Call Admission Control for Adaptive Multimedia in Wireless/Mobile Networks

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Abstract

Recently there is a growing interest in adaptive multimedia networking where the bandwidth of an ongoing multimedia call is variable. In this paper we propose a call admission control framework for adaptive multimedia services in wireless/mobile networks. We introduce the degradation period ratio (DPR) QoS parameter, which represents the portion of a call's lifetime that it is allocated bandwidth below a predefined target bandwidth to the whole service time of the call. Based on DPR, we present how to guarantee QoS to users analytically. Simulations reveal that adaptive multimedia framework outperforms the non-adaptive multimedia services.

1 Introduction

With the increase in the demand on wireless/mobile communications and the emergence of bandwidth-intensive multimedia applications, quality of service (QoS) provisioning in wireless/mobile networks is becoming more and more important. A most significant QoS parameter in wireless networks is forced-termination probability – the probability that an accepted call will be forced to terminate before the completion of service. In general, forced-termination of an ongoing call is more unbearable than the blocking of call. So far, call admission control (CAC) has focused on how to block originating calls to reduce the forced-termination probability[1, 2, 3]. However, with the introduction of adaptive multimedia[4, 5], forced-termination probability can be reduced to a negligible level in normal traffic load.

Originally, the concept of adaptive multimedia was introduced in fixed networks. In fixed broadband networks like ATM, 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 paradigm, network congestion can cause fluctuations in the availability of network resources and thereby can result in severe degradation of multimedia services. To overcome this problem, many adaptive multimedia schemes are proposed such as hierarchical encoding[6] and network filters[7] to mitigate the effect of fluctuation in the network resources.

We advocate in this paper that an adaptive multimedia paradigm can play an important role to mitigate the highly-varying resource availability in wireless/mobile networks. Compared to fixed networks, the fluctuation in resource availability in wireless/mobile networks is much more severe and results from two inherent features of such networks : fading and mobility.

The fading in a wireless channel is highly varying with time and spatial dependencies and interference. The second reason for the fluctuation in resource availability is mobility (or equivalently handoff). We assume that the effect of fading can be mitigated by rich-function transmission/reception wireless subsystem. Our adaptive multimedia framework takes into consideration only handoff. That is, adaptive multimedia call changes its bandwidth only when there is a new call arrival, a call completion, or a handoff.

In this framework, we propose a CAC algorithm that can guarantee QoS to users. Also, we introduce a reallocation algorithm that manages the allocation of bandwidth of every call in a cell. Here reallocation means the bandwidth allocation of incoming calls and/or the change of bandwidth of the existing calls in a cell.

In the case of non-adaptive multimedia networking, the incoming handoff call to a given cell will be forced to terminate if there are no available channels. However, in this adaptive multimedia paradigm, the reallocation algorithm reduces the bandwidth of the existing calls in the given cell, thereby freeing bandwidth for additional channels. Also, the bandwidth of the incoming call can be adjusted according to the situation of the given cell. In conclusion, there is a trade-off between having adaptive bandwidth and reducing the forced-termination. That is, the problem of forced-termination is moved to bandwidth adaptation which is more bearable.

Sen et al.[8] have investigated the tradeoff between the carried traffic and bandwidth degradation; however, the degradation mode is restricted to the case in which only one channel of a call is released. Lu et al. have proposed a general adaptation framework in [9] where the bandwidth of a call can be adjusted continuously within its bound. Still, the discrete bit rate of the adaptive multimedia is not taken into consideration.

The rest of this paper is organized as follows. A model of an adaptive multimedia is presented in Section 2. Our CAC framework is proposed in Section 3. Section 4 describes our reallocation algorithm. Simulation results are shown in Section 5. In Section 6, we discuss the forced-termination probability in adaptive multimedia. Finally, we conclude this paper in Section 7.

2 Adaptive Multimedia

2.1 Model Description

According to the adaptive multimedia paradigm, a multimedia call can dynamically change its bandwidth depending on the situation throughout its lifetime. We assume that the bandwidth of a call takes its value from the set $B = \{b_1, b_2, ..., b_n\}$ where $b_i < b_{i+1}$ for $i = 1, ..., n-1$. We also assume that all calls belong to a single class of calls and all of them assume (varying) bandwidth values from the same set B. Here we define bandwidth in terms of units of bandwidth.

Let us take an example of adaptive video stream with four different values of bandwidth. If video stream is encoded by H.263, it can be b_1. An video stream consisting of only MPEG I frame can be b_2. The whole MPEG frame can make b_3. If forward error correction (FEC) code is added into b_3, this video stream can be b_4.

Our aim is to allocate as much bandwidth as possible to every call. Equivalently, a call wants to be allocated with bandwidth b_n (the maximum bandwidth) whenever possible. However, network congestion may occur, in which case a cell cannot accomodate all calls with their maximum bandwidth b_n. In this case, one or more calls should be "degraded" to a lower bandwidth. To choose which calls to degrade and how much bandwidth of the chosen call to change is a role of our reallocation algorithm.

A reallocation algorithm that manages the allocation of bandwidth of each call is necessary in this adaptive multimedia framework. According to the different QoS objective, there can be diverse reallocation algorithms. Here we adopt a simple objective of reallocation algorithm – to minimize the number of calls with lower than predefined "target bandwidth". The target bandwidth will be denoted by b_{tar} and it is assumed to take a predetermined value from the set B. In our single class network, all calls are assumed to have the same b_{tar}.

Our adaptive multimedia framework tries to allocate at least the target bandwidth to every call in a cell. More exactly, the proposed CAC enforces the cell overload probability to be less than predetermined value P_{qos}. The cell overload probability, P_{CO}, represents the steady state probability that there is at least one call with lower than the target bandwidth.

2.2 Degradation Period Ratio

With our adaptive framework, we could ignore the forced-termination probability in normal traffic load, as it can be made practically zero (see Section 6). Instead, we define another QoS parameter : degradation period ratio. The degradation period ratio (DPR) represents the portion of a call's lifetime that a call is allocated bandwidth that is lower than the target bandwidth. For example, if a call's DPR is 0.5, the period while the call is allocated lower than the target bandwidth is half of the call's lifetime.

We define a state as the number of calls in each cell at each instant. A call may experience a number of states throughout its lifetime. The residence time in a state represents the time interval between every reallocation point. As mentioned before, reallocation happens whenever there is: (1) a new call arrival, (2) a call completion, or (3) a handoff. We assume that the time between every reallocation follows exponential distribution with rate r. Here r is a state transition rate which reflects how fast the state of the system (a cell) will move to another state.

The state transition rate is a function of the new call arrival rate λ, the handoff call arrrival rate λ_h, the call service rate μ, and the handoff rate h. Here the handoff rate means how fast a call will handoff and takes the inverse of cell residence time of a call. As is usual in the literature, we assume that call arrivals to each cell form a Poisson process with mean λ. Also, call service time and cell residence time is assumed to follow exponential distributions with mean $1/\mu$ and $1/h$ respectively. Then, the rate r can be calculated from the effective new call arrival rate, the handoff call arrival rate (see [2]), the call service rate, and the handoff rate as in Equation 1.

$$r = \lambda(1 - P_B) + \lambda_h + E[i]\mu + E[i]h \tag{1}$$

In Equation 1, P_B is the call blocking probability and $E[i]$ denotes the average number of calls in each cell. Furthermore, according to [2], λ_h can be expressed by Equation 2 where P_{HD} is the handoff dropping probability.

$$\lambda_h = \lambda(1 - P_B)\frac{h}{\mu + hP_{HD}} \tag{2}$$

An accepted call will experience a number of state transitions as (1) a new call is accepted, (2) a call is handed-off from adjacent cells to a given cell, (3) calls are terminated, or (4) a call in a given cell is handed-off to adjacent cells. Using r, we can calculate the probability distribution of how many states a call will experience throughout its lifetime. Suppose that the call resides in the i-th state for a time period t_i $(i = 1, 2, ...)$. Then, the time between state change follow the exponential distribution with rate r

$$P[t_i \le t] = 1 - e^{-rt} \tag{3}$$

Let us denote the call service time by the random variable τ and the number of state transitions experienced by a call by the random variable K. Then, the probability of a k-state call (a call will experience k states during its lifetime) is given by:

For $k = 1$,

$$\begin{aligned} P[K = k] &= P[\tau < t_1] \\ &= \int_{t_1=0}^{\infty}\int_{\tau=0}^{t_1} \mu e^{-\mu\tau} r e^{-rt} d\tau dt_1 \\ &= \frac{\mu}{\mu + r} \end{aligned} \tag{4}$$

For $k \ge 2$,

$$\begin{aligned} & P[K = k] \\ &= P[\sum_{i=1}^{k-1} t_i < \tau \le \sum_{i=1}^{k} t_i] \\ &= \int_0^{\infty}\int_0^{\infty} \ldots \int_0^{\infty}\int_{\tau=t_{k-1}}^{t_k} \mu e^{-\mu\tau} r e^{-rt_k} \ldots r e^{rt_2} r e^{rt_1} \\ & \quad d\tau dt_k \ldots dt_2 dt_1 \\ &= \left(\frac{r}{\mu + r}\right)^{k-1} \frac{\mu}{\mu + r} \end{aligned} \tag{5}$$

In general, probability of a k-state call is $(\frac{r}{\mu+r})^{k-1}\frac{\mu}{\mu+r}$ for $k = 1, 2, \ldots$.

At each state, a call will be allocated with bandwidth which may be lower than the target bandwidth or not. Let P_D be the "degradation probability" that a call will be allocated with bandwidth lower than the target bandwidth in a state. If we assume that the state transition process of a call is a discrete-time Markov process and every state is independent of each other, then, for a k-state call, the number of states with lower than target bandwidth will follow the binomial distribution $B(k, P_D)$. Let us denote X_k as the random variable of the above distribution which represents the number of degraded states of a k-state call. Furthermore, if X_k is multiplied by $1/k$, it expresses the DPR of a k-state call obviously. Then we can calculate the final DPR as in Equation 6. Here X represents the expected DPR of a call.

$$X = \sum_{k=1}^{\infty} P[K = k] X_k \frac{1}{k} \tag{6}$$

Equation 6 takes the form of summation of binomial distributions. In [10], the sum of random variables following binomial distribution can be approximated by normal distribution $N(m, \sigma^2)$. Accordingly, m and σ can be approximated by Equation 7 and 8 respectively where $c_k = P[K = k]\frac{1}{k}$.

$$m = \sum_{k=1}^{\infty} c_k k P_D \tag{7}$$

$$\sigma^2 = \sum_{k=1}^{\infty} {c_k}^2 k P_D (1 - P_D) \tag{8}$$

Therefore, $P[X \geq a]$ can be calculated by tail function[11] of standard normal distribution : $Q(\frac{a-m}{\sigma})$.

We think that the ultimate QoS to users in adaptive multimedia framework can be expressed by

$$P[X \geq a] \leq \beta \tag{9}$$

In a standard normal distribution, we can find an α such that $P[Z \geq \alpha] = \beta$. Here Z is a random variable of a standard normal distribution which is derived from Equation 6. Finally, what we have to do is to find the maximum P_D that satisfies Equation 10, which in turn satisfies Equation 9. Here a is a given QoS value, and both m and σ are a function of P_D from Equation 6.

$$\frac{a-m}{\sigma} \geq \alpha \tag{10}$$

3 Call Admission Control

Now the problem is how CAC can guarantee QoS with respect to DPR to users. We basically adopt the CAC algorithm proposed in [1]. In this algorithm, the number of calls in the given cell after estimation period T is expected probablistically. That is, the number of handoff calls which is expected to hand-off from the neighboring cells before T is estimated. Also, the number of the remaining calls in the given cell until T is estimated. Recall that the cell overload probability as the probability that there are at least one call with lower than target bandwidth in a cell. If the cell overload probability, P_{CO}, is expected to be greater than predetermined QoS value (P_{qos}) after T time units, then the originating call is rejected. Thereby our CAC can enforce P_{CO} to be less than P_{qos}.

Thus, our CAC is based on P_{CO}. However, QoS in adaptive multimedia is guaranteed by means of P_D as duscussed in Section 2. Therefore, we should calculate the ratio of P_{CO} to P_D by

$$ratio = \frac{P_{CO}}{P_D} = \frac{\sum_{i=N_{th}}^{N_{max}} \pi(i)}{\sum_{i=N_{th}}^{N_{max}} \pi(i)\frac{y(i)}{i}} \tag{11}$$

Here N_{th} is the minimum number of calls where there is at least a call with less than b_{tar} in our adaptive framework. Also, N_{max} is the maximum number of calls in a cell and is calculated by $\lfloor \frac{C}{b_{min}} \rfloor$ where C is a total bandwidth capacity of a cell. The parameter $y(i)$ denotes the number of calls with lower than the target bandwidth when there are totally i calls in the given cell. Note that the value of $y(i)$ is uniquely determined for every state, which can be calculated by finding out the minimum $y(i)$ that satisfies the following equation:

$$C = [i - y(i)]b_{tar} + y(i)b_{min} + r, \ (0 \leq r < b_{tar} - b_{min}) \tag{12}$$

Finally, after calculating the *ratio*, we can compute the maximum P_{CO} value that satisfies the QoS with respect to DPR by multiplying *ratio* by P_D. Recall that P_{qos} is the upper bound value of P_{CO} in our CAC algorithm. Therefore, P_{qos} equals the maximum P_{CO} as calculated above.

Table 1: Notation for Reallocation Algorithm

b_{tar}	target bandwidth
b_{min}	minimum bandwidth (b_1)
b_{max}	maximum bandwidth (b_n)
B_A	available bandwidth in the given cell
B_T	amount of squeezable bandwidth by changing all calls with more than b_{tar} into calls with b_{tar}
B_M	amount of squeezable bandwidth by changing all calls with more than b_{min} into calls with b_{min}

4 Reallocation Algorithm

Our reallocation algorithm (RA) tries to minimize the number of calls with lower than target bandwidth. Equivalently, it tries to maximize the number of calls with equal to or more than b_{tar} at any instant.

There are two kinds of RAs in each cell: for reduction and for expansion. RA for reduction applies to the case where a new call or an incoming handoff call arrives in the given cell. Depending on the situation, RA allocates the suitable bandwidth to the incoming call (new call or handoff call) and reallocates the bandwidth of the existing calls, if necessary. RA for expansion tries to expand the calls with lower than target bandwidth to b_{tar} or more when there is a outgoing handoff call or call completion in the given cell. The notation of our algorithm is summarized in Table 1.

The description of RA for reduction when a call (new or handoff) arrives in the given cell is decribed below. There are six cases in RA for reduction. Below, the operation ReduceT(b_{wanted}) squeezes the calls with more than b_{tar} to b_{tar} until B_A exceeds the b_{wanted}. Similarly, ReduceM(b_{wanted})

squeezes the calls with more than b_{min} to b_{min} until B_A exceeds the b_{wanted}.

1) if ($B_A \geq b_{tar}$),
   assign maximum b_i to the incoming call
   ($b_i \leq B_A, b_{tar} \leq b_i \leq b_{max}$)

2) else if ($B_A < b_{tar}$ and $B_A + B_T \geq b_{tar}$),
   ReduceT(b_{tar})
   assign maximum b_i to the incoming call
   ($b_i \leq B_A, b_{tar} \leq b_i \leq b_{max}$)

3) else if ($B_A \geq b_{min}$ and $B_A + B_T < b_{tar}$),
   assign maximum b_i to the incoming call
   ($b_i \leq B_A, b_{min} \leq b_i < b_{tar}$)

4) else if ($B_A < b_{min}$ and $B_A + B_T \geq b_{min}$),
   ReduceT(b_{min})
   assign maximum b_i to the incoming call
   ($b_i \leq B_A, b_{min} \leq b_i < b_{tar}$)

5) else if ($B_A < b_{min}$ and $B_A + B_M \geq b_{min}$),
   ReduceM(b_{min})
   assign maximum b_i to the incoming call
   ($b_i \leq B_A, b_{min} \leq b_i < b_{tar}$)

6) else drop/block the call

When a call leaves the cell, the B_A will increase. This change in B_A may enable one or more calls to expand their bandwidth. The RA for expansion follows. Below, b_{cur} denotes the currently allocated bandwidth of the corresponding call and b_{req} denotes the required bandwidth to upgrade the bandwidth of the corresponding call.

1) order the remaining calls by decreasing DPR

2) for each call whose bandwidth is less than b_{tar}

   a) $b_{req} = b_{tar} - b_{cur}$

   b) if $B_A >= b_{req}$,
      assign maximum b_i to the call
      ($b_i \leq B_A, b_{tar} \leq b_i \leq b_{max}$)

   c) else if $B_A < b_{req}$ and $B_A + B_T \geq b_{req}$,
      ReduceT(b_{req})
      assign maximum b_i to the call
      ($b_i \leq B_A, b_{tar} \leq b_i \leq b_{max}$)

3) go to next call

5 Numerical Results

In this section, we present the performance of our CAC using an adaptive framework and compare it to the same CAC in the fixed multimedia paradigm where the bandwidth of on-going call is fixed throughout its lifetime. The bandwidth of a call in fixed multimedia paradigm is 10 (bandwidth units) in this simulation experiment. Whereas, there are 3 different bandwidth values in adaptive multimedia paradigm (see Table 2). In the fixed paradigm, P_{qos} represents the upper bound of cell overload probability; that is, the probability that the number of expected calls after T is greater than the maximum allowable number of calls in a cell.

The experimental results here are based on the simulation of a system consisting of 10 cells arranged on a circle. The probability of a user handing off to any adjacent cell is equally likely. The system parameters for our simulation are shown in Table 2.

Table 2: Simulation Parameters

C	200
b_1	1 (b_{min})
b_2	9 (b_{tar})
b_3	10 (b_{max})
$1/\mu$	500 seconds
$1/h$	100 seconds
T	20 seconds

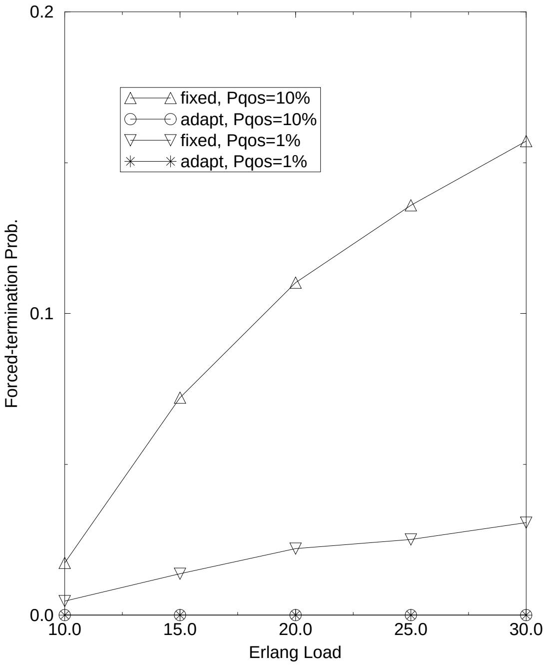

Figure 1: Forced-termination Probability

In these experiments, our performance measures are the forced-termination probability P_F and call blocking probability P_B. Figure 1 shows that forced-termination probability is always 0 in our CAC in adaptive multimedia, which highlights the advantage of adaptive multimedia.

In Figure 2, the call blocking probability of CAC in both paradigms is shown accordingly as load (in Erlangs) increases. When the cell overload probability is relatively high (P_{qos}=10%), the call blocking probability of fixed multimedia is lower than that of adaptive multimedia. It is mainly because there are already many calls in each cell in the adaptive multimedia paradigm as there is no forced-termination. However, in the case that P_{qos} is relatively low (P_{qos}=1%), adaptive multimedia shows better performance. We think that this phenomenon results from two facts : 1) when P_{qos} is low, the number of calls in each cell is already suitably low and 2) b_{tar} in adaptive multimedia is less than the bandwidth of a fixed multimedia call.

We define another performance measure in this paper : utilization. Here utilization represents the ratio of the bandwidth used by completely serviced calls to the total bandwidth capacity. If a call is forced-terminated, the bandwidth used by the call is not taken into account. Figure 3 shows that utilization of adaptive multimedia outperforms that of fixed multimedia.

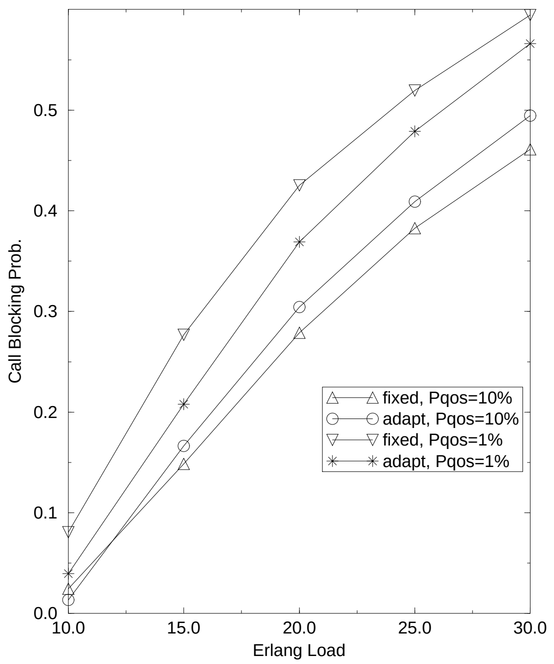

Figure 2: Call Blocking Probability

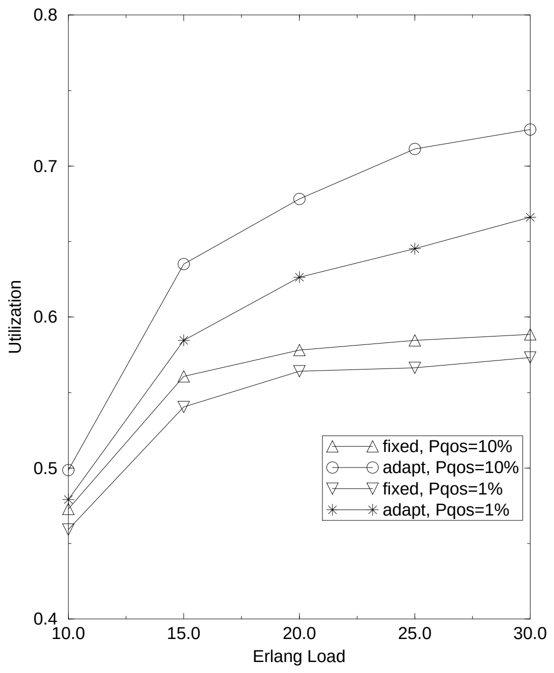

Figure 3: Utilization

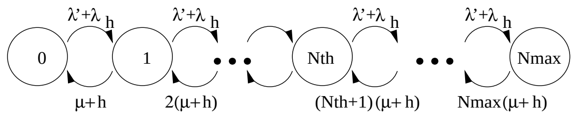

Figure 4: State Transition Diagram

6 Forced-termination Probability

In this section, we discuss the forced-termination probability in adaptive multimedia paradigm. According to [2, 12], the forced-termination probability is directly proportional to handoff dropping probability. Thus we consider handoff dropping probability instead of forced-termination probability. Here handoff dropping probability means the probability that an attempt of handoff will fail.

As mentioned earlier, the handoff dropping probability in our adaptive framework is negligible in normal case. However, theoretically it is possible that a handoff fails if a given cell is full of calls with b_{min} ($B_A = 0$) and an incoming handoff call arrives. That is, in our adaptive framework, the handoff dropping probability equals the steady state probability of $\pi(N_{max})$. The state-transition diagram in a cell can be described by Figure 4.

Recall that λ_h denotes the handoff call arrival rate into a cell and can be expressed by Equation 2. Furthermore, λ' denotes the reduced new call arrival rate. It can be approximated by fixed point approximation[13, 14] as follows.

$$\lambda' = \lambda(1 - P_B) \tag{13}$$

Here P_B represents the probability that a newly arriving call will be blocked; it can be calculated by considering steady state probability of three cells (the given cell and two adjacent cells) in one dimensional cellular network. Let the function $test(i,j,k)$ be the CAC function which returns 0 when the newly arriving call should be rejected (see [1]). Then, the equation for P_B is expressed by

$$P_B = \sum_{i=0}^{N_{max}} \sum_{j=0}^{N_{max}} \sum_{k=0}^{N_{max}} state[i] * state[j] * state[k]$$
$$where\ test(i,j,k) = 0$$

Accordingly, we can figure out the steady state probabilities of a cell and thereby calculate the handoff dropping probability by

$$P_{HD} = \pi(N_{max}) = \frac{\prod_{i=0}^{N_{max}-1} \frac{\lambda' + \lambda_h}{(i+1)(\mu+h)}}{\sum_{j=0}^{N_{max}-1} \prod_{i=0}^{j} \frac{\lambda' + \lambda_h}{(i+1)(\mu+h)}} \tag{14}$$

Similar to the fixed point approximation in [13, 14], we can solve the above equations by repeated substitution.

7 Conclusion

It is anticipated that multimedia applications with adaptive framework where the bandwidth of an ongoing call may vary will become widespread. Although the forced-termination probability can be ignored in this adaptive multimedia paradigm, a CAC to guarantee QoS to users is highly required. In this paper, we have proposed a novel QoS parameter : degradation period ratio (DPR). We have discussed how to guarantee QoS on DPR to users. Simulation is conducted to highlight the advantage of adaptive multimedia in comparison with non-adaptive multimedia. In this study, only a single class of adaptive multimedia has been investigated. In the future we will extend our CAC scheme for multiple classes where there are various adaptive multimedia streams in an integrated services network.

References

[1] Mahmoud Naghshineh and Mischa Schwarz, "Distributed Call Admission Control in Mobile/Wireless Networks," IEEE PIMRC '95, pp. 289-293, 1995.

[2] Yi-bing Lin et al., "Queueing Priority Channel Assignment Strategies for PCS Hand-off and Initial Access," IEEE Trans. on Vehicular Technology, Vol. 43, No. 3, pp.704-712, Aug., 1994.

[3] Arak Sutivong and J. M. Peha, "Novel Heuristics for Call Admission Control in Cellular Systems," IEEE ICUPC '97, pp. 129-133, 1997.

[4] Mahmoud Naghshineh and Marc Willebeek-LeMair, "End-to-End QoS Provisionging in Multimedia Wireless/Mobile Networks Using an Adaptive Framework," IEEE Communications Magazine, Vol. 35, No. 11, pp. 72-81, Nov., 1997.

[5] A. Alwan et al., "Adaptive Mobile Multimedia Networks," IEEE Communications Magazine, Vol. 34, No. 4, pp. 34-51, Apr., 1996.

[6] Nachum Shacham, "Multipoint Communication By Hierarchically Encoded Data," IEEE INFOCOM '92, pp. 2107-2114, 1992.

[7] Nicholas Yeadon, "Filters: QoS Support Mechanisms for Multipeer Communications," IEEE Journal on Selected Areacs in Communications, Vol. 14, No. 7, pp. 1245-1262, Sep., 1996.

[8] Sanjoy Sen et al., "Quality of Service Degradation Strategies in Multimedia Wireless Networks," IEEE VTC '98, pp. 1884-1888, Ottawa, 1998.

[9] Songwu Lu et al., "Adaptive Service in Mobile Computing Environments," IWQOS '97, New York, 1997.

[10] R. Howard, "Dynamic Probabilistic Systems, Volume 1 : Markov Models," John Wiley and Sons, 1971.

[11] Alberto Leon-Garcia, "Probability and Random Processes for Eletrical Engineering," Addison Wesley, 1994.

[12] Stephen S. Rappaport, "The Multiple-Call Hand-off Problem in High-Capacity Cellular Communications Systems," IEEE Trans. on Vehicular Technology, Vol. 40, No. 3, pp. 546-557, Aug. 1991.

[13] F. P. Kelly, "Fixed Point Models of Loss Networks," Journal of Australian Mathemathical Society, Vol. B, No. 31, pp. 204-218, 1989.

[14] D. L. Pallant, "A Reduced Load Approximation for Cellular Mobile Networks Including Handovers," Australia Telecommunications Research, Vol. 26, No. 2, pp. 21-29, 1992.