Novel Method for Adaptive Filter Algorithm with Shadow Mechanism for Speech Signal

Vaishali Popatrao Akolkar[1]
[1] Student
Dept. of Electronics & Communication engineering,
Marathwada Institute of Technology, Aurangabad,
Maharashtra

A. A. Somani[2]
[2] Asst.Professor,
Dept. of Electronics & Communication engineering,
Marathwada Institute of Technology, Aurangabad,
Maharashtra

***Abstract*— Adaptive filters are successfully using in cancellation of echo and noise signals in signal processing, radar communications and in speech communication systems. Here we are proposing an Adaptive Filter with LMS Algorithm based on Shadow concept. Which is useful for the cancellation of the noise component overlap with speech signal in the same frequency range, but fixed LMS algorithm produces minimum convergence rate and fixed steady state error. So we presents design, implementation and performance of adaptive FIR filter, based on Shadow concept, which produces minimum mean square error compare to fixed LMS, and we also obtains de-noised speech signal at output, and also we propose to calculate SNR values of Adaptive Filter with LMS algorithm with and without Shadow concept. And also observe the output for adaptive filter using LMS and RLS algorithms.**

Keywords—Adaptive algorithm, LMS algorithm, RLS algorithm, shadow concept.

I. ADAPTIVE FILTERS

An adaptive filter is a system with a linear filter. Adaptive filters are successfully using in removal of artifacts presenting in ECG signal [1]. It has a transfer function which is controlled by variable parameters and a means to adjust those parameters according to an optimization algorithm. These optimization algorithms are complex and because of that almost all adaptive filters are digital filters. The performance of active noise control system which uses linear adaptive filter algorithm was degraded by the non-linear saturation [2]. Adaptive filters are used in devices that are mobile phones, other communication devices, camcorders, digital cameras, medical monitoring etc. The LMS algorithm is widely used in many applications as an effect of its simplicity and robustness [3]. Filters with adjustable coefficient are called adaptive filters. Although both FIR and IIR filters have been considered for adaptive filtering, but FIR filter is commonly used. Various adaptive filter de-noising methods were analyzed with modulated signal as reference signal to achieve a better SNR [4]. Adaptive filters provide performance excellence due to their inherent pole-zero structure as compared with adaptive finite impulse response (FIR) filters that have an all-zero form, in active noise control application [5]. The stability of filters depends critically on the algorithms for adjusting its coefficients. RLS Filters [6].The adaptive filters are widely used in areas such as control systems, communications, signal processing, acoustics, and others to deal with random signals with stationary or quasistationary statistics [7]. An adaptive filters are using in neuro processing systems [8]. Adaptive noise cancelling is used for the noise cancellation and it is produce a signal that is equal to a disturbance signal in amplitude and frequency but has opposite phase. These two signals results in the cancellation of noise signal. LMS based adaptive filters used in all sparse systems for noise Cancellation [9]. Adaptive LMS filters are employed in the design of mechanical, electronic systems [10]. Adaptive filtering technique has been shown to be useful in many biomedical applications [11].

II. SHADOW CONCEPT

In shadow filter mechanism uses two filter one is in forward path and second one is feedback filter, by this arrangement the spectral characteristics forward path filter improves by varying the shadow factor ‘β’. Shadow Mechanism is successfully used in improving the spectral characteristics of windows [12]. Shadow based filters are used in cardiac signal processing for elimination of noises. Shadow mechanism interprets window characteristics which are used in the design of FIR filter. And also it is used in the design of tunable FIR filter. Shadow based filters are used in cardiac signal processing for elimination of noises [13].

III. DESIGN OF ADAPTIVE FILTER WITH FIXED LMS ALGORITHM

The Figure 1 shows the block diagram of Adaptive filter with Fixed LMS Algorithm which processes the noised speech signal through it. Where

$s(n)$ = clean speech signal
$v(n)$ = noise signal
h = Low pass FIR Filter
$v1(n) = h * v(n)$
$d(n)$ = noised speech signal, $[s(n)+v1(n)]$
$y(n)$ = Filtered Noise signal
$e(n) = d(n)-y(n)$, [Original speech signal]

the adjustable weights are typically determined by the LMS Algorithm, the weight update equation is

$w(n+1) = w(n)+\mu*e(n)*v1(n)$
$y(n) = w(n)+e(n)*v1(n)$

Steps to design adaptive Filter with Fixed LMS

1. Create or record actual speech signal.
2. Create or record a noise signal.
3. Correlate noise by passing through a low pass filter.
4. Merge Noise signal with actual Noise signal.

5. Pass this merged signal to Adaptive filter using Fixed LMS Algorithm.
6. Calculate error e(n)
7. Update weight equation w(n)
8. Repeat step 7 and calculate adaptive output y(n) until error is minimized.
9. Calculate input SNR and output SNR

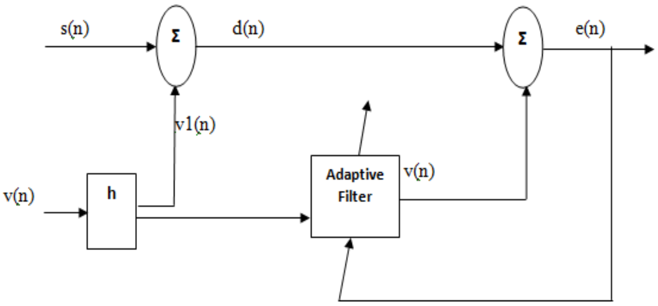

Fig1. Block diagram of Fixed LMS Adaptive Filter

IV. DESIGN OF ADAPTIVE FILTER WITH FIXED LMS ALGORITHM BASED ON SHADOW CONCEPT

The Figure 2 shows the block diagram of Adaptive filter with Fixed LMS Algorithm with Shadow concept. In shadow filter mechanism the Low pass filter output is feedback either positively or negatively by a shadow filter of same type or different type. Here we used the shadow mechanism to find best combination for different values of 'β'. Hence we can derive expression of the transfer function for the shadow mechanism with positive feedback connection is,

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

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

$$v1(n) = v(n) * \bar{h}\,(n)$$

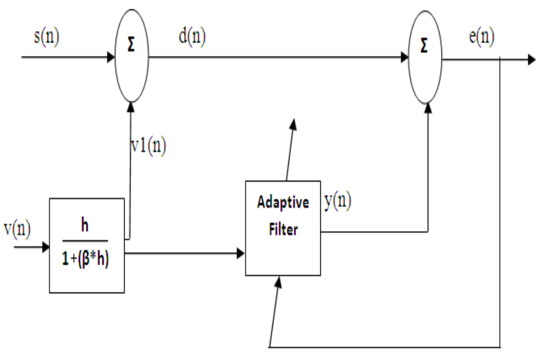

Fig 2. Block diagram of Shadow based LMS Adaptive Filter

The input SNR, output SNR and MSE are calculated and tabulated in table 1. The equations used to calculate input SNR and output SNR and MSE are

Input SNR_{dB} = 10 log $_{10}$ ((original. speech) ^2 / (ref. noise) ^2)

Output SNR_{dB} = 10 log $_{10}$ ((denoised. speech) ^2 / (ref. noise) ^2)

$$MSE = (1/N) * \sum_{k=0}^{N} \left(original.speech\,(k) - denoised.speech\,(k)\right)^2$$

V. RESULT AND IMPLEMENTATION

We implement these algorithms in MATLAB by using Matlab codes for adaptive filtering. The figure3 shows the response of the Adaptive filter with fixed LMS Algorithm with shadow concept. And we applied a noise signal to Speech and compare the signal to noise ratio of Noised signal before and after the filtering for Kaiser Window. When the Noised speech is filtered with Adaptive Filter with Fixed LMS algorithm the whole noise was removed, producing a near clean signal with different 'β' values of shadow FIR Filter for Kaiser Window. SNR, Steady state error are computed for adaptive filter based on without shadow and with shadow concept. The results also show responses of the adaptive filter with LMS and RLS algorithms.

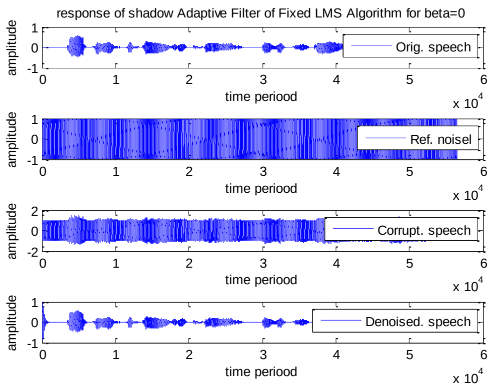

Fig 3: Response of shadow adaptive filter of fixed LMS algorithm for β=0

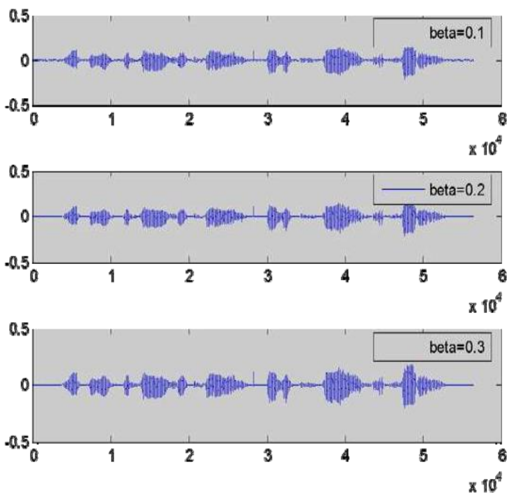

Fig 4: De-noised speech for β= 0.1, 0.2, 0.3

International Journal of Engineering Research & Technology (IJERT)
ISSN: 2278-0181
Vol. 10 Issue 09, September-2021

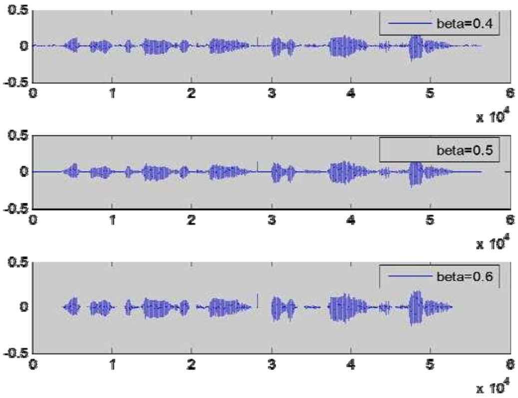

Fig 5: De-noised speech for β= 0.4, 0.5, 0.6

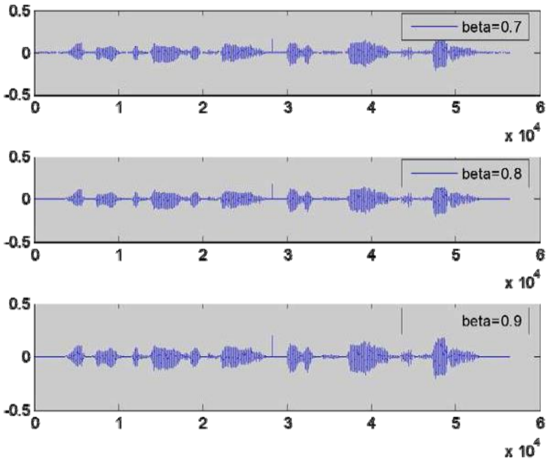

Fig 6: De-noised speech for β= 0.7, 0.8, 0.9

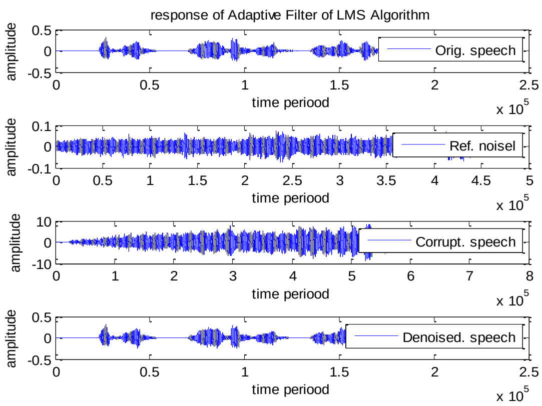

Fig 7: Response of adaptive filter of LMS algorithm

The above figure 4 shows the denoised speech at the output for β = 0.1, 0.2, 0.3. The figure 5 shows the denoised speech at the output for β = 0.4, 0.5, 0.6 and the figure 6 shows the denoised speech at the output for β = 0.7, 0.8, 0.9. The below figure 7 shows the response of adaptive filter of LMS (least mean square) algorithm. And we also show the response of adaptive filter for RLS (recursive least square) algorithm as shown in figure 8.

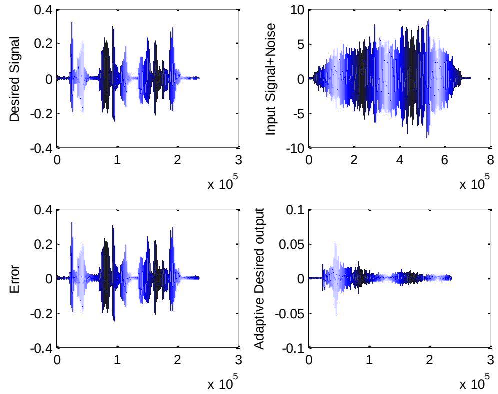

Fig 8: Response of adaptive filter of RLS algorithm

Below table shows the input and output SNR values for shadow based fixed LMS algorithm for different values of β, also calculate the MSE.

Table1. Comparison of SNR of before and after filtering of speech signal and MSE

Sr no	Window	SNR before filtering in dB	SNR after filtering in dB	MSE(Mean Square Error)
1	Kaiser	0.0020	0.0020	1.5176e-012

Table 2. Comparison of SNR and MSE for Kaiser Window and shadow factors

Sr. no	Window	β	SNR before filtering in dB	SNR after filtering in dB	MSE(Mean Square Error)
1	Kaiser Window	0.1	0.0022	0.0023	1.2204e-012
2		0.2	0.0025	0.0026	1.0022e-012
3		0.3	0.0028	0.0029	8.3850e-013
4		0.4	0.0032	0.0033	7.1330e-013
5		0.5	0.0036	0.0037	6.1598e-013
6		0.6	0.0040	0.0041	5.3923e-013
7		0.7	0.0045	0.0046	4.7795e-013
8		0.8	0.0050	0.0052	4.2850e-013
9		0.9	0.0056	0.0058	3.8824e-013
10		1.0	0.0062	0.0064	3.5522e-013

V. CONCLUSION

The implementation of adaptive FIR filter using shadow mechanism with fixed LMS algorithm for Kaiser Window was performed and show response of shadow adaptive filter from figure 3. We also observed response of shadow adaptive filter for different shadow factors as shown from figure 4 to figure 6. We compared SNR, mean square error (MSE) at input and output which are shown from table 1 and table 2. Later we show the response of adaptive filter by using LMS and RLS algorithms as shown from figure 7 and figure 8. From the above discussion it is concluded that shadow based fixed LMS adaptive filter produces better responses in terms of SNR and MSE as compare to adaptive filter using LMS and RLS algorithms.

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] Sen M. Kuo, *Senior Member, IEEE,* and Hsien-Tsai Wu, *Member, IEEE,''* Nonlinear Adaptive Bilinear Filters for Active Noise Control Systems*'',*ieee transactions on circuits and systems—i: regular papers, vol. 52, no. 3, march 2005 617

[3] 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

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

[5] Jong-Yih Lin _, Ching-Wen Liao "New IIR filter-based adaptive algorithm in active noise control applications: Commutation error-introduced LMS algorithm and associated convergenceassessment by a deterministic approachI"- Automatica 44 (2008) 2916_2922-Elsevier Publications

[6] Ibtissam Constantin,Regis Lengelle-"performance Analysis of Kernal Adaptive Filters Based on LMS Algorithm"-procedia Computer science 20(2013) 39-45,sciencdirect.

[7] Garanayak, P., & Panda, G. (2016). Fast and accurate measurement of harmonic parameters employing hybrid adaptive linear neural network and filtered-x least mean square algorithm. IET Generation, Transmission & Distribution, 10(2), 421-436.

[8] Protocoles, Qualite´ des soins e300 (II) / Revue d'E´ pide´miologie et de Sante´ Publique 55S (2012) e297–e303

[9] Bijit Kumar Das and MrityunjoyChakraborty, *Senior Member, IEEE,''* Sparse Adaptive Filtering by an Adaptive Convex Combination of the LMS and the za-lms algorithm*'',*ieee transactions on circuits and systems—i: regular papers, vol. 61, no. 5, may 2014 1499

[10] Márcio H. Costa, José Carlos M. Bermudez, *Member, IEEE*, and Neil J. Bershad*, Fellow, IEEE,''* Stochastic Analysis of the LMS Algorithm with a Saturation Nonlinearity Following the Adaptive FilterOutput*'',*ieee transactions on signal processing, vol. 49, no. 7, july 2001

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

[12] 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

[13] N Mohana Rao, AS Srinivasa Rao, PV Muralidhar, Venkata LN Sastry D. "Noise Removal in Cardiac Signal by Shadow Digital Filters". *International Journal of Engineering Research & Technology (IJERT)*. 2013; 2(10), issn: 2278-0181