

Approximate performance analysis of slotted downlink channel in a wireless CDMA system supporting integrated voice and data services

Jacek Świdorski

Abstract—This paper is concerned with the performance analysis of a slotted downlink channel in a wireless CDMA communication system with integrated packet voice and data transmission. The system model consists of mobile terminals (MT) and a single base station (BS). It is assumed that the voice (data) packet error rate (PER) does not exceed 10^{-2} (10^{-5}). With this requirement the number of simultaneous transmissions over the downlink channel is limited. Therefore, the objective of the call admission control is to restrict the maximum number of CDMA codes available to voice and data traffic. Packets of accepted voice calls are transmitted immediately while accepted data packets are initially buffered at the BS. This station distinguishes between silence and talkspurt periods of voice sources, so that data packets can use their own codes for transmission during silent time slots. Data packets are buffered in queues created separately for each destination. Discrete-time Markov processes are used to model the system operation. Statistical dependence between queues is the main difficulty which arises during the analysis. This dependence leads to serious computational complexity. The aim of this paper is to present an approximate analytical method based on the restricted occupancy urn model which enables to evaluate system performance despite the dependence. Numerical calculations compared with simulation results show excellent agreement for the average system throughput and the blocking probability of data packets for higher system loads. On the other hand, when the average data packet delay is considered, analytical results underestimate simulation and therefore only approximate system performance evaluation is possible.

Keywords—*wireless CDMA system, synchronous downlink, voice and data integration, queueing analysis.*

1. Introduction

The code division multiple access (CDMA) technique has been under intense research recently [1–3]. This technique can successfully be applied as an air interface in personal communication systems (PCS). In general, a PCS should provide integrated voice and data service to its users equipped with voice and data MTs. In such a system requirements for quality of service (QoS) are different for distinct traffic types. For example, a voice traffic requires real-time delivery (i.e., a guaranteed upper bound on delay with negligible fluctuations) and reasonably low packet er-

ror rate (PER). On the other hand, for a data traffic higher packet delays are allowed (i.e., packets are allowed to be buffered) and much lower PER is required.

In most papers on voice and data integration in CDMA systems call admission control for the uplink channel is analyzed [4–10]. In [4–8], silence/talkspurt period detection of the voice traffic is included in system models, while in [9, 10] speech activity is omitted. A downlink slotted CDMA channel transmitting voice and data traffic is considered in [11]. The silence and talkspurt periods of voice sources are not modeled (the voice activity factor is taken into account instead) and only one queue concentrating the whole data traffic at the BS is analyzed. This means that different destination MTs are not distinguished by the system model. Finally, in [12], approximate performance analysis of heavily loaded slotted downlink channel in a wireless CDMA system supporting integrated voice and data services is presented. The silence and talkspurt periods of voice sources are modeled and multi-packet data messages are assumed as an input, non-symmetric data traffic.

This paper is concerned with the performance analysis of a slotted downlink channel in a wireless CDMA communication system with integrated packet voice and data transmission. Its objective is to present an approximate analytical approach, which includes speech detection and makes the multi-queueing analysis of the downlink channel computationally tractable for the possibly wide range of the system load. Therefore, this paper can be considered as a contribution to the family of papers [4–12] devoted to performance analysis of voice and data integration in wireless CDMA systems.

2. System model

The system consists of MTs and a single BS. An MT can receive voice and/or data packets transmitted from the BS. It is assumed that a MT, which is able to receive voice and data traffic in the same time slot, has two receivers: one to receive voice and one to receive data packets. It is assumed that there are k MTs which are able to receive data packets in the system. Both voice and data packets are transmitted with the same rate and with the same energy per bit, thus the same pool of CDMA codes for both traffic

types is used. Voice and data packets are of the same size (n bits long) and the system is synchronized to slots of duration equal to the packet transmission time h_s . It is also assumed that bit interleaving and forward error correction codes with codeword length n are used.

The maximum of correctable bit errors for each voice (data) packet is assumed to be t_v (t_d). Typically, $t_v < t_d$ due to different quality of service requirements for voice and data traffic. It is assumed that the number of simultaneous packet transmissions is limited and accepted t_v (t_d) assures that the probability of receiving an uncorrectable voice (data) packet is less than 10^{-2} (10^{-5}).

Any downlink channel model which enables to evaluate for bit error rate (BER) as a function of k_s CDMA codes used simultaneously (i.e., BER(k_s)) can be employed. An example of such downlink channel model is presented in [13]. This model is used in this paper. Since the downlink channel is considered, all k_s simultaneously transmitted signals are both time and phase synchronous. They consist of independent and identically distributed data sequences taking values at ± 1 with equal probability. These data sequences represent voice and data packets and they are spread out with spreading codes—one code assigned to one transmitted signal. Spreading codes are sequences of independent and identically distributed random variables taking values at ± 1 with the same probability. These random variables are called chips and it is assumed that they have a rectangular pulse shape. It is also assumed that bit errors are caused only by other-user interference, allowing the effects of thermal noise on system performance to be ignored. Finally, equal received power for all signals is assumed.

Voice calls arrive at the BS according to Poisson process with a rate λ_v calls/slot. The duration of a voice call is exponentially distributed with an average $1/\mu_v$ slots and it is assumed that $1/\mu_v \gg 1$ and $1/\lambda_v \gg 1$. With the requirement that the voice PER does not exceed 10^{-2} and with proper choice of t_v , the number of simultaneous transmissions over the downlink channel is limited up to L_v^{\max} (L_v^{\max} is the maximum number of CDMA codes available for the voice traffic, i.e., the direct admission policy is accepted). As a result, voice call is accepted at the BS if there are less than L_v^{\max} voice connections being established. Rejected voice calls are lost. If a voice call is accepted, a voice source is described by the well known ON-OFF model [5, 7, 8]. The ON state represents a voice source in a talkspurt mode (a user is speaking) while the OFF state represents a silence period during conversation. The length of both ON and OFF states is assumed to be geometrically distributed with means $1/P_{on-off}$ and $1/P_{off-on}$ slots, respectively. In the ON state, a voice source generates a stream of voice packets with constant rate equal to one packet per time slot. To serve this stream (i.e., to transmit voice packets to a given destination MT) a single CDMA code is used.

Channel capacity which is not occupied by the voice traffic can be used to serve data packets. The BS distinguishes between silence and talkspurt periods of voice sources, so that

accepted data packets can use their own codes for transmission during silent time slots. Therefore, apart L_v^{\max} CDMA codes used to serve voice traffic, the BS also uses k mutually orthogonal CDMA codes to serve data packets destined for k MTs.

Packets of accepted voice calls are transmitted immediately, while accepted data packets are initially buffered at the BS. Thus, voice packets can be treated as higher priority packets with respect to data packets considered as lower priority packets. Data packets are buffered in k FIFO queues created separately for each destination (i.e., for k MTs which are able to receive data packets). It is assumed that all queues have the same maximum length L_q^{\max} (in packets) and that the external data packet traffic is symmetric. More precisely, a data packet destined for a given MT appears at the BS with probability λ_d in each time slot (λ_d is considered to be a data packet arrival rate in a single queue input). If the packet arrives at the BS and its queue is full, it is rejected and removed from the system. Moreover, since the probability of unsuccessful data packet transmission (because of downlink interference) is assumed to be less than 10^{-5} , retransmission of uncorrectable data packets is not considered (although it is possible in real systems).

The scheduling scheme enabling a given queue to use a CDMA code at the beginning of the next time slot t ($t = 1, 2, \dots$) is described as follows. The scheduler calculates the number $n_v(t)$ of active voice sources in time slot t at first. Then, the algorithm determines the number on non-empty queues $k_{q \neq 0}(t)$ and the number of queues which can be served simultaneously in the same time slot t . This number is equal to $k_q(t) = \min[k_{q \neq 0}(t), L_v^{\max} - n_v(t)]$, which means that $k_q(t)$ CDMA codes can be used simultaneously to transmit data packets. Finally, the scheduler randomly determines $k_q(t)$ out of $k_{q \neq 0}(t)$ queues which can be served. The probability that a given queue can be served in time slot t is equal to $P_q(t) = \frac{k_q(t)}{k_{q \neq 0}(t)} = \min\left[1, \frac{L_v^{\max} - n_v(t)}{k_{q \neq 0}(t)}\right]$ (i.e., this probability is the same for all queues). Thus, this random mechanism assures that the scheduler is fair with respect to all non-empty queues.

With the assumed scheduler no more than L_v^{\max} simultaneous voice and data transmissions in time slot t are allowed. As a result, a voice and data frame in the CDMA code domain is transmitted in each time slot. This frame has a movable-boundary for voice and data traffic. Assuming the number of accepted voice calls to be constant and equal to L_v , this voice and data boundary moves in the time slot domain within its range, following slot by slot changes of the number of active voice sources $n_v(t)$.

3. Analysis

For the assumed forward error correction codes with codeword length n and the maximum of correctable bit errors equal to t_v (t_d) for voice (data) traffic in conjunction with the interleaving technique, the probability of receiving an uncorrectable packet (i.e., the PER) when there are k_s si-

multaneous signals (i.e., k_s simultaneous CDMA codes) is given by [13, 14]:

$$P_{p,x}(k_s) = 1 - \sum_{i=0}^{k_s} \binom{n}{i} [P_e(k_s)]^i [1 - P_e(k_s)]^{n-i}, \quad x = v, d, \quad (1)$$

where index $x = v$ ($x = d$) is used for the voice (data) traffic and $P_e(k_s)$ is the known bit-error rate BER (k_s) on the downlink channel.

The PER expressed by Eq. (1) is an approximation which stems from the fact that this formula is exact only when bit errors are statistically independent. However, if bit interleaving is used (as it is assumed) one can expect that bit-to-bit error independence is hold [13]. Formula (1) is used to determine L_v^{\max} (the maximum number of simultaneous voice and data packet transmissions in the downlink) for assumed transmission parameters. In order to calculate the probability $P_{p,x}(k_s)$ expressed by formula (1) it is necessary to know the probability $P_e(k_s) = \text{BER}(k_s)$. With the accepted assumptions this probability can be expressed as [13]:

$$P_e(k_s) = \frac{1}{\sqrt{2\pi}} \int_{\text{SNR}(k_s)}^{\infty} \exp\left(-\frac{u^2}{2}\right) du, \quad (2)$$

where $\text{SNR}(k_s) = \left(\frac{k_s-1}{G}\right)^{-\frac{1}{2}}$ and G is the processing gain. The BER expressed by Eq. (2) is only an approximation obtained using the standard Gaussian approximation technique [13, 14]. This approximation can be good even for small number of simultaneous signals k_s , assuming relatively low processing gain G . A discussion about the accuracy of these approximation can be found in [13].

Formula (2) is used to calculate $P_{p,d}(k_s)$ expressed by Eq. (1). Moreover, the average BER for voice (data) traffic after decoding can be evaluated using the following formula [15]:

$$P_{e,x}(k_s) = \frac{1}{n} \sum_{i=i_x+1}^n \binom{n}{i} [P_e(k_s)]^i [1 - P_e(k_s)]^{n-i}, \quad x = v, d, \quad (3)$$

where index $x = v$ ($x = d$) is used for the voice (data) traffic. Since it is assumed that the length of a data packet (this length is equal to one time slot) is much shorter than the average duration of a voice call (voice connection), i.e., $1/\mu_v \gg 1$, and it is also much shorter than the average interarrival time between two consecutive voice calls, i.e., $1/\lambda_v \gg 1$, the CDMA system under consideration can be modeled at either the voice call level or at the data packet level, depending on the particular system performance aspect being investigated [5, 12].

3.1. System performance at the data packet level

Due to the mentioned assumptions, it is reasonable to assume that for a constant number L_v of accepted voice

calls the queuing system reaches the steady state before L_v changes. In order to evaluate efficiency of the system, some performance measures for the steady state are given below.

The average data packet throughput (in packets per time slot) can be expressed as:

$$\hat{\lambda}_{d|L_v} = \sum_{n_d=0}^{kL_q^{\max}} \Pr(n_d) \sum_{l=1}^{\min(n_d, L_v^{\max}, k)} l t(n_d, l), \quad (4)$$

where $\Pr(n_d)$ is the probability that there are n_d data packets at the BS, and $t(n_d, l)$ is the conditional probability of l simultaneous data packet transmissions given n_d packets at the BS (index $d|L_v$ means that data throughput is obtained by conditioning on the number of accepted voice calls L_v).

Next, the data packet blocking probability can be obtained as:

$$PB_{d|L_v} = 1 - \frac{\hat{\lambda}_{d|L_v}}{k\lambda_d}. \quad (5)$$

Finally, using Little's formula, the average data packet delay can be calculated as:

$$T_{d|L_v} = \frac{\bar{m}}{\hat{\lambda}_{d|L_v}}, \quad (6)$$

where $\bar{m} = \sum_{n_d=1}^{kL_q^{\max}} n_d \Pr(n_d)$ is the mean number of data packets at the BS (also including data packets in service).

In order to calculate the mentioned performance measures it is necessary to obtain the following probability distributions:

$$\{t(n_d, l) : n_d = 0, 1, \dots, kL_q^{\max}, l = 0, 1, \dots, \min(L_v^{\max}, k)\},$$

$$\{\Pr(n_d) : n_d = 0, 1, \dots, kL_q^{\max}\},$$

$$\{\Pr(n_v) : n_v = 0, 1, \dots, L_v\}.$$

These probability distributions can be obtained using an analytical approach described below.

Two discrete-time Markov processes are used to model the system operation. Namely, with the accepted assumptions, once voice calls are admitted at the BS, their behavior depends only on their traffic sources and therefore the state evolution for the voice traffic is independent not only of the data traffic, but of the states of queues at the BS as well. Therefore, the system is considered as two statistically independent subsystems, each of which can be described by a discrete-time Markov chain. The first chain represents behavior of voice sources, while the second chain describes all (k) queues at the BS.

The state space of the chain representing the voice sources can be expressed as $\mathbf{S}_v(t) = \{n_v(t) : n_v(t) = 0, 1, \dots, L_v\}$, where $n_v(t)$ is the number of active voice users (i.e., speaking users) in time slot t . The probability of a transition

from state $n_v(t)$ to state $n_v(t+1)$ can be expressed by the following formula [12]:

$$\Pr(n_v(t+1)|n_v(t)) = \sum_{x=0}^{\alpha} \binom{\alpha}{x} p^x (1-p)^{\alpha-x} \times \left(\begin{matrix} L_v - \alpha \\ |n_v(t+1) - n_v(t)| + x \end{matrix} \right) q^{|n_v(t+1) - n_v(t)| + x} (1-q)^{\beta-x}, \quad (7)$$

where

$$\{\alpha, \beta, p, q\} =$$

$$= \begin{cases} \{n_v(t), L_v - n_v(t+1), P_{on-off}, P_{off-on}\}, & n_v(t+1) \geq n_v(t) \\ \{L_v - n_v(t), n_v(t+1), P_{off-on}, P_{on-off}\}, & n_v(t+1) < n_v(t) \end{cases}$$

One can notice that from a given state $n_v(t)$ there are always $L_v + 1$ possible transitions to all feasible states $n_v(t+1) \in \mathbf{S}_v(t)$.

Since all k queues at the BS are served by one common scheduler, these queues are always statistically dependent. Therefore, these queues cannot be analyzed separately and a system of $(1 + L_q^{\max})^k$ linear equations has to be solved [12, 16]. However, in practice, the problem is intractable because of its computational complexity. In this situation an approximate analytical approach is used. The queueing system which stores and serves data packets at the BS is analyzed by expanding a linear approximate model proposed in [16] and then used in [17]. It is assumed that the system state depends only on the aggregate number n_d of data packets in the queueing system. This assumption allows to construct a Markov chain with $kL_q^{\max} + 1$ linear equations. The state space of this Markov chain can be expressed as $\mathbf{S}_d(t) = \{n_d(t) : n_d(t) = 0, 1, \dots, kL_q^{\max}\}$, where $n_d(t)$ is the number of data packets at the BS in time slot t . The probability of a transition from state $n_d(t) = i$ to $n_d(t+1) = j$ can be obtained as:

$$\Pr(n_d(t+1) = j | n_d(t) = i) = P_{i,j} = \sum_{l=0}^{\min(k,i)} a_{j-i+l}(i) t(i,l), \quad (8)$$

where $a_j(i)$ is the probability of j packets arriving in a time slot given i packets in the queueing system (i.e., at the BS), and $t(i,l)$ is the conditional probability of l data packet transmissions given i packets in the queueing system. To calculate the transition probabilities the restricted occupancy urn model is used [16].

Let $R(i, k, L_q^{\max})$ be the number of ways of distributing i indistinguishable balls (or data packets) among k distinguishable urns (or queues) under L_q^{\max} -(occupancy) restriction, given by:

$$R(i, k, L_q^{\max}) = \sum_{j=0}^k (-1)^j \binom{k}{j} \binom{i+k-j(L_q^{\max}+1)-1}{k-1}. \quad (9)$$

Let also $R(i, k, L_q^{\max} | b_r = u)$ be the number of ways distributing i indistinguishable balls (or data packets) among k distinguishable urns (or queues), so that exactly

u urns (or queues) have exactly r balls (or packets) under L_q^{\max} -restriction, derived as:

$$R(i, k, L_q^{\max} | b_r = u) = \binom{k}{u} \sum_{j=0}^{k-u} (-1)^j \binom{k-u}{j} R(i - (u+j)r, k - (u+j), L_q^{\max}). \quad (10)$$

Therefore:

$$\Pr(b_r = u | i) = \frac{R(i, k, L_q^{\max} | b_r = u)}{R(i, k, L_q^{\max})}. \quad (11)$$

Using the restricted occupancy urn model, the probability $a_j(i)$ can be expressed as:

$$a_j(i) = \sum_{u=j}^k \frac{R(i, k, L_q^{\max} | b_{L_q^{\max}} = k-u)}{R(i, k, L_q^{\max})} \binom{u}{j} \lambda_d^j (1-\lambda_d)^{u-j}. \quad (12)$$

Next, the conditional probability $t(i,l)$ can be calculated as:

$$t(i,l) = \sum_{d=l}^{\min(k,i)} \frac{R(i, k, L_q^{\max} | b_0 = k-d)}{R(i, k, L_q^{\max})} s(d,l), \quad (13)$$

where $s(d,l)$ is the probability of l data packet transmissions in a time slot, given d non-empty queues. For the analyzed queueing system this probability can be obtained as:

$$s(d,l) = \begin{cases} 1, & (d=l) \wedge (l \leq \gamma) \\ \sum_{n_v=0}^{\min(L_v^{\max}-l, L_v)} \Pr(n_v), & (d=l) \wedge (l > \gamma) \\ \Pr(n_v = L_v^{\max} - l), & (d > l) \wedge (l \geq \gamma) \\ 0, & \text{otherwise} \end{cases}, \quad (14)$$

where $\gamma = L_v^{\max} - L_v$.

3.2. System performance at the voice call level

Due to the accepted assumptions, at the voice call level the downlink CDMA channel can be modeled as a $M/M/m/m$ queueing system, where $m = L_v^{\max}$. Applying this model, the probability that there are L_v voice connections served in the system is given by [18]:

$$\Pr(L_v) = \frac{(\lambda_v / \mu_v)^{L_v}}{L_v!} \frac{1}{\sum_{i=0}^{L_v^{\max}} \frac{(\lambda_v / \mu_v)^i}{i!}}, \quad (15)$$

and the blocking probability of voice calls $PB_v(\rho_v)$ is given by Erlang's loss formula, i.e., $PB_v(\rho_v) = \Pr(L_v = L_v^{\max})$, where $\rho_v = \lambda_v / \mu_v$. On the other hand, the data packet throughput, the data packet blocking probability, and the average data packet message delay which have been determined by conditioning on L_v at the data packet level have to be obtained now by taking the distribution of L_v into

account. Therefore, the basic performance measures at the voice call level can be expressed as follows:

$$\hat{\lambda}_d(\rho_v, \lambda_d) = \sum_{L_v=0}^{L_v^{\max}} \hat{\lambda}_{d|L_v}(\lambda_d) \Pr(L_v), \quad (16)$$

$$PB_d(\rho_v, \lambda_d) = \sum_{L_v=0}^{L_v^{\max}} PB_{d|L_v}(\lambda_d) \Pr(L_v),$$

and finally

$$T_d(\rho_v, \lambda_d) = \sum_{L_v=0}^{L_v^{\max}} T_{d|L_v}(\lambda_d) \Pr(L_v),$$

where $\hat{\lambda}_{d|L_v}(\lambda_d)$, $PB_{d|L_v}(\lambda_d)$, and $T_{d|L_v}(\lambda_d)$ are determined by formulas (4), (5), and (6), respectively. One can notice that system performance evaluation at the voice call level seems to be a simple task and therefore only system performance at the data packet level is farther considered.

4. Numerical example

In order to illustrate the presented approximate performance analysis and to verify its validity, an integrated wireless voice and data CDMA system described by the parameters given in Table 1 is considered.

Table 1
System parameters

Parameter	Symbol	Quantity	Unit
Processing gain	G	16	
Voice and data packet length	n	255	[bit]
Maximum number of correctable bit errors per data packet	t_d	18	
Maximum number of correctable bit errors per voice packet	t_v	12	
Slot duration	h_s	20	[ms]
Average voice call duration	$1/\mu_v$	7500	[slot]
Maximum number of voice connections	L_v^{\max}	5	
Transition probability of a voice source from ON- to OFF-state	P_{on-off}	1/17	
Transition probability of a voice source from OFF- to ON-state	P_{off-on}	1/22	
Data packet arrival rate in a single queue input	λ_d	0.1–0.9	[pack./slot]
Number of MTs accepting data	k	10	
Maximum length of a queue of data packets	L_q^{\max}	4	
Number of voice connections served by the system	L_v	1, 3, 5	

It is assumed that the processing gain is relatively low ($G = 16$) and only five signals can be transmitted simultaneously (this is explained below). For these values the approximate BER analysis is accurate enough [13], but on the other hand relatively strong error-correction capabilities of coding is required (e.g., BCH code can be used with $n = 255$ and with $t_v = 12$ and $t_d = 18$, for voice and data packets, respectively) to satisfy different quality of service for both traffic types. Namely, it is assumed that the system has to ensure the probability of receiving an uncorrectable voice (data) packet to be less than 10^{-2} (10^{-5}). One can find that $P_{p,v}(5) = 6.3 \cdot 10^{-3}$ and $P_{p,d}(5) = 8.2 \cdot 10^{-6}$ provided that formula (1) is used. Since $P_{p,v}(6) > 10^{-2}$, therefore in the sequel it is assumed that at most five simultaneous transmissions are allowed on the downlink (i.e., $L_v^{\max} = 5$). For $k_s = 5$ the average BER after decoding for voice (data) traffic calculated using (3) is equal to $P_{e,v}(5) = 3.3 \cdot 10^{-4}$ ($P_{e,d}(5) = 6.2 \cdot 10^{-7}$). Queuing model validity is evaluated by simulation. The BS is simulated as a system of k queues with slotted service time and the single common scheduler, which enables/disables the use of CDMA codes for transmission of data packets in each time slot. The method of independent replications is used [19]. A 95% confidence interval is generated by applying standard arguments based on the t -distribution.

A systematic investigation of the accuracy of the analytical approach is presented in Figs. 1, 2, and 3. More precisely, numerical calculations are compared with the simulation results for different numbers of voice connections, i.e., $L_v = 1, 3$, and 5. In all calculations the maximum queue length is chosen to be $L_q^{\max} = 4$. The total system load of data packet traffic $k\lambda_d$ changes with respect to $\lambda_d \in [0.1, 0.9]$. High system load is assumed, since the number k of destination MTs accepting data packets is chosen to be 10. An excellent agreement between numerical and simulation results is obtained for the average data throughput (Fig. 1). A different situation occurs when the average data packet delay is analyzed. Namely, Fig. 2 shows that numerical calculations underestimate simulation results for the assumed range of λ_d . It can also be noticed that the higher the system load, the better the approximation

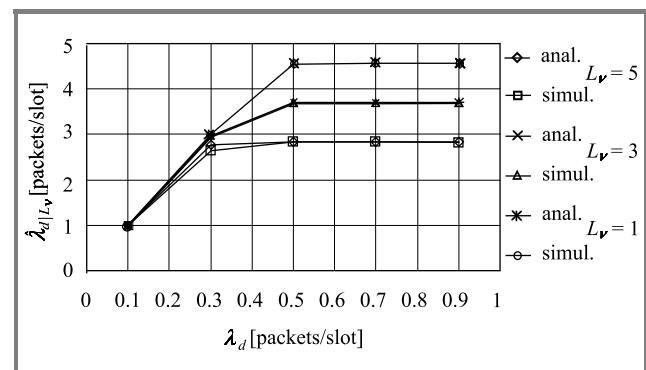


Fig. 1. Average data packet throughput $\hat{\lambda}_{d|L_v}$ versus system load λ_d ($L_q^{\max} = 8$, $L_v = 1, 3, 5$).

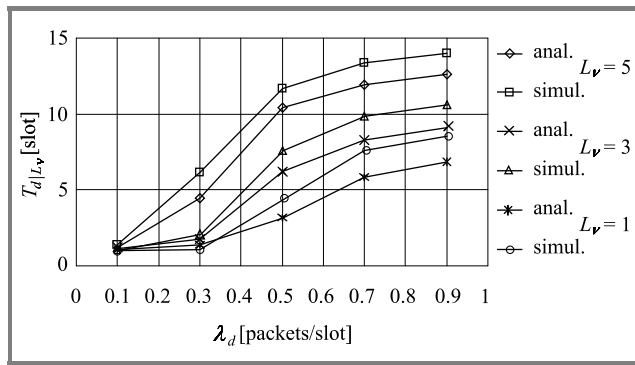


Fig. 2. Average data packet delay $T_{d|L_v}$ versus system load λ_d ($L_q^{\max} = 8$, $L_v = 1, 3, 5$, $h_s = 20$ ms).

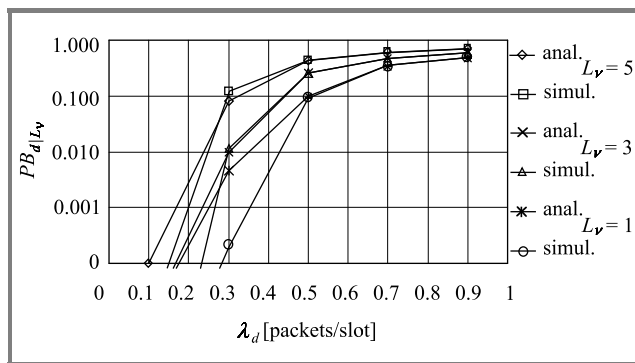


Fig. 3. Data packet blocking probability $PB_{d|L_v}$ versus system load λ_d ($L_v = 1, 3, 5$).

(when $\lambda_d = 0.9$ the approximation error is less than 20%). A similar observation can be done when the blocking probability of data packets is considered. However, one can notice that the approximation error is very low (it is less than 2%) only when the average packet input rate is higher than the output packet service rate, i.e., the total system load $k\lambda_d$ is such that $k\lambda_d > [L_v^{\max} - \sum_{n_v=1}^{L_v} n_v \Pr(n_v)]$, and the probability that queues are not empty is very low. This occurs for $\lambda_d > 0.46, 0.37, 0.28$ and for $L_v = 1, 2, 5$, respectively.

5. Conclusions

An approximate performance analysis of slotted downlink channel in a wireless CDMA system transmitting voice and data packets has been presented. Unfortunately, analyzed queues at the BS are statistically dependent, since a common scheduler allows them to use their allocated CDMA codes in subsequent time slots. The Markov analysis requires a system of linear equations to be solved. In practice however, with the accepted system model, the problem is computationally intractable, because the number of equations is unacceptably large. In order to make the analysis computationally tractable the restricted occupancy urn

model has been used to estimate the probability distribution of data packets in the analyzed system. Numerical calculations, compared with simulation results, show excellent agreement for the average system throughput and the blocking probability of data packets for higher system loads. On the other hand, when the average data packet delay is considered, analytical results underestimate simulation and therefore only approximate system performance evaluation is possible.

It has also been shown that performance results strongly depend on the number of simultaneous voice connections L_v in the system, even when silence periods of voice sources are used for data transmission. These results can be averaged, using probability distribution of the number of simultaneous voice connections. Therefore, system performance as a function of both data and voice traffic load can easily be evaluated.

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Jacek Świdorski was born in Gdańsk, Poland, in 1951. He received the M.S. and Ph.D. degrees in electronic engineering from the Technical University of Gdańsk in 1974 and 1982, respectively. He is now with the Institute of Power Engineering, Gdańsk, as a Research Staff Member. His main current interest is in the area

of computer communication systems including modeling, analysis, and performance evaluation.

e-mail: j.swiderski@ien.gda.pl

Institute of Power Engineering, Div. Gdańsk

Mikołaja Reja st 27

80-870 Gdańsk, Poland