An FDD Wideband CDMA MAC Protocol for Wireless Multimedia Networks

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Abstract—**A medium access control (MAC) protocol is developed for wireless multimedia networks based on frequency division duplex (FDD) wideband code division multiple access (CDMA). In this protocol, the received power levels of simultaneously transmitting users are controlled by a minimum-power allocation algorithm such that the heterogeneous bit error rates (BERs) of multimedia traffic are guaranteed. With minimum-power allocation, a multimedia wideband CDMA generalized processor sharing (GPS) scheduling scheme is proposed. It provides fair queueing to multimedia traffic with different QoS constraints. It also takes into account the limited number of code channels for each user and the variable system capacity due to interference experienced by users in a CDMA network. The admission of real-time connections is determined by a new effective bandwidth connection admission control (CAC) algorithm in which the minimum-power allocation is also considered. Simulation results show that the new MAC protocol guarantees QoS requirements of both real-time and non-real-time traffic in an FDD wideband CDMA network.**

I. Introduction

To date, multiple radio transmission technologies have been proposed for the next generation wireless networks. Some of them are still under investigation, while others have been selected as a candidate technology for the next generation wireless networks. A typical example is wideband code-division multiple-access (CDMA). It has been chosen as the basic access technology for UMTS/IMT2000 [1]. Compared to the narrow-band CDMA, wideband CDMA can support services with much higher rate. It is also flexible to deliver multimedia traffic. However, in order to fully make use of such improvements, a new medium access control protocol (MAC) is required to efficiently manage packet access in wideband CDMA wireless networks. Wideband CDMA can be categorized into pure wideband CDMA and wideband time-division (TD) CDMA. Pure wideband CDMA uses frequency division duplex (FDD) to organize the uplink and downlink transmissions, while wideband TD-CDMA uses time division duplex (TDD). TDD mode is well-suited for indoor environments with high traffic density and applications with highly asymmetric traffic [2], while FDD mode is advantageous for applications in public macro- and micro-cell environments. In terms of resource allocation, more differences exist between FDD mode pure wideband CDMA and wideband TD-CDMA. In wideband TD-CDMA, resource units include both time slots in a frame and codes in a time slot due to slotted frame structure. Multi-code (MC) operation and multi-slot operation are normally used to allocate resources [3]. However, FDD mode wideband CDMA does not have the option of multi-slot operation because the resource units only include codes in a frame. In order to enhance the flexibility of resource allocation in FDD mode wideband CDMA, orthogonal-variable-spreading-factor (OVSF) is used [3]. A mobile terminal can have multiple codes and the OVSF of each code can be variable. Thus, both MC operation and OVSF operation are used in FDD mode wideband CDMA. This paper is focused on the MAC protocol for FDD wideband CDMA wireless multimedia networks.

Several MAC protocols have been proposed for FDD wideband CDMA networks. In [4], a proposal for an RLC/MAC protocol for wideband CDMA is presented. This proposal does not consider how to allocate resources to different services. An uplink MC CDMA system architecture is proposed to support heterogeneous traffic with diverse QoS requirements [5]. The received power level of each code channel is controlled by a power allocation algorithm so that BERs of all services are guaranteed. Moreover, the scheduling scheme for non-real-time traffic is based on first-in first-out (FIFO) queueing and round-robin queueing. The average message transmission delay of non-real-time traffic can be very large when a lower priority is assigned. Another MAC protocol for FDD wideband CDMA is proposed in [6], where the power allocation algorithm and the token bucket traffic regulator are not designed interactively. In addition, the power allocation algorithm does not consider both MC and OVSF operation. For UMTS/IMT-2000 based on wideband CDMA, the performance of a multiple access protocol for integration of variable bit rate multimedia traffic is analyzed in [7]. However, a PRMA-like MAC protocol is assumed for wideband CDMA networks. The 3rd Generation Partnership Project (3GPP) standardization committee has also released a specification on MAC protocol [8]. In this specification, the MAC architecture, channel structures, services, and MAC functions are defined.

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However, no specific access scheme is specified for the MAC protocol. Thus, the design of the MAC protocol for wideband CDMA is still an open problem. Recently, an access scheme is developed in [9] for the voice/data traffic transmissions over common packet channel (CPCH) in a wideband CDMA network. However, it does not include an access scheme for dedicated channels (DCHs).

In this paper, a new MAC protocol is proposed for multimedia traffic transmission in uplink DCHs of wideband CDMA wireless networks. Similar but simpler procedures can be applied to multimedia traffic transmission in downlink DCHs, thanks to the broadcast nature of downlink. In the new MAC protocol, a minimum-power allocation algorithm is first derived to determine the minimum received power level for each code channel when a target signal-to-interference-plus-noise-ratio (SINR) is given. In this algorithm, both MC and OVSF transmissions is considered. Also, channel fading and noise are taken into account to establish the relationship between the required BER of a service type and the SINR at the receiver. The minimum received power levels are assumed to be maintained through open- or close-loop power control algorithms [10], [11]. How to design a power control algorithm is out of the scope of this paper. Based on the minimum-power allocation algorithm, a wideband CDMA generalized processor sharing (GPS) scheduling scheme is proposed. This GPS scheduling scheme is novel and different from other GPS scheduling schemes [12], [13], because all the following features are considered:

1) The system capacity considered in the scheduler varies with respect to the interference level experienced by CDMA users.
2) Both MC-CDMA and VSF-CDMA are considered in the scheduling scheme.
3) Both power constraint of code channels and the limited number of code channels for each user are considered in the scheduling scheme.

The GPS scheduling scheme provides fair queueing to multimedia traffic with heterogeneous QoS requirements. It also guarantees the diverse BER values of multimedia traffic.

In order to admit real-time connections, a CAC scheme is needed. One of the challenging issues of a CAC algorithm is that the bandwidth requirement of a connection cannot be determined at connection admission time due to the traffic burst. Thus, effective bandwidth-based CAC schemes are generally used to solve this problem. For a CDMA system, another challenging issue is that the system capacity is interference-sensitive. Thus, the effective bandwidth-based CAC algorithm for FDD wideband CDMA networks must consider minimum-power allocation so that both maximum system capacity and BER guarantees are taken into account by the CAC algorithm. Such a CAC scheme improves the system performance by greatly reducing packet loss and packet delay of multimedia traffic. The CAC algorithm in [14] does not consider minimum-power allocation. Although the CAC algorithm proposed in [15] takes into account the minimum-power allocation, it does not include the concept of effective bandwidth, and thus it is not feasible for connections with bursty traffic. In addition, MC transmission is not considered in [15].

In this paper, simulations are also carried out to evaluate the performance of the MAC protocol. Results show that excellent performance is achieved by the MAC protocol.

The paper is organized as follows. The overall MAC protocol is described in Section II. In Section III, the minimum-power allocation algorithm is derived for an FDD wideband CDMA network. Based on the minimum-power allocation algorithm, the multimedia wideband CDMA GPS scheduling scheme is developed in Section IV, and the effective-bandwidth CAC scheme is derived in Section V. In Section VI, performance of the new MAC protocol is evaluated through simulation. Comparisons between the new MAC protocol and the FDD mode CDMA MAC protocol in [5] is presented in Section VII. The conclusion of this paper is drawn in section VIII.

II. Protocol Description

A. System Specifications

In this paper, a FDD mode wideband CDMA system is considered. Due to the broadcast nature, downlink transmission of multimedia traffic can be supported more easily than uplink transmission. Thus, when a MAC protocol is designed, the focus is on the uplink transmission.

The following transport channels defined in the 3GPP specification [8] are used in the MAC protocol:

- *Random access channel (RACH).* This channel is used by mobile terminals to send control packets, e.g., connection requests.
- *Broadcast control channel (BCCH).* This channel is a point-to-multipoint channel, which is used to convey system information from the base station to mobile terminals. For example, the feedback about resource allocation is transmitted in this channel.
- *Dedicated channel (DCH).* The DCH is point-to-point channel, which is used to transmit data from mobile terminals to the base station or vice versa.

DCHs, RACH and BCCH are multiplexed in the code division. In DCHs of FDD mode wideband CDMA, both MC transmission and VSF transmission are used. A DCH can have variable transmission rate depending on the spreading factor, and the basic transmission rate of the DCH corresponds to the maximum spreading factor used in this channel. Thus, a variable-length MAC packet is accommodated in a DCH. However, in order to simplify packet segmentation in the interface between link access control (LAC) layer and the MAC layer, a fixed-length packet, called radio link control (RLC) packet data unit (PDU) is defined [1], [4]. The size of a fixed-length RLC packet, denoted by l_r, relates to the basic transmission rate r_b of a DCH according to $l_r = r_b \cdot t_{fr}$, where t_{fr} is the frame length. When a packet is generated in the LAC layer, it is segmented into multiple RLC PDUs.

Those RLC PDUs may be transmitted in one frame or several frames, depending on available DCHs and transmission rates of these DCHs. Thus, the variable length of a MAC packet is represented by the variable number of RLC PDUs transmitted during a frame.

In order to reduce the complexity of a mobile terminal, the number of DCHs that can be used by the mobile terminal is limited. The limited number varies with service type supported in a mobile terminal. For example, a mobile terminal transmitting video traffic needs several DCHs, while a mobile terminal transmitting voice traffic can be satisfied with one DCH.

B. The MAC Protocol

The MAC protocol supports both real-time and non-real time services:

- When a mobile terminal wants to support real-time service, it needs to send a *connection request* in the RACH. When this request is received at the base station, an effective-bandwidth CAC scheme, which is based on the minimum-power allocation algorithm as will be derived in Section III, is used to check the admission of the connection request. If the answer is positive, the connection request is accepted and become ready to transmit real-time traffic. However, how the packets of this connection are transmitted in each frame is determined by the wideband CDMA GPS scheduling scheme, which will be derived in Section IV.
- When a mobile terminal wants to deliver non-real-time service, no admission control is used. Whenever packets become available in this terminal, they are ready to be transmitted as long as the resources are allocated by the base station.

Packet transmissions in real-time connections and a non-real-time traffic flows follow the same procedure:

1) In a mobile terminal, when a packet in the network layer is generated, its virtual arrival time and virtual departure time are determined by the wideband CDMA GPS scheduling scheme. Since this packet must be segmented into smaller RLC PDUs in the MAC layer, all those RLC PDUs hold the same virtual arrival time and virtual departure time. It should be noted that the wideband CDMA GPS scheduler takes into account the interference-sensitive system capacity in the CDMA networks, because it is derived by considering minimum-power allocation.
2) The priority of packet transmission is determined according to the virtual departure time. Packets in all real-time connections and non-real-time traffic flows are scheduled from the highest priority towards the lowest one, until system capacity is exhausted or no packet waits for scheduling. In the wideband CDMA GPS scheduling scheme, the limited number of code channels in a connection and the variable rate of a code channel must be considered. The system capacity in a frame is determined by the minimum-power allocation algorithm through which the interference-sensitive CDMA system capacity is maximized.
3) The received power levels of RLC PDUs that are scheduled for transmission are determined based on the minimum-power allocation algorithm.
4) After scheduling is finished in a frame, feedback information is sent back from the base station to mobile terminal through the BCCH. It includes the connection or traffic flow ID, the allocated number of code channels for each connection or traffic flow, the corresponding VSF of each code channel, the received power levels of those code channels, and the number of RLC PDUs to be transmitted in each code channel.
5) Feedback information is received by mobile terminals. The transmitted power level of a DCH is determined based on the received power level and estimated channel gain. Then, packets are transmitted in a DCH by using allocated VSF and desired transmitted power level.
6) If errors are detected but cannot be corrected in non-real-time packets received at the base station, selective-repeat ARQ is used to re-transmit these erroneous packets.

III. Minimum Power Allocation

As described in Section II-B, both the effective-bandwidth based CAC scheme and the wideband CDMA GPS scheduling scheme are based on the minimum-power allocation algorithm. The objective of this algorithm is, given a number of code channels of different users with heterogeneous BER requirements, to find the minimum received power level of each code channel such that the heterogeneous BER values of different users are satisfied. Compared to other related work [6], [16], [17], the algorithm considers all the following features of FDD wideband CDMA:

- Multiple service types with heterogeneous BER requirements are supported.
- Both MC operation and OVSF operation are taken into account when allocating resources.

However, MC operation is not considered in [6], [16], while OVSF operation is not included in [17] where only one service type is supported.

A. The Minimum-Power Allocation Algorithm

To focus on the MAC protocol design, a single cell in an FDD mode wideband CDMA wireless network is considered. There are K type of services supported in the cell, and that service type k requires SINR to be γ_k in order to have desired BER. The relationship between SINR and BER will be discussed in Section III-B. Of service type k, it is assumed that there are N_k users. The set of DCHs allocated to user n_k is represented by a vector $C^{n_k} = [C_1^{n_k}, \cdots, C_M^{n_k}]$, which must be chosen from an OVSF code tree. Such a OVSF tree has M levels of orthogonal codes, and the SF of m-th level is assumed to be $G_m = 256/2^{m-1}, m = 1, 2, \cdots, M$. Thus, the transmission rate of the a DCH using a code at m-th level is $r_m = W/G_m$, where W is the bandwidth of the wideband CDMA system. $P^{n_k} = [P_1^{n_k}, \cdots, P_M^{n_k}]$ denotes the received power levels that corresponds to DCHs of C^{n_k}. Thus, the overall transmission rate r_{n_k} and received power level of user n_k are $\sum_{m=1}^{M} C_m^{n_k} \frac{W}{G_m}$ and $\sum_{m=1}^{M} C_m^{n_k} P_m^{n_k}$, respectively.

Considering a specific user n_ζ with service type ζ, one of its DCHs at the ν-th level of the OVSF code tree experiences interference $I_\nu^{n_\zeta}$ at the receiver of the base station. $I_\nu^{n_\zeta}$ consists of two components: one is the interference from DCHs of other users in the same system, denoted by I_I, and the other is the noise, denoted by N_o. Thus, the SINR of one of the ν-th level DCHs, denoted by $\gamma_\nu^{n_\zeta}$, can be described as

$$\gamma_\nu^{n_\zeta} = \frac{P_\nu^{n_\zeta}/r_\nu}{(I_I + N_o)/W}, \tag{1}$$

where $\nu \in \{1, \cdots, M\}$, $\zeta \in \{1, \cdots, K\}$, $n_\zeta \in \{1, \cdots, N_\zeta\}$, and r_ν is the transmission rate of a DCH at the ν-th level. Interference I_I is contributed by the power levels of DCHs of all users except those of user n_ζ, so

$$I_I = \underbrace{\sum_{k=1}^{K} \sum_{n_k=1}^{N_k} \sum_{m=1}^{M} C_m^{n_k} P_m^{n_k}}_{power\ levels\ of\ all\ users} - \underbrace{\sum_{l=1}^{M} C_l^{n_\zeta} P_l^{n_\zeta}}_{power\ levels\ of\ user\ n_\zeta}. \tag{2}$$

Combining (1) and (2) and considering $\gamma_\nu^{n_\zeta} \geq \gamma_\zeta$ yield

$$\frac{P_\nu^{n_\zeta} G_\nu}{\sum_{k=1}^{K} \sum_{n_k=1}^{N_k} \sum_{m=1}^{M} C_m^{n_k} P_m^{n_k} - \sum_{l=1}^{M} C_l^{n_\zeta} P_l^{n_\zeta} + N_o} \geq \gamma_\zeta, \tag{3}$$

where $G_\nu = W/r_\nu$. To minimize the power levels of each DCH, the equality in (3) must hold. This yields

$$P_\nu^{n_\zeta} \frac{G_\nu}{\gamma_\zeta} = \sum_{k=1}^{K} \sum_{n_k=1}^{N_k} \sum_{m=1}^{M} C_m^{n_k} P_m^{n_k} - \sum_{l=1}^{M} C_l^{n_\zeta} P_l^{n_\zeta} + N_o. \tag{4}$$

Note that (4) is satisfied for all $\nu \in \{1, \cdots, M\}$ of user n_ζ. Thus, for the 1-st level DCHs of user n_ζ, the right side of (4) is the same as that for the ν-th level channel. Thus,

$$P_\nu^{n_\zeta} G_\nu = P_1^{n_\zeta} G_1, \tag{5}$$

i.e., $P_\nu^{n_\zeta} = P_1^{n_\zeta} \frac{G_1}{G_\nu}$. According to this equation, all power levels in (4) can be represented by the power level $P_1^{n_\zeta}$ or $P_1^{n_k}$. Defining Γ_{n_k} as $\sum_{m=1}^{M} \frac{G_1}{G_m} C_m^{n_k}$ and Γ_{n_ζ} as $\sum_{l=1}^{M} \frac{G_1}{G_l} C_l^{n_\zeta}$, and from (5), (4) becomes

$$\left(\frac{G_1}{\gamma_\zeta} + \Gamma_{n_\zeta}\right) P_1^{n_\zeta} = \sum_{k=1}^{K} \sum_{n_k=1}^{N_k} \Gamma_{n_k} P_1^{n_k} + N_o. \tag{6}$$

It should be noted that (6) is also satisfied for any user n_k, i.e., the left side of (6) can be $(\frac{G_1}{\gamma_k} + \Gamma_{n_k}) P_1^{n_k}$. Thus,

$$\left(\frac{G_1}{\gamma_\zeta} + \Gamma_{n_\zeta}\right) P_1^{n_\zeta} = \left(\frac{G_1}{\gamma_k} + \Gamma_{n_k}\right) P_1^{n_k}, \tag{7}$$

i.e., $P_1^{n_k} = \frac{\frac{G_1}{\gamma_\zeta} + \Gamma_{n_\zeta}}{\frac{G_1}{\gamma_k} + \Gamma_{n_k}} P_1^{n_\zeta}$. Putting this into (6) yields

$$P_1^{n_\zeta} = \frac{N_o}{(\frac{G_1}{\gamma_\zeta} + \Gamma_{n_\zeta})(1 - \sum_{k=1}^{K} \sum_{n_k=1}^{N_k} \frac{\Gamma_{n_k}}{\frac{G_1}{\gamma_k} + \Gamma_{n_k}})}. \tag{8}$$

According to the definition of Γ_{n_k} and $r_{n_k} = \sum_{m=1}^{M} C_m^{n_k} \frac{W}{G_m}$, $\Gamma_{n_k} = G_1 \frac{r_{n_k}}{W}$ and $\frac{\Gamma_{n_k}}{\frac{G_1}{\gamma_k} + \Gamma_{n_k}} = \frac{1}{1 + \frac{W}{\gamma_k r_{n_k}}}$. Putting these results back into (8), then

$$P_1^{n_\zeta} = \frac{N_o/G_1}{(\frac{1}{\gamma_\zeta} + \frac{r_{n_\zeta}}{W})(1 - \sum_{k=1}^{K} \sum_{n_k=1}^{N_k} \frac{1}{1 + \frac{W}{\gamma_k r_{n_k}}})}. \tag{9}$$

(9) gives the minimum-required power level of one of the 1-st level DCHs of user n_ζ. For DCHs on other levels of the OVSF code tree of user n_ζ, this equation also holds, i.e.,

$$P_m^{n_\zeta} = \frac{N_o/G_m}{(\frac{1}{\gamma_\zeta} + \frac{r_{n_\zeta}}{W})(1 - \sum_{k=1}^{K} \sum_{n_k=1}^{N_k} \frac{1}{1 + \frac{W}{\gamma_k r_{n_k}}})}, \tag{10}$$

as long as an m-th level channels is allocated to user n_ζ. Since each DCH has power constraint, it is required that the received power level of one of the m-th level DCHs be less than P_m^{max}. Thus, $1 - \sum_{k=1}^{K} \sum_{n_k=1}^{N_k} \frac{1}{1 + \frac{W}{\gamma_k r_{n_k}}} \geq \frac{N_o/G_m}{P_m^{max}(\frac{1}{\gamma_\zeta} + \frac{r_{n_\zeta}}{W})}$ for all $m \in \{1, \cdots, M\}$ and $\zeta \in \{1, \cdots, K\}$. Thus,

$$\sum_{k=1}^{K} \sum_{n_k=1}^{N_k} \frac{1}{1 + \frac{W}{\gamma_k r_{n_k}}} \leq 1 - \max_{\substack{m=1,\cdots,M \\ \zeta=1,\cdots,K}} \frac{N_o/G_m}{P_m^{max}(\frac{1}{\gamma_\zeta} + \frac{r_{n_\zeta}}{W})}. \tag{11}$$

Define $\Delta = \max_{\substack{m=1,\cdots,M \\ \zeta=1,\cdots,K}} \frac{N_o/G_m}{P_m^{max}(\frac{1}{\gamma_\zeta} + \frac{r_{n_\zeta}}{W})}$, (11) becomes

$$\sum_{k=1}^{K} \sum_{n_k=1}^{N_k} \frac{1}{1 + \frac{W}{\gamma_k r_{n_k}}} \leq 1 - \Delta, \tag{12}$$

which must be satisfied in order to have minimum-power allocation for each DCH and satisfy BER of each user in a wideband CDMA system. Since $\frac{P_m}{N_o/G_m}$ is the SNR, so $\frac{P_m^{max}}{N_o/G_m}$ must be the maximum SNR that can be achieved by the receiver on the base station. For the base station receiver of current 5 MHz wideband CDMA, it is normal that SNR can be as high as $60dB$. Thus, a very small value of Δ in (12) can guarantee that the power level of each code channel does not exceed its maximum value.

In (12), $\frac{1}{1 + \frac{W}{\gamma_k r_{n_k}}}$ can be viewed as the *normalized transmission rate* of user n_k whose service type is k, transmission rate in a frame is r_{n_k}, and required SINR is γ_k. Thus, (12) means that the overall normalized transmission rates of all users in a frame cannot exceed $1 - \Delta$, which is called the *normalized system capacity*.

B. Relationship between BER and SINR

In a wireless network, the relationship between BER and SINR is determined by several factors such as error control algorithms, modulation schemes, and channel fading. In this paper, the modulation scheme is assumed to be binary phase shifting keying (BPSK), and channels are assumed to have Rayleigh fading. Error control schemes are different for real-time and non-real-time traffic.

- *Real-time traffic.* At the receiver, convolutional, RS, and CRC decoders are used sequentially to correct the errors in packets. The BER and SINR relationship of real-time

TABLE I
SINR VALUES AND BER REQUIREMENTS.

Service Type	BER	SINR (dB)
Voice	10^{-3}	2.54
Audio	10^{-4}	2.72
CBR Video	10^{-5}	2.91
VBR Video	10^{-6}	3.08
Data or Email	≈ 0	2.75

traffic is derived using the same formulas for class-I traffic in [5].

- *Non-real-time traffic.* Only the convolutional and RS decoders are used to correct errors in packets. The residual errors are detected by the CRC decoder. Selective ARQ is used for re-transmission of detected erroneous packets. The probability P_r of packet re-transmission triggered by the CRC decoder is approximately equal to the bit error probability at the output of the RS decoder [5]. Suppose the transmission rate of a non-real-time mobile terminal is r_{nrt} and its allowed normalized transmission rate is R_{nrt}. Thus, according to the definition of *normalized transmission rate* in Section III-A, $r_{nrt} = \frac{W}{(\frac{1}{R_{nrt}}-1)\gamma_{nrt}}$, where γ_{nrt} is the required SINR of non-real-time traffic. Thus, the throughput ρ of the mobile terminal is $(1-P_r)r_{nrt}$, i.e.,

$$\rho = (1-P_r)\frac{W}{(\frac{1}{R_{nrt}}-1)\gamma_{nrt}}. \qquad (13)$$

  P_r increases as γ_{nrt} decreases, so there is an optimum SINR γ^*_{nrt} that achieves maximum throughput of the mobile terminal. Such an optimum SINR is determined by maximizing $(1-P_r)/\gamma_{nrt}$.

Suppose $\frac{1}{2}$ convolutional coding with $d_{free} = 10$ and $(256, 240, 256)$ RS coding with error correction capability $t = 8$ are used, then the SINR values corresponding to the typical BERs of different services are listed in Table I.

IV. GPS SCHEDULING SCHEME FOR WIDEBAND CDMA

A. The wideband CDMA GPS scheduling scheme

When packets in a frame are available for transmission, they need to be scheduled according to their QoS and BER requirements. The scheduling scheme must satisfy several constraints in order to achieve high performance in the FDD mode wideband CDMA system:

- *Power/BER constraint.* When packets are transmitted in a wideband CDMA frame, minimum-power allocation must be used in order to achieve maximum capacity and guarantee BER requirements. Therefore, the constraint in (12) must be satisfied in the scheduling scheme.
- *QoS Requirement.* The scheduling scheme must support heterogeneous QoS requirements of multimedia traffic. It needs to be fair and provide QoS guarantees to multimedia traffic.
- *Code channel constraint.* Not all DCHs on the OVSF code tree can be used simultaneously by a user. One reason is that the DCHs used simultaneously must be orthogonal. The other reason is that the number of DCHs available to a user is generally limited, which reduces the complexity of a mobile terminal by reducing the transceiver units [18]. Since scrambling codes are user-specific [1], code blocking that exists in the downlink [18] does not occur in the uplink of wideband CDMA systems. In order to take into account code channel constraint, the scheduling scheme must check if the permitted number of DCHs in a mobile terminal is exceeded.

Having these constraints in mind, a new scheduling scheme, called wideband CDMA GPS scheduling, is proposed in this section. The wideband CDMA GPS scheduling scheme is based on GPS. The significant feature of GPS is that it treats different traffic types differently according to their QoS requirements [12]. GPS also assumes that multiple traffic flows with variable traffic rates can be served simultaneously. This was considered as a drawback of GPS because the classical packet-based systems are TDMA-based and thus do not permit parallel packet transmissions. However, in a CDMA system, it is natural to simultaneously serve multiple traffic flows with variable transmission rates. Thus, GPS helps to design a high performance scheduling scheme for CDMA systems. In [13], a GPS-based scheduling scheme is proposed for hybrid CDMA/TDMA systems. The scheme in [13] assumes that only VSF transmission is used. The scheduling scheme under this assumption is easy to derive, because the power/BER constraint has a simpler form and no code channel constraint exists. However, in FDD mode wideband CDMA systems, both MC and VSF transmissions need to be considered. Thus, the wideband CDMA GPS scheduling scheme designed here is much more general and flexible to support multimedia traffic.

The FDD mode wideband CDMA does not have a TDMA frame. Mobile terminals with different BER requirements do not have separate time slots to accommodate their packets. The lack of such a flexibility results that the BER-scheduling schemes used in [19] cannot be applied in FDD mode wideband CDMA networks.

The wideband CDMA GPS scheduling scheme is operated as follows:

1) *Determine the virtual finishing time of MAC packets.* When a LAC PDU of a mobile terminal arrives, the base station should be informed of the arrival time of this packet. Then, the virtual finishing time of this LAC PDU is determined, which will be derived in Section IV-B. RLC PDUs belonging to the same LAC PDU have the same virtual finishing time.
2) *Serve RLC PDUs according to virtual finishing times.* In a wideband CDMA frame, the RLC PDUs from different mobile terminals are serviced from the highest priority towards the lowest one. The smaller the virtual finishing time of RLC PDUs, the higher the priority. In addition, two constraints need to be considered:

- *Check system capacity.* The power constraint in (12) must be checked. As long as this constraint is satisfied, packets are scheduled until all RLC PDUs are serviced or no system capacity is left.
- *Check code channel constraint.* Since each mobile terminal has a limited number of DCHs, an algorithm is necessary to select appropriate DCHs and transmit as many packets as possible by using these DCHs. Such an algorithm will be discussed in Section IV-C.

3) *Calculate received power levels.* After the code channels has been assigned to each mobile terminal, the received power level of each assigned code channel is calculated according to (10).

B. Determining Virtual Finishing Time

As in Section III-A, a cell in a FDD mode wideband CDMA network can be considered as a queueing system with capacity $1-\Delta$. Denote $\Omega_i(t_1,t_2)$ as the amount of traffic of session i served in the time interval $(t_1,t_2]$, and $\omega_i(t)$ as the work rate of a session, i.e., $\omega_i(t)=\frac{d}{dt}\Omega_i(0,t)$. ϕ_i is a positive number associated with session i. According to the definition of GPS and its work conserving characteristics [12], the GPS for a cell of FDD mode wideband CDMA network must have the following two features:

- The work rate of each backlogged session i is guaranteed to be
$$\Omega_i=\frac{\phi_i}{\sum_{j\in A}\phi_j}(1-\Delta), \tag{14}$$
where A is the set of all the accepted sessions in the system.
- Fair resource sharing is guaranteed, i.e., for any two backlogged sessions i and j, $\frac{\omega_i(t)}{\omega_j(t)}=\frac{\phi_i}{\phi_j}$. Thus,
$$\omega_i(t)=\frac{\phi_i}{\sum_{j\in\beta(t)}\phi_j}(1-\Delta),\ \forall i\in\beta(t), \tag{15}$$
where $\beta(t)$ is the set of all backlogged sessions.

For the m-th LAC PDU of session n_k, assume it arrives at $a^m_{n_k}$, starts service at $S^m_{n_k}$, and finishes service at $d^m_{n_k}$. According to the definition of *normalized transmission rate* in Section III-A, $\omega_{n_k}(t)=\frac{1}{1+\frac{W}{\gamma_k r_{n_k}}}$ during $(S^m_{n_k},d^m_{n_k}]$. Combining this result with (15) when $i=n_k$, then $\frac{1}{1+\frac{W}{\gamma_k r_{n_k}}}=\frac{\phi_{n_k}}{\sum_{j\in\beta(t)}\phi_j}(1-\Delta)$, i.e.,

$$\begin{aligned} r_{n_k} &= \frac{\frac{W}{\gamma_k}}{\frac{\sum_{j\in\beta(t)}\phi_j}{\phi_{n_k}(1-\Delta)}-1}, \\ &= \frac{W}{\gamma_k}\frac{\phi_{n_k}(1-\Delta)}{\sum_{j\in\beta(t)}f_j\phi_j}, \end{aligned} \tag{16}$$

where $n_k\in\beta(t)$ and $f_j=1$ if $j\neq n_k$; otherwise, $f_j=\Delta$.

For any given busy period $(t_1,t_2]$ in GPS, the virtual time $v(t)$ of session i is defined as [20]

$$v(t_2)-v(t_1)=\frac{\Omega_i(t_2,t_1)}{\Omega_i},\forall i\in\beta(t_1,t_2), \tag{17}$$

where $v(0)=0$. Thus, the virtual time of the m-th LAC PDU of session n_k is

$$v(d^m_{n_k})-v(S^m_{n_k})=\frac{\Omega_{n_k}(S^m_{n_k},d^m_{n_k})}{\Omega_{n_k}} \tag{18}$$

Considering that $\omega_{n_k}(t)=\frac{1}{1+\frac{W}{\gamma_k r_{n_k}}}$ during $(S^m_{n_k},d^m_{n_k}]$, $\Omega_i(S^m_{n_k},d^m_{n_k})$ can be described as

$$\begin{aligned} \Omega_i(S^m_{n_k},d^m_{n_k}) &= \int_{S^m_{n_k}}^{d^m_{n_k}}\omega_{n_k}(t)dt, \\ &= \frac{d^m_{n_k}-S^m_{n_k}}{1+\frac{W}{\gamma_k r_{n_k}}}, \\ &= \frac{L^m_{n_k}}{r_{n_k}+\frac{W}{\gamma_k}}, \end{aligned} \tag{19}$$

where $L^m_{n_k}=r_{n_k}(d^m_{n_k}-S^m_{n_k})$ is the length of the m-th LAC PDU of session n_k. Putting (16) into (19) yields

$$\Omega_i(S^m_{n_k},d^m_{n_k})=\frac{L^m_{n_k}}{\frac{W}{\gamma_k}\frac{\sum_{j\in\beta(t)}\phi_j}{\sum_{j\in\beta(t)}f_j\phi_j}}. \tag{20}$$

Thus, (18) becomes

$$v(d^m_{n_k})-v(S^m_{n_k})=\frac{L^m_{n_k}}{\frac{W}{\gamma_k}\frac{\sum_{j\in\beta(t)}\phi_j}{\sum_{j\in\beta(t)}f_j\phi_j}\Omega_{n_k}}. \tag{21}$$

From (14), $\Omega_{n_k}=\frac{\phi_{n_k}}{\sum_{j\in A}\phi_j}(1-\Delta)$. Thus, (21) becomes

$$v(d^m_{n_k})-v(S^m_{n_k})=\frac{L^m_{n_k}}{\frac{W}{\gamma_k}\frac{\sum_{j\in\beta(t)}\phi_j}{\sum_{j\in\beta(t)}f_j\phi_j}\frac{\phi_{n_k}(1-\Delta)}{\sum_{j\in A}\phi_j}}. \tag{22}$$

Because $v(S^m_{n_k})$ is defined to be $v(S^m_{n_k})=\max\{v(d^{m-1}_{n_k}),v(a^m_{n_k})\}$ [20], $v(d^m_{n_k})$ is thus derived as

$$\begin{aligned} v(d^m_{n_k}) &= \max\{v(d^{m-1}_{n_k},v(a^m_{n_k})\} \\ &\quad +\frac{L^m_{n_k}}{\frac{W}{\gamma_k}\frac{\sum_{j\in\beta(t)}\phi_j}{\sum_{j\in\beta(t)}f_j\phi_j}\frac{\phi_{n_k}(1-\Delta)}{\sum_{j\in A}\phi_j}}, \end{aligned} \tag{23}$$

where $v(a^m_{n_k})$ is determined from $\frac{dv(t)}{dt}=\frac{\sum_{j\in A}\phi_j}{\sum_{j\in\beta'(t)}\phi_j}$. It should be noted that $\beta'(t)$ is the set of all backlogged sessions before the arrival of the m-th LAC PDU. (23) determines the virtual finishing time of a LAC PDU once it arrives. Such virtual time is used as a priority for fair queueing of packets. In a wideband CDMA frame, all packets are serviced from the queue with the smallest virtual finishing time towards the one with the largest virtual finishing time, until system capacity is exhausted.

C. Checking Code Channel Constraint

In order to decrease the complexity of the transceiver of mobile terminals, the maximum number of DCHs for user n_k, denoted by M_{n_k}, is normally much less than the available codes on the OVSF tree. Thus, in the wideband CDMA GPS scheduling scheme, this constraint must be always checked so that it is not violated.

Given M_{n_k} DCHs for user n_k and an OVSF tree with M levels of codes, the available transmission rates for such a user need to be determined by considering the following two factors:

1) The total number of DCHs is not larger than M_{n_k}.
2) The codes of two DCHs cannot be on the same path to the root of the OVSF tree.

One approach to this problem is to check if a transmission rate can be decomposed into l $(l \leq M_k)$ smaller transmission rates such that each of these smaller transmission rates is $2^{i-1}(i = 1, \cdots, M)$ times of the basic transmission rate r_b, i.e. the transmission rate of a DCH using the M-th level code on the OVSF code tree. As an example, the set of available transmission rates when $M_{n_k} = 2$ and $M = 7$ is $\{1,\ 2,\ 3,\ 4,\ 5,\ 6,\ 8,\ 9,\ 10,\ 12,\ 16,\ 17,\ 18,\ 20,\ 24,\ 32,\ 33,\ 34,\ 36,\ 40,\ 48,\ 64\}$, where the transmission rates are in unit of r_b. Such an approach can be used off-line to determine the set of available transmission rates.

Suppose the number of packets scheduled by the wideband GPS scheduling scheme for user n_k is g_{n_k}. Then, the transmission rate corresponding to g_{n_k} packets must be $g_{n_k} r_b$, since one packet needs a basic transmission rate. If such a transmission rate lies in the set of available transmission rates for user n_k, all the scheduled packets can be transmitted. Otherwise, the maximum transmission rate that is less than $g_{n_k} r_b$ is used. Suppose this allowed transmission rate $h_{n_k} r_b$, then $(g_{n_k} - h_{n_k})$ packets cannot be served in the current frame due to the constraint of available code channels. However, since some packets are not served, a certain amount of system capacity is not utilized by user n_k. Therefore, under this situation, the wideband CDMA GPS scheduling scheme needs to be performed again until system capacity is fully utilized or all packets are serviced.

V. Effective-Bandwidth CAC

Since single cell is considered when designing a MAC protocol, user mobility is not taken into account in the CAC algorithm.

A. The CAC Algorithm

The CAC algorithm is only applied to real-time connections. Non-real-time connections are always accepted. In order to take into account the contribution of non-real-time traffic to the overall *normalized transmission rates*, a minimum *normalized transmission rate*, denoted by R_{nrt}, is reserved for non-real-time traffic. $R_{n_k} = \frac{1}{1+\frac{W}{\gamma_k r_{n_k}}}$ denotes the *normalized transmission rate* of a real-time connection n_k, then, from (12), the following constraint must be satisfied for real-time connections:

$$\sum_{k=1}^{K} \sum_{n_k=1}^{N_k} R_{n_k} \leq 1 - \Delta - R_{nrt}, \tag{24}$$

where N_k is the number of real-time connections in the system.

During the life time of the connection n_k, R_{n_k} is a random variable because r_{n_k} varies from frame to frame. Thus, a satisfaction factor α is used to evaluate the probability that (24) is satisfied, i.e., for $0 < \alpha \leq 1$, if

$$\Pr(\sum_{k=1}^{K} \sum_{n_k=1}^{N_k} R_{n_k} \leq 1 - \Delta - R_{nrt}) > \alpha, \tag{25}$$

then (24) is satisfied with probability α.

Given a satisfaction factor α, the admission region of real-time connections can be determined based on (25). In what follows, Gaussian approximation [14] is used to derive the admission region.

Of the same service type k, different connections are independent and follow the same traffic characteristics. Considering the connection n_k, denote the mean and variance of R_{n_k} as μ_k and σ_k^2, respectively. According to the central limit theorem, when N_k is large, $\sum_{n_k=1}^{N_k} R_{n_k}$ can be approximated by a Gaussian random variable $\mathcal{G}_k$ with mean and variable equal to $N_k\mu_k$ and $N_k\sigma_k^2$, respectively. Since $\{\mathcal{G}_k, k = 1, \cdots, K\}$ are independent Gaussian random variables, $\sum_{k=1}^{K} \mathcal{G}_k$ can also be approximated by a Gaussian random variable $\mathcal{G}$ whose mean and variance are $\sum_{k=1}^{K} N_k\mu_k$ and $\sum_{k=1}^{K} N_k\sigma_k^2$, respectively. Thus, (25) becomes

$$\Pr(\mathcal{G} < 1 - \Delta - R_{nrt}) > \alpha. \tag{26}$$

According to the characteristics of Gaussian random variable, (26) is satisfied if and only if

$$\frac{1 - \Delta - R_{nrt} - E[\mathcal{G}]}{\sqrt{Var[\mathcal{G}]}} \geq \beta, \tag{27}$$

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

$$\frac{1}{\sqrt{2\pi}} \int_{\beta}^{\infty} e^{-t^2/2} dt = 1 - \alpha. \tag{28}$$

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, (27) becomes

$$\sum_{k=1}^{K} N_k\mu_k + \beta\sqrt{\sum_{k=1}^{K} N_k\sigma_k^2} \leq 1 - \Delta - R_{nrt}, \tag{29}$$

which determines an admission region $(N_1, \cdots, N_K)$. When a connection of service type k arrives, N_k is increased by one. Whether or not this connection can be accepted is decided by checking if the new value of N_k satisfies (29). If the answer is positive, the connection is accepted. Otherwise, it is rejected. When a connection of service type k is terminated, N_k is decreased by one, which equivalently releases the capacity occupied by the connection. Therefore, in the CAC algorithm given by (29), the effective bandwidth of each new arrival connection does not need to be explicitly determined, which decreases the complexity of the CAC algorithm.

VI. Performance Evaluation

Simulation is carried out to evaluate the performance of the MAC protocol.

A. Traffic Models and System Parameters

Six types of traffic models [19] are considered in the simulation:

- *Voice.* The duration of a voice connection is exponentially distributed with the average equal to 180.0 s. The average length of talkspurts and gaps are 1.00 s and 1.35 s, respectively. When a connection is in the talkspurt, the average length of minispurts and gaps are 0.235 s and 0.050 s, respectively. BER of voice traffic is 10^{-3}, timeout value of a voice packet is 2 frames, and the maximum number of DCHs for a voice connection is 1.
- *Audio.* The duration of an audio connection is also exponentially distributed with the average equal to 180.0 s. The bit rate is 128 kbps. BER of audio traffic is 10^{-4}, timeout value of an audio packet is 4 frames, and the maximum number of DCHs for a connection is 2.
- *CBR video.* The duration of a CBR video connection is exponentially distributed with the average equal to 360.0 s. The bit rate is 220 kbps. BER of CBR video traffic is 10^{-5}, timeout value of a packet is 4 frames, and the maximum number of DCHs for a connection is 3.
- *VBR video.* The duration of a VBR video connection is exponentially distributed with the average equal to 180.0 s. A multiple-state model is used to simulate the traffic in each connection. Duration of each state of the model is also exponentially distributed with the average equal to 160 msec. The traffic rate in each state is obtained from a truncated exponential distribution with the maximum and minimum bit rates equal to 120 and 420 kbps, respectively. BER of VBR video traffic is 10^{-6}, timeout value of a packet is 6 frames, and the maximum number of DCHs for a connection is 3.
- *Computer data.* The length of a computer data message is exponentially distributed with the mean size equal to 30 kbytes. BER of data traffic is approximately equal to zero, and the maximum number of DCHs for a call is 1. Packets cannot be dropped because of timeout.
- *Email.* The email model proposed in [19] is used to simulate emails. BER of email is approximately equal to zero, and the maximum number of DCHs for a call is 1. Packets cannot be dropped because of timeout.

Among all calls, 60%, 5%, 6%, 9%, 15%, and 5% are generated by voice, audio, CBR video, VBR video, computer data, and email traffic. For the wideband CDMA system, the level of the OVSF tree M is 7, the basic transmission rate r_b is 19.5 kbps, system bandwidth W is 5 MHz, satisfaction factor α is 0.990, minimum capacity for non-real-time traffic R_{nrt} is 0.02, and and Δ in (12) is 0.0005. The SINR values corresponding to different BERs are given in Table I. Frame length t_{fr} is 10 ms.

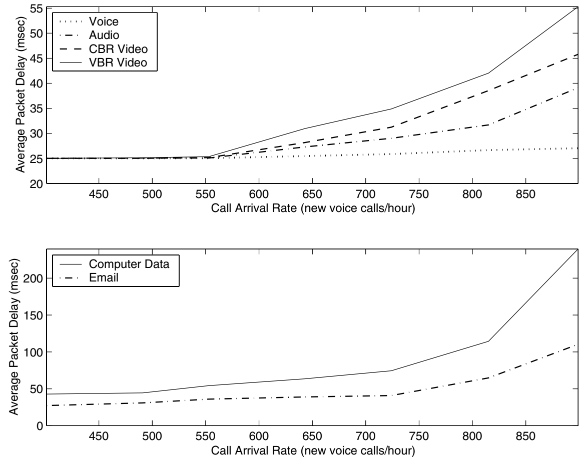

Fig. 1. Average packet delay versus call arrival rate without CAC.

The performance metrics for the MAC protocol are the average packet delay d_p, packet loss ratio l_p, throughput t_r, and call blocking probability b_c. The average packet delay consists of three components, i.e., $d_p = d_r + d_a + d_t$, where d_r is the average time of successfully sending a packet transmission request, d_a is the queueing time before a code channel is allocated to a packet, and d_t is the transmission time of a packet after the code channel is allocated. In the simulation, the code channels in RACH are assumed to have a large number, so the request transmission is collision-free. Thus, the average value of d_r is equal to half a frame because the generation time of a request is uniformly distributed in a frame. Since pure CDMA is used in FDD mode wideband CDMA, a packet is transmitted within a whole frame. Thus, d_t is equal to the frame length. l_p of a connection is defined as $l_p = \frac{N_l}{N_l+N_t}$, where N_l is the number of lost packets due to timeout, and N_t is the number of packets being successfully transmitted. r_t is defined as the total packets being transmitted in a frame. b_c of a service type is defined as the $b_c = \frac{C_b}{C_b+C_a}$, where C_b and C_a are the number of blocked calls and accepted calls, respectively, of a service type.

B. Numerical Results

1) Experiments without CAC: Three performance metrics such as average packet delay, packet loss ratio, and throughput are used. To show the performance versus traffic load, different experiments are performed by varying the call arrival rates.

The average packet delay versus the call arrival rate is shown in Fig. 1. The results of packet loss ratio and throughput are shown in Fig 2 and Fig. 3, respectively. As shown in 1, the average packet delay of each service type increases as the traffic load becomes high. The average packet delay of non-real-time traffic is much larger than that of real-time traffic because: a) the virtual finishing times of non-real-time packets

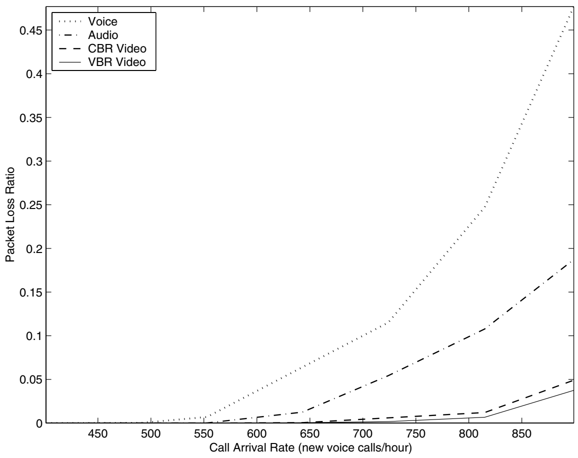

Fig. 2. Packet loss ratio versus call arrival rate without CAC.

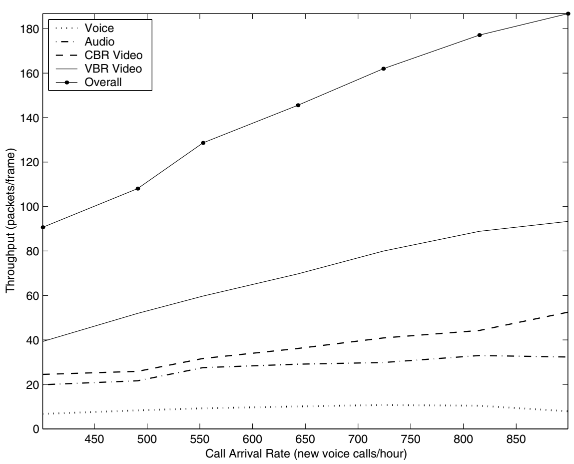

Fig. 3. Throughput versus call arrival rate without CAC.

are much larger; b) no timeout occurs in non-real-time traffic. Among the service types of real-time traffic, voice has the smallest average packet delay. The average packet delay of audio is smaller than that of video traffic, because the bit rate of a video connection is larger. VBR video has the largest average packet delay due to its bursty characteristics. It should be noted that the packet delay of real-time traffic is guaranteed to be lower than a bound, because each real-time service has a certain timeout value and the delay of each packet cannot exceed this value.

As shown in Fig. 2, the packet loss ratio of voice traffic is much larger than those of other service types, which is reasonable since voice traffic is less sensitive to packet loss than other types of traffic. As shown in Fig 3, the increment of throughput becomes lower and lower when the traffic load increases. The reason is that the system capacity is gradually approached as traffic load increases, and thus, the packet loss becomes higher and higher. This can be illustrated by the comparison between the throughput in Fig. 3 and the packet loss ratio in Fig. 2.

As shown in Fig. 2, when the packet arrival rate is very

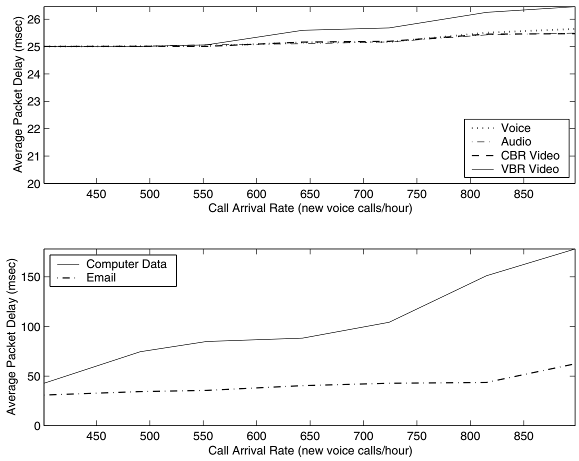

Fig. 4. Average packet delay versus call arrival rate with CAC.

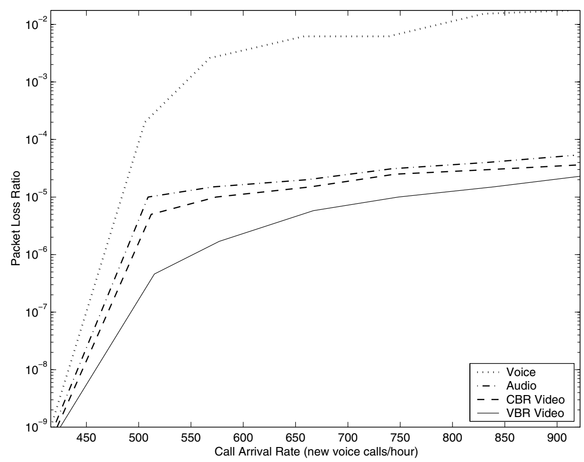

Fig. 5. Packet loss ratio versus call arrival rate with CAC.

high, the packet loss ratio of a service type becomes too large to be acceptable. In order to resolve this issue, CAC must be used so that some connections are blocked to ensure that traffic load in the system does not exceed the system capacity.

2) Experiments with CAC: In this experiment, the CAC scheme proposed in Section V is employed to admit connections of real-time traffic. With CAC, the average packet delay, packet loss ratio, and throughput are shown in Figs. 4, 5, 6, respectively. In addition, the connection blocking probability of each real-time service type is shown in Fig. 7.

As illustrated by the comparison between Fig. 1 and Fig. 4, the average packet delay is greatly decreased. For real-time traffic, the average packet delay is almost decreased to 25 msec. Comparisons between the results in Fig. 5 and Fig. 2 show that the CAC algorithm greatly reduces packet loss ratio of real-time traffic. Such improvement is because CAC ensures that the traffic load does not exceed the system capacity. It should be noted that the minimum value of average packet delay is 25 msec, because the average packer delay at least consists of half a frame of request delay, one frame of queueing delay, and one frame of transmission delay.

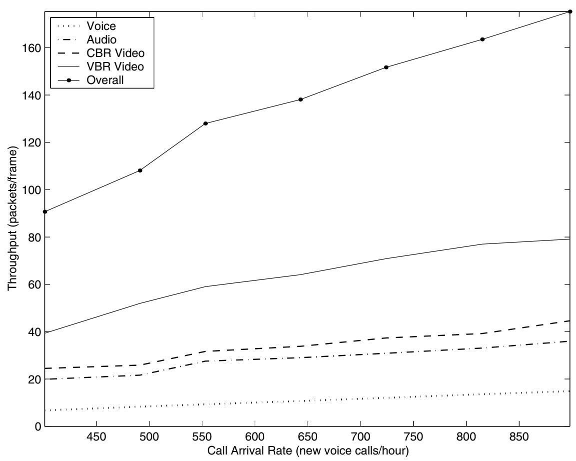

Fig. 6. Throughput versus call arrival rate with CAC.

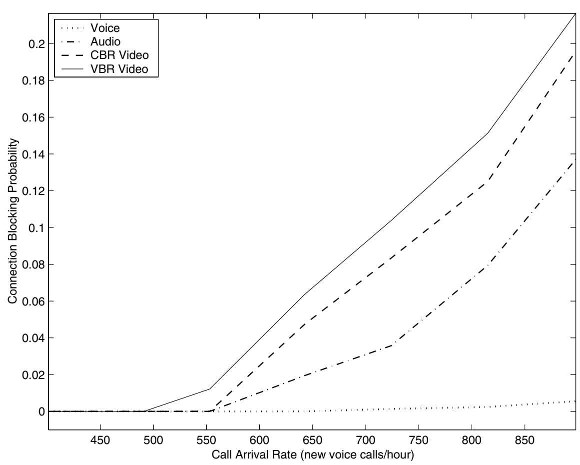

Fig. 7. Connection blocking probability versus call arrival rate.

As shown in Fig. 6, the system throughput is lower than that in 3. The reason is that the rejected connections reduce overall offered load in the system. However, the throughput reduces by less than 10% for two orders of magnitude of reduced packet loss ratio.

As shown by the connection blocking probability in Fig. 7, connections with higher traffic rate are easier to be rejected. Thus, the CAC scheme is fair to different service types.

VII. Comparisons

The protocol proposed in [5] and the new MAC protocol proposed in this paper have similar features. For example, multimedia traffic with diverse QoS requirements can be supported by both protocols. Moreover, power allocation to a code channel is considered in the CAC and scheduling schemes. However, in the new MAC protocol, the minimum-power allocation algorithm and the wideband CDMA GPS scheduling scheme are used. Therefore, interference-sensitive system capacity can be utilized more efficiently, and packets of different services can be fairly serviced according to their heterogeneous BER and QoS requirements. In order to show

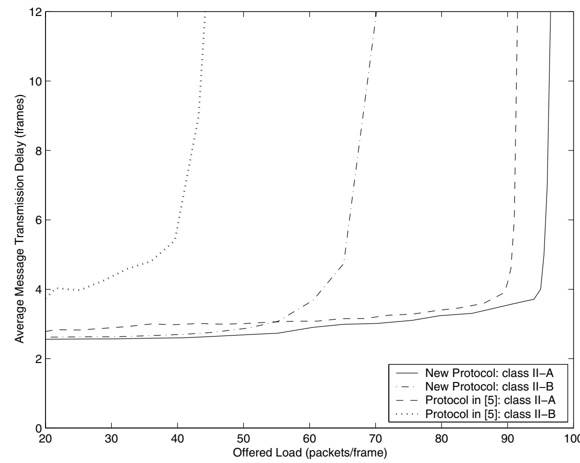

Fig. 8. Average message transmission time versus offered load.

such advantages, the new MAC protocol is compared with that in [5]. Since peak rate is assigned to a real-time connection in [5], QoS of real-time connections is always guaranteed. Thus, the average message transmission delay of non-real-time traffic (i.e., the class II traffic in [5]) is the only performance metrics used in the comparison between two protocols.

Assumptions, traffic models, and system parameters are summarized as follows, which are same as those in [5]:

1) There are two types of class-II traffic. Class II-A traffic is delay sensitive, while class II-B is delay-tolerable.
2) The number of packets in each message of class II-A is geometrically distributed with average 2. Same distribution is used for class II-B, but the average message size is equal to 18 packets.
3) 50 mobiles are generated for each traffic type. In each mobile, the message generation rate is a Poisson process. The average generation rate of class II-A is $0.9\lambda/50$, while that of class II-B is $0.1\lambda/50$. Thus, each traffic type has the equal load (i.e., 1.8λ), and the total offered load is $0.9\lambda \cdot 2 + 0.1\lambda \cdot 18 = 3.6\lambda$.
4) Fading is not considered in signaling and control channels.
5) The system bandwidth is equal to 128×8 kbps, i.e., 1.024 Mbps.
6) 50 out of 107 bandwidth units are reserved for class II traffic. Real-time connections (i.e., class I traffic in [5]) do not change throughout the simulation. Thus, they do not affect the performance of class II traffic.
7) The average message transmission delay consists of three components: request access delay, queueing delay, and transmission delay.

In addition, a large number of request access code channels are used so that request collision rarely occurs. In the simulation, when this number is equal to 25, the collision is almost free.

Under these assumptions and system parameters, the average message transmission delay versus the offered load is shown in Fig. 8. In our protocol, although a wideband CDMA GPS scheduling scheme is used, the average message transmission delay of two traffic types is different. The reason is that a larger weight (i.e. ϕ_i in (23)) is assigned to class II-A traffic than that to class II-B traffic.

As shown in Fig. 8, for class II-B traffic, the new MAC protocol achieves a lower average message transmission delay. When the traffic load is lower than 40 packets/frame, the difference of the average message transmission delay is approximately between 1.5 and 3.5 frames. However, when the offered load is between 45 and 65 packets/frame, the average message transmission delay achieved by the protocol in [5] is more than 10 frames larger than that of the new protocol. For class II-A traffic, when the offer load is less than 90 packets/frame, the average message transmission delay of the new protocol is a little smaller than that achieved by the protocol in [5]. However, when the offered load is between 90 and 95 packets/frame, the average message transmission delay of the new MAC protocol is more than 10 frames smaller. The main reasons for the better performance achieved by the new protocol are as follows:

- *Minimum-power allocation algorithms.* The new MAC protocol uses minimum-power allocation for each traffic type. Although the protocol in [5] allocates different power levels to different traffic type, minimum-power allocation is not considered. Therefore, given the same offered load, the new protocol can transmit more packets in a frame.
- *Fair scheduling schemes.* A wideband CDMA GPS scheduling scheme is used in the new MAC protocol to allocate the code channels to mobile terminals. The transmission order of packets are determined based on their virtual arrival times. Thus, packets of class II-B are not necessarily transmitted later than those of class II-A traffic. However, in [5], higher priority is always given to class II-A traffic. Thus, the message transmission delay of class II-B traffic is large and increases abruptly when the offered load is as low as 45 packets/frame.
- *Better re-transmission mechanism.* In the protocol in [5], although the packets to be re-transmitted are put into a FIFO queue and have higher priority than packets in the round-robin queue, packets in the FIFO queue of class II-B traffic is still lower than packets in the FIFO and the round-robin queues of class II-A traffic. Thus, for class II-B traffic, the packets to be re-transmitted have a large queueing delay. In our protocol, a packet to be re-transmitted participates in scheduling according to its virtual arrival time. Thus, this packet does not need to be transmitted later than a class II-A traffic.

VIII. Conclusions

In this paper, a MAC protocol was proposed for next generation wireless networks based on FDD mode wideband CDMA. First, a minimum-power allocation algorithm was derived with the consideration of both MC and VSF transmissions. Based on this algorithm, a wideband CDMA GPS scheduling scheme was proposed. It took into account different constraints such as interference-limited system capacity, BER and QoS requirements of multimedia traffic, and limited number of codes in each mobile terminal. An effective bandwidth-based CAC algorithm was also proposed with the consideration of minimum-power allocation. The MAC protocol proposed in this paper guarantees heterogeneous QoS of multimedia traffic and improves the overall system throughput.

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