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The Study of Acoustic Source Localization using a Small Microphone Array for Condition Monitoring

THE STUDY OF ACOUSTIC SOURCE LOCALIZATION USING A SMALL MICROPHONE ARRAY FOR CONDITION MORNITORING

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ABSTRACT

The aim of this research is to investigate microphone array techniques based condition monitoring (CM). In this paper, a cost effective microphone array configured with 4-sensors and data acquisition systems is investigated theoretically and experimentally to evaluate its performance in characterising sound sources in different acoustic environments. Time delay estimation (TDE) is used as a basic tool for calculating the time delay of the sound source arrived at each of the 4-sensors and hence the localization parameters of the sound source such as azimuth, elevation and distance can be calculated from the TDE information. Results are presented for experimental studies both in a semi-anechoic chamber, an outdoor environment and an enclosure. Comparing the results obtained under different conditions, it has confirmed that the microphone array system can catch the location of the main sound source and also has ability to track and give the azimuthal trajectory of a moving target with high degree of accuracy.

Keywords: sound localization, microphone array, time delay estimation

1 INTRODUCTION

Recently, microphone array technologies have develop rapidly for various applications including environmental monitoring, object detection and location and speech recognition. Measurement using microphone array has become very vital for investigating the acoustic field in many experimental environments, both in an open space measurement [1] and an enclosure [2] like in cabin [3]. In particular, a passive acoustic location technology based on the properties of sound propagation is gaining popular in acoustical measurements because of its advantages of good permeability, good stability and richness of information included in acoustic signals.

In condition monitoring, comparison with the dominative techniques of vibration based monitoring, acoustic CM has the potential to become a generic approach because it has a number of unique features such as generality of acoustic signals in the majority of machines, the richness of information in acoustic signals and simplicity in sensor placement. Latest works [4-10] have demonstrated a number of outstanding capabilities of acoustic CM on engines, electric motors, pumps and gear transmission when the acoustic signals were processed along with new enabling techniques such as time-frequency-analysis, adaptive filtering, wavelet transforms, independent component analysis (ICA), self-encoding techniques etc.

Although there have been considerable progress in acoustic CM, accurate characterization of acoustic sources in reverberant fields (i.e. industrial environments) is still a primary obstacle in promoting CM performance. Recent advances in array measurement technology provide great potential to overcome the problem as a multiple-sensor based array measurement can produce a much more detailed picture of acoustic sources than a single microphone does. So it is possible to extract more detailed information from the microphone array measurement for condition monitoring and hence achieve a higher accuracy of detection and diagnosis.

This paper is to investigate the location information of acoustic sound sources and try to track the main sound source of a moving target using a small microphone array. Time delay estimation method is employed to calculate the delay time of the sound propagated to each of the 4 sensors in the array. Based on the delay time, the location parameters of the azimuth, elevation and distance are estimated for the purpose of detection. A mathematic model of microphone array is also presented to show how the location information can be calculated from the array geometry with the local 4-sensor microphone array. In the mean time, the performance of this array is evaluated in three typical acoustic environments: in a semi-anechoic chamber, an outdoor environment and an enclosure with higher reverberation.

2 MATHMATIC MODEL FOR SPACE ARRAY ORIENTATION

The location parameters of the source illustrated Figure 1(a) can be obtained based on the model which has M-sensor used for space array orientation [11]. Assuming that there is a single band limited acoustic source P whose location is denoted by azimuth φ and elevation θ, the distance r_i between the source and the array is denoted in the coordinate system as:

$$r_i = [x_i, y_i, z_i] \tag{1}$$

Assume the 1th sensor is the reference sensor, and then the time delay vector τ can be defined as:

$$\tau = [\tau_{12}, \tau_{13}, \cdots \tau_{1j}, \cdots, \tau_{1M},] \tag{2}$$

Where, τ_{1j} donated the time delay of the sound propagation between the j:th sensor and 1th sensor.

The direction vector in space is defined as:

$$k = \begin{bmatrix} k_x \\ k_y \\ k_z \end{bmatrix} = \begin{bmatrix} \sin\theta\cos\varphi \\ \sin\theta\sin\varphi \\ \cos\theta \end{bmatrix} \tag{3}$$

Therefore, the time delay can be denoted as:

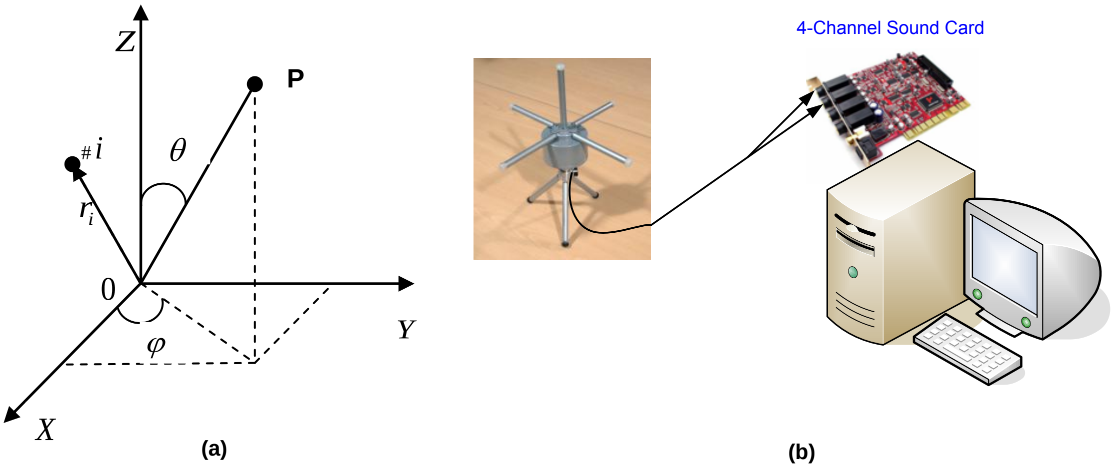
$$\tau = -\frac{R \cdot k}{c}; \qquad R = \begin{bmatrix} r_2 - r_1 \\ \vdots \\ r_M - r_1 \end{bmatrix} \tag{4}$$

Where, c is the speed of sound, R is the distances between each sensor and the reference sensor. From equation (4), it can be concluded that direction parameter k can be calculated from equation (5) after calculating the time delay estimation$\hat{\tau}$.

$$\hat{k} = -c(R^T R)^{-1} R^T \hat{\tau} \tag{5}$$

Thus if the sensors of the array are distributed in three-dimensional space, the azimuth and elevation of single sound source P can be calculated by:

$$\begin{cases} \hat{\varphi} = \tan^{-1}(\hat{k}_y/\hat{k}_x) \\ \hat{\theta} = \tan^{-1}((\hat{k}_x^2 + \hat{k}_y^2)/\hat{k}_z) \end{cases} \tag{6}$$



(a) (b)

Figure 1: (a) Mathematic model of space array and (b) Microphone array location system

For the moving target, its space location can be tracked by estimating the azimuth and elevation of the sound source in a specified time interval.

3 CROSS-CORRELATION TIME DELAY ESTIMATION ALGORITHM

As shown in section 2, the estimation of time delay is a critical step in source localisation. If the signals $x_1(t)$ and $x_2(t)$ are measured by two different sensors M_1 and M_2 is from the sound source $s(t)$ [12], there is a time difference between these two signals because the two sensors' location is different. The difference as τ_0 is known as the time delay that needs to be estimated properly because signals contain evitable noise. Assuming that $n_1(t)$ and $n_2(t)$ are additive noise and $n_1(t)$, $n_2(t)$ and $s(t)$ are independent with each other, the signal models for the two sensors received can be expressed as:

$$\begin{cases} x_1(t) = s_1(t) + n_1(t) = s(t) + n_1(t) \\ x_2(t) = s_2(t) + n_2(t) = s(t - \tau_0) + n_2(t) \end{cases} \tag{7}$$

Cross-correlation function is usually employed to estimate the time delay between two related signals. The cross-correlation function of signal $x_1(t)$ and $x_2(t)$ is:

$$R_{12} = E[x_1(t)x_2(t+\tau)] = R_{ss}(\tau - \tau_0) + R_{sn_1}(\tau - \tau_0) + R_{sn_2}(\tau) + R_{n_1n_2}(\tau) \tag{8}$$

In equation (8), R_{ss} is the autocorrelation function of $s(t)$, R_{sn_1} is the cross-correlation function of $s(t)$ and $n_1(t)$, R_{sn_2} is the cross-correlation function of $s(t)$ and $n_2(t)$, and $R_{n_1n_2}$ is the cross-correlation function of $n_1(t)$ and $n_2(t)$. Because of $n_1(t)$, $n_2(t)$ and $s(t)$ are independent with each other, so the noise related cross-correlation:

$$R_{sn_1}(\tau - \tau_0) = R_{sn_2}(\tau) = R_{n_1n_2}(\tau) = 0 \tag{9}$$

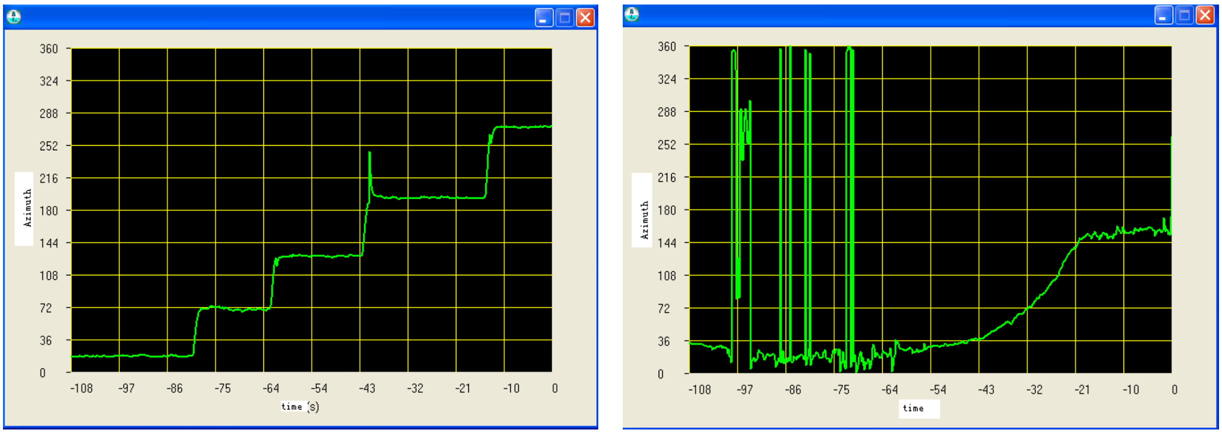
Therefore, the cross-correlation between the signals:

$$R_{12} = R_{ss}(\tau - \tau_0) \tag{10}$$

As the maximum value of the cross-correlation function is appeared when $\tau - \tau_0 = 0$, so the time delay value is equate to the time according to the peak of the cross-correlation function [13].

4 EXPERIMENTAL STUDY AND RESULTS

To evaluate the performance of sound source location system, tests are carried out in three different acoustical environments: a semi-anechoic chamber, an open space and a car cabinet respectively. The small microphone array used in the experiments is illustrated in figure 1(b). The aperture of the small array is 20mm and the frequency response range of microphones is from 20Hz to 20 KHz. A multiple channel sound card collected the signals from the 4 microphones simultaneously into the host PC which implements the algorithms in section 2 and 3 in real time and present the results in different styles of graphs.



(a) **(b)**

Figure 2: Location results in (a) semi-anechoic chamber and (b) open space

Firstly, the experiment is carried out in the semi-anechoic chamber. The result of single sound source localization is shown in figure 2(a). It can be seen that the azimuth of the sound source can be estimated accurately because the background noise level is lower in the semi-anechoic chamber. In the experiment, the location of the sound source is determined, but the microphone array is rotated every few seconds. It simulated that the sound source is moving around the microphone array. Figure 2(a) demonstrated that the azimuth of the sound source is changed according to the moving target.

In the open space, the microphone array system is used to measure the sound of a passenger car. The car is running from one side of the array system and then across it to the other side.

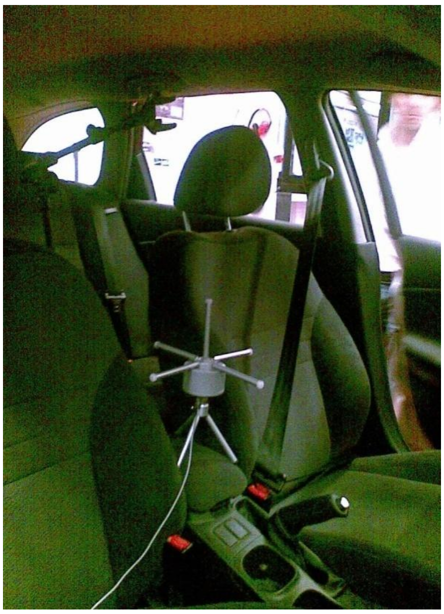


Figure 3 Array setup in car cabinet

Figure 2(b) is the azimuth estimated result. It is clear that the accuracy is high when the car is near the microphone array when the signal to noise ratio (SNR) is high but when the car is far from the system and with some disturb by some other sound sources, the accuracy of location result is low and shows some high jumps in the estimation results.

To evaluate the array performance, it has been tested in a car cabinet to detect abnormal sounds such as squeaks etc. The abnormal noises generated during the driving of car highly influence the conformability of driving and could also be the symptoms of incipient faults. In practice, it is often very difficult to eliminate the noises due to the fact that these noises sometimes can only be heard under certain driving conditions and only experienced engineers can effectively localise the noise source. During the test typical recorded sounds are reproduced through a speaker that can be moved to different positions such as underneath of a seat or in the booth and the speaker is then localized using the array system. Figure 3 shows the setup of the array in the cabinet. A typical location result is shown in Figure 4, which shows that this speaker is found in the passenger side indicated by the red dot in the both the two-dimension and three-dimension graphs. In the mean time the A-weighted noise level and raw signals is also displayed for noise intensity indication. In general, detection can achieve more than 90% correct rate when the speaker is placed in different positions, which is still not high enough for real applications because of the high acoustic reverberation in the cabinet.

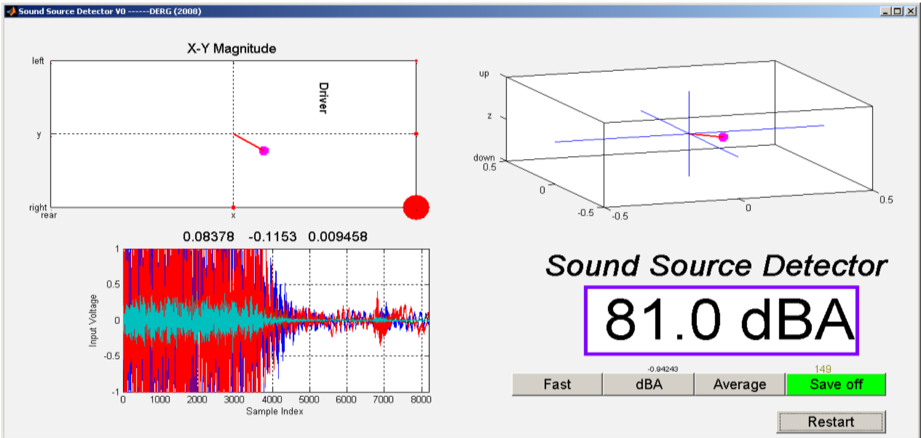


Figure 4: Location result in car cabinet

5 CONCLUSION

A cost effective microphone array configured with 4-sensors and sound card data acquisition system has been studied in this research. In theory, time delay estimation using cross-correlation can be based on for calculating the time delay of the sound source arrived at each of the 4-sensors and hence the localization parameters of the sound source such as azimuth, elevation and distance. Results are presented for experimental studies both in a semi-anechoic chamber, an outdoor environment and an enclosure. Comparing the results obtained under different conditions, it has confirmed that the microphone array system can catch the location of the main sound source and also has ability to track and give the azimuthal trajectory of a moving target. However, more research has to be conducted on the direction of reverberation cancellation for more accurate localization.

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