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IMPLEMENTATION OF DIRECTIONAL CHARACTERISTICS BY REAL-TIME PROCESSING OF SOUNDS OBSERVED BY TWO MICROPHONES

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Abstract. *Focusing on the difference in sound powers observed by two microphones depending on the location of the sound sources, we propose a method of forming a sharp directivity by two microphones. The implementation of directional characteristics by real-time processing of sounds observed by two microphones was carried out using mbed microcontroller.*

Keywords: Directional microphone, Real-time processing, Joint distribution, mbed

1. **Introduction.** Conventional microphone technique is by forming a super-directional microphone using a delay-and-sum beamformer or an adaptive microphone-array, and it can record a target sound with low noise [1]. However, a large number of microphones are required in order to sharpen the directivity, and it becomes expensive. In our recent study, it is shown that the estimation of sound source was possible by statistical processing using the power difference of the sound recorded by the plurality of microphones [2]. Moreover, focusing on the difference in sound powers observed by two microphones depending on the location of the sound sources, we have proposed a method of forming a sharp directivity by two microphones. The validity of the method was shown by the computer simulation [3]. This method can be very easy to implement by digital processing, and has a feature that can record target sound at real time.

In this study, the implementation of directional characteristics by real-time processing of sounds observed by two microphones is presented for validation of the theory. The directional microphone is realized by very simple calculation in consideration of the joint distribution of the sound source using mbed microcontroller. The directional characteristics and dependence of the distance between sound source and microphones are also presented.

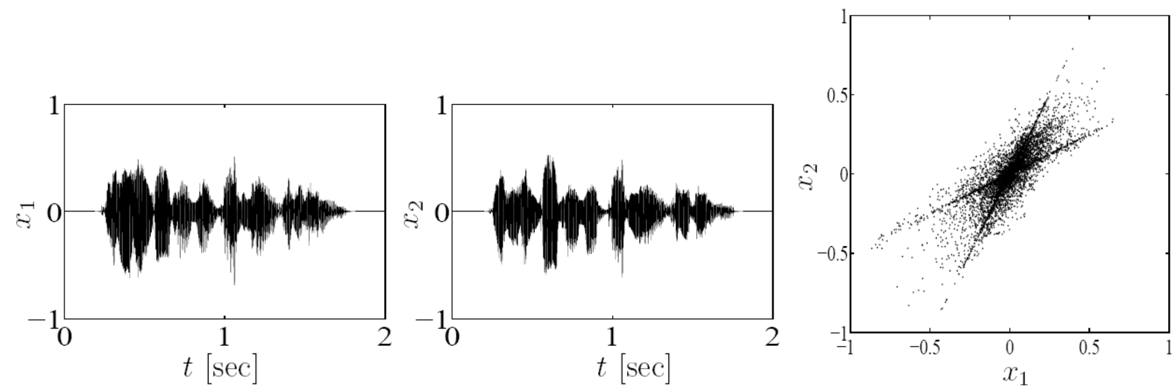
2. **Emphasis of the Target Speech.** Microphone array processing is a technique for creating a directivity in the target signal direction based on the arrival direction of the desired sound or noise. When N speakers are speaking at the same time, the speaker's voice $s_n(t)$ $(n = 1, 2, \ldots, N)$ are observed by M microphones. Observed speech $x_m(t)$

$(m = 1, 2, \ldots, M)$ is expressed as

$$x_m(t) = \sum_{n=1}^{N} a_{mn} s_n(t) \tag{1}$$

Here, a_{mn} are unknown transfer functions corresponding to the damping coefficients from the original signals to the microphones when the influences of the time delay and reflection are ignored. The estimation methods of the original signal from the observed signal have been proposed by some kinds of algorithms including the independent component analysis (ICA). It is known that the estimation is possible when the number of original signals and that of microphones are the same [4,5].

Assuming that the probability distribution of the audio signal, the amplitude of the $x_1(t)$ for horizontal axis and the amplitude of $x_2(t)$ for vertical axis are plotted as a joint distribution. Figures 1(a) and 1(b) show the waveform of the example observed signals and that of the joint distribution, respectively.



(a) Waveforms (b) Joint distribution

Figure 1. Waveform of the example observed signal and its joint distribution

The relations of the $x_1(t)$ and $x_2(t)$ are expressed as,

$$x_2(t) = \phi x_1(t) \tag{2}$$

Here, the ϕ denotes the slope of the joint distribution. By the weighting for the inclination $\phi = x_2/x_1$ of the joint distribution, the desired sound can be emphasized and the noise as another sound can be suppressed. By focusing on the damping ratio for transmission of two sound sources observed by two microphones, we proposed a noise suppression method based on the location of the sound sources [3].

Applying this method, directional microphone can be realized by receiving the target sound and suppressing the unexpected sound from different direction. Consider realizing the microphone just having a directivity in the front. When the sound source is present in front of the microphones, the distance from the sound source to each microphone is the same, and it becomes $x_2/x_1 = 1$ $(x_1 = x_2)$. On the other hand, when the sound source is not in front of the microphones, different damping ratios are from the sound source to each microphone, and it is expressed as $x_2/x_1 \neq 1$ $(x_1 \neq x_2)$. From the above facts, the error ε of slope by inclined sound at the joint distribution is defined by Equation (3),

$$\varepsilon = 1 - \frac{x_2}{x_1} \tag{3}$$

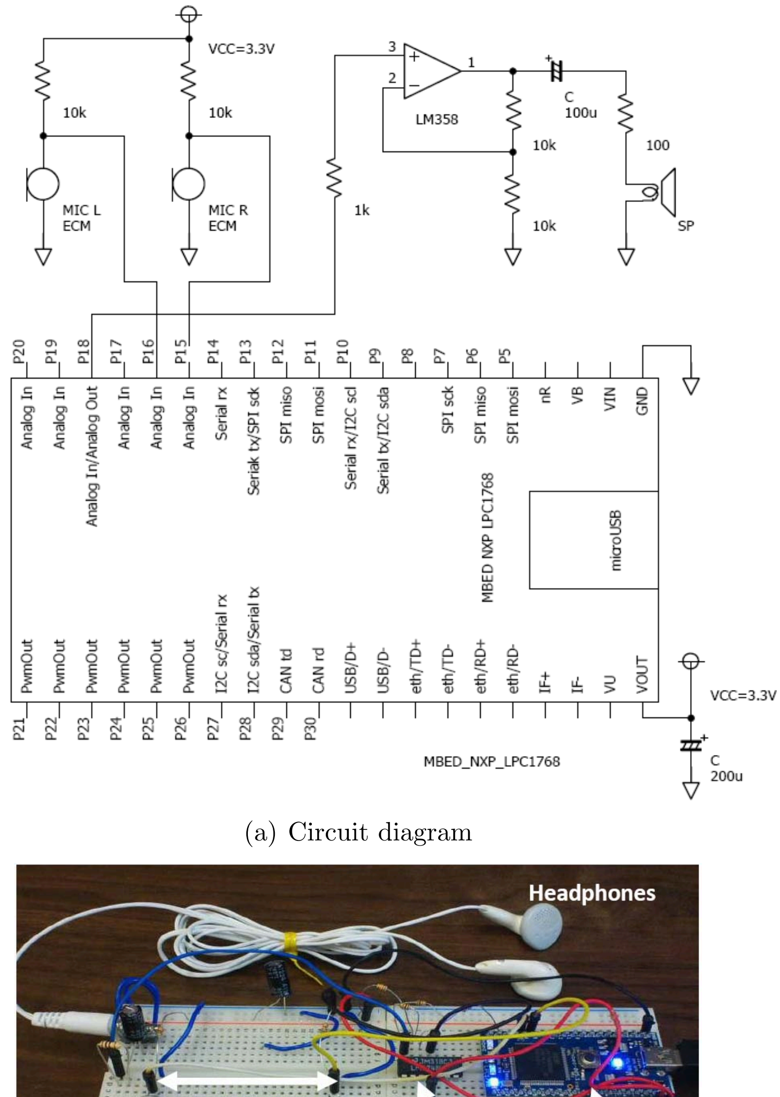
The output signal $y(t)$ having the directivity to the front is calculated using the ε,

$$y(t) = \frac{1}{1 + \alpha\varepsilon^2} x_m(t) \tag{4}$$

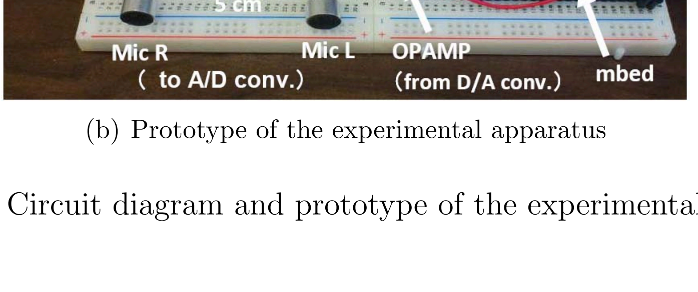
Here, the α is a parameter that determines the suppression factor of noise. Therefore, the sound in front of the two microphones with the same distance is emphasized and the sound form inclined direction can be suppressed. Even though using a conventional microcontroller, Equation (4) can be calculated in a sufficiently short time.

3. **Experiment.** The experimental circuit was fabricated using mbed microcontroller which is known as one of the rapid prototyping tools [6]. Figures 2(a) and 2(b) are the circuit diagram and photograph of the prototype of the experimental apparatus, respectively.

Two condenser microphones are used in the circuit, and the microphones are directly connected to the analog port of the mbed (12 bit A /D converter) in order to capture the left and right phase relations as faithfully as possible. The two channel audio signals convert 12 bit continuous data by 8 kHz sampling. The distance of the microphones is 5 cm, which is enough close distance in order to ignore the phase difference at the sampling



(a) Circuit diagram



(b) Prototype of the experimental apparatus

FIGURE 2. Circuit diagram and prototype of the experimental apparatus

frequency. The calculated signal is output continuously using the mbed built-in 10 bit D/A converter, and the signal is amplified by the operational amplifier and plays the headphones for monitoring the result.

4. **Results and Discussion.** Figure 3 shows the experimental result of the directional pattern at different distances of the reference speaker and microphone. The 1 kHz square wave is used for reference sound.

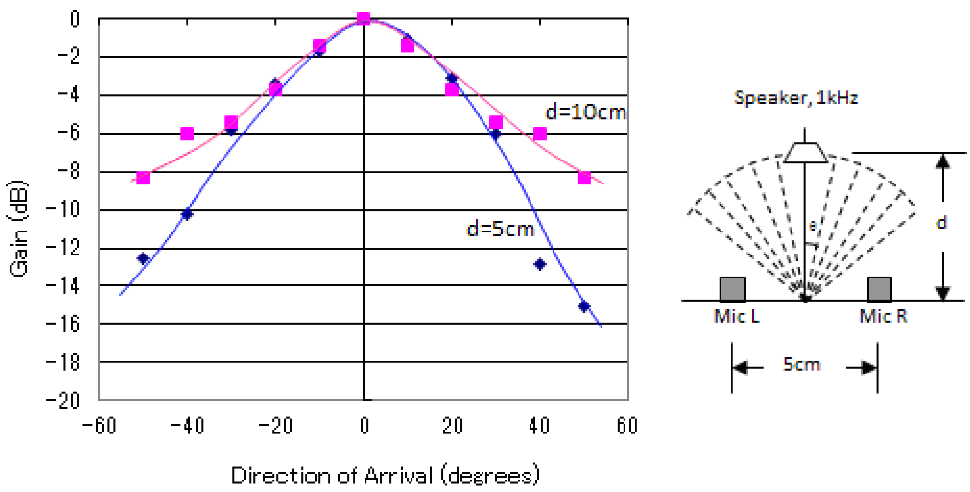


FIGURE 3. Experimental result of the directional pattern

Clear sound is obtained when speaking in front of the microphones at the equal distance of about 5 cm, and the output was reduced when the incident direction of the sound is inclined to the left and right. Even if the output is reduced, the noise is remained. The noise is considered to be due to a phase difference between the left and right sounds. In addition, as the distance of the sound source and the microphone is far away, directivity became unclear. This reason is the amplitudes of the x_1 and x_2 decreases by increasing the distance, and the difference between x_1 and x_2 becomes smaller.

5. **Conclusions.** We have presented the implementation of the proposed directional microphone using mbed microcontroller. It has advantages of modularization of the directional microphone without personal computer, and it would be possible to provide an inexpensive directional microphone module. In our future work, we will develop the directional microphone module by optimization of the distance of microphones and several parameters for calculation.

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