ISSN: 0974-5823 Vol. 7 No. 4 April, 2022

International Journal of Mechanical Engineering

ENHANCEMENT OF SPEECH SIGNAL BY USING VARIABLE STEP SIZE NORMALIZED DIFFERENTIAL LMS (VSSNDLMS) ALGORITHM

G. Sunil Kumar[1], Dr. S.M. Ramesh[2], Dr. A. A. Ansari[3]

[1,2]Department of Electronics and Communication Engineering

[1]Research Scholar, SSSUTMS, Sehore, MP, India.

[2]Professor, KPRIET(Autonomous), Coimbatore, India.

[3]Professor, Department of EEE, SSSUTMS, Sehore, MP, India.

Abstract - Different adaptive filters, such as LMS and Normalized LMS are used to minimize the speech signal noise, in order to achieve steadiness. To increase the SNR of the method output through a faster convergence time, filter weights are varied according to the unique behavior. This research shows a new strategy based on the variable step size LMS (VSSLMS) to decrease the noise in distorted voice by utilizing LMS algorithm, which is the fundamental adaptive algorithm even though it has the major drawback of larger mean square error which enhances linearly along with required signal power. The Variable Step Size Normalized Differential LMS algorithm (VSSNDLMS) was implemented in the present paper to wipe out the noise in the speech. The outcomes reveal the existing VSSNDLMS algorithm which has reduced steady-state excess mean square error (MSE). It is also constantly progressing SNR over an extensive variety of SNR-inputs. In addition to this range, the projected technique provides additional stable presentation and improves convergence speed which can reduce steady state error. It also provides extremely minor error values between the weights of the filter and target channel.

Keywords — *LMS, Normalized Differential LMS, SNR improvement,Variable Step Size LMS, MSE, Adaptive Filter*

I. INTRODUCTION

Speech is usually noise-influenced. Acoustic noise elimination is the most significant cleaning applications for noise speakers with adaptive filters. In 1975, Widrow was regarded as the first to present the concept of adaptive filters [1]. The aim was to remove the filtered input noise from the loud, damaged voice. This approach controls the subtraction processes, without any previous knowledge of noise signals, by preventing the processes of noise amplification or voice attenuations.

In noise cancellation operation the major problems concerns with different signal works i.e., voice recognition, speakers' verification and speech recognition etc. due to background noise and other different noises. The various methods implemented to nullify noise produced by speech signal i.e., linear and non-linear filtering, adaptive cancellation of noise for variations. Speech enhancement assists to enhance the quality of voice signal by decreasing background noise. Intelligibility, clarity and happy plays a vital factor in the quality of speech [5-7]. Mainly speech consists of recognition, interpretation, coding and synthesis improves the basic method in the field of speech processing. Speaking signals are mutilate with short duration's noises i.e., an impulsive noise in the communications systems [8].

The above-mentioned interferences are exclusively disagreeable to listeners which should be wiped out speech signals which are intelligible. The maximum number of algorithms used for processing voice signals which are in integrated in their design which follow Gaussian distribution. Though the non-Gaussian probability is distinctive for noises viz. impulsive noise. The impulsive noise gradually decreases the quality of speech processing systems [9-10]. The present article analyses and compare various speech processing algorithms i.e. LMS, NLMS, VSSNLMS and their demerits applied to proposed algorithm (NDLMS) that enhances the present output of adaptive noise cancellers in the treatment of speech.

II. RELATED WORKS

Zayed Ramadan and et.al [11] implemented new vacillating step size LMS algorithms and its efficiency has examined in the expression of misadjustment (M) and large MSE in stationary and movable environment. They examined the efficiency of present algorithms which is far better than of NLMS algorithm.

Paulo A.C. Lopes [12] introduced a Kalman referenced normalized LMS algorithm for speech improvement and examined the efficiency of algorithm in OFDM equalization system. The present algorithm depicted from the benefits of both Kalman filter and NLMS algorithm which is expressed in the form of speed of convergence and stability.

Ching-Ta Lu and et.al [13] implemented the impact of excess and scarcity of estimation of noise while speech enhancement and also implemented the technology for enhancement of minima-controlled-recursive-averaging algorithm for noise calculation. Besides, many researchers are doing research in the arena of speech improvement and existing various algorithms referenced on statistical and signal processing techniques for enhancement of Signal to Noise Ratio (SNR), intelligibility, decreasing noise and Mean Square Error (MSE). The next will provide complete study of literature review on statistical and signal processing techniques.

M. Gorriz et al [14] proposed modified LMS algorithm referenced on reducing the Euclidean squared norm of a weight differential vector comes under stability limit implemented by a posteriori calculation error. The algorithm efficiency calculation is examined utilizing AURORA 2 and 3 speech databases.

Rey Vega L and et.al [15] proposed robust fluctuating step size NLMS algorithm. The algorithm referenced on minimizing the square of posteriori error. They examined the efficiency of algorithm utilizing various noise environments for noise cancellation and system identification besides theoretical model.

Ramsey Faragher [16] presented a review article on Kalman filter and its applications. The author describes computational requirement, recursive properties and using as optimal estimator for one-dimensional system with Gaussian noises.

III.PROBLEM STATEMENT

- LMS Drawback: LMS is direct forward in mathematical expression, because of stochastic & nonlinear implementation of filter, therefore its computational method is largely difficult and complex.
- The phase size is implemented to utilize NLMS algorithm. This phase size variation is monitored by SNR to calculate which is given by ratio between average power of original signal and signal power.

IV. MATERIALS AND METHODS ADAPTIVE ALGORITHMS

4.1 Adaptive Algorithms

Adaptive threshold algorithms can be thought as a method in which the parameter was using to filter signals change based on some criteria. The estimated average squared error or correlation is commonly used as a criterion for evaluation.

The adaptive filters are distinct in their parameters are dynamically adjusted to meet the demands of the application. In this context, an adaptive filter is unique that performs the approximation step on the fly. Efficient fixed-filter design normally requires the presence of a reference signal that is usually hidden during the approximation phase. Consider the Least Medium Squares Algorithms (LMS) in more detail. The LMS variations of two more algorithms are also analyzed here. NLMS and standardized variable-step size algorithms for minimum mean squares are the algorithms in question (VSS-NLMS).

Adaptive filtering algorithms illustrate the usage of implementation of signal difference. The mathematical squared error or correlation is basically executed by adaptive filters and are differed because of parameters changes to meet the needs. A filter that operates on-line calculation phase of fixed filter design that is fixed of efficiency. As of now, we will understand Least Medium Squares Algorithm (LMS) in vividly explained. In the present process, the estimation of LMS magnitudes of two algorithms such as Normalized Lowest Mean Squares (NLMS) & Standardized Variable-Step Size algorithms for Minimum Mean Squares (VSS-NLMS).

The elementary notion for acoustic noise removal is a device with a double sensor depicted in Figure1.

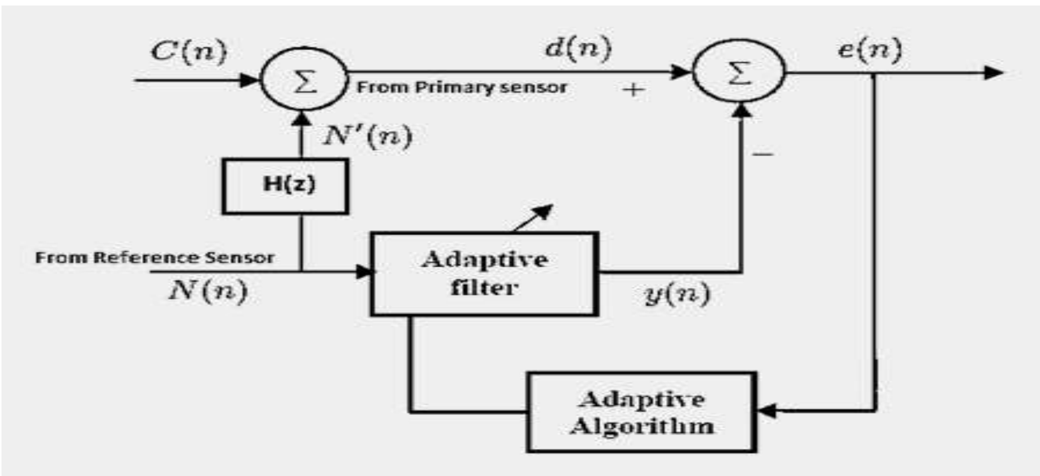

Figure.1 Application of speech enhancement using Adaptive Filter

The noisy speech from a main sensor is recognized and this is regarded as desired signal d(n), n represents the index of time. The transmitted clean C(n) with additive noise N´(n) are received from this sensor (n). The referral signal is regarded to be the filter input for the noise source N(n). Further, the channel H(z) replicates correlation among the source and actual noise additional to the speech, e(n) is the signal we improved or we can characterize our enhanced speech as

$$e(n) = d(n)-y(n) \tag{1}$$

$$e(n) = c(n)+N(n)-y(n) \tag{2}$$

Thus, MSE can be given as:

$$E[e^2(n)] = E[(N'(n)-y(n))^2] + E[C^2(n)] + 2E[C(n)(N'(n)-y(n))] \tag{3}$$

Where the expected value is meant by $E\{\cdot\}$.The third element in the equation is seen because neither N´(n) noise nor y(n) filter output has any link with clear speech C(n) .Furthermore, the constant energy and vitality of a clean signal is shown by the second term. To reduce the MSE, the y(n) filter must be close to the added noise N´(n) as possible. The vector of Filter weights W(n) = $[W_0(n), W_1(n)\ldots., W_{L-1}(n)]^T$ is a parameter that determine MSE and its order is L.

4.2Variable Step Size LMS Algorithm

ANC utilize VSSLMS technique as an Adaptive Filter to wipe out noise from discourse signal. Depending upon error squared force, a reduced VSSLMS algorithm is designed. The power mirrors intermix the situation of adaptive channel to join framework that has greater mistake during the intermixing framework which is more blunder power. In the same way, scalar step size reduces the squared blunder allowing adaptive channel to implement variations in the framework and gives greater modest consistent mistake.

In VSSLMS algorithm, the step is controlled by the square of error. A large mistake shall vary step size to increment to provide a less future error that shall reduce in step size to get modest maladjustment. By implementing the adaptive framework ID problem, channel loads are varied to get an ideal signal.

$$d_n = X_n^T W_n^* + e_n \tag{4}$$

Hence e_n is referred as autonomous grouping Zero Mean Gaussian with interaction of information X_n. As a result, there arise two conditions,

1. Which equals to a constant W,

2. Randomly varying W according to the prescribed equation.

4.3. Normalized Differential Least Mean Square Algorithm

ANC utilizing Normalized Differential LMS Algorithm is utilized as an AF to eliminate the noise from the discourse signal. For STT change model, noiseless discourse needed for STT Transformation.

In NDLMS Algorithm an alternate methodology is considered for weight change. The inspiration is to plan a LMS that can deal with both the solid and the feeble objective signs. Consequently, at whatever point the channel data sources and yields vary pretty much, the loads ought to be changed appropriately.

Some unfriendly impacts may happen when we utilize a LMS algorithm as portrayed in (1). Under the situation that the weight has been moving toward the ideal worth, generally it is huge for some time.

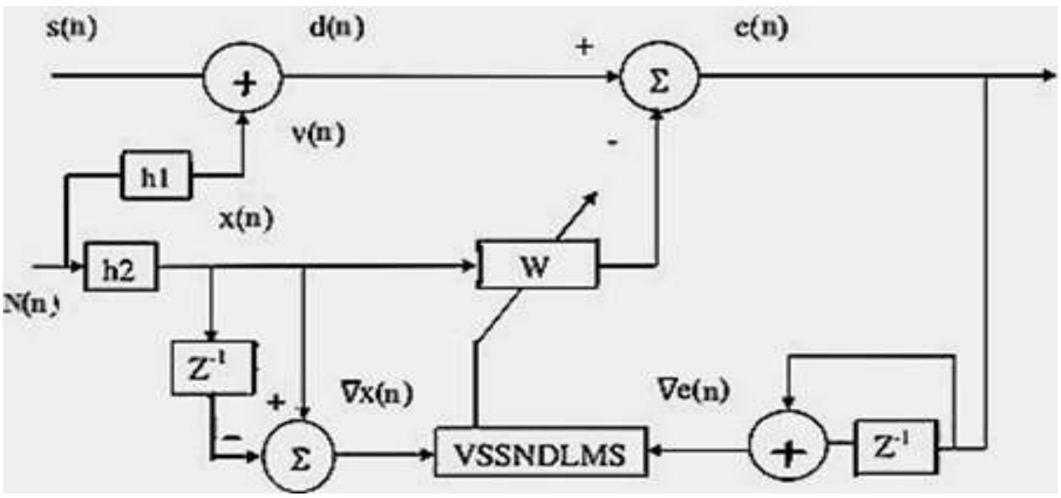

Figure.2 Projected VSSNDLMS Adaptive Filter

Condition (5) suggests that will stay away from the ideal load in a moderately huge way. This makes the adaptive loads vary around their ideal qualities for a generally longer period and it is beyond the realm of imagination to expect to make the consistent state MSE discretionarily little by diminishing the value.

It is principally utilized for finding the error signal and the channel loads. For this situation the algorithm improves the consistent state execution for dropping noise in discourse preparing. NDLMS Algorithm proposes to change the load as per the distinction of the signs, rather than the actual signs. Like the normalized LMS plot, the weight updates.

The weighs of Filter are employed to their optimum and best values done the iterations by using the algorithm of steepest decent technique [2]:

$$W(n+1)=W(n)+\frac{\mu}{\varepsilon+||\nabla x(n||)^2}\nabla x(n) * \nabla e(n) \qquad (5)$$

The instantaneous value substitutes for the expectation. When μ is represented as a variable step size which is given as

$$\mu = \begin{cases} \text{maximum value of } \mu \text{ if } \mu_i > \mu\ min \\ \text{minimum value of } \mu \text{ if } \mu_i < \mu\ min \end{cases} \qquad (6)$$

The aim of the research analysis is to reduce the Mean Square Error As a result, it is necessary to evaluate MSE of an adaptive Filter, which is calculated as

$$\text{MSE} = E\{||\, d(n) - đ(n)\, ||^2\} \qquad (7)$$

V. PROPOSED VARIABLE STEP SIZE NORMALIZED DIFFERENTIAL LMS ALGORITHM

ANC utilizing Variable Step Size Normalized Differential LMS Algorithm is utilized as an AF to eliminate the noise from the discourse signal. For STT change model, noiseless discourse needed for STT Transformation.

The NDLMS and VSSLMS are consolidated together and an improved productive algorithm called Variable Step Size Normalized Differential LMS (VSSNDLMS) algorithm is proposed to upgrade discourse handling. The target of proposing this algorithm is to plan a viable adaptive channel to eliminate the noise and to improve the nature of discourse signals. Figure 3.4 shows the proposed algorithm utilizing VSSNDLMS technique.

The Normalized Differential LMS Algorithm is especially appropriate for gradually changing signs and is less delicate to the craving signal force variety contrasted with the current Algorithms. Additionally, the abundance mistake and maladjustment by NDLMS are substantially less than that of existing Algorithms. VSSLMS Algorithm is utilized to diminish the compromise among maladjustment and following capacity of the fixed step size LMS Algorithm. The VSSLMS likewise diminishes affectability of the maladjustment to the degree of non- fixed. The highlights of NDLMS and VSSLMS are consolidated, VSSNDLMS Algorithm is proposed.

In the event of LMS algorithm under non-fixed climate, blunders happen which prompts deviation of channel loads from the ideal load of the channel. The proposed algorithm fulfills this standard by changing the step size. The VSSLMS algorithm combines quicker and NDLMS algorithm has negligible MSE. By joining the VSS and NDLMS, the VSSNDLMS algorithm combines quickly with MMSE. VSSNDLMS Algorithm are represented in the following as,

Step 1: Input the discourse signal in (*.wav) design
Step 2: Select the VSSNDLMS Algorithm
Step 3: Select the step size
Step 4: Update weight vector
Step 5: Compute information and yield
Step 6: Compute MSE
Step 7: Check blunder e(n) = 0, if off limits to step3
Step 8: Stop

VI. EXPERIMENTAL RESULT AND DISCUSSION

In this section we discuss about the performance of the proposed method with variable step size LMS and Normalized Differential LMS algorithm. The Experimental Setup of Adaptive Filter using Adaptive Filter Algorithm is shown in Figure.3, which consists of user voice Microphone, a sound signal in the *.wav format file, Proposed algorithm and the output as a speaker.

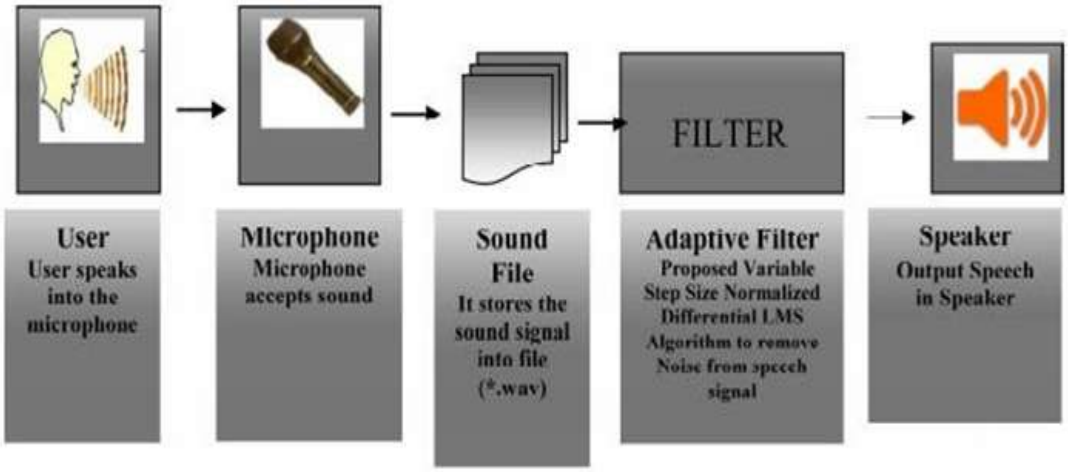

Figure.3**:** Experimental Setup of Adaptive Filter using VSSNDLMS Algorithm

Initially, VSSLMS Algorithm is utilized in Adaptive channels to remove Noise from the Speech signal. Speaker creates the yield of the discourse signal and reproduction results are performed utilizing MATLAB. Figure.4 and Figure.7 shows the reproduced MATLAB information, yield and MSE of the discourse signal.

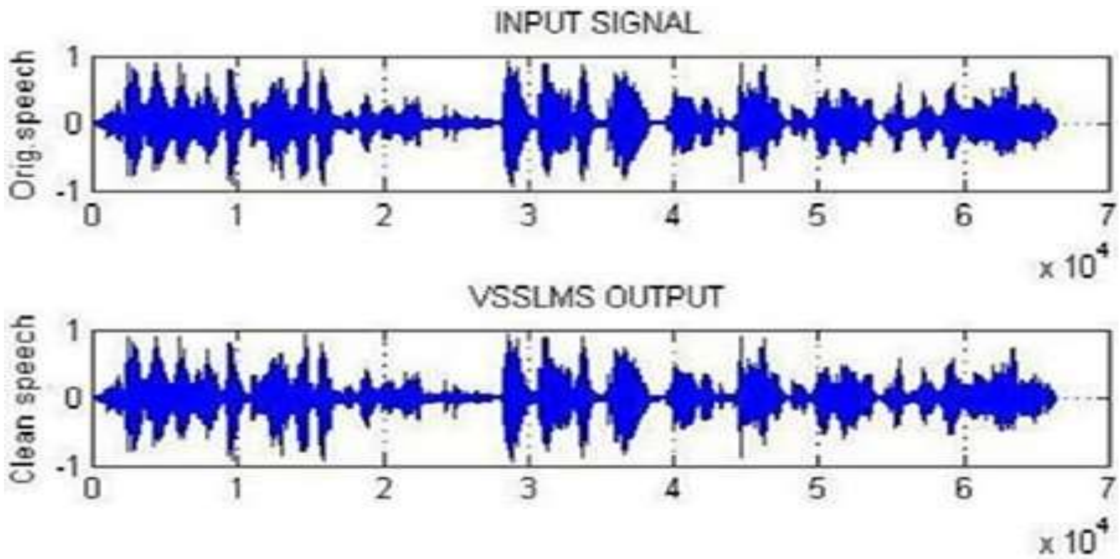

Figure.4**:** VSSLMS Output of Speech Signal (Input speech signal corrupted)

Succeeding the above processes, the discourse signal is evaluated with NDLMS Algorithm. Here NDLMS Algorithm is utilized in Adaptive channels to eliminate Noise. The speaker creates the yield of the discourse signal and recreation results are performed utilizing MATLAB, which are examined in the Figure.5, and Figure.8 shows the reproduced MATLAB input yield and MSE of the discourse signal.

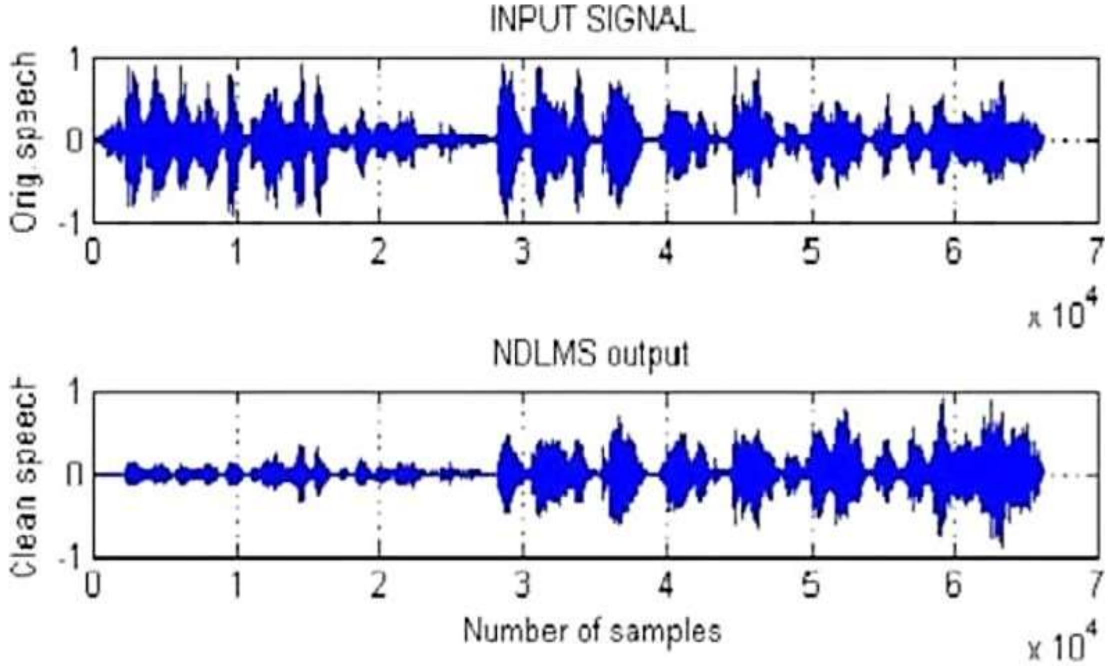

***Figure.5*:** NDLMS Output of Speech Signal (Input speech signal corrupted)

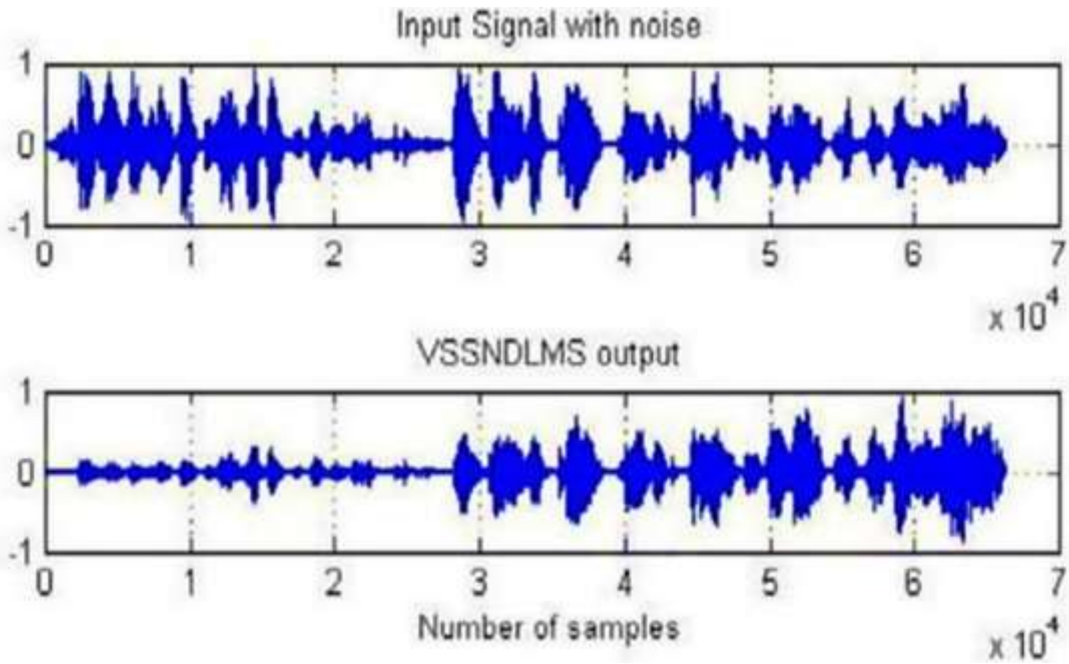

***Figure.6*:** VSSNDLMS Output of Speech Signal (Input speech signal corrupted)

Further the Discourse Signal is evaluated with VSSNDLMS Algorithm. Figure 6 and Figure 9 shows the square outline of ANC utilizing Improved Adaptive channel Based Noise cancellation Techniques for discourse signals. Here Proposed VSSNDLMS Algorithm is utilized in Adaptive channels to eliminate Noise from Speech signal. The speaker creates the yield of the discourse sign and reenactment results are performed utilizing MATLAB, which are examined in the Next Chapter. Figure 5 shows the reproduced MATLAB information, yield and MSE of the discourse signal.

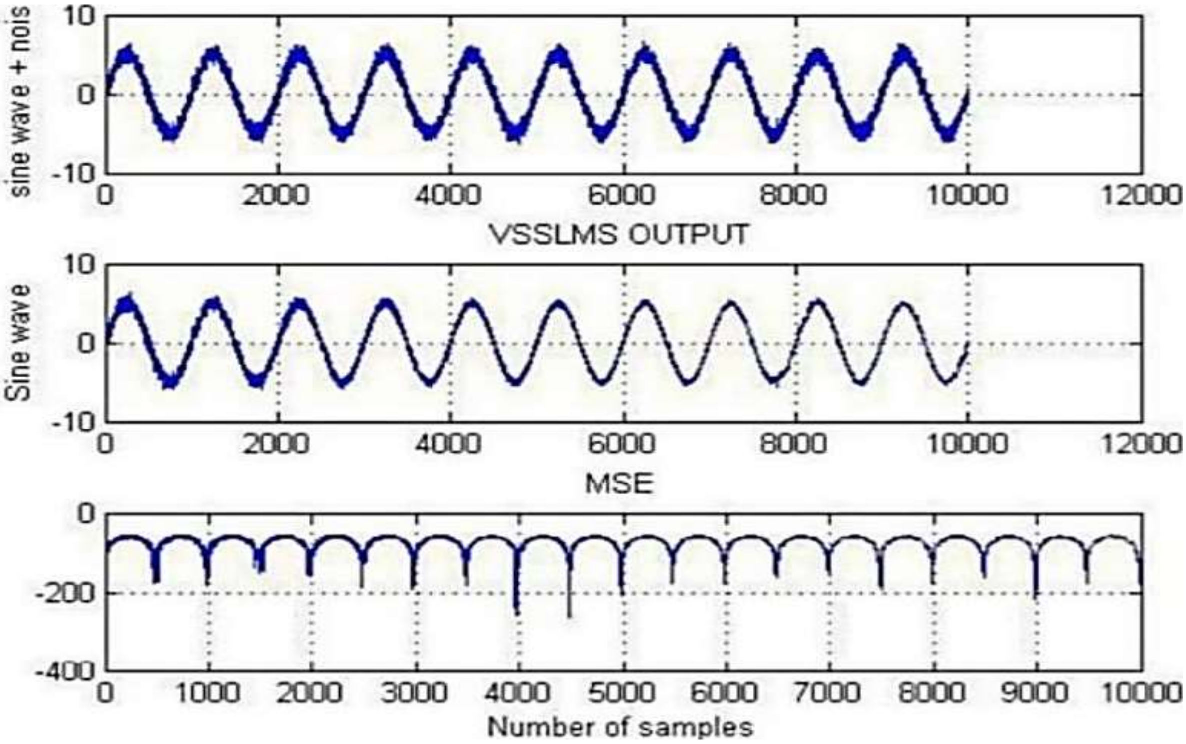

Figure.7: Sinusoidal Wave MSE and the Output of VSSLMS

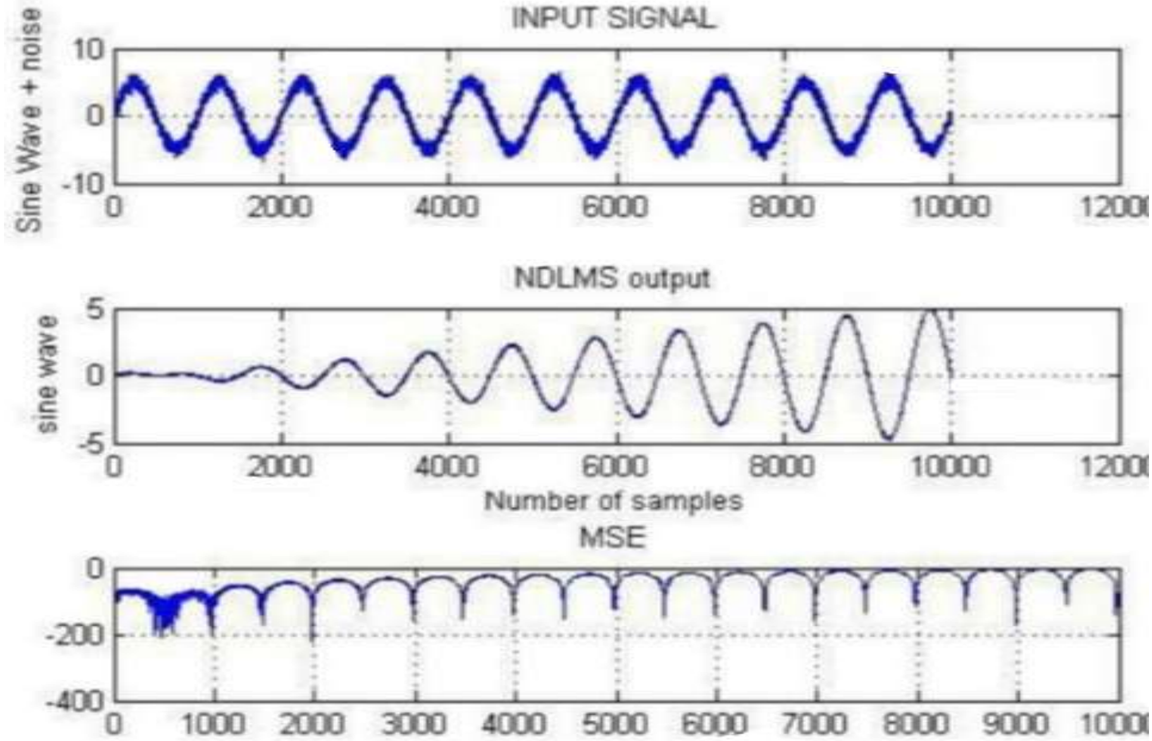

Figure.8: Sinusoidal Wave MSE and the Output of NDLMS

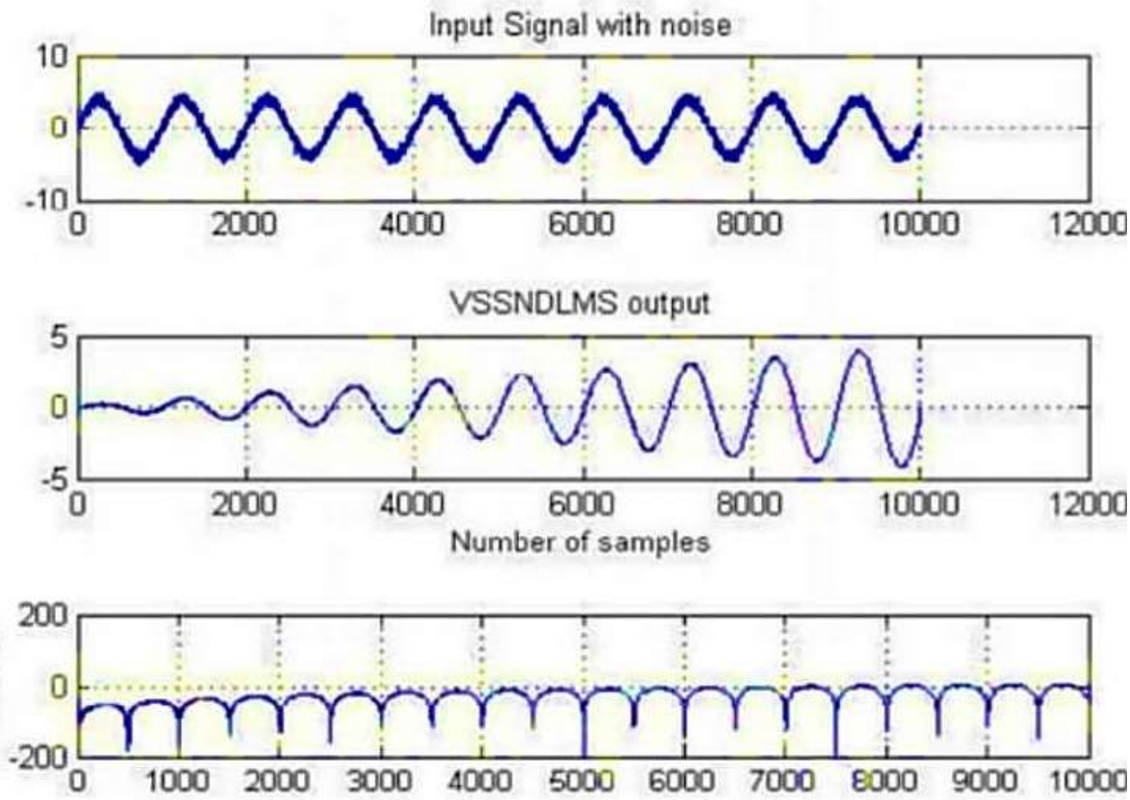

Figure.9: Sinusoidal Wave MSE and the Output of VSSNDLMS

VII. CONCLUSION

The proposed algorithm pointed toward planning a powerful adaptive channel to eliminate the noise and to improve the nature of discourse signals. The noiseless discourse signal is needed during the time spent change of discourse signals into text which is to be utilized by hearing impeded individuals. The Variable Step Size Normalized Differential LMS algorithm proposed which consolidates the highlights of NDLMS and VSSLMS algorithm. The property of VSS Normalized Differential LMS algorithm is quicker assembly and MMSE. Among different LMS Algorithms like Normalized Differential LMS, VSSLMS the proposed algorithm ends up being of more noteworthy productivity in noise cancellation. Execution examination of Various Adaptive Algorithms is talked about with Simulation Results.

REFERENCES

[1] Khan S, Usman M, Naseem I, Togneri R, Bennamoun M. A robust variable step size fractional least mean square (rvss-flms) algorithm. In2017 IEEE 13th International Colloquium on Signal Processing & its Applications (CSPA) 2017 Mar 10 (pp. 1-6). IEEE.

[2] Dixit S, Nagaria D. LMS Adaptive Filters for Noise Cancellation: A Review. International Journal of Electrical & Computer Engineering (2088- 8708). 2017 Oct1; 7(5).

[3] Mohapatra S, Kar A. An improved subband sigmoid function based feedback Active Noise Cancellation. In2015 Annual IEEE India Conference (INDICON) 2015 Dec 17 (pp. 1-6). IEEE.

[4] Li S, Wu S, Wang Y, Guo W, Zhou Y. An improved NLMS algorithm based on speech enhancement. In2015 IEEE Advanced Information Technology, Electronic and Automation Control Conference (IAEAC) 2015 Dec 19 (pp. 896-899). IEEE.

[5] Quadri A, Manesh MR, Kaabouch N. Noise cancellation in cognitive radio systems: a performance comparison of evolutionary algorithms. In2017 IEEE 7th Annual Computing and Communication Workshop and Conference (CCWC) 2017 Jan 9 (pp. 1-7). IEEE.

[6] Chaudhary NI, Raja MA, Khan AU. Design of modified fractional adaptive strategies for Hammerstein nonlinear control autoregressive systems. Nonlinear Dynamics. 2015Dec;82(4):1811-30.

[7] Adnan Quadri , M. R. Manesh , and N. Kaabouch. Denoising signals in cognitive radio systems using an evolutionary algorithm based adaptive filter. Computer Science, Engineering. 2016 IEEE 7th Annual Ubiquitous Computing, Electronics & Mobile Communication Conference (UEMCON) 2016 Oct 1 (pp. 1-6).

[8] Olutayo O. Oyerinde. Regularized VSSNLMS-based iterative channel estimation for MC-IDMA systems. 2016 IEEE Global Conference on Signal and Information Processing, IEEE GlobalSIP 2016 Dec 7 (pp. 1418-1422).

[9] Kumar S, Kohli AK. Adaptive Polynomial Filtering for System Identification Using Modified Sigmoid Variable Step-Size LMS Algorithm (Doctoral dissertation).

[10] Taghia J, Martin R. A frequency-domain adaptive line enhancer with step-size control based on mutual information for harmonic noise reduction. IEEE/ACM Transactions on Audio, Speech, and Language Processing, 2016 Mar 23, 1140-1154.
[11] Zayed Ramadan, & Alexander Poularikas-Performance Analysis of a New Variable Step-Size LMS Algorithm with Error Nonlinearities, IEEE 2004 pp 384-388.
[12] Zayed Ramadan and Alexander Poularikas ― A Variable Step Size Adaptive Noise Canceller Using Signal to Noise Ratio as the Controlling Factor ‖ IEEE 2004, pp456-461.
[13] Ching-Ta Lu, Chung-Lin Lei, Jun-Hong Shen, Ling-Ling Wang and Kun-Fu Tseng Estimation of Noise Magnitude for Speech Denoising Using Minima-controlled Recursive-Averaging Algorithm Adapted by Harmonic Properties‖ Appl.Sci.2017,7,9; doi:10.3390/app7010009, pp1-22.
[14] Gerald―New Normalized LMS algorithm Based on the Kalman Filter‖1-4244-0921-7/07-2007, IEEE, pp 117-120.
[15] Leoardo Rey Vega, Hernan Rey, Jacob BenestyJ and Sara Tressens ― A New Robust Variable Step Size NLMS Algorithm, IEEE transactions on signal processing, Vol.56, Issue 5, May 2008, pages1878-1893.
[16] Ramsey Faragher ―understanding the basis of the Kalman filter via a simple and intuitive derivation, IEEE signal processing magazine, 2012, pp 128-132.