Wide-Band TD-CDMA MAC With Minimum-Power Allocation and Rate- and BER-Scheduling for Wireless Multimedia Networks

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***Abstract*—A wide-band time-division–code-division multiple-access (TD-CDMA) medium access control (MAC) protocol is introduced in this paper. A new minimum-power allocation algorithm is developed to minimize the interference experienced by a code channel such that heterogeneous bit-error rate (BER) requirements of multimedia traffic are satisfied. Further, from analysis of the maximum capacity of a time slot, it is concluded that both rate and BER scheduling are necessary to reach a maximum capacity. Based on the new minimum-power allocation algorithm as well as on rate and BER scheduling concepts, a new scheduling scheme is proposed to serve packets with heterogeneous BER and quality of service (QoS) requirements in different time slots. To further enhance the performance of the MAC protocol, an effective connection admission control (CAC) algorithm is developed based on the new minimum-power allocation algorithm. Simulation results show that the new wide-band TD-CDMA MAC protocol satisfies the QoS requirements of multimedia traffic and achieves high overall system throughput.**

***Index Terms*—Bit-error rate (BER), medium access control (MAC), minimum-power allocation, quality of service (QoS), wide-band TD-CDMA.**

I. INTRODUCTION

THE capacity of a code-division multiple-access (CDMA) system varies with interference levels experienced by code channels. In order to maximize the capacity of such an interference-sensitive system, the power levels allocated to code channels need to be minimized. Thus, packet-scheduling schemes and admission control algorithms in a medium access control (MAC) protocol must take the minimum-power allocation for each code channel into account. Compared to a narrow-band CDMA system, a wide-band CDMA system [1] supports much higher traffic rates and can provide satisfactory multimedia services. The MAC protocol design for such a system is challenging because multimedia services have heterogeneous bit-error rates (BERs) and quality-of-service (QoS) requirements.

In this paper, we introduce a new MAC protocol for wide-band time-division CDMA (TD-CDMA) wireless networks. We develop a new minimum power allocation algorithm to reduce power levels of code channels. In this algorithm, we take the intra-cell and inter-cell interference and channel fading into account. The heterogeneous BER requirements of multimedia traffic are guaranteed by satisfying the target SINR values of different services. Thus, the new minimum-power allocation algorithm is not directly based on BER, which it is different from [2]. We establish the relationship between target SINR values and BER requirements by considering the error control schemes and channel fading. Different from the power control algorithms in [3]–[5], our new algorithm considers both heterogeneous BER requirements of multimedia services and multiple code channels in a mobile terminal. Although a novel power control algorithm is developed in [6], it is based on an asymptotic analysis in which bandwidth and mobile terminals are assumed to be very large.

Based on the minimum-power allocation algorithm, we analyze the maximum capacity of a time slot and conclude that the packet scheduler must be based on joint rate- and BER-scheduling in order to reach the maximum capacity of each time slot. We determine the variable system capacity and available code channels in a time slot by our new minimum-power allocation algorithm, while the transmission priorities of packets are determined according to the QoS requirements of multimedia traffic. Therefore, the new joint scheduling scheme increases the system throughput, guarantees heterogeneous BER requirements, and satisfies QoS requirements of multimedia traffic.

We also introduce a new connection admission control (CAC) algorithm for real-time connections. It is an effective bandwidth-based scheme [7], but it considers the minimum power allocation for each code channel. Compared to other power-control-based CAC algorithms [8], [9], our solution considers the variable number of code channels in a mobile terminal, which is necessary when traffic is bursty.

Several MAC protocols have been proposed for TD-CDMA systems in recent years [10]–[12]. A MAC protocol with BER scheduling (called WISPER) is proposed for wireless multimedia networks in [10]. Power levels of all code channels are assumed to be equal in WISPER, i.e., the power levels are not minimized. Although power control is used in [11] to determine the transmitted power levels of mobile terminals in Universal Mobile Telecommunication System (UMTS) TD-CDMA networks, the capacity of a time slot is assumed to be fixed. The resource allocation algorithm in [12] considers different received power levels for various service types, but the power level of a service type is assumed to be fixed. Thus, the power level is not minimized in [12]. None of the existing solutions [10]–[12]

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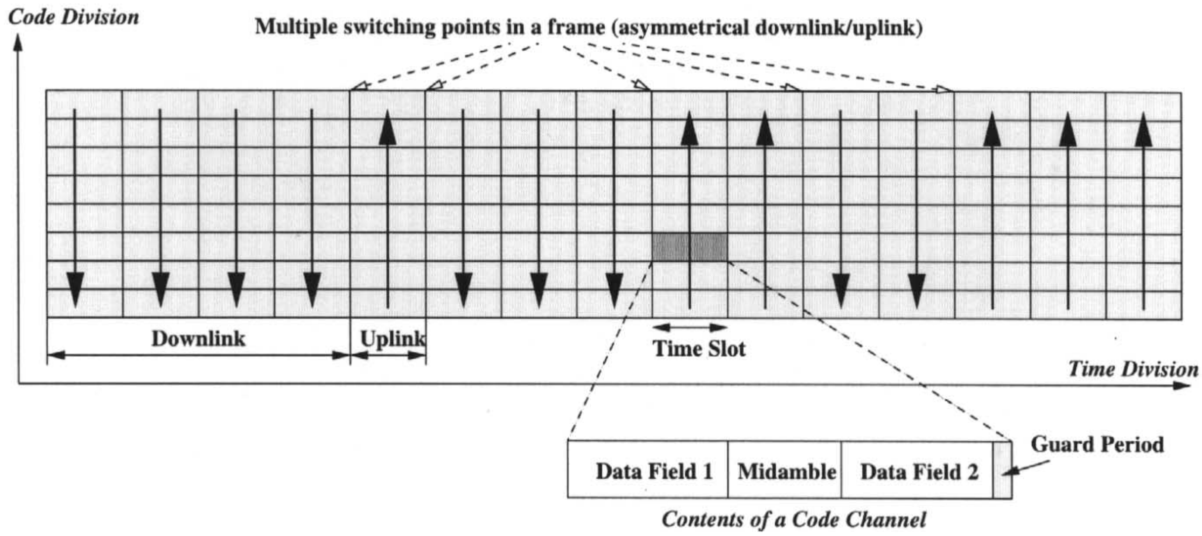

Fig. 1. The structure of one wide-band TD-CDMA frame.

address the CAC issue. General concepts of resource management strategies and the MAC protocol functions of wide-band TD-CDMA are specified by the Third Generation Partnership Project (3GPP) in [13] and [14], but to date and to our knowledge, no concrete MAC protocol has been developed to satisfy the above specifications.

This paper is organized as follows. In Section II, we introduce the basic wide-band TD-CDMA MAC protocol. In Section III, we develop the new minimum-power allocation algorithm. In Section IV, we find that the joint rate- and BER-scheduling is necessary in order to achieve the maximum capacity of a time slot. Based on the joint rate- and BER-scheduling concept and the new minimum-power allocation algorithm, we develop a new packet-scheduling scheme in Section V. Further, we introduce an effective bandwidth-based CAC algorithm in Section VI. We evaluate the performance of the wide-band TD-CDMA MAC protocol in Section VII, and conclude the paper in Section VIII.

II. New MAC Protocol

A. System Model

Wide-band TD-CDMA has the following features [15]–[18]:

- *Multiple switching points between uplink and downlink*. Due to TDD mode, additional interference comes from heterogeneous uplink/downlink traffic asymmetry in different cells [19]. A dynamic channel assignment algorithm [13], [20] is usually used to adjust the switching points between uplink and downlink so that such interference is reduced.
- *Open-loop power control architecture in the uplink*. Since TDD mode is used in the wide-band TD-CDMA, the path loss in the uplink can be measured from the downlink signal. Thus, open-loop power control is adopted in the uplink. It does not have stability issues.
- *Discontinuous transmission*. A code channel does not transmit packets in all time slots of a frame. The same code channel in different time slots can be allocated to different mobile terminals.

The frame structure of a wide-band TD-CDMA system is shown in Fig. 1, where uplink and downlink are multiplexed in time division [16]. In general, the traffic load of uplink and downlink is asymmetric, and multiple switching points exist between uplink and downlink transmissions. Multiple code channels co-exist in one time slot, and each code channel consists of two data fields, one midamble field, and a guard period [18].

As in [13], we assume multicode (MC) operation in both uplink and downlink. Thus, different transmission rates of a mobile terminal can be achieved by varying the number of code channels for the mobile terminal. In the MC operation, the codes of a mobile terminal are generated by subcode concatenation [21]. Thus, the self-interference among the codes of the same user is avoided [21].

Many transport channels are defined for wide-band TD-CDMA networks in 3GPP specifications [14], [18]. Here we focus on packet transmissions in dedicated channels (DCHs). Since signaling channels are required to assist DCHs, we also consider two other transport channels, i.e., the random access channel (RACH) to send requests from a mobile terminal to the base station and the broadcast channel (BCH) to send feedback of resource allocation from the base station to mobile terminals. Mappings of DCH, BCH, or RACH onto a physical channel (i.e., a code channel in a time slot) are specified in [18].

B. Basic Procedures

The uplink operation procedures of the new MAC protocol are shown in Fig. 2. Due to the broadcast nature, a similar but simpler procedure can be applied to the downlink transmission. The maximum number of code channels in a time slot is determined by the new minimum-power allocation algorithm. The switching points between uplink and downlink are determined by a dynamic channel allocation (DCA) algorithm [13], [20]. A MAC packet data unit (PDU) in DCHs, BCH, or RACH is mapped onto a code channel in a time slot through procedures such as CRC attachment, channel coding, interleaving, and rate matching [15], [17], [22]. As shown in Fig. 2, the MAC protocol includes the following main functions:

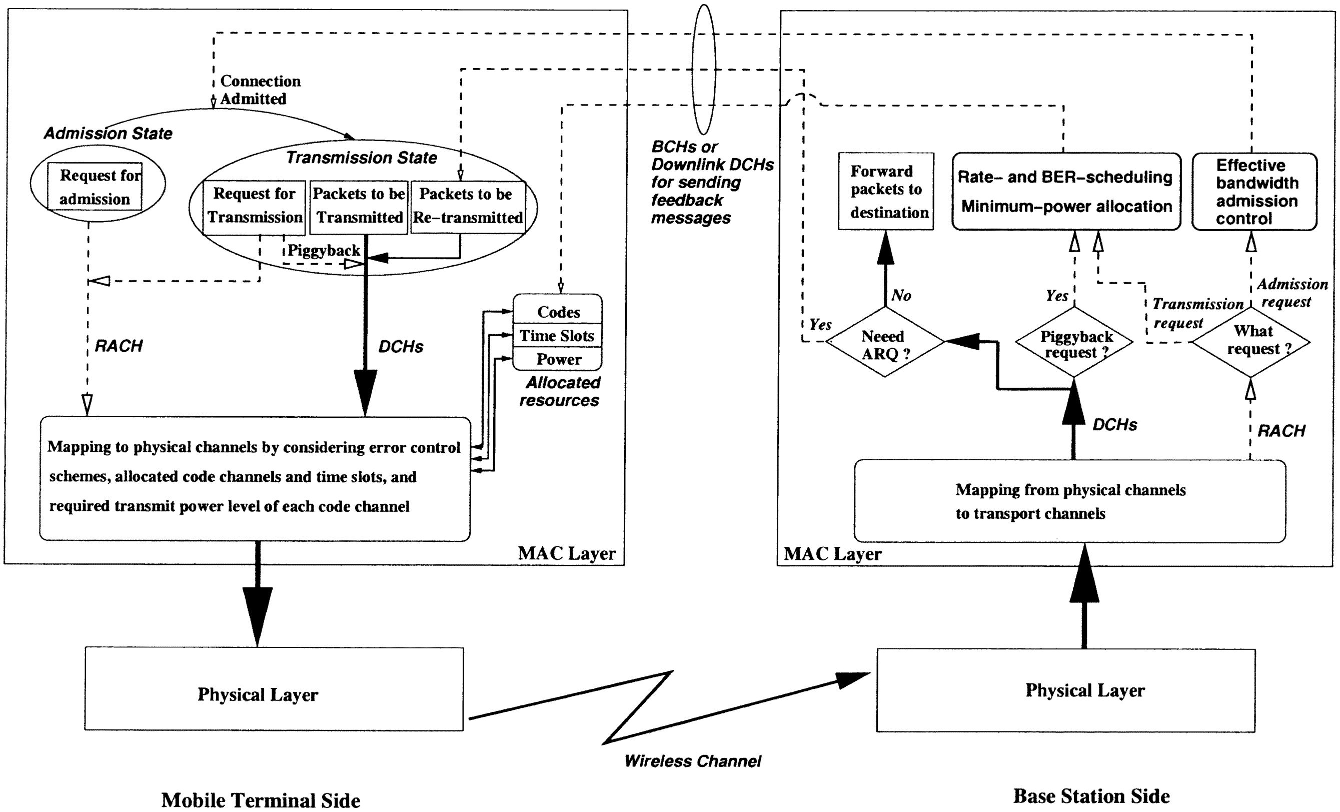

Fig. 2. Operation procedures of the MAC protocol.

- *Admission control*: This step only applies to mobile terminals with real-time traffic. When communication is initiated, a mobile terminal enters the *admission state* and sends an admission request in RACH by spreading signal with a randomly selected primary PN code. The request should include the necessary information such as the average traffic rate, the maximum transmission capability in a time slot, and BER requirement of the mobile terminal. When the base station receives such a request, a new effective bandwidth-based CAC algorithm is invoked to check whether enough bandwidth is available in the system. The result is sent back to the mobile terminal through BCH. If the answer is positive, the request is accepted, and the primary PN code is reserved for the mobile terminal. The mobile terminal then enters the *transmission state*. Otherwise, the admission request is blocked, and the mobile terminal will try to request for admission again after a random duration.
- *Packet scheduling*: In the *transmission state*, a mobile terminal sends a transmission request before it transmits a batch of packets. The transmission request can be sent in RACH or piggybacked in a DCH. This request specifies the mobile terminal ID, the number of packets that need to be transmitted in the next frame, and the timeout value of a batch of packets. The timeout value is specified according to the delay requirement of a service. After the base station collects the transmission requests from all mobile terminals during an entire frame length, it uses a packet-scheduling scheme to determine how the packets of multimedia applications are accommodated to each time slot. The packet scheduler is a joint rate- and BER-scheduling scheme based on minimum-power allocation for each code channel. It maximizes the system capacity and satisfies the heterogeneous BER requirements of multimedia traffic. Other QoS metrics such as delay and packet loss ratio are also considered in the scheduling scheme. After scheduling is completed, the base station sends the results back to mobile terminals through BCHs or piggybacks those results in downlink DCHs. The feedback information for each terminal includes the terminal ID, assigned time slots, the number of code channels in each assigned time slot, and the transmitted power level of each code channel. Such information can be sent back to a mobile terminal very fast, because multiple downlink/uplink switching points exist in a wide-band TD-CDMA frame.
- *Packet transmission*: Based on the permission feedback from the base station, the mobile terminal transmits packets in the specified time slots with the allocated power level. The base station receiving these packets forward them to the desired destination. Since erroneous packets are not allowed by nonreal-time loss-sensitive services, ARQ is applied. All packets that have not been transmitted in the current frame wait for permission in the next frame unless timeout occurs.
- *Mapping between physical and transport channels*: Based on the applied error-control scheme and the allocated resources (codes, time slots, and power levels), the transport channels are mapped onto physical channels or *vice versa* as specified in [18] and [22].

III. Minimum-Power Allocation

A. SINR of a Code Channel

For the uplink of a wide-band TD-CDMA system, we use the MC operation in the dedicated channels [13]. Given a code channel allocated to a mobile terminal in a time slot, its experienced interference at the receiver on the base station includes both intra-cell and inter-cell interference. Intra-cell interference results from code channels allocated to other mobile terminals in the same cell. Suppose the transmitted power level of a code channel of the mobile terminal n of service type k is $P_{k,n}$, where $n = 1, \ldots, N_k$, $k = 1, \ldots, K$, K is the number of service types supported in the system, and N_k is the total number of mobile terminals of type k that have code channels in the time slot. The number of code channels allocated to mobile terminal n is assumed to be $m_{k,n} \in \{1, \ldots, M_k\}$, where M_k is the maximum allowed number of code channels. We also assume that the path loss between mobile terminal n of service type k and the base station is $a_{k,n}$. Thus, the received power level in the base station of one of the $m_{k,n}$ code channels, denoted by $S_{k,n}$, is $P_{k,n}/a_{k,n}$. In addition, the overall received power level of mobile terminal n of service type k is $m_{k,n}S_{k,n}$. Thus, considering a specific mobile terminal j of service type i, the intra-cell interference I_{intra} experienced by one of its code channels at the base station is

$$I_{\text{intra}} = \sum_{k=1}^{K}\sum_{n=1}^{N_k} m_{k,n}S_{k,n} - m_{i,j}S_{i,j}, \tag{1}$$

where $S_{i,j}$ is the received power level of a code channel of mobile terminal j of service type i, and thus, $S_{i,j} = P_{i,j}/a_{i,j}$.

The inter-cell interference in the base station in the cell of interest, denoted by I_{inter}, exists for several reasons [19], [20]: 1) the users in neighboring cells transmit signals in the same time slot that is used by the user in the cell of interest; 2) different uplink/downlink asymmetry in the neighboring cells; and 3) power leaks due to improper time synchronization. The third type of interference becomes very small when cells are synchronized, and the second type of interference can be avoided by a slow DCA algorithm [20]. Thus, only the first type of inter-cell interference is considered in this paper.

Within a single time slot, both I_{inter} and I_{intra} are approximately constant, and can be considered deterministic [4], [20]. Assume that the thermal noise is N_0B_s, where N_0 is the power spectrum density and B_s is the spread bandwidth. Then, the SINR of a code channel of mobile terminal j of service type i, denoted by $\lambda_{i,j}$, is given by

$$\lambda_{i,j} = \frac{S_{i,j}}{\dfrac{(I_{\text{intra}} + I_{\text{inter}})}{1.5G} + \dfrac{N_0B_s}{G}} \tag{2}$$

where coefficients 1.5 arise because the thermal noise is assumed to be Gaussian and rectangular pulses are used in the code waves [3]. G is the processing gain.

B. Minimum Power Level

In order to guarantee the BER of mobile terminal j of service type i, the SINR $\lambda_{i,j}$ must satisfy

$$\lambda_{i,j} \geq \gamma_i, \tag{3}$$

where γ_i is the target SINR value. When each mobile terminal is allocated one code channel, it is proved in [4] that the minimum power level of a code channel is achieved when the equality of (3) holds. Such monotonicity can be generalized to the scenario where one mobile terminal has multiple code channels. Thus, $\lambda_{i,j} = \gamma_i$ must hold in order to have minimum power allocation in all the code channels, i.e.,

$$\frac{S_{i,j}}{\dfrac{(I_{\text{intra}} + I_{\text{inter}})}{1.5G} + \dfrac{N_0B_s}{G}} = \gamma_i. \tag{4}$$

After some derivations (see Appendix A), the minimum power level of a code channel is

$$S_{i,j} = \frac{1.5GN_0B_b + I_{\text{inter}}}{\left(1 - \sum_{k=1}^{K}\sum_{n=1}^{N_k} \dfrac{m_{k,n}}{m_{k,n} + \frac{1.5G}{\gamma_k}}\right)\left(m_{i,j} + \frac{1.5G}{\gamma_i}\right)} \tag{5}$$

where $B_b = B_s/G$. For the purpose of clarity, we substitute $1.5G/\gamma_i$ by ρ_i in what follows.

Suppose the maximum transmitted power level of a code channel of service type i is P_i^{max}. Then, $S_{i,j} \leq P_i^{max}/a_{i,j}$ must hold. Putting this requirement back into (5) and after some algebra, we have the following constraint:

$$\sum_{k=1}^{K}\sum_{n=1}^{N_k} \frac{m_{k,n}}{m_{k,n} + \rho_k} \leq 1 - \frac{a_{i,j}(1.5GN_0B_b + I_{\text{inter}})}{P_i^{max}(m_{i,j} + \rho_i)}. \tag{6}$$

Otherwise, no solution exists.

If

$$\Delta \equiv \max_{\substack{j=1,\ldots,N_i \\ i=1,\ldots,K}} \frac{a_{i,j}(1.5GN_0B_b + I_{\text{inter}})}{P_i^{max}(m_{i,j} + \rho_i)}$$

then, from (6), the following constraint must hold:

$$\sum_{k=1}^{K}\sum_{n=1}^{N_k} \frac{m_{k,n}}{m_{k,n} + \rho_k} \leq 1 - \Delta. \tag{7}$$

Considering a time slot in a TD-CDMA network, if such a constraint is satisfied, the BER requirements of all mobile terminals can be guaranteed by using the minimum power level in each code channel. If we define $W_{k,n} = (m_{k,n}/(m_{k,n} + \rho_k))$ as the normalized capacity consumed by a mobile terminal that uses $m_{k,n}$ code channels to transmit packets of service type k, then (7) becomes

$$\sum_{k=1}^{K}\sum_{n=1}^{N_k} W_{k,n} \leq 1 - \Delta \tag{8}$$

which means that in a time slot the overall normalized capacity consumed by mobile terminals of all service types must be less than $1 - \Delta$.

IV. Maximum Capacity of a Time Slot

For service type k, if $T_{k,l}$ denotes the number of mobile terminals that are allocated l code channels in the same time slot, then the overall normalized capacity of those mobile terminals is $T_{k,l}(l/(l+\rho_k))$. For any mobile terminal of service type k, the number of its allocated code channels must be within

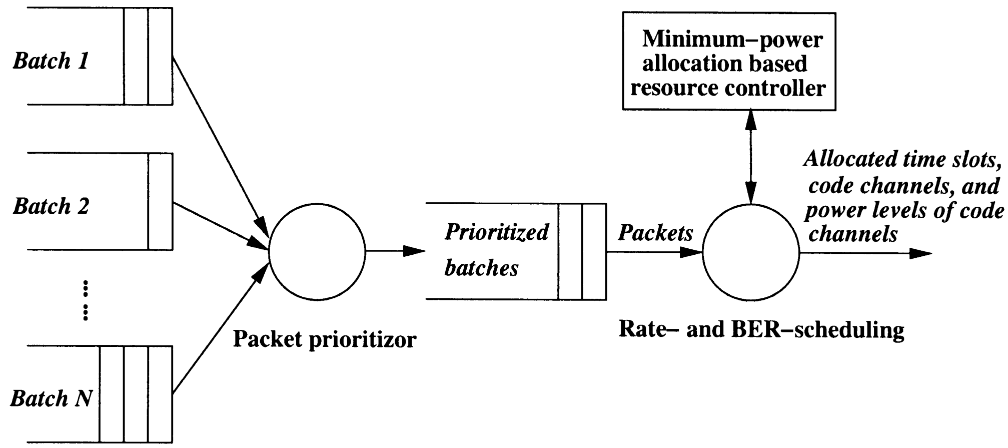

Fig. 3. New packet scheduler.

$\{1,\ldots,M_k\}$. Thus, the overall normalized capacity consumed by all mobile terminals of service type k, i.e., $\sum_{n=1}^{N_k} W_{k,n}$, is equal to $\sum_{l=1}^{M_k} T_{k,l}(l/(l+\rho_k))$. Therefore, the constraint in (8) becomes

$$\sum_{k=1}^{K}\sum_{l=1}^{M_k} T_{k,l}\frac{l}{l+\rho_k} < 1-\Delta. \tag{9}$$

According to the definition of $T_{k,l}$, the number of mobile terminals of all service types in a time slot can be described by a vector $[T_{1,1},\ldots,T_{k,l},\ldots,T_{K,M_K}]$. If C_t denotes the number of code channels of all mobile terminals in a time slot, it is given by

$$C_l = \sum_{k=1}^{K}\sum_{l=1}^{M_k} lT_{k,l}. \tag{10}$$

When minimum-power allocation is used, $T_{k,l}$ must satisfy the constraint in (9). As a result, in order to maximize C_t, we need to solve the following optimization problem: Find a vector $[T_{1,1},\ldots,T_{k,l},\ldots,T_{K,M_K}]$ that satisfies

$$\begin{cases} \sum_{k=1}^{K}\sum_{l=1}^{M_k} T_{k,l}\frac{l}{l+\rho_k} < 1-\Delta \\ [T_{1,1},\ldots,T_{k,l},\ldots,T_{K,M_K}] \geq \mathbf{0} \end{cases} \tag{11}$$

and maximizes the objective function C_t in (10).

According to the theory of linear programming, C_t is maximized at one of the extreme points given by (11). By investigating (11), we know the set of extreme points consists of $[(1-\Delta)(1+\rho_1),0,\ldots,0],\ldots,[0,\ldots,0,(1-\Delta)(l+\rho_k)/l,0,\ldots,0],\ldots,[0,\ldots,0,(1-\Delta)(M_K+\rho_K)/M_K]$. Substituting an extreme point, e.g., $[0,\ldots,0,(1-\Delta)(l+\rho_k)/l,0,\ldots,0]$, into (10), the value of C_t is $(1-\Delta)(l+\rho_k)$. Thus, the maximum value of C_t, denoted by C_t^*, must be

$$C_t^* = \max\{(1-\Delta)(l+\rho_k)|\, l=1,\ldots,M_k,\, k=1,\ldots,K\} \tag{12}$$

i.e.,

$$C_t^* = \max\{(1-\Delta)(M_k+\rho_k)|\, k=1,\ldots,K\} \tag{13}$$

which implies that C_t^* is achieved when all mobile terminals use the maximum allowed number (i.e., M_k) of code channels to transmit packets. This phenomenon can be explained as follows. When mobile terminals use the maximum number of code channels, the number of different mobile terminals in a time slot is minimized. Considering subcode concatenation, mutual interference between code channels is also minimized. Thus, if a mobile terminal wants to transmit packets in a time slot, its transmission rate must be as high as allowed. This is called rate scheduling. According to this concept, a time slot with more available code channels is reserved for a mobile terminal with more packets waiting for transmission.

From (13), C_t^* must be achieved at M_{k^*} and γ_{k^*} with $M_{k^*}+\rho_{k^*} = \max\{M_k+\rho_k|k=1,\ldots,K\}$. If the maximum value is achieved at one extreme point, then the vector $[T_{1,1},\ldots,T_{k,l},\ldots,T_{K,M_K}] = [0,\ldots,0,T_{k^*,M_{k^*}},0,\ldots,0]$. This implies that service k^* is the only service that can be accommodated in the same time slot; otherwise, the maximum capacity cannot be achieved. In order to guarantee this, BER scheduling is necessary in the packet scheduler, i.e., packets with different BER requirements must be accommodated in different time slots. If the maximum value, C_t^*, is achieved at two or more extreme points, then the entire edge joining those points also achieves the maximum value. This means that the different services corresponding to those extreme points are equivalent when BER scheduling is performed.

In summary, in order to efficiently utilize the capacity of a time slot in a wide-band TD-CDMA system, both BER scheduling and rate scheduling are required.

V. Packet Scheduler

The packet scheduler consists of the packet prioritizer, the joint rate- and BER-scheduling scheme, and the resource controller based on minimum-power allocation, as shown in Fig. 3. The prioritizer determines which batch of packets is served first according to QoS requirements. After a batch is selected by the packet prioritizer, the joint rate- and BER-scheduling scheme allocates the appropriate time slots and code channels to accommodate packets of this batch. The capacity of each time slot is controlled by the resource controller. After scheduling is completed for the whole frame, the minimum-power allocation algorithm is used to allocate the received power level to each code channel in each time slot. The transmitted power level is then determined based on the measured path loss and the received power level.

A. Packet Prioritizer

In order to reduce packet delay and packet loss ratio, the priority of a batch of packets needs to be proportional to the number of remaining packets in the batch and inversely proportional to its remaining time before timeout. For ith batch of packets, its priority $\phi_b^{(i)}$ is simply calculated as

$$\phi_b^{(i)} = \frac{N_b}{T_d^{(i)} - T_c + T_{fr}} \tag{14}$$

where $T_d^{(i)}$ and T_c are the due time of the ith batch and the current time, respectively, N_b is the number of remaining packets in a batch, and T_{fr} is the length of a frame. The difference between $T_d^{(i)}$ and T_c captures the remaining time of the ith batch. Since packets are discarded if they cannot be transmitted before timeout, no packet is overdue by T_{fr}. Thus, $\phi_b^{(i)}$ is guaranteed to be a positive value. Based on the priority $\phi_b^{(i)}$, all batches are in an order from the highest priority to the lowest.

Given a batch with the highest priority (e.g., batch i), the prioritizer also needs to satisfy the requested transmission rate of the connection that owns batch i. If batch i belongs to nonreal-time service, only packets of this batch are served. However, if batch i belongs to a real-time connection, packets from other batches also need to be served until the requested transmission rate is satisfied. For a constant bit rate (CBR) connection, it will not be involved in the next iteration of packet prioritizing. For a variable bit rate (VBR) connection, it will still be involved in the next iteration of packet prioritizing, because only minimum transmission rate is satisfied in the current iteration.

B. Joint Rate- and BER-Scheduling

Suppose that batch i is the first being selected according to the packet prioritizer. The scheduling scheme needs to allocate appropriate time slots and code channels to the packets of this batch. In order to transmit as many packets as possible in a time slot, the joint rate- and BER-scheduling scheme is used to allocate code channels in different time slots to batch i. The available code channels in a time slot for batch i is determined according to the minimum-power allocation algorithm. After scheduling is completed, the received power level of each code channel is computed.

To simplify descriptions, the notation used in the scheduling scheme is defined in Table I. We also define several types of time slots as follows:

- *Empty Time Slot*. If no packets are accommodated in a time slot, the time slot is called an empty time slot.
- *BER_i Time Slot*. A time slot is called a BER_i time slot if the most stringent BER required by the packets in this time slot is BER_i.

TABLE I
NOTATION IN THE SCHEDULING SCHEME

t	Iteration number of the scheduling scheme.
BER_i	BER requirement of the service type of batch i.
$C_a^{(t)}$	Number of codes that can be used in a time slot by batch i in each iteration.
$m_i^{(t)}$	Number of code channels used by the mobile terminal owning batch i in a time slot before each iteration.
$J_a^{(t)}$	Number of packets accommodated in each iteration.
$J_i^{(t)}$	Number of packets in batch i in each iteration.

- *Nonavailable Time Slot*. A nonempty time slot is called a nonavailable time slot for a mobile terminal if the mobile terminal has used the maximum allowed number of code channels in this time slot. According to this definition, a nonavailable time slot is not available to a specific mobile terminal instead of to all mobile terminals.

The joint rate- and BER-scheduling scheme works iteratively until all packets in batch i are accommodated or no code channels are available for batch i in the current frame. If no code channel is available and there are remaining packets in batch i, these packets are served in the next frame. However, if timeout occurs in batch i in the next frame, the remaining packets need to be discarded. As shown in Fig. 4, in each iteration (e.g. iteration t) of packet accommodation with code channels in different time slots, there are four possible cases depending on the availability of the required number of code channels in a specific time slot.

- *Case 1: An empty or nonempty BER_i time slot is found and $J_i^{(t)} \geq C_a^{(t)}$ in this time slot.* $C_a^{(t)}$ packets in batch i are accommodated in the time slot. In such a way, the mobile terminal that owns batch i uses as many code channels in a time slot as possible. How to calculate $C_a^{(t)}$ will be explained in Section V-C. After this step, $J_a^{(t)} \leftarrow C_a^{(t)}$, i.e., all code channels left for batch i are used, and thus, the corresponding time slot becomes a nonavailable time slot to the mobile terminal that owns batch i. In this case, both rate and BER scheduling is performed.
- *Case 2: Empty or nonempty BER_i time slots are found, but none of them satisfies $J_i^{(t)} \geq C_a^{(t)}$.* $J_i^{(t)}$ packets must be accommodated in the empty or nonempty BER_i time slot in which $C_a^{(t)}$ is the smallest among all empty and nonempty BER_i time slots. Thus, time slots with larger $C_a^{(t)}$ are reserved for batches with more packets, which is the objective of rate scheduling. After this step is successfully executed, $J_a^{(t)} \leftarrow J_i^{(t)}$, i.e., no packet is left in batch i.
- *Case 3: Neither an empty nor a nonempty BER_i time slot is found but $J_i^{(t)} \geq C_a^{(t)}$ is satisfied in other types of time slots.* $C_a^{(t)}$ packets of batch i are accommodated in the time slot in which $C_a^{(t)}$ is the smallest, i.e., $J_a^{(t)} \leftarrow C_a^{(t)}$. In addition, this time slot becomes a nonavailable time slot to the mobile terminal that owns batch i. Only rate scheduling is performed in this case.
- *Case 4: No time slot satisfies $J_i^{(t)} \geq C_a^{(t)}$.* $J_i^{(t)}$ packets are accommodated in the time slot in which $C_a^{(t)}$ is the smallest. Thus, $J_a^{(t)} \leftarrow J_i^{(t)}$.

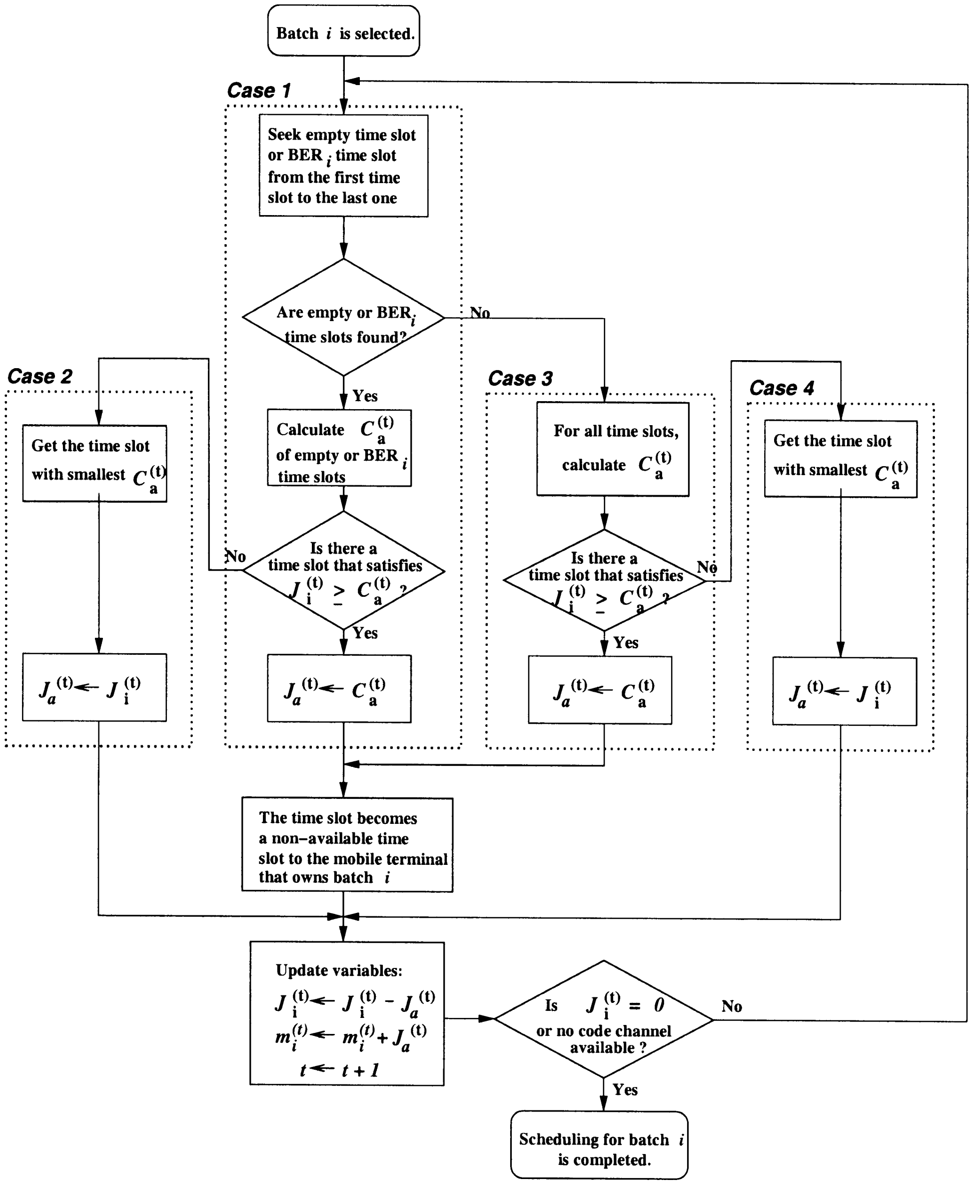

Fig. 4. Joint rate- and BER-scheduling scheme.

When the tth iteration is completed, $J_a^{(t)}$ packets are removed from batch i, i.e., $J_i^{(t)} \leftarrow J_i^{(t)} - J_a^{(t)}$. For the time slot where $J_a^{(t)}$ packets of batch i are accommodated, $m_i^{(t)} \leftarrow m_i^{(t)} + J_a^{(t)}$.

C. Resource Controller

In each iteration of the scheduling scheme, $C_a^{(t)}$ is determined as follows. According to the minimum-power allocation algorithm, the set of used code channels $\{m_{k,n} | n = 1, \ldots, N_k^{(t)}, k = 1, \ldots, K\}$ and $C_a^{(t)}$ must satisfy the constraint in (8), i.e.,

$$\sum_{k=1}^{K} \sum_{n=1}^{N_k^{(t)}} W_{k,n} - \frac{m_i^{(t)}}{m_i^{(t)} + \rho_i} + \frac{C_a^{(t)} + m_i^{(t)}}{C_a^{(t)} + m_i^{(t)} + \rho_i} \leq 1 - \Delta. \tag{15}$$

Thus, the maximum allowable value of $C_a^{(t)}$, denoted by $C_i^{(t)}$, is

$$C_i^{(t)} = \frac{\rho_i}{\sum_{k=1}^{K} \sum_{n=1}^{N_k^{(t)}} W_{k,n} - \dfrac{m_i^{(t)}}{m_i^{(t)} + \rho_i} + \Delta} - \rho_i - m_i^{(t)}. \tag{16}$$

In addition, for the mobile terminal that owns batch i, the number of code channels is limited to M_i, i.e., $C_a^{(t)}$ must satisfy $m_i^{(t)} + C_a^{(t)} \leq M_i$. As a consequence, $C_a^{(t)}$ is given by

$$C_a^{(t)} = \min \left\{ C_i^{(t)}, M_i - m_i^{(t)} \right\}. \tag{17}$$

VI. New CAC Algorithm

To enhance the QoS of multimedia traffic, CAC is required. However, the bursty traffic characteristic of a connection causes its bandwidth requirement unpredictable at connection admission time. One widely accepted approach to this problem is to use effective bandwidth-based algorithms [7]. In this section, a new effective bandwidth-based CAC algorithm is developed considering the minimum-power allocation scheme derived in Section III.

Considering a time slot in the uplink frame, we assume that N_k mobile terminals belong to service type k. Suppose mobile terminal n of service type k uses $m_{k,n}$ code channels. According to (8), the normalized capacity consumed by mobile terminals of all service types must be less than $1-\Delta$. During the lifetime of a connection of mobile terminal n of service type k, $W_{k,n}$ varies frame-by-frame. Thus, $W_{k,n}$ is a random variable when it is observed at the connection level. The satisfaction of (8) needs to be evaluated stochastically. Define a satisfaction factor α as

$$P\left(\sum_{k=1}^{K}\sum_{n=1}^{N_k} W_{k,n} < 1-\Delta\right) \geq \alpha \tag{18}$$

and $0 < \alpha \leq 1$. From (18), the larger α is, the higher the probability that (18) is satisfied. Given a satisfaction factor α, an admissible region can be determined based on (18). In what follows, Gaussian approximation is used to derive the admissible region.

For the same service type k, the traffic of different mobile terminals is independent and follows the same characteristics. Denote the mean and variance of $W_{k,n}$ as μ_k and σ_k^2, respectively. Also, the probability that a mobile terminal of service type k has m code channels is denoted by $p_{k,m}$ (estimation of its value is discussed in Section VII-C2). Thus, $W_{k,n}$ is a random variable with probability mass function (PMF)

$$P\left(W_{k,n} = \frac{m}{m+\rho_k}\right) = p_{k,m} \; for \; m = 0,\ldots,M_k \tag{19}$$

where M_k is the maximum allowed number of code channels for a mobile terminal of service type k. As a consequence, the mean μ_k and the variance σ_k^2 of $W_{k,n}$ are

$$\mu_k = \sum_{m=1}^{M_k} p_{k,m}\frac{m}{m+\rho_k} \tag{20}$$

and

$$\sigma_k^2 = \sum_{m=1}^{M_k} p_{k,m}\left(\frac{m}{m+\rho_k}\right)^2 - \mu_k^2 \tag{21}$$

respectively. According to the central limit theorem, when N_k is large, $\sum_{n=1}^{N_k} W_{k,n}$ can be approximated by a Gaussian random variable $\mathcal{G}_k$ with mean and variance equal to $N_k\mu_k$ and $N_k\sigma_k^2$, respectively. Since $\{\mathcal{G}_k | k = 1,\ldots,K\}$ are independent for different service types, $\sum_{k=1}^{K}\mathcal{G}_k$ can also be approximated by a Gaussian random variable $\mathcal{G}$ with mean and variance equal to $\sum_{k=1}^{K} N_k\mu_k$ and $\sum_{k=1}^{K} N_k\sigma_k^2$, respectively. Therefore, $\sum_{k=1}^{K}\sum_{n=1}^{N_k} W_{k,n}$ can be approximated by a Gaussian random variable $\mathcal{G}$. Consequently, (18) becomes

$$P(\mathcal{G} \leq 1-\Delta) \geq \alpha. \tag{22}$$

According to the characteristics of Gaussian random variables, the constraint in (22) is satisfied if and only if

$$\frac{1-\Delta-E[\mathcal{G}]}{\sqrt{Var[\mathcal{G}]}} \geq \beta \tag{23}$$

where $E[\mathcal{G}]$ and $Var[\mathcal{G}]$ are the mean and the variance of $\mathcal{G}$, respectively, and β is defined by

$$\frac{1}{\sqrt{2\pi}}\int_{\beta}^{\infty} e^{-\frac{t^2}{2}}dt = 1-\alpha. \tag{24}$$

With $E[\mathcal{G}] = \sum_{k=1}^{K} N_k\mu_k$ and $Var[\mathcal{G}] = \sum_{k=1}^{K} N_k\sigma_k^2$ and after some algebra, the constraint in (23) becomes

$$\sum_{k=1}^{K} N_k\mu_k + \beta\sqrt{\sum_{k=1}^{K} N_k\sigma_k^2} \leq 1-\Delta. \tag{25}$$

For a state $(N_1, N_2, \ldots, N_K)$, if (25) is satisfied, it is called an admissible state. If a new connection lies in the admissible state, then it can be admitted.

VII. Performance Evaluation

A. Simulation Models

Six traffic models of multimedia traffic are developed.

- *Voice Traffic*: The same model as in [10] is used. The average durations of principal talkspurts and gaps are 1.000 and 1.350 s, respectively. In the talkspurts, the average durations of minispurts and minigaps are 0.235 and 0.050 s, respectively. The data rate in minispurts is 16.5 kb/s. The average length of the conversation is assumed to be 180.0 s.
- *Audio Traffic*: The constant bit rate is 32 kb/s and the holding time of each audio stream is assumed to be exponentially distributed, with a mean equal to 180.0 s.
- *CBR Video Traffic*: This model is actually a constant bit stream with the rate equal to 64 kb/s. For different CBR video users, the transmission time is assumed to be exponentially distributed, with a mean equal to 360.0 s.
- *VBR Video Traffic*: The model consists of various states [10]. Each state holds a certain duration which is exponentially distributed, with a mean equal to 160 ms. The data rate in each state is constant and obtained from a truncated exponential distribution with the maximum and minimum bit rates equal to 16 and 64 kb/s, respectively. The mean video transmission time is assumed to be 180.0 s.
- *Remote Login Traffic*: This model is used to simulate a nonreal-time low-delay service. The length of each message is geometrically distributed with the average equal to two MAC packets, and the message generation rate is a Poisson process with average equal to 0.2 message/frame. During the lifetime of a remote login session, the overall length of all messages is exponentially distributed with the mean equal to 30 kB.
- *E-mail Traffic*: An empirical distribution is used to generate e-mail traffic [10].

We consider a seven-cell model in a wide-band TD-CDMA wireless networks. However, in this section, we give the performance results only for the center cell. The effect of other cells to the center cell is taken into account through inter-cell interference I_{inter}. The radius of a cell is assumed to be 30 m. To simulate the attenuation, fading, and shadowing in a wireless channel, we use the same path loss model as that in [23]. The

TABLE II
PARAMETERS OF DIFFERENT TRAFFIC TYPES

Service Type	BER	SINR	t_{out}	M	r_b	p_c	Ns
Voice	10^{-3}	5.31	2	1	12.2	65	8
Audio	10^{-4}	7.31	6	3	12.2	5	5
CBR Video	10^{-5}	9.32	5	6	12.2	8	4
VBR Video	10^{-6}	11.34	4	5	12.2	12	3
Remote Login	10^{-9}	8.00	100	1	8.1	5	5
Email	0	2.94	∞	1	12.2	5	13

TABLE III
PARAMETERS OF WIDE-BAND TD-CDMA SYSTEM

t_{fr}	N_{fr}	U_{du}	W	W_c	G
10 ms	15	1.5	5.0 MHz	3.84 Mcps	16.0

thermal noise is simulated by using the Gaussian process. Based on I_{inter}, thermal noise, path loss, and the maximum power range, the value of Δ can be determined.

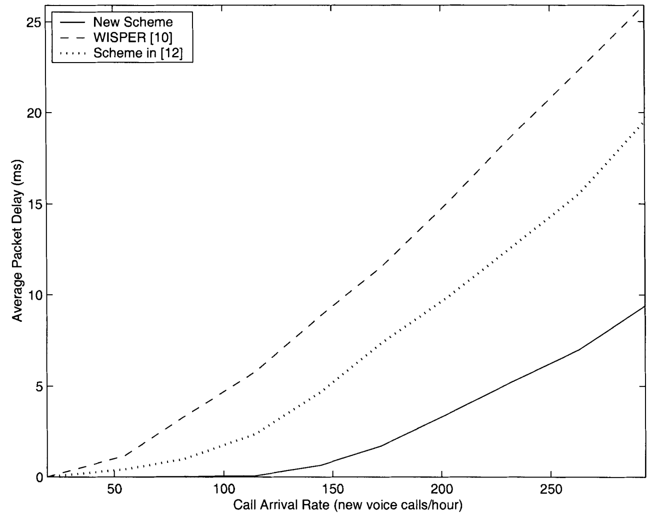

Fig. 5. Packet delay of VBR video traffic (no CAC).

B. System Parameters

The parameters related to various traffic types consist of BER requirements, target SINR in decibels, basic transmission rate of a code channel in units of kilobits per second (r_b), the maximum allowable number of code channels for a mobile terminal in a time slot (M), timeout value in unit of frames (t_{out}), and the call percentage of a traffic type (p_c). These are specified in Table II. The target SINR values are determined by considering channel fading, error control schemes, modulation schemes, and BER requirements (see Appendix B). We determine r_b of different traffic types according to the number of bits in a MAC packet and the coding rate of the error control scheme. Since the error control scheme for remote login traffic has higher redundancy, r_b of remote login is lower than those of other services. p_c in Table II is a simulation parameter to control the traffic composition of different services. Its absolute value varies depending on individual simulation points.

Parameters of the wide-band CDMA system include processing gain (G), frame length (t_{fr}), the number of time slots in a frame (N_{fr}), downlink/uplink asymmetry ratio (U_{du}), system bandwidth (W), and chip rate of a spread signal (W_c). These parameters are shown in Table III. U_{du} is assumed to be fixed, but the locations of switching points are assumed to be randomly distributed in a frame.

The average packet loss ratio r_l, the average packet delay d_p, the throughput t_r, and the connection blocking probability b_p are the metrics used to evaluate the performance of the protocol. Given a traffic type, r_l is computed from $r_l = (N_l/(N_l + N_t))$, where N_l is the number of lost packets due to timeout, and N_t is the number of packet transmitted successfully. The average packet delay d_p consists of two components, i.e., $d_p = d_r + d_t$, where d_r is the time being spent on allocating code channels to mobile terminals, and d_t is the transmission time of a packet after time slots are allocated. Since the location of uplink/downlink switching point is assumed to randomly distributed in a frame, d_t is approximately half of the frame length. The throughput t_r is defined to be the total number of packets being successfully transmitted in a frame. When CAC is used, some connections of real-time traffic will be blocked. If C_b and C_a denote the number of connections being blocked and being accepted, respectively, then $b_p = (C_b/(C_b + C_a))$.

C. Numerical Results

We simulated the proposed MAC protocol with all new algorithms for a TD-CDMA wireless system. Based on the system input parameters and the traffic models, we obtained several simulation results, which show that the average packet delay, the average packet loss ratio, and the call blocking probability have a similar behavior for real-time services. In order to avoid repetitive figures for these cases, we only illustrate the results for VBR video traffic. Moreover, since e-mail and remote login traffic types have similar average packet delays, we show only the average packet delay of remote login traffic.

1) Experiments Without CAC: In this set of experiments, we do not incorporate the new CAC algorithm so that we can compare our new MAC protocol with the existing ones that do not have CAC algorithms either [10], [12]. However, we consider all other new algorithms of the MAC protocol in the simulations. The inter-cell interference level, represented by Δ, is controlled at 0.005. Our new MAC protocol is compared with WISPER protocol [10] and the resource allocation algorithm proposed in [12]. For the WISPER protocol, the BER scheduling is applied to wide-band TD-CDMA networks. According to the target SINR values given in Table II, the maximum number of code channels that can be supported per time slot for each service, denoted by Ns, is determined by employing the same method used in [10]. The result is shown in Table II. In the scheme [12] for wide-band TD-CDMA networks, the ratio of the power level of service i to that of service j is $P_i/P_j = (Ns_j - 1)/(Ns_i - 1)$. Here Ns_i and Ns_j are the maximum code channels that can be supported in a time slot for services i and j, respectively. Their values are the same as Ns given in Table II. Other parameters used by WISPER and the scheme in [12] are the same as those described in Section VII-B.

The average packet delays of VBR video traffic and remote login traffic are shown in Figs. 5 and 6, respectively. During the operation region, the average packet delays of voice, audio,

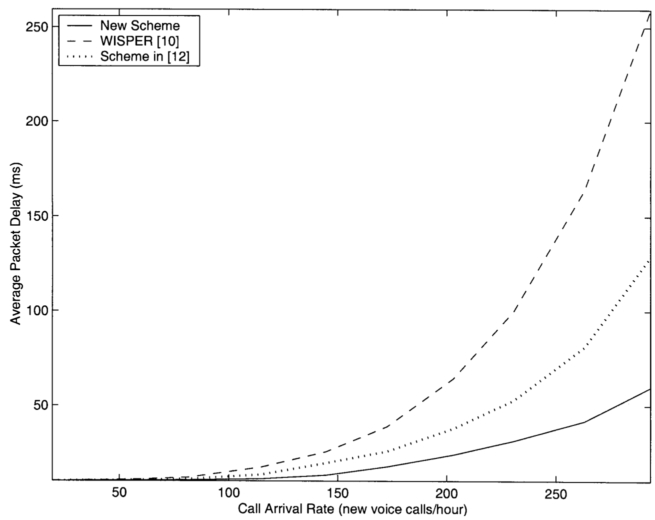

Fig. 6. Packet delay of remote login traffic (no CAC).

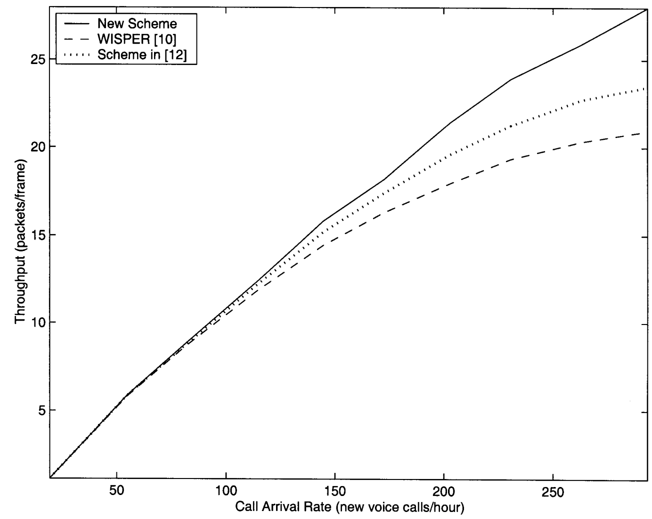

Fig. 8. System throughput (no CAC).

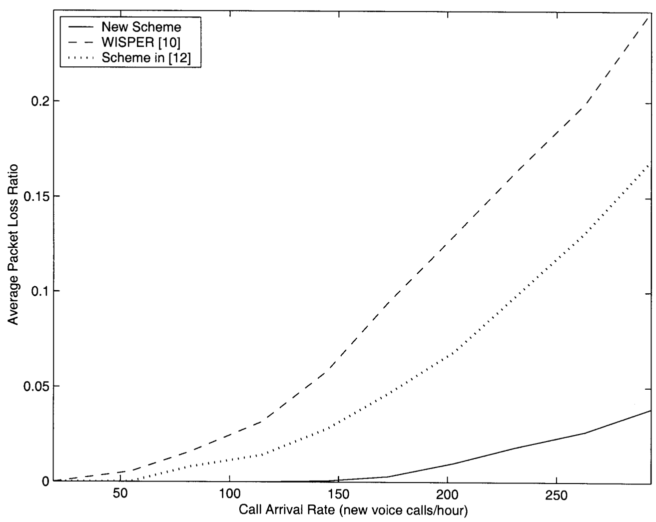

Fig. 7. Packet loss ratio of VBR video traffic (no CAC).

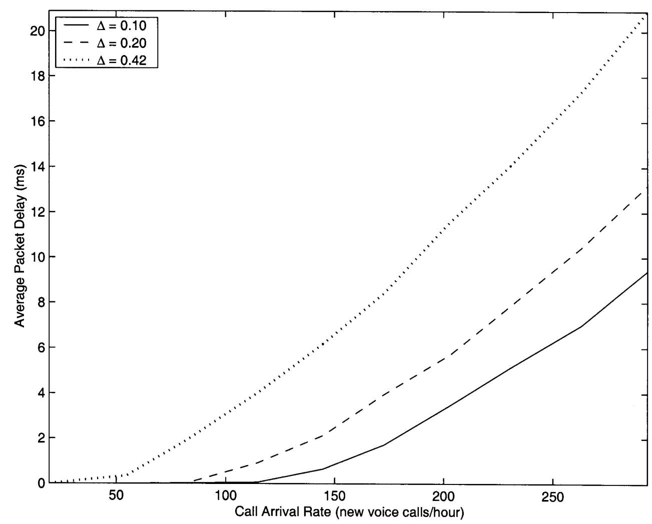

Fig. 9. Packet delay of VBR video traffic.

CBR traffic have a similar trend as the VBR video traffic, with the maximum value for our new scheme equal to 2.3, 12.3, and 11 ms, respectively. Also, the e-mail traffic has a similar trend as the remote login traffic. The difference is that the delay of e-mail traffic is two magnitudes larger than that of remote login traffic. For our proposed scheme, the maximum delay is 9.0×10^3 ms. The average packet loss ratio of VBR video traffic is illustrated in Fig. 7. The average packet loss ratios of voice, audio, and CBR traffic have a similar trend, and the maximum value for our proposed scheme is 0.33, 0.25, and 0.05, respectively. Due to large timeout values in e-mail and remote login traffic, no packet loss is assumed to occur in nonreal-time traffic. The throughput of the system is shown in Fig. 8.

Comparisons of output parameters between different schemes show that the proposed MAC protocol significantly outperforms both WISPER and the scheme in [12]. The reason is that the minimum-power allocation and the joint rate- and BER-scheduling reduce capacity consumption in each time slot. It is also shown that the scheme proposed in [12] outperforms WISPER, because different power levels are used for different services in [12]. However, due to lack of minimum-power allocation and an effective scheduling scheme, its performance is much lower than that of the new MAC protocol.

The effect of inter-cell interference to the performance of the MAC protocol is also investigated in this set of experiments. The inter-cell interference is mainly varied by the number of floors in the path loss model. With respect to different inter-cell interference, represented by the value of Δ, the average packet delay, the packet loss ratio, and the system throughput are illustrated in Figs. 9–12. Again, here we show only the results for the VBR video traffic.

During the operation region, the maximum packet delays of voice, audio, and CBR video traffic for $\Delta = 0.20$ are 6.9, 21.1, and 17.6 ms, respectively. The maximum packet loss ratios of voice, audio, and CBR video traffic are 0.35, 0.45, and 0.21, respectively, for $\Delta = 0.20$. For nonreal-time services, e-mail traffic has the similar trend in average packet delay as remote login traffic. When $\Delta = 0.20$, its average packet delay is two magnitudes larger with the maximum value equal to 3.9×10^4 ms.

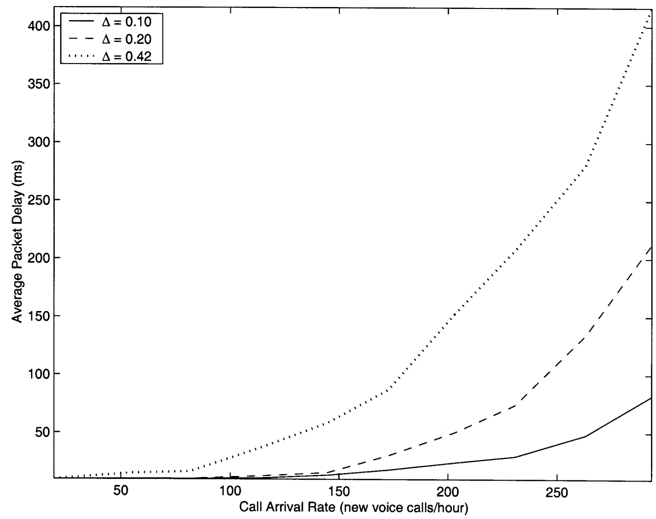

Fig. 10. Packet delay of remote login traffic.

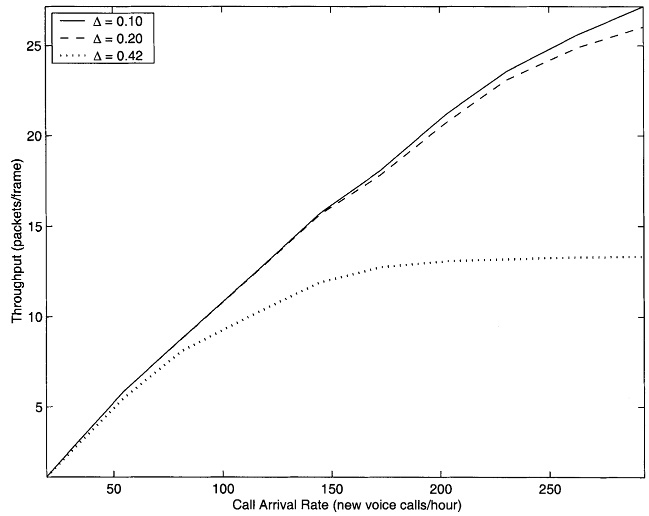

Fig. 12. System throughput.

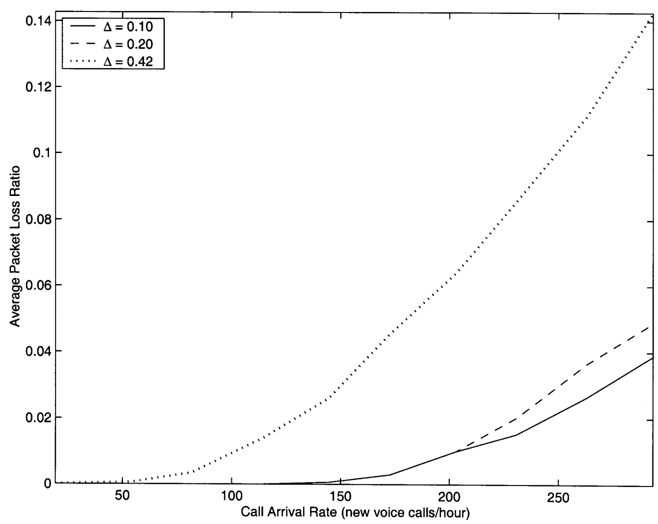

Fig. 11. Packet loss ratio of VBR video traffic.

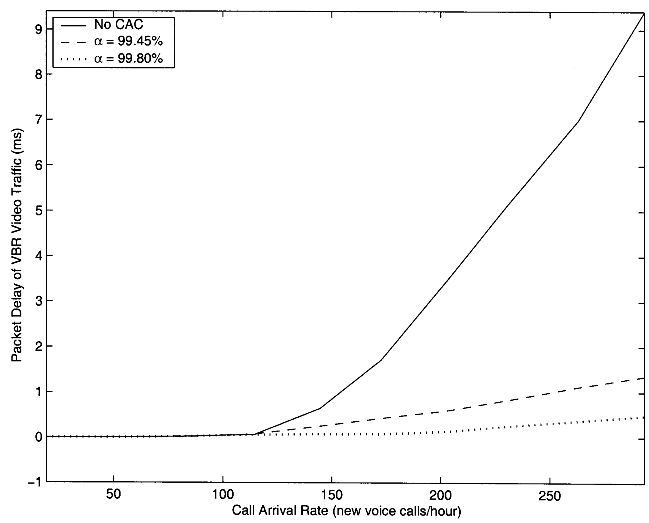

Fig. 13. Packet delay of VBR video traffic (with CAC).

Comparing these results with those in Figs. 5–8 where $\Delta = 0.005$, we find that that the performance degradation of the MAC protocol is negligible when Δ is less than 0.10. In this situation, the performance of the new MAC protocol proposed in this paper does not rely on an inter-cell interference canceler. However, when inter-cell interference is severe (e.g., $\Delta = 0.42$), the performance of the MAC is significantly degraded. As a consequence, a high-quality inter-cell interference canceler is needed.

2) Experiments With CAC: At run-time of the CAC algorithm, the probability $p_{k,m}$ needs to be determined in order to calculate μ_k and σ_k^2. Such a parameter is determined by the distribution of code channels in a frame. However, for any connection, such a distribution is unknown, because it depends on the scheduling scheme and traffic characteristics. To determine a parameter that depends on a random variable with unknown distribution, on-line estimation can be used. In particular, when the relationship between the scheduling scheme and the random variable can be established through a formula, a stochastic approximation algorithm can estimate the parameter [24]. Unfortunately, it is nontrivial to find such a formula in the joint rate- and BER-scheduling scheme. However, in all frames before a new connection is requested, the results of code channel allocation are known. Suppose we observe the code channel allocation results for L frames before a new connection request arrives, the probability $p_{k,m}$ can be approximately calculated according to the statistics of code channel allocation results of all connections of each service type. This estimated probability is updated on-line as the L-frame observation window slides. This method can capture the feature of the scheduling scheme and traffic characteristics. Simulations results will show that it has satisfactory accuracy.

In this set of experiments, the inter-cell interference level is $\Delta = 0.005$. For various values of the satisfaction factor α, the effect of the CAC algorithm to the performance of the MAC protocol is illustrated in Figs. 13–17. For real-time traffic, the average packet delay and packet loss ratio of voice, audio, and CBR video have similar trends as those of VBR video. When $\alpha = 99.45\%$, the maximum delay of voice, audio, and CBR traffic is 1.9, 5.7, and 1.8 ms, respectively, and the maximum

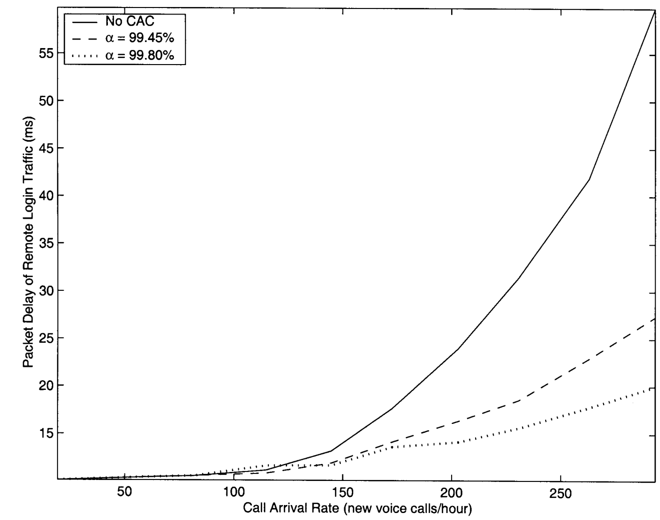

Fig. 14. Packet delay of remote login traffic (with CAC).

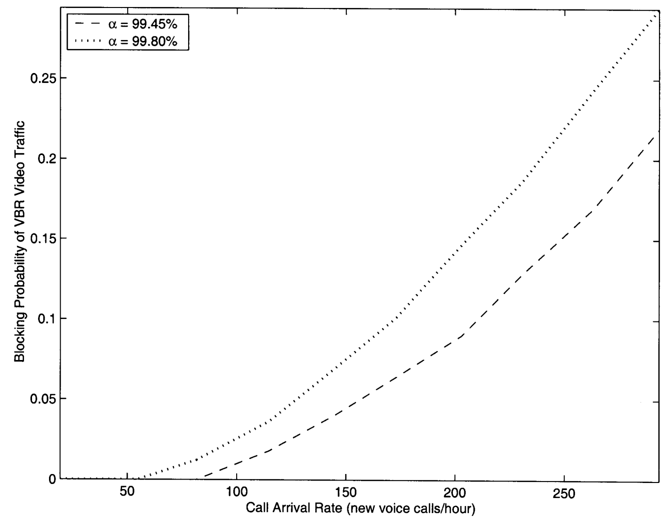

Fig. 16. Connection blocking probability.

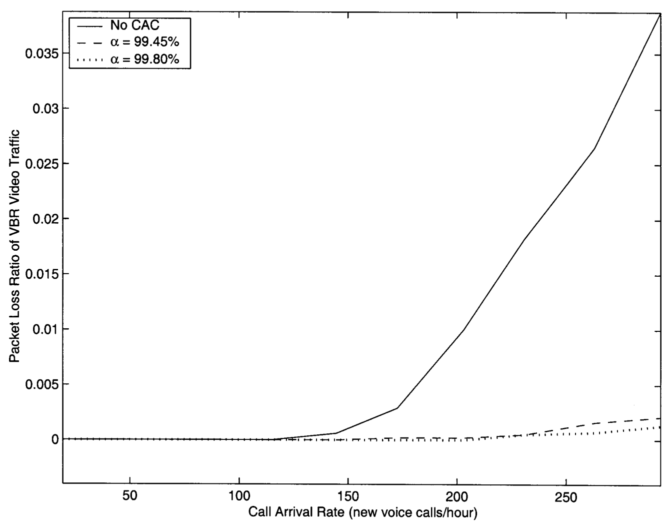

Fig. 15. Packet loss ratio of VBR video traffic (with CAC).

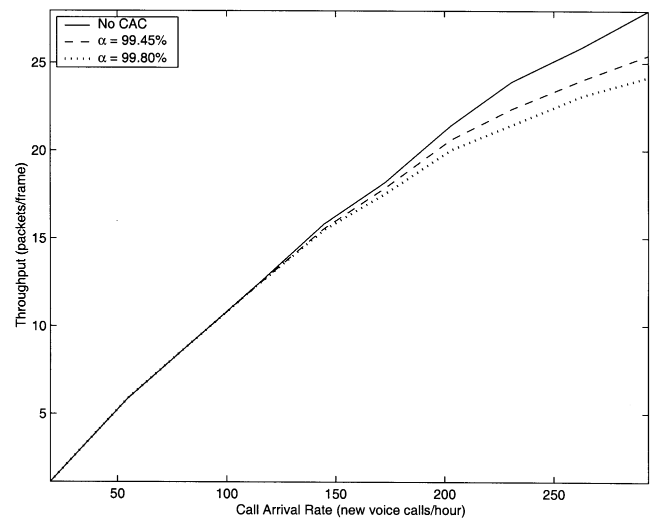

Fig. 17. System throughput (with CAC).

packet loss ratio of these three traffic types is 0.12, 5.3×10^{-2}, and 8.5×10^{-3}, respectively. For nonreal-time traffic, the average packet delay of e-mail traffic has a similar trend as that of remote login traffic, but it is two magnitudes larger. When $\alpha = 99.45\%$, the maximum delay within the operation region is 3.7×10^{3} ms.

With the help of CAC, the performance of the MAC protocol is greatly enhanced, as shown in Fig. 15. When traffic load is high, the packet loss ratio of VBR video traffic is less than 0.5%. Those of audio and voice traffic are also reduced by a large percentage. Moreover, the average packet delays of both real-time and nonreal-time traffic are significantly reduced, which is illustrated in Figs. 13 and 14.

Simulation also shows that the connection blocking probability of video and audio traffic is much higher than that of voice traffic, because video and audio calls consume much higher bandwidth than voice calls. As α increases, CAC becomes more conservative, and thus, more connections are blocked under the same traffic load. Thus, lower average packet delay and packet loss ratio are achieved.

As illustrated in Fig. 17, the system throughput is reduced when the CAC algorithm is used. However, compared to the significant improvement in packet loss ratios and packet delays of different traffic types, such reduction is rather small, even when the satisfaction factor is equal to 99.80%. This result also illustrates a satisfactory accuracy of the estimation method for $p_{k,m}$. Otherwise, either too many connections will be blocked, which causes significant throughput reduction, or the packet delays and packet loss ratios cannot be significantly improved.

VIII. Conclusion

In this paper, we derived the minimum-power allocation algorithm for mobile terminals that transmit multimedia traffic in wide-band TD-CDMA networks. We also proved that joint rate- and BER-scheduling is required in order to reach the maximum capacity of a time slot. With joint rate- and BER-scheduling and minimum-power allocation, we proposed a new packet scheduler for the MAC protocol of TD-CDMA wireless system. Furthermore, to enhance the performance of the MAC protocol, we developed a new CAC algorithm based on minimum-power allocation. The new MAC protocol satisfies QoS requirements of multimedia traffic and increases the system throughput. It is a promising protocol for wide-band TD-CDMA wireless multimedia networks.

Appendix A

Putting (1) into (4) gives

$$\sum_{k=1}^{K}\sum_{n=1}^{N_k} m_{k,n}S_{k,n} - \left(m_{i,j} + \frac{1.5G}{\gamma_i}\right)S_{i,j} = -(1.5GN_0B_b + I_{\text{inter}}) \tag{26}$$

where $B_b = B_s/G$. (26) is derived for the mobile terminal j of service type i. Considering mobile terminal n of service type k, a new equation similar to (26) can be derived, except that the second term on the left side of (26) is replaced with $(m_{k,n} + 1.5G/\gamma_k)S_{k,n}$ in the new equation. Therefore, comparing mobile terminals j and n, the following condition must hold:

$$\left(m_{i,j} + \frac{1.5G}{\gamma_i}\right)S_{i,j} = \left(m_{k,n} + \frac{1.5G}{\gamma_k}\right)S_{k,n} \tag{27}$$

i.e.,

$$S_{k,n} = \frac{m_{i,j} + \frac{1.5G}{\gamma_i}}{m_{k,n} + \frac{1.5G}{\gamma_k}}S_{i,j}. \tag{28}$$

Substituting (28) into (26) and after some algebra, we have

$$S_{i,j} = \frac{1.5GN_0B_b + I_{\text{inter}}}{\left(1 - \sum_{k=1}^{K}\sum_{n=1}^{N_k}\frac{m_{k,n}}{m_{k,n} + \frac{1.5G}{\gamma_k}}\right)\left(m_{i,j} + \frac{1.5G}{\gamma_i}\right)}$$

which is (5).

Appendix B

Given the BER requirement of a service, the target SINR is related to channel fading, modulation, and error-control schemes. In this paper, Rayleigh fading is considered, and the modulation scheme is QPSK. According to 3GPP specifications [17], three options of channel coding are available for DCHs: convolutional coding (rate 1/2 or 1/3), 1/3 Turbo coding, and no coding. In this paper, the channel coding schemes for different services are chosen as follows:

- *Real-time service*. 1/2 convolutional coding is used. Its constraint length is 3, and the free distance d_{free} is 5. Short constraint length and free distance are chosen to increase the coding speed for real-time service. Voice, audio, and video belong to this service type.
- *Nonreal-time delay-tolerant service*. 1/2 convolutional coding is used to achieve 10^{-2} of block error rate (BLER). Selective-repeat ARQ is used to recover the residual errors. The constraint length of convolutional coding is 9 and the free distance d_{free} is 12. A typical example of this service type is e-mail.
- *Nonreal-time low-delay service*. 1/3 Turbo coding is used so that the BER is less than 10^{-9}. It consists of parallel concatenated convolutional code (PCCC) with eight-state constituent encoder and one Turbo code internal interleaver is used. ARQ is not used in order to reduce the transmission delay. Remote login is an example of this service type.

Considering BPSK modulation and Rayleigh fading for convolution coding, the BER P_b at the decoder output of convolutional coding has been derived in [25]. Since QPSK has the same bit-error probability as that of BPSK, the result in [25] can be applied here. Thus

$$P_b \le \frac{1}{2}\sum_{d=d_{\text{free}}}^{\infty} b_d\left(\frac{1}{1+\gamma_0}\right)^d \tag{29}$$

where b_d is the number of nonzero information bits on all weight-d paths on the trellis code tree of convolutional code, and γ_0 is the required average SINR. The values of b_d are studied in [26]. Therefore, for real-time service, given a BER P_b, the average SINR γ_0 can be calculated from (29). For nonreal-time delay-tolerant service, the required BER can be derived from BLER. Given a block size of L bits, the relationship between BLER P_{bler} and BER P_b is $P_{\text{bler}} = \sum_{i=1}^{L}\binom{L}{i}P_b^i(1-P_b)^{L-i}$. Suppose $L = 228$ bits as specified for low-rate (3.84 kb/s) packet data service in [22], then then $P_b = 0.000\,45$ if $P_{\text{bler}} = 10^{-2}$. Thus, based on (29), the required SINR γ of nonreal-time low-delay service is 2.94 dB. For nonreal-time low-delay service, when Rayleigh fading is considered, the required SINR can be obtained from [27], and it is 8 dB if BER of less than 10^{-9} is required at the decoder output of 1/3 Turbo coding. Having the above choices of coding schemes, the SINR-BER relationship of different service types are given in Table II.

Acknowledgment

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