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Acoustic Echo Cancellation with Adaptive Filtering Using Texas Instrument DSK6713

Hareetaa Mallani[1], Santosh Kumar Prajapati [2]
[1]Asst Prof., [2,]Student, Dept. Of Electronic and Communication Engineering,
MLV Textile & Engineering College, Bhilwara, Rajasthan, India

Abstract: Acoustic echo cancellation is common in modern telecommunications systems. It occurs when the audio source and receiver are in full-duplex mode. Echo signal interference can distract users and degrade connection quality. This paper investigates and implements the signal processing algorithms least mean square LMS, Recursive least squares RLS and normalized least mean square NLMS and has found the best method for suppressing sound echo. The algorithm was first developed in MATLAB and then implemented in C using Code Composer Studio IDE on the DSK6713 DSP card from Texas Instruments. The paper aims to suppress the acoustic Echo generated by a system that includes an adjacent speaker and a speaker that transmits the flow signal.
Keywords: NLMS, LMS, RLS filters, Code composer studio IDE

I. INTRODUCTION

The Acoustic Echo Cancel (AEC) algorithm removes unwanted Echo, reverberation, and additional sound from the signal passing through the acoustic space. As shown in fig 1, the remote person's voice (Far End In) is transmitted parallel to the DSP and acoustic paths. The acoustic path Consists of a sound amplifier/loudspeaker, any audio environment, and a microphone that signals to the DSP. The AEC is based on an adaptive filter. The algorithm continuously updates update this filter to model the acoustic path. The filter output is subtracted from the acoustic path signal to produce a "clean" signal output; most of the linear parts of the acoustic echoes are removed. The AEC also calculates residual signals containing linear acoustic signals. The signal is sent to the remaining echo cancellation, which retrieves the input signal. The signal (optional) generates output via a noise reduction function called a "remote output.

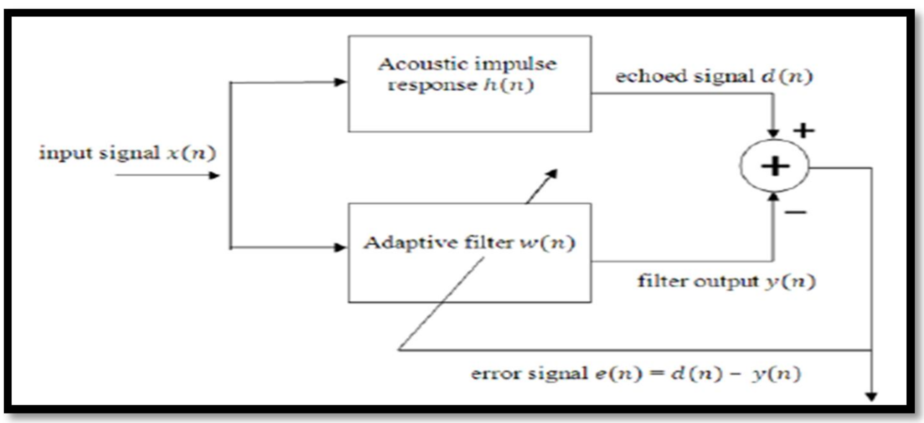

Fig. 1 Common Architecture of Adaptive Acoustic Echo Cancellation

If the filter detects sounds in the sound path, it is not related to the distance and stops converting. This allows incoming sound to be added remotely. For example, when one speaks directly into the microphone in a hands-free system or speaker, the fitting stops. People in the distance can only hear the voice of the local speaker, but there are no echoes and reverberation from the far end. This is essential for clear and full-duplex conversations over the communication channel. The AEC removes the Echo and the linear part of the Echo. To design such a model we are using hardware board to implement Acoustic Echo Cancellation Using Texas Instrument board

II. TEXAS INSTRUMENTS TMS320C6713 Board

The Texas Instruments TMS320C6713 and TMS320C6416 Digital Signal Processing Starter Kits are low-cost development platforms for real-time digital signal processing applications. The kit uses USB communications for true plug-and-play functionality. The DSK also serves as a hardware reference design for the TMS320C6713 DSP.

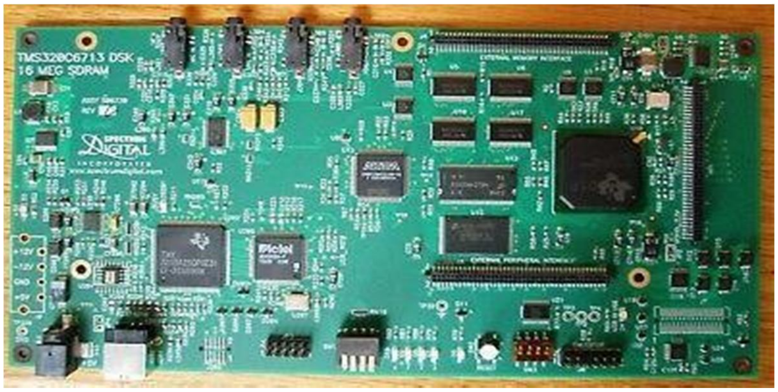

Fig. 2 Top view of TMS320DS6713 DSK kit

III. APPROACHES

There are three main filters which are commonly used for Echo Cancellation

1) Least-Means-square (LMS)
2) Recursive Least Squares (RLS)
3) Normalized Least-mean-square (NLMS)

A. LMS

The least-mean-square (LMS) algorithm is one of the most popular echo cancellation adaptation methods and is used. In the adaptive filter, the filter coefficients are updated as follows:

$$h\hat{}(n+1) = h\hat{}(n) + \Delta h\hat{}$$

The LMS algorithm updates the filter coefficients according to the negated gradient of mean square error as follows:

$$\hat{h} = -\frac{\mu}{2}\nabla E[e^2(n)]$$

Where μ is the step size according to the integration rate, ∇ is the slope, and E is the expected square error, but for the LMS algorithm, the instantaneous value is changed. After some operation, the improved filter coefficient:

$$h\hat{}\ (n+1) = h\hat{}\ (n) + \mu\ e(n)\ x\ (n)$$

B. RLS

Recursive Least Squares (RLS) is an adaptive filtering mechanism that repeatedly finds coefficients that reduce the weighted linear minimum square cost function associated with the input signal. This method contradicts other methods, such as least mean squares (LMS), which aims to reduce the root mean square error. Compared to most of its competitors, RLS shows a very fast integration rate. However, this advantage comes at the expense of higher computational complexity. The RLS method is very effective and robust, but it is computationally expensive and therefore unsuitable for embedded systems.

C. NLMS

The Normalized Least Means Squares (NLMS) algorithm has some drawbacks; For example, the input μ depends on the vector; Which means that for different values of the input signal, the step size must be changed, another is proportional to the error input signal x (n), then the input signal x (n), which is proportional to the refractive power of the filter coefficient, and is affected by the noise. One way to solve this problem in the default LMS is to use the power of the input signal x (n) to normalize the update area:

$$\hat{h}(n+1) = \hat{h}(n) + \frac{\mu}{\|x(n)\|^2 + a}$$

Where a is a protection value to avoid division by zero, in this approach, μ can be between 0 < μ <1

IV. LITERATURE SURVEY

Acoustic Echo Cancellation Algorithm has been developed using the Adaptive Filtered Algorithm in Embedded Systems and with standalone devices. These experiments were performed using techniques like RLS filter, LMS filter, and NLMS filter. This Adaptive filtering provides the challenge of reducing the Echo and the noise and how it can be improved in future Communication systems.

V. DESIGN METHODOLOGY

Acoustic Cancellation (AEC) Removing the barriers of unwanted and resonant sounds with the acoustic signal passing through the developer. As shown in the figure depicts one, there is a voice on the part of the person to talk about far away, it is known to the end, which was inserted in the path of the path, and DSP parallel to external acoustics. The path consists of an acoustic amplification/loudspeaker, an acoustic film, and a microphone and is returned to the DSP signal.

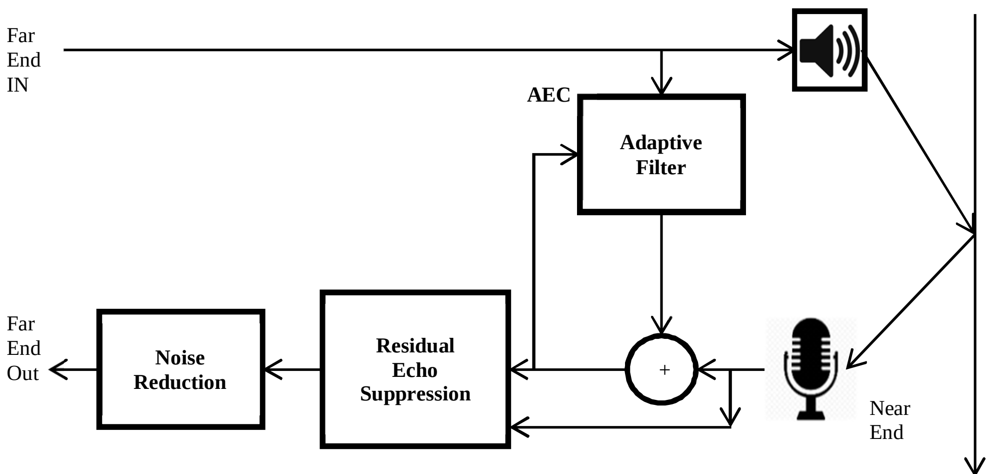

Fig. 3 Main Architecture of Adaptive Acoustic Echo Cancellation

Adaptive filters are useful when the behavior of the channel varies over time. Before implementing the concept on the DSP kit we need to look after following parameters

1) Prototype in the MATLAB software to find the Simulink results and the voice graphs
2) Finding the suitable adaptive filtering technique
3) Choosing the best adaptive techniques suitable for the DSP kit

Implementation of LMS algorithm leads to few drawbacks

a) μ (step size) depends on the input vector; it means that for different values of the input signal, the step size should be changed,
b) The error is proportional to the input signal x(n), then the updated part of the filter coefficients is proportional to the power of the input signal x(n), and it suffers from the presence of noise.

Implementation of RLS algorithm to design Adaptive Acoustic Echo Cancellation whose results are shown in Table below which are also not upto mark. Although the RLS method is the most effective and robust technique, its only drawbacks are that more computational heavy, which makes it unsuitable for use in an embedded system like the DSK 6713 kit.

One way to avoid this problem in the LMS, RLS is to normalize the updated part with the power of the input signal

- Implementing in the MATLAB Simulink using the suitable filters and also changing the step size μ to get better output results. Finding the computational time and also the Echo is measured as echo return loss (ERL)
- Build the model in the DSK6713 kit Using the embedded coder package Provided by Texas instrumentation in the MATLAB Simulink software

All this process is shown in Flow Diagram.

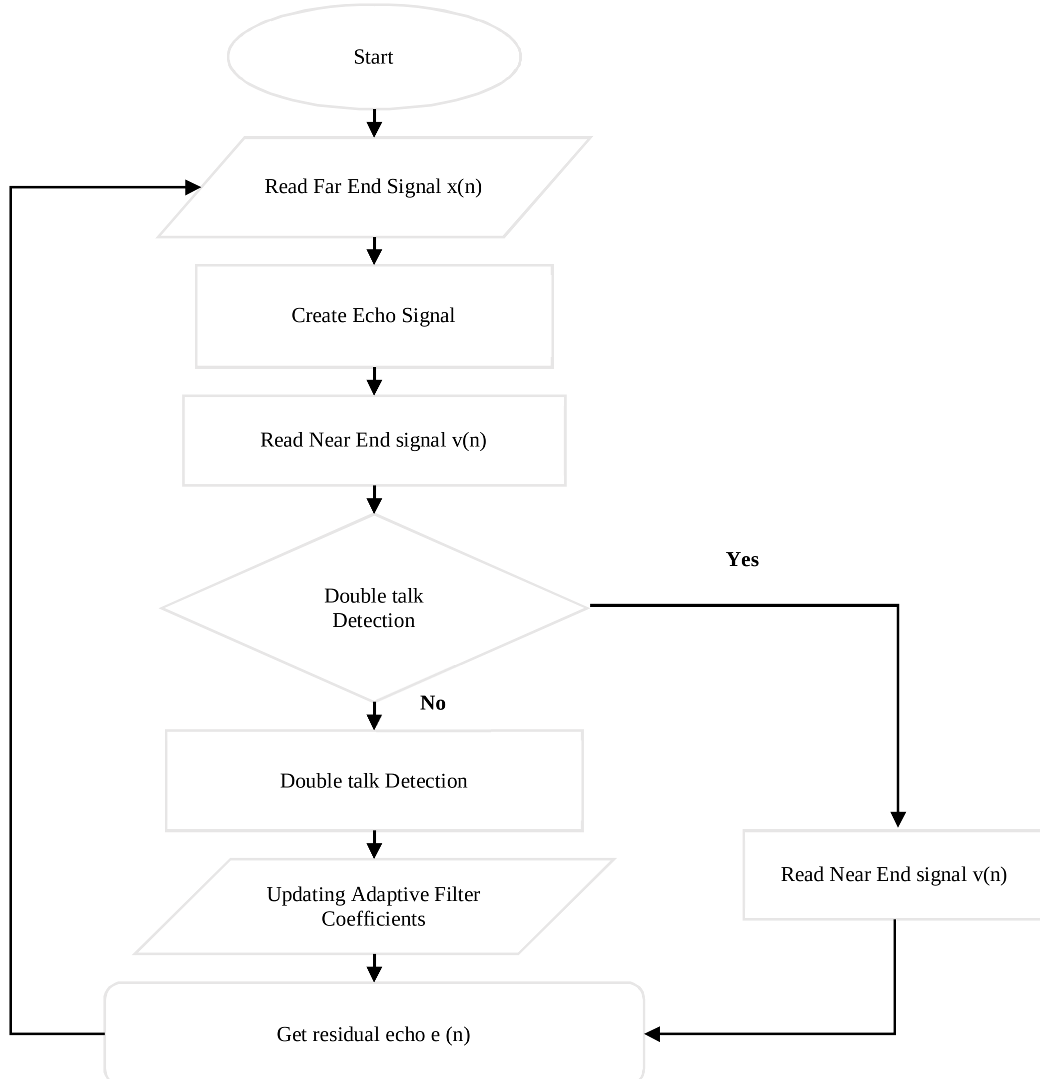

Fig. 4 Flow Diagram

VI. HARDWARE SIMULATION

For the hardware implementations on the DSK6713 kit

1) *Step 1:* Implement the acoustic echo cancellation Using the adaptive filter algorithm Simulink model on MATLAB 2013a.
2) *Step 2:* Connect the pc to the Texas Instrumentation DSK6713 kit using the USB cable and ensure that the kit is properly connected to the PC, checking all the compatibility that the CCS code composer studio communicates to the Matlab Simulink using the "file setup" command on Matlab
3) *Step 3:* Using the embedded code software provided by Texas instrumentation to directly built the Matlab Simulink model to the DSK6713 kit by using the ".out" file from the Matlab Simulink blocks.
4) *Step 4:* We have to Generate the desired signal for this purpose using the function generator to generate the Sine wave with a particular frequency and make it an input to the dsk6713 kit
5) Step 5: To add noise, we use an earphone mic to add some noise in the Sine wave and make another input to the DSK 6713 kit for the project.

Figure 5 Earphone mic to introduce far end noise

6) *Step 6:* In the presence of external noise, Check the output results from the Dsk6713 through the output pin using the aux to video/audio cable. Using connectors and connecting to the DSO/CRO and finding the result in the DSO that is still in the presence of noise, we get output filter sine wave.

Figure 6 Function generator to generate desired Sine Wave

VII. RESULT ANALYSIS

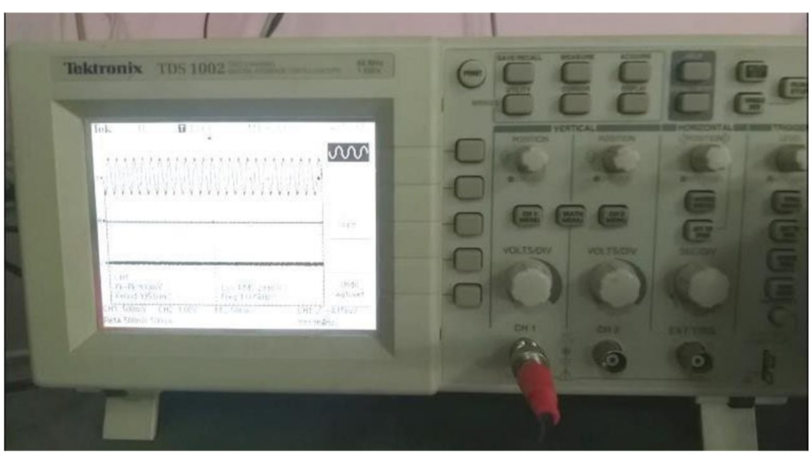

Figure 7 CRO for checking the results of filtered Sine wave

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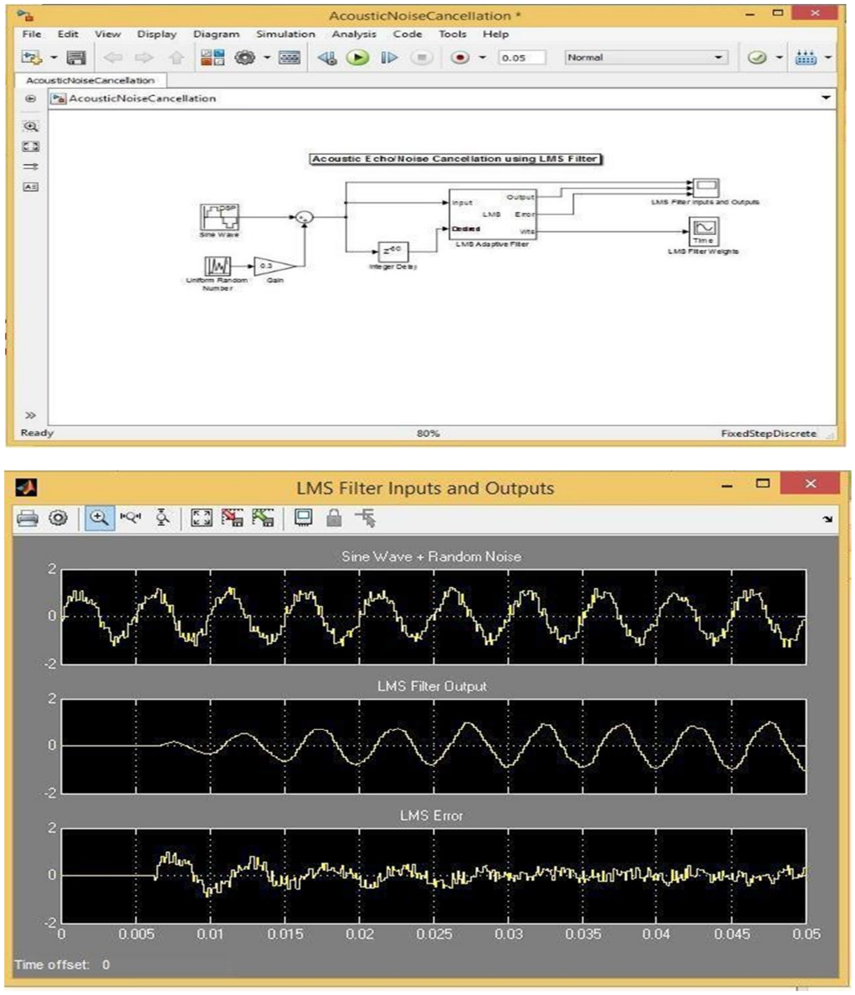

Fig 8 - LMS filter Simulink Model

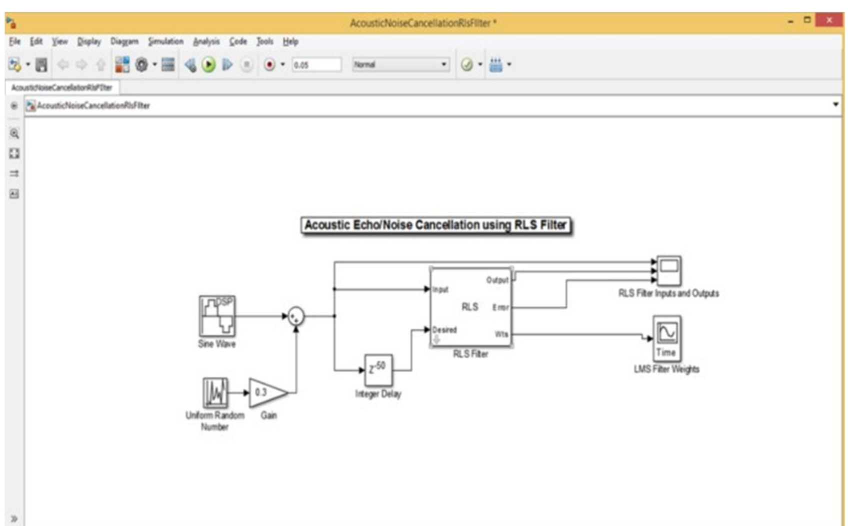

Figure 10 RLS filter Simulink model

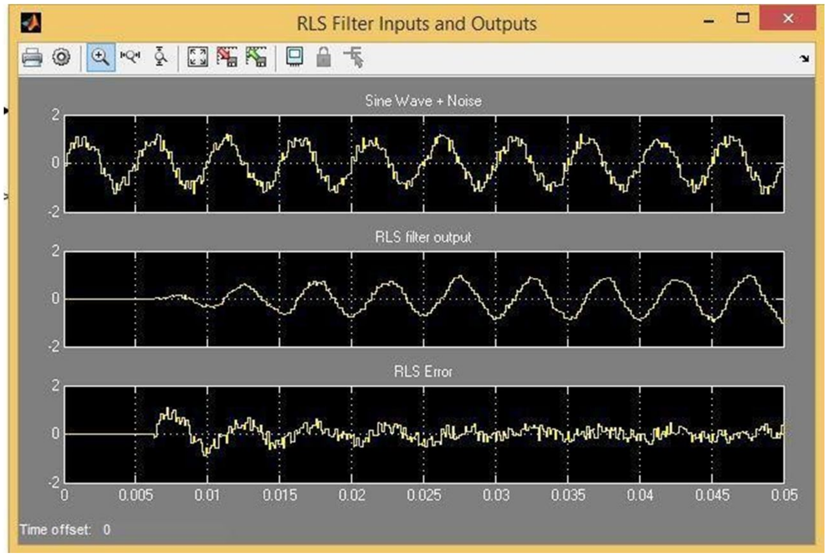

Figure 11 RLS filter outputs with RLS error

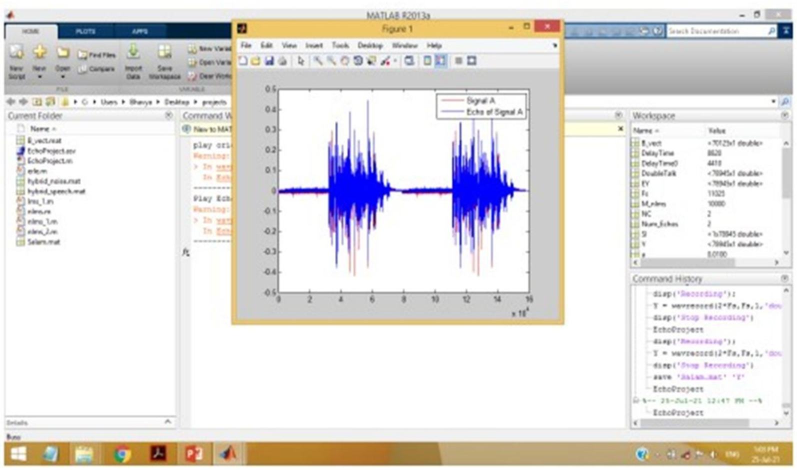

Fig 12 Graph show signal A and Echo of signal A

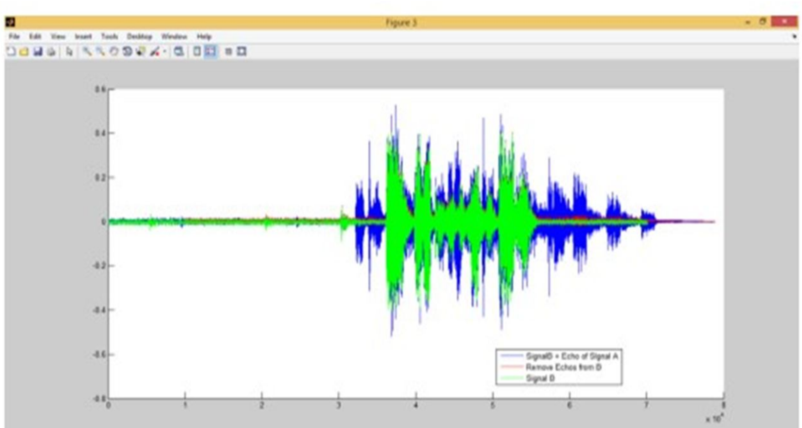

Fig 13 Graph show the Filtered B signal by using NLMS Filter

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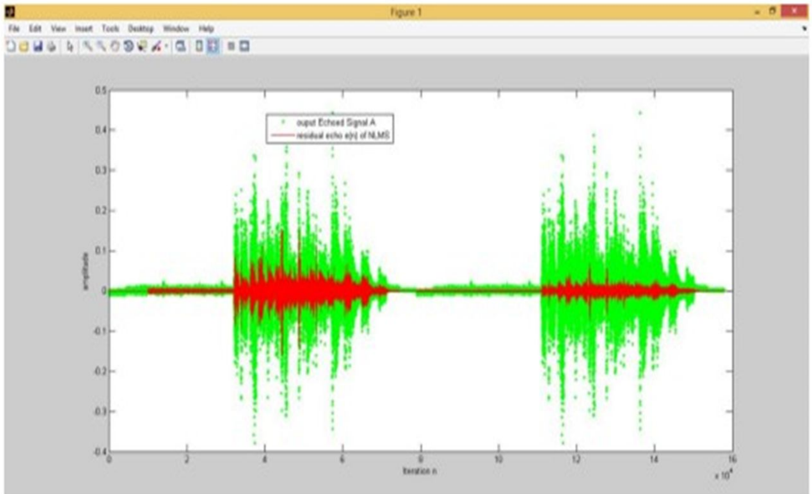

Fig 14 Graph show Echoed Signal A vs NLMS weights

Algorithms	Convergence rate	Mean Square Error	Computation Complexity	Stability	Signal to Noise Ratio
LMS	More iteration required to converge.	Mediate	Simplest	Sensitive to scaling of its input, therefore it's hard to choose step size to give stability	Lower (50db max)
RLS	Less iterati[illegible] than LMS t[illegible] converge	[illegible]	[illegible]	[illegible]	Moderate (55db max)
NLMS	Less iteration to converge, Faster than LMS and RLS	Lowest	Simple	Stable with a small value of step-size	High (80db max)

Fig 15 Table showing Comparison of Various Algorithm

Fig 12 Graph show signal A and Echo of signal A

VIII. FURTHER WORK

A. Inbuilt Echo Cancellation Filters In-Car System

As the communication inside the car produced Echo especially when we use Bluetooth device of call for calling therefore if therefore if there is an adaptive technology in the inbuilt system of the car, then the listening to the other side is very much benefited.

B. Placing AEC (acoustic echo cancellation) in the subsystem of teleconferencing.

Suppose in the subsystem there is already placed adaptive algorithms technology for the echo cancellation, then during the teleconferencing communication. In that case, the listeners will have received the authentic voice of the speaker, and there is no chance of destruction due to the Echo produced by the speaker side.

C. Online Meeting

When the echo cancellation technique in the real-time is faster, then during the online meeting, the echo producers can be reduced on a real-time basis, and the listeners will listen to the accurate voice, which is sometimes it destructed due to the Echo produced.

Fig 13 Graph show the Filtered B signal by using NLMS Filter

IX. CONCLUSION

The aim of the algorithms is to achieve higher ERLE (Echo return loss enhancement) at a higher rate of convergence with less complexity.

The adaptive algorithms for echo cancellation were successfully implemented in MATLAB. The NLMS algorithm has less computational complexity and is very easy to implement in real-time.

The RLS algorithm also gives the best results. Still, it is not used because it requires 4n2 multiplications per iteration; as for echo cancellation systems, the order is usually in the thousands in real-time. Thus, the number of multiplications required is very large, making the RLS algorithm too costly to implement.

The identification system was successfully implemented in a Digital Signal Processor since it not only was easy to mount but also exploited at maximum the development board DSK C6713 specifications; besides, despite the few resources for research and hardware fabrication, this technological tool was appropriate and convenient due its low cost and its compatibility with Matlab platform.

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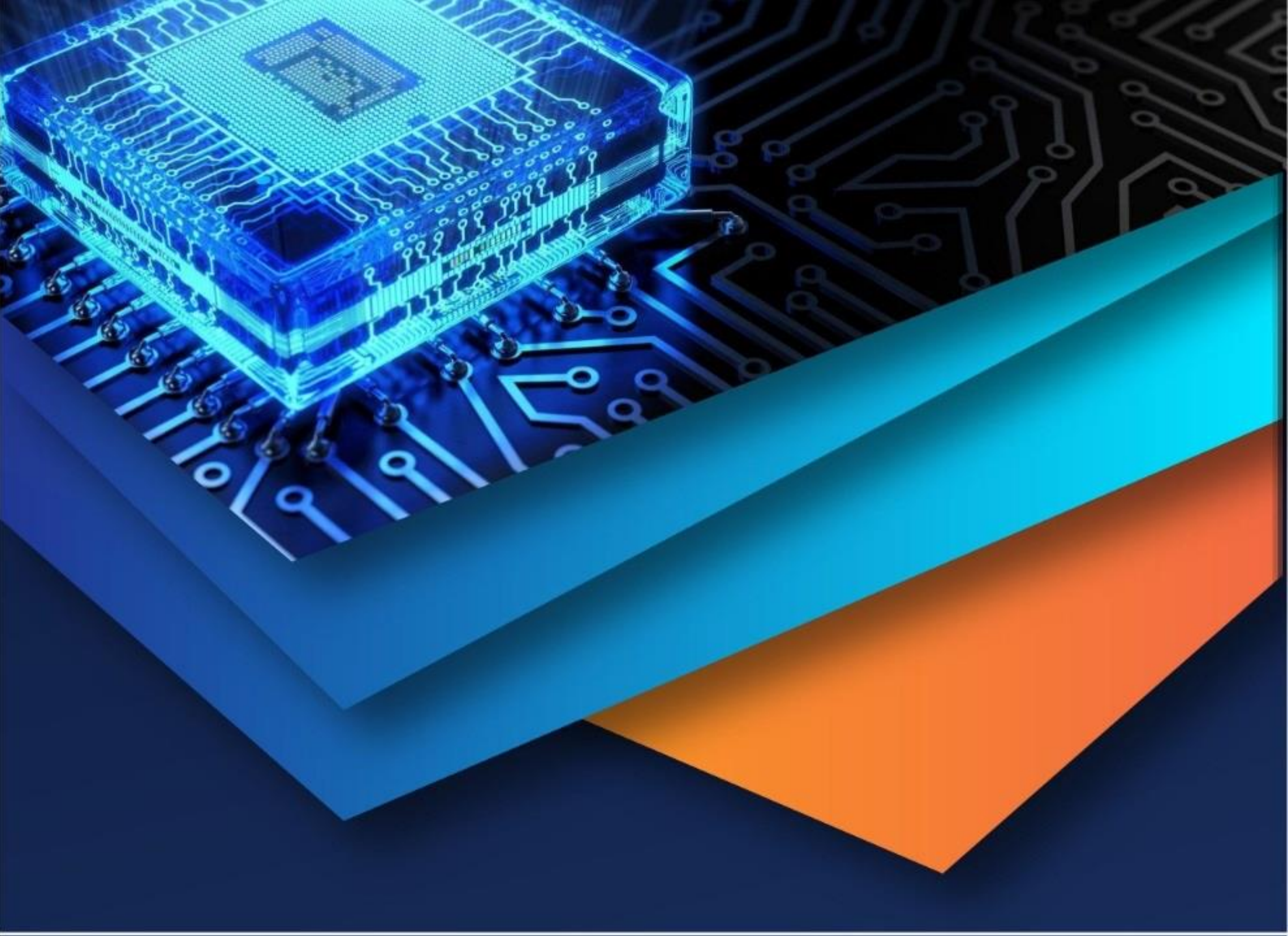

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