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COMPARATIVE PERFORMANCE ANALYSIS OF ADAPTIVE ALGORITHMS FOR ARTIFACTS REMOVAL FROM EEG SIGNAL

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Abstract—

The objective of this paper is to de-noise and analyze the EEG signal which is contaminated by artifact signals by adaptive filtering techniques in software MATLAB in simulink. Simulation results of adaptive algorithms for filtering gives better results as compared with hardware processing of EEG data. There are three adaptive algorithms namely Least Mean Squared (LMS),Normalized Least Mean Squared (NLMS) algorithms and Recursive least square (RLS) and to compare their relative performance with a EEG signal. Other parameters such as stability, convergence rate, MSE,PER are calculated for two algorithms in this paper

Keywords- EEG, Ocular Artifacts (EOG),Myogram Artifacts(EMG), Least Mean Squared, Normalized Least Mean Squared, MSE,SNR, convergence, stability EEG signal

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I. Introduction

The Adaptive filtering constitutes one of the core technology in the field of digital signal processing and finds numerous application in the areas of science and technology viz. echo cancellation, channel equalization, adaptive noise cancellation, adaptive beam-forming, biomedical signal processing etc. Noise problems in the environment have gained attention due to the tremendous growth in upcoming technologies which gives spurious outcomes like noisy engines, heavy machinery, high electromagnetic radiation devices and other noise sources. Therefore, the problem of controlling the noise level in the area of signal processing has become the focus of a vast amount of research over the years. In this particular work an attempt has been made to explore the adaptive filtering techniques for noise cancellation using Least Mean Square (LMS), Normalized Least Mean Square (NLMS) algorithms. The mentioned algorithms have been simulated in MATLAB and compared for evaluating the best performance in terms of Mean Squared Error (MSE), Signal to noise ratio (SNR), convergence rate, and Percentage noise removal (PER), Power spectral density (PSD), computational complexity and stability. In the specific example of tone signal, LMS has shown low convergence rate, with low computational complexity while NLMS provides a trade-off in convergence rate and computational complexity which makes it more suitable for hardware implementation. The results have also been compared with the LMS algorithm to prove the superiority of NLMS algorithm.

The discrete adaptive filter process the reference signal n1(n) to produce the output signal y(n) by a convolution with filter's weight, w(n). Then, a desired signal reference signal d(n) is compared with the output y(n) to obtain an estimation error e(n).The objective here is to minimize the error signal e(n) which is equivalent to minimize the error signal e(n) which is equivalent to minimize d(n) – y(n).This error signal is used to incrementally adjust the filter's weights for next time instant. The algorithms for weight update are: wiener filter, least-mean square (LMS) algorithm, Normalized-least-mean square (NLMS) algorithm, recursive-least-square (RLS) algorithm, and Kalman filter. Rest of the paper is organized as follows: Section II is about related work done on the adaptive filter. Section III is explains methodology of the paper. Section IV is the basic of adaptive filter algorithms. Section V is about simulation diagram and the result discussed in Section VI. Conclusion is discussed in Section VII.

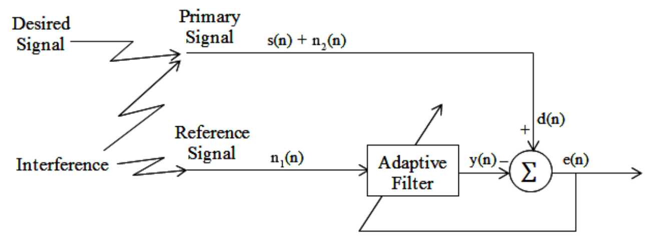

Figure 1 Adaptive filter for noise cancellation [1]

II. RELATED WORK

Jan vanus [3] describes a proposition of the method for optimal adjustment parameters of the adaptive filter with LMS algorithm. [4] Results reveal that the LMS and NLMS are better thanthe other adaptive filters. However, the obtained SNR ofreconstructed signal of LMS filter is lower than that of theNLMS filter. So, the NLMS adaptive filter is more appreciable for removing the power line interference from the EEG signal.

III. METHODOLOGY

It is difficult to apply filters with fixed coefficients to reduce Biomedical Signal noises, because human behavior is not exact known depending on the time. The goal for EEG signal enhancement is to separate the valid signal components from the undesired artifacts. The basic idea for the adaptive filter is to predict the amount of noise in the primary signal and then subtract that noise from it. The prediction is based on filtering the reference signal, which contains a solid reference of the noise present in the primary signal.The noise in the reference signal is filtered to compensate for the amplitude, phase and time delay and then subtracted from the primary signal. This filtered noise is the system's prediction of the noise portion of the primary signal. The resulting signal is called error signal and it presents the output of the system. Ideally, the resulting error signal would be only the desired portion of the primary signal. Two types of adaptive filter algorithms are used. Least mean square (LMS) and Normalized Least mean square (NLMS) algorithms are used to de-noise the EEG signal by using Simulink in MATLAB.

IV. ADAPTIVE FILTER ALGORITHMS

A. *Least Mean Square Algorithm*

First, The filter tap weights of the adaptive filter are updated according to the following formula: $w(n + 1) = w(n) + 2\mu e(n)x(n)$;

Where x(n) is the input vector of time delayed input values, and is given by

$x(n) = [x(n)\ x(n - 1)\ x(n - 2)....x(n - N + 1)]T$

&$w(n) = [w0(n)\ w1(n)\ w2(n)....w\ N - 1\ (n)]T$

Represents the coefficients of the adaptive FIR filter tap weight vector at time n and μ is known as the step size parameter and is a small positive constant.

B. *Normalized Least Mean Square Algorithm*

When the convergence factor μ is large, the algorithm experiences a gradient noise amplification problem. In order to solve this difficulty we can use the NLMS algorithm. The correction applied to the weight vector w(n) at iteration n+1 is "normalized" with respect to the squared Euclidian norm of the input vector x(n) at iteration n. Convergence factor μ-

$\mu(n) = \alpha / (c + \|x(n)\|^2)$;

Where c is constant less than 1, and α is NLMS adaption constant and should satisfy $0<\alpha<2$. It obeys principle of minimum disturbance.

V. SIMULATION DIAGRAM

In this we required the recorded EEG signal, and then it will be given as an input to the simulink model. Recorded EEG Noise also taken and then it will be added with original signal. The recorded noise taken is EMG. Added signal is subtracted from the output of the adaptive filter and then given to the display.

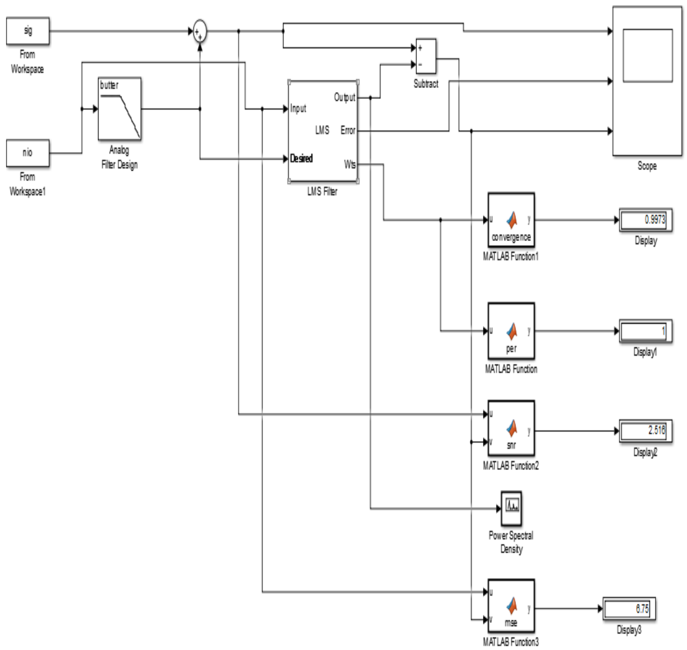

Figure 2 Simulink model for LMS filter algorithm

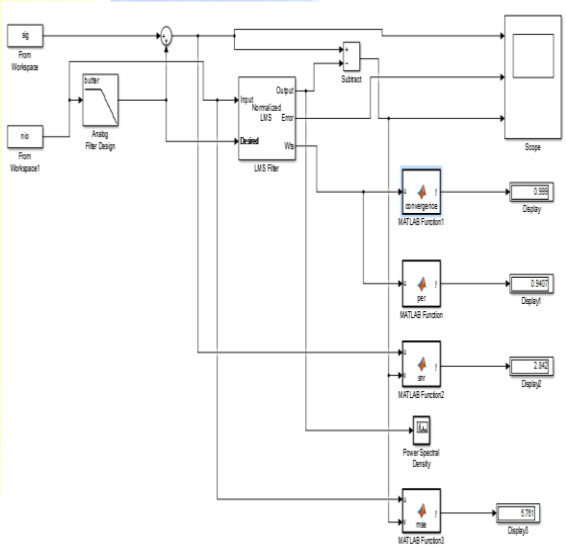

Figure 3 Simulink model for NLMS filter algorithm

The figure 2 shows Simulink model of LMS filter algorithm. And the figure 3 shows Simulink model of NLMS filter algorithm.

VI. Results

The Simulation is done to remove the noise of an EEG signal which is corrupted by various types of interferences & distortions. Figure 4 shows de-noise EEG signal by LMS filter algorithm. Figure 5 shows de-noise EEG signal by NLMS filter algorithm.

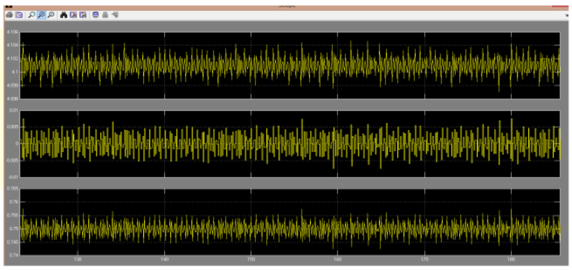

Figure 4 De-noise EEG signal by LMS filter a) corrupted EEG b) error signal c)de-noised EEG signal

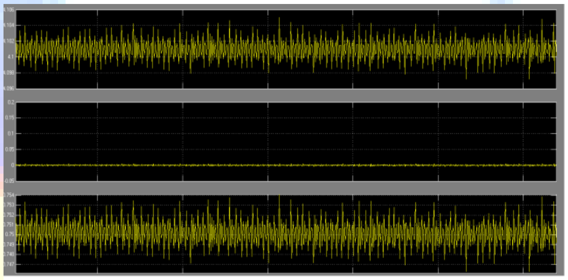

Figure 5 De-noise EEG signal by NLMS filter a) corrupted EEG b) errorsignal c)de-noised EEG signal

Figure 6 shows power spectral density graph for LMS. And figure 7 shows power spectral density graph for NLMS.

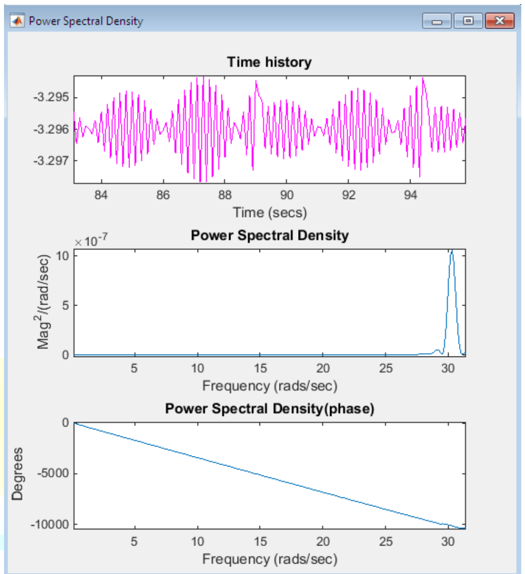

Figure 6 Power Spectral Density graph for LMS filter

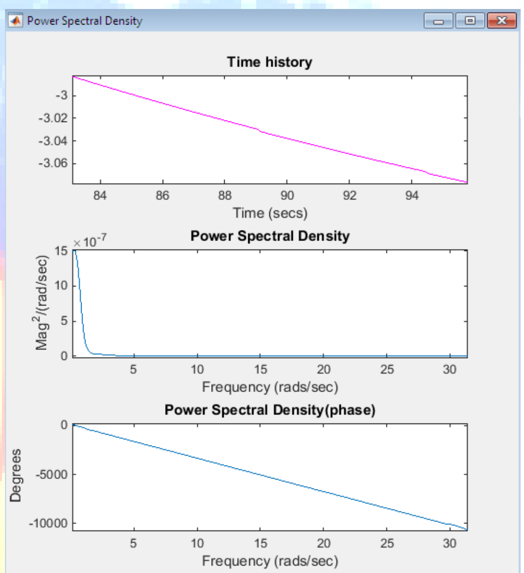

Figure 7 Power Spectral Density graph for LMS filter

Table: 1 Comparison of LMS and NLMS algorithm based on Statistical Parameters

Sr. No.	Algorithms	SNR	PER	MSE	Con-Speed	PSD
1	LMS	2.511	0.9995	6.788	0.997	10*10^-7
2	NLMS	2.542	0.9407	6.761	0.999	15*10^-7

VII. Conclusion

The simulation of LMS & NLMS MATLAB simulink for de-noising EEG signal has been done successfully and the results are compared in the terms of SNR, PER, MSE, Convergence speed, PSD. During the simulation for NLMS algorithm, we got the SNR Improvement of 2.53dB. During the simulation for LMS algorithm, we got the SNR improvement of 2.43dB. LMS is simple algorithm which is used for less complexity. NLMS gives better results in terms of SNR so it is more preferable. Convergence speed or rate of NLMS is more as compared with LMS algorithm Also Percentage of error rate and mean square error is low for NLMS compared to that of LMS. By observing table 1, we got high convergence rate and power spectral density in the NLMS filter. Hence we conclude that NLMS algorithm is better than that of LMS algorithm.

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