Adaptive Filter Design for ECG Noise Reduction using LMS Algorithm

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***Abstract*—This paper presents the adaptive filters algorithms for removing noise from the Electrocardiogram to receive noise less pure embryo signals. Filtering ECG signals requires a filter which can automatically adapt according to changing input and noise. Adaptive filtering has been used to reduce the noise from the desired ECG signals by using LMS algorithm. Other algorithms like NLMS and RLS can also be used but LMS gives least MMSE amongst them so it can be used where accuracy is required. The measures of performance contains the optimization between the rate of convergence and MMSE by the help of MATLAB. The experimental results have shown that for small values of step size the rate of convergence increases.**

***Keywords* — ADAPTIVE, ANC, ECG, FILTER, LMS, MATLAB, NLMS.**

I. Introduction

Various fields like audio, video, medical and cellular industries are now a days depends on digital technology which in short is dealt by one of the most upcoming field of engineering i.e. digital signal processing. This area of engineering has achieved high interest, by the use of all DSP chips [1]. Digital signal processing basically deals with digitized representation of signals or data and use of digital sub systems to study, manipulate save or collect meaningful data from these signals [2]. An electrocardiogram is one of the such application which is a painless testing process which measures our heart beats in the form of electrical parameters. The data is recorded either on the paper or on screen and this data is then analyzed by the experts. With every heartbeat vibrations are produced across the heart and it tends muscles to contract and extract blood from the heart. But since many other signals are surrounding us, there is a large probability of interference created by those signals tending our ECG signal to lose information which in turn provides incorrect diagnosis [3]. So to avoid interference from our ECG signals is our prime concern. To design such kind of systems we need to utilize maximum assets. For achieving reliable system two things are to be considered in mind that accuracy and processing time should be appropriate [4]. Basically we can classify these methods as adaptive and non adaptive filtering. In non adaptive filtering or fixed filtering, comes IIR filters, FIR filters and notch filters [5]. Adaptive filters are used for noise cancellation in variety of areas. Adaptive filters are able to manage their impulse response consequently and with small knowledge of signal or information we can design them to extract our information signal from the unwanted data. This helps in improving the signal to noise ratio [6]. Adaptive filters are used for those applications where some parameters of required processing operation are not initially known or can be changed. Figure 1 represents the general block representation of adaptive FIR filter. Where x(k) is the data, y(k) is filter response which is not constant, and d(k) is the required signal. In this figure 1, the data waveform is linked to the adaptive filter and generates an output waveform. The error waveform can reduce the error and remove the received output and required signal by manipulating the filter coefficients [7].

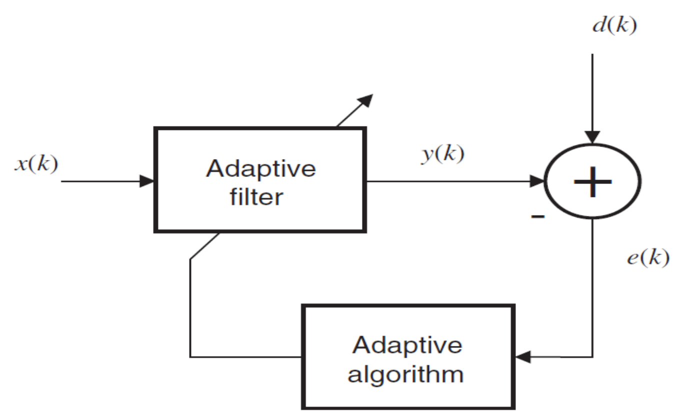

Fig. 1 General Adaptive Filter Structure

II. Adaptive Algorithm

An Adaptive filter finds its application where the stable characteristics are not known or cannot be fulfilled by the time invariant filters. Adaptive filter direct modeling or system identification and adaptive inverse modeling or channel equalization have wide applications in telecommunication, control system, instrumentation, power system engineering and geophysics [8]. So for this type of situations adaptive filters are considered to be the best. Adaptive filters uses various algorithms to reduce the noise from the desired signal and can be utilized on the basis of application and requirements. To design an adaptive filter it should be kept in mind which algorithm suits the desired requirement [9].

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Amongst various adaptive algorithms we have considered LMS algorithm.

LMS ALGORITHM :- Least mean squares (LMS) algorithms are used in adaptive filters to find the filter coefficients and produces the least mean squares of the error signal [10].It is commonly used algorithm for adaptive filtering. The important features of LMS algorithm are simplicity and robustness, effective tracking capabilities both with respect to computational load and easiness of implementation [11]. Least mean square algorithm manipulates the coefficients of the filter and generate changes by a quantity equivalent to the current approximate value of the error surface [12]. LMS algorithm doesn't require matrix inversion technique nor it needs correlation function calculation due to which this algorithm becomes the simplest and easiest amongst all. Decreased Mean square error is obtained because of iterative method incorporate in it to design repeated corrections in negative direction of gradient vector expressed as [13, 14].

$$a(n) = b(n).c(n) \qquad \text{(i)}$$

$$e(n) = d\,(n) - a\,(n) \qquad \text{(ii)}$$

$$b\,(n+1) = b\,(n) + \mu.\, c\,(n).\, e\,(n) \qquad \text{(iii)}$$

Where, a(n) = filter output, b(n) = input signal, E(n) = error signal, d(n) = other observed signal.

The get the optimum filter weights (R^{-1} P), by considering the initial value to be equal to zero in most of the cases and calculating the MSE gradient, the weights are managed. Now if the value of the MSE gradient is negative, the value of the weights should have to be increased whereas if the value of MSE gradient is positive then the value of weights has to be decreased. Hence the equation for the updated weights can be defined as:

$$W_{n+1} = W_{n-}\ \mu\ \ [n] \qquad \text{(iv)}$$

Where, mean square error is denoted as [1].

For p^{th} order :

Parameters: P = filter order

μ = step size

Initialization: •(0) = 0

Computation: For n = 0, 1, 2…

$X(n) = [x(n), x(n-1), \ldots, x(n-p+1)]^T$

$e(n) = d(n) - \bullet^H(n)\, X(n)$

$\bullet(n+1) = \bullet(n) + \mu e^*(n)\, X(n)$

NLMS ALGORITHM :- The drawback of pure LMS algorithm of being sensitive to the scaling of the input is overcome by the variant type of LMS algorithm i.e. Normalized Least Mean Square algorithm which normalizes the power of the input signal. It is a type of Least Mean Square algorithm [15].

For p^{th} order :

Parameters: P = filter order

μ = step size

Initialization: •(0) = 0

Computation: For n = 0, 1, 2…

$X(n) = [x(n), x(n-1), \ldots, x(n-p+1)]^T$

$e(n) = d(n) - \bullet^H(n)\, X(n)$

$$\bullet(n+1) = \bullet(n) + \frac{\mu e*(n) X(n)}{X^H(n) X(n)}$$

III. ELECTROCARDIOGRAM

The Electrocardiogram plays very important part in giving information about the heart. Here the signal is converted into electrical waveforms and that can be viewed on displays like oscilloscope [16]. The Embryo Electrocardiogram displays the electrical format of embryo heartbeat. The embryo waveform denoted by d[n] can be viewed as superposition of two waveforms: one is the embryo waveform f[n] and the mother heartbeat waveform $s_0[n]$. The waveform $s_0[n]$ can be considered as noise signal because it consists of all the maternal noise including mother heart beat. So to get pure embryo waveform , $s_0[n]$ has to be calculated with accuracy and then should be removed from the abdominal waveform [17]. The noise introduced can be of two types narrow band and broad band.. Examples of narrow band noise are baseline wander (BW), motion artifacts (MA), power line interference (PLI), etc. These types of noise can be attenuated by FIR notch filter or by adaptive noise canceller (ANC) [1]. The Adaptive Noise Canceller (ANC) purpose is to remove noise from the signal and hence improving the signal to noise ratio. The figure 2 shows the Adaptive Noise Canceller.

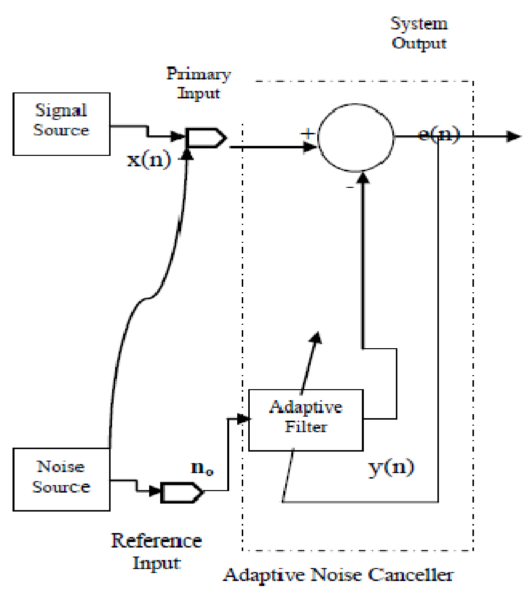

Fig:2 Adaptive filter based Noise Cancellation System [1]

The ANC have two inputs named as the primary input and the reference input. The primary input signal gets 'x ' corrupted by the noise 'n' which is uncorrelated to the signal whereas the reference signal get mixed up with noise n_0 which is uncorrelated to the signal but is correlated to the noise n. To get the close estimation of the input noise signal y(n) the n_0 is passed through adaptive filter and the estimated signal is then removed from the corrupted signal which then gives the estimation of error e(n).

IV .LMS ECG SIMULATIONS

The concept is used to reduce the noise in Electrocardiogram by using Least Mean Square (LMS) algorithm. This work has used MATLAB tool for designing electrocardiogram with reduced noise. In the process we have developed an embryo waveform, a mother heartbeat and desired reference signal. As while going through the system the waveform gets added with the noise. The figure 3 shows the mother heartbeat waveform.

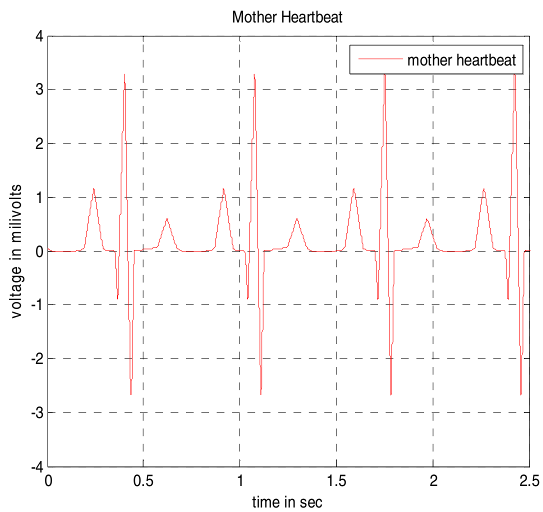

Fig 3. Mother heartbeat waveform

Figure 4 represents Waveform of the embryo heartbeat.

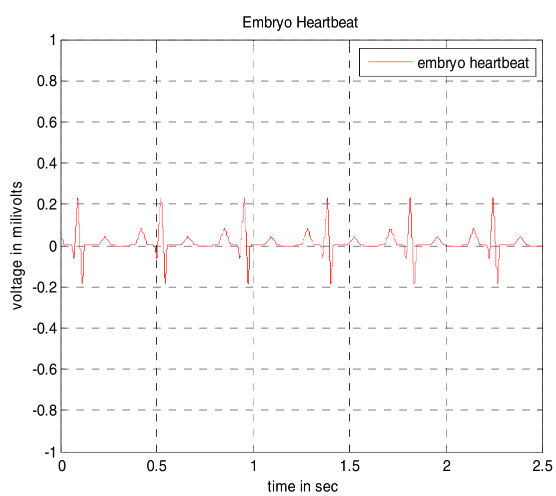

Fig. 4 Embryo heartbeat waveform

In figure 5 the desired embryo waveform is generated with MATLAB.

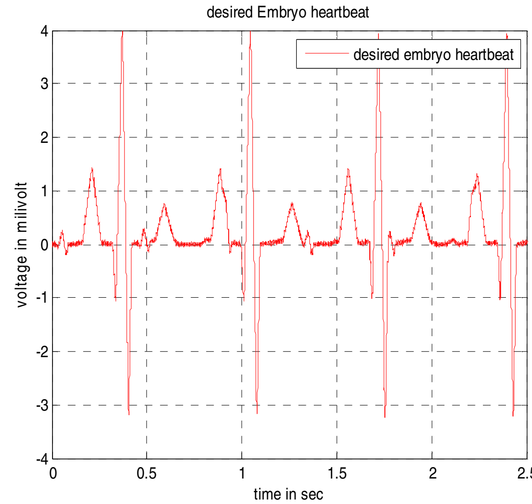

Fig. 5 Desired embryo heart beat waveform

Figure 6 is the mother heartbeat which has been considered as the reference waveform.

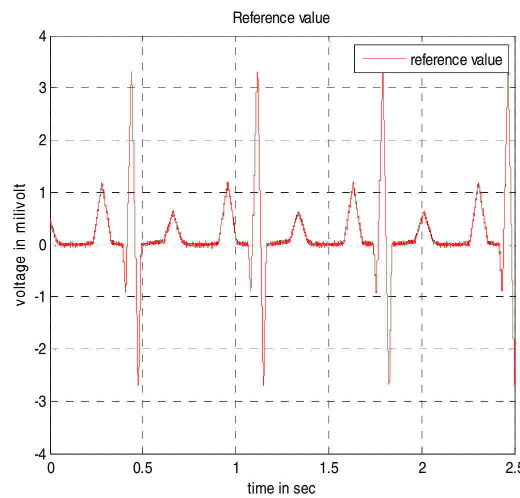

Fig. 6 Reference waveform for mother heart beat

Figure 7 shows measured waveform with noise introduced as error.

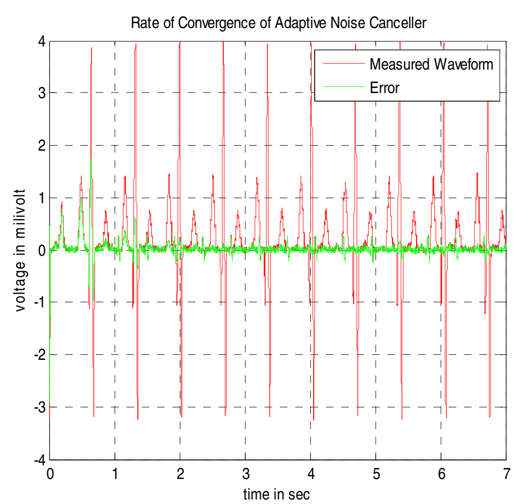

Fig. 7 Rate of convergence of measured and error signal

Now designing the adaptive filter with 15 number of taps and step size .00007 which shows steady state error signal in figure 8..

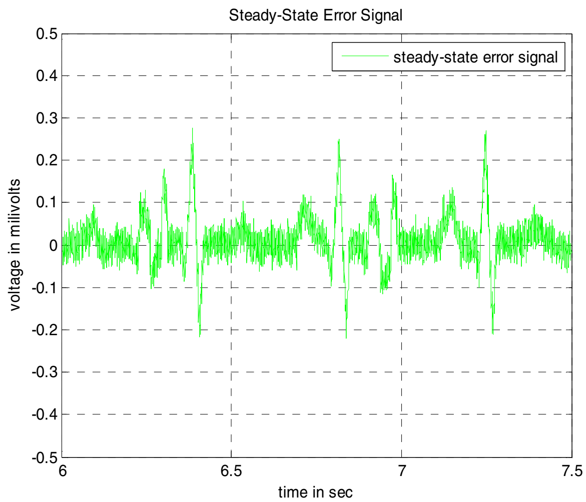

Fig. 8 15 taps filter with step size .00007

Whereas in figure 9 the same adaptive filter is designed using step size .00001 which reduces the MMSE but rate of convergence has also increased.

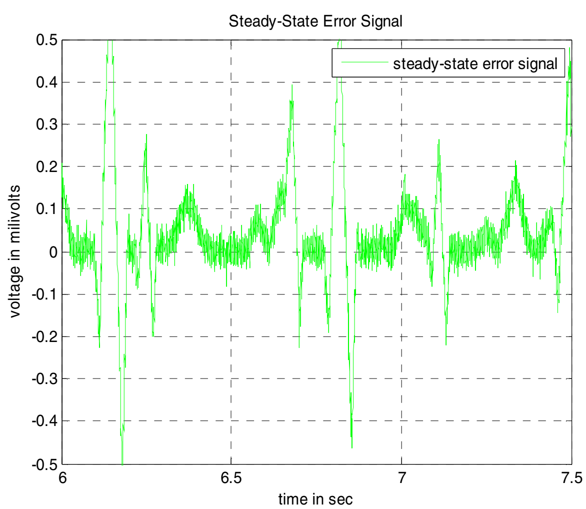

Fig. 9 15 taps filter with step size .00001

In figure 10 step size is increased to .00004 which has improved rate of convergence a bit but MMSE has also increased. Similarly other results are designed for step size .00020 which is shown in figure 11, with step size .00500 in figure 12 and with step size .01 in figure 13.

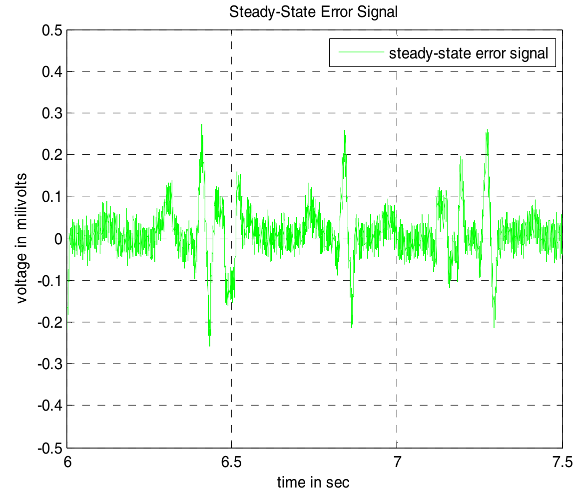

Fig. 10 15 taps filter with step size .00004

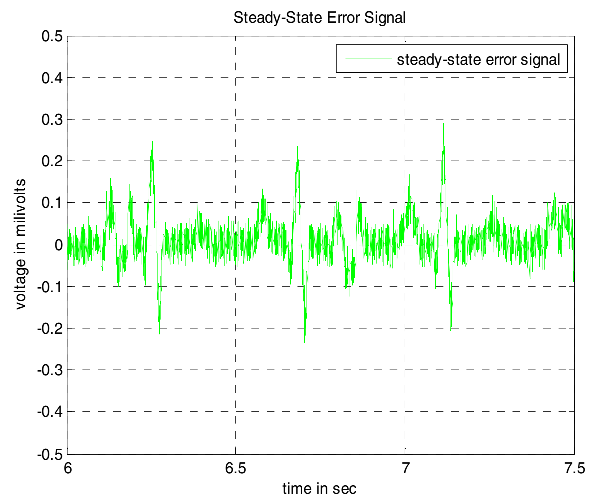

Fig. 11 15 taps filter with step size .00020

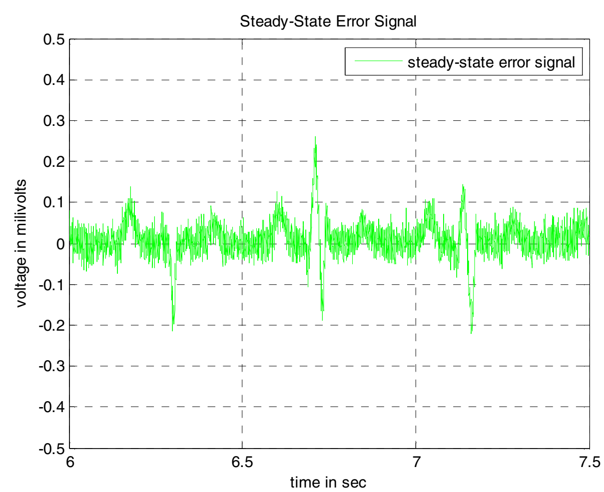

Fig. 12 15 taps filter with step size .00500

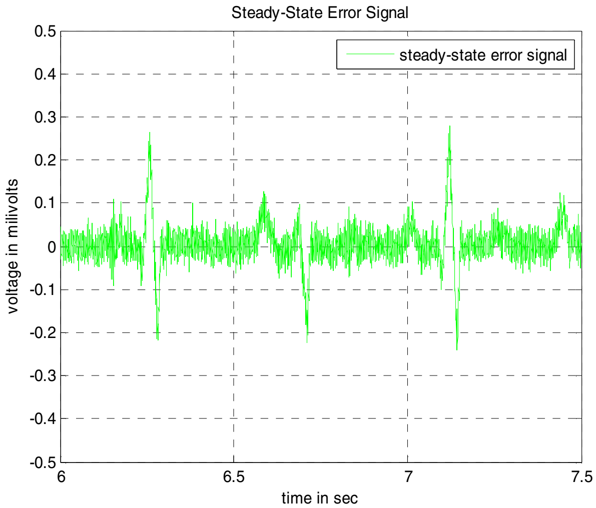

Fig. 13 15 taps filter with step size .01

IV. CONCLUSION

This paper presents the designing of adaptive filter using LMS algorithm for reducing noise in the Electrocardiogram waveform so to generate a waveform with reduced noise level. Thus from the above results it can be concluded that as the step size is increasing the noise is also increasing as well as the rate of convergence is also increasing. The rate of convergence and MMSE is having trade off between each other i.e. the optimum result is produced when the step size is considered to be as small as possible but it should also be kept in mind while designing adaptive filter that step size should not be taken very small because it will increase the value of rate of convergence. Future work can be done by using other LMS algorithms of adaptive filters.

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