ISSN: 2277-3754
ISO 9001:2008 Certified
International Journal of Engineering and Innovative Technology (IJEIT)
Volume 7, Issue 3, September 2017

Evaluation and Comparison the Performance of LMS and NLMS Algorithm in Acoustic Echo Cancellation using the8 sub-bands Techniques

Mahmod. A. Al Zubaidy, Sura. Z. Thanoon
Ninevah University, Mosul, Iraq School of Electronics Engineering, Mosul, Iraq

Abstract— ***The Acoustic echo is the most common problem in Acoustic and communication systems such as telephone system. This phenomena occurs when the signal reflected from the environments and loudspeaker coupled with the microphone. Hence an adaptive filter used to defeat this problem. In this paper a sub-band technique with LMS and NLMS algorithm adaptive filter used to cancel the echo from the received signal and the result of the two algorithms are compared.***

Index Terms— **Sub-band adaptive filter, echo canceller, LMS, NLMS.**

I. INTRODUCTION

The Acoustic Echo Cancellation (AEC) system treats the Echo problem where the adaptive filter represents the important part of AEC. Because the speech signal varying with time need a filter adjust his parameter continuously to get the required performance, an adaptive filter is a dynamic and non-linear system and self-adjusting where with these properties achieve the purpose of AEC[1][2].

An adaptive filter weights are adjusting according to algorithms like LMS, NLMS and RLS, each one of these algorithm try to realize the optimal desired output by minimizing the difference between the desired signal d(n) and the practical output of the filter y(n), the difference is known as error signal e(n).The echo signal is an image of the original signal but with attenuation and delay. Let x(t) is an speech signal then the echo of this signal express by the following equation:

$x_d(t) = \alpha x(t - t$……………………………………..….. (1)

Where α is an attenuation factor and t_d is time delay [3]. The interface between the echo of the far-end signal and the near-end signal as shown in figure (1) usually distribute the user.

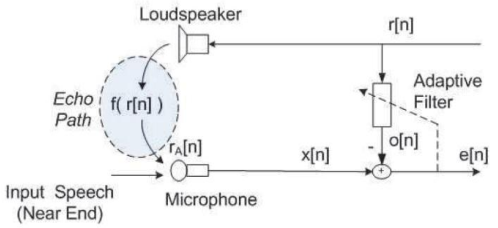


Figure (1): Acoustic echo cancellation

II. SUB-BAND ADAPTIVE FILTER

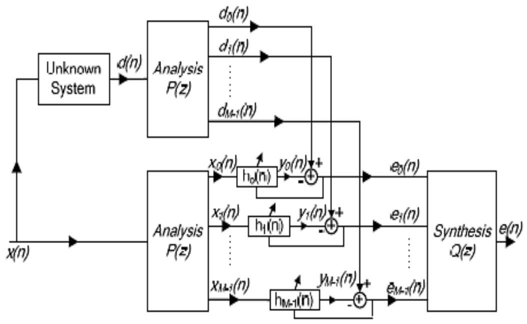


Fig (2):sub-band system identification

The idea of sub-band is to split the signal in to N bands by using sub-band analyzer and deal with each band separately then the N bands combined together by sub-band synthesizer. This property of sub-band technique acted on reduces the range of frequency where each adaptive filter deals with, so the adaptive filter converges faster to the desired performance. Figure (2) present sub-band system identification [4].

III. LEAST MEAN SQUARE (LMS) ALGORITHM

The first and most popular algorithm used in many applications, the reason for his fame back to ease of implementation and lack of complexity in calculations.

The LMS adaptive filter is described by the equations [5]:

W (n+1) = W (n) + μ (n) e (n) X (n) …… ……………….. (2)

e (n) = d(n) − WT (n)X(n) …………………….…….. (3)

Where

W (n) =[w0(n) w1(n) ….. wL−1(n)] ^{T}is the coefficient vector, *X(n)* = [*x(n) x*(*n* −1) *x*(*n*−*L*+1)]^{T}is the input signal vector, *d(n)* is the desired signal, *e(n)* is the error signal, and *μ(n)* is the step size.

IV. NORMALIZED LEAST MEAN SQUARE (NLMS) ALGORITHM

Normalized Least Mean Square (NLMS) generally used in acoustic echo cancellation system because the signal power

ISSN: 2277-3754
ISO 9001:2008 Certified
International Journal of Engineering and Innovative Technology (IJEIT)
Volume 7, Issue 3, September 2017

change continually with time so it need a variable step size to converge to the desired signal by cancelling the echo, this feature mean it has good convergence speed, also NLMS like LMS it easy to implement [6].

The formula to compute the normalized step size:

$$\mu = \frac{\beta}{c+\|x(n)\|^2} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots \quad (4)$$

Where

μ (n)= step size, β = Normalized step size (0 <β< 2),c= small positive constant

The following equation to clarify how the filter calculates his weights [7]:

$$w(n + 1) = w(n) + \mu (n)e(n)x(n) \quad \ldots\ldots\ldots \quad \ldots\ldots\ldots\ldots\ldots \quad (5)$$

V. PERFORMANCE EVALUATION

To evaluate the AEC system, the echo return loss enhancement (ERLE) parameter is used, where ERLE is ratio of the expected value of microphone signal to the expected value of the residual echo or error signal. ERLE clarify the amount of echo that the adaptive filter canceled and it's value depend on the algorithm that was chosen to the adaptive filter, however it's quantity is high it mean that an adaptive filter cancel the major or whole echo [8]. ERLE calculated as in the equation:

$$\mathrm{ERLE} = 10 \log \frac{\mathrm{Pd}}{\mathrm{Pe}} \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots \quad (6)$$

VI. SIMULATION DIAGRAMS AND RESULTS

A MATLAB_SIMULINK was used to simulate the Acoustic Echo Cancellation System. The AEC models using LMS algorithm and NLMS algorithm in adaptive filter depend on the following parameters of the filter: - the length of the filter=1024, step size= 0.002 for both models to show the difference in the performance. Figure (3) show the MATLAB_SIMULINK design for AEC system by using LMS and NLMS algorithm applying 8 sub-bands technique.

In order to test this simulation systems, speech signal was recorded with sampling rate 8000 as shown in figure (4), while the echo signal was generated from this signal by make attenuation (0.5) and time delay (100 msec), then the echo signal and the original signal has been added to get the sound with echo signal which presented in figure (5).

The sound with echo signal represented the input port of the adaptive filter while the estimated echo signal represents the desired port of the adaptive filter.

The output which take from the error port represent the real echo signal, by subtracting the error signal from the sound with echo signal we can get the sound without echo, the performance of the system differ according to the algorithm that was chosen to the adaptive filter which was shown in the output figures (6) and figure (7) .

The error signal and ERLE clarify more the difference in the performance between LMS and NLMS algorithm; however the error signal in NLMS system is smaller than in the LMS system that means the performance is better in NLMS system, also ERLE for NLMS is better than LMS. Figure (8), (9) and (10) show the different in the error signal between the two systems.

VII. CONCLUSION

The system of AEC in this paper was realized using 8 sub-bands adaptive filter method and applied LMS algorithm and NLMS algorithm to evaluate the optimum performance between them. The ERLE factor determines the performance of AEC system and when ERLE is higher this mean a good AEC system. The ERLE value for LMS was 11.4 dB and for NLMS was 13.78 dB, From these values NLMS has the better performance than LMS, so this paper improve that 8 sub-bands NLMS is better than 8 sub-bands LMS and is a good choice algorithm for AEC system.

REFERENCES

[1] P. S. R. Diniz, E. A. B. da Silva, and S. L. Netto, "Digital Signal Processing: System Analysis and Design", Cambridge University Press, Cambridge, UK, 2002.

[2] T. Bose, John Wiley& Sons, "Digital Signal and Image Processing", NewYork, NY, 2004.

[3] Rafid Ahmed Khalil, "Adaptive Filter Application in Echo Cancellation System and Implementation using FPGA", College of Engineering, University of Mosul.Alrafidain, Engineering Jornal Vol.16, No.5 Dec. 2008.

[4] Irina Dornean, Marina Topa, BotondSandorKirei and Marius Neag, "Sub-band Adaptive Filtering for Acoustic Echo Cancellation", IEEE Xplore, 810- 813, 2009.

[5] Widrow, B. and Hoff, M.E., "Adaptive switching circuits", IRE WESCON Conv. Rec.,4, 96–104, Aug. 1960.

[6] Monsoon H. Hayes, "Statistical Digital Signal Processing And Modeling", John Wiley & Sons, Inc., 1992.

[7] J. Dhiman, S. Ahmad and K. Gulia, "Comparison between adaptive filter algorithms (lms, nlms and rls)", International Journal of Science, Engineering and Technology Research, vol. 2, no. 5, pp. pp-1100, 2013.

[8] J. B. Allen and D. A. Berkley, "Image method for efficiently simulating small-room acoustics", The Journal of the Acoustical Society of America, vol. 65, no. 4, pp. 943 - 950, 1979.

AUTHOR BIOGRAPHY

Name: Dr. Mahmod Ahmed Mahmod AL_Zubaidy

Birth date: 1967

Professional: lecturer in Communication Eng. Department, Electronics College, Ninevah University.

Degrees (Qualifications): Phd in communication Engineering From university of Mosul

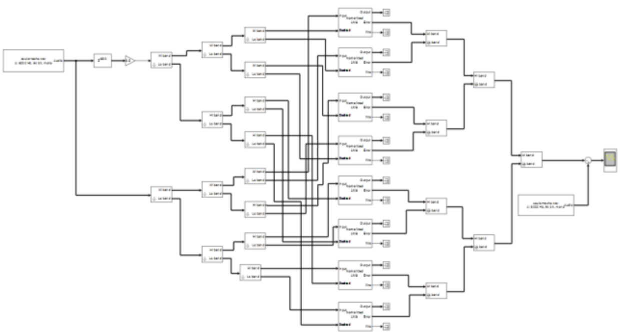
Phd Date: 2007

ISSN: 2277-3754
ISO 9001:2008 Certified
International Journal of Engineering and Innovative Technology (IJEIT)
Volume 7, Issue 3, September 2017

Interests: Digital signal processing, acoustic, Radar systems, Mobile systems, digital communications.

Name: Sura Zeno Thanoon
Birth date: 7/7/1990
Professional: Master student
Degrees (Qualifications): BSC in Communication Engineering
Graduate Date: 2012
Interests: Digital signal processing, acoustic

APPENDIX



Fig(3): Simulink model for 8 sub-band LMS/ NLMS adaptive filter

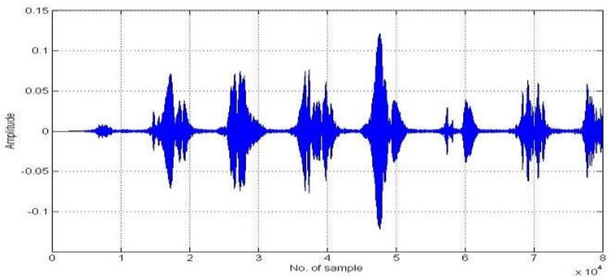


Fig (4): The speech signal

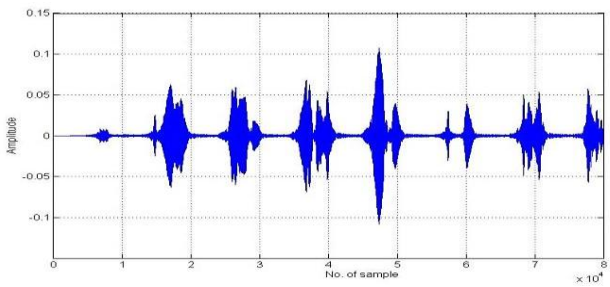


Fig (5): The echo signal

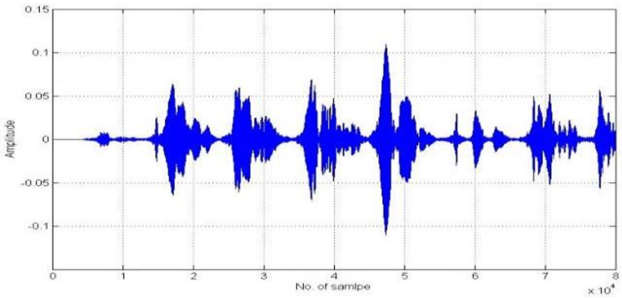


Fig (6): The output signal of LMS 8 sub-band adaptive filter

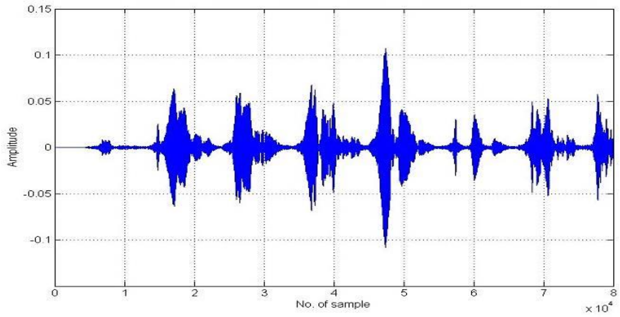
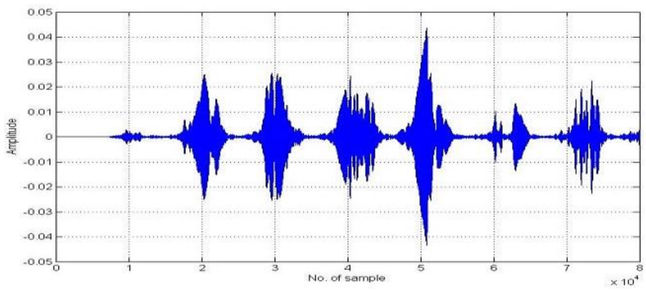


Fig (7): The output signal of NLMS 8 sub-band adaptive filter

Fig (8): The error signal for LMS 8 sub-bands adaptive filter



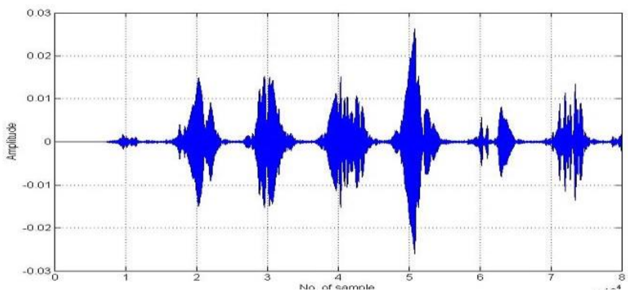


Fig (9): The error signal for NLMS 8 sub-bands adaptive filter

ISSN: 2277-3754
ISO 9001:2008 Certified
International Journal of Engineering and Innovative Technology (IJEIT)
Volume 7, Issue 3, September 2017

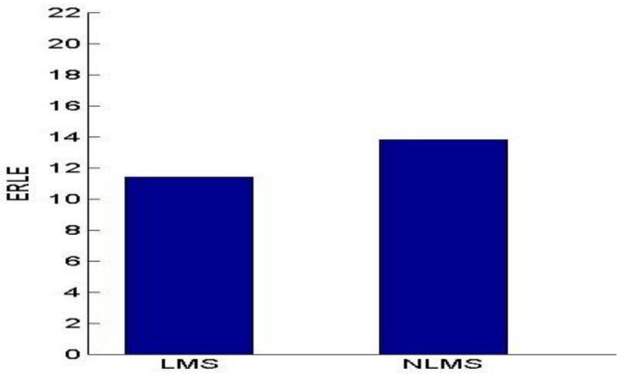


Fig (10): The ERLE for LMS and NLMS 8 sub-bands