

A New Fair Call Admission Control for Integrated Voice and Data Traffic in Wireless Mobile Networks

Young Ha Hwang*, Sung-Kee Noh*, and Sang-Ha Kim**

Abstract: It is essential to guarantee a handoff dropping probability below a predetermined threshold for wireless mobile networks. Previous studies have proposed admission control policies for integrated voice/data traffic in wireless mobile networks. However, since QoS has been considered only in terms of CDP (Call Dropping Probability), the result has been a serious CBP (Call Blocking Probability) unfairness problem between voice and data traffic. In this paper, we suggest a new admission control policy that treats integrated voice and data traffic fairly while maintaining the CDP constraint. For underprivileged data traffic, which requires more bandwidth units than voice traffic, the packet is placed in a queue when there are no available resources in the base station, instead of being immediately rejected. Furthermore, we have adapted the biased coin method concept to adjust unfairness in terms of CBP. We analyzed the system model of a cell using both a two-dimensional continuous-time Markov chain and the Gauss-Seidel method. Numerical results demonstrate that our CAC (Call Admission Control) scheme successfully achieves CBP fairness for voice and data traffic.

Keywords: CAC, QoS, Fairness, Integrated service, Wireless mobile networks

1. Introduction

Because network resources are limited, today's wireless mobile networks need to become more efficient by using a dynamic allocation of the bandwidth to satisfy different service demands. Integrated voice and data traffic results in the existence of two major QoS metrics, CDP and CBP, in wireless mobile networks. As mobile users move from one cell to another during a call, all handoff calls should maintain an acceptable level of service quality. If no channel is available for the new call, the handoff call is blocked or the on-going call is forcefully terminated. However, from the point of view of mobile users, the forced termination of an on-going call is more critical than the blocking of a new call. As a result, guaranteeing the CDP constraint has been a major challenge for QoS in wireless mobile networks.

The CAC problem in wireless mobile networks has been the focus of many recent studies [1-8]. In [1-3], the QoS of only one service class was considered. On the other hand, [4-8] introduced an adaptive call admission mechanism in which the admission threshold for different classes of traffic is updated periodically to adapt to changing traffic conditions. However, as previously mentioned, research up to now has only concentrated on CDP constraints and the performance parameters related to CBP have received little attention, even though CBP performance is very

closely related to resource utilization and handoff calls. One of the performance parameters related to the CBP problem in integrated wireless mobile networks is that of the fairness problem between services. Since integrated data and voice traffic inherently require different bandwidths, the current admission control policies create a bias against data traffic; data traffic receives a higher CBP than voice traffic since data traffic requires more bandwidth than voice traffic. For this reason, in most schemes data traffic packets are hardly admitted, leading to serious CBP unfairness. Thus, a new CAC algorithm has recently been developed to admit all types of service fairly.

To overcome serious CBP unfairness between data and voice traffic in wireless networks, Epstein et al. [9] suggested a fair CAC algorithm via a blocking probability measurement function (BPMF), which enables the control of the relative admitting probability between wideband and narrowband calls. Such a BPMF algorithm serves to block users of an "overly privileged" class in order to accommodate users of the "underprivileged" classes. To achieve this, independent multi-class one-step prediction-complete sharing and reservation (IMOSP-CS and IMOSP-RES) is incorporated with a new resource management, which partitions the available bandwidth to reflect the desired blocking probability profile. Greater bandwidth is allocated to underprivileged calls if the CBP ratio between services is greater than the predetermined threshold. The numerical results demonstrate that BPMF actually achieves CBP fairness between wideband and narrowband calls.

However, IMOSP controls the reservation partition by a simple resource management algorithm so that it often leads to system abnormalities depending on traffic behavior. Above all, IMOSP cannot guarantee short-term

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Corresponding Author: Young Ha Hwang

* Electronics and Telecommunications Research Institute, Daejeon, Korea (hyh@etri.re.kr, sknoh@etri.re.kr)

** Dept. of Computer Science, Chungnam National University, Daejeon, Korea (shkim@cnu.ac.kr)

fairness in normal traffic conditions, much less guarantee long-term fairness under heavy traffic conditions. To cope with its weakness, we developed a new CAC algorithm and resource management using the biased coin method [10-13] to guarantee both short-term fairness and long-term fairness, as well as improve resource utilization.

In this paper, we propose a new fair call admission control algorithm for integrated data and voice traffic in wireless mobile networks. This algorithm is based on an additional buffer for underprivileged data traffic and on the biased coin method [10-13], which is used to balance unfairness. Significantly different from voice traffic, data packets do not have rigid delay constraints. So, the proposed buffer will queue data traffic packets instead of immediately rejecting them when the residual bandwidth is insufficient. Unlike data traffic, voice traffic is immediately blocked if no available bandwidth remains. In addition to the buffer system, long-term as well as short-term CBP fairness concepts are applied. Long-term CBP fairness is balanced depending on each short-term CBP fairness level. This algorithm is based on the biased coin method [10-13]. If short-term CBP is unfair, privileged service packets are blocked during the next short-term so that more underprivileged traffic can be admitted. Otherwise, each service packet competes for the available resources. We analyze the system model of a cell using a two-dimensional continuous-time Markov chain and the Gauss-Seidel method [14]. The numerical results demonstrate that our CAC (Call Admission Control) scheme successfully achieves CBP fairness for voice and data traffic and improves resource utilization.

The remainder of this paper is organized as follows. In the next section, related works and the theoretical background to wavelet thresholding are presented; in Section 3 we describe our noise reduction algorithm; in Section 4 we present some experimental results of our proposed scheme; and, finally, our conclusions are offered in Section 5.

2. Fair Call Admission Control

In this research, we consider the call admission control problem with fair CBP for integrated data and voice traffic in wireless mobile networks. In Fig.1, the system model is depicted for each cell. There are three types of traffic channels: voice, data, and shared. Voice and data channels (V_k and D_k) are designated specifically for voice and data traffic, respectively, while shared channels (S_k) can be used for either type of traffic. The admission control policy is based on the following rules:

- A newly arriving voice call (arrival rate: λ_{voice}) is accepted if channels designated for voice traffic can accommodate a new voice call.
- When there is no available channel for voice traffic, i.e. when the number of existing voice calls is greater than or equal to the number of channels designated for

voice traffic, a newly arriving voice call can still be accepted if the current rate of channel occupancy is lower than the predetermined threshold.

- A newly arriving data call (arrival rate: λ_{data}) is accepted if the current buffer occupancy is lower than the predetermined threshold.
- Handoff voice calls (arrival rate: λ_{hvoice}) are accepted as long as resources remain available.
- Handoff data calls (arrival rate: λ_{hdata}) are accepted as long as the buffer (b_k) is not fully occupied.

The above admission call control policy can guarantee the CDP constraint for designated resource amounts. However, CBP fairness is not taken into account in this policy. For CBP fairness, an additional operation must be appended to the basic admission control policy. This additional operation is only applied to the admission decision regarding new calls. Let M be the number of points needed for the function to achieve an adequate level of significance, determined using the input variables. A blocking threshold that defines the level of resolution to which we would like to match the call blocking probabilities of the different traffic classes is assumed. M is defined as one short-term interval.

After M occurs, we then compare the value of the CBP among the classes. The blocking probability profile defines the desired relationship between the blocking probability function of the traffic classes in equation (1), where b_{voice} and b_{data} are the number of blocked calls and r_{voice} and r_{data} indicate the number of requesting calls during time $(t-1, t)$ at update time t :

$$CBP_ratio(t) = \alpha * CBP_ratio(0, t-1) + (1-\alpha) * \frac{b_{voice} / r_{voice}}{b_{data} / r_{data}} (t-1, t) \quad (1)$$

If this CBP ratio is less than the predetermined FIF (Fairness Indication Factor), then both data and voice traffic are admitted fairly. Thus, the usual CAC policy is applied as above. Otherwise, the admission controller initializes a new variable for each traffic class, called as REJECTED_CALLS, which indicates the number of calls that should be rejected in the next short-term interval. This variable is computed and is given by equation (2):

$$\begin{aligned} REJECTED_CALLS_{voice} &= M * (1 - CBP_{data}(t-1, t)) \\ REJECTED_CALLS_{data} &= M * (1 - CBP_{voice}(t-1, t)) \end{aligned} \quad (2)$$

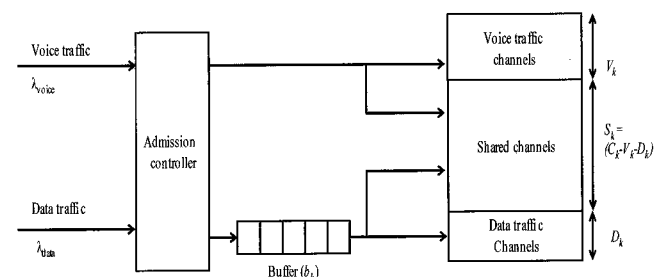


Fig. 1. System model for a cell

This calculation implies the usage of the biased coin method concept [9-12]. In other words, in order to adjust unfairness, more calls in the traffic class with the lower CBP should be blocked while calls with higher CBP should be blocked less. Thus, long-term CBP fairness is adjusted depending on short-term CBP and gradually becomes balanced. Once the REJECTED_CALLS have been calculated, the admission controller applies these conditions to the aforementioned call admission control policy. The fair call admission control policy is based on the following rules:

- If the number of blocked calls in this short-term interval is fewer than the number of REJECTED_CALLS, the requesting data and voice calls are blocked without any conditions.
- If the number of blocked calls already exceeds the number of REJECTED_CALLS, new data or voice calls are accepted or rejected depending on the previous call admission policy. That is, a two-tier call admission policy is accomplished for new data and voice calls where the FIF (Fairness Indication Factor) is larger than the predetermined threshold.

These additional conditions are applied only when the cell is congested. If these conditions are applied when the cell is not congested, low resource utilization may occur because a number of calls equal to the number of REJECTED_CALLS would be dropped even if adequate resources were available.

2.1 The Model

In this section, we present a model to evaluate the network performance in terms of CBP fairness. It is

assumed that, in each cell, the inter-arrival times of all new and handoff voice and data calls are exponentially distributed with the rates λ_{voice} , λh_{voice} , λ_{data} and λh_{data} ; that each cell behaves independently of other cells and that all cells are uniform (for example, it is assumed that the transition and handoff rate are the same for all cells in the network); that the duration of each call for voice and data traffic is exponentially distributed with a mean of $1/\mu_{\text{voice}}$ and $1/\mu_{\text{data}}$ respectively; and, finally, that the period of time that a call stays in the wireless cell before moving into other wireless cells follows an exponential distribution with means $1/\eta_{\text{voice}}$ and $1/\eta_{\text{data}}$.

For each cell, the system is modeled by a continuous-time model with different arrivals (voice and data), multiple designated channels (voice, data, and shared channels), and a finite buffer. In this model, we show that voice and data traffic are exponentially distributed with arrival rate $\lambda_1 = \lambda_{\text{voice}} + \lambda h_{\text{voice}}$, $\lambda_2 = \lambda_{\text{data}} + \lambda h_{\text{data}}$ and that the channel occupancy times for voice and data traffic are summed with the means $1/\mu_1 = 1/(\mu_{\text{voice}} + \eta_{\text{voice}})$ and $1/\mu_2 = 1/(\mu_{\text{data}} + \eta_{\text{data}})$ respectively.

2.2. Analysis

The system model shown in Fig. 1 can be described by a two-dimensional (i, j) continuous-time Markov chain where i and j denote the number of existing voice and data calls in the cell, respectively. The state space of the system is defined as follows:

$$S = \{(i, j) \mid 0 \leq i < V_k, 0 \leq j \leq (C_k - V_k)/BU + b_k\} \text{ and } V_k \leq i \leq C_k - D_k, 0 \leq j \leq ((C_k - i)/BU + b_k),$$

where C_k is the total number of channels in cell and BU is the required basic bandwidth unit of data traffic.

The state (i, j) changes to $(i+1, j)$ with the rate λ_1 if there

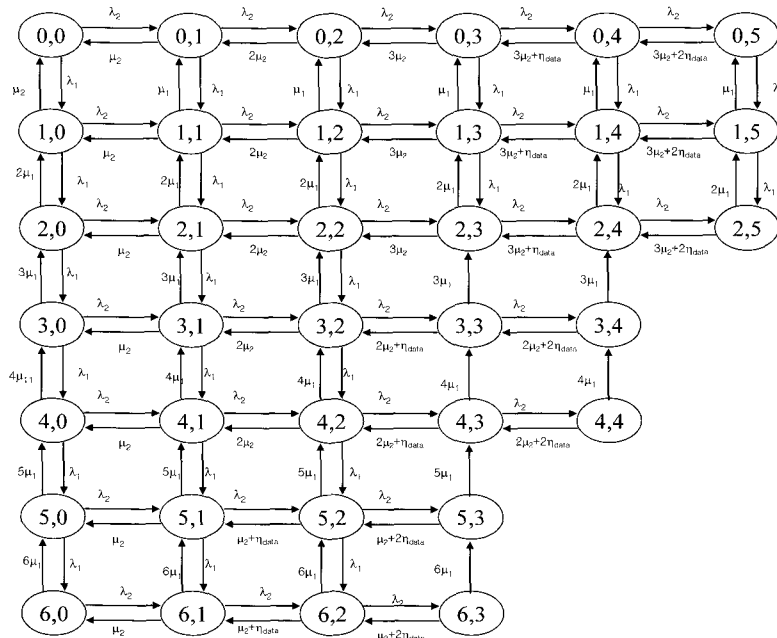


Fig. 2. Two-dimensional Markov chain for $C_k=8$, $V_k=2$, $D_k=2$, $b_k=2$, and $BU=2$

are available channels remaining, and changes to $(i, j+1)$ with the rate λ_2 if the data traffic buffer is not full. When one of the channels occupied by i voice traffic calls is released (with rate $i^* \mu_{\text{voice}}$), the state (i, j) changes to $(i-1, j)$. Unlike the previous three cases, here two processes contribute to the transition from (i, j) to $(i, j-1)$: the release of channels occupied by either a data call or a handoff departure of queued data calls in the buffer. If $i < V_k$ and $j > C_k - V_k$, the state changes from (i, j) to $(i, j-1)$ with the rate $(C_k - V_k) \mu_2 + (j - (C_k - V_k) / BU) \eta_{\text{data}}$. Also, another transition occurs with the rate $(C_k - i) \mu_2 + (j - (C_k - i) / BU) \eta_{\text{data}}$ if $i > V_k$ and $j > C_k - i$. Figure 2 shows an example for $C_k=8$, $V_k=2$, $D_k=2$, $b_k=2$, and $BU=2$ (2 data traffics = 4 units).

Let P_{ij} be the steady state probability that i voice calls and j data calls exist simultaneously in the cell. The corresponding balance equations are shown as four distinct cases.

- Case 1: The number of existing voice calls in the cell is less than V_k ; that is, $0 <= i <= V_k - 1$
- Case 2: The number of existing voice calls in the cell is equal to V_k , i.e. $i = V_k$.
- Case 3: The number of existing voice calls exceeds V_k but is less than the total number of channels for voice traffic: i.e. $V_k + 1 <= i <= C_k - V_k - 1$.
- Case 4: The number of existing voice calls in the cell is equal to $C_k - D_k$; i.e. $i = C_k - D_k$.

For example, we can define the corresponding balance equations for case 3 as follows:

$$\begin{aligned} & [\lambda_1 + \lambda_2 + i\mu_1 + j\mu_2] P_{ij} = \lambda_1 P_{i-1j} + (i+1)\mu_1 P_{i+1j} + \lambda_2 P_{ij-1} \\ & + (j+1)\mu_2 P_{ij+1}, \text{ for } 0 <= j <= (C_k - i) / BU - 1; \\ & [\lambda_2 + i\mu_1 + j\mu_2] P_{ij} = \lambda_1 P_{i-1j} + (i+1)\mu_1 P_{i+1j} + \lambda_2 P_{ij-1} + [((C_k - i) \\ & \mu_2 + (j - (C_k - i) / BU) * \eta_{\text{data}})] P_{ij+1}, \text{ for } j = (C_k - i) / BU; \\ & [\lambda_2 + i\mu_1 + ((C_k - i) / BU) \mu_2 + (j - (C_k - i) / BU) \eta_{\text{data}}] P_{ij} = \\ & (i+1)\mu_1 P_{i+1j} + \lambda_2 P_{ij-1} + [((C_k - i) / BU) * \mu_2 + (j - (C_k - i) / \\ & BU + 1) * \eta_{\text{data}}] P_{ij+1}, \text{ for } ((C_k - i) / BU + 1) <= j <= ((C_k - i) / \\ & BU + b_k - 1); \\ & [i\mu_1 + ((C_k - i) / BU) \mu_2 + b_k \eta_{\text{data}}] P_{ij} = \lambda_2 P_{ij-1}, \text{ for } 0 <= j <= (C_k - i) / BU + b_k. \end{aligned}$$

The steady state probabilities satisfy the following normalization condition:

$$\sum_{i=0}^{V_k} \sum_{j=0}^{(C_k - V_k) / BU + b_k} P_{ij} + \sum_{i=V_k+1}^{C_k - D_k} \sum_{j=0}^{(C_k - i) / BU + b_k} P_{ij} = 1. \quad (3)$$

Also, these steady state probabilities can be solved by using one of the classical iterative methods, the Gauss-Seidel method [14]. Both the arrival and departure rate of voice and data traffic can be presented for each state (i, j) . By using both the normalization condition and the iterative procedure for steady state probability, we can obtain converged P_{ij} values.

The new call and handoff blocking probability can be

estimated by these steady state probabilities. Consider the blocking probability under normal conditions. A new voice call is accepted if the number of existing voice calls is less than V_k . However, when the number of existing voice calls is greater than or equal to V_k , a new voice call is admitted only when the channel occupancy in a cell is smaller than the predetermined threshold, T_{voice} . Therefore, the voice new-call blocking probability, P_B^{voice} , is given by:

$$P_B^{\text{voice}} = 1 - \left[\sum_{i=0}^{V_k-1} \sum_{j=0}^{(C_k - V_k) / BU + b_k} P_{ij} + \sum_{i=V_k}^{C_k - D_k - 1} \sum_{j=0}^{i + j * BU < T_{\text{voice}}} P_{ij} \right]. \quad (4)$$

A new handoff voice call is accepted if available channels remain. Thus, the handoff dropping probability of voice calls is given by:

$$P_D^{\text{voice}} = 1 - \left[\sum_{i=0}^{V_k-1} \sum_{j=0}^{(C_k - V_k) / BU + b_k} P_{ij} + \sum_{i=V_k}^{C_k - D_k - 1} \sum_{j=0}^{i + j * BU < C_k} P_{ij} \right]. \quad (5)$$

Furthermore, a new data call is accepted if the buffer occupancy is less than or equal to the predetermined threshold, T_{data} . Therefore, the new call blocking probability of data traffic, P_B^{data} , is given by:

$$P_B^{\text{data}} = \sum_{i=0}^{V_k} \sum_{j=T_{\text{data}}}^{b_k} P_{i(C_k - V_k) / BU + j} + \sum_{i=V_k+1}^{C_k - D_k} \sum_{j=T_{\text{data}}}^{b_k} P_{i(C_k - i) / BU + j}. \quad (6)$$

A handoff data call is blocked only when the buffer is full. Hence, the handoff dropping probability of data traffic can be expressed as:

$$P_D^{\text{data}} = \sum_{i=0}^{V_k} P_{i(C_k - V_k) / BU + b_k} + \sum_{i=V_k+1}^{C_k - D_k} P_{i(C_k - i) / BU + b_k}. \quad (7)$$

2.3. Iterative Algorithm

In this section, we propose an iterative algorithm to analyze and evaluate a new, fair CAC policy in this paper. Using equations (4) and (6), the expected CBP of both voice and data traffic is calculated. If this CBP of both voice and data traffic is unfair, the recovery procedure is performed. This procedure starts by initializing the REJECTED_CALL variable. During the next short-term, the number of calls that can be accepted from each traffic service is bounded as the average $\lambda * M$ events * $(1 - \text{CBP of either data or voice traffic})$. To analyze the fairness ratio in our CAC policy, the CBP of each traffic service can be obtained by equation (2). Once each CBP has been computed, the arrival rate can be calculated by the above method. We iterate this procedure until the CBP ratios converge and use these quantities as estimates for the long-term CBP fairness ratio. The iterative algorithm is shown as follows:

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program CBP fairness ratio
{obtains long-term CBP ratio between data
and voice traffic service with steady state
probability by iterative algorithm};
begin
repeat
  Compute  $P_B^{voice}$  using equation(4);
  Compute  $P_B^{data}$  using equation(6);
   $\lambda_{voice} = (1 - P_B^{data}) * \lambda_{voice} + \lambda h_{voice}$ ;
   $\lambda_{data} = (1 - P_B^{voice}) * \lambda_{data} + \lambda h_{data}$ ;
until  $\lambda_{voice} / \lambda_{data}$  converge;
end.

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3. Numerical Results

This section presents our numerical analysis for the performance of our scheme from the perspective of CBP fairness and resource utilization. The analysis is conducted with IMOSP[9]. The cell capacity accommodates 20 units. The analysis environments are designed according to the offered traffic load as shown in Table 1, where $V_k = 2$, $D_k = 4$, $b_k = 8$, and $BU = 2$.

As can be seen in Fig.3-(a) and Fig.3-(b), both short-term and long-term CBP fairness are only achieved under our method. IMOSP shows slow convergence for long-term CBP fairness. So, there is some difficulty in guaranteeing short-term CBP fairness. These figures also indicate that ours has a lower CBP of wideband calls than IMOSP in the long-term period. This is because more narrowband (voice) calls are blocked than others so that the remaining bandwidth can be used for more wideband (data) calls. The results for case 2 are very noticeable.

Table 1. Traffic values for analysis

Type	Parameters	Case1	Case2
Voice (narrow)	Call arrival rate (calls/sec)	0.5	0.1
	Required Bandwidth (units)	1	1
	Service time in a cell (sec)	4	2
Data (wide)	Call arrival rate (calls/sec)	0.1	1
	Required Bandwidth (units)	2	2
	Service time in a cell (sec)	4	4

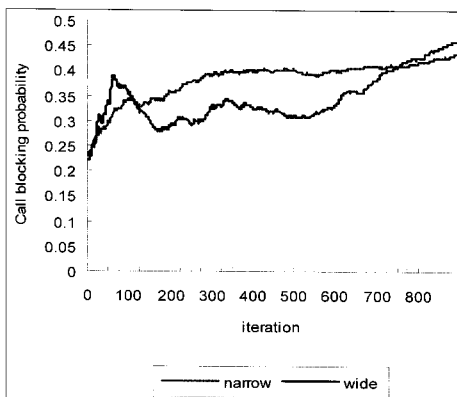


Fig. 3-(a). IMOSP in case 1

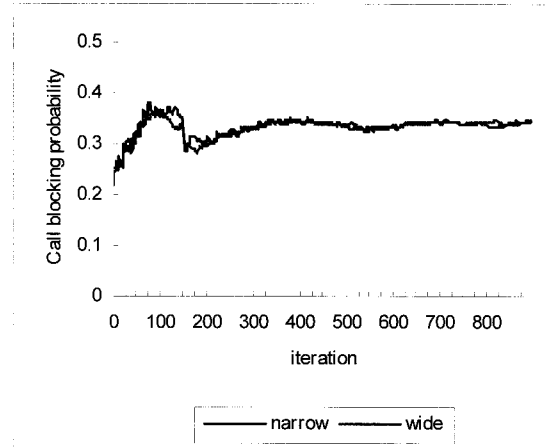


Fig. 3-(b). Ours in Case 1

As can be seen in Fig. 4-(a) and Fig. 4-(b), there is a big difference between IMOSP and ours in terms of CBP fairness. In Case 2, the traffic intensity between wideband (data) and narrowband (voice) is wide. In particular, the wideband call arrives with large traffic. IMOSP shows an obvious CBP unfairness between wideband calls and narrowband calls. On the other hand, our scheme shows a fair CBP between the two services. We can observe from Fig. 4-(b) that the CBP of wideband decreases as the CBP of narrowband increases. After all, the two CBP are converged into their average CBP value.

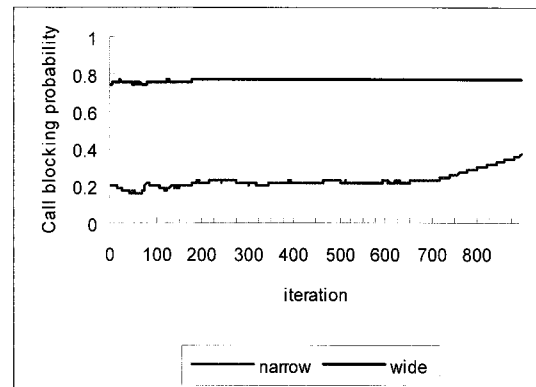


Fig. 4-(a). IMOSP in Case 2

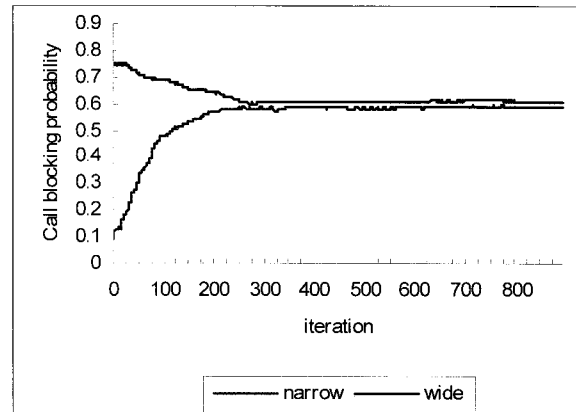


Fig. 4-(b). Ours in Case 2

With regard to resource utilization, the analysis results are shown in Fig. 5-(a) and Fig. 5-(b). As can be seen from these figures, the most efficient resource utilization is achieved in our scheme. On the other hand, IMOSP shows the worst resource utilization. In case 2, there are few differences between the two schemes. However, regarding IMOSP in case 1, the resource utilization of IMOSP is dramatically reduced. This means that IMOSP is affected by traffic behavior to such a great extent that it cannot manage resources efficiently and promptly. So, excessive resources are allocated to underprivileged services and wasted.

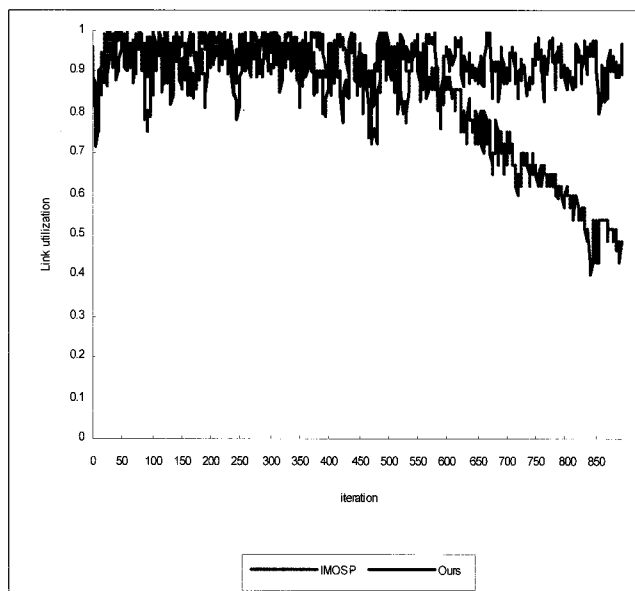


Fig. 5-(a) Utilization in Case 1

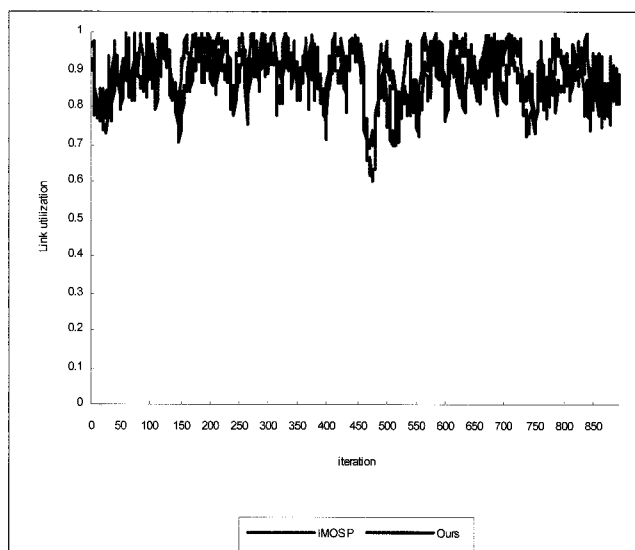


Fig. 5-(b) Utilization in Case 2

Fig. 6-(a) and Fig. 6-(b) show the resources occupied by narrowband (voice) and wideband (data) vs. link capacity. This factor can in part be considered as resource fairness. In IMOSP, narrowband calls occupy 10% of total capacity

in the latter part. However, ours shows fairer resource usage than IMOSP, mainly because we make use of the minimum channel pool concept, which cannot be occupied by other services. It prevents all resources from being occupied by one service class.

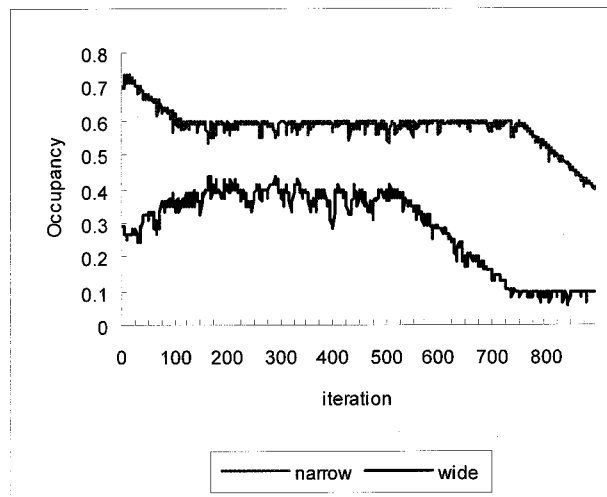


Fig. 6-(a) Resource Occupancy in IMOSP

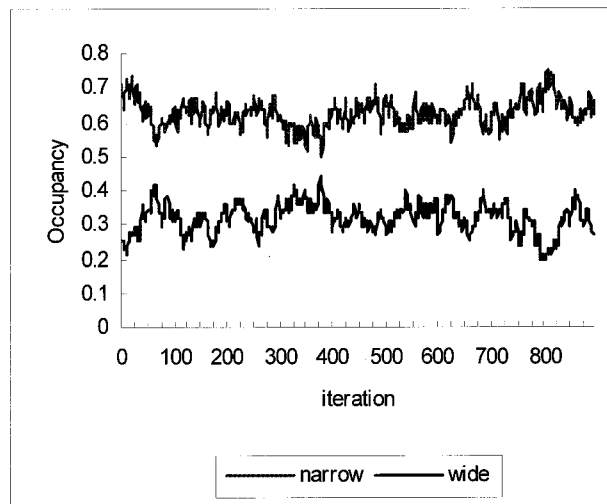


Fig. 6-(b) Resource Occupancy in Ours

4. Conclusion

Guaranteeing CDP constraints irrespective of the movement between cells is an essential issue for mobile wireless network technology. Previous studies have only concentrated on the CDP QoS metric. However, CBP is also a critical QoS metric and is closely related to resource utilization as well as to the handoff rate to neighboring cells.

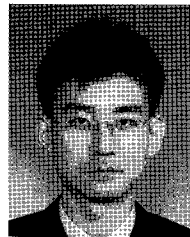
This paper proposed a novel CAC scheme and resource management algorithm that guarantee both short-term and long-term fairness between heterogeneous services with different traffic properties and enhance resource utilization

of the system. By numerical analysis, we demonstrated that our CAC scheme actually achieves a fair admission probability for voice and data traffic and also improves resource utilization regardless of traffic behavior.

Future work will include an analysis of the performance of CAC algorithms that considers real user's demands, and non-Poisson traffic could also be the focus of significant research.

Reference

- [1] D. Hong et al., "Traffic Model and Performance Analysis for Cellular Mobile Radio Telephone Systems with Prioritized and Nonprioritized Handoff Procedures," *IEEE Trans. Vehicular Technology*, vol. 35, no. 3, pp. 77 – 92, 1986.
- [2] O.T.W. Yu and V.C.M. Leung, "Adaptive Resource Allocation for Prioritized Call Admission over an ATM-Based Wireless PCN," *IEEE J. Selected Areas Comm.*, vol. 15, no. 7, pp. 1208-1225, 1997.
- [3] M. Naghshineh and M. Schwartz, "Distributed Call Admission Control in Mobile/Wireless Networks," *IEEE J. Selected Areas Comm.*, vol. 14, no. 4, pp. 711-717, 1996.
- [4] W.B. Yang and E. Geraniotis, "Admission Policies for Integrated Voice and Data Traffic in CDMA Packet Radio Networks," *IEEE J. Selected Areas Comm.*, vol. 12, no. 4, pp. 654-664, 1994.
- [5] M. Naghshineh and A.S. Acampora, "QoS Provisioning in Micro-Cellular Networks Supporting Multiple Classes of Traffic," *Wireless Networks*, vol. 2, pp. 195-203, 1996.
- [6] J. Y. Lee et al., "Realistic Cell-Oriented Adaptive Admission Control for QoS Support in Wireless Multimedia Networks," *IEEE Trans. Vehicular Technology*, vol. 52, no. 3, May 2003.
- [7] F. Prihandoko et al., "Adaptive call admission control for QoS provisioning in multimedia wireless networks," *Journal of Computer Communications*, Elsevier Publisher, November 2002.
- [8] Y. Xiao et al., "Optimal Admission Control for Multi-Class of Wireless Adaptive Multimedia Services", *IEICE Transactions on Communications*, Special Issue on Mobile Multimedia communications, vol. E84-B, no.4, pp.795-804, April 2001.
- [9] B. M. Epstein et al., "Predictive QoS-based admission control for multiclass traffic in cellular wireless networks," *IEEE JSAC*, vol.18, no. 3, pp.523-534, March 2000.
- [10] B. Efron, "Forcing a Sequential Experiment to be balanced," *Biometrika*, vol. 58, pp. 403 – 417.
- [11] L. J. Wei, "The Adaptive Biased Coin Design for Sequential Experiments," *Journal of Annals of Statistics*, vol. 6, pp. 92 – 100, Jan. 1978.
- [12] J. M. Steele, "Efron's Conjecture on Vulnerability to Bias in A Method for Balancing Sequential Trials," *Biometrika*, vol. 67, pp. 503 – 504.
- [13] S.J. Pocock, *Clinical Trials : A Practical Approach*, John Wiley & Sons Ltd., pp. 79 – 80, 1991.
- [14] R. Cooper, *Introduction to Queueing Theory*, 1981.



Young-Ha Hwang

He received M.S. degree in Industrial Engineering from Chonnam National University, Korea, in 2001. He is a member of INCOSE and a senior engineer at Electronics and Telecommunications Research Institute (ETRI), Korea. His research areas are system

and software engineering, communications system performance analysis, and QoS in Wireless Mobile Networks.



Sung-Kee Noh

He received M.S. degree in Industrial Engineering from POSTECH, Korea, in 1992. He received Ph.D. degree in Computer Science from Chungnam National University, Korea, in 2006. He is a senior engineer at Electronics and Telecommunications Research

Institute (ETRI), Korea. His research areas are QoS in BcN Networks, communications system performance analysis, and QoS in Wireless Mobile Network.



Sang-Ha Kim

He received the B.S. degree in chemistry from Seoul National University, Seoul, Korea, in 1980. He received the M.S. degree and Ph.D. degrees in quantum scattering and computer science from University of Houston, in 1984 and 1989, respectively.

From 1990 to 1991, he was in Supercomputing Center, SERI, Korean Institute of Science and Technology (KIST) as senior researcher. Since joining Chungnam National University, Korea, in 1992, he is currently a professor. His current research interests include wireless network, ad hoc network, QoS, optical network, network analysis.