A Blind Equalization Method For Multipath Rician Fading Channels

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Abstract— **A novel blind equalization method is proposed for multipath Rician fading channels. In the proposed method, based on blindly estimated characteristics of Rician fading channel, a realizable DFE is designed to minimize the mean square error (MSE) from the frequency domain perspective. DFE's filters are implemented in an adaptive structure to enable dealing with rapid time-varying channels. The method shows better convergence performance compared to similar existing methods. Furthermore, it works satisfactorily even for channels with severe dispersions.**

Key Words— **Blind Equalization, Adaptive Equalization, Decision Feedback Equalizer (DFE), Multipath Rician Fading Channel, Convergence Rate**

I. INTRODUCTION

The recovery of transmitted messages in the presence of noise and intersymbol interference (ISI) is an important research topic in digital communications. In low quality channels, ISI increases symbol error rate drastically. In meeting the increasing demand of high-speed communications, equalizer employed at the receiver plays an important role in overcoming ISI. Therefore, equalization of communication channels has been a problem attracting intensive research activities over past few decades. Many equalization methods have been proposed. The optimal approach to channel equalization is based on Maximum Likelihood Sequence Estimation (MLSE) which may be efficiently implemented using Viterbi algorithm provided the channel impulse response is available. If the channel response is long, MLSE faces significant implementation complexity. Probably the simplest equalizer is a linear Finite Impulse Response (FIR) filter, called a Linear Equalizer (LE). Its simplicity however makes it vulnerable in severe propagation conditions. On the other hand, Decision Feedback Equalizer (DFE) is proved to be very efficient in the case of supervised adaptation. It approaches the performance of MLSE without prohibitive computational complexity and its performance exceeds by far that of an LE [1].

In [2], a fixed robust DFE which gives an acceptable performance over a range of perturbations for channel parameters is proposed. Multipath fading channels with Rician (or Rayleigh) distribution have been used to characterize statistical properties of the time-varying channels for a long time [3-7]. These channels are encountered in many digital communication applications such as mobile radio communications [1]. In [6], based on [2], a realizable transversal filter is constructed to minimize the mean square error from the frequency domain perspective for Rician fading Channels. In [7], this strategy was expanded in order to design an optimal DFE in the frequency domain which is called DFE-FD here. In [7], a training sequence was used in order to estimate statistical characteristics of the fading channels and, consequently a fixed equalizer is designed. The first problem arises if the channel undergoes sudden drastic characteristic change (after the design process) such as variation of its order, in which equalizer certainly fails to track the channel because of its fixed form. On the other hand, transmitting a training sequence decreases the effective bit rate and is not always possible. Hence, adaptive and unsupervised (blind) methods are necessary.

The first class of the blind equalization methods utilizes a transversal filter and a MSE cost function [8-10]. In [8], Godard proposed a new criterion called Constant Modulus Algorithm (CMA). Due to its low computational complexity, CMA has been so popular in SISO and SIMO systems [11-12]. The CMA steady state MSE performance, however, may not be sufficiently high to guarantee satisfactory bit error rate performance because of its restricted transversal structure. Unfortunately, in a recursive equalizer such as a DFE, the error propagation phenomenon limits the use of unsupervised adaptation. In [13] equalizer is split into a cascade of several linear filters including at least a recursive one. In [14], based on [13], a blind Adaptive DFE (ADFE) is proposed in which both the structure and adaptation algorithm, according to some performance index such as the MSE, are modified to blindly deal with fast time-varying channels. The substantial performance of ADFE has been attracted by many researchers [15-17]. However, the receiver has no information for proper choices of the recursive and transversal filters lengths. Long lengths result in magnified noise, and short lengths may not compensate ISI. Moreover, in severe channels ADFE may not open eye and therefore, MSE can not be sufficiently low.

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The second class of the blind equalization methods is blind sequence estimation [18]. In this group, the approximate maximum likelihood estimates for the channel and data are obtained. In [18], Chen proposed a fast blind equalization based on Bayesian DFE for joint data and channel estimation. However, Chen's algorithm may not converge to the correct estimate without a suitable initialization requiring some pre-known channel characteristics. In [19], Parallel Bayesian DFE (PBDFE) was proposed to tackle this problem which has high accuracy and reliability. Moreover, its fast convergence to the real channel is considerable. In time-varying channels the equalizer converges to paths' means. Although in this class, the computational complexity is drastically increased if alphabet size of the modulation is high or channel impulse response length is long. In this study, channels with short lengths are considered. So, the latter is not noticeable here. On the other hand, PBDFE with the small alphabet size faces reasonable computational complexity with high accuracy.

In the present work, the channel is assumed to be multipath Rician fading. The channel parameters are blindly estimated initially, and two optimal filters are consequently designed in the frequency domain [7]. Based on these results, two FIR filters (feedforward and feedback) are constructed. Finally, they are implemented in an adaptive structure that can follow severe variations [14]. Furthermore, filters lengths can be properly selected. Simulations illustrate a fast convergence of the proposed method to the desired MSE leading to open eye in severe conditions where other methods fail.

The organization of this paper is as follows: In section 2, mathematical description is presented and some modeling aspects of the frequency-selective Rician fading channel are discussed. In section 3 the proposed method is introduced. To demonstrate the performance of the proposed design method, illustrative simulations are presented in section 4, and section 5 concludes the paper.

II. MATHEMATICAL DESCRIPTION

A digital Communication system is depicted in Fig.1. The channel, which is the combination of the transmitter filter, transmission medium, and the receiver filter, is modeled as a FIR filter. The backward shift operator is shown by q^{-1} i.e. $q^{-1}u(n) = u(n-1)$. The i.i.d., equiprobable zero mean input sequence $u(n)$ is transmitted every T_s seconds. The additive white circular complex-valued Gaussian noise with zero mean and variance of σ_w^2 is denoted by $w(n)$.

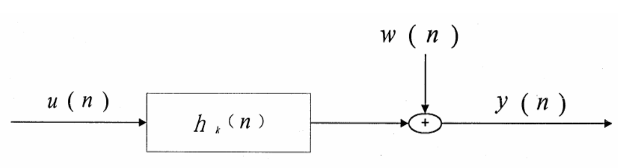


Figure 1. Communication system

The time-varying channel impulse response $h_k(n)$ is expressed in the following form [6-7]:

$$h_k(n) = h_k + \tilde{h}_k(n) \tag{1}$$

h_k is a constant mean and $\tilde{h}_k(n)$ satisfies wide-sense stationary and uncorrelated scattering (WSSUS) properties. For each k, $\tilde{h}_k(n)$ is a circular complex valued Gaussian sequence governed by [6]:

$$\tilde{h}_k(n) = \rho_k \tilde{h}_k(n-1) + v_k(n) \tag{2}$$

Where $v_k(n)$ is an i.i.d., circular complex-valued Gaussian sequence with zero mean and variance of $\sigma_{v_k}^2$ and ρ_k is arbitrarily chosen close to unity so that $\tilde{h}_k(n)$ represents a generic low-pass process. Finally the received discrete-time signal $y(n)$ is given by:

$$y(n) = \sum_{k=0}^{L} h_k(n)u(n-k) + w(n) \tag{3}$$

III. PROPOSED ALGORITHM

The design method is based on minimizing the following MSE regarded as the reconstruction performance index [7]:

$$\begin{aligned} J &= E_{h,u,w}\left[\left|e(n-l)\right|^2\right] \\ &= E_{h,u,w}\left[\left|u(n-l) - \hat{u}(n)\right|^2\right] \end{aligned} \tag{4}$$

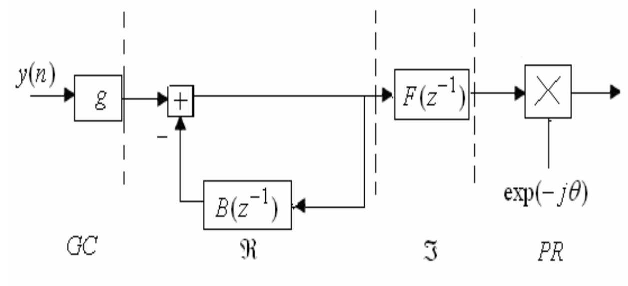
l is the number of smoothing lags, $\hat{u}(n)$ is a detected symbol in the output of receiver and $e(n)$ denotes the reconstruction error. Parseval's theorem, Calculus of variation and spectral factorization are deployed in the design procedure exactly like the method proposed in [7]. For designing the optimal filters, h_k, ρ_k, σ_w^2 and $\sigma_{v_k}^2$ must be known [7]. In [7] a training sequence is used to estimate h_k, ρ_k and $\sigma_{v_k}^2$. In this study, transmitting a training sequence is not desirable. So, In order to blindly estimate channel parameters, we use PBDFE [19] which blindly estimates h_k and data jointly. Based on the blindly reconstructed data, ρ_k and $\sigma_{v_k}^2$ are estimated via the method introduced in [6]. In other words, [7] uses a training sequence to estimate channel parameters, but we employ PBDFE, and utilize the blindly estimated data instead. In fact, the detected data can be considered as a training sequence with high accuracy provided PBDFE converges to the right channel. The right convergence is guaranteed in high Signal to Noise Ratio (SNR). However, due to the adaptive structure of the proposed method, optional values for ρ_k and $\sigma_{v_k}^2$ don't have adverse effects (ρ_k must be selected in a manner which $\tilde{h}_k(n)$ represents a generic low-pass process). As mentioned before, PBDFE suffers from high computational complexity when alphabet size of the modulation is large. Hence, In order to estimate channel parameters, firstly a binary sequence is utilized. Then larger alphabet size can be selected. In fact, firstly a binary PAM with a short period (because of the fast convergence of PBDFE) is transmitted, and the required parameters are estimated. Then, larger constellation can be selected. In the blind practical systems, channel is usually estimated by binary sequences. Hence, this strategy is justifiable. Based on the blindly estimated parameters, an optimal DFE is designed in the frequency domain exactly like the method proposed in [7]. DFE filters are stable and have IIR forms. The order of their transfer function denominator is 1 (because of the unity order of the AR model of eq.2). IIR forms are converted to FIR ones as follows [20] ($b(k)$ is pole location of the filter):

$$\frac{r(k)}{1-b(k)z^{-1}} = r(k)\sum_{i=0}^{Q}(b_k z^{-1})^i \tag{5}$$

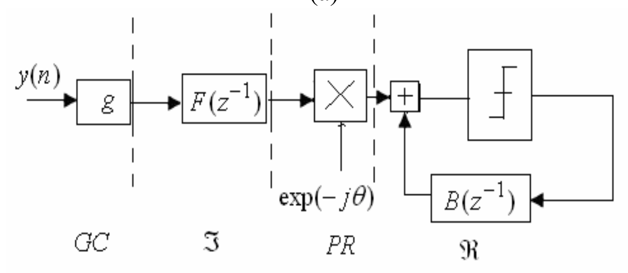
$$where, \quad Q = \left[\left| \log(\xi)/\log(b(k)) \right| \right] + 1$$

The parameter ξ is an accuracy control factor which can be normally selected as 0.01 [20]. These FIR vectors are called "Initial Vectors". We use the ADFE structure [14]. Its major feature is the ability to deal with severe fast time varying channels. The equalization process is divided into *starting* and *tracking* periods. In each period, the equalizer has special configuration which is shown in Fig.2. In this figure GC is an absolute gain controller which sets dynamic range of the received signal. Also $\Re$ and $\Im$ are recursive and transversal filters respectively with FIR forms. A phase rotator shown by PR compensates the phase distributed by both channel and equalizer. From the structure in the *starting* period, one may reasonably hope that convergence speed of the transversal filter is high [14]. To control the running modes (*starting* or *tracking* mode) a criteria is required. Since the proposed method is blind, the true transmitted data is unknown at the receiver. Hence, an estimation of the (true) output MSE is calculated [14]. The rule for changing the mode or the structure re-organization is as follows (M_0 is a reasonably small threshold):

- If $MSE(k_0) \geq M_0$, the equalizer works in the starting mode for $k > k_0$
- If $MSE(k_0) < M_0$, the equalizer works in the tracking mode for $k > k_0$



(a)



(b)

Figure 2. a) *starting* period, b) *tracking* period

All filters are updated in each iteration. Special adaptation rules govern for each period. Adaptation algorithms used in this study are the same as [14]. In [14], recursive and transversal filters start from $[0,0,...,0]$ and $[0,0,...,1,0,...,0]$ respectively but, we propose the "Initial Vectors" instead. In fact, the optimal DFE is used to initialize ADFE. The computational complexity of the proposed method is a little more than ADFE (because of finding a good initialization).

The proposed algorithm is summarized as follows:

1- First a short binary sequence is transmitted (In our simulation, 400 iterations are used). Then, the channel mean vector is blindly estimated and the transmitted data is reconstructed via PBDFE [19].

2- By the means of blindly detected data in the first step, ρ_k and $\sigma^2_{v_k}$ can be estimated based on method proposed in [6]. Because of the adaptive structure of the proposed method, we can neglect this step.

3- Based on the estimated parameters, a realizable DFE is constructed to minimize the mean square error from the frequency domain perspective [7].

4- The IIR forms are converted to FIR vectors via the method proposed in [20].

5- FIR vectors are implemented in the ADFE structure [14] (in recursive and transversal filters).

6- Estimated MSE is obtained recursively and compared with M_0 in order to decide on the running mode. Adaptation algorithm for each iteration is based on the rules described in [14].

IV. SIMULATION

In this section some illustrative simulations are given. In each simulation mean vector of the channel is normalized and MSE (db) is averaged over 50 independent simulations. The symbol scheme considered in each simulation is binary PAM except simulation 5, and the channel is a multipath Rician fading one.

Sim.1. For illustrating the superior performance of the proposed method three channels are selected. Mean vector for each channel is as follows:

$$h(Channel1):[0.0427,\ -0.0938,\ -0.2555,\ 0.7357,\ -0.6187]$$
$$h(Channel2):[0.5802,\ -0.1160,\ -0.6614,\ -0.1404,\ 0.4389]$$
$$h(Channel3):[0.1479,\ 0.0118,\ -0.5386,\ 0.7425,\ -0.3696]$$

The zero locations are shown in Fig.3. Due to proximity of the zeros to the unit circle, all channels suffer from severe fading, and are hard to be equalized. For designing procedure, the parameters ρ_k, σ_w^2 and $\sigma_{v_k}^2$ are required, but in our simulation, the real values are not chosen (real and assumed values are summarized in Tabel.1 of app.). In fact, neglecting the estimation of these parameters can lead to reduce the computational complexity without serious destructive effects. We compare our method with ADFE [14], and DFE-FD [7]. The reason for selecting these methods is the similarity of the proposed method to the combination of ADFE and DFE-FD. The proposed method and ADFE are implemented in the adaptive forms, but DFE-FD has a fixed form. The adaptation steps are the same for both the proposed method and ADFE. These steps are as large as possible (stability of the adaptive method must be guaranteed). The number of transmitted symbols is 5000. The results for different methods with $SNR = 30db$ are compared in Fig.4.

It can be inferred the ADFE is not able to achieve the desired MSE or, at least, needs a significant long time to approach it, for all three channels. Because of the fixed structure of DFE-FD, it fails in equalizing these severe channels. On the other hand, the proposed method is able to approach the desired MSE in a few iterations. Therefore, both the convergence speed and steady-state MSE are significantly improved. In other words, the performance is improved significantly while the cost is slightly higher than ADFE.

Sim.2. As it was mentioned before, we can select optional values for ρ_k, σ_w^2 and $\sigma_{v_k}^2$ instead of estimated parameters (In order to reduce the computational complexity). In this simulation, we illustrate that it doesn't have significant adverse effects. Channel (c) in fig.3 is selected. Once with real parameters of the channel and another time with the assumed ones the design procedure is done (real and assumed values are summarized in Tabel.2 of app.). The comparison results are shown in Fig.5. It can be realized that using optional values instead of estimated ones does not result in considerable lack of performance.

Sim.3. In [19] it is shown that the accuracy of estimated channel impulse response in PBDFE depends on the SNR level. In this simulation we examine effect of the accuracy of estimated mean vector on the performance of the proposed method. Again, channel (c) in Fig.3 is selected. Fig.6 shows the MSE at three different SNRs, for ADFE and proposed method. It can be concluded that the higher accuracy of the estimated mean vector leads to a better performance of the proposed method.

Sim.4. The aim of this simulation is to demonstrate the adaptive property of the proposed method in presence of severe variations of the channel. Two scenarios are presented. In the first one, a minimum phase channel is considered (Fig.7.a). After 1000 iterations, one zero is added and the previous zero locations are changed significantly (Fig.7.b). Because we use 400 iterations for estimating the mean vector of the channel, it is clear that these variations are not considered in the design procedure. $\rho_k = 0.99$, $\sigma_{v_k}^2 = 2\times10^{-5}$, $SNR = 30db$ and M_0= 0.25 are selected. The comparison results for ADFE and the proposed method are shown in Fig.7.c. The number of the transmitted symbols is 4000.

In the second scenario a non-minimum phase channel is considered (Fig.8.a). After 1000 iterations, one zero is added and the previous zero locations are changed significantly (Fig.8.b). $\rho_k = 0.95$ and other parameters are the same with the pervious simulation. MSE for ADFE and the proposed method are shown in Fig.8.c. For choosing the running mode (*starting* and *tracking* period), MSE is continually calculated and compared to M_0. According to the simulation results, high ability of the proposed method to adapt with severe channel variations without a training sequence can be realized.

Sim.5. In this simulation, modulation with larger alphabet size is used (After transmitting the binary sequence to estimate channel parameters). 4-QAM and 8-psk are considered. MSE is depicted for three channels mentioned in Sim.1. Again, real parameters of the channel are not considered in the simulation. Real values of the channel and assumed values in our simulation are summarized in Table.3. of app. It is clear that the proposed method yields much better results compared to two previous methods (Fig.9).

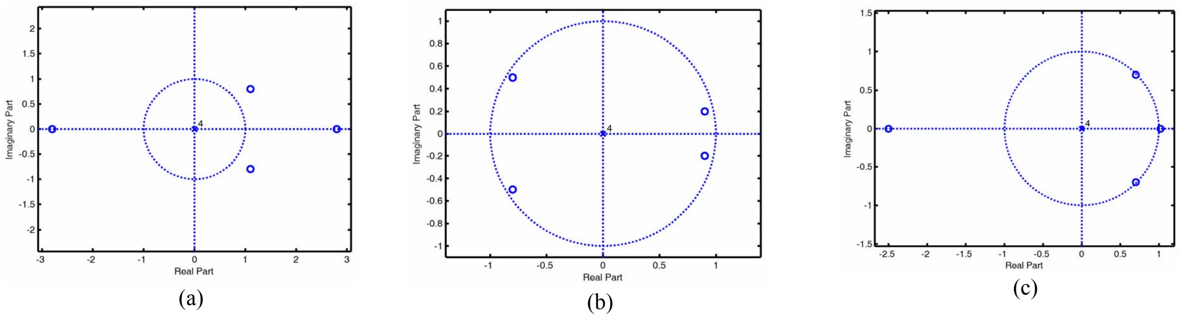


Figure.3. Zeros of each channel, a) channel 1, b) channel 2, c) channel 3

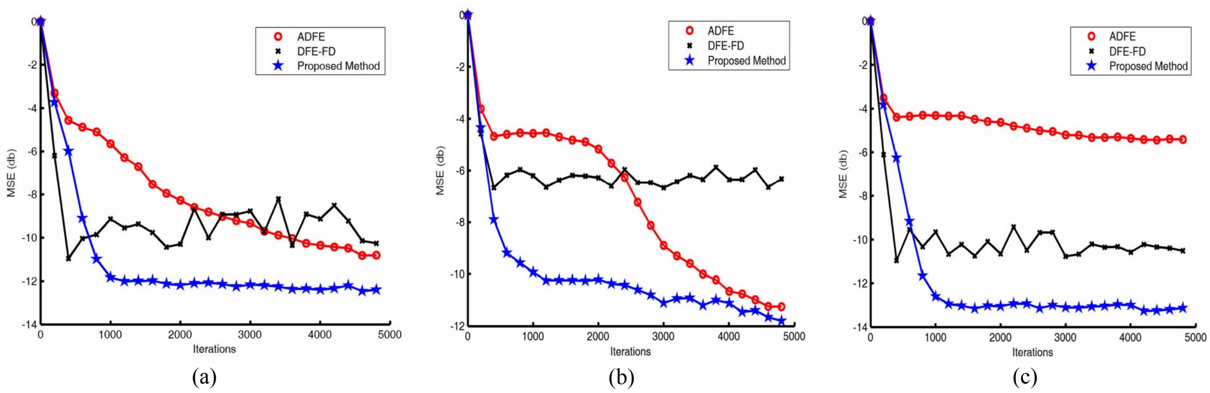


Figure.4. Compared performance (MSE) of the proposed method with ADFE and DFE-FD
a) channel 1, b) channel 2, c) channel 3 (the channel dynamic parameters are summarized in table.1 of app. for each simulation)

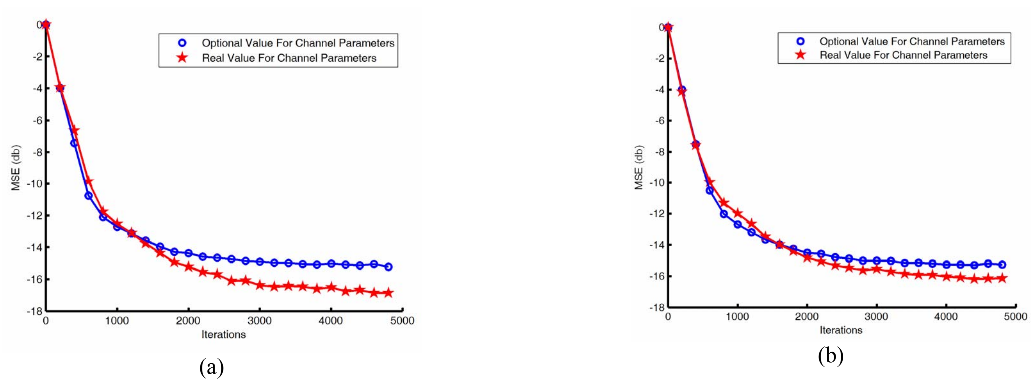


Figure.5. Compared performance (MSE) of a situation in which real values of the channel dynamic parameters are considered in the design procedure with the one that they are neglected (the channel dynamic parameters are summarized in table.2 of app. for each simulation)

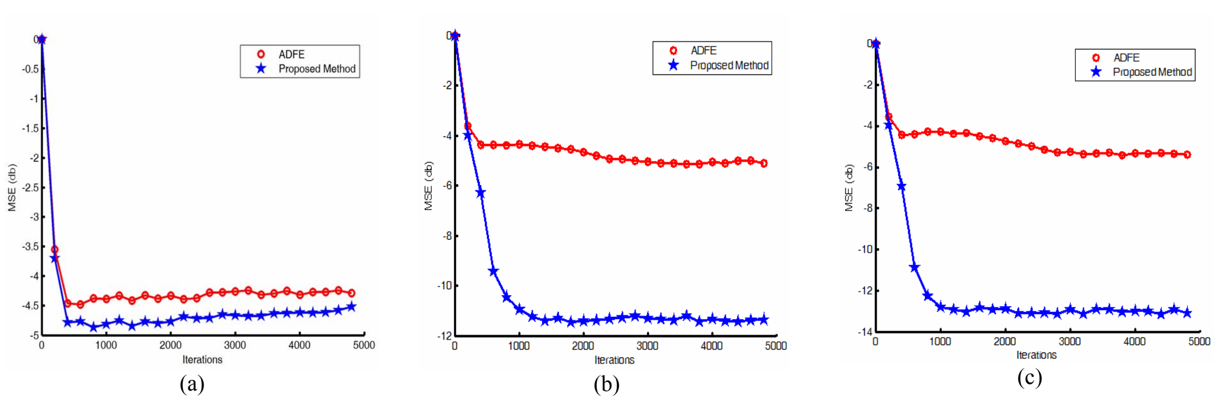
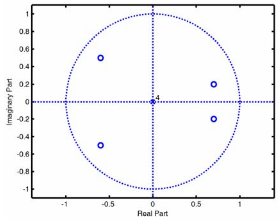


Figure.6. Compared performance (MSE) of the proposed method with ADFE in different SNRs,
a) SNR=10 db, b) SNR=20 db, c) SNR=30 db



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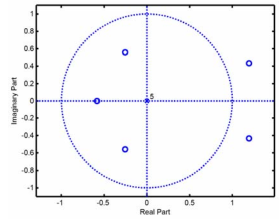

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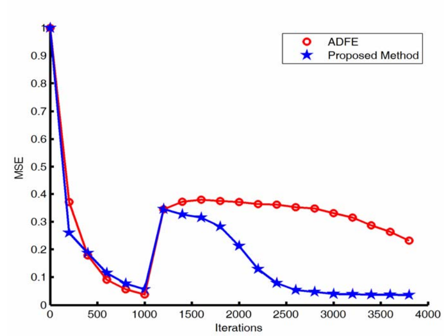
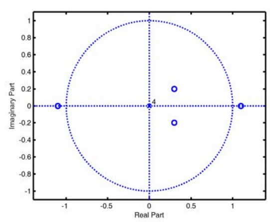
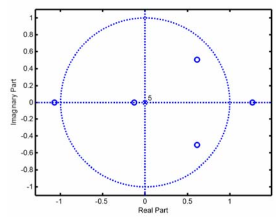


(c)

Figure.7. The First Scenario, a) Zeros of the channel before 1000 iterations, b) Zeros of the channel after 1000 iterations
c) Compared performance (MSE) of the proposed method with ADFE



(a)



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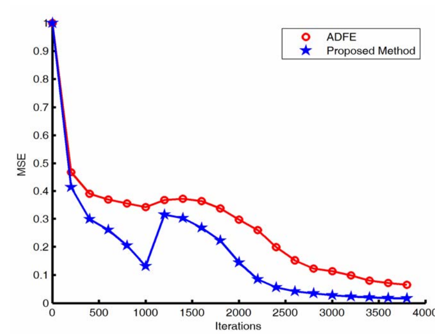
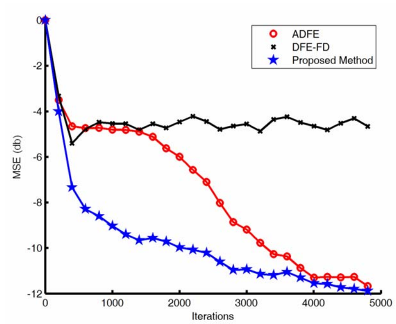
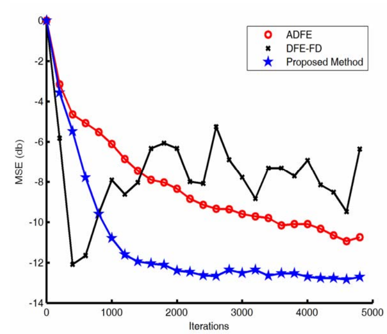


(c)

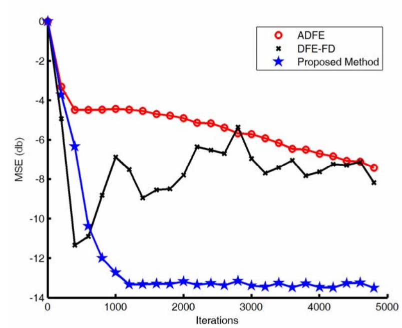
Figure.8. The Second Scenario, a) Zeros of the channel before 1000 iterations, b) Zeros of the channel after 1000 iterations
c) Compared performance (MSE) of the proposed method with ADFE



(a)



(c)



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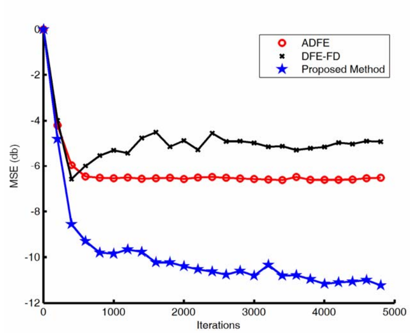

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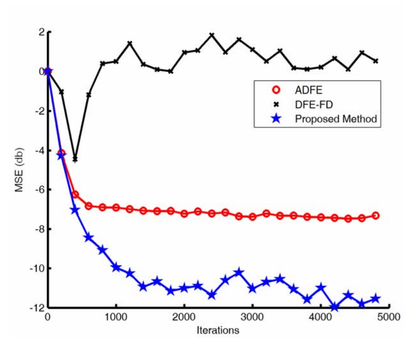

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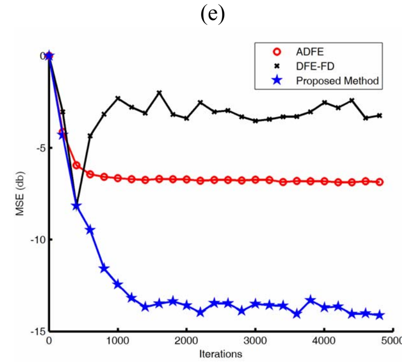

(f)

Figure.9. Compared performance (MSE) of the proposed method with ADFE and DFE-FD for three channels mentioned in Sim.1.,
a) the first channel with 4-QAM, b) the first channel with 8-PSK, c) the second channel with 4-QAM,
d) the second channel with 8-PSK, e) the third channel with 4-QAM, f) the third channel with 8-PSK
(the channel dynamic parameters are summarized in table 3 of app. for each simulation)

V. CONCLUSION

In this paper, a novel blind equalization method is presented. First, a short binary sequence is transmitted, and channel information is blindly estimated via PBDFE [19]. Based on the estimated parameters, an optimal DFE is constructed to minimize the MSE from the frequency domain perspective [7]. These optimal filters are converted to FIR vectors. These vectors are implemented in the ADFE structure [14]. It was shown at a cost of slightly more than ADFE, the performance is improved significantly and because of channel information, one can select the appropriate lengths for feedforward and feedback filters. Our simulations demonstrate that the proposed method converges to the desired MSE fast, compared to the similar methods, and due to its structure, functions properly in the fast varying channels. As a consequence, a wide field of applications can be thought of, such as radio wave transmission, TV broadcasting, voice band modems, asymmetrical digital subscriber line (ADSL), and so on.

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VII. APPENDIX

Tabel.1: Real and assumed values of the channel dynamic parameters in simulation 1

	ρ_k	$\sigma^2_{v_k}$	σ^2_w
Real value in channel (a)	0.95	0.0002	0.001
Assumed value in simulation (a)	0.8	0.001	0.0001
Real value in channel (b)	0.95	0.0004	0.001
Assumed value in simulation (b)	0.8	0.001	0.0001
Real value in channel (c)	0.95	0.0002	0.001
Assumed value in simulation (c)	0.8	0.001	0.0001

Table.2: Real and assumed values of the channel dynamic parameters in simulation 2

	ρ_k	$\sigma^2_{v_k}$	σ^2_w
Real value in channel (a)	0.9	0.00008	0.001
Assumed value in simulation (a)	0.8	0.01	0.0001
Real value in channel (b)	0.8	0.0002	0.001
Assumed value in simulation (b)	0.99	0.01	0.0001

Table.3: Real and assumed values of the channel dynamic parameters in simulation 5

	ρ_k	$\sigma^2_{v_k}$	σ^2_w
Real value in channel (a)	0.99	0.00004	0.001
Assumed value in simulation (a)	0.8	0.01	0.0001
Real value in channel (b)	0.99	0.00002	0.001
Assumed value in simulation (b)	0.8	0.01	0.0001
Real value in channel (c)	0.99	0.00004	0.001
Assumed value in simulation (c)	0.8	0.01	0.0001
Real value in channel (d)	0.99	0.00002	0.001
Assumed value in simulation (d)	0.8	0.01	0.0001
Real value in channel (e)	0.99	0.00008	0.001
Assumed value in simulation (e)	0.8	0.01	0.0001
Real value in channel (f)	0.99	0.00004	0.001
Assumed value in simulation (f)	0.8	0.01	0.0001