Uplink CDMA Systems with Diverse QoS Guarantees for Heterogeneous Traffic *

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Abstract

Addressed is the uplink access control problems for cellular CDMA systems that service heterogeneous traffic with various types of quality-of-service (QoS) and use Multi-Code CDMA (MC-CDMA) to support variable bit rates. Considering its distinct QoS requirements, class-I real-time traffic (e.g., voice and video) is differentiated from class-II non-real-time traffic (e.g., data). Connection-oriented transmission is achieved by assigning transmitter-oriented code channels for class-I traffic where each corresponding mobile client needs to pass an admission test. Class-II traffic is transmitted in a best-effort manner using the bandwidth left unused by class-I traffic and a transmission-rate request access scheme: whenever a mobile has class-II messages to transmit, the mobile requests code channels via a receiver-oriented transmission-request code channel, then, according to the base station scheduling, the transmission is scheduled and permitted. Addressed are the admission test for class-I connections, power control, and how to maximize the aggregate throughput for class-II traffic. The admission region of voice and video connections and the optimum target signal-to-interference ratio (SIR) of class-II traffic are derived numerically. The performance of class-II traffic transmissions in terms of average delay is also evaluated and discussed based on simulations.

Index Terms—CDMA systems, wireless/mobile communication, Multi-Code CDMA, admission control, power control, QoS guarantees, transmission-rate request access protocol, Reed-Solomon/convolutional concatenated code, Automatic Retransmission reQuest (ARQ).

1 Introduction

Code-Division Multiple-Access (CDMA) is emerging as a promising technique for future cellular mobile/wireless communication systems. As claimed by many researchers [1, 2], CDMA in cellular environments offers several advantages over other wireless access techniques such as high spectral efficiency, soft capacity, soft handoff, and increased system capacity.

Here, we consider a direct-sequence (DS) CDMA uplink (mobile-to-base) system to provide three types of QoS for heterogeneous traffic: (1) bounded packet-delivery delay; (2) guaranteed transmission rate; (3) packet error probability (or bit error probability, BER). Table 1 summarizes the classification of heterogeneous traffic. Basically, class I and II are differentiated according to the required delay and rate performance. Class-I traffic requires bounded delay and guaranteed rate, but "looser" error performance. Voice and video fall into this class, although video requires better error performance and higher rate than voice. Currently, low-rate voice coding schemes like QCELP and VSELP are being used to deliver speech over a wireless link at a rate of 8 Kbps or less [3]. There has also been significant research on low-rate video coding techniques to transmit video at a rate lower than 64 Kbps including the new ITU-T recommendation H.263 [4]. Class-II traffic, like the conventional data service, does require virtually loss-free transmission, but does not require guaranteed rate nor bounded delay. Class II is divided further into two subclasses: (1) class II-A which is delay-sensitive like FTP and remote log-in; and (2) class II-B which is delay-tolerant like paging and e-mail. Class II-A receives priority over class II-B.

To support multi-rate transmissions with DS-CDMA systems, many techniques have been proposed such as multi-modulation CDMA, multi-processing gain CDMA, and multi-code CDMA [5–7]. We adopt the multi-code (MC) CDMA because it has some advantages over other methods; the multi-modulation CDMA degrades the performance for the users with high data rates [5], and the multi-processing gain CDMA is expected to cause problems to users with very high source rate to have too small a processing gain to maintain good cross correlation among different user codes [7]. For class-I real-time traffic, connection-oriented transmission is achieved by assigning a set of transmitter-oriented *code channels* and a corresponding receiver, like assigning circuits in a circuit-switched network, to each corresponding mobile after it passes the admission test. (One can imagine a pseudo-noise (PN) code as a channel through which information can be transmitted, hence called a *code channel*. Assigning a code channel to a mobile means that the base station gives the mobile permission to use a particular PN code, and gets ready to receive packets by tuning

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	Class I		Class II-A	Class II-B
Examples	voice	video	remote login	e-mail
Delay	bounded		sensitive	tolerable
Rate	guaranteed		not guaranteed	
	0-8 Kbps	0-64 Kbps	8-128Kbps	
BER	$\leq 10^{-3}$	$\leq 10^{-5}$	$\leq 10^{-7}$	

Table 1: Traffic classification.

a receiver to the PN code.) On the other hand, class-II non-real-time traffic does not require bounded delays nor guaranteed rates. Moreover, it is likely to be bursty and have a high average-to-peak ratio, so it is inefficient to assign a transmission-oriented code channel and a receiver to each mobile for class-II transmissions. We, therefore, adopt a transmission-rate request access scheme, similar, in principle, to the Distributed-Queueing Request Update Multiple Access (DQRUMA) [8], for class-II traffic, in which a mobile, upon generation of a class-II message, requests its transmission in a transmission-request mini-slot, according to a link access scheme similar to the slotted ALOHA, and the base station permits the transmission according to some scheduling principle. The transmission-request scheme is also used for setting up new class-I connections. The third QoS related to error performance is supported by the combination of power control, error control codes, and Automatic Retransmission reQuest (ARQ) scheme (only for class II). A Forward Error Correction (FEC) scheme is used for class I due to the difficulty of ARQ schemes in making delivery-delay guarantees.

Our previous work [9] focused on various QoS guarantees for heterogeneous traffic in wireless LANs using a time-slot allocation-based access protocol. Other researchers have also looked into CDMA to support different types of traffic. The authors of [10] considered how to support voice and data in a CDMA system using the slotted ALOHA for data transmission. A demand-assignment access control scheme was proposed in [8] to support efficient multi-rate transmissions according to the mobile's demand. The authors of [11] proposed a receiver-oriented code channel and receiver scheme for data traffic using the ALOHA access. But, all of these have some form of shortcomings: (1) multi-rate is not supported [10, 11], or (2) class-I traffic is not supported [8].

The paper is organized as follows. Section 2 presents the specification and assumptions of the CDMA system as well as the traffic models under consideration. The communication protocols for both class I and II traffic are described in Section 3. The error control schemes and their related equations are presented in Section 4. Section 5 describes run-time control schemes of the base station including power control of the uplink access, admission control of class-I connection requests, and scheduling of class-II packet-transmission requests. In Section 6, we present both the analytical and simulation results. Finally, the paper concludes with Section 7.

2 Modeling and Specifications

In this section, we describe the MC-CDMA transmitter which is equipped in each mobile, and present the class-I traffic models, which are essential to establish the admission test of class-I connections.

2.1 MC-CDMA Transmitter

Only the uplink (mobile-to-base) is considered in this paper; the downlink (base-to-mobile) can be supported in a similar way but more easily, thanks to the broadcast nature of the downlink. Figure 1 shows the block diagram of a transmitter in each mobile client. The system is based on the MC-CDMA [7, 12] and the Reed-Solomon/convolutional concatenated coding scheme [13, 14]. Each of these components is described below in detail.

A mobile client can transmit its packets at integer multiples of the *basic rate* R_b up to $M \cdot R_b$. In this paper, we assume $R_b = 8$ Kbps, which is equivalent to the rate for a voice transmission, and $M = 16$ to make $M \cdot R_b = 128$ Kbps, the maximum data transmission rate. We start with an m-rate[1] transmitter module, shown in Figure 1 (a), which accepts an incoming message stream at rate mR_b. First, the stream is converted into m basic-rate streams. Each basic-rate stream is encoded by an (N, K, q) Reed-Solomon (RS) encoder. The encoder maps K symbols into N symbols each of which consists of a b-binary sequence, where $b = \log_2 q$. An RS codeword corresponds to a packet, and accommodates $K \cdot b - H$ information bits assuming that an H-bit header is added in each packet. A packet header is appended to all m basic-rate streams once every $K \cdot b - H$ bits to form packets in the serial-to-parallel converter. Note that the RS encoder's input rate is $R_b' = R_b(Kb)/(Kb - H)$. The output of the RS encoder is interleaved by an outer-interleaver on a symbol-by-symbol basis, then encoded by a rate $1/n$ convolutional code with memory size M_{mem}. The output of the convolutional encoder is interleaved by an inner-interleaver on a bit-by-bit basis. The inner-interleaver renders the fading channel memoryless while the outer-interleaver randomizes burst errors at the output of the convolutional (Viterbi) decoder.

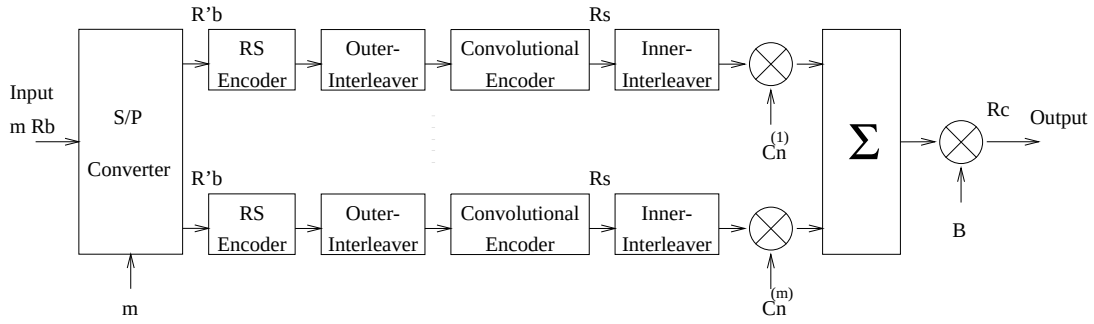
When a mobile enters the cell under consideration,[2] it is assigned a primary pseudo-noise (PN) code, which is used exclusively by the mobile as long as it remains within the cell. Each of m streams is multiplied by one of orthogonal codes. The set of orthogonal codes is generated by a sub-code concatenation scheme: when C_n^{PN} is the primary PN code of a mobile n, the spreading codes $\{C_n^{(i)}, i = 1, 2, \cdots, M\}$ are obtained by

$$C_n^{(i)} = C_n^{PN} \times D_i, \quad D_i \perp D_j, \quad i \neq j, \tag{1}$$

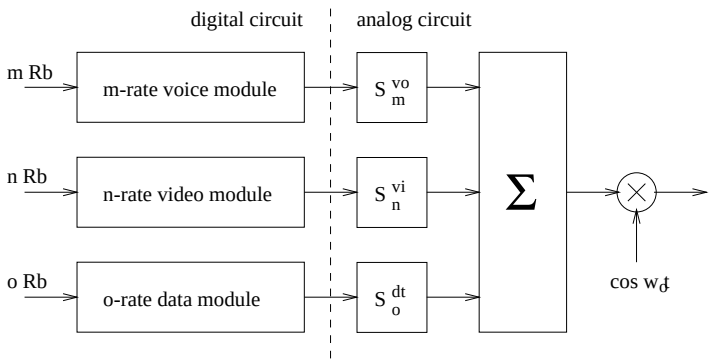
where D_i's are from a set of Walsh (i.e., orthogonal) codes. Note that $C_n^{(i)} \perp C_n^{(j)}, i \neq j$, is also guaranteed. This scheme avoids self-interference [7] because the codes arrive simultaneously at the base station, thus preserving the orthogonality. Thanks to the absence of this self-interference, multi-rate transmission can be achieved with a smaller power than

[1] m-rate means 'rate mR_b' throughout this paper.

[2] Physically, when the user turns on the switch of the handset or the mobile is handed off from an adjacent cell.



(a) an m-rate transmitter module



(b) the integrated transmitter

Figure 1: Block diagram for a Multi-Code CDMA transmitter: (a) an m-rate transmitter module with RS/convolutional concatenated encoders and interleavers, where $R_b' = R_b(Kb)/(Kb-H)$; (b) the integrated transmitter, where $m+n+o \leq M$.

the basic rate [7].[3] After being spread by orthogonal codes, all parallel streams are summed together, then multiplied by a base station code, which is used to indicate which base station the mobile is communicating with.

Each mobile can transmit voice, video, and data traffic at a time with the condition that the transmission rate in total is not over M. As shown in Figure 1 (b), for each traffic type, separate transmitter module is used because each traffic type requires different transmission power level due to different rates and BER requirements as described later. Note that when transmitter modules are in use, orthogonal codes $C_n^{(i)}$ from the same set of M codes will be used. The blocks S_m^{vo}, S_n^{vi}, and S_o^{dt} represent the analog amplifiers with those values of power level for each of m-rate voice traffic, n-rate video traffic, and o-rate data traffic, respectively. Note that blocks before the amplifiers are implemented with a digital circuit while the others are with an analog circuit. All the amplified outputs of modules are summed together, multiplied by the carrier frequency, then transmitted.

2.2 Class-I Traffic Modeling

In our proposed scheme, class-I traffic is handled by assigning specific code channels to mobiles similarly to assigning frequency slots in FDMA, time slots in TDMA, and more generally, circuits in a circuit-switched network. However, the underlying concept is totally different because in CDMA systems, all mobiles share the entire frequency and time slots. Depending on their traffic characteristics, we can admit many more mobiles simultaneously into the system than the physical link capacity without violating the required BER because the CDMA system is interference-limited. In a sense, the CDMA system achieves statistical multiplexing. However, to achieve the required error performance, we need to control the admission of class-I traffic which, in turn, requires appropriate traffic modeling. Class-II traffic does not affect the admission control of class-I traffic because the transmission of each class-II packet is permitted by the base station, depending on the available bandwidth left unused by class-I traffic in our protocol.

It is well-known that voice traffic is characterized by an "on-off" model since a speech signal is in either *talking* or *silent* mode during the conversation of a mobile client. Each voice mobile is modeled by a two-state Markov chain where one state represents ON (transmitting at rate R_b) and the other represents OFF (idle). The utilization ρ_{vo} (i.e., fraction of time during which the voice mobile is active) is assumed to be 0.5. Given the number K_{vo} of voice mobiles, the number K_{vo}^a of active voice mobiles at current time can be computed by a binomial distribution:

$$P_{K_{vo}^a}(k) = \binom{K_{vo}}{k}(0.5)^{K_{vo}}. \qquad (2)$$

Since there doesn't exist any proper model for low-rate video coding, we adopt a very simple model to represent the variable bit rate (VBR) characteristics of video traffic: each video mobile is modeled by a three-state Markov chain where in the first state, the mobile transmits packets at the rate of $4R_b$; in the second, at the rate $6R_b$; and in the third, at the rate $8R_b$ (= 64 Kbps). The mobile will reside in

[3] If multi-path fading is considered, the orthogonality is not assured to be preserved. But there is still some interference advantage for multi-rate transmissions because the orthogonality is preserved among the multi-path components which arrive at the base station simultaneously.

each of the three states with probability 0.25, 0.5, and 0.25, respectively. Given the number K_{vi} of video mobiles, the number $K^a_{vi}(j)$ of j-rate video mobiles is given by a 3rd-order multinomial distribution:

$$P_{K^a_{vi}(4),K^a_{vi}(6),K^a_{vi}(8)}(k_4,k_6,k_8) = \binom{K_{vi}}{k_4,k_6,k_8}(0.5)^{k_6}(0.25)^{k_4+k_8}, \quad (3)$$

where $k_4+k_6+k_8 = K_{vi}$. For other video traffic models, this method can be used for admission control.

3 Protocol Description

Connection-oriented transmission is achieved by assigning a set of transmitter-oriented code channels for each class-I mobile. On the other hand, whenever a mobile has class-II messages to transmit, it requests code channels via a receiver-oriented transmission-request code channel (or a piggyback rate-request code channel), then, according to the base station's scheduling policy, the transmission is scheduled and permitted. Note that for a mobile to send a packet through a code channel, the base station should be ready to receive it by tuning a receiver to the corresponding PN code. As shown in Figure 2, our scheme is based on the uplink and downlink frame formats, which are drawn basically from [8]. The uplink time-axis is divided into frames which consist of a mini-slot for ACK/NAK, a contention-based transmission-request mini-slot, a piggyback rate-request mini-slot, and a packet-transmission slot. On the other hand, the downlink time-axis is divided into frames which consist of a mini-slot for ACK/NAK transmission, a mini-slot for result announcements of contention-based transmission-requests, a mini-slot for uplink packet-transmission permissions, and a slot for downlink packet transmission.

3.1 Contention-Based Transmission-Request Accesses

A set of receiver-oriented code channels in each contention-based transmission-request mini-slot are used for class-II transmission-request access: a finite number, say, N^{rqst}_{rcvr}, of receivers (and so, code channels) are dedicated for mobiles' transmission-request accesses. An ALOHA-like access protocol is used for the request accesses. This contention-based request is basically used by a mobile which is not permitted to transmit any packet during the next frame and falls into one of the following three cases: (1) when the mobile wants to establish a new class-I connection; (2) when it generated new class-II messages after transmitting all messages for which it previously requested; and (3) when it wishes to transmit class-II messages at a higher rate than the rate it previously requested due to the generation of new messages. Such a mobile waits for the next mini-slot then decides whether to send the request or not with probability p_r (i.e., in a p_r-persistent manner). If it decided to send, the mobile chooses one of N^{rqst}_{rcvr} code channels, and transmits the request via the code channel in the mini-slot. Each request includes the relevant information such as mobile ID, class type (voice, video, class II-A, or class II-B), and the required transmission rate, up to MR_b (or equivalently, the number of packets, up to M packets, to be transmitted in the next frame) if it is for class-II traffic. The packets to be retransmitted by the ARQ scheme will not be included in the calculation of the required transmission rate.

A request can be received erroneously by the base station for the following three reasons. First, the total interference level exceeds the link capacity[4] and many of the requests in the mini-slot may be received with error. Second, the request collided with another request transmitted in the same request code channel, and only the collided requests are received with error. Third, a request may be received erroneously due to random noise and interference. Only successful requests are broadcast to the mobiles through the downlink channel before the next mini-slot. The value of p_r is determined by each mobile in a distributed fashion. For the first attempt of a message transmission request, p_r is set to 1. Upon failure of a request, however, the value of p_r is decreased to

$$p_r := \frac{p_r}{p_r+1}, \quad (4)$$

following $1, 1/2, 1/3, \cdots$ (i.e., the "harmonic" backoff) [8]. Each failed request should be retransmitted in a p_r-persistent manner until it is successfully received, and a mobile with a failed request is said to be *backlogged*.

3.2 Collision-Free Piggyback Class-II Rate Requests

When a mobile is scheduled and permitted to transmit class-II packets in the packet-transmission slot of the subsequent frame, it is supposed to update the required transmission rate (for the next scheduled transmission-slot of that mobile) through the piggyback rate-request mini-slot. These piggyback requests are collision-free because, during the piggyback mini-slots, mobiles use code channels dedicated to the mobiles in the subsequent transmission slot. Through the piggyback mini-slot, the currently-transmitting mobile can request a transmission rate including the rate of newly-generated messages without involving the contention-based request accesses, thus reducing the average request access delay. A class-I mobile also can request the transmission of newly-generated class-II messages through the piggyback rate-request mini-slot, so class-I mobiles do not need the contention-based transmission-request accesses. A mobile which has voice and/or video connections currently can request to transmit class-II data up to rate $15R_b$ (with a voice connection), $8R_b$ (with a video connection), and $7R_b$ (with both voice and video connections).

Each piggyback request includes the information such as the priority level (class II-A or II-B) and requested transmission rate. As shown in Figure 2, as many piggyback requests as the number of packet transmissions permitted for a mobile during a frame can be transmitted within a piggyback mini-slot. However, up to two piggyback requests will, in reality, be transmitted: one for each of class II-A and II-B. If there is only one packet transmission scheduled, only one request can be made where class II-A is given priority over class II-B.

3.3 Class-I Connection Establishment

Upon receiving a request to establish a class-I connection in a mini-slot, the base station performs an admission test on the request according to the link capacity. The admission test checks if the total required bandwidth for class-I traffic (or *equivalent bandwidth* BW_{eq} defined in Section 5) is less than, or equal to, the link capacity C. If the test result is positive, the base station admits the mobile via the transmission-permission mini-slot, and assigns a receiver to the corresponding mobile while the receiver tunes to the mobile's code channels using the mobile's primary code and the

[4] Due to the CDMA's soft capacity characteristic, the request might be received correctly even under an over-threshold situation.

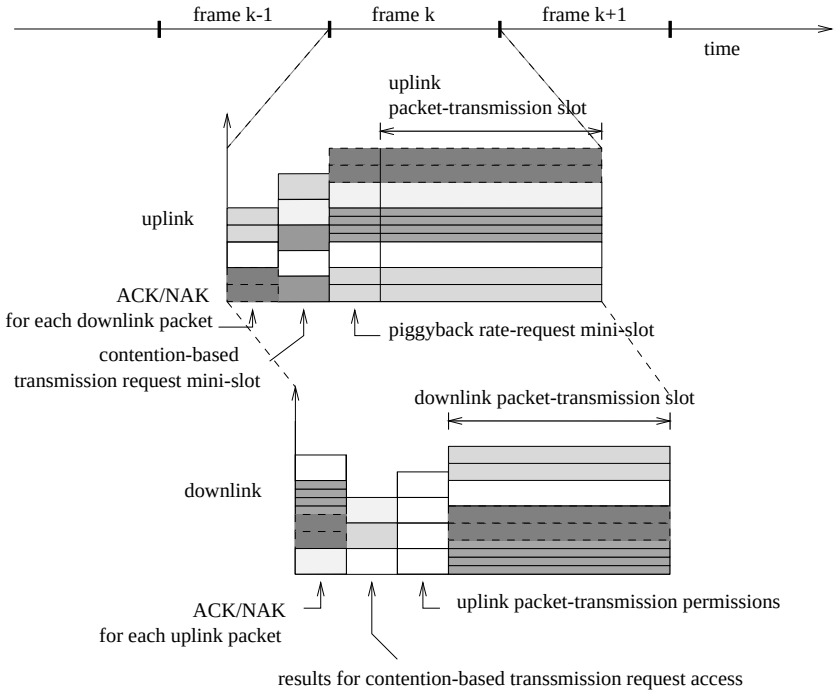


Figure 2: The uplink and downlink frames along the time-axis.

corresponding orthogonal codes, then the mobile continues to transmit packets whenever it wants. If the test result is negative, the base station declares *connection blocking*, and notifies it to the mobile.

An admitted mobile will transmit its packets in each packet-transmission slot with the power level controlled by the base station, depending on the traffic condition. The transmission rate (or the number of packets transmitted in a slot) can vary according to the amount of traffic generated. Because each class-I mobile can transmit packets at a rate up to the contracted value (i.e., R_b for voice and $8R_b$ for video) after its admission into the system, there is no queueing delay, so the delivery delay is bounded, i.e., the first QoS is guaranteed.

3.4 Class-II Message Transmission

Each of successful transmission/rate requests is queued in one of two base station service queues, i.e., one for class II-A and the other for class II-B. The base station serves (i.e., gives the transmission permission to) the requests according to a transmission scheduling policy with the bandwidth available for class-II, BW_{dt} $(= C - BW_{eq})$, which depends on the admitted pair (K_{vo}, K_{vi}) of voice and video mobiles. The relation between BW_{dt} and the transmission scheduling of class-II traffic is detailed in Section 5.3. Transmission permissions for the subsequent frame are announced to mobiles in the middle of each slot via the downlink as shown in Figure 2. After sending the transmission permissions, the base station assigns a number of receivers, so that they tune to the scheduled mobiles' code channels. Each permission includes such information as mobile ID and transmission rate.

Since class-II traffic uses a selective-repeat ARQ scheme, the feedback from the base station should be announced. For each transmission of packets, ACK (acknowledge) or NAK (non-acknowledge) is fed back. NAKed packets are retransmitted with priority over other packets in the next frame during which the mobile is permitted to send packets; if a mobile has NAKed packets, only after retransmitting all of them, it will transmit other newly-generated packets. An erroneous packet transmitted in the i-th frame can be retransmitted as early as in the $(i+2)$-th frame because the corresponding NAK is fed back to the mobile during the $(i+1)$-th frame.

4 Error Control Schemes

As described in Section 2, an RS/convolutional concatenated coding is adopted for error protection. For class-I traffic, an error-correction-only decoder is used while for class-II, an error-correction-and-detection decoder is used for retransmission of corrupted packets by selective-repeat ARQ.

When the binary phase shift keying (BPSK) and the soft-decision Viterbi decoder are used, the bit error probability at the output of the convolutional decoder can be upper-bounded [15] by:

$$P_b < \sum_{d=d_{free}}^{\infty} b_d Q(\sqrt{2\gamma_o d}), \tag{5}$$

where d_{free} is the free distance of the convolutional code, b_d is the total number of nonzero information bits on all weight-d paths in the trellis diagram of the convolutional code, and γ_o is the signal-to-interference ratio of the received convolutionally-coded symbol.

We adopt a bounded distance RS decoder of the (N, K, q) RS code over GF(q) with the error correction capability t $(= \lfloor (N-K)/2 \rfloor)$. A symbol is mapped to a b-binary symbol, so $q = 2^b$. The decoder looks for a codeword within distance t of the received word; if there is such a codeword, the decoder finds it, and if not, the decoder reports "decoding failure,"

and initiates retransmission of the packet. If P_s denotes the symbol error rate at the input of the RS decoder, then the probability P_T of total error of the RS decoder which includes both decoding error (i.e., decoding into a wrong codeword) and decoding failure (i.e., inability to decode) is given by

$$P_T = \sum_{i=t+1}^{N} \binom{N}{i} P_s^i (1-P_s)^{N-i}, \tag{6}$$

where $t = \lfloor (n-k)/2 \rfloor$ is the RS error-correction capability, and P_s can be upper-bounded by

$$P_s \leq bP_b, \tag{7}$$

with the bit error probability P_b of the Viterbi decoder given in Eq. (5). Now, the probability P_E of RS decoding error and the probability P_F of RS decoding failure are given [16] by:

$$P_E \leq P_T \sum_{i=0}^{t} \binom{N}{i} (2^b-1)^{i-(N-K)}, \tag{8}$$

$$P_F = P_T - P_E. \tag{9}$$

The decoding failure will be informed to the mobile via the downlink, and will trigger the retransmission of the packet by the selective-repeat ARQ scheme.

Meanwhile, the symbol error probability due to the RS decoder error (in contrast to decoding failure) can be obtained as:

$$P_{es,II} = \sum_{i=t+1}^{N} P_{es}(i) P_E(i) q_i, \tag{10}$$

where $P_{es}(i)$ is the symbol error probability given i channel errors and decoder error, $P_E(i)$ is the decoder error probability given i channel errors, and q_i is the probability that the error pattern has weight i. $P_{es}(i)$ is simply expressed as $P_{es}(i) = i/N$ while $q_i = \binom{N}{i} P_s^i (1-P_s)^{N-i}$ because the channel is symmetric. $P_E(i)$ is upper-bounded by $\sum_{s=0}^{t} \binom{N}{s} (q-1)^{s-(N-K)}$ [16]. Now, Eq. (10) is reduced to

$$P_{es,II} = \sum_{i=t+1}^{N} \frac{i}{N} \binom{N}{i} P_s^i (1-P_s)^{N-i} \cdot \sum_{s=0}^{t} \binom{N}{s} (q-1)^{s-(N-K)}. \tag{11}$$

Finally, we obtain the bit error probability at the output of the RS decoder in the error-correction-and-detection mode for class-II traffic as [15]:

$$P_{eb,II} = \frac{2^{b-1}}{2^b-1} P_{es,II}. \tag{12}$$

When the decoder is used for class-I traffic, the decoder will be switched to the error-correction-only mode, in which, instead of declaring a decoding failure, the decoder will just bypass the received word to the output of the decoder. In this case, the symbol error probability is obtained as:

$$P_{es,I} = \sum_{i=t+1}^{N} \frac{i}{N} \binom{N}{i} P_s^i (1-P_s)^{N-i}, \tag{13}$$

and accordingly, the bit error probability at the output of the RS decoder in the error-correction-only mode for class-I traffic is

$$P_{eb,I} = \frac{2^{b-1}}{2^b-1} P_{es,I}. \tag{14}$$

5 Run-Time Control Schemes

At run-time, the base station exercises the following three types of control: (1) power control of each packet transmission; (2) admission control of newly-requested class-I connections; and (3) class-II traffic transmission-permission control. For the following schemes, a *unit of bandwidth* is the amount of bandwidth to support an active voice mobile.

5.1 Power Control

We exploit the property of sub-code concatenation while meeting different error performance requirements for different traffic classes. Given modulation and coding schemes, the BER of a traffic class is determined directly from the received signal-to-interference ratio (SIR) as shown in the previous section. So, for each type of traffic, a *target* SIR of the convolutionally-coded symbol received is assigned according to its required BER. Since there is no self-interference for higher-rate mobiles, each code channel of higher-rate mobiles can transmit packets with less power than the basic-rate mobile with the same target SIR. So, when the system can accommodate K_{dt}^1 1-rate data-transmission mobiles without any other mobiles, it can accommodate more than K_{dt}^n/n n-rate data-transmission mobiles.

Now, we assume that there are K_{vo} voice mobiles with target SIR $\hat{\gamma}_o^{vo}$, and K_{vi} video mobiles with target SIR $\hat{\gamma}_o^{vi}$. In the next frame, $K_{dt,k}^a$ k-rate data mobiles, $k = 1, 2, \cdots, M$ $(= 16)$, with target SIR $\hat{\gamma}_o^{dt}$ are scheduled to transmit packets. Note that a single mobile can have all of voice, video, and data traffic at a time. First, the required convolutionally-coded symbol energies at the receiver for a voice-only (not video nor data) mobile, can be obtained by modifying the result from [8] which didn't consider any traffic classification:

$$\hat{E}_s^{vo} = \frac{1.5PGN_o}{(\hat{\gamma}_o^{vo} + 1.5PG)/\hat{\gamma}_o^{vo} - I_{est}}, \tag{15}$$

where I_{est} is the total estimated interference level during the subsequent slot as a function of K_{vo}, K_{vi}, and $K_{dt,k}^a$, as defined later, N_o is the single-side power density of white Gaussian noise, and PG is the processing gain of the CDMA system. The processing gain is defined as $PG = R_c/R_s$, where R_c is the chip rate of the PN sequence and R_s is the convolutionally-coded symbol rate (as shown in Figure 1).

For a mobile which will transmit voice at i-rate, video at j-rate, and data at k-rate, where $i + j + k \leq M$, during the following slot, the corresponding symbol energies (i.e., $E_s^{vo}(i,j,k)$ for voice, $E_s^{vi}(i,j,k)$ for video, and $E_s^{dt}(i,j,k)$ for data) relative to $\hat{E}_s^{vo}$ can be given by

$$\begin{aligned} E_s^{vo}(i,j,k) &= \beta_{vo}(i,j,k)\hat{E}_s^{vo}, \\ E_s^{vi}(i,j,k) &= \beta_{vi}(i,j,k)\hat{E}_s^{vo}, \\ E_s^{dt}(i,j,k) &= \beta_{dt}(i,j,k)\hat{E}_s^{vo}, \end{aligned} \tag{16}$$

where the relative energy ratios are

$$\beta_{vo}(i,j,k) = \frac{\hat{\gamma}_o^{vo}/1.5PG + 1}{(i \cdot \hat{\gamma}_o^{vo} + j \cdot \hat{\gamma}_o^{vi} + k \cdot \hat{\gamma}_o^{dt})/1.5PG + 1},$$

$$\beta_{vi}(i,j,k) = \frac{\hat{\gamma}_o^{vi}}{\hat{\gamma}_o^{vo}}\beta_{vo}(i,j,k),$$

$$\beta_{dt}(i,j,k) = \frac{\hat{\gamma}_o^{dt}}{\hat{\gamma}_o^{vo}}\beta_{vo}(i,j,k). \quad (17)$$

Note that $\hat{E}_s^{vo} = E_s^{vo}(1,0,0)$. Now, the value of I_{est} is given by

$$I_{est} = BW_{eq} + \sum_{k=1}^{M} k\beta_{dt}(0,0,k)K_{dt,k}^a, \quad (18)$$

where BW_{eq} is the equivalent bandwidth which was reserved for class-I traffic, and $K_{dt,k}^a$ is the scheduled number of k-rate data mobiles. The equivalent bandwidth BW_{eq} is determined for a given pair (K_{vo}, K_{vi}) of admitted voice and video mobiles during the admission control phase as described in the next subsection. The power level to be calculated is for the next frame, so the base station does not know the actual rates of the class-I connections in that frame, nor whether mobiles currently with scheduled data-transmissions will transmit voice and/or video at a time. Hence, the base station just assumes that BW_{eq} is always consumed by class-I connections and the mobiles with scheduled data-transmissions will not transmit voice and video at a time (hence, $\beta_{dt}(0,0,k)$ is included in Eq. (18)). We err on the safe side because $\beta_{dt}(0,0,k) \geq \beta_{dt}(i,j,k)$ for all i and j.

In Eq. (15), the following condition should hold:

$$I_{est} \leq \frac{\hat{\gamma}_o^{vo} + 1.5PG}{\hat{\gamma}_o^{vo}} - \Delta_1 = C, \quad (19)$$

where Δ_1 is called the *link capacity reservation factor* as it determines the bandwidth that should remain unallocated, and the value C represents the link capacity. Note that (i) if $\Delta_1 = 0$ then $\frac{\hat{E}_s^{vo}}{N_o}$ could be infinite when Eq. (19) is satisfied with the equality, and (ii) using Δ_1, we can bound the maximum transmission energy. The required power levels of a voice-only mobile for the subsequent frame calculated using Eq. (15) is announced to the mobiles through the downlink, and the mobiles will calculate their corresponding powers using Eqs. (16) and (17).

5.2 Admission Control of Class-I Connections

Before admitting each class-I connection, the corresponding mobile needs to pass the admission test, determining if the new connection can be established without affecting the QoS of the existing connections. The admission test can be derived easily by modifying Eqs. (18) and (19) as:

$$BW_{eq} \leq \frac{\hat{\gamma}_o^{vo} + 1.5PG}{\hat{\gamma}_o^{vo}} - \Delta_1 - \Delta_2 = C - \Delta_2, \quad (20)$$

where Δ_2 is the link capacity reservation factor for class-II traffic which is a design or dynamic parameter that depends on the traffic condition. BW_{eq} is a function of the pair (K_{vo}, K_{vi}) including the newly-requested connection.

Suppose there are K_{vo}^a active voice mobiles and $K_{vi,j}^a$ j-rate video mobiles, such that $\sum_{j=0}^{8} K_{vi,j}^a = K_{vi}$ in a given slot. Assuming that all the available bandwidth BW_{dt} $(= C - BW_{eq})$ for class-II traffic is used by class-II traffic and a mobile transmit only one type of traffic (i.e., the worst case in terms of required power level), for a specific value of BW_{eq}, SIRs for voice mobiles, j-rate video mobiles, and k-rate data mobiles, respectively, are given by:

$$\gamma_o^{vo} = \left[\frac{N_o}{\hat{E}_s^{vo}} + \frac{K_{vo}^a - 1 + \sum_{j=1}^{8} \beta_{vi}(0,j,0) j K_{vi,j}^a + BW_{dt}}{1.5PG}\right]^{-1}, \quad (21)$$

$$\gamma_o^{vi,j} = \frac{1.5PG}{\left(\frac{1.5PG}{\gamma_o^{vo}} + 1\right)/\beta_{vi}(0,j,0) - j}, \quad (22)$$

$$\gamma_o^{dt,k} = \frac{1.5PG}{\left(\frac{1.5PG}{\gamma_o^{vo}} + 1\right)/\beta_{dt}(0,0,k) - k}. \quad (23)$$

The BERs for each state $(K_{vo}^a, K_{vi,j}^a, j = 0, \cdots, 8)$ can be obtained by using the equations in Section 4. Because we can calculate the distributions of active voice mobiles and actual video rates from Eqs. (2) and (3), the average BERs can be obtained for a given set of $(K_{vo}, K_{vi}, BW_{eq})$. Using an iterative search, the minimum BW_{eq} satisfying the admission condition in Eq. (20) and all of the required BERs can be obtained for a given pair (K_{vo}, K_{vi}). If there does not exist such a BW_{eq}, the corresponding numbers of voice and video mobiles cannot be admitted into the system. Because it is computationally expensive to calculate BW_{eq} for each pair of voice and video mobiles, one can calculate it off-line and store the results in the base station to be used at run-time.

5.3 Class-II Transmission Permission Control

Class-II traffic transmission-permission is controlled under a round-robin scheduling policy with a set of four prioritized queues. When there are M_o mobiles in the cell, the base station is equipped with the following four queues (in that order of priority): (1) a first-in-first-out (FIFO) queue for class II-A retransmissions; (2) a class II-A round-robin queue (with M_o entities, the number of mobiles in the cell); (3) a FIFO queue for class II-B retransmissions; and (4) a class II-B round-robin queue (with M_o entities). The i-th entity of a round-robin queue, corresponding to mobile i, contains its requested transmission rate $R_{req}[i]$, and is served in a round-robin fashion. If a number of transmitted packets by a mobile results in an error, the mobile ID and the number of the packets (R_{ret}) will be placed at one of two FIFO queues, depending on the packets' priority, for retransmission. Figure 3 shows a pair of FIFO and round-robin queues for M_o mobiles in the cell.

First, the permitted transmission-rate index $R_{per}[i]$ for mobile i, and the residual bandwidth BW_{res} are defined. For the subsequent frame, transmission-permissions are scheduled recursively as follows.

1. With the value BW_{dt} of the available bandwidth for class-II traffic, the residual bandwidth $BW_{res} := BW_{dt}$, and for all i, $R_{per}[i] := 0$.

2. For the next step, a mobile is selected according to the order of the class II-A FIFO queue, the class II-A round-robin queue, the class II-B FIFO queue, and the class II-B round-robin queue, where the requested transmission rate is determined by R_{ret} for FIFO queues and $R_{req}[i]$ for round-robin queues, respectively.

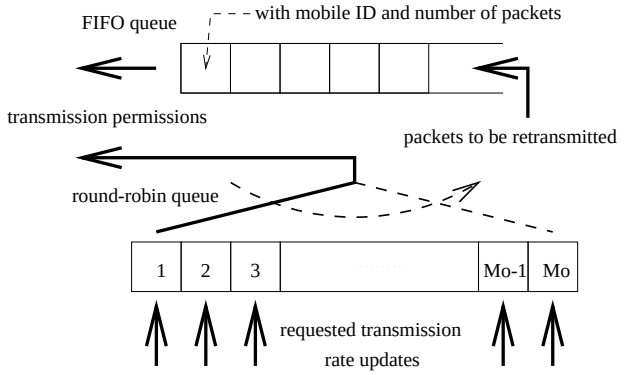


Figure 3: A FIFO queue for retransmission and a round-robin queue.

3. For an m-rate transmission-request of mobile i, $R_{tmp} := R_{per}[i]$. If $R_{tmp} \neq 0$, then the residual bandwidth is increased as:

$$BW_{res} := BW_{res} + R_{tmp} \cdot \beta_{dt}(0, 0, R_{tmp}). \quad (24)$$

4. The available rate R_{ava} using the residual bandwidth BW_{res} is calculated as:

$$R_{ava} = \frac{(1.5PG/\hat{\gamma}_o^{dt})BW_{res}}{1 + 1.5PG/\hat{\gamma}_o^{vo} - BW_{res}}. \quad (25)$$

5. Check if $m + R_{tmp} \leq R_{ava}$, then the transmission-permission rate is set accordingly (i.e., if yes, then $R_{per}[i] := m + R_{tmp}$, else $R_{per}[i] := \lfloor R_{ava} \rfloor$).
Then, the residual bandwidth is recalculated as:

$$BW_{res} := BW_{res} - R_{per}[i] \cdot \beta_{dt}(0, 0, R_{per}[i]). \quad (26)$$

6. If $BW_{res} \geq \beta_{dt}(0, 0, 1)$ then go to Step 3 after selecting the next mobile using Step 2, else go to the next step.

7. The value of $R_{per}[i]$ is the transmission-rate to be permitted for mobile i, so the values of $K^a_{dt,k}$, $k = 1, 2, \cdots, M$, can be calculated. Then, calculate the voice-only mobile's power level using Eqs. (15) and (18). The transmission permission and power level are announced to mobiles through the downlink.

The reason why the residual bandwidth is increased in Step 3, and decreased again in Step 5 is that the necessary bandwidth is not linearly dependent on the transmission rate. In order to err on the safe side, the base station assumes that the scheduled data mobiles will transmit class-II messages only, not voice nor video, and hence $\beta_{dt}(0, 0, k)$'s are used.

6 Numerical Results and Discussions

To derive numerical results, we used the best rate 1/2 convolutional code with $d_{free} = 10$ and $(256, 240, 256)$ extended RS code with the error correction capability $t = 8$. The values of $\{b_d\}$ in Eq. (5) are obtained from [17]. Notice that the total code rate is $1/2 \cdot 240/256 = 15/32 \approx 0.47$. The PN codes with the processing gain $PG = 128$ and the link capacity reservation factor $\Delta_1 = 1$ are used, so we the link capacity $C \approx 205$ (units of bandwidth) is obtained from Eq. (19).

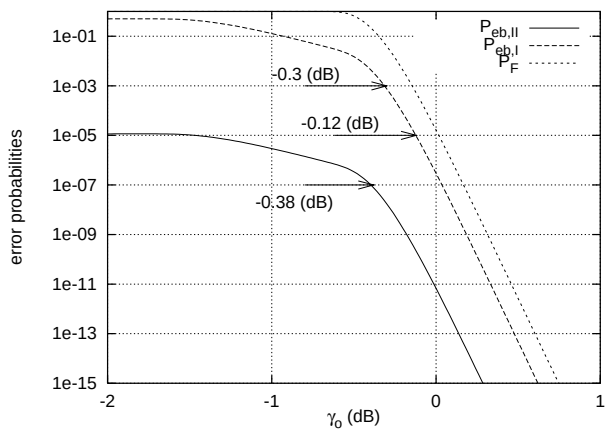


Figure 4: Error probabilities vs. SIR γ_o for the convolutional/RS concatenated coding system.

6.1 Error Performance and Target SIRs

In Figure 4, we plotted the following three error probabilities as the SIR γ_o increases:

Bit error probability $P_{eb,I}$ at the output of the RS decoder when the error-correction-only mode for class-I traffic is used;

Bit error probability $P_{eb,II}$ at the output of the RS decoder when the error-correction-and-detection mode for class-II traffic is used;

Decoding failure probability P_F (i.e., the probability of packet retransmission) when the error-correction-and-detection mode for class-II traffic is used.

For a given SIR, $P_{eb,II}$ is lower than $P_{eb,I}$ because the decoding failure is declared for the packet with uncorrectable errors, thus triggering its retransmission. According to each BER requirement in Table 1, we can determine the target SIR for each traffic type. For class-I voice and video, we obtain $\hat{\gamma}_o^{vo} = -0.3$(dB) and $\hat{\gamma}_o^{vi} = -0.12$(dB), respectively, from the $P_{eb,I}$ curve because the error-correction-only scheme is used in the RS decoder. For class-II data, we obtain $\hat{\gamma}_o^{dt} = -0.38$(dB) from the $P_{eb,II}$ curve where the error-correction-and-detection scheme is used.

Now, we study the relationship between the power level and the throughput for class-II traffic. From Figure 4, we obtained $\hat{\gamma}_o^{dt} = -0.38$(dB), but observed that the decoding failure (thus retransmission) probability is very high (larger than 0.1) at that SIR level. So, to make the throughput higher, it is necessary to increase $\hat{\gamma}_o^{dt}$. Assume that there are only the mobiles with data to be transmitted at the basic rate, and the ideal selective-repeat ARQ is used. We obtain the maximum total throughput at peak traffic for class II, which is defined as the expected number of the total packets successfully received at the receiver during one slot in the presence of more class-II traffic demands than the capacity available for class II:

$$W_{max} = (1 - P_F)K^*_{dt,1}, \quad (27)$$

where $K^*_{dt,1}$ is the maximum number of the mobiles which can transmit the data simultaneously. For a specific value of BW_{dt}, $K^*_{dt,1}$ can be expressed, by modifying Eq. (19), as follows:

$$K^*_{dt,1} = \lfloor BW_{dt}/\beta_{dt}(0, 0, 1) \rfloor. \quad (28)$$

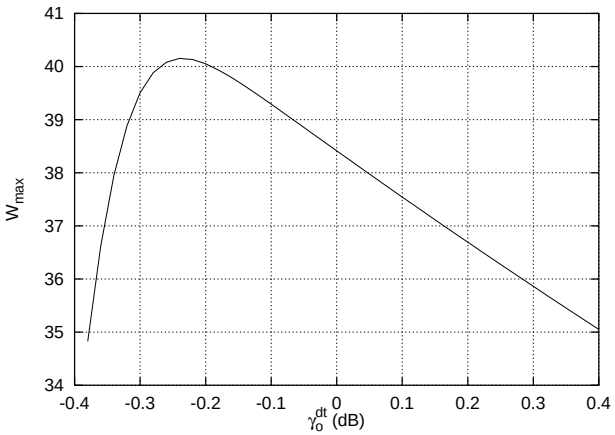


Figure 5: Maximum total throughput at peak traffic W_{max} vs. class-II traffic target SIR $\hat{\gamma}_o^{dt}$ with $BW_{dt} = 0.2C$.

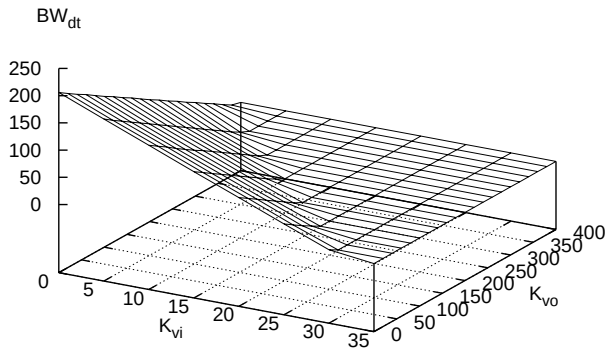


Figure 6: Admission regions for the pairs (K_{vo}, K_{vi}).

There is a tradeoff between $\hat{\gamma}_o^{dt}$ and $K_{dt,1}^*$, i.e., for a low $\hat{\gamma}_o^{dt}$, $K_{dt,1}^*$ would be high, and vice versa. But, because P_F would be high for low $\hat{\gamma}_o^{dt}$, there might exist an optimum $\hat{\gamma}_o^{dt}$ that maximizes W_{max}. In Figure 5, we find the optimum $\hat{\gamma}_o^{dt}$ to be -0.24 (dB). We can easily see that the result will be the same for different transmission rates and BW_{dt} values. The target BER of class II is also modified to 4.5e-9 from 1.0e-7 by observing Figure 4 to maximize the throughput of class-II traffic.

6.2 Admission Region for Class-I Connections

Using three target SIRs and the procedure in Section 5.2, we obtained the admission region shown in Figure 6, where the link capacity $C \approx 205$. The surface represents the bandwidth, BW_{dt}, which can be used for class II, given the admitted voice and video mobiles (K_{vo}, K_{vi}). For a given capacity reservation factor Δ_2 for class II, the pairs (K_{vo}, K_{vi}) with $BW_{dt} \geq \Delta_2$ can be admitted to the system. From the numerical results, we found that a voice mobile consumes roughly 0.55 unit of bandwidth while a video mobile consumes 6.6 units of bandwidth, and we can use the following equation for the admission test:

$$\lceil 0.55K_{vo} + 6.6K_{vi} \rceil \leq C - \Delta_2. \qquad (29)$$

It can reduce the storage cost in the base station at the expense of some possible connection blockings due to the inaccuracy caused by using the ceiling function in the above equation.

6.3 Class-II Performance

Finally, we consider class-II performance in terms of the average message transmission delay via simulations. A message transmission delay, where a message consists of a number of packets, is defined by the time duration from the generation of a message to the completion of the last packet transmission in the message. We used the following assumptions and parameters for the simulations:

1. There are 50 mobiles which generate only class-II traffic throughout the simulation.
2. N_{rcvr}^{rqst} $(= 25)$ transmission-request access channels are used.
3. Each transmission request is received correctly unless it collides with another request in the same code channel.
4. Throughout the simulation, the value of BW_{dt} is kept constant; the class-I connections are not changed.
5. The number of packets in a class II-A (II-B) message is geometrically distributed with the average 2 (18).
6. For a total message generation rate λ (messages/frame), the class II-A (II-B) messages are generated independently in each of 50 mobiles according to a Poisson process with rate $0.9\lambda/50$ $(0.1\lambda/50)$.

Note that the offered load, defined by the overall expected number of the packets to be transmitted within a frame, for the overall message generation rate λ (messages/frame) is $(0.9 \cdot 2 + 0.1 \cdot 18)\lambda = 3.6\lambda$ (packets/frame), in which a half is for class II-A and the other half is for class II-B.

A message transmission delay consists of three components: (1) a request access delay, i.e., from the generation of a message to the time at which the mobile notifies the required transmission rate to the base station successfully through a contention-based transmission-request mini-slot or a contention-free piggyback rate-request mini-slot. Physically, a mobile requests the required transmission rate, not the transmission of each message, but from an analytical point of view, we can imagine that a successful transmission of a rate request means the successful transmission-requests for all the messages generated already; (2) a queueing delay of the request in the base station queues, or equivalently, the delay from the successful rate request after the generation to the transmission of the first packet in the message; and (3) a transmission delay, i.e., the time from the transmission of the first packet to the completion of the last packet transmission.

First, we consider the request access delay of the protocol. For a comparative evaluation, we also consider an ideal retransmission probability control scheme. In conventional ALOHA-type data networks, the optimal control schemes are shown to have the retransmission probability p_r as an inverse function of the number of the backlogged mobiles [18]. In a given contention-based request-access mini-slot, N_{rcvr}^{rqst} requests can be received successfully. So, if the number of

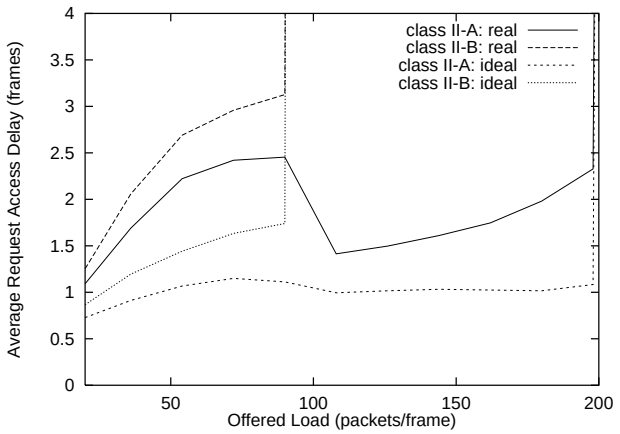


Figure 7: Average request access delay vs. offered load with $BW_{dt} = 100$.

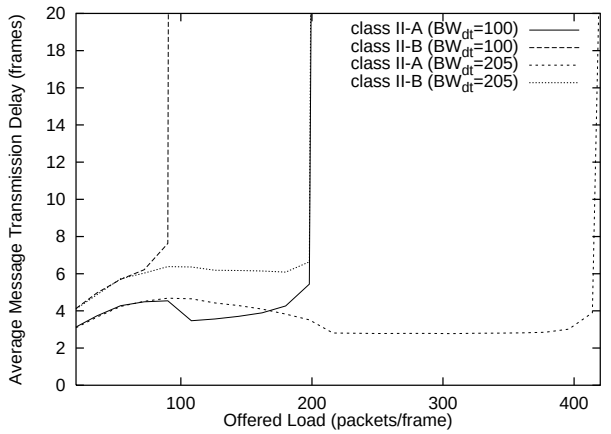


Figure 8: Average message transmission delay vs. offered load.

backlogged mobiles N_{back} is known to the mobiles — which is not practical with our system — the transmission probability p_r can be dynamically assigned to each mini-slot as:

$$p_r = \min\left\{\frac{N_{rcvr}^{rqst}}{N_{back}},\ 1\right\}. \tag{30}$$

Figure 7 plots the average request access delay as the offered load increases for each combination of the real and ideal schemes with class II-A and II-B. The average request access delay is always greater than, or equal to, 0.5 because the message generation times are uniformly distributed over a frame time. Because a request for class II-A is given priority over that for class II-B, the request access delays appear to reflect that fact for both schemes. We observe that for the marginal offered load, i.e., around 100 for class II-B, and around 200 for class II-A, the delays go to infinity. That is because at these loads, the existing messages are rarely served, so the mobiles can rarely request newly-generated messages. The reason why the marginal load for class II-A is twice that for class II-B is somewhat obvious because the each half of the offered load is from each subclass. The optimal scheme is shown to perform pretty well throughout all ranges of the offered load until reaching the marginal loads. We can see that the delay for class II-A increases until around offered load of 90 is reached, then decreases somewhat abruptly, and starts to increase slowly. The reason why it decreases in the middle is that the role of the piggyback rate-request becomes effective as the number of packet transmissions in a frame increases. However, as the offered load increases further, the number of transmitted packets in a frame also increases, then the total number of mobiles which transmit in a frame will decrease. So, the chance for a mobile to request through a piggyback rate-request mini-slot decreases, so the request access delay increases. The delay eventually goes to infinity at the marginal load.

Figure 8 plots the average delay as the offered load increases for $BW_{dt} = 100$ and 205. $BW_{dt} = 205$ corresponds to no class-I connections, and $BW_{dt} = 100$ corresponds to about a half of the total bandwidth is used by class-I connections, e.g., 190 voice mobiles. Note that the average delay should be at least 2.5 because the average request access delay is at least 0.5, the queueing delay is at least 1, and the transmission delay is also at least 1. Here, we can also observe the effect of the request access delay behavior as the offered load increases. We see that for $BW_{dt} = 100$, the delay of class II-B gets infinite at around 100 of offered load and that of class II-A does at around 200 of offered load similar to the request access delay cases. For $BW_{dt} = 205$, we observe that the offered loads at which the delays get infinite are almost doubled, respectively, because the bandwidth for class-II traffic is doubled.

7 Conclusion

In this paper, we studied a CDMA uplink system to provide diverse QoS guarantees for heterogeneous traffic, where the MC-CDMA is used to support the variable bit rates. To satisfy the pre-defined BER QoS metrics for each traffic class, we used a concatenated RS/convolutional code scheme and developed a power control scheme. Connection-oriented transmission is achieved by assigning transmitter-oriented code channels for class-I traffic where each corresponding mobile client needs to pass the admission test. We calculated an admission region for pairs of voice and video mobiles. For class-II traffic, the best-effort transmission is supported: whenever each class-II mobile has messages to transmit, the mobile requests code channels via a transmission-request code channel, then the transmission is permitted according to the base station scheduling policy. We obtained the target SIR to maximize the aggregate throughput of class-II traffic and evaluated the performance of class-II transmissions using simulations.

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