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Performance Evaluation of channel estimation in TD-SCDMA system

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Abstract-

In previous B. Steiner channel estimation algorithm adopted in TD-SCDMA system, but there are some errors in estimated channel due to noise. In this paper, we evaluate the performance of Time Division-Synchronous Code Division Multiple Access (TD-SCDMA) systems with channel estimation. In channel estimation we implement the channel estimation algorithm to equalize the channel and reduce the BER in the received signal. The Channel estimation is carried out at the receiver. In TD-SCDMA, the uplink and downlink transmissions use the same frequency band for duplex transmission by using synchronized time intervals. The simulation result shows that the system performance is effectively increased.

Keywords: TD-SCDMA, Channel Estimation, Steiner channel estimation

I. Introduction

TD-SCDMA, or Time Division-Synchronous Code Division Multiple Access, is a 3G mobile telecommunications standard, being developed initially for People's Republic of China. TD-SCDMA is a time division system that uses an unpaired bandwidth structure; the same bandwidth allocation is used for both downlink and uplink in a time synchronized manner. Two most key parameters are used in TD-SCDMA, 1) Smart Antennas [1] 2) Joint Detection etc, as key technologies which depend on the fast and precise estimation of wireless CIRs. Traditional channel estimation methods of CDMA system [2] cannot meet the requirements of rapidity and real-time, because all these methods usually perform massive matrices inverse operations. B. Steiner presented a low-cost channel estimation algorithm based on accuracy and synchronization. This method simplified the complicated linear convolution operation, which increased the operation speed of channel estimation algorithm. Precisely know the numbers of multi-paths of channel and the time delay of every path), the estimated channel condition is closely to meet with the real ones. However, the receivers hardly know the information of channel under practical receiving system. In this case, if the B. Steiner algorithm is adopted directly, the results of channel estimation will be greatly affected by the background noises, and the signal-to-noise ratio (SNR) in output terminals is decreased. In this paper, an improved method based on standard model with transfer function is proposed after analyzing the B. Steiner channel algorithm. The results of simulation show that, enhance the accuracy of estimation and the performance of system.

II. Steiner channel estimation in TD-SCDMA system

In TD-SCDMA system, the data are transmitted in the form of burst. The lasting time of burst is a timeslot. The structure of burst in TD-SCDMA system is showed as figure 1. The burst consists of two data symbol fields with length of 352 chips respectively, a midamble sequence of 144 chips and a guard period of 16 chips. Among these parts, midamble sequence is used for channel estimation. For fast channel estimation, TD-SCDMA system defines 32 basic midamble groups, specifying one cellular can only uses one of these groups. The different user's midamble sequence in same cellular and same timeslot is produced by one midamble code group m_{basic} after cyclic shift. Concrete construction method of midamble sequence can be found in reference [4].

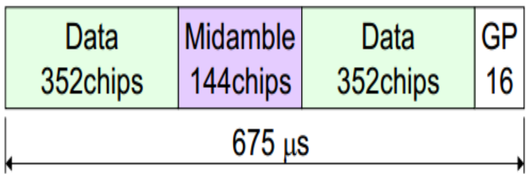

Figure 1 Time slot Burst structure

After each user's midamble sequence was produced, it is transmitted with user's data and reaches the receiver through wireless channel. The k_{th} user's CIR is defined as:

$$h^{(k)} = (h_1^{(k)}, h_2^{(k)}, \cdots, h_W^{(k)}) \qquad k = 1, 2, \cdots K \qquad (1)$$

The transmitted midamble sequence is defined as:

$$m^{(k)} = (m_1^{(k)}, m_2^{(k)}, \cdots, m_{L+W-1}^{(k)})^T \quad k = 1, 2, \cdots K \quad (2)$$

After the midamble sequence passed through channel, the result of impulse response $e^{(k)}$ is defined as:

$$e^{(k)} = m^{(k)} * h^{(k)} + n^{(k)} \tag{3}$$

The convolution operation expressed by (3) can be represented by k_{th} user's $m^{(k)}$ through matrix:

$$e^{(k)} = \begin{bmatrix} m_1^{(k)} & 0 & \cdots & 0 \\ m_2^{(k)} & m_1^{(k)} & \cdots & \cdots \\ \vdots & \vdots & \vdots & 0 \\ m_W^{(k)} & m_{W-1}^{(k)} & \cdots & m_1^{(k)} \\ \vdots & \vdots & \vdots & \vdots \\ m_{W+L-1}^{(k)} & m_{W+L-2}^{(k)} & \cdots & m_L^{(k)} \\ 0 & m_{W+L-1}^{(k)} & \cdots & m_{L+1}^{(k)} \\ \vdots & \vdots & \vdots & \vdots \\ 0 & 0 & \cdots & m_{W+L-1}^{(k)} \end{bmatrix} \bullet \begin{bmatrix} h_1^{(k)} \\ h_2^{(k)} \\ \vdots \\ h_W^{(k)} \end{bmatrix} + \begin{bmatrix} n_1^{(k)} \\ n_2^{(k)} \\ \vdots \\ n_W^{(k)} \end{bmatrix}$$

$$= G^{(k)} * h^{(k)} + n^{(k)}$$

$$k = 1, 2, \cdots K; n^{(k)} \text{ is additive noises;} \tag{4}$$

After the W(window length of channel response) is given, the situation that midamble sequence is located between two data fields causes the interference which former W-1 bits midamble codes of receiver is affected by first data field and last W-1 bits midamble codes affect second data field, so, the only codes decided by midamble sequence is $W \sim W+L-1$.

In TD-SCDMA system, midamble sequences of all K users are transmitted simultaneously. Received signals are commonly determined by midamble sequences of all K users and additive noises, so, cyclic matrix of $K \times LW$ dimensions comprehensively acquired by K users is defined as:

$$G = (G^{(1)}, G^{(2)}, \cdots, G^{(K)})_{K \times LW} \tag{5}$$

At the same time, channel response vector h of all users is defined as:

$$h = ([h^{(1)}]^T, [h^{(2)}]^T, \cdots, [h^{(K)}]^T)^T \tag{6}$$

In the case of considering additive noise into account, the total received signals can be expressed as:

$$e = Gh + n \tag{7}$$

The CIR of vector h can be obtained from (7):

$$\hat{h}^{(k)} = G^{-1} e = h + G^{-1} n \tag{8}$$

In general case, W is set to be 16, when W=16 and after removing the bits that can be interfered by data fields, the rest of the matrix G is just a 128X128 dimensions cyclic matrix:

$$G = \begin{bmatrix} m_{128} & m_{127} & \cdots & m_1 \\ m_1 & m_{128} & \cdots & m_2 \\ \vdots & \vdots & \ddots & \vdots \\ m_{127} & m_{126} & \cdots & m_{128} \end{bmatrix} \tag{9}$$

Thus FFT/IFFT fast algorithm formula can be used:

$$\hat{h} = IFFT \frac{FFT(e)}{FFT(m_{basic})} \tag{10}$$

Solve the equation (10), taking out 16 values from h by order, thus the 16 values are the estimation of each user's CIR.

III. Improvement of Channel Estimation algorithm

In TD-SCDMA system, various key technologies are built on the basis of precise estimation of CIRs, so the existence of errors can seriously affect the system performance. Therefore, the impact of these noises must be eliminated or be weakened. It is clearly can be found from (8) that the existence of errors between the estimated CIRs and the real ones is generated by the noises in channels.

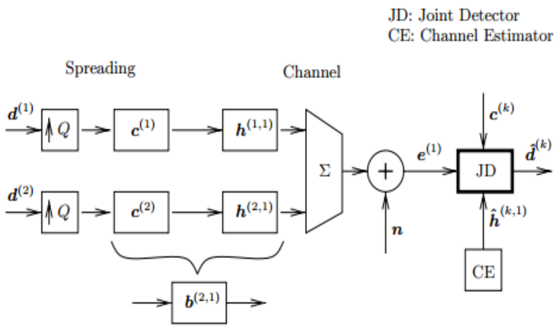

Figure2 Model of a TD-SCDMA transmission system

Channel estimation algorithms are applied on the receiver side. The signal is passing through the channel and is received by the channel estimator algorithms. Figure 2 explains the general idea behind TD-SCDMA transmission system. The receiving signal is not in the original shape. If Q spreading sequence and N data sequence are used, the spreading code vector and the transmitted data symbol vector of the k-th user are represented as, respectively,

$$\begin{aligned} \mathbf{c}^k &= [c_1^k c_2^k \cdots c_Q^k]^T, \\ \mathbf{d}^k &= [d_1^k d_2^k \cdots d_N^k]^T, \end{aligned} \tag{11}$$

Where T denotes the transpose of a vector.

If the channel impulse response is assumed to have a finite length of W, the multipath impulse response vector for the k-th user is expressed as

$$\mathbf{h}^k = [h_1^k h_2^k \cdots h_W^k]^T \quad (12)$$

The convolution vector of the estimated channel impulse response vector and the spreading code vector is expressed as

$$\begin{aligned} \mathbf{b}^k &= \hat{\mathbf{h}}^k \otimes \mathbf{c}^k \\ &= [b_1^k b_2^k \cdots b_{Q+W-1}^k]^T \end{aligned} \quad (13)$$

If G^k and C^k denote the estimated channel impulse matrix and the spreading code matrix of the k-th user, respectively, the received signal vector for the k-th user is written as

$$\mathbf{e}^k = \mathbf{G}^k\mathbf{C}^k\mathbf{d}^k = \mathbf{A}^k\mathbf{d}^k \quad (14)$$

The total received signal vector for total K users are expressed as

$$\begin{aligned} \mathbf{e} &= \sum_{k=1}^{K}\mathbf{e}^k + \mathbf{n} = \sum_{k=1}^{K}\mathbf{A}^k\mathbf{d}^k + \mathbf{n} \\ &= [\mathbf{A}^1\mathbf{A}^2\cdots\mathbf{A}^K][\mathbf{d}^{1^T}\mathbf{d}^{2^T}\cdots\mathbf{d}^{K^T}]^T + \mathbf{n} \\ &= \mathbf{A}\mathbf{d} + \mathbf{n}, \end{aligned} \quad (15)$$

If a zero forcing-block linear equalizer (ZF-BLE) is used at the receiver of BS, the transmitted data symbol vector is estimated as

$$\hat{\mathbf{d}} = (\mathbf{A}^H\mathbf{A})^{-1}\mathbf{A}^H\mathbf{e} \quad (16)$$

During transmission the signal is affected by many kinds of noises and fading effects. Since the channel estimation algorithm estimate the channel parameter with help of the training sequence and then this information is used in the receiver to extract the original data. First random data is generated and spreaded by Walsh code, the spreaded data is then scrambled by PN code and the scrambled data is then modulated and passed through the channel. The channel model used in the simulation is AWGN and multipath channel.

The main advantages of these algorithms are that it gives us efficient data from a mixture of different kind of signal and noise etc. These algorithms reduce the error noise ratio and also probability of errors. In Telecommunication the radio channel often consists of multipath fading channels. This will cause intersymbol interference (ISI) in the received signal. To remove the ISI, many kind channel equalizers are used to equalize the channel and reduce the bit error rate (BER) of the system. Figure 3 explains the adaptive channel estimation. The adaptive filter [H] adjusts its coefficients to minimize some kind of cost function between an unknown system and its output [5]. The adaptive filter and unknown system process the same input signal x [n] and have outputs d [n] and y [n] [5]. The d[n] signal is also referred to as the desired signal [5].

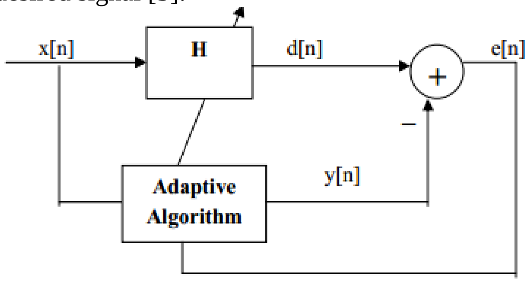

Figure 3 General Adaptive channel estimation

The error signal e [n] is computed as e [n] = d [n] - y [n], which measures the difference between the output of the unknown system and output of the adaptive filter [5]. On the basis of this measurement, the adaptive filter will change its coefficients to reduce the bit error rate [5].

IV. Simulation results and its analysis

According to the improved method proposed by this paper, its performance is simulated. Figure 4 shows the Comparison between B. Steiner Channel and estimated channel.The results of simulations are compared with the original channel estimation and the estimated channel estimation. Figure 5 Comparison between original Channel and estimated channel The maximum number of users accessing channels simultaneously is set to be 16;(Table 1 gives the parameters of channels) each user's length of window is set noises in receivers are additive white Gaussian noises, and their power are known, powers delivered by each user is equivalent and uniformly. We can find B. Steiner estimation algorithm is greatly affected by the noises.

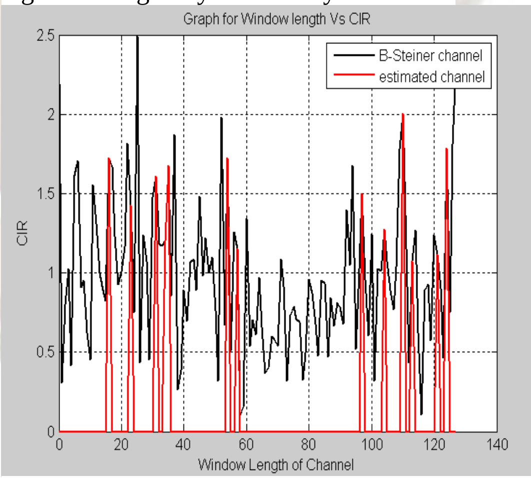

Figure 4 Comparison between B. Steiner Channel and estimated channel

Number of Multipath	Vehicle Speed : 120 km/h	
	Relative Time Delay (ns)	Average Power (dB)
1	0	0
2	310	-1
3	710	-9
4	1090	-10
5	1730	-15
6	2510	-20

Table 1 gives the parameters of channels.

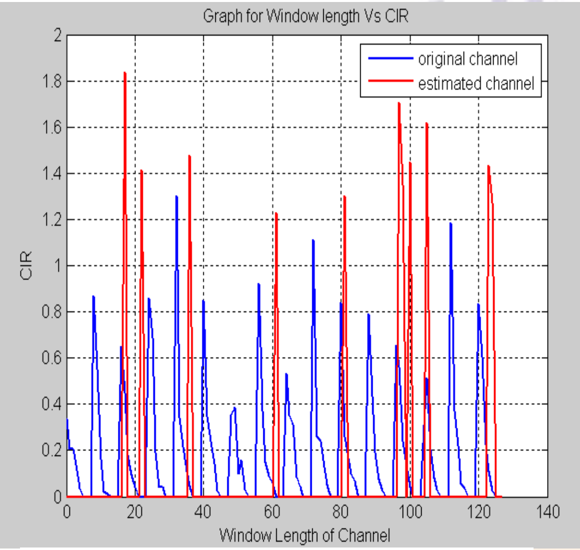

Figure 5 Comparison between original Channel and estimated channel

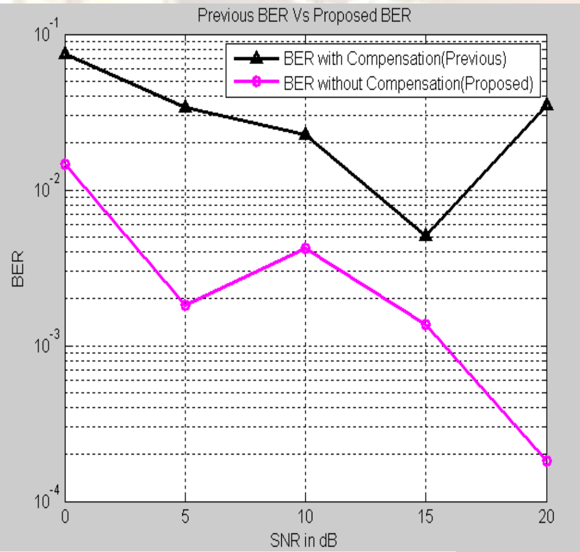

Figure 6 BER with and without CE

V. Conclusions

In channel estimation we implement the channel estimation algorithm to equalize the channel and reduce the BER in the received signal. The Channel estimation is carried out at the receiver. This improved method can not only estimate the channel impulse response accurately, but also increase the performance of system. Simulations proved the effectiveness of the proposed algorithm over the conventional channel estimator, in terms of normalized mean error, correlation coefficient and BER performance. Finally it has been concluded form simulation result that adaptive cannel estimation algorithm implemented on Receiver side reduce the BER of the system in great extent. Considering the weakness of B. Steiner channel estimation algorithm which is greatly affected by noises provides BER 10^{-1} at 0 dB SNR, $10^{-2.5}$ at 8dB SNR, this paper proposes an improved method based on standard model with transfer function. This improved method can not only estimate the channel impulse response accurately, but also increase the performance of system figure 6 shows BER $10^{-1.9}$ at 0 dB SNR, $10^{-3.8}$ at 20dB SNR. Simulations proved the effectiveness of the proposed algorithm over the conventional channel estimator, in terms of normalized mean error, correlation coefficient and BER performance.

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