Performance of Voice/Data Integration for Two MAC Protocols in DS-CDMA Wireless Networks *

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ABSTRACT

Code Division Multiple Access (CDMA) has become an attractive technique for media access control in Personal Communication Networks (PCN). In this paper, two MAC protocols, namely Preamble Signaling Access (PSA) and Minislot Signaling Access (MSA) are defined and evaluated for voice/data integration at the base station. It assumes that there are two types of mobile hosts, voice mobiles and data mobiles. The base station integrates the data and voice transmission access so that there are no contentions between voice and data users. An adaptive access control is proposed with parameters which affect the system performance. Markovian chain models have been developed to analyze the performance of PSA and MSA protocols. Performance results based upon simulations are also provided.

Keywords: Personal Communication Network (PCN), CDMA, medium access control (MAC), voice/data integration.

1 Introduction

In recent years, Code Division Multiple Access (CDMA) has become an attractive technique for medium access control in cellular networks and Personal Communication Systems (PCS). Previous research has shown that, CDMA enjoys advantageous features such as efficient spectrum utilization, simple frequency planning, soft handoff, graceful degradation, and tight security, when compared to Time Division Multiple Access (TDMA) and Frequency Division Multiple Access (FDMA) [6, 16]. A second-generation system standard IS-95 [16] and a third-generation system testbed CODIT [1] have been proposed or built, which demonstrates the feasibility of applying CDMA technology to personal communications. A CDMA personal communication network (PCN) consists of multiple *cells*, each of which is an area covered by the transmission range of a *base station* (BS). A large number of *mobile hosts* (MH) are supported within a cell. One of the most important issues for the base station is how to control the mobile hosts' uplink access to the shared wireless spectrum.

In this paper, we are only concerned with a receiver-oriented Direct-Sequence CDMA (DS-CDMA) PCN, in which a BS have several receivers, each of which listens to a specific code [11]. Only the receiver which listens to this code can recover the bits from the symbols; other receivers perceive them as noise. It is important that two MHs should use different codes for transmission at the same time; otherwise, the receiver cannot distinguish them.

Future CDMA PCNs are expected to provide a wide range of services, including both voice services and data services such as file transfer. These services will generate various types of traffic with different quality of service (QOS) requirements. For example, voice packets have stringent delay constraint but can tolerate certain loss while data packet has no delay requirement but cannot suffer any loss.

There has been extensive research done in the area of voice/data integration in CDMA wireless network. In [17], CDMA and TDMA methods are compared in an integrated PCN. Its CDMA protocol allows completely unconstrained transmission of voice and data packets. As an enhancement to this protocol, Capone and Merakos propose an integration protocol to control data traffic [4]. Data users send packet transmission requests to the BS through uplink control channels. The BS queues these requests and then schedules data packet transmission based on its observation of uplink interference level. It instructs data users when to start transmission by sending permissions and codes through downlink control channels. In [14] Soroushnejad and Geraniotis consider multiple-access strategies for an integrated FH-CDMA packet radio network. Three schemes, based on feedback information from the BS, are proposed for voice and data users to control their transmission. Admission control policies in DS-CDMA packet radio networks are discussed in [18]. Voice traffic is admitted independent of data traffic. For data traffic, two models of Multiple Access Interference (MAI) are considered, *viz.* the threshold model and the graceful degradation model. Its admission policies depend on voice admission policies as well as the MAI models. In [3] Brand and Aghvami propose a joint CDMA/PRMA protocol for the third-generation mobile systems. The time is organized into frames, each of which contains a fixed number of slots. In each slot, the access to the wireless medium is governed by transmission permission probabilities, which are calculated based on the number of users transmitting in the same slot of the previous frame. In [15], the reservation code multiple-access protocol is proposed for voice and data users to share a limited number of codes on a contention basis.

In our previous research [5], two MAC protocols, namely the Preamble Signaling Access (PSA) protocol and the Minislot Signaling Access (MSA) protocol, are proposed and compared for CDMA wireless environment. We compared the performance of these two protocols under data-only and voice-only systems. In this paper, we extend the voice/data integration protocol proposed in [19] for both the PSA and the MSA protocols. The integration protocol can be described briefly as follows: We consider a slotted system where the access control is done in a centralized manner at the the BS. For the voice traffic, we used the silence

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detection method [2]. Code assignment is considered in order to improve the system performance. Also, we consider the problem of how to control the Multiple Access Interference. In this paper, we use an adaptive access control method which is different from the approach proposed in [19].

We can compare our integration protocol with the previously proposed ones as follows: Slotted systems are considered in [3, 14, 15, 18] but not in [4, 17]. For voice services, a silence detection method is utilized in [3, 4, 15] but not in [14, 17, 18]. The code assignment problem is covered in [15]. However, it differs from our protocol in that the voice and data users contend for the same codes. In [3, 14, 17, 18], either transmitter-oriented protocol is used or the code assignment is not specified. In [4], the code assignment for voice users is not mentioned and the code assignment for data users is simplified by assuming no contention. The MAI control problem is the focus of [3, 4, 14, 18] but ignored in [15, 17]. Distributed approaches,which differ from our protocol, are used in [3, 14] while the control on a slot basis is not performed in [4, 18]. For these distributed approaches, the MHs could have outdated or incomplete information and thus the MAI could be very large. The admission controls in [18] are too coarse to cope with data traffic variation effectively.

The rest of the paper is organized as follows. In Section 2, the main features of the two MAC protocols proposed in [5] are briefly presented. Then, in Section 3, the integration protocol is introduced. Markovian models are developed and solved for these two protocols in Section 4. Section 5 presents numerical results, used to compare the performance of the two MAC protocols for voice/data integration. Finally, conclusions are given in Section 6.

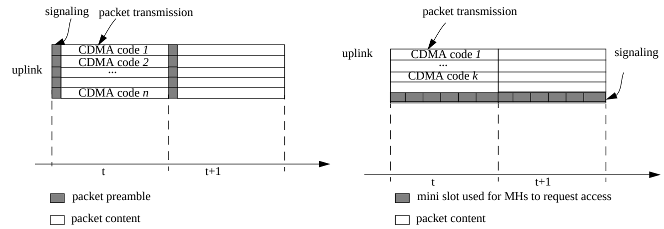
2 Two MAC Protocols for CDMA Wireless Networks

In this section, two MAC protocols, namely, Preamble Signaling Access protocol (PSA) and Minislot Signaling Access (MSA) protocol [5] will be presented.

2.1 System Architecture and Basic Assumptions

We consider a single micro-cell in a receiver-oriented CDMA PCN. We assume that the uplink (MH to BS) and downlink (BS to MH) transmissions are physically separated by means of Frequency Division Duplex. In this paper, only the uplink transmission in the CDMA network is considered and investigated. This is because the downlink access control is carried out solely by the BS while the uplink control depends on the "snapshot" status of the MHs and its control design is much more challenging. Two kinds of MHs, voice MHs and data MHs, are in the cell, which generate voice and data packets, respectively. Voice and data packets have the same size. The activities of voice MHs follow a pattern of interleaved talkspurts and silence gaps. Only during talkspurts can voice MHs generate packets, at a rate of one per slot. Voice packets can tolerate some loss but have stringent delay requirement, which prohibits retransmissions. In this paper, we assume the tolerable voice loss rate is 1% and the allowed voice delay is one slot. On the other hand, data packets have no delay constraint, and have to be retransmitted until correctly received.

An uplink transmission in a CDMA wireless packet network can be summarized as following 4 stages: (1) The MH sends a REQUEST signal to the BS, asking for a CDMA code to transmit its packets with; (2) The BS, after receiving the REQUEST from an MH, decides whether to grant the MH a code; (3) The MH, whose REQUEST is acknowledged by the BS, uses this code to transmit its packets; (4) The BS informs the MH whether the transmission is successful; If not, the MH might try to retransmit the packets. We call stages (1), (2), and (4) *signaling*, the CDMA codes the BS or the MH use to send signaling information are called *signaling codes*. In step (3), the MH is doing *packet transmission*, using the CDMA codes called *packet transmission codes*.



(a) Preamble Signaling Access (b) Minislot Signaling Access

Figure 1. Two approaches in CDMA MAC

2.2 Two Protocols for CDMA Medium Access Control

2.2.1 Preamble Signaling Access (PSA) Protocol

The first MAC protocol, Preamble Signaling Access protocol, is similar to the MAC protocols proposed in [8, 19] as depicted in Figure 1(a). Each slot of the CDMA system has a preamble part for the purpose of signaling and a packet transmission part for the packet content transmission. Using the preamble, a BS informs MHs which codes are available in next slot. An MH randomly picks up one of these codes and contends for it for packet transmission. For the PSA protocol, the preamble is what we call *signaling code*. It corresponds to one of the CDMA codes associated. The format of the preamble is as

$$Preamble = (MHid, EC),$$

where $MHid$ is the identification of the MH assigned to it when it is registered to the CDMA cell during setup or handover. EC is the error correcting code. In our research, we use BCH code for packet error protection.

If its REQUEST is acknowledged by the BS, the MH uses the same CDMA code as the preamble to transmit its packets. In contrast to previous research, in PSA, instantaneous REQUEST acknowledgment is not assumed. A REQUEST can only be acknowledged to the mobile host in the next time slot.

2.2.2 Minislot Signaling Access (MSA) Protocol

As depicted in Figure 1(b), in Minislot Signaling Access protocol, the system slot is no longer divided into preamble and packet transmission parts. Instead, during each slot, some of the CDMA codes are assigned by the BS as *signaling-purpose codes*. If a code is specified as *signaling-purpose code*, its time duration is further partitioned into a number of *minislots*, each of which is associated with one of the available codes currently in the system.

In the MSA protocol, a minislot is a *signaling code*. An MH randomly chooses one of these minislots to contend for packet transmission.

If a code is assigned as signaling-purpose code, it will be well error-protected. Throughout this paper, we assume the signaling-purpose code for the MSA protocol is so well protected that no MAI error will occur for signaling.

During different time slots, BS can specify different number of signaling-purpose codes. When the traffic is light, the BS can assign more signaling-purpose codes, and vice versa. This is because that, if there are more signaling-purpose codes, the MHs' contention will be more likely to succeed.

The format of a minislot is the same as that of preamble in the PSA protocol.

When an MH's REQEUST is acknowledged, it will use the CDMA code associated with the signaling minislot to transmit its packet. A look-up table is maitained in the BS for mapping each signaling minislot to its associated CDMA code.

As already exploited in [5], the MSA protocol performs better than the PSA protocol for many system circumstances of voice-only and data-only CDMA systems. In the following section we will focus on the protocol for the integration of data and voice traffic.

3 Voice/Data Integration

In this section, we propose an extension to the protocols we summarized in Section 2 so as to support both voice and data traffic. We first discuss how to integrate traffic of voice and data, then propose a dynamic control scheme with parameters which can be used to control the system performance.

3.1 Integrating Voice and Data Traffic

In order to assign CDMA codes to data and voice MHs, we look at the characteristics of voice and data traffic. In a voice MH, voice packets are generated constantly during talkspurt periods, which average about one second a period. Compared with the short duration of a slot, typically tens of milliseconds, the average of a talkspurt is fairly long. Besides, voice packets are sensitive to delay and will be dropped if not transmitted immediately. Given these considerations, it is not effective for a voice MH in a talkspurt to contend for a code in every slot. Rather, it seems more reasonable for the voice MH to contend for a code only at the start of the talkspurt and reserve the code for the entire talkspurt. On the other hand, the packets generated by a data MH are neither continuous nor delay sensitive. Thus the data MH will contend for a code whenever it has a packet to send.

For reasons which have been pointed out in [19], we use the approach which assigns two distinguished code sets, one for voice MHs contention, the other for data MHs contention. This is to control the packet loss due to contention; otherwise, voice packet loss would be affected by the variation of data traffic [9]. We will partition the whole set of codes associated with the BS, S, into three non-overlapped subsets, S_{rsv}, which is used for reserved voice and data MHs, S_{vc}, used for voice MH contentions at the starts of talkspurts, and S_{dc}, which is used for data MH contentions. The elements of these sets are updated and broadcast downlink by the BS at the end of every slot. To control the loss due to MAI, a limit L is set on how many request acknowledgments can be sent back by the BS for successfully acquired packets. In addition, if the number of acquired packets is beyond this limit, the voice packets should be favored by the BS in acknowledgment because of their delay constraint.

Flowcharts of the underlying procedures in our CDMA systems are shown in Figure 2, Figure 3, and Figure 4 for data MH, voice MH and BS, respectively. These flowcharts contain the operations in one slot while the dashed lines indicate the transitions into the next slot.

For a data MH (Figure 2), it has three states: (1) Thinking state (TH), in which it has no packet to transmit; (2) Acquiring state (AC), when it has a packet to transmit and is asking the BS for a code to use; and (3) Backlogged state (BL), when a data MH is assigned a code by the BS and is transmitting its packet. Once a packet is generated, a data MH will send out a REQUEST with probability p_d in the subsequent slots, asking for a CDMA code to transmit its data packet. The signaling code used is randomly selected from S_{dc}. The data MH will transmit its data packet when the REQUEST is acknowledged by the BS. It will try to retransmit the data packet unless both the REQUEST acknowledgment (QACK) and the data packet acknowledgment (DACK) have been received.

As to a voice MH (Figure 3), it does nothing during a silence gap. Once a talkspurt starts, the voice MH sends REQUEST using a signaling code randomly selected in S_{vc} with probability p_v in every slot. The code associated with the first REQUEST which is acknowledged (by QACK) is reserved for the voice MH during the whole talkspurt. A voice packet is lost if it is not transmitted in the next slot after its generation or corrupted in transmission due to MAI.

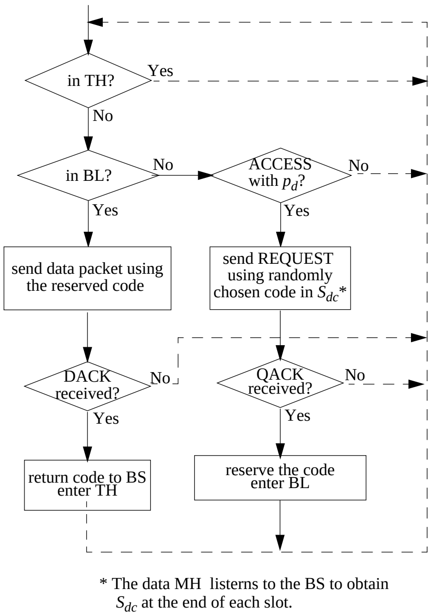


Figure 2. Flowchart for Data Terminal

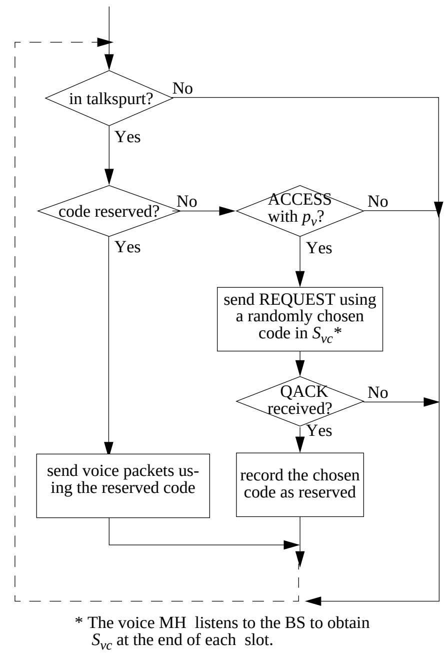
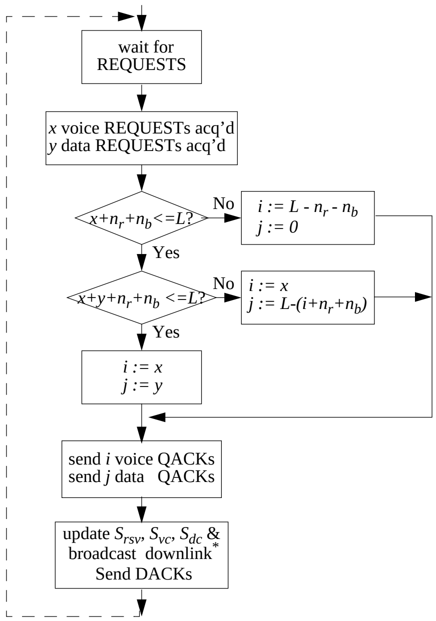


Figure 3. Flowchart for Voice Terminal

The BS (Figure 4) is the central point of control. It takes the responsibilities of: (1) determining which acquired packet preambles to acknowledge and (2) updating the signaling code sets. Let x and y be the numbers of the acquired voice and data packet preambles, respectively, and n_r the number of reserved codes. If x, y and n_r add up to less than L, all REQUESTs are acknowledged. Otherwise, the voice packet REQUESTs are acknowledged with priority. Then the BS updates the set S_{rsv} using the following two steps: (1) delete from S_{rsv} the codes associated with finishing talkspurts. (2) add to S_{rsv} the codes in S_{vc} which



* The BS also listens to reserved codes, either to catch voice packets or take back those unused codes. This is an underlying process not shown above.

Figure 4. Flowchart for BS

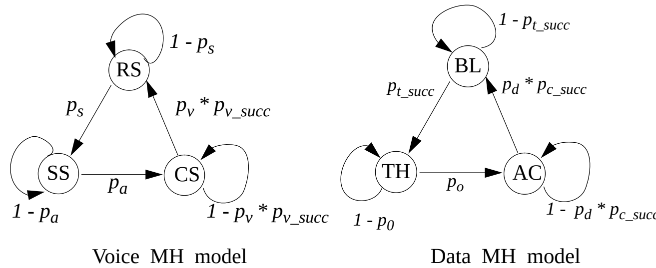


Figure 5. Models of state and transition probability of voice MHs and data MHs

are to be used in the acknowledgment of voice packet REQUESTs.

3.2 Adaptive Voice/Data Contention Control

During a certain time slot, there might be some voice MHs which just enter the talkspurt and want to require a CDMA code for packet transmission. An important system parameter is the maximum number of CDMA codes for those voice MHs to contend during one time slot. This number, C, is chosen using the following equation,

$$C = \alpha N_v - \beta N_d + c_0. \tag{1}$$

When $\alpha = 0, \beta = 0$, $C = c_0$, this is the stable method as described our previous protocol [19], which means C will remain the same value no matter what the system configuration is. In this case the number of codes for voice MHs to contend is restricted by a fixed number c_0. When $\alpha = 0$, $C = -\beta N_d + c_0$, as N_d increases, C will become smaller, until C reaches its minimum value, *e.g.* 0. When $\beta = 0$, $C = \alpha N_v + c_0$, as N_v increases, C will be larger, and thus can reduce possible contention for the voice users, until C reaches its maximum permissible value. The tuning and control of C is one of the main focuses of this paper.

Equation 1 might not be used directly in practical system design, but it shows the basic feature of choosing an appropriate value of C.

4 Analysis of the voice/data integration protocol

In this section, we present analytical models to evaluate the performance of the MAC protocols proposed in Section 2 and Section 3. First, we establish the voice and data traffic models and define the system parameters. Then, Markovian models are developed for the mixed voice/data traffic CDMA system, respectively. Several performance measures are defined based on these analytical models.

4.1 System variables and terminal models

The voice traffic and data traffic models are depicted in Figure 5. A voice MH stays in the Silence State (SS) without generating any packet. Upon the arrival of a talkspurt, it enters the Contention State (CS), where it sends its packets with probability p_v in the next slot. The voice MH moves into the Reservation State (RS) once an acknowledgment to a signaling code is received, regardless of whether the packet is correctly received or not. The probability of this is evaluated by $p_v \times p_{v\_succ}$, where p_v is the access probability of voice MHs in the CS state while p_{v_succ} is the probability of a successful contending. In the RS, the MH continuously generates and transmits packets, which may be lost due to MAI. We assume the lengths of the talkspurt and the silence gap for a voice MH to be exponentially distributed with average durations $1/\lambda$ and $1/\mu$. We also assume the probability that a voice user returns to SS before it enters RS is zero, as can be seen in Figure 5. Given a slot duration time τ, the probabilities of transitions from SS to CS and from RS to SS can be calculated as $p_a = 1 - e^{-\lambda\tau}$ and $p_s = 1 - e^{-\mu\tau}$, respectively.

A data MH is in the Thinking State (TH) if it has no packet to transmit. It generates data packets following the Bernoulli distribution with the rate of p_o. After a packet is generated, the MH moves into Acquiring State (AC) and contends for a CDMA code. It moves to the Backlogged State (BL) after obtaining a code from the BS with probability $p_d \times p_{c\_succ}$, where p_d is the access probability while p_{c_succ} denotes the contending success probability. In the BL, the data MH keeps on transmitting the packet in subsequent slots until an acknowledgment is received. The probability of a successful transmission is evaluated by p_{t_succ}. Then it moves back to the TH state.

System variables are listed in Table 1. K is a system parameter representing the number of codes associated with the BS. K can be chosen between 60 and 80, depending upon the tradeoff between the BS complexity and system capacity [19]. L is the limit imposed on the number of packets transmitted in one slot. As discussed in Section 2, we wish to determine what the optimal value of C will be for each of the protocols. For the PSA protocol, C is chosen as $C = \alpha N_v - \beta N_d + c_0$. For the MSA protocol, C can be expressed as

$$C = min(W \times H, \alpha N_v - \beta N_d + c_0)$$

where W is the number of minislots available per signaling-purpose code. As W decreases, the size of a signaling minislot will increase and thus the MAI loss for the REQUEST will decrease. H is the number of codes which are assigned by BS to be the signaling-purpose code. For the MSA protocol $W \times H$ is the total number of signaling minislots which can be used to carry REQUEST of the MHs.

4.2 Markov Model for the Voice/Data Integrated System

Given N_v voice MHs and N_d data MHs in a cell, the system can be described by four-state variables $\{N_r, N_s, N_b, N_t\}$. In order to

Variable	Description
K	number of codes associated with the BS
C	maximum number of codes for voice MHs to contend
C'	current number of codes available for voice MHs to contend
L	limit of simultaneous packet transmission
c_0, α, β	associated with C, $C = \alpha N_v - \beta N_d + c_0$
W	number of minislots per signaling code (for MSA)
H	number of signaling codes (for MSA)
τ	slot duration time
N_v	number of voice MHs in the cell
N_d	number of data MHs in the cell
N_r	number of voice MHs in the reservation state
N_s	number of voice MHs in the silence state
N_c	number of voice MHs in the contention state
N_t	number of data MHs in the thinking state
N_a	number of data MHs in the acquiring state
N_b	number of data MHs in the backlogged state
$1/\lambda$	average of talkspurt duration
$1/\mu$	average of silence gap duration
p_v	access probability of voice MHs in the CS state
p_o	packet generation rate of data MHs in the TH state
p_d	access probability of data MHs in the AC state

Table 1. System variables

find the stationary distributions of these variables, a Markov chain embedded at the slot boundary can be considered. This is because the system state, denoted by $\{N_r, N_s, N_b, N_t\}$, at the $(v+1)$st embedded point (*i.e.* the $(v+1)$st time slot) depends only on that at the (v)st point. This Markov chain is ergodic since it is irreducible, aperiodic and with finite states. To solve this four-dimensional Markov chain, a simplification can be made [12] to split it into two sub-processes: a speaking-silence process and a reservation process. The first process has one-state variables $\{N_s\}$, where N_s is the number of voice MH in the Silence State. Its stationary distribution can be calculated as follows [12]:

$$\pi(n_s) = \binom{N_v}{n_s} \frac{(\frac{\lambda}{\mu})^{n_s}}{(1+\frac{\lambda}{\mu})^{N_v}}$$

where $\pi(n_s)$ stands for the probability that n_s voice MHs stay in Silence State.

The second process is an three-dimension process with state variables $\{N_r, N_b, N_t\}$. Its evolvement depends on N_s. The stationary distribution of this process, $\{\Pr[N_r = n_r, N_b = n_b, N_t = n_t \mid N_s = n_s]\}$ or $\{\pi(n_r, n_b, n_t \mid n_s)\}$, is assumed to exist [12]. The stationary distribution of the original two-dimension Markov chain, $\Pr[N_r = n_r, N_s = n_s, N_b = n_b, N_t = n_t]$, can be calculated as follows:

$$\pi(n_r, n_s, n_b, n_t) = \pi(n_r, , n_b, n_t \mid n_s)\pi(n_s) \qquad (2)$$

In order to obtain the solution to $\{\Pr[N_r = n_r \mid N_s = n_s]\}$, we first list two functions defined in [12] for our later discussion.

- $B(n,m,p)$: Binomial distribution,

$$B(n,m,p) = \binom{n}{m} p^m (1-p)^{n-m}$$

- $\Phi(a,m,q)$: the probability that q REQUEST are acknowledged when a MHs make access to m codes in one slot.

$$\Phi(a,m,q) = \frac{(-1)^q m! a!}{q! m^a} \sum_{k=q}^{\min(m,a)} (-1)^k \frac{(m-k)^{a-k}}{(k-q)!(m-k)!(a-k)!}$$

Based on the description of our protocol, the following two functions can be used.

- $P_E(z)$: probability of packet loss due to MAI when z packets are attempted to be transmitted in one slot. Suppose the packet size is n bits and at most t error bits are correctable. If we assume the bit errors are *i.i.d.* within a packet,

$$P_E(z) = \sum_{k=t+1}^{n} B(n,k,P_b(z))$$

where function $P_b(z)$ is used to approximate the probability of bit error.

- $\Psi(c_v, c_d, a_v, a_d, n_r, n_b)$: the probability that a_v REQUEST of voice packets and a_d REQUEST of data packets are acknowledged given that c_v voice packets and c_d data packets make access to the medium in a slot while n_r voice users are in RS and n_b data users in BL. This function's parameters should satisfy the following constraints: $a_v \le c_v$, $a_d \le c_d$, $a_v \le C'$, $a_d \le K - C' - n_r$ and $a_v + a_d \le L - n_r - n_b$. C' is the current number of codes available for voice MHs to contend.

$$\Psi(c_v, c_d, a_v, a_d, n_r, n_b) = \begin{cases} \Phi(c_v, C', a_v)\Phi(c_d, K - C' - n_r - n_b, a_d) \\ \quad \text{if } (a_v + a_d < L - n_r - n_b) \\ \Phi(c_v, C', a_v) \sum_{i=L-n_r-n_b-a_v}^{c_d} \Phi(c_d, K - C' - n_r - n_b, i) \\ \quad \text{if } (a_v < L - n_r - n_b) \wedge (a_d = L - n_r - n_b - a_v) \\ \sum_{i=a_v}^{c_v} \Phi(c_v, C', i) \\ \quad \text{if } (a_v = L - n_r - n_b) \wedge (a_d = 0) \end{cases}$$

Using the above functions, the one-step transition probability of the contending process, conditioned on $N_s = n_s$, can be evaluated by

$$\begin{aligned} &Pr[N_r(t+1) = r', N_t(t+1) = i', N_b(t+1) = j' \\ &\qquad |N_r(t) = r, N_t(t) = i, N_b(t) = j] = \\ &\sum_{m=max(r'-r,0)}^{N_r-r-n_s} \sum_{l=max(0,i-i')}^{min(i,N_d-i'-j')} B(r, r'-m, 1-p_s) \\ &\quad B(i,l,p_o)B(j, i'-i+l, 1-P_E(j+r+H)) \\ &\quad \sum_{\bar{m}=m}^{N_v-r-n_s} \sum_{\bar{g}=i'+j'-i-j+l}^{N_d-i-j} \Big(\Psi(\bar{m}, \bar{g}, m, i'+j'-i-j+l, r, j) \\ &\qquad B(N_v - r - n_s, \bar{m}, p_v) B(N_d - i - j, \bar{g}, p_d)\Big) \end{aligned}$$

With the transition probability, we can first construct the one-step transition matrix $\mathbf{P}$ and then solve the equations of $\Pi = \Pi\mathbf{P}$ and $\sum \pi_i = 1$ to get $\{\pi(n_r \mid n_s)\}$.

Once $\{\pi(n_r \mid n_s)\}$ is obtained for every value of N_s, the stationary distribution $\{\pi(n_r, n_s)\}$ can be calculated by Eq. (2). It can be used to derive many performance measures which will be introduced in next subsection.

4.3 *Performance measures*

The important system performance measures include the voice packet loss rate, data packet through[put and delay. In this section, they will be derived based on the Markov chain model discussed above.

The average number of voice MHs in the Contention State (CS), $E(N_c)$, the average number of voice MHs in the Reservation State (RS), $E(N_r)$, the average number of data MHs in the Thinking State (TH), $E(N_t)$, and the average number of data MHs in the Backlogged State (CS), $E(N_c)$, can be calculated as follows,

- $E(N_c) = \sum_{n_r=0}^{N_v} \sum_{n_s=0}^{N_v-n_r} \sum_{n_t=0}^{N_d} \sum_{n_b=0}^{N_d-n_t} (N_v - n_r - n_s)\pi(n_r, n_s, n_b, n_t)$
- $E(N_r) = \sum_{n_r=0}^{N_v} \sum_{n_s=0}^{N_v-n_r} \sum_{n_t=0}^{N_d} \sum_{n_b=0}^{N_d-n_t} n_r\pi(n_r, n_s, n_b, n_t)$
- $E(N_t) = \sum_{n_r=0}^{N_v} \sum_{n_s=0}^{N_v-n_r} \sum_{n_t=0}^{N_d} \sum_{n_b=0}^{N_d-n_t} n_t\pi(n_r, n_s, n_b, n_t)$
- $E(N_b) = \sum_{n_r=0}^{N_v} \sum_{n_s=0}^{N_v-n_r} \sum_{n_t=0}^{N_d} \sum_{n_b=0}^{N_d-n_t} n_b\pi(n_r, n_s, n_b, n_t)$

Let $\mathcal{L}_C$ and $\mathcal{L}_M$ be the number of voice packet loss per slot due to REQUEST contention and MAI, respectively, and $\mathcal{R}$ be the number of data packets which are correctly received per slot. Their averages can be computed as follows:

- $E(\mathcal{L}_C) = \sum_{n_r=0}^{N_v} \sum_{n_s=0}^{N_v-n_r} \sum_{n_t=0}^{N_d} \sum_{n_b=0}^{N_d-n_t} \Bigg(\sum_{\bar{m}=0}^{N_v-n_r-n_s} \sum_{\bar{y}=0}^{N_d-n_b-n_t} B(N_v - n_r - n_s, \bar{m}, p_v) B(N_d - n_b - n_t, \bar{y}, p_d) \sum_{m=0}^{\bar{m}} \sum_{y=0}^{\bar{y}} (N_v - n_r - n_s - y)\Psi(\bar{m}, \bar{y}, m, y, n_r, n_b) \Bigg) \pi(n_r, n_s, n_b, n_t)$
- $E(\mathcal{L}_M) = \sum_{n_r=0}^{N_v} \sum_{n_s=0}^{N_v-n_r} \sum_{n_t=0}^{N_d} \sum_{n_b=0}^{N_d-n_t} \Bigg(\sum_{\bar{m}=0}^{N_v-n_r-n_s} \sum_{\bar{y}=0}^{N_d-n_b-n_t} B(N_v - n_r - n_s, \bar{m}, p_v) B(N_d - n_b - n_t, \bar{y}, p_d) \sum_{m=0}^{\bar{m}} \sum_{y=0}^{\bar{y}} (n_r P_E(n_b + n_r + H))\Psi(\bar{m}, \bar{y}, m, y, n_r, n_b) \Bigg) \pi(n_r, n_s, n_b, n_t)$
  (note: for PSA protocol, $H = 0$.)
- $E(\mathcal{R}) = \sum_{n_r=0}^{N_v} \sum_{n_s=0}^{N_v-n_r} \sum_{n_t=0}^{N_d} \sum_{n_b=0}^{N_d-n_t} \Bigg(\sum_{\bar{m}=0}^{N_v-n_r-n_s} \sum_{y=0}^{N_d-n_b-n_t} B(N_v - n_r - n_s, \bar{m}, p_v) B(N_d - n_b - n_t, \bar{y}, p_d) \sum_{m=0}^{\bar{m}} \sum_{y=0}^{\bar{y}} n_b(1 - P_E(n_b + n_r + H))\Psi(\bar{m}, \bar{y}, m, y, n_r, n_b) \Bigg) \pi(n_r, n_s, n_b, n_t)$

Based on the above averages, the averages of the performance measures, namely, the voice loss rate due to REQUEST contention, VLR_C, the voice loss rate due to MAI, VLR_M, the total voice packet Loss rate, VLR, the data packet throughput, T, and the data packet average delay, D, can be calculated as follows:

- $E(VLR_C) = \frac{E(\mathcal{L}_C)}{E(N_c) + E(N_r)}$
- $E(VLR_M) = \frac{E(\mathcal{L}_M)}{E(N_c) + E(N_r)}$
- $E(VLR) = E(VLR_C) + E(VLR_M)$
- $E(T) = E(\mathcal{R})$.
- $E(D) = \frac{N_v - E(N_t)}{E(\mathcal{R})}$

5 Numerical Results

The purpose of this section is three folds: (1) to evaluate the performance of the two CDMA MAC protocols proposed in Section 2 and (2) to investigate impacts of some of the system design parameters.

5.1 Performance Evaluation Configurations

The configurations for performance evaluation are depicted in Table 2.

We consider a cell with a number of active voice or data MHs. The processing gain of the system, G, is 31. Time is divided into slots of 16 ms and 17.9 ms long for MSA and PSA respectively. The average durations of talkspurts and silence gaps are 1.0 and 1.35 seconds, respectively. BCH-codes are used for error correction for both voice and data packets. A packet is taken as a code of 255 bits. We use (255, 131, 18) BCH-code for voice and data packets except for voice packets under PSA protocol, a (255, 147, 14) code is used. A signaling code uses BCH code (31, 11, 5) [7] which means the identification number for a MH can have up to 11 bits, which is enough for a microcell environment. We use the Gaussian Approximation to determine the bit error probability [10, 13]:

$$P_b(z) = \frac{1}{2}\text{erfc}\left(\left(\frac{2(z-1)}{3G} + \frac{N_0}{E_b}\right)^{-\frac{1}{2}}\right)$$

where $erfc$ is the complementary error function [13]. E_b/N_0 is 10 dB. The voice and data transmission probabilities, p_v and p_d, are both chosen as 0.9 because as pointed out in [8], a fixed retransmission probability of 0.9 can achieve near optimal performance.

For all our configurations, we choose W equal to 8, so for the MSA, there can be 8 minislots in one code, each minislot occupies $\lfloor 255/8 \rfloor = 31$ bits.

5.2 Numerical Results

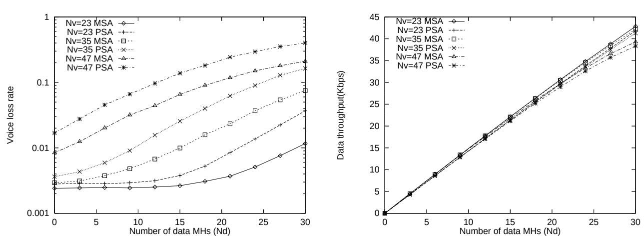
Similar to [5], we performed simulation to investigate the performance of our integration protocol. Each simulation point is an average of 3 runs lasting 3000 seconds.

Figure 6 to Figure 9 show the comparison of the performance of PSA and MSA when integrating voice and data services. From

Variable	PSA protocol	MSA protocol
CDMA processing gain G	31	31
CDMA base rate (bps)	16,000	16,000
Voice data rate (bps)	8,000	8,000
Slot length (ms)	17.9	16
Packet length (ms)	15.94	15.94
BCH code (voice)	(255,147,14)	(255,131,18)
BCH code (data)	(255,131,18)	(255,131,18)
BCH code (signaling)	(31,11,6)	(31,11,6)
$1/\lambda$(second)	1.0	1.0
$1/\mu$(second)	1.35	1.35
p_v	0.9	0.9
p_d	0.9	0.9

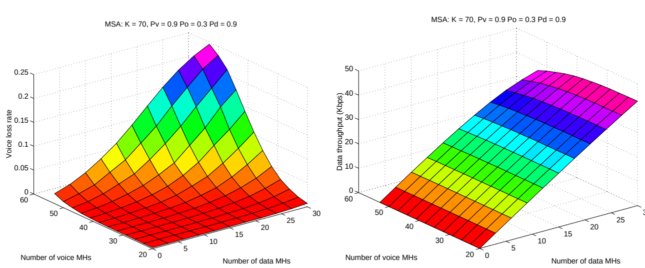
Table 2. Performance evaluation configurations

these figures we can see that for most of the time, the MSA protocol performs much better than the PSA protocol with respect to the average voice packet loss rate and data throughput. For example, in Figure 9, it is shown that, given a number of voice MHs, what is maximum number of data MHs that can be supported in the system such that the voice loss will be smaller than 0.01. when $N_v = 35$, 15 data MHs can be served with average voice packet loss rate to be less than 1% if MSA is used. However, for PSA, only 9 data MHs can be supported.



(a) Voice packet loss rate (b) Data packet throughput

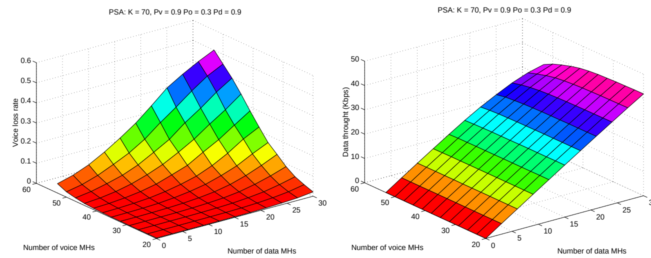
Figure 6. Integration performance of MSA and PSA (1), $K = 70, L = 40, c_0 = 5, \alpha = 0, \beta = 0, p_o = 0.3$



(a) Voice packet loss rate (b) Data packet throughput

Figure 7. Performance for MSA, $K = 70, L = 40, c_0 = 5, \alpha = 0, \beta = 0, p_0 = 0.3$

Figure 10 and Figure 11 show the impact of α and β on the integration protocols. As we discussed in Section 3.2, as α increases, the voice loss rate will drop but the data throughput will also de-



(a) Voice packet loss rate (b) Data packet throughput

Figure 8. Performance for PSA, $K = 70, L = 40, c_0 = 5, \alpha = 0, \beta = 0, p_0 = 0.3$

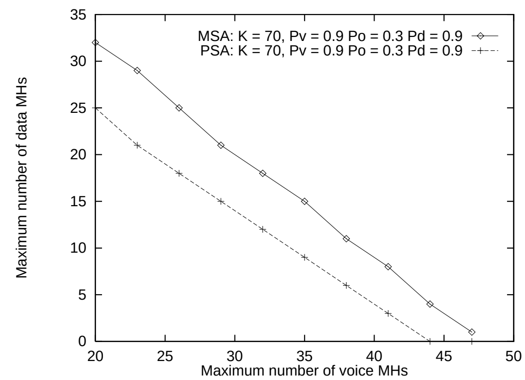


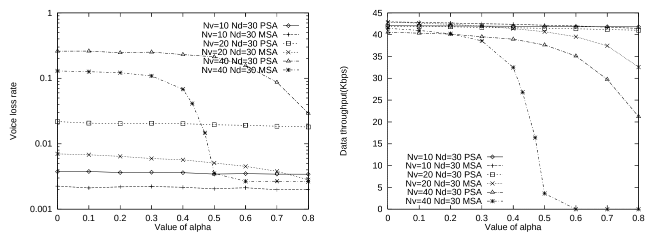
Figure 9. Voice loss < 0.01**,** $K = 70, L = 40, c_0 = 5, \alpha = 0, \beta = 0, p_0 = 0.3$

crease. Larger β, on the other hand, can help the data throughput increase but will cause more voice access contention and thus increases the voice packet loss rate. Take, for example, in Figure 10, for $K = 70, L = 40, c_0 = 5, \beta = 0, p_o = 0.3$, when $N_v = 40, N_d = 30$, as α increases to 0.47, the voice loss rate will drop to less than 1%. But the data throughput will drop from 30 Kbps to only 15 Kbps. One can find out that, when α increases to 0.5, the voice loss rate will even becomes less than the case when $N_v = 20, N_d = 30$. This is due to the impact of N_v in $C = \alpha N_v - \beta N_d + c_0$. When N_v becomes large, the C will become large, too, and will lead to significant decrease in voice packet loss rate. However, in this case, the data throughput will also decrease dramatically, as shown in Figure 10(b). These results show that in order to maintain the voice loss at some level, α can be tuned according to the current system configuration, with the sacrifice of data throughput.

For $K = 70, L = 40, c_0 = 8, p_o = 0.3, N_v = 40, N_d = 20$, one can use Figure 11 to see the impact of both α and β on the MSA protocol. It can bee seen that there is no significant interaction between α and β. Both the voice loss rate and the data throughput will drop dramatically when α reaches to some value (about 0.5 in the figure). On the other hand, there no significant change in the voice loss rate and the data throughput when the value of β

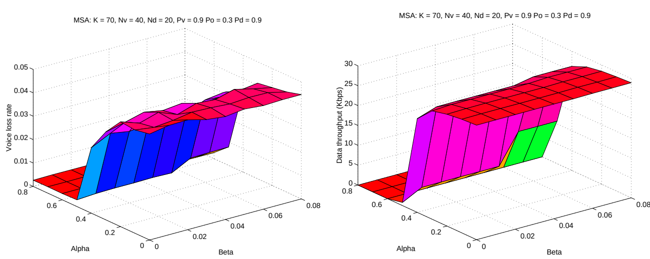
changes for a fixed value of α.

Also from these figures one can see that the MSA performs better than the PSA, which confirms the conclusion drawn from clear previous figures.



(a) Voice packet loss rate (b) Data packet throughput

Figure 10. Impact of α (1), $K = 70, L = 40, c_0 = 5, \beta = 0, p_o = 0.3$



(a) Voice packet loss rate (b) Data packet throughput

Figure 11. Impact of α and β on MSA, $K = 70, L = 40, c_0 = 8, p_o = 0.3, N_v = 40, N_d = 20$

6 Summary

In this paper, we have studied two voice/data integration MAC protocols for CDMA PCNs. One is MSA, which uses mini-slot techniques to solve the problems of code assignment and MAI control. The other one, PSA, has been proposed in previous CDMA MAC research. The integration protocol separates data and voice access contention. An adaptive access control is proposed with parameters which can be used to control the system performance. We have used Markovian chain models to analyze the performance of these two protocols. Based on the expected values obtained from the modeling, we have compared the performance of these two protocols and found that in many cases, the MSA protocol performs better than the PSA protocol, especially for voice service. Future work can be conducted on integration issues for other wireless traffic such as wireless video. More comprehensive research regarding the adaptive control access can also be investigated. For example, the control of the threshold number of simultaneously transmitted packets, L, is conservative in our current approach. How to relax this threshold to improve the system capacity is very challenging.

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