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Designing of Digital Adaptive Filter for Removal of Artifacts in PCG Signal

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Abstract: The objective of this paper is to serve as how Noise can be combated using adaptive filter for PCG signal. The problem of controlling the noise level has been one of the research topic over the years. This paper focuses on Adaptive filtering algorithms and some of the applications of adaptive filter. The main concept is to use the LMS (Least-Mean-Square) algorithm to develop an adaptive filter that can be used in Adaptive noise Cancellation (ANC) application. In this paper we will learn the various algorithms of LMS (Least Mean Square), NLMS (Normalized Least Mean Square) and RLS (Recursive Least Square) on MATLAB platform with the intention to compare their performance in noise cancellation. The adaptive filter in MATLAB with a noisy tone signal and white noise signal and analyze the performance of algorithms in terms of MSE (Mean Squared Error), percentage noise removal, Signal to Noise Ratio, computational complexity and stability. The Adaptive Filter maximizes the signal to noise ratio & minimize the Mean Squared Error and compare their performance with respect to stability. Adaptive Noise Canceller is useful to improve the S/N ratio. The Adaptive Filter minimizes the mean squared error between a primary input, which is the noisy PCG, and a reference input, which is either noise that is correlated in some way with the noise in the primary input or a signal that is correlated only with PCG in the primary input.

Keywords: LMS (Least Mean Square), NLMS (Normalized Least Mean Square), RLS (Recursive Least Square), MSE (Mean Squared Error).

I. INTRODUCTION

The noise cancellation technique is a part of Optimal filtering that can be applied only when we have predetermined knowledge about the reference noise level.. Some of its applications are:- Speech processing, echo cancellation and enhancement, antenna array processing, biomedical signal and image processing and so on. There are numerous denoising techniques used in speech processing. Most of them include hypotheses on the original signal, as well as SNR ratio and distortion. However these techniques do not cover all the explicit speech models. Each of them is associated with a particular type of distortion while maximizing noise-reduction effects. There have been several methods used to study the noise cancellation problems. One of the basic and important noise cancellation methods is adaptive filtering. Adaptive filters have several applications in acoustics, controls, communications, and coding.

Its structure varies from a very simple to complex one. A Digital communication system consists of a transmitter, channel and receiver connected together. The channel has two major problems, Namely, Intersymbol interference and Noise. The basic principle of noise cancellation is to have an estimate of the interfering signal and subtract it from the corrupted signal. Adaptive noise cancellation is an interference cancellation technique in itself which relies on the use of noise cancellation by subtracting noise from a received signal, an operation controlled in an adaptive manner for the purpose of improved signal to noise ratio.

II. PHONOCARDIOGRAM (PCG): A NEW BIOMETRIC

A Phonocardiogram or PCG is a plot of high fidelity recording of the sounds and murmurs made by the heart with the help of the machine called phonocardiograph, or "Recording of the sounds made by the heart during a cardiac cycle." These sounds are due to vibrations created by closure of the heart valves. There are at least two: the first is when atrio ventricular valves close at the beginning of systole and the second one is when the aortic valve and pulmonary valve close at the end of systole. PCG detects these sub audible sounds and murmurs, and makes a permanent record of these events. In contrast, the ordinary stethoscope cannot detect such sounds or murmurs, and provides no record of their occurrence. This measurement of the sounds made by the heart provides information not readily available from more sophisticated tests, but at the same time it also provides vital information about the effects of certain cardiac drugs upon the heart. It is also an effective and important method for tracking the progress of the patient's disease.

Heart auscultation is a fundamental tool in the diagnosis of heart diseases. But now a day it has been less focused due to the emergence of ECG and echocardiography; still there are some cardiac defects that are best detected by heart sounds.

The human heart is a four-chamber pump with two auricles for the collection of blood from the veins and two ventricles for pumping out the blood to the arteries. The

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mechanical functionality of the cardiovascular system is governed by an electrical signal originated in specialized pacemaker cells in the right atrium (the sino-atria node), and is propagated through the atria to the AV-node (a delay junction) and to the ventricles. The periodic beating of the heart is due to complex interaction among pressure gradients, the dynamics of blood flow, and the compliance of cardiac chambers and blood vessels.

The flow of blood is controlled by two sets of valves control: the AV-valves (mitral and tricuspid) between the atria and the ventricles, and the semilunar valves (aortic and pulmonary) between the ventricles and the arteries. These mechanical processes results in vibrations and acoustic signals that can be recorded over the chest wall. The cardiac cycle events are demonstrated in Figure1.1

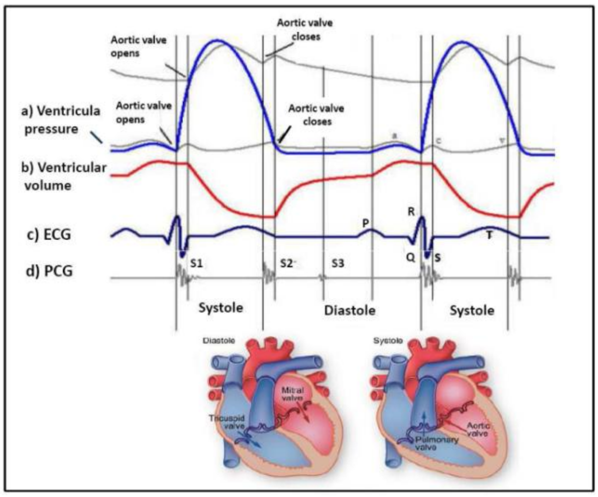

Figure 1.1: The cardiac cycle, (a) Ventricular Pressure, (b) Ventricular volume, (c) ECG trace, (d) PCG signal

The cardiac cycle consists of two periods, systole and diastole respectively. Both are periods of relatively high activity, alternating with comparatively long intervals of low activity. The major audible components of the PCG are short beats which are recognized as the primary components (S1, S2, S3, and S4). The other classes of sounds are murmurs, clicks, and snaps. However, the two major audible sounds in a normal cardiac cycle are the first and second heart sound, S1 and S2 as depicted in Figure 1.2.

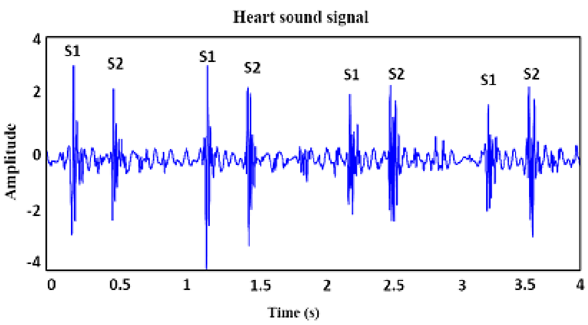

Figure1.2: Different components of a normal PCG signal.

S1: occurs at the onset of the ventricular contraction during the closure of the AV valves. It contains a series of low-frequency vibrations, and is usually the longest and loudest component of the PCG signal. The audible sub-components of S1 are those associated with the closure of each of the two AV-valves. S1 lasts for an average period of 100ms − 200ms and its frequency components lie in the range of 25Hz − 45Hz. It is usually a single component, but may be prominently split with some pathology.

S2: is heard at the end of the ventricular systole, during the closure of the semilunar valves. S2 lasts about 0.12s, with a frequency of 50Hz which is typically higher than S1 in terms of frequency content and shorter in terms of duration. It has aortic and pulmonary sub-components: A2 and P2 corresponding to the aortic part and pulmonary part respectively. Usually A2 and P2 are closed together, but a split S2 can occur if A2 and P2 are just far enough apart that they can be heard as two beats within S2.

S3: is the third low-frequency sound that may be heard at the beginning of the diastole, during the rapid filling of the ventricles. Its occurrence can be normal in young people (less than 35 years of age).

S4: is the fourth heart sound that may occur in late diastole during atrial contraction shortly before S1. It is always considered as an abnormality within the cardiac cycle.

Click and Snaps: are associated with valves opening and indicate abnormalities and heart defects. Opening snaps of the mitral valve or ejection sound of the blood in the aorta may be heard in case of valve disease (stenosis, regurgitation). The most common click is a systolic ejection click, which occurs shortly after S1 with the opening of the semilunar valves. The opening snap when present, occurs shortly after S2 with the opening of the mitral and tricuspid valves.

Murmurs: Is high-frequency, noise-like sounds that are heard between the two major heart sounds during systole or diastole. They are caused by turbulence in the blood flow through narrow cardiac valves or reflow through the atrio ventricular valves due to congenital or acquired defects. They can be innocent, but at times also indicate certain cardiovascular defects.
Pathologies of the cardiovascular system occur due to different etiologies, e.g. congenital heart valve defects, stenotic valve, and regurgitated valve. These pathologies affect the normal sounds with respect to intensity, frequency content, and timing of components (splitting).

In a medical context the heart sound signal is collected from four main regions on the chest wall as demonstrated in Figure 4.3. The aortic (A), between the second and third intercostal spaces at the right sternal border; mitral (M), near the apex of the heart between the fifth and sixth intercostal spaces in the mid-clavicular line; pulmonic (P), between the second and third intercostal spaces at the left sternal border; and tricuspid (T), between the third, fourth, fifth, and sixth intercostals space at the left sternal border.

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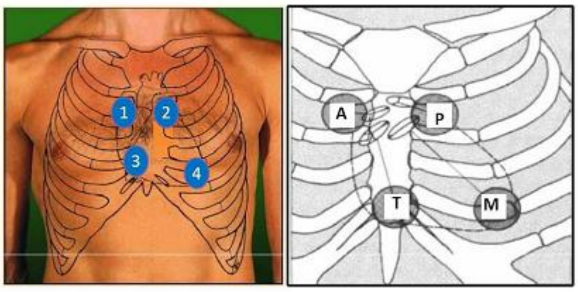

Figure 1.3: Four auscultations cites

However, presence of noise in recording the PCG signals is a major problem which is inevitable in practice. The noise contained in the PCG is introduced by a variety of sources such as ambient noise (e.g. background noise and stomach rumbling), respiratory sounds, thoracic muscular noise, peristaltic intestine noise, fetal breath sounds (if the subject is pregnant), non-optimal recording sites, weak sounds (obese patients), and so on.

III. ADAPTIVE FILTER: CONCEPT OF ADAPTIVE FILTER

The first work in Adaptive Noise Cancelling was done by two scientists Howells and Applebaum and their colleagues at General Electrical Company. They designed a system for antenna side lobe cancelling that used a reference input derived from an auxiliary antenna and a basic two-weight adaptive filter[1].
An adaptive filter is a system with a linear filter that has a transfer function controlled by variable parameters and a means to adjust those parameters according to an optimization algorithm. Because of the complex nature of the optimization algorithms, most adaptive filters are digital filters. Adaptive filters are required for some applications because some parameters of the desired processing operation are not known in advance or are changing continuously. The closed loop adaptive filter uses feedback in the form of an error signal to refine its transfer function.
The first adaptive noise cancelling system was designed at Stanford University and built in 1965. And its purpose was to cancel 60 Hz interference at the output of an electrocardiographic amplifier and recorder[1] .The basic concept of Adaptive noise cancelling is shown in figure.

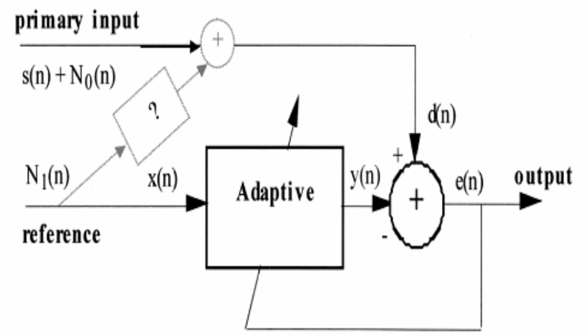

Fig1: Adaptive Noise Canceller

s(n) – Source signal
d(n) – Primary signal
No(n) – Noise signal
$N_1(n)$ – Noise reference input
y(n) – Output of Adaptive Filter
e(n) – System Output Signal

IV. ADAPTIVE FILTERING

Main of an adaptive filter in noise cancellation is to remove the noise from a signal adaptively to increase signal to noise (SNR) ratio and finally to improve the quality of signal.
A signal s(n) is transmitted through a channel to a sensor and also receives noise No(n) uncorrelated with the signal. The combined signal and noise (s(n) + No(n)) form the primary input to the canceller. Now a second sensor receives a noise $N_1(n)$ uncorrelated with the signal but correlated in some unknown way with the noise No(n). Now this sensor provides the reference input to the canceller. The noise $N_1(n)$ is filtered to produce an output y(n) that is in resemblance (not exactly) with No(n).

This output is subtracted from the primary input [s(n) + No(n)] to produce the output of the system.

$$e(n) = s(n) + No(n) - y(n)$$

The reference input is processed by an adaptive filter, an adaptive filter is a system with a linear filter that has a transfer function controlled by variable parameters and a means to adjust those parameters according to an optimization algorithm. The adaptive filter automatically adjusts its own parameters through an algorithm and responds to an error signal, which depends on the filter's output. In noise cancellation using adaptive filter its objective to produce an error signal that is fit in the least square sense to the signal s(n). This is achieved by feeding back the system output to the adaptive filter using an LMS algorithm to minimize the error signal until it reaches the value (e(n) = s(n)).
Assume that the s(n) is the signal which is to be transmitted. No(n), $N_1(n)$, and y(n) are statistically stationary signal. Let assume that s(n) is uncorrelated with No(n) and $N_1(n)$, and $N_1(n)$ is correlated with No(n).
The output e(n) is given by

$$e(n) = s(n) + No(n) - y(n) \quad \ldots\ldots\ldots\ldots..(1)$$

Taking square both sides,

$$e^2(n) = \{s(n) + No(n) - y(n)\}^2 \; e^2(n) = s^2(n) + \{No(n) - y(n)\}^2 + 2\, s(n)\{No(n) - y(n)\} \quad \ldots\ldots\ldots\ldots\ldots(2)$$

Taking the expected value both sides, we get

$$E[e^2(n)] = E[s^2(n)] + E[\{No(n) - y(n)\}^2] + 2E[\, s(n)\{No(n) - y(n)\}] \quad \ldots\ldots\ldots\ldots\ldots(3)$$

Realizing that s(n) is uncorrelated with No(n) and $N_1(n)$ and with y(n), then

$$E[e^2(n)] = E[s^2(n)] + E[\{No(n) - y(n)\}^2] \quad \ldots\ldots\ldots\ldots\ldots (4)$$

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The signal power $E[s^2(n)]$ will be unaffected as the filter is adjusted to minimize $E[e^2(n)]$. The minimum output power will be given by

$$\min E[e^2(n)] = E[s^2(n)] + \min E[\{No(n) - y(n)\}^2] \quad \ldots\ldots\ldots\ldots\ldots (5)$$

When the filter is adjusted so that $E[e^2(n)]$ is minimized, $E[\{No(n) - y(n)\}^2]$ is also minimized. When $E[\{No(n) - y(n)\}^2]$ is minimized then $E[\{e(n) - s(n)\}^2]$ is also minimized, and hence

$$[e(n) - s(n)] \approx [No(n) - y(n)] \quad \ldots\ldots\ldots\ldots\ldots (6)$$

Here, The output e(n) will contain the signal s(n) and noise. From (1), the output noise is given by [No(n) – y(n)]. Since minimizing $E[e^2(n)]$ minimizes $E[\{No(n) - y(n)\}^2]$, and since the signal in the output remains constant, minimizing the total output power maximizes the output signal-to-noise ratio. From (3), it seems that the smallest possible output power is $E[e^2(n)] = E[s^2(n)]$ and that will lead to $E[\{No(n) - y(n)\}^2] \approx 0$. Therefore

y(n) = No(n), and e(n) = s(n)

Hence minimizing output power causes the output signal to be noise free.

V. METHODOLGY

The idea behind a closed loop adaptive filter is that a variable filter is adjusted until the error (the difference between the filter output and the desired signal) is minimized. The Least Mean Squares (LMS) filter, Normalized Least Mean Square (NLMS) filter and the Recursive Least Squares (RLS) filter are types of adaptive filter. All adaptive algorithms LMS has probably become the most popular for its robustness, good tracking capabilities.

V.I: LMS algorithm:

An adaptive filter is a computational device that iteratively models the relationship between the input and output signals of a filter. An adaptive filter self-adjusts the filter coefficients according to an adaptive algorithm. Figure 1 shows the diagram of a typical adaptive filter.

The LMS is one of the simplest algorithms used in the adaptive structures due to the fact that it uses the error signal to calculate the filter coefficients. The output y(n) of FIR filter structure can be obtain from Eq.

$$y(n) = \sum_{m=0}^{N-1} w(m)x(n-m) \ldots\ldots\ldots\ldots\ldots (1)$$

Where n is no. of iteration

The error signal is calculated by Eq. (2)

$$e(n) = d(n) - y(n) \ldots\ldots\ldots\ldots\ldots (2)$$

The filter weights are updated from the error signal e(n) and input signal x(n) as in Eq. (3).

$$w(n+1) = w(n) + \mu\, e(n)x(n) \ldots\ldots\ldots\ldots\ldots (3)$$

Where: w(n) is the current weight value vector, w(n+1)is the next weight value vector, x(n) is the input signal vector, e(n) is the filter error vector and μ is the convergence factor which determine the filter convergence speed and overall behavior.

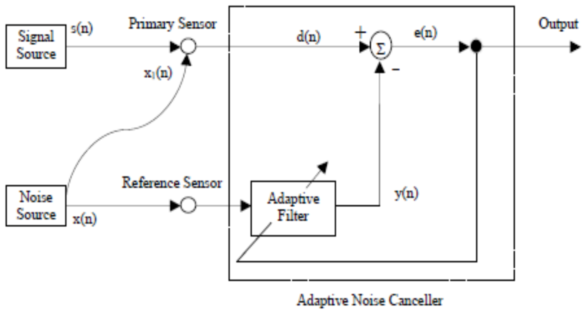

Fig 2: Adaptive Noise Canceller

s(n) – Source signal
d(n) – Primary signal
$X_1(n)$ – Noise signal
X(n – Noise reference input
y(n) – Output of Adaptive Filter
e(n) – System Output Signal

V.II: NLMS Algorithm:

In the standard LMS algorithm, when the convergence factor μ is large, the algorithm experiences a gradient noise amplification problem. In order to solve this difficulty, we can use the NLMS (Normalized Least Mean Square) algorithm. The correction applied to the weight vector w(n) at iteration n+1 is "normalized" with respect to the squared Euclidian norm of the input vector x(n) at iteration n. We may view the NLMS algorithm as a time-varying step-size algorithm, calculating the convergence factor μ as in Eq. (4)[7].

$$\mu(n) = \frac{\alpha}{c + (\|x(n)\|)^2} \ldots\ldots\ldots\ldots\ldots (4)$$

Where: α is the NLMS adaption constant, which optimize the convergence rate of the algorithm and should satisfy the condition 0< α<2, and c is the constant term for normalization and is always less than 1. The Filter weights are updated by the Eq. (5).

$$w(n+1) = w(n) + \frac{\alpha}{c + (\|x(n)\|)^2}\, e(n)x(n) \ldots\ldots\ldots\ldots\ldots (5)$$

V.III: RLS Algorithm:

The RLS algorithms are known for their excellent performance when working in time varying environments but at the cost of an increased computational complexity and some stability problems. In this algorithm the filter tap weight vector is updated using Eq. (6) [8].

$$w(n) = \bar{w}^T(n-1) + k(n)\bar{e}_{n-1}(n) \ldots\ldots\ldots\ldots (6)$$

Eq. (7) and (8) are intermediate gain vector used to compute tap weights.

$$k(n) = u(n) / [\lambda + x^T(n)u(n)] \quad \ldots\ldots\ldots\ldots(7)$$

$$u(n) = \overline{w_\lambda^{-1}}(n-1)x(n) \quad \ldots\ldots\ldots\ldots(8)$$

Where: λ is a small positive constant very close to, but smaller than 1.
The filter output is calculated using the filter tap weights of previous iteration and the current input vector as in Eq. (9).

$$\bar{y}_{n-1}(n) = \bar{w}^T(n-1)x(n) \quad \ldots\ldots\ldots\ldots(9)$$

$$\bar{e}_{n-1}(n) = d(n) - \bar{y}_{n-1}(n) \quad \ldots\ldots\ldots(10)$$

In the RLS Algorithm the estimate of previous samples of output signal, error signal and filter weight is required that leads to higher memory requirements.

VI. CONCLUSION

In this work, different Adaptive algorithms were analysed and compared. The basic adaptive algorithms which widely used for performing weight updating of an adaptive filter are: the LMS (Least Mean Square), NLMS (Normalized Least Mean Square) and the RLS (Recursive Least Square) algorithm. Among all adaptive algorithms LMS has probably become the most popular for its robustness, good tracking capabilities and simplicity in stationary environment. RLS is best for non-stationary environment with high convergence speed but at the cost of higher complexity.

Advantages & disadvantages of LMS algorithm:
(1) Simplicity in implementation
(2) Stable and robust performance against different s
(3) slow convergence (due to Eigen value spread)

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BIOGRAPHIES

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