Adaptive Filter Design for Noise Reduction in Electrocardiography

[1]Anamika Sharma, [2]Rajesh Mehra
[1]ME Scholar, [2]Associate Professor,
[1,2]Department of Electronics & Communication Engineering
National Institute of Technical Teachers' Training & Research, Chandigarh, India

ABSTRACT
Electrocardiogram is commonly used to detect abnormal heart rhythm. An electrocardiogram (ECG) records the electrical activity of the heart. ECG signal is corrupted by noise signal which restricts the correct diagnose of heart beats. Adaptive filters are used to reduce the noise signal from the ECG signal. In this paper an Adaptive filter has been designed to eliminate the noise from ECG signal by using LMS algorithm. In diagnosing the Fetus heartbeat, mother's heartbeat can interfere as a noise. The designed filter will reduce this noise from measured signal by a reference signal which is highly correlated with the noise signal. It can be observed from simulated results that developed filter can separate out noise from fetus heartbeat signal.

Keywords: *Adaptive Filter, DSP, ECG, FIR, IIR*

I. INTRODUCTION
Digital signal processing (DSP) is the process of analyzing and modifying a signal to optimize or improve its efficiency or performance. It involves applying various mathematical and computational algorithms to analog and digital signals to produce a signal that's of higher quality than the original signal. DSP is primarily used to detect errors, and to filter and compress analog signals in transit. It is a type of signal processing performed through a digital signal processor or a similarly capable device that can execute DSP specific processing algorithms. Typically, DSP first converts an analog signal into a digital signal and then applies signal processing techniques and algorithms. For example, when performed on audio signals, DSP helps to reduce noise and distortion.The main applications of DSP are audio signal processing, audio compression, digital image processing, video compression, speech processing, speech recognition , digital communications, RADAR, SONAR, and Financial Some of the applications of DSP include audio signal processing, digital image processing, speech recognition, digital communication, RADAR, SONAR[1].

One of the most widely used complex signal processing function is filtering, whose main objective is to alter spectrum according to given specifications. The system implementing this operation is filter, the filter may be designed to pass a certain frequency components in a signal through system and to block other frequency components .The frequency range allowed to pass through filter is pass band and the blocked range of frequency is stop band. Various types of filters can be defined depending on the nature of filtering operations. Filtering operation for analog signals is performed by LTI filter. There are various methods to remove the noise of ECG signal by FIR, IIR and Adaptive filters each has its own advantages and disadvantages. FIR has finite Impulse response but it has more number of coefficients so, it requires more memory space to store its coefficients. IIR filter has less coefficients but it is unstable sometimes because there is a feedback loop [2].
Linear filtering is required in many applications. A filter will be optimal only if it is designed with some knowledge about the input data. If this information is not known, then adaptive filters are used. Adaptive filter finds its application in Adaptive noise cancelling, Line enhancing, frequency tracking, Channel equalizations, Echo cancellation in audio signals, Linear prediction etc [3].

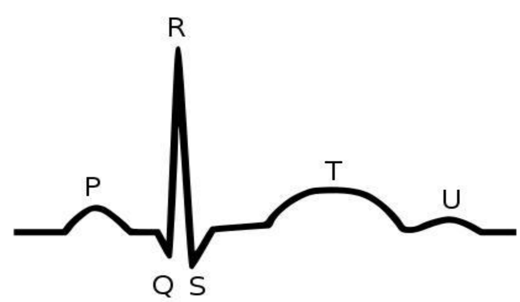

Fig. 1 ECG Waveform

Cardiac cycle consists of one beat or P–QRS-T sequence. The cardiac cycle is measured on ECG from one R wave to next R wave. The P wave shows the first deflection of the cardiac cycle. PR interval represents the time required for electrical impulse to leave the SA node and travel through the atria, AV node. QRS complex represents ventricular de-polarization. ST segment represents the end of ventricular conduction. T wave represents ventricular recovery or re polarization. U wave represents the recovery period of fibers. Above figure shows general ECG waveform. In biomedical application of a dsp it is used to diagnose the heartbeats.

II. ADAPTIVE FILTER

The usual method of estimating a signal corrupted by additive noise is to pass it through a filter that tend to reduce the noise while leaving the signal unchanged. The design of such a filter is the domain of optimal filtering. Adaptive filters have the ability to adjust their own parameters automatically, and their design requires little or no priori knowledge of signal characteristics [3]. Noise cancelling is the variation of optimal filtering that is highly advantageous in various applications. It makes use of a reference input signal is used. The purpose of an adaptive filter in noise cancellation is to remove the noise from a signal adaptively Fig.1 shows the diagram of a typical Adaptive Noise Cancellation (ANC) system. Here the adaptive filter is used to cancel the unwanted signal in a primary signal, with the cancellation being optimized in some sense. The primary signal serves as the desired response for adaptive filter. A reference is employed as the input to the filter [6].

In this paper an adaptive filter is designed for biomedical application of dsp by using LMS algorithm. Generally, the recorded ECG is contaminated by noise and artifacts that can be within frequency band of interest. Here maternal heartbeat signal is adaptively removed from the fetus heartbeat signal by ANC [4].

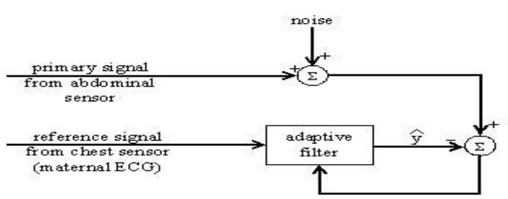

Fig. 2 Adaptive Filter

Here primary signal from abdominal sensor is corrupted by noise signal and a reference signal from chest sensor or maternal ECG are applied to adaptive filter. Adaptive filter separates the noise signal from the original signal. Adaptive filter works on LMS algorithm to reduce noise. This algorithm is proposed by Window and Hoff [5].

LMS Algorithm: The LMS algorithms a linear adaptive filtering algorithm, which consists of two processes A filtering process, which involves computing the output of a linear filter in response to an input signal and generating an estimation error by comparing this output with a desired response. An adaptive process, which involves the automatic adjustment of parameters of the filter in accordance with estimated error [5].

$$h_M(n+1) = h_M(n) + \Delta e(n)\, X^*_M(n)$$

III. ADAPTIVE FILTER AND SIGNALS TYPE

Heartbeat Maternal Mother's heart might produce assuming a 4000 Hz sampling rate .The heart rate for this signal is 89 beats per minute, and the amplitude of the 2mV.

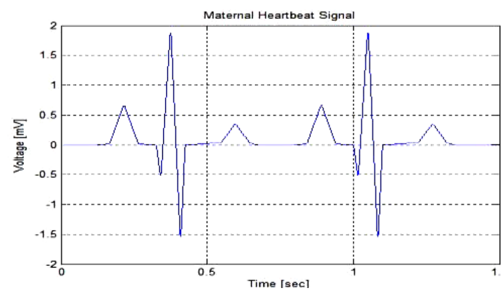

Fig.3 Maternal Heartbeat

The heart of a fetus beats noticeably faster than that of its mother, with rates ranging from 120 to 160 beats per minute. The amplitude of the fetal electrocardiogram is also much weaker than that of maternal ECG.

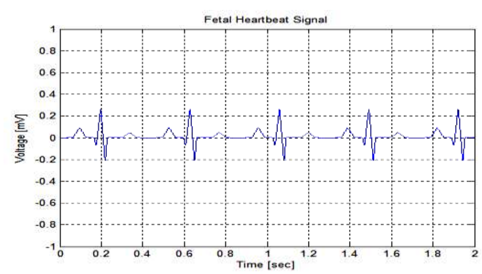

Fig.4 Fetal Heartbeat

The measured fetal electrocardiogram signal from the abdomen of the mother is usually dominated by the maternal heartbeat signal that propagates from the chest cavity to the abdomen. This propagation path is described as a linear FIR filter, and an amount of uncorrelated Gaussian noise is added.

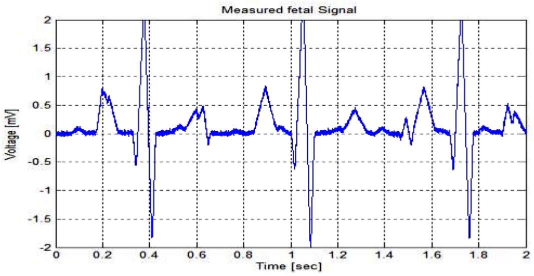

Fig.5 Measured Fetal ECG

The maternal electrocardiogram signal is obtained from the chest of the mother. The goal of the adaptive noise canceller is to adaptively remove the maternal signal from the fetal ECG. A reference signal generated from a maternal electrocardiogram .Fetal heartbeat is recovered from the maternal heartbeat signal. Fetal heartbeat is much more than the mother's heartbeat signal. Maternal ECG and Fetal ECG both are contaminated with noise, by applying ANC noise as well as maternal signal is removed from fetal signal.

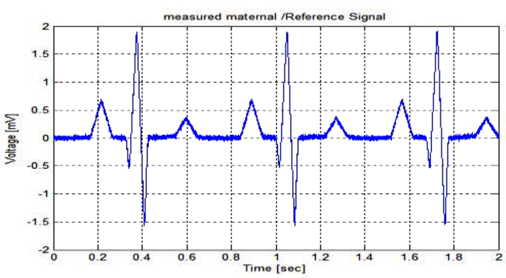

Fig.6 Measured Maternal ECG

IV. NOISE SEPARATION

The output of adaptive filter contains fetal signal and noise. Steady state error signal contains both fetal signal and noise signal.

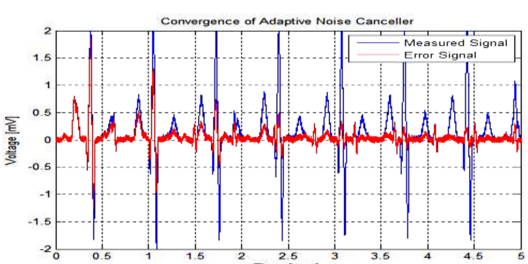

Fig.7 Measured ECG with Noise

The adaptive filter use the least-mean-square algorithm, (LMS) adaptive filter with 15 coefficients and a step size of 0.00007.

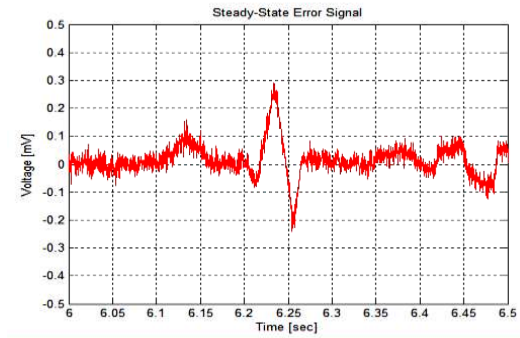

Fig.8 Fetal Signal with Noise

The error signal contains both ECG fetal and noise signal. Adaptive filter is used to separate out the noise and finally original fetal signal is recovered through Mat lab.

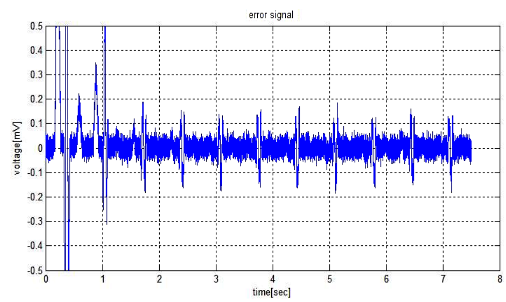

Fig.9 Noise Signal

The noise is removed from the fetal heartbeat and we get the desired fetal signal by adaptive noise cancelling.

Fig.10 Recovered Fetal Signal

V. CONCLUSION

From the above results and discussion the original fetal signal is recovered from the noise signal by applying adaptive noise canceller through LMS. Here primary signal from abdominal sensor is corrupted by noise signal and a reference signal from chest sensor or

maternal ECG are applied to adaptive filter. Adaptive filter separates the noise from the fetal ECG.
Maternal as well as noise signals are removed from the fetal ECG.

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AUTHORS

Anamika Sharma received the Bachelors of Technology degree in Electronics and Communication Engineering from College of Engineering and Rural Technology, Meerut, India in 2010. She is pursuing Master of Engineering in Electronics and Communication Engineering from National Institute of Technical Teachers' Training & Research, Panjab Univsrsity, Chandigarh, India. She is an Assistant Professor with the Department of Electronics & Communication Engineering, ABSS Institute of Technology, Meerut, India. Her current research and teaching interests are in Electronics Engineering and in signal processing.

Rajesh Mehra, received the Technology degree in Electronics and Communication Engineering from National Institute of Technology, Jalandhar, India in 1994 , Masters of Engineering in Electronics and Communication Engineering from National Institute Of Technical Teachers' Training & Research, Panjab Univsrsity, Chandigarh, India in 2008. He is pursuing Doctor of Philosophy degree in Electronics and Communication Engineering from National Institute of Technical Teachers' Training & Research, Panjab Univsrsity, Chandigarh, India. He is an Associate Professor with the Department of Electronics & Communication Engineering, National Institute of Technical Teachers' Training & Research, Ministry of Human Resource Development, Chandigarh, India. His current research and teaching interests are in Signal, and Communications Processing, Very Large Scale Integration Design. He has authored more than 175 research publications including more than 100 in Journals. Mr. Mehra is member of IEEE and ISTE.