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Dealing with VoIP Calls During "Busy Hour" in LTE

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1. Introduction

Long Term Evolution (LTE) is an evolving wireless standard developed by the 3rd Generation Partnership Project (3GPP) which, along with 3GPP HSPA+, 3GPP EDGE Evolution and Mobile WiMAX (IEEE 802.16e), opens the road to 4G technologies. The standard is focused on delivering high data rates for bandwidth-demanding applications and on improving flexibility and spectral efficiency, thus constituting an attractive solution for both end users and mobile operators. An important feature of LTE that differentiates it from conventional mobile standards is the all-IP packet based network architecture, which further ensures the seamless integration of internet applications and facilitates the convergence between fixed and mobile systems.

The radio interface of LTE is based on Orthogonal Frequency Division Multiplexing (OFDM) and supports Multiple-Input-Multiple-Output (MIMO) technology. The standard defines asymmetrical data rates and modulations for uplink and downlink, using different access schemes for each link. In particular, Orthogonal Frequency Division Multiple Access (OFDMA) is employed in the downlink, while the technically similar but less power-demanding Single Carrier – Frequency Division Multiple Access (SC-FDMA) is used in the uplink. In terms of the wireless spectrum allocation, LTE supports variable channel bandwidths that vary from 1.4 to 20 MHz and can be deployed in different frequency bands.

The LTE architecture, referred to as Evolved Packet System (EPS) comprises the Evolved Radio Access Network (E-UTRAN) and the Evolved Packet Core (EPC), illustrated in Fig. 1 (3GPP, 2010). The E-UTRAN consists of a network of enhanced base stations called evolved Nodes B (eNBs) whose main role is to manage the radio resource and mobility in the cell in order to optimize the communication among all User Equipments (UEs). The eNBs can communicate with each other through the X2 interface and can access the EPC by means of the S1 interface. On the other hand, the EPC consists of a control plane node called the Mobility Management Entity (MME) and two user plane nodes, the Serving Gateway and the Packet Data Network Gateway (PDN Gateway or P-GW). These control planes handle the data packet routing within the LTE and towards non-3GPP data networks, respectively.

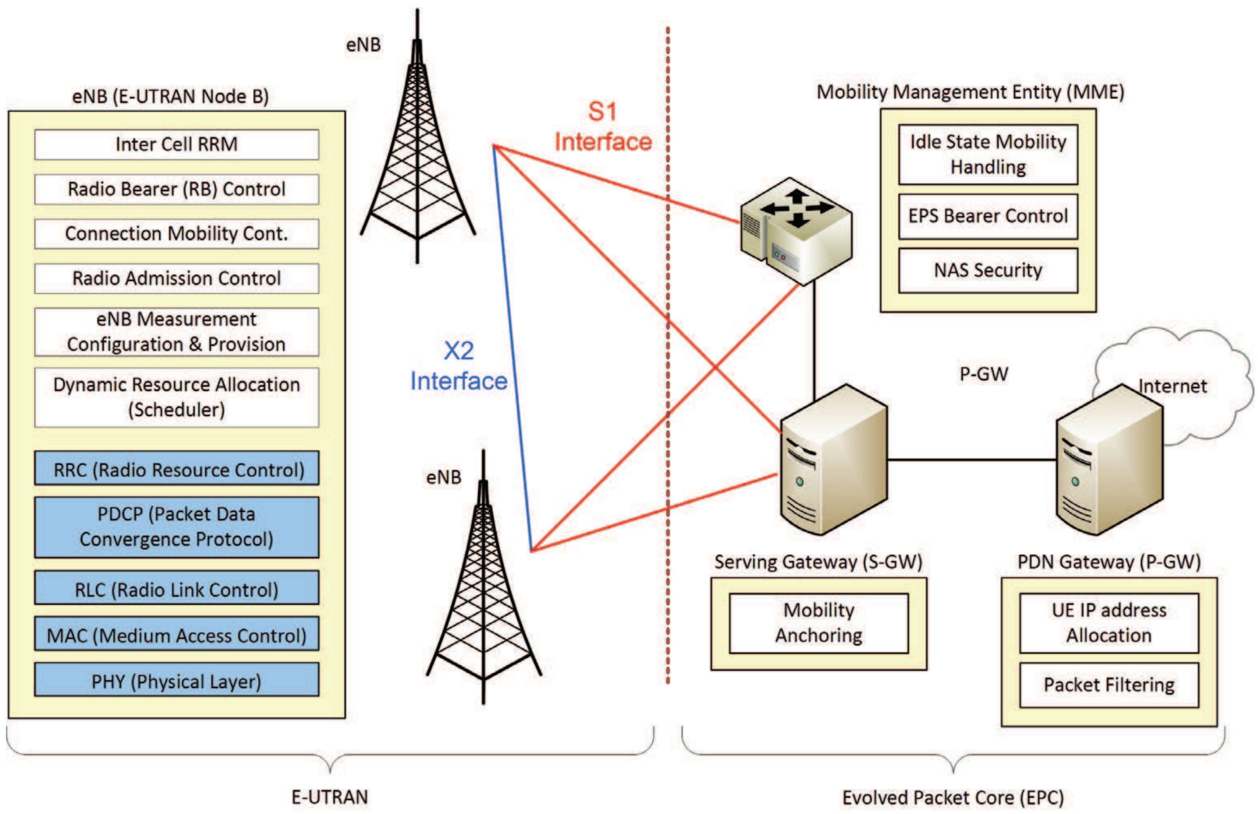

Fig. 1. LTE Architecture and functional split between E-UTRAN and EPC

LTE provides service differentiation by adopting a class-based Quality of Service (QoS) concept. In particular, each data flow between the user equipment and the P-GW (called EPS bearer) is assigned a QoS profile. A total of nine profiles are defined in the specification (3GPP, 2011) that can be mapped to different types of applications such as real time video and voice services, online gaming, etc. Each profile involves the bearer type, the flow priority, an upper bound for the packet delay and the packet error rate. The bearer type indicates whether a Guaranteed Bit Rate (GBR) will be provided to the bearer by permanently allocating network resources during the data session.

The essential difference between GBR and non-GBR bearers is that, in the first case, a connection may be blocked if the network does not have the resources to guarantee the desired QoS of these connections. This concept is known as Call Admission Control (CAC) and it is an important component of radio resource management. CAC algorithms are usually implemented in eNBs and their role is to determine whether a new connection request should be accepted or rejected, depending on the available network resources.

This chapter is focused on CAC policies for handling the admission of Voice over IP (VoIP) calls in an LTE system. Particular interest is laid on the "busy hour" phenomenon, which is defined as the "uninterrupted period of 60 minutes during the day when the traffic offered is the maximum" (Weber, 1968). It is during these intervals of increased traffic that CAC mechanisms play a significant role in the system performance and stability. However, most works in the literature either implement fixed bandwidth reservation schemes or give priority to real time services once they have been admitted to the system. In this chapter, the authors present two different approaches that take into consideration the "busy hour" phenomenon, namely, a dynamic bandwidth reservation scheme and a dynamic CAC mechanism that adapts to the incoming traffic load.

The chapter is organized as follows. Section 2 outlines the related work on admission control schemes for 4G networks found in the literature. The two proposed CAC algorithms are described in Section 3 and their performance is discussed in Section 4. Finally, Section 5 is devoted to conclusions.

2. Related work

Although the admission control concept has been extensively studied in the literature, only a limited number of contributions are developed within the context of 4G networks. The objective of this section is to highlight the recent work on this field in order to provide the reader with an up to date State of the Art.

One of the first attempts towards introducing admission control in the fourth-generation cellular mobile networks has been made by Jeong et al. (Jeong et al., 2005). In their work, the authors present a CAC scheme that supports the QoS requirements of the accepted connections in IEEE 802.16e wireless systems. The objective of the proposed CAC is to maximize the utilization of the resources, considering as basic parameter the capacity estimation of the cell. Furthermore, during the admission and scheduling process, the base station distinguishes the delay-sensitive real-time (RT) from the delay-tolerant non-real-time (NRT) connections. The proposed scheme achieves to fulfil the QoS demands of the connections, but in temporary overloaded situations, only NRT class connections can be admitted, thus excluding entirely the RT traffic.

Qian et al. (Qian et al., 2009) propose a novel radio admission control scheme for multiclass services in LTE systems. The authors introduce an objective function to maximize the number of admitted users and propose a CAC algorithm that implements a service degradation scheme whenever a limitation of resources occurs. In their paper, there is a service differentiation approach, with different portions of bandwidth devoted to each traffic class. However, in the presented numerical results there is no plot that distinguishes the blocking probabilities for the different types of traffic, thus not providing any information about the actual handling of the multiclass services.

Anas et al. propose an admission control algorithm for LTE utilizing the fractional power control (FPC) formula agreed in 3GPP (Anas et al., 2008). In their work, GBR is the only considered QoS of the bearer, while each user is assumed to have a single-bearer. The main idea of their proposed algorithm is that the current resource allocation can be modified in order for the new user to be admitted without violating the power restriction for the physical uplink shared channel (PUSCH).

Lei et al. (Lei et al., 2008) introduce a resource allocation algorithm along with a connection access control scheme for LTE systems with heterogeneous services. Their proposed CAC assigns different portions of bandwidth for real-time and non-real-time connections, thus balancing the ongoing connections of different traffic classes and facilitating the support to potential handoff users. However, the results show that the cell throughput remains the same whether the proposed admission control scheme is applied or not.

In (Kwan et al., 2010) a novel predictive admission control scheme is presented. The authors propose a new cell load measurement method and mechanisms for predicting the load increase due to the acceptance of new connections. In the same content, a resource-estimated CAC algorithm is proposed in (Bae et al., 2009). Specifically, whenever a service request occurs, the resource-estimated CAC algorithm calculates the required amount of resources in order for the request to be served. This amount is determined based on the service type, the modulation and the coding scheme level of the particular user. However, the results show that the proposed CAC is beneficial only in terms of packet delay, since the average data rate and the cell utilization are decreased.

Regarding the bandwidth reservation, a downlink CAC algorithm with look-ahead calls for 3GPP LTE mobile networks is presented in (Sallabi and Shuaib, 2009). The proposed algorithm handles the advance resource reservations, providing a high probability that the advance calls will be immediately served once their session is ready to start. Nevertheless, it is hard to derive useful conclusions since there is no reference or comparison to other admission control methods.

Finally, there are various works on the scheduling phase of LTE that grant priority to the VoIP traffic service (Choi et al., 2007; Puttonen et al., 2008; Saha and Quazi, 2009). The main idea behind these contributions is that the prioritization of the voice packets takes place once the connection has been admitted in the system. This results in higher satisfaction of the VoIP users since the QoS of the voice traffic remains in high levels.

3. Admission control schemes

As mentioned in the introduction, LTE defines a class-based QoS concept, thus providing a simple but effective solution to operators in order to offer differentiation between packet services. Furthermore, recent studies have shown that the proportion of VoIP users show a continuous growth from 28% of users in 2008 (up from 20% of users in 2007) to more than 50% in 2010 (Report Study, 2009). Due to this fact, the proposed schemes focus on voice flows, giving them higher priority comparing to the other types of traffic of the standard.

The problem becomes more intense if we take under consideration the variation of daily traffic volume, where there is a peak during the "busy hour". In Fig. 2 the mean number of calls per minute to a switching centre taken as an average for periods of 15 minutes during 10 working days (Monday-Friday) is depicted (Iversen, 2010).

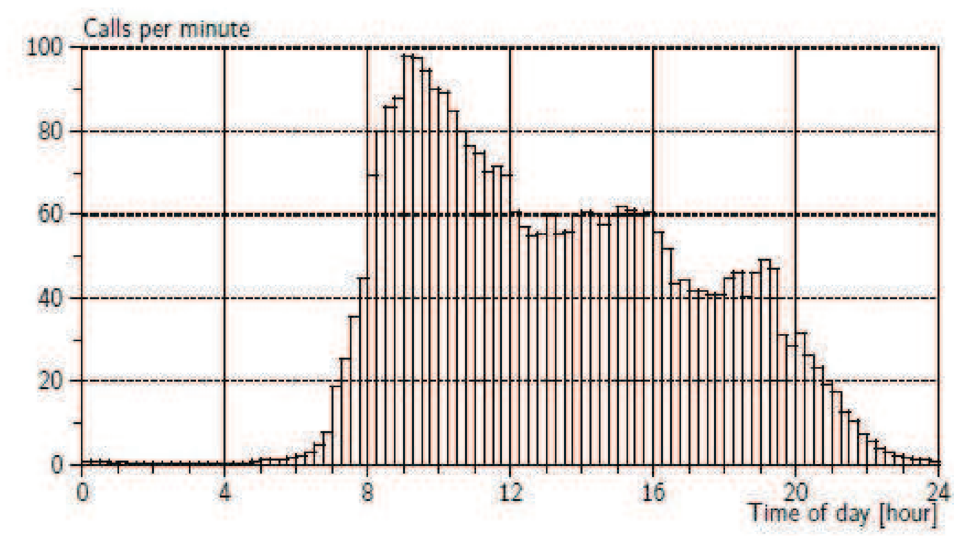

Fig. 2. Typical 24-hour traffic variation

In this section, two CAC algorithms to handle the admission of VoIP calls are presented. The target of the both schemes is to provide enhanced Grade of Service (GoS) to voice traffic flows by improving the acceptance rate of the VoIP calls. Grade of Service is defined as the probability of a call being blocked (BP) or delayed more than a specified interval. From a practical aspect it could be also defined as the probability of a user receiving a network busy signal in a telephone service and can be measured using the following equation:

$$GoS = BP = \frac{Number\_of\_lost\_calls}{Number\_of\_offered\_calls} \tag{1}$$

3.1 Bandwidth reservation-based CAC mechanism (BR CAC)

Our first proposed admission control mechanism is based on the bandwidth reservation concept and is executed under “busy hour” conditions. Under these conditions (i.e. for high arrival rate of VoIP calls), once a connection request arrives at the system, it is mapped onto the corresponding service class. Three main service classes are considered in our scheme: i) the voice GBR ii) the non-voice GBR and iii) the non-GBR traffic types. The two first classes are included in the GBR family, while the third includes the connections that do not require any Guaranteed Bit Rate. In case of voice connections, the request is accepted if the total available bandwidth (BW_T) suffices to serve the incoming connection. On the other hand, restricted bandwidth (BW_T - BW_R) is provided to the other GBR classes, as the algorithm’s aim is to prioritize VoIP calls over other types of connections. In order to deal with the connections that do not require any QoS guarantees (non-GBR), the requests are always admitted, but no bandwidth allocation is considered. The portion of the reserved bandwidth for voice traffic is dynamically changed according to the traffic intensity of the VoIP calls:

$$BW_R = \lfloor \rho_1 \times \beta \rfloor \times BW_1 \tag{2}$$

In the above expression, the traffic intensity ρ_1 is a measure of the average occupancy of the base station during a specified period of time. It is denoted as $\rho = \lambda_1 / \mu_1$, where λ_1 is the mean arrival time for VoIP connections and μ_1 represents their mean service rate (duration). Furthermore, BW_1 is the bandwidth needed for each VoIP call, while $\beta \in [0,1]$ denotes the bandwidth reservation factor.

Formula (2) implies that traffic intensity has an impact on the blocking probabilities of both voice and non-voice connections. It makes sense that applying this bandwidth reservation scheme, the blocking probability for the VoIP connections is decreased, since a portion of bandwidth is exclusively dedicated to this service type. On the contrary, the available bandwidth for the connections of the other service types is decreased and consequently the blocking probability for the specific types increases.

In bandwidth reservation schemes, one of the main difficulties is to avoid the inefficient utilization of system resources. However, in our case, the daily traffic variation establishes the ability to predict an increase in VoIP calls, thus enabling us to tackle this problem. Therefore, our scheme outperforms classic bandwidth reservation mechanisms.

3.1.1 Analytical model

In this section an analytical model for the proposed bandwidth reservation scheme is developed, to derive the blocking probabilities for the different class types. The results are further verified by extensive simulations, presented in the following section.

In order to simplify the analysis, the non-voice connections (e.g. video, data etc) are treated as a single class type with the same characteristics (i.e. arrival rate, bandwidth demand). In this point we must clarify that this simplification takes place only in the admission control process since, after being accepted, the connections are treated according to their different priorities. Furthermore, non-GBR connections are not included in the model as they are always accepted without any QoS guarantees.

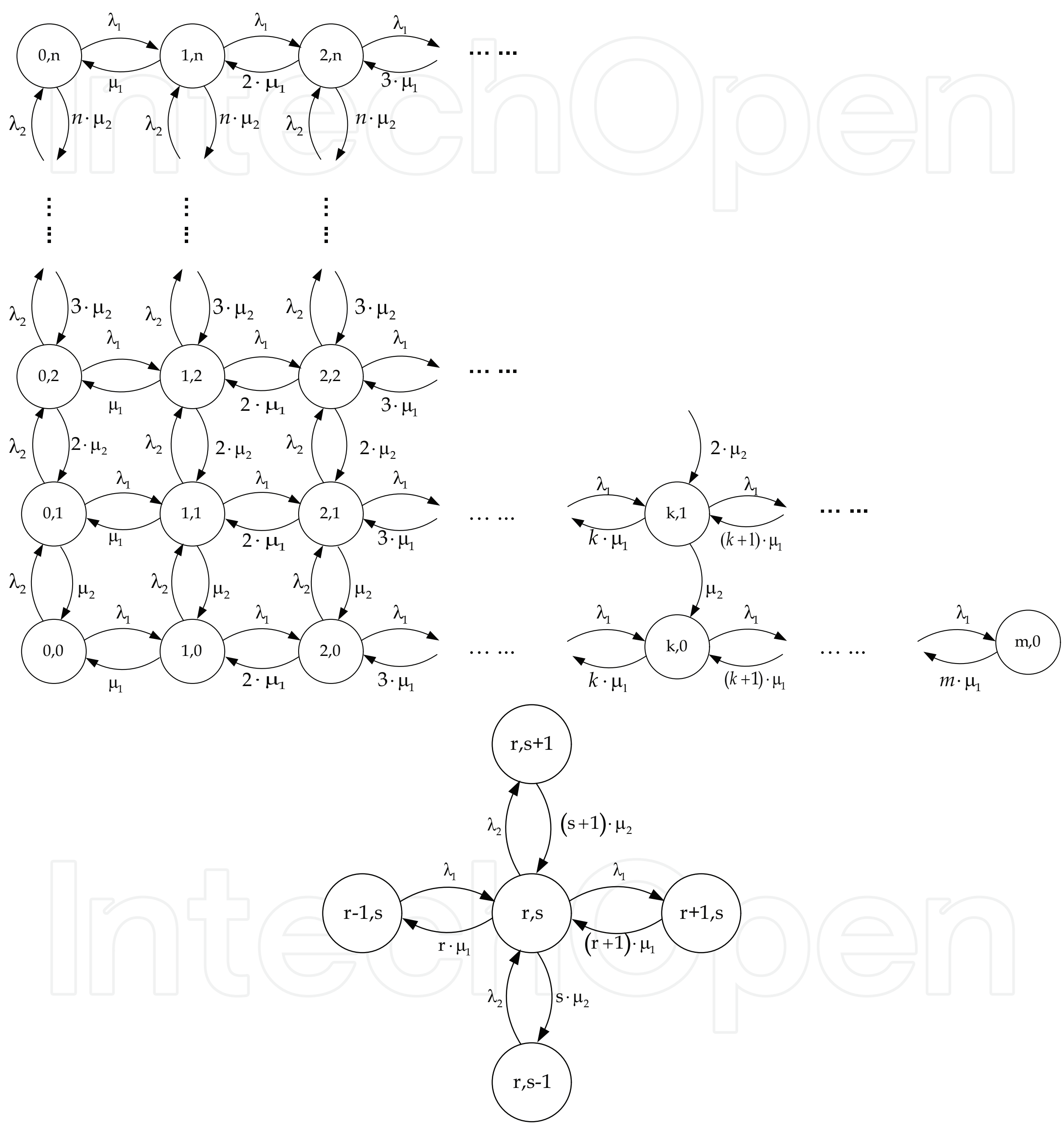

Fig. 3. The two-dimensional Markov model's state transition diagram

Thus, the 2-dimensional continuous Markov model (Fig. 3) can be used to analyze the performance of the proposed scheme. The state space of this Markov model is

$$S = \{(r,s) | 0 \le r \le m, 0 \le s \le n, r \cdot BW_1 + s \cdot BW_2 \le BW_T\}, \tag{3}$$

where $m = \left\lfloor \frac{BW_T}{BW_1} \right\rfloor$ and $n = \left\lfloor \frac{BW_T - BW_R}{BW_2} \right\rfloor$. The number of VoIP and non-VoIP connections is represented by r and s, respectively. Additionally, BW_T and BW_R represent the overall and the reserved bandwidth, while BW_1 and BW_2 represent the bandwidth that is needed in order to serve each VoIP and non-VoIP connection, respectively. We also define other parameters as follows:

λ_1	Arrival rate of VoIP connections
λ_2	Arrival rate of non-VoIP connections
$1/\mu_1$	Service time for VoIP connections
$1/\mu_2$	Service time for non-VoIP connections

The state transmission diagram of the Markov model is shown in Fig. 3.
Its steady state equation is the following:

$$\begin{aligned} & p_{r,s} \cdot \left(\lambda_1 \cdot \varphi_{r+1,s} + \lambda_2 \cdot \varphi_{r,s+1} \cdot \theta_{r,s+1} + r \cdot \mu_1 \cdot \varphi_{r-1,s} + s \cdot \mu_2 \cdot \varphi_{r,s-1} \right) = \\ & = \lambda_1 \cdot p_{r-1,s} \cdot \varphi_{r-1,s} + \lambda_2 \cdot p_{r,s-1} \cdot \varphi_{r,s-1} \cdot \theta_{r,s} + (r+1) \cdot \mu_1 \cdot p_{r+1,s} \cdot \varphi_{r+1,s} + (s+1) \cdot \mu_2 \cdot p_{r,s+1} \cdot \varphi_{r,s+1} \end{aligned} \tag{4}$$

where $p_{r,s}$ denotes the steady state probability of the system lying in the state (r,s) and $\phi_{r,s}$, $\theta_{r,s}$ denote characteristic functions:

$$\varphi_{r,s} = \begin{cases} 1, & (r,s) \in S \\ 0, & otherwise \end{cases} \tag{5}$$

$$\theta_{r,s} = \begin{cases} 1, & r \cdot BW_1 + s \cdot BW_2 \le BW_T - BW_R \\ 0, & otherwise \end{cases} \tag{6}$$

The above functions are used in order to prevent a transition into an invalid state, according to the previously defined restrictions. Furthermore, considering the normalization condition $\sum_{(r,s)\in S} p_{r,s} = 1$, the steady state probability for each possible state can be obtained.

The blocking probabilities for VoIP and non-VoIP connections are given by:

$$BP_{VoIP} = \sum_{(r+1)\cdot BW_1 + s\cdot BW_2 > BW_T} p_{r,s} \tag{7}$$

$$BP_{non-VoIP} = \sum_{r\cdot BW_1 + (s+1)\cdot BW_2 > BW_T - BW_R} p_{r,s} \tag{8}$$

3.1.2 Operational example

In order to clarify the mathematical analysis above, we provide two possible states of the system's Markov Chain. Fig. 4 depicts the exact form of the chain in each of the two cases. The first represents the state where there is no available bandwidth for non-voice connections, hence not permitting the transition from *s* to *s+1*. On the other hand, the second represents an equivalent situation along with the assumption that only voice connections are served in the system (*s*=0), thus not allowing the transition from *s* to *s-1* and vice versa.

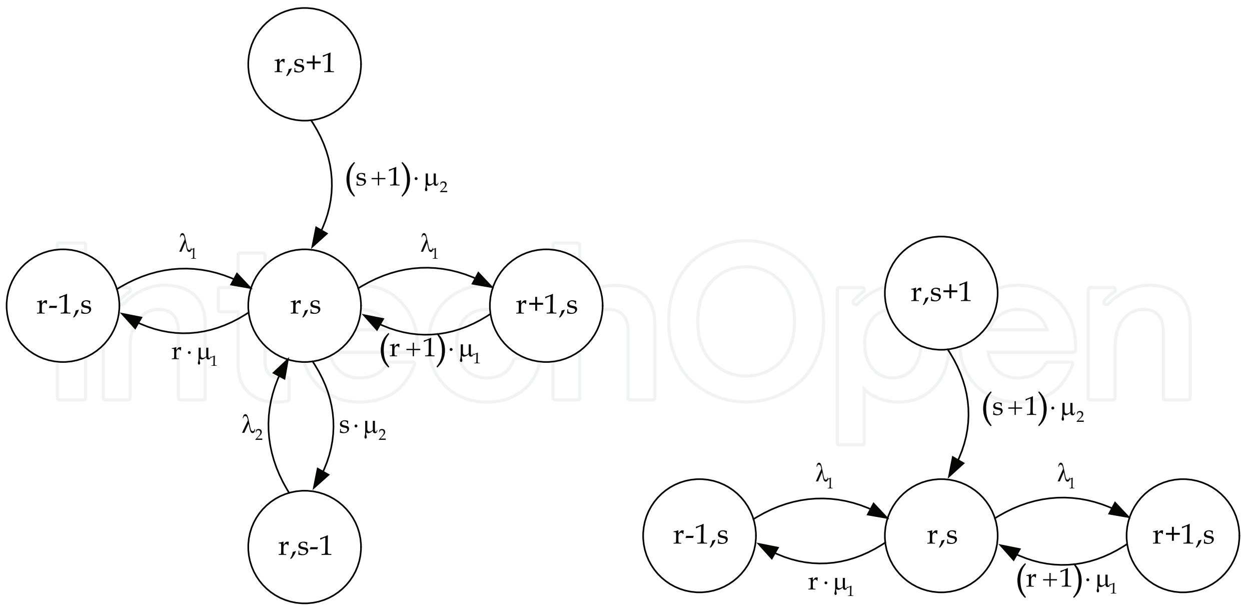

Fig. 4. Two examples of possible states of the system

First case: We assume that the system lies in the state (r, s), subject to the following constraints:

$$\{(r,s),(r+1,s),(r,s+1),(r-1,s),(r,s-1)\} \in S \tag{9}$$

$$r\cdot BW_1 + (s+1)\cdot BW_2 > BW_T - BW_R \tag{10}$$

$$r\cdot BW_1 + s\cdot BW_2 < BW_T - BW_R \tag{11}$$

Under these assumptions and using the definitions of $\phi_{r,s}$ and $\theta_{r,s}$, we derive the steady state equation for the specific case:

$$p_{r,s}\cdot(\lambda_1 + r\cdot\mu_1 + s\cdot\mu_2) = \lambda_1\cdot p_{r-1,s} + \lambda_2\cdot p_{r,s-1} + (r+1)\cdot\mu_1\cdot p_{r+1,s} + (s+1)\cdot\mu_2\cdot p_{r,s+1} \tag{12}$$

Second case: In this case we assume that the system lies in the state (r,s), subject to the following constraints:

$$\{(r,s),(r+1,s),(r,s+1),(r-1,s)\} \in S \tag{13}$$

$$(r,s-1) \notin S \tag{14}$$

$$r\cdot BW_1 + (s+1)\cdot BW_2 > BW_T - BW_R \tag{15}$$

$$r\cdot BW_1 + s\cdot BW_2 = BW_T - BW_R \tag{16}$$

Considering again the definitions of $\phi_{r,s}$ and $\theta_{r,s}$, we derive the respective steady state equation for this case, that is:

$$p_{r,s}\cdot(\lambda_1 + r\cdot\mu_1) = \lambda_1\cdot p_{r-1,s} + (r+1)\cdot\mu_1\cdot p_{r+1,s} + (s+1)\cdot\mu_2\cdot p_{r,s+1} \tag{17}$$

3.2 Dynamic call admission control algorithm (DCAC)

In the same context, we propose a second CAC algorithm that gives priority to the VoIP calls during the "busy hour". In this scheme, unlike the previous one, no bandwidth reservation takes place, while there is an effort towards a fairer handling of all connections.
According to this CAC scheme, the eNB accepts all the VoIP flows if the available bandwidth suffices in order for the calls to be served. In the case of non-VoIP flows there is an outage probability that depends both on the arrival rate of VoIP requests as well as on the available bandwidth. The requests of non-GBR connections are always admitted, but no bandwidth allocation is considered, since non-GBR flows do not need any QoS guarantees.
The proposed algorithm has two main parameters: the arrival rate of VoIP requests and the available bandwidth of the system. The outage probability for the non-VoIP connections increases either when the arrival rate of the VoIP calls grows or when the available bandwidth decreases. The capacity required in order to serve all the upstream connections can be approximated with the following expression:

$$C_{need} = \sum_{i=1,2} \rho_i \times BW_i \tag{18}$$

All the parameters in the above expression have been already defined. However, it should be stressed that the index i corresponds to different service types and can take values 1 and 2 for VoIP and non-VoIP traffic, respectively.
In case that the system bandwidth suffices to serve the flows of all service types, the outage probability is equal to zero. Due to this fact, the proposed admission control has the same output as classic admission control schemes under light traffic conditions in the network. On the contrary, in overloaded environments where the bandwidth is not sufficient for all connections, an admission control algorithm is required in order to provide different levels of priority to the various connections.
Let us consider the arrival rate of the VoIP requests, defined as λ_1. If this rate is higher than a specific threshold there will be an outage probability for the requests of the other GBR service types. This threshold is defined by the administrator/operator of the network, by considering the network parameters, e.g. the arrival rate of VoIP calls during "busy hour". The value of the outage probability fluctuates between $Pout_{\min}$ and $Pout_{\max}$, depending on the available system bandwidth. In the extreme case that we have no available bandwidth, the overall outage probability becomes $Pout_{\max}$. Adversely, when the total bandwidth of the system is available and no connections are being served, i.e., $BW_{\text{available}}/BW_T = 1$, the outage probability becomes $Pout_{\min}$, since there is enough bandwidth in order for the connections of all types to be served. These borderline values are selected by the system's operator according to each traffic class' desired level of priority. On the other hand, whenever the arrival rate of VoIP connections is smaller than this arrival rate threshold, we assume that we are out of "busy hour" and, therefore, the outage probability equals zero.
The flowchart in Fig. 5 depicts the connection acceptance/rejection procedure in the proposed Dynamic Connection Admission Control (DCAC) algorithm. The basic process of the connection request flow has been described above. In the last part of the algorithm, there is an estimation of the available bandwidth ratio in order to derive the exact value of the outage probability (the higher the ratio, the lower the probability). In particular, the $Pout_{\min}$ is a system parameter, designated by the operator, which determines the desirable level of priority to be assigned to the voice calls. By adding this value to the normalized bandwidth ratio, the outage probability for the specific connection is derived.

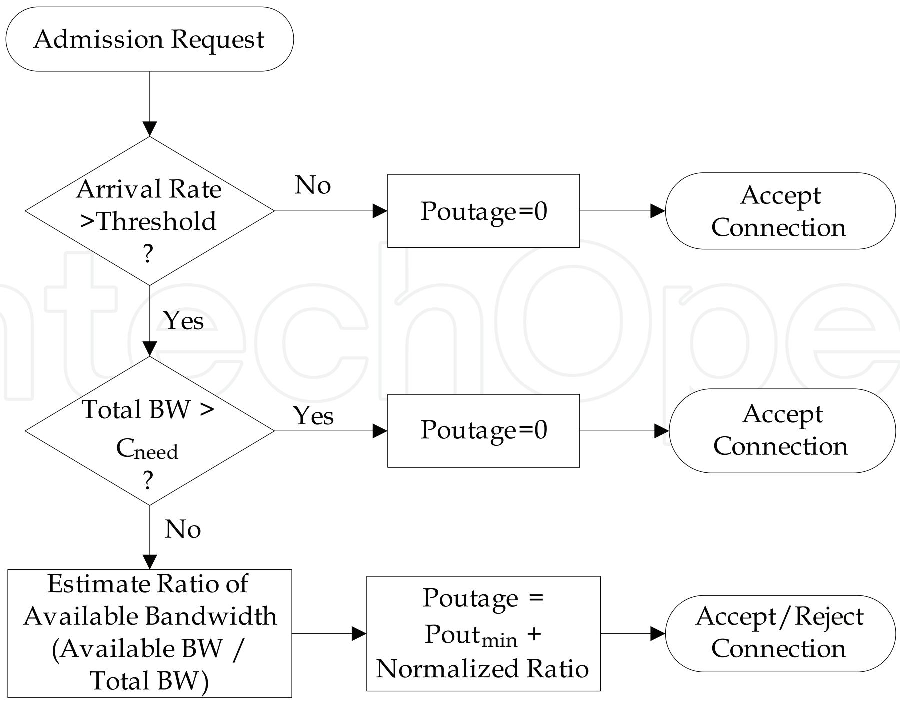

Fig. 5. Dynamic connection admission control (flowchart)

4. Performance evaluation

In order to evaluate the performance of the proposed CAC schemes and verify the validity of the analytical formulation, corresponding event-driven C++ simulators that execute the rules of the algorithms have been developed. In this section, the simulation set up is described, followed by a discussion of the obtained results.

4.1 Simulation scenario

Based on the physical capabilities of the LTE technology, we assume that the overall bandwidth for the uplink traffic is 4 Mb/s. Assuming that the non-VoIP traffic consists mainly of audio and video data, an average bandwidth of 128 kb/s for each connection is considered (Koenen, 2000). The codec chosen to generate VoIP traffic is the G.711, resulting to a constant bit rate of 64 kb/s. Each result was produced by running the simulation 100 times using different seeds, while we simulate 3600 seconds of real time in order to be in accordance with the definition of "busy hour".

In order to evaluate the efficiency of the proposed algorithms, a research on the state-of-the-art admission control mechanisms for the LTE standard has been conducted. Several schemes in the literature accept a new connection when the following condition is satisfied:

$$C_{reserved} + TR_i^{service} \leq C_{total} \tag{19}$$

where $C_{reserved}$ represents the capacity reserved by the already admitted connections in the system, $TR_i^{service}$ denotes the traffic rate that should be guaranteed to the new connection i of service type *service* and C_{total} is the total available capacity.

We refer to these methods as capacity-based (CB) algorithms in order to distinguish from our proposed algorithms which are either based on the bandwidth reservation (BR) concept or follow a dynamic approach (DCAC). In order to study the performance of our mechanisms we have carried out simulation tests by varying the VoIP requests arrival rate, thus providing a large range of voice traffic that fluctuates between 15 and 240 connections/min. However, it should be clarified that the rate request of the voice connections remains constant during the busy hour. The system parameters that are presented in Table 1, define that the arrival rate of all connections follows a Poisson distribution, while the mean service time for the connections is exponentially distributed.

Parameter	Value
Bandwidth	4 Mb/s
λ_2	Poisson (1 connection/s)
$1/\mu_1$	Exponential (mean 50 s)
$1/\mu_2$	Exponential (mean 50 s)
BW_1	64 kb/s (G.711)
BW_2	128 kb/s
Threshold	0.2 calls/s
β (BR)	1/3
$Pout_{min}$ (DCAC)	0.6
$Pout_{max}$ (DCAC)	0.85

Table 1. System parameters

Under these assumptions and considering λ_1 = 1 connection/s, the system can serve about 98% of the VoIP calls if all the requests of the other classes are rejected, which means that the network is overloaded. Furthermore, in the specific case we use a single admission control based on bandwidth availability (CB) where all the requests are accepted if there is enough bandwidth to serve them, regardless of the class that they belong to, the system serves about 57% of the VoIP flows and 34% of the other flows.

Finally, before proceeding to the simulation results, let us recall that the aim of the proposed schemes is to serve more voice traffic by reducing the GoS, and consequently the blocking probability, of VoIP calls.

4.2 Performance results

Simulation results are compared to those obtained with the mathematical model presented in section 3.1.1. First, it can be observed that the simulation results verify the mathematical analysis, with the difference varying in a range of less than 2% (Fig. 6). Comparing the first proposed admission control to traditional schemes for different values of arrival rates for the VoIP connections, we observe that the BR CAC outperforms single admission control methods in terms of GoS, without any deterioration in the overall system performance. Fig. 6 depicts the GoS among various arrival rates of VoIP calls. It is observed that, using our proposed CAC, a better system performance in terms of voice communication is achieved, as there is a significant enhancement in GoS (10-40%) of VoIP traffic.

On the other hand, the GoS of the other types of connections is increased as expected, but examining the system considering the total number of connection requests (both VoIP and non-VoIP) we achieve a more efficient utilization of system resources as we observe an enhancement in the total GoS ratio for high arrival rates of VoIP connections (i.e. rates greater than 1 connection/s).

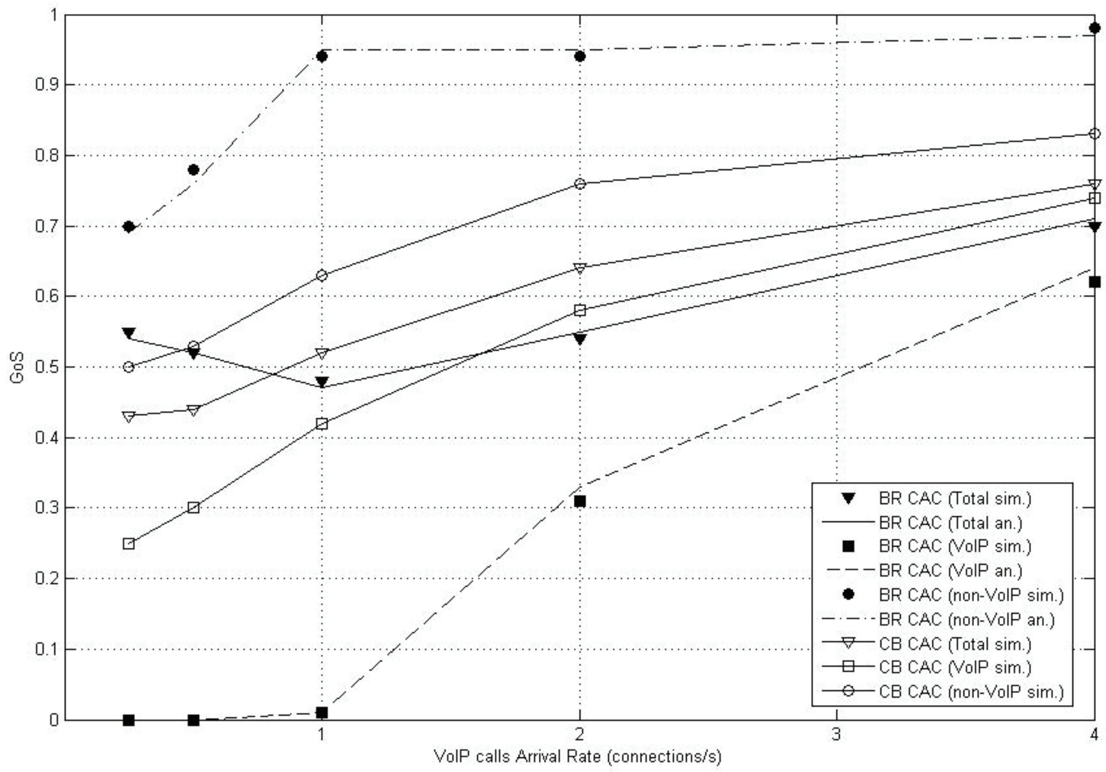

Fig. 6. GoS vs. VoIP Calls Arrival Rate (proposed Bandwidth Reservation (BR) CAC vs. Capacity-based (CB) CAC including analytical results)

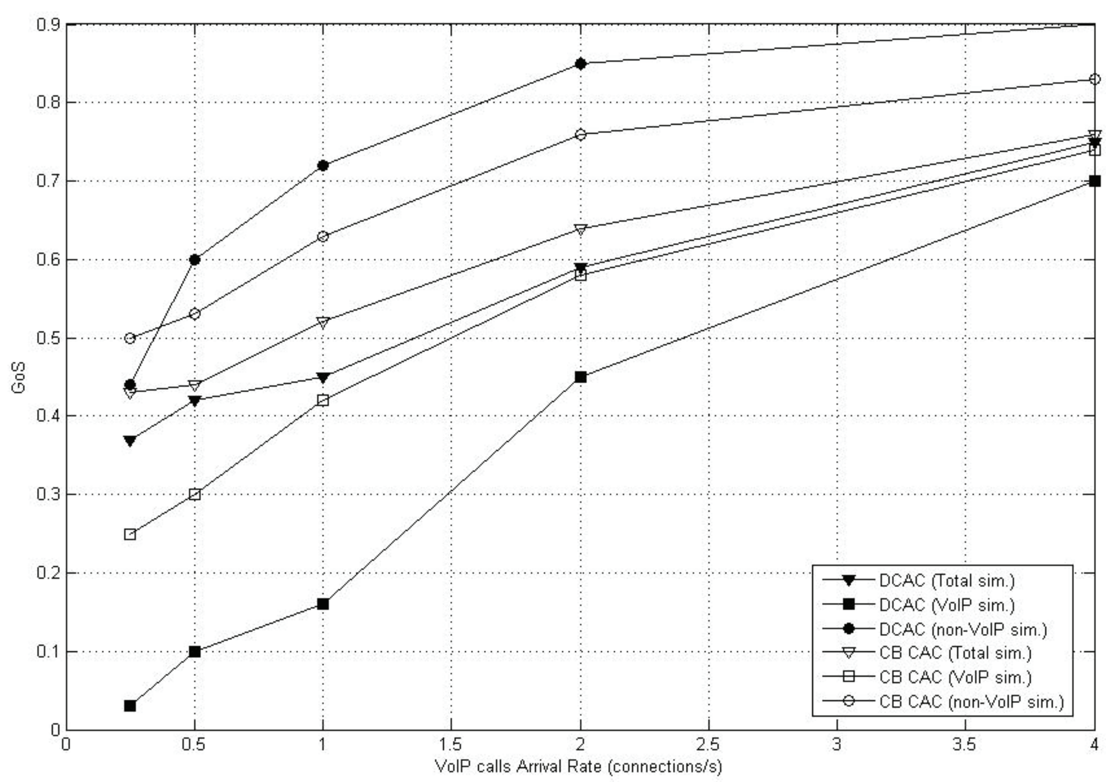

Fig. 7. GoS vs. VoIP Calls Arrival Rate (proposed DCAC vs. Capacity-based (CB) CAC)

The simulation results of the proposed Dynamic Call Admission Control (DCAC) algorithm comparing to the Capacity-based (CB) algorithm are presented in Fig. 7. This algorithm not only improves the voice traffic service, but also enhances the overall system performance. However, in this case the level of prioritization of the VoIP calls over the other type of traffic is lower compared to the bandwidth reservation scenario, thus resulting in a fairer distribution of the system resources.
Furthermore, it is interesting to observe that even for the lower arrival rates of VoIP calls (i.e. 0.25 and 0.5 calls/s) the DCAC handles efficiently the system’s bandwidth, due to its flexibility, while the BR scheme fails to overcome the Capacity-based algorithm. The comparison between the two proposed schemes is given in Fig. 8. In this figure, even if there is no further information provided, it can be clearly seen how the two proposed schemes deal with the different types of traffic, as well as their overall performance. An interesting observation is that, in this particular scenario, the curves for the total GoS for the two schemes cross when the arrival rate is approximately 1.3 connections/s. Below this threshold (i.e. for relatively low traffic conditions) the DCAC outperforms the proposed BR scheme, while above this threshold (i.e. for relatively high traffic conditions) the BR scheme handles the total connections in a more efficient way.
The system’s bandwidth is a main parameter of the DCAC. In Fig. 9 the provided Grade of Service for various values of bandwidth is plotted. As far as networks with restricted bandwidth capabilities are considered, we observe that our proposed dynamic admission control algorithm outperforms single methods, as it improves the GoS of both VoIP calls (11-27%) and of the total number of connections (8-10%) as well.

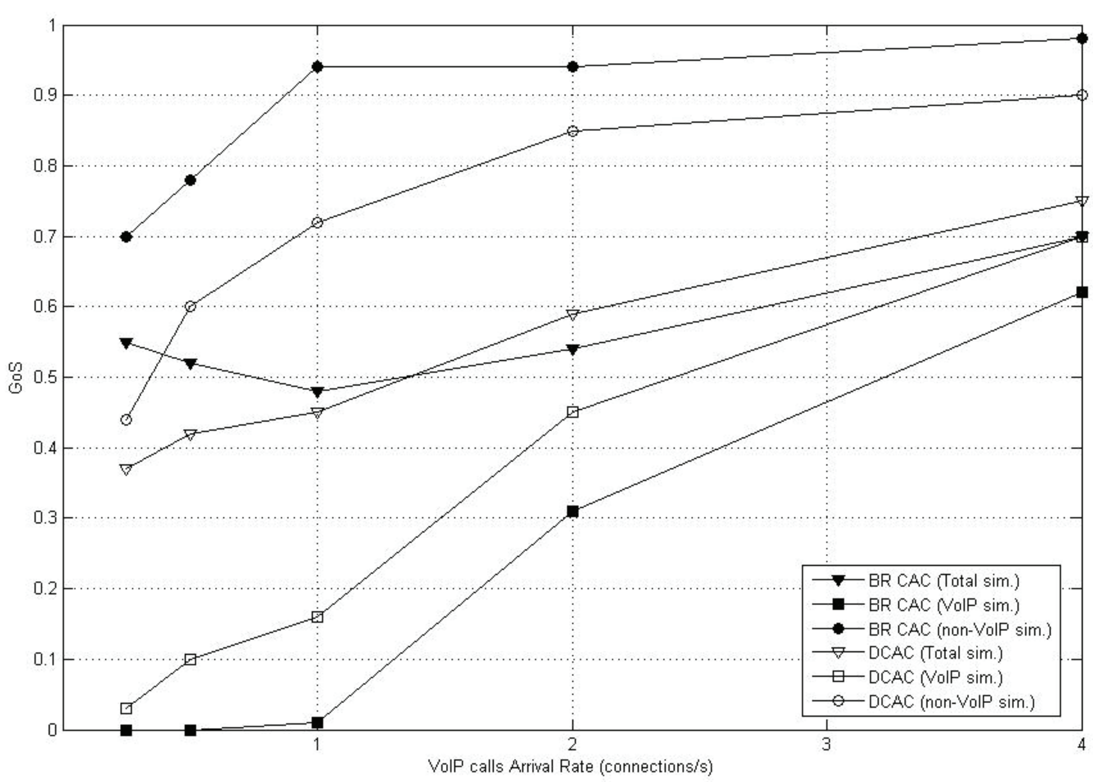

Fig. 8. GoS vs. VoIP Calls Arrival Rate (proposed Bandwidth Reservation (BR) CAC vs. proposed DCAC)

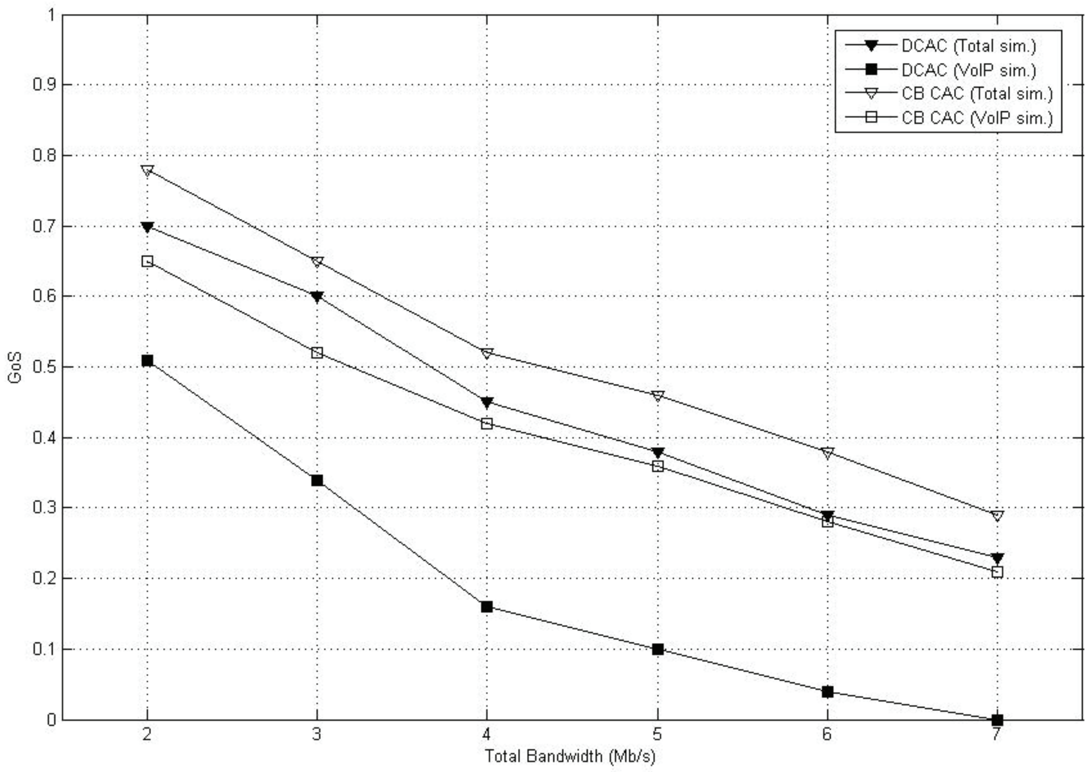

Fig. 9. GoS vs. Total System's Bandwidth (proposed DCAC vs. Capacity-based (CB) CAC)

5. Conclusion

In this chapter, two new admission control schemes for the LTE architecture have been presented. The first mechanism (BR CAC) is based on bandwidth reservation concept, while the second (DCAC) reacts dynamically, depending on the available system's bandwidth. Compared to simple, Capacity-based (CB) admission control methods for 4G networks, the proposed solutions improve the Grade of Service of the voice traffic, without deteriorating the total system performance. The main idea of the proposed schemes is that the base station serves more VoIP calls by considering the "busy hour" phenomenon. Finally, although both the proposed algorithms have been designed with LTE infrastructure in mind, the flexibility of the schemes enables their adaptation to other similar technologies such as IEEE 802.16 (WiMAX).

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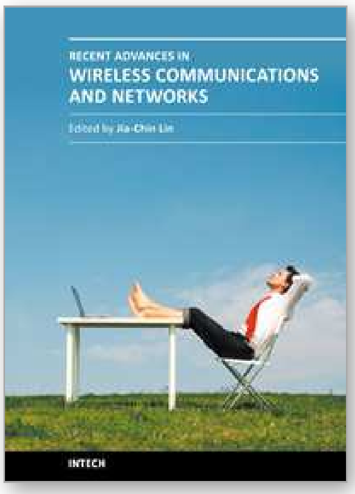

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