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ESTIMATION OF DIRECTION OF ENVIRONMENTAL NOISE SOURCES BY USING ARRAY MICROPHONE SYSTEM

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ABSTRACT

In our living environment, there are many environmental noises that affect our daily life such as factory engines, construction machinery and so on. Although it might be possible to reduce an environmental noise under a direction of the environmental noise source is known, it is difficult to estimate the direction of the environmental noise sources in complex environmental conditions. In this paper, we propose a method for estimating a direction of an environmental noise source by using MUSIC algorithm. In order to evaluate the method, experiments in anechoic chamber are carried out. In our experiments, a linear array microphone system composed of 5 microphones and number of the sound source is assumed to 4. The results of the experiments are that the direction of the sound sources respectively is estimated.

1 - INTRODUCTION

Over the past many years, research for reducing environmental noise that audio sound is considered mainly in this paper has been studied and many methods have been developed simultaneously, for example, active noise control method, noise control by use of enclosure or by bulk absorbing materials, particularly partial enclosure, silencers and mufflers are set up around an environmental noise and so on. These methods are applicable to that localization and characteristics of an environmental noise is known. However, it is difficult to estimate a position of the environmental noise in complex environment that there are many environmental noises, in presence of reflection and diffraction. On the other hand, in many sensor systems, array signal processing technology such as maximum likelihood method, maximum entropy method and high resolution direction-of-arrival (DOA) estimation methods have been widely applied to estimate DOA of signal. Also, it is possible to estimate the DOA of signal in presence of reflection, in real time with digital signal processing technology. In this paper, we used the MUSIC (Multiple Signal Classification) algorithm that is one of the high-resolution DOA technologies to estimate direction of DOA of environmental noise referenced to sound sources in this paper. The problems in the experiments and in data processing are discussed also in this paper.

2 - MUSIC ALGORITHM

MUSIC applies to a wide variety of problem focus on DOA estimation. However, it is necessary for a few assumptions to make the problem analytically tractable. These assumptions are that the transmission medium is isotropic and non-dispersive so that the radiation propagates in straight lines, and the sound sources are assumed to be in far field of the linear array microphone. Consequently, the radiation impinging on the array microphone is in the form of a sum of plane waves [1], [2].

2.1 - Data model

Assume that there are m microphones, n narrow-band sound sources ($n \leq m$). The received signal data at t time can be expressed as next equation.

$$\mathbf{X}(t) = \mathbf{AS}(t) + \mathbf{N}(t) \tag{1}$$

where

- $\mathbf{S}(t)$: amplitude of received signal data
- $\mathbf{A}$: steering vector; $\mathbf{A} = [\mathbf{a}(\theta_1), \mathbf{a}(\theta_2), \ldots, \mathbf{a}(\theta_n)]$
- θ_n: DOA of nth sound source
- $\mathbf{N}(t)$: measurement noise.

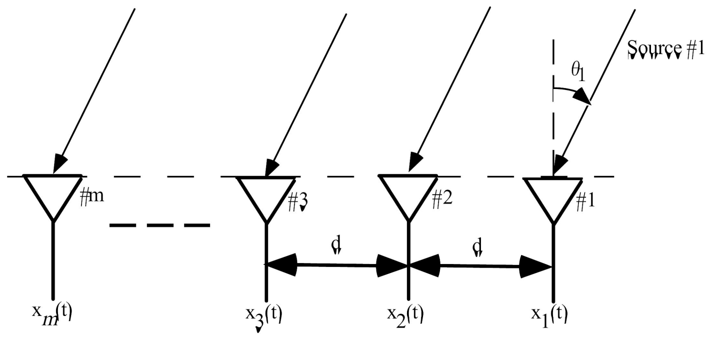

Figure 1: Linear array (a sound source is drawn only).

2.2 - MUSIC algorithm

MUSIC algorithm is shown as the following procedure.

• Collect signal data $\mathbf{X}$ by a linear array microphone.

• Calculate $\mathbf{R}_{XX}$ that is covariance matrix of signal data $\mathbf{X}$ is shown in Equation 2.

$$\mathbf{R}_{XX} = E\left[\mathbf{X}(t)\mathbf{X}^H(t)\right] = \mathbf{AR}_{ss}\mathbf{A}^H + \sigma^2\mathbf{I} \tag{2}$$

where

- $\mathbf{A}^H$: transpose of $\mathbf{A}$ matrix
- $\mathbf{R}_{SS} = E\left[\mathbf{S}(t)\mathbf{S}^H(t)\right]$
- σ^2: power of measurement noise
- $\mathbf{I}$: identity matrix

• Solve for the eigensystem of $\mathbf{R}_{XX}$ and estimated value is expressed by $\hat{\mathbf{R}}_{XX}$.

$$\hat{\mathbf{R}}_{XX}\bar{\mathbf{E}} = \bar{\mathbf{E}}\mathbf{\Lambda} \tag{3}$$

where

- $\mathbf{\Lambda} = \mathrm{diag}\{\lambda_1, \lambda_2, \ldots, \lambda_m\}, \lambda_1 \geq \lambda_2 \geq \ldots \geq \lambda_n = \ldots = \lambda_m = \sigma^2$
- $\bar{\mathbf{E}} = [\mathbf{e}_1, \mathbf{e}_2, \mathbf{e}_m]$

• Estimate the number of sound sources.

It is possible to estimate the number of sound sources by using Akaike Information Criteria method or Minimum Description length. In this paper, the number of sound sources is assumed to be known.

• Evaluate MUSIC spectrum shown in Equation 4.

$$P_M(\theta) = \frac{\mathbf{a}^H(\theta)\mathbf{a}(\theta)}{\mathbf{a}^H(\theta)\mathbf{E}_N\mathbf{E}_N^H\mathbf{a}(\theta)} \tag{4}$$

where $\mathbf{E}_N = [\mathbf{e}_{n+1}, \mathbf{e}_m]$.

• Find peaks from MUSIC spectrum and the angle correspond to the peak value is DOA of sound sources.

3 - EXPERIMENT

The experimental system is shown in Figure 2. The following experiments are carried out in an anechoic chamber (Length=7.3m, Width=3.5m and Height=3.5m). The number of observed signal data is 4096 and the others experimental conditions are shown in section 3.1.

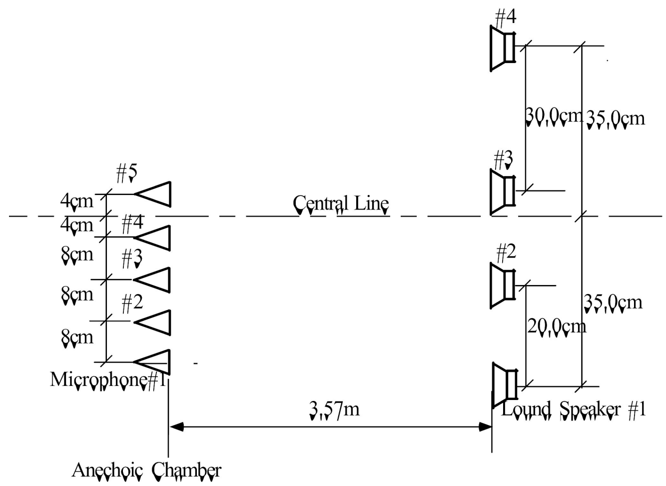

Figure 2: Experimental system.

3.1 - Experimental conditions

The number of sound sources is 4, central frequency is 2520.0Hz, 1000.1Hz, 1587.5Hz and 2000.1Hz respectively. The sound sources were located at 3.57m over from the linear array microphone in the experiment. The space between sound source #1 and sound source #2 is 10.0cm, sound source #2 and sound source #3 is 20.0cm, and sound source #3 and sound source #4 is 10.0cm. The DOA of sound source #1, #2, #3 and #4 is 1.12(deg), -2.08(deg), -5.28(deg) and -10.01(deg) for the linear array microphone.

3.2 - Experimental results and discussion

In figure 3, estimated results by using MUSIC algorithm are shown. The DOA of sound source #1 is 1.0(deg) approximately and estimated error is 0.12(deg), estimated errors of the others exceed 15.0(deg) due to microphone gain and phase errors in the experiments and so on. It is necessary for correction on microphone of array microphone in gain and phase before apply MUSIC algorithm to estimate DOA of sound source. We are discussing on correction methods about microphone gain and phase errors.

4 - CONCLUDES

As above described, DOA estimation methods have been applied widely in sensor systems [3]. In this paper, we introduced MUSIC algorithm for estimating a direction of sound sources. Although the experimental results show that it is possible to estimate a direction of 4 sound sources, it is necessary to correct errors of microphones in phase and gain.

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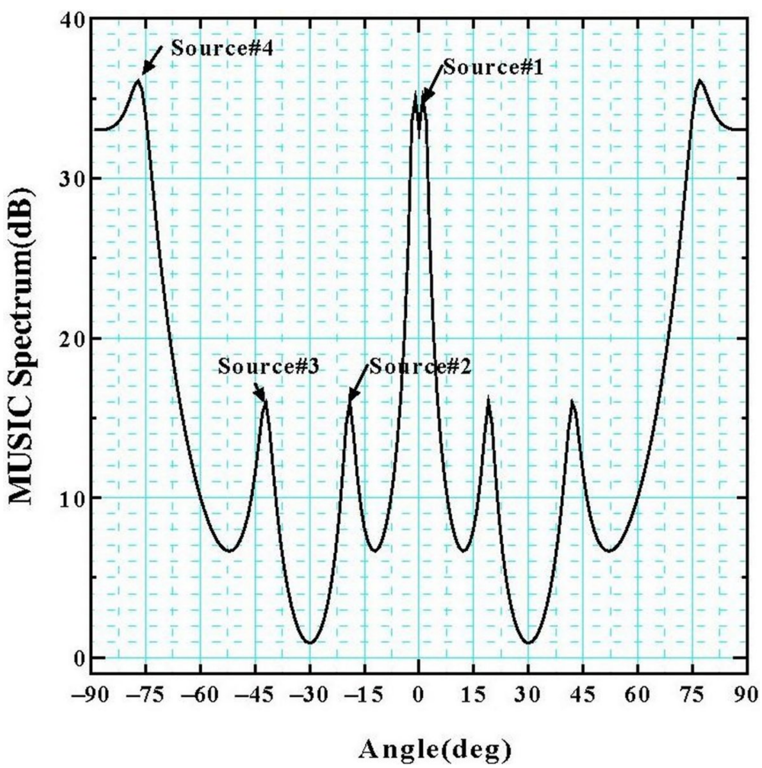

Figure 3: Estimated result (MUSIC spectrum).