Implementation of directional characteristics by real-time processing of sounds observed by three microphones

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Abstract

In this study, focusing the difference between the sound powers observed by microphones is depending on the location of the sound source, we propose a method of forming a sharp directivity by three microphones. The implementation of directional characteristics by real-time processing of sounds observed by three microphones were carried out using mbed microcontroller.

Keywords: directional microphone, real-time processing, mbed

1. Introduction

Conventional microphone technique is by forming a super-directional microphone using a delay-and-sum beamformer or an adaptive microphone-array, it can be recorded 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. Directional microphone that can separate the speech of only specific speaker by suppressing the external noise, is useful for applications such as the telephone transmitter, the voice recognition, etc. Not only to provide the directivity of the microphone, but also to limit the distance of the speaker, removing effect of the external noise can be expected.

In our recent study, focusing the difference between the sound powers observed by two microphones is depending on the location of the sound sources, we have proposed a method of forming a sharp directivity by two microphones, and the validity of the method was shown by the computer simulation⁽²⁾. This method can be very easy to implement by digital processing, and the method has a feature that can be recorded target sound at real-time.

In this study, we propose new type of the directional microphone that have the limitation of distance to the microphone, by using the dependence of the sound power on the location of the sound sources. The implementation of the directional microphone by real-time processing of sounds observed by three microphones is also presented.

2. Emphasis of the target speech

Figure 1 shows an illustration of the directional characteristics of the microphone proposed theory. The directivity characteristic of standard two microphones are shown Fig. 1(a). The X_1 and X_2 are the directional characteristics of the microphones of MIC1 and MIC2, respectively. The distance of the microphones must be enough close distance in order to ignore the phase difference at the sampling frequency. When calculating the difference of the signal power from two microphones $X_2 - X_1$, the directional characteristics of the combined microphones is shown in Fig. 1(b). When the sound source located just front of the two microphones, the distance from the sound source to the microphones are equal, therefore the output signal of the combined microphones becomes silent. Large output can be obtained when the sound source is located much different distance from the microphones. Though the output tends to decrease with increasing distance, the output further decreases. Because the difference of the distance becomes negligible. As shown in Fig. 1(c), by applying a damping coefficient α to the input X_1 , it is possible to extend the directivity toward to the microphone X_2 .

Fig. 1(d) shows the configuration of the directional microphone which is proposed in this study. Three microphones are arranged one in front center (MIC2) and placed other two microphones on the left (MIC3) and right

Fig. 1. Illustration of the directional characteristics of the proposed theory.

(MIC1) rear. The output is combined with the difference of the signals between front (MIC2) and right (MIC1), and front (MIC2) and left (MIC3). Applying the damping factor of α and β to the output signal X_1 and X_3 , it is possible to extend the directivity toward to the front. Similar to the case using two microphones mentioned above, the output decreases with increasing distance from the sound source. That is the directional microphone with the limited distance.

3. Experiment

Figure 2 (a) and 2(b) are the circuit diagram and photograph of the prototype of the experimental apparatus, respectively.

Fig. 2. Circuit diagram and prototype of the proposed directional microphone.

The experimental circuit was fabricated using mbed microcontroller which is well known as one of the rapid prototyping tool⁽³⁾. Online development tool is provided for mbed. Many kind of libraries can be used on the mbed official site, so the software is very easy to code (see Appendix). There are some kind of mbed platform. We have chosen the enough specification of mbed LPC1768 for this study, which is based on the NXP LPC1768, with a 32-bit ARM Cortex-M3 core running at 96MHz, included 512KB FLASH and 32KB RAM⁽⁴⁾.

Three condenser microphones are used in the circuit, the microphones are directly connected to the analog input port (p15, p16, p17) of the mbed in order to capture the relations of left and right phase as faithfully as possible. The three channel audio signals (X_1 , X_2 and X_3) are read as the continuous analog value by 12 bit A/D converter by means of the subtracted of the dc offset level. The 8 kHz sampling was done by the timer interrupt setting every 125 μ s. The distance of the rear two microphones is 5 cm, which is enough close distance in order to ignore the phase difference at the sampling frequency. The output signal Y is calculated by following equation at real-time,

$$Y=(X_2-\alpha X_1)+(X_2-\beta X_3) \quad (1)$$

The signal output continuously using the mbed built-in 10 bit D/A converter at the same term with the calculation. The signal is amplified by the operational amplifier and plays the headphones for monitoring the result.

4. Results and discussion

Figure 3 shows the example of the voice recording result using the experimental apparatus. The sensitivity of the microphone seems to be enough in order to validate the proposed theory.

Figure 4 shows the experimental result of the directional pattern at different frequency of the reference sound. The sine wave from the 3.5 cm diameter dynamic speaker is used for the reference. The characteristics of single microphone is also shown in the figure. The output was reduced when the incident direction of the sound is inclined to the left and right. It becomes poor directivity at the frequency of 1 kHz. Because the phase difference of the microphones cannot be negligible at high frequency.

Figure 5 shows the sound level versus the distance from the microphones at different frequency. The remarkable

reduction of the sound level is observed. This means the microphone system can be useful for the limited distance.

Fig. 3. Example signal of the recorded voice by the center microphone (X_2).

Fig. 4. The directional pattern at different frequency.

Fig. 5. Sound level versus the distance from the microphones.

When uttered a voice just front of the center microphone at a distance of about 10cm, clear output was obtained for resulting the calculation of Eq. (1) from the three signals. When the incidence direction of the sound source moved to the left and right, the output was reduced. However, it was not silence even if the output is reduced. Because the noise due to a phase difference between the left and right signals was remained. In addition, when the distance between sound source and microphones becomes far, the directivity of the microphone becomes unclear. This is because the difference of the distance between X_1 , X_2 and X_3 is reduced to each other.

To evaluate the difference of the magnitude of the signals, the target voice is uttered from the right 45 degree for the distance about 10 cm, and then the signals recorded by three microphones. Figure 6 shows the calculated signals of X_2-X_3 and X_2-X_1 , corresponding the distances are different and almost same, respectively. Compared with these signals, the magnitude of the calculated signals are different enough, and the difference causes the directional characteristic of the combined microphone.

Fig. 6. Example signals of X_2-X_3 and X_2-X_1 for the different distance from sound source.

5. Conclusions

We have proposed the new theory of the directional characteristics by the combined three microphone. The implementation of the proposed directional microphone using mbed microcontroller, and the experimental result of the directivity were presented. It has advantages to modularization of the directional microphone without personal computer, and it would be possible to provide an inexpensive directional microphone module.

References

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Appendix: Example of the mbed source code for the proposed directional microphone.

```
#include "mbed.h"

#define DEF_ALFA 0.8
#define DEF_BETA 0.8
volatile float micRB, micLB, micCB;
volatile float x1, x2, x3, y;
volatile float alfa, beta;
AnalogIn micR(p15);
AnalogIn micL(p16);
AnalogIn micC(p17);
AnalogOut spOut(p18);
BusOut leds( LED4, LED3, LED2, LED1 );
Ticker timer;

void sampling(){
    x1=micR-micRB; x2=micC-micCB; x3=micL-micLB;
    y=(x2-alfa*x1)+(x2-beta*x3);
    spOut=y*10.0+0.5;
}

int main() {
    alfa=DEF_ALFA;
    beta=DEF_BETA;
    wait(2);

    micRB=micLB=micCB=0;
    for(int i=0;i<1000;i++){
        micRB+=micR; micLB+=micL; micCB+=micC;
    }
    micRB/=1000; micLB/=1000; micCB/=1000;
    timer.attach_us(&sampling,125);

    while(1){
        leds=abs(y*1024);    // for level meter
    }
}
```