Combining MUSIC Algorithm and Adaptive Beamforming to Improve Online Call Quality

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Abstract. In this paper, proposing a uniform circular microphone array (UCA) model that combines the multiple signal classification algorithm (MUSIC) and the adaptive beamforming technique to improve the quality of online calls. MUSIC algorithm is used to accurately detect the direction of arrival (DOA) of signal sources, while adaptive beamforming using the least mean square (LMS) algorithm can eliminate unacceptable sources and noise. As a result, the UCA system can actively select the desired signal source. Based on simulation results for three narrowband sinusoidal signal sources, the proposed system shows that it meets the requirements of direction detection, changing appropriate adaptive weight values, and limiting the influence of unwanted sources and noise. From there, the desired signal is accurately filtered with consistent filtered power.

Keywords: DOA Estimation · MUSIC Algorithm · Adaptive Beamforming · LMS Algorithm · UCA

1 Introduction

With the global growth of the Internet and communication technologies, online calling has become one of the most popular methods of communication. New innovations and applications need to be launched regularly to improve call quality and user experience, improving sound quality is an important issue. Although simple frequency filters can filter the wideband signals, using an adaptive filter with the adaptive beamforming technique is always preferable when sources with the same or nearly the same frequency need to be filtered. To use this technique, we must first determine the angle at which the signals are received by using the DOA estimation method, which is high-resolution and plays an important role in array signal processing research. Certain common approaches, including the MUSIC algorithm, the minimum variance distortionless response (MVDR) algorithm, and the conventional beamforming (CB) algorithm, can be used to adequately detect the direction of the received signals [1–12, 23, 25]. Under identical input conditions, the simulation shows that the MUSIC algorithm always has a minimum main beamwidth at −3 dB and has a higher peak to average power ratio (PAPR) than the

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CB and MVDR algorithms [7–9]. Adaptive beamforming is one of the most important applications of the LMS algorithm [14–18], which, together with the Wiener filter and the recursive least square (RLS) algorithm, is a well-known optimal filtering technique used in many different fields [14]. Unlike the Wiener filter, which only works in an environment with a priori statistical information, the LMS algorithm is considered stable and effective even in the presence of unknown environmental factors, since it does not require statistical features in the system's operating environment [14]. While the RLS algorithm is based on the least squares method, the LMS algorithm is based on the stochastic gradient descent (SGD) method [14]. Moreover, the LMS algorithm is less complex than the RLS algorithm because it does not require inversion of the correlation matrix of the received signals [14, 19]. The model combining DOA estimation method and the adaptive beamforming technique is a popular model for uniform linear array (ULA) antenna, the performance of this model is demonstrated in [20–25]. However, ULA antenna provides only 180° of coverage in the azimuth plane, while the UCA antenna geometry provides up to 360° of coverage, which is the distinct advantage of UCA over ULA [10–13]. In this paper, we propose to combine the MUSIC algorithm and the adaptive beamforming technique with the LMS algorithm to reduce the influence of unwanted sources and noise on the UCA microphone. From there, optimizing the microphone array's response to the target signal to enhance audio quality for online conversations.

The goal of this research is to provide the basis for developing a compact, high-performance beamforming microphone that can be integrated into smart mobile devices. The simulation results show that the proposed combination of the MUSIC algorithm and the adaptive beamforming technique on a 6-element UCA microphone for three sinusoidal signals in a narrowband frequency does not take much time to determine the direction of the signal sources and then eliminate noise and unwanted signals. The target signal is determined quite accurately and reliably after filtering. In the following sections, the mathematical model of the UCA antenna, the theories of the algorithm, the implementation approach, and the simulation results of the proposal are explained.

2 Signal Model and UCA Configuration

Figure 1 shows the UCA antenna model, which consists of N_e isotropic and in-phase elements uniformly distributed around the circle. The UCA geometry provides a coverage of 360° in the azimuth plane. The radius R of the antenna array is calculated according to Eq. (1), where λ_{max} is the largest wavelength of the received signal:

$$R = \frac{0.5 \times N_e \times \lambda_{\max}}{2\pi} \tag{1}$$

In fact, it is necessary to reduce the dimensions of the microphone to save money while maintaining reliable performance. The number of antenna elements N_e must be greater than the number of received signal sources D. The received signals are narrowband signals and come from the angles $\theta_1, \theta_2, \ldots, \theta_D$, each signal source has a center frequency $f_c = c/\lambda$, where λ is the wavelength of the signal and $c = 340$ m/s is the

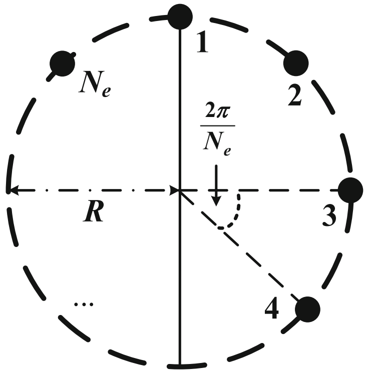

Fig. 1. Geometric structure of UCA antenna

propagation speed of sound in air. The angular coefficient of the signals k_D is calculated by $k_D = 2\pi/\lambda_D$.

If *s(t)* is the amplitude of the received signals, *A(θ)* is the received signal's steering matrix at the array, and *n(t)* is the white Gaussian noise in the received signals, the total received signals at the array is written as:

$$x(t) = A(\theta).s(t) + n(t) \tag{2}$$

where:

$$A(\theta) = [a(\theta_1), a(\theta_2), \ldots, a(\theta_D)]$$
$$= \begin{bmatrix} e^{jk_1 R\cos(\theta_1)} & e^{jk_2 R\cos(\theta_2)} & \ldots & e^{jk_D R\cos(\theta_D)} \\ e^{jk_1 R\cos(\theta_1 - 2\pi\frac{1}{Ne})} & e^{jk_2 R\cos(\theta_2 - 2\pi\frac{1}{Ne})} & \ldots & e^{jk_D R\cos(\theta_D - 2\pi\frac{1}{Ne})} \\ \vdots & \vdots & \ddots & \vdots \\ e^{jk_1 R\cos(\theta_1 - 2\pi\frac{Ne-1}{Ne})} & e^{jk_2 R\cos(\theta_2 - 2\pi\frac{Ne-1}{Ne})} & \ldots & e^{jk_D R\cos(\theta_D - 2\pi\frac{Ne-1}{Ne})} \end{bmatrix} \tag{3}$$

3 MUSIC Algorithm

The MUSIC algorithm is based on the decomposition of the covariance matrix of signals received from antenna array without scanning its beam along angles in space. Where X is the set of signals received by each element of the antenna array, the covariance matrix R_x is represented as:

$$R_x = E\left[X.X^H\right] \tag{4}$$

Since Eq. (2) and Eq. (4):

$$R_x = A.E\left[s.s^H\right].A^H + E\left[n.n^H\right] = A.R_s.A^H + R_n \tag{5}$$

where $R_s = E[s.s^H]$ and $R_n = E[n.n^H]$ are the corresponding covariance matrixes of signals s and noise n, and the matrix A^H is the conjugate transpose matrix of matrix A. The matrix R_n is described as:

$$R_n = \sigma^2.I \tag{6}$$

where I denotes the identity matrix and σ^2 represents the noise variance. Analyze the R_x matrix to obtain D large eigenvalues and $Ne - D$ extremely small eigenvalues for signals and noise, respectively. For each eigenvalue, the matrix space is divided into two subspaces: the signal space and the noise space, which respectively include the signal eigenvector $a(\theta_l)$ and the noise eigenvector E_n, with $l = 1, 2, \ldots, D$. With $a(\theta)$ is the steering matrix of array, scan θ in the range from -180° to 180° according to Eq. (7) to obtain the spectrum P_{music} containing peaks corresponding to the angles of incidence $\theta_1, \theta_2, \ldots, \theta_D$ of the signal sources for the UCA:

$$P_{music}(\theta) = \frac{1}{a^H(\theta).E_n.E_n^H.a(\theta)} \tag{7}$$

4 Adaptive Beamforming

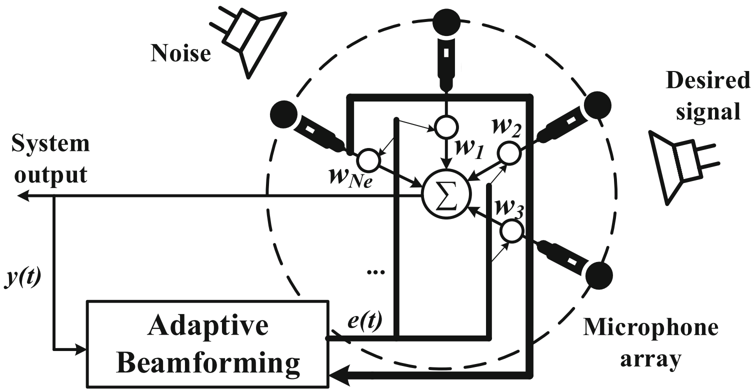

Fig. 2. Structure of Adaptive Beamforming system using LMS algorithm

Figure 2 shows the construction of an adaptive beamforming system on the UCA using the LMS algorithm. Re-express the total received signals by the antenna array using the following equation:

$$x(t) = A(\theta_0).s_0(t) + \sum_{i=1}^{D-1} A(\theta_i).s_i(t) + n(t) \tag{8}$$

where, $A(\theta_0)$, $s_0(t)$ are correspondingly the desired signal's steering vector and amplitude; $A(\theta_i)$, $s_i(t)$ are the i^{th} noise's steering matrix and amplitude, respectively, with $(i = 1, 2, \ldots, D - 1)$; and $n(t)$ is the white Gaussian noise included inside the received signals.

The output response of the UCA antenna is written as:

$$y(t) = w^H.x(t) \tag{9}$$

where w means the adaptive weight vector, and $x(t)$ denotes the total received signals by the UCA. If $d(t) = s_0(t)$ is the desired signal at time t, an error $e(t)$ occurs as:

$$e(t) = d(t) - y(t) = d(t) - w^H.x(t) \tag{10}$$

The mean square error (MSE) is minimized when the adaptive weight vector w has the optimal value, which is the error $e(t)$ used to calibrate w using the SGD method. The following equation is used to calculate the updated value of w at time $t + 1$:

$$w(t+1) = w(t) + \mu.x(t).e(t) \tag{11}$$

where μ is the learning rate, which governs the system's convergence speed and accuracy. Normally, it is a constant that is chosen within the given range of $0 < \mu < 2/\lambda_{max}$. However, choosing $\mu < \; < 0.01$ helps enhance the system's accuracy throughout processing. The following equation shows the response spectrum of the LMS algorithm on the UCA:

$$p_{LMS}(\theta) = \left| w^H.a(\theta) \right| \tag{12}$$

5 Evaluate the Effectiveness of the Proposal Through Simulation

5.1 Simulation Setups

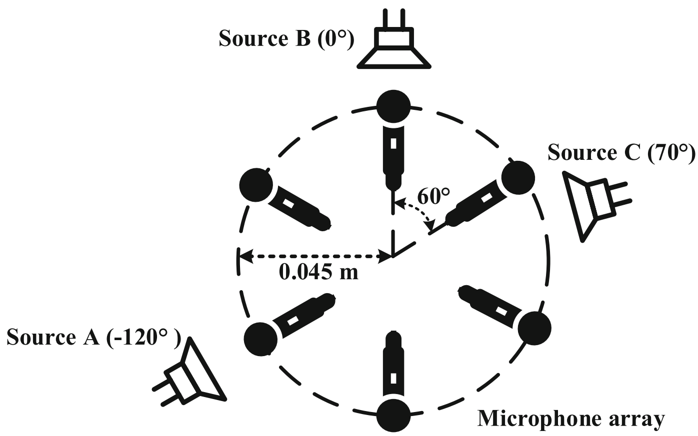

Fig. 3. General simulation model

Figure 3 shows the general simulation model of the system, using a UCA microphone with six elements equidistant from 60°. The signal sources for the simulation consist of three narrowband sinusoidal signals with the same amplitude of 1 mV, signal A at −120°, signal B at 0°, and signal C at 70°. This configuration takes full advantage of the UCA in terms of coverage.

Choosing $SNR = 30$ dB; $R = 0.045$ m for the radius of the microphone array helps to satisfy the requirement of Eq. (1) and optimize the dimensions for the actual devices; $F_s = 48$ kHz – this is the standard sampling frequency, which is sufficient to regenerate the sound in near-original quality and widely used in research on digital signal processing and realism; and $\mu = 0.001$ to ensure stability and accuracy in processing. The center frequency of each signal is $f_c = c/\lambda$.

Matlab software is used for this processing and simulation. First, calculate the covariance matrix R_x using Eq. (4) to identify the direction of the signals using the MUSIC algorithm. Determine the eigenvalues and eigenvectors of the matrix R_x by applying $[u, v] = eig(R_x)$. Create a MUSIC directed spectrum plot using the eigenvalues and eigenvectors according to Eq. (7). After determining the direction of the signals, use the LMS algorithm to eliminate unacceptable sources and noise so that the desired signal is filtered. Adjust the weight vector w indefinitely until the optimal value of Eq. (11) is reached. According to Eq. (12), the spectrum plot represents the response of the LMS algorithm to the microphone array. Assess the effectiveness of the system in filtering audio signals by comparing the value of PAPR in the power spectrum of the filtered signal with SNR = [5:50] dB.

5.2 Simulation Results

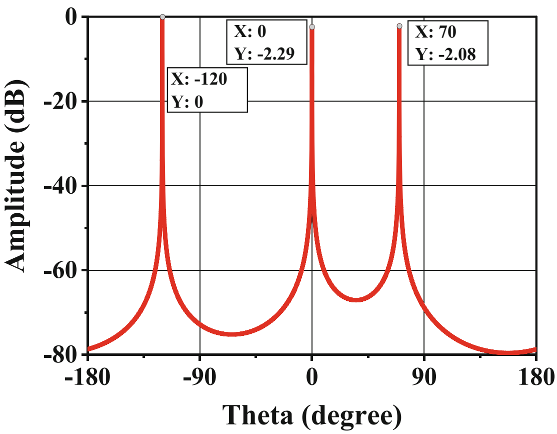

Fig. 4. Estimation the DOA of signal sources

Figure 4 shows the plot of the directional spectrum for the microphone array used by the MUSIC algorithm to detect the signal direction. The main beamwidth at -3 dB is less than 0.05°, so the directions of the three signal sources can be easily identified. As a result, with SNR = 30 dB, this estimated result shows the performance of the system. Use the MUSIC algorithm's above incidence angle estimation results to continue. Consider sources A and B as unwanted signals and source C as the desired signal. Then, sample the signals to change the weight vector *w* before examining the response spectrum of the LMS algorithm on the microphone array in Fig. 5.

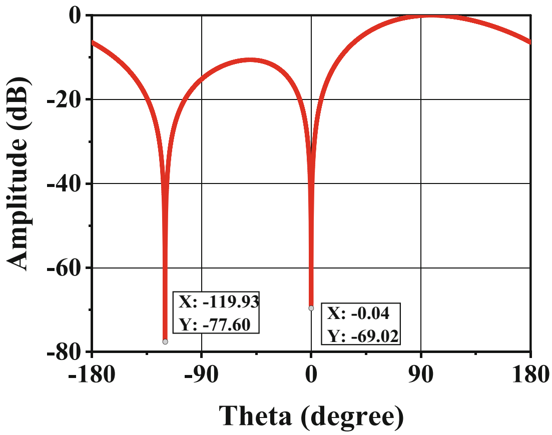

Fig. 5. LMS algorithm response spectrum

Figure 5 shows the response spectrum of the LMS algorithm for the microphone array, which contains two peaks in the negative direction with amplitudes of less than − 65 dB, corresponding to two noises. There is also a large region behind 0°, showing that the two noises were reduced and only the desired signal was retained. Figure 6 shows the mixed signal *x(t)*, the desired signal *d*(*t*), and the output signal *y(t)* in the time range from 0.4 s to 0.45 s. The result is that after convergence, the system has virtually reduced the noise while retaining only the desired signal.

Figure 7 illustrates the value of adaptive weight *w* in the time domain for each element of the microphone array; it increases dramatically throughout the time the system is converging before optimization with the LMS algorithm. Figure 8 shows the comparable error *e(t)* after the adaptive weights are updated. The system has a convergence time of less than 0.15 s, error values of less than 0.07 mV, and the value of *w* at each array element is saturated. After convergence, the weight *w* moves within a very small range and is considered stable for the duration of the operation.

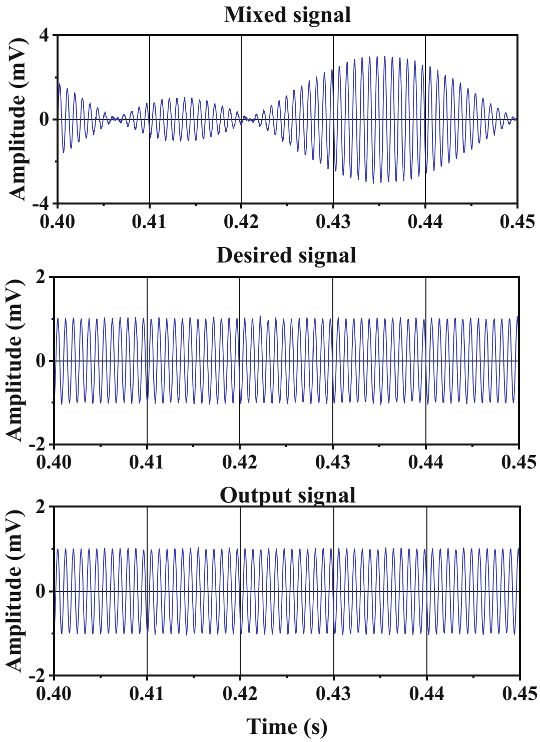

Fig. 6. Mixed signal, desired signal and output signal in the time domain

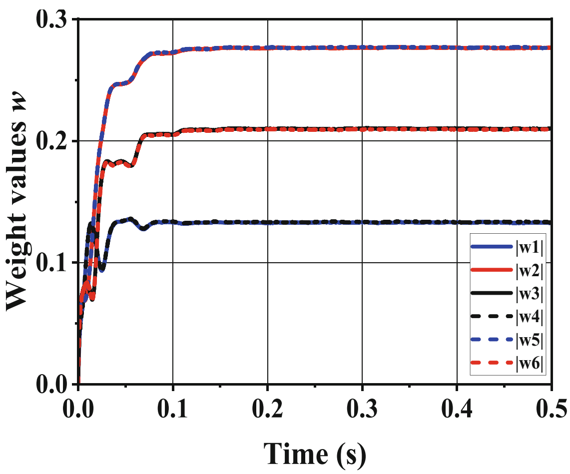

Fig. 7. Weight values in the time domain

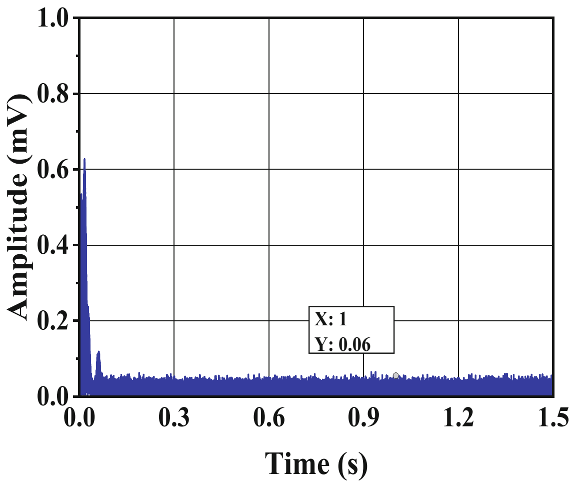

Fig. 8. Amplitude error in the time domain

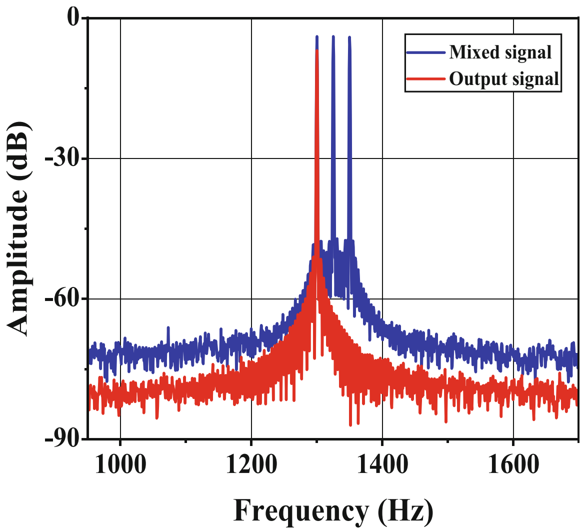

Fig. 9. Power of mixed signal and output signal in the frequency domain

The power spectrum in the frequency domain of the mixed signal *x(t)* and the output signal *y(t)* are shown in Fig. 9. It can be seen that the signals before filtering contains both desired signal and undesired signals and noise. The filtered signal has two principal components: The center frequency is about 1300 Hz with a maximum amplitude of about −7 dB, and the average noise power is less than -80 dB, which means that the noise is not overly significant in the important frequency range of the system and the performance of the system is excellent. Change the SNR from 5 dB to 50 dB to examine the performance of the system when filtering the audio signal through the PAPR ratio

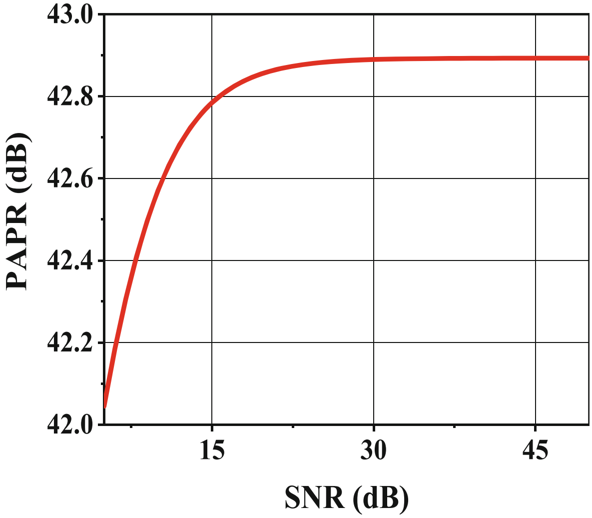

Fig. 10. PAPR versus SNR

on the power spectrum of the filtered signal. Figure 10 shows how the PAPR starts at SNR = 5 dB with about 42.04 dB and steadily progresses before reaching optimization at SNR = 30 dB with about 42.89 dB. This shows that the system can run consistently from SNR = 30 dB.

6 Conclusion

In this research, a small 6-element UCA microphone is proposed to filter audio signals for online calls by combining the MUSIC algorithm and adaptive beamforming technique using the LMS algorithm. The above simulation results and performance evaluation for narrowband sinusoidal signals show that the signal incidence angle estimation process has small beamwidths and small errors, which creates favorable input conditions for signal filtering in the next step. The performance of the LMS algorithm is shown by its fast adaptation speed and simple and effective filtering of the desired signal. The power of the signal after filtering is not significantly lost. This makes it possible to design a microphone model for practical applications. The limitation of this proposal is that the MUSIC algorithm works correctly only when the number of signal sources to be determined is less than the number of elements in the microphone array. In addition, it is necessary to continue to perform simulations and experiments under many different audio signal conditions, such as effects of multipath noise, color noise, etc., and to combine Deep Learning and AI to automate the identification of noise on the fly.

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