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Performance Evaluation of Directional Microphones

Wen-Kung Tseng
Graduate Institute of Vehicle Engineering, National Changhua University of Education, Taiwan

Abstract

Directional microphones are used in popular car multimedia systems, such as CAR PLAY and ANDROID AUTO. The driver can command the microphone to control the car multimedia system. The directional microphone can also improve speech recognition accuracy and prevent the microphone from receiving noise from outside the car. This research improves microphone array directivity. Delay-and-sum beamforming combined with the optimization method is used in this study to achieve the desired effect of the directional microphone array. An array of 57 circular microphones is used in the study. The results show that the proposed method significantly improves microphone array directivity.

Keywords – *Microphone array, directivity, delay-and-sum beamforming, optimization method, circular microphones.*

I. INTRODUCTION

The environment is filled with all kinds of noises and echoes. This noise interference greatly reduces speech recognition performance. It is very important to prevent the microphone from being affected by background noise in automobiles, conference rooms, and hands-free audio communications. In most cases, the voice signal is too poor to receive, especially when the speaker is far away from the microphone. The most effective way to improve SNR in the voice input system is to place the microphone closer to the speaker. However, this may not be the ideal solution [1] [2]. An array signal processing technique is used in this study [3]. This processing technique overcomes ambient noise and voice signal echo and restores the sound signal without noise. Hands-free audio communication is used to reduce room reverberation and noise. In the past, most of the work was to achieve the directional microphone array effect [4] [5].

A novel approach is proposed to control the beam-width of the main lobe and the level of the side lobe. The beam pattern level is controlled using the optimization technique. Compared with a system using a single microphone array, the microphone array and voice acquisition beam-forming technology combination is expected to be significantly improved [9]. Ohyama, Sasagawab, Takayama, and Kobayashia proposed a closed microphone array system with a complex weighting method [7]. Adaptive microphone arrays facilitate simple built-in instrumentation and environmental calibration. The scheme provides several advantages, such as simple calibration procedures, suppression of directional sources, multi-function robust beamforming, and target signal distortion reduction. This analysis adopts the non-causal wiener filter and produces a compact and effective theoretical suppression limit [8].

This paper investigates the directivity performance using a circular array composed of fifty-seven microphones as well as the influence on the sound beam diffusion angle. The optimal weighting value of the microphone corresponding to the specific sound beam extension angle can be calculated. The circular array consisting of fifty-seven microphones has been used to control the directivity of the beam-width.

II. METHOD

The delay-and-sum beamforming combined with the optimization method is used in the study. The delay-and-sum beamforming is to be accepted as a simple but powerful array signal processing algorithm [2, 3]. Assume that a group of M microphones, which is arranged in a uniform linear array (ULA) with an inter-element spacing of d. An observation point is set in the far-field of the array at an angle θ concerning the normal of the microphone array aperture. If each microphone is weighted with a weighting, w_n for n=0, 1, 2, . . ., M-1, the array response function can be derived as [2, 3]:

$$H(\omega\tau)=\frac{1}{M}\sum_{n=0}^{M-1}w_n e^{im\omega\tau} \tag{1}$$

where

τ = (d/c)sinθ is the time delay.

ω is the frequency.

w_n is weightings for each microphone.

θ is the angle concerning the axis of the beam.

From (1) it shows that the maximum of the main lobe exists on the broadside of the ULA(θ=0). However, the maximum of the main lobe can be changed by adding a phase shift or delay to each microphone. If the ULA is to be steered in the direction θ_0, time delay ($n\tau_0$) has to be added to n the microphone. The time delay τ_0 can be calculated as τ=(d/c)$\times$sinθ_0, and the array response of the delay-and-sum beamforming becomes:

$$H(\omega\tau) = \frac{1}{M}\sum_{n=0}^{M-1} w_n e^{im\omega(\sin\theta - \sin\theta_0)d/c} \quad (2)$$

Then (2) can be expressed as:

$$H(\omega\tau) = \frac{1}{M}\sum_{n=0}^{M-1} w_n e^{jn\omega(\tau-\tau_0)} \quad (3)$$

Then the far-field directivity of the weighted primary sources array for frequency ω_a $D_{1a}(\theta)$ can appear:

$$D_{la}(\theta) = D_1(k_a, \theta)H(k_a, \theta) \quad (4)$$

Where $D_1(k_a, \theta)$ is the aperture directivity for frequency ω_a, and the far-field array response $H(k_a, \theta)$ is indicated in (1) with w_{am} and ω_a instead of w_m and ω, similarly, the far-field directivity for the primary frequency ω_b with w_{bm} instead of w_{am}? Therefore, the beam pattern of the sound frequency can be expressed as:

$$D(\theta) = D_1(k_a, \theta)H(k_a, \theta)D_1(k_b, \theta)H(k_b, \theta) \,. \quad (5)$$

Generally speaking, one microphone directivity can be expressed as [2]:

$$D(\theta) = 0.5 + 0.5\cos(\theta) \quad (6)$$

Therefore the directivity of the microphone array can be expressed as:

$$D\_(\theta) = [0.5 + 0.5\cos(\theta)] \cdot \frac{1}{M}\sum_{n=0}^{M-1} \omega_n e^{jn\omega(\tau-\tau_0)} \quad (7)$$

(7) can also be expressed as:

$$D\_(\theta) = [0.5 + 0.5\cos(\theta)] \cdot \frac{1}{M}\sum_{n=0}^{M-1} \omega_n e^{jn2\pi d(\sin\theta - \sin\theta_0)d/c} \quad (8)$$

The formulation of the optimization approach proposed in the work for designing directivity microphone sound systems can be expressed as:

Minimize σ

Subject to

$$\|D\_(\theta_1)\|_2^2 + \|D\_(\theta_2)\|_2^2 \langle \sigma$$

$$\|D\_(\theta_3) - D\_(\theta_1)\|_\infty > \delta$$

$$\|D\_(\theta_3) - D\_(\theta_2)\|_\infty > \delta \quad (9)$$

Whereσ is a real number, θ_1 and θ_2 are the angle the side lobe, θ_3 is the angle of the main lobe, δ is the predefined value between the main lobe and side lobes.

(9) can also be expressed as:

Minimize σ

Subject to

$$\left\|[0.5 + 0.5\cos(\theta_1)] \cdot \frac{1}{M}\sum_{n=0}^{M-1} \omega_n e^{jn2\pi f(\sin\theta_1 - \sin\theta_0)d/c}\right\|_2^2 + \left\|[0.5 + 0.5\cos(\theta_2)] \cdot \frac{1}{M}\sum_{n=0}^{M-1} \omega_n e^{jn2\pi f(\sin\theta_2 - \sin\theta_0)d/c}\right\|_2^2 \langle \sigma$$

$$\left\|[0.5 + 0.5\cos(\theta_3)] \cdot \frac{1}{M}\sum_{n=0}^{M-1} \omega_n e^{jn2\pi f(\sin\theta_3 - \sin\theta_0)d/c} - [0.5 + 0.5\cos(\theta_1)] \cdot \frac{1}{M}\sum_{n=0}^{M-1} \omega_n e^{jn2\pi f(\sin\theta_1 - \sin\theta_0)d/c}\right\|_\infty \rangle \delta$$

$$\left\|[0.5 + 0.5\cos(\theta_3)] \cdot \frac{1}{M}\sum_{n=0}^{M-1} \omega_n e^{jn2\pi f(\sin\theta_3 - \sin\theta_0)d/c} - [0.5 + 0.5\cos(\theta_2)] \cdot \frac{1}{M}\sum_{n=0}^{M-1} \omega_n e^{jn2\pi f(\sin\theta_2 - \sin\theta_0)d/c}\right\|_\infty \rangle \delta \quad (10)$$

The optimal values ω_n can be calculated using the function *fmincon*() in MATLAB. By substituting the optimal values w_n into (8), the directivity of the microphone sound beam $D\_(\theta)$ can be obtained.

III. RESULTS

In this section, the directivity of the microphone array created by using the optimization method as shown in (10) is presented. The numbers of microphones used in the system are 8 and 52 with a frequency response at 1 kHz. The speed of sound *c* is 344 ms^{-1}. The weighting functions w_n are calculated for different angle's beam-width for $\theta = 30$, 40, and 60 degrees using the proposed method.

Fig. 1 is the directivity of eight microphone array only using the delay-and-sum beamforming. It can be seen that the amplitude difference between the main and side lobe is about 20 dB. Fig. 2 shows the directivity for the beam-width of 30 degrees with an array of 52 microphones by using the delay-and-sum bam-forming combined with the optimization method. From Fig. 2 we can observe that better directivity of the microphone array is created using the optimization method proposed in this study. We can also observe that the amplitude difference between the main lobe and side lobe is about 150 dB. Fig. 3 shows the directivity for the beam-width of 30 degrees with an array of 52 microphones by using the delay-and-sum bam-forming and the optimization method in 3D. From Fig. 3 the amplitude difference between the main lobe and side lobe is about 150 dB over the whole angle.

Fig. 4 shows the directivity for the beam-width of 40 degrees with an array of 52 microphones by using the delay-and-sum bam-forming and the optimization method in 3D. From Fig. 4 the amplitude difference between the main lobe and side lobe is about 200 dB. From the figures, it can be seen that the amplitude in the sidelobe using the proposed method is lower than that for the beam-width of 30 degrees. Fig. 5 shows the directivity for the beam-width of 60 degrees with an array of 52 microphones by using the delay-and-sum bam-forming and the optimization method in 3D. From Fig. 5 we can observe that the amplitude difference between the main lobe and side lobe is about 250 dB over the whole angle. From the figures, it can be seen that the directivity of the microphone array with 52 microphones using the optimization method proposed in the study is better than that with 8 microphones using the delay-and-sum beamforming method. This is because the optimization method tried to find the optimal weightings which minimize the sum of the squared amplitude of the sidelobe and subject to the amplitude difference between the main lobe and the side lobe. Therefore the better directivity of the microphone array used in this study could be obtained.

IV. CONCLUSIONS

This study investigated the performance of the directivity for the microphone array using the optimization method. The theoretical derivation of the directivity for the microphone array has been presented. Also, the formulation of the beam width control for the directional microphone array using the optimization method has been described. As can be seen from the results the beam width of the directional microphone array could be controlled and the amplitude of the side lobe could be minimized using the optimization method.

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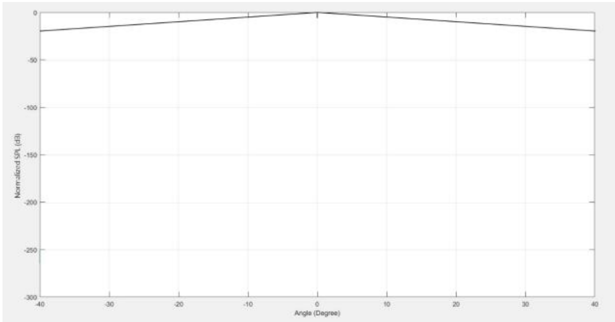

Fig. 1. Directivity for the microphone array with 8 microphones for $\theta = 30^0$ using the delay-and-sum beam-forming method.

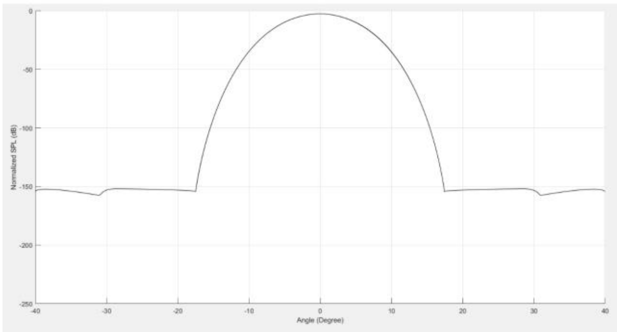

Fig. 2. Directivity for the circular microphone array with 57 microphones for $\theta = 30^0$ using delay-and-sum beam-forming and optimization method.

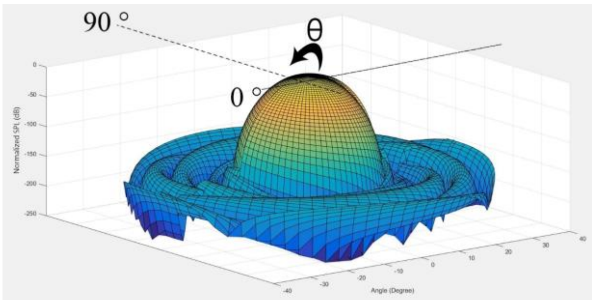

Fig. 3. Directivity for the circular microphone array with 57 microphones for $\theta = 30^0$ using delay-and-sum beam-forming and optimization method. (3D)

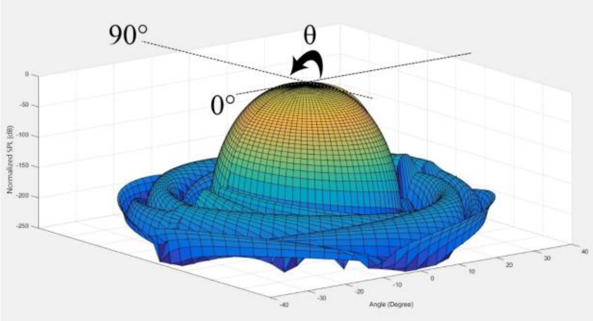

Fig. 4. Directivity for the circular microphone array with 57 microphones for $\theta = 40^0$ using delay-and-sum beam-forming and optimization method. (3D)

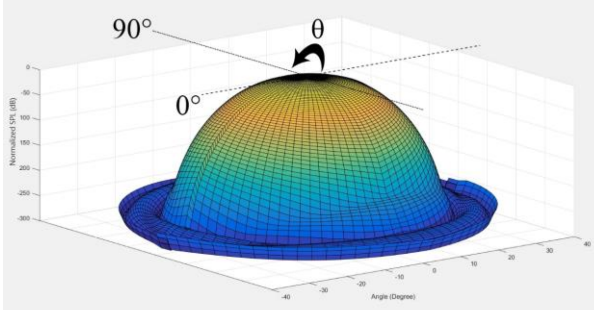

Fig. 5. Directivity for the circular microphone array with 57 microphones for $\theta = 60^0$ using delay-and-sum beam-forming and optimization method. (3D)