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Adaptive Noise Cancellation from Speech Signals using Variablestep Sizealgorithm

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Abstract**:** ***Noise cancellation from the speech signal is the most importanttask in applications like communications, hearing aids, speech therapy and many others. This allows providing good resolution speech signal to the user. The speech signals are mostlycontaminated due to the several natural as well as manmade noises. As the characteristics of these noises random in its nature filtering techniques with fixed coefficients are not suitable for noise cancellation task. Hence, in this work an adaptive noise canceller algorithmhas driven for enhancement of speech signal applications which has the capability to update its weight coefficients based on the statistical nature of the undesired component in the actual speech signal. In our experiments in order to achieve better convergence rate as well as filtering capability we propose Step Variable Least Mean Square (SVLMS) algorithm instead of constant step parameter. The computational complexity of the speech enhancement process is also a key aspect due to the excessive length of the speech signals in realistic scenario. Hence, to reduce the computational complexity of the proposed mechanism we used Sign Regressor SVLSM (SRSVLMS), which is a hybrid realization of familiar sign regressor algorithm and the proposed SVLMS. Using these two techniques noise cancellation models are developed and tested on real speech signals with unwanted noise contaminations. The experimental outputsconfirm that the SRSVLMS based speech signal enhancement unit performs better than its counterpart with respect to convergence rate, computational complexity and signal to noise ratio increment.***

Keywords**:** ***Adaptive Algorithms, Convergence rate, Computational Burden, Speech Signal, Speech Enhancement, Step Size Parameter.***

I. INTRODUCTION

Speech signal enhancement is a key phenomenon in the present-day technological evaluation scenario. It has several applications in the fields of cellular communications, war field communications, hearing aids, speech re-habitation, air traffic control, speech recognition, speech biometrics, tele-conference systems, and so forth. Many researchers have contributed several mechanisms for speech signal enhancement.

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According to fixed step size in [1] Leonardo Rey Vega et al. scrutinized a strong variable step-size NLMS algorithm can lower the square of a posteriori fault and also proved the relation among the proposed technique and the other obtained by utilizing a strong guides advance and found better results.In extension they introduced in [2] Scott C. Douglas et al.projected an NLMS technique is widespread, leading to a group of ledge-like technique supported on the L_p-reduced purifier coefficient variance.

A total origin of group of algorithm is set, and results are carried out to explain the union characteristics of the methods.And in [3] Dr. D. Deepa et al. analyze clattery, wanted voice signals were double altered with discrete cosine and hadamard changes and applied to adaptive filter by utilizing NLMS method as a result, speed assembly of clattery voice to wanted voice with nice act when evaluated to conventional LMS method. A novel innovative mechanism is futured in [4] Jwu-sheng Hu et al. proposed a new advance with a beam forming technique which reduces the least functioning and provides best strength beside a random but norm-bounded wanted wave routing vector variance. Also, second-order extended (SOE) H∞ removes the noise. Adiffernt methods have been used to categorize the actual speech signal from the artifact contaminated speech signal is explained briefly in [5] Yue Xian Zou et al. discussed an efficient spatial-frequency area voice development technique wiener post-filtering (WPF) methods in which WPF is the potent approximator reduces the spatial distortion. Not only LMS we have a latest adaptive filtering technique is also shown in [6] L. Zao et al. presented a novel technique to remove noise. Here EMD and Hurst-based (EMDH) analysis is used to estimate the voice improvement researches taking surroundings echo artifacts in various parameters of mobility which gets better results.To extract the speech signal a new algorithm is proposed in [7] Brady N. M. Laska et al. proposed that by utilizing element clean outline lets the needed technique to form the voice spectral voltages as an autoregressive method with laplace spread excitation. Two variables, one is interacting various schemes for changes and the other permits for angle variations and hence developing slicing competence.In [8] Upal Mahbub et al. justified a two level method to pact with acoustic echo cancellation (AEC), firstly the holdup edition of echo and cleaned wave is made as reference in the next level a gradient-based adaptive purifier method and minimum wiener-hopf result is formed and gave better results when compared with TIMIT database.

They introduced a novel implementation in [9] Amin Zehtabian et al. developed a new advancewith voice signal development SVD along withGA which proceed outcomes in a required decrement of the distorted wave among drawing the worth of the unique voice.In[10] Sharon Gannot et al. elucidated on sensor array need for constructing the wave indistincted by nosiness. They mentioned the arbitrary transfer functions (TFs) along with generalized sidelobe canceler (GSC) drawn a subminimal method by taking TFs ratios in the place of TFs which forms the wanted wave.

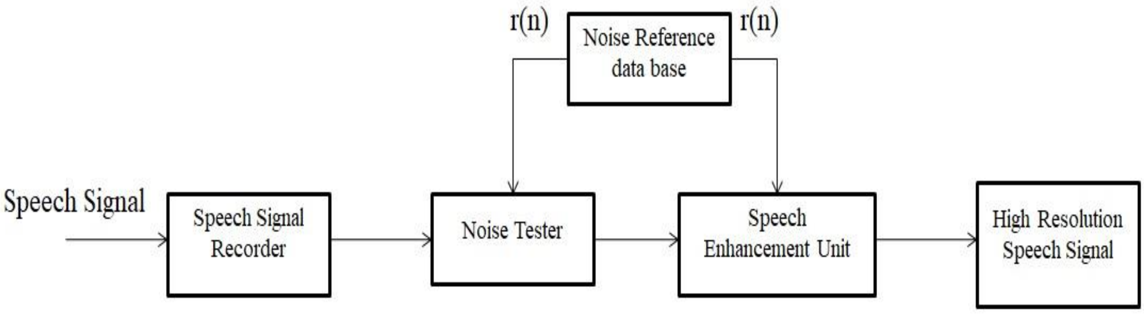

Fig.1: Schematic diagram of Speech Enhancement process used in our experiments.

In [11] Wai Pang Ng et al. proposed aminimum adaptation step-size mechanism to reduce the seperation of voice signal with the help of small step-sizes at undesired time and is measured with perfect divergence detector (PDD) by utilizing wiener weights provides best development.In [12] S. Abolfazl Hosseini et al. demonstrated an modern adaptive filtering technique by using VSS adaptive filter for elimination of distortion in the development of voice which provides speed meeting time and less steady state mean square fault.In [13] M. Senthil Kumar et al. explained atechnique based on leaky VSS used in speech analysis.In [14] Salvador Olmos et al. described two principles are analyzed in favor of block-wise method of wave, the way out of sharpest reduction plan for decreasing the MSE. Also BLMS is alike the block recursive least squares(BRLS).In [15] Jingen Ni et al. investigated on a technique to bring back the strengths of the subband scheme. In [16] Hsu-Chang Huang et al. demonstrated a statistic variable step-size normalized least mean-square (VSS-NLMS) technique that presents the mean-square fault along with the guessed fault strength to manage the step-size inform which results in better accord with examined outcomes. In [17] Yunfei Kan et al. discussed a modern changeable step LMS technique which leveled the step parameter in time area while processing of iteration so that it removes the control on the total drift when modernizing the step and found better results.In [18] Liu Wei et al. enlightened a modern strong voice identification technique mixing adaptive filtering related on changing the step size LMS method along with intuitive uniform calculation cepstral guessing which determines the concert of this technique is very good than that of the voice identification way. In [19] Vinayshankar et. al. justified a technique which finds the good correlating fraction of the artifact wave in clattery voice and can be utilized as basic for voice development in adaptive ways and also found VSSLMS and Recursive Least Squares (RLS) techniques.In [20] M. Raghu Ram et al. described an easy and capable advance with the help of adaptive step LMS to lessen the motion artifacts at distorted photoplethysmographic (PPG) waves competently. In recent times, because of the usage of high data rates in mobile communications and video calls, the speech signal enhancment techniques should have less computational complexity. If the computations of the speech signal enhancment algorithm is huge, the samplesof the input data overlap at the input end of the speech signal

analysis block and t leads to inter symbol interference (ISI). This creates ambiguity in the analysis of speech signal at the receiver end. Therefore, the number of computations essential forthe enhancement of speech signal processing is also a key aspect. So for to the best of authors knolwedge this aspect is not presented in literature in the aspect of removal of undesired noise from speech signal. So, in our work we targeted to reduce the computational compexity of the noise removalapproach with the aid of combining the proposed SVLMS with Sign Regressor mechanism.Various experimnets are achieved to enhance speech signals using SVLMS, SRSVLMS and the performance is compared with reference to the conventional LMS based speech enhancement technique.

II. ADAPTIVE ALGORITHMS FOR SPEECH SIGNAL ENHANCEMENT

In order to eliminate the noise components, form the speech signals we propose a Speech Enhancement Unit (SEU) based on adaptive algorithm in our work. The main theme of this SEU is the proposed adaptive algorithm for the operation of speech enhancement minimizes the noise contamination in the actual speech signal. Figure 1 shows a schematic sketch of SEU for analysis. The input to the SEU is the recorded speech signal with noise contamination. The noise tester measures type of contamination using power spectral estimation and choose a correlated reference signal. If the correlated reference signal is not available then a random signal is given reference signal to the SEU. The adaptive algorithm used in the filtering process has the innate ability to change the filter coefficients to train the reference signal such that it becomes close to the noise contamination buried in the speech signal statistically. This process finally facilitates high resolution speech signal without noise contaminations. Based on the steepest descent algorithm [21], $\nabla x(n)$.

$$\mathbf{v}(n+1) = \mathbf{v}(n) - Q\nabla x(n) \quad (2)$$

$$x(n) = E[m^2(n)]$$

The steepest descent algorithm then becomes

$$\mathbf{v}(n+1) = \mathbf{v}(n) - Q\nabla x(n)$$

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(3)

Wherever $x(n) = m^2(n)$

The weight update recursion for the LMS algorithm is.

$$\mathbf{v}(n+1) = \mathbf{v}(n) + Q\mathrm{m}(n)\mathbf{f}(n) \quad (4)$$

Where, $\mathbf{v}$(n) is weight vector, Q is step size, m(n) is error component, $\mathbf{f}$(n) is data vector.

The mathematical expression for the Sign Regressor LMS (SRLMS) is,

$$\mathbf{v}(n+1) = \mathbf{v}(n) + Q\mathrm{m}(n)sign(\mathbf{f}(n)) \quad (5)$$

In critical situations when the signal power is varying with respect to input sample value, and varies with respect to the instantaneous value. In such scenarios, Step Variable LMS (SVLMS) is a better candidature. In this algorithm the step size parameter varies with respect to the instantaneous time value. The analysis of this algorithm is presented in [20]. Step size of the proposed filter is updated at each step as,

$$\mathbf{v}(n+1) = \mathbf{v}(n) + \rho \times \mathbf{f}(n) \times \mathrm{m}(n) \times \gamma(n) \quad (6)$$

Here ρ is a small positive constant and $\gamma(n)$ is defined as the partial derivative of tap weight vector.

$$\gamma(n) = \frac{\delta \mathrm{P(n)}}{\delta \mathrm{s(n)}}$$

The sign regressor version of this algorithm is written as,

$$\mathbf{v}(n+1) = \mathbf{v}(n) + \rho \times sign(\mathbf{f}(n)) \times \mathrm{m}(n) \times \gamma(n) \quad (7)$$

This SR is computationally less complex and the number of multiplications required for carrying the noise cancellation process.

III. EXPERIMENTAL RESULTS AND DISCUSSION

In order to test the ability of the speech enhancement units using SVLMS and SRSVLMSalgorithms inpractical we developed two SEUs and experimentation is carried. In all the SEUs the tap length is chosen as ten. In this experiment initially the concept of noise cancellation is proved by applying additive Gaussian noise and then several speech signals with real noise are applied. To prove the ability of the proposed adaptive algorithms speech signals are chosen for filtering. For that purpose, five sample speech signals are taken from the data base. Both synthetic and real noises are taken to prove the performance analysis of the proposed adaptive algorithms and the non-stationary tracking performance of the algorithms. These speech signals are contaminated with various types of noises namely cockpit noise, elevator noise, high voltage murmuring, gun firing noise and random noise. The experimental results for cockpit noise in battle filed scenario are shown for all the five considered samples in the figures 2-6. The performance of these techniques is calculated in terms of signal to noise ratio improvement (SNRI) and is tabulated in Table I. The comparison of calculated SNRI is shown in Figure 7.

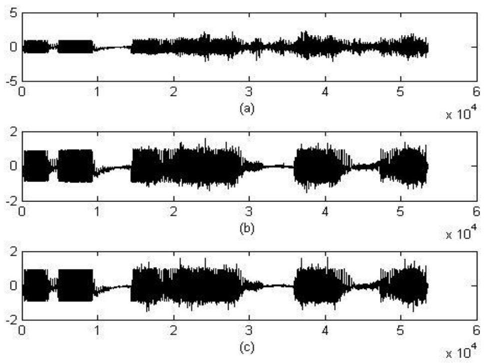

Fig.2: Experimental results during Speech Wave-I enhancement, (a). Speech wave with cockpit noise, (b). enhanced wave using SVLMSbased Speech Enhancement Unit, (c). enhanced signal using SRSVLMS algorithm-based Speech Enhancement Unit.

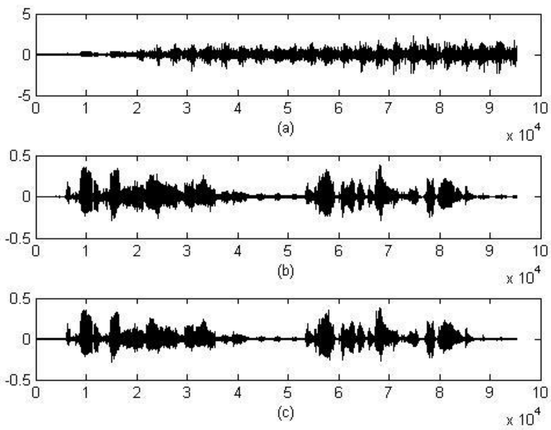

Fig.3: Experimental results during Speech Wave-II enhancement, (a). Speech wave with cockpit noise, (b). enhanced wave using SVLMSbased Speech Enhancement Unit, (c). enhanced signal using SRSVLMS algorithm-based Speech Enhancement Unit.

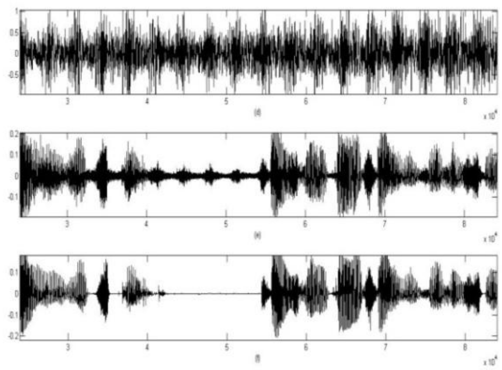

Fig.4: Experimental results during Speech Wave-III enhancement, (a). Speech wave with cockpit noise, (b). enhanced wave using TVLMS based Speech Enhancement Unit, (c). enhanced signal using SRTVLMS algorithm-based Speech Enhancement Unit.

Table I: SNRImprovementcomputations for the proposed Speech Wave Enhancement based on LMS, SVLMS and SRSVLMS algorithms (all values are in dBs)

S.No	Noise type	Sample	LMS	SVLMS	SRSVLMS
1.	Cockpit Noise	Speech Wave-I	8.5795	26.3782	24.3911
		Speech Wave-II	8.1673	26.4882	24.3209
		Speech Wave-III	8.8705	26.4979	24.2390
		Speech Wave-IV	8.3691	26.5973	24.4364
		Speech Wave-V	8.7753	26.9387	24.7656
2.	Elevator Noise	Speech Wave-I	6.1468	24.3291	22.5468
		Speech Wave-II	6.3617	24.4382	22.3239
		Speech Wave-III	6.5784	24.3902	22.3490
		Speech Wave-IV	6.8665	24.2198	22.2129
		Speech Wave-V	6.0346	24.8650	22.4398
3.	High Voltage Murmuring	Speech Wave-I	5.6385	23.6529	21.1296
		Speech Wave-II	5.1893	23.3996	21.2167
		Speech Wave-III	5.3866	23.3243	21.1275
		Speech Wave-IV	5.9582	23.9845	21.4476
		Speech Wave-V	5.7418	23.3243	21.4936
4.	Gun Firing Noise	Speech Wave-I	7.9127	25.2369	23.9302
		Speech Wave-II	7.0836	25.7649	23.2940
		Speech Wave-III	7.3353	25.2795	23.2144
		Speech Wave-IV	7.7538	25.8120	23.4929
		Speech Wave-V	7.5253	25.2193	23.3448
5.	Random Noise	Speech Wave-I	9.0314	27.3498	25.2345
		Speech Wave-II	9.9735	27.2192	25.3454
		Speech Wave-III	9.2946	27.4910	25.3241
		Speech Wave-IV	9.5904	27.2183	25.2343
		Speech Wave-V	9.7733	27.4198	25.2484

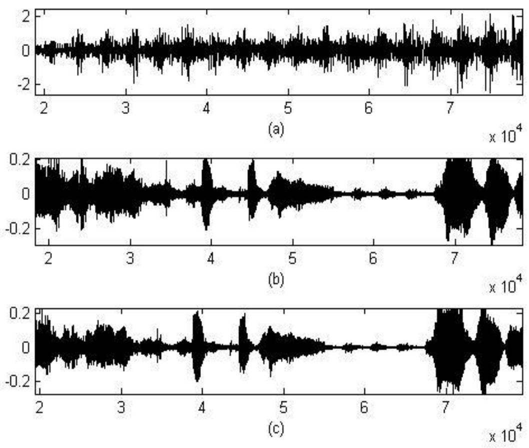

Fig.5: Experimental results during Speech Wave-IV enhancement, (a). Speech wave with cockpit noise, (b). enhanced wave using SVLMS based Speech Enhancement Unit, (c). enhanced signal using SRSVLMS algorithm-based Speech Enhancement Unit.

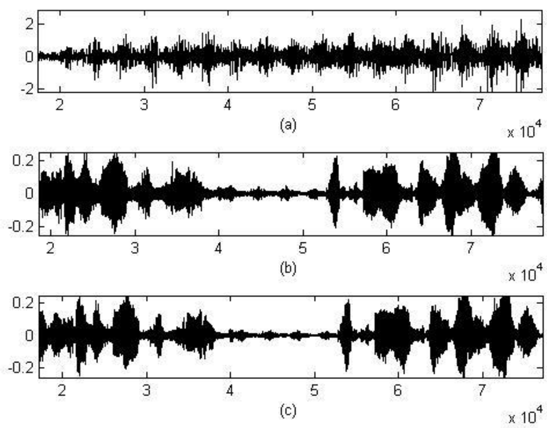

Fig.6: Experimental results during Speech Wave-V enhancement, (a). Speech wave with cockpit noise, (b). enhanced wave using SVLMS based Speech Enhancement Unit, (c). enhanced signal using SRSVLMS algorithm-based Speech Enhancement Unit.

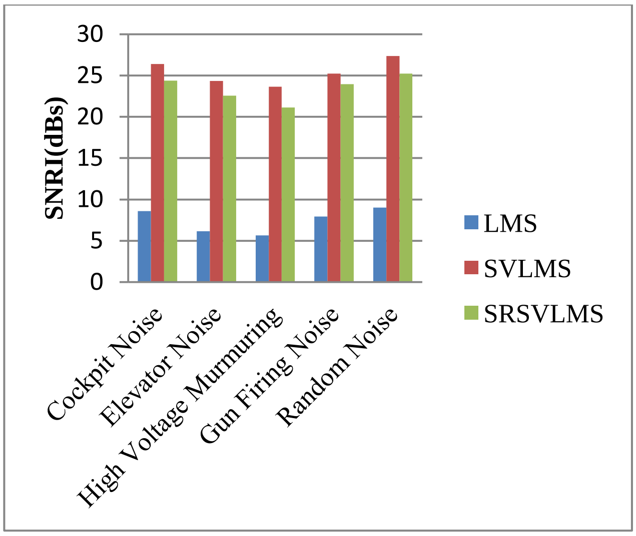

Fig.7: Comparison of performance measures in speech signal filtering due to various adaptive filters.

IV. CONCLUSION

This research comprises of adaptive noise cancellation of voice waves by overcoming the composition of different categories of noises. With selected step-size the conventional LMS technique changes the gradient noise component. In order to cope up the drawbacks associated with constant step size parameter, in this work a variable step size parameter with respect to time variable is developed in the contest of speech signal enhancement. To overcome the problem of inter symbol interference at the input port of the noise canceller, a strategy is applied to minimize the computational burden of the noise canceller. This is possible by developing the hybrid realization of adaptive step variable LMS and sign regressor algorithm. Therefore, in our realizations, we developed SVLMS and SRSVLMS algorithms for the development of proposed speech signal enhancement. The experimental results carried during the noise elimination process are shown in Figures 2-6, the calculated signal to noise ratio improvement in these experiments are showed in Table I. From these results it is clear that SRSVLMS based SEU achieved little bit inferior SNRI than SVLMS. But, due to the signum function applied to the data vector we are able to minimize number of multiplications equal to the tap length of SEU. Therefore, based on this it is conclude that SRSVLMS based SEU performs better than SVLMS as well as conventional LMS. Hence, SRSVLMS based SEU can be used in all practical applications in real time environment.

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