Sub Band Beamforming for Speech Enhancement using Various Beamforming Techniques

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ABSTRACT

The purpose of this project is to provide an efficient Speech Enhancement technique using various beam forming techniques. Speech is a broadband signal. Most speech signal receiving systems operate in noisy environments, where the desired speech signal is corrupted by interfering signals such as competing speakers and noise sources Speech enhancement can be achieved in a number of ways. By means of a literature study, three beamformers were investigated and presented: Delay and Sum (DAS), Minimum Variance Distortionless Response (MVDR) and the Generalized Sidelobe Canceller (GSC). Due to time constraints the DAS and MVDR were chosen for further simulations and testing, based on the terms of reference. Both beamformers were extensively simulated and tested in Matlab. The results were used to compare the two beamformers based on their white noise suppression, punctuated noise source suppression, frequency range and computational complexity. We implemented beamforming using two microphones. Through this project, we observed that Generalized Sidelobe Canceller(GSC) Technique is easy to implement on real time due to less complexity of calculating beam weights and the performance is on par with Minimum Variance Distortionless Response (MVDR) technique but the results are not as accurate because of limitations in real world.

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1.INTRODUCTION

Speech is a broadband signal. Most speech signal receiving systems operate in noisy environments, where the desired speech signal is corrupted by interfering signals such as competing speakers and noise sources. In many applications, there is a need to separate the multiple sources or extract a source of interest while minimizing undesired interfering signals and noise**. Speech enhancement** aims to improve speech quality by using various algorithms. The objective of enhancement is improvement in intelligibility and/or overall perceptual quality of degraded speech signal using audio signal processing techniques.

Speech enhancement can be achieved by

- Single channel
- Multi channel

Beamforming or **spatial filtering** is a signal processing technique used in sensor arrays for directional signal transmission or reception. This is achieved by combining elements in a phased array in such a way that signals at angles experience constructive interference while others experience destructive interference. It can be used for a myriad 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. *Beamforming techniques basically approach the problem from a spatial point of view. 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*

Types of Beam forming

Based on computational complexity and filters used:

- Fixed beamforming
- Partial adaptive beamforming
- Fully adaptive beamforming

Based on range of frequencies of interest:

- Narrowband
- Broadband

Based on Beam width:

- Directive beam
- Wide beam

1.1 PROBLEM FORMUALTION:

Consider a uniform linear microphone array in a noisy environment, and a Sound(speech) source is present on the broadside of the array and located along the axis of the array. A noise source is present in the end fire direction of the array and is also stationary. The speech signal is to be enhanced by reducing the background noise. Beam forming is susceptible to spatial aliasing

The main goal is to create a beam forming algorithms and compare them which spatially filters noise significantly. By means of simulations and tests we can then give a verdict on the feasibility of beam forming with 2 microphones, and check the validity of the assumptions and choices.

1.2 PROBLEM SOLUTION:

Having looked at the goals and restrictions of the algorithm, there are three possible solutions. Firstly, we will look at the simplest possible solution, a Delay-and-Sum(DAS) beam former. Secondly, we will investigate the possibilities of a Minimum Variance Distortionless Response (MVDR) beam former. Thirdly, we will explore the Generalized Sidelobe Canceller (GSC) beam former. Detailed discussion will be in Chapter 2, and Chapter 4 will further discuss the advantages and disadvantages of the given solutions.

2.IMPLEMENTATION

The implementation of this project is done in two stages

- MATLAB Simulation.
- Real Time Implementation on DSP board.

2.1 MATLAB IMPLEMENTATION:

We Implemented three beam forming algorithms in Matlab and reviewed the advantages of them

The three beamforming algorithms are

a) Delay and Sum beamformer
b) Minimum variance Distortion less beamformer
c) Generalized side lobe canceller

while implementing on Matlab we made some theoretical assumptions they are as follows.

- Number of microphones are assumed to be 2
- Speech source is considered to be in the broad side of the microphone array located along the axis
- Noise is assumed to come from end-fire directions.
- A block size of 32 samples is assumed with a sampling frequency of 8Khz.

A) Delay and Sum Beamformer:

In this beam former where the signals at the microphones are delayed and then summed in order to combine the signal arriving from the direction of the desired source coherently, expecting that the interference components arriving from off the desired direction cancel to a certain extent by destructive combining[1].

The delay and sum beamformer as shown in figure is simple in its implementation and provides for easy steering of the beam towards desired source.

DRAWBACKS:

- The performance of the delay-and-sum Beam former in reverberant environments is often insufficient.
- The DSB yields an improvement in SNR in the target direction, but its fixed choice of weights limits its ability to achieve optimum output.

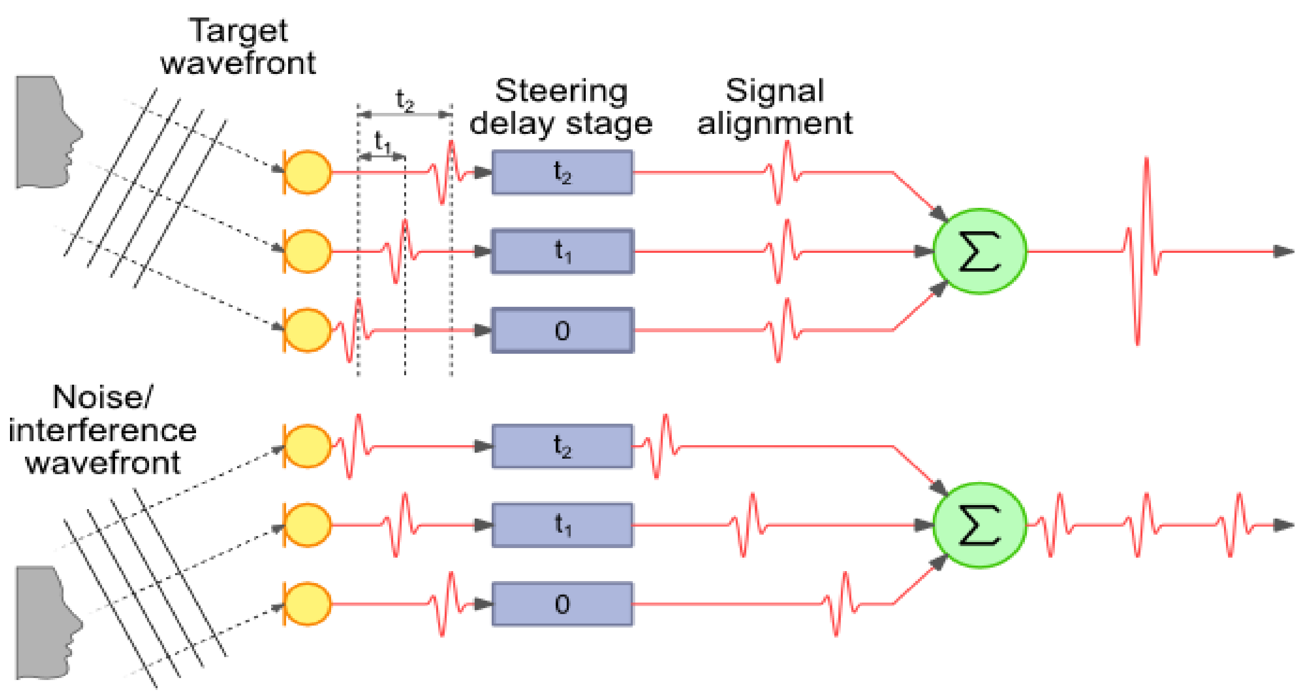


Fig .2.1. Block Diagram of Delay and Sum

B) Minimum Variance Distortion-less Beamformer (MVDR):

In MVDR beamformer the power of the output signal is minimized under the constraint that signals arriving from the assumed direction of the direction of the desired speech source are processed without distortion[2].

In MVDR, we employ adaptive beamforming (i.e. optimizing the beam weights such that minimizing the output power along with maintaining unity gain in the target direction)

The most famous constrained Minimum variance algorithm is Frost's Algorithm this algorithm requires knowledge of desired signal location and the array geometry in order to define a constraint on the filter weights such to ensure that the response to the signal coming from the desired direction as constant gain and linear phase. This is achieved in conjunction with a minimization of the received energy components originating from other directions.

DRAWBACK:

The input signal correlation Matrix(RXX) is not stationary and not known ahead of time for an adaptive array. The beam weight computation is difficult to achieve.

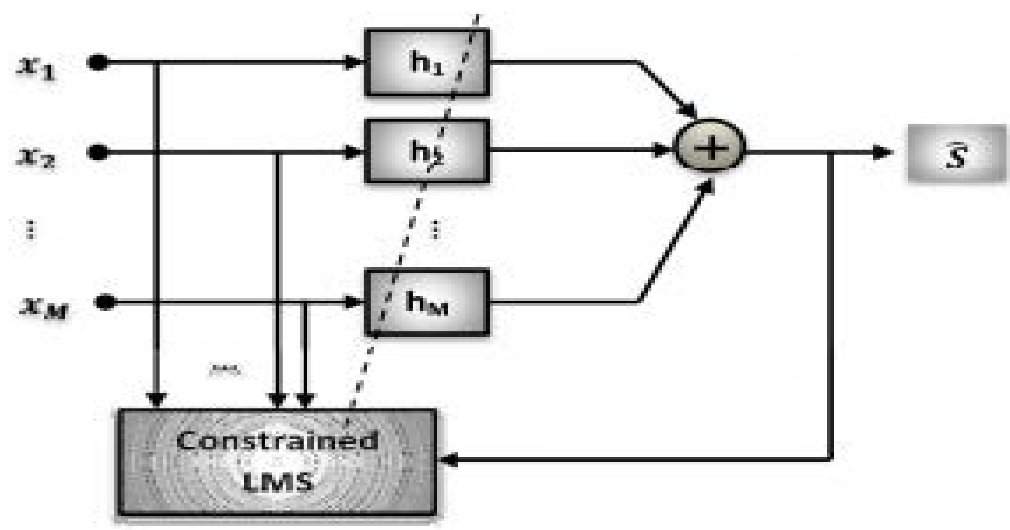


Fig .2.2. Block Diagram of Minimum Variance Distortionless Response

C) Generalized Side Lobe Canceller:

An improved solution to constrained adaptive beamforming problem decomposes the adaptive filter and sum beamformer into a fixed beamformer and adaptive multichannel noise canceller the resulting system is termed as generalized side lobe canceller which is shown below[3].

Here the constraint of a Distortionless response in loop direction is established by the fixed beamformer the noise canceller can then be adapted without a constraint

The Fixed beamformer is implemented using delay and sum beamformer to avoid distortions of the desired signals, the input to the adaptive noise canceller must not contain the desired signal therefore a blocking matrix is employed such that the noise signals are free of the desired signal.

The adaptive noise canceller then estimates the noise components at the output of the fixed beamformer and subtracts the estimates.

DRAWBACKS:

- As the speech is broad band spectrum, it is not possible to completely eliminate the desired signal. So, the output signal we obtained is attenuated to some extent
- As the environment is not completely non-reverberant medium. The microphones obtain some part of noise signals from the desired direction.

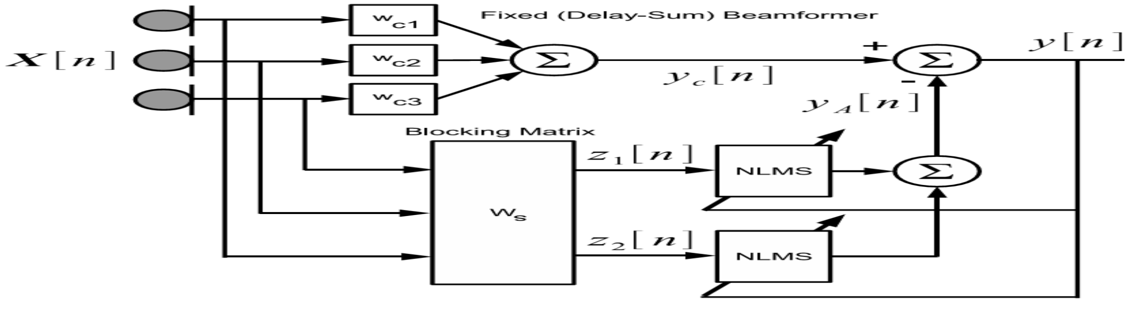


Fig .2.3. Block Diagram of Generalized of Side Lobe Canceller

2.2 Real Time Implementation:

The real-time implementation of this project is achieved in visual Dsp++ environment, the programming of DSP processor is written in C language.

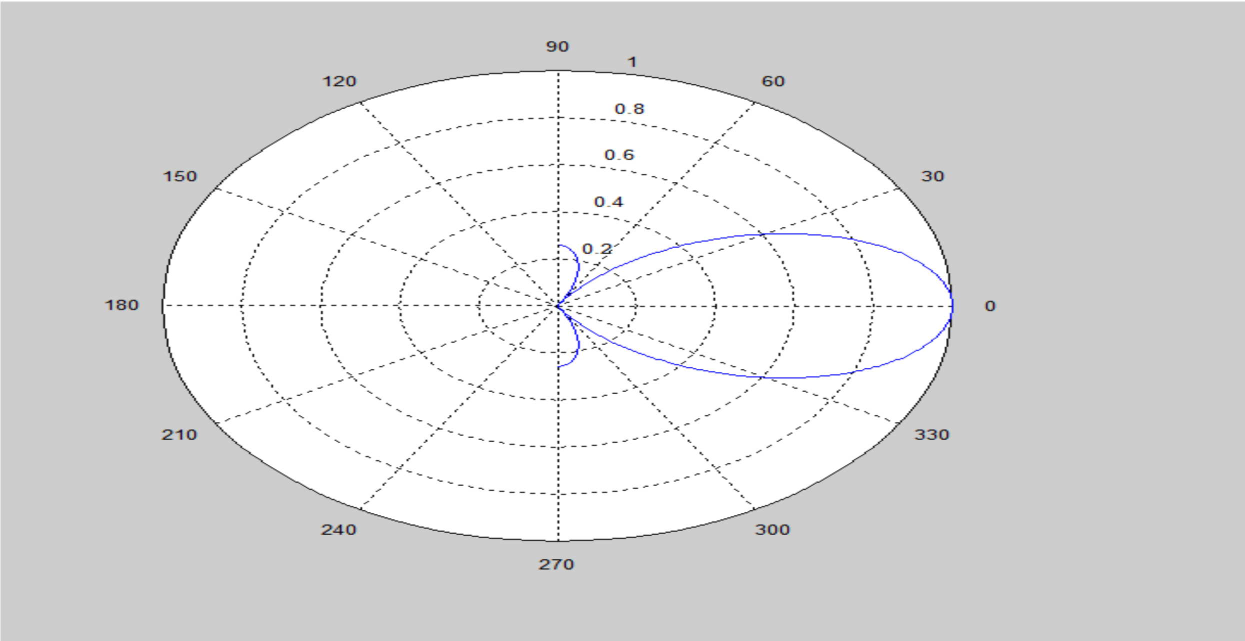
The block size of the system is assumed to be 32 samples and the sampling frequency is set to 8Khz the input is given through microphone array the blocks of input are acquired using an audio interrupt to the Dsp processor

The whole operation of GSC is written in a function called process the process function was initiated whenever there is an audio interrupt a pair of push button is used for enabling and disabling the GSC algorithm whenever necessary the final NLMS weights from the Matlab implementation are used as a FIR filter instead of adaptive NLMS filter and so no feedback is given and Voice Activity Detector (VAD) is not used while implementing the algorithm, this is done because there is a stability problem in the NLMS filter in real time implementation.

3.Results

3.1 Matlab Results

Beamformer Output using Different Arrays of Microphones:



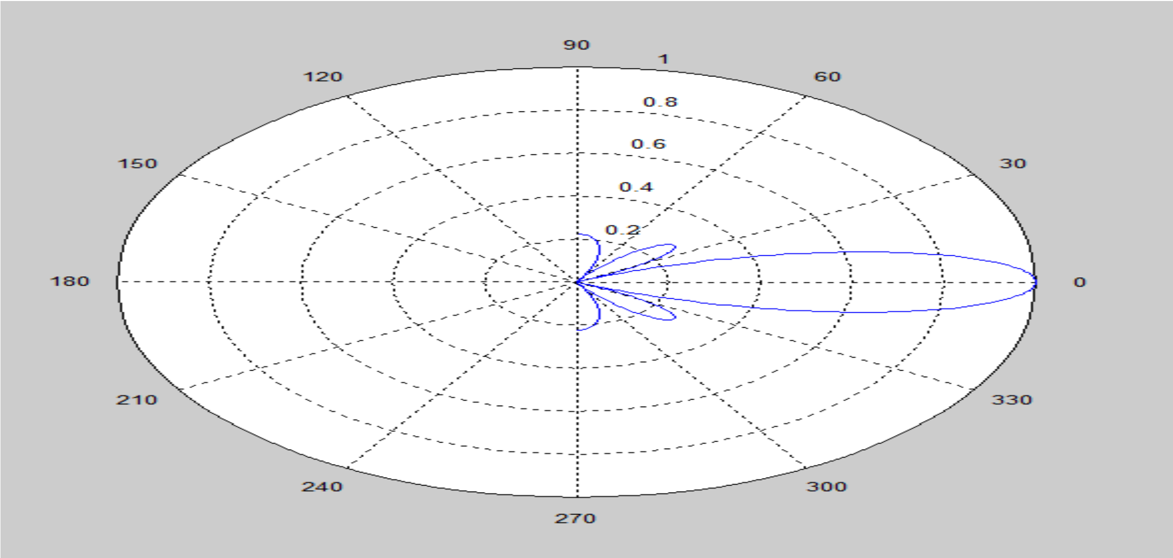


Fig .3.1.1 Beamforming output for (a) 2 microphones (b) 4 microphones

Normalized Output for Different Microphone Arrays:

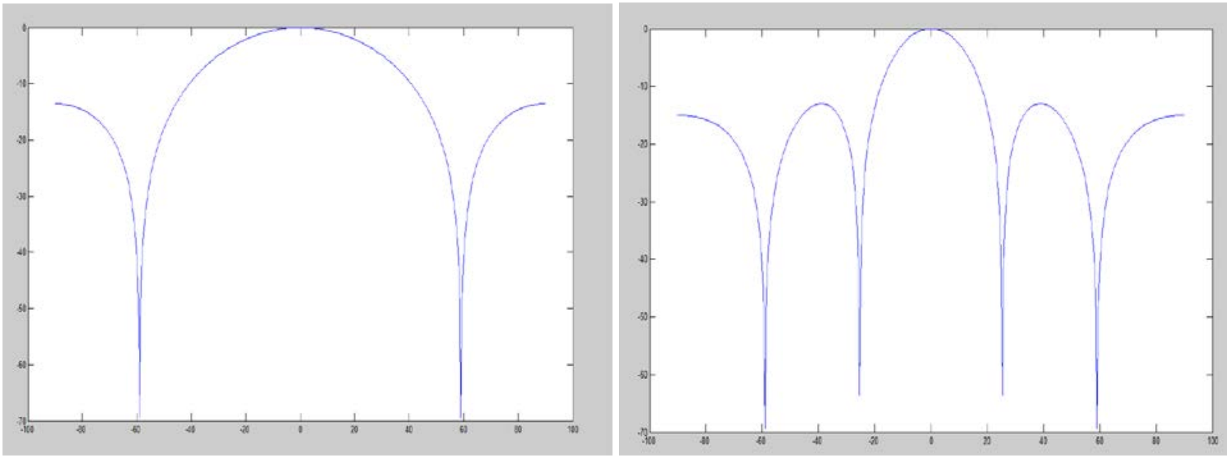


Fig.4.1.2 Normalized output for (a)2 microphones (b) 4 microphones

The MATLAB results show that increase in microphones increases the accuracy of the of receiving signals in the target direction while attenuating signals from the other directions.

Delay and Sum Output:

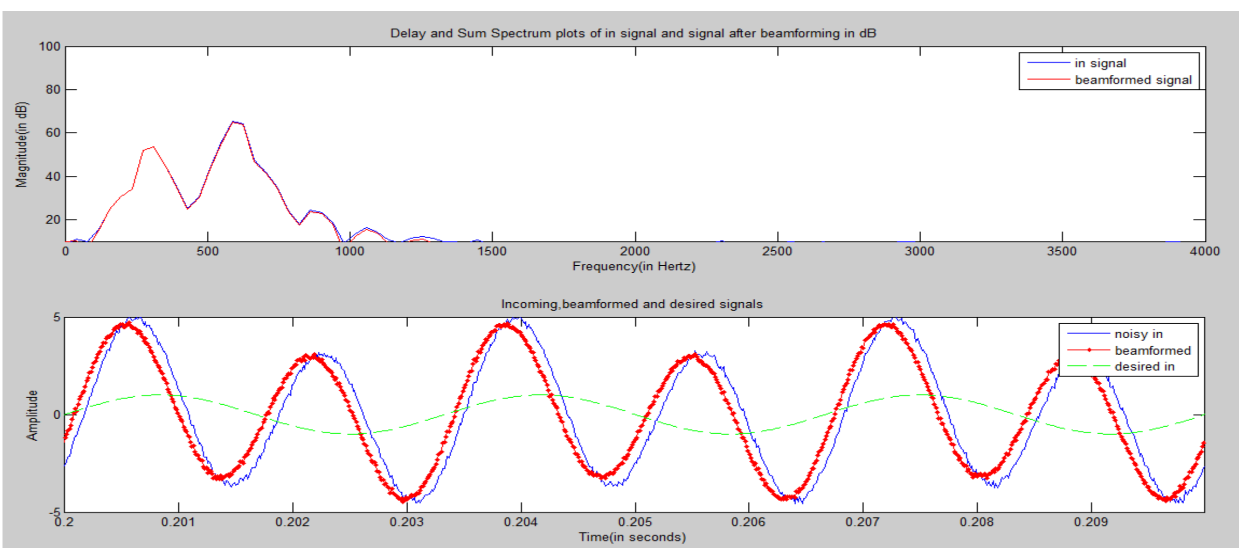


Fig.3.1.3 Delay and Sum spectrum plots of in signal & signal after beamforming in dB

In the Delay and Sum Beamformer, the obtained beamformed signal(red) though contains desired signal(green) the noise signal is not completely attenuated due to fixed beam weights.

Minimum Variance Distortionless Response Output:

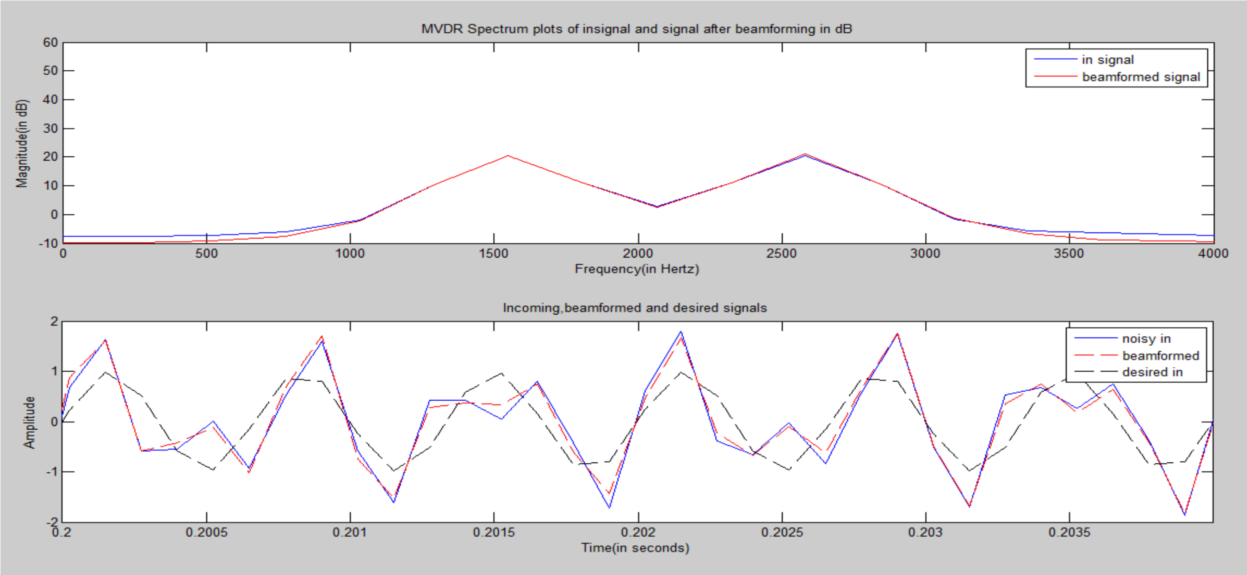


Fig. 3.1.4 MVDR spectrum plot of in signal and signal after beamforming in dB

In MVDR beamformer, the beam formed signal (red dotted) is approximately similar to desired signal (blue dotted) because we use adaptive beam weights based on spatial characteristics of the target signal.

3.2 DSP Implementation Results:

While implementing the MVDR technique on DSP boards we came across problem for calculating the beam weights adaptively because the input signal correlation matrix(RXX) is not stationary and not known ahead of time for an adaptive array. So, we implemented Generalized Side lobe Canceller(GSC) in DSP boards due less complexity in calculating beam weights.

Simulating the Matlab equivalent on a Dsp simulator using Dsp++ environment by giving a block of data input from Matlab.

```
E[0] = 0.301700
E[1] = 0.307877
E[2] = 0.308285
E[3] = 0.305238
E[4] = 0.298028
E[5] = 0.283623
E[6] = 0.259508
E[7] = 0.225134
E[8] = 0.185460
E[9] = 0.138767
E[10] = 0.087920
E[11] = 0.036648
E[12] = -0.014722
E[13] = -0.064246
E[14] = -0.117096
E[15] = -0.169508
E[16] = -0.218535
E[17] = -0.261705
E[18] = -0.296628
E[19] = -0.319253
E[20] = -0.326302
E[21] = -0.319592
E[22] = -0.303620
E[23] = -0.281931
E[24] = -0.256456
E[25] = -0.231427
E[26] = -0.207867
E[27] = -0.189141
E[28] = -0.176526
E[29] = -0.168910
E[30] = -0.168353
E[31] = -0.176704
```

3.3 Problems Faced During Implementation:

- While hearing to the audio output of the system at different stages it was observed that blocking filter was not as efficient as required and the desired speech signal is still present in the blocking matrix output which depicts the signal leakage in the system leading to a distortion in the enhanced speech output.
- As the speech is broad band spectrum, it is not possible to completely eliminate the desired signal. So, the output signal we obtained is attenuated to some extent.
- As the environment is not completely non-reverberant medium. The microphones obtain some part of noise signals from the desired direction.

4. Conclusion and Future Scope

Through this project, we observed that Generalized Sidelobe Canceller(GSC) Technique is easy to implement on real time due to less complexity of calculating beam weights and the performance is on par with Minimum Variance Distortionless Response (MVDR) technique but the results are not as accurate because of limitations in real world.

We obtained better results in Adaptive Beamforming (GSC) compared to fixed Beamforming (Delay and Sum).

In our project, we implemented beamforming using two microphones the performance of the system can further be improved by using more number of microphones in the array and different array configurations.

Blocking output can further be improved by using adaptive blocking filter in GSC technique.

Bibliography

[1] Pallavi Agrawal, "Dual Microphone Beamforming Algorithm for Acoustic Signals," in International Journal of Computer Applications, Vol. 129-No. 11, November 2015.

[2] Zohra Yermeche, "Sub-band Beamforming for speech enhancement in Hands-Free Communication," Department of Signal Processing, School of Engineering, December 2004.

[3] Hidri Adel, "Beamforming Techniques for Multichannel Audio Signal Separation" National Engineering School of Tunis, December 2004

Individual Contribution Towards Project

Chaithanya Kumar Reddy Veluru:

Generalised Side Lobe Canceller and Report Writing

Harindra Sai Tej Muchrala:

Minimmum Varriance Distorionless Response and Report Writing

Raj Sekhar Sana:

Delay and Sum Beamformer, Beam pattern and Report writing