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Application of Microphone Array Beamforming Approaches for Acoustic Noise Reduction in Speech Signals

***N.K.Manikanta Reddy*[1], *A S N Murthy*[2], *Dr.D Elizabeth Rani*[3]**
[1]*Master of Technology, DSSP, GITAM University, Visakhapatnam. Email: nkmreddy29891@gmail.com*
[2]*Sr. Assistant Professor, ECE Department, GITAM University, Visakhapatnam. Email: m_andhe@yahoo.com*
[3]*Professor, EIE Department, GITAM University, Visakhapatnam. Email: kvelizabeth@gitam.edu*

ABSTRACT

In this paper, proposed techniques aretoreduce acoustic noise in the speech using Microphone Array Beamforming. In the day to day life acoustic noise in speech communications giving trouble to several applications. The problem with microphones is that they not only capture the intended speech signal but also capture all acoustic sounds that are in the range of the microphones. All the unwanted acoustics sounds are referred as noise and interferences, here in this paper different approaches of the Beamforming, is verified in terms of noisy environment, tested by the noisy speech signals in both subjective and objective ways.

Key words: Microphone array beamforming, Delay and Sum Beamforming (DSB), GCC_PHAT, Generalized Side lobe Canceller (GSC).

1. INTRODUCTION

The decrease in speech quality is due to the pickup of background noise. Hands-free audio communication is now a major feature in communication systems as well as audio and video conferencing systems [1].

Using microphones nearer to the person or source, this idea is good but it has its own drawbacks. Firstly we can't place microphone very close to the person or to the source. Second thing is whenever the position between the microphone and the person changes the corresponding variations in sound takes place. Forthese drawbacks microphone array beamforming is the solution, gives high directional gain which result in same operation as the microphone placed near the person or source [2]. Some of the main beamforming techniques are delay and sum beamforming andGriffiths-Jim Beamformer commonly known as Generalized Side lobe Canceller (GSC).

These techniques improve the speech quality and intelligibility as well reduces the noise content in the speech signal. Noise reduction involves in large applications in the hands free communications [3].

Figure 1 shows the effect of noise on the speech signals arriving at the different microphones.

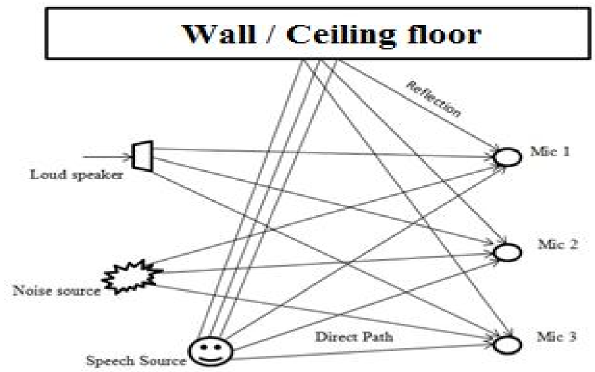

Figure 1: Illustration of different sounds and noise arriving at microphones.

2. DELAY AND SUM BEAMFORMER

A uniform linear array of microphones with equal spacing between them has been considered [4]. Delays are inserted at each microphone in order to compensate the incoming time differences of the speech signal. The aligned signals are added at the output. The signals after time aligned added constructively which are correlated and others will cancelled out. The output of the delayandsum beamformer is given by

$$y(t) = \frac{1}{N}\sum_{n=1}^{N} x_n(t-\tau_n) \qquad (1)$$

Where τ_n is the delay samples, sincethe delay in between the microphones having a non-integer value of time samples, delay is done by phase change in fourier domain instead of time domain [5].The interspacing between microphones (d) is given by

$$d = \frac{c}{2f_{max}} \qquad (2)$$

Where c is the speed of sound and $f_{max}$is the maximum speech signal frequency, Interspacing of microphone should not exceed the lowest wavelength of the signal.

Maximum number of delay samples is given by

$$\tau_n = \frac{(N-1)df_s}{c} \qquad (3)$$

Where f_s =sampling frequency of the signal.

Time delay of arrival (TDOA) estimation is done using Generalized Cross-Correlation (GCC_PHAT), so GCC_PHAT for two microphones m, n is given by

$$R_{m,n}(\tau) = \frac{1}{2\pi}\int_{-\infty}^{\infty} \frac{1}{|X(\omega)X(\omega)|} X(\omega)X^*(\omega)e^{j\omega\tau}d\omega \quad (4)$$

The beamformer gives the output of reduced noise in speech signal.

Figure 2 shows the block diagram of Delay and sum beamformer.

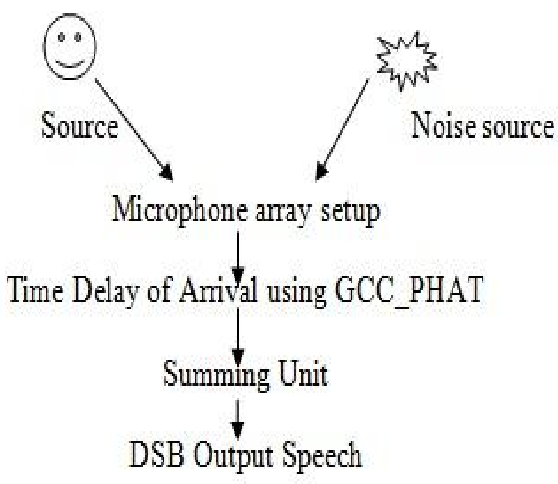

Figure 2: Block diagram of Delay and Sum Beamformer

The Delay and Sum Beamforming have its own drawback, in order to improve SNR, large number of microphones to be placed.

3. GRIFFITHS-JIM BEAMFORMER

Generalized side lobe canceller is a flexible structure, since of its fixed and adaptive blocks are separated and individually manipulated. The Figure 3 shows the Generalized Sidelobe Canceller in which itconsists delay and sum Beamformer and blocking matrix blocks along with adaptive block [6].
Adaptive part is simply group of filter that minimizes the power of the output, whereas blocking matrix (BM) used for minimize the noise power.

Blocking matrix should be having (N-1) rows which are linearly independent microphones, the sum of the rows is zero and the rows are linearly independent. Thus the dimensions of BM must be (N-1) or less than that [7].

Standard Griffiths-Jim Blocking Matrix (BM) is defined as

$$w_s = \begin{pmatrix} 1 & -1 & 0 & 0 \\ 0 & 1 & -1 & 0 \\ 0 & 0 & 1 & -1 \end{pmatrix} \quad (5)$$

Figure 3 shows the structure of the generalized side lobe canceller.

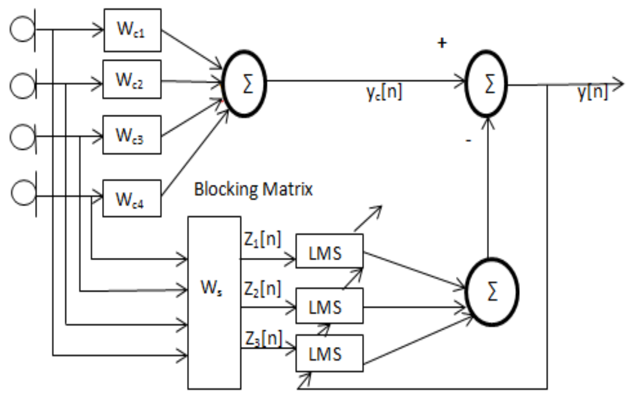

Figure 3: Structure of Generalized Side lobe Canceller

The output of BM is calculated as the matrix product of BM and the matrix of current input data.

$$z[n] = w_s x[n] \quad (6)$$

While in the adaptive section weight updating is done by using LMS algorithm and reference signal as y[n].

$$w_k[n+1] = w_k[n] + \mu y^*[n] z_k[n] \quad (7)$$

Where “*” represents the conjugate value and “μ” is the step size.

The final output for the GSC is given as

$$y[n] = y_c[n] - \sum_{K=1}^{N-1} w_k[n] z_k[n] \quad (8)$$

Where $w_k[n]$ is the k^{th} column of the tap weight matrix $W_s$and $z_k[n]$ is the k^{th} blocking matrix output and these two matrixes has same length [8].

4. SIMULATION RESULTS

In this section, performance of the two techniques is recorded, through whichthe noise levels are reduced. It is clearly observed by the spectrograms and the output speech. Both the DSB and GSC techniques improve the speech signal by reducing the noise content in the speech. In the experiment utterance of male voice taken as “seven zero six three four zero eight "which is sampled at 8 kHz frequency, and the speech is corrupted by car noise and pink noise. A uniform four linear array of microphones with equal spacing between them is considered. Here four microphones having spacing (d) between them is 0.0433 mts, total array width as 0.1298 mts.The results of the delay and sum beamformer and the Generalized Side lobe Canceller have been given.

Figure 4 is the input clean speech where its time domain and its spectrum and spectrogram are shown. Figure 5 is the car noise. Figure 6 is the noisy speech signal received at the microphone and figure7and 8 are the results after applying the beamforming techniques.

4.1 Car Noise:

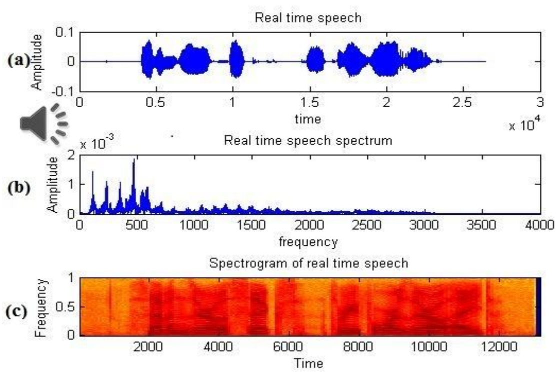

Figure 4: (a) Original clean speech (b) Its spectrum (c) Its Spectrogram.

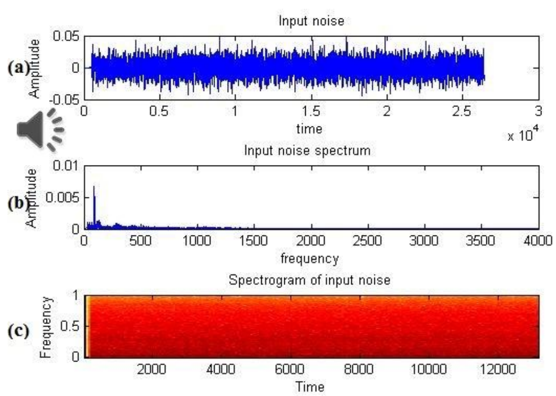

Figure 5: (a) Car Noise (b) Its Noise spectrum (c) Its Spectrogram.

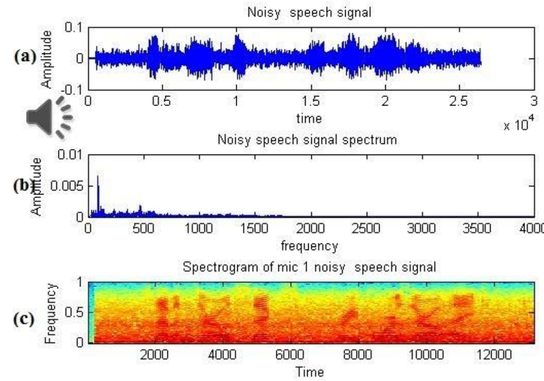

Figure 6: (a) Noisy speech signal (b) Its Spectrum (c) Its Spectrogram.

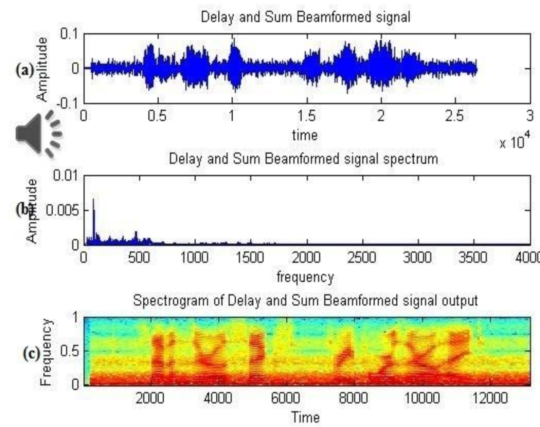

Figure 7: (a) Delay and Sum Beamformer output (b) Its Spectrum (c) Its Spectrogram.

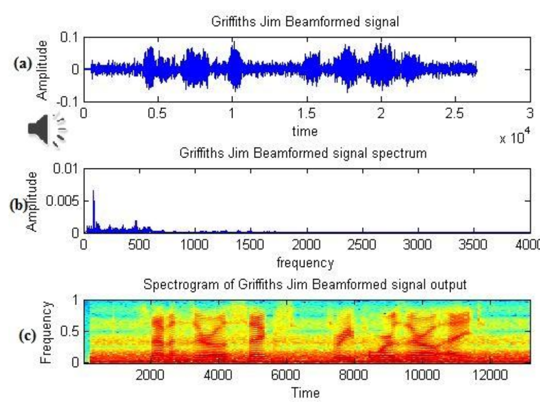

Figure 8: (a) Griffiths JimBeamformer output (b) Its Spectrum (c) Its Spectrogram.

Table 1 shows Signal to Noise Ratio of Input clean Speech and the output of each technique.

Table 1: Comparison between Input and Output Speech Signals for different Beamforming techniques

Sl. No.	Number of Microphones (N)	Input Speech SNR (dB)	Output Speech SNR (dB) DSB	Output Speech SNR (dB) GSC
1	2	-7.7973	-1.3653	-1.2678
2	4	-7.7973	-1.1964	-0.9919
3	6	-7.7973	-1.1129	-0.8710
4	8	-7.7973	-1.0231	-0.7639
5	10	-7.7973	-0.6044	0.8766

Figure 9 shows the SNR of output speech of GSC is greater than the DSB technique.

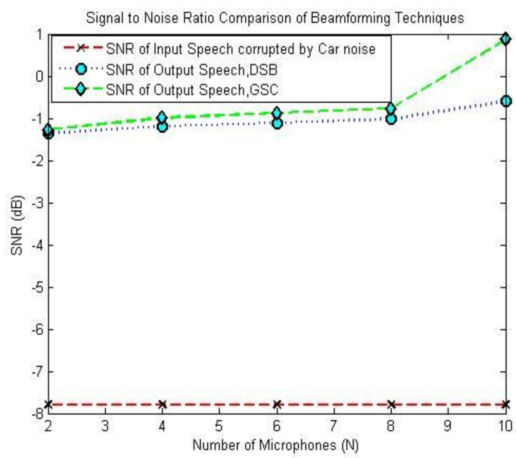

Figure 9: Signal to Noise Ratio Comparison of DSB vs GSC

The following table 2 tabulates the SNR values for all the simulations carried out along with the Angle of Arrival of the incoming source Signal.

Table 2: Comparison SNR vs Angle of different techniques, speech corrupted by Car noise

Sl. No.	Angle of Arrival(θ)	Input Speech SNR (dB)	Output Speech SNR (dB) DSB	Output Speech SNR (dB) GSC
1	0	-7.7973	-1.2959	-1.2724
2	30	-7.7973	-1.2959	-1.1600
3	60	-7.7973	-1.2356	-1.0618
4	90	-7.7973	-1.1964	-0.9919
5	120	-7.7973	-1.2356	-1.0618
6	150	-7.7973	-1.2959	-1.1600
7	180	-7.7973	-1.2959	-1.2724

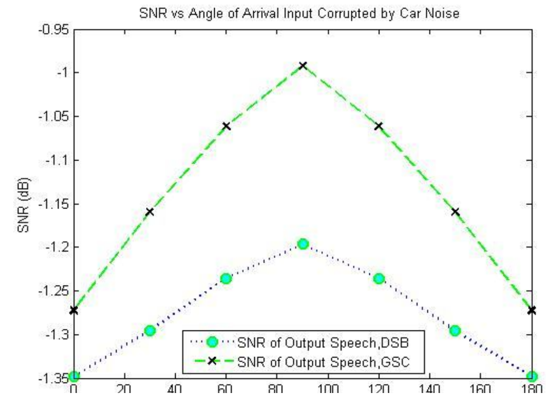

Figure 10: SNR vs Angle of Arrival of speech corrupted by Car noise.

The following table 3 tabulates the SNR values for all the simulations carried out along with the Step size (mu).there is an increase in the SNR as step size is increasing Figure 11 shows its graphical representation.

Table 3: Comparison SNR vs Step size speech corrupted by Car noise

Sl. No.	Output Speech SNR (dB) GSC mu=0.1	Output Speech SNR (dB) GSC mu=0.3	Output Speech SNR (dB) GSC mu=0.5	Output Speech SNR (dB) GSC mu=0.7	Output Speech SNR (dB) GSC mu=0.9
1	-1.3914	-1.2895	-1.2051	-1.1332	-1.0711
2	-0.9367	-0.6633	-0.5446	-0.4827	-0.4460
3	-0.7762	-0.5156	-0.4249	-0.3821	-0.3580
4	-0.6537	-0.4256	-0.3549	-0.3229	-0.3057
5	0.7745	1.2271	1.4252	1.5485	1.6360

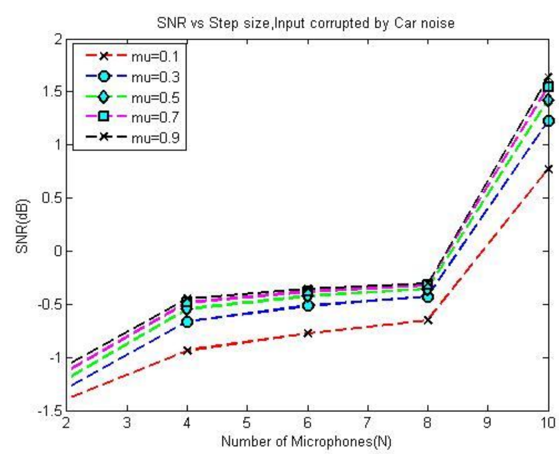

Figure 11: SNR vs Step size, speech corrupted by Car noise.

4.2Pink Noise:

The speech is corrupted by the Pink noise,Figure12 is the input clean speech where its time domain and its spectrum and spectrogram are shown. Figure 13 is the pink noise. Figure 14 is the noisy speech signal received at the microphone and figure15and 16 are the results after applying the beamforming techniques.

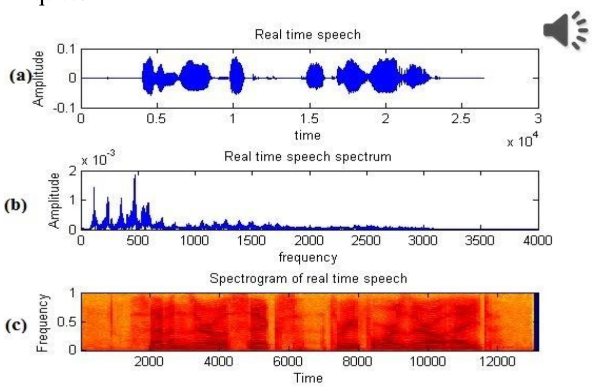

Figure 12: (a) Original clean speech (b) Its spectrum (c) Its Spectrogram.

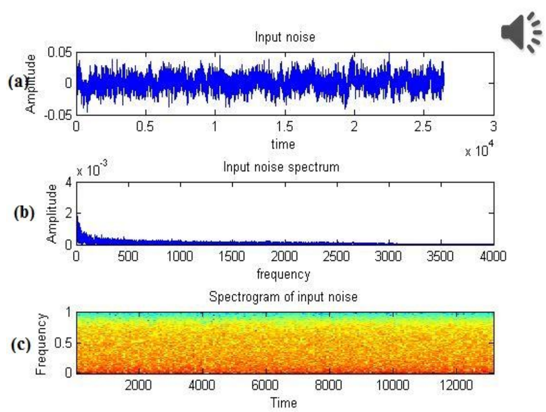

Figure 13: (a) Pink Noise (b) Its Noise spectrum (c) Its Spectrogram.

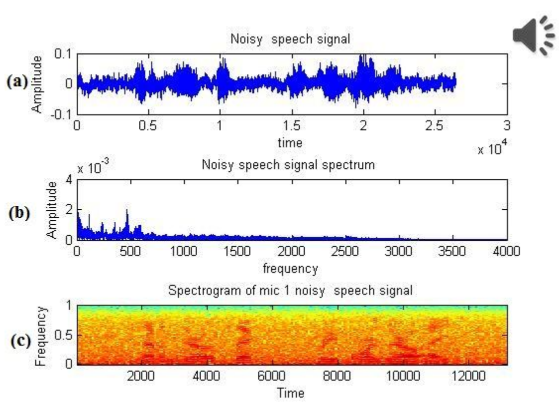

Figure 14: (a) Noisy speech signal (b) Its Spectrum (c) Its Spectrogram.

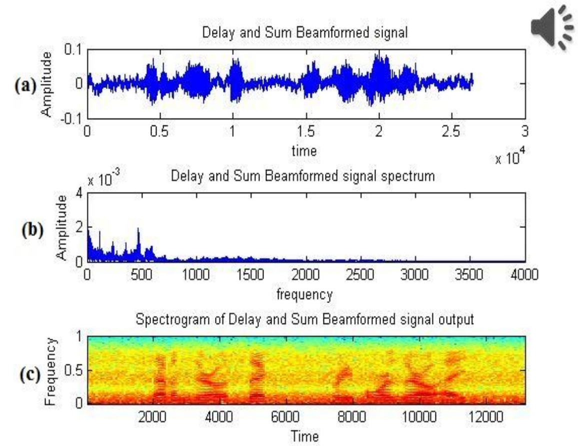

Figure 15: (a) Delay and Sum Beamformer output (b) Its Spectrum (c) Its Spectrogram.

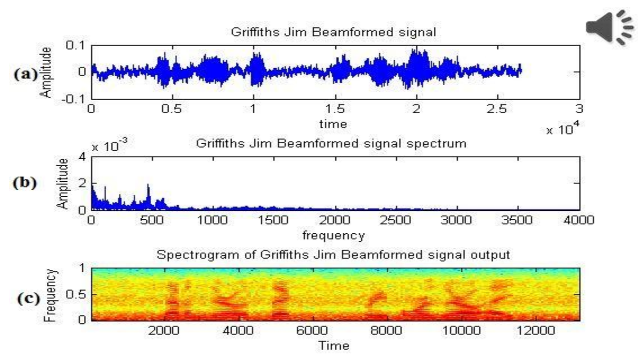

Figure 16: (a) Griffiths Jim Beamformer output (b) Its Spectrum (c) Its Spectrogram.

Table 4 shows Signal to Noise Ratio of Input clean Speech and the output of each technique.

Table 4: Comparison between Input and Output Speech Signals for different Beamforming techniques

Sl. No.	Number of Microphones (N)	Input Speech SNR (dB)	Output Speech SNR (dB) DSB	Output Speech SNR (dB) GSC
1	2	-7.4225	-1.3143	-1.2112
2	4	-7.4225	-1.0972	-0.7362
3	6	-7.4225	-1.0555	-0.6866
4	8	-7.4225	-1.0185	-0.6482
5	10	-7.4225	-0.9818	-0.6052

Figure 17 shows the SNR of output speech of GSC is greater than the DSB technique.

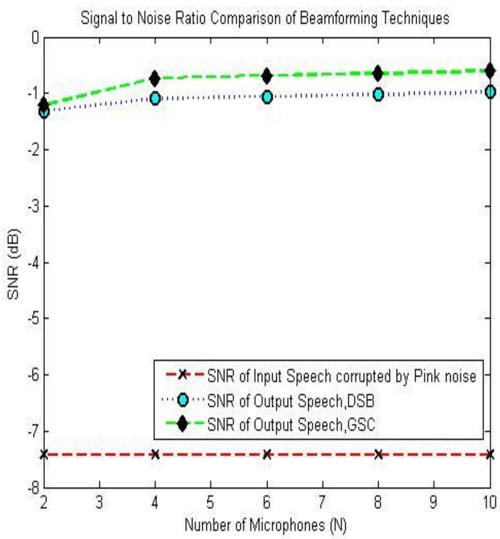

Figure 17: Signal to Noise Ratio Comparison of DSB vs GSC

The following table 5 tabulates the SNR values for all the simulations carried out along with the Angle of Arrival of the incoming source Signal.

Table 5: Comparison SNR vs Angle of different techniques, speech corrupted by Car noise

Sl. No.	**Angle of Arrival(θ)**	**Input Speech SNR (dB)**	**Output Speech SNR (dB) DSB**	**Output Speech SNR (dB) GSC**
1	0	-7.4225	-1.1998	-0.8257
2	30	-7.4225	-1.1817	-0.7962
3	60	-7.4225	-1.1324	-0.7302
4	90	-7.4225	-1.0972	-0.7310
5	120	-7.4225	-1.1324	-0.7302
6	150	-7.4225	-1.1817	-0.7962
7	180	-7.4225	-1.1998	-0.8257

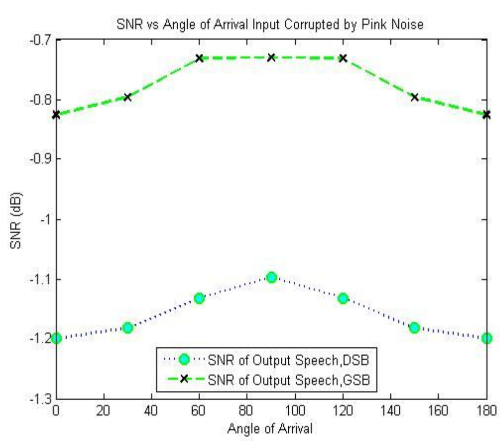

Figure 18: SNR vs Angle of Arrival of speech corrupted by Pink noise

The following table 6 tabulates the SNR values for all the simulations carried out along with the Step size (mu).there is an increase in the SNR as step size is increasing Figure 19 shows its graphical representation.

Table 6: Comparison SNR vs Step size speech corrupted by Pink noise

Sl. No.	**Output Speech SNR (dB) GSC mu=0.1**	**Output Speech SNR (dB) GSC mu=0.3**	**Output Speech SNR (dB) GSC mu=0.5**	**Output Speech SNR (dB) GSC mu=0.7**	**Output Speech SNR (dB) GSC mu=0.9**
1	-1.1594	-0.9190	-0.7929	-0.7101	-0.6497
2	-0.4956	-0.0021	0.2557	0.4204	0.5352
3	-0.4422	0.0512	0.3006	0.4562	0.5630
4	-0.4072	0.0805	0.3221	0.4710	0.5725
5	-0.3818	0.1054	0.3417	0.4857	0.5830

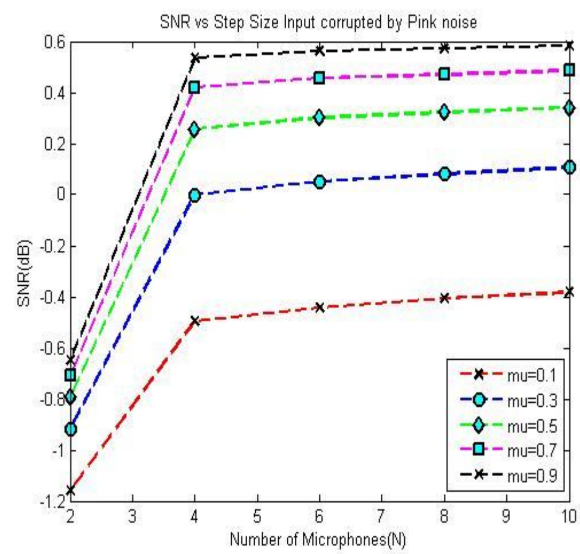

Figure 19: SNR vs Step size, speech corrupted by Pink noise.

5. CONCLUSION

In this paper, approaches for the microphone array beamforming techniques like Delay and Sum Beamforming (DSB), generalized side lobe canceller (GSC) are proposed. The experiential results show these are used for the acoustic noise reduction in the speech signal, SNR improvement is more in the case of GSC than in the DSB.

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