ISSN: 2277-5528
Impact Factor: 4.015

INTERNATIONAL JOURNAL OF ENGINEERING SCIENCES & MANAGEMENT

SIMULATION AND ANALYSIS OF NLMS AND RLS ALGORITHMS USING AWGN CHANNEL

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ABSTRACT

Reduction of background noise is still a challenging task for speech enhancement which affects the performances of communication systems like mobile phones, voice communications etc. The performance of the speech recognition system is also reduces if the speech signal is corrupted by noise. To remove the noise present in the speech signal, the adaptive filters shown the improvement in increasing the Signal to Noise Ratio values. The simulations are done using MATLAB simulation toll R2013a with communication block, speech corpus for different SNR values using Normalized Least Mean Square (NLMS) and Recursive Least Squares (RLS) filter.

Keywords: NLMS, RLS, Speech Enhancement, SNR, PSNR.

INTRODUCTION

Digital filtering of the speech signal is complicated in dynamically changing environment. The main frequency components of the speech signal are situated in the range from 5 Hz to 4 kHz [1]. Simultaneously the audible noise is situated in the frequency range from 20 Hz to 20 kHz [1]. Speech enhancement is pertained with improving the perceptual part of speech that has been dissipated by various noises. In many applications, such as mobile telephony, speech communication, information forensics, speech recognition, video conferences etc [2], the speech enhancement algorithms improve the quality of dissipated speech. In Fig.1. Clean speech that gets corrupted with background noise is sent through a speech enhancement block. The purpose of such a block is to estimate the speech magnitude from the noisy observation. Most common speech enhancement techniques operate in the frequency domain. The basic operation of adaptive filter involves two processes [6].

1. Filtering process that produces an output signal in response to a given input signal and
2. Adaptation process that aims to adjust the filter parameters to the environment.

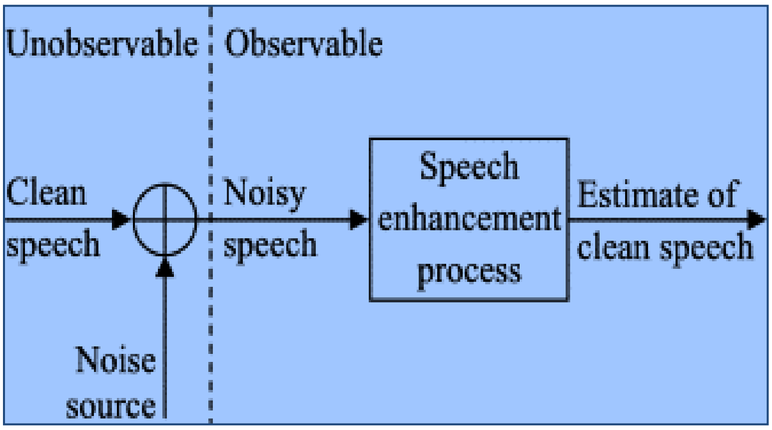

Fig. 1: Block diagram of basic speech enhancement in the additive noise [3].

ADAPTIVE FILTERS

Adaptive filters that use simple in implementation algorithms, such as Least Mean Squares (LMS), requires many iterations to adapt filter coefficients. More advanced algorithms, such as Normalised Least Mean Squares (NLMS) or Recursive Least Squares (RLS), are more complicated in implementation. Consequently the system load increases and it leads to the more strict limitations to the maximum filter order that can be implemented in DSP.

Recursive Least Squares (RLS) algorithm uses all the information present in the input signal. It is recursive because the present coefficients update by using the past coefficients. It provides faster convergence rate than the LMS algorithm. The RLS adaptive algorithm is mostly used for determining the coefficients of an adaptive filter [2], for application where the fast convergence rate is vital, NLMS algorithm is not applicable. The more complex RLS algorithm maintains a good rate of adaptation but the prize to be paid is an order-of magnitude increase in complexity. Moreover RLS algorithm is known to have stability issues due to the recursive covariance update formula [4].

A. *Active Noise Cancelling*

The active noise cancelling, also called adaptive noise cancelling or active noise canceller belongs to the interference cancelling class. The aim of this algorithm, as the aim of any adaptive filter, is to minimize the noise interference or, in an optimum situation, cancel that perturbation [2]. The approach adopted in the ANC algorithm, is to try to imitate the original signal x(n). A scheme of the ANC can be viewed in Fig.2 [5]. In the ANC, as explained before, the aim is to minimize the noise interference that corrupts the original input signal. In the figure above, the desired signal d(n) is composed by an unknown signal, that s(n) is called corrupted for an additional noise n2(n), generated for the interference. The adaptive filter is then installed in a place that the only input is the interference signal n1(n)[5].

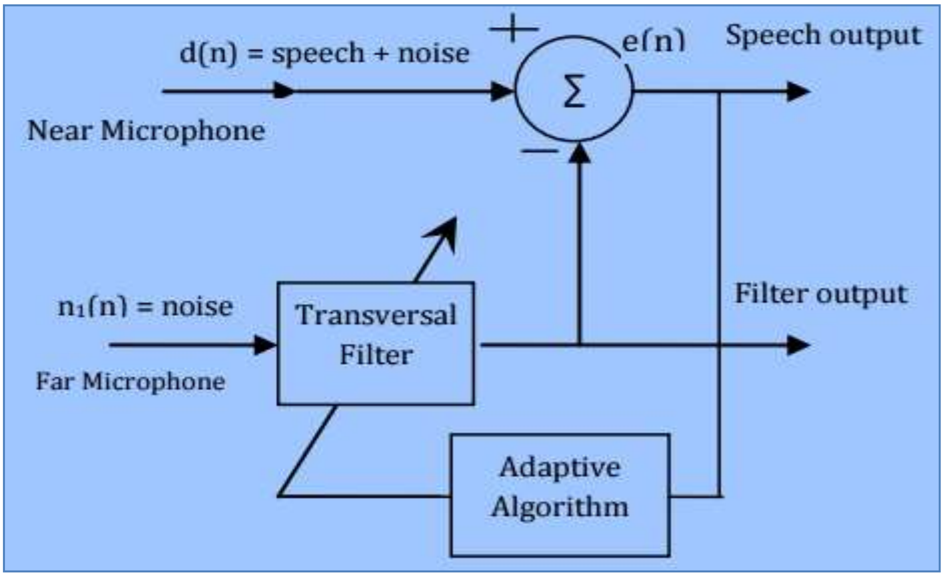

Fig. 2: Adaptive Noise Canceller (ANC) [5]

QUALITY MEASURING PARAMETERS

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.

A. *SNR (Signal-to-Noise Ratio)*

SNR is the ratio of the amplitude of a desired analog or digital data signal to the amplitude of noise in a transmission channel at a specific point in time. SNR is typically expressed logarithmically in decibels (dB). SNR measures the quality of a transmission channel or an audio signal over a network channel. The greater the ratio, the easier it is to identify and subsequently isolate and eliminate the source of noise. A SNR of zero indicates that the desired signal is virtually indistinguishable from the unwanted noise. SNR also is abbreviated as *S/N*.
The signal to noise ratio is the ratio between the wanted signal and the unwanted background noise.

$$SNR = \frac{P_{signal}}{P_{noise}} \tag{1}$$

It is more usual to see a signal to noise ratio expressed in a logarithmic basis using decibels:

$$SNR_{dB} = 10\, log_{10}\left(\frac{P_{signal}}{P_{noise}}\right) \tag{2}$$

B. *PSNR (Peak Signal-to-Noise Ratio)*

Peak signal-to-noise ratio, often abbreviated PSNR, is the ratio between the maximum possible power of a signal and the power of corrupting noise. Because many signals have a very wide dynamic range, PSNR is usually expressed in terms of the logarithmic decibel scale. PSNR is most commonly used to measure the quality of reconstruction of lossy signal. Although a higher PSNR generally indicates that the reconstruction is of higher quality, in some cases it may not. One has to be extremely careful with the range of validity of this metric; it is only conclusively valid when it is used to compare results from the same codec (or codec type) and same content. PSNR is most easily defined via the mean squared error (MSE). The PSNR (in dB) is defined as:

ISSN: 2277-5528
Impact Factor: 4.015

$$PSNR = 10log_{10}\left(\frac{Max\_Signal^2}{MSE}\right)$$
4

$$PSNR = 20log_{10}\left(\frac{Max\_Signal}{\sqrt{MSE}}\right)$$
5

SIMULATION RESULTS ANALYSIS

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.

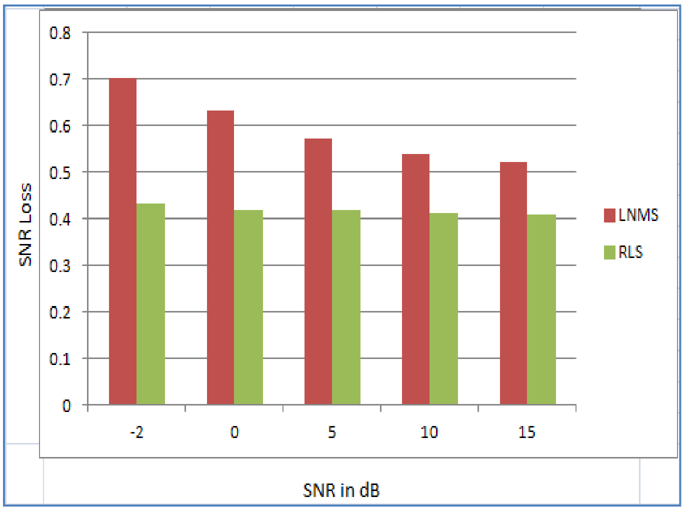

Fig. 3: Comparison of NLMS and RLS filters performance for AWGN Channel using Amplitude modulation

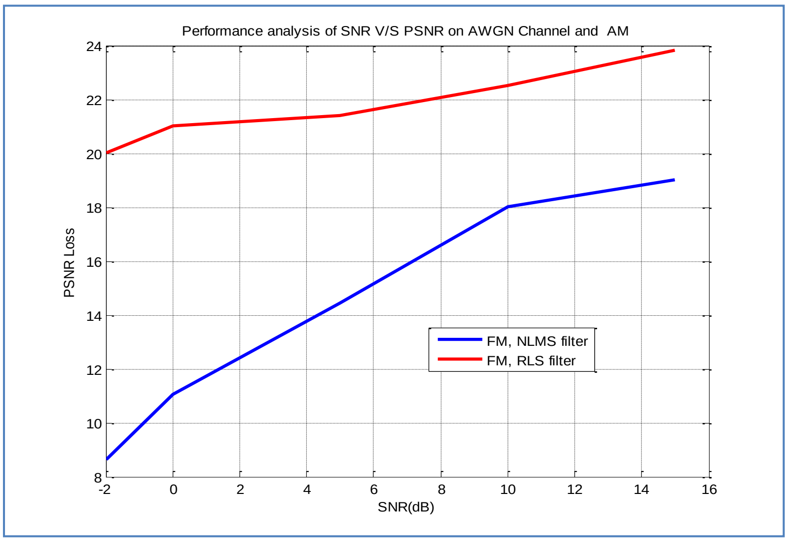

Fig.4: Performance of PSNR for AWGN Channel using AM

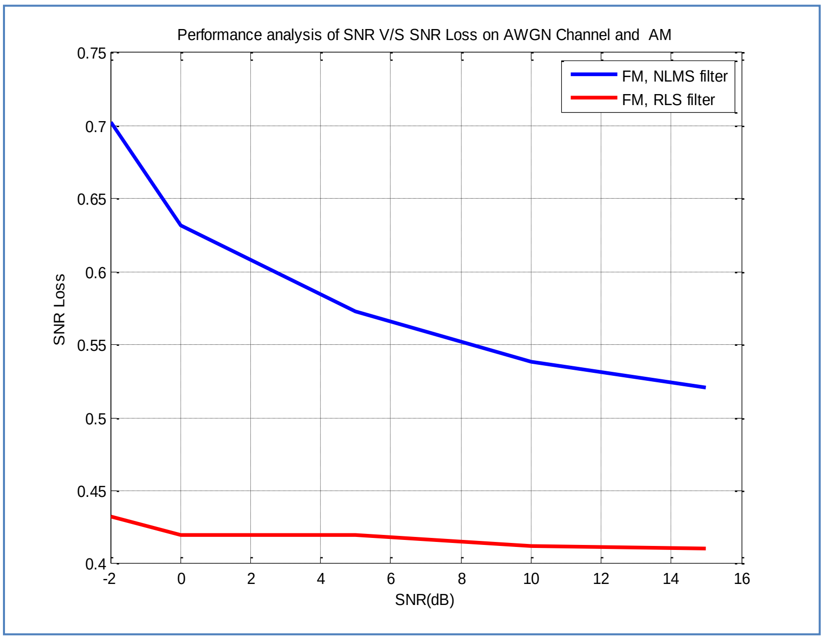

Fig. 5: Comparatively analysis of SNR V/S SNR Loss for AWGN Channel using Amplitude Modulation

CONCLUSION

The convergent rate of NLMS is the fastest but it is not good resolution. Adaptive filters and algorithm are described in overview emphasizing the applications. The RLS algorithm is two times faster than NLMS algorithm. Amount of Nose is reduced 10dB more than NLMS. The Mean Square Error of NLMS algorithm is 0.0002312, Mean Square Error of RLS algorithm is 0.0001359 at 15dB SNR.

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