International Journal of Advances in Engineering and Management (IJAEM)
Volume 4, Issue 11 Nov. 2022, pp: 1236-1241 www.ijaem.net ISSN: 2395-5252

Noise removal technique for speech signal in Adaptive Filter using Shadow concept

Mr. Amol P. Bodhane, Dr. V. R. Ratnaparkhe

[1,] *Student, Dept. of Electronics & Telecommunication Engineering, Government College of Engineering, Aurangabad, Maharashtra*

[2] *Associate Professor, Dept. of Electronics & Telecommunication Engineering, Government College of Engineering, Aurangabad, Maharashtra.*

Date of Submission: 25-11-2022 Date of Acceptance: 06-12-2022

ABSTRACT: Adaptive filters take a leading role in digital communication, where it successfully removes the noise and echo present in original information signal. Here, this paper suggested an Adaptive filter for LMS algorithm in accordance with Shadow theory. An effective method of adaptive filtering is used to reconstruct the signal by improving SNR & minimizing MSE. LMS algorithm effectively removes the noise element overreach with original information signal. Fixed LMS algorithm has least possible convergence rate and unvarying steady state error. Here the spectral characteristics of the filter are improved by the shadow technique, as well as obtain a noise free signal at the output side. Here the SNR values of the adaptive filter for the LMS algorithm depending on the shadow concept and without the shadow concept is calculated. Also observes the response of the adaptive filter using the RLS algorithm.

KEYWORDS: Adaptive algorithm, LMS, RLS, shadow theory, SNR, MSE, MATLAB.

I. INTRODUCTION ADAPTIVE FILTER USED IN ADAPTIVE NOISE CANCELLING

Adaptive filters are digital filters and linear types of filters. For elimination of artifacts present in Electro Cardio Gram signal [1] adaptive filters are used. Adaptive filtering uses an adaptive algorithm. According to these adaptive algorithms the filter coefficient is adjusted and uses to control the parameters of the filter. Adaptive filters also have transfer function to control the parameters. Adaptive filters have the capability to customize their impulse response and they don't have any previous idea of the noise and signal qualities. Adaptive filters are digital in nature because these optimization algorithms are complicated. One of the applications of adaptive filter is Adaptive Noise Cancellation (ANC). To get an enhanced SNR (signal to noise ratio) by using a different de-noising method where a modulated signal is used as a reference signal [2]. Here we use Adaptive Noise Cancelling (ANC) to remove the noise which is present in our original information signal [3]. In ANC there are two inputs one is noise corrupted speech signal and second is reference noise input. In ANC the reference noise input passes through the filter and make it equal to the noise present in the corrupted speech signal. After subtracting these two signals the noise gets removed due to same amplitude and same frequency but opposite phases. Here two sensors are used to receive the noise signal and target signal separately. The application of adaptive filters is also in neuro processing systems [4]. Further adaptive filtering with least mean square algorithm is used in various electronics systems. [5].This technique is widely used in many biomedical equipment applications [6]. The LMS algorithm is robust, so it is widely used in many applications [7]. There are two types of impulse response, one is finite and another is infinite i.e. FIR and IIR. Between them FIR is commonly used because FIR have only adjustable zero's on the other hand IIR have both adjustable zero's and pole's[8]. Adaptive Filters have adjustable coefficient [9]. It also has low computational complexity and a fast convergence rate [10]. The structure of LMS algorithm is simple and due to this advantage various researchers use this for hardware implementation [11][12]. Adaptive filters are used in signal processing, medical controlling, channel equalization in cell phones, eco removal etc [13]. The noise used in this system is a random process and doesn't have any prior knowledge [14].

International Journal of Advances in Engineering and Management (IJAEM)
Volume 4, Issue 11 Nov. 2022, pp: 1236-1241 www.ijaem.net ISSN: 2395-5252

II. CONCEPT OF SHADOW FILTER

Shadow type of filter approach uses two filters. One of the types of filters is used in feedback path whereas second type of filter is used in forward path. The spectral distinctiveness of forward path filter can be enhanced by changing the shadow factor 'β' of feedback path filter [15]. Shadow filters are also used in medical monitoring. To design FIR filter the shadow concept is used. In cardiac signal processing noise elimination is done with the help of Shadow based filters. [16].

III. LMS ALGORITHM FOR DESIGNING ADAPTIVE FILTER

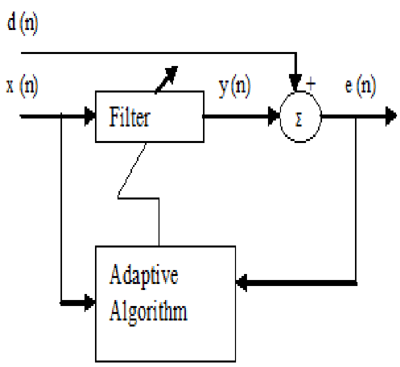

Fig1. Basic block diagram of Adaptive filtering

This above figure 1 shows the basic block plan of adaptive filtering. The input digital signal $x(n)$ is filtered to produce output $y(n)$and minimize the error signal $e(n)$ when subtracted from $d(n)$. Here, $e(n)$ is an error signal playing a significant role in adaptive filtering. This error signal impels the adaptive algorithm which generates the filter coefficient in such a way that minimizes the error signal. The Figure 2 indicates the Adaptive filtering using Least Mean Square LMS Algorithm. Where,

$e(n)$ = original information signal

$x1(n)$ = noise signal (which get added to the information signal to make it corrupted signal)

$H(z)$ = Low pass FIR Filter

$d(n)$ = corrupted signal, $[s(n) + x1(n)]$

$x(n)$ = reference noise signal

$e(n)$ = error signal, $[d(n) - y(n)]$

The weight equation for adaptive filter is,

$w(n+1) = w(n) + \mu * e(n) * x1(n)$, where μ is step size

$y(n) = w(n) + e(n) * x1(n)$

In the design of Adaptive filter with Least Mean Square (LMS) algorithms following steps are implemented:

1. Record or create an original signal and noise signal.
2. Correlate noise with another signal by passing it through an LPF.
3. Mix the noise signal with the original information signal.
4. Passes these signals to the adaptive filter, which utilises the LMS algorithm for determining error $e(n)$.
5. Edit the weight equation $w(n)$
6. Repeat this step until the error get minimum.
7. Then calculate output $y(n)$, input SNR before filtering and output SNR after filtering.

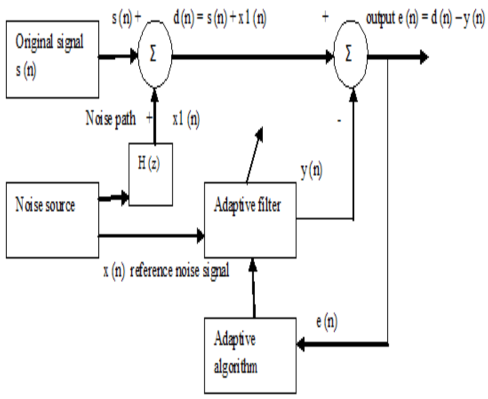

Fig 2. Adaptive Filter with LMS algorithm

IV. SHADOW CONCEPT BASED ADAPTIVE FILTERING WITH LMS ALGORITHM

The below Figure 3 shows the block diagram of shadow adaptive filter. Here the output of Low pass filter is given as feedback (positive or negative) by a shadow filter of high pass or low pass filter. We utilized the shadow concept to find the output for different values of 'β'. Where, β is the shadow factor.

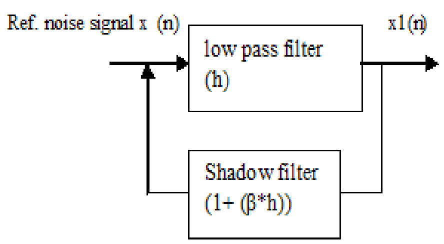

Fig 3. Block diagram of shadow adaptive filter

Following is the equation for the shadow filter with positive feedback relationship.

$$\bar{h}(n) = \frac{\text{low pass filter}}{1 + (\beta * \text{low pass filter})}$$

International Journal of Advances in Engineering and Management (IJAEM)
Volume 4, Issue 11 Nov. 2022, pp: 1236-1241 www.ijaem.net ISSN: 2395-5252

$$\bar{h}(n) = \frac{h}{1+(\beta * h)} \quad , \quad 0 \le \beta \le 1$$

$$x1(n) = x(n) * \bar{h}(n)$$

In above equation the $\bar{h}(n)$ is the output of shadow filter.

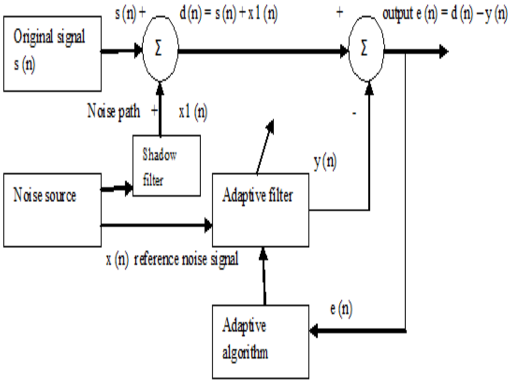

Fig 4. Shadow concept based Adaptive filtering with LMS algorithm

The Figure 4 shows the shadow concept based adaptive filtering with LMS algorithm. The input and output Signal Noise Ratio and Mean Squared Error (MSE) are determined and indicated in table number 1. Below equations are used to determine input and output Signal to Noise Ratio and Mean Square Error, which are:

$$Input\ SNR_{db} = 10\ log_{10}\ (\frac{(original\ \ speec\ h)^2}{(ref\ .noise\ )^2})$$

$$Output\ SNR_{db} = 10 log_{10} (\frac{(denoised\ \ speec\ h)^2}{(ref\ .noise\ )^2})$$

$$\text{MSE} = \left(\frac{1}{N}\right) * \sum_{k=0}^{N} \big(\text{original speech (k)} -$$
denoised speech k2

V. RESULTS AND EXECUTION

MATLAB codes are used for the implementation of these algorithms which is used for adaptive filtering process. The figure 5 indicates the output of the Adaptive filter with Least Mean Square Algorithm by shadow concept. Figure no.5 shows the output of LMS adaptive filter for β=0. Here, the applied original speech signal gets mixed with noise and given to the adaptive filter at the input side and give denoised speech signal at the output side. Here, Kaiser Window is used to observe the output. Also determine and compare the SNR values for shadow factor β which is varying from 1.1 to 2.0. The observed de-noised speech at the output in case of Adaptive filtering with LMS algorithm using shadow factor is clearer than the de-noised speech obtained without shadow factor. Signal to Noise Ratio is also determined for adaptive filter based on shadow concept and without shadow concept. Here we indicate responses for adaptive filtering by using RLS and LMS algorithms.

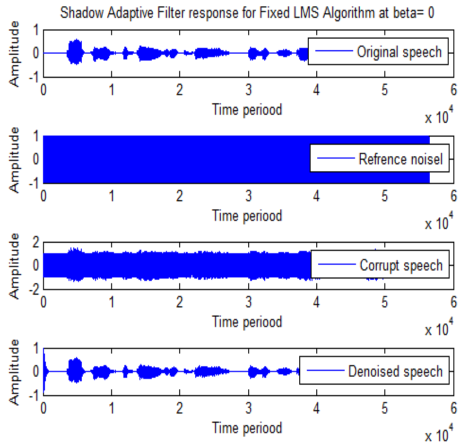

Fig 5: Output Signal of LMS adaptive filter for β=0

The above figure no. 5 shows the waveforms at the output side when the shadow factor β considered as zero. The value of β is equal to zero indicates that we filtered the corrupted signal by using adaptive filtering with the help of Least Mean Square (LMS) algorithm without shadow concept. The original speech and the reference noise are given at the input side to our system. This original signal gets corrupted and we filtered this signal through adaptive filtering. After filtering we observed de-noised speech at the output side of our system. After that we observed the output for various values of shadow factor (β). Below we see that responses.

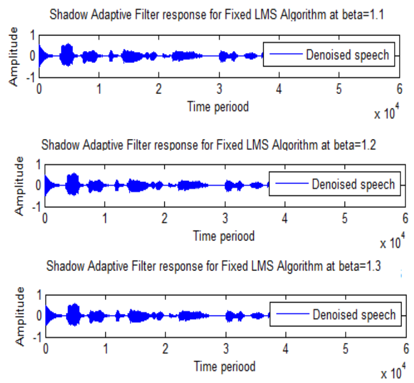

Fig 6: De-noised speech for β= 1.1, 1.2, 1.3

International Journal of Advances in Engineering and Management (IJAEM)
Volume 4, Issue 11 Nov. 2022, pp: 1236-1241 www.ijaem.net ISSN: 2395-5252

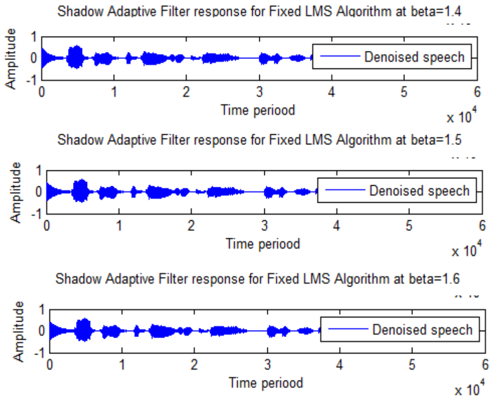

Fig 7: De-noised speech for β= 1.4, 1.5, 1.6

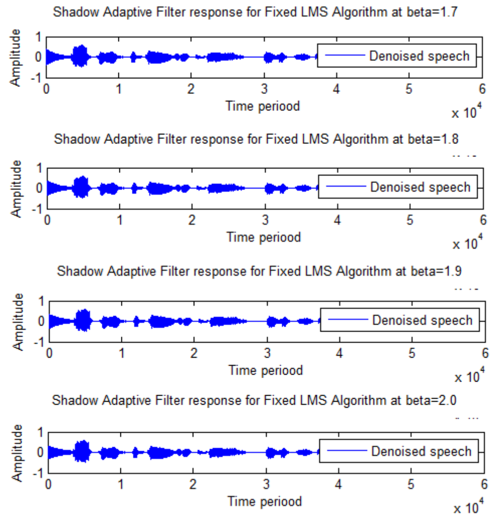

Fig **8:** De-noised speech for β= 1.7, 1.8, 1.9, 2.0

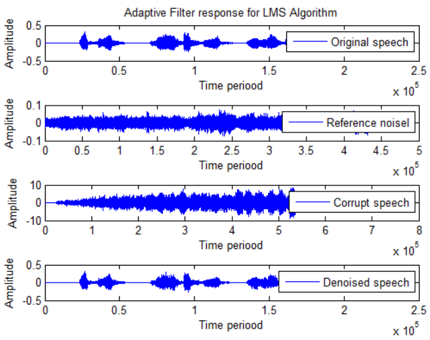

Fig 9: Output of adaptive filtering using Least Mean Square algorithm

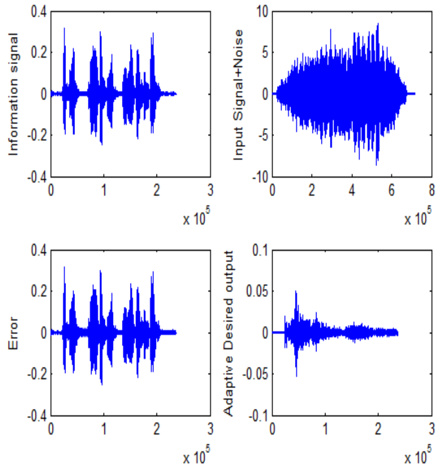

Fig 10: Output Signal of adaptive filtering using RLS algorithm

Figure No. 6 indicates the de noised speech at the output for β = 1.1, 1.2, 1.3. The figure No.7 indicates the de noised speech at the output for β = 1.4, 1.5, 1.6 and the figure number 8 indicates the de noised speech at the output for β = 1.7, 1.8, 1.9, 2.0. Figure number 9 shows the output of adaptive filtering by using LMS algorithm. In addition, we indicate the output of adaptive filtering by using RLS algorithm as indicated in figure no. 10. Following table indicates the input and output Signal Noise Ratio values for the adaptive filtering based on shadow concept and varying β from 1.1 to 2.0, also determined the Mean Squared Error.

Table1. Differentiation of SNR value before filtering and after filtering of speech signal and Mean Squared Error

SSr. .No	Name of Window	Value of SNR before filtering in Decibel (dB)	Value of SNR after filtering in Decibel (dB)	Value of MSE (Mean Square Error)
1	Kaiser	0.0030	0.0035	1.5174e-014

Table2. Comparison of Signal Noise Ratio value before filtering and after filtering and Mean Square Error for Kaiser Window for different shadow factors

Sr. No	Name of Window	Value of β	Value of SNR before filtering in Decibel	Value of SNR after filtering in Decibel	Value of MSE (Mean Square Error)
1	Kaiser Window	1.1	0.0070	0.0072	1.2204e-011
2		1.2	0.0078	0.0080	1.0022e-011
3		1.3	0.0087	0.0090	1.3849e-014
4		1.4	0.0098	0.0101	7.1430e-012
5		1.5	0.0110	0.0114	1.1578e-012
6		1.6	0.0124	0.0128	1.3922e-012
7		1.7	0.0141	0.0146	1.6795e-012
8		1.8	0.0161	0.0166	1.2851e-012
9		1.9	0.0184	0.0190	2.8823e-012
10		2.0	0.0213	0.0220	2.5512e-012

VI. CONCLUSION

Above described system investigated the responses to adaptive filtering using the LMS algorithm, the RLS algorithm, and the LMS algorithm with shadow factor. Also calculated signal to noise ratio (SNR) and mean square error (MSE) for the shadow factor, varying values of β from 1.1 to 2.0. So, the conclusion is that, the use of the shadow concept for the elimination of noise components present in the original information signal gives better SNR and low convergence rate as compared to any other method.

REFERENCES

[1]. Wang an-dong, liu lan ,wei qin," An adaptive morphologic filter applied to ECG de-noising and extraction of r peak at real-time", 2012 aasri conference on computational intelligence and bioinformatics .

[2]. G.M.Wenz, " Acoustic Ambient Noise in Ocean: Spectra and Sources", Journal of the Acoustical Society of America, Vol.34, pp.1936-1956, 1962.

[3]. Thamer M. Jamel Haider Abd Al-Latif Mohamed "Performance Improvements of Adaptive Noise Canceller Using New Adjusted Step Size LMS Algorithm." 3rd IEEE International Conference on Signal Processing Systems (ICSPS 2011), August 27-28, 2011, Yantai, China.)'

[4]. B. Widrow, J . R. Glover, J . M. McCool, et al., "Adaptive noise cancelling: principles and applications." Proc. IEEE. vol. 63, pp .1692-1716, 1975

[5]. Yonggang Zhang and Jonathon A. Chambers, Senior Member, IEEE,'' Convex Combination of Adaptive Filters for a Variable Tap-Length LMS Algorithm'', IEEE signal processing letters, vol. 13, no. 10, october 2006

[6]. N.Mohana Rao, A.S.Srinivasa Rao, P.V.Muralidhar , Venkata L N Sastry.D ,'' Noise Removal in Cardiac Signal by Shadow Digital Filters'', International Journal of Engineering Research & Technology (IJERT)Vol. 2 Issue 10, October – 2013,issn:2278-0181

[7]. Noor K. Muhsin, " Comparison of the RLS and LMS Algorithms to Remove PowerLine Interference Noise from ECG Signal" Al-Khwarizmi Engineering Journal, Vol. 6, No. 2, PP 51- 61 (2010)

[8]. 1B. Harinath, 2U. Rajya Lakshmi, 3K. Krishnam Raju, "A new Family of Shadow Filters-IIR Filters" IJECT Vol. 3, Issue 4, Oct - Dec 2012

[9]. Sekata Olika , Dr.A.Rajani, "Adaptive Noise Cancellation For Speech Signal", International Journal of Scientific and Research Publications, Volume 10, Issue 9, September 2020.

[10]. Deepanjali Jain, Dr. Poonam Beniwal, "Review Paper on Noise Cancellation using Adaptive Filters", International Journal of Engineering Research & Technology (IJERT) IJERT,Vol. 11 Issue 01, January-2022.

[11]. M. Koteswara Rao1, I. Santhi Prabha2," Introducing Adaptive filters Based on Shadow Concept for Speech Processing" Journal of Telematics and Informatics (JTI) Vol.2, No.2, September 2014, pp. 67~73ISSN: 2303-3703

[12]. Shubhra Dixit, Deepak Nagaria, "LMS Adaptive Filters for Noise Cancellation: A Review", International Journal of Electrical and Computer Engineering

(IJECE)
Vol. 7, No. 5, October 2017.

[13]. Rodrigo M. S. Pimenta, Leonardo C. Resende, Newton N. Siqueira, Idiego B. Haddad," A New Proportionate Adaptive Filtering Algorithm with Coefficient Reuse and Robustness Against Impulsive Noise" IEEE , 2018 26th European Signal Processing Conference (EUSIPCO)

[14]. Shifei Hui, Mingjiang Wang, Yufei Han, "Research on Adaptive Algorithm for Weight Partial Update of ANC" IEEE, 2019 IEEE 2nd International Conference on Information Communication and Signal Processing (ICICSP)

[15]. Eppili Jaya, P. Krishna Rao, "Analysis of Shadow Filter With Hybrid Windows" International Journal of Engineering Research & Technology (IJERT) IJERTIJERT ISSN: 2278-0181Vol. 3 Issue 5, May – 2014

[16]. G.Aishwarya1, Abhishek Singh2," A Review on Adaptive Algorithms" International Research Journal of Engineering and Technology (IRJET), Volume: 07 Issue: 07 | July 2020