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ENHANCED PRECISION IN SOURCE LOCALIZATION BY USING 3D-INTENSITY ARRAY MODULE

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ABSTRACT

The sound intensity technique has advantages of using a small sensor spacing and small number of sensors compared to the other source localization methods. Particularly, a tetrahedral array module for measuring 3D sound intensity can be used for the omnidirectional detection of the sound source. However, it yields a considerable error, fluctuating in spectrum, in estimating the bearing angle. Preliminary test suggests that the cause is the boundary reflection. In this study, the reflected signal from a known rigid wall is analysed by considering the cross-correlation function as a series of impulse functions. Consequently, intensity can be calculated by using the cross power spectral density function (CPSD) expressed as a series of sine functions, among which the high amplitude components invoke the spectral fluctuation. An experiment is conducted to verify the method with reflective panel and two microphones. The result indicates that the fluctuating spectral pattern is proportional to the difference of distance between direct and reflective paths. By compensating such spectral component of CPSD in the measurement by a 3D-intensity module, composed of four microphones with 30 mm spacing, it is shown that the localization errors of all bearing angles are less than 1° in an anechoic chamber.

1 INTRODUCTION

Studying on the typical array methods for sound source localization, as like beamforming, TDOA, sound intensity techniques, one can find that the localization performance depends on the arrangement of sensors in the array, the spacing between adjacent sensors, the overall size or span of the whole array, and the frequency range of the source sound. In comparison with other methods, the sound intensity technique has an advantage in the localization error when the distance between adjacent sensors is small. In the previous works, to measure the sound intensity vector indicating the three-dimensional direction angle, a compact localization system using 4 microphones arranged in a tetrahedral configuration or a system incorporating double tetrahedral array configuration have been suggested [1, 2]. However, the problem related with the fluctuating bias errors in frequency and spectral domains have been the great difficulty to solve. To overcome the spectral fluctuation of the measured 3D sound intensity, the use of octave or 1/3-octave band averaging has been suggested [2], but it cannot solve the problem intrinsically and has yet left intensity fluctuation. In this paper, the measured cross power spectral density functions (CPSD) between sets of two sensors are approximated by a function, and the function is used to compensate such bias error due to the fluctuation of measured intensity spectrum.

2 METHODOLOGY

2.1 Cause analysis of the spectral fluctuation phenomenon

If the measurement proble utilizes a configuarion of 4 microphones arranged in the tetrahedron [3], azimuthal and elevation angles, indicating the source bearing angle, are calculated by using the measured 3D sound intensity which is determined by the matrix combination of CPSD of signals of the sets of microphone pairs [4]. However, the estimated bearing angles exhibit bias errors in the form of spectral fluctuation in narrow frequency band as shown in Fig. 1. From the preliminary experimental study, it is found that such phenomenon is caused by the interaction of incident and reverberant sound appearing as the unwanted peaks in cross-correlation function as depicted in Fig. 2. Because the cross-correlation function can be expressed as a series of impulsive functions the CPSD can be expressed in Fourier transform pair of cross-correlation function as

$$\begin{aligned} G_{12}(f) &= A_0 e^{j2\pi f\tau_0} + \sum_{n=1}^{k} A_n e^{j2\pi f(\tau_0-\tau_n)} + \sum_{n=1}^{k} A_n' e^{j2\pi f(\tau_0+\tau_n)} \\ &= A_0 \cos 2\pi f\tau_0 + \sum_{n=1}^{k} A_n \left\{\cos 2\pi f(\tau_0-\tau_n) + \cos 2\pi f(\tau_0+\tau_n)\right\} \\ &\quad + j\left[A_0 \sin 2\pi f\tau_0 + \sum_{n=1}^{k} A_n \left\{\sin 2\pi f(\tau_0-\tau_n) + \sin 2\pi f(\tau_0+\tau_n)\right\} \right]. \end{aligned}$$

(1)

Here, A_n, A_n' denote amplitude of cross-correlation functions, τ_0 is the time difference between two microphones, G_{12} indicates CPSD of sensor 1 & 2, and $A_n \cong A_n'$. In Eq. (2), the spectral fluctuation phenomenon is expressed by the series of harmonic functions.

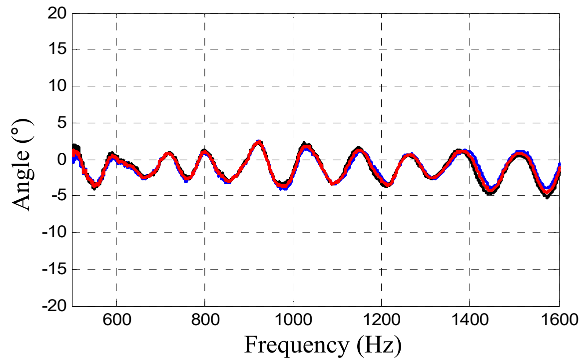


Fig. 1. Spectral fluctuation of the measured sound intensity magnitude in localizing a source located at 0°. This result is obtained by using the twisted-double sound intensity module [2].

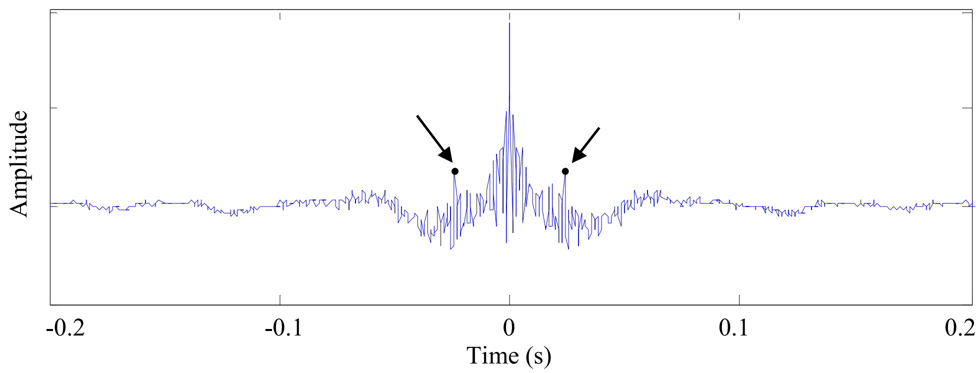


Fig. 2. Unwanted peaks in cross-correlation function of two microphones signals caused by the interaction between incident and "reverberant" sound. "←" indicates the peak component of concernt.

2.2 Error compensation

To compensate for the bias error caused by the spectral fluctuation in frequency domain, the correlated component between direct and reflected sound signals in the cross-correlation function should be removed. To this end, the filtering of cross-correlation function with a digital filter can be adopted. A low-pass filter to eliminate the unwanted correlation component is employed and the filter cut-off time is defined as $\tau_c = d/c$, in which d means the distance between microphones, and c the speed of sound. The designed example of the filter is illustrated in Fig.3.

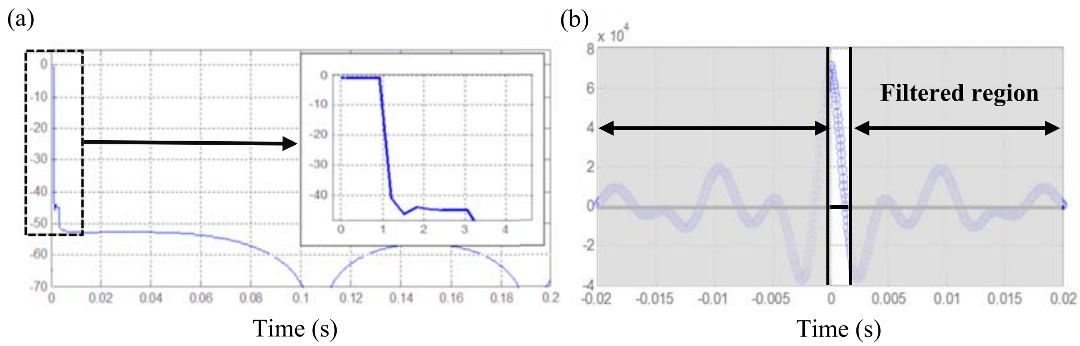


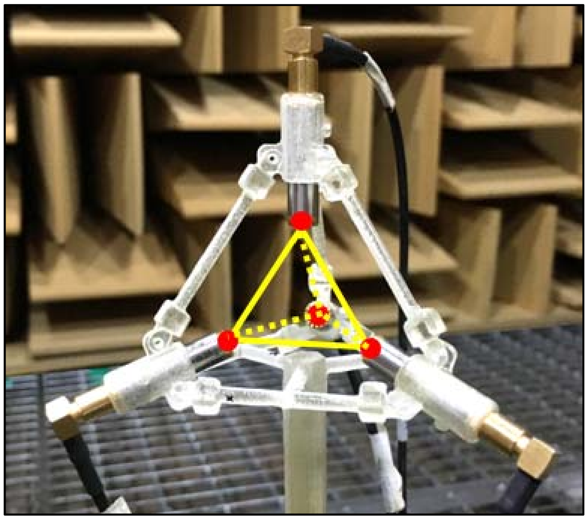
Fig. 3. (a) Designed filter for compensating the spectral fluctuation of sound intensity, (b) filtered cross-correlation function.

3 EXPERIMENTAL VALIDATION

3.1 Measurement set up and data processing condition

Experiments are conducted in a full anechoic chamber by using 4 quarter-inch microphones (B&K 4958) arranged in a tetrahedron to verify the effect of bias error compensation on the performance of the sound source detection. The effective frequency range is determined to be 0.500-4.55 kHz considering the distance between adjacent microphones [5], 30 mm, and the cut-off frequency of the anechoic chamber (~150 Hz), and the cut-off frequency of the speaker (~500 Hz). Sound pressure is measured for 5 s through data acquisition unit (NI-cDAQ-9178 and NI-9234), in which the sampling frequency is 25.6 kHz, and the data is averaged by 20 times with applying the Hanning window. The white noise is used as input signal, which is radiated through an amplifier (B&K 2734) and a horn driver (Selenium D250-X) positioned 1 m apart from the probe. For the measurement of angle difference between the laser-measured and probe-estimated localization results, a system of encoder and digital protractor (Bosch GLM80) is employed. The system configuration for the measurement is depicted in Fig. 4.

(a)



(b)

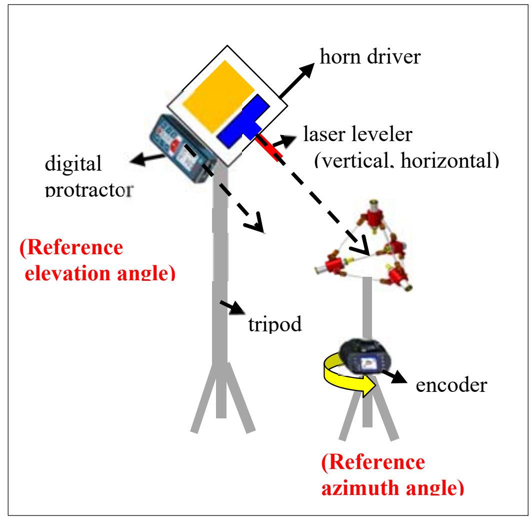


Fig. 4. (a) Sound intensity probe composed of four microphones arranged in a tetrahedron, (b) system configuration for measuring of reference bearing angle.

3.2 Experimental result

The mean localization error for azimuth or elevation angle is given by

$$\Delta\varepsilon_\theta = \frac{\sum_{f=f_L}^{f_H} \left|\theta_e(f) - \theta_0\right|}{f_H - f_L + 1}, \quad \Delta\varepsilon_\phi = \frac{\sum_{f=f_L}^{f_H} \left|\phi_e(f) - \phi_0\right|}{f_H - f_L + 1}, \tag{2, 3}$$

where θ_0, ϕ_0 denote the reference, i.e., laser-measured, azimuth and elevation angle, θ_e, ϕ_e the estimated, i.e., probe-measured, azimuth and elevation angle, and f_H, f_L the upper and lower frequency limit, respectively. Although the bias error related with the spectral fluctuation of measured intensity magnitude occurs in all experimental results, it is thought that the reduction of such error is possible by adopting the compensation method explained in the preceding chapter. Figure 5 illustrates the experimental results . The thin solid line indicates the estimated angle spectrum without compensation, and the thick solid line signifies that with compensation by using the filtered CPSD, while the thick dotted line shows the reference angle data measured by the laser and angle-protractor instrument system. One can find that, after compensation, the errors in azimuth and elevation angles are diminished to the values less than 1° in the anechoic chamber.

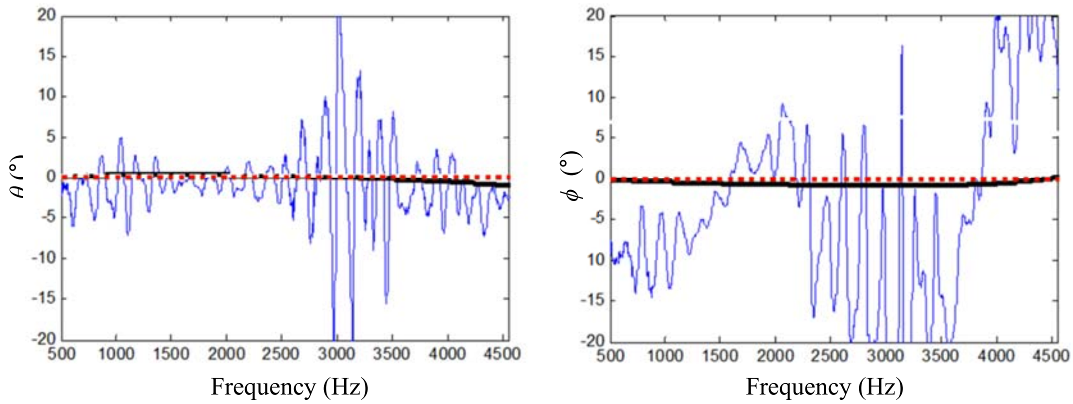


Fig. 5. Experimental result for localizing a white noise source positioned at a reference angle. ·······, measured reference angle by laser system; ——, initial localization result without any compensation; ——, compensated result by adopting the filtered CPSD. (a) Azimuthal angle ($\theta_0 = 0°$), (b) elevation angle ($\phi_0 = 0°$).

4 CONCLUSIONS

The purpose of this study is to establish a compensation method for the bias error related to the spectral fluctuation in the measured intensity value by identifying the physical nature causing such phenomenon in using the three-dimensional intensity probe. Mathematical analysis of cross power spectral density measured between sets of two microphones in the probe, being used for calculating the sound intensity at each direction, it is found that the spectral fluctuation phenomenon is caused by the correlation between incident and the reflected, or in general, reverberant, sounds. A compensation method for this kind of bias error is suggested, which applies a low-pass IIR filter to the measured cross-correlation function. To validate the effectiveness of the suggested compensation method, the experiments are conducted in an anechoic chamber with a source radiating the white noise signal, and the 3D intensity measurement probe in a tetrahedral configuration of microphone arrangement with 30 mm in microphone spacing. In the experimental results, the estimated bearing angles for the sound source localization, without any compensation, bear errors larger than 10° due to the bias error related to the spectral fluctuation of measured intensity magnitude. It is noted that, after compensating the bias error adopting the filtered cross-power spectral density, the localization errors in terms of bearing angles are dramatically reduced to be less than 1° within the effective frequency range set as .5-4.55 kHz in this work.

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