing requirement, passive mid/high speaker systems were utilized. As digital processing becomes increasingly more cost effective, this may become less of an issue in the future. If a double wide version of six drivers is used, only the addition of speakers are required as the amplifiers and signal processing is already available and being used. For very little extra cost (compared to the average complete system), doubling the low frequency capabilities is easily within reach.

System Configurations and Variations

Again, the goal of this systems approach is to provide greatly improved broad band constant directivity down to the lower frequencies, that can be easily implemented. One of the greatest advantages of this approach is the integration of the low frequency array into the mid/high array as a single speaker cluster unit that is easily rigged and aesthetically pleasing. Two such configurations are shown in Figures 20 and 21. Shown are modular systems of three and four mid/high frequency modules wide with one or two low frequency line arrays. Additional short throw components and asymmetrical horn configurations are being developed that can be integrated into the system as well.

Conclusion

The array design approach as described here provides some dramatic improvements over typical more traditional systems designs in directivity control. Lobing and interference are minimized, and system directivity is closely maintained down to well within the low frequency band.

However, this should be only the beginning of a more innovative "systems" approach to speaker design. Additional component and systems variations are being developed. Improvements to modeling programs should enable more insightful and accurate predictions for complete array performance. As systems are designed, installed and tested, additional data will become available to aid in the further development and refinement of these techniques.

Acknowledgments

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Thanks to TOA Electronics, USA for the use of a DP-0204 software for graphical presentation of low frequency processing.

References

[1] ALS-1 Array Lobing Program, Renkus-Heinz, and Rex Sinclair of Sinclair Consultants, 1985.

[2] H. F. Olson, *Acoustical Engineering* (Van Nostrand, Princeton, NJ, 1957)

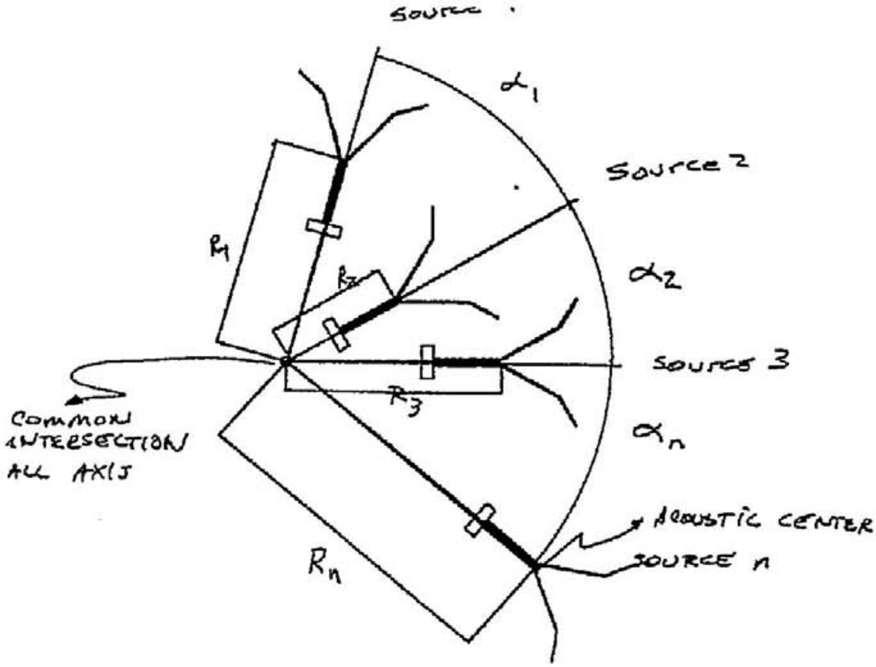

Figure 1. Array prediction device relationship.

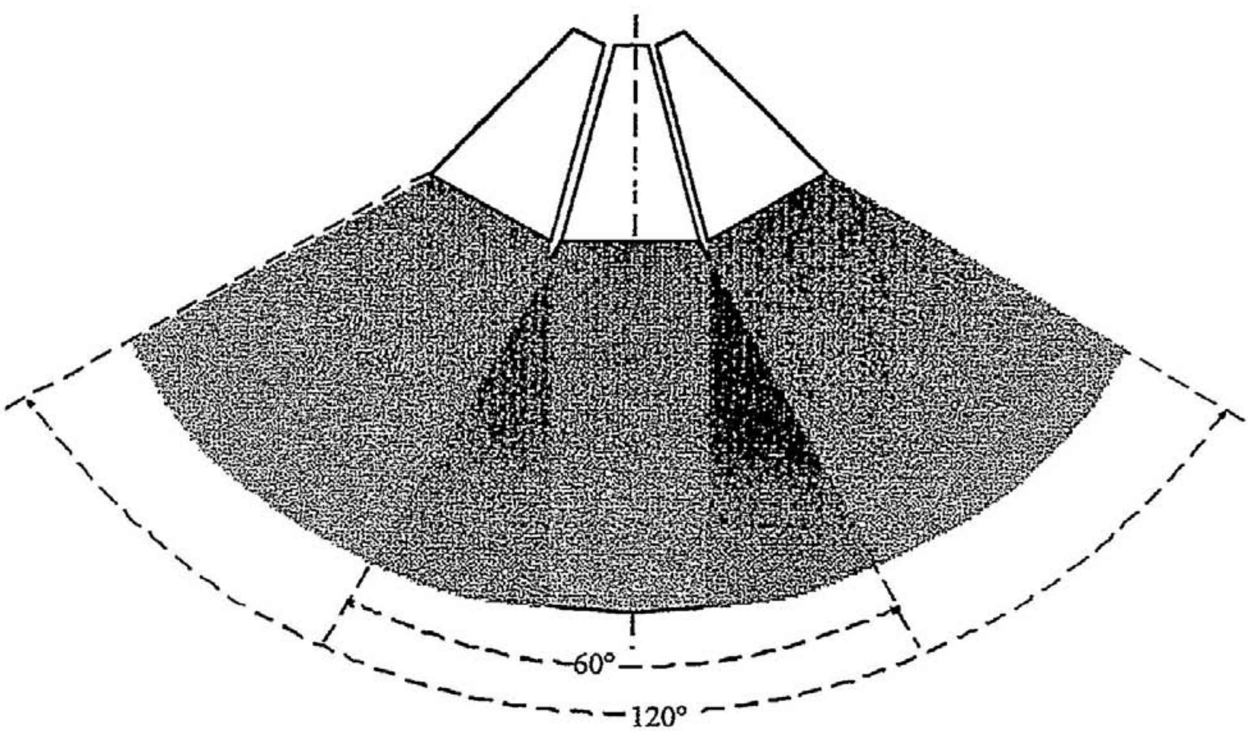

Figure 2. A typical conventional convex circular array.

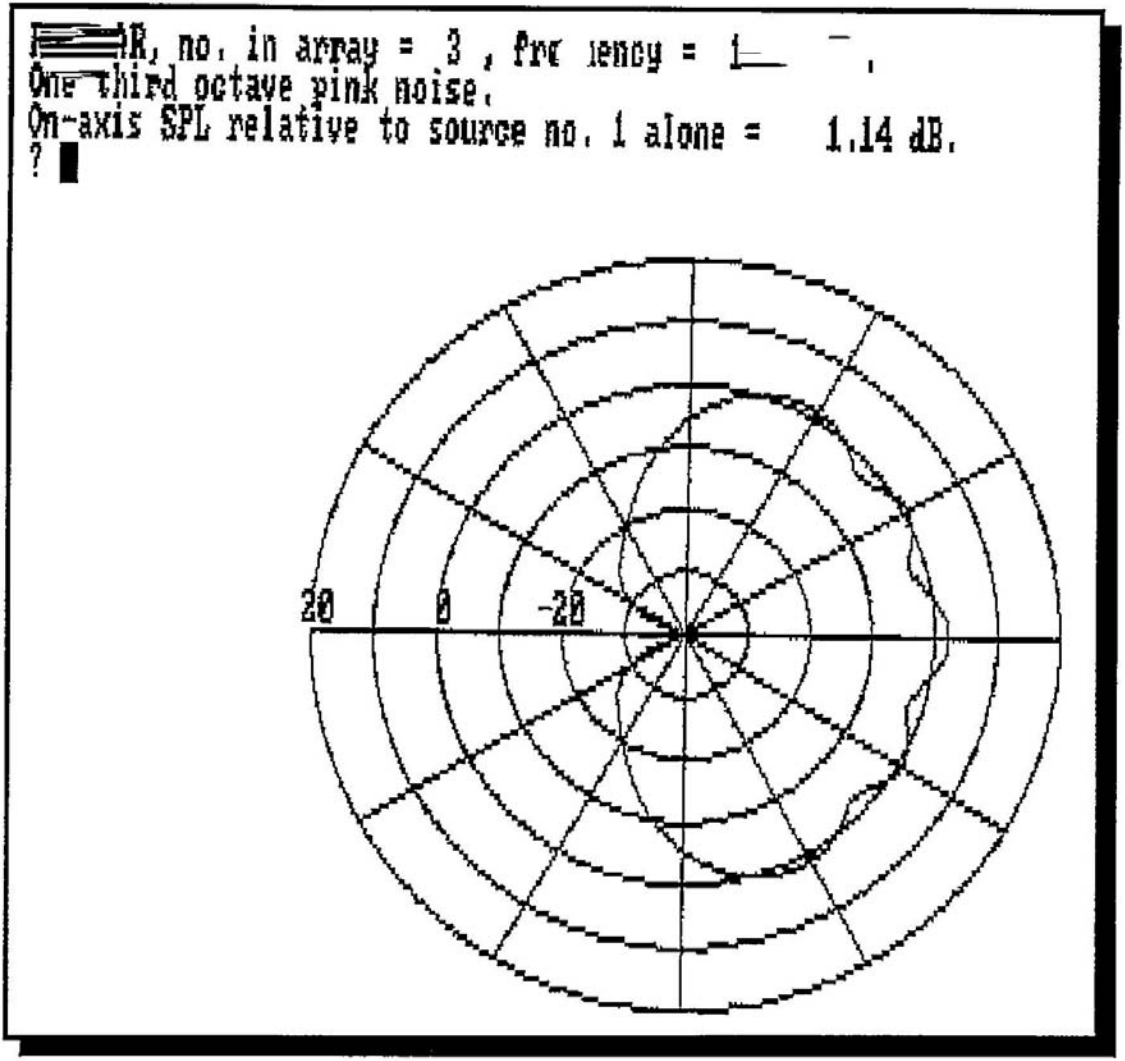

Figure 3a. Predictions for a conventional convex circular array at 1 kHz.

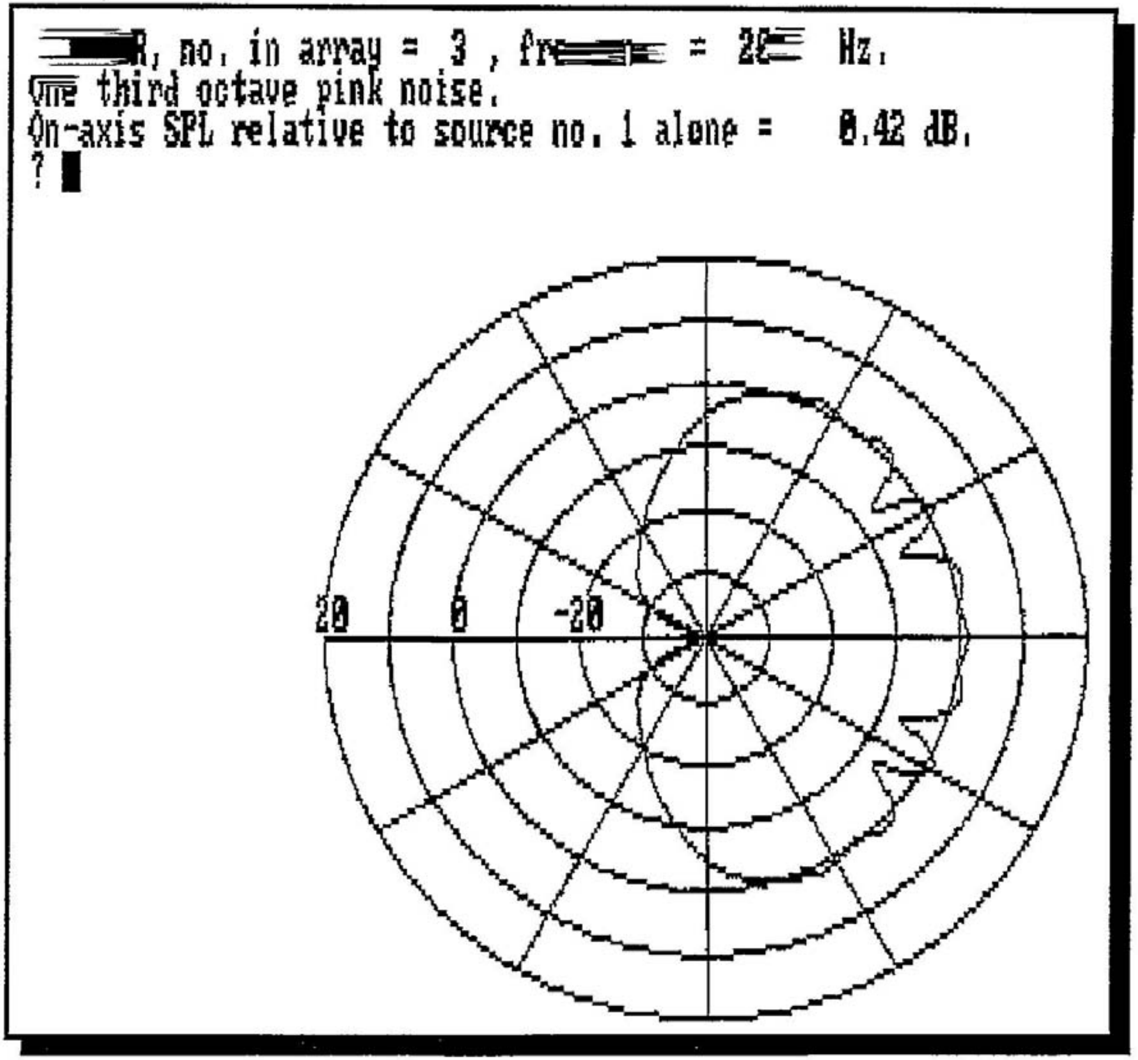

Figure 3b. Predictions for a conventional convex circular array at 2 kHz.

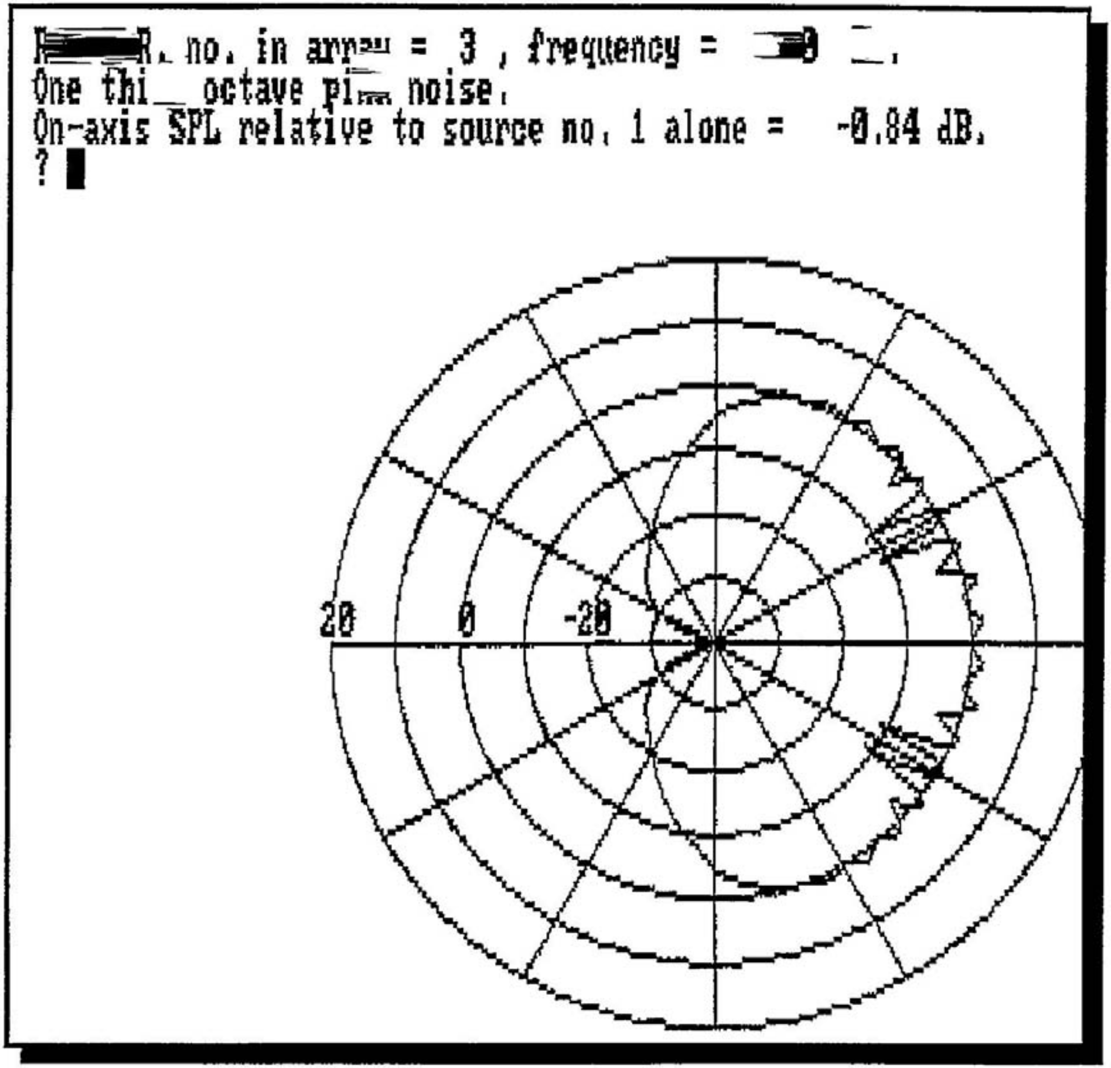

Figure 3c. Predictions for a conventional convex circular array at 4 kHz.

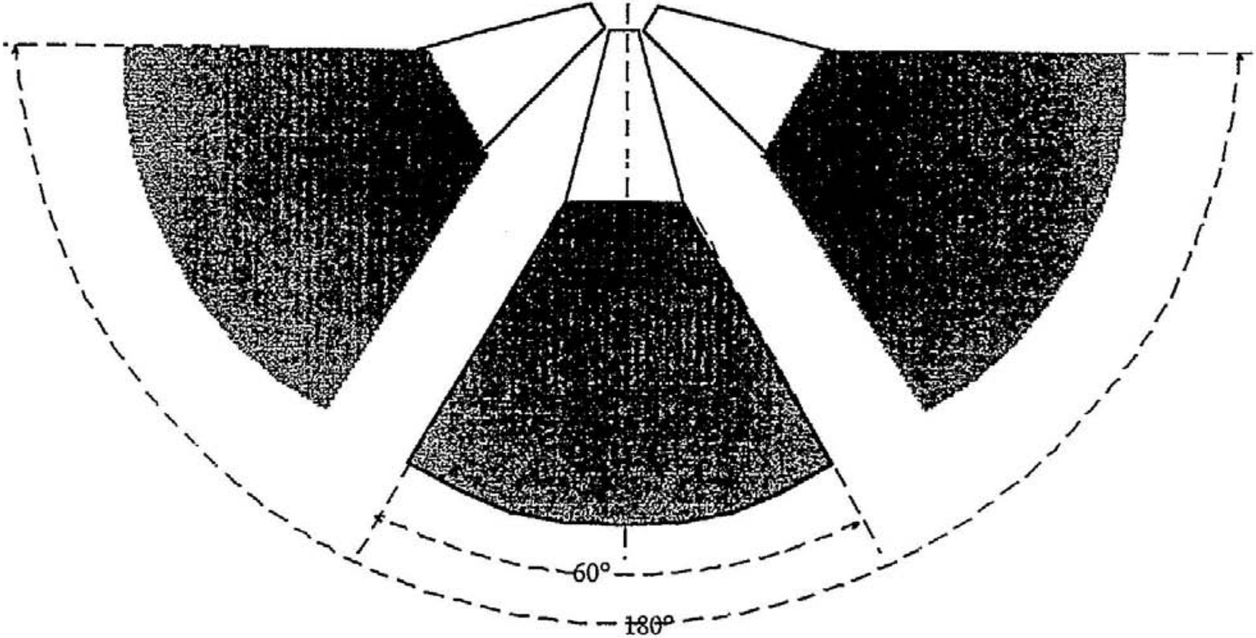

Figure 4. A conventional convex circular array with cabinets splayed 60 degrees.

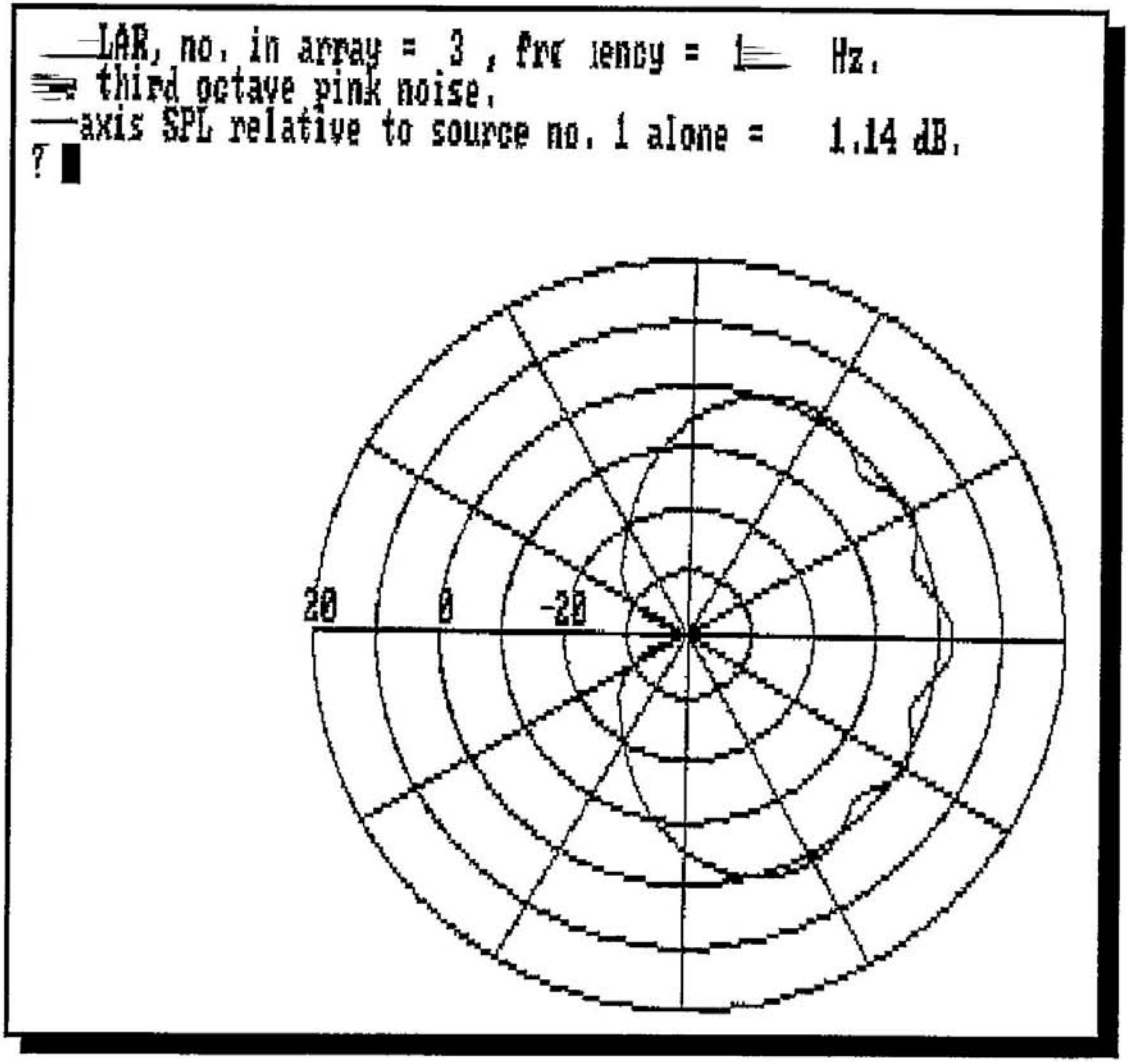

Figure 5a. Predictions for a conventional convex circular array with cabinets splayed 60 degrees at 1 kHz.

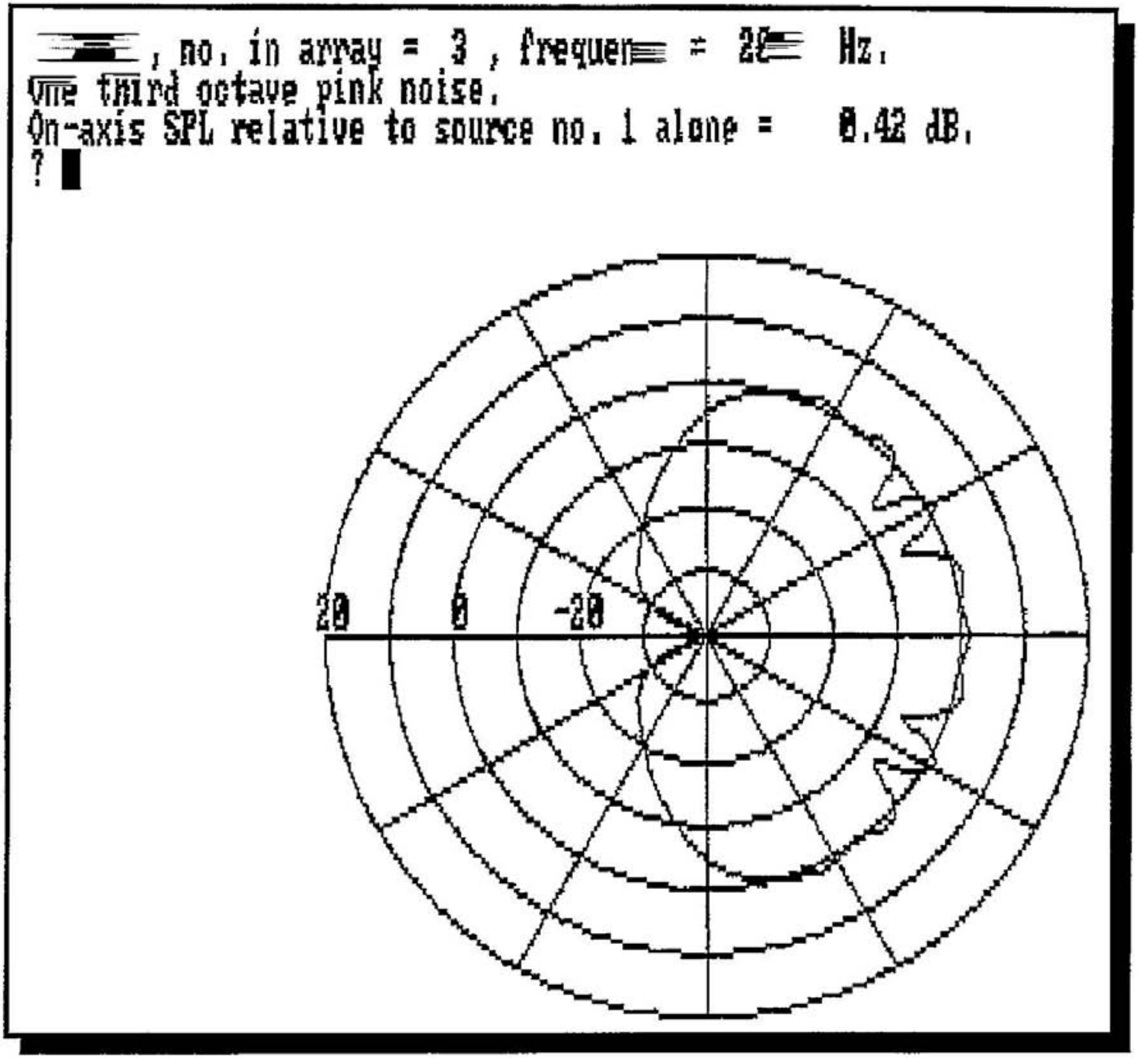

Figure 5b. Predictions for a conventional convex circular array with cabinets splayed 60 degrees at 2 kHz.

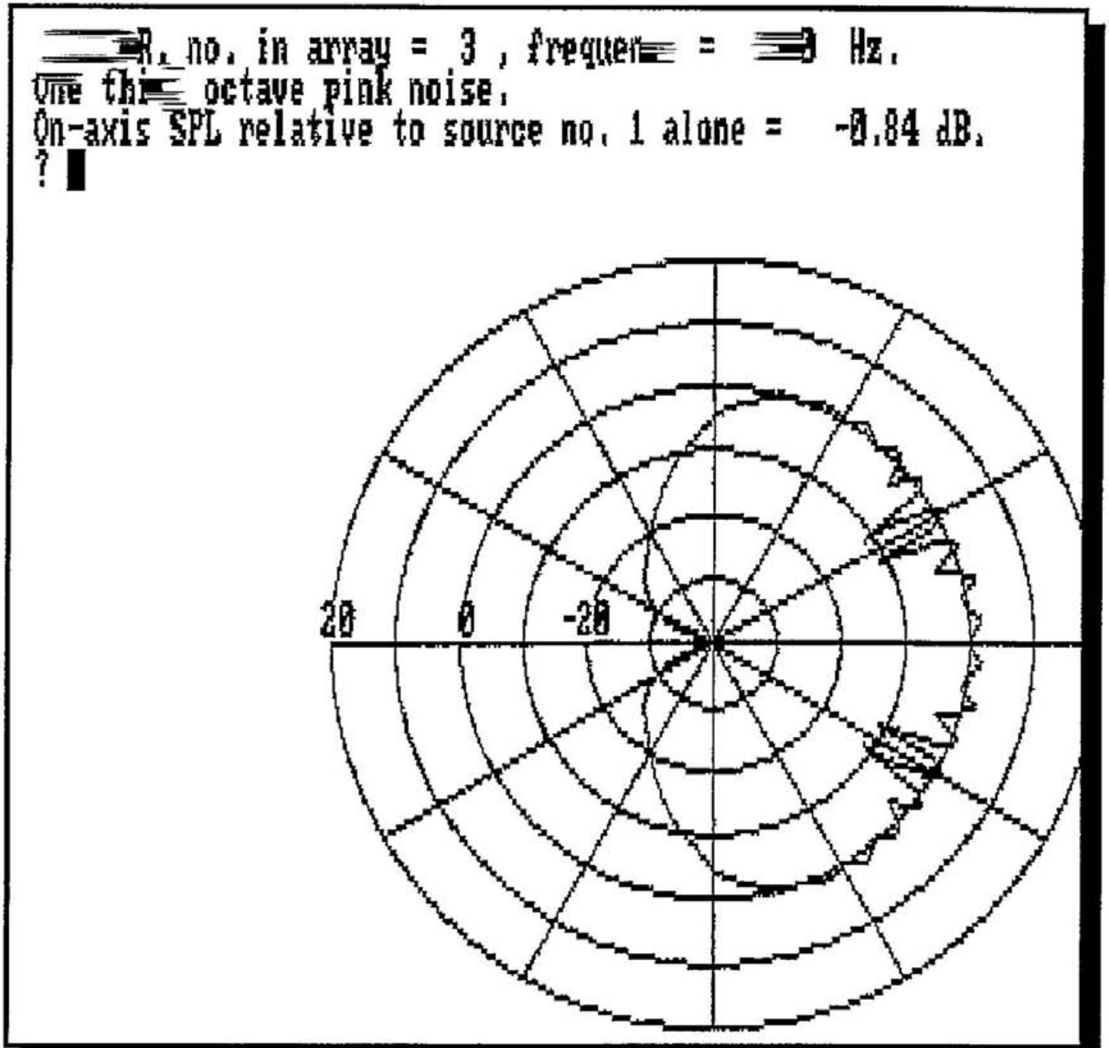

Figure 5c. Predictions for a conventional convex circular array with cabinets splayed 60 degrees at 4 kHz.

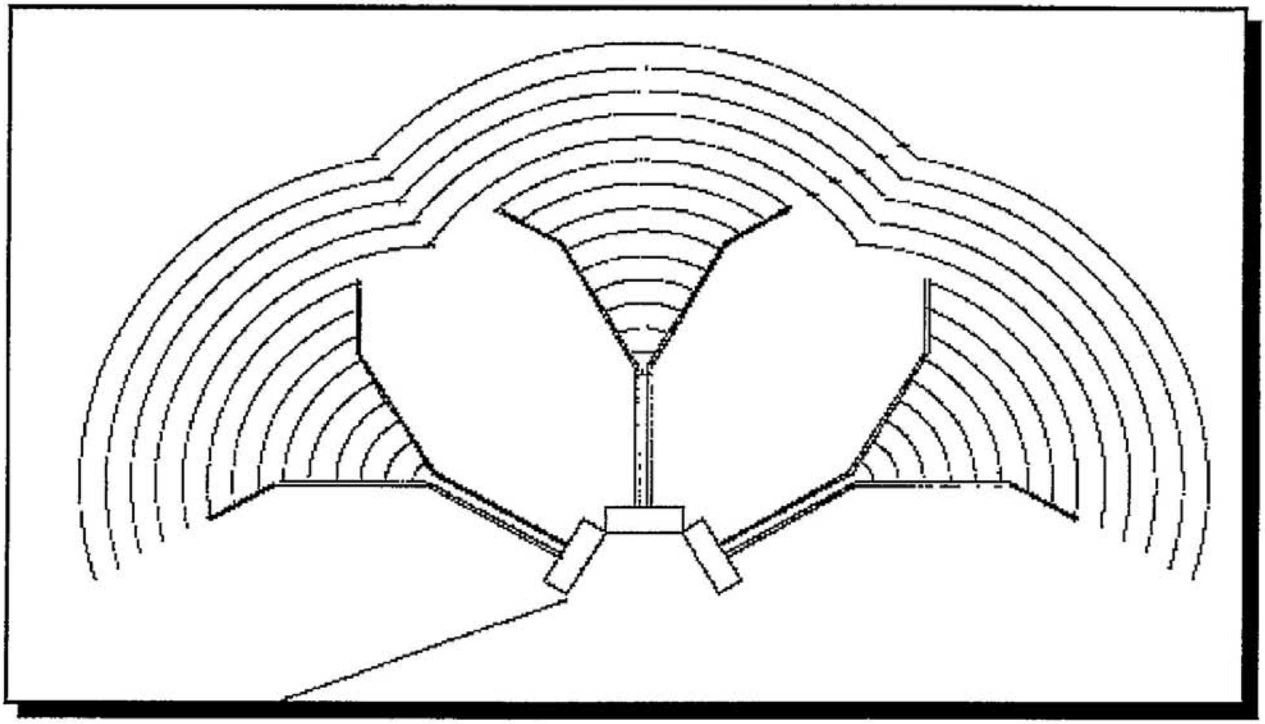

Figure 6. Illustration of wave front radiation from noncoincident points of acoustic origin.

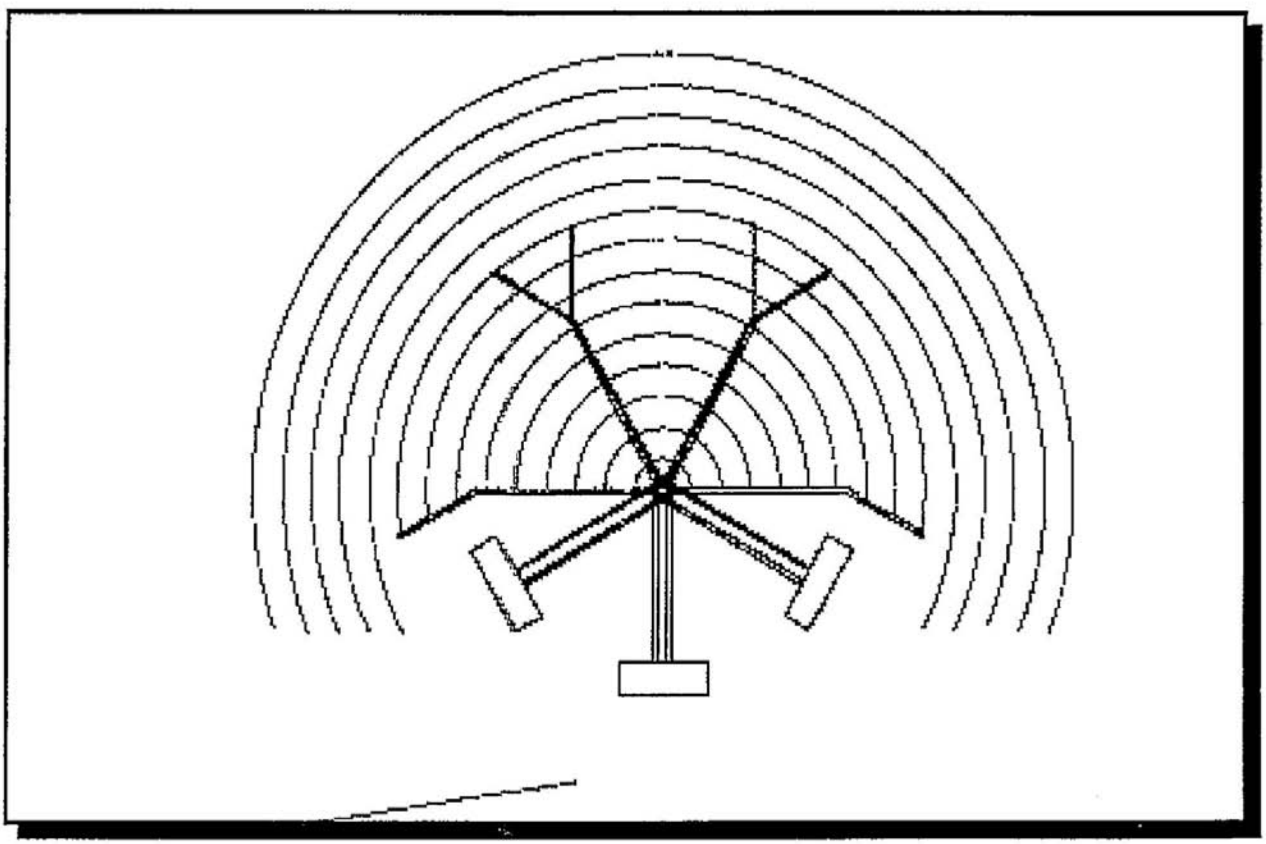

Figure 7. Illustration of wave front radiation from coincident points of acoustic origin.

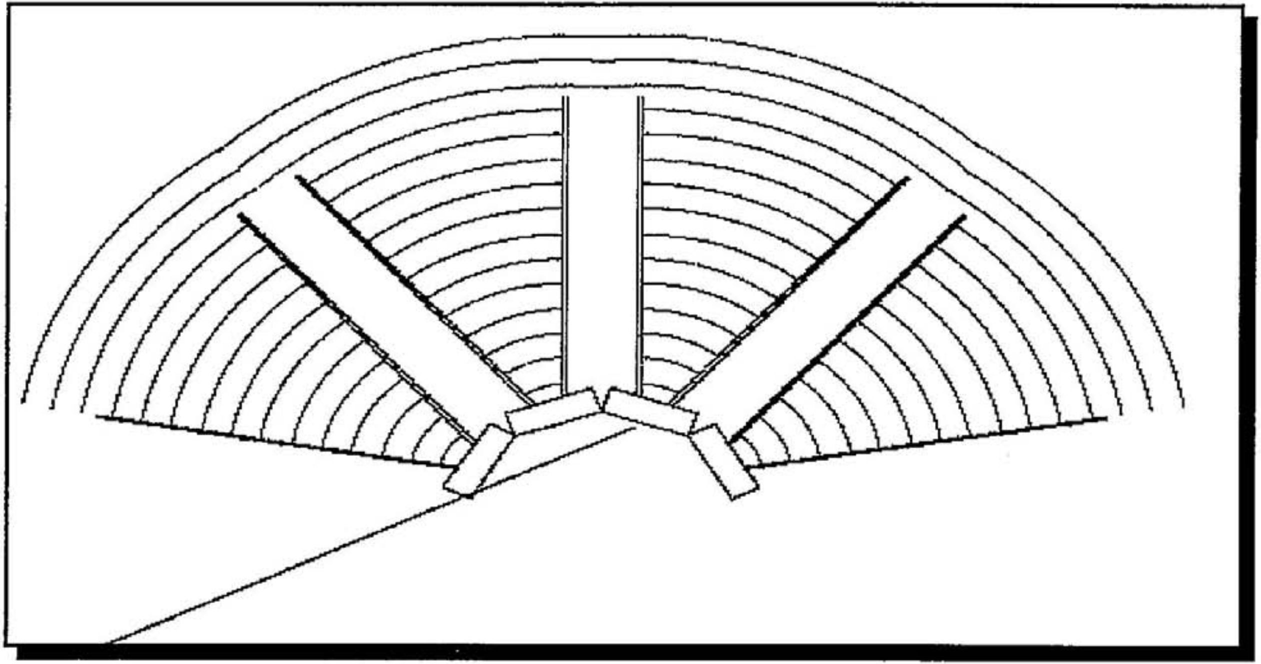

Figure 8. Illustration of closely packed acoustic origin points.

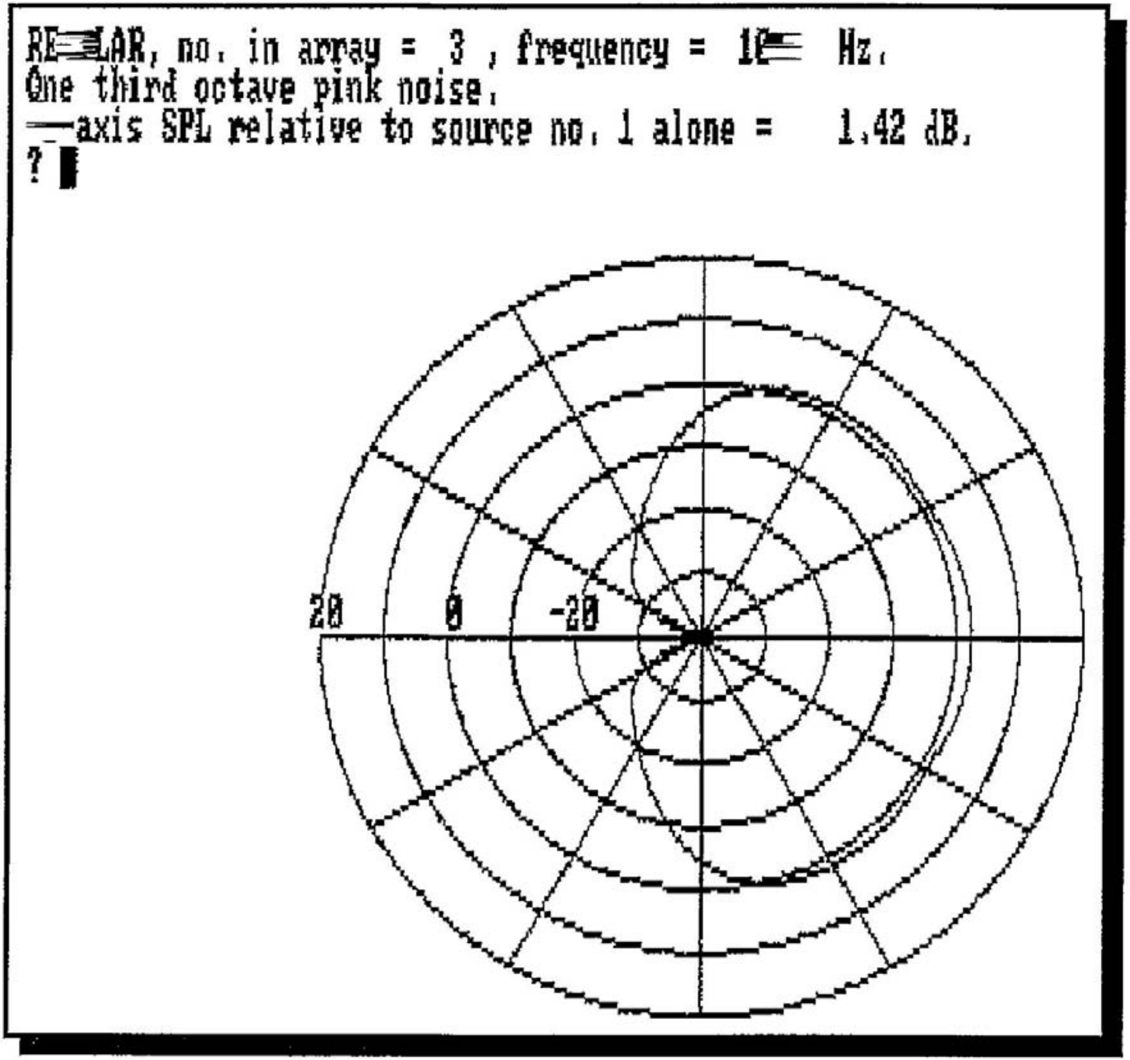

Figure 9a. Predictions for a coincident acoustic origin array at 1 kHz.

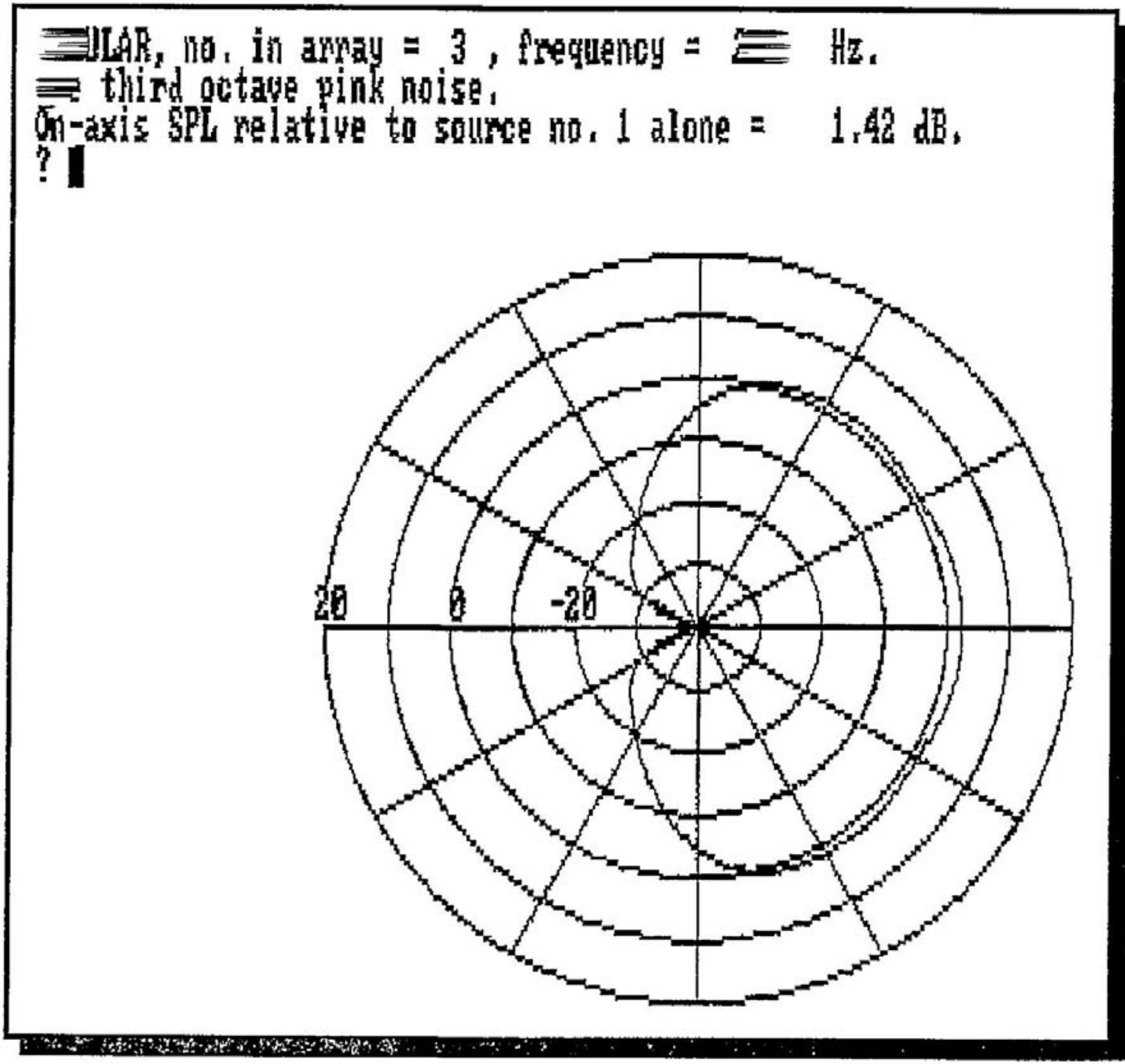

Figure 9b. Predictions for a coincident acoustic origin array at 2 kHz.

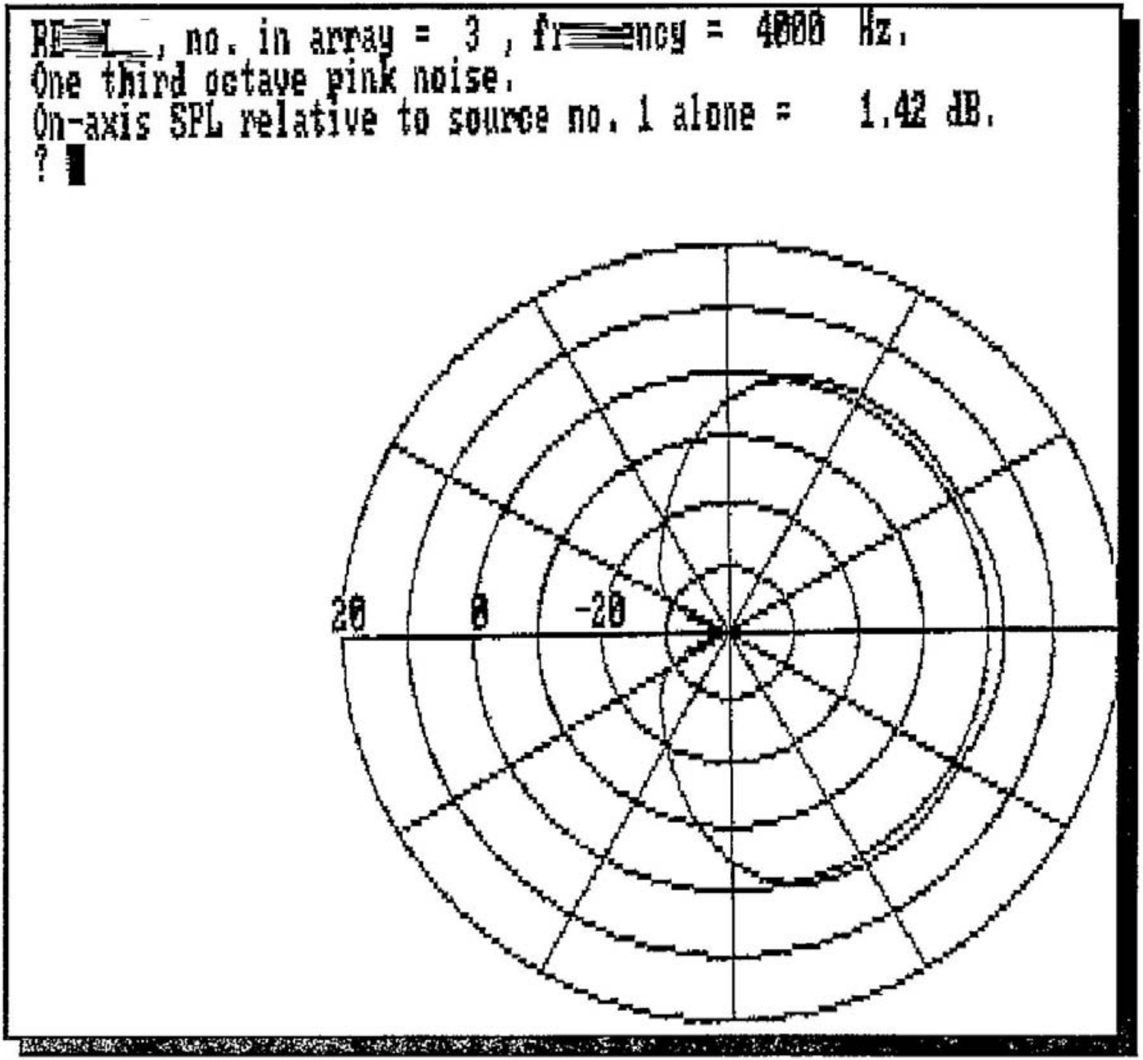

Figure 9c. Predictions for a coincident acoustic origin array at 4 kHz.

Figure 10. Single speaker cabinet

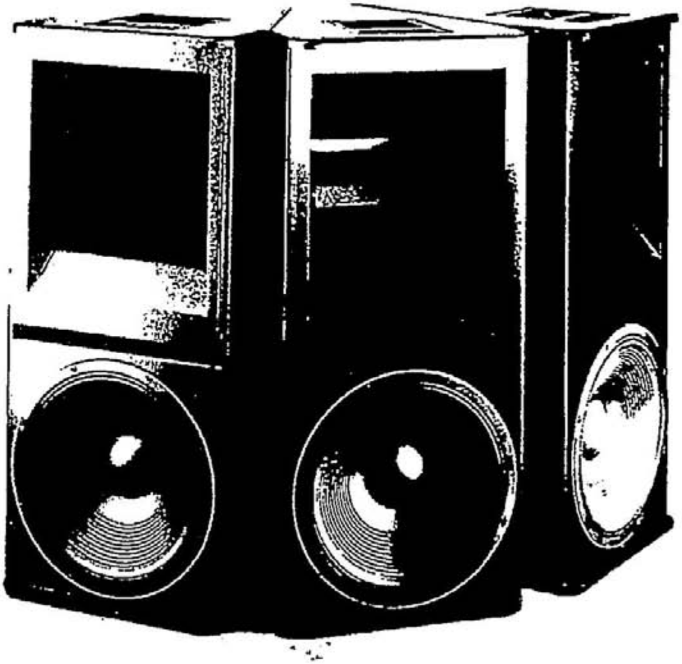

Figure 11. Array of 3 full range speaker cabinets without low frequency line array.

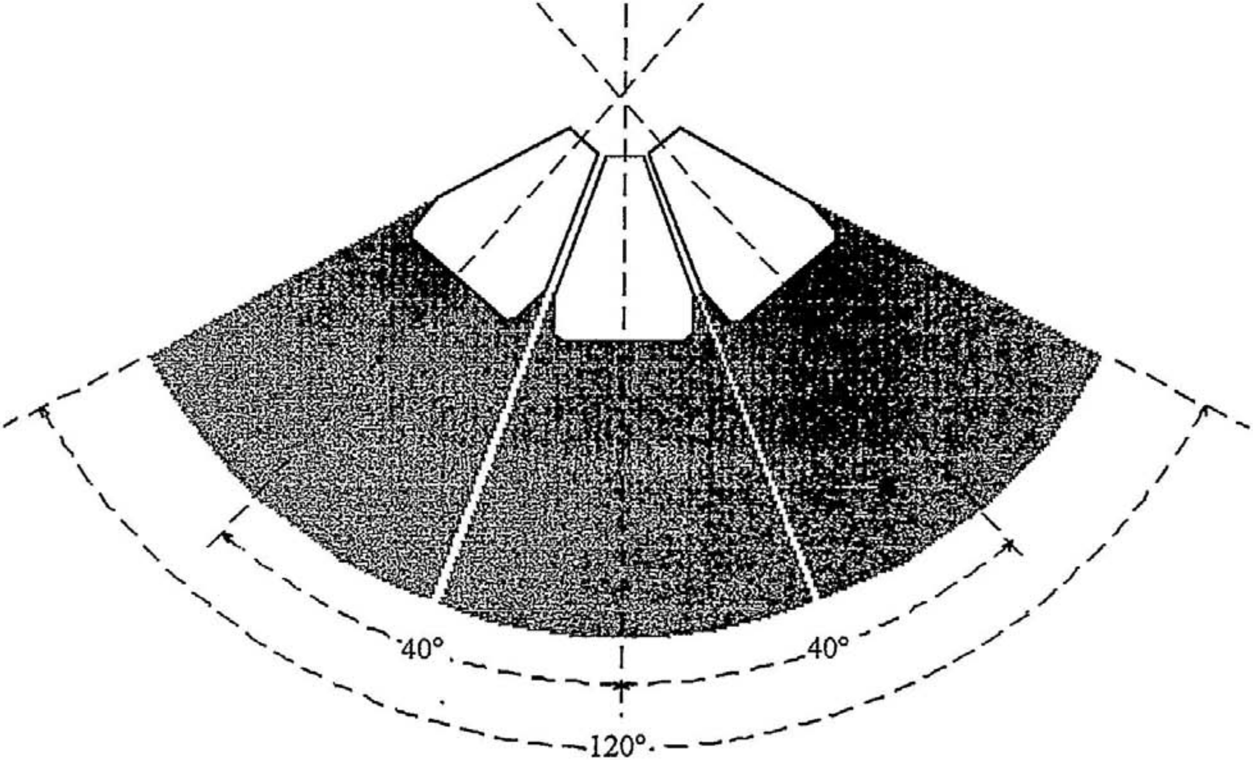

Figure 12. Illustration of 3x40 degree speaker array.

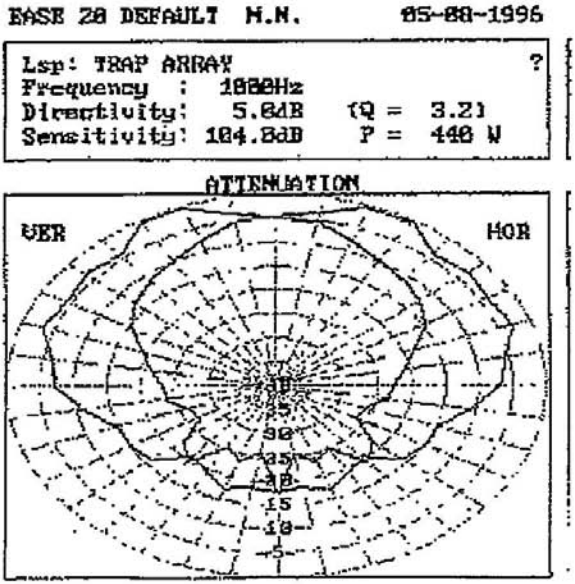

Figure 13a. Measurements of a 3x40 degree array at 1 kHz.

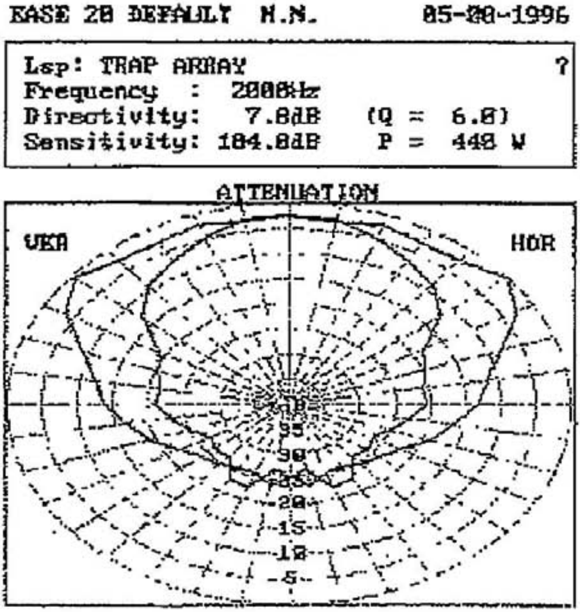

Figure 13b. Measurements of a 3x40 degree array at 2 kHz.

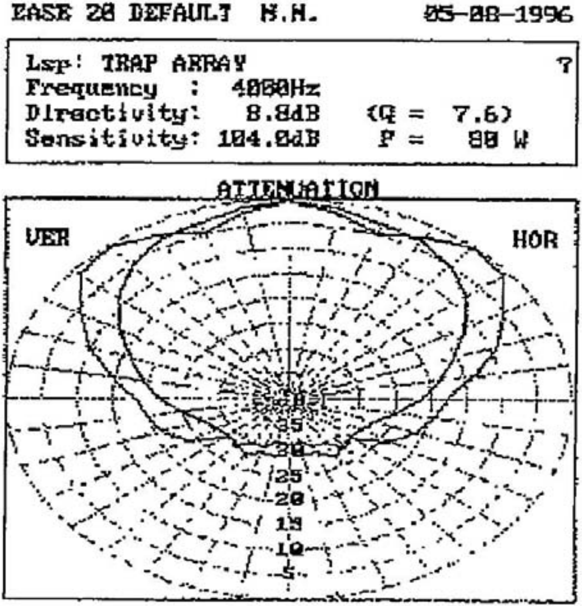

Figure 13c. Measurements of a 3x40 degree array at 4 kHz.

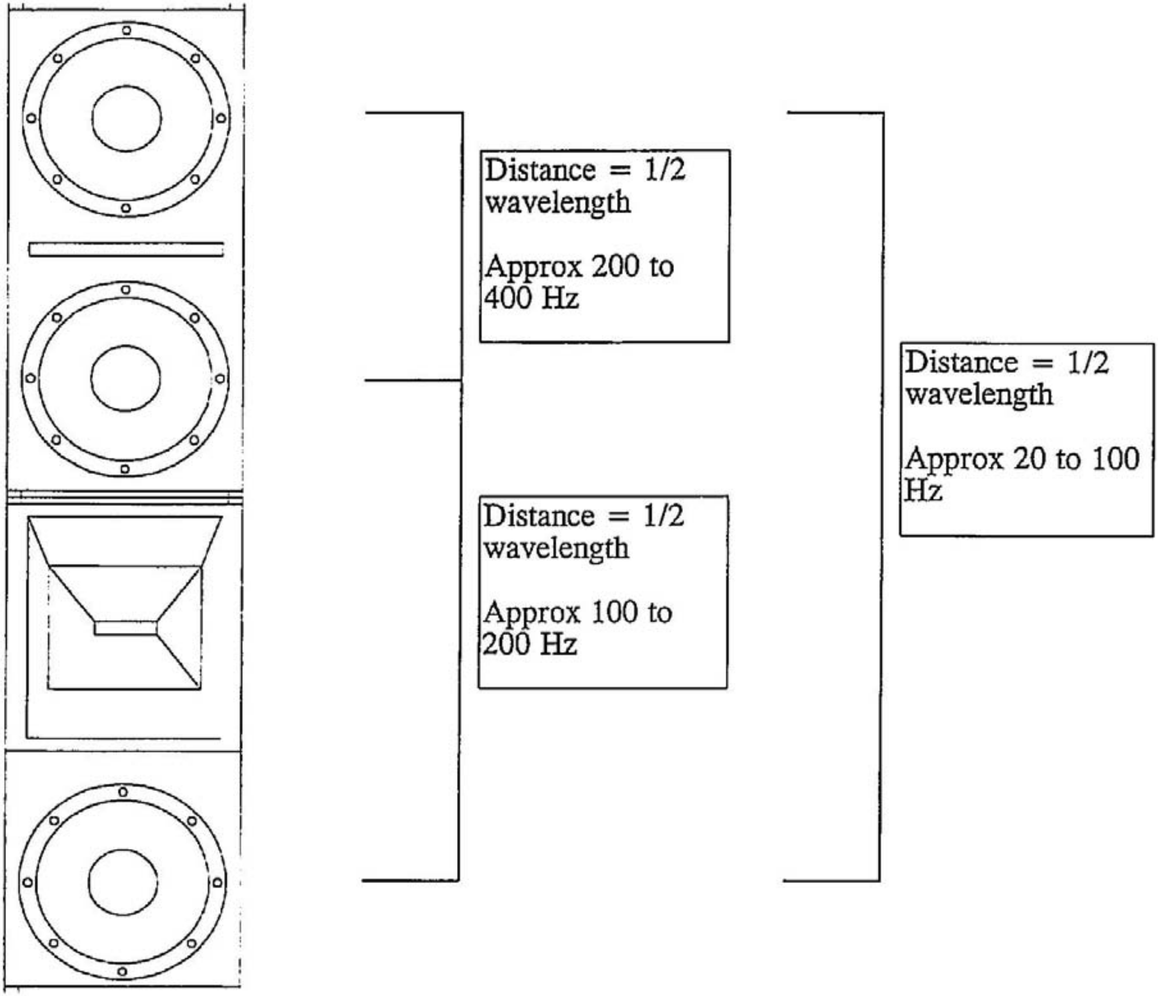

Figure 14. Illustration of low frequency line array driver and processing relationship.

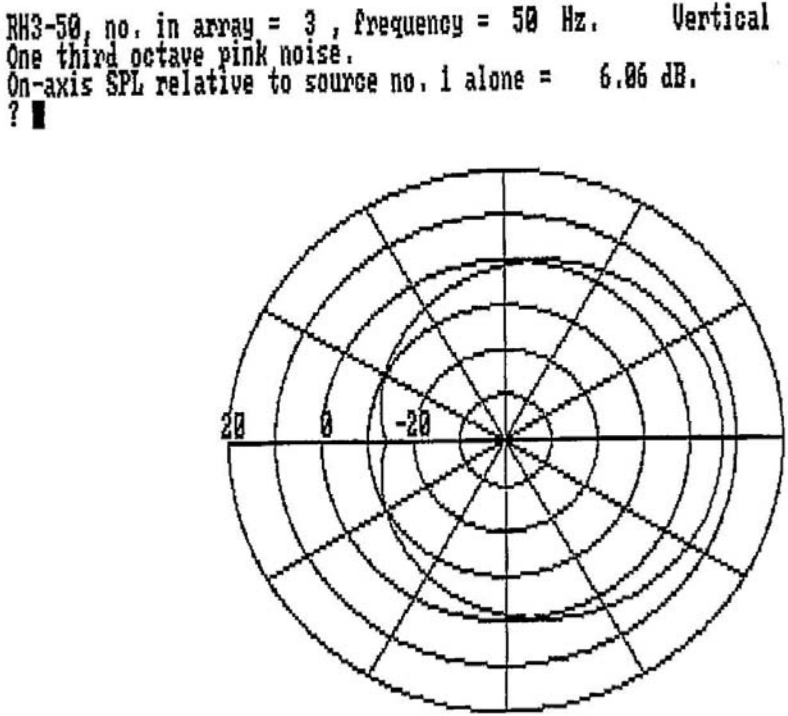

Figure 15a. Low frequency line array vertical polar prediction at 50 Hz.