Int. J. Electron. Commun. (AEÜ) 83 (2018) 150–155

Contents lists available at ScienceDirect

International Journal of Electronics and Communications (AEÜ)

journal homepage: www.elsevier.com/locate/aeue

Regular paper

# A robust adaptive weighted constant modulus algorithm for blind equalization of wireless communications systems under impulsive noise environment

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ARTICLE INFO

Article history:
Received 17 April 2017
Accepted 21 August 2017

Keywords:
Impulsive noise
Inter symbol interference
Blind equalization
Robust adaptive weighted constant modulus algorithm
Wireless communication

ABSTRACT

A robust adaptive weighted constant modulus algorithm is proposed for blind equalization of wireless communication systems under impulsive noise environment. The influence of the impulsive noise is analyzed based on numerical analysis method. Then an adaptive weighted constant modulus algorithm is constructed to adaptively suppress impulsive noise. Theoretical analysis is provided to illustrate that the proposed algorithm has a robust equalization performance since the impulsive noise is adaptively suppressed. Moreover, the proposed algorithm has stable and quick convergence due to avoidance of large misadjuntment and adoption of large step size. Simulation results are presented to show the robust equalization performance and the fast convergence speed of the proposed algorithm under both impulsive noise and Gaussian noise environments.



## 1. INTRODUCTION

In a band-limited digital communication system, the equivalent discrete-time linear finite impulse response (FIR) channel causes inter symbol interference (ISI) [1–3]. During the past years, various equalization methods and channel estimation algorithms [4–17] have been developed to overcome the effect of ISI and compensate the channel distortions.

The training-based methods were presented in [4–8]. Biguesh and Gershman presented the novel scaled least square and relaxed minimum mean square error techniques which require less knowledge of the channel second-order statistics [4]. Based on the sparsity of the channel, Li et al. proposed some channel estimators that improve the convergence speed of the estimators [5–7]. Those approaches have better performance than the conventional lest mean square (LMS) and minimum mean square error (MMSE) channel estimators. Das, Pattnaik and Padhy applied artificial neural network (ANN) trained by particle swarm optimization (PSO) to the problem of channel equalization [8]. As compared with other ANN based equalizers as well as Neuro-fuzzy equalizers, the Das' method optimizes not only the weights but also transfer function and topology of the ANN. Moreover, the algorithm has a better equalization performance than the other ANN based equalizers at the expense of higher computational complexity.

Blind equalization is a popular method to the reconstruction of transmitted symbols based on the noise-corrupted channel output without knowing the underlying FIR channel [9]. Constant modulus algorithm (CMA) [10] may be the most classical blind equalization algorithm due to its simplicity, stability and efficiency. It is worth mentioning that CMA has also been used in other blind signal processing fields, such as blind beamforming [18]. Li et al. formulated blind equalization of short burst signals by the twin support vector regression (TSVR) framework [11]. The iterative re-weighted least square (IRWLS) algorithm is used to achieve fast convergence. In addition, data-reusing method is utilized for small amounts of data samples to reach stable convergence. Fernandes provided a comparative evaluation of the performance of interior-point, active-set and simplex methods [12], which applied to the optimization process involving the blind equalizer.

To date most of the works on equalization, including blind equalization, assumes that the channel ambient noise is Gaussian, such as the work [8–12]. However, many physical channels are non-Gaussian, such as urban and indoor radio channels, underwater acoustic channels, man-made electromagnetic interference and a great deal of natural noise as well [19–21]. In an impulsive noise environment, most of blind learning algorithms suffer from large

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http://dx.doi.org/10.1016/j.aeue.2017.08.036

misadjustment or slow convergence [22]. It has been shown that significant performance improvements can be achieved if the statistical characteristics of the impulsive noise are considered during receiver design.

It is found that $\alpha-$ stable distribution can be considered as the greatest potential distribution to characterize various impulsive noises when different exponent parameter $\alpha$ is selected. An important characteristic of the $\alpha-$ stable distribution is that the moments of order higher than $\alpha$ tend to infinity. Under investigation, it is found that the norm constant modulus algorithm (NCMA) [13] is proposed to handle the impulsive noise by normalizing the samples. Li et al. proposed a robust constant modulus algorithm (RCMA) [14], which transforms the equalizer output error of the original constant modulus algorithm nonlinearly to suppress the $\alpha-$ stable noise. Li et al. proposed a novel CMA based on fractional lower order statistics (FLOS_CMA) [15] to avoid the stochastic infinity of the high order moment of $\alpha-$ stable noise. To decrease steady-state mean square error of the FLOS_CMA, Li proposed a concurrent equalizer, in which a decision-directed least mean $p$ norm (DD_LMP) equalizer operates cooperatively with a FLOS_CMA equalizer [16]. Wang and Chen developed a Bayesian equalizers based on the Gibbs sampler for both Gaussian ISI channel and impulsive ISI channel [17]. A salient feature of the proposed blind Bayesian equalizers is that they can incorporate the a priori symbol probabilities. To the best of our knowledge, these works are the state-of-the-art techniques about equalization applied to suppress impulse noise.

In this paper, we analyze the influence of impulsive noise for blind equalization and propose a robust adaptive weighted constant modulus algorithm (RAWCMA) for the wireless communication systems employing quadrature amplitude modulation (QAM) signal. The proposed method adaptively adjusts the weight coefficient according to the blind equalizer (BE) output error. Compared to the existing methods [4,8,11,12], RAWCMA can handle not only Gaussian noise but also impulsive noise. Moreover, the proposed algorithm has the following superiorities compared with other counterparts [13–17]. The proposed algorithm has not the problem of parameter selection and can efficiently handle the Gaussian and impulsive noise. Theoretical analysis shows that the RAWCMA has robust equalization performance, stable calculation performance and fast convergence speed. The RAWCMA is an adaptive weighting method in which the weights are large if there are small output errors, while the weights are small if there are large output errors. It is seen that channel distortion is mainly compensated through the samples with the small output errors. In addition, the RAWCMA avoids the lager misadjustments and can take the step size relatively larger than that of the constant modulus algorithm (CMA). Hence, the proposed algorithm has a significantly faster convergence speed than CMA. Simulation results demonstrate that RAWCMA has a good equalization performance, a stable computational property and a fast convergence speed under both impulsive noise environments and Gaussian noise environments.

## 2. System description

In the presence of additive impulsive noise, the resulting received signal for the $k$th symbol interval $x(k)$ can be written as

$$x(k) = \sum_{l=0}^{\bar{L}-1} h(l)s(k-l) + n(k). \tag{1}$$

Here, $h(k)$ with order $\bar{L}$ is the channel impulse response (CIR) that includes the responses of pulse shaping filter, propagation path and receiver and so on. $s(k)$ is the transmitted signal that takes the value from the QAM signal set. $n(k)$ is the additive impulsive noise and satisfies $\alpha-$ stable distribution which does not have a close form of probability density function (p.d.f.). It can be described by its characteristic function as

$$\psi(u) = \exp\left\{jau - \gamma|u|^{\alpha}[1 + j\beta \mathrm{sgn}(u)\phi(u,a)]\right\} \tag{2}$$

where $\phi(u,a) = tan\frac{\alpha\pi}{2}$, if $\alpha \neq 1$; $\phi(u,a) = \frac{2}{\pi}\ln|u|$, if $\alpha = 1$. Please see the Ref. [15] for the specific meanings of parameters $a, \gamma, \beta$ and $\alpha$. Set the order of BE to be $L$, then the resulting equalizer output $y(k)$ can be expressed as

$$y(k) = \sum_{l=0}^{L-1} w^*(l)x(k-l) = \mathbf{w}^H\mathbf{x}(k) \tag{3}$$

where $\mathbf{w} = [w(0), w(1), \ldots, w(L-1)]^T$ is the weight vector of the BE with order $L$ and $\mathbf{x}(k) = [x(k), x(k-1), \ldots, x(k-L+1)]^T$ is the received sample vector. Our objective is adjusting the weight vector $\mathbf{w}$ to make $\tilde{C}y(k) \approx a(k-\tau)$ , where $\tilde{C}$ is a constant, $\tau$ is the delay time.

## 3. The proposed adaptive weighted constant modulus algorithm

### 3.1. The adaptive weighted constant modulus algorithm

Constant modulus algorithm (CMA) is perhaps the best known and most popular scheme for blind channel equalization due to its simplicity and relatively stable convergence property [23]. The cost function of CMA [24] is given as

$$J(\mathbf{w}) = E\left[\left(|y(k)|^p - R\right)^2\right] \tag{4}$$

where $R = E\left[|s(k)|^{2p}\right]/E[|s(k)|^p]$ and $p$ is a positive integer. And the updating formula is

$$\mathbf{w}_{k+1} = \mathbf{w}_k - \mu\left(|y(k)|^p - R\right)|y(k)|^{p-2}y^*(k)\mathbf{x}(k) \tag{5}$$

where $\mu$ is the step size.

However, CMA is most likely to fail under the impulsive noise. Because impulsive noise can causes a serious misadjustment by the large output error $e(k) = \left(|y(k)|^p - R\right)|y(k)|^{p-2}y^*(k)$. Moreover, serious misadjustment is produced both in the Gauss noise and impulsive environment at the beginning of iteration. The step size $\mu$ must be significantly smaller than the other least mean square (LMS) method to ensure a stable operation due to this misadjustment [25]. Hence, the convergence speed of CMA is always very slow.

To overcome the above problems of CMA, we expect to design a scheme that the weight vector of the BE is hardly adjusted when the output error of the BE is large, and is caused by the large impulsive noise or misadjustment at the beginning of the iterations. On the other hand the weight vector of the BE is appropriately adjusted according to the samples corresponding to the small output error to ensure the equalization performance.

It is obvious that $e(k) = \left(|y(k)|^p - R\right)|y(k)|^{p-2}y^*(k)$ is a scalar. Hence, its absolute value does not determine the iteration direction. Inspired by this conclusion, we hope adaptively adjusting the BE according to the absolute value of $e(k)$ to realize robust blind equalization.

Based on the above analysis, we design the following blind equalization cost function

$$J(\mathbf{w}) = E\left[\ln((|y(k)|^p - R)^2 + C)\right] \tag{6}$$

where the parameter $p$ is set to be 1 and $C$ is a small constant trivially set to be 1.