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NOISE REDUCTION IN SPEECH SIGNALS USING RECURSIVE LEAST SQUARE ADAPTIVE ALGORITHM

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ABSTRACT: *In many types of communications that involve speech systems it has been observed that the speech signals are easily interfered with by noise to corrupt it and this tampers with the output accuracy and performance of the systems. It is important that the noise is filtered out from the corrupt speech signal to improve the performance and accuracy of the speech systems or quality of the speech for listening. Filtering out the noise in practice without compromising the integrity of the speech signal is not very simple. Several speech filtering algorithms have been used by various researchers to filter out noise present in speech. In this paper a finite impulse response adaptive filter of order 32 and forgetting factor of λ=1.0, based on recursive least square adaptive algorithm for coefficient update is designed to filter out additive white Gaussian noise from a speech signal. The speech signal is produced by converting a real voice statement "Recursive Least Square Adaptive Algorithm is Very Efficient in the Processing of Speech Signals" to speech signal with a microphone and stored in a file in a computer system. An "audioread" command is used to load the stored speech signal into a matlab edit window and a 10.5db additive white Gaussian noise is generated with matlab and used to corrupt the loaded speech signal. Applying the corrupt speech signal to the designed filter indicates that the noise is drastically reduced. We used four properties to evaluate the performance of this algorithm and they are the output sound, signal morphology, frequency distribution and the filter attenuation strength. The matlab program code for the simulation is provided in this paper.*

KEYWORDS: adaptive algorithm, speech signal, RLS, additive white Gaussian noise, Power spectral density.

INTRODUCTION

In speech communication systems, the processing of the signals involves dealing with noise which the speech signals are contaminated with before getting to the stage of processing. The noise components compromise the quality of the speech signals and result in substantial loss of information contained in it. Therefore it is important to remove these noise components if the message or information content of the signals is to be preserved or its good property for listening purpose is to be maintained. Different researchers have used different adaptive filtering algorithms to process the speech signals. The researcher in [1] used LMS adaptive filter to remove additive white Gaussian noise, traffic noise and airplane noise from voice signals. The phenomenon of fading was also demonstrated with the adaptive filter in [1]. The order of the filter and the algorithm step size depend on the noise to be removed. What appears to be a shortcoming in [1] is the order of the filter which seems to be unnecessarily high for such application. In [2] the authors applied RLS adaptive algorithm in the filtering of additive white Gaussian noise present in transmitted audio signal in a graphic user interface (GUI) or filter builder platform of matlab and compared the performance and that of non-adaptive linear filters. The result shows that for the same noisy audio signal the RLS based filter output has a better filtered signal waveform almost devoid of the noise component. Fast block least mean square (FBLMS) algorithm is also very effective in cancelling white Gaussian noise in speech [3]. The original noise free signal is a recorded audio signal, and a white Gaussian noise generated with matlab is added to the original speech signal to form a noisy audio/speech signal. When the designed adaptive filter is used to filter the noisy signal result shows that the algorithm can remove the different levels of noise more efficiently and effectively and may exhibit faster response. In addition it has a low computational complexity property than LMS algorithm. The authors in [4] applied LMS algorithm-based FIR adaptive filter with a step size of 0.006 to cancel out from a voice signal, a combination of additive white Gaussian noise and another random noise of 0.25 amplitude. The result shows that the LMS algorithm effectively cancelled out the composite random noise. In [5] a comparison of three different adaptive filters designed with least mean square (LMS), normalised least mean square (NLMS) and recursive least square (RLS) algorithms to denoise an audio signal of pink noise, is carried out. The result shows that each algorithm cancels the pink noise in the contaminated audio signal but the output from the NLMS algorithm has the highest signal to noise ratio. Enhancing speech signal with RLS based adaptive filtering method was demonstrated by [6]. Here a noisy data is prepared by adding Babble and pink noise to a clean speech samples. The noisy speech which is sampled at a frequency of 8 kHz was filtered with RLS based adaptive filter, and result shows a significant reduction in the noise content of the speech signal. In [7] LMS-based adaptive filter of order 29 designed with a sampling frequency of 8000Hz and step size of 0.006 was used to reduce additive white Gaussian noise in audio signal. The result shows that the algorithm performed effectively.

In [8] a new adaptive filtering algorithm known as modified adaptive filtering with averaging (MAFA) algorithm was developed by the researchers and used to cancel white Gaussian noise in speech signals. They modified an existing algorithm known as adaptive filtering with averaging (AFA) [9] developed to improve on the high computational and stability issues inherent in RLS algorithm. The result of the experiment shows that the new algorithm (MAFA) improved the stated shortcomings in RLS algorithm and provided higher signal to noise ratio than AFA algorithm when denoising speech signals of white Gaussian noise. The authors in [10] compared the performance of least mean square and recursive least square algorithms in adaptive systems for noise reduction in radio (mobile) communication network. Number of samples equal to 50,000 samples was used for each algorithm. Results show that both algorithms reduced the noise. In [11] the authors demonstrated the performance of LMS, NLMS and RLS algorithms in the cancellation of noise in speech signals. They obtained a real time speech signal from Hindi Speech Database as a test sample and added a car noise from NOISE-92 database to it to constitute the noisy version of the speech signal. In the case of LMS a step size of 0.007 was used whereas for the NLMS a step size of 0.5 was used. Also for the RLS a forgetting factor of 0.9 and initializing constant of 0.3 were used. In each of the three algorithms the authors considered filter lengths of 50, 100, 200 and 1000

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and determined in each case the mean square error, time of convergence and complexity of execution. According to the authors results revealed that LMS is least stable unlike NLMS which is just stable and RLS which is highly stable. Also, convergence time taken by LMS is the highest and reduces in NLMS and RLS with that of RLS being the least. The result further revealed that the RLS performed best out of the three algorithms in terms of the noise level reduction and minimization of the mean square error though at the expense of highest computational complexity. No researcher has demonstrated the use of RLS algorithm on speech signal of windows media audio (.wma) format which is a double column vector. In this paper, reducing additive white Gaussian noise in speech signal of windows media audio (.wma) format with RLS based adaptive filter is proposed. Four properties which include listening, signal morphology, frequency distribution and the attenuation strength of the filter are used in evaluating the performance of the filter. The matlab code for the simulation is provided in this paper.

DESIGN OF RLS BASED ADAPTIVE FILTER

The sampling frequency of the recorded voice signal is 44.10 kHz. The parameters for the RLS based filter design are as follows:

Filter order L=32
Forgetting factor λ=1.0
Constant for initializing P0 is δ=0.5
Inverse covariance of the input signal=P0
P0=δ^{-1}P1, where P1 identity matrix of rank L+1.

With the above parameters the object of the filter is created in matlab platform as in (1). Based on the object the instantaneous responses of the filter including impulse response, magnitude response, phase response and z domain response are generated as shown fig.1, fig. 2, fig. 3 and fig. 4 respectively.

ha=adaptfilt.rls(L+1,lam,P0); (1)

where the parameters are as explained above.

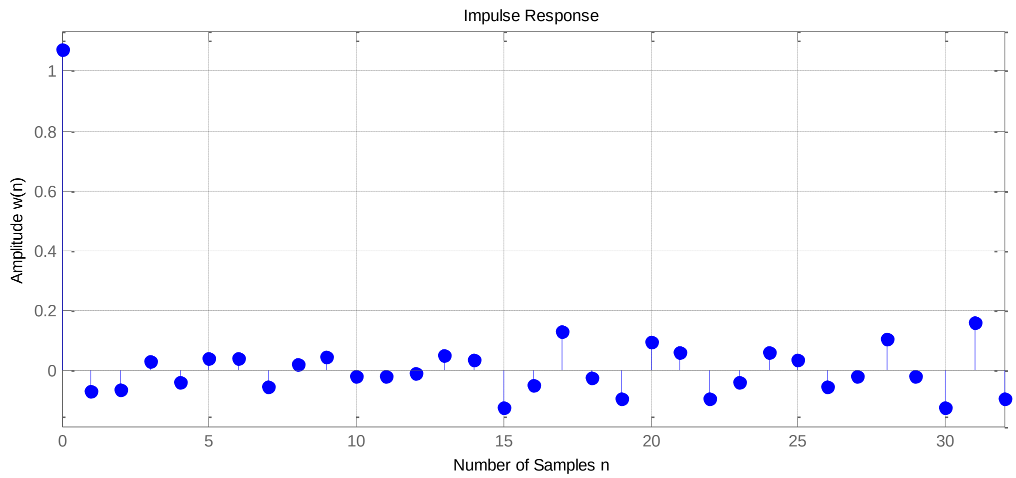

Fig. 1: Impulse Response of the Filter

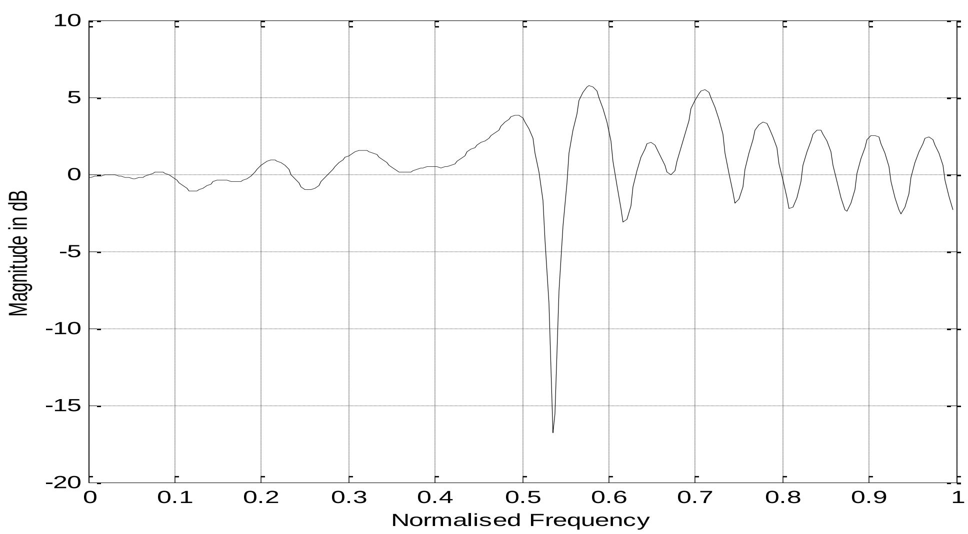

Fig. 2: Magnitude Response of the Filter

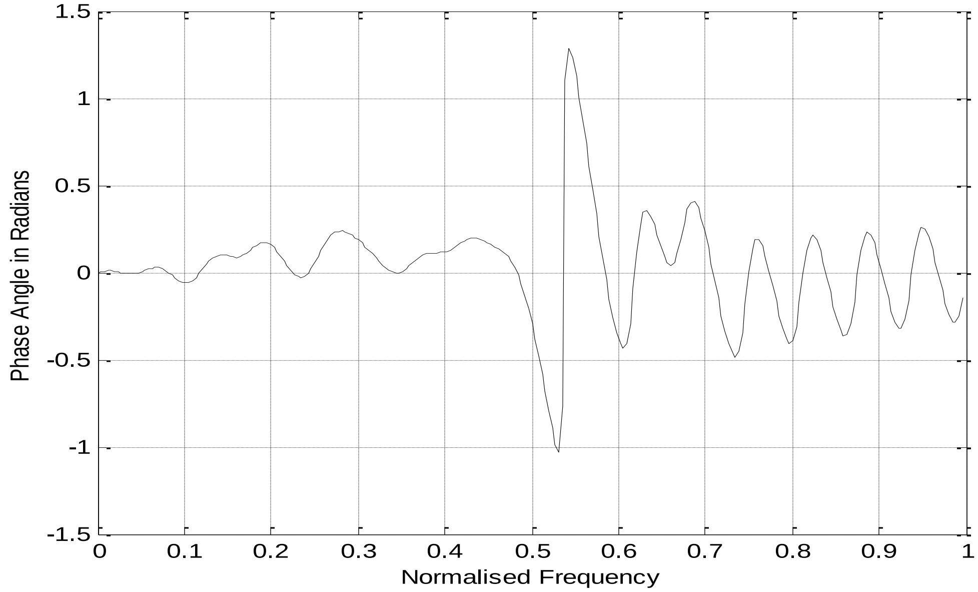

Fig. 3: Phase Response of the Filte

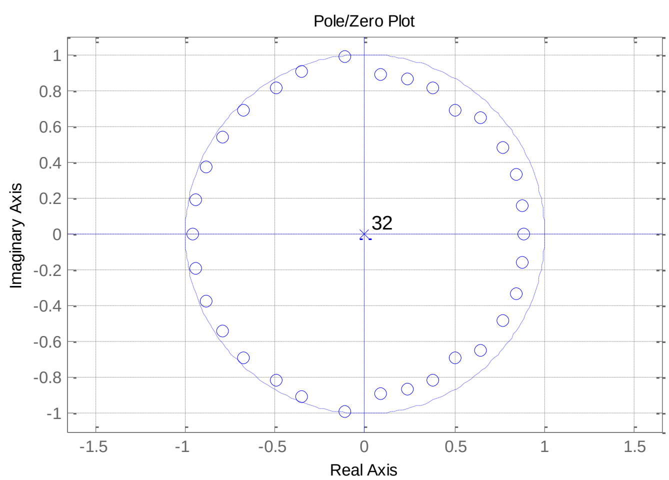

Fig. 4: z domain Response of the Filter

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The impulse response shows that the designed filter may not be highly stable because even though the response reduces in amplitude from origin it does not exhibit continuous gradual reduction to the end of the response as can be seen from fig. 1. The magnitude response is also showing some degree of sustained oscillations which implies compromised stability. The phase response shows the filter is not very linear. On the other hand, in fig.4 the poles and zeros are confined within a unit circle and the zeros are in alignment but this is not maintained in all trials in this work. These properties imply that the RLS based filter may have stability and linearity issues when denoising speech signal of .wma format of additive white Gaussian noise, which to some extent agrees with the view of the researchers in [9].

RESULT

A real voice statement "Recursive Adaptive Filtering Algorithm is Very Efficient in the Processing of Speech Signals" is converted to speech signal with a microphone and stored in a file of a computer system. With "audioread" command the stored speech signal is loaded into a matlab edit window as original speech signal. A 10.5dB additive white Gaussian noise component is generated with matlab and added to the signal to constitute a corrupt or contaminated speech signal. Fig. 5 depicts the original speech signal whereas the corrupt speech signal is depicted as fig. 6. The corrupt speech signal is thereafter used as input to the designed adaptive filter and outputs recorded. Fig. 7 indicates the filtered speech signal and fig. 8, the noise estimate in the corrupt speech signal. Also generated for purposes of result evaluation are the frequency distributions of the signals obtained by taking the fast Fourier transform (FFT) of the original, corrupt and filtered speech signals, represented as fig. 9, fig. 10 and fig. 11 respectively, as well as the power spectral densities of the original, corrupt and filtered speech signals as depicted in fig. 12, fig. 13 and fig. 14 respectively.

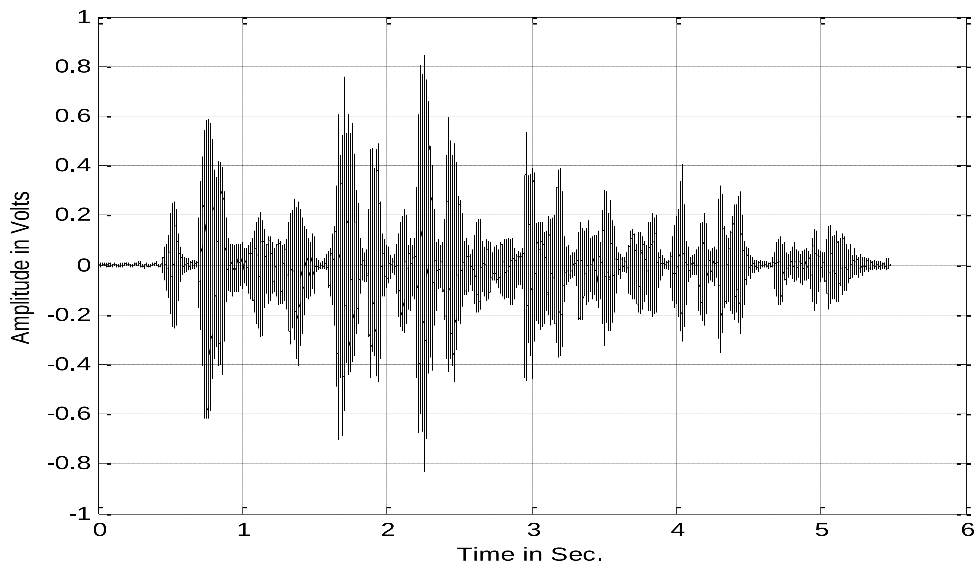

Fig. 5: Original Speech Signal

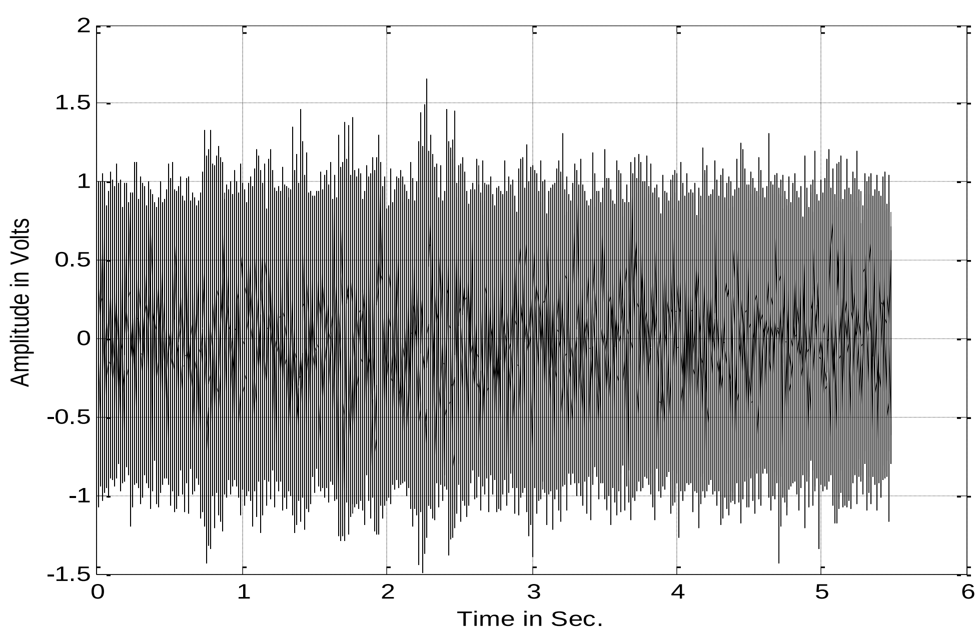

Fig. 6: Contaminated Speech Signal

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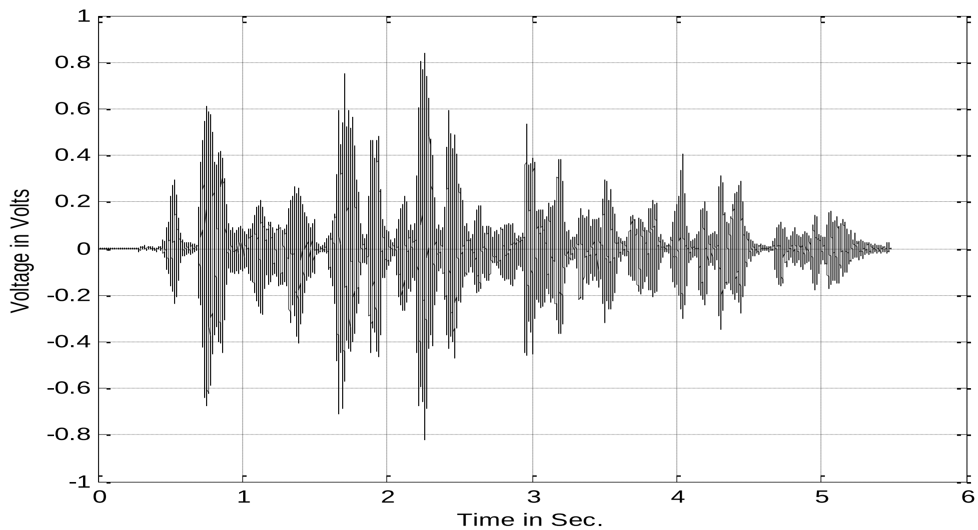

Fig. 7: Filtered Speech Signal

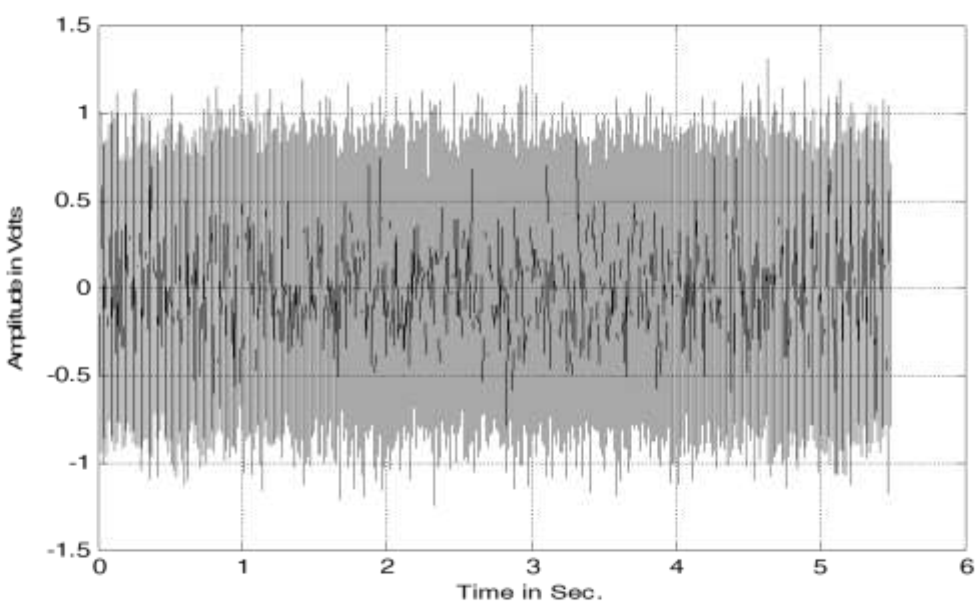

Fig. 8: Estimated Noise in the Corrupt Speech Signal

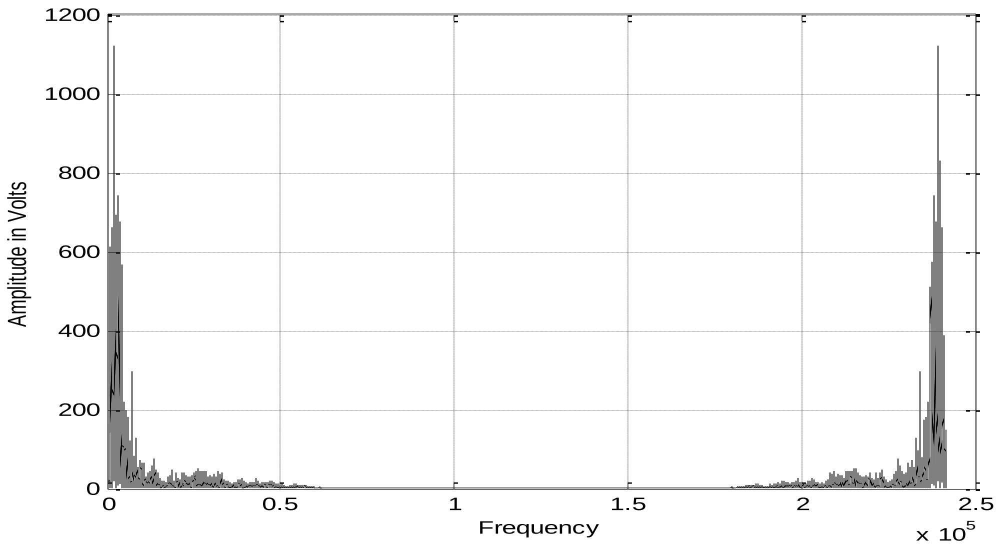

Fig. 9: Frequency Distribution of Original Speech Signal

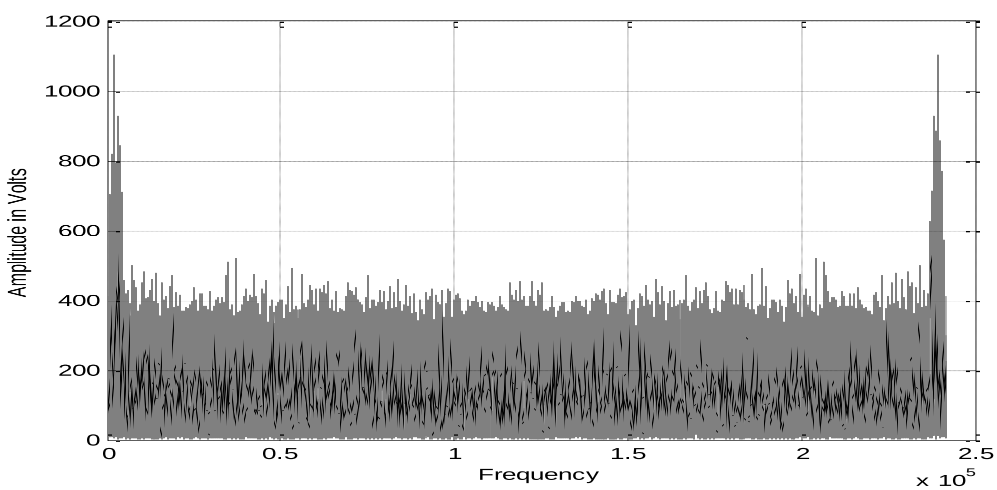

Fig. 10: Frequency Distribution of Contaminated Speech Signal

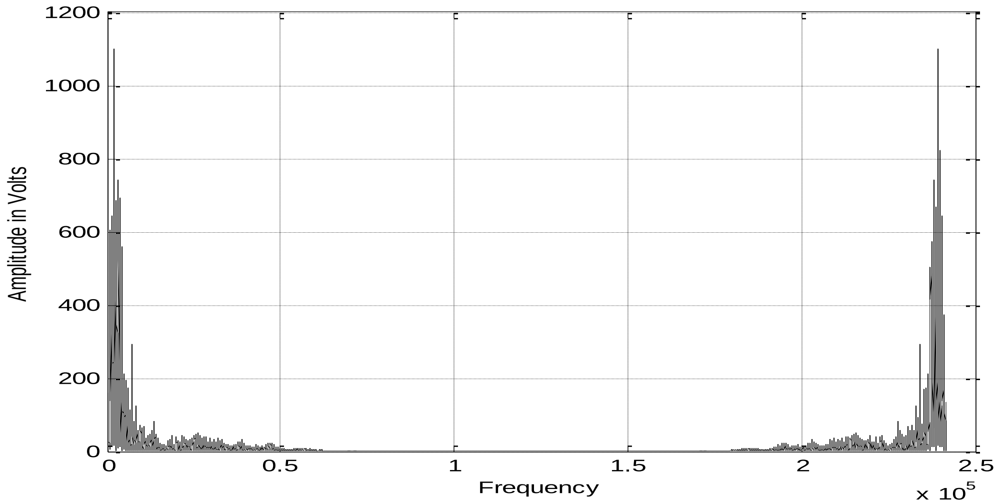

Fig. 11: Frequency Distribution of Filtered Speech Signal

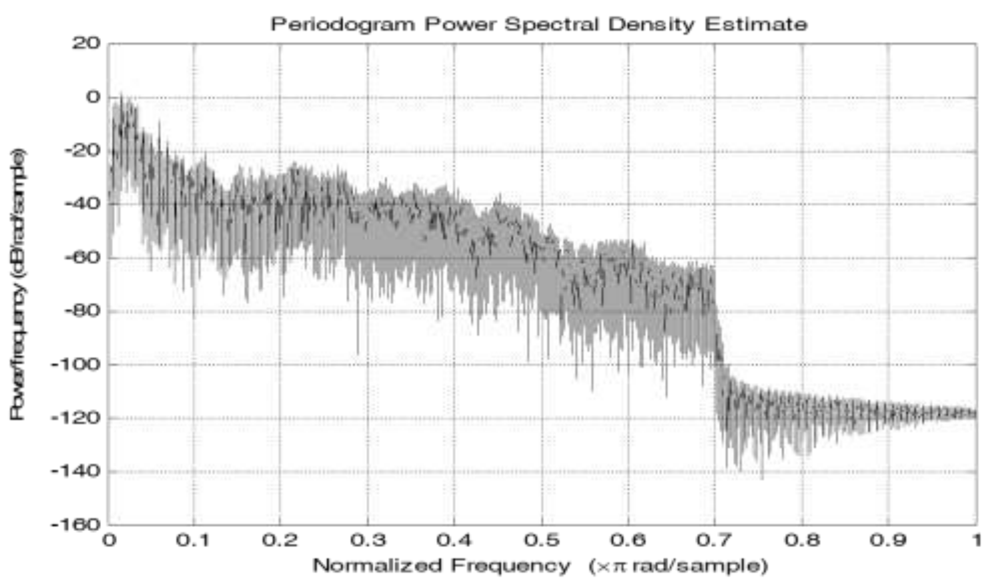

Fig. 12: Power Spectral Density of Original Speech Signal

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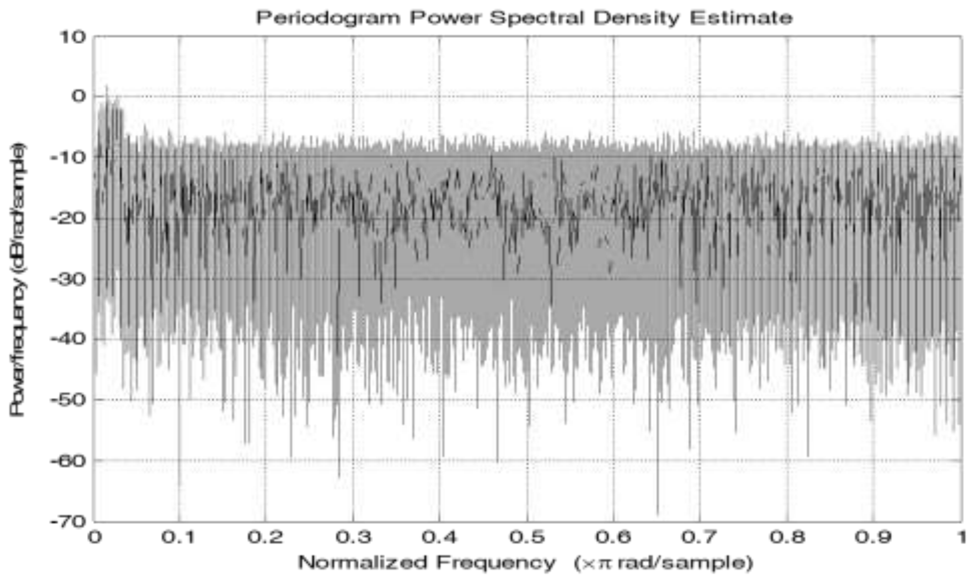

Fig. 13: Power Spectral Density of Corrupt Speech Signal

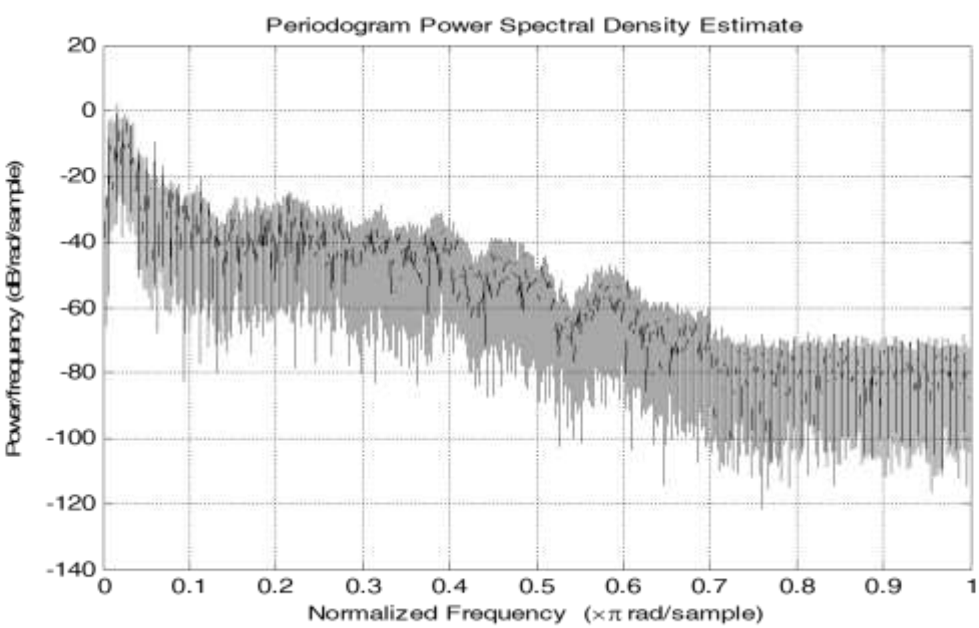

Fig. 14: Power Spectral Density of Filtered Speech Signal

EVALUATION OF RESULTS

We used four properties to evaluate the results of this work; (a) listening to both input and out sounds (b) observing the signal morphologies (c) observing the frequency distribution of the signals and (d) attenuation strength of the filter. In (a) by listening to the original speech signal, the corrupt speech signal and the filtered speech signal from a sound speaker reveals that the filtered speech signal is as clear as the original speech signal and conveys the same message content, whereas the contaminated speech signal is very noisy, and this implies that the adaptive filter drastically reduced the noise component in the corrupt speech signal.

In (b), looking at the appearances of the original and filtered speech signals it can be seen that they are the same, which implies that the filter did not alter the filtered speech signal.
In (c) the frequency distribution of the original, corrupt and filtered speech signals are generated by performing descrete Fourier transforms on them using the formula (2) [12]-[14]

$$X(K) = \sum_{n=0}^{N-1} x(n) e^{-j2\pi nk/N} \qquad (2)$$

Where N is the number of samples, and n and k are integer values that vary from a reference point to equivalent number of samples in N. That is, n and k can vary from 0 to N-1, 1 to N, 2 to N+1, etc.

The matlab code for the transform is as in (3) [15]

$$Y = fft(X, N); \qquad (3)$$

where X is the function and N, the number of samples desired. From figures 9, 10 and 11 above it can be seen that the original and filtered signals have the same frequency distributions devoid of noise components whereas the frequency distribution of the corrupt signal displays much noise content.
In (d) the attenuation strength of the filter is obtained from the power spectral densities of the original, contaminated and filtered speech signals considered at six different frequencies in order to appreciate the degree of attenuation of the white Gaussian noise by the filter and the noise proportion still present in the filtered signal. Table 1 shows the signal power magnitudes at the six different normalised frequencies including 0.1, 0.3, 0.5, 0.7, 0.8 and 0.9, as obtained from signal power spectral densities of fig. 12, fig. 13 and fig. 14. The noise power present in the filtered speech signal is calculated from (4) [16], [17]

$$N_0 = S - S_F \qquad (4)$$

where S is the power of the original signal and S_F, the power of the filtered voice signal.

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By comparing the power magnitudes for the original, contaminated and filtered speech signals at the six frequency positions in table 1, it clearly reveals that the adaptive filter drastically attenuated the noise at each of the frequencies. The remaining noise power not attenuated in the filtered signal is shown in the table which in each case is very small enough to translate to high signal to noise ratio for the filtered signal. Notice that once the filtered signal attenuation level is more than the power spectral density of the original signal the noise power present in the filtered signal at the concerned specific frequency is equal to zero. This is because no noise is expected to remain in the filtered signal under such circumstance. Notice also that the contaminated signal power at each of the six frequency positions is very low; meaning that noise present in them is high. The noise power present in the filtered signal from frequency of 0.8 and above cannot disturb the filtered speech signal because they are not part of the active speech section during recording. It is the dummy section of the speech.

Table 1: Signal Power at Different Frequencies

Normalised Frequency	0.1	0.3	0.5	0.7	0.8	0.9
Original Signal Power in dB (S)	-22.72	-49.94	-43.84	-63.52	-111	-113.80
Contaminated Signal Power in dB	-7.035	-6.56	-7.132	-7.619	-7.438	-7.194
Filtered Signal power in dB (S_F)	-25.02	-33.55	-44.95	-60.00	-71.19	-69.10
Noise Power Present in the Filtered Signal in dB (N_0)=S-S_F	0.0	-0.78	-4.99	-3.52	-39.81	-44.70

CONCLUSION

From table 1, it can be deduced that RLS adaptive filtering algorithm is very efficient when it is used in finite impulse response filter to reduce additive white Gaussian noise in speech signals. The optimum values of the variable parameters such as filter order, forgetting factor, initializing constant and the inverse covariance matrix depend on the type and magnitude of noise to be reduced as well as the format of the speech signal. The noise power present in the filtered signal from frequency of 0.8 and above cannot disturb the filtered speech signal because they are not part of the active speech section during recording. The frequency distribution of the signals clearly reveals that the original and filtered signals have no significant difference which means that RLS-based adaptive filter is very effective here. However the instantaneous responses of the filter reveal that the RLS algorithm has stability and linearity issues when it is used in driving FIR filters.

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APPENDIX

MATLAB PROGRAM

```
clc,clear
[y,Fs]=audioread('C:\Users\Test\Desktop\MYVOICE.wma');%loads speech signal  into workspace
sound(y,Fs)
pause(10)
d=awgn(y,10.5);%Corrupts speech with 10.5dB AWGN
sound(d,Fs)
k=241580;%Length of the speech signal=number of samples=number of iterations
t=1/Fs:1/Fs:1/Fs*k;%Time range and incremental value
figure(1)
plot(t,y,'k');%Plots original speech signal in black colour
grid on
ylabel('Amplitude in Volts')
xlabel('Time in Sec.')
figure(2)
plot(t,d,'k');%Plots corrupt speech signal in black colour
grid on
ylabel('Amplitude in Volts')
xlabel('Time in Sec.')
L=32;%Order of the filter
q=0.5;%Initialisation constant for inverse covariance matrix
P0=q^-1*eye(L+1);%Inverse of the input signal covariance matrix
lam=1.0;%Forgetting factor for the adaptation process
ha=adaptfilt.rls(L+1,lam,P0);%Creates the adaptive filter object
[y1,e]=filter(ha,y(:,2),d(:,2));%Filters the corrupt speech signal using second column vectors
sound(y1,Fs)
[h,w]=freqz(ha,256);%Returns 256 samples of filter vector
HdB=20*log10(abs(h));%Magnitude response of the filter
Phaseangle=unwrap(angle(h));%Phase response of the filter
impz(ha);%Plots the instantaneous impulse response of the filter
```

```
grid on
ylabel('Amplitude w(n)')
xlabel('Number of Samples n')
figure(4)
plot(w/pi,HdB,'k')%Plots the instantaneous magnitude response of the filter
grid on
ylabel('Magnitude in dB')
xlabel('Normalised Frequency')
figure(5)
plot(w/pi,Phaseangle,'k')%Plots the instantaneous phase response of the filter
grid on
ylabel('Phase Angle in Radians')
xlabel('Normalised Frequency')
zplane(ha);%Plots the pole-zero response of the filter
grid on
ylabel('Imaginary Axis')
xlabel('Real Axis')
figure(7)
plot(t,y1,'k')%Plots filtered speech signal
grid on
ylabel('Voltage in Volts')
xlabel('Time in Sec.')
figure(8)
plot(t,e);%Plots estimated noise
grid on
ylabel('Amplitude in Volts')
xlabel('Time in Sec.')
figure(9)
periodogram(y(:,2))%Power spectral density of the original speech signal
figure(10)
periodogram(d(:,2))%Power spectral density of the corrupt speech signal
figure(11)
periodogram(y1)%Power spectral density of the filtered speech signal
n=1:k;%Ploting range for frequency distribution
x1=fft(y,k);%Transforms original speech signal to frequency domain
x2=fft(d,k);%Transforms corrupt speech signal to frequency domain
x3=fft(y1,k);%Transforms filtered speech signal to frequency domain
figure(12)
plot(n,abs(x1),'k')%Plots the frequency distribution of the original speech signal
grid on
ylabel('Amplitude in Volts')
xlabel('Frequency')
figure(13)
plot(n,abs(x2),'k') )%Plots the frequency distribution of the corrupt speech signal
```

```
grid on
ylabel('Amplitude in Volts')
xlabel('Frequency')
figure(14)
plot(n,abs(x3),'k') )%Plots the frequency distribution of the filtered speech signal
grid on)
ylabel('Amplitude in Volts')
xlabel('Frequency')
```