Performance Comparison of Clipping Technique with Adaptive Filters for Impulsive Noise Reduction in AWGN Environment

Sumrin Mehak Kabir, Alina Mirza, Shahzad Amin Sheikh
College of E&ME, National University of Sciences and Technology (NUST),
Islamabad, Pakistan
E-mail: sumrin.mehak75@ee.ceme.edu.pk, alinamirza2002@yahoo.com, sheikh.shahzadamin@gmail.com

***Abstract*— *Impulsive* noise is a non-Gaussian noise that appears in the communication and it may affect the information badly. In the past, methods have been investigated to mitigate this noise. In this paper, clipping based impulsive noise cancellation technique is compared with a previously existing technique comprised of Normalized Least Mean Square (NLMS) and Recursive Least Square (RLS) algorithms. The scheme is tested on binary data modulated over different types of constellation schemes. The performance of the adaptive filters is quite better than that of clipping for different modulation schemes namely Quaderature Phase Shift Keying (QPSK), 16-Quaderature Amplitude Modulation (QAM) and 32-QAM. The convergence characteristics of both methods are demonstrated by the simulation results in terms of Bit Error Rates (BER).**

Keywords- Impulsive Noise; Adaptive Filter; NLMS; RLS; QPSK; 16QAM; 32QAM; BER.

I. INTRODUCTION

Adaptive Filters with non-stationary statistical characteristics, low cost, and their ability to adapt to the unknown environment make them most suitable for the control applications and signal processing [1]. Therefore adaptive filters have been successfully used in numerous signal processing applications over the past decades.

The adaptive filter systems have general characteristics i.e., an output signal is generated by the adaptive filter and is compared with a desired signal, to generate an error. That error is then used to modify the adjustable coefficients of the filter, generally called filter tap weights, in order to minimize the error.

However practically, noise is impulsive in nature which is non-Gaussian generated by human activities [2] [3] and has more catastrophic effects in communication systems. Nowadays active area of research is to inspect the impulsive noise behavior and suggest solutions to improve the performance of systems by suppressing it. For noise cancellation, clipping technique is implemented in literature which attempt to recover the original transmitted signal [5].

In [6], the performance comparison of the adaptive filter algorithms such as the least mean square (LMS), Normalized LMS (NLMS) and Recursive least squares (RLS) were carried out to remove the noise from the audio signal. Impulsive noise has been removed using NLMS filter over different modulation schemes in [7] on the basis of step size and likelihood probabilities.

In this paper, a comparison of the already existing NLMS and RLS filters with clipping method for impulsive noise cancellation has been presented. Though clipping method is simple in terms of implementation and carrying out the parameters, it can be used for impulsive noise removal in the presence of Additive White Gaussian Noise (AWGN).

The paper is organized as: Section II briefly describes the basic principle of noise cancellation. Section III gives the review of clipping method which is followed by discussion of different adaptive filters in Section IV, supported with the simulation results in Section V. In the end, Section VI concludes the paper followed by the references.

II. IMPULSIVE NOISE GENERATOR MODEL

Impulsive noise has been generated using the model given in Figure 1 in MATLAB/Simulink [2]. It includes the zero-order hold, data source, and sign as a comparator. The output is multiplied by a random number to generate fixed or 1 unit width and variable amplitude impulses as shown in Figure 2.

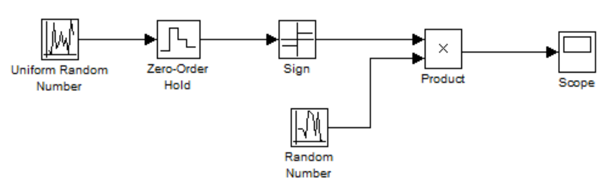

Figure 1. Impulsive noise generator model [2]

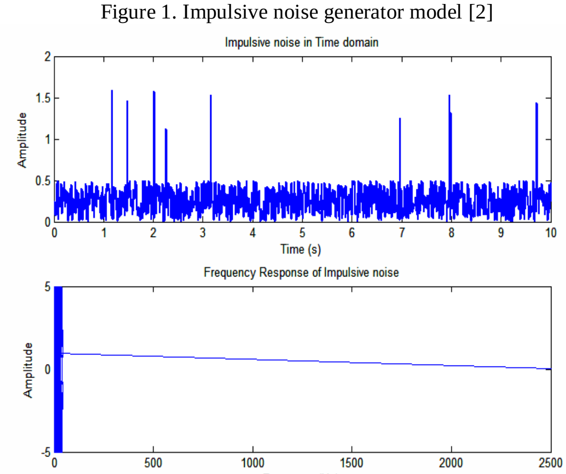

Figure 2. Impulsive noise signal and spectrum

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The frequency response of the generated impulsive noise is also shown in Figure 2. The impulsive noise being a non-Gaussian noise, has a flat response having all the frequencies in equal amount.

III. NOISE CANCELLATION

Adaptive filter have an adaptation algorithm that monitors the environment and vary the filter transfer function accordingly. Each adaptive filter depends on the error signal computed from the adaptive filter to update its filter taps.

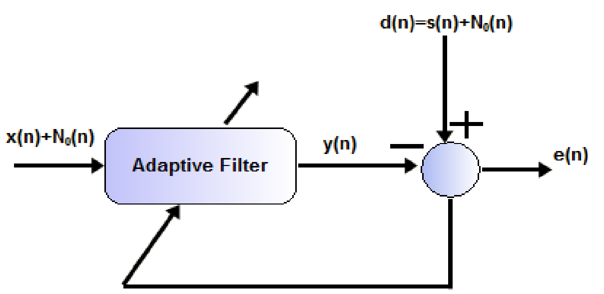

Figure 3. Block Diagram of noise cancellation in adaptive filters

IV. CLIPPING

In practical applications, clipping method is usually used for impulsive noise mitigation due to its simplicity. A clipping algorithm is employed at the receiver end of the AWGN channel, where we presume that impulsive noise is being added by the channel itself during the communication [5]. It is to reckon that clipping method only changes the amplitude of the data without changing the other parameters such as phase.

$$y_k = \begin{cases} r_k & , if\ r_k < T_c \\ T_c\, e^{j arg(r_k)} & , if\ r_k \geq T_c \end{cases} \quad (1)$$

T_c is the Clipping Threshold that can be set according to the maximum value of data if known. And r_k is a sample of the signal to be clipped over T_c. The $arg(r_k)$ is used due to the possibility of the existence of complex valued samples in the signal.

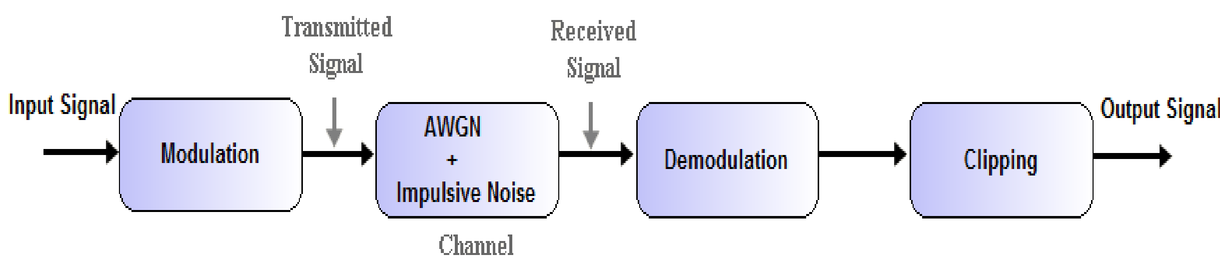

Figure 4. Block diagram of Clipping method

The output of the adaptive filter is compared with the desired signal *d(n)*.The desired signal is a random signal corrupted with another noise source or system noise, as shown in Figure 3. The coefficients of adaptive filters adapt recursively to make the error signal *e(n)* to be minimum so that the output signal *y(n)* approaches to be equivalent to the desired signal having minimum error.

The interference cancellation application of the adaptive filters is employed to cancel the noise from the signal. For that, the adaptive filter updates recursively in order to remove the noise from the input signal using the noise in the desired signal by subtracting it from the output signal.

In this method the amplitude of the received data is clipped or limited by the threshold and no other change occurs to the signal/data that has been received. Figure 4 shows the simple model carried for the clipping method.

V. ADAPTIVE FILTERS

There are many adaptive algorithms used for noise removal. The brief summaries of adaptive algorithms which are used in this research are as follows.

A. NLMS

The Normalized Least Mean Square (NLMS) is one of the variants of LMS algorithm whose convergence ate is faster than LMS. The drawback of LMS algorithm is its sensitivity.

To avoid gradient noise amplification problem NLMS algorithm is used in which the tap weight *w(n)* at

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n+1 is normalized with the Euclidean norm of the square of the input to the filter so that the convergence is stable. The filter tap weights are updated using following equation during the recursive procedure:

$$w(n+1) = w(n) + \frac{\mu e(n)x(n)}{\epsilon+\|x(n)^2||} \tag{2}$$

Where ϵ is a small number added for algorithm stability, μ is the step size and $e(n)$ is error signal.

B. RLS

The Recursive least squares (RLS) adaptive filter belongs to the least square family of the adaptive filters. It tends to minimize a linear least cost function related to the input signal by finding the coefficients recursively. Also the input signal for RLS is deterministic. The convergence rate of RLS is far higher than many other adaptive algorithms. However, it costs in higher computational complexity. The weights of the filter are updated by these equations:

$$w(n+1) = w(n) + k(n)x(n) \tag{3}$$

$$k(n) = \frac{\lambda^{-1}\Phi^{-1}(n-1)x(n)}{1+\lambda^{-1}x^T(n)\Phi^{-1}(n-1)x(n)} \tag{4}$$

$$\Phi^{-1}(n) = \lambda^{-1}\Phi^{-1}(n-1) - \lambda^{-1}k(n)x^T(n)\Phi^{-1}(n-1) \tag{5}$$

Where λ is the forgetting factor. Φ^{-1} is the cross correlation matrix.

VI. SIMULATION RESULTS

In the first part of simulations, the binary data perturbed by impulsive noise in Figure 5, is recovered using clipping method. This is shown by initially generating impulsive noise by following steps mentioned in [2] and depicted in Figure 2.

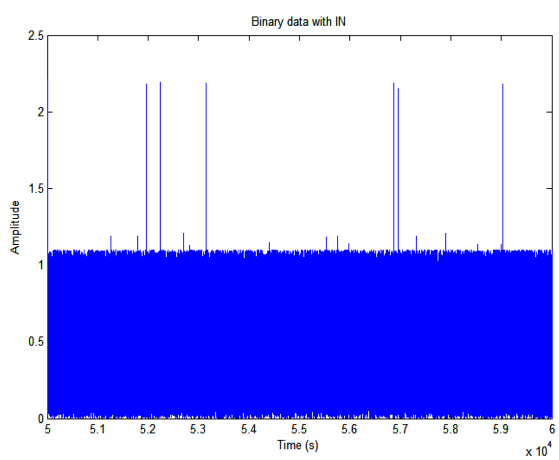

Figure 5. Binary input signal with Impulsive noise

The binary data is generated randomly, comprised of 100,000 randomly generated bits, in MATLAB. This binary data is added with the impulsive noise after modulation as shown in Figure 5 and is then transmitted over an AWGN channel, pretending that impulsive noise has been added during the communication.

For clipping method, the received binary data is clipped over a threshold of amplitude 1 after demodulation. It can be put in the way that the clipping method can clip or limit the data above the threshold but it cannot remove the noise that lie below the threshold. The noise below the threshold is usually caused by additive white Gaussian noise from the channel.

In the second part of the simulations, in adaptive algorithms the demodulated data is filtered using the adaptive filters (NLMS & RLS). The basic scenario is that the output $y(n)$ of the filter is compared with the desired signal *s(n).* Their difference produces the error signal *e(n)* and the filter updates its weights recursively using the adaptive weight update equations (2) and (5), such that the error signal is minimized.

$$\begin{aligned} e(n) &= d(n) - y(n) \\ &= s(n) + n_2(n) - n_2(n) = s(n) \end{aligned} \tag{6}$$

In an optimum sense, the system output signal should contain the original signal as in (6). For these simulations the length of the two adaptive filters is fixed to 32. The step size parameter μ for NLMS Algorithm is chosen to be 0.1 and forgetting factor λ for RLS is 0.98.

Figure 6 represents the bit error rate (BER) plot of clipping method, NLMS algorithm and RLS algorithm for QPSK modulation. The performance of IN reduction methods is compared through the BER plots of the received signal that does not contain impulsive noise and is considered to show the minimum number of errors occurred in the signal, due to the channel only.

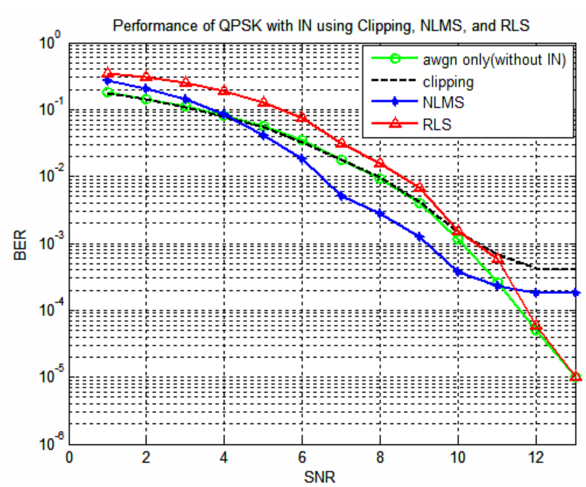

Figure 6. Comparison of BER (dB) over QPSK modulation

It can be seen that among the three algorithms, the performance of RLS is quite better than the rest of the methods for an SNR of up to 20 dB.

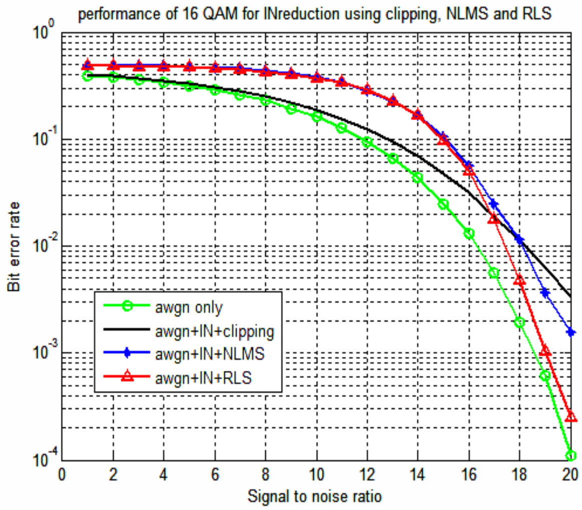

Figure 7. Comparison of BER (dB) over 16-QAM modulation

For 16-QAM modulations or square constellation, the results show that among the performances of the three methods, RLS is better as shown in Figure 7. Similarly, Figure 8 shows the BER comparison for 32-QAM modulation that is a rectangular QAM constellation and the results are hereby proven that RLS depicts far better performance in comparison with the clipping method.

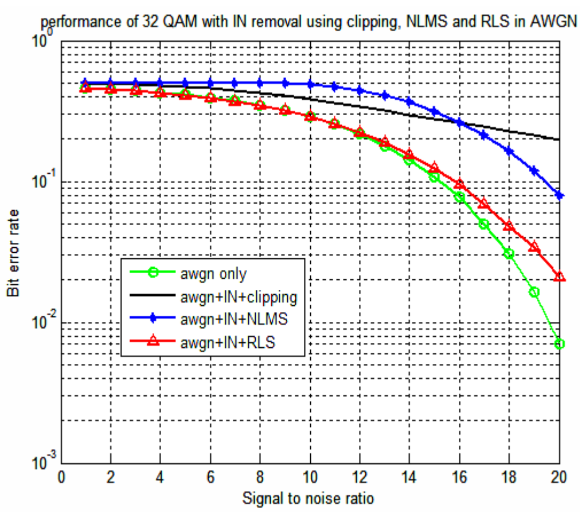

Figure 8. Comparison of BER (dB) over 32-QAM modulation

The performance metric bite error rate in decibel indicates that RLS filter has lowest BER compared to other used algorithms. The BER of RLS filter is close to the performance of minimum possible errors for the removal f impulsive noise. Whereas the BERs of NLMS and clipping method are much greater than the RLS BER and the AWGN only performance that has no impulsive noise.

VII. Conclusion

In this paper, adaptive noise cancellation technique based on Normalized Least Mean Square (NLMS) and Recursive Least Square (RLS) algorithm, are compared with another previously existing technique named as clipping. Due to recursive parameters, the adaptive filters require the reference signal and exhibit better impulsive noise cancellation when compared with the clipping technique. The conducted comparison guarantees that adaptive RLS filter is an efficient noise canceller for different modulation schemes such as QPSK, 16 QAM (square constellation) and 32 QAM (rectangular constellation). It ensured the better performance of RLS in terms of convergence speed and lower BER and is verified by the simulation results.

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