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Design of a Wideband, Constant Beamwidth, Array Microphone for Use in the Near Field

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Directional microphones have long been proposed for the removal of room reverberation. An array microphone would seem ideal for this purpose, since theoretically it can be aimed anywhere within the room. However, microphone pattern beamwidth is related to wavelength and aperture size. For a fixed-size aperture, as wavelength goes down so does beamwidth. The change in beamwidth over a decade change in wavelength would seem to be unacceptable for this application. We discuss the design of a constant beamwidth array microphone for the frequency range 300 to 3000 Hz. Because the microphone-to-talker distance is assumed to be about 3 ft while the array has a 9-ft aperture, the microphone is optimized for near field. We also discuss the use of a nonlinear optimization program for choosing the array parameters.

I. INTRODUCTION

Directional microphones have been proposed for the removal of room reverberation on the assumption that a properly aimed microphone would pick up the direct path speech energy and reject the reverberant energy. Indeed, highly directive microphones are often employed in "press conference" situations to pick up questions from an auditorium. An array microphone would seem to be an ideal replacement for the "directional" mike in such a situation since it would offer a quick automatic aiming capability and, if so designed, either multiple-speaker monitoring or the ability to correlate multiple reverberant paths from a single speaker.

Conventional array microphones, however, suffer from a number of drawbacks which must be considered. If we consider speech in the frequency range 300 to 3000 Hz, then the microphone designer must consider the decade change in frequency (and wavelength). In particular, the beamwidth of the radiation pattern is related to wavelength and microphone aperture size (microphone dimensions). As wavelength goes down, so does beamwidth. The change in beamwidth over a decade change in wavelength would seem to be unacceptable for this application.

If we would like to employ our microphone for small conference room dereverberation, then we must also consider the near-field/far-field transition region of the microphone. Note that at 300 Hz the wavelength of sound is about 4 ft. A microphone with a two-wavelength aperture at this frequency is therefore 8 ft wide. However, most rooms have an 8 to 10 ft ceiling. As a result, speakers in such a situation are between 1 wavelength (standing) and 2 wavelengths (seated) away from a ceiling-mounted microphone. We consider a ceiling-mounted mike because of symmetry with respect to aiming the microphone and because there is always a direct path between the microphone and talker. This, of course, does not mean that it is an optimal location. We note, however, that TV studios use overhead boom mikes for sound pickup.

Finally, most array design texts concentrate on single-frequency (or narrowband) designs. Because of the near-field constraints, however, the design equations are highly nonlinear and thus difficult (or impossible) to solve analytically. As a result, some type of optimization approach must be taken to select the proper design parameters, such as element position and gain, for the microphone. We discuss this problem later. First, we discuss the microphone design problem in more detail.

II. MICROPHONE DESIGN

Microphone design, that is, the selection of position and gain for each element of an array of microphones, is *not* a simple task. Most texts on array design, e.g., Ref. 1, start out with a uniformly illuminated aperture (opening) and develop the radiation pattern from basic physical constraints. An array microphone then corresponds to a spatially (uniform) sampled aperture. The analogy to sampling theory is usually drawn at this point since, in the *far* field, the radiation pattern is the (discrete) Fourier transform of array illumination (element gains). Since we are constrained to working in the near-field/far-field transition region, the exact response r, at observation point o and wavelength λ, for an *array* of N omnidirectional microphones is:

$$r(o, \lambda) = \sum_{k=1}^{N} \frac{A(k)}{d(k)} e^{-j(2\pi/\lambda)(d(k)+\phi(k))}$$

where $d(k)$ is the distance from the kth microphone to the observation point o.

$A(k)$ is the gain of the kth microphone

$\phi(k)$ is the phase shift (delay) of the kth microphone
λ is the wavelength.

Using this equation, we have calculated the response for the array consisting of five elements uniformly spaced over the range $-\lambda_0$ to $+\lambda_0$ depicted in Fig. 1. The element gains have a triangular weighting and no phase shift. The polar response, at a distance of $10\lambda_0$, of this array is plotted on a log scale in Fig. 2 for the range $-90°$ to $+90°$. (This corresponds to measuring the response along the circle BAC.) Curve (a) is the response at $\lambda = \lambda_0$; curve (b) at $\lambda = 0.75\lambda_0$ shows that the main lobe response has narrowed appreciably. Curve (c) at $\lambda = 0.5\lambda_0$ shows a marked increase in the sidelobes. Curve (d) at $\lambda = 0.4\lambda_0$ shows spatial foldover since the element spacing is only $0.5\lambda_0$. Finally, curve (e) at $\lambda = 0.3\lambda_0$ shows severe foldover.

Figure 3 shows the polar response of this same array as a function of distance. Curves (a), (b), and (c) are for distances of $10\lambda_0$, $5\lambda_0$, and $3\lambda_0$, respectively. The flattening of the array response is obvious. Curves (d) and (e) for distances of $2\lambda_0$ and $1.5\lambda_0$ show even more severe flattening. Curve (f) for a distance of $1.1\lambda_0$ illustrates a serious problem associated with plotting the near-field response of a microphone array. Figure 1 illustrates the array configuration and two distances, $d = 10\lambda_0$ and $d = 1.1\lambda_0$, for which responses are being measured. For the circle at $d = 10\lambda_0$ and $\theta = 0°$ (point A), all elements are at least $10\lambda_0$ away from the observation, while at $\theta = \pm 90°$ (points B, C) the array elements are between $9\lambda_0$ and $11\lambda_0$ away from the observation point (a 10-percent change). However, for $d = 1.1\lambda_0$ at $\theta = 0°$ (point D), all elements are at least $1.1\lambda_0$ away while, at $\theta = \pm 90°$ (points E, F), the array elements are between $0.1\lambda_0$ and $2.1\lambda_0$ away (a 90-percent change).

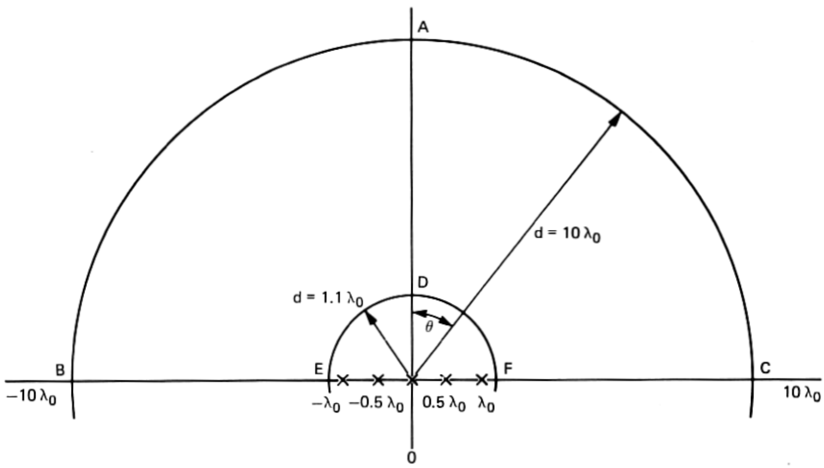

Fig. 1—Array response observation paths.

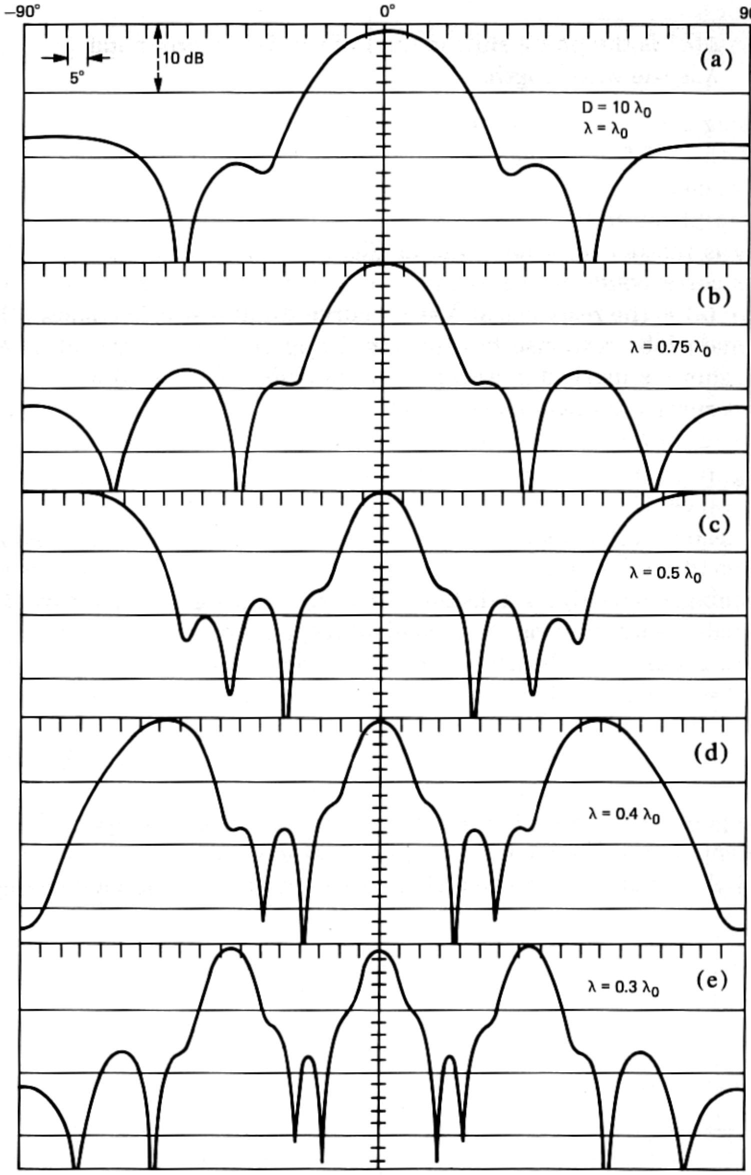

Fig. 2—Response variation with λ.

In essence, what results is a proximity effect. At $d = 1.0\lambda_0$ and $\pm 90°$, the observation point and the microphone locations coincide. The resulting response plot is useless because the A/d term in the response equation becomes infinite. Because we are most interested in room responses, we have modified the subsequent response plots to be the microphone response in a plane parallel to the microphone. As a result, the response at ±90° falls to zero since this corresponds to an infinite distance from the array.

III. MICROPHONE DIRECTIVITY

Microphone directivity index (DI) may be defined as the ratio of the maximum microphone sensitivity at the peak of the main lobe to the

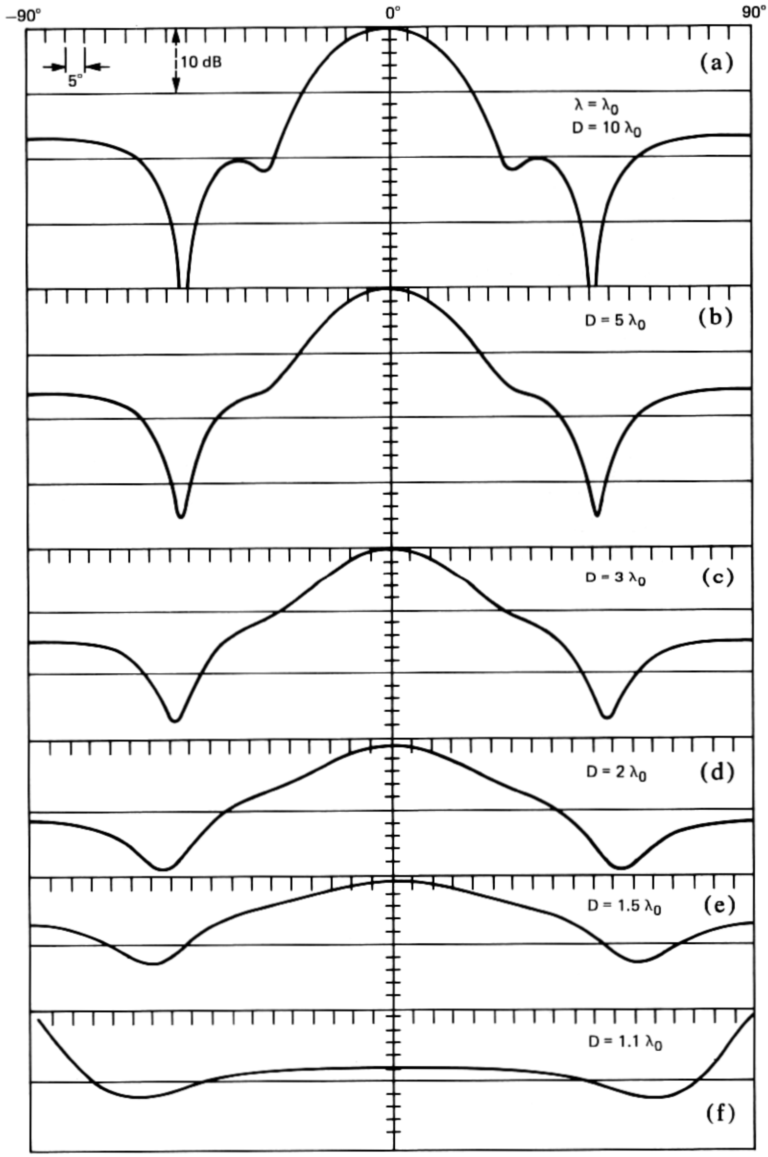

Fig. 3—Response variation with D.

sensitivity of an isotropic (uniform response in all directions) microphone with the same average sensitivity.[2] However, because we are dealing with a wideband microphone, directivity is also a function of frequency. This introduces a problem since averaging implies uniform sampling across the band which unfairly weights one band edge over the other. For comparison purposes, we have calculated directivity by averaging over both frequency and wavelength for both polar and planar response functions. The average directivity indices we have obtained range from 8.8 to 5.1. For comparison, a second-order gradient microphone has a directivity index of 7, a cardioid of 4.8, and a limacon of 6.0.[2]

IV. NONLINEAR OPTIMIZATION GOAL PROGRAMMING

As discussed previously, microphone design must be done by a nonlinear optimization program. However, in terms of nonlinear optimization, microphone design poses a problem. Typically, we begin with a desired response, for example, the ideal specifications of Fig. 4 and then try to build a microphone which has that response "*or better.*" The "or better" rules out any kind of least-squares optimization technique since responses which are better (below the sidelobe requirement, for example) are counted in as part of the error term. Ignizio[3] has developed a technique which he refers to as "goal programming," which deals with this problem. In this case, when the current solution meets or exceeds the goal, the penalty function is zero. If the solution does not meet the goal, then the penalty is nonzero. The optimization program then seeks to minimize the penalty function(s) by varing the

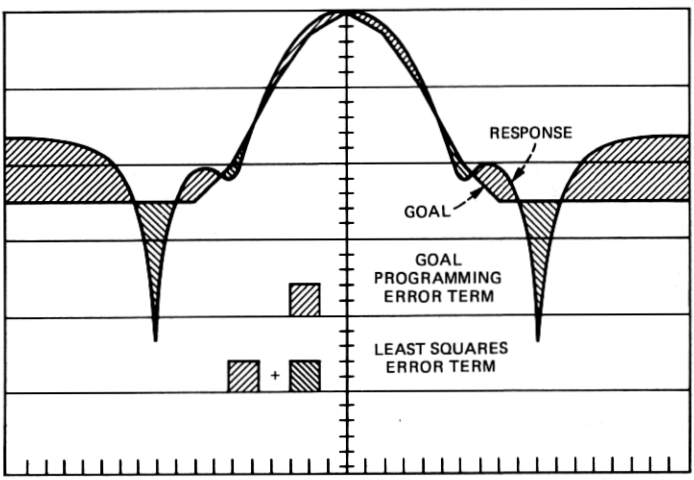

Fig. 4—Goal programming error comparison.

microphone parameters (called decision variables). For our microphone design problem, the 180-degree response field of the microphone is evaluated at 19 separate points. A penalty function is written for each point. This allows different points of the response to be weighted separately and to be constrained in different ways. The optimization program then seeks to minimize an achievement function which is a combination of each of the penalty functions.

Unlike most nonlinear optimization, goal programming does not need gradients or derivatives, and it does not need an initial valid solution. Unfortunately, like most nonlinear optimization techniques, there is no guarantee of global maximum. A version of Ignizio's optimization program was programmed for our Data General Eclipse Computer and used to optimize the microphone designs presented here. Our experience with this algorithm has been very good. It has enough flexibility with respect to the optimization goal, variable weighting, and penalty function construction to make it practical to use. Running time is linearly proportional to the number of variables, as is storage space required. Optimization of a microphone involving 20 decision variables with 57 goal points and 72 terms in the achievement function typically takes 10 to 15 minutes.

As with most optimization programs, one must carefully pose the problem. The optimization program will take advantage of any "loopholes" left in the problem statement, usually with disastrous results. When considering designs for a single frequency, the optimized result was often a superdirective design (a design with close element spacing and phase alternations between elements). Superdirectivity, unfortunately, is an extremely narrowband phenomenon and is of little interest for the intended application.

V. MICROPHONE DESIGN RESULTS

Because of the decade range of frequencies involved, an array properly sampled at the highest frequency is grossly (10:1) oversampled at the lowest frequency. In addition, the number of elements implied by such oversampling is excessive, impractical, and also unnecessary, as we shall see. As a result, we must now consider a nonuniformly sampled array (also called a thinned or random array).

The technique which we have developed to handle this problem makes use of superposition. Essentially, three arrays are designed, one for midband and one for each band edge as in Fig. 5. These three arrays are then combined (superimposed), with suitable filtering, to give a single array which covers the full frequency band. This approach is essentially similar to that taken in Refs. 4 and 5, except that the array is constrained to operate in the near field. Other approaches which are suitable only in the far field are detailed in Refs. 6 and 7.

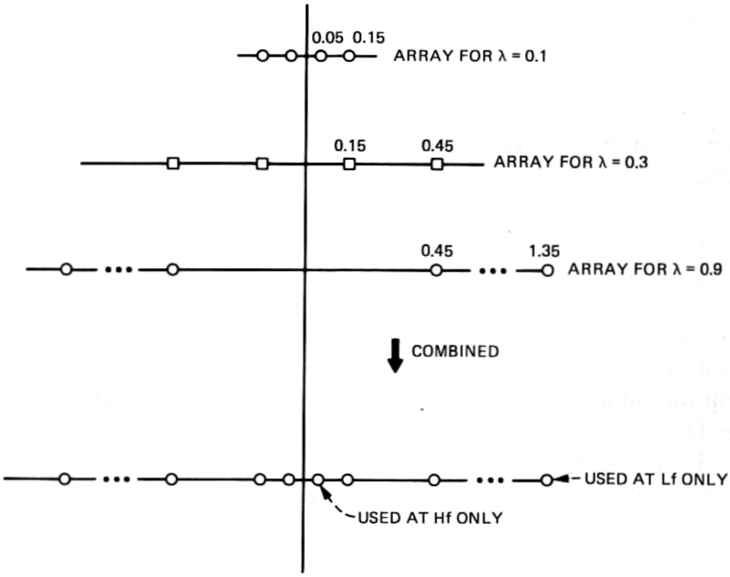

Fig. 5—Wideband array prototype.

Our first approach was to attempt to design a microphone optimized at both ends of the frequency band. However, without any frequency-dependent components in the microphone, the result was optimized for midband with very poor response at band edges. It became obvious that some frequency-dependent element would have to be introduced into the microphone design to allow the microphone *aperture* to vary with frequency. Since each microphone will need a pre-amplifier of some sort, we decided to introduce some frequency shaping into the amplifier design. A second order 12dB/octave filter was incorporated into the microphone preamplifier and the goal programming model was modified to set the cutoff frequencies accordingly. When the amplifier phase shift is ignored, the results of Fig. 6 are obtained. This design was optimized using a polar response function at a distance of $3\lambda_0$. For these designs, λ_0 corresponds to an f_0 of 300 Hz. Here the filter cutoff frequencies correspond to approximately what would be expected for the given frequency band. This figure illustrates the microphone response at the band edges and the average responses for both frequency and wavelength averaging. The set of plots on the left of the figure are polar responses, while those on the right are planar. Figure 7 illustrates the response of a microphone array designed using polar response at a distance of $1\lambda_0$. Amplifier phase shift, however, has been included in this design and the cutoff frequencies have shifted to compensate for this. Figure 8 illustrates the responses of a microphone optimized using planar response instead of polar response.

VI. DISCUSSION OF ARRAY RESULTS

Initially, a steerable array was proposed for removing room reverberation. However, the directivity factors and beamwidths attainable with the arrays we have been able to design indicate that perhaps these arrays might better be applied as fixed microphones. In this case, the directivity of the microphone would allow it to be located over a conference table, while rejecting (or reducing) room noise and reverberation from the balance of the room. An additional factor which

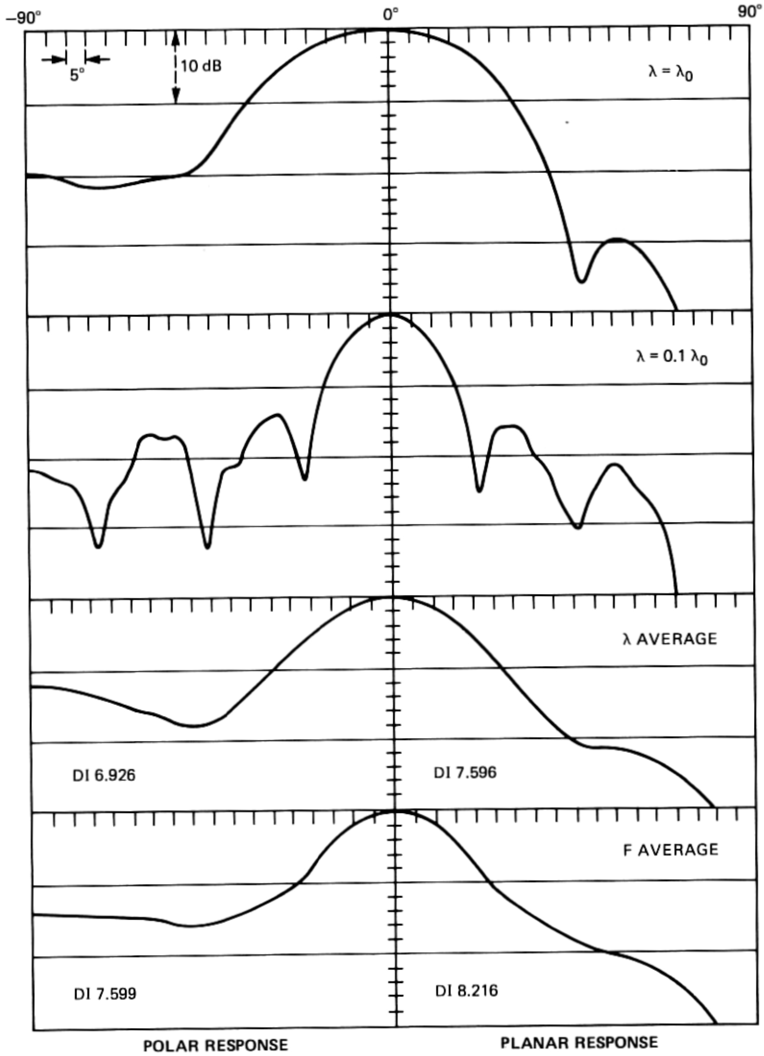

Fig. 6—Array response ignoring phase shift.

favors a fixed microphone is the cost and complexity of the hardware required to steer the array.

The 8-element linear arrays presented here imply approximately 64 elements in a two-dimensional array. A steerable array would require a phase shifter (variable delay) for each element of the array. Current CCD technologies allow the construction of fairly cheap analog delay lines (several commercial grade circuits are already available). A multibeam microphone, however, would require multiple tapped delay

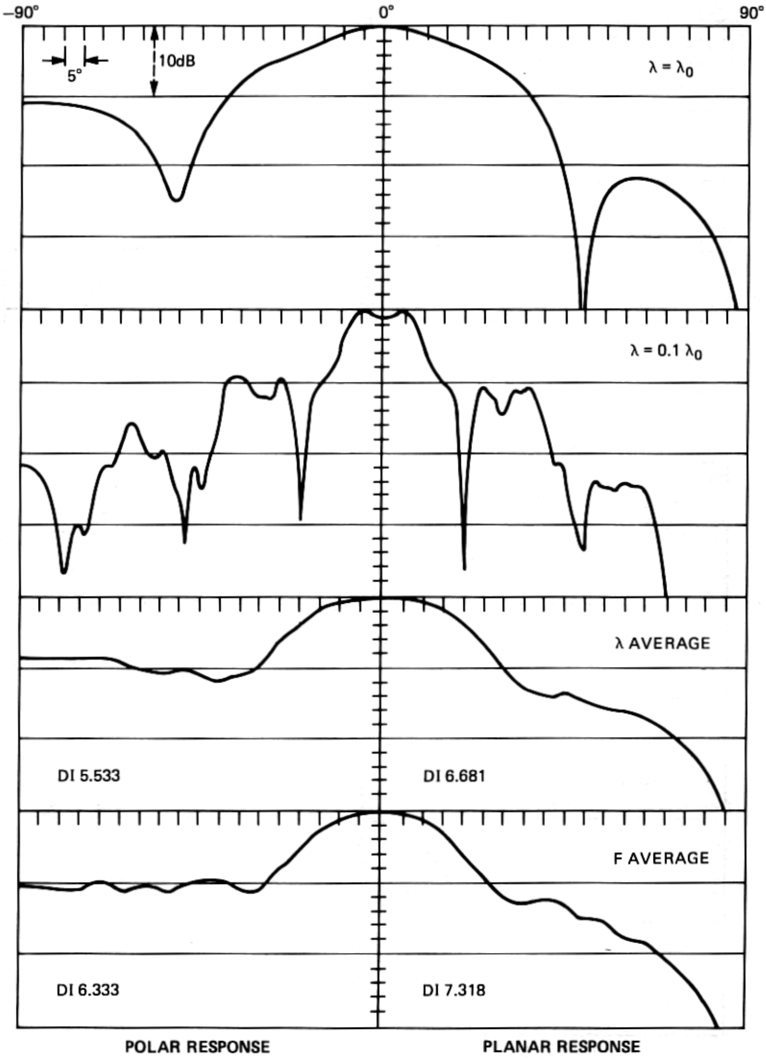

Fig. 7—Array response polar optimization.

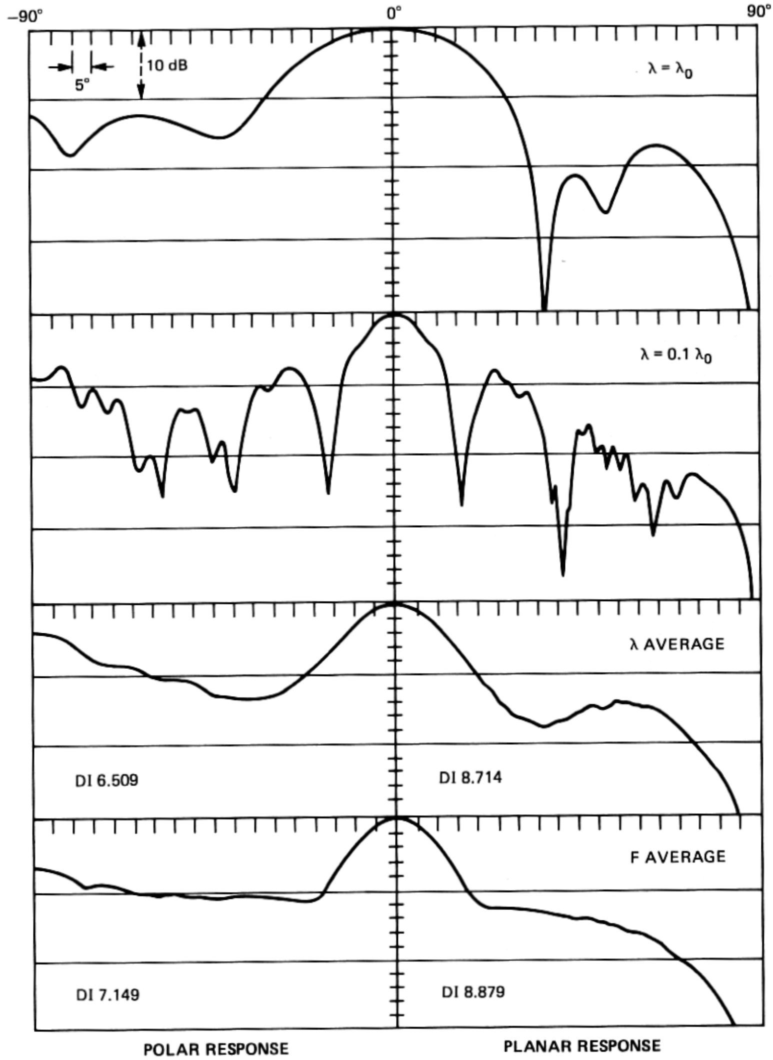

Fig. 8—Array response planar optimization.

lines which are difficult to realize in analog form. The summation and switching circuits would also be fairly complex. A multitapped digital delay line made from an A/D converter, RAM storage, and D/A converter could probably be implemented for approximately $100/channel (prototype quantities). A typical array then begins to cost almost $5000, which is enough to make it unattractive.

Since the microphone is not *significantly* directive, simple geometric constraints indicate that a microphone with perhaps four to nine beam positions might be fabricated using analog techniques. One of these beam positions could then be chosen (dynamically perhaps) during a conference as the active speaker(s) move about in a conference room.

VII. CONCLUSION

We have shown that it is possible to design a wideband, approximately constant beamwidth array microphone for use in the near field. We conjecture that using more elements in the array would improve the beamwidth/sidelobe ratio and overall response. We have yet to attempt to steer this array design. Because of the poor sidelobe response, we do not see much prospect of pencil point focusing of this type of microphone for small room dereverberation unless, as mentioned previously, the number of elements is increased to improve response. We note, however, that the array steering mechanism then increases in complexity and may become prohibitively expensive. This work, and that cited in Refs. 4 to 7, however, indicate that an array microphone might indeed be suitable for large room (auditorium) dereverberation.

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