BAUET JOURNAL
Published by
Bangladesh Army University of Engineering & Technology
Journal Homepage: http://journal.bauet.ac.bd

Effects of Impulse Noise on Various Input Signals and Its Mitigation for Time Varying Channel

Rubaiyat Yasmin[1,*], Mithun Kumar[1]

[1]*Department of Information and Communication Engineering, Faculty of Engineering, University of Rajshahi, Rajshahi-6205, Bangladesh*

Abstract: Communication system has its diversified application areas. Different types of input signals are used in service-oriented communication system. Impulse noise may be incorporated during signal transmission through channel and can corrupt the signal. Hence, it is necessary to investigate the effect of impulse noise on different types of input signals. In this paper, we investigated the effect of impulse noise on various types of input signals passing through a time varying channel. We investigated the effect for three different input signals by simulation. It is observed that different types of input signal shows different types of effects on impulse noise. The graphs show that there is an abrupt rise in output signal due to impulse noise which causes an abrupt rise in error figure. In this paper, different adaptive algorithms have been incorporated to reduce the effect of impulse noise. We used mean square error (MSE) and convergence as the performance measures in our simulation. Some research works related to impulse noise shows that using of existing clipping technique and median filter individually is not enough to mitigate the impulse noise effect for time varying channel. Our simulated result shows that combining clipping technique with order statistics based median filter can effectively reduce the impulse noise effect for time varying channel.

Keywords: *Adaptive filter; AWGN Channel; Impulse noise; MSE; LMS*

Introduction: Wireless communication plays a vital role in present world. Present era has enormous needs of context-based and service-oriented communication systems. Different types of input signals are used in service-oriented communication systems. Noise may be incorporated during signal transmission through channel and can corrupt the signal. Hence communication signal processing addresses different current issues to be solved [1]. Adaptive filter has the ability to adjust with the environment that makes it suitable for different applications [1]. They have important property that modifies the values of their parameters during the processing of the input signal, in order to generate output signal. The goal of the adaptation is to adjust the goal of the adaptation is to adjust the characteristics of the filter through an interaction with the environment inorderto reach the desired values.The operation of adaptive filters is based on the estimation of the statistical characteristics of the input signal. The input signal is corrupted with various environmental signals that are undesired at output. Adaptive filter removes the undesired response at the filter output to compare with the desired response. However sometimes, noise is environmental signals that are undesired at output. Adaptive filter removes the undesired

Article history:
Received 19 April 2017
Received in revised form 20 May 2017
Accepted 4 June 2017
Available online 30 June 2017

Corresponding author details:
E-mail: sheba.ict@gmail.com
Tel: +88 0721 751473

response at the filter output to compare with the desired response. However sometimes, noise is impulsive in nature [2] which is sporadic, non-contiguous, short duration irregular pulses and relatively high amplitude. It may generate from various sources like switching, lighting strikes, power line load switching, different channel environments communication systems etc. In communication systems, impulse noise originates at some point in time and space. Impulse noise is added with the information and propagates through the channel that changes the original information, so increase the error at the receiver. Nowadays impulse noise is an active area of research and its mitigation is needed due to better system performance.

There are many adaptive algorithms for adaptive filter [1,3-5], commonly popular algorithm is Least Mean Square (LMS) algorithm. LMS algorithm is mostly used algorithm due to its less complexity. Clipping is a technique that limits a signal when it exceeds a threshold. For impulse noise reduction, clipping technique is implemented which removes the impulse above a threshold [2]. However only clipping itself is not enough for different types of input signals. The median filtering is a technique that often reduces noise from signals. Median filter is widely used due to its effective noise reduction [6-8]. Clipping is a course technique; on the other hand median filter provides good result for preserving the fine details. However none of them addresses the time varying effect of a channel. For reliable communication system, convergence is a crucial issue for real time communication. The system learns quickly when it achieves faster convergence. Context-oriented and service based communication system need the minimum error during recovery of the signal. Hence we are motivated to introduce a robust adaptive algorithm which is targeted to provide faster convergence, optimum error and can mitigate the impulse noise effectively compared to the existing conventional methods.

In this paper, by investigating the effect of impulse noise on different input signals, we concentrated on mitigating the impulse noise effect from the received signal for time varying channel. We compared conventional LMS adaptive algorithm and only clipping techniques. We investigated the combined use of the simplicity of clipping technique and effectiveness of noise reduction of median filter. LMS adaptive algorithm is computationally competent to RLS algorithm because RLS needs to have around 0.9 forgetting factor for good MSE level which mean the last iteration uses almost all the previous iteration effect.. At the same time, selecting threshold and clipping also has low computational complexity. On the other hand, median filtering increases the computational complexity. However the proper choice of filter length keeps it reasonable as very good performance can be achieved with this tittle increase in computational complexity. Simulation result shows that order statistics based median filter with clipping technique outperforms among the compared methods for time varying channel.

The proposed technique is applicable in many real environments specially for communicating discrete messages reliably on a real-world channel. This may involve wireless communications and wired communication as transmission over twisted-pair telephone wires or shielded cable-

TV wires. This technique is also applicable for mitigating impulse noise present in images. The application area may cover from indoor radio to deep-space radio.

Following the introduction above, the remainder of this paper is organized as follows. Next sections describe the basic operation of adaptive filter and the adaptive algorithm section briefly describes the LMS algorithm. Next two sections present simulation model for noise reduction and simulation results respectively. Finally, we conclude the paper with significant noise reduction in conclusion section.

Adaptive Filter: An adaptive filter is a time variant filter whose coefficients are adjusted in a way to optimize a cost function or to satisfy some predetermined optimization criterion. The brief summaries of an adaptive algorithm are follows:

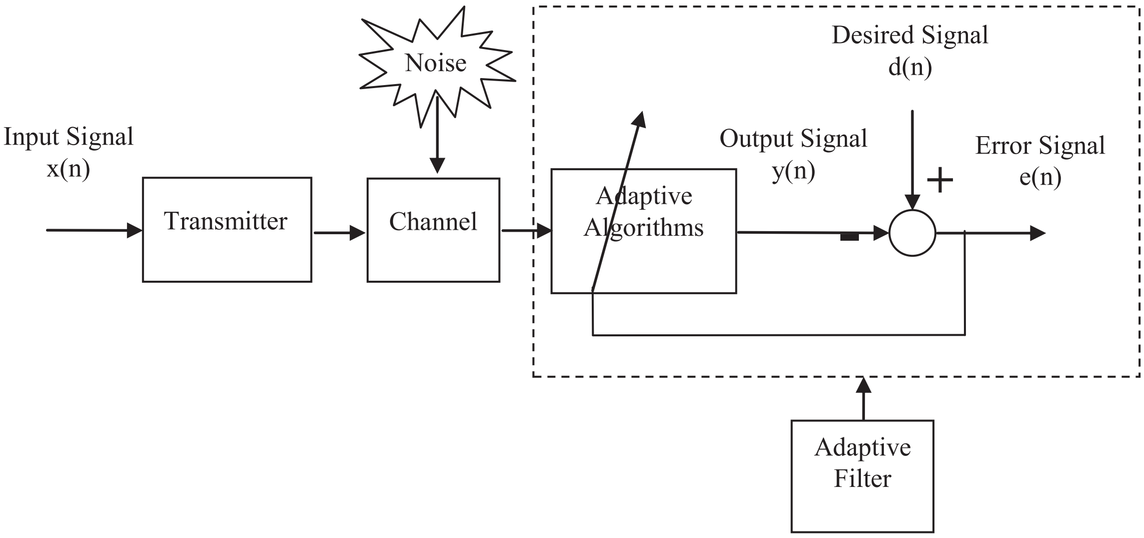

Fig.1: Schematic diagram of adaptive filter

Schematic Diagram: Figure 1 shows the basic diagram of an adaptive filter. The different input signals are transmitted that is corrupted with Gaussian*g(n)* having mean μ =0, variance σ^2=1at channel and its probability density function is shown in figure 2. The corrupted signal is received by the receiver which is the input of adaptive filter. The output signal *y(n)* of adaptive filter is compared with the desired response *d(n)* that generates the error signal *e(n)*. The error is used to modify the adjustable filter coefficients, in order to minimize the error. The filter output signal y(n) and error signal e(n) is obtained by the following equations:

$$y(n) = w^T(n)x(n) + g(n) \quad \text{eq. 1}$$

$$e(n) = d(n) - y(n) \quad \text{eq. 2}$$

where w(n) is the filter weight vector in equation (1) and error signal e(n) is obtained by using equation (2). To analysis the effect, impulse noise is also incorporated with the output signal having arbitrary value at a random iteration.

Performance Parameters: An adaptive filter has several parameters that measure the performance of the filter and it is an important factor because required fast filter adaptation with the environment as well as stable behavior with the systems.

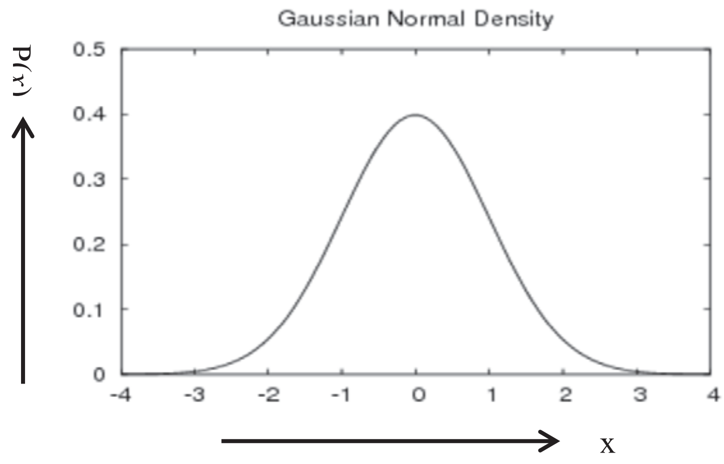

Fig.2: Probability density function for Gaussian noise

Some performance parameters for the adaptive filter are given below:

Convergence: Convergence corresponds to the coefficients of the adaptive filter that converse to the desired values. Where the convergence rate determines the rate at which the filter converges to its resultant state. Usually a faster convergence rate is a desired characteristic of an adaptive system.

Minimum Mean Square Error: The minimum mean square error (MSE) is a metric indicating how well a system can adapt to a given solution. A small minimum MSE is an indication that the adaptive system has accurately modeled, adapted and converged to a solution for the system. We need to minimize $E\{e^2(n)\}$, where E denote the expectation value of error signal.

Adaptive Algorithm: The Least Mean Square (LMS) algorithm is widely used adaptive algorithm. It is an adaptive algorithm that uses gradient based steepest decent method. LMS is an iterative procedure that makes successive corrections to the filter weight vector in the direction of the negative of the gradient vector that goes to minimum MSE. LMS is based on transversal filter which is responsible for performing the filter process. A weight control mechanism is used for adaptation of the filter. The algorithm depends on the statistics of input and desired signal.

The adaptation of LMS algorithm that updated the filter taps weights by using the following equation:

Adaption step: $w(n+1) = w(n) + \mu e(n)x(n)$ eq.3

Where, μ is known as the step size and w is the filter tap weight vector.

However, the step size is the important parameter for the adaptation equation that regulates the convergence. Convergence and MSE is an important criterion for the algorithm that depends on the step size. So choosing the value of step size is an important issue. A large step size may lead to fast convergence but big misadjustment, on the other hand, small step size may provide small misadjustment but slow convergence [5]. So expectation criterion is that fast convergence as well as small misadjustment.

Simulation Model: This paper shows a technique for effective impulsive noise reduction as well as error minimization based on clipping method and median filter with adaptive algorithm. Figure 3 shows the simulation model.

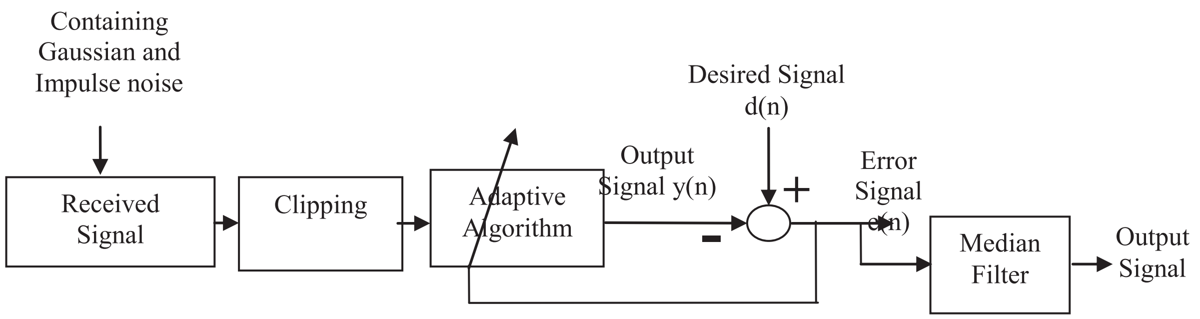

Fig.3: Block diagram for Impulse noise and MSE reduction

The received signal contains Gaussian noise and impulse noise that clipped by using clipping method. Here we defined a clipping threshold to remove impulse noise above the threshold value. For clipping, threshold value choosing is an important factor. The clipping threshold (T_{cl}) is set comparing with the maximum value of the received signal. The clipping method only removes the amplitude of the impulse noise above the threshold while all other characteristics are remaining unchanged [2],[6]. After performing the clipping method, adaptive filter is used to recover the original data from noisy data. Adaptive filter is used to remove the noise from the data that still noise below the threshold remains, because clipping method only removes the noise peaks above the threshold. After performing adaptation by using adaptive algorithm the error signal may also contain some error as a result we obtained slow convergence and large MSE. So, we integrate order statistics based median filtering technique to reduce noise and MSE. The error signal e(n) obtained from adaptive filter is used as input to the median filter. Then performing the median filtering operation and obtain the output signal which is more likely to reduce the impulse noise effect. Our approach consists of different individual techniques that are cascading to give better result than each individual method. In this approach, combined effect of clipping and median filter based adaptation has been investigated.

Algorithm of the proposed technique is given below

----

1. *Iterarion starts*
2. *Take input signal x(n)*
3. *Pass it through the Gaussian channel*
4. *Introduce impulse noise at any iteration*
5. *Get the output signal y(n) from the channel*
6. *Select optimum threshold*
7. *Do clipping*
8. *Adapt filter coefficient with LMS algorithm*
9. *Calculate error signal e(n)*
10. *Calculate median from the error vector*
11. *Replace the error value for impulse with median*
12. *Repeat for next iteration*

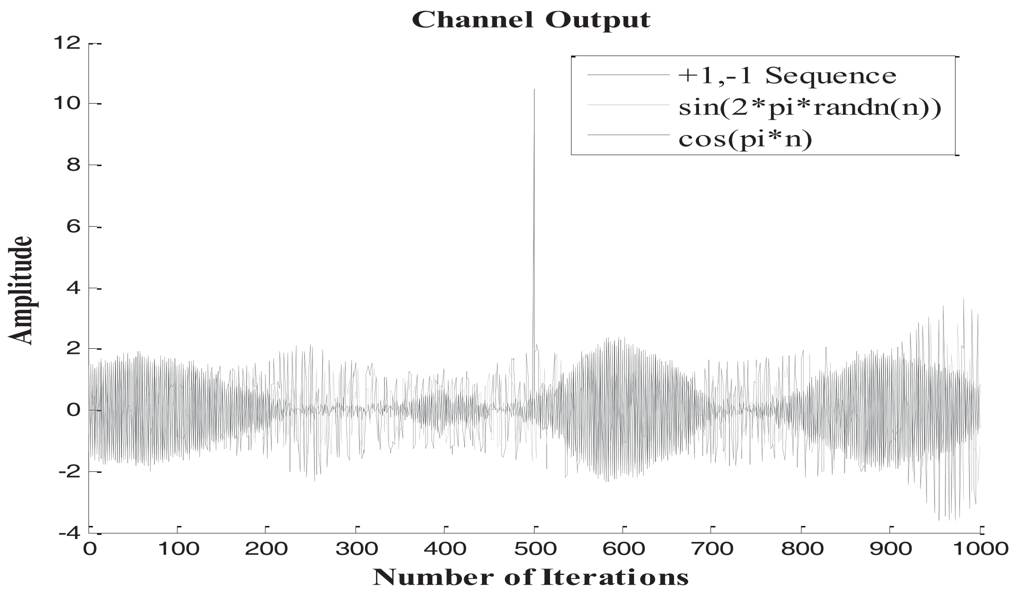

Fig. 4: Channel output for different input signals with impulse noise

Simulation Results: In our simulation results, we have tried to show the effects of impulse noise on different types of input signals and effectively mitigate the impulse noise effect. Computer simulations have been done using Mat Lab. The simulation is done for 1000 iterations and averaged for 100 trials to get the stable output. The filter length was fixed and it was set to be filter_ length L=5 throughout the simulations of this research. The step size for the LMS algorithm is set to be μ = 0.05 as this value provides optimum result for both the convergence and mean square error performance. For larger step size, the convergence will be faster. However it degrades the MSE level. Therefore we chose the step size to such a value that can care for both the convergence and MSE level at a time.

In figure 4, we investigated the behavior of three different input signals corrupted with impulse noise. The input signals are (i) *+1,-1* sequences, (ii) *sin (2*pi*randn(n))* and (iii) *cos(pi*n)*: where *n* denotes the number of iteration and *n*=1000. All the simulations are run 100 times and then averaged to get steady results. Figure 3 shows the output signal or received signal passing through the channel. All the output signals contain impulse noise and the signal is deviated thereby. It is observed from the figure that the deviation pattern is different for different input signals.

Figure 5 represents the effectiveness of LMS algorithm for different input signals with impulse noise. We observed the MSE variation and convergence after arrival of impulse noise. The signal $\cos(\pi n)$ shows the first convergence and lower MSE than other signals. The step size is chosen to be μ = 0.05 and all other parameters are same for all input signals. In this figure, convergence obtained around 10-30 iterations for the input$\cos(\pi n)$, where for other signals it may be required 60-80 iterations after the appearance of impulse noise. From the figure it is observed that only LMS algorithm cannot suppress the impulse noise effect from the output signal. A performance comparison for different techniques of impulse noise effect mitigation has been shown in figure 6. Three different methods have been compared; (i) conventional LMS, (ii) conventional clipping technique with LMS algorithm and (iii) our combined clipping-LMS-Median filter approach. Total 900 iterations are performed and the impulse noise is introduced in around 500 iteration point which is kept same for all techniques. The input sequences are chosen to be +1,-1, the channel to be time varying and the adaptive algorithm had step size μ = 0.05.

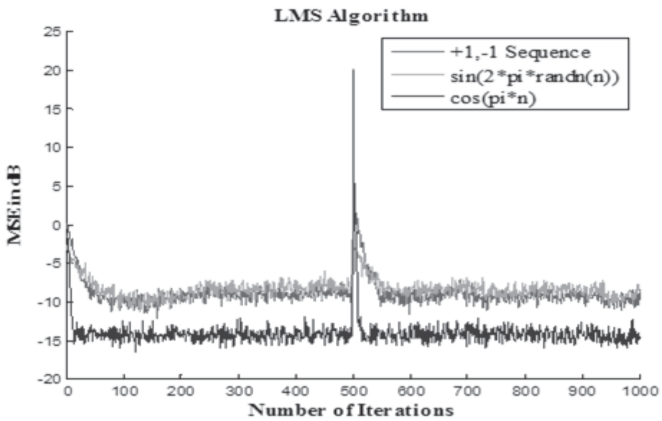

Fig.5: Performance comparison of LMS algorithm for different input signals

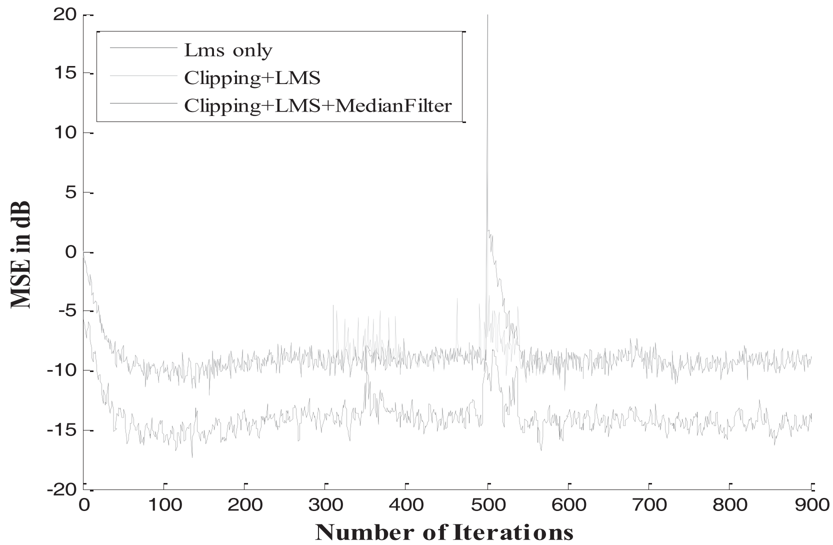

Fig.6: LMS based method with clipping and median filter

From Figure. 6, it is observed that conventional LMS algorithm shows the poor performance. It gives slow convergence and high impulse effect remains after filtering. When we added clipping technique, it gives reduces the impulse effect and converges faster than conventional LMS only. However the MSE level is not reduced. Figure 5 shows that almost all the effects of impulse noise has been mitigated by our combined clipping-LMS-Median filter techniques. It is also observed that this combined method provides around 5 dB MSE level improvement which is a very nice figure for time varying channel.

Conclusion: In this paper, we observed the effects of impulse noise for different input signals. Simulation results show that only the conventional LMS algorithm is not suitable for impulse noise mitigation. We present a combine method for impulse noise and MSE reduction. Though the conventional clipping method reduces the impulse noise to some extent, however it cannot reduce the overall MSE level. Median filter itself can suppress the impulse noise effect with fast convergence. Furthermore, we implement median filter that gives the better results with combining both clipping and adaptive algorithm. Computer simulation results showed that both the convergence and MSE is in better condition for this approach than conventional methods individually. Hence it may be a good alternative for the applications in impulse noise environment for time varying channel.

References:

[1] Simon Haykin, “Adaptive Filter Theory”, Prentice Hall, 1986.

[2] Sumrin M. Kabir, Alina Mirza, and Shahzad A. Sheikh, “Impulsive Noise Reduction Method Based on Clipping and Adaptive Filters in AWGN Channel”, International Journal of Future Computer and Communication, Vol. 4, No. 5, October 2015.

[3] Saeed Gazor, “Prediction in LMS-Type Adaptive Algorithms for Smoothly Time Varying Environments”, IEEE Transactions on Signal Processing, Vol. 47, No. 6, June 1999.

[4] Kumudini Sahu, Rahul Sinha, “Simulation of NLMS Adaptive Filter for Noise Cancellation”, International Journal of Engineering and Applied Sciences (IJEAS) ISSN: 2394-3661, Volume-2, Issue-1, January 2015.

[5] Zhao Shengkui, Man Zhihong, KhooSuiyang, “Modified LMS and NLMS Algorithms with a New Variable Step Size”, 9th IEEE International Conference on Control, Automation, Robotics and Vision (ICARCV'06), pp. 1-6, 2006.

[6] S. Al-Mawali and Z. H Hussain,“Adaptive-threshold clipping for impulsive noise reduction in OFDM-based power line Communications,” in Proc. International Conference on Advanced Technologies for Communications, Oct 12-14, 2009, pp. 43-48.

[7] Monika Kohli and Harmeet Kaur, “Noise Removal in Image Processing using Median, Adaptive Median and Proposed Median Filter and Respective Image Quality Comparison”, International Journal of Engineering Technology, Management and Applied Sciences, November 2015, Volume 3, Special Issue, ISSN 2349-4476

[8] N.Sakthivel andL. Prabhu, “Mean – Median Filtering For Impulsive Noise Removal”, International Journal of Basic and Applied Science, Vol. 02, No. 04, pp. 47-57, 2014.