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Enabling seamless multimedia wireless access through QoS-based bandwidth adaptation

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Summary

Effective support of real-time multimedia applications in wireless access networks, *viz.* cellular networks and wireless LANs, requires a dynamic bandwidth adaptation framework where the bandwidth of an ongoing call is continuously monitored and adjusted. Since bandwidth is a scarce resource in wireless networking, it needs to be carefully allocated amidst competing connections with different Quality of Service (QoS) requirements. In this paper, we propose a new framework called QoS-adaptive multimedia wireless access (QoS-AMWA) for supporting heterogeneous traffic with different QoS requirements in wireless cellular networks. The QoS-AMWA framework combines the following components: (i) a threshold-based bandwidth allocation policy that gives priority to handoff calls over new calls and prioritizes between different classes of handoff calls by assigning a threshold to each class, (ii) an efficient threshold-type connection admission control algorithm, and (iii) a bandwidth adaptation algorithm that dynamically adjusts the bandwidth of an ongoing multimedia call to minimize the number of calls receiving lower bandwidth than the requested. The framework can be modeled as a multi-dimensional Markov chain, and therefore, a product-form solution is provided. The QoS metrics—new call blocking probability (NCBP), handoff call dropping probability (HCDB), and degradation probability (DP)—are derived. The analytical results are supported by simulation and show that this work improves the service quality by minimizing the handoff call dropping probability and maintaining the bandwidth utilization efficiently.

KEY WORDS: QoS provisioning; priority classes; multimedia; bandwidth adaptation; wireless cellular networks; multi-dimensional Markov chain

1. Introduction

Wireless communication technology has been making significant progress in the recent past and will be playing a more important role in access networks, as evidenced by the widespread adoption of cellular networks, wireless local area networks (WLANs), and wireless home networks. Users of these wireless access networks are expecting high quality, reliability, and easy access to high-speed services anytime, anywhere, and in any form. The expected services will include multimedia applications that need real-time guarantees. A wireless multimedia application enables the simultaneous transmission of voice, data, text, and

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images through radio links by means of future wireless access technologies. Wireless multimedia applications include mobile commerce, geographical and location information, Web services, cooperation group work, streaming media and entertainment, voice and gaming. These and other emerging wireless multimedia applications come with high user expectations and increased demands on network resources. Different wireless multimedia applications have diverse bandwidth and Quality of Service (QoS) requirements that need to be guaranteed by wireless networks. To achieve this goal, QoS provisioning in wireless multimedia networks is critical.

In addition to packet level QoS issues (related to packet-delay bound, throughput, and packet loss probability) considered in References [1,2], connection-level QoS issues (related to connection establishment and management) need to be addressed, as users are expected to move around during communication sessions, causing handoffs between cells. One of the most important connection-level QoS issues is how to provide mobile users a means of 'seamless handoff' communication and thus reducing handoff drops due to lack of available resources in the new cell, since mobile users should be able to continue their ongoing sessions.

In this paper, we focus on the provisioning of connection-level QoS in the multimedia wireless networks. Infrastructure of these networks implement the cellular architecture. Connection-level QoS in wireless cellular networks is usually expressed in terms of *new call blocking probability* (*NCBP*) and *handoff call dropping probability* (*HCDP*) [3]. A new call results when a user requests a new connection, while a handoff call occurs when an active user moves from one cell to another. Thus, NCBP is the probability of a new arriving call being rejected while the HCDP is the probability that an accepted call is terminated before the completion of its service, that is, the probability that a handoff attempt fails [3].

Provisioning of QoS in wireless cellular networks is complicated by the limited resources (e.g., radio link bandwidth), the highly fluctuated wireless environment and the user's mobility. The problem becomes even more challenging as recent wireless cellular networks have been implemented based on small-size cells (i.e., microcells or picocells [4]) to allow higher transmission capacity, and thus to achieve better performance. However, small-size cells increase the handoff rate, and result in rapid changes in the network traffic conditions, making QoS guarantees difficult [5].

Recently, adaptive multimedia networking is introduced [6] where the bandwidth of an ongoing call can vary during its lifetime. Originally, the concept of the adaptive multimedia networking was introduced in wired networks. In broadband ATM networks, for instance, once a call is admitted to the network, a contract between network and application is established. Then, they both try to keep the contract throughout the call's lifetime. In such a model, network congestion can cause fluctuations in the availability of network resources, thereby resulting in severe degradation of multimedia services. To overcome this problem, many adaptive multimedia encoding and/or networking schemes are proposed, such as hierarchical encoding [7] and network filters [8], to mitigate the effect of fluctuation in the network resources.

Adaptive multimedia networking becomes very attractive due to the scarcity of wireless resources, the availability of a wide range of bandwidth granulites, and their ability to mitigate the highly fluctuating link bandwidth in wireless networks. In multimedia wireless networks using the adaptive framework, it is possible to overcome the link overload condition by dynamically adjusting the bandwidth of individual ongoing calls, which we call 'bandwidth adaptation.' With the help of such adaptive framework, the dropping probability of handoff calls can be reduced to a negligible level under normal traffic load.

In this paper, we focus on both the new call blocking probability (NCBP) and the handoff call dropping probability (HCDP) as the main QoS requirements. In addition to the NCBP and the HCDP, we introduce a new QoS parameter: the call degradation probability (DP). It represents the average ratio of the number of degraded calls to the number of calls at any state of the system. We refer to a call as degraded if its assigned bandwidth is below its requested bandwidth.

1.1. Related Work and Motivation

Several adaptive multimedia frameworks in wireless networks have been introduced in the literature [9–15]. In References [9–12] where it is assumed that all calls belong to a single class of adaptive multimedia traffic and receive varying bandwidth assignments from a discrete set of integer bandwidth values.

Multiple classes of adaptive multimedia services in cellular wireless networks have been introduced in References [13–15] without considering the prioritization between new call arrivals and handoff calls for each class of traffic. However, in our framework we provide a bandwidth allocation policy that takes into account the separation between incoming traffic for each class and prioritizes handoff calls over new calls. Xiao *et al*. in References [13,14] proposed a fair adaptive bandwidth algorithms for multiple classes of connections. Fairness among classes is achieved by partitioning the bandwidth according to arrival rates. The disadvantage of these algorithms is that they cause all ongoing connections to receive reduced bandwidth. In order to avoid this problem, we design a bandwidth adaptation algorithm that aims to minimize DP as well as to reduce the complexity of the bandwidth adaptation process where not all the connections are reduced. A prioritization in the process of bandwidth adaptation among multiple classes of multimedia services is presented in Reference [15] where the bandwidth of calls with lower priority is preferably adapted. However, in this approach the authors assume no handoff dropping which makes their work impractical. This is not the case in our work since a handoff call can be dropped if it does not satisfy the adaptation condition.

In this paper, we model the proposed framework as a multi-dimensional Markov chain. Multi-dimensional Markov chains may not have product-form solutions and can be computationally expensive to solve due to the explosion of their state-space. However, we show in our model that product-form results prevail. Also, we provide a mathematical derivation for the NCBP, the HCDP and the DP for each traffic class. To the best of our knowledge, no mathematical analysis for such adaptive framework has been previously reported in the literature.

1.2. Contributions

The main contribution of this paper is the introduction and analytical modeling of a novel adaptive multimedia framework for multimedia wireless networks. The framework operates at the connection-level where the bandwidth of ongoing connections can be dynamically adjusted. This framework supports multiple classes of adaptive multimedia services that have diverse QoS requirements. Our proposed work integrates a threshold-based bandwidth allocation policy, a connection admission control (CAC) algorithm, and a bandwidth adaptation algorithm (BAA) into a new framework called QoS-adaptive multimedia wireless access (QoS-AMWA). The objective of QoS-AMWA is to provide the mobile users with seamless handoff communication through the wireless network and to distinguish between handoff calls and new calls for each class of traffic, while minimizing the NCBP and the HCDP.

A block diagram of our proposed QoS-AMWA system architecture is depicted in Figure 1. The QoS

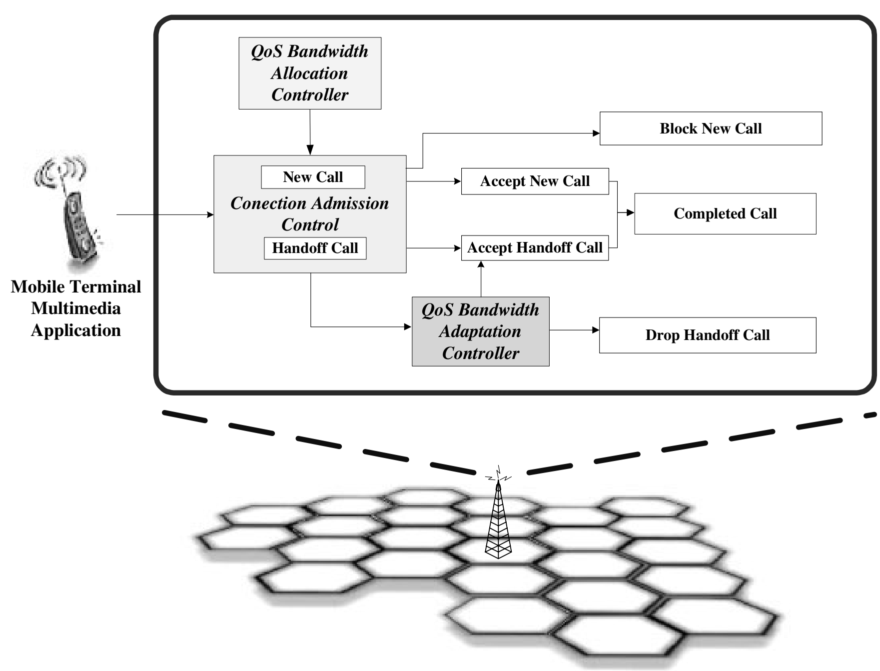

Fig. 1. QoS-AMWA system architecture.

bandwidth allocation controller implements the threshold-based bandwidth allocation policy that priorities connections according to their QoS constraints by assigning a threshold value to each traffic class. The policy is based on reserving bandwidth for aggregate handoff connections, thus giving them priority over new connections for each class of traffic and providing them with low HCDP. We require that during call setup, a mobile terminal (running a user multimedia application) defines its requirements in a traffic profile before it is transmitted to the CAC at the base station. The traffic profile consists of the traffic type (new or handoff) and the bandwidth requirements. The traffic profile is then sent to the CAC, which determines the acceptance or rejection of a call based on traffic type and the amount of available bandwidth in the system. If the available bandwidth is not sufficient to satisfy the new call requests, the call will be blocked. However, for a handoff call if the available bandwidth is not enough to accommodate this call, then the QoS bandwidth adaptation controller is invoked to execute a BAA. BAA sensibly reduces the bandwidth of ongoing calls in the system and assigns the saved bandwidth to the incoming handoff call. Hereafter, a call of any class is referred as 'degraded' if and only if the assigned bandwidth of the call is lower than its requested bandwidth. Note that the QoS bandwidth adaptation controller will also invoke and execute BAA when a call (new or handoff) completion occurs in the system. In this case, the BAA will use the released bandwidth to upgrade the degraded calls if any.

1.3. Organization of the Paper

The remainder of this paper is organized as follows. In Section 2, we describe the network and the adaptive multimedia traffic models. Section 3 presents a detailed description of the adaptive multimedia framework with its main components. In Section 4, we develop an analytical model for the proposed framework. In addition, we provide derivations of several QoS metrics. Section 5 provides the simulation model, analytical model validation, and the performance evaluation. Finally, in Section 6, we conclude the paper and discuss future work directions.

2. Network Description

We consider a system model that supports multiple classes of adaptive multimedia services, which have diverse bandwidth and QoS requirements. This model is frequently used in the literature [16–19]. This section describes our system model and the adaptive multimedia traffic model.

2.1. Cell Configuration

We consider a multimedia wireless network with a cellular infrastructure, comprising a wired backbone and a number of base stations (BSs). The geographical area controlled by a BS is called a *cell*. A mobile, while staying in a cell, communicates with another party, which may be a node connected to the wired network or another mobile, through the BS in the same cell. When a mobile moves into an adjacent cell in the middle of communication session, a handoff will enable the mobile to maintain connectivity to its communication partner, that is, the mobile will start to communicate through the new BS, hopefully without noticing any difference.

In this paper, we are concerned with the QoS guarantees for new and handoff calls in each cell. Therefore, we decompose the cellular network into individual sub-systems, each corresponding to a single cell. The correlation between these sub-systems, results from handoff connections between the corresponding cells, which is re-introduced as an input to each submodel. Under this assumption, each cell can be modeled and analyzed individually. A same model is used for all cells in the network, but the model parameters may be different, reflecting the mobility and traffic conditions in individual cells, as well as the channel assignment policy employed by the network. Therefore, we can model the system at single-cell level.

We assume the system uses fixed channel allocation (FCA), which means each cell has a fixed capacity. No matter which multiple access technology (FDMA, TDMA, or CDMA) is used, we can interpret system capacity in terms of bandwidth. Hereafter, whenever we refer to the bandwidth of a connection, we mean the number of basic bandwidth units (bbu) that is adequate for guaranteeing desired QoS for this connection with certain traffic characteristics.

2.2. Adaptive Multimedia Traffic Model

We assume a fixed capacity in each cell. The fixed total capacity is B (bbu). Traffic arriving at the cell is partitioned into K separate classes based on bandwidth requirements. The bandwidth of a multimedia call can be dynamically adapted depending on the network load situation during its lifetime.

In this paper, we adopt the layered coding approach where the bandwidth of a call can take a set of discrete values as in Reference [20]. Each class-i connection requires a discrete bandwidth value, $b_{i,j}$, where $b_{i,j}$ belongs to the set $B_i = \{b_{i,1}, b_{i,2}, \ldots, b_{i,j}, \ldots, b_{i,K_i}\}$ for $i = 1, 2, \ldots, K$, and $b_{i,j} < b_{i,j+1}$ for $j = 1, 2, \ldots, K_i - 1$. Here, we express discrete bandwidth values in terms of basic bandwidth units. Thus, $b_{i,j}$ is an integer number of basic bandwidth units (bbu). The minimum and maximum values that class-i connection can take are $b_{i,1}$ and b_{i,K_i}, respectively. Also, K_i denotes the number of possible values of bandwidth that a class-i call can be allocated. The requested bandwidth of a class-i connection is denoted as $b_{i,\text{request}}$, where $b_{i,\text{request}} \in B_i$. We assume that all the connections in the same class use the same requested bandwidth, $b_{i,\text{request}}$ and $b_{i,\text{request}} > b_{i,1}$, where $b_{i,1}$ is the minimum bandwidth of class-i. Hence, the requested bandwidth for each class is predetermined.

Let $b_{i,\text{assigned}_j}$ denote the assigned bandwidth for call j, where for new calls $1 \le j \le m_i$ and for handoff calls $1 \le j \le n_i$, of class-i, $1 \le i \le K$. The non-negative integer m_i and n_i denote the number of ongoing new call and handoff call class-i connections, respectively. A call j of class-i is called a 'degraded call' if $b_{i,\text{assigned}_j} < b_{i,\text{request}}$.

3. Adaptive Multimedia Framework

Our adaptive multimedia framework consists of three main components: a threshold-based bandwidth allocation policy, a connection admission control algorithm, and a bandwidth adaptation algorithm.

3.1. Threshold-Based Bandwidth Allocation Policy

As aforementioned, a cell has a fixed total capacity of B (bbu) and traffic arriving at the cell is partitioned into K separate classes based on bandwidth requirements. The classes are indexed in an increasing order according to their bandwidth requirements, such that:

$$b_{1,j} \le b_{2,j} \le \ldots \le b_{K,j}, \qquad \text{for } j = 1, 2, \ldots, K_i$$

New calls and handoff calls are further segmented into separate sub-classes, each representing connections with different QoS requirements.

The main principle of our bandwidth allocation policy is based on reserving channels for aggregate handoff connections, thus giving them priority over new connections, and providing them with lower handoff dropping probability. In addition, the policy prioritizes between different classes of handoff connections according to their QoS constraints by assigning a series of bandwidth thresholds $t_0, t_1, \ldots, t_K$, such that

$$t_0 \le \ldots \le t_i \le t_{i+1} \le \ldots \le t_K$$

where, t_0 denotes the maximum number of total bandwidth units that can be allocated to new connections and t_i, $1 \le i \le K$, denotes the maximum number of total bandwidth units that can be allocated to class-i handoff connections. It should be noted that if the different handoff connections were allowed to completely share the bandwidth, then connections with lower bandwidth requirements will have a better chance at occupying the bandwidth than those with higher bandwidth requirements.

In this paper, the values for these thresholds will be tuned to satisfy the handoff dropping probabilities for each individual traffic class. Note that $\sum_{i=0}^{K} t_i$ may be greater than B, which also allows further sharing of bandwidth. Figure 2 shows the accessible bandwidth regions for a two-class system.

3.2. Connection Admission Control Algorithm

The objectives of any CAC are to satisfy the QoS requirements and to utilize the system resources efficiently [21–23]. In this section, we provide a simple threshold-based CAC algorithm. The algorithm uses the threshold values that are assigned in the bandwidth allocation policy, described in Subsection 3.1, to make the admission/rejection decision for an incoming connection request.

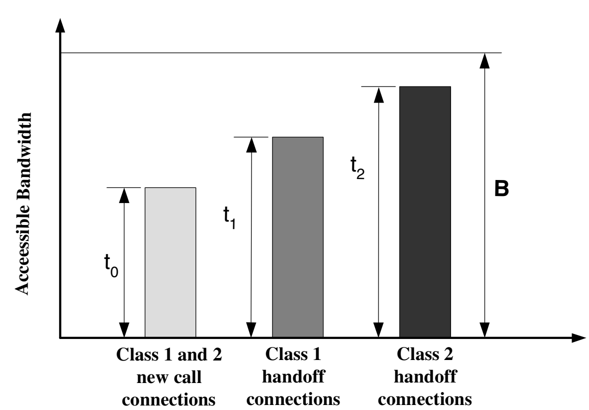

Fig. 2. Accessible bandwidth for a two-class system.

A newly arriving call of class-i is blocked if its requested bandwidth plus the current total bandwidth of ongoing new connections for all classes is greater than t_0, or no more bandwidth is available in the given cell to accommodate the new connection. On the other hand, we allow an incoming handoff connection of class-i to be always momentarily accepted regardless of its bandwidth requirements. If necessary, bandwidth adaptation is performed. Final acceptance of a handoff connection is constrained by the following condition: *The bandwidth adaptation algorithm, described in the following section, can allocate enough bandwidth for the 'accepted' request*. If this is not satisfied then the handoff connection will be dropped.

3.3. The Bandwidth Adaptation Algorithm

A BAA increases or decreases assigned bandwidth of ongoing connections in the cell depending on the network load situation. BAA performs two main procedures: reduction and expansion. The reduction procedure is activated when an incoming handoff call arrives to an overloaded cell. On the other hand, the expansion procedure is activated when there is an outgoing handoff call or a call completion in the given cell.

In this paper, our BAA seeks to satisfy the following objectives at any time instance:

1. Minimizing the DP.
2. Minimizing the number of calls of class-i with lower assigned bandwidth than requested.

We assume that the requested bandwidth of the arriving call (new and handoff) of class-i, $b_{i,\mathrm{request}}$, is predetermined.

We do not apply bandwidth adaptation for incoming new calls as our main objective is to minimize the handoff dropping probability. Therefore, the bandwidth adaptation reduction will only be performed for the incoming class-i handoff calls, $b_{i,\mathrm{request}}$, according to the following cases:

Case 1: no more bandwidth is available in the cell to accommodate this connection

$$b_{i,\mathrm{request}} + \left(\sum_{i=1}^{K} \sum_{j=1}^{m_i} b_{i,\mathrm{assigned}_j} + \sum_{i=1}^{K} \sum_{j=1}^{n_i} b_{i,\mathrm{assigned}_j} \right) > B$$

Case 2: no more bandwidth is available under class-i to accommodate this connection, that is, the current total bandwidth of ongoing class-i handoff connections is greater than t_i, $1 \le i \le K$

$$b_{i,\mathrm{request}} + \sum_{j=1}^{n_i} b_{i,\mathrm{assigned}_j} > t_i, \forall i$$

Table I. Bandwidth reduction procedure decisions.

Case 1	Case 2	Action
False	False	The call is accepted
False	True	Execute single class reduction algorithm (SCRA)
True	False	Execute multiple classes reduction algorithm (MCRA)
True	True	Execute (MCRA) if (SCRA) is successfully executed

Based on these two cases the system will decide which reduction algorithm should be invoked as shown in Table I. The main objective of both algorithms is to accommodate incoming handoff calls and to assign the maximum bandwidth (b_{i,K_i}) for such connections. As aforementioned, the application of BAA may result in the rejection of handoff calls if all tests fail.

The selection of the algorithm will reduce the complexity of the reduction procedure since in the single class reduction algorithm (SCRA) only the handoff connections of class-i are affected in the reduction process, while in multiple classes reduction algorithm (MCRA) all connections in the system may be affected. The detailed description of the SCRA and the MCRA are shown in Table II, where B_{A_i} represents the free available bandwidth of class-i and B_{Tot_A} represents the free available bandwidth in the given cell. Also, $\deg_{nc_i}$ is the number of degraded new calls of class-i, $\deg_{\mathrm{h}_i}$ is the number of degraded handoff calls of class-i, and $\mathrm{d}_{\mathrm{SCRA}_i}$ and $\mathrm{d}_{\mathrm{MCRA}_i}$ are number of handoff dropped calls of class-i after the execution of SCRA and MCRA, respectively.

When a call of class-i departs the cell (outgoing handoff call or a call completion), the available bandwidth of class-i, $\mathrm{B}_{\mathrm{A}_i}$, increases. The system will invoke the expansion procedure to increase the bandwidth for one or more calls of the degraded calls of class-i starting from class 1 and ending with class K. The detailed description of the expansion algorithm is shown in Table III, where b_{upgrade} denotes the required bandwidth to upgrade the bandwidth of the corresponding call and $b_{i,\mathrm{current}}$ denotes the

Table II. Bandwidth reduction algorithms.

```
SCRA(int b_{i,request}, int i, int caller) // i: class index,
                                           // caller: this determine the caller to SCRA,
                                           // caller=0: main program is the caller (default)
                                           // caller=1: MCRA is the caller
{
  d_{SCRA_i}=0;
  if (B_{A_i} ≥ b_{i,request})
  {
    allocateBandwidth(b_{i,j}, b_{i,request}, b_{i,K_i});
    allocationIsDone=1; // the call is accepted and allocated a bandwidth
  }
  else
  {
    reduceAll(b_{i,request}, caller);
      if (B_{A_i} ≥ b_{i,request})
          {
          allocateBandwidth(b_{i,j},b_{i,request}, b_{i,K_i});
          allocationIsDone=1;
        }
      else
          {
            reduceAll(b_{i,1}, caller)
            if(B_{A_i} ≥ b_{i,1})
            {
                allocateBandwidth(b_{i,j},b_{i,request}, b_{i,K_i});
              allocationIsDone=1;
            }
            else
            {
                allocationIsDone=0; // the call is dropped, cannot offer a sufficient bandwidth
                d_{SCRA_i}++;
            }
          }
    }
  return allocationIsDone;
}// end of SCRA
allocateBandwidth(int assignedValue, int minBW, int maxBW)
{
  allocate assignedValue to the incoming call such that
  (assignedValue ≤ B_{A_i}) and (minBW ≤ assignedValue ≤ maxBW)
}// end of allocateBandwidth function
reduceAll(int minLevel, int caller)
{
  For class-i, order the calls decreasingly by bandwidth;
  while(B_{A_i} < b_{i,request} and a call's bandwidth > minLevel exists)
  {
      if (m_i ≠ 0 and caller==1) // new call of class-i is not empty
      {
        reduce the calls with more than minLevel to minLevel
                                     starting with the largest bandwidth;
      add the extra bandwidth to B_{A_i};
      deg_{nc_i} ++;
      m_i --;
      }
    else if (n_i 0) // handoff call of class-i is not empty
    {
        reduce the calls with more than minLevel to minLevel
                                      starting with the largest bandwidth;
      add the extra bandwidth to B_{A_i};
      deg_{h_i} ++;
      n_i --;
      }
  }
}// end of reduceAll function
```

Continues

Table II. Continued.

```
MCRA ()
{
  classIndex=1;
  d_MCRA_i=0;
  while (classIndex ≤ K and B_Tot_A b_i,request)
  {
    tryNextClass=SCRA(b_i,request,classIndex, 1);
    if (tryNextClass==0) // need to degrade calls of other classes
    {
        B_Tot_A=B_Tot_A + B_A_i;
        classIndex ++;
      }
      else break; // the call is accepted
  }
  if(classIndex > K)
    {
      drop the call;
      d_MCRA_i ++;
    }
} // end of MCRA
```

Table III. Bandwidth expansion algorithm.

```
Expansion Algorithm ()
{
  for (classIndex=1; classIndex ≤ K; classIndex ++)
  {
    while( exist degraded calls of class classIndex to upgrade)
    {
      i=classIndex;
      order the most degraded calls increasingly by bandwidth;
      starting with the most degraded call from above step;
        b_upgrade=b_i,request - b_i,current;
            if (B_A_i ≥ b_upgrade)
          {
              if (deg_nc_i≠0)
                {
                  allocate Bandwidth(b_i,j,b_i,request, b_i,K_i);
                  deg_nc_i --;
                }
              else if (deg_h_i ≠ 0)
              {
                  allocate Bandwidth(b_i,j,b_i,request, b_i,K_i);
                  deg_h_i --;
                }

          }
    }// end of while
  }// end of for
}// end of expansion algorithm
```

currently allocated bandwidth of the corresponding call.

4. Framework Analytical Model

In this section, we show that our framework can be modeled as a multi-dimensional Markov chain. Indeed, the major contribution of this paper is the analytical model. From the steady state distribution of the Markov chain, performance measures of interest, NCBP, HCDP, and DP, can be computed.

4.1. Assumptions

In our system (cell) model, we assume that any arriving call (new or handoff) is never buffered. Thus, an arriving call (new or handoff) is either blocked/dropped or accepted depending on the bandwidth availability to accommodate this call.

For traffic characterization, we assume a simple model from a cell's perspective [24]. New call arrivals and handoff call arrivals of class-i connections are assumed to follow a Poisson process with rates λ_{nc_i} and λ_{h_i}, respectively. The call holding time (CHT) of a class-i call is assumed to follow an exponential distribution with mean $1/\mu_i$.

For mobility characterization, the cell residence time (CRT), that is, the amount of time that a mobile user stays in a cell before handoff, is assumed to follow an exponential distribution with mean $1/h$ [25]. CRT is independent of the service class. Hence, connections in any class follow the same CRT distribution. Note that the parameter h represents the call handoff rate.

The channel occupancy time is the minimum of the CHT and the CRT. As the minimum of two exponentially distributed random variables is also exponentially distributed, the channel occupancy time for new calls and handoff calls for class-i are therefore assumed to be exponentially distributed with means $1/\mu_{nc_i}$ and $1/\mu_{h_i}$, respectively, where $\mu_{nc_i} = \mu_{h_i} = \mu_i + h$.

4.2. Model Analysis

A *state* of the model representing the considered *cell* is determined by the non-negative integer number of ongoing new call and handoff call class-i connections, m_i and n_i, respectively. As a consequence, the state of the cell can be defined as the vector:

$$\boldsymbol{s} = (m_1, m_2, \ldots, m_K, n_1, n_2, \ldots, n_K)$$

Also, let S denote the state space required to represent the system (a cell). Then, $\boldsymbol{s}$ will belong to S if and only if the following conditions are satisfied:

$$\sum_{i=1}^{K}\sum_{j=1}^{m_i} b_{i,\text{assigned}_j} \le t_0 \tag{1}$$

$$\sum_{j=1}^{n_i} b_{i,\text{assigned}_j} \le t_i, \forall i \tag{2}$$

$$\sum_{i=1}^{K}\sum_{j=1}^{m_i} b_{i,\text{assigned}_j} + \sum_{i=1}^{K}\sum_{j=1}^{n_i} b_{i,\text{assigned}_j} \le B \tag{3}$$

The result is a truncated chain. The truncation of this system is still a Markov chain having the same Markov state diagram with the only difference being that some states have been eliminated. A Markov state diagram for a small single traffic class system is shown in Figure 3 where $B=6$, $t_0=4$, and $t_1=6$.

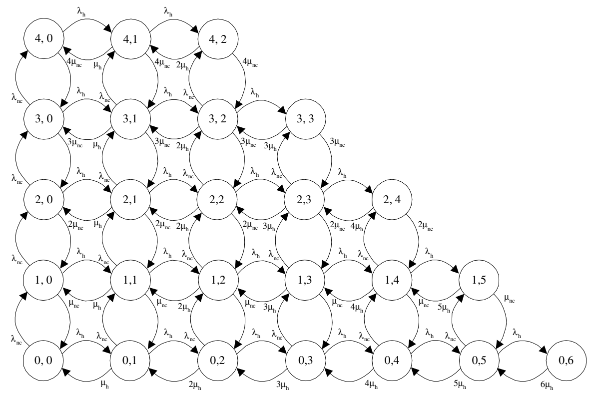

Fig. 3. Markov state diagram for a system with $B=6$, $t_0=4$, and $t_1=6$.

Based on the above assumptions, the system (a cell) follows an M/M/∞ queuing discipline. As aforementioned, we consider two types of traffic, new call and handoff call, which are each further separated into K classes. Thus, our framework can be modeled as a truncation of 2K independent M/M/∞ queues, and admits, therefore, a product-form solution.

From Equations (1–3), we can define the state space of the system, S, as follows:

$$S = \{\boldsymbol{s} = (m_1, m_2, \ldots, m_K, n_1, n_2, \ldots, n_K) \mid \sum_{i=1}^{K}\sum_{j=1}^{m_i} b_{i,\text{assigned}_j} \le t_0 \wedge \sum_{j=1}^{n_i} b_{i,\text{assigned}_j} \le t_i \forall i \wedge \sum_{i=1}^{K}\sum_{j=1}^{m_i} b_{i,\text{assigned}_j} + \sum_{i=1}^{K}\sum_{j=1}^{n_i} b_{i,\text{assigned}_j} \le B\}$$

Also, let ρ_{nc_i} denote the load generated by class-i new connections, and let ρ_{h_i} denote the load generated by class-i handoff connections. Then, $\rho_{nc_i} = \frac{\lambda_{nc_i}}{\mu_{nc_i}}$ and $\rho_{h_i} = \frac{\lambda_{h_i}}{\mu_{h_i}}$.
Define $a(s)$ as, $a(s) = \prod_{i=1}^{K} \frac{\rho_{nc_i}^{m_i}}{m_i!}\frac{\rho_{h_i}^{n_i}}{n_i!}$.
Then the steady state probability that the system is in state s, $p(s)$, is given by: $p(s) = a(s)/G$, where G is a normalization constant given by: $G = \sum_{s \in S} a(s)$.

Let $S_{b_i} \subset S$ denote the set of states at which a class-i new connection is blocked as follows:

$$S_{b_i} = \{s \in S \mid b_{i,\text{request}} + \sum_{i=1}^{K}\sum_{j=1}^{m_i} b_{i,\text{assigned}_j} > t_0 \vee b_{i,\text{request}} + \sum_{i=1}^{K}\sum_{j=1}^{m_i} b_{i,\text{assigned}_j} + \sum_{i=1}^{K}\sum_{j=1}^{n_i} b_{i,\text{assigned}_j} > B\}$$

Then the blocking probability for class-i new connections, P_{block_i}, is given by:

$$P_{\mathrm{block}_i} = \frac{\sum_{s \in S_{b_i}} a(s)}{\mathrm{G}}$$

The output of the bandwidth adaptation algorithm at state s is the number of degraded new calls of class-i, $\deg_{nc_i}(s)$, the number of degraded handoff calls of class-i, $\deg_{h_i}(s)$ and the number of handoff dropped calls of class-i, $d_i(s)$. The value of $\mathrm{d_i}(s)$ depends on which reduction algorithm is executed. If the SCRA is executed, then $d_i(s) = d_{\mathrm{SCRA}_i}$ while if the MCRA is executed, then $d_i(s) = d_{\mathrm{MCRA}_i}$. As a result, the call DP of class-i, $P_{\deg_i}$, is expressed by:

$$P_{\deg_i}(s) = \sum_{s \in S} p(s) \frac{\deg_{nc_i}(s) + \deg_{h_i}(s)}{m_i + n_i}$$

Also, the handoff call dropping probability of a class-i call, P_{drop_i}, is given by:

$$P_{\mathrm{drop}_i}(s) = \sum_{s \in S} p(s) \frac{d_i(s)}{n_i}$$

5. Numerical Results

In this section, we investigate the performance of our adaptive multimedia framework. The connection-level QoS parameters: the NCBP, the HCDP, and the DP were evaluated. We define bandwidth utilization as the ratio of the bandwidth used by completely serviced calls to the total bandwidth capacity. We also compare the bandwidth utilization of our adaptive multimedia framework with the non-adaptive case.

Since the new call arrival rate, mobility (handoff rate, h) and new call threshold (t_0) of each class could significantly affect these parameters, different experiments sets of numerical results are shown for these factors under various settings of the system parameters. Before proceeding with evaluating the performance of our adaptive multimedia framework, we first describe the simulation model that is used in this paper. Then, we validate the accuracy of our analytical model, developed in Section 4, by comparing it with a simulation model.

5.1. Simulation Model

We built a simulation model for the wireless cellular environment. The assumptions for our simulation study are as follows.

- The simulated system consists of seven cells arranged in a circle with identical mobility and traffic conditions. The diameter of each cell is 1 km (micro cellular environment). The BS resides at the center of each cell.
- The total capacity of each cell is $B = 100$ (bbu). It is assumed that 1 (bbu) corresponds to a channel of 64 Kbps rate. Therefore, the total bandwidth is $B = 100 \times 64\,\mathrm{Kbps} = 6.5\,\mathrm{Mbps}$.
- Maximum number of total bandwidth units that are allocated to new connections, $t_0 = 50$ bbu (3.25 Mbps) and to class-i handoff connections, $t_{\mathrm{i}} = 100$ bbu (6.5 Mbps), for $i = 1, 2, 3$.
- Three classes of adaptive multimedia services are considered: voice (class 1), low-quality video (class 2), and high-quality video (class 3). Table IV shows the bandwidth set values and the requested bandwidth value of each class (in kbps).
- Call requests are generated according to a Poisson process with rate λ (calls/second) in each cell. A newly generated connection can appear anywhere in the cell with an equal probability. Note that $\lambda = \lambda_{nc_1} = \lambda_{nc_2} = \lambda_{n_3}$ and the handoff call arrival rate of class-i is assumed to be proportional to the new call arrival rate of class-i by $\lambda_{h_i} = \alpha \lambda_{nc_i}$ for $i = 1, 2, 3$, where α is set to 0.5 in all experiments. The range of offered load (call arrival rate) varies from 0 to 6.0.
- Mobiles can travel in one of eight directions with equal probability. A constant randomly selected speed is assigned to a mobile when it enters a cell either at call initiation or after handoff. The speed is obtained from a uniform probability distribution function ranging between $\mathrm{V_{min}} = 10$ km/h and $\mathrm{V_{max}} = 60$ km/h.
- The lifetime of a class-i connection (i.e., CHT) is exponentially distributed with mean (in seconds) $\mu_1^{-1} = 500$, $\mu_2^{-1} = 700$, and $\mu_3^{-1} = 900$. Also, we assume the CRT is exponentially distributed with mean $h^{-1} = 100$.

Table IV summarizes the simulation parameters, where T is the simulation time.

5.2. Analytical Model Validation

Table V presents a numerical comparison of the NCBP, the HCDP, and the DP obtained from the analytical model (denoted ANA) and simulation (denoted SIM). In this experiment, only the first two classes (class 1 and class 2) in Table IV are studied. The results in Table V show the effect of varying the

Table IV. Simulation parameters.

Service/class index	Bandwidth Set (kbps)	$b_{i,\text{request}}$ (kbps)	h^{-1}(s)	μ_i^{-1}(s)
Voice/1	{64, 80, 96, 112, 128}	96	100	500
Low-quality video/2	{128, 160, 192, 224, 256}	192	100	700
High-quality video/3	{256, 288, 320, 352, 384}	320	100	900

Other parameters

B (Mbps)	T (s)	t_0 (Mbps)	t_1 (Mbps)	t_2 (Mbps)	t_3 (Mbps)
6.5	1000	3.25	6.5	6.5	6.5

Table V. Comparison of analytical and simulation results.

Handoff rate (h)	NCBP				HCDP				DP			
	ANA		SIM		ANA		SIM		ANA		SIM	
	Pb1	Pb2	Pb1	Pb2	Pd1	Pd2	Pd1	Pd2	DP1	DP2	DP1	DP2
0.6	0.147	0.275	0.139	0.273	5.62E-06	1.62E-05	0	0	4.50E-03	5.80E-03	4.44E-03	5.71E-03
1.2	0.062	0.128	0.059	0.126	3.80E-05	8.13E-05	0	0	7.70E-03	9.40E-03	7.65E-03	9.34E-03
1.8	0.032	0.076	0.031	0.072	1.26E-04	2.95E-04	1.22E-04	2.89E-04	1.06E-02	1.24E-02	1.00E-02	1.21E-02
2.4	0.021	0.044	0.021	0.043	3.16E-04	7.94E-04	3.08E-04	7.90E-04	1.29E-02	1.49E-02	1.22E-02	1.44E-02
3	0.017	0.031	0.016	0.029	6.76E-04	1.38E-03	6.66E-04	1.31E-03	1.47E-02	1.65E-02	1.45E-02	1.60E-02
3.6	0.015	0.022	0.013	0.019	1.31E-03	2.18E-03	1.01E-03	2.13E-03	1.62E-02	1.77E-02	1.59E-02	1.72E-02
4.2	0.012	0.017	0.009	0.014	1.58E-03	3.09E-03	1.49E-03	3.07E-03	1.73E-02	1.87E-02	1.69E-02	1.81E-02
4.8	0.011	0.015	0.010	0.013	2.07E-03	3.63E-03	2.00E-03	3.59E-03	1.82E-02	1.93E-02	1.81E-02	1.89E-02
5.4	0.011	0.013	0.009	0.011	2.34E-03	3.89E-03	2.29E-03	3.87E-03	1.87E-02	1.98E-02	1.84E-02	1.97E-02
6	0.010	0.012	0.010	0.011	2.29E-03	4.16E-03	2.28E-03	4.14E-03	1.90E-02	2.02E-02	1.87E-02	2.01E-02

handoff rate on the performance measures where the new call arrival rate $\lambda_{nc_1} = \lambda_{nc_2} = 3$ (call/s). The comparison illustrates that difference between the two models are negligible, and that our analytical model is accurate. Moreover, all the results shown in Figures 4–7 have also been compared with simulation results with negligible differences observed. The simulation results obtained in all experiments have a 95% confidence level with 5% confidence intervals.

5.3. Performance Evaluation

The effect of varying the call arrival rate on the NCBP and the HCDP is shown in Figure 4. From this figure, we observe that the new call blocking probabilities and handoff call dropping probabilities of all classes increase as the call arrival rate increases. However, the HCDP is always lower than the NCBP as a result of the 50 bandwidth units ($B - t_0$) reserved exclusively for the handoff connections. Obviously, by applying the CAC and BAA algorithms, the HCDP is minimized (lower than 0.00048) and therefore satisfies the objective of this work. Furthermore, the results in Figure 4 show that the HCDP increases as the class index increases. This is due to the complete sharing between the handoff connections ($t_1 = t_2 = t_3 = 100$). Therefore, the prioritization among different handoff classes is achieved.

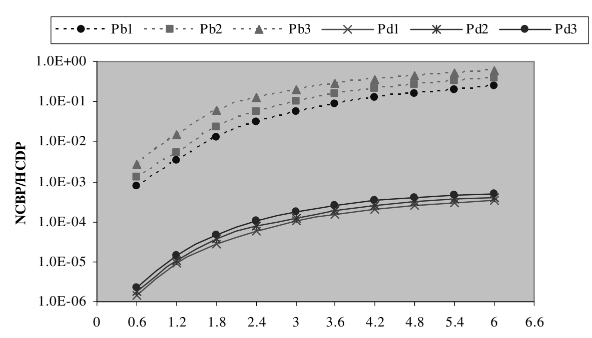

Fig. 4. Effect of varying the call arrival rate ($\lambda = \lambda_{nc_1} = \lambda_{nc_2} = \lambda_{nc_3}$) on the connection-level QoS.

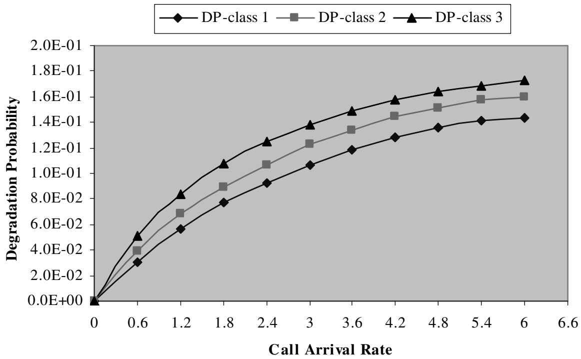

Fig. 5. Degradation probability versus call arrival rate ($\lambda = \lambda_{nc_1} = \lambda_{nc_2} = \lambda_{nc_3}$).

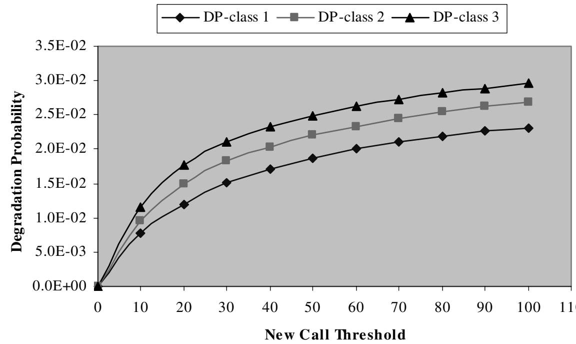

Fig. 7. Degradation probability versus new call threshold.

Figure 5 shows the DP versus the call arrival rate for all classes. As expected, DP increases for all the three evaluated classes as the call arrival rate increases. However, the DP of the different classes can be clearly distinguished, where class 1 has the lowest DP values while class 3 has the highest DP values, $DP_1 < DP_2 < DP_3$. This behavior can be explained as follows: the requested bandwidth of class-i increases as the class index increases and since all the three classes are sharing the same bandwidth threshold, $t_1 = t_2 = t_3 = 100$, there are more calls to share the degradation under class-i as the new arrival rate increases.

In Figure 6, we demonstrate the effect of varying the new call threshold, t_0, on the NCBP and the HCDP where the new call arrival rate $\lambda_{nc_1} = \lambda_{nc_2} = \lambda_{nc_3} = 2$ (calls/s). At low new call threshold values, the NCBP for the three evaluated classes is high, while the HCDP is low. As the threshold value, t_0, increases, the NCBP decreases while the HCDP increases until they converge almost to the same value. This is because the new calls are given more access to the available bandwidth. While the HCDP increases as a result of the higher degree of sharing between the new and the handoff calls, it is always lower than the corresponding NCBP.

Figure 7 illustrates the performance of the new call threshold, t_0, in terms of the DP where the new call arrival rate $\lambda_{nc_1} = \lambda_{nc_2} = \lambda_{nc_3} = 2$ (calls/s). The usage of the BAA allows the system to offer bandwidth for incoming handoff calls. However, in the case where the new call threshold increases, the degree of sharing between the new and handoff calls increases. This results in a slight increase of the HCDP (Figure 6) which means that the system was unable to offer sufficient bandwidth for the incoming handoff calls. Therefore, compared to Figure 5, a fewer number of ongoing calls is degraded.

Figure 8 shows the bandwidth utilization versus the offered load (call arrival rate) for adaptive multimedia

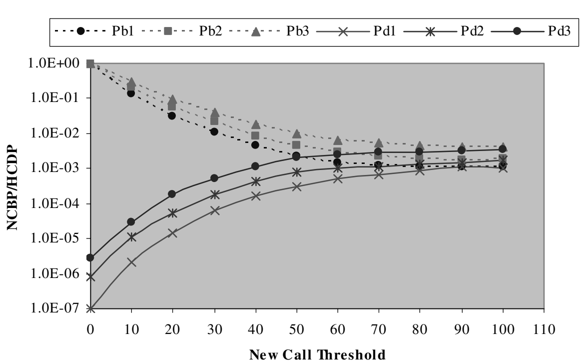

Fig. 6. Effect of varying the new call threshold on the connection-level QoS.

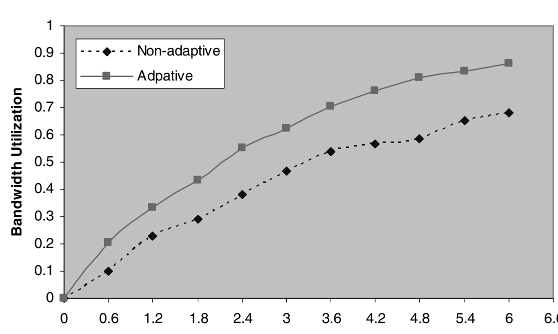

Fig. 8. Effect of varying the call arrival rate ($\lambda = \lambda_{nc_1} = \lambda_{nc_2} = \lambda_{nc_3}$) on bandwidth utilization for non-adaptive/adaptive multimedia.

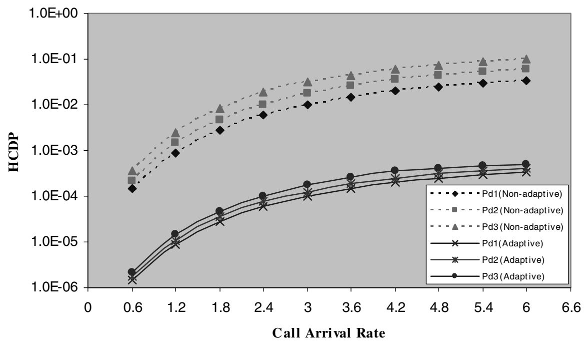

Fig. 9. Effect of varying the call arrival rate ($\lambda = \lambda_{nc_1} = \lambda_{nc_2} = \lambda_{nc_3}$) on HCDP for non-adaptive/adaptive multimedia.

framework as opposed to a non-adaptive multimedia framework. Clearly, the bandwidth utilization of the adaptive multimedia framework outperforms that of the non-adaptive multimedia framework. As the offered load increases, the advantage is more evident. This follows since a highly restrictive threshold-bandwidth allocation policy (non-adaptive case) can result in high call dropping before their completion. Whereas for the adaptive multimedia framework, the usage of the bandwidth adaptation algorithm allows the system to offer services whenever there is sufficient amount of bandwidths by intelligently adjusting bandwidth allocation which results a near zero HCDP. Therefore, more calls are able to complete their services and as a result better effective utilization is obtained.

Figure 9 illustrates the performance of the proposed adaptive multimedia framework and non-adaptive multimedia framework in terms of the HCDP. It is apparent that by applying the CAC and BAA algorithms in the manner described above that the HCDP of the adaptive multimedia framework is minimized (lower than 0.00048) and, therefore, it surpasses the non-adaptive multimedia framework. However, as the traffic load increases, the HCDP increases. The reason is that even the CAC will accept the handoff calls all the time but when the traffic load becomes larger, the threshold-bandwidth allocation policy can still reject some handoff arrivals due to the restrictive threshold occupancy. In the non-adaptive multimedia framework, an incoming handoff connection of class-i is immediately dropped if the current total bandwidth of ongoing class-i connections is greater than t_i, $1 \leq i \leq \quad K$, or no more bandwidth is available in the cell to accommodate this connection. This result in increasing the HCDP of class-i as the call arrival rate increases.

6. Conclusions and Future Work

Wireless access networks in the form of cellular networks, wireless local area networks, and home networks are expecting to support real-time multimedia applications such as streaming media and entertainment, video-conferencing and tele-medicine, and gaming. Different multimedia applications have diverse bandwidth and quality of service (QoS) requirements that need to be guaranteed by wireless access networks. In this paper, a novel adaptive multimedia framework called QoS-AMWA for the wireless cellular network has been presented. The proposed QoS-AMWA system considers the different priorities of the traffic classes and assures a guaranteed QoS for heterogeneous traffic.

Three related components comprise the main building blocks of the QoS-AMWA system: (i) a threshold-based bandwidth allocation policy, (ii) a threshold-based connection admission control algorithm, and (iii) a bandwidth adaptation algorithm. As the mobile user may visit different cells during a connection's lifetime, the system components work together to provide the mobile user a means of 'seamless handoff' communication through the wireless network. Thus, the proposed QoS-AMWA system maintains the QoS of ongoing connections in the network, and at the same time minimizes the new call blocking probability and the handoff call dropping probability. In addition, our framework is cell-oriented, meaning that all its components are implemented on a cell-by-cell basis. It thus has an extremely low complexity, making it practical for real wireless cellular networks.

The framework is modeled as a multi-dimensional Markov chain and a product-form solution is provided. We argue that this work is a powerful and novel contribution in wireless cellular network design as it can be used to derive many performance parameters of interest—NCBP, HCDP, and DP. To the best of our knowledge, our work presents a first attempt towards the analytical modeling of such adaptive framework. Indeed, no such mathematical analysis has been previously reported in the literature.

A simulation model was also developed. Our results show the conformance of the analytical and the simulation results under various traffic and mobility conditions. The overall performance results of our adaptive multimedia framework are very attractive in that the handoff call dropping probability is near zero (negligible) resulting in a stable performance levels during heavy load periods. It is also demonstrated that the proposed framework maintains its bandwidth utilization efficiently.

This work can be extended in several ways. One straightforward extension is to develop an adaptive threshold-based bandwidth allocation policy, which dynamically computes and changes the threshold values based on the traffic and mobility parameters. Thus, satisfying the QoS requirements of all connection classes and maximizing the bandwidth utilization, that is, minimizing the new call blocking probability.

Our framework is attractive as it can be implemented not only in wireless cellular networks, but also in heterogeneous wireless access networks. In heterogeneous wireless access networks, handoffs among networks with different air interfaces takes place during a connection's lifetime. This type of handoff is called a vertical handoff. As each wireless access network has limited bandwidth capacity, we believe with the help of our adaptive framework, the bandwidth of an ongoing call can be dynamically adjusted to adapt to the various network environments. Thus, a seamless vertical handoff that satisfies the QoS requirements of all connection classes can be achieved, while the bandwidth of each network is efficiently utilized.

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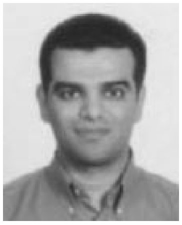
Authors' Biographies



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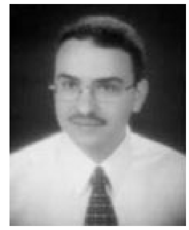

Hossam Hassanein is a leading researcher in the School of Computing at Queen's University in the areas of broadband, wireless and variable topology networks architecture, protocols, control and performance evaluation. Before joining Queen's University in 1999, he worked at the department of Mathematics and Computer Science at Kuwait University (1993–1999) and the department of Electrical and Computer Engineering at the University of Waterloo (1991-1993). Dr. Hassanein obtained his Ph.D. in Computing Science from the University of Alberta in 1990. He is the founder and director of the Telecommunication Research (TR) Lab http://www.cs.queensu.ca/~trl in the School of Computing at Queen's. Dr. Hassanein has more than 150 publications in reputable journals, conferences and workshops in the areas of computer networks and performance evaluation. Dr. Hassanein has served on the program committee of a number international conferences and workshops. He is the editor of the IEEE 802.6e standard for Slot Reuse in Distributed Queue Dual Bus networks in 1992. Dr. Hassanein is the recipient of Communications and Information Technology Ontario (CITO) Champions of Innovation Research award in 2003. He is also the recipient of the 1993 IEE ''Hartee Premium'' best paper award, for the IEE proceedings on Computers.