*Journal of Information & Communication Technology*
Vol. 9, No. 2, (Fall 2015) 42-54

# REAL TIME ADAPTIVE FILTER IMPLEMENTATION FOR SYSTEM IDENTIFICATION AND NOISE ELIMINATION

**Ali Akbar Siddiqui[1] and S Hassan Ahmed[2]**
*Department of Telecommunication Engineering at Sir Syed University of Engineering & Technology*

**Habib Shaukat[3], M. Shafiq[4] and Dr. M. Tahir Qadri[5]**
*Department of Electronic Engineering at Sir Syed University of Engineering & Technology*

## ABSTRACT

*Filters are the most important instrument that play a vital role in signal processing application such as audio equalizers, image processing etc. So in our proposed algorithm, we have utilized an adaptive filter for system identification and noise elimination. Identification of the system is necessary for the filter implementation; this is accomplished using Least Men Square (LMS). Noise is a feature that varies continuously if the device is used remotely, so we require an adaptation technique to cater any kind of noise that can affect our system and negate the effects of that noise for providing a better signal quality. Mean Squared Error is reduced at a certain interval of time as it adapts to the original signal. Our proposed algorithm performed according to the desired expectations and can be implemented in numerous audio applications.*

**Key Words:** Adaptive Filter, Least Mean Algorithm (LMS), IIR Chebyshev Filter.

**INSPEC Classification :** A9555L, A9630, B5270

---

* The material presented by the author does not necessarily portray the viewpoint of the editors and the management of the Institute of Business & Technology (IBT)

1 Ali Akbar Siddiqui : ali124k@hotmail.com
2 S Hassan Ahmed : shahmed@ssuet.edu.pk
3 Habib Shaukat : habbib_shaukat@hotmail.com
4 M. Shafiq : ssuet.shafiq@hotmail.com
5 Dr. M. Tahir Qadri : mtahirq@hotmail.com

# 1. INTRODUCTION

In this fast developing Technology era where the high performance computing is desired, the advancement in the field of communication can also be witnessed. For Digital Processing System the flexibility, high speed and parallel computation are highly desired which cannot be availed by ASIC due to its limited flexibility and operating capabilities simultaneously. There are widely been used in the applications where high performance along with the complex algorithms are involved. The arenas where FPGA's can be seen but are not limited to the Renewable Energy Systems, Power Converters and Filters etc.

The presence of a transfer function in the auxiliary- path following the adaptive filter and/or in the error-path, as in the case of active noise control, has been shown to generally degrade the performance of the LMS algorithm [2-6]. The method based on Stochastic is present in the above mentioned paper as solution. It is worthwhile to mention here that the LMS algorithm applied in this research for noise Reduction does provide any evidence on the efficiency of algorithm [7-9].

This paper investigates the audio noise reduction in the Digital Systems by the help of an Adaptive Filters. In this paper the behavior of an unknown system was estimated to be a chebyshev low pass filter. The added noise was then eliminated by the help of adaptive Filter's using Least Mean Square algorithm (LMS) [3-5].

# 2. SYSTEM IDENTIFICATION

It is imperative to identify the unknown system as it provides us with information to design a filter to eliminate noise from the system. There are various means of identifying an unknown system. In our proposed system we have implemented LMS (Least Mean square) algorithm for the identification process. Unknown system is often unpredictable and fairly hard to operate, so after identification we know the system frequency response.

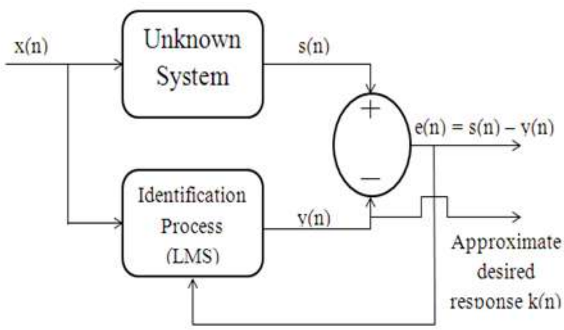


Figure1: Block diagram of adaptive system identification

In Figure 1, s(n) represents an unknown systems output provided with an input signal x(n) to estimate the process usually an Impulse function is used as an input. LMS algorithm block estimated the modeled output by using an error signal e(n) and estimated the response which is approximately equal to the unknown system by minimizing the error. The approximated output is represented by k (n).

$$Q_{n+1}= Q_n+ \mu.e(n).x(n)(1)$$

Equation (1) is an updating equation of LMS, where $\mu$ is a convergence factor and $Q\_n$ are the adjustable coefficient. Error signal is given in equation (2)

$$e(n)=s(n)-y(n) \quad (2)$$

If takes square of equation (2) we gain equation (3) where $E(e^2(n))$ is basically an expected mean squared error, this is the parameter that needs to be reduced constantly so as to get an approximated response of an unknown system k(n) as shown in Figure 1.

$$E(e^2(n))=E(s^2(n))-2QE(s(n)x(n)+Q^2E(x^2(n)) \quad (3)$$

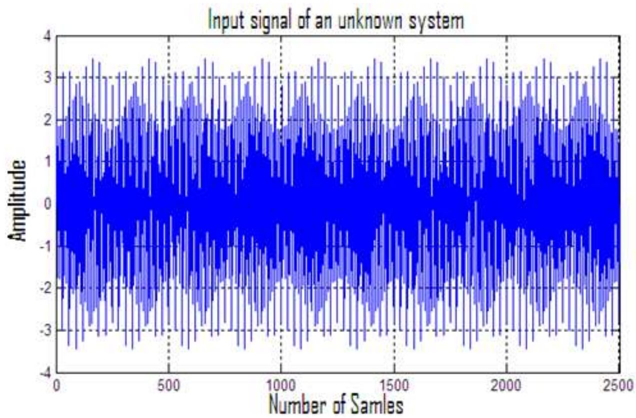


Figure 2: Input Signal of an unknown signal in Time Domain

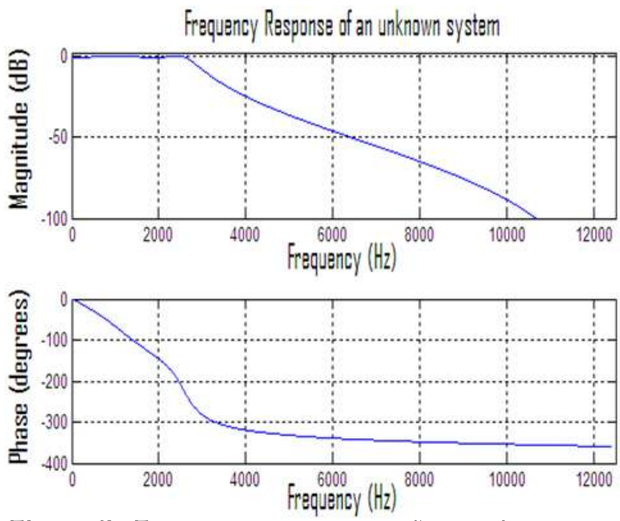


Figure 3: Frequency response of an unknown system

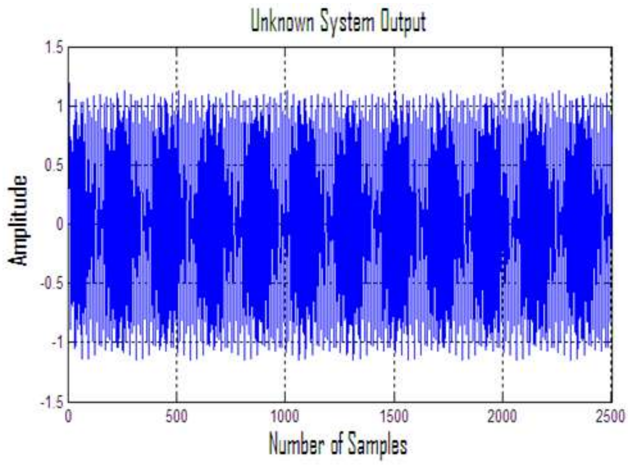


Figure 4: Unknown system Output in Time Domain

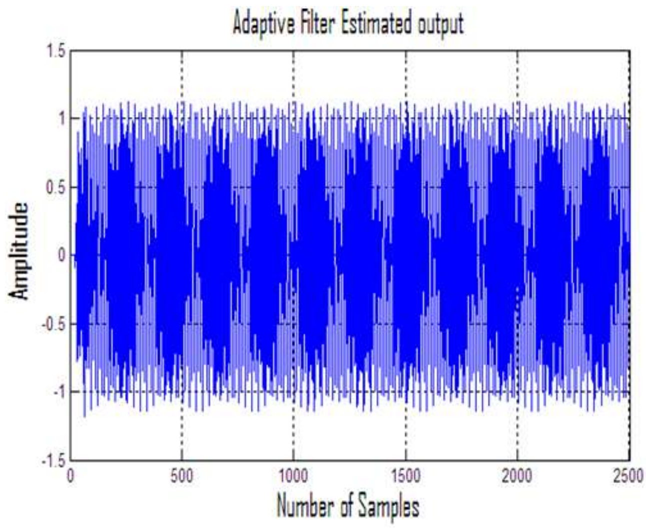


Figure 5: Adaptive Filter Estimated Output in Time Domain

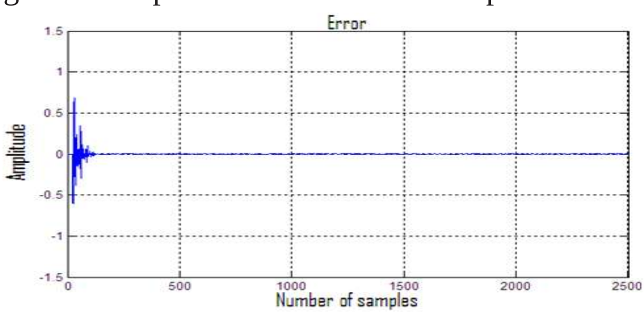


Figure 6: Error Response

Figure 2 represents the input signal to our unknown system which is basically the combinations of different tones. Figure3 represents the frequency response of an unknown system which resembles to that of an IIR Chebyshev low pass filter,

$$G(Z)= G(Z) = \frac{\sum_{k=0}^{N} b_k Z^{-k}}{1+\sum_{l=1}^{N} a_l Z^{-l}} (4)$$

Equation (4) is a general equation for the IIR Filter. Figure 4 is the output signal of our unknown system denoted by s (n) and figure 5 is an Adaptive filter output, we can observe that both the signals are almost identical. These initial samples are basically the learning process or adaptation period in which it tends to minimize the error by learning the approximate predicted weighted value of Q_(n+1) from equation (1). Figure .6 represents an error signal that is minimizing gradually due to the learning process after some samples as shown in Table. 1.

| Number of Samples | Mean Squared Error (MSE) for 50 samples |
|---|---|
| 0-49 | -0.2027 |
| 50-99 | 0.1808 |
| 100-149 | 0.1192 |
| 150-199 | -0.0122 |
| 200-249 | $-1.892e^{-03}$ |
| 250-299 | $1.064e^{-04}$ |
| --- | --- |
| 2450-2499 | $-1.782e^{-04}$ |

Table 1: Mean Squared Error (MSE) of 50 samples

So the cumulative Mean Squared error (MSE) is $-1.3114e^{-04}$ and the Variance is 1.3437. Variance obtained from our estimation is very small; hence we can conclude that we have achieved our desired system response.

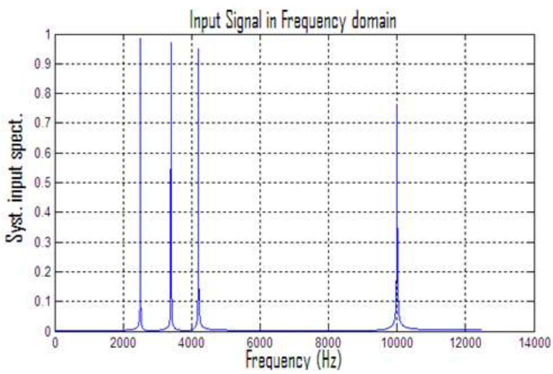


Figure 7: Input Signal in Frequency Domain

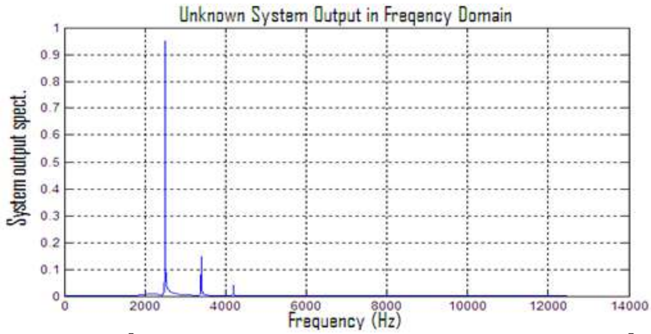


Figure 8: Unknown System's Output in Frequency domain

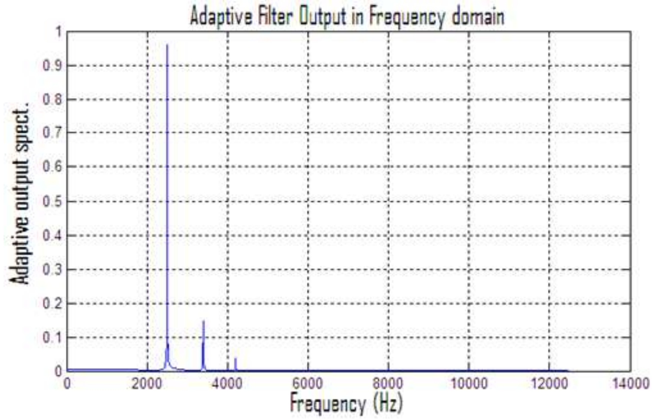


Figure 9: Adaptive Filter output Signal in Frequency Domain

Figure 7, Figure 8, and Figure 9 represents the Input Signal, Unknown system output and Adaptive Filter Output respectively in frequency domain.

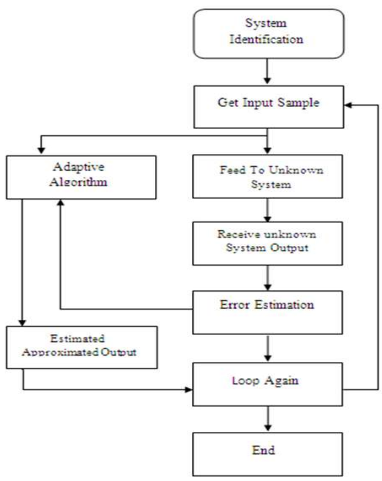


Figure 10: System Identification Software Model

Figure 10 represents the software model of the System identification algorithm. In the first step system is initialized and input signal is generated using a mike or any other audio input device to feed it to the unknown system and in Adaptive Filter simultaneously. An unknown system will generate its output; we know the frequency response of the system's output. Using that output and an adaptive filter output an error signal will be generated, which feedback to the adaptive filter, and we acquired an Estimated Approximate Output for our desired samples.

## 3. NOISE ELEMINATION

After the identification it is possible that our system output is affected by some random noise, so to negate the effect of this noise, we have implemented another Adaptive algorithm for noise elimination.

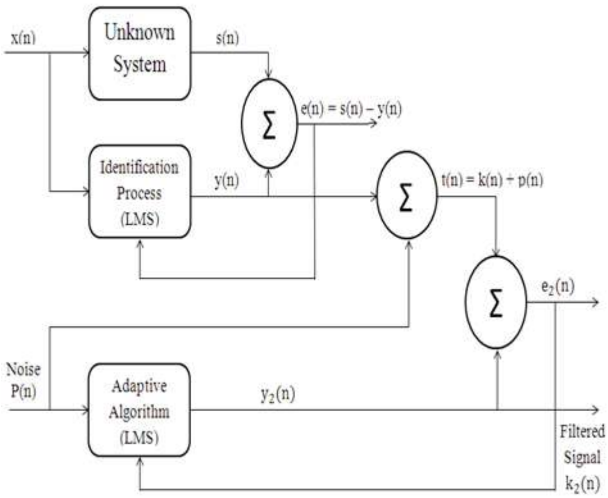


Figure 11: Noise Elimination Model

In Figure (11), y(n) is an estimated response of an unknown system and p(n) is a reference noise affecting output response of an unknown system. Here k_2 (n) is a filtered output response of our desired system which has recently identified using the LMS algorithm.

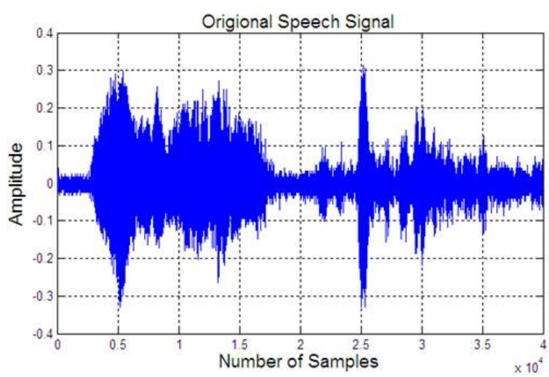


Figure 12: Original Speech Signal in Time Domain

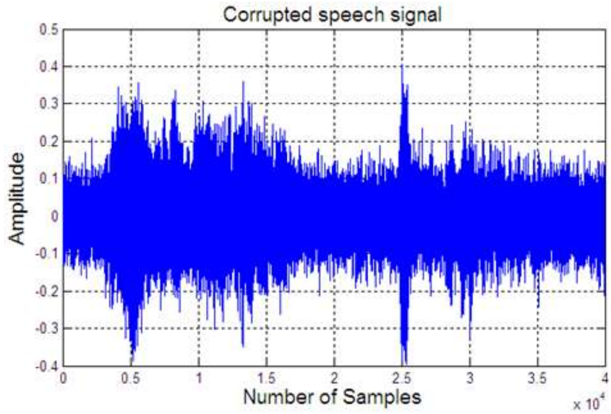


Figure 13: Corrupted Speech Signal in Time Domain

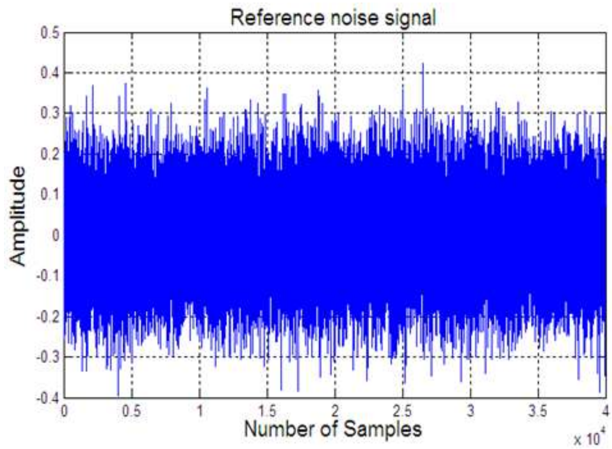


Figure 14: Reference Noise Signal in Time Domain

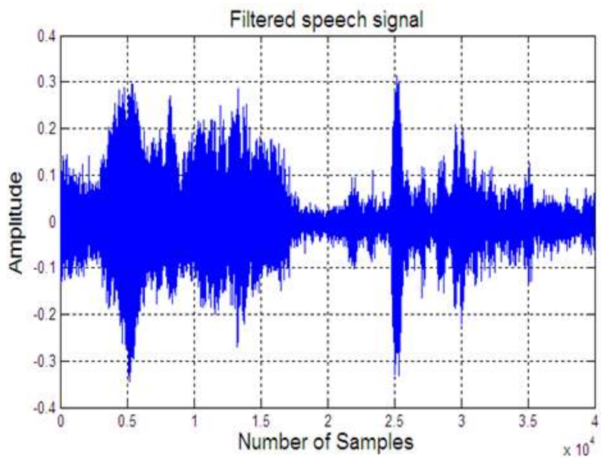


Figure 15: Filtered Speech signal in Time Domain

In Figure 12, we have taken a speech signal, and some random noise is affecting t represented in Figure13and Figure14 is a reference noise. Figure15 is a filtered output response.

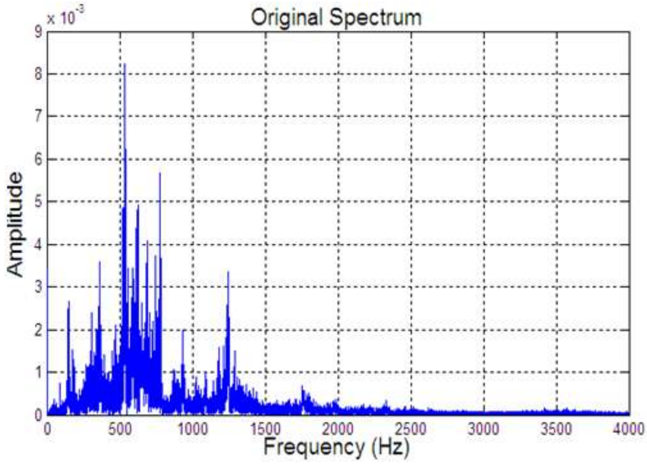


Figure 16: Original Spectrum in frequency domain

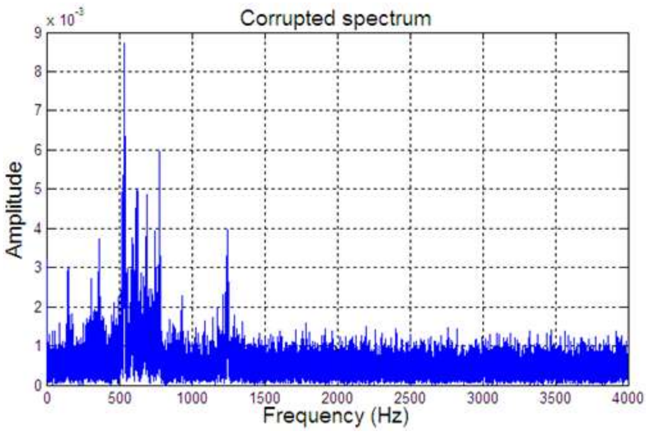


Figure 17: Corrupted Spectrum in Frequency Domain

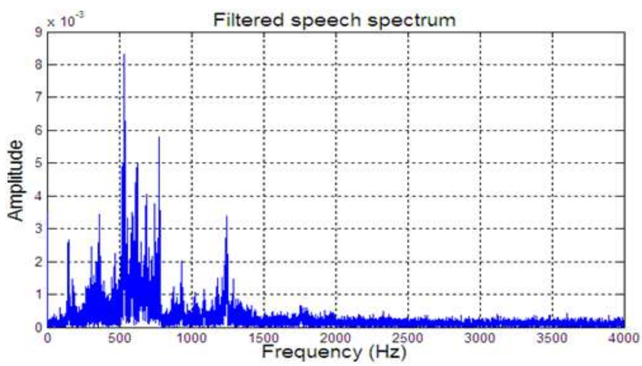


Figure 18: Filtered Spectrum in Frequency Domain

Figures 16, 17 and 18 represent the response of the original signal, corrupted signaland Filtered signal in frequency domain.

## 4. RESULTS & DISCUSSION

In light of results accumulated, we have added some more conclusive results of MATLAB build in sound signals for noise elimination and the results were acceptable as they were for the real time audio signals. Errors are reduced as it searches for its actual signal and tends to find the desired coefficients for the input signal with respect to the system response. Figure.19 and 20 are the representation of the MATLAB sound in time and frequency domain, we can see the input of reference noise and its removal.

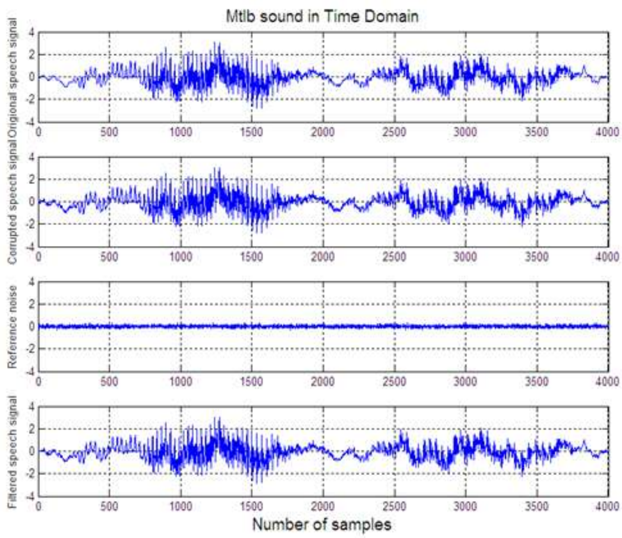


Figure 19: MATLAB sound filtering in Time Domain

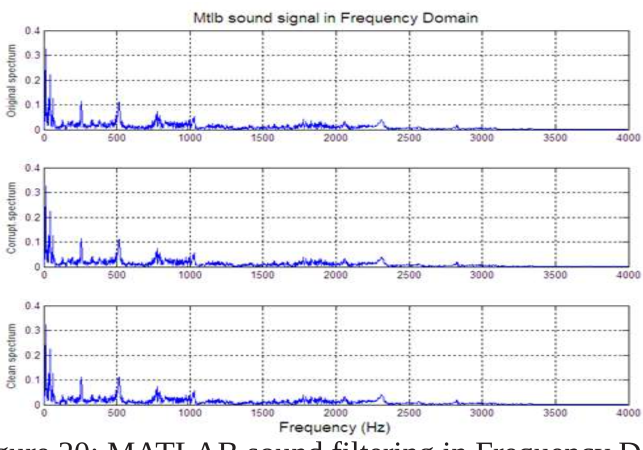


Figure 20: MATLAB sound filtering in Frequency Domain

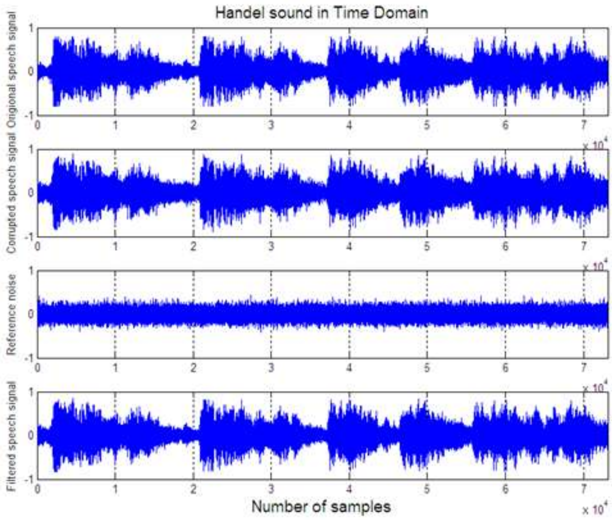


Figure 21: Handle sound filtering in Time Domain

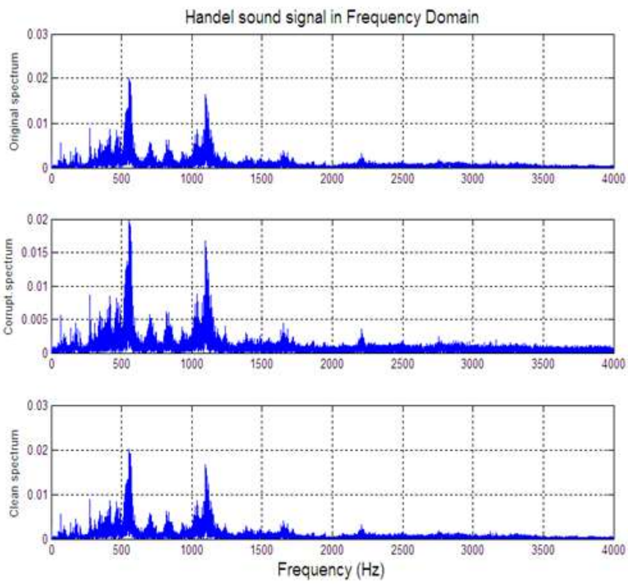


Figure 22: Handle sound filtering in Frequency Domain

## CONCLUSION

The conclusion drawn from the analysis yield the following result, Initially the process of Prediction was performed via Least Mean Square (LMS) algorithm where the system was identified by the help of its input and output response. The prediction showed some acceptable variance from the original signal. However, the overall prediction yields the expected results. The second phase of the research was the removal of Gaussian noise from the Original Signal, which was done by the help of Adaptive Filter. For adaptive filtering, Least Mean Square algorithm was implemented and the results obtained were close enough to the desired results.

## ACKNOLEGEMENT

First of all with a profound gratitude, we are thankful to Almighty Allah forgiving us success, knowledge and understanding without which we would not been capable of completing this research paper.

We are also profoundly grateful to all our family members whose endurance and understanding have played a significant role in our success by sacrificing the important family time and supporting us all over the research work.

We are finally thankful to the editor, reviewers and IBT specially who provided us with the opportunity to publish our research paper in this esteemed journal.

## REFERENCE

[1] Mrs. A B Diggikar, Mrs. SS Ardhapurkar, Design and Implementation of Adaptive Filtering Algorithm For Noise Cancellation in Speech Signal on FPGA , International Conference on Computing, Electronics and Electrical Technology ,(2012),PP.22-25.

[2] Elias Bjamason, Analysis of Filtered X-LMS Algorithm, IEEE Transaction on Speech and Audio Processing, Volume No 3, Issue No 6, ,(2012),pp.504- 514.

[3] Guoz-hu Long, Fuyun Ling, Proakis, J.G, The LMS algorithm with delayed coefficient adaptation, IEEE Transaction on Speech Signal and Processing, Volume 37, Issue 09,(2002), pp. 1397-1405.

[4] C. Mosquera, l.A. Gomez "Adaptive Filters for Active Noise Control", Sixth international congress on sound and vibration Copenhagen, Denmark,(2008),pp.23-28.

[5] U. Meyer-Baese, G. Botella, D. E. T. Romero and Martin Kumm, “Optimization of high speed pipelining in FPGA-based FIR filter design using genetic algorithm,” Proc. of SPIE, Vol. 8401, (2012),pp.54-56.

[6]Elliott, Stephen J., Ian M. Stothers, and Philip Nelson. "A multiple error LMS algorithm and its application to the active control of sound and vibration."Acoustics, Speech and Signal Processing, IEEE Transactions,(2013), pp.1423-1434.

[7] Kwong, Raymond H., and Edward W. Johnston. "A variable step size LMS algorithm." Signal Processing, IEEE Transactions on 40.7 ,(1992),pp. 1633-1642.

[8] Tarrab, Moshe, and ArieFeuer. "Convergence and performance analysis of the normalized LMS algorithm with uncorrelated Gaussian data." Information Theory, IEEE Transactions on 34.4 (1988): 680-691.
[9] Macchi, Odile M., and Neil J. Bershad. "Adaptive recovery of a chirped sinusoid in noise. I-Performance of the RLS algorithm. II-Performance of the LMS algorithm." IEEE Transactions on Signal Processing , (1991),pp. 583-602.
[10]Widrow, Bernard, et al. "Stationary and nonstationary learning characteristics of the LMS adaptive filter." Proceedings of the IEEE, (1976),pp.1151-1162.
[11] Simon Haykin. "Adaptive Filters Theory" Pearson Education, ACM Transaction on Speech Signal and Processing, Volume 24, Issue 8, (2008), pp. 129-135.