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Analysis for Denoising of ECG Signals Using NLMS Adaptive Filters

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Abstract

Electrocardiogram (ECG) signal is the signal which consists of the parameters which reflects the electrical representation of heart activity. The main components which are shown by the ECG signal have some important attributes of human heart as well as some hidden information of heart. The information which is found from the ECG signal is so much meaningful to derive various vital parameters related to heart. But the ECG signal can easily be affected with the Noise. Noise is the signal which distorts or interfere the actual power level of ECG signal, this can be due to the motion artifacts or due to the power sources which are resided where this ECG had been taken The ECG system which is typically based on computer have some units, the very first unit is used for pre-processing of ECG signal, second unit is used to detect the heart beats, third unit is used for feature extraction & the last one is used for classification. Signal processing unit which is used for ECG signal is much important for both research & clinical experiments. The artifacts which are analyzed due to the motion & shown in the heart beat processing can effectively remove & the ECG signal is cleaned after using LMS, NLMS and Notch processing. This paper comprises of result and analysis of ECG signal processing using NLMS algorithm & shows its important role in the biomedical applications.

1 Introduction

It is so much complicated to find out the accurate parameters as a result of the patients who are monitored under the processes like ECG, EEG, EOG & EMG. The problem with this type of processing is the Electromagnetic field which is generated by the machines which are placed near to the equipment which is used for the measurement [1].

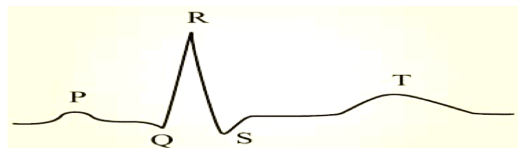

Figure 1 Electrocardiogram Signal

First of all if we want to find out the source of noise in the ECG signal then the primary source of the noise is electrical power system, this causes noise at the time of recording & monitoring of Electrocardiogram signal. If we try to differentiate this noise according to the range of frequencies so there are two types of noises exists in this type of systems. First one is the low frequency noise which disturbed the Electrocardiogram signal this is the prime noise in the Electrocardiogram signals. Sometimes ECG signal disturbed or we can say interfered by the high frequency noise. In today's era main cause of this type of noise are cellular phones [3].

So the interference or the noise is unwanted signal, so our prime objective is to reduce or remove this noise. So noise cancellation is widely used in the applications like voice and audio signal processing, data communication, signal acquisition techniques to calculate vital parameters etc. Various methods are used for noise cancellation; the basic procedural flow of noise cancellation is given below in Fig. 1.2. The two methods described majorly in the literature one is Adaptive and second one is non-adaptive technique. The problem with non-adaptive type of technique is that this is time-invariant technique. So to overcome this problem adaptive method is widely used [6].

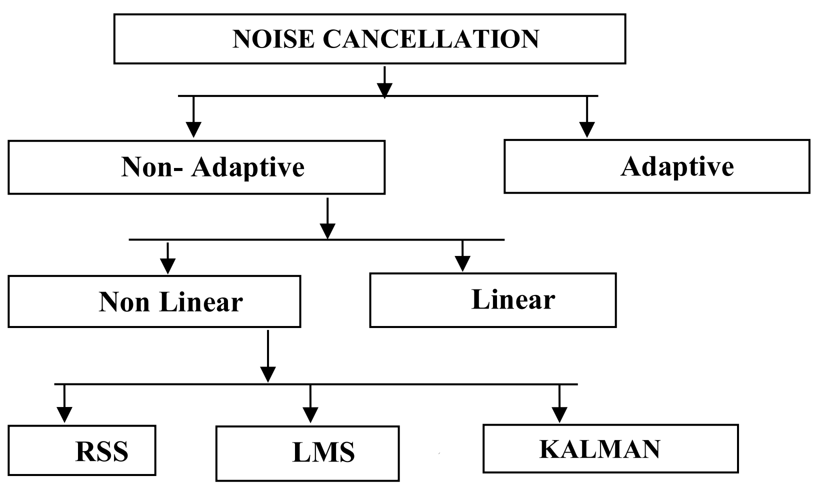

Figure 2 Different Methods of Noise Cancellation

1.1 Adaptive Filter

Adaptive filter have a property that the transfer function of Adaptive filter is self-adjustable, this self-adjustable transfer function is based upon the optimization algorithm and is controlled by error signal. Problem which is arises in these adaptive filters is complexity of the algorithms which is used for optimization. Due to this reason most of the adaptive filter which are used are digital in nature.

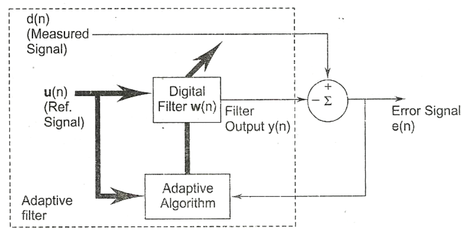

Figure 3 Block diagram of an adaptive filter

1.2 Digital Filter

Digital filter is nothing but a mathematical formulation or much precisely an algorithm which is used to implement desired hardware and software. Input and output both are digital type in digital filters, to achieve the objective of digital filters. The main objective to adopt a digital filter is that if there is a recovery of the signal and this recovery signal is distorted by noise, so the separation between two signals is highly needed which is provided by the digital filters.

1.3 Normalized LMS (NLMS) algorithm

Modification in the Least Mean Square algorithm is known as Normalized Least Mean Square (NLMS).As we all know that rate of convergence depends upon step size of two neighboring coefficients of filter. From the literature survey we found that the step size between the neighboring elements changes due to the change in input signal power as input signal power changes with time. Two types of signals affected the convergence rate one is small in nature and another one is large in nature, the signal which is large in nature produces an error in the signal. That’s why the main objective of the NLMS algorithm is to maintain the step size with respect to the power of input signal. So this process is called normalization of step size. Therefore the step-size parameter is said to be normalized. In the designing of adaptive filter one issue arises which is the selection of step size parameter μ. LMS algorithm have a severe problem which effects the stability of algorithm. This is due to the sensitivity of algorithm based on stability of input u (n). So due to this problem the stability of the algorithm is little bit affected.

The question arises here is that is there any other advantage of NLMS algorithm then yes it reduces the gradient noise amplification issue. This problem arises when the value of When the convergence factor is too large. Normalized algorithm has an advantage that it has lower value of step size than the traditional LMS algorithm. Due to the lower value of step size & utilization of variable convergence factor normalized algorithm is much faster than the conventional LMS algorithm. It also minimizes instantaneous output error.

ECG signal is easily distorted by noise signal so reduce this signal an efficient algorithm is used which is known as NLMS algorithm.

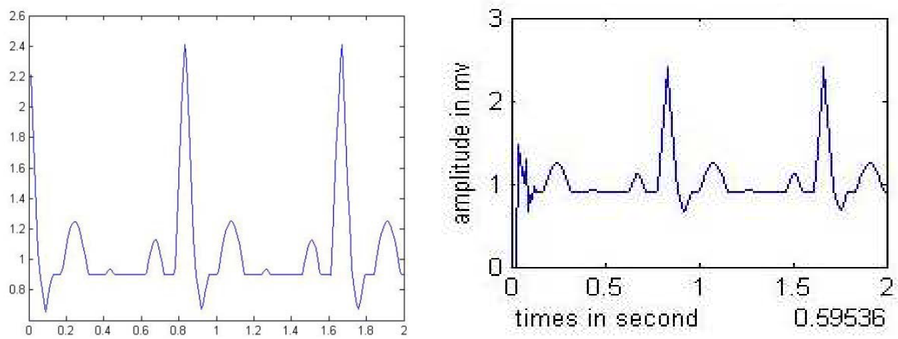

Figure 5 (a) ECG Signal (b) NLMS Algorithm

The work which is mentioned above is implemented using MATLAB, as it is very useful software product of Math works. It reduces complexity of manipulation as it based on the matrix manipulations. By using this software we can easily plot the data and functions & can easily implement the algorithms & generations of user interfaces. This software has a capability to interface their own program with other languages like (C, C++, FORTRAN & Java).

2 Methodology

This paper comprises of NLMS algorithm. First of all the detail of this algorithm is given in the paper. Then the validation and development of this algorithm also mentioned in the paper. The main objective of NLMS algorithm in this paper is to remove the non- trivial noise in ECG signal. First of all the component analysis which are important and can effectively separate out artifacts in ECG which are induced by motion done. The set of parameters which are selected for the better and quicker response also play an important role for the recognition & recovery.

2.1 Implementation of NLMS algorithm

If we want to apply the LMS filter to any signal then a standard format is desired for the application of LMS filter. The adjustment which is applied to the tap weight vector of the filter has various iterations and it consists of three terms:

i. The filter designing parameter which is known as step size μ can be chosen by the designer.
ii. Second parameter is the source of information which is supplied to the input. It is denoted by tap input vector u (n).
iii. Third parameter related to error. Which is called as estimation error e (n), this error is calculated on each iteration (n).

The improvement which is done in the step size has a direct correlation with tap input vector u (n). So if the value of step size is high then Least Mean Square filter has some limitations due to gradient noise amplification problem. The maximum value of step size μ gives a highly stable signal u (n). The best method to optimize the speed of convergence with tap vector adjustment kept in mind; the numbers of iterations are n+1 in the normalized way with respect to the squared Euclidean norm of the tap –input vector u (n) at iteration n –hence the term “normalized”. It can be briefly described as follows:

- Initialization: Suppose that the value of tap weight vector w (n) is available, then we algorithm is used to find out an appropriate value of **w** (0), else set **w** (0) = 0.

Consider: $$0 < \mu < 2(E[|u(n)|^2 D(n) / E[|e(n)|^2])$$

Here $E[|u(n)|^2]$= Signal power of input signal.

$E[|e(n)|^2]$= Error signal power.

D (n) = Mean square deviation in weight vector

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- Data: Given u (n) =M by 1 tap input vector at time n.
  d (n) = desired response at time n.
  To be computed **w** (n) = estimate of tap-weight vector at time n.

- Computation:

$$y(n) = w(n)^T .u(n)$$
$$e(n) = d(n) - y(n)$$
$$w(n+1) = w(n) + \mu.e(n).u(n)$$

In this algorithm there are 2*M+1 addition & 2*M+1 multiplication, the number of iterations are n & tap length is denoted as M, it is also considered as order of the filter. Filter order describe the complexity of computation & it must be chosen very carefully. The table which is given below briefly describes the procedural flow.

Table 1 Variables used in Normalized Least Mean Square (NLMS) algorithm

Variable	Description
N	The current time index
u(n)	The vector of buffer input sample at step n=[u(n).u(n-1)…u(n-M-1)]^T
u*(n)	Complex conjugate of the vector of buffered input sample at step n
w(n)	Estimated value of filter weight with step size
y(n)	Output value of filter with step size n
e(n)	The error signal with step size n
d(n)	Expected value of desired input signal with step size n
M	Length of the filter
μ	The adaption step size
C	Constant with a small value.
A	Adaption Constant for NLMS algorithm.

Table 2 Normalized Least Mean Square (NLMS) algorithm

S.NO	NLMS ALGORITHM	
1	**Initial Conditions:**	**0< α < 2 and c is small constant. Length of Adaptive Filter=L Input vector: u[0,0,0…0]^T Weight vector[0,0,0..0]^T**
For each instant of Time,n=1,2,…,compute:		
2	**Output signal:**	**y(n) = w^T . u(n);**
3	**Estimation Error:**	**e(n) = d(n)-y(n);**
4	**adaption step size:**	**w(n)+[2 α/c+u(n)^T.u(n)]e(n)u(n) w (n+1) = w + 2 μ .e(n).u(n);**

3 Results

The result which we get from the simulation of algorithm mentioned in this paper given below. The simulation depicts performance analysis of various adaptive filters with different value of step size. The prime objective of this comparison is to check error cancellation capability of the algorithms. This error cancellation capability depends upon length of the filter, step size & total number of iterations. The simulations which are given below have different parameters such as step size, iterations & tap of filter.

3.1 Results & Analysis for Normalized Least Mean Square (NLMS) algorithm

For case c = 0.019, α = 0.102, Filter length is 15, 20, 30 and 1400 iterations Figure 6,7,8 given below.

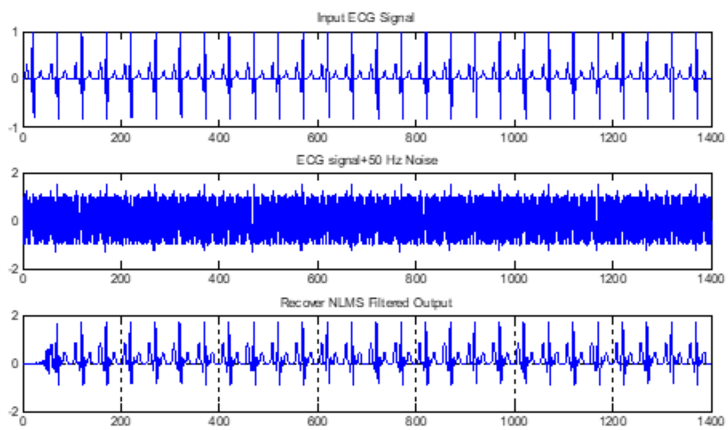

Figure 6 NLMS algorithm where c=0.019, α=0.102, Filter length = 15

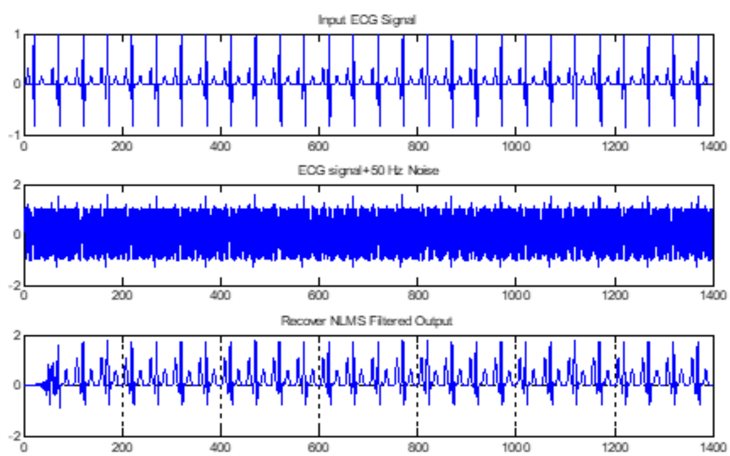

Figure 7 NLMS algorithm where c=0.019, α=0.102, Filter length = 20

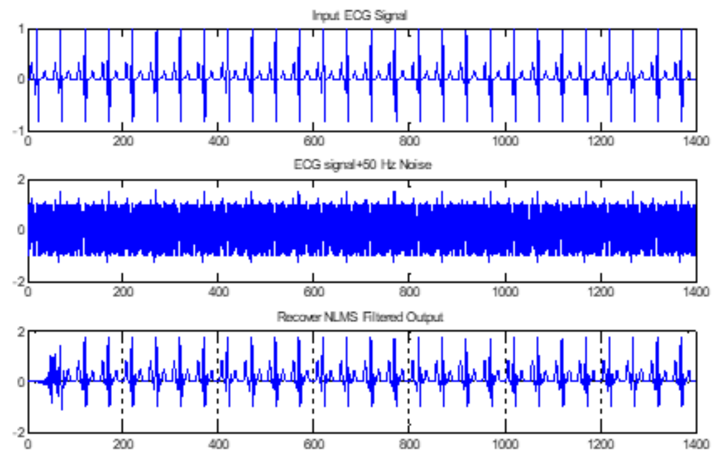

Figure 8 NLMS algorithm where c=0.019, α=0.102, Filter length = 30

3.2 NLMS Algorithm

Case with c= 0.019, α=0.102, Filter length is 40 and 1400 iterations in figure 9.

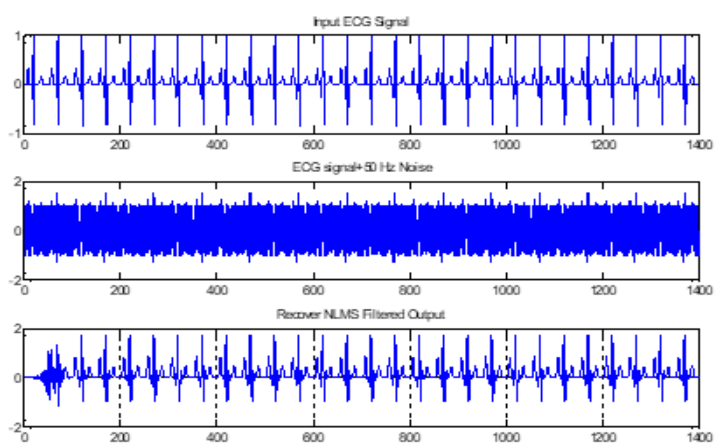

Figure 9 NLMS algorithm where c=0.019, α=0.102, Filter length. = 40

4 Conclusion

The prime motive of different types of adaptive filter algorithm is to reduce the Noise in the signal. In this paper Normalized Least Mean Square (NLMS) algorithm analyzed. After the simulation we get that NLMS algorithm can vary the step-size according to the input signal energy. Both stationary & non stationary signals can be processed using NLMS algorithm. Implementation process of algorithm is successfully done for different values of parameters. The parameters are as c =

0.019, α = 0.102, filter length is 15, 20, 30, 40 and 1400 iterations. As the order of the filter increased the noise level reduced in ECG signal.

5 Scope

According to our work which is done for NLMS various aspects with result given in this paper. There are other adaptive algorithms also present in the literature, we can simulate those algorithms too & make a comparison among all the algorithms. For this particular work we can use other algorithms too as Recursive Least Squares. In this paper real time processing of finite- length filters not considered, so we can also consider real time processing with its casual approximation. Another algorithms such as LMS and RLS algorithm and wiener filter can also be used to achieve the desired results.

References

[1] R.S Khandpur “Biomedical instrumentation hand book), 11th reprint 2008 Tata McGraw –Hill publication company Limited New Delhi. ISBN-13: 978-0-07-0473355-3.

[2] Stephen J. chapman “MATLAB programming for engineers” 3rd Reprint Edition 2003 by Thomson Asia Pvt. Ltd., Singapore ISBN: 981-240-606-9.

[3] Malindi, P.(2002) “Cancelling power line interference in electrophysiological signals”. *ECT Research Journal*, 2.

[4] Widrow B. and Hoff M.E. (1960), “Adaptive switching circuits”, *In IREWESCON Convention Record,* pp. 96-104, New York.

[5] Zhang Jiashu, Tai Heng-Ming, “Adaptive Noise Cancellation Algorithm for Speech Processing”, IEEE Transactions, pp 2489-2492, 2007.

[6] Jigram H. Shah, Jay M. Joshi, “Digital signal processing” University science press laxmi publication company Limited New Delhi.

[7] Suzanna M. M. Martens , Massimo Mischi, S. Guid Oei, ‘An Improved Adaptive Power Line Interference Canceller for Electrocardiography’ IEEE transactions on biomedical engineering, vol. 53, no. 11, November 2006.

[8] Syed Zahurul Islam, Syed Zahidul Islam, Razali Jidin, Mohd. Alauddin Mohd. Ali, “*Performance Study of Adaptive Filtering Algorithms for Noise Cancellation of ECG Signal”,* IEEE 2009.

[9] Chhavi Saxena, P.D.Murarka,Hemant Kumar Gupta,*” Noise Reduction in ECG Signals Using Notch Filter”*,IJCSE Volume-5, Issue-8 E-ISSN: 2347-2693.