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REVIEW ARTICLE

ADAPTIVE BLIND CHANNEL EQUALIZATION IN MULTIPATH ENVIRONMENT

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ABSTRACT

In this paper, we develop the Multi-Target Least Square Constant Modulus Algorithm as adaptive blind channel equalizer for transmission of sinusoidal signal through a communication channel. The channel coefficients are considered random, countenancing for both constructive and destructive interferences, leading to multipath effect. Simulation result shows that considerable improvement is achievable in the reconstruction of the received signal in terms of signal to noise power ratio. It is observed that we get better performance in terms of error rate as compared to conventional methods. The speed of equalization in terms of convergence rate is simulated and compared with least mean square equalizer and better performance is apprehended.

KEYWORDS

Adaptive algorithm, blind channel equalizer, Constant Modulus Algorithm (CMA), sinusoidal signal.

1. INTRODUCTION

A method of elimination of multipath random channel interference in presence of Gaussian noise is proposed in this paper, by using a modified Constant Modulus Algorithm (CMA), forming a blind channel equalization process. A multipath channel is considered with random coefficients, which degrades the input sinusoidal signals. We use an improved least square CM algorithm as adaptive equalizer at the receiver end to reconstruct back the original transmitted signal. MATLAB based simulation results show that in terms of signal to noise (SNR), significant enhancement of performance is possible in the reconstruction of the received signal. It is also observed that we get better performance in terms of error rate and speed of convergence as compared with the existing Least Mean Square (LMS) equalizer.

1. In the last few decades, many adaptive equalization techniques and algorithms for channel estimation have been proposed to overcome the effect of interference and to minimize the channel distortions (Treichler and Agee, 1983, Agee, 1986, Ganguly et al., 2019, Liu et al., 2016). The training signal based non-blind algorithms are being introduced in (Treichler and Agee, 1983, Agee, 1986).

Based on the channel sparsity, A study proposed channel estimators that increase the convergence rate of the algorithms (Bakhshi and Shah Talebi, 2018). The performance of these approaches is better than the minimum mean square error (MMSE) and conventional least mean square (LMS) techniques. Constant modulus algorithm (CMA) [2, 6-7] is a standard blind equalization algorithm because of its stability, efficiency and simplicity (Agee, 1986, Sharma and Mathur, 2016, Saxena and Kothari, 2014). Earlier, recent study introduced a technique for an adaptive digital

filtering that compensates modulated signals that comprises both interference and frequency-selective multipath, on a constant envelope (Treichler and Agee, 1983). This method realizes the point that multipath reception and several interference sources generate related amplitude modulation on the received signal. CM algorithm has also been used in different blind signal processing fields, such as blind adaptive beamforming (Gross, 2015). Least Square CM algorithm (LSCMA) that was proposed by Agee uses the least square method and minimizes the cost function over a single block of data vectors, which are updated on a block-by-block basis. Thus, LSCMA obtain faster convergence rate and global stability as compared to conventional CMA (Agee, 1986, Ganguly et al., 2019).

In this paper we make use of the Multi-Target LSCMA (MT-LSCMA) based method as blind adaptive equalization in multipath channel interference model with sinusoidal waves. MT-LSCMA exploits the constant amplitude property of the envelope of the received sinusoidal signals and generates two sets of weights. It optimizes and selects the weights in such a way that suppresses the multipath generating coefficients, thus reconstructing back the original waveform.

The paper is prepared as follows, section (2) introduces a mathematical model of random coefficient multipath channel, in section (3) we develop MT-LSCMA algorithm as blind equalization for random multipath channel. Simulation results providing comparisons of the input-output sinusoidal waveforms in terms of SNR (dB) and variation of error rate with SNR (dB) are presented in section (4). Also in this section, we provide the comparison between the convergence rate of MT-LSCMA algorithm and least mean square (LMS) algorithm. Section (5) completes this paper with conclusion and discussion.

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2. MATHEMATICAL MODEL OF MULTIPATH RANDOM CHANNEL

Figure 1: A multipath channel model

One of the major distortions that occur due to multipath phenomenon is the Inter Symbol Interference (ISI). In addition, due to randomly varying characteristics of the multipath channel, the associated coefficients of the channel also become time varying. This leads to misinterpreting of the received signal.

Figure 1 depicts a model of a multipath channel, with additive white Gaussian noise (AWGN) vector, $n(k)$. Let us consider that there are L distinct paths, including the direct line-of-sight (LOS) path. Other than this LOS, all other paths introduce a finite time delay τ_i , (where $1 \leq i \leq L-1$) to the signal transmitted along that path. From Figure 1), the output of the channel, $y(k)$ can be written as:

$$y(k) = x(k)h^T + n(k) \quad (1)$$

where, $x(k) = [x(k) \ x(k - \tau_1) \ x(k - \tau_2) \ \dots \ x(k - \tau_{L-1})]$ is the input signal vector.

$h = [h_0 \ h_1 \ h_2 \ \dots \ h_{L-1}]$ is the associated channel coefficient vector of the multipath.

$n(k)$ is the AWGN vector of zero mean and unity variance.

h^T indicates transpose of the vector h , while the symbol k is used for time unit.

Expanding equation (1), we get:

$$y(k) = h_0 x(k) + h_1 x(k - \tau_1) + \dots + h_{L-1} x(k - \tau_{L-1}) + n(k) \quad (2)$$

In the above equation, the first term of the RHS represents the direct LOS path. In the other RHS terms, the expressions $x(k - \tau_i)$ denote the time delay associated with the input signal at each multipath. $n(k)$ is the AWGN vector. Thus, the output of the channel or the receiver input comprises of the direct and various delayed components of the input signal, together with the noise.

3. MT-LSCMA AS BLIND ADAPTIVE EQUALIZATION

Non-blind adaptation requires a training signal to adjust the weights of the Wiener filter or the equalizer. In hostile environment, severe fading disrupts the continuous communication, thus leading to interruption of the training sequence. In these cases, blind adaptation provides a far better alternative as it does not require any training or desired signal.

It is a significant substitute to non-blind techniques because the channel is equalized and communication is re-established without any information of the transmission source. In comparison to the non-blind adaptive channel equalization technique, channel equalization of blind algorithm does not need any training or desired sequence. Instead, the statistical properties of the transmitted signals are exploited.

The CM array receives $x(k)$ and form blocks of data vector as $X = [x(1) \ \dots \ x(p)]$ where p is the number of data samples in each block. As an alternative of directly estimating the correlation matrix R_x , it estimates $S_i(k)$ of $S_m(k)$ and source direction vector a_i in a block iterative process

and hence the angle of arrival θ_i . From fig.1, the output of the weight-and-sum beamformer is given by:

$$y(k) = w^H(k)X(k) \quad (3)$$

where $w(k) = [w_1(k) \ \dots \ w_N(k)]^T$ are the adaptive weights accustomed by MT-LSCM algorithm and $(\cdot)^H$ is the Hermitian transpose. To determine the adaptive weights, the dynamic nonlinear least-square CM optimizes

$$\min_{w^*, \bar{S}} \left\| \bar{w}^* \bar{X} - \bar{S} \right\| \quad \text{s.t.} \quad |S_i(t)| = 1 \quad t = 1, \dots, N$$

where $\|\cdot\|$ is the vector 2-norm. The minimization is achieved by projecting $w^* X$ into the set of constant modulus signals through dividing entry-wise by modulus by means of the projection operator $r(S) = \begin{bmatrix} \frac{S(1)}{|S(1)|} & \dots & \frac{S(n)}{|S(n)|} \end{bmatrix}$, where n is the number of iterations. Finally, we can optimize over w and find the least square solution as $w^* = SX^\dagger$, where $\dagger$ denotes the Moore-Penrose pseudo inverse. Given an initial point $w^{(0)}$, the value of the weights updated by block iteration is given by:

$$\begin{aligned} S^{(n+1)} &= r(S)(w^{(n)*} X) \\ w^{(n+1)} &= S^{(n+1)} X^\dagger \end{aligned} \quad (4)$$

Equation (3) gives the weight updating calculation of MT-LSCMA. In a changing environment, the data blocks are updated for each iteration, resulting in dynamic MT-LSCMA. The iterations continue until appropriate convergence is satisfied. Characteristically, this is somewhat similar to the Sample Matrix Inversion (SMI) algorithm [7-8].

Figure 2 shows the flow chart of MT-LSCMA for adaptive equalization

Figure 2: Channel equalization with MT-LSCMA

Figure 3 below shows the complete multipath with adaptive equalization in a communication channel.

Figure 3: A complete multipath communication system with MT-LSCMA adaptive equalization

4. SIMULATION RESULTS

A MATLAB based testbed is developed to assess and compare the performances of MT-LSCMA adaptive equalization. Table-1 below depicts the various parameters used for simulation purpose.

Low frequency sinusoids are considered as transmitting signals to channel with random coefficients, leading to multipath effect. The step size of CM algorithm is assumed as 0.01. The channel coefficients are generated randomly to get the multipath effect. The length of the weights are considered as 12.

The table below illustrates the simulation parameters used.

Table 1: Simulation parameter for MT-LSCM algorithm			
Parameter	Simulation data	Parameter	Simulation data
Length of weight	12	Step size	0.01
Channel coefficient	Random value	No. of iterations	200
Transmitted signal	$\sin(2\pi fn)$	SNR of transmitted signal	-10 dB to 10 dB
Transmitted signal frequency (f)	100 Hz	Sampling frequency	500 Hz

Figure 4: Transmitted, received and equalized signal at 10 dB

Figure 5: Transmitted, received and equalized signal at 5 dB

Figure 6: Transmitted, received and equalized signal at -5 dB

Figure 7: Transmitted, received and equalized signal at -10 dB

Figure 8: Error plot with the variation of SNR values

Figures 4, 5, 6 and 7 shows the simulation results of with and without equalization at various values of SNR levels. Clearly, at higher SNR values, the sinusoids are almost accurately reconstructed back at the receiver. At lower SNR too, MT-LSCMA based equalizer partially recover the transmitted signal. This is clearer if we look at the error plot of Figure 8 above.

Figure 9 depicts the convergence plot with the variation of SNR values. It is observed that at higher values of SNR, the converging rates are higher.

Figure 9: Convergence plot with the variation of SNR values with iteration number

5. CONCLUSION

In this paper, we propose Multi-Target LSCMA as a blind adaptive equalizer to mitigate the multipath effect in a communication channel. Simulation results show that at considerable SNR values, the received signal after equalization almost accurately reconstructs back the transmitted signal. At lower SNR levels too, the received signal can be reconstructed back partially. This is well observed in the variation of error with SNR plot. Finally, the convergence plot shows speed of the algorithm is quite high at higher values of SNR.

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