LOUDSPEAKER ARRAY PREDICTION

R Schwenke Meyer Sound Laboratories, Berkeley CA, USA.

1. INTRODUCTION

The chief concern of sound designers is adequate and evenly distributed sound power with flat frequency response over an audience area. The sound pressure level (SPL) at any point in the audience will be dominated by the direct sound and the first few reflections. Also, the ear integrates the first (approximately) 50 milliseconds after the direct sound into a direct sound sensation, and the sounds arriving after 50 milliseconds into a reverberant sound sensation. The subjective sound level and frequency response experienced by the audience will be dominated by the direct sound and first few reflections, and therefore by the nulls and lobes of the loudspeaker array. Low resolution, magnitude only loudspeaker polar data can yield adequate predictions of scalar, long term averaged quantities, such as reverb time or clarity. This information is useful for designing a room. However, to design an array of loudspeakers, high resolution complex polar data is necessary to accurately model the frequency response and direct field caused by the interaction of loudspeakers in the array. MAPP Online is an acoustic prediction program specifically made for designing loudspeaker arrays. It uses high resolution, complex loudspeaker data and has been validated with measurements of real speaker arrays.

2. MAPP COMPUTATIONAL ALGORITHM

MAPP utilizes complex loudspeaker transfer functions measured with 1/24 octave frequency resolution in one degree angular increments. The far field model of acoustic propagation is applied to each individual loudspeaker in an array and summed to find the transfer function at any point in space. MAPP has been validated both empirically and analytically [1,2,3]

3. A SIMPLE EXAMPLE

The directivity of two omni directional sources separated by a distance d is [4]:

$$H_B(\theta) = \cos\left(\frac{1}{2}kd\sin(\theta)\right)$$

At the frequency where d is equal to one half wavelength there will be two nulls in the directivity, when equal to one wavelength there will be four nulls and so on. Figure 1 shows the sound field of two omnis spaced the width of an MSL4, at 500 Hz (which is approximately where the separation equals one wavelength). Figure 2 shows the frequency and impulse response of a microphone placed in one of the nulls, which exhibits deep comb filtering as expected. The product theorem, as defined by [4], says that the directivity of an array of identical directional sources can be computed by multiplying the directivity of an individual element by the directivity of the array with all the sources replaced with omni-directional sources:

$$H_t(\theta) = H_a(\theta)H_e(\theta)$$

where H_e is the directivity of a single element of the array, H_a is the directivity of an array of omnis with the same spacing and weighting, and H_t is the total directivity of the array.

Unless otherwise specified, all sound field plots will be normalized with the highest SPL value set to 0 dB, and a dynamic range of 42dB. The corresponding SPL colour palette is shown in Figure 4.

Figure 3 shows the sound field from a single MSL4 at 500Hz. Figure 5 shows a parallel array of two MSL4. Figure 6 shows the frequency and impulse response from a microphone placed in the null, which again exhibits the same comb filtering as seen for the omnis in Figure 2. Figure 7 shows the sound field of a pair of tight packed MSL4s. The spatial null is no longer so deep, but the improvement is more easily seen in the frequency and impulse response in Figure 8 where there is significantly less combing. Note how the arrival time is later for the measured MSL4s versus the theoretical omni directional source. This represents the internal delay of the speaker due to the crossovers and mechanical transfer function of the drivers. This delay will be different for different models of loudspeaker. In order to correctly predict spatial nulls and comb filtering special care must be taken to retain the internal delay of the speaker when it is measured.

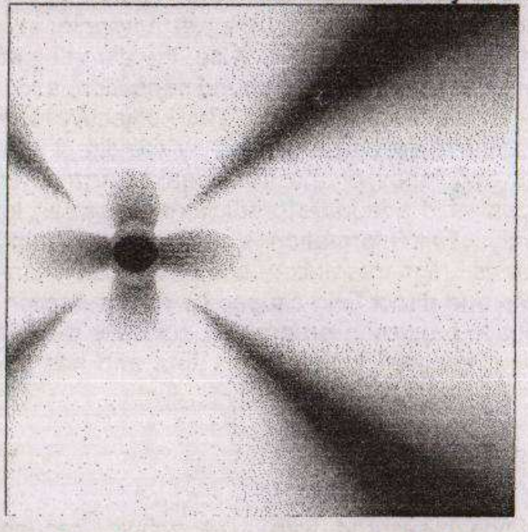

Figure 1 Two omnis spaced the width of an MSL4, at 500 Hz

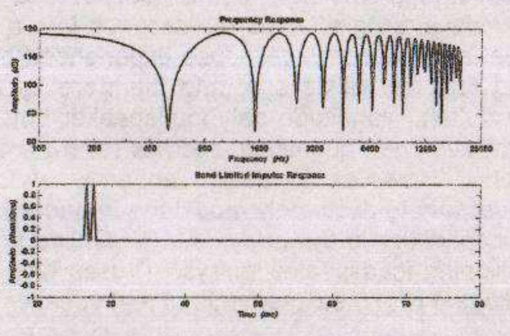

Figure 2 Frequency and Impulse Response of Two omnis 45 degrees off axis

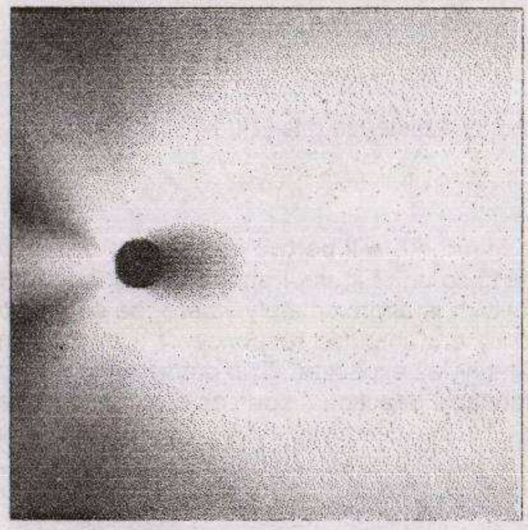

Figure 3 A Single MSL4 at 500 Hz

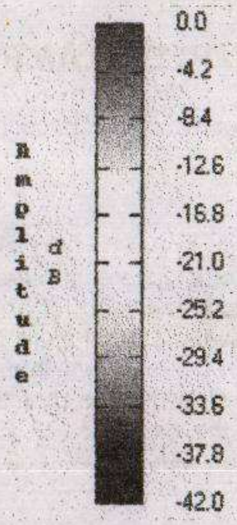

Figure 4 Default SPL Palette for Sound Field Plots

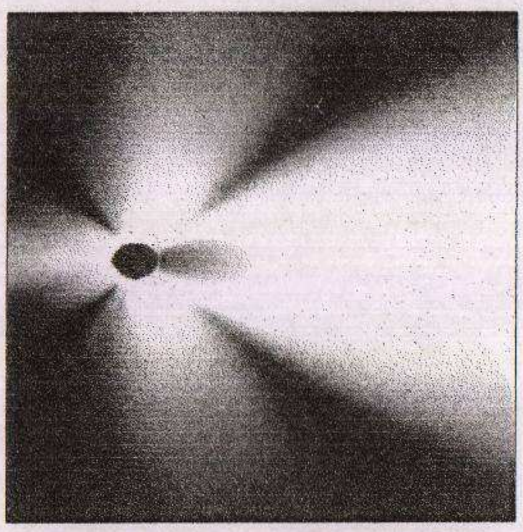

Figure 5 Two parallel MSL4s at 500 Hz

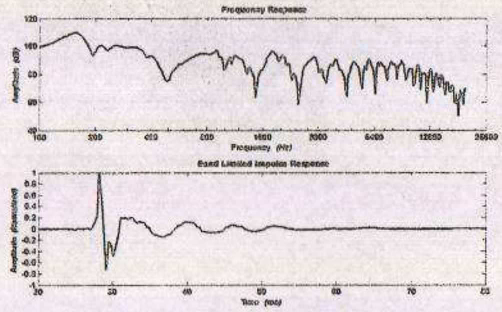

Figure 6 Frequency and Impulse Response of Two parallel MSL4s 45 degrees off axis

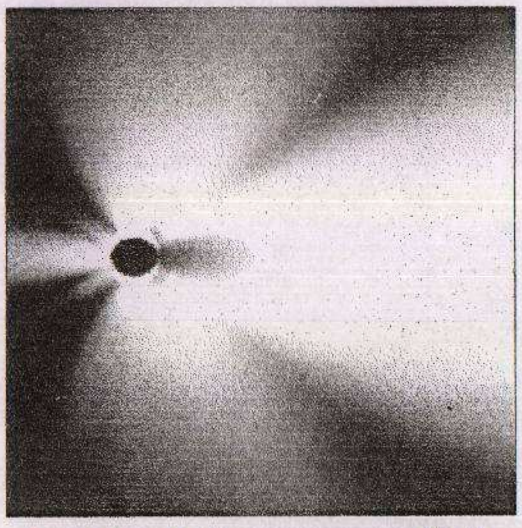

Figure 7 Two MSL4s tight packed at 500 Hz

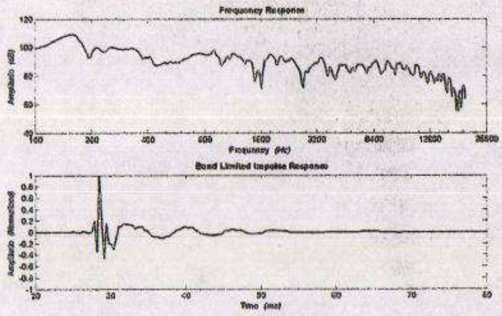

Figure 8 Frequency and Impulse Response of Two MSL4s 45 degrees off axis.

4. A PRACTICAL EXAMPLE

To demonstrate how this might be used in practice, we will now work through the theatre coverage example from the Meyer Sound Design Reference [5]. Our goal is to create a theatre sound system using MSL4 for left and right mains, UPA2P as an inside fill, and UPM1P as front fills which can reach an average SPL of 95dB everywhere in the room.

4.1 ALIGNING AN INSIDE FILL

Figure 9 shows the coverage from the mains and the inside fill, by placing a microphone in the back centre of the room we measure an average SPL of96.3dB. Because these are two different models of speaker, they will not necessarily have the same phase in the seam where their coverage patterns overlap. Figure 10 shows the frequency and impulse response of a microphone placed in the spatial null. Note the large notch at 500Hz, and the fact that there are two impulses separated by about 2.4 milliseconds (this delay can be measured more exactly by doing the impulse response of one speaker and then the other). Figure 11 shows the sound field when the inside fill is properly time aligned by delaying it 2.4 milliseconds. This steers the spatial null into the area that will be covered by the front fills, and greatly reduces the notch in the frequency response.

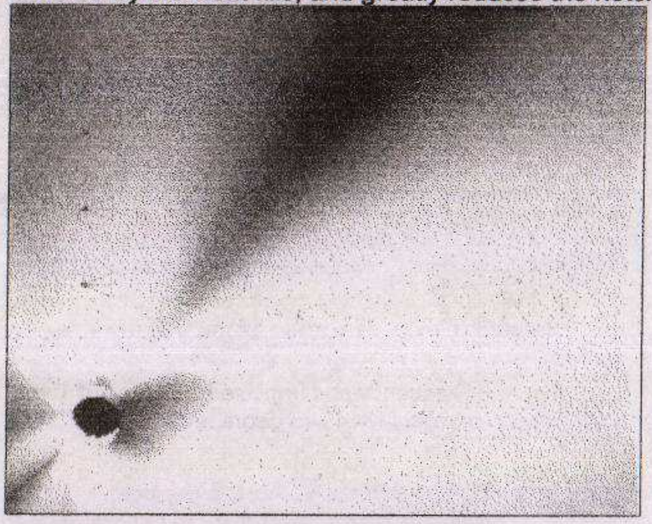

Figure 9 Mains and inside fill of an example Theatre system

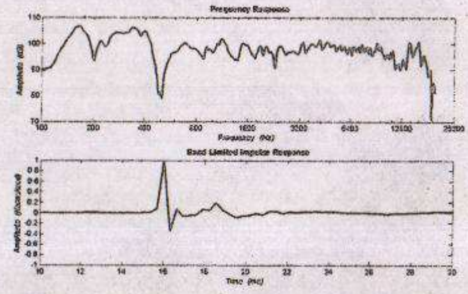

Figure 10 Frequency and Impulse Response of the mains and inside fill in the spatial null

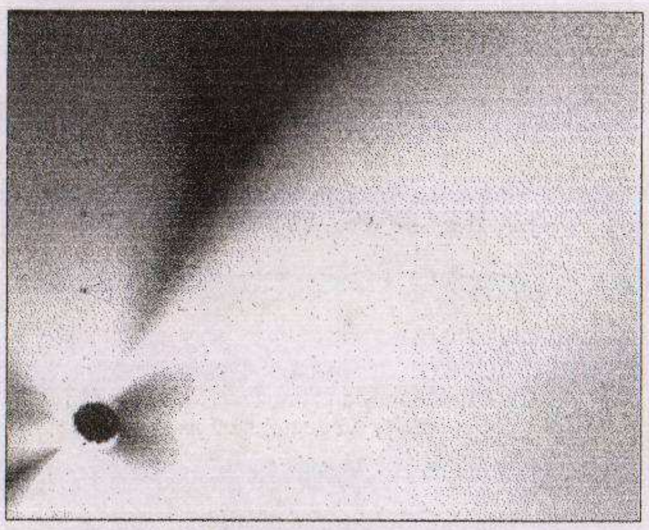

Figure 11 Time-aligned mains and inside fill

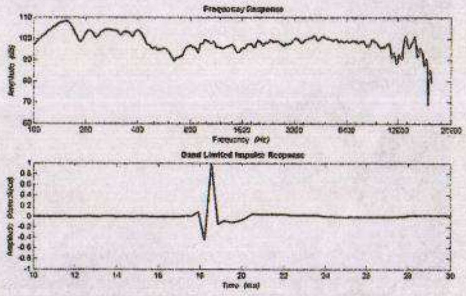

Figure 12 Frequency and Impulse Response of time-aligned mains and inside fill.

4.2 FRONT FILL

Figure 13 shows the sound field from a row of front fills. Note that, by default, MAPP automatically normalizes sound field plots to the loudest point in the field, and has a fixed dynamic range. One must place a virtual microphone in the room in order to see the total SPL at that point. Doing so we see that we reach 99.54dB SPL at 10 feet from the front fills, which is close enough that the speakers are not interacting with each other. Beyond that distance the speakers will interact with each other and combing, but this is not a concern because, as shown in Figure 14, this area will be overwhelmed with sound from the main system.

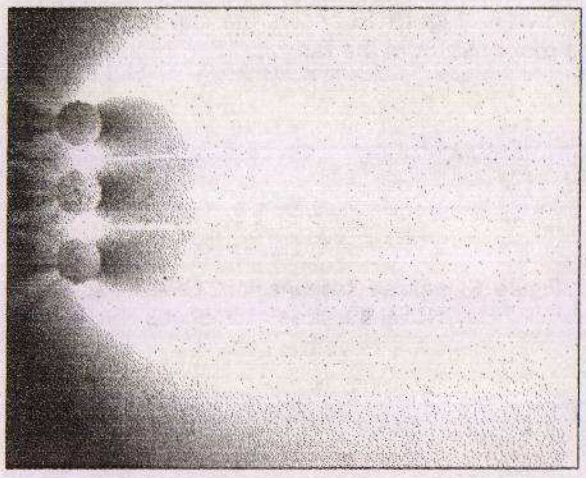

Figure 13 Front Fills only

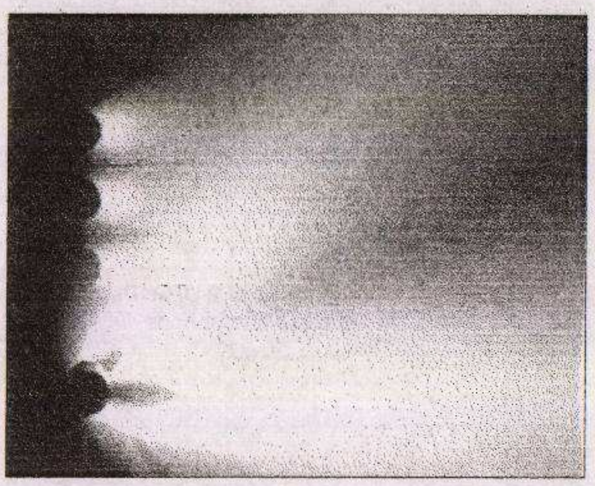

Figure 14 Mains, Inside Fill, and Front Fill all together.

5. WALLS

For any hall less than about 50 feet across there is an area in the front and centre of the audience where the left and right reflections arrive later than 50 milliseconds, and thus decrease clarity (C50). By selecting a loudspeaker system whose off axis response is significantly attenuated over the front part of the side walls, one can increase the clarity in the front centre portion of the room, while still maintaining strong side wall reflections to audience members in the back of the room.

Figure 15 shows the sound field from two UPA1Ps splayed 85 degrees with walls that are 80 feet apart. In this example only the reflection from the right wall is modelled. Figure 17 shows the impulse response to the near microphone. The reflection from the right wall appears 60 milliseconds after the direct arrival, and thus will decrease C50. Figure 19 shows the impulse response of the far microphone. In this case the reflection appears within 40 milliseconds of the direct arrival and thus will contribute to C50.

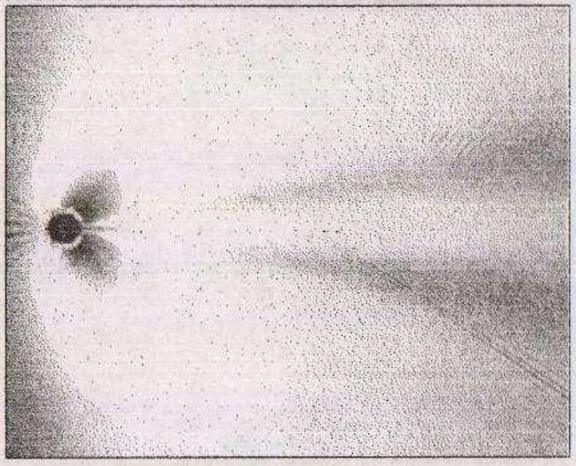

Figure 15 two UPA1Ps splayed 85° at 2 kHz

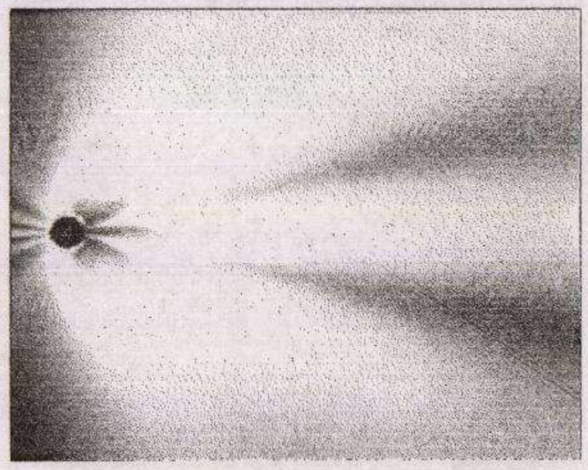

Figure 16 two UPA2Ps splayed 45° at 2 kHz

By using more directional loudspeakers with a smaller splay angle the reflection that arrives at the near mic can be greatly attenuated while not significantly altering the reflection that arrives at the far mic. Figure 16 shows the sound field from two UPA2Ps splayed at an angle of 45 degrees, at

2000Hz. Note that the audience area is still covered well. Figure 18 shows the impulse response from the near mic, and Figure 20 shows the impulse response from the far mic.

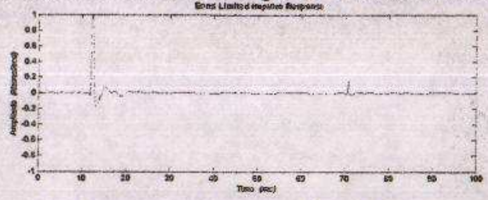

Figure 17 Impulse Response of 2 UPA1Ps splayed 85° at the close microphone

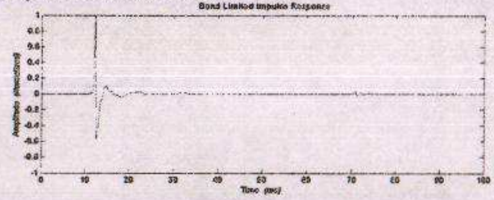

Figure 18 Impulse Response of 2 UPA2Ps splayed 45° at the close microphone

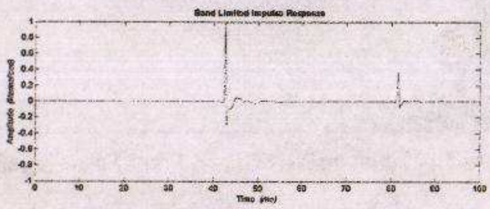

Figure 19 Impulse Response of 2 UPA1Ps splayed 85° at the far microphone

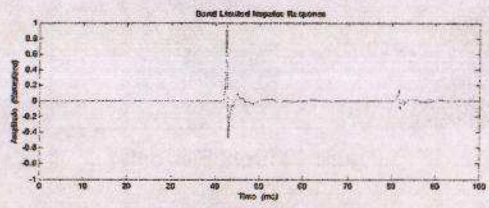

Figure 20 Impulse Response of 2 UPA2Ps splayed 45° at the far microphone

6. LINE ARRAYS

Though MAPP uses the far field model to find the pressure from an individual loudspeaker, it does accurately compute the near field of arrays of speakers. The exact pressure from a line array of omni directional speakers (no near field approximation) is:

$$p(r,\theta,t) = j\frac{\rho_0 c U_0 ka}{2}\sum_{n=1}^{N}\frac{1}{r_n}e^{j(\omega t - kr_n)}$$

Figure 21 shows the on axis pressure of an array of 16 omni directional speakers at 100Hz spaced 0.5 meters apart computed from the equation above in Matlab, and modelled in MAPP. Figure 22 shows the same configuration at 250Hz. Similar results were presented last year at IOA-RS16 by Duran Audio [6].

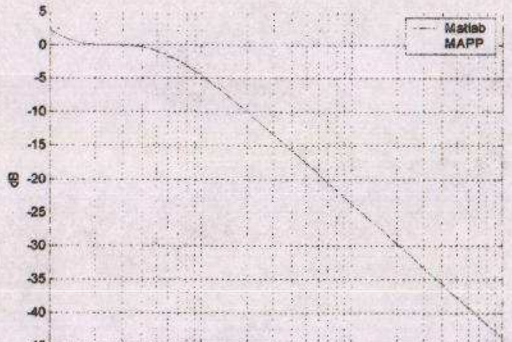

Figure 21 On Axis pressure of a Line Array 16 Omnis spaced 0.5 meters at 100Hz

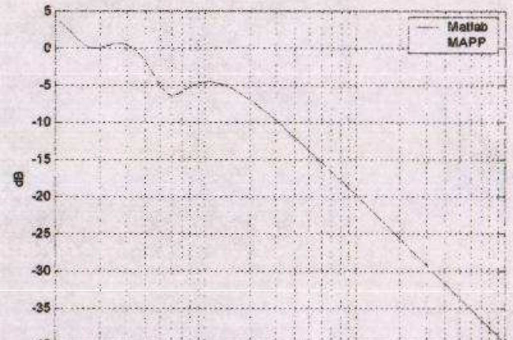

Figure 22 On Axis pressure of a Line Array 16 Omnis spaced 0.5 meters at 250Hz

Much has been made of the fact that antenna theory predicts 3dB loss per distance doubling, instead of the 6dB that we've come to expect from most loudspeakers. Unfortunately, antenna theory makes several assumptions that are generally not valid for acoustics. It is assumed that the array is a continuous line (or that the spacing between the sources is small compared to a wavelength) which is infinitely long (or many wavelengths long). One can see from the above figures that there is in fact less than 6dB loss per distance doubling in the near field, but that this breaks down quite easily as wavelength approaches the dimensions of the array.

This is not to say that one cannot create a useful line array of speakers. If one properly matches the horn directivity of the high drivers to the line array directivity of the low drivers, a full bandwidth, highly directional array can be created. Figure 23 shows the sound field of an array of 16 M3Ds at 2kHz.

For the purposes of comparing available power the all of the figures in this section will not be normalized, but shown with an absolute 1/3 octave SPL palette shown in Figure 24.

The Product Theorem also has important implications for line arrays. It has long been known that the directivity of a line array of omnis can be steered using delay [7]. Real world speakers are generally nowhere near omni in the high frequencies, and the resulting directivity is the multiplication of the line array directivity and the horn directivity. Using delay to steer an array outside of the main lobe of the horn results in grating lobes and enormous losses of available power. Figure 25 shows the sound field at 2 kHz of an array of omni sources spaced 0.5 meters apart and delay steered -30 degrees (angles measured clockwise). Notice the grating lobes spaced 20 degrees apart centred on the main lobe. Figure 26 shows the directivity from a single M3D cabinet at 2kHz, note the null at +/-30 degrees where we intend to steer the beam of the array. The sound field we expect from an array of M3Ds steered to -30 degrees will be the directivity of a single M3D multiplied by the directivity of the line array. Figure 27 shows that this is so. The null from the M3D has cancelled out the lobe that we tried to steer to -30 degrees while the on axis lobe of the M3D has amplified the grating lobe at –10 degrees.

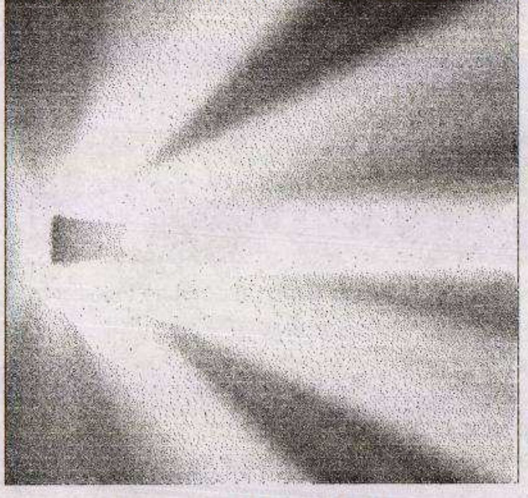

Figure 23 16 M3Ds at 2 kHz

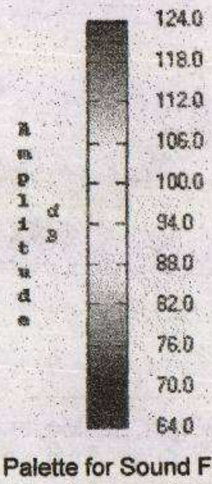

Figure 24 SPL Palette for Sound Field Plots in the Line Array section

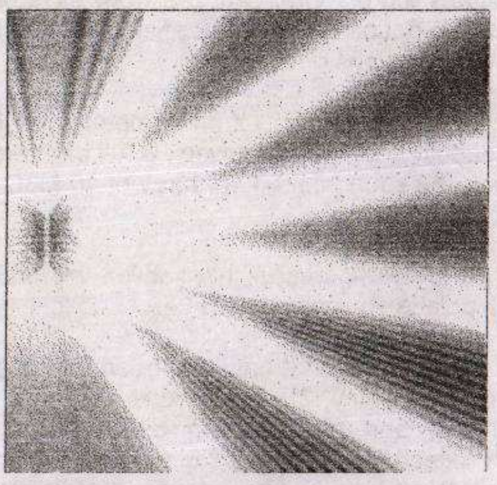

Figure 25 16 Omnis steered –308 at 2 kHz

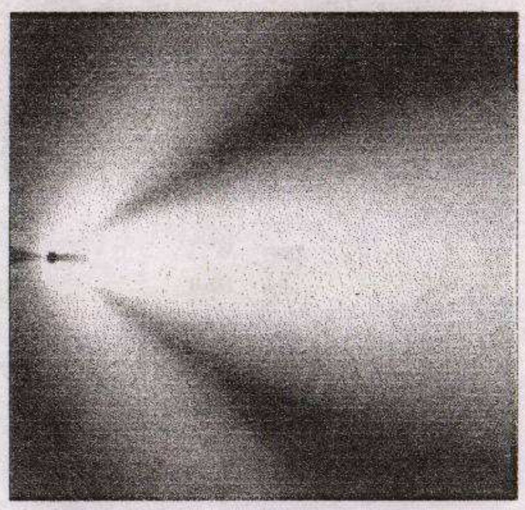

Figure 26 A single M3D cabinet at 2 kHz.

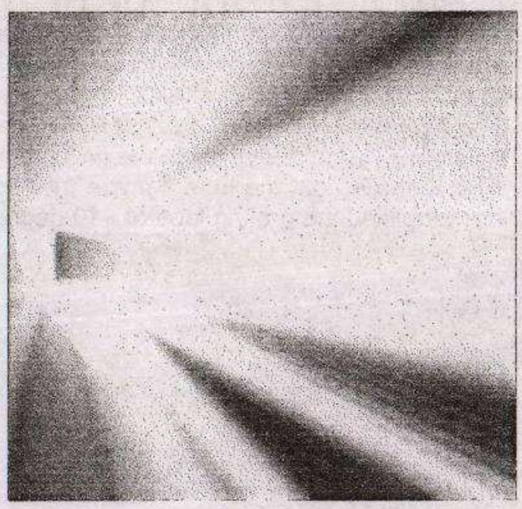

Figure 27 16 M3Ds at 2 kHz attempting to steer to –30°.

1 J. Baird, P. Meyer, J. Meyer, "Far-Field Loudspeaker Interaction: Accuracy in Theory and Practice", in 110th Audio Engineering Society Convention (Amsterdam) Preprint no. 5309, 2001. http://www.aes.org

2 J. Baird, P. Meyer, "The Analysis, Interaction, and Measurement of Loudspeaker Far-Field Polar Patterns", in 106th Audio Engineering Society Convention (Munich) Preprint no. 4949(Q4), 1999. http://www.aes.org

3 P. Meyer, J. Meyer, "Multi Acoustic Prediction Program (MAPP ™) Recent Results", Proceedings of the IOA Vol. 22, Pt 6, 2000.

4 L. Kinsler, et al., "Fundamentals of Acoustics", 3rd ed., John Wiley and Sons, 1982.

5 B. McCarthy, "Meyer Sound Design Reference For Sound Reinforcement", Meyer Sound Laboratories, 1997.

6 G. van Beuningen, E. Start, "Optimising Directivity Properties of DSP Controlled Loudspeaker Arrays", Proceedings of the IOA Vol. 22, Pt 6, 2000.

7 H. Olson, "Acoustical Engineering", Professional Audio Journals, 1991.