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Admission Policies for Integrated Voice and Data Traffic in CDMA Packet Radio Networks

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Abstract

In this report, we derive optimal admission policies for integrated voice and data traffic in packet radio networks employing code-division multiple-access (CDMA) with direct-sequence spread-spectrum (DS/SS) signaling. The network performance is measured in terms of the average blocking probability of voice calls and the average delay and packet loss probability of data messages. Our admission scheme determines the number of newly arrived voice users that are accepted in the network so that the long-term blocking probability of voice calls is minimized. in the longer-term. In addition, new data arrivals are rejected, if the mean delay or the packet loss probability of data exceeds a desirable prespecified level. A semi-Markov decision process (SMDP) is used to model the system operation. Then a value-iteration algorithm is used to derive the optimal admission control. Two models for the other-user interference of the CDMA system are considered: one based on thresholds and another based on the graceful degradation of the CDMA system performance, and their performance is compared. These admission policies can be employed by either terrestrial or satellite CDMA networks and are currently being extended to networks of LEO satellites and multi-rate multi-media CDMA communications.

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CONTENTS

1 Introduction

Code-division multiple-access (CDMA) techniques find today many commercial applications beyond their traditional use in military communications. Cellular systems, mobile satellite networks, and personal communication networks (PCN) that use CDMA have been proposed and are currently under design, construction, or deployment [1]-[4]. Moreover, networks of LEO (Low Earth Orbit) satellites for world-wide (global) communications such as Loral/Qualcom's Globalstar and TRW's Odyssey that use CDMA have been proposed and are under design [5]-[6].

Desirable features of CDMA include: increased received signal quality due to protection offered against multipath fading, narrowband in-band interference (other radio signals), and intentional interference (jamming); reduced synchronization requirements (users do need to have a common clock); graceful performance degradation (to gradual addition of users or other interference); flexibility in accommodating multi-media (voice/data) users and users with data rates; privacy is provided; and low-interference coexistence with other CDMA or narrowband systems operating in the same band.

Although much is known about the multiple-access capability (MAC), that is, the number of simultaneous transmissions of distinct users that can be supported at a given bit error rate, and the throughput of CDMA networks for Poisson or binomial populations, not much has been published about policies for admitting voice and data users into the network and allocating CDMA codes to them. This report addresses exactly this fundamental issue: when should voice and data users be admitted to the CDMA network ? How should this be done efficiently so that some suitable performance measure is optimized?

The voice and data users have different traffic characteristics and requirements. Voice requires real-time delivery (i.e., negligible delay but more important delay without any variability) but can accommodate higher bit error rates. Data does not require real-time delivery (it can be queued) but it requires lower bit error rates. The admission control policy must take these features/requirements into account.

Another issue that must be taken into account in the admission policy is the model for the effects of other-user interference. Two models are available: the threshold model and the graceful degradation model which are elaborated upon later. These models were used in [7] and [8] to enable the addition of users and the mitigation of errors in protocol operation or in network monitoring (feedback) without a catastrophic disruption of the system operation.

Our previous work of [7] dealt with multiple-access protocols for voice/data integration in pure CDMA packet radio networks; the MAC of CDMA was used for the voice users and ALOHA with retransmission control was used for the data users; several feedback schemes were introduced and analyzed. This work assumes that the voice and data users have already been admitted in the system and the emphasis is on performance evaluation; no policy for admitting new users in an optimal manner is considered. The follow-up work of [8] extended that of [7] to hybrid (satellite/terrestrial) integrated voice/data networks. CDMA was used in the ground subnetwork and framed ALOHA was used on the satellite. Again performance evaluation rather than admission policy optimization was the emphasis of that paper. In this report, we consider a CDMA network of voice and data users that employ direct-sequence spread-spectrum (DS/SS) signaling. The performance measures are the average blocking probability of the voice calls and the average delay and packet loss probability of data messages as functions of the offered voice and data traffic loads. We derive an optimal admission control scheme that determines the number of newly arrived voice users that are accepted in the network so that the long-term blocking probability of voice calls is minimized. The new data arrivals are rejected, if the mean data delay (or the packet loss probability) exceeds a desirable prespecified level. A semi-Markov decision process (SMDP) is used to model the system operation. Then a value-iteration algorithm is used to derive the optimal admission control. Two performance of the threshold and the graceful degradation models for the other-user interference of the CDMA system is compared.

This report is organized as follows. In Section 2, the system models are described. The traffic analysis and admission control policies for the voice and data traffic are presented in Sections 3 and 4, respectively. Numerical results and conclusions are in Sections 5 and 6.

2 System Model

The CDMA network may have any generic architecture. The basic ingredients of our work are applicable to terrestrial (cellular, PCN) or satellite networks.

The voice call or data message of each user admitted in the network is packetized with the same fixed length packet. Time is divided into slots of duration equal to the transmission of one packet. In our model, packet transmissions start at common clock instances and packets have constant length. The typical packet length is 1000 to a few thousand bits.

2.1 Model for Voice Traffic

The population of voice users is assumed to consist of maximum number of M_v users. The traffic generated by each voice user is modeled as a two-state discrete-time Markov chain with the same parameters. The transition probabilities between the two states (ON/OFF or idle/active periods) at any packet-slot are α_v and β_v, as shown in Figure 1. Thus, the steady-state probability of k active voice users is

$$P\{k \text{ active voice users}\} = \binom{M_v}{k}(p^v_{active})^k(1-p^v_{active})^{M_v-k} \tag{1}$$

where

$$p^v_{active} = \frac{\alpha_v}{\alpha_v+\beta_v}, \quad p^v_{idle} = 1 - p^v_{active} \tag{2}$$

The mean duration of the idle and active periods are $1/\beta$ are $1/\alpha$ (in packet-slots); the integer parts should be taken here to ensure that an integer number of packets results. The probability that the duration of a voice call is k packets is

$$P\{\text{voice call lasts for } l \text{ slots}\} = (1-\beta_v)^l\beta_v \tag{3}$$

This model only describes the ON/OFF (call initiation/call termination) behavior of voice traffic; the silent and talkspurt periods of the calls are not modeled. It is assumed that the CDMA system monitoring can not distinguish between silent and talkspurt periods of the calls and thus can not use the silent periods to transmit data messages or other voice calls. However, due to the fact that the voice activity factor of half-duplex is 0.4 (and of full-duplex is 0.8) the effective MAC of the CDMA system may be increased by a factor of 2.5 (or 1.25). By this we mean that although the silent periods can not be distinguished and exploited, we can take advantage of the graceful degradation of the CDMA system and assign codes to $2.5K_v$ rather than to K_v (=MAC) users in anticipation of only $0.4 \times (2.5K_v) = K_v$ being in talkspurt mode on the average.

2.2 Model for Data Traffic

The model for data traffic considered in this report is appropriate for **forward links** in CDMA wireless network scenarios; for example it is suitable for the link from the gateway to the mobiles (through the bent-pipe satellite) for satellite networks or from the base station to the mobiles for cellular networks. The key principle for the applicability of the data traffic analysis of this report is the concentration of data traffic at the site (e.g., gateway or base station) prior to transmission using CDMA. We can also analyze the data traffic over the forward link (mobile to satellite gateway or to base station) by using the data model and methodology of [7] and [8]; however, it will be more complicated to derive the admission policies for data than it is in this report. Thus we defer such analysis to a future paper. By contrast, we emphasize that the traffic analysis and derivation of admission policies for voice traffic is applicable to both the forward and return links of CDMA networks.

The number of arriving data packets (from all data users) is assumed to be Poisson distributed with mean rate λ_d (in packets per time-slot). We can also consider a finite population model (e.g., a binomial population model) but this will complicate significantly the data traffic analysis. Since, when the population of data users is large or when they have packets to transmit with small probability the binomial model approaches the Poisson model, our modeling assumption is not so restrictive. All data packets are first stored in buffers and then transmitted at the next time-slot under a CDMA multiple-reception policy. If the data packets are not successfully received, the packets are retransmitted.

2.3 CDMA System and Interference Models

Direct-sequence code-division multiple-access (DS/CDMA) is employed by all users in the network. The same frequency band is shared by voice and data traffic. Voice and data traffic have the same data rate; thus the same pool of CDMA codes is used. This assumption can be relaxed and a truly multi-rate system can be considered; this is the subject of a forthcoming paper. Each user (data or voice node) employs a distinct code for the transmission of its packets.

In the CDMA several packets form distint users can be received correctly (multiple-reception capability) according to the threshold model and the graceful degradation model described next.

We define the **multiple-access capability (MAC)** index K_v as the number of voice users that can be accommodated simultaneously, such that the expected packet error probability of voice traffic remains below a specified threshold. Similarly, the MAC index K_d for data users is the number of data users that can transmit simultaneously with a tolerable packet error. Thus we have

$$P_E(k) \leq P_E^v, \quad \forall\, k \leq K_v \tag{4}$$

$$P_E(k) \leq P_E^d, \quad \forall\, k \leq K_d \tag{5}$$

where P_E^v and P_E^d are the maximum tolerable voice and data packer error probabilities, respectively, and $P_E(k)$ is the packet error probability in the presence of k simultaneous packet transmissions, where k includes both burst voice users and active data users. In practice, $P_E^v > P_E^d$ (e.g, 10^{-3} versus 10^{-5}), since voice can tolerate higher bit error rates than data, and, therefore, $K_v > K_d$. If the total number of simultaneous users is $k \leq K_d$, then all k data or voice packets are received with acceptable error probabilities; if $K_d < k \leq K_v$, then, among the k packets, the voice packets are received with acceptable error probabilities, whereas the data packets have higher than the acceptable error probability; finally, if $K_v < k$, then all voice or data packets are received with unacceptable error probabilities. The model described above is referred to as the **threshold model**; its usefulness and simplicity lies in characterizing acceptable operating conditions in terms of maximum allowable numbers of simultaneous users. Shown in Figure 2 are the boundaries (moving) for allocation of CDMA codes to voice and data users. In our CDMA network voice packets have priority over data packets and the threshold model is appropriate for the other-user interference due to voice traffic.

For the multiple-reception of data packets both the threshold model and the **graceful degradation model** can be used. According to the latter model, there is a non-zero probability of correct reception for any arbitrary number of packets (even for numbers exceeding the MAC index), which depends on the total number of simultaneously transmitted packets (data and voice) in the CDMA network. This takes the form

$$
\begin{aligned}
&P_{gdm}(l \text{ successful data packets} \mid m \text{ voice and } n \text{ data packets are transmitted })\\
&\approx \begin{cases} \begin{pmatrix} n \\ l \end{pmatrix} [1 - P_E(m+n)]^l [P_E(m+n)]^{n-l}, & \text{if } 0 \leq l \leq n \\ 0, & \text{elsewhere} \end{cases}\\
&= P_{gdm}(n, l, P_E(m+n)) \qquad (6)
\end{aligned}
$$

where P_E is the standard probability of packet error for a direct-sequence CDMA system. From [9], we can obtain an upper bound on the packet error probability by considering a chip- and phase-synchronous system. If binary convolutional codes of rate 1/2, constraint length 7, and Viterbi decoding with hard-decisions are used (as for example in [9]), the upper bound of packet error probability with the parameters $N = 127$, $K = 12$, $L = 1000$, and $E_b/N_0 = 12dB$ is approximately 4.10×10^{-6}, where N is the chip rate per bit, K is the number of simultaneous spread spectrum signals, and L is the packet length.

The approximation in (6) is termed the **independence of receiver operation assumption (IROA)**; and assumes a binomial form. For frequency-hopped CDMA systems closed-form expressions were derived in [10] for the multi-reception packet probabilities and the accuracy of the IROA was verified. For direct-sequence CDMA the accuracy of the IROA is verified for first time in [11] which is also submitted to the JSAC special issue.

3 Voice Traffic Analysis and Admission Control

The performance measures of interest are the blocking probability of voice calls, the average throughput, the packet loss probability, and the delay of data users. Because voice calls have priority over data users, each accepted voice call is assigned one code for its packets; data packets successful transmissions (multiple-reception) are analyzed according to both the threshold model and graceful degradation model.

In Section 3.1, we analyze a **direct admission policy** according to which all voice arrivals are accepted if there are available codes for them. That is, if there are i active voice calls and the MAC index is K_v, we accepted only up to $K_v - i$ new voice arrivals; thus if there are more than $K_v - i$ new arrivals the surplus (above $K_v - i$) will be blocked (denied access to the system). In Section 3.2 a semi-Markov decision process is introduced for call admission control. This policy minimizes the long-term rejection rate of voice calls and is termed **optimal admission policy**. Then in Section 3.3 we compare the performance of the direct and optimal admission policies with respect to the call blocking probability.

3.1 Voice Traffic Analysis with Direct Admission Policy

Recall that M_v is the total number of voice users in the CDMA network. K_v is the MAC (multiple access capability) and it represents the maximum number of voice calls tat can be admitted in the network at any instant o time. In general $M_v >> K_v$. Actually, as discussed in Section 2.1, we may use K'_v ($\in [1.25K_v, 2.5K_v)$ instead of K_v to take advantage of the voice activity factor (talkspurt periods/total call duration) being smaller than 1. The total number of CDMA codes that are available for allocation to the voice and data users is K'_v. In the subsequent analysis K_v is used in all relevant equations, but it should be kept in mind that K'_v can replace K_v in any of them.

According to the direct admission policy analyzed in this section, if a new voice call there is an available code, then the new voice call is accepted immediately and can transmit its first packet in the beginning of the next time-slot; otherwise, the new voice call is rejected. Let the number of active voice calls be s^v, where $0 \leq s^v \leq K_v$. Thus the voice states form a Markov chain as in Figure 3 with certain transition probabilities.

Let p_{ij}^v be the transition probability from the current state $s^v = i$ to the next state $s^v = j$ and π^v be the steady state probability of being at state i, where $0 \le i, j \le K_v$. Thus we have a set of linear equations

$$\begin{cases} \pi_j^v = \sum_{i=0}^{K_v} \pi_i^v p_{ij}^v \\ \sum_{i=0}^{K_v} \pi_i^v = 1 \end{cases} \tag{7}$$

for which the transition probabilities, if we use the above admission scheme to manage the voice traffic, are given by

$$p_{ij}^v = \begin{cases} b(M_v, j, \alpha_v), & i = 0,\ 0 \le j < K_v \\ \sum_{k=K_v}^{M_v} b(M_v, k, \alpha_v), & i = 0,\ j = K_v \\ \sum_{k=0}^{i} b(i, k, \beta_v) \cdot b(M_v - i, j + k - i, \alpha_v), & 0 < i \le K_v,\ 0 \le j < K_v \\ \sum_{k=0}^{i} \sum_{m=K_v-i+k}^{M_v-i} b(i, k, \beta_v) \cdot b(M_v - i, m, \alpha_v), & 0 < i \le K_v,\ j = K_v \\ 0, & \text{elsewhere} \end{cases} \tag{8}$$

where $b(M, m, p)$ denotes the binomial distribution with parameters M and p, where $0 \le p \le 1$.

$$b(M, m, p) = \begin{cases} \begin{pmatrix} M \\ m \end{pmatrix} p^m (1-p)^{M-m}, & \text{if } 0 \le m \le M \\ 0, & \text{elsewhere} \end{cases} \tag{9}$$

where $b(0, 0, p) = 1$. The above equation is a direct consequence of the Markov model of voice traffic in Figure 1, the MAC index K_v representing the upper bound for the number of calls in the system, and the direct admission policy.

After we obtain the steady-state probability distribution by solving the above set of linear equations, we can compute the probability of blocking voice calls under the direct admission policy as follows:

$$P_B^v = \frac{\sum_{i=0}^{K_v} \sum_{k=K_v-i+1}^{M_v-i} (k - K_v + i)\ \pi_i^v\ b(M_v - i, k, \alpha_v)}{\sum_{i=0}^{K_v} \sum_{k=1}^{M_v-i} k\ \pi_i^v\ b(M_v - i, k, \alpha_v)} \tag{10}$$

3.2 An SMDP Approach To Call Admission Control

The goal of the optimal call admission policy minimize the long-run average cost per unit time. A policy that rejects certain voice calls at some instance of time (which would otherwise be accepted) may admit more calls on the average than a policy that always accepts calls, whenever there are available channels.

In order to find an optimal policy (in some practical sense) for the admission control problem we introduce a semi-Markov decision process (SMDP). A Markov decision process does not suffice here because the times between consecutive decision epochs are not identical but random.

The optimal admission control strategy, which determines the number of newly accepted voice calls, is found by minimizing the average rejected number of voice calls per unit time.

Let the state space of the CDMA network be

$$I = \{x_k = (i^{(v)}, l^{(v)}) \mid 0 \leq i^{(v)} \leq K_v, 0 \leq l^{(v)} \leq M_v - i^{(v)}\} \tag{11}$$

where $i^{(v)}$ is the number of the currently active voice calls, whereas $l^{(v)}$ is the number of new voice calls. The action space is $A^{(v)} = I \times A_k^{(v)}$, where $A_k^{(v)} = \{0, 1, 2, ..., \min(l^{(v)}, K_v - i^{(v)})\}$. It is assumed that a central controller has access to this information about the currently active (ON) and the arriving voice calls.

Let the action vector be a_k, then $a_k \in A_k^{(v)}$ which corresponds to the state k, where $k \in I$. Here the service completion epochs are introduced as fictitious decision epochs in addition to the real decision epochs, which are the arrival epochs of voice calls. Clearly, the actions a_k is equal to 0 on fictitious decision epochs, if there are no new arrivals at the epoch slot.

The transition probabilities that at the next epoch the system will be in state j, if action a_k is chosen at the present state i, are the following:

$$\begin{aligned} P_{ij}(a_k) &= P[j = (j^{(v)}, m^{(v)}) \mid i = (i^{(v)}, l^{(v)}), a_k] \\ &= P[(j^{(v)}, m^{(v)}) \mid (i^{(v)}, l^{(v)}), a_k^{(v)}] \end{aligned}$$

$$= \quad b(i^{(v)}, i^{(v)} + a_k - j^{(v)}, \beta_v) \cdot b(M_v - i^{(v)} - a_k, m^{(v)}, \alpha_v) \tag{12}$$

The above relationship follows from the fact that the events of the various voice users becoming active or turning idle are independent.

Let $C(x_k, a_k)$ be the expected cost incurred until the next decision epoch, if action a_k is selected at the present state x_k. The cost (the number of rejected calls) at the state x_k is defined as follows:

$$C(x_k, a_k) = l^{(v)} - a_k \tag{13}$$

In addition, the expected time until the next decision epoch, if action a_k is selected at the present state x_k, is $\tau(x_k, a_k)$ in packet-slots, where

$$\tau(x_k, a_k) = \lceil \{1 - (1 - \alpha_v)^{M_v - i^{(v)} - a_k} \cdot (1 - \beta_v)^{i^{(v)} + a_k}\}^{-1} \rceil \tag{14}$$

Here it is assumed that $\tau(x_k, a_k) > 0$, for all states and actions. In the above expression the entity in $\{\ \}$ represents the probability of no new arrival and no new service completion at the current system state (x_k, a_k) in any packet-slot; then $1/\{\ \ \}$ stands for the mean of the geometrically distributed number of time-slots that the system remains unchanged (holding time).

The **value-iteration algorithm** [13] of a semi-Markov decision model is applied to derive the optimal admission policy as described below.

step 0 : Choose $V_0(x_k)$ and τ such that $0 \le V_0(x_k) \le \min_{a_k}\{C(x_k, a_k)/\tau(x_k, a_k)\}$, for all x_k and $0 \le \tau \le \min_{x_k, a_k} \tau(x_k, a_k)$, for all x_k and a_k. Let $n := 1$.

step 1 : Compute the recursive function of $V_n(x_k)$ for all x_k, from

$$V_n(x_k) = \min_{a_k \in A_k} \left\{ \frac{C(x_k, a_k)}{\tau(x_k, a_k)} + \frac{\tau}{\tau(x_k, a_k)} \cdot \sum_{\overline{x}_k} P_{x_k \overline{x}_k}(a_k)\ V_{n-1}(\overline{x}_k) + [1 - \frac{\tau}{\tau(x_k, a_k)}]\ V_{n-1}(x_k) \right\}$$

and determine $R(n)$ as a stationary policy, actions maximize the right side of the above equation.

step 2 : Compute the bounds

$$l_n = \min_{x_k}\{V_n(x_k) - V_{n-1}(x_k)\} \tag{15}$$

and

$$L_n = \max_{x_k}\{V_n(x_k) - V_{n-1}(x_k)\} \tag{16}$$

The algorithm terminates and outputs policy $R^v(n)$, when $0 \le (L_n - l_n) \le \varepsilon l_n$, where ε is a prespecified bound on the relative error (accuracy).

Otherwise, go to *step 3*.

step 3 : $n := n + 1$ and go to *step 1*.

From the above algorithm, we obtain an optimal control scheme R^v for managing voice traffic. The optimal action a_k corresponding to the state x_k is obtained according to the policy R^v.

3.3 Performance Analysis of Optimal Admission Policy

In Section 3.2 we obtained the optimal call admission control policy R^v which requires taking action a_k when the network state is $x_k = (i^{(v)}, l^{(v)})$. In this section we analyze the performance of voice and data traffic under the optimal admission policy. This together with the performance analysis of the direct admission policy in Section 3.1 enable the comparison of the two policies in Section 5 (Numerical Results).

To analyze the optimal admission policy, we still have to solve the set of linear equations in (7). However, due to the rejection of some of the newly arrived voice calls (which would be accepted under the direct admission policy, under the optimal direct admission policy), the transition probability in (8) must be changed under the optimal call admission policy. The values of p_{ij}^v, where $0 \le i, j \le K_v$, become

$$\begin{aligned} p_{ij}^v &= \sum_{l=0}^{M_v - i} P[j \mid i, l]\ P_r(l|i) \\ &= \sum_{l=0}^{M_v - i} P[j \mid i, l]\ b(M_v - i, l, \alpha_v) \\ &= \sum_{l=0}^{M_v - i} b(i, i + a_k - j, \beta_v)\ b(M_v - i, l, \alpha_v) \end{aligned} \tag{17}$$

where $P_r(l)$ is the probability distribution of new voice arrivals and the action a_k corresponds to the state $x_k = (i, l)$, according to the optimal call admission policy R^v. Finally, after solving (7) with p_{ij}^v obtained from the above equation, we are able to calculate the steady-state probability distribution of the voice state π_i^v $(i = 0, 1, \ldots, K_v)$. Then the probability of blocking voice calls under the optimal admission policy is obtained from

$$P_B^v = \frac{\sum_{i=0}^{K_v} \sum_{l=1}^{M_v - i} (l - a_k)\ \pi_i^v\ b(M_v - i, l, \alpha_v)}{\sum_{i=0}^{K_v} \sum_{l=1}^{M_v - i} l\ \pi_i^v\ b(M_v - i, l, \alpha_v)} \tag{18}$$

4 Data Traffic Analysis and Admission Control

Recall that the data packet arrivals on the **forward link** [e.g, from gateway to mobiles (for satellite network) or from base station to mobiles (for cellular network)] are assumed to be Poisson with mean rate λ_d. The packet length is fixed and the transmission time of one packet is D, which is equal to the time of one packet-slot. It is assumed that the base station or gateway is equipped with a buffer of size K (in packets) for the data packets accumulated (concentrated) there. Since, voice calls have priority over data users, the number of successfully transmitted data packets strongly depends on the number of transmitted voice packets at each time-slot. In other words, the number of servers available for data packets is dependent on the number of servers already occupied by voice packets.

To evaluate the performance for data traffic in terms of the average data delay and average packet loss probability, we make the assumption that the state of voice traffic varies considerably more slowly than that of the data traffic; and thus the data traffic can reach its steady-state distribution while the value of the voice state has not yet changed. This assumption enables us to analyze the data traffic conditioned on the event that the voice traffic assumes a certain state (number of active calls in the network).

Since both the threshold and the graceful degradation models are useful for data traffic and two admission policies were already considered for the voice calls we must analyze the data traffic for the following four situations:
(i) direct admission policy for voice traffic and threshold model for data traffic,
(ii) optimal admission policy for voice traffic and threshold model for data traffic,
(iii) direct admission policy for voice traffic and graceful degradation model for data traffic, and
(iv) optimal admission policy for voice traffic and graceful degradation model for data traffic.

4.1 Data Traffic Analysis Under Threshold Model

Recall that under the threshold model the MAC indices for voice and data packets are K_v and K_d. That is, if there are s active voice calls at a particular time-slot, (with $0 \leq s \leq K_d$), then the maximum number of successful data packets in this time-slot is $K_d - s$; otherwise, no data packets are transmitted successfully. From Sections 3.1 and 3.3, we are able to obtain the steady state probability distribution π_s^v, $s = 0, 1, ..., K_v$, of the number of voice packets at any time-slot under the **direct admission policy** and the **optimal admission policy**. Let the random variable c represent the number of servers available for data packets and let $q(c)$ denote the probability distribution of c. Then under the assumption that the voice state changes much more slowly than the data state, with probability $q(c)$ an $M/D/c/K$ queue model (as shown in Figure 4) is valid for the data analysis under the threshold model.

The probability distribution $q(c)$ of the number of available servers for data packets at any time-slot can be obtained as follows:

$$q(c) = \begin{cases} \pi^v_{K_d - c}, & \text{if } 1 \leq c \leq K_d \\ \sum_{s=K_d}^{K_v} \pi_s^v, & \text{if } c = 0 \\ 0, & \text{elsewhere} \end{cases} \tag{19}$$

Assume that the data buffer at the gateway (for satellite networks) or the base-station (for cellular networks) has size K. Let the number of data packets in the buffer at the ith time-slot be n_i, where $0 \leq n_i \leq K$, for any time-slot. The number of arriving data packets at each time-slot is Poisson with mean rate λ_d. Let a_i^d be the number of arrival packets at ith time-slot, which is independent of n_i; then we have

$$n_{i+1} = n_i + a_{i+1}^d - d_{i+1} \tag{20}$$

where d_{i+1} is the number of departing data packets that are transmitted at the beginning of the $(i+1)$th time-slot; these depend upon n_i and the number of available channels for data users as $d_{i+1} = \min\{n_i, c\}$. The states $n_i = 0, 1, ..., K$ follow discrete Markov chain and their steady-state probability distribution π^d is obtained from the solution of the following set of $K+1$ linear equations

$$\begin{cases} \pi^d = \pi^d \cdot P_1^d \\ \sum_{k=0}^{K} \pi^d(k) = 1 \end{cases} \tag{21}$$

where P_1^d is the time-slot transition probability matrix for data under the threshold model. The entries of the transition matrix P_1^d are obtained from the relationships

$$P_1^d(n_{i+1}|n_i) = \sum_{a_{i+1}=0}^{\infty} \sum_{c=0}^{K_d} P(n_{i+1}|n_i, a_{i+1}^d, c) \cdot P(a_{i+1}^d) \cdot q(c) \tag{22}$$

where $P(a_{i+1}^d)$ is independent of n_i and is given by

$$P(a_{i+1}^d = k) = \frac{e^{-\lambda_d} {\lambda_d}^k}{k!}, \;\; k = 0, 1, 2, ..., \tag{23}$$

for the Poisson arriving data traffic model (at gateways or base stations). The random variable c only depends on the number of channels used for voice calls and is independent of n_i and a_{i+1}. Thus we have

$$P(n_{i+1}|n_i, a_{i+1}^d, c) = \begin{cases} 1, & \text{if } n_{i+1} = \min\{K, n_i + a_{i+1}^d - \min(n_i, c)\}, \\ 0, & \text{elsewhere.} \end{cases} \tag{24}$$

After obtaining the time-slot transition matrix P_1^d, we can compute the steady-state probability distribution $\pi^d(n_i)$, for all $0 \le n_i \le K$. From the latter we can easily obtain the average data delay and the packet loss probability, which is due to the finite buffer size at the gateways or the base stations. First we obtain the average data throughput as

$$\eta_d = \sum_{c=1}^{K_d} \sum_{n_i=1}^{K} \min(c, n_i) \pi^d(n_i) q(c) \tag{25}$$

and the average total data delay

$$D_d = 1 + \frac{1}{2} + \frac{1}{\eta_d} \sum_{n_i=1}^{K} n_i \pi^d(n_i) \tag{26}$$

where the value of D_d is in packets (time-slots), 1 is the service time of one packet, $\frac{1}{2}$ is the average waiting time from the arrival time of one packet to the beginning of the next time-slot, and the last term in the right-hand side represents the average data queueing delay.

For gateways connecting satellites and mobiles the round trip propagation time between earth stations and satellites may have to be added. If the satellites are located around 1389km and 10370km above the earth, as is the case, for example, with the Globalstar and the Odyssey system, then the respective values of R_p is about 9 msecs and 69 msecs, which are respectively equivalent to $9 \times 10^{-3}/D$ and $69 \times 10^{-3}/D$ time-slots. Finally, the packet loss probability is given by

$$P_L = \frac{\lambda_d - \eta_d}{\lambda_d} \tag{27}$$

4.2 Data Traffic Analysis Under Graceful Degradation Model

Under the graceful degradation model described in Section 2, the multi-reception probabilities $P_{gdm}(n, j, P_E(s+n))$ that j data packets are transmitted successfully given that there are s packets from active voice users and n data packets in the system are needed. These are computed for binary convolutional codes with rate 1/2 and constraint length 7 and Viterbi decoding and hard-decisions as follows. First we evaluate an upper bound on the packet error probability of a single-receiver system $P_E(s+n)$. If N is the number of chips per bit and K the number of simultaneous DS/CDMA signals that are both time- and phase-synchronous; then the symbol error probability ρ under a Gaussian noise channel from [9] is

$$\rho \approx Q\left(\left[\frac{N_0}{E_b} + \frac{(K-1)}{N}\right]^{-1/2}\right) \tag{28}$$

where E_b is the energy per information bit in the received signal, $N_0/2$ is the two-sided spectral density of Gaussian noise, and

$$Q(x) = \frac{1}{\sqrt{2\pi}} \int_x^\infty exp\left(-\frac{u^2}{2}\right) du. \tag{29}$$

From ρ, we can obtain an upper bound of packet error probability $P_E(s+n)$ from (3.29) and (4.1) in [9] and Table 1 in [12]. Then the upper bound for $P_E(s+n)$ is plugged in the expression (5) to obtain the desired P_{gdm}.

As in Section 4.1 (for the threshold model) we again have the following three equations

$$n_{i+1} = n_i + a_{i+1}^d - d_{i+1} \tag{30}$$

$$\begin{cases} \pi^d = \pi^d \cdot P_2^d \\ \sum_{k=0}^{K} \pi^d(k) = 1 \end{cases} \tag{31}$$

and

$$P(a_{i+1}^d = k) = \frac{e^{-\lambda_d} {\lambda_d}^k}{k!}, \ k = 0, 1, 2, ..., \tag{32}$$

that determine the evolution of n_i the number of data packets in the queue, the $K+1$ linear equations involving the steady-state probability distribution of $\pi^d(n_i)$, and the Poisson distribution of the number of arriving data packets. However, under graceful degradation model the time-slot transition probability matrix P_2^d is now determined by the following equations:

$$P_2^d(n_{i+1}|n_i) = \sum_{a_{i+1}=0}^{\infty} \sum_{s=0}^{K_v} P(n_{i+1}|n_i, a_{i+1}^d, s) \cdot P(a_{i+1}^d) \cdot \pi_s^v \tag{33}$$

where $P(a_{i+1}^d)$ is independent of n_i and s and

$$P(n_{i+1}|n_i, a_{i+1}^d, s) = \begin{cases} P_{gdm}(n_i, n_i + a_{i+1}^d - n_{i+1}, P_E(s + n_i)), & \text{if } a_{i+1}^d \leq n_{i+1} < K, \\ \sum_{l=0}^{n_i + a_{i+1}^d - K} P_{gdm}(n_i, l, P_E(s + n_i)), & \text{if } n_{i+1} = K, \\ 0, & \text{elsewhere.} \end{cases} \tag{34}$$

After obtaining the slot transition matrix P_2^d, we can compute the steady state probability distribution $\pi^d(n_i)$, for all $0 \leq n_i \leq K$, over cases (iii) and (iv). We also need the steady state probability distribution π_s^v, $s = 0, 1, ..., K_v$, of the number of voice packets at any time-slot under the **direct admission policy** and the **optimal admission policy** can be obtained from Sections 3.1 and 3.3. When the above entities are computed the average data throughput, the average data delay, and the packet loss probability, due to the finite buffer size at the gateways or base stations, are evaluated by

$$\eta_d = \sum_{s=0}^{K_v} \sum_{n_i=1}^{K} \sum_{l=1}^{n_i} l \cdot P_{gdm}(n_i, l, P_E(n_i + s^v)) \cdot \pi^d(n_i) \cdot \pi_s^v \tag{35}$$

$$D_d = 1 + \frac{1}{2} + \frac{1}{\eta_d} \sum_{n_i=1}^{K} n_i \pi^d(n_i) \tag{36}$$

and

$$P_L = \frac{\lambda_d - \eta_d}{\lambda_d} \tag{37}$$

4.3 Admission Control Policy for Data Traffic

Data users have lower priority over voice users. Voice packets are transmitted first and use all the CDMA codes they need (provided their number is smaller than K_v); if after these allocations there are still available codes for the data users, then the data packets are transmitted; otherwise no data packets are transmitted at all and they are queued or lost (when the buffer size is exceeded). Therefore, it makes sense to choose a threshold admission policy for data. The threshold value depends upon the voice traffic load, the admission policy, and the multiple-reception (threshold or graceful degradation) model. According to this admission policy, we find the threshold data arrival rate λ_t which corresponds to a prespecified tolerable average data delay (or packet loss probability) and compare it to the sum of the rate λ_i of data that is currently in the system and the rate of new data arrivals λ_a; in particular the policy $R^d(\lambda_i)$ is

$$R^d(\lambda_i, \lambda_a) = \begin{cases} 1 \text{ (accept)}, & \text{if } \lambda_i + \lambda_a \leq \lambda_t \\ 0 \text{ (reject)}, & \text{if } \lambda_i + \lambda_a > \lambda_t \end{cases} \tag{38}$$

5 Numerical Results

In our system, CDMA codes are first assigned to voice packets, since voice packets have priority over data packets. In this way, the performance of voice packets is not affected by data packets. In the threshold model, there are K_v codes assigned to voice packets at each time-slot, if the number of assigned voice packets is less than K_d; then the remaining ones out of the K_d can be assigned to data packets. Generally, $K_d < K_v$, since the packet loss probability of voice packets is larger than that of data packets. In our numerical evaluations, we work with $K_v = 20$ and $K_d = 15$.

Binary convolutional codes with rate 1/2 and constraint length 7, Viterbi decoding with hard-decisions are employed. $E_b/N_0 = 10dB$ and $N = 255$ are used in the numerical computations for the threshold and graceful degradation models of CDMA interference. The corresponding packet error probabilities of $K_v = 20$ and $K_d = 15$ are 1.5×10^{-3} and 6×10^{-5}.

The blocking probabilities of voice calls under the direct admission policy and the optimal admission control vs. the offered voice load are shown in Figure 5. The offered load of voice traffic is defined as the average of the active (= currently in the system + newly arrived) voice calls.

$$\begin{aligned} G_v &= \sum_{s=0}^{K_v} [(M_v - s)p^v_{active} + s]\pi^v_s \\ &= M_v p^v_{active} + (1 - p^v_{active}) \sum_{s=0}^{K_v} s\pi^v_s \end{aligned} \tag{39}$$

The blocking probability for the optimal policy is better than that in the direct policy, in particular, under heavy voice loads.

The performance measures of data traffic in the numerical results are evaluated for an offered voice load is $G_v = 11.54$ and data buffer size of 30 packets.

In Figure 6, we show the average data throughput over the four different cases of Section 4. versus the offered (= currently in the system + newly arrived) data load [i.e., $\lambda_i + \lambda_a$ in (38)]. The average data throughputs of the two admission policies (direct and optimal) for voice traffic are almost the same. The main difference of the data throughput is the model for multiple-receptions of data traffic. The data throughput under the graceful degradation model is much higher than that under the threshold model, when the data arrival rate is between 4 and 9.

The average delay of data traffic vs. the offered data rate is displayed in Figure 7. Again there is no perceptible difference between the direct and the optimal admission policies for voice traffic. However, the average data delay under the graceful degradation model is substantially smaller than that under the threshold model.

Figure 8 shows the packet loss probability of data traffic vs. the offered data load. From the log-scale figure we see that the packet loss probability under the optimal admission policy for voice is a little worse than that under the direct admission policy. Moreover, the packet loss probability under the threshold model is slightly smaller than that under graceful degradation model when the data traffic load is light. On the other hand, there is a substantially higher packet loss probability under the threshold model when the data traffic load is heavy.

For all figures on data performance (Figures 6 to 8), we may use as pre-specified threshold rates for the data admission policy of (38) values obtained from Figures 7 (for data delay) or 8 (data packet loss probability). For example for a tolerable data-packet loss probability of around 10^{-4} (Figure 8) we obtain the threshold values of 2.2 under the threshold model and 5.5 under the graceful degradation model.

6 Conclusions and Discussion

In this paper, we have developed and analyzed the performance of admission policies for integrated voice and data traffic in wireless CDMA packet radio networks. Our models and policies find applications in many commercial CDMA networks including cellular and satellite networks.

In our voice/data integration scheme, voice packets have priority over data packets. Therefore, the voice traffic analysis and derivation of admission policies was conducted first without any influence from data traffic considerations. A direct admission policy that accepts voice calls when there are sufficient unused resources (CDMA codes) and an optimal admission policy that minimizes the long-term rejection rate of arriving voice calls were derived and their performance analyzed in terms of the average blocking probability of voice calls. The optimal admission scheme was shown to outperform the direct one especially for heavy voice loads.

For data traffic two models for multiple-reception were analyzed the threshold and graceful degradation models for CDMA interference. An admission policy for data was derived which strongly depends on the voice admission scheme and the average throughput, delay, and packet loss probability of data were evaluated. The data performance was shown to change little with the different voice admission policies but it depends drastically on the the interference model used; the graceful degradation model was shown to exhibit superior performance over a broad range of data loads.

The work described in this report can be applied to both terrestrial networks (e.g., cellular) and satellite networks (e.g., mobile satellite or networks of LEOs). It is actually currently being extended by our team to networks of LEO (or MEO) satellites Globalstar and Odyssey. In this work which nears completion and will be reported in [14] and [15], path diversity due to double coverage (each mobile is seen by at least two satellites) is exploited. However, the effects of unequal received signal powers and of power control still need to be incorporated into the analysis and the design of the admission schemes.

The work described in this report is also being extended to other directions. Admission schemes deal with new arrivals (of voice calls or data messages) based on the number of current (ongoing) calls or messages. On the other hand, once the number (and type) of active users is known, the problem of dynamic allocation of CDMA codes to the voice and data users emerges under a different formulation. This problem is dealt with in our new work to be reported in [16].

Finally, extension of our admission policies to true multi-media traffic (not just voice and data) is possible. We are currently studying two scenarios: according to the first, an (arbitrary) number of groups of users each with different (but low to moderate) data rates (e..g., several Kbps) is served by a CDMA network; the application in mind is PCS communications (both terrestrial and satellite). Some preliminary results on this work will be reported in [17]. The other scenario is wireless video (of rate 256 Kbps or 348 Kbps) together with voice and data (rates lower than 32 or 64 Kbps) served by CDMA. For the latter work is under way and will be reported in forthcoming papers.

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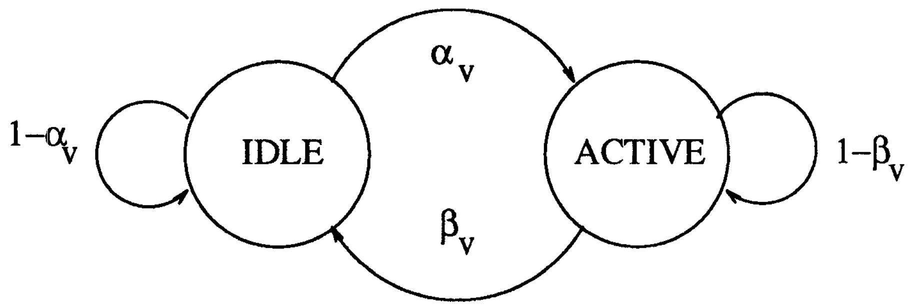

Figure 1: Model for Voice Users

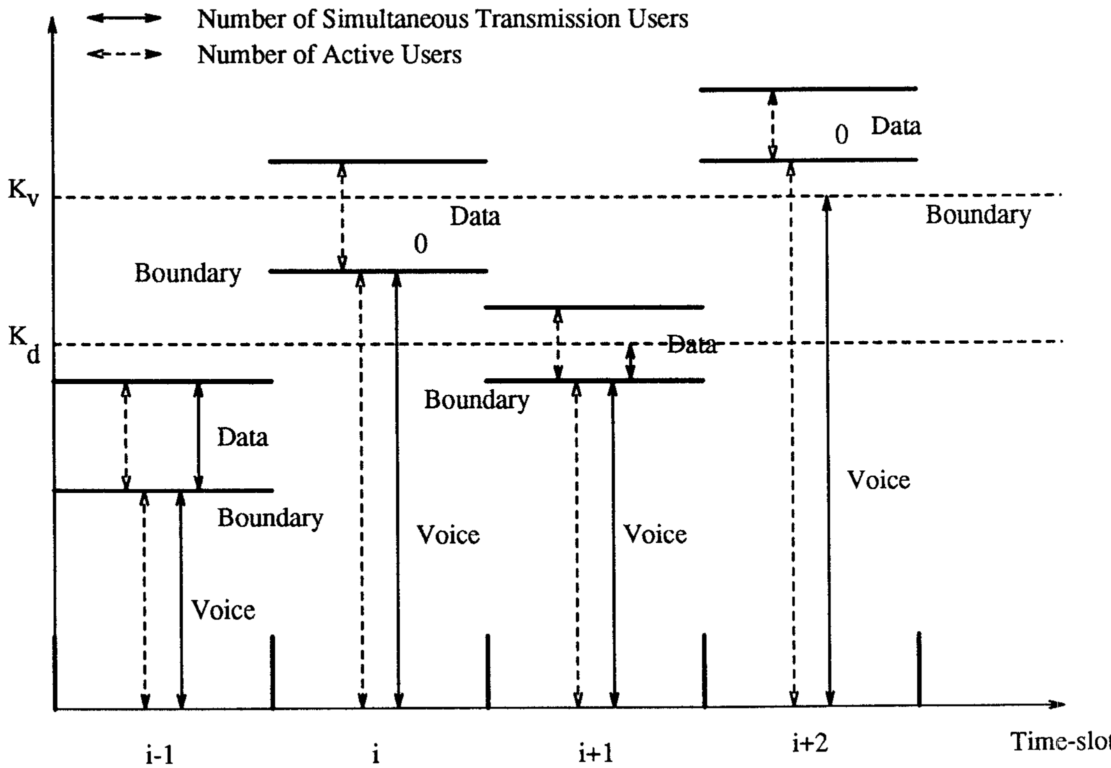

Figure 2: Threshold Model of CDMA

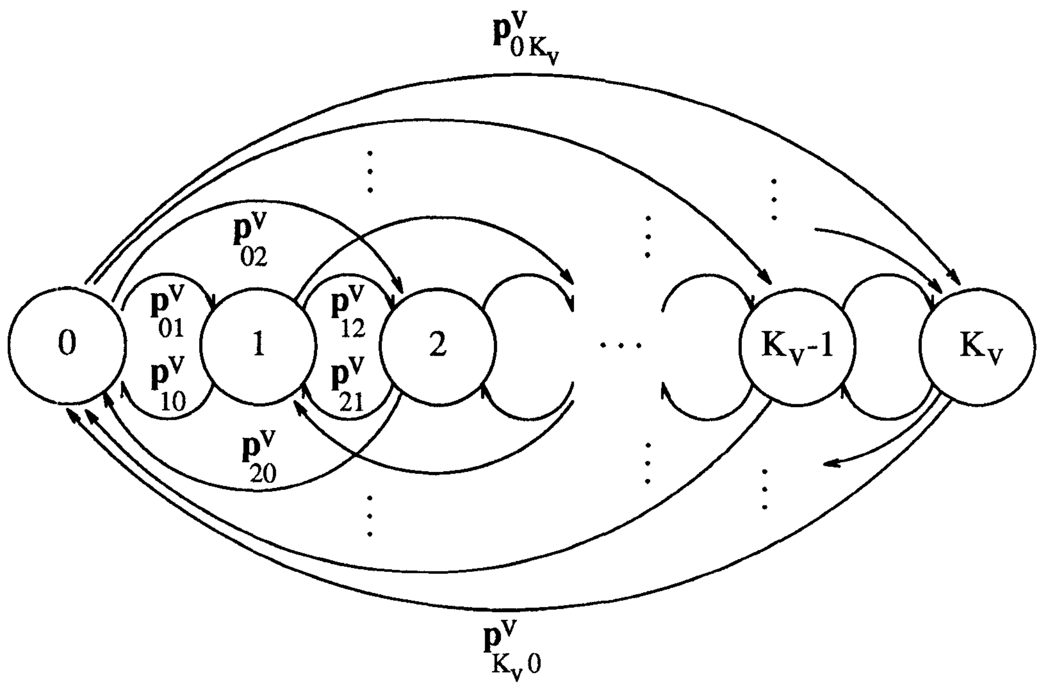

Figure 3: Markov Chain for Voice Users in CDMA Channels

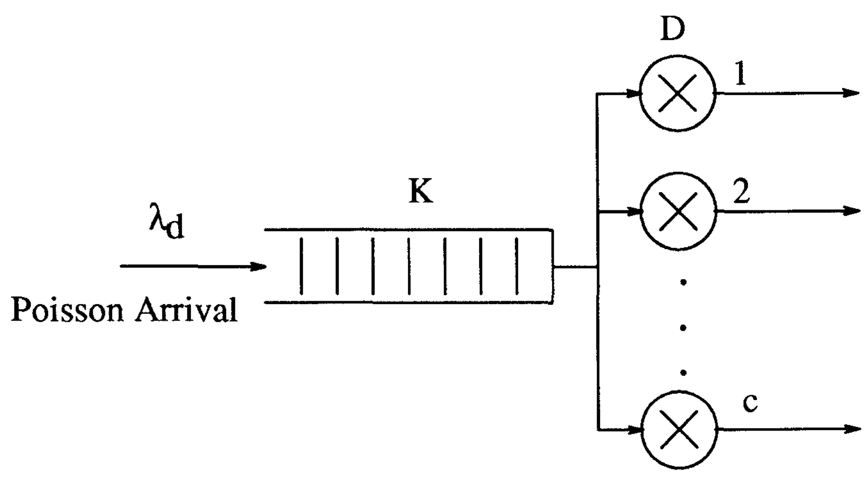

Figure 4: M/D/c/K Queue with probability q(c)

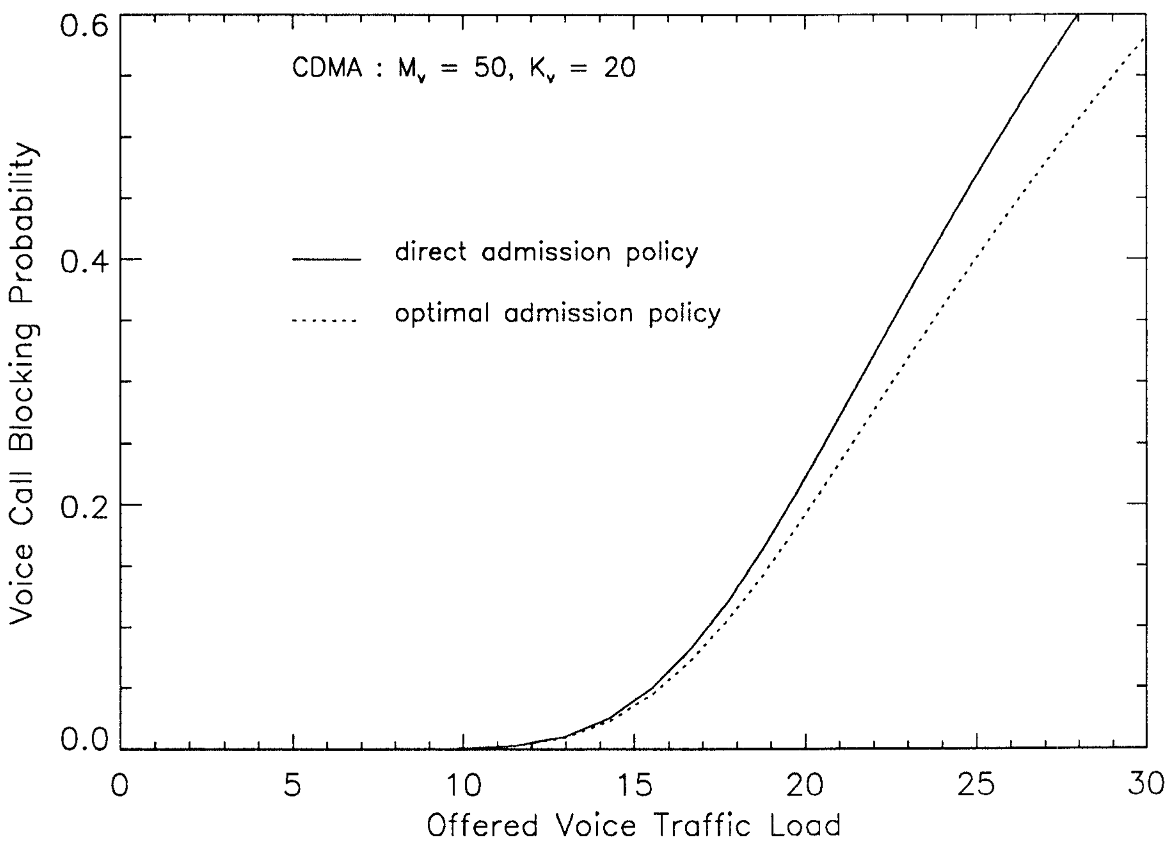

Figure 5: Average Blocking Probability of Voice Calls.

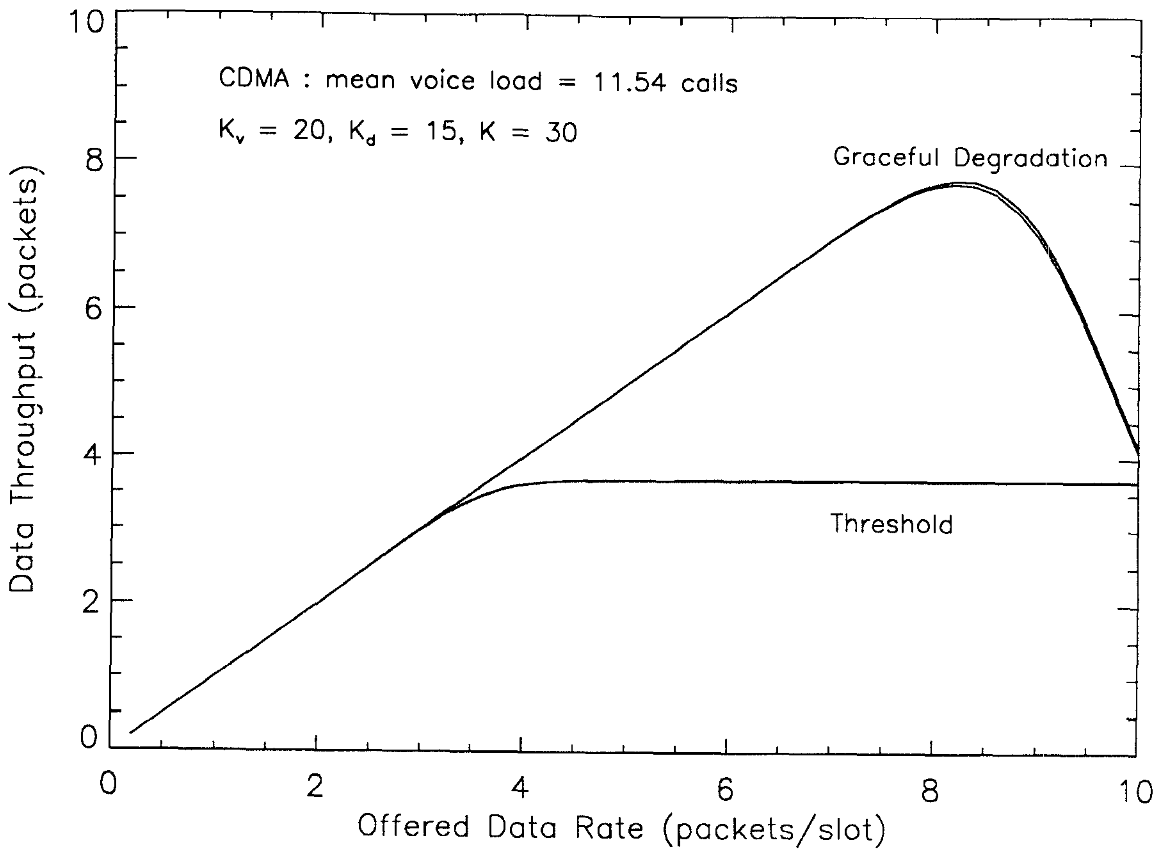

Figure 6: Average Data Throughput.

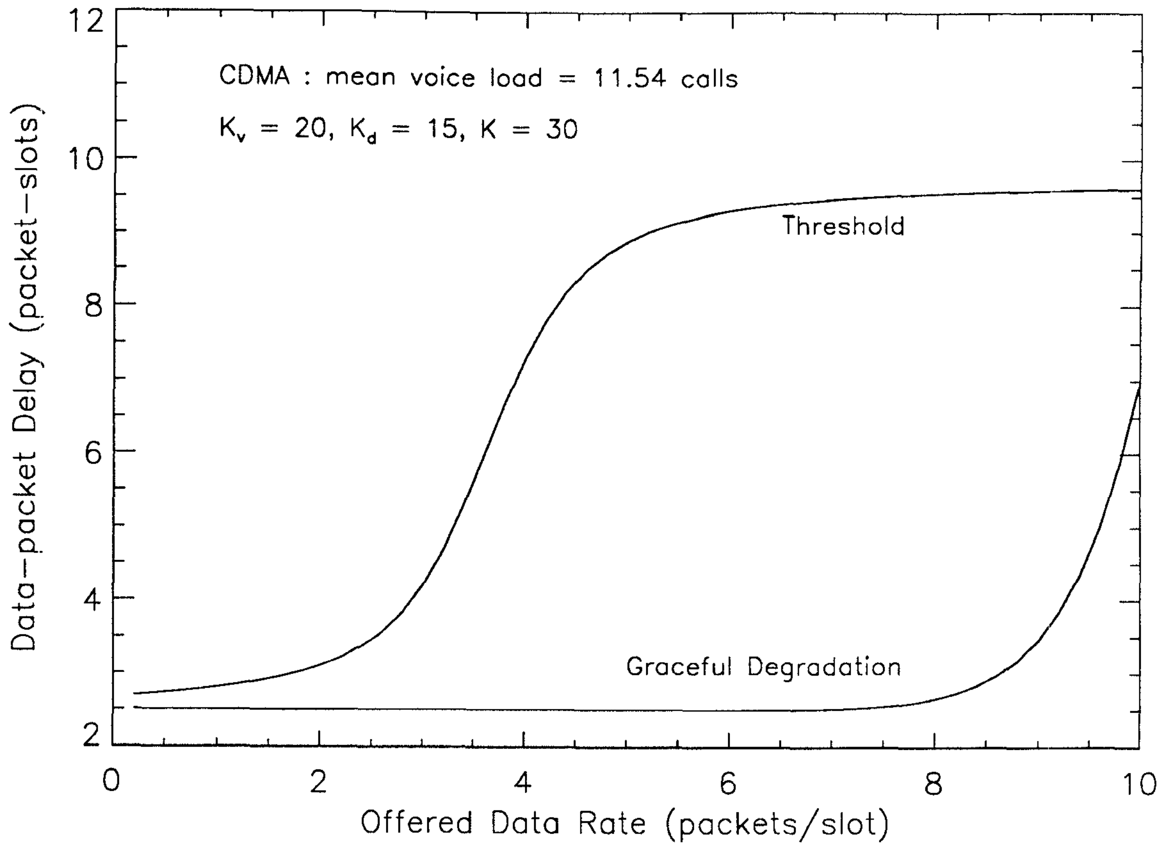

Figure 7: Average Data-Packet Delay.

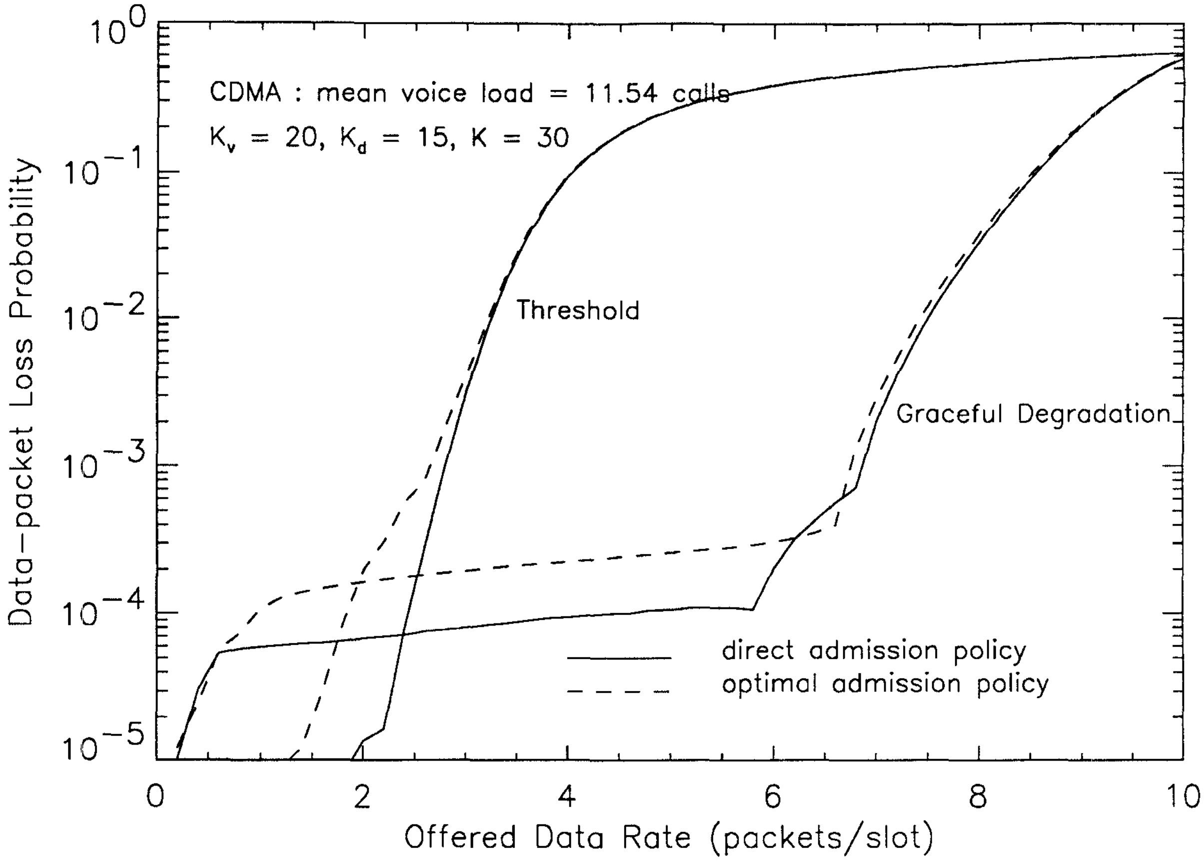

Figure 8: Average Data-Packet Loss Probability.