The Design and Implementation of Line Arrays Using Digital Signal Processing

David Gunness
Eastern Acoustic Works, Whitinsville, MA USA

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Abstract

The requirements for a wide bandwidth, steerable loudspeaker array are presented, and used to establish physical criteria for a broadly useful system. An implementation is described which meets these criteria with a high-density, multi-way source array and integrated processing and amplification. The directional characteristics of digitally steered arrays are exlpored, including some which offer unique advantages. Practical limits of directional performance are also established.

0 INTRODUCTION

The capability of digitally steered line array loudspeakers to create directional patterns of varying beamwidth, and to steer those patterns off the primary axis of the device is well known. The ability to cover an audience from an elevated position without physically tilting the loudspeaker is a significant benefit in itself. Additional acoustical benefits may be realized by employing non-traditional coverage patterns, matched to the particular dimensions of the listening area. It will be shown that the horizontal invariance of the vertical patterns results in more precisely covered typical audience areas.

In order for all of these benefits to be realized over a broad range of applications, an implementation is required with wide enough bandwidth to be considered a legitimate musical sound reinforcement loudspeaker system. In rough terms, this requires a system with a bandwidth of about 7 octaves, or from less than 100 Hz to greater than 8 kHz.

1 IMPLEMENTATION

To achieve a useful degree of steerability to above 8 kHz, a source spacing of less than 50 mm is required. This can be accomplished by using very small sources. However, to achieve useful output levels and directional control as low as 100 Hz requires larger transducers and a relatively long array. The solution to these conflicting requirements is to create a two-way system with separate arrays of low frequency and high frequency transducers.

Modeled sources were evaluated at various spacings to determine the optimum arrangement for steerability. Effective steering is dependant upon source spacing that is proportional to the minimum wavelength reproduced. Should the source spacing exceed the minimum wavelength, off axis lobes can result that are equal in magnitude to the primary lobe. Consequently, source spacing must be kept to a minimum. To further prevent off axis lobes at high frequencies, an edge-to-edge linear array of horn mouths was found to be much more effective than a linear array of direct radiating sources.

1.1 Acoustical Requirements

1.1.1 High Frequency Analysis

To determine the effect of source size and spacing on the directional behavior of a steered array, it is instructive to compare the performance of a single, ideal, continuous, line source with a linear array made up of multiple sources. Figure 1 illustrates an ideal, continuous, 30° arc source. The source has been equalized and filtered to be representative of a real high frequency source. Magnitude response of the source is shown at 0°, 10°, and 60° off axis. The polar response plots show that the source exhibits an ideal 30° pattern above 2 kHz.

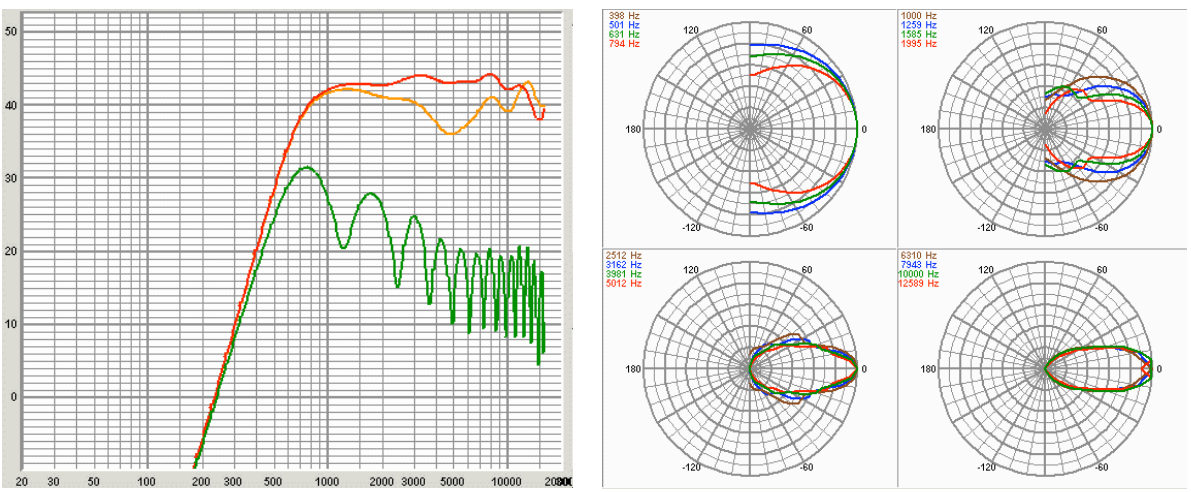

Fig. 1. Ideal, continuous, 30° arc source. Magnitude - Red: 0°, orange: 10°, green: 60°. Polar - 400 Hz to 12.5 kHz, 1/3 octave.

Figure 2 illustrates the response of this source at the 60° location. For clarity the magnitude response from figure 1 is displayed again. The impulse response exhibits a positive-going pulse corresponding to the nearest edge of the source and a negative-going pulse corresponding to the far edge of the source, spaced about 0.8 ms in time. The details of this topic are outlined in "Loudspeaker Acoustic Field Calculations with Application to Directional Response Measurement", presented at the 109th Convention of the Audio Engineering Society, 2000[1]. Note that "peaks" and "valleys" in the magnitude response occur approximately every 1.2 kHz, corresponding to a 0.8 ms period.

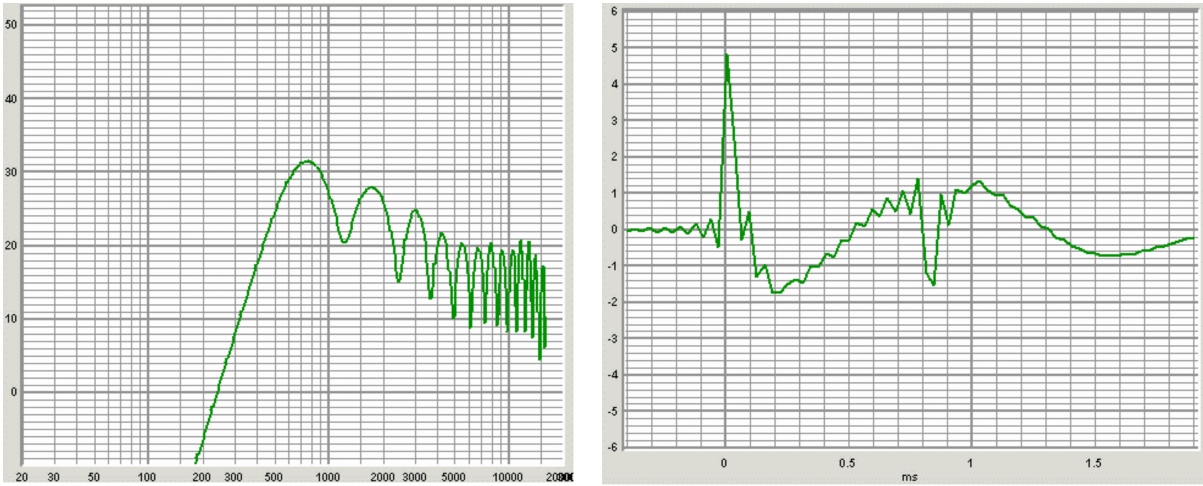

Fig. 2. Response at 60° off axis: ideal, continuous, 30° arc source.

Figure 3 illustrates an array of eight point sources spaced on 41 mm centers. Point sources are used in this example to emulate the characteristics of a very small transducer. The magnitude response of the array is shown at 0°, 10°, and 60°. Proper delay has been applied to individual sources to achieve a 30° pattern. The polar response shows that the array exhibits a nominal 30° pattern above 2 kHz. Note the similarities to the ideal source from figure 1 within the array's 30° pattern. However, outside this pattern, the array deviates significantly from the ideal source. Above 6.5 kHz, the array suffers from off axis lobes, a result of the source spacing exceeding the reproduced wavelength. Lobes are clearly seen in the polar response at 6.3 kHz, 8 kHz, 10 kHz, and 12.5 kHz.

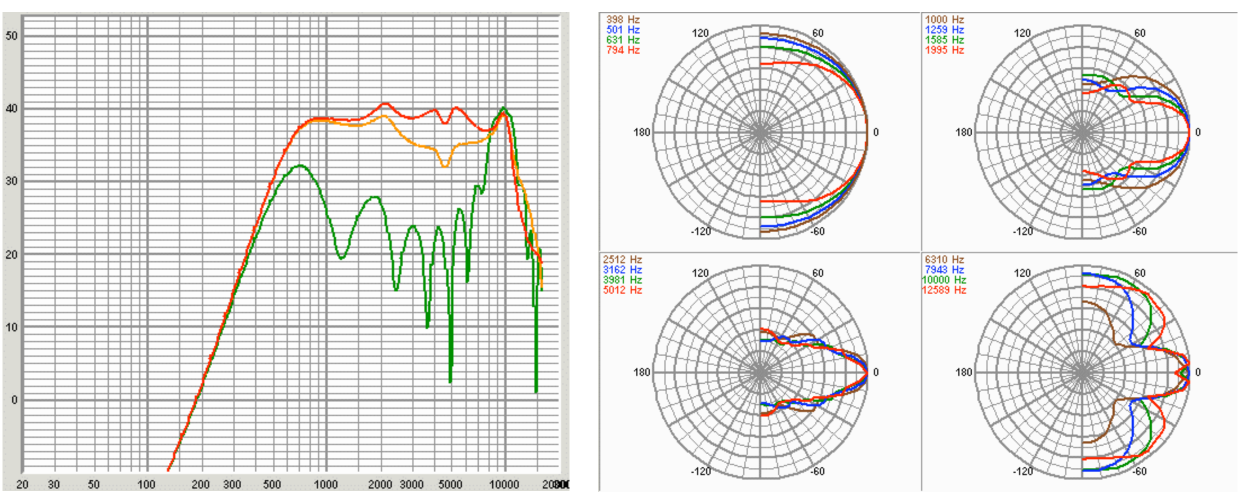

Fig. 3. Array of eight point sources on 41 mm centers, processed to 30° pattern. Magnitude - Red: 0°, orange: 10°, green: 60°. Polar - 400 Hz to 12.5 kHz, 1/3 octave.

Perhaps more interesting is the impulse response of the array at 60°, shown in figure 4. Here, rather than the two distinct positive and negative pulses shown in figure

2, a positive and negative pulse for each source is discernible. Each pulse's negative trailing edge lines up in time with the positive leading edge of the next pulse.

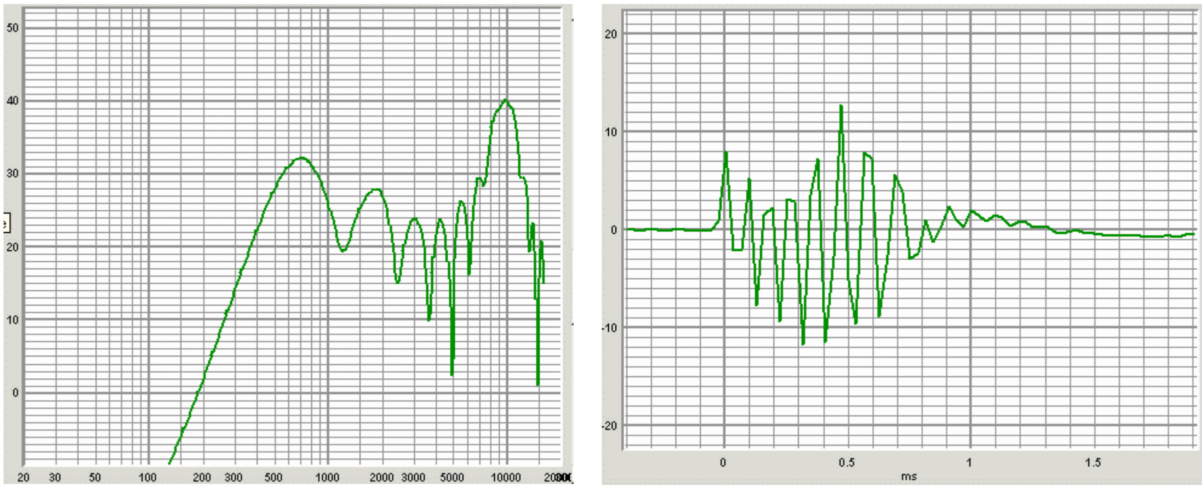

Fig. 4. Response at 60° off axis: array of eight point sources on 41 mm centers, processed to 30° pattern.

The directional characteristic of a line of directional sources is known to be the mathematical product of the directional characteristic of a single element of that line, and the directional characteristic of a line of point sources with the same arrangement. To achieve spacing as small as possible, very small sources are required. Since the directional characteristics of a small source mimics that of a point source within the audible frequency range, the only means of overcoming the limitations of the array of point sources shown in figures 3 and 4 above would be to decrease the source spacing beyond what is physically possible, or alter the pattern of the individual sources.

Figures 5 and 6 illustrate an edge-to-edge array of high frequency horns on 41 mm centers. Again, proper delay has been applied to individual sources to achieve a 30° pattern. The polar response shows that the array exhibits a nominal 30° pattern above 2 kHz. Unlike the array of point sources, the vertical pattern of each source in this array is controlled by its horn. The result is an overall polar response that more closely matches the behavior of the ideal continuous source outlined above. Additionally, the impulse response at 60° off axis, while still displaying evidence of each individual source, displays a much more prominent leading edge pulse and greater similarity to the impulse response of the ideal source.

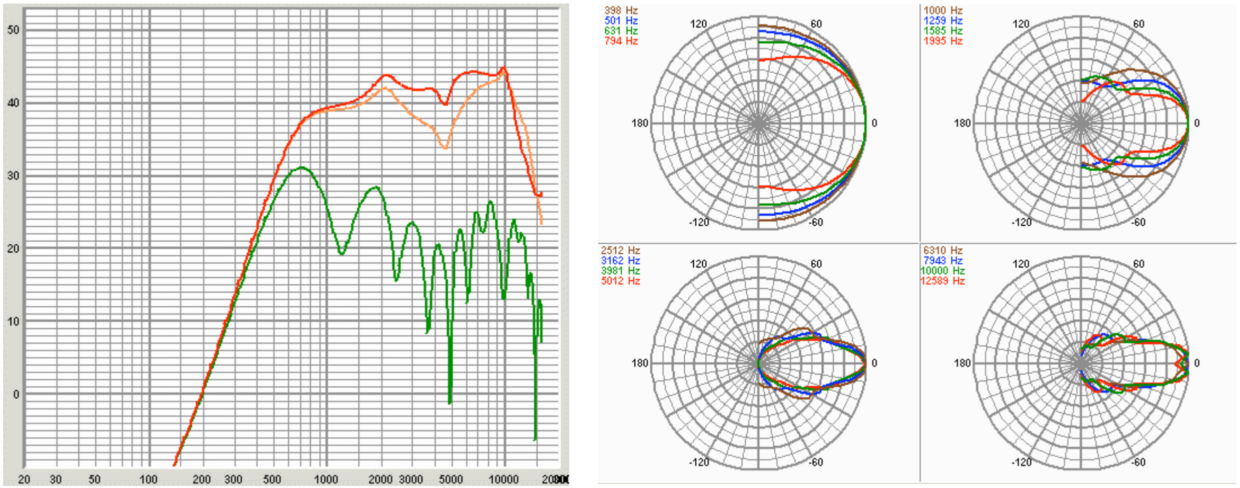

Fig. 5. Array of eight horns on 41 mm centers, processed to 30° pattern. Magnitude - Red: 0°, orange: 10°, green: 60°. Polar - 400 Hz to 12.5 kHz, 1/3 octave.

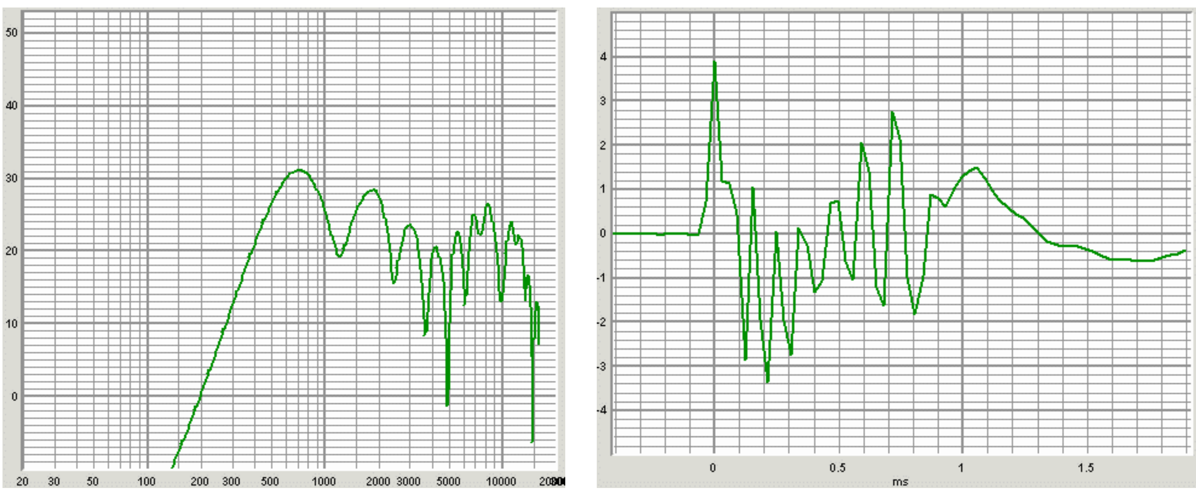

Fig. 6. Response at 60° off axis: array of eight horns on 41 mm centers, processed to 30° pattern.

1.1.2 Low Frequency Analysis

In order to successfully steer the upper frequency range of a linear array of low frequency devices, the optimum spacing calls for small cone transducers to be overlapped, a physical impossibility.

Recall that the directional characteristic of a line of directional sources is known to be the mathematical product of the directional characteristic of a line of point sources with the same arrangement, and the directional characteristic of a single element of that line. Therefore, it is imperative that the size of the transducer not be so large that its inherent directionality limits its usefulness at the upper end of its operating range. If the pattern of the source collapses at too low a frequency, it will be ineffective in integrating with neighboring sources to allow steering in that frequency range. At the same time, it is important that the size of the device be large enough to reproduce low frequencies.

Additionally, steerability is dependant upon source spacing that is proportional to the minimum wavelength reproduced. Assuming the maximum frequency could approach 4 kHz, the ideal source spacing would be around 85 mm.

In contrast, in order to achieve successful pattern control at lower frequencies, a line source length proportional to the maximum wavelength reproduced is necessary.

As with the analysis of the high frequency sources, performance of a single, ideal, continuous, line source was compared to a linear array of sources. In figure 7, the 0°, 15° and 60° off axis responses are displayed for an ideal, continuous, 30° arc source with an included arc of 30°. The polar response displays a well-defined 30° beam, with very little leakage beyond 30° above 500 Hz.

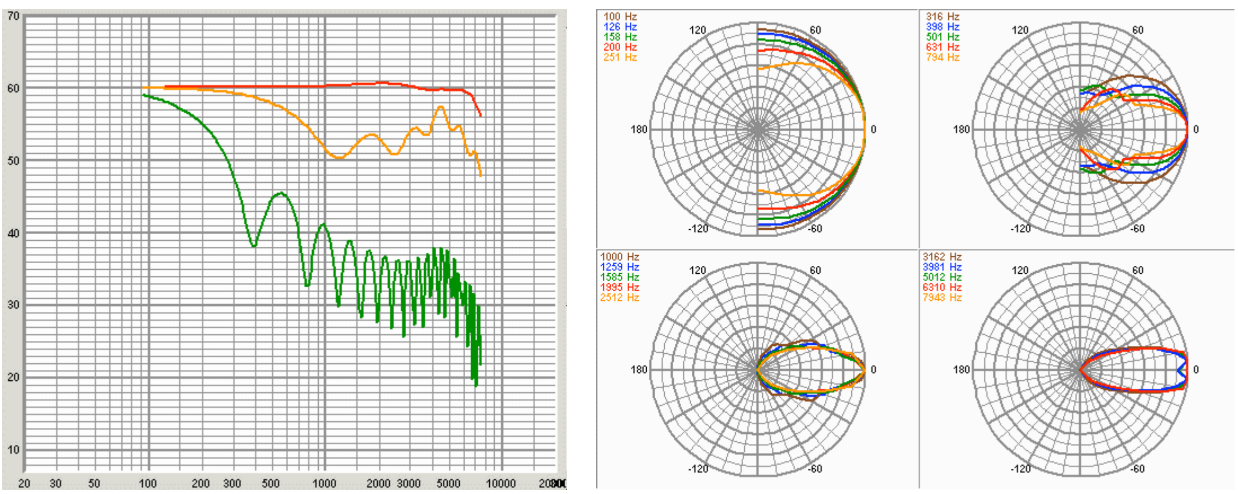

Fig. 7. Ideal, continuous, 30° arc source. Magnitude - Red: 0°, orange: 10°, green: 60°. Polar - 100 Hz to 8 kHz, 1/3 octave.

In figure 8, the 0°, 15°, and 60° off axis responses are displayed for a steered array with the same dimensions as the ideal source, but made up of eight 100 mm cone transducers on 125 mm centers. Both the magnitude and polar responses are very similar to those of the ideal source below 1.25 kHz. However, there is an abrupt deviation from ideal behavior beginning at 2 kHz.

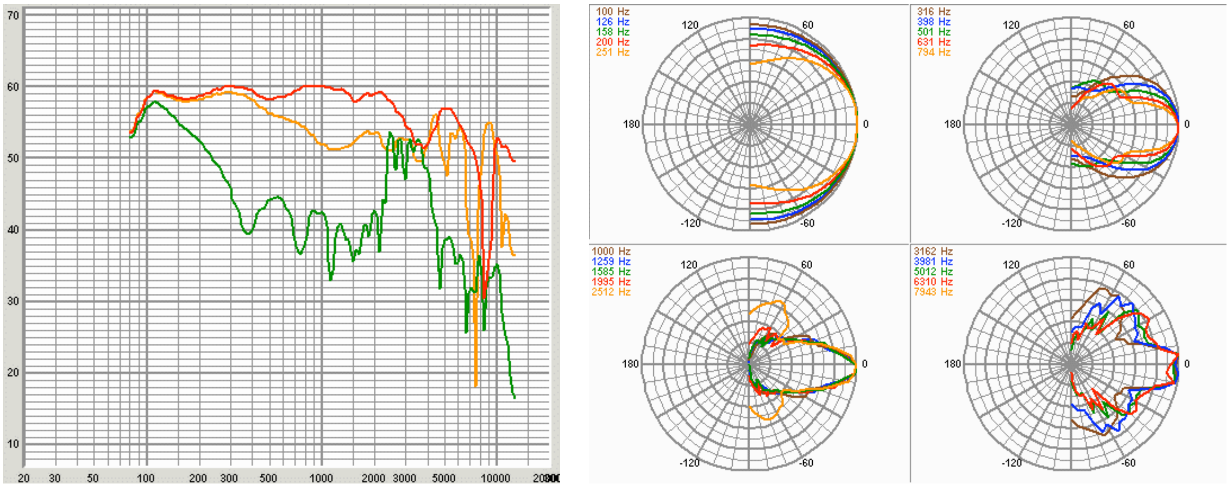

Fig. 8. Array of eight 100 mm cones on 125 mm centers, processed to 30° pattern. Magnitude - Red: 0°, orange: 10°, green: 60°. Polar - 100 Hz to 8 kHz, 1/3 octave.

The reason for this deviation can be understood by inspecting the impulse responses at 60° off axis. Figure 9 shows the 60° off axis response of the ideal source, while figure 10 shows the 60° off axis response of the steered array.

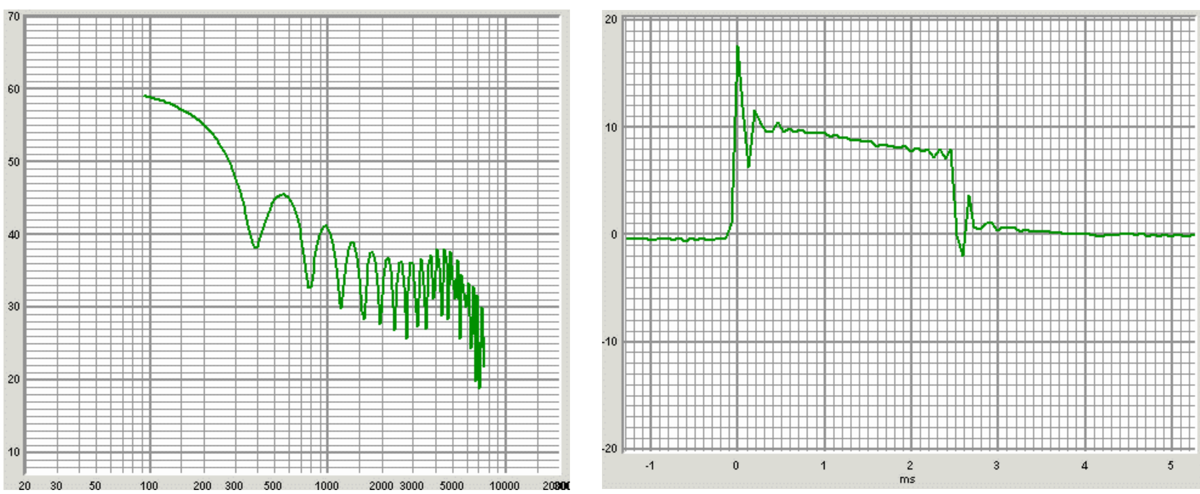

Fig. 9. Response at 60° off axis: ideal, continuous, 30° arc source.

The ideal source exhibits a positive-going pulse corresponding to the nearest edge of the source, a raised portion due to the continuous strength of the source, and finally a negative-going pulse corresponding to the far edge of the source[1].

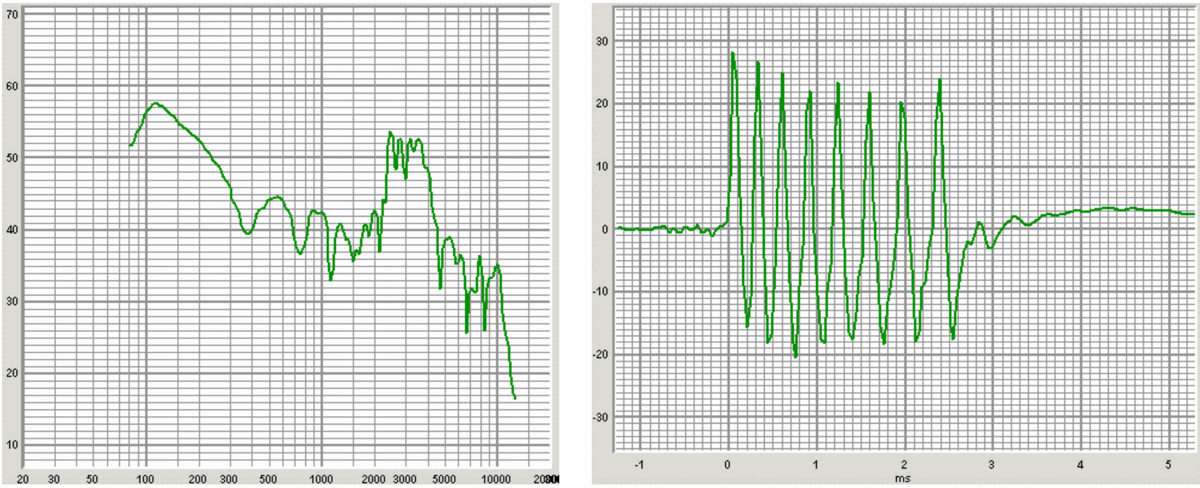

Fig. 10. Response at 60° off axis: array of eight 100 mm cones on 125 mm centers, processed to 30° pattern.

The steered array clearly consists of eight distinct arrivals, each with its own positive going leading edge, and negative-going trailing edge. This straight line of multiple small transducers provides the length necessary to achieve adequate control to 500 Hz; but although the small transducer size should allow for steerability above 2.5 kHz, the spacing of these devices limits their effectiveness in this range.

If the spacing is reduced to 75 mm, the impulse responses of each individual loudspeaker begin to "meld together", with the negative-going, trailing edge of one impulse nearly canceling the positive-going, leading edge of the following impulse. Figure 11 illustrates the magnitude and impulse responses of the same set of sources with their spacing reduced to 75 mm. Note that because the delays applied to the sources approximate a 30° arc, the initial impulses arrive closer together in time than the final ones. Hence, the first few impulses meld more than the later ones.

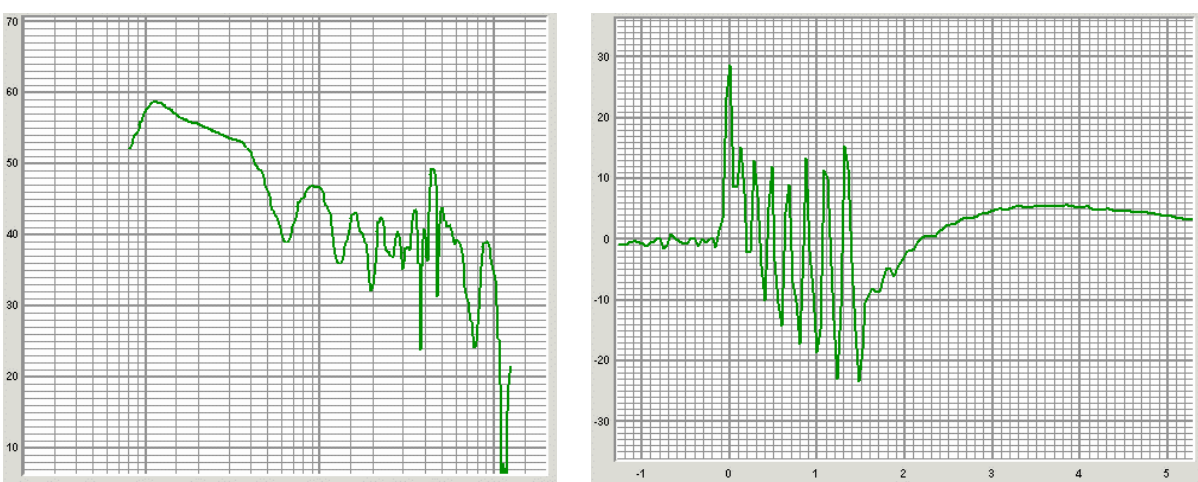

Fig. 11. Response at 60° off axis: array of eight 100 mm cones on 75 mm centers, processed to 30° pattern.

Figure 12 shows that the magnitude and polar responses are much closer to the ideal source than the same set of sources with 125 mm spacing.

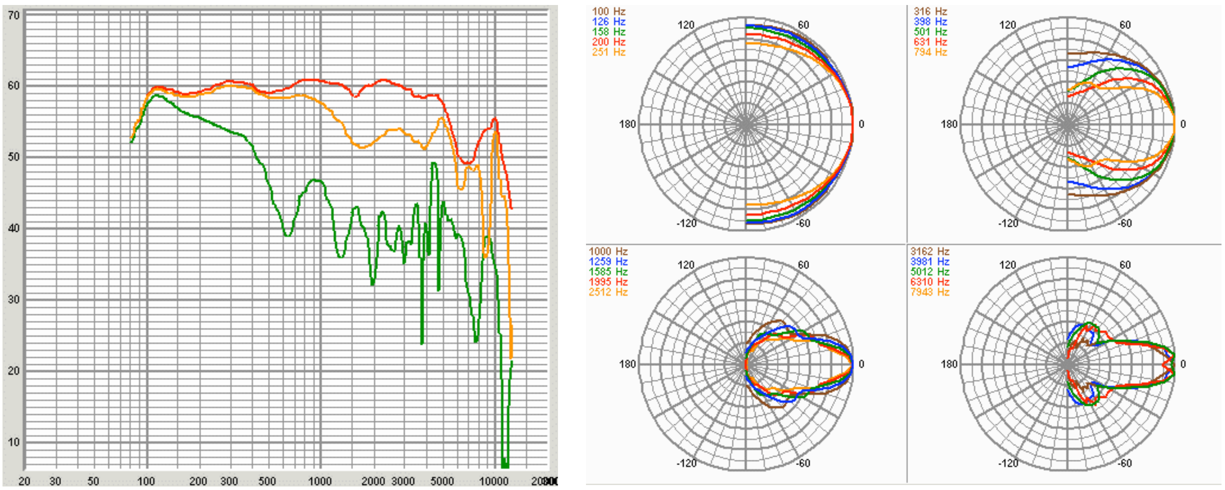

Fig. 12. Array of eight 100 mm cones on 75 mm centers, processed to 30° pattern. Magnitude - Red: 0°, orange: 10°, green: 60°. Polar - 100 Hz to 8 kHz, 1/3 octave.

It might appear that the solution to this problem would be simply to space the transducers 75 mm apart, as modeled here. However, the transducers that produced these results are 100 mm units, with a smallest external dimension and minimum linear spacing of 115 mm.

Fortunately however, a staggered array provides the spacing density required for upper frequency steering. This solution offers the additional benefit of slightly improved control in the horizontal plane, with negligible detriment to the directional results in the vertical plane.

1.1.3 Implemented Design

The chosen implementation utilizes a high frequency section of eight 25 mm transducers coupled to an edge-to-edge array of horns spaced 41 mm on center. The low frequency section utilizes a staggered array of eight 100 mm cone transducers. Two models currently exist, one utilizing both the high and low frequency arrays, the other containing just the low frequency array. The software-based modular design approach allows multiple loudspeakers to be combined, allowing arrays to be tailored to the specific application. This approach allows longer line lengths to extend pattern control to lower frequencies and/or achieve additional output. The goal of these systems is to perform steering in the vertical plane. Therefore, they are designed for installation flat against a wall in their vertical orientation. Examples are shown in figure 13.

Fig. 13. Examples of digitally steered loudspeaker arrays. Two-way module, one-way module, multi-enclosure system for extended pattern control.

1.2 Hardware Implementation

The loudspeakers shown above contain integral digital signal processing (DSP), amplification, and communications. The techniques used for steering an array of transducers require that each have its own processing and amplification channel. The full-range loudspeaker shown above contains sixteen DSP and amplifier channels, shown in figure 14. Likewise, the low frequency loudspeaker contains eight DSP and amplifier channels.

Each transducer is relatively small, so its associated amplifier has a relatively low power rating. But because sixteen of these amplifiers are used (in the case of the full-range model), significant output levels are still possible. Since fan cooling is not desirable for a loudspeaker system, the high efficiency amplifiers utilize the entire aluminum enclosure for convection cooling.

Steering is accomplished via dedicated DSP using a combination of traditional and proprietary digital filters. Crossover functions and user adjustable input parameters such as EQ, delay, gain, and compression are handled with additional DSP.

Control of these DSP parameters, as well as a comprehensive monitoring system, is accomplished via an integrated EIA 485 (RS 485) transceiver. This allows the loudspeaker to be connected to an EIA 485 network for control via a personal computer.

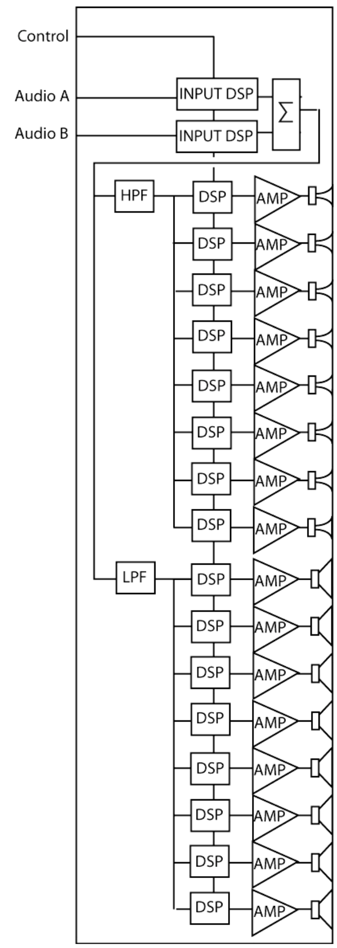

Fig. 14. Block diagram of DSP

1.3 Software Implementation

Digitally steered arrays must be accompanied by software that allows the users to determine the settings. For a particular array, the position and orientation of each individual enclosure must be communicated to the software, as well as the desired coverage area. After the optimum signal processing settings are calculated, they are uploaded from the computer to the signal processors in the array so that the intended design may be realized. Traditionally, sound system designers have specified loudspeaker coverage in terms of aiming angles and direction. While the traditional specification is effective in some situations, it is unnecessarily limiting. To realize the full potential of digitally steered arrays, we introduce a new method of specifying the desired coverage – which we refer to as the "audience coverage" method..

1.3.1 Traditional Directionality Specification

Several common terms have been used to characterize loudspeaker directionality over the years. "Directivity index," while a useful measure, was not sufficiently precise for detailed system layout work because it ignored the differences between the horizontal and vertical behavior. The horizontal and vertical beamwidth was more descriptive, and served both as an intuitively useful specification and as a figure of merit, when plotted against frequency. Because many practitioners are comfortable and familiar with beamwidth specifications, the software supports a simple, traditional, method for specifying the vertical coverage pattern. The beamwidth is specified, along with an off axis steering angle labeled, "steering" – to substitute for the physical aiming of the loudspeaker. Figure 4 defines the terms.

The third parameter in the traditional method is "focal distance," which has two effects on the end result. It creates a beam whose edges correspond to the nominal beamwidth at the specified distance; and, it adjusts the steering angles so that the beams from the high frequency and low frequency sections converge at the focal distance, even though the arrays that produce them are located in slightly different places. Figures 6 illustrates these two effects. In normal use, the primary effect of this parameter will be on the pattern edge. If the entered focal distance is too long, the lower pattern edge might be a bit brighter than normal, because the low frequency array would not be steered down enough. If it is set too short, the lower pattern edge might lack brightness.

The focal distance parameter also allows the system to create a beam smaller than the array that produced it. In other words, a zero-degree entry for beamwidth will actually produce a converging beam which achieves maximum pressure at the focal distance.

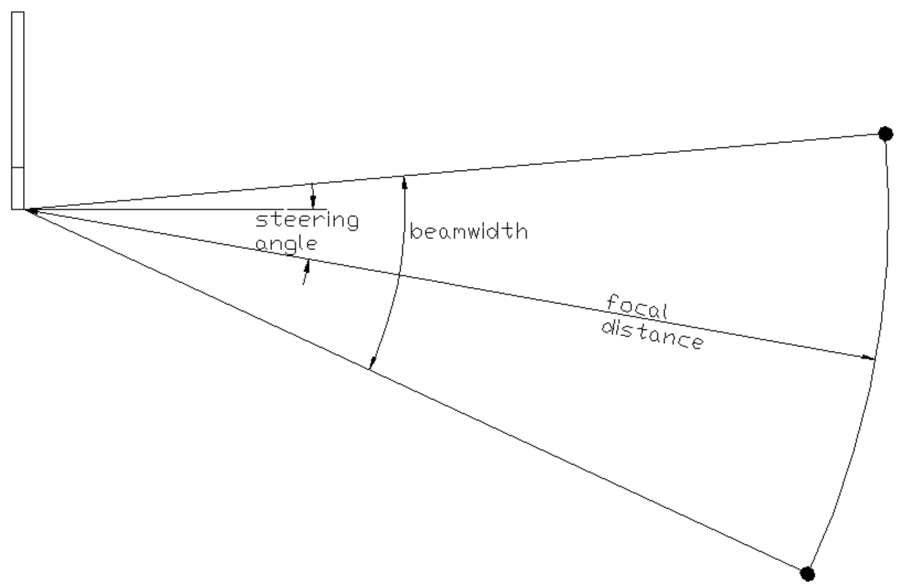

Fig. 4: Traditional Coverage Specification Method Terminology

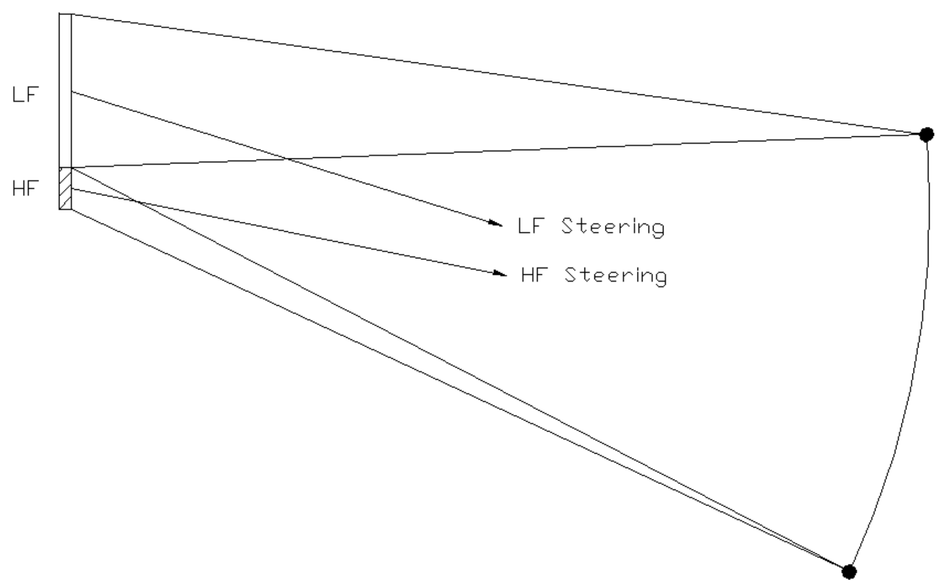

Fig. 5: Beam Convergence at the Focal Distance

1.3.2 Audience-Coverage Directionality Specification

While the traditional specification is effective in some situations, it is unnecessarily limiting. Digital steering is capable of producing a beam which varies in intensity within the angular constraints of the beam. All that is required is a definition of the desired coverage area, and the steering algorithms can attempt to fit the coverage in both direction and distance. For the vast majority of applications, a sufficient level of detail can be obtained with a simple, two-segment approximation of the cross-section.

The two segments are defined by a start point, an inflection point, and an end point. Such a cross section is illustrated in figure 6.

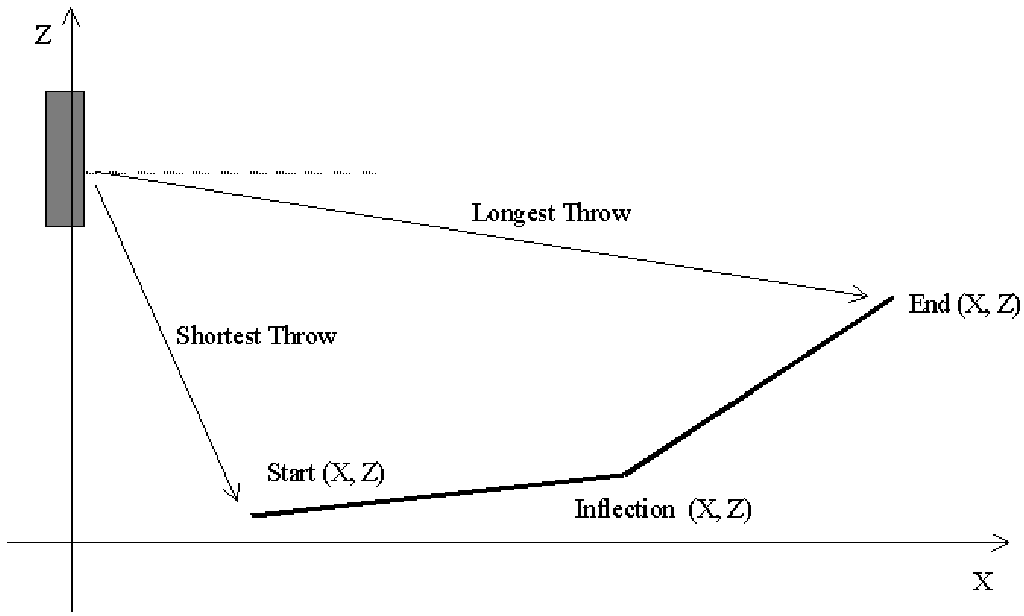

Fig. 6. Two-Segment Definition of Listening Area

The software interface, illustrated in figure 7, implements both the traditional method and the audience coverage method. In the upper left hand corner are the parameters that define the two-segment listening area. The three points defined by these parameters are represented graphically by the blue lines in the venue cross section. The last point might represent the end of a contiguous listening area, or it might represent the last seat in a balcony. In the case of a room with a balcony, the inflection point may be placed somewhat arbitrarily, to either encourage or discourage aggressive coverage of the back of the main floor, under the balcony.

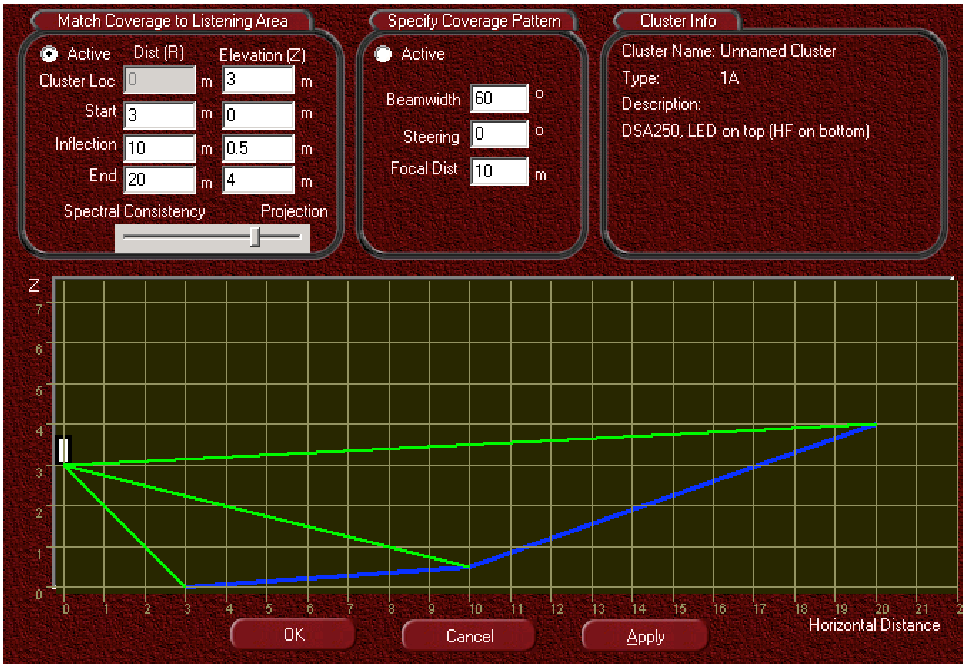

Fig. 7. Sample software interface

The defined listening area is displayed even when the traditional beamwidth method of specifying coverage is active. This allows the traditionally defined beam and its off-axis steering to be matched to the profile of the room.

Having established the coverage goals for the loudspeaker cluster, the next task is to determine a set of digital signal processing parameters that achieve the desired results. While delay adjustments alone are capable of achieving significant beneficial steering and beam forming, one or more of a variety of secondary techniques can be used to improve the results over a simple delay scheme. These techniques affect the precision of the polar response, the relative smoothness and consistency of the frequency response in various directions, and can alternately enhance either intelligibility or musicality. By varying the degree to which each technique is employed, the software allows the steering parameters to be optimized according to the specific needs and objectives of a particular installation.

The user communicates those objectives by way of a slider control which allows the objective of the steering algorithm to be varied from “Spectral Consistency” to “Maximum Projection,” with several intermediate settings in between. The affect of this setting will be explored in Section 2.4, below.

2 CHARACTERISTICS OF DIGITALLY STEERED ARRAYS

Digitally steered arrays are capable of creating directional patterns of varying beamwidth and steering those patterns off the primary physical axis of the device. A major advantage of such an array is the character of the radiation pattern. The pattern produced by a digitally steered array is not the same as that produced by tilting a conventional loudspeaker with the same horizontal and vertical beamwidths.

A typical loudspeaker that is tilted down to cover a listening area creates several problems. First, the geometry of the down-tilted loudspeaker's pattern is arc-shaped with a main lobe emanating from its primary physical axis. The width and directionality of the beam results in inadequate coverage across the front area of the intended listening area. In addition to this, the pattern produced by a down-tilted loudspeaker projects a distinct band of energy onto the side walls of the venue, which of course arrives at the listener as a late reflection. This reflection can be nearly as strong as the primary arrival which reaches the listeners directly. An example of this behavior is shown in Figure 8a.

2.2 Suppressing Lateral Reflection

On the other hand, digitally steering an array produces a far more consistent SPL across the intended coverage area. With the long dimension of the array vertical and signal processing used to direct the energy downward, the side radiation projects downward at the same angle as the front radiation. This provides far more uniform sound levels in the listening area. In addition to this, the energy that is directed to the side walls is much lower in level and creates a very different reflection pattern off of those walls. The reflected sound is directed into the floor or the nearby seats along the wall early enough in relation to the direct sound that it can actually enhance intelligibility. See Figure 8b.

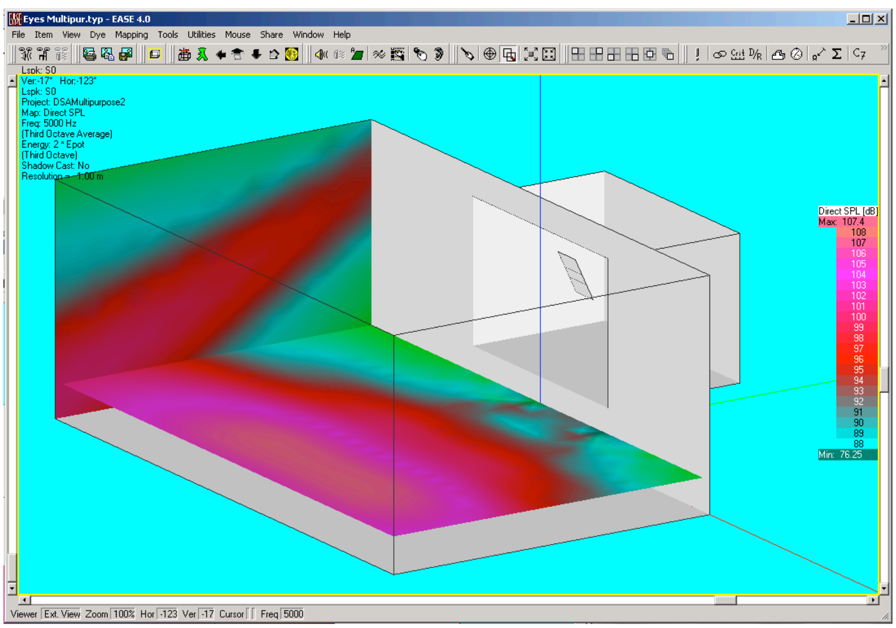

Fig. 8A. Conventional Line Array Coverage, Tilted Down

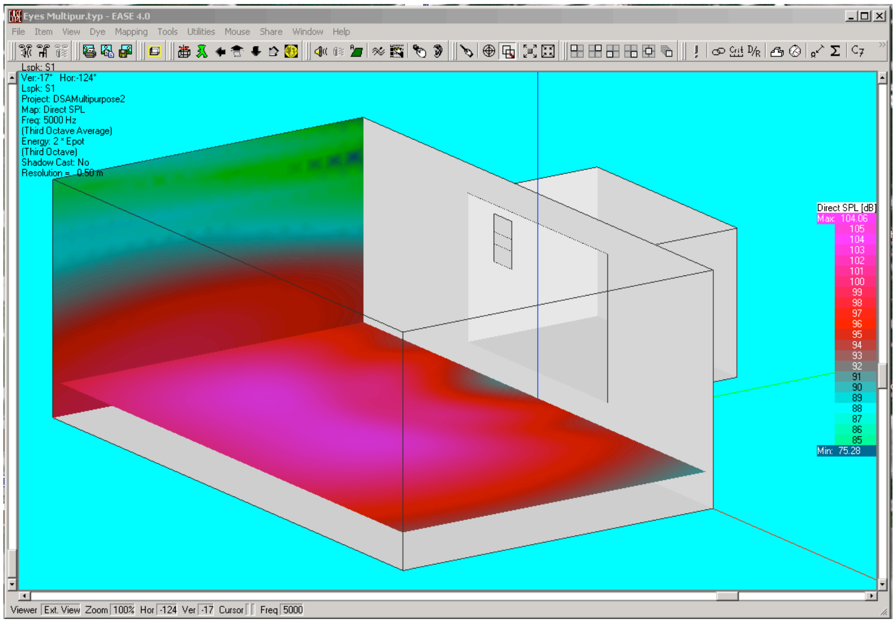

Fig. 8B. Digitally Steered Coverage

2.3 Back Radiation

It is important to be aware of the back radiating characteristic of a steered array. Because the enclosures are less than 25 cm wide, the horizontal beamwidth does not narrow to 180 degrees until 800 Hz. At frequencies lower than 800 Hz, the back radiation will be increasingly strong. Furthermore, the backward radiating beam will be steered down at the same angle as the front radiation. This can be an important consideration if these arrays are suspended in mid-air, rather than mounted to a wall. When mounted to a wall, the back radiation reflects back into the front hemisphere with the same downward angle as the front radiation. Consequently, its effect is relatively benign, particularly through the range of frequencies relevant to vocal reinforcement.

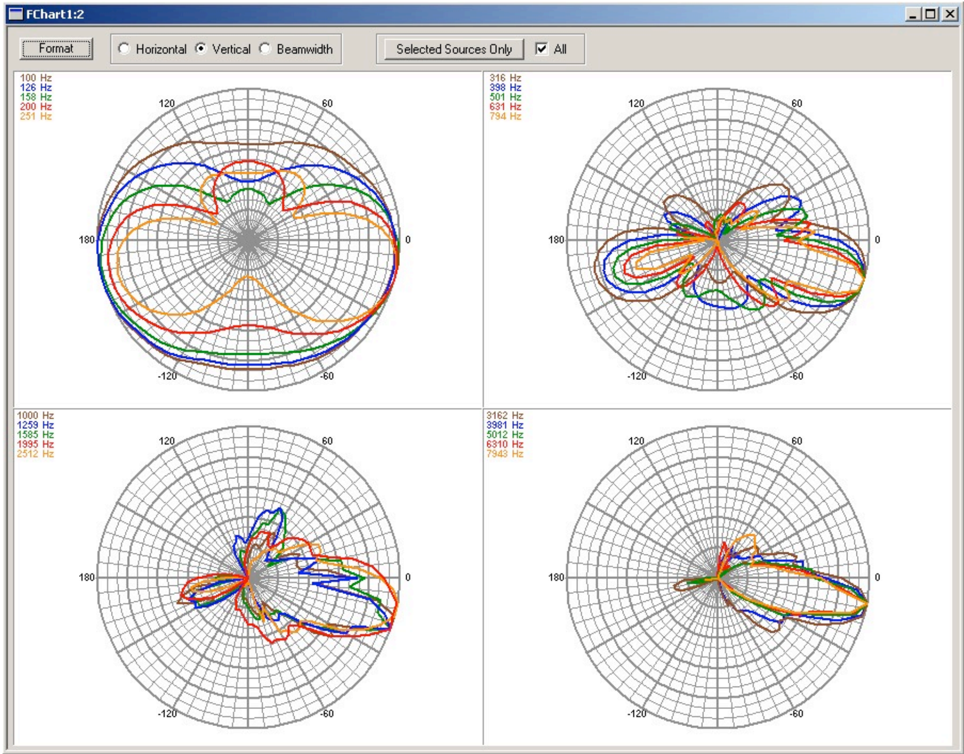

Fig. 9: 15 degree beam steered –10 degrees, showing back radiation

Figure 9 displays the back radiation for a 15 degree beam steered down 10 degrees. Note that the level of the back lobe at 316 Hz is within about 5 dB of the level of the front lobe. At 800 Hz, there is about a 12 dB differential. Figure 10 shows the same array mounted to a wall. Note that the polar response above 300 Hz is not appreciatively affected by the presence of the wall, even though there is a significant contribution from the back radiation from 300 Hz to 800 Hz. The apparent widening of the 200 Hz beam is actually an indication of destructive on-axis interference between the front lobe and the reflected back lobe.

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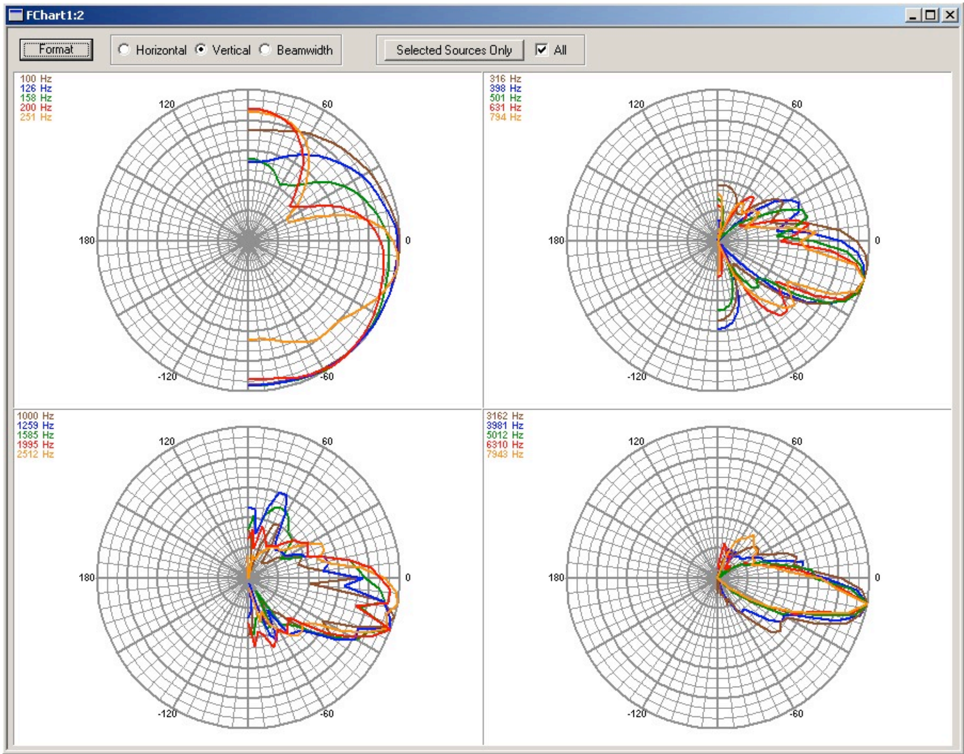

Fig. 10: Steered array mounted on a wall, same settings as figure 10

2.4 Spectral Consistency vs. Projection

In a well damped room with naturally high intelligibility, considerable flexibility exists as to how a sound system may be tuned. One might accept a significant degree of variability in the SPL produced in different parts of the room, in order to obtain a consistent spectral balance throughout. In a hard, reverberant, challenging room, on the other hand, one might accept a significant degree of spectral variation, in order to obtain the maximum possible direct-to-reverberant ratio. This is the trade-off offered by the spectral consistency vs. projection slider. As illustrated in the uppermost chart in figure 11, the high frequency coverage can be matched very precisely to the requirements of the room, in order to obtain nearly equal SPL throughout. However, the same degree of control is not possible at low frequencies. As a result, the direct sound delivered to the farthest listener will be considerably brighter than that delivered to the closest listener.

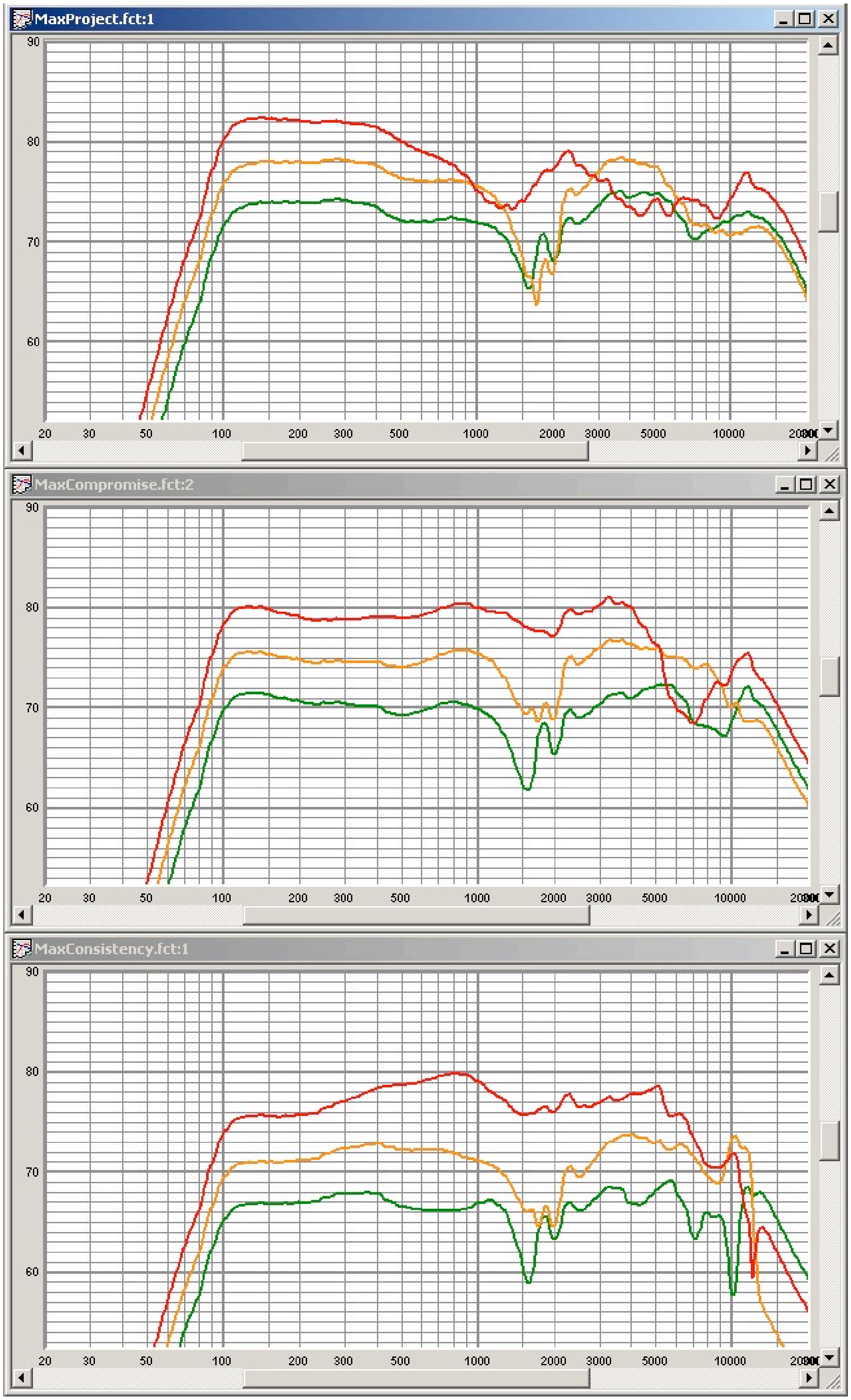

Fig. 11: Varying The Objective from Maximum Projection to Maximum Spectral Consistency, response calculated at 5 m, 15 m, & 25 m

With maximum spectral consistency selected (the lowest chart in figure 12), the spectrum of the direct sound is quite consistent up to about 7 kHz, and the response at the nearest microphone (shown in red) is somewhat smoother.

This consistency and response smoothness, however, comes at the expense of maximum output. All of the charts are normalized to represent equal drive level to the hardest-working transducer. So, the approximate 7 dB reduction in the long-throw response (shown in green) is representative of the maximum level achievable at the back of the room. The reason for this is that the spectral consistency setting allows the use of steering filters which affect the overall drive level to the various transducers. Consequently, some of them will be running at less than maximum level when the hardest working one reaches its maximum. With the slider in the maximum projection position, the transducers are all driven equally hard. Driving all the transducers with equal level has been found to produce the most clearly defined focus at the farthest listener, which results in maximum intelligibility. It also has the side effect of maximizing the total sound power output capability.

2.1 Performance Limits

Regarding the performance of digitally steered systems, two of the most frequently asked questions are, “How far off axis can the system be steered?” and “What range of beamwidths can it achieve?” Unfortunately, these are very difficult questions for which to provide simple answers. In order to provide a performance limit, one must define what constitutes a failed setting. This definition is not always obvious.

2.1.1 Off Axis Steering Limits

It is possible to create a focal point as much as 90 degrees off axis. However, the primary limitation for off-axis steering is out-of-beam leakage. To define a limit for off-axis steering capability, then, one must define the limit of acceptable out-of-pattern response. In particular, narrow beams steered radically off axis produce severe out-of-beam leakage. The acceptability of such leakage may depend on the the acoustical properties present in a prospective application.

As a starting point, let’s look at a relatively narrow beam, steered at various angles off axis. Figure 12 shows the polar response of a small array (one of the two-way enclosures) projecting a 30 degree beam, and steered 10 degrees off axis.

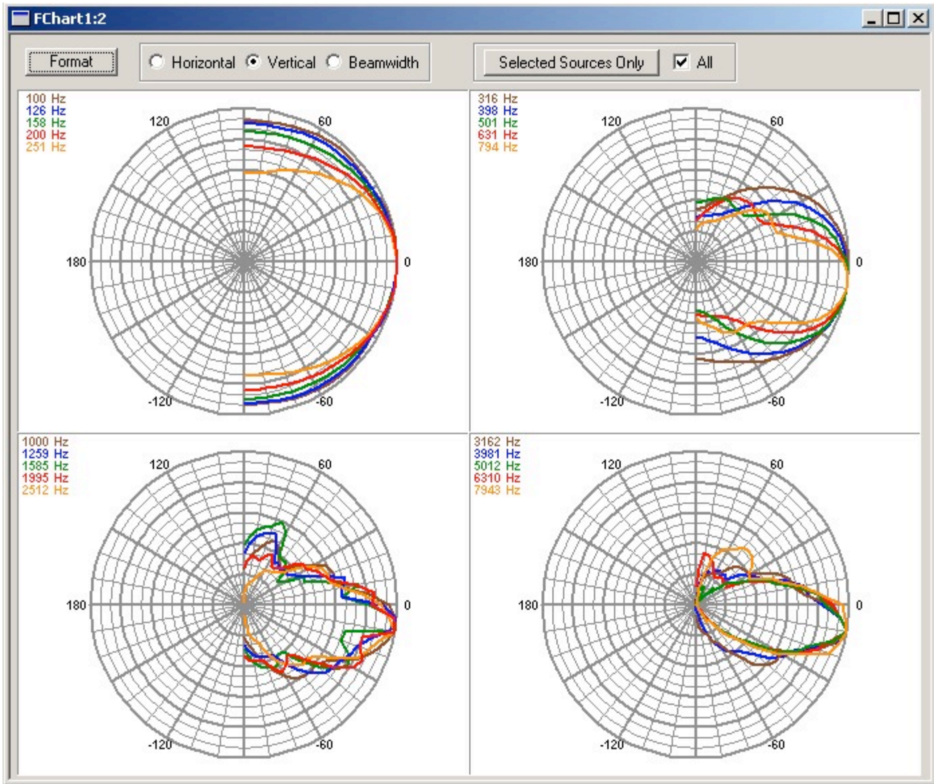

Fig. 12: Small Array, 30-degree beam steered – 10 degrees

There is little to criticize in this set of polars, though there is a hint of grating lobes beginning to form between +60 and +90 degrees at 1250 Hz to 1600 Hz.

Figure 13 displays the polars for a 30 degree beam steered down 30 degrees. In this example, the grating lobes have become more obvious, particularly at 8 kHz, at which frequency the out-of-pattern lobe nears –6 dB from the main beam.

Figure 14 displays the polars for a 30 degree beam steered down 50 degrees. With the grating lobe at 8 kHz reaching the same level as the main lobe, this setting is clearly "beyond the steering limit". However, this could still be a useful configuration in an application in which the upward-directed high frequency energy was deemed not to be a problem, or in which the system could be low-passed at 5 kHz (i.e., speech only).

Based on this series of data, one might state the limit of steering as –30 degrees; with the criteria being that the largest grating lobe is at least 6 dB lower in level than the main beam. Of course, the choice of criteria is purely arbitrary.

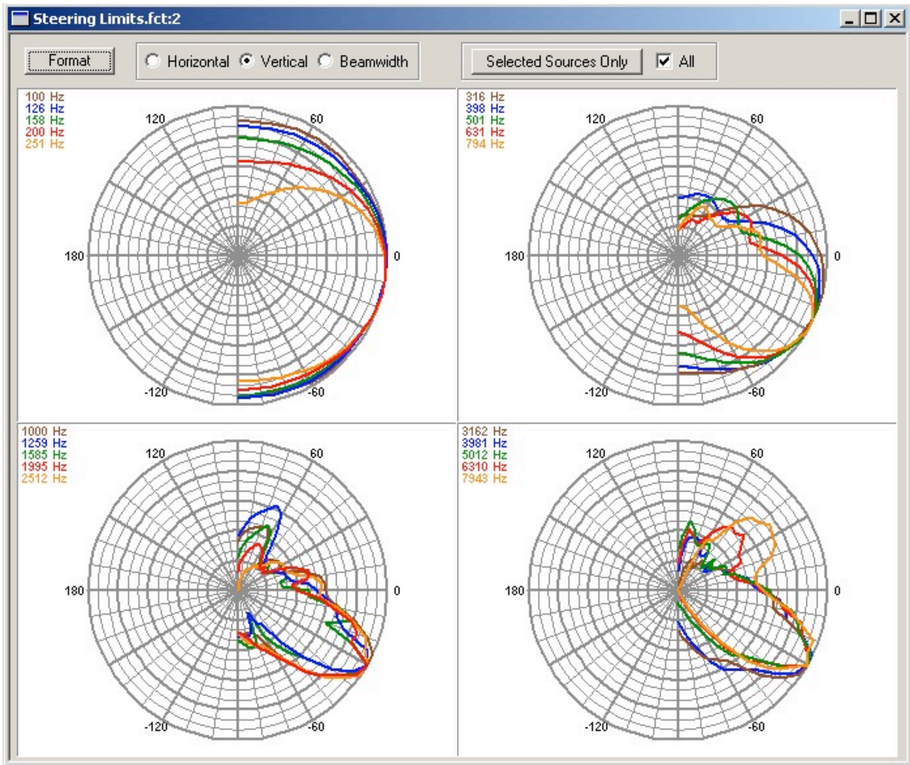

Fig. 13: 30-degree beam, steered – 30 degrees

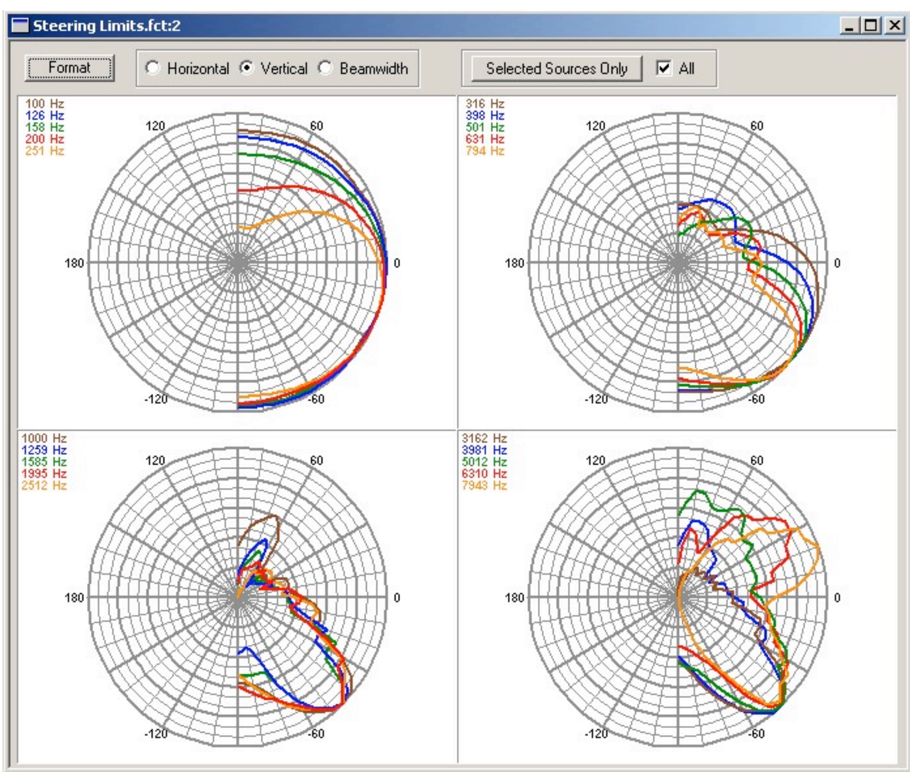

Fig. 14: 30-degree beam, steered –50 degrees

Having established that the limit of off-axis steering is 30 degrees for a relatively narrow beam, let's look at an example of an audience-coverage steering setting. In the example presented in Figure 15, the coverage extends to below –60 degrees; yet, there is barely a hint of a grating lobe at 1250 Hz. Clearly, the previous analysis does not hold for a complex audience-coverage beam. In this configuration, most of the energy is directed nearly axially, with only a small portion of the energy directed in the extreme off axis direction

In fact, experience has shown that the system is useable down to –90 degrees when the audience-coverage steering algorithm is employed.

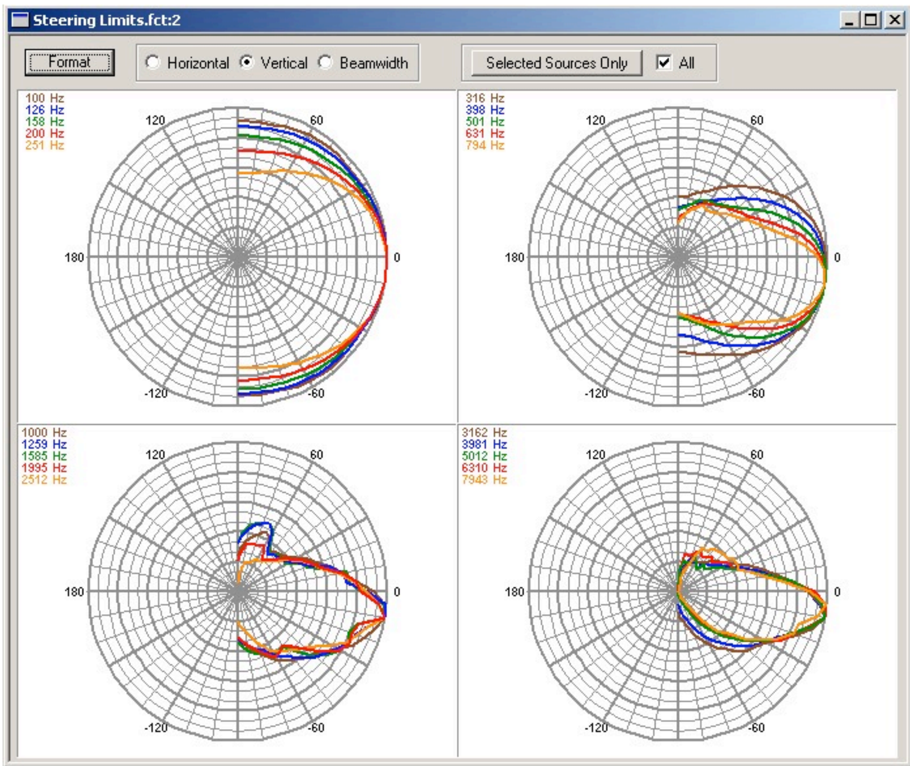

Fig. 15: Audience Coverage beam, The listening area (relative to loudspeaker) is Start Point: (3.5 m, -6 m), Inflection Point: (20 m, -5 m), and End Point: (30 m, -2 m)

2.1.2 Beamwidth Limits

The beamwidth limits are much easier to quantify. The customary practice when specifying the effective frequency range of control for constant directivity horns is to specify the lower frequency at which the beamwidth is 1.5 times nominal, and the upper frequency at which the beamwidth collapses to 2/3 of nominal. If we model a 150 degree beam, and inspect the polars (Figure 16), we see that the beamwidth collapses to 100 degrees at about 5 kHz. If 5 kHz is sufficient bandwidth for a given application, then a 150-degree beam could reasonably be used.

If we model a 90 degree beam (Figure 17), we see that the beamwidth stays wider than 60 degrees beyond 10 kHz, which exceeds the limit of horizontal control.

Consequently, it is reasonable to state the maximum beamwidth as 90 degrees, without stating a frequency limitation.

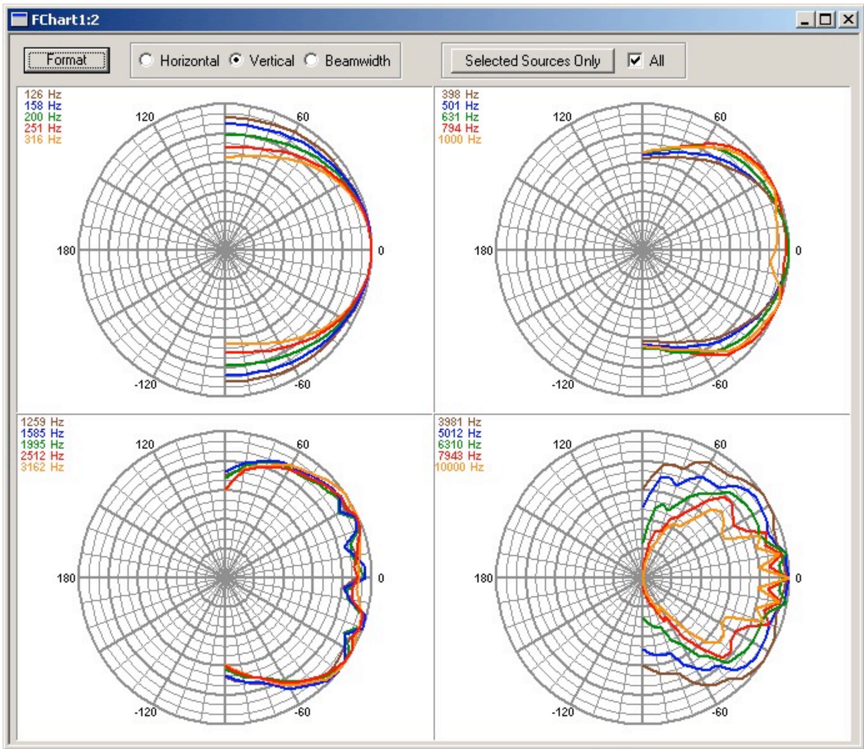

Fig. 16: Small array configured for 150 degree beam

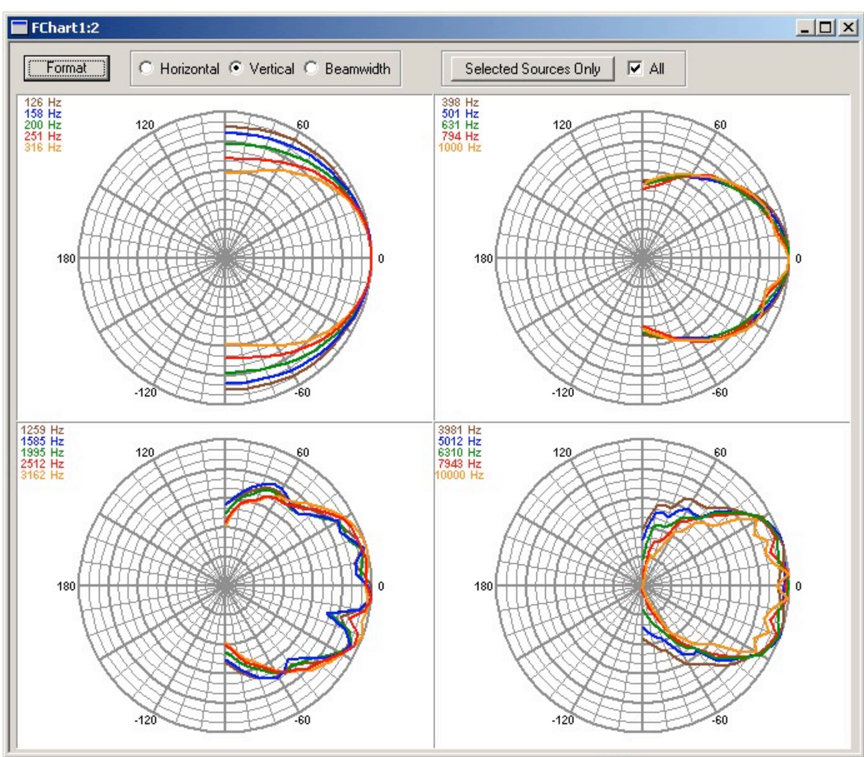

Fig. 17: Small array configured for 90 degree beam

3 CONCLUSION

The requirements for successfully steering a wide-bandwidth array of transducers and an implementation thereof were described. This two-way implementation specifies a high frequency section of small transducers coupled to an edge-to-edge array of tightly spaced horns. The low frequency section requires a staggered array of small cone transducers.

Details of the internal hardware were also presented. The number of processing and amplifier channels required for a digitally steered array necessitates integral electronics. This significantly simplifies use and setup. Communication circuitry for control of DSP parameters and monitoring is also described.

An associated software interface implements a method of specifying coverage which incorporates the coordinates of the listening area, rather than limiting input to the traditional terms, "beamwidth", and "aiming angle".

Some of the unique characteristics of digitally steered arrays were explored. A valuable benefit is the inherent advantage of a pattern which is directed downward both on axis and off axis. This pattern tends to reduce lateral reflections in a room, and provide broader coverage in the nearfield. The unique nature of these arrays' back radiation was also noted.

Finally, the limits of acoustical performance were explored, and the limiting factors identified. The audience-coverage method of specifying the directionality target was determined to be especially immune from grating lobes.

4 ACKNOWLEDGEMENTS

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