IJCSN International Journal of Computer Science and Network, Volume 4, Issue 5, October 2015
ISSN (Online) : 2277-5420 www.IJCSN.org
Impact Factor: 0.417

Develop and Validate RLS and Refined RLS Adaptive Filter Algorithm for Speech Enhancement

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Abstract - One of the important means of communication is speech signal. In case of long distance communication it is important to maintain the quality of speech signal. The speech signal may get corrupted due to different types of noise. Hence, becomes a challenge to maintain high quality of speech signal [2]. Noise Cancellation is a technique used for reducing undesired noise signal. Communication has become an integral part of our life. The paper aims to investigate performance of recursive least square adaptive algorithms in speech enhancement application. Objective of Implementing and analyzing the algorithms is to modify the algorithm to improve convergence behaviour, reduce computational requirements and decrease steady state mean square error. Experimental results revel that modified RLS algorithms perform better than the existing algorithms.

Keywords - *RLS, Modified RLS, Speech Enhancement, SNR, Convergence.*

1. Introduction

The process to remove unwanted interference is common and occurs in many situations. The technique of adaptive filtering is a method by which signal enhancement or noise reduction can be accomplished. An adaptive filter self adjusts its transfer function according to an optimizing algorithm [1].Many noise measures performance indices have been defined to measure noise in signal processing, in absolute terms, relative to some standard noise level, or relative to the desired signal level. The process to remove unwanted interference is common and occurs in many situations. The technique of adaptive filtering is a method by which signal enhancement or noise reduction can be accomplished. An adaptive filter self adjusts its transfer function according to an optimizing algorithm [5].Many noise measures performance indices have been defined to measure noise in signal processing, in absolute terms, relative to some standard noise level, or relative to the desired signal level [6]. They include, Dynamic range, often defined by inherent noise level, Signal to Noise Ration (SNR), ratio of noise power to signal power, Peak signal-to-noise ratio, maximum SNR in a system. The development became prominent after establishment of Bell Telephone Laboratories at New Jersey, USA. Since then the growing discrete time signal processing techniques along with the development in digital hardware and software technologies have helped the rapid growth of purely digital speech signal processing applications. The digital signal processing areas of applications are as shown in figure 1.1.The major application areas are speech coding, speech synthesis, speech recognition, speaker verification and identification and speech enhancement [1].

2. Recursive Least Mean Square Algorithm

An important class of adaptive filter algorithm is Recursive Least Square algorithm (RLS).The central problem in estimation is to recover, to good accuracy, a set of unobservable parameters from corrupted data. Several optimization criteria is used for estimation purpose over years, but the most important is based on the quadratic cost function. The most important is linear least square criterion, which was developed by Gauss[11].The properties of least square solution is ,it can be evaluated in closed forms, it can be recursively updated as more input data are made available and are maximum likelihood estimators in the presence of Gaussian measurement noise. Thus by setting some predefined constant values the algorithm can be initialized and considering some higher values and then taking its inverse this equation can be adjusted, for this purpose forgetting factor is used, it decides the step size as well as convergence speed of the filter. The weight update equation for RLS algorithm is given as below,

$$z(n) = w(n-1) \times y^{T}(n) \quad (1)$$

$$e(n) = d(n) - z(n) \quad (2)$$

$$k(n) = \frac{P(n-1) \times z(n)}{\lambda + z^{H}(n) \times P(n-1) \times z(n)} \quad (3)$$

$$P(n) = \frac{P(n-1) - P(n-1) \times z^{H}(n) \times k(n)}{\lambda} \quad (4)$$

$$w(n) = w(n-1) + e(n) \times k(n) \quad (5)$$

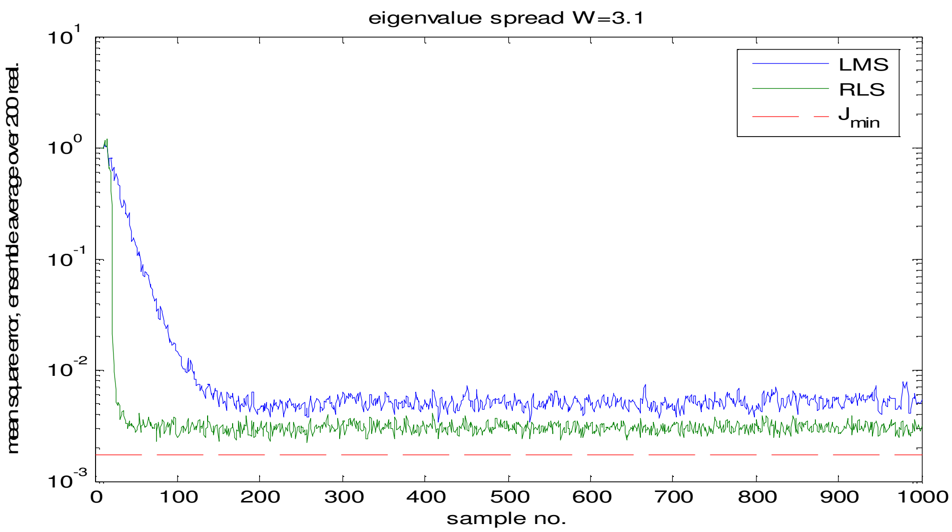

Figure 1 MSE ,Jmin for LMS and RLS algorithm

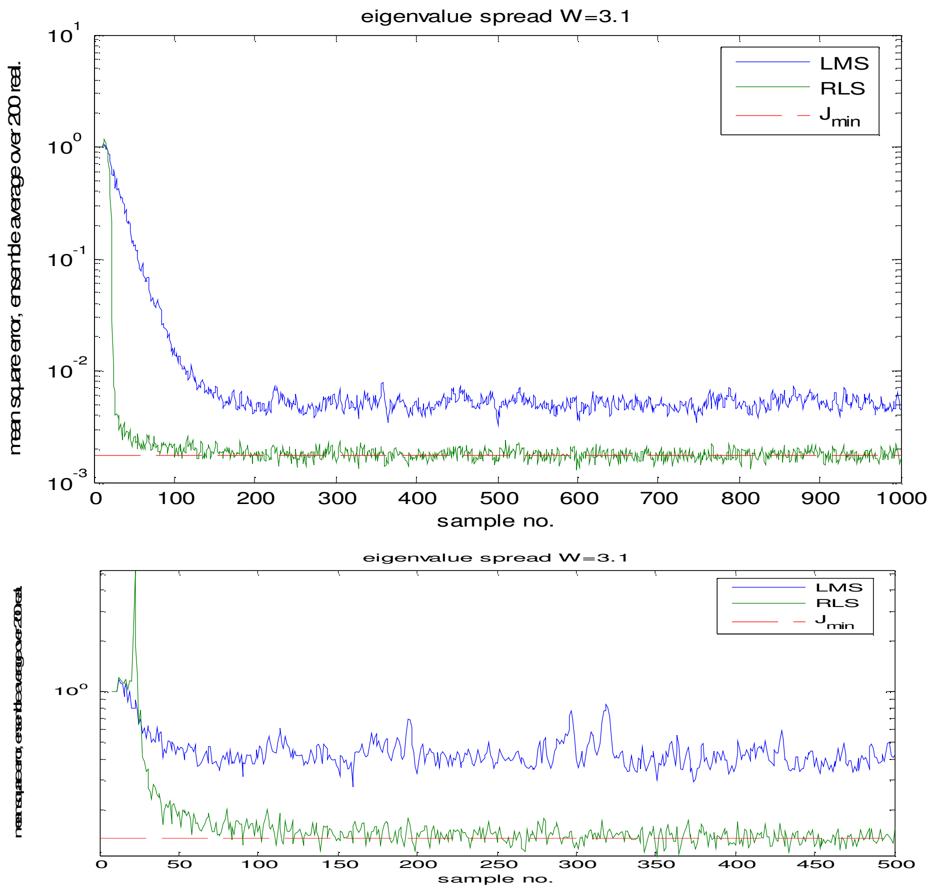

Figure 2 MSE, Jmin for LMS and RLS algorithm

Algorithms insensitivity to the eigen value spread of the correlation matrix

Least Mean Square (LMS)

Table 1 Results for LMS algorithm

METHOD	NOISE TYPE	NOISE LEVEL	SNR	MSE	TIME
LMS	AIRPORT	0DB	11.1708	0.000218	2.39115
LMS	AIRPORT	5DB	9.9084	0.000193	0.24389
LMS	AIRPORT	10DB	9.1533	0.000186	0.24631

Table 2 Results for RLS algorithm

METHOD	NOISE TYPE	NOISE LEVEL	SNR	MSE	TIME
RLS	AIRPORT	0DB	24.7705	1.14E-05	0.39084
RLS	AIRPORT	5DB	22.6959	1.27E-05	0.34427
RLS	AIRPORT	10DB	21.6435	1.35E-05	0.34369

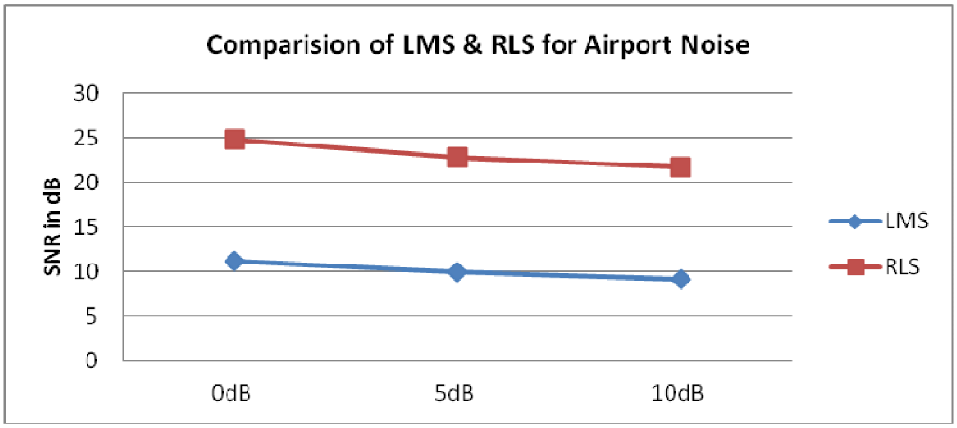

Figure 3 SNR Performance comparisons of LMS and RLS in presence of airport noise

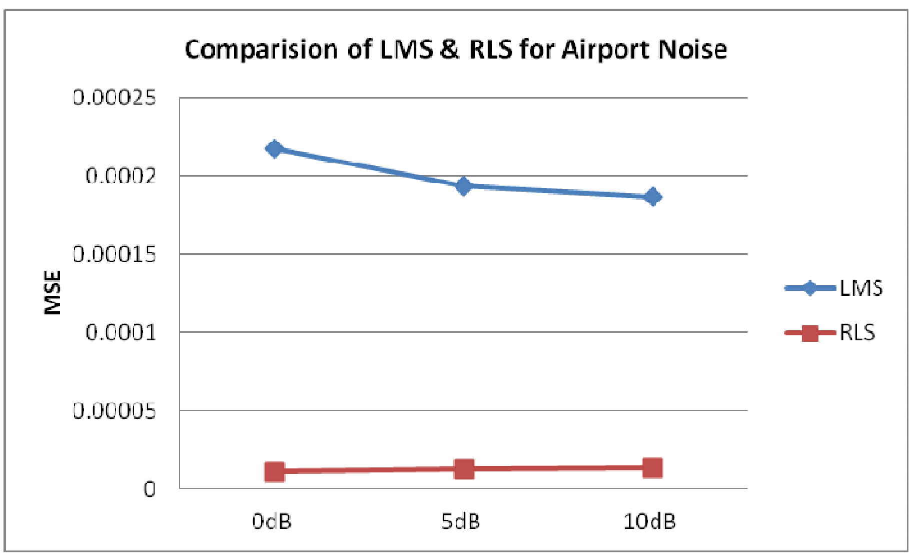

Figure 4 MSE Performance comparisons of LMS and RLS in presence of airport noise

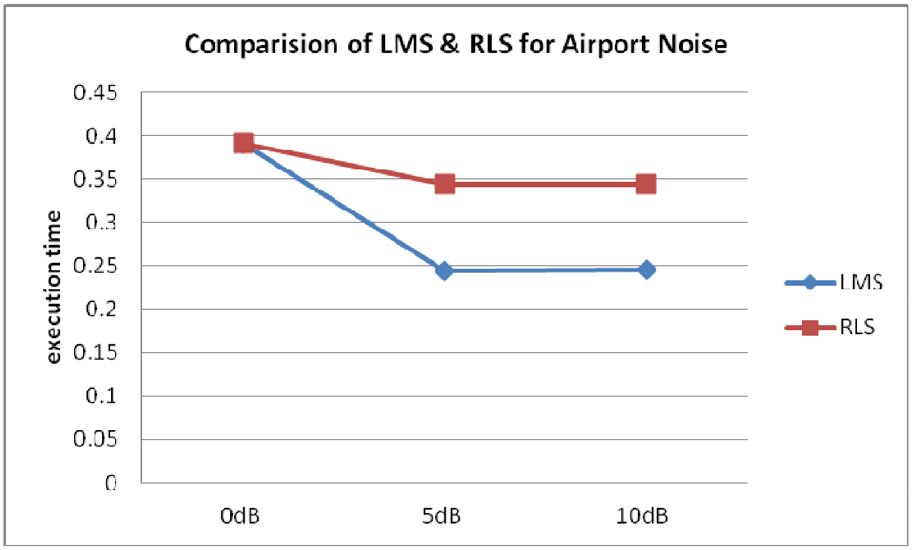

Figure 5 MSE Performance comparisons of LMS and RLS in presence of airport noise

Table 3 Results for LMS and RLS algirithm

PARAMETER	LMS	RLS
SNR	GOOD	BETTER
MSE	AT PAR	LESS ERROR
EXECUTION TIME	CONVERGES FAST	CONVERGES SLOW

A. Page Layout

Your paper must use a page size corresponding to A4 which is 210mm (8.27") wide and 297mm (11.69") long. The margins must be set as follows:

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3. Modified Recursive Least Mean Square Algorithm

The weight update equation for modified RLS algorithm is given as,

$$w(n+1) = w(n) + e(n) \times zz(n) \quad (6)$$

$$e(n) = d(n) - y(n) \quad (7)$$

$$y(n) = w^T(n) \times x(n) \quad (8)$$

$$z(n) = v \times z(n) \quad (9)$$

$$v = \frac{1}{1+q} \quad (10)$$

$$z(n) = R(n) \times x(n) \quad (11)$$

$$q = x^T(n) \times z(n) \quad (12)$$

$$R(n+1) = R(n) - zz(n) \times z(n) \quad (13)$$

Where e is the output error, d is the desired signal and x is the input signal.

Table 4 SNR performance analysis using RLS and modified RLS algorithms

SNR	0dB	5dB	10dB
RLS	24.7705	22.6959	21.6435
Modified RLS	26.5192	23.5748	22.3697

Table 5 MSE performance analysis using RLS and modified RLS algorithms

MSE	0dB	5dB	10dB
RLS	0.0000114	0.0000127	0.0000135
Modified RLS	0.0000077	0.0000106	0.0000118

Table 6 Time performance analysis using RLS and modified RLS algorithms

TIME	0dB	5dB	10dB
RLS	0.39084	1.46482	1.467
Modified RLS	1.53531	0.34427	0.34369

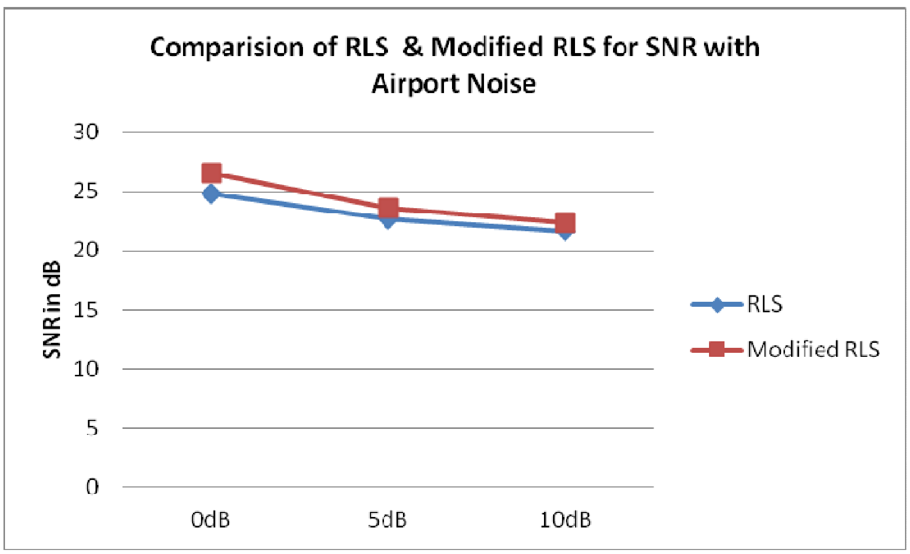

Figure 6.17 Comparisons of RLS & Modified RLS for SNR with airport noise

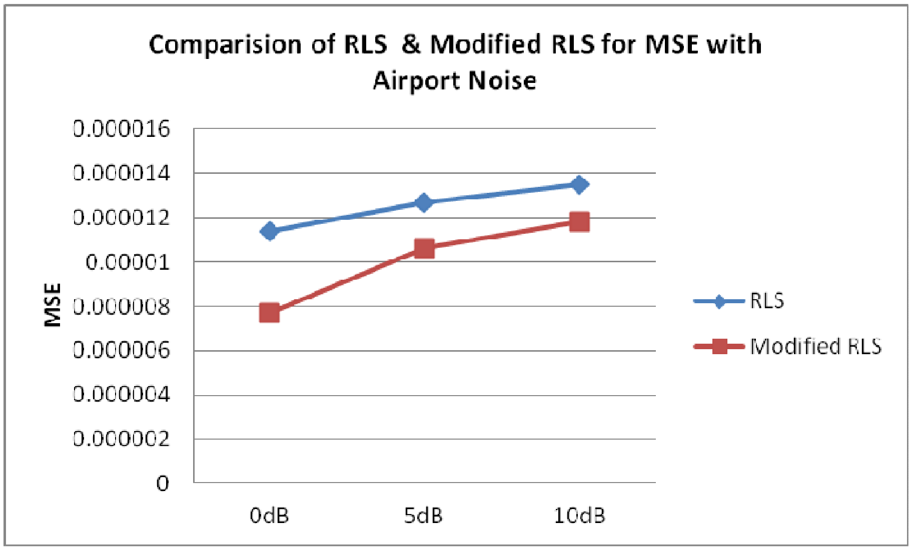

Figure 6.19 Comparisons of RLS & Modified RLS for MSE with airport noise

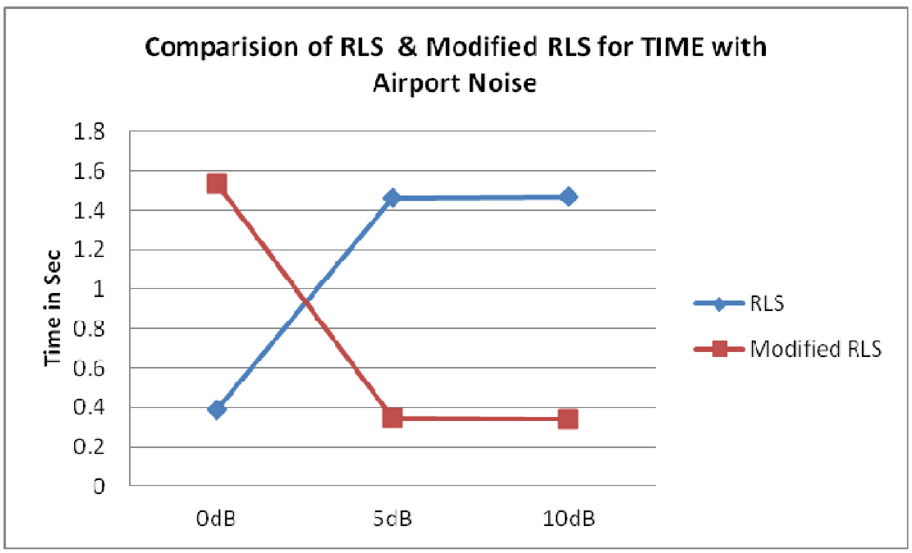

Figure 6.20 Comparisons of RLS & Modified RLS for MSE with airport noise

Table 7

Parameter	**RLS**	**Modified- RLS**
SNR	Good	Better
MSE	At par	Less error
Execution Time	Converges slow	good

4. Conclusions

The Experimental results reveal that the RLS algorithm has minimum MSE and maximum SNR as compared to LMS but at a cost of increased computational complexity. Modified RLS provides even better SNR as compared to existing RLS algorithm. The test is performed at 0dB, 5dB and 10dB airport noise. The experimentation and validation are carried out for Mean Square Error (MSE) ,SNR and execution time. The experimentation and validation is carried out for modified RLS and is compared with existing methods and it is observed that modified method performs better as compared to existing methods.

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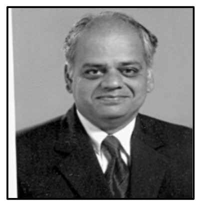

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