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Call Admission Control using Road Maps over Wireless Cellular Networks

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Abstract

Currently, wireless communications is one of the fields with the largest activity in the field of telecommunications. Although the respective research started in the 1960s, the past decade included a burst of research activity on this field. As the popularity of wireless technologies increases, the scientific community faces new challenges. The unified transmission of voice, data and video traffic over a wireless network is one of these.

In this thesis, we study three mechanisms which control the admission of users moving from cell to cell in a wireless network. The Call Admission Control (CAC) mechanism is defined as a sequence of activities that are realized from the network in order to check if a user's request, to use a specific service of the network, can be admitted or not. This request will be admitted if the desirable level of quality of service (QoS) can be accomplished without causing violation on the QoS that existing users enjoy. The design and simulation of the CAC mechanism were realized for the simultaneous use of the channel both from voice and e-mail users and video users downloading movies encoded with MPEG4. The CAC mechanism aims to maximize channel throughput without allowing network congestion which would lead to the violation of the users' QoS requirements. This is difficult to achieve because of the contradictory nature of the different types of traffic. Our work is focused on the case when there are no adequate traffic models that estimate the behavior of video traffic, and we especially consider the case of significant handoff loads.

Our results show the advantages and disadvantages of all three proposed CAC mechanisms, and reaches some significant conclusions on when each type of mechanism should be used.

List of Abbreviations

3G	Third Generation Communication Systems
CAC	Call Admission Control
FGC	Fractional Guard Channel
GoP	Group of Pictures
HQ	High Quality
LQ	Low Quality
MAC	Medium Access Control
MQ	Medium Quality
MSC	Mobile Switching Center
QoE	Quality of Experience
QoS	Quality of Service

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1. Introduction

The percentage of people that make use of wireless multimedia services, is continuously increasing, hence increasing the need to satisfy the users' QoS requirements.

There are two techniques which are used widely for the control of traffic that users transmit and receive through the wireless network: Call Admission Control and Traffic Policing. The second one checks if a user respects its contract with the provider in terms of the transmitted traffic.

Call Admission Control (CAC) is defined as a sequence of activities which are realized by the network in order to check if a user's request to use a specific service can be accepted or not. A new request becomes acceptable when the desirable QoS can be achieved for this service without causing violation on the QoS that other users enjoy.

CAC mechanisms aim to avoid congestion in the network while ensuring the maximum possible use of its resources. The above goals become even more difficult to achieve due to the fact that cellular providers have significantly reduced the size of wireless cells (picocells) [1], in order to be able to offer enough bandwidth to multimedia users. In this cellular structure, due to the mobility of users, the mobility conditions in a cell can change very rapidly. Furthermore, when users move far from the coverage area of a cell, the point of connection changes (handoff), while users keep on expecting the same QoS. So, there is a large challenge to plan simple CAC mechanisms which will improve the performance of the network in the case of highly mobile usres.

CAC mechanisms for wired and wireless networks have been studied thoroughly in the literature and the relevant approaches can be broadly classified as follows.

1.1 First Classification of CAC Mechanisms

This classification concerns the type of services that are offered by the network and classifies the services into three subcategories [4]:

The traditional service model is defined in [5, 6], as guaranteed service. CAC algorithms for the guaranteed service use the a priori features of sources for the computation of the worst-case scenario, i.e, the computation of the maximum bandwidth that existing and newly arriving users will need. The CAC mechanism ensures, in this case, that a user will be accepted only if its maximum requirements can be guaranteed. When traffic is bursty, guaranteed service leads unavoidably to very low exploitation of the network resources [7].

Probabilistic service which is described in [8], doesn't anticipate the worst scenario, but guarantees a limit on the rate of packet dropping, which is based on the statistical features of the traffic load. The typical method for this kind of service is implemented when an equivalent bandwidth, bigger than the average rate but smaller than the peak rate, is associated each source. In most cases equivalent bandwidth is computed based on a statistical model [9]. However, in case that an accurate enough statistical model for each user is absent, the guarantee of the equivalent bandwidth leads to an important overestimation of the real demands of traffic sources and as a result to a conservative CAC mechanism that can't use the available bandwidth efficiency [11-15]. Results in [16] also showed that a CAC mechanism based on the computation of equivalent bandwidth in [10] for video transmission is so conservative that it achieves a smaller bandwidth utilization than a simple traffic policing mechanism which is also suggested in [16]. However, traffic policing is not enough by itself, as it can only correct problems that are caused by traffic contracts' breaking, without being able to prevent their appearance.

In case of applications which are tolerant to occasional violations of their delay limit, a third kind of service is suggested in [4], which is called predictive service. The measurement-based CAC mechanism, which proposed in [4] uses the a priori characteristics of sources only for incoming flows and flows that have very recently been admitted. Flows which have been admitted longer are approximated by measurements of their transmitted traffic. This approximation can't ever provide accurate delay limits which are required for the guaranteed or probabilistic service, so approximations based on measurements can only be used when the network offers loose commitments to the user.

In this thesis we will propose and study, in the next chapters, a CAC mechanism which belongs in the category of probabilistic service mechanisms.

1.2 Second Classification of CAC Mechanisms

The second classification splits CAC mechanisms into two large categories based on their handoff priority policy [17]:

A. Guard Channel (GC) mechanisms

In this type of mechanisms, some of the channels are reserved for handoff calls. Four subcategories of GC CAC mechanisms have been proposed. These are:

a. The cutoff priority mechanism, which reserves a percentage of channel bandwidth for handoff calls. When a channel is released, it is returned to the channel pool [18].

b. The fractional guard channel mechanism, which admits a call with a specific probability that depends on the number of occupied channels [19].
c. The rigid division-based mechanism [20], where all channels of a cell are separated into two groups, one for the common use of all calls and the other for handoff calls.
d. The new call bounding mechanism [21], which imposes a threshold on the number of calls that are admitted in the cell.

B. Queuing Priority (QP) mechanisms

In this type of mechanisms, calls become admitted if available channels exist. Depending on the approach, either new calls are blocked and handoff calls are queued [22] or vice versa [23] or all calls are queued and the queue is reordered based on certain priorities [24].

In our thesis we have initially studied a CAC mechanism that falls in the category of Queuing Priority mechanisms based on its handoff priority policy. In our mechanism, no channels are reserved for handoff calls. Every time that a handoff call needs to be served, our mechanism admits the call if it is able to guarantee the necessary bandwidth for the handoff user. Our mechanism treats new calls generated within the cell in the same manner.

In our work, we also propose a hybrid CAC mechanism, which works as a QP scheme for all traffic types except video, for which we propose a fractional guard channel (FGC) mechanism. Our work continues the study of Call Admission Control for cellular networks which was recently presented in [41].

2. System Model

In a picocell, spatially dispersed terminals share a wireless channel that connects them to a fixed base station (BS). The BS allocates channel resources and provides feedback information. The BS also serves as an interface with the Mobile Switching Center (MSC). The MSC provides access to the fixed network infrastructure. Our study focuses on the downlink channel, i.e., the transmission from the BS to the wireless terminals. Time in the downlink channel is divided into time frames of equal length. Each frame has a duration of 12 ms and can accommodate 566 information time slots. The number of time slots is derived by the channel transmission rate which is 20 Mbps. Each information slot accommodates exactly one packet of fixed size, equal to 53 bytes of which 48 bytes contain information (payload) and the remaining 5 bytes correspond to the header. The use of this packet size is indicative (it was chosen because it has been used widely in the literature) and does not affect the results of our study.

In this work, we focused on three types of traffic: video, data and voice. In the rest of Section 2 we will present each type of traffic and the respective models used.

In [25] it was shown that the use of a small portion of the channel bandwidth (less than 3%) for requests usually suffices for the high performance of the system. We use the same assumption in our work.

2.1 Channel Error Model

The most popular and widely used error model in wireless channels is the Gilbert-Elliot model [26, 27]. It is a two state Markovian model, where the channel alternates between a “good” and “bad” state. By “good state” we refer to the absence of noise in the channel and by “bad state” we refer to the existence of noise in the channel.

Transmission of a packet is considered successful in the “good state”, otherwise it fails.

The probability of moving to the “bad state” given that the channel is in “good state” is $P_{good\text{-}bad} = 0.0000086$.

The probability of moving to the “good state” given that the channel is in “bad state” is $P_{bad\text{-}good} = 0.172$.

The steady-state probability for the channel to be in the “good state” is $P_{good} = 0.99995$ and the steady-state probability for the channel to be in the “bad state” is $P_{bad} = 0.00005$.

The reason that we chose this value of P_{bad} is that we assume that video users have very strict QoS requirements and cannot lose over 0.01% of their packets. As a result, if P_{bad} becomes bigger than 0.0001, then the QoS requirement of video users will be violated automatically even if the channel bandwidth is used ideally.

As we mentioned, in the case of noise any transmission fails; this leads to voice or video packet dropping, and to the retransmission of data packets that are corrupted by noise.

2.2 Voice Traffic Model

1. Voice terminals are equipped with a voice activity detector (VAD) [1]. Voice sources follow an alternating pattern of talkspurts and silence periods (on and off), and the output of the voice activity detector is modeled by a two-state discrete time Markov chain (Figure 1). The mean talkspurt duration is 1 s and the mean silence duration is 1.35 s.

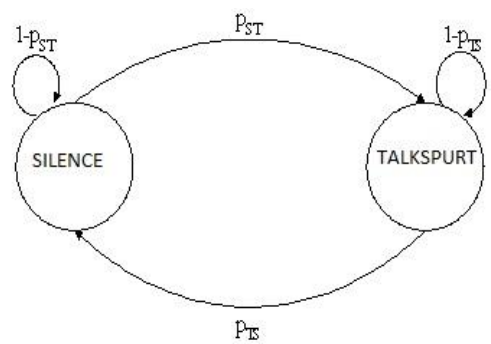


Figure 1. Markovian model for the activity of voice sources

2. The number of active voice terminals N in the system is assumed to be constant over the period of interest.
3. All of the voice source transitions (e.g., talk to silence) occur at the frame boundaries. This assumption is reasonably accurate, taking into consideration that the duration of a frame is equal to 12 ms here, while the average duration of the talkspurt and silence periods exceeds 1 s.
4. The allowed voice packet dropping probability is set to 0.01, and the maximum transmission delay for voice packets is set to 40 ms [28, 29].
5. Reserved slots are deallocated immediately. This implies that a voice terminal holding a reservation signals the BS upon the completion of its talkspurt (the same assumption is made for slots reserved by data terminals).
6. The speech codec rate is 32 kbps. By adding the header length, we get a transmission rate of each voice terminal equal to 35.33 kbps. Each voice terminal is in talkspurt for

42.5% (1/2.35 sec) of the total time that it is active (on average). Therefore, the average transmission rate of a terminal is equal to 35.33*0.425 = 15.015 kbps.

Hence, our CAC mechanism assumes that a voice terminal, which sends a connection request to the network, needs to be assigned a rate of 15.015 kbps.

2.3 Data Model

We adopt the data traffic model based on statistics collected on e-mail usage from the Finnish University and Research Network (FUNET) [30]. The probability distribution function f(x) for the length of the e-mail data messages of this model was found to be well approximated by the Cauchy (0.8, 1) distribution. The packet interarrival time distribution for the FUNET model is exponential, and the average e-mail data message length has been found (by simulation) to be 80 packets. A quite strict (considering the nature of this type of traffic) upper bound is set on the average e-mail transmission delay, equal to 5 s. The reason for this strict bound is that mobile users sending e-mails will probably be quite demanding in their QoS requirements, as they will expect service times similar to those of short message service traffic.

2.4 Video Model

The MPEG initiated the new MPEG-4 standards in 1993 with the goal of developing algorithms and tools for high efficiency coding and representation of audio and video data to meet the challenges of video conferencing applications. The standards were initially restricted to low bit rate applications but were subsequently expanded to include a wider range of multimedia applications and bit rates. The most important addition to the standards was the ability to represent a scene as a set of audiovisual objects. The MPEG-4 standards differ from the MPEG-1 and MPEG-2 standards in that they are not optimized for a particular application but integrate the encoding, multiplexing, and presentation tools required to support a wide range of multimedia information and applications. In addition to providing efficient audio and video encoding, the MPEG-4 standards include such features as the ability to represent audio, video, images, graphics, text, etc. as separate objects, and the ability to multiplex and synchronize these objects to form scenes. Support is also included for error resilience over wireless links, the coding of arbitrarily shaped video objects, and content-based interactivity such as the ability to randomly access and manipulate objects in a video scene.

We use three statistics of actual MPEG-4 streams from [31, 32]. Video sources have Weibull-distributed sessions with a mean duration of 10 minutes and variance equal to 5 minutes (in [33] it is noted that users of cellular video applications will use the service for either three to five minutes or twenty to thirty minutes).

Video packets need to be transmitted within 40 ms (before the arrival of the next video frame), otherwise they are dropped. The allowed video packet dropping probability is equal to 0.01% [29].

We used six traces (Silence of the Lambs, Star Wars IV, Die Hard III, Robin Hood, Simpsons, Formula 1, which correspond to the numbers 1 to 6) for the simulation of video users. For each movie we studied three types of quality: High Quality (HQ), Medium Quality (MQ) and Low Quality (LQ). The statistical parameters used were the mean, the variance and the peak bit rate. The respective values are shown in Table 1.Table 2 presents the number of frames, the size and the duration of each movie.

Movies	**Average Bit Rate (Mbps)**			**Variance Bit Rate (Mbps)**			**Peak Bit Rate (Mbps)**		
	HQ	**MQ**	**LQ**	**HQ**	**MQ**	**LQ**	**HQ**	**MQ**	**LQ**
1	0,58	0,18	0,11	0,208	0,044	0,031	4,4	2,4	2,3
2	0,28	0,078	0,053	0,033	0,008	0,008	1,9	0,94	0,94
3	0,7	0,25	0,15	0,196	0,044	0,034	3,4	1,6	1,6
4	0,91	0,33	0,19	0,212	0,052	0,052	3,3	2	2
5	1,3	0,42	0,23	0,288	0,104	0,104	8,8	4,4	4
6	0,84	0,29	0,17	0,124	0,034	0,038	2,9	1,4	1,4

Table 1. Statistical values of the traces

Movies	**Number of Frames**			**File Size (MB)**			**Run Time (sec)**
	HQ	**MQ**	**LQ**	**HQ**	**MQ**	**LQ**	
1	89998	89998	89998	260	79	48	3600
2	89998	89998	89998	120	35	24	3600
3	89998	89998	89998	310	110	69	3600
4	89998	89998	89798	410	150	84	3600
5	30334	30335	30335	200	64	35	1200
6	44998	44998	44998	190	66	39	1800

Table 2. Other characteristics of the traces

2.5 Network and Mobility Models

The following mobility assumptions are adopted from [38, 39].

We consider an hexagonal cell architecture, as shown in Figure 2. We consider in all cells the uplink (wireless terminals to BS) wireless channel. Each cell has six neighbors. The cell diameter is 300 meters. Roads are modeled by straight lines. Each road is assigned a weight (w_j for road *j)*, which represents the traffic volume. Each new call is generated with a probability of 50% to be moving on the road and 50% to be stationary. Moving users are assumed to be traveling only on the roads, and are placed on each road *i* with probability $w_i / \sum^{j=N} w_j$ (1), where N is the total number of roads.

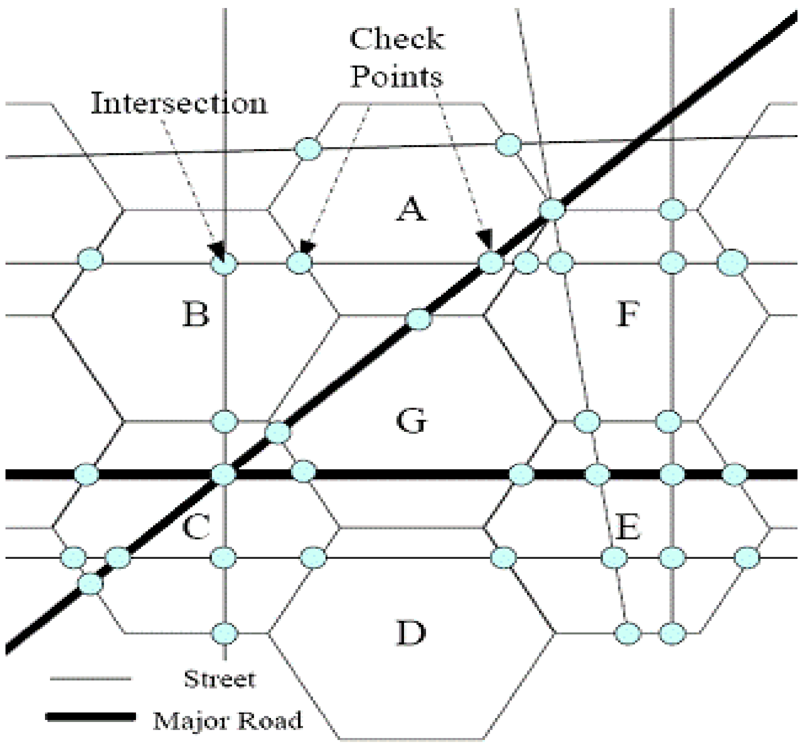


Figure 2. Road map and cellular network model.

The initial location of a moving user on a particular road is a uniform random variable between zero and the length of that road. During their call, stationary callers remain stationary and mobile users travel at a constant speed. Mobile users can travel in either of the two directions of a road with an equal probability, and with a speed chosen randomly in the range of [36, 90] Km/h. At the intersection of two roads, a mobile user might continue to go straight, or turn left, right, or around with probabilities 0.55, 0.2, 0.2 and 0.05, respectively. If a mobile user chooses to go straight or turn right at the intersection, it needs to stop there with probability 0.5 for a random time between 0 and 30 seconds due to a red traffic light. If the user chooses to turn left or around, it needs to stop there for a random time between 0 and 60 seconds due to the traffic signal. Each base station is loaded with the road map of its coverage area and its neighboring cells. Mobile stations report their position to the BS of their cell through a control channel. The position information includes the mobile user's exact location (cell and road), moving direction, and speed, and can be provided with an accuracy of 1m through GPS [38, 39, 40].

2.6 Adaptive Bandwidth Reservation Scheme [42]

We adopt the proposed scheme in [42], in our study. The bandwidth reservation schemes in [38, 39, 40] require mobile stations to report their location information to the BS every T seconds (1 second in [40], 10-45 seconds in [38] and 60 seconds in [39]). For this period of time if there is a probability based on the mobile's trajectory that it will move to a new cell, then a certain amount of bandwidth is reserved in all possible future cells that the mobile may move to.

In [38] the mobile's recorded moving history is also used for the prediction. In [39] a prediction is made by the system based on each updated position information and the road map. In [37], the authors do not use a standard road map for their study, but generate a random map for each simulation. The common disadvantage of these approaches is that the length of the report period yields a tradeoff between the prediction accuracy and the computational load imposed on the system. If mobile stations report their position frequently, the computational load of processing this information and using it for bandwidth reservation will be very high; on the other hand, if mobile users' position reports are very infrequent, the prediction on the mobile's trajectory can be untrustworthy and lead to an unnecessary waste of the reserved bandwidth in neighboring cells. Because of this tradeoff, the authors in [38, 40] use additional dynamic mechanisms in their schemes in order to adjust the amount of reserved bandwidth for users in neighboring cells, based on the quality of the system performance; we will explain in the following discussion why this “corrective” approach, which imposes a further computational load on the system, is not needed in our adopted scheme from [42].

In order to eliminate the problems introduced by the time-based location information reports of the mobile station to the BS, [42] proposes the use of *distance-based* information reports. More specifically, the authors set the road intersections and cell boundaries to be the “check-points” of the system, as shown in Figure 2. Each cell boundary represents a unique check-point, while around each road intersection four check points are set, one in each possible direction that the user may choose after reaching the intersection. Each of the check points is assumed to be placed at a distance of 10m from the intersection, which is a distance that even the slowest moving vehicles considered in this work (with a speed of 36 Km/h) will cover within 1s after passing the intersection (naturally, if there are less than four possible directions for a mobile to follow after reaching an intersection, the number of check points needed is smaller than four). Mobile stations only need to update their position information to the BS of their cell when they arrive at a check-point. Check-points are assumed to be configured by the wireless provider into the mobile terminals' GPS maps.

If the BS of the current cell of a mobile station predicts, based on the station's location (at a check point) and speed that the station is going to move to another cell (i.e., that the next check point for the station will be a cell boundary), it sends a notification to the BS of that cell, including the current bandwidth used by the station and the estimated arrival time at the next check point. Hence, the proper amount of bandwidth is reserved for the station. For regular video terminals, the bandwidth that is reserved in the next cell is equal to the remaining bandwidth that the terminal will need to complete its transmission (this bandwidth is declared in each video user's initial request to the BS). For all other types of users, the bandwidth that is reserved in the next cell is equal to their current bandwidth, so that they will seamlessly continue its transmission. The proposed approach in [42] not only guarantees the existence of adequate resources in the new cell for handoff users but also reduces the information update frequency.

The prediction and adaptive reservation process executed by the BS can be summarized as follows:

For each new user in the system do

  If the mobile station is not stationary then

    When the mobile station arrives at a check point

    Update the position information

    Estimate the next arrival check-point and calculate the arrival time at that check point

    If The next check point is a cell boundary then

      Reserve the proper amount of bandwidth for the station in the next cell

The proposed in [42] adaptive bandwidth reservation scheme, based on *distance-based location updates,* has two major advantages:

1. The position update duration is unique for every mobile user, based on their different initial position and speed. This is impossible to "capture" with the time-based location updates proposed in [38, 39, 40]. Additionally, it creates less computational load than the

time-based updates which are, in most proposals in the literature, synchronized and therefore require simultaneous information transmission from the terminals to the BS.

2. An important parameter used to evaluate a bandwidth reservation scheme is bandwidth efficiency [39], which is calculated by $f = Nr/Nq$, where Nr is the reserved bandwidth and Nq is the actual bandwidth utilized by handoff users. The closer f is to 1, the higher is the efficiency achieved by the bandwidth reservation scheme, since this would mean that there is neither lack nor waste of bandwidth in the reservation procedure. The scheme in [39] outperformed other schemes with which it was compared, achieving at best a f = 1.047379 (the schemes with which [39] was compared achieved a bandwidth efficiency in the range of 1.25).

The distance-based approach in [42] leads the bandwidth efficiency of their scheme to asymptotically reach 1. The reasons are:

▪ The scheme in [42] guarantees that there is no waste of bandwidth. For all users ready to handoff, the exact amount of bandwidth they need is reserved in the new cell and this cell is known with precision, with the use of the four check points around each intersection. The only case when bandwidth can be wasted, with the use of the adaptive bandwidth reservation approach, is when a mobile station which is predicted to move to another cell makes a stop before entering this cell (this case is not included in our adopted mobility model). Given the fact that the distance between a check point set close to an intersection and a cell boundary is in all cases much smaller than the cell diameter of 300 meters, this case can be considered a rare exception.

▪ The scheme in [42] also guarantees that there will be no lack of bandwidth for handoff users. The bandwidth needed for all types of regular mobile stations which do not transmit video is close to 35 Kbps (just one slot per channel frame). Therefore, this bandwidth is very small compared to the total channel capacity of 20 Mbps and can generally be reserved in the next cell with the rare exception of cases when the channel is overloaded with traffic.

3. The Proposed Call Admission Control Mechanism

Our CAC mechanism attempts to take advantage of the video users' willingness to experience degraded service for cheaper connection rates, in order to increase the number of users that are accepted in the network, and hence to increase channel throughput. Without loss of generality, we assumed that in each user category the percentage of users who accept to be downgraded is equal to 50%.

Every second, the system checks for new requests by video users to enter the network. The choice of one of the six MPEG-4 traces is made with equal probability. Without loss of generality (we experimented with different values and there was no qualitative change in our results) we assume that the probability of a user choosing the HQ version of the trace is 10%, the MQ version 40% and the LQ version 50%. Also, the probability that the user accepts degradation to be immediate next Quality is 50%.

The CAC mechanism needs to calculate the equivalent bandwidth of the superposition of the MPEG-4 traces, i.e., to estimate the bandwidth that will be needed to satisfy the existing video users, plus the new requesting user. We use the following formula, proposed in [10] and used widely in the relevant literature:

$$t\sqrt{\sum_i \sigma_i^{\,2}} + \sum_i \mu_i < BW \tag{2}$$

The constant t is defined as h(t) = 1- PLR, where PLR is the acceptable packet loss rate, BW is the available bandwidth, h is the cumulative normalized normal distribution, σ_i^2 and μ_i are the variance and mean of each trace, respectively.

If the equivalent bandwidth of the traces' superposition does not exceed the available bandwidth, the new user is accepted into the network. If it does exceed the available bandwidth and the user's contract defines that the user does not accept degradation, then the user's request is rejected. If, however, the equivalent bandwidth exceeds the available bandwidth but the user accepts degradation, then the user is degraded to the immediate next Quality (HQ -> MQ or MQ -> LQ) and the CAC mechanism checks again, with the use of (2) whether the user can be accepted, under the new Quality. If not, the user's request is rejected. Similarly to the degradation procedure, our mechanism also allows user upgrades (to the immediate next superior Quality), in order to improve user QoS and achieve an increase in network throughput, when possible.

A slightly different approach is necessary in the case of requests coming from handoff video users. It is well-known that user annoyance, in cellular networks, is significantly higher when a call in-progress is terminated, compared to the case when a user is unable to initiate a call. Therefore, the minimization of call dropping is more important than that of call blocking, and for this reason handoff users need to be treated by the network with absolute priority, over new calls initiated from within the cell.

In the case of handoff video users, our CAC mechanism checks whether there is enough available bandwidth to accept the handoff user into the network. If so, the user is accepted. If not, the mechanism checks whether the user accepts degradation, in which case the user is degraded to the immediate next inferior Quality and the CAC checks once again if the user can be accepted. In this case, either the degraded user is accepted, or a new cycle of degradations starts as explained below.

In the case where a degraded user still cannot be accepted due to bandwidth unavailability, or in the case where a handoff user does not accept degradation, the CAC mechanism looks for video users which have already been accepted in the network and which accept degradation. The mechanism degraded these users in a serial manner (after each degradation the mechanism checks if the "released" bandwidth is enough to accept the handoff user) and continues to degrade as many of them as needed until enough bandwidth is available to accept the handoff user. In case all users who can tolerate degradation are degraded and the available bandwidth is still not enough to accept the handoff user, then the handoff video call is terminated. The same is true for handoff voice and data users, when there is not enough bandwidth from video degradations to accommodate them.

3.1. The Upgrade/Downgrade Implementations

We study two implementations of the CAC mechanism.

In the first, for each trace all three versions (HQ, MQ, LQ) from [32] are available. When a user is degraded/upgraded, the user continues to receive the trace in the downlink channel, but in the new, inferior/superior Quality. This is theoretically plausible, but in practice quite difficult, as every video that the user wishes to watch will need to be available in different qualities and cached in the cell or in neighboring cells, so that the transition between different Qualities can be done seamlessly. Additionally, it has been shown in [35, 43] that the classic equivalent bandwidth approach leads to very conservative estimations of the bandwidth that will be needed to satisfy the users' QoS requirements, and hence to a significant under-utilization of the channel bandwidth. Therefore, the difficulty of such an implementation, together with its inefficiency, led us to propose a second, more practical degradation/upgrade implementation of the mechanism, in which the MQ and LQ versions of each trace are defined differently.

The decoding of an I frame in a typical MPEG-4 trace is independent of other video frames. The decoding of P frames depends on the successful decoding of the I frame. The decoding of B frames depends on the successful decoding of I and P frames. Therefore we have implemented the upgrade/downgrade procedure as follows, so that the existence of just one version of the trace (the HQ version) is enough.

A HQ user receives all video frames of the movie that the user requested. A MQ user receives all I and P frames (B frames are dropped). A LQ user receives only the I frames, i.e., the basic information of each GoP(P and B frames are dropped). This implementation depends on the re-calculation of the statistical parameters needed in (2), to estimate the equivalent bandwidth of the new MQ and LQ versions of each trace.

3.2. Fractional Guard Channel Scheme

As mentioned in the introduction, in the second part of our thesis we used a fractional guard channel (FGC) mechanism. The FGC scheme falls into the category of the new call thinning schemes; the idea behind these schemes is to smoothly throttle the new call stream as the network traffic is building up. Hence, when the network is approaching congestion, the admitted new call stream becomes thinner [17]. Our scheme is enforced only on video users and admits a new video call with a specific probability; otherwise the request is rejected.

The reason that the FGC scheme is used only on video calls is that video traffic is by far the burstiest traffic in the network, and therefore needs to be controlled. We cannot use FGC for handoff video calls, as handoff traffic needs to be accepted in the new cell with absolute priority. As we will see in Section 4, where we present and discuss our results, the use of the FGC scheme just for new video calls suffices for a significant improvement in system performance when a strict CAC needs to be enforced and the QoS of handoff users is of primary concern.

The implementation of our mechanism is as follows: For each new video user who tries to enter the network the BS computes the ratio of the equivalent bandwidth of the video users, versus the maximum bandwidth that video users are allowed to use (in order to allow bandwidth for voice and data users).The BS accepts a new video call that does not violate (2) with a probability P_{acc} equal to $P_{acc} = 1 – EB / mabv$, where EB is the equivalent bandwidth of the video terminals calculated by (2) and mabv is the maximum allowed bandwidth for video traffic. Hence, in the case when video users are largely absent from the system or underutilize the bandwidth that is dedicated to them (e.g., LQ users), video calls will be accepted with high probability. However, when the system is becoming congested by video traffic, video calls will be rejected with a high probability.

It needs to be emphasized that the FGC mechanism is clearly stricter than the first CAC mechanism that we study in this work. This is due to the fact that the first CAC always accepts a call if (2) is satisfied.

4. Results and Discussion

Each simulation is the result of an average of 10 independent runs (Monte-Carlo simulation). The duration of each simulation is equal to 300000 channel frames (1 hour) while the duration of each frame is 12 msec.

We studied all versions of our CAC mechanism in the case of simultaneous use of the channel by voice, data (email) and video (mpeg-4 movies) users. The final results are separated into two categories (according to the implementation of the video users' degradation).

4.1. First Implementation of Video Users' Downgrades/Upgrades

In this implementation we assume that each movie is available in each cell in 3 different quality formats (HQ, MQ and LQ). In the following sections, we studied the network performance metrics when the channel is used by voice, video and data terminals, when it is used only by video users and the metrics which refer to data users for a given volume of voice and video traffic.

4.1.1 Results for Voice, Data and Video Users

In order to study the system's behavior for different traffic loads, we used transmission rates for voice, data and video which correspond to: 40%, 50%, 60%, 70%, 80%, 90%, 100%, 110%, 120%, 130%, 140% of the channel bandwidth. For example, 40% means that the traffic which is generated, on average, equals to 8 Mbps, since the channel bandwidth is 20 Mbps. For each one of the above - mentioned eleven rates we created 3 different traffic combinations for voice, data and video (each combination generates the same traffic load). The rates and their combinations are shown in Tables 3 and 4. In all of these traffic scenarios we assume that the number of voice users is constant during the simulation. The email arrival rate changes in each traffic scenario; we assume a high average arrival rate of video users in all of the scenarios, equal to 10 users per minute.

Furthermore, a threshold was defined for the maximum bandwidth that video users can utilize. This means that our CAC scheme allows video users to enter the channel until a specific amount of bandwidth has been reserved. If that bandwidth is exceeded, the requests of video users are rejected. Therefore, the number of video users in the system changes constantly, depending on the traffic mix. A video user's average sojourn time inside the network is 10 minutes. For the calculation of the equivalent bandwidth of the video users' superposition, we use (2).

Table 3 presents all the simulation scenarios that we used that correspond to traffic generation with rates equal to 40%-100% of the channel bandwidth. The parameter "Maximum Video Capacity" of Table 3 denotes the maximum possible bandwidth that video users can reserve. Table 3 presents, for each traffic scenario, the real bandwidth that all users reserve and the respective prediction of the CAC scheme.

The first line of Table 3 shows that the bandwidth which theoretically would be reserved by the 3 types of traffic is 40% of the total (16%+4%+20%). Because of the CAC algorithm, which, as in [16], is shown to be very conservative in its estimations, the real amount of bandwidth that users manage to reserve is 29.3%. It should be noted that this result is very close, quantitatively and qualitatively, with the results derived in [41], despite the fact that [41] did not use our adaptive bandwidth reservation scheme. A comparison between the adaptive bandwidth reservation scheme and [41] will follow in this section. We also observe, from Table 3, that as the traffic load increases, the difference between the theoretical and the real bandwidth utilized can exceed 20%. It should also be noted that only for a 100% traffic load are the QoS requirements of very few email messages (delay less than 5 seconds) violated. Voice and video traffic QoS requirements are satisfied in all the scenarios, at the cost of low channel utilization.

The CAC's conservatism is shown clearly in Table 4, where we present again, in the first three columns, the percentage of channel bandwidth, which the three types of traffic (voice, data, video) try to reserve. In the next three columns, the percentage of bandwidth that is really reserved by the three traffic types is shown; it is significantly lower, for video and data users, than the requested values.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	CAC Bandwidth Estimation (Mbps)	Real Bandwidth (Mbps)	Dropped Voice Packets (%)	Average Number of messages with excessive delay	Dropped Video Packets (%)
214 (16%)	24 (4%)	4 (20%)	7.86	5.86 (29.3%)	0	0	0
267 (20%)	24 (4%)	3.2 (16%)	7.88	6.2 (31%)	0	0	0
320 (24%)	24 (4%)	2.4 (12%)	7.89	6.52 (32.6%)	0	0	0
267 (20%)	30 (5%)	5 (25%)	9.83	7.5 (37.5%)	0	0	0
334 (25%)	30 (5%)	4 (20%)	9.88	7.89 (39.5%)	0	0	0
400 (30%)	30 (5%)	3 (15%)	9.89	8.25 (41.3%)	0	0	0
320 (24%)	36 (6%)	6 (30%)	11.82	9.18 (45.9%)	0	0	0
400 (30%)	36 (6%)	4.8 (24%)	11.85	9.62 (48.1%)	0	0	0
480 (36%)	36 (6%)	3.6 (18%)	11.69	9.74 (48.7%)	0	0	0
374 (28%)	42 (7%)	7 (35%)	13.78	10.91 (54.6%)	0.0000005987	0	0
467 (35%)	42 (7%)	5.6 (28%)	13.82	11.38 (56.9%)	0.000000479	0	0
560 (42%)	42 (7%)	4.2 (21%)	13.87	11.81 (59.1%)	0	0	0
427 (32%)	48 (8%)	8 (40%)	15.73	12.66 (63.3%)	0.00005166	0	0
534 (40%)	48 (8%)	6.4 (32%)	15.82	13.08 (65.4%)	0.00001131	0	0
640 (48%)	48 (8%)	4.8 (24%)	15.88	13.59 (68%)	0.000001399	0	0
480 (36%)	54 (9%)	9 (45%)	17.7	14.38 (71.9%)	0.00117	0	0
600 (45%)	54 (9%)	7.2 (36%)	17.8	14.85 (74.3%)	0.000385	0	0
720 (54%)	54 (9%)	5.4 (27%)	17.87	15.35 (76.8%)	0.0001947	0	0
534 (40%)	59 (10%)	10 (50%)	19.42	15.72 (78.6%)	0.009356	0	0
667 (50%)	59 (10%)	8 (40%)	19.49	16.27 (81.4%)	0.005513	1.286	0
800 (60%)	59 (10%)	6 (30%)	19.59	16.76 (83.8%)	0.003464	2.714	0

Table 3. Simulation scenarios with loads between 40%-100% of the total bandwidth

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Voice Real Bandwidth	Data Real Bandwidth	Video Real Bandwidth
214 (16%)	24 (4%)	4 (20%)	16.09%	3.91%	9.31%
267 (20%)	24 (4%)	3.2 (16%)	20.09%	3.93%	6.98%
320 (24%)	24 (4%)	2.4 (12%)	24.06%	3.94%	4.62%
267 (20%)	30 (5%)	5 (25%)	20.09%	4.95%	12.48%
334 (25%)	30 (5%)	4 (20%)	25.12%	4.89%	9.43%
400 (30%)	30 (5%)	3 (15%)	30.07%	4.94%	6.22%
320 (24%)	36 (6%)	6 (30%)	24.04%	5.92%	15.92%
400 (30%)	36 (6%)	4.8 (24%)	30.09%	5.91%	12.12%
480 (36%)	36 (6%)	3.6 (18%)	35.14%	5.89%	7.68%
374 (28%)	42 (7%)	7 (35%)	28.1%	6.9%	19.53%
467 (35%)	42 (7%)	5.6 (28%)	35.12%	6.93%	14.83%
560 (42%)	42 (7%)	4.2 (21%)	42.12%	6.8%	10.14%
427 (32%)	48 (8%)	8 (40%)	32.08%	7.89%	23.32%
534 (40%)	48 (8%)	6.4 (32%)	40.17%	7.87%	17.36%
640 (48%)	48 (8%)	4.8 (24)	48.12%	7.79%	12.05%
480 (36%)	54 (9%)	9 (45%)	36.08%	8.54%	26.27%
600 (45%)	54 (9%)	7.2 (36%)	45.12%	8.62%	20.51%
720 (54%)	54 (9%)	5.4 (27%)	54.15%	8.51%	14.13%
534 (40%)	59 (10%)	10 (50%)	40.16%	7.97%	30.48%
667 (50%)	59 (10%)	8 (40%)	50.15%	7.93%	23.32%
800 (60%)	59 (10%)	6 (30%)	60.15%	7.65%	16%

Table 4. Bandwidth that is really reserved by each type of traffic, with loads between 40%-100% of the total bandwidth.

Tables 5 and 6 present the same results as Tables 3 and 4, but in this case the total traffic load that tries to enter the network is higher than 100% of the channel bandwidth (between 110% and 140%). From Table 5 we observe that, for loads larger than 130% the voice QoS requirement of mean packet dropping less than 1% is violated. For channel loads equal or greater than 120% there isn't any QoS requirement violation for data users; this can be explained by Table 6, where it is shown that if the channel load becomes greater than 110%, the real percentage of bandwidth that data terminals use approaches zero. Therefore, the QoS requirement of data users is not violated simply because very few data users are allowed to enter the network. As far as video users are concerned, there is no violation of their QoS requirements. This is due to:

a. The conservatism of the CAC algorithm, which leads to the reservation of significantly higher bandwidth than that which video terminals actually need. This means that even in cases when video terminals receive bursty information, there will be available bandwidth in order for the user to receive that information without packet dropping.

b. Mpeg-4 movies have the highest priority in comparison to the other types of traffic. This means that, after all video packets have been transmitted, voice packets will be transmitted in the remaining time slots and the transmission of data packets will take place last.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	CAC Bandwidth Estimation (Mbps)	Real Bandwidth (Mbps)	Dropped Voice Packets (%)	Number of messages with excessive delay	Dropped Video Packets (%)
587 (44%)	65 (11%)	11 (55%)	19.84	16.1 (80.5%)	0.124	1.234	0
734 (55%)	65 (11%)	8.8 (44%)	19.97	16.6 (83%)	0.08504	0	0
880 (66%)	65 (11%)	6.6 (33%)	20.06	17.16 (86.8%)	0.08315	0	0
640 (48%)	71 (12%)	12 (60%)	21.07	17.13(85.7%)	0.7703	0	0
800 (60%)	71 (12%)	9.6 (48%)	21.23	17.77 (88.9%)	0.7951	0	0
960 (72%)	71 (12%)	7.2 (36%)	21.34	18.38 (91.9%)	0.9018	0	0
694 (52%)	77 (13%)	13 (65%)	22.72	18.37 (91.9%)	3.154	0	0
867 (65%)	77 (13%)	10.4 (52%)	22.96	18.99 (95%)	3.841	0	0
1040(78%)	77 (13%)	7.8 (39%)	23.1	19.4 (97%)	4.475	0	0
747 (56%)	83 (14%)	14 (70%)	24.32	19.22 (96.1%)	9.273	0	0
934 (70%)	83 (14%)	11.2 (56%)	24.65	19.6 (98%)	9.938	0	0
1120(84%)	83 (14%)	8.4 (42%)	24.86	19.86 (99.3%)	11.28	0	0

Table 5. Simulation scenarios with loads between 110%-140% of total bandwidth.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Voice Real Bandwidth	Data Real Bandwidth	Video Real Bandwidth
587 (44%)	65 (11%)	11 (55%)	44.07%	2.47%	33.96%
734 (55%)	65 (11%)	8.8 (44%)	55.15%	2.27%	25.56%
880 (66%)	65 (11%)	6.6 (33%)	66.15%	1.93%	17.71%
640 (48%)	71 (12%)	12 (60%)	47.73%	0.53%	37.38%
800 (60%)	71 (12%)	9.6 (48%)	59.7%	0.32%	28.86%
960 (72%)	71 (12%)	7.2 (36%)	71.5%	0.23%	20.16%
694 (52%)	77 (13%)	13 (65%)	50.55%	0.49%	40.84%
867 (65%)	77 (13%)	10.4 (52%)	62.7%	0.27%	32%
1040(78%)	77 (13%)	7.8 (39%)	74.7%	0.11%	22.2%
747 (56%)	83 (14%)	14 (70%)	50.95%	0.47%	44.69%
934 (70%)	83 (14%)	11.2 (56%)	63.25%	0.27%	34.47%
1120(84%)	83 (14%)	8.4 (42%)	74.7%	0.07%	24.55%

Table 6. Bandwidth that is really reserved by each kind of streaming with loads between 110%-140% of total bandwidth.

In Figure 3 we present graphically the CAC estimation for the required bandwidth (average of three combinations for each traffic load), as well as the real bandwidth that users need. Once again, it is obvious that the CAC mechanism's estimation is very conservative. This conservatism comes with the advantage that the users who enter the system enjoy very high QoS. However, it leads to significant underutilization of the channel bandwidth.

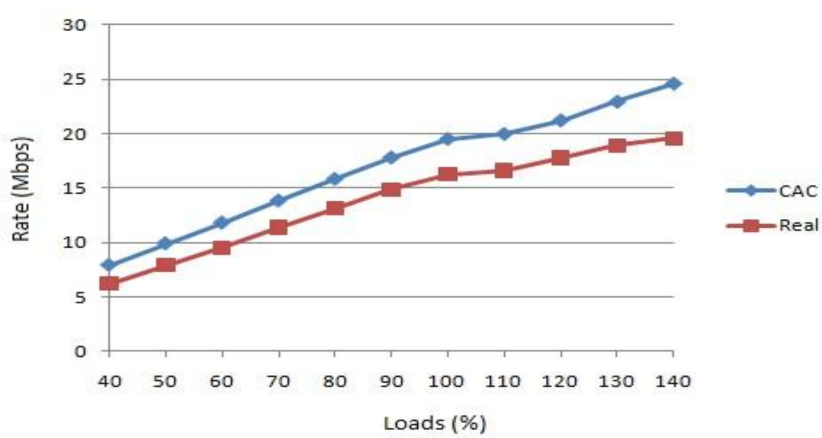


Figure 3. Comparison between real bandwidth needs and the respective CAC estimation for different loads.

Finally, we compare the case where road map-based adaptive bandwidth reservation is used, with the case studied in [41] (only one cell, without adaptive bandwidth reservation). For this comparison we use three metrics:

a. The percentage of video users who enter the network without degradation.
b. The percentage of video users who enter the network either by being degraded or due to the degradation of other video users.
c. The percentage of video users who do not manage to enter the network, due to the lack of adequate bandwidth.

The comparison of the results presented in Tables 7 and 8 shows that in most cases (and on average over all the scenarios studied) the use of the adaptive bandwidth reservation leads to a higher percentage of handoff video users being accepted without degradation and a smaller percentage of handoff video users being rejected. Tables 9 and 10 show that the adaptive bandwidth reservation scheme clearly improves the QoS of non-handoff users as well.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Handoff Video Users without Degradation	Handoff Video Users Degraded by CAC	Handoff Video Users rejected by CAC
214 (16%)	24 (4%)	4 (20%)	21.43%	3.06%	74.49%
267 (20%)	24 (4%)	3.2 (16%)	17.86%	2.38%	78.57%
320 (24%)	24 (4%)	2.4 (12%)	11.54%	2.56%	85.9%
320 (24%)	36 (6%)	6 (30%)	31.09%	5.88%	62.18%
400 (30%)	36 (6%)	4.8 (24%)	26.32%	5.26%	67.37%
480 (36%)	36 (6%)	3.6 (18%)	21.59%	4.55%	72.73%
427 (32%)	48 (8%)	8 (40%)	40.48%	7.94%	50.79%
534 (40%)	48 (8%)	6.4 (32%)	30.91%	6.36%	61.82%
640 (48%)	48 (8%)	4.8 (24)	23%	5%	72%
534 (40%)	59 (10%)	10 (50%)	37.04%	5.19%	57.04%
667 (50%)	59 (10%)	8 (40%)	34.13%	3.97%	61.11%
800 (60%)	59 (10%)	6 (30%)	25.23%	3.74%	69.16%

Table 7. Metrics for handoff video terminals for different loads, using adaptive bandwidth reservation.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Handoff Video Users without Degradation	Handoff Video Users Degraded by CAC	Handoff Video Users rejected by CAC
214 (16%)	24 (4%)	4 (20%)	16.98%	1.89%	81.13%
267 (20%)	24 (4%)	3.2 (16%)	7.55%	1.89%	90.57%
320 (24%)	24 (4%)	2.4 (12%)	9.09%	1.82%	89.09%
320 (24%)	36 (6%)	6 (30%)	34.43%	8.2%	57.38%
400 (30%)	36 (6%)	4.8 (24%)	20%	6.67%	73.33%
480 (36%)	36 (6%)	3.6 (18%)	16.36%	5.45%	78.18%
427 (32%)	48 (8%)	8 (40%)	40.68%	6.78%	52.54%
534 (40%)	48 (8%)	6.4 (32%)	32.84%	10.45%	56.72%
640 (48%)	48 (8%)	4.8 (24)	17.86%	7.14%	75%
534 (40%)	59 (10%)	10 (50%)	38.71%	4.84%	56.45%
667 (50%)	59 (10%)	8 (40%)	29.41%	3.92%	66.67%
800 (60%)	59 (10%)	6 (30%)	23.87%	3.02%	73.13%

Table 8. Metrics for handoff video terminals for different loads, for one cell, without using adaptive bandwidth reservation.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Non Handoff Video Users without Degradation	Non Handoff Video Users Degraded by CAC	Non Handoff Video Users rejected by CAC
214 (16%)	24 (4%)	4 (20%)	16.88%	1.45%	81.49%
267 (20%)	24 (4%)	3.2 (16%)	12.12%	1.25%	86.45%
320 (24%)	24 (4%)	2.4 (12%)	10.47%	1.08%	88.27%
320 (24%)	36 (6%)	6 (30%)	22.7%	2.88%	74.41%
400 (30%)	36 (6%)	4.8 (24%)	18%	2.23%	79.59%
480 (36%)	36 (6%)	3.6 (18%)	13.61%	2.18%	84.21%
427 (32%)	48 (8%)	8 (40%)	30.37%	3.33%	66.3%
534 (40%)	48 (8%)	6.4 (32%)	24.18%	2.91%	72.73%
640 (48%)	48 (8%)	4.8 (24)	18.41%	2.48%	78.94%
534 (40%)	59 (10%)	10 (50%)	40.18%	3.06%	56.58%
667 (50%)	59 (10%)	8 (40%)	32.73%	3.25%	64.01%
800 (60%)	59 (10%)	6 (30%)	25.37%	2.43%	72.01%

Table 9. Metrics for non handoff video terminals, using adaptive bandwidth reservation.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Non Handoff Video Users without Degradation	Non Handoff Video Users Degraded by CAC	Non Handoff Video Users rejected by CAC
214 (16%)	24 (4%)	4 (20%)	12.8%	2.01%	85.19%
267 (20%)	24 (4%)	3.2 (16%)	9.76%	1.39%	88.85%
320 (24%)	24 (4%)	2.4 (12%)	6.95%	0.71%	92.34%
320 (24%)	36 (6%)	6 (30%)	19.73%	2.11%	78.16%
400 (30%)	36 (6%)	4.8 (24%)	14.11%	1.92%	83.97%
480 (36%)	36 (6%)	3.6 (18%)	11.24%	1.14%	87.62%
427 (32%)	48 (8%)	8 (40%)	24.8%	3.52%	71.68%
534 (40%)	48 (8%)	6.4 (32%)	18.32%	2.99%	78.69%
640 (48%)	48 (8%)	4.8 (24)	15.04%	1.63%	83.33%
534 (40%)	59 (10%)	10 (50%)	35.21%	2.25%	62.55%
667 (50%)	59 (10%)	8 (40%)	29.39%	4.08%	66.53%
800 (60%)	59 (10%)	6 (30%)	21.09%	2.07%	76.84%

Table 10. Metrics for non handoff video terminals, for one cell, without using adaptive bandwidth reservation.

4.1.2 Results for Video Terminals

In this section our study focuses on video terminals. For this reason, in the following simulations we assume that the only type of traffic present in the network is video. More specifically, for each traffic load, users are inserted into the network until the CAC mechanism starts rejecting them, when the maximum allowable video load is exceeded. The number of users does not remain constant during the simulation as video users move from cell to cell and can either exit the network topology or terminate their call.

We have derived results for different average arrival rates λ of video terminals and different maximum traffic loads. In Table 11, where we present results for a video arrival rate λ = 10 users/minute, we notice that the increase in the maximum allowable video load increases the real transmission rate and the average number of video users in the network. Still, the bandwidth utilized falls significantly short of the maximum allowable load; the reason is, again, the conservatism of the CAC mechanism, which however ensures (as shown in the Table) that only for a maximum allowable video load larger than 90% of the channel bandwidth are the QoS requirements of video users violated.

Maximum Allowed Video Load (Mbps)	Video Arrival Rate (λ) (users/min)	CAC Bandwidth Estimation (Mbps)	Real Bandwidth (Mbps)	Average Number of Users	Dropped Video Packets (%)
8 (40%)	10	7.69	4.65 (23.3%)	30	0
10 (50%)	10	9.55	6.13 (30.7%)	35	0
12 (60%)	10	11.43	7.69 (38.5%)	41	0.000003065
14 (70%)	10	13.3	9.29 (46.5%)	48	0.00002028
16 (80%)	10	15.16	10.91 (54.6%)	50	0.0001879
18 (90%)	10	16.88	12.4 (62%)	55	0.004046
20 (100%)	10	18.71	14.06 (70.3%)	61	0.04385
22 (110%)	10	20.42	15.54 (77.7%)	64	0.3245

Table 11. Average number of active users in the system with λ = 10 users/minute.

In Figures 4-7, for an average video arrival rate equal to 1 user/minute/cell, and in Figures 8-11, for an average video arrival rate equal to 10 users/minute/cell, we present for various video loads the bandwidth that is really used by video terminals per second and the bandwidth that the CAC algorithm estimates as required. All of the figures show that the CAC mechanism overestimates significantly the required bandwidth, and the gap increases for higher λ.

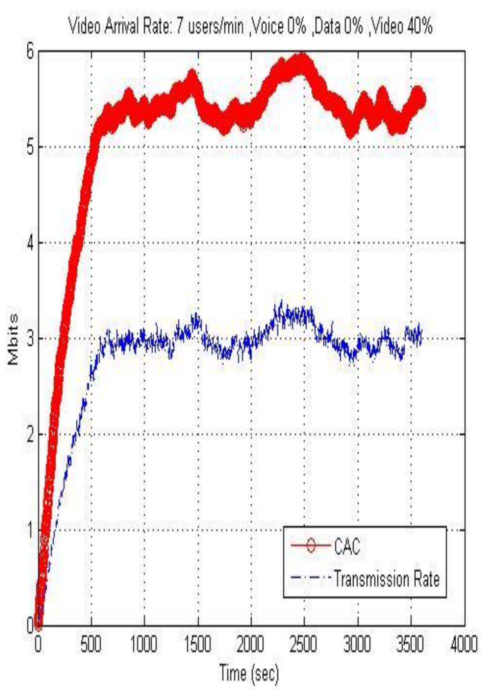


Figure 4

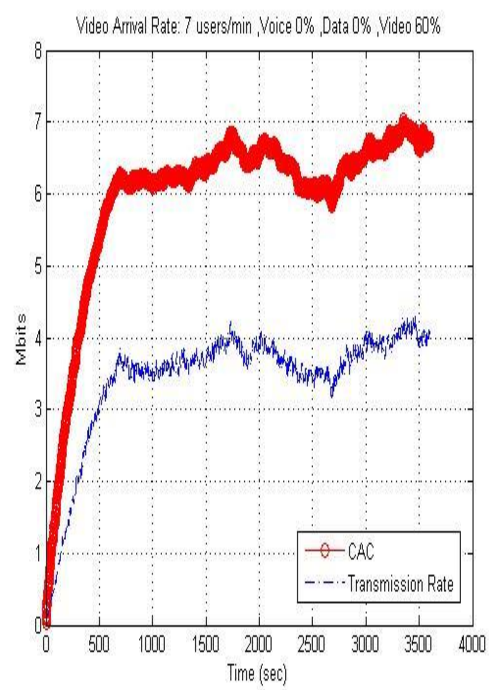


Figure 5

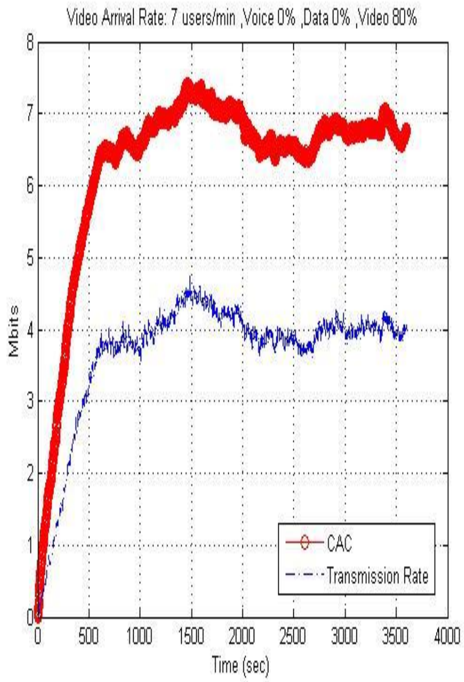


Figure 6

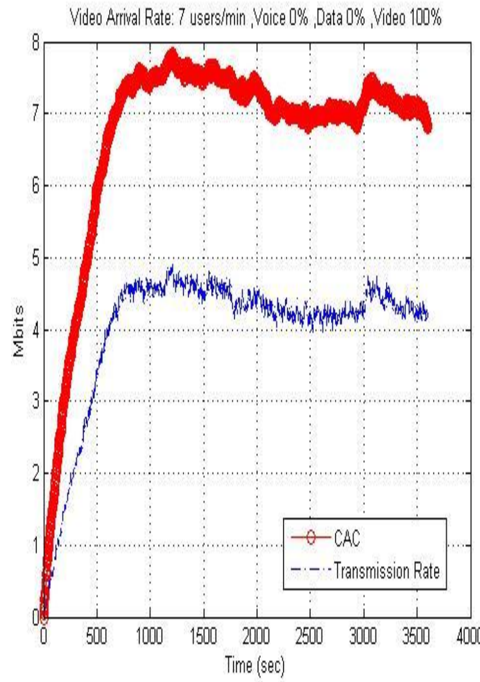


Figure 7

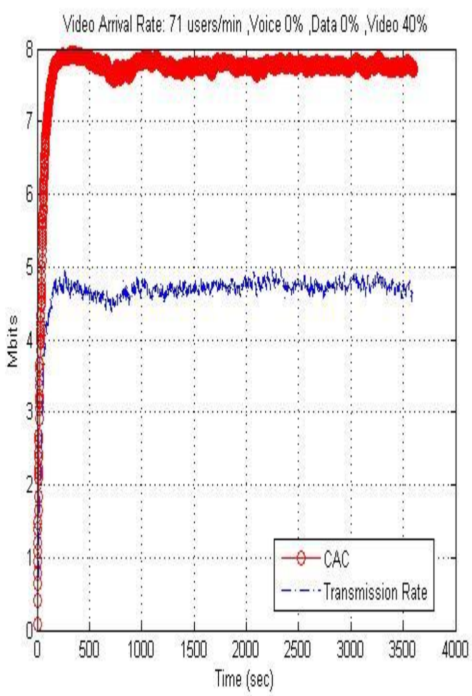


Figure 8

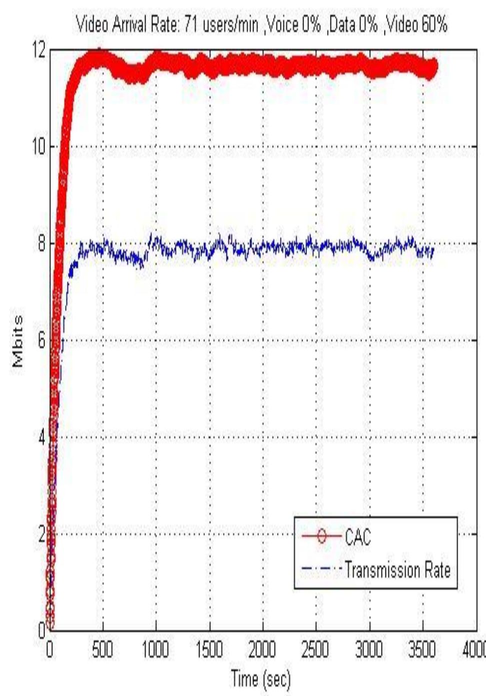


Figure 9

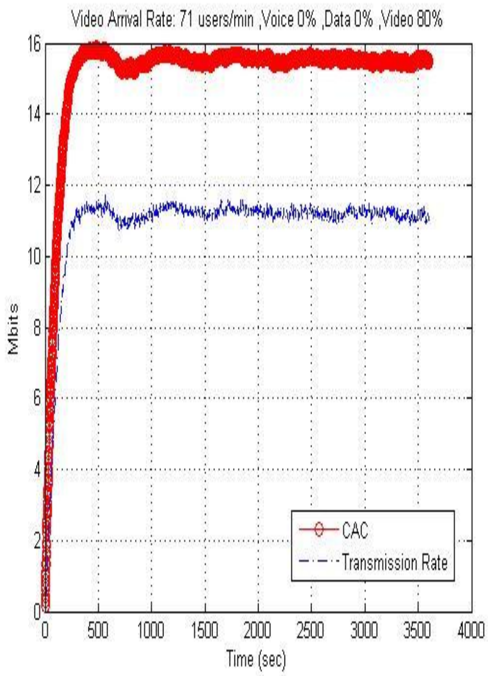


Figure 10

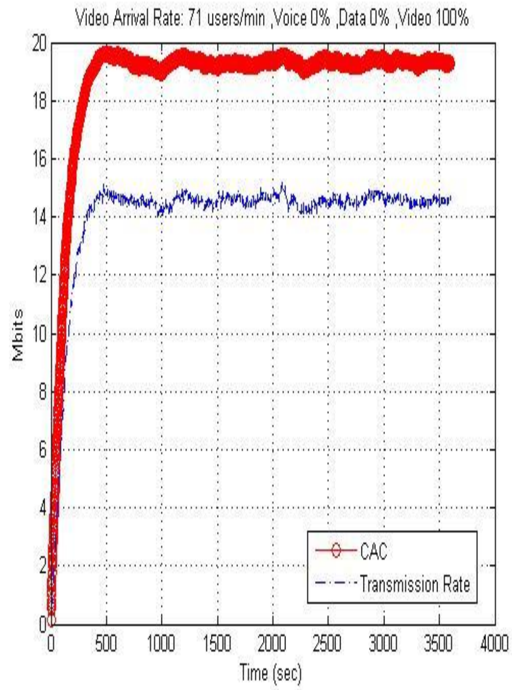


Figure 11

4.1.3 Downgrades/Upgrades of Video Terminals

Table 12 shows results for different video loads with λ = 1 user/minute. The column "Active Users" of Table 12 presents the total number of users that have been admitted in the network. The values of the third and fourth column of Table 12 show the percentage of HQ and MQ video terminals that have been downgraded (this percentage is computed over the number of users that can be downgraded based on their contract). Values in the last two columns of the Table present the degradation ratio, which is the average percentage of time that a video user remains downgraded. As the maximum allowable video load increases, the downgrade percentage is shown to be practically negligible.

Maximum Allowed Video Load (Mbps)	Active Users	Downgrade percentage HQ	Downgrade percentage MQ	Degradation Ratio HQ	Degradation Ratio MQ
8 (40%)	69	31.86%	21%	36%	24.86%
10 (50%)	74	20.14%	16.29%	22.57%	22.86%
12 (60%)	73	11.71%	7.143%	17.29%	12.43%
14 (70%)	77	3.429%	4.857%	7.429%	12.29%
16 (80%)	75	2.429%	2.571%	3.143%	9.429%
18 (90%)	77	0.5714%	0.4286%	2.857%	1.571%
20 (100%)	78	1%	0.1429%	0.8571%	0.8571%
22 (110%)	76	0%	0%	0%	0%

Table 12. Percentage of degradations and degradation ratio for λ=1 user/minute/cell.

Tables 13 and 14 show that, for higher video arrival rates, the percentage of users that need to be degraded is significantly higher than that when λ = 1 user/minute. Also, users that are degraded remain degraded for the largest part of their call duration. Regardless of the video arrival rate, the percentage of degradations is greater for HQ than MQ movies. This is expected due to the much higher bandwidth requirements of HQ movies; this makes it difficult to find the necessary resources to transmit HQ movies without degradation.

Maximum Allowed Video Load (Mbps)	Active Users	Downgrade percentage HQ	Downgrade percentage MQ	Degradation Ratio HQ	Degradation Ratio MQ
8 (40%)	188	88.43%	80.57%	70.14%	66.43%
10 (50%)	217	83.14%	67.71%	71.71%	61%
12 (60%)	252	74.29%	59.71%	71.14%	58.57%
14 (70%)	270	63.43%	51.14%	67%	52.43%
16 (80%)	291	57.86%	45.29%	66.43%	51.71%
18 (90%)	310	49.43%	38.57%	60.14%	46.14%
20 (100%)	328	41.14%	34.86%	53.14%	47.71%
22 (110%)	337	36.29%	30.43%	47.71%	44.86%

Table 13. Percentage of degradations and degradation ratio for λ=5 users/minute/cell.

Maximum Allowed Video Load (Mbps)	Active Users	Downgrade percentage HQ	Downgrade percentage MQ	Degradation Ratio HQ	Degradation Ratio MQ
8 (40%)	245	95.57%	92.71%	75.86%	80%
10 (50%)	300	96.29%	90.14%	77.14%	75.86%
12 (60%)	345	94.43%	86.71%	73.71%	69.86%
14 (70%)	381	90.14%	81%	74%	64.57%
16 (80%)	412	84%	73.86%	69.57%	61.57%
18 (90%)	444	79.29%	68.14%	69.43%	59.14%
20 (100%)	486	73.14%	59.57%	70.71%	55.71%
22 (110%)	509	69.29%	56%	70.29%	55.86%

Table 14. Percentage of degradations and degradation ratio for λ = 10 users/minute/cell.

Figures 12-23 illustrate the number of upgrades and degradations of mpeg-4 movies that correspond to the elements of Tables 12-13 for arrival rates equal to 1, 5 and 10 users/minute/cell. In Figures 12-15 we notice that the total number of upgrades is much higher than that of the downgrades, due to the low arrival rate that makes it easy for users to be admitted by the network.

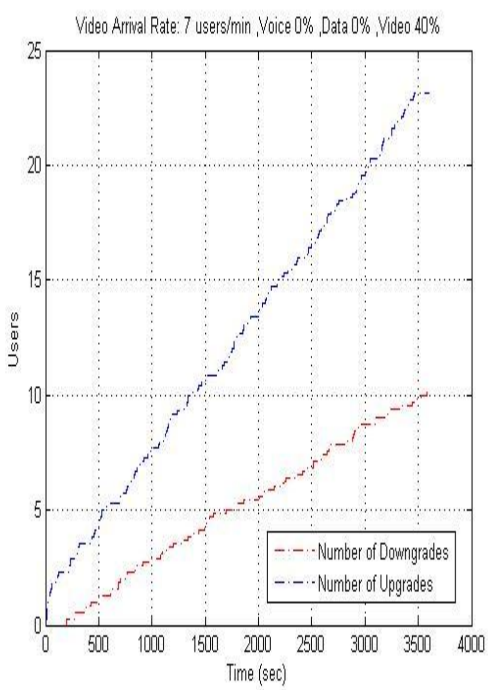


Figure 12

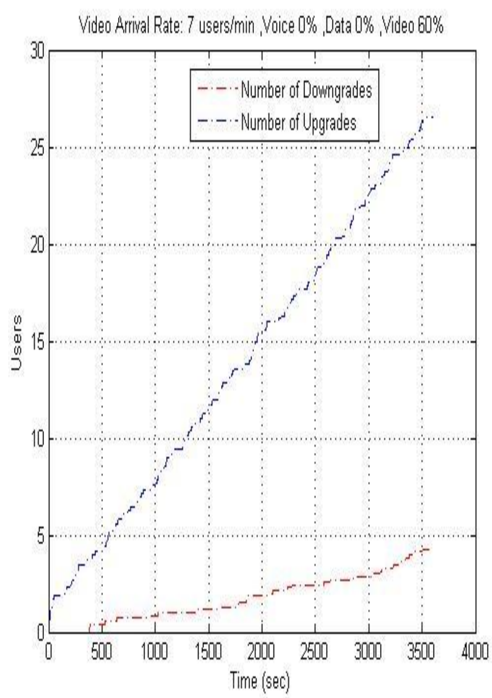


Figure 13

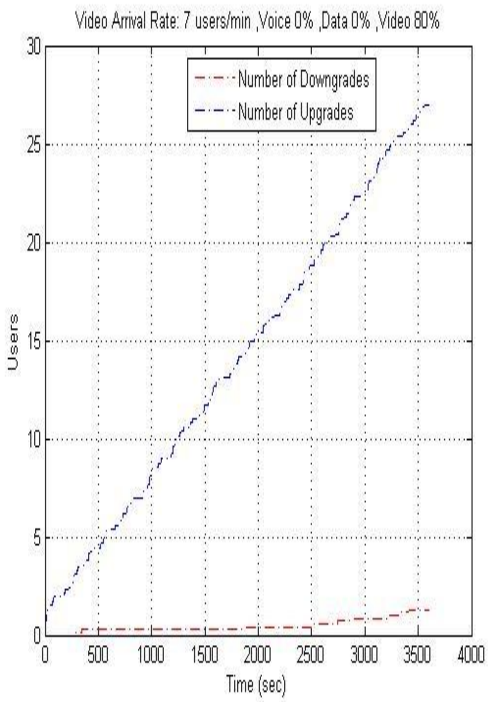


Figure 14

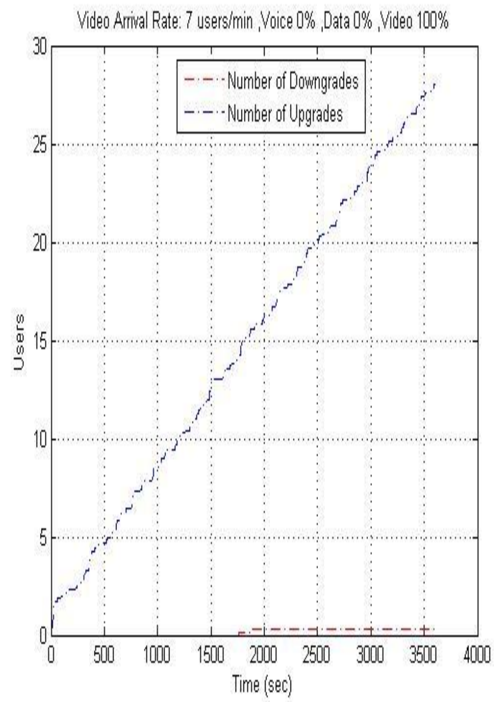


Figure 15

Figures 16-19 illustrate the number of upgrades and degradations of mpeg-4 movies for a video arrival rate equal to 5 users/minute/cell. Figure 16 shows that when the available bandwidth (40%) for video terminals is small, the CAC mechanism degrades many users in order to allow more video terminals to enter the network. When the maximum allowable video load increases, we notice that the total number of upgrades is larger than the total number of degradations. Generally the increase in available bandwidth leads both to the increase of the total number of video users who enter the network and to the decrease of the total number of video users who are downgraded.

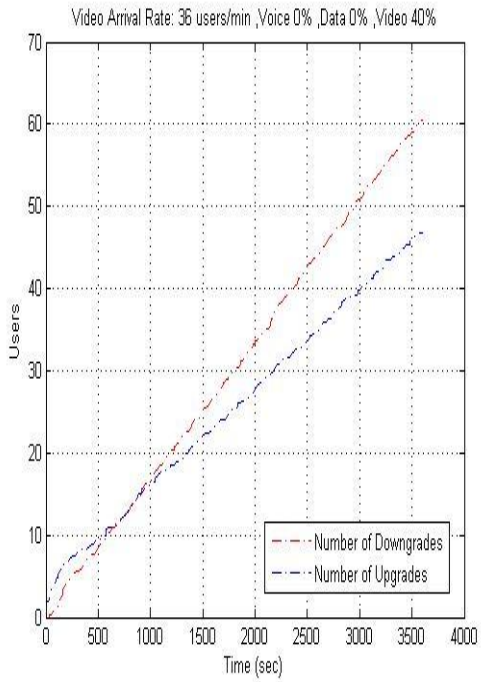


Figure 16

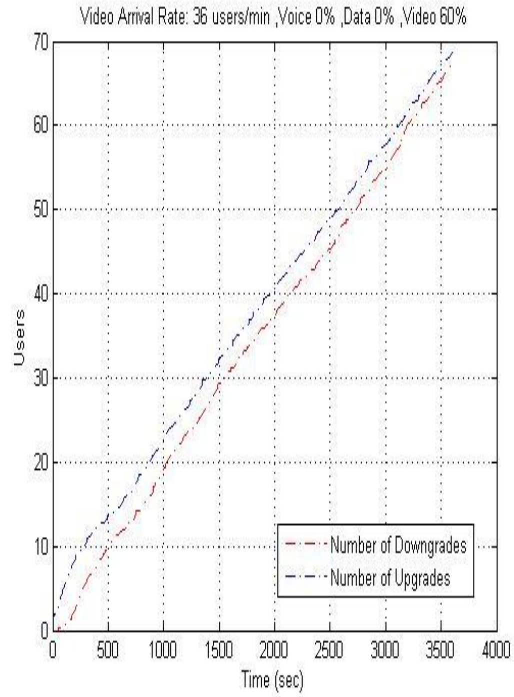


Figure 17

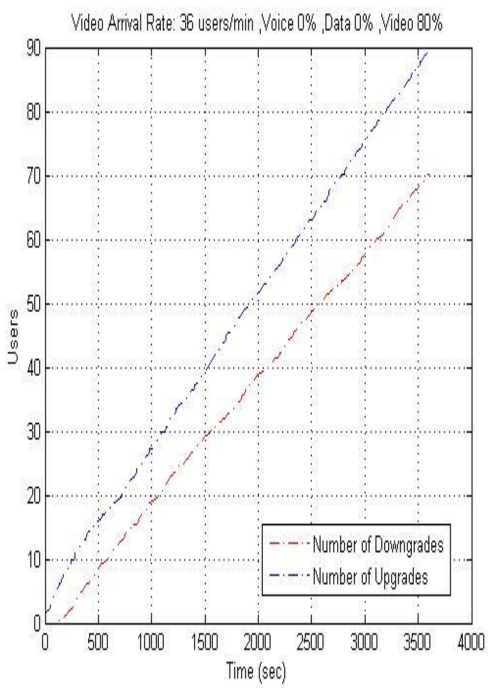


Figure 18

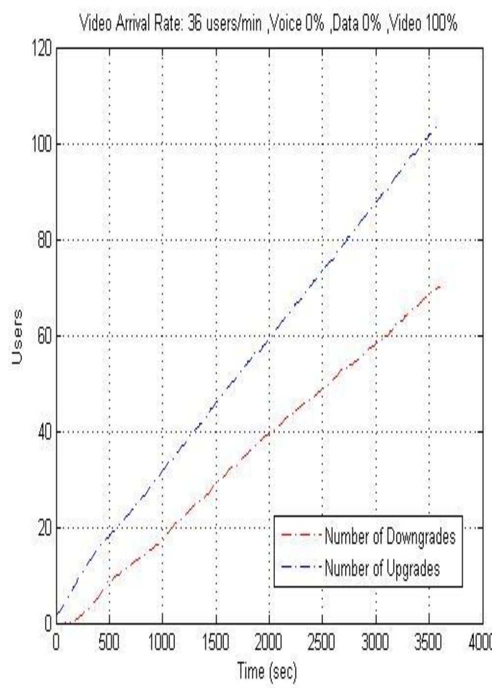


Figure 19

Figures 20 to 23 illustrate the number of upgrades and degradations of mpeg-4 movies for a video arrival rate equal to 10 users/minute/cell. We can see that when the available bandwidth is less than 100%, the total number of degradations is larger than the total number of upgrades. The high video arrival rate forces the CAC mechanism to downgrade all the users, who can be downgraded based on their contract to allow more users to enter the network. Only in the case that the allowable video load becomes equal to 100% of the total bandwidth, the number of upgrades is very close to the total number of degradations (Figure 23).

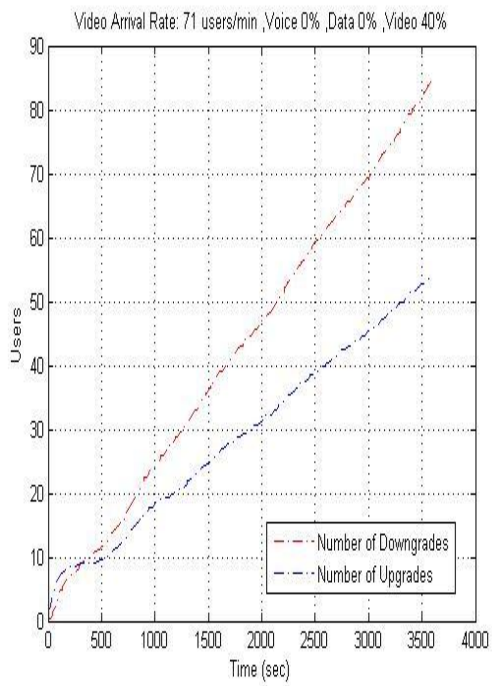


Figure 20

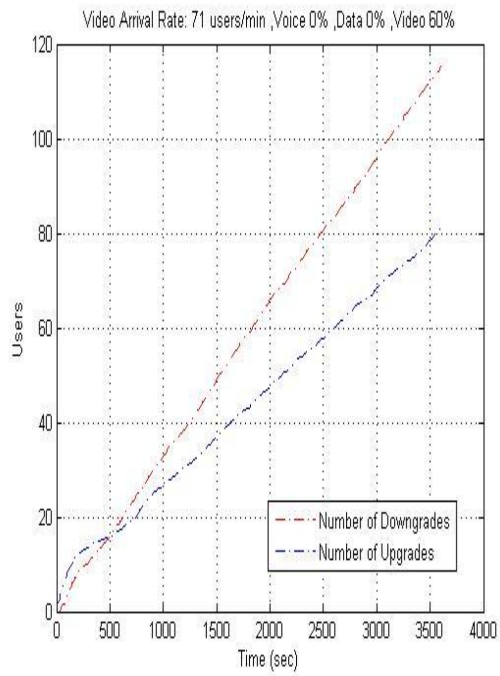


Figure 21

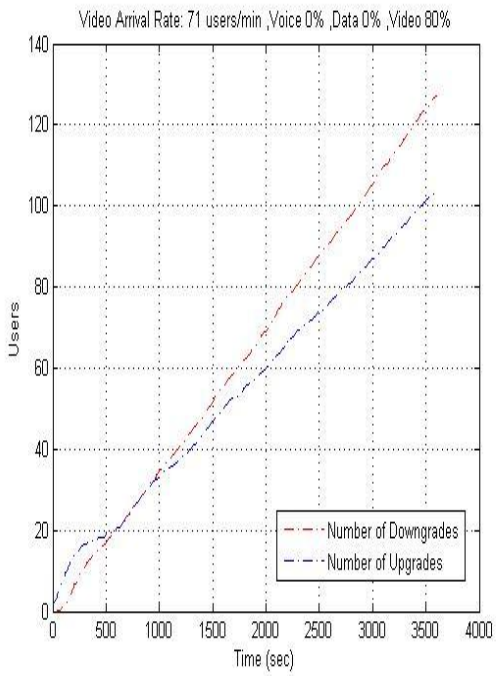


Figure 22

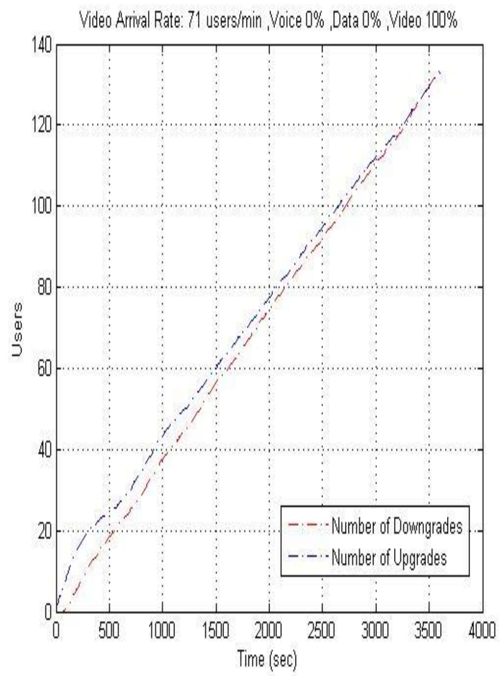


Figure 23

Figures 24-31 illustrate the number of active video users that are admitted into the system and the total number of video users who have attempted to enter the system since the beginning of the simulation (either successfully or their requests have been rejected by CAC mechanism). With the term "active users" we refer to all video terminals who have managed to enter successfully the network since the beginning of the simulation. The bigger the distance between the two graphs, the less are the terminals who succeeded in entering the network. This distance is obviously larger in Figures 28-31 than in Figures 24-27, as the arrival rate λ is ten times larger in the former than in the later, hence leading to the rejection many more requests.

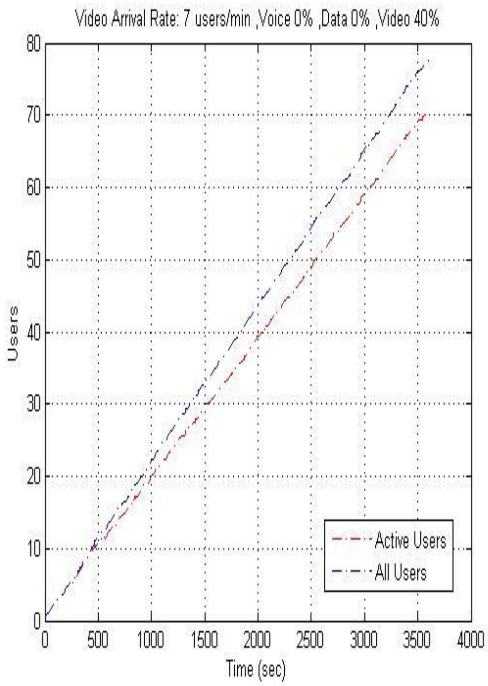


Figure 24

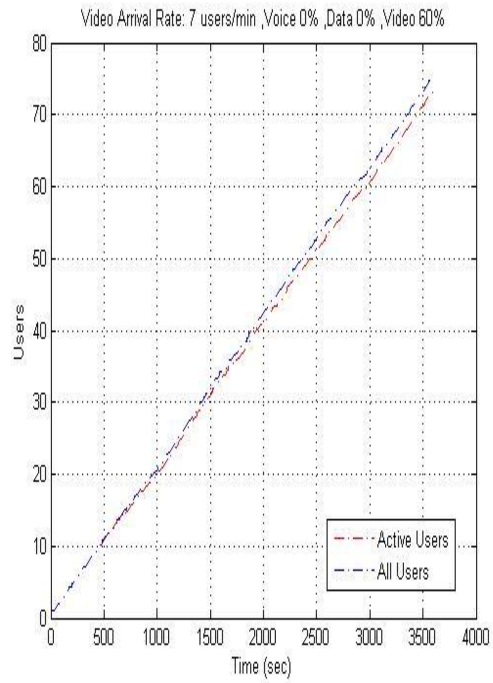


Figure 25

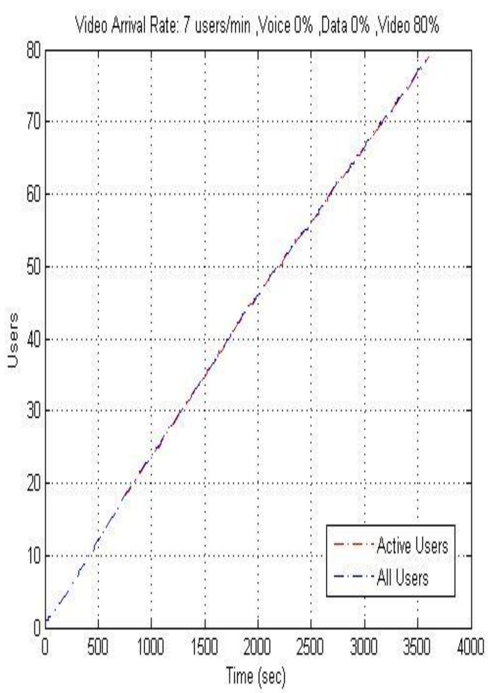


Figure 26

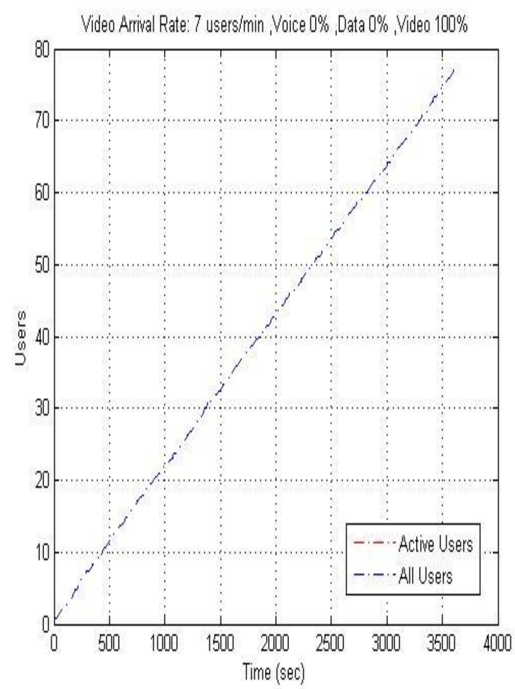


Figure 27

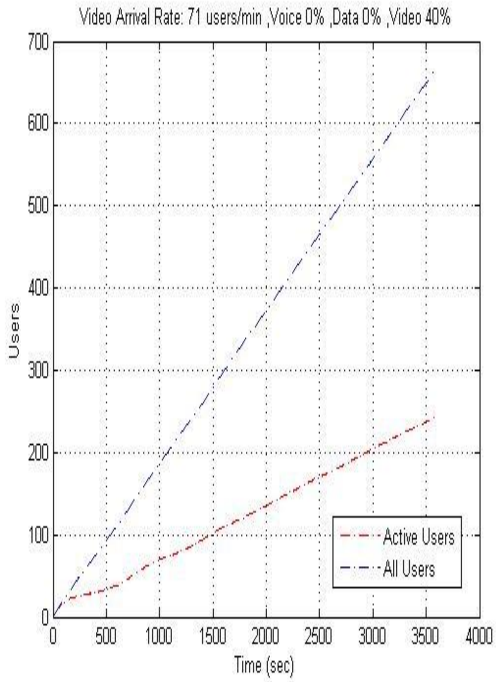


Figure 28

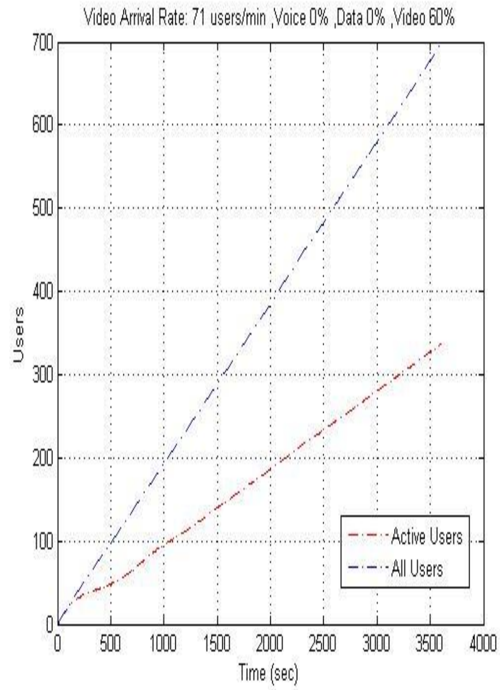


Figure 29

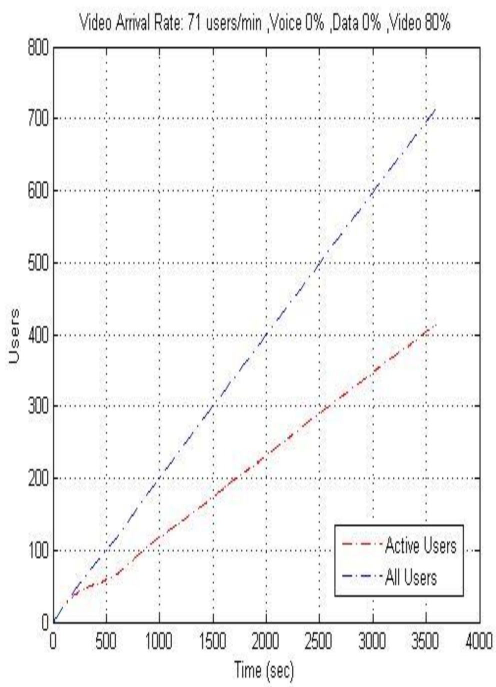


Figure 30

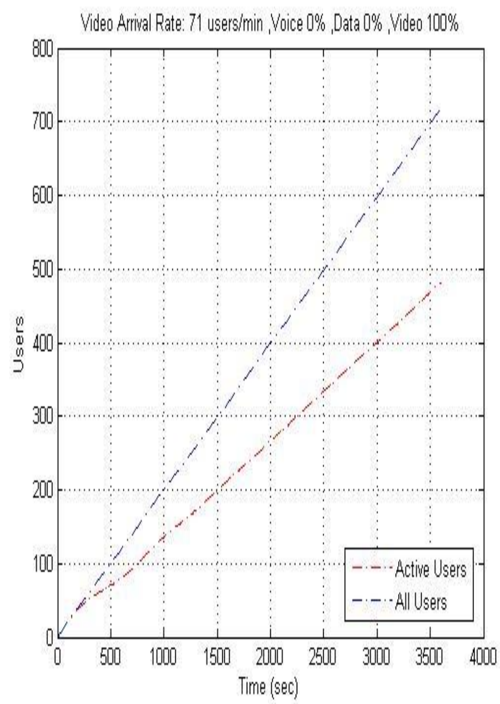


Figure 31

4.1.4 Results for Data Terminals

In this section we focus on the study of the influence of the CAC mechanism on data terminals. In the simulations that follow, the number of voice terminals is constant and equal to 3738 (534 per cell) and they take up bandwidth equal to 40% of the channel bandwidth in each cell (8 Mbps). The average video arrival rate is equal to 10 users/minute/cell, and the average sojourn of a video terminal in the network is equal to 10 minutes. The maximum allowable bandwidth for video users is assumed equal to 40% of the channel bandwidth (8 Mbps). Therefore, voice and video terminals can cover up to 80% of the channel bandwidth, but, due to the conservatism of CAC algorithm, the real bandwidth that is consumed by video terminals is significantly smaller. This is confirmed by the results presented in the fourth column of Table 12; voice and video users consume about 63% of the channel bandwidth, therefore data terminals, theoretically, could use the remaining 37%. However, regardless of the data load, data users never consume more than 18.4% of the total bandwidth, as the CAC mechanism does not allow more than 100–80 = 20% bandwidth utilization for data. Still, the proposed mechanism is shown (last column of the Table) to be able to guarantee the QoS of data users, as none of the messages exceeds the transmission delay limit of 5 seconds (messages are transmitted within less than half a second). It needs to be emphasized, however, that these results concern only the data users who are admitted in the network.

Data Arrival Rate (Messages/sec)	CAC Bandwidth Estimation (Mbps)	Real Bandwidth for Data Users (Mbps)	Voice+Video Real Bandwidth (Mbps)	Avg Message delay (msec)	Number of messages with excessive delay
50 (8%)	1.72	1.61 (8.1%)	12.64 (63.2%)	139.3	0
70 (12%)	2.41	2.19 (11%)	12.68 (63.4%)	183.1	0
90 (15%)	3.07	2.72 (13.6%)	12.62 (63.1%)	223.9	0
110 (18%)	3.62	3.17 (15.9%)	12.58 (62.9%)	255.7	0
130 (22%)	3.98	3.47 (17.4%)	12.68 (63.4%)	290.1	0
150 (25%)	4.17	3.59 (18%)	12.66 (63.3%)	292	0
170 (29%)	4.26	3.66 (18.3%)	12.54 (62.7%)	292.7	0
190 (32%)	4.3	3.68 (18.4%)	12.55 (62.8%)	297.9	0

Table 15. Used bandwidth by data, voice and video terminals for different data loads.

Figures 32-35 present the active data terminals that are admitted in the network and the total number of data terminals (including those whose requests were rejected by the CAC mechanism). The larger the distance between the two lines, the fewer terminals managed to enter the network. It is clear from the Figures that, as the data message arrival rate increases, the number of rejected data users increases significantly as well.

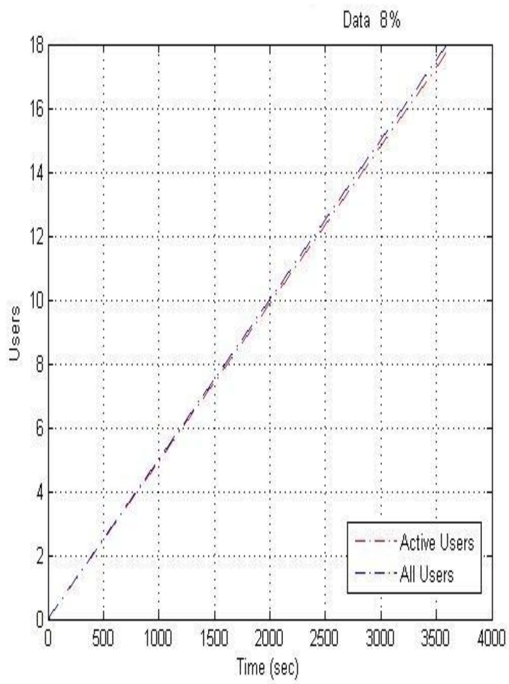


Figure 32

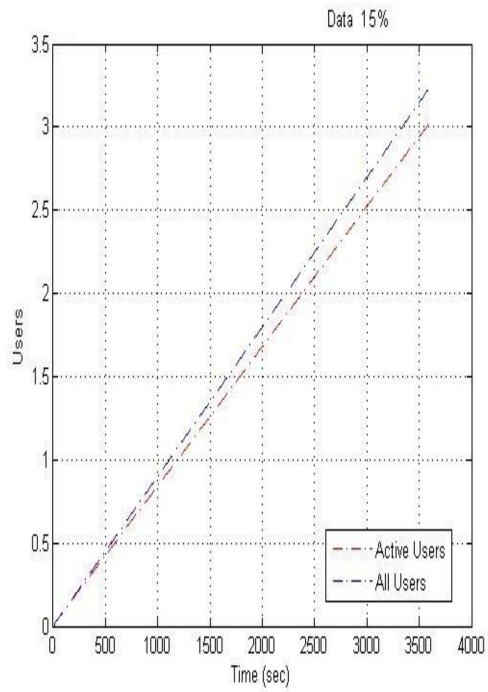


Figure 33

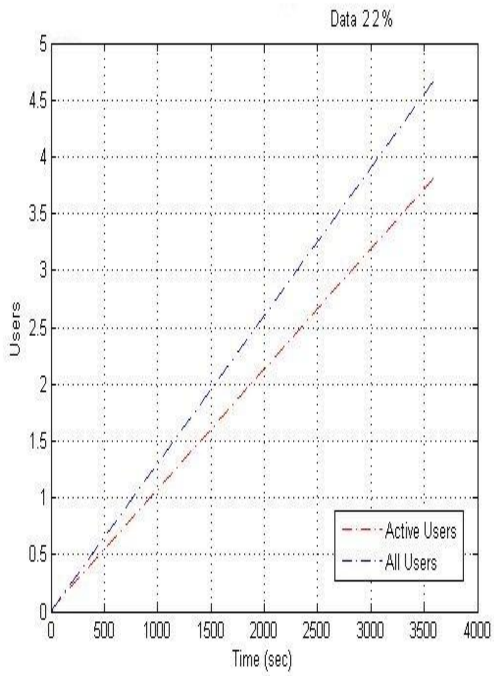


Figure 34

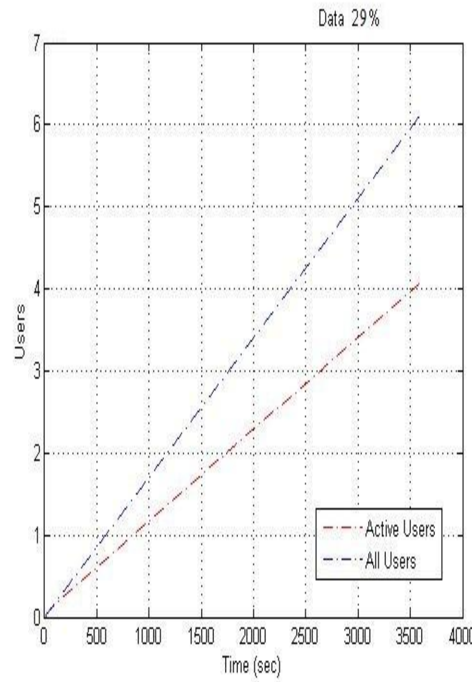


Figure 35

The results presented in Table 16, show that the increase in data message arrival rate leads initially to an increase in rejected handoff video requests. When data arrival rate becomes greater than 110 messages/second/cell, the percentage of rejected video requests is stabilized around 62-64%. The reason, as explained earlier, is that the maximum available bandwidth for data users cannot exceed 20%; hence, for low data message arrival rates, data users occupy almost all of the bandwidth they request, thus leading to an increase of rejected video requests. However, for higher data arrival rates, the CAC mechanism rejects the additional data requests, keeping the data transmission rate stable. As a result, the percentage of rejected video handoff calls is stabilized. The same is true for the percentage of downgraded video users. The comparison of the results presented in Table 16 with those in Table 17, containing the respective results from [41], shows the significant improvement offered by the adaptive bandwidth reservation scheme for data loads larger than 15% of the total bandwidth. The comparison of Tables 18 and 19 shows that the use of the adaptive bandwidth reservation scheme improves the system performance also for non-handoff video calls. The reason is that, as users move from cell to cell and at times out of the 7-cell topology, extra bandwidth becomes available for non-handoff video users.

Data Arrival Rate (Messages/sec)	Hand Off Video Users without Degradation	Hand Off Video Users Degraded by CAC	Hand Off Video Users rejected by CAC
50 (8%)	38.4%	7.2%	53.6%
70 (12%)	32.58%	5.51%	55.12%
90 (15%)	35.71%	6.35%	57.14%
110 (18%)	33.91%	2.61%	62.61%
130 (22%)	34.75%	3.39%	60.17%
150 (25%)	31.15%	3.28%	64.75%
170 (29%)	33.83%	2.26%	63.91%
190 (32%)	31.5%	3.94%	64.57%

Table 16. Influence of the data message arrival rate on handoff video calls (for λ=10 video terminals/minute/cell).

Data Arrival Rate (Messages/sec)	Hand Off Video Users without Degradation	Hand Off Video Users Degraded by CAC	Hand Off Video Users rejected by CAC
50 (8%)	44.11%	8.82%	47.05%
70 (12%)	40.35%	10.52%	49.12%
90 (15%)	33.33%	7.94%	58.73%
110 (18%)	29.23%	4.62%	66.15%
130 (22%)	26.22%	3.28%	70.49%
150 (25%)	26.14%	4.29%	69.57%
170 (29%)	24.13%	3.45%	72.41%
190 (32%)	24.57%	4.76%	70.70%

Table 17. Influence of the data message arrival rate on handoff video calls without adaptive bandwidth reservation (for only one cell and λ=10 video terminals/minute).

Data Arrival Rate (Messages/sec)	Non Hand Off Video Users without Degradation	Non Hand Off Video Users Degraded by CAC	Non Hand Off Video Users rejected by CAC
50 (8%)	29.86%	3.6%	66.37%
90 (15%)	32.21%	3.37%	64.42%
130 (22%)	32.85%	3.47%	63.5%
170 (29%)	32.43%	3.44%	63.95%

Table 18. Influence of the data message arrival rate on non-handoff video calls (for λ=10 video terminals/minute).

Data Arrival Rate (Messages/sec)	Non Hand Off Video Users without Degradation	Non Hand Off Video Users Degraded by CAC	Non Hand Off Video Users rejected by CAC
50 (8%)	21.36%	1.46%	77.18%
90 (15%)	26.22%	3%	70.79%
130 (22%)	27.09%	2.41%	70.5%
170 (29%)	26.4%	3%	70.6%

Table 19. Influence of the data message arrival rate on non-handoff video calls without adaptive bandwidth reservation (for only one cell and λ=10 video terminals/minute).

4.2 **Second** *Implementation of Video Users' Downgrades/Upgrades*

In this implementation, as explained in Section 3.1, downgrades take place by a selective transmission of the video frames of a GoP. Given that, in [41] it was shown that the second implementation clearly outperformed the first, we will indicatively present in the following Sections some of the respective results, in order to focus more, in Section 4.3, on the results with the use of the FGC scheme. The results in Tables 20, 21 have been derived in the same way as Tables 3, 4 for the first downgrade/upgrade implementation.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	CAC Bandwidth Estimation (Mbps)	Real Bandwidth (Mbps)	Dropped Voice Packets (%)	Number of messages with excessive delay	Dropped Video Packets (%)
214 (16%)	24 (4%)	4 (20%)	7.85	7.18 (35.9%)	0.002379	0	0
267 (20%)	24 (4%)	3.2 (16%)	7.88	7.27 (36.35%)	0.00001174	0	0
320 (24%)	24 (4%)	2.4 (12%)	7.9	7.42 (37.1%)	0.000001048	0	0
267 (20%)	30 (5%)	5 (25%)	9.83	8.94 (44.7%)	0.0001115	0	0
334 (25%)	30 (5%)	4 (20%)	9.84	9.15 (45.75%)	0.00002681	0	0
400 (30%)	30 (5%)	3 (15%)	9.89	9.3 (46.5%)	0.00002213	0	0
320 (24%)	36 (6%)	6 (30%)	11.78	10.83 (54.15%)	0.001854	0	0
400 (30%)	36 (6%)	4.8 (24%)	11.81	10.99 (54.95%)	0.001242	0	0
480 (36%)	36 (6%)	3.6 (18%)	11.86	11.19 (55.95%)	0.0003603	0	0
374 (28%)	42 (7%)	7 (35%)	13.71	12.63 (63.15%)	0.0372	0	0
467 (35%)	42 (7%)	5.6 (28%)	13.8	12.87 (64.35%)	0.042619	0	0
560 (42%)	42 (7%)	4.2 (21%)	13.85	13.13 (65.65%)	0.01875	0	0
427 (32%)	48 (8%)	8 (40%)	15.7	14.36 (71.8%)	0.4079	0	0.0003689
534 (40%)	48 (8%)	6.4 (32%)	15.81	14.78 (73.9%)	0.2549	0	0.00004535
640 (48%)	48 (8%)	4.8 (24)	15.88	15.01 (75.05%)	0.08941	0	0
480 (36%)	54 (9%)	9 (45%)	17.69	16.19 (80.95%)	1.063	18.57	0.002944
600 (45%)	54 (9%)	7.2 (36%)	17.8	16.31 (81.55%)	0.7208	16.57	0.0002095
720 (54%)	54 (9%)	5.4 (27%)	17.89	16.76 (83.8%)	0.4338	23.14	0
534 (40%)	59 (10%)	10 (50%)	19.27	17.34 (86.7%)	3.677	21.57	0.06229
667 (50%)	59 (10%)	8 (40%)	19.47	17.79 (88.95%)	2.33	45.57	0.0002985
800 (60%)	59 (10%)	6 (30%)	19.57	18.13 (90.65%)	1.673	63.29	0

Table 20. Simulation scenarios with loads between 40%-100% of the total bandwidth.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Voice Real Bandwidth	Data Real Bandwidth	Video Real Bandwidth
214 (16%)	24 (4%)	4 (20%)	16.09%	3.91%	15.89%
267 (20%)	24 (4%)	3.2 (16%)	20.06%	3.95%	12.35%
320 (24%)	24 (4%)	2.4 (12%)	24.07%	3.96%	9.09%
267 (20%)	30 (5%)	5 (25%)	20.08%	4.95%	19.69%
334 (25%)	30 (5%)	4 (20%)	25.11%	4.94%	15.69%
400 (30%)	30 (5%)	3 (15%)	30.08%	4.9%	11.55%
320 (24%)	36 (6%)	6 (30%)	24.05%	5.86%	24.22%
400 (30%)	36 (6%)	4.8 (24%)	30.06%	5.93%	18.95%
480 (36%)	36 (6%)	3.6 (18%)	36.08%	5.97%	13.94%
374 (28%)	42 (7%)	7 (35%)	28.11%	6.91%	28.14%
467 (35%)	42 (7%)	5.6 (28%)	35.08%	6.84%	22.43%
560 (42%)	42 (7%)	4.2 (21%)	42.1%	6.87%	16.69%
427 (32%)	48 (8%)	8 (40%)	31.97%	7.73%	32.1%
534 (40%)	48 (8%)	6.4 (32%)	40.05%	7.87%	25.98%
640 (48%)	48 (8%)	4.8 (24)	48.11%	7.82%	19.12%
480 (36%)	54 (9%)	9 (45%)	35.74%	8.39%	36.84%
600 (45%)	54 (9%)	7.2 (36%)	44.78%	8.38%	28.41%
720 (54%)	54 (9%)	5.4 (27%)	53.9%	8.35%	21.55%
534 (40%)	59 (10%)	10 (50%)	38.67%	7.66%	40.4%
667 (50%)	59 (10%)	8 (40%)	48.97%	7.43%	32.57%
800 (60%)	59 (10%)	6 (30%)	59.15%	7.33%	24.17%

Table 21. Bandwidth that is really committed by each type of traffic, with loads between 40%-100% of the total bandwidth.

The comparison of the results of Tables 20-21 with those of Tables 3-4 shows a significant improvement (4%-8%) in the utilization of the total channel bandwidth. This increase comes at the cost of worse QoS for all users; however, we note that, even for a channel load equal to 80%, the QoS requirements of voice terminals are not violated, while the QoS requirements for video terminals are not violated for channel loads up to 90%. Regarding data users, when the channel load is equal or greater than 90%, a large number of messages is not transmitted before their deadline expires.

In Figure 36 we present graphically the CAC estimation for the required bandwidth (average of three combinations for each traffic load) as well as the real bandwidth that users need. It is clear, once again, that the CAC estimation is much closer to the actual bandwidth needs of the users. There are two reasons for which the second implementation excels in comparison to the first:

a) The selective transmission of just I and P frames (for MQ users) and just I frames (for HQ users) leads to a significant decrease in the value of the standard deviation σ, hence the estimation with (2) becomes more accurate and less bandwidth is lost due to the conservatism of the estimation.

b) The MQ and the LQ versions of the traces, in the first implementation case, contain I, P and B frames which have smaller sizes than the respective I, P and B frames of the HQ version. On the contrary, in the case of the second implementation, B frames are dropped but the I and P frames which are used by MQ and LQ users are those of the initial HQ trace, therefore they are larger in size.

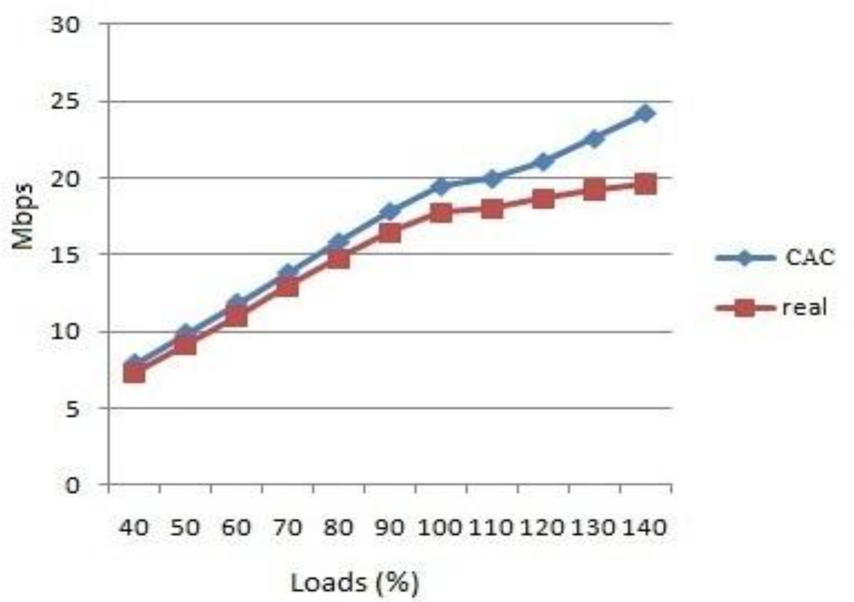


Figure 36. Comparison between real bandwidth needs and the respective CAC estimation for different loads.

4.3 FGC Scheme Using the Second Implementation of Video Users' Upgrades/Downgrades

4.3.1 Results on Voice, Data and Video Users

In order to avoid repetition, we only present our results for the second implementation of video users' upgrades/downgrades, and we comment on the differences with the results of the first implementation, which we have also derived.

Similarly to Chapter 4.1.1 we used transmission rates for voice, data and video which correspond to: 40%, 60%, 80%, 100%, 120% and 140% of the channel bandwidth. For each one of the above-mentioned 6 rates we created 3 different traffic combinations for voice, data and video (each combination generates the same traffic load). The rates and their combinations are shown in Tables 22 and 23. In all of these traffic scenarios we assume that the number of voice users is constant during the simulation. The email arrival rate changes in each scenario; we assume a high average arrival rate of video users in all of the scenarios, equal to 10 users per minute.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	CAC Bandwidth Estimation (Mbps)	Real Bandwidth (Mbps)	Dropped Voice Packets (%)	Number of messages with excessive delay	Dropped Video Packets (%)
214 (16%)	24 (4%)	4 (20%)	8	6.46 (32.3%)	0.000001045	0	0
267 (20%)	24 (4%)	3.2 (16%)	8.14	6.82 (34.1%)	0.0000004193	0	0
320 (24%)	24 (4%)	2.4 (12%)	8.25	7.13 (35.6%)	0.0000006993	0	0
320 (24%)	36 (6%)	6 (30%)	11.75	9.8 (49%)	0.0007126	0	0.00001858
400 (30%)	36 (6%)	4.8 (24%)	11.81	10.16 (50.8%)	0.0001396	0	0
480 (36%)	36 (6%)	3.6 (18%)	12.19	10.62 (53.1%)	0.0007579	0	0
427 (32%)	48 (8%)	8 (40%)	15.32	12.99 (64.9%)	0.01501	0	0.00002823
534 (40%)	48 (8%)	6.4 (32%)	15.71	13.54 (67.7%)	0.009521	0	0
640 (48%)	48 (8%)	4.8 (24)	15.93	14.17 (70.8%)	0.02611	0	0
534 (40%)	59 (10%)	10 (50%)	18.2	15.8 (79%)	0.6902	0	0.004425
667 (50%)	59 (10%)	8 (40%)	18.7	16.4 (82%)	0.5294	18.14	0.00003933
800 (60%)	59 (10%)	6 (30%)	19.09	17.14 (85.7%)	0.322	26.71	0

Table 22. Simulation scenarios with loads between 40%-100% of the total bandwidth.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Voice Real Bandwidth	Data Real Bandwidth	Video Real Bandwidth
214 (16%)	24 (4%)	4 (20%)	16.1%	3.92%	12.27%
267 (20%)	24 (4%)	3.2 (16%)	20.07%	3.95%	10.07%
320 (24%)	24 (4%)	2.4 (12%)	24.06%	3.93%	7.67%
320 (24%)	36 (6%)	6 (30%)	24.06%	5.9%	19.03%
400 (30%)	36 (6%)	4.8 (24%)	30.07%	5.87%	14.88%
480 (36%)	36 (6%)	3.6 (18%)	36.09%	5.87%	11.16%
427 (32%)	48 (8%)	8 (40%)	32.09%	7.82%	25.05%
534 (40%)	48 (8%)	6.4 (32%)	40.14%	7.88%	19.68%
640 (48%)	48 (8%)	4.8 (24)	48.11%	7.81%	14.95%
534 (40%)	59 (10%)	10 (50%)	39.85%	8.77%	30.37%
667 (50%)	59 (10%)	8 (40%)	49.87%	8.15%	23.97%
800 (60%)	59 (10%)	6 (30%)	59.9%	7.72%	18.04%

Table 23. Bandwidth that is really committed by each type of traffic, with loads between 40%-100% of the total bandwidth.

The comparison of the results of Tables 22-23 with those of the FGC scheme when using the first implementation of video users' upgrades/downgrades shows a significant improvement (2.7%-4.9%) in the utilization of the total channel bandwidth. This increase comes at the cost of worse QoS for all users; however, we note that, even for a channel load equal to 70%, the QoS requirements of voice terminals are not violated, while the QoS requirements for video terminals are not violated in any of the studied scenarios. Regarding data users, when the channel load is equal or greater than 100%, a significant number of messages is not transmitted before their deadline expires.

However, the comparison of the results of Tables 22-23 with those of Tables 20-21 shows a considerable decrease (1.5%-7.7%) in the utilization of the total channel bandwidth. Comparing Table 20 to Table 22, we notice that the decrease on the exploitation of the channel bandwidth by video terminals using the FGC scheme leads to better QoS for all types of users whose calls are accepted.

In Figure 37 we present graphically the FGC CAC estimation for the required bandwidth (average of three combinations for each traffic load) as well as the real bandwidth that users need. It is clear, once again, that the FGC estimation is much closer to the actual bandwidth needs of the users. The reasons for which the second implementation excels in comparison to the first are (as explained in Section 4.2.1.) the selective transmission of just I and P frames and the larger size of these selectively transmitted video frames. However, by comparing Figure 37 to Figure 36 we notice that the first CAC mechanism's estimation is much closer to the actual bandwidth needs of the users than the FGC estimation.

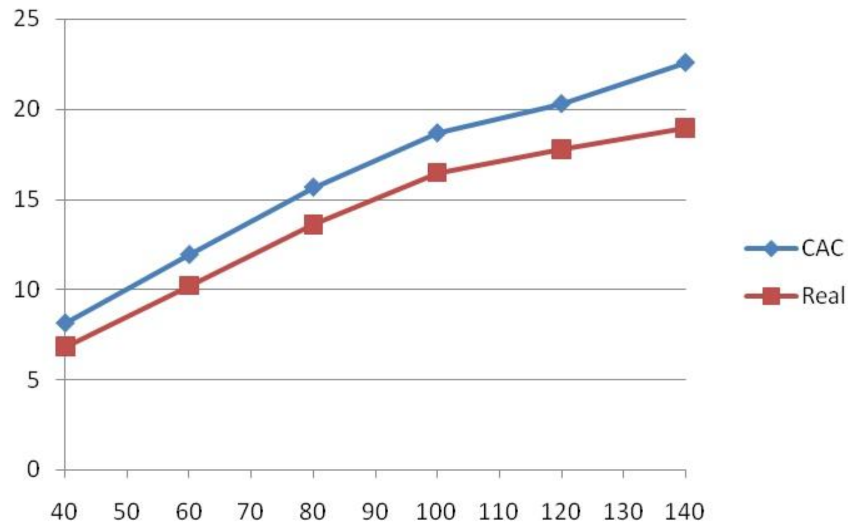


Figure 37. Comparison between real bandwidth needs and the respective CAC estimation for different loads.

Finally, we compare the case where road map-based adaptive bandwidth reservation is used, with the case studied in [41] (only one cell, without adaptive bandwidth reservation). For this comparison we use again three metrics:

a. The percentage of video users who enter the network without degradation.

b. The percentage of video users who enter the network either by being degraded or due to the degradation of other video users.

c. The percentage of video users who do not manage to enter the network, due to the lack of adequate bandwidth.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Handoff Users without Degradation	Handoff Users Degraded by CAC	Handoff Users rejected by CAC
214 (16%)	24 (4%)	4 (20%)	53.25%	6.49%	40.26%
267 (20%)	24 (4%)	3.2 (16%)	48.68%	6.58%	44.74%
320 (24%)	24 (4%)	2.4 (12%)	32.86%	4.29%	61.43%
320 (24%)	36 (6%)	6 (30%)	75%	9.52%	15.48%
400 (30%)	36 (6%)	4.8 (24%)	57.83%	8.43%	32.53%
480 (36%)	36 (6%)	3.6 (18%)	48%	4%	46.66%
427 (32%)	48 (8%)	8 (40%)	79.35%	9.78%	9.78%
534 (40%)	48 (8%)	6.4 (32%)	77.65%	9.41%	12.94%
640 (48%)	48 (8%)	4.8 (24)	61.36%	6.82%	30.68%
534 (40%)	59 (10%)	10 (50%)	87.5%	4.81%	7.69%
667 (50%)	59 (10%)	8 (40%)	76.53%	6.12%	16.33%
800 (60%)	59 (10%)	6 (30%)	69.05%	4.76%	25%
Average			63.92%		

Table 24. Metrics for handoff video terminals for different loads, using adaptive bandwidth reservation.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Non Handoff Users without Degradation	Non Handoff Users Degraded by CAC	Non Handoff Users rejected by CAC
214 (16%)	24 (4%)	4 (20%)	6.33%	0%	93.49%
267 (20%)	24 (4%)	3.2 (16%)	5.35%	0%	94.65%
320 (24%)	24 (4%)	2.4 (12%)	4.63%	0%	95.19%
320 (24%)	36 (6%)	6 (30%)	7.03%	0%	92.79%
400 (30%)	36 (6%)	4.8 (24%)	7.77%	0 %	92.04%
480 (36%)	36 (6%)	3.6 (18%)	5.03%	0%	94.79%
427 (32%)	48 (8%)	8 (40%)	10.24%	0%	89.57%
534 (40%)	48 (8%)	6.4 (32%)	8.11%	0%	91.89%
640 (48%)	48 (8%)	4.8 (24)	6.73%	0%	93.09%
534 (40%)	59 (10%)	10 (50%)	18.07%	0%	81.93%
667 (50%)	59 (10%)	8 (40%)	16.03%	0%	83.79%
800 (60%)	59 (10%)	6 (30%)	13.69%	0%	86.31%
Average			9.08%		

Table 25. Metrics for non handoff video terminals, using adaptive bandwidth reservation.

The comparison of the results of Table 24 with those of Table 25 shows that the use of the FGC scheme and the adaptive bandwidth reservation scheme leads to a higher percentage of handoff video users being accepted without degradation and a smaller percentage of handoff video users being rejected. It is also useful to compare the results of FGC with those of the first CAC mechanism, in terms of the mean percentage of accepted handoff and non-handoff users without degradation. The respective mean percentages are (63.92%, 9.08%) and (24.82%, 17.24%) for FGC and for the first CAC mechanism respectively.

These results show that, as expected from the qualitative comparison of the two mechanisms made in Section 3.2., the FGC scheme is even more conservative than the first CAC mechanism, which results in much better QoS for handoff users (more bandwidth is left available for them to use, hence they are more easily accepted) and worse QoS for non-handoff users.

4.3.2 Results for Video Terminals

In this section our study focuses on video terminals. For this reason, in the following simulations we assume that the only type of traffic present in the network is video. More specifically, for each traffic load, users are inserted into the network until the CAC mechanism starts rejecting them, when the maximum allowable video load is exceeded. The number of users does not remain constant during the simulation as video users move from cell to cell and can either exit the network topology or terminate their call.

We have derived results for different average arrival rates λ of video terminals and different maximum traffic loads. In Table 26, where we present results for a video arrival rate λ = 10 users/minute, we notice that the increase in the maximum allowable video load increases the real transmission rate and the average number of video users in the network. Still, the bandwidth utilized falls significantly short of the maximum allowable load; the reason is, again, the conservatism of the CAC mechanism, which however ensures (as shown in the Table) that only for a maximum allowable video load larger than 80% of the channel bandwidth are the QoS requirements of video users violated. The comparison of the results of Table 26 with those of FGC scheme using the first implementation of video users' upgrades/downgrades show a significant increase in the utilization of the total channel bandwidth and in the average number of video users. However, the comparison of the results of Table 26 with those of the first CAC mechanism using the second implementation of video users' upgrades/downgrades shows that the

FGC scheme is more conservative than the first CAC mechanism, rejecting more video requests and exploiting less bandwidth, but achieves better QoS for video terminals.

Maximum Allowed Video Load (Mbps)	Video Arrival Rate (λ) (users/min)	CAC Bandwidth Estimation (Mbps)	Real Bandwidth (Mbps)	Average Number of Users	Dropped Video Packets (%)
8 (40%)	10	7.30	4.96 (24.8%)	22	0.00004993
12 (60%)	10	10.32	7.37 (36.85%)	28	0.009696
16 (80%)	10	12.84	9.50 (47.5%)	33	0.1441
20 (100%)	10	15	11.33 (56.65%)	36	0.6967

Table 26. Average number of active users in the system with λ=10 users/minute.

In Figures 38-41, for an average video arrival rate equal to 1 user/minute/cell, and in Figures 42-45, for an average video arrival rate equal to 10 users/minute/cell, we present for various video loads the bandwidth that is really used by video terminals per second and the bandwidth that the CAC algorithm estimates as required. All of the figures show that the CAC mechanism overestimates significantly the required bandwidth, and the gap increases for higher λ. This gap is smaller than in the case of using the FGC scheme with the first implementation of video users' upgrades/downgrades, but similar or larger than using the first CAC mechanism with the second implementation of video users' upgrades/downgrades.

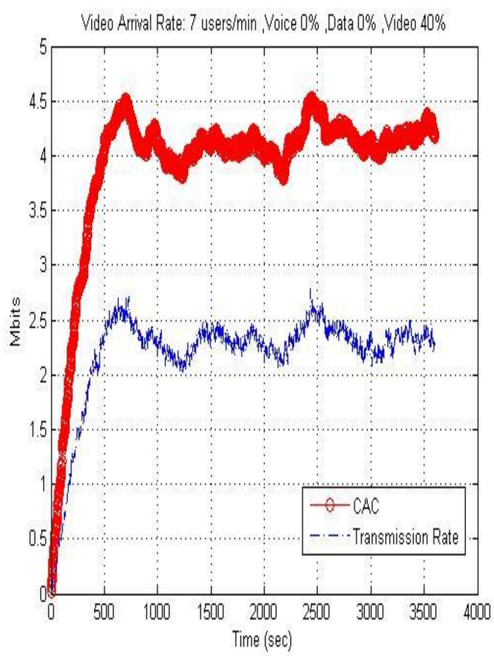


Figure 38

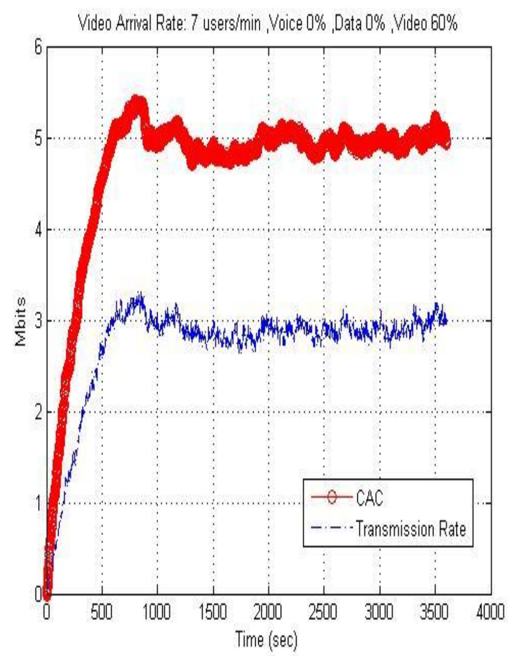


Figure 39

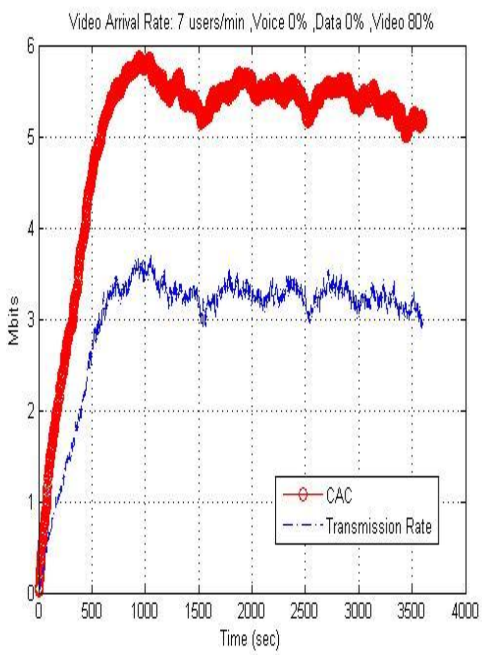


Figure 40

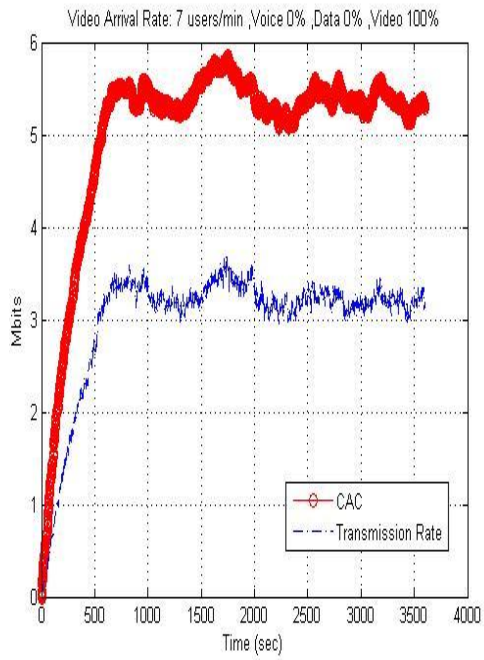


Figure 41

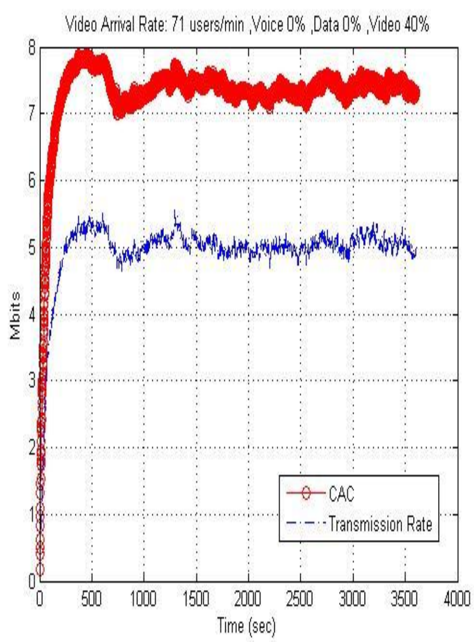


Figure 42

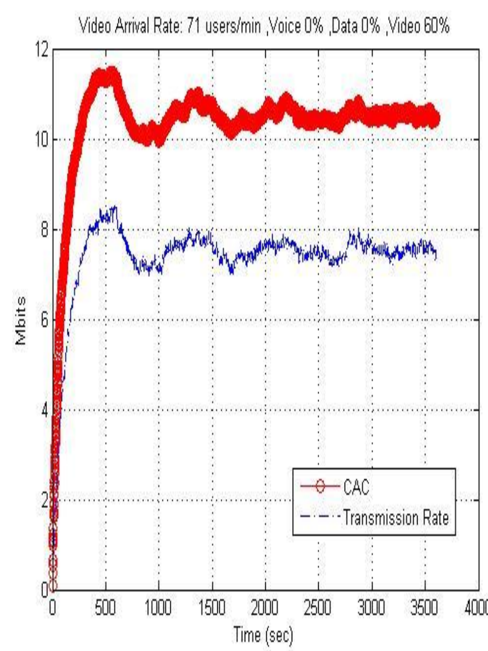
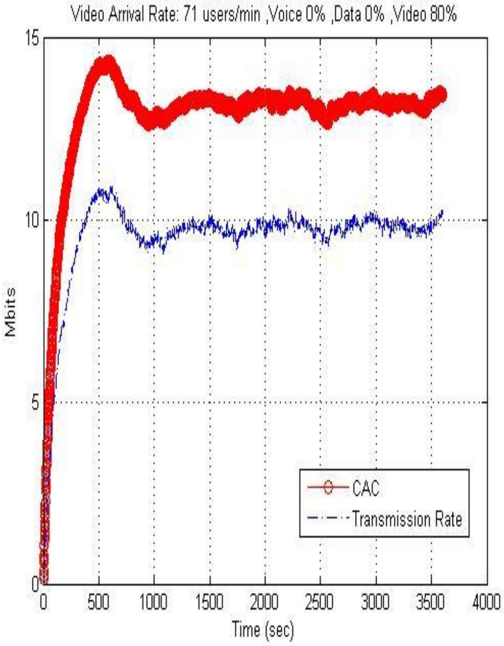


Figure 43



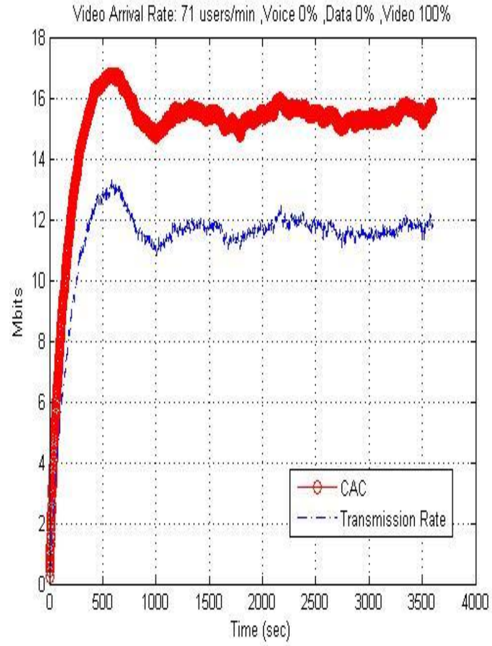


Figure 44

Figure 45

4.3.3 Degradations/Upgrades of Video Terminals

Table 27 shows results for different video loads with λ = 5 users/minute. The column "Active Users" of Table 27 presents the total number of users that have been admitted in the network. The values of the third and fourth column of Table 27 show the percentage of HQ and MQ video terminals that have been downgraded (this percentage is computed over the number of users that can be downgraded based on their contract). Values in the last two columns of the Table present the degradation ratio, which is the average percentage of time that a video user remains downgraded.

The comparison of the results of Table 27 with those of the FGC scheme using the first implementation of video users' upgrades/downgrades show that:

a) The number of video terminals that are admitted into the system is greater.
b) The percentage of HQ terminals that are downgraded and the degradation ratio are slightly lower.
c) The percentage of MQ terminals that are downgraded and the degradation ratio are similar.

The reason for all of the above results is that the FGC mechanism is conservative and rejects many video requests, hence allowing accepted HQ users a higher available bandwidth and not "pressuring" them to downgrade in order to free bandwidth for new users. This is not the case for MQ users, because the bandwidth that is freed by their degradation is not as significant as that of HQ users.

	Active Users		Downgrade percentage HQ		Downgrade percentage MQ		Degradation Ratio HQ		Degradation Ratio MQ	
Maximum Allowed Video Load (Mbps)	**FGC Scheme using 1st implementation**	**FGC Scheme using 2nd implementation**	**FGC Scheme using 1st implementation**	**FGC Scheme using 2nd implementation**	**FGC Scheme using 1st implementation**	**FGC Scheme using 2nd implementation**	**FGC Scheme using 1st implementation**	**FGC Scheme using 2nd implementation**	**FGC Scheme using 1st implementation**	**FGC Scheme using 2nd implementation**
8 (40%)	95	102	43.71%	44.29%	44.57%	47.29%	38.14%	40.57%	48.57%	47.14%
12 (60%)	128	134	26%	24%	23.57%	23%	21.14%	20.57%	28%	30.14%
16 (80%)	153	158	14.43%	14%	14.57%	14.29%	15.86%	17.29%	17.71%	16.71%
20 (100%)	176	177	10.86%	8.71%	10.14%	10.14%	10.57%	10.14%	14.14%	14.14%

Table 27. Percentage of degradations and degradation ratio for λ=5 users/minute/cell.

Table 28 shows that, for higher video arrival rates, the percentage of users that need to be degraded is significantly higher than that when λ = 5 user/minute. Also, users that are degraded remain degraded for the largest part of their call duration. Regardless, of the video arrival rate, the percentage of degradations is greater for HQ than MQ movies. This is expected due to the much higher bandwidth requirements of HQ movies; this makes it difficult to find the necessary resources to transmit HQ movies without degradation.

Maximum Allowed Video Load (Mbps)	Active Users	Downgrade percentage HQ	Downgrade percentage MQ	Degradation Ratio HQ	Degradation Ratio MQ
8 (40%)	135	74.29%	72.86%	57.43%	60%
12 (60%)	175	43.57%	45.29%	40.71%	45%
16 (80%)	212	27.29%	27.86%	28.29%	33.14%
20 (100%)	252	20.29%	21.14%	21.43%	21.71%

Table 28. Percentage of degradations and degradation ratio for λ = 10 users/minute/cell.

However, the comparison of the results of Table 27 and 28 with those of the first CAC mechanism using the second implementation of video users' upgrades/downgrades, show that the percentages of downgrades and degradation ratios of HQ video terminals are significantly lower with FGC, because of the conservatism of the mechanism (active users are much fewer).

Figures 46-57 illustrate the number of upgrades and degradations of mpeg-4 movies that correspond to the elements of Tables 27-28 for arrival rates equal to 5 and 10 users/minute/cell. In Figures 46-49 we notice that the total number of upgrades is much higher than that of the downgrades, due to the low arrival rate that makes it easy for users to be admitted by the network. These results are very close quantitatively with the results of FGC scheme using the first implementation of video users' upgrades/downgrades. However, comparing the results of Figures 46-57 to those of the first CAC mechanism using the second implementation of video users' upgrades/downgrades, we notice that the number of downgrades is significantly lower with FGC, because of the conservatism of FGC mechanism.

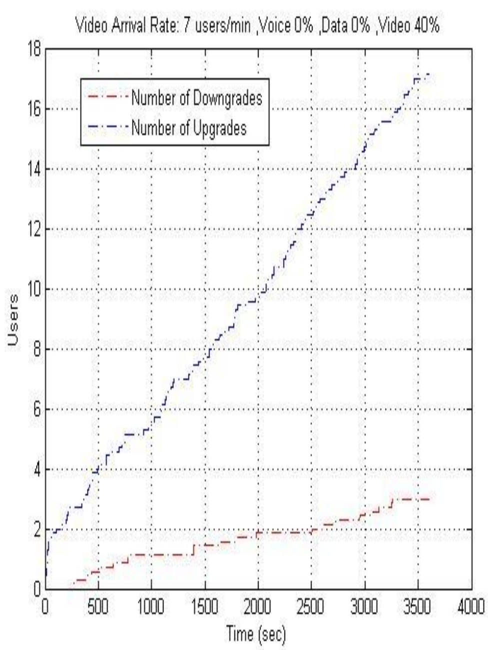


Figure 46

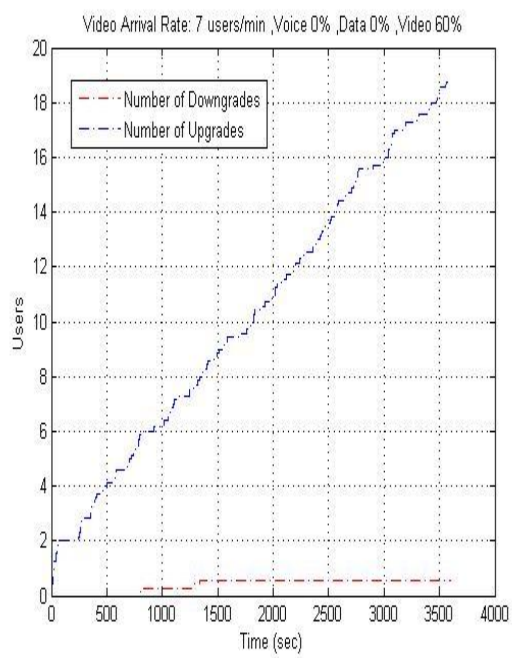


Figure 47

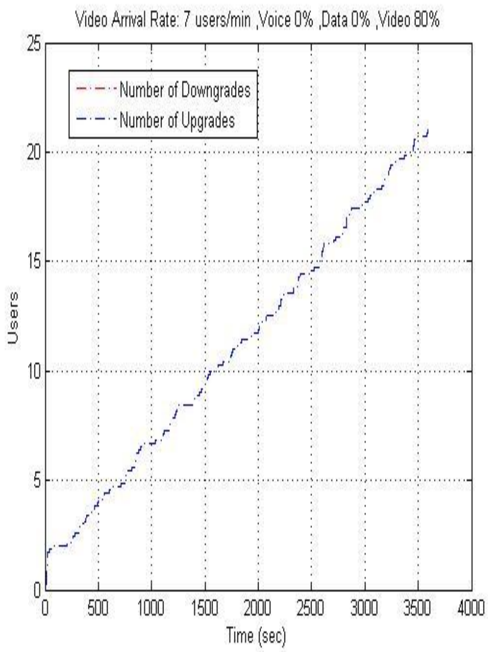


Figure 48

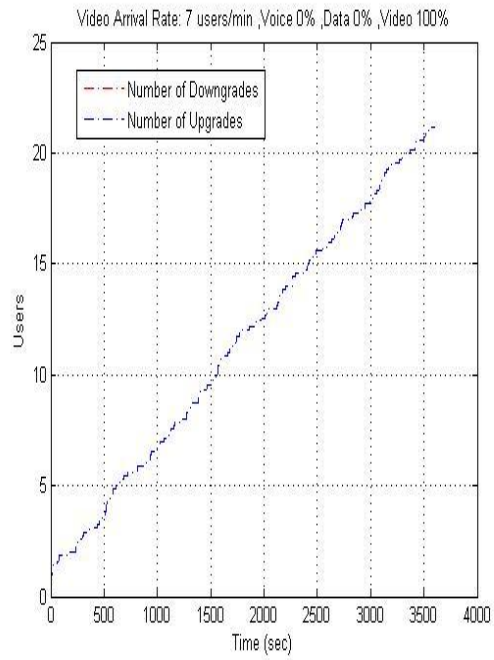


Figure 49

Figures 50-53 illustrate the number of upgrades and degradations of mpeg-4 movies for a video arrival rate equal to 5 users/minute/cell. Figure 16 shows that when the available bandwidth (40%) for video terminals is small, the CAC mechanism degrades many users in order to allow more video terminals to enter the network. When the maximum allowable video load increases, we notice that the total number of upgrades is larger than the total number of degradations. Generally, the increase in available bandwidth leads both to the increase of the total number of video users who enter the network and to the decrease of the total number of video users who are downgraded (similarly to the respective results of the first CAC mechanism).

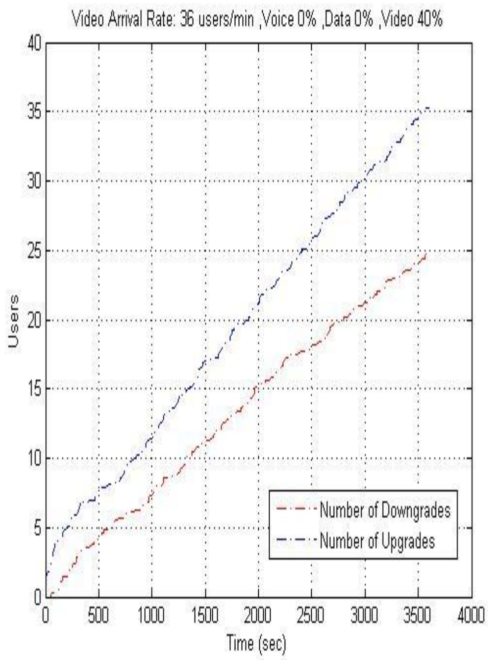


Figure 50

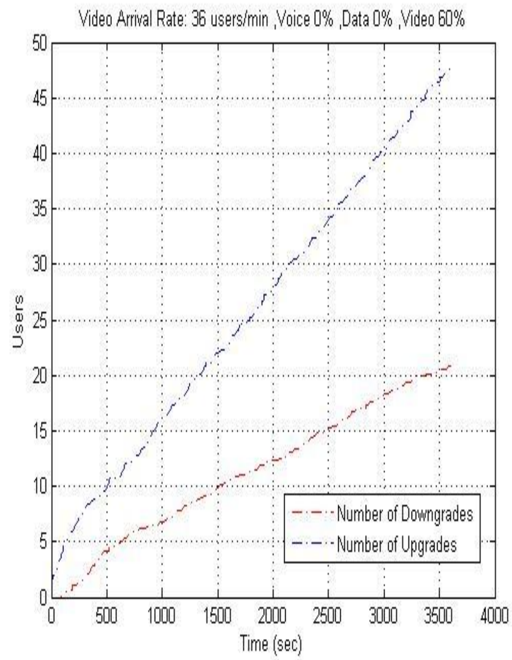


Figure 51

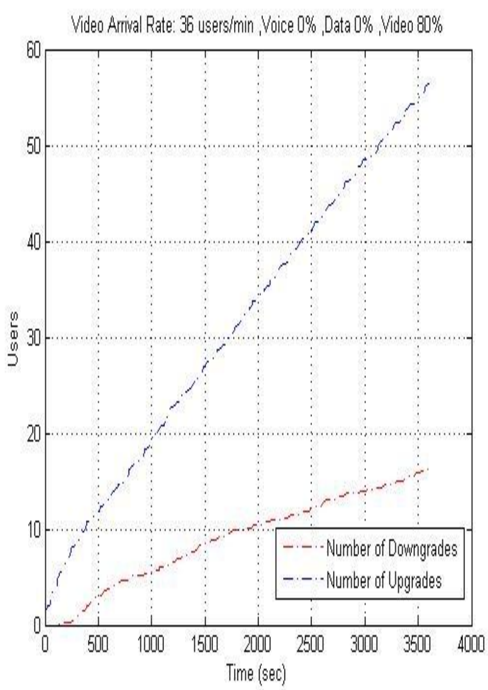


Figure 52

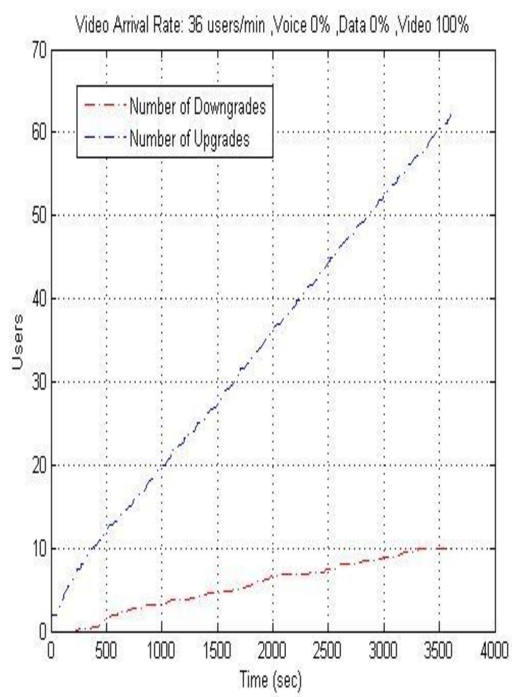


Figure 53

Figures 54 to 57 illustrate the number of upgrades and degradations of mpeg-4 movies for a video arrival rate equal to 10 users/minute/cell. We can see that when the available bandwidth is equal to 40%, the total number of degradations is larger than the total number of upgrades. The high video arrival rate forces the CAC mechanism to downgrade all the users, who can be downgraded based on their contract to allow more users to enter the network.

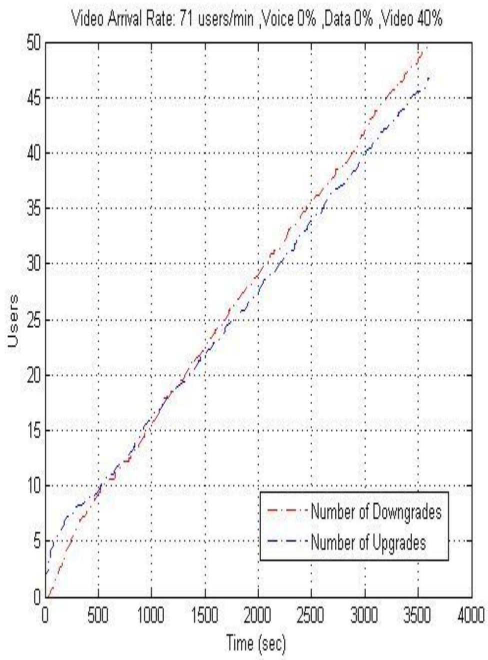


Figure 54

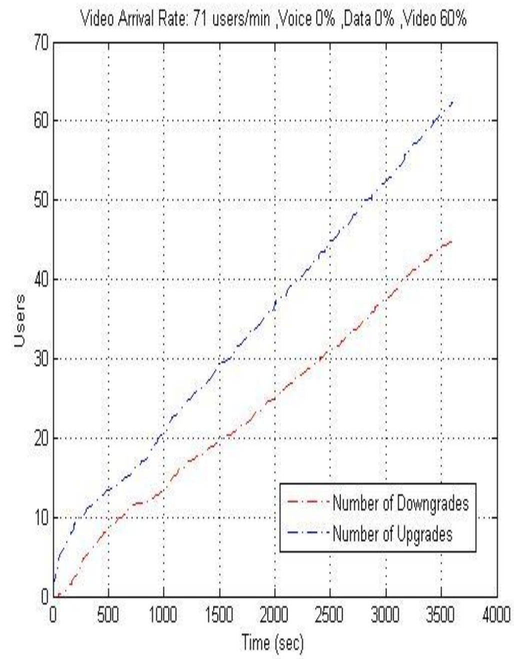


Figure 55

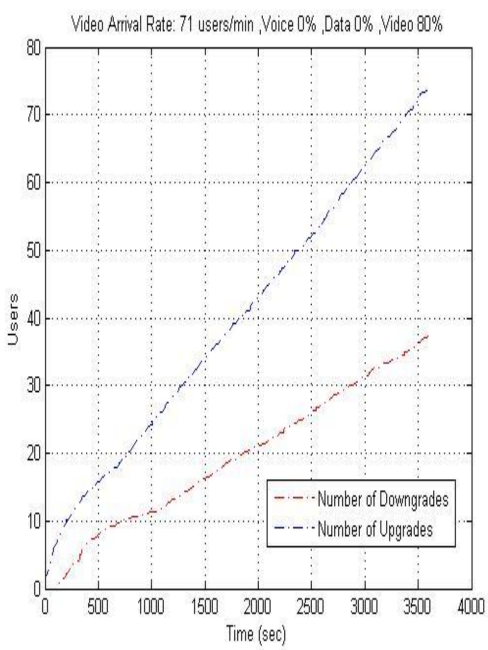


Figure 56

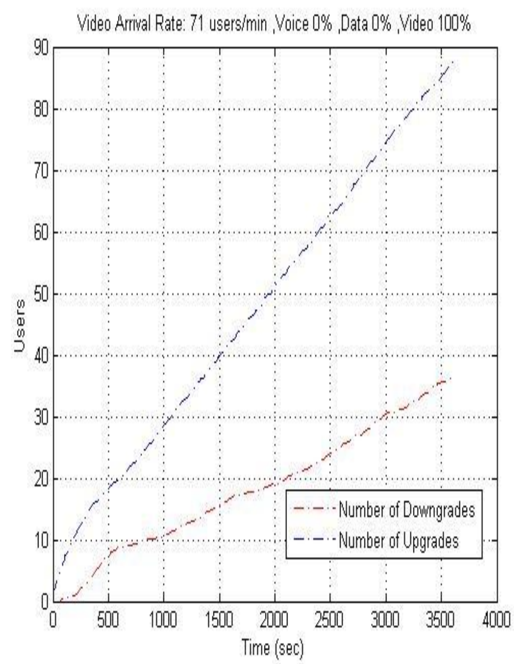


Figure 57

4.3.4 Results For Data Terminals

In this section we focus, as in Section 4.1.4, on the study of the influence of the FGC scheme on data terminals. In the simulations that follow, the number of voice terminals is constant and equal to 3738 (534 per cell) and they take up bandwidth equal to 40% of the channel bandwidth in each cell (8 Mbps). The average video arrival rate is equal to 10 users/minute/cell, and the average sojourn of a video terminal in the network is equal to 10 minutes. The maximum allowable bandwidth for video users is assumed equal to 40% of the channel bandwidth (8 Mbps). Therefore, voice and video terminals can cover up to 80% of the channel bandwidth, but, due to the conservatism of FGC scheme, the real bandwidth that is consumed by video terminals is significantly smaller. This is confirmed by the results presented in the fourth column of Table 29; voice and video users consume about 64% of the channel bandwidth, therefore data terminals, theoretically, could use the remaining 36%. However, regardless of the data load, data users never consume more than 22%. Still, the proposed mechanism is shown (last column of the Table) to be able to guarantee the QoS of data users, as only 2.7 messages on average, for a data load equal to 32%, exceed the transmission delay limit of 5 seconds. It needs to be emphasized, however, that these results concern only the data users who are admitted in the network.

Data Arrival Rate (Messages/sec)	CAC Bandwidth Estimation (Mbps)	Real Bandwidth for Data Users (Mbps)	Voice+Video Real Bandwidth (Mbps)	Avg Message delay (msec)	Number of messages with excessive delay
50 (8%)	1.73	1.61 (8.05%)	12.85 (64.25%)	160.3	0
70 (12%)	2.44	2.19 (10.95%)	13.01 (65.05%)	227.7	0
90 (15%)	3.12	2.72 (13.6%)	12.82 (64.1%)	262.9	0
110 (18%)	3.71	3.18(15.9%)	12.86 (62.3%)	299.3	0
130 (22%)	4.22	3.57 (17.85%)	12.77 (63.85%)	332.4	0
150 (25%)	4.61	3.89 (19.45%)	12.76 (63.8%)	352.7	0
170 (29%)	4.99	4.1 (20.5%)	12.75 (63.75%)	394.1	0.4286
190 (32%)	5.13	4.23 (21.15%)	12.79 (63.95%)	388.6	2.714

Table 29. Used bandwidth by data, voice and video terminals for different data loads.

Data Arrival Rate (Messages/sec)	CAC Bandwidth Estimation (Mbps)	Real Bandwidth for Data Users (Mbps)	Voice+Video Real Bandwidth (Mbps)	Avg Message delay (msec)	Number of messages with excessive delay
50 (8%)	1.77	1.58 (7.9%)	14.36 (71.8%)	229.1	3.857
70 (12%)	2.49	2.12 (10.6%)	14.46 (72.3%)	305.1	3.86
90 (15%)	3.15	2.59 (13%)	14.41 (72.1%)	359.1	7.86
110 (18%)	3.68	2.97 (14.9%)	14.38 (71.9%)	401.3	9.857
130 (22%)	4.04	3.23 (16.2%)	14.35 (71.8%)	429.6	12.71
150 (25%)	4.25	3.37 (16.9%)	14.4 (72%)	441.1	18.43
170 (29%)	4.32	3.41 (17.1%)	14.38 (71.9%)	451.1	14.57
190 (32%)	4.41	3.48 (17.4%)	14.29 (71.5%)	448.6	17.14

Table 30. Used bandwidth by data, voice and video terminals for different data loads for the first CAC mechanism using the second implementation.

The comparison of the results of Table 29 to those of Table 30 shows that, once again, the conservatism of the FGC scheme concerning video users, creates more available bandwidth for use by the data users. Hence, at the expense of bandwidth underutilization from the most bursty and needy type of traffic (video), data users experience much smaller transmission delays and negligible violation of their QoS requirements, whereas the first CAC mechanism leads to much worse QoS for data traffic.

4.4 Balancing the Tradeoff: Towards a More Efficient FGC Scheme

By studying the two CAC mechanisms presented in Sections 4.1.4.2, and 4.3, respectively, we arrived at the conclusion that the FGC scheme offers much higher QoS to handoff video users and to data users, at the cost of a significant degradation of the QoS of non-handoff video users. The reason is, as explained earlier, the fact that FGC builds upon the conservatism of the first CAC scheme, by allowing only a fraction of the already "scrutinized" video calls to enter the network; and the "scrutiny" comes from the rather unfairly conservative inequality in (2).

This led us to question whether a different equivalent bandwidth estimation can be used for FGC, in order to alleviate the conservatism of (2), given the already conservative nature of this type of CAC mechanism. The simplest idea is to disregard the first addend, which contains the standard deviation, and, more importantly, is heavily influenced by the acceptable packet loss rate. Hence, we used the sum of the mean rates of each video source in the system as an indication of the equivalent bandwidth. This is clearly not an optimal solution, as it disregards the burstiness of the traces. However, the results in Tables 31-35 show that even this simple implementation leads to impressively better results than both other CAC schemes that were studied earlier. More specifically, Tables 31 and 32 show that the use of just the mean rates of the video sources to compute their equivalent bandwidth leads to a much higher bandwidth utilization; this is true both for the case when only video users are present in the network, and the case when all traffic types are present. This increase in bandwidth utilization is combined in most cases of Tables 31-32 with a slight underestimation, by the new FGC CAC scheme, of the bandwidth that video users will need; the underestimation is expected, because of the removal of the first addend of (2). This underestimation, however, comes at no serious practical cost, since:

a) the FGC scheme "corrects" by its nature most underestimation errors, by probabilistically rejecting new video calls even if they could be accepted, based on the estimation.
b) Video packet dropping with the new FGC implementation is similar to that of the first CAC scheme, and leads to a violation of the QoS of video users only in two cases (Table 31, 60% maximum allowed video load and Table 33, 100% total load) in which the original FGC scheme that we proposed manages to satisfy the respective QoS requirement. The results in Table 33 reveal that the same is true for voice and data QoS: the new FGC scheme has comparable (only slightly worse) results with the first CAC scheme; it does not lead to voice QoS requirements violation in any case when the other schemes satisfy the respective requirements, and it only leads to a violation of data QoS requirements for a total load of 100%.

The results presented in Tables 34-35 also show that the new FGC CAC scheme improves the results for both handoff and non-handoff users, in comparison to the initial FGC. More specifically, the new FGC scheme achieves the highest percentage of handoff users being admitted into the network without degradation and the lowest percentage of handoff users being rejected from the network, among all three schemes. It also achieves better results in terms of non-handoff users degradation and rejection in comparison to the original FGC, but worse results than the first CAC scheme.

All of these results show that the main disadvantage behind the first CAC mechanism and the first FGC implementation is the overly conservative equivalent bandwidth estimation; this, as explained earlier, is a common disadvantage among all CAC mechanisms in the literature that have used an equivalent bandwidth estimation formula. Our proposed second implementation of the FGC scheme has the advantage of being very simple and excels in comparison to the other two schemes; however, a CAC scheme which underestimates (by 6-11%) the bandwidth needed by video sources is certainly not the most desirable choice. We comment on this further in our Conclusions section.

	CAC with 2nd implementation			FGC with 2nd implementation			FGC with 2nd implementation without considering standard deviation		
Maximum Allowed Video Load (Mbps)	**CAC Bandwidth Estimation (Mbps)**	**Real Bandwidth (Mbps)**	**Dropped Video Packets (%)**	**CAC Bandwidth Estimation (Mbps)**	**Real Bandwidth (Mbps)**	**Dropped Video Packets (%)**	**CAC Bandwidth Estimation (Mbps)**	**Real Bandwidth (Mbps)**	**Dropped Video Packets (%)**
8 (40%)	7.66	6.42 (32.1%)	0.004242	7.30	4.96 (24.8%)	0.00005	6.79	7.56 (37.8%)	0.005119
12 (60%)	11.41	9.6 (48%)	0.1344	10.32	7.37 (36.85%)	0.009696	9.22	10.18 (50.9%)	0.1342
16 (80%)	15.05	12.65 (63.3%)	0.9915	12.84	9.50 (47.5%)	0.1441	11.54	12.63 (63.2%)	1.156
20 (100%)	18.63	15.36 (76.8%)	3.661	15	11.33 (56.65%)	0.6967	13.26	14.13 (70.7%)	3.899

Table 31. Simulation scenarios with loads between 40%-100% of the total bandwidth only for video traffic loads.

Traffic Loads			CAC with 2nd implementation		FGC with 2nd implementation		FGC with 2nd implementation without considering standard deviation	
Number of Voice Users	**Messages per second**	**Maximum Video Capacity (Mbps)**	**CAC Bandwidth Estimation (Mbps)**	**Real Bandwidth (Mbps)**	**CAC Bandwidth Estimation (Mbps)**	**Real Bandwidth (Mbps)**	**CAC Bandwidth Estimation (Mbps)**	**Real Bandwidth (Mbps)**
214 (16%)	24 (4%)	4 (20%)	7.85	7.18 (35.9%)	8	6.46 (32.3%)	7.77	8.12 (40.6%)
320 (24%)	36 (6%)	6 (30%)	11.78	10.83 (54.15%)	11.75	9.8 (49%)	11.37	11.88 (59.4%)
427 (32%)	48 (8%)	8 (40%)	15.7	14.36 (71.8%)	15.32	12.99 (64.9%)	14.05	13.54 (67.7%)
534 (40%)	59 (10%)	10 (50%)	19.27	17.34 (86.7%)	18.2	15.8 (79%)	17.56	17.56 (87.8%)

Table 32. Used and estimated bandwidth for simulation scenarios with loads between 40%-100% of the total bandwidth.

Traffic Loads			CAC with 2nd implementation			FGC with 2nd implementation			FGC with 2nd implementation without considering standard deviation		
Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Dropped Voice Packets (%)	messages with excessive delay	Dropped Video Packets (%)	Dropped Voice Packets (%)	messages with excessive delay	Dropped Video Packets (%)	Dropped Voice Packets (%)	messages with excessive delay	Dropped Video Packets (%)
214 (16%)	24 (4%)	4 (20%)	0.00238	0	0	0	0	0	0.0002	0	0
320 (24%)	36 (6%)	6 (30%)	0.00185	0	0	0.00071	0	0.00002	0.0785	0	0.0012
427 (32%)	48 (8%)	8 (40%)	0.4079	0	0.00037	0.01501	0	0.00003	0.6523	2.14	0.0013
534 (40%)	59 (10%)	10 (50%)	3.677	21.57	0.06229	0.6902	0	0.00443	3.218	46.54	0.038

Table 33. Violations of the QoS of users for simulation scenarios with loads between 40%-100% of the total bandwidth.

Traffic Loads			CAC with 2nd implementation			FGC with 2nd implementation			FGC with 2nd implementation without considering standard deviation		
Number of Voice Users	Messages per second	Max Video Capacity (Mbps)	Handoff Video Users without Degradation	Handoff Video Users Degraded by CAC	Handoff Video Users rejected by CAC	Handoff Video Users without Degradation	Handoff Video Users Degraded by CAC	Handoff Video Users rejected by CAC	Handoff Video Users without Degradation	Handoff Video Users Degraded by CAC	Handoff Video Users rejected by CAC
214 (16%)	24 (4%)	4 (20%)	21.43%	3.06%	74.49%	53.25%	6.49%	40.26%	71.26%	12.64%	16.09%
320 (24%)	36 (6%)	6 (30%)	31.09%	5.88%	62.18%	75%	9.52%	15.48%	78.49%	10.75%	8.6%
427 (32%)	48 (8%)	8 (40%)	40.48%	7.94%	50.79%	79.35%	9.78%	9.78%	84.76%	5.71%	8.57%
534 (40%)	59 (10%)	10 (50%)	37.04%	5.19%	57.04%	87.5%	4.81%	7.69%	85.71%	3.36%	10.08%

Table 34. Metrics for handoff video terminals for different loads, using adaptive bandwidth reservation.

Traffic Loads			CAC with 2nd implementation			FGC with 2nd implementation			FGC with 2nd implementation without considering standard deviation		
Number of Voice Users	Messages per second	Max Video Capacity (Mbps)	Non Handoff Users without Degradation	Non Handoff Users Degraded by CAC	Non Handoff Users rejected by CAC	Non Handoff Users without Degradation	Non Handoff Users Degraded by CAC	Non Handoff Users rejected by CAC	Non Handoff Users without Degradation	Non Handoff Users Degraded by CAC	Non Handoff Users rejected by CAC
214 (16%)	24 (4%)	4 (20%)	16.88%	1.45%	81.49%	6.33%	0%	93.49%	8.76%	0%	91.24%
320 (24%)	36 (6%)	6 (30%)	22.7%	2.88%	74.41%	7.03%	0%	92.79%	11.93%	0%	87.88%
427 (32%)	48 (8%)	8 (40%)	30.37%	3.33%	66.3%	10.24%	0%	89.57%	17.03%	0%	82.97%
534 (40%)	59 (10%)	10 (50%)	40.18%	3.06%	56.58%	18.07%	0%	81.93%	26.16%	0%	73.84%

Table 35. Metrics for non handoff video terminals for different loads, using adaptive bandwidth reservation.

Conclusions and Future Work

The problem of efficient call admission control for multimedia (voice, video, data) traffic transmission over wireless cellular networks is of major importance, but is difficult to solve due to the contradictory QoS requirements of the different traffic types and the unpredictability of video traffic behavior.

A CAC mechanism aims at maximizing channel throughput without allowing network congestion, which would lead to the violation of users' QoS requirements. Our work focused on the admission control of video traffic and studied the case where no satisfactory video traffic models are available. We studied three mechanisms which control the admission of users moving from cell to cell in a wireless network. Our results have confirmed (as noted in previous literature) that the use of equivalent bandwidth estimation methods leads the CAC mechanism to very conservative estimates of the bandwidth that video users really need. This causes a large number of video calls to be rejected and a significant percentage of channel bandwidth to remain unused. On the other hand, users who succeed to enter the network enjoy very high QoS.

The first CAC mechanism that we studied was an extension of previous work, where users accept degradation for a discount. The use of sophisticated video upgrades and downgrades, combined with an adaptive bandwidth reservation scheme, is shown to improve channel throughput without any significant effect on the QoS of video users, despite the large number of handoffs taking place in the 7-cell structure used in our study, where users move along various roads.

To study possible improvements of the first CAC mechanism, we proposed and implemented a Fractional Guard Channel scheme, which was shown to provide better results for handoff video users as well as data users, but also to lead to worse QoS for non-handoff users. To alleviate this problem, we proposed a third CAC mechanism, using the same FGC scheme, but a simpler and less conservative equivalent bandwidth estimation. This third mechanism was shown to generally excel over the other two mechanisms, providing the highest channel throughput and the best results for handoff users; it also provides better results for non-handoff users than the second proposed mechanism, but worse than the third. Still, this third CAC mechanism has the opposite problem than the first two mechanisms: it steadily underestimates the bandwidth needed by video sources. This is not a negligible problem, as a CAC mechanism needs to make an accurate prediction of the total bandwidth that all types of traffic will need. For this reason, our future work will focus on the development of an equivalent bandwidth estimation method that will more accurately make this prediction. This accurate estimation will help the FGC mechanism that will be based on it to outperform all previous mechanisms and provide the highest QoS for both handoff and non-handoff users.

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