

LMS, DLMS, AND TVLMS ADAPTIVE FILTER EVALUATION

¹LIZARANI PARIDA,

Gandhi Institute of Excellent Technocrats, Bhubaneswar, India

²SONALI SUBHADARSINI BEHERA,

Black Diamond College of Engineering & Technology, Jharsuguda, Odisha, India

ABSTRACT

The aim of this paper is to implement the adaptive digital Least Mean Square (LMS) and delayed-LMS (DLMS) for typical noise cancellation applications. Noise reduction from given sound can be achieved by extraction using LMS algorithms with MATLAB. We are comparing these algorithms on the basis of sound wave provided with working MATLAB. Sound can transfer in high rate but noise added in that signal it becomes a noisy signal. The noisy sound we could not recognize the original sound. This technique can be used to reduce noise level from noisy signal without reducing the characteristic of the signal. The practical work using MATLAB it prove that LMS algorithm better than DLMS and TVLMS on the basic of result showing in the form of wave in training and also in evaluation section. It shows the result in PSNR format which obtained from comparing the original sound and denoisy sound. So that LMS obtained highest PSNR value as compared to the DLMS and TV-LMS.

Key words: LMS, DLMS, TVLMS, MATLAB, adaptive filter

Cite this Article: Pradnya D. Mendhe and Nilesh P. Bodne, Performance Evaluation of LMS, DLMS and TVLMS Adaptive Filter, International Journal of Electronics and Communication Engineering and Technology, 7(5), 2016, pp. 68–76.
<http://www.iaeme.com/IJECET/issues.asp?JType=IJECET&VType=7&IType=5>

1. INTRODUCTION

Objective of this project is to determine the performance of different algorithm of adaptive digital filters. The determinations of coefficients involve noise cancellation application also comparing the LMS, DLMS and TV-LMS algorithm by using MATLAB. And determine PSNR value but PSNR value obtained from comparison between the original sound and denoisy sound. This technique can be used to reduce noise level from noisy signal without reducing the characteristic of the signal. The least mean square (LMS) adaptive filter is the most popular and widely used adaptive filter, because of its simplicity and its satisfactory convergence performance. The direct-form LMS adaptive filter involves a long critical path due to its inner-product computation to obtain the output from filter such that the critical path is required to be reduced by pipelined implementation when it exceeds to desired sample period.

Least mean squares (LMS) algorithms are type of adaptive filter used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean squares of the error signal (difference between the desired and the actual signal). LMS algorithms adjust the filter coefficient to minimize the cost function. The LMS algorithms do not involve any matrix operations. Therefore LMS algorithms required fewer computational resources and memory. The implementation of the LMS algorithms is also less complicated.

The conventional LMS algorithm due its recursive behavior, it is modified to a form called as DLMS algorithm. To implement the DLMS algorithm, during each sampling period of training phase, one has to compute a filter output and an error value which equals to the difference between the current filter output and the desired response. The estimated error is then used to update the filter weights in every training cycle. A lot of work has been done to implement the DLMS algorithm in systolic architectures to increase the frequency. They involve an adaptation delay for filter length; this is quite high for large order filters.

The TVLMS algorithms is totally depends on time varying convergence parameter. The basic idea of TVLMS algorithm is to utilize the fact that the LMS algorithm need a large convergence parameter value to speed up the convergence of the filter coefficient to their optimal values, the convergence parameter

should be small for better accuracy. In other words, the convergence parameter is set to a large value in the initial state in order to speed up the algorithm convergence. As time passes, the parameter will be adjusted to a smaller value so that the adaptive filter will have a smaller mean-squared error.

In this, we are comparing the behavior of LMS, DLMS and TV-LMS adaptive algorithms by using MATLAB. They mostly used the removing unwanted noisy from the signal for that they are using some algorithms. In existing work various authors perform different methodology to show their work but as this paper the working and the result shows that the LMS algorithm performs the best result than any other algorithm for comparing on the basis of noise reduction. Various techniques are used to reduce the noise from the given sound signal. LMS algorithm has a faster rate of convergence. Also it shows the result in a short period of time without any interruption. It is cheaper than any other. It has a capacity to perform any task in given time but limitation is important. But during MATLAB work LMS shows the desired result for various systems.

In existing work various authors perform different methodology to show their work but as this paper the working and the result shows that the LMS algorithm performs the best result than any other algorithm for comparing on the basis of noise reduction. Various techniques are used to reduce the noise from the given sound signal.

2. COMPARISON OF EXISTING AND PROPOSED METHODOLOGY

Such work LMS algorithm has a faster rate of convergence. Also it shows the result in a short period of time without any interruption. It is cheaper than any other. It has a capacity to perform any task in given time but limitation is important. But during MATLAB work LMS shows the desired result for various systems.

Existing Methodology	Proposed Methodology
Other algorithms which have a pipelined architecture are faster than LMS algorithm while it uses more chip area due to using extra registers.	When LMS algorithms do not support pipelined architecture but it is faster than any other.
The filter order is based on a trade-off between the MSE performance and algorithm execution time.	The filter order is based on a trade-off between complexity and the convergence speed.
Other algorithms do not replace the random weight.	LMS algorithm replaces the random weight by instant weight till it reaches to optimized weight

3. IMPLEMENTATION OF PROPOSED METHODOLOGY

Noise is estimated and applied to LMS, DLMS and TVLMS. Removing the noise from the signals by using the LMS, DLMS and TVLMS algorithm. These algorithms are related to each other for forming the new sound of the signal. The original sound wave (x) is added to the noise (N) forming the noisy sound (y).

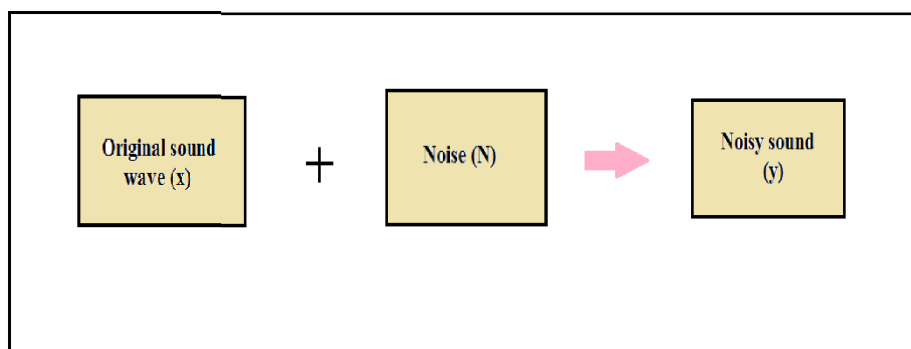


Figure 1 Noisy sound

4. SIMULATION RESULTS

In this section the simulation results of realization adaptive digital FIR filters by using the MATLAB, performance of LMS, DLMS and TVLMS algorithms are compared. Also they are updated the weights of the filters and removing the unwanted noise from the original signal. The results are shown in wave format with MMSE and PSNR for proper evaluation.

1. We used sound signal from airport location, in bubble, in car, in exhibition, in restaurant and from station in wave format that are our input for the MATLAB .first we use the airport sound for evaluation on LMS algorithm and then DLMS and TVLMS resp. The sound is hearable for us for training section. After it noise is added by using software that make it noisy. But when we move toward evaluation section the MATLAB with given software removes the unwanted sound that is noise signal from it. The results are shown in wave format with the MMSE and PSNR for both training and evaluation. The resulted sound is clear to hear .For any circumstances and places the sound can be clear by using this software.

For the LMS Algorithm

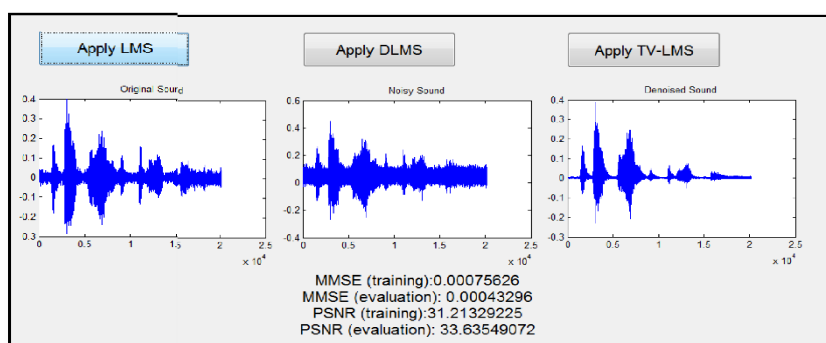


Figure (a) Output of LMS algorithm

Sound is heard by us and the clarity of the input sound by mixing airport noise can be easily notified. After removing the unwanted noise from the input sound the result showing in the wave format as given above. The sound without any noise can be heard.

For the DLMS Algorithm

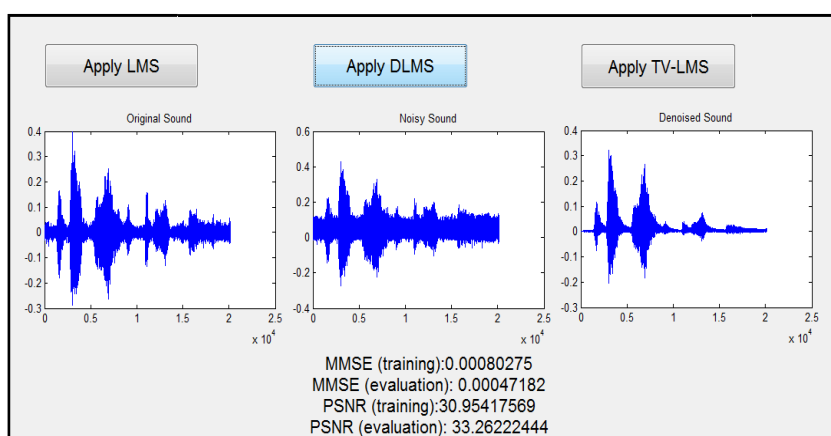


Figure (b) Output of DLMS algorithms

Here also same action takes place, which sound we provide as input in MATLAB it mixes the noise in it and also heard by us but the clarity of sound is weak as compared to LMS algorithm. It is shown as given in above figure in waveform.

For the TVLMSAlgorithm

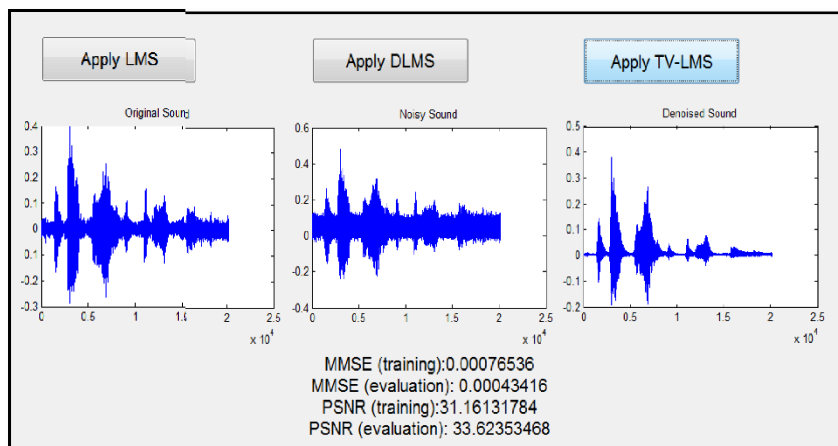


Figure (c) Output of TVLMS algorithm

The input sound given to MATLAB removes the noise but this sound also weak in clarity as compared to LMS algorithm. It is shown as given in above figure in wave format.

Table 1 Performance of comparison between LMS , DLMS & TVLMS adaptive filters for the airport sound wave.

Algorithm	MMSE	PSNR
LMS (Training)	0.00075	31.21
LMS(evaluation)	0.000432	33.63
DLMS(Training)	0.0008	30.95
DLMS (evaluation)	0.0004	33.26
TVLMS (Training)	0.0007	31.16
TVLMS (evaluation)	0.0004	33.62

From the above table it shows that PSNR value of LMS algorithm is highest and then TVLMS algorithm. For this purpose the importance of using LMS algorithm is faster than other.

2. Similarly, we used the bubble sound for the evaluation of LMS, DLMS and TVLMS algorithm. They are removed the unwanted noise from sound wave but in the evaluation Part by using the software noisy is added in the original sound.

For the LMSAlgorithm

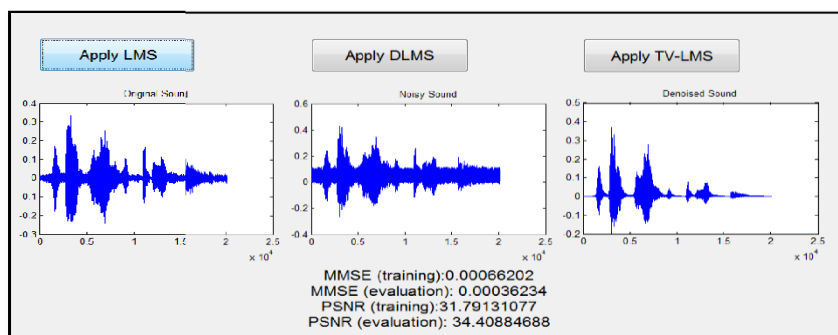


Figure (a) Output of LMS algorithm

In LMS algorithm original sound can be heard thus the original wave form shown but we added the

noise in the original sound wave .That the noise is removed from sound wave gives the denoised sound.

For the DLMS Algorithm

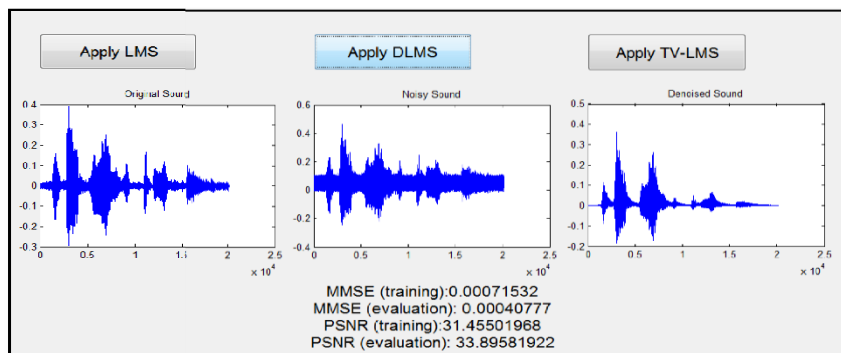


Figure (b) Output of DLMS algorithm

Same procedure should be applied here .Original sound can be heard we mixed noise for that the signal and removed the unwanted noise from that sound.

For the TVLMS Algorithm

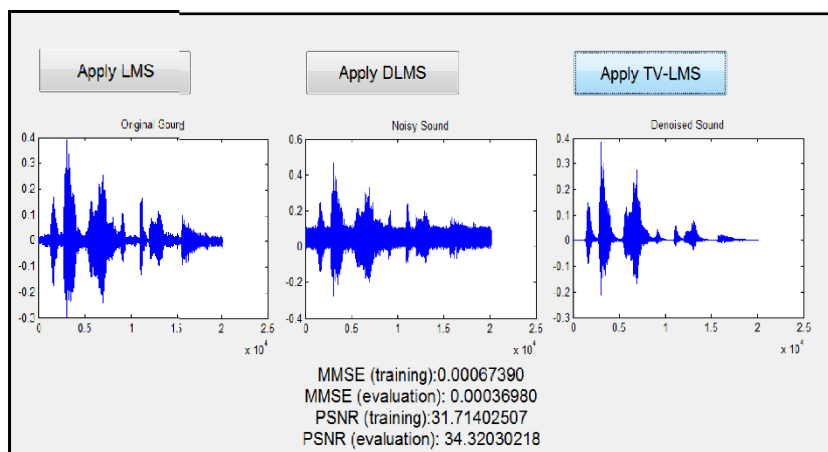


Figure (c) Output of TVLMS algorithm

The input sound given to MATLAB removes the noise but this sound also weak in clarity as compared to LMS algorithm. It is shown as given in above figure in wave format.

Table 2 Performance of comparison between LMS, DLMS & TVLMS adaptive filters for the bubble sound wave.

Algorithm	MMSE	PSNR
LMS (Training)	0.0006	31.79
LMS(evaluation)	0.0003	34.40
DLMS(Training)	0.0007	31.45
DLMS (evaluation)	0.0004	33.89
TVLMS (Training)	0.0006	31.71

TVLMS (evaluation)	0.0003	34.32
--------------------	--------	-------

From the above table it shows that PSNR value of LMS algorithm is highest and then TVLMS algorithm. For this purpose the importance of using LMS algorithm is faster than other.

3. Similarly, we used the Car sound for the evaluation of LMS, DLMS and TVLMS algorithm. They are removed the unwanted noise from sound wave but in the evaluation Part by using the software noisy is added in the original sound.

For the LMS Algorithm

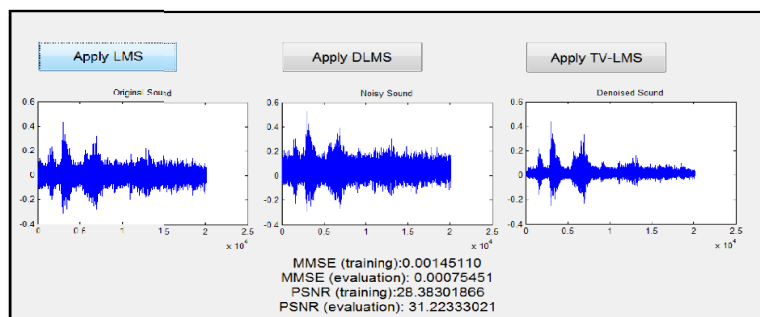


Figure (a) Output of LMS algorithm

In LMS algorithm original sound can be heard thus the original wave form shown but we added the noise in the original sound wave. That the noise is removed from sound wave gives the denoised sound.

For the DLMS Algorithm

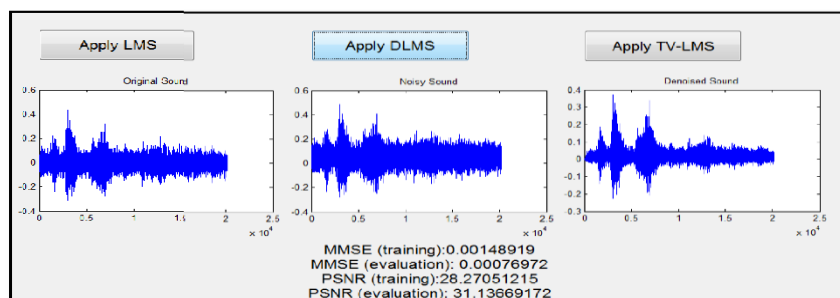


Figure (b) Output of DLMS algorithm

Same procedure should be applied here. Original sound can be heard we mixed noise for that the signal and removed the unwanted noise from that sound.

For the TVLMS Algorithm

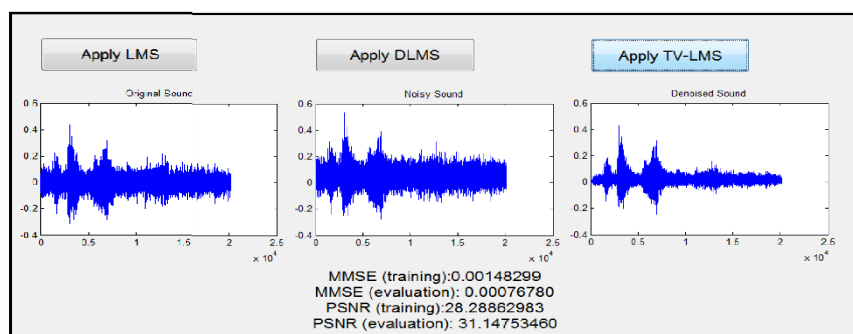


Figure (c) Output of TVLMS algorithm

The input sound given to MATLAB removes the noise but this sound also weak in clarity as compared to LMS algorithm. It is shown as given in above figure in wave format.

Table 3 Performance of comparison between LMS, DLMS & TVLMS adaptive filters for the Car sound wave.

Algorithm	MMSE	PSNR
LMS (Training)	0.0014	28.38
LMS(evaluation)	0.0007	31.22
DLMS(Training)	0.0014	28.27
DLMS (evaluation)	0.0007	31.13
TVLMS (Training)	0.0014	28.28
TVLMS (evaluation)	0.0007	31.14

From the above table it shows that PSNR value of LMS algorithm is highest and then TVLMS algorithm. For this purpose the importance of using LMS algorithm is faster than other.

Above three tables it shows that PSNR value of LMS algorithm is higher than other algorithms .So that the algorithm is totally depends on the PSNR value .Therefore LMS algorithm is faster than DLMS and TVLMS algorithm.

5. CONCLUSION

The primary objective of the thesis was to clear noise from desired sound. Also to improve the sound clarity due to mixing of noise .For this purpose the working of well known filters i.e. LMS DLMS and TVLMS are compared .With using MATLAB a best quality sound is generated by removing the unwanted sound and it gets output of denoisy sound .For this purpose the MATLAB is use .It use a basic software quality a Low Mean Square (LMS) that could be used for wireless applications. LMS was designed such that, it can be capable of removing the unwanted noise. The hardware implementation of adaptive FIR filters is a challenging issue in digital signal processing andcommunications.

REFERENCE

- [1] R.Karma “Performance Comparison of Adaptive Algorithms for Noise Cancellation” IJESRT October 2014.
- [2] T.J.Milna, S.K.Mythili “Survey on DLMS adaptive filters with low delay” IJRCCT ,vol-3 9 september-2014
- [3] C.S.Yadav “ Performance of Wiener Filter and Adaptive Filter for Noise Cancellation in Real-Time Environment” IJCA ,Vol.97-No.15, July2014
- [4] S.Patel,S.R.Panchal“StudyofvariousmethodstoimproveconvergencespeedofLMSalgorithm” IJSHRE vol-2 issues 5 may 2014
- [5] V.Anand, “Performance analysis of various noise cancellation methods” IJAREEIE vol-2 issues may 2013.
- [6] R. P. Deshmukh “Review on implementation of FIR adaptive filter using distributed arithmetic and block LMS algorithm” international journal in2013.
- [7] K. Belpatre , M.R Bachute “Comparative performance study between the time varying LMS (TVLMS) algorithm, LMS algorithm and RLS algorithm”NCIPET-2012
- [8] H.Ariyadoost, Y. S.Kavian, K.Ansari “Performance evaluation of LMS & DLMS digital adaptive FIR filters by realization on FPGA” IJSET vol-1 No. 1 September,2011.
- [9] LilatulFerdouse, NasrinAkhter, TamannaHaqueNipa and FarihaTasminJaigirdar, “Simulation and Performance Analysis of Adaptive Filtering Algorithms in Noise Cancellation”.IJCSI International Journal of Computer Science Issues, Vol. 8, Issue 1, January2011.

- [10] Sayed.A.Hadei and M.lotfizad, "A Family of Adaptive filter Algorithms in Noise Cancellation For Speech Enhancement", International Journal of Computer and Engineering, vol.2, No.2, April 2010, 1793-8163.
- [11] Haykin, Simon, 2003, "Least-Mean-Square Adaptive Filters", ISBN 0-471- 21570-8 Wiley, pp.315.
- [12] L.D.Van and W.S.feng, "An efficient architecture for the DLMS adaptive filters and its application" IEEE trans. Circuits syst. Vol.48, No. 4, pp 359-366 April2001.
- [13] S. Koike, "Analysis of Adaptive Filters Using Normalized Signed Regressor LMS Algorithm", IEEE Transactions on Signal Processing, Vol. 47, No.1, pp.2710-2723,1999.
- [14] B.Widrow, M.Lehr and Michel Bilello, "adaptive signal processing", IEEE International Conference on Neural Networks, IEEE Service Center, New Jersey (1993), pp.1–8
- [15] W. B. Mikhael, F.H.Wu, G. Kang, and L. Fransen, "Optimum adaptive algorithms with applications to noise cancellation," IEEE Trans Circuits Syst., vol. CAS-31, pp. 312-315, Mar.1984.
- [16] R. Marvin, "Adaptive noise canceling for speech signals," IEEE Trans. Acoustics, Speech and Signal Processing, vol. 26, pp.419- 423, October1978.
- [17] Widrow B, et al. "Stationary and non stationary learning characteristics of the LMS adaptive filter." Proc.IEEE,64:1151-1162,1976.
- [18] G. Long, F.Ling and J.G.Proakis, "The LMS algorithm with delayed co-efficient adaptation algorithm".IEEE, Trans.Acoustic,speech signal processing, vol.37, no.9,pp.1397-1405.
- [19] P. K. Meher, S. Y. Park "Low adaptation delay LMS adaptive filters. part-I introducing a novel multiplication cell"IEEE.
- [20] S.Haykin, "Adaptive filter theory, prentice Hall," upper saddle River, NJ,2001
- [21] M.H.Hayes, "Statistical digital signal processing & modeling," John wiley& sons,1996.
- [22] N.Shanbhag and K.parhi "Pipelined adaptive digital filters," Kluwer,1994.
- [23] SantoshPandi P S L Gangadharaiah, FPGA Based Fixed Point LMS Adaptive Filters, *International Journal of Electronics and Communication Engineering and Technology (IJECEt)*, 6(10), 2015, pp. 30–42.
- [24] G.Prasannakumar and K.Indirapriyadarsini, Low Complexity Algorithm for Updating the Coefficients of Adaptive Filter, *International Journal of Electronics and Communication Engineering and Technology (IJECEt)*, 4(3), 2015, pp.63–69.