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PAPER

Performance Analysis of an Integrated Voice/Data Wireless Network with Voice Buffer*

Haw-Yun SHIN†, **Jean-Lien C. WU**†, *Nonmembers, and* **Wei-Yeh CHEN**†, *Student Member*

SUMMARY This paper investigates the performance of an integrated voice/data wireless mobile network where a finite buffer is provided for voice calls since they can endure a tolerable time, or the reneging time, for service. Based on a given humanistic reneging time, we analyze the voice traffic blocking probability. The probability distribution of receiving service within the reneging time is obtained for each buffered voice call and based on this result, an appropriate amount of voice buffer is obtained. To alleviate the impact on data blocking probability caused by the voice buffer and to enhance the efficiency of data service, a dynamic multi-channel allocation scheme with channel de-allocation and guard channels is proposed for data traffic. Compared with the conventional method where the system adopts a single-channel allocation scheme without guard channel for data users, the proposed scheme shows significant improvement in data blocking probability, throughput and the mean service time. Furthermore, a system with an appropriate size of buffer for voice traffic can receive good improvement in voice blocking probability.
key words: *dynamic multi-channel allocation, channel de-allocation, reneging time, guard channel*

1. Introduction

The success of the Internet has brought a growing demand for data communication. Nowadays wireless mobile networks are expected to support wideband data service in addition to voice service. The bandwidth of wireless links is inherently limited and is generally much less than that in its wireline counterpart, efficient channel utilization is thus becoming increasingly important. Resource allocation schemes and scheduling policies are generally employed to achieve these goals [1], [2].

Mobile networks such as GPRS and UMTS nowadays provide both circuit-switched and packet-switched services [3], [4]. In general, real-time applications are best provided via circuit-switched service, while most data applications will be more efficiently provided by packet-switched data service. Both GPRS and UMTS allow multi-channel to be allocated to a user served in either circuit-switched mode or packet-switched mode to fulfill its QoS requirement. For instance, channel allocation in GPRS is flexible where one to eight channels can be allocated to a user or one channel can be shared by several users. Moreover, in UTRA-TDD, according to the specification [5], 240 resource units (15 timeslots in a frame and 16 different code sequences in each timeslots) can be dynamically distributed among users. Cimini et al. [6] presented the call blocking performance of a distributed algorithm based on a novel Erlang-B approximation for dynamic channel allocation (DCA) in microcells. Jordan and Khan [7] demonstrated the performance bound on dynamic channel allocation in cellular systems under uniform load condition. Lin et al. [8], [9] analyzed the performance of DCA in GPRS and Kriengchaiyapruk et al. [10]–[12] provided some practicable algorithms and analyses for DCA in UTRA with different QoS requirement. Although the multi-channel scheme can satisfy the bandwidth requirement of different traffic types, to alleviate the blocking probability at heavy load, users hogging with radio resources should be forced to release some channels for new requests, i.e., channel de-allocation. By adopting DCA with channel de-allocation, both the blocking probability and the wastage probability of radio resource can be very low. However, none of the pervious studies did theoretical analysis of DCA with channel de-allocation scheme.

For integrated voice/data wireless mobile networks, voice traffic encounters losses and data traffic suffers long delays should networks have insufficient bandwidth. Thus, the design of effective and dynamic bandwidth allocation to satisfy different service demands is important in wireless networks. Several bandwidth allocation schemes [8], [9], [13]–[16], [18] have been proposed to improve system performance in integrated voice/data wireless mobile networks. Generally, these schemes are based on the movable-boundary concept with or without reservation.

We propose an analytical model to study the voice and data traffic blocking probability for an integrated voice/data wireless mobile network with voice buffer. Voice users usually do not renege within a tolerable time, the reneging time. Buffers can be provided for voice calls because the transmission times of data packets are typically short. Some related studies used voice buffers [8], [16] and considered the performance of voice traffic without taking the reneging time into consideration. Voice calls in the front of a queue would mostly be served before the reneging time but those in the rear are not so lucky. To ensure that all queued voice

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†The authors are with the Department of Electronic Engineering, National Taiwan University of Science and Technology, Taipei, Taiwan, R.O.C.
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calls can be served before the reneging time, a proper amount of buffer size must be devised. In this paper, we calculate the probability that a tagged voice call buffered in queue can be served before reneging time. As voice load increases, providing buffers to voice may cause the great part of the channels being occupied by voice calls and the data blocking probability would increase because of the long voice call holding time. To compensate the increased blocking probability of data traffic, guard channels are provided to data [9], [16] and dynamic multi-channel allocation scheme with channel de-allocation are used. Numerical results show that the system with voice buffer can indeed decrease voice traffic blocking probability. The multi-channel scheme with possible de-allocation and guard channels for data service can enhance data traffic performance in terms of the mean service time, throughput and blocking probability. The queuing behavior of voice calls and the efficiency of data traffic are the focuses of the analysis, where handoff [17] traffic is considered as a part of voice traffic. The proposed analytical model is based on the fixed channel capacity assumption, mobile networks such as UMTS using CDMA technology are not included because the channel capacity is limited by interference power in CDMA and the capacity changes dynamically.

The paper is organized as follows. Section 2 presents the system description. The performance analysis of the proposed model and the reneging time distribution are given in Sect. 3. Section 4 provides the performance evaluation in numerical results. The benefit the system gains in giving buffers to voice traffic and adopting multi-channel scheme for data users are illustrated. Finally, Sect. 5 concludes this study.

2. The System Description

We propose to provide buffer to voice traffic to decrease the voice blocking probability, meanwhile, to alleviate the impact on data blocking probability caused by voice buffer and to enhance the efficiency of data service, we propose the dynamic multi-channel allocation scheme with channel de-allocation and guard channels for data traffic. Assume that all cells in the wireless system are homogenous. With respect to each cell, the channels are allocated as illustrated in Fig. 1. There are totally C channels in a cell where C_g channels are dedicated to data and $(C - C_g)$ channels are contended by voice and data users with no partiality, each data user can be allocated at most C_{max} free channels. Voice traffic is served in circuit-switched mode and data traffic is served in packet-switched mode. To alleviate the voice traffic blocking probability caused by data traffic, B_v buffers are provided to voice traffic. Upon a new data or voice user arrival, if there are available channels, the new arrival would be accepted immediately, otherwise, some of the data users occupying more than one channel would be forced to release one channel (channel de-allocation) for the new arrival. If none of the data users uses more than one channel, a new voice arrival would be placed in queue, while a new data user would be obstructed by the system. A queued voice call does not renege to wait and has higher priority than data requests. For the channel holding time of data users is relatively short, the queued voice calls have a very high probability of receiving service before the reneging time.

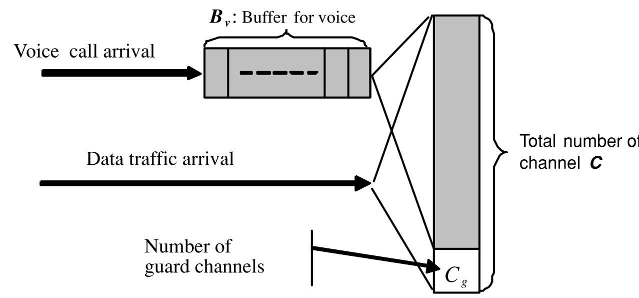

Fig. 1 The channel allocation model.

Assume there are i ($i \leq B_v + C - C_g$) voice users and j ($j \leq C$) data users in the cell simultaneously. To determine if a new arrival is to be accepted, obstructed or buffered by the system, four functions are used and described in the following,

$FC(i, j)$ denotes the number of free channels that can be allocated to the new voice calls, and can be expressed as

$$FC(i, j) = C - i - Min\{Max\{j * C_{max}, C_g\}, C - i\}. \quad (1)$$

$DC(i, j)$ denotes the number of channels that can be de-allocated from data users in service, and can be expressed as

$$DC(i, j) = C - Max\{C_g, j\} - Min\{C - Max\{C_g, j\}, i\}. \quad (2)$$

$BS(i, j)$ denotes the number of empty buffers in the system, and can be expressed as

$$BS(i, j) = B_v + C - Max\{i + Max\{j, C_g\}, C\}. \quad (3)$$

$SC(i, j)$ denotes the total number of channels to be shared by all data users in service, and can be expressed as

$$SC(i, j) = C - Min\{C - C_g, i\}. \quad (4)$$

Figures 2 and 3 depict the flow of the channel allocation process upon a service completion and upon a service request, respectively. The probability distribution that a queued voice call has to wait for service within a tolerable time, i.e., the reneging time distribution, is a significant metrics for the design of buffer size and it should be so designed that each queued voice call can obtain service before the reneging time expires. Assume that voice arrivals form a Poisson process with mean rate λ_v, the holding time of these calls are exponentially distributed with mean $1/\mu_v$. The arrivals of data traffic is also assumed to be a Poisson process with mean rate λ_d and the service time is exponentially distributed with mean $1/\mu_d$ in the single-channel scheme. In the multi-channel scheme, at most C_{max} can be allocated to a data user, the mean service time of a data user can be shortened to an extent of $1/(C_{max} \cdot \mu_d)$.

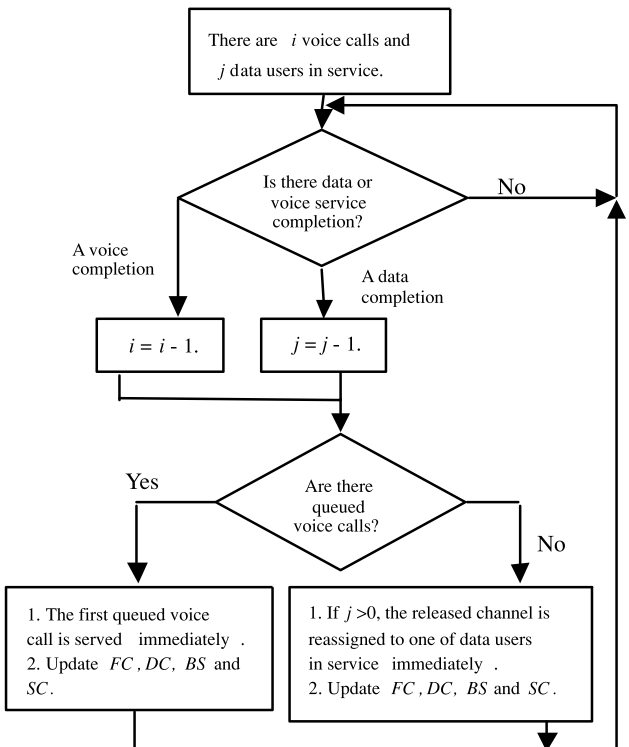

Fig. 2 Channel allocation process upon a service completion.

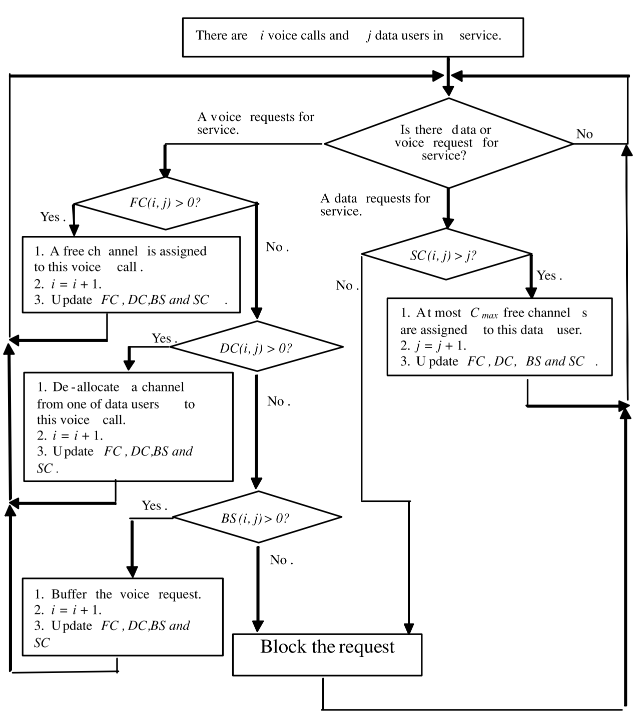

Fig. 3 Channel allocation process upon a new request arrival.

3. Performance Analysis

3.1 The Analytical Model and Performance

In this section, the analytical model of the proposed strategies, guard channels provided to data and dynamic multi-channel allocation scheme with channel deallocation, is depicted and performance metrics such as data traffic blocking probability, voice blocking probability, and mean queuing time of voice are obtained analytically. Note that when $C_{max} = 1$ and $C_g = 0$, the model degenerates to the single-channel scheme without guard channel for data. Assume there are i voice users and j data users in the cell simultaneously. The state space, S, can be denoted as

$$S = \{(i,j)|(0 \le j < C_g, 0 \le i \le C - C_g + B_v) \\ \text{or } (C_g \le j \le C, C_g \le i + j \le C + B_v)\}. \quad (5)$$

Figure 4 depicts the state transition diagram where the transition rate of the Markov process are explained in the following:

1) $\text{Ł}_d(i,j)$ is the transition rate from state (i,j) to $(i, j+1)$. A data user is accepted for service only when either there are guard channels available or there is at least one channel available in the system. Therefore, $\text{Ł}_d(i,j)$ can be written as

$$\text{Ł}_d(i,j) = \begin{cases} \lambda_d & \text{if } j < C_g \text{ or} \\ & (C_g \le j < C, i + j < C), \\ 0 & \text{otherwise.} \end{cases} \quad (6)$$

2) $\text{Ł}_v(i,j)$is the transition rate from state (i,j) to $(i+1, j)$. A voice user is allowed into the system as long as there is at least one idle buffer. Therefore, $\text{Ł}_v(i,j)$ can be written as

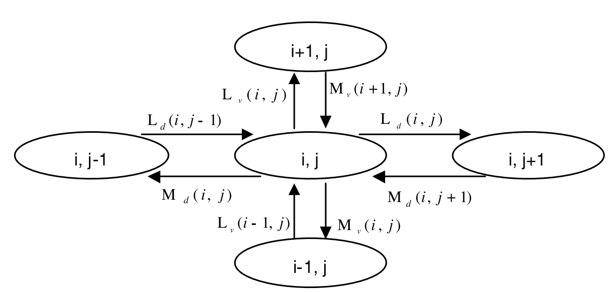

Fig. 4 The state transition diagram. State (i,j) represents that there are i voice users and j data users in the system.

$$Ł_v(i,j)=\begin{cases}\lambda_v & \text{if } (j<C_g, i<C+B_v-C_g) \\ & \text{or } (C_g\le j\le C, i+j<C+B_v), \\ 0 & \text{otherwise.}\end{cases} \quad (7)$$

3) $M_d(i,j)$ is the transition rate from state (i,j) to $(i,j-1)$, i.e. one of the j data users completes transmission, while i voice users are in the system. In the single-channel scheme, only one channel can be allocated to a data user, $M_d(i,j)$ can be represented as

$$M_d(i,j)=\begin{cases}j\mu_d & \text{if } (j<C_g, i\le C+B_v-C_g) \\ & \text{or } (C_g\le j\le C, i+j\le C+B_v), \\ 0 & \text{otherwise.}\end{cases} \quad (8)$$

When the multi-channel scheme is adopted, each data user can be at most allocated a number of C_{max} channels if there are at least channels available, otherwise, all $(C-i)$ available channels are distributed among data users. All guard channels can be used for data service if all $(C-C_g)$ channels are occupied by voice. Therefore, can be written as

$$M_d(i,j)=\begin{cases}Min(C-i, jC_{max})\mu_d & \\ \quad \text{if } (C_g\le j\le C, i+j\le C)\text{or} & \\ \quad (j<C_g, i\le C-C_g), & \\ j\mu_d \quad \text{if } (C_g\le j\le C, & \\ \quad C\le i+j\le C+B_v), & \\ C_g\mu_d \quad \text{if } (j<C_g, & \\ \quad C-C_g\le i\le C+B_v-C_g), & \\ 0 \quad \text{otherwise.} & \end{cases} \quad (9)$$

4) $M_v(i,j)$ is the transition rate from state (i,j) to $(i-1,j)$, i.e. one voice user completes service while j data users are in the system. Bearing in mind that C_g guard channels are reserved for data users, $M_v(i,j)$ can be written as

$$M_d(i,j)=\begin{cases}(C-j)\mu_v & \\ \quad \text{if } (C_g\le j\le C, & \\ \quad C\le i+j\le C+B_v), & \\ (C-C_g)\mu_v & \\ \quad \text{if } (j<C_g, & \\ \quad C-C_g\le i\le C+B_v-C_g), & \\ i\mu_v \quad \text{if } (C_g\le j\le C, i+j\le C)\text{or} & \\ \quad (j<C_g, i\le C-C_g), & \\ 0 \quad \text{otherwise.} & \end{cases} \quad (10)$$

Therefore, let $P(i,j)$ be the probability of state (i,j), the balance equation for the Markov process is expressed as

$$\begin{aligned}&[M_d(i,j)+M_v(i,j)+Ł_d(i,j)+Ł_v(i,j)]P(i,j)\\ &\quad =P(i,j-1)Ł_d(i,j-1)+P(i-1,j)Ł_v(i-1,j)\\ &\quad\quad +P(i,j+1)M_d(i,j+1)+P(i+1,j)M_v(i+1,j).\end{aligned} \quad (11)$$

According to Eqs. (5) to (11), the state-transition matrix can be formed. The steady-state probability $P(i,j)$ can be obtained via matrix inversion subject to

$$\sum_{i=0}^{C-C_g+B_v}\sum_{j=0}^{C}P(i,j)=1$$

Having obtained $P(i,j)$, we can make a number of significant estimations. The data traffic blocking probability, denoted by P_B^d, can be expressed as

$$P_B^d=\sum_{C_g\le j\le C, C\le i+j\le C+B_v}P(i,j) \quad (12)$$

The mean service time of a data user, denoted by W_d, can be obtained by using Little's formula,

$$W_d=\frac{\sum_{i=0}^{C-C_g+B_v}\sum_{j=0}^{C}j\cdot P(i,j)}{\lambda_d(1-P_B^d)} \quad (13)$$

Where the numerator represents the mean number of data users and the denominator is the throughput of data users.

Another significant metrics P_B^v, the voice traffic blocking probability, can be expressed as

$$P_B^v=\sum_{j<C_g, i=C-C_g+B_v}P(i,j)+\sum_{C_g\le j\le C, i+j=C+B_v}P(i,j) \quad (14)$$

Using Little's formula, the mean queuing time of voice users, denoted by W_v^Q can be written as

$$W_v^Q=\frac{\sum_{j<C_g, C-C_g<i\le C-C_g+B_v}P(i,j)[i-(C-C_g)]}{\lambda_v(1-P_B^v)} +\frac{\sum_{C_g\le j\le C, C<i+j\le C+B_v}P(i,j)(i+j-C)}{\lambda_v(1-P_B^v)}, \quad (15)$$

where the numerator represents the mean voice queue length and the denominator is the voice call throughput.

3.2 The Reneging Time Distribution

Users at the front of a queue can obtain service in a short time while the ones in the rear will wait for a long time. Assume T_{reg} is the reneging time of a queued voice user. The reneging time distribution, the probability that the queued voice user acquires a channel for service before waiting for a time of t, is an important metrics collocated with W_v^Q, the mean voice queuing time. A proper amount of buffer for voice users is designed according to the reneging time distribution, so that each queued voice user can get a channel before the reneging time. To observe at a time point that a tagged voice user arrives at a steady-state system while no channel can be provided and the buffer is empty, the tagged voice user would be queued in the first place of the buffer. The random variable T_q represents the time spent waiting for service in queue. $W_{reg,0}(t)$, the probability density function (**PDF**) of the reneging time distribution of the first voice user in queue, can be defined as
$W_{reg,0}(t)=Prob\{Tq=t\}$

$= Prob\{$having waited for time t, a tagged voice obtains a channel for service | no available channel and empty queue upon the arrival of the tagged voice user$\}$

$$=\frac{1}{P_0^Q}\{\textstyle\sum_{C_g<j\le C, i+j=C} P(i,j)[(i\mu_v+j\mu_d)e^{-(i\mu_v+j\mu_d)t}] + \sum_{j\le C_g, i=C-C_g} P(i,j)(i\mu_v e^{-i\mu_v t})\} \quad (16)$$

P_0^Q represents the probability that there is no channel and the queue is empty upon the arrival of the tagged voice user. Since the number of data users can be more than the number of guard channels, i.e., $j > C_g$, or, $j \le C_g$, P_0^Q can be expressed as

$$P_0^Q = \sum_{j\le C_g, i=C-C_g} P(i,j) + \sum_{C_g<j\le C, i+j=C} P(i,j) \quad (17)$$

A channel can be allocated to a queued voice user when either a data or voice user completes its service. If the number of data users in service is less than the number of guard channels, a free channel released by a data user can not be allocated to a queued voice user. In addition, when there are i voice and j data users in service at the same time, with voice holding time exponentially distributed with a mean of $1/\mu_v$ and the data service time exponentially distributed with mean of $1/\mu_d$, therefore, it can be proved that the inter-departure time distribution is also exponentially distributed with mean of $1/(i\mu_v + j\mu_d)$. Hence, the probability distribution that the tagged voice user spent a time t to wait for service can be represented by

$$\begin{cases} P(i,j)[(i\mu_v+j\mu_d)e^{-(i\mu_v+j\mu_d)t}] \\ \qquad\qquad \text{if } C_g < j \le C, i+j=C \\ P(i,j)(i\mu_v e^{-i\mu_v t}) \\ \qquad\qquad \text{if } C_g \ge j, i = C - C_g \end{cases} \quad (18)$$

Consequently, by summing up all possible conditions, the **PDF** of the reneging time distribution, $W_{reg,0}(t)$, can be written as Eq. (16). It is easy to prove that $\int_0^\infty W_{reg}(t)dt = 1$.

Consider a general case when a tagged voice user arrives, n voice users are queued in the buffer. The **PDF** of the reneging time distribution of the tagged voice user, denoted by $W_{reg,n}(t)$, is as follows

$W_{reg,n}(t) = Prob\{T_q = t\}$

$=Prob\{$having waited for time t, a tagged voice obtains a channel for service | n queued voice users upon the arrival of the tagged voice user, $n < B_v$ $\}$.

To solve the problem, Laplace transform is used. The following notations are defined,

$W_{reg,n}(s)$: Laplace transform of $W_{reg,n}(t)$.

$D_{i,j}(s)$: Laplace transform of the probability distribution of the queued voice user waiting for a channel released from one of the i voice or j data users.

$P_{i,j}^v$: the probability that the next service complete is voice while i voice and j data users are in service.

$P_{i,j}^d$: the probability that the next service complete is data while i voice and j data users are in service.

P_n^Q: the probability of no channel and $n (n < B_v)$ queued voice user in the system upon a tagged voice arrival.

When i voice users and j data users are in service, the probability distribution that the time between the arrival of a tagged voice user and the next service complete equal to t can be represented by $(i\mu_v + j\mu_d)e^{-(i\mu_v+j\mu_d)t}$, its Laplace transform is $s/(s + i\mu_v + j\mu_d)$. However, if $j \le C_g$, the tagged voice user can not obtain service until one voice service completion, therefore, $D_{i,j}(s)$ can be written as

$$D_{i,j}(s) = \begin{cases} s/(s+i\mu_v+j\mu_d) & \text{if } j > C_g \\ s/(s+i\mu_v) & \text{otherwise.} \end{cases} \quad (19)$$

Moreover, it is easy to prove that

$$P_{i,j}^v = \begin{cases} i\mu_v/(i\mu_v+j\mu_d) & \text{if } j > C_g \\ 1 & \text{otherwise.} \end{cases}$$
$$P_{i,j}^d = \begin{cases} j\mu_d/(i\mu_v+j\mu_d) & \text{if } j > C_g \\ 0 & \text{otherwise.} \end{cases} \quad (20)$$

P_n^Q is the probability of n queued voice users in the system upon a tagged voice arrival, as explained in Eq. (17), can be written as

$$P_n^Q = \sum_{j\le C_g, i=C-C_g+n} P(i,j) + \sum_{C_g<j\le C, i+j=C+n} P(i,j) \quad (21)$$

When a tagged voice user arrives at the system with n voice users in queue, the tagged voice user has to wait for $(n+1)$ service completions. If $j \le C_g$, the tagged voice user has to wait for $(n+1)$ voice users complete their services, therefore, the probability distribution that the tagged voice user waits for time t for service is $D_{c-c_g,0}(s)^{n+1}$. If $C_g < j \le C$, the probability distribution that the tagged voice user waits for time t for service depends on the kind of service completion. Assume when a tagged voice user arrives, there are i voice and j data users in the system and n voice users in queue, hence, $(C - j)$ voice users are in service. During the tagged voice user waiting for service, the probability distribution of the time the first available channel released from a voice or data user is $D_{C-j,j}(s)$, however, the probability distribution of the time the next channel released would depend on the kind of the previous service completion. With a probability of $(C-j)\mu_v/[(C-j)\mu_v + j\mu_d]$ a channel is released by a voice user, the number of voice and data users in service does not change since the first queued voice starts its service immediately, therefore, the probability distribution of the time the next channel release is $D_{C-j,j}(s)$. On the other hand, with a probability of $j\mu_d/[(C-j)\mu_v + j\mu_d]$ a channel is released by a data user, the number of voice and data users being served would become $(C-j+1)$ and $(j-1)$, respectively, therefore, the probability distribution of the time the next channel release is $D_{C-j+1,j-1}(s)$. Similarly, the probability distribution of the time of the third available channel would be $D_{C-j,.j}(s)$, $D_{C-j+1,.j-1}(s)$, or $D_{C-j+2,j-2}(s)$ implies that previous two service completions are all voice users, one data and one voice user, or all data users, respectively. Therefore, the probability distribution of the time the tagged voice user would wait for service, $(n+1)$ users completed services, can be obtained by integrating the probability from the time the tagged voice user arrived to the time the $(n+1)$th channel available.

A difference equation is used to simplify the expression of the probability distribution of the time the $(n+1)$ channel available. It is expressed as $D_{C-j,j}(s)F_{i,j}(n,s)$, where $D_{C-j,j}(s)$ is the probability distribution of the time the first available channel occurs, and $F_{i,j}(n,s)$ is the probability distribution of the time the following nth available channel occurs. $F_{i,j}(n,s)$ can be obtained by integrating the probability distribution of the time the next user completed service after the $(n-1)$th available channel occurring, and which relies on the kind of user providing the $(n-1)$th available channel. Therefore, can be expressed in a form of difference equation as follows

$$\begin{aligned} F_{i,j}(n,s) &= P^v_{i,j}D_{i,j}(s)F_{i,j}(n-1,s) \\ &\quad + P^d_{i,j}D_{i+1,j-1}(s)F_{i+1,j-1}(n-1,s) \\ F_{i,j}(0,s) &= 1 \end{aligned} \tag{22}$$

Finally, $W_{reg,n}(s)$ can be represented as

$$W_{reg,n}(s)=\frac{1}{P^Q_n}\left[\sum_{C_g<j\le C, i+j=C+n} P(i,j)D_{C-j,j}(s)F_{C-j,j}(n,s) + \sum_{j\le C_g, i=C-C_g+n} P(i,j)D_{C-j,0}(s)^{n+1}\right] \tag{23}$$

Having obtained the **PDF** of the reneging time distribution, we can obtain $W_{reg,n}(t)$ can be obtained using the inverse transform. The cumulative probability distribution of reneging time, $prob\{T_q \le T_{reg}\}$, can be obtained by $\int_0^{T_{reg}} W_{reg,n}(t)dt$. Finally, the important metrics, P_{reg}, the probability that voice traffic can be served before waiting for a reneging time T_{reg}, can be obtained as follows

$$P_{reg} = (1-P^v_B)\left[1-\sum_{i=0}^{B_v-1} P^Q_i\left(1-\int_0^{T_{reg}} W_{reg,i}(t)dt\right)\right] \tag{24}$$

4. Performance Evaluation

In this section, the performance of the proposed strategies, guard channels provided to data and dynamic multi-channel allocation scheme with channel de-allocation, is evaluated and compared with the single-channel scheme without guard channel for data. GPRS environment is adopted because it supports both circuit-switched and packet-switched services and it also supports channel de-allocation. In the numerical calculations, 3 carriers, i.e., a total of 24 channels in a cell is assumed. Among these 24 channels, 0 and 2 guard channels, respectively, are dedicated to data users. Various buffer sizes are considered in order to find an appropriate one for high voice service efficiency. The holding time of a voice user is exponentially distributed with a mean of 120 s, and the voice traffic load is 17.56 Erlang corresponding to a 3% blocking probability for 24 channels in a traditional GSM system. The C2 coding scheme corresponding to a transmission rate of 13.4 kbits/sec is used. The packet call size is exponentially distributed with a mean of 2×13.4 kbits, corresponding to a mean service time of 2 s. The data traffic load ranges from 0.5 to 7.5 packets/sec corresponding to light load to heavy load. When the multi-channel scheme is adopted, as described in the GPRS standard, one carrier ($C_{max}=8$) can be assigned to a data user simultaneously. Parameters used for numerical calculations are shown in Table 1.

Table 1 Parameters used for numerical calculations.

Environment considered	GPRS
Total number of channels	24 (3 carriers)
Mean call holding time	120 sec
Mean packet call size	26.8 kbits
Coding scheme	C2(13.4 kbits/sec)
Voice traffic load	17.56 Erlang
Data traffic load	0.5–7.5 packets/sec

4.1 Efficiency Improvement by Using Voice Buffers and Providing Guard Channels to Data

Figure 5 shows the effects of dedicating guard channel to data users and providing various buffer sizes to voice calls in the single-channel scheme. For the system adopting single-channel scheme without guard channel for data, Figs. 5(a) to 5(c) show the effects in terms of data blocking probability, voice blocking probability and mean queuing time of voice traffic, respectively, with various voice buffers. Figure 5(a) shows the increasing data blocking probability as the voice buffer increases. Figure 5(b) depicts the improvement in voice blocking probability when the system provides some voice buffers. However, providing buffers for voice may cause the voice queuing time to increase as shown in Fig. 5(c). Note that the results of $B_v=0$ (no voice buffer) are not shown in Fig. 5(c) since no queuing time occurs when $B_v=0$. To take a close look at how much the voice blocking probability is improved by providing buffer to voice calls, numerical values of Fig. 5(b) are

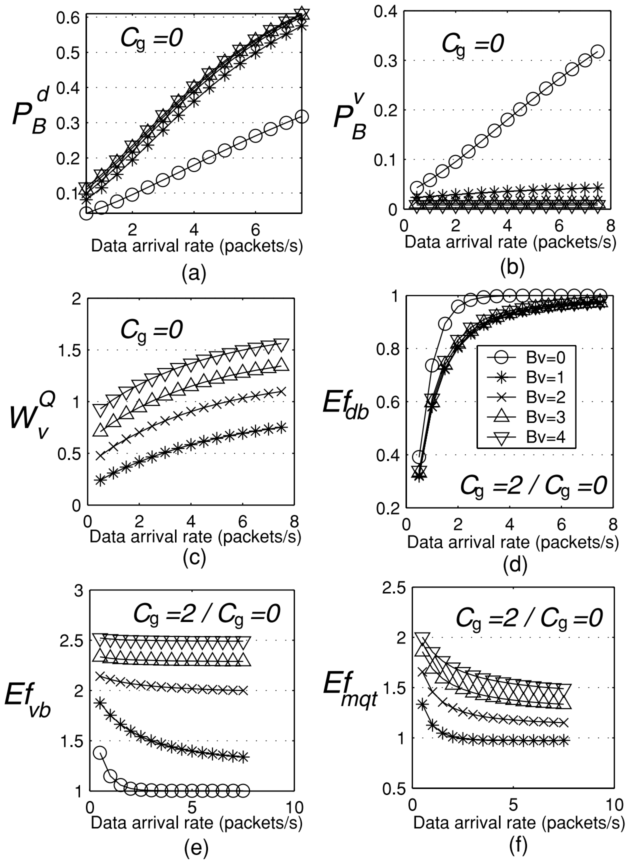

Fig. 5 (a), (b) and (c) show the performance of single-channel scheme without data guard channel. (d), (e), and (f) show the efficiency improvement when 2 data guard channel are used.

Table 2 Numerical values of voice blocking probability in Fig. 5(b).

λ_d	$B_v = 0$	$B_v = 1$	$B_v = 2$	$B_v = 3$	$B_v = 4$
1	0.0585	0.0254	0.0161	0.0113	0.0082
2	0.0955	0.0291	0.0165	0.0114	0.0082
3	0.1371	0.0326	0.0167	0.0115	0.0082
4	0.1800	0.0355	0.0170	0.0115	0.0082
5	0.2222	0.0380	0.0171	0.0116	0.0082
6	0.2624	0.04	0.0172	0.0116	0.0082
7	0.300	0.0417	0.0173	0.0116	0.0082

provided in Table 2. A 3% call blocking probability in the GSM system is acceptable. As shown in Table 2, the voice blocking probability is less than 3% as long as the system provides 2 voice buffers ($B_v = 2$).

Figure 5(d) shows the improvement of data blocking probability and Figs. 5(e) and 5(f) show the impact on the voice blocking probability and mean queuing time of voice traffic, respectively, when the system provides 2 guard channels to data. The following terms are used to represent the efficiency improvement by providing guard channel to data,

Ef_{db} = (Data blocking probability in single-channel scheme with 2 guard channels)/(that in single-channel scheme without guard channel).

Ef_{vb} = (Voice blocking probability in single-channel scheme with 2 guard channels)/(that in single-channel scheme without guard channel).

Ef_{mqt} = (Mean queuing time of voice call in single-channel scheme with 2 guard channels)/(that in single-channel scheme without guard channel).

Figure 5(d) shows that the data blocking probability can be improved by dedicating guard channels to data. However, as shown in Figs. 5(e) and 5(f), a system that dedicates guard channels to data users would increase the blocking probability and mean queuing time of voice traffic. As shown in Fig. 5(e), when $C_g = 2$, the voice blocking probability will increase about 2 and 2.3 times for when $B_v = 2$ and 3, respectively. From table 2, the voice blocking probability is about 3.46% (=0.0173 $*$ 2) when $C_g = 2$ and $B_v = 2$ and is about 2.67% (=0.0116 $*$ 2.3) when $C_g = 2$ and $B_v = 3$, hence, to keep the voice blocking probability below 3%, the system should provide 3 voice buffers when 2 guard channels are dedicated to data users.

4.2 Efficiency Improvement by Using Multi-Channel Scheme with Channel De-Allocation

When the multi-channel scheme is adopted, all free channels are shared by data users. Each data user can be allocated at most 8 channels (one carrier). With no guard channel and 2 guard channels provided to data users, respectively, Figs. 6 and 7 show the gains by adopting the multi-channel scheme. The following terms are used to represent the efficiency improvement by using the multi-channel scheme,

$Ef_{db}^{m/s}$= (Data blocking probability in multi-channel scheme)/(that in single-channel scheme without guard channel).

$Ef_{wd}^{m/s}$ = (Mean service time of data traffic in multi-channel scheme)/(that in single-channel scheme without guard channel).

$Ef_{vb}^{m/s}$= (Voice blocking probability in multi-channel scheme)/(that in single-channel scheme without guard channel).

$Ef_{mqt}^{m/s}$= (Mean queuing time of voice traffic in multi-channel scheme)/(that in single-channel scheme without guard channel).

$Ef_{th}^{m/s}$= (Data packet throughput in multi-channel scheme)/(that in single-channel scheme without guard channel).

With no guard channel dedicated to data users, Fig. 6 shows efficiency improvement versus data traffic for different number of voice buffers. In both schemes, data users will not be blocked at light data load and each data user is basically allocated just one channel at heavy data load, however, at medium data load, free channels are allocated to all data users in service so that the gain of multi-channel scheme in data blocking probability is obvious, which can be found in Fig. 6(a).

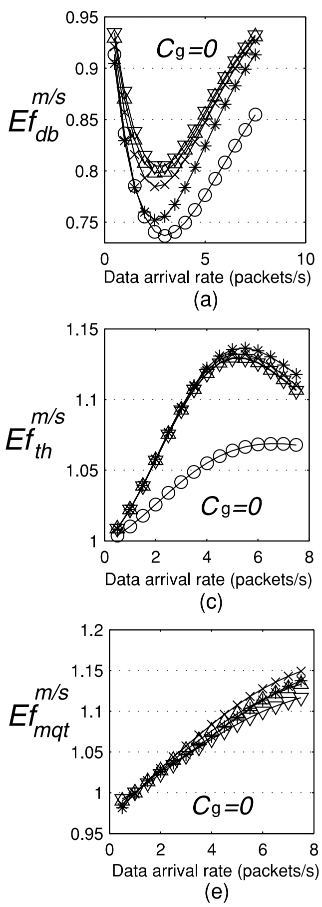

Fig. 6 Efficiency improvement of using multi-channel scheme with $B_v = 0 \sim 4$, $C_g = 0$.

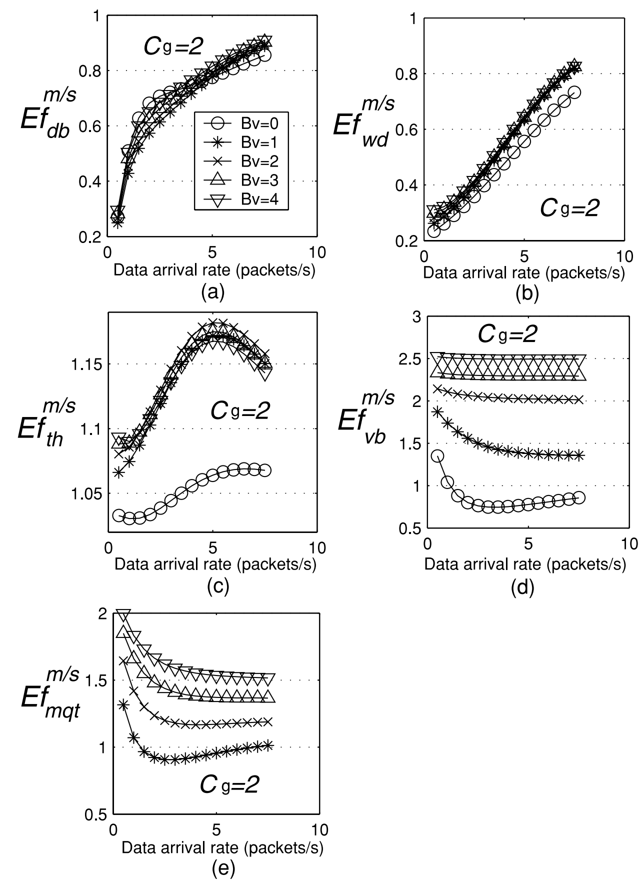

Fig. 7 Efficiency improvement of using multi-channel scheme with $B_v = 0 \sim 4$, $C_g = 2$.

As shown in Fig. 6(b), since more free channels can be shared by data users at light data load, the improvement of data service time is even more obvious. As shown in Fig. 6(c), data packet throughput can be increased since the data blocking probability is improved in the multi-channel scheme. $Ef_{vb}^{m/s}$ and $Ef_{mqt}^{m/s}$ shown in Figs. 6(d) and 6(e), depict the impact on the blocking probability and the mean queuing time of voice traffic, respectively, by using the multi-channel scheme. As shown in Fig. 6(d), there is little impact on voice blocking probability when data traffic increases except the case of $B_v = 0$. When $B_v = 0$, voice users contend the channels with data users with no partiality, data users has a much shorter service time when using the multi-channel scheme, therefore, the number of free channels in the system will increase. It is because the voice blocking probability can be improved when $B_v = 0$. Figure 6(e) shows that the multi-channel scheme also has little impact on mean queuing time of voice traffic when data traffic increases.

With 2 guard channels dedicated to data users, Fig. 7 shows efficiency improvement versus data traffic for different number of voice buffers. By dedicating guard channels to data, the efficiency improvement in both the blocking probability and throughput of data traffic are better and can be found by comparing Figs. 6(a) and 6(c) with Figs. 7(a) and 7(c). However, as shown in Figs. 7(d) and 7(e), a system that dedicates guard channels to data users would increase the blocking probability and mean queuing time of voice call. Moreover, since all free channels (include guard channels) can be shared by data user in service, $Ef_{wd}^{m/s}$ are about the same in despite of the number of guard channels provided to data users, which can be found by comparing Figs. 6(b) and 7(b).

4.3 The Effect of Reneging Time

The probability that a voice arrival can be served before waiting for a reneging time T_{reg}, denoted by P_{reg}, is given in Eq. (24). P_{reg} is important for the system to decide how many voice buffers should be provided. Figure 8 shows the cumulative probability distribution of the reneging time of voice users in queue where 17.56 Erlang voice traffic load and 3 packets/sec data arrival rate are assumed, the multi-channel scheme is adopted, and 4 voice buffers are provided. The probability that voice users in queue can be served before waiting for a reneging time is observed so that an appropriate

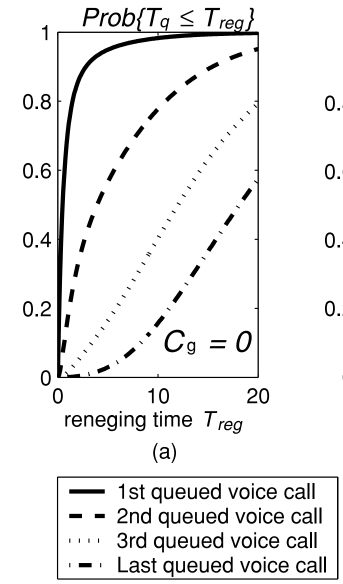

Fig. 8 Cumulative probability distribution of the reneging time of the queued voice users at various buffer positions. (a) $C_g = 0$. (b) $C_g = 2$.

Table 3 The probability that a voice user is queued at various buffer positions.

	P_0^Q	P_1^Q	P_2^Q	P_3^Q
$C_g = 0$	0.1802	0.0402	0.0183	0.0117
$C_g = 2$	0.1487	0.0477	0.0329	0.0259

amount of buffer can be provided for voice traffic to guarantee its QoS. Figures 8(a) and 8(b) show the cumulative probability distribution of the reneging time of the queued voice users in the first, second, third and the last positions. Although, the probability that the queued voice user in the third and the last positions can be served before T_{reg} is small, the probability that voice users will be buffered in the third or last positions is even smaller. The probability that voice users buffered in various positions, P_n^Q (discussed in previous section), is showed in Table 3.

Figure 9 depicts P_{reg} versus voice buffer in an increasing reneging time with a voice traffic load of 17.56 Erlang and a data arrival rate of 3 packets/sec. Multi-channel scheme is adopted by the system. 1 to 4 voice buffers are discussed here with 0 and 2 guard channels for data traffic, respectively. In Fig. 9(a), no guard channel for data traffic, if the reneging time is below 6 s, before waiting for a reneging time T_{reg}, with a probability of 0.96 the voice traffic receives a channel when the system provides 1 buffer. If the reneging time is 10 s, the voice traffic has a probability of 0.97 to receive a channel when 2 buffers are provided. The longer the reneging time is, the more the voice buffer should be provided, and the lower the blocking probability of data traffic. However, $1 - P_B^v$, is the upper bound of P_{reg} irrespective to the reneging time. In Fig. 9(b), 2

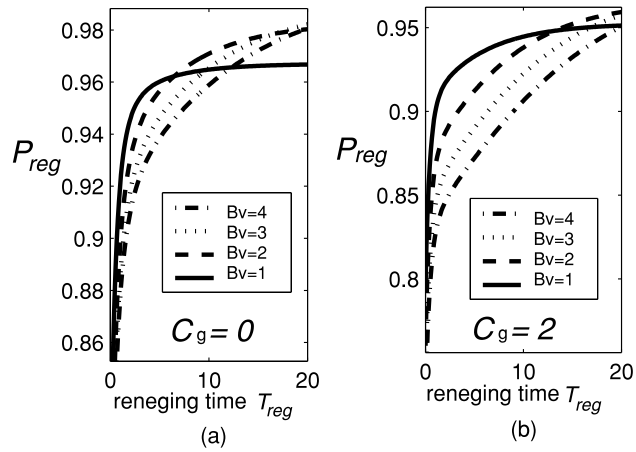

Fig. 9 P_{reg} (defined in Eq. (24)), the probability that voice traffic can be served before waiting for a reneging time T_{reg}, in various voice buffer size. (a) $C_g = 0$. (b) $C_g = 2$.

guard channels are provided to enhance the efficiency of data users. If $T_{reg} < 13$ sec, with a probability of 0.95 the voice traffic receives a channel when 1 buffer is provided.

5. Conclusion

This paper investigates the performance of an integrated voice/data wireless network with finite buffer for voice calls. The analysis of voice call blocking probability is provided based on a humanistic reneging time. In order to have the system to provide an appropriate amount of voice buffer, the probability distribution that the queued voice calls at various buffer positions to be served within the reneging time is obtained. To reduce the impact on the data blocking probability caused by furnishing buffers for voice calls, we employ both the dynamic multi-channel allocation scheme with channel de-allocation and guard channels for data traffic to enhance its service efficiency. Numerical results show that the dynamic multi-channel scheme with channel de-allocation and providing guard channels to data users, compared with the single-channel scheme without guard channel, can enhance data traffic performance significantly in terms of the packet blocking probability, packet throughput and mean service time. In addition, a system with an appropriate amount of voice buffer can receive significant improvement in voice blocking probability.

References

[1] S. Lu, V. Bharghavan, and R. Srikant, "Fair scheduling in wireless packetnetworks," IEEE/ACM Trans. Netw., vol.7, no.4, pp.473–489, Aug. 1999.

[2] X. Liu, E.K.P. Chong, and N.B. Shroff, "Transmission scheduling for efficient wireless utilization," Proc. IEEE INFOCOM 2001, vol.2, pp.776–785, April 2001.

[3] "Digital cellular telecommunications system (Phase 2+);

General packet radio service (GPRS); Service description: Stage 1(GSM 02.60 version 7.2.0 release 1999)," ETSI/TC, Tech. Rep. Rec. GSM 02.60, 1999.

[4] "Selection procedure for the choice of radio transmission technologies of the universal mobile telecommunication system UMTS (UMTS 30.03)," ETSL TR 101 112, April 1998.

[5] "TS 25.22x—Physical layer, general description (TDD), technical report," 3rd Generation Partnership Project, 3GPP, Oct. 1999.

[6] L.J. Cimini, G.J. Foschini, C.-L. I, and Z. Miljanic, "Call blocking performance of distributed algorithms for dynamic channel allocation in microcells," IEEE Trans. Commun., vol.42, no.8, pp.2600–2607, Aug. 1994.

[7] S. Jordan and A. Khan, "A performance bound on dynamic channel allocation in cellular system: Equal load," IEEE Trans. Veh. Technol., vol.43, no.2, pp.333–344, May 1994.

[8] P. Lin and Y.B. Lin, "Channel allocation for GPRS," IEEE Trans. Veh. Technol., vol.50, no.2, pp.375–387, March 2001.

[9] C. Lindemann and A. Thummler, "Performance analysis of the general packet radio service," Comput. Netw., vol.41, pp.1–17, 2003.

[10] T. Kriengchaiyapruk and I. Forkel, "Dynamic allocation of capacity in UTRA TDD system," Second International Conf. on 3G Mobile Communication Technologies, pp.262–266, March 2001.

[11] E.D. Fitkov-Norris and A. Khanifar, "Dynamic pricing in mobile communication systems," First International Conf. on 3G Mobile Communication Technologies, pp.1032–1036, March 2000.

[12] L. Ortigoza-Guerrero and A.H. Aghvami, "Capacity assessment for UTRA," Vehicular Technology Conf. 1999 IEEE 49th, vol.2, pp.1653–1657, May 1999.

[13] Y.R. Haung, Y.B. Lin, and J.M. Ho, "Performance analysis for voice/data integrated on a finite-buffer mobile," IEEE Trans. Veh. Technol., vol.49, no.2, pp.367–378, March 2000.

[14] H. Qi and R. Wyrwas, "Performance analysis of joint voice-data PRMA over random packet error channel," IEEE Trans. Veh. Technol., vol.45, no.2, pp.332–345, May 1996.

[15] J.E. Wieselthier and A. Ephremides, "Fixed- and movable-boundary channel-access schemes for integrated voice/data wireless networks," IEEE Trans. Commun., vol.43, no.1, pp.64–74, Jan. 1995.

[16] H.H. Liu, J.L.C. Wu, and W.C. Hsieh, "Delay analysis of integrated voice and data service for GPRS," IEEE Commun. Lett., vol.6, no.8, pp.319–321, Aug. 2002.

[17] Y.B. Lin and I. Chlamtac, Wireless and Mobile Network Architecture, pp.60–65, John Wiley & Sons, 2001.

[18] D. Calin and D. Zeghlache, "Analysis of a joint voice-data cellular network with queued handoffs and new calls," Proc. IEEE GLOBECOM 1997, vol.3, pp.1657–1661, Nov. 1997.

[19] D. McMillan, "Delay analysis of a cellular mobile priority queuing system," IEEE/ACM Trans. Netw., vol.3, no.3, pp.310–319, June 1995.

Haw-Yun Shin received his B.S. and M.S. degree from National Taiwan University of Science and Technology in 1990 and 1992, respectively. He is working toward the Ph.D. degree in the Department of Computer Science and Information Engineering at the National Taiwan University of Science and Technology. His current research is focused on performance analysis for mobile communication systems.

Jean-Lien C. Wu received the B.S. in electrical engineering from National Taiwan University, Taipei, Taiwan, in 1970, and the Ph.D. degree in electrical engineering from Cornell University, Ithaca, NY, in 1976. Her current research interests are in wireless networks, broadband Internet, and mobile multimedia networks. From 1976 to 1977, she was a lecture in the Dept. of EE in Cornell Univ. From 1978 to 1984, she was Member of the Technical Staff with the Business Switching Systems Dept., Bell Laboratories, Holmdel, NJ. Since 1984, she has been a professor in the Electronic Engineering Department, National Taiwan Univ. of Science and Technology (NTUST), Taipei, Taiwan. She was Head of the department from 1987 to 1990 and Dean of the College of Electrical and Computer Engineering from 1998 to 2001. She is currently Dean of the Academic Affair of the university. Dr. Wu was a member of the editorial advisory board of Elsevier Computer Communications from 1994 to 1999. She was an editor of Journal of Information Science and Engineering (JISE) from 1996 to 2002. She was the Editor-in-Chief of Journal of Chinese Institute of Electrical Engineering (JCIEE) from 2000 to 2003. Since, Dec. 2002, Dr. Wu has been the coordinator of the Communication Engineering Program of the Engineering and Applied Sciences Dept. of the National Science Council (NSC), Taiwan. She is also a member of the advisory board of the Science and Technology Advisory Office of the Ministry of Education, Taiwan. Dr. Wu is a member of the IEEE communication society and computer society. She is also member of the Chinese Institute of Engineers society and the Chinese Institute of Electrical Engineering society.

Wei-Yeh Chen received his B.S. degree from National Cheng Kung University in 1986, and M.S. degree from National Chiao Tung University in 1988. He is working toward the Ph.D. degree in the Department of Information Management at the National Taiwan University of Science and Technology. His current research is focused on performance analysis for mobile communication systems.