SIGNAL DETECTION AND ABRUPT-CHANNEL TRACKING FOR MC-CDMA

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ABSTRACT

In this paper, a channel estimation and multiuser detection method is presented for uplink MC-CDMA systems in time-varying environment. Based on the fact that the channel coefficients of two successive symbols are highly correlated, joint channel tracking and signal detection are carried out iteratively. A major contribution of the proposed approach is that it enables robust error propagation control, and is capable of detecting and tracking abrupt channel changes effectively. Simulation examples demonstrate the robustness of the proposed algorithms.

1. INTRODUCTION

Multicarrier code division multiple access (MC-CDMA) [1] has been identified as a major technique for high speed wireless communications. However, good performance for MC-CDMA is only guaranteed with accurate *channel state information* (CSI) and effective demodulation. As wireless communication at higher frequency bands is becoming possible, relative motion between the mobile, the base station and the surrounding objects introduces significant frequency dispersions. Traditional channel estimation techniques for time-invariant models are no longer applicable.

To track the time-varying channels, a widely used approach is to insert pilot bits in each OFDM symbol, and to estimate the channel coefficients utilizing filtering techniques, see [2,3] for example. These pilot-aided techniques yield good performance in fast fading scenarios but result in a considerable amount of overhead bits.

To improve spectral efficiency, joint channel estimation and signal detection approaches have been proposed to reduce the overhead bits [4, 5]. The main idea is to use the estimated symbols as pseudo-pilot signals for channel estimation, and then improve the subsequent signal detection based on the refined channel estimates. For time-varying channels, joint channel/signal estimation based on Kalman filtering has been representative [6]. Kalman filtering results in the optimal minimum variance estimator for channels characterized by the first-order autoregressive (AR) model, provided that the model parameters are known to the receiver. In most of the existing works utilizing Kalman filtering, time-varying channels are modeled as AR processes with known parameters, and no efforts are taken to estimate them.

In this paper, we propose an alternative joint channel estimation and multiuser detection scheme. Instead of assuming a known AR channel model, we use the general Jakes' model [7], making no assumptions on the knowledge of the channel model parameters. Furthermore, we investigate the more realistic circumstances when some unpredictable abrupt changes are imposed on the channels. Such changes disrupt the channel correlation and cause the error propagation problem. In this paper, effective algorithms are proposed to detect and track the abrupt changes and suppress the error propagation.

2. SYSTEM MODEL

Consider an uplink MC-CDMA system with K users. The transmitter structure of user k is illustrated in Fig. 1. The input binary stream is first mapped to BPSK or QPSK symbols and then grouped into J-symbol blocks. The ith block for user k is denoted as $\mathbf{d}_{k,i}$. Each block of data is spread into $N = PJ$ chips using the user's specific pseudo-random spreading codes $\{c_{k,i}(n), n = 0, 1, \dots N-1\}$. Here, P is the spreading factor.

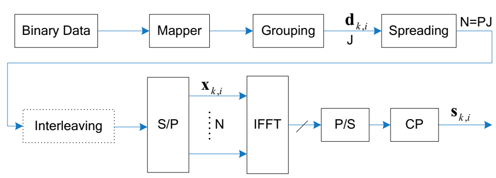


Fig. 1. Transmitter structure in an uplink MC-CDMA system.

The spread signals are interleaved and then modulated on OFDM subcarriers. These signals are saved in a vector $\mathbf{x}_{k,i} = [x_{k,i}(0), \cdots, x_{k,i}(N-1)]^T$. The output of the IFFT block is parallel-to-serial converted, and then appended with cyclic prefix (CP) at the beginning of the modulated symbol. To prevent intersymbol interference (ISI), the length of the CP is assumed to be equal to the maximum channel order L.

The transmitted signal passes through the time-varying channel which is modeled as

$$h(t, \tau) = \sum_{l=0}^{L} \gamma_l(t)\delta(\tau - \tau_l), \quad (1)$$

where $L+1$ is the total number of the multipaths from discrete scatterers; τ_l is the delay of the $(l+1)$th path; $\gamma_l(t)$ is the complex amplitude of the $(l+1)$th path at time instant t, which is assumed to be constant during the transmission of one OFDM symbol and variable on a symbol-by-symbol basis.

The signal at the receiver is sampled at the chip rate and synchronized with the desired user k. After the CP removal and the FFT demodulation, the ith symbol of user k can be written as

$$\mathbf{y}_{k,i} = \mathbf{X}_{k,i}\mathbf{H}_{k,i} + \sum_{j=1, j\neq k}^{K} \mathbf{I}_{j,i} + \mathbf{v}_{k,i}, \quad (2)$$

where $\mathbf{X}_{k,i} = diag(\mathbf{x}_{k,i})$ is the desired signal, $\mathbf{v}_{k,i}$ is the noise term, $\mathbf{I}_{j,i}$ is the interference from user j. $\mathbf{H}_{k,i} = [H_{k,i}(0), \cdots, H_{k,i}(N-1)]^T$ represents the frequency domain channel coefficients of the N subcarriers. If the channel order is L, the corresponding time domain coefficient $\mathbf{h}_{k,i} = [h_{k,i}(0), \cdots, h_{k,i}(L)]^T$

is related to $\mathbf{H}_{k,i}$ by

$$\mathbf{h}_{k,i} = [\mathbf{F}]_{L+1}\mathbf{H}_{k,i}, \tag{3}$$

where $\mathbf{F}$ is the $N \times N$ IFFT matrix defined as $F_{nk} = \frac{1}{\sqrt{N}} e^{j2\pi nk/N}$. $[\mathbf{F}]_{L+1}$ means that only the first $L+1$ rows of the matrix $\mathbf{F}$ is retained.

3. JOINT CHANNEL ESTIMATION AND SIGNAL DETECTION

In this section, the proposed approach for joint channel estimation and multiuser detection is presented. Instead of setting aside pilot carriers in each MC symbol, only one pilot symbol is needed at the beginning of each data frame. For each symbol, parallel interference cancelation (PIC) [8] is exploited to suppress MAI and improve system performance. The proposed algorithm is outlined below:

1. At the beginning of each data frame of user k ($k = 1, \cdots, K$), one pilot symbol $\mathbf{x}_{k,0}$ is transmitted to obtain an initial channel estimate $\hat{\mathbf{h}}_{k,0}$;
2. For the ith ($i > 0$) symbol, based on the channel information estimated from the previous symbol, i.e., $\hat{\mathbf{h}}_{k,i-1}$, a rough coherent detection is performed to obtain a tentative symbol estimate $\tilde{\mathbf{x}}_{k,i}$;
3. With the tentative signal $\tilde{\mathbf{x}}_{k,i}$ and the channel information $\hat{\mathbf{h}}_{k,i-1}$, ($k = 1, \cdots, K$), MAI is reconstructed and canceled out from the composite signal; a more accurate channel estimate for the ith symbol, $\hat{\mathbf{h}}_{k,i}$, is obtained and then utilized in turn to get more accurate detection of the ith symbol, denoted as $\hat{\mathbf{x}}_{k,i}$;
4. $\hat{\mathbf{h}}_{k,i}$ is used as the channel information for the next symbol and Step 2 to Step 3 are repeated to process each symbol recursively in the rest of the data frame.

4. DETECTION AND TRACKING OF ABRUPT CHANNEL CHANGES

The proposed scheme works well for "smooth" channels where no abrupt changes occur to the channel coefficients. However, in practice, the time-varying environment is not as idealistic. Channels may suffer from abrupt changes, which causes severe error propagation and results in unreliable signal detection. For tight error propagation control, we propose effective algorithms to detect and track the changes.

4.1. Detection of Abrupt Channel Changes

We assume channels are uncorrelated and no channel suffers abrupt changes simultaneously with any other channels. For analysis convenience, we introduce the algorithm in a noiseless system with K quasi-synchronous users.

Without loss of generality, the time domain channel coefficient for user l at time i, $\mathbf{h}_{l,i} = [h_{l,i}(0), \cdots, h_{l,i}(L)]^T$, can be expressed as

$$\mathbf{h}_{l,i} = \mathbf{h}_{l,i-1} + \Delta\mathbf{h}_{l,i}, \tag{4}$$

where $\Delta\mathbf{h}_{l,i}$ is the deviation from the channel coefficient $\mathbf{h}_{l,i-1}$. The corresponding frequency response is given by

$$\mathbf{H}_{l,i} = \mathbf{H}_{l,i-1} + \Delta\mathbf{H}_{l,i}. \tag{5}$$

If we assume user v is *the* user whose channel undergoes abrupt changes at time i, then

$$\|\Delta\mathbf{H}_{v,i}\| \gg \|\Delta\mathbf{H}_{l,i}\|, \text{ for } l \neq v. \tag{6}$$

where $\|\Delta\mathbf{H}_{l,i}\| = \sqrt{\sum_{n=0}^{N-1} |\Delta H_{l,i}(n)|^2}$. In the extreme case when the channels are invariant, $\|\Delta\mathbf{H}_{l,i}\| = 0$.

At the receiver end, if it is noiseless, after passing the FFT demodulator, the overall signal at subcarrier n is given by

$$r_i(n) = \sum_{l=1}^{K} H_{l,i}(n) x_{l,i}(n), \tag{7}$$

where $H_{l,i}(n)$ and $x_{l,i}(n)$ denote the channel coefficient and the transmitted signal on subcarrier n, respectively.

Following the PIC procedure, after the cancelation of the reconstructed MAI, the "cleaner" signal for user l, $y_{l,i}(n)$, is given by

$$y_{l,i}(n) = r_i(n) - \sum_{j \neq l} \hat{H}_{j,i-1}(n)\hat{x}_{j,i}(n), \tag{8}$$

where $\hat{H}_{j,i-1}(n)$ and $\hat{x}_{j,i}(n)$ are the estimates of $H_{j,i-1}(n)$ and $x_{j,i}(n)$, respectively. If the estimates are accurate, i.e., $\hat{x}_{j,i}(n) = x_{j,i}(n)$ and $\hat{H}_{j,i-1}(n) = H_{j,i-1}(n)$, then

$$y_{l,i}(n) = H_{l,i}(n)x_{l,i}(n) + \sum_{j \neq l} \Delta H_{j,i}(n) x_{j,i}(n). \tag{9}$$

The estimate of the channel coefficient at time i is given by

$$\hat{H}_{l,i}(n) = \frac{y_{l,i}(n)}{x_{l,i}(n)} = H_{l,i}(n) + \sum_{j \neq l} \Delta H_{j,i}(n) \frac{x_{j,i}(n)}{x_{l,i}(n)}. \tag{10}$$

If $l = v$, the second term on the right-hand side of (10) is approximately 0 when channels undergo slow fading, as $\Delta H_{j,i}(n) \approx 0$. Then

$$\hat{H}_{v,i}(n) \approx H_{v,i}(n) = H_{v,i-1}(n) + \Delta H_{v,i}(n). \tag{11}$$

If $l \neq v$,

$$\hat{H}_{l,i}(n) \approx H_{l,i}(n) + \Delta H_{v,i}(n) \frac{x_{v,i}(n)}{x_{l,i}(n)}. \tag{12}$$

Since we exploit constant envelope modulation such as BPSK or QPSK, (12) can be written as

$$\begin{aligned} \hat{H}_{l,i}(n) &\approx H_{l,i}(n) + e^{j\theta_{lv}(n)} \Delta H_{v,i}(n) \\ &\approx H_{l,i-1}(n) + e^{j\theta_{lv}(n)} \Delta H_{v,i}(n), \ l \neq v, \end{aligned} \tag{13}$$

where $\theta_{lv}(n)$ is the phase difference between $x_{l,i}(n)$ and $x_{v,i}(n)$. From (11) and (13), it is clear that for any l ($1 \leq l \leq K$),

$$|\hat{H}_{l,i}(n) - H_{l,i-1}(n)| \approx |\Delta H_{v,i}(n)|. \tag{14}$$

That is, when one user's channel is subjected to a sudden change, the channel estimates of all other users present a similar abrupt deviation. This finding leads to the following algorithm.

Proposition 1 *An abrupt channel change is considered to have occurred at time instant i, if any one of the following K inequalities holds:*

$$\frac{1}{N} \sum_{n=0}^{N-1} |\hat{H}_{l,i}(n) - \hat{H}_{l,i-1}(n)| > \lambda, \ 1 \leq l \leq K, \tag{15}$$

where λ is a "toleration threshold" which satisfies

$$\max_{l \neq v} \{ \frac{1}{N} \sum_{n=0}^{N-1} |\Delta H_{l,i}(n)| \} < \lambda < \frac{1}{N} \sum_{n=0}^{N-1} |\Delta H_{v,i}(n)|. \quad (16)$$

In practice, the algorithm is carried out in the time domain and the "toleration threshold" λ is obtained through a recursive process. Assuming the abrupt change occurs at time i, it can be detected if any one of the following inequalities holds:

$$\frac{1}{L+1} \sum_{j=0}^{L} |\hat{h}_{l,i}(j) - \hat{h}_{l,i-1}(j)| > \lambda(i), \ 1 \leq l \leq K, \quad (17)$$

where $\lambda(i)$ is recursively obtained by

$$\lambda(i) = \max_{l} \{ \frac{1}{L+1} \sum_{j=0}^{L} |\hat{h}_{l,i-1}(j) - \hat{h}_{l,i-2}(j)| \} + \epsilon. \quad (18)$$

When the channels are "smooth", $|\hat{h}_{l,i-1}(j) - \hat{h}_{l,i-2}(j)|$ does not deviate very much for different i's. ϵ is a small number larger than such deviation. In the simulations, the empirical value of $\epsilon = 0.5$ works well for a large range of Doppler shift when the signal to noise ratio (SNR) is larger than 5dB.

4.2. Extraction and Tracking of the Abrupt Channel

When the receiver detects the abrupt change at time i, it should identify which user's channel is subjected to the change. From (3) and (11), the channel estimate for user v at time i, $\hat{\mathbf{h}}_{v,i} = [\hat{h}_{v,i}(0), \ \cdots \hat{h}_{v,i}(L)]^T$, is given by

$$\begin{aligned} \hat{\mathbf{h}}_{v,i} &= [\mathbf{F}]_{L+1} \hat{\mathbf{H}}_{v,i} \\ &\approx \mathbf{h}_{v,i} = \mathbf{h}_{v,i-1} + \mathbf{\Delta h}_{v,i}. \end{aligned} \quad (19)$$

Here $\hat{\mathbf{h}}_{v,i}$ approximates $\mathbf{h}_{v,i}$ due to the reason that the reconstructed signals of other users are well approximated. While from (13), it is easy to see that for user l $(\neq v)$, $\hat{\mathbf{h}}_{l,i}$ deviates from $\mathbf{h}_{l,i}$ significantly since it is corrupted by the strong interference from user v. Hence it is very natural now that we use $\hat{\mathbf{h}}_{v,i}$ to reconstruct user v's signal and resort to $\{\hat{\mathbf{h}}_{l,i-1}, \ l \neq v\}$ to reconstruct other users' signals. Following the analysis in Section 4.1, it can be seen that after extracting the reconstructed signal of user v, the channel estimate of user l $(l \neq v)$, $\tilde{\mathbf{h}}_{l,i}^{v}$, is given by

$$\tilde{\mathbf{h}}_{l,i}^{v} \approx \mathbf{h}_{l,i-1} + \Delta \mathbf{h}_{l,i}. \quad (20)$$

Therefore, if user v has been identified and its reconstructed signal using $\hat{\mathbf{h}}_{v,i}$ has been extracted, $\|\tilde{\mathbf{h}}_{l,i}^{v} - \hat{\mathbf{h}}_{l,i-1}\| \approx \|\mathbf{\Delta h}_{l,i}\|$ is a small value. However, if the abrupt channel is incorrectly identified to be user k $(\neq v)$, the channel estimate of user l $(l \neq v, k)$, $\tilde{\mathbf{h}}_{l,i}^{k}$, deviates from $\hat{\mathbf{h}}_{l,i-1}$ significantly. Hence, the value of $\|\tilde{\mathbf{h}}_{l,i}^{k} - \hat{\mathbf{h}}_{l,i-1}\|$ $(l \neq k)$ is only minimized when the user who experienced abrupt channel changes is correctly identified, i.e., when $k = v$. The algorithm is analytically presented below.

Proposition 2 *If the abrupt channel change at time instant i has been detected, user v, whose channel undergoes the abrupt change, can be identified through:*

$$v = \arg \min_{\Omega_k} \{ \sum_{l \neq k} \|\tilde{\mathbf{h}}_{l,i}^{k} - \hat{\mathbf{h}}_{l,i-1}\| \}, \quad (21)$$

where $\Omega_k = \{k : 1 \leq k \leq K\}$.

4.3. Discussions

Up till now, for channel estimation at time i, we have assumed that $\{\tilde{\mathbf{d}}_{m,i}, \ 1 \leq m \leq K\}$ is obtained accurately. Although the assumption is reasonable for "smooth" channels, it cannot be guaranteed when abrupt changes occur. An optimal solution to perform joint abrupt channel identification and signal detection is given by

$$(v, \{\tilde{\mathbf{d}}_{m,i}\}) = \arg \min_{(\Omega_k, \Omega_{\mathbf{d}})} \{ \sum_{l \neq k} \|\tilde{\mathbf{h}}_{l,i}^{k} - \hat{\mathbf{h}}_{l,i-1}\| \}, \quad (22)$$

where $\Omega_{\mathbf{d}} = \{\mathbf{d}_{m,i} : \mathbf{d}_{m,i} = \{-1, 1\}^J$ for BPSK; $\mathbf{d}_{m,i} = \{\pm 1 \pm j\}^J$ for QPSK; $1 \leq m \leq K\}$. The complexity grows exponentially with the number JK, making it only applicable to some small systems. Suboptimal solutions could be found when antenna arrays are employed at the base station. When the antennas are employed to minimize the correlation between each other, it is then possible to obtain reliable $\{\tilde{\mathbf{d}}_{m,i}, \ 1 \leq m \leq K\}$ through the channels over which no abrupt changes occur.

5. SIMULATION RESULTS

A total of $K = 4$ asynchronous users are assumed in the simulations. The channel is an 8-ray multipath model with exponential power delay profile and a normalized Doppler frequency of $f_d T = 0.035$ represents the time varying environment. The block size of the original input binary data is $J = 8$ and the spreading factor is $P = 16$. BPSK is used and the entire channel bandwidth is divided into $N = 128$ subchannels.

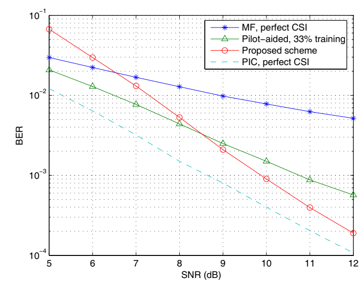
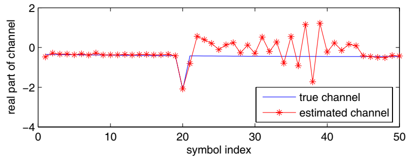


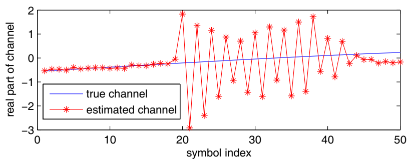
Fig. 2. BER comparison between various schemes, assuming 4 asynchronous users.

Performances of various systems are illustrated in Fig. 2. The dashed line corresponds to the method using PIC with perfect CSI knowledge, which serves as the benchmark for BER comparison. Performances of the matched filter (MF) with perfect CSI and the pilot-aided scheme are illustrated as well. For the pilot-aided scheme, the training pilots occupy 33% of the payload data and the linear interpolation technique is exploited for channel interpolation. It is shown that the proposed scheme delivers a significantly better performance than the MF and the pilot-aided schemes when the SNR is larger than 8.5 dB.

In the case when abrupt changes occur, the proposed detector suffers performance degradation if no error propagation control is applied. In the simulation, two receive antennas are employed at the base station. At the first antenna, *user 1's channel is assumed to undergo an abrupt change at time $i = 20$*. The channels of the other three users are "smooth". In Fig. 3, the real parts of the true channel coefficients and their estimates for user 1 and user 2 are plotted, as illustrated in Fig. 3(a) and Fig. 3(b), respectively.



(a) Channel estimate of user 1 (abrupt channel).



(b) Channel estimate of user 2 (smooth channel).

Fig. 3. Estimates (real part) of user 1 and user 2's channels without error propagation control: SNR=20dB, K=4, a downward "spike" occurs in user 1's channel at $i = 20$.

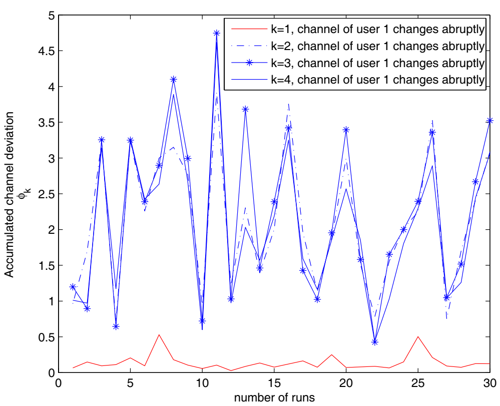
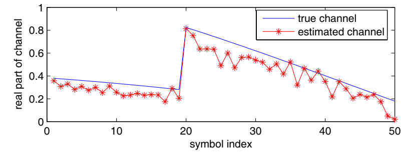
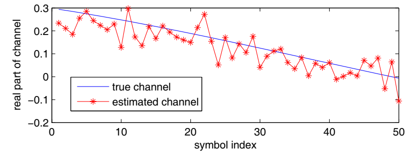


Fig. 4. Accumulated channel deviation ϕ_k when user k (k=1,2,3,4) is identified to be subjected to abrupt channel changes, while user 1's channel undergoes the abrupt changes and other users' channels are "smooth".

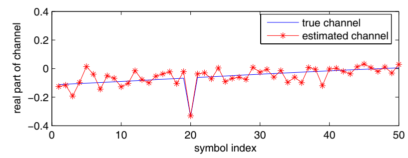


(a) Channel estimate of user 1 (abrupt channel).

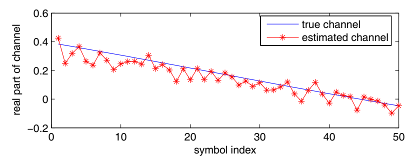


(b) Channel estimate of user 2 (smooth channel).

Fig. 5. Estimates (real part) of user 1 and user 2's channels with the proposed error propagation control: SNR=20dB, K=4, a sudden "jump" occurs in user 1's channel at $i = 20$.



(a) Channel estimate of user 1 (abrupt channel).



(b) Channel estimate of user 2 (smooth channel).

Fig. 6. Estimates (real part) of user 1 and user 2's channels with the proposed error propagation control: SNR=20dB, K=4, a downward "spike" occurs in user 1's channel at $i = 20$.

It shows that both users' channel estimates deviate after abrupt channel changes occur.

With the proposed error control algorithm, the abrupt channel can be successfully detected and identified. In (21), varying k from 1 to K, we have different evaluation of the term $\phi_k = \sum_{l \neq k} \|\tilde{\mathbf{h}}^k_{l,i} - \hat{\mathbf{h}}_{l,i-1}\|$, which corresponds to the *accumulated channel deviation* when user k is identified to be subjected to abrupt channel changes. It is shown in Fig. 4 that ϕ_k is minimized when $k = v = 1$, i.e., when the right user has been identified.

The performance of channel estimation is improved greatly after the abrupt channel is correctly identified. Fig. 5 and Fig. 6 illustrate the channel tracking in two scenarios. In Fig. 5, a sudden "jump" appears in user 1's channel at $i = 20$, while in Fig. 6, there is a downward "spike" in user 1's channel. In both cases, the channels of other users are "smooth". As shown in Fig. 5(b) and Fig. 6(b), no large deviations occur in the estimate of the "smooth" channel of user 2. Compared to the result in Fig. 3, the robustness of the proposed scheme is demonstrated.

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