INTERNATIONAL JOURNAL OF TECHNOLOGY ENHANCEMENTS AND EMERGING ENGINEERING RESEARCH, VOL 3, ISSUE 03 24
ISSN 2347-4289

De-Noising Of Speech Signal Using Adaptive Filter Algorithms

Ramesh Yadav, Mr. Deepak Sharma

M.E. Scholar, ETC Branch, Chhatrapati Shivaji Institute of Technology, Durg, India
Associate Professor, Chhatrapati Shivaji Institute of Technology, Durg,India
ramesh.yaad@gmail.com, deepaksharma@csitdurg.in

Abstract: Interference or Noise is the major issue in any communication system. It has been used LMS and NLMS adaptive filters to de-noise the noisy signal and compare with the performance of these filters. Adaptive algorithms are the most effective and elementary. This paper investigates the noise cancellation capabilities of the adaptive filters.

Keywords: Adaptive algorithm, LMS filter, NLMS Filter, RLS filter, Linear filter, SNR, MSE

I. INTRODUCTION

Adaptive algorithms are the most popular algorithms to use denoising the speech signal, echo cancellation, equalization, beam-forming, line enhances, interference suppression, etc. Basically there are two types of family (i) Least mean square (ii) Recursive least square[1],[3]. The Least Mean Square (LMS) filter is built around transversal filter that is based on tapped delay line. It is the simplest design among all adaptive filter algorithms and its performance is efficiently good. It is frequently used for denosing and interference suppression. The Normalized Least Square (NLMS) filter also the family of least mean square family. It is generally used for echo cancellation. These filters have basic drawbacks like slow in conversion, tacking is poor and comparatively high mean square error. On the other hand recursive family overcomes the limitation of LMS filters, and provides highly efficient and fast adaptation rate so it provides high conversion rate i.e. The Recursive least square (RLS) filter.[3] The price paid for this improvement computation complexity will increase.

II. ADAPTIVE ALGORITHMS

An adaptive algorithm is based on adaptation process, In this process all the taps weight value update automatically according to the output error. This error adjusts by the tap weight value in various adaptive algorithms. This algorithm supports the stochastic process [2],[3],[4]. Fig. 1 shows the basic block diagram of the adaptive filter algorithm.

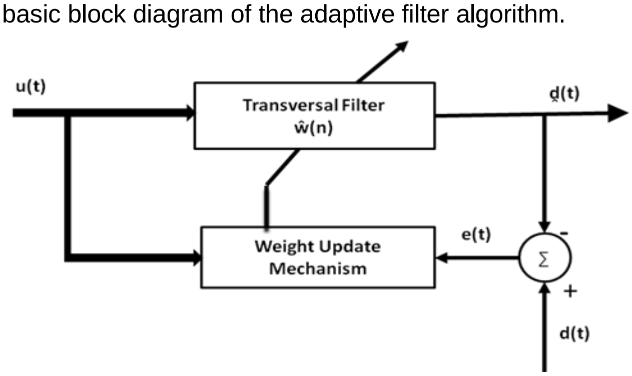

Fig. 1 *Adaptive Filter Algorithm*

Linear filter is used in adaptive filter with adaptive algorithm. The linear filter provides the linear output after the tap weight population. Finite Impulse Response (FIR) or Infinite Impulse Response (IIR) filter generally used for designing linear filters. Based on the adaptive algorithm, adaptive filter will be decided. If we decide a filter on its tap weight updates on the present status basis of the present and prior status based, i.e. w (n) & w (n+1), so it's an LMS filter. If we take it tap weight updates which is based on difference of two consecutive weight values, so it's an NLMS filter. If we taken present and future tap weight value to update present weight value, so it's an RLS filter[2][3].

A. LMS Adaptive Filter

LMS filter is the most popular and simple filter among all the adaptive process. The Least Mean Square (LMS) filter is a linear adaptive algorithm. It consists two main processes a) filtering process – It shows the estimated error by comparing the output signal and desired signal b) adaptation process – according to the estimate error, it automatically adjusts of the tap weights of the filter [2],[6],[7]. There are three basic relations in LMS filter

- Filter output $y(n) = \hat{w}5(n)\, u(n)$ (1)
- Estimated Error $e(n) = d(n)-y(n)$ (2)
- Tap weight adaption $\hat{w}(n+1)=\hat{w}(n)+\mu u(n)e^*(n)$ (3)

Here u(n) is the tap input signal, y(n) is the output signal, d(n) is the desired signal for interference and e(n) is the error signal. $\hat{W}(n)$ is the weight vector of the filter.

μ is step size parameter
$0< \mu< 2/M\ Smax$

Where Smax is the maximum power value of the power spectral density of the tap input u(n) and the M is the filter length.

B. NLMS Adaptive Filter

It is a Normalized Least Mean Square algorithm. This is used to normalize the high input power of input vector u (t). When a high power signal comes in input vector, then LMS filter suffers gradient noise amplification problems. To overcome this problem, adjustment in tap weight vector of the filter at iteration (n+1). Step size of the filter is under the control of the designer. It supports the real value's error e (n) as well as complex conjugate error *e(n)[2],[6],[7]. The mathematical relation of the filter is given bellow.

$e(n) = d(n) - \hat{w}\square(n)\, u(n)$ (4)

$\hat{w}(n+1) = \hat{w}(n) + \tilde{u}/(||u(n)||2)\, u(n)\, {}^*e(n)$ (5)

ISSN 2347-4289

$\hat{w}(n)$ is a old weight vector of the filter with n iteration and $\hat{w}(n+1)$ is the updated weight vector of (n+1) iteration. $\tilde{u}$ is the adaptation constant. The conversion rate is much faster than conventional LMS filter.

III. NOISE CANCELLING FOR SPEECH SIGNAL

Fig. 2 Illustrates the basic principle of noise cancellation system for adaptive filter, Where u (t) taken as an input tap vector i.e. a reference input the system. And d (t) taken as a primary input or desired input for the noise cancellation system [5], [7].

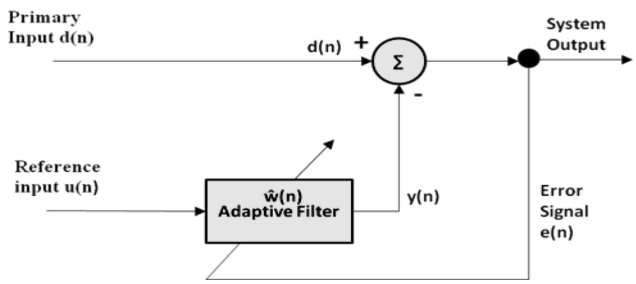

Fig. 2 *Adaptive Noise cancellation System*

It has been considered that the input signal is a voice signal, i.e. stationary signal. Random signal taken as a noise signal to adding in the voice signal uncorrelated. So reference signal d(t) is a noisy signal, and noise signal taken as a reference signal for adaptive filter, where w(n) is a tap weight value or filter coefficient value[4], [6].

IV. SIMULATION AND RESULTS

Simulation has been done in MATLAB software. Where input taken as a low power speech signal and random noise is added, And noisy signal used in LMS algorithm for simulation. Input and output parameters show in table 1.

Table 1. List of Input and Output Parameters

S. NO	Parameter
1.	Filter Length – Input
2.	Step Size – Input
3.	MSE(Mean Square Error) - Output
4.	SNR(Signal to Noise Ratio) - Output

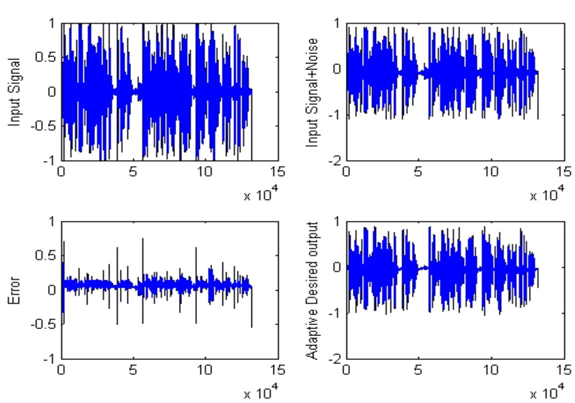

Fig. 3 *Input Signal, Noisy signal, Error Signal and Output signal waveform for Filter Length 12*

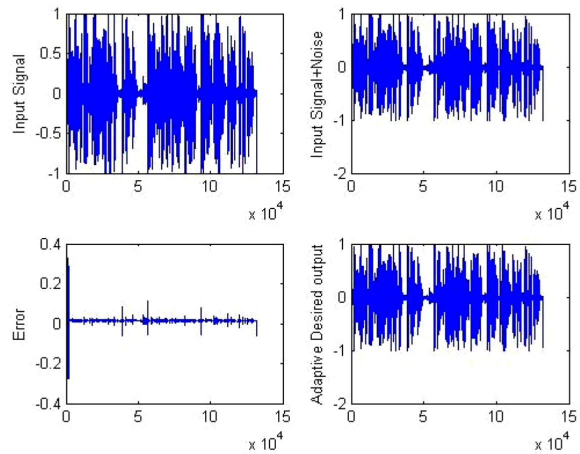

Fig. 4 *Input Signal, Noisy signal, Error Signal and Output signal waveform for Filter Length 18*

Table 2. LMS Filter Length Vs Step Size

LMS Filter			
Filter Length			12
S.No	Step Size	SNR(dB)	MSE
1	0.02	4.1187	0.0083
2	0.03	13.2807	0.0023
3	0.04	6.8627	0.0064
4	0.05	4.8667	0.0066
5	0.06	5.1092	0.0063
6	0.07	3.0503	0.0075
7	0.08	4.6345	0.0062
8	0.09	0.7236	0.000114
9	0.1	0.335	0.0066
10	0.5	1.0472	0.0034

Table 3. LMS Filter Length Vs Step Size

LMS Filter			
Filter Length			18
S.No	Step Size	SNR(dB)	MSE
1	0.02	11.5855	0.0033
2	0.03	17.0119	0.0011
3	0.04	8.0638	0.0055
4	0.05	3.7977	0.007
5	0.06	7.5832	0.0056
6	0.07	1.1617	0.0069
7	0.08	5.6327	0.0064
8	0.09	5.7798	0.0056
9	0.1	0.7568	0.0069
10	0.5	3.6756	0.000079721

ISSN 2347-4289

These tables show the SNR and MSE values, varying according to the variation of step size, where filter length is fixed. Table 2 is showing for filter length 12, where SNR value is highest at Step size is .02, and MSE value is least. Table 3 also shows the similar results for filter length is 18. Where SNR is highest and MSE value is lowest at a particular step size. Figure Number 5 shows the plot of SNR Value and the step size with fixed filter length for adaptive LMS algorithm.

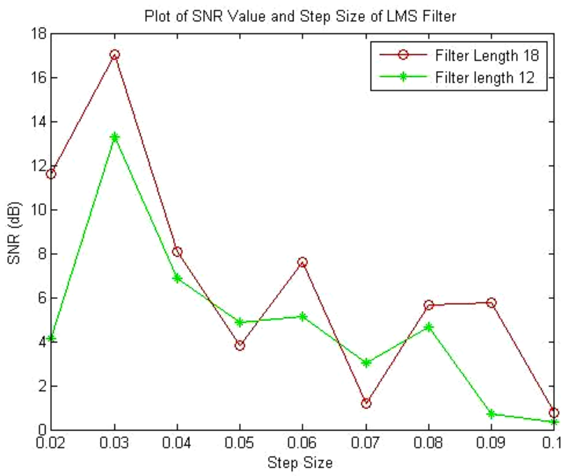

Fig. 5 *SNR (dB) Vs Step Size of LMS Filter*

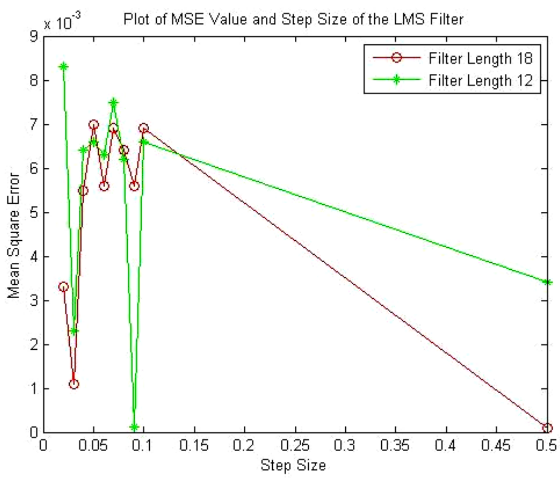

Fig. 6 *MSE Vs Step Size of LMS Filter*

It has been shown low power speech signal as an input signal. Random signal taken as a noise signal with different variance. All the simulation results shown bellow.

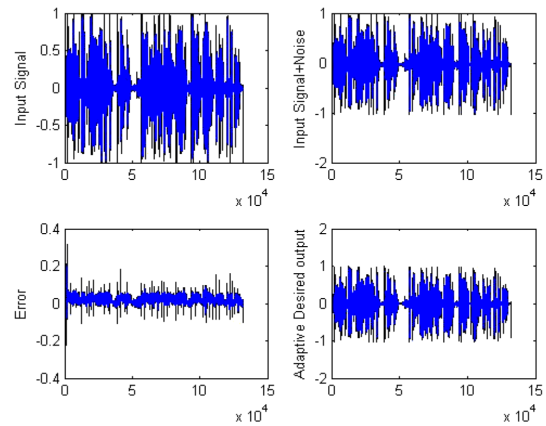

Fig. 7 *Input Signal, Noisy signal, Error Signal and Output signal waveform for Filter Length 12*

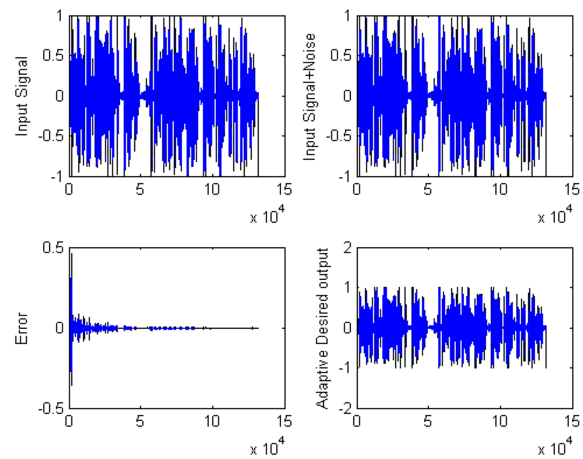

Fig. 8 *Input Signal, Noisy signal, Error Signal and Output signal waveform for Filter Length 18*

Table 4. LMS Filter Length Vs Step Size

NLMS Filter			
Filter Length			12
S.No	Step Size	SNR(dB)	MSE
1	0.02	2.009	.0000407
2	0.03	2.3625	0.0013
3	0.04	6.0838	.00018
4	0.05	6.5158	0.002
5	0.07	8.1484	0.0019
6	0.08	15.8126	.000643
7	0.09	6.8385	.000501
8	0.1	5.1239	.00003.14
9	0.2	5.6477	.000966
10	0.3	18.7002	.00044
11	0.5	9.1607	0.0000083808

ISSN 2347-4289

Table 5. LMS Filter Length Vs Step Size

NLMS Filter			
Filter Length			18
S.No	Step Size	SNR(dB)	MSE
1	0.02	3.4301	0.000104
2	0.03	4.1348	0.00065
3	0.04	4.6144	0.00004
4	0.05	4.8802	0.00010
5	0.07	5.1803	0.00005
6	0.08	10.8299	0.00130
7	0.09	20.1189	0.00019
8	0.1	26.933	0.00001
9	0.2	8.6214	0.00002
10	0.3	20.0299	0.00031
11	0.5	10.8846	0.00001

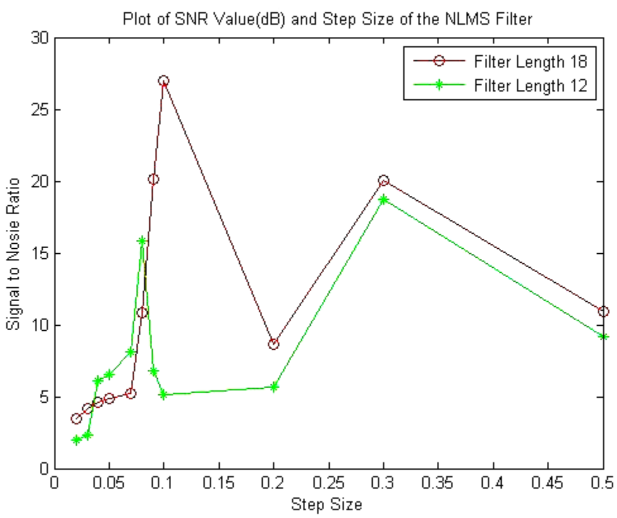

Fig. 9 *SNR (dB) Vs Step Size of NLMS Filter*

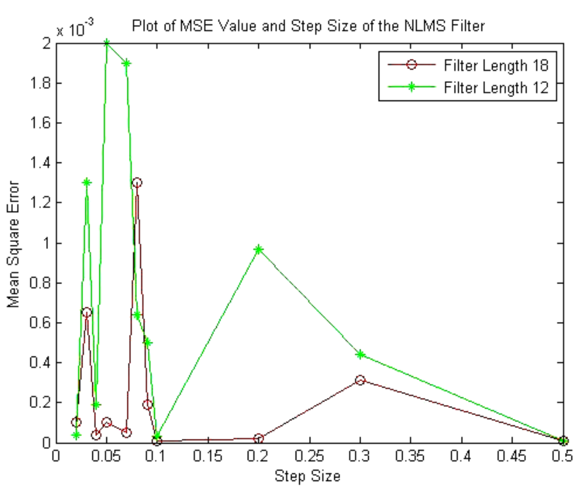

Fig. 10 *MSE Vs Step Size of NLMS Filter*

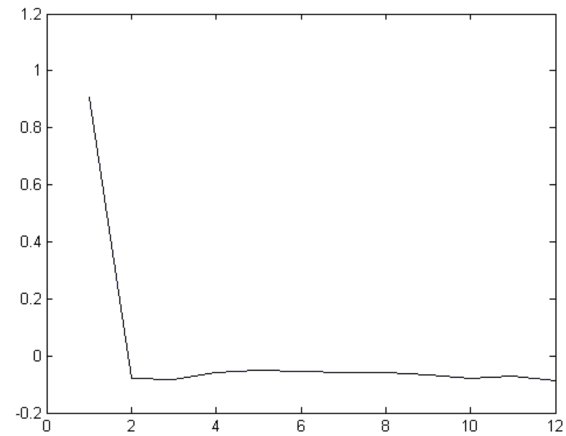

Fig 11 *Weight adaption of the filter length 12*

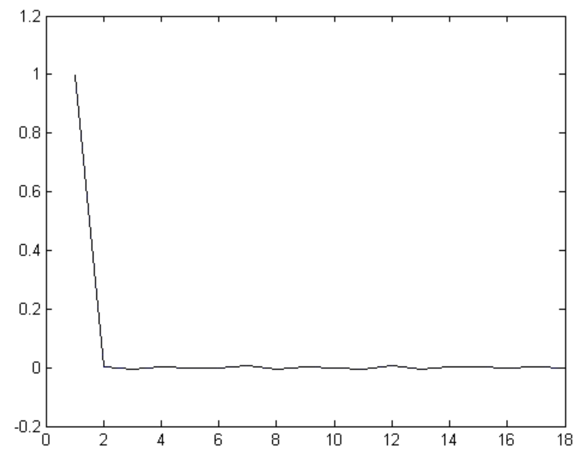

Fig. 12 *Weight adaption of the filter length 18*

V. CONCLUSION

This simulation result shows the performance of two adaptive filters, LMS and NLMS adaptive filters. Two output parameters of the filters are Signal to Noise ratio (SNR) in dB and Mean Square Error (MSE) value. According to this value NLMS adaptive filter shows better result in both parameters. In LMS adaptive filter, better noise cancellation shows in lower step size in higher filter order, whereas NLMS adaptive filter shows better noise cancellation shows in higher step size in higher filter order i.e. filter length.

Overall based on the performance of these two filters, It is concluded that the NLMS adaptive filter has better noise cancellation capability and fever adaption is required in higher order of filter (Filter Length). Higher filter order always shows better noise cancellation. Increases the filter length, increases the noise cancellation capability of the filter. Conversion rate is fast in NLMS filter. LMS adaptive filter is the simplest and reliable filter.

REFERENCES

[1] Sayed. A. Hadei, Student Member IEEE and M. lotfizad, "International Journal of Computer and Electrical Engineering, Vol. 2, No. 2, April 2010, pp. 1793-8163.

ISSN 2347-4289

[2] Windrow, S. Steam, Adaptive Signal Processing. Englewood Cliffs, NJ: Prentice – Hall, 1985.

[3] S. Haykin. Adaptive Filter Theory, 4th Edition Pearson Education, 2008.

[4] Paulo.S.R.Diniz, "Adaptive Filtering Algorithms and Practical Implementations", Kluwer Academic Publishers ,London. 2nd Edition.

[5] Bernard Widrow, et. al., "Adaptive Noise Cancellating Princilples and Application", Proceding of the IEEE, 1975 Vol. 63 No. 12 pp. 1692-1716.

[6] Radhika Chinaboina, D.S.Ramkiran, Habibulla Khan, " Adaptive Algorithms For Acoustic Echo Cancellation In Speech Processing", IJRRAS 7 (1) ● April 2011, pp. 38-42.

[7] Raj Kumar Tuteja, S.K. Agrawal, "Simulation and Performance Analysis of Adaptive filter in Noise Cancellation", International Journel of Engineering and Technology Vol. 2(9),2010,pp. 4373-4378.

[8] Marvin R. Sambur**,** Member, IEEE, "Adaptive Noise Canceling for Speech Signals", IEEE TRANSACTIONS ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING, VOL. ASSP-26, NO. **5,** OCTOBER 1978, pp. 419-423

[9] J. M. Gorriez et.al., " A Novel LMS Algorithm Applied to Adaptive Noise Cancellation", IEEE SIGNAL PROCESSING LETTERS, VOL. 16, NO. 1, JANUARY 2009,pp. 34-37.

[10] Soria, E; Calpe, j; et.al. "A novel approach to introducing adaptive filters based on the LMS algorithm and its variants", IEEE Transactions, vol. 47, pp. 127-133, Feb 2008.

[11] Homana,I; Topa, M.D.;et.al., "Echo cancelling using adaptive algorithms", Design and Technology of Electronics Packages, (SIITME) 15th International Symposium., pp. 317-321, Sept.2009.

[12] Ying He,et.al. "The Applications and Simulation of Adaptive Filter in Noise Canceling", 2008 International Conference on Computer Science and Software Engineering, 2008, Vol.4, Page(s): 1 – 4.