IJCTA, 9(13) 2016, pp. 6057-6067

Comparative Analysis of Filtering Techniques for Impulsive Noise Cancellation from Real-Time Bio-medical Signal

Sarthak Panda* and Mihir Narayan Mohanty*

ABSTRACT

Impulsive Noise Cancellation in practice is a challenging task. In case of biomedical signals like ECG, EMG, EEG are even challenging for noise cancellation. These signals suffer from impulsive noise at the time of acquisition. In this paper, Cardiac signal corrupted with impulsive noise at the time of acquisition is considered. Different filtering algorithms are being applied to remove the noise. Initially the filters like FIR and two types of IIR filters are used for comparison. Further it has been proposed for adaptive filter with different Least Mean Square (LMS) algorithms. The signal has been tested and verified. Finally, the Wilcoxon norm based LMS is chosen which has an attractive result. For all the filtering techniques the signal is applied and their results are compared and are shown in result section. It shows that the proposed algorithm works better than the existing ones.

Keywords: Filtering Algorithm, FIR Filter, IIR Filter, Adaptive Filter, LMS, NLMS, WLMS

I. INTRODUCTION

The characteristics of human organ can be studied by collecting their parameters. These parameters can be collected through different devices for different organs. One such example of bio-medical signal is Electrocardiograph (ECG) that exhibits the characteristics of the heart. From these signals the physicians obtain important information to support their diagnosis, monitoring and treatment. At the time of such data collection, different types of noises may interfere. These noises may due to internal that is from human body, or may be at communication point as the electrical disturbance may occur do may be at the end of the device. Sudden occurrence of this type of noise is said to be Impulsive. It is not Gaussian in nature.

Day-by-day various devices have been developed for faster service with improvement of accuracy. Accordingly the research develops to meet these challenges.

Electrocardiogram (ECG) signals are vital for determination of heart related problems. ECG signals are low frequency signals of around 0.5Hz-100Hz. There are different noises such as noise considered here is impulsive by nature, which get included in these signals and change the initial signal; hence there is a need of cancellation of these noises from the original signal.

The simplest method of noise cancellation is direct filtering. It is the technique for evaluating a signal affected by noise is to pass it through a filter that has a tendency to minimize the noise. In case of direct filtering we can use the type of the filter is to be ‘Fixed’ or ‘Adaptive’.

The configuration of fixed filters requires earlier information of both the signal and the noise so we can plan a digital filter that passes frequencies contained in the signal and rejects the frequencies band that has been tied up with the noises.

* Department of ECE, ITER, S ‘O’ A University, Bhubaneswar, Odisha, India, *E-mail: sarthak216@rediffmail.com; mihirmohanty@soauniversity.ac.in*

The adaptive filters require a little or no knowledge of the signal and noise attributes. They can alter their impulse response to refine the noisy signal. Adaptive filters naturally adjust the filter coefficients to achieve optimum filtering with less amount error.

A little amount of the works related to this area of research has been done. Most of them are cited in the following section. In this work, we have attempted to develop a system which will cancel out the impulsive noise form ECG signal by using adaptive filtering theory. We have compared the SNR and spectral density of different fixed and adaptive filters before filtering and after filtering. Then to decide the best system we have compared the Mean Square Error and the improvement in SNR of different filters.

The paper is composed as follows. Section II, presents the related literature. In section III, Adaptive noise cancellation method has been discussed along with our approach to impulsive noise cancellation from ECG signal by using adaptive filter theory. Section IV presents the result and finally section V concludes this paper.

II. RELATED LITERATURE

Noise removal from bio-medical signal is highly essential before further processing. It has been used by many researchers since some decades; some of those works are cited here.

Finite Impulse Response (FIR) filter has been explored in view of different windows and in case of noise cancellation Infinite Impulse Response (IIR) filters has been utilized for ECG signal. Kaiser Window based FIR filters has found ideal for removal of artifacts [1]. Impulsive noise might generate false positives and it can be treated as outlier in a framework. The proposed system in [2] utilizes adaptive filters to minimize impulsive noise in ECG signal and maintain a strategic distance from the false positives. Also system identification has been achieved by the algorithm.

Impulsive noise can be minimized by the group of nonlinear robust filter. A new approach to myriad filter computation has been studied in [3] for bio-medical signal processing. The method offers the comparative results of filtering signals in impulsive noise case with respect to reference myriad filter. The proposed method in [3] was based on 2^{nd} order polynomial approximation.

Performance analysis based on MSE for different algorithms have been done for Least Mean Square (LMS), Normalized LMS (NLMS), Variable Step size LMS (VSLMS), Recursive Least Square (RLS) and Blind LMS algorithms on application of adaptive noise cancellation. With the help a delay input data was introduced to the filter. This helps for better estimation compared to the conventional LMS algorithm in [4]. Though human behavior is not exactly known it becomes very difficult to reduce noises in bio-medical signal by the help of fixed filters with fixed coefficients so adaptive filtering is required to overcome this type of problem. In [5] two types of adaptive filters were taken in to consideration to reduce noises from the ECG signal. In LMS adaptive Filter Mean Square Error reduces and Signal to Noise Ratio increases in comparison to RLS adaptive filter.

Sub-band decomposition using wavelet analysis has been applied to the corrupted ECG signal with PLI and impulsive noise [6]. For removal of impulsive noise morphological filtering is applied followed by one-dimensional structuring element. For removal of power line interference IIR Butterworth filter is used. As a result the PSD has been reduced significantly in between 50 to 60 Hz.

Least Mean Squares (LMS) and Normalized Least Mean Squares (NLMS) algorithm for cancellation has been implemented for additive white Gaussian noise (AWGN) which is added to the raw ECG signal. A better improvement in SNR has also been obtained in NLMS algorithm as compared to LMS algorithm [7].

Active control of impulsive noise has been performed in [8]. Impulsive noise has been modeled by utilizing non-Gaussian stable process with no second order moments. Due to this reason Fx-LMS algorithm becomes unstable for the impulsive noise. Fx-LMP algorithm is hence used which is based on minimization of least mean p-power (LMP) of the error signal. Modification has been done based on some statistics property and compared with some existing algorithms.

To mitigate impulsive noise, a new algorithm was proposed by authors in [9]. It was the modification of standard RLS in state space representation and was named to SSRLS. It has faster rate of convergence and better tracking performance. Initially authors have tested the algorithm over a sinusoidal signal. Then they have applied the algorithm over an impulsive noise affected ECG signal. On the basis of MSE their proposed algorithm is better convergence rate than other existing algorithms.

The advantages and disadvantages of FX-LMS algorithm and nonlinear FX-LMM algorithm have been discussed. Modification has been done with FX-LMS algorithm to achieve better performance in controlling impulsive noise. When impulsive noise is present proposed modified FXLMM algorithm can achieve better stability without sacrificing the performance of residual noise than FX-LMS and FX-LMM algorithm [10].

III. PROPOSED METHOD FOR NOISE CANCELLATION

Though different techniques have been used in literature, there is scope to enhance the accuracy. From this study the standard filter techniques are used in this work for comparison and explored as follows. In this work, we have tested with the fixed filter in the first stage, and further extended the work with adaptive filters. The FIR and IIR filters are represented as [11].

FIR filter relation can be described as [11]:

$$Y(n) = \Sigma_{k=0}^{M} b_k X(n-k) \tag{1}$$

Where, $X(n)$ is the input signal, $Y(n)$ is the filter output, M is the filter order and b_k is the filter coefficients.

IIR filter can be demonstrated as:

$$Y(n) = \frac{\Sigma_{k=0}^{M} b_k X(n-k)}{1+\Sigma_{l=0}^{N} a_l X(n-l)} \tag{2}$$

Where, $X(n)$ is the input signal, $Y(n)$ is the filter output,

M and N are the feed forward and feedback filter order respectively and b_k is the feed forward filter coefficient and a_l is the feedback filter coefficient.

For IIR filter we have taken two types as:

(1) Butterworth IIR filter

(2) Chevyshev IIR filter

The transfer function of Butterworth IIR filter can be described as:

$$T^2(\omega) = \frac{T_0^2}{1+\left(\frac{\omega}{\omega_c}\right)^{2M}} \tag{3}$$

Where, M is the filter order, ω_c is the cutoff frequency, T_0 is the gain at zero frequency.

Similarly the transfer function of Chevyshev IIR filter can be described as:

$$T^2(\omega) = \frac{1}{\sqrt{1 + R^2 C_M^2 \left(\frac{\omega}{\omega_c}\right)}} \tag{4}$$

Where, M is the filter order, ω_c is the cutoff frequency, R is the ripple factor and C is the Chevyshev polynomial with same as the filter order M.

An adaptive filter might be comprehended as a self-adapting filter that alters its coefficients to minimize the error. The error is the separation estimation between the desired signal and the adaptive output. At the point when the surrounding changes, adaptive filter arranges new features itself. The most essential property of adaptive filter is that it can work for unknown system well [12]. The proposed block diagram of the adaptive noise cancellation for ECG signal is as follows.

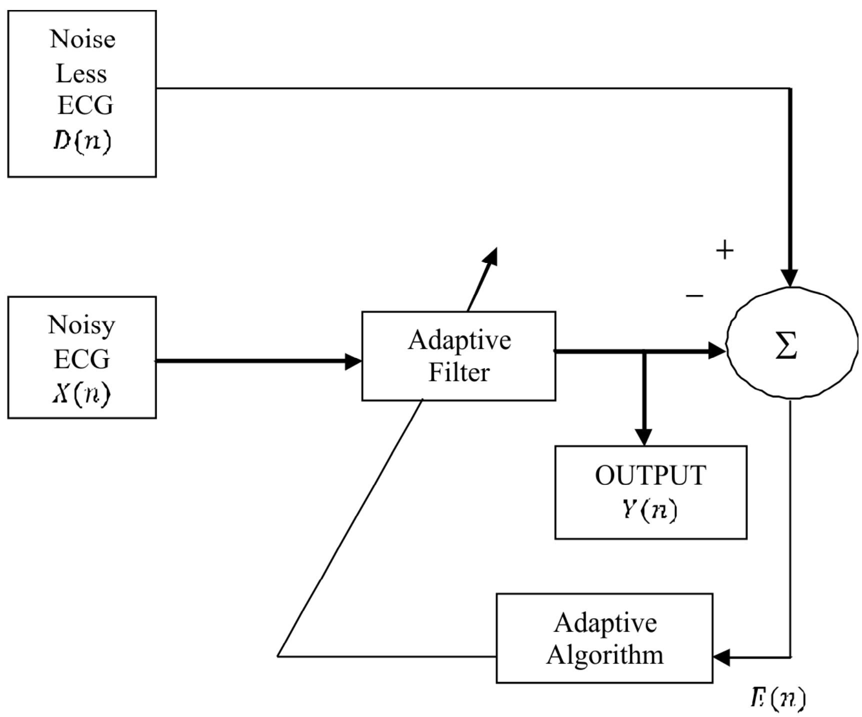

Figure 1: Block Diagram for Adaptive noise cancellation process

It is considered that, the desired signal as $D(n)$ which noise less ECG signal. For input $X(n)$ we have chosen an impulsive noise affected ECG signal. The error signal which is calculated by taking the difference between the desired signal and the output signal collected by passing the input signal through an adaptive filter is denoted by $E(n)$ [12].

$$E(n) = D(n) - Y(n) \tag{5}$$

Then the adaptive filter changes its weights according to the error signal. Whenever the error will be minimum, optimal point will be reached and at this instance of time we will get our reconstructed signal where impulsive noise will be not present. The output signal is given by,

$$Y(n) = W(n)\,X(n)^T \tag{6}$$

$W(n)$ is the adaptive filter co-efficient. Here now we will explore some adaptive algorithms which are used for impulsive noise cancellation purpose.

(A) LMS based adaptive algorithm

By using LMS algorithm system coefficients can be found. It uses gradient descent method in which the filter is adapted. The weight update relation is derived from [12, 16] as:

$$W(n+1) = W(n) + 2\mu E(n)\, X(n) \tag{7}$$

A positive constant μ (learning rate) is taken which controls the stability and rate of convergence of the system.

(B) NLMS based adaptive algorithm

The Normalized LMS algorithm is the personalized version of LMS algorithm. The weight updating relation for NLMS algorithm can be given by [13, 16]:

$$W(n+1) = W(n) + \mu\,.\,E(n).\frac{X(n)}{\delta + \| X(n) \|^2} \tag{8}$$

The step size can support the convergence rate of the filter. The NLMS algorithm produces a higher convergence rate when contrasted with the LMS algorithm with a higher leftover error.

(C) Wilcoxon norm LMS based adaptive algorithm

A norm called Wilcoxon norm is picked up as the cost function in the proposed method. Utilizing standard LMS the weights of the model are upgraded, which diminishes the norm steadily. Wilcoxon Norm of a vector can be characterized by, a score function [14-16]. Here a score function is defined by

$$\Delta(u){:}\ [0, 1] \to R \tag{9}$$

Which is non-decreasing,

$$\int_0^1 \Delta^2(u)\, du < \infty \tag{10}$$

Usually the score function is standardized such that

$$\int_0^1 \Delta^2(u)\, du = 1 \tag{11}$$

And

$$\int_0^1 \Delta^1(u)\, du = 0 \tag{12}$$

The error vector of n^{th} particle at m^{th} generation can be represented as $[E_{1,n}(m), ..., E_{N,n}\,(m)]^T$ for N input samples. Then the errors are distributed in a rising order from which the rank rnk $(E_{N,n}(m))$ each n^{th} error term is achieved. The score correspondent with each rank be stated as:

$$K(n) = \Delta(u))$$

$$=> K(n) = \sqrt{12}\left(\frac{n}{N+1} - 0.5\right) \tag{13}$$

Where, N is a positive number and the rank related to each term can be denoted by (1 ≤ n ≤ N). Now, at generation of each particle the Wilcoxon norm is then calculated as:

$$F_n(n) = \Sigma_{i=1}^{N} K(m) E_{m,n}(n) \tag{14}$$

The learning procedure using LMS Algorithm will continue till the cost function $C_n(t)$ in the equation (14) decrease to the minimum value.

IV. SIMULATION RESULTS AND DISCUSSION

For simulation we have collected the ECG signal from MIT-BIH database [17]. We have tried to cancel out the impulsive noise by considering the following parameters,

Table 1
Parameters for Impulsive Noise cancellation

Filternature	*Parameter*	*Value*
	FIR filter order	50
Fixed Filters	IIR filter order	2
	FIR/IIR filter cut-off frequency	0.5
	Peak-to-Peak ripple in IIR filter	2dB
	μ for adaptive algorithms	0.085
Adaptive Filters	δ for NLMS algorithm	0.001
	No. of iteration	3600
	Tap-weight	16

Fig. 2 and Fig. 3 shows that we have tried with general FIR and IIR filter initially but in this case the impulsive noise is not reducing properly. From Table 2 we can observe that SNR and SNR improvement is also not good in case of FIR and IIR filters. As a result we are moving to adaptive filters.

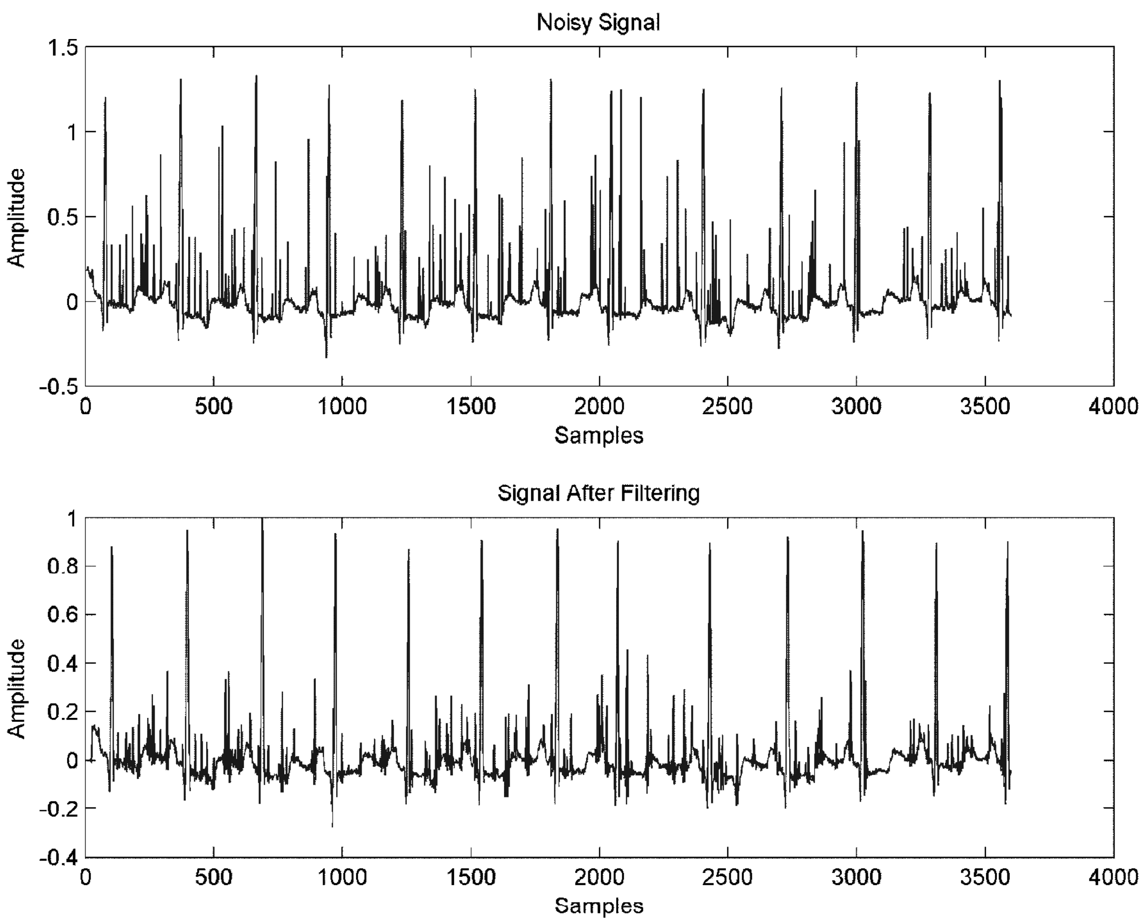

Figure 2: Impulsive noise cancellation using FIR filter

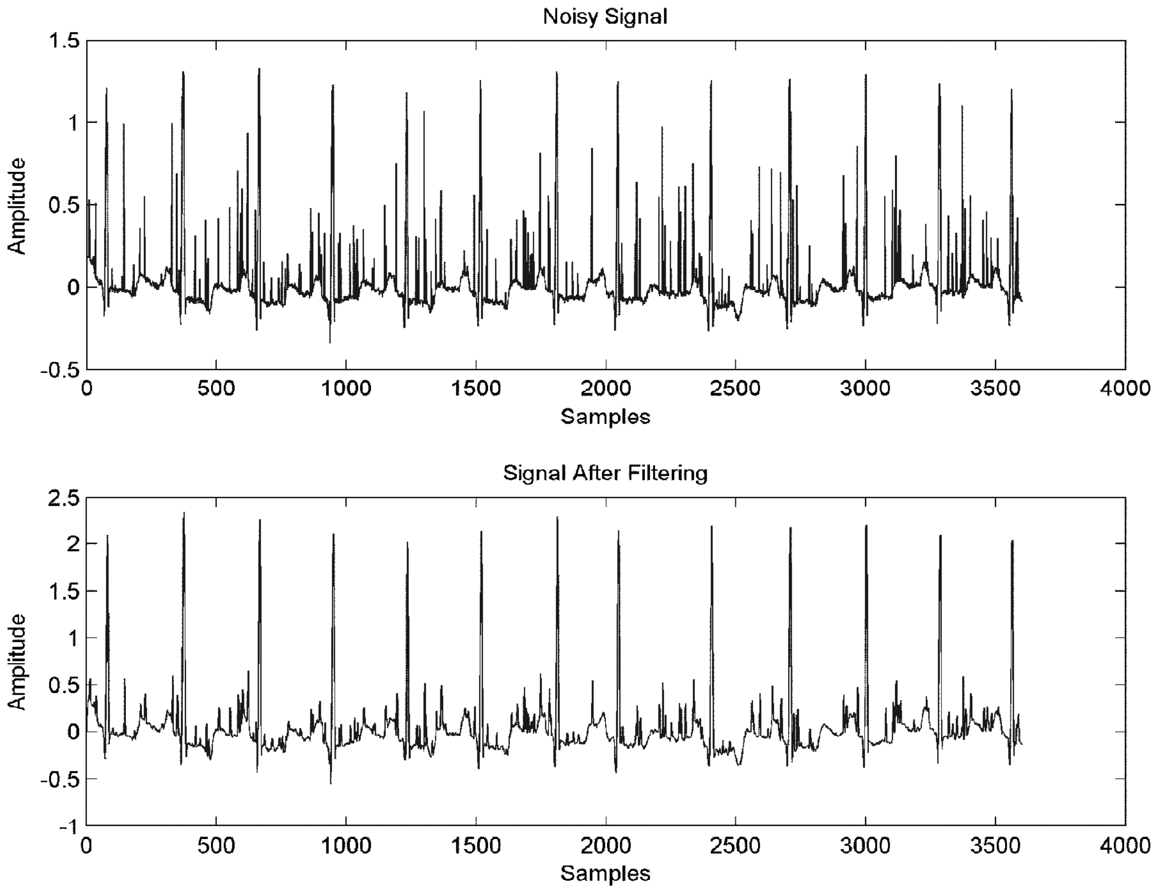

Figure 3: Impulsive noise cancellation using IIR Butterworth filter

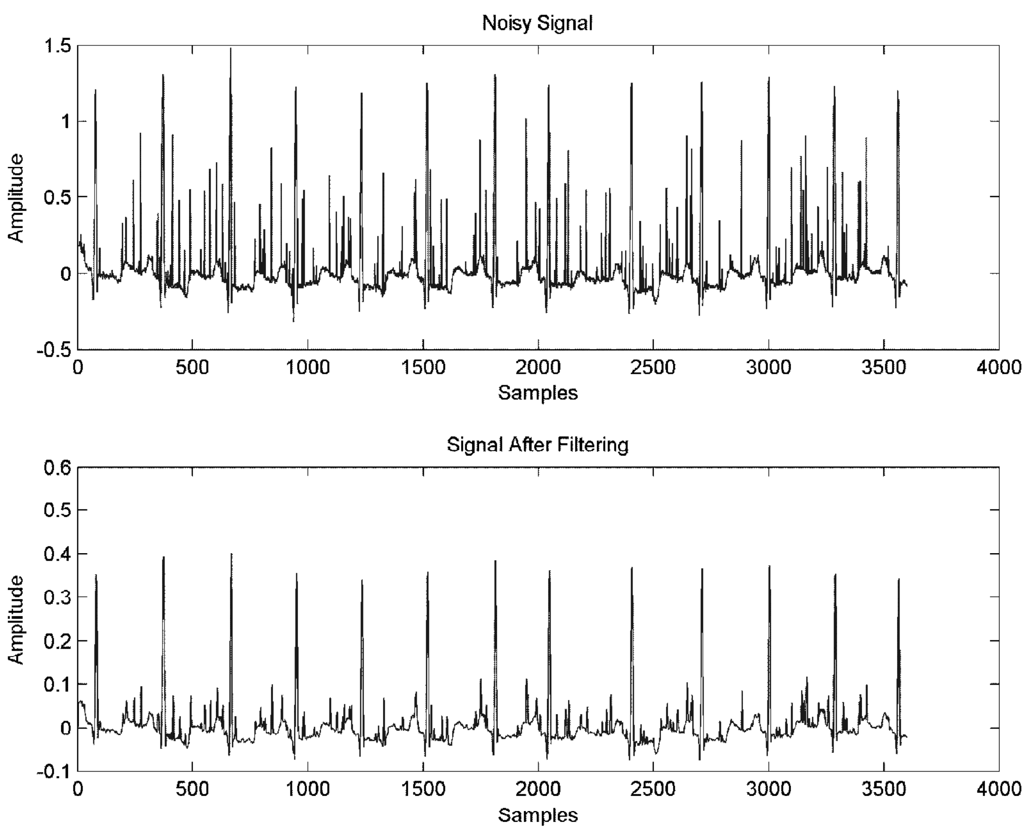

Figure 4: Impulsive noise cancellation using IIR Chebyshev filter

Fig. 5 to Fig. 7 shows the adaptive impulsive noise cancellation of ECG signal using LMS, NLMS and WLMS algorithm respectively. In all cases noise is reducded so to know which algorithm is better we have calculated SNR before filtering and SNR after filtering. Then we have calculated the SNR improvement.From Table 2 WLMS algorithm has better SNR after filtering than conventional LMS and NLMS algorithm.

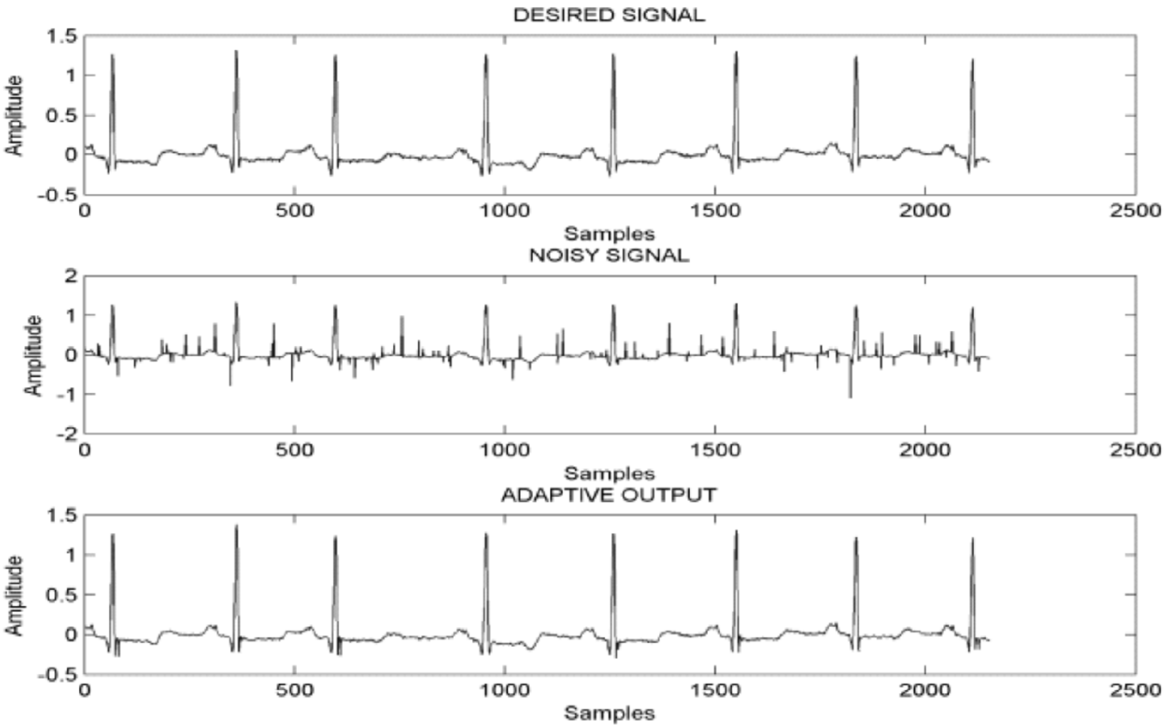

Figure 5: Impulsive noise cancellation using LMS algorithm

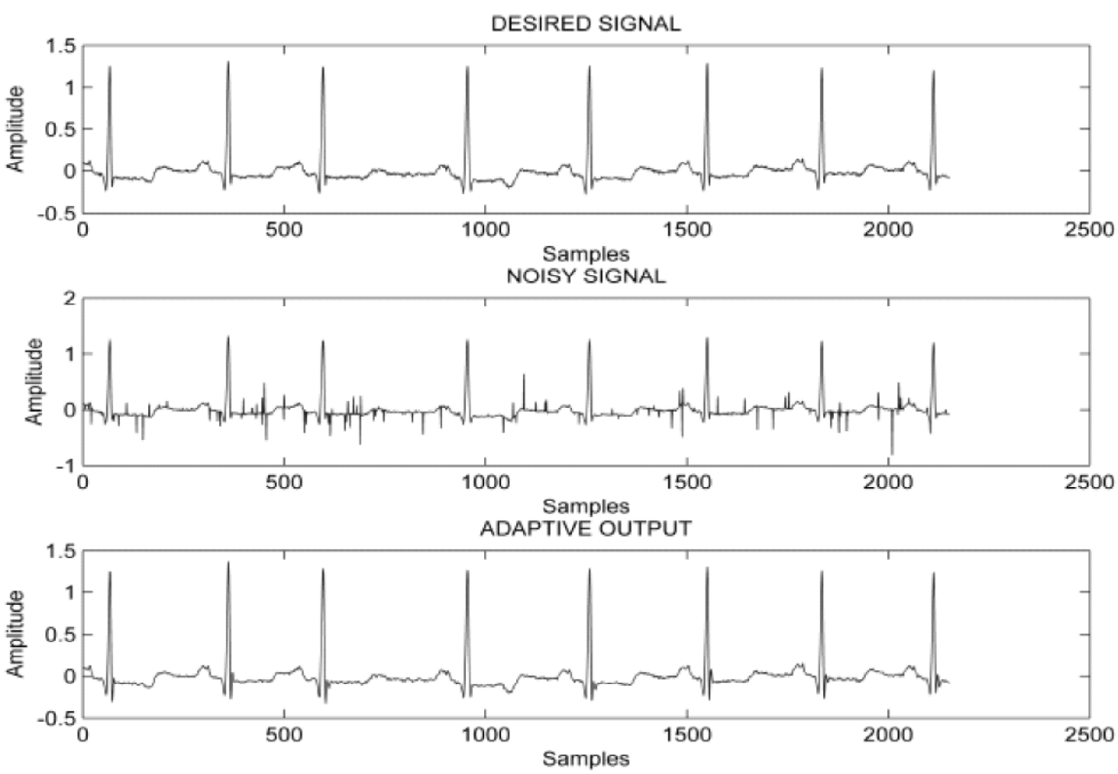

Figure 6: Impulsive noise cancellation using NLMS algorithm

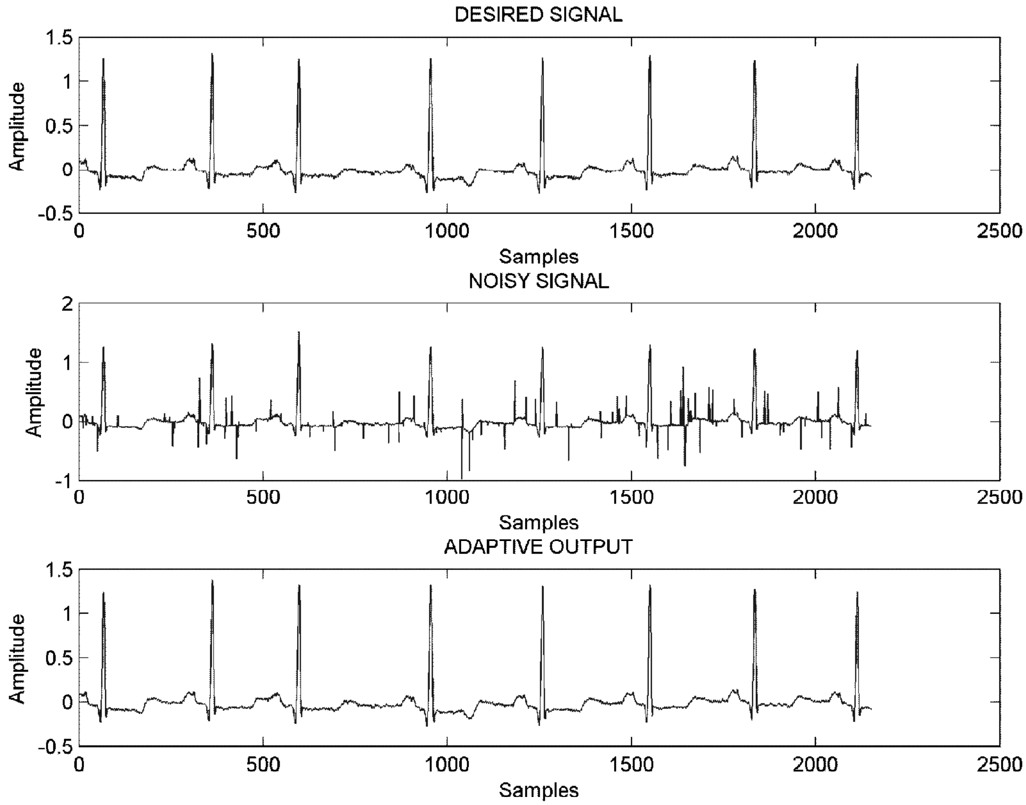

Figure 7: Impulsive noise cancellation using WLMS algorithm

Then we have compared the Mean Square Error of these three algorithms.By observing the mean square error curve WLMS is showing faster convergence than LMS and NLMS.From Fig. 8 we can analyze the MSE of LMS, NLMS and WLMS algorithm on application to impulsive noise cancellation of ECG signal.

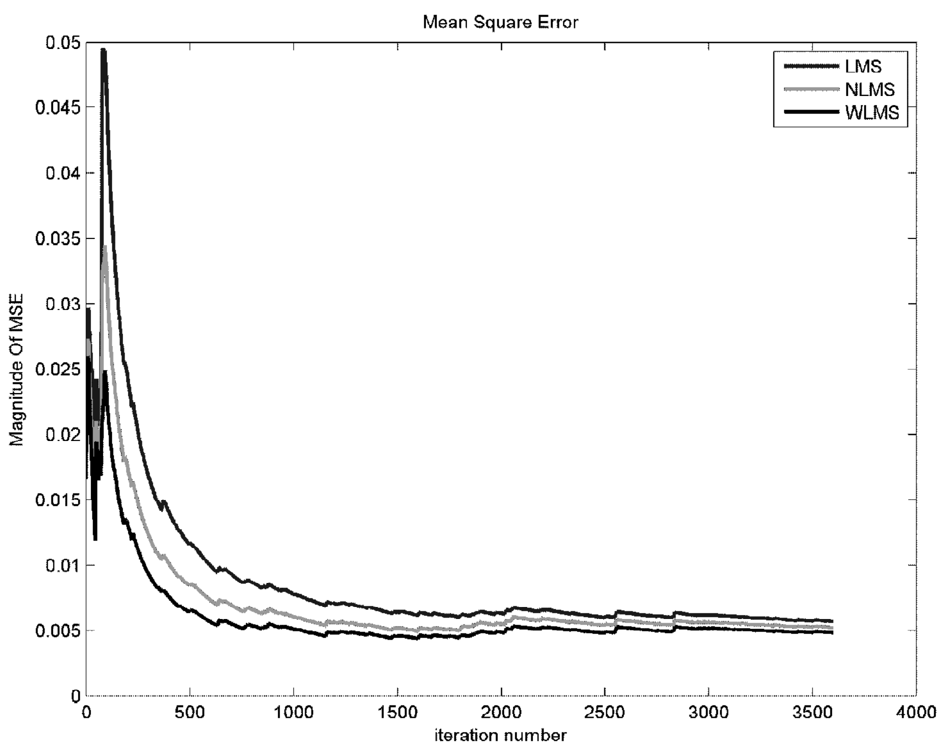

Figure 8: MSE comparision between LMS, NLMS and WLMS

In below Table 2 presents the comparision of spectral density, SNR and SNR improvement before and after filtering. We have compared between 2 types of fixed filters i.e. FIR and IIR and 3 types of adaptive filters based on LMS, NLMS and WLMS algorithm. By observing the spectral density column we can see that power is becoming less after filteing in all case.

Table 2
SNR and Spectral Density Comparison

Filter Nature	*Filter Type*	*Spectral density before filtering in dB*	*Spectral density after filtering in dB*	*SNR Before Filtering in dB*	*SNR after Filtering in dB*	*Final SNR Improvementin dB*
FIXEDFILTER	FIR	-16.9372	-17.0169	4.0362	6.9011	2..8649
	IIRBUTTERWORTH	-13.7124	-13.8219	4.6916	7.0663	2.3747
	IIR CHEBYSHEV	-13.9892	-14.8210	5.5279	7.3220	1.7941
ADAPTIVEFILTER	LMS	-14.4431	-15.5181	7.8499	11.3658	3.5159
	NLMS	-14.2777	-15.3174	7.0176	12.5551	5.5375
	WLMS	-14.0545	-15.5013	7.1397	15.4473	8.3076

V. CONCLUSION

In this paper, we have tested the FIR and IIR filters for the spectral density and SNR comparison. Also the adaptive filters have been used and proposed accordingly. Basically due to the variation of norm of LMS algorithm, the results found better than the other techniques. Further the work can be kept for future that the modification of RLS algorithm may be used and analysis should be done for better accuracy.

REFERENCES

[1] Bhumika Chandrakar, O.P.Yadav, V.K.Chandra, "A Survey of Noise Removal Techniques for ECG Signals", International Journal of Advanced Research in Computer and Communication Engineering, Vol. 2, Issue 3, March 2013.

[2] Shankar Gupta, Ramchandra Manthalkar, Suhas Gajre, "Suppression of Impulse Noise using Adaptive Filters", Computing in Cardiology 2013; Vol 40, pp. 527-530.

[3] Tomasz Pander, "Impulsive Noise Filtering In Biomedical Signals With Application of New Myriad Filter", International Bio-signal Processing Conference 2010, Analysis of Biomedical Signals and Images, Vol 20, pp. 94-101.

[4] Mostafa Guda, Safa Gasser and Mohamed S. El Mahallawy, "MATLAB Simulation Comparison for Different Adaptive Noise Cancelation Algorithms",International Conference on Information Security and Cyber Forensics (InfoSec) , 2014, pp 68-73.

[5] Sachin Singh, K. L. Yadav, "Performance Evaluation Of Different Adaptive Filters For ECG Signal Processing", International Journal on Computer Science and Engineering ,Vol. 02, No. 05, 2010, pp 1880-1883.

[6] Vikrant Bhateja, Shabana Urooj, Rishendra Verma, Rini Mehrotra, "A Novel Approach for Suppression of Powerline Interference and Impulse Noise in ECG Signals", 8TH International Microsystems, Packaging, Assembly and circuits technology Conference, 2013.

[7] Shivani Sharma, Ravindra Pratap Narwaria, "Noise Reduction from ECG Signal using Adaptive Filter Algorithm", International Journal of Engineering Research & Technology (IJERT), Vol. 3, Issue 7, July – 2014.

[8] Muhammad Tahir AKHTAR, Wataru MITSUHASHI, "Improved Adaptive Algorithm for Active Noise Control of Impulsive Noise", International Conference on Software Process, 2008.

[9] Alina Mirza, S.Mehak Kabir,Sara Ayub,Muhammad Arslan, Shahzad Amin sheikh, "Enhanced Impulsive Noise Cancellation based on SSRLS", International Conference on Computer, Communication, and Control Technology, Apr. 2015, pp. 31-35.

[10] Peng Li, XunYu, "Active noise cancellation algorithms for impulsive noise", Elsevier Mechanical Systems and Signal Processing, Vol.36, 2013, pp. 630–635.

[11] John G. Proakis and Dimitris Manolakis, 'Digital Signal Processing', 4th Edition, Pearson Education.

[12] B. Widrow and S. D. Stearns, 'Adaptive Signal Processing', 2nd Edition, Pearson Education.

[13] Simon Haykin, 'Adaptive Filter Theory', Prentice Hall, 2002, ISBN 0-13-048434-2.

[14] Sidhartha Dash, Mihir Narayan Mohanty, "Variable Sign-Sign Wilcoxon Algorithm: A Novel Approach for System Identification", International Journal of Electrical and Computer Engineering (IJECE) Vol.2, No.4, August 2012, pp. 481-486.

[15] Sidhartha Dash, Mihir Narayan Mohanty, "Analysis of outliers in system identification using WLMS algorithm", International Conference on Computing, Electronics and Electrical Technologies (ICCEET), pp. 802-806, Kanyakumari, 21-22 March 2012.

[16] Sarthak Panda, Mihir Narayan Mohanty, "Analysis Of Norms in Adaptive algorithm On Application of System Identification", Information Systems Design and Intelligent Applications, Advances in Intelligent Systems and Computing 433, pp 579-588,2016, DOI 10.1007/978-81-322-2755-7_60.

[17] *https://www.physionet.org/physiobank/database/mitdb/*