IJournals: International Journal of Software & Hardware Research in Engineering
ISSN-2347-4890
Volume 5 Issue 12 December, 2017

Beam Formation for Noise Reductions in Acoustic Application

Author: Pranali Kashid[1]; Balasaheb Patil[2]

Affiliation: PG student, E&TC(Digital System), VPKBIET Baramati, Maharashtra, India[1]; Professor, M.E(Microwave), VPKBIET Baramati, Maharashtra, India[2]

E-mail: kashidpranali1122@gmail.com [1]; b_ash11@rediffmail.com [2]

ABSTRACT

It is always important to a noise free communication during the conferences and meetings. Sound Source Localization and the audio beam formation are the acoustic signal processing techniques implemented for this purpose. Both these methods deal with the microphone array signal processing to reduce the impact of noise on the output of receivers and to deliver a clean signal. Audio beam formation along with the appropriate geometry of microphones in the array combines the incoming signal into a noise free beam of a signal. The beam formation changes the response pattern of the array so that it provides maximum gain only in one angular direction which is previously calculated by the sound source localizing part. Thus the combination of these two techniques can be used to capture the maximum from the speaker and ignoring the noise in the conference rooms. All this work will be done by using microphone array and MATLAB.

Keywords: Audio beam formation, Microphone array processing, MATLAB

1. INTRODUCTION

Noise reduction in the mobile, meeting rooms, conferences is an important topic. This thesis discusses the possibilities of using beam forming with some microphones at the input side. The goal of the project is to realize a beam forming algorithm for conferences and meeting rooms to improve on current Signal to Noise ratios attained by standard noise reduction methods. Meeting or conferences are plays an important part of any profession where it is expected to have a good sound system including microphones. In that part, listeners may lose their attention towards the speaker. Owing to this need and its relative importance, the microphone array processing has gained much more attention in the last few years. When two or more individual microphone sensors are arranged in an array with a proper geometry, and the signals from each sensor are suitably combined, the array functions as a spatial filter capable of suppressing noise, reverberation, and competing speech[6] we have to do work on that. The advancement in signal processing techniques and procedures, various applications have been evolved including RADAR, SONAR, mobile phone location, a navigation system, localization of earthquake epicenters and underground explosions, robots, microseismic events in mines, speaker tracking and sound source tracking [7],etc. In such applications one thing is common that they localize the source of signal whichever may be the signal like sound, radio waves, etc. In acoustics, this concept is used to find the location of the sound source with about some reference frame. Humans are generally able to detect the direction that a sound is coming from using two ears. The combination of the slightly different signals that arrive at the ears enables us to find the direction of the sound. Similarly, a biologically inspired sound localization system can build by making use of an array of microphones [8].This system measures the time delay or phase difference or energy difference between the acoustic signals received at microphones as per the selected algorithm. The estimated difference can then converted into the angular position of the source. Once the location of the speaker tracked, then that in-formation can be provided to the next half of the work which is beam formation. Beam formation is the technique used to increase the SNR (signal-to- noise ratio) of the received signal like sound or radio waves. Beam formation has various applications in the biomedical field, satellite communication, RADAR for getting the noise free beam of the desired output. In acoustics, beam formation estimates the desired signal by using a microphone array. The signals from the various microphones are combined such that signals at particular angles (those angles which are estimated by the localization system) experience constructive interference, while others experience destructive interference. Thus beam formation can be used to enhance the signal quality of received speech which is corrupted by various types of noise and reverberation. It gives different results from microphone array geometry and audio beam formation algorithm used. In general micro-phone array has the linear, circular, spherical geometries while algorithms can be either conventional or adaptive algorithm depending on the application.

2. RELATED WORKS

2.1 Sound Source Localization

As shown in below figure 1, there are three main ways through which the sound source could localize. These are TDOA (Time Difference Of Arrival), DOA (Direction of Arrival) [8] and energy based methods [9]. In TDOA method, the TDE (Time Delay Estimation) is done using the signals from microphones to locate the source.

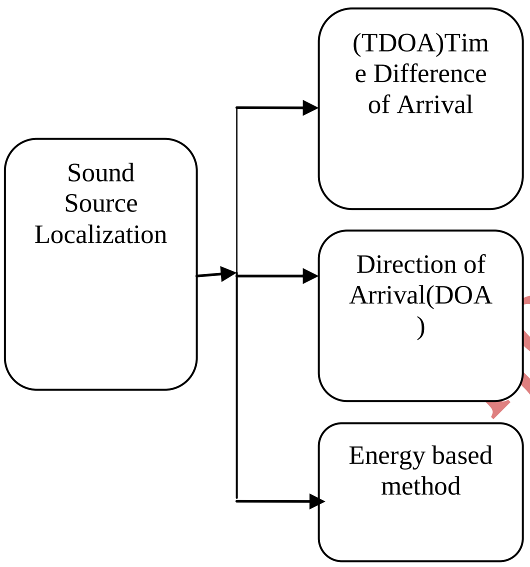

Fig 1: Ways of Source Localization

In DOA, the phase difference between the signals is used to locate the sound source. But this method put restrictions on the frequency of input signal[3]. In energy-based sound localization, the sound energy at different sensor locations is measured to localize the sound source. It saw that for a coherent, narrow band source, the phase difference measured at receiving sensors could be exploited to estimate the bearing direction of the source. For broadband source, like a speech signal, most of the power is spread over a wide range of frequencies. For such cases, time-delay estimation (TDOA) has been quite popular [8].

2.2 Algorithm of Beam Formation

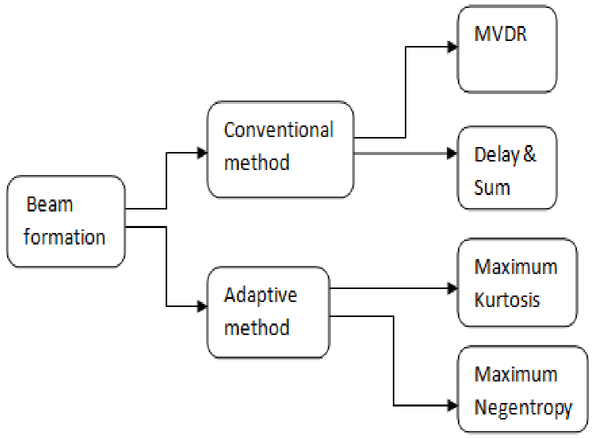

Fig.2. Types of Beam Formation

K. Kumatani; et al..discussed the Beam formation for the audio signal by using maximum kurtosis a criterion for estimating the active weight vectors in a GSC. The kurtosis also measures the degree of super Gaussianity of a pdf [2]. Optimize the active weight vectors of a GSC so as to achieve the output with the maximum kurtosis. After beamforming, Zelinskipost filtering is performed to further enhance the speech by removing residual noise.

The MK beam former can suppress noise and reverberation without the signal cancellation problem encountered in conventional adaptive beamforming algorithms. In contrast to negentropy, kurtosis does not need to knowledge of the actual pdf of subband samples of speech. Rather, kurtosis can be simply calculated in a non-parametric manner [2]. However, the kurtosis measure influenced by samples with a low observation probability. It is worth used the MK criterion to build a multi-microphone speech enhancement system without the demonstrated speech enhancement with relatively little enrollment data and GSC implementation and demonstrated speech enhancement with relatively little enrollment data. Applying the MK criterion to a beam former in GSC configuration enables the beam steered as desired.

K. Kumatani; et al.proposed beam formation with the maximum negentropy. Negentropyas a criterion for estimating the active weight vectors in a GSC. Negentropy indicates how far a probability density function of a particular signal is from Gaussian [3]. The pdf of speech, in fact, super-Gaussian, but it becomes closer to Gaussian when the speech corrupted by noise or reverberation. Hence, in adjusting the active weight vector of the GSC to provide a signal with the highest possible negentropy. The maximum negentropy beamformer can achieve this goal without the signal cancellation problem encountered in conventional beamforming algorithms.

Moreover, our technique can circumvent the permutation and scale ambiguity problems by maintaining a distortionless constraint in the look direction.MN beam former offers the possibility of steering both nulls and sidelobes, the former towards the undesired signal and its reflections, the latter towards reflections of the desired signal. To verify that the MN beam forming algorithm form sidelobes directed towards the reflection of the desired signal [3].

Demba E Ba et al. have proposed the enhanced MVDR beamforming for an array of directional microphone. Minimum Variance Distortionless Response beam formation is a most popular technique for speech enhancement application. Traditional MVDR algorithm does not give a better result over continuous variable gain system e.g. circular microphone geometry which has variable gain. So that traditional MVDR not suitable for circular geometry [12]. Dem-ba E Ba discussed the enhanced MVDR (eMVDR) which is useful for the continuous variable gain system.

3. PROPOSED SYSTEM

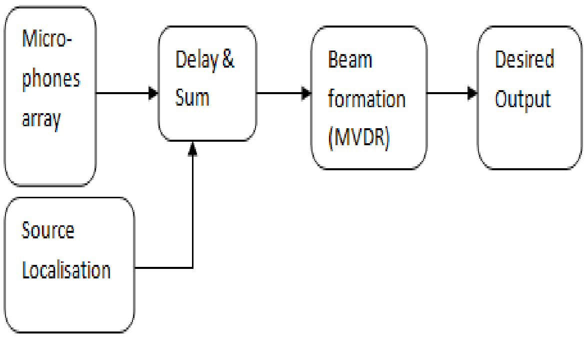

Fig 3: Proposed System Diagram

Figure 3 shows the block diagram of proposed system. The system is thought to implemented by using a microphone array (probably of 4 directional microphones).
Signal reduced by using primary algorithm Delay and Sum. Further reduction of the signal done by using conventional algorithm MVDR and finally, we get desired output.
Here it is assumed that distance between consecutive microphones is constant i.e. 'd'. The number of microphones denoted by Sj Delay(in time) between consecutive microphones is shown by equation (1). Angle to be found by using delay, the distance between microphone and velocity of sound. The distance between two microphones is denoted by di,j here, i is starting a number of microphone and j is the last number of a microphone.

$$Delay_{time} = \frac{d_{i,j} \times \cos\theta}{c} \text{.......................................(1)}$$

Weighting factor for the beam formation using MVDR algorithm is calculated by equation(2)

$$w^H MVDR(\omega) = \frac{v^H(k,\omega)\sum_N^{-1}(\omega)}{v^H(k,\omega)\sum_N^{-1}(\omega)v(k,\omega)} \text{....................................(2)}$$

4. RESULTS AND SIMMULATION

For this project by using MVDR algorithm discussed the effect of number of microphone(S),effect of distance between two microphones(d), effect of frame size on the beam formation response. This effect shown by Simulation results which is generated by MATLAB code.

Table 1. Effect of number of microphones

Sr. No.	Number of Microphones(S)	Mean Sqaure Error (MSE) in (dB)
1	2	-20.0239%
2	4	-31.7035%
3	6	-33.6351%
4	8	-33.8215%
5	10	-33.3862%
6	12	-33.1005%
7	14	-33.4404%
8	16	-33.6207%
9	18	-33.8334%
10	20	-33.8261%

Effect of algorithm of Delay and Sum, MVDR is shown below,

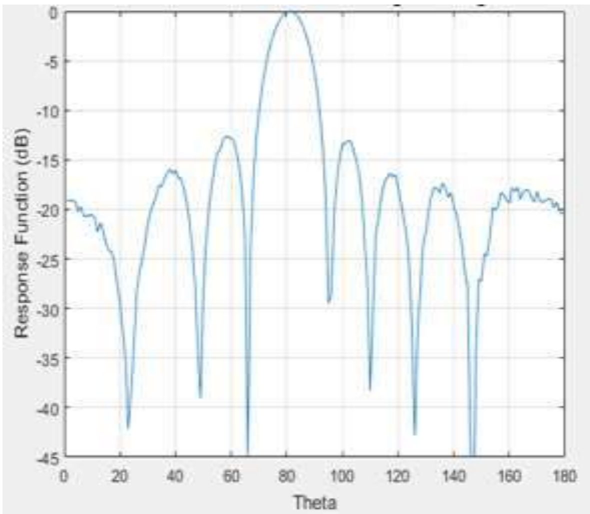

Fig.4. Effect of Delay & Sum

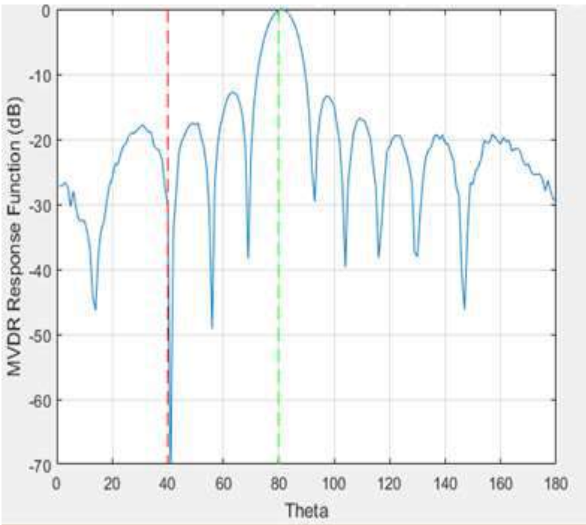

Fig 5 : Effect of MVDR Algorithm

This below effect shown by Simulation results which is generated by MATLAB code.

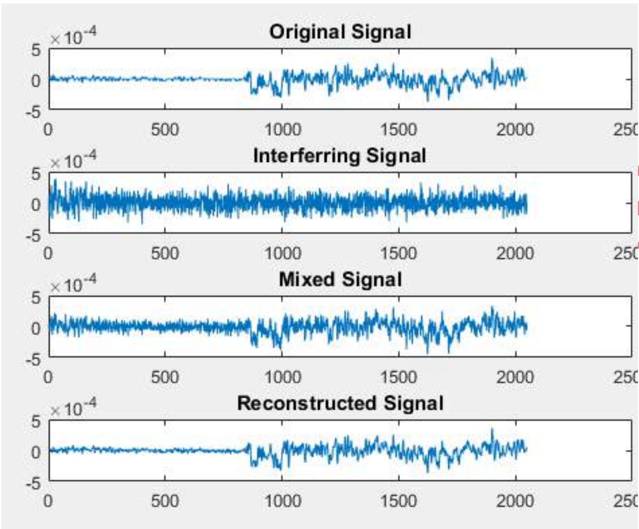

Fig 6 : Effect of MVDR Algorithm on Speech Signal

5. CONCLUSION

This algorithm is less complex than others and is immune up to some extent to the reverberant noise which desired here. A MVDR gives the better results than another algorithm.

Effect of S: As the number of microphones increases, the MSE decreases and width of the desired lobe also decreases giving a sharp peak at desired angle. In above table, at S=20 gives better result.

MVDR algorithm gives better result on speech signal and reconstructed signal matches with the original signal.

6. ACKNOWLEDGEMENT

Patil Balasaheb is currently an Assistant professor in the Department of Electronic & Telecommunication at Vidya Pratishthan's College of Engineering, Baramati, India. He received M.E. in E&Tc (Microwave) from Pune University in 2010. His teaching experience is 10 years. He has 12 International conference papers. His research interest includes light wave systems and optical access networks based on WDM-PON. VLSI and digital image processing.

7. REFERENCES

[1] M.Omologo, M. Matassoni, and P. Svaizer, “Environmental conditions and acoustic transduction in hands-free speech recognition,” Speech Commun., vol. 25, no. 1–3, pp. 75–95, 1998.

[2] M.Wölfel and J. McDonough, "Distant Speech Recognition". Hoboken, NJ: Wiley, 2009.

[3] M. Omologo, “A prototype of distant-talking interface forcontrol of interactive TV,” in Proc. Asilomar Conf. Signals, Systems and Computers (ASILOMAR), Pacific Grove, CA, 2010, pp. 1711–1715.

[4] M. Lincoln, I. McCowan, I. Vepa, and H. K. Maganti, “The multi-channel Wall Street Journal audio visual corpus (MCWSJ-AV): Specification and initial experiments,” in Proc. IEEE Workshop Automatic Speech Recognition and Understanding (ASRU), 2005, pp. 357–362.

[5] J. McDonough, K. Kumatani, T. Gehrig, E. Stoimenov, U. Mayer, S. Schacht, M. Wölfel, and D. Klakow, “To separate speech!: A system for recognizing simultaneous speech,” in Proc. MLMI, 2007, pp. 283–294.

[6] Kenichi Kumatani, John McDonough, and Bhiksha Raj, "Microphone array processing for distant speech recogni- tion", IEEE Signal Processing Magazine, pp. 127-140, ISBN-1053-5888, November 2012.

[7] Ali pourmohammad, "N-dimensional N-microphone sound source localization", EURASIP Journal on Audio, Speech and Music processing 2013.

[8] "Acoustics source localization", International Journal of Advanced Research in Electrical, Electronics and In- strumentation Engineering,Vol.

2, Issue 2, pp. 933-939 , ISSN (Print) : 2320 – 3765, February 2013.

[9] Dan Li and Yu Hen Hu ,"Energy Based Collaborative Source Localization Using Acoustic Micro-Sensor Ar-ray", J. EURASIP Applied Signal Processing, vol. 2003, no.4, pp. 321-337, 2003.

[10] G. C. Carter, ―"Coherence and time delay estimation", Proceedings of the IEEE, vol. 75, no. 2, pp. 236-255, ISSN: 0018-9219, February 1987.

[11] Ivan J. Tashev,"Sound Capture and Processing Practical Approaches", John Wiley & Sons Ltd., edition 2009.

[12] Francis A. Reed, "Time Delay Estimation Using the LMS Adaptive Filter-Static Behaviour", IEEE Transac-tion on Acoustics, Speech and Signal processing, Vol. 29, no. 3, pp.561-571, June 1981.