Superdirective arrays for hearing aids

Article *in* The Journal of the Acoustical Society of America · November 1993
DOI: 10.1121/1.407515 · Source: PubMed

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(Received 28 December 1992; revised 4 June 1993; accepted 8 June 1993)

Microphone arrays are the most effective of the techniques that have been proposed for improving speech intelligibility in noise for the hearing impaired. However, classical delay-and-sum beamforming provides very small amounts of array gain at low frequencies, while adaptive array processing has been shown to cancel the desired signal in the presence of strong room reflections. Superdirective arrays offer a heretofore overlooked solution in which optimal performance can be obtained for a stationary random noise field, but where the desired signal will not be canceled. A short constrained superdirective array suitable for hearing-aid applications is proposed in this paper, and its theoretical performance is evaluated.

PACS numbers: 43.88.Hz, 43.66.Ts

INTRODUCTION

In this paper, a short constrained superdirective microphone array is proposed for hearing-aid applications, and its effectiveness is evaluated using a computer simulation. While superdirectivity is an established array-design technique, it has not been previously applied to the problem of a hearing-impaired individual listening in noise. In this paper, it is shown that a short constrained superdirective array is an effective design approach for use by the hearing impaired.

A short microphone array is attractive since it represents one of the few techniques, among the many that have been proposed, that has actually improved speech intelligibility in noise. The improvement in signal-to-noise ratio (SNR) for a 10-cm-long array using five uniformly spaced directional microphones with delay-and-sum beamforming is 5–12 dB, with the greatest improvement occurring at high frequencies (Soede, 1990). Such an array can be built into an eyeglass frame, and the performance of both end-fire and broadside arrays does not appear to be affected by the presence of the head to any great extent. The directional arrays used by Soede improved the speech reception threshold (SRT) in noise by 7 dB in a diffuse noise field, so the improvement in SNR is directly related to a comparable improvement in speech intelligibility in noise.

The performance offered by delay-and-sum beamforming can be bettered by using superdirective array processing (Cheng, 1971; Newman *et al.*, 1978; Cox *et al.*, 1986) to yield the optimum improvement in SNR for a random stationary noise field. A common assumption, and the one that will be used in this paper, is that the ambient noise field is spherically isotropic. In addition, it is assumed that there is uncorrelated random white noise at each microphone. This additional noise source leads to the sensitivity constraint of Cox *et al.* (1986) in designing the superdirective array weights, which gives a reduction in the sensitivity to microphone position errors, wave-front perturbations, and sensor internal noise, in exchange for a reduction in the superdirective array gain. Given the correspondence between improvements in SNR and improvements in SRT shown for the cited adaptive systems and array processing, one would also anticipate a similar correspondence for a superdirective array.

A superdirective array may also work better than a multimicrophone adaptive array. Adaptive algorithms commonly use a minimum mean-squared error criterion (Monzingo and Miller, 1980), with the consequence that the correlation matrix used in computing the sensor weights includes the signal as well as the noise. For this type of processing, a perturbed signal wave front, as occurs when a strong reflection is present or when there are random displacements in the microphone locations, can lead to cancellation of the desired signal (McDonough, 1972; Cox, 1973). Since the fixed superdirective array weights are based on the correlation matrix of the assumed noise field alone, signal cancellation would not occur for small perturbations of the signal wave front even though the array output power may vary (Cox, 1973). If immunity to perturbed wave fronts cannot be designed into an adaptive algorithm, then the more robust behavior possible with a superdirective array having constrained sensitivity (Newman *et al.*, 1978; Cox *et al.*, 1986) may be preferable for a practical system.

I. OPTIMUM ARRAY PROCESSING

The array geometry of interest is an end-fire linear array having uniformly spaced omnidirectional sensors. The noise correlation matrix is given by the ambient noise combined with the self-noise of the microphones:

$$\mathbf{R}=\sigma_a^2[\mathbf{Q}+(\sigma_n^2/\sigma_a^2)\mathbf{I}], \tag{1}$$

where $\mathbf{Q}$ is the ambient noise correlation matrix, σ_a^2 is the ambient noise power, $\mathbf{I}$ is the identity matrix, and σ_n^2 is the microphone self-noise power. The weight vector that optimizes the SNR while maintaining unity gain in the desired look direction is given by (Monzingo and Miller, 1980)

$$\mathbf{w}=\mathbf{R}^{-1}\mathbf{v}/\mathbf{v}^*\mathbf{R}^{-1}\mathbf{v}, \tag{2}$$

where $\mathbf{v}$ is the steering vector for the look direction, $\mathbf{R}$ is the correlation matrix given by Eq. (1), and the asterisk

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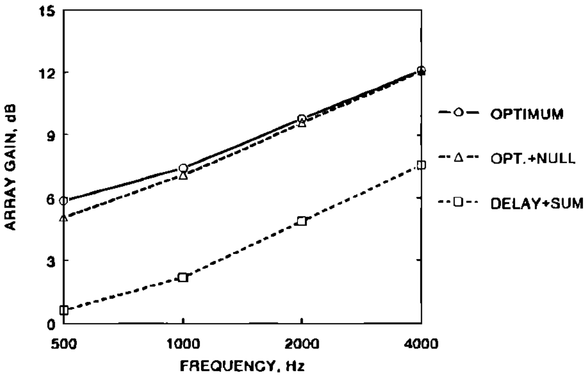

FIG. 1. Array gain as a function of frequency for optimum processing with white noise gain constrained to 0 dB, optimum processing with the constraints of white noise set to 0 dB and a null at 180 deg, and delay-and-sum beamforming.

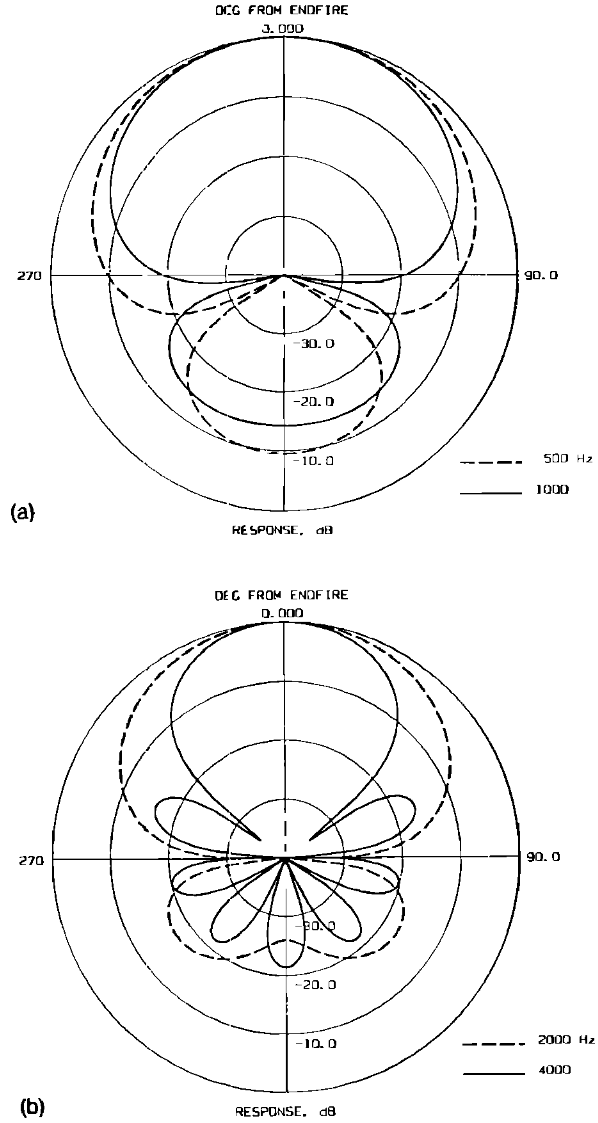

FIG. 2. Array directional response patterns at (a) 500 and 1000 Hz and (b) 2000 and 4000 Hz for the optimum weights constrained to have a white noise gain of 0 dB.

denotes conjugate transpose. Since the correlation matrix contains both ambient and self-noise terms, the solution is equivalent to superdirectivity with the sensitivity constraint of Cox *et al.* (1986) when the assumed sensor self-noise power is used as a design parameter. If $\sigma_n^2 \gg \sigma_a^2$, the optimum solution reduces to classical delay-and-sum beamforming, while the condition $\sigma_a^2 \gg \sigma_n^2$ results in unconstrained superdirectivity. Additional constraints can be introduced to add a null to the array directional pattern (Fenwick, 1971; Frost, 1972) if desired.

The array performance can be characterized by the array gain and the white noise gain (Cox *et al.*, 1987). The array gain indicates the improvement in SNR provided by the array for the ambient noise field. For a five-sensor array in a spherically isotropic random noise field, the maximum possible array gain is 14.0 dB. The white noise gain indicates the improvement in SNR provided by the array for the sensor self-noise. The inverse of the white noise gain indicates the sensitivity of the array response to perturbations in the signal wave front or the assumed microphone positions (McDonough, 1972; Newman *et al.*, 1978; Cox *et al.*, 1987). A usable hearing-aid array must provide an increase in array gain without undue sensitivity to reflections or uncertainties in the microphone positions. These objectives can be met by constraining the array to have a white noise gain of 0 dB. Unity white noise gain means that the microphone array output will have the same self-noise level as that of a single microphone, and the variance in the array output due to wave-front amplitude perturbations will also be the same as for a single microphone (McDonough, 1972).

II. SIMULATION RESULTS

The simulated microphone array is 10 cm long and consists of five uniformly spaced omnidirectional microphones used in an end-fire configuration. This array is short enough that it can fit along the side of an eyeglass frame or be part of a hand-held device the size of a pen or pencil. The array gain as a function of frequency, when the white noise gain is constrained to be 0 dB, is plotted in Fig. 1. The processing strategies represented are the optimum array weights given by Eq. (2) and the optimum weights having a response null positioned at 180 deg. The optimum weights were computed separately at each indicated frequency. Delay-and-sum beamforming is provided for purposes of comparison. The optimum array gain ranges from 5.9 dB at 500 Hz to 12.1 dB at 4000 Hz, while the delay-and-sum beamformer ranges from 0.6 dB of array gain at 500 Hz to 7.6 dB at 4000 Hz. The optimum weights therefore give an average of 5.0-dB-greater array gain than delay-and-sum beamforming. Adding the null at 180 deg reduces the optimum array gain by an average of 0.3 dB, so the penalty for this additional constraint is small.

The differences in array gains for the different array designs are also reflected in the array directional response patterns. These patterns are shown in Fig. 2 for the optimal weights constrained to have a white noise gain of 0 dB. Since the greatest gain occurs in the main lobe of the response pattern, the optimal weights will tend to reduce the width of the main lobe as much as possible in order to minimize the response to the ambient noise. The amount of

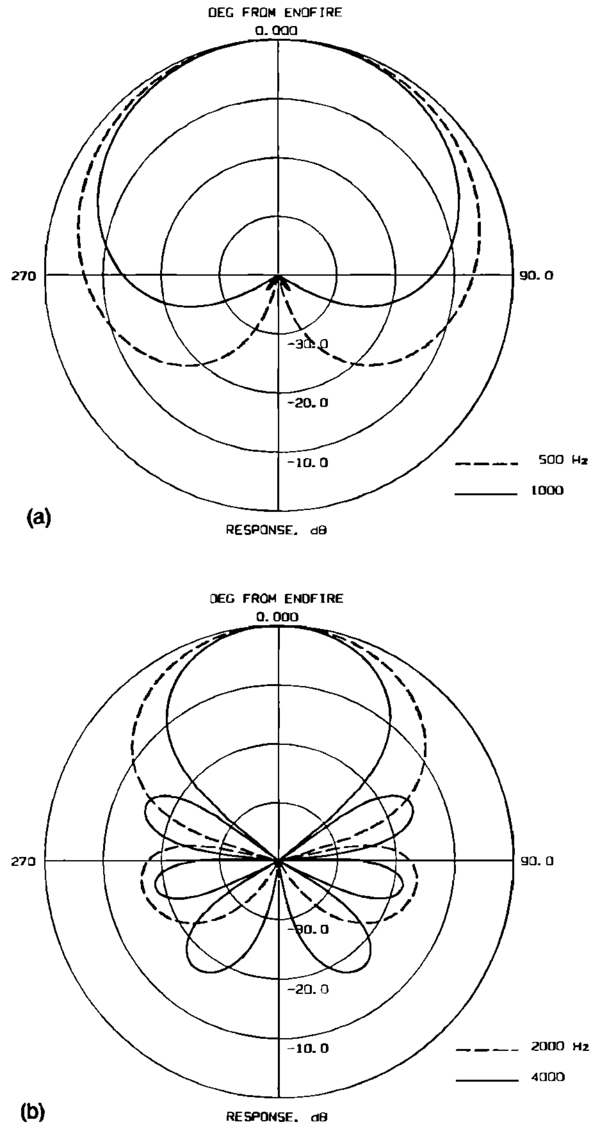

FIG. 3. Array directional response patterns as for Fig. 2 with the additional design constraint of a response null at 180 deg.

reduction in the width of the main lobe, however, is limited by the white noise constraint.

Introducing the additional design constraint of a null at 180 deg results in the directional response patterns of Fig. 3. The effectiveness of the constraint in reducing the response to sounds coming from the rear of the array is clearly evident in this simulation. The null constraint results in a directional pattern at 500 Hz that is similar to that of a cardioid microphone. At higher frequencies the influence of the null constraint on the shape of the main lobe is reduced. However, at all frequencies the main lobe beamwidths are somewhat wider than those of Fig. 2. Thus adding the null reduces the array response to sounds coming from behind the microphone, but at the cost of increasing the width of the main lobe of the response pattern and reducing the array gain by small amounts.

Another processing approach is to combine cardioid directional microphones into an array using delay-and-sum beamforming. This approach was taken by Soede (1990), who found array gains ranging from approximately 5 dB at 500 Hz to 12 dB at 4000 Hz for a 10-cm end-fire array consisting of five directional microphones. The array gain of the delay-and-sum array using cardioid microphones is about the same at all frequencies as that due to the optimized weights constrained to have a white noise gain of 0 dB.

Soede (1990) used a separation of 0.8 cm to create the cardioid responses in the hearing-aid microphones that were combined into his five-microphone array. The white noise gain for the cardioid response having a 0.8-cm sensor separation is about -20 dB at 500 Hz, and rises with a slope of 6 dB/oct. Combining the output of five microphones will increase the white noise gain by 7 dB, so the delay-and-sum combination of five cardioid microphones has a white noise gain of -13 dB at 500 Hz, improving to $+5$ dB at 4000 Hz. Thus, if a delay-and-sum beamforming array of cardioid microphones and a constrained superdirective array are equalized to have the same gain at 500 Hz, the delay-and-sum array will have a 13-dB-higher internal noise level. Only above 2000 Hz do the two array designs have both comparable array and white noise gains; at lower frequencies the optimal weights give substantially better white noise gains than those found for the cardioid microphone array, even though the array gains are similar.

III. CONCLUSIONS

The simulation results presented in this paper indicate that a superdirective array incorporating a sensitivity constraint is an attractive candidate for hearing-aid applications. The optimal weights give the best possible performance in a spherically isotropic noise field, and the 0-dB white noise constraint guarantees that the array self-noise will be no worse than that from a single omnidirectional microphone while still limiting the sensitivity to wave-front perturbations and errors in the microphone positions. The array gain for a 10-cm-long constrained optimal array comprising five omnidirectional microphones ranges from 5.9 dB at 500 Hz to 12.1 dB at 4000 Hz. The array gains average 5.0 dB greater than those for delay-and-sum beamforming using omnidirectional sensors, and it must be remembered that the delay-and-sum array provides very little benefit at or below 1000 Hz. A delay-and-sum array using cardioid microphones gives array gains that are similar to those of the constrained optimal array, but at the expense of greatly increased white noise gains at low frequencies. Thus the constrained optimal array, when compared with other arrays having fixed weights, offers an excellent combination of useful array gain without debilitating reductions in the white noise gain. If desired, a null at 180 deg can be added to the array response with only a small (0.3-dB) average penalty in array gain.

A further consideration would be a comparison between a superdirective array and an adaptive array having the same length and number of sensors. But adaptive arrays have been shown to cancel the desired signal in the presence of strong room reflections, and until this problem is solved in a practical system, the superdirective array may be preferable. In addition, an adaptive array will require greater signal-processing capacity in order to update the weights. The fixed weights of the superdirective array, however, mean that the potential advantages of an adaptive array in steering nulls in the direction of strong noise sources cannot be realized. Thus, improved adaptive arrays for hearing aids should still be a research objective, since an adaptive system can approach the optimal solution for each individual signal and noise situation, while the superdirective array represents an optimal solution only to the generic problem of a spherically isotropic noise field.

ACKNOWLEDGMENT

This work was supported by NIDCD under Grant No. 2P01DC00178.

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