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Revenue and Irritation-based Call Admission Control using Road Maps over Wireless Cellular Networks

Master's Thesis

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Chania, October 2014

Abstract

Bandwidth remains the most valuable commodity in wireless networks, even more so today, as new bandwidth-hungry multimedia applications emerge constantly and their users have high Quality of Service (QoS) requirements. Hence, the ability of a wireless network to efficiently allocate its bandwidth resources is crucial.

In this thesis, we propose a new mechanism which controls the admission of users moving from cell to cell in a wireless cellular 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 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 MPEG-4. 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 one of the first in the relevant literature, to the best of our knowledge, to simultaneously consider both the provider revenue and user irritation in making its admission decisions.

Our results show the resource allocation efficiency of the proposed mechanism in two different implementations regarding the degradation/upgrades of video users. Our CAC mechanism is shown to provide voice and video users long term satisfaction with the QoS they receive, even for very high traffic loads, while at the same time allowing the provider to increase its profit.

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
LUSF	Long User Satisfaction Factor
SUIF	Short User Irritation Factor
EWMA	Exponentially Weighted Moving Average

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1. Introduction

Networks designers face a challenging problem when trying to control the traffic entered into a wireless cellular network, especially traffic coming from a broad range of mobile multimedia applications. The percentage of people that make use of wireless multimedia services is continuously increasing, hence increasing the need to satisfy the users' QoS and Quality of Experience (QoE) 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 [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 was 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.

What the above-described classifications have in common is that they focus on bandwidth reservation and bandwidth availability for requesting users. The vast majority of CAC mechanisms in the literature focus on the QoS that the proposed schemes are able to provide for voice, data and video users. Our proposed mechanism is one of the first in the relevant literature that takes simultaneously into account user satisfaction and provider revenue in order to make its admission decisions.

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 Pgood = 0.99995 and the steady-state probability for the channel to be in the “bad state” is Pbad = 0.00005.

The reason that we chose this value of Pbad 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 Pbad 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.

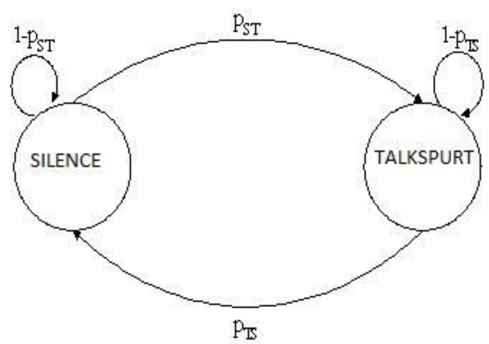

Figure 1. Markovian model for the activity of voice sources

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 1%, 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 Traffic 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.

In the rest of the thesis, we will alternate between using the terms “email” and “data”. Both terms represent the email data generated by the model taken from [30].

2.4 Video Traffic 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. TheMPEG-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 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 Bit Rate (Mbps)**			**Variance Bit Rate (Mbps)**			**Peak Bit Rate (Mbps)**			**Standard Deviation (Mbps)**		
	HQ	**MQ**	**LQ**	**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	0,456	0,2097	0,176
2	0,28	0,078	0,053	0,033	0,008	0,008	1,9	0,94	0,94	0,1816	0,0894	0,0894
3	0,7	0,25	0,15	0,196	0,044	0,034	3,4	1,6	1,6	0,4427	0,2097	0,1844
4	0,91	0,33	0,19	0,212	0,052	0,052	3,3	2	2	0,4604	0,228	0,228
5	1,3	0,42	0,23	0,288	0,104	0,104	8,8	4,4	4	0,5366	0,3225	0,3225
6	0,84	0,29	0,17	0,124	0,034	0,038	2,9	1,4	1,4	0,3521	0,1844	0,195

Table1. Trace Statistics

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 trace characteristics

2.5 Network and Mobility Models

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

We consider a 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.

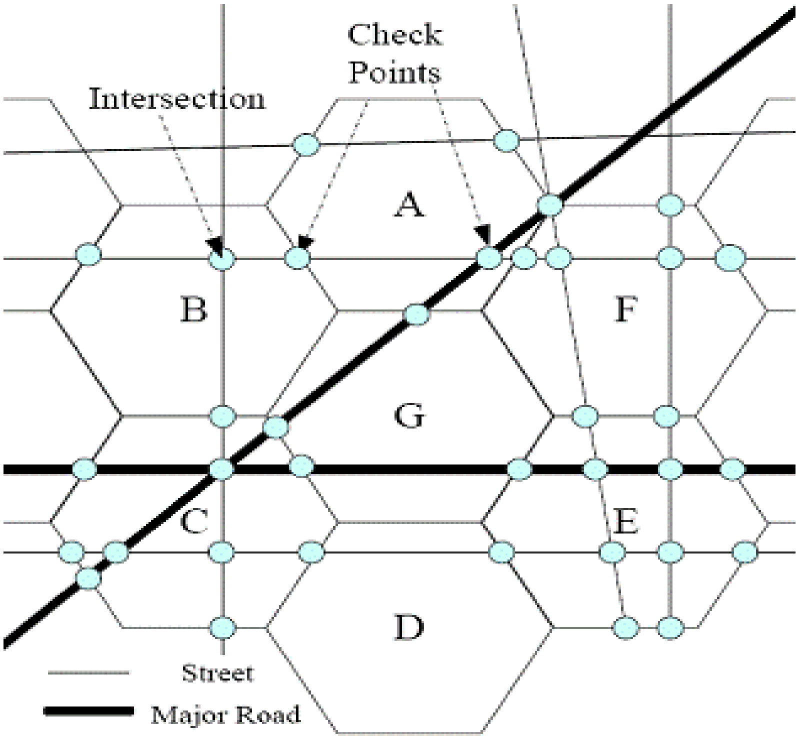

Figure 2. Road map and cellular network model.

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 arandom 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 User Irritation/Satisfaction Factor

An accurate modeling of the client irritation, i.e, what amount of performance degradation the client is ready to suffer without complaining or even considering to terminate their contract with the provider, will enable the service provider to keep and grow its clientele.

In the following, we adopt the proposed method from [44] to model the user irritation and we use the two proposed metrics from that study: the short term user irritation factor (SUIF) and the long term user satisfaction factor (LUSF). Each factor signifies different levels of user satisfaction as described below.

SUIF determines the delay that a user is ready to suffer prior to which the user decides to change or cancel the particular request. On the other hand, LUSF measures the degree of user satisfaction or dissatisfaction with the service over a longer time period, which could result, in the case of low LUSF in the user's decision to cancel service completely.

Sigmoid functions have been used in the literature [48, 49] to approximate the user's satisfaction with respect to service qualities or resource allocation. In [44], authors use the Sigmoid function for modeling the satisfaction/ dissatisfaction of users via SUIF and LUSF.

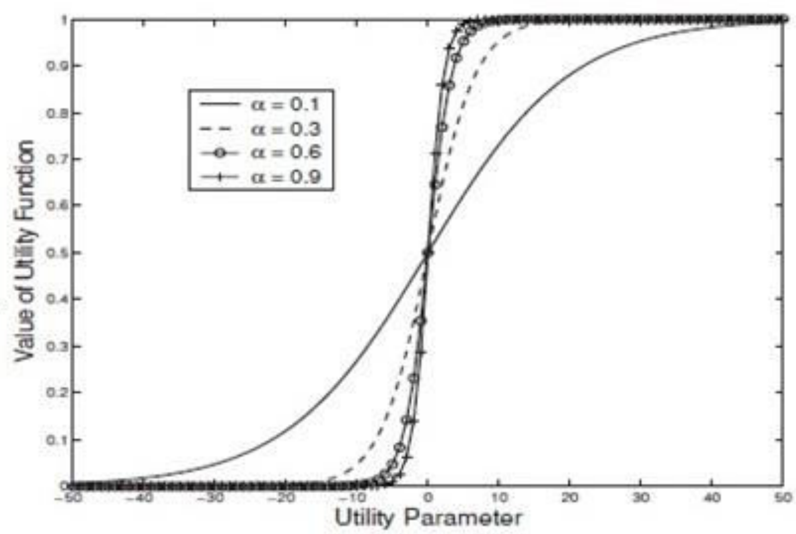

Figure 3. Examples of Utility Functions.

For a random variable x representing a service parameter like coverage or reliability, the corresponding satisfaction U(x) is given by

$$U(x) = \frac{1}{1+e^{-\alpha(x-\beta)}} \quad (2)$$

Hence αand β, determining the steepness and the center of the curve respectively, can be tuned to customize the function for different users. The plots for Equation (2), for different values of α are shown in Figure 3. As we can observe from the Figure, the satisfaction increases with increasing x. But for parameters like price or delay, the satisfaction should be decreased with the increasing x. In such cases we can model satisfaction as:

$$U(x) = 1 - \frac{1}{1+e^{-\alpha(x-\beta)}} \quad (3)$$

The value of α represents user's sensitivity to QoS degradation while β represents the "acceptable" region of operation. We use Equation (3) to compute the LUSF in this study. LUSF, which is computer per channel frame, is a quantitative measure of a user's tolerance to continuous degradation (or continuous excellence) of the service provided. Hence, LUSF keeps track of the long term QoS being provided to each user. The service providers would be able to control the churn rate by judicious manipulation of the LUSF.

2.6.1 Short User Irritation Factor

The requirements of the class of data users are less strict when compared to those of voice and video users. For data users, the delay that a user is willing to tolerate before cancelling a session can be considered as a suitable metric on which the computation of SUIF will be based.

We define the random variable x denoting the SUIF of the data user whose first attempt to enter the network was successful, as:

x= τ * (actual-ideal)/ideal (4),

where by "actual" we denote the message delay, and by "ideal" the message delay that the user would experience if his first connection request were to be accepted and his message were to be transmitted, at one packet per frame, without interruption or error due to noise.

The value of τ is taken to be equal to 0.3, from [44].

In the case of voice and video users, their satisfaction depends on the packet dropping probability: as the dropping probability increases, so does the user irritation. Hence, we define the random variable x denoting the SUIF of the user for voice and video users who are accepted in their first attempt to enter the network as:

$$x = \tau \times P_{drop} \quad (5)$$

where τ =0.3 and P_{drop} is the dropping probability.

Since P_{drop} can range between 0 and 1, Equations (3) and 5 show that the LUSF of users who are accepted in their first attempt to enter the network can range between 0.455 and 0.5, which is the value of the optimal LUSF (corresponding to zero short term irritation).

If, however, a user (voice, video or data) has already been rejected one or more times in his attempt to enter the network, the SUIF is computed as:

$$\text{SUIF} = [1+(2^{\text{Number\_of\_tries} - 1}) *0.1] * \text{SUIF}_{pre} \quad (6)$$

The "Number of tries" represents the number of user attempts to enter they system after the user is initially rejected, i.e., the number of retries after the first attempt. Equation (6) implies that a user's SUIF is increased geometrically after each failed attempt, and as a percentage of the existing SUIF, i.e., it increases by 10%, 20%, 40%, etc. over the existing SUIF.

Since LUSF is calculated on a per-user basis based on all the SUIFs perceived by that user, maintaining SUIF for each and every request for all users become memory and cost extensive. We argue that the QoS received in the distant past would have less significant impact (though not zero) on the user's overall long time irritation than the QoS received recently. We use an exponentially weighted moving average (EWMA) mechanism to maintain a continuous estimation of the SUIFs for each user. Let the stored SUIF be x_{n-1} and the SUIF to be computed be x_n; the input to the system, i.e., the current SUIF be x_i. Then x_n is calculated as follows:

$$x_n = \rho * x_{n-1} + (1-\rho) * x_i \quad (7)$$

where ρ is the weight given to the cumulative SUIF up to that point and x_n denotes the random variable which is used to estimate the SUIF at the n^{th} request. The value of ρ needs to be experimentally determined. However, in EWMA mechanisms, ρ=0.2 or 0.3 is generally chosen. In our study, we choose ρ=0.3.

2.7 Calculation of Revenue

2.7.1 Revenue from Video Users

Some recent studies (e.g. [50-52]) have suggested dynamic pricing strategies in mobile networks, as opposed to static pricing which has also been proposed in other works in the literature (e.g. [47], which however does not consider video traffic). Dynamic pricing gives negative incentives to users, based on current network conditions (e.g. congestion), in an attempt to shape the aggregate traffic in the network. However, in our study we prefer static revenue to dynamic revenue, since the last one does not take into account the fact that frequent changes in the price of a call can cause user dissatisfaction.

We use the idea of video "modes", as it is used in [35]. We denote as modes for the MPEG-4 video users the sets of traffic parameters presented in Table 1. Hence, we use eighteen modes for MPEG-4 video traffic, one high, one medium and one low quality mode for each one of the six traces. High-paying users adopt the high quality (HQ) modes, due to the increased bandwidth these modes offer. Low-paying users adopt the low quality modes, as these modes provide non-expensive prices.

With our static pricing approach, revenue weights are determined based on the traffic parameters of the traces i.e. based on the mean, standard deviation and peak. As revenue weight unit we choose the smallest sum $\mu+U*\sigma$, where μ is the mean of the video trace, σ is the standard deviation and U is the smallest constant equal to μ/σ among all modes (the values of U vary from 0.595 to 2.423 as the mean is for some modes larger and for some modes smaller than the standard deviation). The smallest value of U among the modes is related to the LQ mode of trace 2, leading to the smallest sum ($\mu+U*\sigma$) for the LQ mode of that trace. This is used as our revenue weight unit, and by computing the ratio of each one of the rest of the sums $\mu+U*\sigma$ versus the revenue weight unit, we get the "Revenue Weight" Column, containing the revenue weights for all modes, with the use of this approach. These values are presented in Table 3.

Movies	U=μ/σ			μ+Uσ			Revenue Weight		
	HQ	**MQ**	**LQ**	**HQ**	**MQ**	**LQ**	**HQ**	**MQ**	**LQ**
1	1,272	0,857	0,625	0,851	0,305	0,215	8,104	2,895	2,044
2	1,538	0,876	0,595	0,388	0,131	0,105	3,692	1,247	1
3	1,58	1,19	0,815	0,963	0,375	0,26	9,171	3,565	2,471
4	1,978	1,447	0,833	1,184	0,466	0,326	11,273	4,434	3,101
5	2,423	1,304	0,714	1,62	0,611	0,422	15,418	5,824	4,015
6	2,386	1,576	0,872	1,05	0,4	0,286	9,994	3,804	2,718

Table 3. Statistics for the high, low and medium quality versions of the video traces.

The current revenue R is computed as

$$R = \sum_i N_i * W_i \qquad (1)$$

where N_i is the total number of video users of mode *i*, and W_i is the revenue weight presented in Table 3.

2.7.2 Charging of Users

We propose in this work to allow the LUSF of voice and video users to play an important role in how users are charged, on a frame-by-frame basis, as well as for the whole duration of the simulation. The reason for this choice is that the overall user quality of experience (QoE) from a provider is always related not only to the user's QoS but also to the amount of money that the user is charged with for the specific QoS.

2.7.2.1 Data Users

The LUSF of data users is not included in the pricing process, due to the fact that the vast majority of users are not expected to be willing to pay more to improve their email or SMS download rates. So in each frame, the data users are charged 0.1E/message [45, 46].

2.7.2.2 Video Users

The pricing of video involves the video users' LUSF. We initially calculate the average users' LUSF for each cell. These average values are used in each frame, to define if a user is satisfied or dissatisfied, comparing the cell average with the LUSF of each individual user. If a user has lower LUSF than the average LUSF of the corresponding cell, the user is considered to be dissatisfied, otherwise the user is satisfied. Based on this user classification, the price that the user is charged with in the current channel frame is determined.

Video users are charged as follows:

```
1.      if user does not have special discount
2.          if LUSF of user < cell average LUSF (dissatisfied user)
3.                  The user will be charged as a user of the immediate lower quality without experiencing any
        quality degradation. This is a special discount which will end when the user has higher or equal LUSF to
        the cell average. The value of α (sensitivity) is decreased from its default value of 0.6 to 0.4.
4.          else
5.                  The user will be charged according to its current quality, as he is satisfied.
6.  end if
7.      else
8.          if LUSF of user >= cell average LUSF
9.                 End of special discount. The user will pay according to the charge of its quality as he is satisfied.
10.         else
11.             The special discount is still valid. The user will be charged as a user of the immediate lower
        quality without experiencing any quality degradation.
12.         end if
13.     end if
14. end if
```

The revenue from video users is computed from Equation (1).

2.7.2.3 Voice Users

The pricing of voice users involves the users' LUSF, similarly to the case of video users. The only difference between the way video and voice users are charged, as shown from the pseudocode below, has to do with the fact that voice users are charged based on the duration of their call. The values, in Euros, used in our work, have been taken from [45, 46], per minute, and are presented in the pseudocode as the respective charges per frame.

Voice users are charged as follows:

```
1.      if user does not have special discount
2.          if LUSF of user < cell average LUSF (dissatisfied user)
```

```
        The user will be charged 0.00006 Euros/frame. This is a special discount which will end when the user has higher or equal LUSF to the cell average.
    else
        The user will be charged with the regular pricing of 0.00007445 Euros/frame, as he is satisfied.
end if
   else
     if LUSF of user >= cell average LUSF
        End of special discount. The user will be charged with the regular pricing of 0.00007445 Euros/frame.
     else
        The special discount is still valid. The user will be charged 0.00006 Euros/frame.
     end if
   end if
end if
```

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 the 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:

$$t\sqrt{\sum_i \sigma_i^{\,2}} + \sum_i \mu_i < BW \qquad (7)$$

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 this is not possible, the algorithm attempts to degrade the requesting user, if the user accepts degradation. The rationale behind this decision is that the arrival of a new user should cause the minimum possible number of degradations, and hence irritation, to users who are already in the system, therefore is preferable that the new user is accepted with degradation. One point which needs to be stressed here is that in most of the relevant works in the literature it is commonly accepted that handoff calls have absolute priority in the obtaining an equal amount of channel bandwidth as the one they were occupying in the previous picocell location, i.e. handoff calls are not expected to endure any quality degradation, as this would lead to user dissatisfaction. We take a different approach in this work. It is indeed crucial for a handoff user not to experience call dropping when moving from one picocell to the next, as this would lead to significant user irritation (call dropping is much more irritating than the blocking of the call of a new user who attempts to transmit). However, if the mobile user experiences, during handoff, a degradation for which he has agreed in his contract, this should not be a cause for user irritation and therefore is allowed in our algorithm.

If after degradation the acceptance of the call is still not possible, there are two options based on the user's state (new or handoff). If a user is a handoff one, the CAC mechanism checks all possibilities of degrading users of the same or lesser priority and of higher LUSF than the handoff user, in order to accommodate the request. If such a possibility exists, the user is accepted.

More specifically, in the case of a HQ video handoff request for which there is not enough available bandwidth to be accepted, the CAC mechanism checks all HQ and MQ users who accept quality degradation based on their contract. If a user accepts degradation based on his contract AND the user's LUSF is higher than the LUSF of the handoff user, then we proceed with the degradation. That is, both the above conditions need to be satisfied for the degradation to take place.

A slightly different approach is necessary in the case of requests coming from a new call originating from within the picocell. If after degradation of the requesting user his request can still not be accepted, the request is rejected (leading to an increase in SUIF and decrease in LUSF). Then, the user will transmit his request again after a specific period of time. This time the CAC mechanism treats the call in the same manner as a handoff call; i.e., calls that have been rejected at least once are treated with equal priority with handoff calls, in order not to further increase the already once rejected users' irritation.

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, i.e., when the bandwidth availability exists.

As mentioned above, our CAC mechanism focuses mainly on the local cell. The reason is that, as a good as the mobility prediction might be, it is impossible to ever predict with perfect accuracy to which cell a user might move, and additionally, video calls can be degraded, therefore even if the necessary bandwidth to accommodate a handoff user may not be available at a given time in its entirety, the additional needed bandwidth can be "squeezed" out of existing calls.

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) 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 underutilization of the channel bandwidth. Therefore, the difficulty of such an implementation, together with its inefficiency, led us to use 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 Equation (1), to estimate the equivalent bandwidth of the new MQ and LQ versions of each trace.

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 CAC Implementation

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 4 and 5. In all of these traffic scenarios we assume that the number of voice users, the number of data users and the number of video users are constant during the simulation, i.e., if a user exits our hexagonal cell architecture shown in Figure 2, another user (belonging to the same "mode", in the case of video) immediately arrives.

Furthermore, a threshold was defined for the maximum bandwidth that each type of users can utilize. This means that our CAC scheme allows users to enter the channel until a specific amount of bandwidth has been reserved. If that bandwidth is exceeded, the requests of users are rejected. If the user cannot enter the network, he tries again after a specific time period until he manages to be accepted by the CAC, although in this case the user's irritation has increased, as explained in Sections 2 and 3.

The first line of Table 4 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 is conservative in its estimations as shown in [16], the real amount of bandwidth that users manage to reserve is 26.22%. We also observe, from Table 4, 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 the voice traffic QoS requirements are not violated even for 100% traffic load. Data 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 5, 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 percentages of bandwidth that is really reserved by the three traffic types are shown; it is significantly lower, for video users, than the estimated values. This shows once again that video traffic presents, due to its burstiness, the most significant resource allocation problem for the networks.

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%)	6.67	5.24 (26.22%)	0	0	0
267 (20%)	24 (4%)	3.2 (16%)	6.946	5.71 (28.55%)	0	0	0
320 (24%)	24 (4%)	2.4 (12%)	7.148	6.173 (30.87%)	0	0	0
267 (20%)	30 (5%)	5 (25%)	8.516	6.973 (34.87%)	0	0	0
334 (25%)	30 (5%)	4 (20%)	9.89	7.92 (39.64%)	0	0	0
400 (30%)	30 (5%)	3 (15%)	9.969	7.997 (39.99%)	0	0	0
320 (24%)	36 (6%)	6 (30%)	10.34	9.31 (42.18%)	3.499E-07	0	0
400 (30%)	36 (6%)	4.8 (24%)	10.1	8.917 (44.59%)	1.129E-05	0	0
480 (36%)	36 (6%)	3.6 (18%)	11.96	10.13 (50.65%)	0.0000566	0	0
374 (28%)	42 (7%)	7 (35%)	13.87	10.91 (54.6%)	0.00034	0	0
467 (35%)	42 (7%)	5.6 (28%)	10.49	11.38 (56.9%)	0.0008432	0	0

560 (42%)	42 (7%)	4.2 (21%)	9.477	11.81 (59.1%)	0.0008921	0	0
427 (32%)	48 (8%)	8 (40%)	11.47	11.47 (56.59%)	0.0009052	0	0
534 (40%)	48 (8%)	6.4 (32%)	12.18	11.75 (60.88%)	0.00112	0	0
640 (48%)	48 (8%)	4.8 (24%)	12.98	12.07 (64.88%)	0.002784	0	0
480 (36%)	54 (9%)	9 (45%)	12.95	11.61 (64.77%)	0.00333	0	0
600 (45%)	54 (9%)	7.2 (36%)	14.81	12.76 (69.09%)	0.004707	0	0
720 (54%)	54 (9%)	5.4 (27%)	14.72	12.76 (73.62%)	0.007707	0	0
534 (40%)	59 (10%)	10 (50%)	14.43	12.8 (79.34%)	0.006154	0	0
667 (50%)	59 (10%)	8 (40%)	16.11	15.26 (76.32%)	0.009248	0	0
800 (60%)	59 (10%)	6 (30%)	18.75	16.03 (80.17%)	0.01155	0	0

Table 4. 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%	3.54%	6.60%
267 (20%)	24 (4%)	3.2 (16%)	20.07%	3.43%	5.05%
320 (24%)	24 (4%)	2.4 (12%)	24.06%	3.68%	3.13%
267 (20%)	30 (5%)	5 (25%)	20.09%	5.08%	9.70%
334 (25%)	30 (5%)	4 (20%)	25.11%	4.92%	6.82%
400 (30%)	30 (5%)	3 (15%)	30.08%	5.31%	4.60%
320 (24%)	36 (6%)	6 (30%)	24.06%	6.13%	12%
400 (30%)	36 (6%)	4.8 (24%)	30.08%	5.73%	8.78%
480 (36%)	36 (6%)	3.6 (18%)	34.55%	5.94%	8.65%
374 (28%)	42 (7%)	7 (35%)	28.11%	6.85%	20.56%
467 (35%)	42 (7%)	5.6 (28%)	34.87%	6.72%	10.87%

560 (42%)	42 (7%)	4.2 (21%)	36.18%	4.21%	6.97%
427 (32%)	48 (8%)	8 (40%)	32.10%	7.52%	16.97%
534 (40%)	48 (8%)	6.4 (32%)	40.15%	7.82%	12.92%
640 (48%)	48 (8%)	4.8 (24)	48.11%	7.66%	9.11%
480 (36%)	54 (9%)	9 (45%)	36.09%	9.40%	19.28%
600 (45%)	54 (9%)	7.2 (36%)	45.09%	9.20%	14.81%
720 (54%)	54 (9%)	5.4 (27%)	54.12%	8.66%	10.85%
534 (40%)	59 (10%)	10 (50%)	40.17%	9.31%	22.67%
667 (50%)	59 (10%)	8 (40%)	50.14%	8.88%	17.31%
800 (60%)	59 (10%)	6 (30%)	60.14%	8.40%	11.63%

Table 5. Bandwidth that is really reserved by each type of traffic, with loads between 40%-100% of the total bandwidth.

In Table 6 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. The values in the next column of the Table show the provider's revenue. In the three last columns, the LUSF's averages for each type of terminal are presented. The results presented in Table 6 need to be taken under consideration together with the results of Tables 4-5. Together, the three Tables show that:

a) as the total allowed bandwidth increases, provider revenue also increases, up to the case of 90% traffic load. However, when the traffic load is allowed by the CAC to reach 100%, the revenue decreases in comparison to the 90% load case, as the number of rejected/irritated users increases.

b) among the three cases of the same total traffic load, the highest revenue for the provider is always achieved when the voice load increases at the expense of video load. This is again explained by the burstiness of the video traffic, which as shown in Table 5 manages to enter the network by a fraction of its theoretically requested bandwidth, whereas voice traffic covers all of its allocated bandwidth.

c) In order to explain the average LUSF results for video, we need to take into account Equation (3), as well as the fact that the CAC mechanism is quite conservative. Starting with Equation (3), it is clear that for the minimum SUIF value (i.e., zero), the utility function that represents LUSF gets its highest value, i.e, it becomes equal to 0.5. Therefore, the LUSF values shown in Table 6 for video reveal that all accepted video users in the network have minimal packet dropping, as their LUSF is close to optimal. The reason for this is that the CAC mechanism is conservative. By allowing significantly less video users into the system (the underutilization we discussed in Tables 4 and 5) the CAC mechanism ensures excellent QoE for the accepted video users.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Revenue	Average LUSF		
				VOICE	DATA	VIDEO
214 (16%)	24 (4%)	4 (20%)	4.46E+03	0.3821	0.139	0.4989
267 (20%)	24 (4%)	3.2 (16%)	4.54E+03	0.3906	0.1431	0.4988
320 (24%)	24 (4%)	2.4 (12%)	4.68E+03	0.415	0.1673	0.4989
267 (20%)	30 (5%)	5 (25%)	5.50E+03	0.3887	0.3188	0.4995
334 (25%)	30 (5%)	4 (20%)	6.52E+03	0.4177	0.3891	0.499
400 (30%)	30 (5%)	3 (15%)	6.79E+03	0.4285	0.4638	0.4988
320 (24%)	36 (6%)	6 (30%)	7.23E+03	0.4122	0.4511	0.4993
400 (30%)	36 (6%)	4.8 (24%)	7.79E+03	0.427	0.5	0.4992
480 (36%)	36 (6%)	3.6 (18%)	8.89E+03	0.4428	0.5	0.4992
374 (28%)	42 (7%)	7 (35%)	8.52E+03	0.4236	0.5	0.4995
467 (35%)	42 (7%)	5.6 (28%)	8.89E+03	0.4393	0.5	0.4994
560 (42%)	42 (7%)	4.2 (21%)	9.45E+03	0.4542	0.5	0.4991
427 (32%)	48 (8%)	8 (40%)	9.63E+03	0.4345	0.5	0.4997
534 (40%)	48 (8%)	6.4 (32%)	9.89E+03	0.4499	0.5	0.4996
640 (48%)	48 (8%)	4.8 (24%)	1.00E+04	0.4644	0.5	0.4994
480 (36%)	54 (9%)	9 (45%)	1.11E+04	0.443	0.5	0.4996
600 (45%)	54 (9%)	7.2 (36%)	1.16E+04	0.4624	0.5	0.4995
720 (54%)	54 (9%)	5.4 (27%)	1.21E+04	0.4743	0.5	0.4995
534 (40%)	59 (10%)	10 (50%)	1.07E+04	0.4531	0.5	0.4997
667 (50%)	59 (10%)	8 (40%)	1.13E+04	0.4692	0.5	0.4995
800 (60%)	59 (10%)	6 (30%)	1.19E+04	0.4898	0.5	0.4995

Table 6. Provider Revenue and LUSF for voice, data and video users

In Table 7, we present for voice and data traffic the results of average LUSF *for the accepted* users, for each individual cell, as well as the average LUSF over all cells.

Table 7 shows that the average LUSF for accepted voice users constantly increases, up to 100% traffic load. We do not present a similar Table for video traffic, as we have already discussed in the previous Table's results why the LUSF of accepted video users is very close to optimal (i.e., 0.5). No significant variance was observed in the individual cell values of video users' LUSF; it ranged from 0.497 to 0.5. The respective results for data traffic show that for total loads equal or larger than 60% the average LUSF is again very close to optimal.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Average LUSF for Accepted Voice Users							
			CELL A	CELL B	CELLC	CELLD	CELL E	CELLF	CELLG	TOTAL
214 (16%)	24 (4%)	4 (20%)	0.4026	0.3285	0.4498	0.329	0.4412	0.3962	0.3285	0.382186
267 (20%)	24 (4%)	3.2 (16%)	0.4124	0.333	0.4545	0.333	0.4537	0.4147	0.333	0.390614
320 (24%)	24 (4%)	2.4 (12%)	0.4222	0.3848	0.4656	0.374	0.4656	0.422	0.3742	0.415514
267 (20%)	30 (5%)	5 (25%)	0.4166	0.3333	0.45	0.333	0.4529	0.4026	0.333	0.388771
334 (25%)	30 (5%)	4 (20%)	0.4253	0.3742	0.4757	0.362	0.4792	0.4392	0.3684	0.4177
400 (30%)	30 (5%)	3 (15%)	0.4392	0.3848	0.4812	0.385	0.4812	0.4437	0.3848	0.428529
320 (24%)	36 (6%)	6 (30%)	0.4253	0.374	0.4656	0.375	0.4546	0.4166	0.3748	0.412239
400 (30%)	36 (6%)	4.8 (24%)	0.4412	0.3848	0.4792	0.385	0.4792	0.4222	0.3976	0.427
480 (36%)	36 (6%)	3.6 (18%)	0.4537	0.4083	0.4812	0.408	0.4856	0.4545	0.4083	0.442843
374 (28%)	42 (7%)	7 (35%)	0.4346	0.3848	0.4757	0.385	0.4757	0.4253	0.3848	0.423671
467 (35%)	42 (7%)	5.6 (28%)	0.4412	0.4147	0.476	0.412	0.4812	0.433	0.4166	0.4393
560 (42%)	42 (7%)	4.2 (21%)	0.4656	0.4222	0.4888	0.425	0.4901	0.4656	0.4222	0.454257
427 (32%)	48 (8%)	8 (40%)	0.4437	0.3962	0.4812	0.408	0.4812	0.4333	0.3976	0.4345
534 (40%)	48 (8%)	6.4 (32%)	0.4545	0.4267	0.4901	0.422	0.4901	0.4437	0.4222	0.449929
640 (48%)	48 (8%)	4.8 (24%)	0.4792	0.4431	0.494	0.435	0.4941	0.4712	0.4346	0.4644

480 (36%)	54 (9%)	9 (45%)	0.4529	0.4026	0.4923	0.417	0.4913	0.4431	0.4026	0.443057
600 (45%)	54 (9%)	7.2 (36%)	0.4757	0.4392	0.4951	0.439	0.4941	0.4545	0.4392	0.462429
720 (54%)	54 (9%)	5.4 (27%)	0.4812	0.4545	0.497	0.455	0.4978	0.4812	0.4545	0.474386
534 (40%)	59 (10%)	10 (50%)	0.4656	0.4253	0.4951	0.422	0.4951	0.4437	0.4253	0.453186
667 (50%)	59 (10%)	8 (40%)	0.4792	0.4437	0.4982	0.444	0.4982	0.4656	0.4561	0.469243
800 (60%)	59 (10%)	6 (30%)	0.4911	0.4812	0.4993	0.497	0.4999	0.4792	0.4812	0.489843

Table 7. LUSF per cell for accepted voice users

In Tables 8-9 we present for voice and video traffic the results of "Number of Tries" *for the accepted users* in each cell, as well as the average over all cells. The results in Table 8 show that voice users seldom need to retry; only in cell E, through which a major road passes, do we encounter two cases, one of a total traffic load of 70% and one when the total traffic load is 100%, where a voice user needs on average to retry 0.4 times to make his call. In the vast majority of the other cases (traffic loads and cells) the average number of retries is almost negligible; the same is true for the average number of retries of data users. On the contrary, video users are shown, in Table 9, to need 0.12-0.56 retries for each of their calls. Also, for each traffic load which is generated in three different ways, video users as expected need a larger average number of retries when the allocated voice traffic load is higher. It should be mentioned that, as shown in the Table, the average number of retries for video users decreases as the traffic load increases. The reason for this seemingly counter-intuitive result is that, as traffic load increases, there is a significant increase in the number of video users who are not able to enter the network at all. Hence, those video calls that are accepted need less tries to enter.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Average "Number of Tries" for accepted voice users							
			CELL A	CELLB	CELL C	CELLD	CELL E	CELLF	CELLG	TOTAL
214 (16%)	24 (4%)	4 (20%)	9.34E-06	0	0	0	0.01235	0	0	0.00176562
267 (20%)	24 (4%)	3.2 (16%)	2.55E-05	0	0	0	0.02354	0	0	0.0033665
320 (24%)	24 (4%)	2.4 (12%)	6.78E-05	0	0	0	0	0	0	9.6857E-06
267 (20%)	30 (5%)	5 (25%)	5.66E-05	0	0	0	0	0	0	8.0871E-06
334 (25%)	30 (5%)	4 (20%)	0.00005	1.5E-05	0	0	0	0.796	0	0.11372356

400 (30%)	30 (5%)	3 (15%)	6.66E-05	0	0	0	0	0	0	9.5143E-06
320 (24%)	36 (6%)	6 (30%)	0.000202	0	0	0	0	0	0	2.8857E-05
400 (30%)	36 (6%)	4.8 (24%)	8.74E-05	0	0	0	0	0	0	1.2486E-05
480 (36%)	36 (6%)	3.6 (18%)	0.0515	0.0515	0.0515	0.0515	0.12	0.0515	0	0.05392857
374 (28%)	42 (7%)	7 (35%)	0	0	0.07535	0	0.0754	0	0	0.02153571
467 (35%)	42 (7%)	5.6 (28%)	0.000238	0	0.01962	0	0.01026	0	0	0.00430257
560 (42%)	42 (7%)	4.2 (21%)	2.07E-05	0	0.1244	0	0.4016	0	0	0.07514581
427 (32%)	48 (8%)	8 (40%)	0.000128	0	0	0	0.0513	0	0	0.00734686
534 (40%)	48 (8%)	6.4 (32%)	4.48E-05	0	0.0327	0	0.1048	0	0	0.01964926
640 (48%)	48 (8%)	4.8 (24%)	1.89E-05	0	0	0.0513	0.1026	0	0	0.02198841
480 (36%)	54 (9%)	9 (45%)	0.000024	0	0	0	0	0	0	3.4286E-06
600 (45%)	54 (9%)	7.2 (36%)	2.24E-05	0	0	0	0	0	0	0.0000032
720 (54%)	54 (9%)	5.4 (27%)	0.000059	0	0	0	0	0	0	8.4286E-06
534 (40%)	59 (10%)	10 (50%)	0.000089	0	0	0	0	0	0	1.2714E-05
667 (50%)	59 (10%)	8 (40%)	0.000202	0	0.13193	0	0.4185	0	0	0.078661
800 (60%)	59 (10%)	6 (30%)	0	0	0	0	0	0	0	0

Table 8. “Number of Tries” for accepted voice users.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Average “Number of Tries” for accepted video users							
			CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
214 (16%)	24 (4%)	4 (20%)	0.3137	0.1316	0.6411	0.1853	0.9104	0.3882	0.2108	0.39729
267 (20%)	24 (4%)	3.2 (16%)	0.6469	0.0897	0.7732	0.2325	0.9771	0.2643	0.273	0.46524
320 (24%)	24 (4%)	2.4 (12%)	0.8467	0.03316	0.798	0.4088	1.087	0.683	0.0724	0.56129
267 (20%)	30 (5%)	5 (25%)	0.08085	0.08486	0.3628	0	1.008	0.3267	0.2642	0.30392
334 (25%)	30 (5%)	4 (20%)	0.5634	0.08183	0.8238	0.0628	0.8312	0.1873	0.0167	0.36671
400	30 (5%)	3 (15%)	0.8327	0	0.6391	0	0.5886	0.8933	0.7909	0.53494

(30%)										
320 (24%)	36 (6%)	6 (30%)	0.06167	0.00233	0.5463	0.2101	0.8335	0.074	0.07121	0.25701
400 (30%)	36 (6%)	4.8 (24%)	0.4244	0.40344	0.435	0.061	0.8005	0.214	0.08067	0.34556
480 (36%)	36 (6%)	3.6 (18%)	0.29167	0.03567	0.684	0.1093	0.8179	0.2202	0.08888	0.32109
374 (28%)	42 (7%)	7 (35%)	0.1523	0.02971	0.4076	0.0159	0.7494	0.0876	0.05025	0.21326
467 (35%)	42 (7%)	5.6 (28%)	0.1953	0.1073	0.5361	0.0727	0.7208	0.1028	0.03659	0.25309
560 (42%)	42 (7%)	4.2 (21%)	0.3322	0.08809	0.7628	0.0811	0.7689	0.1959	0.0338	0.32325
427 (32%)	48 (8%)	8 (40%)	0.04722	0.152	0.2906	0	0.5361	0.0519	0.02359	0.15735
534 (40%)	48 (8%)	6.4 (32%)	0.15433	0.03289	0.3468	0.0289	0.6324	0.0375	0.03015	0.18043
640 (48%)	48 (8%)	4.8 (24%)	0.27766	0.0602	0.4099	0.0527	0.698	0.1238	0.07586	0.24257
480 (36%)	54 (9%)	9 (45%)	0.2259	0.00876	0.2592	0.0008	0.5065	0.0242	0.00758	0.14756
600 (45%)	54 (9%)	7.2 (36%)	0.1094	0.05144	0.3045	0.0259	0.529	0.0173	0.00893	0.1495
720 (54%)	54 (9%)	5.4 (27%)	0.4466	0.03308	0.5445	0.0796	0.3698	0.1991	0.03828	0.24442
534 (40%)	59 (10%)	10 (50%)	0.04924	0.01277	0.1888	0.0005	0.5763	0.0107	0.01355	0.12168
667 (50%)	59 (10%)	8 (40%)	0.14459	0.01063	0.3887	0.037	0.5891	0.062	0.03352	0.18078
800 (60%)	59 (10%)	6 (30%)	0.1278	0.04902	0.3883	0.1366	0.6172	0.1353	0	0.20775

Table 9. "Number of Tries" for accepted video users

In Tables 10-12, we present for all three types of traffic the results of average LUSF *for the rejected* users, for each individual cell, as well as the average LUSF over all cells.

As expected, the average voice LUSF is much smaller (Table 10) than the LUSF shown in Table 7. The average LUSF values for video and data users are also much smaller than the close to optimal values associated with accepted video and data users.

In the case of rejected voice users, our results show that as the total allocated bandwidth increases, LUSF increases as well. The reason for this result is that many of the rejected voice users are handoff users which were initially accepted in another cell but during handoff, because of the large traffic load, they were rejected by the BS of the cell to which they were moving. As shown in our earlier results in Table 6, the increase in available bandwidth leads to an increase in LUSF. Hence, these users initially experienced a high LUSF, but due to their rejection their SUIF will now increase. This increase, however, has not yet been computed in their LUSF, which is added to that of the rest of the rejected users, hence increasing the mean LUSF value. The same is true for the data users as well, as shown in Table 11. The cases with LUSF=0 in both Tables correspond to the absence of rejected data users in that cell.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Average LUSF for Rejected Voice Users							
			CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
214 (16%)	24 (4%)	4 (20%)	0.0515	0	0.0343	0	0.01202	0.0601	0.0206	0.025503
267 (20%)	24 (4%)	3.2 (16%)	0.06865	0	0.0455	0	0.02746	0.0275	0.0206	0.027096
320 (24%)	24 (4%)	2.4 (12%)	0.06865	0	0.0455	0	0.02746	0.0507	0.0507	0.034717
267 (20%)	30 (5%)	5 (25%)	0.068667	0	0.0687	0	0.01599	0.016	0.016	0.026473
334 (25%)	30 (5%)	4 (20%)	0.06865	0	0.0343	0	0.06865	0.0507	0.0206	0.034704
400 (30%)	30 (5%)	3 (15%)	0.06865	0	0.0343	0	0.06867	0.0515	0.0687	0.041686
320 (24%)	36 (6%)	6 (30%)	0.068667	0	0.0343	0	0.06865	0.0507	0.0206	0.034706
400 (30%)	36 (6%)	4.8 (24%)	0.06865	0	0.0515	0	0.06865	0.0515	0.0206	0.037268
480 (36%)	36 (6%)	3.6 (18%)	0.0515	0.0515	0.0515	0.052	0.01202	0.0515	0.0515	0.04586
374 (28%)	42 (7%)	7 (35%)	0.0515	0.0515	0.0515	0.052	0.01202	0.0515	0.0515	0.04586
467 (35%)	42 (7%)	5.6 (28%)	0.0515	0.0515	0.0515	0.052	0.01202	0.0515	0.0515	0.04586
560 (42%)	42 (7%)	4.2 (21%)	0	0.0343	0.103	0	0.103	0.103	0	0.049047
427 (32%)	48 (8%)	8 (40%)	0.103	0	0.0343	0	0.0546	0.1373	0	0.047033
534 (40%)	48 (8%)	6.4 (32%)	0.03433	0	0.103	0	0.103	0.103	0	0.049047
640 (48%)	48 (8%)	4.8 (24%)	0.103	0	0.0343	0	0.103	0.103	0	0.049047
480 (36%)	54 (9%)	9 (45%)	0.068667	0.0687	0.0159	0.069	0.01599	0.0687	0.0687	0.053605
600 (45%)	54 (9%)	7.2 (36%)	0.068667	0.0515	0.0343	0.069	0.06867	0.0687	0.0343	0.056403
720 (54%)	54 (9%)	5.4 (27%)	0.05149	0.0687	0.0343	0.069	0.05149	0.0687	0.0687	0.058854
534 (40%)	59 (10%)	10 (50%)	0.03433	0.0449	0.1726	0.034	0.1726	0.0515	0.0515	0.080247
667 (50%)	59 (10%)	8 (40%)	0.068667	0.0515	0.1726	0.045	0.1726	0.0515	0.0687	0.090058
800 (60%)	59 (10%)	6 (30%)	0.068667	0.1726	0.1726	0.051	0.1726	0.0515	0.0515	0.105848

Table 10. LUSF per cell for rejected voice users

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Average LUSF for Rejected Data Users							
			CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
214 (16%)	24 (4%)	4 (20%)	0	0	0	0	0.0241	0	0	0.0034407
267 (20%)	24 (4%)	3.2 (16%)	0	0	0	0	0.1606	0	0	0.0229429
320 (24%)	24 (4%)	2.4 (12%)	0.0807	0	0.0804	0	0.1396	0	0	0.04295
267 (20%)	30 (5%)	5 (25%)	0.0807	0	0.1231	0	0.0804	0	0	0.0405929
334 (25%)	30 (5%)	4 (20%)	0.0807	0	0.0804	0	0.1396	0	0	0.04295
400 (30%)	30 (5%)	3 (15%)	0.0807	0	0.1608	0	0.0804	0	0	0.0459714
320 (24%)	36 (6%)	6 (30%)	0	0	0.0962	0	0.1112	0	0	0.0296229
400 (30%)	36 (6%)	4.8 (24%)	0	0	0.0962	0	0.1112	0	0	0.0296229
480 (36%)	36 (6%)	3.6 (18%)	0	0	0	0	0.0178	0	0	0.0025357
374 (28%)	42 (7%)	7 (35%)	0.1613	0.1612	0.1985	0.1612	0.2511	0.1613	0.0807	0.1678786
467 (35%)	42 (7%)	5.6 (28%)	0.1207	0	0.3659	0	0.3634	0	0	0.1214286
560 (42%)	42 (7%)	4.2 (21%)	0	0	0.2145	0	0.4556	0	0	0.0957286
427 (32%)	48 (8%)	8 (40%)	0	0	0.192	0	0.3719	0	0	0.0805571
534 (40%)	48 (8%)	6.4 (32%)	0.2418	0	0.3446	0	0.3792	0	0	0.1379429
640 (48%)	48 (8%)	4.8 (24%)	0.0605	0	0.4265	0	0.4825	0	0	0.1385
480 (36%)	54 (9%)	9 (45%)	0.3626	0	0.3926	0	0.4076	0.0985	0	0.1801857
600 (45%)	54 (9%)	7.2 (36%)	0.2412	0.1288	0.3813	0	0.404	0.0966	0	0.17884
720 (54%)	54 (9%)	5.4 (27%)	0.3629	0	0.4043	0	0.4044	0.0965	0	0.1811629
534 (40%)	59 (10%)	10 (50%)	0.4817	0.4522	0.4921	0.159	0.4828	0.4077	0.3219	0.3996286
667 (50%)	59 (10%)	8 (40%)	0.4822	0.3893	0.3911	0.3587	0.3587	0.3564	0.3095	0.3779593
800 (60%)	59 (10%)	6 (30%)	0.4817	0.4698	0.4934	0.4829	0.4939	0.2119	0	0.3762286

Table 11. LUSF per cell for rejected data users

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Average LUSF for Rejected Video Users							
			CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
214 (16%)	24 (4%)	4 (20%)	0.32025	0.2159	0.1186	0.2258	0.09514	0.1802	6.1E-06	0.165121
267 (20%)	24 (4%)	3.2 (16%)	0.2314	0.1997	0.1228	0.2595	0.09198	0.181	2.9E-06	0.155198
320 (24%)	24 (4%)	2.4 (12%)	0.1753	0.1848	0.1286	0.1792	0.104	0.1666	6.8E-05	0.134081
267 (20%)	30 (5%)	5 (25%)	0.1918	0.2803	0.1478	0.08065	0.08869	0.1925	5.7E-05	0.14025
334 (25%)	30 (5%)	4 (20%)	0.192	0.3262	0.1257	0.297	0.09767	0.2398	1.5E-05	0.182626
400 (30%)	30 (5%)	3 (15%)	0.1992	0.134	0.1117	0.1972	0.1183	0.1444	9.7E-05	0.129271
320 (24%)	36 (6%)	6 (30%)	0.4125	0.4144	0.1637	0.2697	0.1106	0.2712	0.0002	0.234615
400 (30%)	36 (6%)	4.8 (24%)	0.17376	0.1881	0.1311	0.23342	0.10074	0.1574	0.00011	0.140661
480 (36%)	36 (6%)	3.6 (18%)	0.199975	0.1794	0.1154	0.17301	0.09478	0.2857	2.5E-05	0.149763
374 (28%)	42 (7%)	7 (35%)	0.3714	0.2365	0.1333	0.242	0.10389	0.2967	0	0.197677
467 (35%)	42 (7%)	5.6 (28%)	0.202	0.1733	0.1408	0.08717	0.09869	0.1525	2.4E-05	0.122069
560 (42%)	42 (7%)	4.2 (21%)	0.1211	0.2635	0.1249	0.2913	0.09607	0.3257	2.1E-05	0.174656
427 (32%)	48 (8%)	8 (40%)	0.3562	0.2372	0.1499	0.1613	0.09613	0.4376	0.00013	0.205494
534 (40%)	48 (8%)	6.4 (32%)	0.1974	0.2357	0.1055	0.2696	0.102	0.2746	0.05307	0.176839
640 (48%)	48 (8%)	4.8 (24%)	0.2223	0.2826	0.1354	0.136	0.12046	0.2373	0.00014	0.162029
480 (36%)	54 (9%)	9 (45%)	0.2542	0.0524	0.169	0.1209	0.1174	0.2419	2.2E-05	0.136539
600 (45%)	54 (9%)	7.2 (36%)	0.49155	0.3498	0.1702	0.17084	0.08735	0.363	2.2E-05	0.233241
720 (54%)	54 (9%)	5.4 (27%)	0.4035	0.3443	0.1612	0.188	0.10727	0.1564	5.9E-05	0.194383
534 (40%)	59 (10%)	10 (50%)	0.3897	0.2753	0.1912	0.1613	0.1022	0.1613	8.9E-05	0.183013
667 (50%)	59 (10%)	8 (40%)	0.3478	0.4128	0.1486	0.1597	0.09988	0.1434	2E-05	0.187457
800 (60%)	59 (10%)	6 (30%)	0.1993	0.484	0.1154	0.1479	0.0955	0.2014	0	0.177643

Table 12. LUSF per cell for rejected video users

In Tables 13-15 we present for all three traffic types the results of "Number of Tries" *for the rejected users* in each cell, as well as the average over all cells. It is clear from comparing the results of these Tables with those of Tables 8-9 that, contrary to the case of accepted users, all types of rejected users need to make, on average, a significant number of retries after their initial attempts are rejected.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Average "Number of Tries" voice - rejected							
			CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
214 (16%)	24 (4%)	4 (20%)	0	0	0	0	0	0	0	0
267 (20%)	24 (4%)	3.2 (16%)	0	0	0	0	0	0	0	0
320 (24%)	24 (4%)	2.4 (12%)	0	0	0	0	0.01796	0	0	0.00256571
267 (20%)	30 (5%)	5 (25%)	0	0	0	0	0	0	0	0
334 (25%)	30 (5%)	4 (20%)	0	0	0	0	0	0	0	0
400 (30%)	30 (5%)	3 (15%)	0	0	0	0	1.081	0	0	0.15442857
320 (24%)	36 (6%)	6 (30%)	0	0	0	0	0	0	0	0
400 (30%)	36 (6%)	4.8 (24%)	0	0	0	0	0	0	0	0
480 (36%)	36 (6%)	3.6 (18%)	0	0	0	0	1.27725	0	0	0.18246429
374 (28%)	42 (7%)	7 (35%)	0	0	0.7531	0	0.7531	0	0	0.21517143
467 (35%)	42 (7%)	5.6 (28%)	0	0	0.7297	0	0.7531	0	0	0.21182857
560 (42%)	42 (7%)	4.2 (21%)	0	0	0.7297	0	1.089	0	0	0.25981429
427 (32%)	48 (8%)	8 (40%)	0	0	0	0	0.621	0	0	0.08871429
534 (40%)	48 (8%)	6.4 (32%)	0.01796	0	1.081	0	5.6096	0	0	1.1151
640 (48%)	48 (8%)	4.8 (24%)	0.0265	0	1.27725	0	6.5345	0	0	1.11975
480 (36%)	54 (9%)	9 (45%)	0.04556	0	3.24	0	7.484	0	0	1.53850857
600 (45%)	54 (9%)	7.2 (36%)	0.05123	0	4.61	0	8.84	0	0	1.92874714

720 (54%)	54 (9%)	5.4 (27%)	0.6172	0	6.56	0	8.056	0	0	2.17617143
534 (40%)	59 (10%)	10 (50%)	2.46	0	5.123	0	9.341	0	0	2.41771429
667 (50%)	59 (10%)	8 (40%)	3.1122	0	6.856	0	10.2	0	0	2.88117143
800 (60%)	59 (10%)	6 (30%)	8.056	0	2.311	0	12.31	0	0	3.23957143

Table 13. "Number of Tries" for rejected voice users

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Average "Number of Tries" data - rejected							
			CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
214 (16%)	24 (4%)	4 (20%)	0	0	0	0	0.5427	0	0	0.0775286
267 (20%)	24 (4%)	3.2 (16%)	0	0	0	0	1.944	0	0	0.2777143
320 (24%)	24 (4%)	2.4 (12%)	0.1667	0	1.877	0	1.418	0	0	0.4945214
267 (20%)	30 (5%)	5 (25%)	0.1667	0	0.2982	0	1.418	0	0	0.26898
334 (25%)	30 (5%)	4 (20%)	0.1666	0	0.2982	0	1.877	0	0	0.3345429
400 (30%)	30 (5%)	3 (15%)	0.1667	0	1.877	0	2	0	0	0.5776643
320 (24%)	36 (6%)	6 (30%)	0	0	3	0	4.425	0	0	1.0607143
400 (30%)	36 (6%)	4.8 (24%)	0	0	3	0	4.425	0	0	1.0607143
480 (36%)	36 (6%)	3.6 (18%)	0	0	0	0	14.062	0	0	2.0088571
374 (28%)	42 (7%)	7 (35%)	0.4344	0.3739	3.325	0.3858	7.303	0.3676	0.1666	1.7651857
467 (35%)	42 (7%)	5.6 (28%)	0.5	0	5.748	0	6.891	0	0	1.877
560 (42%)	42 (7%)	4.2 (21%)	0	0	4.296	0	9.309	0	0	1.9435714
427 (32%)	48 (8%)	8 (40%)	0	0	2	0	6.1965	0	0	1.1709286
534 (40%)	48 (8%)	6.4 (32%)	0	0	4.507	0	9.437	0	0	1.992
640 (48%)	48 (8%)	4.8 (24%)	0.125	0	5.337	0	7.012	0	0	1.782
480 (36%)	54 (9%)	9 (45%)	1.054	0	13.26	0	11.57	1.833	0	3.9595714
600 (45%)	54 (9%)	7.2 (36%)	0.7725	0.667	12.191	0	16.79	2	0	4.6315

720 (54%)	54 (9%)	5.4 (27%)	12.44	0	10.195	0	8.0177	0	0	4.3789857
534 (40%)	59 (10%)	10 (50%)	2.058	4.3243	7.981	1.015	6.8276	7.3546	1.36	4.4172143
667 (50%)	59 (10%)	8 (40%)	2.0058	2.03775	9.286	2.264	8.132	5.5675	1.75	4.4347143
800 (60%)	59 (10%)	6 (30%)	2.068	4.076	4.144	2	3.176	38.25	0	7.6734286

Table 14. “Number of Tries” for rejected data users

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Average “Number of Tries” video - rejected							
			CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
214 (16%)	24 (4%)	4 (20%)	2.333	2.902	4.4565	2.125	5.326	3.6095	3.3627	3.44496
267 (20%)	24 (4%)	3.2 (16%)	3.198	3.094	4.489	2.2285	5.4405	3.4125	3.91235	3.68212
320 (24%)	24 (4%)	2.4 (12%)	3.6875	3.2185	4.4355	1.6552	5.2715	3.924	4.025	3.74531
267 (20%)	30 (5%)	5 (25%)	3.4387	2.54025	4.7703	0.9942	5.3977	3.594	3.032	3.3953
334 (25%)	30 (5%)	4 (20%)	3.8915	2.527	4.63	1.622	5.3995	2.967	1.1065	3.16336
400 (30%)	30 (5%)	3 (15%)	9.7E-05	3.78	4.89	2.977	4.58	4.104	6.771	3.87173
320 (24%)	36 (6%)	6 (30%)	1.511	1	3.6047	2.477	5.0257	2.455	6.525	3.22834
400 (30%)	36 (6%)	4.8 (24%)	3.4072	3.0356	4.1778	0.9924	5.0766	4.44	4.55	3.66852
480 (36%)	36 (6%)	3.6 (18%)	3.0595	3.1375	4.7618	1.2145	5.5038	2.4715	2.4585	3.22957
374 (28%)	42 (7%)	7 (35%)	1.725	1.8248	4.3065	0.5	5.3885	2.2915	0.4054	2.34881
467 (35%)	42 (7%)	5.6 (28%)	2.212	2.464	4.26	1.683	5.123	2.2	2.095	2.86243
560 (42%)	42 (7%)	4.2 (21%)	2.788	2.572	4.762	1.244	5.328	2.29	1.147	2.87586
427 (32%)	48 (8%)	8 (40%)	1.158	2.412	4.0935	0.3333	5.514	1.32	3.7815	2.6589
534 (40%)	48 (8%)	6.4 (32%)	2.855	2.2447	4.216	0.7503	5.0658	1.2465	3.1125	2.78439
640 (48%)	48 (8%)	4.8 (24%)	3.048	2.5535	4.2115	1.2305	4.871	2.95	6.5935	3.63686
480	54 (9%)	9 (45%)	2.6807	0.2499	3.443	0.249	4.729	0.500	1.4647	1.9025

(36%)					2	9	5	3		9
600 (45%)	54 (9%)	7.2 (36%)	1.2422	1.47025	3.6895	0.9906	5.5148	0.75	2.6955	2.33612
720 (54%)	54 (9%)	5.4 (27%)	1.5527	1.4905	3.6653	3.3403	4.9762	3.6082	4.884	3.35959
534 (40%)	59 (10%)	10 (50%)	1.6476	1.2153	3.131	0.3333	5.082	0.333	11.8	3.36318
667 (50%)	59 (10%)	8 (40%)	1.939	1.4905	3.8162	1.2798	5.0753	0.25	19.53	4.76868
800 (60%)	59 (10%)	6 (30%)	2.962	1	4.5	4.858	5.28	3.023	0	3.089

Table 15. “Number of Tries” for rejected video users

For traffic loads between 110%-140%, we present some indicative results of our study in Appendix A. The most important conclusions that can be derived for these cases, are the following:

a) As the traffic load increases, the average voice LUSF decreases significantly, for both accepted and rejected voice users.

b) The decrease is even larger for the average data LUSF, which also shows a very large variance among cells. The reason is that the vast majority of data requests are rejected, and those that are accepted are only accepted after a significant number of retries (8, on average) and when some bandwidth becomes momentarily free, which can happen at any time, hence delays among data users differ greatly. Our results for rejected data users show that they would need, on average, 24-346 retries to enter the network, which practically means that these requests are never accepted, since the users will have long stopped trying to send their messages.

c) The average LUSF for rejected video users decreases significantly, but the average LUSF for accepted video users does not change, compared to the case of lower loads. It remains very high, close to the optimal value of 0.5, due again to the conservatism of the CAC mechanism which leads to an underutilization of the bandwidth but also to the existence of enough "room" for video users to "breathe" inside the network.

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, the number of video users which are created, equals to the result of the division of the bandwidth to the average bandwidth of all type of movies. The users are inserted into the network until the CAC mechanism starts rejecting them, when the maximum allowable video load is exceeded.

In Table 17, where we present our relevant results, 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 equal to 100% of the channel bandwidth the QoS requirements of video users are violated. We also observe, from Table 34, that as the traffic load and the number of users who achieve to enter the network increases, the revenue gained by the provider increases as well.

Maximum Allowed Video Load (Mbps)	**CAC Bandwidth Estimation (Mbps)**	**Real Bandwidth (Mbps)**	**Average Number of Users**	**Dropped Video Packets (%)**	**REVENUE**
8 (40%)	5.376	3.067 (15.34%)	11	0	2.48E+03
10 (50%)	6.87	4.221 (21.1%)	14	0	3.21E+03
12 (60%)	8.132	5.233 (26.16%)	17	0	3.85E+03
14 (70%)	9.406	6.269 (31.35%)	20	2.442E-05	4.52E+03
16 (80%)	9.931	6.769 (33.84%)	23	0.0002784	5.24E+03
18 (90%)	12.12	8.546 (42.73%)	27	0.002003	5.75E+03
20 (100%)	12.49	8.95 (44.79%)	29	0.05	6.43E+03

Table 17. Average number of active video users in the system for various loads

4.1.3 Downgrades/Upgrades of Video Terminals

Table 18 shows results for different video loads. The column "Active Users" of Table 35 presents the total number of users that have been admitted in the network. The values of the third and fourth column of Table 35 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 next 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 decreases, but still remains high, showing the need for downgrades in order to be able to utilize the network bandwidth at least acceptably, in the presence of video traffic.

Maximum Allowed Video Load (Mbps)	Active Users	Downgrade percentage HQ	Downgrade percentage MQ	Degradation Ratio HQ	Degradation Ratio MQ
8 (40%)	97	41.86%	38.29%	46.00%	43%
10 (50%)	121	30.43%	23.29%	43.43%	39.86%
12 (60%)	150	32.14%	31.86%	34.57%	37.00%
14 (70%)	172	28.14%	27.71%	30.86%	30.00%
16 (80%)	202	24.86%	27.00%	30.71%	30.71%
18 (90%)	213	20%	23%	25%	23%
20 (100%)	255	20%	23%	28%	28%
275	275	22%	23%	27%	26%

Table 18. Percentage of degradations and degradation ratio

4.2. Second CAC Implementation

In this implementation, as explained in Section 3, downgrades take place by a selective transmission of the video frames of a GoP. Given that:

a) in [41] it was shown that the second implementation clearly outperformed the first in terms of bandwidth utilization (smaller overestimation by the CAC, much larger percentage of the allocated bandwidth actually used), which is also confirmed by our results,

and

b) our qualitative results are almost identical as those in the first implementation, we present our results for this CAC implementation in Appendix B.

The only significant difference between the results of the first and the second CAC implementation has to do with the average LUSF of accepted voice and data users, which is smaller in the second implementation and exhibits higher variance in the case of voice users. The reason is that, by selectively dropping video frames of various movies, the bandwidth utilized by video users exhibits significant fluctuations. P frames and especially B frames have high variance, therefore depending on which video frames are discarded the QoS of voice and data users may improve or deteriorate significantly. This is the cost for improving the overall bandwidth utilization and it is shown in the results in Tables B1-B5 of the Appendix. The LUSF of video users remains excellent in this CAC implementation as well, as the CAC mechanism remains conservative, even if less so in this case. Also, the provider revenue steadily increases, as the traffic load increases, without any significant deterioration in the LUSF of voice users.

5. Conclusions and Future Work

The problem of efficient resource allocation 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. In this thesis, we proposed and studied a novel CAC mechanism which makes admission decisions not only on bandwidth availability but also on short and long term user satisfaction and on provider revenue. We studied two implementations of our mechanism, which controls the admission of users moving from cell to cell in a wireless network. Our results have confirmed 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. The second implementation of our mechanism partly alleviates this problem, but most importantly both implementations allow users to experience high long term satisfaction even for high traffic loads, while the provider revenue simultaneously increases.

In our mechanism, a percentage of the users accept degradation for a discount; the use of sophisticated video upgrades and downgrades 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. Still, this is not the only discount offered to users. We use Sigmoid utility functions and EWMA to estimate the long and short term user satisfaction and in the case of dissatisfied users the CAC either upgrades them for no additional charge, or users are charged less for the duration of their call, until their satisfaction improves, based on the algorithm's estimates.

Based on these results, we intend to focus our future work on the development of an equivalent bandwidth estimation method that will more accurately estimate the required bandwidth. This accurate estimation will help the CAC mechanism not only to accept a larger number of video users, but also to provide higher long-term user satisfaction to voice and data users, which "suffer" in the case of high loads because of the burstiness of video traffic.

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Appendix A

First CAC Implementation - Results for Loads 110%-140%

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)								
			CELL A	CELL B	CELLC	CELLD	CELLE	CELLF	CELLG	TOTAL
587 (44%)	65 (11%)	11 (55%)	0.4273	4965	0.4931	0.4286	0.4898	0.4958	0.4285	**0.4656**
734 (55%)	65 (11%)	8.8 (44%)	0.3717	0.4339	0.4924	0.3726	0.4902	0.4956	0.3726	**0.4327**
880 (66%)	65 (11%)	6.6 (33%)	0.4461	0.4254	0.4929	0.3548	0.4899	0.4215	0.3547	**0.4269**
640 (48%)	71 (12%)	12 (60%)	0.4957	0.4241	0.4905	0.4257	0.486	0.4234	0.4256	**0.453**
800 (60%)	71 (12%)	9.6 (48%)	0.4329	0.4331	0.491	0.3713	0.4885	0.4321	0.372	**0.4316**
960 (72%)	71 (12%)	7.2 (36%)	0.3859	0.4402	0.4896	0.3862	0.4876	0.4402	0.386	**0.4308**
694 (52%)	77 (13%)	13 (65%)	0.4925	0.424	0.4252	0.4252	0.4877	0.4234	0.4245	**0.4426**
867 (65%)	77 (13%)	10.4 (52%)	0.4316	0.4322	0.4401	0.4141	0.4855	0.4431	0.4335	**0.44**
1040(78%)	77 (13%)	7.8 (39%)	0.432	0.4931	0.4888	0.247	0.4844	0.4923	0.2464	**0.412**
747 (56%)	83 (14%)	14 (70%)	0.37	0.4309	0.4841	0.3091	0.4807	0.4311	0.31	**0.4023**
934 (70%)	83 (14%)	11.2 (56%)	0.3686	0.3692	0.4867	0.3398	0.4832	0.3701	0.3394	**0.3939**
1120(84%)	83 (14%)	8.4 (42%)	0.4909	0.3693	0.4857	0.1847	0.4834	0.3692	0.1843	**0.3668**

Table A1. LUSF per cell for accepted voice users

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)								
			CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
587 (44%)	65 (11%)	11 (55%)	0.5	0.361	0.5	0.5	0.5	0.5	0.5	0.48
734 (55%)	65 (11%)	8.8 (44%)	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
880 (66%)	65 (11%)	6.6 (33%)	0.4592	0.5	0.5	0.457	0.5	0.5	0.4561	0.4817
640 (48%)	71 (12%)	12 (60%)	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
800 (60%)	71 (12%)	9.6 (48%)	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
960 (72%)	71 (12%)	7.2 (36%)	0.3869	0.4045	0.5	0.4011	0.5	0.3739	0.409	0.425
694 (52%)	77 (13%)	13 (65%)	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
867 (65%)	77 (13%)	10.4 (52%)	0.3387	0.2244	0.5	0.1407	0.5	0.2412	0.237	0.3117
1040(78%)	77 (13%)	7.8 (39%)	0.09767	0.02276	0.4796	0.0963	0.4629	0.1465	0	0.1865
747 (56%)	83 (14%)	14 (70%)	0.2459	0.1784	0.5	0.0933	0.4505	0.1918	0.1863	0.2638
934 (70%)	83 (14%)	11.2 (56%)	0.08945	0	0.5	0	0.4468	0.26	0.09263	0.1984
1120(84%)	83 (14%)	8.4 (42%)	0	0.0938	0.39895	0	0.2012	0	0	0.0991

Table A2. LUSF per cell for accepted data users

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)								
			CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
587 (44%)	65 (11%)	11 (55%)	0.4999	0.5	0.4997	0.5	0.4987	0.5	0.5	0.4997
734 (55%)	65 (11%)	8.8 (44%)	0.4999	0.5	0.4994	0.4999	0.4987	0.4999	0.5	0.4996
880 (66%)	65 (11%)	6.6 (33%)	0.4998	0.5	0.4988	0.4999	0.4983	0.4997	0.4999	0.4994857
640 (48%)	71 (12%)	12 (60%)	0.5	0.5	0.4995	0.5	0.4988	0.5	0.5	0.4997
800 (60%)	71 (12%)	9.6 (48%)	0.4999	0.5	0.4994	0.4999	0.4986	0.4998	0.4999	0.4996
960 (72%)	71 (12%)	7.2 (36%)	0.4999	0.4999	0.4987	0.5	0.4986	0.4998	0.4999	0.4995
694 (52%)	77 (13%)	13 (65%)	0.5	0.5	0.4996	0.5	0.499	0.5	0.5	0.4998
867 (65%)	77 (13%)	10.4 (52%)	0.5	0.5	0.4994	0.5	0.4986	0.4999	0.5	0.4997
1040(78%)	77 (13%)	7.8 (39%)	0.4995	0.5	0.4991	0.5	0.4986	0.5	0.4999	0.4996
747 (56%)	83 (14%)	14 (70%)	0.5	0.5	0.4995	0.5	0.498	0.5	0.5	0.499
934 (70%)	83 (14%)	11.2 (56%)	0.5	5	0.4994	0.5	0.4989	0.5	0.5	0.4997
1120(84%)	83 (14%)	8.4 (42%)	0.4999	0.5	0.4988	0.5	0.4987	0.5	0.5	0.4996

Table A3. LUSF per cell for accepted video users

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)								
			CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
587 (44%)	65 (11%)	11 (55%)	0	0.02861	0.0286	0	0.0106	0.0286	0	**0.01377**
734 (55%)	65 (11%)	8.8 (44%)	0.05147	0.05149	0.0583	0.02574	0.0412	0.0515	0.02574	**0.04363**
880 (66%)	65 (11%)	6.6 (33%)	0.02942	0.02942	0.0446	0.02942	0.0444	0.0294	0.02942	**0.03373**
640 (48%)	71 (12%)	12 (60%)	0.0294	0.0294	0.0138	0	0.0156	0.0294	0	**0.0168**
800 (60%)	71 (12%)	9.6 (48%)	0.00523	0.01287	0.0382	0.01288	0.0386	0.0129	0.01287	**0.01907**
960 (72%)	71 (12%)	7.2 (36%)	0.09155	0.09155	0.0417	0.06866	0.0241	0.0687	0.06866	**0.06498**
694 (52%)	77 (13%)	13 (65%)	0.03433	0.03433	0.0343	0.03433	0.0343	0.0343	0.03433	**0.03433**
867 (65%)	77 (13%)	10.4 (52%)	0.02942	0.02942	0.0386	0.02942	0.1315	0.0589	0.02942	**0.04953**
1040(78%)	77 (13%)	7.8 (39%)	0.0588	0.0588	0.1393	0.0588	0.0978	0.0588	0.0588	**0.0759**
747 (56%)	83 (14%)	14 (70%)	0.0588	0.0882	0.0756	0.0588	0.0553	0.0883	0.05885	**0.05914**
934 (70%)	83 (14%)	11.2 (56%)	0.05885	0.05885	0.045	0	0.0432	0.0589	0	**0.03782**
1120(84%)	83 (14%)	8.4 (42%)	0.07243	0.103	0.0532	0.103	0.0545	0.103	0.103	**0.08459**

Table A4. LUSF per cell for rejected voice users

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
587 (44%)	65 (11%)	11 (55%)	0.1746	0.2397	0.4842	0.2234	0.4244	0.178	0.2266	0.2787
734 (55%)	65 (11%)	8.8 (44%)	0.3045	0.234	0.3479	0.3742	0.3461	0.2929	0.186	0.2979
880 (66%)	65 (11%)	6.6 (33%)	0.2713	0.3721	0.4047	0.3116	0.3364	0.2137	0.3181	0.3183
640 (48%)	71 (12%)	12 (60%)	0.224	0.2165	0.3187	0.3018	0.3959	0.1977	0.2427	0.271
800 (60%)	71 (12%)	9.6 (48%)	0.2346	0.1974	0.2612	0.2548	0.2915	0.2046	0.2206	0.2378
960 (72%)	71 (12%)	7.2 (36%)	0.2467	0.1585	0.1245	0.234	0.1583	0.194	0.2105	0.1895
694 (52%)	77 (13%)	13 (65%)	0.2454	0.2211	0.2091	0.1878	0.1097	0.1002	0.1688	0.1774
867 (65%)	77 (13%)	10.4 (52%)	0.18303	0.14227	0.13804	0.0761	0.1237	0.0955	0.13372	0.1275
1040(78%)	77 (13%)	7.8 (39%)	0.1781	0.05623	0.3	0.0452	0.138	0.0494	0.1292	0.128
747 (56%)	83 (14%)	14 (70%)	0.1873	0.0461	0.2915	0.1326	0.2175	0.1156	0.18071	0.1673
934 (70%)	83 (14%)	11.2 (56%)	0.1825	0.06472	0.1916	0.0657	0.1322	0.0948	0.1311	0.1232
1120(84%)	83 (14%)	8.4 (42%)	0.2281	0.1238	0.2976	0.1184	0.2343	0.0491	0.1596	0.173

Table A5. LUSF per cell for rejected data users

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
587 (44%)	65 (11%)	11 (55%)	0.2957	0.1981	0.25	0	0.1018	0.2992	0.06914	0.17342
734 (55%)	65 (11%)	8.8 (44%)	0.3666	0.1421	0.1572	0.1955	0.0955	0.2962	1.9E-05	0.179
880 (66%)	65 (11%)	6.6 (33%)	0.484	0.09798	0.1576	0.1382	0.09764	0.2138	0	0.1699
640 (48%)	71 (12%)	12 (60%)	0.2074	0.09371	0.151	0.06914	0.1057	0.3457	3.8E-05	0.1389
800 (60%)	71 (12%)	9.6 (48%)	0.1901	0.02153	0.19306	0.31048	0.112	0.3328	1.5E-05	0.165735
960 (72%)	71 (12%)	7.2 (36%)	0.3494	0.2335	0.1348	0.237	0.1095	0.2192	5.7E-05	0.183351
694 (52%)	77 (13%)	13 (65%)	0.4444	0.06814	0.191143	0	0.10515	0.1383	7.7E-05	0.135469
867 (65%)	77 (13%)	10.4 (52%)	0.484	0.0605	0.15955	0.121	0.10139	0.0854	0.00495	0.1452
1040(78%)	77 (13%)	7.8 (39%)	0.3459	0.1419	0.1634	0.142	0.0779	0.2803	0.0606	0.1731
747 (56%)	83 (14%)	14 (70%)	0.1815	0.20065	0.2059	0.1815	0.786	0.121	0	0.2395
934 (70%)	83 (14%)	11.2 (56%)	0.4235	0.1164	0.1945	0	0.10008	0.0852	6.7E-05	0.1314
1120(84%)	83 (14%)	8.4 (42%)	0.4152	0.1741	0.1356	0.242	0.0917	0.0977	9.2E-05	0.1652

Table A6. LUSF per cell for rejected video users

Appendix B

Second CAC Implementation - Most Important Results

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	CAC Bandwidth Estimation (Mbps)	Real Bandwidth (Mbps)	Dropped Voice Packets Packets (%)	Average Number of messages with excessive delay	Dropped Video Packets Packets (%)
214 (16%)	24 (4%)	4 (20%)	6,144	5.45(27.25%)	0,000000593	0	0
267 (20%)	24 (4%)	3.2 (16%)	6,046	5.748 (28.74%)	0,0002054	0	0
320 (24%)	24 (4%)	2.4 (12%)	6,65	6.36 (31.84%)	2,622E-07	0	0
267 (20%)	30 (5%)	5 (25%)	7,453	7.033 (35.17%)	0,000001053	0	0
334 (25%)	30 (5%)	4 (20%)	7,63	7.43 (37.14%)	0,0584	0	0
400 (30%)	30 (5%)	3 (15%)	7,27	7.22 (36.12%)	0,8129	0	0
320 (24%)	36 (6%)	6 (30%)	8,068	8.021(40.36%)	0,3041	0	0
400 (30%)	36 (6%)	4.8 (24%)	8,669	8.256 (41.35%)	0,8185	0	0
480 (36%)	36 (6%)	3.6 (18%)	8,97	8,86 (48,90%)	1,3136	0	0
374 (28%)	42 (7%)	7 (35%)	9,445	9.167 (45.84%)	0,6317	0	0,000000163
467 (35%)	42 (7%)	5.6 (28%)	10,75	10.23(51.19%)	0,81	0	0
560 (42%)	42 (7%)	4.2 (21%)	9,12	9.058(45.21%)	2,11	0	0
427 (32%)	48 (8%)	8 (40%)	10,95	10.14(54.78%)	0,5315	0	0,00000998
534 (40%)	48 (8%)	6.4 (32%)	11,49	11.32 (57.48%)	1,577	0	0
640 (48%)	48 (8%)	4.8 (24%)	10,48	10,09 (52,46%)	2,756	0	0
480 (36%)	54 (9%)	9 (45%)	13,14	11,98 (65,74%)	1,735	0	0,0001048
600 (45%)	54 (9%)	7.2 (36%)	13,85	12,39 (69.27%)	2,0311	0	0,0000397
720 (54%)	54 (9%)	5.4 (27%)	11,67	10,98 (%)	3,774	0	0,0000324
534	59 (10%)	10 (50%)	14,22	12,87 (71,11%)	2,30093	0	0,00098

(40%)							
667 (50%)	59 (10%)	8 (40%)	12,93	15,76 (78,81%)	4,156	0	0,0000348
800 (60%)	59 (10%)	6 (30%)	13,02	11.66 (65.12%)	4,67	0	0

Table B1. 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%)	15,40%	4,04%	7,81%
267 (20%)	24 (4%)	3.2 (16%)	18,96%	4,04%	5,74%
320 (24%)	24 (4%)	2.4 (12%)	25,03%	4,01%	3,77%
267 (20%)	30 (5%)	5 (25%)	19,49%	4,87%	10,81%
334 (25%)	30 (5%)	4 (20%)	24,67%	4,91%	7,56%
400 (30%)	30 (5%)	3 (15%)	27,73%	3,31%	5,07%
320 (24%)	36 (6%)	6 (30%)	21,78%	5,57%	13,02%
400 (30%)	36 (6%)	4.8 (24%)	27,14%	4,61%	9,56%
480 (36%)	36 (6%)	3.6 (18%)	33,57%	4,52%	6,70%
374 (28%)	42 (7%)	7 (35%)	25,29%	5,31%	15,78%
467 (35%)	42 (7%)	5.6 (28%)	33,39%	4,77%	12,48%
560 (42%)	42 (7%)	4.2 (21%)	3,05%	4,55%	10,12%
427 (32%)	48 (8%)	8 (40%)	30,07%	6,09%	19,65%
534 (40%)	48 (8%)	6.4 (32%)	36,29%	5,04%	15,10%
640 (48%)	48 (8%)	4.8 (24%)	37,81%	4,53%	10,12%
480 (36%)	54 (9%)	9 (45%)	34,73%	8,20%	22,82%
600 (45%)	54 (9%)	7.2 (36%)	44,57%	7,51%	17,15%
720 (54%)	54 (9%)	5.4 (27%)	42,94%	3,97%	11,43%
534 (40%)	59 (10%)	10 (50%)	38,76%	7,04%	25,30%
667 (50%)	59 (10%)	8 (40%)	42,93%	5,71%	19,70%
800 (60%)	59 (10%)	6 (30%)	47,23%	4,45%	13,42%

Table B2. Bandwidth that is really reserved by each type of traffic, with loads between 40%-100% of the total bandwidth.

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Revenue
214 (16%)	24 (4%)	4 (20%)	5,14E+03
267 (20%)	24 (4%)	3.2 (16%)	5,03E+03
320 (24%)	24 (4%)	2.4 (12%)	5,02E+03
267 (20%)	30 (5%)	5 (25%)	6,11E+03
334 (25%)	30 (5%)	4 (20%)	6,22E+03
400 (30%)	30 (5%)	3 (15%)	4,22E+03
320 (24%)	36 (6%)	6 (30%)	7,02E+03
400 (30%)	36 (6%)	4.8 (24%)	5,90E+03
480 (36%)	36 (6%)	3.6 (18%)	5,77E+03
374 (28%)	42 (7%)	7 (35%)	6,74E+03
467 (35%)	42 (7%)	5.6 (28%)	6,07E+03
560 (42%)	42 (7%)	4.2 (21%)	5,87E+03
427 (32%)	48 (8%)	8 (40%)	6,39E+03
534 (40%)	48 (8%)	6.4 (32%)	7,76E+03
640 (48%)	48 (8%)	4.8 (24%)	5,80E+03
480 (36%)	54 (9%)	9 (45%)	1,04E+04
600 (45%)	54 (9%)	7.2 (36%)	9,80E+04
720 (54%)	54 (9%)	5.4 (27%)	5,84E+03
534 (40%)	59 (10%)	10 (50%)	3,68E+05
667 (50%)	59 (10%)	8 (40%)	7,31E+04
800 (60%)	59 (10%)	6 (30%)	5,87E+03

Table B3. Provider Revenue for various loads

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Average LUSF for accepted voice users							
			CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
214 (16%)	24 (4%)	4 (20%)	0,4026	0,45	0,4499	0,4499	0,35	0,4994	0,4999	0,35
267 (20%)	24 (4%)	3.2 (16%)	0,45	0,4	0,4497	0,3	0,4988	0,45	0,3	0,4069
320 (24%)	24 (4%)	2.4 (12%)	0,5	0,5	0,5	0,5	0,5	0,5	0,5	0,5
267 (20%)	30 (5%)	5 (25%)	0,4998	0,4999	0,4994	0,4	0,4995	0,4999	0,3999	0,4569
334 (25%)	30 (5%)	4 (20%)	0,4374	0,4375	0,4985	0,2499	0,4977	0,4374	0,25	0,4012
400 (30%)	30 (5%)	3 (15%)	0,25	0,25	0,4958	0,25	0,496	0,3125	0,25	0,3291
320 (24%)	36 (6%)	6 (30%)	0,3123	0,3748	0,4978	0,1849	0,496	0,3747	0,1875	0,3472
400 (30%)	36 (6%)	4.8 (24%)	0,3748	0,4062	0,499	0,2811	0,496	0,4061	0,2811	0,392
480 (36%)	36 (6%)	3.6 (18%)	0,3885	0,3885	0,4963	0,2776	0,4951	0,3883	0,2775	0,3874
374 (28%)	42 (7%)	7 (35%)	0,3882	0,333	0,4969	0,2776	0,4941	0,3327	0,2776	0,3714
467 (35%)	42 (7%)	5.6 (28%)	0,4439	0,4437	0,4973	0,3885	0,4955	0,4437	0,3885	0,443
560 (42%)	42 (7%)	4.2 (21%)	0,4984	0,2907	0,4128	0,0831	0,49	0,3742	0,0831	0,3189
427 (32%)	48 (8%)	8 (40%)	0,4431	0,3881	0,4927	0,2218	0,4929	0,4431	0,2219	0,3862
534 (40%)	48 (8%)	6.4 (32%)	0,4427	0,4431	0,442	0,3878	0,4937	0,4429	0,3877	0,4343
640 (48%)	48 (8%)	4.8 (24%)	0,4424	0,2765	0,385	0,2766	0,492	0,3318	0,2766	0,3544
480 (36%)	54 (9%)	9 (45%)	0,4424	0,4572	0,49554	0,4677	0,4945	0,4656	0,4678	0,475
600 (45%)	54 (9%)	7.2 (36%)	0,497	0,4973	0,4947	0,4424	0,4948	0,4968	0,4425	0,4808
720 (54%)	54 (9%)	5.4 (27%)	0,4417	0,3314	0,4378	0,2124	0,4885	0,3477	0,2211	0,3611
534 (40%)	59 (10%)	10 (50%)	0,4966	0,4961	0,4919	0,4427	0,4902	0,4958	0,4426	0,4794
667 (50%)	59 (10%)	8 (40%)	0,3857	0,3859	0,4913	0,3307	0,4908	0,4402	0,3314	0,408
800 (60%)	59 (10%)	6 (30%)	0,2479	0,4958	0,4938	0,2482	0,4867	0,3719	0,2482	0,3703

Table B4. LUSF per cell for accepted voice users

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Average "Number of Tries" for accepted voice users							
			CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
214 (16%)	24 (4%)	4 (20%)	0,0000332	0	0	0	0	0	0	4,74E-06
267 (20%)	24 (4%)	3.2 (16%)	0,00001294	0	0	0	0,005017	0	0	0,0007186
320 (24%)	24 (4%)	2.4 (12%)	0,00003095	0	0	0	0	0	0	1,75E-06
267 (20%)	30 (5%)	5 (25%)	0,000008721	0	0	0	0	0	0	1,246E-06
334 (25%)	30 (5%)	4 (20%)	0,0000279	0	0	0	0,004371	0	0	0,0006285
400 (30%)	30 (5%)	3 (15%)	0,000016	0	0,3096	0	0,2164	0	0	0,07516
320 (24%)	36 (6%)	6 (30%)	0,00000355	0	0,0248	0	0,02669	0	0	0,007362
400 (30%)	36 (6%)	4.8 (24%)	0,00002521	0	0	0	0,2492	0	0	0,0356
480 (36%)	36 (6%)	3.6 (18%)	0,00002523	0	0,1591	0	0,291	0	0	0,06431
374 (28%)	42 (7%)	7 (35%)	0,00001096	0	0,0377	0	0,174	0	0	0,03024
467 (35%)	42 (7%)	5.6 (28%)	0,00004672	0	0,1303	0	0,2496	0	0	0,05427
560 (42%)	42 (7%)	4.2 (21%)	0,00006779	0	0,107	0	0,5284	0	0	0,09081
427 (32%)	48 (8%)	8 (40%)	0,00001992	0	0,6321	0	0,2172	0	0	0,1213
534 (40%)	48 (8%)	6.4 (32%)	0,00003709	0	0	0	0,2324	0	0	0,0332
640 (48%)	48 (8%)	4.8 (24%)	0,00003439	0	0,1056	0	0,4164	0	0	0,07458
480 (36%)	54 (9%)	9 (45%)	0,00000084	0	0	0	0,05262	0	0	0,007518
600 (45%)	54 (9%)	7.2 (36%)	0,0000127	0	0,0943	0	0,03668	0	0	0,01871
720 (54%)	54 (9%)	5.4 (27%)	0,0000667	0	0,5191	0	0,9971	0	0	0,2166
534 (40%)	59 (10%)	10 (50%)	0,0000597	0	0,3447	0	0,2929	0	0	0,0911
667 (50%)	59 (10%)	8 (40%)	0,0000597	0	0,2547	0	0,3035	0	0	0,07975
800 (60%)	59 (10%)	6 (30%)	0,00001373	0	0	0	0,8072	0	0	0,1153

Table B5. "Number of Tries" for accepted voice users

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Average "Number of Tries" for accepted video users							
			CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
214 (16%)	24 (4%)	4 (20%)	0,3137	0,1316	0,6411	0,1853	0,9104	0,3882	0,2108	0,397292857
267 (20%)	24 (4%)	3.2 (16%)	0,554	0,03497	0,9205	0,2199	1,061	0,4079	0,2179	0,4881
320 (24%)	24 (4%)	2.4 (12%)	0,3902	0,05437	1,116	0,4707	1,066	0,8088	0,4883	0,6278
267 (20%)	30 (5%)	5 (25%)	0,2393	0,06305	0,4925	0,1446	0,7177	0,1476	0,06852	0,2676
334 (25%)	30 (5%)	4 (20%)	0,0922	0,02996	0,2736	0,0811	0,6128	0,05337	0,01409	0,1653
400 (30%)	30 (5%)	3 (15%)	0,3179	0,06279	0,6887	0,22032	0,8736	0,36117	0,19992	0,3892
320 (24%)	36 (6%)	6 (30%)	0,2144	0,02944	0,6782	0,03282	0,7029	0,1862	0,0241	0,2669
400 (30%)	36 (6%)	4.8 (24%)	0,2425	0,04698	0,7145	0,07069	0,8693	0,2523	0,0241	0,3197
480 (36%)	36 (6%)	3.6 (18%)	0,4077	0,07899	0,5762	0,1706	0,9212	0,3524	0,1572	0,3806
374 (28%)	42 (7%)	7 (35%)	0,2039	0,1524	0,5081	0,05359	0,5524	0,0917	0,0051	0,2067
467 (35%)	42 (7%)	5.6 (28%)	0,215	0,0462	0,4268	0,04317	0,7778	0,1476	0,1006	0,251
560 (42%)	42 (7%)	4.2 (21%)	0,4086	0,08399	0,6977	0,1306	0,7012	0,2547	0,05284	0,3328
427 (32%)	48 (8%)	8 (40%)	0,1159	0,1291	0,3008	0,01064	0,6255	0,0619	0,01135	0,1648
534 (40%)	48 (8%)	6.4 (32%)	0,1731	0,01335	0,5175	0,05511	0,7317	0,1355	0,06948	0,2422
640 (48%)	48 (8%)	4.8 (24%)	0,3709	0,09625	0,6189	0,14131	0,7545	0,2039	0,06427	0,3214
480 (36%)	54 (9%)	9 (45%)	0,0493	0,02209	0,2801	0,00788	0,5166	0,07519	0,00163	0,1361
600 (45%)	54 (9%)	7.2 (36%)	0,145	0,02804	0,3426	0,05658	0,6681	0,07891	0,01689	0,1909
720 (54%)	54 (9%)	5.4 (27%)	0,2917	0,06167	0,469	0,04317	0,8336	0,2514	0,04223	0,2848
534 (40%)	59 (10%)	10 (50%)	0,0433	0,02483	0,2777	0,0071	0,5275	0,03302	0,01098	0,132
667 (50%)	59 (10%)	8 (40%)	0,0922	0,02996	0,2736	0,0811	0,6128	0,05337	0,01409	0,1653
800 (60%)	59 (10%)	6 (30%)	0,1961	0,07418	0,3733	0,1139	0,7512	0,1228	0,0677	0,2428

Table B6. "Number of Tries" for accepted video users

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Average LUSF for rejected voice users							
			CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
214 (16%)	24 (4%)	4 (20%)	0,0206	0,02059	0,0206	0	0,0412	0,4119	0	0,02059
267 (20%)	24 (4%)	3.2 (16%)	0,0412	0,0206	0,04119	0	0,0618	0,0412	0	0,02942
320 (24%)	24 (4%)	2.4 (12%)	0	0	0	0	0	0	0	0
267 (20%)	30 (5%)	5 (25%)	0,206	0,206	0,206	0,206	0,206	0,206	0,206	0,206
334 (25%)	30 (5%)	4 (20%)	0	0	0,02574	0	0,0863	0	0	0,016
400 (30%)	30 (5%)	3 (15%)	0,0322	0,03218	0,0655	0,0322	0,067	0,0386	0,0322	0,04282
320 (24%)	36 (6%)	6 (30%)	0,0515	0,05149	0,103	0,0515	0,0977	0,0772	0,0515	0,06912
400 (30%)	36 (6%)	4.8 (24%)	0,0644	0,09011	0,09011	0,0386	0,0679	0,0901	0,0386	0,06854
480 (36%)	36 (6%)	3.6 (18%)	0,0458	0,02288	0,07367	0,0229	0,1085	0,0229	0,0229	0,03092
374 (28%)	42 (7%)	7 (35%)	0,0687	0,09154	0,07589	0,0458	0,0697	0,0687	0,0458	0,06657
467 (35%)	42 (7%)	5.6 (28%)	0	0	0,00982	0	0,035	0	0	0,0064
560 (42%)	42 (7%)	4.2 (21%)	0,0858	0,03433	0,1217	0,0343	0,1514	0,0858	0,3433	0,07826
427 (32%)	48 (8%)	8 (40%)	0,0229	0,02288	0,1046	0,0229	0,154	0,0458	0,0229	0,05657
534 (40%)	48 (8%)	6.4 (32%)	0,0458	0,04577	0,04577	0,0458	0,0182	0,0458	0,0458	0,04182
640 (48%)	48 (8%)	4.8 (24%)	0,0687	0,04576	0,08556	0,0458	0,0637	0,0458	0,0458	0,05729
480 (36%)	54 (9%)	9 (45%)	0,1287	0	0,01287	0	0,0049	0	0	0,00438
600 (45%)	54 (9%)	7.2 (36%)	0	0	0,02396	0	0,024	0	0	0,00685
720 (54%)	54 (9%)	5.4 (27%)	0,0458	0,04577	0,07764	0,0469	0,0646	0,0458	0,0229	0,05314
534 (40%)	59 (10%)	10 (50%)	0	0	0,02396	0	0,0479	0	0	0,01027
667 (50%)	59 (10%)	8 (40%)	0,0458	0,04577	0,0111	0,0229	0,0415	0,0229	0,0229	0,0304
800 (60%)	59 (10%)	6 (30%)	0,103	0,103	0,06709	0,0515	0,028	0,103	0,0858	0,07258

Table B7. LUSF per cell for rejected voice users

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Average LUSF for Rejected Data Users							
			CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
214 (16%)	24 (4%)	4 (20%)	0	0	0	0	0	0	0	0
267 (20%)	24 (4%)	3.2 (16%)	0	0	0	0	0,05465	0	0	0,0078071
320 (24%)	24 (4%)	2.4 (12%)	0	0	0	0	0	0	0	0
267 (20%)	30 (5%)	5 (25%)	0	0	0	0	0	0	0	0
334 (25%)	30 (5%)	4 (20%)	0,06037	0	0,05825	0	0,1624	0	0	0,04016
400 (30%)	30 (5%)	3 (15%)	0,06037	0	0,2201	0	0,2154	0	0	0,07085
320 (24%)	36 (6%)	6 (30%)	0,06047	0	0,1826	0	0,276	0	0	0,07416
400 (30%)	36 (6%)	4.8 (24%)	0	0	0,0905	0	0,1723	0	0	0,03755
480 (36%)	36 (6%)	3.6 (18%)	0,2148	0	0,1648	0	0,1325	0	0	0,07316
374 (28%)	42 (7%)	7 (35%)	0,05373	0	0,1694	0	0,1942	0	0	0,05961
467 (35%)	42 (7%)	5.6 (28%)	0	0	0,07238	0	0,2493	0	0	0,04596
560 (42%)	42 (7%)	4.2 (21%)	0,2012	0	0,2038	0	0,3052	0	0	0,1014
427 (32%)	48 (8%)	8 (40%)	0,2687	0	0,3662	0	0,388	0	0	0,1461
534 (40%)	48 (8%)	6.4 (32%)	0,05367	0	0,3218	0	0,4075	0	0	0,1118
640 (48%)	48 (8%)	4.8 (24%)	0,1611	0	0,3212	0	0,3269	0	0	0,1156
480 (36%)	54 (9%)	9 (45%)	0,3917	0,03018	0,4701	0	0,4977	0,4516	0	0,263
600 (45%)	54 (9%)	7.2 (36%)	0,1612	0,05366	0,455	0	0,4943	0,1609	0	0,1893
720 (54%)	54 (9%)	5.4 (27%)	0,2684	0,1606	0,3805	0	0,3508	0,2677	0	0,2008
534 (40%)	59 (10%)	10 (50%)	0,4817	0,4837	0,3459	0,4529	0,2644	0,4237	0,3857	0,1059
667 (50%)	59 (10%)	8 (40%)	0,3747	0,3184	0,3571	0,2702	0,4182	0,3067	0,1071	0,3074
800 (60%)	59 (10%)	6 (30%)	0,2399	0,3493	0,4273	0,2408	0,2819	0,2117	0,1204	0,2674

Table B8. LUSF per cell for rejected data users

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Average LUSF for rejected video users							
			CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
214 (16%)	24 (4%)	4 (20%)	0,32025	0,2159	0,1186	0,2258	0,09514	0,1802	6,11E-06	0,1651
267 (20%)	24 (4%)	3.2 (16%)	0,2026	0,1304	0,129	0,2629	0,09262	0,2077	1,294E-05	0,1464
320 (24%)	24 (4%)	2.4 (12%)	0,169	0,1639	0,1283	0,1481	0,09458	0,1798	3,095E-05	0,1262
267 (20%)	30 (5%)	5 (25%)	0,2505	0,1825	0,1376	0,2139	0,09995	0,3058	8,721E-06	0,17
334 (25%)	30 (5%)	4 (20%)	0,2149	0,1469	0,1211	0,1247	0,09149	0,1678	0,0000279	0,2391
400 (30%)	30 (5%)	3 (15%)	0,165	0,1653	0,1257	0,3055	0,0866	0,1697	1,612E-05	0,1457
320 (24%)	36 (6%)	6 (30%)	0,217	0,1851	0,13	0,136	0,09887	0,239	0,0000056	0,1437
400 (30%)	36 (6%)	4.8 (24%)	0,2216	0,2424	0,1225	0,2374	0,08843	0,2431	0,000025	0,165
480 (36%)	36 (6%)	3.6 (18%)	0,1953	0,2466	0,1402	0,1349	0,08835	0,1919	2,525E-05	0,1425
374 (28%)	42 (7%)	7 (35%)	0,3145	0,2379	0,1425	0,1436	0,1018	0,3637	1,992E-05	0,1863
467 (35%)	42 (7%)	5.6 (28%)	0,1908	0,1738	0,1587	0,2025	0,1038	0,261	0,0000467	0,1458
560 (42%)	42 (7%)	4.2 (21%)	0,1791	0,24	0,1379	0,4001	0,1033	0,2388	6,767E-05	0,1865
427 (32%)	48 (8%)	8 (40%)	0,2278	0,19001	0,1416	0,2151	0,1114	0,3407	1,992E-05	0,1752
534 (40%)	48 (8%)	6.4 (32%)	0,2818	0,1343	0,1416	0,3134	0,1037	0,1762	3,709E-05	0,1644
640 (48%)	48 (8%)	4.8 (24%)	0,2284	0,2084	0,1333	0,1738	0,09256	0,2376	0,000034	0,1534
480 (36%)	54 (9%)	9 (45%)	0,4419	0,1947	0,164	0,0852	0,1038	0,3614	0,0000084	0,193
600 (45%)	54 (9%)	7.2 (36%)	0,1773	0,2917	0,1545	0,1203	0,09674	0,1892	0,0000127	0,1471
720 (54%)	54 (9%)	5.4 (27%)	0,194	0,2905	0,1344	0,0878	0,0991	0,21	0,0000791	0,1451
534 (40%)	59 (10%)	10 (50%)	0,2279	0,2908	0,1707	0,0538	0,09531	0,2055	0,0000597	0,1497
667 (50%)	59 (10%)	8 (40%)	0,3257	0,1925	0,2012	0,2099	0,0856	0,4302	0,01209	0,2156
800 (60%)	59 (10%)	6 (30%)	0,3382	0,256	0,1381	0,2015	0,09039	0,276	0,000013	0,1863

Table B9. LUSF per cell for rejected video users

Number	Messages	Maximum	Average “Number of Tries” voice - rejected

of Voice Users	per second	Video Capacity (Mbps)	CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
214 (16%)	24 (4%)	4 (20%)	0	0	0	0	0	0	0	0
267 (20%)	24 (4%)	3.2 (16%)	0	0	0	0	9,193E-05	0	0	1,313E-05
320 (24%)	24 (4%)	2.4 (12%)	0	0	0	0	0	0	0	0
267 (20%)	30 (5%)	5 (25%)	0	0	0	0	0	0	0	0
334 (25%)	30 (5%)	4 (20%)	0	0	0	0	0,125	0	0	0,0699
400 (30%)	30 (5%)	3 (15%)	0	0	0,089	0	0,08613	0	0	0,025
320 (24%)	36 (6%)	6 (30%)	0	0	0,0005	0	0,06195	0	0	0,00089
400 (30%)	36 (6%)	4.8 (24%)	0	0	0	0	0,4817	0	0	0,0688
480 (36%)	36 (6%)	3.6 (18%)	0	0	0,6719	0	0,4878	0	0	0,1656
374 (28%)	42 (7%)	7 (35%)	0	0	0,3113	0	0,3015	0	0	0,08753
467 (35%)	42 (7%)	5.6 (28%)	0	0	0,285	0	0,5613	0	0	0,121
560 (42%)	42 (7%)	4.2 (21%)	0	0	0,4865	0	1,7475	0	0	0,3191
427 (32%)	48 (8%)	8 (40%)	0	0	1,3079		1,09066	0	0	0,334
534 (40%)	48 (8%)	6.4 (32%)	0	0	0	0	0,5657	0	0	0,0808
640 (48%)	48 (8%)	4.8 (24%)	0	0	0,4466	0	0,908	0	0	0,1935
480 (36%)	54 (9%)	9 (45%)	0	0	0	0	0,1695	0	0	0,02422
600 (45%)	54 (9%)	7.2 (36%)	0	0	0,3243	0	0,3243	0	0	0,09266
720 (54%)	54 (9%)	5.4 (27%)	0	0	1,007	0	1,6376	0	0	0,3778
534 (40%)	59 (10%)	10 (50%)	0	0	0,3243	0	0,6486	0	0	0,139
667 (50%)	59 (10%)	8 (40%)	0	0	0,76	0	0,9527	0	0	0,2446
800 (60%)	59 (10%)	6 (30%)	0	0	0,7795	0	1,648	0	0	0,3468

Table B10. “Number of Tries” for rejected voice users

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Average “Number of Tries” data - rejected							
			CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
214 (16%)	24 (4%)	4 (20%)	0	0	0	0	0	0	0	0
267 (20%)	24 (4%)	3.2 (16%)	0	0	0	0	0,5021	0	0	0,071728571
320 (24%)	24 (4%)	2.4 (12%)	0	0	0	0	0	0	0	0
267 (20%)	30 (5%)	5 (25%)	0	0	0	0	0	0	0	0
334 (25%)	30 (5%)	4 (20%)	0,2463	0	0,6586	0	4,191	0	0	0,7275
400 (30%)	30 (5%)	3 (15%)	0,2395	0	5,773	0	4,6482	0	0	1,523
320 (24%)	36 (6%)	6 (30%)	0,1521	0	1,775	0	19,68	0	0	3,087
400 (30%)	36 (6%)	4.8 (24%)	0	0	3,468	0	5,023	0	0	1,21
480 (36%)	36 (6%)	3.6 (18%)	0,663	0	55,27	0	113,24	0	0	24,16
374 (28%)	42 (7%)	7 (35%)	0,1605	0	4,283	0	11,056	0	0	2,214
467 (35%)	42 (7%)	5.6 (28%)	0	0	5,893	0	25,97	0	0	4,55
560 (42%)	42 (7%)	4.2 (21%)	0,8198	0	1,754	0	18,4	0	0	2,996
427 (32%)	48 (8%)	8 (40%)	0,6435	0	27,52	0	59,47	0	0	12,52
534 (40%)	48 (8%)	6.4 (32%)	0,2222	0	12,92	0	15,51	0	0	4,094
640 (48%)	48 (8%)	4.8 (24%)	0,4804	0	18,57	0	149,41	0	0	24,07
480 (36%)	54 (9%)	9 (45%)	1,592	0,125	9,152	0	16,56	4,2006	0	4,52025
600 (45%)	54 (9%)	7.2 (36%)	0,3752	0,2222	16,49	0	14,38	1	0	4,63
720 (54%)	54 (9%)	5.4 (27%)	0,9502	0,7381	172,105	0	215,237	1,1601	0	55,764
534 (40%)	59 (10%)	10 (50%)	2,077	7,3235	9,523	1,7478	178,29	4,9444	0,6938	29,22
667 (50%)	59 (10%)	8 (40%)	1,64	1,91	16,71	4,79	16,15	3,001	0,4758	6,387
800 (60%)	59 (10%)	6 (30%)	1,235	2,391	7,969	1,119	442,08	3,001	0,56	65,47

Table B11. “Number of Tries” for rejected data users

Number of Voice Users	Messages per second	Maximum Video Capacity (Mbps)	Average "Number of Tries" video - rejected							
			CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
214 (16%)	24 (4%)	4 (20%)	2,333	2,902	4,4565	2,125	5,326	3,6095	3,3627	3,444957143
267 (20%)	24 (4%)	3.2 (16%)	3,314	2,743	4,526	3,028	5,58	3,168	0,8198	3,311
320 (24%)	24 (4%)	2.4 (12%)	3,85	3,432	4,428	3,316	5,541	3,471	2,276	3,759
267 (20%)	30 (5%)	5 (25%)	2,252	2,041	4,302	2,003	5,207	2,302	0,4389	2,65
334 (25%)	30 (5%)	4 (20%)	2,2421	1,3175	3,4196	1,6005	5,3924	0,8888	7,014	3,106
400 (30%)	30 (5%)	3 (15%)	2,96	2,16	4,412	2,23	5,699	3,75	1,759	3,28
320 (24%)	36 (6%)	6 (30%)	3,209	1,374	4,4715	1,23025	5,319	2,248	1,092	2,7065
400 (30%)	36 (6%)	4.8 (24%)	3,141	2,33	4,54	1,4	5,54	2,72	2,232	3,138
480 (36%)	36 (6%)	3.6 (18%)	3,504	2,497	4,2733	1,6245	5,617	3,257	1,6003	3,196
374 (28%)	42 (7%)	7 (35%)	2,4046	0,6969	4,125	1,038	5,25	1,3463	1,8983	2,3996
467 (35%)	42 (7%)	5.6 (28%)	2,786	2,215	3,763	0,994	5,155	2,561	3,933	3,058
560 (42%)	42 (7%)	4.2 (21%)	3,347	2,7315	4,153	1,878	5,267	2,816	6,931	3,875
427 (32%)	48 (8%)	8 (40%)	2,588	1,0221	3,9973	0,4444	4,85	1,0551	2,67	2,375
534 (40%)	48 (8%)	3,106	1,211	4,098	0,9974	5,1136	2,2713	5,3313	3,1613	3,1691
640 (48%)	48 (8%)	4.8 (24%)	3,17	2,36	4,45	1,21	5,49	2,91	2,66	3,18
480 (36%)	54 (9%)	9 (45%)	1,4396	1,1762	3,6205	0,4971	5,1035	1,5781	0,9451	2,052
600 (45%)	54 (9%)	7.2 (36%)	2,222	0,666	3,73	1,097	5,21	1,535	0,6733	2,17
720 (54%)	54 (9%)	5.4 (27%)	2,8467	1,4735	4,3117	1,3243	5,3002	3,1126	3,7487	3,1602
534 (40%)	59 (10%)	10 (50%)	1,3206	0,8863	3,5028	0,1111	5,2128	0,9918	5,4444	2,495
667 (50%)	59 (10%)	8 (40%)	2,2421	1,3175	3,4196	1,6005	5,3924	0,8888	7,014	3,106
800 (60%)	59 (10%)	6 (30%)	2,071	0,1913	4,2845	2,234	5,508	2,4495	0,754	2,74

Table B12. "Number of Tries" for rejected video users

Maximum Allowed Video Load (Mbps)	CAC Bandwidth Estimation (Mbps)	Real Bandwidth (Mbps)	Average Number of Users	Dropped Video Packets (%)	REVENUE
8 (40%)	5.667	3.813 (19.06%)	11	2.16E-07	2.55E+03
10 (50%)	6.916	4.969 (24.85%)	14	0.00%	3.18E+03
12 (60%)	8.14	6.07 (30.35%)	17	0.01%	3.94E+03
14 (70%)	9.374	7.251 (36.25%)	20	0.06%	4.60E+03
16 (80%)	10.61	8.409 (42.04%)	23	0.21%	5.23E+03
18 (90%)	11.78	9.575 (47.86%)	26	0.54%	5.789.5
20 (100%)	12.38	10.11 (50.59%)	29	0.78%	6.42E+03
22 (110%)	13.39	10.05 (50.10%)	32	0.48%	6.87E+03

Table B13. Average number of active users in the system for various loads

Maximum Allowed Video Load (Mbps)	Active Users	Downgrade percentage HQ	Downgrade percentage MQ	Degradation Ratio HQ	Degradation Ratio MQ
8 (40%)	96	37.57%	34.86%	43.57%	46%
10 (50%)	119	30.57%	33.57%	41.29%	42.57%
12 (60%)	149	32.00%	31.29%	39.43%	33.43%
14 (70%)	171	24.43%	28.14%	31.86%	31.14%
16 (80%)	200	22.71%	27.00%	30.00%	29.14%
18 (90%)	226	22%	23%	27%	27%
20 (100%)	250	18%	22%	25%	24%
22 (110%)	278	20%	23%	27%	26%

Table B14. Percentage of degradations and degradation ratio

Maximum Allowed Video Load (Mbps)	Average Long User Satisfaction Factor video -rejected							
	CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
8 (40%)	0.2172	0.1925	0.1522	0.1124	0.09211	0.2409	0	**0.1439**
10 (50%)	0.416	0.2336	0.1689	0.0484	0.1089	0.2904	0	**0.1809**
12 (60%)	0.4553	0.0484	0.1691	0.0484	0.1018	0.1875	0	**0.1444**
14 (70%)	0.191	0.06899	0.1742	0.0484	0.1005	0.1864	0	**0.1099**
16 (80%)	0.2904	0.06939	0.1853	0	0.1036	0.09679	0	**0.1065**
18 (90%)	0.424	0	0.295	0	0.1021	0	0	**0.09125**
20 (100%)	0.1874	0	0.2371	0	0.1052	0.1037	0	**0.0905**
22 (110%)	0.1815	0	0.2805	0	0.1134	0.0605	0	**0.09084**

Table B15. LUSF for rejected video users.

Maximum Allowed Video Load (Mbps)	Average "Number of Tries" video - rejected							
	CELL A	CELL B	CELL C	CELL D	CELL E	CELL F	CELL G	TOTAL
8 (40%)	1.673	1.443	3.875	0.5395	5.45	2.017	0.5479	**2.221**
10 (50%)	1.647	0.8173	4.017	0.1	5.037	0.6	0.214	**1.776**
12 (60%)	1.198	0.1	3.546	0.1	5.206	0.8756	0	**1.575**
14 (70%)	0.8929	0.2	3.884	0.1	5.186	0.8835	0	**1.592**
16 (80%)	0.6	0.2	3.769	0	5.112	0.2	0	**1.41**
18 (90%)	0.5	0	2.9216	0	5.1733	0	0	**1.2278**
20 (100%)	0.5671	0	3.2224	0	5.004	0.2142	0	**1.2868**
22 (110%)	0.375	0	2.9047	0	4.994	0.125	0	**1.199**

Table B16. Number of tries for rejected video users