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Performance Enhancement of Information Hiding in FM and AM with Rician Channel

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ABSTRACT

In hands-free telephony and in teleconference systems, the main aim is to provide a good free voice quality when two or more people communicate from different places. The problem often arises during the conversation is the creation of acoustic echo. This problem will cause the bad quality of voice signal and thus talkers could not hear clearly the content of the conversation, even though lost the important information. This acoustic echo is actually the noise which is created by the reflection of sound waves by the wall of the room and the other things exist in the room. The main objective for engineers is the cancellation of this acoustic echo and provides an echo free environment for speakers during conversation. For this purpose, scientists design different adaptive filter algorithms. Our paper is also to study and simulate the acoustics echo cancellation by using different adaptive filter algorithms, to compare and analyze the performance of LMS, NLMS and UNANR on the basis of SNR and PSNR, using MATLAB R2012a.

Keyword: *LMS, NLMS, UNANR, Speech, Channel*

I. INTRODUCTION

Speech is most natural form of human communication. It existed since human civilizations began and even till now. The perception of speech signal is usually measured in terms of its quality and intelligibility. The quality is a subjective measure that indicates the pleasantness or naturalness of the perceived speech. Intelligibility is an objective measure which predicts the percentage of words that can be correctly identified by listeners. Enhancement means the improvement in the value or quality of something. When applied to speech, this simply means the improvement in intelligibility and/or quality of a degraded speech signal by using signal processing tools. By speech enhancement, it refers not only to noise reduction but also to de reverberation and separation of independent signals.

The whole dimension of communications has been changed by the rapid growth of technology. Today people are more interested in hands-free communication, which makes use of loud speaker and high gain microphone, in place of the old modelled wired telephone. The main advantage of wireless system is that, more than one person can participate in conversation while freely moving in the room.

A. *Speech Enhancement*

The main objective of speech enhancement technique is to improve the quality and minimize the loss in intelligibility of the signal and listener fatigue. The basic overview is shown in Figure 1.

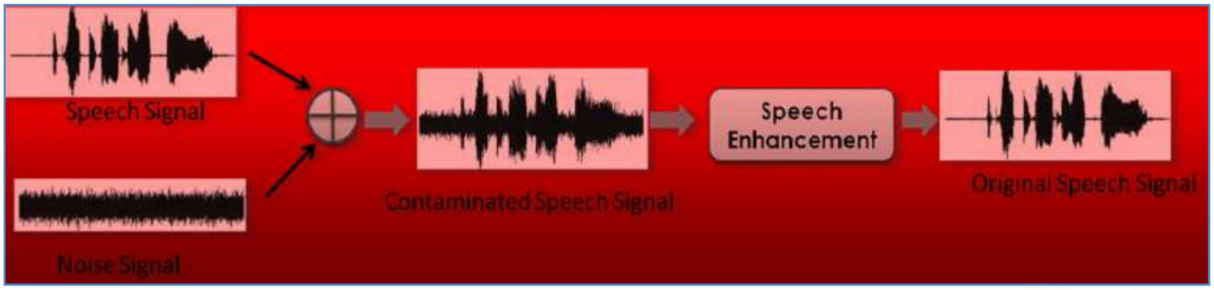

Fig. 1: Basic Speech Enhancement System

Continuous improvement of communication and multimedia systems has led to the widespread use of speech recording and processing devices, e.g., mobile phones, speech recognition tools. In most practical situations, these devices are being used in environments where undesirable background noise exists. Degraded speech can cause problems for both mobile communication and speech recognition systems. Nowadays, all the people use the communication devices almost as a primary good: telephones, mobiles, internet and the customers demand a high coverage and quality.

II. ADAPTIVE FILTER STRUCTURE

The basic idea of an adaptive noise cancellation algorithm is to pass the corrupted signal through a filter that tends to suppress the noise while leaving the signal unchanged. This is an adaptive process, which means it does not require a priori knowledge of signal or noise characteristics. Adaptive noise cancellation (ANC) efficiently attenuates low frequency noise for which passive methods are ineffective. Suppose an adaptive filter with a primary input i(n), that is noisy speech signal S(n) with additive noise C(n). While the reference input is noise r(n), which is correlated in some way with C(n). If the filter output is f(n), the output of the summer O(n) is nothing but the error signal and it is written as, filter error e = {S(n) + C(n)} - f(n).

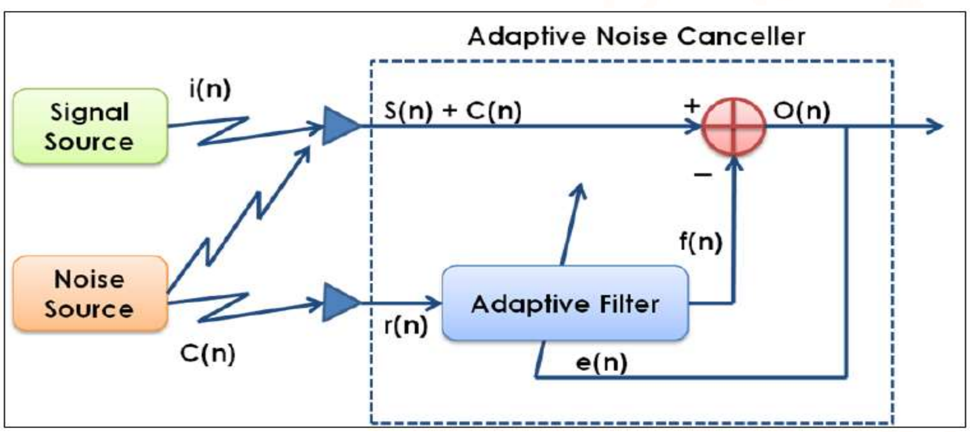

Fig. 2: Adaptive Filter Structure

III. SIMULATION BLOCK DIAGRAM

This project is all about the speech enhancement of voice signal using different adaptive filters.

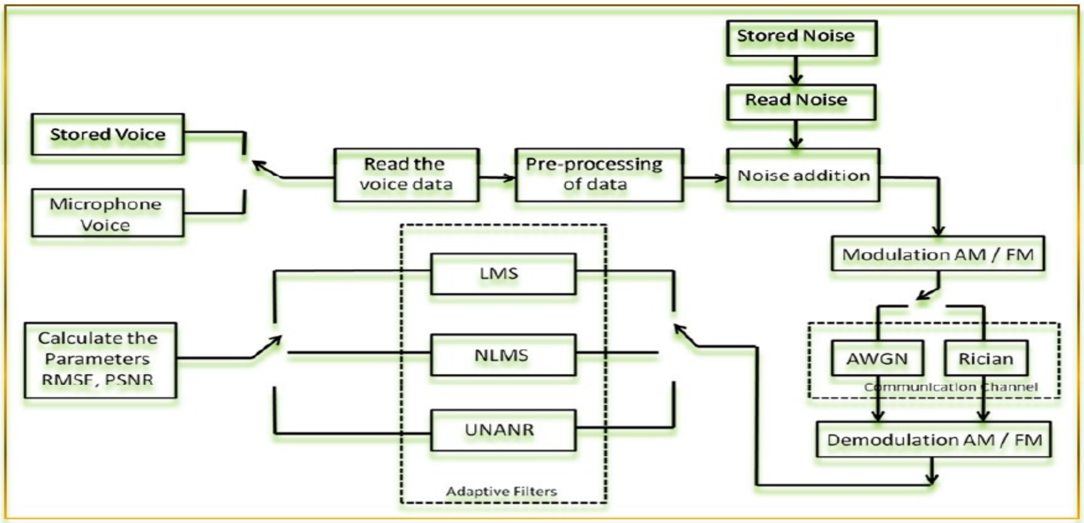

Fig. 3: Simulation Block

The speech signal is first mixed with a noise signal then it is modulated with two of the analog modulation techniques i.e. AM and FM; one at a time. Then AWGN is chosen as a communication channel in configuration with one of the modulation technique. Then at the receiver side demodulation if performed and filtered with adaptive filters. The same process is also done with Rician fading channel. The filters which are used are: LMS, NLMS and UNANR. The following block diagram gives the complete idea of the project. The major units are modulation, communication channel and adaptive filters. This project comprises of two types of input voice signal: stored voice signal and microphone voice signal.

IV. SIMULATION RESULTS

The performances of the adaptive filters are compared with respect to the variation in SNR (dB). The used modulation techniques are AM and FM and the considered channels are AWGN and Rician fading channel. Under speech enhancement techniques, for improving quality of adaptive filters a newly emerging filter is used i.e. UNANR. This filter's performance is compared with two traditionally used adaptive filters; LMS and NLMS. The above considered technologies have been combined using the MATLAB R2013a software (Version 8.1.0). Now for different cases for the performance evaluation, the selected range of SNR is -5 to 40. However there is no restriction of the SNR range. But, if SNR range increases then the simulation time will increase and the considered noise removal capability may decrease. It is necessary to evaluate the performance of the system, and PSNR and RMSE provide a base for comparing the performances of different filters.

A. AM with Rician fading channel

In this case AM is selected to transmit the whole speech signal after addition of background noise at the transmitter side. Rician fading channel is selected as a communication channel for transferring the speech signal. In Rician fading channel, channel noise gets added to the speech signal. At the receiver side first AM demodulation is performed then speech signal is passed through one of the adaptive filter. Firstly LMS filter is selected and PSNR and RMSE signal parameters are recoded. Secondly NLMS filter is selected for the same received demodulated speech signal. And at the last UNANR filter is selected for the same received demodulated speech signal. Graphs have been plotted to check the performance of the adaptive filters.

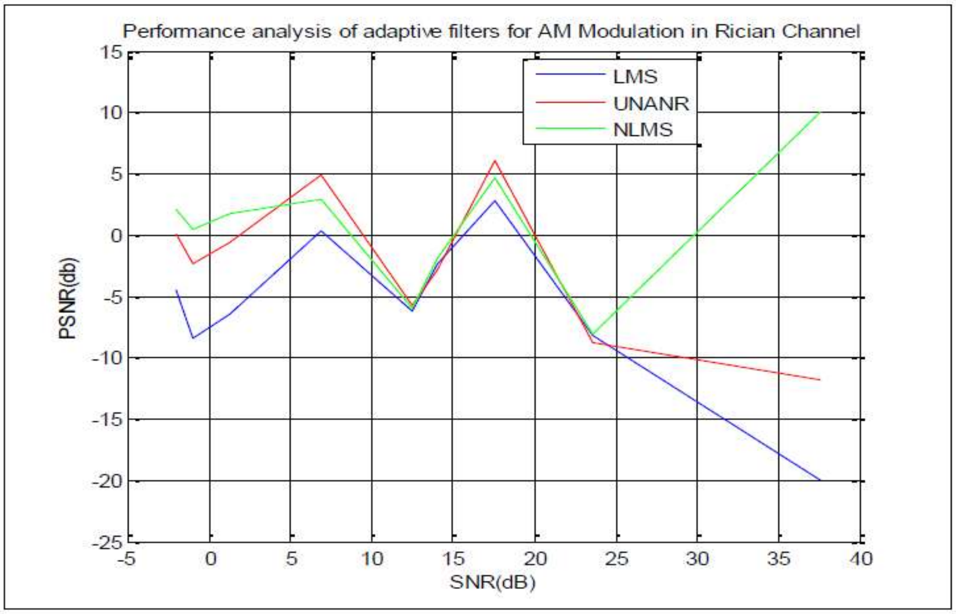

Fig. 4: Adaptive filtering on AM with Rician channel for stored voice

Communication channel for transferring the speech signal. In Rician fading channel, channel noise gets added to the speech signal. At the receiver side first AM demodulation is performed then speech signal is passed through one of the adaptive filter. Firstly LMS filter is selected and PSNR and RMSE signal parameters are recoded. Secondly NLMS filter is selected for the same received demodulated speech signal. And at the last UNANR filter is selected for the same received demodulated speech signal. Graphs have been plotted to check the performance of the adaptive filters.

B. FM with Rician fading channel

In this case FM is selected to transmit the whole speech signal after addition of background noise at the transmitter side. Rician fading channel is selected as a communication channel for transferring the speech signal. In Rician fading channel, channel noise gets added to the speech signal. At the receiver side first FM demodulation is performed then speech signal is passed through one of the adaptive filter. Firstly LMS filter is selected and PSNR and RMSE signal parameters are recoded. Secondly NLMS filter is selected for the same received demodulated speech signal. And at the last UNANR filter is selected for the same received demodulated speech signal.

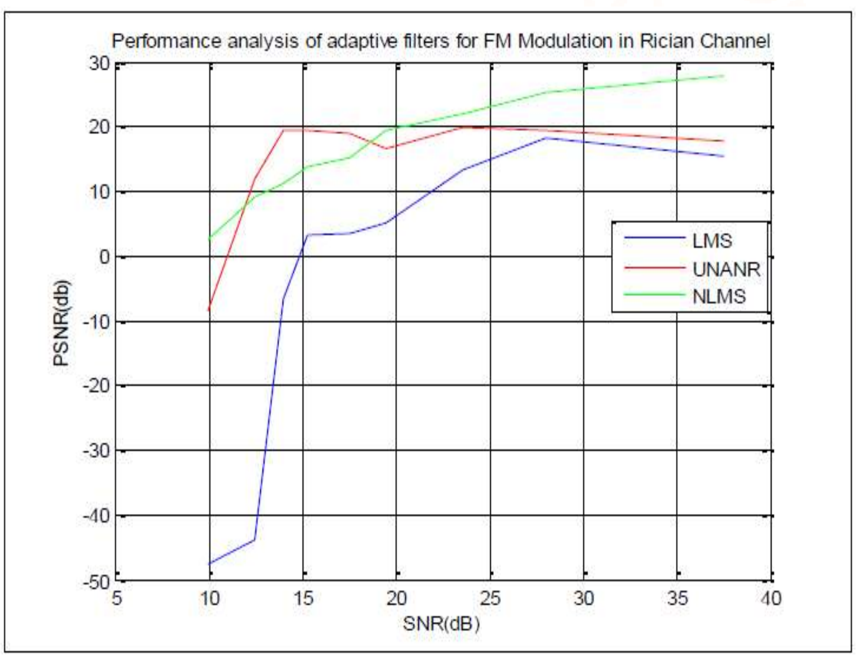

Fig. 5: Adaptive Filtering on FM with Rician Channel For Stored Voice

V. CONCLUSION

From all the performed experiments it is apparent that NLMS and UNANR filter have better performance than adaptive LMS filters. When the online voice signal is taken as input to the speech enhancement system then UNANR performed slightly better than NLMS filter. Though UNANR takes little more time to filter speech signal from noise in comparison with LMS and NLMS but has better convergence rate than other two. As soon as the Signal to noise ratio in dB improves performances of the filters get improved and when background noise level gets increased then performances of LMS, UNANR and NLMS filters gets degraded.

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