Iterative Equalizer for OFDM-CDMA Multiple Access Communication Systems

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***Abstract*—In this paper, the receivers with adaptive equalizer and turbo decoder are investigated for orthogonal frequency division multiplexing-code division multiple access (OFDM-CDMA) systems in multipath channels. The de-spreading equalizer which combines de-spreading and equalization is used to mitigate the MAI and ISI. The stochastic gradient based LMS algorithm is used as the adaptive algorithm for the equalizer. Three receiver architectures are investigated: The first one is the traditional receiver called de-spreading equalizer disjointed with turbo decoder. In this scheme, the equalizer and turbo decoder operate independently and the equalizer relies on slicer output to adapt its filter coefficients. However, the extrinsic information from turbo decoder can provide more reliable information of the transmitted data. Hence the second receiver architecture is called as de-spreading equalizer jointed with turbo decoder, where turbo decoder gives the extrinsic information as feedback to the adaptive equalizer as the desired information. Finally, the equalizer jointed with turbo decoder is extended to differential coded signaling in OFDM-CDMA system and its performance is presented and discussed.**

***Index Terms*—Iterative Equalizer; OFDM;CDMA; MAI;ISI**

I. Introduction

In mobile wireless communication systems, much attention had been paid to code division multiple access (CDMA) scheme due to its capabilities to provide higher capacity over conventional time division multiple access (TDMA) and frequency division multiple access (FDMA) schemes, to cope with asynchronous nature of multi-media data traffic, and to combat the adverse channel frequency selectivity. But single carrier CDMA system needs complex equalizer to mitigate ISI. Therefore, next generation wireless communication system such as Long Term Evolution (LTE) or Worldwide Interoperability for Microwave Access (WiMAX), attention has been focused on multi-carrier (MC) modulation scheme such as Orthogonal Frequency Division Multiplexing (OFDM) due to its capability of providing wideband multimedia service and efficiency of combating multipath fading when compared with CDMA counterpart. Recently, new CDMA schemes which combine orthogonal frequency division multiplexing (OFDM) technique, which combine the advantages of OFDM and CDMA, have been proposed, such as OFDM-CDMA and MC-DS/CDMA schemes. With the remarkable advance in digital signal processing techniques, they have become a topic of research [1], [2].

Error correcting code is a indispensable element is modern communication system to overcome channel impairments such as noise and multipath propagation. Turbo code [3] has the characteristic of approaching Shannon limit, offer very low BER under lower SNR. Hence turbo code has been specified in the standard such as University Mobile Telephone Service (UMTS), its successor LTE and WiMAX. The extraordinary performance turbo code is in part due to a class of suboptimal iterative decoding algorithms that generate soft outputs based on the maximum a posteriori (MAP) principle or BCJR algorithm [4]. At each decoding round, an a posteriori probability (APP) decoder provides extrinsic information for use in the second round as the a priori information. The extrinsic information about an information bit is gathered through message-passing from the channel output samples corresponding to other related bits, and its derivation is based on the structures of the interleaver and the component codes as well as the statistical property of the channel. The interleaver is thus an integrated and critical component of a turbo code and its importance has been well documented.

Although turbo code is a powerful correcting errors, the MAP or BCJR algorithm [5] has the drawback of high complexity and long decoding delay. Therefore many simplified forms of MAP algorithm were proposed such as LogMAP, MaxLogMAP, and soft output Viterbi algorithm (SOVA). The performance and complexity comparisons of all these algorithms were well documented in the literatures [5]. In this work, we use the MaxLogMAP algorithm for turbo decoder in all

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investigated schemes, because of its lower complexity than that of the MAP algorithm and better performance than the SOVA algorithm.

Utilizing the amazing performance gains of the turbo decoding algorithm, turbo equalization is an iterative equalization and decoding technique that can achieve equally impressive performance gains for communication systems that send digital data over channels that require equalization, i.e. those suffer from ISI. But in previous study of turbo equalization [6], convolutional codes were the error correction code and the channel impulse response is usually claimed to be known. Since MAP algorithm is used in equalizer and convolutional code decoder, it has the drawback of very high complexity if the delay spread is very long. In [7, 8, 9], the equalizer has been replaced with linear equalizer to reduce the complexity, but the channel impulse response still needs to be estimated. Since turbo code has been specified in the standard of many wireless systems, we investigate the turbo equalization receivers and their performance of turbo coded data transmission over ISI channels with adaptive equalizer. In our proposed receivers, the equalizers not only equalize the distorted signal but also cancel the multiple access interference (MAI), hence called de-spreading equalizer.

This paper is organized as follows. In Section II, the system model used in our work is briefly introduced. The LMS based adaptive equalizer and turbo decoder are described in section III. Section IV presents the architecture of conventional and proposed iterative receivers. Simulation results and discussions of the proposed receivers are presented in section V. Finally, conclusions are drawn in section VI.

II. System Model

Consider the downlink OFDM-CDMA system in multipath channels. The baseband structure of the OFDM-CDMA transmitter is shown in Fig. 1. Assume that there are K active users accessing this multicarrier system simultaneously, and the source information sequence for the k-th user is $m^{(k)}(n) \in \{0,1\}$. After turbo encoder, the k-th user's coded data stream is $\mathbf{x}^{(k)}(n)$ which consists of information bit $x_i^{(k)}(n)$ and two parity bits $x_{p1}^{(k)}(n)$ and $x_{p2}^{(k)}(n)$. Assume QPSK modulation (mapper) and parity bit puncture are adopted such that the effective code rate is 1/2, the k-th user's symbol is $d^{(k)}(n) \in \{\pm 1 \pm j\}$. Each user is assigned a unique spreading signature sequence $\mathbf{c}^{(k)}$ of length Q which is denoted as

$$\mathbf{c}^{(k)} = [c_1^{(k)} \quad c_2^{(k)} \quad \cdots \quad c_Q^{(k)}]^T . \tag{1}$$

The spread signal of the k-th user can be written as

$$\mathbf{s}^{(k)}(n) = \mathbf{c}^{(k)} d^{(k)}(n) = [s_1^{(k)} \quad s_2^{(k)} \quad \cdots \quad s_Q^{(k)}] . \tag{2}$$

Consequently the Q chips are guaranteed to be transmitted over approximately independently faded sub-channels. The resulting chips for the k-th user $\mathbf{s}^{(k)}(n)$ are modulated over Q subchannels with separation $\Delta f = 1/T_S$, where T_S is the bit duration, which can be implemented by taking an inverse fast Fourier transform (IFFT) of $\mathbf{s}^{(k)}(n)$. The time domain samples of the n-th OFDM block symbol of k-th user can be computed as following:

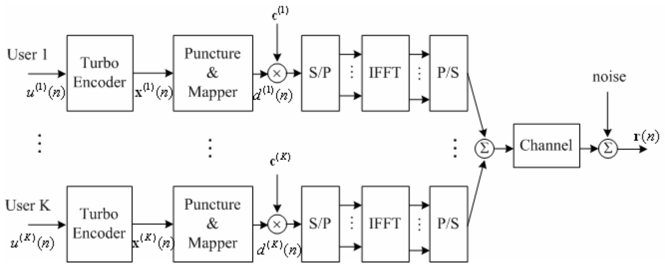

Figure 1. The structure of OFDM-CDMA transmitter

$$\mathbf{y}^{(k)}(n) = \mathbf{F}^H \mathbf{s}^{(k)}(n) \tag{3}$$

where $\mathbf{F}$ is the Fourier transform matrix defined in equation (4) and the IFFT matrix is denoted as $\mathbf{F}^H$.

$$\mathbf{F} = \begin{bmatrix} 1 & 1 & \cdots & 1 \\ 1 & e^{-j2\pi(1)(1)/Q} & \ddots & e^{-j2\pi(1)(Q-1)/Q} \\ \vdots & \vdots & \ddots & \vdots \\ 1 & e^{-j2\pi(Q-1)(1)/Q} & \cdots & e^{-j2\pi(Q-1)(Q-1)/Q} \end{bmatrix} \tag{4}$$

Note that Q denotes the total number of OFDM subcarriers. A guard interval longer than the maximal channel delay spread in the form of a cyclic-prefix (CP) is inserted in front of the P/S converted samples of the n-th OFDM block to avoid interference between consecutive OFDM symbols such that each OFDM symbol is ISI free. The guard interval insertion in the transmitter and guard interval removal in the receiver are implicitly assumed to simplify the diagram in Fig. 1 and also the derived equations in next sections.

Finally the signals of all K users are added up synchronously and then transmitted through the channel. The transmitted signal is denoted as

$$\mathbf{y}(n) = \sum_{k=1}^{K} \mathbf{y}^{(k)}(n) . \tag{5}$$

The transmission channel is modeled as a time invariant channel with N_p paths, each of which is characterized by path gain factor α_k and relative delay time $\tau_k = N_p T_c$, where T_c is the chip time. The channel impulse response can be characterized by

$$h(n) = \sum_{p=0}^{N_p - 1} \alpha_k \delta(n-p) . \tag{6}$$

Within one chip duration, the individual channel measurement samples are assumed to be non-resolvable because of the high chip rate of CDMA systems, and they are combined together as a single path.

III. THE DE-SPREADING EQUALIZER AND TURBO DECODER

In this section, the principles of LMS-based de-spreading equalizer and turbo decoder used in this work for OFDM-CDMA multiple access system are described. The equalizer, called de-spreading equalizer which has been proposed for CDMA system in [10] not only equalizes the distorted signal in the frequency domain but also de-spreads the received signal and removes the multiple access interference (MAI).

A. LMS based Adaptive Equalizer for Coherent System

Because of cyclic prefix insertion, the received signal $\mathbf{r}(n)$ in the time-domain can be described as the circular convolution of channel impulse response $h(n)$ and the transmitted signal with additive white Gaussian noise as follows

$$\mathbf{r}(n) = \sum_{l=1}^{N_P} \alpha_p \mathbf{y}(n;l) + \mathbf{g}(n)\,, \tag{7}$$

where

$$\mathbf{y}(n;1) = [y_1(n), y_2(n), \ldots, y_Q(n)]^T,$$
$$\mathbf{y}(n;2) = [y_Q(n), y_1(n), \ldots, y_{Q-1}(n)]^T,$$
$$\mathbf{y}(n;l) = [y_{Q-(l-2)}(n), \ldots, y_Q(n), y_1(n), \ldots, y_{Q-(l-1)}(n)]^T.$$

That is, $\mathbf{y}(n;l)$ denotes the signal of the l -th path delayed $(l-1)$ chip time, i.e. $(l-1)T_c$, and $\mathbf{g}(n)$ is the additive noise vector containing zero-mean, uncorrelated complex Gaussian noise samples during the n-th OFDM symbol block interval. From the expression of (7), the distorted signal through a dispersive channel can be viewed as a combination of contents of all the attenuated paths. After the samples corresponding to the CP are discarded, an FFT of size Q is performed over the received signal. The OFDM demodulation output can be described as

$$\begin{aligned}\mathbf{z}(n) &= \mathbf{F}\mathbf{r}(n)\\ &= [z_1(n) \quad z_2(n) \quad \cdots \quad z_Q(n)]^T.\end{aligned} \tag{8}$$

where $\mathbf{z}(n)$ is OFDM demodulation output vector.

The weight vector $\mathbf{w}(n)$ of the de-spreading equalizer vector for the desired user k was initialized as the spreading signature vector $\mathbf{c}^{(k)}$. From now on the super script (k) to denote the desired user k will be dropped for the briefness of equation when there is no evidence of ambiguity.

The de-spreading equalizer output signal of desired user can be formulated as

$$q(n) = \mathbf{w}^H(n)\mathbf{z}(n) \tag{9}$$

The weights here will be updated using LMS algorithm based on Minimum Mean Squared Error (MSE) criterion [11]. The error signal $e(n)$ thus can be denoted as the difference between the equalized data symbol $q(n)$ and the desired signal $d(n)$, and is denoted as

$$e(n) = d(n) - q(n) = d(n) - \mathbf{w}^H(n)\mathbf{z}(n)\,, \tag{10}$$

The cost function of adaptive algorithm known as the complex LMS algorithm is the norm of signal error. So, the cost function can be donpted as follows

$$J = [\|d(n) - q(n)\|^2] = [\|e(n)\|^2]. \tag{11}$$

To find the minimum of the cost function we need to take a step in the opposite direction of $\partial J/\partial \mathbf{w}^*$. The gradient vector of the cost function by the weight vector is manipulated as follows

$$\begin{aligned}\frac{\partial J}{\partial \mathbf{w}^*} &= \frac{\partial}{\partial \mathbf{w}^*}\|e(n)\|^2\\ &= \frac{\partial}{\partial \mathbf{w}^*}\left\|d(n) - \mathbf{w}^H(n)\mathbf{z}(n)\right\|^2\\ &= -2\mathbf{z}(n)e^*(n).\end{aligned} \tag{12}$$

Therefore, the weight vector update equation of the dispreading equalizer at the (n+1)-th iteration can be given by the following equation

$$\mathbf{w}(n+1) = \mathbf{w}(n) - \mu\frac{\partial J_{LMS}}{\partial \mathbf{w}^*}, \tag{13}$$

$$= \mathbf{w}(n) + 2\mu\mathbf{z}(n)e^*(n), \tag{14}$$

where μ is the step size. In equation (12), the desired signal $d(n)$ in non-training or decision-directed period of data transmission is not available and usually replaced by estimated value from slicer or error correction decoder. In this paper, the extrinsic information of information bits and parity bits from turbo decoder is used as the desired estimate in order to exploit the information from decoder.

B. LMS Based Adaptive Equalizer for Differentially Coded System

Differentially coded system has the advantage that it does not need to generate synchronized carrier reference in order to demodulate the received signal. Therefore, differentially coded system which uses non-coherent detection has lower system complexity when compared to its coherent counterpart.

In differentially coded system, the receiver generate the estimated symbol $\hat{d}(n)$ which is defined as

$$\hat{d}(n) = q(n)q^*(n-1)\,, \tag{15}$$

and the error signal is defined as

$$e_D(n) = d(n) - q(n)q^*(n-1)\,. \tag{16}$$

In order to apply LMS algorithm for the differentially coded system, the cost function J_D is defined as

$$\begin{aligned}J_D &= \|e_D(n)\|^2\\ &= \left\|d(n) - q(n)q^*(n-1)\right\|^2\end{aligned} \tag{17}$$

Following the derivation of the LMS algorithm, we can manipulate the cost function J_D as the following

$$\begin{aligned}J_D = [&\mathbf{z}^H(n)\mathbf{w}(n)q^*(n-1)q(n-1)\mathbf{w}^H(n)\mathbf{z}(n)\\ &-\mathbf{z}^H(n)\mathbf{w}(n)q^*(n-1)d^*(n)\\ &-d(n)q(n-1)\mathbf{w}^H(n)\mathbf{z}(n)\\ &+d(n)d^*(n)]\end{aligned} \tag{18}$$

The gradient vector of the cost function is obtained as

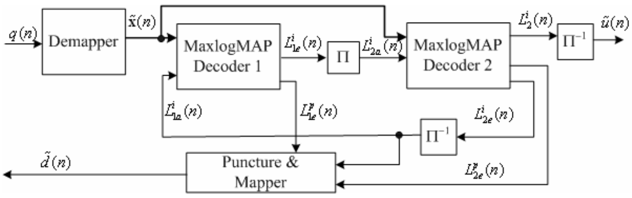

Figure 2. Turbo decoder with signal mapper and de-mapper

$$\frac{\partial J_D}{\partial \mathbf{w}^*} = -2\mathbf{z}(n)q^*(n-1)e_D^*(n). \tag{19}$$

Therefore, the code update equation at the (n+1)-th iteration can be given by the following equation

$$\begin{aligned}\mathbf{w}(n+1) &= \mathbf{w}(n) - 2\mu \frac{\partial J_D}{\partial \mathbf{w}^*} \\ &= \mathbf{w}(n) + 2\mu\mathbf{z}(n)q^*(n-1)e^*(n).\end{aligned} \tag{20}$$

C. Turbo Decoder with Mapping and De-mapping

The turbo decoder used in this work is a variant of original turbo decoder [3] which only provides the extrinsic information of information bit. The block diagram of turbo decoder with two component MAP decoders is shown in fig. 2. In this figure, the interleaver and de-interleaver are shown as Π and Π^{-1}, respectively. The equalized signal is de-mapped as estimate of coded vector $\tilde{\mathbf{x}}(n)$ which consists of information bit signal $\tilde{x}_i(n)$ and parity bit signals $\tilde{x}_{p1}(n)$ and $\tilde{x}_{p2}(n)$. Note that in each component decoder, the extrinsic information of both information and parity bits is generated. Although the component decoder needs only the extrinsic information of information bits from the other component decoder as the a priori information for next iteration, the de-spreading equalizer needs reliable estimates of information bits parity bits to adaptively adjust the weight vector.

Decoder 1 generates the log-likelihood ratios of information and parity bits conditioned on the equalized turbo coded sequence $\tilde{\mathbf{x}}(k)$. These log-likelihood ratio are denoted as $L_1^i(n)$ and $L_1^p(n)$, respectively and are defined as follows.

$$L_1^i(n) = \log\frac{p[x_i(n)=1\,|\,\tilde{\mathbf{x}}(n)]}{p[x_i(n)=0\,|\,\tilde{\mathbf{x}}(n)]}, \tag{21}$$

$$L_1^p(n) = \log\frac{p[x_{p1}(n)=1\,|\,\tilde{\mathbf{x}}(n)]}{p[x_{p1}(n)=0\,|\,\tilde{\mathbf{x}}(n)]}, \tag{22}$$

In order to reduce error propagation, only extrinsic information is sent back to the other component decoder and the de-spreading equalizer. The extrinsic information for information and parity bits for decoder1 are denoted as $L_{e1}^i(n)$ and $L_{e1}^p(n)$ which are calculated as follows,

$$L_{e1}^i(n) = L_1^i(n) - L_{a1}(n) - L_C\tilde{x}_i(n), \tag{23}$$

$$L_{e1}^p(n) = L_1^p(n) - L_C\tilde{x}_{p1}(n), \tag{24}$$

where the $L_{a1}(n)$ initialized as zero is the *a priori* information of information bit, and that is the extrinsic information of information bit from the 2nd component decoder after interleaving and L_C is the channel reliability factor defined as

$$L_C = \frac{4E_S}{N_0}, \tag{25}$$

where E_S is the energy per symbol and N_0 is the spectrum density of additive white Gaussian noise.

The log-likelihood ratios from decoder 2 are denoted as $L_2^i(n)$ and $L_2^p(n)$,.which are defined similar to equation (21) and (22). $L_2^i(n)$ is hard decision detected to give the source information estimate $\tilde{u}(n)$ after several decoding iterations. The extrinsic information for information and parity bits generated from decoder 2 is denoted as $L_{e2}^i(n)$ and $L_{e2}^p(n)$ respectively and are calculated as follows,

$$L_{e2}^i(n) = L_2^i(n) - L_{a2}(n) - L_C\tilde{x}_i(n). \tag{26}$$

$$L_{e2}^p(n) = L_2^i(n) - L_C\tilde{x}_{p2}(n) \tag{27}$$

where $L_{a2}(n)$ is the *a priori* information of information bit, and is the extrinsic information of information bit from the first component decoder after interleaving.

When adaptive equalizer is operating in decision directed mode, it will need reliable desired signal estimate to operate. In order to reduce error propagation, the extrinsic information for information bit $L_{e2}^i(n)$ and parity bit $L_{e2}^p(n)$ and $L_{e2}^p(n)$ are punctured and mapped to generate $\tilde{d}(n)$ which will be used as the desired signal estimate for the de-spreading equalizer.

IV. THE ADAPTIVE TURBO RECEIVERS

Having described the proposed architecture and operation of de-spreading equalizer and turbo decoder used in this work, we will present the receiver architectures that combine the functions of adaptive equalizer and turbo decoder in this section.

A. Aaptive Equalizer Disjointed with Turbo Decoder

Fig. 3 shows the block diagram of adaptive equalizer disjointed with turbo decoder. In this conventional architecture, the de-spreading equalizer relies on the hard decided output of equalizer as the estimated desired information to generate the error signal and adaptively adjust its weight vector. The turbo decoder will iteratively decode the equalized signal by passing extrinsic information between component decoders without sending extrinsic information to the equalizer. This architecture has the drawback that information embedded in the signal is not fully utilized. Therefore, the receiver with jointed functions is proposed in next sub-section.

B. Adaptive Equalizer Jointed with Turbo Decoder

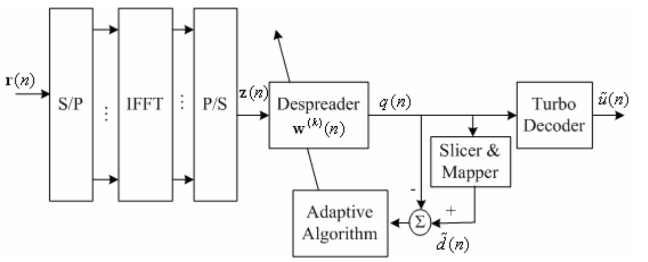

Figure 3. Equalizer with disjointed turbo decoder for OFDM-CDMA system

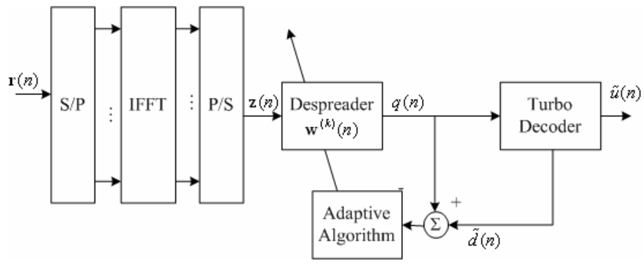

Figure 4. Equalizer with jointed turbo decoder for OFDM-CDMA system.

Fig. 4 shows the block diagram of adaptive equalizer jointed with turbo decoder. In this proposed architecture, the de-spreading equalizer relies on the extrinsic information from the turbo decoder used as the estimated desired information to generate the error signal and adaptively adjust its weight vector. The turbo decoder will iteratively decode the equalized and de-mapped signal by passing extrinsic information between component decoders and the de-spreading equalizer. This architecture has the advantage that information embedded in the signal is explored in the turbo decoder and exploited in de-spreading equalizer. It has been shown in [7] that the soft estimates from turbo decoder can be calculated from extrinsic information using the following equation

$$\hat{d}(n) = \tanh\left(\frac{1}{2}L_e(d(n))\right), \tag{28}$$

where $L_e(d)$ is the extrinsic information of coded bit $d(n)$. The function tanh in (26) needs complex hardware or software to implement. In our study, the sign function is adopted to approximate equation (26) in order to reduce the complexity and defined as follows

$$\hat{d}(n) = \text{sign}\left(\frac{1}{2}L_e(d(n))\right). \tag{29}$$

C. Adaptive Equalizer Jointed with Turbo Decoder for Differentially Coded System

Differentially coded system has the advantage that it can tolerate the carrier phase offset and reduce the cost to implement carrier phase synchronization sub-system. Fig. 5 shows the block diagram of adaptive equalizer jointed with turbo decoder for differentially coded OFDM-CDMA system. In this figure, a differential detector (DD) which performs the differential detection operation of equation (15) is included. The differential detector generates the estimate $\hat{d}(n)$ using $q(n-1)$ as reference. Although differentially coded system is expected to have worse performance than coherent system when compared with the coherent system, it still finds application where the operation condition is difficult to implement carrier phase tracking function.

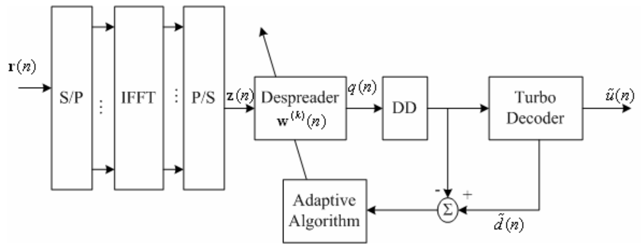

Figure 5. Equalizer with jointed turbo decoder for differential coded OFDM-CDMA system

IV. SIMULATION RESULTS AND DISSUSSIONS

In this section, simulation results of the proposed adaptive receivers are presented and compared. The generating polynomial of the turbo code is $[1 \quad (1+D^2)/(1+D+D^2)]$ and the encoded code words are punctured to $1/2$ code rate. The interleaver adopted in the simulations is the random permutation interleaver. The codeword length of turbo code after puncture is 1024 bits. The spreading code is pseudo-noise (PN) code of length 63. QPSK and $\pi/4$-DQPSK are the modulation schemes for coherent and differentially coded OFDM-CDMA systems. The channel model is 3-path fading channel with **h** = [-0.3241+j*0.4721 0.0323+j*0.3430 -0.0311+j*0.1676].

Fig. 6 and fig. 7 depict the bit error rate performance of the equalizer with disjointed and jointed turbo decoder, respectively. The performance of equalizer disjointed with turbo decoder saturates at about third iteration, while the equalizer jointed with turbo decoder still has performance gain up to 5 iterations. Fig. 8 compares the performance for the equalizers jointed and disjointed with turbo decoder when the receivers have made 5 iterations of processing. Fig. 8 shows that when equalizer jointed with turbo decoder, it has higher performance gain when compared with equalizer disjointed with turbo decoder. Fig. 9 depicts the performance of differentially coded OFDM-CDMA system with equalizer jointed with turbo decoder. It has almost 2.7dB performance difference when compared with the result in figure 8 for coherent OFDM-CDMA system.

V. CONCLUSIONS

In this paper, three iterative de-spreading equalizer schemes with turbo decoding algorithm for OFDM-CDMA system have been presented and verified with computer simulations. The conventional equalizer disjointed with turbo decoding performs worse than the equalizer jointed with turbo decoding. When the equalizer is jointed with turbo decoder, the equalizer has more reliable reference information from the turbo decoder to adaptive adjust the weight vector and the ISI and MAI can be cancelled effectively. The extrinsic information from turbo decoder is hard decided and mapped to be used as the reliable reference for the adaptive de-spreading equalizer. For differentially coded OFDM-CDMA system, the equalizer jointed with turbo decoding is about 2.7 dB worse than the receiver of the coherent OFDM-CDMA system, although the differentially coded OFDM-CDMA system has the advantage in system complexity.

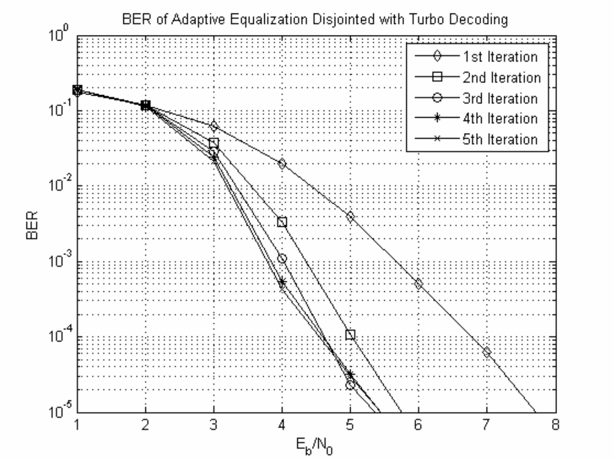

Figure 6. BER of adaptive equalizer disjointed with turbo decoding for OFDM-CDMA system with 5 active users

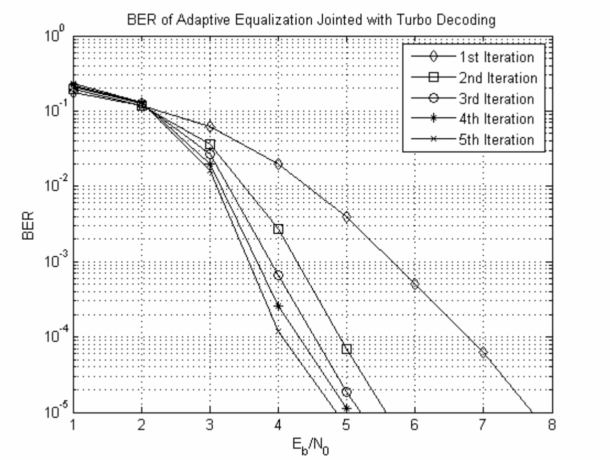

Figure 7. BER of adaptive equalizer jointed with turbo decoding for OFDM-CDMA system with 5 active users

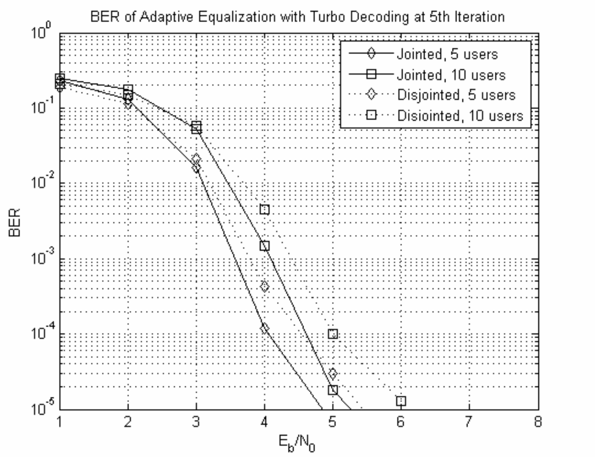

Figure 8. BER of adaptive equalizer with turbo decoding at 5th iteration for OFDM-CDMA system

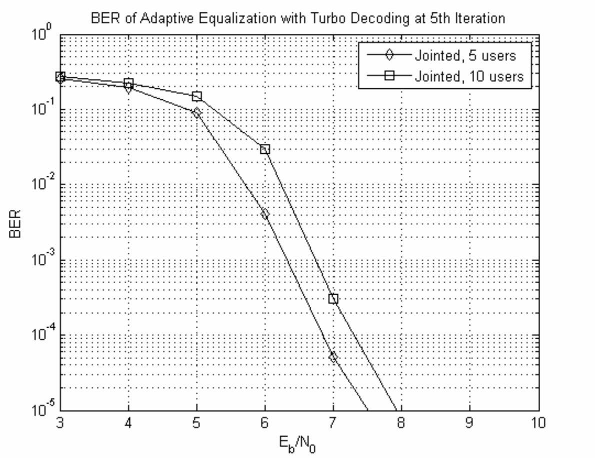

Figure 9. BER of adaptive equalizer jointed with turbo decoding for differentially coded OFDM-CDMA system

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