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Accurate Position Detection of Sound Source by LabView

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ABSTRACT

This paper introduces a method to detect an accurate position, which includes its distance and direction, of a sound source based on a new T- type microphone array having four sound sensors in a LabView development environment. The T-type microphone array consists of three linear microphones in the front side and one in the center of the rear side. Existing linear microphone array types have difficulties in detecting a sound source which exists in the back side of the array. However the type proposed in this paper makes omnidirectional detection possible. The T-type also estimates a source distance by comparing two pressure levels from a front sound sensor and a rear sound sensor respectively. The reduction rate of a sound pressure level is used to calculate the source distance. In general, the direction of a sound source is estimated by calculating the time delay between separate sensors, but the technique proposed in this paper uses both the time delay approach mentioned above and the geometrical method with triangular surveying relations for a higher probability value to obtain the high level accuracy of the direction calculation. Sound-related hardware design requires complex signal processing techniques with a difficulty of desired algorithm verification. In this case, LabView can be used to help intuitive PC-based programming and effective algorithm verification.

1. Introduction

Sound source localization has been used as a main method for determining the position of a sound source to be applied to various fields such as military surveillance to detect flying objects or armored vehicles and a humanoid robot auditory system. In sound source localization, microphone array signal processing is one of the most applicable techniques. The primary goal of microphone array signal processing is to enhance and extract information carried by acoustic waves received from a number of microphones [1]. Based on this method, we can obtain sound source information for sound localization about level density and time delay.

In this paper, we propose an efficient hardware design method for determining sound source position. In general microphone array system has a difficulty in hardware design because of signal processing and analysis for many inputs. In addition, it is not easy to consider algorithm relevance for the signal processing. To overcome the limitations mentioned above for hardware development, we use a LabView developing environment which will allow easy verification of signal processing and convenient adaptation to hardware design process. The T-type microphone array proposed in this paper has four sound sensors consisting of three in the front side and one in the center of the rear side. The three sound sensors are used for both detection of a sound source and enhancement of noise robust. One in the rear side is used for extension of direction range to make omnidirectional detection possible. In general, it is not easy to get position information at rear side when a linear microphone is just used. In this paper, we consider a method which combines a triangular surveying technique for near field sound and a sound pressure-level reduction rate technique for far field sound to estimate accurate distance of a sound source.

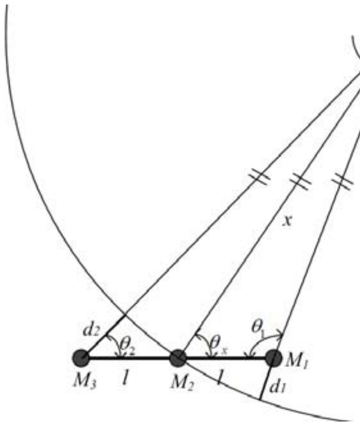
The paper is organized as follows; Section2 explains the principles about time delay and distance estimation of a sound source. Section3 presents a design method using a LabView environment and shows experimental results compared with actual position.

2. Fundamental principle for sound source position estimation

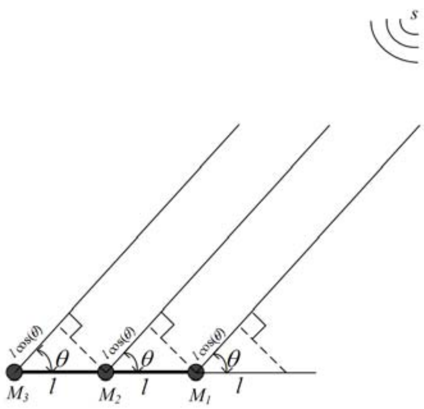
Sound source localization method can be accomplished actively or passively depending on the situation. In the active system, a pulse is transmitted to the target and then an echo is received. In the passive system, a signal generated by the target is considered for the accomplishment [2]. In this paper, we consider the passive system for the estimation of target distance and direction, in which the accuracy of the time delay is critical to that of direction estimation.

2.1 Sound source direction estimation

In order to estimate accurate sound source position, we need to consider near and far field models separtely. The difference of geometrical configuration between the near field model and the far field model can be seen in Figure 1. The near field model as shown in Fgure 1.(a) is applied when the distance between a sound source and a far microphone is simlar or shorter than the distance between two microphones. On the other hand, the far field model is used when a sound source is far apart from the microphone array. In this case, we don't need to consider gaps between microphones as shown in Figure 1.(b) [1].



(a)



(b)

Figure 1.Geometrical configuration of three sound sensors: (a) near field model, and (b) far field model

In the Figure 1.(a), l indicates the distance between two near microphones and x is the distance from a sound source to the central microphone M_2. θ_x is the sound source target direction at the near field model [3]. d_1 and d_2 are distance differences from a radius of the central microphone (M_2) to M_1 and M_3, respectively . We can compute d_1 and d_2 as follows;

$$d_1 = c\cdot\tau_{12} \tag{1}$$

$$d_2 = c\cdot\tau_{23} \tag{2}$$

where τ_{12} and τ_{23} are time delays between M_1 and M_2, between M_2 and M_3, respectively, and c is the speed of sound which is assumed as constant (in m s^{-1})

In the near field geometrical model, following mathematical expressions are considered to estimate θ_x;

$$\cos\theta_1 = \frac{l^2+(x+d_1)^2-x^2}{2l(x+d_1)} \quad \left(\theta_1 = \cos^{-1}\frac{l^2+(x+d_1)^2-x^2}{2l(x+d_1)}\right) \tag{3}$$

$$\cos\theta_2 = \frac{l^2+(x+d_2)^2-x^2}{2l(x+d_2)} \quad \left(\theta_2 = \cos^{-1}\frac{l^2+(x+d_2)^2-x^2}{2l(x+d_2)}\right) \tag{4}$$

$$\cos(\theta_1+\theta_2) = \frac{2l^2-(x+d_1)^2-(x+d_2)^2}{2l(x+d_1)(x+d_2)} \tag{5}$$

Equations (3), (4), and (5) are expressions for triangles M_1M_2S, M_2M_3S, M_1M_3S in Figure 1.(a), respectively. Based on the expressions, we can drive x as follows;

$$x = \frac{l(l^2-{d_1}^2)+l(l^2-{d_2}^2)}{2l(d_1+d_2)} \tag{6}$$

And we can get as θ_x follows;

$$\theta_x \doteqdot 90^\circ - \frac{\theta_1}{2} + \frac{\theta_2}{2} \tag{7}$$

In the far field geometrical model shown as in Figure 1.(b), the following expression is considered to estimate θ;

$$\tau_{12} = \tau_{23} = \frac{l\cdot\cos\,\theta}{c} \tag{8}$$

where τ_{12}, τ_{23}, l and c have the same meaning as mentioned in the above equations (1),(2) and (6).

$$\theta = \cos^{-1}\left(\frac{c\cdot\tau}{l}\right) \tag{9}$$

We can get the value of θ if we just know τ which is based on cross correlation $R_{ij}(\tau)$ between two microphone input signal S_{Mi} and S_{Mj}. In this case, $R_{ij}(\tau)$ can be expressed as

$$R_{ij}(\tau) = \sum_{k=0}^{K} S_{Mi}(k)S_{Mj}(k-i) \tag{10}$$

$$\Delta T = \underset{\tau}{argmax}\, R_{ij}(\tau) \tag{11}$$

where ΔT is the time delay of arrival between two microphones. ΔT can be found by locating the peak in the cross correlation.

2.2 Sound source distance estimation

To estimate the sound source distance, we can use two methods like a sound-level reduction rate model and a trigonometric identity model mentioned above in the equation (6). The trigonometric method is used for the near field model while the sound-level reduction rate method is applied for a far field model.

The sound-level reduction rate method uses acoustics characteristics in which sound level intensity is reduced by 6dB if the distance of a sound source is two times reduced. In this case, we assume a first-order linear equation model like $Y=mX+b$ where X is $log_2 x$. In this equation, m is the slope of the first-order linear equation. Figure2 shows these characteristics, m can be expressed as follow;

$$m = \frac{S_r - S_f}{D_r - D_f} = \frac{S_r - S_f}{log_2 D_r - log_2 D_f} = \frac{S_r - S_f}{log_2 D_f + d - log_2 D_f} \quad (11)$$

$$D_f = \frac{d}{2^{\frac{S_f - S_r}{m}} - 1} \quad (12)$$

Where D_f and D_r are sound source distances from front-central and rear microphones to a sound source. d is the distance between front-cental and rear microphones.

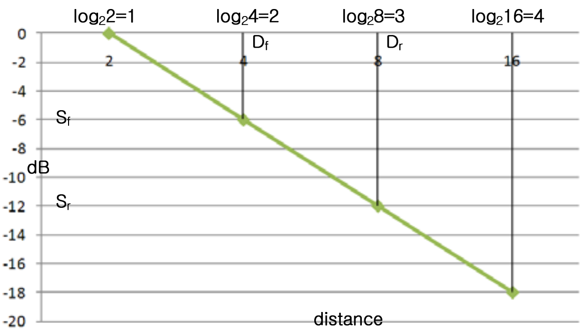


Figure 2.Sound-level linear prediction according to a target distance.

3. LabView-based sound acquisition and data processing

3.1 System design for LabView

The T-type microphone array proposed in this paper is shown with four microphones in Figure3. To input four-channel signals, two-line inputs with two PCI sound cards are used. These signal processing is performed in a LabVIEW environment.

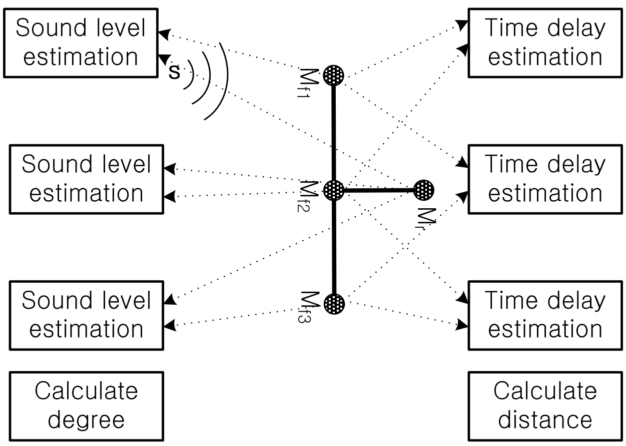


Figure 3.T-type Microphone array position and sound processing.

In the LabVIEW environment, the sound collection toolkit is used to input sound signals with block diagram window. These input signals passes a bandpass filter (100~5Khz) for cancelling noises with high frequency. Correlation factors between M_{f1} and M_{f2}, $M_{f2}$and M_{f3}, $M_{f1}$and M_{f3} are obtained based on cross correlation toolkit. And then statistics toolkit is used to get the time delay values between two microphones. This system proposed in this paper consists of four main blocks such as cross correlation blocks, sound level compare blocks, sound target direction calculate blocks, and distance calculate blocks. To increase the accuracy of position estimation, we consider the near and far field models separately. Figure 4 shows the overall configuration of this system.

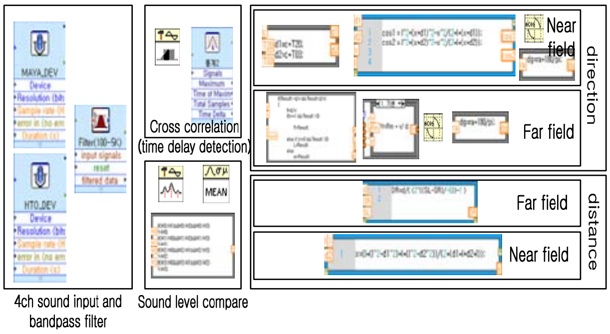


Figure 4.Main block in LabView environment.

3.2 Experiment results

In the experiment, we used the T-type microphone array having four microphones with 15cm gaps each. A sound codec is used for sound signal input and then processed through a digital filter using a FPGA chip. These processed signals are verified in the LabVIEW and displayed in the front panel. Figure 5 shows the hardware configuration for the system proposed in this paper.

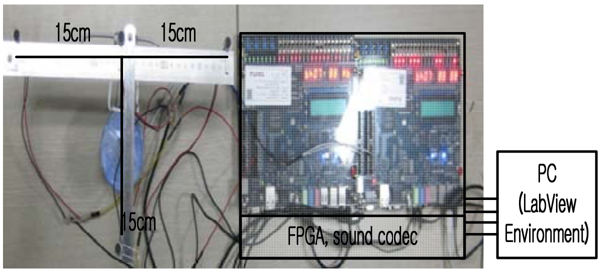


Figure 5.Hardware configuration for sound localization.

The experimental result gives 2Khz sinusoidal wave for 2m distance and 120° direction as shown with LabVIEW front panel in Figure 6. 4 LEDs on the left of Figure 6 indicate 4 microphones. The input with largest sound level among them turns on its corresponding LED. If MIC1 is turned on, there is a source signal generated from the back side of the microphone array. The middle shows sound source direction, which is currently 124°. The Right side shows filtered signals about 2Khz sinusoidal wave and source distance with 2m at red-pointing Y-axis.

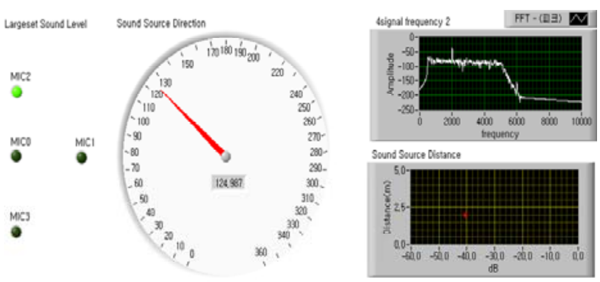


Figure 6.Experiment result in front panel (LabVIEW)

4. Conclusions

Conclusively, the goal of this paper is to get the two following achievements: First, the proposed T-type microphone array provides effectiveness and accuracy in detecting the sound source direction and distance in comparison with existing systems. Second, we show that LabView-based design makes easier and faster verification for desired algorithms before modelling the hardware for sound localization. As a further study, the last goal of this system is to achieve an actual hardware design based on the method proposed in this paper.

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