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Performance Comparison of Adaptive Algorithms with Improved Adaptive Filter Based Algorithm for Speech Signals

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Abstract: Noise cancellation is a technique applied to offset the interference in the signals. Noise Cancellation finds a pile of applications like mobile telephones, hearing aids bluetooth receivers and more. It is difficult to find out any signal, especially speech signal in a noisy environment.An adaptive filter plays a lively role in cancelling the noise in the speech signal. Adaptive filters use several algorithms for scaling down the interferences in the signal. Thus an improved Adaptive filter based Noise cancellation for speech signals approach named Variable Step Size Normalized Differential LMS (VSSNDLMS) Algorithm that incorporates the performance and features of Variable Step Size LMS (VSSLMS) and Normalized Differential LMS (NDLMS) is offered. It is to analyze and compare the performance of the proposed VSSNDLMS Algorithm with various Least Mean Square adaptive algorithms. This algorithm aims in applying the proposed algorithm for various real time applications like auditorium, automobile, seminar hall, etc. The analysis indicates that the proposed adaptive algorithm has fast convergence rate, tracking ability, reduced Mean Square Error (MSE) and maladjustments which is the required characteristics of an adaptive filter. For performance, analysis different input speech signals and sound signal are analyzed. The simulation results show that the proposed VSSNDLMS algorithm converges fast with Minimum MSE and is useful in anticipating the performance of adaptive filters.

Key words: Adaptive filter, VSSNDLMS, VSSLMS, NDLMS, MSE

INTRODUCTION

The primary idea of this study is to propose Improved adaptive filter based noise cancellation techniques for speech signals.This study investigates the existing algorithms towards noise cancellation function. Granting to the analysis, the existing system fails to call off the noise effectively. The proposed system offers an improved adaptive algorithm based on noise cancellation for speech signals which overcomes the drawbacks of the existing algorithms. Although the existing LMS filter is trouble-free in computational terms, its mathematical analysis is profoundly complicated because of its stochastic nonlinear nature. The improved Adaptive Filter based Noise cancellation technique for speech signals using Variable Step Size Normalized Differential LMS Algorithm has various real time applications like automotive, auditorium, seminar hall, railway station , bus stand and more.

In Variable Step Size LMS (VSSLMS) algorithm the step size adjustment is controlled by the square of the prediction error. A large prediction error will cause the step size to increase to provide faster tracking while a small prediction error will result in a decrease in the step size to yield smaller maladjustment. In Normalized Differential LMS (NDLMS) algorithm a different approach is considered for weight adjustment. The motivation is to design an LMS algorithm that can handle both the strong and the weak target signals. Hence, whenever the filter inputs and outputs fluctuate more or less, the weights should be adjusted accordingly.

The NDLMS and VSSLMS are combined together and an improved, efficient algorithm called Variable Step Size Normalized Differential LMS (VSSNDLMS) algorithm is proposed to enhance speech processing. The objective of proposing this algorithm is to design an effective adaptive filter to remove the noise and to improve the quality of speech signals.

Literature review: In this literary survey on adaptive filter theory, adaptive algorithms, signal processing, speech processing and the performance of adaptive algorithms are discussed. Widrow *et al*. (1975, 1976) has proposed an adaptive filter which is highly efficient in reducing interference when a second sample of the noise is available. The LMS algorithm introduced by Widrow and Hoff is one of the simplest algorithms used in the adaptive structures. Kwong and Johnston (992) proposed new variable step size algorithm.

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Shengqian *et al.* (2011) analysed on adaptive noise cancellation of an improvement LMS algorithm and proposed a modified LMS algorithm of variable step length based on Kwong's LMS algorithm for Noise Cancellation.

Widrow *et al* (1975) proposed a adaptive noise cancelling principles which is an alternative method of estimating signals corrupted by additive noise or noise. Many adaptive algorithms were proposed for noise cancellation (Untwale and Degaonkar *et al*., 2015) in this study some algorithms are discussed. There are conflicting objectives between fast convergence and low EMSE for NLMS with fixed regularization parameter. There are many variable step size NLMS algorithms been proposed to solve this dilemma of the conventional NLMS. Kwong and Johnston (1992) applied the power of instantaneous error to introduce a VSSLMS filter. This VSSLMS had a larger step size when the error is large and a smaller step size when the error is small. Aboulnasr and Mayyas (1997) modelled a modified VSSLMS algorithm to alleviate the influence of uncorrelated disturbance to overcome the sensitivity of VSSLMS algorithm.

The step-size update of MVSS is adjusted by using an estimate of the autocorrelation of errors at adjacent time samples (Proakis and Manolakis, 2007). Shin *et al.* (2004) used the norm of filter coefficient, error vector as a criterion for optimal variable step-size and proposed a Variable Step Size Affine Projection Algorithm (VS-APA) and a variable step size NLMS as easily. Benesty *et al.* (2006) projected a non-parametric VSSNLMS algorithm, which need not tune many parameters as that of many Variable Step Size algorithms. Park *et al.* (2007) have proposed a suitable VSSLMS algorithm of ANC in an Automobile environment. The ANC algorithm presented in this work provides improvements in both rates of convergence and estimation accuracy over previously presented LMS algorithms in an automotive environment, such as colored input signal.

Ryu *et al.* (2008) presented his work on an Adaptive Noise Canceller with DSP Processor. The variable step size normalized least mean square algorithm is proposed to settle the conflicting requirement of fast convergence and low maladjustment. Adaptive filter is applied in medical fields for denoising (Makwana and Gupta, 2015; Sultana *et al*., 2015; Yazdanpanah *et al*., 2015) of ECG signals. ECG signals consist of noise signal which is non stationary that affects the reliability of ECG signals.

Mota *et al.* (2015) has provided sufficient conditions for perfect signal reconstruction at each time instant. Rewadkar and Madhukar (2014) has suggested a new adaptive algorithm for information collection. Radhika and Arumugam (2014) has proposed An Affine Projection Algorithm (APA) with decomposed weight vector and variable step size. The optimal value of variable step size is obtained by the largest decrease in the mean square deviation error. Fravolini *et al.* (2014) has proposed Adaptivecontroller for easy verification procedures that relate design parameters to time domain performance especially during transients. Gui *et al.* (2013) in his work explained the difference between VSS-ZA-NLMS and ISS-ZA-NLMS algorithms for channel estimation. Adaptive algorithm is proposed for motion detection (Maghsoumi *et al*., 2015) based on significant robustness against shadow, noise, lighting changes, etc.

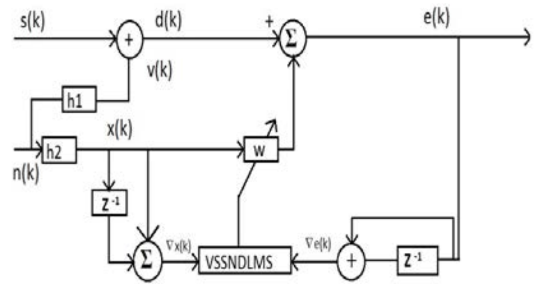

Fig. 1: Proposed VSSNDLMS algorithm

The critique of literature has been made out in the sphere of the ANC. Several stages of research students and scientists have been put forward and the advantages and drawbacks are analyzed. Hence the paper on ANC, adaptive filter theory, adaptive algorithms (Kumar and Gill, 2015), signal processing and the carrying out of adaptive algorithms have been discussed and their advantages and disadvantages have been studied. It is understood that while some of the papers have contrasting advantages and disadvantages, some proposed their work effectively able to negate the disadvantages of their twins.

MATERIALS AND METHODS

Proposed adaptive algorithm: Adaptive Noise Canceller (ANC) using Variable Step Size Normalized Differential LMS Algorithm is applied as an Adaptive Filter to capture rid of the noise from the speech signal. The NDLMS and VSSLMS are combined together and an improved, efficient algorithm called Variable Step Size Normalized Differential LMS algorithm is proposed to enhance speech processing. The objective of proposing this algorithm is to design an effective adaptive filter to get rid of the disturbance and to ameliorate the tone of speech signals. Figure. 1 shows the proposed algorithm using VSSNDLMS method.

The Normalized Differential LMS algorithm is particularly suited for slow varying signals and is less sensitive to the desire signal power variation compared to the existing algorithms. Moreover, the excess error and maladjustment by NDLMS are much less than that of existing Algorithms. VSSLMS Algorithm is applied to cut the trade off between maladjustment and tracking ability of the fixed step size LMS Algorithm. The VSSLMS also reduces sensitivity of the maladjustment to the point of non-stationary. The features of NDLMS and VSSLMS are combined, VSSNDLMS Algorithm is proposed.

In case of LMS algorithm under non-stationary environment, errors occur which leads to deviation of filter weights from the optimal weight of the filter. The proposed algorithm satisfies this criterion by adjusting the step size. The VSSLMS algorithm converges faster and NDLMS algorithm has minimal MSE. By combining the VSS and NDLMS, the VSSNDLMS algorithm converges fast with MMSE.

Figure 1 shows a typical Adaptive Noise Canceller which has two inputs: a primary input, d (k), composed of the desired signal, s (k), corrupted by a filtered additive noise signal v (k) and a reference input, x (k) which is a different filter noise and the input to the adaptive filter. X (k) is assumed to be correlated with v (k) and uncorrelated with s (k). The output of Adaptive Noise Canceller is e (k), the difference between the primary input and the output of the Adaptive filter. The interference signal is represented by n (k).The difference in input x(k) and difference in output e(k) is given to VSSNDLMS Algorithm. From Eq. 1, 2 and 3 VSSNDLMS algorithm is aimed and the equation for updating the coefficient is made by:

$$w(n+1) = w(n) + \frac{\mu_{var}}{\varepsilon + \|\nabla X(k)\|^2} \times \nabla x(n) \times \nabla e(n) \quad (1)$$

where:

$$\nabla e(k) = e(k) - e(k-1) \quad (2)$$

$$\nabla x(k) = x(k) - x(k-1) \quad (3)$$

and the μ_{var} is the variable step size which is given by:

$$\mu_{var} = \begin{cases} \mu_{max} \ldots if \mu_i > \mu_{max} \\ \mu_{min} \ldots if \mu_i < \mu_{min} \end{cases} \quad (4)$$

where: $\mu_i\ \alpha \times \mu + \gamma \varepsilon^2_k$ and by selecting the proper value of α and γ we can get better performance. Depending upon the speech signal, the value of α and γ vary. The constant μ_{max} is chosen to ensure that the Mean-Square Error (MSE) of the algorithm remains bounded.is chosen to provide a minimum level of tracking ability. Usually, μ_{min} will be near the value of μ that would be chosen for the Fixed Step Size (FSS) algorithm. The purpose of the adaptive filter is to minimize the MSE .Thus, the MSE quantity is essential to assess the operation of the adaptive filter. The quality of adaptation can be measured by the maladjustment, M which is the dimensionless ratio of the EMSE to the MMSE in the steady state environment:

$$M = \frac{EMSE_{SS}}{MMSE} \quad (5)$$

The MMSE is the optimum filtering error which represents the percentage of primary signal that cannot be cancelled by optimal weight. It can be determined by averaging the output signal power over samples after which the algorithm extends to the steady state, that is:

$$MMSE = \frac{1}{K-P} \sum_{K=P}^{K} |s(k)|^2 \quad (6)$$

where:

K = The total number of the speech samples

P = The number of samples at which the algorithm reaches steady state

The EMSE, caused by the difference of the weights from their optimal values, is defined equally:

$$EMSE(k) = \frac{1}{N} \sum_{J=0}^{N-1} | e_1(k-j) | \quad (7)$$

Where:

(k)e (k) -s (k) = The residual error and

N = The number of samples used in the appraisal. The steady state EMSE (EMSESS) is defined by taking the average of the EMSE over certain durations

$$EMSE_{SS} = \frac{1}{K-P} \sum_{k=p}^{K} EMSE(k) \quad (8)$$

The Signal-to-noise ratio (SNR) compares the amount of signal with the amount of background noise in a particular signal. Higher SNR indicates the background noise is less noticeable. The decibel is defined in such a way that the SNR may be applied to any signal, regardless of its source The general SNR is defined as:

$$SNR = 10\log_{10} \frac{\sum_n s^2(n)}{\sum_n (s(n) - s(n))^2} \quad (9)$$

Thus an improved Adaptive Filter Based Noise Cancellation technique for speech signals is proposed.

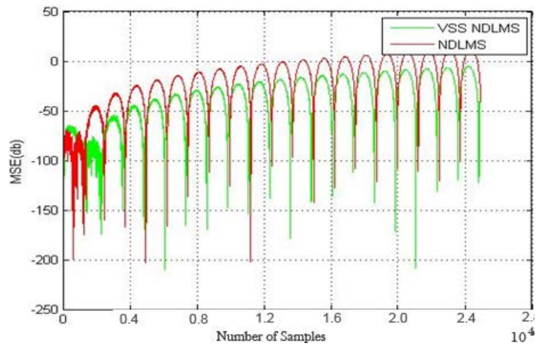

Fig. 2: MSE comparisons of NDLMS and VSSNDLMS for Bus stand Speech signal

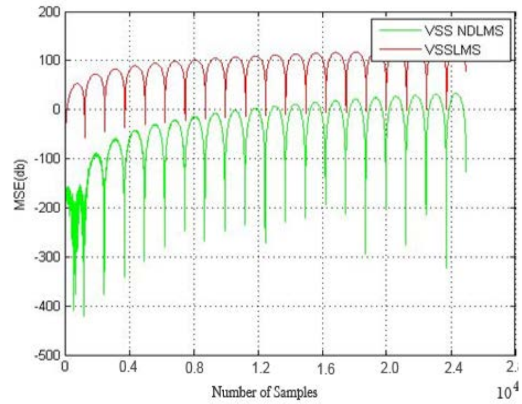

Fig.3: MSE comparison of VSSLMS and VSSNDLMS for Bus stand Speech signal

RESULTS AND DISCUSSION

Performance comparison of various least mean square adaptive algorithms with the proposed vssndlms algorithm: This section analyses the performances of VSSLMS, Normalized Differential LMS and Proposed VSSNDLMS algorithms, in terms of MSE, maladjustment, spectral response and Convergence rate. Simulation is performed with reference sine wave signal added with random noise. In one case the interference is removed effectively, then speech signal corrupted with noise is applied as input. The results demonstrate that proposed algorithm performance better in terms of fast convergence rate and MMSE. It also improves the performance on the caliber of the speech corrupted with stationary and non stationary noise. So the proposed VSSNDLMS algorithm is a promising LMS method for interference cancellation.

Results and discussion carried out with different speech signal in different environments. In this section, rate of convergence, MSE, maladjustment are analyzed and the results are discussed

Performance analysis of MSE: In performance measure MSE is considered. Different inputs taken in different environments are considered for analysis. Proposed and improved adaptive filter based noise cancellation techniques for speech signals using VSSNDLMS algorithm is compared with NDLMS and VSSLMS algorithm. Sine wave, speech signal observed in the bus stand, drums signal observed in the studio, speech signal observed in seminar hall and speech signal observed in the classroom are considered for analysis.

In Fig. 2 the simulation results of speech signal observed in bus stand is taken. The MSE of Proposed VSSNDLMS Algorithm is compared with NDLMS Algorithm. Fig. 3 simulation result of proposed VSSNDLMS Algorithm is compared with VSSLMS Algorithm. In both simulation results the MSE value is less. In Fig. 4 the simulation results of speech signal

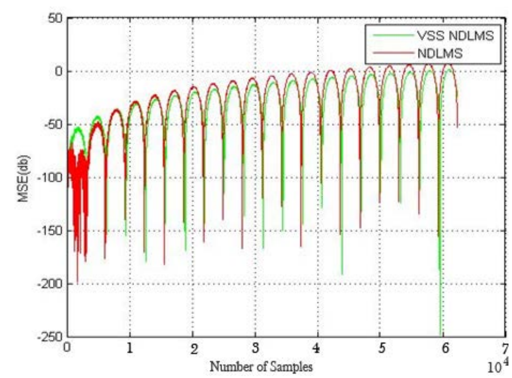

Fig. 4: MSE comparisons of NDLMS & VSSNDLMS for Seminar Hall speech signal

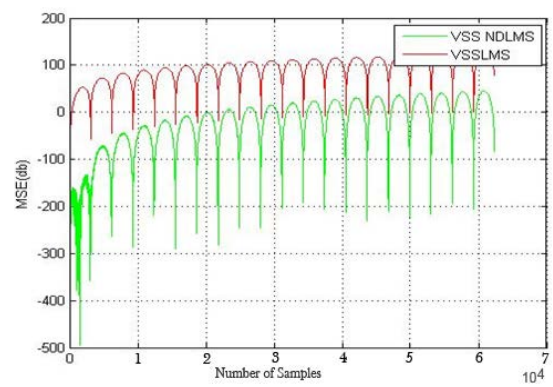

Fig. 5: MSE comparison of VSSLMS & VSSNDLMS for Seminar Hall speech signal

observed in seminar hall is believed. The MSE of Proposed VSSNDLMS Algorithm is compared with NDLMS Algorithm. Figure 5 simulation results of proposed VSSNDLMS Algorithm are compared with VSSLMS Algorithm. In both simulation results the MSE

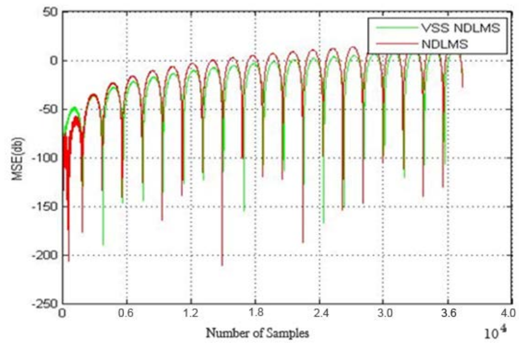

Fig. 6: MSE comparisons of NDLMS& VSSNDLMS for class room speech signal

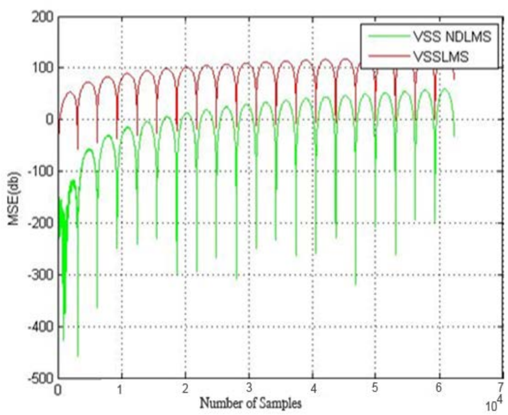

Fig. 7: MSE comparisons of NDLMS& VSSNDLMS for class room speech signal

value is less. In Fig. 6, the simulation results of speech signal observed in the classroom is seen. The MSE of Proposed VSSNDLMS algorithm is compared with NDLMS Algorithm. Figure 7 simulation results of proposed VSSNDLMS algorithm are compared with VSSLMS Algorithm. In both simulation results the MSE value is less.

Simulation results show that the results obtained for the VSSLMS algorithm and NDLMS algorithm are not acceptable. The most significant element, which decides whether the algorithm is suitable to be used for Noise cancellation The experimental results prove that the operation of the existing VSSLMS Algorithm and NDLMS Algorithm is quite tender to the noise disturbance. Since measurement noise is a reality in any practical system, the usefulness of any adaptive algorithm is evaluated by its performance in the presence of this disturbance.

The performance of the VSS algorithm deteriorates in the presence of measurement noise. Hence, a Improve Adaptive filter based noise cancellation techniques for speech signals using VSSNDLMS algorithm is proposed, where the step size of the algorithm is adjusted according to the Step size. The constant μ_{max} is chosen to ensure that the Mean-Square Error (MSE) of the algorithm remains bounded. $\mu_{min}$is chosen to provide a minimum level of tracking ability.

Performance analysis of maladjustment and convergence rate: In this study, the performance of the speech signal and music signal are analyzed for maladjustment and convergence rate. Performance analyses of the maladjustment and convergence rate are being estimated by the rate of convergence and error. When a mistake is estimated earlier the maladjustment will be consumed and the pace of convergence will be high, If the maladjustment is low the tracking speed of the algorithm will be quicker. By the performance analysis, It is clear that the MSE is very low in novel Adaptive algorithm. Thus the pace of convergence of the novel VSSNDLMS algorithm is high compared with VSSLMS and NDLMS algorithm. In the speech signals both stationary and non-stationary noise is present. The speech signal is corrupted by the Stationary Noise which is assumed to be white Gaussian with zero mean and constant variance, whereas speech signal corrupted by the non-stationary noise is white Gaussian noise which has variance varying from 0.000-0.15.

Table 1 comparisons of Maladjustment for speech signal observed in various places are tabulated. In Bus Stand Speech Signal when VSSLMS and NDLMS are compared, the maladjustment of the proposed algorithm is less. When a noise variance is 0.15, the maladjustment of VSSNDLMS is less than 52% of VSSLMS and 55% less than NDLMS algorithm. So, it can be understood that the proposed algorithm shows figurative difference in the values of the maladjustment.

The Maladjustment for speech signal Observed in the seminar hall, when a noise variance is 0.1, the maladjustment of VSSNDLMS is less than 32% VSSLMS algorithm and 75% less than NDLMS algorithm. So, it can be understood that the proposed algorithm shows figurative difference in the values of the maladjustment. The Maladjustment for speech signal Observed in the bus stand when VSSLMS and NDLMS are compared, the maladjustment of the suggested algorithm is less.When noise variance is 0.01, the maladjustment of VSSNDLMS is<80% VSSLMS Algorithm and 85% less than NDLMS Algorithm. So, it can be understood that the proposed algorithm shows figurative difference in the values of the

Table 1: Performance measures for different noise variance in different applications

Noise variance	Bus stand			Seminar hall			Classroom		
	VSSLMS	NDLMS	VSSNDLMS	VSSLMS	NDLMS	VSSNDLMS	VSSLMS	NDLMS	VSSNDLMS
0.001	0.072	0.0788	0.001	0.107	0.0152	0.005	0.102	0.140	0.010
0.01	0.071	0.0788	0.006	0.102	0.01516	0.0037	0.101	0.150	0.012
0.1	0.05	0.0760	0.039	0.094	0.0416	0.031	0.095	0.151	0.005
0.15	0.081	0.045	0.042	0.089	0.0131	0.037	0.089	0.152	0.010

Table 2 : Comparison of Signal to Noise Ratio for various algorithms

Algorithm	SNR value (dB)		
	Bus stand	Seminar hall	Classroom
VSSLMS	13.59	12.79	14.66
NDLMS	16.86	15.67	16.83
VSSNDLMS	20.03	19.86	19.93

Table 3 Performance analysis for Bus Stand sppech in terms of MSE and maladjustments

Noise variance	VSSLMS		NDLMS		VSSNDLMS	
	Emsess	m(%)	Emsess	m(%)	Emsess	m(%)
0.001	-24.45	7.54	-38.43	0.29	-45.92	0.031
0.01	-24.34	7.89	-24.47	8.65	-44.83	0.042
0.1	-24.12	8.15	-21.78	18.65	-44.21	0.048
1.0	-24.01	8.97	-13.87	63.47	-43.34	0.067

maladjustment. Lower maladjustment is one of the central advantages of VSSNDLMS Algorithm when compared with VSSLMS and NDLMS Algorithm.Table 2 shows the performance analysis of existing and proposed algorithms in terms of SNR values, where the existing alsogithm finds a better way. Table 3 shows the Maladjustment and MSE analysis for different variances in bus stand speech , which depicts the outperformance of VSSNDLMS algorithm.

CONCLUSION

The performance comparison of various LMS adaptive algorithms with the proposed VSSNDLMS algorithm is analyzed. Performance analyses of novel adaptive algorithm with different speech samples are implemented. By properly analyzing the merits and faults of the algorithms a novel algorithm named variable Step size normalized differential LMS is proposed for cancelling the noise.

The algorithm is analyzed with test signal in the presence of interference.The outcomes show that the algorithm effectively removes the interference when compared to the other adaptive algorithms. Thereafter the signals recorded in various noisy environments like bus stands, drum music in the studio, seminar hall speech, classroom lecturer are considered and the proposed VSSNDLMS effectively removes the noise in all instances.

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