Estimation of Channel States for Adaptive Code Rate Change in DS-SSMA Communication Systems: Part 2. Estimation of Fading Environment

Youngkwon Ryu, Iickho Song, Kwang Soon Kim, and Jinsoo Bae

Abstract

In this series of two papers, adaptive code rate change schemes in DS-SSMA systems are proposed. In the proposed schemes, the error correcting code rate is changed according to the channel states. Two channel states having significant effects on the bit error probability are considered: one is the effective number of users considered in Part 1, and the other is the fading environment considered in Part 2. These channel states are estimated based on retransmission requests. The criterion for the change of the code rate is to maximize the throughput under given error bound. Simulation results show that we can transmit maximum amount of information if we change the code rate based on the channel states.

I. Introduction

Because of its usefulness and many possible attractive application areas, there have been many investigations on the design and performance analysis of code division multiple access (CDMA) systems such as direct sequence (DS) spread spectrum (SS) and frequency hopping (FH) SS systems [e.g., 1-5].

When the number of hopping frequencies and the packet error probability are given, optimum code rate and optimum number of users for maximum throughput can be obtained in FH-SSMA systems. In DS-SSMA systems, however, neither optimum code rate nor optimum number of users has been obtained, since packet error probability and channel throughput are merely increasing functions of the code rate and number of users.

Among the various channel states, knowledge of the number of users is essential for the performance analysis in DS-SSMA systems. In Part 1 [6], we considered a transmission scheme which controlled the error correcting code rate adaptively according to the estimated effective number of users to maximize throughput when the packet error probability was fixed in DS-SSMA communication systems. The effective number of users was estimated based on retransmission requests.

Another channel state having major effects on the bit error performance is the fading environment. In this paper, we consider estimation of the fading environment when the effective number of users is known. A rationale of the investigation is that the characteristics of a channel felt by a mobile terminal change as the terminal moves, which result in different fading conditions over a period of communication.

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II. The System Model

Let $a_k(t)$, $b_k(t)$, and θ_k be the random signature signal, encoded signal, and phase of the kth carrier, respectively. The common carrier (angular) frequency ω_c is known, and K is the number of users. The signals $a_k(t)$ and $b_k(t)$ can be written as

$$a_k(t)=\sum_{i=-\infty}^{\infty} a_i^{(k)} P_{T_c}(t-iT_c) \tag{1}$$

and

$$b_k(t)=\sum_{i=-\infty}^{\infty} b_i^{(k)} P_T(t-iT) \tag{2}$$

where $\{a_i^{(k)}\}_{i=-\infty}^{\infty}$ is the signature sequence assigned to the

kth user, $\{b_i^{(k)}\}_{i=-\infty}^{\infty}$ is the data bit stream, T is the bit duration, and T_c is the chip duration. In (1) and (2), $P_\tau(t)=1$ for $0\le t\le\tau$, and $P_\tau(t)=0$ otherwise. We assume that each signature sequence has an integer period $N=T/T_c$. The transmitted signal for the kth user can be written as

$$s_k(t)=\sqrt{2P}\,\mathrm{Re}[\ a_k(t)b_k(t)\exp(j\omega_c t+j\theta_k)]\ . \tag{3}$$

where P is the signal power. Assuming a Rician fading channel, the faded signal can be written as

$$y_k(t)=\ \mathrm{Re}\{u_k(t-\tau_k)\}+n_{K(t)}, \tag{4}$$

where $n_k(t)$ is the additive white Gaussian noise (AWGN) term,

$$u_k(t)=\gamma\int_{-\infty}^{\infty}\beta_k(\tau,t)s_k(t-\tau)d\tau+s_k(t) \tag{5}$$

with γ the transmission coefficient of the slow-fading channel and $\beta_k(\tau,t)$ a zero-mean complex Gaussian random process.

In the receiving end, it is assumed that the phase of the despreading signal is matched to the received signal. For computational convenience, we use Reed-Solomon for error control, although use of convolutional codes might be a more realistic investigation.

Among the three methods to change the code rate, we adopt the method in which the code rate is changed by changing the information bit length with the code word length fixed. We also assume that backloged packets, requested through an ideal backward channel, are retransmitted in the next packet transmission slot with probability 1.

1. Bit Error Probability

Assuming the number of users is large enough to satisfy the central limit theorem, it has been shown in [7] that the approximation of multiple access interference (MAI) to AWGN results in quite close calculation of the bit error probability: the bit error probability is

$$P_b=Q[\ \overline{\mathrm{SNR}}\ ], \tag{6}$$

where

$$Q(x)=\frac{1}{\sqrt{2\pi}}\int_x^{\infty}\exp(-u^2/2)du \tag{7}$$

and $\overline{\mathrm{SNR}}$ is the effective signal to noise ratio (SNR) with the approximation taken into account. The effective signal to noise ratio at the input of the decoder in the receiver has been calculated in [8]. The estimation procedure in Section 3.2 is based on the observation that $\overline{\mathrm{SNR}}$ (6) can be expressed as

$$\overline{\mathrm{SNR}}=(CK'+D)^{-1/2}, \tag{8}$$

where K' is the effective number of users and C and D are determined by the fading types explained in [9]. As shown in [9], the parameters C and D are functions of the fading intensity, fading range, channel covariance, random signature sequence length, and SNR.

2. Packet Error Probability

Let P_c, P_d, and P_u denote the probabilities that a decoded packet contains no error, detectable errors, and undetectable errors, respectively. Then we have $P_c+P_d+P_u=1$, and it is easy to see that

$$P_c=\sum_{i=0}^{\lfloor (n-k)/2\rfloor}\binom{n}{i}P_b^i(1-P_b)^{n-i}, \tag{9}$$

where $\lfloor x\rfloor$ is the largest integer not greater than x, n is the codeword length, and k is the information bit length (i.e., the code rate is k/n). We also have

$$P_u=\sum_{i=n-k+1}^{n}A_iP_b^i(1-P_b)^{n-i}, \tag{10}$$

where A_i is the weight distribution defined as [6]

$$A_i=\binom{n}{i}\sum_{j=0}^{i-n+k-1}(-1)^j\binom{i}{j}[\ (n+1)^{i-n+k-j}-1]\ . \tag{11}$$

The probability P_E that the receiver commits an error when the ARQ strategy is used is

$$\begin{aligned}P_E&=\sum_{i=1}^{\text{infty}}P_uP_d^{i-1}\\&=\frac{P_u}{P_{c+P_u}}\ .\end{aligned} \tag{12}$$

III. A Code Rate Change Algorithm

In general, the error performance of the DS-SSMA system is worse than that would be caused by K real users counted by the base station because of near cell interference. The added error performance degradation effects result in increase of the 'apparent' number of users, the effective number of users K'. Note that P_E defined in (12) is now a function of the code rate k/n, the effective number of users K', fading environment, and the ratio of pure AWGN power and signal power. In [6], we have considered an estimation scheme of the effective number of users when the fading environment is known.>From the estimate of the effective number of users, we could change the code rate for optimum performance.

In this paper, we consider an estimation scheme of the fading environment when the effective number of users is known. The estimate is then used to change the code rate for optimum performance.

1. Maximum Throughput Code Rates

In this paper, our performance criterion is to maximize the throughput when P_E is limited to a fixed value, where the throughput is defined as

$$E=\frac{K}{N}\frac{k}{n}(P_u+P_c). \quad (13)$$

Thus to get a higher throughput, we should use a higher code rate and allow more users to communicate. Since P_E and the throughput are merely increasing functions of the effective number of users and code rate the code rate which satisfies the P_E bound maximizes the channel throughput.

When the effective number of users is given and the fading environment is known, we calculate P_E for all code rates (k = 1,2, $\cdots$,n) by making use of (6), (9), (10), and (12). We then select the highest code rate which results in less P_E than the given bound. As an example, Figure 1 shows the code rate for maximum throughput when the effective number of users and P_E bound are given and the fading is frequency selective (similar figures for time selective fading were shown in [6]). In this figure, n=15, N=1023, the signal power $\varepsilon=PT$, and the pure AWGN power is N_0. As expected, the code rate becomes high as the effective number of users decreases, and the effect of the pure AWGN on P_E is more severe when the fading intensity of MAI γ is smaller.

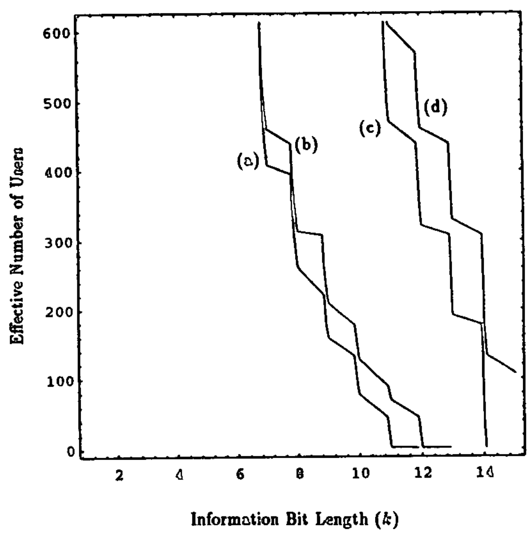

Fig. 1. Code Rate Bound for Frequency Selective Fading and Triangular Channel Covariance Function. ($P_E=10^{-4}$)

(a) $\gamma=1.0, \lambda=1.5, \frac{\varepsilon}{N_0}=10dB$
$\Rightarrow C=9.775\times 10\text{-}4,\ D=2.157\times 10\text{-}1$

(b) $\gamma=1.0, \lambda=1.5, \frac{\varepsilon}{N_0}=0$
$\Rightarrow C=9.775\times 10\text{-}4,\ D=1.657\times 10\text{-}1$

(c) $\gamma=0.05, \lambda=1.5, \frac{\varepsilon}{N_0}=10dB$
$\Rightarrow C=3.544\times 10\text{-}4,\ D=5.797\times 10\text{-}2$

(d) $\gamma=0.05, \lambda=1.5, \frac{\varepsilon}{N_0}=0$
$\Rightarrow C=3.544\times 10\text{-}4,\ D=7.974\times 10\text{-}3$

2. Estimation of Fading Environments

Let us assume that the effective number of users is known while the fading environment is not. Our estimation method of fading environment is based on the observation that $\overline{\text{SNR}}$ can be written as

$$\overline{\text{SNR}}=[\ CK'+D]^{-1/2}, \quad (14)$$

irrespective of the fading types and the covariance functions. The implication of (14) is that knowledge of the values C and D allows us to characterize the bit error probability regardless of the two fading types, time selective or frequency selective, and regardless of the two channel covariance functions, triangular and truncated exponential.

Figure 2 shows our estimation scheme, where $T_M=LT_s$ is the channel monitoring interval (CMI), R_i is the number of backloged packets of a slot in the ith CMI, K'_i is the estimate of K' in a transmission slot of the ith CMI, and $T_O=VT_M$ is the channel observation interval (COI), with T_s the slot interval and L the number of slots in a CMI.

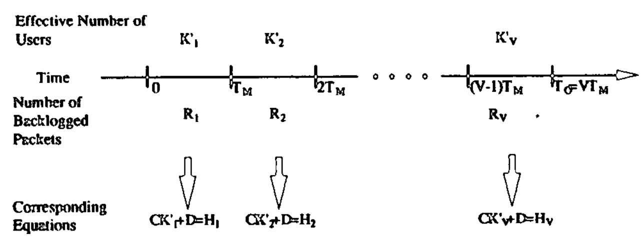

Fig. 2. Estimation of Fading Parameters.

We assume that the code rate is constant during a CMI and can be changed at the start of every CMI. We assume the number of users K is constant over a CMI, which can be achieved by not accepting new users until the end of a CMI and allowing them to start communication at the beginning of the next CMI. It is assumed that the code rate is adjusted based on the known effective number of users at the beginning of a CMI to satisfy given P_E limit.

First note that we have

$$KP_R(K'_i|k)=R_i, \quad (15)$$

where k is the information bit length trimmed to satisfy P_E limit for the previously estimated effective number of users and $P_R(K'_i|k)$ denotes the retransmission probability for K'_i effective users when the code rate is r=k / n. This probability can be calculated using (9) and (10), or

$$\begin{aligned} P_R(A|k) &= P_d|_{K'=A'} \\ &= 1-P_u|_{K'=A}-P_c|_{K'=A} \end{aligned} \quad (16)$$

for Reed-Solomon error correcting codes. For i=1,2, $\cdots$,V, we randomly choose one transmission slot out of the ith CMI and get the equation $CK'_i+D=H_i$ from (15) based on the available values K'_i, K, R_i, and k_i, where k_i is the information bit length in the ith transmission slot trimmed to satisfy given P_E bound according to the known K'_i value. Here $H_i = Q^{-1}(x)$ and x is the solution to

$$1-\sum_{j=n-k_i}^{n} A_j x^j(1-x)^{n-j} - \sum_{j=0}^{n-k_i/2} \binom{n}{j} x^j(1-x)^{n-j} = \frac{R_i}{K}, \quad (17)$$

Equation (17) is obtained from (15) and (16) using (9) and (10).

Thus we obtain V equations for 2 unknown variables, C and D. The least squares solution of the overdetermined system is

$$\begin{pmatrix} C \\ D \end{pmatrix} = (K^T K)^{-1} K^T H \quad (18)$$

where

$$K = \begin{pmatrix} K'_1 & 1 \\ K'_2 & 1 \\ \vdots & \vdots \\ K'_V & 1 \end{pmatrix} \quad (19)$$

and

$$H = \begin{pmatrix} H_1 \\ H_2 \\ \vdots \\ H_V \end{pmatrix} \quad (20)$$

Table 1. Fading Parameter Estimates for C=0.001 and D=0.05.

V	When $k_{opt}-1$ is used		When k_{opt} is used		When $k_{opt}+1$ is used	
	C	D	C	D	C	D
50	0.00050640	0.287456	0.00115249	0.014337	0.00094524	0.025939
100	0.00055415	0.263153	0.00108640	0.016887	0.00088561	0.028352
200	0.00080218	0.144957	0.00107890	0.028470	0.00104876	0.038388
500	0.00099970	0.037690	0.00107655	0.039075	0.00107509	0.047346

Table 2. Fading Parameter Estimates for C=0.0005 and D=0.03.

V	When $k_{opt}-1$ is used		When k_{opt} is used		When $k_{opt}+1$ is used	
	C	D	C	D	C	D
50	0.0076246	0.011153	0.00064714	0.013003	0.00047755	0.041151
100	0.00066074	0.010066	0.0006[illegible]213	0.022993	0.00046200	0.049053
200	0.00596008	0.025660	0.00055969	0.030665	0.00048291	0.039301
500	0.0055651	0.03550	0.00053432	0.043302	0.00049088	0.034612

Table 3. Fading Parameter Estimates for C=0.0015 and D=0.5.

V	When $k_{opt}-1$ is used		When k_{opt} is used		When $k_{opt}+1$ is used	
	C	D	C	D	C	D
50	0.00101218	0.253417	0.00112681	0.267840	0.00184825	0.321456
100	0.00110897	0.237248	0.00118833	0.264230	0.00179880	0.244140
200	0.00130293	0.245685	0.00140602	0.297850	0.00172529	0.389184
500	0.00140289	0.337690	0.00146359	0.378995	0.00148317	0.419127

Tables 1-3 show some simulation results. In general, the larger V is, the more accurate the estimated values are.

Once the values of C and D are estimated, we can adaptively change the code rate using the results of Section 3.1 to achieve maximum throughput in the next COI. A simulation result for V=500 is shown in Figure 3. In Figure 3, we let $K'=K \in (480,500)$ (so that the code rate does not change due to the variation of the number of users) and assumed that the fading environments estimation is accomplished every 4 COIs, with $T_M = T_s$ (that is, a CMI is composed of one transmission slot) for simulation simplicity.

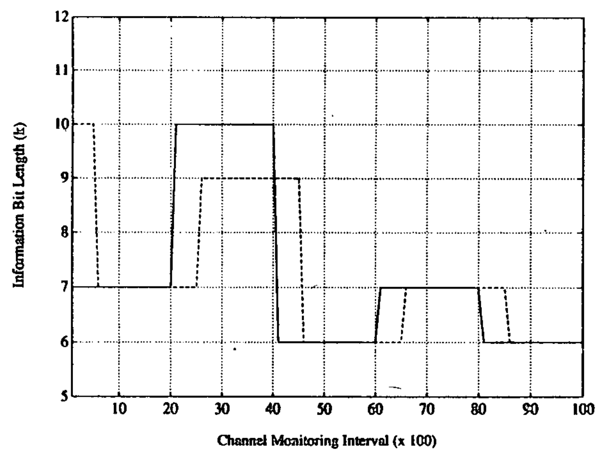

Fig. 3. A Simulation Result(real line-theory, dshed line-simulation).

IV. Conclusion

In this series of two papers, we proposed adaptive code rate change schemes in DS-SSMA systems. In the proposed schemes, the error correcting code rate was changed according to the channel states. Two channel states having significant effects on the bit error probability were considered. One was the effective number of users, and the other was the fading environment. We estimated the channel states by making use of the retransmission requests. The criterion for the change of the code rate was to maximize the throughput under given error bound.

The transmitters and receivers using the proposed schemes should have the capacity to encode and decode data at any code rate. This means hardware complexity of the communication equipments. The proposed schemes may therefore be useful under data communication conditions where high accuracy is the most important requirement.

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