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# Impact of HSUPA Implementation on UMTS Capacity and Cell Coverage

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*To my parents*

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# Abstract

This thesis analyses the HSUPA deployment on top of a UMTS network. A theoretical model was developed in order to predict the impact on capacity and cell coverage. The model outputs were then compared with 3G's OPNET Modeler simulator results.

The theoretical model was implemented in a simple C++ program. It is mainly focused on Admission Control algorithms, and supports multiservice traffic and dynamic carrier allocation. The network's behaviour analysis is made for an instant in time and for a single sector, addressing number of users' throughput, cell radius, and processed traffic. Release 99 was assessed with OPNET Modeler, a powerful simulator, which allows studying a huge set of parameters, like application's latency.

For HSUPA's theoretical study, the network's behaviour was analysed until it reaches saturation. It is important mentioning that the model respects the minimum requirements for applications. So, the main limitation is the loading factor and not the spreading factor. HSUPA allows good applications' performance, however, only few data users can be served in a single sector.

Using OPNET Modeler, the number of users was gradually increased, for lighter and heavier service scenarios. Network's saturation at the sector level occurs for around 15 users, with Interactive and Background classes being strongly penalised. Also network latency is too high, which has a huge impact on VoIP and other data applications.

## Keywords

WCDMA, HSUPA, OPNET Modeler, Multiservice Traffic, Capacity

# Resumo

Esta tese analisa a implementação de HSUPA sobre uma rede UMTS. Um modelo teórico foi desenvolvido de maneira a prever o impacto na capacidade e na cobertura. Estes resultados foram comparados com os de uma rede 3G simulada pelo OPNET Modeler.

O modelo teórico foi implementado em C++. Incide especialmente sobre algoritmos de controlo de admissão, suporta multi-serviço e portadora dinâmica. A análise de parâmetros sobre o comportamento da rede, como o número de utilizadores servidos e respectivo débito binário, o raio da célula e o tráfego processado, é efectuado para um instante de tempo e para uma célula com um único sector. A Release 99 foi implementada com o OPNET, um poderoso simulador que permite estudar uma grande gama de parâmetros, como a latência das aplicações.

No estudo teórico do HSUPA, o comportamento da rede foi analisado até atingir a saturação. É importante mencionar que o modelo impõe condições mínimas para o bom funcionamento das aplicações de dados, de maneira que a maior limitação é o factor de carga e não o de espalhamento. Embora o HSUPA permita bons desempenhos das aplicações, deve-se referir que só poucos utilizadores podem ser servidos nas condições anteriormente mencionadas.

Utilizando o OPNET, o número de utilizadores foi gradualmente aumentado, para um cenário mais leve e outro mais pesado. A saturação da rede ocorre para 15 utilizadores, sendo as classes Interactive e Background fortemente penalizadas. A latência da rede é bastante elevada, o que gera grande impacto no VoIP e nas outras aplicações.

# Palavras-chave

WCDMA, HSUPA, OPNET, Tráfego Multiserviço, Capacidade

# Table of Contents

# List of Figures

# List of Tables

# List of Acronyms

| | |
|---|---|
| 3GPP | 3rd Generation Partnership Project |
| ACK | Acknowledge |
| ARQ | Automatic Repeat Request |
| ATM | Asynchronous Transfer Mode |
| BPSK | Binary Phase-Shift Keying |
| BS | Base Station |
| BLEP | BLocking Error Probability |
| BLER | BLocking Error Rate |
| BCH | Broadcast Channel |
| BoD | Bandwidth on Demand |
| CDMA | Code Division Multiple Access |
| CN | Core Network |
| CPCH | Common Packet CHannel |
| CPICH | Common Pilot CHannel |
| CS | Circuit Switched |
| DC | Data Centric |
| DCH | Dedicated CHannel |
| DL | Downlink |
| DPCCH | Dedicated Physical Control CHannel |
| DPDCH | Dedicated Physical Data CHannel |
| DS-CDMA | Direct Sequence CDMA |
| DTX | Discontinuous Transmission |
| DVB-H | Digital Video Broadcasting - Handheld |
| E-AGCH | E-DCH Absolute Grant CHannel |
| E-DCH | Enhanced uplink Dedicated Channel |
| E-DPCCH | E-DCH Dedicated Physical Control CHannel |
| E-DPDCH | E-DCH Dedicated Physical Data CHannel |
| E-HICH | E-DCH Hybrid ARQ Indicator Channel |
| EIRP | Equivalent Isotropic Radiated Power |
| E-RGCH | E-DCH Relative Grant CHannel |
| FACH | Forward Access CHannel |
| FDD | Frequency Division Duplex |
| FRC | Fixed Reference Channel |

| | |
|---|---|
| FSM | Finite State Machine |
| GBR | Guaranteed Bit Rate |
| GGSN | Gateway GPRS Support Node |
| GMSC | Gateway MSC |
| GPRS | General Packet Radio System |
| GSM | Global System for Mobile Communications |
| HARQ | Hybrid ARQ |
| HLR | Home Location Register |
| HSDPA | High Speed Downlink Packet Access |
| HS-DSCH | High-Speed Downlink Shared Channel |
| HSPA | High Speed Packet Access |
| HSUPA | High Speed Uplink Packet Access |
| IP | Internet Protocol |
| IR | Incremental Redundancy |
| ISDN | Integrated Services Digital Network |
| LAN | Local Area Network |
| LTE | Long Term Evolution |
| MAC | Medium Access Control |
| ME | Mobile Equipment |
| MNT | Maximum Number of Transmissions |
| MSC | Mobile Services Switching Centre |
| MT | Mobile Terminal |
| MUD | Multiuser Detection |
| NACK | Negative ACK |
| OVSF | Orthogonal Variable Spreading Factor |
| PCH | Paging CHannel |
| PDU | Packet Data Unit |
| PSCH | Physical Shared CHannel |
| PS | Packet Switched |
| PSTN | Public Switched Telephone Network |
| QoS | Quality of Service |
| RAB | Radio Access Bearer |
| RACH | Random Access CHannel |
| RLC | Radio Link Control |
| RNC | Radio Network Controller |
| RNS | Radio Network Sub-system |
| RRM | Radio Resource Management |
| RSN | Retransmission Sequence Number |
| SF | Spreading Factor |
| SGSN | Serving GPRS Support Node |

| | |
|---|---|
| SNR | Signal to Noise Ratio |
| SPI | Scheduling Priority Indicator |
| TTI | Time Transmission Interval |
| UE | User Equipment |
| UL | Uplink |
| UMTS | Universal Mobile Telecommunication Services |
| USIM | UMTS Subscriber Identity Module |
| UTRA | UMTS Terrestrial Radio Access |
| UTRAN | UMTS Terrestrial Radio Access Network |
| VLR | Visitor Location Register |
| VoIP | Voice over IP |
| WCDMA | Wideband CDMA |
| WiFi | Wireless Fidelity |
| WiMAX | Worldwide Interoperability for Microwave Access |
| WLAN | Wireless LAN |

# List of Symbols

| | |
|---|---|
| $\alpha$ | DL orthogonality factor |
| $\Delta f$ | Signal bandwidth |
| $\eta_{DL}$ | DL load factor |
| $\eta_{UL}$ | UL load factor |
| $\nu$ | Activity factor |
| $r$ | Physical layer throughput |
| $r_a$ | Application layer throughput |
| $r_{UL}$ | Total uplink throughput |
| $l$ | Calls per hour |
| $l_H$ | Calls per hour for heavy profile |
| $l_i$ | Calls per hour for service $i$ |
| $l_L$ | Calls per hour for light profile |
| $d$ | Cell radius |
| $E_b$ | Energy per bit |
| $E_c$ | Energy per chip |
| $EIRP$ | Equivalent isotropic radiated power |
| $f$ | Frequency |
| $F$ | Receiver's noise figure |
| $G_{div}$ | Diversity gain |
| $G_e$ | Emitting antenna gain |
| $G_P$ | Processing gain |
| $G_r$ | Receiving antenna gain |
| $G_{rdiv}$ | Receiving antenna gain plus diversity gain |
| $G_{SHO}$ | Soft handover gain |
| $h_b$ | Base station height |
| $h_m$ | Mobile terminal height |
| $i$ | Inter to intra-cell interferences ratio |
| $L_0$ | Free space loss |
| $L_c$ | Cable losses between transmitter and antenna |
| $L_{int}$ | Indoor penetration losses |
| $L_p$ | Path loss between Node B transmitter and MT |
| $L_{tm}$ | Approximation for the multi-screen diffraction loss |

| | |
|---|---|
| $L_{tt}$ | Rooftop-to-street diffraction loss |
| $L_u$ | Body losses |
| $M_{FF}$ | Fast fading margin |
| $M_I$ | Interference margin |
| $M_{SF}$ | Slow fading margin |
| $N$ | Total noise power |
| $N_i$ | Number of users per service $i$ |
| $N_0$ | Noise spectral density |
| $N_s$ | Number of services |
| $N_{served}$ | Number of users served |
| $N_u$ | Number of users |
| $N_{u_{HSUPA}}$ | Number of HSUPA users |
| $N_{u_{R99CS}}$ | Number of R99 CS users |
| $N_{u_{R99PS}}$ | Number of R99 PS users |
| $P_r$ | Available receiving power at antenna port |
| $P_{Rx}$ | Received power at receiver input |
| $P_{Rx\min}$ | Receiver sensitivity |
| $P_t$ | Power fed to transmitting antenna |
| $P_{Tx}$ | Node B transmission power |
| $p_y$ | Probability for each service |
| $R_b$ | Bit rate |
| $R_c$ | Chip rate |
| $SF_{effective}$ | Effective Spreading Factor |
| $SF_{ratio}$ | Spreading Factor ratio |
| $SF_{target}$ | Target Spreading Factor |
| $T$ | Global traffic |
| $V_i$ | Mean file volume for service $i$ |

# 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.

## 1.1 Overview

Mobile Communication Systems caused a revolution in terms of mobility. People could be reachable and connected to the world with a simple cellular phone, anytime and anywhere. The First Generation was analogue and only allowed voice calls, but it was the beginning in personal and mobile communications. Then arrived Second Generation digital systems. There is no doubt Global System for Mobile communications (GSM) was the one. In 1990 the first releases were published and already in 1991 the first network was launched in Finland. GSM offered better voice quality, low-cost alternatives to making calls, like SMS, and also other data services for the first time. However, the data rates were very low, so new technologies like General Packet Radio Service (GPRS) and Enhanced Data rates for GSM Evolution (EDGE) appeared, enabling access to data networks and better multimedia services. There is no doubt GSM was (and still is) a huge success, as it is estimated 2 billion people worldwide use it.

As the interest of public grows around multimedia services, new technologies arise, providing access to those kinds of services with improved quality, which benefits both end users and providers. Looking at Figure 1.1, one sees the mobility and data rate of different wireless technologies, based on [Alte07]. One should also mention the increasing interoperability among these technologies.

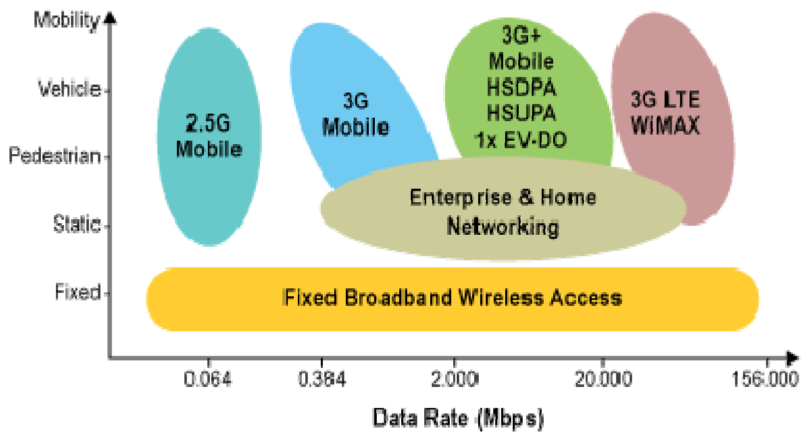



Figure 1.1 – Evolution of mobile wireless technologies (extracted from [Alte07]).

The request for improved networks supporting data communications was very high. So, the 3rd Generation Partnership Project (3GPP) published the first releases in 1999 for the Universal Mobile Telecommunication System (UMTS), which turned out to be the most global mobile access technology with deployments covering Europe, Asia, USA, and nowadays in huge markets like China, India and South America. UMTS networks are designed from the beginning for flexible delivery of any type of service, instead of being designed for efficient delivery of voice like GSM. UMTS represents an evolution in terms of capacity, data rates and new service capabilities, being based on Wideband

Code Division Multiple Access (WCDMA), which operates in a band around 2GHz (Europe).

From a technical point-of-view, WCDMA brings advanced capabilities, such as higher bit rates, Quality of Service (QoS) differentiation for higher efficiency of service delivery, simultaneous voice and data capability, low delays with packet round trip, and an essential inter-working with existing GSM/GPRS networks. Nevertheless, it also improves basic voice due to interference control mechanisms, like frequency reuse, fast power control and soft handover.

Although UMTS offered bit rates up to 384 kbps, they were not high enough for customers needs. In order to offer improved bandwidth to end-user, improved network capacity to the operator and improved interactivity for multimedia applications, 3GPP published Releases 5 and 6, also known as High Speed Packet Access (HSPA). Release 5 was first in March 2002 for the Downlink (DL), enabling peak rates initially up to 1.8 Mbps, increasing to 3.6 Mbps and 7.2 Mbps, and probably beyond 10 Mbps. In December 2004, Release 6 was set up for the Uplink (UL), also known as High Speed Uplink Packet Access (HSUPA). The first HSUPA networks are being deployed during 2007, achieving peak rates around 1-2 Mbps, and in a near future around 3-4 Mbps. It is very important mentioning the need for higher UL data rates, which HSUPA satisfies, as the HSDPA can not provide UL broadband. Higher data rates will enable more complex applications and a richer end user experience, as one can see in Figure 1.2, based on [FrSc07].

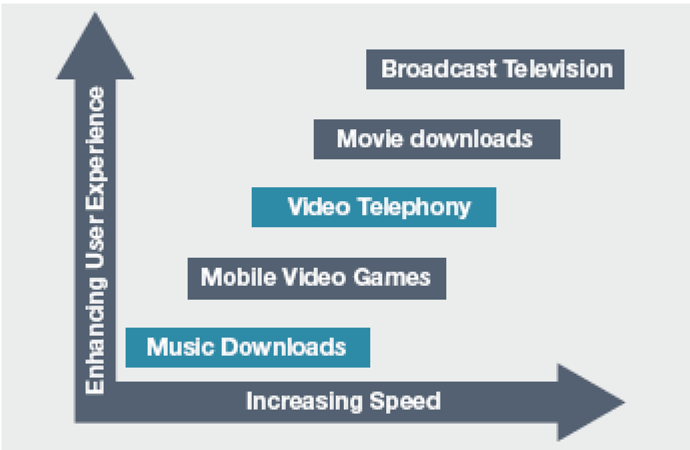



Figure 1.2 – Multimedia applications for HSPA (extracted from [FrSc07]).

HSPA is deployed on the top of WCDMA network, which means it is not as expensive as a standalone data network. It uses the same carrier, or for higher capacity and higher bit rates, it can use other carriers. It shares most of WCDMA network elements, just needing new software packages and enhanced Node B and RNC hardware.

HSUPA presents some new features like a new physical channel, the Enhanced Dedicated CHannel (E-DCH), fast physical layer Hybrid Automatic Repeat Request (HARQ), fast Node B based scheduling and a shorter UL Transmission Time Interval (TTI) length of 2 ms, while higher order modulation showed no potential gains.

## 1.2 Motivation and Contents

Although many studies and analysis about HSUPA have been done by manufacturers, operators and technical consortia, this thesis presents an independent view over this technology. The main objective is to study the impact of HSUPA implementation on UMTS capacity and cell coverage. In order to do so a theoretical model was developed based on a Admission Control algorithm for a single sector, and presents results for the number of users served, the achievable throughput, cell radius and traffic processed. Later in this work, the comparison between theoretical HSUPA results and OPNET Modeller simulation results is done. However, only Release 99 is simulated, as OPNET does not include HSUPA implementation.

This thesis is composed of 5 chapters, including introduction and conclusion, and they are summarised next.

Chapter 2 describes the UMTS basic concepts, mainly services and applications, network architecture, air interface, and capacity and interference. It does an overview over HSUPA, summarising important considerations for future work, like new channels, performance analysis and some radio resource algorithms. In the end, it presents HSUPA state of the art.

Chapter 3 describes the theoretical model and assesses scenarios and applications. It is also done the analysis of results focused on users' distribution, then moving to Release 99 and HSUPA parameters, and in the end it presents considerations about the global traffic on the network.

Chapter 4 is focused on the OPNET Modeller. First of all, it describes its main features and functionalities, and the UMTS Model as well. Considerations about scenarios, network architecture, applications and profiles are presented next. Node and Global results are processed and compared with HSUPA's theoretical model.

# 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. HSUPA is presented in Section 2.3, which describes the new channels, performance and radio resource management. Finally, in Section 2.4, the state of the art is presented.

## 2.1 The system

### 2.1.1 Services and applications

Some years ago, when UMTS started, several organisations attempted to regulate UMTS services, like 3GPP [3GPP99], UMTS Forum [UMTS99], ITU-T and ETSI, but only the first two had an important role.

3GPP QoS was designed to prioritise the different services according to their requirements. There are 4 different QoS classes, also referred to as traffic classes: Conversational, Streaming, Interactive and Background. There are some important factors that distinguish them, such as how delay-sensitive traffic is (the most important one), traffic delay, guaranteed bit rate, and different priority categories:

- Conversational: The most demanding class. It is mainly intended for real-time services, such as speech (CS or Voice over Internet Protocol (VoIP)) and video telephony, which means that a maximum end-to-end delay is needed, given by the human perception of video and audio conversation.
- Streaming: The main difference between this class and the former one is how delay sensitive traffic is. It is intended for services like video streaming (web broadcast or video on demand). This class tolerates some delay variations that are hidden by the jitter buffer in the receiver.
- Interactive: Applications for this class include web browsing, database retrieval, access of server, etc. Interactive traffic is a communication scheme that is characterised by the request response pattern of the end user. Round trip delay is the most important attribute, another important one being the error rate that should be very low in data transfer.
- Background: It is the less demanding class, assuming that the destination is not expecting the data within a certain time, which is why it is suitable for email.

Table 2.1 shows the main differences among the 3GPP classes.

According to [UMTS98], services can be divided into six different classes. The most relevant parameters of the UMTS Forum classes are user’s bit rate, asymmetry factors (the relation between downlink (DL) and uplink (UL) traffic), and switching mode. Although, response and delay time requirements were not taken into account, they will affect the mode of operation at the air interface, and will consequently impact on the efficiency. This perspective became somehow inadequate, and is no longer used.

Table 2.1 - Services and Applications according to 3GPP (extracted from [3GPP99]).

| Service Class | Conversational | Streaming | Interactive | Background |
|---|---|---|---|---|
| Real Time | Yes | Yes | No | No |
| Symmetric | Yes | No | No | No |
| Switching | CS | CS | PS | PS |
| Guaranteed Rate | Yes | Yes | No | No |
| Delay | Minimum Fixed | Minimum Variable | Moderate Variable | High Variable |
| Buffer | No | Yes | Yes | Yes |
| Bursty | No | No | Yes | Yes |
| Example | Speech | Web Broadcast | www | Email |

## 2.1.2 Network architecture

This section gives an overview of the UMTS architecture, which uses the same architecture of GSM. The system consists of three functional network elements [HoTo04]:

- UMTS Terrestrial Radio Access Network (UTRAN)
- Users Equipment (UE)
- Core Network (CN)

The first two network elements have protocols created for UMTS, which means that the design is based on the needs of WCDMA. CN is an upgrade from the current GSM/GPRS version.

Figure 2.1 shows the UMTS network architecture.

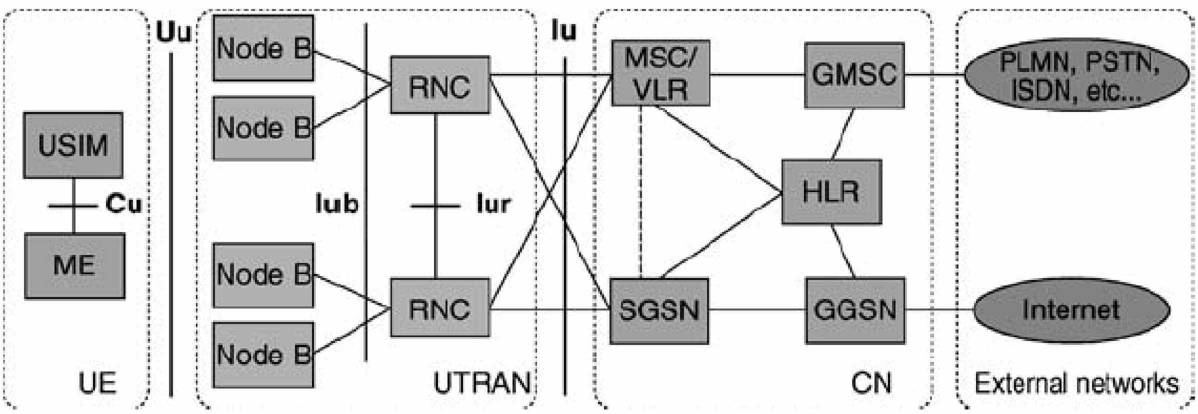



Figure 2.1 - UMTS network architecture (extracted from [HoTo04]).

The UE interfaces the user with the radio interface. It consists of two parts:

- The Mobile Equipment (ME) is the mobile radio terminal used for radio communication over the Uu interface, i.e. the Mobile Terminal (MT);

- The UMTS Subscriber Identity Module (USIM), which is a smart card that holds the subscriber identity, performs authentication algorithms, and stores authentication and encryption keys and some subscription information that is needed at the terminal.

UTRAN handles all radio-related functionalites. The main task of UTRAN is to create and maintain Radio Access Bearers (RABs) for communication between UE and CN. It consists of one or more Radio Network Sub-systems (RNSs). Each RNS has one Radio Network Controller (RNC) responsible for managing the radio resources in its domain for one or more Nodes B, the Base Stations (BSs). The Nodes B and the RNC are connected to each other by the Iub interface. RNCs may be connected by the Iur interface. Uu interface connects the UE and the UTRAN, while the Iu interface connects the UTRAN to the CN.

UTRAN respects the main requirements for the design of the architecture, functions and protocols. They can be summarised as: maximisation of the commonalities with GSM; maximisation of the commonalities in the handling of CS and PS data; use of the Asynchronous Transfer Mode (ATM) transport; use of the IP-based transport as an alternative transport; and support of soft handover and of WCDMA-specific Radio Resource Management (RRM) algorithms.

As mentioned before, UMTS CN is an upgrade from GSM's. The CN is responsible for switching and routing calls and data to the external networks, such as the Internet (PS network), the Public Service Telephone Network (PSTN) and the Integrated Services Digital Network (ISDN) (these last two are CS networks). The main elements are the Home Location Register (HLR), the Mobile Services Switching Centre/Visitor Location Register (MSC/VLR) for the CS domain, the Gateway MSC/Serving GPRS Support Node (GMSC/SGSN) for the PS one, and the Gateway GPRS Support Node (GGSN).

### 2.1.3 Air Interface

In UMTS, the air interface is based on WCDMA [HoTo04], which is a wideband Direct-Sequence Code Division Multiple Access (DS-CDMA), meaning that user information bits are spread over a wide bandwidth by multiplying the user data with quasi-random bits (called chips) derived from CDMA spreading codes. This requires a variable spreading factor and multicode connections in order to support variable bit rates, especially the higher ones. The chip rate is 3.84 Mcps, leading to a carrier bandwidth of approximately 4.4MHz, adjustable in 200 kHz steps. Table 2.2 shows the bands for UMTS Europe:

Table 2.2 - European UMTS spectrum (extracted from [HoTo04]).

| | Uplink | Downlink | Total |
|---|---|---|---|
| UMTS-Frequency Division Duplex (FDD) | 1920-1980 (MHz) | 2110-2170 (MHz) | 120 MHz |

WCDMA accepts Bandwidth-on-Demand (BoD), which means that it supports highly variable user data rates. The data rates are not constant during the transmission. In fact, they can change every 80, 40, 20 and 10ms frame. However, the data capacity among users can change from frame to frame.

The data generated at higher layers is carried over the air interface with transport channels, which are mapped onto the physical layer by different physical channels. There are two different types of transport channels, which are known as common and dedicated ones. The main difference between them is that a common channel is shared among a group of users in a cell, whereas a dedicated channel is reserved for a single user only.

There are six different common channels defined for UTRA in Release '99 [HoTo04], Table 2.3. There is one more common channel, defined later on in Release 5 [HoTo06], being called High-Speed Downlink Shared Channel (HS-DSCH). Common channels do not support soft handover, but some of them have fast power control.

DCH is the only dedicated transport channel carrying all the information intended for a given user coming from layers above the physical one. The DCH is mapped onto physical channels: the Dedicated Physical Data Channel (DPDCH) carries higher layer information, including users' data, while the Dedicated Physical Control Channel (DPCCH) carries the necessary physical layer control information. The DCH supports some features, such as, fast power control, fast data rate change, and soft handover.

Table 2.3 - Common Channels (extracted from [HoTo04]).

| Channel | Link | Description |
|---|---|---|
| Broadcast Channel (BCH) | UL | Transmits information specific of UTRAN for a given cell |
| Forward Access Channel (FACH) | DL | Carries control information to MTs known to be located at a given cell |
| Paging Channel (PCH) | DL | Carries data for the paging procedure (network wants to initiate communication with the MT) |
| Random Access Channel (RACH) | UL | Carries control information from the MT |
| Uplink Common Packet Channel (CPCH) | UL | An extension to the RACH that is intended to carry packet-based user data |
| Downlink Shared Channel (PSCH) | DL | Transport channel intended to carry dedicated user data and/or control information |

The transmission of the physical channels consists of three main procedures: spreading, scrambling and modulation.

The spreading operation is the multiplication of each user bit data by a sequence of bits called chips (spreading code). The resulting spread data has the same random appearance as the spreading code. This process is applied to physical channels, consisting of two operations, Channelisation and Scrambling, Figure 2.2.

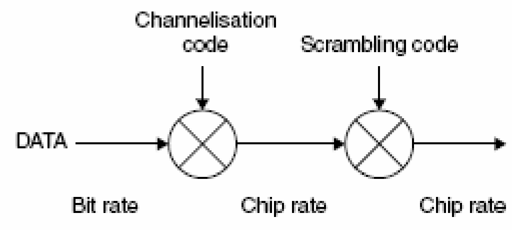



Figure 2.2 - Relationship between spreading and chip rate (extracted from [HoTo04]).

Channelisation codes are used to separate transmissions from a single source, which means DL connections within one sector and the dedicated physical channel in the UL. They are based on the Orthogonal Variable Spreading Factor (OVSF) technique, which allows the spreading factor to be changed, and the orthogonally between different spreading codes of different lengths to be maintained. These codes are picked up from the channelisation code tree. The data rate depends on the spreading factor.

Scrambling codes are used to separate MTs or BSs from each other. They are used on top of spreading, so that the signal bandwidth is not changed, only making the signals from different sources separable from each other. This allows using identical spreading codes for several transmitters. The symbol rate is not affected by the scrambling (see Figure 2.2).

As mentioned before, one of the main features of WCDMA is Power Control. Without it, a single MT could block a whole cell, because of the near-far problem of CDMA. In WCDMA, MTs adaptively adjust their power level so as not to swamp all the others in the system. There are two different types of power control:

1. Open loop power control is only used to provide a coarse initial power setting of the MT at the beginning of a connection.
2. Fast closed loop power control is mainly used in UL, but it can also be used in DL (there is no near–far problem due to the one-to-many scenario).

### 2.1.4 Capacity and Interference

While planning capacity, it is important to remember that all users share the same frequency, which means interference among them. So, capacity is affected by the number of users in the cell and the requested service.

Since there is asymmetry between UL and DL, there are different load factors. These parameters are used for predicting cell capacity and planning noise raise. They are given by the following equations [HoTo04]:

$$h_{UL} = (1+i)\sum_{j=1}^{N_u} \frac{1}{1+\dfrac{R_c / R_{b_j}}{\left(E_b/N_0\right)_j \cdot n_j}} \tag{2.1}$$

$$h_{DL} = \sum_{j=1}^{N_u} n_j \times \frac{\left(E_b/N_0\right)_j}{R_c / R_{b_j}}\left[\left(\left(1-\alpha_j\right)+i_j\right)\right] \tag{2.2}$$

where:

- $N_u$ : number of active users per cell;
- $\left(E_b/N_0\right)_j$ : energy per user bit divided by the noise spectral density for user *j*;
- $n_j$ : activity factor of user *j* at physical layer;
- $R_c$ : WCDMA chip rate;
- $R_{b_j}$ : bit rate of user *j*;
- $i$ : ratio of inter to intra-cell interferences;
- $a_j$ : average orthogonality factor in the cell;
- $i_j$ : ratio of inter-to-intra-cell interferences for user *j*.

The load equation predicts the amount of noise rise over thermal noise due to interference. The noise rise is equal for both links, being given by:

$$M_{I_{UL,DL}[\mathrm{dB}]} = -10 \cdot \log(1 - h_{UL,DL}) \tag{2.3}$$

The total power received at the BS is given by:

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

where:

- $N_0$ is the noise spectral density of the MT equipment
- $\overline{L_{p_j}}$ is the average cell path loss

## 2.2 HSUPA

### 2.2.1 New features

In order to offer improved bandwidth to the end-user, improved network capacity to the operator and improved interactivity for data applications, a new technology arrived. It is referred to as High Speed Packet Access (HSPA), and it is an upgrade from the initial version of WCDMA.

HSPA was standardised as part of 3GPP Release 5 [HoTo06] and Release 6 [HoTo06]. Release 5 refers to improvements made in DL, referred as High Speed Downlink Packet Access (HSDPA). Release 6 refers to the improvements made in UL, referred as High Speed Uplink Packet Access (HSUPA) and also known as Enhanced Dedicated Channel (E-DCH).

HSUPA had its commercial launch in 2007, with peak data rates around 1-2 Mbps, and in a near future around 3-4 Mbps.

It is important to refer the key technical capabilities introduced by HSUPA, i.e., dedicated UL channel E-DCH, the introduction of Hybrid Automatic Repeat Request (HARQ), and the fast Node B scheduling.

Table 2.4 shows the main differences between DCH (WCDMA), HSDPA and HSUPA. As one can see, HSUPA does not support adaptive modulation like HSDPA – it uses BPSK. Instead, it uses a variable spreading factor in order to achieve variable data rates.

Table 2.4 - HSDPA, HSUPA and DCH comparison table [HoTo06].

| Feature | DCH | HSDPA (HS-DSCH) | HSUPA (E-DCH) |
|---|---|---|---|
| Variable Spreading Factor | Yes | No | Yes |
| Fast Power Control | Yes | No | Yes |
| Adaptive modulation | No | Yes | No |
| Node B based scheduling | No | Yes | Yes |
| Fast L1 HARQ | No | Yes | Yes |
| Soft handover | Yes | No | Yes |
| TTI length [ms] | 80, 40, 20, 10 | 2 | 10, 2 |

Because in a cellular system the UL connection is many-to-one, HSUPA remains based on a dedicated channel. Though, a new UL transport channel was created, being known as E-DCH, supporting fast physical layer HARQ, fast Node B scheduling, and even a shorter Transmission Time Interval (TTI) of 2 ms.  A series of new channels are introduced for both signalling and traffic to improve overall UL capabilities, Figure 2.3.

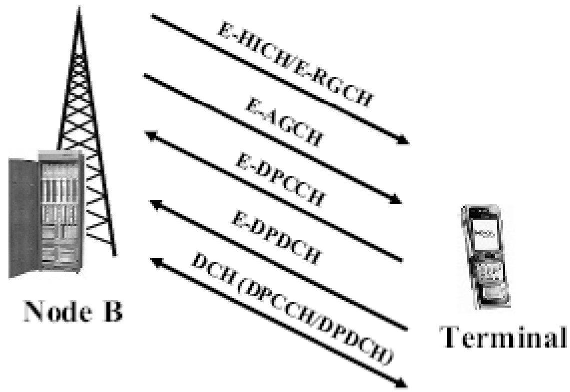



Figure 2.3 - Channels needed for HSUPA DL is in Release 99 DCH [HoTo06].

The new HSUPA channels are:

- E-DCH Dedicated Physical Data Channel (E-DPDCH): carries user's data, being parallel to all the UL dedicated channels released before. It is very similar to DPDCH of Release 99. They both support OVSFs and BPSK modulation, transmit multiple channels in parallel in order to go beyond the data rate that one physical channel can support, and even follow the same fast power control loop. The biggest difference for E-DPDCH is the support of the SF2. E-DPCCH supports simultaneous transmission of two SF2 and two SF4 (DPCCH supports six parallel SF4 codes) and the new TTI length of 2 ms.
- E-DCH Dedicated Physical Control Channel (E-DPCCH): like DPCCH, it delivers the information needed to decode the corresponding data channel transmission. The big difference between them is that E-DPCCH provides only information about E-DPDCH. E-DPCCH has only one possible slot format, SF256, and is capable of delivering 10 information bits coded to 30 physical channel bits in an E-DPDCH's 2ms frame. The 10 information bits consist of a 7 bit E-DCH transport format combination indicator (E-TFCI), a 2 bit retransmission sequence number (RSN), and the happy bit that indicates if the UE is happy with the current data rate.
- E-DCH HARQ Indicator Channel (E-HICH): this new DL physical channel transmits HARQ feedback information. It is used for transmitting positive (ACK) or negative (NACK) acknowledgments for UL packet transmission and has BPSK modulation. If the Node B receives TTI correctly or incorrectly, responds with ACK (+1) or NACK (-1). If TTI is not detected, then it does nothing (Discontinuous Transmission (DTX)). Each E-HICH transmits one bit of information in 3 slots, in case of 2 ms TTI, or in 12 slots, in case of 10 ms TTI length. The spreading factor is 128.
- E-DCH Absolute Grant Channel (E-AGCH): this new DL physical channel is used for transmitting Node B scheduler's decision about UE's relative transmission power it is allowed to use for E-DPDCH. This means, the allowed maximum transmission data rate it may use. E-AGCH uses a fixed SF of 256. The absolute grant consists of a 5 bit grant

value, indicating the exact power level for E-DPDCH may use in relation to DPDCH, and one bit indicating the scope of the grant (absolute grant is valid for one specific HARQ process or for all of them). There is also a primary and a secondary UE-id for identifying the intended receiver and delivering one additional bit of information.

- E-DCH Relative Grant Channel (E-RGCH): DL channel used for adjusting the UL rate up/down. This relative grant can take values UP (+1), DOWN (-1) or HOLD (0). It has exactly the same channel structure as E-HICH. The exception is that E-RGCH is transmitted from cells not belonging to the serving E-DCH radio link set. It always transmits a 10 ms long message.

### 2.2.2 Performance analysis

When discussing HSUPA performance analysis, one must consider parameters like bit rate, capacity and coverage. It is also very important to keep in mind the dependence on the selected scenarios and on deployment and service parameters. The most important performance metrics are average cell throughput (amount of data in a cell in a certain amount of time), 10% packet call throughput (90% of end users will experience a better packet call throughput then this), the required power per information bit over noise ($E_b/N_0$), and the required power per chip bit over noise ($E_c/N_0$).

There are six different UE categories defined for HSUPA. The categories differ mainly in terms of the maximum of E-DPDCH codes, the minimum spreading factor transmitted, and the supported TTI length. These categories are shown in Table 2.5.

Table 2.5 - HSUPA terminal categories [HoTo06].

| Category | Maximum number of E-DPDCH and smallest spreading factor | Supported TTIs [ms] | Maximum data rate with a 10 ms TTI [Mbps] | Maximum data rate with a 2 ms TTI [Mbps] |
|---|---|---|---|---|
| 1 | 1×SF4 | 10 | 0.72 | N/A |
| 2 | 2×SF4 | 2 and 10 | 1.45 | 1.45 |
| 3 | 2×SF4 | 10 | 1.45 | N/A |
| 4 | 2×SF2 | 2 and 10 | 2 | 2.91 |
| 5 | 2×SF2 | 10 | 2 | N/A |
| 6 | 2×SF2+2×SF4 | 2 and 10 | 2 | 5.76 |

Although there are 6 HSUPA terminal categories, the following considerations are based on the assumptions agreed in the 3GPP for testing purposes. So, Fixed Reference Channels (FRC) are used in this chapter. Table 2.6 shows FRCs defined for E-DCH.

Figure 2.4 shows the data rate for FRC 5 (first-phase HSUPA UE), FRC 2 and FRC 6 (these last two

FRCs represent future releases) without power control. $E_b/N_0$ must achieve high values in order to reach high data rates, for example, one needs an $E_c/N_0$ higher then 0 dB to reach 2 Mbps.

Table 2.6 - FRCs defined for E-DCH [HoTo06].

| FRC | TTI length [ms] | Codes | Maximum bit rate [kbps] | UE category |
|---|---|---|---|---|
| 1 | 2 | 2×SF4 | 1353 | 2 |
| 2 | 2 | 2×SF2 | 2706 | 4 |
| 3 | 2 | 2×SF4+2×SF2 | 4059 | 6 |
| 4 | 10 | 1×SF4 | 508 | 1 |
| 5 | 10 | 2×SF4 | 980 | 2 and 3 |
| 6 | 10 | 2×SF2 | 1960 | 4 and 5 |
| 7 | 10 | 1×SF16 | 69 | 1 |

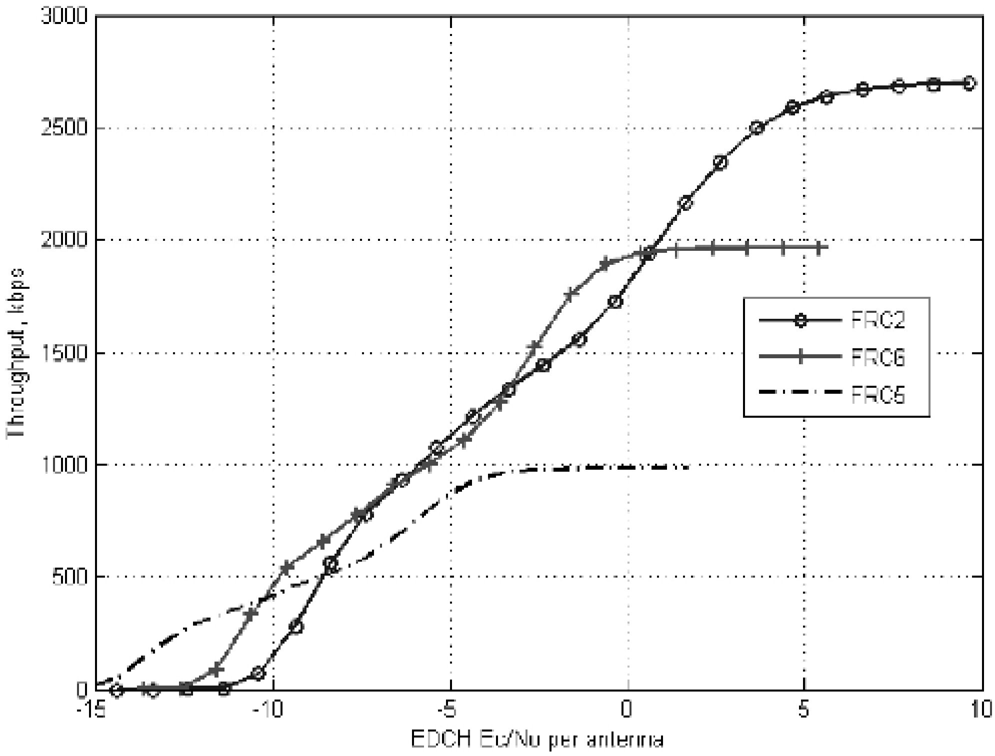



Figure 2.4 - HSUPA throughput in Vehicular A at 30 Km/h [HoTo06].

Other important information is the relation between $E_c/N_0$ and the noise rise caused by a single user, Figure 2.5.

One can see that the uplink noise rise for a single user reaches higher values when $E_c/N_0$ increases. Considering the former example, for an $E_c/N_0$ equal to 0 dB the noise rise should be 3 dB.

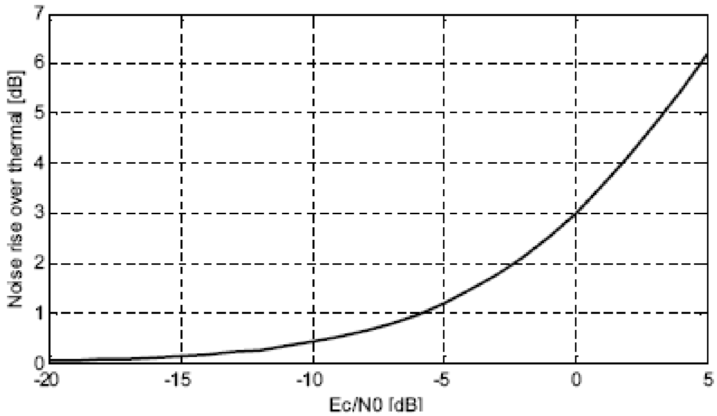



Figure 2.5 - Uplink noise rise caused by single user as a function of required $E_c/N_0$ [HoTo06].

It is very important to keep in mind that the cell coverage area decreases for other simultaneous users when noise rise increases. So, it may happens that high $E_c/N_0$ are not possible for macro-cells.

Concerning interference-limited cell capacity due to L1 HARQ and to Node B based scheduling, there is a capacity gain from HSUPA.

The introduction of L1 HARQ allows faster retransmissions, so, one has a higher retransmission probability while maintaining the same user delay performance, which means a decrease in the required $E_b/N_0$, hence, it is possible to increase the spectral efficiency. One can achieve cell throughput gains from L1 HARQ between 15% and 20%, assuming realistic operations.

Node B based scheduling with L1 control signalling provides the two main advantages: tighter power control of total received UL power, and faster reallocation of radio resources among users. These improvements allow an increase of the overall QoS and a reduction in the required power headroom to prevent the system from entering in an unwanted load region. Hence, it is possible to improve the cell throughput performance. Comparing with RNC-based scheduling, it is expected to observe gains between 6% and 9%.

## 2.2.3 Radio Resource Management

Release 5 and Release 6 are responsible for huge enhancements in Radio Resource Management (RRM) algorithms. Hybrid ARQ, Node B based scheduling and even shorter TTI provide improved performance for packet data services in terms of reduced delays, higher data rates and increased cell capacity. RRM enhancements occur in functions located in the RNC, Node B and UE. Figure 2.6 shows the HSUPA RRM architecture in Release 6:

The main RNC algorithms are resource allocation, QoS parameterisation, and admission control. Although being important, RNC mobility management is out of range of this dissertation.

Resource allocation is in RNC's algorithm responsible for establishing the maximum received wideband power for the Node B. This received power is the sum of thermal noise, inter-cell interference, intra-cell interference from DCH (scheduled and non scheduled connections) and intra-cell interference from E-DCH connections. The RNC can also send a congestion indication to the Node B, whether there is no congestion, delay build-up or even lost packets, for a certain UE. The Node B can then lower the bit rate of at least this particular user, until the congestion situation is solved.

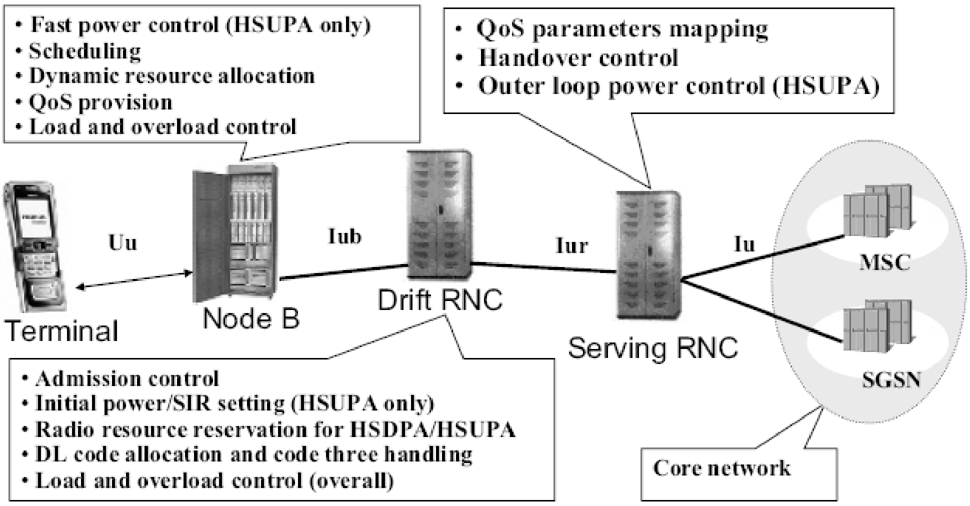



Figure 2.6 - HSUPA RRM architecture in Release 6 [HoTo06].

QoS parameterisation is a set of parameters given to the Node B that can be used in packet scheduling:

- Scheduling Priority Indicator (SPI) takes values between 0 and 15, where the higher number gets the higher priority;
- Guaranteed Bit Rate (GBR) indicates the guaranteed number of bites per second to be delivered over the air interface;
- Maximum Number of Transmissions (MNT) defines the maximum number for L1 HARQ transmissions.

The admission control algorithm is responsible for accepting, or not, a new user to HSUPA. The RNC needs information about the number of active HSUPA users, the UL interference levels, the SPI, the GBR, the provided bit rate for each E-DCH class, the provided bit rate on the DCH and of course the DL limitations. If there is a new user trying to access the cell, he/she will send a common pilot channel (CPICH) $E_c/N_0$ measurement to the RNC, so that it can estimate if there is HSUPA capacity available without changing other users' QoS requirements.

HARQ and packet scheduling are the two main Node B's algorithms, having a very important role in HSUPA.

The Hybrid-ARQ (HARQ) algorithm uses the turbo-codes and allows the Node B to rapidly request retransmission of erroneously received data, substantially reducing delay, and also to combine information from the original transmission with that of later retransmissions. HARQ can operate in two different ways: chase or soft combining if one has the same rate matching between retransmissions; non-identical retransmission, also known as 'incremental redundancy' (IR), if the retransmissions have different parity bits from the original transmission. Node B will continue requesting retransmission until the packet is received correctly or the maximum number of transmissions is achieved. HSUPA HARQ is fully synchronous, supports soft handover and even with IR versions retransmissions can be predetermined.

Like HSDPA, HSUPA scheduling was moved from the RNC to the Node B. This allows scheduling decisions with lower latency and closer to the radio interface. Node B scheduling manages the shared UL resource, UL noise rise, makes sure the HSUPA system does not overload and even tries to maintain UL capacity close to the maximum. So, the Node B can upgrade or can downgrade UE capacity allocation: it will upgrade based on happy bit and on UE available transmission power, and it will downgrade if capacity grants are not maximised and if channel utilisation is too low. Capacity grants can be absolute or relative and they are both DL channels.

## 2.3 State of the art

Nomor Research, a company in the area of real time system emulation and specialised in the implementation of radio access networks, proposed, [SeAL06], a dynamic Real-time Network Simulator Platform (RealNeS) that is capable of simulating any kind of packet-switched network in real-time, and also supports several wireless systems, such as UMTS and WiMAX. It is a powerful tool that allows evaluating the performance of all types of IP-based multimedia services, like video Streaming, VoIP, Gaming, WWW, and E-mail. Hence, it is possible to maximise end user perception in a network and optimise RRM, in order to improve QoS. This platform also allows developing new protocols and scheduling algorithms. The network simulator is divided into three components: the server, the client and the network client. Each of them is implemented in a single PC, connected to the others through Ethernet or Wireless Local Access Networks (WLANs). It supports several clients, and each of them mayt use multiple applications in parallel, which means different parameter settings, like user topology, flow priorities, and link quality, being monitored in the user interface. The system level simulator supports simulation of a whole mobile environment with specific cellular layout, base station layouts and transmitter configurations. In UL $E_c/N_0$ is computed considering path loss, shadowing, fast fading, and intra- and inter-cell interferences. Several output parameters can be analysed, such as throughput, delay and Block Error Probability (BLEP). Protocol-specific statistics are also possible, e.g., number of retransmissions, interference values, power measurements, etc.

A recent Ph.D. work on heterogeneous services for HSUPA, [Das06], provided valuable data for this thesis. A discrete-event simulation model was created in order to analyse the effect of physical parameters, soft parameters and the different RRM strategies adopted. A single carrier WCDMA system with uniformly distributed users has been chosen. Cells are hexagonal and the omni-directional Node Bs' antennas are at the centre of the cell (a 19 cell system was studied). Although the effect of mobile velocity is considered, user locations are not changed in order to avoid the need for simulating multiple radio links per user and soft handover, hence, slow fading was excluded from the simulation model. Fast power control at 1500Hz and both 2ms and 10ms TTIs were adopted for the simulations. Different services are considered for simulation purposes: speech, video and data are distributed uniformly in a cell. The simulation was optimised for MPEG-4 coded video. In order to simulate these services, a random number generator capable of generating several independent random streams is used to generate packet sizes, inter-arrival times, and talk spurt windows. The following outputs were considered:

- The effects of the rate upgrade threshold and the rate scheduling period on the proposed RRM strategy.
- The effects of the target Signal to Noise Ratio (SNR) and the error coding scheme in terms of multiplexing efficiency and system throughputs, packet transmission delay, and output power.

In [WBMB06], a dynamic network simulator is proposed, used to analyse the effect on cell throughput of increasing the BLock Error Rate (BLER) target and the target number of transmissions. In order to simulate a large variety of user behaviours and propagation environments, this simulator includes parameterised models of data traffic, user mobility and radio wave propagation. Because power levels are constant during one slot, the time resolution is 667µs (1 slot). The receiver chain is modelled according to Actual Value Interface tables (see [HSHL04]) that map signal to noise levels onto frame error probabilities. Maximum ratio combining is used in case of Rx diversity. Although it is possible to simulate all kinds of traffic, the only traffic considered in this study was MMS. It is described as an exponentially distributed packet call size with a mean of 12.7 kB, a 5 ms cutoff, and no TCP protocol effects included. It has been concluded that the increase of BLER and Redundancy Sequence Numbers (RSN) targets, which together control the average number of transmissions, lead to a lower required $E_b/N_0$, hence, a higher L3 cell throughput. This means that more capacity is available, which can be used by either increasing the bit rates of the existing users or allowing more users into the system.

Another interesting work about HSUPA is presented in [ChKM06]. It studies the system performance of VoIP on HSUPA, which is investigated by semi-analytical prediction and system level simulations; the former shows the capacity gain of VoIP over HSUPA to 12.2 kbps on DCH in Release 99, while the latter confirms semi-analytical results and analysis and even presents more details. It is expected that enhanced RRM algorithms provide a lower required UE power and then increase capacity. Nevertheless, due to VoIP sensitivity, a compromise between HARQ retransmissions and delay budget is necessary, in order to optimise the capacity of VoIP on HSUPA. A quasi-static level simulator, where all necessary RRM algorithms are modelled in order to investigate the performance of VoIp on HSUPA, is used. This tool includes a detailed simulation of the users within multiple cells. Fast fading is modelled according to the ITU Vehicular-A profile. Otherwise, it is simulated according to 3GPP2 Technical Specification Group C. The duration of each VoIP call is described by a negative exponential distribution function with an average call length of 60 s. Transmission and silent periods have a probability of 50 %, and both are negative exponentially distributed with a 3 s average. The VoIP packet rate is approximately 12.2 kbps and arrives directly at the Node B every 20 ms (during on periods).

Another work was developed by [Rodr06]. Link level simulations in Matlab for HSUPA applications, based on ITU macro-cellular channels, were done. The increase of throughput, due to Multi-User Detection (MUD) schemes with Interference Cancellation (IC) and Node B controlled HARQ was studied. Simulations were done considering ITU Vehicular-A channels, with different mobility scenarios and with a packet rate of 64 kbps.

Although there is plenty of work done on simulating HSUPA, several things have not been considered yet, e.g., different users profile. Besides that, OPNET still has to work on HSPA software. One should keep in mind that the first HSDPA release was presented only in OPNETWORK 2006. So, this is only the kick off!

# Chapter 3

# Capacity and Coverage 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 study the impact of HSUPA implementation on UMTS capacity and cell coverage a simple theoretical model was developed. It is mainly based on the Admission Control algorithm for a single sector, and it presents results about the network's behaviour, like: number of served users; available throughput per user; maximum cell radius. Outputs are analysed for different scenarios, environments and algorithms calculations being based on the link budget presented in Annex A and described in the following pages. As discussed before, there are two limitations for the network: the number of available codes and the load factor. Although the load factor has been increased for HS technology, it is still the main limitation. The following results are a reflection of those limitations. There can be a third limitation to the number of users: the number of users served in DL, which it is not studied in this thesis.

HSUPA's snapshot analysis it is an improvement from the work by developed [Lope07] and [Salv07]. They have created a monoservice snapshot analysis that supports R99, HSDPA and HSUPA users. This work presents the following features: multiservice (CS voice and video-telephony, and PS streaming, FTP, HTTP, e-mail and MMS) and dynamic carrier (in order to support, at the same time, both R99 and HSUPA clients) for the UL. [Antu07] developed the calculations for DL i.e. for HSDPA. This is a snapshot calculator which means that output values for one section (or cell) are only for a moment in time, so, it is not possible to analyse the behaviour of the network in time. Another limitation is the fact that each user only supports one service in the same instant of time. Although it has the former limitations, it can be a very important and useful tool to study the behaviour and response of the network, while increasing the number of users.

The calculator has the following inputs:

- Total number of users;
- UL transmission power;
- UL frequency;
- UL inter to intra cell interference ratio;
- UL load factor
- Priority police algorithm
- Service profile
- Environment

Other parameters, such as the geometrical description of the landscape and the link budget parameters are default implemented.

After collecting the service profile and the total number of users, it works out the number of users for each service and the requested SF (global) based on Table 3.1. If one keeps in mind that each carrier supports 256 codes and that HSUPA signalling needs two codes, there are 254 codes available. However, the main restriction to serve more HSUPA clients is not the number of SFs but the UL load factor!

The UL SF allocation algorithm has the following features: the contribution of the HTTP users to the global SF is not considered, because they only use UL signalling channels (so their number is limited by DL capacity); the algorithm always allocates voice users first, then video-telephonies', and at last the high speed users; the lowest HSUPA SF is SF32, in order to have a data rate high enough to satisfy data clients; all HSUPA users are allocated with the same SF.

R99 PS users are not considered here. They are only allocated, when there is not enough UL load factor for HS users. Although they have the same SF as the lowest HSUPA user, they request a lower throughput and a different $E_b/N_0$. Only MMS clients can be considered as R99 PS clients, because they support a 64 kbps data rate.

Table 3.1 – Spreading factor allocation priority table.

| # priority | Service | Switching | Throughput [kbps] | SF | # equivalent SF256 codes |
|---|---|---|---|---|---|
| 1 | Voice (R99) | CS | 12.2 | 128 | 2 |
| 2 | VT (R99) | | 64 | 32 | 8 |
| 3 | HSUPA | PS | 1450 | 2 | 128 |
| | | | 725 | 4 | 64 |
| | | | 360 | 8 | 32 |
| | | | 180 | 16 | 16 |
| | | | 90 | 32 | 8 |
| 4 | MMS (R99) | | 64 | 32 | 8 |

One can see that 1450 kbps is correlated with SF2. In fact, this is not exactly true as 3GPP's category 3 points, not a SF2, but a double SF4. So what one really has is 1450 kbps correlated with a 2*SF4. Nevertheless the behaviour is the same.

The required load factor is allocated for all R99 CS clients. If it is higher than the system load factor, normally 0.9 for HSUPA, clients are rejected while the condition is not satisfied (first the video-telephony users, only next the voice customers). From what is left of the input load factor, HSUPA users are allocated and the former procedure goes again. They are all allocated with the same and maximum data rate as possible. If MMS clients are rejected, then they become R99 PS (SF32) users; once again, they can be allocated or not depending on the free load factor. This way, it is possible to achieve a dynamic or dedicated carrier, depending on the scenario requested and the total number of users. The load factor equation is given by (2.1) and the link budget can be seen at Annex A.

After allocating the clients, we shall study which applications are being served. We already now how many R99 and HSUPA users we have. What is missing, is which and how many users are being served by each application. The simulator supports two algorithms for client's selection. Both of them pretend to select clients, and exclude the others, when there is not enough load factor for everyone. They are:

- QoS Priority
- Proportional Reduction

The QoS algorithm respects the priority in Table 3.2. Streaming users will go first, followed by FTP, e-mail and at the end by MMS users. HTTPs' are not considered. The Proportional Reduction algorithm reduces, at every iteration, 10% of each application probability, until the left users respect the allowed number of HSUPA clients.

Table 3.2 – QoS priority table.

| **Priority** | **QoS class** | **Application** |
|---|---|---|
| 1 | QoS1 | Streaming |
| 2 | QoS2 | FTP |
| - | QoS2 | HTTP |
| 3 | QoS3 | E-mail |
| 4 | QoS3 | MMS |

The calculator has the following outputs:

- Load factor (R99, HSUPA and Global)
- Number of users (R99, HSUPA, PS, CS and applications)
- Radius (R99 and HSUPA)
- SF ratio
- Throughput (HSUPA)
- Throughput ratio

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

$$L_{0[\mathrm{dB}]} = 32.4 + 20\log\left(d_{[\mathrm{km}]}\right) + 20\log\left(f_{[\mathrm{MHz}]}\right), \tag{3.1}$$

being:

- $d$ : cell radius;
- $f$ : frequency.

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

$$L_{P[\mathrm{dB}]} = L_{0[\mathrm{dB}]} + L_{tt[\mathrm{dB}]} + L_{tm[\mathrm{dB}]} , \tag{3.2}$$

where:

- § $L_{tt}$ : losses due to propagation over roof tops;
- § $L_{tm}$ : losses due to diffraction from the roof to the MT.

Manipulating these two expressions, the cell radius can be determined by:

$$d_{[\mathrm{km}]} = 10^{\frac{L_P - 32.4 - \log\left(f_{[\mathrm{MHz}]}\right) - L_{tt} - L_{tm}}{20}} , \tag{3.3}$$

which can also be written as function of power:

$$d_{[\mathrm{km}]} = 10^{\frac{EIRP_{[\mathrm{dBm}]} - P_{Rx\min[\mathrm{dBm}]} - L_{u[\mathrm{dB}]} - 32.4 - \log\left(f_{[\mathrm{MHz}]}\right) - L_{tt[\mathrm{dB}]} - L_{tm[\mathrm{dB}]}}{20}} , \tag{3.4}$$

where:

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

Based on the work developed by [Lope07] and [Salv07], there are two rings of users, which means HSUPAs' and R99s' have different distances to the Node B. Of course, all HSUPA clients distance the same from the Node B ($Radius_{HSUPA}$). The same goes with the R99 ($Radius_{R99}$).

Number of served users is a parameter studied in this chapter, being given by:

$$N_{served} = \sum_{x=1}^{N_u} \sum_{y=1}^{N_s} x \cdot p_y , \tag{3.5}$$

being:

- § $N_{served}$ : number of served users;
- § $N_s$ : number of services;
- § $p_y$ : probability for each service.

It is known that different throughput values exist for the physical and application layers. Although the load factor is estimated based on the physical layer, the real application throughput has a lower value. Table 3.3 shows the differences. Although throughput values are not exact, they can be considered good approximations.

Table 3.3 - Physical and application layer throughput.

| Physical layer | Application layer |
|---|---|
| 1450 | 1250 |
| 725 | 625 |
| 360 | 310 |
| 180 | 155 |
| 90 | 75 |

The HSUPA throughput is based on equation (2.1). Knowing the available load factor in UL, then it is possible assessing the number of HSUPA users, and their throughput. The total UL throughput is given by:

$$r_{UL[\text{kbps}]} = N_{u_{R99CS}} \cdot 12.2 + N_{u_{R99PS}} \cdot 64 + N_{u_{HSUPA}} \cdot r_{a[\text{kbps}]} , \tag{3.6}$$

being:

- $r_{UL}$ : total UL throughput;
- $N_{u_{R99CS}}$ : number of R99 CS users;
- $N_{u_{R99PS}}$ : number of R99 PS users;
- $N_{u_{HSUPA}}$ : Number of HSUPA users;
- $r_a$ : application layer throughput.

The SF ratio it is a very important parameter. There are two main parameters that can restrict the number of HSUPA (and R99) users, i.e. the number of available codes (correlated with the SF) and the load factor. The SF ratio gives the relation between these two parameters, using the number of codes allocated to each of them. It shows how the load factor can be restrictive, being given by:

$$SF_{ratio} = \frac{SF_{effective}}{SF_{target}} \tag{3.7}$$

where:

- $SF_{effective}$ : global SF assigned to all allocated users according to the load factor
- $SF_{target}$ : global SF assigned to all allocated users according to the 254 available codes

Another important parameter assessed is the global traffic, being given by:

$$T_{[\text{MB/h}]} = \sum_{i=1}^{N_s} l_i \cdot N_i \cdot V_{i[\text{MB}]} , \tag{3.8}$$

where:

- $T$ : global traffic;
- $l_i$ : number of calls per hour for service $i$ ;
- $N_i$ : number of users per service $i$ ;
- $V_i$ : mean volume per service $i$ ;

## 3.2 Scenario

This section presents scenarios' configuration. Figure 3.1 shows the MTs location:

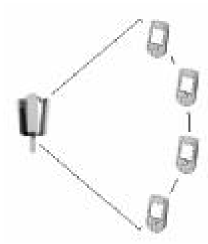


Figure 3.1 - All users at maximum distance [Antu07].

These MTs are requesting different applications. The probability of each application has different values, depending on the profile, Table 3.4.

Table 3.4 – Profiles.

| Applications | Voice Centric [%] | Data Centric [%] | High Speed Only [%] |
|---|---|---|---|
| Voice | 0.485 | 0.223 | 0 |
| VT | 0.002 | 0.003 | 0 |
| Streaming | 0.071 | 0.106 | 0.14 |
| FTP | 0.169 | 0.255 | 0.33 |
| HTTP | 0.118 | 0.177 | 0.23 |
| E-mail | 0.105 | 0.159 | 0.20 |
| MMS | 0.050 | 0.077 | 0.10 |

Table 3.5 presents the target throughput values for both R99 and HSUPA users. Table 3.6 presents the minimum required throughput considered in this analysis (based on [CoLa06]). Table 3.7 presents the fast and slow fading margins, and indoor penetration. These values are based on [EsPe06]. $E_b/N_0$ values used in R99 are based on [CoLa06], and summarised in Table 3.8.

Table 3.5 - Target throughput.

| Service | Target Throughput [kbps] |
|---|---|
| Voice | 12.2 |
| Video-Telephony | 64 |
| HSUPA | 1450 |
| R99 PS | 64 |

Table 3.6 - Minimum required throughput, [CoLa06].

| Service | Minimum Throughput [kbps] |
|---|---|
| Voice | 12.2 |
| Video-Telephony | 64 |
| Streaming | 75 |
| FTP | 75 |
| E-mail | 75 |
| MMS | 64 |

Table 3.7 - Fading margins and indoor penetration values, [EsPe06].

| Environment | Indoor | Pedestrian | Vehicular |
|---|---|---|---|
| $M_{SF\ [dB]}$ | 7.0 | 4.5 | 7.5 |
| $M_{FF\ [dB]}$ | 0.3 | 0.3 | 1.0 |
| $L_{int\ [dB]}$ | 21.0 | 0 | 11.0 |

Table 3.8 - $E_b/N_0$ values for the different services considered in Release 99, [CoLa06].

| Application Data Rate [kbps] | Indoor[dB] | Pedestrian[dB] | Vehicular[dB] |
|---|---|---|---|
| 12.2 (CS) | 5.8 | 5.8 | 6.9 |
| 64 (CS) | 4.1 | 4.2 | 5.8 |
| 64 (PS) | 2.5 | 2.5 | 2.5 |

Table 3.9 presents the values for the activity factor for the different services in Release 99 and HSUPA. Table 3.10 presents the default values used in the link budget. Table 3.11 summarises the frequencies of the different UL carriers, the first one being used as a shared carrier and the other three as dedicated ones, [EsPe06].

Table 3.9 - Activity factor values for the different services considered in Release 99 and HSUPA.

| | Applications | Activity Factor |
|---|---|---|
| CS | Voice | 0.65 |
| | Video-telephony | 1.0 |
| PS | HTTP | 0.5 |
| | Other applications | 1.0 |

Table 3.10 - Default values used in link budget.

| Parameters | | Value |
|---|---|---|
| UL transmission power [dBm] | | 24.0 (Power Class 3) |
| UL frequency [MHz] | | [1920; 1980] |
| MT antenna gain [dBi] | | 0 |
| Node B antenna gain [dBi] | | 17.0 |
| Soft handover gain [dB] | Release 99 | 1.5 |
| | HSUPA | Not considered |
| UL inter to intra cell interference ratio | | [0.391; 0.65] |
| Cable losses between emitter and antenna [dB] | | 3.0 |
| Losses due to user [dB] | Voice | 3.0 |
| | Data | 1.0 |
| UL Noise Factor [dB] | | 5.0 |
| Losses between transmitter and antenna | | 2.0 |
| Diversity gain | | 3.0 |

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

| Carrier | Frequency [MHz] |
|---|---|
| 1 | 1922.5 |
| 2 | 1927.5 |
| 3 | 1932.5 |
| 4 | 1937.5 |

## 3.3 Analysis of results

This section presents the analysis of results given by the HSUPA's theoretical model

### 3.3.1 Number of users

Considering a Pedestrian environment and a maximum UL load factor of 0.9, the analysis of user's distribution is presented. Figure 3.2 presents the number of served users for different scenarios.

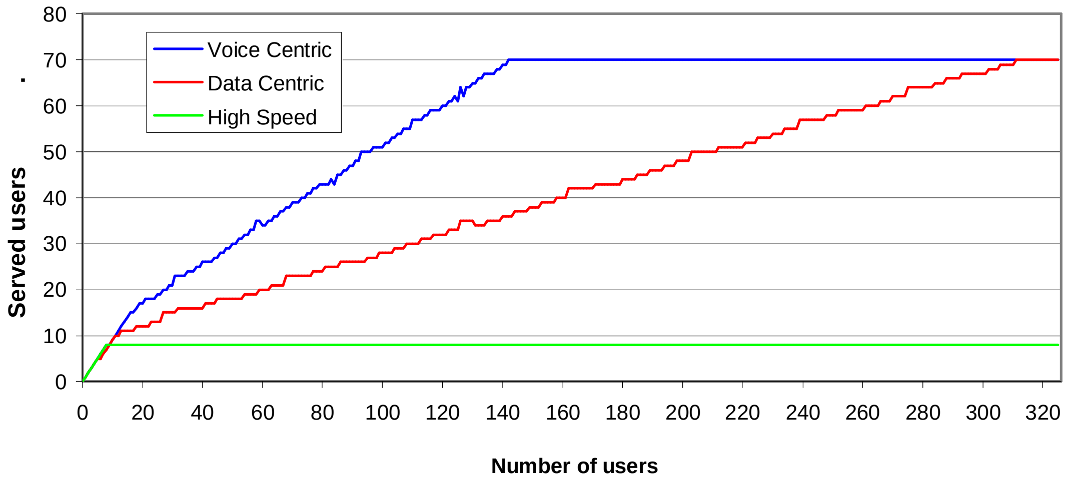



Figure 3.2 - Number of served users for different profiles.

There are two basic different kinds of profiles: with CS voice (Voice and Data Centric), and without (High Speed profile). The latter is very useful when simulating a dedicated high speed carrier. Analysing Figure 3.2 and taking into consideration all the assumptions described before, concerning load factor limitation, one observes that it is only possible to serve a maximum number of 8 HSUPA users per carrier and sector. Nevertheless, this it is not such a low number because tri-sectoriesed cells allow something around 24 HSUPA users. This number can go higher, as the network can support 4 carriers.

Although both Voice and Data Centric profiles have similar behaviours, Figure 3.2 shows they describe different routes. They have different slopes, between the point where they start rejecting clients – Voice Centric profile users are rejected when the total number of users is more than 19, while for Data Centric it is more than 13 - and the saturation point, due to the percentage of voice users. In fact, one R99 voice user requests such a low load factor, that it is possible to allocate up to 70 users, while it is only possible to serve 8 HSUPA users requesting only 90kbps (physical layer). Reminding that the snapshot model always satisfies voice users first, what happens in Figure 3.2 is that one has a lower percentage of voice users for DC profile. So, in that case, it takes longer to satisfy the maximum number of voice users (70). For Voice and Data Centric profiles, when the saturation point is achieved, there are only voice users being served. Data applications are gradually expelled by the admission control algorithm, as one can see in Figures 3.3, and 3.4.

It is important mentioning that, in a real network, no HSUPA users are rejected due to low throughput available (for obvious reason). While the network has available codes allocates them, even if it means data rates around 1 and 2 kbps per user. However, this thesis takes Table 3.6 into consideration in order to satisfy applications requirements.

Figures 3.3 and 3.4 show some differences between them. While for the Voice Centric profile there are HSUPA users served until the number of users goes up to 127, for the Data Centric one that number is much higher, around 274 users. Another difference is the fact that almost no MMS users are served in the Voice Centric Profile. It does not mean that there are no MMS requests, but rather that MMS percentage is very low for such a low number of HSUPA users supported.

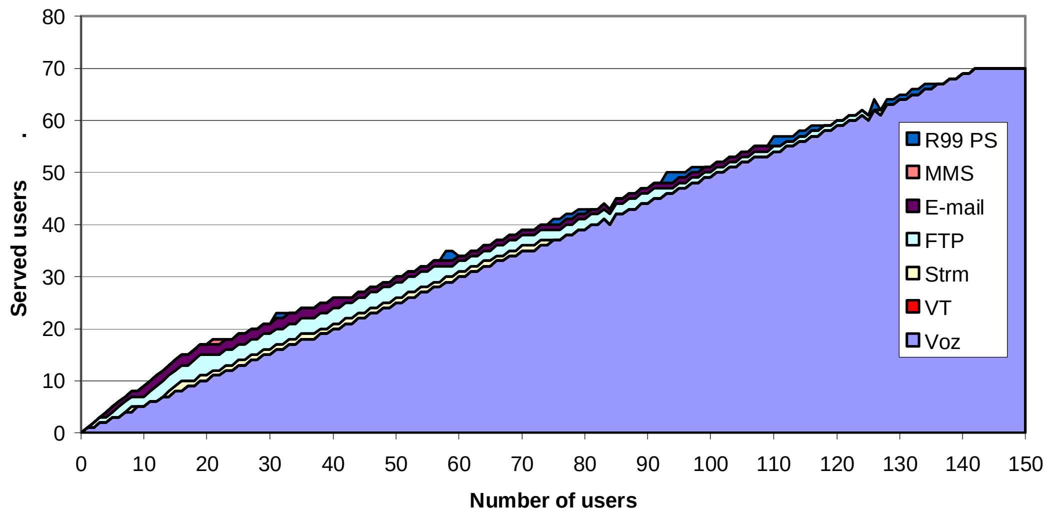



Figure 3.3 - Users distribution for the Voice Centric profile and proportional priority policy.

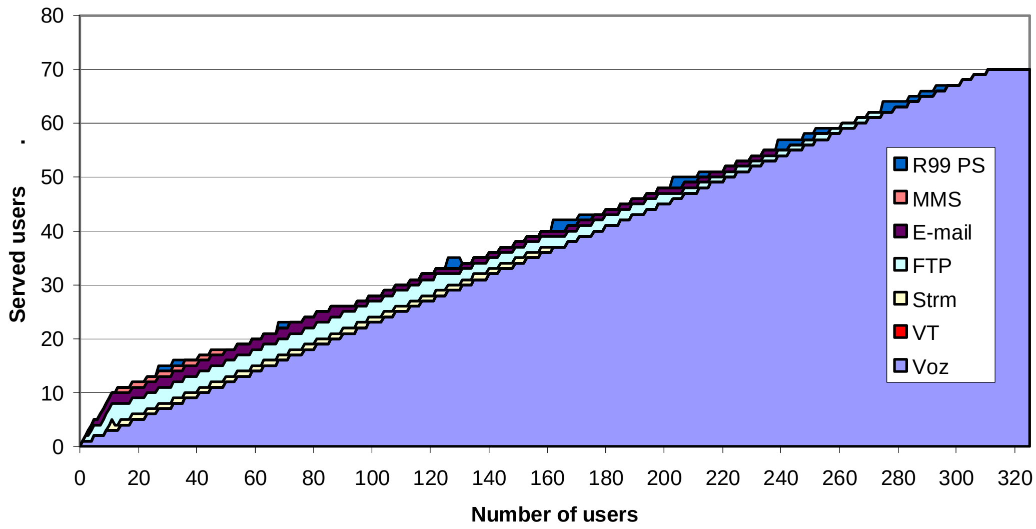



Figure 3.4 - Users distribution for the Data Centric profile and proportional priority policy.

Considering a low blocking probability, as practiced by operators, Figures 3.5 and 3.6 present the users' distribution for the previous scenarios. Although HTTP users are not considered in the UL, in order to analyse the number of rejected users, Figures 3.5 and 3.6 show the allocated DL HTTP users. Both Voice and Data Centric profiles support MMS clients as R99 PS, no matter the number of users requesting service. They do not need a high load factor, so they can be allocated whenever there is available space.

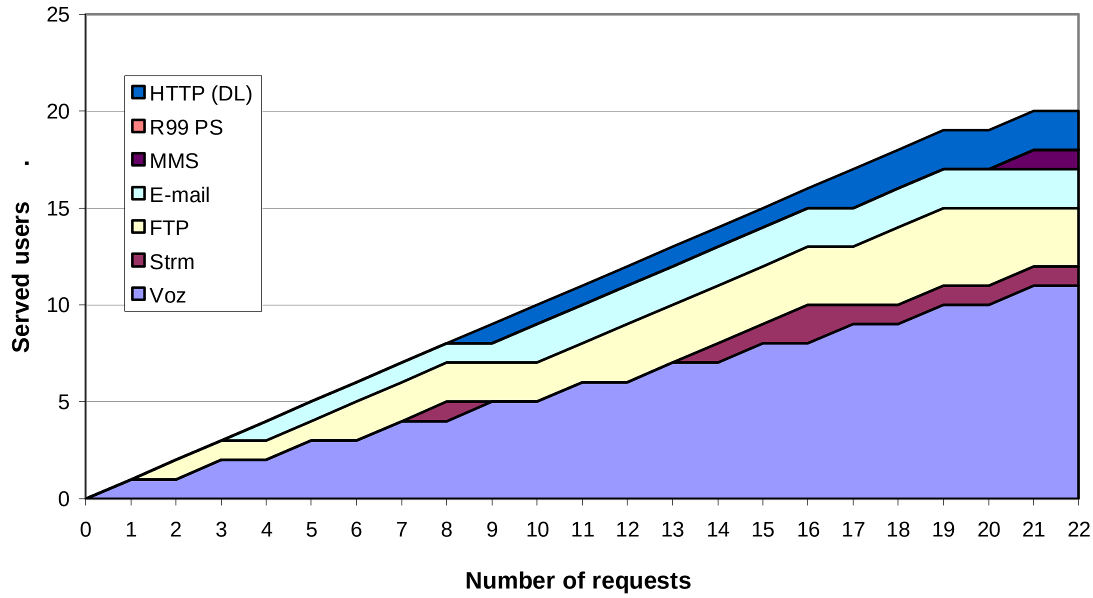



Figure 3.5 - Users distribution with low blocking probability for the Voice Centric profile and proportional priority policy.

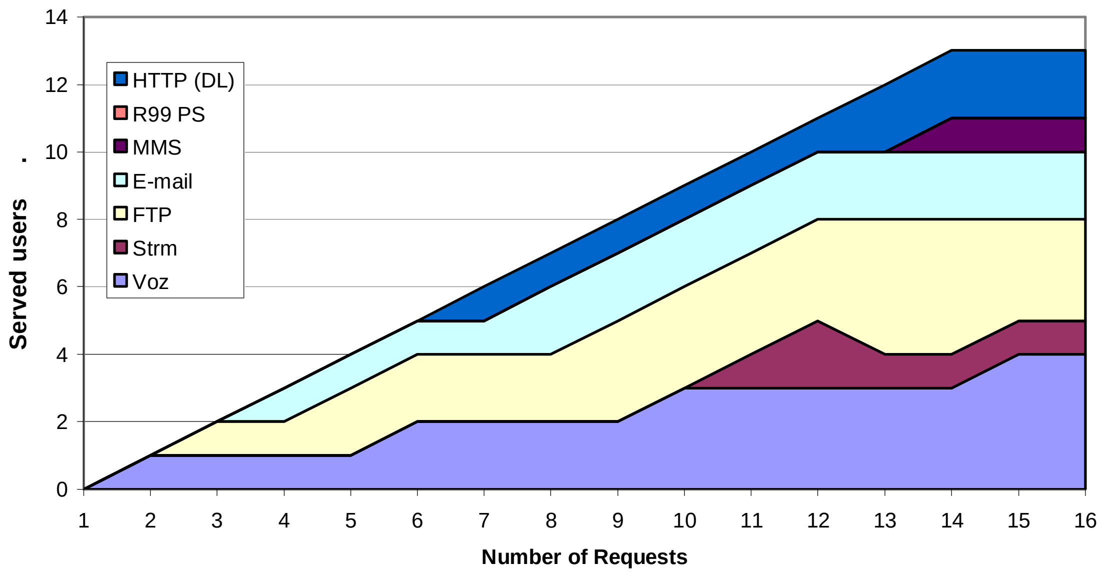



Figure 3.6 - Users distribution with low blocking probability for the Data Centric profile and proportional priority policy.

From Figure 3.7 one can see the following average user's distribution for Proportional Reduction algorithm and High Speed scenario: 1 MMS, 2 E-mail, 3 FTP and 1 Streaming. The UL snapshot model is not very sensitive to different priority policy algorithms, mainly due to the low achievable throughput (at least for Category 3). However, it may have same impact in user's distribution scenario. Choosing QoS priority algorithm and High Speed profile, Figure 3.8 plots the user's distribution scenario.

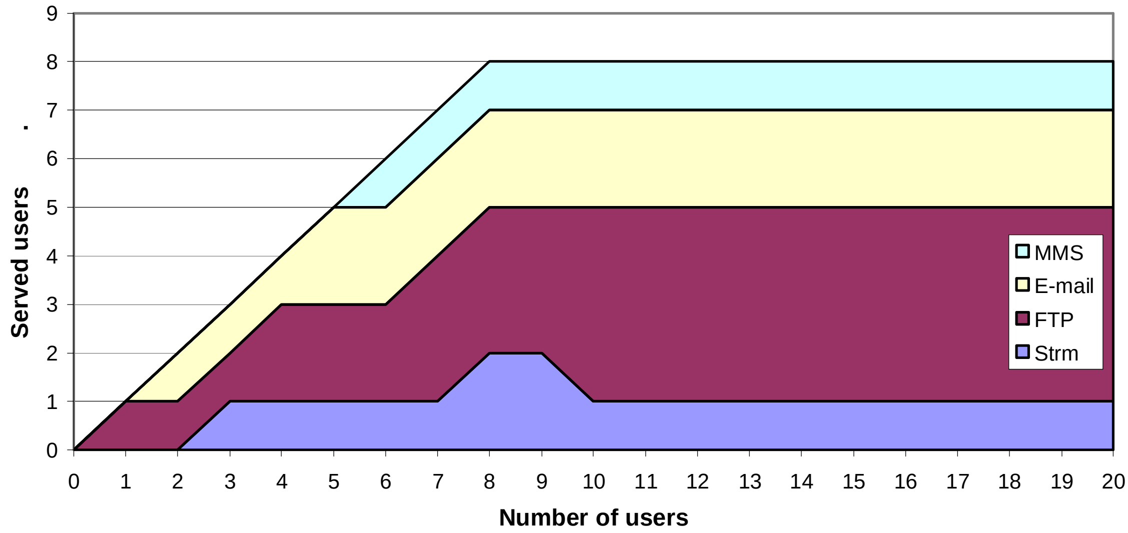



Figure 3.7 - Users distribution for the High Speed profile and Proportional Reduction policy.

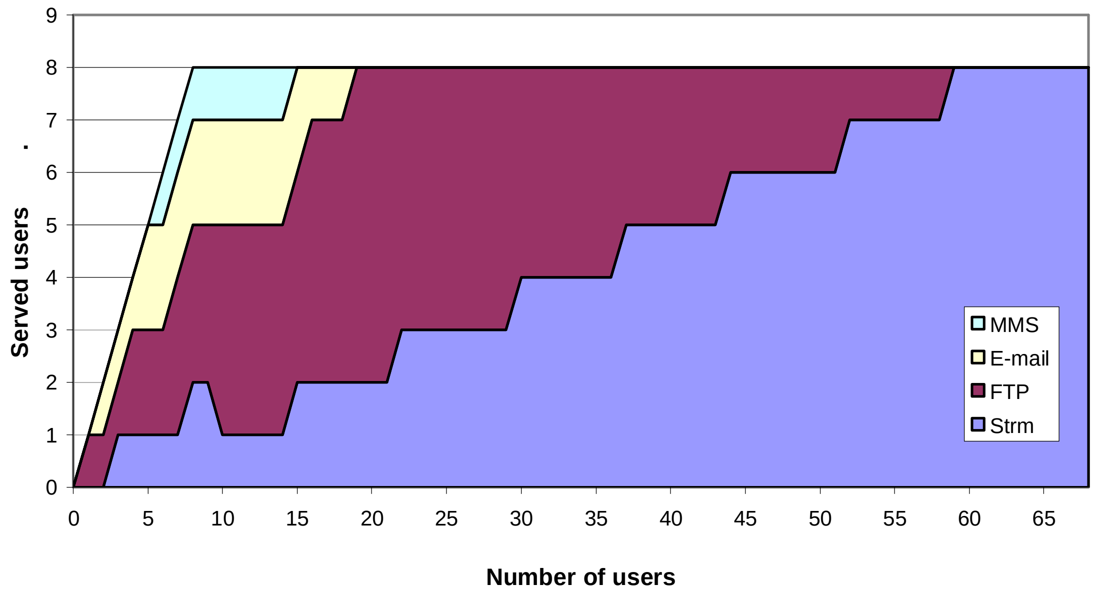



Figure 3.8 - Users distribution for the High Speed profile and QoS priority policy.

There are two main differences between Figures 3.7 and 3.8. QoS priority requires more users to achieve the final configuration, and in the end it only serves Streaming clients (while Proportional Reduction priority quickly achieves the final configuration and serves different applications). Comparing Figures 3.3, 3.4 and 3.7, one realises the xx axis has much shorter scope for the latter figure, because there are no voice users. However, one can see that for few users all profiles respect services' distribution.

### 3.3.2 Profile variation

Considering the Pedestrian environment, Proportional Reduction algorithm and different profiles, results are presented for the following parameters: load factor, radius, throughput, and $SF_{ratio}$ .

Observing Figure 3.9, it is easy to realise how fast the load factor achieves the maximum value of 0.9, no meter the profile. After achieving the top, the system enters in an oscillatory regime that will be explained later in this chapter. In the end, after reaching the final users configuration, the load factor is constant, and has a final value very near to 0.9. Looking at the High Speed profile, Figure 3.9, one sees that the saturation line could be nearer the 0.9 value. However, one HSUPA user at 75 kbps (application layer) requests a load factor around 0.107, while one voice user only requests 0.020. So Voice and Data Centric profiles have a higher efficiency concerning the load factor. When the system achieves the maximum allowed load factor, it starts rejecting users, preferably HSUPAs' and accepts voices'. Figure 3.10 confirms this.

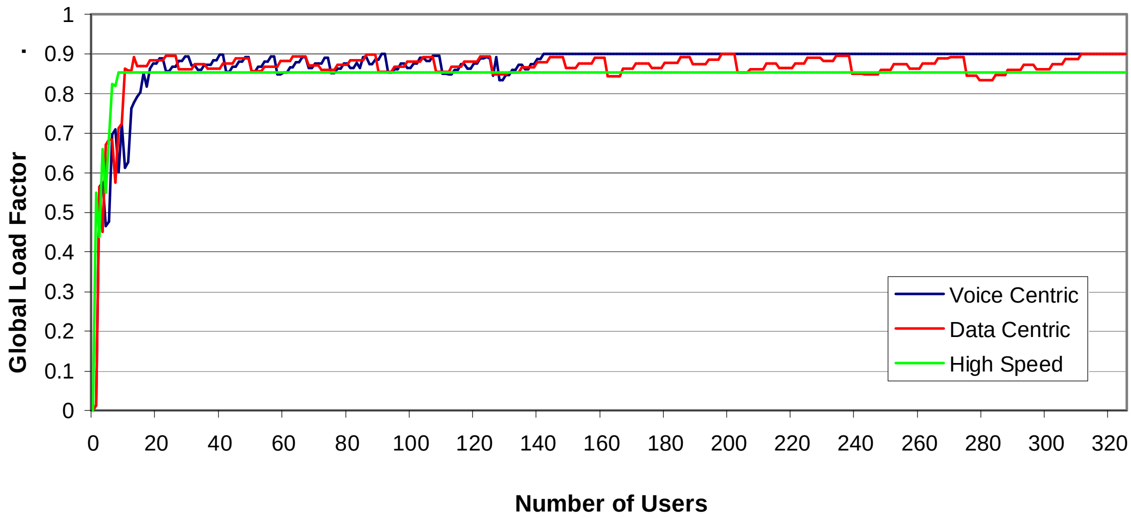



Figure 3.9 - Global Load Factor.

Figure 3.10 shows the load factor occupied by R99 and HSUPA users. HSUPA curves show a downstairs behaviour, where each floor depends on the number of HSUPA users served and throughput, Figure 3.11. Comparing big HSUPA steps and small R99 steps, one realises how restrictive the load factor can be for the former clients. The High Speed profile does not considerer any R99 users, so there are always 8 HSUPA users allocated.

Figure 3.11 shows HSUPA throughput. First of all, these values are for the application layer and not for the physical one, Table 3.3. The latter one is used for the load factor calculations, while the former one is what clients get. It has a maximum peak value around 1250 kbps and a minimum around 75 kbps. For Voice and Data Centric profiles, one can see that it reaches minimum throughput for less than 8 users, and, after that, it can goes up for 155kbps. Once again it just depends on the number of active voice and data users and their impact on the load factor. Although high UL data rates are only available for a few users, considering a tri-sectoriesed cell and 3 dedicated high speed carriers, then, it is possible to serve 9 users at 1250 kbps, or 27 users at 625 kbps, or even 45 users at 312 kbps, and at the same time have free load factor for lower HSUPA data rates.

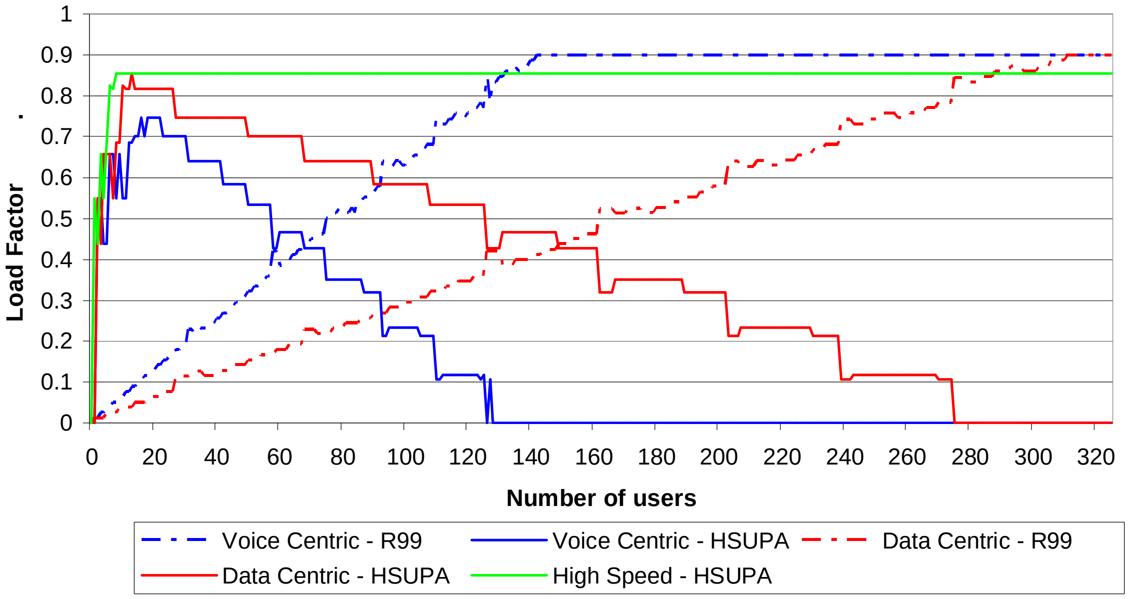



Figure 3.10 - HSUPA and R99 load factor.

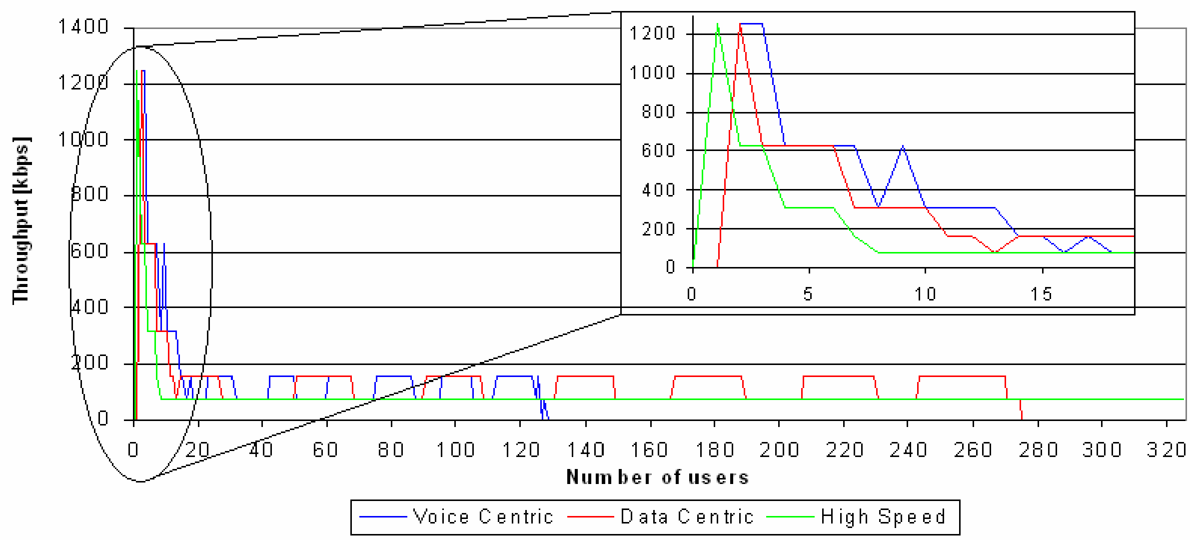



Figure 3.11 - HSUPA throughput.

Figure 3.12 presents the total throughput in a sector or cell. It almost reaches 2.0 Mbps in all profiles. Voice and Data profiles present saturation throughput values around 855 kbps, while High Speed 600 kbps.

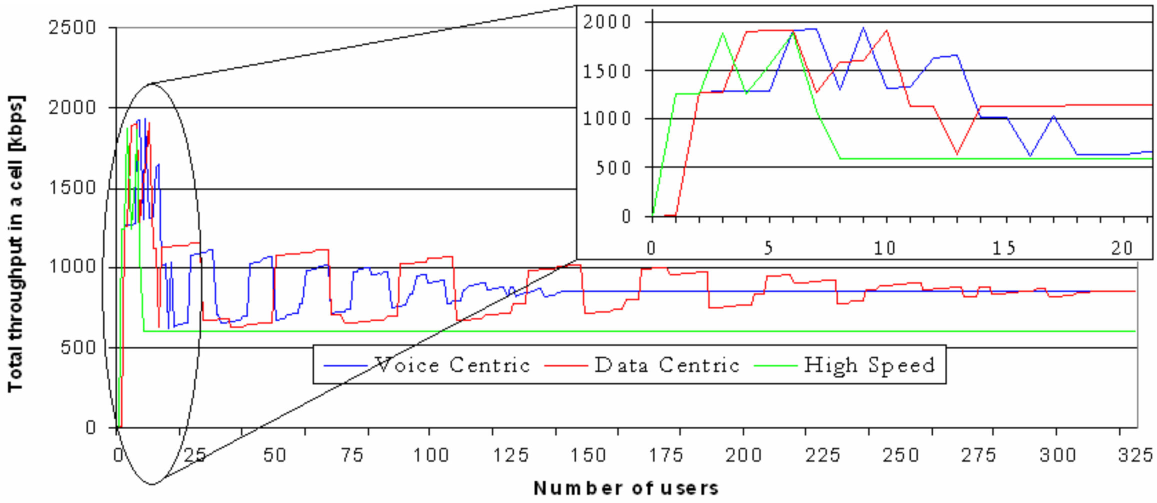



Figure 3.12 - Total UL throughput in a sector or cell.

The values of R99 and HSUPA radius for different profiles are presented in Figure 3.13. The R99 radius understandably decreases while the number of voice users is increasing. The HSUPA radius has the opposite behaviour. While the available load factor for HSUPA decreases, the number of users and their throughput also decreases. So, considering a HSUPA radius, it increases until there is no more HSUPA clients. In this scenario, the HSUPA radius goes from 0.58 km (1user at 1250 kbps) to 1.18 km (1 user at 75 kbps), and as an average 0.84 km radius for 8 users at 75 kbps. At the same time the R99 radius downgrades from 1.69 km in the beginning, to 0.93 km while serving 70 voice users.

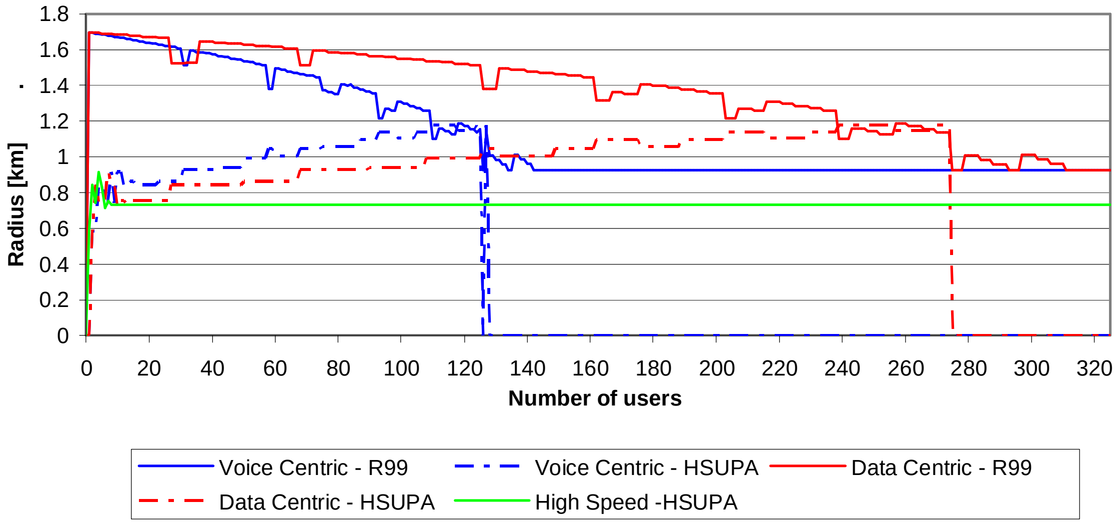



Figure 3.13 - R99 and HSUPA radius.

Figure 3.14 presents the relation between the allocated number of codes according to the load factor, and according to the number of available codes. Figure 3.14 shows that, the load factor is always the main restriction for assigning new users. Although in the beginning the $SF_{ratio}$ equals 1 and one could think that $SF_{target}$ is limiting the number of users, the load factor still is the main reason for rejecting or downgrading new clients. Table 3.12 presents a good example.

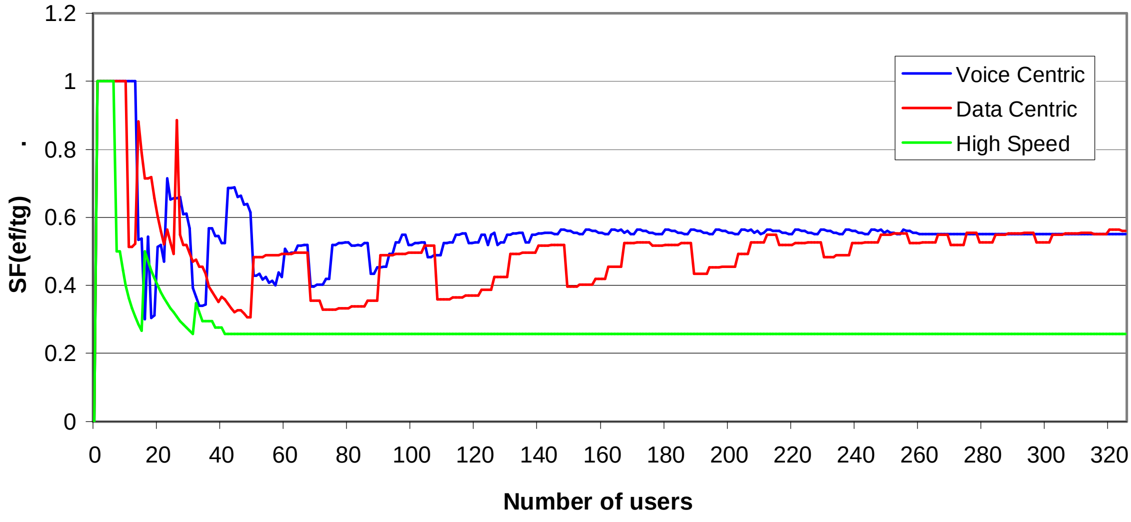



Figure 3.14 - $SF_{ratio}$ .

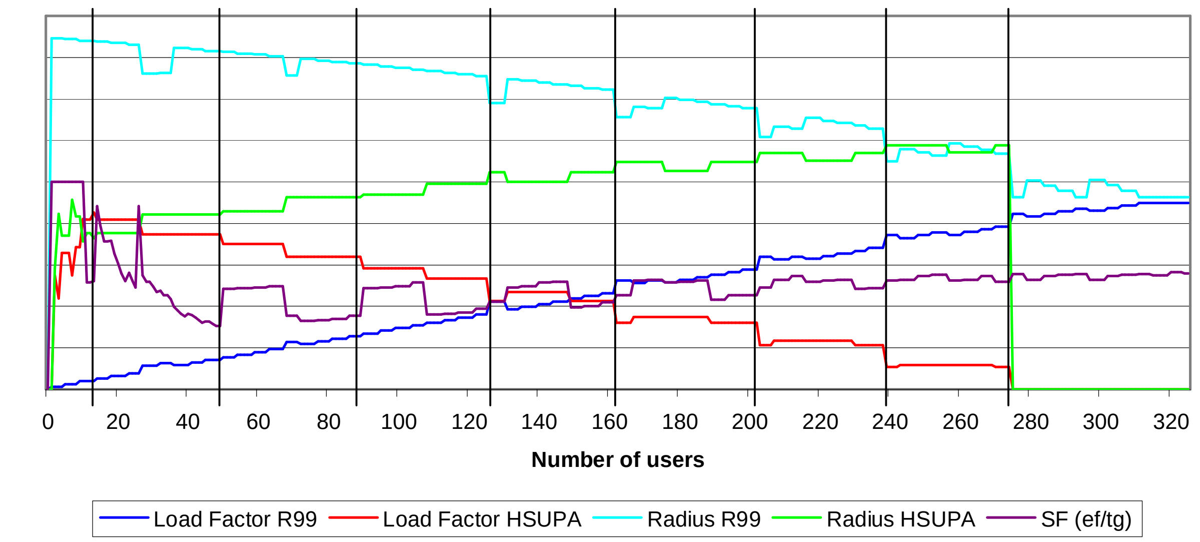
Table 3.12 - Load Factor vs Spreading Factor.

| Request | Spreading Factor | Load Factor |
|---|---|---|
| Request 1:<br>2 users @ 1450 kbps | 2×128=256 < 254 False | 2×0.55=1.1 < 0.90 False |
| Request 2:<br>2 users @ 625 kbps | 2×64=128 < 254 True | 2×0.22=0.44 < 0.90 True |

From Table 3.12, Request 1 ($SF_{target}$ =256) is not fulfilled because there are not enough codes and load factor. However, downgrading the requested throughput (Request 2), and refreshing the $SF_{target}$ value (128), the load factor is fulfilled. While $SF_{ratio}$ =1, the previous conditions always go on; when $SF_{ratio}$ <1 the only limitation is the load factor. So one could say the Spreading Factor is not restrictive.

Regarding figure 3.14, one realises that $SF_{target}$ gets stable when the network serves more voice users, due to the low number of codes and the low load factor required.

Previously one has seen a periodic behaviour, for several outputs, between the point where the system stars rejecting users and the saturation point. Figure 3.15 shows the correlation between those outputs and the network response, every time one HSUPA client is expelled. The following scenario was considered: Pedestrian environment, Data Centric profile, and Proportional Reduction algorithm.




.Figure 3.15 - Correlation between load factor, radius and $SF_{ratio}$ .

Looking at the $SF_{ratio}$ trace, every time it crosses one of the black vertical lines, it gives a jump up, because one HSUPA user has just been expelled in order to release load factor and serve new voice requests. Although the simulator always serves the biggest number of users as possible, if there is enough load factor it upgrades the others HSUPA users' throughput to the upper SF level. This is why the $SF_{ratio}$ has big jumps up. If the jump is shorter, then the throughput was not increased, in order to allocate MMS users as R99 PS, or because it just there was not enough available load factor. At the same time, the radius will be probably upgraded.

For the same scenario, Figure 3.16 describes the global load factor while UL inter-to-intra-cell interference ratio ranges from 0.391 to 0.65. Although there is no big difference among them, lower inter-to-intra-cell interferences ratio generates lower load factors. This means more users will be served.

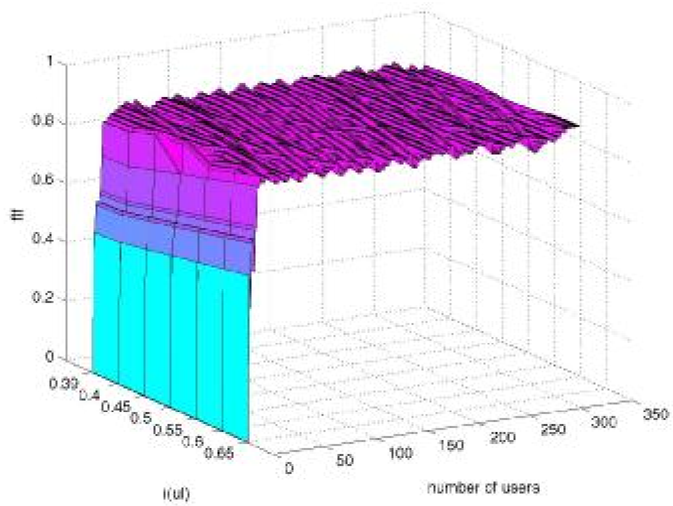



Figure 3.16 - Relation between global load factor and inter-to-intra-cell interference ratio.

### 3.3.3 Environment variation

Considering Proportional Reduction priority algorithm, Data Centric profiles, and different environments, results are presented for the following parameters: load factor, radius, and inter-to-intra-cell interferences ratio.

Figures 3.17 and 3.18 present HSUPA and R99 radius for different environments. These three environments produce several differences among them, the Pedestrian being the most favourable, and the Indoor the less one. Indoor and Vehicular environments present big HSUPA and R99 radius losses, compared to the Pedestrian one.

Traces Ped-Veh (R99), Ped-Ind (R99), Ped-Veh (HSUPA) and Ped-Ind (HSUPA) plot those radius differences:

- Ped-Veh (R99): R99 users in a Vehicular environment have less 400 m radius coverage, than in a Pedestrian one. Saturation point comes earlier, so around less 16 users are served.
- Ped-Ind (R99): R99 users in an Indoor environment have an average loss of 400m radius coverage, compared to a Pedestrian one. It goes from 1.2 km (1 user) to 0.62 km (70 users).
- Ped-Veh (HSUPA): HSUPA users in a Vehicular environment have an average loss of 150 m radius coverage, compared to a Pedestrian one. HSUPA users are expelled earlier from the network (while talking about number of requests, not time).
- Ped-Ind (HSUPA): HSUPA users in an Indoor environment have an average loss of 650 m radius coverage, compared to a Pedestrian one. It goes from 0.50 km to 0.80 km.

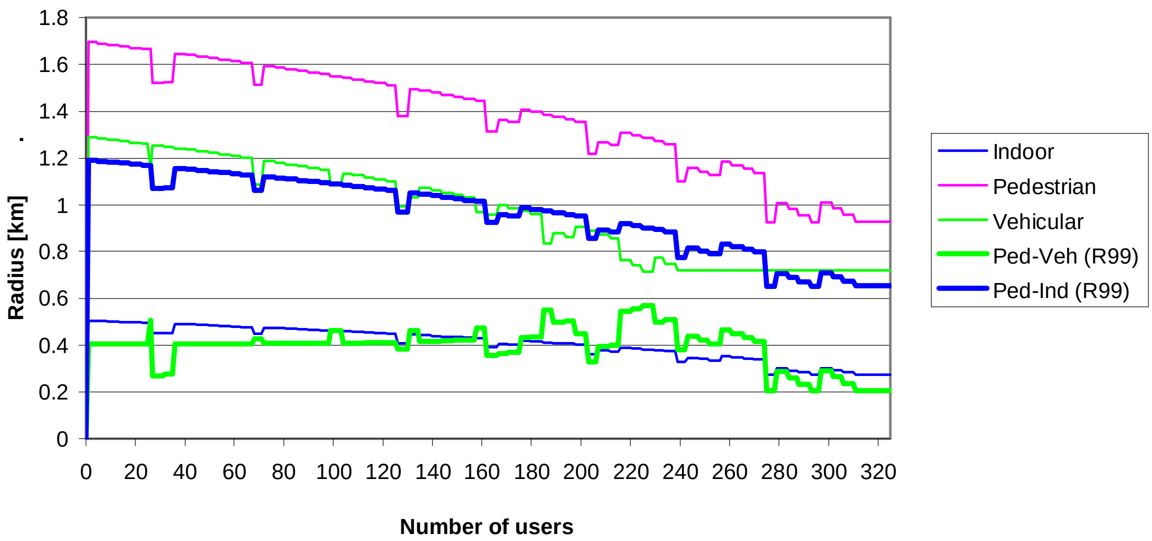



Figure 3.17 - R99 radius for different environments.

Figures 3.19 and 3.20 present HSUPA and R99 load factors for different environments. Indoor and Pedestrian environments (they have the same R99 $E_b/N_0$ values) describe the same route, while the Vehicular environment requires a higher R99 load factor (for the same number of requests), therefore, lower load factor available for HSUPA. As mentioned before, for the Vehicular environment, only 54 voice users can be served, and the HSUPA users will be excluded earlier from the system.

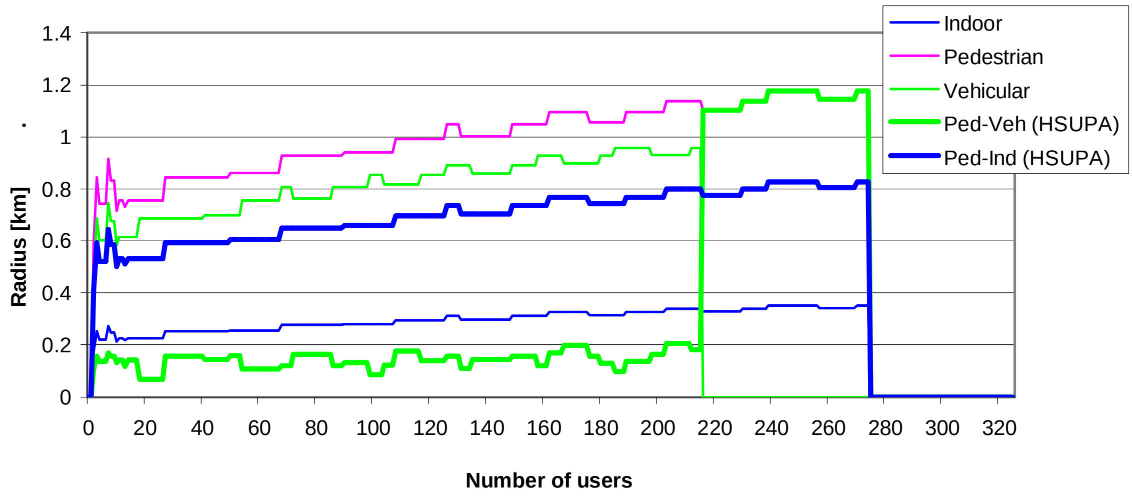



Figure 3.18 - HSUPA radius for different environments.

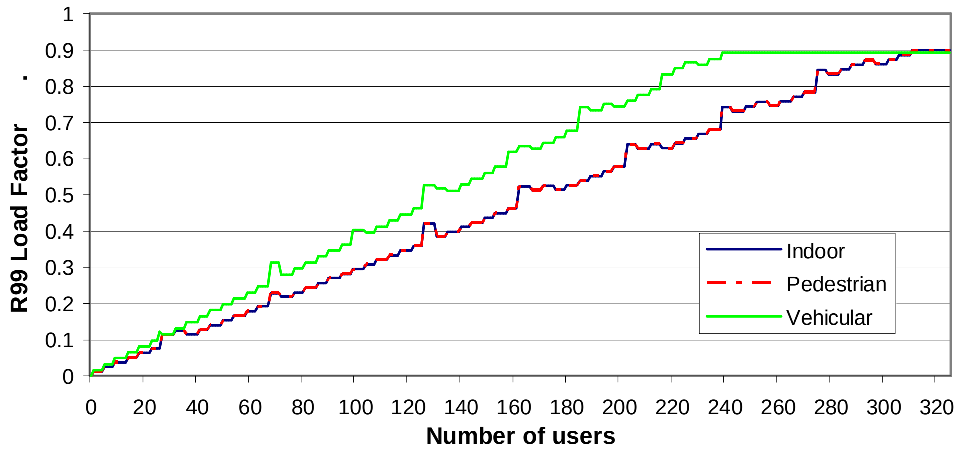



Figure 3.19 - R99 load factor for different environments.

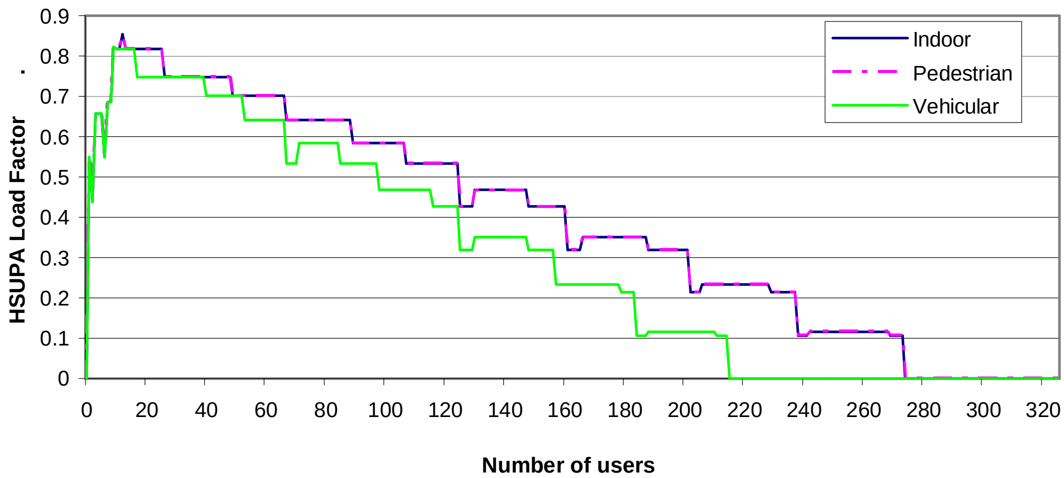



Figure 3.20 - HSUPA load factor for different environments.

## 3.4 Traffic processed by Node B

It is of interest to analyse the total traffic processed by the Node B during one hour (one sector). In order to do so, one must define the mean call duration for voice users and the file size for data users. Table 3.13 shows this information, based on [Seba07].

Table 3.13 – Mean call duration and mean file size.

| Service | Parameter | Value |
|---|---|---|
| Voice | Mean Call Duration [s] | 120 |
| Streaming | Mean File Size [kB] | 17500 |
| FTP | Mean File Size [kB] | 2550 |
| E-mail | Mean File Size [kB] | 100 |
| MMS | Mean File Size [kB] | 165 |

Two scenarios were defined, with the main difference emphasising the data usage, called Light and Heavy, defined by the calls per hour ( $l$ ) parameter, Table 3.14.

Table 3.14 – Light and Heavy scenarios.

| Service | Scenario | |
|---|---|---|
| | Light | heavy |
| | $l$ | $l$ |
| Voice | 6 | 6 |
| Streaming | 2 | 4 |
| FTP | 3 | 12 |
| E-mail | 4 | 12 |
| MMS | 2 | 12 |

One also considers the percentage of rejected users, in order to have a realistic network's behaviour. Considering 5% rejected users, one can determine the maximum scope for this analysis, Figure 3.21.

Figures 3.22 and 3.23 show the UL total traffic processed by Node B for Voice and Data Centric respectively. Tendency lines are observed. This way it is possible to obtain the UL total traffic equations.

For Light and Voice Centric scenario:

$$T_{[\mathrm{MB/h}]} = 4.23N_u - 6.0708 \tag{3.5}$$

For Heavy and Voice Centric scenario:

$$T_{[\text{MB/h}]} = 10.485 N_u - 5.1109 \tag{3.6}$$

For Light and Data Centric scenario:

$$T_{[\text{MB/h}]} = 6.1113 N_u - 10.598 \tag{3.7}$$

For Heavy and Data Centric scenario:

$$T_{[\text{MB/h}]} = 15.891 N_u - 15.121 \tag{3.8}$$

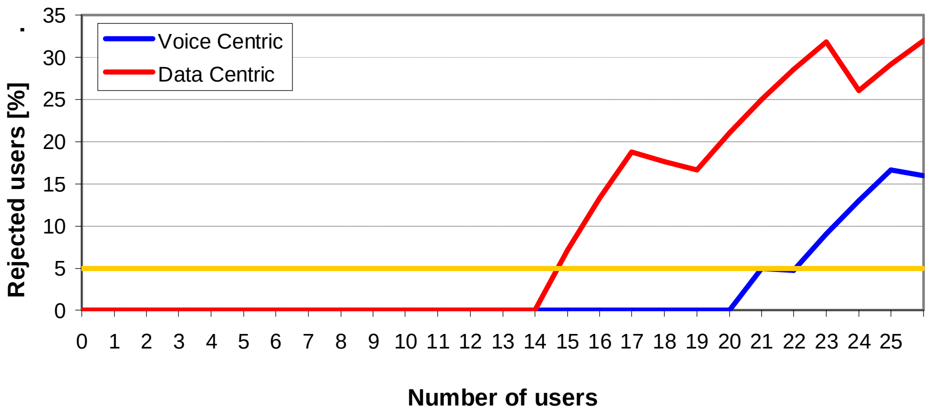



Figure 3.21 – Rejected users.

Regarding Figures 3.22 and 3.23, one can observe that global traffic is three times higher for the Heavy scenario. As expected, for a 5% rejecting probability, the global traffic processed by the Node B is almost the same for Voice and Data Centric, being: 75/80 MB/h for Light scenario, and 205/215 MB/h for the Heavy one. The only difference between VC and DC is the number of served users. If one looks carefully to Figures 3.22 and 3.23, some peaks are easily noticeable because of streaming users (being served or, being rejected) as this application has a huge size (17500 KB). It means the number of users is not proportional to the global traffic processed. This is why the trend lines make so much sense! Finally, one realise that QoS Priority and the Proportional Reduction algorithms are not significantly differences in the network's behaviour.

It is obvious that the theoretic approach is always very optimistic, enhancing the outputs. This study has limitations, due to the scope of this thesis, because it does not consider many network features and the transmission channel quality. This thesis is more focused on admission control. Concerning the UL total traffic processed by the Node B estimation, it is important mentioning these values do not include the average throughput per application, but just the number of users, the mean call duration, and the mean file size.

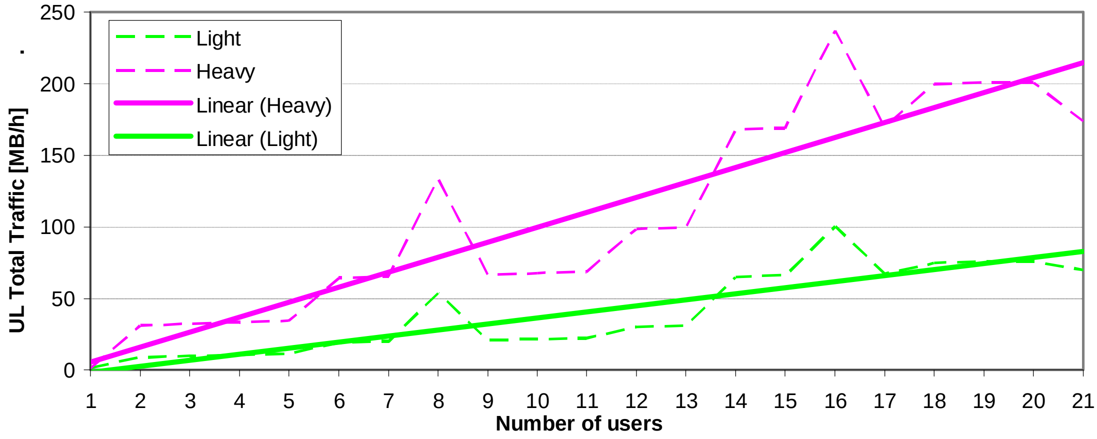



Figure 3.22 – UL total traffic processed by Node B for Voice Centric.

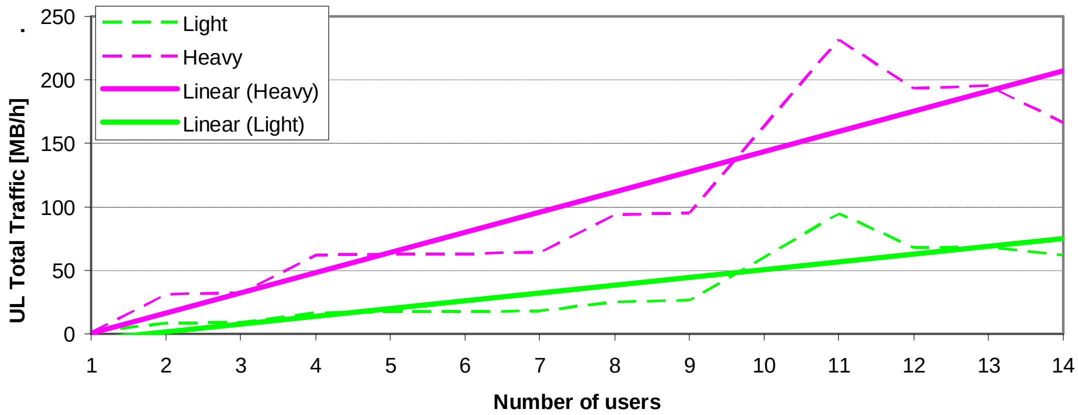



Figure 3.23 – UL total traffic processed by Node B for Data Centric.

# Chapter 4

# Simulator

This chapter gives an overview of the OPNET simulator that was 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 Modeler

OPNET is a powerful network simulation tool [OPN06a]]. It has several solutions for Application Performance Management, Network Operations, Capacity Planning and Design, and Network Research and Development (R&D). These solutions fit into different market segments: Enterprise IT, Defence, Service Providers, Network Equipment Manufacturers, and of course for Universities. OPNET Technologies, Inc. has free research software licences for the former one, and that is why OPNET Modeler, an industry's environment for network modelling and simulation, has been used for this thesis. It allows to design and study communication networks, equipment, protocols, and applications, such as UMTS, WiMAX, IPv6, MPLS, and many more. Network topology is a combination of nodes and links, and both of them can be user defined in order to answer to researchers' needs. Simulations can then be executed and the results analysed for any network element in the simulation network. The OPNET Modeler 12.0 has also other key features, like: a very scalable and efficient simulation engine, open model source code, different simulation technologies, object-oriented modelling, comprehensive graphical user interface, integrated debugging and analysis, 64-bit fully parallel simulation kernel, etc..

Modeler is based on a series of hierarchical editors that can be very efficient simulating real networks, equipment and protocols [OPN06b]. The Project Editor is the main one, being where one can create a network model, choose statistics (from a simple node or from the whole network), execute a simulation, and view results. To create node and process models, build packet formats, create filters and parameters, one needs to use additional editors. The most important ones are described in the following paragraphs.

The Node Editor captures the architecture of a network device. Its behaviour is defined by the flow of data between different functional elements called modules. These modules represent applications, protocol layers, algorithms and physical resources. They are connected through packet streams or statistic wires in order to generate, send and receive packets to perform their function within the node.

The Process Editor is used to describe the processes (protocols, resources, applications, algorithms and queuing policies) that run inside the modules. These process models are represented by a powerful Finite State Machine (FSM), and are created with icons, that represent states, and lines, that represent transitions between states. Each state contains open source C/C++ code, supported by a library of functions designed for protocol programming (Proto-C: an OPNET variant on the C language). So, with the Process Editor it is possible to develop new process models.

Figure 4.1 maps the hierarchical structure among the main models.

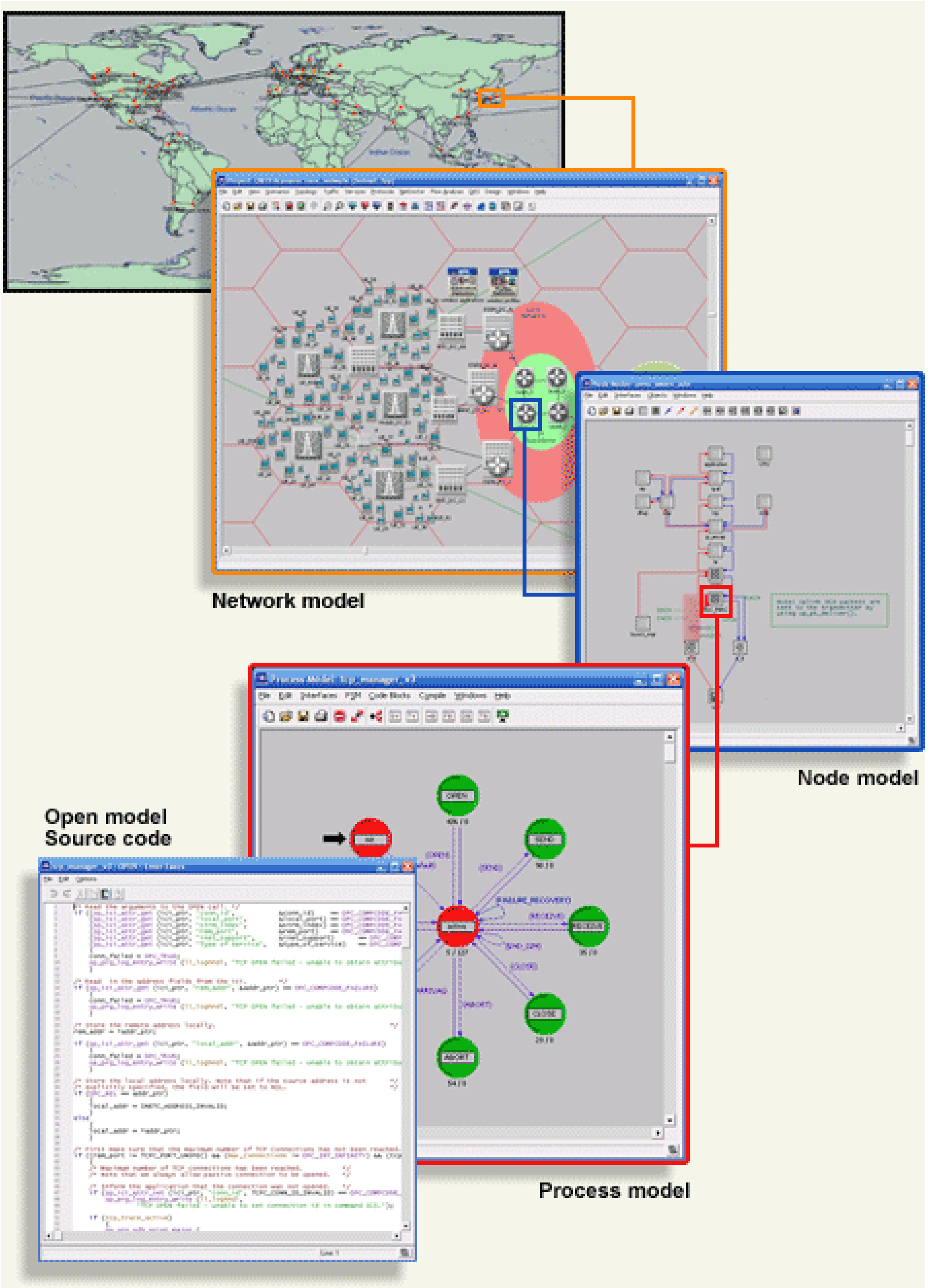



Figure 4.1 - Hierarchical structure among the main editors (extracted from [OPN06b])

There are many more editors in OPNET Modeler [OPN06c], e.g.: the Path Editor creates new path objects in order to define a traffic route; the Packet Format Editor defines the internal structure of a packet as a set of fields; the Link Model Editor creates new types of link objects, etc., and of course there are important editors for simulating and analysing the results.

While creating a new network model [OPN06d], one must first create a new project and scenario. The project is a group of related scenarios that explore a different aspect of the network. The scenario is defined by the topology, the scale and size, and the background scenario for the network. There one can use nodes (representation of real-world network objects) and links (communication medium that connects nodes to one another). These take one to the next level: creating network topologies. It is possible to import the network or manually construct it. There are three different ways of doing it:

- Import the topology: although the model library that comes with OPNET provides models of various devices used in today's networks, it is possible to import topologies directly from a number of vendor products.
- Rapid Configuration: allows to select a network configuration, the types of nodes and the types of links;
- Place each individual node from the Object Palette into the workspace.

Once the general network topology has been built, one may need to add a server. One can find it in the object palette; if not, one can always drag a universal server and configure it towards the needs. In the end one needs to connect the server to the network.

Finally, one needs to add configuration objects to specify the application traffic that will exist on the network. In order to do so, the Application Config and the Profile Config objects should be dragged into the workspace. The question is how can one configure them?

Communication networks enable applications to exchange data, and each of these applications generates its own sort of traffic (it is possible to import traffic or model application traffic), [OPN05a]. This means that, different types of traffic cause and experience a different set of problems, so one may want to accurately model the traffic patterns generated by a variety of applications. Each application can be enabled or disabled on the client nodes through the use of one or more profiles, and each can be specified as a supported application service type on the server nodes. FTP, E-mail, Remote Login, Video Conferencing, Database, HTTP, Print, Voice and Custom (user-definable multi-tier application) are the components of the Application Model. Each of these applications has configurable attributes, such as start time, duration and repeatability.

In order to configure a workstation to model the behaviour of a user or group of users, one needs to describe their profile. The profile is a set of applications used by that group, on how long and how often applications are used through the day. These profiles can represent different user groups. One can execute profiles at the same time and repeatedly and also configure applications within a profile. Applications can be executed at the same time, one after the other, etc., in fact one have the freedom to simulate what really happens in a communication network.

One can choose which statistics [OPN06d] one wishes to analyse. Statistics can be collected from individual nodes in the network (object statistics) or from the entire network (global statistics). One can "Choose Individual DES Statistics" to collect single node statistics or one can go to the Probe Editor. The Probe Editor lets one specify the statistics to be collected during simulation, as they can be global statistics, node statistics, link statistics, attribute statistics, etc…

OPNET Modeler supports three kinds of simulation technologies [OPN05b]:

- Discrete Event Simulation (DES) provides highly detailed models that explicitly simulate packets and protocol messages. In fact, DES executes the protocol almost as a production environment, providing accurate results. Though, simulation runtimes are longer than other techniques.

- Flow Analysis uses analytical techniques to model steady state network behaviour. This simulation technique can be very useful to study routing and reach ability across the network in steady scenarios. Execution runtimes can be much faster.
- Hybrid Simulation combines the former simulation techniques to provide accurate and detailed results for targeted flows. In order to achieve reasonable times, it is possible to “fine-tune” the balance between discrete and analytical models.

Although there are several simulation techniques, DES has been continuously enhanced to deliver faster and more efficient simulations that scale significantly with the amount of traffic in the model. DES is the recommended tool to study application performance, network capacity planning and resource utilisation analysis. To run a simulation select “Run Discrete Event Simulation”, and then one can configure the Duration, the Update Interval (how often the simulation calculates events/second data) and the Simulation Kernel. Run it and a simulation progress dialog box will appear. Check Simulation Log for errors.

There are several ways to see the information collected for each statistic [OPN06d]. Although one can view simulation results in the Project Editor, the best way is to use the Analysis Tool. This feature can, for example, create scalar graphics and parametric studies, define template to which you apply statistical data, and create analysis configurations that can be saved and viewed late. The Filter Editor is used to define filters to mathematically process, reduce, or combine statistical data.

OPNET Modeler has more important features, while analysing the simulation results, e.g.:

- The Animation Tool can be very interesting. One can “Record Packet Flow Animation for Subnet” or “Record Node Movement Animation for Subnet” before executing the simulation.
- It is possible to generate and launch the HTML web report of your simulation, which shows all graphs and other statistics.

The process model describes the module’s behaviour [OPN06e]. There are features to keep in mind:

- A process model is a finite state machine (FSM);
- A FSM defines the states of the module and why it changes the states;
- FSM use states (the condition of a module) and transitions (a change of state due to an event);
- OPNET allows attaching fragments of C/C++ code and OPNET-specific functions to each part of an FSM. It is called Proto-C;
- The three primary places to use Proto-C are: “Enter Executive” when the module moves into a state, “Exit Executive” when the module leaves a state, and “Transition Executive” in response to an event;

- Process models respond to events and can schedule new ones;
- There are forced (green state does not return control, but executes the exit executives and transitions to another state) and unforced (red state returns control of the simulation to the Simulation Kernel after executing it's enter executives) states.

The first thing to do while implementing the process model is to open the Process Editor and place the model's states ("File->New… Process Model" and click the "Create State" tool button). Give each state a unique name. Every new state is, by default, unforced. Edit state attributes to "Make State Forced". The second step is to "Create Transition(s)" between the states. There are unconditional and conditional (the transition must evaluate to true before control passes from the source state to the destination state) transitions. The next step is to create the state executives needed in the FSM. Although there are so many functions, OPNET provides a summary of the most frequently used Kernel Procedures (choose "Help->Essential Kernel Procedures"). At the end one can control attributes visible at the node level editing the "Process Interfaces".

The first step in implementing a node model is to create one (choose "File->New… Node Model"). Use the "Create Processor" tool button to create processor modules, the "Create Packet Stream" and the "Create Statistical Wire" to connect the modules. The second step is changing the attributes of each module. At this point one can set the module with your own process model creation or use the module's library ("process model" attribute). Choose "Interfaces->Node Statistics" to make a statistic available at the node level.

At last, create the network model using your own node model and other nodes that might be required. Add your own node model to the Object Palette, place the node in the workspace, edit node attributes, select the statistics for collection, configure the simulations, run them, and analyse the results.

## 4.2 UMTS Model

OPNET's UMTS model suite is based on 3GPP Release 99 standards supporting only PS traffic [CIS05]. It allows modelling UMTS networks to evaluate end-to-end QoS, throughput, drop rate, end-to-end delay, and delay jitter through the radio access network and core packet network. It can also evaluate the feasibility of offering a mix of service classes given QoS requirements. It is possible to configure a UMTS network model in two different ways:

- UMTS network using application traffic,Figure 4.2: UMTS workstation nodes routing application traffic (voice, ftp,…) to other UMTS workstation nodes or server nodes.
- UMTS network using raw packet generator, Figure 4.3: UMTS station nodes send generic data traffic (conversational, streaming, interactive, and background classes) to other UMTS node stations through a single SGSN node. One can not have application

traffic sent to a UMTS station node, nor can you send traffic from an UMTS station node to an UMTS workstation node. This network topology should be used when one wants to model raw traffic data within the UMTS network, is not interested in the external IP network, and does not wish modelling CN to CN data transfer.

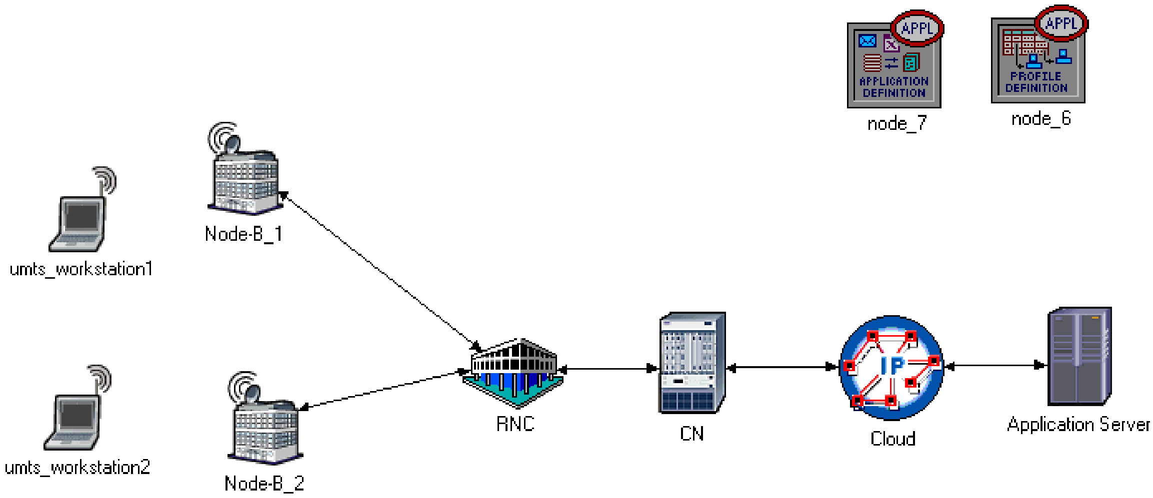



Figure 4.2 - UMTS network using application traffic (extracted from [CIS05]).

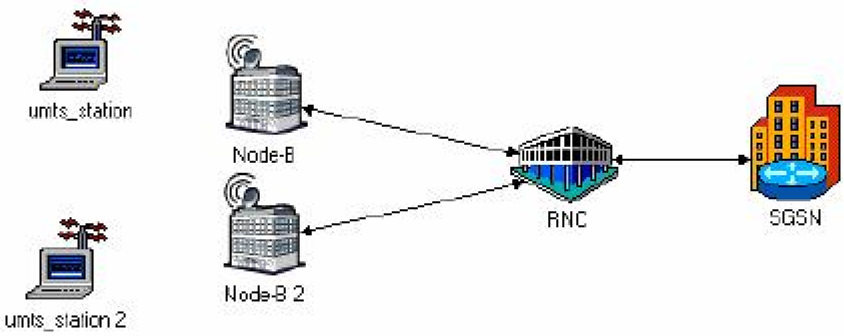



Figure 4.3 - UMTS network using raw packet generator (extracted from [CIS05]).

When the UE has traffic ready to go and queued on one of four QoS queues, it sends an Activate Packet Data Protocol (PDP) Context Request to the SGSN, which includes the QoS request. The SGSN can grant or reject access. So, the SGSN sends a Radio Access Bearer (RAB) and the QoS request to the RNC. If there is sufficient UL and DL capacity, the RNC sends a Radio Bearer Setup request to the UE. The former one set up the channel as specified and sends a Radio Bearer Complete back to the RNC. At the end, the RNC sends a RAB assignment, which includes the granted QoS to the SGSN. The SGSN sends an Activate Packet (PDP) Context Accept message to the UE, so it can send packets to the destination.

There are three different node models for the User Equipment: simple mobile stations (umts_station), advanced workstations (umts_wkstn), and advanced servers (umts_server). Due to the scope of this thesis only the second node station is considered. Advanced workstations stations can be modelled as pedestrian outdoor, vehicular outdoor and indoor office. Looking at the UMTS station model, Figure 4.4, it includes the full TCP/IP protocol stack between the application layer and the GMM one. There are also a RLC/MAC layer, a radio receiver, a radio transmitter, and an antenna. Transport channels link the RLC/MAC layer, the transmitter and the receiver.

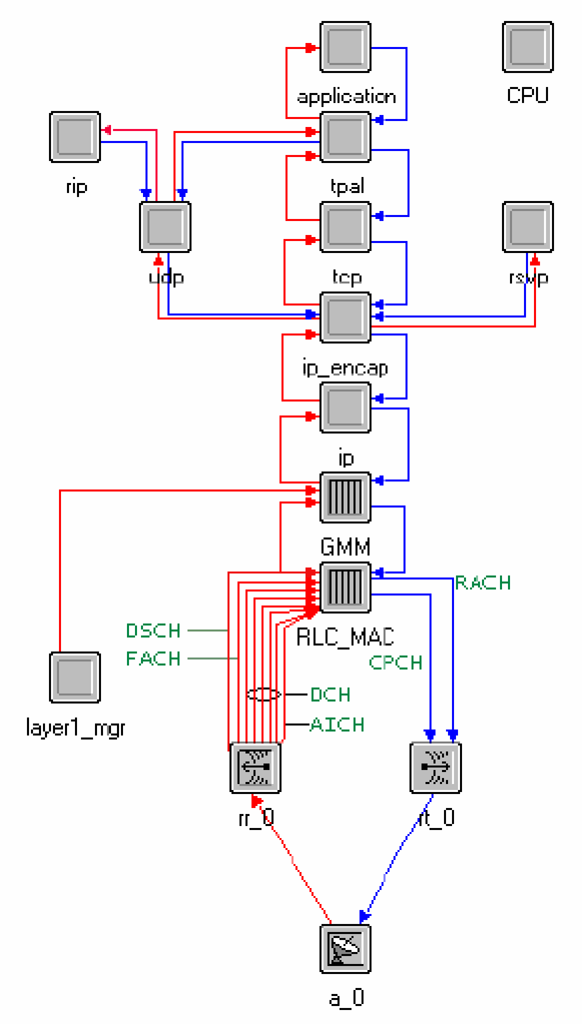



Figure 4.4 - UMTS station model (extracted from [CIS05]).

The GPRS Mobility Management (GMM) layer contains mobility management functions, session management functions, and radio resource control functions. The RLC/MAC layer contains the RLC and MAC layers, including priority handling of data flows, the three types of RLC modes, and segmentation and reassembly of higher-layer packets.

In order to manage the network's air interface, the UMTS model suite includes two Node-B models, the single-sector Node-B and the three-sector Node-B. Figure 4.5 represents the single-sector Node-B model. There is one Node-B processor module, connected to an ATM stack, a transmitter module (pr_0) and a receiver module (pt_0). Through the UL and DL, all packets travel over the transport channels.

The RNC is connected to one or more Node-Bs, and manages the resources of the air interface of all the UEs. Particularly, it coordinates the admission control process, manages handovers between its Node-Bs, buffer packets per QoS class, communicates with the SGSN, and manages the radio bearers. The RNC node model has nine ATM node stacks, each one of them is connected to the SGSN. More Node-Bs can be supported by adding more ATM stacks to the node structure.

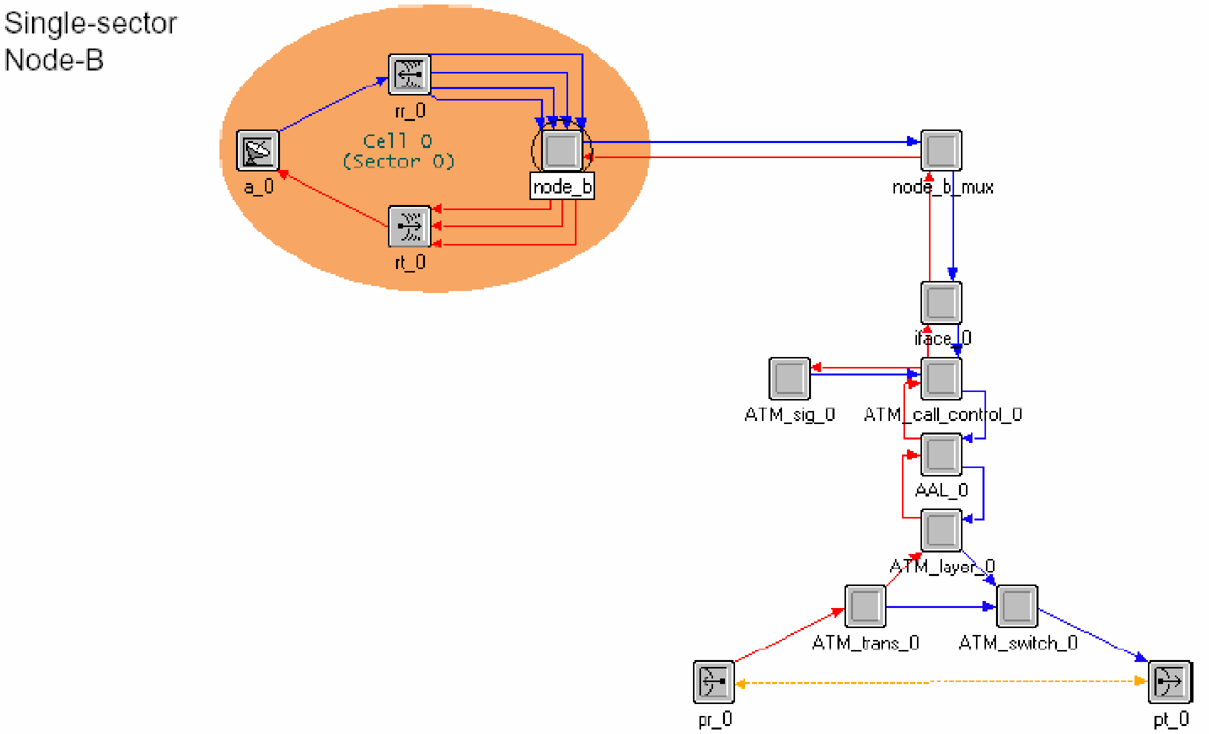



Figure 4.5 – Single-sector Node-B node model (extracted from [CIS05]).

It is possible to model the air interface between the UE and the UTRAN by modifying OPNET's 13 pipeline stages. Some of the most important pipelines are: the received power pipeline stage includes a path loss model and shadow fading model that depends on the environment; the propagation path loss models are based on Recommendation ITU-R M.1225 Guidelines for Evaluation of Radio Transmission Technologies for IMT-2000, 1997 (the Hata model and the free space are also supported); thermal noise and noise figure of the mobile and base station receiver can be modified due to the background noise pipeline stage; the interference noise pipeline stage will include same-cell and other-cell interference calculation; the bit-error rate pipeline stage includes the SNR versus BLER.

## 4.3 Scenarios

In this section the network architecture and the applications and profile configuration will be described.

### 4.3.1 Network architecture

Figure 4.5 presents the network architecture used in the simulations.

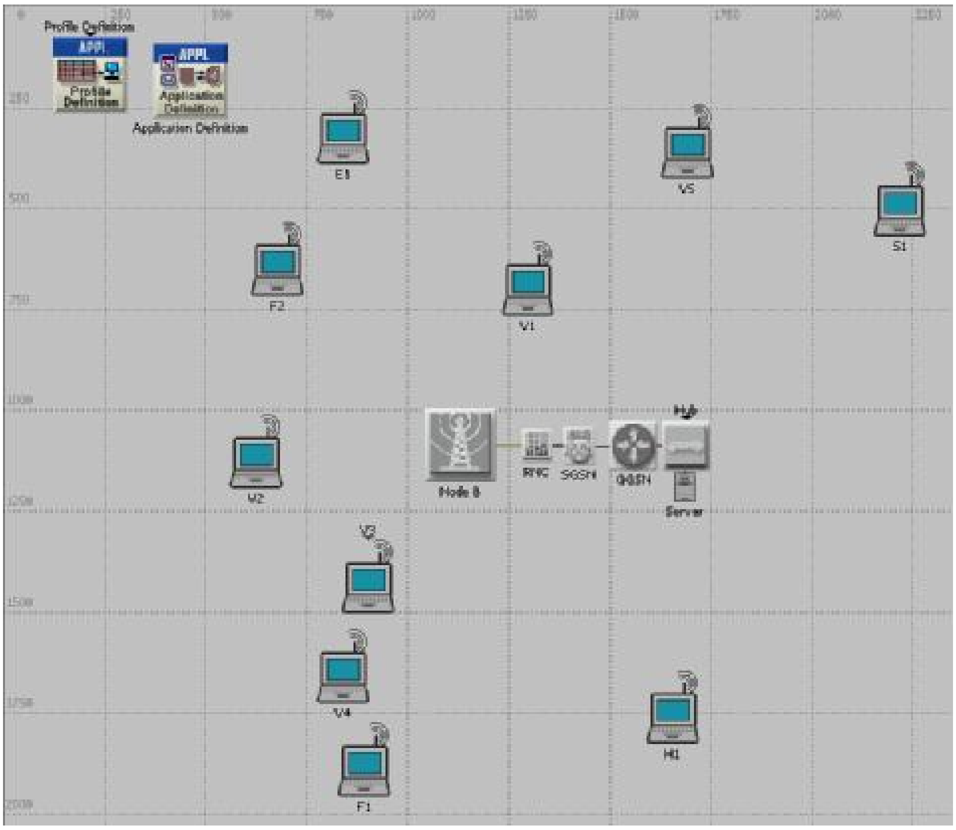



Figure 4.6 – Network Architecture considering 10 MTs.

The nominal cell area is 2.25×2.25 km². The Walfish-Ikegami propagation model was selected, as well the Outdoor to Indoor and Pedestrian Environment pathloss models, in order to compute the power at the receiver. The chosen pathloss is a typical urban and suburban environment, assuming 10.5m between the mean building height and the mobile terminal antenna height, 15m between the mobile terminal and the diffracting edges, and 80m as the average separation between rows of buildings. The BS antenna height is assumed near mean rooftop level. Important UMTS parameters are defined at the RNC, such as: the activity factor (0.65 for conversational class and 1.0 for the other classes); the loading factor (0.7 for DL and 0.6 for UL); the 20ms TTI; ⅓ rate convolutional coding (intended to be used with relatively low data rates). The other elements are the SGSN, the GGSN and a hub connecting them to a server.

### 4.3.2 Applications and Profiles

Seven different applications are considered: VoIP (OPNET does not consider CS), Streaming, FTP, HTTP, E-mail, MMS and Video-Conference. Although video-conference was initially defined, it was not considered due to the very small percentage of users. It is also important mentioning that OPNET does not distinguish between e-mail and MMS, so, the only difference between them is mainly the file size. All applications, except MMS, were defined by [Seba07] and they can be seen at Annex B. The Profile configuration has one main parameter, the inter-repetition time, which is the time between sessions per user (one user, one application only). By using the inter-repetition parameter, one can determine the number of calls per hour. Two profiles were defined, Heavy and Light, which can be seen at Annex C; the former one was also defined by [Seba07]. Table 4.1 represents important application and profile parameters.

Table 4.1 – Application and profile main parameters.

| QoS Class | Service | Parameter | Value | |
|---|---|---|---|---|
| | | | Scenario | |
| | | | Light | Heavy |
| Conversational | VoIP | Mean Call Duration [s] | 120 | 120 |
| | | Calls/Hour | 6 | 6 |
| Streaming | Streaming | Mean File Volume [kB] | 17500 | 17500 |
| | | Calls/Hour | 2 | 4 |
| Interactive | FTP | Mean File Volume [kB] | 2550 | 2550 |
| | | Calls/Hour | 3 | 12 |
| | HTTP | Mean Page Volume [kB] | 34.4 | 34.4 |
| | | Calls/Hour | 4 | 12 |
| Background | E-mail | Mean File Volume [kB] | 100 | 100 |
| | | Calls/Hour | 4 | 12 |
| | MMS | Mean File Volume [kB] | 165 | 165 |
| | | Calls/Hour | 2 | 12 |

OPNET limitations considering UMTS were mentioned before. One of them is the throughput limitation to each QoS class. Table 4.2, shows those limitations, where one can see that the maximum data rate for streaming is 12.2 kbps, while for Interactive and Background classes it is only 64 kbps. These values could be acceptable for UL (mainly because many applications only have signalling traffic this way), but for DL they are just too low. So having low data rates UL generates lower traffic in UL.

In order to study network behaviour, several scenarios with different number of users were simulated, considering Voice and Data Centric (their distribution respects Table 3.4) and of course Light and Heavy profiles. The number of users ranges from 10 to 40 (5 users steps). The lower limit is a situation where everyone should be served, while the higher one presents a significant rejection probability. At the end 28 different scenarios were assessed. Table 4.3 presents the number of calls per hour, per profile and per application for the Voice Centric scenario, while Table 4.4 presents for the Data Centric one. One can see that some scenarios are really demanding.

Table 4.2 – Maximum bit rate per QoS class (UL &DL).

| QoS Class | Throughput [kbps] |
|---|---|
| Conversational | 12.2 |
| Streaming | 12.2 |
| Interactive | 64.0 |
| Background | 64.0 |

Table 4.3 – Number of calls per hour, per profile and per service for Voice Centric.

| $N_u$ | VoIP | | Streaming | | | FTP | | | HTTP | | | E-mail | | | MMS | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| | $N_i$ | $l_{L/H}$ | $N_i$ | $l_L$ | $l_H$ | $N_i$ | $l_L$ | $l_H$ | $N_i$ | $l_L$ | $l_H$ | $N_i$ | $l_L$ | $l_H$ | $N_i$ | $l_L$ | $l_H$ |
| 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 | 36 | 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 |

where:

- $l_L$ : calls per hour for Light profile;
- $l_H$ : calls per hour for Heavy profile.

Table 4.4 – Number of calls per hour, per profile and per service for Data Centric.

| $N_u$ | VoIP | | Streaming | | | FTP | | | HTTP | | | E-mail | | | MMS | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| | $N_i$ | $l_{L/H}$ | $N_i$ | $l_L$ | $l_H$ | $N_i$ | $l_L$ | $l_H$ | $N_i$ | $l_L$ | $l_H$ | $N_i$ | $l_L$ | $l_H$ | $N_i$ | $l_L$ | $l_H$ |
| 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 |

## 4.4 Analysis of Results

As mentioned before OPNET offers a huge set of network parameters that can be analysed. In this thesis, one has selected some Node or Object Statistics (throughputs, packet delays, number of retransmissions, requests granted and queued), Global Statistics (applications' throughput and applications' delay) and Point-to-Point Statistics (throughputs) . In order to present valid results, each scenario was simulated ten times [Seba07].

### 4.4.1 Node Statistics

From Figure 4.7, one can see the total UL throughput at Node B, which is the total throughput of the traffic received by the Node B in UL. These are the mean values in one hour for each scenario and for the Light profile. Although, the Voice Centric average throughput is higher than the Date Centric one, one must take the high standard deviation values into consideration, Figure 4.8, which means those throughput values are not so different from each other, so, more simulations should have been done in order to reduce the standard deviation and increase the accuracy of these results. The Voice Centric throughput is higher then the Data Centric one for a main reason: OPNET always provides service first to Conversational and Streaming classes, so for Data Centric scenarios there are more Interactive and Background users not being served, because of VoIP and Streaming clients.

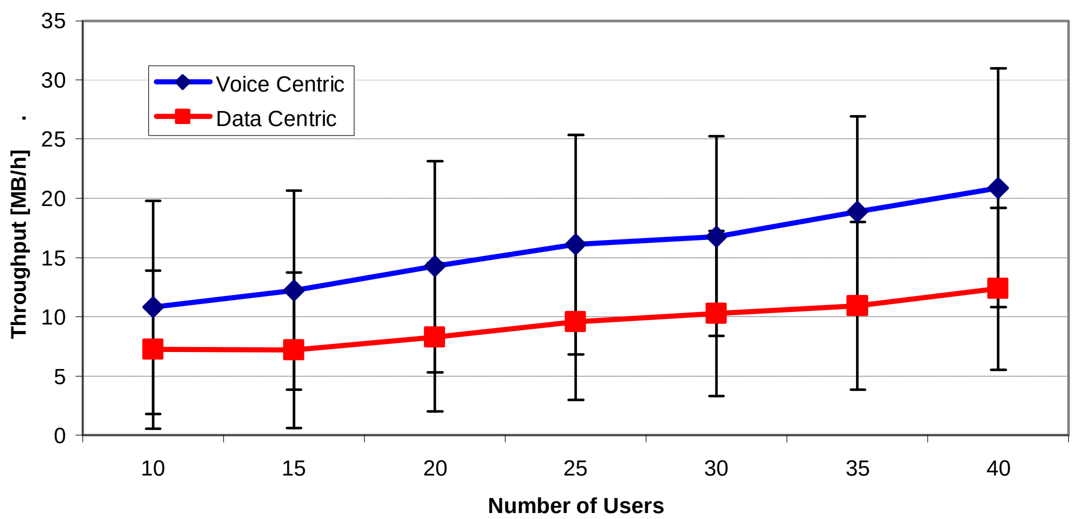



Figure 4.7 – Node B total UL throughput for Light scenario.

As one expected, the average throughput increases with a higher number of users, until it reaches around 20 MB/h (Voice Centric). These values are not higher mainly due to OPNET limitations, because data rate values are limited to 12.2 kbps and 64 kbps, and also because of DL limitations (applications have a much heavier profile in DL). It is also important keeping in mind that these results are for a single sector, and for 3G, and not for High Speed. From Figure 4.9, one can see the total DL throughput at Node B, which is the total throughput of the traffic transmitted by the Node B in DL, and

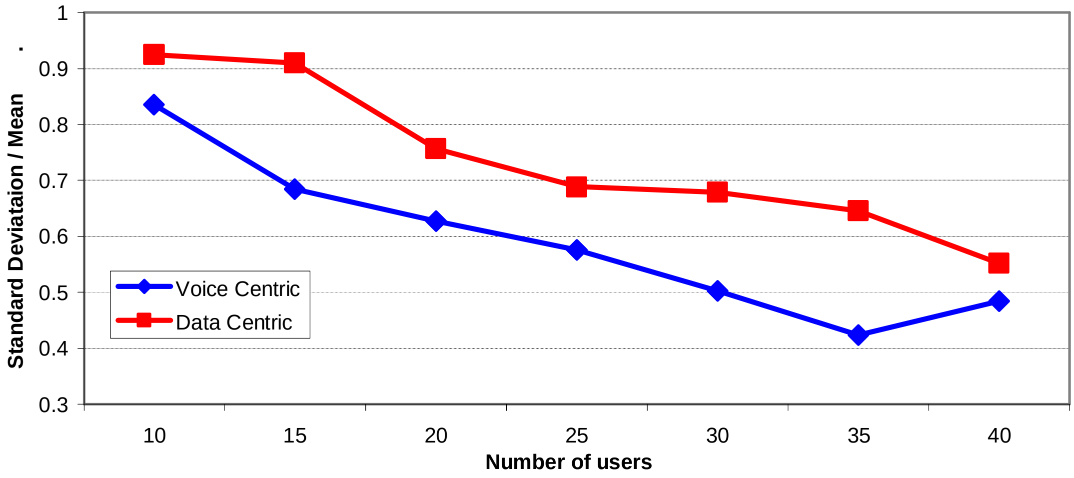



Figure 4.8 – Standard Deviation over Mean for Node B total UL throughput (Light profile).

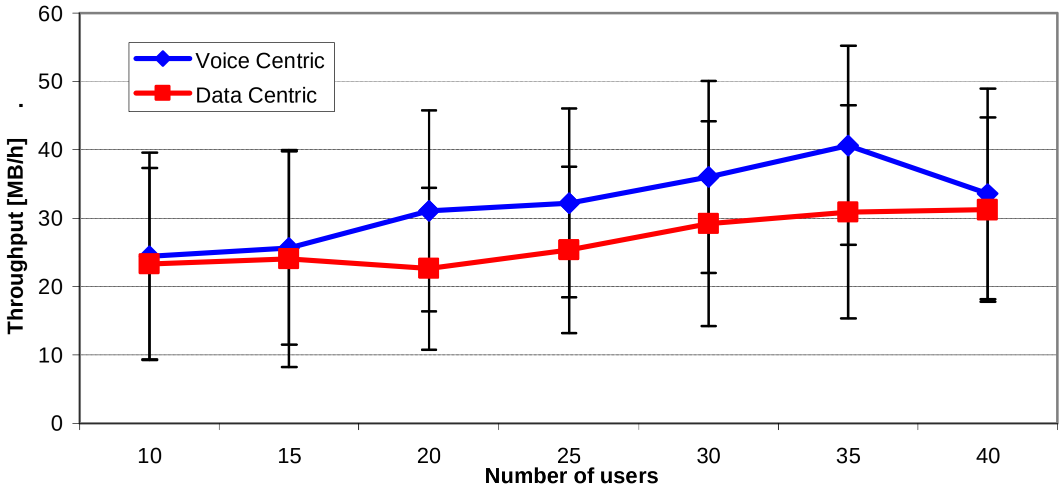



Figure 4.9 – Node B total DL throughput for the Light scenario (extracted from [Antu07]).

realise the global throughput is around two times higher in DL. It happens for two main reasons: a higher loading factor, and because generally the applications are more demanding in DL direction.

All previous results were for the Light profile. Figure 4.10, shows a comparison between the Heavy and Light profiles: there is not much difference between them, because the Light profile is already very demanding. So, this chapter only presents results for the Light scenario.

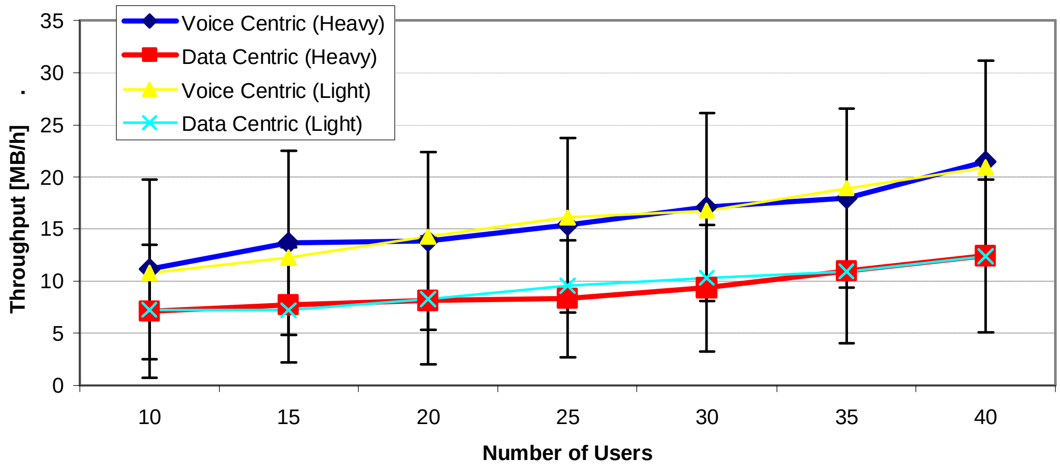



Figure 4.10 – Comparison between Light and Heavy profiles.

In order to study network's behaviour, one needs to check if it is really the channel between the MTs and the Node B, that is limiting the communication. Figure 4.11 presents data from three different connections (Voice Centric only): MTs->NodeB (presented before as Node B total UL throughput), NodeB->RNC and RNC->SGSN. It is the connection between MTs and Node B the biggest obstacle to a better communication. The NodeB->RNC throughput achieves much higher values, around two times higher. As expected (only one Node B for the SGSN), the RNC->SGSN throughput is limited by the Node B total UL throughput.

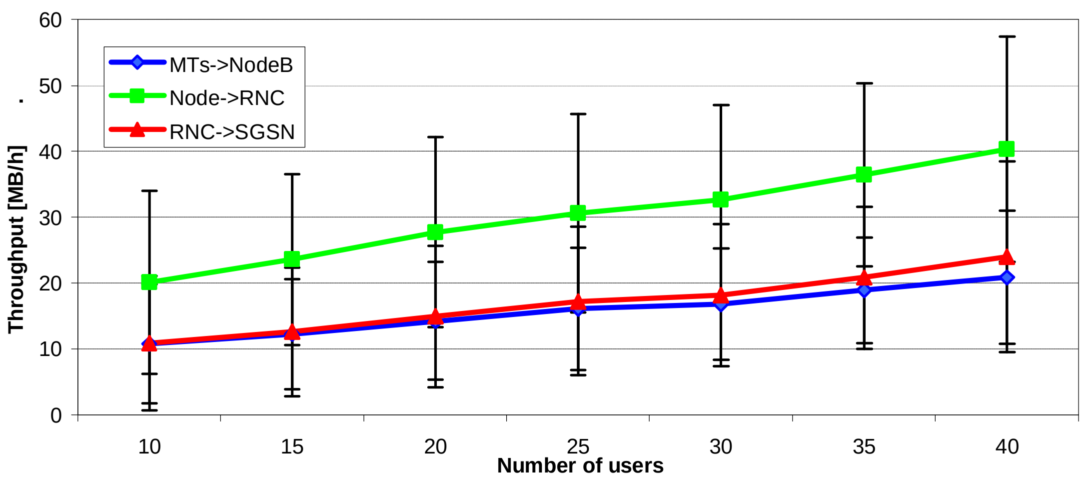



Figure 4.11 – Throughput between different network nodes (Voice Centric only).

Looking at Figure 4.12, one can justify some previous results, like the throughput limitation (why it does not achieve higher values). There are very high percentages of queued requests (in opposition to the decrease of the granted requests), mainly for more than 10 users, for both Data and Voice Centric scenarios. Light and Heavy profiles have similar results.

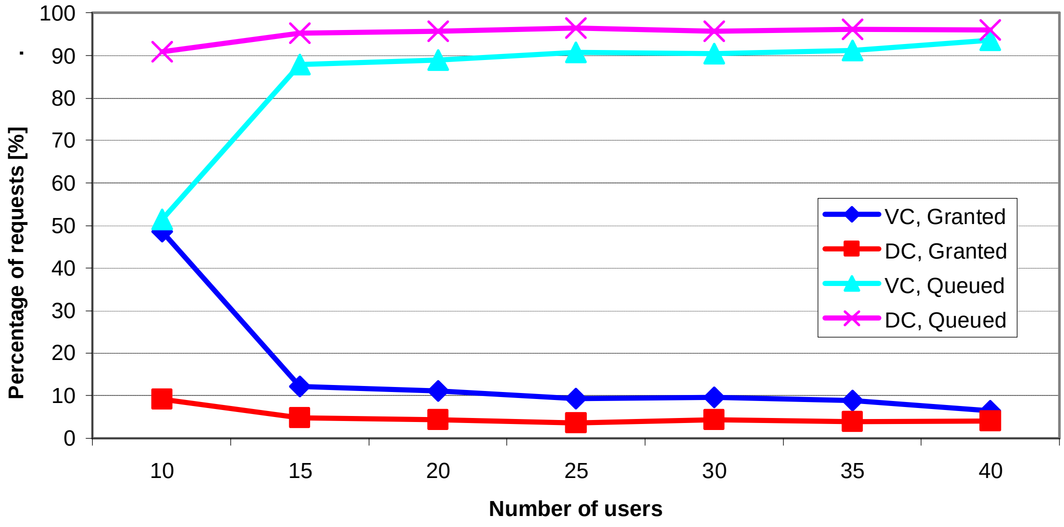



Figure 4.12 – Percentage of requests granted and released.

It is also possible to analyse the percentage of requests granted, Figure 4.13 and queued, Figure 4.14, for the different QoS classes. The presented values are a percentage of the total number of requests, for the Voice Centric distribution [Antu07]. The results do not show any surprise. All VoIP (Conversational) and Streaming requests are granted and none is queued, justifying Figure 4.7, and why Voice Centric throughput is higher than the Data Centric one.

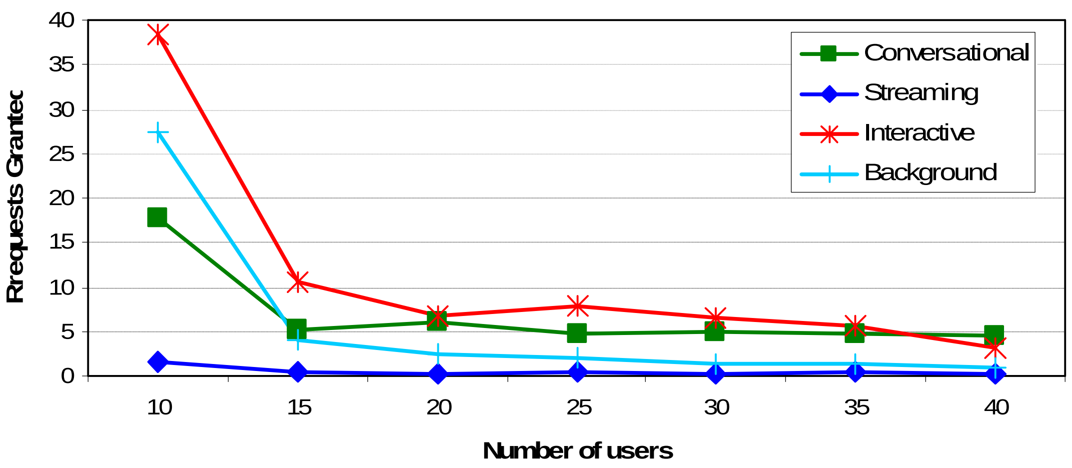



Figure 4.13 – Percentage of requests granted per QoS.

There is another reason that can justify previous results: the number of UL transmissions required for sending a PDU (average value of all transport channels), Figure 4.15 presents. It is slightly higher than one, so the connection between the MTs and the Node B can be considered a good channel. Because it is higher than one (if only one transmission required, it would be a perfect channel), around an average value of 1.025 (low standard deviation, between ± [0.11, 0.15]), it contributes to downgrading the throughput.

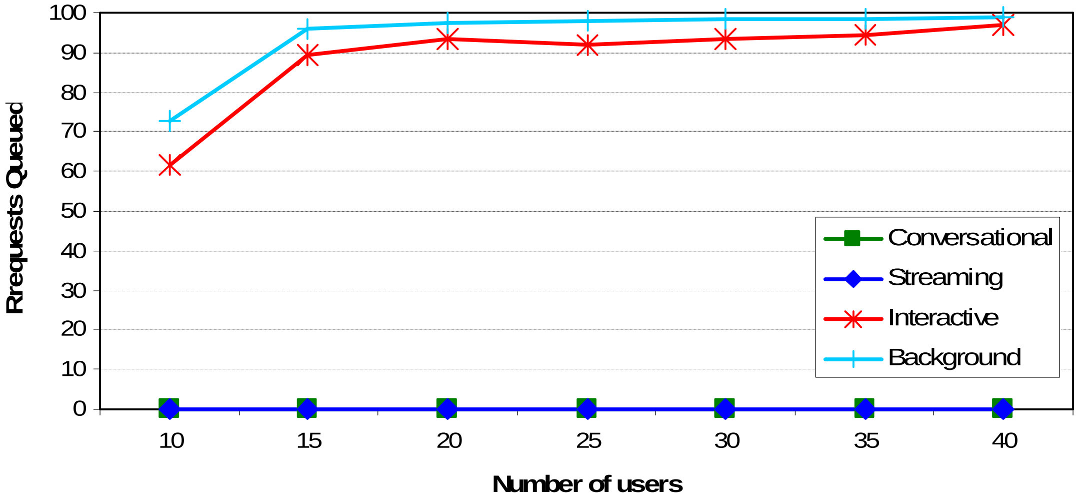



Figure 4.14 – Percentage of requests queued per QoS.

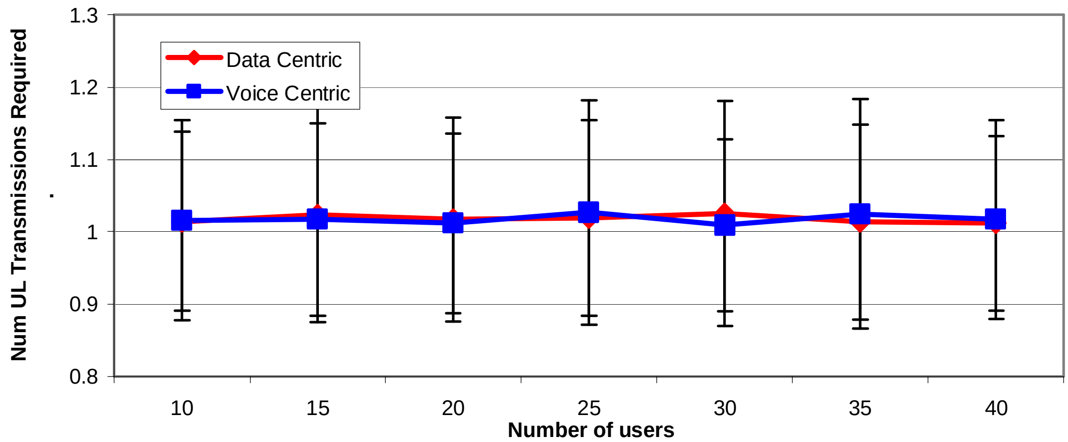



Figure 4.15 – Average number of UL transmissions required.

Analysing the maximum number of UL transmissions required, Figure 4.16, one realises this number can go up to 8 retransmissions. It is a very high number, which increases significantly the average number of UL required transmissions. If one limits the number of retransmissions, then one could upgrade the data rate. At the same time there would be some miss detections.

Finally, one can study the UL retransmission delay (average value for all MTs), which is the time required to retransmit successfully a PDU in RLC acknowledge mode. Considering the Interactive class one can see in Figure 4.17 how high this value is, almost achieving half a second. The UL retransmission delay is almost the same, no matter if it is Voice or Data Centric.

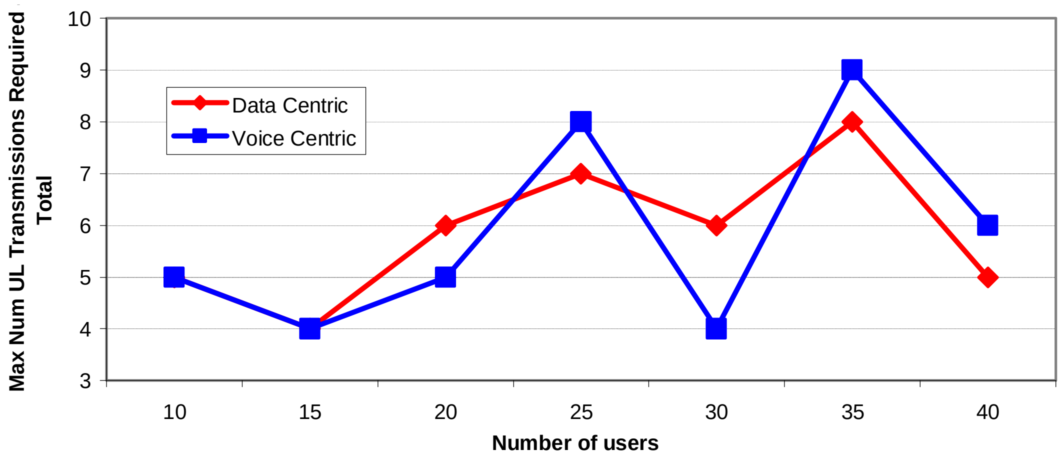



Figure 4.16 – Maximum number of UL retransmissions required.

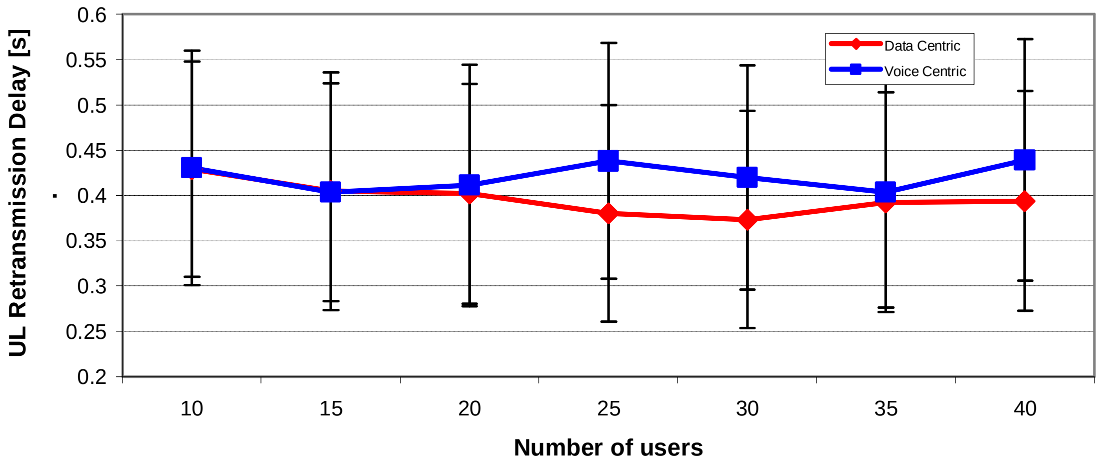



Figure 4.17 – UL Retransmission Delay for Interactive class.

## 4.4.2 Global Statistics

In order to study application's behaviour through the network, one should analyse global statistics. Multiple processes, all at different locations in the models system, can contribute to the same shared statistic. In other words, it means global statistics can be used to gather information about the network as whole.

### 4.4.2.1 Conversational Class

One starts with the highest priority class: Conversational. Figure 4.18 represents the average value of the total transmitted throughput by all VoIP users. As mentioned before, the network always tries to serve first VoIP clients', which is why the average throughput is constant, nevertheless the number of users, being around 2 kbps.

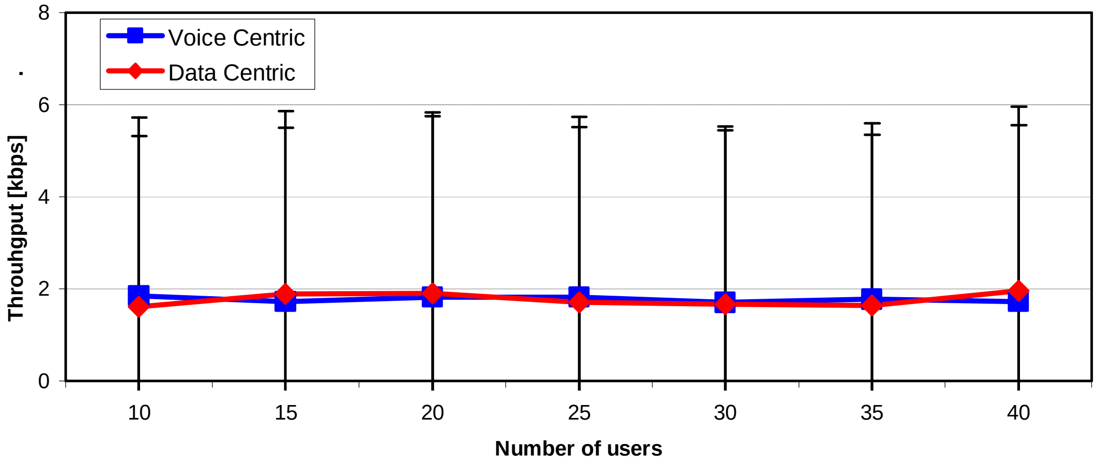



Figure 4.18 – Average value of the total transmitted throughput by VoIP users.

By analysing the average transmitted throughput per user, one can also study some parameters that give a big contribution to understand the network's performance to VoIP users, like the packet end-to-end delay (or mouth-to-ear delay, being measured from the time it is created to the time it is received), and the packet delay variation (variance among end-to-end delays for voice packets). The former is represented in Figure 4.19 and the latter in Figure 4.20.

Results are not good enough for VoIP, because it has to be real time. First of all, the average transmitted throughput is too low, even if it is the average value over one hour, even if the clients may not be doing calls all the time, even if it is PS, and even if many packets are sent at a maximum throughput of 11.8 kbps. There is also the very high packet end-to-end delay, around 1.175 s, three times bigger than the one tolerable by human perception. Although the average transmitted throughout is constant despite the number of users, one can see in Figure 4.20 how fickle the network's behaviour can be with many VoIP users.

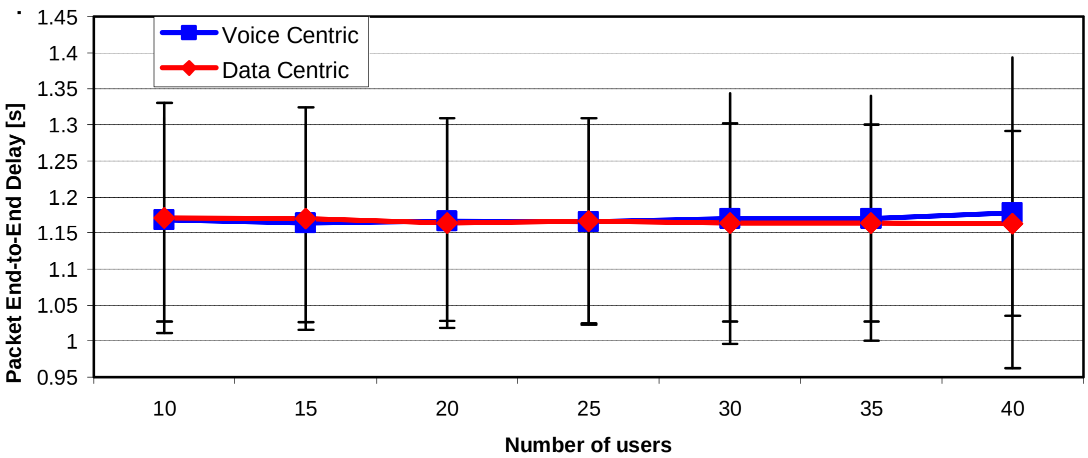



Figure 4.19 – Packet end-to-end delay for VoIP users.

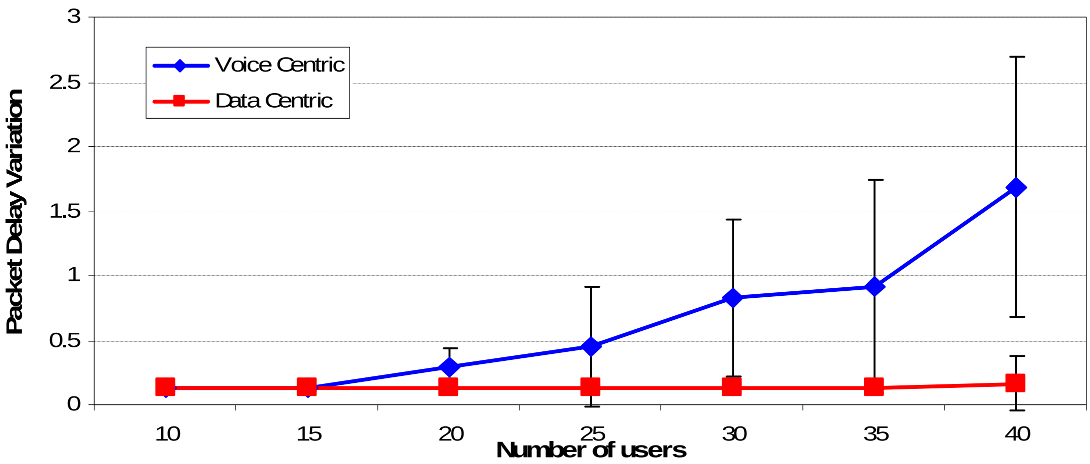



Figure 4.20 – Packet delay variation for VoIP users.

One can say 3G networks do not provide the best quality for VoIP customers. However, why should operators concern about this if they provide CS voice? Probably, if thinking in a near future about Skype, or later on vertical handover with different systems beside 2G. Comparing the average transmitted/received throughput (Annex D has important DL results like the average DL throughputs per application, extracted from [Antu07]) one realises, not surprisingly, that the same result has been achieved.

#### 4.4.2.2 Streaming Class

Figure 4.21 shows the average value of the total transmitted throughput by all streaming users. It is a very low throughput, meaning the UL channel is only used for signalling. If one looks at the average DL throughput (Annex D), one can see that it has a constant 2 kbps, which means the network gives priority to Streaming, instead of Interactive and Background classes.

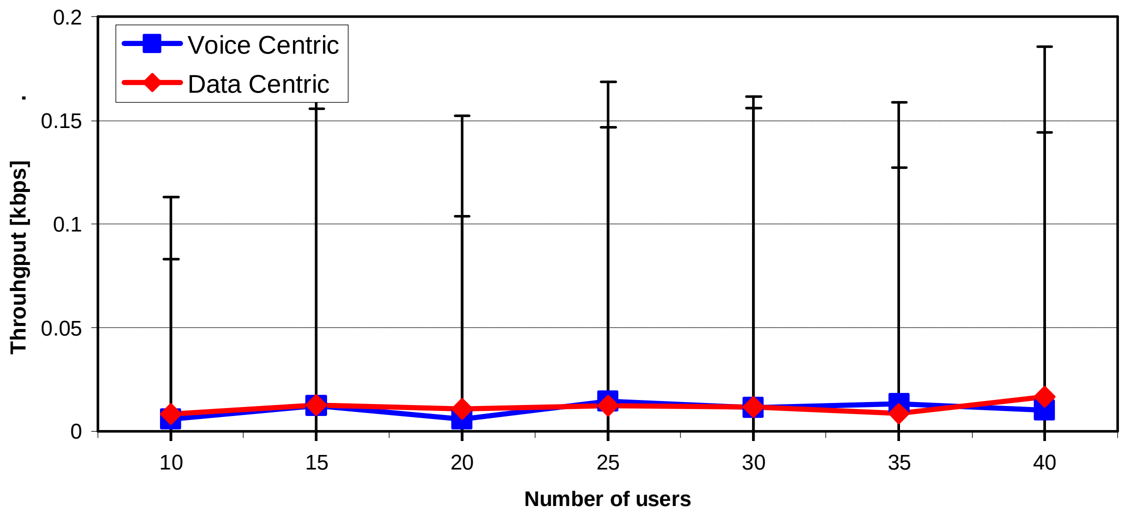



Figure 4.21 – Average value of the total transmitted throughput by Streaming users.

Figures 4.22 and 4.23 present packet end-to-end delay and packet delay variation respectively. The former one presents a low value, around 0.35 s, which can be considered a very good value considering streaming definitions. The latter is very low, which means the network provides a good service to the Streaming clients (they are not too many) despite the number of users in the sector/cell requesting calls. However, the standard deviation is very high, so, there can be unexpected network behaviour for streamers.

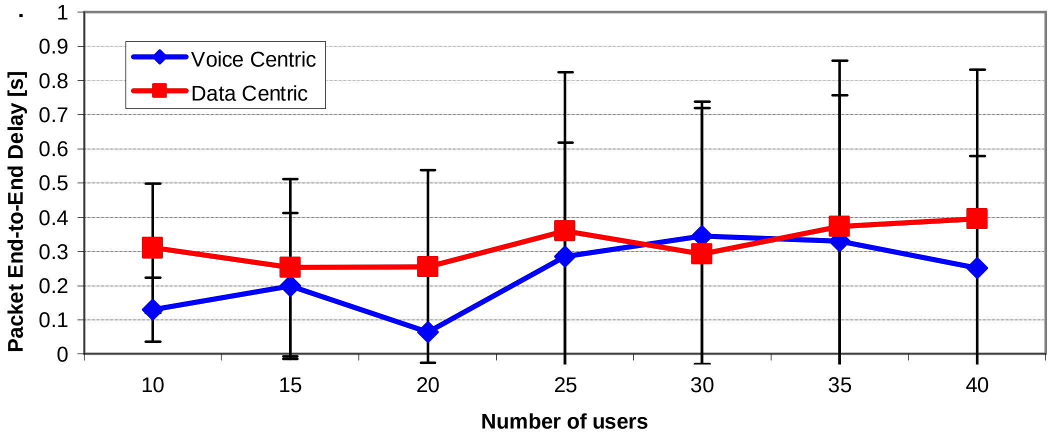



Figure 4.22 – Packet end-to-end delay for Streaming users.

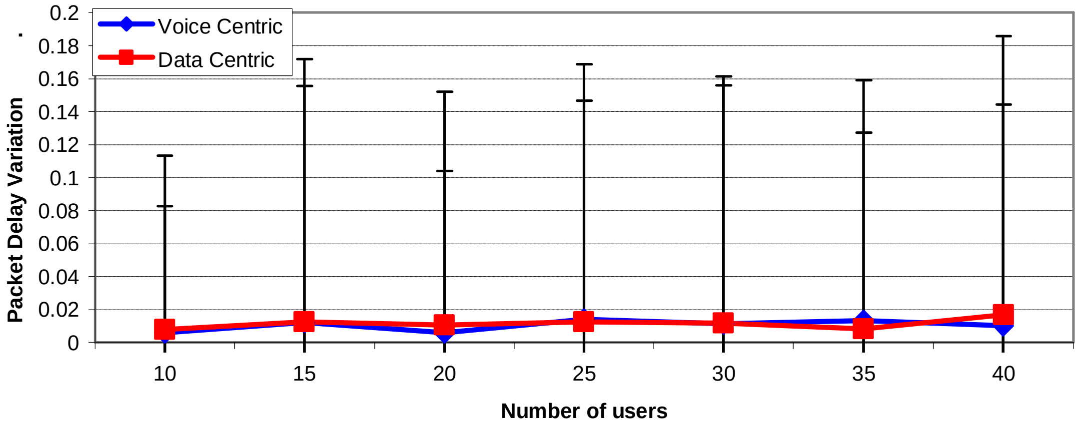



Figure 4.23 – Packet delay variation for Streaming users.

The network's behaviour is acceptable for streaming users, even if the average received throughput is not too high (streaming is ideal for low speed internet access). The delay sensitivity is not a problem because it is quite low.

#### 4.4.2.3 Interactive Class

Figures 4.24 and 4.25 present the average value of the total transmitted throughput by FTP and HTTP users respectively. As expected, the average transmitted throughput is not high for FTP and HTTP. First of all, either applications' attributes do not require a demanding UL channel, or in other words, they do not need to transmit heavy files (they are only doing signalisation and control). Of course demanding FTP uploads could be considered, but in order to simulate a 3G cell this option was not selected. Other restrictions to a higher throughput are the increase of users, and OPNET's priority policy, which favours Conversational and Streaming classes. It is why one has low and decreasing data rates for both applications. If one looks at Annex D, it is easy to understand the differences between DL and UL profiles, as the average throughput is much higher. However, it is not enough to satisfy a file size around 2.5 Mbyte (Annex B).

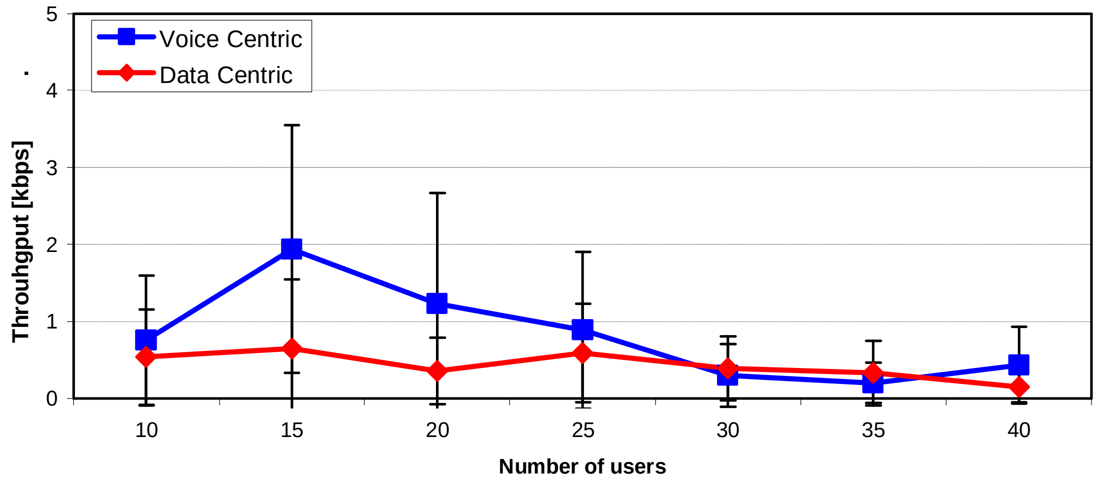



Figure 4.24 – Average value of the total transmitted throughput by FTP users.

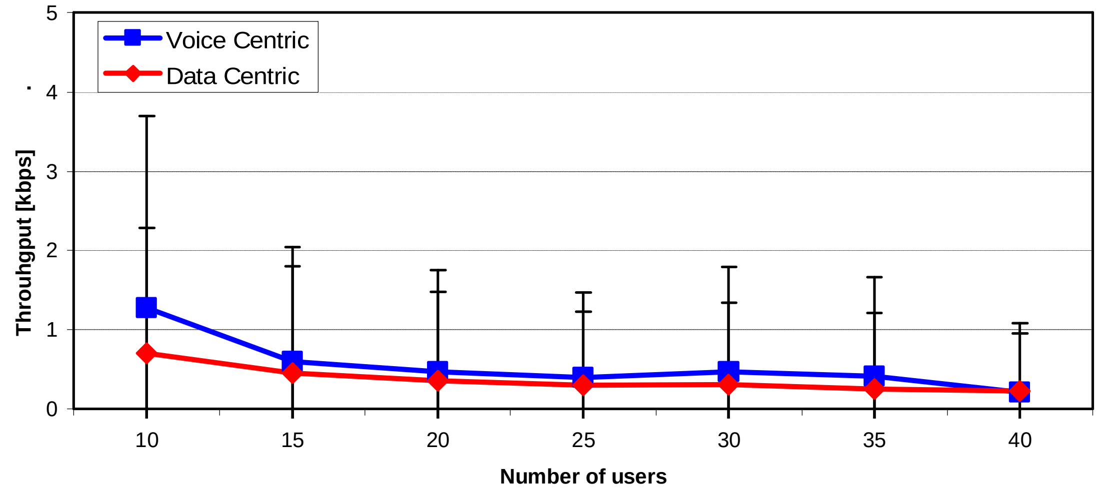



Figure 4.25 – Average value of the total transmitted throughput by HTTP users.

OPNET also allows studying some more important parameters about these applications, like the upload response time (time elapsed between requesting a file and the upload is concluded) for FTP, and the page response time (time required to retrieve the entire page with all the contained inline objects). Unfortunately, due to OPNET malfunction, it was not possible to present solid results for the former parameter. However, it respects the download response time, which means a time domain between 6 to 14 minutes along all users' scenarios (see Annex E). Figure 4.26 represents the page response time. Although it increases while the number of users gets higher, it still has acceptable values for some one doing HTTP.

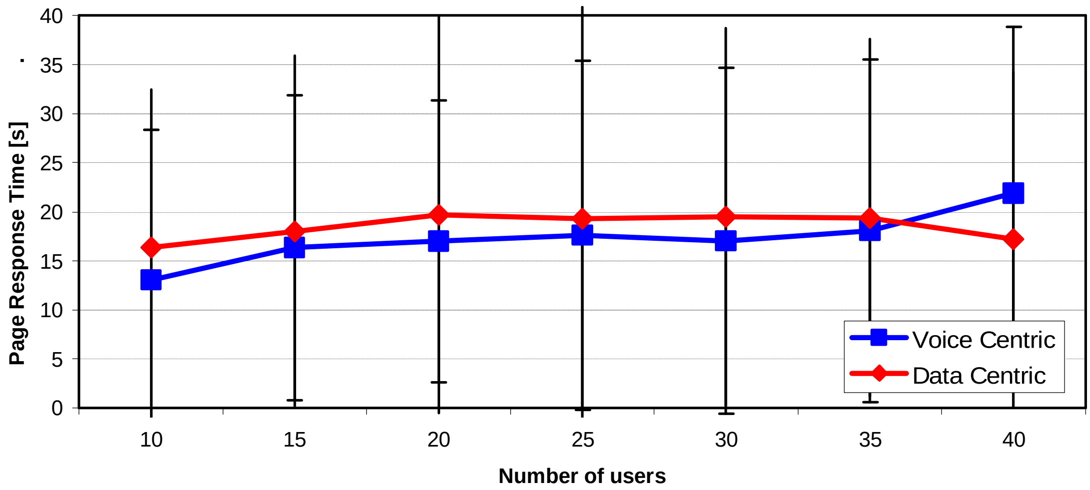



Figure 4.26 – Page response time.

One can make a different analysis for FTP and HTTP. Of course 3G is not a good solution for the former, because it is not able to satisfy clients' expectations, as the throughput is too low and the response time is too high for files of this size. However, it could be a practical solution if doing HTTP.

#### 4.4.2.4 Background Class

Figures 4.27 and 4.28 present the average value of the total transmitted throughput by E-Mail and MMS users, respectively. It is important mentioning that the E-Mail application is configured for receiving and sending e-mails around 300 kbyte, and the low number of MMS users, even in many users' scenarios.

For few users, the E-Mail throughput has relative high values. However, once again due to OPNET's priority policy, by neglecting the Background class, when increasing number of users, it quickly goes down to very low and not acceptable values. Although the same thing goes with MMS, it is not such a problem, because MMS size is very small. Nevertheless, it is very important to mention the very high standard deviation, which has severe impact on the reliability of results.

Figure 4.29 shows the upload response time (time elapsed between sending e-mails to the server and receiving acknowledgment from the e-mail server). This parameter represents E-Mail and MMS together, because OPNET does not distinguish them. Once again, results given by OPNET do not have enough accuracy (and no results for 35 and 40 users in Data Centric). For instance, the upload response time should be increasing with the number of users, and it is not. Nevertheless, UL response time seems to achieve very high values for a 100 kB file size.

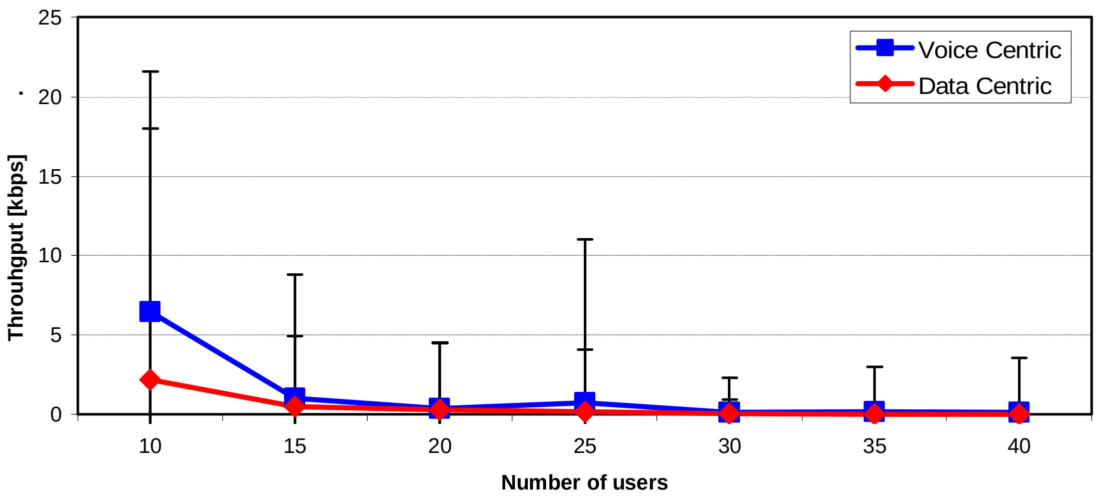



Figure 4.27 – Average value of the total transmitted throughput by E-Mail users.

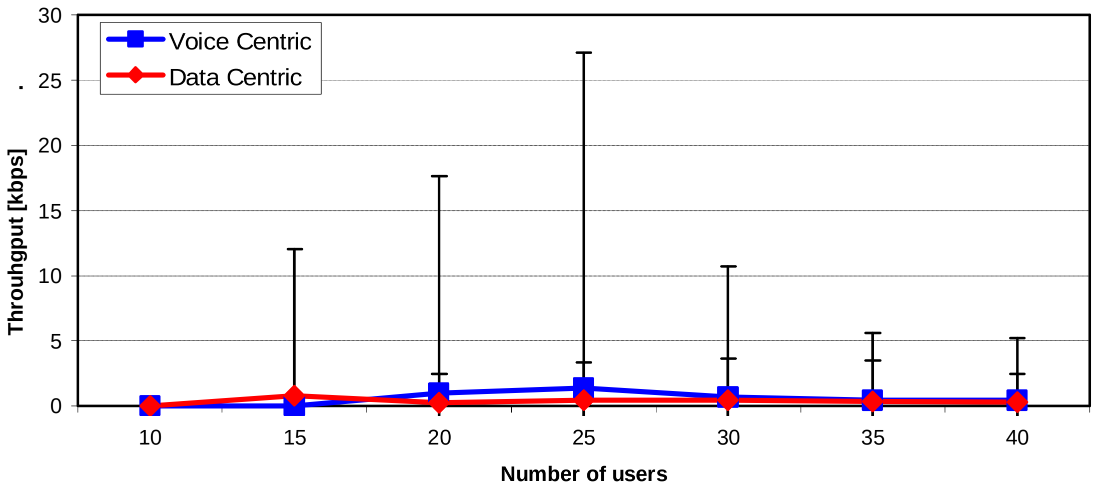



Figure 4.28 - Average value of the total transmitted throughput by MMS users.

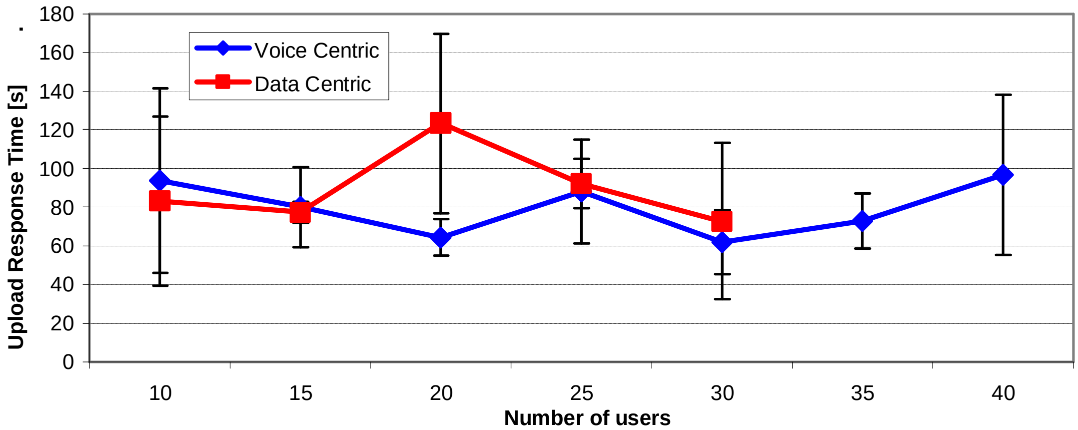



Figure 4.29 - Upload response time for E-Mail and MMS.

The E-Mail throughput has relative high values for few users. However, once again due to OPNET's priority policy and the increasing number of users, it quickly goes down to very low and not acceptable values, even if it is not a real-time application. Throughput is not a problem for MMS, because its size is very small. E-Mail UL response time achieves very high values for a 100 kB file size. It is obvious that the network has severe difficulties serving Background class. It would be very interesting for customers a good e-mail service.

## 4.5 Comparison with theoretical model

It is important to perform a comparison between 3G's practical model and HSUPA's theoretical one. As mentioned before, both models have limitations which have impact on final results. For instance, the theoretical model does not consider several networks' features and the transmission channel quality (admission control focussed), being defined to support applications just in case minimum throughput requirements are fulfilled (SF must be 32 or higher). On the other hand, OPNET does not support High Speed, CS voice (only VoIP), and RABs higher than 64 kbps.

It is also important mentioning the application's profile is not the same, because HSUPA being an enhancement from 3G, it has more ambitious data objectives and it is designed to satisfy different customers' needs. So, FTP uploads and UL streaming were not considered for 3G. As mentioned before, the UL HTTP was neglected in theoretical model, because it is only used for signalisation and control.

The best parameter for comparison is the UL traffic processed by the Node B. For a better analysis one has obtained trend lines for each scenario, given by (3.5), (3.6), (3.7) and (3.8).

For Light and Voice Centric scenario:

- $$T_{[\mathrm{MB/h}]} = 0.3315 N_u + 7.17 \tag{3.5}$$

For Heavy and Voice Centric scenario:

- $$T_{[\mathrm{MB/h}]} = 0.3053 N_u + 7.8501 \tag{3.6}$$

For Light and Data Centric scenario:

- $$T_{[\mathrm{MB/h}]} = 0.1785 N_u + 4.7691 \tag{3.7}$$

For Heavy and Data Centric scenario:

- $$T_{[\mathrm{MB/h}]} = 0.1695 N_u + 4.7365 \tag{3.8}$$

Figures 4.30 and 4.31 present the comparison between the theoretical model and OPNET's UL total traffic processed by the Node B for Voice and Data Centric, respectively. The xx axis' number of users is limited in order to respect a low rejection probability.

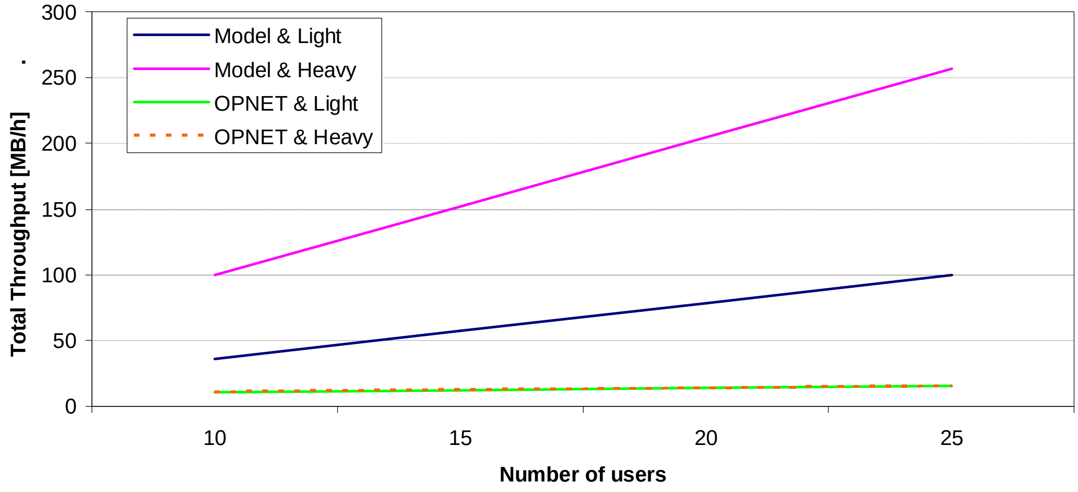



Figure 4.30 - Total traffic processed by the Node B for Voice Centric.

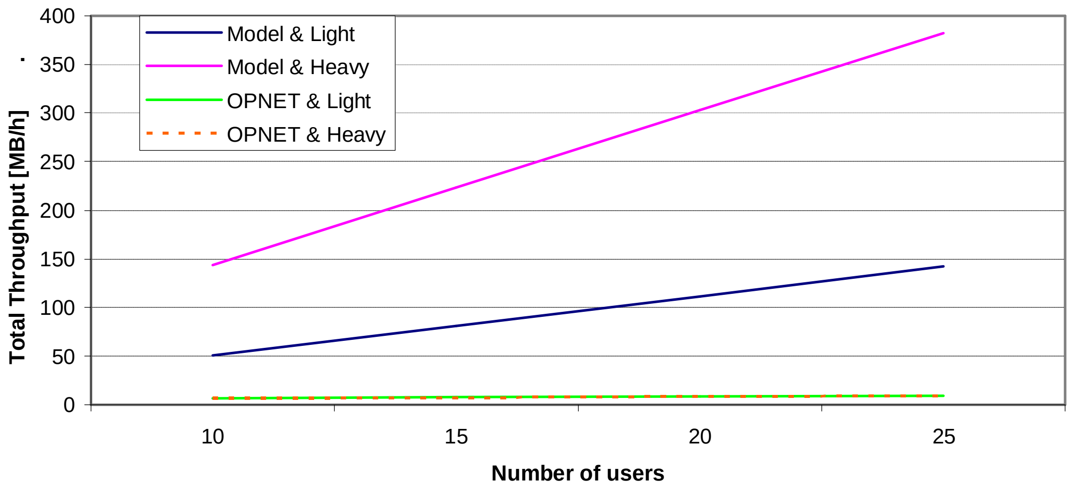



Figure 4.31 - Total traffic processed by the Node B for Data Centric.

Figures 4.30 and 4.31 show the HSUPA enhancements, due to the new physical channel E-DCH, HARQ and fast Node B scheduling. However it is important reminding the theoretical model and OPNET limitations mentioned before. The model estimation of the total throughput processed by the Node B does not include the average throughput per application parameter.

Figure 4.32 shows the HSUPA theoretical model global traffic gain comparing with the one obtained with simulation (basic 3G), for both profiles and scenarios. However, Figure 4.32 looses accuracy with the increasing number of users, specially for Data Centric profile because it has a lower number of users served (around 15 users) a maximum 5% blocking probability.

Although 3G basic version presents good results for same applications, it is not a good solution for someone who needs an internet connection. Nevertheless HSUPA and HSDPA ([Antu07]) can provide interesting data rates, and they can be a good solution when no cable internet is available.

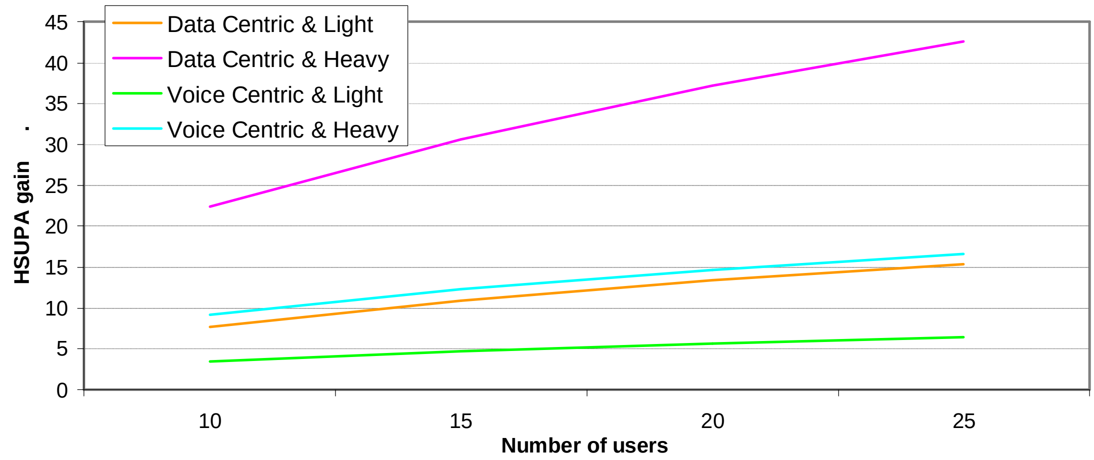



Figure 4.32 – HSUPA global traffic gain over basic 3G.

# Chapter 5

# Conclusions

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

This thesis analyses the HSUPA deployment on top of a UMTS basic network. A theoretical model was developed in order to predict the impact on interference and cell coverage. The model outputs were then compared with 3G's OPNET Modeler simulator results.

The theoretical model was implemented in a simple C++ program, considering link budget's COST 231 Walfisch-Ikegami propagation model and WCDMA/HSUPA admission control algorithms. Enhanced from the work developed by [Lope07] and [Salv07], it supports multiservice and dynamic carrier for UL, while the DL work was carried out by [Antu07]. The model supports CS voice and video-telephony and PS Streaming, FTP, HTTP, E-Mail and MMS. It is important mentioning that data applications have minimum requirements, i.e., they require at least SF32, in order to achieve a good performance and client's satisfaction. The network's behaviour analysis is made for an instant in time (snapshot), reacting to the increasing number of users in a single cell, where all users are at the maximum distance. Two different profiles were considered, the first being based on voice users, and the second on data applications (Voice and Data Centric respectively). The analysis was carried out by varying parameters like the environment, frequency, interference, the priority policy algorithm, etc.. The theoretical model presents network behaviour outputs like, the number of users served, average throughput per users, maximum cell radius for R99's and HSUPA's, Based on these outputs, considerations about the global traffic processed by the Node B in the UL direction are made.

The OPNET Modeler was used to simulate the network. Network architecture was defined considering the following nodes: MTs, Node B, RNC, SGSN, GGSN, and a Hub connecting to the Server. Also applications and profiles were defined considering important parameters, like mean call duration, file size, inter-repetition time, type of service, destination, etc.. Then, two more profiles were created, the Light profile (used as a reference) and the Heavy one. The main difference between them is the amount of data to be processed. In order to statistical accuracy, each scenario was simulated ten times.

Concerning the theoretical model, it is very important mentioning the factors that affect the admission of new users. There are two technical limitations: the number of available codes and the loading factor. Although the load factor has been increased for HS technology (from 0.6 to 0.9 in UL), it is still the main limitation, due to the minimum SF32 required for data applications. There can be a third limitation to the number of users: the number of users served in DL. Although the latter factor it is not studied in this thesis, considering the work developed by [Antu07], it should not be a problem, as HSDPA is a much more mature technology.

Although Voice and Data Centric profiles have different services' distributions, when the maximum loading factor is achieved, data applications are gradually expelled by the admission control algorithm, because the model always satisfies first voice users. Regarding the number of users served, due to the load factor limitation, it is only possible serving a maximum number of 8 HSUPA data users per carrier. These HSUPA users are requesting 90 kbps at the physical layer, and in good conditions they can achieve 75 kbps peak. Considering voice users, they request such a low load factor that it is possible to allocate up to 70 users. Due to the percentage of voice users, admission control algorithm starts rejecting data clients, for the Voice Centric profile, when the total number of users is more than 19, while for the Data Centric one it is more then 13. It is important mentioning that in a real network, no HSUPA users are rejected due to low throughput available (for obvious reason). While the network has available codes, it will allocate them, even if it means data rates around 1 and 2 kbps per users. It is also important reporting that the UL theoretical model is not very sensitive to different priority policy algorithms, mainly due to the low achievable throughput (at least for HSUPA's terminal category 3).

These results can look like not being very interesting. Nevertheless, these numbers are not that low because tri-sectorised cells allow something around 24 HSUPA users. This number can go 4 times higher, as the network can support 4 carriers. This way, operators can supply real broadband Internet, because there is enough capacity for higher data rates and a higher number of users is served.

It was referred how fast the system achieves loading factor saturation. After that, the loading factor enters an oscillatory regime, allocating voice users while expeling HSUPA's. If one looks at the required loading factor by one user at 1250 kbps throughput, one sees it has a value around 0.55 (more then 50% system's capacity, meaning only one user can be served with maximum throughput), and at 75kbps goes around 0.107, while one CS voice user only needs 0.020. Considering these values, the system achieves a maximum 2.0 Mbps at Node B. For a dedicated High Speed carrier, the system reaches 600 kbps when load factor saturation occurs. If one simulates the system ranging the inter-to-intra-cell interferences from 0.391 to 0.65, one can see there is no big difference between them. However, lower inter-to-intra-cell interferences ratio generates lower load factors. This means more users will be served.

Considering the Pedestrian environment, the maximum cell radius (all users at the same distance from Node B) for R99 users ranges from 1.7 to 0.9 km, while for HSUPA users ranges from 1.2 to 0.6 Km. In the Voice Centric profile and before the system starts rejecting clients, the radius is around 1.64 and 0.85 km respectively, while for the Data Centric one it is 1.68 km and 0.73 km. In the Vehicular environment, there are losses around 400m for R99 and 200m for High Speed. In the Indoor environment, the losses can be up to 3 times higher than the latter one.

By analysing the global traffic processed by the Node B in UL, one realises that the traffic for the Light scenario is 3 times lower than the Heavy one, which is about 200 MB/h. However it is important mentioning these values do not include the average throughput per application, just the number of users, the mean call duration, and the mean file size.

It is obvious that the theoretic approach is always very optimistic, enhancing the outputs. This study has limitations, because it does not consider many network features and also does the transmission channel quality. Besides that, the model can not focus on applications' latency, as the analysis is not made in time. However, one can conclude that High Speed technology can be a serious wireless Internet connection. Since this model only regards one single carrier, if more carriers are available, then it would enhance system capacity.

OPNET Modeler is a simulation tool for communication networks that allows the study and design of network devices, protocols and applications. However, the UMTS module based on Release 99 from 3GPP has some limitations, which have impact on results. Due to the scope of this thesis, the main one is the throughput limitation. In fact, OPNET only allows 12.2 kbps for Conversational and Streaming classes, while Interactive and Background is limited to 64 kbps. The network behaviour is studied by increasing the number of users from 10 to 40 (5 user's steps). Another important limitation is the fact that no CS can be simulated, so voice is represented by VoIP.

Considering the total traffic received by the Node B in UL (mean value in one hour) one realises it only has a slight increase until it reaches 20 MB/h. This value is not higher mainly due to OPNET data rates limitations. Although, the Voice Centric average throughput is higher than the Date Centric one, one must take the high standard deviation values into consideration, which means those throughput values are not so different from each other. The Voice Centric throughput is higher than the Data Centric one for a main reason: the OPNET priority policy always provides service first to Conversational and Streaming classes. So Interactive and Background classes are always passed over, meaning the Data Centric profile with many more data users is punished.

Considering the total DL throughput transmitted by Node B, it is around two times higher than the UL one, for two main reasons: a higher loading factor, and because generally applications are more demanding in DL. Comparison between the Heavy and Light profiles' global transmitted throughput in UL shows that there is not much difference between them, because the latter is already very demanding. One also presents results showing that the connection between MTs and Node B is the biggest obstacle to a better communication.

Previous results can be justified by analysing the number of granted and queued requests. There are very high percentages of queued requests (in opposition to the decrease of the granted requests), for more than 10 users. It is also possible looking at the percentage of requests granted and queued for the different QoS classes. So, all VoIP and Streaming requests are granted, while Interactive and Background classes have high probability of being queued, which proves OPNET's priority policy. The number of UL transmissions required (average 1.025) shows the connection between the MTs and the Node B can be considered a good channel. However, the UL retransmission delay is very high, almost half a second.

For the Conversational class, one can say 3G networks do not provide the best quality for VoIP customers. The average transmitted throughput is too low (even if it is constant), and there is also very high packet end-to-end delay, around 1.175 seconds, three times bigger than the one tolerable by human perception. However, why should operators concern about this if they provide CS voice?

The network's behaviour is acceptable for streaming users, even if the average received throughput is not too high, around 2 kbps and constant, because it is not a real-time application (streaming is ideal for low speed internet access). The delay sensitivity is not a problem, because it is quite low.

Considering the Interactive class, one should make a different analysis for FTP and HTTP. Of course, 3G is not a good solution for the former, because it is not able to satisfy clients' expectations, as the throughput is too low and the response time is too high for files this size. However, it could be a practical solution if doing HTTP.

Regarding Background class, the E-Mail throughput has relative high values for few users. However, once again due to OPNET's priority policy and the increasing number of users, it quickly goes down to very low and not acceptable values, even if it is not a real-time application. Throughput is not a problem for MMS, because its size is very small. E-Mail UL response time achieves very high values for a 100 kB file size. It is obvious that the network has severe difficulties serving Background class. It would be very interesting for customers a good e-mail service.

The comparison between HSUPA theoretical model and OPNET Modeler R99 simulations for the achievable UL traffic processed by the Node B is presented. Of course, the traffic achieved theoretically is much higher than the one obtained from simulation, but both approaches tend to increase with the number of users. The main reason is pretty obvious: HSUPA is an improvement from basic 3G. Another important factor is both models have limitations which have impact on final results. While theoretical model limitations enhance the outputs, OPNET's downgrade the results. Although 3G presents good results for some applications, it is not a good solution for someone who needs an Internet connection. Nevertheless HSUPA and HSDPA ([Antu07]) can provide interesting data rates, and they can be a good solution when no cable Internet is available.

For future work, it would be very interesting a time domain HSUPA simulation using OPNET Modeler. Besides this, a more advanced release would also allow a more accurate WCDMA study. It would also be very interesting to perform a comparison with a system composed by Mobile WiMAX and local WiFi, because it can be a solid competitor against 3G and all its enhancements. Another interesting study would be a comparison between applications served by WCDMA and very focused technologies on specific services, like DVB-H a technical specification for bringing broadcast services to mobile handsets. And of course, the analysis of 3GPP's Release 7 and 8, also known as Long Term Evolution (LTE).

# Annex A

## Link Budget

In this annex, the link budget is described.

The total path loss is defined by [Corr06]:

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

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.

If diversity is used (3 dB considered for this thesis) the $G_r$ in (A.1) is substituted by:

$$G_{rdiv[\text{dB}]} = G_{r[\text{dB}]} + G_{div[\text{dB}]} \tag{A.2}$$

The Equivalent Isotropic Radiated Power (EIRP) can be estimated for UL by:

$$EIRP_{[\text{dBm}]} = P_{Tx[\text{dBm}]} - L_{u[\text{dB}]} + G_{e[\text{dBi}]} \tag{A.3}$$

where:

- $P_{Tx}$: transmitted power;
- $L_c$: cable losses between emitter and antenna;
- $L_u$: body losses.

The received power can be calculated by:

$$P_{Rx[\text{dBm}]} = P_{r[\text{dBm}]} - L_{c[\text{dB}]} \tag{A.4}$$

where:

- $P_{Rx}$: received power at receiver input.

The UMTS receiver sensitivity can be approximated by:

$$P_{Rx\min[\text{dBm}]} = N_{[\text{dBm}]} - G_{P[\text{dB}]} + SNR_{[\text{dB}]} \tag{A.5}$$

where:

- $N$: total noise power given by (A.6);
- $G_p$: processing gain given by $R_c/R_b$;
- SNR: signal to noise ratio given by $E_b/N_0$.

The total noise power is:

$$N_{[\text{dBm}]} = -174 + 10 \cdot \log(\Delta f_{[\text{Hz}]}) + F_{[\text{dB}]} + M_{I[\text{dB}]} \tag{A.6}$$

where:

- $\Delta f$: signal bandwidth, in UMTS it is equal to $R_c$;
- $F$: receiver's noise figure;
- $M_I$: interference margin.

Some margins must be taken into account, to adjust additional losses due to radio propagation:

$$M_{[\mathrm{dB}]} = M_{SF[\mathrm{dB}]} + M_{FF[\mathrm{dB}]} + L_{int[\mathrm{dB}]} - G_{SHO[\mathrm{dB}]} \tag{A.7}$$

where:

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

The total path loss can than be calculated by:

$$L_{p\,total[\mathrm{dB}]} = L_{p[\mathrm{dB}]} + M_{[\mathrm{dB}]} \tag{A.8}$$

The total path loss is used as an input in the COST 231 Walfisch-Ikegami propagation model, described in detail in [CoLa06], in order to calculate the cell radius as in Section 3.1. COST 231 Walfisch-Ikegami propagation model is valid for [Corr06]:

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

As mentioned before, the SNR used is the required $E_b/N_0$, and the E-DPDCH throughput is a continuous function of the $E_b/N_0$ at the Node B. The values of the $E_c/N_0$, as function of the throughput, Figure A.1, were calculated by interpolation of the curve for FRC6 in Figure 2.4 [Lope07]. The results presented are for a Vehicular A channel, but since there are no other results, the same values were used for all environments. For the same reason was considered FRC6.

To minimize approximation errors, the interpolated function is stepwise:

$$E_c / E_b = \begin{cases} 10 \cdot \rho - 12.5, & 0 \le \rho \le 0.1 \\ 4.2 \cdot \rho - 11.9334, & 0.1 < \rho \le 0.45 \\ 5.187 \cdot \rho^2 + 0.659 \cdot \rho - 11.3473, & 0.45 \le \rho \le 1.0 \\ 6.8 \cdot \rho - 12.3, & 1.0 < \rho \le 1.22 \\ 4.3 \cdot \rho - 9.235, & 1.220 < \rho \le 1.45 \end{cases} \tag{A.9}$$

where:

- $r$ : physical layer throughput.

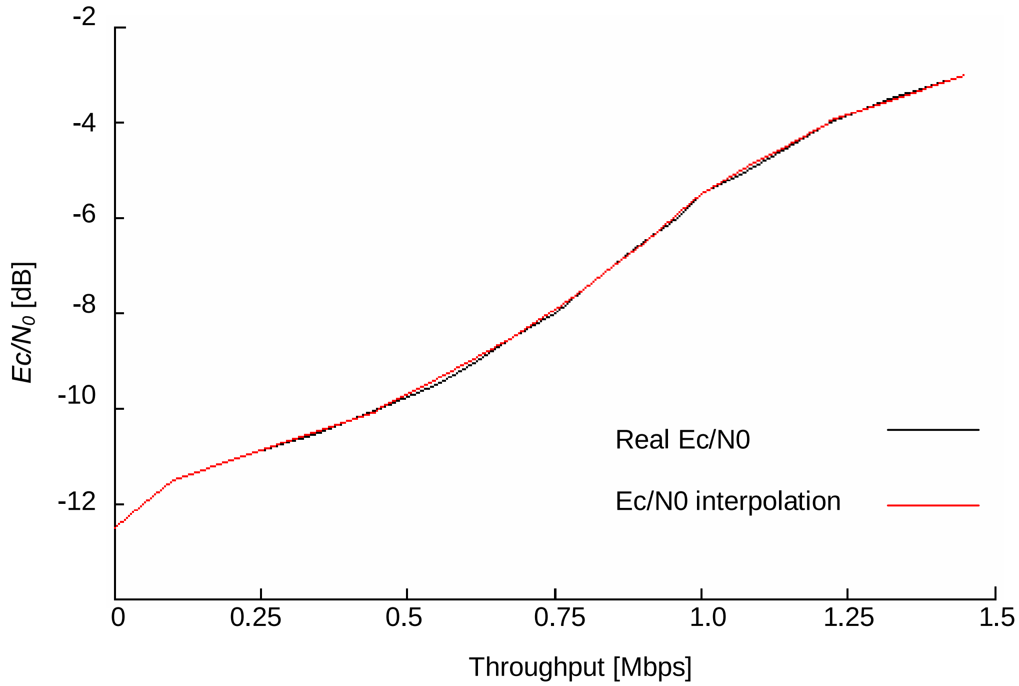



Figure A.1 - Interpolation for the HSUPA FRC6 curve (extracted from [Lope07]).

To ensure approximation validity, the relative mean error value and variance were calculated and are depicted in Table A.11. The values are within the acceptable interval and so, the approximation was used to obtain the $E_c / N_0$ as a function of the intended throughput.

Table A.1. HSUPA relative mean and variance errors (extracted from [Lope07]).

| | **Relative Error [%]** | **Variance** |
|---|---|---|
| HSUPA | 0.38 | $2.04\times10^{-5}$ |

For the sensitivity calculation, the $E_b/N_0$ *should be used*, which can be obtained from $E_c / N_0$ by (A.15):

$$E_b / N_{0[\mathrm{dB}]} = E_c / N_{0[\mathrm{dB}]} + G_{P[\mathrm{dB}]} \tag{A.10}$$

Regarding the situation where it is necessary to calculate the achievable throughput due to the distance that the user is from the Node B, the first step is to determine the path loss associated with the user distance, described in [CoLa06] and [Sant06]. Then, with the path loss calculated, the receiver sensitivity is determined, resulting:

$$P_{Rx[\mathrm{dBm}]} = EIRP_{[\mathrm{dBm}]} - L_{P[\mathrm{dB}]} + G_{r[\mathrm{dB}]} - L_{u[\mathrm{dB}]} \tag{A.11}$$

Considering (A.5) and (A.10), the $E_c/N_0$ given by the user distance is:

$$E_c / N_{0[\mathrm{dB}]} = P_{Rx[\mathrm{dBm}]} - N_{[\mathrm{dBm}]} \tag{A.12}$$

From Figure A.2 is now possible to calculate the user's throughput considering the user's distance.

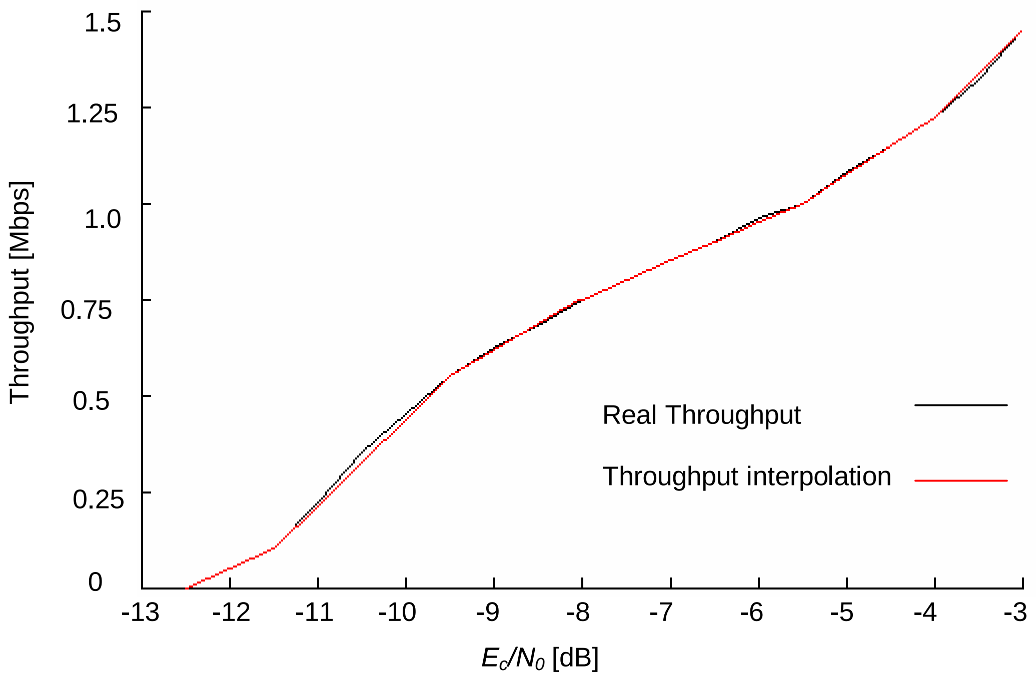



Figure A.2 - HSUPA throughput as function of the $E_c/N_0$ (extracted from [Lope07]).

The expressions in (A.13) were determined using linear interpolation and limiting the error to ensure an error below 5%. The relative mean error and variance of this interpolation are presented in Table A.2.

$$\rho = \begin{cases} 0, & E_c/N_0 < -12.5 \\ 0.1 \cdot E_c/N_0 + 1.25, & -12.5 < E_c/N_0 \le -11.5 \\ 0.225 \cdot E_c/N_0 + 2.6875, & -11.5 < E_c/N_0 \le -9.5 \\ 0.1333 \cdot E_c/N_0 + 1.8167, & -9.5 < E_c/N_0 \le -8 \\ 0.1 \cdot E_c/N_0 + 1.55, & -8 < E_c/N_0 \le -5.5 \\ 0.1467 \cdot E_c/N_0 + 1.8067, & -5.5 \le E_c/N_0 \le -4 \\ 0.23 \cdot E_c/N_0 + 2.14, & -4 < E_c/N_0 \le -3 \\ 1.45, & E_c/N_0 > -3 \end{cases} \tag{A.13}$$

Table A.2. Relative error and variance for the interpolated curves in Figure A.2 (extracted from [Lope07]).

| | **Relative Error [%]** | **Variance** |
|---|---|---|
| HSUPA | 0.70 | $2.56 \times 10^{-4}$ |

# Annex B

# Application Definition

In this annex, the applications used to characterise OPNET are described.

As mentioned before all applications, except MMS, were defined by [Seba07].

Table B.1 – VoIP definition.

| VoIP | |
|---|---|
| **Attribute** | **Value** |
| Incoming Silence Length (seconds) | Exponential (0.456) |
| Outgoing Silence Length (seconds) | Exponential (0.456) |
| Incoming Talk Spurt Length (seconds) | Exponential (0.854) |
| Outgoing Talk Spurt Length (seconds) | 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) |
| RSVP Parameters | None |
| Traffic Mix (%) | All Discrete |
| Signalling | None |
| Compression Delay (seconds) | 0.02 |
| Decompression Delay (seconds) | 0.02 |
| | |
| **Encoder Scheme Used** | |
| **Attribute** | **Value** |
| Codec Type | CS-ACELP |
| Name | G.279.A (silence) |
| Frame Size | 10ms |
| Lookahead Size | 5ms |
| DSP Processing Ratio | 1.0 |
| Coding Rate | 12.2 kbps |
| Speech Activity Detection | Enabled |
| Equipment Impairment Factor | Unknown |
| Packet Loss Robustness Factor | Default |

Table B.2 – Streaming definition.

| Streaming | |
|---|---|
| **Attribute** | **Value** |
| Incoming Stream Interarrival Time (seconds) | Constant (0.04) |
| Outgoing Stream Interarrival Time (seconds) | None |
| Incoming Stream Frame Size (bytes) | Constant (2000) |
| Outgoing Stream Frame Size (bytes) | Constant (2000) |
| Symbolic Destination Name | Video Destination |
| Type of Service | Streaming Multimedia (4) |
| RSVP Parameters | None |
| Traffic Mix (%) | All Discrete |

Table B.3 – Video-Conference definition.

| Video-Conference | |
|---|---|
| **Attribute** | **Value** |
| Frame Interarrival Time Information | 15 frames/sec |
| Incoming Stream Frame Size (bytes) | Constant (533) |
| Outgoing Stream Frame Size (bytes) | Constant (533) |
| Symbolic Destination Name | Video Destination |
| Type of Service | Interactive Multimedia (5) |
| RSVP Parameters | None |
| Traffic Mix (%) | All Discrete |

Table B.4 – HTTP definition.

| HTTP | | | | |
|---|---|---|---|---|
| **Attribute** | | **Value** | | |
| HTTP Specification | | HTTP 1.1 | | |
| Page Interarrival Time (seconds) | | Exponential (39.5) | | |
| RSVP Parameters | | None | | |
| Type of Service | | Best Effort (0) | | |
| Page Properties | | | | |
| Object size (bytes) | Number of objects | Location | Back-End … | Object Group … |
| Lognormal(20000,50000) | Constant(1) | HTTP Server | Not used | Not used |
| Lognormal(14400,252000) | Gamma(47.258,0.232) | HTTP Server | Not used | Not used |

| HTTP (cont.) | |
|---|---|
| **Attribute** | **Value** |
| Server Selection | |
| Initial Repeat Probability | Browse |
| Pages per Server | Exponential (10) |

Table B.5 – FTP definition.

| FTP | |
|---|---|
| **Attribute** | **Value** |
| Command Mix (Get/Total) | 0.95 |
| Inter-Request Time (seconds) | Exponential (600) |
| File Size (bytes) | Lognormal (100000,5000000) |
| Symbolic Server Name | FTP Server |
| Type of Service | Background (1) |
| RSVP Parameters | None |
| Back-End Custom Application | Not used |

Table B.6 – E-Mail definition.

| E-Mail | |
|---|---|
| **Attribute** | **Value** |
| Send Interarrival Time (seconds) | Exponential (360) |
| Send Group Size | Uniform_int (1,5) |
| Send Interarrival Time (seconds) | Exponential (360) |
| Receive Group Size | Uniform_int (1,5) |
| E-Mail Size (bytes) | Lognormal (100000,660000) |
| Symbolic Server Name | Email Server |
| Type of Service | Background (1) |
| RSVP Parameters | None |
| Back-End Custom Application | Not used |

Table B.7 – MMS definition.

| E-Mail | |
|---|---|
| **Attribute** | **Value** |
| Send Interarrival Time (seconds) | Exponential (1200) |
| Send Group Size | None |
| Send Interarrival Time (seconds) | Exponential (1200) |
| Receive Group Size | None |
| E-Mail Size (bytes) | Lognormal (30000,300000) |
| Symbolic Server Name | Email Server |
| Type of Service | Best Effort (0) |
| RSVP Parameters | None |
| Back-End Custom Application | Not used |

# Annex C

# Profile Definition

In this annex, the applications' profiles are described.

Based on the work developed by [Seba07].

Table C.1 – Common configuration for all profiles.

| **All Profiles** | |
|---|---|
| **Attribute** | **Value** |
| Operation Mode | Serial (Ordered) |
| Start Time (seconds) | Uniform (0,120) |
| Duration (seconds) | End of Simulation |
| Repeatability | |
| Inter-Repetition Time (seconds) | Constant (300) |
| Number of Repetitions | Constant (0) |
| Repetition Pattern | Serial |

For each application, two profiles are defined (Light and Heavy). Video-Conference profile was not created for reasons explained before.

Table C.2 – Voice and Streaming profiles.

| **Voice and Streaming Profile** | | |
|---|---|---|
| **Start Time Offset (seconds)** | **Duration(seconds)** | |
| No Offset | Uniform (100,140) | |
| **VoIP Repeatability** | | |
| | **Light** | **Heavy** |
| Inter-Repetition Time (seconds) | Exponential (600) | |
| Number of Repetitions | Unlimited | |
| Repetition Pattern | Serial | |
| **Streaming Repeatability** | | |
| | **Light** | **Heavy** |
| Inter-Repetition Time (seconds) | Exponential (1800) | Exponential (900) |
| Number of Repetitions | Unlimited | |
| Repetition Pattern | Serial | |

Table C.3 – FTP, HTTP, E-Mail and MMS profiles.

| **FTP, HTTP, E-Mail and MMS Profile** | | |
|---|---|---|
| **Start Time Offset (seconds)** | **Duration(seconds)** | |
| No Offset | End of Profile | |
| **FTP Repeatability** | | |
| | **Light** | **Heavy** |
| Inter-Repetition Time (seconds) | Exponential (1200) | Exponential (300) |
| Number of Repetitions | Unlimited | |
| Repetition Pattern | Serial | |
| **HTTP Repeatability** | | |
| | **Light** | **Heavy** |
| Inter-Repetition Time (seconds) | Exponential (900) | Exponential (300) |
| Number of Repetitions | Unlimited | |
| Repetition Pattern | Serial | |
| **E-Mail Repeatability** | | |
| | **Light** | **Heavy** |
| Inter-Repetition Time (seconds) | Exponential (900) | Exponential (300) |
| Number of Repetitions | Unlimited | |
| Repetition Pattern | Serial | |
| **MMS Repeatability** | | |
| | **Light** | **Heavy** |
| Inter-Repetition Time (seconds) | Exponential (1800) | Exponential (300) |
| Number of Repetitions | Unlimited | |
| Repetition Pattern | Serial | |

# Annex D

## Important DL Results

This annex presents important DL results, which are useful for the UL analysis.

Annex D presents important DL results, which are useful for the UL analysis. Looking at Figures D.1, D.2, D.3, D.4, D.5 and D.6 one sees VoIP, Streaming, FTP, HTTP, E-Mail and MMS average values of the total received throughput per application respectively.

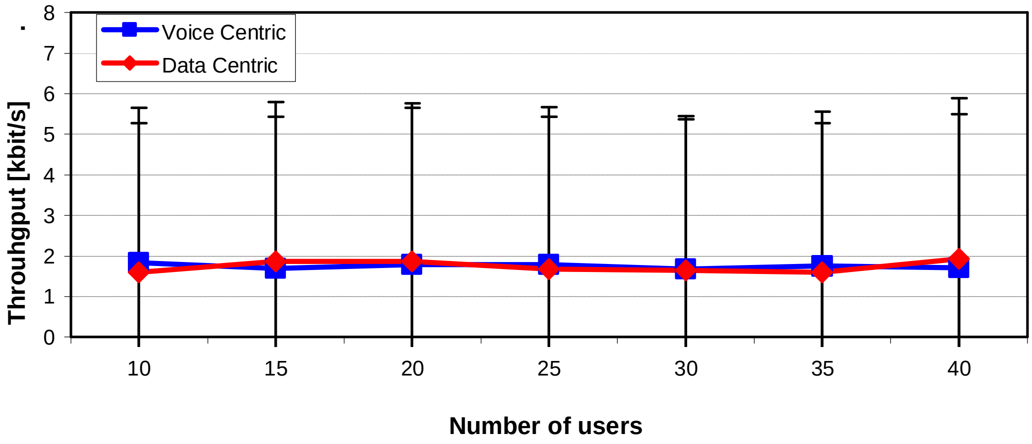



Figure D.1- Average value of the total received throughput by VoIP users.

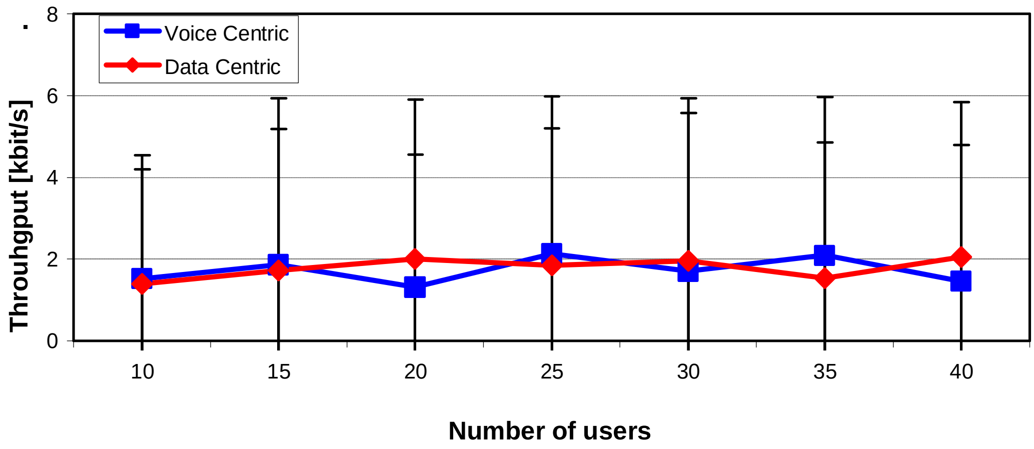



Figure D.2- Average value of the total received throughput by Streaming users.

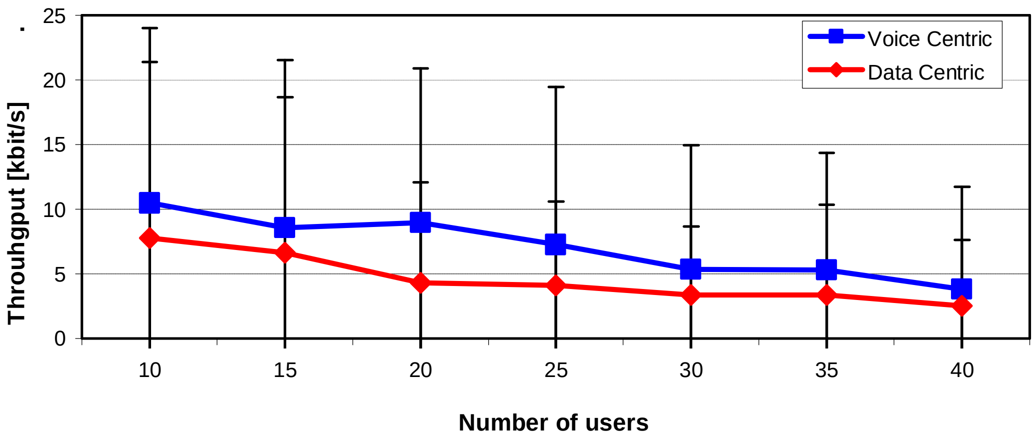



Figure D.3 - Average value of the total received throughput by FTP users.

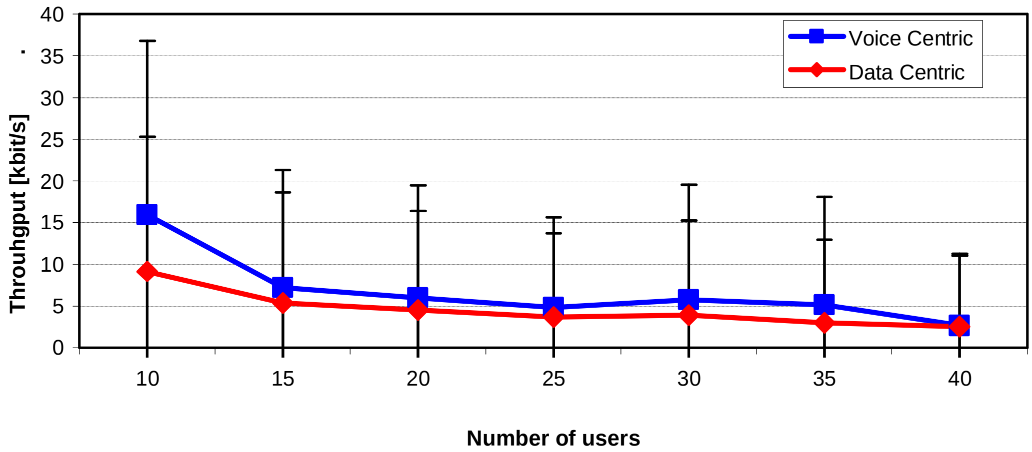



Figure D.4 - Average value of the total received throughput by HTTP users.

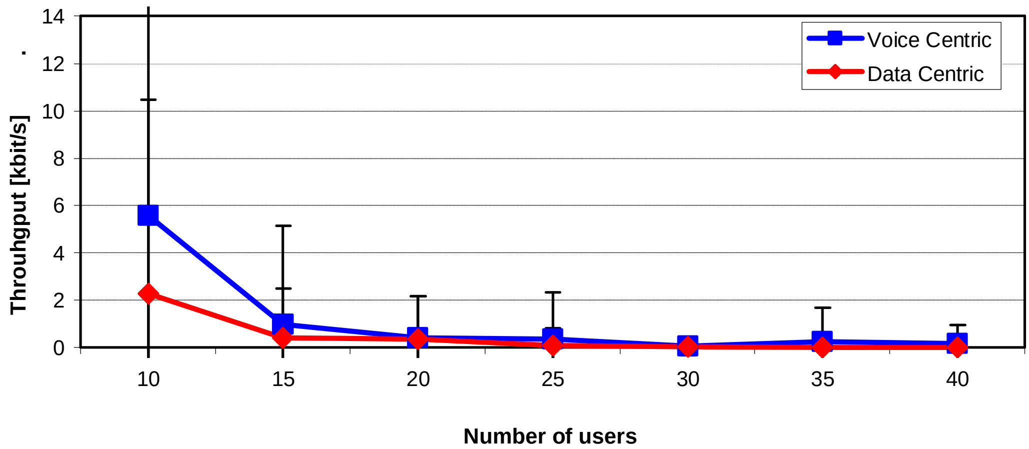



Figure D.5 - Average value of the total received throughput by E-Mail users.

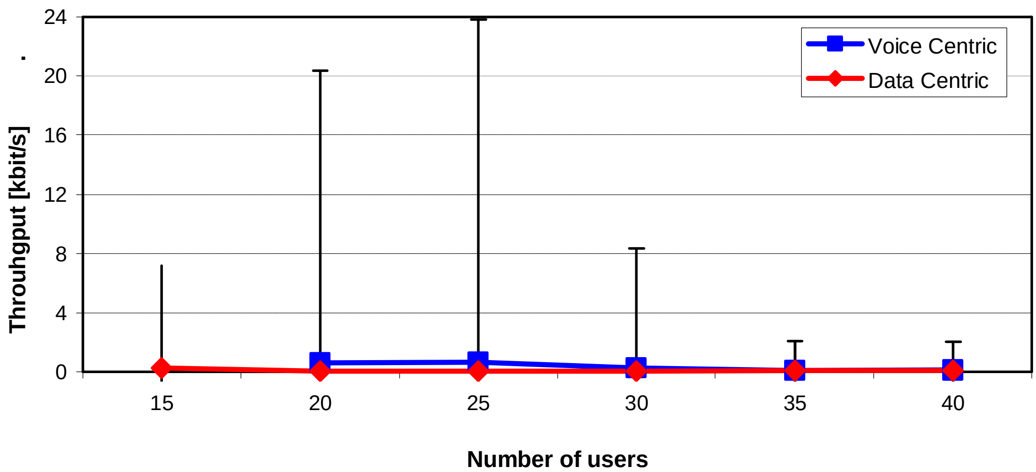



Figure D.6 - Average value of the total received throughput by MMS users.

Looking at Figure D.7 one can see the DL response time (for FTP files), which is the time elapsed between requesting a file and the download is concluded.

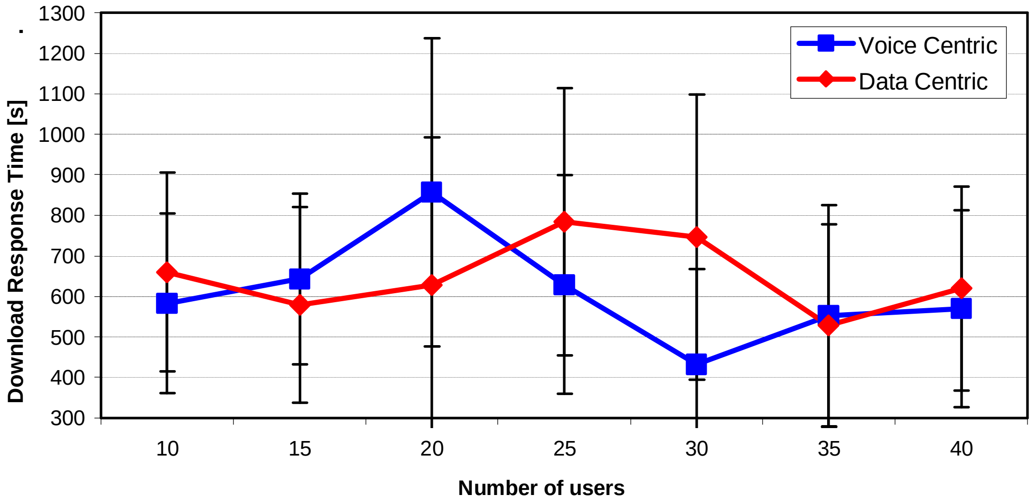



Figure D.7 - DL Response Time.

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