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Quality of service management efficient scheme for the universal mobile telecommunications system

Esquema eficiente de administración de la calidad de servicio para el sistema de telecomunicaciones móviles universales

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Abstract

This research work proposes a new Radio Resource Management (RRM) scheme in order to accomplish the Quality of Service (QoS) management for the Universal Mobile Telecommunications System (UMTS). The solution is based on UMTS standardization and a performance evaluation is presented to demonstrate its efficiency.

Keywords: *Universal Mobile Telecommunications System (UMTS), Radio Resource Management (RRM), Quality of Service (QoS), Management Scheme (3G), service class, Connection Admission Controller (CAC), Traffic Classifier (TC), Radio Resource Scheduler (RRS), Traffic Dispatcher (TD), 3GPP, simulation, efficiency and congestion control.*

Resumen

Este trabajo de investigación propone un nuevo esquema Administrador de Recursos de Radio (RRM), para llevar a cabo la administración de la Calidad de Servicio (QoS) en el Sistema de Telecomunicaciones Móviles Universales (UMTS). El planteamiento de solución que se presenta está basado en la estandarización de UMTS y se presenta una evaluación de desempeño para demostrar su eficiencia.

Descriptores: Sistema de Telecomunicaciones Móviles Universales (UMTS), Administración de Recursos de Radio (RRM), Calidad de Servicio (QoS), Esquema de Administración (3G), clase de servicio, Controlador de Admisión de Conexión (CAC), Clasificador de Tráfico (TC), Planificador de Recursos de Radio (RRS), Despachador de Tráfico (TD), 3GPP, simulación, eficiencia y control de la congestión.

Introduction

While the Second Generation (2G) cellular systems are capable of offering internet access, they do it in circuit mode, which represents enormous limitations not only in terms of the bit rate employed but also in the resource use efficiency, since this type of applications are known for generating information in bursts, which means that during a considerable time interval, the circuit is not used. On one side, these limitations result in a reduced capacity to offer this type of services, while on the

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other hand, the connection cost for the users is higher than the cost offered by a fixed network.The new broadband communications networks should be capable of offering all kinds of multimedia services. One of the most important requirements to offer this type of services is that the network guarantees, at the moment of establishing a connection, certain parameters of QoS that should be kept during the whole connection. In complex topology networks, like the ones used for mobile communications, it can be difficult to evaluate the quality parameters of the traffic in order to guarantee, for example: low delay, low jitter and low packet loss.

Particularly, the radio resources management is a function of special importance in the design of the Third Generation mobile communications systems (3G). In this sense, there is the need of designing, studying and analyzing the distinct service disciplines that each base station of the network must have, while being able to guarantee that the communication link can comply with the established QoS criteria. This aspect will be essential to be able to correctly design admission and congestion control strategies over the network, which will allow to maintain acceptably high values in the overall system´s efficiency.

The UMTS system

Through a new terminal, the new mobile technology will provide not only the voice communications, but also the exchange of data and images. Basically, the 3G services combine the high data rate mobile access with the services based on the Internet Protocol (IP) [1]. The 3G mobile communication systems focus on offering a greater range of data rates than the present mobile communications systems. For the deployment of an UMTS network, different operation environments or scenarios are considered, associated to different coverage environments and types of mobility for the terminals. The target bit rates being offered to the users depend on the environment, as shown in the table 1 [2, 3]. It's a fact that with greater area coverage and faster mobile speed, the target bit rate becomes lower.

In this sense, figure 1 shows the different scenarios considered in UMTS (Huidoro) and (ETSI,1998), being the global coverage one of the main features in this new system.

QoS ASPECTS

Bearer services architecture

The services support in UMTS is based on the hierarchical bearer services architecture defined in [4] and (Koodli et al., 2001). A bearer service is defined as the medium through which the information is transmitted. This architecture considers the decomposition in layers of the end-to-end service offered to the user, taking into account the different segments that are involved. Also this architecture is recurrent, so that the bearer services inside a layer are supported by the services offered by the lower layers. This architecture is shown in figure 2.

From figure 2: Terminal Equipment (TE), Mobile Terminal (MT), UMTS Terrestrial Radio Access Network (UTRAN), Core Network (CN), Interface be-

Table 1. Service targets in UMTS

Type of cell	Environment	Speed of mobiles	Target bit rate
Macrocell	rural outdoor	High (up to 500 km/hr)	144 kbps
Macrocell	urban/suburban outdoor	Medium (up to 120 km/hr)	384 kbps
Macrocell	low range Indoor/outdoor	Low (up to 10 km/hr)	2048 kbps

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tween UTRAN and CN (Iu), Frequency Division Duplex (FDD), Time Division Duplex (TDD) and Radio Access Bearer (RAB).

Service classes in UMTS

From the point of view of the QoS requirements and fundamentally considering the delay tolerance criteria, there have been four service classes in UMTS, from which the most prominent features are mentioned to continuation and are also more extensively detailed in [4] and (García, 2002).

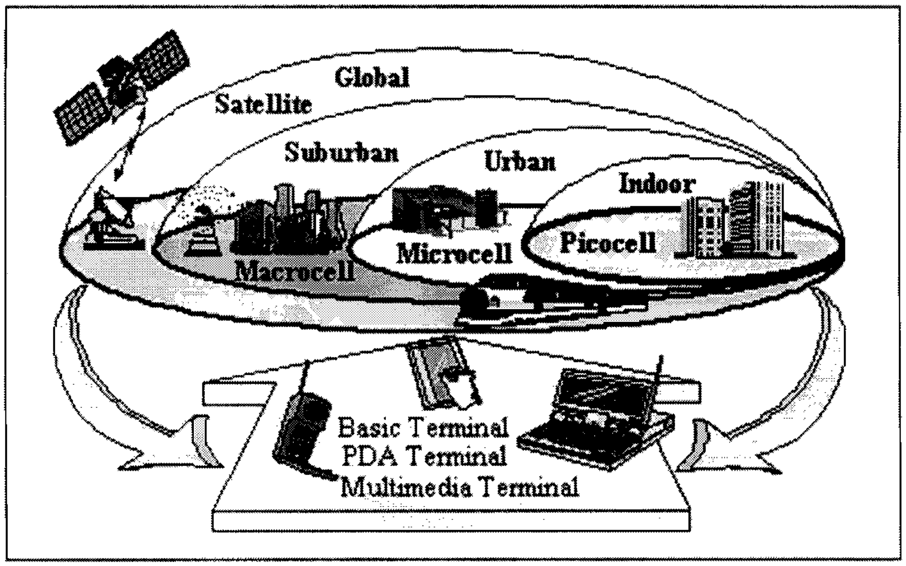

Figure 1. Scenario for UMTS [3]

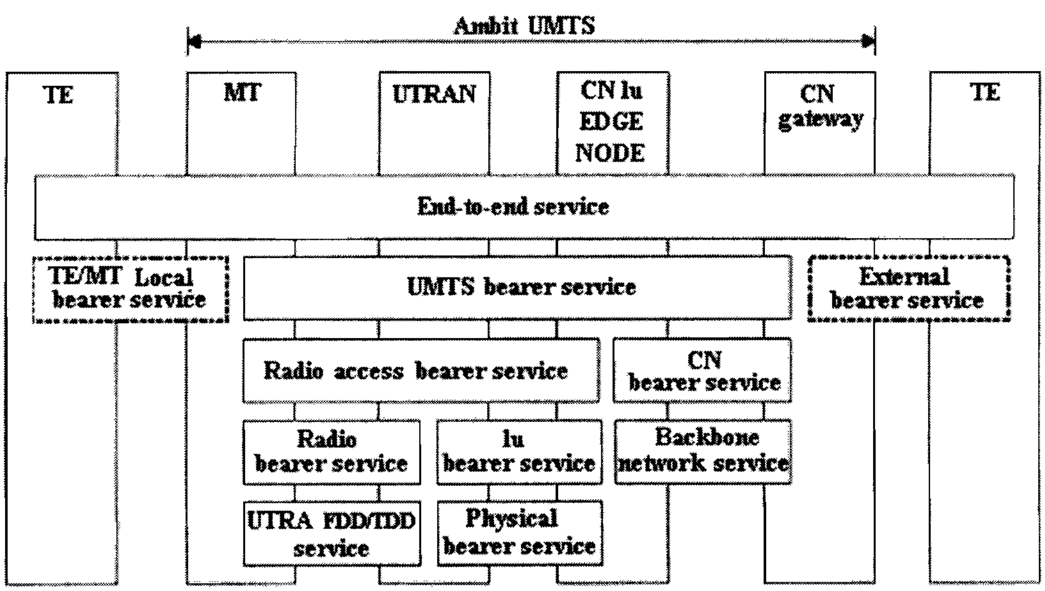

Figure 2. Quality of Service architecture for UMTS [7]

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Conversational class

The transmissions of this type are identified for being practically symmetrical and for requiring very small end to end transmission delays. The most used application of the conversational class is the voice service.

Streaming class

In this class, multimedia information is transferred in unidirectional way so it can be processed like a stable flow of data. Basically, this type of application considers the transmission of video and audio sequences in real time.

Interactive class

They are services that in general present a strong asymmetry, since while the user of an end point is only sending small commands or requests, these ones unleash a much larger information download. Examples of applications under this category are: web navigation and database queries, as well as remote access to computers.

Background class

The e-mail, the short message sending and, the database information download or the remote measurements reading, are typical examples of applications of this service class. The data transmission delay can be in the order of seconds, tens of seconds or even minutes. Although the delay is not a restriction in these cases, the integrity of the data is an indispensable requirement for these connections.

In general, the first two classes consider the so called real time services, while the last two represent the services which are not in real time.

QoS Attributes

In [2], some QoS requirement values are given as examples of final applications. Nevertheless, these requirements should be someway translated to values to be imposed in the different sections of the UMTS environment, since as it was previously mentioned, in each section there is a corresponding bearer with different characteristics. In this sense, the definition of an attribute group can be found in [4], the definition of a set of attributes (parameters) can be found, as well as the margin of values that these can take for the UMTS bearer services and of radio access. The parameters (Dixit, 2001) that have most impact in the QoS are:

– Residual bit error rate: indicates the quantity of bits erroneous inside the Service Data Units (SDUs) delivered, due to not detected errors.

– Erroneous SDU rate: is the fraction of SDUs which are lost or detected as erroneous.

– Transfer delay: the SDU's transfer delay is the time that elapses since its transfer request at the point of access, until it is received in the other end. The Third Generation Partnership Project (3GPP) specifies the maximum values for the 95% of the delay distribution of the SDUs delivered. This attribute is defined only for the service classes in real time.

Proposed scheme

The proposed scheme for the QoS management is shown in figure 3. Although the scheme was selected from diagrams found in the state of the art, the solution, the procedures and the criteria that are implemented in this scheme, correspond to a new approach of solution (Zaleta, 2004). The most important criteria in the selection of the scheme are:

– The decision to utilize the RRM scheme is supported mainly in the importance that the radio resources management has for the access technology used in the new wideband networks.

– Another reason why the RRM was selected in this work is because an important radio

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resource project was found, in which a great amount of information related to the topic is found. This work is called Advanced Radio Resources Management for Wireless Services (ARROWS) project developed in the Technical University of Catalunia (Arrows, 2001-2002).

– Also, in the ARROWS project, some strategies for the QoS management that helped to give solution to RRM scheme, since at the beginning the foundations to carry out the evaluation of the scheme were insufficient.

Therefore, the solution proposal is the following:

– To carry out the strategies implementation for each element that conform the RRM. These strategies are given by solution algorithms.

–· With an IP packets generator as source of traffic, to obtain four files (each one representing one service class), which will serve like data packets source for the simulation of the scheme (Castañeda, 2002) in MATLAB version 6.0.

– Evaluate the efficiency of the RRM considering the three following clauses:

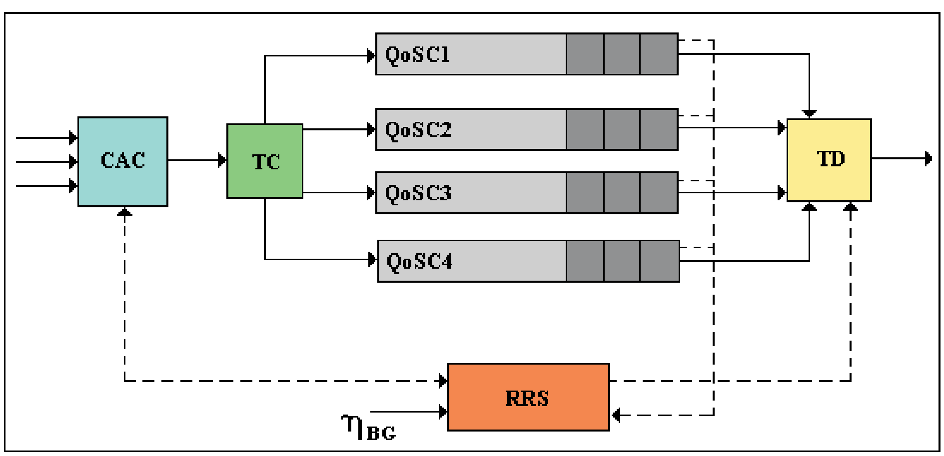
a) Maintain the RRM under stable conditions (this is, out of congestion the majority of the time).

b) Have the lowest packet loss rate possible, in function of the maximum delay permitted parameter, which will depend on the logic implemented in the Traffic Dispatcher (TD).

c) Accept the highest quantity of users.

Connection Admission Controller (CAC)

The admission control strategy is considered in this element. The admission control strategy is implemented by an algorithm which determines if a connection request should be accepted or rejected in function of the interference or load that this connection adds to the existing ones. Therefore it is responsible for deciding if a new Radio Bearer (RB) can be set-up. The considered admission control makes use of the load factor and the load increment estimation that is generated in the network radio by the connection request set-up. In the case that the load factor η is estimated in statistical terms and assuming that they have k users admitted in the system, the user $k+1$ should verify



CAC = Connection Admission Controller
TC = Traffic Classifier
QoSC1 = Quality of Service/class 1
TD = Traffic Dispatcher
RRS = Radio Resource Scheduler
η_{BG} = Interference and noise measurements

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the equation (1); (Sallent *et al*., 2003) and (Arrows-D09, 2002).

$$(1+f)\sum_{i=1}^{k}\frac{1}{\dfrac{SF_i}{\nu_{i}\cdot\left(\dfrac{E_b}{N_o}\right)_i\cdot r}+1}$$

$$+(1+f)\sum_{i=1}^{k}\frac{1}{\dfrac{SF_{k+i}}{\nu_{k+1}\cdot\left(\dfrac{E_b}{N_o}\right)_{k+1}\cdot r}+1}\leq\eta_{max} \quad (1)$$

Where:

SF_i is the Spreading Factor,

f the intercellular interference,

ν_i the activity factor of the traffic source,

r the coding rate,

η_{max} the threshold of admission control and

$\left(\dfrac{E_b}{N_o}\right)$ the bit energy-to-noise density ratio.

Traffic Classifier (TC)

This element is the responsible of classifying the traffic classes that are accepted into the RRM. The traffic is divided into four different types of service (conversational, streaming, interactive and background). Once the traffic (packets) is classified, is sent through buffers to the TD. Thus the classified traffic gives us the possibility to differentiate each one of them, considering that its characteristics are not the same and to be able to give priorities in order to dispatch. The Type of Service (TOS) field (Blake *et al*.) in the header of the IP packets is utilized for this process and contains a different value for each class of service. In this work the table 2 summarizes the values and characteristics of this field.

On the other hand, the quantity of packets that will be delivered depending on transmission rate requested for each user is obtained with the equation (2).

$$\text{Quantity of packets} = \text{Transmission rate} / \text{Packets size (in bits)} \quad (2)$$

This procedure is applied for the four service classes implemented in the RRM. Therefore, the quantity of packets delivered to the mobiles will be in function of the transmission rate requested without caring of the type of service. The table 3 shows the necessary quantity of packets in the different transmission rates and packets size considered.

Table 2. TOS field values and characteristics

Service classes	Binary value	Hex value	Transmission characteristics	Priority
Conversational	0 0 1 1 0 0 0 0	0C	Minimum delay	4
Streaming	1 1 0 0 1 0 0 0	13	Maximum capacity	3
Interactive	0 1 0 0 0 0 0 0	02	Normal service	2
Background	1 0 0 0 0 1 0 0	21	Maximum reliability	1

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Table 3. Packets for considered transmission rates

Transmission rates	Packets size		
	256 bytes	512 bytes	1024 bytes
64 kbps	31 packets	16 packets	8 packets
144 kbps	70 packets	35 packets	18 packets
384 kbps	188 packets	94 packets	47 packets
2048 kbps	1000 packets	500 packets	250 packets

Radio Resource Scheduler (RRS)

The congestion control strategy is considered in this element. The congestion control algorithm includes the following (Sallent *et al*., 2003):

1. - Congestion detection: some criteria should be established to decide if the network is in congestion or not. The criterion to detect if the system has entered a situation of congestion is when the load factor is over a certain threshold ($\eta \geq \eta_{CD}$) during a specific time period ΔT_{CD}.

2. - Congestion resolution: certain actions should be carried out to recover the stability when it is assumed that a congestion exists. The congestion resolution algorithm takes certain measures in order to try to solve these situations. Multiple possibilities exist at the moment of carrying out a resolution for the congestion, but in general, three steps can be differentiated: prioritization, load reduction and load verification.

a) Prioritization: in this case, traffic prioritization is considered for the packets dispatch but not exactly to reduce the congestion, although it helps to the stability and mainly avoids packets to be discarded by exceeding its maximum permitted delay.

b) Load reduction: It is carried out by means of not accepting any incoming connection until the load level achieves certain permitted value. This being, the RRS will indicate the CAC not to accept any calls during a short period of time because it is congested.

c) Load verification: after the step (b), the verification of the load consists on checking the four buffers condition (one for each class of service) and to activate or deactivate the con -gestion control. If the congestion persists, it is necessary to return to (b). It is considered that the congestion has been solved if the load factor is lower than certain threshold ($\eta \geq \eta_{CR}$) during a specific time interval ΔT_{CR}.

3. – Congestion recovery: a step for congestion recovery results necessary. This step consists of accepting calls again once the congestion has been solved: at this time, the RRS will indicate to the CAC that it can accept more calls.

On the other hand, the RRS will also be responsible for monitoring the wireless channel status in function of η_{BG} (background noise that indicates the conditions of the wireless channel). This will serve so that the RRS reports to the TD about the channel's conditions allowing it to dispatch the packets contained in the buffers. If the wireless channel's conditions are adequate, the TD will dispatch the packets at the maximum dispatch or transmission rate R_i allowed. Otherwise, the TD will reduce the transmission velocity according to the value of η_{BG}. If the channel's bad conditions persist, the alternative is to not accept calls during a "very small period of time", since not doing this would certainly cause a congestion. In this sense,

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the minimum monitored noise by the RRS is obtained from the thermal noise power general formula (Wayne, 1996) of the equation (3).

$$N_T \quad 10\log_{10}(KTB) \qquad (3)$$

Where:

N_T is the thermal noise power in (dBW),

K the Boltzmann constant ($1.38*10^{-23}$ J/°K),

T the temperature (290 °K) and

B the bandwidth (5×10^6 Hz).

From calculation, we obtain:

$$N_T = -136.98\ dBW = -106.98\ dBM,$$

being this value a reference for the maximum dispatch rate. Therefore, for a value of $_{BG}$ monitored by the RRS, there is a corresponding value of packets to be dispatched.

Traffic Dispatcher (TD)

This element considers the following steps. Firstly, the part to calculate the background noise added by a new connection and it is realized by the evaluation of the equation (4) which is obtained from (Moustafa *et al*., 2002 and 2003), and subsequently, the part to execute the packet dispatch. Thus, this element is in charge of the buffer's packets dispatch and, as it was mentioned before, the dispatch rate will be subject to the value of $_{BG}$ measured by the RRS.

$$_{BG} \quad \frac{\dfrac{G_{bi}p_i}{R_i}}{\left(\dfrac{E_b}{N_o}\right)_i} \quad W \quad \sum_{j\ i}^{M} G_{bj}P_j \qquad (4)$$

Where:

p_i is the power level,

p_j the power level as a function of the number of mobiles,

G_{bi} the link losses,

G_{bj} the link losses as a function of the number of mobiles,

R_i the transmitted bit rate,

W the bandwidth,

M the number of mobiles and

$_{BG}$ the background noise.

For the dispatch, a solution is presented in which the packet dispatch is considered under a priority scheme while the services work under stricter delay characteristics, such as the conversational service which will have the greatest priority (first to be dispatched). On the opposite, the background service will be the one with the lowest priority (last to be dispatched).

Thus, also the quantity of packets that will be dispatched for each class of service will be in function to this priority; this means that for a certain quantity of dispatched packets, the 40% of these will be of conversational type, the 30% of streaming type, the 20% of interactive type and the remaining 10% of background type. The transmission delay is the main criteria considered for this case.

Radio propagation model

The radio propagation model considered is the Manhattan described in (Moustafa, 2001). For this model, the calculation of the path losses is obtained with the equation (5).

$$P_L(R_K) \quad 142.37 \quad 29.74\log f_G \quad 50.37\log R_K \qquad (5)$$

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Where:

P_L is the path loss,

R_K the distance between mobile station and base station in $Km(0.05 < R_K < 3)$ and

f_G the central frequency in $GHz(0.9 < f_G < 2)$.

Applying the equation (5), the path losses for the three UMTS scenarios used for the calculation of η_{BG} are:

- Picocell (0.1 *Km*): $P_L = G_{bi} = 100.970\ dB$.
- Microcell (1 *Km*): $P_L = G_{bi} = 151.160\ dB$.
- Macrocell (3 *Km*): $P_L = G_{bi} = 175.192\ dB$.

Parameters to evaluate the efficiency

The presented values are obtained in their great majority from the 3GPP specifications [5] and from the ARROWS project (2001-2002). The parameters considered for the different service classes are presented in table 4, while in table 5 they are presented by scenario.

Scheme evaluation

Parameters based on the specifications

The simulations of the RRM scheme for the QoS management were carried out in MATLAB version 6.0 and the generated packets were obtained with

Table 4. Parameters by service class

Parameters	Services			
	Conversational	Streaming	Interactive	Background
f	0.6	0.6	0.6	0.6
SF_i	De 4 a 256	De 4 a 256	De 4 a 256	De 4 a 256
$\left(\frac{E_b}{N_o}\right)_i (dB)$	4.57	4.25	4.69	4.69
ν_i	0.67	0.57	0.47	0.37
r	1/3	1/3	1/3	1/3
$R_i(Kbps)$	64, 144, 384 y 2048	64, 144, 384 y 2048	64, 144, 384 y 2048	64, 144, 384 y 2048
$\eta_{CD}(\%)$	0.8	0.8	0.8	0.8
$\eta_{CR}(\%)$	0.7	0.7	0.7	0.7
η_{max}	0.6	0.6	0.6	0.6

Table 5. Parameters by scenario

Parameters	Scenario		
	Picocell	Microcell	Macrocell
$p_i(dBm)$	14	14	21
$p^{max}(dBm)$	21	21	21
$R_K(km)$	0.1	1	3
$f_G(GHz)$	1.975	1.975	1.975
$W(MHz)$	5	5	5

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the *"Ultra Network Sniffer"* Software (2002) [6]. The obtained results are based on the representation of the admission and congestion control strategies, while including the load behavior for each service class.

In what the admission control concerns, the obtained graphics represent the admission efficiency to the RRM (user's petitions against users accepted). At the congestion control, the load behaviors at the different status considered by this strategy were obtained (out of congestion, stable state and in congestion).

Finally, the load behavior for each service class shows how the strictest delay characteristic (conversational) service class is kept with low values when compared with the most delay tolerant service (background).

Next, the simulation results considering the table 6 parameters were obtained, which served as base for finding the most adequate values for a more efficient scheme; this represents the improvement to the developed QoS management scheme.

The figure 4 shows the load behavior before the congestion control strategy. As it is observed, the congestion control is being applied correctly according to the steps that it should be following with the specified load value in the buffers. What is important to consider in this graphic is the time that elapses between two states of congestion, since with the greater the time is, the better is the performance obtained. The time among congestion is defined as the time that elapses between two consecutive states in which the congestion control strategy operates. It is important to recall that the congestion control acts only when the load remains above the threshold η_{CD} a time ΔT_{CD}. This is commented because as the figure shows the congestion control is not activated at every instant that the load surpasses the η_{co} threshold.

Figure 5 shows the efficiency in the admission control. This efficiency is obtained from the users who requested to enter the RRM (petitions) scheme against that ones that were finally admitted. The restrictions to be able to access the scheme are: the parameters requested by the user (these must correspond with the transmission rate and scenario), the η_{max} threshold and at last, that there are enough locations available for storing the packets of the class of service requested. Therefore, when these restrictions have been verified and accomplished, the user can access to the RRM. It is

Table 6. Considered foundation parameters

Parameter	Value
Separation between thresholds η_{CD} and η_{CR}	10%
Level of the congestion thresholds η_{CD}	80%
Level of the out-of-congestion threshold η_{CR}	70%
Time to activate the congestion control ΔT_{CD}	3 sec.
Time to deactivate the congestion control ΔT_{CR}	1 sec.
Admission threshold η_{max}	0.6
Spreading Factor SF_i	256
Packet size	512 bytes
Time between dispatches	3 sec.
Length of each buffer	1000 localities
Simulation time	3 minutes

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considered that in the case of none of the restrictions being fulfilled, there should exist an option in which the user could access to the RRM, this is, if for example a user requests a transmission rate that cannot be supported, due to the fact that the required resources do not exist, the CAC will offer him a smaller rate and it will be a decision of the mobile user to agree or to refuse to this available resource. The sudden fall that both curves (petitions and accepted ones) suffer appears when the value of is obtained, in this moment the users decide randomly to remain connected or not. After this process, it is possible to accept more users again.

The figure 6 shows the amount of users admitted per class of service. Recalling that this is a matter of a random process, the amount of users that can be admitted to the RRM does not follow a defined pattern, this is, for a specific simulation there can be more users admitted from the conversational class than from any other class, while in a new simulation this preference can correspond to the background class of service.

Several simulations were carried out where the great majority of these presented an efficiency value over 70%. Thus, from the graphic (see figure 5) that shows the admission control it is concluded that: the values of efficiency obtained are acceptable whenever they are maintained above the 70% of efficiency. In the following figures, the load behavior for the four classes of service is shown. Figure 7 shows the load for the background service class and as it was mentioned before, the most important characteristic to consider in this service class is to maintain the data integrity without considering the transmission delay. It is because of this that the load remains in high values, which indicates that the packet delivery is slow and in small quantities. It is necessary to observe that even when it seems that in some of the graphics the maximum possible value of the load is reached, this does not happen, since that would indicate packets loss by overflow of the buffer.

Figure 8 presents the load for the interactive service class where, just as the background service class, the data integrity is more important than the transmission delay. Nonetheless, this figure shows a faster delivery of packets by presenting major load changes and lower levels.

The figure 9 shows the load for the streaming service class and this service class is not so strict in delay; nevertheless, it is necessary that the information is delivered in acceptable values. In this sense, it is possible to observe that the load level does not reach the maximum and that its changes are major.

Finally, figure 10 presents the load of the conversational service class; this class of service is the strictest in delay and therefore the one that should be delivered to the users as soon as possible. Respect to this, it can be observed that during most of the time, the load levels are the lowest of the four classes of service. This is accomplished thanks to the priority assigned to this service and also to the bigger quantity of dispatched packets.

From the previous statement one concludes that the lower the load values of the conversational service, the better QoS the users will perceive, whereas for the background service class, the most important thing is the information integrity. Besides, it highlights the importance that treating the traffic in differentiated form has, since with this, one of the peculiarities from UMTS is solved, where a great number of applications are expected across service classes with its very own characteristics.

Best performance parameters

To be able to find an improved scheme it was necessary to run several simulations trying to find the parameters that most impact in the behavior of the RRM. Therefore, the following simulations were ran:

1. Impact while changing the Spreading Factor: SF=128 (reduced by half), Admission Control

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Strategy = smaller amount of users (approximately 95).

2. Impact while changing the admission threshold: 0.95 (augmented), Admission Control Strategy = bigger amount of users (approximately 300).

3. Impact while varying the separation percentage between η_{CD} and η_{CR} thresholds: 20% (increased to double) = smaller number of times entering con- gestion (time between congestions = 48 seconds approximately).

4. Impact while changing the time between dispatches (2 and 4 seconds): dispatches every 2 seconds = congestion happens very few times (admission efficiency = 85% approximately); dispatches every 4 seconds = bigger number of times to enter in congestion (time between congestion = 22 seconds approximately).

5. Impact while varying the level of the η_{CD} and η_{CR} thresholds: 30% (superior), 20% (inferior), dispatches occurring every second = longer times between congestions (56 seconds approximately) and bigger transmission efficiencies (91% approximately).

6. Impact while changing the packet size (256 and 1024 bytes): 256 bytes (half) = no impact (time between congestions = 33 seconds approximately, 190 users approximately. And admission efficiency = 77% approximately); 1024 bytes (double) = admission efficiency increases (85% approximately).

The proposed values show themselves in table 7 (to compare them with table 6) and they are those for which a better behavior appears in the considered strategies. The proposed values are based on the analysis realized from the cases earlier exposed.

Figure 4 shows the congestion control strategy, where the load presents wider intervals between congestions as efficiency measurement for this strategy.

With regard to the admission efficiency, figure 5 shows a value of 72.28%, which is greater than the 70% which is the reference value to establish whether it complies or not with a good percentage of admitted users. In figure 6, the quantity of users by service class can be observed (more users).

Table 7. Proposed parameters for the best performance

Parameter	Value
Separation between thresholds η_{CD} and η_{CR}	20%
Level of the congestion threshold η_{CD}	80%
Level of the out-of-congestion threshold η_{CR}	70%
Time to activate the congestion control ΔT_{CD}	3 sec.
Time to deactivate the congestion control ΔT_{CR}	1 sec.
Admission threshold η_{max}	0.95
Spreading Factor SF_i	256
Packet size	512 bytes
Time between dispatches	3 sec.
Length of each buffer	1000 localities
Simulation time	3 minutes

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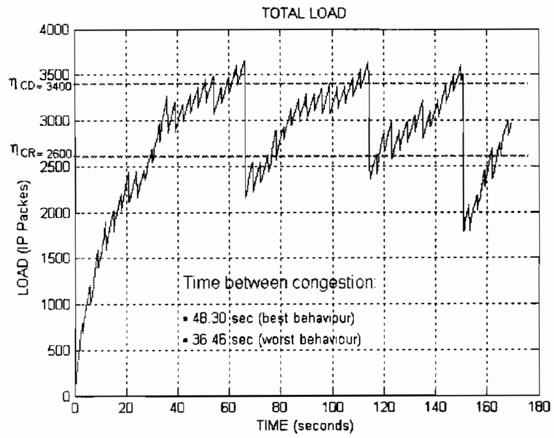

Figure 4. Congestion control strategy

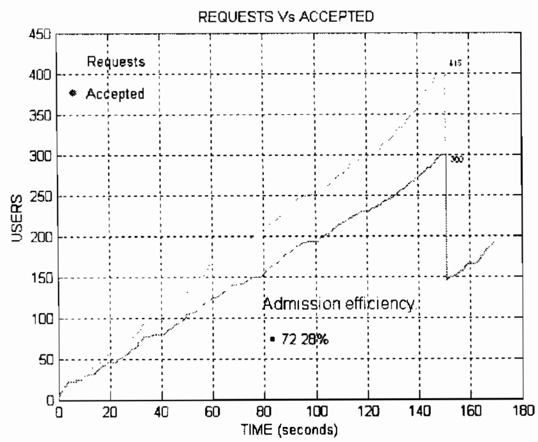

Figure 5. Admission control strategy

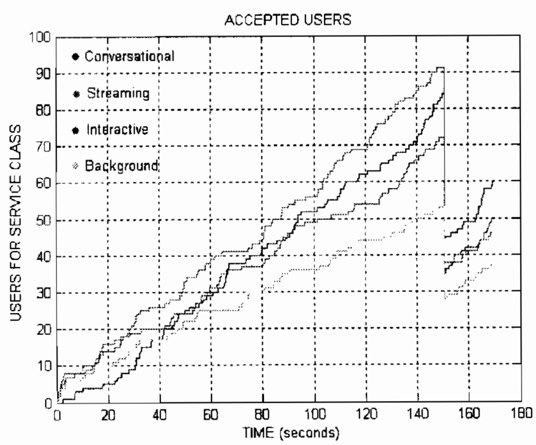

Figure 6. Accepted users for class of service

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In figure 7, the load behavior for the background service class is represented again. As it can be observed, the proposed parameters do not affect the behavior of the load, which continues to be in function to the self characteristics of each of the four service classes. In figure 8, the load behavior of the interactive service class is shown, the streaming service class is shown in figure 9 and figure 10 presents the conversational service class.

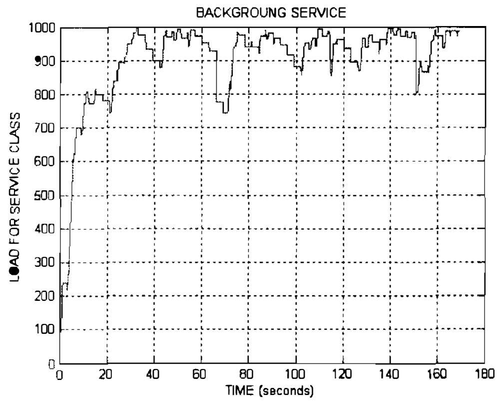

Figure 7. Background service

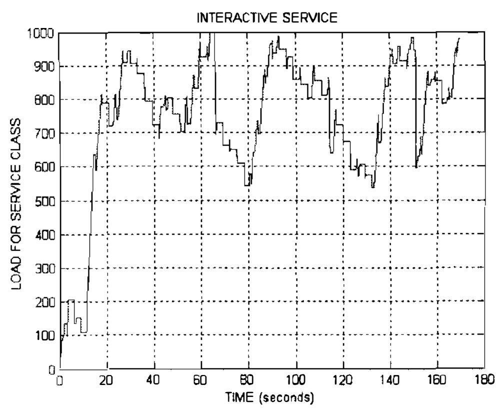

Figure 8. Interactive service

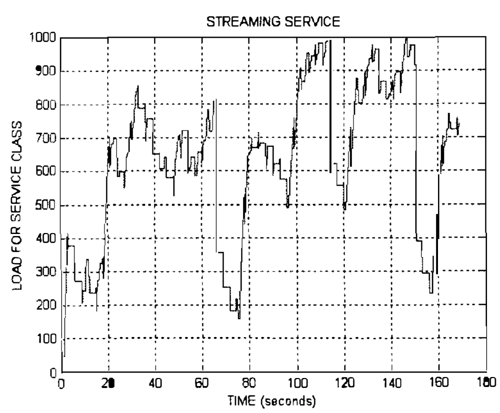

Figure 9. Streaming service

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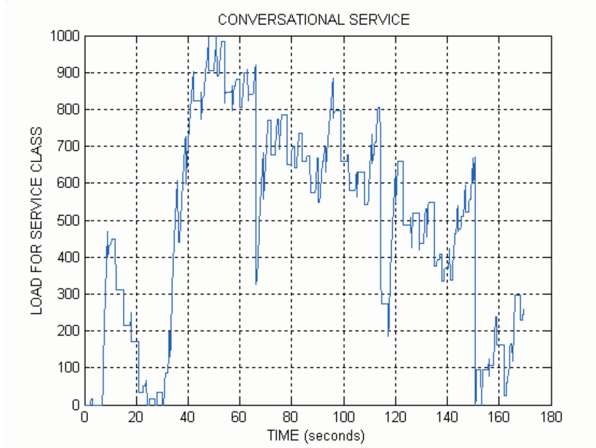

Conclusions

Nowadays, the internet and the mobile phone system receive the greatest interest inside the world of telecommunications, as shown by the great growth in the number of users that presently opt for using this type of services. In the particular case of the UMTS system, it is expected that for the year 2010, 2,000 million mobile users will exist approximately, which will be able to use this technology in any part of the world according to figures reported by the UMTS Forum [3]. For this reason, these subjects are still under research worldwide. The following is concluded from this research work:

– The proposed scheme of Radio Resource Management – RRM was implemented to manage the QoS of the four different service classes in UMTS, considering the parameters reported in the specifications from the 3GPP.

– This RRM scheme presents a solution to this crucial aspect of the new packet networks, with acceptable efficiency of user admission and with a practical congestion control algorithm.

– The advantage that the differentiation of the services has, was made clear as a form of traffic prioritization, this as an action from the Traffic Classifier - TC.

– The logic implemented at the traffic dispatcher is based on the parameter of maximum authorized delay.

– In the proposed RRM scheme, it was possible to verify the impact that the Spreading Factor has as the parameter that more reverberates in the number of users that can be admitted into the system.

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