SPHERICAL MICROPHONE ARRAYS FOR ANALYSIS OF SOUND FIELDS IN BUILDINGS

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1. INTRODUCTION

In evaluating room acoustics, sound transmission through walls and floors, or other building acoustics problems, a directional sound detector can potentially be of great value. Beamforming microphone arrays have become increasingly practical and affordable in recent years, and can make flexible and highly directional detectors. This paper describes two recently developed arrays, and application to some building acoustics problems.

2. ARRAY DESIGN ALGORITHM

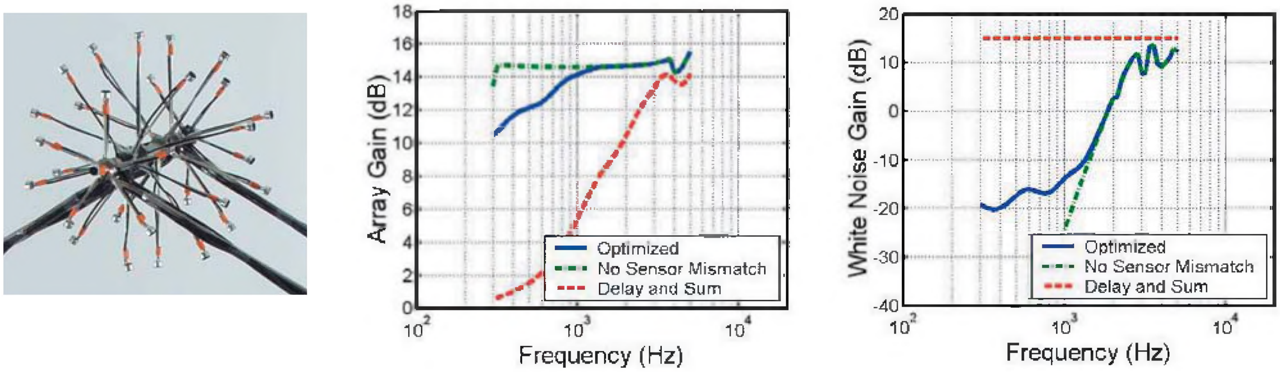
For a given geometrical arrangement of sensors (omnidirectional microphones), any number of beamformers can be implemented to realize a variety of beampatterns. A simple approach is to "delay and sum" the element signals, with no magnitude scaling. A more powerful approach is to "filter and sum" to implement a frequency-dependent magnitude and phase weighting for each sensor signal. For a given set of sensor weights $\mathbf{w}(\omega)$, the *array gain* $G(\omega)$ at frequency ω is given by

$$G(\omega)=\frac{\mathbf{w}^{H}\mathbf{R}_{SS}\mathbf{w}}{\mathbf{w}^{H}\mathbf{R}_{NN}\mathbf{w}}, \tag{1}$$

where $\mathbf{R}_{SS}$ is the signal correlation matrix, determined by the steering direction, and $\mathbf{R}_{NN}$ is the noise correlation matrix, determined by the noise at the sensors. One robust beamformer design seeks to maximize G, subject to constraints on the white noise gain [1]. The white noise gain is given by Eq. (1) for the case $\mathbf{R}_{NN} = \mathbf{I}$, and is a measure of the ability of the beamformer to tolerate uncorrelated noise. In the current work, an estimate of the sensor magnitude and phase mismatch is used to construct a noise constraint [2]. The signal and noise correlation matrices in Eq. (1) are replaced with the expected matrices, incorporating the mismatch, and the sensor weights $\mathbf{w}$ that maximize G are determined for a range of frequencies. This procedure specifies, in the frequency domain, the filter weights $\mathbf{w}$.

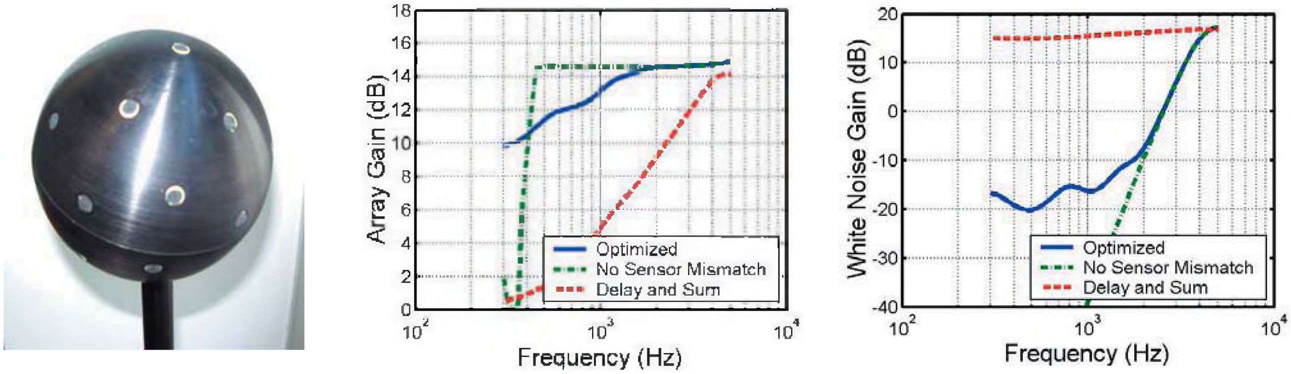
3. ARRAYS

Two types of arrays have been designed and constructed. Shown in Fig. 1(a) is a 16 cm diameter "open" or "free field" array, in which 32 omnidirectional electret microphones (6 mm diameter) lie on the surface of a (notional) sphere. This array has been discussed in Ref. [3]. The geometry is that of a "pentakis dodecahedron", related to the familiar 32-faced soccer ball. Fig. 1(b) shows the array gain versus frequency assuming a 0.1 dB sensitivity mismatch among microphones (solid curve). Shown for comparison are: the maximum gain possible assuming no sensor mismatch (dash-dot curve), and the gain for a delay and sum beamformer (dashed curve). Fig. 1(c) shows the white noise gain for all three beamformer designs. The higher the white noise gain, the better the ability to tolerate noise. Fig. 2 shows comparable results for a newly-constructed 10 cm diameter rigid array, in which 32 omnidirectional electret microphones (6 mm diameter) are flush mounted in the surface of a hollow aluminum sphere. The scattering of sound by the sphere is taken into consideration in the beamformer design. The rigid sphere increases the apparent separation of the microphones, and also serves to smooth the white noise gain.



(a) (b) (c)

Fig. 1: 16 cm diameter "free field" array: (a) Photograph, (b) array gain and (c) white noise gain for: optimal design assuming 0.1 dB sensor mismatch (solid curve), no sensor mismatch (dash-dot curve), and delay and sum (dashed curve).



(a) (b) (c)

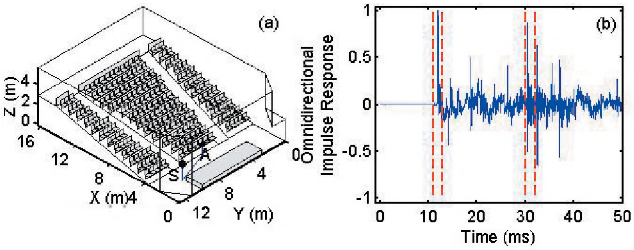
Fig. 2: 10 cm diameter rigid array: (a) Photograph, (b) array gain and (c) white noise gain for: optimal design assuming 0.1 dB sensor mismatch (solid curve), no sensor mismatch (dash-dot curve), and delay and sum (dashed curve).

It can be seen for both arrays that the optimal design "trades off" some robustness to noise for increased gain, or alternately, "trades off" some of the maximum theoretical gain for increased robustness to noise. For both arrays there is a frequency range of about 1.7 octaves over which the array gain is flat, and greater than 14 dB or so. The beam pattern in this range (not shown) has a main lobe beamwidth of about 28 degrees.

4. APPLICATIONS

4.1. Analysis of sound fields in rooms

Shown in Fig. 3 are results from Ref. [4] using a free field array to measure sound arriving at a point in a lecture theatre. The room is shown in (a), and the omnidirectional impulse response at one of the array elements is shown in (b). In panels (c)–(e), the radius of the surface in a given direction indicates the level arriving from that direction. Panel (c) shows the levels arriving at the array position over the entire time record, allowing analysis of the isotropy of the sound field. Panels (d), (e) indicate the directions of incidence of individual arrivals or reflections.



(c) Full-time: [0,600] ms (d) Direct Sound: [11,13] ms (e) Reflection: [30,32] ms

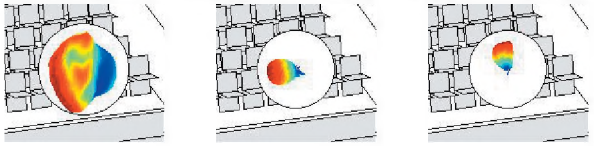


Fig. 3: Measurement in lecture theatre: (a) room, array position (A), and source position (S); (b) omnidirectional impulse response; (c) sound arriving at array position, integrated over entire time record; (d) sound arriving at array between 11–13 ms, corresponding to direct arrival; (e) sound arriving at array between 30–32 ms, corresponding to ceiling reflection.

4.2. Detection of sound leaks in walls

Figure 4 shows results from Ref. [5] indicating the levels arriving at a free field array, projected onto a wall that separated the array from the room containing the sound source. The wall contained a pair of back-to-back electrical boxes, which caused increased levels at the array position, arriving from their direction. The boxes constituted a sound leak in the wall, which was identified by the array.

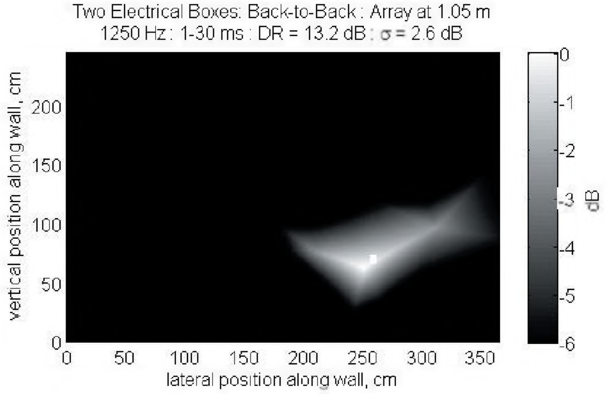


Fig. 4: Levels arriving at array position, projected onto wall that contained back-to-back electrical boxes. Light feature indicates increased sound transmission through the defect.

5. CONCLUSIONS

Beamforming microphone arrays are promising diagnostic tools for the analysis of sound in buildings. The array designs can be highly directional and robust.

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