Hardware Implementation of Adaptive Algorithms for Noise Cancellation

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***Abstract*—In this work an attempt has been made to de-noise a sinusoidal tone signal and an ECG signal, with the help of LMS based adaptive algorithms, implemented on TMS320C6713 DSP processor in real-time environment. A SIMULINK model is developed then linked to code composer studio through embedded target and Real Time workshop facility to generate corresponding C code. The generated C code is used for the DSP processor to perform adaptive noise cancellation. The designed system is tested at three level of noise and shows a considerable level of improvement in Signal to Noise Ratio (SNR).**

***Index Terms*—Adaptive noise cancellation (ANC), digital signal processor (DSP), mean squared error (MSE), normalized least mean square (NLMS), real time workshop (RTW).**

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 [1]-[6] over the years. Bernard Widrow *et. al.* [1] developed a model for noise cancelation with the help of adaptive filter and employed for variety of practical applications like the cancelling of various forms of periodic interference in electrocardiography, the cancelling of periodic interference in speech signals, and the cancelling of broad-band interference in the side-lobes of an antenna array.

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 [4]. 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.

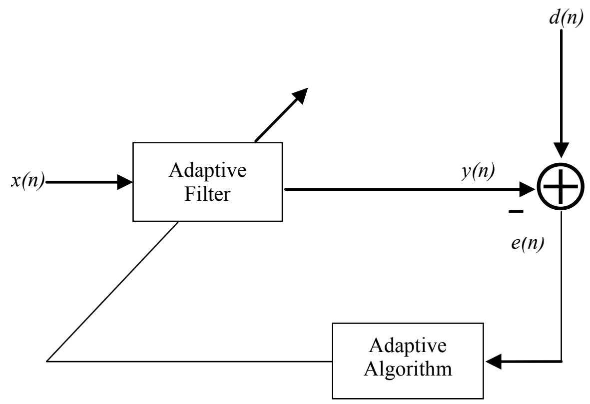

Fig. 1. General Adaptive filter configuration.

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 biomedical 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.

This paper investigates the performance of LMS and NLMS adaptive algorithms when implemented on Texas Instruments (TI) TMS320C6713 DSP hardware [8]-[10] and tested for two types of signals; sinusoidal tone signal and ECG signal. The obtained results from DSP kit are analyzed with the help of Digital Storage Oscilloscope (DSO) and shows considerable improvement in SNR level of a filtered signal.

II. Adaptive Algorithms

The two classes of adaptive filtering algorithms namely Least Mean Squared (LMS) and Recursive Least Squares

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(RLS) are capable of performing the adaptation of the filter coefficients. However, the LMS based algorithms are simple to understand and easy to implement whereas RLS based algorithm are complex and requires much memory for implementation. Therefore this work is focused on LMS based algorithms.

A. Least Mean Square Algorithm

The LMS algorithm [1-2], is a 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:

$$w(n+1) = w(n) + 2\mu e(n)x(n) \tag{1}$$

where, $x(n)$ is the input vector of time delayed input values, $w(n)$ represents the coefficients of the adaptive FIR filter tap weight vector at time n and μ is known as the step size.

Selection of a suitable value for μ is imperative to the performance of the LMS algorithm, if the value is too small, the time adaptive filter takes to converge on the optimal solution will be too long; if μ 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 μ is large, the algorithm experiences a gradient noise amplification problem. This difficulty is solved by 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.

The NLMS algorithm can be viewed as a time-varying step-size algorithm, calculating the convergence factor μ as in Eq. (2)[4].

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

where α 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, which is always less than 1.

The filter weights using NLMS algorithm are updated by the Eq. (3).

$$w(n+1) = w(n) + \frac{\alpha}{c + \|x(n)\|^2} e(n)x(n) \tag{3}$$

III. ANC Model Design

Adaptive noise cancellation as shown in Fig.2 is performed by subtracting predicted noise from a received signal, and continues the process by updating filter weights adaptively in a controlled manner to get an improved signal-to-noise ratio.

The ANC system composed of two separate inputs, a primary input i.e. source signal s(n) and a reference input i.e. noise input $x(n)$. The primary signal is corrupted by a noise $x_1(n)$ which is highly correlated with noise signal $x(n)$. The 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). The output of the summer block is then fed back to adaptive filter to update filter coefficients. The above process is run recursively to obtain the noise free signal which is supposed to be the same or very similar to primary signal $s(n)$.

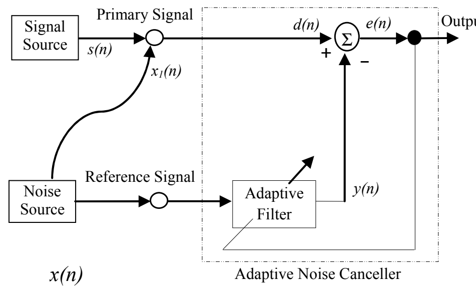

Fig. 2. Adaptive Noise Cancellation system.

A. ANC Simulink Model

The ANC Simulink model as shown in Fig.3 is designed using LMS and NLMS algorithms for generating C code and to download this code on DSP target processor. The ANC model is designed with help of inbuilt library of Simulink and the blocks are reconfigured as per the requirements of TMS320C6713 DSP processor [7], [10].

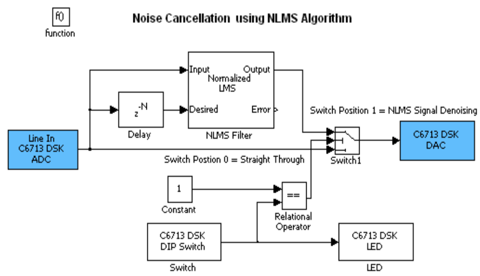

Fig. 3. Simulink model for ANC system.

B. C code Generation with Simulink Model

The DSP development software, Code Composer Studio can accept either C or assembly code to generate output (.out) file, which can be load on DSP chip. The programming in C or assembly is not easy for every researcher. Thus the designed Simulink model facilitates to generate C code automatically corresponding to the desired problem with the help of embedded target and RTW facility provided in Simulink as shown in Fig. 4 The real-time experimental setup using DSP processor is illustrated in Fig. 4 (b).

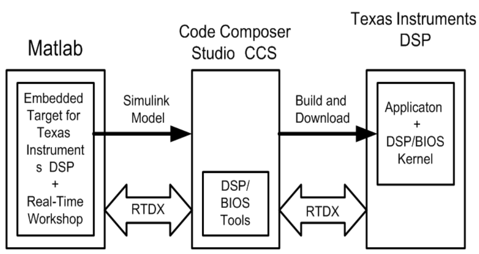

Fig. 4(a). Interfacing between MATLAB, CCS and DSP.

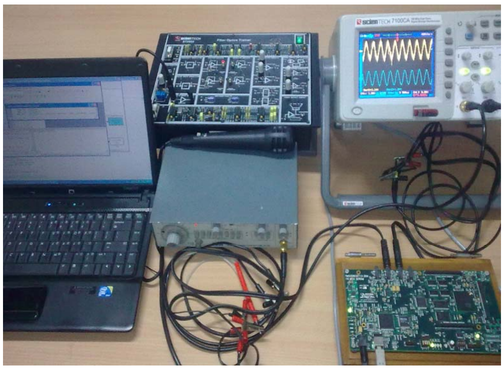
Fig. 4 (b). Real-time experimental setup using DSP processor.

IV. RESULTS

The designed model is tested with two types of signals. First of all the system is tested for a standard sinusoidal tone signal with an average power of 2dB and 1 kHz of frequency. Later the system is tested with a real-time biomedical ECG signal having 1000 samples with the amplitude of 260mV and frequency of 35 Hz, generated through twelve lead configurations.

Three types of test performed for each type of signal to examine the capability of the designed system. First test is done for the low power noise corrupted signals. And other two tests considered for broadband white Gaussian noise corrupted signals with different high noise powers. The filtered output is measured with the help of DSO and the output is saved in *.csv* file format, that file is utilized later in MATLAB for further processing. A MATLAB program is written to calculate SNR of noisy signal before and after filtering, that help the reader to observe the filtering capability of the system in terms of SNR Improvement (SNR after filtering –SNR before filtering).

When the designed system is tested with sinusoidal tone signal as shown in Fig.5, it shows SNR Improvement of 13dB at low noise, 11dB at medium noise and 10dB for high noise environment. The ANC model performance is further tested by making variations in the amplitude and frequency of the input tone signal, which affect the SNR improvement of the system. When frequency of signal is varied (1kHz-5kHz); the SNR improvement varies (11dB-13dB) as per the correlation of noise signal frequency, with the frequency of clean signal as illustrated in Table I.

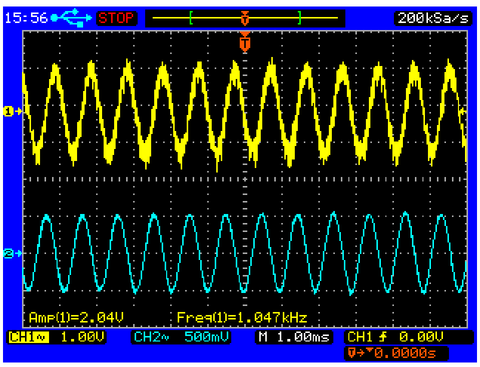

Fig. 5(a). Filtered tone signal at low noise.

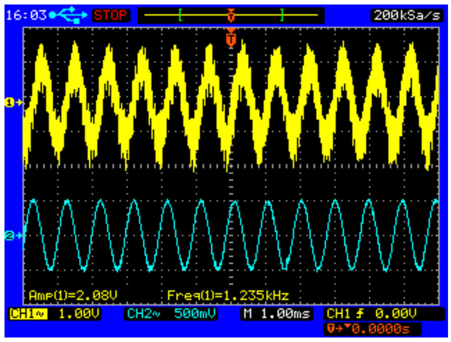

Fig. 5 (b). Filtered tone signal at medium noise.

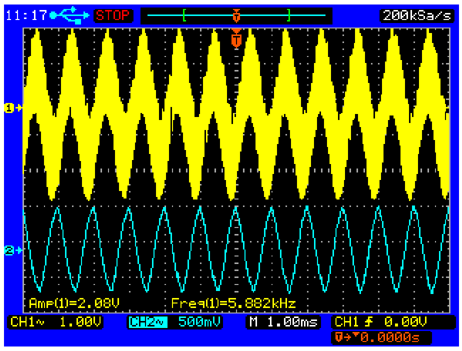

Fig. 5 (c). Filtered tone signal at high noise

Now the system is tested for a real-time biomedical ECG signal as shown in Fig.6. The ECG signal is composed of various frequencies hence de-noising of such signal is a challenging task. When ECG signal get corrupted by noise, it loses the characteristics of the signal which are generally used for diagnosis purpose. In real-time ECG signal, noise corrupts the QRS complex of ECG signal as shown in Fig.7 (b) & Fig.7(c), which create complications in the measurement of heart rate of a patient.

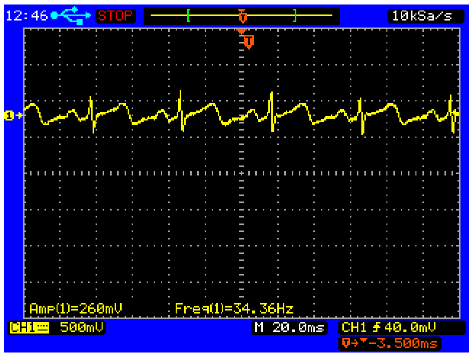

Fig. 6. Clean ECG signal *s(n)*.

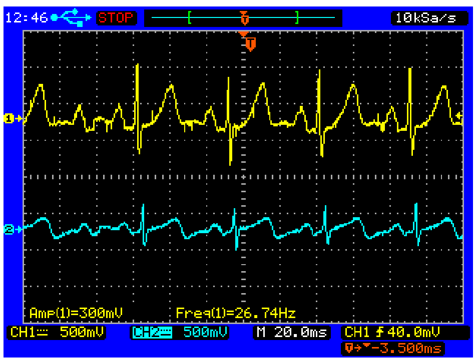

Fig. 7(a). Filtered output at low level noise

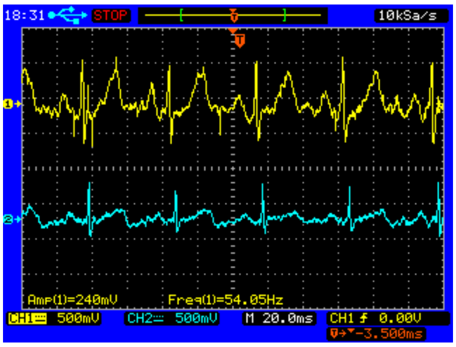

Fig. 7(b). Filtered output at medium level noise.

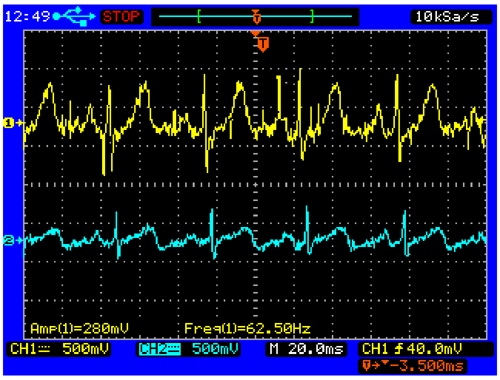

Fig. 7(c). Filtered output at high level noise.

The proposed system is capable of removing the peaks of noise from the noisy ECG signal and preserves the R waves of ECG signal as shown in Fig.7 (b) & Fig.7(c) which is useful for diagnosis purposes. To de-noise ECG signal both LMS and NLMS based ANC models are designed and results for both algorithms are analyzed. The SNR Improvements for LMS algorithm are 8.85dB at low noise, 7.55dB at medium noise and 5.12dB at high noise. Similarly for NLMS algorithm SNR Improvements are 9.89dB at low noise, 8.62dB at medium noise and 6.38dB at high noise which are illustrated in Table II. It is clear from the Table II that the NLMS algorithm removes more amount of background noise from an ECG signal when compared with LMS algorithm and gives an average SNR improvement difference of 1dB at low noise and 1.3 dB at high noise environment.

TABLEI: ISNR IMPROVEMENT VERSUS VOLTAGE AND FREQUENCY VARIATIONS FOR TONE SIGNAL

S.N.	Amplitude (V)	Frequency (kHz)	SNR Improvement (dB) NLMS
1.	2	1	11.00
2.	3	1	11.52
3.	4	1	11.93
4.	5	1	12.80
5.	2	2	11.58
6.	2	3	11.93
7.	2	4	12.08
8.	2	5	11.66

TABLE II: SNR IMPROVEMENT VERSUS NOISE VARIANCE FOR ECG SIGNAL

S.N.	Noise Variance	Sampling Rate (kHz)	SNR Improvement (dB) NLMS	SNR Improvement (dB) LMS
1.	0.02	1.5	9.89	8.85
2.	0.05	1.5	8.62	7.55
3.	0.1	1.5	6.38	5.12

The results that obtained for the designed system shows a good performance for the system and considerable improvement of signal to noise ratio was achieved.

V. CONCLUSION

The implementation of adaptive noise cancellation system on DSP processor is done successfully and the real-time processor results are analyzed for two types of signals; tone signal and ECG signal, with the help of DSO. And the filter performance is measured in terms of SNR improvement. The tone signal is analyzed at various frequencies and voltage level to check the effect of noise on the filtering and observed that the effect of noise gets more prominent when the frequency of noise and clean signal is highly correlated or noise signal has higher amplitude. The designed system is further tested for three ECG signals of different noise level for NLMS algorithm and LMS algorithm, a fair amount of SNR improvement is achieved in both cases and the filtered ECG signal found useful for medical diagnosis purposes .When the results of two algorithms were compared it is proved that the NLMS algorithm has the best performance.

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