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Noise Cancellation and Speech Enhancement for Hearing Impaired Person

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Abstract: A major complaint of individuals with normal hearing and hearing impairments is a reduced ability to understand speech in a noisy environment. This paper is proposed to remove the noise from the speech signal in various real time environments and to make the hearing impaired to understand the information by transforming speech in to text format. The quality of audio signal can be improved by filtering the degraded speech signal through adaptive filters. For Noise cancellation widrow & hoff's Least Mean Square (LMS) algorithms are being used for simplicity in implementation. The LMS algorithm was simple in it's derivation and robust in a number of applications but it also have limitation in selection of a certain values such as step size and results in computational complexity. Hence to overcome all the limitations and to present an enhanced speech signal, a novel approach named VSSNDLMS algorithm is proposed. The performance of algorithms like Variable Step Size Normalised LMS , Normalised Differential LMS with proposed VSSNDLMS with different input signals are analysed in this paper . Finally, through simulation results the proposed VSSNDLMS algorithm converges fastly with minimum mean square error and it is useful in predicting the adaptive filter performance of various algorithms and the implementation indicate the improvement in quality of the speech signal and it appears to be favourable for hearing impaired.

Keywords: VSSNDLMS, LMS Algorithm, Noise Cancellation, Speech Processing, Adaptive Filter

I. INTRODUCTION

Hearing impairment is not only the most prevalent communicative disorder, it is also the number one chronic disability affecting people in the UnitedStates. A major complaint of those with hearing impairments is a reduced ability to understand speechin everyday communication in a noisy environment. Even with the absence of hearing impairment, the addition of background noise can signifiantly reduce the intelligibility of speech. In order to overcome all such complaints the concept of adaptive filters is introduced. Adaptive filter is a digital filter that has self-adjusting characteristics which can adjust its filter coefficients automatically and gets adapted to the input signal. Adaptive filters work generally for adaptation of signal-changing environments like spectral overlap between noise and signal, noise were presented which included adaptive noise cancelling and noise suppression like telephone echo cancellation, equalization of communications channels, biomedical signal enhancement, active noise control, and adaptive control systems. Adaptive filtering is one of the approaches used to remove the noise from the desired signal. Adaptive filtering (AF) finds application in noise cancellation in speech called as Adaptive Noise cancellation (ANC) which involves in time-varying signals and systems. In adaptive filter least mean square (LMS) algorithm is the most popular algorithm. Because of its simplicity, robustness, and low computational complexity, it has been widely used in noise cancellation, linear prediction and so on. Least Mean Square (LMS) algorithm is one of the well-known adaptive algorithms which is based on stochastic gradient approach.

Various adaptive algorithms have been proposed for noise cancellation. In 1956, Widrow pro-posed an adaptive filter which can be used to reduce interference when a second sample of the noise is available. This technique was developed at Stanford University in 1959 and appliedto a pattern-recognition scheme known as Adaline.In 1965 the first adaptive noise cancelling system was built by two students at Stanford University.In 1972, the first all-digital adaptive filter was built by McCool and Widrow at the Naval Undersea Center in Pasadena, California. R.Bilcu et.al has proposed a new variable length LMS algorithm theoritical and implemented [4],[5]. The sub band adaptive filters have been proposed [7] to analyse the convergence and complexity in adaptive filters. Gorriz et.al, proposed a novel LMS algorithm [9] for filtering speech sounds in the adaptive noise cancellation. J. E. Greenberg has modified the LMS algorithm and used to cancel the noise in speech signal [8]. Y.Gong and C.Cowan proposed a variable tap-length algorithm based on MSE output from different filter segments, which combines the traditional segmented filter approach with a gradient descent-based method [10]. Krstajic. B et.al proposed a combination of the LMS based algorithms, which results in an adaptive system that takes the favourable properties of these algorithms [13].All manuscripts must be in English. These guidelines include complete descriptions of the fonts, spacing, and related information for producing your proceedings manuscripts.

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II. ADAPTIVE ALGORITHMS

Adaptive signal processing in variable and noisy non stationary environments is usually accomplished by means of adaptive filters. In these adaptive filters, it needs two inputs to remove noise; 1) primary input needs to obtain noise-corrupted speech and 2) reference input needs to obtain component of noise included primary input. Signal of reference input make noise signal component through passing adaptive filter to remove the noise of included primary input. The correlation between the additive noise that corrupts the clean speech (primary signal) and the random noise in the reference input(adaptive input) is necessary to adaptively cancel the noise of the primary signal. In this paper, the various LMS adaptive algorithms were analysed and a novel algorithm named Variable Step Size Normalized Difference LMS (VSSNDLMS) algorithm is proposed to remove the noise in signal recorded in various environments like inside a theatre or in an auditorium.

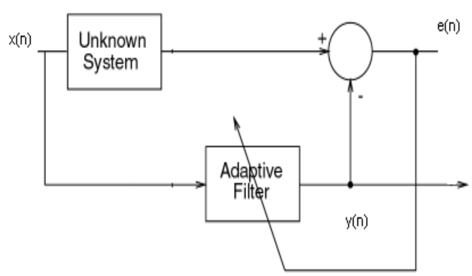

Figure 1: System identification

A. LMS Algorithm:

The Least Mean Square algorithm is one of the simplest algorithms used in the adaptive structures. It is based on the minimization of the the squared Euclidean norm of the difference weight vector under a Stability constraint over the aposteriori estimation error. LMS algorithm expressions for finding the error signal and the filter weights.

$$e(n)=d(n)- \mathbf{W}^T(n) * u(n). \tag{1}$$

$$W(n+1)=W(n)+\mu*u(n)*e*(n). \tag{2}$$

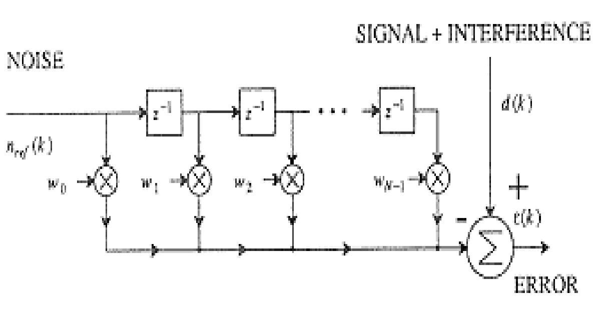

Figure 2: An Adaptive LMS Filter

The widely used least-mean-square (LMS) algorithm has been successfully applied to many ltering applications, including signal modeling, equalization, control, echo cancellation, biomedicine, or beam forming.

B. NDLMS Algorithm:

Many ANCs have been proposed in the past years using modifed LMS algorithms in order to simultaneously im-prove the tracking ability and speed of convergence. Bershad has studied the performance of the normalized LMS (NLMS) algorithm with an adaptive step size showing advantages in convergence time and steady state. Later J Zhang and H.M.Tai made a slight modification in LMS algorithm equations and they proposed the Normalised Differential LMS Algorithm. It is mainly used for finding the error signal and the filter weights. In this case the algorithm improves the steady state performance for cancelling noise in speech processing [15].

$$e(n)=d(n)- \mathbf{W}^T(n) *x(n)*d(n). \tag{3}$$

$$w(n+1)=w(n)+\mu\times\nabla x(n)\times\nabla d(n) \tag{4}$$

C. VSSNLMS Algorithm:

Adaptive filtering has been, and still is, an area of active research that plays an important role in an ever increasing number of applications, such as noise cancellation, channel estimation, channel equalization and acoustic echo cancellation. The least mean square (LMS) and its normalized differential version (NDLMS) are the workhorses of adaptive filtering. In the presence of various corrupted input signals, the LMS and the NDLMS algorithms have extremely slow convergence rates. In order to overcome the problem of convergence speed and estimation accuracy in real time environment, the variable step size normalised LMS algorithm is proposed. The **LMS** type adaptive algorithm is a gradient search algorithm which computes a set of weights $\boldsymbol{W_k}$ that seeks to minimize $\boldsymbol{E\ (d_k - X_k^T\ W_k)^2}$**,** The algorithm is of the form

$$w_{k+1}=w_k+\mu_k X_k \varepsilon_k \tag{6}$$

Where,

$$\varepsilon_k = d_k - X_k^T W_k \tag{7}$$

And μ_k is the step size. In the standard **LMS** algorithm [1], μ_k is a constant. In [9], μ_k is time varying with its value determined by the number of sign changes of an error surface gradient estimate. Here, we propose a new algorithm, which we shall refer to as the variable step size or **VSS** algorithm, for adjusting the step size μ_k :

$$\mu'_{k+1}=\alpha\mu_k+\gamma\varepsilon_k^2 \tag{8}$$

with

$$0<\propto<1,\quad \gamma>0$$

and

$$\mu = \begin{cases} \mu_{\min} \ldots ifSNR(n) > SNR_{\max} \\ \mu_{man} \ldots ifSNR(n) > SNR_{\min} \\ aSNR(n) + b \ldots ifSNR_{\min} \leq SNR(n) \geq SNR_{\max} \end{cases} \quad (9)$$

Where $0 < \mu_{\min} < \mu_{\max}$. The initial step size μ_0 is usually taken to be $\mu_{\max}$ **a**lthough the algorithm is not sensitive to the choice. As **it** can be seen from (8),the step size μ_k is always positive and is controlled by the size of the prediction error and the parameters $\propto$ and γ. Intuitively speaking, a large prediction error increases the step size to provide faster tracking. If the prediction error decreases, the step size will be decreased to reduce the misadjustment.

III. PROPOSED ALGORITHM

The features of NDLMS and VSSNLMS are combined together and an efficient algorithm called Variable Step Size Normalized Differential LMS (VSSNDLMS) algorithm is proposed to enhance speech processing for hearing impaired. The objective of proposing this algorithm is to design an effective adaptive filter to remove the noise and to improve the quality of speech signal.

In case of LMS algorithm, under non-stationary environment some errors occur leading to deviation of filter weights from the optimal weight of the filter. It is efficient to have lesser value of SNR because such a value gives the maximized step size that provides faster tracking. At the same time, the larger value of SNR results in minimized step size producing smaller mis-adjustment. Hence the step size must be controlled to the requirement. The proposed algorithm satisfies this criteria by adjusting the step size. The variable step size LMS algorithm converges fastly and NDLMS algorithm have minimised mean square error. By combining the VSS and NDLMS, the VSSNDLMS algorithm converges fastly with minimum mean square error.

As per the VSSNDLMS algorithm, the expression for updating the coefficient is given by

$$w(n+1) = w(n) + \frac{\mu_{\text{var}}}{\varepsilon + \|\nabla X(k)\|^2} \nabla x(n) * \nabla e(n) \quad (10)$$

Where ,

$$\nabla e(k) = e(k) - e(k-1) \quad (11)$$

$$\nabla x(k) = x(k) - x(k-1) \quad (12)$$

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

$$\mu^{\text{var}} = \begin{cases} \mu_{\min} \ldots ifSNR(n) > SNR_{\max} \\ \mu_{man} \ldots ifSNR(n) > SNR_{\min} \\ aSNR(n) + b \ldots ifSNR_{\min} \leq SNR(n) \geq SNR_{\max} \end{cases} \quad (13)$$

The purpose of the adaptive filter is to minimize the Mean Square Error MSE.

$$MSE = E\left\{\| d(n) - \hat{d}(n)\|^2\right\} \quad (14)$$

Therefore, the graph of MSE quantity is essential to evaluate the performance of the adaptive filter.

IV. EXPERIMENTAL RESULTS

The theoretical results presented in this paper are justified by computer simulations using Matlab 7 version. The comparison results of two speech signals recorded in various environments with NDLMS, VSSNLMS and VSSNDLMS Algorithm are presented in this section. Figure 3 shows the responses of the input speech signal which is recorded inside an auditorim. Initially the speech signal with noise is presented which is then compared with the proposed VSSNDLMS output. It is evident that maximun noise is removed at this stage. Similarly Figure 4 shows the responses for another speech signal which is recorded in theatre environment. Two different speech signals are considered to analyse the mean square errors found using all the above mentioned existing algorithms. The comparison of MSE for LMS, NDLMS and VSSNDLMS algorithms is shown in table 1.

The results shows that VSSNDLMS algorithm have least mean square error and converges effectively thus gives better performance when compared to all other algorithms.

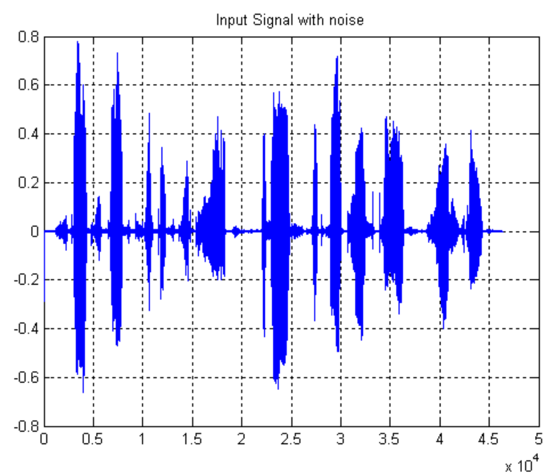

Figure 3 : Input speech signal with street noise level 1

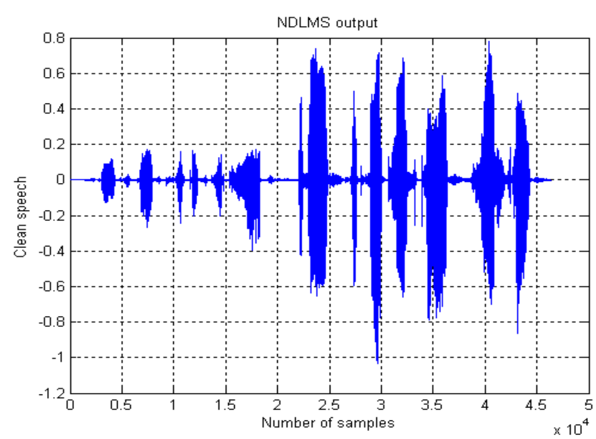

Figure 4 : NDLMS response for Input speech signal with street noise level 1

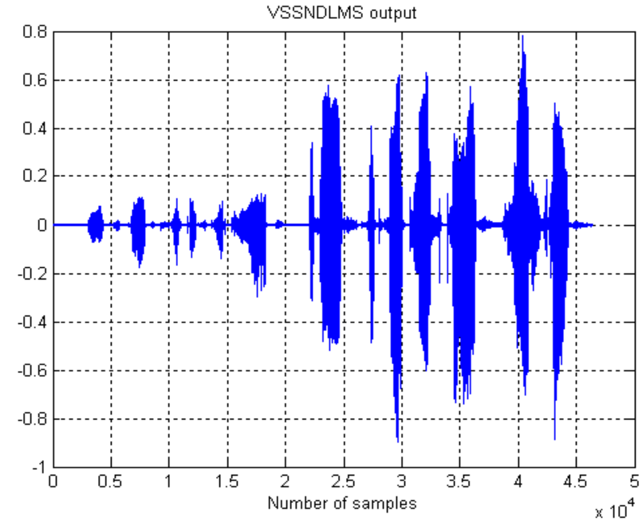

Figure 5 : VSSNDLMS response for Input speech signal with street noise level 1

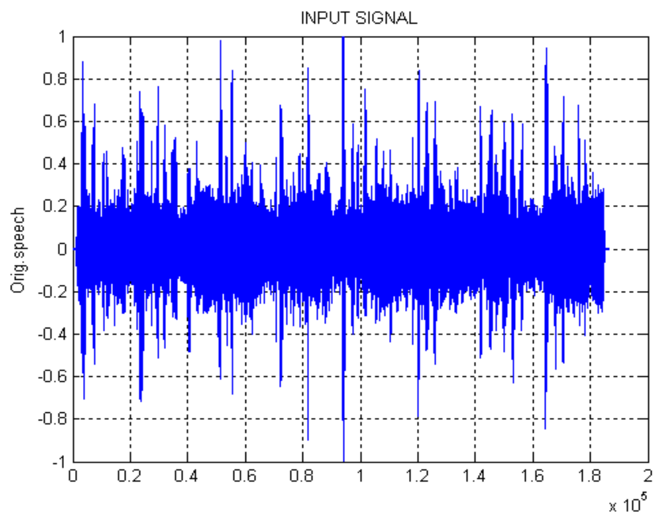

Figure 6 : Input speech signal with street noise level 2

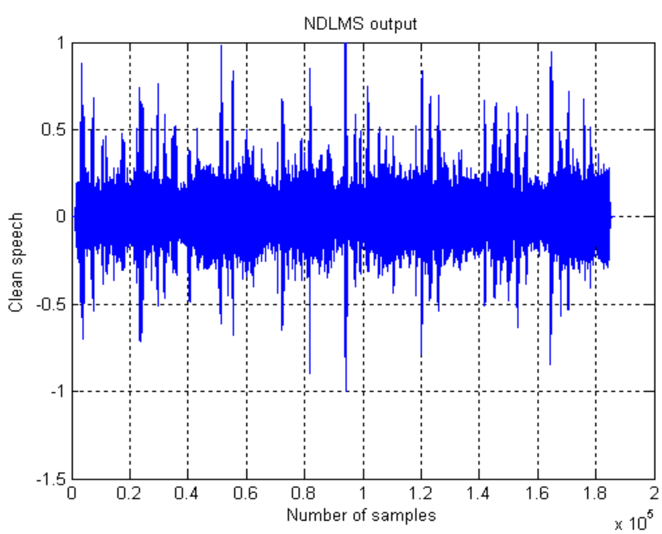

Figure 7 : NDLMS response for Input speech signal with street noise level 1

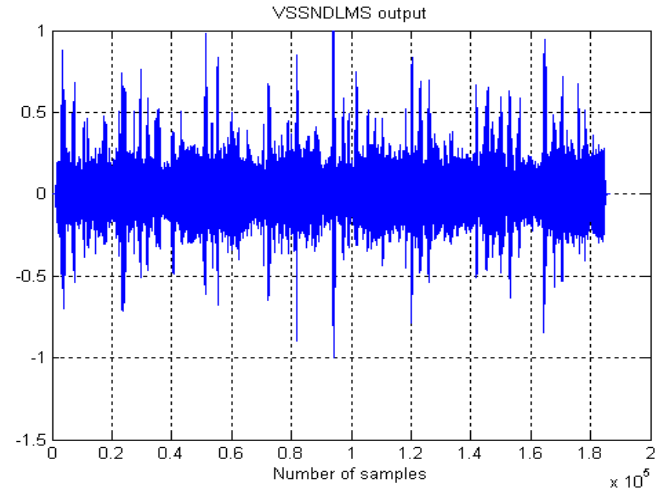

Figure 8 : VSSNDLMS response for Input speech signal with street noise level 1

Table 1: Comparison of Mean Square Error

Variance	Mean Square Error		
	LMS	NDLMS	VSSNDLMS
0.01	0.266	0.10	0.104
0.1	0.529	0.21	0.16
1.0	0.6714	0.354	0.31

V. CONCLUSION

In this paper we have applied VSSNDLMS algorithm on adaptive noise cancellation setup to enhance speech signals for hearing impaired. The simulation results were compared with the classical adaptive filters such as LMS, NLMS, and VSSNLMS algorithms for attenuating noise in speech signals. In each algorithm the output of filter were analysed. The simulation results show that the convergence rate of this algorithm is comparable with the existing algorithms. Also, the optimum values of the VSSNDLMS algorithm were calculated through experiments. In this algorithm, the number of iterations to be performed at each new sample time is a user selected parameter giving rise to an attractive and explicit tradeoff between convergence/tracking properties and computational complexity. Our Future enhancement is to analyse the VSSNDLMS by varying the parameters and to compare with other adaptive algorithms.

VI. REFERENCES

[1]. R.Bilcu, P.Kuosmanen, K.Egiazarian, “A new variable length LMS algorithm: theoretical analysis and implementations”, 9th International Conference on Electronics, Circuits and Systems, Vol.3, 2002, pp.1031–1034.

[2]. G.M. Vetterli, “Adaptive filtering in subbands with critical sampling: analysis, experiments, and applications to acoustic echo cancellation”, IEEE Trans. Signal Process. 40 (August 1992) 1862–1875.

[3]. J. E. Greenberg, "Modified LMS algorithms for speech processing with an adaptive noise canceller," IEEE Trans. Speech audio process vol 6, no4, pp 338-358 ,july 1998

[4]. Gorriz, J.M. Ramirez, J. Cruces-Alvarez, S. Puntonet, C.G. Lang, E.W. Erdogmus, "A Novel LMS algorithm applied to adaptive noise cancellation" IEEE signal Processing Letters Vol 16, pp 34-37.

[5]. Y.Gong,C.Cowan, "A LMS style variable tap-length algorithm for structure adaptation", IEEE Transactions on Signal Processing, 2005,pp 2400–2407.

[6]. Krstajic. B, Stankovic. Lj, Uskokovic.Z, Combined Adaptive Filter with LMS-Based Algorithms, Int. J. Electronics and Communications. (AEU) 57 (2003) No. 4, 295−299.

[7]. Jiashu Zhang and Heng-Ming Tai, "Adaptive Noise Cancellation Algorithm for Speech Processing", The 33rd Annual Conference of the IEEE Industrial Electronics Society (IECON) Taipei, Taiwan.

[8]. QU Yan-bin, MENG Fan-gang , GAO Lei, "A New Variable Step Size LMS Adaptive Filtering Algorithm" IEEE International Symposium on Industrial Electronics, 2007.

[9]. J.P Joonwan Kim, J. Lee, S. Kim, C. Lee, "Performance analysis of adaptive noise canceller in real-time automobile environments", 39th South eastern Symposium on System Theory, Mercer University, March 2007.

