SPEECH SIGNAL IMPROVEMENT USING ADAPTIVE ALGORITHM BY CONVERGENCE FACTOR VARIATION

M. Tahir Qadri, Ali Akbar Siddiqui, Noman Ahmed Siddiqui, Fahad Farooq.

Abstract —over the past few decades, controlling of noise has become a monumental amount of research. In our proposed work we utilized the concept of adaptive filter for noise cancellation by varying the convergence parameters in order to obtain the improvise communication signal (speech signal), it can be considered as an alternative method for estimation the corrupted signal by additive noise. Signals are very sophisticated to work with as they include multiple frequencies within them and some algorithms are required to amplify them at a point where they can be of any use. We utilize an adaptive filter on MATLAB with changing value of converging factor "μ" and analyze the algorithms in terms of LMS (Least Mean Square) algorithm.

Keywords: Adaptive Algorithm, Signal Processing, Noise cancellation.

I. INTRODUCTION

Noise, distortion and interference in the signal have been a major factor and a source for corrupting the signals present in the environment and have gained attention of the researchers due to the recent growth of technology leading to heavy machinery, high speed wind buffeting, noisy engines, and any other unwanted noise sources [1]. The information transmitted to the receiver side, noise automatically added to the signal from the surrounding, then this acoustic noise carried by the headphone which produce distortion and destructive interference signal, ultimately it resist the intelligibility of the audio signal. Therefore an active area of research has playing a role to prevent the noise to devastate the desired signal.

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The characteristic of adaptive filter is one of the most popular proposed methods utilized to reduce the corruption caused by unwanted signal (noise) [3].

Algorithm used widely for updating the weighted values are: the LMS (Least Mean Square), NLMS (Normalized Least Mean Square) and the RLS (Recursive Least Square) algorithm [6]. LMS is basically the one most utilized for its simplicity in stationary environment, better tracking capabilities, robustness [5].

This error Utilization of an adaptive technique in noise elimination is to reduce the noise to a level that it may not be of any consequence and also to improve the signal to noise ratio (SNR).

An adaptive filter is capable of adjusting the coefficients by itself and change the behavior of the process in time. Adaptation allows the filter to adapt the response of coefficients with respect to the input signal.

Signal generated from the process output to the system input is used to adjust the filter's weights for the next incoming sample incoming [4].

Therefore in this paper we will be utilizing the Least Mean Square (LMS) adaptive filter algorithm while changing the convergent factor (μ) in order to gather precise and approximate signal responses.

II. CONCEPT OF NOISE

It is important to reduce the noise from the signal till it almost negligible because it may cause disturbance in a long term. There are many ways for the noise to corrupt our signal and to eliminate them we require some algorithm which is capable enough to tackle all kind of noises in almost any environment. If we are to reduce with frequencies that are known to us, in which case there is no need for implementing Adaptive Algorithm, a simple filter will suffice for such application.

This much is true that noise is never constant or remains the same; it tends to vary according to the environment. An adaptive filter aim is to adjust its impulse response to filter the desired signal; they may or may not require prior knowledge based on signal and noise characteristic. Basically an

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adaptive filter can adjust itself even in non-stationary conditions.

III. ADAPTIVE ALGORITHM

In this paper, we implemented Least Mean Square (LMS) as our Adaptive filter. They basically falls in the class of an adaptive filter which intend to find the weighted values that produce minimum error to its output. Eq(1) is a mathematical representation of an Adaptive LMS Algorithm

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

Where,
N is the number of iterations,
w(m) are the weighted vector.

$$w(n+1) = w(n) + \mu e(n)x(n) \qquad Eq(2)$$

In eq(2), w(n+1) is a next weighted value to be calculated. It utilizes the previous value of w(n), error e(n) and the input signal x(n) along with convergence factor μ to predict the upcoming value. It tends to reduce the error by evaluating weighted values of previous iterations. μ determines the convergence speed and its overall behavior [7-10].

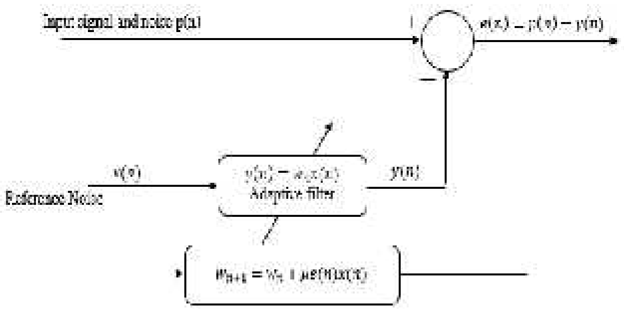

Figure.1: Block Diagram

Fig-1 is a block representation of the adaptive system, the signal p(n) is basically the sum of the original signal with the reference noise. The error generated from the difference of input signal with noise and y(n) which is the output of our adaptive filter. This Adaptive filter uses a Least mean squared (LMS) technique based on Weiner solution represent in eq-2. This is the most commonly used adaptive algorithm for enhancing the input signal or speech. Basically convergence factor is a rate at which the coefficients tend to change. We can vary it to make our system more robust but at a cost of not providing the algorithm enough time to cater the error signal, due to which its output has a lot of disturbance in terms of noise.

We can identify the best weighted vectors at the given time instant for N number of samples with eq(3), where W is a weights to be identified, R is the correlation of an input signal to itself and P is a correlation of the corrupted signal with the input signal.

$$W = R^{-1}P \qquad Eq(3)$$

In fig-2, we considered a simple sin wave for our input signal corrupted with some randomly generated noise, and the clean signal obtained with 7.87% approximate error when the value of μ is 0.1. Fig-2 represents the speech signal with the convergence rate of 0.01 generating an output signal with an approximate error of 6.2%. I fig-3 the convergence factor is tuned to 0.001; hence the output signal generated has an approximate error of 3.3%.

By analyzing the signal with different convergence rates, we can observer that if the convergence rate is set to 0.1, the adaptive filter will become more robust and try to reduce the error as fast as possible by utilizing little number of samples. This cause the adaptive filter even less time to tackle the errors in signal, hence an extensive part of the noise will still corrupt the signal. If it is set to 0.01 the error is reduced to 6.2%, which means it has more time now to tackle the error but there is a tradeoff, if the convergence rate is reduced, it will take more time for the filter to reduce the noise in our speech signal.

Table.1: Convergence Rates with approximate error

S.No	μ	Number of Samples	Error (Approximate)
1	0.1	1	7.87%
2	0.01	2.5	6.8%
3	0.001	5	3.3%

Table.1 displays the number of samples taken to reduce the noise with respective convergence rates, and the approximate error is generated with those rates.

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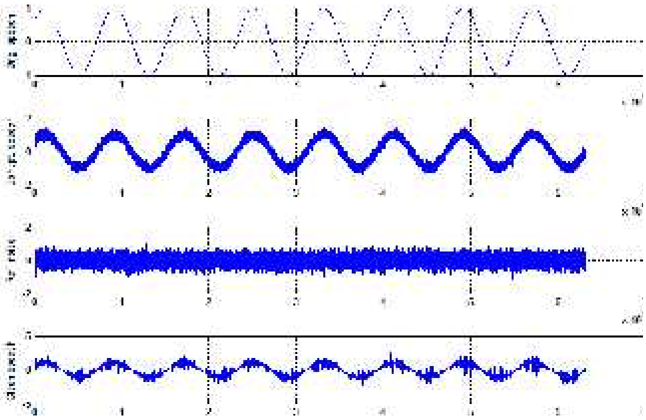

Figure.2: Speech signal with noise at μ = 0.1

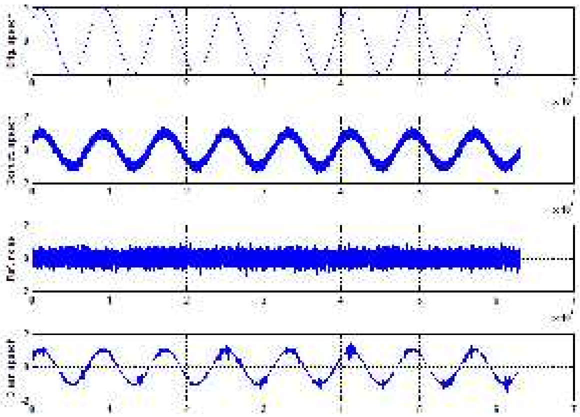

Figure.3: Speech signal with noise at μ = 0.01

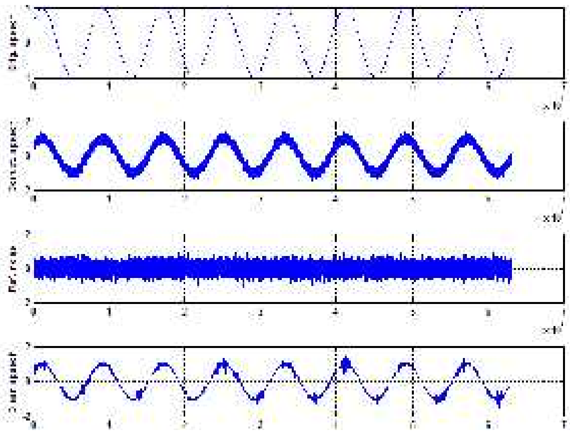

Figure.4: Speech signal with noise at μ = 0.001

IV. RESULTS

Results are also accumulated with the build-in chirp signal in MATLAB with three different convergence factors in Fig-5,6 and 7. The results obtained were not different than that of the sin wave.

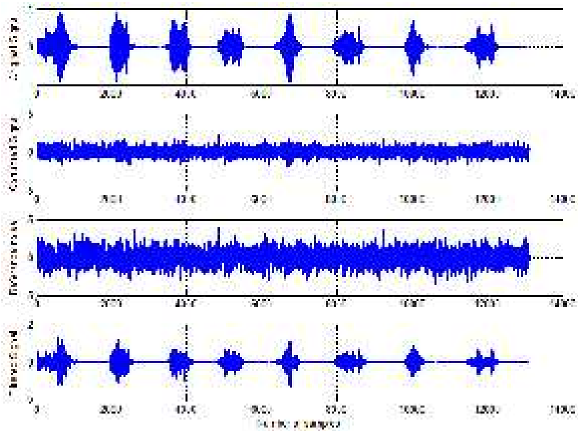

Figure.5: Chirp signal with 0.01 convergence factor.

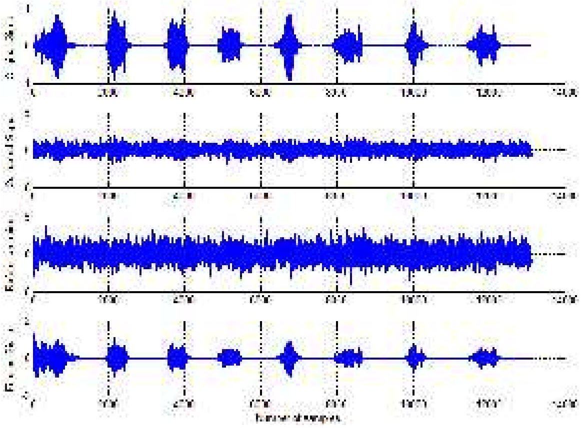

Figure.6: Chirp signal with 0.001 convergence factor.

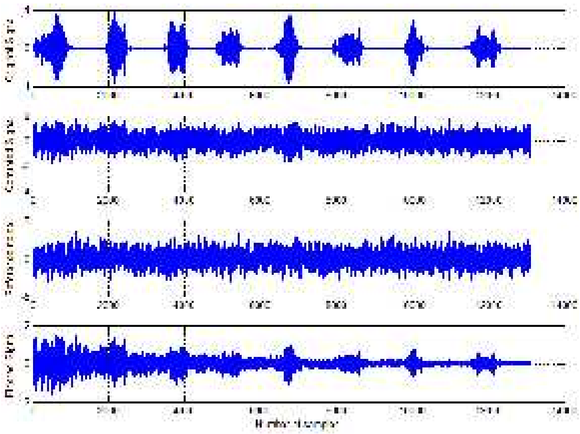

Figure.7: Chirp signal with 0.0001 convergence factor.

In fig.7, convergence rate is set to 0.0001 which must produce an output signal with minimum error but in turn is seems to generate much greater error. The reason for this is the convergence rate is set so low that it is utilizing almost whole signal to reduce the noise.

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V. CONCLUSION

In this paper, we have seen that the convergence parameter has a vital role in improvising a signal However; a certain changes in convergence factor may cause even more distortion in a speech signal. The whole implementation of algorithms was successfully achieved, and carried out using MATLAB. By observing the results we can conclude that if step size is chosen correctly so our communication signal can become suitable for non-stationary environment or in other words a respective signal can be achieved.

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