On a Estimation Method of Sound Source Directions Using a Reflection Board

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Synopsis

Recently, many methods using more than one microphone have been widely studied to estimate the sound source directions. Especially the methods using the time and frequency differences at both ears are well known. But we humankind can distinguish the various directions of sound sources with only one ear. Here a new estimation method of sound source directions by using only one microphone and a reflection board is proposed. By the board behind a microphone, we can get the variations of the signal which comes from the changes of the directions. This system collecting sounds imitates the man's auditory system with an auricle. The sound direction is estimated by recognizing the characteristics made by the reflection board.

KEYWORDS: sound source direction, reflection board, characterization, direction estimation,

1. Introduction

The sense of hearing can use the various physical clues to decide the position of the sound source. At the study inside the laboratory, usually they deal with only one or two clues. Most studies are based on a microphone array [1)] and the time or power difference at both ears [2)]. But we propose a new method to distinguish the various directions of sound sources with only one microphone corresponding to one ear. The time difference is useful in the low frequency area, and the power difference is useful in the high frequency area. But, because the former depends on the transitional part of the sound, it is not suitable for periodic sounds[3)]. The method used in this paper characterizes throughout the sound. And the advantage of our method is to avoid the difficulties about the synchronism of more than one microphone. In this paper, a reflection board is installed so that the reflection sound may change corresponding to the direction of the sound source.

2. Modeling

To think easily, suppose that the sound source is a sinusoidal wave. Let $s(t)$ the sound source, then we obtain

$$s(t) = \sin(t). \tag{1}$$

Next, let us think about the reflection sound made by the reflection board installed as the **Fig.1**. The microphone input includes both of the direct sound and the reflection sound. Let $s_M(t)$ the microphone input, then we obtain

$$s_M(t) = \sin(t) + k \bullet \sin(t + \Delta t). \tag{2}$$

Here, k ($0<k<1$) is the reflection coefficient and Δt is the time interval to go to and from the reflection board. Let $S(\omega)$ denote the Fourier transform of the sound source and $S_M(\omega)$ the Fourier transform of the microphone input, then we obtain

$$S(\omega) = \int_{-\pi}^{\pi} s(t) \bullet e^{-j\omega t} \mathrm{d}t = -j\pi + \frac{1}{4\omega}\left(e^{2j\omega\pi} - e^{-2j\omega\pi}\right) \tag{3}$$

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$$S_M(\omega) = \int_{-\pi}^{\pi} s_M(t) \bullet e^{-j\omega t}\mathrm{d}t$$
$$= \{-j\pi + \frac{1}{4\omega}\left(e^{2j\omega\pi} - e^{-2j\omega\pi}\right)\} + \{-j\pi \bullet e^{j\omega\Delta t} + \frac{1}{4\omega}\left(e^{2j\omega\pi} - e^{-2j\omega\pi}\right) \bullet e^{-j\omega\Delta t}\}. \quad (4)$$

Compared with Eq.(3), the second term in Eq.(4) is the influence given by the reflection board. By the way, with the simulation, we see that the frequency data rather than time data is changed greatly with the change of parameter k or Δt. And we don't have to take the synchronism into consideration in the frequency data. So we make a basic experiment according to the fact to distinguish these frequency influences.

3. Basic Experiment

Fig.2 shows the environment of this basic experiment. The microphone and the speaker are located 33cm apart in consideration for the sound speed, and the microphone and the speaker 1cm apart, just like **Fig.1**. Here we treat two cases: (a) without a reflection board and (b) with a reflection board and it is tried to distinguish these two cases each other. **Fig.3** shows the flowchart in this basic experiment, and we will use the system in the main experiment. In this flowchart, the left part shows the process to estimate the condition, and the right part shows the process to record the sample data. At the beginning of this basic experiment, it is necessary to record some sample data under each condition - with a reflection board or without a reflection board -. In this time, we respectively repeat ten times in the right part to record ten sample data. **Table1** shows parameters used in Fourier transform and recording with a microphone in this experiment. In the estimation process, the system proceeds in the same way as "Recording sample data" until the "Preservation as the data in frequency". The new data is compared with sample data. Then the best matching sample data is selected and shown as the result. Though the actual condition has the reflection board, the sample data without the reflection board is chosen. Then the result may be mis-estimated.

Experiment results show the estimation rate to be the condition in real time is close to 100%. But mis-recognitions sometimes occur. Because the data includes both informations made in two conditions. In other words, the precision is about 100% under the condition that a reflection board is completely static.

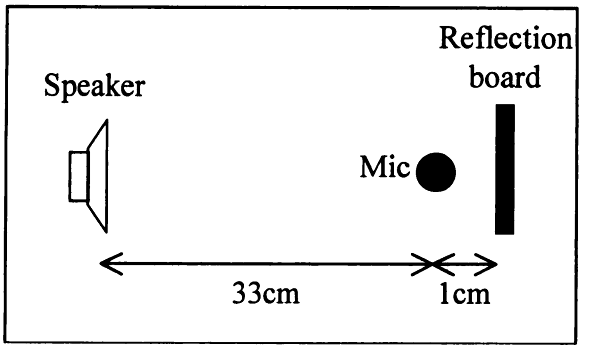

Fig.1 Positions of the microphone, speaker and reflection board.

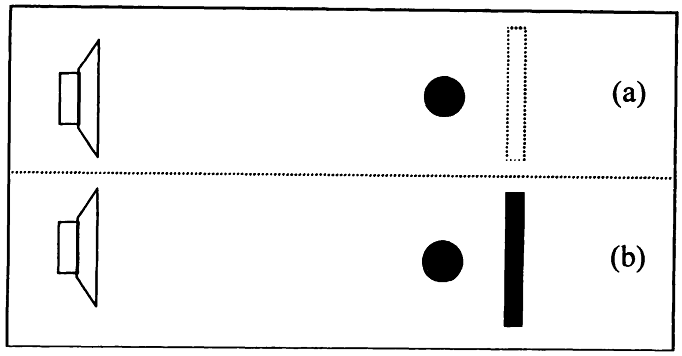

Fig.2 Environment of the basic experiment.

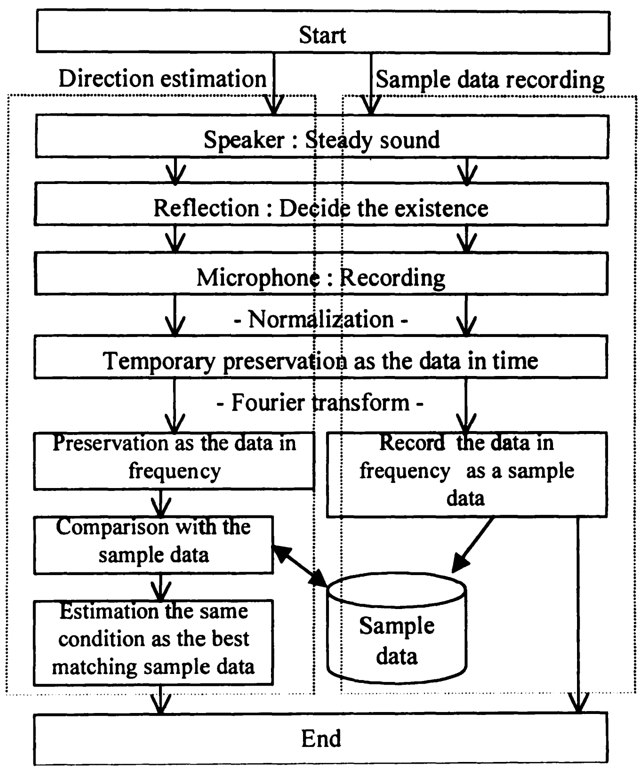

Fig3 Flowchart in the basic experiment.

Table1 Parameters used in recording and Fourier transform.

Sampling frequency	22,050 Hz
Sound source frequency	220 or 440 or 880 Hz
Observation time	100 ms
Measurement frequency interval	20Hz
Maximum frequency	10000 Hz
Thread of input sound	1000

4. Experiment of Estimation of Sound Source Directions

4.1 Outline of the Experiment

Fig.4 shows the positions of the microphone, speaker, reflection board. In this experiment, the reflection board is fixed, and the position of a microphone takes one among five directions (0°~90°). For simplicity, here a planar rectangular board is used as a reflection board. It is made of thick paper. This system covers only the range (0°~90°) in consideration for the relation of positions.

Fig.5 shows the flowchart. The parameters are shown in **Tablel**. The frequency of the sound source is only 440Hz. The outline of the experiment is almost like that of the previous basic experiment. First, we record ten sample data in five cases. The acquired data are normalized the amplitude. Because the amplitude changes according to the power of the sound source. The time data is changed to the frequency data and preserved as sample data. In the estimation process, this system proceeds in the same way as "Recording sample data" until "Preservation as the data in frequency". And the new acquired data is compared with sample data. The best matching sample data is selected and indicated as the result.

4.2 Results

We show relations between the direction estimated by this system and the actual direction of the sound source in **Table2** and **Table3**. In **Table2**, we only count the number. In **Table3**, the data is converted into probability. Here a column shows the actual direction of the sound source and a row shows the direction estimated by this system. When the value of a column is equal to that of a row, this system outputs a right result. And when not so, this system outputs a wrong result.

According to the result shown in **Table2** and **Table3**, the precision of this system is about 90%. It is shown with the bold-faced type. More, the precision is 99% if the 22.5° error is allowed. It is shown with the underline. Though the 22.5° error may be so big that we can't say it the precise estimation, we should pay attention to it to increase the precision from now on. The precisions at two positions (0° and 90°) are better than others. Because these two positions have only one side error, and the precision includes the probability of another side error.

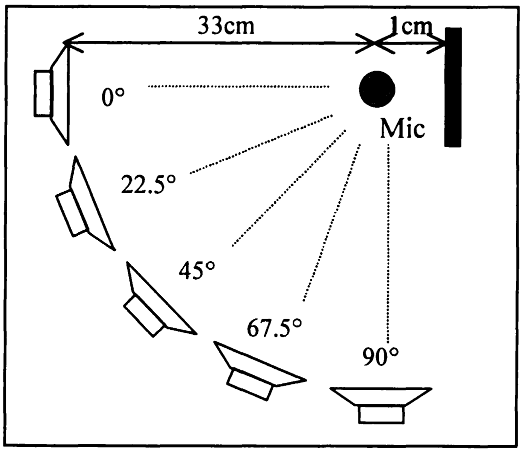

Fig.4 Positions of the microphone, speaker and reflection board.

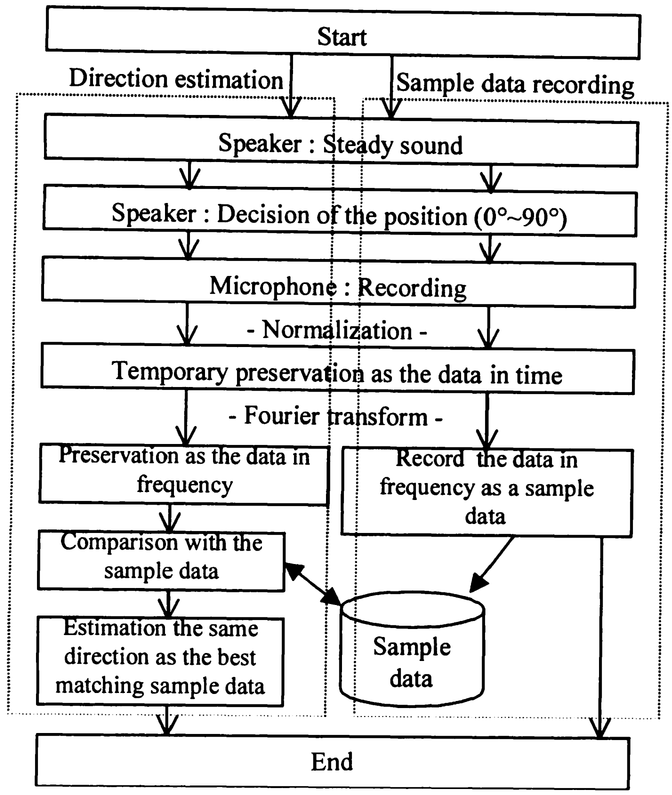

Fig.5 Flowchart in the experiment of estimation of sound source directions.

Table2 Result of the experiment (The data are shown in the number).

actual \ estimated	0°	22.5°	45°	67.5°	90°
0°	**114**	<u>5</u>	-	-	1
22.5°	<u>6</u>	**106**	<u>7</u>	1	-
45°	-	<u>8</u>	**103**	<u>9</u>	-
67.5°	-	2	<u>6</u>	**107**	<u>5</u>
90°	-	1	1	<u>7</u>	**111**

Table3 Result of the experiment (The data are shown in the probability).

actual \ estimated	0°	22.5°	45°	67.5°	90°
0°	**0.95**	<u>0.04</u>	-	-	0.01
22.5°	<u>0.05</u>	**0.88**	<u>0.06</u>	0.01	-
45°	-	<u>0.07</u>	**0.86**	<u>0.08</u>	-
67.5°	-	0.02	<u>0.05</u>	**0.89**	<u>0.04</u>
90°	-	0.01	0.01	<u>0.06</u>	**0.92**

5.Conclusion

In this paper, the method to estimate the sound source directions by a reflection board is proposed. The experimental result shows the efficiency of the method, though it is tried under some limitations, and rough and simple. We must think about three points as the future subject. The first one is the range for estimation. The method using a time difference at both ears covers the range 180°. So we want this system to cover the equal range. The second one is the sound source. Before applying this method to the sound required in daily life, it must be adapted for the various sound sources. The third one is the shape of the reflection board. Though it is the simplest shape in this paper, it is changed to the various shapes.

References

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