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BER Analysis ofImpulse Noise inOFDM System Using LMS,NLMS&RLS

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Abstract*:Orthogonal Frequency Division Multiplexing (OFDM) scheme is also used as a digital multi carrier modulation method .Alarge number of closely spaced orthogonal sub-carrier signals are used to carry data on several parallel data streams or channels. While OFDM is ideally suited to deal with frequency selective channels and AWGN, its performance may be dramatically impacted by the presence of impulse noise. In fact, very strong noise impulses in the time domain might result in the erasure of whole OFDM blocks of symbols at the receiver. BER analysisis done by using various adaptive algorithms and Impulse noise impact can be observed.*

I. Introduction

Orthogonal Frequency Division Multiplexing (OFDM) is a modulation scheme widely used in wired systems (DSL, PLC) and wireless systems (WiMAX, 3GPP LTE). Various issues are linear distortion due to inter symbol interference(ISI),Inter-carrier Interference(ICI) additive white Gaussian noise (AWGN) and impulse noise. OFDM is ideally suited to handle ISI by frequency domain transmission and equalization using the Inverse Discrete Fourier Transform (IDFT)/ Discrete Fourier Transform (DFT) cyclic prefix and the effect of AWGN is eliminated using an appropriate level of coded modulation, impulse remains a limiting factor.

II. OFDM System

In the OFDM system coding techniquessuch as forward error control/coding (FEC) coding and interleaving is added to the OFDM system needed to protect against burst errors(robustness). An OFDM system is called codded OFDM(COFDM) system when both channel coding and interleaving are done.

In a digital domain binary input data are collected and FEC coded with schemes such as convolution codes. To obtain diversity gain bit stream are interleaved. Afterwards, 16-QAM a group of channel codded bits are mapped to corresponding constellation points. Then at this point the data are represented as complex numbers and they are serial. Known pilot symbol are mapped with known mapping scheme can be inserted at this moment. A serial to parallel converter is applied and the IFFT operation is performed on the parallel complex data. The transformed data are grouped together again, as per the number of required transmission subcarriers. Cyclic prefix is inserted in every block of OFDM data according to the system specification and the data are multiplexed in a serial fashion. At this point of time the data are OFDM modulated and ready to be transmitted. A time-domain digital data is transform to time-domain analog data using DAC. RF modulation is performed and the signal is up-converted to transmission frequency.

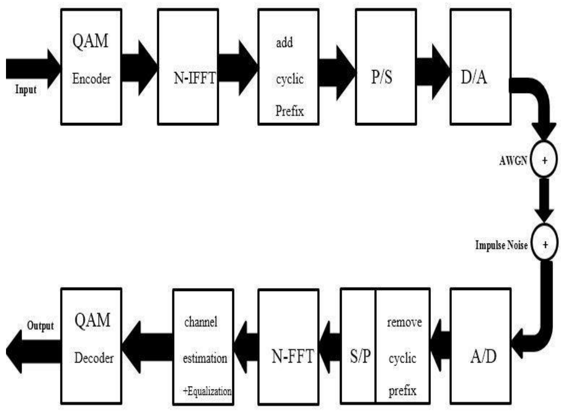

Fig1 Basic OFDM System with AWGN & Impulse Noises

After OFDM signal transmitted from the transmitter antenna, the signals go through all the anomaly and hostility of wireless channels. After receiving the signal, the receiver downconverts the signal and converts to digital domain using ADC convertor, symbol timing and synchronization are performed. After ADC conversion, symbol timing and synchronization is achieved. Then OFDM signal is demodulated using an FFT block. After that, channel estimation is performed using the demodulated pilots. Using the estimation, the complex received data are obtained which are demapped according to the transmission constellation diagram. At this moment the originally transmitted bit stream are recovered using FEC decoding and de-interleaving.

III. Impulse Noise

Impulse noise is generated by spurious sources such as switching of electrical AC sources.
Impulse noise can be broadly classified as

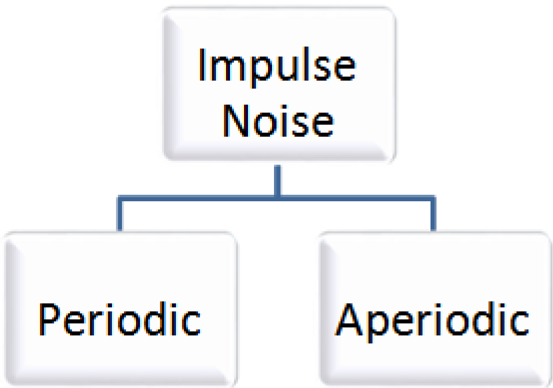

Fig2 Types of Impulse Noise

Periodic impulses consists of impulses of longer duration &occurring periodically in time.Aperiodic impulse noise is characterized by impulses occurring at random time. In this paper we concentrate on aperiodic impulse noise.

There are different statistical model available for modelling aperiodic impulse noise .The most commonly used models are Gaussian mixture model, Middleton's Class A model and symmetric alpha stable mode [1].

IV. Algorithms

1) Least Mean Square (LMS)

In the Lest Mean Square(LMS) algorithme(n) is minimized in the mean square sense with changes in the filter tap weights. The algorithm converges to a set of tap-weightswhen the processes $X(n)$&$d(n)$ are jointly stationary, which on average, are equal to the Wiener-Hopf solution. The LMS algorithm is a practical scheme for realizing Wiener filters, without explicitly solving the Wiener-Hopf equation

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

$$e(n) = d(n) - y(n)$$

The cost function $J(t)$ chosen for the steepest descent algorithm determines the coefficient solution obtained by using adaptive filter. If the MSE cost function is chosen, the resulting algorithm depends on statistics of $s(t)$ &$d(t)$ because of the expectation operation that defines this cost function is the least square cost function given by

$$J(t) = \sum_{i=0}^{t} ά(i)(d(i) - w'(t)s(i))^2$$

The weight update equation for LMS can be represented as

$$W(t+1) = W(t) + \mu e(t)S(t)$$

Where μ is learning factor, requires only multiplications and additions to implement. In effect, the iterative nature of the LMS coefficient updates is a form of time-averaging that smoothers the errors in the instantaneous gradient calculation to obtain a more reasonable estimate of the true gradient.

2) Normalized Least Mean Square (NLMS)

The NLMS algorithm is the extension of the standard LMS algorithm, the NLMS algorithms implementation is very similar to that of the LMS algorithm. Each iteration of the NLMS algorithm requires these steps to be followed

- The adaptive filter output is calculated by the equation

$$y(n) = \sum_{i=0}^{N-1} w(n)x(n-i)$$
$$= W'(n)X(n)$$

- An error signal is calculated as the difference between the desired signal and the filter output

$$e(n) = d(n) - y(n)$$

- The input vector step size value is calculated as

$$\mu(n) = \frac{1}{X'(n)X(n)}$$

- The filter tap weights are updated in preparation for the next iteration

$$W(n+1) = W(n) + \mu(n)e(n)X(n)$$

Each iteration of the NLMS algorithm requires 3N+1 multiplications, this is only N more than the standard LMS algorithm. This is an acceptable increase considering the gains in stability and echo attenuation achieve [3].

3) Recursive Least Square (RLS)

Recursive least squares algorithms have a faster convergences speed and do not exhibit the Eigen value spread problem. RLS algorithms calculate J(n) by using the following equation

$$J(n) = \frac{1}{N}\sum_{i=0}^{N-1} \lambda^i e^2(n-i)$$

Where N is the filter length and λ is the forgetting factor. This algorithm calculates not only the instantaneous value $e^2(n)$ but also the past values, such as $e^2(n–1)$, $e^2(n– 2)$, ..., $e^2(n–N+1)$. The value range of the forgetting factor is (0, 1]. When the forgetting factor is less than 1, this factor specifies that this algorithm places a larger weight on the current value and a smaller weight on the past values. The resulting $E[e^2(n)]$ of the RLS algorithms is more accurate than that of the LMS algorithms.

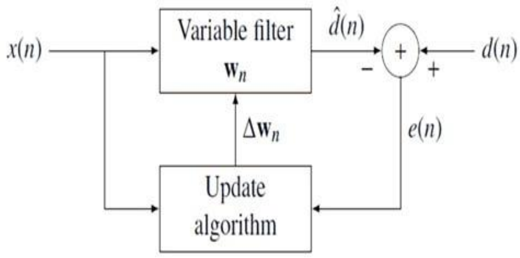

Fig3 RLS Block Diagram

- Calculate the adaptive filter output signal y(n).
- The error signal e(n) is calculated by using the following equation

$$e(n) = d(n)–y(n).$$

- The filter coefficients are updated by using the following equation

$$\bar{w}(n+1) = \bar{w}(n) + e(n).\bar{k}(n)$$

Where $\bar{w}(n)$ the filter coefficient is vector and $\bar{k}(n)$ is the gain vector $\bar{k}(n)$ is defined by the following equation

$$\bar{k}(n) = \frac{p(n).u(n)}{\lambda + \bar{u}^T(n).p(n).\overline{\overline{u(n)}}}$$

Where λ is the forgetting factor and P(n) is the inverse correlation matrix of the input signal.

V. Simulation Results

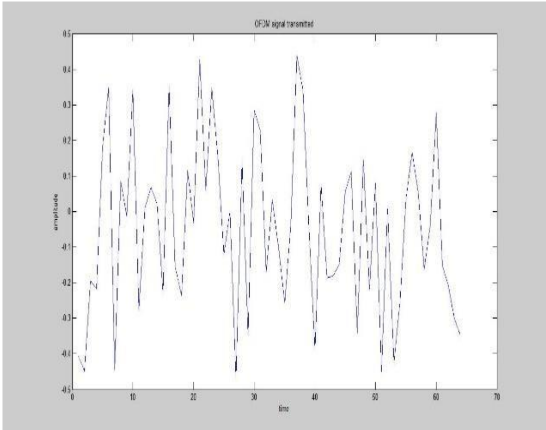

Fig4 Transmitted OFDM Signal

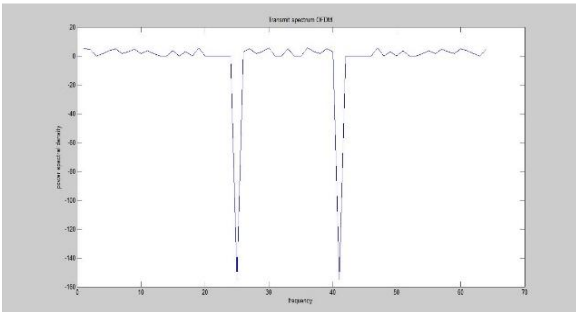

Fig5OFDM Transmitted Spectrum

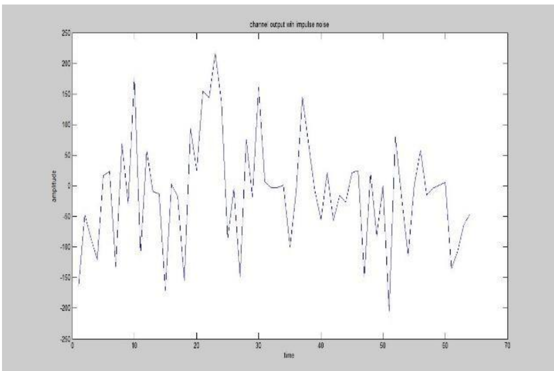

Fig6 Channel Output with Impulse Noise

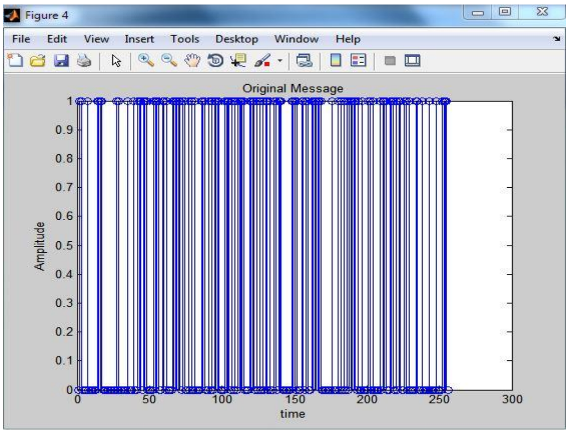

Fig7 Original Message Signal

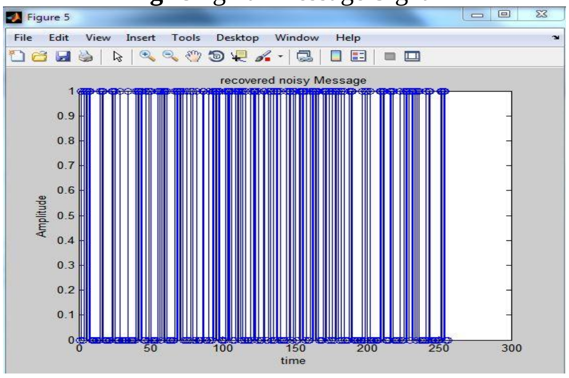

Fig8 Recovered Noisy Signal Using RLS

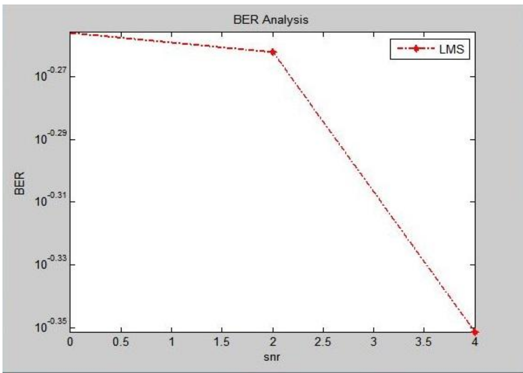

Fig9BER Analysis using LMS Algorithm

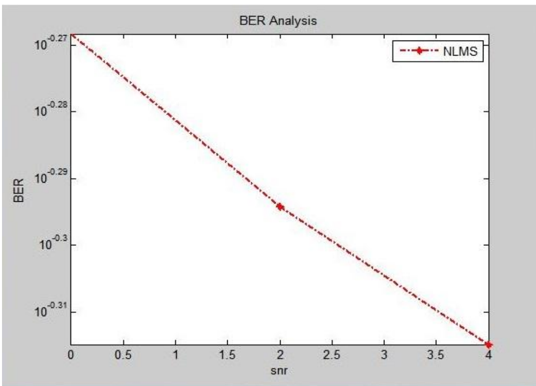

Fig10BER Analysis Using NLMS Algorithm

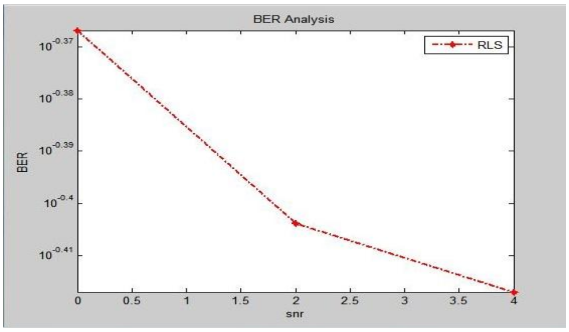

Fig11BER Analysis Using RLS Algorithm

VI. Simulation Description

The graphs shown above contains BER plots for the LMS,NLMS and RLS algorithms. The graphs are plotted for the various SNR values. By varying the SNR values we can observe the variation in BER values as shown in above plots. As we take the SNR between 0 and 4, the BER of individual algorithms increased over a period of internal and then decreased. Of those algorithms we executed RLS showing minimum BER when compared with the LMS and NLMS algorithms, and after RLS algorithm, NLMS algorithm showing better performance than LMS algorithm.

VII. Conclusion

In this research the above algorithms, LMS algorithm is the most popular adaptive algorithm, because of the low computational complexity. Even though, the LMS algorithm suffers from slow and data dependent convergence behaviour. Where NLMS algorithm, an equally simple, but more robust variant of the LMS algorithm, exhibits a better balance between simplicity and performance than the LMS algorithm. Due to its good characteristics the NLMS has been largely used in real-time applications but RLS has more sustainability to higher SNR compared to LMS & NLMS, RLS is better among the algorithms used.

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