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## **Localization of noise source and primary reflections in a room using a spherical array and aggregate beamforming technique**

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**[N275] Localization of noise source and primary reflections in a room using a spherical array and aggregate beamforming technique**

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**ABSTRACT**

Aggregate beamforming is described as an alternative to conventional beamforming for reducing the hardware complexity of arrays with many elements. A 32-element spherical array has been designed for the analysis of directional and spatial variations of sound fields in rooms. These two technologies are combined for the purpose of localizing early reflections within a room. The alternative beamforming method and conventional methods, such as delay and sum and superdirective beamforming, are compared. The effectiveness of the methods for identifying significant reflectors and, ultimately, assessing noise control strategies is examined.

**KEYWORDS:** Beamforming, Microphone array, Source localization

## INTRODUCTION

A noise source localization system within a room must be capable of distinguishing direct and reflected sounds. For noise control indoors it is advantageous to have accurate knowledge of the location of both the noise source and its major reflections. Beamforming with microphone arrays [1] can be simpler than holographic techniques and is more easily applied to large volumes. In this study we demonstrate that simplified beamforming methods can provide useful source localization information.

A method for reducing the hardware requirements for a beamforming array has been proposed [2,3]. This method, called *aggregate beamforming*, makes it possible to eliminate the analog anti-aliasing filters that are normally required on each channel of a microphone array. The method offers other simplifications to array beamforming as well, like reducing the total sampling rate and achieving higher beam-steering resolution without data interpolation.

A 32-element spherical array has been demonstrated [4] for estimating the direction of arrival of sound and sound field anisotropy in a room. This system was carefully calibrated and the array data processing was highly optimized for the application. Here we use the same microphone array with noisier data acquisition equipment and simpler beamforming algorithms.

The ability to resolve noise sources with the 32-element spherical array is compared for:

- (a) the optimized beamforming method,
- (b) basic delay and sum beamforming, and
- (c) the aggregate beamformer.

## THE BEAMFORMERS

### Optimized Beamformer

Optimized beamforming is discussed, for example in [5]. The optimized beamformer output  $B_{\text{opt}}$  is

$$B_{\text{opt}} = \sum_{m=0}^{M-1} h_m \otimes x_m, \quad (1)$$

where  $\otimes$  indicates convolution,  $h_m(n)$  are the optimized FIR filter coefficients, and  $x_m$  is the

signal on sensor  $m$ . The beamforming time delays are applied implicitly by the phase response of the FIR filter and the delays are not constrained to integral multiples of the sampling interval. Each beam direction requires its own set of optimized filter coefficients. For this study, the beamformer was optimized assuming a ratio of signal to uncorrelated noise of 5 dB over the frequency band 1000 to 3300 Hz. Designs for lower snr, which would theoretically provide better array gain at lower frequencies, were found to be ineffective at the relatively high noise and inter-channel variations of the data acquisition system.

### Delay and Sum Beamformer

The delay and sum beamformer output  $B_{DS}$  is obtained by summing the weighted and delayed sensor channel signals,

$$B_{DS}(n) = \sum_{m=0}^{M-1} W_m x_m(n - t_m). \quad (2)$$

The beamforming time delay  $t_m$  must be quantized to the sampling interval (for *basic delay and sum beamforming*) unless the sensor data is first interpolated to a higher sampling resolution (which is not done here). Each beam steering direction requires its own set of time delays. The channel weights  $W_m$  are all equal and sum to unity for this study.

### Aggregate Beamformer

The aggregate beamformer [2,3] employs a random channel-sampling method. The aggregate beamformer output  $B_{agg}$  is obtained by decimating the time-aligned sequence of all the sampled data,

$$B_{agg}(n) = \sum_k h_d(k) Z'(nK_{os} - k), \quad (3a)$$

where  $h_d$  is a decimation filter,  $K_{os}$  is the decimation (or oversampling) factor given by the ratio of the total sampling rate of the array to the final desired signal sampling rate, and  $Z'$  is derived from the time-aligned sequence of sampled data  $Z_{agg}$

$$Z_{agg}(n) = x_{s'(n)}(n - t_{s'(n)}), \quad (3b)$$

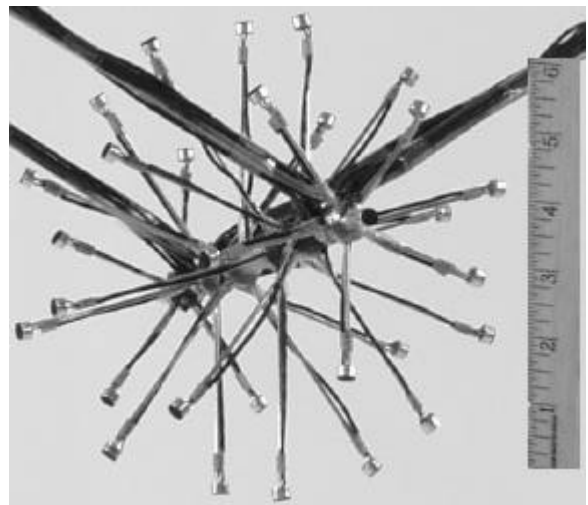
as explained below. The time delays for each channel  $t_m$  (with  $m=s'(n)$ ) are determined as for delay and sum beamforming but quantized to the higher sampling rate. The index  $s'(n)$  is a (modified) random sequence of channel indices obtained from a uniformly distributed sequence of channel indices  $s$  modified so as to ensure that only one channel is sampled at a time. This is done by initially setting  $s'$  equal to  $s$  and, for each successive a value of  $n$ , if

there exist  $n' < n$  such that  $n - t_{\sigma(n)} = n' - t_{\sigma(n')}$  then, based on a random choice, either  $\mathbf{S}'(n')$  or  $\mathbf{S}'(n)$  is set to an undefined value (such as  $\infty$ ). This process is referred to as collision management. Since the modified sequence  $\mathbf{S}'$  has undefined values, so will  $Z_{\text{agg}}$ . The final sequence of sample data  $Z'$  is obtained from  $Z_{\text{agg}}$  by a process of 'void filling', wherein the undefined entries are simply filled with the most recent previously defined value.

The aggregate beamformer is described in more detail in [2,3]. It provides the same signal output (on average) as the delay and sum beamformer but has an additional small amount of white Gaussian noise. The time delays of the aggregate beamformer are quantized to the total sampling rate of the collected data, thereby avoiding the costly signal interpolation necessary to obtain the same delay quantization with the delay and sum beamformer. An important advantage of the aggregate beamformer is that the incoming signal is reconstructed at a higher average sampling rate (in  $Z_{\text{agg}}$ ), making it unnecessary to apply anti-aliasing filters to each input channel of the array.

## THE MICROPHONE ARRAY

The microphone array has 32 elements and is illustrated in Fig. 1. Each sensor is a 6-mm omni directional electret microphone with an essentially flat frequency response over the audio spectrum. They are positioned on the surface of a sphere 16 cm in diameter at the vertices of a regular geodesic with isosceles triangular faces. The separations of microphones in the geodesic are 5.1 and 5.7 cm. The structure of the array is quite simple, with the electret capsules mounted on spokes radiating cylindrically from a shaft that defines the main axis of the array. The array, its design, and its application to sound field characterization are discussed in [4]. For this



*Figure 1 32-Element spherical array.*

study, the beamforming directions are constrained to the horizontal plane, with 0 degrees being in the direction of the array's main axis. The beam width varies with frequency and beamforming method but is nominally about 30 degrees. A total of 36 beams are formed at 10-degree increments on the horizontal plane.

## THE EXPERIMENTAL SETUP

The microphone array is suspended at the surface of the floor in a large test room. Two 4' by 8' (1.2 by 2.4 m) sheets of plywood,  $\frac{3}{4}$ " (19 mm) thick, are positioned vertically on opposite sides of the array so as to form parallel reflecting surfaces 3 m apart. A point source, constructed from a 1.44 m tube 25 mm in diameter attached to a compression driver, is positioned on the floor 1.5 m from the array. The array and source are equidistant from the sheets. Figure 2 shows the configuration of microphone array, source and one of the panels. The walls and ceiling of the test room are far enough away that reflections from their surfaces can be removed by time-gating techniques.

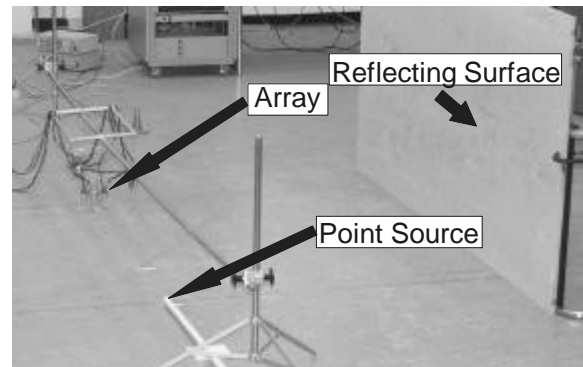


Figure 2 Experiment setup.

The source signal is a 2.5 ms burst consisting of a smoothly and linearly swept tone from about 1.5 kHz to about 5 kHz. The direct arrival at the array is followed by simultaneous arrival of the first pair of reflections from the panels. Note that these coincident signals would not be resolved by an intensity probe; intensity in this case would only reveal the net energy transport in the direction of the direct sound. A series of reflections follow the direct arrival and, although several orders of these can be resolved (Fig. 3), only the first and second pairs are considered in this study.

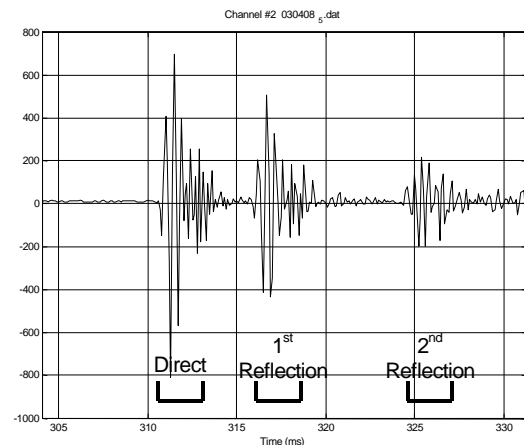


Figure 3 Direct signal and first two reflections.

Each channel of the array has a pre-amplifier with 40 dB gain and an analog filter with a 500 Hz highpass and a 4 kHz, 12<sup>th</sup> order anti-aliasing lowpass section. The filter outputs are connected to a switching network that selects array channels under computer control. The output of the switcher goes to a 12-bit analog to digital converter sampling at 320 000 samples/s (10 000 samples/s/channel). The digitized data is transferred to a 233 MHz PC for file storage. The signal processing is done off-line using the raw array data stored on disk.

For aggregate beamforming, the filters are bypassed and the switcher channel is selected in a pre-determined random sequence. The optimized and delay and sum beamforming use sequential sampling and share the same dataset. For the optimized beamforming only, microphone and pre-amplifier calibration corrections are applied.

## SIGNAL ANALYSIS

Figure 4 shows the rms signal level in each beam for each of the beamformers during the direct arrival pulse. (The time of the direct pulse is indicated in Fig. 4.) All of the beamforming methods produce a distinctive peak in the 90 degree direction, which corresponds to the direction of the source. Similarly, Fig. 5 shows the directional response during the arrival of the first reflection pair. In this case, there is little signal arriving from the source but the reflections are clearly seen at about 30 and 150 degrees. The two reflected signals arrive at the array simultaneously yet the beamformers are able to resolve them.

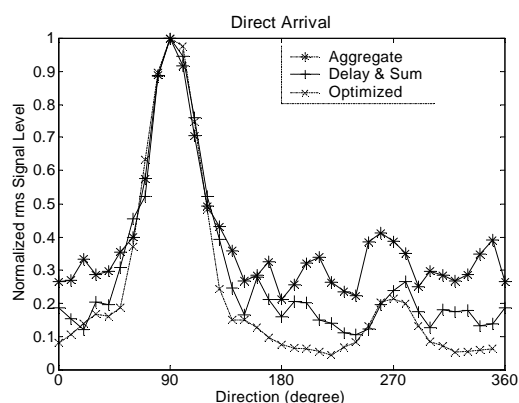


Figure 4 Directional response during direct arrival.

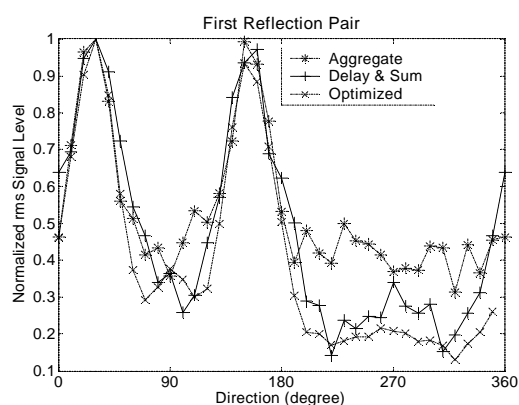


Figure 5 Directional response during 1<sup>st</sup> reflection.



The arrival times of the direct signal and its reflections are estimated by detecting peaks in the cross-correlation of a template of the pulse signal and the beamformed signals. The template is obtained from the waveform of the direct arrival on the 90-degree beam. If the template is normalized to unit power and the co-variance is computed, rather than the cross-correlation, then the height of the covariance pulse measures the rms power relative to that of the direct arrival.

Beamforming and cross-correlation allows the separation of signals by both direction and time of arrival. Figures 6, 7, and 8 show the direct and image source location estimates for the optimized beamformer, the conventional delay and sum beamformer, and the aggregate beamformer, respectively. A representation of the true room geometry is overlaid. The reflecting panels are indicated with heavy lines, the spherical array with an open box, and the source and image positions with solid boxes. A straight line is drawn through the images and the source. Only the optimal beamformer located the two second reflections; the D&S beamformer misses the second reflection on the left while the aggregate beamformer produces a spurious reflection. Nonetheless in each case, the position estimates intersect the actual location of the source and the first reflection pair.

## DISCUSSION AND CONCLUSIONS

Although the angular resolution may be improved by using a larger array size, better position estimation precision can also be obtained if the

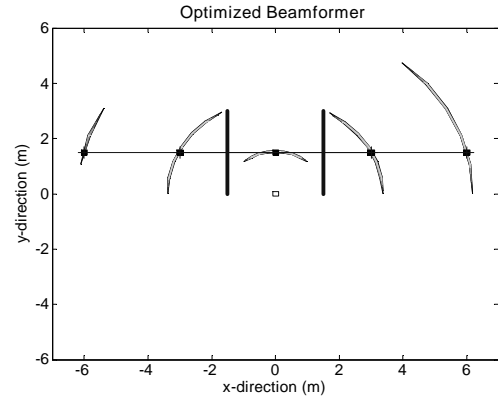


Figure 6 Optimal beamformer position estimates from cross-correlation peaks.

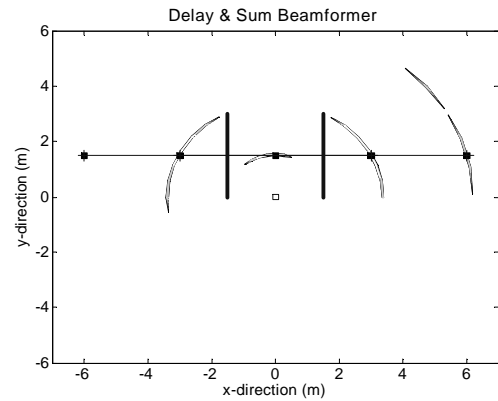


Figure 7 Delay & Sum beamformer position estimates from cross-correlation peaks.

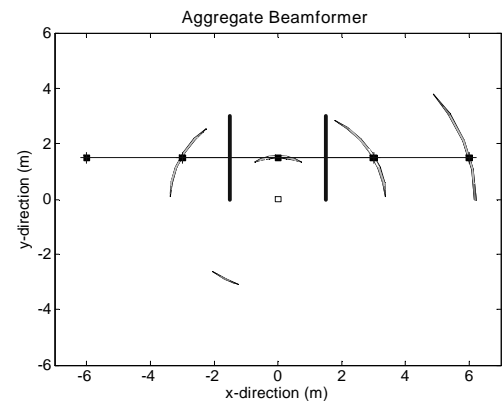


Figure 8 Aggregate beamformer position estimates from cross-correlation peaks.

procedure is repeated with the array located at two or more positions within the room. The intersection of the resulting arcs would then more accurately indicate the source and image positions. The magnitude of the covariance function can be used to eliminate spurious peaks due to the side-lobes of the beamformers. The magnitude, together with the propagation time, can also provide information on the surface reflection coefficient. As demonstrated here, the physical geometry of the room can be used to localize the source and its images. The template correlation method may work poorly for sources with high anisotropy. For continuous broadband noise sources, a longer template may be needed to reduce the correlation with the background, reverberant noise energy. As the nature of the source noise becomes more tonal, the range resolution will degrade but it will still be possible to localize sources by triangulating measurements from multiple array positions.

It has been shown that by using beamforming and template matching, received signals can be resolved in both time and direction of arrival so that the physical position of the source and individual reflection images can be resolved and their relative strengths determined. Coincident, coherent arrivals are also resolved with the microphone array whereas this is not possible with intensity probe measurements. The complexity of array processing can be greatly reduced without seriously affecting the efficacy of the method. The aggregate beamformer is an effective alternative beamforming method. Although it requires a decimation filter for each beam, it avoids the use of anti-alias filters, digital interpolation filters, or snr-dependent optimized FIR coefficients on each input channel. Furthermore, for arrays with many elements, it can be operated at a lower total sampling rate without being constrained by the Nyquist sampling condition on individual input channels.

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