Close talk Speech Enhancement in Linear Microphone array for laptop application
Using Wiener beamformer, GSC with LMS and N-LMS

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ABSTRACT

Now a day’s communication through laptops is drastically increasing in numerous fields. During the communication between person to person through laptops the speech signals is contaminated by the other speech interference signals. In order to enhance the desired speech signal from the noisy environment there are many algorithms are proposed in speech signal processing.

This thesis work studies about the suppression of interference signals produced by the surrounding environments for the close talk applications of laptop. In this thesis work uses the three microphones of linearly equi spaced in 3D-co-ordinate system. The speech enhancement algorithms implemented in microphone array were wiener beamforming, Generalized sidelobe canceller using LMS and N-LMS. In order to enhance the desired speech signal with good quality, compares the result of each algorithm using quality metrics like SNR, SNRI and PESQ.

The implementation and validation of the algorithms is simulated in Matlab. The quality metrics taken is SNRI and PESQ. In PESQ the output signal is compared with the original clean speech signal and gives the quality measure of the output signal. The SNR tests were conducted for the different input SNR values according to 0dB, 5dB, 10dB, 15dB, 20dB and 25dB.

The Simulation result shows that the wiener beamformer effective noise suppression i.e SNRI is 27.9869dB and maintains the speech quality i.e PESQ measurement is 1.459. The effective noise suppression i.e SNRI of the GSC using LMS is 6.0206 dB higher than the wiener beamformer and speech quality is slightly incremental. Comparing the results of GSC using LMS and N-LMS algorithms, The GSC using N-LMS gives the effective noise suppression 3.48dB higher than the GSC using LMS and speech quality i.e PESQ is slightly decreases.
To My Parents
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LIST OF ACRONYMS AND ABBREVIATIONS:

FD Fractional delay
GSC Generalized sidelobe canceller
MA Microphone Array
SNR Signal to Noise ratio
SNRI Signal to Noise ratio Improvement
WBF Wiener beamformer
LMS Least Mean Square Algorithm
N-LMS Normalized Least Mean Square Algorithm
FIR Finite Impulse Response
DMI Direct Matrix Inversion
GSC-L Generalized sidelobe canceller using LMS
GSC-N Generalized sidelobe canceller using N-LMS
1.1 Overview

In closed talk laptop applications usually the speech signals are contaminated by the interference signals. In some conditions both desired and interference signals are used to have the same temporal frequency, in this situation general temporal filtering cannot works to separate the interference signals from desired speech signal. In this case spatial filtering is used to exploit the desired and interference signals depending on their spatial location distributed in the space [1].

In speech enhancement process the use of microphone array is having a key role to implement. This thesis work studies about the implementation of microphone array with the wiener beam forming algorithms that which is used to enhance the signal in desired direction and considers the remaining spatial location as noisy environment. The concept of wiener beamformer uses in the implementation of Generalized sidelobe canceller. This thesis work studies the working of generalized sidelobe canceller for both stationary and non-stationary signal environments using the least mean square algorithm (LMS) and normalized least mean square algorithms (N-LMS) for adaptive noise cancellation.

The implemented algorithms are measured for the different positions of source and interference signals and validate the result measuring with the quality metrics like SNR, SNRI and PESQ.

1.2 Research question:-

In generalized side lobe canceller for adaptive noise cancellation purpose comparing the two algorithms Least mean square algorithm (LMS) and Normalized least mean square algorithm (N-LMS) and which algorithm is works effectively under different positions of source and interference signals.
1.3 Objective of thesis:

The primary objective of this thesis is to suppress the noise present in desired speech signal by using speech enhancement algorithms without losing the quality of speech.

The amount of noise suppressed is measured by using the certain metrics like SNR, SNRI and PESQ. Using these metrics to successfully figured out the noise suppression range for different implementations of algorithms.

1.4 Motivation of thesis:

The motivation of this thesis is to implement the algorithms that which enhances the desired speech signals effectively without losing its speech quality and also suppress the noise signals by exploiting the spatial location of speech and interference signals.

1.5 Thesis outline:

This thesis report consists of 8 chapters each chapter contains subsections. This thesis is presented as a part of double degree in Master of Science in electrical engineering with emphasis on signal processing. Chapter-1 introduces about the introduction of thesis work. It explains the research question, objective and motivation of the thesis work. Chapter-2 discusses about the introduction of fractional delay filters, Implementation of fractional delay in microphone array and fractional delay measurements for source and interference signals. Chapter-3 discusses about the introduction of microphone array, limitation of linear equi spaced microphone array, microphone array speech processing, and beamforming techniques. Chapter-4 discusses about the generalized sidelobe canceller structure, time domain formulation of GSC and Implementation of LMS and N-LMS algorithms. Chapter-5 explains about the microphone array positioning in Matlab and fractional delay measurements in Matlab. Chapter-6 gives explanation about the Evaluation and comparison of simulated result in Matlab. Chapter-7 explains the conclusion and future work of the thesis. Chapter-8 gives the Bibliography.
CHAPTER-2

2 Fractional delay filters

2.1 INTRODUCTION

Fractional delay filters are very essential for beam steering process in microphone arrays. The incident waves at each microphone are delayed with sum of integer and fractional part of the Filters, Fractional delay filters are form of digital filters with band limited interpolation [2].

Band limited interpolation is the process that is used to analyze the signal sample at an arbitrary point in time, sometimes it is used to analyze the signal sample when it is located somewhere between the sampling points [2].

The value of sample that which is obtained with band limited interpolation is accurate with the half the sampling rate frequency (Fs/2). According to this process it can be exactly reconstruct continuous time signal from the sampled data. When the continuous time signal representation is known then it can be easy to evaluate the sampled value at any arbitrary time and even if the signal is fractionally delayed from the last integer multiple of the sampling interval. Generally FIR or IIR filters are used for this process to implement fractional delay filters [2].

2.2 Ideal fractional delay:-

Normally the ideal fractional delay is digital form of continuous time delay. The delayed system function is band limited with the ideal low pass filter then the delay slowly shifts in time domain. So according to that impulse response of ideal fractional delay filter is shifted and sampled sinc function. It can be represented as h (n) = sinc (n-D), hear ‘n’ is integer and ‘D’ is delay of integral part floor (D) from this fractional part is derived as d=D-floor(D) [4].
2.3 FIR - Fractional delay filters:-
There are five different approaches to implement dynamic fractional delay FIR filters [3]. They are
1- Windowed sinc function.
2- Weighted least squares method.
3- Oetken’s method.
4- Lagrange interpolation method.
5- Low pass FD with a smooth transition band.

From these methods windowed sinc function technique is selected to implement in this thesis work.

2.4 Fractional Delay using FIR filters:-
According to Nyquist Shannon sampling theorem original signal is reconstructed by multiplying the each sampled data with scaled sinc function. The only condition that the original waveform is band limited to have the maximum frequency should be less to the half of the sampling rate [22].

\[ h(x) = 2f_t \text{sinc} (2\pi f_t x) \]  \hspace{1cm} (1) \\
\[ = 2f_t \frac{\sin(2\pi f_t x)}{2\pi f_t x} \]  \hspace{1cm} (2) \\
\[ = \frac{\sin(2\pi f_t x)}{\pi x} \]  \hspace{1cm} (3)

In this thesis fractional delay filter is implemented by using FIR filter sinc function.
2.5 Geometrical locations of source and interference signals:

In this thesis work uses the three equi spaced microphone arrays and they located in 3D-coordinate system. Hence it considers as $M_1=(X_1, Y_1, Z_1)$, $M_2=(X_2, Y_2, Z_2)$, and $M_3=(X_3, Y_3, Z_3)$. The distance between source, primary interference signal and secondary interference signals were measured with respect to three equi spaced microphone arrays. The distance between each side by side microphone consider as ‘d’. Hence the $M_1 M_2 = d$ and $M_2 M_3 = d$. The distance between each side by side microphone d should be avoiding the aliasing effect. The co-ordinate axis points of Source, primary interference signal and secondary interference signals are considers as $S=(X_s, Y_s, Z_s)$, $I_1=(X_{i11}, Y_{i11}, Z_{i11})$, and $I_2=(X_{i12}, Y_{i12}, Z_{i12})$. Here the calculation of the distance between microphones to source signal, primary interference signal and secondary interference signals is calculated by using the distance between two points formulae.

$I_1 = (X_{i11}, Y_{i12}, Z_{i13})$  \hspace{1cm} $S = (X_s, Y_s, Z_s)$ \hspace{1cm} $I_2 = (X_{i12}, Y_{i12}, Z_{i12})$

Primary interference \hspace{1cm} Source signal \hspace{1cm} Secondary Interference

Mic1 \hspace{1cm} Mic2 \hspace{1cm} Mic3

![Figure 1: Microphone array to source and interference signals distance](image-url)
2.5.1 Calculation of distance:

Distance between successive microphones

\[ M_{1M2} = \sqrt{(X_2 - X_1)^2 + (Y_2 - Y_1)^2 + (Z_2 - Z_1)^2} \] \hspace{1cm} (4)

\[ M_{2M3} = \sqrt{(X_3 - X_2)^2 + (Y_3 - Y_2)^2 + (Z_3 - Z_2)^2} \] \hspace{1cm} (5)

Distance between the source to microphones

\[ S_{M1} = \sqrt{(X_s - X_1)^2 + (Y_s - Y_1)^2 + (Z_s - Z_1)^2} \] \hspace{1cm} (6)

\[ S_{M2} = \sqrt{(X_s - X_2)^2 + (Y_s - Y_2)^2 + (Z_s - Z_2)^2} \] \hspace{1cm} (7)

\[ S_{M3} = \sqrt{(X_s - X_3)^2 + (Y_s - Y_3)^2 + (Z_s - Z_3)^2} \] \hspace{1cm} (8)

Distance between primary interference signal to microphones.

\[ I_{1M1} = \sqrt{(X_{11} - X_1)^2 + (Y_{11} - Y_1)^2 + (Z_{11} - Z_1)^2} \] \hspace{1cm} (9)

\[ I_{1M2} = \sqrt{(X_{11} - X_2)^2 + (Y_{11} - Y_2)^2 + (Z_{11} - Z_2)^2} \] \hspace{1cm} (10)

\[ I_{1M3} = \sqrt{(X_{11} - X_3)^2 + (Y_{11} - Y_3)^2 + (Z_{11} - Z_3)^2} \] \hspace{1cm} (11)
Distance between secondary interference signal to microphones.

\[ I_{2M1} = \sqrt{(X_{i2} - X_1)^2 + (Y_{i2} - Y_1)^2 + (Z_{i2} - Z_1)^2} \]  \hspace{1cm} (12)

\[ I_{2M2} = \sqrt{(X_{i2} - X_2)^2 + (Y_{i2} - Y_2)^2 + (Z_{i2} - Z_2)^2} \]  \hspace{1cm} (13)

\[ I_{2M3} = \sqrt{(X_{i2} - X_3)^2 + (Y_{i2} - Y_3)^2 + (Z_{i2} - Z_3)^2} \]  \hspace{1cm} (14)

### 2.6 Fractional delay calculation:

In microphone array delaying data in fraction of samples is essential. Hear calculating the fractional delay is depends on the distance between source to microphones and speed of the sound. In figure 2: The signals reached at M2 and M3 is delayed with respect to the microphone M1. In order to steer the signals in a plane wave’s direction the signals received at M1 and M2 is needs to be delayed.

![Fractional delay measurement](image)

According to the signal that which the fractional delay function needs to be calculated, first take those signal generation positional point and calculate the distance from the microphone array to the signal by using formulae mentioned in equations (6-14). The distance between two points is given as SM1, SM2, SM3, I1M1, I1M2, I1M3, I2M1, and I2M3. These are the distance from the microphone array to source, primary interference signal and secondary interference signals.

\[ \text{Delay} = \frac{\text{distance}}{v} \]  \hspace{1cm} (15)

\[ v = \text{velocity of sound} \]  \hspace{1cm} (16)
CHAPTER-3
3 Microphone Array

3.1 Introduction

In speech processing desired speech signal is contaminated by interference signals. The spatial location of desired source signal is separated from the interference signals. In order to exploit the exact desired source signal location microphone arrays are essential. For beam forming applications microphone array are further classified into two types’ Narrow band and Broad band processing. Narrow band is the process in which the signal incident directly on microphone array and it is called narrowband assumption. Broad band is the process in which the signal sources are far away from microphone array and its wave fronts are touching the microphone and it is called far field assumption. Far field assumption is applicable for the most of the microphone array applications [21].

In the process of understanding the reasoning behind problems with narrow band array for broad band signals, let us consider a linear array with fixed number of arrays with fixed some of length. The important term in the measurement is its size with respect to its operating wavelength. In high frequency signal condition the array looks large and the beam will be narrow in spite of that for low frequency signals array looks small and beam will be widen.

Microphone array working is completely different than antenna array working in the field of radar and sonar technology. The reasons are [20].

1.) Speech signal is very long.
2.) Very high room reverberation.
3.) Very high non stationary signals.
4.) Both speech and noise used to have same spectral characteristics.
5.) Very restricted number of sensors.
6.) The hearing capacity of human is approximately 120 dB. It is sensitive with weak tails of channel impulse response for this filtering length should be long [20].

Generally most of the algorithms used in microphone array are borrowed from the narrow band array processing. The biggest advantage of this is most of the algorithms are used in antenna array from several years. So it can be used with less effort. The disadvantages of these algorithms are failed to execute in certain room acoustic conditions and its result performance is very limited. To overcome these problems microphone array should need broad band processing.

Microphone array broad band process has certain potential characteristics to solve conditions compared to narrow band array processing. Those are [20].

1.) Noise reduction.
2.) Echo cancellation.
3.) De reverberation.
4.) Localization of the single source.
5.) Estimation of the number of sources.
6.) Localization of the multiple sources.
7.) Source separation.
8.) Cocktail party.

3.2 Limitations of Linear equi spaced microphone array:-

The linear equi spaced array suffered with three significant problems. First problem is its equi spaced arrange only useful for narrowband range of frequencies. The condition used for this is spatial aliasing. The analog nyquist rate applied to beam forming, which is stated as

\[ d < \frac{\lambda_{\text{min}}}{2} \]  \hspace{1cm} (17)
The second limitation occurs with linear equi spaced array is the incident wave is steered with only single parameter $\theta$; it is called angle of incidence with respect to array axis. In this type of steering sound sources of array is collinear with array steering so it cannot be resolved. In the case of rotational symmetry of array sound waves from different heights cannot be resolved for linear array.

The third limitation is in some conditions linear equi spaced array cannot be feasible to construct. The factors influencing on microphone array construction are lightening ventilation systems, metal ceiling and tile grid.

### 3.3 Microphone array speech processing:-

Microphone array speech processing is more efficient technique due to its high SNR gain compared to single channel speech enhancement. So many speech enhancement techniques prefer microphone array structure. In microphone array enhancement mainly two methods are used those are spatiotemporal filtering and beam forming [25].

### 3.4 Beamforming:-

Beamforming is the process that which is used to exploit the both spatial and temporal distribution of source and interference signals. According to the specified enhancement requirements beamformers are classified in to two types. They are fixed beam forming technique and adaptive beam forming technique.

Fixed beam formers are depends on the model based assumptions of source and noise signals. When source and noise signals are determined based on that these optimal beamforms are designed. These beamformers are exists when there is no mismatch between source and noise signals [25].
The role of adaptive beam formers are designed for even when there is mismatch between source and noise signals these beamformers can work effectively. Adaptive beam formers are constructed depends on spatial and statistical information of source and noise signals continuously received to microphones. Adaptive beam formers are very complex to design and implement compared to optimal beamformers.

In this thesis work implemented the optimal wiener beamformer technique for the designing of Generalized side lobe canceller using LMS algorithm and N-LMS algorithms.

3.5 Wiener Beamformer :-

Optimal beamformers are designed on the basis of wiener filtering concept. Optimal beamformers are designed to enhance the desired part of the signal from the noisy, interference corrupted signals. In real time conditions it is impossible to implement the optimal beamformer. The reason is the correlation matrix ‘R’ is unknown. To resolve this problem Direct Matrix Inversion (DMI) technique was introduced [12].

3.5.1 Direct Matrix Inversion Technique:-

Here in Direct Matrix Inversion (DMI) technique, estimation of correlation matrix $R_{xx}$ is depends on the average of multiple time samples of the received data. As long as the interference does not shows the any changes significantly during the length of averaging process then it can be represented as maximum likelihood correlation matrix [24].

In this thesis the correlation matrix $R_{ss}$ and $R_{nn}$ are calculated offline for the source and interference signals separately. The correlation matrix $R_{ss}$ is calculated when interference signal is absent and the correlation matrix $R_{nn}$ is calculated when the source signal is absent. So the mathematical formulation of the correlation matrix $R_{ss}$ and $R_{nn}$ is calculated offline separately is represented in equations (19) and (20) respectively.
The correlation matrix ‘\( R_{xx} \)’ is estimated by

\[
R_{xx} = \frac{1}{K} \sum_{K=1}^{K} X_K X_K^H \tag{18}
\]

Here \( x(n) = s(n) + v(n) \)

Hence \( S(n) \) is source signal and \( v(n) \) is noise signal

Similarly

\[
R_{ss} = \frac{1}{K} \sum_{K=1}^{K} S_K S_K^H \tag{19}
\]

\[
R_{nn} = \frac{1}{K} \sum_{K=1}^{K} n_K n_K^H \tag{20}
\]

The cross correlation matrix \( R_{dx} \) is estimated by

\[
R_{dx} = \frac{1}{K} \sum_{K=1}^{K} d_k^* X_k \tag{21}
\]

Where \( K \) is represented as known transmitted symbols or training sequence. This is the procedure for training based beam forming.

Then

\[
W_{opt} = R_{xx}^{-1} R_{dx} \tag{22}
\]

\[
W_{opt} = (R_{ss} + R_{nn})^{-1} R_{dx} \tag{23}
\]

The output signal of the wiener beamformer is obtained by convolution of the \( W_{opt} \) and the input signal \( X(k) \).

\[
Y_{out} = W_{opt}^T * X(k) \tag{24}
\]
CHAPTER-4
4 Generalized sidelobe canceller

4.1 Generalized sidelobe canceller (GSC):-

In this thesis work generalized side lobe canceller structure was implemented by using the wiener beam forming concept. Here the spatial environment is considers as three lobes according to the generation of source signal, Primary interference signal and Secondary interference signals. The environment surrounded by the source signal is considers as main lobe. It contains the major part of desired speech signal and small amount of interference signals. The environments surrounded by primary and secondary interference signals consider as side lobes and it contains the minimum amount of source signal.

In the designing of generalized side lobe canceller, wiener beam forming implementation is essential. The construction of wiener beam forming is completely depends on predefined source and interference signals. The optimal coefficient $W_{opt}$ is measured by implementing the direct matrix inversion techque on source and interference signals [14]. Here the implementation process of wiener beam forming out put signal is depends on desired ‘$r_{dx}$’ matrix [13]. If the ‘$r_{dx}$’ matrix is choses from the middle column of the source signal then the output of the beamformer contains the maximum amount source signal and if the $r_{dx}$ matrix is chosen from the middle column of the noise signal correlation matrix then the out put signal contains the major amount of noise signal.

The main lobe signal (desired) is trapped and processed by the wiener beamforming-1 block (WBF-1). In this the desired source signal is consider as ‘$r_{dx}$’ matrix. Side lobes signals was trapped and processed by the wiener beamforming-2(WBF-2) and wiener beamforming-3 (WBF-3) blocks. The outputs of WBF-2 and WBF-3 consist of interference signals. By using adaptive LMS and N-LMS algorithms filters the small amount of noise presented in the output of WBF-1 and gives the adaptively filtered desired source signal.
4.2 Least mean square algorithm (LMS):

Least mean square algorithm was proposed by Windrow and Hoff in 1959. It is an adaptive algorithm. LMS algorithm uses the gradient based method of steepest decent and also it uses to estimate of gradient vector on available data [15].

*Figure 3: Block diagram of generalized sidelobe canceller*
The signals received from the wiener beam forming one is considered as desired speech signal and the signals received from the wiener beamforming-2 and wiener beamforming-3 is considered as noise signals to the LMS algorithm. LMS algorithm is adaptively filters the noise present in the speech signal received from the wiener beamformer-1.

4.2.1 Mathematical formulation of LMS algorithm:

The weight vector equation is obtained by estimate of the gradient vector with the method steepest descent algorithm is given by [15].

\[ W(n+1) = W(n) + \mu \left[ -\nabla (E[e^2(n)]) \right] \]  \hspace{1cm} (25)

Where \( \mu \) is the step size parameter, it is used to control the convergence phenomenon. Here \( e^2(n) \) is the measure of error obtained with the difference between beamformer output \( y(n) \) and reference signal \( x(n) \).

This is shown as

\[ e^2(n) = \left[ d^*(n) - W^h X(n) \right]^2 \]  \hspace{1cm} (26)
The gradient vector for the above weight update equation is calculated as.

$$\nabla_w (E[e^2(n)]) = -2r + 2RW(n) \quad (27)$$

Practically in adaptivization process calculation of r and R is not possible.

So the weight vector update equation is given as.

$$W(n+1) = W(n) + \mu x(n)[d^*(n) - X(n)W(n)] \quad (28)$$

$$W(n+1) = W(n) + \mu X(n)e^*(n) \quad (29)$$

The initial arbitrary value is $W(0)$ at $n = 0$

The repeated corrections occurred with weight vector gradually tends to minimum value of the mean square error.

The summarized points of the LMS algorithm are.

$$\text{Output} \quad Y(n) = W^hX(n) \quad (30)$$

$$\text{Error} \quad e(n) = d^*(n) - Y(n) \quad (31)$$

$$\text{Weight} \quad W(n+1) = W(n) + \mu X(n)e^*(n) \quad (32)$$

### 4.2.2 Convergence factor and stability of LMS algorithm:

The convergence and stable equation of the LMS algorithm with some initiated arbitrary value. The equation is given as.

$$0 < \mu < \frac{1}{\lambda_{\text{max}}} \quad (33)$$

Where $\lambda_{\text{max}}$ is sum of the diagonal elements of the correlation matrix R and it also called as maximum eigenv value of the correlation matrix R. If the given eigenv values of R is wide spread convergence is slow.so the Eigen value of R is inversely proportional to the convergence. The largest Eigen value $\lambda_{\text{max}}$ to the smallest eigenv value $\lambda_{\text{min}}$ ratio is defined as the eigenv value spread.so
the convergence is directly proportional to the $\mu$. If the $\mu$ is taken very small then the convergence is less and if the $\mu$ is large convergence is fast and may be stability is less around the minimum value.

4.3 Normalized least mean square algorithm (N-LMS):

There are some difficulties in the implementation of the step size $\mu$ in LMS algorithm. In stationary environments LMS algorithm converges to $0 < \mu < \frac{2}{(P+1)E[|x(n)|^2]}$ (34)

Hear $E[|x(n)|^2]$ is power of the signal $x(n)$

The power is estimated by using the time average equation [19].

$$E[|x(n)|^2] = \frac{1}{P+1} \sum_{k=0}^{P} |x(n-k)|^2$$ (35)

Bounds of mean square convergence is

$$0 < \mu < \frac{2}{x^H(n)x(n)}$$ (36)

Using this bound in LMS algorithm it is represented as

$$\mu(n) = \frac{\beta}{x^H(n)x(n)} = \frac{\beta}{\|x(n)\|^2}$$ (37)

Where $\beta$ is normalized step size which varies $0 < \beta < 2$, replacing the $\mu$ in LMS update equation with $\mu(n)$ is represented as Normalized Least Mean Square algorithm [18].

The Normalized LMS algorithm is represented as

$$W_{n+1} = W_n + \beta \frac{x(n)}{\|x(n)\|^2} e^*(n)$$ (38)

Where $\|x(n)\|^2$ does not show any effect on the direction it only changes the magnitude.

Mean square convergence of N-LMS algorithm is $0 < \beta < 2$ (39)
CHAPTER - 5

5 Test setup

5.1 Microphone array positioning:-
In this thesis work three microphones of equi spaced distance is linearly positioned in 3D- coordinate system. In the array of microphones the successive distance between two microphones is separated by the distance ‘d’. The source signal is sampled to 16 KHZ and hence the distance ‘d’ must be less than 4.5 cms in order to avoid the aliasing effect discussed in the equation (17). The microphone array measurements are fixed and represent as following sequence. Hear the measurements are calculated in meters.

\[
\text{M1} = (2.46, 2.50, 1.24); \quad \text{M2} = (2.50, 2.50, 1.24); \quad \text{M3} = (2.54, 2.50, 1.24);
\]

From the above 3D- coordinate Microphone array measurements used to find out the distance between the successive microphones as discussed in the equations (4-5), hence ‘d ’ is 4 cms.

5.2 Fractional delay measurements:-
In this thesis work fractional delay concept is performed by using FIR sinc function windowing filters as mentioned in the equation (3).

<table>
<thead>
<tr>
<th>Source signal position (X,Y,Z)</th>
<th>Delay received at microphone-1 (D1)</th>
<th>Delay received at microphone-2 (D2)</th>
<th>Delay received at microphone-3 (D3)</th>
</tr>
</thead>
<tbody>
<tr>
<td>(2.50, 2.60,1.34)</td>
<td>6.8497</td>
<td>6.5911</td>
<td>6.3325</td>
</tr>
</tbody>
</table>

*Table 1: Fractional delay measurement*
Figure 5: (a) Original speech signal (b) speech signal received at microphone-1 after FD (c) speech signal received at microphone-2 after FD. (d) Speech signal received at microphone-3 after FD.
CHAPTER-6

6 Evaluation

The evaluation of noise suppression with wiener beam former, Generalized side lobe canceller using LMS algorithm and N-LMS algorithms under different positions of source and interference signals were implemented in matlab. In this thesis work used a clear speech signal with a combination of male and female voices is having the length of 182824 samples. The same length of another speech signal is taken as primary interference signal. The secondary interference signal is a random noise signal, which is generated in matlab with the same length as source signal. In order to suppress the noise under different positions of source and interference signals, here experiment is conducted and implemented for the positional values of the source and interference signals in 3D-co-ordinate system.

In the first position the distance between the microphone array to the source signal and interference signals are kept close. So the co-ordinate axis is measured as.

<table>
<thead>
<tr>
<th>Distance between Ref microphone to source and interference signals</th>
<th>Source signal position (2.90, 2.90, 1.74)</th>
<th>Primary Interference signal position (3.00, 3.00, 2.00)</th>
<th>Secondary interference signal (3.20, 3.20, 1.50)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Reference microphone-2 Position Mic-2 (2.50, 2.50, 1.24)</td>
<td>75 cms</td>
<td>103 cms</td>
<td>102 cms</td>
</tr>
</tbody>
</table>

Table 2: Measurement of distance from microphone array to source and interference signals in position-1

In second position the distance between the microphone arrays to the source signal is kept close and interference signals are kept long distance from the Microphone array. Hence the co-ordinate axis is measured as.
Distance between Ref microphone to source and interference signals | Source signal position (2.90, 2.90, 1.74) | Primary Interference signal position (4.00, 4.00, 2.24) | Secondary interference signal (4.20, 4.10, 2.00) |
Reference microphone-2 Position Mic-2 (2.50, 2.50, 1.24) | 75 cms | 234 cms | 245 cms |

*Table 3: Measurement of distance from microphone array to source and interference signals in position-2*

In third position the distance between interference signals to microphone array keep close and source signal is away from microphone array. Hence the co-ordinate axis measurements are taken as.

Distance between Ref microphone to source and interference signals | Source signal position (4.20, 4.10, 2.00) | Primary Interference signal position (2.90, 2.90, 1.74) | Secondary interference signal (3.20, 3.20, 1.50) |
Reference microphone-2 Position Mic-2 (2.50, 2.50, 1.24) | 245 cms | 75 cms | 102 cms |

*Table 4: Measurement of distance from microphone array to source and interference signals in position-3*

In fourth position the distance between microphone array to source and primary interference speech signals are keep close and secondary interference signal is away from the microphone array so the co-ordinate axis is measured as.

Distance between Ref microphone to source and interference signals | Source signal position (2.90, 2.90, 1.74) | Primary Interference signal position (3.20, 3.20, 1.50) | Secondary interference signal (4.00, 4.00, 2.24) |
Reference microphone-2 Position Mic-2 (2.50, 2.50, 1.24) | 75 cms | 102 cms | 234 cms |

*Table 5: Measurement of distance from microphone array to source and interference signals in position-4*
The output speech signals of wiener beamformer, GSC with LMS and GSC with N-LMS were analyzed by using the signal to noise ratio (SNR), signal to noise ratio improvement (SNRI) and PESQ.

6.1 Signal to noise ratio:-

Signal to noise ratio is the ratio of signal power to the noise power. SNR is measured between the desired speech signal to the interference signals. The formula of the SNR measurement is the ratio of variance of speech signal to the variance of noise signal, the measurement scale of SNR is in dB.

To measure the SNR values of output speech signal for the given input SNR the derived formulae is given as

\[ \alpha = \frac{1}{\sqrt{10^{\frac{SNR_{input}-SNR}{10}}}} \]  

(40)

Here the \( SNR_{input} \) is according to the 0,5,10,15,20, 25 dB.

The interference signals received at each microphone is multiplied with the ‘\( \alpha \)’ value.

Here the reference microphone is M2.

\[ M2 (n) = s(n) + \alpha \left[ I1 (n) + I2 (n) \right] \]  

(41)

The SNR measurement for the microphone M2 is given as.

\[ M2 = 10 \log_{10} \left( \frac{\text{variance}(s2(n))}{\text{variance}(I1(n)+I2(n))} \right) \]  

(42)

The signal to noise ratio improvement is measured from the difference between outputs SNR to the input SNR.

\[ SNRI = 10 \log_{10} \left( \frac{\sigma_{\text{speech}_{\text{output}}}^2}{\sigma_{\text{noise}_{\text{output}}}^2} \right) - 10 \log_{10} \left( \frac{\sigma_{\text{speech}_{\text{input}}}^2}{\sigma_{\text{noise}_{\text{input}}}^2} \right) \]  

(43)
6.2 Perceptual evaluation of speech quality (PESQ):

Perceptual evaluation of speech quality gives the perfect and exact evaluation of speech degradation occurring in the system. It compares the degraded speech signal time alignment with the original speech signal time alignment. Then it identifies the bad intervals occurring in it and gives the degraded speech quality score. The PESQ values range is varies from 0 to 5. If the tested speech signal score is 4.5 then it is clear speech signal. If the speech signal score is 3 to 4 then it is almost near to clean speech and it is called good quality. If the speech signals score is less than 2 then it is fully degraded signal.

Perceptual evaluation of speech quality improvement is measured by the difference between output speech signal PESQ score to the input signal at mic-2 PESQ score.

6.3 Simulation result of Wiener beamformer:

The simulated results of wiener beam former at different positions of source and interference signals were calculated by using the measurements of SNR, SNRI and PESQ.

In Position-1 the source signal and interference signals are close to microphone array the noise suppression of the wiener beam former average of SNR Improvement is 16.1340dB and the average PESQ Improvement score is 1.816.

In position-2 the source signal is near to microphone array and interference signals are long distance from the microphone array. The average SNR of Improvement is 19.6348 dB and the average PESQ Improvement is 2.067.

In position-3 the interference signals are close to the microphone array and source signal is keep long distance from the microphone array. The average of SNR Improvement is 27.9869dB and the average PESQ Improvement is 1.459.

In position-4 the primary speech interference signal and source signal is close to microphone array and secondary interference signal is long distance from the microphone array. The average SNR Improvement is 17.0501dB and the average PESQ Improvement is 1.593.
From the observation of the simulation results of the wiener beamformer the conclusion is when interference signals is near to microphone array the average of SNR Improvement is high 28.1423dB and the average PESQ Improvement is 1.459.

<table>
<thead>
<tr>
<th>Microphone positions</th>
<th>Input SNR</th>
<th>Output SNR</th>
<th>SNRI</th>
<th>PESQ Improvement</th>
</tr>
</thead>
<tbody>
<tr>
<td>Position-1</td>
<td>0</td>
<td>17.0255</td>
<td>17.0255</td>
<td>1.795</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>21.3075</td>
<td>16.3075</td>
<td>1.990</td>
</tr>
<tr>
<td></td>
<td>10</td>
<td>26.0267</td>
<td>16.0267</td>
<td>1.976</td>
</tr>
<tr>
<td></td>
<td>15</td>
<td>30.9003</td>
<td>15.9003</td>
<td>1.892</td>
</tr>
<tr>
<td></td>
<td>20</td>
<td>35.8072</td>
<td>15.8072</td>
<td>1.751</td>
</tr>
<tr>
<td></td>
<td>25</td>
<td>40.5827</td>
<td>15.5827</td>
<td>1.493</td>
</tr>
<tr>
<td>Position-2</td>
<td>0</td>
<td>19.9517</td>
<td>19.9517</td>
<td>1.913</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>24.7003</td>
<td>19.7003</td>
<td>2.209</td>
</tr>
<tr>
<td></td>
<td>10</td>
<td>29.6014</td>
<td>19.6014</td>
<td>2.258</td>
</tr>
<tr>
<td></td>
<td>15</td>
<td>34.5571</td>
<td>19.5571</td>
<td>2.193</td>
</tr>
<tr>
<td></td>
<td>20</td>
<td>39.5246</td>
<td>19.5246</td>
<td>2.091</td>
</tr>
<tr>
<td></td>
<td>25</td>
<td>44.4550</td>
<td>19.4550</td>
<td>1.739</td>
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<tr>
<td>Position-3</td>
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<td>28.1453</td>
<td>28.1453</td>
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</tr>
<tr>
<td></td>
<td>5</td>
<td>33.1019</td>
<td>28.1019</td>
<td>1.654</td>
</tr>
<tr>
<td></td>
<td>10</td>
<td>38.0766</td>
<td>28.0766</td>
<td>1.705</td>
</tr>
<tr>
<td></td>
<td>15</td>
<td>43.0351</td>
<td>28.0351</td>
<td>1.553</td>
</tr>
<tr>
<td></td>
<td>20</td>
<td>47.9121</td>
<td>27.9121</td>
<td>1.196</td>
</tr>
<tr>
<td></td>
<td>25</td>
<td>52.6333</td>
<td>27.6333</td>
<td>0.886</td>
</tr>
<tr>
<td>Position-4</td>
<td>0</td>
<td>18.1938</td>
<td>18.1938</td>
<td>1.810</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>22.2605</td>
<td>17.2605</td>
<td>1.813</td>
</tr>
<tr>
<td></td>
<td>10</td>
<td>26.8637</td>
<td>16.8637</td>
<td>1.705</td>
</tr>
<tr>
<td></td>
<td>15</td>
<td>31.6988</td>
<td>16.6988</td>
<td>1.534</td>
</tr>
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<td></td>
<td>20</td>
<td>36.6110</td>
<td>16.6110</td>
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</tr>
<tr>
<td></td>
<td>25</td>
<td>41.4262</td>
<td>16.4262</td>
<td>1.235</td>
</tr>
</tbody>
</table>

*Table 6: Simulation result of wiener beamformer.*
Table 7: Average SNR Improvement for wiener beamformer at each position.

<table>
<thead>
<tr>
<th>positions</th>
<th>Average SNR Improvement</th>
<th>Average PESQ Improvement</th>
</tr>
</thead>
<tbody>
<tr>
<td>Position-1</td>
<td>16.1340</td>
<td>1.816</td>
</tr>
<tr>
<td>Position-2</td>
<td>19.6348</td>
<td>2.067</td>
</tr>
<tr>
<td>Position-3</td>
<td>27.9869</td>
<td>1.459</td>
</tr>
<tr>
<td>Position-4</td>
<td>17.0501</td>
<td>1.593</td>
</tr>
</tbody>
</table>

Figure 6: Bar plot between input and output SNR for wiener beamformer at 4 different positions.
Figure 7: Bar plot between input SNR and output SNR Improvement for wiener beamformer at 4 different positions.

Figure 8: Bar plot between input SNR and PESQ Improvement at 4 different positions for wiener beamformer.
6.4 Simulation result of Generalized sidelobe canceller (GSC) using LMS algorithm:

The simulated results of generalized side lobe canceller using LMS algorithm at different positions of source and interference signals were calculated by using the measurements SNR, SNRI and PESQ.

In Position-1 the source signal and interference signals are close to microphone array. The noise suppression of the average of SNRI is 22.1542 dB and the average PESQ Improvement is 1.815.

In position-2 the source signal is near to microphone array and interference signals are long distance from the microphone array. The average of SNR Improvement is 25.6893 dB and the average PESQ Improvement is 2.060.

In position-3 the interference signals are close to microphone array and source signal is keep long-distance. The average of SNR Improvement is 34.0074 dB and the average PESQ Improvement is 1.462.

In position-4 the primary interference signal and source signal is close to microphone array and secondary interference signal is long distance from the microphone array. The average of SNR Improvement is 23.1446 dB and the average PESQ Improvement is 1.592.

Generalized side lobe canceller using LMS algorithm achieves the maximum signal to noise ratio when the interference signals is near to microphone array. The average of SNR Improvement is 34.0074 and its average PESQ Improvement score is 1.462.
Step size $\mu = 1.00 \times 10^{-4}$

Order=10

<table>
<thead>
<tr>
<th>Microphone positions</th>
<th>Input SNR</th>
<th>Output SNR</th>
<th>SNRI</th>
<th>PESQ Improvement</th>
</tr>
</thead>
<tbody>
<tr>
<td>Position-1</td>
<td>0</td>
<td>23.0461</td>
<td>23.0461</td>
<td>1.795</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>27.3281</td>
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<td>10</td>
<td>32.0473</td>
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<td>15</td>
<td>36.9209</td>
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</tr>
<tr>
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<td>20</td>
<td>41.8278</td>
<td>21.8278</td>
<td>1.750</td>
</tr>
<tr>
<td></td>
<td>25</td>
<td>46.6033</td>
<td>21.6033</td>
<td>1.491</td>
</tr>
<tr>
<td>Position-2</td>
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<td>26.0184</td>
<td>26.0184</td>
<td>1.915</td>
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<td>5</td>
<td>30.7717</td>
<td>25.7717</td>
<td>2.210</td>
</tr>
<tr>
<td></td>
<td>10</td>
<td>35.6549</td>
<td>25.6549</td>
<td>2.213</td>
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<tr>
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<td>15</td>
<td>40.6252</td>
<td>25.6252</td>
<td>2.196</td>
</tr>
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<td></td>
<td>20</td>
<td>45.5743</td>
<td>25.5743</td>
<td>2.092</td>
</tr>
<tr>
<td></td>
<td>25</td>
<td>50.4758</td>
<td>25.4758</td>
<td>1.739</td>
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<tr>
<td>Position-3</td>
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<td>34.1659</td>
<td>34.1659</td>
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<td>34.1225</td>
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<td>44.0972</td>
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<td>15</td>
<td>49.0557</td>
<td>34.0557</td>
<td>1.554</td>
</tr>
<tr>
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<td>20</td>
<td>53.9327</td>
<td>33.9327</td>
<td>1.197</td>
</tr>
<tr>
<td></td>
<td>25</td>
<td>58.6539</td>
<td>33.6539</td>
<td>0.898</td>
</tr>
<tr>
<td>Position-4</td>
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<td>24.3578</td>
<td>24.3578</td>
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<td></td>
<td>25</td>
<td>47.4787</td>
<td>22.4787</td>
<td>1.234</td>
</tr>
</tbody>
</table>

*Table 8: Simulation result of generalized side lobe canceller using LMS algorithm.*
Table 9: Average of SNR Improvement for GSC-L at each position.

<table>
<thead>
<tr>
<th>Positions</th>
<th>Average SNRI</th>
<th>Average PESQ Improvement</th>
</tr>
</thead>
<tbody>
<tr>
<td>Position-1</td>
<td>22.1542</td>
<td>1.815</td>
</tr>
<tr>
<td>Position-2</td>
<td>25.6893</td>
<td>2.060</td>
</tr>
<tr>
<td>Position-3</td>
<td>34.0074</td>
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</tr>
<tr>
<td>Position-4</td>
<td>23.1446</td>
<td>1.592</td>
</tr>
</tbody>
</table>

Figure 9: Bar plot between input and output SNR at 4 different positions for GSC-L.
Figure 10: Bar plot between input SNR and output SNR improvement at 4 different positions for GSC-L.

Figure 11: Bar plot between input SNR and PESQ improvement at 4 different positions for GSC-L.
6.5 Simulation result of Generalized side lobe canceler (GSC) using N-LMS algorithm:

The simulated results of generalized side lobe canceler using N-LMS at different positions of source and interference signals were calculated by using the measurements of SNR, SNRI and PESQ.

In Position-1 the source signal and interference signals are close to microphone array the noise suppression of average of SNR Improvement is 25.1921 and the average PESQ Improvement score is 1.817.

In position-2 the source signal is near to microphone array and interference signals are long distance from the microphone array. The average of SNR Improvement is 27.6526 dB and the average PESQ Improvement score is 2.083.

In position-3 the interference signals are close to microphone array and source signal is keeps long-distance from the microphone array. The average of SNR Improvement is 37.4858 and the average PESQ Improvement score is 1.458.

In position-4 the primary interference signal and source signal is close to microphone array and secondary interference signal is long distance from the microphone array. The average of SNR Improvement is 25.9917 and the average PESQ Improvement score is 1.577.

From the observation of the simulation results of the GSC with N-LMS conclude that when interference signals is near to microphone array the average of SNR Improvement is high 37.4858 and with the average PESQ Improvement score is 1.458.
Normalized step size $\beta = 1.00 \times 10^{-6}$

order = 10

<table>
<thead>
<tr>
<th>Microphone positions</th>
<th>Input SNR</th>
<th>Output SNR</th>
<th>SNRI</th>
<th>PESQ</th>
</tr>
</thead>
<tbody>
<tr>
<td>Position-1</td>
<td>0</td>
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<td>25.8177</td>
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<td>2.074</td>
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<td>51.6193</td>
<td>26.6193</td>
<td>1.719</td>
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<td>Position-3</td>
<td>0</td>
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<td>38.1293</td>
<td>1.754</td>
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<td></td>
<td>5</td>
<td>42.5742</td>
<td>37.5742</td>
<td>1.698</td>
</tr>
<tr>
<td></td>
<td>10</td>
<td>47.2609</td>
<td>37.2609</td>
<td>1.650</td>
</tr>
<tr>
<td></td>
<td>15</td>
<td>51.7355</td>
<td>36.7355</td>
<td>1.554</td>
</tr>
<tr>
<td></td>
<td>20</td>
<td>56.3827</td>
<td>36.3827</td>
<td>1.196</td>
</tr>
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<td></td>
<td>25</td>
<td>60.7842</td>
<td>35.7842</td>
<td>0.897</td>
</tr>
<tr>
<td>Position-4</td>
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<td>27.4835</td>
<td>27.4835</td>
<td>1.792</td>
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<td>31.3046</td>
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<td>1.709</td>
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<td>25.6470</td>
<td>1.516</td>
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<td>20</td>
<td>45.3758</td>
<td>25.3758</td>
<td>1.457</td>
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<td>25</td>
<td>49.8039</td>
<td>24.8039</td>
<td>1.211</td>
</tr>
</tbody>
</table>

*Table 10: Simulation result of generalized side lobe canceller using N-LMS algorithm.*
Table 11: Average of SNR Improvement for GSC-N at each position.

<table>
<thead>
<tr>
<th>positions</th>
<th>Average SNRI at each position</th>
<th>Average PESQ Improvement</th>
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</thead>
<tbody>
<tr>
<td>Position-1</td>
<td>25.1921</td>
<td>1.817</td>
</tr>
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<td>Position-2</td>
<td>27.6526</td>
<td>2.083</td>
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<tr>
<td>Position-3</td>
<td>37.4858</td>
<td>1.458</td>
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<td>Position-4</td>
<td>25.9917</td>
<td>1.577</td>
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</table>

Figure 12: Bar plot between input and output SNR at 4 different positions for GSC-N.
Figure 13: Bar plot between input SNR and SNR Improvement at 4 different positions for GSC-N.

Figure 14: Bar plot between input SNR and PESQ Improvement at 4 different positions for GSC-N.
6.6 Comparing the result of wiener beamforming, GSC - L and GSC- N:-

Comparison of the simulation result of the wiener beamforming, GSC- L and GSC- N at position-3, when interference signals are near to the microphone array and source signal is far away from microphone array is given as.

The average of signal to noise ratio improvement achieved by the GSC- L at position-3 is 6.0206 dB higher than the wiener beam forming. In PESQ Improvement after the implementation of GSC –L there is no considerable changes.

In the GSC-N algorithm the average of signal to noise ratio improvement achieved at position-3 is 9.4858 dB higher than the wiener beamformer and PESQ Improvement is slightly decreases while compared to wiener beamformer.

Comparing the both simulated values of GSC-L and GSC-N, the average of SNR Improvement of GSC-N at position-3 is 3.48 dB higher than the GSC-L. PESQ Improvement is slightly decreases compared to GSC-L.
<table>
<thead>
<tr>
<th></th>
<th>Input SNR</th>
<th>Output SNR</th>
<th>SNR Improvement</th>
<th>PESQ Improvement</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Wiener beamformer</strong></td>
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<td>28.1453</td>
<td>1.760</td>
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<td>33.1019</td>
<td>28.1019</td>
<td>1.654</td>
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<td></td>
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<td>38.0766</td>
<td>28.0766</td>
<td>1.705</td>
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<td>43.0351</td>
<td>28.0351</td>
<td>1.553</td>
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<td>1.196</td>
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<td>52.6333</td>
<td>27.6333</td>
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<td><strong>GSC using LMS</strong></td>
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<td>34.1659</td>
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<td>39.1225</td>
<td>34.1225</td>
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<td>1.712</td>
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<td></td>
<td>15</td>
<td>49.0557</td>
<td>34.0557</td>
<td>1.554</td>
</tr>
<tr>
<td></td>
<td>20</td>
<td>53.9327</td>
<td>33.9327</td>
<td>1.197</td>
</tr>
<tr>
<td></td>
<td>25</td>
<td>58.6539</td>
<td>33.6539</td>
<td>0.898</td>
</tr>
<tr>
<td><strong>GSC using N-LMS</strong></td>
<td>0</td>
<td>38.1293</td>
<td>38.1293</td>
<td>1.754</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>42.5742</td>
<td>37.5742</td>
<td>1.698</td>
</tr>
<tr>
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<td>10</td>
<td>47.2609</td>
<td>37.2609</td>
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</tr>
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<td>15</td>
<td>51.7355</td>
<td>36.7355</td>
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<td>20</td>
<td>56.3827</td>
<td>36.3827</td>
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<tr>
<td></td>
<td>25</td>
<td>60.7842</td>
<td>35.7842</td>
<td>0.897</td>
</tr>
</tbody>
</table>

*Table 12: Comparing the result of wiener beamforming, GSC-L and GSC-N at position-3.*
<table>
<thead>
<tr>
<th>Speech Enhancement algorithms</th>
<th>Average SNRI at position-3</th>
<th>Average PESQ Improvement at position-3</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wiener beamformer</td>
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<td>1.459</td>
</tr>
<tr>
<td>GSC using LMS</td>
<td>34.0074</td>
<td>1.462</td>
</tr>
<tr>
<td>GSC using N-LMS</td>
<td>37.4858</td>
<td>1.458</td>
</tr>
</tbody>
</table>

Table 13: Average of SNR Improvement for wiener beamformer, GSC-L and GSC-N at position-3.

Figure 15: Barplot between input and output SNR for wiener beamformer, GSC-L and GSC-N at position-3.
Figure 16: Bar plot between input SNR and SNR Improvement for wiener beamformer, GSC-L and GSC-N at position-3.

Figure 17: Bar plot between input SNR and PESQ score for wiener beamformer, GSC-L and GSC-N at position-3.
Chapter-7

7 CONCLUSIONS AND FUTURE WORK

7.1 CONCLUSION

This thesis work discusses the working and implementation of the wiener beam forming, generalized side lobe canceller using LMS and N-LMS algorithms for the noise suppression and speech enhancement in linear equi spaced microphone array. The experiments were conducted under different positions of speech and interference signals. The output signals obtained from each algorithm is measured with speech quality metrics like SNR, SNRI and PESQ. The result obtained from each metric is compared and analyzed under which condition the more SNR Improvement is achieved and which algorithm is capable of suppressing the interference signals effectively without losing quality of speech was studied.

The simulated result of different positions of source and interference signals were shows the effect of SNR Improvements achieved by the implemented algorithms. The generalized side lobe canceller using N-LMS algorithm gives the better result compared to the GSC-L and Wiener beam forming techniques.

7.2 FUTURE WORK

In this thesis work implementation of microphone array does not considers in acoustic environments. So there is a further possibility to implement the room impulse response (RIR) function to resolve the effect of reverberation noise occurred in acoustic environments. In this thesis system implementation and simulation is done in offline mode. So there is further possibility to implement the system in real-time environment.
Chapter-8

8. BIBLIOGRAPHY


