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Universidade Técnica de Lisboa

Impact of HSDPA Implementation on UMTS Interference and Cell Coverage

Mónica Rute Cardoso Antunes

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Supervisor: Prof. Luís M. Correia

President: Prof. António Rodrigues

Members: Prof. António Topa

Eng. Carlos Caseiro

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To Rui

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Abstract

The goal of this thesis was to analyse how UMTS behaves when HSDPA is implemented. So, a theoretical model was developed calculating network capacity, for a sector of a cell, focusing on the analysis of parameters, such as number of served users, throughput, and cell radius. Two scenarios were assessed with different users' distribution. With OPNET Modeler, Release 99 was assessed by increasing the number of users, and for two different scenarios that were distinguished by the weight assigned to each service.

For the theoretical model, two analyses are done: one until the network saturates, and other for a percentage of rejected users of 5%. For the latter, one verifies that the limiting factor is the spreading factor. An analysis of the global traffic processed by the Node B was also made, obtaining 350 MB/h as a maximum for a heavy data profile, and 140 MB/h for a light one, for a service profile based on voice.

In OPNET Modeler, a single-sector cell was evaluated for an increasing number of users, and for two different service scenarios. Global and application related parameters were assessed. As OPNET gives priority to Conversational and Streaming classes, above 15 users, the percentage of queued requests is higher than 80%, the remaining being granted. Only Interactive and Background classes contribute for the percentage of queued requests. As expected, the application throughput is low, and response times are high, not only for data services but also for voice ones, thus, considering VoIP not being recommendable in Release 99.

Keywords

UMTS FDD, HSDPA, OPNET Modeler, Multiservice Traffic, Capacity

Resumo

O objectivo desta tese foi a análise do comportamento da rede UMTS com a implementação de HSDPA. Um modelo teórico do cálculo da capacidade foi desenvolvido, analisando apenas um sector da célula, focando-se na análise de parâmetros como o número de utilizadores servidos pelo Node B, o ritmo de transmissão e o raio da célula. Foram definidos dois cenários com diferentes disposições de utilizadores. Recorrendo ao simulador OPNET Modeler, a Release 99 foi analisada variando o número de utilizadores, e definindo dois cenários que se diferenciavam entre si pelo peso atribuído a cada serviço.

No modelo teórico, a análise foi feita de duas formas: até alcançar a saturação da rede e até um grau de serviço de 5%. Para o último caso, verificou-se que o factor mais limitativo é o factor de espalhamento. Foi ainda efectuada uma análise do tráfego global, obtendo-se no máximo 350 MB/h para um perfil centrado em dados e cerca de 140 MB/h para um perfil de serviços centrado na voz.

Foi simulada uma célula com um sector. Foram analisados resultados em termos globais e por aplicação. Dando o OPNET prioridade às classes *Conversational* e *Streaming*, a partir de 15 utilizadores, a percentagem de pedidos em espera é superior a 80% sendo que os restantes pedidos foram admitidos. O ritmo de transmissão para serviços de dados é baixo, e o tempo de espera é elevado, este último não só para os dados mas também para a voz, tornando VoIP uma tecnologia pouco recomendável sobre Release 99.

Palavras-chave

UMTS FDD, HSDPA, OPNET, Tráfego Multiserviço, Capacidade

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List of Acronyms

3GPP	3rd Generation Partnership Project
AAL	ATM Layer Adaptation
AICH	Acquisition Indicator Channel
AM	Acknowledged Mode
AMC	Adaptive Modulation and Coding
AMR	Adaptive Multi Rate
ARP	Allocation/Retention Priorities
ARQ	Automatic Repeat Request
ATM	Asynchronous Transfer Mode
BCH	Broadcast Channel
BS	Base Station
BWA	Broadband Wireless Access
CDMA	Code Division Multiple Access
CN	Core Network
CPCH	Common Packet Channel
CPICH	Common Pilot Channel
CPU	Central Processing Unit
CQI	Channel Quality Indicator
CS	Circuit Switched
DCH	Dedicated Channel
DES	Discrete Event Simulation
DL	Downlink
DSCH	Downlink Shared Channel
DSL	Digital Subscriber Loop
DTX	Discontinuous Transmission
EDGE	Enhanced Data rates for GSM Evolution
EIRP	Equivalent Isotropic Radiated Power
FACH	Forward Access Channel
FDD	Frequency Division Duplex
FSM	Finite State Machine
GGSN	Gateway GPRS Support Node
GMM	GPRS Mobility Management
GMSC	Gateway MSC
GPRS	General Packet Radio System

GSM	Global System for Mobile Communications
GUI	Graphical User Interface
HARQ	Hybrid Automatic Repeat Request
HLR	Home Location Register
HSDPA	High Speed Downlink Packet Access
HSOPA	High Speed OFDM Packet Access
HSPA	High Speed Packet Access
HSUPA	High Speed Uplink Packet Access
HS-DPCCH	High-Speed Dedicated Physical Control Channel
HS-DSCH	High-Speed Downlink Shared Channel
HS-PDSCH	High-Speed Physical Downlink Shared Channel
HS-SCCH	High-Speed Shared Control Channel
IEEE	Institute of Electrical and Electronics Engineers
IP	Internet Protocol
ITU	International Telecommunication Union
LTE	Long Term Evolution
MAC	Medium Access Control
MAC-hs	Medium Access Control – high speed
ME	Mobile Equipment
MIMO	Multiple Input Multiple Output
MMS	Multimedia Message Service
MT	Mobile Terminal
MS	Mobile Station
MSC	Mobile Services Switching Centre
NBAP	Node B Application Protocol
OFDM	Orthogonal Frequency Division Multiplexing
OPNET	Optimized Network Engineering Tool
OVSF	Orthogonal Variable Spreading Factor
PCH	Paging Channel
PDCP	Packet Data Convergence Protocol
PDP	Packet Data Protocol
PRACH	Physical Random Access Channel
PS	Packet Switched
PSTN	Public Switched Telephone Network
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature Phase-Shift Keying
RAB	Radio Access Bearer
RACH	Random Access Channel
RAN	Radio Access Network

RIP	Routing Information Protocol
RLC	Radio Link Control
RNC	Radio Network Controller
RNS	Radio Network Sub-system
RRM	Radio Resource Management
RSVP	Resource Reservation Protocol
SCCPCH	Secondary Common Control Physical Channel
SF	Spreading Factor
SGSN	Serving GPRS Support Node
SINR	Signal-to-Interference-plus-Noise Ratio
SIP	Session Initiation Protocol
SIR	Signal-to-Interference Ratio
SMS	Short Message Service
SRB	Signalling Radio Bearer
TCP	Transport Control Protocol
TDD	Time Division Duplex
TPAL	Transport Adaptation Layer
ToS	Type of Service
TTI	Time Transmission Interval
UDP	User Datagram Protocol
UE	User Equipment
UL	Uplink
UMTS	Universal Mobile Telecommunications System
USIM	UMTS Subscriber Identity Module
UTRAN	UMTS Terrestrial Radio Access Network
VLR	Visitor Location Register
VoIP	Voice Over IP
WCDMA	Wideband CDMA
WiBro	Wireless Broadband
Wi-Fi	Wireless Fidelity
WiMAX	Worldwide Interoperability for Microwave Access

List of Symbols

a	DL orthogonality factor.
a_j	Orthogonality factor for user *j*.
h	Load Factor.
h_{DL}	Global load factor of a cell in DL.
h_{HS}	HSDPA throughput efficiency.
h_{N_u}	Number of served users ratio.
h_{UL}	Global load factor of a cell in UL.
Δf	Signal bandwidth.
ΔR	Distance between users.
l	Number of calls per hour.
l_{ef}	Effective calls per hour.
l_i	number of calls per hour for service i.
r	Signal-to-interference-plus-noise ratio
q_C	Throughput per code.
q_u	Throughput per user.
y	Orientation angle.
C_u	Number of HS-PDSCH codes per user.
d	Distance between the cell radius for the pedestrian and indoor environments.
E_b	Bit Energy
f	Frequency.
F	Noise figure.
F_{a_j}	Activity factor for user *j*.
G_P	Processing gain.
G_r	Gain of receiving antenna.
G_{SHO}	Soft handover gain.
G_t	Gain of transmitting antenna.

H_B	Node B height.
h_b	Building height.
h_m	MT height.
i	DL Inter-to-intra cell interference ratio.
I_{inter_n}	Normalised inter-cell interference.
$I_{\mathrm{inter}_{nj}}$	Normalised inter-cell interference of user j.
L_C	Losses in cable between transmitter and antenna.
L_{int}	Indoor penetration.
L_P	Path loss.
$\overline{L}_{p_j}$	Average cell path-loss.
$L_{P_{total}}$	Total path loss.
L_{tm}	Losses due to diffraction from the roof to the MT.
L_{tt}	Losses due to propagation over roof tops.
L_u	Losses due to user.
M_{FF}	Fast fading margin.
M_I	Interference Margin.
M_{SF}	Slow fading margin.
N_0	Noise power spectral density.
N_{au}	Number of active users per cell.
N_C	Number of connections per cell.
N_i	Number of served users for service i.
N_{MAX}^{HS}	Maximum number of HS-PDSCH codes.
N_{RF}	Average noise power.
N_s	Number of users of the corresponding service.
N_u	Number of users.
N_u^{HS}	Number of HSDPA users.
N_u^{serv}	Number of served users.
p_{HSDPA}	HSDPA power percentage.

P_r	Power available at the receiving antenna.
P_{Rx}	Power at the input of the receiver.
$P_{Rx\min}$	Receiver sensitivity.
P_t	Power fed to transmitting antenna.
P_{Tx}	Transmitter output power.
P_{Tx}^{BS}	Total transmission power.
R	Cell radius.
R_b^{ef}	Effective throughput.
R_b^{tg}	Target throughput.
R_{b_j}	Bit rate of user j.
R_c	WCDMA chip rate.
$R_{\max}$	Maximum cell radius.
T	Global traffic processed by the Node B.
V_i	Mean file volume for service i.
w_B	Building separation.
w_s	Street width.

List of Programmes

MatLab

Microsoft Office Excel

Microsoft Office Word

Microsoft Visual Studio

OPNET Modeler

Chapter 1

Introduction

This chapter gives a brief overview of the work. Before establishing work targets and original contributions, scope and motivations are brought up. At the end of the chapter, the work structure is provided.

The development of the Mobile Communication Systems caused a revolution in the way people were used to communicate. One of the most important advantages that mobile systems brought over the fixed ones was mobility. Nowadays, people can always be reachable, anytime, anywhere.

Mobile Communication Systems developed from the so-called First Generation systems to the ones used nowadays, Third Generation systems. First Generation systems were analogue and only allowed voice communications. With the arrival of the Second Generation systems, already digital ones, although the main interest of the users was still voice communication, the importance of other types of service has grown. Examples of these services can be short text messaging and enabling the access to data networks.

As the interest of the public grows around multimedia services, new technologies arise, providing an access to these kinds of services with an improved quality, turning the telecommunications business a very competitive and lucrative one, which benefits both end users and providers. Figure 1.1 depicts the evolution of technologies and the trend in mobile wireless access, based on [UMFo07]. As an example of this spreading of mobile communications, according to ANACOM, the First Generation appeared in Portugal in 1989, and at the end of that year 2800 subscribers were counted. This number has grown constantly, and at the end of 2006 there were 12.2 million subscribers in Portugal, with a penetration rate of 115.7%, [ANAC07].

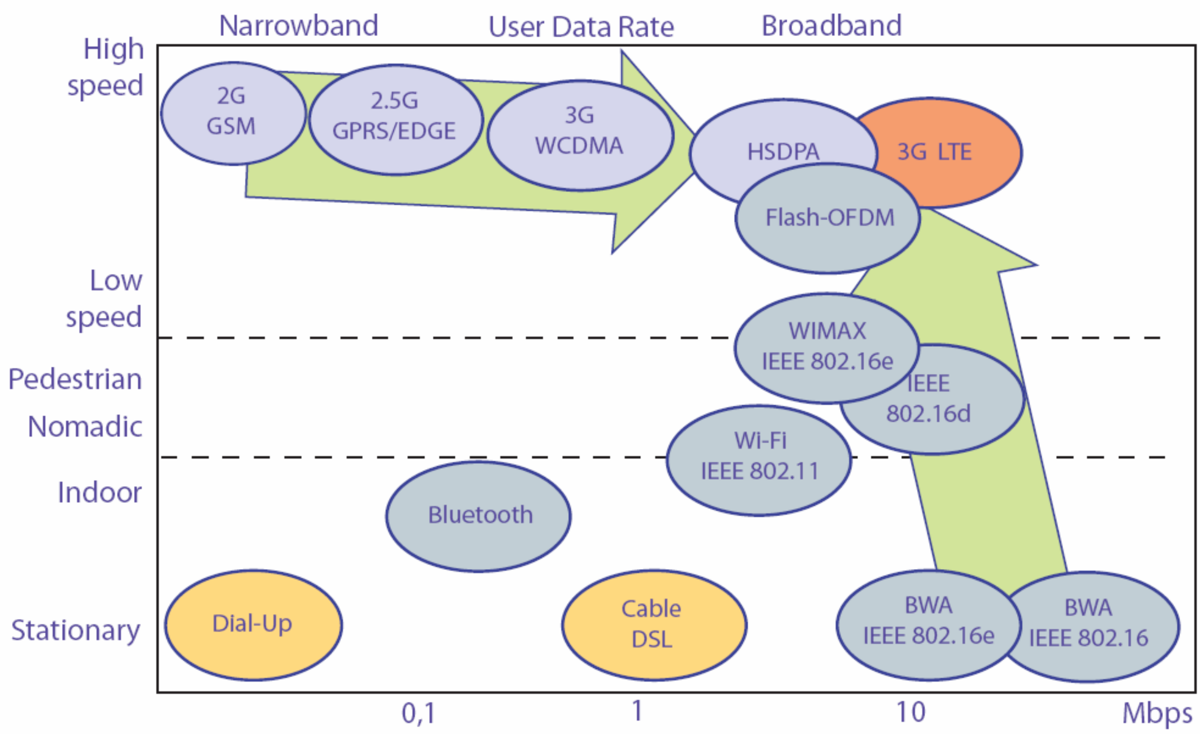


Figure 1.1. Evolution of mobile wireless access technologies, [UMFo07].

With this growing interest, networks supporting data communication with better quality started arising. With Global System for Mobile Communications (GSM), data services were supported, although the throughputs were very low. So, General Packet Radio Service (GPRS) appeared bringing a new variety of multimedia services and enabling a better quality in the access to data networks.

This quality improved with the appearance of Universal Mobile Telecommunication System (UMTS), which is the European standard for the Third Generation systems. UMTS Release 99 enabled a very good access to data networks, providing a maximum of 384 kbit/s data rate in downlink (DL), [HoTo04], which brought a new component to the mobile communications market, turning mobile operators also into Internet providers, competing with fixed operators not in terms of throughput but in terms of mobility. Figure 1.2 shows the Wideband Code Division Multiple Access (WCDMA) subscribers growth from January 2004 to February 2006.

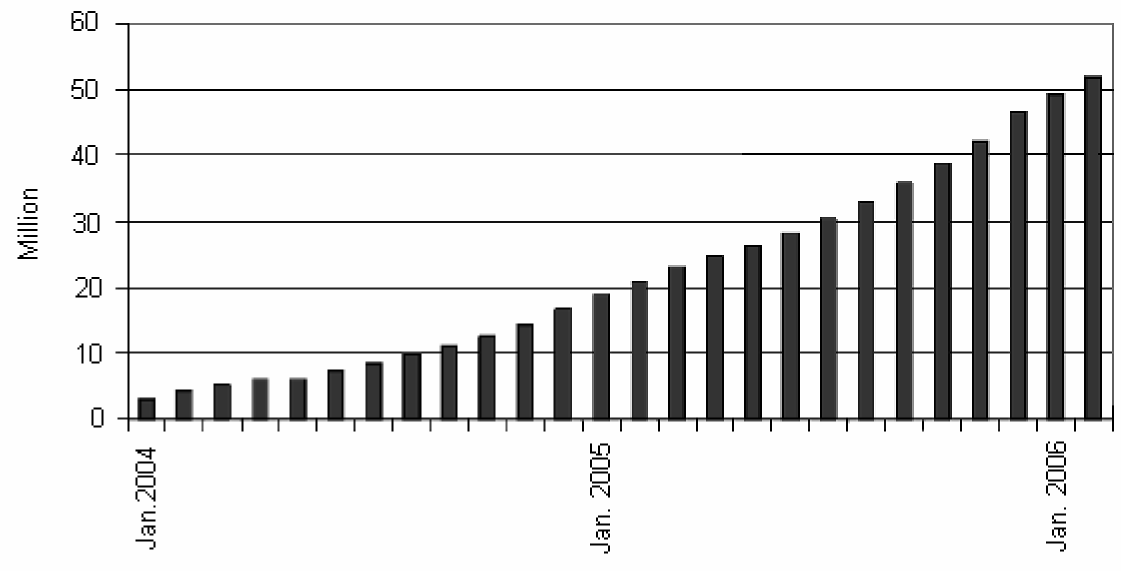


Figure 1.2. Number of WCDMA subscribers growth, [HoTo06].

With the increasing interest on data based services, High Speed Downlink Packet Access (HSDPA) was developed, its first specification being integrated as part of 3GPP Release 5, [HoTo06], and was launched on March 2002. When the first HSDPA networks became available at the end of 2005, data rates were able to reach 1.8 Mbit/s, increasing to 3.6 Mbit/s in 2006, and nowadays 7.2 Mbit/s are already available. According to the UMTS Forum, Portugal is among the top ten countries ranked by WCDMA penetration, with 25% WCDMA, and 1% HSPA, [UMFo07]. With these improvements in the DL channel, and, hence, the growing number of users using data based services, Release 99 has not enough capacity to answer the expectations for the uplink (UL) channel. So, in December 2004 a new specification was launched, Release 6, which characterised High Speed Uplink Packet Access (HSUPA), enabling data rates up to 2 Mbit/s in an initial phase, [HoTo06].

Some new features introduced with HSDPA, which enhances its performance, are the adaptive modulation and coding (AMC), adjusting the data rate concerning the available channel quality, the Hybrid Automatic Repeat Request (HARQ) mechanism, performing retransmission in the physical layer, and the medium access control layer (MAC-hs) located in the Node B. Also a new Transmission

Time Interval (TTI) of 2 ms is implemented, allowing the system to be more reactive to changes in the radio channel conditions, thus, performing a better allocation of resources among users. The main characteristic that enables this technology is the new supported modulation, 16-QAM, contrarily to Release 99, which only supported Quadrature Phase Shift Keying (QPSK). Only for HSDPA use, a new channel is implemented, which can be shared among various users, [HoTo06]. Concerning HSUPA, the most important introduced characteristics are the fast Node B scheduling, a new dedicated channel, and, similarly to HSDPA, the introduction of the HARQ mechanism.

With the new enhanced characteristics brought by High Speed Packet Access (HSPA), new services arise, as Voice over IP (VoIP) or multiplayer gaming, and existing ones are improved, as peer-to-peer applications and streaming video. Figure 1.3 depicts the evolution enabled by HSPA to the services supported by this technology.

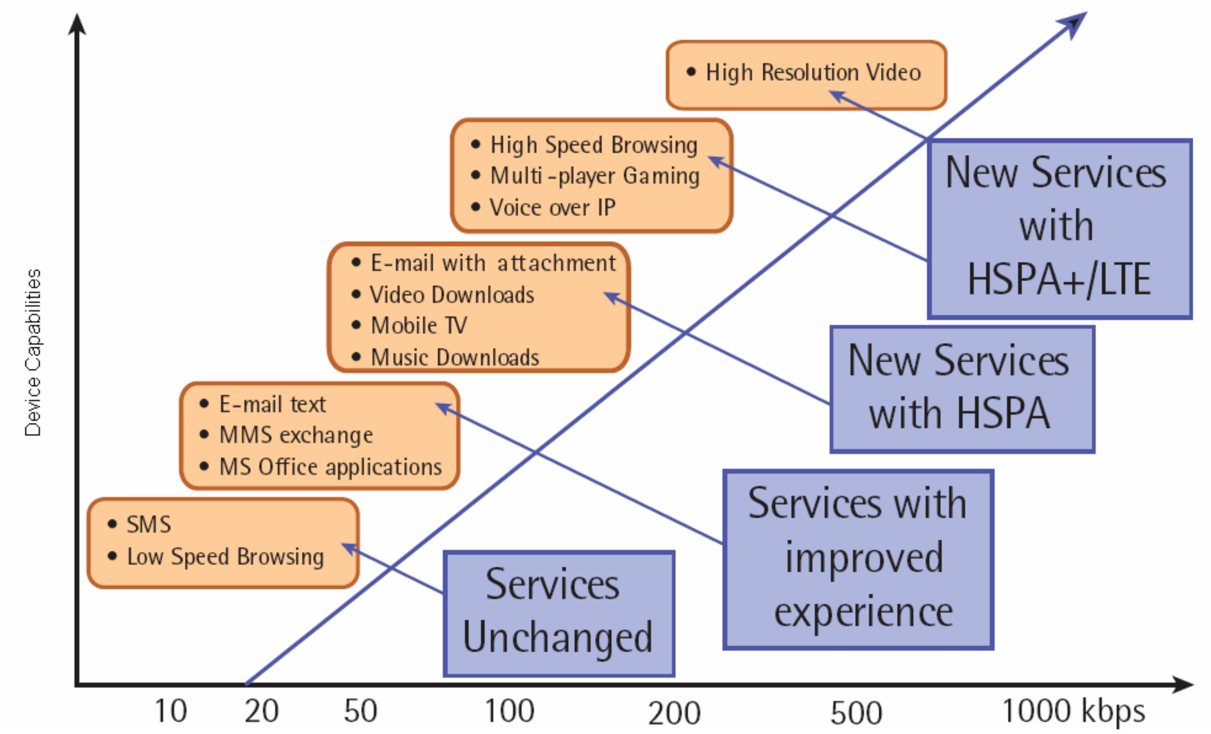


Figure 1.3. Evolution in services with HSPA, [UMFo07].

The evolution in technology improvements is constant, so, combining Orthogonal Frequency Division Multiplexing (OFDM) and Multiple Input Multiple Output (MIMO) antennas, High Speed OFDM Packet Access (HSOPA) appeared, enabling throughputs even higher than HSPA and a significant decrease in the cost per bit, and should become an attractive alternative to providers and end-users, [Nort05].

Already considering a Fourth Generation technology, 3GPP is already working in Long Term Evolution (LTE) technology, which will enable increased peak data rates, with instantaneous maximums of 100 Mbit/s in DL and 50 Mbit/s in UL, both in a 20 MHz frequency band. Another advantage is the compatibility with earlier releases and with other systems, [3GPP07].

The goal of this thesis is to analyse the impact of HSDPA implementation on the behaviour of UMTS networks, in terms of capacity and interference. For this, a model that calculates network capacity in terms of served users, throughput and cell radius was developed and implemented. The obtained results are compared with simulation results, which were obtained by using a modelling and simulating tool named OPNET Modeler. Only Release 99 was simulated, since OPNET does not have HSDPA implemented.

This thesis is composed of 5 chapters, including this Introduction.

Chapter 2 presents an overview of UMTS basic aspects, describing how the system works focusing on network architecture, the radio interface, and capacity and interference. Nowadays in use services and applications are presented. HSDPA is described, focusing on the new channels that had to be introduced, the system performance and on radio resource management. Finally, a performance analysis is made describing the state of the art.

In Chapter 3 the model for theoretical calculations is presented. The chosen scenarios and correspondent parameters are described and an analysis of results is made, firstly analysing the users distribution, then Release 99 and HSDPA and finally, gathering the obtained results, analysing the global traffic of the network.

Chapter 4 introduces the OPNET Modeler simulator, describing its functionalities focusing in the UMTS modules. Then the options made for simulations, as network architecture, profiles and applications parameters, and scenarios are presented. The analysis of results is made separating global results of the network and applications results. Finally, a comparison between the results of the simulator and the theoretical model is made.

Finally in Chapter 5 some conclusions are drawn and suggestions of future work presented.

Chapter 2

UMTS Basic Aspects

This chapter provides an overview of UMTS basic aspects. In Section 2.1, the system is described, focussing on its architecture, aspects of the radio interfaces, capacity and interference. In Section 2.2, services and applications are presented. HSDPA is presented in Section 2.3, which describes the new channels, performance and radio resource management. Finally, in Section 2.4, the current state of the art is presented.

2.1 The system

2.1.1 Network Architecture

The elements that compose the UMTS network can be divided into three groups, [HoTo04], as depicted in Figure 2.1., according to their functions: the User Equipment (UE), the UMTS Terrestrial Radio Access Network (UTRAN) and the Core Network (CN).

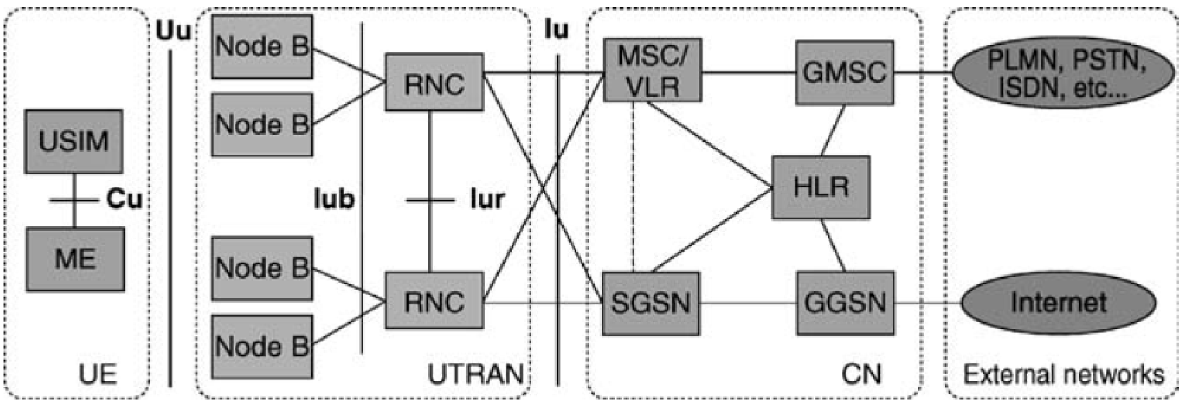


Figure 2.1. UMTS Network Architecture, [HoTo04].

The UE is responsible for the radio communications between the user and the network, including two elements: the UMTS Subscriber Identify Module (USIM), which is a smartcard containing information about the subscriber and authentication operations, and the Mobile Equipment (ME), which is the radio terminal that makes the connection between UE and UTRAN possible, over the Uu Interface, i.e., the WCDMA radio interface. The USIM and the ME are connected to each other via an electrical interface, named Cu.

UTRAN deals with the radio related functionalities, comprising one or more sub-systems, called Radio Network Sub-systems (RNS). Each RNS is composed of one Radio Network Controller (RNC) and one or several Node Bs, i.e., Base Stations (BSs). The RNC is responsible for the control of the radio resources, and the Node B processes the information flow between UE and RNC.

UE and UTRAN protocols were designed for the needs of the WCDMA radio technology, but the UMTS CN is an adaptation of the GSM one. The CN consists of the following elements:

- Home Location Register (HLR): a database where the operator subscribers' information is stored, consisting mainly of the users' service profile and their location.

- Mobile Service Switching Centre (MSC): dealing with the switching in the Circuit Switched (CS) domain.
- Visitor Location Register (VLR): database with information on active users.
- Gateway MSC (GMSC): linking the CN to CS external networks.
- Serving GPRS Support Node (SGSN): performing a function identical to MSC/VLR, but concerning Packet Switched (PS) services.
- Gateway GPRS Support Node (GGSN): linking the CN to PS external networks.

The CN switches and routes calls or data connections to external networks, which can be of two types: CS networks, such as Public Switched Telephone Network (PSTN), or PS ones, such as the Internet.

One of the main factors for the new design of the UE and UTRAN modules is that the latter was required to support soft handovers. In GSM, only hard handover occurs, in which the connection from a Mobile Terminal (MT) is reassigned from one BS to another without being connected simultaneously to both. In UMTS, hard handover can be either inter-frequency or inter-system: in the former the MT is transferred from one frequency carrier to another, which involves the use of several carriers per BS; in the latter, the MT is transferred from UMTS to another system, currently GSM, enabling load balancing and coverage enhancements. UMTS supports the other two types of handover:

- soft handover, where the MT is connected to more than one BS at the same time;
- softer handover, similar to soft handover, but only when the MT is transferred from one sector of a cell to another sector of the same cell.

2.1.2 Radio interface

In UMTS, the air interface is based on WCDMA, which was specified by the 3rd Generation Partnership Project (3GPP), [3GPP07]. There can be two modes of operation: Frequency Division Duplex (FDD) and Time Division Duplex (TDD). In this work, only the FDD mode is considered, since the TDD mode, which was designed to provide high data rates, was replaced by HSPA in FDD (which is addressed later).

WCDMA is a wideband multiple access technique based on CDMA, [HoTo04], which leads to a very wide bandwidth, a carrier bandwidth of 4.4 MHz being achieved with a chip rate of 3.84 Mcps. Another important characteristic of WCDMA, mainly for PS data services, is the possibility of performing variable user data rates; the data rate is constant over each 10 ms frame, but it can be different from frame to frame.

Spreading is used to separate the physical data and control channels in UL and to distinguish the connections to different users within one cell in DL, consisting of two operations: channelisation and scrambling. The former originates a wideband signal by multiplying the user's data by a sequence of chips, being associated to a spreading factor. The use of the Orthogonal Variable Spreading Factor (OVSF) technique to obtain the channelisation code enables to maintain different spreading codes orthogonal to each other, and also to change the spreading factor between them.

The scrambling operation is used over spreading, but it does not modify the signal bandwidth. In DL, it differentiates the sectors of the cell, and in UL, it separates MTs from each other. The scrambling code can be either a short or a long one, the latter being a 10 ms code based on the Gold family, and the former being based on the extended S(2) family. UL scrambling uses both short and long codes, while DL one only employs long codes. Codes characteristics are summarised in Table 2.1.

Table 2.1. Functionality of the channelisation and scrambling codes, [Corr06].

	Channelisation	**Scrambling**
Use	DL: MT separation UL: Channel separation	DL: Sector separation UL: MT separation
Duration	DL: 4 – 512 chip UL: 4 – 256 chip	38 400 chip
Number	Spreading Factor (SF)	DL: 512 UL: several millions
Family	OVSF	Gold or S(2)
Spreading	Yes	No

The signal resulting from the multiplication of the user's data by the channelisation code is again multiplied by the scrambling code, Figure 2.2., which gives the final chip rate that is transmitted.

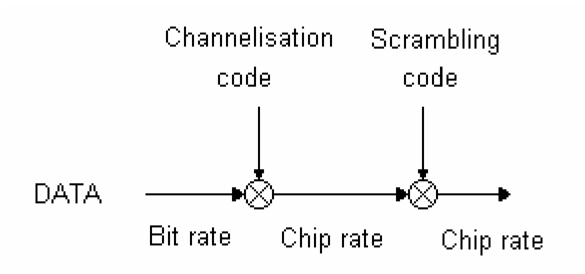


Figure 2.2. Relation between spreading and scrambling, [Corr06].

Concerning the channels, there can be four types: radio, physical, logical and transport. Radio channels are related to the carrier frequency, being separated by 5 MHz. The frequency bands used for FDD are [1920, 1980] MHz for UL, and [2110, 2170] MHz for DL.

Logical channels assure the exchange of specific information between the user and the network. UTRAN protocols define the kind of information generated in higher layers that must be transferred, which is carried through the air interface by transport channels, being then mapped onto physical channels in the physical layers.

Transport channels can be of two types: a common channel, which is a resource shared by all users in the cell, and a dedicated channel, which is a resource intended to a given user. The Dedicated Channel (DCH) is the only dedicated transport channel, being responsible for the transport of user's information from higher layers. The kind of transferred information can be data for the actual service, and control information provided by the higher layers. The common transport channels are the Broadcast Channel (BCH), the Forward Access Channel (FACH), the Paging Channel (PCH), the Random Access Channel (RACH), the Uplink Common Packet Channel (CPCH) and finally the Downlink Shared Channel (DSCH). Common channels essential for the basic network operations are FACH, PCH and RACH. In a cell, there can be one or more FACHs, which are used to carry control information for locating a known MT in a specific cell. The required data for the initiation of a connection between the network and a user is carried by the Paging Channel. The Random Access Channel carries the necessary control information from the MT, for example to initiate a connection.

Physical channels can be used to map transport channels, and also to carry information of physical layer procedures. Physical channels enssure the transfer of information through the radio interface, each one being associated to a different code. The Secondary Common Control Physical Channel (SCCPCH) is used to map the FACH and PCH transport channels, and RACH is mapped onto the Physical Random Access Channel (PRACH).

One of the most important aspects in WCDMA is power control. This feature aims at maximising capacity by equalling the received power of all MTs. Power control can be of two kinds: open-loop or closed-loop. The former, also called outer loop, is used to supply the initial power to the MT that is initiating a connection, while the latter, also named inner loop, is crucial in a WCDMA system, occurring at a rate of 1.5 kHz. Open loop power control sets a target Signal-to-Interference Ratio (SIR) that is compared with the estimated received SIR at the BS. Depending on this comparison, the BS sends a command to the MT to increase or decrease power. In DL, the used technique is very similar, although it is desirable to allocate additional power to MTs located at the cell edge, due to the inter-cell interference to which they are exposed to.

2.1.3 Capacity and Interference

In UMTS, system's capacity depends on the number of MTs that can be connected to a certain BS, being limited mainly by three factors:

- The number of available codes, as it might not be sufficient for identifying all MTs within a certain cell, which usually is not a real constraint.
- BS transmission power, as it is restricted and shared among all MTs in the cell.
- System load, as it affects the cell coverage, hence, the interference margin must be taken into account, it being the most important of the three factors.

The interference margin is given by

$$M_{I[\mathrm{dB}]} = -10 \cdot \log\left(1-h\right), \quad (2.1)$$

where h represents the load factor.

The system load is used to estimate the interference margin and the cell capacity, so, the global load factor specifies the amount of supported traffic per BS. If the load factor increases, the interference margin also increases, which reduces coverage. In case the load factor approaches unity, interference goes to infinity, leading the system to achieve its pole capacity. The load factor per user depends on the link and on the type of service.

The UL load factor for a given user *j* is given by

$$h_{UL} = \left(1 + I_{\mathrm{inter}_n}\right) \sum_{j=1}^{N_{au}} \frac{1}{1 + \frac{R_c / R_{b_j}}{\left(E_b / N_0\right)_j F_{a_j}}}, \quad (2.2)$$

where:

- N_{au} : number of active users per cell;
- F_{a_j} : activity factor for user *j*;
- $\left(E_b / N_0\right)_j$: energy per bit over noise spectral density, equivalent to signal-to-noise ratio, required to meet a predefined QoS;
- R_c : WCDMA chip rate;
- R_{b_j} : bit rate of user *j*;
- I_{inter_n} : normalised inter-cell interference.

The DL load factor is given by

$$h_{DL} = \sum_{j=1}^{N_C} F_{a_j} \frac{\left(E_b / N_0\right)_j}{R_c / R_{b_j}} \left[\left(1 + a_j\right) + I_{\mathrm{inter}_{n_j}}\right], \quad (2.3)$$

where:

- N_C : number of connections per cell;
- a_j : code orthogonality factor of user *j*;
- $I_{\mathrm{inter}_{n_j}}$: normalised inter -cell interference of user *j*.

The main difference between UL and DL load equations is the a_j parameter. For perfectly orthogonal codes, the orthogonality factor would be 1; nevertheless, with multipath propagation, the orthogonality factor may vary between 0.4 and 0.9, and the MT might mistake the BS signal for multiple access interference. Also in DL, N_C stands for the number of connections per cell, since soft-handover transmissions are accounted as extra connections.

The number of available codes in a certain cell depends on the number of users and on the bit rate required by the type of service each user is accessing. The number of channelisation codes is given by the Spreading Factor. The higher the bit rate is, the smallest the SF will be, so, when the number of users increases, the bit rate of the bearer service also increases, leading to a decrease of the available SF, hence, of the number of codes. The number of available scrambling codes may be a limitative factor only in DL, as there are only 512 available codes, Table 2.1. Although this is the less important factor of the three previously listed, it must be taken into account.

In DL, coverage is more dependent on the load than in UL, since the BS has a maximum transmission power, despite of the number of users in the cell. The total transmission power is expressed by

$$P_{Tx}^{BS} = \frac{N_0 \cdot R_c}{1-\bar{\eta}_{DL}} \sum_{j=1}^{N_u} F_{a_j} \overline{L_{p_j}} \frac{\left(E_b / N_0\right)_j}{R_c / R_{b_j}} , \tag{2.4}$$

where

- § N_0 being the noise spectral density of the MT receiver;
- § $\overline{L}_{p_j}$ the average cell path loss.

Even if the load in DL is low, coverage decreases with the increase of the number of users. So, coverage is limited by UL and the capacity is limited by DL.

2.2 Services and Applications

One attribute of UMTS is the wide range of available services. WCDMA characteristics, like practical data rates above 2 Mbit/s with 3GPP Release 5, enable the appearance of new services and the enhancement of the existing ones. In order to allow efficient ways of accessing services, and to optimise system capacity, 3GPP defined different classes of services based on their Quality of Service (QoS) requirements. These classes are distinguished by how delay-sensitive traffic can be, being used to prioritise traffic when the system load is too high. The four identified QoS classes are [3GPP06a]: conversational, streaming, interactive and background. The conversational class is the most delay-sensitive one, while the background class is the less delay-sensitive. UMTS QoS classes' main parameters and characteristics are summarised in Table 2.2.

In each class, three different allocation/retention priority (ARP) categories are defined. When the network has low resources available, ARP is used to prioritise bearers when performing admission control.

Table 2.2. QoS classes main parameters and characteristics, [3GPP06a].

Class	Conversational	Streaming	Interactive	Background
Maximum Transfer Delay	80 ms	250 ms	-	-
Guaranteed bit rate	Yes	No	No	No
Traffic Handling Priority	-	-	1,2,3	-
Allocation/Retention Priority	1,2,3	1,2,3	1,2,3	1,2,3
Real-Time	Yes	Yes	No	No
Symmetric	Yes	No	No	No
Switching	CS / PS	CS / PS	PS	PS
Examples	Voice	Video streaming	www	MMS

The conversational class is intended for real-time conversation based on both CS traffic, as the speech service, and PS one, as VoIP. The maximum delay on end-to-end real-time conversations (both audio and video) is set by human perception, thus, it must be below 400 ms. For CS speech services, the Adaptive Multirate (AMR) technique is applied (AMR is a speech codec that has eight different source rates, which can vary every 20 ms frame upon a command during an active connection). The Radio Access Network (RAN) controls the bit rate variations concerning the air interface loading and the quality of the speech connection. As the conversational class traffic is characterised by being symmetric, or almost, in a voice conversation both users occupy each link, on average, 50% of the time, so discontinuous transmission (DTX) is employed. With DTX, the required bit rate can decrease, which makes it possible to achieve lower levels of interference, thus, increasing network capacity. It also allows that the MT battery lasts longer. For Video Telephony in CS connections, the used specification is ITU-T Rec. H.324M, [ITUT06], and in PS connections, as IP Multimedia Applications, the Session Initiation Protocol (SIP) is used.

Streaming class services are based on the multimedia streaming technique, which enables the end user to access the data before the transfer is complete, because information is transferred in a continuous stream and is buffered. Streaming applications do not require a delay as stringent as conversational ones; DL traffic is the most significant. Examples of services in the streaming class are audio and video streaming, like broadcast or video on demand.

In the interactive class, based on PS connections, traffic is very asymmetric and tolerant to delay, being mainly applied to end-users requesting data from a remote equipment. The end user may be human or machine. A characteristic of this class is the request response pattern of the end user. In order to schedule traffic according to the bearer capabilities, traffic handling priorities are defined. Applications intended to human interaction consist essentially of web browsing, push-to-talk applications, location-based services and network games, while those intended to machine interaction can be automatic data base enquiries.

The background class is very similar to the former one, but delay can be higher, because the user is not expecting the information within a certain time. Services in the background class only spend the network resources when they are not needed for applications from the other classes. Examples of applications are e-mail delivery, Short Message Service (SMS) and Multimedia Message Service (MMS).

2.3 HSDPA

The following subsections contain a description of the HSDPA channels, performance analysis and Radio Resource Management (RRM) protocols [HoTo06].

2.3.1 Channels

One of the differences introduced with the implementation of HSDPA are the new channels needed for its operation, as shown in Figure 2.3. HSDPA is always operated with the DCH from Release 99, DL packet access operation running in parallel, which can be used to carry CS services and the Signalling Radio Bearer (SRB). The new channels are:

- High-Speed Downlink Shared Channel (HS-DSCH): transport channel carrying actual user data, being mapped onto the High-Speed Physical Downlink Shared Channel (HS-PDSCH). This transport channel is characterised by a TTI of 2 ms, shorter than in Release 99, in which it is 10 ms. Besides QPSK, it also supports 16QAM modulation. Another characteristic is the lack of fast power control and the lack of soft handover, as only one HS-DSCH serving cell is used. For multicode operation, a fixed SF of 16 is used. When a user has no data to be transmitted in the HS-DSCH, there is no transmission and the resource is allocated to another user during the 2 ms TTI.
- High-Speed Shared Control Channel (HS-SCCH): logical channel carrying time-critical signalling information, divided in two parts. The use of a SF=128 enables carrying 40 bits per slot, being divided into 3 slots and having a 2 slots offset compared with the HS-DSCH. The first part contains the information that is needed before the reception of the HS-DSCH, as which codes to de-spread and modulation information, while the second part contains less urgent information.
- High-Speed Dedicated Physical Control Channel (HS-DPCCH): physical channel containing control information for UL. It has a 3 slot structure, a fixed SF of 256 and is divided into two parts. The first one, consisting of one slot, carries the HARQ information. The second one, consisting of the remaining two slots, carries the Channel Quality Information (CQI) feedback.

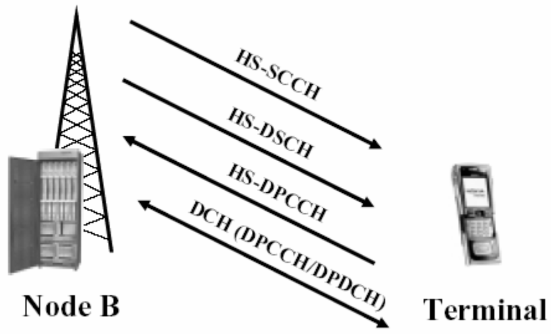


Figure 2.3. Channels needed for HSDPA operation in Release 5, [HoTo06].

Although HSDPA uses a fixed SF of 16, from these 16 available codes, only 15 can be allocated to data transmission, as 1 code is needed for HS-SCCH transmission. From the BS point of view, all 15 codes can be allocated; however, for the MT, the allocated codes can vary within 5, 10 or 15, depending on the MT category.

Terminals supporting HSDPA are divided into 12 categories, which were specified by 3GPP, being differentiated from each other by the modulation, number of supported codes, minimum inter-TTI interval, and ARQ type at maximum data rate, Table 2.3. The first one to be implemented was Category 12, which only supports QPSK modulation and has 5 codes available for users. Then, category 6 was implemented, also with only 5 codes available, but supporting 16-QAM modulation, which enables a theoretical throughput of 3.6 Mbit/s. More recently, Category 8 was already released, with 10 available codes and reaching a theoretical throughput of 7.2 Mbit/s.

Table 2.3. HSDPA terminal capability categories, [HoTo06].

Category	Supported Modulation	Maximum number of parallel codes per HS-DSCH	Minimum inter-TTI interval	Achievable maximum data rate [Mbit/s]
1	16QAM	5	3	1.2
2	16QAM	5	3	1.2
3	16QAM	5	2	1.8
4	16QAM	5	2	1.8
5	16QAM	5	1	3.6
6	16QAM	5	1	3.6
7	16QAM	10	1	7.2
8	16QAM	10	1	7.2
9	16QAM	15	1	10.2
10	16QAM	15	1	14.4
11	QPSK	5	2	0.9
12	QPSK	5	1	1.8

2.3.2 HSDPA Performance

Another difference between HSDPA and Release 99 are the metrics used to assess network performance. In Release 99, the most well-known metric is the signal-to-noise ratio (E_b/N_0), but this is not appropriate for HSDPA, since the bit rate may change every TTI. The metric used to evaluate HSDPA performance, fundamental for link budget planning and network dimensioning, is the average HS-DSCH signal-to-interference-plus-noise ratio (SINR) after HS-PDSCH de-spreading, being given by:

$$r = SF_{16} \frac{P_{Rx,DSCH}}{(1-a) \cdot I_{intra} + I_{inter} + N}, \tag{2.5}$$

where:

- § SF_{16} : HS-PDSCH spreading factor of 16;
- § $P_{Rx,DSCH}$: received power of the HS-DSCH summing over all active HS-PDSCH codes;
- § I_{intra} :intra cell interference;
- § I_{inter} : inter cell interference;
- § N : received noise power.

When analysing single-user performance, which is strongly related to the chosen modulation and coding scheme, link adaptation makes the choice concerning the instantaneous SINR, in order to optimise throughput and delay. QPSK modulation is chosen for a low SINR and 16QAM is employed for a high one, which is necessary to provide the higher data rates. Figure 2.4 illustrates the difference between the use of QPSK and 16QAM modulations for a single user supporting 5 HS-PDSCH codes, and for two channel profiles: Vehicular A and Pedestrian A.

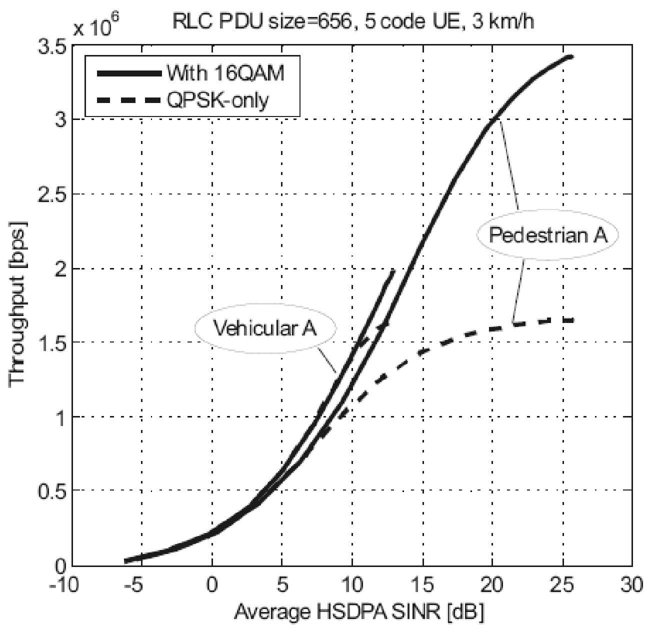


Figure 2.4. Single-user performance with 16QAM/QPSK and with QPSK-only, [HoTo06].

When 5-code transmission is used, the required SINR is higher than with 10 or 15 codes. It can also be seen that with 16-QAM the throughput obtained is much higher, mainly in the Pedestrian A channel. The calculations for the SINR value, for a given inter-to-intra-cell interference ratio, must take several parameters into account such as orthogonality and UE receiver capabilities, since SINR is not a constant value.

The HS-DSCH transmit power is adjusted regarding the HS-DSCH SINR observed by the different users in the cell, which should be within the HS-DSCH dynamic range. For 16QAM modulation, and when using 5 HS-PDSCH codes, the HS-DSCH dynamic range is 20dB, in the interval between -3dB and 17dB. In case the HS-DSCH transmit power is too high, excessive interference is experienced in the network; if the HS-DSCH transmit power is too low, users cannot achieve the highest data rates.

Another measurement to evaluate HSDPA performance is the pilot E_C/I_0, which is a measurement from the CPICH. It is possible to estimate the average single user's throughput resorting to the average P-CPICH E_C/I_0, once knowing the value of the average HS-DSCH SINR and considering Figure 2.5., which represents the data rate as a function of the average HS-DSCH SINR for the cases where 5, 10 or 15 codes are supported, [HoTo06]. The average HS-DSCH SINR is given by:

$$r = SF_{16} \frac{P_{Tx,DSCH}}{P_{Tx,pilot}/r_{pilot} - a\,P_{Tx,tot}}, \tag{2.6}$$

where:

- $P_{Tx,DSCH}$: transmit power form the serving HS-DSCH cell;
- $P_{Tx,tot}$: total Node B transmit power;
- P_{pilot} : P-CPICH transmit power;
- r_{pilot} : P-CPICH E_C/I_0 when HSDPA power is on.

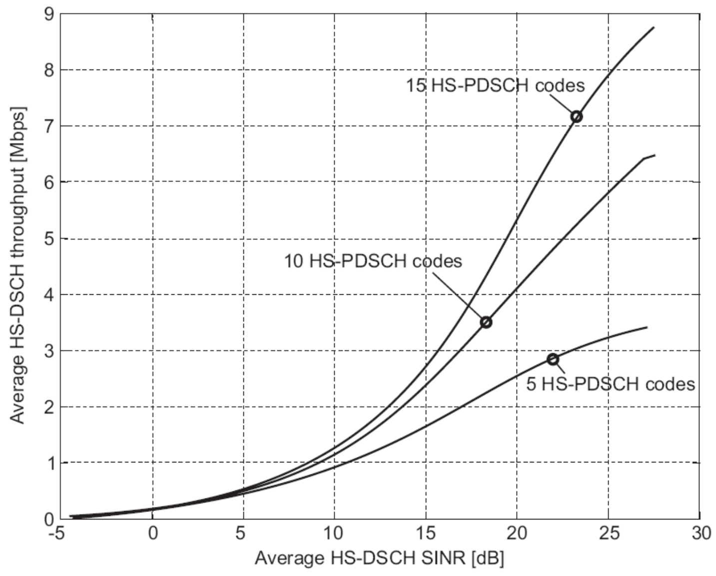


Figure 2.5. HSDPA data rate as a function of the average HS-DSCH SINR, [Pede05].

2.3.3 Radio Resource Management

With the arrival of HSDPA, RRM suffered some changes compared to Release 99. The most relevant RRM algorithms take place in the Iub interface, within the RNC and the Node B. The former deals with resource allocation, admission control and mobility management, while the latter is responsible for HS-DSCH link adaptation, HS-SCCH power control and packet scheduling.

When a HS-DSCH transmission is initiated in the Node B, the RNC must have enough power allocated and channelisation codes so that the communication is possible. The Node B supports from 10 to 15 HS-PDSCH codes, being recommendable to allocate as many codes as possible in spite of improving the spectral efficiency of the HS-DSCH. However, these codes can only be used for HSDPA transmission, and sometimes call blocking of Release 99 users may occur if the codes reserved to the HS-PDSCH are too many. In terms of allocating transmission power for HSDPA, there are two options: the first, and less attractive, is the one in which the RNC allocates a fixed value of power per cell; the second, and most attractive, is the one in which the power allocated is not discriminated by the RNC, hence, the Node B can adjust the power of HSDPA transmission based on measurements of the power used by non-HSDPA channels.

Admission control is a functionality performed by the RNC to determine if new HSDPA users can access the cell, and if they will be served by the HS channels or by DCH. The admission control algorithm needs to consider the available resources in the network, the type of service, and its QoS requirements. In order to perform admission, the Node B must provide three measurements to the RNC: total carrier power, non-HSDPA power, and HS-DSCH required power. With these measurements, the RNC can predict the power available for HSDPA transmissions in the cell. When a new user is trying to access a cell, the MT sends a pilot power measurement to the RNC, so that it can verify if there is enough HSDPA capacity in the cell to accept that new user, while maintaining all the QoS attributes of existing users.

The Node B is responsible for the HS-DSCH link adaptation algorithm, which depends on the user's reception quality, adjusting the bit rate for the HS-DSCH every TTI. The algorithm uses the CQI sent regularly by the UE to the serving HS-DSCH cell. The CQI informs about the maximum transport block size that can be correctly received.

Another algorithm for which the Node B is responsible for is the HS-SCCH power control. Enough power is needed to ensure a reliable reception of the HS-SCCH, which is fundamental for the decoding of the HS-DSCH, hence, to allow communication. It is also important that power is as low as possible, so that the interference level decreases. This algorithm adjusts the transmission power every TTI, power regulation being based on CQI reports or on DPCCH power control commands.

Finally, the packet scheduler is an important part of the Medium Access Control – high speed (MAC-hs), being responsible for sharing the available resources among HSDPA users; a good packet scheduler must consider the available cell resources (power and codes), the number of users in the cell, their radio channel quality, and QoS requirements, thus, maximising cell capacity and allowing a good end user experience. There are many algorithms to perform packet scheduling, the most used ones being known as Round Robin (RR), which schedules all HSDPA users with equal probability, not considering their QoS requirements; another often considered algorithm is the Proportional Fair (PF) scheduler, which provides a fairer split of the available resources among users, also enabling the enhancement of HSDPA cell throughput and coverage.

The scheduling process must take cell parameters into account, such as the codes allocated for HS-SCCH and HS-DSCH, HSDPA power, and user parameters (scheduling priority indicator (SPI), guaranteed bit rate (GBR), discard timer (DT), UE category, and amount of data buffered in the Node B). The packet scheduler elaborates metric calculations concerning these parameters, the scheduling principle, and the operator service strategy, which leads to the scheduling decision.

When code multiplexing is needed, the Node B can use up to 15 HS-PDSCH codes. In order to enable the maximisation of the spectral efficiency, and as the UE only supports 5 codes at a time, the code multiplexing should schedule 5 codes to each one of the three parallel users. Code multiplexing is only used when more than one HSDPA user is scheduled in a single TTI in a cell.

2.4 Performance Analysis

In this section, special emphasis is given to earlier work developed on the subject of this thesis. A common goal of this state of art work is on how to predict system performance.

In [BWBW04], a dynamic WCDMA system level simulator was extended to support HSDPA and enable the analysis of PS services, by collecting performance indicators in different scenarios. The simulation environment consists of multiple moving users in multiple cells where soft and softer handovers are supported; it is also possible to transmit several packet data services simultaneously. The traffic generated by each active MT consists of a sequence of packets with different sizes and times of arrival. An important performance indicator considered in this study is the maximum achievable user throughput distribution within the coverage areas, which is calculated based on link level simulations. Two scenarios are analysed:

- for the micro-cellular scenario, the average maximum data rate achieved for the whole simulation area is 2.8 Mbit/s;
- for the macro-cellular scenario the average peak data rate is 2.14 Mbit/s.

Network performance in the micro-cellular scenario is influenced by the modulation, as 16QAM modulation cannot be used, due to the high interference levels existing in a large area of the simulation. In the macro-cellular scenario, the influence of the modulation is even worse, because of the decreased channel conditions at the cell border. The choice of the scheduling algorithms used in RRM may influence the maximisation of system performance, by allocating the available network resources according to the user QoS requirements. Some work has been developed in the analysis of various scheduling algorithms.

In [Pede05], a new quality based HSDPA access algorithm is proposed and analysed. This algorithm only allows the admission of new users if network resources are enough to guarantee that the new user requirements are fulfilled, and that the already existing users' requirements are maintained. This enables the allocation of resources to users according to their priority; hence, users with high priority have a higher probability of being admitted, compared to users with lower priorities. The new algorithm has been submitted to dynamic system level simulations and only streaming traffic is considered. It is evaluated according to user's satisfaction, which is measured by the time that the user has to wait to have access to the required service. Simulation results show a cell capacity gain of 30% when the algorithm is enabled, comparing to the case when the algorithm is not in use. The parameters used to assess performance are the unsatisfied users probability and the Cumulative Distribution Function (CDF) of the estimated required power to serve new users requesting access.

Another important study, examining the performance of different scheduling algorithms used for HSDPA, is [GoAg05]. The investigated scheduling algorithms are:

- The Maximum Signal-to-Interference Ratio scheduling algorithm (Max-SIR),
- The PF algorithm;
- An algorithm based on a Cost Function (CF), which takes all the three types of traffic into account, allocating different weights to each one of them.

Different media types are considered in the modelling, being spread among three kinds of traffic: block, transaction and streaming. Performance measures used to evaluate system's behaviour are the average delay and the percentage of dropped packets. The average delay is estimated for a given traffic type, quantifying the overall average time taken to download the data units. If the delay exceeds a maximum value allowed by the QoS parameters, then, packets are dropped. Performance results show that, in terms of the average delay, the Max-SINR and PF algorithms performe better than the CF one for block and transaction traffic; however, for streaming traffic, the CF algorithm shows better results. In terms of dropped packets, the CF algorithm has a higher percentage for block traffic and a lower percentage for transaction and streaming ones compared to the Max-SIR and PF simulation values. Another performance measure analysed during this study is the average throughput achieved by a user for different classes of traffic. The CF algorithm results in a higher value of average throughput per user, especially when the coefficient that weights the streaming traffic is high.

With the purpose of analysing performance and optimising the Iub interface of UTRAN, a model for simulating HSDPA was developed with OPNET, [WLTG06]. The model was not based on the OPNET UMTS modules, but on OPNET ATM ones, mainly for simplicity and simulation performance reasons, and because the research is focused on UTRAN. This investigation consists of a system level dynamic simulation, with multiple users but a limited number of services, web traffic being the main application. The research compares performance results for two different flow control mechanisms. The first simulation scenario is the one where the ON/OFF Flow Control scheme is used, which is the simplest mechanism that can be applied for managing the flow control in HSDPA traffic on Iub. In the other scenario the flow control mechanism is Provided Bit Rate (PBR) based Flow Control also called Enhanced Flow Control (EFC). The analysed performance indicators are the ATM throughput, which illustrates the utilisation at link level, the IP throughput, which points out the performance of end users, and delay statistics and buffer occupancy, which show the impact of the two schemes over the Iub interface.

Less relevant studies can be referred to as well, e.g., [RiZh05], which makes a performance evaluation of the VoIP service with an OPNET based simulation, using a fixed-to-mobile scenario with multiple users in a single cell. The metrics used to assess performance are delay, jitter and packet loss. The results are presented as a function of system outage, i.e., the percentage of sessions experiencing more than 2% packet losses for different delay budgets, and system capacity under different conditions. In [TKSG04], a performance analysis is made for HSDPA with radio network planning purposes. The simulation is static, the snapshot simulation technique being used, in a urban scenario with multiple cells. The goal is to estimate signal and interference levels, and resources scheduling. The metric used to evaluate performance is the user throughput, achieved with two different scheduling algorithms, both based on Round Robin.

The mentioned studies illustrate the importance of estimating system performance using HSDPA in PS communications. For the scope of this thesis, the analysed studies are an important basis, but more work has to be done and some differences must be mentioned. A cell level dynamic simulation is used to evaluate HSDPA performance, some of the inputs that are considered being the types of PS services, number of users randomly positioned in the cell, and user profile. Other system characteristics must also be taken into account, as modulation, available network resources and their scheduling (code and power allocation), CQI reporting, and radio channel conditions. The output parameters most used in previous studies are the average packet delay and average throughput. Other parameters may be analysed, as coverage, retransmissions, packet losses, offered and carried traffic.

Performance evaluation simulation is essential for radio network planning, as it offers a prediction of the network behaviour over time, and a prediction of capacity and coverage.

Chapter 3

Capacity and Interference Model

This chapter gives an overview of the model used for assessing network behaviour, a brief description of its implementation, and an analysis of the obtained results.

3.1 Model

In order to understand how HSDPA influences the behaviour of UMTS networks, it is important to make a theoretical approach of the technology. This approach is not done for the network as a whole, but for a sector of a cell. So, this analysis focuses on some parameters, as the number of users that can be served, the throughput available for each of them, according to the service they are performing, and the maximum cell radius that can be achieved. For this purpose, a simple theoretical model was developed and implemented. The objective of the model is to assess how many users are served and the resources they get from the network. The calculations are based on the link budget presented in Annex A, and are described as follows.

The program developed in this work is a simple calculator based on [Lope07] and [Salv07], which assesses a mono-service scenario, having been modified to assess a multi-service scenario as well. It has a snap-shot approach, which means that it evaluates the behaviour of the network in a sector of a cell in a certain instant of time, for DL, where a user is only doing one service. The inputs for radius calculation are:

- N_u : number of users;
- P_{Tx}^{BS} : Node B DL Transmission power;
- f : frequency;
- i : DL Inter-to-intra cell interference ratio;
- p_{HSDPA} : HSDPA power percentage (only for shared carrier analysis).

Three different environments can be chosen: indoor, pedestrian and vehicular. The differences between these three environments are the values used for E_b/N_0 , the fading margins and indoor penetration. It is possible to choose two service profiles, which have different percentage of users in each service: voice centric and data centric. Other parameters can be modified, as BS antenna gain, diversity and soft handover gains, additional losses, activity factor, traffic power percentage, and DL orthogonality factor.

Firstly, based on the total number of users and on the service profile chosen, the number of user for each service is calculated. The number of users available for HSDPA is limited, so if the number of PS users goes beyond this limit, the users that cannot be served with HSDPA are served with Release 99. These values are also limited by the available spreading factors. The algorithm to allocate SFs is very simple. First, it allocates SFs for signalling and control channels, then, for each service, it verifies if there are still SFs available and allocates them according to the priority of each user, Table 3.1. In this verification, HSDPA users are not distinguished according to their service, nor are PS Release 99 ones, except for HTTP users, which have a different activity factor than the other PS Release 99 services.

For Release 99, the first step is to calculate the load factor (2.3), for which each service is associated to a value of E_b / N_0 and activity factor, according to Table A.3 and Table A.4, respectively. After obtaining the global load factor as the sum of the load factor of each service, the *EIRP* for DL (A.2), the mean throughput and the processing gain (A.6), the total path loss is calculated using (A.9).

Two factors can be a limitation for the number of served users by Release 99. One is the SF, which is most limitative, mainly for PS, but in general, the most limitative is the load factor. The limiting load factor considered for this analysis is 0.7, but the case of the maximum load factor is 0.8 also studied.

Table 3.1. Spreading factor allocation priority table.

# priority	Service	Throughput [kbit/s]	Switching Type	SF	# equivalent SF256 codes
1	Voice (R99)	12.2	CS	128	2
2	VT (R99)	64	CS	32	8
3	HSDPA	-	-	16	16
4	PS (R99)	384	PS	8	32
5		128	PS	16	16
6		64	PS	32	8

When the load factor exceeds the maximum, the number of users is re-calculated, based on the limit load factor, by manipulating (2.3), and then, based on the new number of users of each service that can be supported, a new global load factor is obtained, from which the total path loss is calculated.

The radius is obtained according to the COST-231 Walfisch-Ikegami propagation model, [DaCo99], which is presented in Annex B. The cell radius can be determined by:

$$R_{[\mathrm{km}]} = 10^{\frac{L_{P[\mathrm{dB}]} - 32.4 - \log\left(f_{[\mathrm{MHz}]}\right) - L_{tt[\mathrm{dB}]} - L_{tm[\mathrm{dB}]}}{20}}, \tag{3.1}$$

being:

- R : cell radius;
- L_P : total cell path loss;
- L_{tt} : losses due to propagation over roof tops;
- L_{tm} : losses due to diffraction from the roof to the MT;

which can also be written as a function of power:

$$R_{[\mathrm{km}]} = 10^{\frac{EIRP_{[\mathrm{dBm}]} - P_{Rx\min[\mathrm{dBm}]} - L_{u[\mathrm{dB}]} - 32.4 - \log\left(f_{[\mathrm{MHz}]}\right) - L_{tt[\mathrm{dB}]} - L_{tm[\mathrm{dB}]}}{20}}, \tag{3.2}$$

being:

- $EIRP$: equivalent isotropic radiated power;
- $P_{Rx\min}$: receiver sensitivity;

For HSDPA, the number of users is limited by the number of HS-PDSCH codes and the available SFs. If a shared carrier is used, the maximum number of HS-PDSCH codes is 10, which corresponds to a maximum of 10 users. In this case, the maximum allowed throughput is 6.72 Mbit/s, and the percentage of the total power for traffic is an input that can be changed. For calculations, 30% of the total traffic power for HSDPA is used as a default value. For the dedicated carrier case, the maximum number of HS-PDSCH codes is 14, as 2 lower level codes are needed for signalling and control, and the maximum allowed throughput is 9.4 Mbit/s.

In order to obtain the corresponding users of each service, for HSDPA, two priority policies are defined:

- according to QoS, in which users are allocated according to their service correspondent QoS class, Table 3.2;
- proportional, in which the number of users is proportional to their original percentage.

Afterwards, the target throughput of each service is set, being used to obtain the average target value.

Table 3.2. QoS priority classes.

#priority	QoS class	Service
1	Streaming	Streaming
2	Interactive	FTP
3	Interactive	HTTP
4	Background	E-mail
5	Background	MMS

To evaluate the allowed throughput per user, so that all users can be served, the number of available codes per user must be calculated:

$$C_u = N_{MAX}^{HS} / N_u^{HS} , \tag{3.3}$$

where:

- C_u : number of HS-PDSCH codes per user;
- N_{MAX}^{HS} : maximum number of HS-PDSCH codes;
- N_u^{HS} : number of HSDPA users.

Table 3.3. Throughput per code and maximum number of HS-PDSCH codes.

Carrier	Throughput per code [Mbit/s]	Maximum number of HS-PDSCH codes
Shared	0.672	10
Dedicated	0.6714	14

The throughput per user is given by:

$$q_u = q_C \cdot C_u \tag{3.4}$$

where:

- § q_C : throughput per code.

When the target throughput is lower than q_u, the radius is calculated according to (3.1), and when it is higher, the target throughput of each service is reduced until all users can be served. The ratio for the reduction algorithm is:

$$k = {R_b^{tg}}/{q_C} , \tag{3.5}$$

where:

- § R_b^{tg} : target throughput;

the new effective throughput being:

$$R_b^{ef} = (k-1) q_C . \tag{3.6}$$

When the effective throughput is lower than q_u, all users can be served, and the radius is calculated according to (3.1).

For a better assessment of the network behaviour two new parameters are introduced. The number of served users ratio, h_{N_u}, and the HSDPA throughput efficiency, h_{HS}. The former is given by:

$$h_{N_u} = {N_u^{serv}}/{N_u} , \tag{3.7}$$

where:

- § N_u : number of users,
- § N_u^{serv} : number of served users;

and the latter is given by:

$$h_{HS} = {R_b^{ef}}/{R_b^{tg}} . \tag{3.8}$$

Other important parameter that is assessed is the global traffic, T , processed by the Node B:

$$T_{[\mathrm{MB/h}]} = \sum_{i=1}^{N_S} l_i \cdot N_i \cdot V_{i[\mathrm{MB}]} \ , \tag{3.9}$$

being:

- § l_i : number of calls per hour for service i;
- § N_i : number of served users for service i;
- § V_i : mean file volume for service i;
- § N_S : number of different services.

3.2 Scenarios

In order to calculate the cell radius, two scenarios are taken: one considers that all users are at a maximum distance of the BS, Figure 3.1(a), and the other considers that all users are placed in a line between the BS and the cell border, equally displaced with a ΔR gap from each other, Figure 3.1(b):

$$\Delta R = R_{\max} / N_u^{serv} \ , \tag{3.10}$$

where:

- § $R_{\max}$ - Maximum radius.

(a) Scenario 1 - with all users at the maximum distance.

(b) Scenario 2 - with all users equally displaced.

Figure 3.1. Scenarios for capacity calculation.

In Table 3.4, the percentage of users for each service profile is presented, [Voda07]. These values determine the number of users that is doing each service, being important to notice that the percentage of Video Telephony is very low, so that there are no users of this service in the Voice Centric profile, and there is only one user in the Data Centric one.

Table 3.4. Voice centric and data centric service profiles.

Service	Users [%]	
	Voice Centric	Data Centric
Voice	48.6	22.3
Video Telephony	0.2	0.3
Streaming	7.1	10.6
FTP	16.9	25.5
HTTP	11.8	17.7
E-mail	10.5	15.9
MMS	4.9	7.7

Table 3.5 shows the different target values of throughput of PS services considered for both scenarios for the HSDPA analysis.

Table 3.5. PS services target throughput, in HSDPA.

Service	Throughput [Mbit/s]				
	Scenario				
	1	2			
		A	B	C	D
Streaming	0.4	0.4	0.4	0.4	0.4
FTP	7.2	1.8	1.5	1.2	1.0
HTTP	1.8	1.0	0.8	0.6	0.5
E-mail	1.8	1.0	0.8	0.6	0.5
MMS	1.8	1.0	0.8	0.6	0.5

The CS services throughput is presented in Table 3.6.

Table 3.6. PS services target throughput.

Service	Throughput [kbit/s]
Voice	12.2
Video Telephony	64

In Table 3.7, the default values used for link budget assessment are presented, and in Table 3.8 the fast and slow fading margins are presented, as well as the indoor penetration one; these values are based on [EsPe06].

Table 3.7. Default values used in link budget.

Parameters		Value
Node B DL transmission power [dBm]		44.7
DL frequency [MHz]		[2110; 2170]
MT antenna gain [dBi]		0
Node B antenna gain [dBi]		17
Soft handover gain [dB]	Release 99	1.5
	HSDPA	Not considered
DL inter to intra cell interference ratio		[0.223; 0.65]
Load Factor (Release 99)		0.7
Cable losses between emitter and antenna [dB]		3
Losses due to user [dB]	Voice	3
	Data	1
Noise Factor [dB]		9
DL orthogonality factor		0.7
Percentage of power used for traffic		0.65

Table 3.8. Fading margins and indoor penetration values, [EsPe06].

Fading margins and Indoor penetration [dB]			
Environment	**Indoor**	**Pedestrian**	**Vehicular**
$M_{SF\ [\mathrm{dB}]}$	7.6	7.6	5.0
$M_{FF\ [dB]}$	2.0	2.0	0.0
$L_{int\ [dB]}$	20.0	0.0	8.0

The values of E_b/N_0 used in Release 99 are based on [CoLa06], and summarised in Table 3.9.

Table 3.9. E_b/N_0 values for the different services considered in Release 99, [CoLa06].

Switching	**Application Data Rate [kbit/s]**	E_b/N_0 **[dB]**		
		Environment		
		Indoor	Pedestrian	Vehicular
CS	12.2	7.7	7.7	8.0
CS	64	7.8	6.7	6.7
PS	64	6.6	6.6	7.3
PS	128	5.9	5.9	6.3
PS	384	6.5	6.7	6.5

Table 3.10 presents the values for activity factor for the different services in Release 99.

Table 3.10. Activity factor values for the different services considered in Release 99.

Service		Activity factor
CS	Voice	0.65
	Video Telephony	1.0
PS	HTTP	0.5
	Other services	1.0

Finally, Table 3.11 summarises the frequencies of the different used DL carriers, the first one being used as a shared carrier and the other used as dedicated one, [EsPe06].

Table 3.11. Corresponding frequencies of each carrier used for simulations.

Carrier	Frequency [MHz]
1	2112.5
2	2117.5

3.3 Analysis of results

In this section, results are presented. The objective is to understand network behaviour for an increasing number of users, when each of them is trying to make a different service. This assessment is made for the pedestrian environment and for two different values of the maximum Release 99 load factor, 0.7 and 0.8.

3.3.1 User distribution

Figure 3.2 shows the number of served users ratio, h_{N_u}, for the two service profiles and for the different load factor limits. For the Voice Centric profile, users are rejected only when the total number of users is more than 30, however, for the Data Centric one, when the number of users is higher than 22 the network does not have enough capacity to serve them all. This happens because, in the Data Centric profile, the percentage of PS services is higher than in the Voice Centric one, so, the number of HSDPA users increases more rapidly, and when the number of HSDPA users reaches the maximum, the users that cannot be served with HSDPA become Release 99 PS ones; when the allocation of SFs is done, there is a point when the number of SFs is not enough for Release 99 PS users, leading to users being rejected.

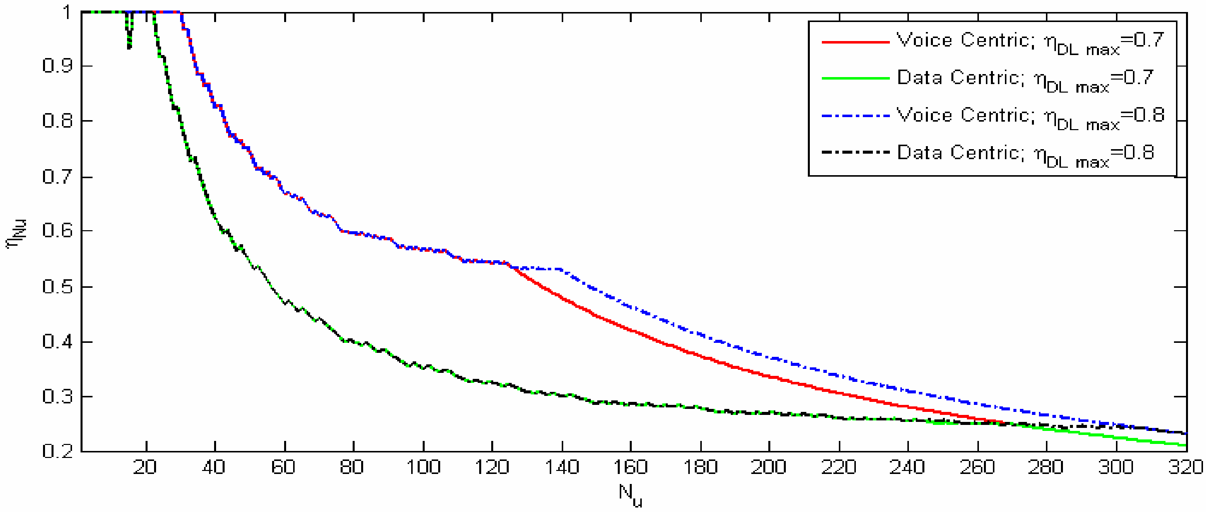


Figure 3.2. User efficiency, as a function of the number of users.

There are two limitations for the network, one being the available SFs, and the other the load factor for Release 99. For the Data Centric profile and for 15 users, one user is rejected not because of the SF but because of the load factor, i.e., there are 11 PS users, and as a shared carrier is in use, the maximum number of HSDPA users is 10, so 1 user becomes a Release 99 one; as there are no restrictions in the SF algorithm, this PS user uses the 384 kbit/s (PS) bearer, which increases the load factor to a value higher than the limit, so the PS user is rejected due to the load factor restriction. Users start being rejected due to the load factor for Voice Centric profile around a total of 130, and for Data Centric one around 268 for $h_{DL_{max}}$ = 0.7 and 303 for $h_{DL_{max}}$ = 0.8. These values can be seen in the following figures, which illustrate the user distribution for the priority policies.

Figure 3.3 shows the users distribution for the Voice Centric profile, when the load factor limit is 0.7, and when the HSDPA priority policy is done according to the QoS of each service.

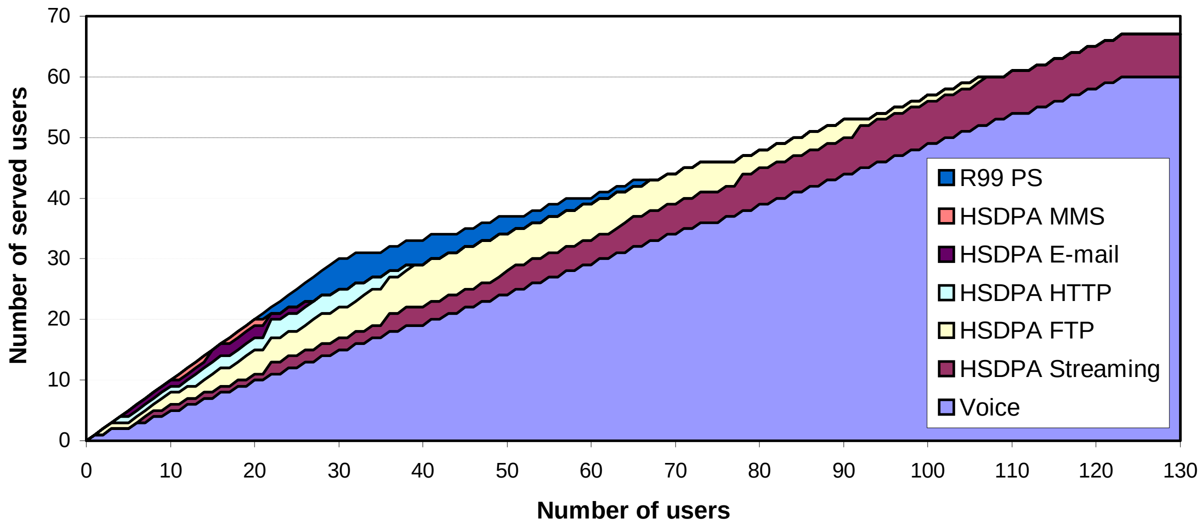


Figure 3.3. User distribution for Voice Centric profile and QoS priority policy.

Figure 3.4 presents this distribution also for the Voice Centric profile when the load factor limit is 0.7, but when the HSDPA priority policy is done proportionally to each service original percentage. The difference between Figure 3.3 and Figure 3.4 is the users' distribution in HSDPA: in the former, it is done according to the service QoS, based on Table 3.2, and as the number of users increases, the users with high priority services also increase and the low priority users become Release 99 PS.

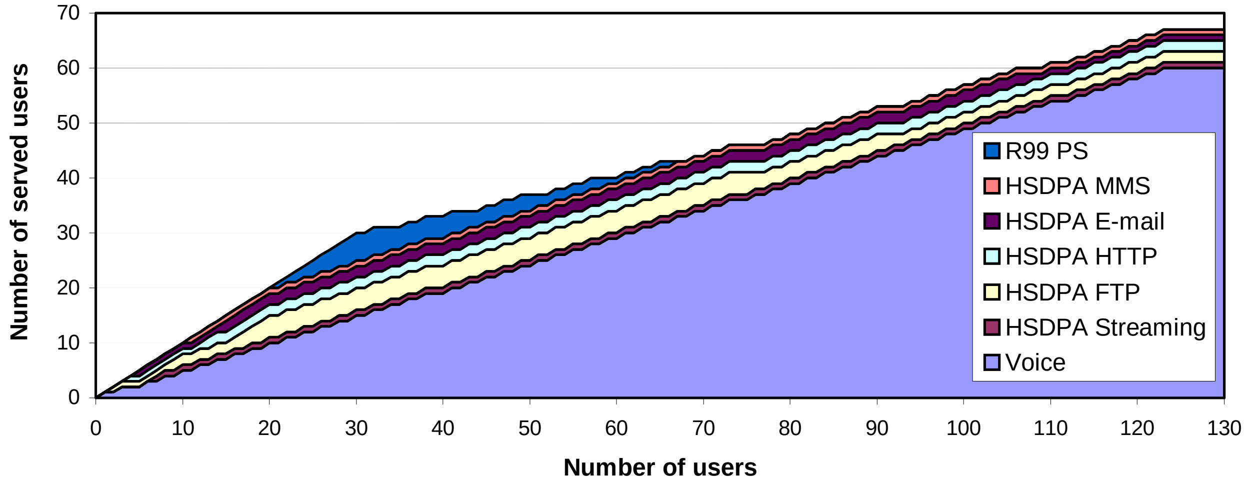


Figure 3.4. User distribution for Voice Centric profile and Proportional priority policy.

In Figure 3.4, users are distributed according to the percentages defined in Table 3.4. All users are served until 30, for the Voice Centric profile, and until 22 for the Data Centric one. When the maximum number of HSDPA users is attained, users with PS services become Release 99 ones or are rejected, depending on the available SFs. The saturation of the network is achieved when the network cannot support more users due to the load factor limitation: for the Voice Centric profile, this happens for 123 users, when the number of voice users is 60.

In Figure 3.5, the saturation of the network can be seen for all different profiles and load factor limits.

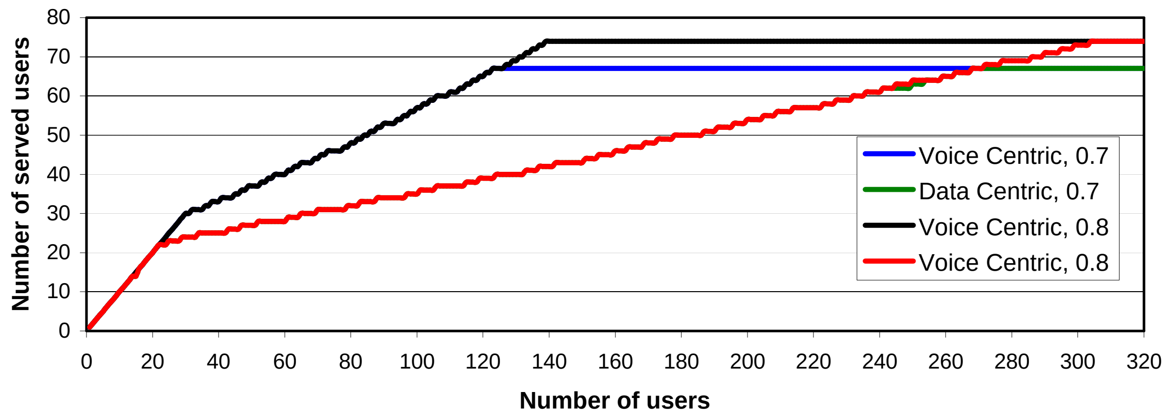


Figure 3.5. Number of served users for different profiles.

For the case when the maximum load factor is 0.8, the network supports 15% more users than with a maximum load factor of 0.7. The limitation due to the load factor happens when the number of users of the Voice Centric profile is more than the double of the number of users of the Data Centric one, because only Release 99 users contribute to the increase of the load factor, and, for a certain number of users, only Voice users are accepted for Release 99 (due to the priorities in Table 3.1); comparing the values in Table 3.4, for the Data Centric profile, voice users percentage is more than half the value of the Voice Centric one, thus, one can say that voice users are the main limitative factor of the network capacity.

3.3.2 Release 99

Figure 3.6 shows the throughput for Release 99. The peaks in both curves are due to the sudden appearance of PS users in Release 99.

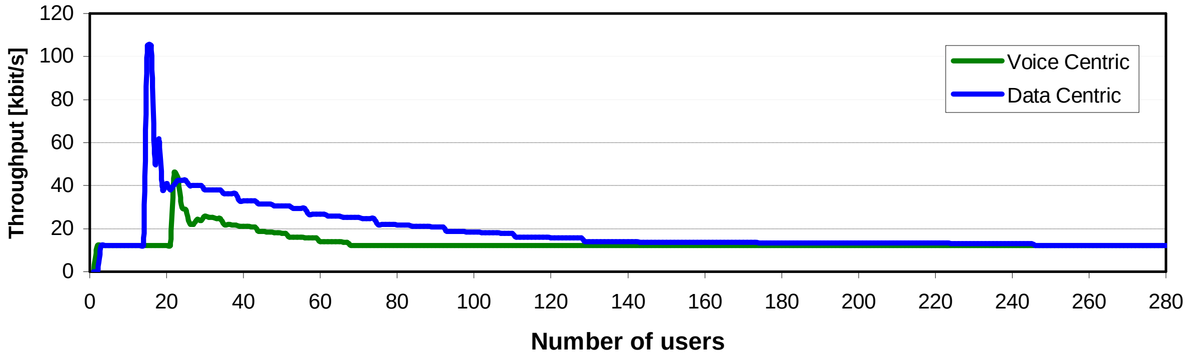


Figure 3.6. Average throughput for Release 99 users.

PS users' throughput is as high as the available SFs enables. Thus, as the number of PS users increases, the SF allocation algorithm imposes a lower throughput for PS users, which cause a decrease of the global throughput until all users are voice users and the throughput stabilises at 12.2 kbit/s.

Figure 3.7 presents the variation of the load factor with the number of users and with the interference for Voice Centric profile and Proportional priority policy.

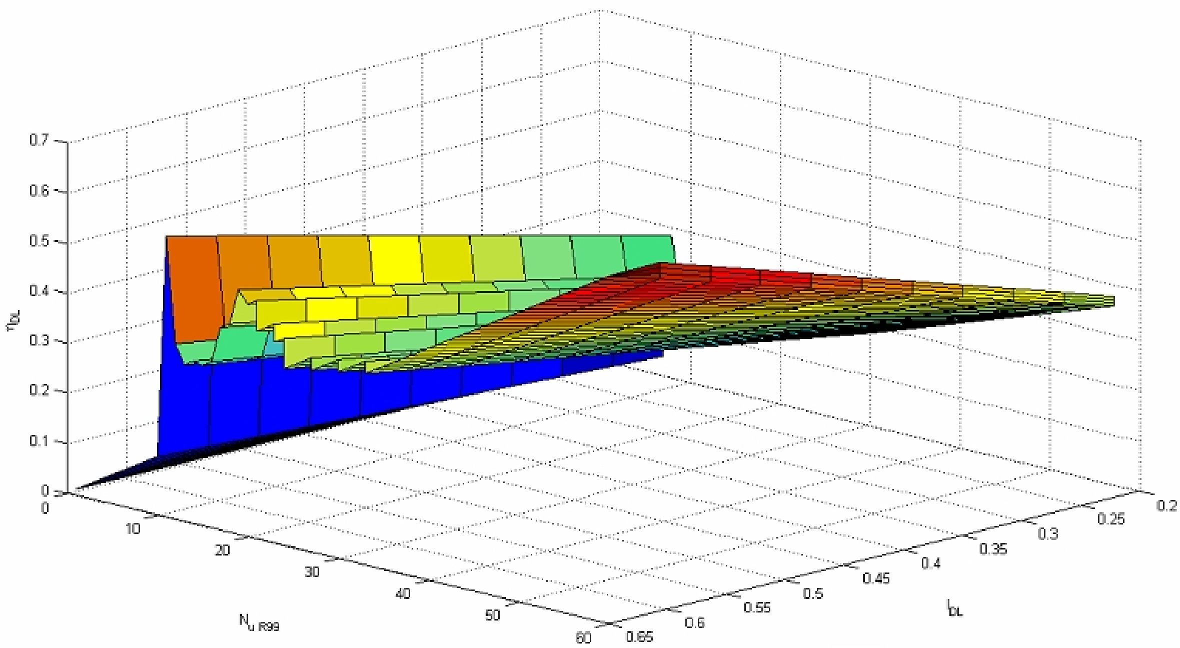


Figure 3.7. Load factor for Voice Centric profile and Proportional priority policy.

Although the priority policy accounts only for HSDPA users, it is important to be considered in this analysis, due to the number of HTTP users, which have a different value of activity factor compared to other PS users. So, concerning the priority policy used in HSDPA, the number of HTTP users in Release 99 is different, which influences the load factor and the radius in Release 99. The load factor increases proportionally to DL interference, but in the following figures only the worst case is analysed, results being presented only for the maximum value of interference, 0.65. The values of load factor of the different profiles are compared in Figure 3.8 and 3.9.

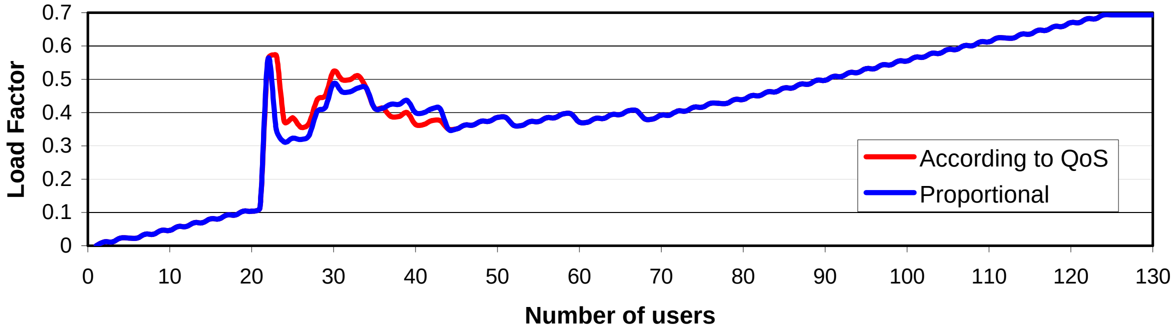


Figure 3.8. Load Factor for Voice Centric profile.

For the Proportional priority policy, the number of HTTP users is constant, however, for the QoS priority policy, the number of HTTP users increases first and then decreases, which causes the variation in the values of the load factor for the two priority policies. These variations can also be seen in the values for the cell radius. The drops that can be seen at around 50, 60, and 70 users for the Voice Centric profile, and at around 50, 70, 90, 110, and 130 users for the Data Centric one happens when one more PS Release 99 user cannot be served and is rejected.

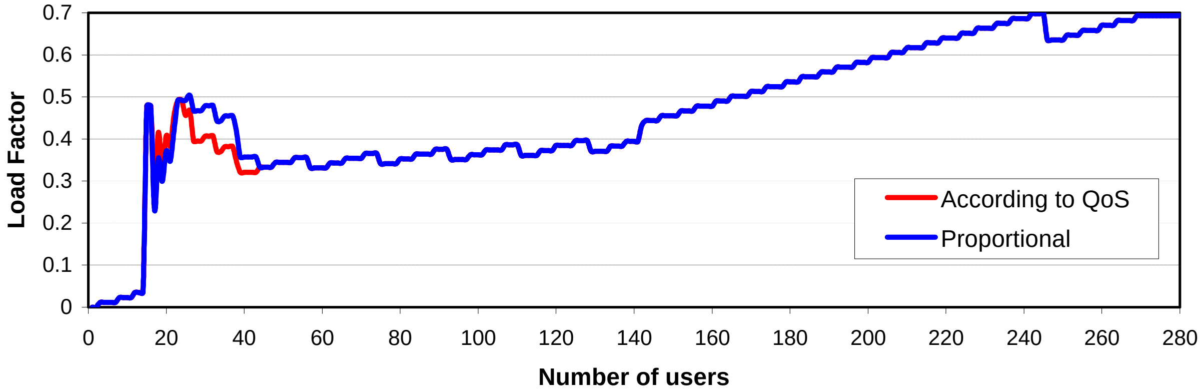
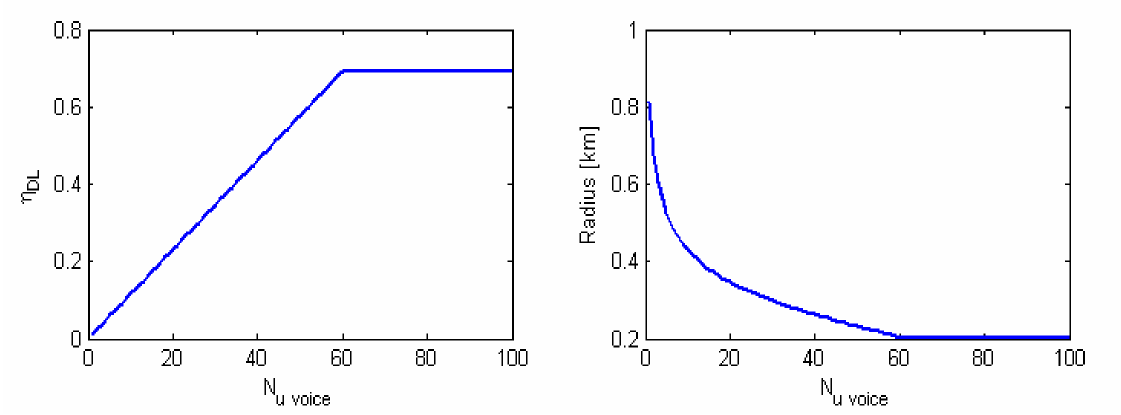


Figure 3.9. Load Factor for Data Centric profile.

The load factor for the Data Centric profile declines rapidly, because Video Telephony users are rejected due to the load factor limitation, and then starts increasing again only with the voice users contribution.

To perform a better analysis, Figure 3.10 presents the load factor and the cell radius for mono service, all users performing the Voice service. The load factor increases proportionally to the growth of the number of users, until it becomes constant for approximately 60 users, which means that saturation is achieved.

In Figure 3.10, the load factor limitation effect on the network capacity is clear. The same effect can be observed for the cell radius, when all users are performing the Voice service. For other Release 99 services, the behaviour is identical to the voice service.



(a) Load factor (b) Cell radius

Figure 3.10. Characteristics for Voice single service.

Figure 3.11 represents the cell radius for Release 99 for the different profiles and priority policies. Similarly to the load factor behaviour, the differences in the cell radius between the two priority policies around 20 to 40 users are due to the variation in the number of HTTP users. The peaks in both profiles that can be seen between 70 and 140 users are due to the decrease in the number of PS Release 99 users.

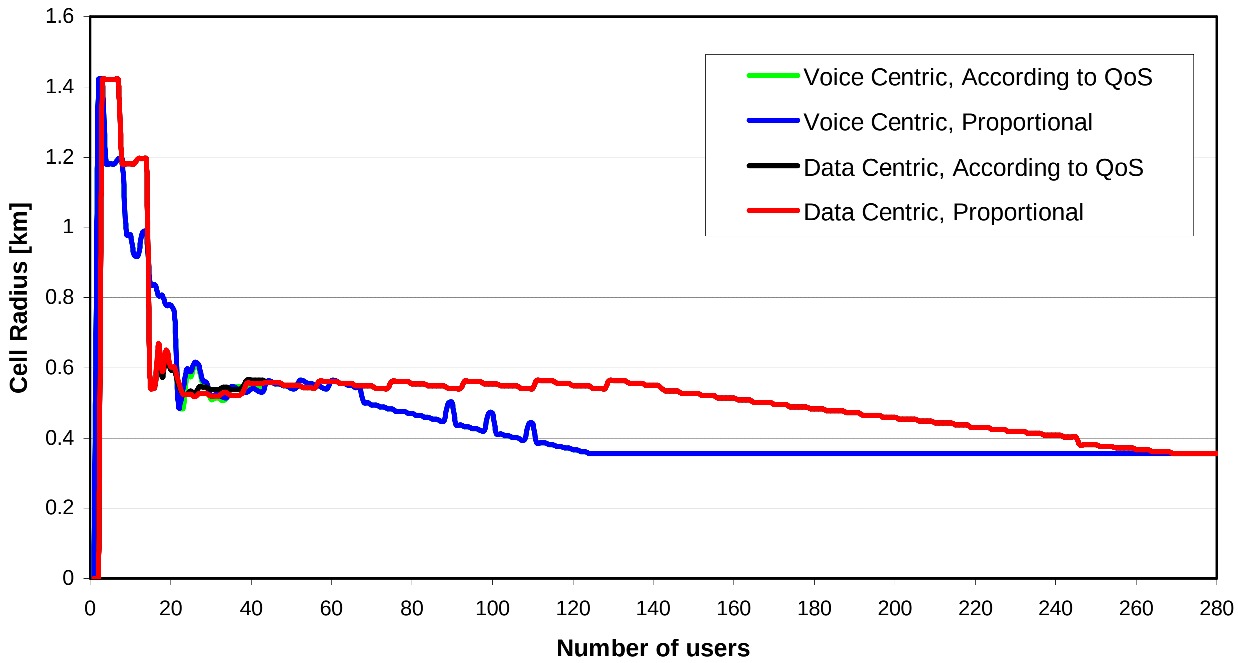


Figure 3.11. Cell radius for Release 99.

The environment where users are placed is also important for the radius analysis. Figure 3.12 illustrates the dependency of the radius with the three different environments: pedestrian, indoor and vehicular.

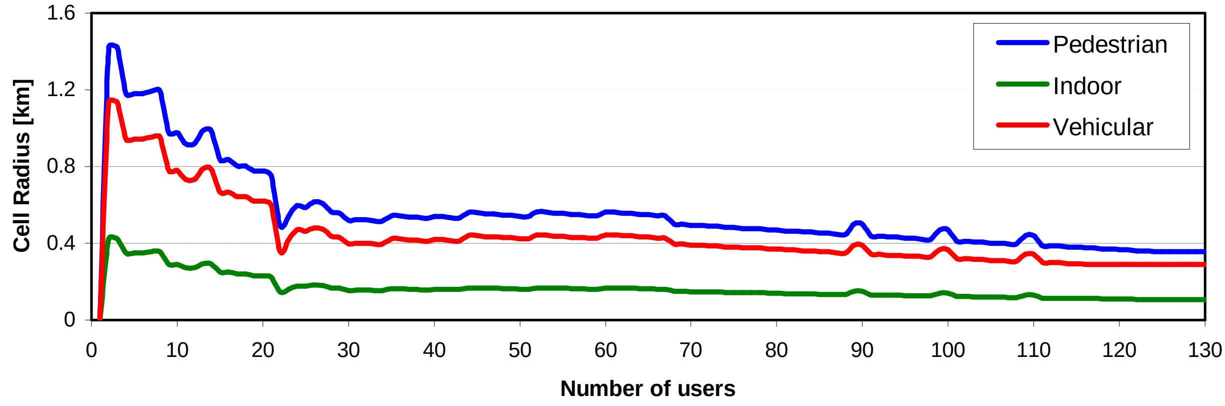


Figure 3.12. Cell radius for different environments in Release 99, for Voice Centric profile.

The variation in this curves is very similar, the main difference in the values is caused mainly by the attenuation, given the high values of indoor penetration for vehicular and especially indoor environments. It is important to mention that the maximum cell radius for the indoor environment is, for some cases, around three times smaller than for the pedestrian one, which can be a very limitative factor, and has to be taken in account when making cell planning.

3.3.3 HSDPA Shared Carrier

For the first scenario, the goal is to serve all users, even if in order to achieve this goal the throughput has to be sacrificed. So, in order to serve all users in HSDPA, the throughput per user is decreased until none of the users is rejected.

In Figure 3.13, two throughput parameters are analysed, the target throughput and the effective one. The target throughput per user is summarised in Table 3.5, it being the mean value of all users and representing the throughput that users would like to be served with. The effective throughput is the highest value with which users can be served in order not to reject any user.

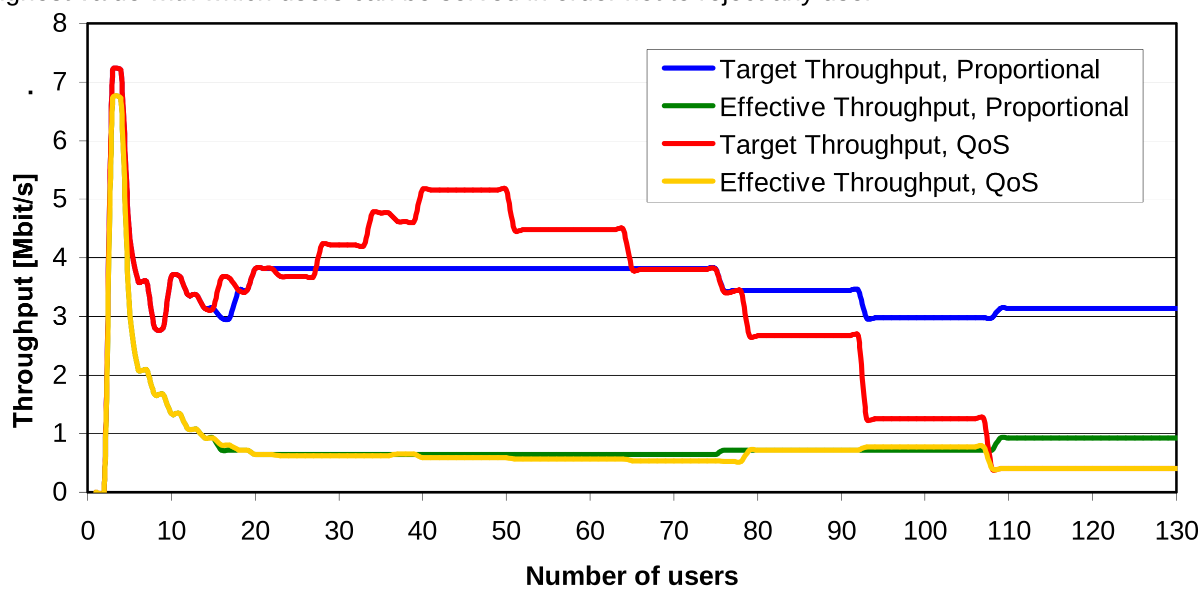


Figure 3.13. Target and effective throughput for both priority policies, and for the Voice Centric profile.

Figure 3.14 presents the throughput efficiency (3.8) for the Voice Centric profile and for the two priority policies. The variation is higher in the QoS algorithm than in the proportional one. For a high number of users trying to establish a connection, the number of Streaming ones increases, and as Streaming is the service with highest priority, and its target throughput is lower than the approximation of the throughput per code, when all HSDPA users are Streaming, the target throughput is equal to the effective throughput for the QoS algorithm. However, this does not happen for the Proportional algorithm, because the number of users of each service is proportionally distributed, which is the reason why the effective throughput becomes constant with the increase of the number of users from a certain point on, and the throughput efficiency variation is not significant.

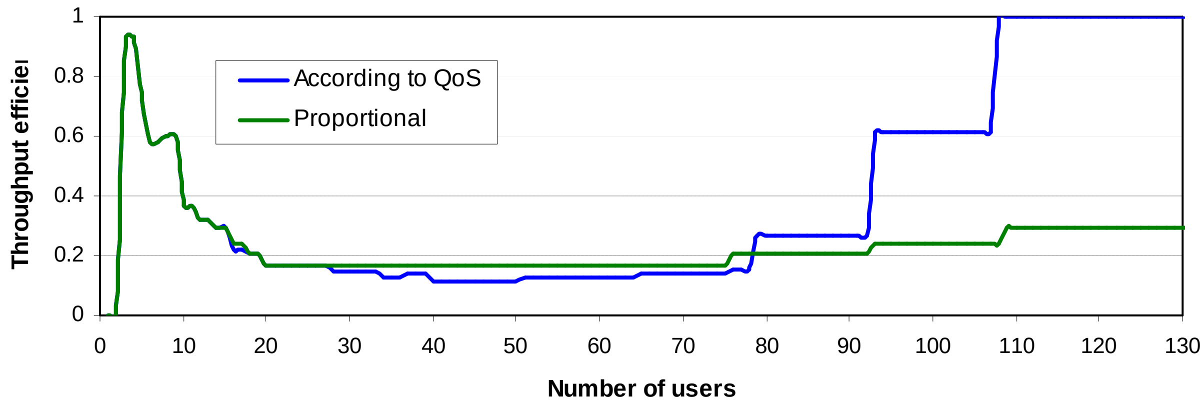


Figure 3.14. Throughput efficiency for Voice Centric profile.

Figure 3.15 presents the maximum cell radius for scenario 1 for the two different profiles and priority policies.

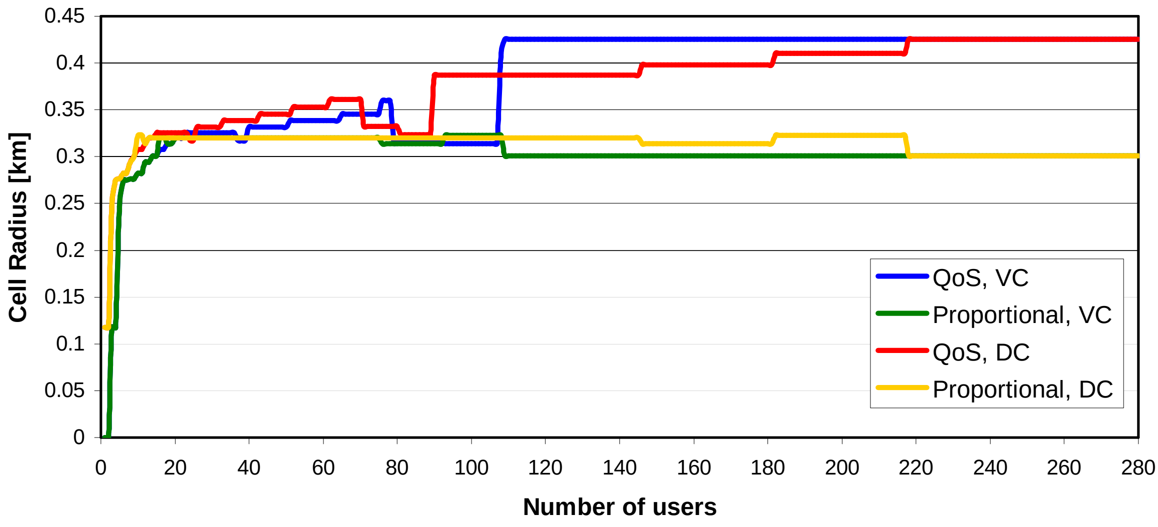


Figure 3.15. Maximum cell radius as a function of the number of users.

The increase of the cell radius is expected, as the throughput decreases, which leads to a decrease in the corresponding SINR, thus, to a lower attenuation. For the QoS algorithm, there is a decrease around 90 users due to the increase of the number of Streaming users and decrease of FTP ones, which leads to an increase of the throughput, becoming constant when all FTP users are rejected, thus, inverting the radius curve and making it increase again.

Figure 3.16 illustrates the dependency of the cell radius on the three different environments (pedestrian, indoor and vehicular), and in Figure 3.17, the difference of the maximum cell radius between pedestrian and indoor environments is presented.

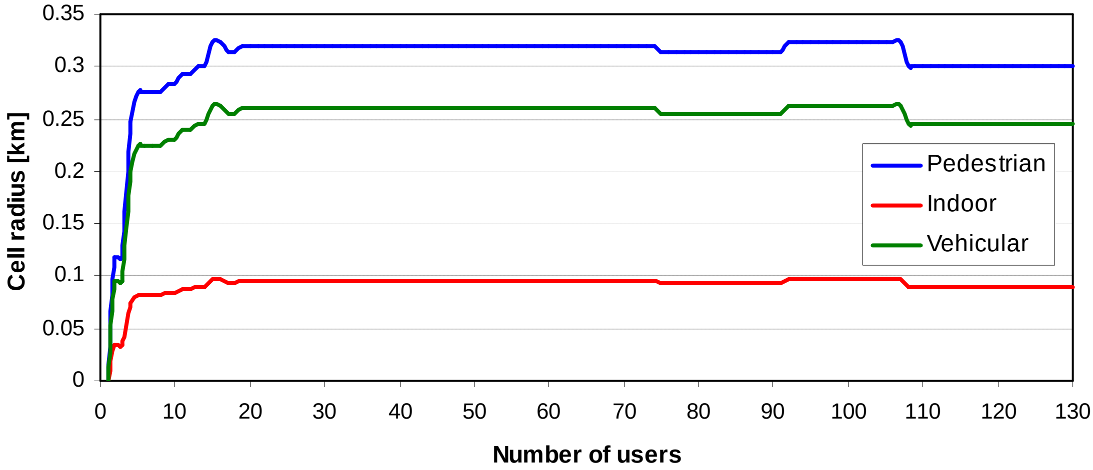


Figure 3.16. Cell radius for different environments in HSDPA, for Voice Centric profile.

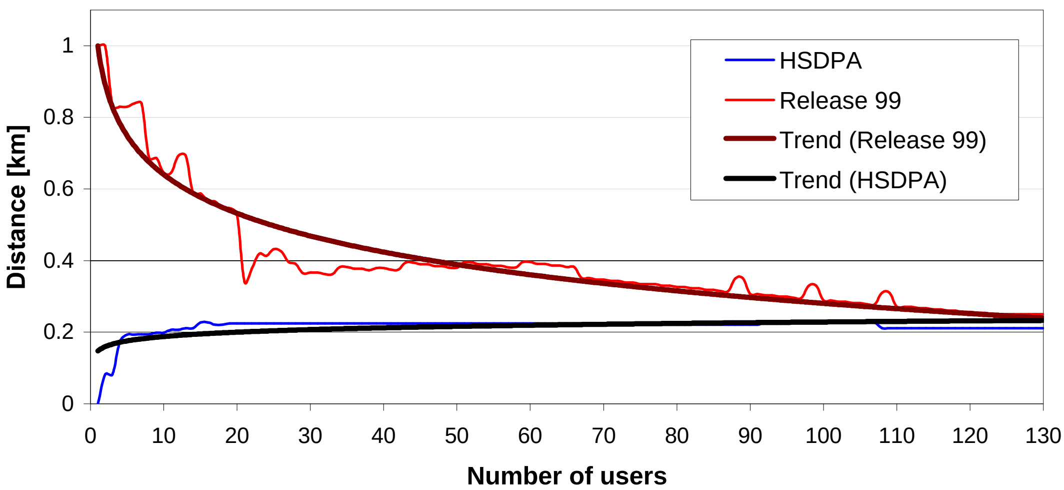


Figure 3.17. Difference between the cell radius for pedestrian and indoor environment.

Trend lines were obtained for the curves in Figure 3.17 with a logarithmic approximation, automatically given by Excel, the Release 99 trend being given by

$$d_{[\mathrm{km}]} = -0.1562\, ln\left(N_u\right) + 1.0001\,, \tag{3.11}$$

where:

- § d : distance that corresponds to the difference between the cell radius for the pedestrian and indoor environments,

with a regression coefficient of R^2 = 0.9108; and for HSDPA the trend is given by:

$$d_{[\mathrm{km}]} = 0.0176\, ln\left(N_u\right) + 0.1473\,, \tag{3.12}$$

with a regression coefficient of R^2 = 0.375.

For HSDPA the variation is almost negligible, and there is a reduction of approximately 200 m from the pedestrian environment to the indoor one. However, for Release 99, when there is a low number of users, thus, a higher variety of services using Release 99, there is a huge difference in the distances between the environments, which tends to decrease as long as the number of users increases and only Voice users are served by Release 99.

3.3.4 HSDPA Dedicated Carrier

For the case of using a dedicated carrier, the frequency for Carrier 2 is need, Table 3.11. The distribution of users can be seen in Figures 3.18 and 3.19, where the difference between the two priority policy algorithms is notorious.

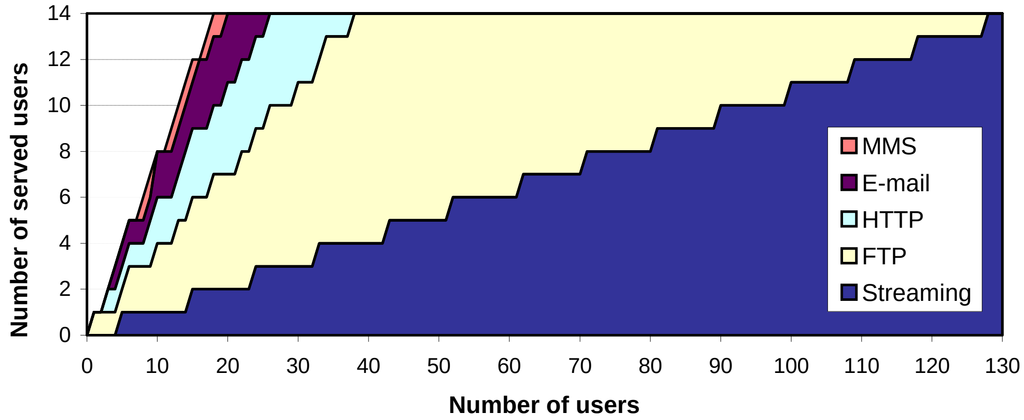


Figure 3.18. User distribution for Data Centric profile and QoS priority policy.

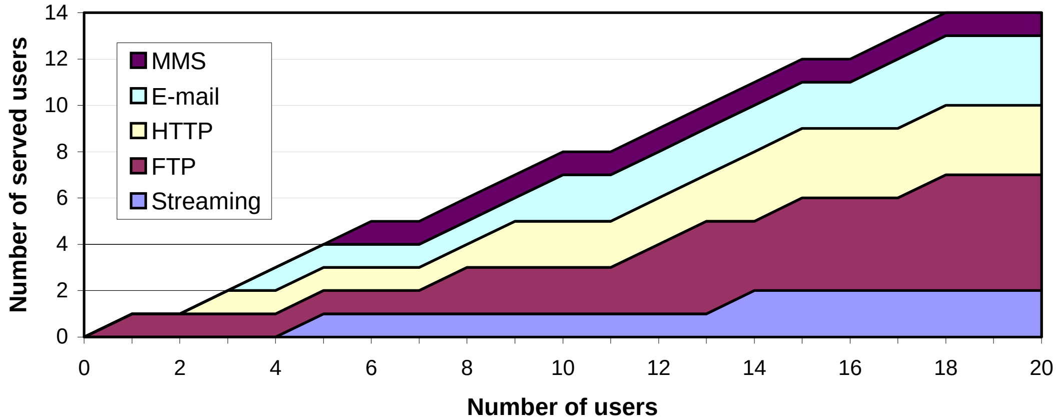


Figure 3.19. User distribution for Data Centric profile and Proportional priority policy.

As for the shared carrier case, Figures 3.20 and 3.21 present the two throughput parameters. The behaviour of the throughput for the two priority policies is very similar to the shared carrier case. The slopes in the target throughput for the QoS priority policy correspond to the different distributions of users by the services depicted in Figure 3.20. The same effect is visible for the Proportional priority policy, in which case, the throughput becomes constant after 18 users, which is the point when saturation is reached

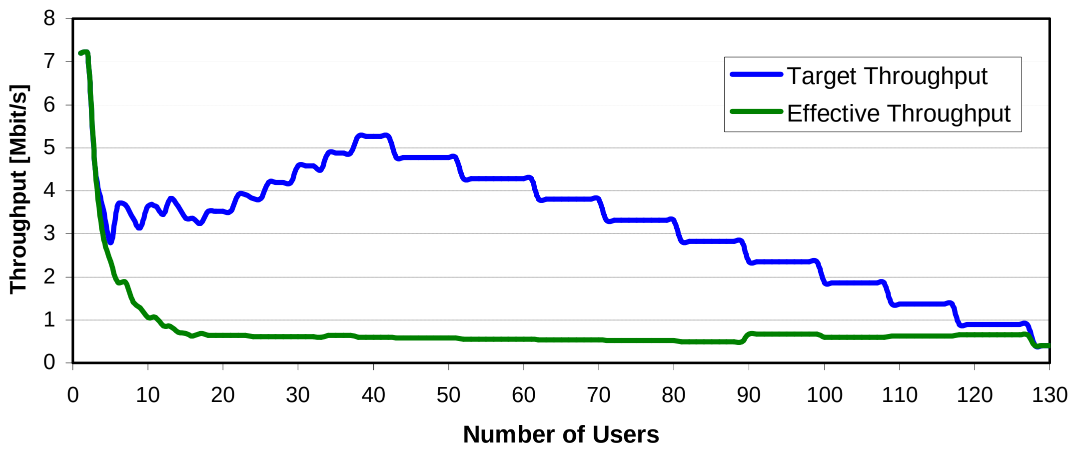


Figure 3.20. Target and effective throughput profile and for QoS priority policy.

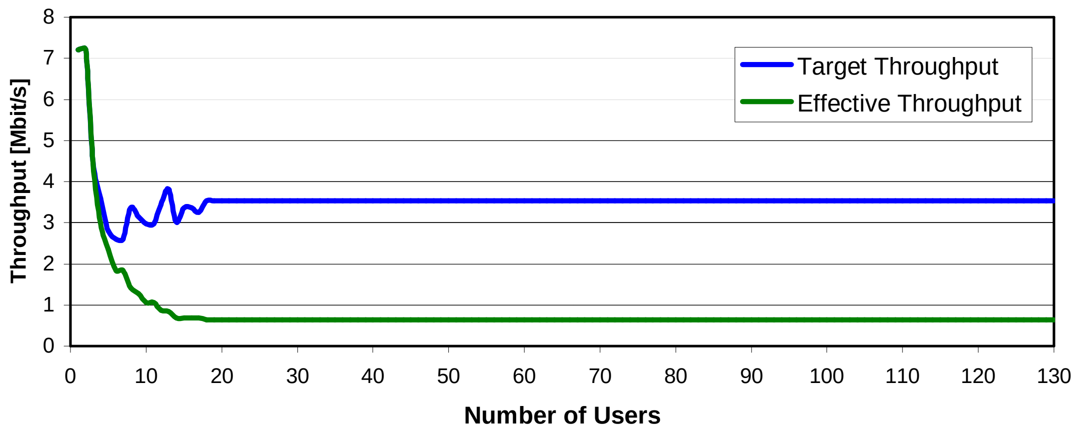


Figure 3.21. Target and effective throughput for Proportional priority policy.

The behaviour of the throughput efficiency (3.8) for the dedicated carrier is very similar to Figure 3.16, and is not presented here.

Figure 3.22 presents the maximum cell radius for the Data Centric profile and for the two different priority policies. Initially, as the number of served users increases, the growth of the cell radius is fast, due to the decrease in the throughput, which corresponds to lower values of SINR, and a smaller value of path loss.

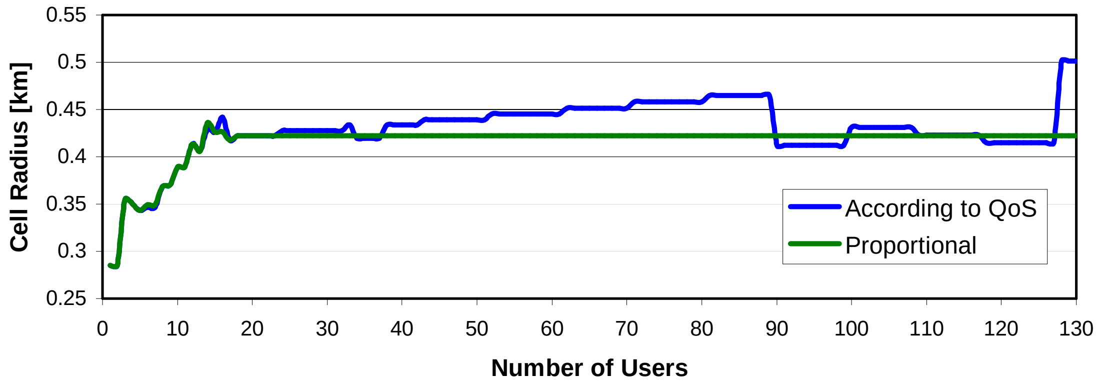


Figure 3.22. Maximum cell radius for Data Centric profile.

As the attenuation decreases, the maximum distance where an MT can establish a connection increases. This can be seen by comparing the slopes from the cell radius and effective throughput curves for the QoS priority policy algorithm. Figure 3.23 illustrates the dependency of the cell radius for Proportional algorithm with the three different environments: pedestrian, indoor and vehicular. The difference in the cell radius for the three environments is higher in the dedicated carrier case than in the shared carrier one. This difference is around 100 m for all environments.

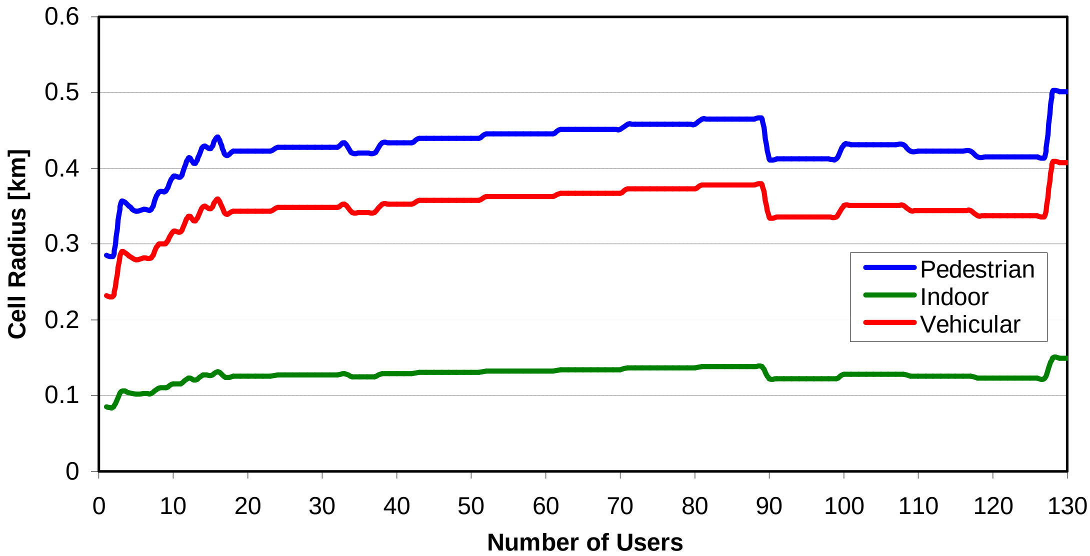


Figure 3.23. Cell radius for different environments in HSDPA, for Data Centric profile.

It is important to mention that a maximum of 14 users are served in the dedicated carrier case, all the resources of the network being distributed among them, so it would be expected that the results, both in terms of throughput and cell radius, would be better for this case. Concerning the case of the shared carrier, although the number of served users is higher, by taking into account both Release 99 and HSDPA, the resources available for HSDPA are only 30% of the ones available for Release 99.

3.3.5 Scenario 2

The goal of this second scenario is to evaluate the maximum distance when users are displaced within the cell. Two approaches are taken for evaluation: worst and best cases. It is considered that users doing the same service are grouped, and for the former case, users with smaller values of throughput are put closer to the BS and users with the highest throughput are put further away to the BS; for the latter the opposite happens, i.e., users with the highest throughput are closer to the BS and users with the smaller throughput are further away to the BS. The values of throughput are presented in Table 3.5.

The purpose is to serve all users with their target throughput, hence, calculations are done for one user at a time, and the distance ΔR, (3.8), grows slowly until all users can be served. Table 3.12 presents the maximum cell radius for given distributions of HSDPA users.

Table 3.12. Maximum cell radius for Scenario 2.

Number of users						Maximum Cell Radius [km]								
						Scenarios								
						1	2							
HSDPA	Services						A		B		C		D	
	S	F	H	E	M		W.C.	B.C.	W.C.	B.C.	W.C.	B.C.	W.C.	B.C.
1	0	1	0	0	0	0.117	1.355	1.355	1.467	1.467	1.602	1.602	1.711	1.711
2	0	1	1	0	0	0.247	0.778	1.074	0.868	1.188	0.982	1.334	1.074	1.424
3	0	1	1	1	0	0.275	0.525	0.780	0.603	0.882	0.699	1.014	0.780	1.098
4	1	1	1	1	0	0.276	0.204	0.808	0.404	0.928	0.532	0.988	0.604	0.988
5	1	2	1	1	0	0.283	-	0.125	0.155	0.390	0.380	0.845	0.485	0.845
5	1	1	1	1	1	0.283	-	0.250	0.155	0.710	0.380	0.845	0.485	0.752
6	1	2	1	1	1	0.294	-	-	-	0.138	0.186	0.558	0.366	0.732
7	1	2	2	1	1	0.301	-	-	-	-	-	0.252	0.203	0.644
8	1	2	2	2	1	0.323	-	-	-	-	-	0.112	-	0.376
8	1	3	2	2	0	0.308	-	-	-	-	-	-	-	0.248
9	1	3	2	2	1	0.314	-	-	-	-	-	-	-	0.126

S: Streaming: **F**: FTP; **H**: HTTP; **E**: E-mail; **M**: MMS; **W.C.**: Worst Case; **B.C.**: Best Case.

There is a significant difference between the Best and Worst Cases on almost all scenarios. As the number of users increases, there is a point when the cell radius obtained in Scenario 2 is lower than the one obtained in Scenario 1, due to the fact that throughput is much lower in Scenario 1 than in Scenario 2. The results lower to 100 m are not shown, as it is considered that they are not relevant enough to this analysis.

3.4 Global Traffic

In order to evaluate the global behaviour of the network, the global traffic that is processed by the Node B was calculated, (3.9). For Voice applications, calculations were based on the mean duration of a call, and for the remaining services, calculations were based on the mean volume of each file and on the number of users that are served by the network.

Two different scenarios are analysed, a heavy scenario, where applications are mostly data based, and a light one, which is mostly based on Voice traffic. Table 3.13 presents the number of “calls per hour” of each application for both scenarios.

Table 3.13. Number of calls per hour for each scenario.

Services	l	
	Light	Heavy
Voice	6	6
Streaming	2	4
FTP	3	12
HTTP	4	12
E-mail	4	12
MMS	2	12

In Table 3.14, the values used for mean call duration and mean file volume corresponding to each service are presented.

Table 3.14. Mean call duration and mean file volumes.

Service	Parameter	Value	
		Light	Heavy
VoIP	Mean Call Duration [s]	120	120
Streaming	Mean File Volume [kB]	17500	17500
FTP	Mean File Volume [kB]	2550	2550
HTTP	Mean Page Volume [kB]	34.4	34.4
E-mail	Mean File Volume [kB]	100	100
MMS	Mean File Volume [kB]	165	165

In order to obtain a realistic view of the network behaviour, this analysis is made for the cases where the percentage of rejected users is around 5%. Nowadays, the acceptable blockage probability is around 1%, so that certain limits of quality are maintained.

Figure 3.24 presents the percentage of rejected users for Voice and Data Centric profiles as a function of the total number of users. So, the Voice Centric profile is analysed for the maximum of 33 users, and the Data Centric one for 24 users.

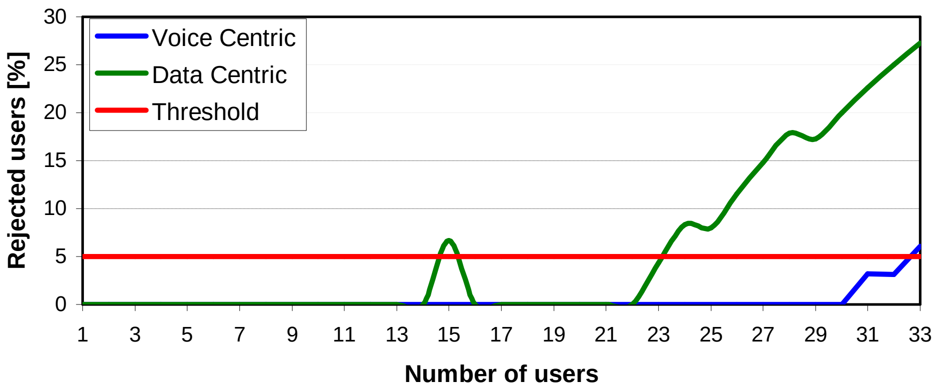


Figure 3.24. Percentage of rejected users.

Figures 3.25 and 3.26 present the total traffic processed by the Node B for the Voice and Data Centric profiles, for the Heavy and Light scenarios, and for QoS and Proportional priority policy algorithms. The differences between the curves for the priority policies are due to the number of Streaming users, which have a big weight as the mean file size is very high compared to the other services; the difference is higher for the QoS algorithm than for the Proportional one, and it is very notorious especially for the Data Centric profile, where the difference between the number of users is higher.

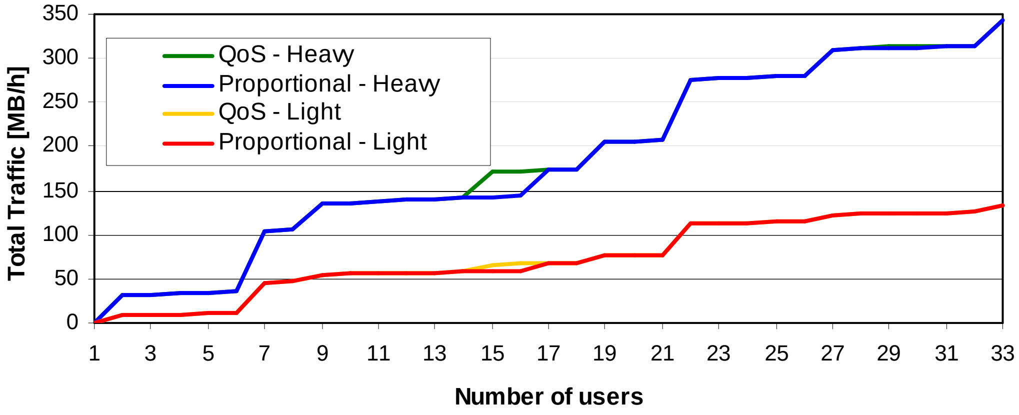


Figure 3.25. Global traffic processed by the Node B for Voice Centric profile.

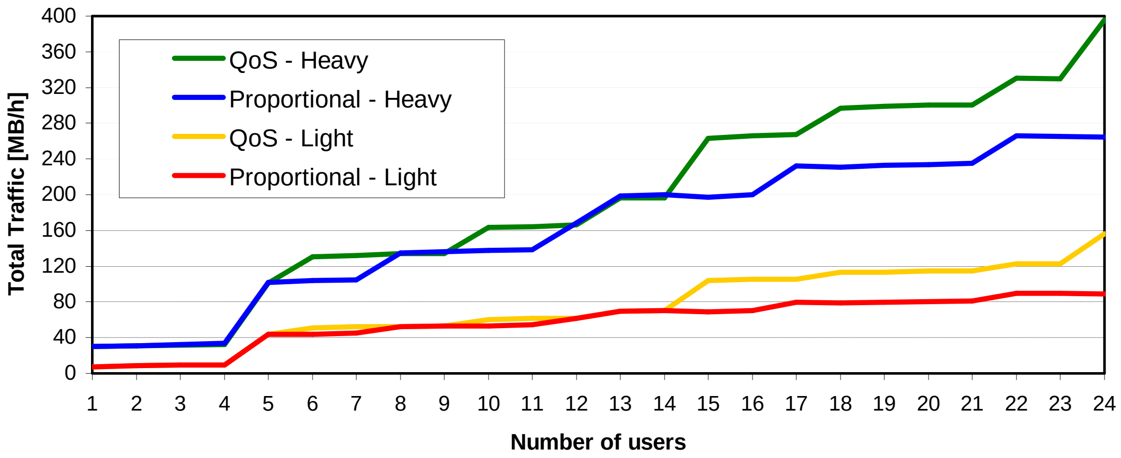


Figure 3.26. Global traffic processed by the Node B for Data Centric profile

The traffic for the Light scenario is almost 3 times lower than for the Heavy one. These are theoretical values, and do not account for retransmissions, which is a reality in practice, and causes a decrease in throughput. The global traffic was calculated not taking into account the throughput for each service, but only the number of users supported by the Node B and the mean volume of each file or the mean call duration.

Chapter 4

Simulator

This chapter gives an overview of the OPNET simulator that will be used during this work, focusing on the UMTS models aspects. All the chosen options for simulations are described and the obtained results are analysed. Finally, a comparison between the results obtained with OPNET and the results from the theoretical model is made.

4.1 OPNET Overview

OPNET Modeler is a modelling and simulation tool for the analysis of communication networks, [OPNE07]. It enables the use of different network configurations, allowing the design and study of network devices, protocols and applications. Some of the OPNET Modeler features, which makes it an attractive environment for network performance evaluation, are the already implemented models, which includes many protocols and device models, an object-oriented modelling, and the possibility of changing the implemented models and their corresponding parameters, due to the open source code feature. Another characteristic is the Graphical User Interface (GUI), which allows an intuitive design of the system under analysis.

OPNET Modeler supports four simulation technologies:

- Discrete Event Simulation (DES), being the best option when the aim is to analyse the dynamic behaviour of the network;
- Flow Analysis, for evaluating the network in a steady state, based on analytical techniques and algorithms;
- Hybrid Simulation, based on the combination of two modelling techniques, the analytical and discrete event ones;
- ACE QuickPredict, for the assessment of application response times, being based on analytical techniques.

For the aim of this work, DES is the most appropriate one, as it enables dynamic simulations. This technique provides highly reliable results, even though the simulation runtime is higher than other techniques, and allows the study of parameters such as dynamic protocol behaviour, network convergence times, and application performance. It also makes possible the analysis of the effect of different network environments on the behaviour of these parameters.

OPNET Modeler is based on different editors hierarchically distributed; its workflow is centred on the Project Editor, Figure 4.1, in which the topology of the network is represented. This editor enables creating a network model, choosing statistics from each network object (Node Statistics) or from the network as a whole (Global Statistics), running simulations, and viewing and analysing the obtained results. Network objects can be node or link objects that are configurable through the Node Editor. The first step when creating a new network model is to create a new project, which can contain one or more scenarios, each of them exploring a different aspect of the network.

After creating the network, profiles and applications can be configured. Applications can be defined, or common applications already defined can be used, such as database access, e-mail, file-transfer, file print, Telnet session, video conferencing, VoIP call, and web browsing. A profile is associated to workstations and servers. In the profile, user applications are defined, and for each profile one or more applications are associated to, and classified as heavy or light. For example, when defining an Engineering profile, the associated applications can be light use of e-mail and heavy use of file transfer. However, there are cases when there is no need to define profiles or applications, as services and applications can be chosen from within the QoS classes available for workstations from certain types of network environments.

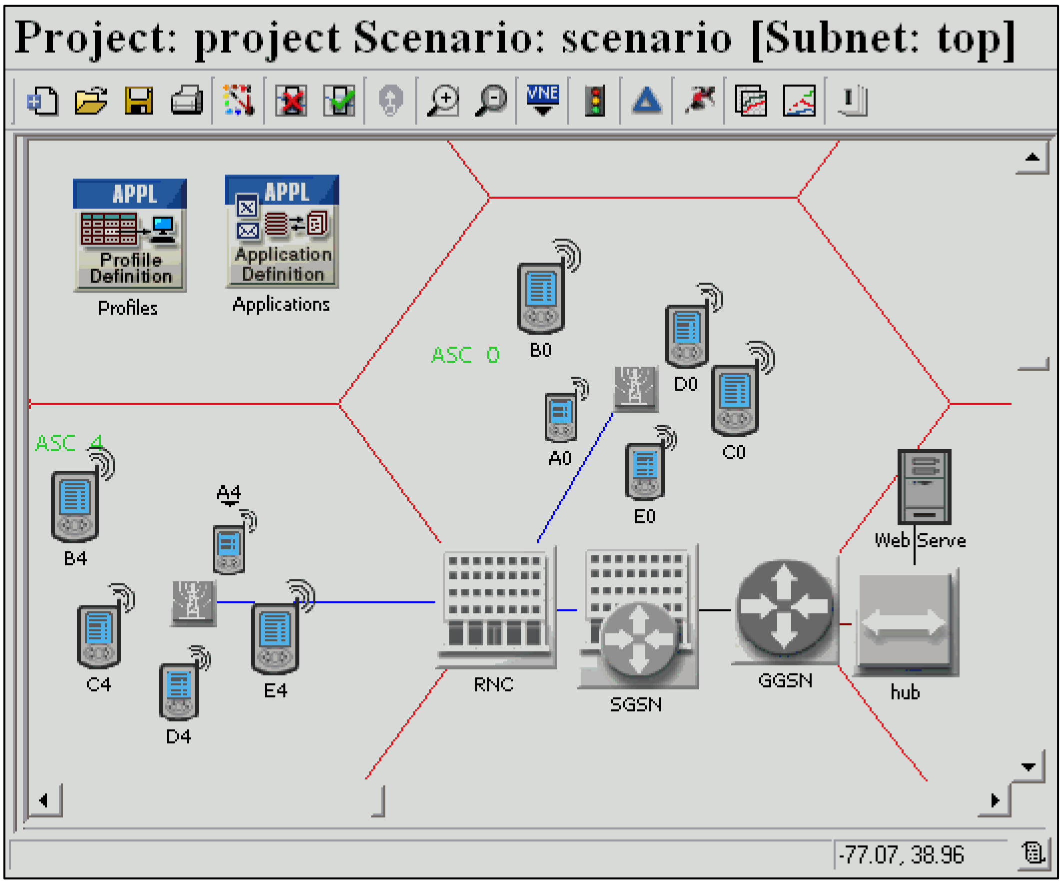


Figure 4.1. Example of a scenario using the Project Editor.

In the Node Editor, the architecture of the network device is based on the flow of data among functional elements, called modules, Figure 4.2. Each module represents applications, protocol layers, algorithms or physical resources, being connected to each other through packet streams. This editor enables defining the behaviour of each module with the help of the Process Editor.

The Process Editor creates the process models, these process models being represented by Finite State Machines (FSMs) to support protocols, resources, applications, algorithms and queuing policies details, Figure 4.3. Process models are comprised of states and transitions, and all the operations performed in each state or transition are described in C or C++ code blocks. This method enables the creation of new process models, or editing the already implemented ones, hence, adapting these models to network needs.

Once the whole network is configured, and all the statistics are chosen, simulations can be run to obtain the results. With DES, it is possible to choose one of two kernels to run simulations, development or optimisation, simulation times being lower for the latter.

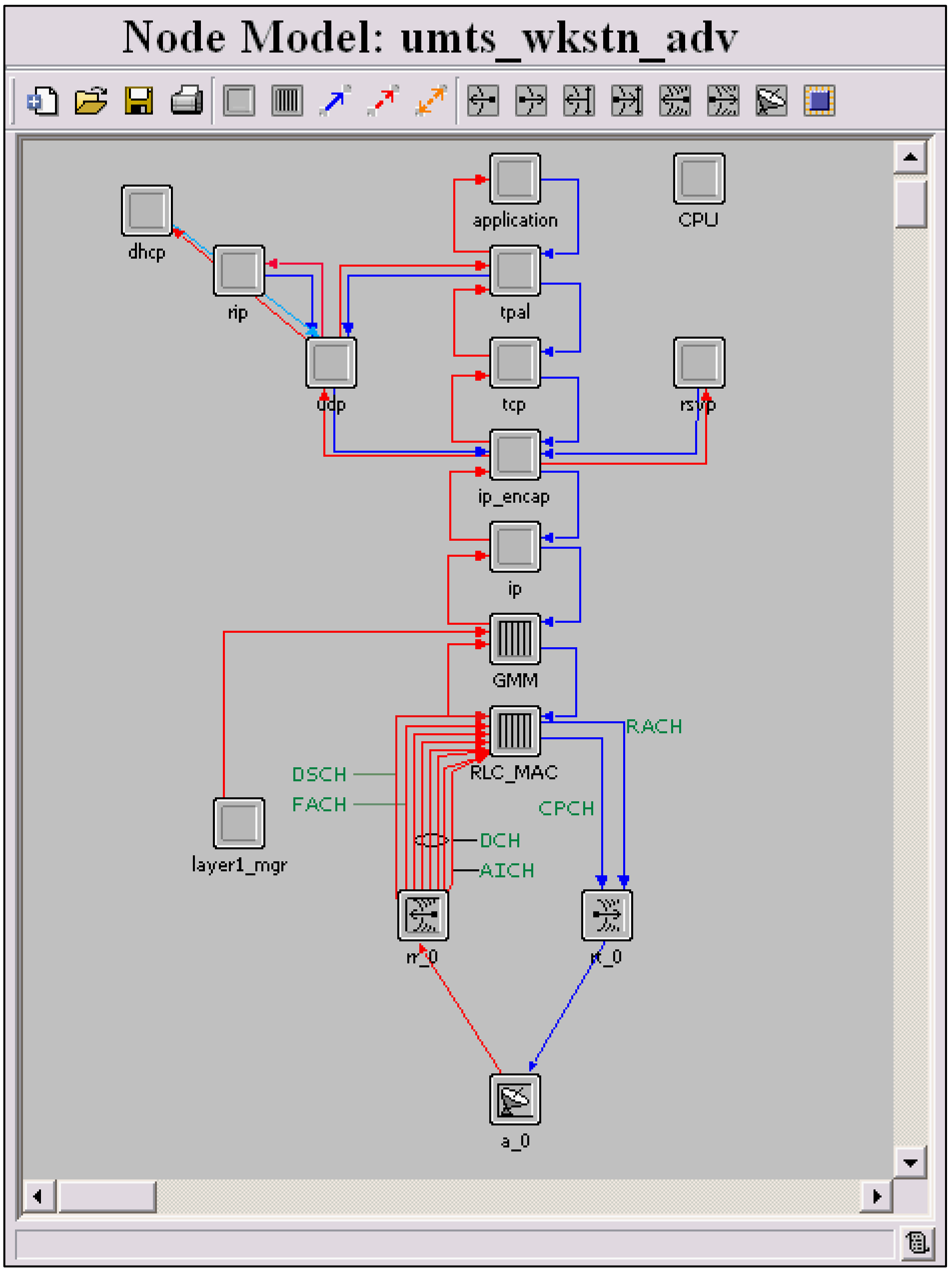


Figure 4.2. Example of UMTS workstation architecture in the Node Editor.

4.2 UMTS

OPNET Modeler has the UMTS module already implemented, [OPNE07], being based on 3GPP Release 99 and supporting only PS traffic. It enables the evaluation of an UMTS network through parameters as end-to-end QoS, throughput, drop rate, end-to-end delay, and delay jitter for the RAN and CN, including the possibility of combining different services and applications for given QoS requirements. Many statistics are available for assessing network performance, which can be associated to network nodes or can be global statistics associated to the network as a whole.

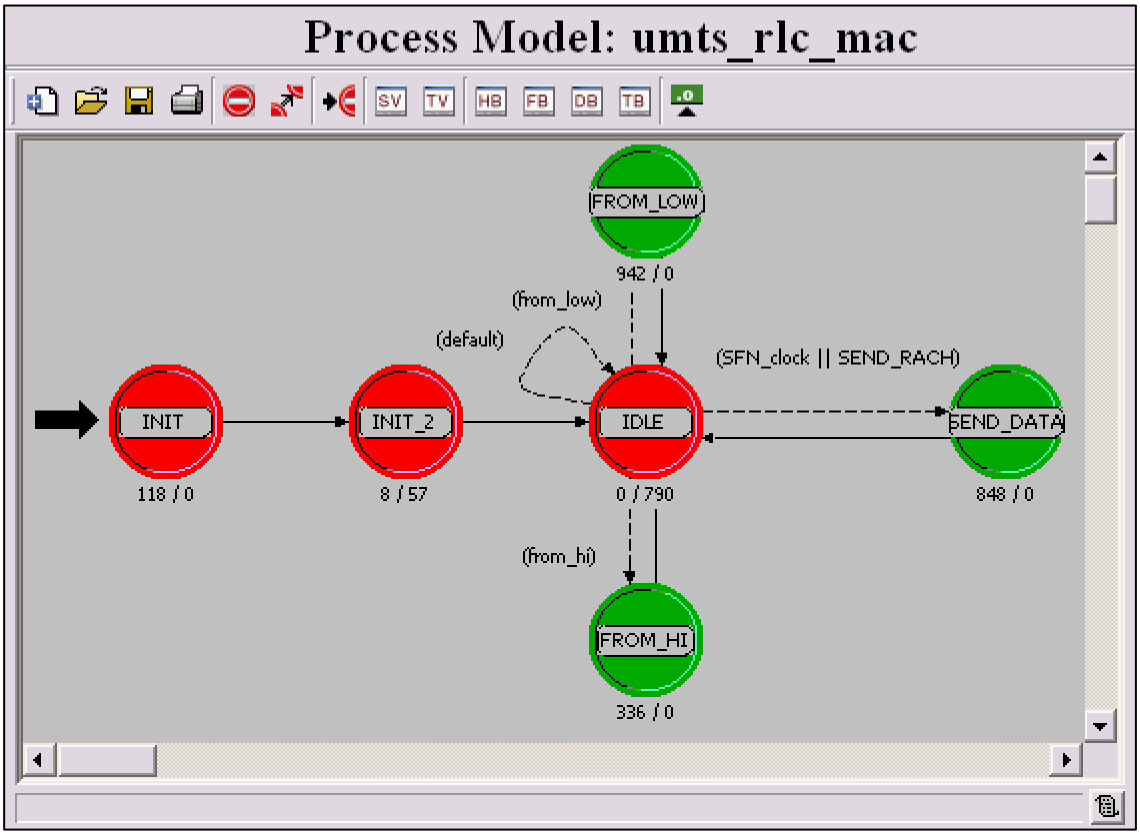


Figure 4.3. Example of process model in the Process Editor.

The UMTS model supports only the FDD mode, the TTI has a 10 ms length divided into 15 slots, and SF varies from 4 to 512 in DL and from 4 to 256 in UL.

Some features of the UMTS model are the four supported traffic classes (conversational, streaming, interactive and background), which can be combined for each UE. Admission control is also supported, and two algorithms for performing admission are available, one default algorithm and another based on throughput, as well as three RLC modes (acknowledged, unacknowledged and transparent mode), MAC priority handling of data flows based on the corresponding traffic class, outer loop power control, modelling UE movement within a cell among other features.

Two simple network configurations are supported: one using raw traffic generation, and another using application traffic. In the former, generic data traffic is sent between MTs through the same SGSN node; this configuration is applied when the goal is not to evaluate external IP networks or modelling CN to CN data transfer, Figure 4.4. The latter is more complex, and should be used when the goal is to analyse the effect of workstation nodes routing application traffic (specified in the application and profile definitions) to one or more CNs or other workstations or servers, which can be based on UMTS or other technologies, as Ethernet or WLAN. The chosen configuration is represented in Figure 4.5.

The network architecture of the available module can model RAN and CN.

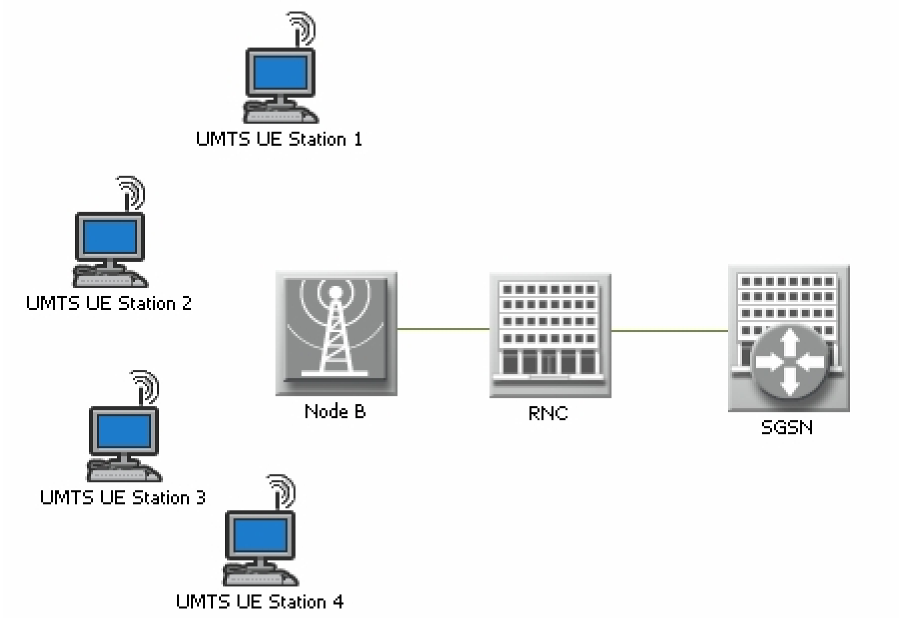


Figure 4.4. Simple UMTS network using raw traffic generation.

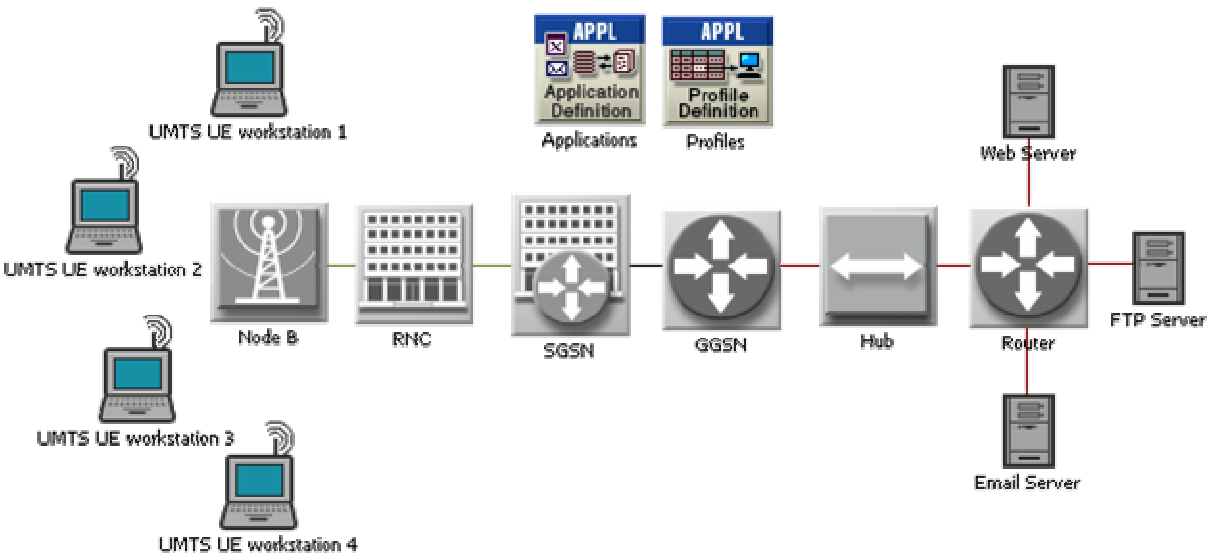


Figure 4.5. Simple UMTS network using application traffic.

The model strategy to emulate network behaviour during a simulation is as follows. First, when a user powers-on, a PS signalling connection is established and lasts all the simulation time, synchronisation is also established, which enables the user to access the provided services, as a UMTS GPRS attach is done with the SGSN. Packets arriving from higher layers are queued on one of the four QoS queues; if no Packet Data Protocol (PDP) is active, then an Activate PDP Context Request is sent to the SGSN, which consists of a message containing the requested QoS. The SGSN consults the RNC, and either accepts or rejects the request. In parallel with this consulting, the SGSN also sends a Radio Access Bearer (RAB) Assignment Request, which is granted after the UTRAN performs the admission control and verifies that UL and DL have enough capacity available. After accepting this request, a Radio Bearer Setup is sent to the UE by the RNC, so that the UE can arrange the channel according to the request. When the channel arrangement is done, a Radio Bearer Complete is sent to the RNC, which sends a RAB Assignment Response to the SGSN. An Activate PDP Context Accept message is sent by the SGSN so that the UE is allowed to send the packets to their destination, this message including the granted QoS.

UMTS models support three types of MTs: simple mobile stations, advanced workstations and advanced servers. In this work, only UMTS advanced workstations are used. The MT can be fixed or mobile, the latter being used when the MT moves within the cell during simulation. UMTS Station is a general workstation node that contains the MT, including client/server application functionality. This node is able to send and receive applications to other workstations in the same network or to the server, which can be connected to other UMTS or other technology based networks. As depicted in Figure 4.6, the workstation model includes an application layer connected to the GMM layer, the RLC/MAC layer, a receiver, a transmitter, one antenna, and a full TCP/IP protocol stack between the GMM and application layers.

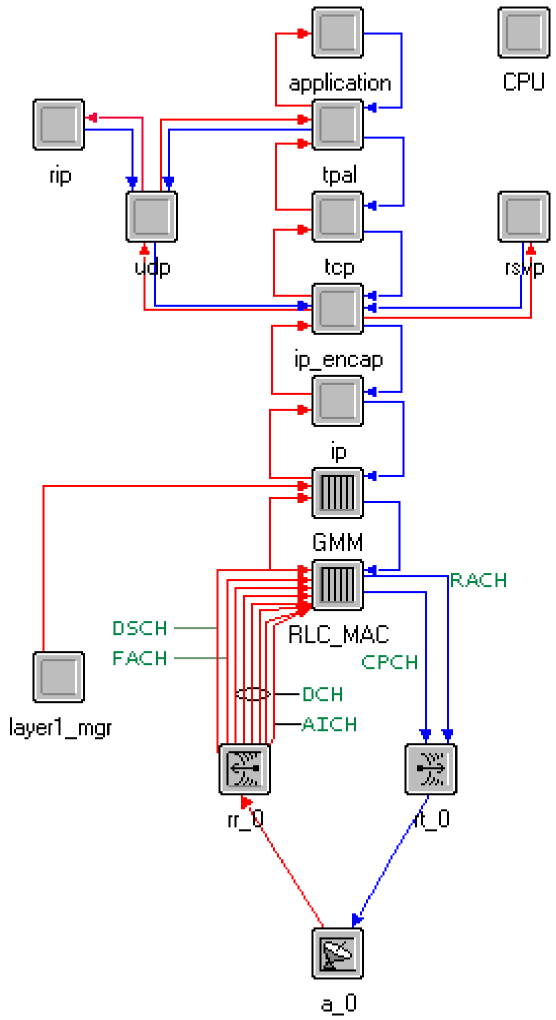


Figure 4.6. UMTS workstation node model.

The GMM layer is responsible for mobility management, as GPRS attach and handovers, session management (PDP Context Activation, service requests and the managing the radio bearers) and Packet Data Convergence Protocol (PDCP) compression. The RLC/MAC layer is responsible for the segmentation and reassembly of higher layer data, contains the three RLC modes, DCH and RACH.

Two Node B models can be used: single-sector, and three-sector, with three directional antennas. The Node B node model is represented in Figure 4.7.

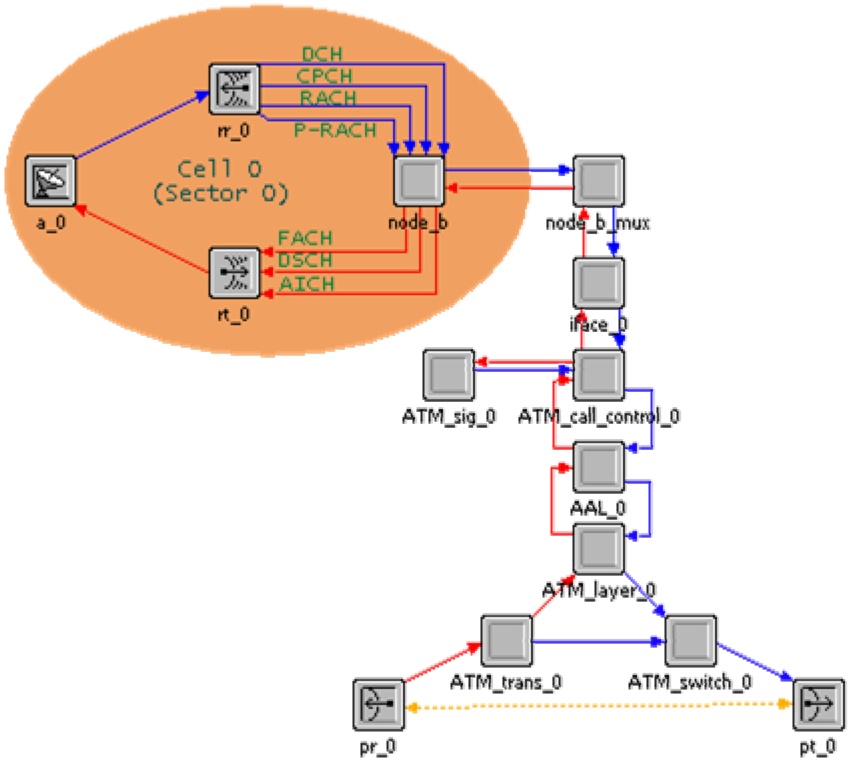


Figure 4.7. Node B node model.

The Node B is responsible for cell representation, RRM, assisting the RNC through Node B Application Protocol (NBAP) signalling messages, and radio interface with UEs. The Node B is connected to the RNC through an Asynchronous Transfer Mode (ATM) layer.

The RNC is responsible for admission control, radio bearer assignment and release, interface between Node-B and SGSN through ATM layer, buffer packets to the UE according to their QoS priority. The RNC node model contains a module with the RNC functionality, which is connected to various ATM stacks. One of the ATM connects to the SGSN, while the others can be connected to Node B's ATM stacks.

The CN architecture consists of a SGSN and a GGSN nodes. The SGSN node can be connected up to eight RNCs and to one GGSN. SGSN is responsible for GPRS attach, HLR registration and PDP context activation, and is modelled as various queues. The number of queues depends on the number of users in the cell and on how many QoS classes each user supports.

4.3 Input Parameters

In this section, the chosen configurations to run the simulations and the used scenarios are described.

The network architecture is comprised of a single-sector Node B, which employs an outdoor to indoor and pedestrian path loss model. This model represents a typical urban to suburban environment assuming a 10.5 m distance between the mean building height and MT height, 15 m between the MT and diffracting edges, and 80 m as the average separation between rows of buildings. The BS height is assumed near roof top level. The RNC defines admission control parameters, the DL load factor being 0.7. The other elements are the SGSN, GGSN, the server and a hub connecting the server to the GGSN, and several MTs. There are also two blocks defining the profiles and applications in usage. Figure 4.8 depicts an example of a simulation scenario.

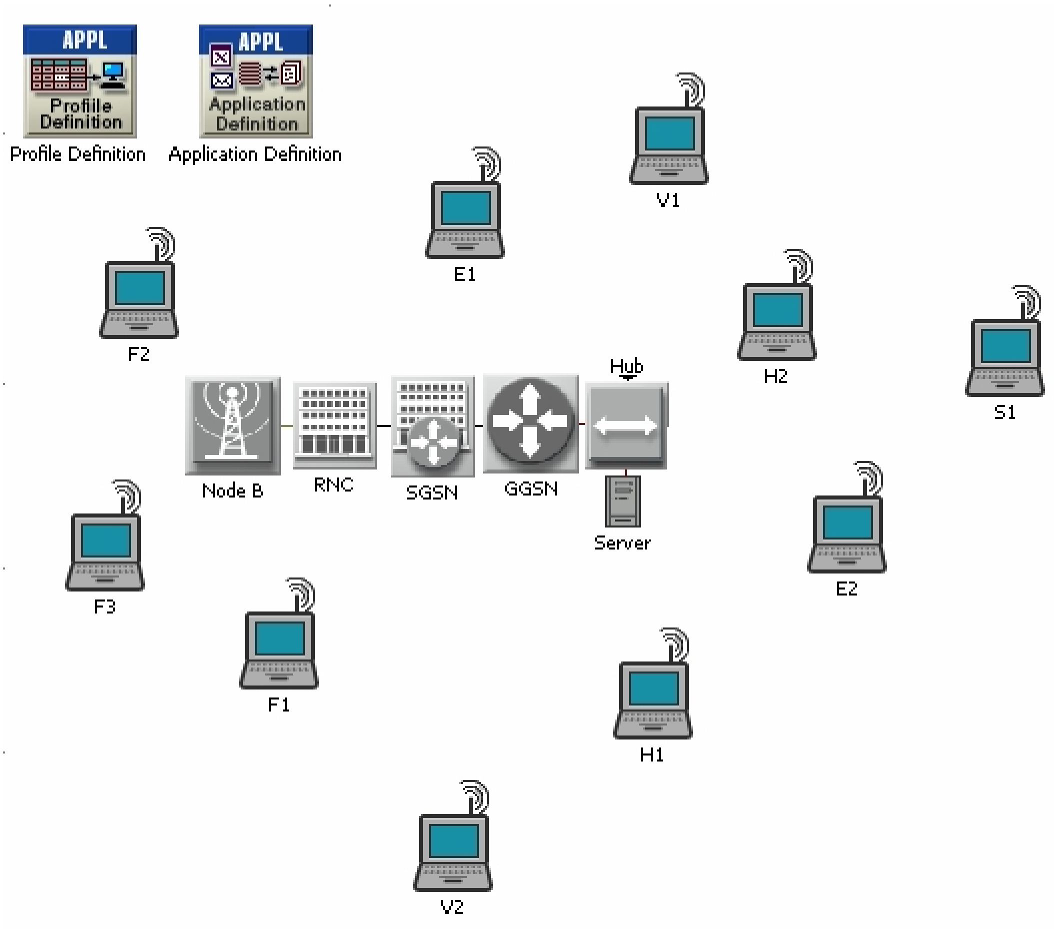


Figure 4.8. Example of the network architecture.

A Type of Service (ToS) class is associated to each application, and these classes are mapped onto QoS classes. OPNET has a throughput limitation to each QoS class; the corresponding values are presented on Table 4.1.

Table 4.1. Throughput limitation of each QoS class.

QoS class	Throughput [kbit/s]
Conversational	12.2
Streaming	12.2
Interactive	64
Background	64

The application and profile definitions are used to establish services characteristics. Table 4.1 presents some important parameters used for simulations, a detailed description of the remaining parameters is presented in Annex D.

Table 4.2. Application and profile main parameters.

Service	Parameter	Value	
		Light	Heavy
VoIP	Mean Call Duration [s]	120	120
	Inter-repetition Time [s]	600	600
Streaming	Mean File Volume [kB]	17500	17500
	Inter-repetition Time [s]	1800	900
FTP	Mean File Volume [kB]	2550	2550
	Inter-repetition Time [s]	1200	300
HTTP	Mean Page Volume [kB]	34.4	34.4
	Inter-repetition Time [s]	900	300
E-mail	Mean File Volume [kB]	100	100
	Inter-repetition Time [s]	900	300
MMS	Mean File Volume [kB]	165	165
	Inter-repetition Time [s]	1800	300

Inter-repetition time represents the time between repeating an application session during a profile, and it is associated to the number of calls per hour.

The behaviour of the network is assessed by increasing the number of users in each scenario, from 10 to 40. The higher limit was chosen based on the theoretical model, in which the maximum percentage of rejected user should be approximately 5% in order to approximate this assessment to a realistic scenario.

For the simulation scenario, the number of users for each different type of service is calculated according to the percentages presented on Table 3.4. Figure 4.9 represents the variation of the number of effective calls per hour (E.1), i.e., the number of times each application is repeated during a profile, for each data service, for the Data Centric profile and the Light scenario. The remaining scenarios are presented in Annex E.

4.4 Analysis of Results

In order to obtain satisfactory results, with an acceptable error, 10 simulations with different seeds were ran for each scenario, [Seba07], which makes a total number of 280 simulations.

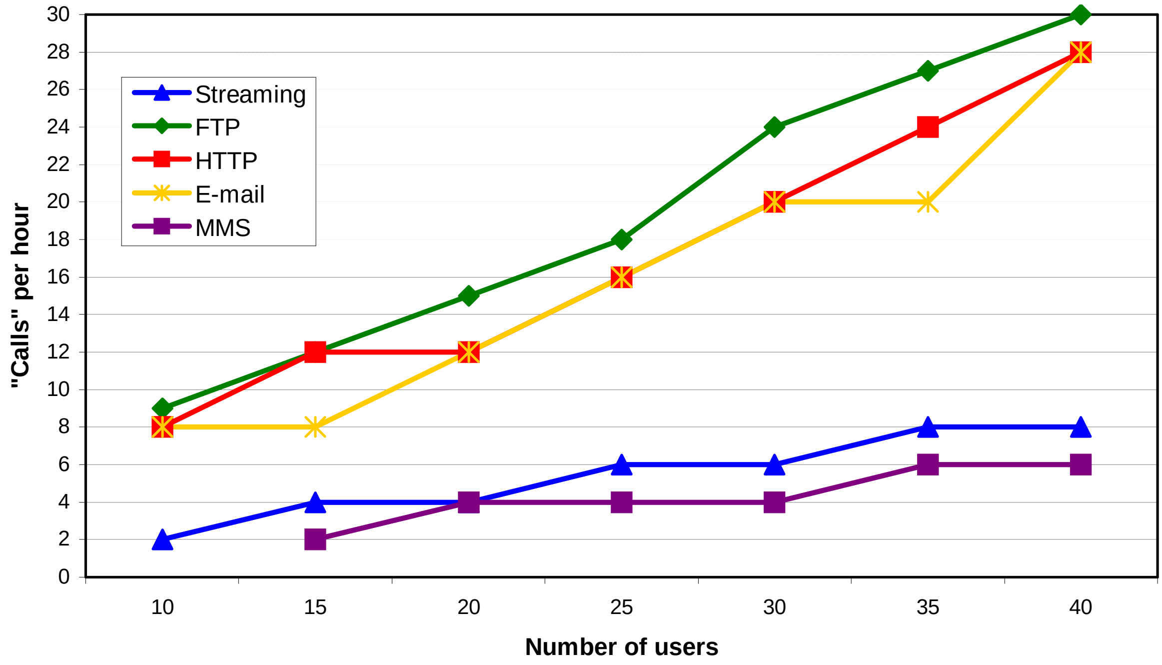


Figure 4.9. Calls per hour for Data Centric profile and Light scenario.

OPNET enables the analysis of an extensive list of parameters. In this work, just the parameters concerning the network behaviour as a whole, global parameters related to global traffic, and the traffic processed by each group of services are assessed, their definitions is presented in Annex F.

4.4.1 Global Results

Global Results includes only node statistics, which enables the analysis of each network element.

Figure 4.10 presents the percentage of requests granted and queued during admission control. As the number of users increases, the percentage of requests queued also increases, opposed to the decrease of the granted requests. It is visible that the network starts becoming in a saturation state for more than 15 users.

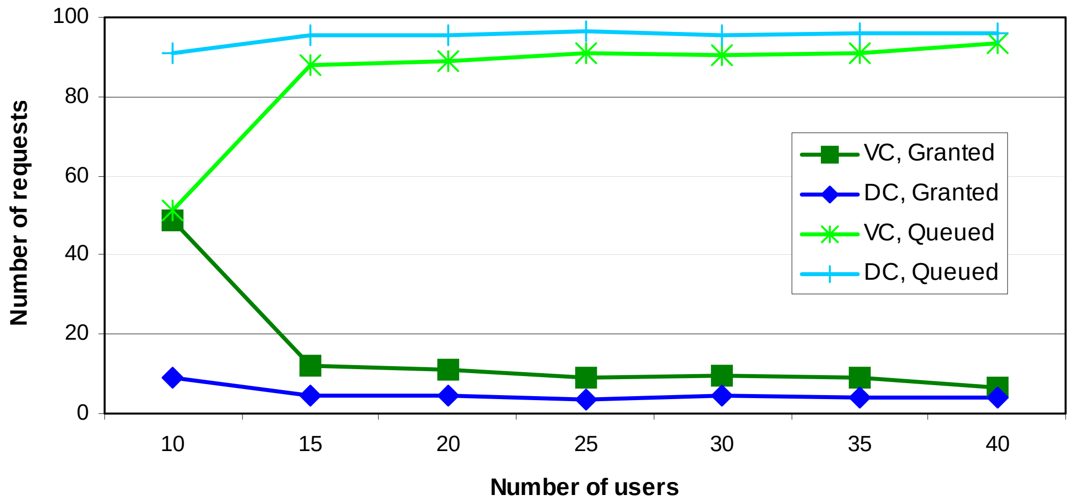


Figure 4.10. Percentage of requests granted and queued.

To complement this graphic, it is important to illustrate, for the different classes, the requests granted, Figure 4.11, and queued, Figure 4.12. The values presented are a percentage of the total number of requests, for the Voice Centric distribution.

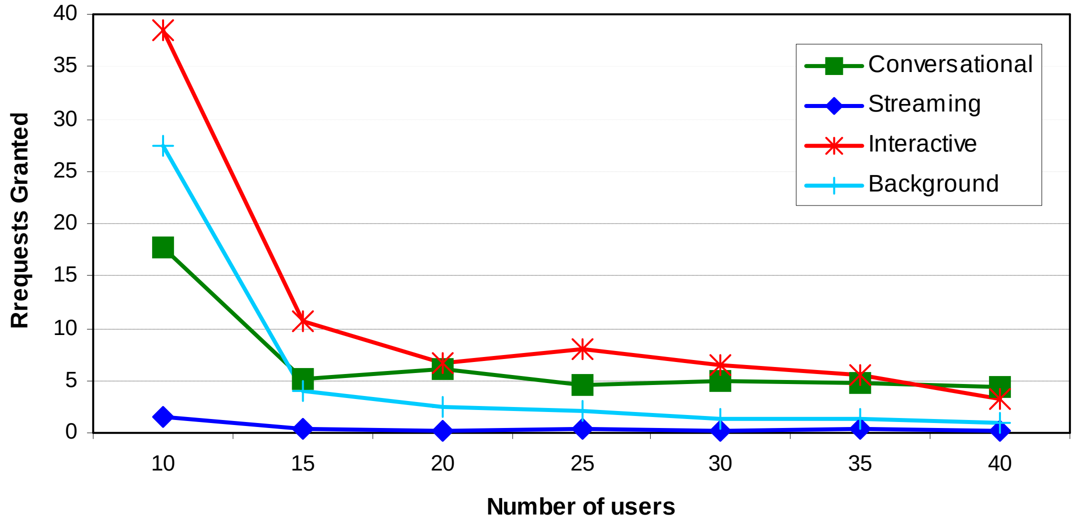


Figure 4.11. Percentage of requests granted per QoS class.

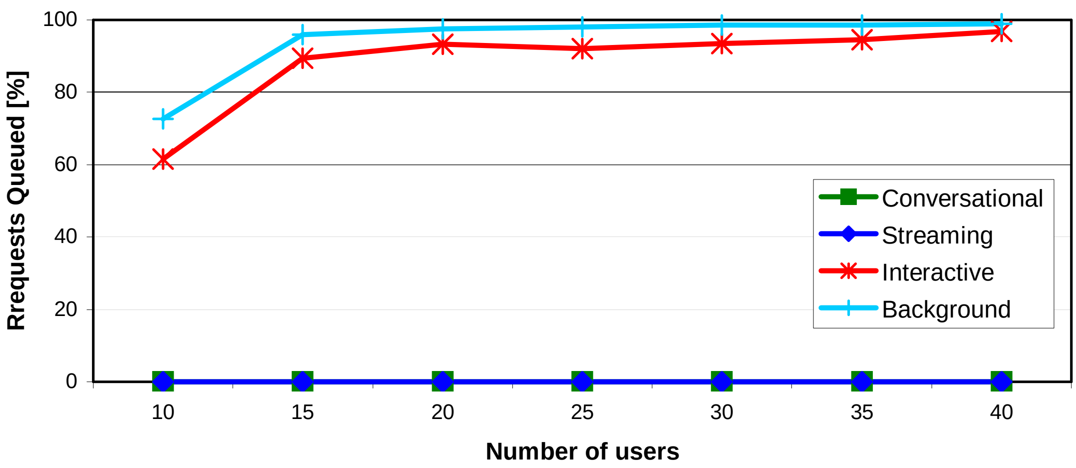


Figure 4.12. Percentage of requests queued per QoS class.

OPNET gives priority to Conversational and Streaming classes over Interactive and Background ones, which is the reason why Conversational and Streaming requests are all granted, and none is queued.

This behaviour influences the amount of traffic processed by the Node B. In Figure 4.13, the total DL throughput in the Node B is presented.

Although the standard deviation is very high, the Node B throughput tends to increase with the number of users, as it would be expected. However, the mean values for the Voice Centric profile are slightly higher than those for the Data Centric one. This behaviour is due to the priority policy performed by OPNET, Figure 4.10, where the percentage of granted requests is higher for the Voice Centric distribution than for the Data Centric one.

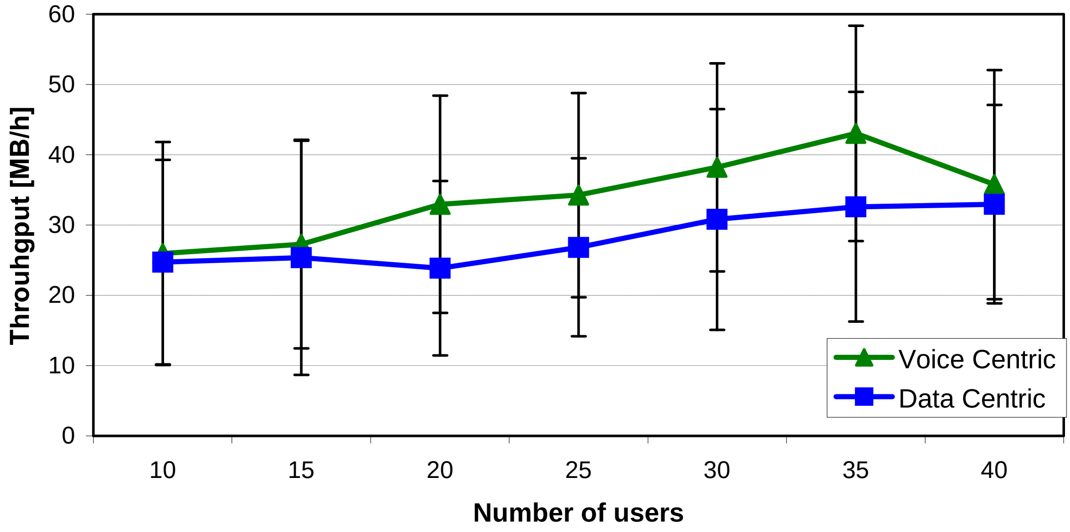


Figure 4.13. Node B total DL throughput.

The low values of the total Node B DL throughput can be explained by two factors. The first is due to the OPNET limitations concerning the UMTS modules. The second is due to the number of transmissions in DL, as shown in Figure 4.14, which is the average value of all transport channels. This statistic represents the total number of transmissions required to successfully transmit a PDU in the RLC acknowledge mode, for each transport channel. It was not possible to obtain results for the scenarios with a higher number of users.

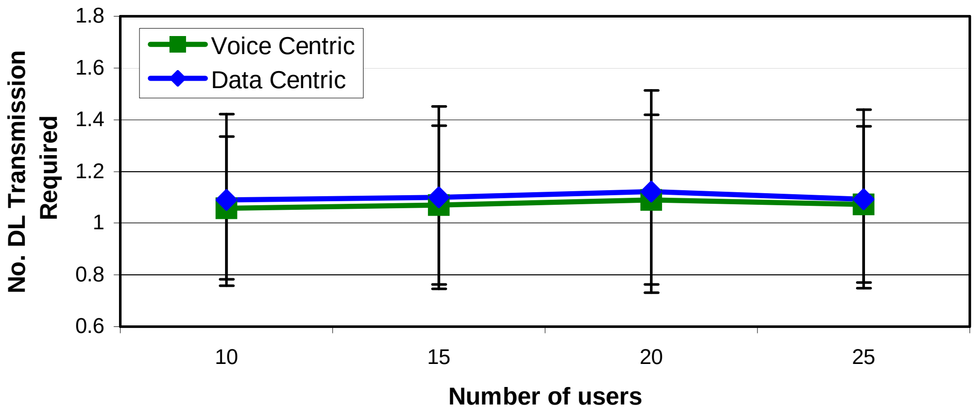


Figure 4.14. Average number of DL transmissions required

The average number of transmissions is approximately one, which means that the PDU is sent successfully at the first time, but there are cases where the maximum number of transmissions is very high, so, for this statistic, it is also important to present the results for the maximum number of transmissions per scenario, Figure 4.15.

Although the average results are satisfactory, there are cases where the number of retransmissions is very high, which means that the transmission of the same packet is repeated many times. This situation causes the global throughput to drop, comparing to an ideal situation where there would be no need to retransmit at all.

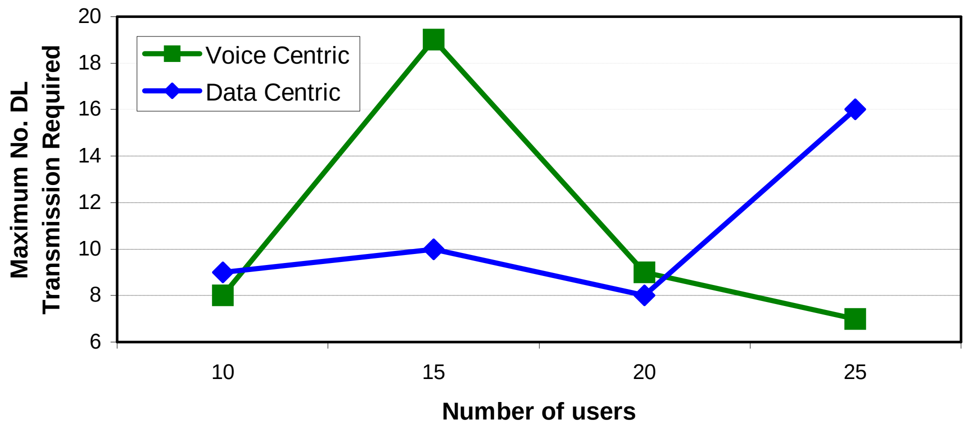


Figure 4.15. Maximum number of DL transmissions required.

In conclusion, the network starts to saturate for a low number of users, around 15, as one can see by the huge percentage of queued requests when comparing to the granted ones.

Also, the global traffic supported by the Node B seems to be a low value comparing to the expected traffic of around 70 MB/h, reference value of practical observation of a network. However, it is important to remind that this analysis is made for a single sector Node B, which should be equivalent to only a sector in a three-sectorised cell. For this case, the corresponding mean values of Node B DL throughput would vary between 75 and 120 MB/h, which corresponds to more acceptable values, although one can say that, hence, it is a light profile, the data base service volume is high compared to Voice, being, nowadays, still the most used service.

4.4.2 Applications

The group of results that are presented includes global parameters, which analyses each group of services, and node parameters, enabling an analysis of the nodes that can be grouped by applications.

Starting with the analysis of the node statistics, 4.16 and 4.17 show the average value of the total received throughput by VoIP and Streaming users, respectively. Due to OPNET's priority policy, the mean throughput of these users is constant, despite the increase in the number of users.

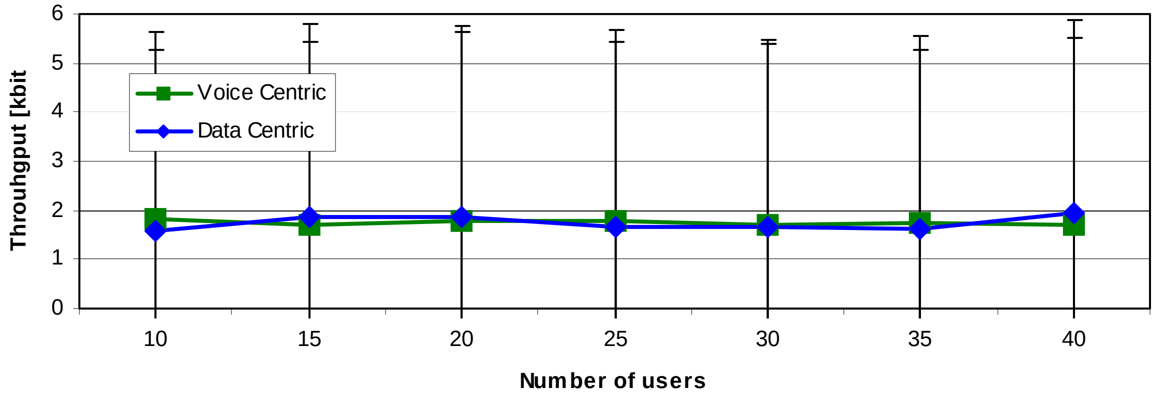


Figure 4.16. Average throughput received in the MT, for VoIP applications.

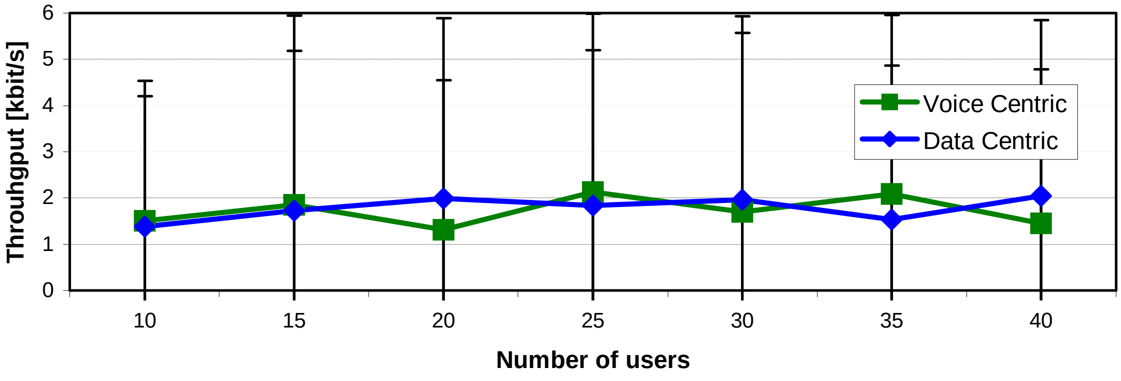


Figure 4.17. Average throughput received in the MT, for Streaming applications.

Results are similar for both classes. Although the mean value is low, approximately 2 kbit/s, which is not enough for these kind of applications, the maximum throughput achieved for both services is in the order of 11.8 kbit/s. The mean value is so low because OPNET calculates the mean of the entire hour, including the time when there are no communications established.

To complement this analysis delay is presented. Figures 4.18 and 4.19 characterise the packet end-to-end delay for VoIP and Streaming applications.

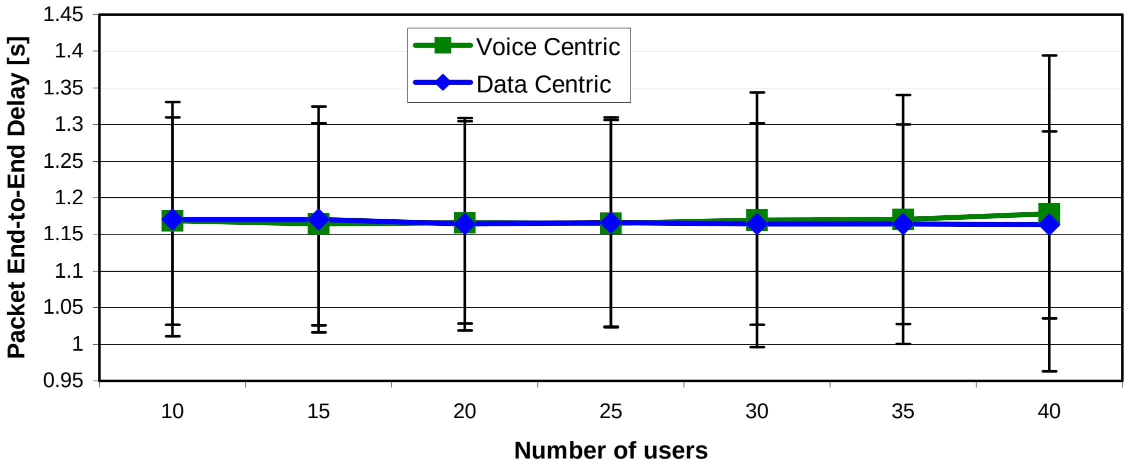


Figure 4.18. Packet end-to-end delay, for VoIP applications.

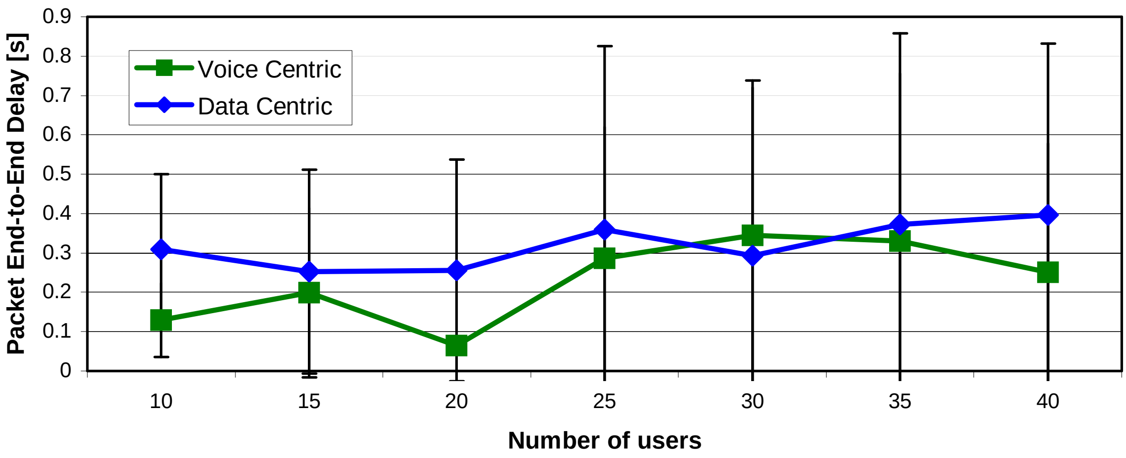


Figure 4.19. Packet end-to-end delay, for Streaming applications.

Concerning VoIP applications, a delay of approximately 1.2 s is extremely high. The Voice delay can reach a maximum of 400 ms in order not to be detected by human perception, so the obtained delay shows that VoIP supported by Release 99 has a very bad quality.

For Streaming applications, this delay is not so severe, as the mean value is around 300 ms, and in Streaming Video the delay is more tolerable than in Voice, because the user is not interacting with other user and it is not a real-time application.

The applications considered in the Interactive class are FTP and HTTP. Firstly, the throughput received by the MT is analysed, as shown in Figures 4.20 and 4.21.

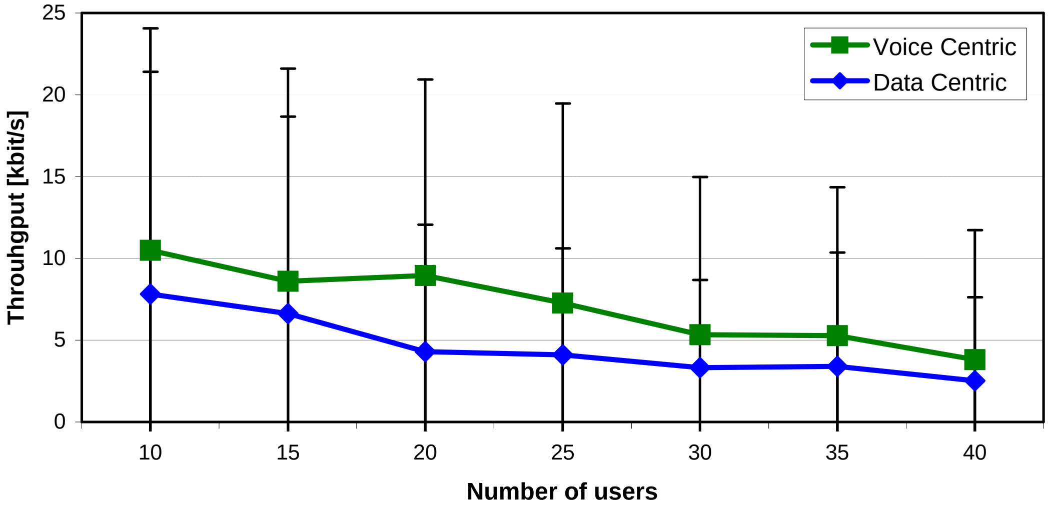


Figure 4.20. Average throughput received in the MT, for FTP application.

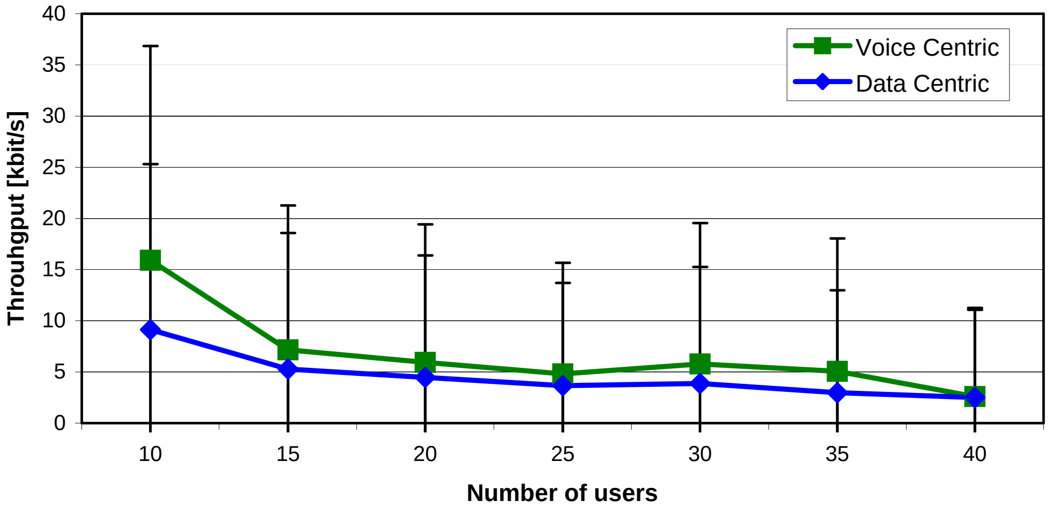


Figure 4.21. Average throughput received in the MT, for HTTP application.

As it would be expected, the throughput of the MT tends to decrease, due to the increasing number of users, which means that the available resources in the network must be shared with these users. Another factor that justifies this decrease is the priority given to VoIP and Streaming applications.

Figure 4.22 present the Download Response Time associated to FTP applications, and Figure 4.23

the Page Response Time for HTTP applications.

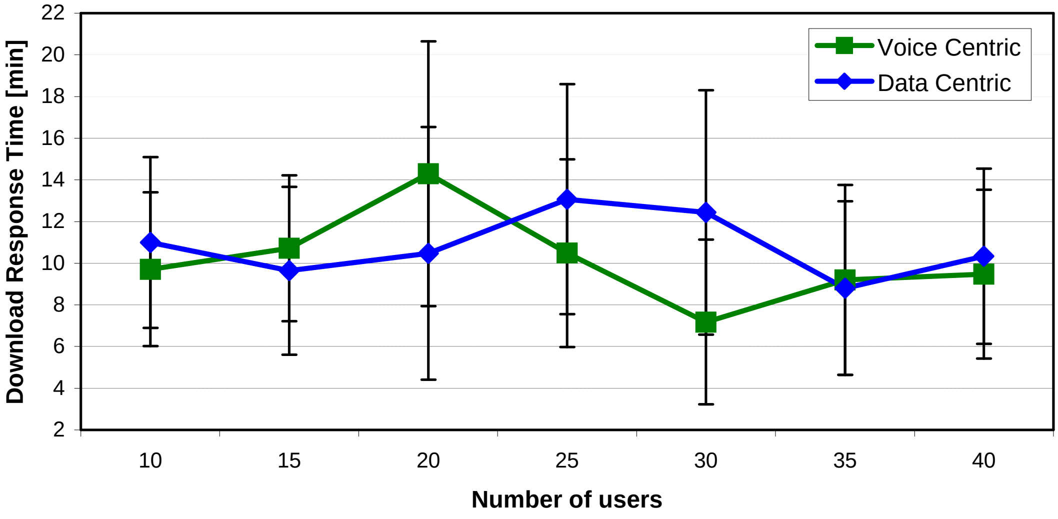


Figure 4.22. Download response time, for FTP application.

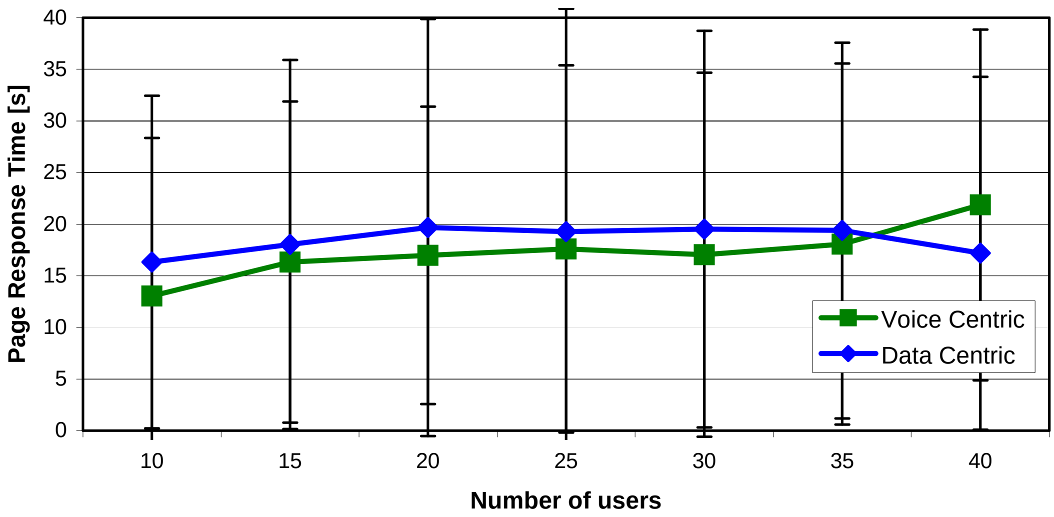


Figure 4.23. Page response time, for HTTP application.

Concerning the OPNET limitations and the mean file size of FTP applications, the results are not surprisingly high. It would be expected that the download time would increase as more users share resources, but at some point these users are queued during the admission control, which justifies the decrease of the time needed to download.

Regarding HTTP applications, the time needed to retrieve an entire page is in accordance to the throughput obtained and with the mean volume of each page. As expected, the page response time tends to increase with a growing number of users.

Release 99 is not the best choice to perform data based services, especially heavy ones, as FTP, which needs many available resources in order to have enough quality to attend client expectations, throughput is very low and response times are too high compared with other available technologies.

Background class applications include E-mail and MMS. The throughput analysis is made separately for both applications, as shown in Figures 4.24 and 4.25, but for the global parameters the two applications are analysed together, because there is no Messaging available in OPNET to characterise MMS applications, and it was characterised as E-mail, due to the similarities between them.

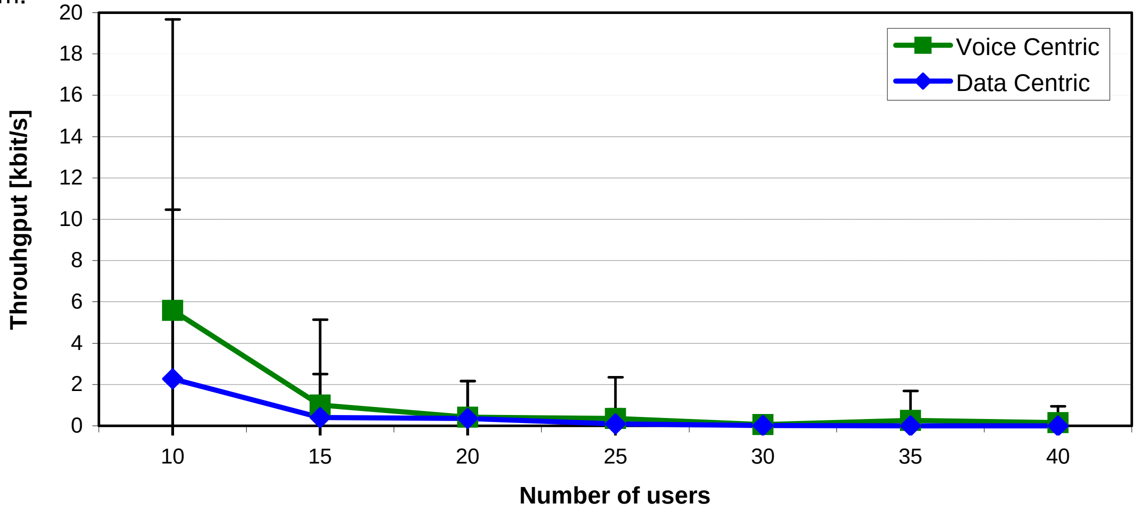


Figure 4.24. Average throughput received in the MT, for E-mail application.

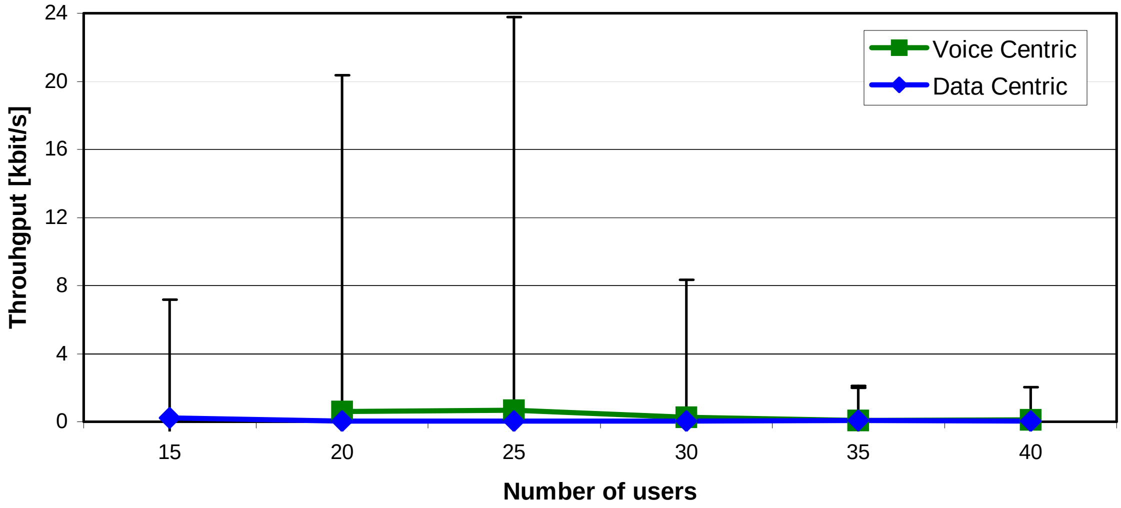


Figure 4.25. Average throughput received in the MT, for MMS application.

These results confirm a trend in data services for the throughput to decrease; however, given the high standard deviation compared to the mean value, mainly in the MMS case, the results, for some scenarios, are not reliable enough to allow to set a characteristic behaviour of the network.

The mean throughput values are very low, which would cause a long time to complete the request, as shown in Figure 4.26.

Included in the Background class, these applications are last prioritised, and being of less importance, the throughput needed is not significant as if they were a real-time application, nor is the time spent to send an e-mail or a MMS.

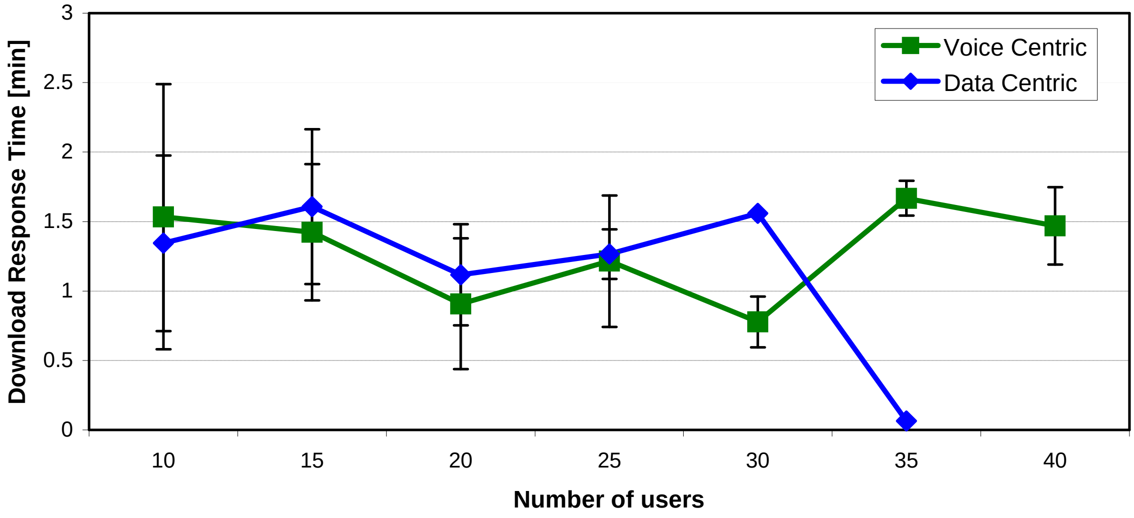


Figure 4.26. Download response time, for E-mail and MMS applications.

From this analysis, one can conclude that Release 99 is not the best technology to perform data based services, especially services that require high data rates and low response times. The solution for this problem lies in the appearance of HSDPA, which, enabling higher data rates, and, hence, lower response times, bring a new way of performing quality data-based services with the mobility that a mobile network, as UMTS, can offer.

4.5 Comparison with Theoretical Model

The comparison between the theoretical model and the results obtained in OPNET is made in terms of the traffic that is processed by the Node B, for the tendency of each scenario.

The results are presented for the range of 10 to 35 users, the lower limit being due to OPNET results and the higher to the proximity to the blocking probability of 5%.

Figure 4.27 compares the trends for the global traffic processed by the Node B with the theoretical model and with OPNET simulator, for the Voice Centric profile, and Figure 4.28 presents the same comparison but for the Data Centric one.

For both the theoretical model and OPNET results, the trend lines were obtained as linear approximations given by:

$$T_{[\mathrm{MB/h}]} = c \cdot N_u + c_0 \, , \tag{4.1}$$

being:

- T : global traffic processed by the Node B.

The values for constants c and c_0 depend on the profile and scenario, being presented in Table 4.3.

Table 4.3. Trend constants for OPNET results and theoretical model.

Approach	Profile	Scenario	Constant	
			c	c_0
Theoretical Model	Voice Centric	Light	4.258	1.072
		Heavy	10.781	1.072
	Data Centric	Light	5.454	7.471
		Heavy	13.677	29.883
OPNET	Voice Centric	Light	0.686	18.2
		Heavy	0.328	27.12
	Data Centric	Light	0.336	19.84
		Heavy	0.2	23

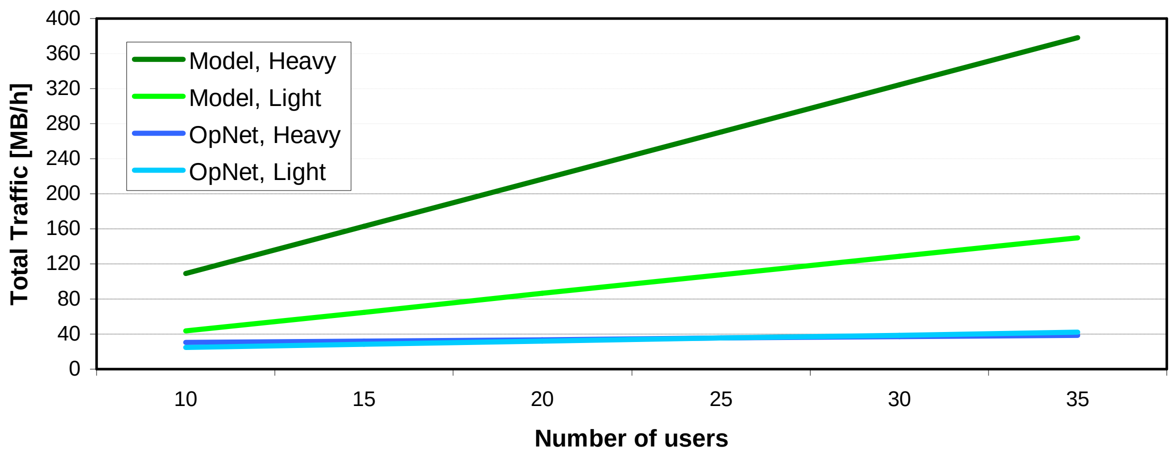


Figure 4.27. Global traffic in the Node B as a function of the number of users, for Voice Centric profile.

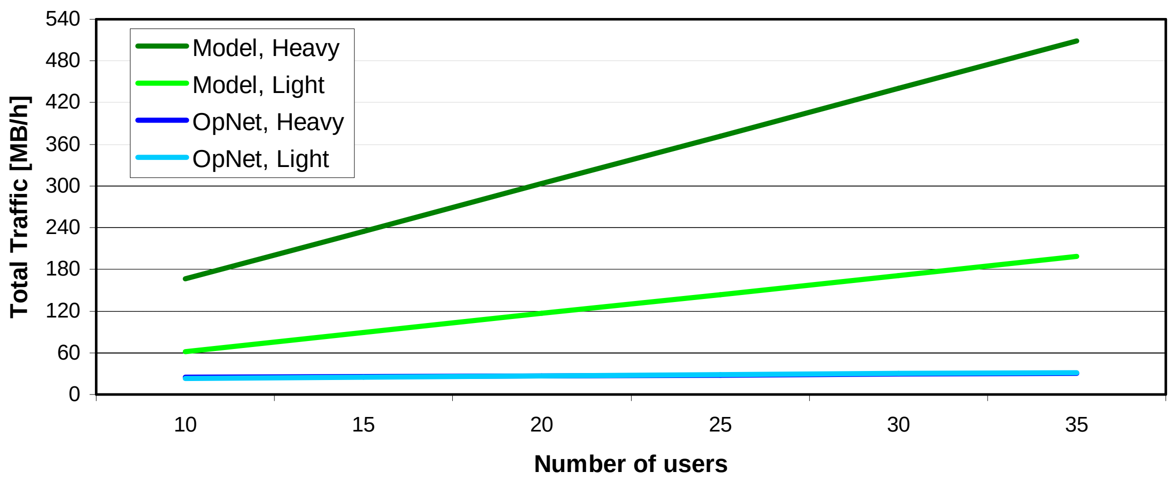


Figure 4.28. Global traffic in the Node B as a function of the number of users, for Data Centric profile.

As it can be seen from both figures, the total traffic obtained theoretically is much higher than the one obtained with the simulator. Both approaches, analytical and simulated, tend to increase with the number of users. The same effect is even more notorious for the Data Centric profile.

It is important to remember that OPNET does not have HSDPA implemented, which is considered for the theoretical model, although it is only related with the admission of new users. Another important factor that is not assessed in the theoretical model is the number of retransmissions, which is considered in OPNET. And finally, the OPNET limitations in terms of throughput; for the theoretical model all RABs are considered for all services except Voice. However, in OPNET it is only considered the 64 kbit/s RAB for applications included in the Interactive and Background classes, as described in Table 4.2.

Figure 4.29 shows how the global traffic obtained with simulations is reduced comparing to the one obtained theoretically by a ratio between the theoretical values and the OPNET ones, for both profiles and both scenarios.

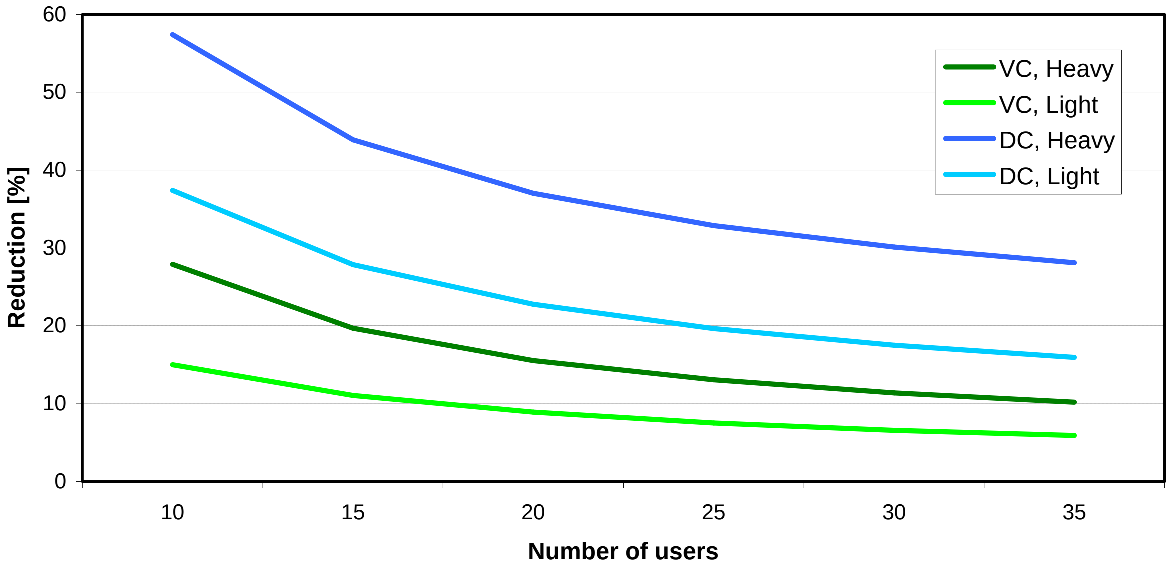


Figure 4.29. Reduction in the traffic comparing Heavy and Light scenarios.

As expected, the ratio for the Data Centric profile is higher than for the Voice Centric one, and the Heavy profile is higher than the Light one, which, accounting with the increase in the number of users, means that when the volume of data based service is higher, network performance decreases, and, once more, one can say that the answer to this problem lies in the HSDPA implementation, thus, complementing Release 99 when the tendency nowadays is to increase the use of data based services.

Chapter 5

Conclusions

In this chapter some conclusions are drawn, and some ideas for future work are presented.

The goal of this thesis was to assess UMTS networks performance and which is the impact in terms of global capacity due to the implementation of HSDPA. For this purpose, a model that predicts the theoretical behaviour of the system was developed and implemented with a snapshot approach, for multiple users performing multiple services. Two different profiles are analysed, one mostly based on voice services and other on data ones. Calculations were made varying parameters as frequency, interference, priority policies, and users' location in a sector of a cell, among others. With the obtained results it was possible to evaluate the global trend on how the network employs the available resources as the users that can be served, the throughput available to each of them, and the cell radius.

Initially, the goal in using OPNET was to implement HSDPA based on OPNET's existing UMTS modules. This turned out to be a hard task due to the extreme complexity of OPNET. The idea was that, with the help of the flag identifying potential HSDPA users, only a block in the RNC process model should suffer alterations, the admission control, but all the changes that were made never produced any results in practice. As mentioned, OPNET has many limitations, not only in the way channels are allocated but also in the maximum throughput that is possible to achieve. Hence, other approach was chosen, using OPNET Modeler to assess Release 99 only.

After obtaining analytical results, OPNET Modeler was used to evaluate Release 99 behaviour. The two service profiles used for the analytical model were used for these simulations, and also two more scenarios, the Light scenario, taken as a reference, and the Heavy one, which only difference lies in the number of calls per hour corresponding to each service.

Regarding the developed model, when analysing the network for a growing number of users, the conclusion is that there can be two limitative factors in the admission of new users. The SF is the first one, and its effect is the one that is felt earlier, around 22 users for Data Centric profile and 30 for the Voice Centric one. The other is the maximum allowed load factor, which effect is felt around 130 users for the Voice Centric profile and 270 for the Data Centric one. Two cases were studied concerning the load factor, one where the limit is 0.7, and the other 0.8. The differences between the two cases is the maximum number of served users, and for a load factor of 0.8 the network serves around 15% more users, at this point being only voice users, except for a few HSDPA users, since the rest has all been rejected due to the SF.

The percentage of rejected users in a real network is not higher than 1%, hence, the analysis was made for scenarios where the percentage of rejected users goes until around 5%, corresponding to a maximum of 30 users in the Voice Centric profile and 24 in the Data Centric one. For these values, the analysis of two different limitations of the load factor makes no sense. The load factor limitation just influences the attribution of a RAB to PS Release 99 users. There are cases where, although there are enough available SFs to use a 384 kbit/s RAB, the load factor is higher than the limit, so a lower level RAB is assigned, for all the cases analysed, PS Release 99 users being only served with the 384 kbit/s and 128 kbit/s RABs.

On the Release 99 analysis, throughput and cell radius is also calculated. The average throughput obtained for Release 99 maintains a constant value of 12.2 kbit/s until no PS user appears, and when this happens, a sudden increase in the throughput is verified. The maximum achieved throughput for the Data Centric profile is around 60% higher than for the Voice Centric one. Regarding the cell radius, it is important to mention that the maximum values for the indoor environment is, for some cases, around three times smaller than for the pedestrian one, which should be a factor to be taken into account when making cell planning.

Two scenarios are assessed, one where all users are at a maximum distance of the BS, and another where all users are placed in line between the BS and the cell border, equally displaced. For the first scenario, the goal is to serve all users, even if in order to achieve this goal the throughput is sacrificed. So, in order to serve all users in HSDPA, the throughput per user is decreased until none of the users is rejected, hence, the effective throughput is sometimes much lower than the target throughput. They have similar values until around 4 users in the network, which corresponds to 2 HSDPA users, the number of codes being equally divided, each user having 5 codes for the shared carrier case; when the number of users increases, the number of codes per user decreases, and so does the effective throughput, until it reaches the minimum of 0.672 Mbit/s per user, which corresponds to the case where each user has only 1 code.

Concerning the cell radius, HSDPA’s 300 m is much lower than Release 99’s 1 km for about 20 users. For the HSDPA shared carrier case, the radius for the indoor environment is more than 200 m lower than for the pedestrian one. This difference is even higher for the dedicated carrier case, which has no difference in behaviour, except for the maximum number of users that becomes 14, hence, all the resources of the network are distributed among them; it would be expected that the results, both in terms of throughput and of cell radius, would be better for this case.

Concerning the case of the shared carrier, although the number of served users is higher, it is taking both Release 99 and HSDPA into account, the resources available for HSDPA are only 30% of the ones available for Release 99.

As previously mentioned, the goal of the second scenario is to evaluate the maximum distance when users are displaced within the cell always maintaining their target throughput. Two approaches are taken for evaluation, the worst and the best cases; there is a significant difference between the two cases on almost all scenarios. As the number of users increases, there is a point when the cell radius obtained in Scenario 2 is lower than the one obtained in Scenario 1, due to the fact that throughput is much lower in Scenario 1 than in Scenario 2.

To finalise the analytical model assessment, the global traffic processed by the Node B was analysed, based on the number of served users that was calculated. The traffic for the Light scenario is almost 3 times lower than for the Heavy one. These are theoretical values and do not account for retransmissions, which is a reality in practice and causes the throughput to decrease. The global traffic was not calculated taking into account the throughput for each service, but only the number of users the Node B supports and the mean volume of each file or the mean call duration.

For Release 99 assessment, OPNET Modeler was used, which is a modeling and simulation tool for the analysis of communication networks. It enables the use of different network configurations, allowing the design and study of network devices, protocols and applications. OPNET Modeler has the UMTS module already implemented, being based on 3GPP Release 99 and supporting only PS traffic. OPNET has some limitations, the most important in the aim of this thesis is the throughput one. For Conversational and Streaming classes the throughput is limited to 12.2 kbit/s, and for Interactive and Background classes it is limited to 64 kbit/s.

The behaviour of the network is assessed by increasing the number of users in each scenario, from 10 to 40. OPNET enables the analysis of an extensive list of parameters; in this work, just the parameters concerning the network behaviour as a whole, global parameters related to global traffic, and the traffic processed by each group of services are assessed.

By analysing the number of requests granted and queued, as the number of users increases, the percentage of requests queued also increases opposed to the decrease of the granted requests. OPNET gives priority to Conversational and Streaming classes over Interactive and Background ones, which is the reason why Conversational and Streaming requests are all granted, and none is queued.

Node B DL throughput tends to increase with the number of users, however, the mean values for the Voice Centric profile is slightly higher than for the Data Centric one. This behaviour is due to the priority policy performed by OPNET. The low values of the total Node B DL throughput are due to the OPNET limitations concerning the UMTS modules, and to the number of transmissions in DL direction.

Globally, one can say that the network starts to saturate for a low number of users, as seen by the huge percentage of queued requests comparing to the granted ones. Also, the global traffic supported by the Node B seems to be a low value comparing to the expected traffic of around 70 MB/h, reference value of practical observation of a network. However, it is important to remind that this analysis is made for a single sector Node B, which should be equivalent to only a sector in a three-sectorised cell.

For Conversational and Streaming classes, the average throughput per users is constant with the increase in the number of users. Concerning VoIP applications, a delay of approximately 1.2 s is observed, which is extremely high. The Voice delay can reach the maximum of 400 ms not to be detected by human perception, showing that VoIP supported by Release 99 has a very bad quality. For Streaming applications, the mean delay is around 300 ms, which is more tolerable than in Voice, because the user is not interacting with another user and it is not a real-time application.

Regarding the Interactive class, Release 99 is not the best choice to perform data based services, especially heavy ones, as FTP, which needs many available resources in order to have enough quality to attend client expectations; throughput is very low and response times are too high compared with other available technologies.

Similarly to the Interactive class, Background class results confirm a tendency in data services for the throughput to decrease with the increase in the number of users. For this case, the response times are very high, as for the Interactive class, but, being of less importance in terms of priority, the throughput needed is not significant as if they were a real-time application, nor is the time spent to send an e-mail or a MMS.

For the comparison between the theoretical model and results obtained in OPNET, the traffic which is processed by Node B is analysed, for the tendency of each scenario. The total traffic obtained theoretically is much higher than the one obtained with the simulator, both approaches, analytical and simulated, tend to increase with the number of users. These can be justified remembering that OPNET does not have HSDPA implemented, which is considered for the theoretical model, although it is only related with the admission of new users. Another important factor that is not assessed in the theoretical model is the number of retransmissions, which is considered in OPNET, and OPNET limitations in terms of throughput.

From this analysis, one can conclude that Release 99 is not the best technology to perform data based services, especially services that require high data rates and low response times. The solution for this problem lies in the HSDPA implementation, which, enabling higher data rates, and, hence, lower response times, bring a new way of performing quality data-based services with the mobility that a mobile network, as UMTS, can offer, thus, complementing Release 99 when the tendency nowadays is to increase the use of data based services.

For future work, an analysis like the one performed with OPNET but including HSDPA would show how the network responds more clearly than only with a snapshot simulator. Given the 2 ms TTI in HSDPA, the conditions are always changing, and the low values of throughput do not correspond to reality in time. Another suggestion would be the analysis of more scenarios and profiles with OPNET, changing the characteristics of profiles and applications, as the percentage of users for each service and including other new services, and also analysing the case where users would be communicating with each other in the same cell and not only with the server. Given the enhancements in the communicating quality brought by HSUPA, analysing HSDPA and HSUPA together, 3GPP Release 6, would give a more realistic vision of a near future way of performing data based services. It would be also interesting to compare Release 6 with other technologies as WiMAX and WiBro (Mobile WiMAX).

Annex A

Link Budget

In this annex, the link budget is described.

Link budget calculations are based on [CoLa06] and [EsPe06].

The total path loss is defined by [Corr06]:

$$L_{P[\mathrm{dBm}]} = P_{t[\mathrm{dBm}]} + G_{t[\mathrm{dBi}]} - P_{r[\mathrm{dBm}]} + G_{r[\mathrm{dBi}]} = EIRP_{[\mathrm{dBm}]} - P_{r[\mathrm{dBm}]} + G_{r[\mathrm{dBi}]}, \qquad \text{(A.1)}$$

being:

- § L_P : path loss;
- § P_t : power fed to transmitting antenna;
- § G_t : gain of the transmitting antenna;
- § P_r : power available at the receiving antenna;
- § G_r : gain of the receiving antenna.

In DL, Equivalent Isotropic Radiated Power (EIRP) is given by:

$$EIRP_{[\mathrm{dBm}]} = P_{Tx[\mathrm{dBm}]} - L_{C[\mathrm{dB}]} + G_{t[\mathrm{dBi}]}, \qquad \text{(A.2)}$$

where:

- § P_{Tx} : transmitter output power;
- § L_C : losses in cable between transmitter and antenna.

The received power is calculated by:

$$P_{Rx[\mathrm{dBm}]} = \mathrm{P}_{r[\mathrm{dBm}]} - L_{u[\mathrm{dB}]}, \qquad \text{(A.3)}$$

being:

- § P_{Rx} : power at the input of the receiver;
- § L_u : losses due to user.

The average noise power is estimated by:

$$N_{RF[\mathrm{dBm}]} = -174 + 10\log\left(\Delta f_{[\mathrm{MHz}]}\right) + F_{[\mathrm{dB}]}, \qquad \text{(A.4)}$$

where:

- § Δf : signal bandwidth, in UMTS it s equivalent to the chip rate, R_C ;
- § F : noise figure,

and the noise power is:

$$N_{[\mathrm{dBm}]} = N_{RF[\mathrm{dBm}]} + M_{I[\mathrm{dB}]}, \qquad \text{(A.5)}$$

being:

- § M_I : interference margin.

The interference margin is defined in (2.1) and is not considered for HSDPA. For HSDPA the processing gain is fixed (12 dB) and for Release 99 the is given by:

$$G_P = 10\log\left(R_C / R_b \right), \qquad \text{(A.6)}$$

where:

- § R_b : service bit rate.

So the receiver sensitivity is :

$$P_{Rx\min[\text{dBm}]} = N_{[\text{dBm}]} - G_{P[\text{dB}]} + SNR_{[\text{dB}]}\,, \tag{A.7}$$

being:

- SNR : signal to noise ratio, representing E_b/N_0 for Release 99, and SINR for HSDPA.

Some margins must be defined to consider other losses due to radio propagation:

$$M_{[\text{dB}]} = M_{SF[\text{dB}]} + M_{FF[\text{dB}]} + L_{int[\text{dB}]} + G_{SHO[\text{dB}]}\,, \tag{A.8}$$

where:

- M_{SF} : slow fading margin;
- M_{FF} : fast fading margin;
- L_{int} : indoor penetration;
- G_{SHO} : soft handover gain.

Finally the total path loss is given by:

$$L_{P_{total}[\text{dB}]} = L_{P[\text{dB}]} + M_{[\text{dB}]}\,. \tag{A.9}$$

The frequency validation limits are exceeded by UMTS in DL, but the error is considered acceptable.

The values of SINR for HSDPA were obtained as an interpolation of the curves in Figure 2.6, [Lope07]. For 10 HS-PDSCH codes the SINR is approximated by:

$$r_{[\text{dB}]} = 0.0382 \cdot R_b^{\,5} - 0.6722 \cdot R_b^{\,4} + 4.4891 \cdot R_b^{\,3} - 14.2023 \cdot R_b^{\,2} + 24.3795 \cdot R_b - 4.6875\,. \tag{A.10}$$

and for 15 HS-PDSCH codes the SINR is approximated by:

$$r_{[\text{dB}]} = \begin{cases} 0.0061 \cdot R_b^{\,5} - 0.1663 \cdot R_b^{\,4} + 1.6581 \cdot R_b^{\,3} - 7.8530 \cdot R_b^{\,2} + 18.9881 \cdot R_b - 3.9237, R_b \le 5.4 \\ \\ 0.0952 \cdot R_b^{\,4} - 2.7432 \cdot R_b^{\,3} + 29.4923 \cdot R_b^{\,2} - 138.1340 \cdot R_b - 257.0166, 5.4 < R_b \le 10.8 \end{cases} \text{.(A.11)}$$

The interpolation curves and the real curves are depicted in Figure A.1. To validate the obtained expressions the relative mean error and variance were calculated, being presented in Table A.1.

Table A.1. Relative mean error and variance for SINR values.

	Relative Error [%]	Variance
10 HS-PDSCH codes	2.1	$4.0\text{x}10^{-2}$
15 HS-PDSCH codes	3.3	$5.3\text{x}10^{-2}$

To obtain the throughput that is possible to achieve with a certain value of SINR, the following interpolation is used, obtained by inverting the values of Figure A.1, as depicted in Figure A.2. For 10 HS-PDSCH codes the throughput is approximated by:

$$R_b = \begin{cases} 0.085 \cdot r_{[\mathrm{dB}]} + 0.34, r \le -2 \\ 0.0167 \cdot r_{[\mathrm{dB}]} + 0.2034, -2 < r \le 1 \\ 0.076 \cdot r_{[\mathrm{dB}]} + 0.144, 1 < r \le 6 \\ 0.0085 \cdot r_{[\mathrm{dB}]}^{\;2} + 0.0271 \cdot r_{[\mathrm{dB}]} + 0.1141, 6 < SINR \le 24 \\ 0.1667 \cdot r_{[\mathrm{dB}]} + 1.599, 24 < r \le 30 \\ 0.08 \cdot r_{[\mathrm{dB}]} + 4.2, 30 < r \end{cases}, \tag{A.12}$$

And for 15 HS-PDSCH codes the throughput is approximated by:

$$r = \begin{cases} 0.0367 \cdot SINR_{[dB]} + 0.183, SINR \le 1 \\ 0.09 \cdot SINR_{[dB]} + 0.13, 1 < SINR \le 3 \\ 0.1296 \cdot SINR_{[dB]} + 0.014, 3 < SINR \le 10 \\ 0.3 \cdot SINR_{[dB]} - 1.7, 10 < SINR \le 16 \\ 0.54 \cdot SINR_{[dB]} - 5.5, 16 < SINR \le 25 \\ 0.3 \cdot SINR_{[dB]} + 0.5, 25 < SINR \le 30 \\ 0.1 \cdot SINR_{[dB]} + 6.5, 30 < SINR \end{cases}. \tag{A.13}$$

The relative mean error and variance of this interpolation are presented in Table A.2.

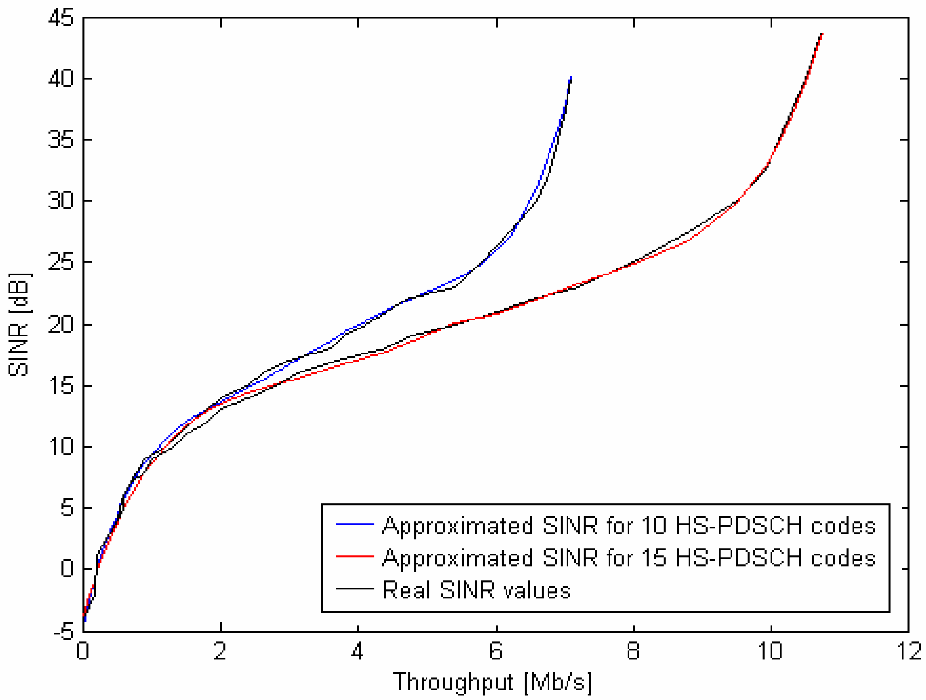


Figure A.1. Approximated *SINR* values for 10 and 15 HS-PDSCH codes.

Table A.2. Relative mean error and variance for throughput values.

	Relative Error [%]	Variance
10 HS-PDSCH codes	3.0	$1.3x10^{-2}$
15 HS-PDSCH codes	4.9	$5.0x10^{-2}$

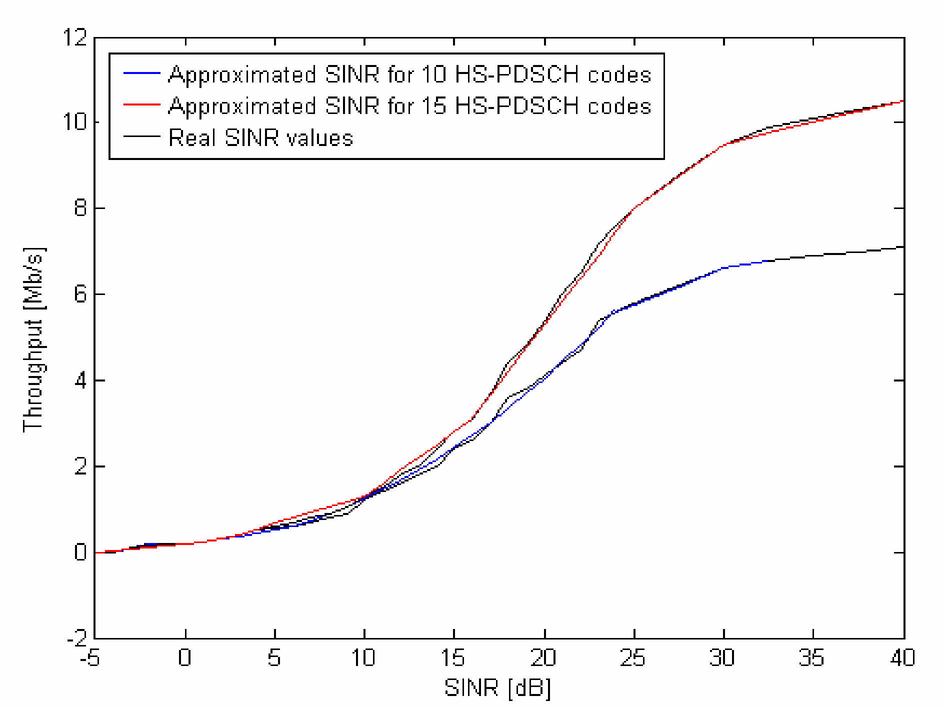


Figure A.2. Approximated throughput values for 10 and 15 HS-PDSCH codes.

Annex B

Model

In this annex, the used propagation model to obtain the model cell radius equations is described.

The radius is obtained according to the COST-231 Walfisch-Ikegami propagation model, [DaCo99]. The path loss for free space propagation is, [Corr06]:

$$L_0 = 32.4 + 20\log\left(R_{[\mathrm{km}]}\right) + 20\log\left(f_{[\mathrm{MHz}]}\right). \tag{B.1}$$

And the total path loss according to the propagation model is:

$$L_{P[\mathrm{dB}]} = L_{0[\mathrm{dB}]} + L_{tt[\mathrm{dB}]} + L_{tm[\mathrm{dB}]}. \tag{B.2}$$

Accounting for propagation from the roof tops to the MT, and considering non line of site propagation (nLoS), Node B height higher than the building height, and a dense urban area, one has:

$$L_{tm} = -16.9 - 10\log\left(w_{s[\mathrm{m}]}\right) + 10\log\left(f_{[\mathrm{MHz}]}\right) + 20\log\left(H_{B[\mathrm{m}]} - h_{m[\mathrm{m}]}\right) + L_{ori}, \tag{B.3}$$

where:

- § w_s : street width;
- § H_B : Node B height;
- § h_m : MT height;

being:

$$L_{ori} = 4 - 0.114\left(90^{\mathrm{o}} - y\ \left[^{\mathrm{o}}\right]\right), \tag{B.4}$$

where:

- § y : orientation angle.

Accounting for propagation over roof tops:

$$L_{tt} = L_{bsh} + k_a + k_d \cdot \log\left(R_{[\mathrm{km}]}\right) + k_f \cdot \log\left(f_{[\mathrm{MHz}]}\right) - 9 \cdot \log\left(w_{B[\mathrm{m}]}\right), \tag{B.5}$$

where:

- § w_B : building separation;

being:

$$L_{bsh} = -18 \cdot \log\left(h_b - H_B + 1\right), \tag{B.6}$$

where:

- § h_b : building height;

and

$$k_f = -4 + 1.5\left(\frac{f_{[\mathrm{MHz}]}}{925} - 1\right). \tag{B.7}$$

According to the considerations earlier mentioned, k_a and k_d are constants and have the values of 54 and 18, respectively.

The values used for the calculations are presented in Table B.1. This model is valid for, [Corr06]:

- $f \in [800; 2000]$ MHz;
- $R \in [0.02; 5]$ km;
- $h_b \in [4;50]$ m;
- $h_m \in [1;3]$ m.

Table B.1. Parameters used for propagation model.

Parameter	Value
Street width [m]	24
Building Separation [m]	48
Node B height [m]	26
Building height [m]	24
MT height [m]	1.8
Orientation angle [º]	90

Manipulating (B.1) and (B.2), the cell radius can be determined by:

$$R_{[\mathrm{km}]} = 10^{\frac{L_{P[\mathrm{dB}]} - 32.4 - \log\left(f_{[\mathrm{MHz}]}\right) - L_{tt[\mathrm{dB}]} - L_{tm[\mathrm{dB}]}}{20}}, \tag{B.8}$$

also, being L_P given by (A.1) and taking into account the receiver sensitivity, (A.7), and the received power, (A.3), the radius can be determined as a function of the power:

$$R_{[\mathrm{km}]} = 10^{\frac{EIRP_{[\mathrm{dBm}]} - P_{Rx\min[\mathrm{dBm}]} - L_{u[\mathrm{dB}]} - 32.4 - \log\left(f_{[\mathrm{MHz}]}\right) - L_{tt[\mathrm{dB}]} - L_{tm[\mathrm{dB}]}}{20}}. \tag{B.9}$$

Annex C

Capacity and Interference Model Results

In this annex, the results for the cell radius calculator are presented.

Figure C.1 and Figure C.2 presents the user distribution for the Data Centric profile, when the load factor limit is 0.7.

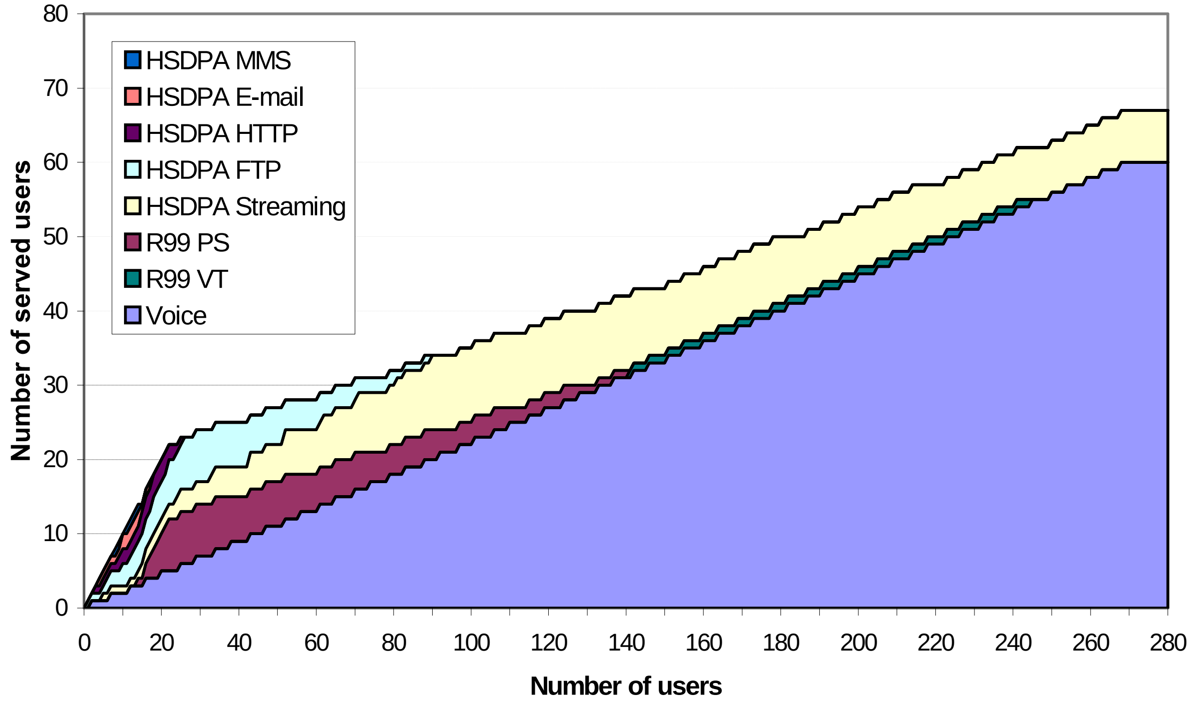


Figure C.1. User distribution for Data Centric profile, QoS priority policy and limit load factor 0.7.

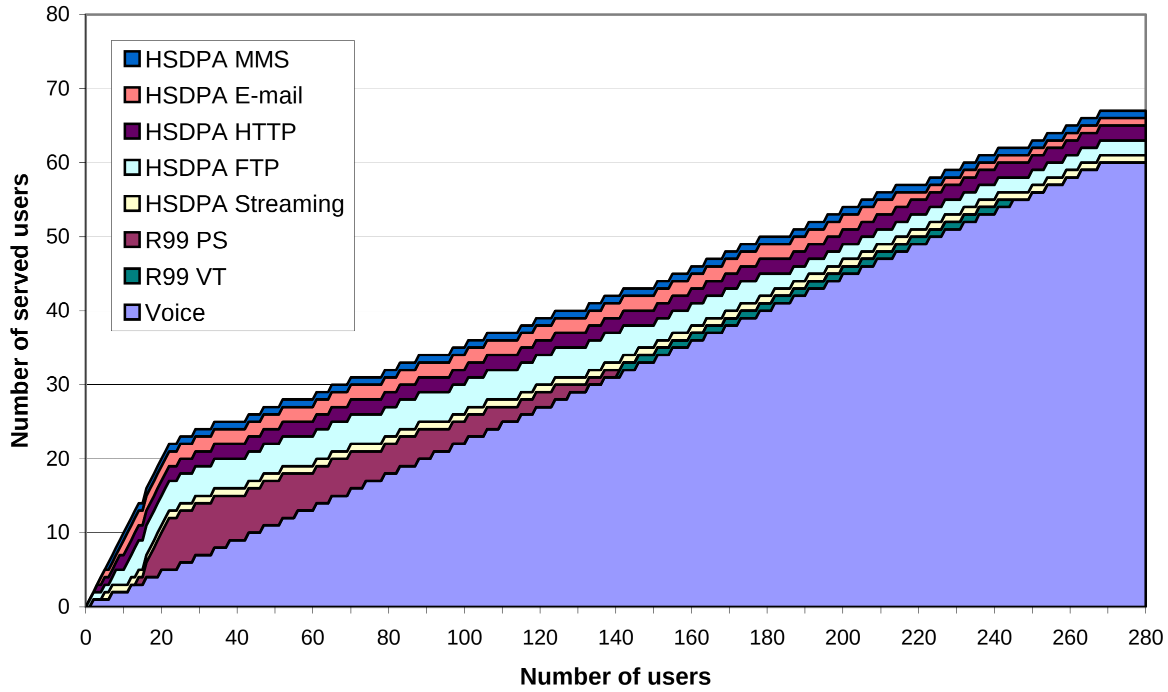


Figure C.2. User distribution for Data Centric profile, Proportional priority policy and limit load factor 0.7.

Figures C.3 and C.4 represent the target and effective throughputs, and throughput efficiency for the HSDPA shared carrier case.

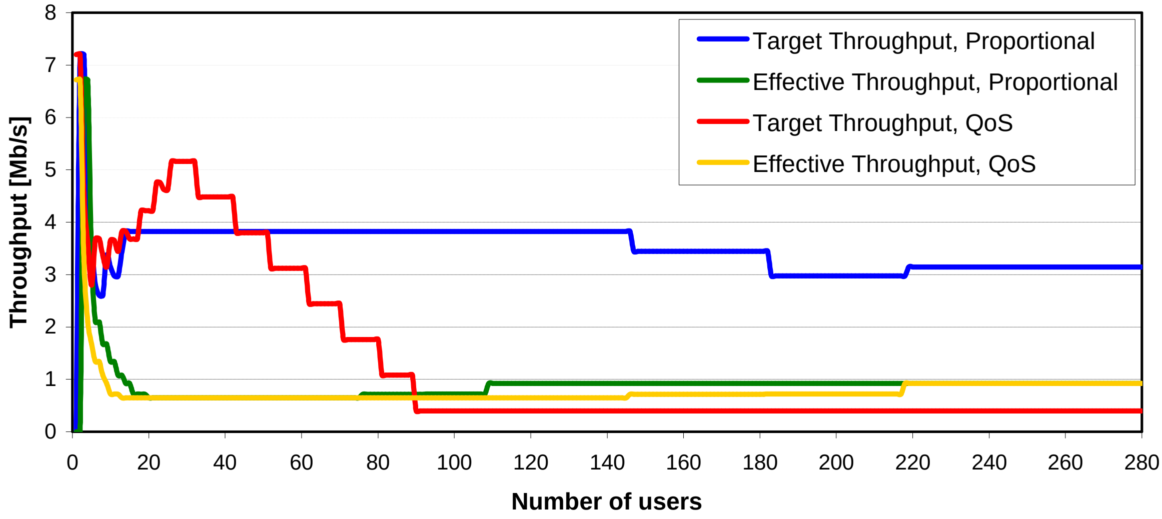


Figure C.3. Target and effective throughput profile and for QoS priority policy, for the Data Centric profile.

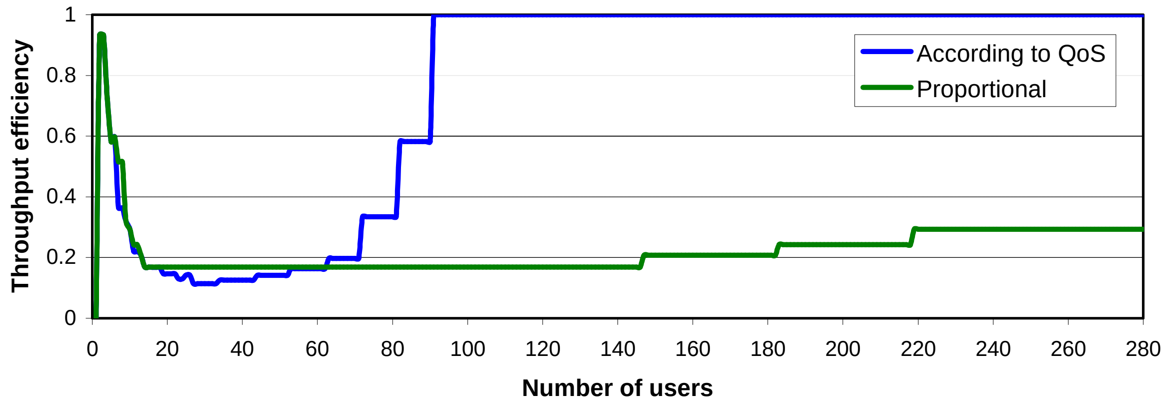


Figure C.4. Throughput efficiency for Data Centric profile.

For the HSDPA dedicated carrier case, the user distribution for the Voice Centric profile and for both QoS and Proportional priority policy algorithms is represented in Figure C.5 and C.6 respectively.

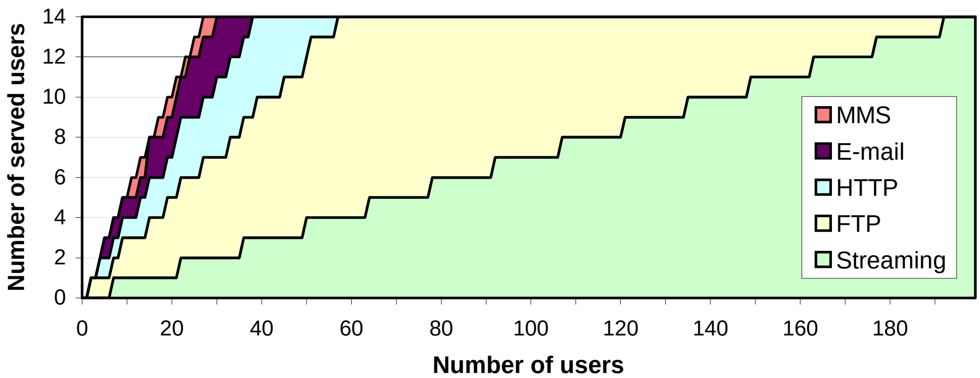


Figure C.5. User distribution for Voice Centric profile and QoS priority policy.

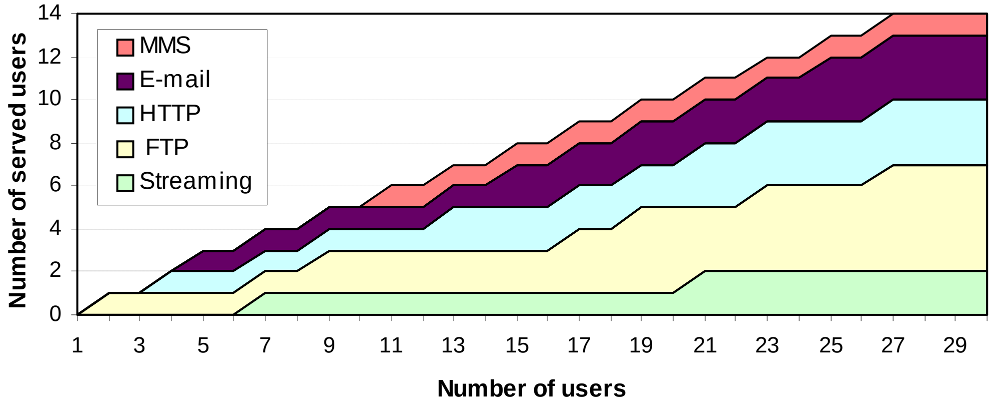


Figure C.6. User distribution for Voice Centric profile and Proportional priority policy.

The following results correspond to the cases where the percentage of rejected users is around 5%.

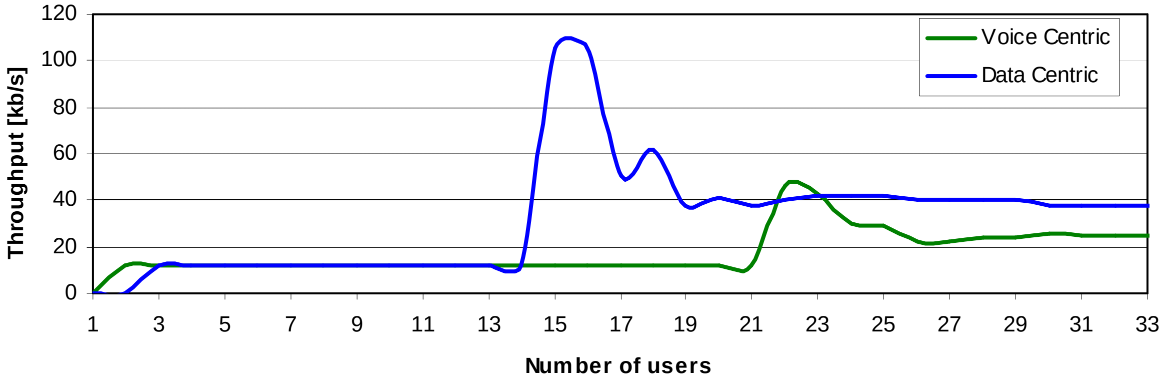


Figure C.7. Throughput for Release 99, for the limitation of users rejected of 5%.

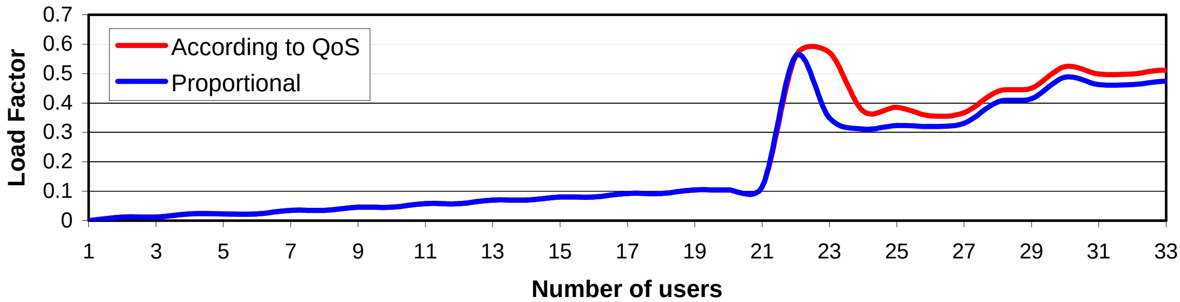


Figure C.8. Load factor for Voice Centric profile, for the limitation of users rejected of 5%.

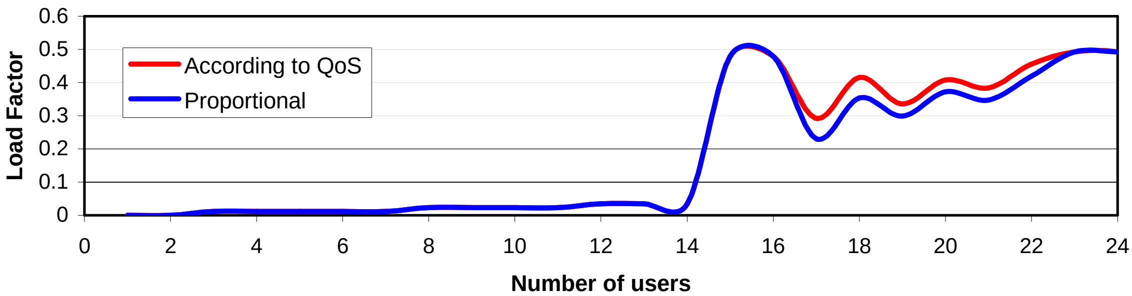


Figure C.9. Load factor for Data Centric profile, for the limitation of users rejected of 5%.

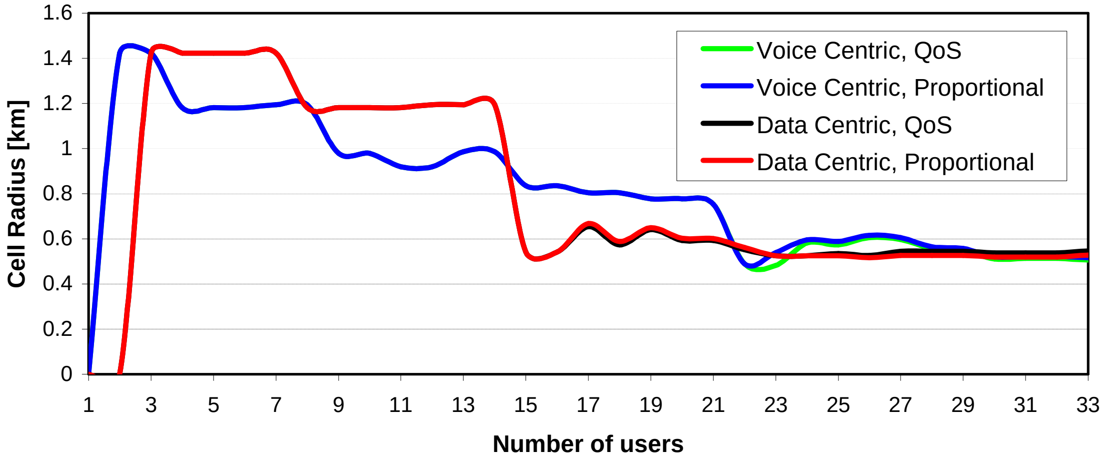


Figure C.10. Cell radius for Release 99, for the limitation of users rejected of 5%.

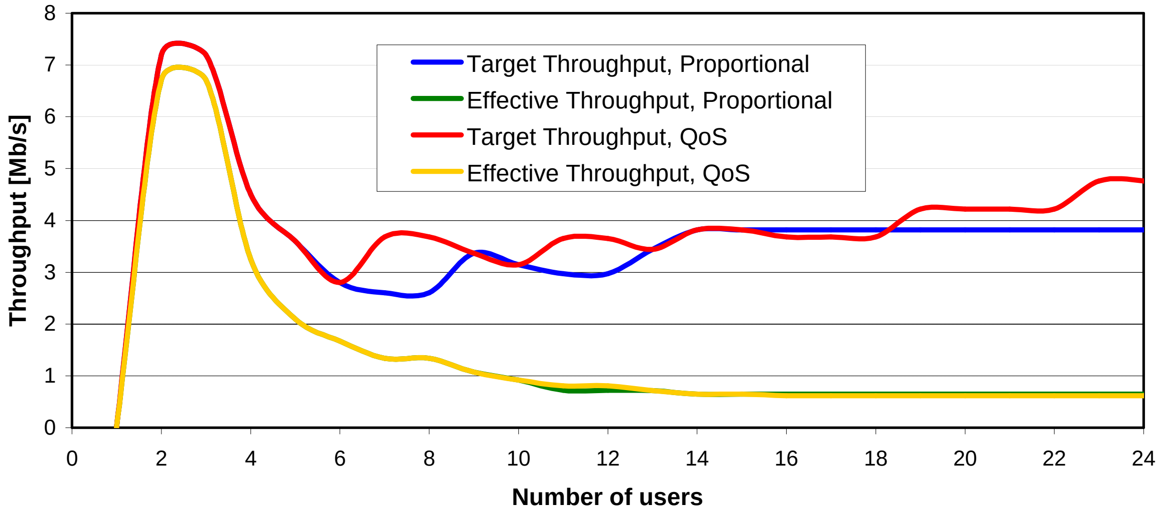


Figure C.11. Target and effective throughput for QoS priority policy, for the Data Centric profile, and for the limitation of users rejected of 5%.

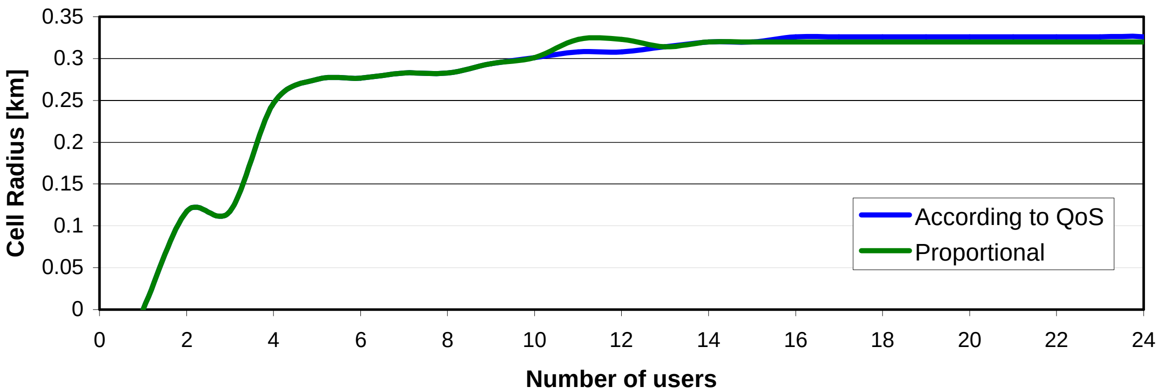


Figure C.12. HSDPA cell radius for Data Centric profile, and for the limitation of users rejected of 5%.

Annex D

Application and Profile Definition

In this annex, the application and profiles used do characterise the services used in OPNET are described.

The applications are used to set the characteristics of each service, and can be repeated during the profile according to the number of calls per hour presented on Table 4.1.The profiles associate the different applications to each user, according to the corresponding service. The chosen parameters are based on [Seba07].

Tables D.1 to D.6 present all the parameters used to define the applications, and Tables D.7 to D.12 define the profile parameters, for the OPNET simulator.

Table D.1. Voice application definitions.

Attribute	Value	
Name	VoIP	
Description	Voice	
Silence Length	Incoming Silence Length [s]	exponential (0.456)
	Outgoing Silence Length [s]	exponential (0.456)
Talk Spurt Length	Incoming Talk Spurt Length [s]	exponential (0.854)
	Outgoing Talk Spurt Length [s]	exponential (0.854)
Symbolic Destination Name	Voice Destination	
Encoder Scheme	G.729 A (silence)	
Voice Frames per Packet	1	
Type of Service	Interactive Voice (6)	
Traffic Mix (%)	All Discrete	
Signaling	None	
Compression Delay [s]	0.02	
Decompression Delay [s]	0.02	

Table D.2. Streaming application definitions.

Attribute	Value	
Name	Video Streaming	
Description	Video Conferencing	
Frame Interarrival Time Information	Incoming Stream Interarrival Time [s]	constant (0.04)
	Outgoing Stream Interarrival Time [s]	None
Frame Size Information	Incoming Stream Frame Size [Byte]	constant (2000)
	Outgoing Stream Frame Size [Byte]	constant (2000)
Symbolic Destination Name	Video Destination	
Type of Service	Streaming Multimedia (4)	
Traffic Mix (%)	All Discrete	

Table D.3. FTP application definitions.

Attribute	Value
Name	FTP
Description	Ftp
Command Mix (Get/Total)	0.95
Inter-Request Time [s]	exponential (600)
File Size [Byte]	uniform_int (100000, 5000000)
Symbolic Server Name	FTP Server
Type of Service	Standard (2)
Back-End Custom Application	Not Used

Table D.4. HTTP application definitions.

<table>
<tr><th>Attribute</th><th colspan="2">Value</th></tr>
<tr><td>Name</td><td colspan="2">HTTP</td></tr>
<tr><td>Description</td><td colspan="2">Http</td></tr>
<tr><td>HTTP Specification</td><td colspan="2">HTTP 1.1</td></tr>
<tr><td>Page Interarrival Time [s]</td><td colspan="2">exponential (39.5)</td></tr>
<tr><td rowspan="3">Page Properties</td><td>Object Size [Byte]</td><td>Number of Objects
(objects per page)</td></tr>
<tr><td>lognormal (20000, 50000)</td><td>constant (1)</td></tr>
<tr><td>lognormal (14400, 252000)</td><td>gamma (47.258, 0.232)</td></tr>
<tr><td rowspan="2">Server Selection</td><td>Initial Repeat Probability</td><td>Browse</td></tr>
<tr><td>Pages Per Server</td><td>exponential (10)</td></tr>
<tr><td>Type of Service</td><td colspan="2">Excellent Effort (3)</td></tr>
</table>

Table D.5. E-mail application definitions.

Attribute	Value
Name	Email
Description	Email
Send Interarrival Time [s]	exponential (360)
Send Group Size	uniform_int (1, 5)
Receive Interarrival Time [s]	exponential (360)
Receive Group Size	uniform_int (1, 5)
E-Mail Size [Byte]	lognormal (100000, 660000)
Symbolic Server Name	Email Server
Type of Service	Background (1)
Back-End Custom Application	Not Used

Table D.6. MMS application definitions.

Attribute	Value
Name	MMS
Description	Email
Send Interarrival Time [s]	exponential (1200)
Send Group Size	None
Receive Interarrival Time [s]	exponential (1200)
Receive Group Size	None
E-Mail Size [Byte]	uniform_int (30000, 300000)
Symbolic Server Name	Email Server
Type of Service	Best Effort (0)
Back-End Custom Application	Not Used

Table D.7. VoIP user profile definition.

<table>
<tr><th>Attribute</th><th colspan="3">Value</th></tr>
<tr><td>Profile Name</td><td colspan="3">VoIP user</td></tr>
<tr><td rowspan="5">Application</td><td>Name</td><td colspan="2">VoIP</td></tr>
<tr><td>Start Time Offset [s]</td><td colspan="2">No Offset</td></tr>
<tr><td>Duration [s]</td><td colspan="2">uniform_int (100,140)</td></tr>
<tr><td rowspan="2">Repeatability</td><td>Inter-repetition Time</td><td>exponential (600)</td></tr>
<tr><td>Number of Repetitions</td><td>Unlimited</td></tr>
<tr><td>Start Time [s]</td><td colspan="3">Uniform (0, 120)</td></tr>
<tr><td>Duration [s]</td><td colspan="3">End of Simulation</td></tr>
</table>

Table D.8. Streaming user profile definition.

<table>
<tr><th>Attribute</th><th colspan="3">Value</th></tr>
<tr><td>Profile Name</td><td colspan="3">Video Streaming user</td></tr>
<tr><td rowspan="5">Application</td><td>Name</td><td colspan="2">Video Streaming</td></tr>
<tr><td>Start Time Offset [s]</td><td colspan="2">No Offset</td></tr>
<tr><td>Duration [s]</td><td colspan="2">uniform (100,600)</td></tr>
<tr><td rowspan="2">Repeatability</td><td>Inter-repetition Time</td><td>exponential (1800)</td></tr>
<tr><td>Number of Repetitions</td><td>Unlimited</td></tr>
<tr><td>Start Time [s]</td><td colspan="3">Uniform (0, 120)</td></tr>
<tr><td>Duration [s]</td><td colspan="3">End of Simulation</td></tr>
</table>

Table D.9. FTP user profile definition.

Attribute	Value		
Profile Name	FTP user		
Application	Name	FTP	
	Start Time Offset [s]	No Offset	
	Duration [s]	End of Profile	
	Repeatability	Inter-repetition Time	exponential (1200)
		Number of Repetitions	Unlimited
Start Time [s]	Uniform (0, 120)		
Duration [s]	End of Simulation		

Table D.10. HTTP user profile definition.

Attribute	Value		
Profile Name	HTTP user		
Application	Name	HTTP	
	Start Time Offset [s]	No Offset	
	Duration [s]	End of Profile	
	Repeatability	Inter-repetition Time	exponential (900)
		Number of Repetitions	Unlimited
Start Time [s]	Uniform (0, 120)		
Duration [s]	End of Simulation		

Table D.11. E-mail user profile definition.

Attribute	Value		
Profile Name	Email user		
Application	Name	Email	
	Start Time Offset [s]	No Offset	
	Duration [s]	End of Profile	
	Repeatability	Inter-repetition Time	exponential (900)
		Number of Repetitions	Unlimited
Start Time [s]	Uniform (0, 120)		
Duration [s]	End of Simulation		

Table D.12. MMS user profile definition.

Attribute	Value		
Profile Name	MMS user		
Application	Name	MMS	
	Start Time Offset [s]	No Offset	
	Duration [s]	End of Profile	
	Repeatability	Inter-repetition Time	exponential (1800)
		Number of Repetitions	Unlimited
Start Time [s]	Uniform (0, 120)		
Duration [s]	End of Simulation		

The presented values are for the Light scenario, the only difference to the Heavy scenario is the inter-repetition time, presented in Table 4.1.

Annex E

Variation of the Number of Users per Scenario

In this annex, the difference distributions of the users per scenario is presented.

The number of effective calls per hour, l_{ef}, for each service, being:

$$l_{ef} = l \cdot N_s, \tag{E.1}$$

where:

§ N_s : Number of users of the corresponding service.

Table E.1 presents the effective number of calls per hour for Voice Centric and Data Centric scenarios and for the Light and Heavy profiles.

Table E.1. Effective number of calls per hour.

		VoIP		Streaming			FTP			HTTP			E-mail			MMS		
	N_u	**N**	L/H	**N**	L	H	**N**	L	H	**N**	L	H	**N**	L	H	**N**	L	H
Voice Centric	**10**	**5**	30	**1**	2	4	**2**	6	24	**1**	4	12	**1**	4	12			
	15	**7**	42	**1**	2	4	**3**	9	36	**2**	8	24	**2**	8	24			
	20	**10**	60	**1**	2	4	**4**	12	48	**2**	8	24	**2**	8	24	**1**	2	12
	25	**12**	72	**2**	4	8	**4**	12	48	**3**	12	36	**3**	12	36	**1**	2	12
	30	**15**	90	**2**	4	8	**5**	15	60	**4**	16	48	**3**	12	36	**1**	2	12
	35	**17**	102	**3**	6	12	**6**	18	72	**4**	16	48	**3**	12	36	**2**	4	24
	40	**20**	120	**3**	6	12	**7**	21	84	**3**	12	36	**5**	20	60	**2**	4	24
Data Centric	**10**	**2**	12	**1**	2	4	**3**	9	36	**2**	8	24	**2**	8	24			
	15	**3**	18	**2**	4	8	**4**	12	48	**3**	12	36	**2**	8	24	**1**	2	12
	20	**5**	30	**2**	4	8	**5**	15	60	**3**	12	36	**3**	12	36	**2**	4	24
	25	**6**	36	**3**	6	12	**6**	18	72	**4**	16	48	**4**	16	48	**2**	4	24
	30	**7**	42	**3**	6	12	**8**	24	96	**5**	20	60	**5**	20	60	**2**	4	24
	35	**8**	48	**4**	8	16	**9**	27	108	**6**	24	72	**5**	20	60	**3**	6	36
	40	**9**	54	**4**	8	16	**10**	30	120	**7**	28	84	**7**	28	84	**3**	6	36

Annex F

Parameter Definitions

In this annex, the parameters assessed with OPNET are defined.

As global parameters one has:

- § **Packet End-To-End Delay [for Video Conferencing application]:** The time taken to send a video application packet to a destination node application layer. This statistic records data from all the nodes in the network.
- § **Packet End-To-End Delay [for Voice application]:** The total voice packet delay, also called "mouth-to-ear" delay.
- § **Page Response Time [for HTTP application]:** Specifies time required to retrieve the entire page with all the contained inline objects.
- § **Download Response Time [for FTP application]:** Time elapsed between sending a request and receiving the response packet, measured from the time a client application sends a request to the server to the time it receives a response packet, i.e., time spent between sending the request and completing the download. Every response packet sent from a server to an FTP application is included in this statistic.
- § **Download Response Time [for E-mail and MMS application]:** Time elapsed between sending request for emails and receiving emails from email server in the network. This time includes signalling delay for the connection setup.

As node parameters one has:

- § **Total Number Requests Granted:** Total number of requests granted during admission control.
- § **Total Number Requests Queued:** Total number of requests queued during admission control.
- § **Node B Total DL Throughput:** Total throughput of the traffic sent by this Node-B in the DL direction (i.e. to the UEs camping in any cell/sector of this Node-B).
- § **Number DL Retransmission Required [per transport channel]:** Total number of transmissions required to successfully transmit a PDU in RLC acknowledged mode (AM).
- § **MT Total Received Throughput:** Total received throughput on the DL, i.e. from UTRAN to UE.

References

[3GPP06a] 3GPP, *Quality of Service (QoS) concept and architecture*, Report TS 23.107, Ver. 6.4.0, Mar. 2006 (http://www.3gpp.org).

[3GPP07] 3GPP Web site, (http://www.3gpp.org).

[ANAC07] ANACOM Web site, (http://www.anacom.pt).

[BWBW04] Buddendick,H., Wölfe,G., Burger,S. and Wertz,P., "Simulator for Performance Analysis in UMTS FDD Networks with HSDPA", *Proc. of PIMRC 2004 -15th IEEE International Symposium on Personal, Indoor and Mobile Radio Communications*, Barcelona, Spain, Sep. 2004.

[CoLa06] Costa,P. and Ladeira,D., *Planning of UMTS Cellular Networks for Data Services Based on HSDPA* (in Portuguese), Graduation Project, IST-TUL, Lisbon, Portugal, June 2006.

[Corr06] Correia,L.M., *Mobile Communication Systems – Course Notes*, IST-TUL, Lisbon, Portugal, Mar. 2006.

[DaCo99] Damasso,E. and Correia,L.M., *Digital Mobile Radio Towards Future Generation*, COST 231 Final Report, 1999 (http://www.lx.it.pt/cost231).

[EsPe06] Esteves,H. and Pereira,M., *Impact of intra- and inter-cellular interference on capacity in UMTS-FDD*, Graduation Project, IST-TUL, Lisbon, Portugal, June 2006.

[GoAg05] Golaup,A. and Aghvami,H., *A multimedia traffic modelling framework for simulation-based performance evaluation studies*, Computer Networks, vol. 50, London, UK, Nov. 2005.

[HoTo04] Holma,H. and Toskala,A., *WCDMA for UMTS*, John Wiley & Sons, Chichester, UK, 2004.

[HoTo06] Holma,H. and Toskala,A., *HSDPA/HSUPA for UMTS*, John Wiley & Sons, Chichester, UK, 2006.

[ITUT06] ITU-T Recommendation H.324, *Terminal for low bit rate multimedia communication*, Sep. 2005.

[Lope07] Lopes,J., *Performance of UMTS/HSDPA/HSUPA at the cellular level*, M. Sc. Thesis, IST-TUL, Lisbon, Portugal, Dec. 2007.

[Nort05] *The Nortel Technical Journal*, Issue 2, Nortel, July, 2005.

[OPNE07] OPNET Modeler 12.0 PL5 Documentation, OPNET Technologies Inc., Bethesda, Maryland, USA, 2007 (http://www.opnet.com).

[Pede05] Pedersen,K.I., “Quality Based HSDPA Access Algorithms”, in *Proc of VTC Fall 2005 – 62nd IEEE Vehicular Technology Conference*, Dallas, TX, USA, Sep. 2005.

[RiZh05] Rittenhouse,G. and Zheng,H., “Providing VoIP Service in UMTS-HSDPA with Frame Aggregation”, in *Proc. of ICASSP’05 – IEEE International Conference on Acoustics, Speech and Signal Processing*, Philadelphia, PA, USA, Mar. 2005.

[Salv07] Salvado,L. , *UMTS/HSDPA Comparison with WiMAX/IEEE802.16e in Mobility Scenarios*, M. Sc. Thesis, IST-TUL, Lisbon, Portugal, Dec. 2007.

[Seba07] Sebastião,D., *Algorithms for Quality of Service in a WiFi Network*, M. Sc. Thesis, IST-TUL, Lisbon, Portugal, Dec. 2007.

[TKSG04] Türke,U., Koonert,M., Schelb,R. and Görg,C., “HSDPA Performance Analysis in UMTS Radio Network Planning Simulations”, in *Proc. of VTC 2004 Spring – 59th IEEE Vehicular Technology Conference*, Milan, Italy, May 2004.

[UMFo07] UMTS Forum Web site, (http://www.umts-forum.org).

[Voda07] Private Communication, May 2007.

[WLTG06] Weerawardane,T.L., Li,X., Timm-Giel,A. and Görg,C., *Modeling and Simulation of UMTS HSDPA in OPNET*, in Proc. of OPNETWORK 2006, Washington, DC, USA, Sep. 2006.