Denoising OF ECG Signals Using CSLMS Adaptive Filter Algorithm

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Abstract: Adaptive filter is a primary method to filter ECG signal, because it does not need the signal statistical characteristics. In this project we present a novel adaptive filter for removing the artifacts from ECG signals based on Constrained Stability Least Mean Square (CSLMS) algorithm. The adaptive filter essentially minimizes the mean-squared error between a primary input, which is the noisy ECG, 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 ECG in the primary input. To show that CSLMS algorithm is really effective in clinical situations, the method has been validated using several ECG recordings with a wide variety of wave morphologies from MIT-BIH arrhythmia database. The applications are Noise cancellation, linear prediction, Adaptive feedback cancellation and Echo cancellation.

Keywords: Adaptive filter, LMS algorithm, CSLMS algorithm

1. INTRODUCTION

Baseline wander and power line interference reduction is the first step in all electrocardiographic (ECG) signal processing. The baseline wander is caused by varying electrode-skin impedance, patient’s movements and breath. This kind of disturbance is especially present in exercise electrocardiography, as well as during ambulatory and Holter monitoring. The ECG signal is also degraded by additive 50 or 60 Hz power line (AC) interference. This kind of disturbance can be modelled by a sinusoid with respective frequency and random phase. These two artifacts are the dominant artifacts and strongly affect the ST segment, degrades the signal quality, frequency resolution, produce large amplitude signals in ECG that can resemble PQRST waveforms and masks tiny features that are important for clinical monitoring and diagnosis.

Cancellation of these artifacts in ECG signals is an important task for better diagnosis. Hence the extraction of high-resolution ECG signals from recordings which are contaminated with background noise is an important issue to investigate. The goal of ECG signal enhancement is to separate the valid signal components from the undesired artifacts, so as to present an ECG that facilitates easy and accurate interpretation. Many approaches have been reported in the literature to address ECG enhancement using both adaptive and non-adaptive techniques, adaptive filtering techniques permit to the detect time varying potentials and to track the dynamic variations of the signals. Apart from these several adaptive signal processing techniques are also published, e.g., NLMS algorithm with decreasing step size, which converge to the global minimum, a variable step size NLMS algorithm with faster convergence rate. Recently, Rahman presented several less computational complex adaptive algorithms in time domain, but these algorithms exhibits slower convergence rate.

2. ADAPTIVE FILTER

An adaptive filter is a filter that self-adjusts its transfer function according to an optimization algorithm driven by an error signal. Because of the complexity of the optimization algorithms, most adaptive filters are digital filters. By way of contrast, a non-adaptive filter has a static transfer function. Adaptive filters are required for some applications because some parameters of the desired processing operation (for instance, the locations of reflective surfaces in a reverberant space) are not known in advance. The adaptive filter uses feedback in the form of an error signal to refine its transfer function to match the changing parameters.

Generally speaking, the adaptive process involves the use of an cost function, which is a criterion for optimum performance of the filter, to feed an algorithm, which determines how to modify filter transfer function to minimize the cost on the next iteration.

As the power of digital signal processing has increased, adaptive filters have become much more common and are now routinely used in devices such as mobile phones and other communication devices, camcorders and digital cameras, and medical monitoring equipment.
Applications of adaptive filters

- Noise cancellation
- Linear prediction
- Adaptive feedback cancellation
- Echo cancellation

3. EXISTING METHOD

3.1 Least Mean Square Filter

Least mean squares (LMS) algorithms are a class of adaptive filter used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean squares of the error signal (difference between the desired and the actual signal). It is a stochastic gradient resource method in that the filter is only adapted based on the error at the current time. It was invented in 1960 by Stanford university professor Bernard windrow and his first Ph.D. student, Ted Hoff.

In order to remove the noise from the ECG signal, the ECG signal SI (n) corrupted with noise signal PI (n) is applied as the desired response d (n) to the adaptive filter shown in Fig.3.1. If the noise signal P2 (n), possibly recorded from another generator of noise that is correlated in some way with PI (n) is applied at the input of the filter, i.e., x(n) = P2(n) the filter error becomes e(n) = [sI(n) + pI(n)] - y(n). Where, y(n) is the filter output and it is given by,

\[ y(n) = w^T(n)x(n) \]  \hspace{1cm} (4.1.2)

Since the signal and noise are uncorrelated, the mean squared error (MSE) becomes

\[ E[e^2(n)] = E[(sI(n) - y(n))^2] + E[pI^2(n)] \]

Minimizing the MSE results in a filter output which is the best least-squares estimate of the signal sI(n). Normalized LMS (NLMS) algorithm is another class of adaptive algorithm used to train the coefficients of the adaptive filter. This algorithm takes into account variation in the signal level at the filter output and selecting the normalized step size parameter that results in a stable as well as fast converging algorithm. The weight update relation for NLMS algorithm is as follows

\[ w(n + 1) = w(n) + \frac{\mu}{p + x^T(n)x(n)} x(n)e(n) \]  \hspace{1cm} (4.1.3)

The variable step can be written as,

\[ \mu(n) = \frac{\mu}{p + x^T(n)x(n)} \]

Equation 4.1.4

Here f.L is fixed convergence factor to control maladjustment A common major drawback of adaptive noise canceler based on LMS and NLMS algorithms is the large value of excess mean-square error which results in signal distortion in the noise-cancelled signal. In the CSLMS algorithm the time-varying step-size that is inversely proportional to the squared norm of the difference between two consecutive input vectors rather than the input data vector as in the NLMS. This algorithm provides Significant improvements in decreasing mean-squared error (EMSE) and consequently minimizing signal distortion [22]. The weight update relation for CSLMS algorithm is as follows,

\[ w(n + 1) = w(n) + \frac{\delta x(n)}{||x(n)||^2} \]  \hspace{1cm} (4.1.5)

Where \( \delta x(n) = x(n) - x(n - 1) \) is the difference between two consecutive input vectors. Also \( \delta e(n) = e(n) - e(n - 1) \) is the difference in the priori error sequence. The weight adaptation rule can be made more robust by introducing a small P and by
multiplying the weight increment by a constant step size \( f.L \) to control the speed of the adaptation. This gives the weight update relation for CSLMS algorithm in its final form as follows,

\[
\mathbf{w}(n + 1) = \mathbf{w}(n) + \left[ \frac{\mathbf{\delta x}(n) \mathbf{\delta e}(n)}{\|\mathbf{\delta x}(n)\|^2} \right]
\]

Equation 4.1.6

The parameter \( P \) is set to avoid denominator being too small, step size parameter too big and to prevent numerical instabilities in case of a vanishingly small squared norm. The convergence characteristics of both the algorithms are shown in Fig. 2.

### 5. CONCLUSION

In this paper the process of noise removal from ECG signal using CSLMS based adaptive filtering is presented. For this, the input and the desired response signals are properly chosen in such a way that the filter output is the best least squared estimate of the original ECG signal. The proposed treatment exploits the modifications in the weight update formula and thus pushes up the speed over the respective LMS based realizations. Our simulations, however, confirm that the performance of the CSLMS is better than the LMS algorithm in terms of SNRI, MSE and maladjustment, this is shown in tables I and II. Hence CSLMS based adaptive noise canceller may be used in all practical applications.

### REFERENCES


