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

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Abstract - *In signal processing, radar communications and in speech communication systems these Adaptive Filters are being used successfully in cancellation of echo and noise signals. Here we are proposing an Adaptive Filter with LMS Algorithm based on Shadow concept in the same frequency range it is useful for the cancellation of the noise component overlap with speech signal, but fixed LMS algorithm produces minimum convergence rate and fixed steady state error. So here we present 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 obtain de-noised speech signal at output. And here we also 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.*

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***Key Words*: Adaptive algorithm, LMS algorithm, RLS algorithm, shadow concept.**

1. INTRODUCTION

1.1 Adaptive filters

An adaptive filter can be called as a system with a linear filter. Adaptive filters are successfully being used in removal of artifacts present in ECG signal [1]. It has a transfer function that is controlled by different variable parameters and a means to adjust these 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 like mobile phones, camcorders, digital cameras, medical monitoring, other communication devices 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]. Due to their inherent pole-zero structure, Adaptive Filters provides excellent performance compared to Adaptive Finite Impulse Response (FIR) filters that have an all-zero form, in active noise control application [5]. The filters' stability depends critically on the algorithms for adjusting its coefficients of RLS Filters [6]. The adaptive filters are widely used in areas via control systems, communications, signal processing, acoustics and others to deal with random signals with stationary or quasi-stationary 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 cancellation of noise signals. For noise cancellation, LMS based adaptive filters are used in all sparse systems [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].

1.2 Shadow Mechanism

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]. For elimination of noises in cardiac signal processing the Shadow based filters are used [13]. Shadow mechanism interprets window characteristics which are used in the design of FIR filter. And also it is used in the design of tuneable FIR filter.

2. DESIGN OF ADAPTIVE FILTER WITH FIXED LMS ALGORITHM

The Fig 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

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

$$y(n) - w(n)+e(n)*v1(n)$$

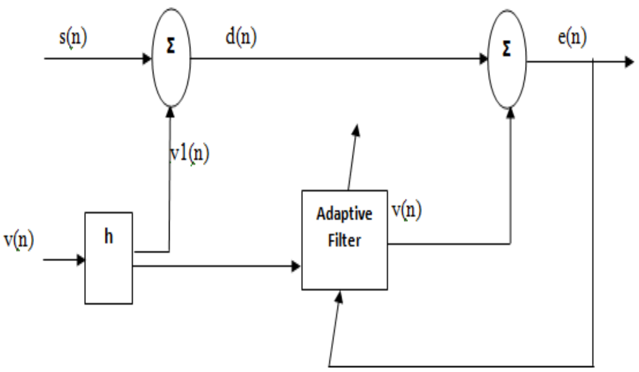

Fig -1: Block diagram of Fixed LMS Adaptive Filter

2.1 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

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

The Fig 2 below shows the block diagram of Adaptive filter with Fixed LMS Algorithm along with Shadow concept. In shadow filter mechanism the LPF output is feed backed either positively or negatively by a shadow filter of same or different types. Here we have used the shadow mechanism to find best combination for different values of β. Hence we can derive expression of the TF for the shadow mechanism with positive feedback connection is,

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

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

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

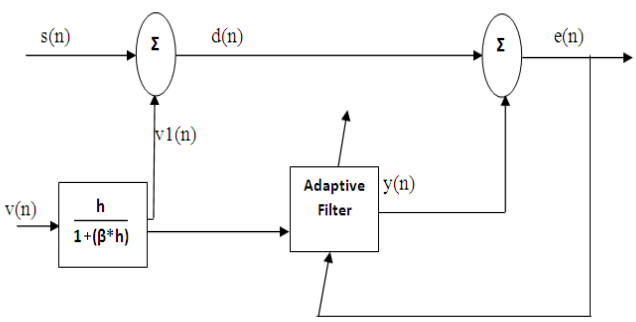

Fig -2: Block diagram of Shadow based LMS Adaptive Filter

The input SNR, output SNR and MSE are calculated and tabulated in table 1. The equations that are 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=
$$(1/N) * \sum_{k=0}^{N} (\text{original. speech}(k) - \text{denoised. speech}(k))^2$$

4. RESULT AND IMPLEMENTATIONS

We implement these algorithms in MATLAB by using Matlab codes for adaptive filtering. The Fig 3 shows the response of the Adaptive filter with fixed LMS Algorithm with shadow concept and we have applied a noise signal to Speech and compare the SNR 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 eliminated producing a near clean signal with different 'β' values of shadow FIR Filter for Kaiser Window. Signal-to-Noise Ratio, 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.

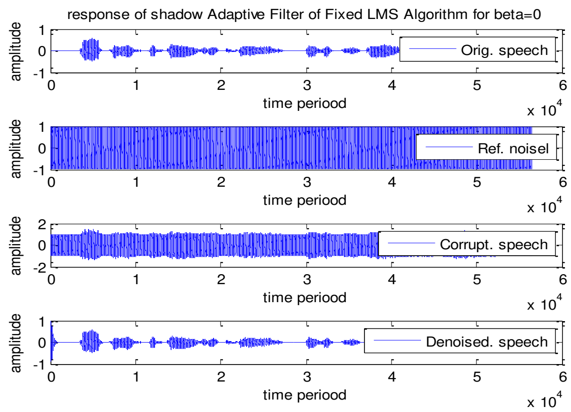

Fig -3: Response of shadow adaptive filter of fixed LMS algorithm for β=0

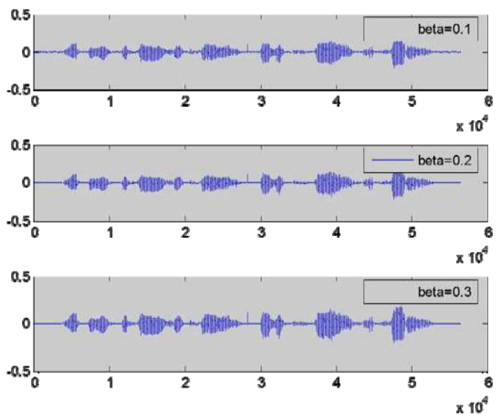

Fig -4: De-noised speech for β=0.1, 0.2, 0.3

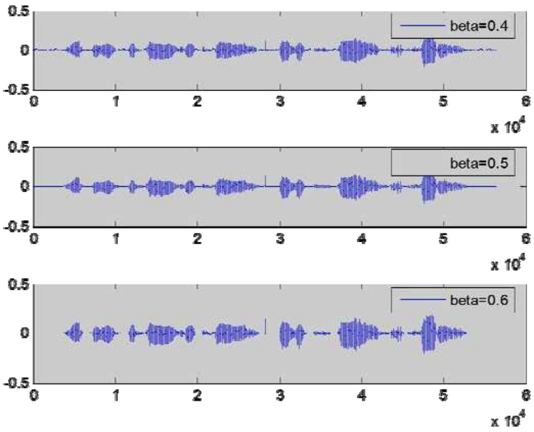

Fig-5: De-noised speech for β= 0.4, 0.5, 0.6

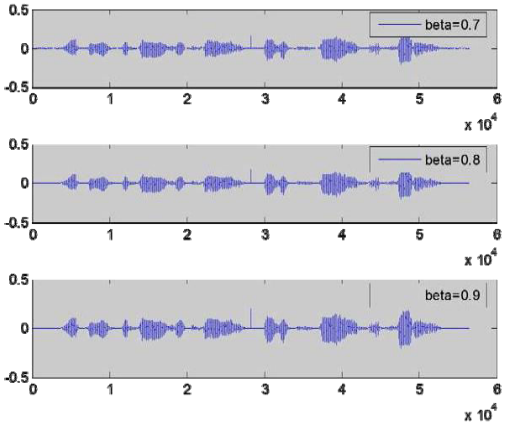

Fig-6: De-noised speech for β= 0.7, 0.8, 0.9

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

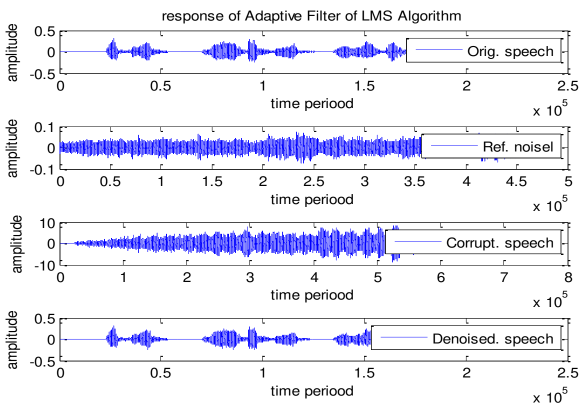

Fig-7: Response of adaptive filter of LMS algorithm

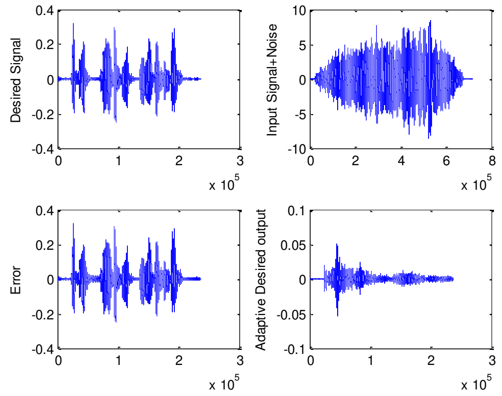

Fig-8: Response of adaptive filter of RLS algorithm

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

Table -1: 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 Window	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

3. 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 Fig. 3. We also observed response of shadow adaptive filter for different shadow factors as shown from Fig. 4 to Fig. 6. We compared SNR, Mean Square Error (MSE) at input & output which are shown from Table 1 and Table 2. Later we have shown the response of Adaptive Filter by using LMS and RLS algorithms as shown from Fig. 7 and Fig. 8. From the above discussion it can be concluded that shadow based fixed LMS Adaptive Filter produces better responses in terms of SNR and MSE as compared to Adaptive Filter using LMS and RLS algorithms.

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