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AN ADAPTIVE BEAM-STEERING MICROPHONE ARRAY IMPLEMENTED ON THE MOTOROLA DSP56000 DIGITAL SIGNAL PROCESSOR

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INTRODUCTION

Adaptive microphone arrays seem to offer the potential of outstanding directional performance. Often the figures achieved by, seemingly equivalent, antennae arrays are quoted as examples. However, whereas antenna arrays operate in near anechoic environments, microphone arrays do not. In fact, the presence of a reverberant field significantly reduces the maximum performance achieved by such an array. It also alters the optimum adaptation strategy. The purpose of this paper is to describe results from an adaptive microphone in both anechoic and "real acoustic" environments.

BACKGROUND

An adaptive microphone array is shown in Figure 1. It consists of several microphones spaced by a distance that should be less than $\lambda/2$ at the highest frequency being used. The outputs of these microphones are combined via individual adaptive filters. The function of these filters is to;

(i) compensate for the inevitable inter-microphone variation

(ii) Modify the polar pattern of the array as required by the system.

It is important to note that the system is a linear one so that any modification of the weight of the filters will modify the effective polar pattern of the system.

There are two possible ways of adapting this system.

(i) **Beam forming:** the array tries to maximise the energy it receives in a particular direction, the look direction. The number of microphones determines the gain in the look direction.

(ii) **Null steering:** the array tries to place nulls in particular directions to remove interfering noise sources. The number of microphones determines the number of nulls that can be steered.

Null steering is often used in an anechoic environment because it offers excellent rejection of interfering sources. However, in a reverberant environment this is not the case, here beam steering seems to offer the best solution. However, the improvement factor to uncorrelated interference is only proportional to $\sqrt{N}$ where N is the number of microphones. One can argue that a diffuse reverberant field is essentially uncorrelated. This limits the maximum achievable performance of a microphone array.

To verify the utility of beam steering we implemented a beam steering array so that we could compare its performance with both theoretical results and a single directional microphone.

THE IMPLEMENTED SYSTEM

An array of seven omni-directional microphones was interfaced to a Motorola 56001 digital signal processor, and DSP code was written to directly implement the standard Least Mean Squares (LMS) adaptation algorithm [1]. It uses an eighth input for the "desired output" reference and software generated tapped delay lines. The iterative LMS algorithm involves recursively updating the FIR filter coefficients to minimise the error between the training signal input, and that received by the array. The configuration is shown schematically in Figure 2. The adaptation phase is separate from the operating one. Following adaptation, the filter coefficients are stored for each steered direction, to enable further array analysis. In a more practical set-up, the training phase could be easily simulated by using a computer to electrically feed array inputs to the processor. This would generate the coefficient values without using an anechoic chamber. This was necessary in our implementation to avoid the effects of multi-path interference. Once the coefficients for different look directions have been they can be selected as part of a track-while-scan system for speaker seeking or tracking [2]. This aspect, however, was not investigated.

After coefficient generation, under anechoic conditions, polar plots were generated of the array's directivity using a Maximal Length System Sequence Analyser (MLSSA). Figures 3 and 4 illustrate array performance for a broad-steered beam and 30 degree beam. These directly relate to the theoretical plots and give a direct comparison between theory and practice. Below 2kHz, in the broad-steered case, there is very strong correlation between theoretical and practical curves, suggesting numerical evaluation of directivity from Flanagan's equations [2,3] accurately predict array performance. At higher frequencies, greater error is observed. This may be largely explained by the resolution of polar plot measurement. They were only 5 degrees and so simply missed the narrow lobes. The 30 degree steered plots are less conclusive, though still indicate reasonable accuracy given the nature of the experiments.

Having verified that the system broadly performed as predicted, the array response was analysed under reverberant conditions. The room used represented a non-ideal operating environment. The room being small, square, cluttered, with no acoustic treatment, and significant interfering sources. As before, plots were generated with two array beam directions, the results being illustrated in Figures 5 and 6, with an extra plot showing the average response across the speech frequency bandwidth. The plots no longer represent the directivity of the array, since each measurement includes energy received from all directions, i.e. from reflections, and the diffuse field. However, they do indicate greater sensitivity to sources in the steered direction, suggesting some degree of useful rejection. The average plot indicates this to be about 8dB in the axis of the array for both cases.

ADAPTIVE BEAM-STEERING MICROPHONE ARRAY

For the purposes of comparison, similar plots were obtained in the same room for an AKG cardioid microphone (the AKG C414EB-P48), the average results being illustrated in figure 7. The array is shown to be considerably more directive, though this is only true in the plane, that contains all the array microphones. Secondly, measurements were made of the received direct-to-reverberant ratios, and are presented in Table 1. The improvement of the array over the cardioid microphone (about 1.5dB) is significant because of the one dimensional nature of the test array. Simple extension to two dimensions, by swapping each array element for a column of microphones that are summed externally to the processor to reduce the processing power required, would considerably improve this. This arrangement could possibly challenge the directivity response of a commercial Gun microphone (about 8dB), without further digital hardware or DSP software.

CONCLUSION

The commercial realisation of a practical, useful, steerable, beam-forming microphone array is still some way off. The principle problem is not, however, the complexity of implementation, but the adverse acoustic environments in which such systems are expected to operate. The ultimate limitation to directivity improvement in a diffuse acoustic field is the number of array transducers. This implies, to some extent, that computing power is also a limitation because it is proportional to the number of inputs. However, summing transducer elements in the analogue domain provides some scope for reducing the processing required.

The first stage of confirming theoretical predictions of array responses in anechoic conditions have been successful. The next stage must be the analysis of the achievable performance, for reverberant field rejection, of array beam-forming. In acoustic environments that suffer predominantly from discrete reflection interference, null-steering systems may be more appropriate. The performances of these systems are highly application dependent, and hence difficult to predict. In the authors' opinion, the immediate future for array beam-forming is in applications that improve sound transduction in acoustically treated environments where the improved directionality enables microphones to be placed further from sources. One could then use null-steering to provide some rejection of external, and hence uncorrelated, interfering sources, for example other speakers in a teleconference.

REFERENCES

1 Widrow, B and Stearns, S "Adaptive Signal Processing," Prentice Hall, 1985

2 Flanagan, J. L. "Beamwidth and Usable Bandwidth for Delay-Steered Microphone Arrays," *AT&T Technical Journal,*, **64**, No 4, pp 983-995.

3 Flanagan, J. L. "Computer-Steered Microphone Arrays For Sound Transduction in Large Rooms," *Journal of the Acoustical Society of America*, **78**, No. 5, November

ADAPTIVE BEAM-STEERING MICROPHONE ARRAY

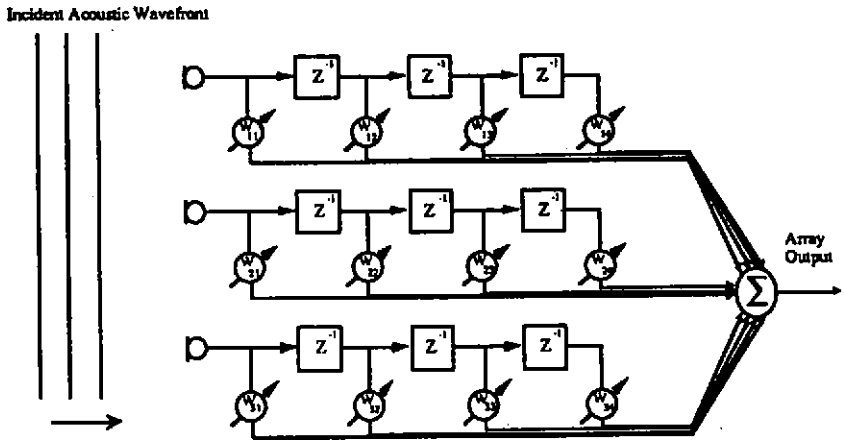


Figure 1: Microphone Array with FIR Filters for Delay Elements

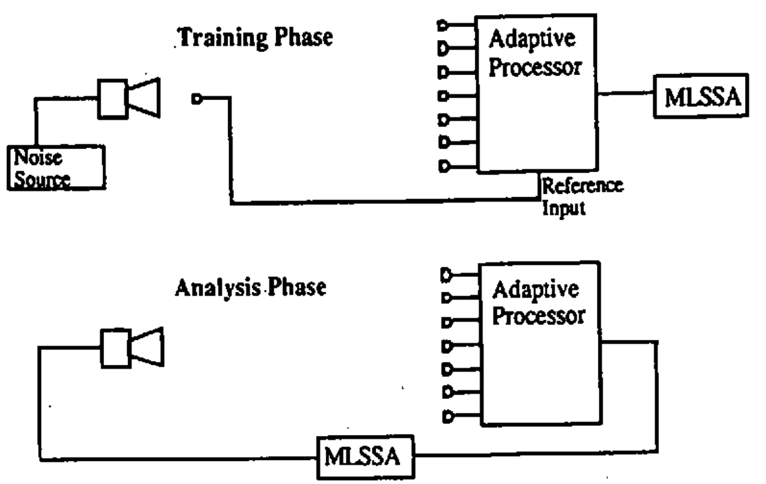
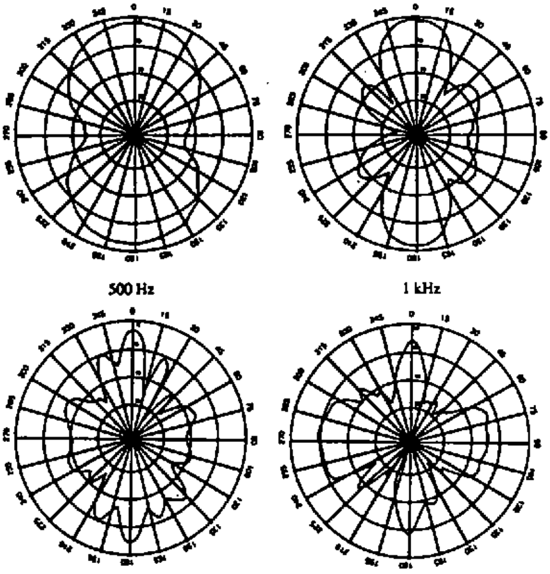


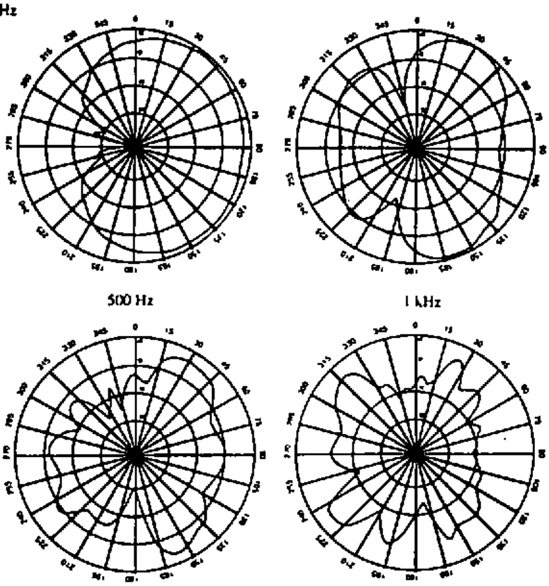
Figure 2: System Configurations for Both Array Adaptation and System Analysis

ADAPTIVE BEAM-STEERING MICROPHONE ARRAY



500 Hz 1 kHz 2 kHz 3.4 kHz

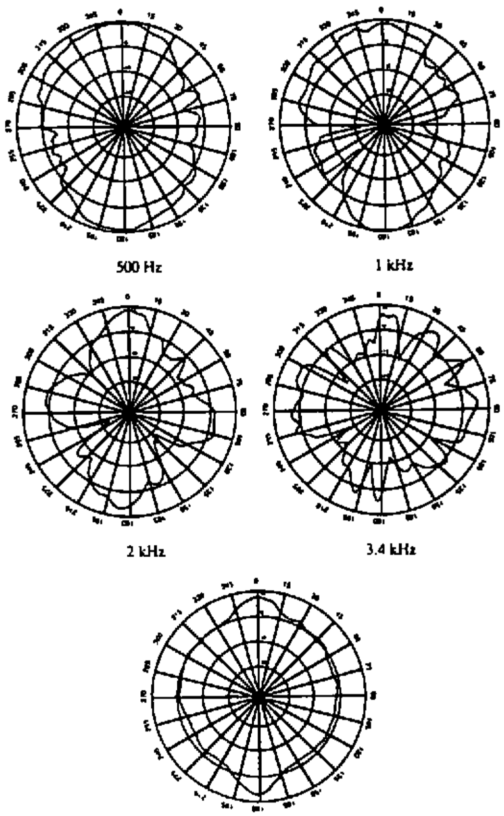
Figure 3: Directivity Plots for a Broad-Steered Array in an Anechoic Chamber



500 Hz 1 kHz 2 kHz 3.4 kHz

Figure 4: Array Steered to 30° Inside an Anechoic Chamber

ADAPTIVE BEAM-STEERING MICROPHONE ARRAY



500 Hz   1 kHz

2 kHz   3.4 kHz

Average For All Frequencies from 500 Hz to 3.4 kHz

Figure 5: Broad-Steered Array Results in a Real Room

ADAPTIVE BEAM-STEERING MICROPHONE ARRAY

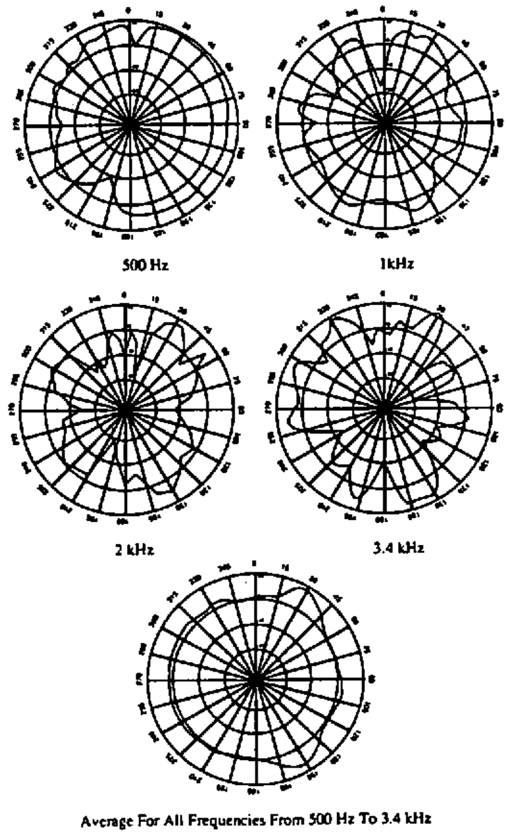
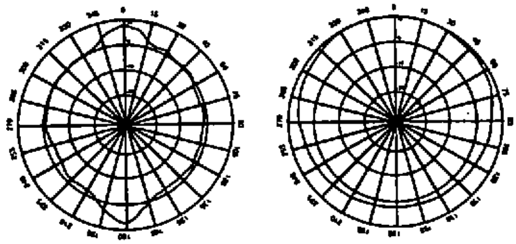


Figure.6: Steered Array Results in a Real Room



Array AKG Cardioid
Average For All Frequencies From 500 Hz To 3.4 kHz

Figure 7: **Directivity Responses of a Seven Microphone Array Compared to an AKG C414EB-P48 Cardioid Microphone**

Real Room	DRR	Source - Array Dist.
AKG 414EB P48	0.17 dB	1.20 m
Array Steered To 0°	1.64 dB	1.20 m
Array Steered To 30°	1.69 dB	1.20 m
Array Steered To 60°	0.42 dB	1.20 m
Anechoic Chamber	**DRR**	**Source - Array Dist.**
Single OMNI-Microphone	1.88 dB	2.58 m
Array Steered To 0°	4.72 dB	2.58 m
Array Steered To 30°	4.7 dB	2.58 m

Table 1: **Direct-to-Reveberant Ratios for Two Test Environments**