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Performance Analysis of Cascaded Adaptive Algorithms for Noise Cancellation in Speech Signal

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Abstract: In this paper, a novel algorithm for cancelling noise from the speech signal is proposed. Least Mean Square (LMS) adaptive noise cancellers are widely used to recover signal corrupted by additive noise due to its simplicity in implementation. But it has limitation when the desired signal is strong, that the excess mean-square errors increase linearly with the desired signal power. This results in poor performance when the desired signal exhibits large power fluctuations. Later several algorithms were proposed to achieve maximum SNR value with minimum distortion, such as normalized least mean square algorithm (NLMS) and variable step size least mean square algorithm (VSSLMS). But in NLMS algorithm, selection of step size and filter length of adaptive filter for different type of noise with different noise level (dB) that gives maximum SNR is difficult. This needs various trials of step size and filter length to get optimum solution. In the proposed algorithm we use the benefits of both variable step size (VSS) LMS algorithm and Normalized LMS (NLMS) algorithm to deal with this situation. Finally, the proposed (cascaded VSSNLMS) algorithm yields maximum signal to noise ratio (SNR) with minimum mean square error (MSE) in simulations which were carried out using MATLAB software with different noise signals.
Key words: Adaptive Noise Canceller, Step Size, Mean Square Error, NLMS, SNR, C-VSSNLMS.

I. INTRODUCTION

The goal of any filter is to extract useful information from noisy data. A normal fixed filter is designed in advance with knowledge of the statistics of both signal and the unwanted noise but if the statistics of the noise are not known priori, or change over time, the coefficients of the filter cannot be specified in advance. In these situations, adaptive algorithms are needed in order to continuously update the filter coefficient. Adaptive filtering finds application in noise cancellation in speech called as Adaptive Noise cancellation (ANC) which involves in time-varying signals and systems. ANC is an effective method for recovering a signal corrupted by additive noise and it is an important core area of the digital signal processing. Fig.1 shows the basic problem and the adaptive noise cancelling solution to it. A signal s(n) is transmitted over a channel to a sensor that also receives a noise n0 uncorrelated with the signal. The primary input to the canceller is combination of both signal and noise s + n0. A second sensor receives a noise n1 uncorrelated with the signal but correlated with the noise n0. This sensor provides the reference input to the canceller. This noise n1 is filtered to produce an output y(n) that is as close a replica of n0. This output of the adaptive filter is subtracted from the primary input to produce the adaptive filter error e(n) = d(n) – y(n).

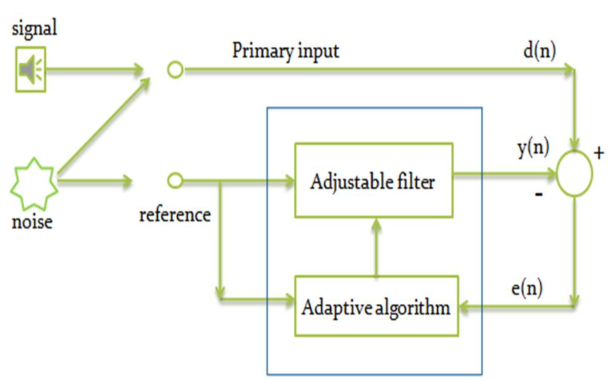


Fig:1 Adaptive Noise Canceller

Probably, one of the well-known algorithms in the field of adaptive filtering is the Least Mean Square (LMS) algorithm. Simplicity and easy implementation are the main reasons for the popularity of LMS algorithm. Some successful applications of the LMS filters are:

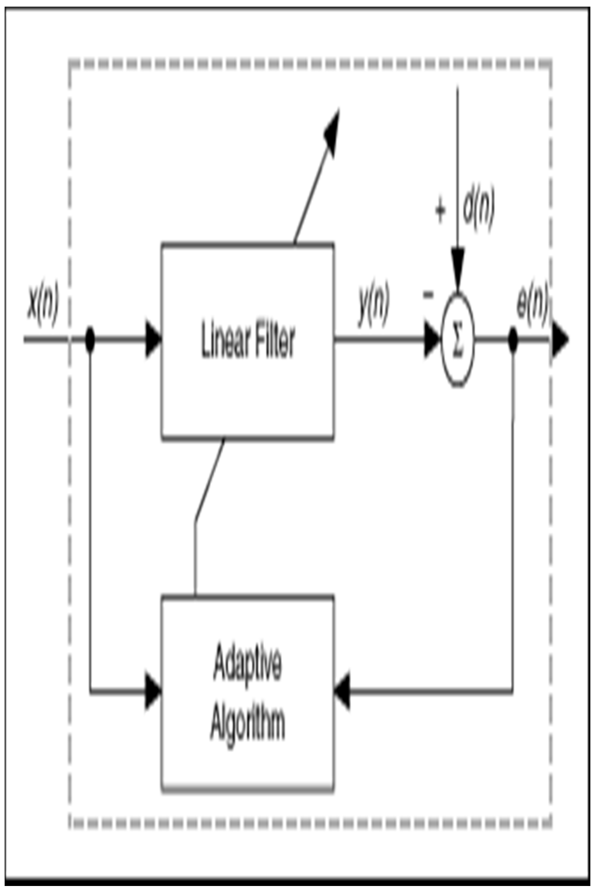
1) System identification.
2) Noise cancellation in speech signals.
3) Signal prediction.
4) Interference cancellation.
5) Channel equalization.
6) Echo cancellation and it has been widely used in noise cancellation.

II. ADAPTIVE ALGORITHMS

Several modified LMS algorithms have been proposed in the past years in order to simultaneously improve the tracking ability and speed of convergence of corrupted signal. They provide an extensive performance evaluation Compared to standard LMS algorithms, including the NLMS, VSSLMS and other recently proposed LMS algorithms.

A. Least Mean Square (LMS) Algorithm

The Least Mean Square (LMS) algorithm was first developed by Widrow and Hoff in 1959 through their studies of pattern recognition (Haykin 1991, p. 67). The least-mean-square (LMS) algorithm is the most widely used among various adaptive algorithms because of its simplicity and robustness. The block diagram illustrates the general LMS adaptive filtering algorithm. Here the adaptation process of the filter parameters is based on minimizing the mean squared error between the filter output and a desired signal.

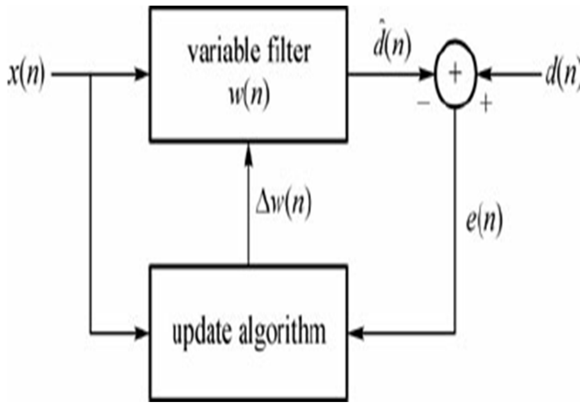


B. Normalised Least Mean Square (NLMS) Algorithm

One of the primary disadvantages of the LMS algorithm is having a fixed step size parameter for every iteration. This requires an understanding of the statistics of the inputsignal prior to commencing the adaptive filtering operation. In practice this is rarely achievable. Even if we assume the only signal to be input to the adaptive echo cancellation system is speech, there are still many factors such as signal input power and amplitude which will affect its performance. The normalised least mean square algorithm (NLMS) is an extension of the LMS algorithm which bypasses this issue by selecting a different step size value, μ(n), for each iteration of the algorithm. This step size is proportional to the inverse of the total expected energy of the instantaneous values of the coefficients of the input vector **x**(n).

C. Variable Stepsize Least Mean Square (VSSLMS) Algorithm

In the variable step-size algorithm for LMS adaptive filtering, Commonly the basic ideas for the variable step-size algorithm of LMS are as follows: In the initial stage of convergence, step size should be bigger, this makes the algorithm has faster convergence speed. Then with the deepening of convergent gradually reduce the step size to reduce the static error. In the process of research for the variable step-size algorithm of LMS, also proposed to make μ(n)is proportional to e(n),and proposed to make μ(n)is proportional to the evaluation of the cross-correlation function for e (n) and x(n),and so on. Practice shows that these algorithms can give attention to faster convergence rate and smaller maladjustment to a certain extent. Can effectively remove the irrelevant noise interference, and have fewer parameters and smaller amount of calculation of the algorithm itself.



D. Variable Stepsize Least Mean Square (VSSLMS) Algorithm

The VSLMS algorithm still has the same drawback as the standard LMS algorithm in that to guarantee stability of the algorithm, a statistical knowledge of the input signal is required prior to the algorithms commencement. Also, recall the major benefit of the NLMS algorithm is that it is designed to avoid this requirement by calculating an appropriate step size based upon the instantaneous energy of the input signal vector. It is a natural progression to incorporate this step size calculation into the variable step size algorithm, in order incII. rease stability for the filter without prior knowledge of the input signal statistics. This is what I have tried to achieve in developing the Variable step size normalized least mean square (VSNLMS) algorithm.

III. ANALYSIS OF ADAPTIVE ALGORITHMS

A. Least Mean Square (LMS) Algorithm

The LMS algorithm is a type of adaptive filter known as stochastic gradient-based algorithms as it utilises the gradient vector of the filter tap weights to converge on the optimal wiener solution. With each iteration of the LMS algorithm, the filter tap weights of the adaptive filter are updated according to the following formula (Farhang-Boroujeny 1999, p. 141).

$$\boldsymbol{w(n+1) = w(n) + 2\mu e(n)x(n)} \quad (1)$$

Here $\mathbf{x}(n)$ is the input vector of time delayed input values, x(n) = [x(n) x(n-1) x(n-2) .. x(n-N+1)]T. The vector w(n) = [w0(n) w1(n) w2(n) .. wN-1(n)] T represents the coefficients of the adaptive FIR filter tap weight vector at time n. The parameter μ is known as the step size parameter and is a small positive constant.This step size parameter controls the influence of the updating factor. Selection of a suitable value for μ is imperative to the performance of the LMS algorithm, if the value is too small the time the adaptive filter takes to converge on the optimal solution will be too long; if μ is too large the adaptive filter becomes unstable and its output diverges.

For each iteration the LMS algorithm requires 2N additions and 2N+1 multiplications (N for calculating the output, y(n), one for 2μe(n) and an additional N for the scalar by vector multiplication).

B. Normalised Least Mean Square (NLMS) Algorithm

The recursion formula for the NLMS algorithm is

$$\boldsymbol{w(n+1) = w(n) + \frac{1}{(x(n)x(n)^T)}\mathrm{x}(n)\mathrm{e}(n)} \quad (2)$$

The output of the adaptive filter is calculated as

$$y(n) = \sum_{j=0}^{N-1} w(n)x(n-1) = w^T(n)x(n) \qquad (3)$$

An error signal is calculated as the difference between the desired signal and the filter output.

$$e(n) = d(n) - y(n) \qquad (4)$$

The step size value for the input vector is calculated as

$$\mu(n) = \frac{1}{(x(n)x(n)^T)} \qquad (5)$$

The filter tap weights are updated in preparation for the next iteration.

$$\mathbf{w}(\boldsymbol{n}+\mathbf{1}) = \mathbf{w}(\boldsymbol{n}) + \boldsymbol{\mu}(\boldsymbol{n})\boldsymbol{e}(\boldsymbol{n})\mathbf{x}(\boldsymbol{n}) \qquad \textbf{(6)}$$

Each iteration of the NLMS algorithm requires 3N+1 multiplications, this is only N more than the standard LMS algorithm this is an acceptable increase considering the gains in stability and echo attenuation achieved.

C. Variable Stepsize Least Mean Square (VSSLMS) Algorithm

The NLMS algorithm exhibits a good balance between computational cost and performance. However, a very serious problem associated with both the LMS and NLMS algorithms is the choice of the step-size (μ) parameters. A small step size (small compared to the reciprocal of the input signal strength) will ensure small mis-adjustments in the steady state, but the algorithms will converge slowly and may not track the non-stationary behavior of the operating environment very well. On the other hand a large step size will in general provide faster convergence and better tracking capabilities at the cost of higher misadjustment. Any selection of the step-size must therefore be a trade-off between the steady-state misadjustment and the speed of adaptation. VSSNLMS algorithm overcomes the problem of convergence speed and estimation accuracy in real time environment. The signal to noise (SNR) ratio is defined as the ratio of the average power of the original signal to that of the noise signal. The main aim of proposing this algorithm is that with the help of SNR, the step size adjustment can be controlled. 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.

In the Variable Step Size Least Mean Square (VSLMS) algorithm the step size for each iteration is expressed as a vector, μ(n). Each element of the vector μ(n) is a different step size value corresponding to an element of the filter tap weight vector, w(n).

The output of the adaptive filter is calculated as

$$y(n) = \sum_{j=0}^{N-1} w(n)x(n-1) = w^T(n)x(n) \qquad (7)$$

An error signal is calculated as the difference between the desired signal and the filter output.

$$e(n) = d(n) - y(n) \qquad (8)$$

The gradient, step size and filter tap weight vectors are updated as

$$w_i(n+1) = w_i(n) + \mu_i(n)e(n)\mathrm{x}(n) \qquad (9)$$

For i= 0,1,2,3……….N-1.

Where if $\mu_i(n) > \mu_{max}(n), \quad \mu_i(n) = \mu_{max}(n)$

$\mu_i(n) < \mu_{min}(n), \quad \mu_i(n) = \mu_{min}(n)$

D. Modified Variable Stepsize Least Mean Square (VSSLMS) Algorithm

In the VSNLMS algorithm the upper bound available to each element of the step size vector, μ(n), is calculated for each iteration. As with the NLMS algorithm the step size value is inversely proportional to the instantaneous input signal energy.

The output of the adaptive filter is calculated as

$$y(n) = \sum_{j=0}^{N-1} w(n)x(n-1) = w^T(n)x(n) \quad (10)$$

An error signal is calculated as the difference between the desired signal and the filter output.

$$e(n) = d(n) - y(n) \qquad (11)$$

The gradient, step size and filter tap weight vectors are updated using the following equations in preparation for the next iteration.

For i= 0,1,2,3……….N-1.

$$g_i(n) = e(n)x(n-i)$$

$$g(n) = e(n)x(n)$$

$$\mu_i(n) = \mu_i(n-1) + \rho g_i(n) g_i(n-1) \qquad (12)$$

$$\mu_{max}(n) = \frac{1}{(2x(n)x(n)^T)} \qquad (13)$$

If $\mu_i(n) > \mu_{max}(n)$, $\mu_i(n) = \mu_{max}(n)$

If $\mu_i(n) < \mu_{min}(n)$, $\mu_i(n) = \mu_{min}(n)$

$$w_i(n+1) = w_i(n) + 2\mu_i(n) g_i(n) \qquad (14)$$

ρ is an optional constant the same as is the VSLMS algorithm. With ρ =1, each iteration of the VSNLMS algorithm requires 5N+1 multiplication operations.

Different types of adaptive filtering algorithms have been analyzed in the above section. In LMS algorithm, signal to noise ratio (SNR) is good in lower orders until the signal power is moderate. If the desired signal is having high power the excess mean square error of the adaptive filter is increased linearly with signal power. This results in the decrease in SNR value of the desired signal. To overcome this problem NLMS algorithm was proposed But in NLMS algorithm, selection of step size and filter length of adaptive filter for different type of noise with different noise level (dB) that gives maximum SNR is difficult. This needs various trials of step size and filter length to get optimum solution. For that reason VSSLMS algorithm was proposed.

E. Proposed Algorithm

Cascaded Structural Implementation of Variable Step Size Normalized Least Mean Square Algorithm (C-VSSNLMS)

This algorithm is a modified version of the variable step size normalized least mean square algorithm (VSSNLMS). The below figure clearly explains the operation of proposed algorithm.

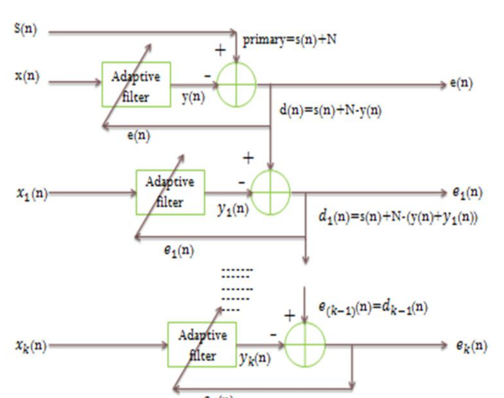


Fig. cascaded structure of VSSNLMS

The variable step size normalized lms algorithm the combination of the both NLMS and VSSLMS algorithm. The C-VSSNLMS algorithm is modification of the VSSNLMS algorithm. In this algorithm the input signal given to the adaptive filter is consists of original signal s(n) and some noise 'N' with in it. That can be given as primary signal to the adaptive filter block. The secondary noise or the reference noise given the adaptive filter is consists of noise signal x(n) and some reference noise.The output of the adaptive filter is y(n) ,which is the impulse response of the input x(n). y(n) is subtracted from the primary signal, and gives the desired signal or the error signal as below.

$$e(n)= \text{primary} - y(n)$$

$$e(n)= s(n) + N - y(n) \quad \text{e.q} \qquad (15)$$

The e(n) consists of both original signal and the noise signal after any reduction (N-y(n)). That means the desired signal consists original signal and noise. The noise in the desired output signal is may be changed time to time due to the filter output y(n), but the original signal can't be lossed in this process. The desired output is again given as the primary input to the next stage of the adaptive filter. In the second stage another reference noise is given to the adaptive filter such as x1(n). the output of the filter is y1(n) is subtracted from the desired signal 1 . the error signal generated in this state is e1(n) . and the process is repeated until the desired output is obtained.The error signal at each stage of the filtering process is obtained, at which the system give the maximum snr is the state where we can obtain maximum likely approximate original signal.

This is about the execution of the cascaded structuralimplementation of the variable step size normalized least mean square algorithm(C-VSSLMS). In LMS algorithm the excess mean square error of the filter is increases along with the increase in the signal power of the desired signal. To overcome this problem, NLMS algorithm has been proposed. The NLMS algorithm provides the better mean square error compared to the LMS. But the selection of step for different signal with differ powers is difficult. To overcome this problem a VSSLMS algorithm has been proposed in this algorithm the step size parameter mu is changed in accordance with the signal to noise ratio (SNR). If the step size is chosen as small it gives minimum mean square errors but the convergence rate is very low. In other side if the step size is chosen as large as possible the convergence rate is improved. But the mean square error is increased. So there must be a tradeoff between the mean square error and convergence rate.

To overcome this we use the benefits of both NLMS and VSSLMS algorithms and designed the VSSNLMS algorithm. In this algorithm the snr and mean square values are better than the previously proposed algorithms, but the snr value of the adaptive filtering algorithm is fluctuated drastically whenever there is any small change in the random noise added to the system. The cascaded structural implementation of the VSSNLMS algorithm improves the SNR value even such cases also. This newly proposed algorithm based on cascaded structure to remove noise from speech signal has been compared with other algorithms, in order to examine the performance enhancement. In the proposed algorithm the weights of adaptive filter is updated based on variable step size and signal to noise ratio of speech and reference noise signal. The simulations (in next setion) show that the behavior of the new algorithm provides an improved performance in terms of SNR and MSE Hence, the proposed C-VSSNLMS algorithm is a promising method for noise cancellation.

IV. SIMULATION RESULTS

A. Specifications

1) Voice signal: song (15 sec length).
No. of samples in voice signal (N): 661504.
Voice signal size: (661504 X 2 double).

2) Noise signal: applause noise, kids play noise, river noise and babble noise .
No. of samples in noise signal (M): 456457.
Noise signal size: (661504 X 2 double).

3) Step size parameter (mu): 0.1.

4) Initial weight vector (w): zeros of vector size (128 X 1).

5) Primary signal = voice signal+ noise signal.
Primary signal size: (661504 X 2 double).

6) Reference signal= noise signal + 0.25*random noise.
Reference signal size: (661504 X 2 double).

7) No. of iterations =N – order of the system.
=661504-128 =331372

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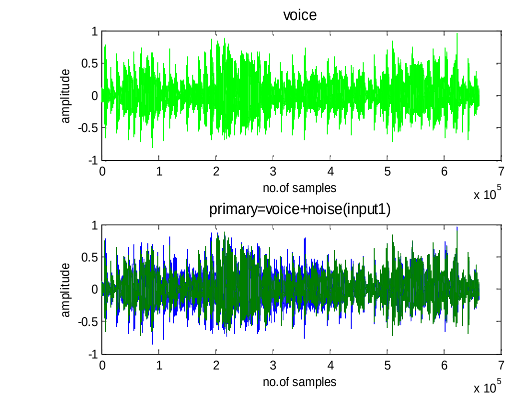
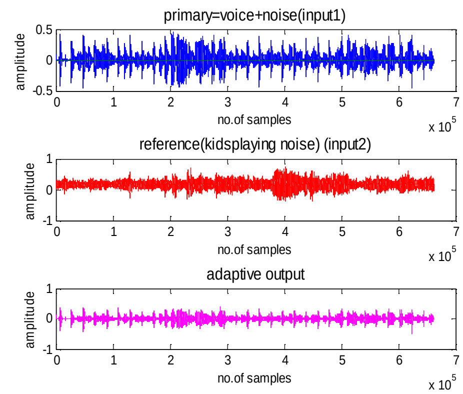
B. Wave forms

1) Simulation Results For Cas-Lms Algorithm For N=12





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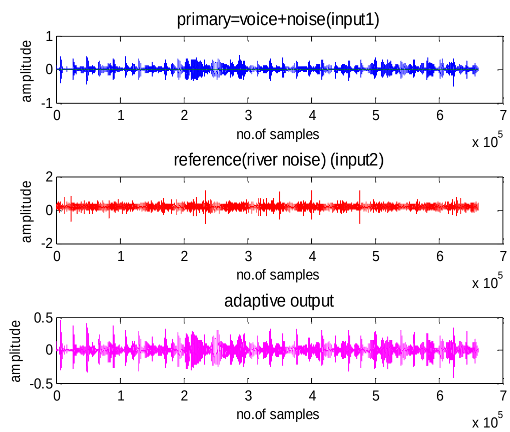
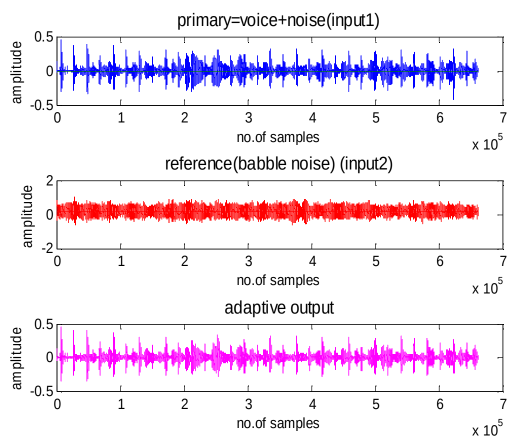
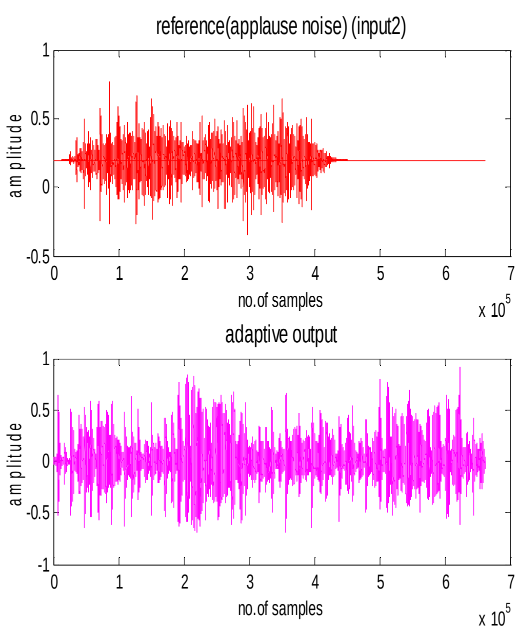
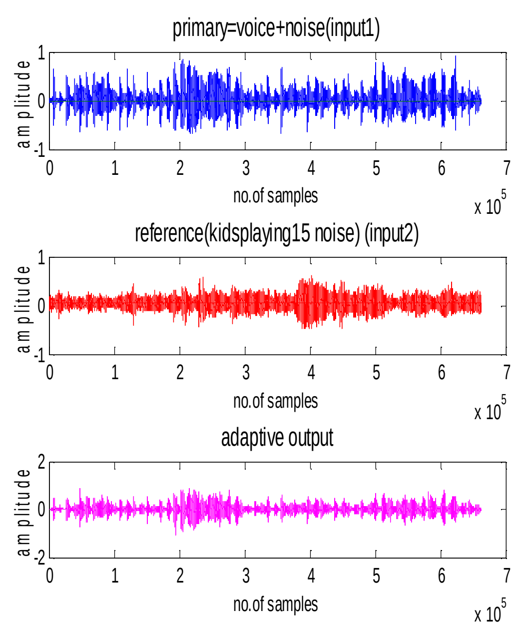




2) Simulation Results For Cas -Nlms Algorithm For N=128





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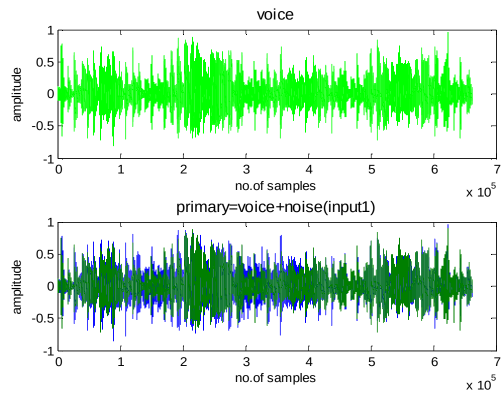
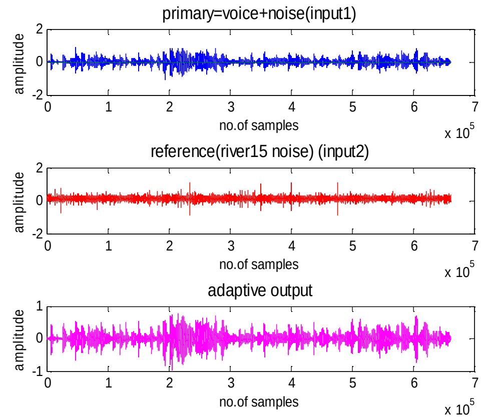
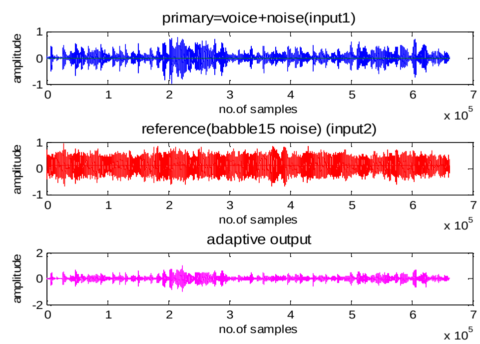
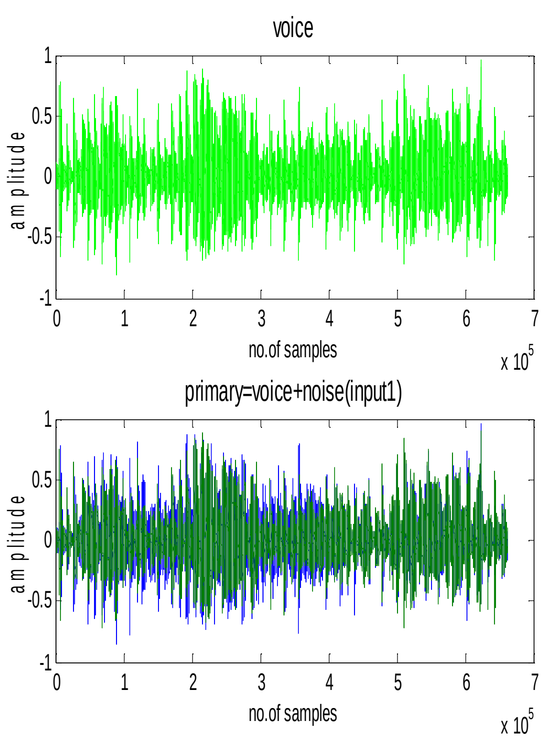
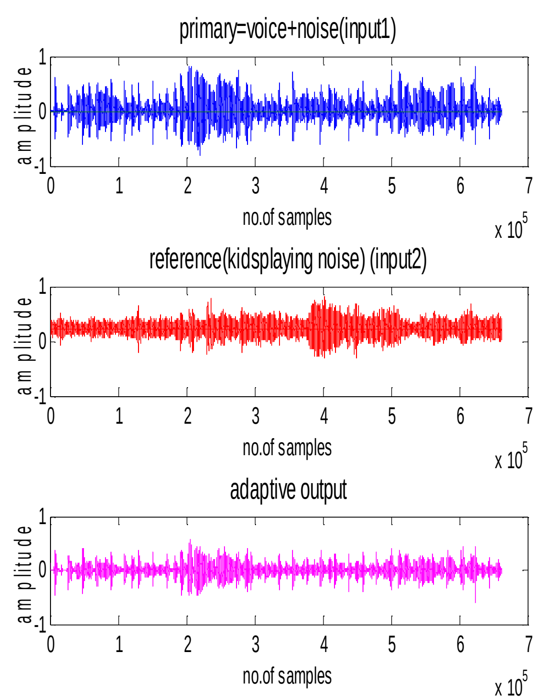


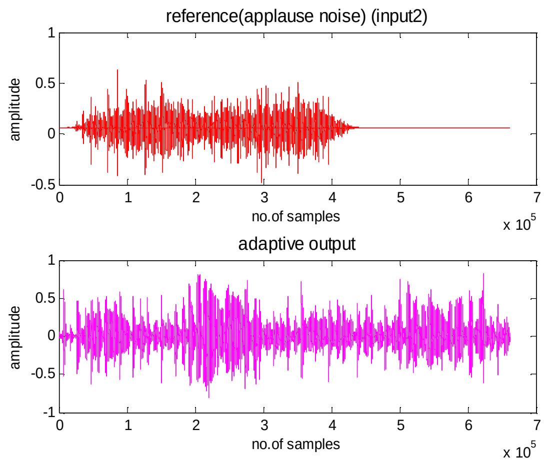


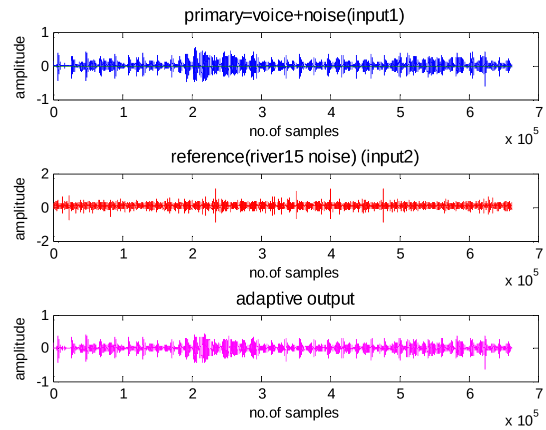


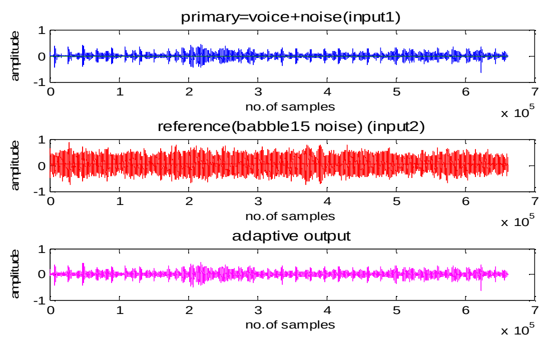
3) Simulation Results For Cas -Vsslms Algorithm For N=128



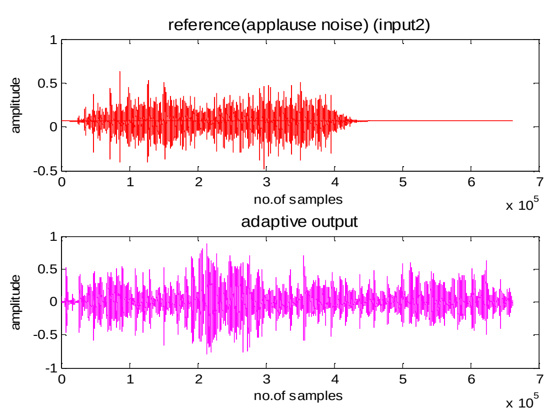


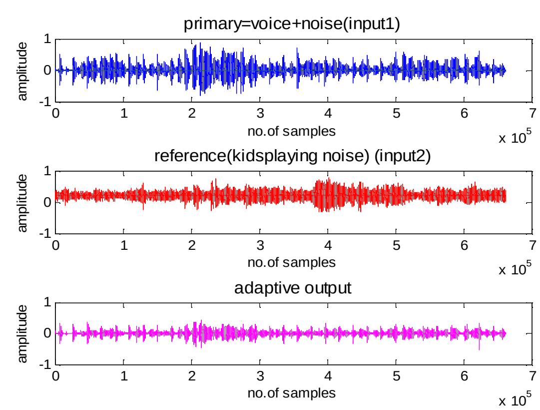


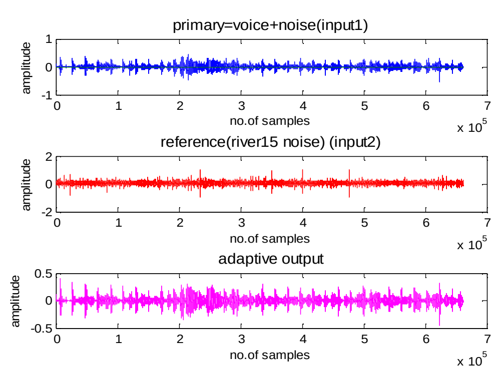


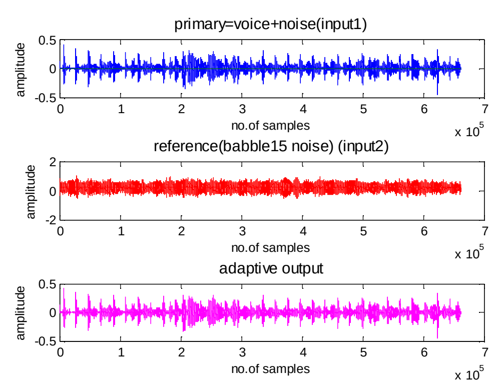


4) simulation results for cas -vssnlms algorithm for n=128









5) Performance comparasion table:

NOISE	Applause	Kidsplaying	River	Bable	AVG SNR
LMS	69.697	24.0454	25.3468	37.6504	39.1849
NLMS	72.7523	35.296	34.0695	39.6533	45.4427
VSSLMS	69.0778	38.5084	49.2665	22.9105	44.9408
VSSNLMS	72.7751	64.3793	53.7249	21.4947	53.0935
CASLMS	70.3161	70.6454	70.6713	70.9804	70.6533
CASNLMS	72.7764	54.7838	64.8205	71.5745	65.9888
CASVSSLMS	64.4768	71.7737	69.4814	31.1798	59.22793
CASVSSNLMS	68.1045	72.6104	70.4496	71.1533	70.57945

This newly proposed algorithm based on Cascaded structure to remove noise from speech signal has been compared with other algorithms, in order to examine the performance enhancement. In this proposed algorithm the weights vectors of the adaptive filters are updating based on variable step size and signal to noise ratio of speech and reference noise signal.

V. CONCLUSION

This newly proposed algorithm based on cascaded structure to remove noise from speech signal has been compared with other algorithms, in order to examine the performance enhancement. In this proposed algorithm the weights vectors of the adaptive filters are updating based on variable step size and signal to noise ratio of speech and reference noise signal. From rigorous matlab experimental analysis we conclude that proposed algorithm outperforms LMS NLMS and VSSLMS algorithms in terms of SNR, MSE and distortion.

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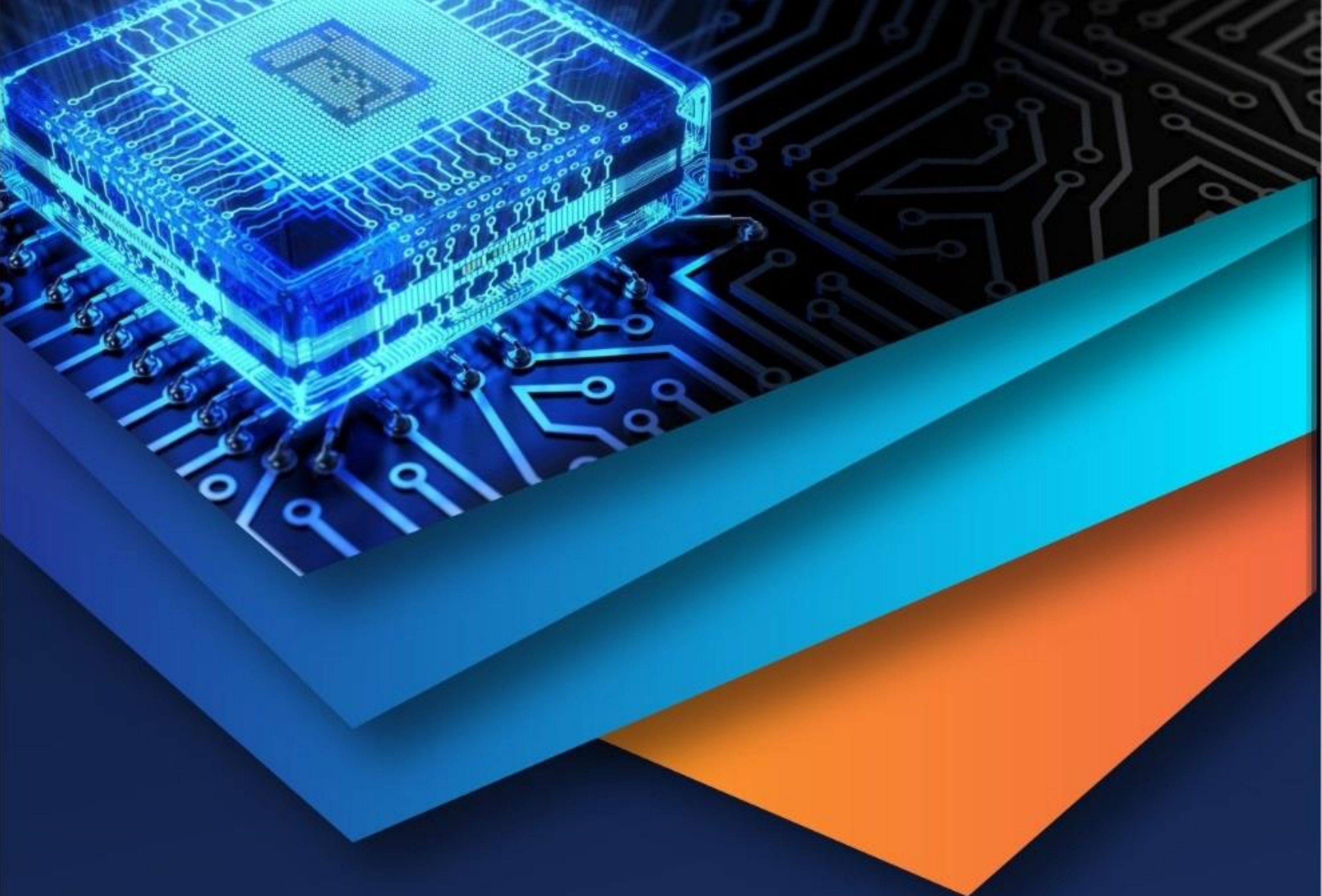

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