Global Journal of Advance Engineering Technologies and Sciences

PERFORMANCE OF NOVEL RLS ADAPTIVE FILTERING FOR SPEECH ENHANCEMENT USING FREQUENCY MODULATION

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ABSTRACT

Speech recognition is one of the most challenging applications of signal processing. It is an adaptive filter which can adjust itself its transfer function corresponding to the best adaptive algorithm conveyed by an error or corrupted signal. The Recursive Least Squares (RLS) algorithm has established itself as the "ultimate" adaptive filtering algorithm in the sense that it is the adaptive filter exhibiting the best convergence behaviour. In this paper, we propose a speech imprudent based on Recursive Least Squares (RLS) adaptive filter of speech signals. Experiments were performed on noisy data which was prepared by adding AWGN, to clean speech samples at -2dB, 0dB, 5dB, 10dB and 15dB SNR levels with Frequency modulation. We then compare the noise cancellation performance of proposed RLS algorithm with existing NLMS algorithm in terms of Mean Squared Error (MSE), Pick to Signal to Noise ratio (PSNR) and SNR Loss. Based on the performance evaluation through the simulation Matlab

INTRODUCTION

There are many speech enhancement method proposed for noise reduction and to improve the noise quality and intelligibility. The earliest work in adaptive noise cancelling known to the author was performed by Howells and Applebaum and their colleagues at the General Electric Company between 1957 & 1960. The best known commercial application of adaptive filtering grew from the work during these periods of Lucky at the Bell Laboratories. The adaptive noise cancelling system at Stanford University was design and built in 1965 by two students. Since 1965, adaptive noise cancelling has been successfully applied to a number of application. Several methods have been reported so far in the literature to enhance the performance of speech processing systems some of the most important ones are: In the another things Continuous improvement of communication and multimedia system has led to the widespread use of speech recording and processing device, e.g., mobile phones, speech recognition tools.

FILTER TECHNIQUES

Adaptive techniques use algorithms, which enable the adaptive filter to adjust its parameter to produce an output that matches the output of unknown systems. This algorithm employs an individual convergence factor that is updated for all the adaptive filter coefficient at each iteration. The Filters used for direct filtering can be either fixed filter or Adaptive filter.

A. **Adaptive Filters: -** Adaptive filters, on the other hand, have the ability to adjust their impulse response to filter out a correlated signal in the input. They require little or no a priori information of the signal and noise characteristics. (If the signal is narrowband and noise broadband, which is generally the case, or vice versa, no a priori information is needed; otherwise they require a signal (desired response) that is correlated in some sense to the signal to be estimated.) Moreover adaptive filters have the capability of adaptively tracking the signal under non-stationary conditions. Rather than use a system identification application to demonstrate the RLS adaptive algorithm, or a noise cancellation model, this example use the inverse system identification model shown in here.

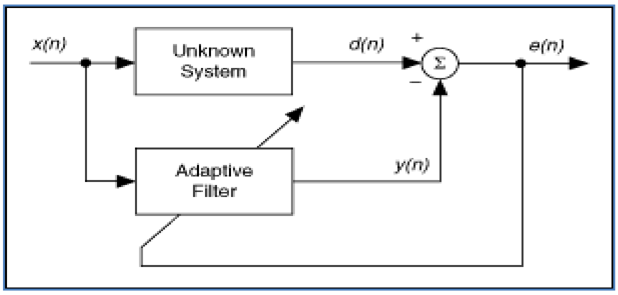

Fig. 1: Adaptive Filter

ISSN 2349-0292
Impact Factor (PIF): 2.675

B. **Fixed Filters: -** The design of fixed filters requires a priori knowledge of both the signal and the noise, i.e. if we know the signal and noise beforehand, we can design a filter that passes frequencies contained in the signal and rejects the frequency band occupied by the noise.

PROJECT BLOCK DIAGRAM

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. The stored voice signal is a wave file which is stored in the computer and microphone voice signal is a speech input from microphone. At a time only one type of input signal is selected. This input speech signal is processed and audible with signal graph in graph window of the project UI. A noise signal is mixed with this speech signal. The SNR value can be of user's choice as it is controlled by user while mixing noise signal. In other words noise signal level is set by user while mixing the noise into the speech signal. After addition of noise and pure speech signal modulation technique is applied. This modulated signal can be seen in graph window of the project UI. Then this modulated signal is sent through the one of the communication channel. At the receiver side received signal is first demodulated and the filtered with one of the three adaptive filters.

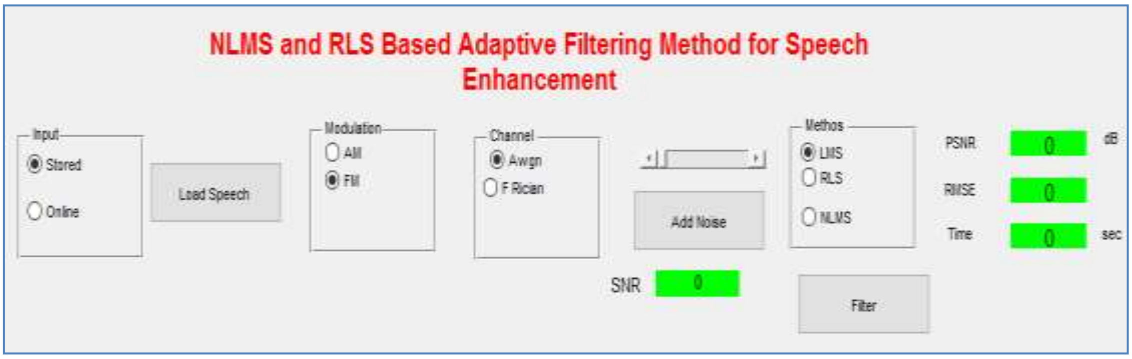

Fig. 2: Simulation block diagram of NLMS and RLS

SIMULATION Results

The primary objective of the adaptive filter is minimization of the error signal e(k) which clearly depends on the nature of the input signals and the adaptive algorithm used. The performance of these algorithms are measured based on the metrics namely MSE, SNR and SNR Loss which are explained below. The audio signal is first mixed with a noise signal then it is modulated with two of the analog modulation techniques. 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 filters which are used are NLMS and RLS. It is necessary to evaluate the performance of the system, and PSNR and RMSE provide a base for comparing the performances of different filters. In figure 3 comparison of NLMS and RLS filters performance for AWGN Channel using Amplitude modulation.

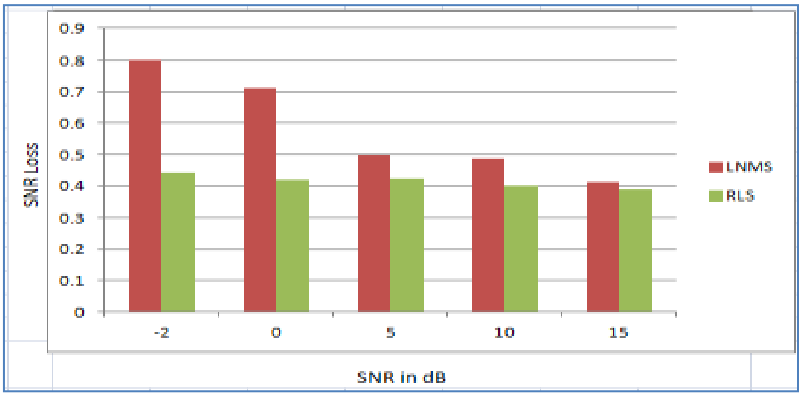

Fig. 3: Comparison of NLMS and RLS filters performance for AWGN Channel using FM

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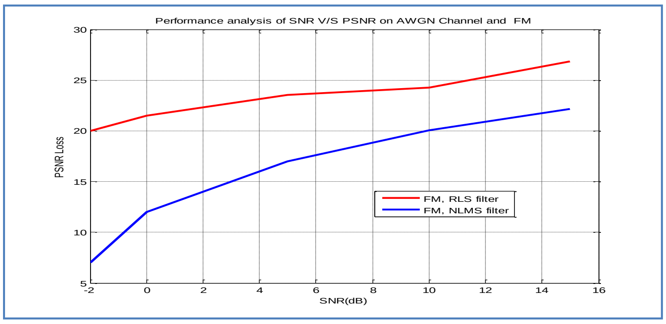

Fig.4: Performance of SNR Loss v/s SNR for AWGN Channel using FM

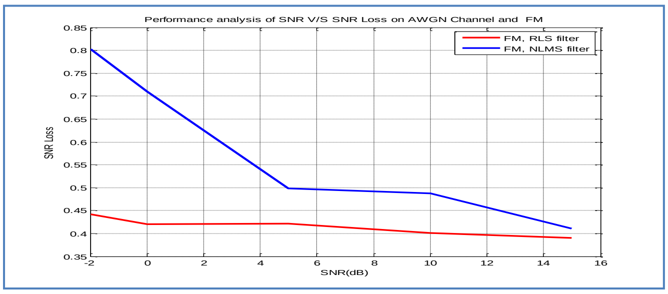

Fig.5: Performance of PSNR v/s SNR for AWGN Channel using FM

CONCLUSION

In this work different adaptive algorithms are tested for two different types of noisy signals. First speech contains the random noise and the second speech contains the AWGN noise. The convergent rate of NLMS is the fastest but it is not good resolution. The RLS algorithm is two times faster than NLMS algorithm. Amount of Nose is reduced 15 dB more than NLMS. The MSE of NLMS algorithm is 0.0001703, MSE of RLS algorithm is 0.000413 at 15dB SNR, and PSNR is 22.1550 and 26.8742 respectively NLMS and RLS algorithm.

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