Comparative Analysis of Planar Acoustic MVDR Beamformer

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Abstract—In this paper an ambiance invariant, data length invariant, real time experimental set up for testing Minimum Variance Distortionless Response (MVDR) beamformer under noisy reverberant environment is presented. To the best of our knowledge, for the first time an attempt is made to deploy MVDR beamforming algorithm on planar topologies, viz. rectangular and circular array. The sound source is localized using Time Delay of Arrival (TDOA) method. We present comparison between analytical performance of MVDR beamformer having up to 12 microphones in the array for reported topologies. An experimental investigation of MVDR beamformer on a dedicated hardware for a greater number of microphones in an array is performed. We show that using MVDR beamformer, in reverberant environment, the noise suppression up to 15 to 16 dB can be achieved using 5 to 9 microphones in a linear array.

Keywords—Audio Beamforming, MVDR, Planar topologies, Audio Signal Processing.

I. Introduction

NOW a days arrays are becoming popular for acquisition of audio signals transmitted by sources located at distinct locations. Signal processing technique that strengthens the desired signal based on the directional information is known as Beamforming. Beamformers find applications in wireless networking [1], radio telescopes [2]. In acoustic signal processing beamformers are extensively used for source localization [3], automatic speech recognition [4], hearing aid applications [5], [6], noise and echo cancellation [7] and source separation [8]. Array designs are primarily focused towards optimizing one of the narrow-band measures like signal to noise ratio (SNR), white noise gain (WNG) [9], array gain, beampattern, directivity factor (DF) [10].

Many classical beamformers such as Delay and Sum (DS) beamformer, sub-array beamformer are popularly used depending upon directivity patterns and array gain [11]. Delay and sum is the simplest beamformer derived by maximizing WNG subject to distortionless constraint on steering vector. However it has a narrow beamwidth. Subarray beamformers are broadband beamformers and give frequency invariant response.

Adaptive beamformers adapt dynamically to maximize one of the parameter such as SNR by strengthening the signal power [12] and suppressing the noise [13]. Cox et al proposed an algorithm for robust adaptive beamformer by adding quadratic inequality constraint on array gain [14]. One of the most popular and most explored beamformer is Capon beamformer also known as Minimum Variance Distortionless Response (MVDR). Many modified versions of MVDR such as MMSE MVDR [15], eMVDR [16], generalized sideband cancellation (GSC) beamformer [17] are proposed in literature [18]. Performance of MVDR in the presence of estimation error [19] and different types of noise is studied in detail [20] and in [21]. Superdirective beamformers maximize either the directivity or the array gain [22]–[25].

Numerous uniform and non uniform geometries are proposed in literature with the objective to obtain maximum DF, provide more antenna gain and better unwanted signal suppression. Simplest array geometry is a linear geometry.Though linear arrays are simple in implementation, they are large in size and do not have three dimensional spacial resolution. Many other planar topologies like rectangular [26], circular [27]–[29] and non planar topologies like spherical [30]–[33] and spiral [34] have been proposed. Ioannides et al have discussed analysis and implementation of planar uniform rectangular and circular geometries.

Microphone arrays are indispensable while working on localization of audio sources. Different localization techniques based on spectral estimation, maximum steered response of beamformer or time delay estimation in microphone arrays are discussed in literature [35], [36]. Recently many algorithms for sound source localization using machine learning methods are proposed [37]–[39]. TDOA estimation using different types of arrays under variety of environmental conditions such as indoor, outdoor, echoic, reverberent, noisy [40] and [41] are widely discussed.

So far, several topologies are proposed for different types of beamformers. MVDR beamformer is explored in detail too. Although lots of theoretical analysis is carried out in the literature, the measured performance differs from theoretical estimates, especially in reverberant environment. Reducing effect of noise for robust performance of beamformer is a big challenge. Majority of the commercial beamformers use limited number of microphones in the array as the performance of beamformer degrades drastically with increasing number of microphones in real life scenario. We have designed and developed an ambiance invariant, high resolution experimental setup to test the performance of beamformer. As far as our knowledge goes, this is the first of its kind attempt to deploy MVDR beamforming algorithm for planar topologies viz Uniform Rectangular Array (URA) and Uniform Circular Array (UCA). Their performance is analyzed for the ability to suppress undesired signal and compared with conventional Uniform Linear Array (ULA). Further, TDOA estimation-based localization is implemented.

The paper is organized as follows: Section II presents brief overview of different topologies of array. MVDR Beamformer

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is discussed in unit III. Localization of a sound source using TDOA method is discussed in section IV. Simulation and experimental results are discussed in section V followed by conclusion in section VI.

II. Conventional Topologies of Planar Array

Lets assume that the source signal s(t) reaches microphones at different time intervals. For a linear array having S sensors, time-delayed signals reaching each microphone with respective delay can be represented by a vector.

$$s(t) = [s(t-t_0) \qquad s(t-t_1)\dots s(t-t_{S-1})]^{\mathrm{T}} \tag{1}$$

Mathematical formulation for estimation of time delay of arrival (TDOA) is different for different geometries of beamformer. In this section conventional topologies are reviewed. We have restricted ourselves to planar arrays because of their simplicity and lesser computational complexity.

A. Uniform Linear Array

Under far field assumption the incident wavefront is treated as planar. The TDOA expression is given by equation 2.

$$t_{m,n}(\theta) = \frac{d_{m,n}cos(\theta)}{c} \tag{2}$$

where $d_{m,n}$ is the distance between the m^{th} and n^{th} microphones. θ is angle between the direction of arrival (DOA) and the line connecting two sensor positions. Linear arrays are simple to implement but large array size is required to attain decent directivity.

B. Uniform Rectangular Array

Uniform M x N rectangular array arranged in X-Y plane contains M elements along X axis with uniform distance d_x and N elements along Y axis with uniform distance of d_y between the sensors.

For a planar wavefront arriving from direction (θ, ϕ), time delay vector is partially linear and given by

$$t_{mn} = -\frac{md_x.sin\theta cos\phi + nd_y.sin\theta sin\phi}{c} \tag{3}$$

In this expression, θ is elevation angle and ϕ is azimuth angle. Rectangular beamformers can localize a sound source placed in 3 dimensional space due to inclusion of elevation angle θ in the TDOA calculation.

1) Uniform Circular Array: As the name suggests, the sensors are arranged in circular manner to form circular array. The array consists of M concentric rings with variable radius a_m forming a symmetric geometry. A uniform distance is maintained between the neighboring elements along the circumference of the circle. Therefore the number of microphones in each circle is different. Any m^{th} ring contains N_m sensors as shown in figure 1

For n^{th} microphone in m^{th} ring, the relative TDOA with respect to origin is described by

$$t_{mn} = -a_m\frac{cos(\phi - \phi_{mn}).sin\theta}{c} \tag{4}$$

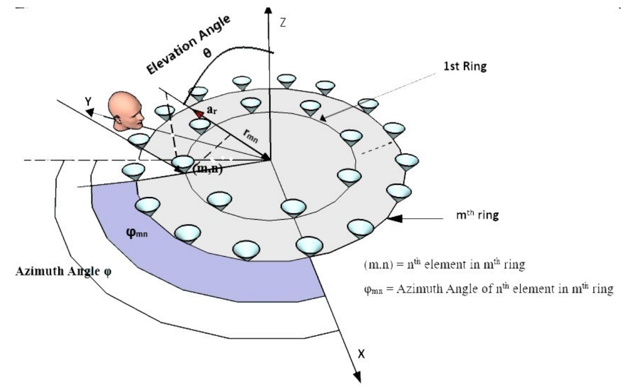

Fig. 1. Uniform circular array with M rings and each ring containing N_m equidistant elements

where θ_{mn} is angular position of n^{th} element in m^{th} ring and is calculated by

$$\phi_{mn} = 2\pi\frac{n}{N_m} \tag{5}$$

where n varies from 1 to N_m.

The biggest advantage of circular topology is that the perfect symmetry can be maintained in the geometry. Shape of the beam does not change significantly even if the array is rotated in the plane of array. Delay vector obtained in any of these topologies is fed to beamforming algorithm for weight calculations.

III. MVDR beamformer

MVDR is an adaptive technique of beamforming with objective of minimizing the variance of the noise signal. In case of indoor source localization, a beamformer which can suppress spatially correlated noise is desirable. MVDR beamformer is selected for experimentation. We have deployed an adaptive MVDR beamformer as described by Kumatani in [4] for our experimentation.

Time delayed signal as described by equation 1 is represented by phase delayed signal in frequency domain. It is called as array manifold vector v_ω.

$$v(\omega) = [e^{-i\omega t_0} \qquad e^{-i\omega t_1} \qquad \dots e^{-i\omega t_{S-1}}]^T \tag{6}$$

In frequency domain the output of a beamformer can be represented as

$$Y(\omega) = W^H(\omega).X(\omega) \tag{7}$$

where $X(\omega)$ is frequency domain representation of input to array of microphones and $W^H(\omega)$ is a vector of frequency dependent weight vectors corresponding to each microphone of array. In absence of interference and noise output of beamformer $Y(\omega)$ is same as original input $X(\omega)$. In equation form it can be represented as

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

Any weight vector $W(\omega)$ achieving equation 8 satisfies the distortionless constraint. Performance of beamformer can be improved by adaptively suppressing spatially correlated noise and interference. This can be achieved by adjusting the weights of beamformer do as to minimize the variance of noise and interference at the output. More particularly, $W(\omega)$ should satisfy

$$argmin_w W^H(\omega)\Sigma_N(\omega)W(\omega) \tag{9}$$

subjected to condition imposed by equation 8

$\sum_N = E\{N(\omega)N^H(\omega)\}$ is variance of noise and interference and E is the expectation operator. $\sum_N$ is computed recursively by averaging the noise covariance matrix. The weight vectors obtained under these conditions correspond to the MVDR Beamformer. Its solution is given by

$$w^H_{MVDR}(\omega) = \frac{v^H(k,\omega)\sum_N^{-1}(\omega)}{v^H(k,\omega)\sum_N^{-1}(\omega)v(k,\omega)} \tag{10}$$

The beamformer response provides maximum gain in the direction of desired signal and deep nulls in the direction of interfering signal thereby minimizing effect of interference.

Sometimes the adaptive nature of beamformer unknowingly gives rise to large sidelobes. The robustness of the beamformer against noise is affected too. To address this issue in addition to distortionless constraint, a quadratic constraint is inflicted. Under the quadratic constraint $||w||^2 \leq \gamma$ where $\gamma > 0$ the solution is modified to

$$W^H_{DL}(\omega) = \frac{v^H(\sum_N + \sigma_d^2 I)^{-1}}{v^H(\sum_N + \sigma_d^2 I)^{-1}v} \tag{11}$$

which is referred to as diagonal loading. σ_d^2 represents the loading level.

IV. Source Localization using TDOA

Audio source localization aims at automatic identification of direction of source based on certain parameters of sound field. Localization methods can be classified in four major categories.

1) Subspace algorithms where source is localized based on spectral estimation technique
2) Localization based on maximum steered response power (SRP) of beamformer
3) Estimation of location of source based on Time delay of Arrival (TDOA) at different microphones in an array
4) Methods based on Machine learning techniques

Time delay in the equation 2 is very important parameter for estimation of direction and location of source. Numerous algorithms are proposed to estimate TDOA and can be broadly classified as cross-correlation-based methods and system identification-based approaches [42]. For multichannel systems adaptive eigenvalue decomposition methods prove to be robust in noisy and reverberant environment demanding more number of sensors. However multichannel cross correlation can prove promising at higher sampling frequencies. Cross correlation function is most popularly used for delay estimation. Estimated time delay is fairly accurate in the presence of low to moderate noise and in the absence of multi-path effect. A generalized frequency weighing function is introduced in the computation of cross correlation to improve performance of TDE in the presence of noise and multipath effect. This method is known as Generalized Cross Correlation (GCC). GCC sequence of two microphone outputs can be computed by equation 12.

$$G_{x_1x_2}(p) = IFFT(\phi(\omega_k)X_1(\omega_k)X_2^*(\omega_k)) \tag{12}$$

where $\phi(\omega_k)$ is the weighing function. The time delay between two signals is calculated as

$$\tau_{12} = arg(max_p(R_{x_1x_2}(p))) \tag{13}$$

This delay between the samples is used for estimation of location of audio source. Selection of appropriate weighing function is a trade off between good resolution and stability of filter. The weighing function should give sharp peak at the output of cross correlator ensuring more accurate TDOA estimation. Some weighing functions are effective for estimation of TDOA in the presence of high additive noise while some are effective in computation of TDOA against multi-path effect. In Smoothed Coherence Transform (SCOT) and Eckart functions, the weights are assigned according to the characteristics of signal and noise. Both functions suppress frequency bands of higher noise. SCOT can be viewed as cross correlator preceded by pre-whitening filter. The drawback of SCOT function is broad correlation because of its inadequacy to pre-whiten the cross power spectrum.

The Eckart function is described by

$$\frac{P_a(\omega_k)}{P_{b1}(\omega).P_{b2}(\omega)} \tag{14}$$

Eckart function maximizes the correlator output due to signal alone by minimizing the effect of noise. It assigns zero weight to bands where cross power spectrum between input signals is zero i.e. when they are not correlated. Eckart function performs better in the noisy environment compared to other functions.

V. Experimentation

A. Details of Proposed Hardware Setup

We have developed an ambiance invariant dedicated experimental setup for testing and evaluating linear and planar topologies of array in noisy conditions. The microphones can easily be rearranged in any of the reported linear topology without much delay. The experimental set up with sample linear arrangement of microphones is as shown in the figure 2.

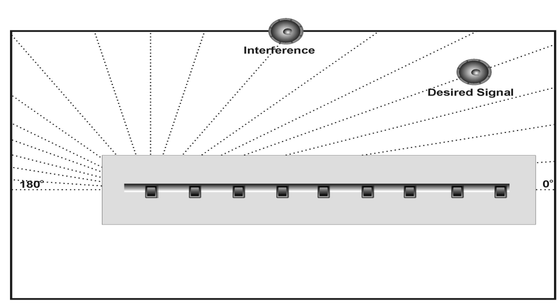

Fig. 2. Experimental Setup

The experimentation is carried out in a laboratory having dimensions 18ft x 12ft x 9ft under noisy reverberant conditions. The data acquisition system consists of SPW2430 MEMS microphones, cDAQ 9188 chassis and NI 9232 C series sound and vibration input cards. The audio signals are acquired by SPW2430 analog, top port, omnidirectional microphones with frequency range from 100 Hz to 10KHz. These are low power and light weight microphones which makes them best suited for portable devices. cDAQ 9188 is a compact data acquisition chassis having Ethernet connectivity and is designed specifically for small distributed sensor measurement system. NI 9232 can acquire three analog or digital signals in a single card. It reads all input channels simultaneously with variable sampling rate and with high speed digital data transfer on Ethernet. Each channel has built in anti-aliasing filter which automatically adjusts the sampling rate. The ADC has 24 bit resolution. Labview, a system design platform and development environment, is used to process the acquired data. Execution is determined by the structure of a graphical block diagram, known as source code, on which the programmer connects different function-nodes by drawing wires.

For experimentation, all the channels are sampled at uniform sampling frequency of 102.4KHz/channel and acquired signal is transferred to PC with average Ethernet speed of 5.2 Mb/sec. The processing unit and the NI-9188 cDAQ is connected via Ethernet cable, both sharing same IP network. Distance between microphone array and sound source is at least 1 meter. The biggest advantage of this system is multi-processing and multi-threading hardware is exploited automatically by the built-in scheduler, which multiplexes multiple threads over the nodes ready for execution. Another special feature of our system is that unlike many other systems, there is no constraint on the data length that can be processed. MVDR beamforming algorithm is implemented and performance is analyzed for all geometries viz linear rectangular and circular.

B. Simulation and deployment of MVDR beamformer

The MVDR beamformer algorithm is simulated initially in Matlab and then source coded in Labview for experimentation. The number of microphones are varied from 2 to 16. Speed of sound (s) is 330 m/s and inter-microphone distance (d) is maintained constant at 5 cm. Frequency for calculations is assumed to be 3300 Hz. Desired signal direction is assumed to be 80^o while interfering signal direction is 40 degrees. Diagonal loading factor is 0.3. Each frame consists of 256 samples.

Samples of signals received by data acquisition systems are processed using Tukey window of length 256 samples. The signal is transformed into frequency domain using 256 point FFT. Delay vector, dependent on placement of microphones, is generated from the array and is fed to MVDR algorithm. Array Manifold vector (AMV) which is frequency domain equivalent of time domain delay is generated. Two stationary sound sources are considered for experimentation purpose. MVDR weight vectors are derived according to equation 10 from AMV to minimize the variance of noise and interference at the output. Signal is recovered back by taking 256 point IFFT and by applying overlap and add method. The time delay changes with the geometry of microphone array and distance between the adjacent microphones. Delay vector is non uniform for planar geometries.

While building source code on Labview, enough precautions were taken to address the issues of noise during acquisition of signal. A high pass filter of 100Hz is added in each channel to prevent power line disturbances and to remove dc bias. To avoid system delays, all channels are multiplexed and processed Parallelly.

1) Localization: GCC uses various weighing functions which are referred to as prefilters. Prefiltering step helps in better estimation of time delay thereby improving performance of localization. Some of the weighing functions make the delay estimates more immune to additive noise, while others improve the robustness of TDE against the multi-path effect. Figure 3 shows localization process.

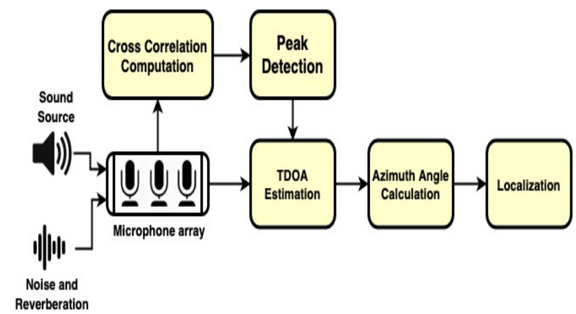

Fig. 3. Steps Involved in Localization

The raw signal received from microphones is windowed by using window of size 256. The windowing process is repeated 15 times to obtain 15 readings of the estimated delay value. Frequency domain GCC for 2 channels at a time is obtained to find delay. The delay is calculated as a deviation from center index. To obtain direction of arrival, the obtained delay is converted into angle.

VI. Results and Discussion

A. Simulation and Experimental Results

In this section, the results of simulation and experimentation on our designed set up are stated. MVDR beamforming is simulated in Matlab for linear and planar geometries namely, i. Uniform Rectangular and ii. Uniform Circular array. MVDR response for all three topologies is compared. For experimentation purpose in all array geometries desired source is placed at 80^0 and interfering source is placed at 40^0. Spacing of 5 cm between two neighboring microphones is maintained. In planar geometry, the elevation angle is 70^o.

1) Results of ULA: MVDR beamformer response of uniform linear array is studied by varying different array parameters such as number of microphones, spacing between the successive array elements, loading factor, frequency etc. The MVDR response of ULA for varying number of microphones is as shown in figure 4. The diagonal loading factor is varied between 0 to 1 during the simulation. MVDR response of 10

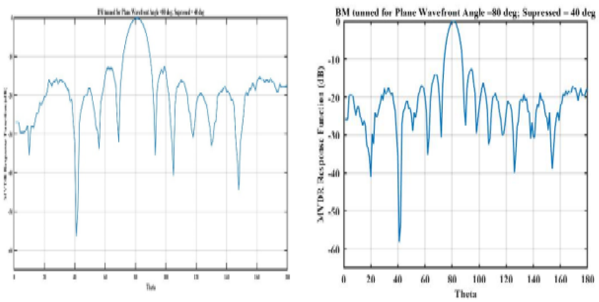

Fig. 4. Output of Linear Beamformer for variable no of microphones a. with 9 microphones and b. with 12 microphones

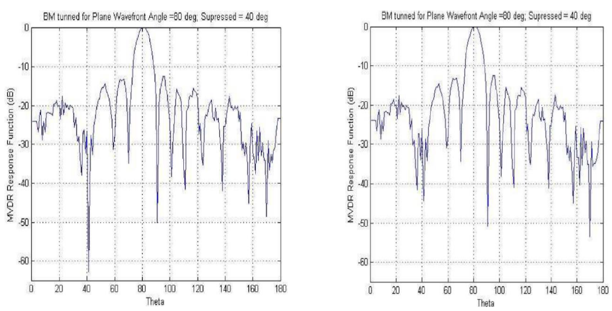

Fig. 5. a. Simulation Result for loading factor 0.1 b. Simulation Result for loading factor 0.9

microphone linear array with loading factor 0.1 and 0.9 is seen in figure 5

The experimental results of MVDR beamformer for linear array with 3 and 5 microphones is as shown in figure 6 and 7 respectively.

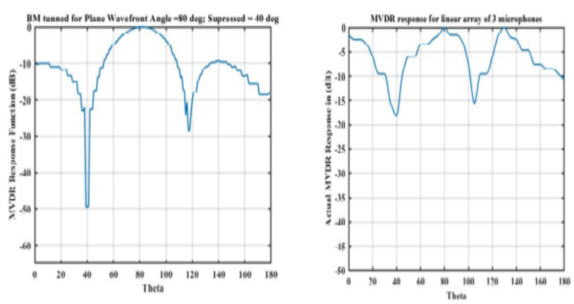

Fig. 6. a. Simulation Result b. Experimental Result for Linear Beamformer with 2 Microphones

2) Results of URA and UCA: The MVDR beamformer response shown in the Figure 8 is obtained by arranging 12 microphones in rectangular array of two rows with interspacing of 5 cms along x and y directions. Beamformer response for uniform circular array of single ring with chord length 5 cm is shown in figure 9.

3) Results of Localization: For better estimation of TDOA in the presence of noise, different weighing functions are used. Most suitable weighing function for our noisy and reverberant testing environment is chosen by simulating GCC function for two microphones. Delay of 10 samples was given between the microphones for simulation purpose. The accuracy in the estimation of delay for different SNR values is checked for

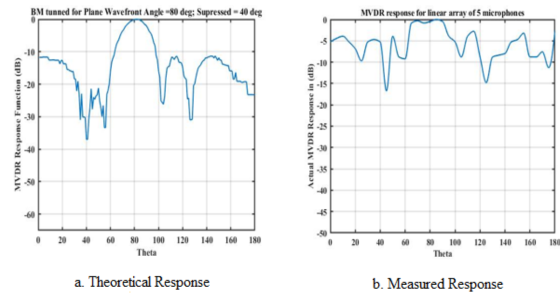

Fig. 7. a. Simulation Result b. Experimental Result for Linear Beamformer with 5 Microphones

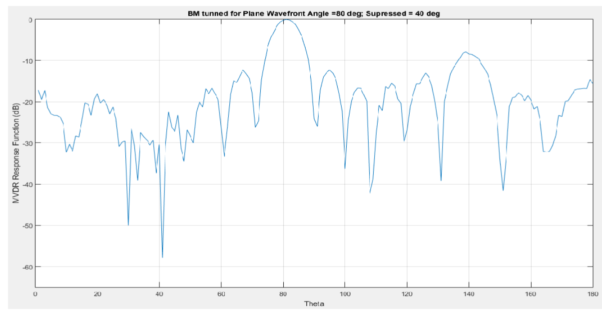

Fig. 8. Rectangular Beamformer output with 12 Microphones arranged in 2X6 array

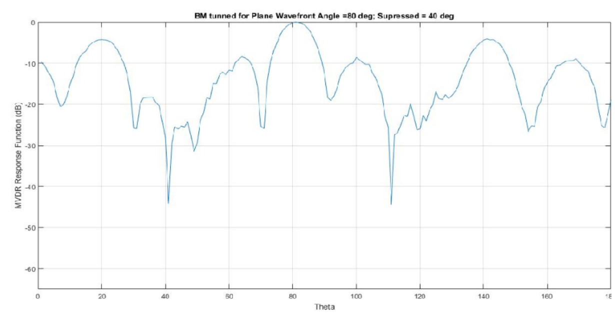

Fig. 9. Output of Circular array with 12 Microphones

different weighing functions as shown in the observation table I

TABLE I
WEIGHING FUNCTION ANALYSIS

Name of method	SNR in dB	% Accuracy
PHAT	5	50
	10	70
	15	90
Smoothed Coherance Transform	5	70
	10	70
	15	85
ECKART	5	90
	10	85
	15	90

B. Observation and Discussion

The array performance is analyzed for varied number of microphones while maintaining same distance of 5 cm between neighboring microphones. The performance measures

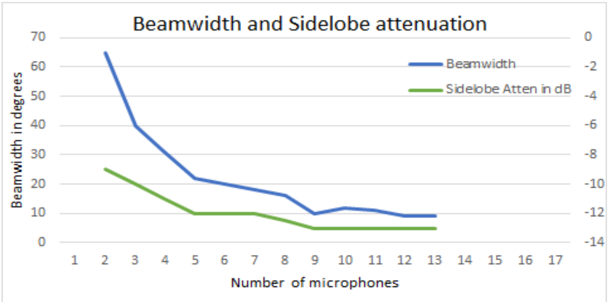

Fig. 10. Effect of increasing number of microphones on beamwidth and sideband attenuation

TABLE II
MEASURED PERFORMANCE OF MVDR BEAMFORMER FOR ULA

Number of Mics	**Interference Suppression in dB**	**Beamwidth in degrees**
3	7.2	32
5	16.74	27
9	15.72	12

considered for comparison are the beamwidth, sidelobe attenuation and undesired signal suppression capability of the beamformer. It can be observed from figure 4 the interference attenuation increases from 52dB for 9 microphones to 58 dB for 12 microphones. All performance measures were observed as a function of number of microphones. As expected though the initial increase in undesired signal suppression is more, it changes slowly giving almost constant suppression for more number of microphones. Figure 10 shows beamwidth and sidelobe attenuation against number of microphones.

As can be seen sidelobe attenuation remains almost constant after 9 microphones. There is no change in beamwidth even if we increase number of microphones more than 12. The directivity and side lobe attenuation improve with increasing no of microphones till maximum 12 microphones. The beamwidth decreases with increasing array length. However risk of system becoming unstable is more for greater number of microphones.

Studying effect of loading factor from figure 5, it is seen that interference attenuation is more than 60dB for loading factor 0.1 and it is decreased to 44 dB for loading factor 0.9for same number of microphones in an array. It indicates that the robustness against noise increases with loading factor. Clearly there is trade off between the number of microphones used, interference suppression capability of the beamfomer and its robustness against the noise.

We have experimentally validated the simulation result for linear beamformer. The comparison of analyzed and measured performance for variable no of microphones is as shown in table II.

For two microphones the interference attenuation is 7.2 dB which improves to 16.74 dB for 5 microphones. We observe that with 9 microphones we achieve fairly good suppression of 15.72 dB. These suppression levels are are at par with the levels achieved by the generic beamformer proposed in [27]. The deviation of measured performance from analyzed performance is attributed to experimentation ambiance.

TABLE III
COMPARISON OF MVDR BEAMFORMER FOR ULA, URA AND UCA TOPOLOGIES

Array Topology	**Interference Suppression in dB**	**Beam width in degrees**	**Sideband attenuation in dB**	**Size of Array in cm**
ULA	58	11	15	55
URA	58	13	8	5 X 25
UCA	42	16	5	radius = 10

The performance of MVDR for all three topologies is compared for 12 microphones in the table III. Our simulation results of URA and UCA may be cross verified with the results mentioned in [26]. As can be observed from table III, we achieve acceptable interference attenuation and moderate sideband suppression with compactness in size and with lesser no of microphones using URA.

Table I indicates that the Eckart function produces maximum precision in the argument value of correlation for our environment. Spatial resolution, defined as the minimum angular separation between the two sources to identify them separately, is observed. Using Eckart function, spatial resolution of $3^o 86''$ is achieved with sampling frequency of 102.4 kHz.

VII. CONCLUSION

In this paper we have conceptualized, designed an ambiance invariant experimental set up for testing MVDR beamformer under noisy reverberant environment. MVDR algorithm is deployed on URA and UCA for the first time to the best of our knowledge. The performance of MVDR for all three geometries viz ULA, URA and UCA is compared, in particular, for its ability to suppress interference and beamwidth. MVDR beamformer with uniform linear geometry was presented and tested on the proposed novel experimental set up of microphone array. The experimental results show fairly good interference suppression of 15 to 16 dB for array comprising upto 9 microphones, in real time acoustic environment. An audio source was localized using TDOA estimation by GCC method. A good spatial resolution was achieved by applying Eckart weighing function to GCC. Further performance of MVDR beamformer for uniform rectangular and uniform circular topologies was analyzed. When compared with linear array, rectangular array gives at par interference rejection but lesser sidelobe attenuation at a smaller array size. Uniform circular beamformers may prove to be optimum solution for a very large number of microphones in an array. Still some anomaly persists regarding the directivity of beam formed towards the desired source. This might be improved by incorporating 3-dimensional arrays with increased computational complexity.

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