

Hardware Implementation of Adaptive Noise Cancellation over DSP Kit TMS320C6713

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Abstract

In noisy acoustic environment, audio signal in speech communication from mobile phone, moving car, train, aero plane, or over a noisy telephone channel is corrupted by additive random noise. The noise is unwanted signal and it is desirable to remove noise from original signal. Since noise is random process and varying at every instant of time, we need to estimate noise at every instant to remove it from original signal. There are many schemes for noise removal but most effective scheme to accomplish noise cancellation is to use adaptive filters. In this paper, we have carried out simulations for different adaptive algorithms (LMS, NLMS and RLS) and compared their performance for noise cancellation in noisy environment. Real time implementation of adaptive algorithm over DSP kit (TMS320C6713) is also presented in this paper. Performance of adaptive algorithm over hardware is also presented. Developed system incorporating best performance adaptive filter in any noisy environment can be used for noise cancellation

Keywords: Noise Cancellation, Adaptive filter, LMS, NLMS, RLS, DSP kit TMS320C6713.

1. INTRODUCTION

We experience effects of audio noise in our daily lives when using mobile phone in moving car, train, aero plane, or talking over a noisy telephone channel. This noise contaminates the original information bearing signal with noise from its surrounding environment. We want to remove this noise to have effective communication between two ends [1]. Noise is random process so we need to estimate the noise at every instant so that we can remove noise to get our desired signal from noisy signal. Noise cancellation in variable, noisy and non-stationary environments is usually accomplished by means of adaptive filters. In adaptive filters [2], [3], [4], two inputs are required:

- Noise
- Primary signal + Noise

There are many adaptive filters and its applications are available in literature [4], [5], [6] but most commonly used adaptive filter is Widrow's and Hoff's Least Mean Square (LMS). LMS is used because of its robustness, good tracking capabilities and simplicity both in terms of computational load and easiness of implementation. It is implemented using Finite Impulse Response (FIR) filter

and first order weight updation equation. It has therefore been successfully applied to wide variety of applications [4]. Normalized Least Mean Square (NLMS) is a variation of LMS algorithm that employs normalization of input power to solve problems. Other adaptive algorithms include Recursive Least Square (RLS) whose convergence is fast as compared to LMS. However, computational complexity of RLS is very large as compared to LMS because it involves computation of matrix inverse.

In this paper, we have presented adaptive noise cancellation using LMS, NLMS and RLS. Problem statement is given in section 2. Design and working of adaptive noise canceller is given in section 3. Simulation results are given in section 4. DSP kit introduction is given in section 5. Hardware implementation is presented in section 6 and finally conclusion is given in section 7.

2. PROBLEM STATEMENT

We want to implement a real time Adaptive Noise Canceller (ANC) over DSP kit. In order to build our system and achieve ANC we will consider following example. In airplane pilot's voice is corrupted with the noise from the airplane's engines. Let us consider that desired signal is the pilot's voice which is represented by $x(n)$. A microphone is placed near engine that will record the engine noise from it. Let us call that noise as $v(n)$. Another microphone near pilot will record both pilot voice and noise from engine. Our objective is to develop a system that remove the engine noise from pilots voice and give us output which contain pilot's voice only.

3. DESIGN AND WORKING OF ADAPTIVE NOISE CANCELLER

Adaptive filter works on principle of minimizing mean square error between the desired output and filter output. Weights of the adaptive filter adapt in such a way that error is minimized and desired output is obtained. Error minimization criteria is different for different adaptive filters and hence their performance in different situations. An adaptive filter can be a combination of

- Single-input or multi-input filters
- Linear or nonlinear filters
- FIR or Infinite Impulse Response (IIR) filters

In this paper, we will implement both FIR and IIR filters for ANC. Linear FIR filters are most commonly used because of their stability and relative ease of adaptation.

It is not possible to subtract the engine noise from the pilot's microphone directly, since the engine noise received in the pilot's microphone, and the engine noise received in the reference microphone are not the same signal. These two signals are uncorrelated with each other. There are differences in amplitude and time delay. Also, these differences are not fixed. They change in time with pilot's microphone position with respect to the airplane engine, and many other factors. Above all, noise is a random process and is varying at every instant of time. We need to estimate noise at every instant to remove it from pilot's audio signal. Therefore designing the fixed filter to perform the task would not obtain the desired results. The application requires adaptive solution.

Adaptive filter process the engine noise and make it equal to noise contaminating the speech signal. Then noise is subtracted from noisy signal to get noise free speech signal. In cases such as receiver of a telecommunication system, where there is no access to instantaneous value of contaminating noise and only noisy signal is available, complete noise cancellation isn't possible. However noise reduction can be achieved in an average sense, using the statistics of the signal and the noise process in such cases [1]. We will use the noise cancellation system shown in figure 1.

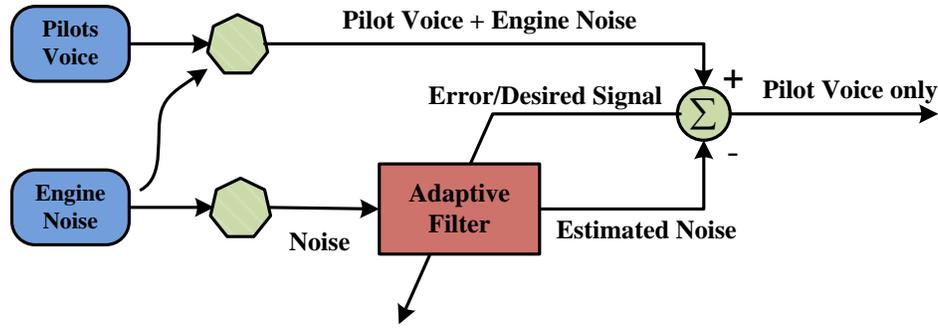


FIGURE 1: Noise Cancellation System.

We will use LMS, NLMS and RLS adaptive filtering for removing the effects of engine noise on pilot's voice. Figure 1 illustrates the basic principles of adaptive noise canceling. The input to the adaptive filter is engine noise signal $v(n)$ and is uncorrelated with the pilots voice signal $x(n)$. The reference signal $v(n)$ is filtered through adaptive filter to produce the output $\hat{v}(n)$ that is an estimate of the noise $v(n)$. This output is then subtracted from the noisy signal $(x(n)+v(n))$ to produce the system output $y(n)$. This is also called error signal. Error is feedback to adaptive algorithm block. Actually error signal is the "desired signal" for noise canceller which eventually becomes the output for noise canceller. Mathematically desired signal is given by (1).

$$Desired_Signal = [x(n) + v(n)] - y(n) \quad (1)$$

where $y(n)$ is output of adaptive filter i.e. estimate of noise at input of adaptive filter. The system output is used to control the adaptive filter and is an estimate of $x(n)$. Provided $x(n)$ is uncorrelated with $v(n)$, and the adaptive filter is adjusted to give a system output $y(n)$ that has the least possible energy, then $y(n)$ is a best least-squares fit to the clean signal $x(n)$. We have used LMS, NLMS and RLS adaptive algorithms for noise cancellation.

4. SIMULATION RESULTS

Simulations for noise cancellation using LMS, NLMS and RLS algorithm were carried out in Matlab/Simulink.

2.1 LMS

Among adaptive algorithms, LMS is most simple and computationally less expensive algorithm. Weight update equation for LMS is given by (2).

$$w(n + 1) = w(n) + \mu v(n)e(n) \quad (2)$$

where $w(n)$ is weight vector for LMS adaptive filter. μ is learning rate and $0 < \mu < 1$. $e(n)$ is error signal and given by (3)

$$e(n) = [x(n) + v(n)] - y(n) \quad (3)$$

where $y(n) = \hat{v}(n)$. For simulations, learning rate for LMS algorithm was set to 0.01 and filter order was chosen 10. Results of noise cancellation using LMS algorithm are shown in figure 2.

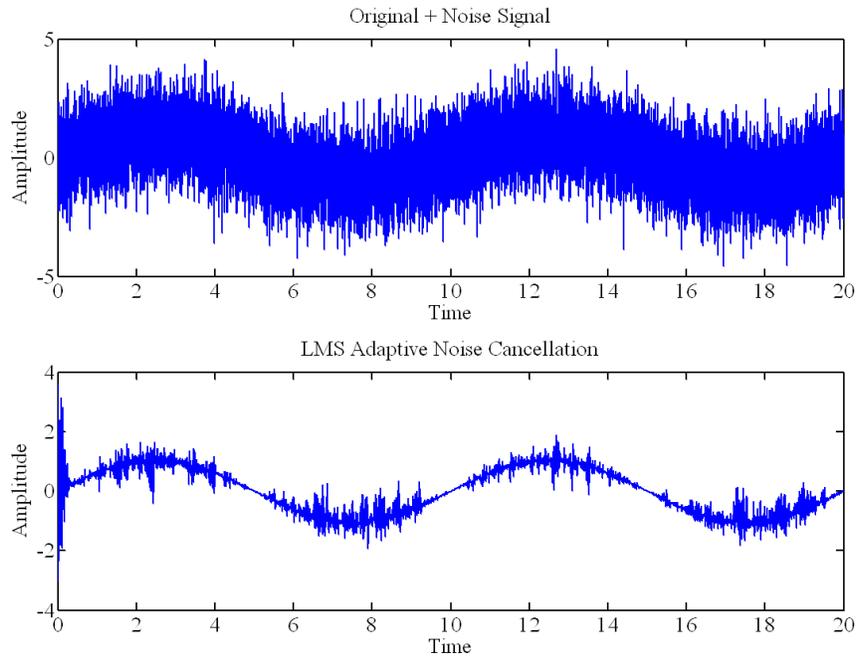


FIGURE 2: Noise Cancellation Using LMS Algorithm.

2.2 NLMS

NLMS is a variation of LMS algorithm having fast convergence as compared to LMS algorithm. Filter coefficient update equation for NLMS algorithm is given by (4).

$$z(n + 1) = z(n) + \mu \frac{v(n)}{|v(n)|^2} e(n) \quad (4)$$

where $z(n)$ is weight vector for NLMS adaptive filter. μ is learning rate and $0 < \mu < 1$. An important limitation of adaptive algorithms is that of selection of certain value for the learning rate μ implies compromise between speed of convergence and steady state mis-adjustments. In simulations, learning rate for NLMS algorithm was set to 0.05 and filter order was chosen 10. Results of noise cancellation using NLMS algorithm is shown in figure 3.

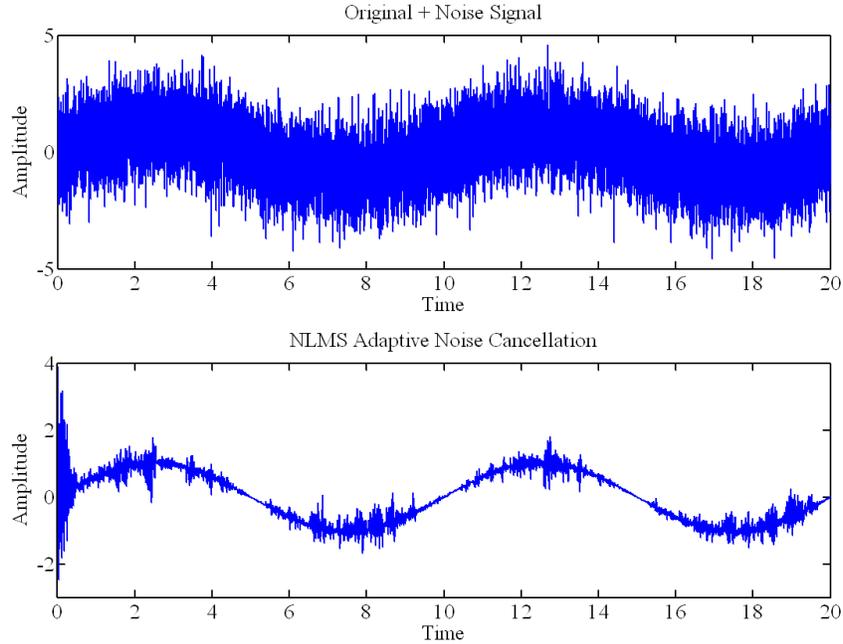


Figure 3: Noise Cancellation Using NLMS Algorithm.

2.3 RLS

RLS is relatively complex algorithm as compared to LMS and NLMS algorithm. Also performance of RLS in terms of convergence and Mean Square Error (MSE) is better than LMS and NLMS. RLS adaptation algorithm with input signals $y(n)$ and $x(n)$ is given below. Initial values for RLS algorithm is given by (5) and (6).

$$\Phi_{yy}(n) = \delta I \quad (5)$$

$$w(0) = w_1 \quad (6)$$

For $m = 1, 2, \dots$ filter gain update vector is given by (7)

$$k(n) = \frac{\lambda^{-1} \Phi_{yy}(n-1) y(n)}{1 + \lambda^{-1} y^T(n) \Phi_{yy}(n-1) y(n)} \quad (7)$$

Error signal equation is given by (8)

$$e(n) = x(n) - w^T(n-1) y(n) \quad (8)$$

Filter coefficient adaptation is given by (9)

$$w(n) = w(n-1) - k(n) e(n) \quad (9)$$

Inverse correlation matrix update is calculated using (10)

$$\Phi_{yy}(n) = \lambda^{-1} \Phi_{yy}(n-1) - \lambda^{-1} k(n) y^T(n) \Phi_{yy}(n-1) \quad (10)$$

Learning rate for RLS algorithm was set to 0.002, forgetting factor was chosen 0.99995 and filter order was chosen 5. Results of noise cancellation using RLS algorithm are shown in figure 4. MSE is also plotted for LMS, NLMS and RLS algorithm in figure 5. It can be seen from figure 5 that RLS performs best in removing noise from original signal and is most suitable for hardware implementation.

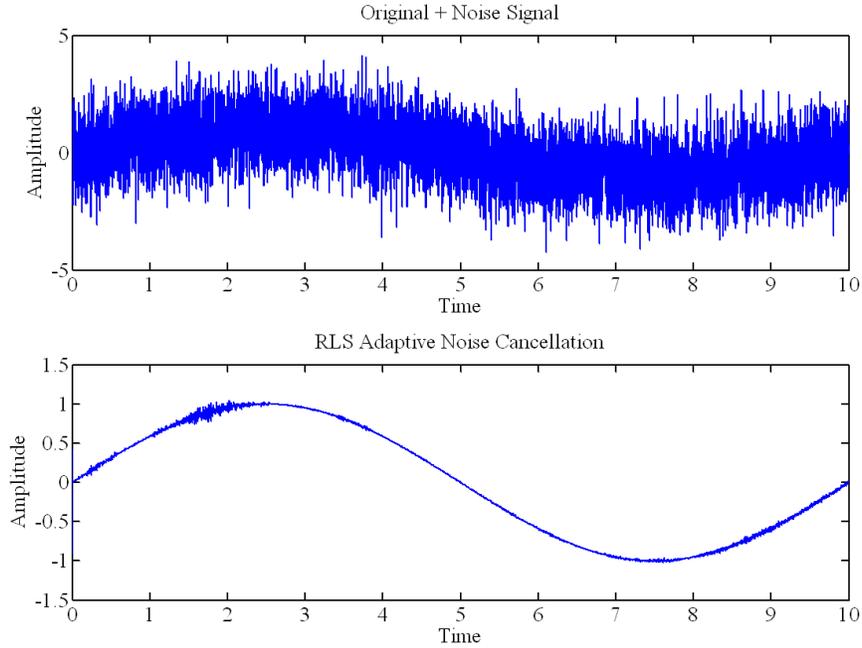


FIGURE 4: Noise Cancellation Using RLS Algorithm.

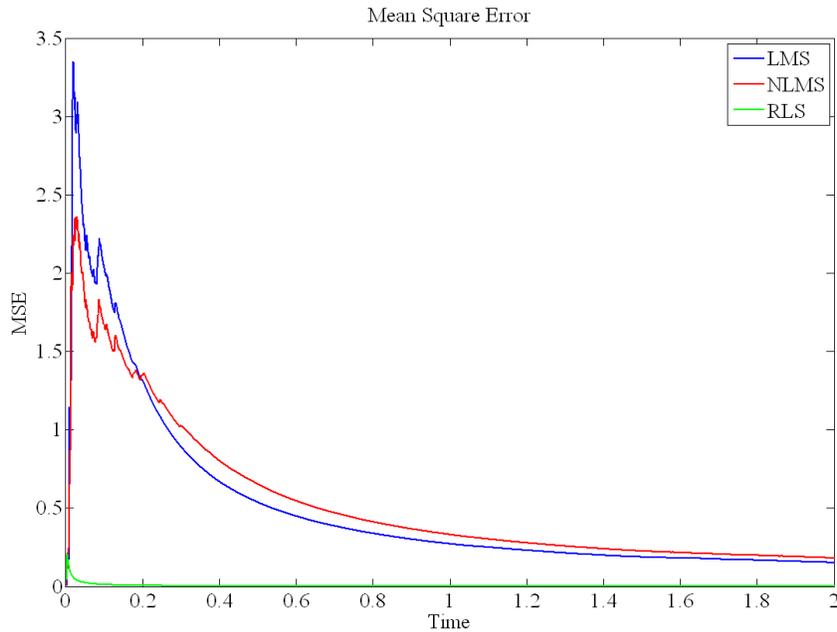


Figure 5: Mean Square Error for LMS, NLMS and RLS Algorithms.

5. DSP KIT

DSP processors are concerned primarily with real-time signal processing [7]. Real-time processing requires the processing to keep pace with some external event, whereas non-real-time processing has no such timing constraint. For non-real time system, data can be recorded on PC and signal processing algorithms can be implemented offline. DSP based systems are less affected by environmental conditions. DSP processors enjoy the advantages of microprocessors

[8]. They are easy to use, flexible, and economical [9]. Perfect reconstruction code was successfully implemented on TMS320C6713 DSP kit.

Embedded Integrated Development Environment (IDE) link between Matlab and code composer studio was used to port the ANC code on DSP kit. ticc objects were used to transfer information to and from code composer studio and with the embedded objects [10]. With embedded IDE link we can get information about data and functions stored in DSP kit processor memory and registers, as well as information about functions in our project. Embedded ANC is comprised of following components and are shown in figure 6.



FIGURE 6: Noise Cancellation Components for Our Implementation via Embedded IDE.

6. HARDWARE IMPLEMENTATION

Best results in simulation were obtained using RLS algorithm therefore we will use RLS algorithm in our hardware implementation. Although computational cost of RLS algorithm is high as compared to LMS and NLMS but our objective is to implement best performance noise cancellation system therefore we will use RLS algorithm. Hardware for ANC was setup for two different methods. Details of both setups and results are given below.

2.1 Method 1

Block diagram of hardware setup for Method 1 is shown in figure 7.

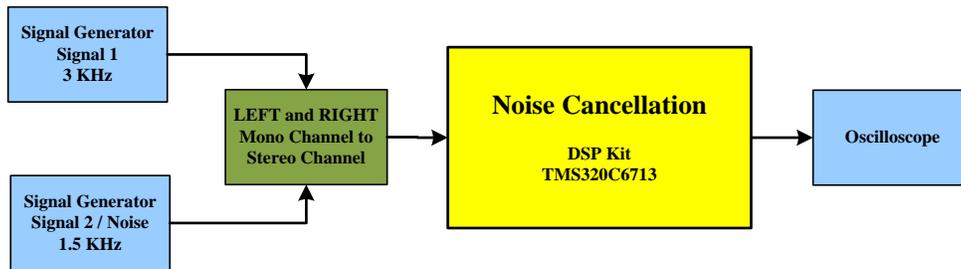


FIGURE 7: Method 1 Hardware Setup Block Diagram.

In this method, two signal generators are used. One is used as original/desired signal source and other as noise source. Mono signals from both signal generators were then converted to stereo signal via stereo connector. Both signals are routed to single cable and is then used to input data to DSP kit. Analog to Digital Converter (ADC) was used to sample the input signal. Block diagram of noise cancellation algorithm running over TMS320C6713 DSP kit for method 1 is shown in figure 8. Signal output from Digital to Analog Converter (DAC) is then routed to oscilloscope for viewing.

Hardware setup for method 1 is shown in figure 9. Two channels of the signal generator provide two required signals i.e. Desired and Noise Signal. Noise cancelled output is shown in figure 10. Oscilloscope displays the output of DSP kit after noise cancellation i.e. noise signal is cancelled by ANC and only desired signal is display at oscilloscope.

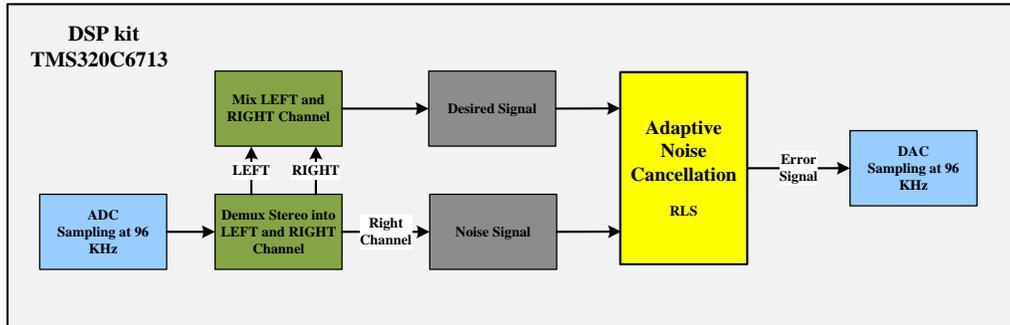


FIGURE 8: Method 1 Noise Cancellation Implementation.

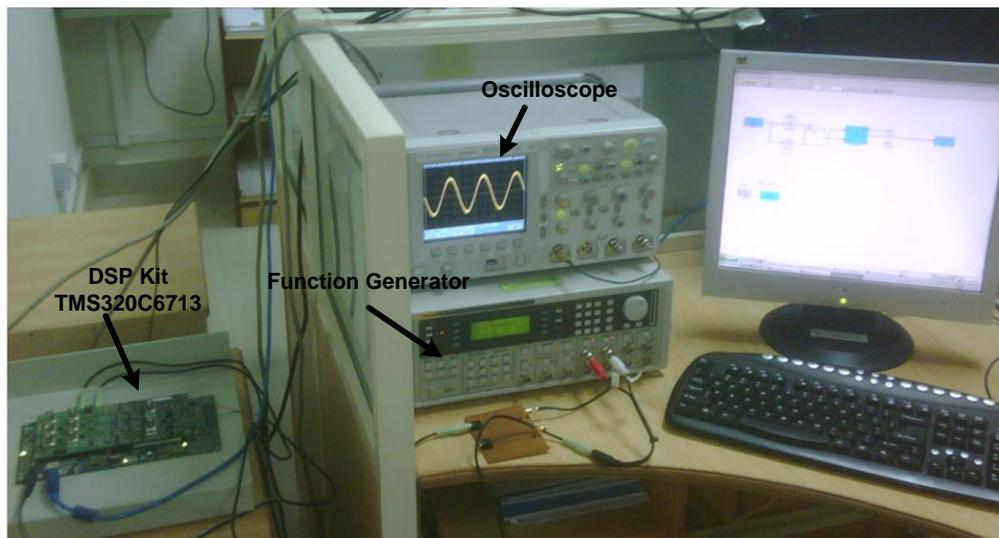


FIGURE 9: Method 1 Hardware Setup.



FIGURE 10: ANC Results for Method 1.

2.2 Method 2

Block diagram of hardware setup for method 2 by microphones is shown in figure 11.

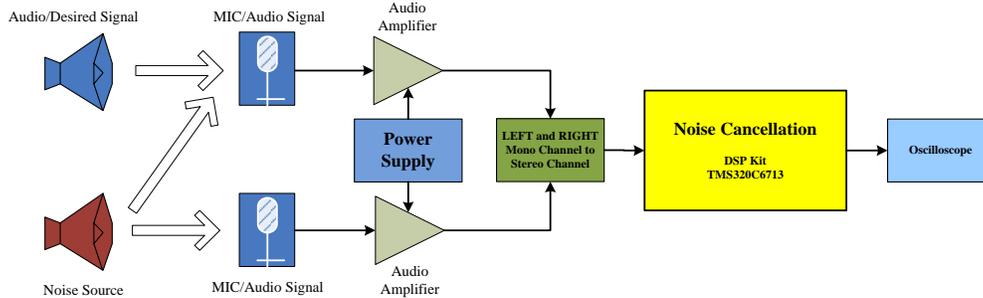


FIGURE 11: Method 2 Hardware Setup Block Diagram.

In this setup, two microphones were used for ANC. In this lab experiment, we have used noise from CPU+AC fan. One microphone captures the noise. Second microphone captures the voice of human + noise from CPU/AC fan. Both signals were then amplified using audio amplifier. Amplified signal then becomes input to DSP kit via "Line In". ADC was used to sample the input signal. Block diagram of noise cancellation algorithm running over TMS320C6713 DSP kit for method 2 is shown in figure 12.

Hardware setup is shown in figure 13. Data captured from microphones before noise cancellation are shown in figure 14. Blue line shows the signal + noise captured from the first microphone. Whereas red line display the noise captured from the fan/AC via second microphone.

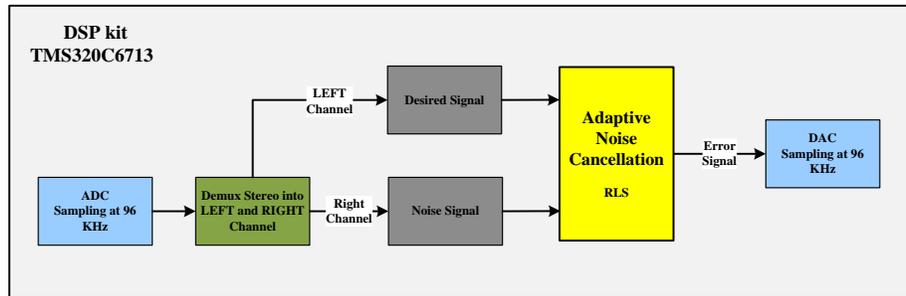


FIGURE 12: Method 2 Noise Cancellation Implementation.



FIGURE 13: Method 2 Hardware Setup.



FIGURE 14: Acquired Data Through Microphones.

Output of noise cancellation from DSP kit TMS320C6713 DAC is shown in figure 15. It can be seen that noise from the fans is suppressed and clean audio signal is obtained from ANC implementation.

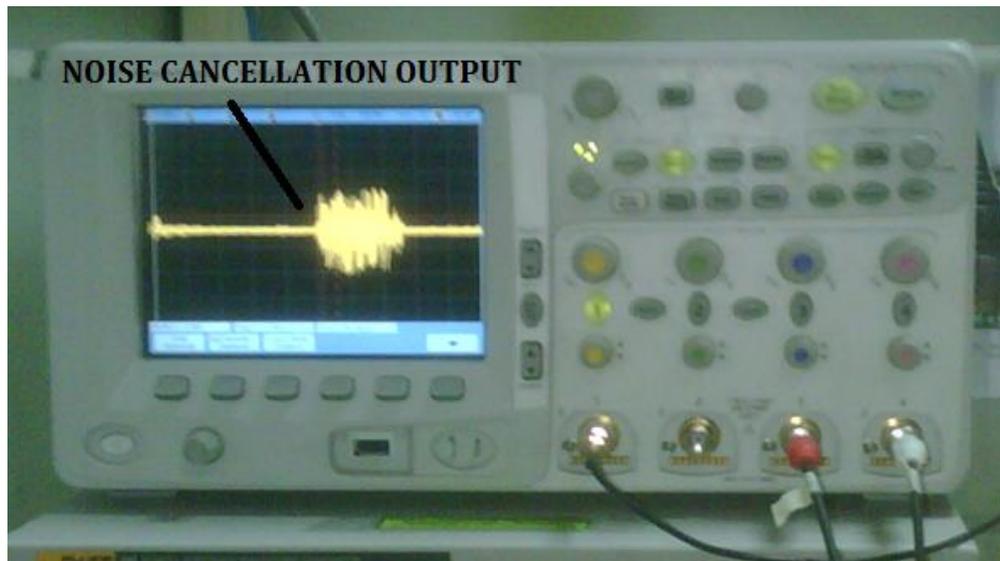


FIGURE 15: Method 2 ANC Result.

7. CONCLUSION

The adaptive noise cancellation system is presented in this paper. LMS, NLMS and RLS adaptive algorithms were used for noise cancellation. Simulation results shows that noise cancellation simulations have been achieved successfully. Real time implementation over DSP Kit TMS320C6713 was also carried out. Implementation was tested using two different approaches i.e. Function generator and Microphones. Experimental results verify effectiveness of implementation. Developed system can also be used for noise cancellation in headphones, cars, aircraft's cockpit etc.

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