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Research Article

Enhancing sound quality through audio beam formation: A MVDR algorithm approach with linear microphone arrays

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ABSTRACT

Achieving high-quality sound in noisy environments is a significant challenge in audio applications, particularly in areas such as communications, broadcasting, and healthcare. This study explores the use of the Minimum Variance Distortionless Response (MVDR) algorithm in conjunction with a linear microphone array to improve audio beamforming. The objective is to enhance the signal-to-noise ratio (SNR) of desired signals while minimizing interference and reverberation. The MVDR algorithm is applied in the research using MATLAB to create directional beams that help isolate the target signal, and background noise is reduced. Comparative analysis against the traditional delay-and-sum beamformer shows that the MVDR scheme is much better in SNR, particularly in the multiple sources of interference environment. The major findings are that as the number of microphones increases to 20, the mean squared error (MSE) reduces by up to 33.8 dB and that the microphones work best when the spacing between them is carefully adjusted. These results indicate that MVDR-based beamforming would be able to offer excellent solutions to improving sound clarity in numerous realistic scenarios, including teleconferencing, sonar, and medical imaging.

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INTRODUCTION

The The act of combining multiple signals captured from the sensors in a sensor array is the main concept in beam forming technique. It is a signal processing technique used in sensor arrays for directional signal transmission or reception. This technique basically approaches the problem from a spatial point of view [1]. It combines the elements in a phased array in such a way that signals at particular angles experience constructive interference while others experience destructive interference. Beam forming can be used at both the transmitting and receiving ends in order to achieve spatial selectivity. This has improved the directivity of elements as compared to the unidirectional reception/ transmission [2].

Beam formation has been studied in many areas such as radar, sonar, seismology and wireless communications, to name but a few. It can be used for a variety of purposes, such as detecting the presence of a signal, estimating the direction of arrival, and enhancing a desired signal from its measurements corrupted by noise, competing sources and reverberation [3]. Typical use of beam forming arise in RADAR, SONAR, communications, imaging, geophysical exploration, biomedical and also in acoustic source localization [4].

In acoustics this concept can be adopted to form a noise free beam of the incoming corrupted signal by using array of microphones. Beam forming combines the output audio signals of the microphone array in such a way that it will make the microphone array to act as a highly directional microphone [5]. Actually, beam forming has been adopted by the audio research society, mostly to separate or extract speech for noisy environment [6]. A microphone array is used to form a spatial filter which can extract a signal from a specific direction and reduce the contamination of signals from other directions. In other words, beam forming provides a "listening beam" which points to a particular sound source while often filtering out other sounds [7, 8].

Spatially propagating signals encounter the presence of interfering signals and noise signals. If the desired signal and the interfering signals occupy the same temporal frequency band, then temporal filtering cannot be used to separate the signal from the interferers. However the desired and the interfering signals generally originate from different spatial locations. This spatial separation can be exploited to separate the signals from the interference using a beam former [9, 10]. A beam former consists of an array of sensors in a particular configuration. The output of each sensor is properly filtered and the filtered outputs of all the sensors are added up. Typically a beam former linearly combines the spatially sampled waveform from each sensor in the same way a FIR filter linearly combines temporally sampled data [11, 12].

When low frequency signals are used, then an array of sensors can synthesize a much larger spatial aperture than a single physical antenna or sensor [13, 14]. A second very significant advantage of using an array of sensors is the spatial filtering versatility offered by discrete sampling. In many applications it is necessary to change the spatial filtering function in real time to maintain effective suppression of interfering signals [15]. Changing the spatial filtering function of a continuous aperture antenna is impractical. Use of an array of sensors makes this possible to implement [16, 17].

In this paper, basic fundamentals of beam formation, certain beam formation techniques, microphone array geometries and effect of various factors on beam former's performance are discussed in subsequent chapters. The aim is to study traditional delay and sum beam former and MVDR (Minimum Variance Distortion less Response) beam former [18, 19]. It consists of performance checking of beam former output by varying various parameters. In this Paper, sound source localization algorithm is not implemented which is required to steer the beam to that direction. Instead of that, certain direction for sound source and interference signal is assumed and then beam formation is implemented in MATLAB [20, 21].

Beamforming is a method that is widely applied in communications and radar systems in order to generate a focused beam of the required signal and minimize interference of background noise [22]. This is also reflected in the science of acoustics, in which good sound in ambient noise and reverberation is very essential. Signal integrity is also paramount in most of the voice applications, such as communications, broadcasting, and healthcare. However, traditional single-channel audio systems often have difficulty improving signal-to-noise ratio (SNR) and improving signal quality [23, 24]. Therefore, techniques such as sound beam generation are needed to solve these problems. Beamforming based on the use of microphone arrays with geometry designs has been shown to have great potential in amplification of the desired signal and reduction of unwanted noise and echo [25, 26]. This article investigates electronic beam forming using the minimum displacement distortion-free response (MVDR) algorithm and linear microphone array configuration (implemented in MATLAB) [27, 28]. This research aims to improve signal clarity and signal-to-noise ratio by focusing on the sound field and optimizing the response, thus demonstrating the effectiveness of beam forming technology in various fields such as sonar, radar, wireless communications, astronomy, etc. and biomedicine [29, 30].

The whole research is organized as follows: the introductory information about the beam formation is presented in Section I. Whole system is presented in section II. In section III deals with the concept of wave front. In section IV and V discuss the delay calculation and MVDR respectively. Finally section VI and VII gives result and conclusion.

Mono audio systems cannot improve signal-to-noise ratio (SNR), making it difficult to achieve good sound quality in noisy environments. This work aims to use the MVDR algorithm and linear microphone array in MATLAB to use electronic beam forming to improve signal quality. By optimizing response to local audio, this method aims to improve the desired signal while reducing noise and reverberation [31].

The ultimate aim of the work is to create and apply an electronic beamforming system based on the Minimum Variance Distortionless Response (MVDR) algorithm and a linear microphone array in MATLAB. The objective of this system is to increase the quality of the desirable audio signals by increasing the signal-to-noise ratio (SNR) in the harsh acoustic surroundings. Additionally, it seeks to optimize the directional response through effective sound source localization, thereby achieving clearer signal transmission while effectively suppressing ambient noise and reverberation [32].

OVERALL SYSTEM

Overall system consists of four parts as shown in figure 1, which are microphone array, sound source localization, beam formation and post filtering. Microphone array geometry can change according to the application. Microphone array has circular, spherical, circular, cylindrical geometry. Here, linear geometry has been selected. The input to the microphone array is the sound signal and interference signal. Its output will be 'S' number of mixed signal which is addition of desired sound signal and interference signal. These signals differ with each other by an amount of delays which are calculated in equation (3.1). These are given to the Fourier transform block and source localization block for calculating the angle (θ_d and θ_i). Sound source localization plays the important role to find direction of sound source and interference. It can calculate by various methods like GCC-PHAT, GCC-NONLIN, MUSIC, cross correlation methods which is not the scope of this work.

On applying the microphone output signals and angles which are calculated by source localization to Fourier transform block, it gives the array manifold vector for desired signal (AMV_d) and interference signal (AMV_ki). These array manifold vectors are required for calculating the MVDR weight vector which are shown in equation (3.9). It is a vector of size S×1. Finally beam formed output signal which is closer to desired signal along with the microphone output signals is applied to post processing block for removing the additional AWGN noise from beam formed signal. In addition to the main features summarized in Figure 1, the system also features significant adaptive filtering in addition to beamform devices. This stage dynamically adjusts the filtering parameters according to the characteristics of the input signal and the required signal. The system can also adapt to the changing environments by incorporating a mechanism of adaptive filtering and thus directly enhancing the beamforming process. This modification increases process versatility and efficiency, providing good performance in different noise environments.

Even though the linear geometry was adopted in the current study, one can design a system that will support various other geometries, which may be circular, spherical, or cylindrical geometry, to fit various requirements. The geometry is a unique advantage in each in spatial coverage, directional sensitivity, and signal capture efficiency, allowing solutions to be found in selected applications. An intermediate step that makes it easier to convert the microphone output signal to frequency.

This modification allows for a more focused analysis of the properties of the signal and helps extract important features for further processing. Using frequency domain analysis, the system will gain insight into signal strength, which will help optimize beamforming performance and increase signal fidelity. and frequency analysis allows solving problems related to environmental noise, interference and flyback. This technique not only improves signal quality, but also makes it flexible and versatile, making it suitable for a variety of applications, from communications to biomedical imaging.

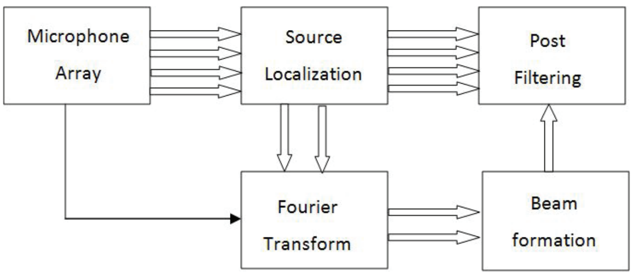


Figure 1. Overall block diagram of system.

CONCEPT OF WAVE FRONT

Microphone Array with linear array geometry is discussed here for generating a microphone output. It has two types of concept i.e. near field system and far field system.

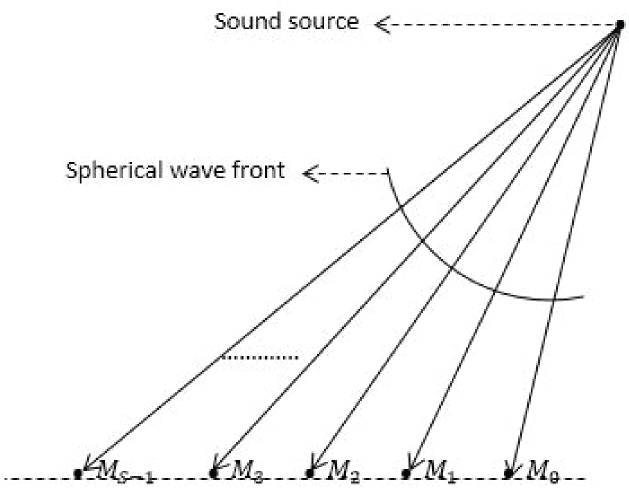


Figure 2. Spherical wave front generated by near field system.

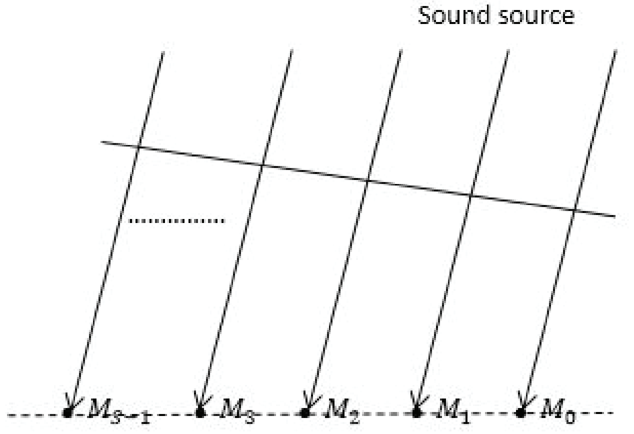


Figure 3. Plane wave front generated by far field system.

If distance between two microphones is greater than distance between sound source and centre of microphone array then it is called as near field system. In near field system circular wave front are generated which is shown in Figure 2.

If distance between two microphones is less than distance between sound source and centre of microphone array then it is called as far field system. In far field system plane wave front are generated which is shown in Figure 3.

For conference or indoor application which is required for this work, distance between sound source and centre of microphone array is greater than distance between two microphones. So, it generates a plane wave front. Hence, a far field pattern is assumed throughout this work and linear microphone array geometry can be used for generation of number of outputs.

Understanding the concept of wave front propagation is important for optimizing the performance of the microphone array, especially when accuracy is critical. In addition to the difference between near-field and far-field, the geometry of the microphone array profoundly affects the properties of the generated wave front.

For example, linear arrays are frequently used in meetings or home environments, while other geometries, such as circular or spherical arrays, offer special advantages in many applications. The distance between the center of the microphone array is generally greater than the distance between the microphones, and the remote system dominates.

This configuration results in a wide plane as shown in Figure 3. The choice of microphone array geometry plays an important role in shaping the wave front characteristics and therefore the performance of the system. Linear arrays offer major advantages in ease of deployment, cost effectiveness, and directional accuracy, making them ideal for applications requiring focused illumination in an area.

The microphone array will be adjustable to the dynamic acoustic environment using sophisticated algorithms and dynamic filtering techniques to achieve consistent performance across a wide range of environments, from the conference room to the concert. The principle of propagation can be applied in the design of microphone arrays in order to optimize their applications. With the understanding of the interaction between array geometry and signal processing algorithms relative to the environment, engineers are able to design and deploy microphone arrays to record real field scenes in an efficient way.

Delay Calculation

In this section we will examine the complex details of signal propagation and delay calculations in linear microphone array geometries. With the center line of “S” microphones, each at a distance “d” from each other, the system captures the signal with the time difference determined by the angle of arrival (α signal). The spacing of consecutive microphones is important for capturing and processing the feedback signal. Taking into account the speed of sound in air (c) and the cosine angle of arrival, the delay time of the microphones will be calculated accurately, as shown in figure 4.

The wavelength (λ) of the signal is important in determining the distance between microphones. This relationship allows the signal to be captured and processed in the array. Moreover, the angle of incidence (α) is closely related to the wave number (K), which means that a change in the angle of incidence will cause a change in the transmission of the wave number and therefore the calculation will be slow. The leaf converts the microphone array output from the time domain to the frequency domain.

This transformation provides a comprehensive analysis of the characteristics of the signal in the frequency range and thus helps extract important features for the next process. The resulting frequency representation of the microphone array output allows signal processing algorithms to be more efficient and optimized to improve signal quality and achieve specific performance requirements.

Suppose there are ‘S’ number of microphones in a linear array geometry i.e. M_0, M_1, M_2, ………M_{S-1}. F is a frequency of signal i.e. 3.3 KHz.‘d’ is a distance between consecutive microphones. ‘c’ is a speed of sound in air i.e. 330 m/s. Assume that at the first microphone M_0 signal $f(t-\tau_0)$ comes at zero delay. So, output of first microphone is $f(t-\tau_0)$ with $\tau_0 = 0$.

By using a linear geometry which is shown in Figure4 delay between first M_0 and second microphone M_1 is $(d * \cos\theta)/c$. Output at second microphone is $f(t-\tau_1)$, here τ_1 is time delay between first and second signal i.e. $(d * \cos\theta)/c$ (by using geometry of Figure4). Delay between first M_0 and third microphone M_2 is $(2d * \cos\theta)/c$. Output at third microphone is $f(t-\tau_2)$, here τ_2 is time delay between first and third signal i.e. $(2d * \cos\theta)/c$ (by using geometry of Figure 4). Delay between m, n microphones is calculated by equation (1)

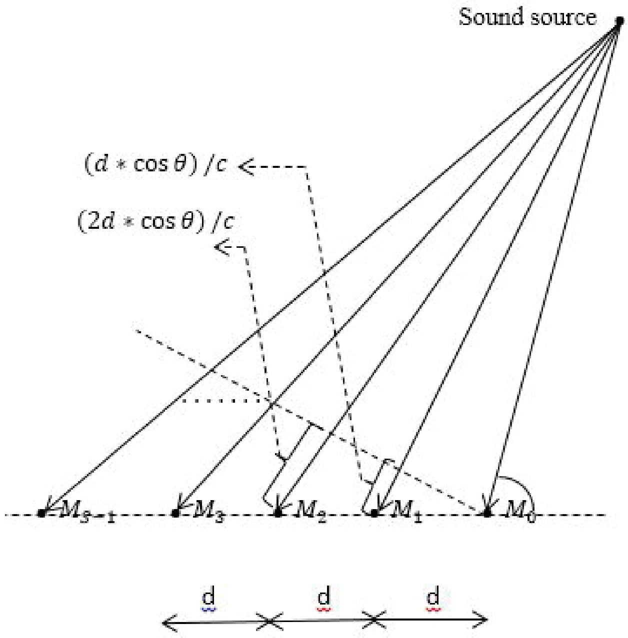


Figure 4. Delay calculation by considering plane wave front.

$$\tau_{m,n}(\theta) = \frac{d_{m,n} * \cos\theta}{c} \tag{1}$$

Here, delay is a function of θ (i.e. angle of arrival direction). Distance between two microphone is adjusted by frequency of sound 'f' and wavelength of signal ' λ'. Distance between two microphones is calculated by

$$\mathrm{d} = \frac{\lambda}{\mathrm{factor}}$$

Where,

$$\lambda = \frac{\mathrm{c}}{\mathrm{f}}$$

In fig. 4 angle θ is a function of 'K' wave number i.e. angle of arrival changes as wave number changes. f(t) is a microphone array output which is calculated by equation (2) and f(t) is a vector quantity.

$$\vec{f}(t) = [\mathrm{f}(\mathrm{t}-\tau_0), \mathrm{f}(\mathrm{t}-\tau_1), , \dots\dots\dots \mathrm{f}(\mathrm{t}-\tau_{S-1})] \tag{2}$$

Converting equation 2 into frequency domain by applying Fourier transform, $\vec{\mathrm{f}}(\mathrm{t})$ in frequency domain is

$$F(\omega) = F.T.\{f(t)\}$$
$$\vec{F}(\omega) = \left[F(\omega)e^{-j\omega\tau_0}, F(\omega)e^{-j\omega\tau_1}, \dots\dots\dots F(\omega)e^{-j\omega\tau_{S-1}}\right] \tag{3}$$
$$\vec{F}(\omega) = F(\omega)\left[e^{-j\omega\tau_0}, e^{-j\omega\tau_1}, , \dots\dots\dots e^{-j\omega\tau_{S-1}}\right]$$

By considering,

$$\vec{V}(k,\omega) = [e^{-j\omega\tau_0}, e^{-j\omega\tau_1}, \dots\dots\dots\dots e^{-j\omega\tau_{S-1}}]$$

$V(k,\ \omega)$ is vector quantity which is function of wave number and frequency

$$\vec{F}(\omega) = F(\omega) \times \vec{V}(k,\ \omega)\, pp \tag{4}$$

Minimum Variance Distortionless Response (MVDR)

If $\vec{\mathrm{X}}(\omega)$ denotes the vector of frequency domain signal for all microphones, and $\vec{\mathrm{Y}}(\omega)$ is the frequency domain output of array, then the operation of the beam formation can be represented as,

$$\vec{Y}(\omega) = W^H(\omega) \times \vec{X}(\omega) \tag{5}$$

Where W(ω) is a vector of frequency dependent sensors (microphone) weights. The difference between various beam formation designs are determined by the specification of the weight vector W(ω). The simplest beam formation is a delay and sum (DS).time aligns the signals for the plane wave arriving from look direction by setting

$$W_{DS} = \frac{\vec{V}(k,\omega)}{S} \tag{6}$$

Substituting $\vec{\mathrm{X}}(\omega) = \vec{\mathrm{F}}(\omega) = \mathrm{F}(\omega) \times \vec{\mathrm{V}}(\mathrm{k},\omega)$ in (5)

$$\vec{Y}(\omega) = W_{DS}^H * F(\omega) \times \vec{V}(k,\omega) = F(\omega) \tag{7}$$

From equation (7), output of the array is equivalent to original signal in absence of any interference or distortion. In general equation (7) will be true for any weight vector achieving

$$W^H(\omega) \times \vec{V}(k,\omega) = 1 \tag{8}$$

Equation (8) shows the condition for distortion less. To improve upon noise suppression performance provided by DS beam formation, it is possible to adaptively suppress specially correlated noise and interference N(ω), which can be achieved by adjusting the weight of beam former, so as to minimize the variance of noise and interference at the output. Weight vector for MVDR algorithm can be calculated by equation (9)

$$W_{MVDR}^H(\omega)\,\frac{V^H(k,\omega)\times\Sigma_{\mathbf{N}}^{-1}(\omega)}{V^H(k,\omega)\times\Sigma_{\mathbf{N}}^{-1}(\omega)\times V(k,\omega)} \tag{9}$$

Where, $\Sigma_N(\omega) = \epsilon\{N(\omega) \times N^H(\omega)\}$ is noise covariance and $\epsilon\{.\}$ Is the expectation operator. In practice $\Sigma_N(\omega)$ is computed by average or recursively updating the covariance matrix. Weight vector obtained under these conditions corresponds to the minimum variance distortion less response (MVDR) beam formation. If N(ω) is a single plane interference with wave number k_1 and has the spectrum

Where,

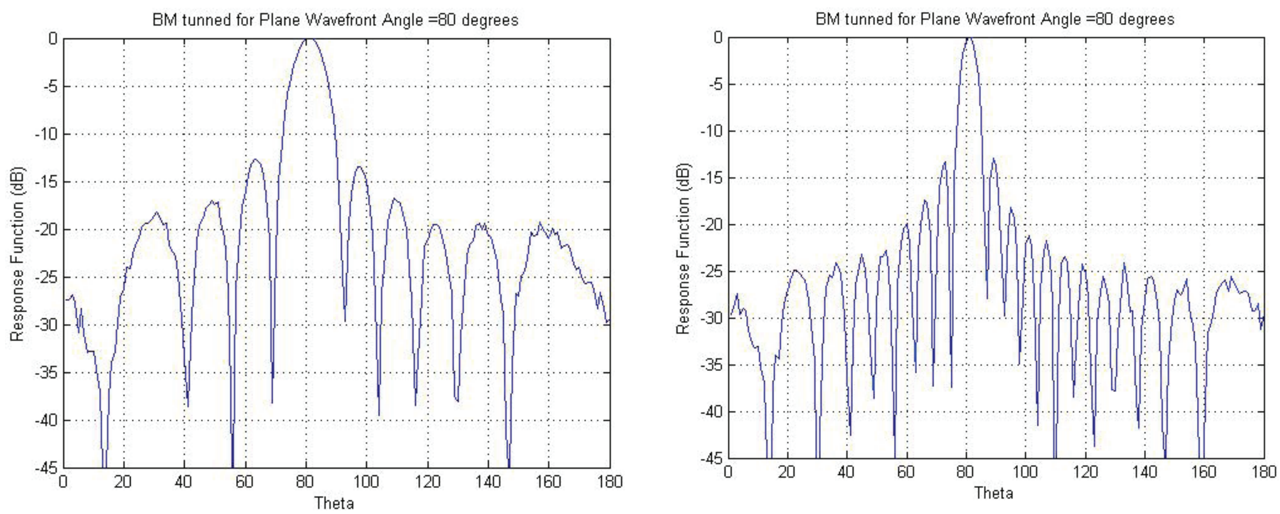
$$\mathbf{N}(\omega) = N(\omega) \times V(k_1, \omega)$$

$$\sum_{.} \mathbf{N}(\omega) = \sum_{.} N(\omega) \times V(k_1, \omega) \times V^H(k_1, \omega)$$

$$\sum_{.} \mathbf{N}(\omega) = \epsilon\{|N(\omega)|^2\}$$

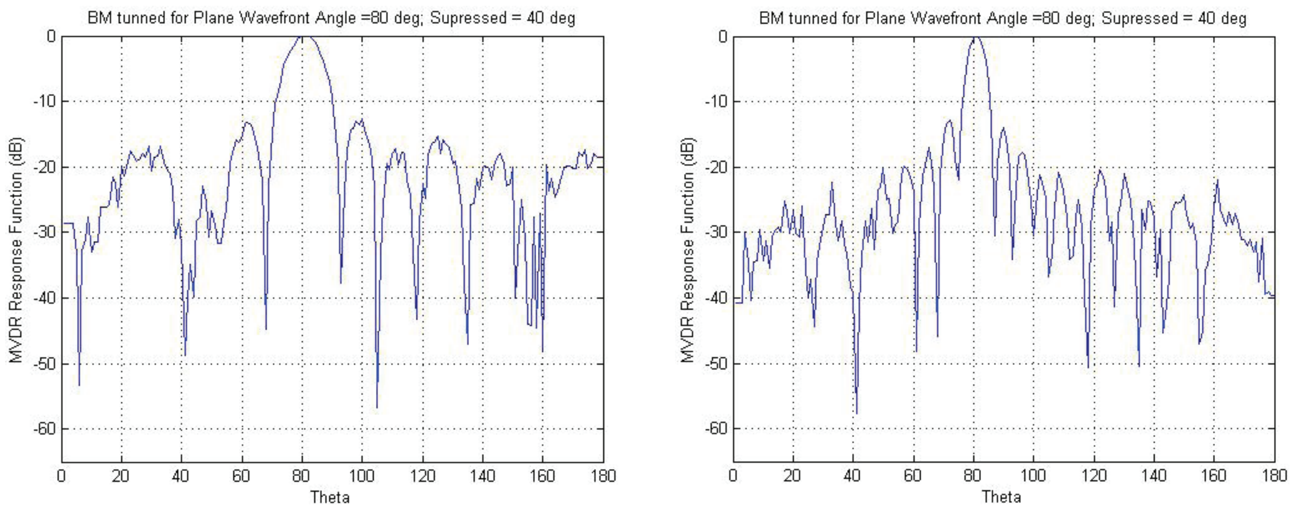
Figure 6 shows the beam pattern of the MVDR BF for the case of two plane wave interferers arriving from directions 80° and 40°. It is apparent from the figure that such a BF can place deep nulls on the interference signals while maintaining unity gain in the look direction. Depending on the acoustic environment, adapting the sensor weights w(ὠ) to suppress discrete sources of interference can lead to excessively large side lobes, resulting in poor system robustness. A simple technique for avoiding this is to impose a quadratic constraint $\|w\|^2 \le \gamma$, for some $\gamma > 0$, in addition to the distortion less constraint six, when estimating the sensor weights. The MVDR solution will then take the form [7, Sec. 13.3.7]

$$w_{DL}^H = \frac{v^H\left(\sum N + \delta_d^2 \mathbf{I}\right)^{-1}}{v^H\left(\sum N + \delta_d^2 \mathbf{I}\right)^{-1} \mathrm{v}} \tag{10}$$



(a) (b)

Figure 5. Delay and sum Response at S=10 and S=20 for sound source f(t) comes from the 80° angle.



(a) (b)

Figure 6. MVDR Response at S=10 and S=20 for sound source f(t) comes from the 40° angle and interference N(ω)comes from the 40° angle.

which is referred to as diagonal loading where δ_d^2 is the loading level; the dependence on ω in (9) has been suppressed for convenience. While (9) is straightforward to implement, there is no direct relationship between c and δ_d^2; hence the latter is typically set either based on experimentation or through an iterative procedure. Increasing δ_d^2 decreases $\|w_{DL}\|$, this implies that the white noise gain (WNG) also increases.

Fig. 5 (a) and (b) shows the response of the delay and sum (DS) at S=10 and S=20 respectively. Fig.6 (a) and (b) shows the response of the MVDR at S=10 and S=20 respectively. It shows that sound source f (t) comes from the 80°and interference comes from the 40° angle. In fig.5 it is shown that maximum lobe comes at 80° and it does not suppress the interference signal i.e. which comes from 40°. In fig. 6 it is shows that maximum lobe comes at 80° and it suppress the interference signal i.e. which comes from 40°, so it is concluded that MVDR suppress the interference signal.

RESULTS AND DISCUSSION

Minimum variance distortion less response (MVDR) is an algorithm used for beam formation. To achieve the goal of maximum SNR at output, experimentation is done by varying parameters like number of microphones (S), distance between two consecutive microphones (d), diagonal loading factor, number of overlapping samples. This section briefly discusses the effect of above parameters on the output signal by calculating mean square error (MSE)

Effect of the number of microphones (S) on the mean squared error (MSE)

Table 1 shows the effect of changing the number of microphones (S) on the mean squared error (MSE) in dB. . As the number of microphones increases from 2 to 20, there is a clear decrease in the MSE value. Reduced MSE means improved signal-to-noise ratio (SNR), thus improving beam forming performance. Obviously, using more microphones leads to a decrease in MSE, indicating better sound quality and better processing of the desired signal.

Table 1. Effect of change in number of microphones(S) on MSE

Sr. No.	Number of microphones (S)	Mean square error (MSE) in dB
1.	2	-20.0239
2.	4	-31.7035
3.	6	-33.6351
4.	8	-33.8215
5.	10	-33.3802
6.	12	-33.1005
7.	14	-33.4404
8.	16	-33.6207
9.	18	-33.8334
10.	20	-33.8561

Table 2. Effect of change in 'factor or distance between microphones' (d) on BM response

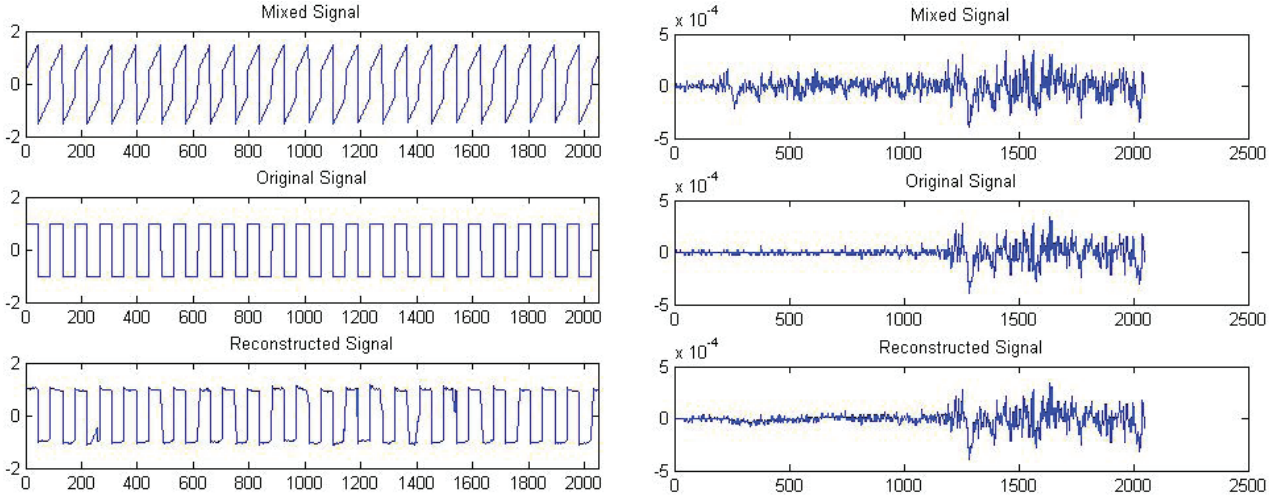
Sr. No.	Factor	Inter-Microphone distance (d) in cm	Mean square Error (MSE) in dB
1.	32	0.3125	-20.4012
2.	16	0.625	-25.9415
3.	8	1.25	-32.4857
4.	4	2.5	-34.0826
5.	2	5	-33.3802
6.	1	10	-29.1044
7.	0.5	20	-20.2985
8.	0.33	30.30	-14.6505
9.	25	40	-11.2208
10.	0.20	50	-7.1474

Effect of Inter-Microphone Distance (d) on Beam forming Response

In table 2 details the effect of different microphone distance (d) on the beam forming response as expected by MSE (in dB). The findings indicate that there is a desirable correlation between inter-microphone distance and beamforming performance. The nearer the microphones, the more resolution contained in the spatial sampling; it results in improved beamforming capabilities and reduced MSE. On the other hand, the greater the distance between microphones, the lower the resolution and thus the higher the MSE values. What this illustrates is the need to adequately pick the spacing between microphones in order to maximize the beamforming capabilities of a certain application.

Waveforms of the Desired Signal, Mixed Signal, and Repeated Signal

Figures 7(a) and 7(b) represent the waveforms of the desired signal, mixed signal, and repeated signal of the square wave and speech signal, respectively. The presence of these waveforms is an indication of how well the beamforming process has separated the desired signal and noise and interference. The reconstructed signal is similar to the desired signal, demonstrating full noise reduction and signal enhancement through beam forming. This fit supports many of the results obtained from the MSE analysis and increases the efficiency of the proposed beam forming algorithm. It is important for formability. Once it is possible to vary parameters (number of microphones and distance between them) to reach the best signal-to-noise ratio and to make the reconstruction signal as faithful to the original as possible. Also, the comparison of the waveforms visually



(a) (b)

Figure 7. Waveforms of the desired signal, mixed signal, and repeated signal for (a) Square wave (b) Speech signal**.**

ascertains the performance of the beamforming algorithm in real-life applications, which reaffirms its potential usefulness in various areas, including communications and audio processing, as well as biomedical imaging.

CONCLUSION

This research demonstrates that the Minimum Variance Distortionless Response (MVDR) algorithm is greater in audio beamforming tasks than the conventional delay-and-sum methods. The MVDR beamformer was used to show improved ability to isolate the desired sound signals and reduce interference of other directions, especially in a complex acoustic environment, by means of applying a linear microphone array. The experimental data indicate that the number of microphones and the spacing between microphones will greatly increase SNR and decrease MSE. The trade-offs associated with beamformer design, however, e.g., the higher computational cost of more area overlap, should be sensitively addressed so as to provide optimum performance. The results highlight the significance of array geometry and parameter optimization to the beamforming performance and offer important ideas in the application to other fields, including telecommunications, sonar, and medical imaging. The proposed work may be done in the future by implementing and testing the MVDR algorithm in real-time in various acoustic settings to once again confirm its efficiency.

This is enhanced by the fact that the main lobe is narrowed; thus, separation of the near field is correct. Additionally, the number of lobes in the response function is directly related to the number of microphones in the array; this highlights the relationship between array geometry and beamformer performance. The impact of beamformer performance is significant. Increasing the crossover reduces the mean square error (MSE), indicating an improvement in signal quality. However, this improvement comes at the expense of reducing interference from other directions. Conversely, increasing the number of overlapping samples in the window results in an increase in MSE and distortion of the reconstructed signal.

According to the results, the trade-offs in the beamformer design are evident, and it is important to optimize performance when optimum performance is required. They also show the nature and restriction in regard to signal boosting and interference. The geometry of the array is also a factor of consideration, and the choice of the parameters and optimization algorithms is critical to the success of beamformer activity in a number of applications, including telecommunications and voice processing. By understanding all these principles, engineers can deduce and implement beamforming systems that can be utilized in particular requirements, which is limiting any further advancement of signal technology.

AUTHORSHIP CONTRIBUTIONS

Authors equally contributed to this work.

DATA AVAILABILITY STATEMENT

The authors confirm that the data that supports the findings of this study are available within the article. Raw data that support the finding of this study are available from the corresponding author, upon reasonable request.

CONFLICT OF INTEREST

The author declared no potential conflicts of interest with respect to the research, authorship, and/or publication of this article.

ETHICS

There are no ethical issues with the publication of this manuscript.

STATEMENT ON THE USE OF ARTIFICIAL INTELLIGENCE

Artificial intelligence was not used in the preparation of the article.

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