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Improvement of Audio Signal Quality Using Adaptive Filtering and It's Performance Advantage over Non - Adaptive (Linear) Filtering

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Abstract: Over the years in signal processing, noise has been a crucial factor that is frequently faced in audio signals transmission and a communication system at large. It is identified to be an unwanted disturbance or wave that tends to disturb the transmission and processing of signals in a communication system. This problem has gained vast consideration due to the essence of a noise-free output signal in several communication systems. The implementation of adaptive methods or techniques in noise cancellation is ways of estimating signals corrupted by additive noise and deduct the unwanted signal from the corrupted signal. Adaptive filters now has many areas of applications such as in the field of signal processing, speech recognition, image processing, medical signal processing, biomedical, radar, sonar, communications, etc. This work investigates the performance advantages of adaptive over non-adaptive filtering techniques in the process of filtering audio signals and improvement of its quality by reviewing various adaptive algorithms more precisely the Recursive Least Square (RLS) and its implementation.

Keywords: Adaptive Filter, Noise, Audio Signals, RLS Algorithm

1. Introduction

In the general sense, a received speech signal contains some form of noise constituent. The noise may be due to the limited accuracy involved in coding the transmitted waveform, or due to the addition of acoustically coupled background noise. Despite the apparent randomness of these signals, however, in most cases they are usually carrying useful information concerning the physical variations. The major task in digital signal processing is then to extract the desired information from discrete random signals. This often requires developing estimated techniques for smoothening, filtering, or predicting, as well as parameter estimation of signal generating models such as autoregressive models [1]. Under such circumstances, there is a need to design more robust filters, such as adaptive filters, to track the changes of signal and noise.

An adaptive filter is a specific type of interference cancellation device that relies on the use of noise cancellation by subtracting noise from a received signal, an operation controlled in an adaptive manner for the purpose of improving signal to noise ratio [2-3]. The so-called adaptive filter, is the use of the result of the filter parameters a moment ago, automatically adjust the filter parameters of the present moment, to adapt to the unknown signal and noise, or over time changing statistical properties, in order to achieve optimal filtering [4]. Adaptive filter has "self-regulation" and "tracking" capacities. An adaptive filter can automatically adjusts its own parameters responding to an error signal dependent on the filter output; therefore, it is capable of removing the noise satisfactorily [5]. The parameters (usually refer to filter coefficients) of an adaptive filter are regularly arbitrarily adjusted or based on the priori knowledge accessible to the system, and they are adjusted each time the new input sample is available. Adaptive filter is a device which uses the filter parameters of a moment ago to automatically adjust the filter parameters of the present moment in order to adapt to the statistical properties that the signal and noise produces and to achieve optimal filtering [6].

Works based on filtering are centered on the pioneering works of Wiener and was extended and enhanced by Kalman, Bucy and others [7-9]. The most important property of adaptive filter is that it can work effectively in an unknown environment, and track the input signal of time-varying characteristics [10]. Adaptive filter has been widely used in communications, control and many other systems. To filter out noise usually means that noise is curbed in the contaminated signal through the filtering while the signal remains relatively unchanged. Also, lessening of noise and distortion removal in certain areas of applications such as speech recognition, image processing, medical signal processing, radar, sonar, and any other application where the preferred signals cannot be accessible from noise and distortion.

The major problem faced in audio signals is removing the unwanted signal components, i.e. noise from a received signal has been the subject of several researches. Noisy received signals prevent accurate interpretations of such signals due to degradation in the signal quality. Non-adaptive filter requires that their impulse response (a set of parameters for discrete-time filters) are adjusted iteratively as data flow through the filters which could be tasking and consumes time whereas the adaptive filters can track the changes in the statistics of the signals or parameters of time-varying systems. The inability of non-adaptive algorithm or systems to operate when limited information is known makes it less suitable in general application.

1.1. Description of Adaptive Filters

Adaptive filters represent a significant part of the subject of statistical signal processing upon which they are founded. Historically, the parametric approach has been the main engineering approach to signal processing and is based on a priori models derived from scientific knowledge about the problem. At the other extreme, the alternative nonparametric approach is based on the use of more general models trained to replicate desired behavior using statistical information from representative data sets. Adaptive filters are actually based on an approach which is somewhere in between these two extremes. When a priori knowledge of a dynamic process and its statistics is limited then the use of adaptive filters can offer performance improvements over the more conventional parametrically based filter designs. Furthermore, they can offer other signal processing benefits that would not be possible otherwise. Consequently, adaptive filters have found application in diverse fields including communications, controls, robotics, sonar, radar, seismology and biomedical engineering to name but a few.

1.2. Recursive Least Square Algorithm (RLS)

RLS algorithm is based on the well-known least squares method. For finding the best fitting curve to a given set of data points, mathematical procedure of least squares method is used [11]. In order to minimizing the sum of the squares of the offsets of the points from the curve, RLS algorithm is used to solve the least squares problem [12]. This is one of the multiple regression methods i.e. iterative method in which with respect to the input signal, the output signal is measured at different instants of time Y_p whereas the input is given as$X_p(i)$.

$$Y_{p=\sum_{i=0}^{Q=1} W(i) X_p(I)+E_p}$$

Here E represents the error signal.

In this algorithm, $W(i)$are updates continuously with each set of new data without solving matrix inversion. In Least Mean Square (LMS) Algorithm, we use expectations whereas in Recursive Least Square (RLS) algorithm, we use time averaging.

2. Methods

2.1. Design Method for Audio Transmission Using Linear Filters

In this paper, audio signal is used to evaluate the performance comparison of linear filter and adaptive filter. The system model used for the simulation of the audio transmission using linear filters is shown in Figure 1. The system consists of audio signal input, modulator, transmit filter (rectangular pulse shaping), additive white Gaussian noise (AWGN) generator, receive filter (integrate and dump), demodulator and audio signal output. Each stage is described as follows:

(a) Audio signal input: The audio signal to be transmitted is acquired through the use of microphone. The microphone signal acquisition is done in the MATLAB environment using MATLAB functions audio recorder and get audio data. The audio signal is converted into double-type numerical data using function wav read. Then the data is transformed into symbols by 8 bit conversion.

(b) Modulate signal with BPSK: The audio symbols are modulated using the binary phase shift keying (BPSK) modulation. The modulated signal is needed for proper transmission over the communication channel.

(c) Rectangular pulse shaping filter: This is a linear filter used at the transmitter to remove unwanted audio signal components.

(d) AWGN: Additive white Gaussian noise (AWGN) generator produces the noise that corrupts the transmitted audio signal. This is achieved by the awgn MATLAB function.

(e) Integrate and dump filter: This is a linear filter at the receiver end of the system. It is used to remove the unwanted components of the received signal.

(f) Demodulate signal with BPSK: The filtered audio signal received is then demodulated back into the original data symbols suitable for hearing.

(g) Audio signal output: This is the earphone that helps to evaluate the audio output quality.

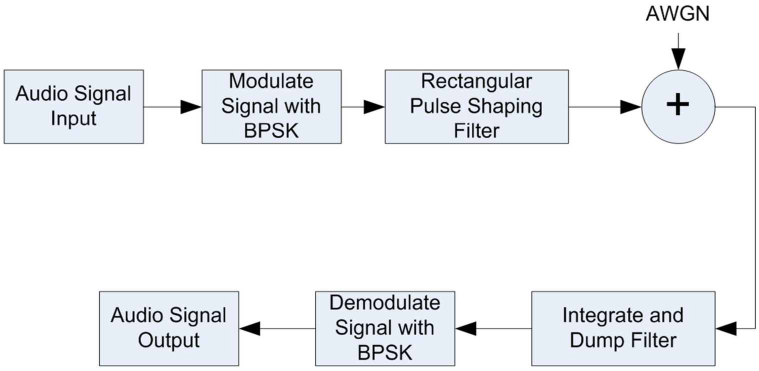

Figure 1. *System model of the audio transmission using Linear Filters.*

2.2. Design of Audio Transmission Using Adaptive Filters

The system model used for the simulation of the audio transmission using adaptive filter is shown in Figure 2. The system consists of audio signal input, modulator, transmit filter (simple Low Pass), additive white Gaussian noise (AWGN) generator, receive filter (RLS adaptive filter), decision block, demodulator and audio signal output. Each stage is as described in section IV above. The Low Pass filter is a Chebyshev 31st order low-pass Finite Impulse Response (FIR) filter. The adaptive filter used is the Recursive Least Square (RLS). The RLS filter uses the error calculated after filtering to determine whether to apply filtering or to stop.

The RLS filtering process is repeated until an acceptable error value is obtained. This helps to remove the noise in the received audio signal as good as possible.

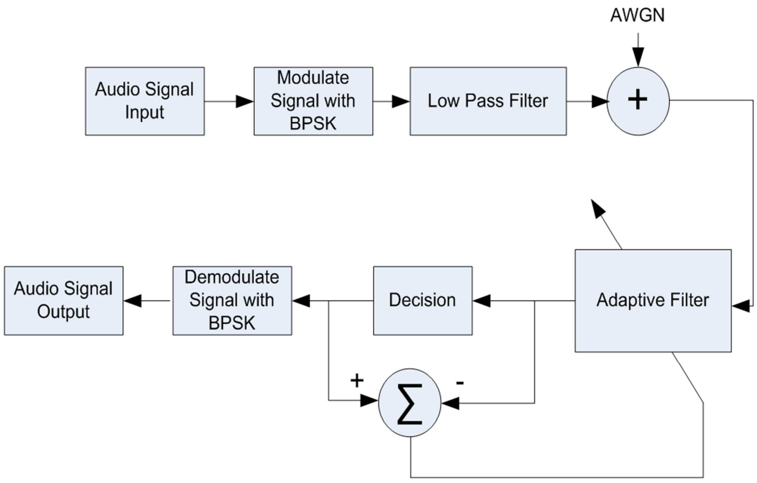

Figure 2. *System simulation model of audio transmission using Adaptive Filters.*

Table 1. *Simulation Parameters.*

Parameter	Specification
Noise	Additive White Gaussian Noise (AWGN)
Modulation	BPSK
Audio format	Wav
Linear filter types	Rectangular pulse (transmitter) and integrate and dump (receiver)
Adaptive filter type	RLS
Audio signal sampling frequency	

3. Results

3.1. Filtering Performance on the Graphic User Interface (GUI)

Figure 3 below shows the GUI for the simulation of audio signal transmitted at a sampling frequency of 8000 Hz. The GUI consists of the Transmitted Signal button, which when pressed gives the original sound of the input signal, the AWGN button gives the sound of the distorted input signal, the Linear Filter button gives the sound output of the received signal filtered by linear filters and lastly, the RLS Adaptive Filter button gives the sound output of the received signal filtered by RLS adaptive filters.

The waveform of the original or transmitted audio signal is shown in Figure 4. The waveform depicts the variation in the audio signal amplitude with respect to the sampling frequency.

Figure 5 represents the audio signal output and the addition of interfering noise. It was observed that the noise corrupted the original signal thus making it unintelligible to the ear. This effect can be observed from the figure as it can be seen that the waveform is very different from the waveform of the original audio signal of Figure 4.

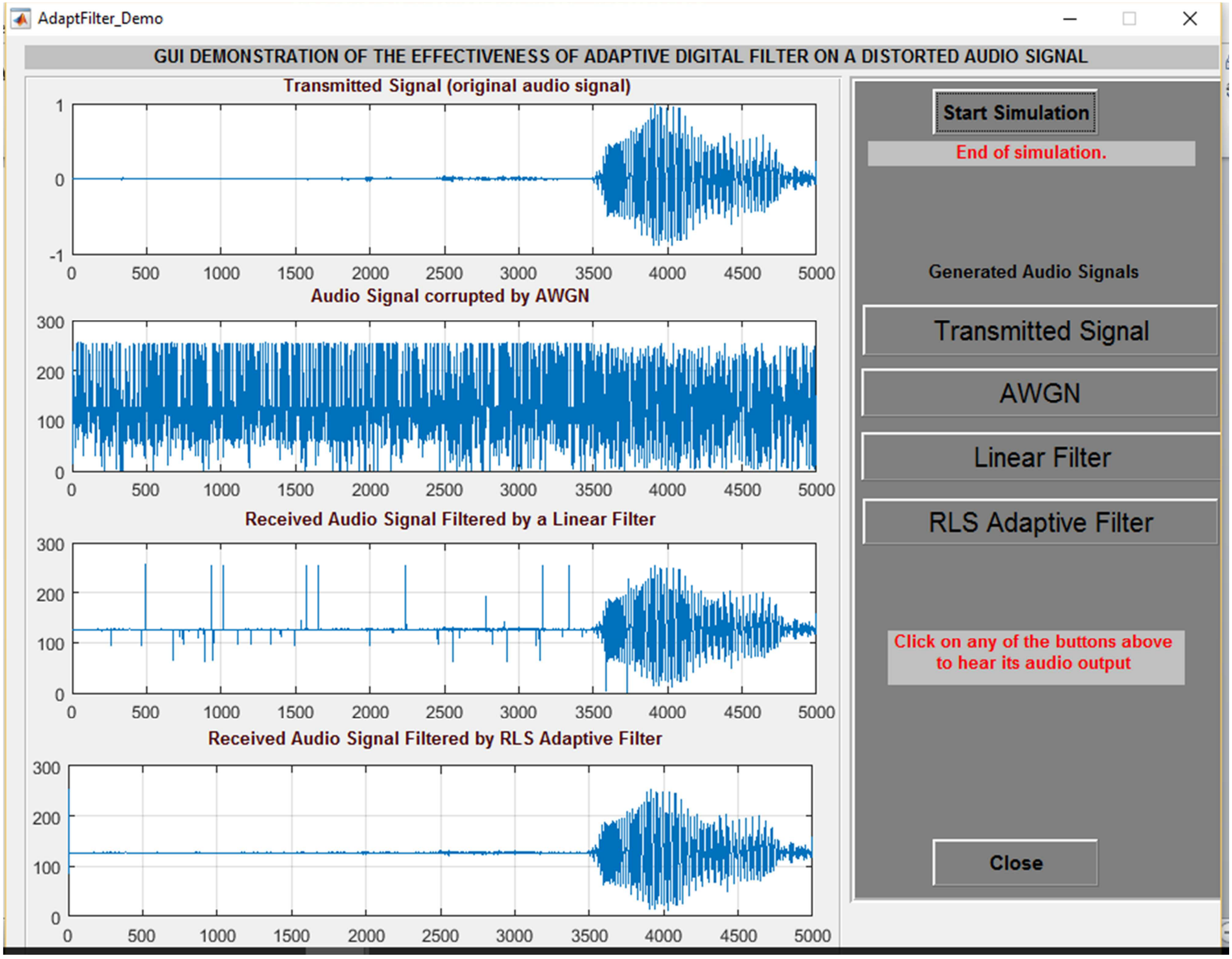

Figure 3. *GUI for the simulation of audio transmission.*

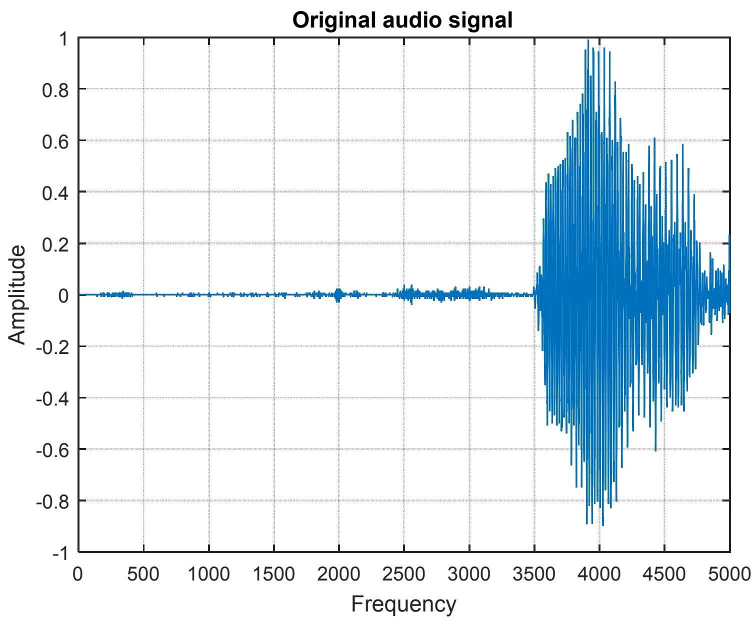

Figure 4. *Waveform of the transmitted audio signal.*

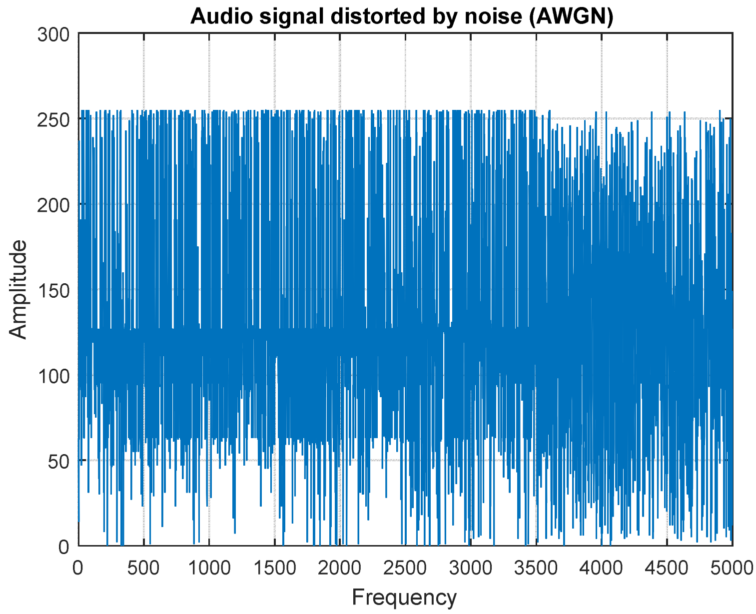

Figure 5. *Waveform of the received audio signal corrupted by interfering noise.*

3.2. Performance of Linear Filters

The audio signal output obtained from the output of the linear filter is shown in Figure 6. It was observed that the linear filter filtered some of the noise components but more noise components were still observed from the received audio signal. The effect of some of the noise components still remaining in the received signal can be seen from the waveform of Figure 6. as it shows some variations in the amplitude as compared to the original or transmitted audio signal.

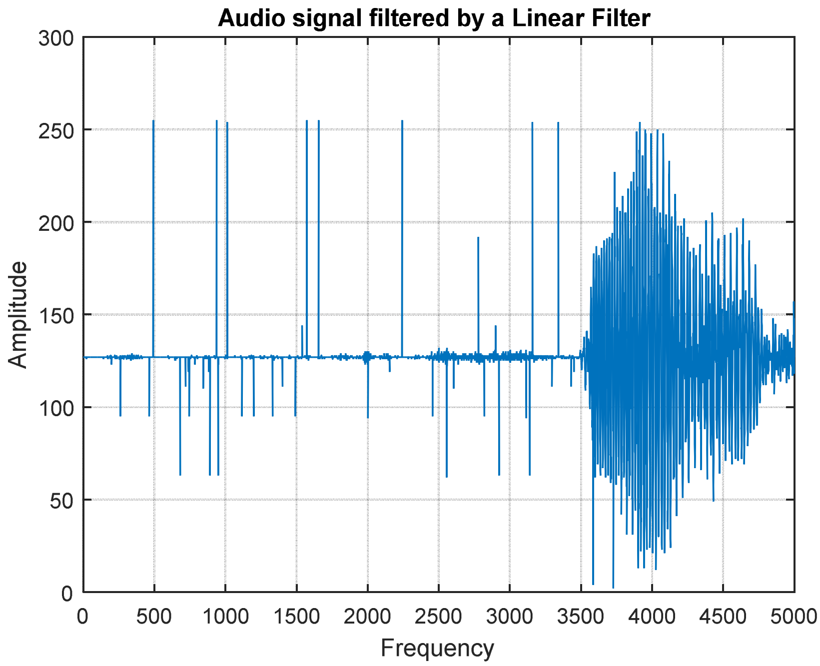

Figure 6. *Waveform of the received audio signal filtered by linear filters.*

3.3. Performance of RLS Adaptive Filter

The result of the use of RLS adaptive filter to cancel the noise picked up during the transmission of the original audio signal is shown in Figure 7. It was observed from the received audio signal quality that the adaptive filter drastically cancelled out more noise components than the linear filter. The resulting waveform of Figure 7 shows the audio output of the adaptive filter which is very much similar to that of the original audio signal. The only amplitude variation observed is found between about 0 to 5 Hz. This filter used the recursive least square algorithm (RLS) to cancel the noise.

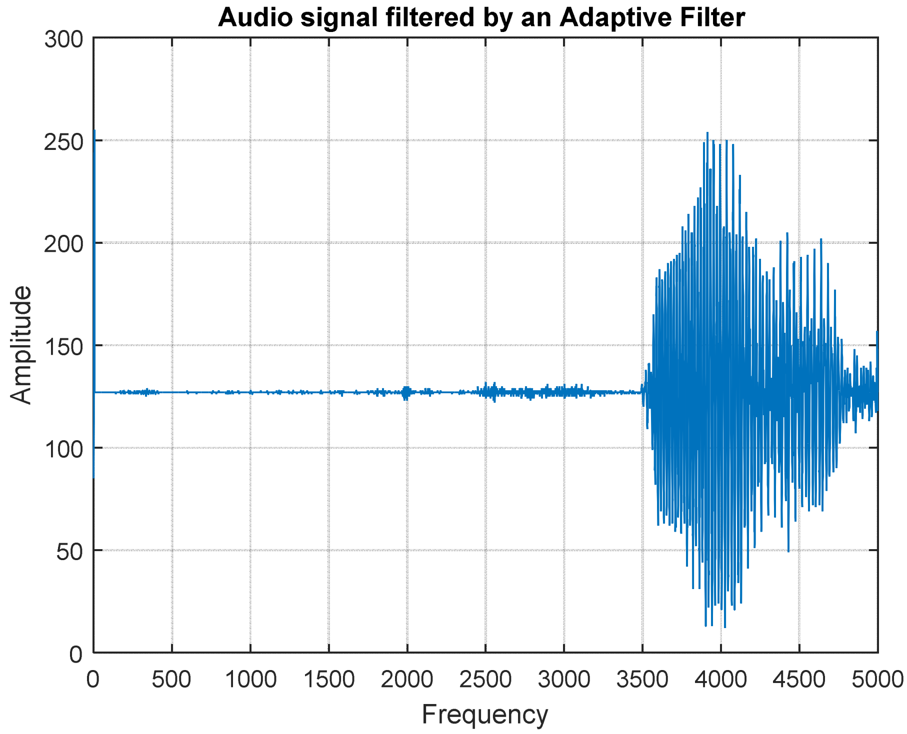

Figure 7. *Waveform of the received audio signal filtered by adaptive filters.*

4. Conclusion

Noise as an important factor which causes distortions or interference to audio signals in communication systems was studied in this work. The performance of the linear (or non-adaptive) filtering technique was compared with that of RLS adaptive filtering technique using an audio/speech signal. The investigation was carried out in the MATLAB software environment. The speech signal was acquired and converted into digital form using signal processing functions in MATLAB. Noise was generated and added to the speech signal to form a corrupted or noisy version of the speech signal. Furthermore, both the linear and RLS filtering algorithms were implemented using the MATLAB programming language, and are then applied to the noisy speech signal separately for the purpose of cancelling out the noise from the corrupted speech signal. The results obtained from the study show that for the same noisy audio signal, the linear filtering technique was able to eliminate some noise components from the noisy signal as was observed from the waveform of the filtered signal. However, there exist more feasible noise components in the filtered signal. On the other hand, applying the RLS adaptive filtering technique to similar noisy speech signal gives better filtered signal waveform that is very close to the original or noiseless speech signal. Furthermore, by audible evaluation of the filtered speech signals, the RLS output gives better audio quality that is very similar to the original speech signal, but the output of the linear filtered signal contains some unwanted audible components. From the performance of the two filtering techniques in terms of both the output waveform and audio quality, the adaptive filtering technique performs better than the linear or non-adaptive filtering technique.

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