Impact Factor (SJIF): 4.542

International Journal of Advance Research in Engineering, Science & Technology

e-ISSN: 2393-9877, p-ISSN: 2394-2444

Volume 4, Issue 7, July-2017

Design & Realization of Adaptive Noise Canceller on TMS320 C6713 Digital Signal Processor

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Abstract — One of the main problems in bio-medical data processing like electrocardiography is the separation of the wanted signal from the noises caused by power line interference, external electromagnetic fields, random body movements and respiration. Different types of digital filter are used to remove signal components from unwanted frequency ranges. It is difficult to apply filter with fixed coefficient to reduce the bio-medical signal noises, because human behavior is not exact known depending on the time. Adaptive filter technique is required to overcome this problem. An Adaptive noise canceller based on an improved algorithm has been designed by DSP. The algorithm is introduced and simulated in MATLAB, then realized on the digital signal processor.

Keywords- Electrocardiography, Adaptive noise canceller, DSP, LMS, FXLMS, SDLMS, PLI

I. INTRODUCTION

Noise problems in the environment have gained attention due to the tremendous growth of technology that has led to noisy engines, heavy machinery, high electromagnetic radiation devices and other noise sources. The problem of controlling the noise level has become the focus of a vast amount of research over the years. Bernard Widrow et.al. Developed a model for noise cancellation with the help of adaptive filter and employed for variety of practical applications like the canceling of various forms of periodic interference in electrocardiography, the canceling of periodic interference in speech signals, and the canceling of broad-band interference in the side-lobes of an antenna array. Power line interference coupled to signal carrying cables is particularly troublesome in medical equipment such as electrocardiograms (ECG). Cables carrying ECG signals from the examination room to the monitoring equipment are susceptible to interference of power frequency (50 Hz or 60 Hz) by ubiquitous supply lines and plugs noise that sometimes the ECG signal is totally masked. Filtering such interference signal is a challenging problem given that the frequency of the time-varying power line signal lies within the frequency range of the ECG signal. There are some other technical difficulties involved, the most important of which is the low sampling frequency at which the ECG signals are taken and the low computational resources available at the level of the device [1].

In the most of practical applications Adaptive filters are used and preferred over fixed digital filters because adaptive filters have the property of self-modifying its frequency response and allowing the filter to adapt the response as the input signal characteristics change.

The general configuration for an Adaptive filter system is shown in Fig.1

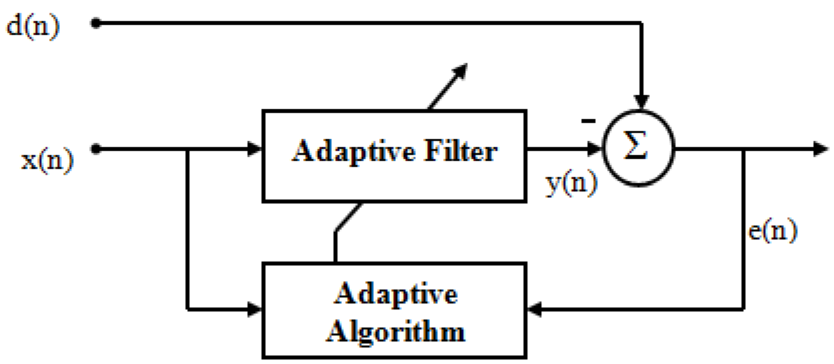

Figure 1 Adaptive Noise Canceller block diagram

It has two inputs: the primary input d(n), which represents the desired signal corrupted with noise, and the reference signal x(n), which is the noise to be filtered out of the system.
The goal of adaptive filtering systems is to reduce the noise portion, and to obtain the uncorrupted desired signal. In order to achieve this, a reference of the noise signal is needed and is called reference signal x(n). However, the reference signal is typically not the same signal as the noise portion of the primary signal; it can vary in amplitude, phase or time. Therefore the reference signal cannot be simply subtract from the primary signal to obtain the desired portion at the output. 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 x(n), which contains a solid reference of the noise present in the primary signal. The noise in the reference signal is filtered to compensate the amplitude, phase and time delay and then subtracted from the primary signal. The resulting signal is called an error signal e(n), and is the output of the system. Ideally, the resulting error signal would be only the desired portion of the primary signal. The adaptive filter can be realize on DSP Processors because they have huge number of applications in today's life, such as audio signal processing, image signal processing, statistical signal processing, and bio-medical signal processing. DSP is widely used in high speed modems and mobile phones also due to availability of low cost DSP chips that can perform extensive computation in real-time.
In this project the adaptive noise canceller is implemented on Texas DSP hardware [8]-[10] and tested for ECG signal record from MIT-BIH Database. The obtained results from DSP kit are analyzed with the help of MATLAB (2007a) and shows considerable improvement in SNR level of a filtered signal. Here we are used Least mean square (LMS) and filter-X adaptive filter are implemented on DSP & MATLB and compare the performance of both this results.

II. LITERATURE REVIEW

Power line is most usual source of interference in the ECG recording. This kind of interference is caused due to power line cords nearby and its effect can be minimized by moving away from such sources of this noise. However, there must be provision in the wearable ECG equipment to minimize the interference. As we know that the power lines have a specific frequency of either 50 or 60Hz. Therefore, the interference can be removed by using a narrow stop band filter centered at the power line frequency. The frequency response of the ECG equipment is usually from 0.05-100Hz. Two different approaches have been proposed in literature for this purpose notch filters and adaptive interference cancellers. Notch filters reduce the power line interference by suppressing predetermined frequencies. Usually, an infinite impulse response (IIR) filter is adopted. However, this leads to problems whenever the power line frequency is not stable or not accurately known, a mismatch between the suppression band and the power line frequency might lead to inadequate reduction of the power line interference. An ideal EMI filter for ECG should act as a sharp notch filter to eliminate only the undesirable power line interference while automatically adapting itself to variations in the frequency and level of the noise. And explain the different adaptive noise canceller.

In this paper, an effective adaptive filter structure is proposed to minimize the residual power-line interference without loss of reality. In order to obtain a satisfactory and acceptable convergence performance, Walsh Hadamard Transform (WHT) is used in the adaptive filter. The backbone is an adaptive cancel filter. The corrupted ECG is the primary input. From this, the component that is correlated with noise is the reference input. To obtain this correlated signal, the signal is processed by FFT and extracted the significant interference bandwidth spectrum. Then IFFT is performed to recover the interference signal and used as the reference input. After transformation the Eigen values are group into M-point outputs and adaptively processed by M stages of sub-band adaptive filters using NLMS algorithm. Throughout many clinical measurements, the result of this structure is effective in eliminating EMI interference [1].

The method proposed in this paper is non–stationary tracking approach for the power line interference by means of Kalman filtering that allows improved discrimination between the ECG signal and the noise. This method properly reduces this interference during its stationary segments and keeps its high performance during amplitude and frequency variations. One of the main shortcomings of the Kalman filter methods, the parameter setting, was proposed to be solved with the implementation of a genetic algorithm obtaining a set of parameters, being optimal in the correlation index sense. The search surface shows that the EKF power line interference suppressor has high performance in a large span of the parameter space [2].

In this paper, a novel algorithm was proposed to subtract power line interference (PLI) from contaminated ECG signal. Firstly, some PLI values were calculated from ECG signal in linear segment. Secondly, using nonlinear regression and least squares estimation, sinusoidal parameters such as frequency and phase were estimated from the calculated PLI values. Thirdly, the PLI values in the nonlinear segment closest to the linear segment were determined by the sinusoid function with the estimated parameters. Lastly, PLI was subtracted from the contaminated ECG signal in nonlinear segment. The experiment results show that the proposed method can more effectively remove PLI from ECG signal than other algorithms when PLI frequency fluctuates, and do not distort the shape of ECG signal [3].

III. METHODOLOGY

3.1 Electrocardiography

ECG is the process of recording the electrical activity of the heart over a period of time using electrodes placed on the skin. These electrodes detect the tiny electrical changes on the skin that arise from the heart muscles electro physiologic pattern of depolarizing and re-polarizing during each heartbeat .It is a very commonly performed cardiology test.

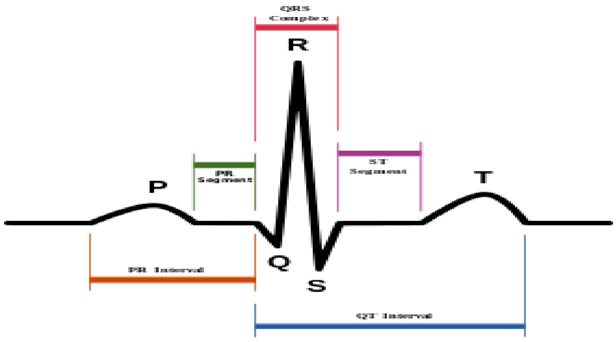

Figure 2 . ECG waveform

3.2 Algorithms for Noise Removal

3.2.1 Noise Cancellation Algorithms

Digital signal processing techniques have rapidly developed in the recent days due to developments in the digital computer technologies and IC fabrication. The use of digital system yields high speed, high reliability and also permits programmable operations. DSP techniques find application in a variety of areas such as speech processing, data transmission on telephone channels, image processing, Bio-medical signal processing and in the processing of signals received from the outer space. In this chapter various adaptive algorithms are discussed that can be applied to noise ECG signal in order to remove PLI and AWGN in MATLAB as well as on TMS320c6713 DSK Processor. Compare the results of both type algorithms in terms of SNR improvement and correlation coefficients, Peak signal to noise ratio for the different records of ECG database and Different frequency PLI with step size parameter u=0.3. Following are the Adaptive noise canceller algorithms used for the analysis and compare their filter performance.

3.2.2 Least Mean Square (LMS) Algorithm

The LMS algorithm is widely used in different application because of low computational complexity , stability and good convergence .The least mean squared algorithm is introduced to minimized the error between a given signal and output of linear filter by adjusting the parameter of linear filter . This algorithm is having its importance due to its stability of faster converged and can range of problem setting, computational restriction and minimum criteria.. Overview of LMS algorithm is shown in fig.1

The PLI x(n) has been taken is the difference input and according to incoming noise signal coefficient are updates control the error signal e(n) . The error signal manipulated by the adaptive algorithm is describe as

$$e(n) = d(n) - y(n) \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (3.1)$$

The equation above shows the desired signal and the filter output, where d(n) is the desired signal and y(n) is the filter output. LMS algorithm works according to the criterion that is supposed to minimize error signal. The input vector is used to update the adaptive coefficient according to following equation.

$$w(n) = w(n-1)+2ue(n-1)x(n-1) \text{----------} (3.2)$$

3.2.3 Filter-x Least mean squared algorithm (FX-LMS)

To facilitate the development of the block filtered-XLMS algorithm, we considered a length L Least mean square (LMS) based adaptive filter shown in Fig. 1 that takes an input sequence x(n) and updates the weights as

$$\mathbf{w} = \mathbf{w}(n+1) + \mu\, \mathbf{x}(n)\, e(n),$$

where,

$$x(n) = [x(n)\ x(n-1) \ldots x(n-L+1)]t,$$

$$e(n) = d(n) - wt(n)\, x(n),$$

In order to remove the noise from the ECG signal, the ECG signal x(n) with additive noise is applied as the desired response d(n) for the adaptive filter. If the noise signal p2(n), possibly recorded from another generator of noise that is correlated in some way with p1(n) is applied at the input of the filter, i.e., x(n) = p2(n) the filter error Becomes

$$e(n) = [s1(n) + p1(n)] \text{ ¡ } y(n) \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots(3.3)$$

The filter output y(n) is given by,

$$y(n) = \mathbf{w}t\,(n)x(n), \ldots\ldots\ldots\ldots\ldots\ldots(3.4)$$

In FXLMS algorithm the filtered version of x(n) is used for weight update process, i.e., the forward path is introduced between the input signal and the algorithm for the adaptation of the coefficient vector. The transfer function of the forward path is assumed to be an I-th order finite impulse response (FIR) system Now the estimation error e(n) can be written as According to the FXLMS algorithm, the filter coefficients are adapted according to the following recursion:

$$w(n+1)= w(n)+ \mu x'(n)e(n) \ldots\ldots\ldots\ldots\ldots\ldots\ldots(3.5)$$

where

$$x'(n) = s(n) * x(n)\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots(3.6)$$

The output of the adaptive filter is computed as,

$$x''(n) = w(n) * x'(n) \ldots\ldots\ldots\ldots\ldots\ldots(3.7)$$

3.2.4. Sign Data LMS Algorithm (SDLMS)

This is a type of Improved LMS algorithm in which sign function is used to replace the input signal and error signal this will reduce the computation and difficulty of implementation in algorithm. In SDLMS algorithm we are not using the input signal but its sign only used. The weight of the filter can be calculated as follows [18]

$$w\,(n)= w\,(n-1) + 2\mu.\ e\,(n).\ \text{Sign}\,[x\,(n-1)]$$

In TMS 320 C6713 DSP System, the SDLMS algorithm of adaptive noise canceller can effectively resolve the problem of eliminating noise signal from the noisy signal and increases signal to noise ratio.

IV. IMPLEMENTATION ON DSP PROCESSOR

Digital signal processors are fast special-purpose microprocessors with a specialized type of architecture and an instruction set appropriate for signal processing. The architecture of the digital signal processor is very well suited for numerically intensive calculations. Digital signal processors are used for a wide range of applications which includes communication, control, speech processing, image processing etc. These processors have become the products of choice for a number of consumer applications, because they are very cost-effective and can be reprogrammed easily for different applications. DSP techniques have been very successful because of the development of low-cost software and hardware support. DSP processors are concerned primarily with real-time signal processing. Real-time processing requires the processing to keep place with some external event, whereas non-real-time processing has no such timing constraint. The external event is usually the analog input. Analog-based systems with discrete electronic components such as resistors can be more sensitive to temperature changes whereas DSP-based systems are less affected by environmental conditions. In this chapter we will learn how we can realize or implement an adaptive filter on hardware for real-time experiments. The model which was designed in previous chapter will be linked to the DSP processor with help of code composer studio v 3.1.

4.1 Introduction to Digital Signal Processor TMS320C6713

The TMS320C6713 is a low cost board designed to allow the user to evaluate the capabilities of the C6713 DSP and develop C6713-based products. It demonstrates how the DSP can be interfaced with various kinds of memories, peripherals, Joint Text Action Group (JTAG) and parallel peripheral interfaces. The board is approximately 5 inches wide and 8 inches long as shown in Fig and is designed to sit on the desktop external to a host PC. It connects to the host PC through a USB port. The processor board includes a C6713 floating-point digital signal processor and a 32-bit stereo codec TLV320AIC23 (AIC23) for input and output. The on board codec AIC23 uses a sigma–delta technology that provides ADC and DAC. It connects to a 12-MHz system clock. Variable sampling rates from 8 to 96 kHz can be set readily. A daughter card expansion is also provided on the DSK board. Two 80-pin connectors provide for external peripheral and external memory interfaces. The external memory interface (EMIF) performs the task of interfacing with the other memory subsystems. Light emitting diodes (LEDs) and liquid-crystal displays (LCDs) are used for spectrum display. The DSK board includes 16MB (Megabytes) of synchronous dynamic random access memory (SDRAM) and 256kB (Kilobytes) of flash memory. Four connectors on the board provide inputs and outputs: MIC IN for microphone input, LINE IN for line input, LINE OUT for line output, and HEADPHONE for a headphone output (multiplexed with line output). The status of the four users DIP switches on the DSK board can be read from a program and provides the user with a feedback control interface. The DSK operates at 225MHz.Also on board are the voltage regulators that provide 1.26 V for the C6713 core and 3.3V for its memory and peripherals.

4.2 DSK Code Composer Studio

Code composer is the DSP industry's first fully integrated development environment with DSP Specific Functionality with familiar environment like MS – Base C++ ,code composer let's you edit ,build, debug ,profile and manage projects from single unified environment. Other unique feature include graphical signal analysis , injection/extraction of data signal via file I/O multi-processor, Debugging automated testing and customization via C- interpretive scripting language and much more

Software - Designers can readily target the TMS32C6713 DSP through TI´s robust and comprehensive Code Composer Studio DSK development platform. The tools, which run on Windows© 98, Windows 2000 and Windows XP, allow developers to seamlessly manage projects of any complexity.

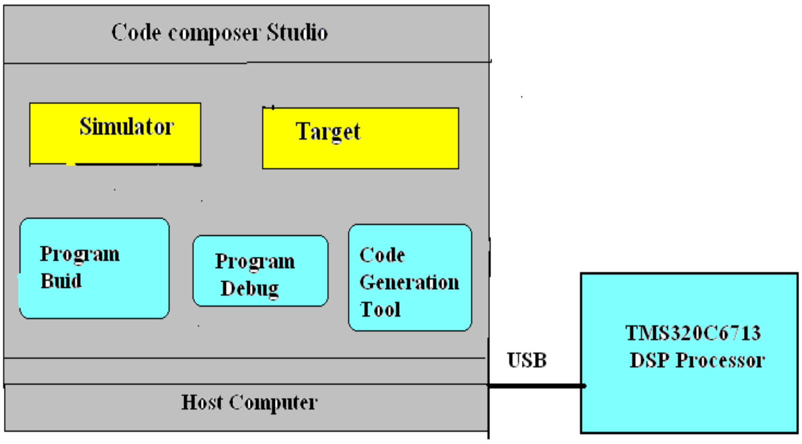

Figure 4. Code Composer Studio Platforms

V. EXPERIMENTAL RESULT

LMS and FX-LMS algorithms are implemented in MATLAB R2007a environment and TMS320C6713DSK Processor. The ECG waveform taken from MIT-BIH database, generated noises and the corrupted ECG signals are also shown. The adaptive filtering techniques are selected with the objective of removing 50/60 Hz noise from ECG signal. The code is written for acquiring clean ECG signal from MIT-BIH ECG_ID Database for different person records further it get corrupted with PLI having variations in frequency, amplitude. The ECG signals are of 1 minute duration with sampling frequency 360 Hz. The simulation results are analyzed for different values of step size parameter to achieve the desire results of LMS and FX-LMS for step size parameter (u=0.3)

To measure the performance of different algorithms at different values of PLI frequency(48hz,50hz,52hz) ,we used signal to noise ratio (SNR dB) or in some time it may called signal to noise interference ratio (SIR dB) , correlation coefficient (C.C.) between the ECG and Filtered ECG. SNR_{in} is the ratio of power of pure ECG signal to the power of noise signal. SNR_{out} is defined as power of pure ECG signal to power of residual error. The performance of LMS AND FX-LMS algorithm can be done by taking the average of 3different ECG signal from database (Person_01rec.m/ Person_02rec.m/ Person_03rec.m) having the different frequency values . Following are the graphs taken from both the MATLAB and DSP result and table shows the performance parameter.

Table 1 shows SNR Improvement in dB for LMS and FXLMS algorithm. In all experiments filter output and system parameters are initialize to zero i.e. w(0)=y(0)=e(0)=0 etc. By analyzing the table we can conclude that the SNR improvement in FXLMS is more than LMS algorithm on both the environment, while LMS algorithm gives approximately same improvement on DSP as well as MATLAB. Table 2 shows Correlation Coefficient for LMS and FXLMS algorithm between the original ECG and filter ECG. Here by observing the result, FX-LMS has better correlation coefficient than LMS algorithm.

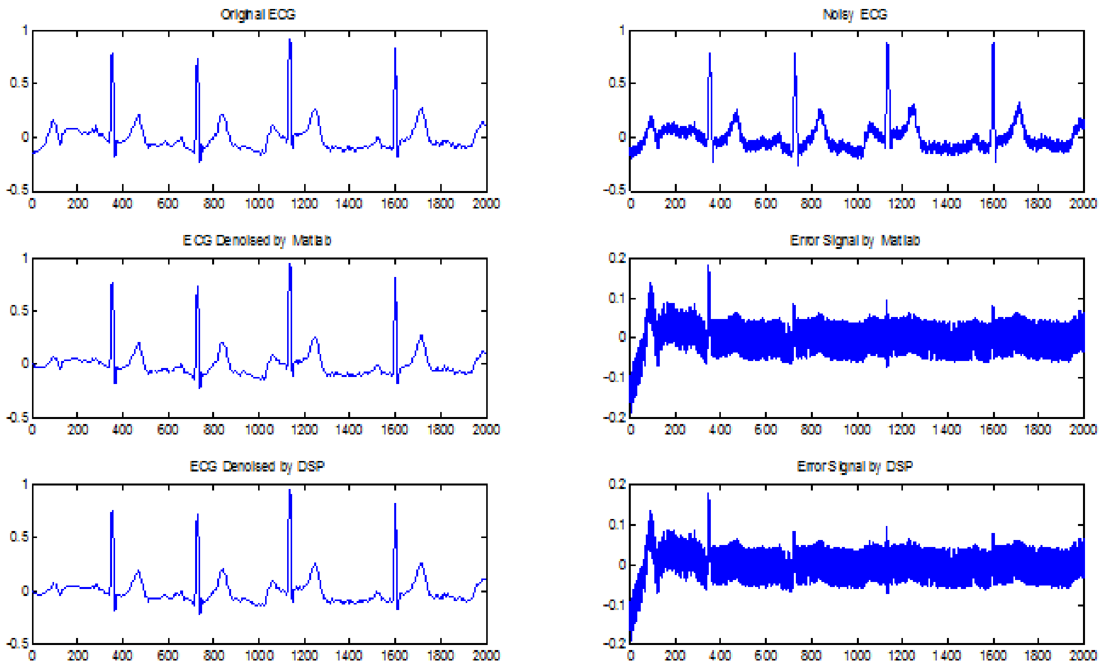

figure 5. Person_01.m / 50 Hz (PLI) / LMS algorithm/ u= 0.3

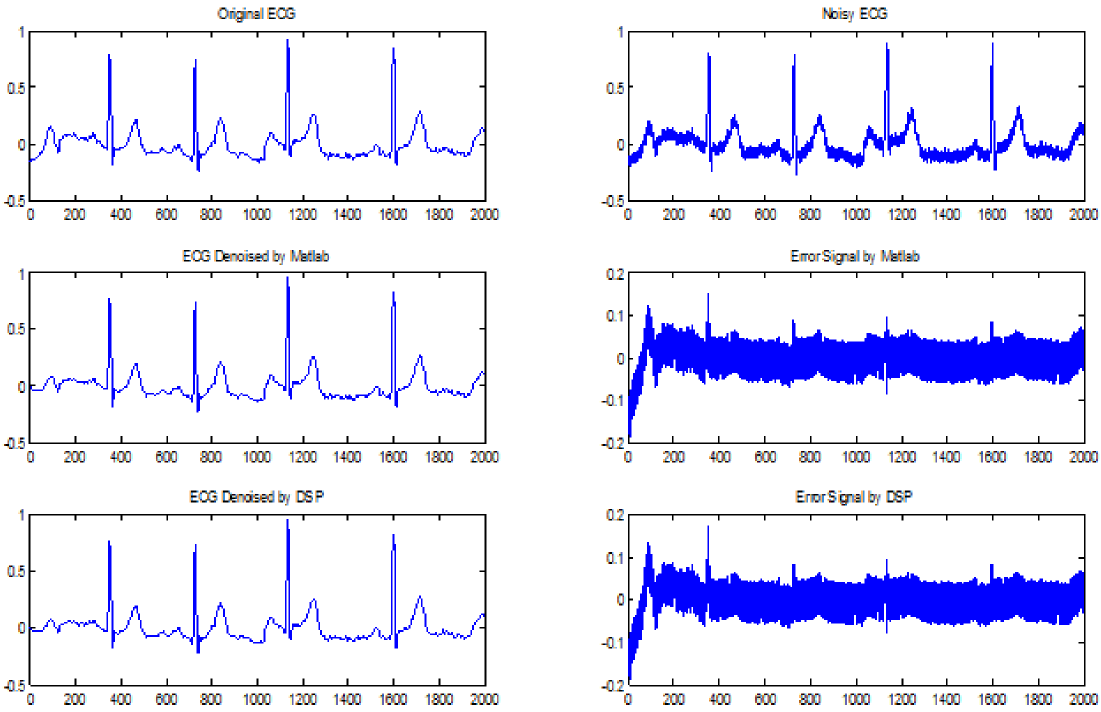

Figure 6. Person_01.m / 50 Hz (PLI) / FX-LMS algorithm/ u= 0.3

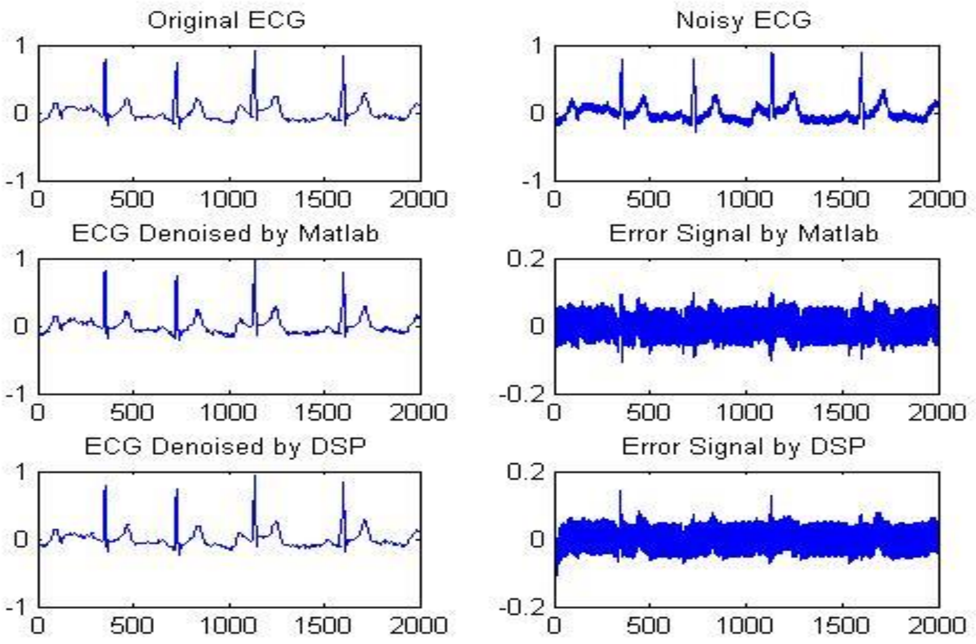

Figure 7. Person_1.m/PLI/ SDLMS Algorithm

Table 1 SNR Improvement of LMS, FXLMS and SDLMS algorithm

Sr. No	Algorithms	SNR IMPROVMENT IN Db	
		MATLAB	**DSP**
1	LMS	16.8933	16.9253
2	FXLMS	17.5655	17.0991
3	SDLMS	20.4954	20.9264

Table 2 Correlation Coefficient of LMS, FXLMS and SDLMS

Noise Frequency	Correlation Coefficient					
	MATLAB			DSP		
	LMS	FXLMS	SDLMS	LMS	FXLMS	SDLMS
48 Hz	0.9907	0.9924	0.9955	0.99073	0.9911	0.9966
50 Hz	0.9902	0.9924	0.9956	0.9908	0.9911	0.9965
52 Hz	0.9907	0.9924	0.9955	0.9907	0.9911	0.9963
Avg	0.99053	0.9924	0.9955	0.99074	0.9911	0.9964

VI. CONCLUSION

This paper is devoted to noise cancellation from ECG signal. The implementation of adaptive noise cancellation algorithms like LMS, FX-LMS & SDLMS done successfully, For denoising of ECG signal in MATLAB R2007.1a environment and TMS320C6713DSK Processor. The designed system is tested for different ECG signal and Noise level. The result analysis shows that SDLMS algorithm gives better improvement than LMS and FXLMS algorithm in both environment MATLAB as well as DSP processor in terms of SNR improvement Correlation Coefficient and PSNR. A fair amount of SNR improvement is achieved in both the cases.

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