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Directivity Improvement of a Microphone Array

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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 array microphone directivity. Delay-and-sum beamforming method is used in most systems to achieve the desired directional microphone array effect. However this study used 57 circular array microphones with delay-and-sum beamforming combined with the optimization method to control the directivity of the microphone array. The results show that the proposed method significantly improves microphone array directivity.*

KEYWORDS: *Directional microphone; microphone directivity; delay-and-sum beamforming; optimization method.*

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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 automobile, conference room 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 the voice input system SNR is to place the microphone closer to the speaker. However, this may not be the ideal solution [1] [2]. An array signal processing technique was used in the literature [3]. This 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 researches used this method to achieve the directional microphone array effect [4] [5].

A novel approach is proposed in this work 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 [6], the directivity of the microphone array was significantly improved. Ohyama, Sasagawab, Takayama and Kobayashia proposed a closed microphone array system with a complex weighting method [7]. In the work 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 presents the simulation results using a circular array composed of fifty-seven microphones to control the sound beam direction, as well as the influence on the sound beam diffusion angle. The optimal weighting values of the microphone arrays corresponding to the specific sound beam could be calculated with the optimization method. The results showed that the directivity of the microphone array could be improved. Also the beam-width of the microphone array could be controlled using the method proposed in the study.

II. THEORY

The delay-and-sum beamforming is a simple and powerful array signal processing algorithm [1]. Assume that a group of M microphone, which is arranged in a uniform linear array (ULA) with an inter-element spacing of d as shown in Fig. 1. An observation point is set in the far-field of the array at an angle, θ with respect to 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:

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

where
τ = (d/c)sin θ is the time delay.
w_n is the weightings for each microphone.
θ is the angle with respect to the axis of the beam.

From Equation (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 th microphone. The time delay τ_0 can be calculated as τ =(d/c)× sin θ_0 , and the array response becomes:

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

Then the equation (2) can be expressed as:

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

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

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

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 equation (1) with w_{am} and ω_a instead of w_m and ω , smilarly, the far-field directivity for primary frequency ω_b with w_{bm} instead of w_{am} . Therefore, the beam pattern can be expressed as:

$$D(\theta) = D_1(k_a,\theta)H(k_a,\theta)D_1(k_b,\theta)H(k_b,\theta) \tag{5}$$

Directivity for one microphone can be expressed as:

$$D(\theta) = 0.5 + 0.5\cos(\theta) \tag{6}$$

By using delay-and-sum beamforming equation (6) becomes:

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

Equation (7) can also be expressed as:

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

This is the general equation for designing directivity of the microphone array to control the beam-width. In general the optimization problem can be defined as follows [9, 10]:

$$\begin{aligned} &\underset{x}{Minimize}\ \ f(x) \\ &\textit{Subject to } c_i(x) = 0 \ \ i = 1, \ldots, m_e \\ &c_i(x) \le 0 \ \ i = m_e+1, \ldots, m, \end{aligned} \tag{9}$$

where $f(x)$ is a nonlinear real scalar function of the optimization parameter vector $x = [x_1, \ldots, x_n]^T$, and $c_i(x)$ are nonlinear scalar constraint functions, which can be written in a vector form as $c(x) = [c_1(x), \ldots, c_m(x)]^T$. $c_i(x)$, $i = 1, \ldots, m_e$ are referred to as equality constraints, and $c_i(x)$, $i = m_e+1, \ldots, m$ are referred to as inequality constraints [9, 10]. The optimal solution for the problem defined in Equation (9) must satisfy a set of equations, known as the Kuhn-Tucker equations [11]. The Kuhn-Tucker equations thus form a necessary condition for the optimality of a solution. Furthermore, when the optimization problem is convex, so that both the objective function f and the constraints c_i shown in Equation (9) are convex, the Kuhn-Tucker equations are also necessary to achieve sufficient conditions for the global minimum point.

The main objective of the optimization method proposed in this study is to control the beam-width of the main lobe and the side lobes. The formulation of the optimization approach proposed in the work for designing directional microphone arrays can be expressed as:

Minimize

$$\left\| D\_(\theta_1) \right\|_2^2 + \left\| D\_(\theta_2) \right\|_2^2$$

Subject to

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

$$\left\| D\_(\theta_3) - D\_(\theta_2) \right\|_\infty > \delta \tag{10}$$

where θ_1 and θ_2 is the angle of the side lobe, θ_3 is the angle of the main lobe, δ is the predefined difference level between the main lobe and side lobe. In this study 50dB is set as the difference level.

By substituting equation (8) into equation (10) the optimization formulation can be expressed as:

Minimize

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

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

Subject to

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

$$\left\| [0.5 + 0.5\cos(\theta_3)] \cdot \frac{1}{M} \sum_{n=0}^{M-1} \omega_n e^{jn2\pi f(\sin\theta_3 - \sin\theta_0)d/c} - [0.5 + 0.5\cos(\theta_2)] \cdot \frac{1}{M} \sum_{n=0}^{M-1} \omega_n e^{jn2\pi f(\sin\theta_2 - \sin\theta_0)d/c} \right\|_\infty \rangle \delta \tag{11}$$

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

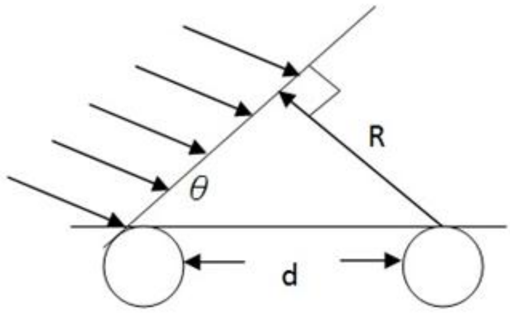

Fig. 1 Time Delay between Microphone Array.

III. PERFORMANCE EVALUATION OF DIRECTIVITY FOR MICROPHONE ARRAY

This section presents the directivity evaluation of the microphone arry generated using the optimization method. A single microphone array consisting of eight microphones is compared with a circular microphone array consisting of 57 microphones. The circular array has a frequency response of 1 KHz. The speed of sound c is 344 ms-1. The beam-width of different angles under 30, 40 and 60 degrees were calculated and compared in the X and Y axis.

Fig. 2 shows the directivity with a single 8 linear microphone array on the X-axis. The beam-width is 30 degrees, and the amplitude difference between the main lobe and side lobe is about 150 db. Fig. 3 shows the directivity with a 57 circular microphone array on the X-axis. The beam-width is 30 degrees, the amplitude difference between the main lobe and side lobe is about 150 db. Fig. 4 shows the directivity with a single 8 linear microphone array on the Y-axis. The beam-width is 30 degrees, and the amplitude difference between the main lobe and side lobe is about 20 db. Fig. 5 shows the directivity with a 57 circular microphone array on the Y-axis. The beam-width is 30 degrees, the amplitude difference between the main lobe and side lobe is about 150 db. From the results above it can be seen that the directivity with a 57 circular microphone array is better than that with a single 8 linear microphone array on the Y-axis. This is because directivity with a 57 circular microphone array could be better controlled with more complicated geometry and more weightings.

Fig. 6 shows the directivity with a single 8 linear microphone array on the X-axis. The beam-width is 40 degrees, and the amplitude difference between the main lobe and side lobe is about 150 db. Fig. 7 shows the directivity with a 57 circular microphone array on the X-axis. The beam-width is 40 degrees, the amplitude difference between the main lobe and side lobe is about 150 db. Fig. 8 shows the directivity with a single 8 linear microphone array on the Y-axis. The beam-width is 40 degrees, and the amplitude difference between the main lobe and side lobe is about 20 db. Fig. 9 shows the directivity with a 57 circular microphone array on the Y-axis. The beam-width is 40 degrees, the amplitude difference between the main lobe and side lobe is about 160 db. From the results above it can be seen that the directivity with a 57 circular microphone array is better than that with a single 8 linear microphone array on the Y-axis. This is because directivity with a 57 circular microphone array could be better controlled with more complicated geometry and weightings.

The 3D directivities for 57 circular microphone arrays with 30 and 40 degree beam-width are analyzed using the delay-and-sum beamforming combined with optimization methods. Figures 10 and 11 show the 3D directivities for 57 circular microphone arrays with 30 and 40 degree beam-width respectively. From the figures we can see that the directivities could be controlled over the X and Y axis. This is since there are more weightings in the 57 circular microphone array and also the geometry is more complicated. Therefore more parameters could be controlled and better directivities could be obtained.

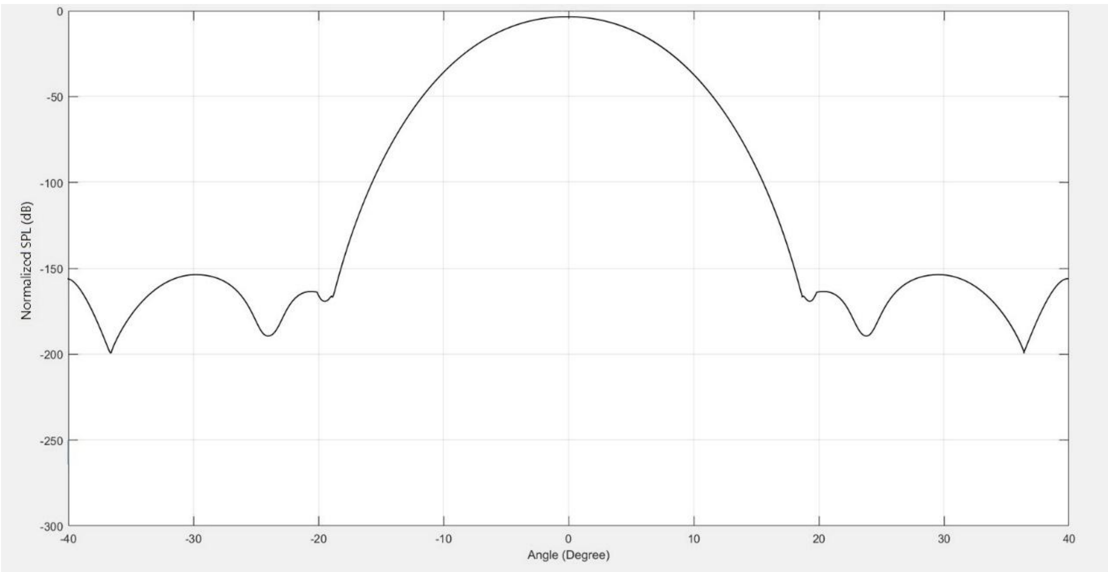

Fig. 2. Directivity for the linear microphone array with 8 microphones for $\theta = 30^0$ using delay-and-sum beamforming combined with optimization method. (X axis)

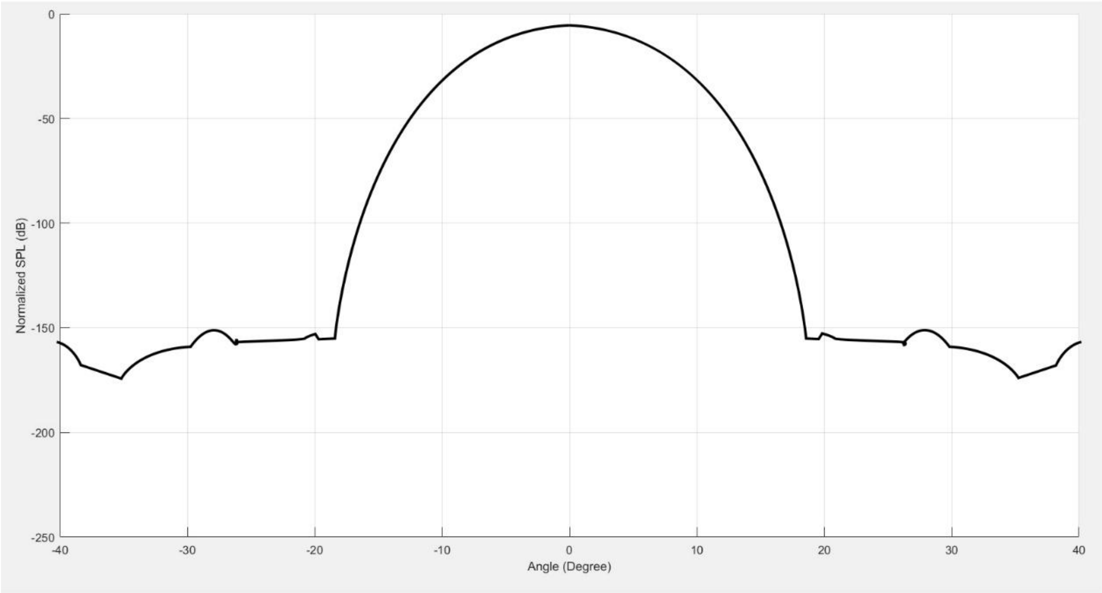

Fig. 3. Directivity for the circular microphone array with 57 microphones for $\theta = 30^0$ using delay-and-sum beamforming combined with optimization method. (X axis)

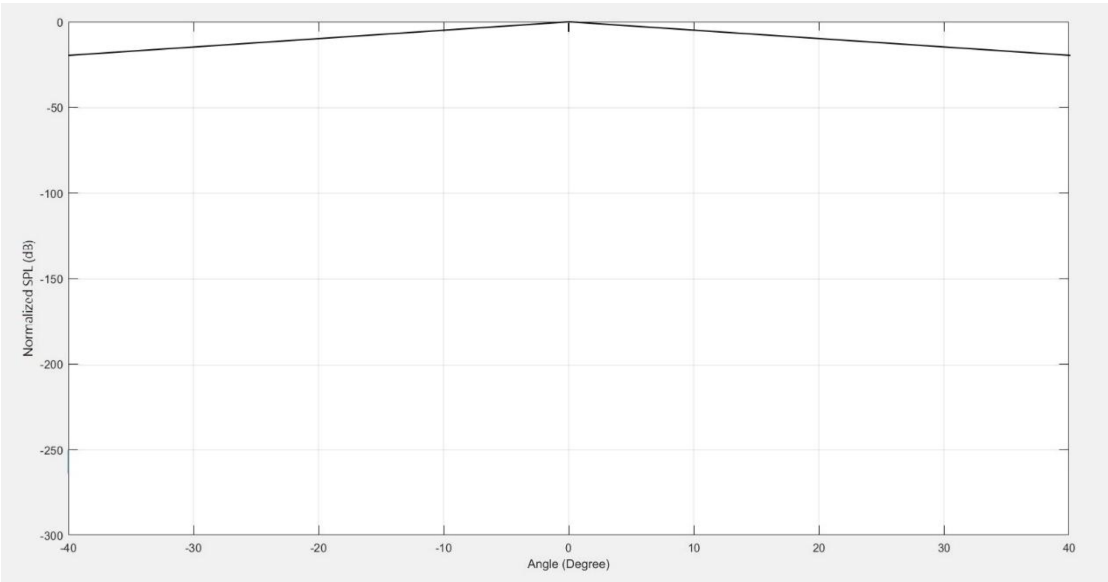

Fig. 4. Directivity for the linear microphone array with 8 microphones for $\theta = 30^0$ using delay-and-sum beamforming combined with optimization method. (Y axis)

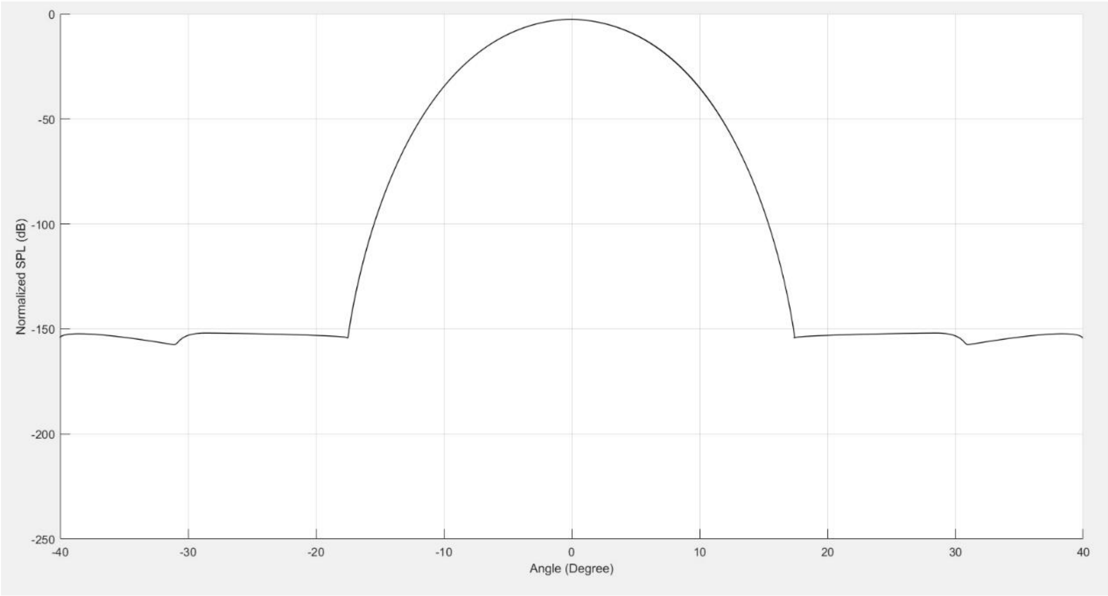

Fig. 5. Directivity for the circular microphone array with 57 microphones for $\theta = 30^0$ using delay-and-sum beamforming combined with optimization method. (Y axis)

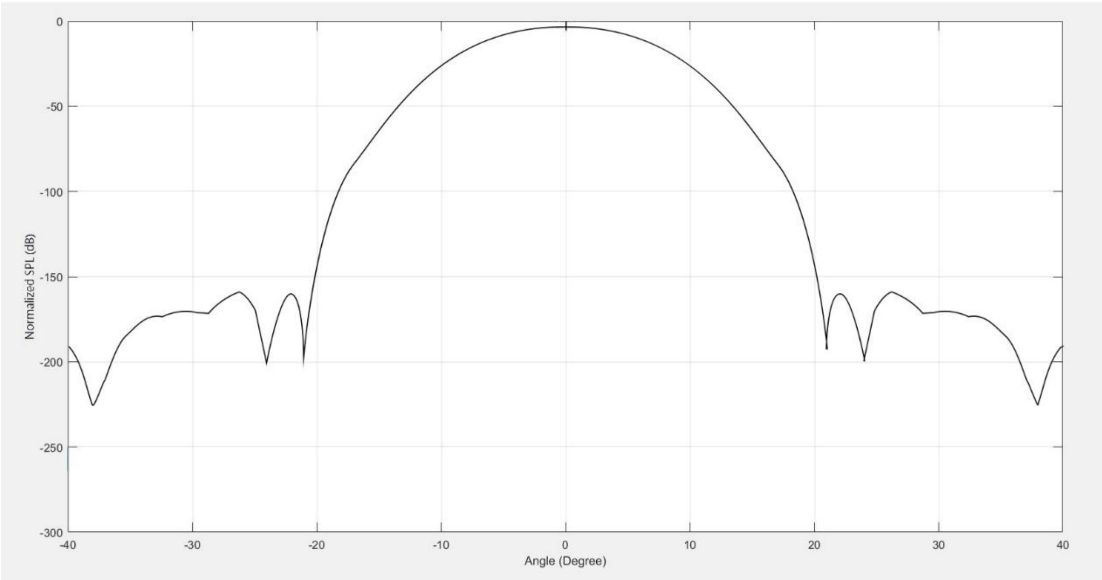

Fig. 6. Directivity for the linear microphone array with 8 microphones for $\theta = 40^0$ using delay-and-sum beamforming combined with optimization method. (X axis)

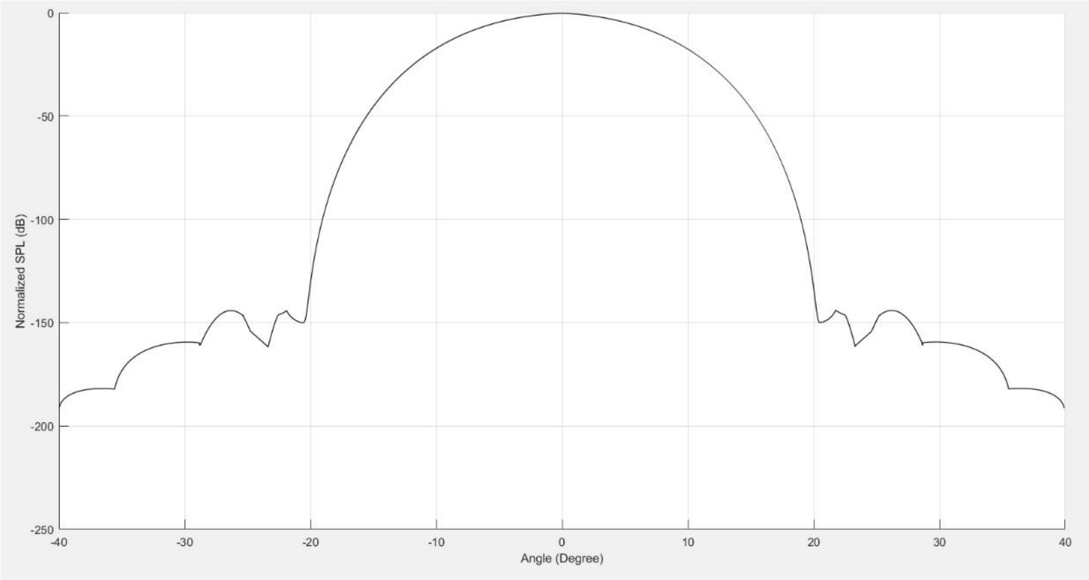

Fig. 7. Directivity for the circular microphone array with 57 microphones for $\theta = 40^0$ using delay-and-sum beamforming combined with optimization method. (X axis)

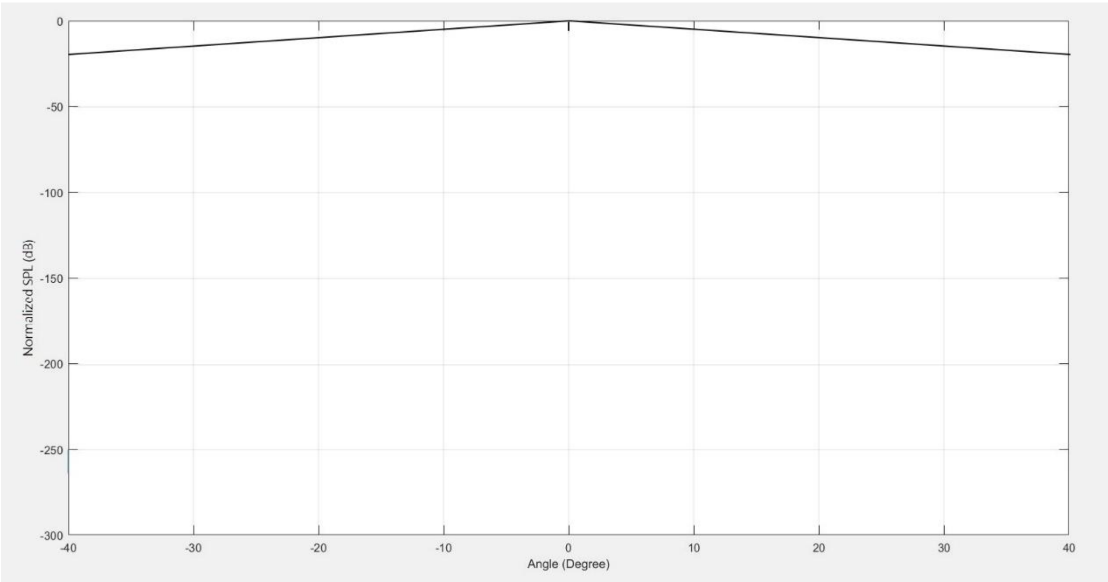

Fig.8. Directivity for the linear microphone array with 8 microphones for $\theta = 40^0$ using delay-and-sum beamforming combined with optimization method. (Y axis)

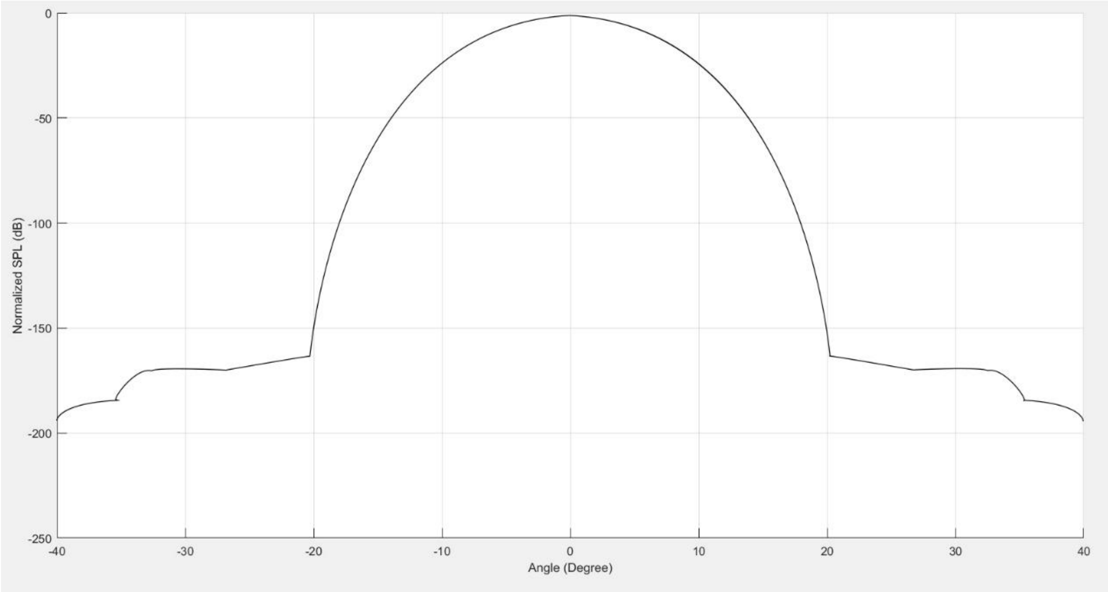

Fig. 9. Directivity for the circular microphone array with 57 microphones for $\theta = 40^{0}$ using delay-and-sum beamforming combined with optimization method. (Y axis)

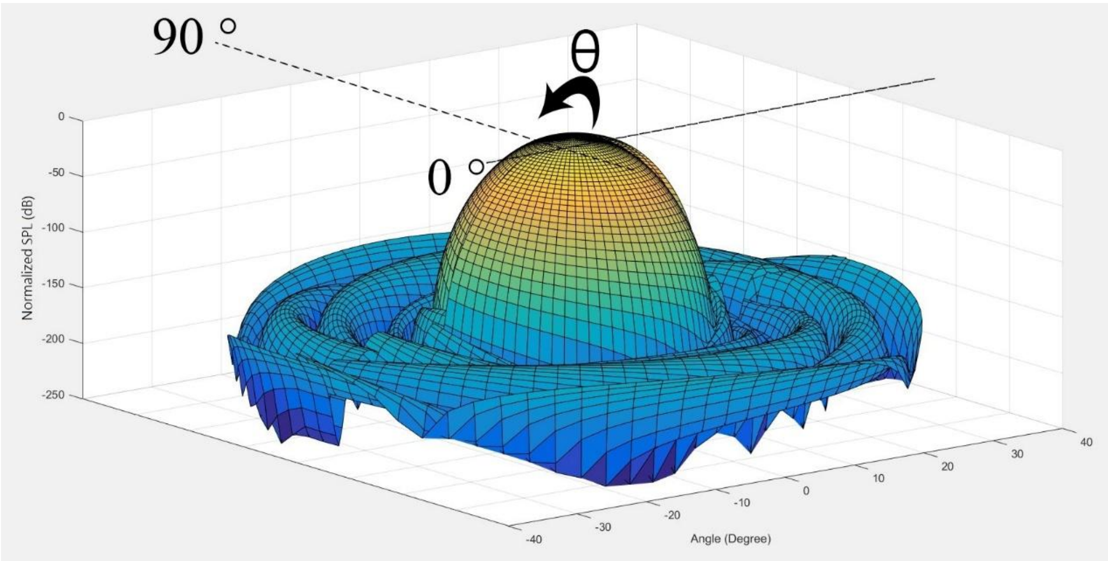

Fig. 10. Directivity for the circular microphone array with 57 microphones for $\theta = 30^{0}$ using delay-and-sum beamforming combined with optimization method.(3D)

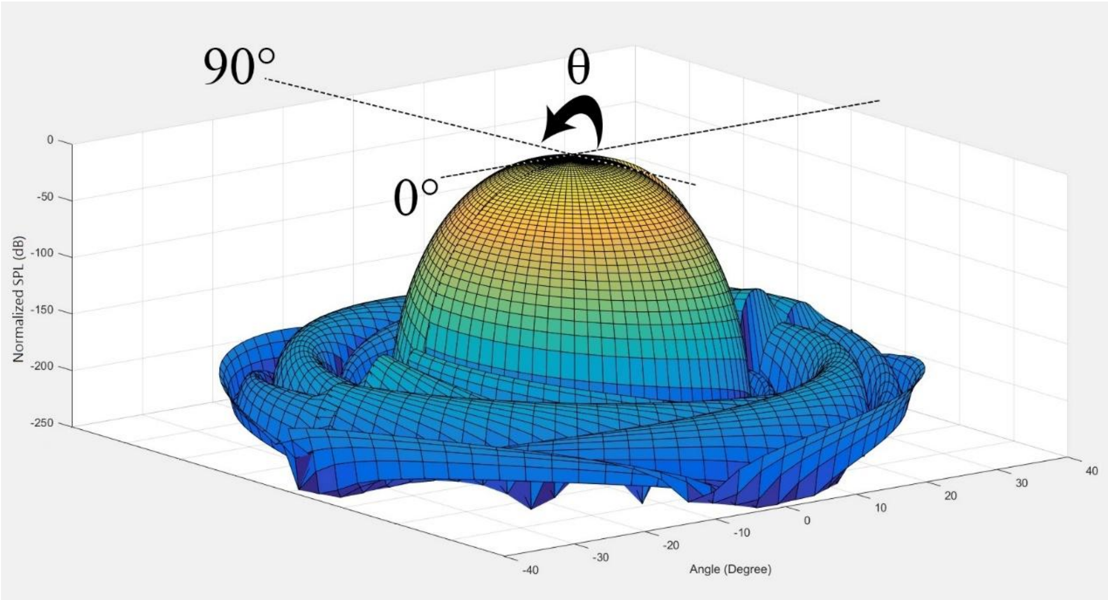

Fig. 11. Directivity for the circular microphone array with 57 microphones for $\theta = 40^{0}$ using delay-and-sum beamforming combined with optimization method.(3D)

IV. CONCLUDING REMARKS

The directivities for the microphone array with 8 and 57 microphones have been evaluated over the X and Y axis in the work. The 3D directivities for the microphone array with 8 and 57 microphones have also been investigated. The results showed that better directivity could be obtained using the circular microphone array with 57 microphones over the X and Y axis. This is due to the fact that the circular geometry is more complicated and there are more weightings in the circular microphone array. Therefore better directivity could be obtained. The beam-width of the microphone array has also been investigated in the work. The results showed that the beam-width of the microphone array could be controlled using the circular microphone array. Therefore the array geometry and optimization method proposed in the work could effectively control the directivity and beam-width of the microphone array.

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