A Novel RLS Based Adaptive Filtering Method for Speech Enhancement

Pogula Rakesh, T. Kishore Kumar

***Abstract*—**Speech enhancement is a long standing problem with numerous applications like teleconferencing, VoIP, hearing aids and speech recognition. The motivation behind this research work is to obtain a clean speech signal of higher quality by applying the optimal noise cancellation technique. Real-time adaptive filtering algorithms seem to be the best candidate among all categories of the speech enhancement methods. In this paper, we propose a speech enhancement method based on Recursive Least Squares (RLS) adaptive filter of speech signals. Experiments were performed on noisy data which was prepared by adding AWGN, Babble and Pink noise to clean speech samples at -5dB, 0dB, 5dB and 10dB SNR levels. We then compare the noise cancellation performance of proposed RLS algorithm with existing NLMS algorithm in terms of Mean Squared Error (MSE), Signal to Noise ratio (SNR) and SNR Loss. Based on the performance evaluation, the proposed RLS algorithm was found to be a better optimal noise cancellation technique for speech signals.

***Keywords*—**Adaptive filter, Adaptive Noise Canceller, Mean Squared Error, Noise reduction, NLMS, RLS, SNR, SNR Loss.

I. Introduction

NOISE is any unwanted disturbance present in a signal and it is normally corrupting the signal of interest and therefore extracting the reliable information from the corrupted signal is always important in any signal processing application. Noise cancellation can be used in areas where noise can be harmful to ones hearing, such as: engine rooms or aircraft runways. The effect of interfering noise in speech is considered as the severe problem in speech signal processing.

Speech signals are often corrupted by various types of degradations. The most common degradations include background noise, reverberation and speech from competing speaker(s). Degraded speech offers poor perceptual quality and poor intelligibility that leads to listener fatigue and degraded performance in tasks like speech and speaker recognition respectively [1].

The aim of speech enhancement is to improve speech quality and intelligibility of degraded speech signal using speech processing techniques. The term intelligibility is related to the amount of speech content that is recognized correctly and quality is related to the aspect of the speech that determines the ease with which one can understand the speech. In many speech communication systems, the presence of background interference causes the quality or intelligibility of speech to degrade. Hence, the signal has to be cleaned up with noise cancellation technique before it is stored, analyzed, transmitted, or processed [2]. The study of cancelling noise from a wanted signal arises from need to achieve stronger signal to noise ratios.

Several noise reduction techniques aim to suppress the noise in a noisy speech signal without distorting original (clean) signal with an underlying assumption that the system itself is ideal (i.e. system is not adding any noise to the signal by itself) and that the only environmental sources of noise are responsible for signal distortion.

In recent years, with the wide development of Digital Signal Processing (DSP) tools, adaptive filtering techniques have become standard solutions for this issue [3]. The main objective of this research work is to implement a novel RLS adaptive filtering algorithm for speech enhancement and compare its performance with existing NLMS algorithm. The NLMS and the proposed RLS algorithms are analyzed both subjectively and objectively to find out the optimal technique for speech enhancement.

This paper is structured as follows. In Section II, the concept of adaptive filtering for speech enhancement is explained. Section III describes the problem formulation and the proposed RLS filtering method. In Section IV, the experimental results are analyzed and Section V deals with the performance evaluation of the proposed RLS algorithm. Finally, the conclusion is summarized in Section VI.

II. Adaptive Filtering Techniques for Speech Enhancement

Adaptive filtering is a technique in which the parameters used for the processing of signals changes according to some predefined criterion like the estimated mean squared error or the correlation. An adaptive filter is a digital filter with coefficients that are determined and updated by an adaptive algorithm. The basic operation of adaptive filter involves two processes: [4]

1. Filtering process that produces an output signal in response to a given input signal and
2. Adaptation process that aims to adjust the filter parameters to the environment.

With this capability, overall performance and construction flexibility, the adaptive filters have been widely used in many different applications, some of the most important are: noise cancellation, signal prediction, telephonic echo cancellation, radar and biometrics signal processing.

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One of the most popular proposed solutions to reduce the signal corruption caused by unpredictable noise is the usage of adaptive filters. The different adaptive algorithms widely used for performing weight updation in an adaptive filter are: the LMS (Least Mean Squares), NLMS (Normalized Least Mean Squares) and the RLS (Recursive Least Squares) algorithm [5].

Adaptive filters are classified into two types namely linear and non linear. In Linear adaptive filters, the computation is done by estimating the desired response with the combination of the available set of observables applied to the input of the filter. Otherwise, the adaptive filter is said to be nonlinear [6]. One of the most common practical applications of adaptive filters is noise cancellation.

A. Adaptive Noise Canceller

Adaptive filter is a digital filter which is widely used as noise canceller [7]. Noise Cancellation is a method to reduce or cancel out undesirable components of the signal. It involves producing an estimate of the noise by filtering the reference input and then subtracting this noise estimate from the primary input containing both signal and noise (Fig. 1).

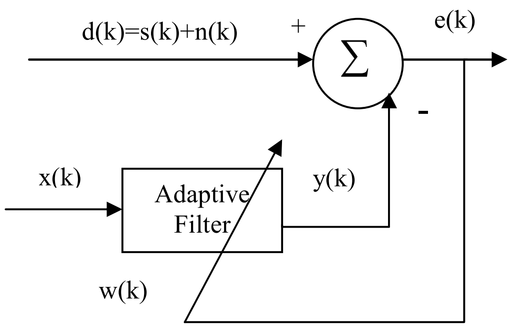

Fig. 1 Adaptive Noise Canceller

The primary input d(k) is the noisy signal consisting of both the desired signal, s(k), and the noise, n(k). It is assumed that the noise is uncorrelated with desired signal. The reference input x(k) is a measure of the distorting signal which is correlated with n(k). x(k) is processed by the adaptive filter to produce an output y(k). An estimate of the desired signal, e(k) is then obtained by subtracting the adaptive filter output, y(k), from the noisy signal, d(k).

The filter output, y(k) and the error, e(k) are given by (1) and (2):

$$y(k) = w^T(k)x(k), \tag{1}$$

$$e(k) = d(k)-y(k) \tag{2}$$

A typical performance criterion is to minimize error signal e(k) i.e., the difference between the filter output signal and a given reference (or desired) signal. Based on the metrics of the adaptive filters, the three widely used algorithms applied to the noisy signals for enhancement are explained below.

B. Least Mean Squares (LMS) Algorithm

The least-mean-squares (LMS) algorithm is widely used in adaptive signal processing for its stable and robust performance and ease of implementation. It is known for its simplicity and its good steady-state performance in stationary context [8].

The steps involved in the standard LMS algorithm are given below:

1. Initialization of filter weight coefficients vector.

$$\bar{w}(k) = [\, w_1(k)\ \ w_2(k)\ \ w_3(k) \ldots w_P(k) = 0] \tag{3}$$

where P is the filter order.

2. Calculate the adaptive filter output $\bar{y}(k)$,

$$\bar{y}(k) = \bar{w}^T(k)\,\bar{x}(k) \tag{4}$$

3. The error signal e(k) is calculated by using (5),

$$e(k) = d(k)-y(k) \tag{5}$$

4. Update the filter tap coefficients by using (6),

$$\bar{w}(k+1) = \bar{w}(k) + \mu e(k)\,\bar{x}(k) \tag{6}$$

where μ is defined as the step-size parameter which affects the convergence behavior of the filter weights.

C. Normalized Least Mean Squares (NLMS) Algorithm

The normalized LMS (NLMS) algorithm is an extension of the standard LMS algorithm [9]. From one iteration to the next, the weight vector of an adaptive filter should be changed in a minimal manner [6]. The NLMS algorithm is a time varying step-size algorithm, calculating the convergence factor μ as:

$$\mu(k) = \frac{\alpha}{c+\|x(k)\|^2}$$

Here α is the adaptation constant, which optimizes the convergence rate of the algorithm

Range of alpha is: $0<\alpha<2$, c is the constant term for normalization and always $c< 1$.

The NLMS algorithm updates the coefficients of an adaptive filter by using

$$\bar{w}(k+1) = \bar{w}(k) + \frac{\alpha}{c+\|x(k)\|^2}\, e(k)\,\bar{x}(k) \tag{7}$$

As here μ is normalized this algorithm converges faster than LMS and the estimated error value between the desired signal and filter output is less than LMS.

D. Recursive Least Squares (RLS) Algorithm

The requirement for the adaptive algorithms which converge rapidly is essential in many applications where the changes in signal characteristics could be quite fast. From this point of view, the best choice is the recursive least squares (RLS) algorithm [10].

The RLS algorithm recursively finds the adaptive filter coefficients that minimize a weighted linear least squares cost function relating to the input signals. RLS algorithm has the potential to automatically adjust the coefficients of a filter, even though the statistical measures of the input signals are not present. This algorithm performs at each instant an exact minimization of the sum of the squares of the desired signal estimation errors [11]. The design of RLS algorithm is given in Fig. 2.

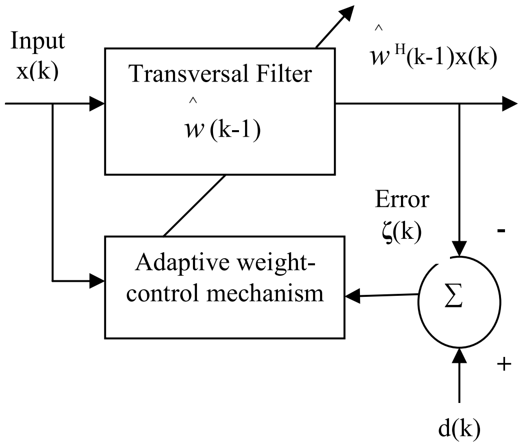

Fig. 2 Design of RLS algorithm

RLS algorithm converges faster than LMS and NLMS algorithms and it has the highest noise cancellation capacity but it involves more complicated mathematical operations and requires more computational resources.

III. Signal Model And Problem Formulation

The speech enhancement problem considered in this paper is to recover a speech signal of interest (SOI) s(k) from the observation signal y(k) which is corrupted by the noise n(k),

$$y(k)=s(k)+n(k) \tag{8}$$

By applying the Discrete Fourier transform (DFT) to (8), we can get the relationship of the signal model in discrete frequency domain. It is assumed that the noise n(k) is uncorrelated with the SOI signal s(k).

A. Speech Enhancement Algorithm Based On Proposed RLS Filter

We propose a noise cancelling scheme based on RLS filtering algorithm of its optimum performance. The block diagram of the proposed noise reduction method is shown in Fig. 3. Most speech signals are time varying signals. In order to achieve effective noise reduction with RLS filtering method, the input signal must be segmented. The unprocessed noisy signal is segmented every ~100ms.

It can be seen from Fig. 3 that the output signal of the proposed method is

$$z_t(n) = \sum_{n=0}^{N-1} y_t(n-k)w(n) \tag{9}$$

where { y_t: t = 1, 2,..., T } be a noisy test signal with T frames and being the frame at time t and N is the number of samples in each frame.

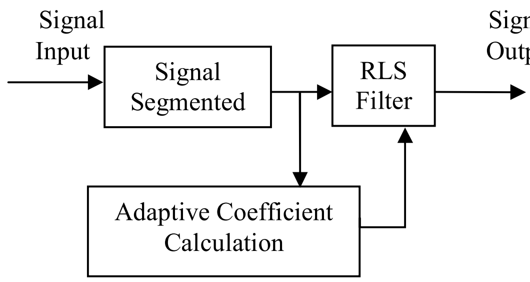

Fig. 3 Block diagram of the proposed method

Steps involved in the proposed RLS Algorithm:

- Initialize the RLS algorithm by setting

$$\hat{w}(0) = [0\ 0\ 0\ldots0]^T$$

as initial values for the filter coefficients.

- For the initial value of the inverse matrix P(0) choose a diagonal matrix with the value δ−1 on the main diagonal.

$$\mathbf{u}(k) = [u(k)\ u(k-1)\ \ldots\ u(k-M+1)]^T \tag{10}$$

where **u**(k) is the input vector.

- For each speech segment and at each time instant, k = 1, 2, . ., compute the RLS update equations given by

$$\Pi(k) = \mathbf{P}(k-1)\mathbf{u}(k), \tag{11}$$

$$R(k) = \lambda^{-1}\,\Pi(k) / (1 + \lambda^{-1}u^H(k)\,\Pi(k)), \tag{12}$$

$$\xi(k) = d(k) - \hat{w}^H(k-1)\mathbf{u}(k), \tag{13}$$

$$\hat{w}(k) = \hat{w}(k\text{-}1) + R(k)\,\xi^*(k) \text{ and} \tag{14}$$

$$\mathbf{P}(k) = \lambda^{-1}\mathbf{P}(k-1) - \lambda^{-1}R(k)\mathbf{u}^H(k)\mathbf{P}(k-1) \tag{15}$$

IV. Experimental Results

In this section, we apply NLMS and the proposed RLS adaptive algorithms to noise reduction in speech applications, and examine their performances. The speech data which sampled at 8 kHz was used to evaluate the proposed method, and the recorded time is ~2s. The speech signal is segmented every 120ms and a frame includes 800 sampled data. The performances of these algorithms are investigated for speech enhancement in different noise conditions. Three types of noises were implemented namely AWGN noise, pink noise and car noise at -5, 0, 5 and 10 dB SNR. The FIR filter order was set to M=32.

The performance of proposed method has been studied with the help of MATLAB simulations. Fig. 4 shows the enhanced speech output for NLMS and the proposed RLS algorithms respectively where the original clean speech is corrupted by pink noise at 10dB. Fig. 5 gives the spectrogram comparison for NLMS and the proposed RLS noise reduction algorithms for the same pink noise at 10dB.

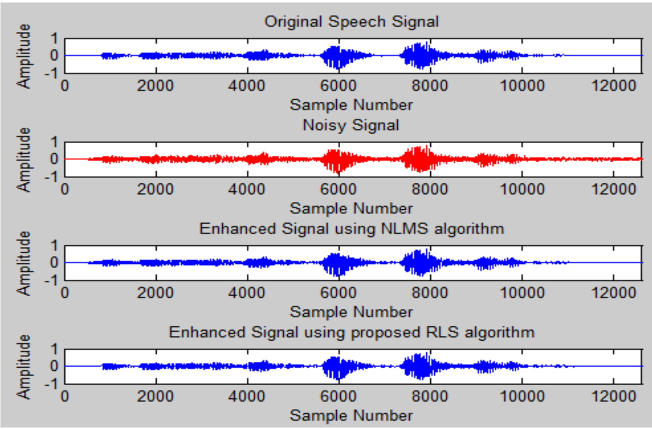

Fig. 4 Results showing the filtered output for pink noise at 10dB SNR using NLMS and proposed RLS algorithm

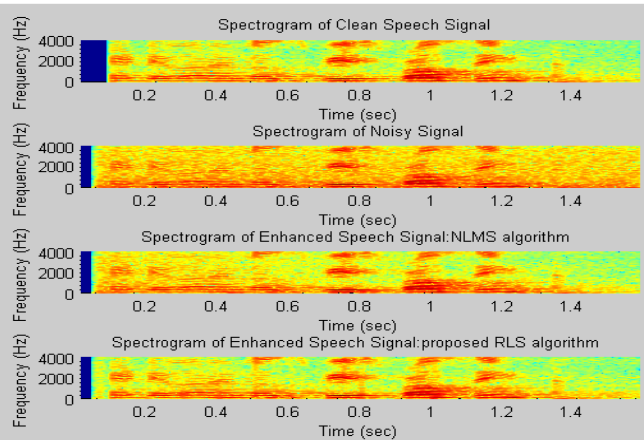

Fig. 5 Spectrogram comparison for pink noise at 10dB SNR using NLMS and proposed RLS algorithm

V. Performance Evaluation

The primary objective of the adaptive filter is minimization of the error signal e(k) which clearly depends on the nature of the input signals and the adaptive algorithm used. The performance of these algorithms are measured based on the metrics namely MSE, SNR and SNR Loss which are explained below.

i) Signal-to-Noise Ratio (SNR)

SNR is defined as the ratio of signal power to the noise power corrupting the signal. One of the most important goals of any speech enhancement technique is to achieve the highest possible SNR. The higher the ratio the better the performance in noise cancellation or reduction. SNR is calculated using the formula

$$SNR(dB) = 10\log 10 \frac{\sum_{k=0}^{N-1} x^2(k)}{\sum_{k=0}^{N-1}\left[\hat{x}(k) - x(k)\right]^2} \tag{16}$$

where $x(k)$ denotes original signal, $\hat{x}(k)$ denotes enhanced signal and N denotes number of samples in original speech signal.

TABLE I
Range of SNR Loss Score and its Remarks

SNR Loss	Degree of SNR Loss	Expected improvement with directional microphones
0-3 dB	Normal/near normal	May hear better than normal hear in noise
3-7 dB	Mild SNR loss	May hear almost as well as normal hear in noise
7-15 dB	Moderate SNR loss	Directional microphones help. Consider array microphones
>15 dB	Severe SNR loss	Maximum SNR improvement is needed.

ii) Mean Squared Error (MSE)

The MSE is defined as the average of the square of the difference between the desired response and the actual system output. It is calculated using

$$MSE = \frac{\sum_{i=1}^{n} (y_i - \hat{y}_i)^2}{n} \tag{17}$$

iii) SNR Loss

SNR loss is the increased signal-to-noise ratio required by an individual to understand speech in noise, as compared to normal performance. It is a new objective measure for predicting the intelligibility of noise-suppressed speech [12]. It provides an efficient way to determine the ability of a person to hear speech in background noise. The following Table I illustrates the range of SNR Loss score and its remarks.

TABLE II
Comparison of NLMS and RLS Algorithms for AWGN Noise Based on MSE, SNR and SNR Loss Values

Type of Noise AWGN Noise		SNR(DB) -5	0	5	10
NLMS	MSE	0.0023	8.6265×10^{-4}	3.3883×10^{-4}	2.9533×10^{-4}
	SNR	7.7885	11.6244	15.4984	15.9010
	SNR Loss	0.6216	0.5833	0.5631	0.5330
RLS	MSE	3.1078×10^{-4}	2.8124×10^{-4}	2.6833×10^{-4}	2.6470×10^{-4}
	SNR	15.6976	16.1172	16.3762	16.4280
	SNR Loss	0.4268	0.4206	0.4249	0.4357

TABLE III
COMPARISON OF NLMS AND RLS ALGORITHMS FOR PINK NOISE BASED ON MSE, SNR AND SNR LOSS VALUES

Type of Noise Pink Noise		SNR(DB) -5	0	5	10
NLMS	MSE	0.0023	9.7817×10^{-4}	5.4487×10^{-4}	3.4010×10^{-4}
	SNR	7.6840	10.9904	13.3827	15.3756
	SNR Loss	0.6932	0.6368	0.5873	0.5428
RLS	MSE	2.6419×10^{-4}	2.6378×10^{-4}	2.6350×10^{-4}	2.6310×10^{-4}
	SNR	16.3881	16.3946	16.3990	16.4054
	SNR Loss	0.4475	0.4535	0.4533	0.4522

TABLE IV
COMPARISON OF NLMS AND RLS ALGORITHMS FOR CAR NOISE BASED ON MSE, SNR AND SNR LOSS VALUES

Type of Noise Car Noise		SNR(DB) -5	0	5	10
NLMS	MSE	0.0013	7.1655×10^{-4}	4.8776×10^{-4}	3.5229×10^{-4}
	SNR	9.8287	12.2341	13.8308	15.0183
	SNR Loss	0.7125	0.6601	0.6174	0.5789
RLS	MSE	3.5529×10^{-4}	3.5211×10^{-4}	3.5042×10^{-4}	3.4904×10^{-4}
	SNR	15.1027	15.1402	15.1598	15.1754
	SNR Loss	0.4826	0.4884	0.4972	0.5034

Figs. 6-8 show the performance evaluation of NLMS and proposed RLS algorithm for AWGN noise based on MSE, SNR and SNR Loss respectively.

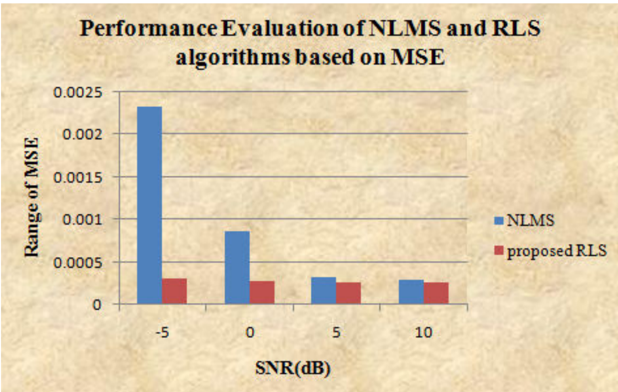

Fig. 6 Performance Evaluation of NMLS and RLS algorithm based on MSE

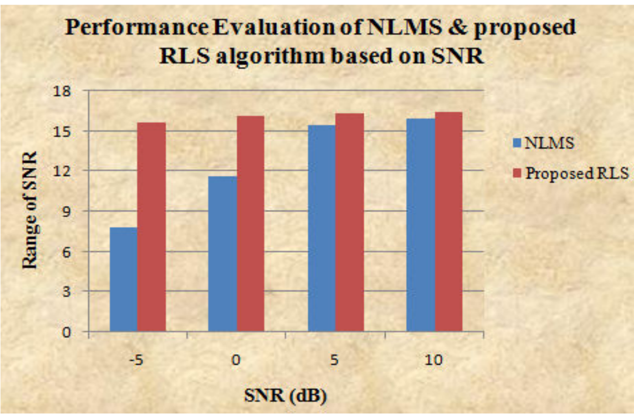

Fig. 7 Performance Evaluation of NMLS and RLS algorithm based on SNR

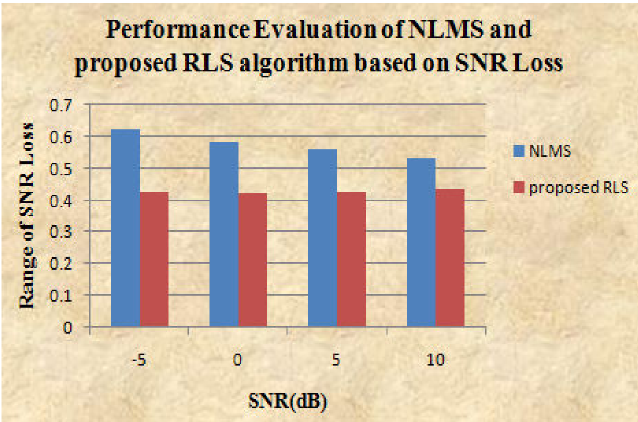

Fig. 8 Performance Evaluation of NMLS and RLS algorithm based on SNR Loss

It is clear from the given tables and figures that the proposed RLS algorithm offers least MSE and SNR Loss values and it also achieves the highest SNR values for all types of noise at different SNR levels.

VI. CONCLUSION

In this paper, we have presented a noise reduction method for speech signals by applying adaptive filtering technique. The noise reduction problem is efficiently solved by using the RLS method. The performance of the proposed RLS algorithm is compared with the NLMS algorithm. We observed that for a particular noise source and algorithm, as the SNR increases the perceived audio quality of the output signal is better. Further, based on the simulation results, we can deduce that the proposed method can improve the performance in quality of noisy speech signal. Through MATLAB simulations, we have demonstrated that the proposed method is quite effective in noise reduction. As a result, with these appropriate settings of the adaptive filter parameters, this optimal signal can be employed for the speech recognition system as a front end.

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