ISSN: 2277-9655
Scientific Journal Impact Factor: 3.449
(ISRA), Impact Factor: 2.114

IJESRT
INTERNATIONAL JOURNAL OF ENGINEERING SCIENCES & RESEARCH TECHNOLOGY

Performance Comparison of Adaptive Algorithms for Noise Cancellation

Rahul Karma*, Umesh Gour
*M. Tech Scholar, SSSIST, Sehore, rahul.karma630@gmail.com, Bhopal (M.P.) India
Asst. Prof., Electronics Department, SSSIST, Sehore, umeshgour11@gmail.com, Bhopal (M.P.) India

Abstracts

Adaptive filtering is widely researched topic in the present era of communication. When the received signal is continuously corrupted by noise or interference where both the noise and signal changes continuously, then arises the need for adaptive filtering. This paper deals with adaptive noise cancellation which is an alternative technique for estimating the signals corrupted by additive noise or interference. An adaptive filter has the property that its frequency response is adjustable or modifiable automatically to improve its performance in accordance with some criterion, allowing the filter to adapt changes in the input signal characteristics. Moreover, adaptive filters have the capability of adaptively tracking the signal under non-stationary conditions it is widely used in many applications because of their self adjusting performance. In this paper we compared performances of adaptive algorithms (LMS, NLMS and SELMS) for noise cancellations. Performance analysis of all three algorithms is presented in term of MSE, SNR before filtering and after filtering, computational complexity and stability. The result of MATLAB simulation shows that our approach provides a good comparison without much degradation in its performance.

Keywords: LMS, NLMS, SELMS, MSE, SNR.

Introduction

Adaptive filtering is widely researched topic in the present era of communication. When the received signal is continuously corrupted by noise or interference where both the noise and signal changes continuously, then arises the need for adaptive filtering. This project deals with adaptive noise cancellation which is an alternative technique for estimating the signals corrupted by additive noise or interference. Its advantage lies in that , with no prior estimates of signal and noise , levels of noise rejection is attained which is difficult to achieve by other signal processing methods of removing noise. It needs two inputs – a primary input containing the corrupted signal & a reference input containing noise correlated in some unknown way with primary noise. The reference input is adaptively filtered & subtracted from the primary input to obtain desired signal. Adaptive filtering before subtraction allows the treatment of inputs that are deterministic or stochastic, stationary or time variable. The signal can be estimated by using proper adaptive algorithms. By using adaptive filter, the signal to noise ratio gets increased. The signal to noise ratio is generally expressed in decibels. Ideally, the SNR is greater than 0dB, indicating that the signal is higher than the noise.

The usual method of estimating a signal corrupted by additive noise is to pass it through a filter that tends to suppress the noise while leaving the signal relatively unchanged .i.e. direct filtering. Filters used for direct filtering can be either fixed or adaptive. Adaptive signal processing is used to process signals whose characteristics are unknown or changing. This can be done using adaptive filters. An adaptive filter has the property that its frequency response is adjustable or modifiable automatically to improve its performance in accordance with some criterion, allowing the filter to adapt to changes in the input signal characteristics.

Theoretical background

A. Adaptive Noise Cancellation

An Active (also called as Adaptive) noise cancellation (ANC) is a technique that effectively attenuates low frequencies unwanted noise where as passive methods are either ineffective or tends to be very expensive or bulky. An ANC system is based on a destructive interference of an anti-noise, which have equal amplitude and opposite phase replica of primary unwanted noise. Following the superposition principle, the result is noise free original sound.

ANC systems are distinguished by their different goals that lead to different architectures. If all ambient sound shall be reduced, a feedback system with its simpler architecture may be used. If, as in our case, single sources of unwanted sound shall be compensated, a feed forward system is required. A feed forward system as shown in figure 1 is characterized by two audio inputs per channel: one reference signal input for the sound to

ISSN: 2277-9655
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be removed, and second error input for the sound after the compensation.

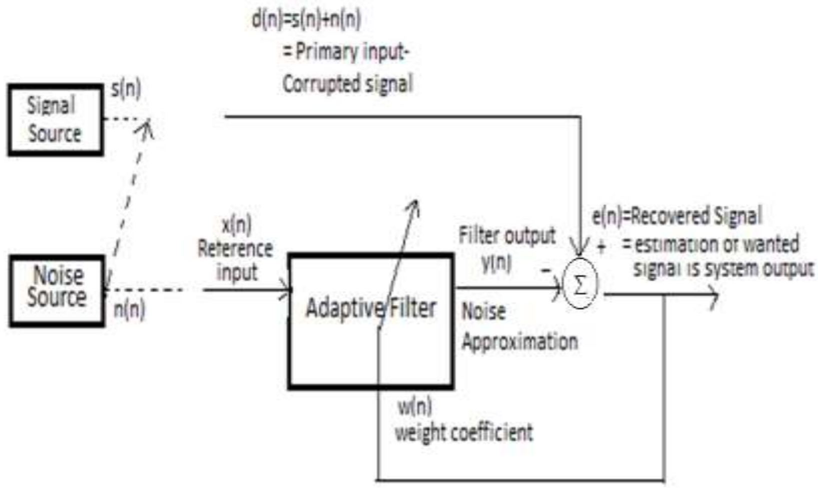

Figure 1: Adaptive Noise Canceller

An adaptive FIR feed forward system is shown in simple way in Figure 1 for the selective cancellation of disturbing noise without affecting other sounds [5]. It is dual input system. The first inputs is primary signal d (n) which is wanted signal (say s (n)) corrupted by noise (say n (n)). The second input is reference signal x(n) can be interfacing noise supposed to be uncorre1ated with the wanted signal but correlated with noise affecting original signal in an unknown way. The filter output signal y (n) is an estimate of the noise signal with inverted sign. This signal and the primary signal are superposed, so that the noise signal is cancelled and error signal e (n) is the result of this superposition which constitutes the overall system output. The adaptive filtering operation achieved the best results when system output is noise free. This goal is achieved by minimizing the mean square of the error signal [3].

B. Adaptive Filter

Adaptive filter is generally defined as a filter whose characteristics can be modified to achieve some objective and is usually assumed to accomplish this adaptation automatically, without the need for substantial intervention by user. Here, an adaptive filter is needed that is capable of altering its impulse response as required by the particular application. Figure 2 shows the general structure of an adaptive filter.

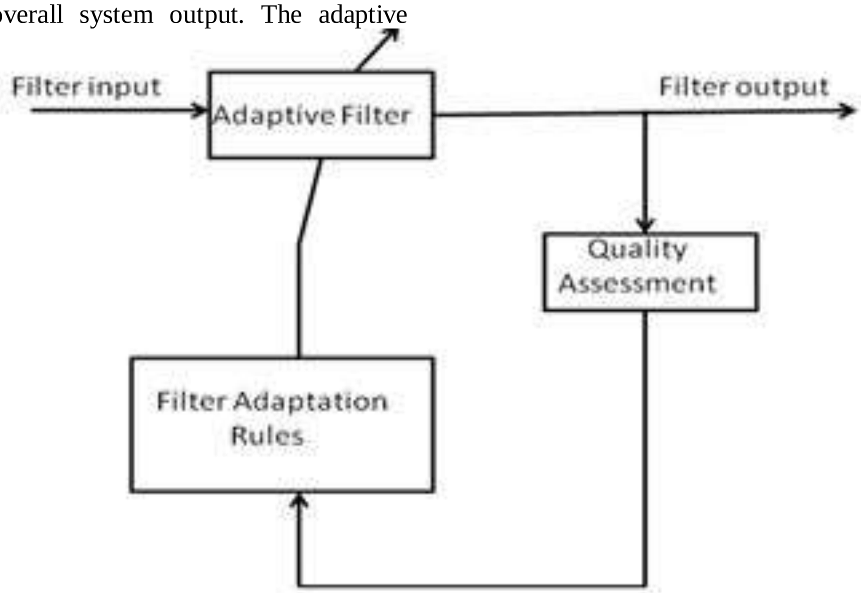

Figure 2: Adaptive filter structure

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Let's take a look at some of the terminology that will be used when we talk about adaptive filters.

Filter Structure – This is the implementation of the filtering algorithm. It is set by the designer of the filter, and may be something like a Direct Form implementation of an FIR filter. This block computes the filter output based on the input. The filter coefficients can be updated by the Adaptive Algorithm.
Criterion of Performance – This block looks at the output of the filter and compares it with some other signal. This other signal is the desired output of the filter. If we know what the desired response is, we can compare it to the actual response and then indicate to the Adaptive Algorithm that something needs to be changed. From the example in the introduction, the criterion of performance would detect the noise at 450Hz and tell the Adaptive Algorithm that it needs to change the filter's cutoff frequency.
Adaptive Algorithm – This is the main part of an adaptive filter. This algorithm decides how to change the filter coefficients in response to the signal given by the criterion of performance. This is the most difficult part of an adaptive filter to design.

Recent scenario

In 2012, Jashvir *et al.* implemented in cancellation of noise on speech signal using two adaptive algorithm LMS algorithm and NLMS algorithm. Choose the algorithms that provide efficient performance with less computational complexity. This paper has described an application in which the use of an LMS and NLMS adaptive filter is particularly appropriate.

In 2011, Ma Shengqian & Xu Guowei has presented an improved LMS algorithm of variable step length based on Kwong least mean-square algorithm. The algorithm proposed is used for an adaptive noise canceller. The sinusoidal signal and audio signal with Gaussian white noise were simulated in noise cancellation system on the MATLAB platform. It has fast convergence and good noise suppression ability than traditional algorithms. This paper presents an improved variable step size LMS algorithm, to study the application in the ANC, analysis the impact of the parameters sensitivity and demonstrate the effectiveness of the algorithm. Finally a hardware system for the ANC is constructed and to achieve noise cancellation of audio signal.

In 2010, Sayed. A. Hadei has given a new approach for noise cancellation in speech enhancement using the two new adaptive filtering algorithms named fast affine projection algorithm and fast Euclidean direction search algorithms for attenuating noise in speech signals. The simulation results demonstrate the good performance of the two new algorithms in attenuating the noise. The simulation results were compared with the classical adaptive filters, such as LMS, NLMS, Recursive Least Squares (RLS) and Fast Affine Projection algorithms, for attenuating noise in speech signals. In each algorithm the time evolution of filter taps, mean square error, and the output of filter were presented. The simulation results show that the convergence rate of these algorithms is comparable with the RLS algorithm.

In 2009, Ioana Homana has proposed an echo canceller based on a system identification scheme with adaptive algorithms is presented. The algorithms consider FIR filters with the taps chosen to minimize an error signal derived from the system according to a stochastic gradient-based method. The classical LMS and its variants adaptive filters are discussed and simulated in MATLAB. The echo was simulated by adding several delayed and attenuated replica to a vocal input. The comparison between the algorithms was made taking into account the mean-square error (MSE) and average attenuation (AV), as well as the required number of multiplications.

In 2005, Antonio Di Stefano has presented a hardware implementation of an Adaptive Noise Canceller. It has been synthesized within an FPGA, using a modified version of the Least Mean Square (LMS) error algorithm. The results obtained so far show a significant decrease of the required gate count when compared with a standard LMS implementation, while increasing the ANC bandwidth and SNR. This novel Adaptive Noise Canceller is then useful for enhancing the SNR data collected from sensors.

Proposed algorithm

Implementation of adaptive filtering algorithms in MATLAB is presented in following flowchart.

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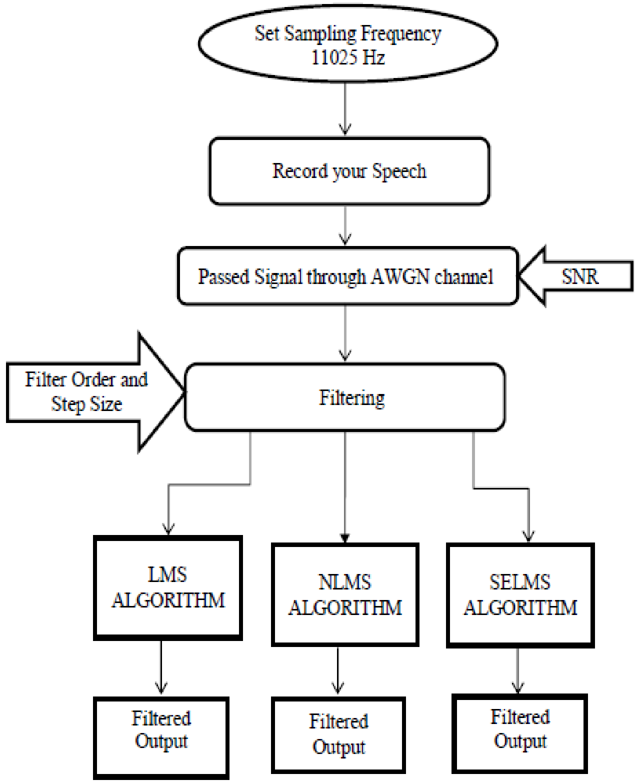

Figure 3: Proposed Algorithm

Results & validation

The adaptive filter is an 8th order FIR filter. The step size was set to 0.02. The MSE shows that as the algorithm progresses the average value of the cost function decreases. The step size μ control the performance of the algorithm, if μ is too large the convergence speed is fast but filtering is not proper, if μ is too small the filter gives slow response, hence the selection of proper value of step-size for specific application is prominent to get good results.

The filter order also affect the performance of a noise cancellation system.MSE is calculated for three algorithms. NLMS has good performance as compared to LMS and SELMS.

Results of LMS

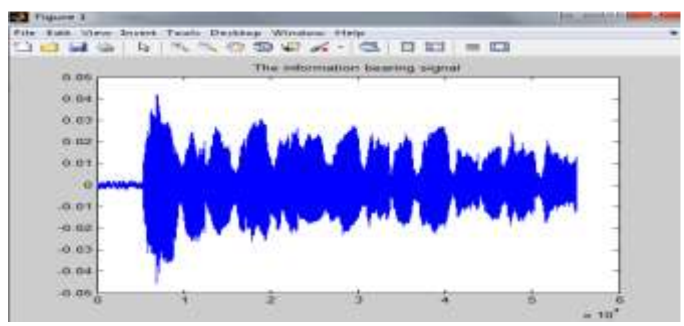

Figure 4: The information bearing signal (speech signal)

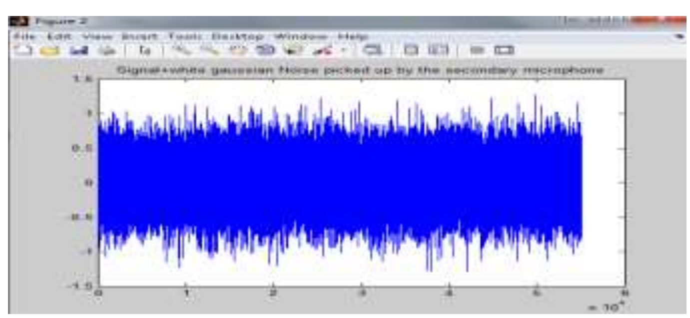

Figure 5: Signal + White Gaussian Noise

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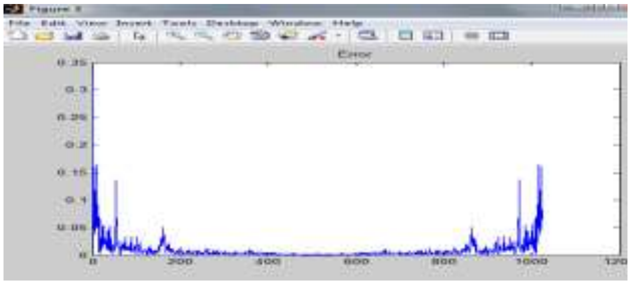

Figure 6: Error Signal

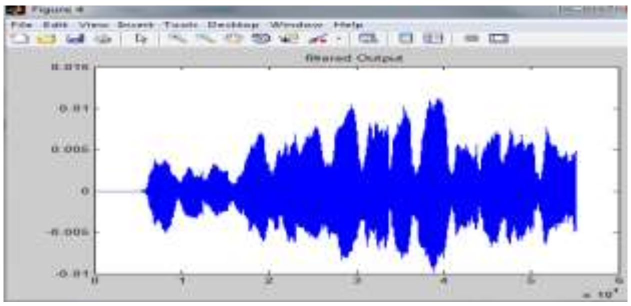

Figure 7: Filter Output

Results of NLMS Algorithm

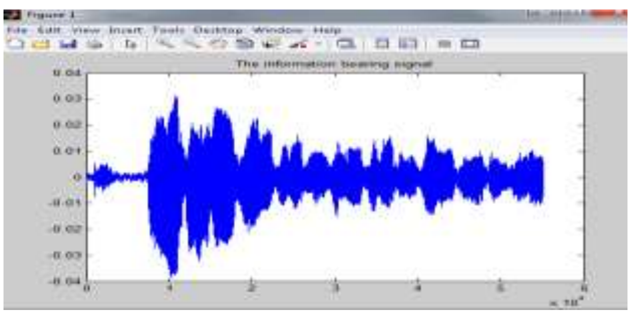

Figure 8: Speech Signal

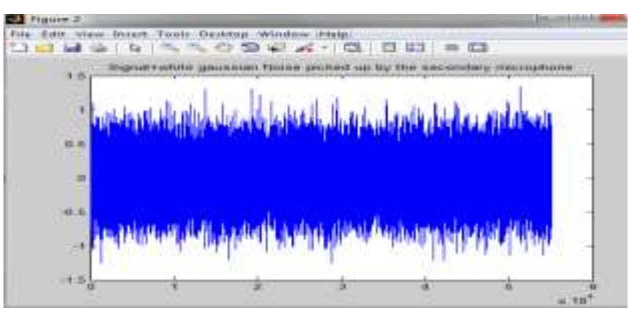

Figure 9: Signal + White Gaussian Noise

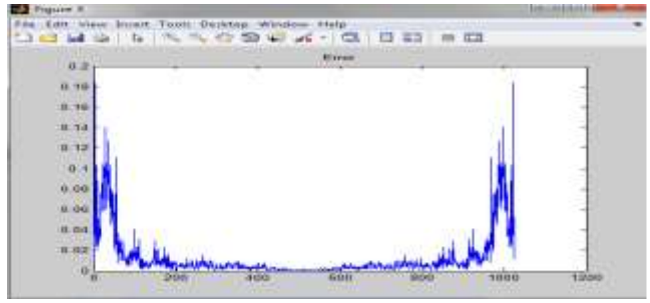

Figure 10: Error Signal

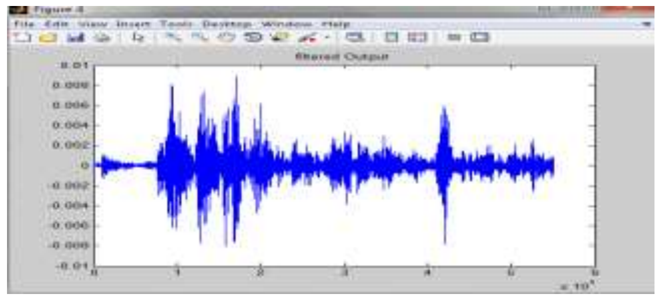

Figure 11: The Filter Output

Results of SELMS Algorithm

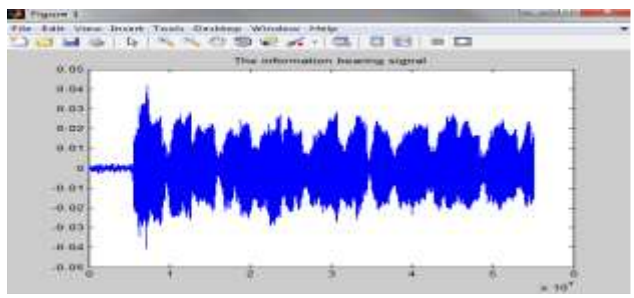

Figure 12: Speech Signal

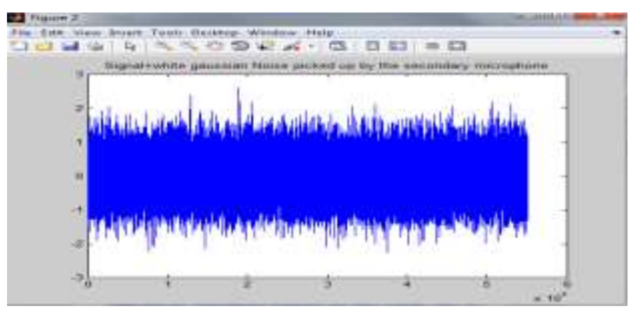

Figure 13: Signal + White Gaussian Noise

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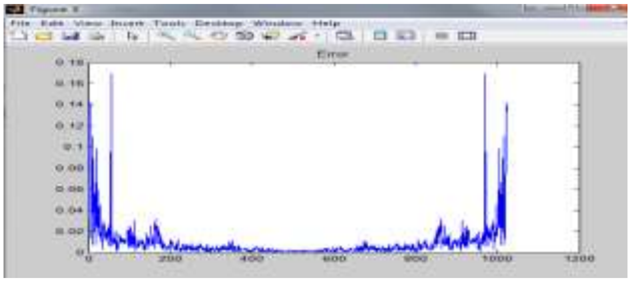

Figure 14: Error Signal

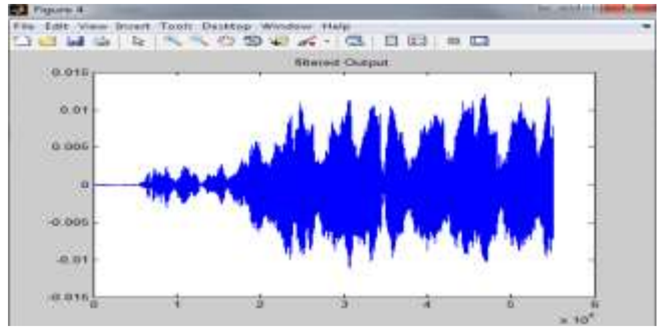

Figure 15: Filter Output

In table 1 performance analysis of all three algorithms is presented in term of MSE, SNR before filtering and after filtering, computational complexity and stability. It is clear from the table, the computational complexity and stability problems increases in an algorithm as we try to reduce the mean squared error.

Sr. No.	Algorithm	Filter Order	μ (Step Size)	MSE	SNR1 (Before Filtering)	SNR2 (After Filtering)	SNR Improvement	Comments
1.	LMS	8	0.02	5.4493×10^{-5}	33.0421dB	47.6267dB	14.5846 dB.	This requires prior knowledge of the input signal which is not feasible for the noise cancellation system.
2.	NLMS	8	0.02	4.4906×10^{-5}	33.5075dB	51.6746dB	18.1671 dB.	Shows very good SNR allows stable performance with non-stationary signals. This is the obvious choice for real time implementation.
3.	SELMS	8	0.02	5.17×10^{-5}	36.6024dB	45.1137dB	08.5113 dB.	Comparing NLMS algorithm displays poor performance, possibly due to non stationary nature of speech signal. Not considered for real time implementation.

Table 1: Performance Analysis

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Conclusion and outlook

In this paper we presented Performance comparison of adaptive algorithms for noise cancellation .It can be seen that when considering the MSE and SNR values and the step size operations for each algorithm, the NLMS algorithm is the obvious choice for the real time adaptive noise cancellation system. Additionally, it does not require a prior knowledge of the signal values to ensure stability. NLMS is the favorable choice for most of the industries due less computational complexity and fair amount of noise reduction.

The algorithms studied in this paper perform best under purely non stationary signal conditions. Strictly speaking speech signal falls into this category. Further work could be done in developing techniques specifically designed for stationary signals.

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