Hardware Implementation of NLMS Algorithm for Adaptive Noise Cancellation

Raj kumar Thenua
Dept. of Electronics & Instrumentation, Anand Engineering College, Agra-UP, India
kumarraj04in@gmail.com

S.K. Agarwal
Dept. of Electronics & Communication, Sobhasariya Engineering College, Sikar-Rajasthan, India
skagarwal5@rediffmail.com

Abstract—In numerous applications of signal processing, communications and biomedical we are faced with the necessity to remove noise and distortion from the signals. Adaptive filter is one of the most important areas in digital signal processing to remove background noise and distortion. In this work an attempt is to be made to de-noise an ECG signal, with the help of adaptive NLMS algorithm, implemented on TMS320C6713 DSP processor in real-time environment. A SIMULINK model is created and linked to TI TMS320C6713 digital signal processor through embedded target for TI C6000 SIMULINK toolbox and REAL-TIME workshop to perform hardware adaptive noise cancellation. A clean (amplified) ECG signal with 1000 samples, of amplitude 260mV and frequency of 35 Hz, generated through two lead configurations, sampled at a frequency of 1.5 kHz is taken as a clean signal. The system is tested for three level of noise and shows a great level of improvement in Signal to Noise Ratio (SNR).

Index Terms—Adaptive filters, LMS, Mean Squared Error (MSE), NLMS, RTDX, TMS320C6713 DSK.

I. INTRODUCTION

An adaptive filter has the property of self-modifying its frequency response to change the behavior in time, allowing the filter to adapt the response to the input signal characteristics change. Due to this capability, the overall performance and the construction flexibility, the adaptive filters have been employed in many different applications, some of the most important are: telephonic echo cancellation, radar signal processing, navigation systems, communications channel equalization and biometrics signals processing [1].

The most common adaptive filters, which are used during the adaption process, are the finite impulse response (FIR) types. These are preferable because they are stable, and no special adjustments are needed for their implementation.

Fig.1 illustrates the general configuration for an Adaptive filter [2]. The adaptive filter has two inputs: the primary input d(n), which represents the desired signal corrupted with undesired noise, and the reference signal x(n), which is the undesired 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 task, a reference of the noise signal is needed. That reference is fed to the system, and it is called a 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 delay. Therefore the reference signal cannot be simply subtracted 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 for the amplitude, phase and time delay, and then subtracted from the primary signal. This filtered noise is the system’s prediction of the noise portion of the primary signal, y(n). The resulting signal is called error signal e(n), and it presents the output of the system. Ideally, the resulting error signal would be only the desired portion of the primary signal.

DSP has huge number of applications in today’s life [3], such as audio signal processing i.e. noise cancellation, system identification, equalization and etc. DSP is widely used in high speed modems and mobile phones due to availability of low cost DSP chips that can perform extensive computation in
real-time. Besides audio signal processing, digital signal processing is also used in other kind of signal processing applications such as image processing, statistical signal processing, biomedical signal processing etc.

In this paper we investigate the performance of an adaptive NLMS algorithm implemented on TI TMS320C6713 hardware [4], when applied to an ECG signal. The obtained results from DSP kit output shows a great improvement in SNR level of a noisy ECG signal. The paper is organized in five sections; section 2 gives an idea of LMS based algorithms, in section 3 an ANC model is designed which results are discussed in section 4 and finally section 5 concludes the work.

II. ADAPTIVE ALGORITHMS

The algorithms used to perform the adaptation, and the configuration of the filter depends directly on the use of the filter. However, the basic computational engine that performs the adaptation of the filter coefficients can be the same for different algorithms, and it is based on the statistics of the input signals to the system. The two classes of adaptive filtering algorithms namely Least Mean Squared (LMS) and Recursive Least Squares (RLS) are capable of performing the adaptation of the filter coefficients. The LMS based algorithms are simple to understand and easy to implement whereas RLS based algorithm are complex and requires so much memory for implementation. So in this work we have focuses on LMS based algorithms.

A. Least Mean Square Algorithm

The LMS algorithm [2], is a type of adaptive filter known as stochastic gradient-based algorithms as it utilizes the gradient vector of the filter tap weights to converge on the optimal wiener solution. With each iteration of the LMS algorithm, the filter tap weights of the adaptive filter are updated according to the following formula:

\[ w(n+1) = w(n) + 2\mu e(n)x(n) \]  

(1)

Here \( x(n) \) is the input vector of time delayed input values, \( x(n) = [x(n)x(n-1)x(n-2)....x(n-N+1)]^T \)  

(2)

The vector \( w(n) = [w_0(n)w_1(n)w_2(n)....w_{N-1}(n)]^T \) represents the coefficients of the adaptive FIR filter tap weight vector at time \( n \).

The parameter \( \mu \) is known as the step size parameter and is a small positive constant. This step size parameter controls the influence of the updating factor. Selection of a suitable value for \( \mu \) is imperative to the performance of the LMS algorithm, if the value is too small the time the adaptive filter takes to converge on the optimal solution will be too long; if \( \mu \) is too large the adaptive filter becomes unstable and its output diverges.

B. Normalized LMS Algorithm

In the standard LMS algorithm, when the convergence factor \( \mu \) 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 \( \mu \) as in Eq. (3)[1].

\[ \mu(n) = \frac{\alpha}{c + \|x(n)\|^2} \]  

(3)

Where: \( \alpha \) is the NLMS adaption constant, which optimize the convergence rate of the algorithm and should satisfy the condition \( 0<\alpha<2 \), and \( c \) is the constant term for normalization and is always less than 1.

The Filter weights are updated by the Eq. (4).

\[ w(n+1) = w(n) + \frac{\alpha}{c + \|x(n)\|^2} e(n)x(n) \]  

(4)

III. ADAPTIVE NOISE CANCELLATION

Adaptive noise cancellation (ANC) is performed by subtracting noise from a received signal, and an operation controlled by an adaptive process, it is possible to achieve a superior system performance compared to direct filtering of the received signal. Fig.2 shows adaptive noise canceling system.

The system composed of two separate inputs, a primary input or ECG signal source which is shown as \( s(n) \) and a reference input that is the noise input shown as \( x(n) \). The primary signal is corrupted by noise \( x_1(n) \). \( x_1(n) \) is highly correlated with noise signal or reference signal \( x(n) \). Desired signal \( d(n) \) results from addition of primary signal \( s(n) \) and correlated noise signal \( x_1(n) \). The reference signal \( x(n) \) is fed into adaptive filter and its output \( y(n) \) is subtracted from desired signal \( d(n) \). Output of the summer block is then fed back to adaptive filter to update filter coefficients. This process is run recursively to obtain the noise free signal which is supposed to be the same or very similar to primary signal \( s(n) \).
IV. EXPERIMENTAL RESULTS

A. Simulink Model

The Simulink model is designed as shown in Fig.3, to generate C code to be downloaded onto DSP chip. The model is designed with help of inbuilt library of Simulink, only the blocks are reconfigured as per the requirements of DSP kit [5-7].

![Fig.3. Simulink model for ANC system](image)

B. Converting the Simulink model to C code

A program needs to be written in Code Composer Studio (CCS) software in C code format to be able to be loaded onto the DSP chip. Therefore, to implement the above SIMULINK model on hardware or DSP board one needs to convert the model to C code first to be compatible with CCS software and then load it on DSP chip [5]. Embedded target for C6000 SIMULINK toolbox provides the link between model and CCS in this project. This toolbox with the aid of Real-time workshop toolbox has the ability of converting and transforming the SIMULINK model into a complete CCS project which then can be implemented on hardware. As it is shown in Fig.4, clicking on “Build model”, the model will start linking to CCS and then creating CCS project.

![Fig.4. C code generation using Real-time workshop (RTW)](image)

C. Hardware Experiment Results

Three types of tests performed for ECG signal to examine the capability of the designed system. First test is done for the power line interference of 50 Hz signal. Other two tests considered for broadband white Gaussian noise but with different noise powers to check the system performance in the presence of different noise powers. In practical demonstration of system DIP switch 1 of the DSP board was used to change the unwanted noise between two different states and thus to observe the system capability to adapt to new noise statistics and as a result cancel it out. A MATLAB program was written to calculate the noise power and signal to noise ratio of signal and noise before and after filtering to help the reader observe the ability of the system to improve the SNR of filtered signal.

Fig.5 shows a clean ECG signal, which is generally corrupted by 50 Hz power line interference and other kind of noise [8].

![Fig.5. Clean ECG signal s(n)](image)

![Fig.6 (a) Filtered signal for 60Hz power line interference.](image)

![Fig.6 (b). Filtered output at medium level noise](image)
Fig.6 (c). Filtered output at high level noise

Fig.6 shows the filtered output results for different conditions. In all figures of Fig.6, the first signal is the noisy signal and the second is the adaptive filtered output signal.

Table 1
SNR Improvement versus Noise Variance

<table>
<thead>
<tr>
<th>S.N.</th>
<th>Noise Variance</th>
<th>Sampling Rate (kHz)</th>
<th>SNR Improvement (dB)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.</td>
<td>0.02</td>
<td>1.5</td>
<td>7.46</td>
</tr>
<tr>
<td>2.</td>
<td>0.05</td>
<td>1.5</td>
<td>6.02</td>
</tr>
<tr>
<td>3.</td>
<td>0.1</td>
<td>1.5</td>
<td>5.22</td>
</tr>
</tbody>
</table>

The results that obtained for the designed system shows a perfect performance of the system and impressive improvement of signal to noise ratio was achieved. SNR improvement for different noise level is shown in Table 1.

V. CONCLUSION

The main objective of this paper was to implement an adaptive filter on hardware and test the performance for biomedical ECG signal at different noise levels. This paper provides background information on FIR adaptive filter structure and adaptive algorithms, LMS and NLMS algorithms. Based on these discussions a transversal FIR adaptive filter with NLMS updating algorithm designed for this paper. The designed adaptive system then implemented on Texas Instruments TMS320C6713 DSP board for hardware implementation. Three noises with different powers were used to test and judge about the system performance in software and hardware. The background noises for ECG signal were eliminated adequately with reasonable rate for all the tested noises. For low and medium power noises the system showed SNR improvement up to 7.46dB and in high power noise test 5.22dB of SNR improvement achieved. Based on these results, the designed system proved to be successful in performing the selected application that is to eliminate unwanted noise form a desired signal such as ECG signal or any other type of biomedical signal.

REFERENCES