A Medium Access Control Protocol for Voice/Data Integrated Wireless CDMA Systems

In-Taek Lim

In this paper, a medium access control protocol is proposed for integrated voice and data services in wireless local networks. Uplink channels for the proposed protocol are composed of time slots with multiple spreading codes per slot based on slotted code division multiple access (CDMA) systems. The proposed protocol uses spreading code sensing and reservation schemes. This protocol gives higher access priority to delay-sensitive voice traffic than to data traffic. The voice terminal reserves an available spreading code to transmit multiple voice packets during a talkspurt. On the other hand, the data terminal transmits a packet without making a reservation over one of the available spreading codes that are not used by voice terminals. In this protocol, voice packets do not come into collision with data packets. The numerical results show that this protocol can increase the system capacity for voice service by applying the reservation scheme. The performance for data traffic will decrease in the case of high voice traffic load because of its low access priority. But it shows that the data traffic performance can be increased in proportion to the number of spreading codes.

Manuscript received March 6, 2000; revised May 2, 2001.
In-Taek Lim is with the Pusan University of Foreign Studies, Pusan, Korea.
(phone: +82 051 640 3423, e-mail: itlim@taejo.pufs.ac.kr)

I. INTRODUCTION

With the rapid development in wireless access technologies, wireless communication networks and mobile terminals have become popular. In particular, a wireless communication network provides a flexible and convenient subscriber interface to the fixed network. An example of a wireless network is a cellular system that is designed to provide voice-oriented communication services [1], [2]. Another example is a wireless LAN, where data communication is the main service [3], [4]. With the general tendency to integrate voice and data traffic into a single wired network, wireless communication networks also need to support integrated voice and data services.

In the wireless communication network, research was carried out on both time division multiple access (TDMA) and code division multiple access (CDMA). Spread-spectrum CDMA systems offer the potential of high spectrum efficiency, soft capacity, multipath-resistance, and inherent frequency diversity [1], [5]. Among many issues related to the wireless communication network, the medium access control (MAC) protocol that coordinates multiple access of a shared medium is one of the most important issues. In an integrated service wireless communication network, the MAC protocol might be efficient and robust enough to support integrated voice/data service and to guarantee the quality of service requirements for both services. Therefore, the purpose of this paper is to present the MAC protocol for voice/data integrated local wireless CDMA systems, called packet-reservation and status-sensing in CDMA systems (PRS^2 - CDMA).

For integrated voice and data services, packet-switched network architecture seems to be the general choice. In conventional packet radio networks based on TDMA, there are several types of MAC protocol for integrated voice/data services.

Packet-reservation multiple access (PRMA) is closely related to the reservation ALOHA, but it is distinguished from the reservation ALOHA by its response time to network congestion and by its short round trip transmission time [6]. In [7], a status sense multiple access (S^2MA) is presented to accommodate heterogeneous classes of customers on a common broadcast channel. Jangi and Merakos [8] consider some other reservation random-access protocol for integrated voice/data services. Mitrou *et al*. propose the implementation of mini-slotted reservation packets within the PRMA protocol, yielding 10-15% improvements over the basic PRMA in terms of the number of users that can be supported [14]. In [15], an improved PRMA (IPRMA) is presented to increase the throughput of data terminals by allowing contending data terminals to reserve idle slots throughout the frame. Clearly, when combined with TDMA, PRMA protocol and other versions of PRMA protocol provide a solution to integrated voice/data communications. However, in all of these protocols, voice packets are transmitted periodically by reserving a slot in a frame, whereas data packets are transmitted with the slotted ALOHA scheme. Those protocols have the potential problem that performance decreases sharply in the case of high offered load because of the limited number of available channels.

In [10], Tan *et al*. proposed a reservation-code multiple access (RCMA) protocol, which applies a reservation random-access (RRA) protocol [8] to slotted spread-spectrum multiple-access systems. The RCMA protocol can simplify the code tracking hardware at the base station by allowing all terminals to share a group of spreading codes on a contention basis. However, because voice terminals have to contend with data terminals in reserving a spreading code, the voice dropping probability increases, especially when voice traffic is dominant. Soroushnejad *et al*. proposed a multiple access strategy for an integrated voice/data CDMA packet radio network [9]. In this protocol, all the mobile terminals have to be assigned a unique spreading code for transmitting packets at the system deployment stage. This transmitter-oriented code assignment has the advantage of no packet collision. However, the required number of spreading codes increases with the growing population of users, so code tracking and synchronization systems at the base station become complex.

The PRS^2-CDMA protocol results from applying PRMA and S^2MA to slotted CDMA systems with a small coverage area. The proposed protocol assumes that the uplink channels are composed of time slots and spreading codes. The time axis of uplink channel is divided into slots that are grouped into frames as in PRMA; a time slot has a group of spreading codes shared by all users.

The rest of this paper is organized as follows. This paper gives a detailed description of the proposed protocol in Section II, followed by the analysis of its voice packet dropping probability and data packet average delay in Section III. The numerical results are investigated in Section IV with a computer simulation. Section V finishes the paper with concluding remarks.

II. DESCRIPTION OF PRS^2-CDMA

In wireless communication networks with a centralized topology, base station and terminals communicate with each other using two logically separated links. The base station uses the downlink to broadcast control traffic and information traffic to terminals, while terminals use the uplink to transmit related traffic to the base station. The base station schedules the downlink traffic without contention. This paper, therefore, focuses attention on the uplink multiple access. The separation scheme of uplink and downlink is categorized into the following two modes: time-division duplex (TDD) mode and frequency-division duplex (FDD) mode. In TDD mode, the same frequency band is used for both uplink and downlink. In FDD mode, two separate frequency bands are assigned to each link. The PRS^2-CDMA is designed as a protocol to control the uplink communication for a centralized local wireless CDMA system with small coverage.

The PRS^2-CDMA protocol is designed with the following three operations. First, uplink and downlink traffic is transmitted simultaneously by FDD mode. Second, the base station synchronizes all the terminals by broadcasting the system timing by which terminals can adjust their timing clock. Since the timing error among terminals in a short-range area is small, it is feasible to use slotted spread-spectrum signaling in the system. Third, the base station broadcasts feedback information including the status of transmitted packets and the spreading codes that are available in the next slot. Because of a short time delay in a small coverage, it is possible for terminals with a transmitted packet to receive feedback from the base station in the same slot in which the packet was sent.

1. Slot and Frame Structure of Uplink

Under PRS^2-CDMA protocol, the uplink stream is divided into fixed-length frames. Frame duration is chosen so that it is identical to the voice packet generation period in voice terminals. Figure 1 shows an example of uplink frame format. A frame consists of k_0 time slots. For each slot, there are m shared spreading codes $\{C_i, i=1, ..., m\}$ that are orthogonal to each other. Therefore, the total number of available channels in an uplink frame is $m*k_0$.

Voice and data terminals share a group of spreading codes to transmit a packet to the base station. Each code can be identified as "reserved" or "available" based on the feedback information from the base station at the end of each slot. A reserved code is exclusively assigned to a voice terminal until a talkspurt ends. An available code is used by either the voice terminal that wants to reserve a code when it begins a talkspurt or the data terminal that has data packets to transmit.

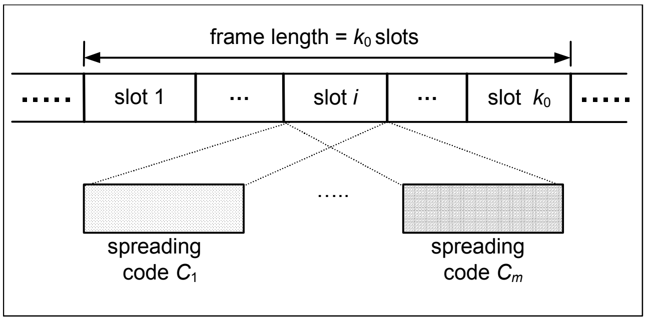

Fig. 1. Uplink frame structure.

A packet generated by voice or data terminal can be transmitted during one slot with one spreading code. The uplink packet structure is illustrated in Fig. 2. The user information is packetized with additional overheads. The preamble field is used for a spreading code acquisition by the base station receiver. The data field is composed of a packet type (i.e., voice, data, or reservation-request packet), a packet header, and an information subfield. The header subfield contains the terminal identifier, the packet sequence number, etc. The unused field in the reservation-request packet is not transmitted.

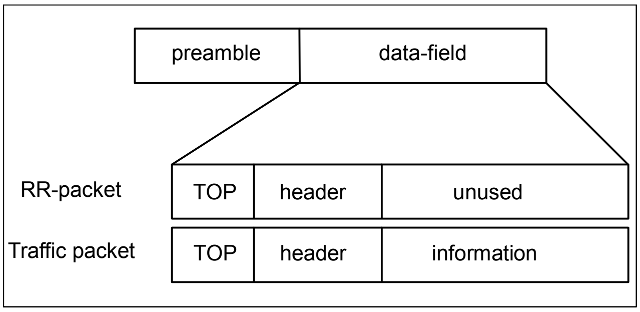

Fig. 2. Packet format; TOP : Type of Packet; RR-packet : reservation-request packet; Traffic-packet : voice or data packet.

2. Operation of Voice Subsystem

Before describing the operation of the voice subsystem, it is necessary to describe the voice traffic model. For the voice terminal with a slow speech activity detector (SAD), its voice source can be characterized by a two-state model, as shown in Fig. 3.

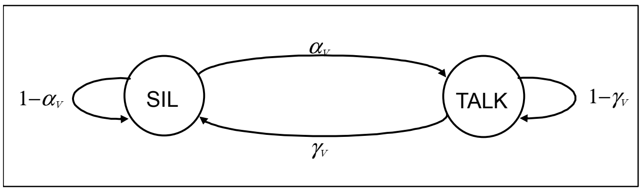

Fig. 3. Voice traffic model.

To describe the voice traffic model, the following four parameters are required. These are the mean silence duration t_1, the mean talkspurt duration t_2, the transition probability from the silence state to the talking state α_v, and the inverse transition probability γ_v. The talkspurt and silence are assumed to have exponentially distributed durations, which are statistically independent of one another. Thus, the parameter α_v and γ_v are represented as [6]

$$\begin{aligned} \alpha_v &= 1 - e^{-\tau/t_1} \\ \gamma_v &= 1 - e^{-\tau/t_2} \end{aligned} \quad (1)$$

where τ is the duration of a slot.

During a talkspurt, the voice terminal generates voice packets with the period identical to the frame duration. To transmit voice packet, therefore, voice terminal needs to be guaranteed one channel (in this system, a spreading code in a slot) per frame.

When beginning a new talkspurt, the voice terminal enters into a reservation-contending state and tries to reserve a spreading code. The terminal in a reservation-contending state randomly selects an available spreading code in the next slot and transmits a reservation-request packet with the transmission permission probability β. If the base station captures the preamble field of a reservation-request packet, it immediately broadcasts an acknowledgement. The acknowledgement contains the terminal identifier and the spreading code information of the current slot. If two or more voice terminals transmit the reservation-request packet with the same spreading code, a packet collision occurs. In this case, the base station cannot capture the preamble field and thus does not send an acknowledgement.

If the voice terminal receives an acknowledgement, it enters into a reservation state. The terminal in a reservation state transmits voice packets over the reserved code in the same slot of subsequent frames until the talkspurt ends. If the terminal decides not to transmit the reservation-request packet (with probability 1-β), or if it fails to reserve even in the case of transmission permission (for instance, by a collision or by a wireless channel error), it tries to reserve a spreading code in the next slot.

After a talkspurt completion, the voice terminal does not access the reserved code dedicated to it. If the base station does not detect a packet from a reserved code (i.e., it fails the code acquisition of the reserved code), it regards the code as an available code, which can be used by any terminal in subsequent frames.

In voice traffic, the transmission delay of voice packets causes a serious problem in the quality of service. If the terminal cannot transmit voice packets until any given speech delay constraint (W_{max} slots), these packets must be dropped. In this paper, the packet dropping probability (P_{drop}) is considered as the performance measure of voice traffic.

3. Operation of Data Subsystem

In general, the length of a data message is variable. To transmit a data message, a terminal first segments the message into several fixed-size packets.

In the PRS^2-CDMA protocol, the operation of the data subsystem is similar to the slotted CDMA_ALOHA scheme [11]. The data subsystem does not use the reservation scheme but the spreading code status-sensing scheme. The data terminal that does not have a message to transmit is said to be in an idle state. When a data terminal in an idle state generates a message, it then enters into a contending state. The data terminal in a contending state selects a spreading code in the current slot which has not been reserved yet by a voice terminal in a reservation state and is not currently reserved by a voice terminal in reservation-contending state. After receiving the code status information of the current slot from the base station, the data terminal transmits a packet with the transmission permission probability β. The transmission of a data packet, therefore, is delayed until the header subfield of reservation-request packet appears. If base station successfully captures a data packet, it also broadcasts an acknowledgement.

If the data terminal receives an acknowledgement and has packets in its buffer, it remains in contending state and tries to transmit another packet in the next slot. The terminal that receives an acknowledgement and does not have any packet to transmit goes into idle state. If the data terminal fails to transmit a packet, it remains in the same state and tries to send the same packet in the next slot.

In data service, to guarantee the quality of service, packets must be delivered although they are delayed. Therefore, the average packet delay (D_d) is considered as the performance measure of data traffic.

III. PERFORMANCE ANALYSIS

1. Voice Subsystem

The voice terminal is independent of the data terminal because the former has a higher access priority than the latter. Let N_v the number of voice terminals.

At a given slot t, a voice terminal is in silence state, reservation-contending state, or reservation state. Therefore, we can define the voice subsystem state $\{X_v^t=(R_v^t, C_v^t)\}$ as the number of voice terminals in reservation state (R_v^t) and in reservation-contending state (C_v^t). Let $Q^v_{jk,rl}$ be the state transition probability from a state (j,k) in a slot to the state (r,l) in the following slot. This probability, which can be derived from the conditional probabilities that v_i terminals in silence state begin talkspurt, v_r terminals in reservation-contending state transmit reservation-request packet with one of x available codes, and s_v packets succeed in reserving a spreading code, is given by

$$Q^v_{jk,rl} = \sum_{x=0}^{m} \sum_{v_i=0}^{N_v-j-k} \sum_{v_r=0}^{k} \begin{Bmatrix} \Phi(x,j,k_0,m) b(N_v-j-k, v_i, \alpha_v) b(k, v_r, \beta) \\ b(j+s_v, r, 1-\gamma_v) S(s_v \mid v_r, x) \end{Bmatrix} \tag{2}$$

where

$$\begin{aligned} & s_v = k + v_i - l, \;\; 0 \le s_v \le \min(v_r, x) \\ & 0 \le \{j, r\} \le \min(N_v, mk_0), \\ & 0 \le k \le N_v - j, \;\; 0 \le l \le N_v - r \\ & b(n,i,p) = \binom{n}{i} p^i (1-p)^{n-i}. \end{aligned}$$

In (2), the term $\Phi(x,j,k_0,m)$ is the probability that x spreading codes are available in a slot under the condition of j reserved voice terminals. The term $b(N_v\text{-}j\text{-}k,v_i,\alpha_v)$ is the probability that v_i terminals in a silence state begin a new talkspurt, and the term $b(k,v_r,\beta)$ is the probability that v_r terminals in a reservation-contending state transmit the reservation request packet. The term $b(j+s_v,r,1\text{-}\gamma_v)$ is the probability that r terminals still remain in a reservation state. And the term $S(s_v|v_r,x)$ is the conditional probability that s_v packets out of v_r are successfully transmitted over one of x available codes.

The probability $S(s|n,x)$ can be derived by conditioning on the number of packets transmitted simultaneously over the first channel (arbitrarily chosen) and using the total probability. This probability can be recursively defined as follow [11]

$$\begin{aligned} S(s|n,x) = \sum_{j=0}^{n} b\left(n, j, \frac{1}{x}\right) \{ & \zeta_j S(s-1 \mid n-j, x-1) \\ & + (1-\zeta_j) S(s \mid n-j, x-1) \} \end{aligned} \tag{3a}$$

where $\zeta_1=1$, and $\zeta_j=0$ ($j\neq 1$). In (3a), j is the number of packets transmitted over the first channel, and $b(n,j,1/x)$ is the probability that j packets out of n are transmitted over the first channel.

ζ_j is the probability that one packet is transmitted successfully if j packets are transmitted simultaneously over a given channel. The initial conditions for (3a) are as follows:

$$\begin{aligned}
&S(0\,|\,0,x)=1,\ S(1\,|\,0,x)=0, && \text{for } x\ge 0\\
&S(0\,|\,n,0)=1,\ S(1\,|\,n,0)=0, && \text{for } n\ge 0\\
&S(0\,|\,1,x)=0,\ S(1\,|\,1,x)=1, && \text{for } x\ge 1\\
&S(1\,|\,n,1)=0,\ S(0\,|\,n,1)=1, && \text{for } n\ge 2\\
&S(s\,|\,n,x)=0, && \text{for } s>\min(n,x).
\end{aligned} \tag{3b}$$

If j is the number of voice terminals in a reservation state at the beginning of a given slot, then the probability $\Phi(x,j,k_0,m)$ is determined by the following. Let $D(j,k_0,m)$ be the number of ways of distributing j indistinguishable voice terminals into k_0 distinguishable slots under the restriction of m-code occupancy per slot (i.e., at any given time, at most m terminals can transmit packet per slot). According to [12], $D(j,k_0,m)$ and $\Phi(x,j,k_0,m)$ are given by

$$D(j,k_0,m)=\sum_{i=0}^{k_0}(-1)^i\binom{k_0}{i}\binom{j+k_0-i(m+1)-1}{k_0-1}, \tag{4}$$

$$\Phi(x,j,k_0,m)=\frac{D(j-m+x,k_0-1,m)}{D(j,k_0,m)} \tag{5}$$

where $j\le mk_0$, $0\le x\le m$, $k_0\ge 1$.

Once the transition probability $Q^v_{jk,rl}$ is obtained, the steady state probability Π^v_{rl} that r voice terminals are in a reservation state and l voice terminals are in a reservation-contending state is derived as [13]

$$\begin{aligned}
\Pi^v_{rl} &= \sum_{j=0}^{\min(N_v,mk_0)}\sum_{k=0}^{N_v-j} Q^v_{jk,rl}\times\Pi^v_{jk}\\
&\sum_{r=0}^{\min(N_v,mk_0)}\sum_{l=0}^{N_v-r}\Pi^v_{rl}=1.
\end{aligned} \tag{6}$$

The performance measure of the voice subsystem in PRS^2-CDMA protocol is the voice packet dropping probability. The descriptive mechanisms for packet dropping are as follows. Let L be the number of packets generated during a talkspurt. In a talkspurt, no packet is dropped if the voice terminal reserves before W_{max} slots have occurred. After waiting W_{max} slots, the terminal drops the initial packet of the talkspurt. The second packet is generated exactly after k_0 slots from the first packet. Therefore, the terminal drops the second packet if it is still in a reservation-contending state after $W_{max}+k_0$ slots. If the terminal can not reserve after $W_{max}+(L-1)k_0$ slots, it drops the all packets of a talkspurt.

The probability that k packets are dropped in a talkspurt of L packets is

$$\Pr\{n_{drop}=k\,|\,L\} = \begin{cases} 1-f^{W_{\max}}, & \text{for } k=0\\ f^{W_{\max}+(k-1)k_0}-f^{W_{\max}+kk_0}, & \text{for } 1\le k\le L-1\\ f^{W_{\max}+(L-1)k_0}, & \text{for } k=L \end{cases} \tag{7}$$

where f is the probability that a reservation-contending terminal fails to reserve a code and defined as

$$f=1-\sum_{r=0}^{\min(N_v,mk_0)}\sum_{l=0}^{N_v-r}\sum_{v_r=0}^{l}\sum_{x=1}^{m}\begin{Bmatrix}\Phi(x,r,k_0,m)b(l,v_r,\beta)\\ S(v_r\,|\,v_r,x)\Pi^v_{rl}\end{Bmatrix} \tag{8}$$

By averaging (7) over L, the mean number of dropped packets in a talkspurt is given by

$$\begin{aligned}
E\{n_{drop}\} &= \sum_{L=1}^{\infty}\sum_{k=0}^{L} k\cdot\Pr\{n_{drop}\,|\,L\}\cdot\Pr\{L\}\\
&=\frac{f^{W_{\max}}}{1-(1-\gamma_f)f^{k_0}}
\end{aligned} \tag{9}$$

where $\Pr\{L\}$ is the probability that there are L packets in a talkspurt. The probability $\Pr\{L\}$ is defined as

$$\Pr\{L\}=\gamma_f(1-\gamma_f)^{L-1} \tag{10}$$

where $L\ge 1$. In (10), γ_f is the probability that a talkspurt ends in a frame and is represented as

$$\gamma_f=1-(1-\gamma_v)^{k_0} \tag{11}$$

where γ_v is given in (1).

The packet dropping probability is defined as the ratio of dropped packets to the total number of generated voice packets. Then, the packet dropping probability is

$$P_{drop}=\gamma_f\cdot\frac{f^{W_{\max}}}{1-(1-\gamma_f)f^{k_0}}. \tag{12}$$

2. Data Subsystem

The data terminal attempts to transmit a packet with an available code that is not used by a voice terminal in a reservation-contending state or reservation state. The state of the data terminal, therefore, is dependent on the state of the voice terminal. In this paper, the data buffer size is assumed to be sufficiently large enough to ignore the data packet blocking caused by the lack of buffer.

Let N_d be the number of data terminals in the system. And let the data subsystem state be $\{X_d^t = (C_d^t, R_v^t, C_v^t)\}$ where C_d^t is the number of the contending data terminals. The state transition probability $Q^d_{ijk,brl}$ that the system state changes from (i,j,k) in a slot to (b,r,l) in the following slot is given by

$$Q^d_{ijk,brl} = \sum_{x=0}^{m} \sum_{v_i=0}^{N_v-j-k} \sum_{v_r=0}^{k} \sum_{d_i=0}^{N_d-i} \sum_{d_b=0}^{i} \left\{ \begin{array}{l} \Phi(x,j,k_0,m) b(N_v-j-k, v_i, \alpha_v) b(k, v_r, \beta) \\ b(j+s_v, r, 1-\gamma_v) b(N_d - i, d_i, \alpha_d) b(i, d_b, \beta) \\ S(s_v \mid v_r, x) S(s_d \mid d_b, x - s_v) \end{array} \right\} \quad (13)$$

where

$$\begin{array}{l} s_v = k + v_i - l,\ s_d = i + d_i - b \\ 0 \le s_v \le \min(v_r, x),\ 0 \le s_d \le \min(d_b, x - s_v) \\ 0 \le \{i,b\} \le N_d \\ 0 \le \{j,r\} \le \min(N_v, mk_0),\ 0 \le k \le N_v - j,\ 0 \le l \le N_v - r. \end{array}$$

In (13), α_d is the data packet arrival rate in a slot. Once the transition probability of the data subsystem is obtained, the steady state probability of the data subsystem can be derived as

$$\begin{array}{l} \Pi^d_{brl} = \sum_{i=0}^{N_d} \sum_{j=0}^{\min(N_v, mk_0)} \sum_{k=0}^{N_v - j} Q^d_{ijk,brl} \times \Pi^d_{ijk}, \\ \sum_{b=0}^{N_d} \sum_{r=0}^{\min(N_v, mk_0)} \sum_{l=0}^{N_v - r} \Pi^d_{brl} = 1. \end{array} \quad (14)$$

The performance measures of data traffic are the system throughput and the average packet delay. Let the system throughput be the total number of data packets successfully transmitted in a slot. Then, the system throughput can be defined as follows:

$$\eta_d = \alpha_d \left(N_d - \sum_{b=0}^{N_d} \sum_{r=0}^{\min(N_v, mk_0)} \sum_{l=0}^{N_v - r} b \Pi^d_{brl} \right). \quad (15)$$

Let the average packet delay be the elapsed time from the generation of a data packet to its successful transmission. Then, the average packet delay can be defined as

$$D_d = \frac{N_d}{\eta_d} - \frac{1}{\alpha_d} + 1. \quad (16)$$

IV. NUMERICAL AND SIMULATION RESULTS

The nominal values of parameters in the numerical analysis and simulation are listed in Table 1. In the PRS^2-CDMA protocol, the frame duration is identical to the packet generation period of voice terminals. With the values of parameters listed in Table 1, slot duration is 4 msec, and thus a frame consists of 5 slots. With the speech delay constraint in Table 1, a voice packet which fails to reserve a spreading code within 10 slots is dropped.

Table 1. Parameter values for performance analysis.

Parameter	value
CDMA channel rate	3.84 Mcps
Uplink channel bit rate	192 Kbps
Frame duration	20 msec
Speech delay constraint	40 msec
Voice sampling rate	32 Kbps
Packet overhead	64 bits
Mean silence duration	1.35 sec
Mean talkspurt duration	1.00 sec
Number of spreading codes	variable
Transmission permission prob.	variable

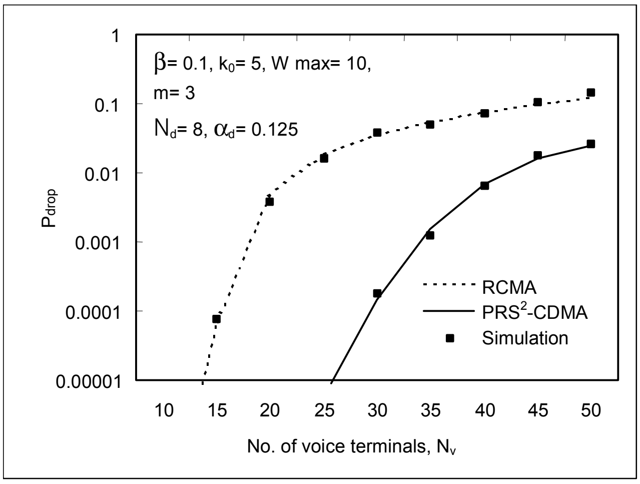

Fig. 4. Packet dropping probability P_{drop} vs. N_v with m=3 (β=0.1, k_0=5, W_{max}=10 slots, N_d=8, α_d=0.125).

Figures 4 and 5 show the voice packet dropping probabilities according to the number of voice terminals when m = 3. Those figures plotted the results compared with the RCMA protocol. The increase in N_v causes collision of reservation-request packets to occur more frequently and increases voice packet dropping. Assuming P_{drop}≤0.01, when the transmission permission probability β is 0.1, the RCMA protocol can accommodate only 22 voice terminals because a reservation-request packet has to contend with data packets. But in the PRS^2-CDMA protocol, reservation-request packets never contend with data packets. The P_{drop} in the proposed protocol, therefore, can accommodate over twice as much as the RCMA protocol. As β increases to 0.3, P_{drop} for RCMA and PRS^2-CDMA protocol increases rapidly. But as shown in Fig. 5, PRS^2- CDMA protocol gives more stable performance than the RCMA protocol in spite of the increased β.

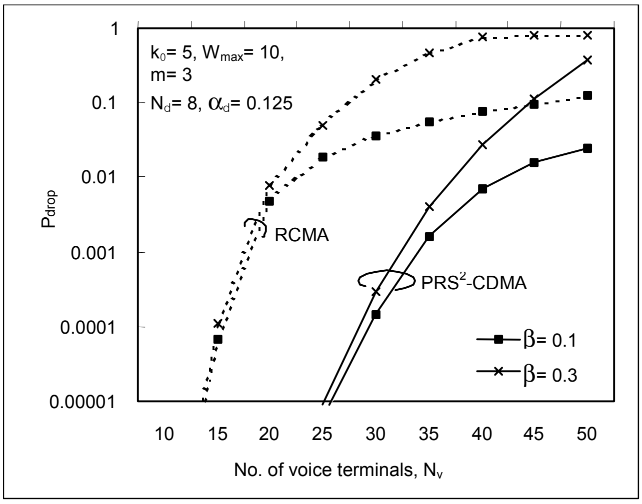

Fig. 5. Effect of transmission permission probability on packet dropping probability according to N_v (m=3, k_0=5, W_{max}=10 slots, N_d=8, α_d=0.125).

Figure 6 shows the maximum number of voice terminals supported by the proposed protocol and the RCMA within the constraint P_{drop}<0.1 at each transmission permission probability. In the proposed protocol, the reservation-request packet of a voice terminal never contends with data packets. Therefore, the proposed protocol can offer twice the system capacity of RCMA.

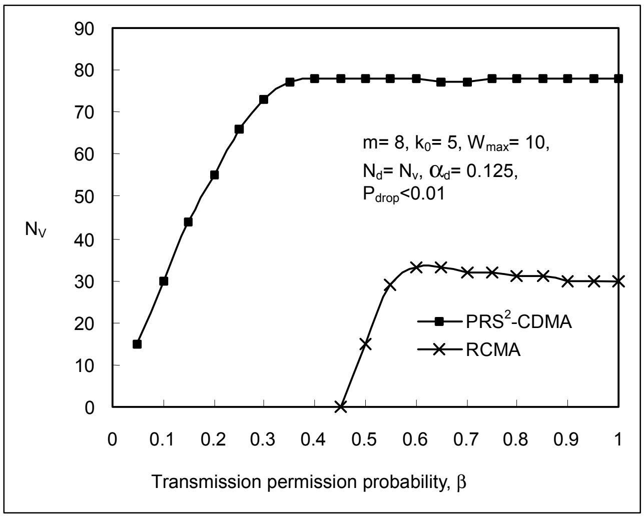

Fig. 6. Maximum number of voice terminals vs. transmission permission probability(m=8, k_0=5, W_{max}=10, N_d=N_v, α_d=0.125, P_{drop}<0.01).

Figure 7 shows the effect of transmission permission probability on the voice packet dropping probability when m=2 and m=3 respectively. The increase in β up to a threshold value causes reservation failure to occur frequently, and therefore the P_{drop} increases rapidly. In Fig. 7, the threshold values for β are 0.4 and 0.8 for the number of spreading codes m=2 and 3 re spectively.

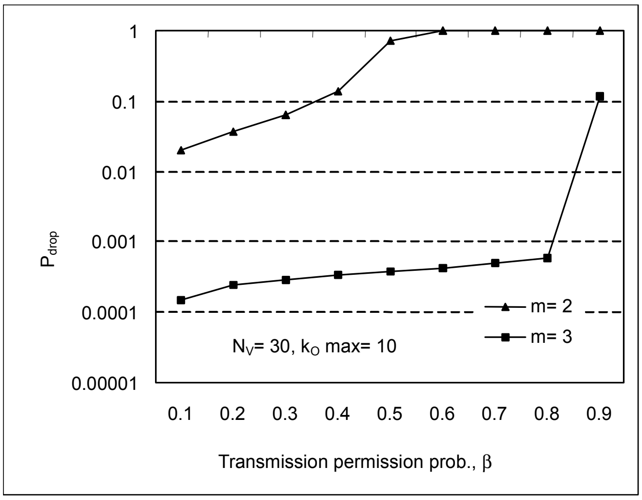

Fig. 7. P_{drop} vs. β according to m (N_v=30, k_0=5, W_{max}=10 slots).

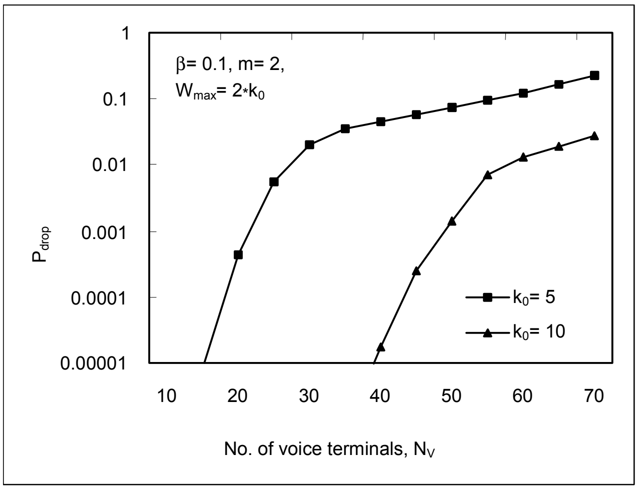

Fig. 8. Packet dropping probability P_{drop} vs. N_v according to number of slots in the uplink frame k_0 (β=0.1, m=2, W_{max}=2*k_0 slots).

Figure 8 shows the effect of the total number of uplink channels on the P_{drop}. If the transmission rate of the voice terminal is reduced from 32Kbps to 16Kbps, the total number of slots in the uplink frame will double. Though we reduce the voice sampling rate to 16Kbps, the quality of voice is still tolerable. The greater the number of spreading codes, the greater the total number of uplink channels. Also, as the number of slots increases, so does the total number of uplink channels. Therefore, P_{drop} decreases for a large number of spreading codes and slots. The number of slots in an uplink frame can be increased by reducing the voice-coding rate.

Figures 9 and 10 show the average data packet delay according to the number of spreading codes when β=0.1 and β=0.3 respectively. If the number of spreading codes is small, almost all of the codes are reserved by voice terminals because of the low access priority of data terminals. Therefore, it leads to excessive data packet collisions. However, as shown in figures, the average delay is significantly improved by increasing the number of spreading codes.

Figure 11 illustrates the effects of the transmission permission probability on the average packet delay. As shown in Fig. 11, when m is small (i.e., m=2), the increase in β up to a threshold value (about 0.3) causes the average delay to decrease continuously. However, the average data packet delay increases rapidly by increasing the transmission permission probability above the threshold. This is because at a given packet generation rate, the collision of data packets occurs frequently as data terminals try to transmit a packet with a high probability. As shown in Fig. 11, these effects do not occur when the number of spreading codes is large (m=4).

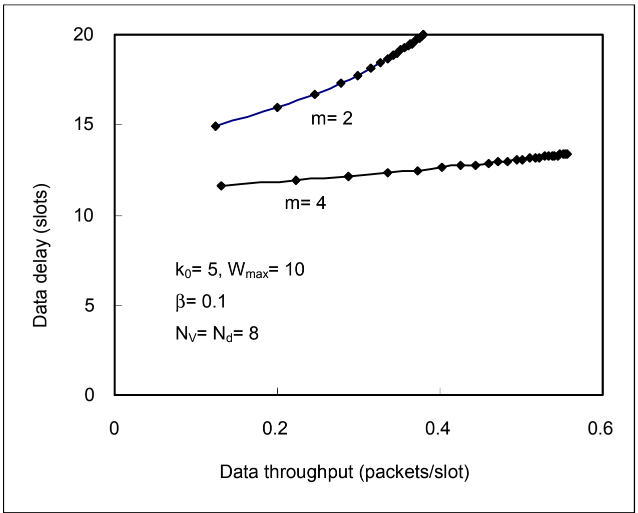

Fig. 9. Average packet delay vs. data throughput for β=0.1 (k_0=5, W_{max}=10 slots, N_v=N_d=8).

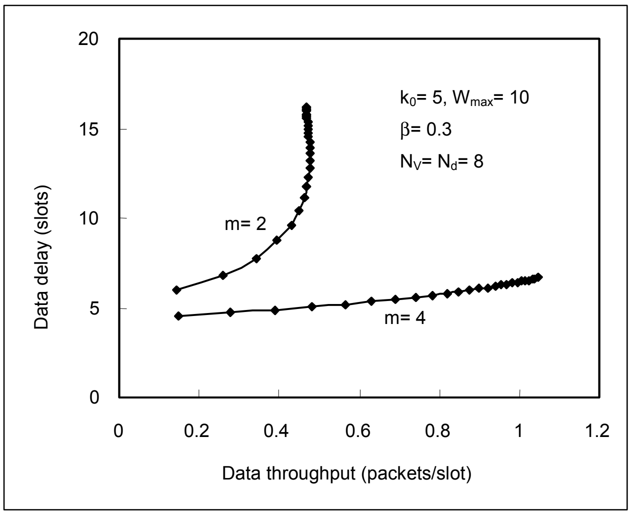

Fig. 10. Average packet delay vs. data throughput β=0.3 (k_0=5, W_{max} =10 slots, N_v=N_d=8).

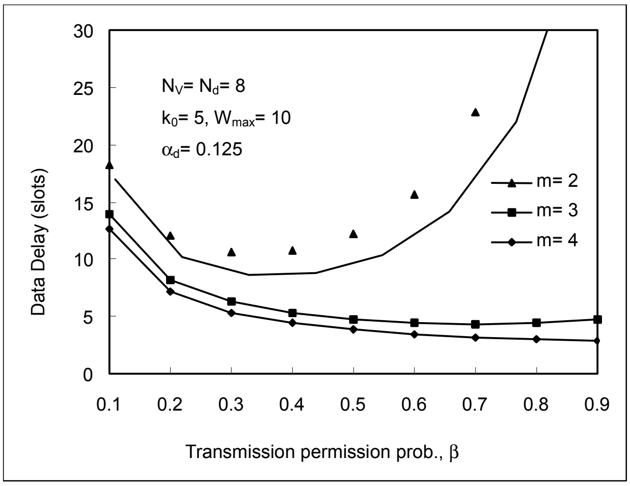

Fig. 11. Effect of transmission permission probability on the average packet delay according to m (k_0=5, W_{max}=10 slots, α_d=0.125, N_v=N_d=8).

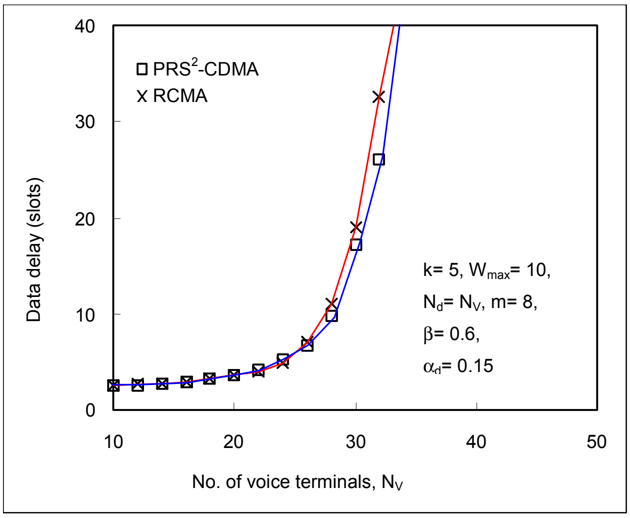

Fig. 12. Delay comparison between PRS^2-CDMA and RCMA (k_0=5, W_{max}=10 slots, α_d=0.15, N_v=N_d, β=0.6).

Figure 12 shows the comparison of average data delay between the proposed protocol and RCMA. In the proposed scheme, the number of spreading codes for a data terminal may be small because of its low access priority. In contrast, a data packet contends with other data packets. In RCMA, a data packet contends with RR packets as well as other data packets. Thus, data performance of the PRS^2-CDMA protocol is similar to RCMA.

V. CONCLUSIONS

In this paper, the PRS^2-CDMA protocol has been presented as a MAC protocol for local wireless CDMA systems supporting integrated voice and data services. The PRS^2-CDMA protocol allows voice and data terminals to share a group of spreading codes. The protocol applied a PRMA protocol and a S^{2}MA protocol to slotted CDMA systems with a small coverage area. A voice terminal can reserve a spreading code for exclusive use in transmitting a whole talkspurt to guarantee the quality of service while a data terminal has to contend for a spreading code for each packet transmission.

This protocol gives a high access priority to voice terminals, where data terminals can access an available spreading code in the current slot that are not reserved by a voice terminal in a reservation state or not used by a terminal in a reservation-contending state. This protocol accommodates the advantages of a reservation scheme for voice traffic, the simplicity of a random access scheme for data traffic, and the multiple access capability of CDMA.

Markov chain analysis is employed to evaluate system behavior, leading to the expressions for the voice packet dropping probability and the average data packet delay. The numerical analysis and simulation results have shown that the voice packet dropping probability and the average data packet delay are strongly dependent on the number of spreading codes. Increasing the number of spreading codes helps to raise the voice system capacity and improve the average data packet delay.

The PRS^2-CDMA protocol has several advantages. The first is its efficient use of the limited resource of spreading codes by allowing all terminals to share a group of spreading codes. Second, it provides a flexible means of integrating voice and data services with different priorities. Finally, the most important advantage is its large system capacity compared with RCMA. Unlike RCMA, the reservation-request packet of a voice terminal never contends with data packets in the PRS^2-CDMA protocol. As shown by numerical analysis and computer simulation, the PRS^2-CDMA protocol can offer twice the system capacity of RCMA. With these advantages, the PRS^2-CDMA protocol can be considered as a medium access control protocol for voice and data integrated local wireless CDMA systems, in which voice services are dominant.

REFERENCES

[1] W. C. Y. Lee, "Overview of cellular CDMA," *IEEE Trans. on Veh.* Tech., Vol.40, 1991, pp.291-302.

[2] A. Baier *et al*., "Design study for a CDMA-based third generation mobile radio systems," *IEEE Journal of Selected Areas in Commun*., Vol.12, No.4, May 1994, pp.733-742.

[3] IEEE 802.11, Standard for wireless LAN medium access control and physical layer specification, 1996.

[4] T. Wilkinson, T. G. C. Phipps, and S. K. Barton, "A report on HYPERLAN standardization," *Int'l Journal of Wireless Information Networks*, Vol.2, No.2, Apr. 1995, pp.99-120.

[5] M. B. Pursley, "The role of spread spectrum in packet radio networks," *Proc. of IEEE*, Vol.75, No.1, 1987, pp.116-134.

[6] D. J. Goodman *et al*., "Packet reservation multiple access for local wireless communications," *IEEE Trans. on Commun*., Vol.37, No.8, Aug. 1989, pp.885-890.

[7] R. Al-Naami and D. Saha, "Status sense multiple-access (S^{2}MA) communication," *IEEE Trans. on Commun*., Vol.40, No.6, Jun. 1996, pp.996-1002.

[8] S. Jangi and L. F. Merakos, "Performance analysis of reservation and random access protocol for wireless access networks," *IEEE Trans. on Commun*., Vol.42, No.2/3/4, Feb./Mar./Apr. 1994, pp.1223-1234.

[9] M. Soroushnejad and E. Geraniotis, "Multiple-access strategies for an integrated voice/data CDMA packet radio networks," *IEEE Trans. on Commun*., Vol.43, No.2/3/4, Feb./Mar./Apr. 1995, pp.934-945.

[10] L. Tan and Q. T. Zhang, "A reservation random access protocol for voice/data integrated spread-spectrum multiple access systems," *IEEE Journal of Selected Areas on Commun*., Vol.14, No.9, Dec. 1996, pp.1717-1727.

[11] Z. Zhang and Y. J. Liu, "Performance analysis of multiple access protocols for CDMA cellular and personal communication services," *IEEE Infocom'93*, 1993, pp.1214-1221.

[12] W. Feller, *An introduction to probability theory and its applications*, Vol.1, John Wiley & Sons, 1968, pp.98-111.

[13] L. Kleinlock, *Queueing Systems Volume 1 : Theory*, John Wiley & Sons, 1975, pp.26-44.

[14] N. M. Mitrou, Th. Orinos, and E. N. Protonotarios, "A reservation multiple access protocol for micro-cellular mobile-communication systems," *IEEE Trans. On Veh. Technol*., Vol. 39, Nov. 1990, pp.340-351.

[15] C. J. Wong and D. J. Goodman, "A packet reservation multiple access protocol for integrated speech and data transmission," *IEE Proc*., Vol.139, No.6, 1992, pp.607-612.

In-Taek Lim received the B.S. degree in computer science from University of Ulsan, Ulsan, Korea, in 1984, and the M.S. degree in computer science and statistics from Seoul National University, Seoul, Korea, in 1986. He received the Ph. D. degree in computer engineering from University of Ulsan, Ulsan, Korea, in 1998. From 1986 to 1993, he was a senior researcher at Samsung Electronics Co., Ltd. From 1993 to 1998, he was an assistant professor at Dong Pusan College. In 1998, he joined the faculty at Pusan University of Foreign Studies, Pusan, Korea, where he is currently an assistant professor in the School of Computer and Electronics Engineering. His research interests include the communication protocol design, high-speed network, and mobile computing.