Hat-type hearing system using MEMS microphone array

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Abstract

For speech communications, it is useful to emphasize the sound coming from a certain direction. To give more freedom to sound directivity, we want to use a large-scale microphone array for hearing aids. However, it is difficult to mount a large-sale microphone array in a natural way. In this paper, we propose and develop a hat-type hearing system using micro electro mechanical systems (MEMS) microphone array with high-speed single-bit signals. From the results of measurement of sound directivity, our system achieves a shape forward directivity. The results of experiment in an actual environment shows that the sound coming from the front direction is emphasized about 10 dB compared with the other directions.

***Index Terms*— Beamforming, Single-bit, Hat, MEMS microphone array, FPGA.**

1. Introduction

It is useful for speech communications to emphasize the sound coming from a certain direction. The beamforming technique is well-known as one of techniques to improve speech intelligibility. Recently, hearing aids using the beamforming technique have been studied and developed all over the world [1]. Only a few microphones are commonly used for beamforming because of the limit of the hardware scale and space to mount the microphones. On the other hand, a large-scale microphone array gives more freedom to sound directivities. When the microphone array is used for the hearing aid system, the system must be small enough to wear. In order to mount a lot of microphones on the small space, we use a micro electro mechanical systems (MEMS) microphones. In addition, we use a hat because we can get space enough to mount the MEMS microphone array.

On the other hand, we have studied the beamforming techniques with high-speed single-bit signals. Previous researches show the achievement of high spatial resolution by just using delay-and-sum beamformer [2]. There are some advantages of high-speed single-bit signals processing [3]. In addition, we can construct extremely simple system using single-bit signals [4].

In this paper, we propose and develop a hat-type hearing system using MEMS microphone array with high-speed single-bit signals. We measured the sound directivity in the anechoic chamber and effectiveness in an actual environment to confirm the effectively of our system.

2. Beamforming

Nomenclature

$s(t)$	Incident plane wave
$z_m(t)$	Signal observed by the m-th microphone
M	Number of microphones
τ_m	Delay time at the m-th microphone
$y(t)$	Output signal

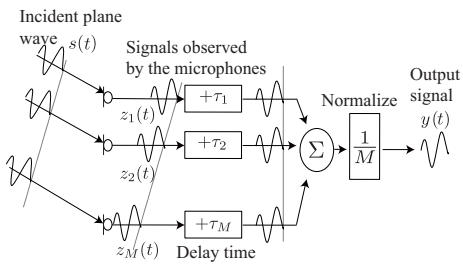

Fig. 1. Signal flow of a delay-and-sum beamforming.

Figure 1 shows the signal flow of a delay-and-sum beamforming technique. This technique can emphasize the sound coming from a steering direction. The incident plane wave travels, and it is measured at the microphone array, which is formed by M microphones. The signal measured by the m-th microphone $z_m(t)$

$$z_m(t) = s\left(t - \tau_m\right), \tag{1}$$

where $s(t)$ is the incident wave. If the delay of τ_m is added to the signal on the m-th microphone, we synchronize all signals and the sum of them is the emphasized signal $y(t)$ from the target direction [5].

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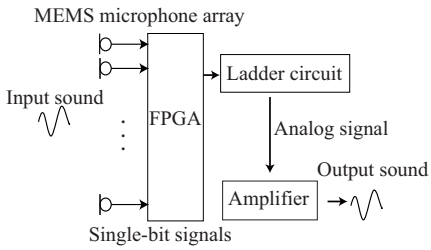

Fig. 2. Diagram of hat-type hearing system.

Fig. 3. Exterior of hat-type hearing system.

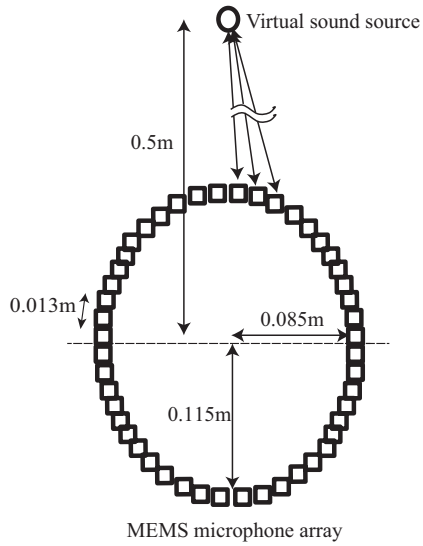

Fig. 4. Arrangement of the microphone array and the virtual source. We calculate delay time for each microphone.

3. Hat-type hearing system

3.1. Supecification

Figure 2 and 3 show the diagram and exterior of our developed hat-type hearing system, respectively. To implement a lot of microphones on a hat, the electric circuits must be simple and small. Therefore, we use 48 MEMS microphones (SPM0405HD4H, KNOWLES ACOUSTICS) of which output signal is an single-bit signal with 2.5 MHz sampling frequency. We can achieve fine delay time without interpolation because the sampling frequency of single-bit signal is extremely high. FPGA(EP4CE15F17C8N, ALTERA) gets the input signal from each microphone and calculates an output signal by the delay-and-sum beamforming technique. The output signal is converted from a digital signal to an analog signal by the ladder circuit, and it is amplified for hearing system. Our developed system operates on 9.0 volt batteries. The batteries are small enough to carry or wear the system. The weight of the proposed system with the hat is 500 g.

3.2. Calculation of delay time for each microphone

We adjust each delay time to make a beam-forming pattern. In this paper, we made a sharp directivity in front of a head. The arrangement of the microphones and the virtual sound source are shown in Fig.4. The microphones are arranged on a hat with elliptical shape. Minor axis and major axis of the ellipse are 0.17 m and 0.23 m, respectively. We assume that a target sound source is located at 0.5 m distant from the center of microphone array. To obtain delay time, we calculate the distance between the target sound source and each microphone and divide it by the speed of sound. We also assume that there are no sound diffraction caused by a head and body and no attenuation for simplification.

4. Measurement of directivity

We measured sound directivity of the proposed system in an anechoic chamber. Figure 5 shows the arrangement of measurement. The loudspeaker (MSP7 STUDIO, YAMAHA) is located at 0.5 m distance from the center of microphone array. The hat-type hearing system is put on the dummy head (KU100, Neumann). The height of the loudspeaker cone and the mouth of dummy head is 1.0 m, respectively. The sounds are recorded with rotating the loudspeaker every two degrees. We use white noise as a measurement signal. The sampling rate is 44.1 kHz. The directivity of hat-type hearing system is calculated from the power of the measured signal at each degree.

Figure 6 shows the sound directivity of the pro-

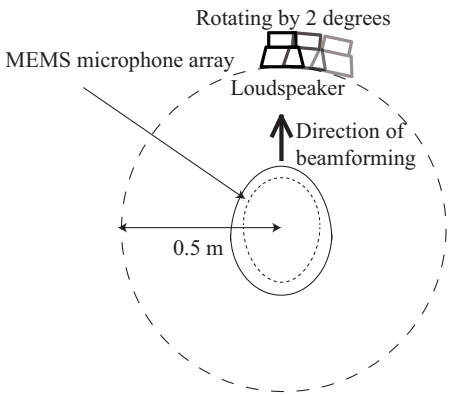

Fig. 5. Arrangement of directivity measurement.

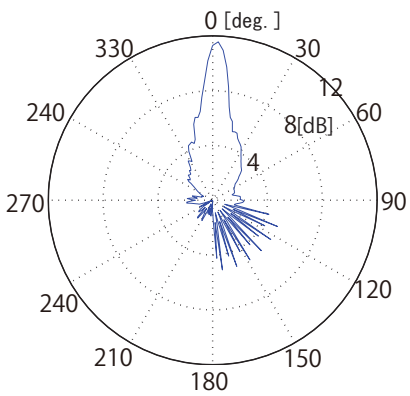

Fig. 6. Sound directivity of proposed system.

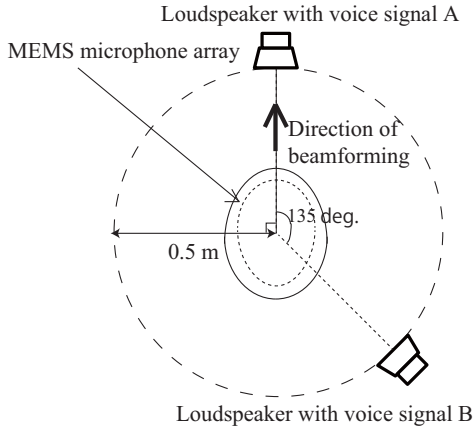

Fig. 7. Arrangement of measurement in actual environment.

Fig. 8. Experiment setup in actual environment.

posed system. Although the arrangement of microphones is not exactly the elliptic shape and the sound diffraction caused by a head is not considered, the directivity is directed forward sharply. Half width at half maximum (HWHM) is 14 degree. The difference between the maximum and minimum values is about 12 dB.

5. Experiment in actual environment

We conducted a experiment in a meeting room 9.0 by 6.0 by 2.3 meters. The background noise was about 50 dB. Figure 7 shows the arrangement of the microphone array and two loudspeakers in this experiment. Experimental setup is shown in Fig.8. The distance between the center of microphone array and each loudspeaker was 0.5 m. The height of the center of loudspeaker cone and the mouth of dummy head is same as the experiment of directivity in Sec.4. One loudspeaker was located at the front of microphone array and generated a voice signal A. Another one was located at right 135 degree direction and generated a voice signal B. We reproduce signals in three patterns in this experiment: playing back each voice signal and both signals. In the case of playing back both signals, we measured both with and without beamforming processing.

The spectrogram of sum of all microphone signals in playing back voice A, B and both signals without beamforming are shown in Fig.9, 10 and Fig.11, respectively. Figure 12 also shows the spectrogram in playing back both signals with beamforming.

As Fig.11 shows, the spectrogram without beamforming includes both components of voice signal A

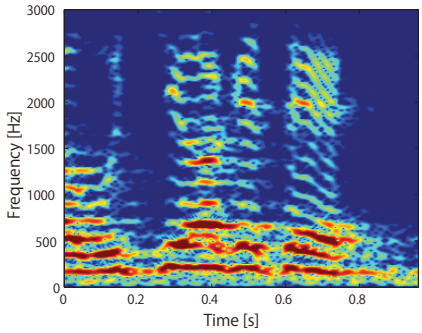

Fig. 9. Spectrogram of sum of signals at all microphones without beamforming. We reproduce voice A using the front loudspeaker.

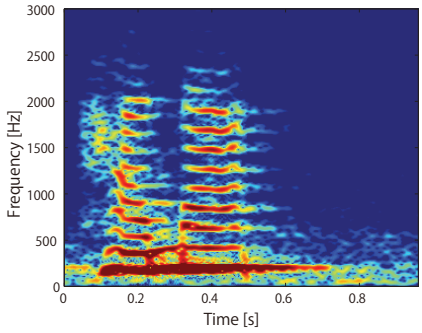

Fig. 10. Spectrogram of sum of signals at all microphones without beamforming. We reproduce voice B using the loudspeaker at 135 degree right.

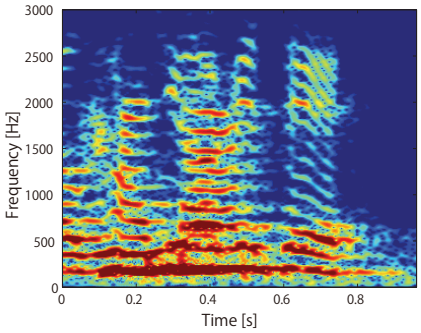

Fig. 11. Spectrogram of sum of signals at all microphones without beamforming. We reproduce both voice A and B using two loudspeakers.

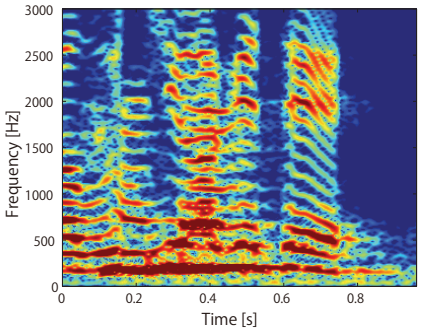

Fig. 12. Spectrogram of sum of signals at all microphones with beamforming. We reproduce both voice A and B using two loudspeakers.

and B. However, as Fig.12 shows, the voice signal A is emphasized and the harmonic components of voice signal B are attenuated with beamforming signal processing.

6. Conclusions

We proposed and developed a hat-type hearing system using MEMS microphone array with high-speed single-bit signals. The delay-and-sum beamforming can make a sound directivity with the microphone array which is mounted on the brim of a hat, and can emphasize the sound traveling from the front 12dB. In addition, experiment in an actual environment was conducted with the proposed system. When the system had front directional directivity based on the delay-and-sum beamforming, voice signal coming from the front direction was emphasized and voice signal coming from 135 degree right direction was attenuated. In the future, we will consider the reflection or diffraction caused by a head, and also continues the studies for application to a hearing system.

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