Performance Analysis Of Regularized Adaptive Algorithms For Noise Suppression In Speech Signals.

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Abstract: *In this Paper, Regularized NLMS algorithm is proposed to enhance the speech signal from the noisy speech. Speech enhancement is a long standing problem with numerous applications like teleconferencing, VoIP, hearing aids and speech recognition degradations. The motivation The motivation behind this research work is to obtain a clean speech signal of higher quality by applying the optimal noise cancellation technique. Real-time adaptive filtering algorithms seem to be the best candidate among all categories of the speech enhancement methods. Experiments were performed on noisy data which was prepared by adding AWGN, We then compare the noise cancellation performance of proposed Regularized NLMS algorithm with existing Spectral Subtraction method and NLMS algorithm in terms of Mean Squared Error (MSE), Signal to Noise ratio (SNR). Based on the performance evaluation, the proposed Regularized NLMS algorithm was found to be a better optimal noise cancellation technique in speech enhancement applications.*

***Keywords*:** *spectral Subtraction, Normalised NLMS, Regularised NLMS, SNR, MSE*

I. INTRODUCTION

Noise is any unwanted disturbance present in a signal and it is normally corrupting the signal of interest and therefore extracting the reliable information from the corrupted signal is always important in any signal processing application. Noise cancellation can be used in areas where noise can be harmful to one’s hearing, such as: engine rooms or aircraft runways. The effect of interfering noise in speech is considered as the severe problem in speech signal processing. Speech signals are often corrupted by various types ofdegraded. The most common degradations include background noise, reverberation and speech from competingspeaker(s). Degraded speech offers poor perceptual quality and poor intelligibility that leads to listener fatigue and degraded performance in tasks like speech and speaker recognition respectively [1].The aim of speech enhancement is to improve speech quality and intelligibility of degraded speech signal using speech processing techniques. The term intelligibility is related to the behind this research work is to obtain a clean speech signal of higher quality by applying the optimal noise cancellation technique. Real-time adaptive filtering algorithms seem to be the best candidate among all categories of the

amount of speech content that is recognized correctly and quality is related to the aspect of the speech that determines the ease with which one can understand the speech. In many speech communication systems, the presence of background interference causes the quality or intelligibility of speech to degrade. Hence, the signal has to be cleaned up with noise cancellation technique before it is stored, analysed, transmitted, or processed [2]. The study of cancelling noise from a wanted signal arises from need to achieve stronger signal to noise ratios.
Several noise reduction techniques aim to suppress the noise in a noisy speech signal without distorting original(clean) signal with an underlying assumption that the system itself is ideal (i.e. system is not adding any noise to the signalby itself) and that the only environmental sources of noise areresponsible for signal distortion.The hearing impaired people experience great difficulties to communicate in the noisy environment. Under thiscondition, the hearing aid technology is used to increase the speech signal quality and reduce the hearing loss in such way that these hearing impaired people hearing the same level of the speech signal which is hear by the normal hearing people. In this technology, speech enhancement methods are widely used to reduce the noise and to enhance speech signal quality with the acceptable hearing loss. Conventional speech enhancement methods such as a Spectral subtraction method, Normalised NLMS and Regularized NLMS etc., are based on variants of Short-Time Spectral Amplitude (STSA) estimates of speech [20], [5].

Noise Model for the speech signal:

A microphone picking up the voice of a local speaker may be corrupted by various sources of noise which come from outside of the speaker’s proximity. An example may be the engine noise in a car interior, the wind ,the rush on a busy street or in a building ,usually the background noise is considered to be additive ,since at any time instant ,the noise is simply added to the clean speech signal

$$x(n) = s(n) + w(n) \quad (1)$$

$w(n)$ is a additive white Gaussian noise whose density function is given by

$$f(x) = \frac{1}{\sqrt{2\pi\sigma_x^2}} e^{\frac{-(x-\mu_x)^2}{2\sigma_x^2}} \quad (2)$$

Where μ_x is the mean of the random noise and σ_x^2 is the variance of the random noise

.This paper is organized as follows: Section II describes about the Spectral Subtraction for speech Enhancement in speech processing applications and Section IIIprovides the existing NLMS algorithms. The proposed Regularized NLMS adaptive noise reduction algorithm is described in Section IV.Section V illustrates the experimental results of subjective and objective measures of the existing and proposed algorithms and Section VI gives the conclusion of this paper.

II.SPECTRAL SUBTRACTION METHOD

In this research, we focused on spectral subtraction noise removal approach in speech processing **[6].**Our experiment involved sampling two different signals: a real-time speech signal "Real graph" and a noise signal generated by a white Gaussian noise. Using Matlab, we digitally added the white Gaussian noise to the speech signal "Real graph", thus obtaining a noisy speech signal. Noise removal cannot be successfully implemented in the time domain; rather, it is performed in the frequency domain. Our spectral subtraction noise removal approach involves segmenting the noisy speech signal into half-overlapped time domain data buffers multiplied by a Hanning window and then transforming the result into the frequency domain using the fast Fourier transform (FFT).Subsequently, noise is removed by subtracting the average magnitude of the noise spectrum from the noisy speech spectrum and zeroing out the negative values using half-wave rectification. Finally, after removing the noise from the noisy speech, we reconstructed the noise-reduced speech back to time domain using the Inverse Fast Fourier Transform (IFFT) [7]. We were able to listen to the reconstructed speech and we observed that the noise had effectively been reduced. Statistical evaluation of the results was accomplished by calculating the Speech to Noise Ratio (*SNR*) [8]. In order to improve the performance, we applied the technique of frames averaging [9]. Moreover, we studied the effect of varying the overlapping lengths of the data buffers and the Hanning windows on improving the *SNR*.

$$\hat{S}[K] = \left|\hat{X}[K]\right| - \mu(K) \quad (3)$$

Where $\left|\hat{X}[K]\right|$ magnitude of the Noisy speech frame during speech activity, and

$$\mu(K) = E\{|N[K]|\}$$

s͂(n)=IFFT{S͂[K] } (4)

Drawback :The major drawback of the spectral subtraction method is that the enhanced speech is accompanied by an annoying tonal characteristic and affects the human listening ,known as musical noise .This noise is sometimes more disturbing not only for the human ear ,but also for speaker recognition systems .

II. NORMALISEDLMS (NLMS) ALGORITHM

The NLMS algorithms, is the most popular adaptive algorithm which has been used for almost 40 years .This algorithm conceptually stems from the Least Mean Squares (LMS)algorithm ,which has been first developed by Widrow and Hoff. The LMS algorithm is a stochastic gradient method that updates iteratively the coefficients of its own adaptive filter(the tap weights) ina direction of the gradient vector .This vector is calculated as a partial derivative of the Mean square Error(MSE) function with respect to the tap weight .The LMS algorithm is described by the following equations

$$e(n) = d(n) - w^T(n)\boldsymbol{x}(\boldsymbol{n}) \quad (5)$$

$$w(n+1) = w(n) + \mu_{LMS}\ x(n)e(n) \quad (6)$$

The term $x(n)e(n)$, which is incorporated in the tap-weight correction term in equation(4)

Is an instantaneous estimate which is given by $\frac{\partial J_w}{\partial w_n} = -2e(n)x(n)$ .Thus the adaptation proceeds along the gradient of the MSE function until a minimum point of the function is reached. The step size µ is fixed basedon the statistics of the input signal which causes slow convergence. Generally in the noisy environment, the statisticsof the input signal are unknown. The iterative, weight adjustment process of the NLMS algorithm is described by $w(n+1) = w(n) + \beta \frac{x(n)}{\|x(n)\|^2} e(n)$ (7)

In this method, the step size$\mu(n) = \frac{\beta}{\|x(n)\|^2}$is normalized and it is expressed as, where, $\beta$is the normalized step size with $0 < \beta < 2$,In this case, the filter coefficients are updated as

Where μ is the step size controlling the convergence rate, stability and misadjustment and it is converged more rapidly than other LMS algorithms

IV.REGULARISED NLMS:

An adaptive filter is a dynamic filter, which self-adjusts its transfer function according to an optimization algorithm driven by an error signal. In adaptive filtering, we always have a linear system of equations, which are over determined or underdetermined. We face an ill-posed problem to solve these equations and also when the observation data is noisy, which is common in all applications. By using regularization concept we optimize this problem and this is done by adding additional information to the existing system. As a result, regularization is an important design part in any adaptive filter to behave properly.
The regularization parameter (δ) is taken as

$$\delta = \beta\sigma_X^2 \qquad (8)$$

Where $\sigma_X^2 = \mathrm{E}[x^2(n)]$ is the variance of the zero-mean input x(n),where E[.] denoting mathematical expectation, and β is a positive constant. The desired signal is given by the speech signal added with the Gaussian noise which is given as the primary signal (or) the desired signal

Table: shows the SNR and MSE of the various

Algorithm	SNR(dB)	MSE	Elapsed Time
Spectral Subtraction	37.90	0.0047	30.55
NLMS	36.28	0.00045	29.87
Regularized NLMS	42.23	0.00018	29.66

techniques for speech signals added with noise.

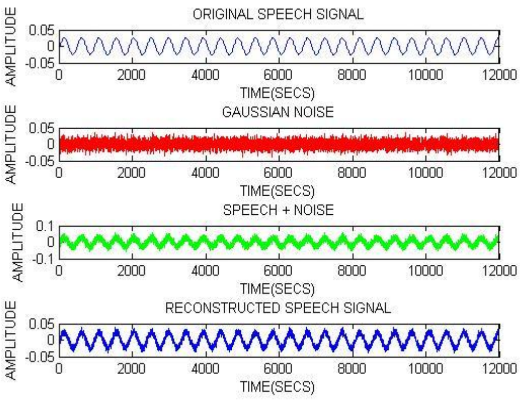

Figure1. Time domain representation ofOriginal speech signal, Gaussian noise, Noisy speech (Speech + Noise) and Reconstructed Signal after Spectral Subtraction.

$$d(n) = x(n) + w(n) \qquad (9)$$

The error signal is given by

$$e(n) = d(n) - y(n) \qquad (10)$$

The weight update equation is given by $h(n) = h(n-1) + \frac{\alpha\, x(n)e(n)}{x^T(n)x(n)+\delta}$ (11)

V.SIMULATIONRESULTS:

This section provides the simulation results and performance evaluation of the proposed speech enhancement algorithm, Normalised Least mean square algorithm and regularised NLMS and its comparisons with the spectral subtraction method .For simulations .we have employed matlab software as the simulation platform, for the experimental purpose, the clean speech signals have been recorded from the microphone for17s and the additive white Gaussian is add
The Normalised NLMS algorithm outputs are shown in the figure 4.

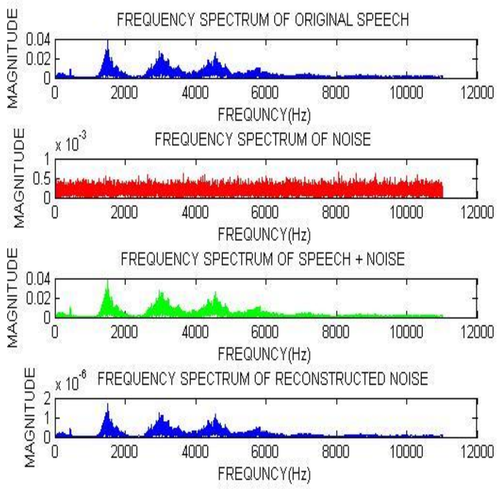

Figure2. Frequency domain representation of Original speech signal, Gaussian noise, Noisy speech (Speech + Noise) and Reconstructed Signal after Spectral Subtraction.

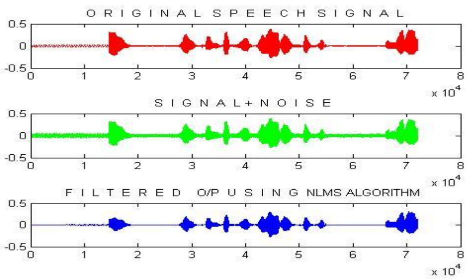

Figure 3. The NLMS filter output

REGULARIZED NLMS:

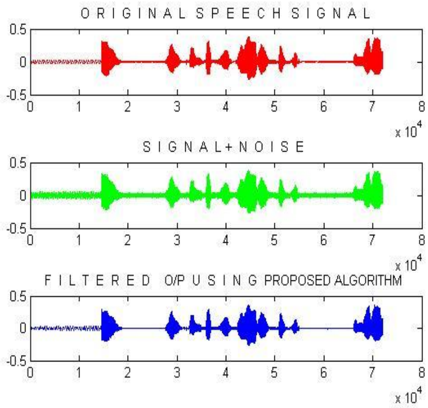

Figure 4. The regularised NLMS filter output

CONCLUSION:

In this work different adaptive algorithms are tested for noisy signal. Spectral subtraction method is first implemented to enhance the noisy speech signal. Then NLMS, Regularised NLMS adaptive algorithms are applied for enhancement. The performance analysis of proposed noise reduction algorithm improves the speech signal quality. Table 1 shows the SNR and MSE values of signals for different algorithms. By comparing the results, Regularised NLMS provides better SNR and MSE is found in case of proposed algorithm.

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