2021 6th International Conference for Convergence in Technology (I2CT)
Pune, India. Apr 02-04, 2021

Speech Enhancement Using Recursive Least Square Based on Real-time adaptive filtering algorithm

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***Abstract*— In real time environment a speech signal is often corrupted and losses its characteristics either by natural disturbances or anything. The key aim of our planned algorithm is toward increase the speech intelligibility and quality. In order to do that a filter has been developed using Recursive Least Squares (RLS) algorithms and Least Mean Square (LMS). Real-time adaptive filtering algorithms are one of the best methods used for the speech enhancement methods. In this research work we have proposed the recursive least square which is under adaptive filtering method for the enhancement of the speech signal. Initially we compare the performance of noise cancellation of the proposed Recursive least square which uses objective evaluations that is based on wavelet based speech enhancement like Signal to noise ratio Loss, Signal to Noise ratio and Mean Squared Error. Based on the Objective and Subjective evaluation, it was found that this algorithm clearly in increases the intelligibility and removes the corrupted noise in the waveforms. There are different types of filters like Kalman filter, Wiener filter, Spectral subtraction, and notch filter and wavelet methods. The performance of every filter depends upon the intelligibility also excellence of the speech signal. The reduction or augmentation in the SNR ratio is the main aim of the most methods. Adaptive filtering is a technique which uses certain predefined criterion like the estimated mean squared error or the correlation has to be considered for the analyses of the waveform. In this adaptive filter, we use coefficients with weights and an adaptive algorithm updates are made available.**

Keywords*— *Digital Signal Processor, Speech Enhancement, Adaptive Filter, Discrete Wavelet Transform, Real-Time, Kalman Filter

I. Introduction

The main aim of all algorithms in the signal processing is to progress the simplicity and value of the speech signal without affecting the superiority of the signal. In order to achieve better noise cancellation estimation of the noise in a particular signal is much important. Removing noise is a major issue in recent decades because it remains extremely complex to calculate the spectral properties of non-stationary noise. In the process of enhancing the speech the noise estimation is the issue which is complicated and treated as noise. Speech processing has spread its wings almost in many applications in telecommunication systems, medical equipment signal processing, hearing devices; ATM processing it is most complicated and interested research area. There are many noise cancellation algorithms, but the features of noise also vary depending on the time and level of interference. Algorithms, Wiener filter, signal Subspace and wavelet enhancement It can amplify sounds but is affected by environmental disturbances or other sounds. From the noisy signal magnitude it undergoes subtraction of the magnitude spectrum then approximating the spectrum of the original signal.

Later, Researcher's made many changes to the kalman filter, but did not reach the expected point and also computational characteristics are more. The paper has solved problem of colored noises and de-noising the random in addition to coefficient estimation. The colored noise was also considered to be an autoregressive creation.. Thus, thus we determined that the signal's AR coefficients is calculated using a linear estimation and the residuals. In our observation, to solve the stated problem Kalman filter base technique is used among preprocessing called digital expander of noisy speech signal which is proposed in order to reconstruct the signal and the addictive noise is model the same as the auto regressive (AR) process. The estimated auto-regressive (AR) speech parameters are derived from a linear model. This paper deals with the calculation of coefficients, including obscuration of coloured signals. The noise is also an auto regressive process. Its variants and coefficients by LPC are estimated in the similar way.

II. Existing Methods

There are different methods individually and also combined with other filters to study and get better the performance of speech signal. In noisy atmosphere the discrete wavelet transform combined with the Kalman filter used for signal speech enhancement. On fixed point digital signal processing (DSP) is implemented to progress the excellence for Texas instruments and transparency of the signal. To filter the noisy speech signal we apply Kalman filter. And then the discrete wavelet transform be conducted toward the filtered frames. Then, toward track the variation of noisy signal, the noisy discrete wavelet coefficients is truncated for soft thresholding according to the input noise level, and then to get the enhanced signal we apply inverse discrete wavelet coefficient. To reduce the white and the colored noise we proposed speech enhancement is adopted properly from the noisy speech. The TIMIT database is applied with proposed method of simulation results are compared with the discrete wavelet transform approach.

Wavelet based methods for de-noising of speech signals are widely used in speech signals which are corrupted with background noise. A filter is eliminates noise by low pass filtering which distorts the noisy features in the speech signal, the methods which uses wavelets for large variety of signals are proves to providing good performance. Wavelet transform which is based on speech enhancement is applied in many applications. There are number of issues for a successful noise suppression application to be treated for the

2021 6th International Conference for Convergence in Technology (I2CT) | 978-1-7281-8876-8/21/$31.00 ©2021 IEEE | DOI: 10.1109/I2CT51068.2021.9417929

real time speech signal corrupted by additive noise to be successful. In the past, DWT enhancement has already dealt with, among other things, with non-Gaussian white noise introduced into real-time audio recordings.

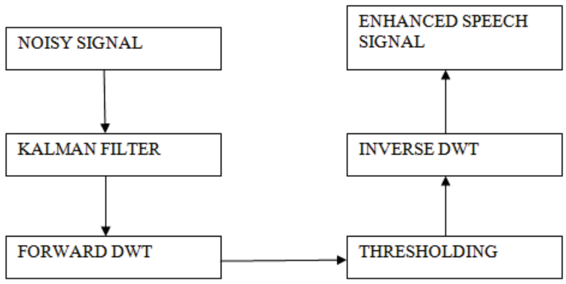

Fig. 1. Block Diagram of Speech Enhancement Algorithm

A. Proposed Methods

Noise cancellation can be used by Fixed or Adaptive filters. Prior knowledge of how noise and fixed filter design rules create signal is essential for implementation of filters. For example, we need to design a filter to remove the frequency band occupied by the noise so we also need to know what kind of signal (frequencies)passes what (frequency) Then, impulse response automatically adjusted by the Adaptive filters, and the little or no knowledge of signal or noise characteristics for their design. The main aim is to separate the noise from a signal of an adaptive filter and to improve the signal toward noise ratio.

Present two inputs for Adaptive Noise Canceller (ANC) one is primary input also orientation input. Non signalienteatory information is fed back to the source that has no relationship to the original input. In contrast to the "tuned input", which has nothing to do with the signal, the "uncorrelated input" has nothing but contributes to the overall background noise in the system. To produce the close estimation input signal the noise is passed through the adaptive filter i.e. y(n). Toward produce the estimation of error e(n), the estimated noise is subtracted from the corrupted signal. Over many years the adaptive filters from the researchers have gained attention. As a result, various algorithms have been developed which are computationally efficient.

$$y(t) = s(t) + n(t) \text{ --} \tag{1}$$

$$S = W(Y) \tag{2}$$

$$Z = D(Y, S) \tag{3}$$

$$S = W^{-1} \tag{4}$$

$$\Psi a,b(t) = [(t-b)/a] \ a, b \ ] \tag{5}$$

III. METHODOLOGY

The proposed method is the RLS algorithm for the effective noise cancellation and block diagram is as shown below. The details of every step have been explained in below. RLS method acts as the best solution for the mean square error solutions. Here, the segments of the decomposition are applied to every frame of the DWT and then segmented to obtain the noise signal. After that we will obtain the mother wavelet as variance of the minimizing error and maximizing the signal to noise ratio in between the original and the reconstructed signal.

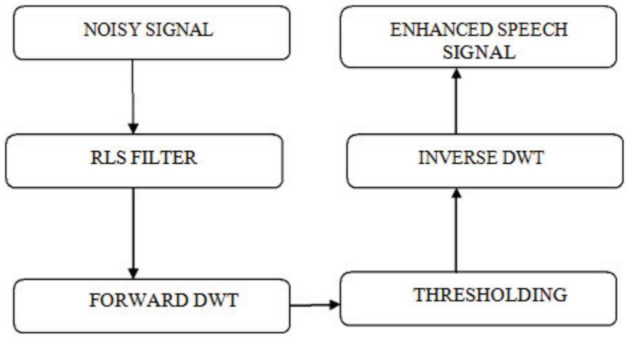

Fig. 2. Block diagram of planned speech enhancement algorithm.

IV. ADAPTIVE ALGORITHMS

In adaptive noise cancellation there are basically two types of algorithms (i) Lease mean square(LMS) (ii)Recursive least square .Generally the least mean square algorithm is one of the most basically used algorithm because of its error coefficients and minimum mean square error. Let us consider the output be y(n). FIR filter can be calculated from Eq. (1).

$$y(n) = \sum_{m=0}^{N-1} w(m)x(n-m) \tag{6}$$

Where N is the order of filter used also n is the no. of iterations n=1,2,…. The output error signal is calculated using

$$e(n) = d(n) - y(n) \tag{7}$$

Here we use current weight value w(n) and obtain the required value

$$w(n+1) = w(n) + \mu e(n)x(n) \tag{8}$$

Error value can be estimated by the tap weight and eq(8) right hand side is the tap adjustment that is a[pplied to w(n).

$$\mu(n) = \frac{\alpha}{c + \|x(n)\|^2} \tag{9}$$

$$w(n+1) = w(n) + \frac{\alpha}{c + \|x(n)\|^2} e(n)x(n) \tag{10}$$

The RLS algorithm has the high performance and computational complexity and it also has some stablity issues inspite of all these it can be considered as one of the finest algorithm for the noise cancellation. Here the filter tap weight vector is also known and then given by Eq(11).

$$w(n) = \overline{w}^T(n-1) + k(n)\overline{e_{n-1}}(n) \tag{11}$$

$$k(n) = \frac{u(n)}{\lambda} + x^T(n)u(n)) \tag{12}$$

$$u(n) = \overline{w_\lambda}^{-1}(n-1)x(n) \tag{13}$$

$$\overline{y_{n-1}}(n) = \overline{w}^T(n-1)x(n) \tag{14}$$

Where λ is an insignificant constant, but consistent . The filter tap weights are calculated by the vector of previous iterationsand also the current input vector as well. This is shown by the Eq(14) and the error signal is given by the Eq(15)

$$\overline{e_{n-1}}(n) = d(n) - \overline{y_{n-1}}(n) \tag{15}$$

V. Recursive Least Squares (RLS) Algorithm

There are certain situations in speech enhancement that the signal and its characteristics change rapidly and cannot be estimated then in that situation the algorithm used has to be very active in predicting the changes and in that case recursive least square algorithm has considered to be one of the fastest algorithm.

The Proposed algorithm repeatedly finds out a coefficient which reduces the cost function of the linear least squares. The main advantage of the RLS is that it has got the ability to adjust its coefficients automatically without our intervention. This formulation returns an exact values, which minimize the sum of the squares of the mean squared magnitude of the signal.

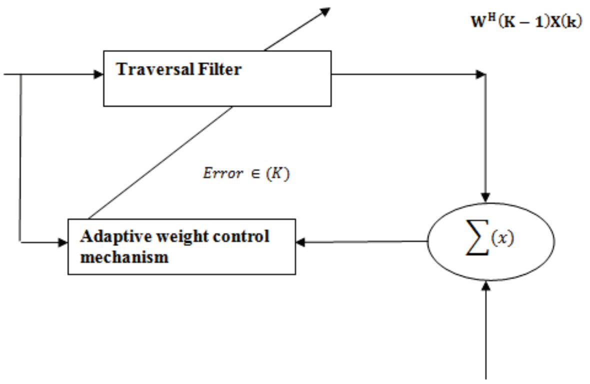

Fig. 3. Design of RLS Algorithm

VI. Simulation Results

We tabulate the values of both error signal and the noisy signal by selecting the type of signal at the top right. We have highlighted the noisy observations and adjusted the measurement settings on the right.

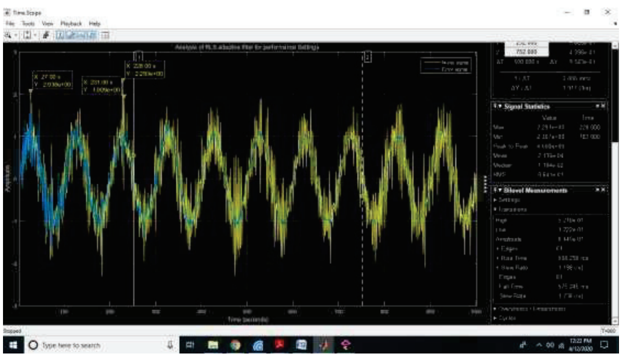

Fig. 4. Noisy signal analysis of RLS Adaptive filter

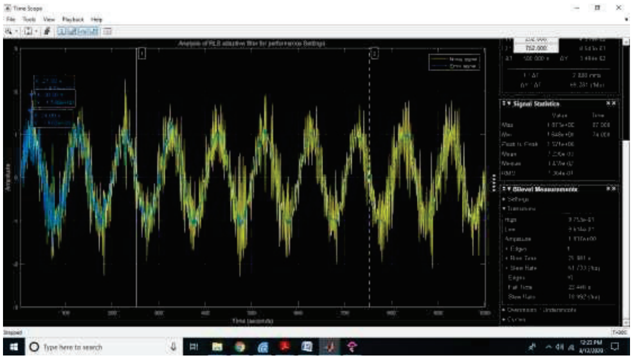

Fig. 5. Noisy signal analysis of RLS Adaptive filter

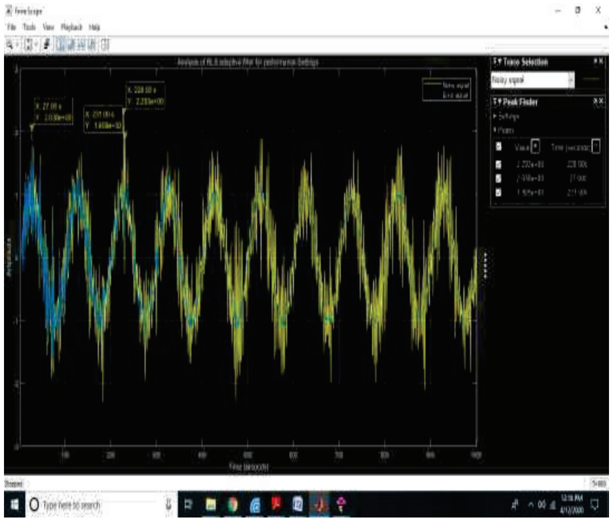

Fig. 6. Error signal analysis of RLS Adaptive filter

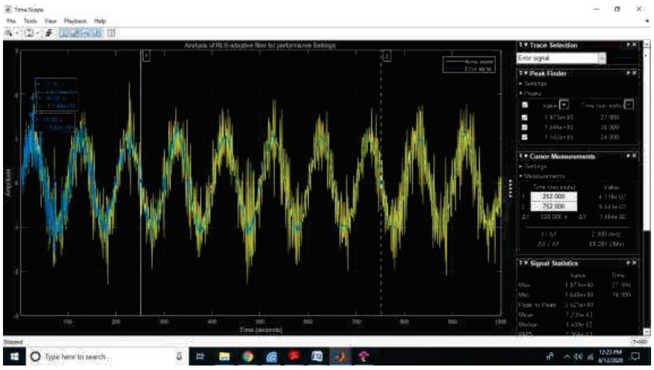

Fig. 7. Error signal analysis of RLS Adaptive filter

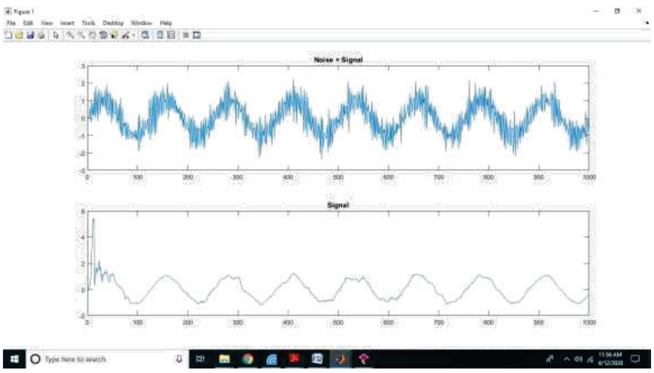

Fig. 8. Analysis of RLS Adaptive filter for performance Analysis

VII. Conclusion and Future Scope

In our project we have studied and performed a new algorithm based on Recursive least square adaptive filtering method. This method removes the noise levels in a particular signal and helps us to analyses the characteristics of the signal easily. The speech signal intelligibility and quality of the signal is increased without affecting the quality of the signal. The performance of the system has been increased when compared to all other algorithms. The mean square error (MSE), SNR ratio etc., have been used for the experimental analysis.

We know that as the SNR increases the output quality of the image or speech is also increased. So using MATLAB code we have performed simulations of various waveforms observed the values and calculated the MSE and SNR. This method appears to reduce noise. Finally, in the real-time test of speech de-noising the proposed method has been successfully implemented and reveals that the proposed algorithm has significantly improved the speech intelligibility.

VIII. Future Scope

As a future work, we will try to implement our system in other real time platforms as ZYBO boards and Raspberry-PI to enhance the speech signals.

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