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Capacity Analysis for Connection Admission Control in Indoor Multimedia CDMA Wireless Communications

JEAN Q.-J. CHAK
Wireless Innovation Lab, Lucent Technologies Canada, 3650 Victoria Park Avenue, Suite 700, Toronto, Ontario, Canada M2H 3P7
E-mail: jchak@canada.lucent.com

WEIHUA ZHUANG
Department of Electrical and Computer Engineering, University of Waterloo, Waterloo, Ontario, Canada N2L 3G1
E-mail: wzhuang@bbcr.uwaterloo.ca

Abstract. In this paper, the capacity analysis for connection admission control is presented for the reverse-link transmission of a packetized indoor multimedia wireless communication system using direct sequence code division multiple access (DS/CDMA). Since CDMA is interference limited, the signal-to-interference-plus-noise ratio criterion is used to check if there is enough system resources (i.e., the CDMA channels and received signal power) for each new connection request. Taking into account the stochastical nature of multimedia traffic, the effective bit rate is used to characterize the resources required by each mobile user and a linear approximation is then used to find the total resources required by all the mobile users already admitted to the system and the new connection request. Transmission errors due to both base station buffer overflow and wireless channel impairments are considered. The capacity of multimedia traffic is determined in such a way that the utilization of the system resources is maximized and, at the same time, the required transmission bit error rate and transmission delay of all users admitted to the system are guaranteed. Computer simulation results are given to demonstrate the performance of the proposed method for capacity analysis.

Keywords: capacity analysis, connection admission control, multimedia wireless communications, code-division multiple access.

1. Introduction

Wireless personal communication networks are becoming more and more important in allowing a tetherless access to communication services in an indoor environment. Although current indoor wireless technologies are mainly focused on voice and low rate data communications, future indoor wireless networks are expected to provide a broad range of multimedia services such as voice, data, graphics, color facsimile, and low-resolution video [1]. To implement a completely wireless network requires a large investment, while partly using co-existing wired networks is more cost efficient. Asynchronous transfer mode (ATM) technology is a promising solution for multimedia communications. It allows various types of traffic to be transmitted over the same network. A network, which uses a wireless network segment as the local user network and an ATM network as the backbone, combines the advantages of wireless and wired networks. The wireless front-end of the network provides flexibility for users who move around buildings, while the ATM backbone provides high transmission quality and capacity. Direct sequence code division multiple access (DS/CDMA) for a wireless environment and

ATM have many common characteristics which offer significant advantages such as zero channel access delay, near-perfect statistical multiplexing of any arbitrary traffic mix, mitigation of channel fading effects and protection of user privacy. As a result, a DS/CDMA indoor wireless network which allows for seamless interconnection to an ATM broadband network is considered in this paper to provide multimedia services to mobile users.

The challenge in internetworking the wireless and wired networks is to guarantee quality of service (QoS) requirements while taking account of (a) the limited radio frequency spectrum, (b) indoor radio propagation impairments resulting in high transmission bit error rates, and (c) user mobility. Resource management for multimedia communications in a wireless mobile environment has been an active area of research in recent years. For a wireless system using frequency-division multiple access (FDMA) and/or time division multiple access (TDMA) protocols, extensive research has been carried out to increase the radio spectrum efficiency. Various channel sharing and assignment schemes have been proposed to allocate the system resources according to traffic loading variations [2–5]. For a wireless system using DS/CDMA protocol, connection admission control (CAC) is one method to adapt the system resources to traffic loading variations and to increase the efficiency of radio spectrum utilization. In particular, the admission control is essential when there are various types of traffic with different QoS requirements and when the system operates in the vicinity of its full capacity. CAC has been attracting a considerable amount of research interests for wired broadband networks [6, 7]. However, the techniques proposed cannot be directly applied to provide consistent QoS guarantees in wireless networks. More recently, the research interests are extended to CAC schemes for wireless networks to deal with the mobility issue and wireless link quality issue respectively. The shadow cluster concept [8] and virtual connection tree [9, 10] have been proposed for call admission control in a wireless network with microcellular architectures. In order to reduce the probability of call dropping due to handoffs, network resources are reserved in all the neighboring base stations. In other words, the mobility problem can be overcome by a proper reservation of system resources. In [11], a CAC algorithm is proposed for a wireless DS/CDMA system with only voice traffic. The algorithm measures the signal-to-interference-plus-noise ratio (SINR) in the reverse link of the DS/CDMA system and compares the ratio to a threshold for voice sources. The SINR threshold is calculated by considering the errors only due to wireless transmission. In [12], the admission-control problem for voice traffic is investigated for FDMA wireless networks. Admission policies for integrated traffic (i.e., voice and data) have been investigated in [13, 14]. To our knowledge, in all the previous work reported in the literature, the characteristics of multimedia traffic and their effects on admission control in a wireless environment have not been studied.

In this paper, we analyze the capacity requirement of multimedia traffic for the CAC decision making process in an indoor DS/CDMA radio system. The traffic types include not only voice, but also data, facsimile and low-rate video. Our focus is on the effects of mixed types of traffic on QoS provisioning, while taking into account the effects of wireless channel impairments and base station buffer overflow on transmission performance. One approach to model the aggregate arrival process of voice, data and video traffic for CAC is the Markov modulated Poisson process (MMPP) model [15, 16]. However, the MMPP model is not suitable for the real-time CAC process due to its high computational cost. The required processing time and memory size by the MMPP model increase significantly as the number of traffic sources increases. As a result, to deal with the stochastical nature of multimedia traffic, we use the effective bit rate to characterize the capacity required by each mobile user and then use the linear approximation to find the total capacity required by all the mobile users already admitted to the system and the new connection request. Since CDMA is interference limited, the SINR criterion is used to check if there is enough system capacity (represented by the CDMA channels and received signal power) for each new connection request. Under the scenario of various user traffic characteristics, the proposed CAC algorithm maximizes the utilization of network resources and, at the same time, guarantees the various QoS requirements of mobile users. The remainder of this paper is organized as follows. Section 2 gives a brief description of the system model and the CAC algorithm. The algorithm checks if there are enough resources for a new connection request by comparing the predicted SINR to a predefined SINR threshold. Section 3 gives the details of how the SINR threshold and the predicted SINR are determined, where bit errors due to wireless transmission and due to buffer overflow at base stations are considered. Section 4 presents numerical analysis results of the proposed method and compares the results with those obtained by computer simulations. Conclusions are given in Section 5.

2. The System Model

The major design objectives of an indoor multimedia wireless network are efficient utilization of radio frequency spectrum, flexible multiservice capability, good quality of service for various service types, a high degree of compatibility with future broadband networks, and low terminal cost, complexity and power consumption. As a result, we consider an indoor wireless network with a base station serving each microcell. Mobile users in each microcell share the radio spectrum through the DS/CDMA protocol. The same total frequency bandwidth is reused in every microcell to increase radio spectrum efficiency and to eliminate the need for frequency coordination. The system operates in a frequency-division duplex mode, i.e., the base-to-mobile forward-link transmission uses a separate frequency band from that of the mobile-to-base reverse-link transmission. The receiving mobile terminals only experience interference from the base stations and the receiving base stations only experience interference from the mobile terminals. Optimizing the system resources and guaranteeing the required QoS for users in the forward-link transmission are relatively simple since the base stations have control over all the system resources and use a broadcast mode to transmit information to the mobile terminals. On the other hand, the reverse link transmission from mobile users to each base station is less cooperative, which makes system resource management and QoS provisioning more challenging as compared to those in the forward link.

In the following, we will focus on the reverse link transmission. Each base station sends a distinct pilot signal to provide initial power control for the mobile terminals. In a multimedia environment with guaranteed QoS, both propagation attenuation (or level of pilot signal power) and resource availability at each base station determine the home base station for each mobile. Details of joint assignment of received power level, transmission rate and home base station in a CDMA network are given in [17, 18]. Multiplexors are used to statistically multiplex the data flows from base stations and to transmit the combined high-rate data traffic onto an ATM network link. Since an ATM network is involved in conveying information between a wireless source terminal and a wireless or wired destination terminal, the multiservice wireless network should have a protocol layer consistent with the ATM protocol structure. Thus, information to be transmitted over the wireless channel is grouped into packets, in order to provide maximum compatibility between the wireless and wired networks and to offer highest flexibility in the information transmission for multimedia services. If the data rate generated by an information source is larger than the peak rate of a DS/CDMA channel, then the information is transmitted in parallel over multiple DS/CDMA channels by using multiple pseudorandom noise (PN) codes. Transmission errors over the wireless link are caused by multipath fading, shadowing, multiple access interference due to other mobile users and additive white Gaussian noise. To reduce the transmission error rate, forward error correction (FEC) such as convolutional coding [19] and BCH (Bose-Chaudhuri-Hocquenghem) coding [20] can be used for real-time traffic. For non-real-time traffic requiring a very low bit error rate (BER), an automatic retransmission request (ARQ) protocol can be used in combination with FEC [19]. Closed loop power control combined with channel estimation can be used for the packetized transmission [21].

A CAC algorithm is needed over the reverse link in order to provide satisfactory QoS to mobile users of different traffic types. The reverse link consists of an admission control channel and a number of data channels. A mobile terminal uses an ALOHA protocol to access the admission control channel for sending the following information to the base station: the service class, traffic type, required transmission BER, maximum transmission delay, peak bit rate, and average bit rate. Upon receiving the information, the base station executes the admission control algorithm to see if the mobile terminal can be admitted to the system. Based on the service class (which relates to service cost) and traffic type, each connection request is assigned a priority to give a user with an emergency situation the opportunity to begin and complete a call. If a connection request is accepted, then the base station will perform resource allocation for the mobile user which includes bandwidth partition and buffer allocation to satisfy the QoS requirements of the user. The bandwidth partitioning algorithm allocates a number of DS/CDMA data channels and the received signal power for the user. The number of DS/CDMA channels (represented by PN code sequences) assigned to the mobile, ξ, is equal to $\lceil R_{b,p}/R_{b,c} \rceil$, where $\lceil \cdot \rceil$ denotes the ceiling function, $R_{b,p}$ and $R_{b,c}$ are the peak transmission bit rate of the user and the transmission bit rate of a DS/CDMA data channel respectively. If the number is larger than one, the mobile terminal will transmit packets in parallel using the assigned PN codes. The allocated received power level depends on the required SINR threshold (to be discussed in Section 3) and interference including multiple access interference and background noise. The buffer allocation algorithm gives each input traffic a fraction of the total buffer space, given the input traffic priority. An existing non real-time connection with a lower priority than the new request may be suspended if not enough resources are available. However, real-time connections with lower priority should not be dropped in order to accommodate the new connection request, as this is not desirable from the users' point of view (forced connection dropping is more disturbing than new connection blocking). A suspended connection means that the base station suspends all transmission from the mobile terminal. The base station keeps a list of suspended connections, and notifies the mobile terminals to resume transmission when enough resources become available.

3. The Decision Making Process for New Connection Request

The key step in the CAC algorithm is to determine whether there are enough resources for each new connection request. This section presents the details of the decision making process. The following assumptions are made:

1. For interactive traffic, the forward link always has enough resources to accommodate a user as long as the reverse link has resources to admit it.

2. The ATM backbone network always has enough resources to accept the call.
3. Transmission errors and delays caused by the ATM portion of the network are negligible as compared to those caused by the wireless network, due to the clean fiber channels and much higher transmission rates in the backbone network.
4. Handoff calls can be treated as new calls. That is, the user mobility issue is not studied here since it has been addressed in detail in [8–10].

The QoS requirements under consideration for admitted connections are BER and transmission delay. Transmission errors are due to radio channel impairments and due to lost packets because of buffer overflow at the receiving base station. Transmission delay occurs at the base station when the total input capacity exceeds the output capacity, and the input packets must be stored in the base station buffers. Since DS/CDMA protocol is interference limited, the SINR is used to check if there are enough system resources for a new connection request. When a new connection request arrives, the base station calculates the SINR threshold for the requesting mobile terminal. The SINR threshold is defined as the minimum SINR required by a traffic type to provide an acceptable BER. The base station then calculates a predicted SINR for each mobile terminal, with the assumption that the requesting mobile terminal is in the system. If the predicted SINR is greater than or equal to the corresponding SINR threshold and the transmission delay requirement is guaranteed for every mobile terminal in the system, then there are enough resources in the system for the requesting mobile terminal and the mobile terminal is admitted into the system.

Let the number of connected mobile terminals in the requesting mobile terminal's microcell be $J - 1$, the requesting mobile terminal be the Jth one, and the requesting mobile terminal's traffic type be type ν. The SINR threshold for the requesting mobile terminal, Γ_ν, can be determined in such a way that the error rate from both wireless transmission and lost packets are below the acceptable BER threshold, and the delay is below a transmission delay threshold for the traffic type. The combined BER must be less than the required BER value P_b. For a given traffic type, the BER threshold P_b can be decomposed into two BER requirements, the wireless BER threshold ($P_{b,w}$) and the buffer BER threshold ($P_{b,b}$), such that $P_b = P_{b,w} + P_{b,b}$. That is, the actual BER from wireless transmission must be less than $P_{b,w}$ and the BER due to lost packets must be less than $P_{b,b}$. In fact, a system with either a higher $P_{b,w}$ and a lower $P_{b,b}$ or with a lower $P_{b,w}$ and a higher $P_{b,b}$ can meet the combined BER threshold P_b. As a result, the thresholds $P_{b,w}$ and $P_{b,b}$ should be chosen in such a way that the total number of mobile users admitted to the system is maximized. The steps to determine the SINR threshold Γ_ν and the predicted SINR are given in the following.

1. *Maximum number, N, of homogeneous sources with acceptable wireless transmission error rate:* This step is to ensure that the error rate due to wireless transmission is less than $P_{b,w}$, if the traffic of all sources is type ν. Let random variable χ_n be 1 if the nth traffic source is transmitting information (in the "on" state), and 0 otherwise (in the "off" state). A traffic source is assumed to have an exponentially distributed time in both "on" and "off" states. Let $R_{b,a}$ be the average bit rate of the traffic source, then

$$\chi_n = \begin{cases} 1, & \text{with probability} \dfrac{R_{b,a}}{\xi R_{b,c}}; \\ 0, & \text{with probability} 1 - \dfrac{R_{b,a}}{\xi R_{b,c}}. \end{cases} \tag{1}$$

Given that there are N homogeneous traffic sources of type ν in the cell and each source has signal power S at the base station receiver, the signal energy per bit to noise density ratio, E_b/N_0, satisfies the following relation

$$\frac{E_b}{N_0} \geq \frac{W/R_{b,p}}{\sum_{n=1}^{N-1} \chi_n + I/S + \eta/S} \tag{2}$$

where W is the total reverse-link bandwidth for the microcell, I is the total other-microcell user interference, and η is the background noise due to spurious interference and thermal noise contained in the total spread bandwidth. The total other-microcell user interference-to-signal ratio, I/S, depends on the number of active mobile terminals in the other microcells, the locations of the active mobile terminals and the types of traffic being transmitted. Based on modulation, coding, diversity schemes used, and channel fading statistics, the BER requirement can be mapped to the required (E_b/N_0) value for real-time traffic without retransmission or to the optimal (E_b/N_0) value for non real-time traffic with retransmission [19]. Taking account of packetized transmission, it can be derived that the required wireless BER threshold is achieved with probability

$$P_T = 1 - \sum_{n=0}^{N-1} \binom{N-1}{n} \left(\frac{R_{b,a}}{\xi R_{b,c}}\right)^n \left(1 - \frac{R_{b,a}}{\xi R_{b,c}}\right)^{N-1-n} Q\left(\frac{\delta - n - E(I/S)}{\sqrt{V(I/S)}}\right) \tag{3}$$

where

$$\delta = \frac{W/R_{b,p}}{(E_b/N_0)_r} - \frac{\eta}{S}, \quad Q(x) = \int_x^{\infty} e^{-y^2/2} dy \ / \ \sqrt{2\pi}$$

$(E_b/N_0)_r$ is the threshold of the received signal energy per bit to noise density ratio to achieve the required wireless transmission BER threshold for traffic type ν, $E(I/S)$ and $V(I/S)$ are the mean and variance of I/S respectively. The upper bounds for $E(I/S)$ and $V(I/S)$ are independent of S and can be calculated as described in [22]. Given the upper bound for η/S and the required value for P_T, N can be computed from (3).

2. *Maximum number, M, of homogeneous sources with acceptable packet loss probability:* This step is to ensure that the BER due to lost packets is less than the buffer BER threshold, $P_{b,b}$, if the traffic of all sources is type ν. Let the output buffer size be K at the base station and the maximum admissible delay for the traffic type be T seconds. If the value of K is chosen such that the maximum delay in the buffer is less than T under the base station service discipline, then the packet delay requirement is always satisfied. Under a FIFO discipline, the buffer size (in units of packets) is

$$K = \frac{TC_o}{L_p} \tag{4}$$

where C_o is the capacity of the base station wired output link to the backbone and L_p is the packet length in bits. The probability of packet loss is the fraction of all transmitted packets that are lost due to buffer overflow, with the length of all packets being the same. Since $P_{b,b}$ represents the tolerable probability of bit errors due to buffer overflow and the bit errors due to wireless transmission should not be included again as errors if a packet is dropped, the packet loss probability threshold, $P_{p,l}$, can be found by

$$P_{p,l} = \frac{P_{b,b}}{1 - P_{b,w}}. \tag{5}$$

From the equivalent capacity and the Gaussian approximation methods [23], the bandwidth required by a single traffic source to satisfy $P_{p,l}$ can be calculated. Using the equivalent capacity method, the required bandwidth c_1 for a traffic source can be computed as

$$c_1 = \frac{R_{b,p}[f - K + \sqrt{(K-f)^2 + 4K\rho f}]}{2f} \tag{6}$$

where

$$\rho = \frac{\alpha^{-1}}{\alpha^{-1} + \beta^{-1}}, \quad g = \ln(1/P_{p,l}), \quad f = g\alpha^{-1}(1-\rho)R_{b,p}$$

α^{-1} and β^{-1} are the mean "on" and "off" periods of the type ν traffic respectively. The maximum number of homogeneous traffic sources allowed in the system with acceptable packet loss probability based on the equivalent capacity method is then

$$M_t = \frac{C_o}{c_1}. \tag{7}$$

Because the equivalent capacity method does not consider the interaction between the traffic sources, c_1 may overestimate the required bandwidth for one traffic source when many traffic sources are multiplexed together. To capture the effects of multiplexing, the Gaussian approximation method is incorporated with the equivalent capacity method. The Gaussian approximation method assumes that the total bit rate for all traffic sources in the system has a Gaussian distribution, which is valid when there are many i.i.d. traffic sources. Let m_t be the mean and σ_t be the standard deviation of the total traffic bit rate in the system, and let α_t be the inverse of the Gaussian distribution representing the total traffic bit rate, then

$$m_t = M_t R_{b,a}, \quad \sigma_t = \sqrt{M_t R_{b,a}(R_{b,p} - R_{b,a})}, \quad \alpha_t = \sqrt{2\ln(1/P_{p,l}) - \ln(2\pi)}. \tag{8}$$

The total bandwidth, C_t, required for M_t traffic sources in the system is given by the minimum bandwidth calculated by using the Gaussian approximation method and by using the equivalent capacity method, respectively,

$$C_t = \min(m_t + \alpha_t\sigma_t, C_o). \tag{9}$$

The equivalent capacity required by a single traffic source, c_e, to satisfy $P_{p,l}$ is then

$$c_e = C_t/M_t. \tag{10}$$

Finally, the maximum number of homogeneous traffic sources allowed in the system is

$$M = C_o/c_e. \tag{11}$$

3. *Maximum number, L, of homogeneous sources:* The maximum acceptable number of multiplexed homogeneous traffic sources of type ν, taking account of the errors due to both wireless transmission error and lost packets, is

$$L = \min(N, M). \tag{12}$$

4. *The effective bit rate $R_{b,e}$:* The effective bit rate is defined as the minimum channel capacity required by a traffic source to satisfy the BER threshold. One method to calculate the effective bit rate is to use linear approximation [24]. Suppose that there is only one type

of traffic source, type ν, sharing the output link capacity, C_o. The effective bit rate of a type ν source is upper bounded by

$$\tilde{R}_{b,e} = \frac{C_o}{L}. \tag{13}$$

The effective bit rate of one type of traffic sources is obtained independently from any other traffic types. If there are different types of traffic in the system and the sum of the effective bit rates of each connected mobile terminal is less than C_o, then the linear approximation method assumes that the combination of different traffic types still satisfies the individual mobile terminal's performance criteria. To take account of the possibility of packet retransmission for non-real-time traffic, the effective bit rate is modified by

$$R_{b,e} = \begin{cases} \left(1 + \dfrac{P_{p,e}}{1 - P_{p,e}}\right) \tilde{R}_{b,e}, & \text{if retransmission allowed} \\ \tilde{R}_{b,e}, & \text{if retransmission not allowed} \end{cases} \tag{14}$$

where $P_{p,e}$ is the probability of received packet error due to wireless transmission and can be calculated depending on the modulation and encoding/decoding schemes used [19]. For combinations of different traffic types in the system, $R_{b,e}$ can be used as the portion of C_o which is used by a single traffic source of type ν.

5. *SINR threshold* Γ_ν*:* The SINR threshold, Γ_ν, is the SINR required for traffic type ν to obtain $(E_b/N_0)_r$ for guaranteeing the BER requirement. Γ_ν can be calculated based on the effective bit rate as

$$\Gamma_\nu = (E_b/N_0)_r (R_{b,e}/W). \tag{15}$$

To find the effect of admitting the requesting mobile terminal on all other admitted mobile users, the base station measures the total interference $(\Psi + I + \eta)$ seen by the requesting mobile terminal if admitted, where Ψ is the total average power received from all existing connections in the microcell. Due to the fact that the existing connections may have "on" and "off" periods, it takes time for the base station to obtain the measured total interference with a reasonable accuracy. As a result, it requires that the base station periodically updates the total interference measurement all the time. Based on the measurement, the initial estimate of the required received signal power $\tilde{S}_J$ from the requesting mobile terminal can be obtained from the threshold Γ_ν

$$\tilde{S}_J = \Gamma_\nu(\Psi + I + \eta). \tag{16}$$

Let $\tilde{S}_j$ be the minimum new received power predicted for the jth connected mobile terminal $(1 \le j \le J - 1)$ which is required to meet the SINR threshold Γ_j, with the effect of the requesting mobile terminal taken into consideration. Under the assumption that the new requesting mobile terminal is in the system, we have $\tilde{S}_j > S_j$, where S_j is the current received signal power from the jth mobile terminal. Indeed, as the number of newly admitted mobile terminals increases, the required minimum received power of the mobile terminals previously admitted into the system should increase correspondingly so that its required SINR threshold can be guaranteed. On the other hand, as $\tilde{S}_j$ $(j = 1, 2, \ldots, J - 1)$ increases, the total interference which the requesting mobile terminal will experience also increases; hence, the required signal power $\tilde{S}_J$ in (16) needs to be increased proportionally. The increase of $\tilde{S}_J$ will further increase the required $\tilde{S}_j$ value. As a result, the predicted value for $\tilde{S}_J$ and $\tilde{S}_j$ needs to be updated recursively based on the thresholds Γ_v and Γ_j and on the interference I and η. Fortunately, the number of different traffic types is usually limited to a few in a practical system even though the total number of mobile terminals in the cell can be large. Consequently, a few iterations are required for $\tilde{S}_J$ and $\tilde{S}_j$ to converge with a reasonable accuracy if the solutions indeed exist. In the case that the converged $\tilde{S}_J$ and $\tilde{S}_j$ values can be found, the requesting mobile user is admitted to the system after all the previously admitted users adjust their transmitted power so that the received power level from the jth mobile is the converged value $\tilde{S}_j$. If the solutions diverge after a few iterations, it means that the system does not have enough remaining capacity to accommodate the requesting mobile terminal. The base station will then check if there are enough resources for the new connection request after suspending some existing lower priority non-real-time connections. The requesting connection will be rejected if not enough system resource is available.

In summary, the CAC algorithm calculates the SINR threshold by first finding the maximum number of homogeneous traffic sources allowed in the microcell given the acceptable BER and then determining the effective bit rate by the linear approximation. Based on the measured interference $(I + \eta)$ and the SINR thresholds, the required minimum received signal powers from all the mobile users are computed recursively. Only if there exist converged solutions to the required receiver powers, should the requesting mobile user be admitted. It should be mentioned that the above capacity analysis is based on the assumption of accurate power control. Any inaccurate power control implementation will lead to a reduction of the system capacity. Details of power control implementation and effect of power control error can be found in [21].

4. Numerical Results and Discussion

This section presents numerical results of the proposed capacity analysis algorithm. Computer simulation results are also given to demonstrate the performance of the proposed algorithm. The parameters used in the numerical calculations and in computer simulations are: $W = 5$ MHz, $C_o = 1.2$ Mbps, $K = 2$ packets, $P_T = 0.95$. Due to wireless propagation impairments and non perfect statistical multiplexing, the spectral efficiency of CDMA systems has been found to be in the neighborhood of 0.12 bps/Hz [19, 20]. As a result, with the C_o value chosen, the output link is not a bottleneck for the input traffic at the base station. This is consistent with the assumptions (b) and (c) made in Section 3, and is further verified by the numerical results to be discussed in this section. Voice, facsimile, low-rate video, and data sources are considered to include a variety of traffic sources. Table 1 lists the parameters for the traffic types. The video sources are assumed to be low-rate video for teleconferencing with motion compensation [15, 25]. Each video source is modeled by 4 i.i.d. subsources. The video parameters in Table 1 are for each such video subsource. The voice, facsimile and video sources have a high probability of being in the “on” state. The mean number of packets transmitted during the “on” state is low for the voice sources, but high for the facsimile and video sources. The data sources are non-real-time traffic, so that retransmission of packets with detected errors is allowed, up to a maximum packet delay of 5 ms, or at most one retransmission per packet. The base station can guarantee a BER of 10^{-6} for the data user without ARQ. The BER can be further reduced to a smaller value by an ARQ protocol. Retransmission increases the number of packets generated during an “on” state, but the effect of retransmission on the

Table 1. Parameters for different types of traffic.

Traffic type	P_b	$(E_b/N_0)_r$ (dB)	α^{-1} (s)	β^{-1} (s)	$R_{b,p}$ (bps)	$R_{b,a}$ (bps)
Voice	10^{-3}	3.6	0.4	0.6	24000	9600
Facsimile	10^{-4}	4.7	1	2	96000	32000
Video	10^{-5}	5.7	0.8	1.765	38400	11977
Data	10^{-6}	7.5	1	5	96000	16000

peak and average bit rates is assumed to be negligible, since retransmission is required only when the FEC used cannot correct the packet errors.

Tables 2 to 4 give the values of N, M and L respectively for the homogeneous traffic sources. The BER threshold P_b is divided into $P_{b,w}$ and $P_{b,b}$ in such a way that the values of N and M are close to each other, resulting in a maximum value for L. The results obtained for N based on the proposed algorithm are equal to (for facsimile) or less than (for the other traffic types) the corresponding simulation results, because the proposed algorithm uses the upper bound (not the exact value) of the mean and variance of noise and interference, $E(I/S)$ and $V(I/S)$. For the value of M, the result of the proposed algorithm is less than the corresponding simulation result for each traffic type considered. This is because the proposed algorithm uses the Gaussian approximation method in finding the M value. However, the method does not fully explore the amount of statistical multiplexing gain, as indicated in [23]. As a result, the L values based on the proposed algorithm are smaller than the corresponding L values from simulation. For the case of heterogeneous traffic sources, we consider (a) one video user, (b) two video users, and (c) one data user, respectively, already in the microcell and determine how many traffic sources of another type can be admitted. The results are shown in Tables 5–7. It is observed that the values obtained using the proposed algorithm are lower than the corresponding simulation results for all traffic types. In general, the simulation results verify the results obtained from the proposed algorithm and, at the same time, indicate that the proposed algorithm is somewhat conservative. On the other hand, it has been shown that capacity analysis based on the MMPP model is even more conservative than the proposed algorithm [26]. In fact, due to the random nature of multimedia traffic flows, capacity analysis for CAC is a complex and challenging issue. So far no better (i.e., more accurate) capacity analysis algorithm has been developed for CAC in wireless multimedia communications, especially if the simplicity of the algorithm is desired. Compared with the MMPP model, the algorithm proposed here has advantages both in simplicity and in accuracy.

5. Conclusions

A capacity analysis algorithm for connection admission control has been developed using the SINR criterion for the reverse-link transmission of the DS/CDMA indoor multimedia wireless network. The SINR threshold is calculated for each traffic type based on the effective bit rate of the traffic type, taking into account the required transmission bit error rate and transmission delay. The linear approximation is used to determine the total capacity needed by a number of traffic sources. The user QoS requirements are guaranteed by ensuring that

Table 2. Maximum number of homogeneous traffic sources with acceptable wireless transmission error rate.

Traffic type	$P_{b,w}$	N (proposed algorithm)	N (simulation)
Voice	5×10^{-4}	73	91
Facsimile	5×10^{-5}	12	12
Video	9×10^{-6}	9	11
Data	9.9×10^{-7}	11	17

Table 3. Maximum number of homogeneous traffic sources with acceptable packet loss probability.

Traffic type	$P_{b,b}$	M (proposed algorithm)	M (simulation)
Voice	5×10^{-4}	76	97
Facsimile	5×10^{-5}	13	17
Video	1×10^{-6}	10	14
Data	1×10^{-8}	15	22

Table 4. Maximum number of homogeneous traffic sources.

Traffic type	L (proposed algorithm)	L (simulation)
Voice	73	91
Facsimile	12	12
Video	9	11
Data	11	17

Table 5. Maximum number of sources allowed with one video source already in system.

Traffic type	Number of sources (proposed algorithm)	Number of sources (simulations)
Voice	64	84
Facsimile	10	12
Data	9	16

Table 6. Maximum number of sources allowed with two video sources already in system.

Traffic type	Number of sources (proposed algorithm)	Number of sources (simulations)
Voice	56	76
Facsimile	9	10
Data	8	14

Table 7. Maximum number of sources allowed with one data source already in system.

Traffic type	Number of sources (proposed algorithm)	Number of sources (simulation)
Voice	66	87
Facsimile	10	11
Video	8	11

the measured SINR is larger than or equal to the SINR threshold. The performance of the proposed algorithm has been verified by computer simulations and, at the same time, it has been shown that the proposed algorithm is somewhat conservative. Compared with the MMPP model, the proposed algorithm has advantages both in accuracy and in computational cost. The required processing time and memory size of the proposed algorithm grow linearly (instead of exponentially as in the MMPP model) with the number of mobile users in the system, which is critical for real-time implementation.

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Jean Q.-J. Chak received the B.A.Sc. degree (1995) in computer engineering and the M.A.Sc. degree (1997) in electrical engineering, both from the University of Waterloo in Canada. She is currently working at Lucent Technologies Canada, in the areas of billing and enhanced services for wireless communications systems.

Weihua Zhuang received the B.Sc. (1982) and M.Sc. (1985) degrees from Dalian Marine University (China) and the Ph.D. degree (1993) from the University of New Brunswick (Canada), all in electrical engineering. Since 1993 she has been a faculty member at the University of Waterloo where she is currently an Associate Professor in the Department of Electrical and Computer Engineering. Her research interests include digital transmission over fading dispersive channels and wireless networking for multimedia personal communications.

Dr. Zhuang is a licensed Professional Engineer in the Province of Ontario, Canada.