Super directivity design for a sphere-baffled microphone array

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Abstract: For the signal extraction from a diffuse sound field, the superdirectional microphone array seems to be more stable and more effective than the noise canceling microphone array system. In this research, for the Sphere-Baffled Microphone array (SBM), the superdirectivity characteristic was designed by the frequency domain beamforming and Least Square Error method.

INTRODUCTION

For the signal extraction from a diffuse sound field and for picking up sound emitted by a distant source, the superdirectional microphone array seems to be the most effective system. In this research, the Sphere-Baffled Microphone array (SBM)[1] of FIGURE 1 is used. SBM is an array which has embedded microphones on rigid diffractive sphere of 8.84 cm radius. The size of SBM is the statistical average of human heads. Diffractive coefficients of SBM is calculated by 1 degree interval. These coefficients can be used as the diffractive dictionary. In this research, we assume that the wave front from sound source is perpendicular to the horizontal plane. Microphones are mounted on the horizontal great circle of sphere.

An advantage of SBM is its isotropy. When the directivity characteristic was designed and fixed for a definite direction, only the shift weights can be used to change the mainlobe direction. This idea can be easily generalized into the 3-dimensional mainlobe control.

FREQUENCY DOMAIN BEAMFORMING

A frequency domain beamformer is used as the structure. This beamformer decomposes each microphone signals into frequency domain by taking a Discrete Fourier Transform. At each frequency bin, weighted sum are calculated as frequency domain output. And an inverse Discrete Fourier Transform produces output signal (FIGURE 2). At k-th bin, received signal \( X(k) \), weight vector \( W(k) \) and output signal \( Y(k) \) are expressed as follows. \( T \) stands for transpose.

\[
X(k) = [X_1(k) \ldots X_m(k) \ldots X_M(k)]^T
\]

\[
W(k) = [W_1(k) \ldots W_m(k) \ldots W_M(k)]^T
\]

\[
Y(k) = X(k)^T W(k) = W(k)^T X(k)
\]

\[
Y = [Y(0) \ldots Y(k) \ldots Y(N-1)]^T
\]
DIRECTIVITY DESIGN AND RESULTS

Because the diffractive coefficients has calculated 1 degree interval, desired characteristics $D(k)$ is also set 1 degree interval.

$$D = [D(k,0), \ldots, D(k,\theta), \ldots, D(k,359)] \tag{5}$$

To design a directivity is to calculate optimum weight $W(k)$ in equation (6) at each frequency bin.

$$\begin{bmatrix} X_1^1 & \ldots & X_m^1 & \ldots & X_M^1 \\ X_1^2 & \ldots & X_m^2 & \ldots & X_M^2 \\ \vdots & \ddots & \vdots & \ddots & \vdots \\ X_1^{360} & \ldots & X_m^{360} & \ldots & X_M^{360} \end{bmatrix} \begin{bmatrix} W_1 \\ \vdots \\ W_m \end{bmatrix} \geq \begin{bmatrix} D_1 \\ \vdots \\ D_m \end{bmatrix} \tag{6}$$

TABLE 1 shows an example of parameters for design. In desired directivity, transient regions (between mainlobe and sidelobe) of 5 degrees are left for "don't care region". As least squares solution, $W(k)$ are designed. A obtained directivity characteristic is shown in FIGURE 3. In the frequency region higher than 2kHz, sharp mainlobe shape is achieved and the frequency characteristic is almost flat.

<table>
<thead>
<tr>
<th>TABLE 1. Parameters for Design</th>
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</thead>
<tbody>
<tr>
<td>The number of Microphone M</td>
</tr>
<tr>
<td>Sampling Frequency $f_s$</td>
</tr>
<tr>
<td>Size of DFT N</td>
</tr>
<tr>
<td>Mainlobe direction</td>
</tr>
<tr>
<td>Desired mainlobe width</td>
</tr>
<tr>
<td>Desired sidelobe level</td>
</tr>
</tbody>
</table>

FIGURE 3. A obtained directivity characteristic

REFERENCES

2. Van Veen, B.D. and Buckley, K.M., IEEE ASSP Magazine April, 4-24 (1988)