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A Novel Approach of Removal of Ocular Artifacts from Electro Encephalogram using Adaptive Filtering

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Abstract: **Artifacts in EEG signals are caused due to various factors like electro-oculogram (EOG) and line interference. Existing regression based methods for removing EOG artifacts require various procedures for pre-processing and calibration that are inconvenient and time-consuming. This paper proposes a method for obtaining a clear EEG by removing ocular artifacts, which are due to eye movement based on adaptive filtering. This method uses two reference inputs vertical EOG and horizontal EOG signals, which are separately recorded. Each reference input is first processed by a finite impulse response filter of length M=3 and then subtracted from the original EEG. The method is implemented by LMS algorithm with a forgetting factor λ=0.9999 for tracking the non-stationary portion of the EOG signals. Results performed on experimental data show that this method is stable, converges fast, easy to implement and is suitable for online removal of EOG artifacts.**

Keywords: Electro Encephalogram(EEG),Electro-Oculogram (EOG), LMS algorithm.

1. INTRODUCTION

1.1 Problem Description

EEG signal is one of the oldest measures of brain activity that has been vastly used for clinical diagnoses and biomedical researches. The medical monitoring devices are more sensitive for the biomedical signal recording and need more accurate results for every diagnosis. It is complicated to get accurate result for every biomedical signal's recording while patient is diagnosed by medical monitoring equipments such as ECG, EEG and EMG.

However, EEG signals are highly contaminated with various artifacts, both from the subject and from equipment interferences. Among these various kinds of artifacts, ocular noise which is due to movement of eye balls and eye blinks is the most important one. Since many applications such as BCI require online and real-time processing of EEG signal, it is ideal if the removal of artifacts is performed in an online fashion.EOG contamination is most serious problem in EEG based analysis.

1.2 Aims and Objectives the paper

- The fundamental purpose of this paper is to remove ocular artifacts from EEG by using adaptive filters based on least mean square (LMS) algorithm optimizing criterion. FIR filters are used in the design of LMS algorithm.
- To simulate these adaptive filtering algorithms using Matlab.

2. INTRODUCTION TO EEG AND EOG WITH INTERFERENCE

2.1 General Overview

EEG signals however especially those recorded from frontal channels often contain strong electro-oculogram (EOG) artifacts produced by eye movements. Existing regression based methods for removing EOG artifacts require preprocessing steps and calibration which are inconvenient and time consuming. This paper describes a method for removing ocular artifacts based on adaptive filtering which uses separately recorded vertical EOG and horizontal EOG signals as two reference inputs. Each reference input is first processed by a finite impulse response filter of length M=3 and then subtracted from the original EEG. The method is implemented by a recursive least squares algorithm that includes a forgetting factor λ=0.9999 to track the non-stationary portion of the EOG signals.

Results from experimental data demonstrate that the method is easy to implement and stable, converges fast and is suitable for on-line removal of EOG artifacts. The first three coefficients were significantly large .The statistical analysis of electrical recordings of the brain activity by an Electroencephalogram is a major problem in Neuroscience. Cerebral signals have several origins that lead to the complexity of their identification. Therefore the noise removal is of the prime necessity to make easier data interpretation and representation and to recover the signal that matches Eye. The EYE forms an electric dipole, where the cornea is positive and the retina is negative. When the eye moves (saccade, blink or other movements), the electric field around the eye changes, producing an electrical signal known as the electro-oculogram (EOG). As this signal propagates over the scalp, it appears in the recorded electro-encephalogram (EEG) as noise or artifacts that present serious problems in EEG interpretation and analysis[2].

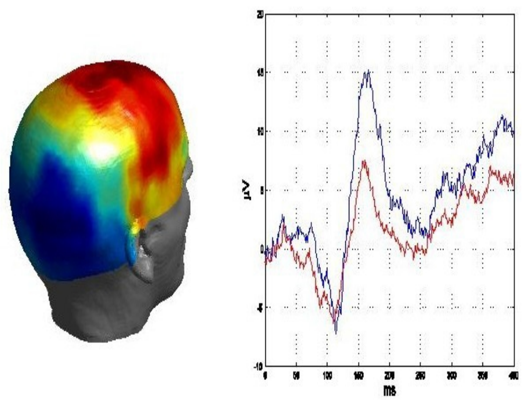

Figure .1 Surface indication with signals

To correct or remove ocular artifacts from EEG many regression-based techniques have been proposed, including simple time-domain regression, multiple-leg time-domain regression and regression in the frequency domain).In all these regression-based approaches, calibration trials are first conducted to determine the transfer coefficients between the EOG channels and each of the EEG channels These coefficients are then used later in the 'correction phase' to estimate the EOG component in the EEG recording for removal by subtraction.

3. ADAPTIVE FILTERS

3.1 Introduction of Adaptive Filter

The adaptive filters are much famous due to their economical quality, fast processing, their short period of time adaptation and residual error is small after adaptation. Adaptive filtering is the most important technique which is used in number of biomedical applications.

Adaptive filtering is properly used due to its esteemed knowledge of signal make up related to the adaptive processing. Literally, the word 'adaptive' means to adjust with other environment (system) by having the same response as the system itself to some phenomenon which is taking place in its surroundings or technically the system which tries to adjust its parameters, depending upon the other system's behavior and its surrounding. The systems which carry out its functionality after undergoing the process of adaptation is called filter. The term 'filter' means to take the unnecessary particles (frequency component) from its input signal and process them to generate required output under certain specific rules.

3.2 Adaptive Filtering

The adaptive filter is a filter which self adjusts its transfer function according to an optimizing algorithm. The adaption is based on the error signal between the primary input signal and the filter output. Widely used optimizing criterion is LMS algorithm.

The basic principal of adaptive filter can be understood by understanding the adaptive filtering which is showed in the figure 3. The error signal e (n) can be generated by the output of the programmable, variable-coefficient digital filter subtracted from a reference signal y (n).

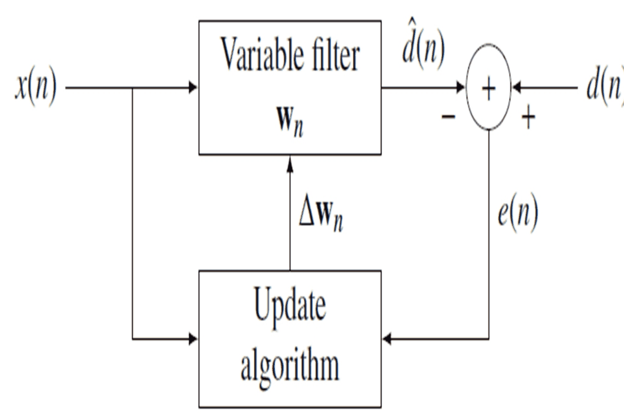

Figure.2 Principle of an Adaptive Filter

The adaptive filter can be classified in the following areas

- The optimization criterion
- The algorithm for coefficient updating
- The programmable filter structure
- The type of signal processed

The choice of filter structure and adaptation algorithm is important for the design of adaptive filter, the structure can be non-recursive or recursive.

The design of digital filter requires the approved specification with fixed coefficients. If this specification is time varying or not accessible then this problem can be manipulated by digital filter with adaptive coefficients, which is known as adaptive filter.

3.3 Performance, Stability and Robustness of the Adaptive Algorithm

The performance of the adaptive algorithm is important for all systems, it is also essential how adaptive system is functioning. For any application the adaptive algorithm provides competent performance evaluations for the structures of various filters and adaptive algorithm. The LMS algorithm is the most popular adaptive algorithm and its performance is dependent on the filter order, signal condition and convergence parameter (μ).The adaptive system is used for the solution for any practical problem the question appears about the stability of adaptive algorithm whether or not the algorithm is stable. In general the adaptive filters based on FIR structure are naturally stable.

To satisfy the robustness of the adaptive algorithm the value of step size μ needs to be small. Robustness is an important criterion which is difficult to measure in a quantitative approach. The satisfaction for the robustness of the adaptive algorithm can be gained by the removal of external noise.

3.4. Principle of removing EOG artifacts by adaptive filtering:

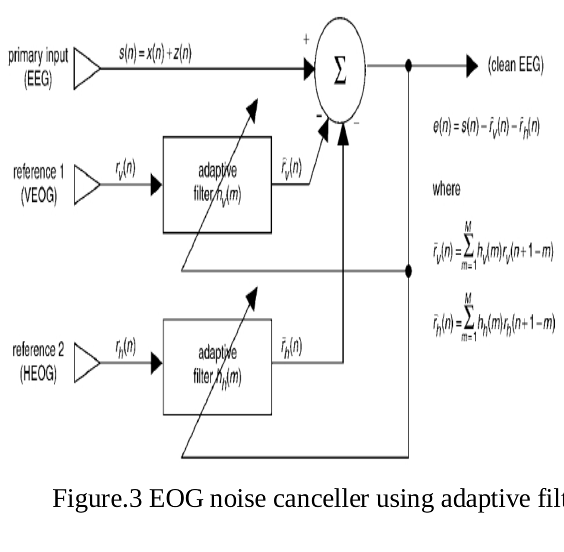

Figure.3 EOG noise canceller using adaptive filter

The block diagram of the EOG noise canceller used in this application is shown in Figure 4. The primary input to the system is the EEG signal s (n).This signal is modeled as a mixture of a true EEG x (n) and a noise component z (n). VEOG (r_v(n)) and HEOG (r_h(n)) are the two reference inputs. r_v(n) and r_h(n) are correlated with the noise component z(n) in the primary input. h_v(m) and h_h(m) represent two finite impulse response (FIR) filters of length M (the two filters can have different lengths).The output from the noise canceller e(n) is clean EEG.

The goal of noise canceller is to produce an output signal e (n) that is as close to x(n) as possible, by adjusting the filter coefficients h_v(m) and h_h(m).In this application we have chosen the LMS algorithm due to its stability and fast convergence.

4. LMS ALGORITHM

LMS algorithm was developed by Widrow and Hoff in 1959.It is widely used in various applications of adaptive filtering. The main features that attracted the use of the LMS algorithm are low computational complexity, proof of convergence in stationary environments and stable behavior when implemented with finite precision The task of LMS algorithm is to estimate the transfer function of the filter. The result of the signal distortion is calculated by convolution .In this case d is the echo and h is the transfer function of the hybrid.

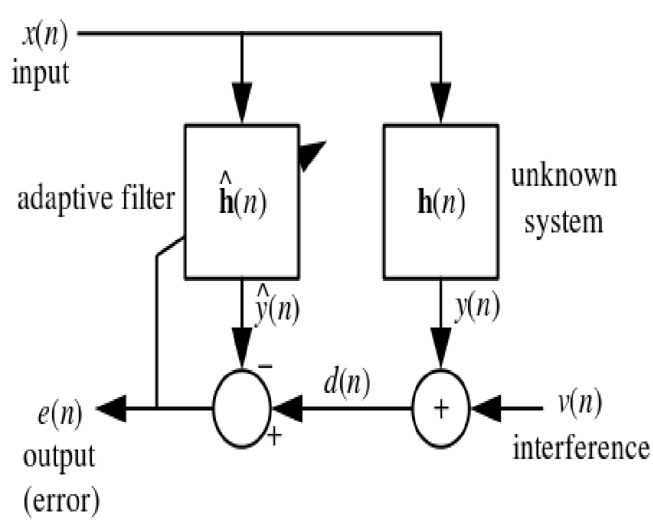

Figure.4 LMS algorithm

The adaptive algorithm tries to create a filter w. The transfer function in turn is used for calculating an estimate of the echo. The echo estimate is denoted by y.

The signals are added so that the output signal from the algorithm is

$$e=d-y \quad (1)$$

where 'e' denotes the error signal.

The error signal and the input signal x are used for the estimation of the filter coefficient vector w. One of the main problems associated with choosing filter weight is that the path h is not stationary. Therefore, the filter weights must be updated frequently so that filter is a FIR filter with the form theadjustment to the variations can be performed. The filter is a FIR filter with the form

$$W= W_0(n)+W_1(n)Z^{-1}+\ldots+W_{N-1}(n)Z^{-(N-1)} \quad (2)$$

The LMS algorithm is a type of adaptive filter known as stochastic gradient based algorithm as it utilizes the gradient vector of the filter tap weights to converge on the optimal Weiner solution. With each iteration of the LMS algorithm, the filter tap weights of the adaptive filter are updated according to the following formula

$$W(n+1)=w(n)+2\mu e(n)x(n) \quad (3)$$

Here x(n) is the input vector of time delayed input values, $\mathbf{x}(n) = [x(n)\ x(n-1)\ x(n-2)\ \ldots..x(n-N+1)]^T$.

The vector $\mathbf{w}(n) = [w0(n)\ w1(n)\ w2(n)\ ..\ wN-1(n)]^T$ represents the coefficients of the adaptive FIR filter tap weight vector at time n. The parameter μ is known as the step size parameter and is a small positive constant. This step size parameter controls the influence of the updating factor. Selection of a suitable value for μ is imperative to the performance of the LMS algorithm, if the value is too small the time the adaptive filter takes to converge on the optimal solution will be too long. If μ is too large the adaptive filter becomes unstable and its output diverges.

4.1 Implementation of the LMS algorithm.

Each iteration of the LMS algorithm requires 3 distinct steps in this order:

1. The output of the FIR filter, y(n) is calculated using equation 4.

$$y(n)=\sum_{i=0}^{N-1} w(n)x(n-i)=\mathbf{w}^T(n)\mathbf{x}(n) \quad (4)$$

2. The value of the error estimation is calculated using equation 5.

$$e(n)=d(n)-y(n) \quad (5)$$

3.The tap weights of the FIR vector are updated in preparation for the next iteration, by equation (6).

$$W(n+1)=W(n)+2\mu e(n)X(n) \quad (6).$$

For each iteration the LMS algorithm requires 2N additions and 2N+1 multiplications (N for calculating the output, y(n), one for 2μe(n) and an additional N for the scalar by vector multiplication).

5. SIMULATIONS RESULTS

5.1 Main Objective

The goal was to implement and analyze different noise removal techniques in EEG signal. Removal of ocular signals from EEG signal and to apply the adaptive LMS algorithm for removing of interference and its harmonic from signals that has been corrupted and the original information contained by unwanted interferences.

5.2 Simulations Results

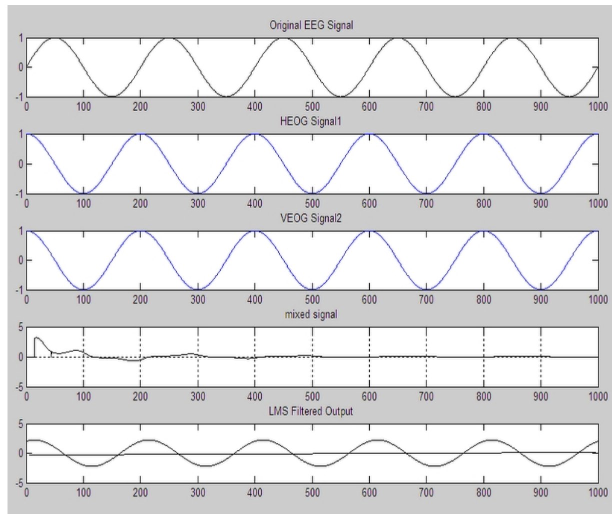

Figure.5 EEG Signal Simulation-1

FilterTap=16;

μ= 0.005;

EEG Signal Simulation-1 has been shown in the figure 5, the value of the filter tap and μ has been taken 16 and 0.005 respectively to generate the graph. The figure shows different plots, first, second, third, fourth and fifth plots are taken as to be EEG signal, HEOG and VEOG signal, mixed signal, and filtered output signal respectively. The EEG signal and noise signal are generated and then mixed together in third plot of the figure. The fifth plot shows the LMS filtered output of the mixed signal which is nearly same to the input EEG signal.

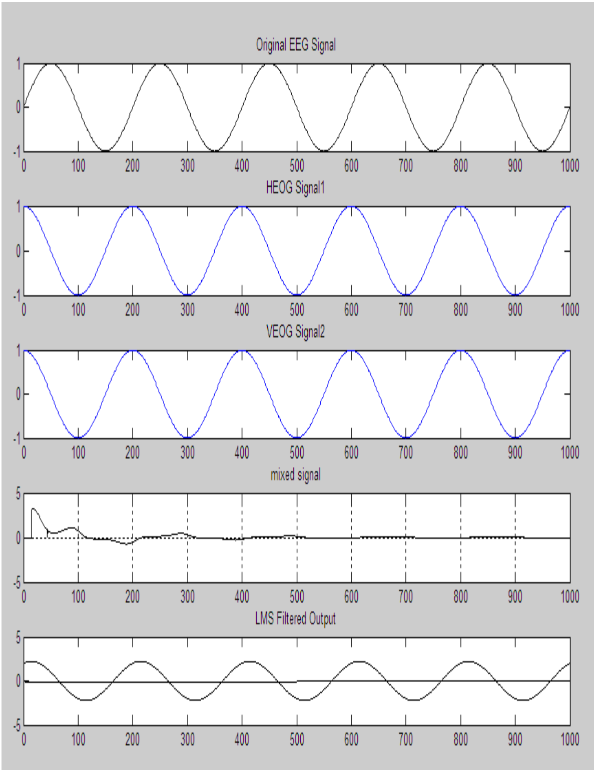

Figure.6 EEG Signal Simulation-2

FilterTap=32;
μ = 0.005;

In figure 6 EEG Signal Simulation-2, the value of filter tap and μ are taken as 32 and 0.005, which shows different results as compared to the EEG Signal Simulation-1. It is concluded that the rate of convergence is changed by changing the value of μ.

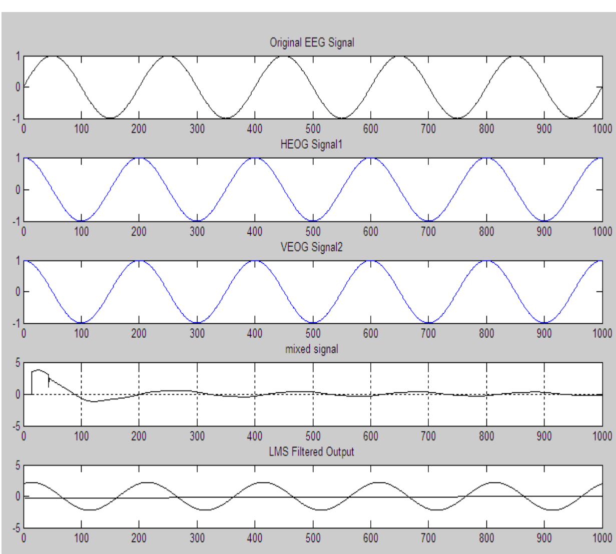

Figure.7 EEG Signal Simulation-3

FilterTap=16;
μ = 0.001;

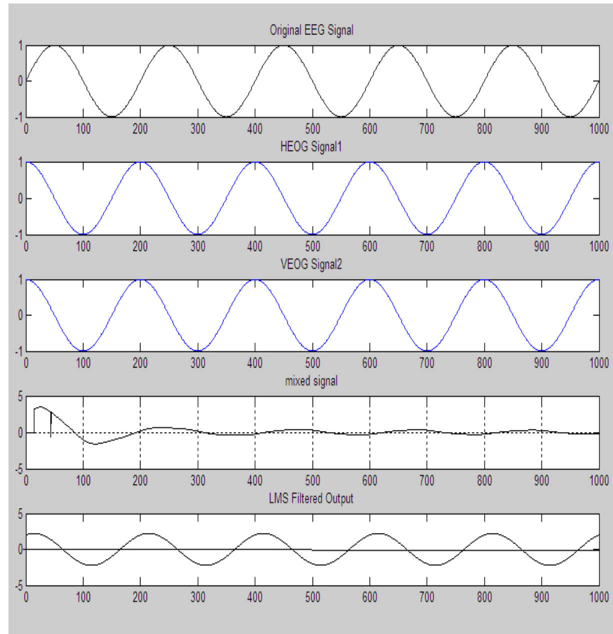

Figure.8 EEG Signal Simulation-4

FilterTap=32;
μ = 0.001;

In the Figure 8 EEG Signal Simulation-3, the value of filter tab has been changed from 16 to 32 and μ is taken as 0.001. This shows the EEG signal and noise signals are mixed together and the filtered out by using LMS adaptive filter. This figure can be compared with the EEG Signal Simulation-3 where the value of filter tab and μ are taken as 16 and 0.001 respectively. In this simulation graph the value of μ is decreased which shows the small change in the coefficient and the convergence of filter act as slowly. So it is concluded that with large step-size the filter convergence takes place fast. It is also concluded that with the large value of μ, the filter convergence act as fast. By changing the results and convergence rates, finally it is concluded that the LMS adaptation take place properly and it performed adaptation.

5.3 Algorithm Implementation and Verification

Biomedical signals play a critical role in the diagnosis of patients. EEG is a medical monitoring device which is used for the diagnosis of heart patient. As

EOG signals is major problem in EEG signal. An algorithm for the LMS adaptive filter was suggested, as the EEG signal can be variously mixed with EOG .The LMS adaptive filter is widely used to filter the EEG signal as its convergence causes good performance.

To verify the performance of the LMS adaptive algorithm, EEG signal has been selected and evaluated the performance of the proposed filtering technique. The MATLAB environment was used for the simulation of EEG signal. The code contents of the source file used in this paper were collected from various sources. To design the FIR filter, these source files were used to configure the DSP technique. After designing of FIR filter the LMS algorithm has been executed and verified.

Two basic processes are involved when the LMS algorithm is applied, a filter process involving the computation of the output of FIR filter produced by a vector of tap inputs and an estimation error signal calculated by comparing this output to a desired response.

LMS algorithm is extensively used in different application of adaptive filtering due to its computational simplicity and FIR filter is also popular because of its simplicity and inherent stability. An important parameter is the step size μ, it affects the convergence rate and stability of the LMS adaptive filter. If μ is small, it will create a slower rate of convergence but is more accurate and stable. On the other hand, if it is large, it will converge faster but will become less accurate and less stable.

Moreover another method has been implemented for the removing of harmonics (known as low frequency noise) and high frequency noise from original EEG signal. The humming and high frequency noise has been removed by using general notch rejection filters and windowed sinc low pass filter.

By investigating this method, it is concluded that the overall result of the technique is achieved. The main goal for the removal of different noise i.e. harmonics and a high frequency noise has been performed satisfactory.

6. CONCLUSION AND FUTURE SCOPE

CONCLUSION

This paper is devoted to the problems and solutions on removal of EOG (HEOG and VEOG) and other Single Frequency tones from Signals. Noise cancellation from EEG signal was explained clearly and the methods and techniques applicable to be used are discussed throughout the paper. It has been proposed a solution for EEG mixed with EOG its respective harmonics and high frequency noise interferences from original EEG signal. The results have been obtained.

An adaptive filter is used in applications that require different filter characteristics in response to variable signal conditions. The speed of adaptation and accuracy of the noise cancellation after adaptation are important measures of performance for noise cancellation algorithm. The goal of the adaptive filter is to match the filter coefficients to the noise so that the adaptive filter can subtract the noise out from signal.

The test EEG signal has been taken. The signal is corrupted by EOG signal. It is observed that the EOG signal which is then mixed with original EEG signal, it is also examined that the mixed signal is displayed on the plot. After passing through LMS algorithm the filtered output is nearly same as the input signal with some acceptable distortion range. The value of step size μ play an important role in determining the convergence speed, stability and residual error after convergence. The convergence rate was controlled by LMS step size μ. The EEG signal graphs described in the simulation results verify the adaptation of the LMS adaptive algorithm by changing various parameters like step size, convergence value (μ) and filter taps have various effects on the output graphs. The result shows that LMS is an effective algorithm used for the adaptive filter in the noise cancellation implementation.

Future Scope

This paper provides the real concepts along with the theoretical backgrounds of removal of EOG signals, single frequency tones and high frequency noise from original EEG signal. The adaptive filtering techniques based on LMS algorithm could be implemented for more signals and also with different algorithms such as NLMS and RLS to achieve the desired results. The implementation of the removal of multiple of harmonics from EEG signal could also be investigated.

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REFERENCES

[1] Stacy Finlay, Carrie Klekta and Ernie Packulak, Adaptive Noise cancellation for EEG Signal, 2002.

[2] Rolf Limacher, Removal of power line interference from the EEG signal by adaptive digital filter, ETH Zurich, laboratory of Electrical Engineering Design Gloriast, Zurich.

[3] Applications of Adaptive Filtering to EEG Analysis, Noise Cancellation and Arrhythmia Detection.

[4] Nitish VThakor, Seiiior Member, IEEE, and Yi-Sheng Zhu, Se~iior Momher, IEEE.

[5] Anna Esposito, Pierluigi D. Andria, An Adaptive Learning Algorithm for EEG Noise.

[6] John Leis, Digital Signal Processing- A MATLAB-Based Tutorial Approach, Research Studies Press Ltd. 2002 .

[7] Hakan Johansson, Synthesis and Realization of High-Speed Recursive Digital Filters, University of Linkoping, Sweden, 1998 .

[8] Philip D. Cha , John I. Molinder , Fundamentals of Signals and Systems, A Building

[9] John G. Proakis, Dimitris G. Manolakis Digital Signal Processing, Principles, Algorithm and Applications

[10] Raman Jamal, Mike Cerna, John Hanks Designing Filters Using the Digital Filter Design Toolkit

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