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# A Review Paper on Adaptive Noise Cancellation Implementation using TMS320C6713 DSP Board

**Rajshree Tiwari[1], Abhinav Singh[2], Mohd Manazir[3], Shalini Singh[4], Pankaj Goel[5]**

*[1,2,3,4]Student, Dept. of Electronics & Communication, IMS Engineering College, Ghaziabad, Uttar Pradesh, India*
*[5]Asst. Professor, Dept. of Electronics & Communication, IMS Engineering College, Ghaziabad, Uttar Pradesh, India*

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**Abstract -** *In a noisy acoustic environment, any form of signal, whether it be audio signal or video signal, is highly affected by additive random noise. Noise is any unwanted signal which when gets mixed with the desired signal, corrupts it. It is very important to remove noise signal so as to obtain the signal in its purest form. This literature review focuses on the removal of the noise corrupting the original signal using adaptive noise cancellation over DSP Kit TMS320C6713. The algorithm is first implemented using either MATLAB M-file or Simulink block diagram which is then converted to a C code. This C code is then converted to an executable file using Code Compose Studio (CCS) software.*

***Key Words***: **Adaptive noise cancellation, Additive random noise, DSP Kit TMS320C6713, MATLAB, CCS, Adaptive filters.**

## 1.INTRODUCTION

Any signal which travels in the environment gets distorted and corrupted by the noise present in it. Removal of this noise, so as to obtain the original signal, is one of the biggest challenges for everyone. One effective technique to remove noise is Adaptive Noise Cancellation (ANC). The basic concept of ANC is to pass the corrupted signal from a filter which cancels the noise signal and passes the desired signal unchanged. This process is an adaptive process, that is, prior knowledge of the signal and noise characteristics is not necessary.

The algorithms used for cancellation of the noise are required to be implemented on DSP platform for actual performance analysis. TheTMS320C6713 (C6713) DSP processor board is one such processor based on very-long-instruction-word (VLIW) architecture which is suitable for signal processing applications which are computationally intensive. Various noise cancellation algorithms are implemented by different researchers for noise cancellation. LMS algorithm is by far the simplest of all which has been considerably used. Apart from this, many variants of LMS algorithm such as Normalized LMS (NLMS) algorithm, Variable Step-size (VSSLMS) algorithm, Block LMS (BLMS) algorithm are implemented for noise cancellation applications.

The general equation for Least Mean Square (LMS) algorithm is as follows:

⇒ Filter output:

$$y(n) = w(n) * x(n) \tag{1}$$

⇒ Estimation error or error signal:

$$e(n) = d(n) - y(n) \tag{2}$$

⇒ Tap weight adaption:

$$w(n+1) = w(n) + \mu \cdot e(n) x * (n) \tag{3}$$

A block diagram of an adaptive filter used for active noise cancellation is given below.

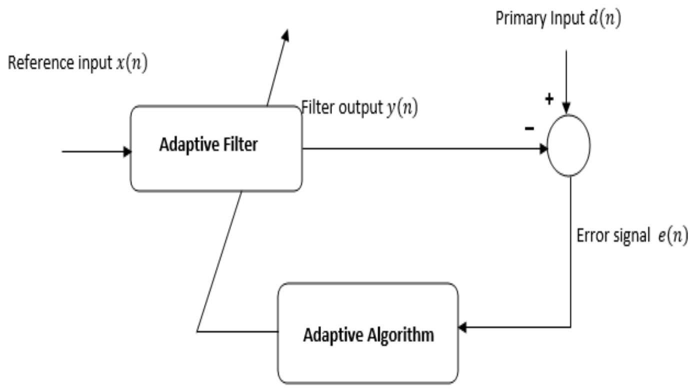


**Fig. 1 Adaptive Noise Canceller System**

## 2. LITERATURE SURVEY

Different researchers used different ANC algorithms to carry out the process of noise cancellation. Some of the important research works are reviewed in this paper.

In voice communication there is a problem of signal interference that can be initiated by acoustic noise and this noise can be removed by the use and implementation of NLMS adaptive filter module over DSP kit in real time which gives the benefits of low cost and easy implantation using Simulink program. This research paper basically focuses on the software based NLMS algorithm to remove the noise in voice communication and it try to achieve clean signal at the output in high fidelity speech network. NLMS filter module was used to remove the noise from the original signal and the result obtained using Simulink simulation shows the effect of modified Variable step size, filtered output and the convergence rate. After that Code Composer Studio (CCS) is used in which algorithm is implemented using C code and then it compiles and prepares a link after that it is loaded into the target DSP processor. The hardware implementation results obtained are similar to the results obtained through Simulink model [1].

Various types of noises are present in the environment which can be removed by implementing adaptive noise canceller (ANC) system using TMS320C6713 DSP Board by the use of different algorithms in automobile environment. In this research paper, the LMS and I-SNRVSS (Instantaneous signal to noise ratio variable step size) algorithms are implemented to achieve ANC. The convergence speed of LMS is found to be slower and the estimated error more as compared to I-SNRVSS algorithm. So I-SNRVSS algorithm is more effective than LMS algorithm for adaptive noise cancellation application [2].

In this research paper implementation of digital adaptive filter on DSPTMS3206713 using a modification of the LMS algorithm, that is, the ECLMS algorithm is discussed. The ECLMS contributes in increasing the convergence speed and decreasing the processing time, hence, reducing the complexity of design because the codified error obtained through the implementation of ECLMS contains integer values. Here, environmental noise canceller is used. In the structure of adaptive filter an input signal, desired signal and the error signal is required which is used to fix adaptive filter parameter. The use of LMS algorithm with codified error (ECLMS) involves the modification to the LMS algorithm, which is done separately and doesn't affect the adaptive filter structure and adaptive algorithm and reduce the complexity of design and increase the convergence factor and also reduce the process time [3].

In this paper the performance of adaptive noise canceller with DSP processor (TI TMS320C6713) using least-mean-square (LMS) algorithm, normalized least-mean-square (NLMS) algorithm and variable step size (VSS-NLMS) algorithm is analysed. Adaptive filters are used to remove noise in non-stationary environments. LMS algorithm is mostly used because of its robustness, good tracking capabilities and simple implementation. If step size (μ) is large, convergence rate is fast but the misadjustment increases. On the other hand if μ is small convergence rate is slow but the misadjustment decreases. To overcome this, NLMS algorithm is introduced which provides the advantage of faster convergence and stable behaviour for a known range of values which are independent of the input data. NLMS algorithm use more computation and to reduce this, the variable step size NLMS algorithm is introduced where the adjustment of step size is monitored by the estimation of signal to noise ratio (SNR). In a room a noise source, a speech source, and primary and reference microphones is located. The signals are collected by real-time using ADC codec of TMS320C6713 DSP board which are sampled at 16 kHz. These signals are processed by the help of LMS, NLMS, and variable step size NLMS algorithms. Variable step size NLMS provides an improvement of 4dB over LMS and 2dB over NLMS algorithm in SNR. VSS-NLMS is more effective algorithm when compared to the LMS and NLMS algorithms in both convergence speed and estimation accuracy [4].

This paper focuses on the use of FLMS (Fast Least Mean Square) adaptive algorithm to remove noise in voice communication systems. The main aim of this paper is to execute a module having the benefit of circular convolution properties of Discrete Fourier transform (DFT) and high computation speed of Fast Fourier Transform (FFT) in frequency domain instead of adaptive algorithms Normalized Least Mean Square (NLMS) and Recursive Least Square (RLS) in time domain which are having more complexity. Also the Simulink implementation is simple. The algorithm used here is the Fast Least Mean Square (FLMS) algorithm. The ANC is modelled with the help of digital filters, specially FLMS algorithm based FFT/IFFT operations and circular convolution frequency domain which requires approximately $N\log_2 N$ real multiplications, thereby, decreasing the computational complexity as compared to RLS and NLMS adaptive algorithms which are having $4N^2$ and 3N+1 real multiplications respectively. The Result of Real-time implementation of the NLMS algorithm is carried out with the following specifications. The Filter order N=32, Variable step size λ= 0.0005 and number of samples per frame in ADC source = 256 [5].

In this research paper RLS algorithm implementation is considered. The recursive least square (RLS) adaptive filter is used to find the filter coefficients that minimize a weighted linear least squares cost function relating to input signals. The RLS is in contrast to other algorithm such as least mean square (LMS) which reduces the mean square error. When compared to the other algorithms, RLS possesses very fast convergence. This paper focusses on noise cancellation which is concerned with removal of noise corrupting the speech signal. The main goal of Least Square algorithm is to minimize the sum of squares of the difference between the required signal and the output at the filter. When each iteration gives new samples of the input signal, it results in the emergence of RLS algorithm. It provides fast convergence and proves to be excellent while working in time varying environment [6].

In this research paper, implementation of ANC on an improved adaptive wiener filter is discussed on Texas Instruments TMS320C6713 DSK and its performance is compared with the Lee's adaptive wiener filter. The National Instruments TI DSP test integration toolkit and adaptive filters toolkit LABVIEW models are used to implement this algorithm. These models are tested with noisy wavelet test data sets and speech/wave files. These models based design of adaptive noise cancellation using Simulink is implemented on TI C6713. Here, the profile statistics of the auto-code generated in the real time using the Simulink models of LMS filter and 'C' implementation of LMS filter on C6713 are compared with each other based on the code length and the computation time. Hence, the signal to noise ratio of the output signal obtained using improved adaptive wiener filter is improved by 2.5 to 4 dB as compared to Lee's adaptive wiener filter. It takes 205ms computation time and 1024 bytes length space by the 'C' implementation whereas 38.95ms computation time and 4032 bytes code length while using the auto-code generated by Simulink [7].

The main objective of this research paper is to reduce the noise content of the signal which is evaluated. There are many algorithms available to reduce the noise content in the signal. The short time spectral amplitude or attenuation (STSA) based algorithm particularly minimum mean square error-log spectral amplitude (MMSE-LSA) is basically used to supress he background noise. However, this algorithm gives rise to musical noise and it has poor performance in low SNR (0-10dB), reverberant and non-stationary noise backgrounds. Another algorithm is the RelAtiveSpecTral Amplitude (RASTA) is available which was originally proposed by Hermanskey and Morgan. This algorithm is used to enhance the performance of the speech recognition systems in reverberating and noisy environment. Hence, the hybrid approach is proposed which uses the combination of both MMSE-LSA and RASTA algorithm which has the dual capacity of enhancing the speech signal corrupted by additive white noise and deteriorated by the reverberation. The hybrid algorithm is more suited for low SNR conditions and reverberant environment. One limitation is that the RASTA algorithm is non-linear and non-causal, hence, less suitable for real time and hardware implementation [8].
In this research paper, a DSP-based oversampling ANC for background noise reduction with the help of RLS (Recursive Least Square) is discussed. Real time experiments were carried out on a DSP board (Texas Instruments TMS320C713 DSK). Sampling rate of 96 KHz was used. The implementation tends to improve the signal to background noise ratio for telephone communication in unfavourable non-stationary acoustic environments. Initial input signal to noise ratios from 0 dB to -10 dB were used for the performance tests and results were analysed in time domain and frequency domain. In this, Mean Opinion Score (MOS) listening test is used along three dimensions following the ITU-T P.835 procedures to personally evaluate the adaptive noise removal output quality [9].

There are several adaptive algorithms of which Least Mean Square (LMS), Normalized Least Mean Square (NLMS) and Recursive Least Squares (RLS) are discussed in this research paper and a comparative analysis is done among them. These algorithms are incited using MATLAB and Simulink and then implemented on TMC320C6713 board. The algorithms are tested on the basis of different parameters such as filter length (N), step size (μ) for LMS and NLMS and forgetting factor (λ) for RLS with performance criteria like MSE (Mean Square Error), SNR (Signal-to-Noise Ratio), Misadjustment and convergence time. It is finally observed that increasing the step size increases the MSE, Misadjustment and convergence in LMS and NLMS algorithms. In case of RLS algorithm, increasing the forgetting factor decreases the MSE, Misadjustment and convergence time and on the other hand increases the SNR. After comparing all the three algorithms, it is concluded that LMS provides low SNR but its implementation is easy, RLS provides the highest SNR but its computational complexity and memory requirement make it costly, hence, not suitable for realization. NLMS lies between LMS and RLS in terms of its performance. Therefore, NLMS is the most suitable algorithm on TMS320C6713 DSP board for real time analysis of the information signals [10].

## 3. COMPARISON TABLE

| S.No. | Title of the paper | Algorithm used | Type of Noise | Input SNR | Output SNR/PESQ/SIG/ OVRL Scores |
|---|---|---|---|---|---|
| 1 | Modeling and Real-Time DSK C6713 Implementation of NLMS for Acoustic Noise Cancellation (ANC) in Voice Communication [1] | NLMS | Acoustic | - | - |
| 2 | Hardware Implementation of an Adaptive Noise Canceller in an Automobile Environment [2] | LMS , I-SNRVSS | Audio | 2.4108dB | -2.4434dB (LMS)<br>-12.6501dB (I-SNR VSS) |
| 3 | Adaptive Noise canceller using LMS Algorithm with Codified Error in a DSP [3] | LMS , ECLMS | Environmental, Electrical Echo | - | - |
| 4 | The Performance of an Adaptive Noise Canceller with DSP Processor [4] | LMS, NLMS, VSS-NLMS | - | 2.0985dB | -0.5567dB (LMS)<br>-2.4662dB (NLMS)<br>-4.2399dB (VSS- NLMS) |
| 5 | Conception and Real Time DSK C6713 of a Low cost Adaptive Acoustic Noise Cancellation (ANC) Based Fast Fourier Transform | LMS | Acoustic | 3.182dB | 20.96dB (T=1sec)<br>30.76dB (T=4sec)<br>26.94dB (T=10sec) |

| | | | | | |
|---|---|---|---|---|---|
| | (FFT) and Circular Convolution for Improving Quality of Voice Communications [5] | | | | |
| 6 | Noise Cancellation in Adaptive Filtering Through RLS Algorithm using TMS320C6713DSK [6] | RLS | - | - | - |
| 7 | Real Time Implementation of Adaptive Noise Cancellation [7] | LMS | - | - | - |
| 8 | Real Time and Embedded Implementation of Hybrid Algorithm for Speech Enhancement [8] | Hybrid of MMSE-LSA & RASTA | Background, Reverberation | 0dB, 5dB, 10dB | 2.6324 (Reverb1)<br>2.8037 (Reverb3)<br>2.9386 (Reverb5) |
| 9 | DSP Based Oversampling Adaptive Noise Canceller for Background Noise Reduction for Mobile Phones [9] | RLS | Background, White Gaussian | 0dB to -10dB | 5(SIG), 4(BAK), 4.1(OVRL) for 0dB<br>5(SIG), 3.9(BAK), 4(OVRL) for -5dB<br>5(SIG), 3.5(BAK), 3.9(OVRL) for -10dB |
| 10 | Comparative Performance Analysis and Hardware Implementation of Adaptive Filter Algorithms for Acoustic Noise Cancellation [10] | LMS , NLMS, RLS | Acoustic | 15dB | 24.4607dB (LMS)<br>25.3211dB (NLMS)<br>37.7446dB (RLS) |

## 3. CONCLUSION

When the information signal travels in the free environment, it gets corrupted by the noise present it. Removing this noise emerges out to be one of the most important concern for everyone. There are many conventional techniques to supress the noise present in the information signal of which Adaptive Noise Cancellation (ANC) is one of the most important technique. ANC uses adaptive filters so as to analyse real time signals which continuously vary with respect to time. ANC involves many algorithms which are implemented so as to cancel the noise. This research paper has focussed on some of these adaptive algorithms.

The basic three algorithms which are implemented are postulated below:

- Least Mean Square (LMS)
- Normalized Least Mean Square (NLMS)
- Recursive Least Square (RLS)

Apart from these three algorithms, various other algorithms are also implemented which, in some or the other way, are derived from these three basic algorithms such as FLMS (Fast Least Mean Square), mean square error-log spectral amplitude (MMSE-LSA), Relative Spectral Amplitude (RASTA), Instantaneous signal to noise ratio variable step size algorithm (I-SNRVSS), LMS algorithm with codified error (ECLMS), Variable Step Size (VSS-NLMS), etc. These modified algorithms tend to optimize several parameters related to the signal, making it more suitable for the noise cancellation.

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## 5. BIOGRAPHIES

**Rajshree Tiwari** is currently pursuing her B.tech degree in Electronics & Communication stream from IMS Engineering College, Ghaziabad. She is a final year student. Born and brought up in Allahabad, Rajshree did her schooling from Mary Lucas School & College, Allahabad. She has a keen interest in electronics field and she aspires to work within an electronics firm after her graduation.

**Abhinav Singh** was born in 1996 in Moradabad. He did his schooling from M.A Inter College, Moradabad. Presently, he is a final year B.tech student. He is pursuing his degree in Electronics and Communication stream from IMS Engineering College, Ghaziabad. He wants to get a reputed job after his graduation.

**Mohd Manazir** is studying in IMS Engineering College, Ghaziabad. He is pursuing his B.tech degree in Electronics & Communication branch. His hometown is Allahabad and he completed his 10th and 12th from Bal Bharti School, Allahabad. His field of interest is to work in electronics domain where he can apply his knowledge and skills to explore and develop himself.

**Shalini Singh,** is born and brought up in Varanasi. She was born in the year 1994. She did her schooling from KV (Kendriya Vidyalaya) BHU. She’s currently pursuing her B.tech degree in Electronics and Communication stream from IMS Engineering College, Ghaziabad. She is a final year student. She wants to work in a good organization after completing her graduation.