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Analysis of Adaptive Filter Algorithms Approach For Speech Enhancement Using Simulink

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ABSTRACT

In this paper, we postulate the analysis of different adaptive filter algorithms, that is Least mean square (LMS), Normalized least mean square (NLMS) and Recursive least square (RLS) for speech enhancement using Simulink tool. This speech enhancement approach is done only through noise suppression, because intelligibility and pleasantness cannot be measured by mathematical calculations. By improving the signal-to-noise ratio during the speech signal processing using different Adaptive Filter algorithms technique, approach for speech enhancement approach can be done. To analyze the adaptive filter algorithms, each algorithm is computationally implemented and signal to noise ratio of algorithms are examined. Criteria of these adaptive filter algorithms are analyzed by varying the step size for minimum mean square error optimization using Simulink tool.

Keywords: Speech Enhancement, Least Mean Square (LMS) Algorithm, Normalized Least Mean Square (NLMS) Algorithm, Recursive Least Square (RLS) Algorithm, Signal To Noise Ratio (SNR), Simulink.

1. INTRODUCTION

Speech enhancement is one of the most important factors during speech signal processing to improve the speech quality. The only approach for speech enhancement is background noise cancellation. Where the suppressing the noise is based on the estimation of the background noise. If the noise is evolving slowly i.e. if noise is more stationary, it will be easy to estimate the noise duration pauses in the speech. While checking the background noise to the current window, pauses in the speech can be estimated.

Many applications are designed in the digital signal processing to approach speech enhancement, but second order statistics applications like noise cancellation, echo cancellation and channel equalization cannot be specified. The transfer functions of the adaptive filter algorithms applications are designed in such a way that the filter coefficients are self-adjustable in according to the optimized algorithm, by adapting the incorporate algorithms of filter coefficients with the input signal statics. Different adaptive filter approaches used in speech enhancement are as following:

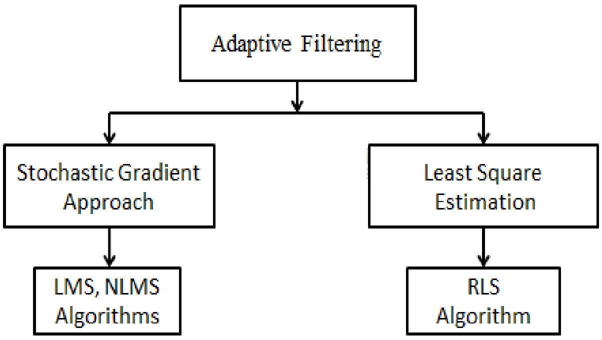

Fig1: Hierarchy of Adaptive Filter

Simulink is a software package for modeling, simulating, and analyzing dynamic systems. It supports linear and nonlinear systems, modeled in continuous time, sampled time, or a hybrid of the two. Systems can also be multirate, i.e., have different parts that are sampled or updated at different rates, which encourages trying new things out.

2. METHODOLOGY

The entire speech enhancement processes objective is to minimize the noise in the background of speech signal. It is simple to reduce the noise when it is in stationary in compression to speech. Never the less, it is very intricate scenario to remove the noise in the acoustic environment.

In Simulink tool, the modeling of acoustic environment is a milieu with the presence of the input speech signal and induced noise is designed due to various external factors. The random variable noise converted from continuous signal to discrete signal by using filters v.i.z., low pass and band pass filters where the frequency is confined to particular value and merged with speech signal to obtain a noisy speech signal, which is fed as desired signal to adaptive filter algorithms along with the noise as the input signal. The filter algorithms change its transfer functions in order to match with the output of an unknown system.

Eventually, the output of linear filter is terminated, and error is extracted while comparing with the original speech signal. The estimated error gets cross correlated to intensify the signal to noise ratio to obtain the best enhanced speech signal.

3. ADAPTIVE FILTERING CONFIGURATION

Various adaptive filter configurations are adaptive inverse system, linear prediction, noise cancellation, and the system identification, where there maybe difference in system configuration but the algorithm implementation is analogy. Let consider x(n) as input, w(n) as transfer function, d(n) be a desired signal and e(n) be error signal is variance of desired and actual outputs. A diagram for specific adaptive filter realizations, they are LMS, NLMS and RLS algorithms are shown in the following figure, because the idea behind the block diagram is taking out estimated signal from desired signal at variable filter.

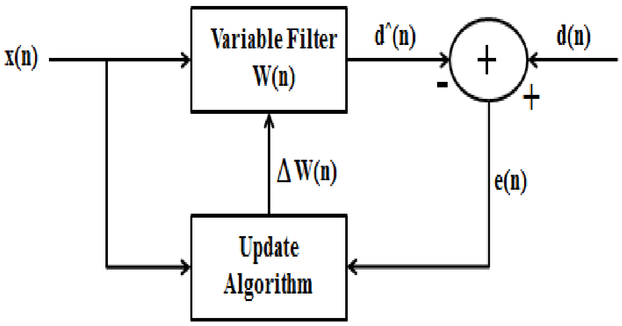

Fig 2: Basic Adaptive Filter

Input signal x(n) is the mixture of the noise v(n) and desired signal d(n) and represented as

$x(n) = d(n) + v(n)$

Variable filter is structure of a finite impulse response (FIR). Error signal is estimated by deduction of estimated signal and the desired signal.

$e(n) = d(n) – d\hat{}(n)$

Desired signal can be estimated by convolving the input signal with impulse response. Vector notation representation of desired signal is given as following

$d\hat{}(n) = W_n * x(n)$

Where

$x(n) = [x(n), x(n01), \text{-------}, x(n – p)]^T$

x(n) can be a input vector value. Time filter coefficients can be updated at every instant by the variable filter and vector notation of input signal is expressed as

$W_{n+1} = W_n + \Delta Wn$

Where ΔWn is correction factor of filter coefficients, which is mainly depends input and the signal from adaptive algorithms (coefficient algorithms are LMS, NLMS & RLS).

4. LMS ALGORITHM

Least Mean Square (LMS) algorithms are class of adaptive filters. In LMS algorithm, desired filter is mimicked by fading filter coefficients that related to producing the least mean square of error signal, which is difference between the desired and the actual signal. It is a stochastic gradient decent method, where filter is adapted based on error at the current time.

Idea behind the LMS filter is optimum filter weights approach by updating the filter weights in a manner to coverage. Basically, the algorithm starts by assuming a small filter weights (zero in most case) at each step. Mean square error gradients are found out by the updated filter weights. If the gradient of mean square error implies positively, error keep increasing positively, we need to reduce the filter weights. If the gradient of mean square error implies negatively, the filter weights should be increased. Filter weights are updated as per the following equation:

$W_{n+1} = W_n - \mu\ e(n)\ x(n)$

Where:

W(n) is current filter weight value,

W_{n+1} is next filter weight value,

e(n) is estimated error and

μ is the convergence factor which determines the convergence time of the filter (step size).

5. NLMS ALGORITHM

Scaling of the input signal is sensitive in the LMS algorithm is the main draw back. In real scenario, input signal power will not remain constant and the step-size (μ) between two adjacent coefficients of the filter will also changes and leads to the gradient noise application problem. This makes hard to choose the learning rate for the stability of the algorithm.

Normalized least mean square algorithm overcomes this problem by normalizing the input power. At instant, the input power is squared to Euclidean norm with respect to the current filter weight value W(n) and the step size parameter (μ) is adjusted with respect to the input signal. NLMS algorithm filter weights are updated as per the following equation:

$W_{n+1} = W_n + \mu(n)\ e(n)\ x(n)$

$$\mu(n) = \frac{\alpha}{c + |x(n)|^2}$$

Where:

α is the NLMS adaption constant aimed to optimize the convergence rate of the algorithm.

Preferably, $0 < \alpha < 2$

C is the constant and is always less than 1.

6. RLS ALGORITHM

Recursive least square aiming to minimize the sum of the squares of the difference between the desired signal and the filter output, RLS algorithm use recursive form to solve least-squares at the latest sampling value is acquired. The RLS algorithm performs the best in time varying environments but at the cost of an increased computational complexity and some stability problems. RLS adaptive filter are known for their excellent performance and the great stability of the algorithm. In this algorithm the filter tap weight vector is updated using following equation:

$W(n) = W^T(n-1) + k(n)\ e_{n-1}(n)$

Intermediate gain vectors used to compute tap weights are calculated as following equation:

$k(n) = u(n) / (\lambda + x^T(n)\, u(n))$

$u(n) = W_{\lambda}^{-1}(n-1)\, x(n)$

Where:

λ is a small positive constant < 1. The filter output is calculated using the filter tap weights of above iteration and the current input vector as following equation:

$y_{n-1}(n) = W^T(n-1)\, x(n)$

$e_{n-1} = d(n) - y_{n-1}(n)$

Higher memory requirements are required for RLS algorithm for the estimation of previous samples of output signal, error signal and the filter weight.

7. RESULT

After modeling the adaptive filter algorithms in the Simulink toll, the noisy speech signal is given as the desired signal to the adaptive filter algorithms (i.e. LMS, NLMS & RLS) along the noise as an input signal with sample rate of 1/8000. The output of the different adaptive filter algorithms is achieved as error signal which is compared with the original speech signal for signal to noise ratio (SNR) estimation.

The output results achieved form the least mean square (LMS) algorithm filter is as follows:

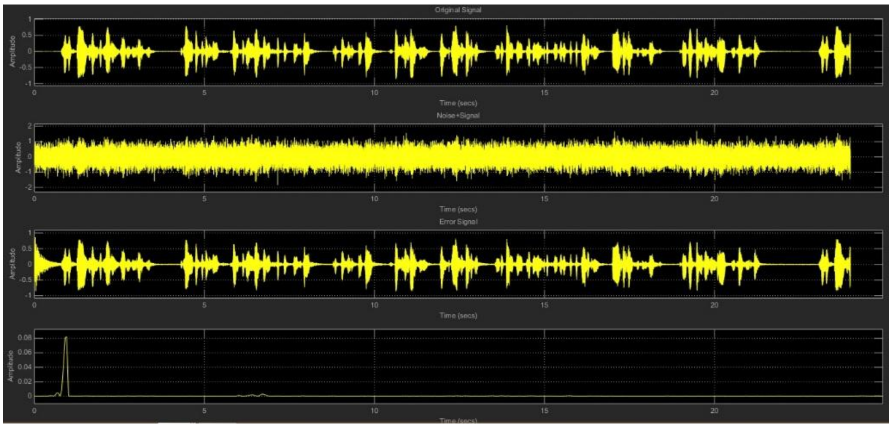

Fig 3: LMS Algorithm Output

1st scope represents the original speech signal which is a .wav audio file format with frequency range of 16 KHz. 2nd scope represents the noisy speech signal which is the combination of original speech audio file and random noise. 3rd scope represents the error signal which is the output of LMS filter and the final 4th scope represents the estimated of all the algorithms. And the estimated signal to noise ratio (SNR) of LMS algorithm filter by cross product relation of the original speech signal and the error signal is as follow:

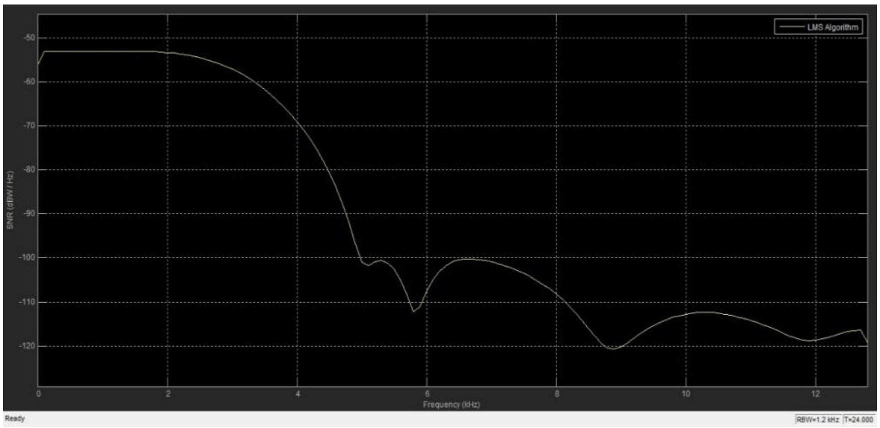

Fig: 4 SNR of LMS Algorithm

The output results achieved from the normalized least mean square (NLMS) algorithm filter is as follow:

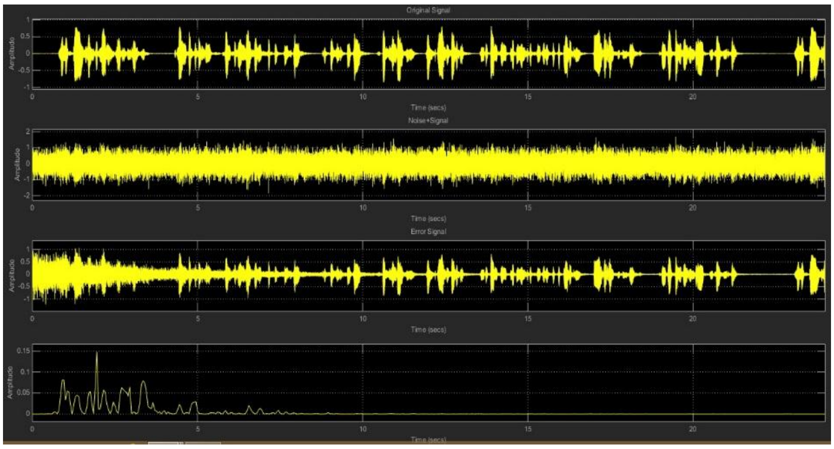

Fig 5: NLMS Algorithm Output

The estimated signal to noise ratio (SNR) of LMS algorithm filter by cross product relation of the original speech signal and the error signal is as follow:

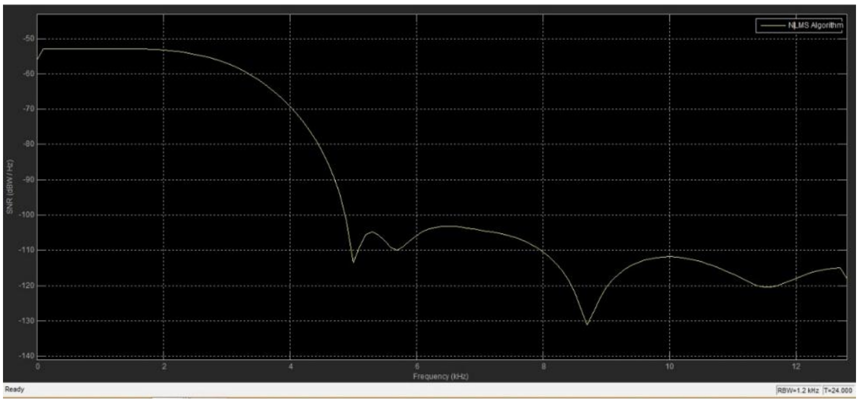

Fig 6: SNR of NLMS Algorithm

The output result achieved from the recursive least square (RLS) algorithm filter is as follow:

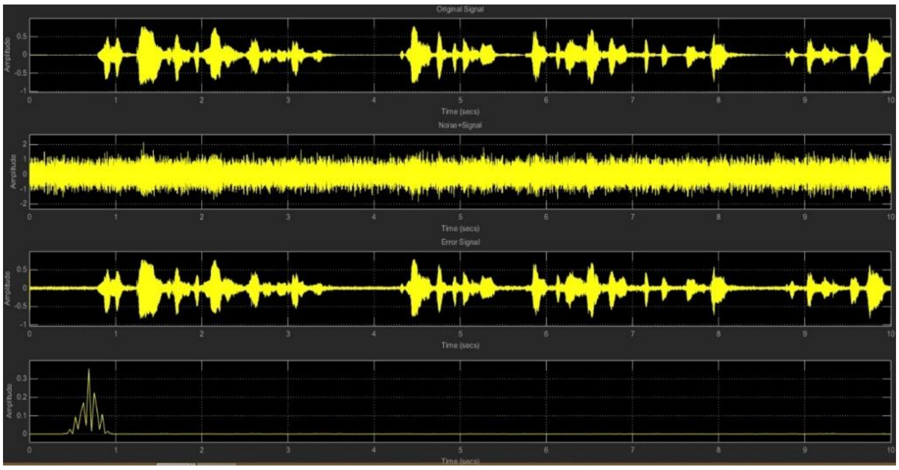

Fig 7: RLS Algorithm Output

The estimated signal to noise ratio (SNR) of RLS algorithm filter by cross product relation of the original speech signal and the error signal is as follow:

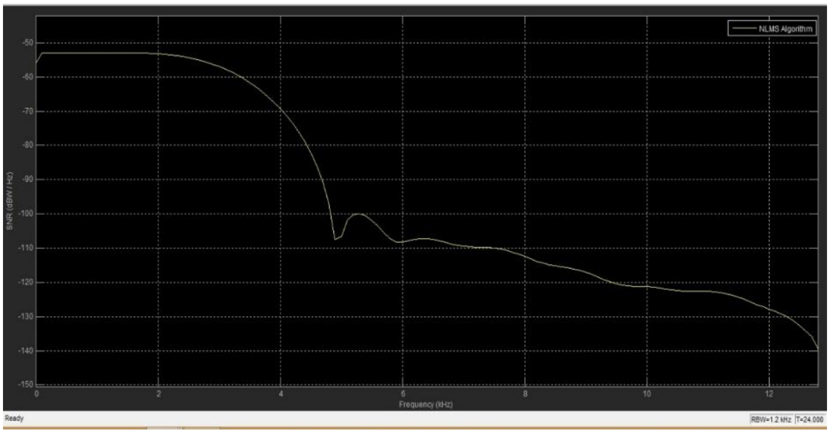

Fig 8: SNR of RLS Algorithm

7. CONCLUSION

From the analysis of different adaptive filter algorithms is given by the signal to noise ratio (SNR) results, the improvement of LMS, NLMS and RLS adaptive algorithms is analyzed for 8 KHz sample rate of the random noise variance at the step size of 0.0008. According to the output results of the LMS, NLMS and RLS adaptive algorithms, the signal to noise ratio (SNR) has improved in dB for given the given 16 KHz audio speech file.

From the analysis of the different adaptive filter algorithms, we can say that RLS algorithm offers the best convergence, algorithm stability and smaller error than the LMS and the NLMS algorithms with respect to the unknown system.

8.REFERENCES

[1] Udawat, A. , Sharma, P.C. ; Katiyal, S., “Performance analysis and comparison of adaptive beam forming algorithms for Smart Antenna Systems”, Next Generation Networks, 2010 International Conference, p.p. 1 – 5, Sept. 2010

[2] Djigan, V.I. , “Adaptive filtering algorithms with quatratized cost function for Linearly Constrained arrays”, Antenna Theory and Techniques (ICATT), 2013 IX International Conference, p.p. 214 – 216, Sep. 2013.

[3] Soumya, R.G., Naveen, N., Lal, M.J., “Application of Adaptive Filter Using Adaptive Line Enhancer Techniques” Advances in Computing and Communications (ICACC), 2013 Third International Conference, p.p. 165 – 168, Aug. 2013.

[4] Yu Xia “Performance analysis of adaptive filters for time- varying systems”, Control Conference (CCC), 2013 32nd Chinese, p.p. 8572 – 8575, July 2013.

[5] Huang, Y.J. Wang, Y.W. ; Meng, F.J. ; Wang, G.L., “A spatial spectrum estimation algorithm based on adaptive beam forming nulling” Intelligent Control and Information Processing (ICICIP), 2013 Fourth International Conference, p.p. 220 – 224, June 2013.

[6] Simulink (Simulation and Model Based Design), Mathworks Product. www.mathworks.com

[7] Yuu Seng lau et. al. “Performance of Adaptive Filtering Algorithms: A Comparative Study”.

[8] Bernard Widrow “Adaptive Noise Cancelling: Principles and Applications”.

[9] Udo Zolzer. “Digital Audio Effects”.

[10] James Brain Richardson. “LPC-Synthesis Mixture: A Low Computational Cost Speech Enhancement Algorithm”.

[11] Philipos C. Loizou “Speech Enhancement Theory and Practice”.