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PERFORMANCE OF REAL TIME VOICE IMPROVEMENT USING LMS, NLMS AND UNANR FILTERS

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ABSTRACT
Speech intelligibility can be enhanced using acoustic properties of "clear speech", the speech produced by a speaker with an intention to improve intelligibility in a difficult communication environment. The research objective is to devise a signal processing technique based on the properties of clear speech for improving perception of stop consonants for use in speech communication devices and hearing aids. This study was made to estimate the frequency of measuring voltage or current signal in presence of random noise and distortion. Here we are first using linear techniques such as least mean square (LMS), algorithm for measuring the frequency from the distorted voltage signal. Then comparing these results with nonlinear techniques such as nonlinear least mean square (NLMS), and UNANR algorithms with different modulation techniques was Amplitude Modulation and Frequency Modulation and communication channels i.e. AWGN Channel. Signal performance parameter PSNR measured and compared with respect to Signal to Noise Ratio. The performances of these algorithms are studied through MATLAB R2012a simulation.

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. This is a very difficult problem for two reasons: First, the nature and characteristics of the noise signals can change dramatically in time and between applications. It is also difficult to find algorithms that really work in different practical environments. Second, the performance measure can also be defined differently for each application.

II. POWER SUBTRACTION
Power Subtraction a power subtraction system with the concept of noise overestimation is introduced. Over subtraction is used to compensate for imperfect noise estimation, this system has interesting points; the variable over subtraction factor is based on the estimated SNR in each band, The rationale is that strong SNR bands are indicative of a strong speech component and there is no need for aggressive subtraction, A spectral floor is used to ensure the presence of a low broadband noise and its effect is to mask any musical noise that may be present.

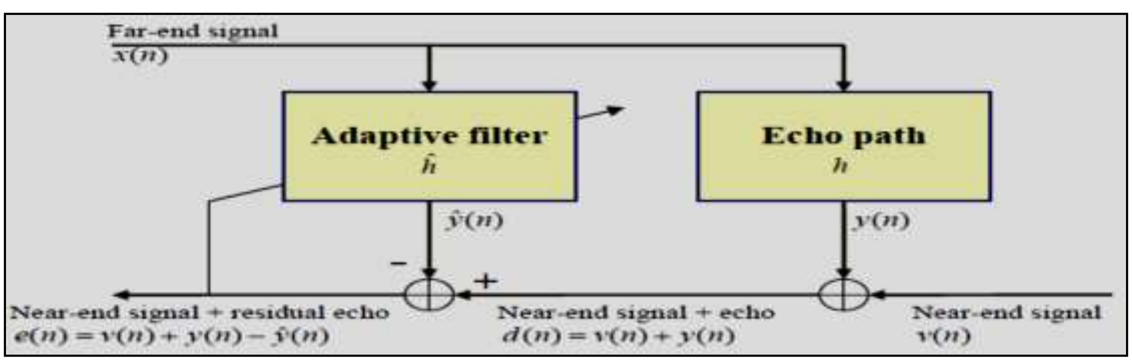

Fig. 1: Block diagram of AEC

Adaptive filtering is the process which is required for echo canceling in different applications. Adaptive filter is such type of filter whose characteristics can be changed for achieving optimal desired output. An adaptive filter can change its parameters to minimize the error signal by using adaptive algorithms. The error is the difference between the desired signal and the output signal of the filter. The figure 1 shows the basic model of adaptive filter used in AEC.

III. 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), then

$$e^2 = \{S(n) + C(n)\}^2 - 2f(n)\ \{S(n) + C(n)\} + f(n)^2 \quad 1$$
$$= \{C(n) - f(n)\}^2 + S(n)^2 + 2S(n)\ C(n) - 2f(n)\ S(n) \quad 2$$

Since the signal and noise are uncorrelated, the mean-squared error (MSE) is:

$$E[e^2] = E[\{C(n) - f(n)\}^2] + E[S(n)^2 \quad 3$$

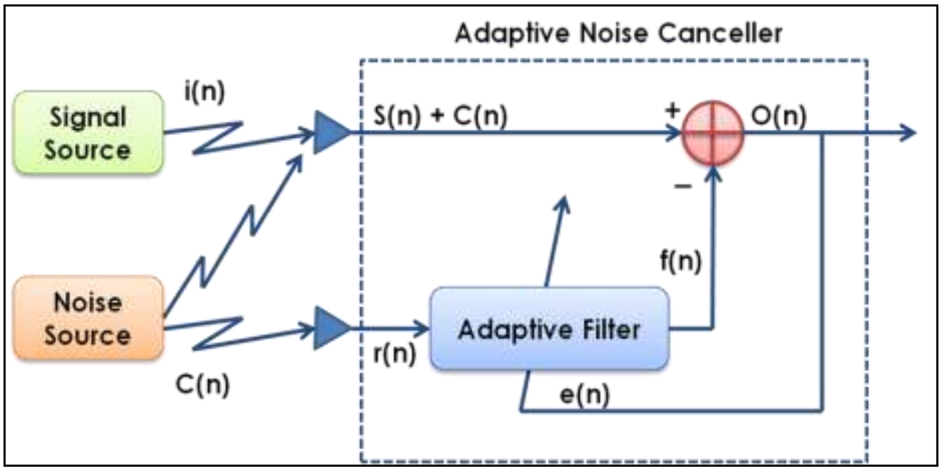

Fig. 2: Adaptive Filter Structure

Minimizing the MSE results in a filter error output that is the best least-squares estimate of the signal S(n). The adaptive filter extracts the signal, or eliminates the noise, by iteratively minimizing the MSE between the primary and the reference inputs. Minimizing the MSE results in a filter error output f(n) that is the best least-squares estimate of the signal S(n).

In Acoustic Echo Cancellation (AEC), the adaptive filter plays the main role to adapt the filter tap weight in order to overcome the echo problem. There are different types of algorithms are used for this purpose such as Least Mean Square (LMS), Normalized Least Mean Square (NLMS), Recursive Least Square (RLS) and Affine Projection Algorithm (APA) and etc. The LMS is widely used algorithm for adaptive application such as channel equalization and echo cancellation. This algorithm is the most simple if we compare it with NLMS and RLS algorithm. The normalized least mean square (NLMS) is also famous algorithm due to its computational simplicity.

IV. SIMULATION RESULTS

Now input had been taken online speech from microphone. For this experiment also there can be total four possible configurations possible, as we have two types of modulation techniques and two types of communication channels. So let’s start analyzing the performances of different filters with all possible combinations.

A. Performance of Amplitude Modulation with AWGN channel for online voice

Consider the case of online input speech signal from microphone. In this case AM is selected to transmit the whole speech signal after addition of background noise at the transmitter side.

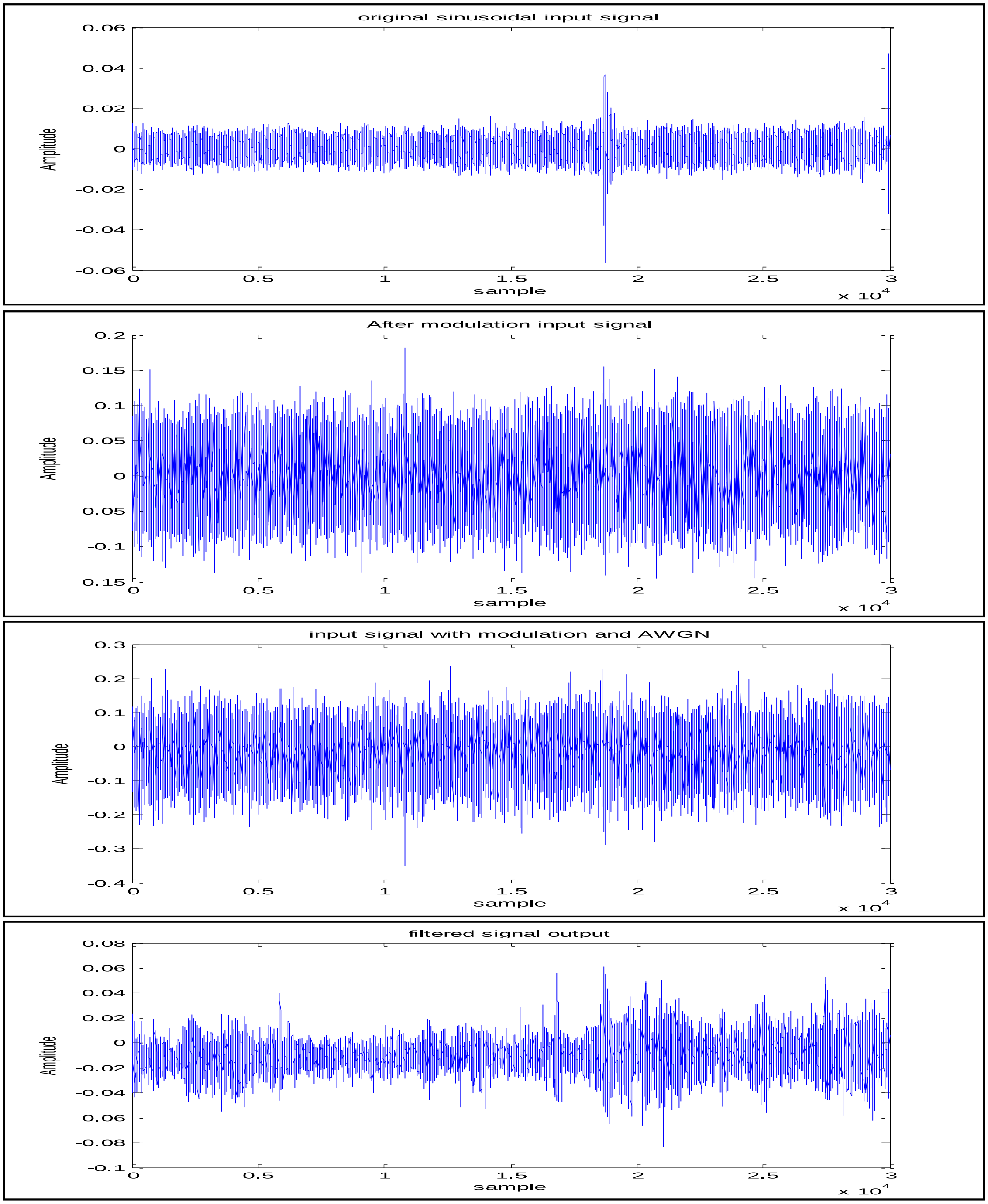

Fig. 3 Online Performance of Amplitude Modulation with AWGN channel , (a) Original stored sinusoidal signal, (b) After modulated input signal, (c) Input signal with modulation and AWGN, (d) Filtered signal output

AWGN channel is selected as a communication channel for transferring the speech signal. In AWGN 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. First, LMS filter is selected and PSNR and RMSE signal parameters are recoded. Second, NLMS filter is selected for the same received demodulated speech signal.

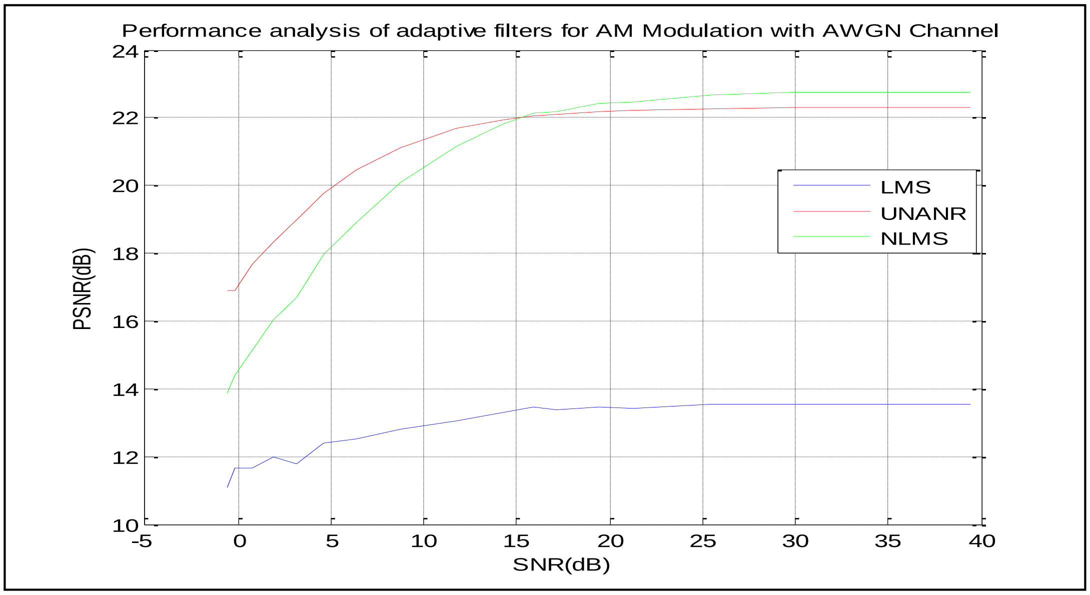

Fig. 4: Adaptive filtering on AM with AWGN channel for online voice

B. Performance of Frequency Modulation with AWGN channel for online voice

In this case FM is selected to transmit the whole online speech signal after addition of background noise at the transmitter side. AWGN channel is selected as a communication channel for transferring the speech signal.

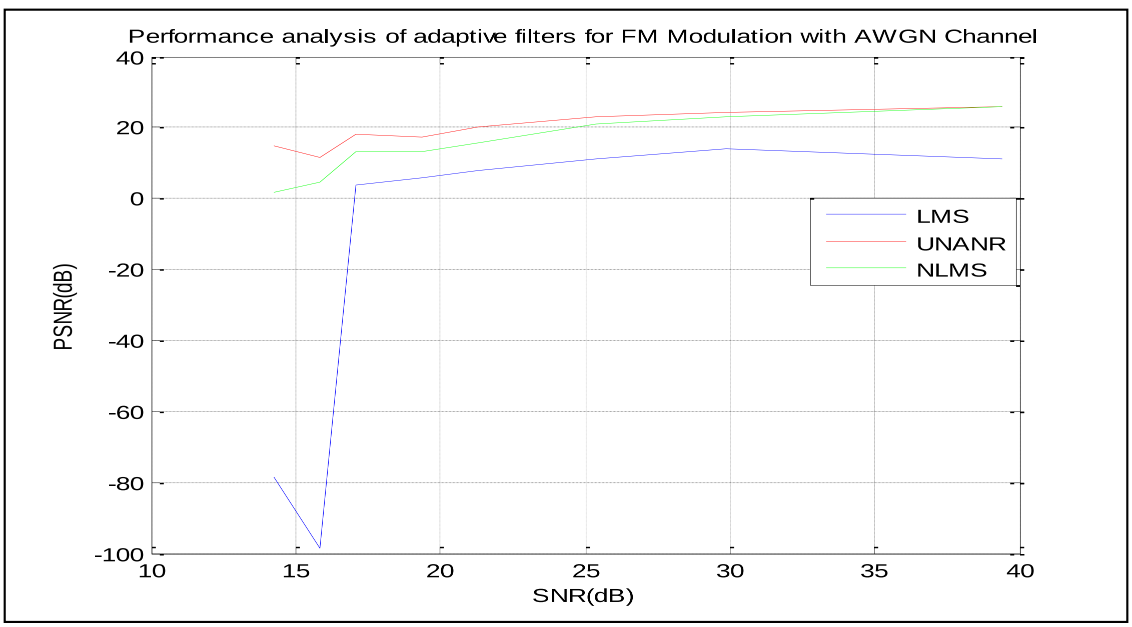

Fig. 5: Adaptive filtering on FM with AWGN channel for online voice

In AWGN 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. Graphs have been plotted to check the performance of the adaptive filters. Graphs are plotted between SNR verses PSNR.

V. CONCLUSION

Performances of these filters are measured with respect to PSNR verses SNR when speech signal is affected by both background and channel noise. The comparison and analysis of performances of these adaptive filters have been done through plotted graphs. From all the performed experiments it is apparent that NLMS and UNANR filter have better performance than adaptive LMS filters. Future work can be extended by implementing the above realization using digital signal processors and for higher filter orders with more computational parameters and to improve speech quality.

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