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MODELING AND REAL-TIME DSK C6713 IMPLEMENTATION OF NORMALIZED LEAST MEAN SQUARE (NLMS) ADAPTIVE ALGORITHM FOR ACOUSTIC NOISE CANCELLATION (ANC) IN VOICE COMMUNICATIONS

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ABSTRACT

In this paper a module consisting of a Normalized Least Mean Square (NLMS) filter is modeled, implemented and verified on a digital signal processor (DSP) TMS320C7613 to eliminate acoustic noise, which is a problem in voice communications. However the acoustic noise cancellation (ANC) is modeled using digital signal processing technique especially Simulink Blocksets.

The main scope of this paper is to implement the module onboard an autonomous DSK C6713 in real time, benefiting to the low computational cost and the easy implementation using Simulink programming.

The needed DSP code is generated in code composer environment under Real Time Workshop. At the experimental level, implementation phase results verify that implemented module behavior is similar to Simulink model.

Keywords: *Adaptive Algorithm, Acoustic Noise Cancellation (ANC), Real Time Implementation, Digital Signal Processing, DSK C6713.*

1. INTRODUCTION

The signal interference initiated by acoustic noise is a major problem in voice communications. However, the longer the channel delay, the more annoying the noise becomes until it renders natural conversation impossible and decreases the perceived quality of the communication service. It is therefore absolutely essential to avoid retransmitting the noise picked-up by the voice gateways [1].

Acoustic Noise Cancellation (ANC) has emerged as an important technology for communication systems. This is then employed to enhance the quality of voice communications by cancellation the undesirable phenomenon, such as acoustic noise.

DSPs are processors where hardware, software, and instruction sets are optimized for high-speed numeric processing applications, somewhat essential for processing digital data and representing analog signals in real time. Also, the TMS320C6x (C6x) processor family are fast special-purpose microprocessors with a specialized type of architecture as they feature appropriate instruction set based on a very-long-instruction-word (VLIW) architecture for signal processing. This family is a form of embedded design that is one of the hottest spot in the field of signal processing and is considered to be the workhorse of choice for many applications.

Different works involving the noise cancellation adaptive algorithm developed across this paper are presented [2-3-4-5-6].

In this work, the method used to achieve noise cancellation is known as adaptive filtering. This method is frequently used to enhance communication quality by removing line noise. This is why adaptative filters were developed and tested long before on analog bench platforms until a digital based technique breakthrough emerged, the DSP. This new technique allows better signal filtering design and found its benefits in High Fidelity audio systems or speech networks.

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This paper will focus on the software based NLMS adaptive algorithm to remove noise in voice communication systems. The Acoustic Noise Cancellation (ANC) is modeled in Simulink using digital filters, especially adaptive Normalized Least Mean Square (NLMS) algorithm. Finally the real-time characteristics of this module are verified on a Digital Signal Processor (DSP) TMS 320 C6713.

The paper is structured as follows: section II presents digital adaptive filters for noise cancelling, section III presents the DSK TMS320C6713 card, section IV presents simulation results, Section V presents module design and Section VI concludes this paper.

2. DIGITAL ADAPTIVE FILTERS FOR NOISE CANCELLING

Developing a filter that is able to comply with the statistics of the signal is the main scope of adaptive filtering. Adaptive algorithm efficiency depends on three criteria that size up:

- The complexity of computation and the amount of computation executed at each stage.
- The behavior of speed adjustment that permits an adaptive filter to reach Weiner solution.
- The estimated error generated by the dissimilarity between the actual Weiner solution and the adaptive algorithm resolution.

Adaptive cancellation of noise is the main pattern
of adaptive filters.

2.1 Adaptive Filters

In this section we first go through an examination of the filter structure with an emphasis on Finite Impulse Reponses (FIR) filters. This is followed by a review of the Wiener filter leading to the development of the Least Mean Squares (LMS) algorithm.

A noise canceller is a closed loop linear adaptive filter used for direct system modeling. (Fig 1) There are many different combinations of filters and algorithms, depending on the requirements of a particular application, from Finite Impulse Response (FIR) to Infinite Impulse Response (IIR) filters, from Least Mean Squares (LMS) to Recursive Least Squares (RLS) algorithms. For noise cancellation, there is a classical standard adaptive filter formation. The filter part is made up of the most commonly used structure: a FIR filter which is also known as a tapped delay line, non-recursive or feed-forward transversal filter, as shown in Fig 2.

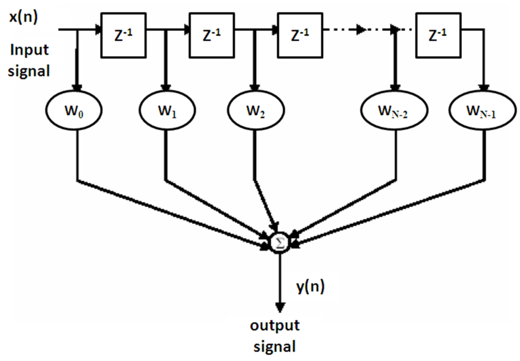

Figure 1: FIR filter structure

The FIR filter consists of a series of delays, multipliers and adders; has one input, x(n), and one output, y(n). The output is known to be a linear combination of the delayed input samples:

$$y(n) = \sum_{k=1}^{N-1} w_K(n)x(n-k) \qquad (1)$$

Where w(n) are the filter coefficients and N is the filter length. Therefore y(n) is the convolution (inner product) of the two vectors w(n) and x(n). This output represents the estimated noise.

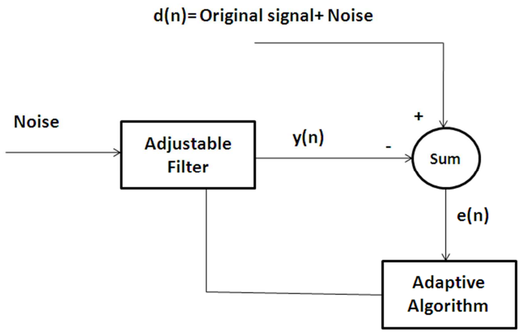

Figure 2: Adaptive filter structure

2.2 Adaptive Noise Cancellation

Among adaptive filters practice, we found the adaptive noise canceller. Fig 3 describes its structure where the requested response is composed of an original signal distorted by the noise, which is uncorrelated with that signal.

The filter input is a sequence of a noised signal which is correlated with the noised signal in the desired signal.

By using the NLMS algorithm within the adaptive filter, the error term e(n) produced by this system is therefore the original signal with the noise signal cancelled [7].

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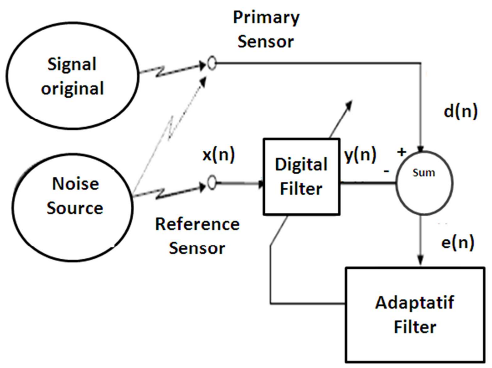


Figure 3: Block diagram of the acoustic Noise canceller

2.3 NLMS Algorithm

The NLMS Filter block shown in Fig 4 implements an adaptive recursively least-square (NLMS) filter, where the adaptation of filter weights occurs once for every block of samples.
The block estimates the filter weights, or coefficients, needed to convert the input signal into the desired signal.

Connect the signal you may want to filter to the Input port. This input signal can be a sample-based scalar or a single-channel frame-based signal.
Connect the signal you expect to model to the desired port. The desired signal must have the same data type, frame status, complexity, and dimensions as the input signal.

The Output port outputs the filtered input signal, which might be sample or frame based. The Error port outputs the result of subtracting the output signal from the desired signal.

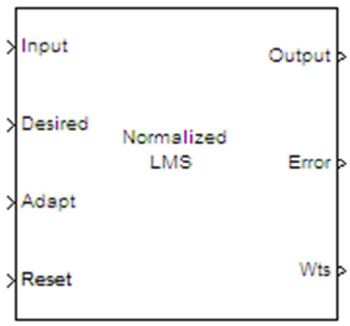


Figure 4: NLMS Filter block [8]

Widrow et al [9] formulated the LMS algorithm for obtaining the minimum output power. Hence, we define the primary input signal to be a delayed one.
We then apply the following modified normalized LMS algorithm which Tracks the most suitable solution.

$$y(n) = w(n).x(n) \qquad (2)$$

$$e(n) = d(n) - y(n) \qquad (3)$$

$$w(n+1) = w(n) + \mu x(n)e(n) \qquad (4)$$

$$w(n+1) = w(n) + \mu \frac{x(n)}{\beta + x(n)^{2}} e^{*}(n) \qquad (5)$$

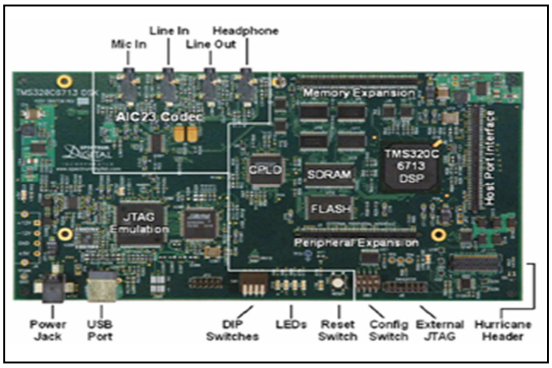
The variables are as follows.

Table 1: Table variable of NLMS Filter

Variable	Description
n	Actual algorithm step
x(n)	Input at step n
ŵ(n)	Array with of adaptive filter values at step n
y(n)	Filtered output at step n
e(n)	Estimated error at step n
d(n)	Desired answer at step n
μ	Step to adjust (must fulfill 0 <μ< 2)
β	Small number inserted in the denominator to avoid division by zero.

3. DSK TMS320C6713

The DSK TMS320C6713 (Figs. 5 and 6) is a development board from Texas Instruments.
It contains the C6713 floating-point digital signal processor (DSP) and a 32 bit stereo codec (AIC23) to handle input and output signals.
The AIC23 codec uses a sigma-delta technology that provides Analog to Digital Conversion (ADC) and Digital to Analog Conversion (DAC) [10].
To develop this application the board must be connected to a PC host, and because it offers a 225 MHz system clock, the variable sampling rates can be set from 8 to 96 kHz.



4. SIMULATION RESULTS

4.1 Noise Canceller Modeling Under Simulink

The overall performance of the module is guaranteed as shown in Fig 7.

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Figure 5: DSK TMS320C6713 board

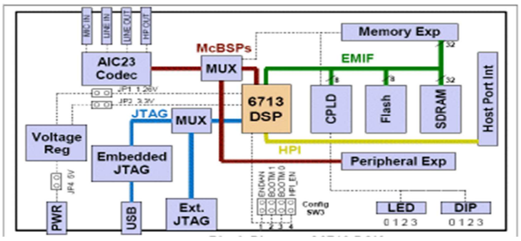

Figure 6: DSK TMS320C6713 block diagram

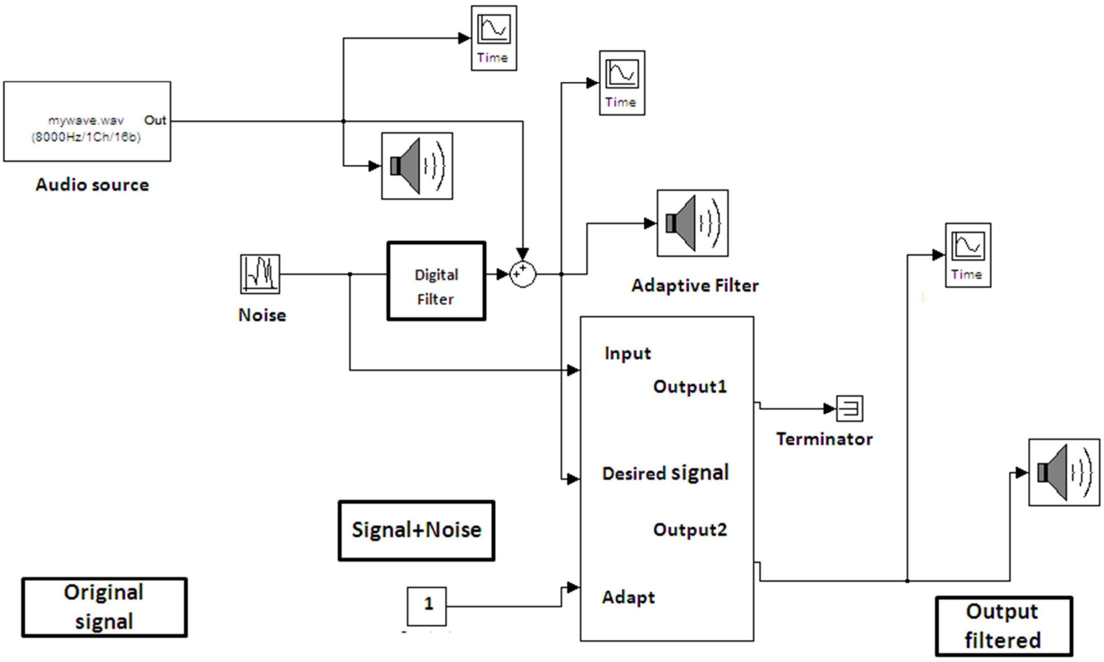

Figure 7: Noise cancelation under Simulink

4.2 Simulink Results

In the following graphics, we observe the input signal, the original signal affected by noise (Figs. 8,9 et 10) and how this noise is removed from the original signal after crossing by the "noise cancellation NLMS Filter" module.

In this work we modeled the system under Simulink Blockset. We also used an audio data with 8000 Hz sampling rate. ANC implementation is setup with NLMS adaptive filter of length 32. The

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variable step size is chosen as μ = 0.002.

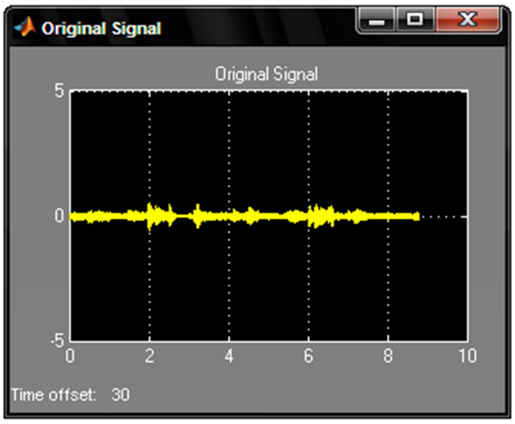

Figure 8: Result obtained using Simulink simulation (Original signal)

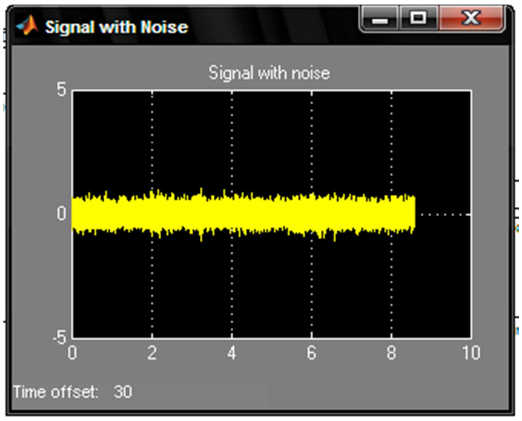

Figure 9: Result obtained using Simulink simulation (Noised signal)

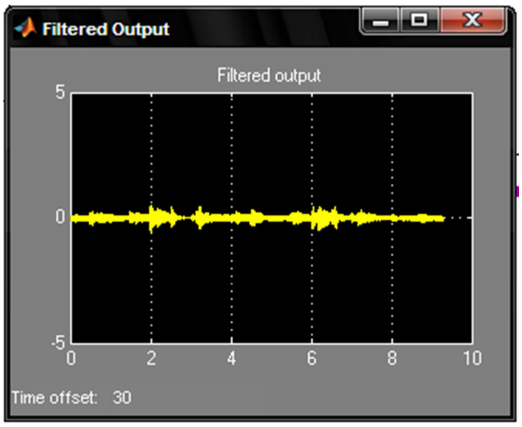

Figure 10: Result obtained using Simulink simulation (Filtered Output)

The effect of modifying the Variable step size, the filter length, the delay value on the convergence rate and obtainable performance is tested [11], [12]. The noise signal is switched between an input signal - a wav file - and a square wave.
It should be verified that a shorter filter length is required to obtain the desired cancellation while using a wav file as the input signal. Informal hearing tests should prove that the system is working properly: the periodic signal is almost cancelled whereas the speech maintains its regular quality.

5. MODULE DESIGN

5.1 Real Time Implementation and Testing

In the following paragraphs, the module implementation on C6713 DSP is discussed. In Layman's terms, the module functionalities are exposed as independent blocks which are thereafter mixed into a single program that integrates C code inside the Code Composer Studio v3.3 (CCS) environment. The CCS compiles it, prepares necessary links, and then loads it into the target processor. Finally, the DSP processes the implemented algorithm and executes the code as shown in fig 11.

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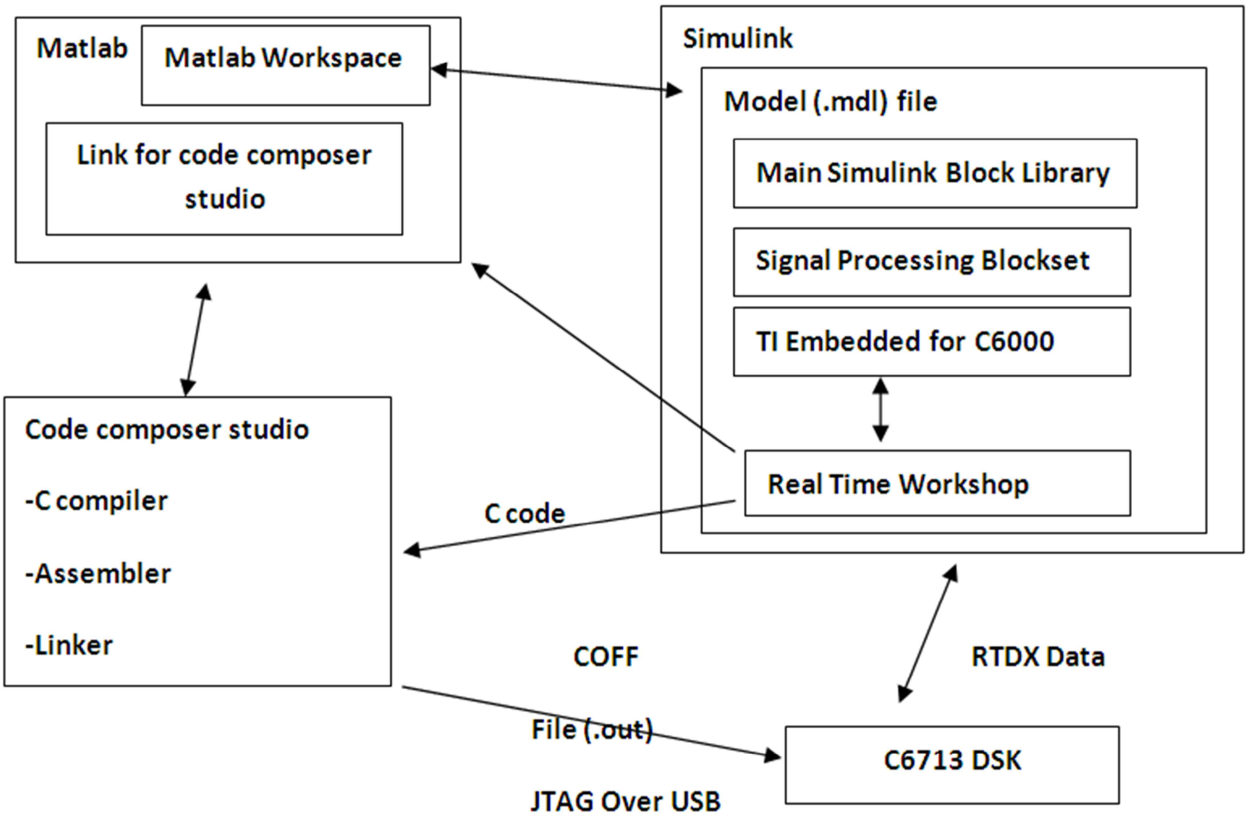

Figure 11: Real-Time Workshop Program compilation, linking and execution

5.2 Real Time Implementation and Testing

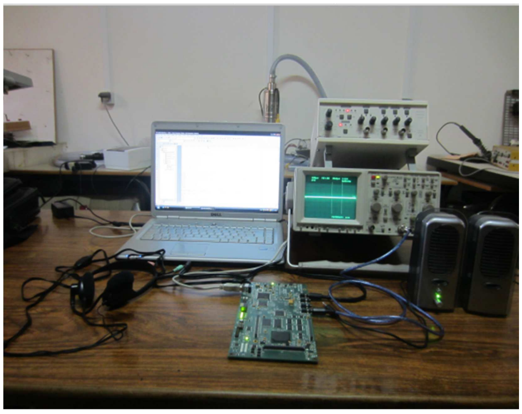

Figure 12: A typical station setup

Using the work workstation setup (Fig. 12), it has been possible to achieve noise cancelation at the experimental level. Result of implementation phase verifies that implemented module behavior is similar to Simulink model.

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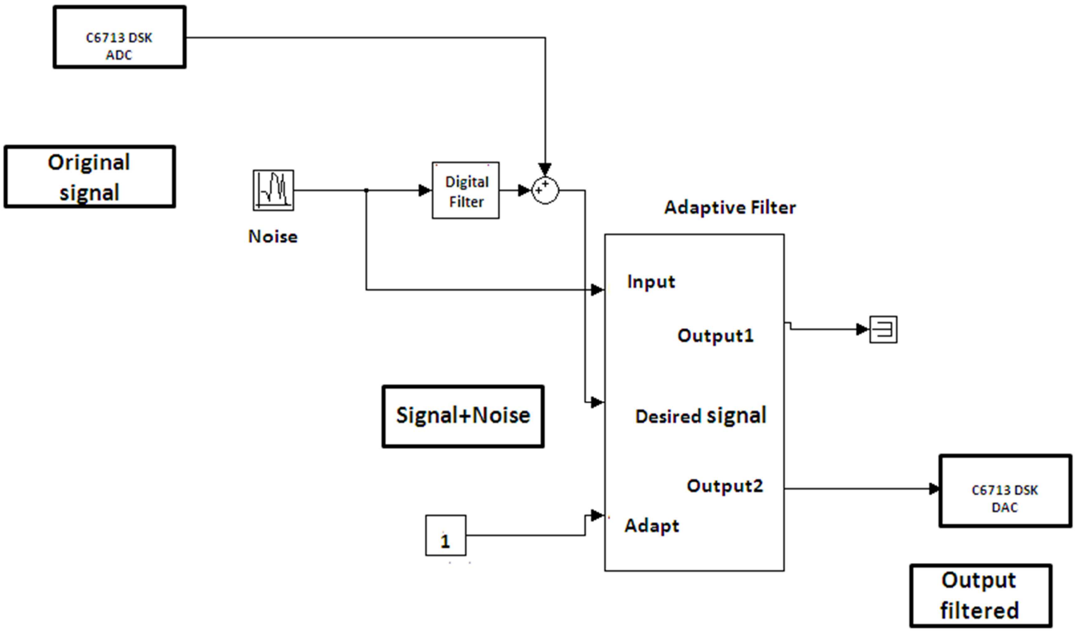

Figure 13: Real-time Noise canceller system implementation

5.3 Experimental Results

In real world application, the module was tested and led to the following results. We notice that this module is a real-time process and that the graphs are similar to those generated using Simulink simulation.
The following figures give an idea of what is produced by the module. Then, in order to verify proper switching of the module input, noised and output signals are probed on an analog oscilloscope as illustrated respectively in Figs 14, 15, 16.
The Result of Real-time implementation of the NLMS algorithm is carried out with the following specifications:
Filter order N=32, Variable step size μ= 0.002

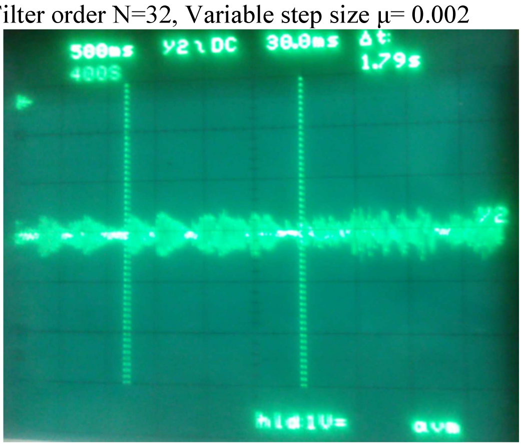

Figure 14: Result of Real-time implementation Original signal in DSK C6713 with CCS environment

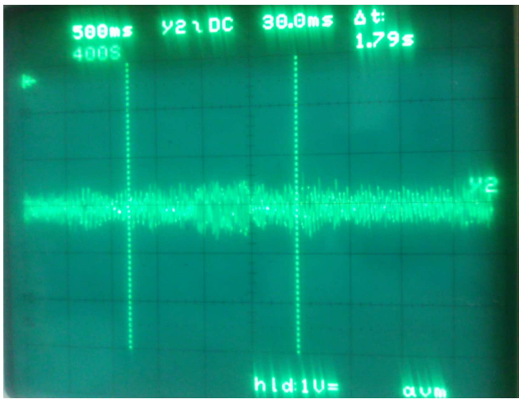

Figure 15: Result of Real-time implementation Noised signal in DSK C6713 with CCS environment

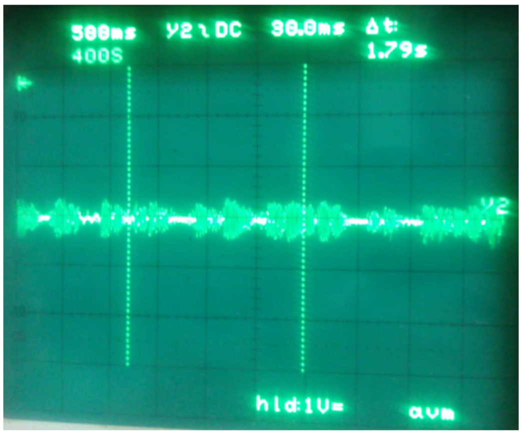

Figure 16: Result of Real-time implementation Filtered output in DSK C6713 with CCS environment

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6. CONCLUSION

In this paper, we have tried to implement a real-time NLMS adaptive filter module within the DSK TMS320C6713 experimentation board. This module, consisting of software blocks rather than electronic blocks, was specifically designed to provide noise cancelation in a voice communications system to achieve ideal sound reproduction as in high-fidelity systems.

The NLMS algorithm has a best capacity of tracking the stationary of signals, such as speech or sound, it also has a low computational cost, compared with the recursive algorithme.This algorithm has very high convergence rate with high computational cost, but it is robust for stationary environment.

In the future work we will focus on adaptive algorithms with low complexity and high computation speed.

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