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Research on the estimation of sound source direction using multi microphone

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1 Introduction

Recently, realization of sound source direction estimation is expected in the various fields. However, reverberations exist in the real environment. It is reported that performance of sound source direction estimation by usual methods reduces in such an environment[1]. This paper proposes a new sound source direction estimation method. This method is robust to the real environment with reverberation as a simple model of hearing mechanisms from the engineering viewpoint.

2 Reverberation

We live environments usually with a reverberation. A sound source direction estimation method has to be applied to such environment. It is difficult to separate a voice from a signal in the environment with reverberation, because the signal is mixture information.

Therefore, using characteristics of the reverberation does a countermeasure to the reverberation. The characteristics of the reverberation are as follows.

- Reflected sound delays from the direct sound.
- Its power is smaller than that of a direct sound.

Reflection sound passes through longer distance than the direct sound, and reach an observation point. Therefore, the reverberation has the above characteristics. Using these characteristics, the direct sound can be extracted from the mixed sound.

3 Estimation of sound source direction

3.1 Microphone array

This paper uses a microphone array that arranges microphones in the apexes of the equilateral triangle. This is the smallest number of microphones that can estimate all direction in the two-dimensional plane. This microphone-array can solve the problem of front-back miss judgment when using a linear microphone-array.

3.2 detection of time difference

One of the most important cue of the direction estimation for mammals is the inter aural time difference (ITD). Distance to each ear from the sound source is different when the sound source moves to right or left from in front of the head and the different distance causes ITD.

The Jeffress model is one of model circuits for the detection ITDs[2]. This model is represented as a circuit, which consists of some coincidence detectors and two-delay line from left and right ears. The detectors fire only when impulse trains coming from both sides though delay lines arrive simultaneously. Thus the model can calculate ITDs with correlation between impulse trains coming from both sides. This model has been approved because of simple theory and structural analogy between the model and the hearing mechanism. Therefore, it is easy to apply the model to ITD detection in the engineering viewpoint.

This paper used the ITD as a significant cue for estimating sound source direction, and measured the ITD by using a simplified model of the Jeffress model.

3.3 Precedence effect

We can determine sound source directions in reverberantal environment. This is because we take the first sound reached. This ability is called the Hass effect. In this paper, the dynamic threshold is proposed to realize this ability. Dynamic threshold detects onset of sound with changing threshold level. The dynamic threshold behaves as follows.

- An onset is detected when amplitude sets over a threshold
- The amplitude at that time is hold as an initial value of a threshold level.
- The threshold level decreases exponentially with time.
- Wherever amplitude gets over a threshold, the initial value of the threshold reset.

Therefore the dynamic threshold does not detect reflected sound, which delays from the direct sound and the power is smaller than the direct sound.

4 Experiments and results

Experiments to confirm the ability of the proposed method were carried out under various conditions. Results of the comparative experiment with the cross-correlation method used for usual research are shown in table 1, figure 1. Here, detection rate is represented that the number of detection points within five degrees against the sound source direction is divided by the number of all detected points. The results show that, detection rate is large for both methods, when a reverberation does not exit. However, detection rate is reduced for using cross-correlation method, when reflected signal is a large level.

Table 1: result of comparative experiment

		echo-to-signal ratio			
		-∞ dB(no-echo)	-14dB	-8dB	-4dB
detection rate [%]	cross-correlation method	80.2	67.9	34.5	32.1
	proposal method	86.6	89.6	89.5	84.0

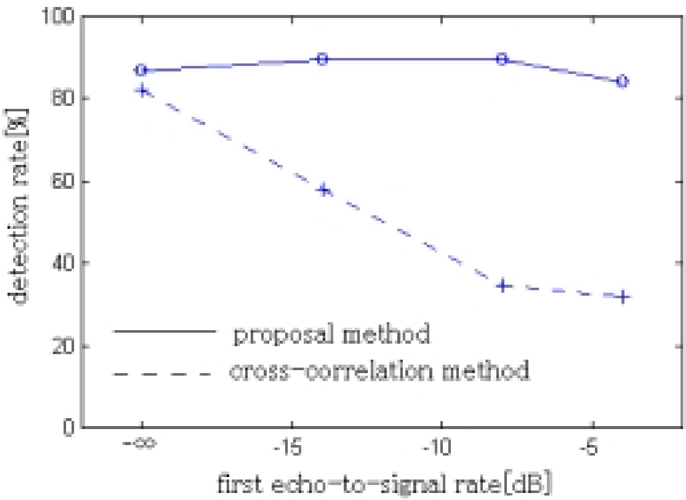

Figure 1: result of comparative experiment

The experiment results using the signal recorded in the environment with reverberation are shown in table 2. Experimental conditions are as follows; the sound source direction is 180 degree and noise existed in the room is used as environmental noise, whose SNR was adjusted by changing the volume of the signal. The reverberation time of this room is 0.56 sec, as a result of the measuring using band passed noise of 500-100Hz, and the environmental noise level was 49.6dB(A).

The result shows , detection rate was over 50% at any condision. However, the performance reduced as noise becomes louder.

Table 2: result of real environment

	SNR 15dB	SNR 19dB	SNR 24dB
detection rate	50.2	59.0	71.3

5 Conclusion

The improvement of the performance using this method in the environment with reverberation was confirmed as results of the comparison with the cross correlation method. This method is robust in the real environment with reverberation.

The results of the experiment using data recorded in real environment show, that performance deteriorated when noise becomes louder. This problem should be solved in further.

References

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