International Journal of Engineering Research & Technology (IJERT)
ISSN: 2278-0181
Vol. 8 Issue 08, August-2019

Interference Cancellation using Different Algorithm of Adaptive Filter

Vaishali
Department of Electronics and Communication,
Guru Jambheshwar University of Science and Technology,
Hisar, India

Dr. Sanjeev Kumar Dhull
Department of Electronics and Communication,
Guru Jambheshwar University of Science and Technology,
Hisar, India

Abstract— **This paper compares different type of adaptive algorithms such as Least mean square (LMS), Normalized least mean square (NLMS), Sign-Sign LMS, sign-error LMS. Here adaptive behavior of the algorithm is examined for the noise or interference cancellation in speech signal. Comparison of LMS variants is based on the SNR calculated. In speech signal or acoustic signal the noise can be reduced by the adaptive filtering, when mixed with the noisy signal at same frequency. So we provide implementation of variants of LMS to compare and get the better result. Various noise signals provide the different impact over the signal during any type of transmission of signal or communicating between different sources like telephonically. As in that case noise gets mixed with the signal and information get lost. To regain the actual information it is necessary to remove the unwanted signal mixed with the original signal. So to remove that adaptive filtering is best way.**

Keywords: Least mean square (LMS), Normalized least mean square (NLMS), Sign-error least mean square (SELMS), Sign-sign least mean square (SSLMS).Mean square error (MSE). Power Spectral Density (PSD).

I.INTRODUCTION

In digital signal processing Adaptive signal is widely used in speech processing [1]. Adaptive filter is also called as self adjusting filter, as it can modify its transfer function depending upon the proposed algorithm [3]. There are numbers of distinct kind of algorithms used in Adaptive filtering that allow the filter coefficient to adjust the signal statics as mentioned in Fig 1. LMS algorithm was developed by Windrow and Hoff [2].
RLS stand for recursive least square and FRLS is fast recursive least square. These are discovered by Gauss.
LMS is the mostly used algorithm in Adaptive filtering because of its simple calculation and preferential performance [2]. In this paper there are different variants of LMS used to remove the noise from desired information [2]. All four variants of LMS are used and there comparison is shown on the basis of step size and SNR [3]. LMS algorithm get required filter coefficient to get the least mean square of the comparison between the required signal and the original signal to remove noise signal. During the communication by telephone or any speech communication

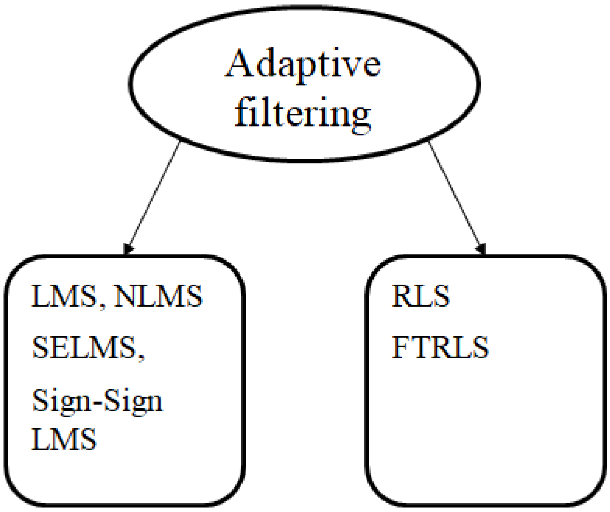


Fig.1 Type of Adaptive algorithms.

different undesired signal got mixed like transport, crowds or it can be electronically (thermal noise) also [12]. There is different type of noise that has different effect on signals according to their quantity.

II.NOISE

There are different type of noise which gives different effects on signal when mix with them during transmission of signal [12]. In general, there are three type of noise-

(a) Impulsive noise
(b) White noise
(c) Colored noise

Impulsive noise: Impulsive unwanted signal introduce to abnormal bursts of undesired signal with comparatively high magnitude. It is generally considered as corrupted Gaussian noise.

White noise: It is a signal having all detectable repetition with same strength. This type of noise range is completely uncertain at each frequency. It is having a value in between 0 to 2π and its esteem at a specific frequency is unconnected to the range at another frequency.

Colored noise: A noise which is having any color is considered as colored noise. White noise is uniform all over the spectrum of noise, where as Colored noise has frequency within a predetermined limited area. There are various categories of the colored noise example blue, pink, brown, orange noise determined by the scale of PSD of noise signal.
The relation between the power of speech signal and the noise signal is called the Signal-to-Noise ratio defined in db

International Journal of Engineering Research & Technology (IJERT)
ISSN: 2278-0181
Vol. 8 Issue 08, August-2019

(decibels). Ideally the ratio should be greater than 0db states that the original signal is powerful instead of noise. This relationship is shown in this paper at a different value of step-size of the adaptive algorithms use to remove the noise signal. The step-size is use to compare the performance of the different variants of LMS to remove the noise. The step-size increment or decrement with the changes in Mean-Square error, so that the filter can detect the variations in the system and to generate the minimum steady-state error.MSE is defined as the difference between the desired signal and the actual signal. It is average squared comparison between the actual signal and the required signal.

III.ADAPTIVE NOISE CANCELLATION

A filter is ideal if it is familiar regarding the input data. If this is unknown, the adaptive filters will be used. The adaptive filters having the characteristics of self-optimization [8]. Whenever we don't know about the characteristics of the input signal at the initial point then we use adaptive filter. The main purpose of noise cancellation is to exceed the snr. The adaptive filter have vast applications in inverse modeling, speech enhancement, speech encoding, acoustic echo cancellation, linear prediction, channel equalization, line enhancing, system identification, jammer suppression, frequency tracking, noise cancellation and image encoding [9-10] etc. Adaptive filtering consists of two different parts-first one is digital filter which can regulate the coefficient and secondly there are adaptive algorithms which can modify the same. Two inputs are given to the filter, one is the original signal x(k) and second is the corrupted signal d(k)+n(k) together.

e(k)= error signal, i.e. e(k)= d(k) - y(k)
x(k) = Input signal,
y(k) = Output signal
d(k)= Desired signal

An adaptive filter is a non-linear filter and time-variant parameters. It can adjust its parameter according to the needs. The block diagram adaptive noise canceller is shown in Fig 2. The adaptive filter finds an approximate of the noise and that is subtracted from the result of the adaptive filter. Adaptive filter get the feedback from its algorithms used and calculate the difference between the error signal produced by subtracting the

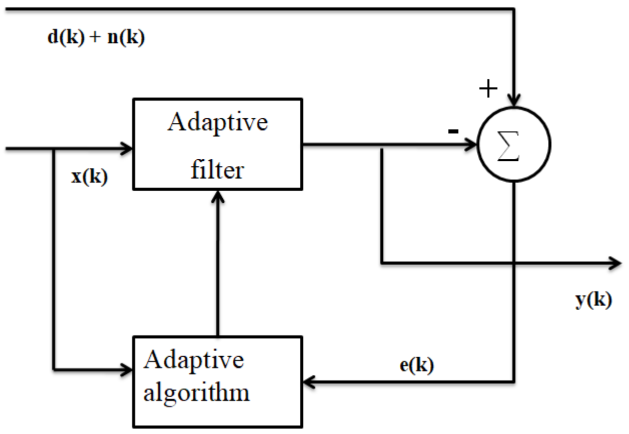


Fig 2. Schematic of adaptive noise cancellation.

y(k) from the desired signal and provide the required output. The adaptive canceller can reduce the mean-square error (MSE) of the complete output, so that output will be the ideal evaluation of the desired signal in the minimum MSE sense.

IV.ADAPTIVE ALGORITHMS

A. LMS Algorithm

In LMS, the coefficients of adaptive filter are adjusted from sample to sample to reduce MSE. The aim of operating under varying condition and regulate itself to reduce the error is fulfilled by the LMS algorithm [3].

$$w(k+1) = w(k) + \mu\left(-\frac{\partial e^2(k)}{\partial w(k)}\right) \quad (1)$$

e(k) = error signal (input signal-adaptive output signal).

$$e(k) = d(k) - w^T(k)x(m) \quad (2)$$

Where d(k)= desired signal
x(k)= original signal.

B. Normalized LMS Algorithm

It is an improved form of standard LMS. Both the equation of LMS and NLMS are same except the NLMS has a time-varying μ. The speed of filter can be improved by this.

$$w(k+1) = w(k) + \frac{\mu}{a + \sum_{k=0}^{p-1} x^2(k-m)} x(k)e(k) \quad (3)$$

$\sum_{k=0}^{p-1} x^2(k-m)$ is the input signal energy
Where **P** is filter length.
μ control the step size.
a is the minor fixed value added to ignore the denominator of the new term becoming zero during input is zero.

The main disadvantage of the standard LMS is, it can be easily affected by the scaling of the input signal.

C. Sign-Error Algorithm

In standard LMS and NLMS algorithms, we can get the coefficient of adaptive filter by calculating the MSE between the desired and the output signal and applying the result to the current filter coefficient [4].

Using sign-error algorithm, it restore the MSE by the sign of error to change the filter coefficient.

When the error is positive:

New coefficient is equal to the pervious one multiplied by the step size μ.

Whereas, in negative error, the new coefficient is equal to the previous one minus the error multiply by the μ [4].

$$w(k+1) = w(k) + x(k)\, \mu\, \mathrm{sgn}[e(k)] \quad (4)$$

Where sgn(.) = sign function

$$\mathrm{sgn}[e(k)] = \frac{d}{dx}|x|, x \neq 0$$

D. Sign-Sign Algorithm

Similarly in the sign–sign LMS, the MSE is replaced by the sign of the input to change the filter coefficient.

Depending upon the sign of the error, the value of new coefficient remains the same as in the case of sign-error.

$$w(k+1) = w(k) + \mathrm{sgn}[x(k)]\, \mu\, \mathrm{sgn}[e(k)] \quad (5)$$

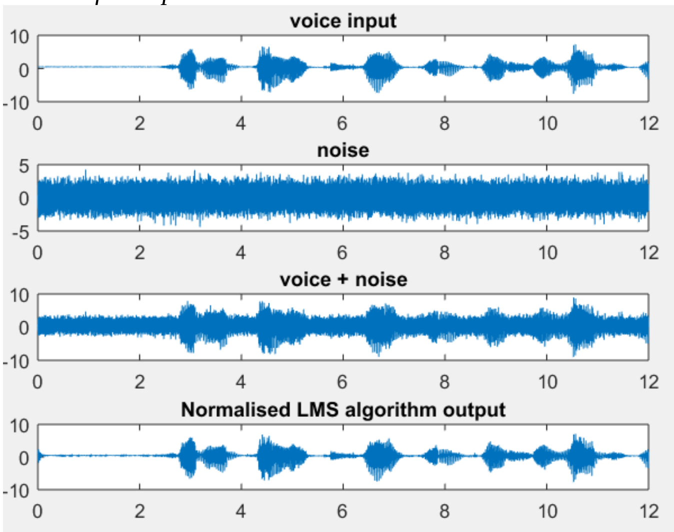
Where sgn(.) = sign function.

International Journal of Engineering Research & Technology (IJERT)
ISSN: 2278-0181
Vol. 8 Issue 08, August-2019

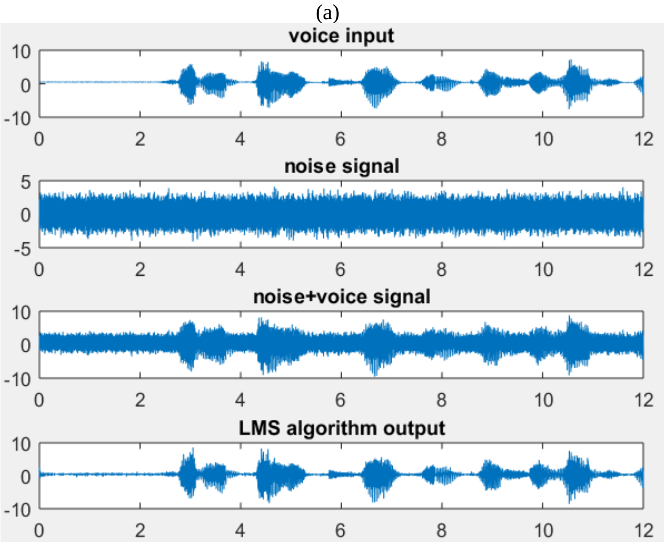
V.RESULT ANALYSIS

OUTPUT WAVEFORMS-

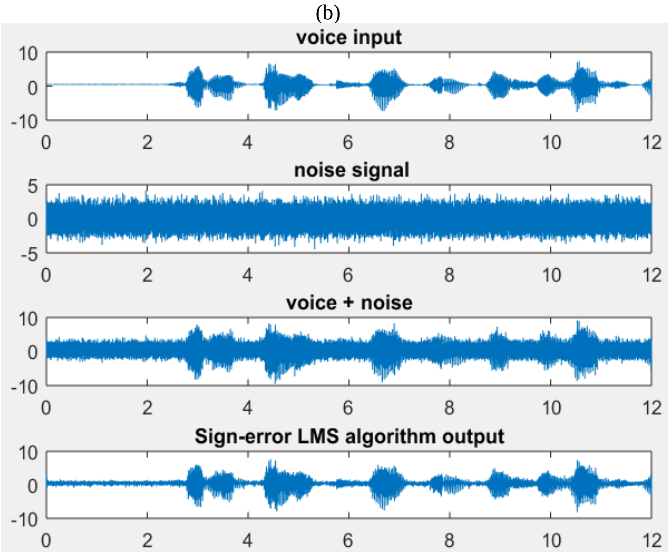
A. for step-size 0.01



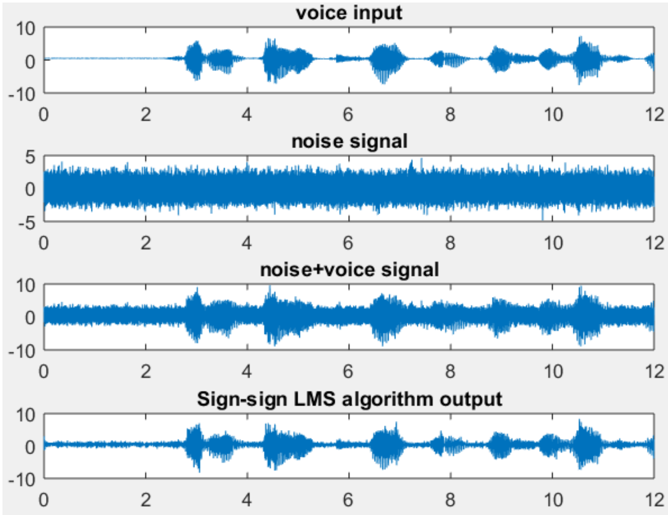
(a)



(b)



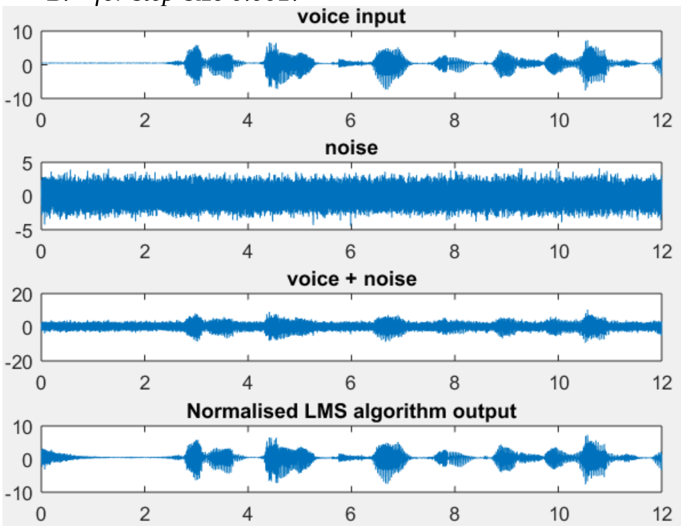
(c)



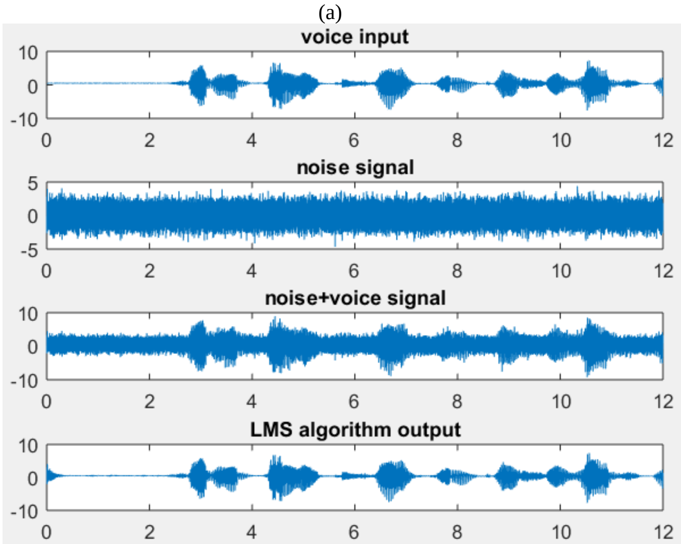
(d)

Fig 3. For step size 0.01 (a) NLMS waveform, (b) LMS waveform, (c) Sign-error LMS, (d) Sign-Sign LMS.

B. for step-size 0.001:

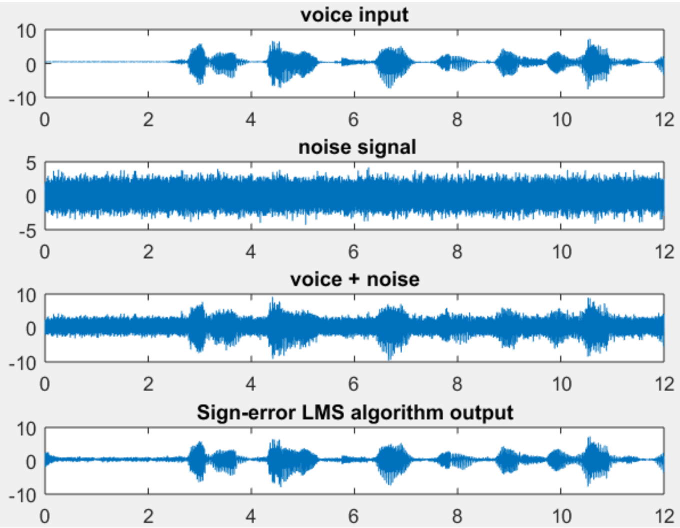


(a)

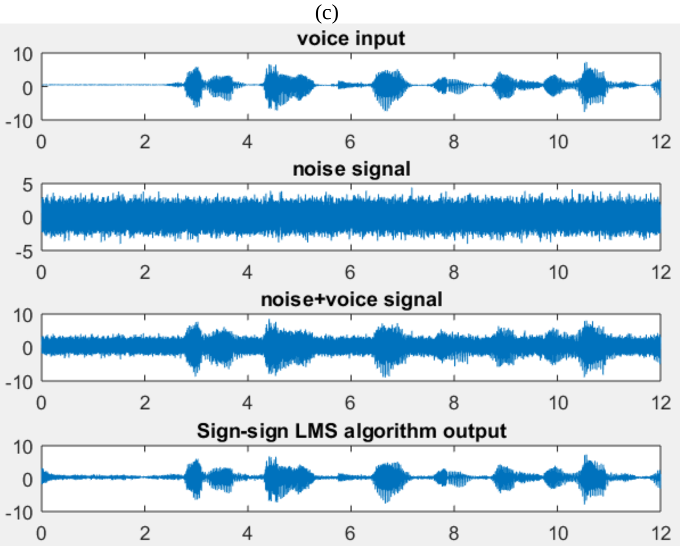


(b)

International Journal of Engineering Research & Technology (IJERT)
ISSN: 2278-0181
Vol. 8 Issue 08, August-2019



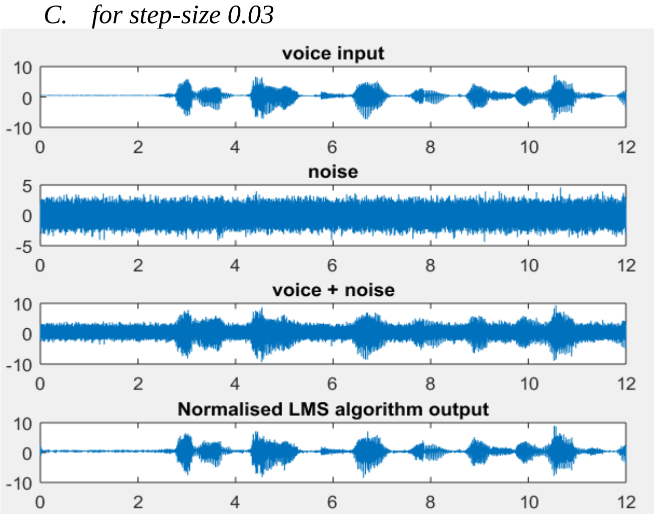
(c)



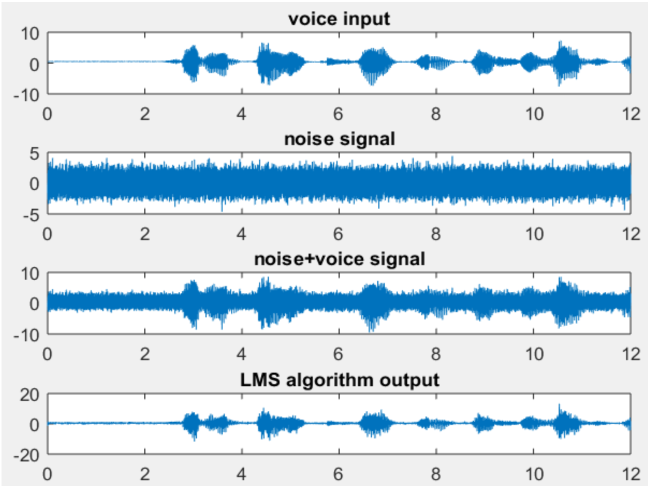
(d)

Fig 4. Waveform for step size 0.001 (a) NLMS, (b) LMS, (c) Sign-error LMS, (d) Sign-Sign LMS.

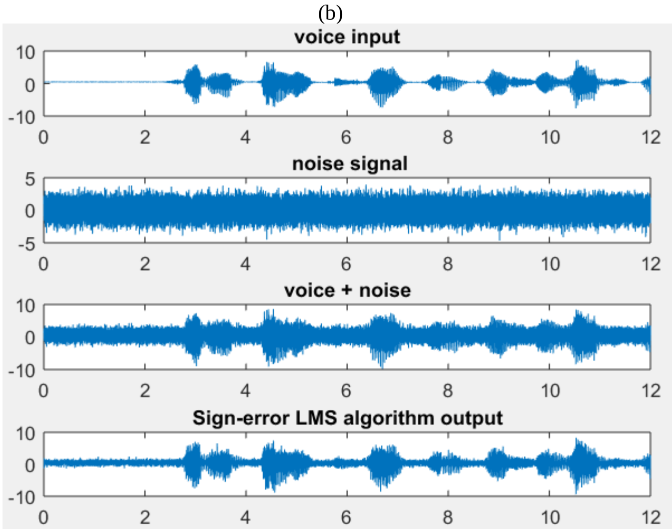
C. for step-size 0.03



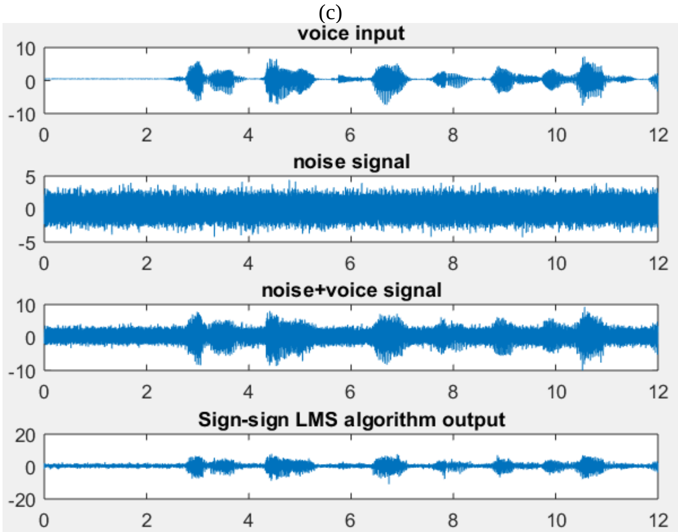
(a)



(b)



(c)

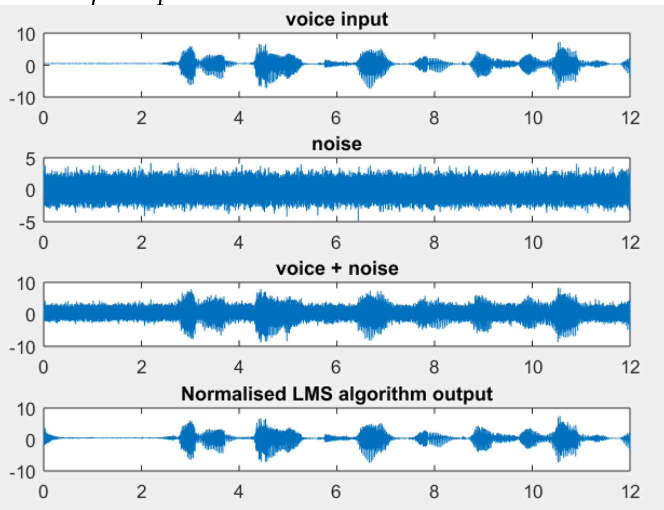


(d)

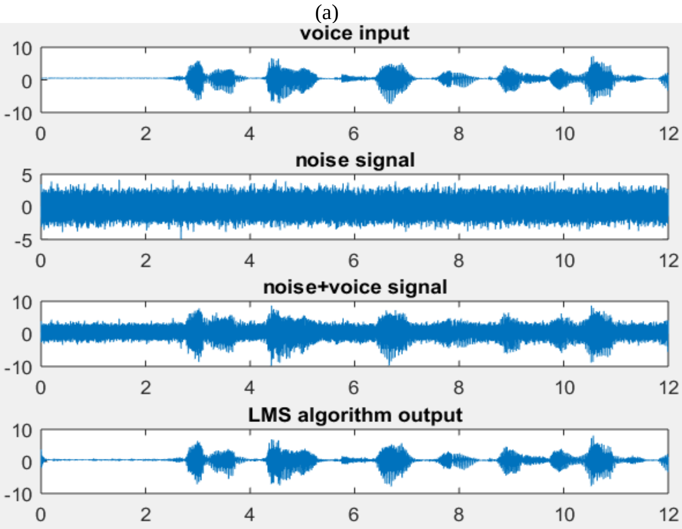
Fig 5. For step size 0.03 (a) NLMS waveform, (b) LMS waveform, (c) Sign-error LMS, (d) Sign-Sign LMS.

International Journal of Engineering Research & Technology (IJERT)
ISSN: 2278-0181
Vol. 8 Issue 08, August-2019

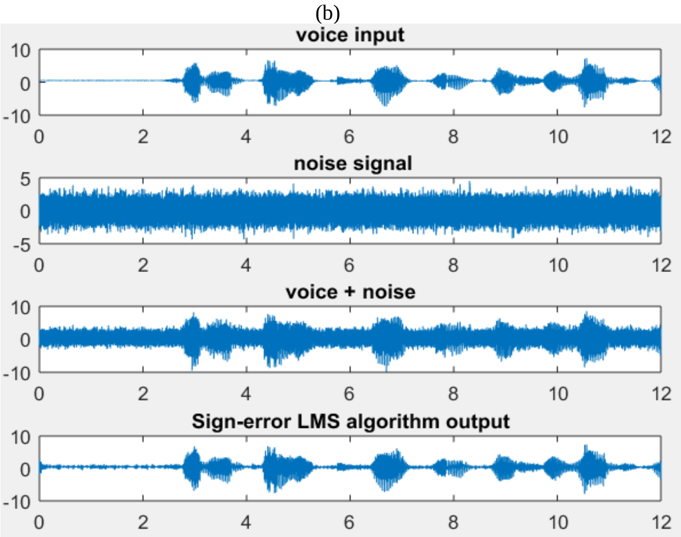
D. for step-size 0.003



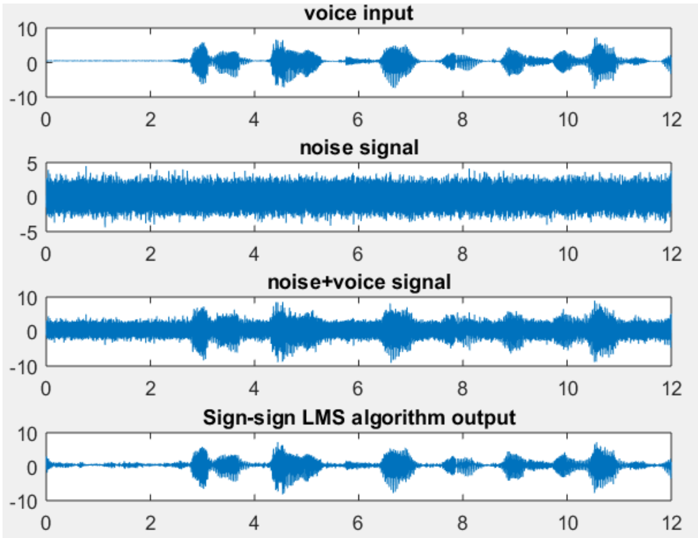
(a)



(b)



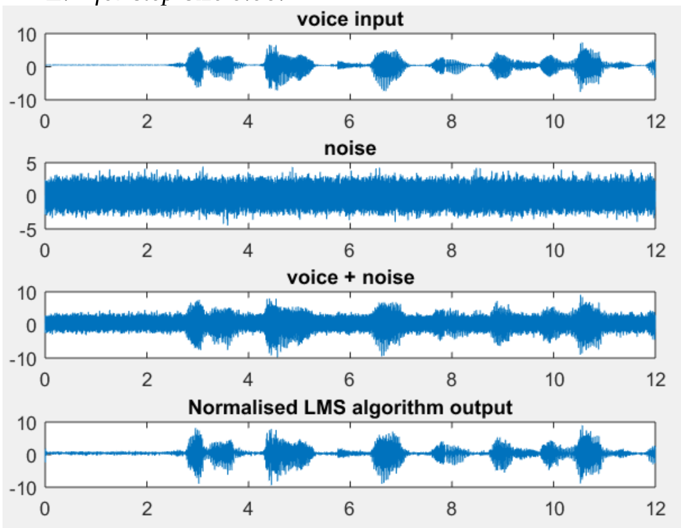
(c)



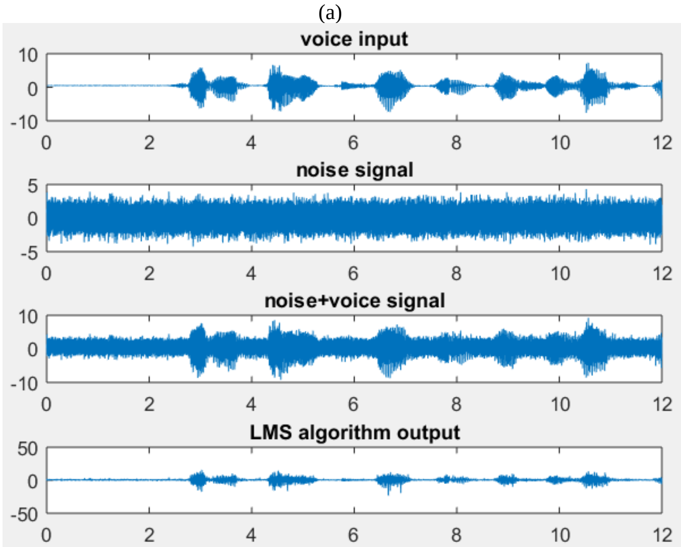
(d)

Fig 6. For step size 0.003 (a) NLMS waveform, (b) LMS waveform, (c) Sign-error LMS, (d) Sign-Sign LMS.

E. for step-size 0.06:

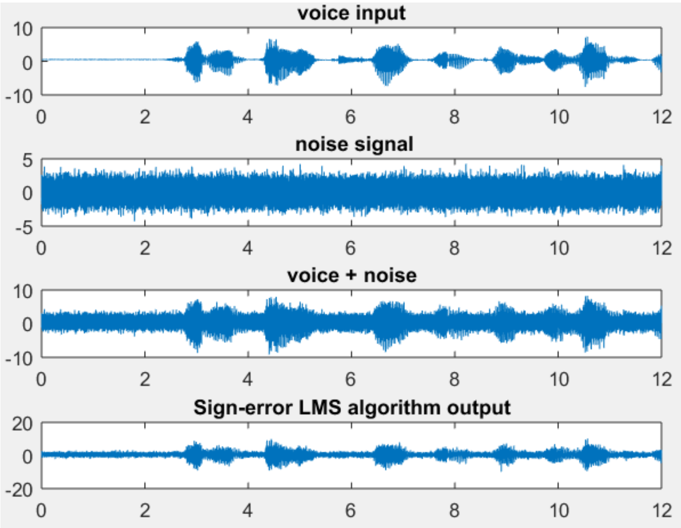


(a)

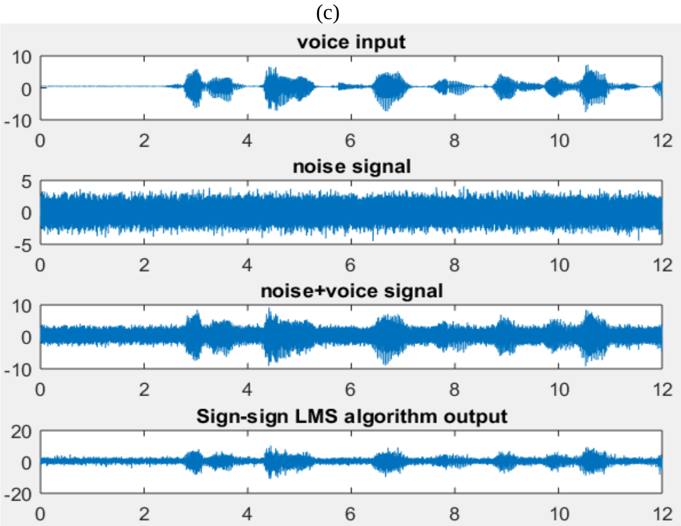


(b)

International Journal of Engineering Research & Technology (IJERT)
ISSN: 2278-0181
Vol. 8 Issue 08, August-2019



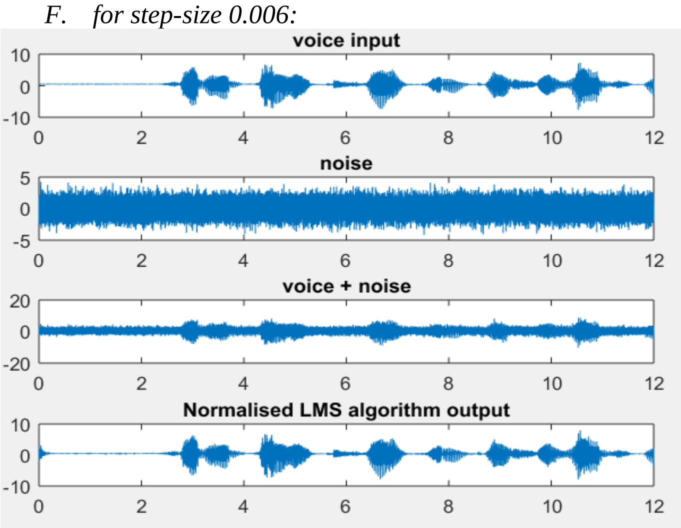
(c)



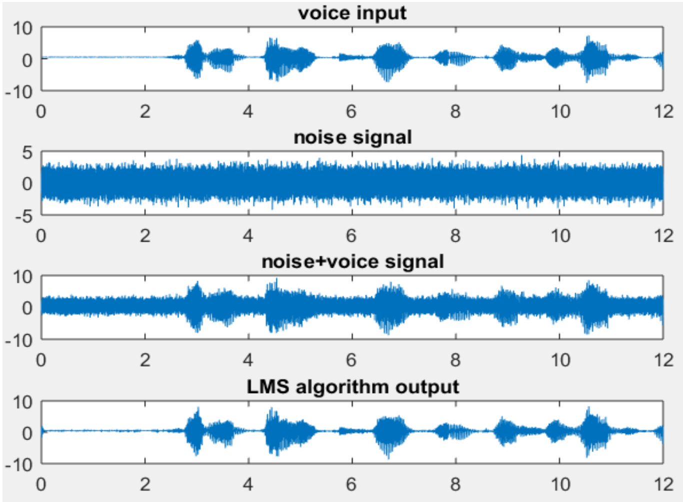
(d)

Fig 7. For step size 0.06 (a) NLMS waveform, (b) LMS waveform, (c) Sign-error LMS, (d) Sign-Sign LMS.

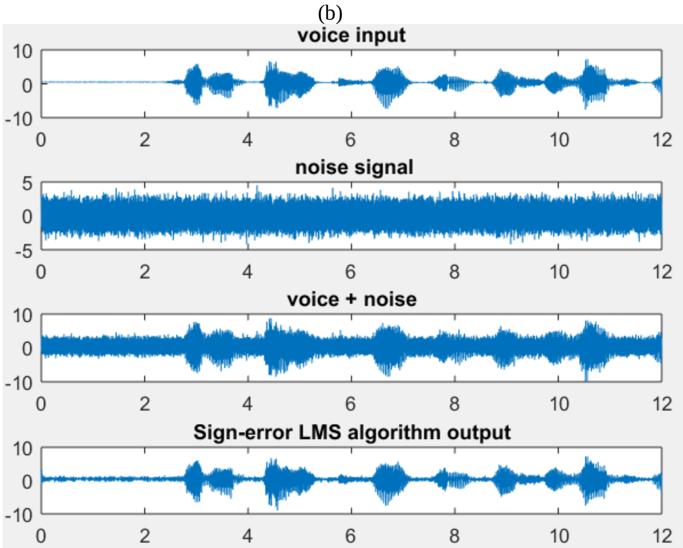
F. for step-size 0.006:



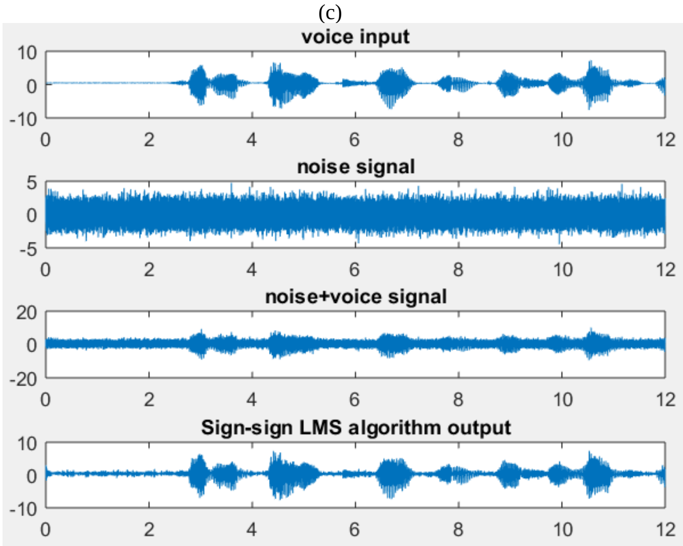
(a)



(b)



(c)



(d)

Fig 8. For step size 0.006 (a) NLMS waveform, (b) LMS waveform, (c) Sign-error LMS, (d) Sign-Sign LMS.

VI.CONCLUSION

Here is the comparison between the different algorithms used for the speech interference cancellation. The SNR values are mentioned in the table below at different step-size. It concludes that Normalized least square mean algorithm is best according to the SNR value for speech noise removal.

REFERENCES

[1] Beniwal, Sandeep Kumar and Sanjeev Kumar Dhull(2013).. “Comparison of Performance of Stereophonic Acoustic Echo Canceller using LMS and NLMS Adaptive Algorithms.

[2] Dhiman, Jyoti, Shadab Ahmad, and Kuldeep Gulia(2013). "Comparison between Adaptive filter Algorithms (LMS, NLMS and RLS)." *International Journal of Science, Engineering and Technology Research (IJSETR)* 2, no. 5: 1100-1103.

[3] Dhull, Sanjeev Kumar and Sandeep K. Arya(2011). “Performance Comparison of Adaptive Algorithms for Adaptive line Enhancer.”.

[4] Dhull, Sanjeev Kumar and Lalita Varma(2013).. “Comparison of different Sub-Band Adaptive Noise Canceller with LMS and RLS.”

[5] Dixit, Shubhra, and Deepak Nagaria(2017).. "LMS Adaptive Filters for Noise Cancellation: A Review." *International Journal of Electrical & Computer Engineering (2088-8708)* 7, no. 5

[6] Ferdouse, Lilatul, Nasrin Akhter, Tamanna Haque Nipa, and Fariha Tasmin Jaigirdar(2011).. "Simulation and performance analysis of adaptive filtering algorithms in noise cancellation." *arXiv preprint arXiv:1104.1962*

[7] Górriz, J.M.; Ramírez, J.; Cruces-Alvarez, S.; Puntonet, C.G.; Lang, E.W.; Erdogmus, D(1, January 2009)., “A Novel LMS Algorithm Applied to Adaptive Noise Cancellation”, IEEE Signal Processing Letters, VOL. 16, NO..

[8] Hadei, Sayed. "A family of adaptive filter algorithms in noise cancellation for speech enhancement." arXiv preprint arXiv:1106.0846

[9] H.G. Hirsch and D. Pearce(2000), "The aurora framework for the performance evaluation of speech recognition systems under noisy conditions," in 6th Int. Conf. on Spoken Language Processing, Beijing, China, pp. 181-188.

[10] Liu, Ming, Ming-Jiang Wang, and Bo-Yang Song(2016). "An efficient architecture of the sign-error LMS adaptive filter." In *2016 13th IEEE International Conference on Solid-State and Integrated Circuit Technology (ICSICT)*, pp. 753-755. IEEE,.

[11] Mahant-Shetti, Shivaling S., Srinath Hosur, and Alan Gatherer(1997). "The log-log LMS algorithm." In *1997 IEEE International Conference on Acoustics, Speech, and Signal Processing*, vol. 3, pp. 2357-2360. IEEE,.

[12] Priyanka Gupta1, Mukesh Patidar2, Pragya Nema3(2015),”Performance Analysis of Speech Enhancement Using LMS, NLMS and UNANR algorithms” IEEE International Conference on Computer Communication and Control,.

[13] Ramli, Roshahliza M., AO Abid Noor, and S. Abdul Samad(2012). "A review of adaptive line enhancers for noise cancellation." Australian Journal of Basic and Applied Sciences 6, no. 6 : 337-352.

[14] Sasaoka, Naoto, Koji Shimada, Shota Sonobe, Yoshio Itoh, and Kensaku Fujii(2009). "Speech enhancement based on adaptive filter with variable step size for wideband and periodic noise." In 2009 52nd IEEE International Midwest Symposium on Circuits and Systems, pp. 648-652. IEEE,.

[15] Thenua, Raj Kumar, and S. K. Agarwal(2010):. "Simulation and performance analysis of adaptive filterin noise cancellation." International Journal of Engineering Science and Technology2, no. 9: 4373-4378.

[16] 11Tienan, L.; Limo, W.; Baochang, X.; Aihua, X.; Hang, Z.(June 10-14, 2002), “Adaptive Noise Canceler and Its Applications for Systems with Time-variant Correlative Noise”, IEEE Proceedings of the 4° World Congress on Intelligent Control and Automation, , Shanghai, P.R.China.