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Speech Enhancement System using FIR Filter

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ABSTRACT: Now-a-days communication having wide range importance. The best communication way is to communicate with voice or speech signal. Speech signals are made up of vowel and consonant. But normally when communication is happening some sort of unwanted sound is also present with the signal. These unwanted sound components are nothing but noise signal. These unwanted signal degrades speech signal quality. Also, due to these noisy signal proper communication does not take place. Hence, speech enhancement is done by various way. Speech Enhancement is necessary for many applications in which clean speech signal is important for further processing.The speech enhancement techniques mainly focus on removal of noise from speech signal.

KEYWORDS: Speech Enhancement, Adaptive Algorithm, HMM Model, FIR Filter, Noise cancellation, *MATLAB*

I. INTRODUCTION

Speech is one of most easy and secure communication mediums. Speech signal are made up of vowel and consonant. Due to easy and secure specification, it is used in number of application or we can say that it have wide range of application. But some problem arises while doing the speech communication, and main problem is interference of noise signal**.**

It happened because speech signal and noise lie on same frequency range. The presence of noise may corrupt the signal. The best way to remove noise signal is by filtering. But, it is not easy task to only remove noise. Hence, proper removal of noise without removing any original or harmonic component of speech signal it gone through various steps. Here, we are using the adaptive algorithm. Adaptive means behavioural of system is changed according to time based on information available. And algorithm is nothing but procedure to solve a problem according to sequence of specific action. So, here we take new algorithms in which we process signal throughvarious stage. Input signal may be any signal with or without noise is taken for the process. Then by using autocorrelation and cross correlation i.e., by comparing these signal with itself provide similarities and difference in autocorrelation. These autocorrelation signal is again comparing with cross correlation and MATLAB function. On the basis of these information we quantize the speech signal. Again for better result we use Hidden Markov Model (HMM).The main purpose is to use HMM Model is to separate speech signal and noise signal. The main properties of HMM are to input sequence depends on only one state. Here, is only present state.

II. RELATED WORK

There are many different methods used to remove noise from speech signal. Many of them achieve different parameter. Firstly, spectral subtraction which works on step size of signal is used to reduce background noise. It subtracts the noise signal but with noise signal some sortof information is also removed or corrupted. The two type of subtraction is done over subtraction and nonlinear substation. Normally, in nonlinear subtraction it produces the distorted signal.

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Dual microphone is another method which uses separate microphone to record noise and speech signal. It achieves fast convergence and stability to some extend but complex circuit makes the damage. Next is active noise cancellation. It uses FL algorithm to remove noise. But, with active noise some information is also lost and hence, many useful harmonic component is lost. Many traditional Least Mean Square (LMS) method is also used as noise removal. Some of them are concept of adaptive algorithm but they do not achieve desired result. In many traditional method they concentrate on step size of speech signal rather than adjusting the parameter according to reference signal. Some of them usewinner filter on perceptual base but when we increase signal power mean square error is also increased. CC-WSLMS is used as a combinational adaptive filter on the basis of updated step width. It takes filter of different tap length. In the proposed system algorithm we define each and every parameter. Here, filtering frequency is defined over 300 Hz to 3.4 KHz. This frequency is nothing but sound producing capability of human. The length of filter tap is decided according to type of speech signal which is needed to filter.

III. PROPOSED ALGORITHM

A. *Design Considerations*

Speech signal are continuous in nature but when it is stored in computer or any storing device it becomes discrete in nature. Then, stored signal is used for computational purpose. In computational processing, the wave of speech signals are directly stored in MATLAB Tool. The voice frequency of human is 300 Hz -3.4 KHz. The audio range of human ear is 20 Hz-20 KHz and many signals lie in the same range. Hence, it is hard to identify the signal. When sound is produced by human lies in the audible frequency range many unwanted sound or noise is present causing signal to get distorted. In order to remove such a serious error from speech signal, FIR filter is used.

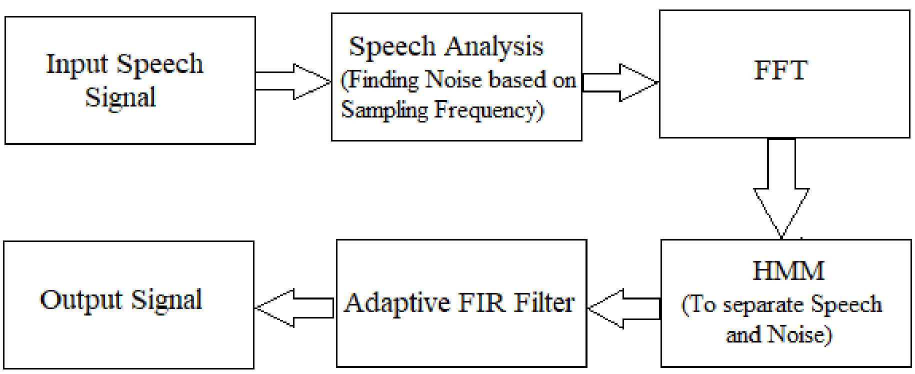

Fig. 1 Process Flow of Speech Enhancement System

B. *Description of the Proposed Algorithm:*

Complete process of speech enhancement goes through six stages. These stages are as follows:

1. Input signal
2. Finding noise(depending on sampling frequency)
3. Apply Fast Fourier Transform (FFT)
4. HMM Model (Separates voice signals and noise)
5. Using adaptive FIR Filter to remove the noise
6. Output signal (parameter value base on SNR ratio)

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Speech Pre-processing and post-processing is carried out using MATLAB library.

Input signal: It is speech signal that we take for processing. It may contain noise or without noise. It may be recorded or chosen in different ways. The signal is converted to wave format.

Finding Noise: There are different type of noise. Some of them are as follows:
1. Babble noise
2. Factory noise
3. Pink noise
4. Volvo noise
5. White noise

Noise can be found out depending on different parameter and sampling frequency. The voice frequency of human is 300 Hz -3.4 KHz. So, it choose on sampling frequency is greater than or equal to twice of modulating frequency.

FFT: Normally in time domain we cannot improve speech signal. So, to improve quality we have to improve frequency component. Hence, signal is converted from time domain to frequency domain in order to remove noise from frame of frequency spectrum.

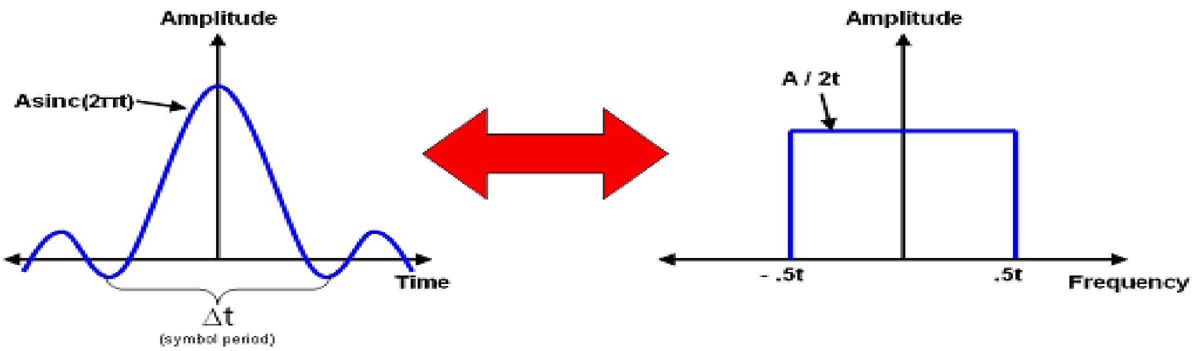

Fig. 2 Basic Fast Fourier Transform

HMM: Hidden Markov Model is used to classify speech signal and noise signal. The most important property is memory less is used i.e., comparing signal with only present stage.

FIR Filter: The adaptive FIR Filter is used to remove the noise from speech signal. Here, FIR Filter is act as low pass filter. It passes low frequency signals up to certain cut-off level and block high frequency component.

Output Signal: It will get the output signal which is noise free and its performance can be calculated.

C. *LMS(LEAST MEAN SQURE):*Comparing results with the LMS and comparing parameter i.e., input,output signal SNR ration and performance of system. Least Mean Squares (LMS) algorithm is a logical choice of subject to examine, because it combines the topics of linear algebra and graphical models, the latter case because we can view it as the case of a single.

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IV. SIMULATION RESULTS

In this section, speech enhancement system is studied to demonstrate the algorithm presented in this paper.Comparison of the adaptive algorithm with the LMS algorithm is shown by the results of simulationsusing MATLAB.

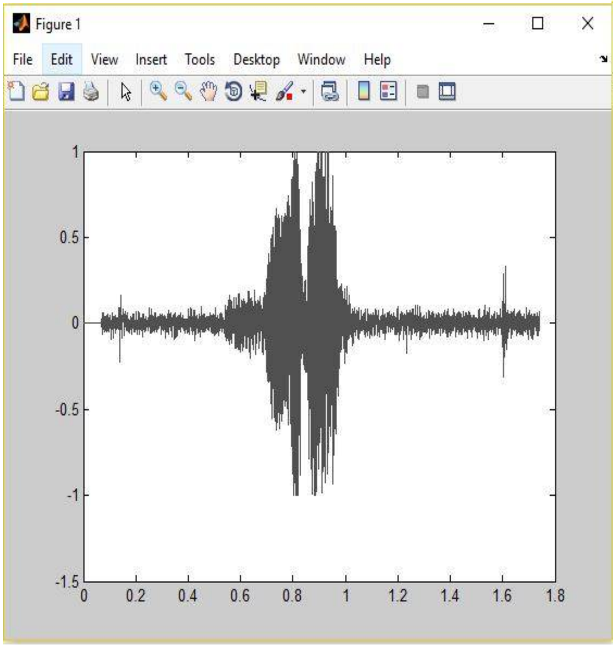

Fig.1. Input signal

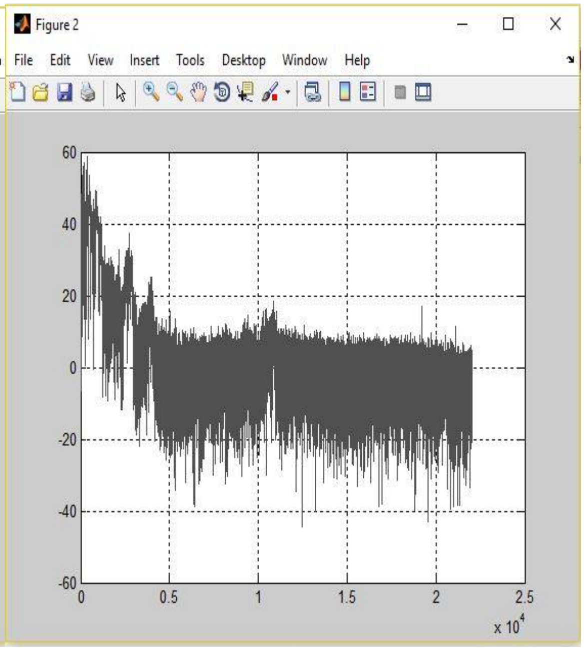

Fig. 2 FFT of Input signal

The Fig.1 shows the input signal. The input signal is nothing but the given signal. The signal is given in the wave format. The speech signal is a continuous signal but, here we use in digital system hence it becomes discrete. Hence, the continuous signal is divided into multiple discrete signals and according to it frame of signal is considered.Fig.2 shows the Fast Fourier Transform of input signal. FFT is nothing but to see the speech signal in another way. In order to enhance the signal, it only possible to enhance frequency component. Also, it helps to reduce the computational time. In frequency domain the unwanted frequency component is reduced (i.e., noise) and it boosts the signal.

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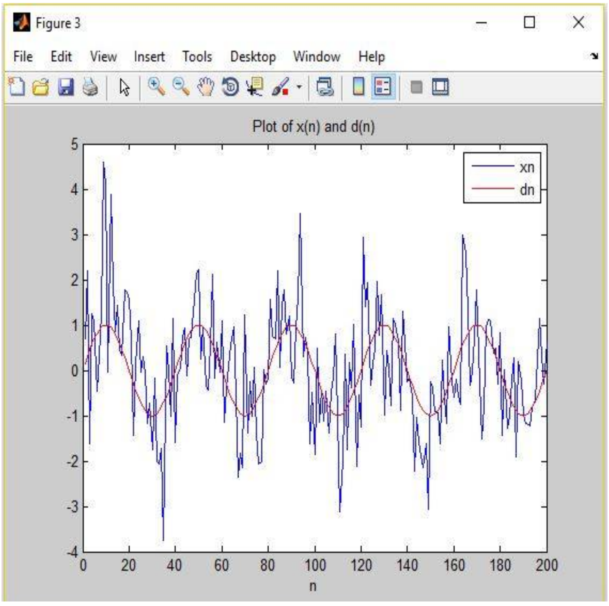

Fig. 3Making a frame with respect to noise

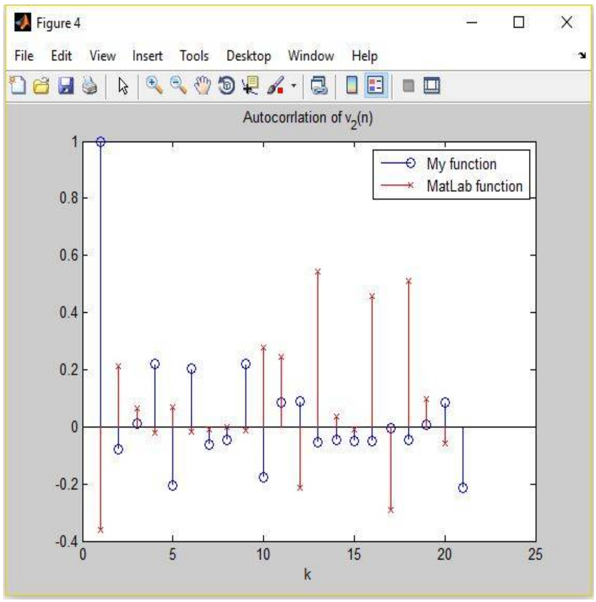

Fig. 4 Autocorrelation signal

The Fig.3 shows making of frame respect to noise. We take particular frame which contain noise and we have to reduce noise from it. Hence, take a sine wave to cut speech signal according to it. The above Fig.4 shows the autocorrelation signal. It is also called as a serial correlation. It is comparing signal with itself providing similarities and difference.

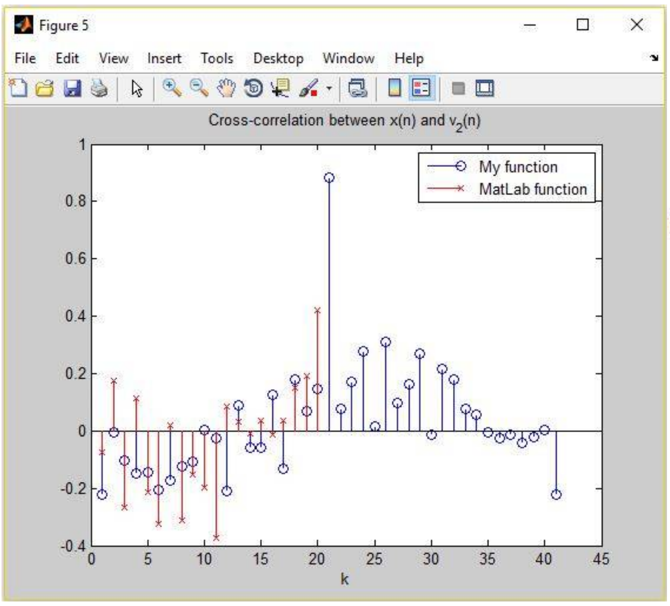

Fig.5cross-correlation between input signal and autocorrelation signal

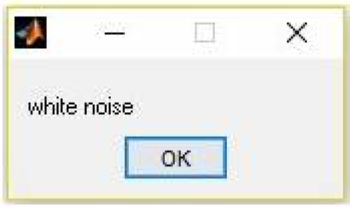

Fig. 6 Type of noise present in Input signal

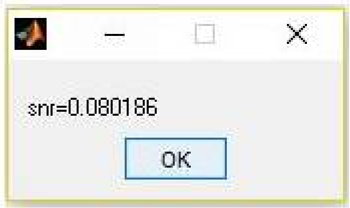

Fig. 7 SNR Ratio of signal

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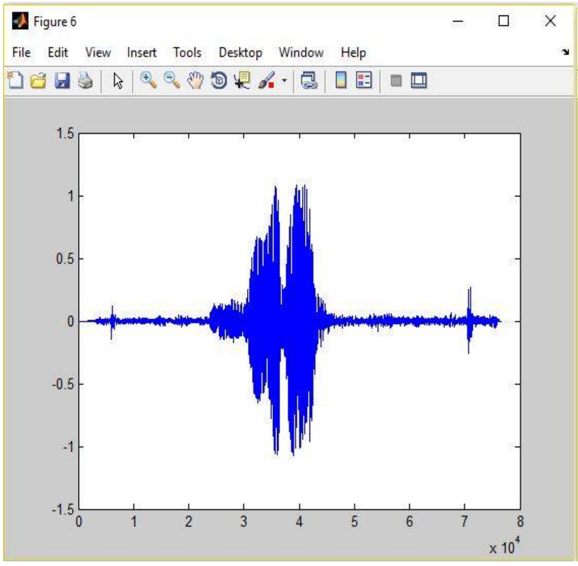

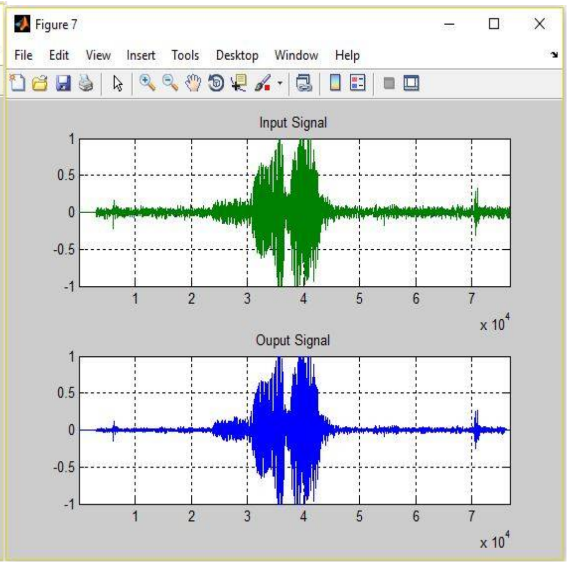

Fig. 8 Output signal

fig. 9 Input signal and Output signal compression

Fig.8 shows the output signal without noise or improved frequency component.Fig .9 shows the compression between input signal and output signal. Input signal shows the noise frequency in output signal. The unwanted edges are removed from input signal.

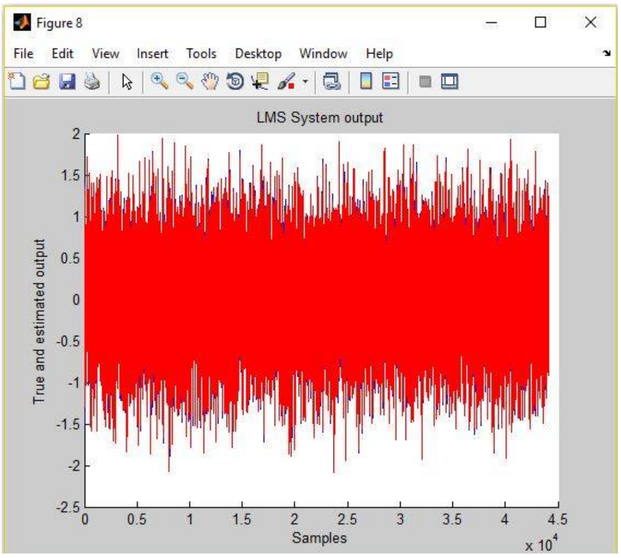

Fig. 10 LMS System output

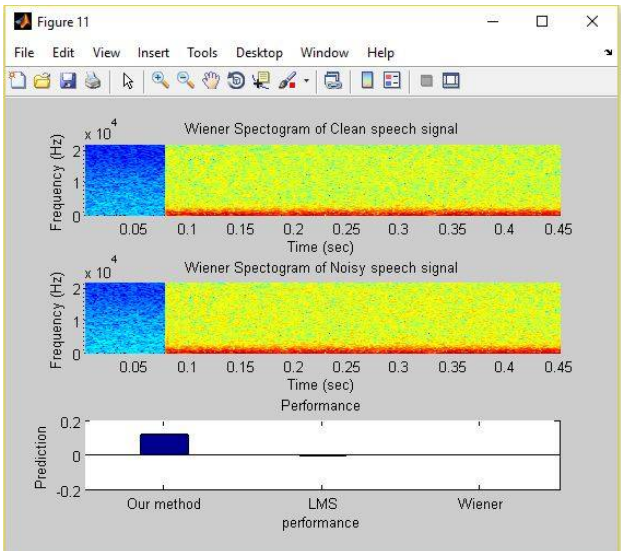

Fig. 11 System performance comparison using spectrogram

Fig 10 shows the output of LMS System showing some noise edges. In the edges noise is present in big extend.Fig 11 shows the comparison of system performance using spectrogram.

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Table No. I Comparison between adaptive FIR Filter and conventional LMS filter

Sr.No.	Signal	Type of noise	SNR of adaptive FIR filter	SNR of conventional LMS filter
1	S1	White noise	0.08018	0.043293
2	S2	White noise	0.13392	0.012447
3	S3	White noise	0.94529	0.793600
4	S4	White noise	0.71944	0.078079
5	S5	White noise	0.53824	0.115340
6	S6	White noise	0.22552	0.937150
7	S7	White noise	0.33464	0.173400

The above table shows the type of noise and comparison SNR result between adaptive FIR Filter and conventional LMS filter. Adaptive FIR Filter achieves higher SNR Ratio hence, we can isolated speech signal from noise easily. We take seven different signals on different condition. From the table it shows the SNR of adaptive FIR filter is achieved high hence system performance is good.

V. CONCLUSION

Speech signal is one of the best communication mediums and it has wide range of application. For proper communication noise free signal is very important. But, when it comes in contact with noise it becomes very hard to find original speech signal. Hence, using speech enhancement technique the noise from speech signal is removed. For removal of noise proposed FIR filter method uses adaptive coefficient for linear and non-linear signal for stability purpose. Taking one speech signal and processing it through different algorithm and recorded there SNR value. Base SNR value performance is calculated. Adaptive FIR Filter algorithm have higher SNR value than traditional LMS. Also, prediction and separation of noise from speech signal is more accurate in adaptive FIR Algorithm. Hence, the performance of system is better than traditional method.

REFERENCES

[1] Caiyun Wu, Xiaoying Wang, Hong Zhao,"An adaptive algorithm using FIR filter for speech enhancement systems" 29th Chinese Control and Decision Conference (CCDC) 2017.

[2] Nishitha Danthi, Dr. A R Aswatha "Single Channel Speech Enhancement Technique An Implementation on MATLAB," International Conference on Innovative Mechanisms for Industry Applications (ICIMIA 2017).

[3] V. K. Ingle and j. g. proakis, "digital signal processing using matlab", cengage learning, 2016 .

[4] Y. Zhang and L. Shi, "An improved perceptual KLT approach for speech enhancement," in Proc. the 4th InternationalConference on Computer Science and Network Technology, 2015.

[5] C. Y. Wu, "Study of Speech Enhancement System Using Combinational Adaptive Filtering," in Proc. IEEE ComputerDistributed Control and Intelligent Environmental Monitoring, 2012.

[6] C. Y. Wu, "Active Noise Control Based on Adaptive Bilinear FL Algorithm," in Proc. the IEEE Computer DistributedControl and Intelligent EnvironmentalMonitoring, 2012.

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