HELSINKI UNIVERSITY OF TECHNOLOGY

Department of Electrical and Communications Engineering

Jose Luis Pradas Adán

Effect of multiple simultaneous HSDPA users on HSDPA end-user performance for non-real time services in one cell system.

Thesis submitted in partial fulfillment of the requirements for the degree of Master of Science in Technology

Supervisor: Professor Sven-Gustav Häggman

Instructor: Jyri Lamminmäki

Espoo, 7 August 2006

HELSINKI UNIVERSITY OF TECHNOLOGY **Abstract of the Master's Thesis**

Author:	Jose Luis Pradas Adán
Name of the Thesis:	Effect of multiple simultaneous HSDPA users on HSDPA end-user performance for non-real time services in one cell system
Date:	07.08.2006 **Number of pages:** 88
Department:	Department of Electrical and Communications Engineering
Professorship:	S-72 Communications Laboratory
Supervisor:	Prof. Sven-Gustav Häggman
Instructor:	M. Sc. Jyri Lamminmäki

HSDPA networks are currently being deployed; however, there is little knowledge about how these networks perform and behave, and which will the Quality of Service and Quality of Experience that users will achieve due to the fact that UEs share the downlink channel. Furthermore, HSDPA planning and dimensioning is being done through the traditional mechanisms to plan and dimension UMTS networks. These mechanisms do not provide, though, accurate results for HSDPA. This thesis will focus on doing progress in these two areas.

A HSDPA simulator was built to find some answers. This simulator used a simplistic model to simulate the radio environment and HSDPA features at Node B. Besides, the simulator dynamically created web browsing traffic according to the traffic patterns specified by the 3GPP. Three main simulations were performed. First, the maximum number of HSDPA users that a HSDPA network can support was obtained for different mean cell throughputs. Results also showed that the relationship between the mean cell throughput and the maximum number of users is linear. Second, the effect of the amount of UEs in a HSDPA network was studied. Results showed how the network and end-user performance changed when the number of UEs differed from the maximum number of UEs. Simulations demonstrated that network and end-user performance decreases rapidly and significantly when the maximum number of UEs was exceeded. Finally, the mean session inter-arrival time was modified to observe how this traffic parameter affected the network and the end-user performance. Furthermore, different sets of number of UEs were used to find out any correlation between the number of UEs and the mean session inter-arrival time. Results showed how the mean session inter-arrival time was much more relevant for the network and end-user performance when the maximum number of UEs had been exceeded.

Results will give a glimpse of how HSDPA can perform in real networks. Besides, this simulator can help operators and providers to plan and dimension HSDPA networks more accurately.

Keywords: HSDPA, UMTS, end-user performance, web browsing.

Contents

List of Abbreviations

3GPP	3rd Generation partnership project
3GPP2	3rd Generation partnership project 2
AICH	Acquisition indicator channel
AM	Acknowledged mode
ARQ	Automatic repeat request
Auc	Authentication centre
BCCH	Broadcast control channel
BCH	Broadcast channel
BMC	Broadcast/multicast control
BPSK	Binary phase shift keying
BSC	Base station controller
BTS	Base transceiver station
CCCH	Common control channel
CDMA	Code division multiple access
CLPC	Closed loop power control
CN	Core network
CPCH	Common physical channel
CPICH	Common pilot channel
CQI	Channel quality indicator
CRNC	Controlling RNC
CS	Circuit switched
CTCH	Common traffic channel
DCCH	Dedicated control channel

DCH	Dedicated control channel
DPCCH	Dedicated physical control channel
DPCH	Dedicated physical channel
DPDCH	Dedicated physical data channel
DS-CDMA	Direct sequence code division multiple access
DSCH	Downlink shared channel
DTCH	Dedicated traffic channel
EIR	Equipment identity register
ETSI	European Telecommunications Standards Institute
FACH	Forward access channel
FBI	Feedback information
FDD	Frequency division duplexing
GERAN	GSM/EDGE Radio Access Network
GGSN	Gateway GPRS support node
GMSC	Gateway MSC
GPRS	General packet radio system
GSM	Global system for mobile communications
HARQ	Hybrid automatic repeat request
HLR	Home location register
HSDPA	High speed downlink packet access
HS-DPCCH	High speed dedicated physical control channel
HS-DSCH	High speed downlink shared channel
HS-PDSCH	High speed physical downlink shared channel
HS-SCCH	High speed shared control channel
HSUPA	High speed uplink packet access
IMS	IP multimedia subsystem
IMT-2000	International mobile telephony 2000
ITU	International Telecommunications Union
Kbps/kbps	Kilobits per second
KPI	Key performance indicator
L1	Physical layer
L2	Data link layer

L3	Network layer
LA	Link adaptation
MAC	Medium access network
MBMS	Multimedia broadcast multicast service
Mbps	Megabits per second
ME	Mobile equipment
MGW	Media gateway
MIMO	Multiple-input multiple-output
MS	Mobile station
MSC	Mobile switching centre
NAS	Non-access stratum
OLPC	Open loop power control
OVSF	Orthogonal variable spreading factor
PCCH	Paging control channel
P-CCPCH	Primary common control physical channel
PCH	Paging channel
P-CPICH	Primary common pilot channel
PDC	Personal digital cellular
PDCP	Packet data convergence protocol
PDU	Packet data unit
PICH	Paging indicator channel
PLMN	Public land mobile network
PRACH	Physical random access channel
PS	Packet switched
QoS/QoE	Quality of Service/Quality of experience
QPSK	Quadrature phase shift keying
RACH	Random access channel
RAN	Radio access network
RLC	Radio link control
RNC	Radio network controller
RNS	Radio network system
RRC	Radio resource control

RRM	Radio resource management
SAP	Service access point
S-CCPCH	Secondary common control physical channel
SCH	Synchronization channel
S-CPICH	Secondary common pilot channel
SDU	Service data unit
SF	Spreading factor
SGSN	Serving GPRS support node
SHCCH	Shared channel control channel
SIR	Signal to interference ratio
SRNC	Serving RNC
SRNS	Serving RNS
TDD	Time division duplex
TF	Transport format
TFC	Transport format combination
TFCI	Transport format combination indicator
TFCS	Transport format combination set
TFS	Transport format set
TM	Transparent mode
TPC	Transmission power control
TTI	Time transmission interval
UE	User equipment
UM	Unacknowledged mode
UMTS	Universal mobile telecommunications system
USIM	UMTS subscriber identity module
UTRA	Universal terrestrial radio access/ UMTS terrestrial radio access
VLR	Visitor location register
WCDMA	Wideband code division multiple access
WLAN	Wireless local area network
WTDMA	Wideband time division multiple access

List of Symbols

Cu	Cu interface
Uu	UMTS radio interface
Iub	Iub interface
Iu CS	Iu interface for circuit switched
Iu PS	Iu interface for packet switched
Iur	Iur interface
T_{prop}	Propagation time
Pmax	Maximum transmission power
$P_{tx\ target}$	Target cell transmission power
n	Session j and UE k.
m_k	UE k.
p	Maximum number of UEs
$page\_download\_time_{i,j,k}$	Download time of page i within session j of UE k
$page\_throughput_{i,j,k}$	Throughput of page i within session j of UE k

List of Figures

List of Tables

Acknowledgements

After all this time in Finland I really have to thank many people who has helped me and supported me not to give up. I have to say that it is a long list so I will try to bring up here the people without whose help, this thesis would not have ever been possible.

First of all, I want to thank my supervisor, Professor Sven-Gustav Häggman, for his continuous support and advices all along the way.

Secondly, I want to thank Jyri Lamminmäki, my instructor, for his guidance, help, support, thesis revision, and specially, for his patience. I also want to mention Florian Reymond who always helped me when I needed it.

This thesis would not have been what it has been without Nokia Networks and without the SBU. Special thanks to my manager, Johanna Kähkönen, for trusting me and giving me the chance of doing my Master Thesis within Nokia and in her team.

I cannot forget all the guys around here: Janne Sipilä, Anders Arte, Sakari Sistonen, Mika Kleemola, Dimitrios Dimaras, Pekka Matilainen, Pauli Aikio, Carlos Crespo, Davide Chiavelli, Svetlana Chemiakina, and all others who, in one way or another, have helped me or cheered me up in some point. Words are not enough to express all my gratitude.

But above all, I want to thank my parents. They have always supported me and helped me. They have tried to understand and respect my decisions. I know how hard it has been for them because decisions were never easy to take and many did not make me happy, but I had to take them. After all, as you can see now, all I have gone through had a meaning. Thanks family. Thanks, parents, for all.

Monday, 07 August, 2006

Jose Luis Pradas Adán

Chapter 1

Introduction

1.1 Background

Mobile networks have evolved significantly during the last 3 decades. The mobile adventure started with the first generation (1G) of mobile networks. These networks were deployed during the 80s. 1G networks were characterized for being analogue, and their technology was optimized for voice. The second generation (2G) of mobile networks came next. The second generation overcame many of the problems that the previous generation had. 2G networks were not analogue any more, digital technology was used instead. Three different systems were mainly standardized in the second generation: Global System for Mobile Communications (GSM), cdmaOne (IS-95), and Personal Digital Cellular (PDC). These technologies were standardized in Europe, US, and Japan respectively. The European standard, though, has been favored by many other countries and it has been deployed widely all around the world. According to [GSM_06], the number of GSM users in the world represents 77 % of the total mobile users.

GSM was first based on circuit switched technology, clearly as a legacy of the previous network technologies. Both voice and data services were offered using circuit switched technology; however, speeds for data services were fairly low. GSM was thought to provide voice services, not data services. GSM was eventually modified to work also with packet switched technology so data services could be offered at higher speeds. These networks were called 2.5G mobile networks. 2.5G networks were improved through High Speed Circuit Switched Data (HSCSD) service and General Packet Radio Service (GPRS). HSCSD was based on circuit switched technology while GPRS was based on packet switched technology.

Enhanced Data Rates for GSM Evolution (EDGE) followed the previous developments. Data speeds were improved significantly. EDGE is able, in theory, to achieve up to 384 Kbps which is far from the 9.6 Kbps offered by the first 2G networks.

The mobile networks evolution did not stop there. EDGE gave way to what is called third generation of mobile networks, 3G. 3G networks have been called differently in different places. Europe named it as Universal Mobile Telecommunication System (UMTS), while in Japan and US, 3G networks were named as International Mobile Telephony 2000 (IMT-2000). UMTS is the name given by the European Telecommunication Institute (ETSI); however, the International Telecommunications Union (ITU) preferred IMT-2000.

3G networks are designed to provide voice and data service in an efficient way. Hence, bit rates have been further improved compared to the other network generations. 3G is able to provide, in theory, up to 2 Mbps as specified in 3GPP Release '99, and up to 10 Mbps as specified in 3GPP Release 5. Not only 3G provides data and voice services, 3G is built over a new technology which is focused in providing better end-user experience through new services and applications, widely known as multimedia services.

Unlike 1G and 2G networks, which provided none or limited compatibility with other networks of the same generation, 3G networks are supposed to provide compatibility and interoperability worldwide with any other 3G network.

Despite the promises of 3G to provide multimedia services at high speed rates, it has turned out that 3G networks could not offer what was expected. Few enhancements have been done along the way resulting in a new mobile generation, 3.5G. These improvements are the result of both the necessity of coping with the new emerging competing technologies, such as CDMA-2000 or WiMAX, and the necessity of providing high bit rates. 3GPP standardized a new technology called High Speed Downlink Packet Access (HSDPA). HSDPA is able to increase the bit rate up to 10 Mbps. Chapter 3 explains in detail HSDPA technology and features.

Although HSDPA achieves high bit rates in the downlink, some applications, such as real time applications, need similar bandwidth in both downlink and uplink. For this reason, the 3GPP is standardizing the natural technology for the uplink, High Speed Uplink Packet Access (HSUPA) [3GPP_308], [3GPP_896]. [Sha_05] presents a brief overview of the technological aspects of HSUPA.

Fourth Generation (4G) of mobile networks is now under development. 4G goes along with two statements: communications convergence, and “All-IP”. 4G networks are supposed to be fully interoperable and compatible to any mobile technology in a seamless way for the users. 4G networks will provide extremely high bit rates, and a vast number of services among other benefits [Eva_00], [San_03].

Figure 1 shows the evolution of the different generations in mobile networks as explained before.

The evolution of the different mobile generations has gone towards cheaper and more flexible technologies, and at the same time, bit rates have been increased generation after generation.

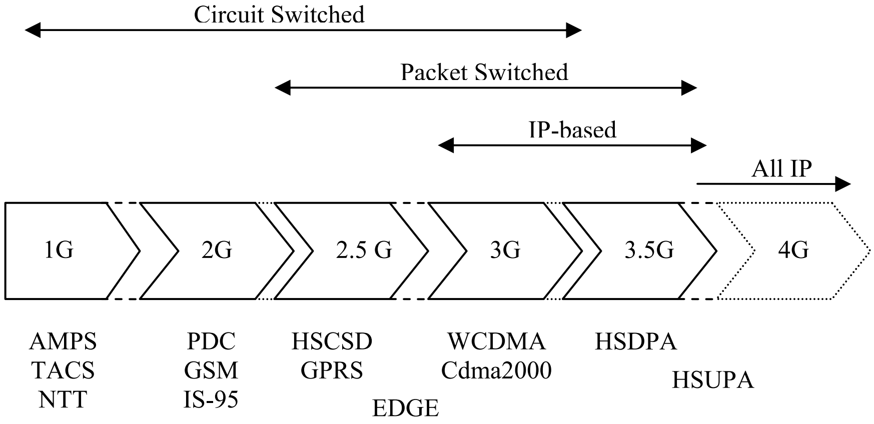

Figure 1 – Cellular Mobile Networks Evolution

But, why mobile networks are obsessed for providing higher bit rates? One possible answer to this question is Internet. The number of Internet users has increased exponentially during the last years [Ran_98], and this trend is likely to continue for the next years [Law_00]. Internet has clearly become indispensable in many people's life [Don_04].

Due to the high penetration of mobile phones, they are likely to be a very useful tool to access the Internet and the services it provides. To make this feasible, bit rates need to be increased to provide to the end users a fair Quality of Service (QoS), and at the same time, make cellular mobile networks competitive against other mobile technologies.

1.2 Research Problem

HSDPA is still a young technology and it is not fully implemented in real networks. Hence, it is difficult to assess how this technology will behave and which its real performance is. Nonetheless, operators as well as manufacturers are investing in this technology. This is clearly a risk which manufacturers and operators need to mitigate as much as possible in order to succeed.

There are many research documents about HSDPA. Many are presented in Chapter 3 and Chapter 4. Most of them present how HSDPA performs for a certain specific algorithm. However, they fail to take into consideration how HSDPA would perform in real situations, when there are certain types of traffic and a certain amount of users. Most of them do not study the network and end-user performance. This thesis will deal with these issues. Section 4.2 describes with more detail what has been already done and what hasn't been done, and it presents a list of relevant documents for this thesis.

Before a network is actually deployed, a planning and dimensioning process needs to be performed. Currently, there are no HSDPA tools for planning and dimensioning; however, these tools are needed. Planning and dimensioning will need to roughly assess how many simultaneous users a HSDPA network can support. The answer will depend, among other things, on the type of traffic and traffic pattern which the network is intended to carry. HSDPA is intended to be used for non-real time applications, such as browsing. Browsing traffic has certain traffic characteristics and, hence, it is interesting to know how the network behaves as well as the end-user performance when several users are simultaneously in the same cell and they interact with different services, in this case, browsing services. A more extensive description of the problem is presented in Chapter 4.

1.3 Objective of the Thesis

This thesis will explain HSDPA and identify the challenges it has to face. It will also provide the answers that the previous section arose. A simulator will be built in order to find some answers. This simulator aims to discover the maximum number of HSDPA users in one cell in different situations, such as for example when all users within that cell are using browsing services. Different parameters will be modified to see the effect of them in the end-user performance. These parameters are the user/session inter-arrival time and the mean cell throughput. Different performance parameters will be collected in order to learn how the network and the users are affected by the types of traffic and number of simultaneous users. These key values will provide the current performance indicators for the simulated environment. Current performance indicators will be compared to some recommended (standardized) Key Performance Indicators (KPI) which, in turn, can be mapped into certain QoS and Quality of Experience (QoE) values. KPIs, QoS and QoE values will limit the maximum number of users that the network is able to give service.

Results will be extremely helpful to glimpse the potential of this technology for browsing and, possibly, for streaming applications, as well as, to provide to operators, service providers, and manufactures a tool to show them whether HSDPA would be a good investment for their portfolio or not. Next to that, the simulator and the results will provide an interesting tool to roughly assess the number of users the network can support offering a good quality of service, i.e. a tool for network planning.

1.4 Structure of the Thesis

This thesis is divided into six chapters. Chapter 2 is an introduction about UMTS system as described in 3GPP Release 99/Release 5. It will describe the architecture, radio access network, radio access technology, radio protocol architecture, and the radio resource management. Chapter 3 introduces HSDPA technology and its main features. Though, HSDPA has been slightly modified in Release 6, this thesis will present HSDPA according to Release 5. Chapter 4 tackles the research problem and it presents previous studies about the problem at hand. Chapter 4 will explain briefly the simulator and the assumptions used in it. Chapter 5 will present the simulation results. Finally, Chapter 6 will summarize what was done and it will present conclusions about the results as well as suggestions for further research.

Chapter 2

Universal Mobile Telecommunications System – UMTS

2.1 Overview

Several drivers have shoved forward the evolution of the 2nd generation towards the 3rd generation. First of all, there was a clear need for higher bit rates and more capacity. In addition, the depletion of the assigned GSM frequencies was obvious. It was in 1992 when ITU identified and allocated frequencies for the future 3G networks. Later on, 3GPP was established. 3GPP aim is to standardize UMTS specifications. 3GPP organization is made up of several standardization groups all around the world, as well as industry players. 3GPP is creating UMTS specifications for Europe and Japan while, a similar organization, 3GPP2, standardizes IMT-2000 in the US.

The first standardized UMTS document was frozen and released in 1999. This standard was strongly based on the previous technology, GSM. Nonetheless, one of the major changes was the Radio Access Network (RAN). The UMTS Terrestrial Radio Access Network (UTRAN) was chosen to be wideband code division multiple access, WCDMA. Figure 2 shows the UMTS network architecture of the Release 99.

Although Figure 2 shows GSM/EDGE radio access network (GERAN), GERAN was not part of the 3GPP Release 99 specifications. It was included and defined in later specifications, though.

It followed Release 4 and Release 5, frozen in 2001 and 2002 respectively. These releases undertook major changes compared to the first release. Release 4 introduced changes in the Core Network (CN) and Release 5 started to specify the IP Multimedia Subsystem (IMS), and high speed data packet access (HSDPA) among others. Besides, Release 5 also started to define the "All-IP" environment. 3GPP Release 4 and Release 5 also introduced EDGE as an alternative way to build the UMTS networks. GERAN is covered in these releases but not in 3GPP Release 99. Figure 3 shows the network architecture of Release 5.

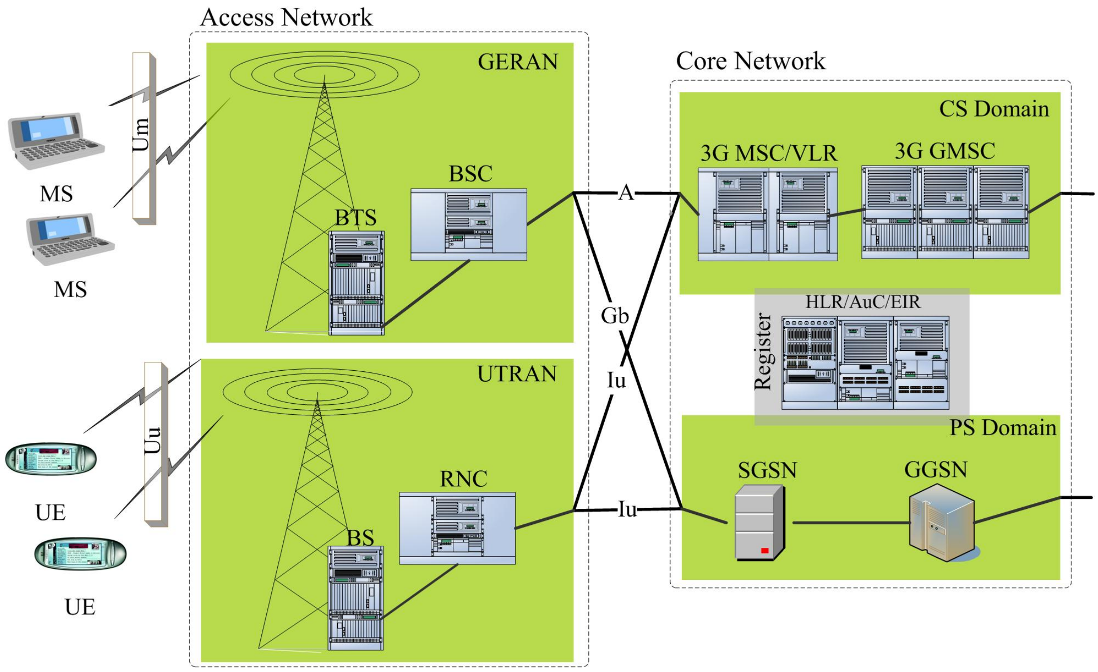

Figure 2 – 3GPP Release 99, network architecture

Release 6 was completely frozen late during 2005. Release 6 continues specifying the IMS as well as other technologies, such as HSUPA or the Multimedia Broadcast Multicast Service (MBMS). It also carries on the process of converging data networks (inter-working with wireless local area networks (WLAN)) and moving towards “All-IP” networks [3GPP_902].

Currently, 3GPP is working on the next release, Release 7. This release will continue enhancing UMTS. Multiple-Input Multiple-Output (MIMO) systems are likely to be tacked in this release. MIMO antenna systems augur better spectral efficiency. Better spectral efficiency in the air interface will provide higher bit rates and better signal quality. As a result, better QoS and larger pool of services will be offered.

2.2 Architecture

As we can see from Figure 2 and Figure 3, the network is divided into CN, UTRAN, and User Equipment (UE). A more detailed description of these elements can be seen in Figure 4.

UE is made of two elements: Mobile Equipment (ME) which is the mobile itself; UMTS subscriber identity module (USIM), which stores subscriber information, 3G services, and home environment or information related to service providers [ETSI_111]. The interface between the ME and the USIM is the Cu interface. UE interfaces with UTRAN through the Uu interface. UE is called Mobile Station (MS) in GERAN terminology.

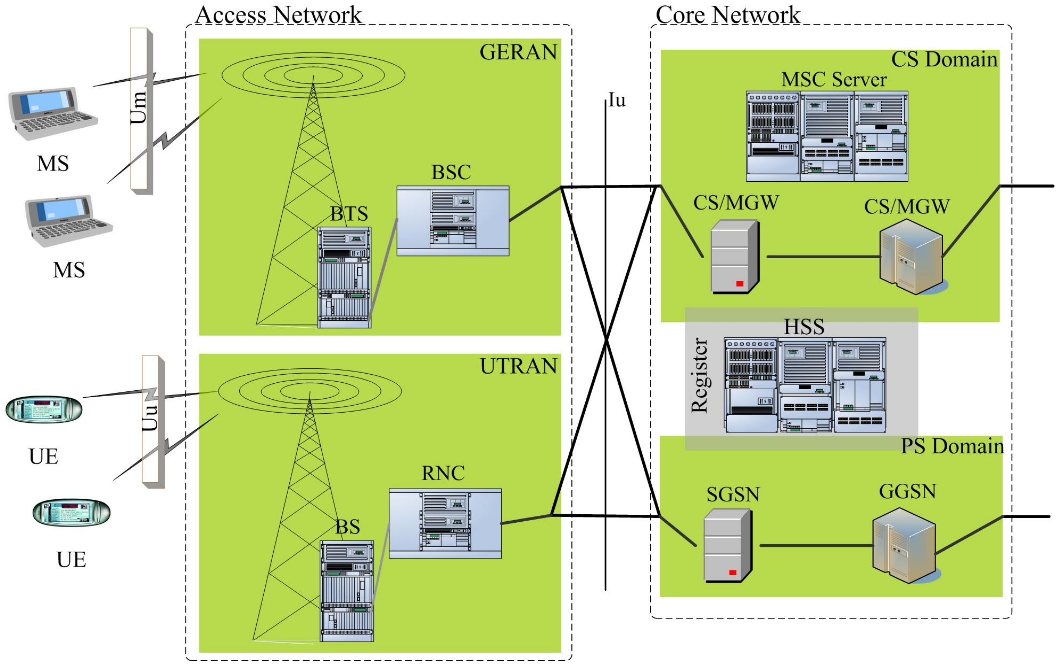

Figure 3 – 3GPP Release 5, network architecture

UTRAN architecture is constituted of one or more Radio Network Systems (RNS). Two elements can be found within a RNS: one (or more) Node B, and one Radio Network Controller (RNC). Node B's are connected to RNC through the Iub interface and RNCs are connected to each others through the Iur interface.

CN is divided into Circuit Switched (CS) domain and Packet Switched (PS) domain. Although the PS domain elements have not changed much along the different specifications, CS domain elements have undergone different modifications. In 3GPP Release 99, the CS domain consisted of a Mobile Switching Centre/Visitor Location Register (MSC/VLR) followed by a Gateway MSC (GMSC). Already, in 3GPP Release 4, CS domain elements were modified. MSC Server and Circuit Switched – Media Gateway Function (CS-MGW) replaced the previous elements.

In the PS domain there are two elements: Serving GPRS Support Node (SGSN) and Gateway GPRS Support Node (GGSN). Although Figure 2 shows different interfaces between the access network and the CN, Release 5 converged the different interfaces to a unique one, Iu interface, as depicted in Figure 3.

Further details about characteristics and functionalities of these elements can be found in [3GPP_002].

2.3 Radio Access Network

As mentioned in the previous section, UMTS RAN (i.e. UTRAN) architecture consists of one or several RNSs. A RNS is subdivided into a RNC and one or several Node B's.

UTRAN interfaces with the UE, with the CN, and with GERAN through the Uu, Iu, and Iur-g interfaces respectively. Uu interface is a well known interface, and it is one of the biggest differences between GSM/EDGE and UMTS. Uu interface is the WCDMA radio interface.

Two interfaces have been defined to connect the different elements within UTRAN. RNCs are connected to each other through the Iur interface while Node B's are connected to RNC through the Iub interface. Figure 4 shows UTRAN architecture, its elements and interfaces to the other part of the network.

UTRAN has the following responsibilities and functions [3GPP_401]:

- Transfer of user data
- Access control, such as for example admission control or congestion control.
- Radio channel ciphering and deciphering (also performed at the UE)
- Mobility, such as for example handover, SRNS Relocation, or paging.
- Radio resource management and control
- Broadcast and multicast services
- Tracing (UE events and activities)
- Volume reporting (for accounting purposes)
- RAN information management

2.3.1 Node B

Node B interfaces with the UE and with RNC through the Uu and Iub interfaces respectively. Node B performs physical layer (L1) functions, such as channel coding and interleaving, modulation, spreading, radio transmission, and reception [3GPP_201]. The radio access scheme chosen in UMTS for the Uu interface is Direct Sequence Code Division Multiple Access (DS-CDMA), and since the information is spread over a wide bandwidth range (5 MHz), the radio access scenario is known as WCDMA.

Node B is also involved in some Radio Resource Management (RRM) functions, such as (inner loop) power control, and it also participates in the O&M functions [3GPP_401].

2.3.2 Radio Network Controller

RNC is responsible of controlling the radio resources, i.e. frequencies, scrambling codes, spreading factors (SF), and power control. RNCs interface with three elements: CN, Node B, and optionally with another RNC. The interfaces used for these connections are Iu, Iub, and Iur respectively. Furthermore RNCs can also interface GERAN through the Ir-g interface in order to provide inter-operation and mobility between both technologies.

RNC can take different roles: Controlling RNC (CRNC), Serving RNC (SRNC), and Drift RNC (DRNC). CRNC controls the logical resources (congestion control, admission control, code allocation control, etc.) of a set of Node B's. On the other hand, SRNC controls the radio connection between the UTRAN and an UE. Besides, SRNC maintains the Iu bearer for that UE. SNRC control mobility and RRM functions, such as (outer loop) power control or radio access bearer allocation.

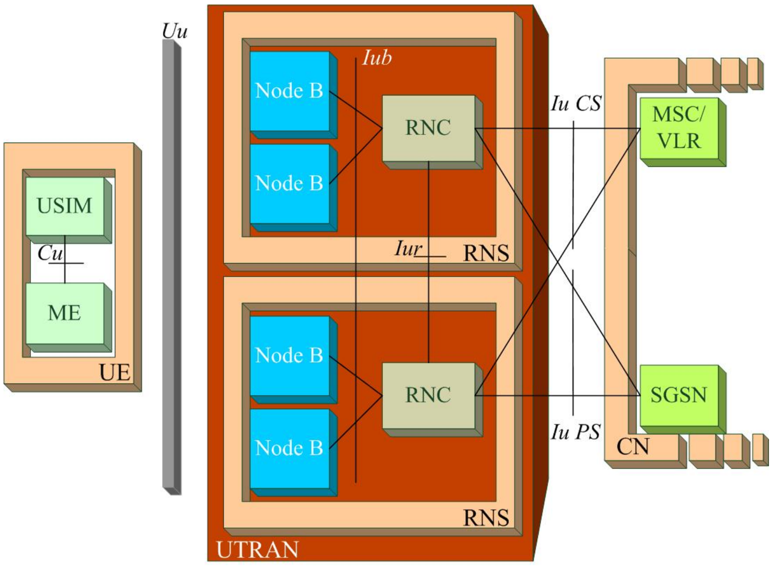

Figure 4 – UTRAN architecture

DRNC role is also linked to a connection between UTRAN and an UE. DRNC operates as a router, routing data from Node B or Node B's (which belong/s to the DRNC) to the SRNC through the Iur interface. This interface is also used for other purposes for common and share channels.

2.4 Radio Access Technology

The UMTS Terrestrial Radio Access (UTRA) technology chosen by ETSI is based on CDMA technology. For paired frequencies, ETSI selected DS-CDMA which is widely known as UTRA-FDD or WCDMA. On the other hand, for unpaired frequencies, WTDMA/CDMA (UTRA-TDD) was chosen.

Table 1 shows the current allocated frequencies for UMTS (FDD and TDD) in Europe. Many other countries allocated their IMT-2000 frequencies as Europe did, however, with slight differences. US allocated frequencies are, though, different than the European context.

In the World Radio-communications Conference 2000 (WRC-2000) new frequencies were identified for IMT-2000. The novelty of these frequencies is that they are not yet allocated in any country and, as a result, these new frequencies would be unique for IMT-2000 all over the world. Last row in Table 1 shows the future bands for IMT-2000. Detailed frequency allocation is available in [3GPP_101].

	Uplink (MHz)	Downlink (MHz)
UMTS-TDD	1900-1920	2010-2025
UMTS-FDD	1920-1980	2110-2170
IMT-2000	*2500-2570*	*2620-2690*

Table 1 – UMTS frequency allocation

2.4.1 WCDMA

WCDMA is also known as direct sequence spread spectrum. WCDMA principle is relatively simple. The signal, before been transmitted to the air interface, is combined with a spreading signal (spreading code). The elements of the data transmitted are called bits while the elements of the spreading signal are called chips. The chip rate of the spreading signal is 3.84 Mchip/s. The result is another signal with bigger bandwidth than the original signal. The data rate of the resulting signal will be the data rate of the spreading signal. The effective bandwidth in UMTS is 3.84 MHz; however, there are security guard bands which increase the bandwidth up to 5 MHz. The spectrum of spread signal will be k times bigger than the original one. K is called spreading factor (SF) or processing gain.

Due to the processing gain, signal-to-noise ratio can be fairly low and still the signal is able to be detected by the receptor. The receptor will need to multiply the received signal by the same spreading signal used in the other end. The result of the previous operation will be integrated obtaining the desired signal.

2.4.1.1 Spreading Codes

UMTS uses different codes: channelization codes, scrambling code, and spreading codes. Spreading codes are the result of the combination of channelization and scrambling codes.

2.4.1.1.1 Channelization Codes

Channelization codes as well as scrambling codes have different missions in the uplink and in the downlink. Channelization codes are used to differentiate data and control channels of a certain UE in the uplink. On the other hand, they are used to differentiate users within a cell or sector in the downlink. Next to that, channelization codes are responsible of spreading the signal bandwidth. Scrambling codes will not affect the signal bandwidth.

Channelization codes are based on Orthogonal Variable Spreading Factor (OVSF). OVSF codes maintain orthogonality between the control and data channels. These codes can be easily built using a channelization code tree (Figure 5). These codes cannot be selected randomly. There are certain limitations whether a new code is to be used. Two rules should apply when selecting a new code: no code which is in the path from the root tree to the code/s already in use can be selected, and no code which belongs to a child branch of the code/s already in use can be selected.

2.4.1.1.2 Scrambling Codes

Unlike channelization codes, scrambling codes are used for differentiating cells or sectors in the downlink while, they are used for differentiating users in the uplink.

There are in theory 262143 scrambling codes (2^{18}-1); however, only 8192 codes are used. These codes are grouped into 512 sets. Each set has 16 codes. One of these codes is a primary scrambling code, and the rest are secondary scrambling codes. Sets are finally disposed into groups. Each group contains 8 sets and, therefore, there are 64 groups.

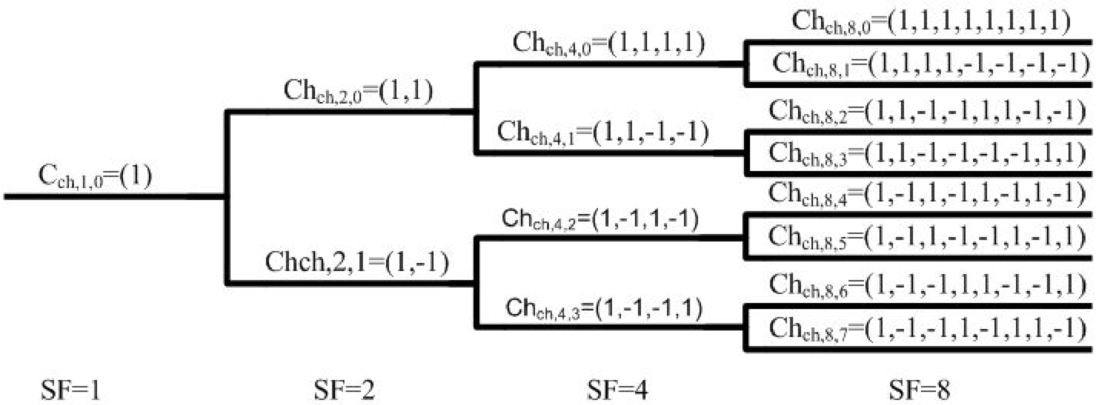

Figure 5 – Channelization code tree

2.5 Radio Protocol Architecture

The UTRAN radio interface (Uu) protocol architecture is shown in Figure 6, as described in 3GPP Release 6 [3GPP_301].

The radio interface architecture is built in three layers:

- physical layer (L1);
- data link layer (L2);
- network layer (L3).

The physical layer provides services, transport channels, to the data link layer. According to [3GPP_211], “a transport channel is defined by how and what characteristics data is transferred over the air interface”. Transport channels are grouped into dedicated channels and common channels. Section 2.5.1.1 describes transport channels as well as their types and characteristics. Transport channels are mapped by the physical layer to physical channels. Physical channels are characterized by a carrier frequency, spreading code (i.e., channelization code and scrambling code), duration, and phase. Physical channels answer the question what is sent. Further details about the physical channels are provided in section 2.5.1.2.

Data link layer is over the physical layer. L2 is divided into four sub-layers: Medium Access Control (MAC), Radio Link Control (RLC), Packet Data Convergence Protocol (PDCP), and Broadcast/Multicast Control (BMC). L2 provides radio bearer service to the upper layer, L3. The service provided to the L3 control plane is designated as signaling radio bearer.

The last radio interface layer in UTRAN is the network layer. The network layer is over L2. L3 is divided into two planes: user plane (U-plane) and control plane (C-plane). Besides, L3 is split into two sub-layers: Radio Resource Control (RRC) and a second sub-layer which provides "duplication avoidance" functionality. This sub-layer is over the RRC layer and it covers the U-plane and the C-plane. Other higher signaling layers, such as call control, mobility management, session management, or GPRS mobility management are handled in the non-access stratum (NAS).

The different layers and sub-layers offer their services to upper layers through Service Access Points (SAP) [3GPP_110].

2.5.1 Physical Layer

The physical layer has undergone several changes along the different releases. L1 functions have changed and the number of services offered has increased as well as the number of physical channels. Latest releases have moved some functions, traditionally done in RNC, to Node-B where L1 is implemented.

The physical layer is implemented in Node B. L1 is the lowest layer in the radio interface protocol architecture stack. This layer is controlled by the RRC, as depicted in Figure 6. L1 provides services to L2. The services which L1 offers are transport channels. MAC sub-layer interfaces L1 and it accesses L1 services through SAPs.

Some functions of the physical layer are [3GPP_201]:

- Macrodiversity distribution/combining and soft handover.
- Transport channel error detection and error report to higher levels.
- Multiplexing of transport channels and demultiplexing of coded composite transport channels.
- Rate matching of coded transport channels to physical channels.
- Power weighting and combining of physical channels.
- Radio characteristics measurements and reporting to higher levels.
- Inner loop power control.

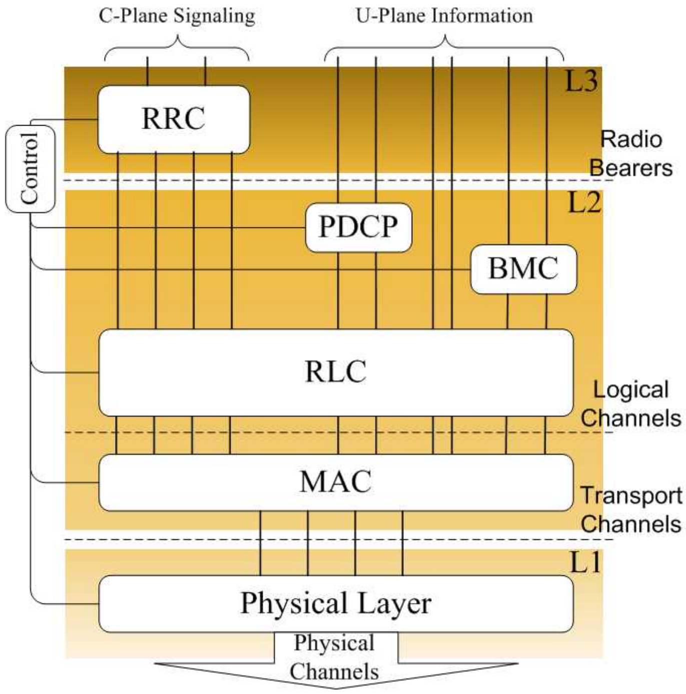

Figure 6 – UTRAN radio interface protocol architecture

L1 interfaces the air through the physical channels using WCDMA (already discussed in Section 2.4.1). Traditionally, the physical channels frame has had 10 ms length and it has been divided into 15 slots. Few transport channels, though, use longer periods as we will see in the following sections.

Before the data reaches the air interface, the physical layer attaches a CRC to the data sent by the MAC. Then, it performs channel coding and interleaving. Next to that, it modulates the data and, finally, it spreads the information over a certain bandwidth.

The modulation scenario differs for the uplink and the downlink. The uplink is modulated using BPSK. On the other hand, the downlink is modulated using QPSK. In the same way, spreading factors vary for uplink channels and downlink channels. For uplink channels, spreading factors vary from 2 to 256 while for downlink channels, spreading factors vary from 4 to 512. Obviously, the channel bit rate will vary depending on the spreading factor and the modulation. Thus, for downlink channels, bit rates will vary from 7.5 K-symbols/s to 960 K-symbols/s. Uplink channels bit rate will vary from 15 K-symbols/s to 1920 K-symbols/s.

Transport channels are mapped into physical channels as described in Figure 7. Next section describes briefly the transport channels. After that, physical channels are described.

2.5.1.1 Transport Channels

There are two types of transport channels: common transport channels and dedicated transport channels. The following two sections explain briefly these channels.

2.5.1.1.1 Common Transport Channels

- Broadcast Channel – BCH is a downlink channel. BCH is broadcasted over a cell and it transports system/cell information, such as frequency information, downlink scrambling codes, or transmission diversity methods. BCH has a unique transport format.
- Random Access Channel – RACH is an uplink channel. This channel is used to send small amounts of information, such as in initial access to the network, non-real time control, or traffic data. Its main function is requesting access to the network.
- Forward Access Channel – FACH is a downlink transport channel. It transports a small amount of data and it is used to broadcast or multicast that data.
- Paging Channel – PCH is a downlink transport channel. It is broadcasted to the whole cell. It transports control information which allows UE to perform the sleep mode procedure.

2.5.1.1.2 Dedicated Transport Channels

- Dedicated Channel – DCH is a transport channel allocated to one UE. It can be used either for uplink or downlink. This channel is controlled through the inner power control. DCH bit rate is variable depending on the channel conditions and the allocated bearer. Bit rate variations can be performed each 10 ms.

2.5.1.2 Physical Channels

Physical channels carry the data sent by the user or by the network through the air interface. As mentioned earlier in section 2.5.1, physical channels are characterized by a carrier frequency, spreading codes, duration, and phase.

Physical channels can be divided into uplink and downlink physical channels and each group is further sub-divided into dedicated and common physical channels.

2.5.1.2.1 Dedicated Uplink Channels

Dedicated data and control uplink channels are multiplexed in phase and quadrature respectively, as described in [3GPP_213]. The dedicated uplink channels are: DPCCH, and DPDCH. Though, Release 6 has introduced new dedicated channels, they are out of the scope of this thesis. The reader is encouraged to read [3GPP_211] to learn about HSUPA physical channels. HSDPA channels are explained in section 3.5.2.

2.5.1.2.1.1 DPCCH/DPDCH

The dedicated physical control channel and the dedicated physical data channel carry L1 control information and data information respectively.

DPCCH uses a fix spreading factor. Its value is 256. Then, DPCCH will contain 10 bits of control information, such as pilot, power control, TFCI, or FBI information. On the other hand, DPDCH channel transports the actual user’s data. DPDCH channel bit rate vary (and so the number of bits in a slot) according to the SF value, which can vary from 4 to 256. This SF range offers a channel bit rate from 960 kbps to 15 kbps.

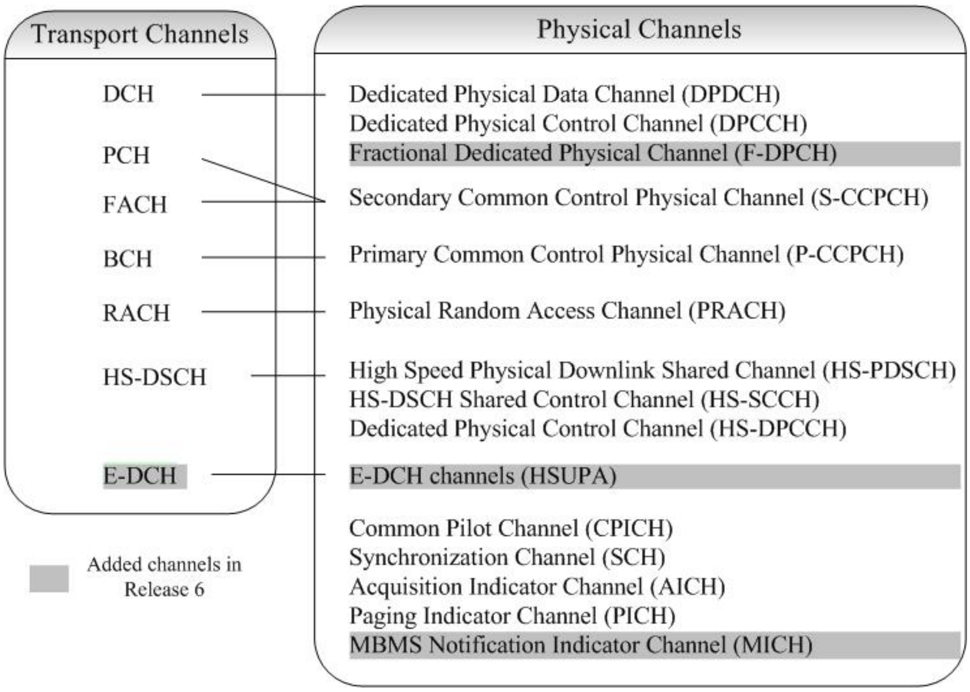

Figure 7 – Mapping of transport channel to physical channels, Release 5

2.5.1.2.2 Common Uplink Channels

2.5.1.2.2.1 PRACH

PRACH channel is made of two parts: a preamble, and a ‘message’. The preamble has to be sent by the UE and it has to be acknowledged by Node B previous any ‘message’ is transmitted. Acknowledgements are sent into the AICH channel.

The ‘message’ consists of a 10 ms radio frame or of the concatenation of two 10 ms radio frames. PRACH radio frames are divided into 15 slots. Slots contain data and control information modulated in phase and quadrature respectively. Data can have different SFs which vary from 32 to 256. These SFs will provide channel bit rates from 120 kbps to 15 kbps. On the other hand, control information uses solely a 256 SF value which provides a 15 kbps channel bit rate.

2.5.1.2.3 Dedicated Downlink Channels

3GPP Release 99 specified only one dedicated downlink physical channel; however, Release 6 increased substantially the number of dedicated downlink channels. Downlink DPCH was specified in Release 99 and it has been inherited by the subsequent releases. Only downlink DPCH is under the scope of this section and, therefore, it will be the only downlink physical channel discussed here. More information about the other three dedicated downlink physical channels can be found in [3GPP_211].

2.5.1.2.3.1 Downlink DPCH

Downlink DPCH is used to carry DCH information as well as L1 control information. L2 and L1 information is multiplex in time. This channel can be seen as the time multiplex of DPDCH and DPCCH.

The downlink DPCH SF value varies from 4 to 512. According to these SF values, downlink DPCH bit rate varies from 1920 kbps to 15 kbps. Downlink channels are modulated using QPSK; hence, downlink DPCH symbol rate varies from 960 Ksymbol/s to 7.5 Ksymbol/s respectively.

2.5.1.2.4 Common Downlink Channels

Common downlink physical channels have been modified along the different releases. Release 5 removed several common downlink channels available in Release 4, such as Physical Downlink Shared Channel (PDSCH), CPCH Access Preamble Acquisition Indicator Channel (AP-AICH), CPCH Collision Detection/Channel Assignment Indicator Channel (CD/CA-ICH), CPCH Status Indicator Channel (CSICH), and Physical Common Packet Channel (PCPCH). On the other hand, Release 5 brought HSDPA in and new common downlink physical channels were introduced. In the same way, Release 6 has introduced new features and, hence, more new physical channels are now supported. This thesis will focus on those channels defined in Release 5.

2.5.1.2.4.1 SCH

Synchronization channel is used for those UEs which need to search for a cell. This is a typical situation when a UE is switched on, in cell re-selection procedure and in handover procedure. Once the UE has detected the SCH, the UE will be synchronized at chip, slot and frame level.

SCH is split into primary and secondary SCH. The primary SCH transmits a code, Primary Synchronization Code (PSC). This same code is transmitted in each slot and in all cells. On the other hand, the secondary SCH transports a sequence of codes called Secondary Synchronization Codes (SSC). This sequence will identify a scrambling group of the 64 available scrambling groups. Section 2.4.1.1.2 described the scrambling codes.

2.5.1.2.4.2 CPICH

Common pilot channel is used as phase reference for several physical channels. CPICH has a fixed SF and a fix bit rate. SF is 256 and its rate is equal to 30 kbps. Moreover, this channel is always scrambled with a primary code.

CPICH is divided into two groups: Primary Common Pilot Channel (P-CPICH), and Secondary Common Pilot Channel (S-CPICH). The P-CPICH uses always the same channelization code. Moreover, primary scrambling codes are used for this channel. This channel is broadcasted over a cell and a cell can only have one P-CPICH. On the other hand, S-CPICH uses any channelization code whose SF is 256. This channel can be scrambled with either primary or secondary codes. S-CPICH is also broadcasted; however, it can be broadcasted either over a cell or over a part of it. S-CPICH is not a mandatory channel. However, if present, there can be more than one S-CPICH channels within a cell.

2.5.1.2.4.3 P-CCPCH

The UE learnt about the primary scrambling code through the SCH. P-CCPCH will provide to the UE the configuration of the other common physical channels.

This channel also has a fix SF equal to 256. Its bit rate is 30 kbps, but due to a “silent” period at the beginning of the slot, the effective bit rate is 27 kbps. During this “silent” period, the UE will read the SCH.

P-CCPCH is transmitted with relatively high power to the terminals. This is due to the fact that if a UE cannot demodulate the P-CCPCH channel, the UE will not be able to access the network. Obviously, the power used for this channel will have a direct effect on the cell capacity.

2.5.1.2.4.4 S-CCPCH

The secondary common control physical channel is the only physical channel which is mapped onto two different transport channels: FACH and PCH. These channels can be carried together into an S-CCPCH or into separated S-CCPCH channels. If there is only one S-CCPCH channel within a cell, its bit rate will be fairly low, i.e. 30 kbps and SF equal to 256. However, if several S-CCPCH are used, each channel can use a different bit rate, from 30 kbps to 1920 kbps. SF values will, thus, vary between 256 and 4. Nonetheless, one S-CCPCH will have to have a SF equal to 256.

2.5.1.2.4.5 AICH

Acquisition indicator channel is tightly associated to PRACH channel (section 2.5.1.2.2.1). This channel is used as an answer to the PRACH preamble. AICH will echo the preamble signature sent in the PRACH.

AICH has a fixed SF and, hence, a fix bit rate. Its SF is 256. Unlike other channels, AICH channel is spread over two radio frames. These 20 ms are divided into 15 access slots. Each access slots is made of two parts: acquisition indicator and an off transmission period.

2.5.1.2.4.6 PICH

Paging indicator channel is used to support the sleep mode procedure. PICH channel will notify the UE when there is paging information for it, so the UE does not need to monitor continuously the S-CCPCH physical channel. PICH SF is fixed to 256, as it is for all the common downlink physical channels (with the exception of the S-CCPCH channel). A PICH is always associated to an S-CCPCH channel.

2.5.2 Data Link Layer

The physical layer, L1, offered services to the upper layer, the data link layer, L2. L2 provides service to L3. L2 offers radio bearer services and signaling radio bearer services to L3. L2 is divided into four sub-layers, as already mentioned in section 2.5. L2 sub-layer stack is illustrated in Figure 6.

MAC sub-layer interfaces with L1 and with RLC sub-layer. MAC sub-layer provides service to the RLC through logical channels and it accesses L1 services, i.e. transport channels, through SAPs. Logical channels answer the following question: what is transported. On the other hand, transport channels answers how is transported. MAC sub-layer is discussed in section 2.5.2.1.

RLC sub-layer is over MAC sub-layer and it interfaces PDCP, BMC, and L3. RLC accesses MAC services (i.e. logical channels) through SAPs. RLC main functionalities are retransmission and segmentation. RLC functionalities are explained in section 2.5.2.2.

PDCP interfaces RLC sub-layer and L3. It provides services to L3 NAS or to the relay at RNC. In contrast to the other layers, PDCP sub-layer can be only found in the PS domain. The main functionality of PDCP is compression. Section 2.5.2.3 describes PDCP.

BMC sub-layer also interfaces with RLC sub-layer and with L3. BMC provides multicast and broadcast services to L3. More details about BMC are presented in section 2.5.2.4.

2.5.2.1 Medium Access Control

As mentioned in the previous section, MAC sub-layer is the first sub-layer over the physical layer. It uses transport channels to communicate with the physical layer and it offers to the upper sub-layer (RLC sub-layer) logical channels. MAC, as the rest of layers, is controlled by the RRC. Figure 6 shows the layers' layout and the relationship between each others.

Through logical channels, MAC is able to send data coming from upper layers to a peer MAC entity. Besides, subject to RRC request, MAC is able to reallocate radio resources and change its own parameters. Finally, MAC presents to the RRC the results of its measurements, such as for example traffic volume and quality indication. Those are the services offered by the MAC sub-layer to upper layers.

2.5.2.1.1 MAC Functions

According to [3GPP_321], the functions performed by MAC sub-layer are:

- Mapping between logical channels and transport channels
- Selection of the appropriate TF for each transport channel depending on instantaneous source rate. The RRC will inform about the TFCS to the MAC sub-layer. MAC will select a TF among the TFS.

- Priority handling between data flows of one UE. Priority handling is done at TFC level, i.e. TFCs within a TFCS will be prioritized. MAC will select a TFC in a way that each data flow can be assigned to a TF. For example, high priority data flows will be assigned to a “high bit rate” TF and low priority data to a “low bit rate” TF.
- Priority handling between UEs by means of dynamic scheduling. UTRAN should maximize the use of the limiting resource, i.e. the spectrum. In order to achieve that, MAC will have to efficiently schedule common, shared, and dedicated transport channels through prioritization.
- Identification of UEs on common transport channels. Common transport channels can be used to send dedicated data to users (or UTRAN). In such cases, the UE (or UTRAN) needs to be identified.
- Multiplexing/demultiplexing of upper layer PDUs into/from transport blocks sets delivered to/from the physical layer on dedicated transport channels.
- Multiplexing/demultiplexing of upper layer PDUs into/from transport blocks delivered to/from the physical layer on common transport channels.
- Traffic volume measurement. MAC measures the logical channels traffic volume and notifies the RRC about the results. These results will be useful to the RRC for switching decision making.
- Transport channel type switching. MAC has to decide what type of transport channel (dedicated or common transport channel) is sent to the physical layer in each moment. Switching decision is based on traffic volume measurements.
- Ciphering for transparent mode RLC.
- Access service class selection for RACH transmission. RACH resources can be prioritized through access services classes. MAC will select the appropriate access service class in each moment.

2.5.2.1.2 Logical Channels

Logical channels describe what is transported. These channels are divided into two groups: control channels (for control plane information) and traffic channels (for user plane information). RLC makes use of these logical channels through SAPs (depicted in Figure 8 as ovals).

Figure 8 shows the relationship between logical channels and transport channels. The upper side of Figure 8 shows the mapping between logical channels and transport channels seen from the UTRAN in the downlink. The lower side presents the mapping seen from the UE in the uplink.

2.5.2.1.2.1 Control Channels

The logical control channels are:

- Broadcast Control Channel (BCCH) – This channel is used for system control information broadcasting. It exists only in the downlink.
- Paging Control Channel (PCCH) – As the BCCH channel, PCCH is a downlink channel which is used for paging purposes.
- Dedicated Control Channel (DCCH) – This is a point-to-point bi-directional channel which is set up in the RRC connection establishment procedure. It carries dedicated control information between RNC and the UE.
- Common Control Channel (CCCH) – CCCH is a bi-directional channel. It carries control information between the network and the UE. This channel is used by those UEs which access a new cell after cell re-selection as well as by UEs which do not have a RRC connection.

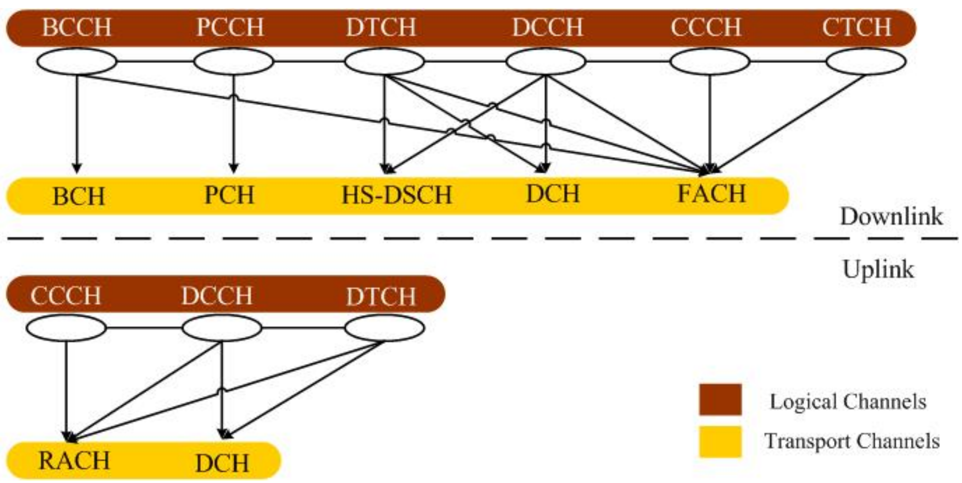

Figure 8 – Mapping between logical channels and transport channels, Release 5

2.5.2.1.2.2 Traffic Channels

Traditionally, there has been only two different logical traffic channel:

- Dedicated Traffic Channel (DTCH) – DTCH is a dedicated point-to-point channel which can be used in the uplink as well as in the downlink. This channel carries user information.
- Common Traffic Channel (CTCH) – This point-to-multipoint channel is used to carry dedicated information in the downlink to all or a group of UEs.

2.5.2.1.3 MAC Architecture

MAC architecture is depicted in Figure 9 as described in Release 5. Although, Release 6 has introduced two new entities, MAC-es/e and MAC-m, they are out of the scope of this thesis and will not be discussed.

- MAC-b is responsible of the BCH transport channel. There is only one MAC-b at Node B for each cell. On the other hand, UE can have one or more than one MAC-b entities.
- MAC-c/sh is responsible of common transport channels, i.e. the PCH, FACH, and RACH transport channels. We can observe in Figure 9 that BCCH logical channel can be also handled by MAC-c/sh which will map the BCCH into a FACH transport channel. There is only one MAC-c/sh entity in the UTRAN which is located in RNC. UE also have only one MAC-c/sh/m entity.
- MAC-d is responsible of the DCH transport channels. Besides, it controls the access to the MAC-c/sh as well as the MAC-hs. This is represented in Figure 9. There is only one MAC-d in the UE. In the UTRAN, there will be as many MAC-d entities as UEs the UTRAN is giving service. MAC-d is implemented in the SRNC.

MAC entities are configured, control and supervised by the radio resource control. RRC is described in section 2.5.3.1.

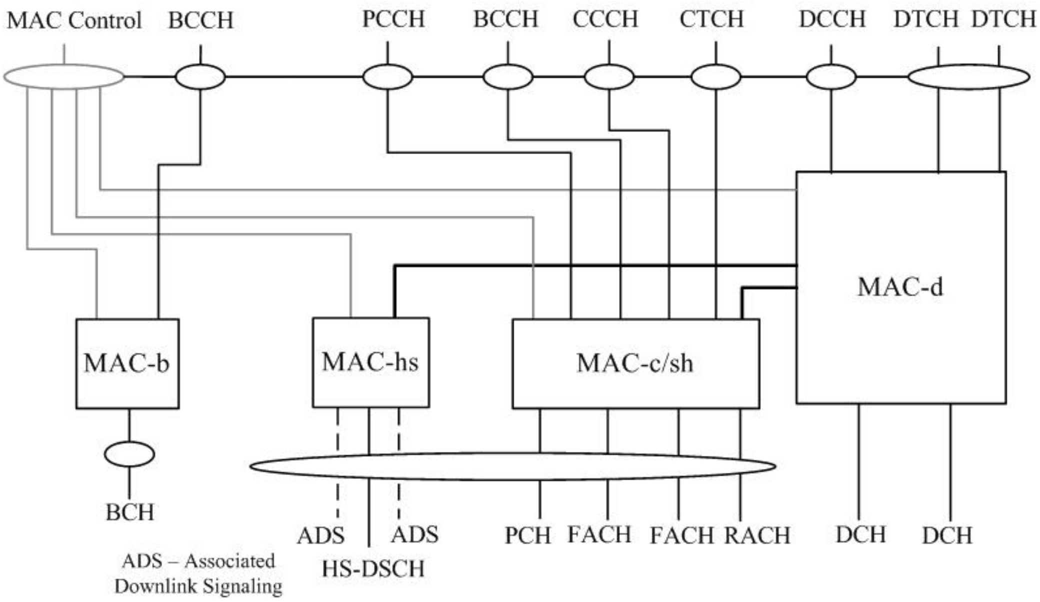

Figure 9 – MAC architecture, Release 5

2.5.2.2 Radio Link Control

RLC is placed on top of the MAC sub-layer. It accesses MAC services, i.e. logical channels through SAPs. As we can observe in Figure 6, RLC handles the control plane as well as the user plane. RLC, as the MAC sub-layer and the physical layer, is managed by the RRC which controls the internal configuration of the different layers.

2.5.2.2.1 RLC Services

RLC provides three types of radio bearer services: Transparent Mode (TM), Unacknowledged Mode (UM), and Acknowledged Mode (AM).

In TM, the RLC receives data from upper layers and sends the data to the MAC, segmenting the data when necessary, without adding any extra headers. The receiver will receive the data, reassemble it when necessary, and send it to upper layers. RLC will discard or mark corrupted data (the physical layer will check for erroneous data through the CRC).

In UM, sender segments, concatenates or pads the data when necessary, and can cipher the information before sending it to the lower layer. The sender will also add a RLC header to the data sent by the upper layer. The receiver will decipher the data (if it was cipher by the sender), remove the RLC headers, and reassemble the data (if it was segmented by the sender). Any erroneous data will be discarded by the receiver.

In AM, the sender segments, concatenates or pads the data when necessary, and can cipher the information before sending it to the lower layer. Besides, the data sent will be stored until an acknowledged is received. This enables the possibility of re-transmission. This mode will then provide error-free data delivery, and unique in-sequence as well as unique out-of-sequence data delivery.

2.5.2.2.2 RLC Functions

RLC performs different functions to provide the services described above. According to [3GPP_322] and [3GPP_301], these functions are:

- Segmentation and reassembly
- Concatenation
- Sequence number check
- Protocol error detection and recovery
- Ciphering
- SDU discard
- Padding
- Transfer of user data
- Error correction
- Duplicate detection
- Flow control
- Out-of-sequence SDU delivery

2.5.2.3 Packet Data Convergence Protocol

PDCP is placed on top of RLC sub-layer, in the U- plane side. Moreover, PDCP only exists in the PS domain. Non-access-stratum at the UE, as well as the relay at RNC, benefits from the PDCP services. The main PDCP functionality is to provide header compression to those IP services which have to go through the air interface. The current compression algorithms are specified in RFC 2507 and RFC 3095 [3GPP_323].

2.5.2.4 Broadcast/Multicast Control

BMC sub-layer is used by broadcast and multicast services. This sub-layer is also on top of the RLC sub-layer, and it only exists in the user plane as depicted in Figure 6. BMC uses UM service provided by the RLC.

BMC functions and operations are described in [3GPP_301] and [3GPP_324]. BMC functions are briefly presented here.

- Storage of cell broadcast messages
- Traffic volume monitoring and radio resource request for CBS.
- Scheduling of BMC messages
- Transmission of BMC messages
- Delivery of cell broadcast messages to NAS.

2.5.3 Network Layer

The network layer is above L2. L3 is made up of two sub-layers. The first sub-layer is the RRC which exists only in the control plane. The second sub-layer, which is on top of the RRC, covers both control and user planes. The functionality of this second sub-layer is duplication avoidance. Figure 6 shows protocol architecture. In it, we can only see the RRC in L3.

The network layer contains protocols which are part of the Access Stratum (AS) and some other protocols which are part of the NAS. Duplication avoidance sub-layer draws the line between AS and NAS. This thesis only covers the AS since NAS is not covered by the 3GPP TSG RAN.

2.5.3.1 Radio Resource Control Protocol

Between the UE and the UTRAN there are, in addition to user information flows, control signaling flows. The RRC manages the later. It also manages the characteristics and properties of L2 and L1.

The RRC uses the services provided by lower layers. These services are, as mentioned before, radio bearers. On the other hand, RRC offers services to upper layers. The services offered by the RRC are: general control, notification, and dedicated control [3GPP_301].

2.5.3.1.1 RRC Functions

The RRC is one of the key entities since it controls most of the signaling. Next to that, it is essential for the right functioning of the radio resource management. Due to this fact, the RRC has a large amount of functionalities. The functions performed by the RRC are shortly listed below ([3GPP_301], [3GPP_331]); however, a detail description of all the functionalities and its characteristics is available in [3GPP_331].

- Broadcast of information provided by the NAS
- Broadcast of information related to the access stratum.

- Establishment, re-establishment, maintenance and release of an RRC connection between the UE and the UTRAN
- Establishment, reconfiguration and release of radio bearers
- Assignment, reconfiguration and release of radio resources for the RRC connection.
- RRC connection mobility functions
- Control of requested QoS
- UE measurement reporting and control of the reporting
- Outer loop power control.
- Paging/notification
- Initial cell selection and cell re-selection

2.6 Radio Resource Management

The radio interface is a shared interface and, thus, its resources are also shared. UMTS has to set a number of algorithms to control and manage the radio resources in order to offer certain QoS, to provide the service level promised to the users, and to improve the network operation. These algorithms are called Radio Resource Management (RRM) algorithms. RRM is placed in the UTRAN (Node B and RNC), and also in the UE. Nonetheless, RNC will take final decisions related to the RRM.

RRM algorithms are grouped into the following algorithm groups: handover, power control, admission control, packet scheduling, and congestion control.

2.6.1 Handover

Handover algorithms have been present since the very early steps of mobile networks. UMTS, though, has implemented new and improved handover algorithms.

Handover algorithms are responsible of creating new connections for a UE, and releasing the non-necessary connections. The reason why to create new connections is due to the fact that the network has to provide mobility support to UEs.

There may be many reasons to create new connections and to release others, for example a certain connection may have poor uplink/downlink quality, measurements done by a UE or by the network may result in a handover request, a UE may request for another service, or the network may need to load traffic [3GPP_922].

The handover process is done in few steps: measurement, decision, and execution [Kaa_05]. Measurements are described in depth in section 8.4 in [3GPP_331]. Handover decision can be performed by the UE (Mobile Evaluated HandOver – MEHO), by the SRNC (Network

Evaluated HandOver – NEHO), or it can be agreed between the UE and the network. Nonetheless, RNC has the final decision about executing or not executing the handover.

There are different handover types. They are briefly explained in the following sections.

2.6.1.1 Hard Handover

Hard handover is characterized by the fact that the connection between the UE and Node B is released before a new connection between the UE and another Node B or BS is established. Hence, there will be a brief cut in the connection.

Hard handover has been the traditional handover in 2G networks. UMTS also supports hard handover and it can happen in three situations. The first situation takes place when a UE performs a handover procedure from a UMTS network to a GSM network. The other way round also requires hard handover. This type of hard handover is called inter-system handover. When a UE performs an inter-system handover from UMTS to GSM, the UE (and sometimes Node B) will need to measure some parameters of the other system. This is achieved through a mechanism called compressed mode. Compressed mode allows introducing a small gap in the transmission, letting the UE to perform those measurements. More information about compressed mode can be found in section 4 in [3GPP_212] and section 6 in [3GPP_215]. The second situation takes place when a UE performs a handover procedure and the SRNC which will take care of the UE connection has no connection through the Iur interface with the SRNC which was supporting the UE connection. This type of handover is known as intra-frequency hard handover if the new Node B is using the same frequency as the previous Node B. If the new frequency differs from the previous frequency used, the handover is called inter-frequency hard handover.

2.6.1.2 Soft and Softer Handover

Soft/softer handover concept was introduced when UMTS was developed. Soft/softer handover maintains the old connection while a new connection is established. Soft/softer handovers allow great flexibility and they highly improve the QoS. Flexibility is due to the fact that RNC will decide, depending on the measurements, whether the old connection is to be released or not. On the other hand, RNC may release the new established connection if, at some point, it does not fulfill RNC expectations. In the meantime, the network will receive the UE signal through different connections. If the old and new connections are set up by the same Node B, the handover is called softer handover. However, if the old and new connections are set up by different Node B’s, the handover is called soft handover. Soft and softer handovers can be either inter and intra-frequency, depending whether the new connection is set up using the same frequency as the previous connection, or not [3GPP_922].

QoS improvement is due to the fact that the UE signals are received by one or several Node B’s and later, these signals are combined. As a result, diversity signal gain will be achieved and the combined signal will have a better SIR.

2.6.2 Power Control

Power control is highly important for the good operation of the network. It is, indeed, critical in the uplink. In general, all cells use the same frequency. The only way to differentiate uses is through the spreading codes. Thus, the interference a UE experiences depends on the number of UEs around it and on the power used in their transmissions. If the power with which the UE transmits is not controlled, some UEs would experience too much interference and, hence, they would not achieve a proper QoS. Obviously, UEs close to Node B need to use less power to achieve the desired SIR than those terminals which are situated far from Node B. This fact can create what is known as 'near-far problem'. The near-far problem is alleviated when all the signals received in Node B have the same power strength. This is difficult to achieve due to the unpredictable behavior of the radio link.

Though power control is not so critical in the downlink, it is also applied to minimize interference with other cells and optimize the capacity of the network.

UMTS implements two power control mechanisms: open loop power control (OLPC), and closed loop power control (CLPC). OLPC mechanism helps UEs to calculate a rough estimate of the power which it has to use when it wants to access the network. OLPC mechanism makes use of the CPICH power measured by the UE. The higher the CPICH power is, the lower the power UEs need to use to their transmissions. Though OLPC is a fair mechanism to roughly assess the power a UE needs to start using at the beginning of the connection, it is not a good mechanism to control accurately the power transmitted by the UE once the connection is set up. Instead, CLPC is used for this purpose.

CLPC is used both in the downlink and uplink. CLPC mechanism are divided into inner (or fast) and outer loop power control mechanisms. Fast CLPC mechanisms assess the SIR (or other key parameters). The calculated value is checked against a certain target SIR. If the calculated SIR is lower than the target SIR, Node B (UE) will request the UE (Node B) to increase the transmitted power. However, if the SIR is higher than the target SIR, Node B (UE) will request the UE (Node B) to reduce the transmitted power. This process is done 1500 times in a second by Node B and the UE. On the other hand, the outer CLPC, which is controlled by RNC, aims to optimize the fast CLPC performance. Basically, RNC will command Node B to increase or reduce the target SIR (or the key parameter used) depending on the quality of the transmission.

Power control mechanisms have to behave slightly different in situations such as soft-handover (in which a UE can receive different commands from different Node B's), site selection diversity, and compressed mode [3GPP_214].

2.6.3 Admission Control

As we mentioned before, the interference within a cell depends, among many things, on the number of users. An excessive number of users within a cell would compromise the network performance and stability.

Admission control will take care that there are not more users than planned. Therefore, each time a radio access bearer is set up or modified, the admission control procedures will need to be executed to verify that the network is able to handle the traffic and offer some certain minimum QoS to the active users. If the admission control decides it cannot accept more users, it will reject new incoming connection.

Admission control is implemented in RNC since RNC controls one or several Node's; hence, RNC has full information about the ongoing traffic, QoS offered, free resources, etc. within the cells under its control.

2.6.4 Packet Scheduler

Packet scheduler is a very important element of the RRM. Besides, it is tightly connected to the congestion control, admission control, and soft handover operation. The packet scheduler is placed in RNC.

Many packet scheduling algorithms have been proposed along the way; however, fair throughput scheduling, fair time scheduling and C/I scheduling are the most common algorithms in UMTS. Fair throughput scheduling and fair time scheduling are favored, though [Hol_04].

Scheduling is based on the QoS classes which UMTS is able to provide. The application will define the type of class used for that service. Interactive and background classes will be used for non-real-time services, while conversational and streaming will be used for real-time services. Real-time and non-real-time services have different QoS requirement; therefore, the packet scheduler will treat them differently. A certain amount of the air interface capacity will be assigned to each service. This is the first task of the packet scheduler. Next to that, the packet scheduler also assigns the transport channel needed for each packet, increases or decreases the user bit rate, handles capacity requests, and implements queuing algorithms.

2.6.5 Congestion Control

Though admission control prevents future congestion situations, there might still be some network congestion situations. The congestion control mechanisms aim to maintain the network performance within acceptable levels and to maintain the network stability.

Congestion algorithms will be activated when congestion is detected in the network. Then, these algorithms will bring the network back to a stable situation through user prioritization, selective connection blocking, transmission rates reduction, inter-system handovers, or power control procedures.

Chapter 3

High Speed Downlink Packet Access - HSDPA

3.1 Overview

As explained in Chapter 1, there has been a clear need for higher bit rates as well as better spectral efficiency in mobile communications. Several technologies, such as CDMA-2000 or WiMAX, provide solutions to offer high bit rates. In order to compete with these new technologies and support the new needs, UMTS added a new feature in 3GPP Release 5. This feature enhanced the bit rates as well as the spectral efficiency. Furthermore, this feature will help to maintain UMTS as a very competitive technology to provide high bit rates in mobile communications. This feature is HSDPA.

HSDPA technology is based in four pillars:

1. Fast link adaptation (LA).
2. Hybrid Automatic Repeat reQuest - HARQ.
3. Fast packet scheduling.
4. Short frame size.

UMTS architecture has not really gone through many changes to introduce HSDPA. Only, a new MAC layer has been introduced. Furthermore, some logical functions traditionally performed in RNC have been moved to Node B. This new MAC layer is called MAC-hs and it is located in Node B. MAC-hs will perform among other things scheduling, HARQ, and flow control. These functions have been traditional performed in RNC. Next to this, the physical layer has been upgraded with new physical channels with shorter frame size (2 ms) and with link adaptation functions, such as adaptative modulation and coding. Link adaptation replaces two important features of the traditional UMTS system. These features are variable spreading factor and fast power control. Another feature disabled in HSDPA is soft-handover.

Thank to these changes, HSDPA is able to achieve a theoretical bit rate of up to 14.4 Mbps. However, due to technological limitations, the maximum bit rate is likely to be closer to 10 Mbps. This fact implies that HSDPA increases the spectral efficiency to the level of being close to the Shannon limit. This is depicted in Figure 10. Moreover, due to the fact that scheduling, retransmissions, modulation and coding decisions are taken nearer of the air interface, HSDPA will reduce significantly system delays. As a result, HSDPA shows to be the best solution to carry non-real time traffic, for instance interactive class or background class. HSDPA will also aim to carry streaming traffic in further releases; however, there may be certain limitations depending on the type of streaming traffic which is sent.

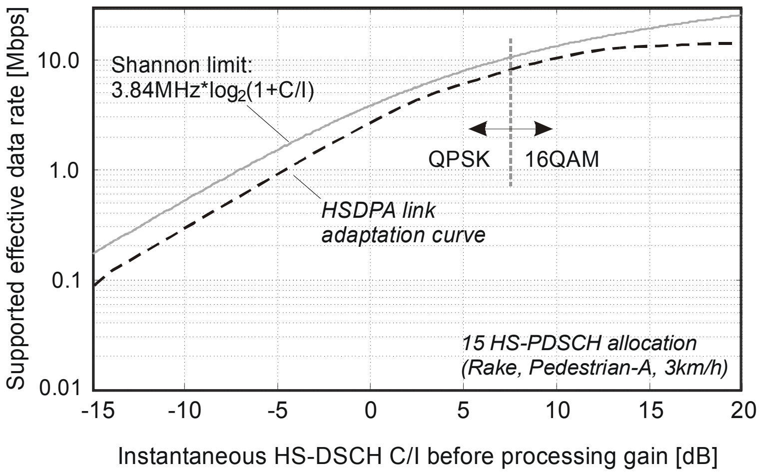

Figure 10 – HSDPA data rate vs. Shannon limit

3.2 Fast Link Adaptation

UMTS, as defined in 3GPP Release 99, uses power control and variable spreading factor to cope with the effects of the air interface. Power control and SF selection are executed each 1500 Hz and each TTI respectively. HSDPA disables these features and tackles the effect of the air interface in a different way. HSDPA uses adaptative modulation and coding, instead. Furthermore, Node B will select the right modulation and coding scheme each HSDPA TTI, i.e. 2 ms. Therefore, link adaptation refers to the adaptative modulation and coding (AMC), and to the number of channelization codes chosen. Fast link adaptation refers to the fact that link adaptation is performed each 2 ms.

Link adaptation decisions are based on the feedback provided by UEs and it is executed by Node B. This feedback comes through the uplink channel HS-DPCCH. This channel carries in its payload a field called channel quality indicator (CQI). CQIs are selected by the UEs depending on the UE's category [3GPP_306], P-CPICH power level, or S-CPICH power level. Moreover, the CQI value chosen by the UE takes into consideration that the transport block error probability does not go beyond 10 % [3GPP_214].

CQIs transmission periods are configured by higher levels. CQI transmission periods are selected by the network and commanded to the UE. Besides, UEs can be configured to send no feedback towards Node B [3GPP_858].

Although, CQIs are meant to help Node B to perform link adaptation, Node B will execute fast link adaptation depending on the implemented algorithm, which is vendor specific. Not only decisions will be based on the CQI (or any other implemented algorithm), but decisions will be also based on the available resources of Node B. In general, regardless of the implemented algorithm, link adaptation algorithms will select a transport block size, a modulation, and the number of codes. This set of parameters is called transport format and resource combination (TFRC).

Though link adaptation algorithms are not specified, it is recommended by the 3GPP the use of the CQIs. [3GPP_214] specifies the CQI mapping for the different UE categories to the TFRC.

3.2.1 Adaptative Modulation and Coding

HSDPA uses two modulation schemes for the HS-DSCH channel. One is the QPSK modulation which was already used for the DCH channel in Release 99. The second modulation scheme is 16QAM. Figure 11 and Figure 12 shows both constellations and how bits are assigned according to [3GPP_213]. As we can observe, 16QAM uses four bits to represent a symbol, while QPSK uses just two bits for the same purpose. Hence, 16QAM doubles the data rate. Table 2 presents the achievable bit rates for HSDPA depending on the modulation, code rate, and the number of channelization codes.

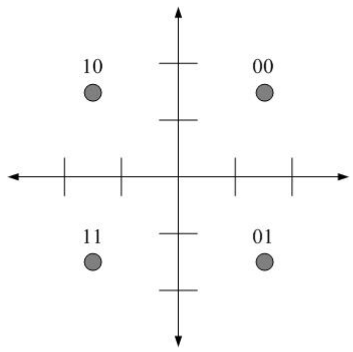

Figure 11 – QPSK constellation

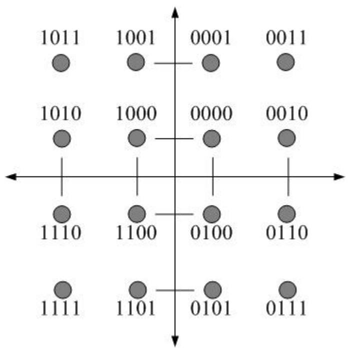

Figure 12 – 16QAM constellation

Node B will adjust the modulation required in each TTI depending on the radio environment of the UE. This fact will minimize the effect of the near-far problem. In other words, UEs nearer Node B will receive better Es/No than those UEs situated far from Node B. Therefore, Node B will select a higher modulation level (16QAM) for the former UEs, while a lower modulation level (QPSK) will be preferred for the later UEs. This is depicted in Figure 13. As we can observe in the picture, higher modulation levels are selected when the Es/No received by the UE is appropriate for those higher modulations levels. For instance, 16QAM will be selected when the Es/No at the UE guarantees a correct demodulation by the UE.

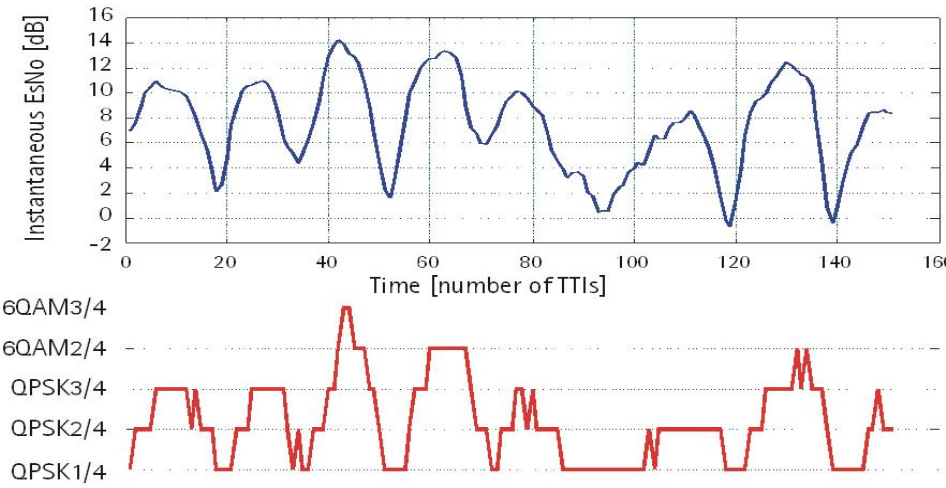

Figure 13 – Fast link adaptation

UMTS system makes use of both convolutional and turbo encoders; however, HSDPA uses only turbo encoders since turbo encoders are more efficient than convolutional encoders [Lee_00]. For backward compatibility reasons, HSDPA still continues using 1/3 rate turbo encoders. The effective code rate (ECR), though, will differ from the turbo encoder rate. After scrambling the data through the turbo encoder, the data rate will be adjusted according to the TFRC. Data is adjusted by means of puncturing or repetition of the turbo encoder output. Puncturing or repetition rate will, then, adjust the ECR [3GPP_212], [3GPP_858].

The task of the turbo encoder module and the puncturing/repetition module is to encode the data sent over the air interface in a way that data is protected against channel noise, i.e. data can be recovered even if the channel noise corrupted it. Puncturing and repetition is a HARQ functionality. The HARQ module is explained in section 3.3.

Table 2 shows the achieved data rates for different modulations, different number of codes and different ECR.

		HS-PDSCH codes		
Modulation	ECR	1 code	5 codes	15 codes
QPSK	1/4	120 kbps	600 kbps	1800 kbps
	2/4	240 kbps	1200 kbps	3600 kbps
	3/4	360 kbps	1800 kbps	5400 kbps
16 QAM	2/4	480 kbps	2400 kbps	7200 kbps
	3/4	720 kbps	3600 kbps	10800 kbps
	4/4	960 kbps	4800 kbps	14400 kbps

Table 2 – HSDPA data rates for the different modulation and rate combinations

3.2.2 Link Adaptations Algorithms

Link adaptation algorithms are not specified by the 3GPP. 3GPP, instead, provides some recommendations to follow. As mentioned before, 3GPP recommended that link adaptation algorithms take into consideration the P-CPICH. Algorithms are likely to use the feedback sent by the UE to adapt the link to the current radio conditions. These algorithms will also be likely to be based on two loops: inner loop and outer loop. The only available feedback in HSDPA is the CQI parameter, L1 ACK/NACK, and DTX. CQI could be used in the inner loop algorithm, while L1 ACK/NACK and DTX can be used for the outer loop [Hol_06]. Nonetheless, real implementations will be vendor specific.

[Pok_05], [Mul_05], [Bau_03], [Wan_05], [Nak_02], and [Dot_04] present and study a variety of link adaptation algorithms and schemes. [Hol_06] also presents several link adaptation algorithms.

3.3 Hybrid ARQ

Hybrid automatic repeat request is an error control method performed in L1. HSDPA uses HARQ stop-and-wait. Stop-and-wait means that the sender will be halt until an acknowledgement is received. After that, the sender will continue sending information. Stop-and-wait is very inefficient unless several processes are handled in parallel. In this way, while a process is halt and waiting for an acknowledgement, another process can send information to another user.

Traditionally in UMTS, retransmissions have been handled by RNC. Due to this fact, RTTs were relatively high. In Release 5, HSDPA retransmissions are handled by Node B. As a result, faster retransmissions will be achieved and so, smaller RTTs. Release 99 achieved RTTs around 110 and 150 ms. However, RTTs in HSDPA will be around 70 ms [Hol_06].

Figure 14 depicts the different retransmission layers. We can observe that the fastest retransmissions will be MAC-hs retransmissions since Node B is the only element involved in the process. After a certain number of retransmissions, RNC will take over Node B and it will perform a RLC retransmission. RLC retransmissions will have longer delays, which are direct effect of the transmission through the Iub interface and the additional processing steps. Finally, if the RLC retransmissions fail or a time-out at the application level happens, a TCP retransmission will be necessary. TCP retransmissions will be the slowest since these retransmissions will need to go through the whole RAN, core network, and the Internet.

The fast retransmission procedure is presented with more detail in Figure 15, as well as the procedure used in Release 99. The procedure can be simplified into four steps. In the first step (1), RNC sends a packet to the UE. This packet will go through Node B where it will be buffered, scheduled, and finally forwarded to the UE. The UE will reply (2) with an ACK, or a NACK. As depicted in Figure 15, in Release 99, the ACK/NACK is forwarded to RNC which would take care of retransmitting (3) the appropriate packet whether it is necessary.

However, in Release 5, the ACK/NACK is received by Node B which will take action to retransmit the appropriate data to the UE. Finally, the UE sends a positive acknowledgement (4) to Node B which will forward the ACK/NACK back to RNC. Whether the UE sends a negative acknowledge to Node B, Node B will retransmit again the correct data. After a certain amount of repetitions, Node B will ask RNC to take action (Figure 14, RLC retransmission), and the process will start again. If after *n* RLC retransmissions the UE still replies with negative acknowledgements, higher levels will take over the retransmission and a TCP retransmission will be performed.

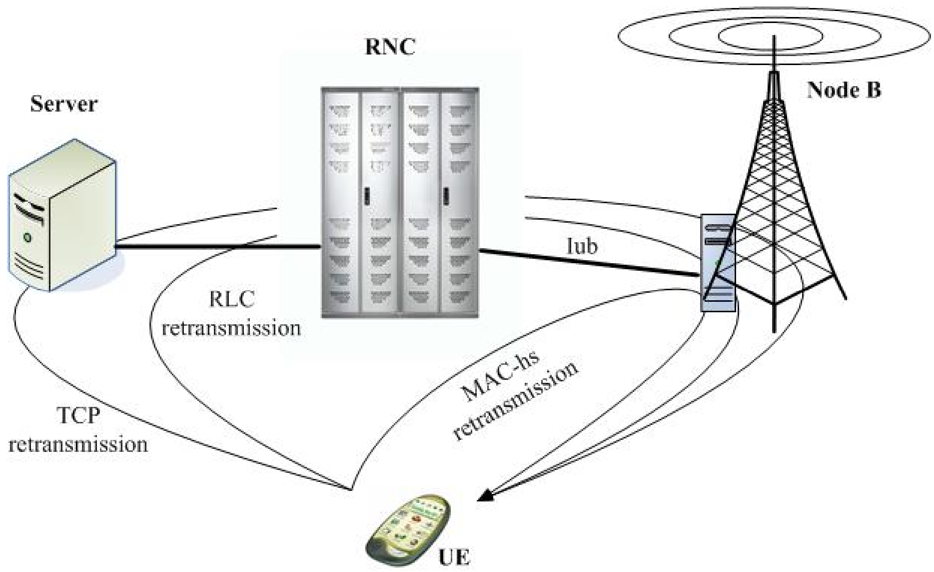

Figure 14 – Retransmission layers

HARQ implements two different retransmission strategies: chase combining and incremental redundancy. “Chase combining” means that retransmissions will be totally identical to the original transmitted data. On the other hand, “incremental redundancy” means that retransmissions will differ from the original transmitted data. Both strategies are described in Figure 16. As it was mentioned in the previous section, the output of the channel coding module is the input of the HARQ module. The output of the HARQ module will depend on the transmission strategy selected and the puncturing/repetition rate.

First transmissions will always be self-decodable. All systematic bits are sent, in addition to some parity bits. The number of parity bits will depend on the puncturing/repetition rate. If data needs to be retransmitted, the sender will either choose to send again a self-decodable stream or a non-self-decodable stream. On one hand, chase combining mechanism will be made of two (identical) self-decodable streams. On the other hand, incremental redundancy will be made of a self decodable stream and a non-self-decodable stream.

Which strategy is to be used will mainly depend on the channel quality and the UE capability class. Chase combining has been proven to be more effective than incremental redundancy when high (over 0.5) effective coding rates are used. Incremental redundancy offers better efficiency for lower effective coding rates. On the other hand, incremental redundancy presents a drawback: UEs will require larger memory capacity [Kol_03], [Fre_01], [Fre_02].

As stated before, HARQ also performs puncturing and repetition of the data obtained in the turbo encoder. Puncturing and repetition function are to adjust the number of bits returned by the turbo encoder to the number of bits of the HS-DSCH physical channel. The detailed HARQ rate match process is explained in [3GPP_858] and [3GPP_212].

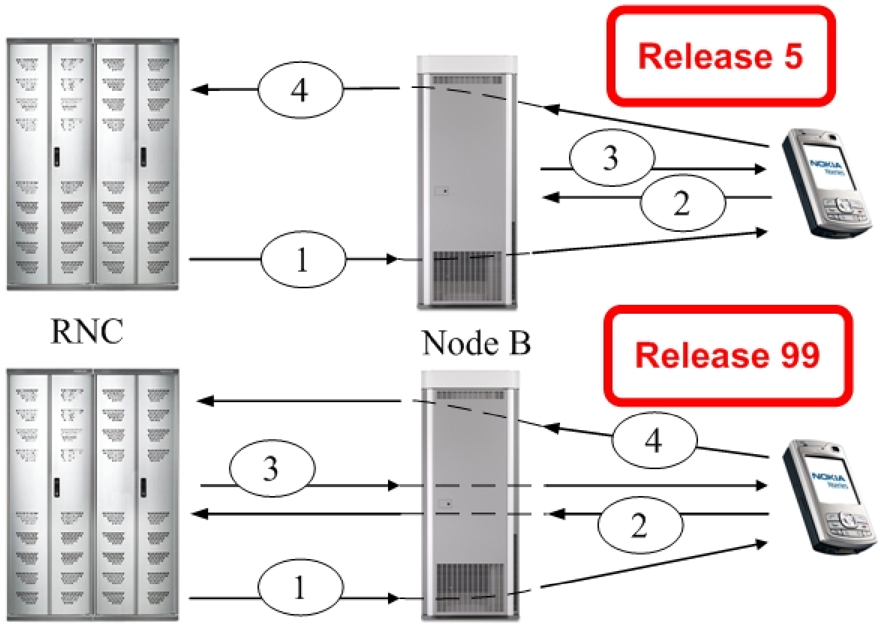

Figure 15 – Retransmission procedure

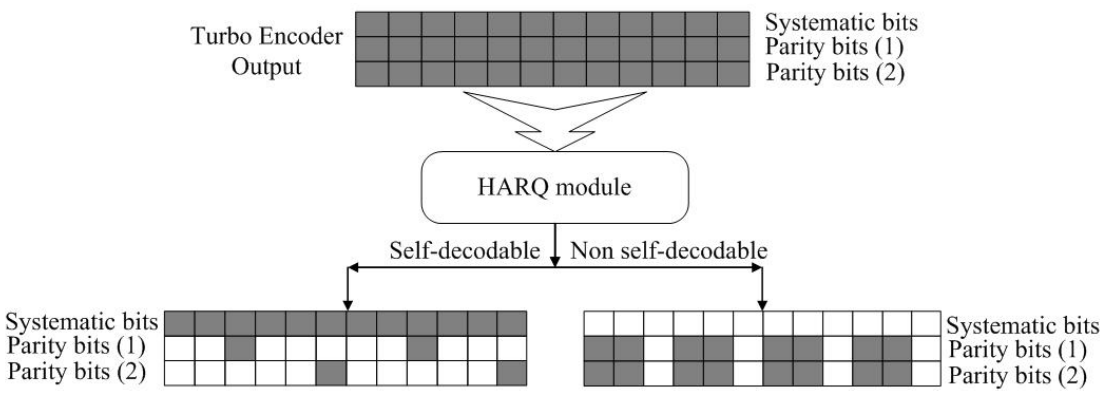

Figure 16 – HARQ retransmission strategies

3.4 Fast Packet Scheduling

Packet scheduling has always been performed in RNC. HSDPA, though, moved packet scheduling decisions to Node B in order to cope and respond to the fast link adaptation decisions and retransmissions. If packet scheduling would have been kept in RNC, little advantage would have been achieved through fast link adaptation and fast retransmission algorithms.

Packet scheduling algorithms are not standardized by the 3GPP; consequently, these algorithms are vendor specific. Nonetheless, packet scheduling algorithms will have to take into account several factors, such as for example free cell/network resources (codes and power), traffic class, UE category and capability, channel conditions, or buffered data in Node B [Lai_06],[Hol_06].

Packet scheduling algorithms are an active research topic in the scientific community and many research papers implementing different scheduling algorithms have been published, such as for example [Ame_04], [Ani_04], [Boa_03], and [Wan_04]. Nevertheless, HSDPA is likely to implement one of the traditional packet scheduler algorithms. These traditional algorithms are round robin, maximum C/I (or maximum throughput), proportional fair, minimum bit rate scheduling, minimum bit rate scheduling with proportional fairness, and maximum delay scheduling [Hol_06], [Kol_03]. Which algorithm will be chosen for HSDPA depends on, as mentioned before, the vendor decision. Nonetheless, the algorithm will aim to achieve a compromise between throughput, coverage, and user fairness.

3.5 HSDPA Architecture

3.5.1 MAC-hs

A new MAC layer was introduced in 3GPP Release 5 to implement HSDPA. Despite all MAC functions have been traditionally performed in RNC, HSDPA has move part of RNC intelligence to Node B. Hence, faster decisions and shorter reaction times can be achieved. Figure 9 shows the general MAC architecture in Release 5, and Figure 17 shows a more detailed structure of the MAC-hs. As depicted in Figure 17, MAC-hs has four main functions [3GPP_308]: flow control, scheduling and priority handling, hybrid ARQ functionality handling, and transport format and resource combination selection. On the other hand, MAC-hs has slightly different functions at the UE side. These functions are HARQ handling, reordering, and de-assembly.

The output of the MAC-hs is the HS-DSCH channel. Furthermore, a HS-DSCH channel is associated to a downlink signaling channel and to an uplink signaling channel, as depicted in Figure 17. Besides of these channels, HS-DSCH channel needs to have a DCH associated channel to carry user and system information in the uplink.

3.5.2 Channel Structure

HSDPA introduced only few new channels in 3GPP Release 5. Nonetheless, HSDPA has continued developing and new channels were added in the following release, 3GPP Release 6, such as for example F-DPCH.

HSDPA can only send user information in the downlink path. Therefore, HSDPA will need, besides its physical channels, an associated dedicated physical channel (DPCH) to provide a complete bi-directional interaction. These associated channels will carry signaling traffic or data traffic depending on the needs. This is depicted in Figure 17.

3.5.2.1 Transport Channels

Figure 7 shows the transport channels as well as the physical channels. All the transport channels depicted in Figure 7 have been presented before except for the High Speed Downlink Shared Channel (HS-DSCH). This is the specific transport channel tied to HSDPA.

Unlike DCH, which is a dedicated transport channel, HS-DSCH is a downlink channel which is shared by all UEs within a cell, i.e. HS-DSCH is a common transport channel. This transport channel can be mapped to either a High Speed Physical Downlink Shared Channel (HS-PDSCH) or to a High Speed Shared Control Channel (HS-SCCH) depending whether the transmitted information is user data or signaling information, respectively.

As stated before, a HS-DSCH channel has a DPCH channel as an associated channel. Next to this, a HS-DSCH channel will have associated one or several HS-SCCH physical channels.

3.5.2.2 Physical Channels

HSDPA defines three new different physical channels [3GPP_211], two for the downlink and one for the uplink. All physical HSDPA channels have a shorter frame structure than the traditional UMTS. While UMTS used, in general, a 10 ms channel frame, HSDPA channel frame uses a 2 ms TTI length.

3.5.2.2.1 High Speed Physical Downlink Shared Channel (HS-PDSCH)

HS-PDSCH is, actually, the physical channel used to carry the user data in the downlink. It does not, though, carry any signaling data. HS-PDSCH has a fix SF equal to 16, and it is not power controlled (see section 3.6.2). Variable SF and power control were the key to provide different bit rates in UMTS. To cope with this, HSDPA has brought another modulation scheme. Thus, HS-PDSCH can use two types of modulations: QPSK or 16-QAM.

Though HS-PDSCH has a SF equal 16, only 15 codes can be allocated for each scrambling code. This is due to the fact that one code is needed for the common channels HS-SCCH and HS-DPCCH. Nonetheless, the final number of codes used will be UE dependant. Proposed UEs can support 5, 10, or 15 codes.

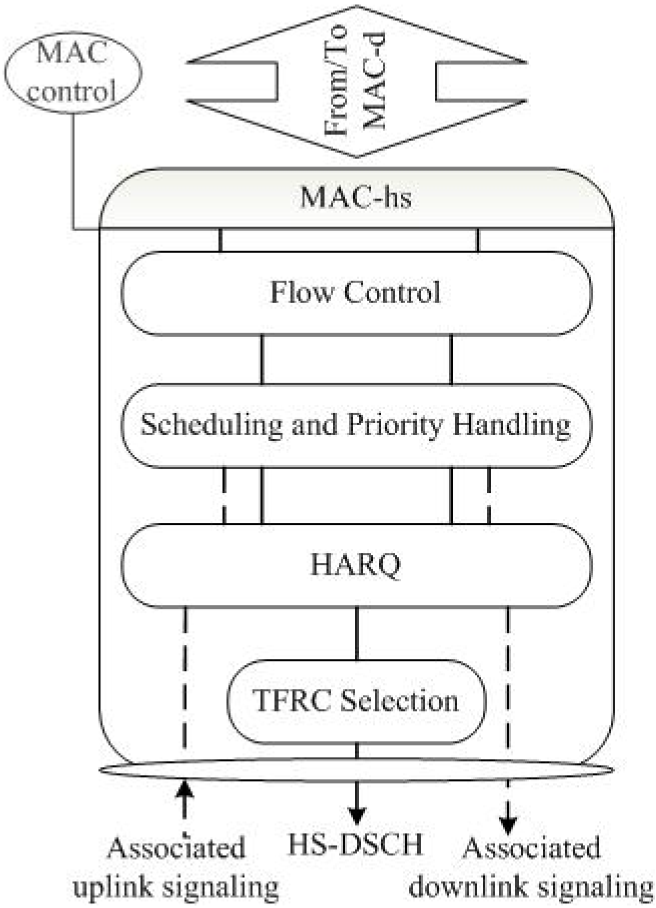

Figure 17 – MAC-hs architecture, UTRAN side

3.5.2.2.2 High Speed Shared Control Channel (HS-SCCH)

HS-SCCH is a downlink channel. This channel arrives 2 slots before the HS-PDSCH, as shown in Figure 18. HS-SCCH channel transports the signaling data associated to a transport channel. Basically, HS-SCCH indicates to the UE how to demodulate the HS-DSCH. HS-SCCH is divided into two parts. The first part, which is more critical, contains the UE identity, modulation used in the HS-DSCH, and coding information. This first part will provide enough information to the UE to decode the HS-PDSCH. The second part of the HS-SCCH transports less critical information, such as for example HARQ information, or redundancy information.

This physical channel uses a SF equal to 128, and the modulation scenario for this channel is QPSK. Unlike HS-PDSCH, HS-SCCH may be power controlled (see section 3.6.2). Nevertheless, this is a vendor specific solution.

Networks are configured with one HS-SCCH channel if HSDPA is time-multiplex based. However, if HSDPA is code-multiplex based, the network will need to be configured with more than one HS-SCCH channel. UEs, though, can monitor at most four HS-SCCH channels.

3.5.2.2.3 High Speed Dedicated Physical Control Channel (HS-DPCCH)

The HS-DPCCH channel is an uplink channel used for signaling purposes. The modulation scheme for this channel is BPSK and the spreading factor is 256; thus, this channel has a bit rate of 15 Kbps. Moreover, HS-DPCCH channel is power controlled.

The signaling information sent through this channel is related to ACK/NACK, and to CQIs. Thanks to this feedback, Node B will be able to perform link adaptation, scheduling decisions, and to retransmit erroneous information.

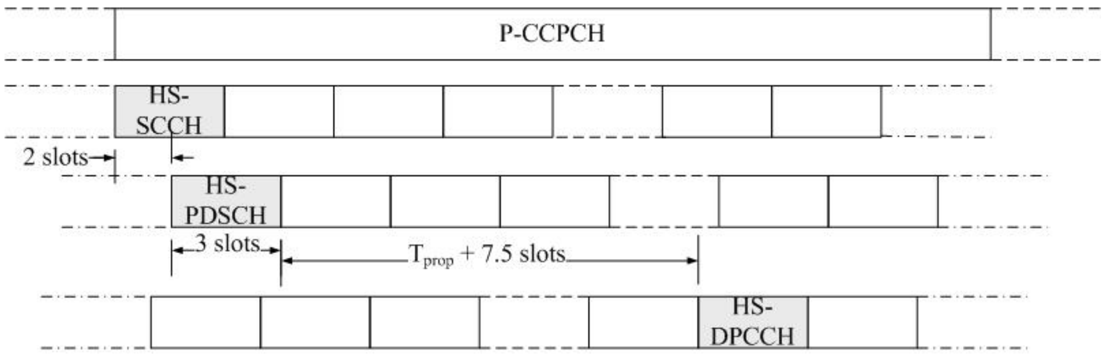

Figure 18 – Timing of HSDPA physical channels

3.6 Radio Resource Management

The previous sections have presented the most relevant HSDPA key features. Some of these new functionalities have required certain modifications in the radio resource management functionality. On the other hand, other radio resource management functionalities have also undergone slight modifications to cope with the new demands of the system. This section will briefly present other relevant modifications that other radio resource management functionalities have undergone.

3.6.1 Handover

HSDPA supports mobility procedures in a seamless way for the user. However, so far, only hard-handover is possible for HSDPA [3GPP_308]. The associated DCH channels, though, can still use soft or hard handover, depending on the situation.

Table 3 summarizes the different handover types applicable for HSDPA as well as their characteristics.

3.6.2 Power Control

As mentioned before in section 3.5.2.2, HSDPA introduced three new physical channels: HS-PDSCH, HS-SCCH, and HS-DPCCH. The HS-PDSCH is not fast power controlled; therefore HS-PDSCH power is kept constant during the HS-DSCH channel frame.

Nonetheless, there is a power adjustment algorithm which is performed in each certain interval to adjust the HS-PDSCH power level to the current conditions. RNC will command Node B to perform this power changes. Link adaptation algorithms are used instead to replace the fast power control functionality.

The HS-SCCH is, on the other hand, fast power controlled. Moreover, Node B will be in charge of the power control functionality of this physical channel. 3GPP has not specified these power control algorithms, so they are vendor specific. Yet, power control algorithms for HSDPA are likely to use and take into account the associated DPCCH power control commands and the CQIs. Furthermore, these algorithms are supposed to follow the technique used for Release 99 which was explained in section 2.6.2, i.e. the use of an inner and outer loop. The associated DPCCH power control commands or the CQIs will be the input for the inner loop; while the outer loop will fine tune the inner loop in order to meet the required block error probability. Fine-tuning will be performed based on the ACK/NACK/DTX [Hol_06].

	Intra Node B HS-DSCH to HS-DSCH	**Inter Node B HS-DSCH to HS-DSCH**	**HS-DSCH to DCH**
Handover measurement	Usually performed by UEs but Node B scheme is also possible.		
Handover decision	SRNC.		
Packet retransmission	Packets forwarded from source MAC-hs to target MAC-hs through the Iub interface.	No forwarding. RLC retransmissions used from SRNC.	RLC retransmissions used from SRNC.
Packet losses	No.	No packet losses if RLC AM is used or duplicate packet are sent in RLC UM.	No packet losses if RLC AM are used.
Uplink HS-DPCCH	HS-DPCCH can be in softer handover.	HS-DPCCH received by one cell.	

Table 3 – HSDPA handover types characteristics [Hol_06]

3.6.3 Admission and Congestion Control

Admission control mechanisms have to take into account the fact that HSDPA also needs certain amount of power. How much power is allocated to HSDPA is also a vendor specific decision. Nevertheless, there are mainly two scenarios for HSDPA power allocation. Either a fixed power is allocated to HSDPA or the unused power in Node B is allocated to HSDPA, i.e. dynamic power allocation. Admission control algorithms as well as load (congestion) control algorithms will need to be adapted to the selected power allocation mechanism.

Figure 19 presents the carrier power breakdown which will help to present an admission control example. Basically, a new HSDPA user would be admitted if [Lai_06]:

$$P_{tx} \leq P_{txTargetHSDPA}$$

Congestion control algorithms will try to prevent to arrive to the Pmax available in Node B. A congestion control mechanism could halt some users' transmissions if [Lai_06]:

$$P_{txNonHSDPA} \geq P_{txTargetHSDPA} + P_{txOffsetHSDPA}$$

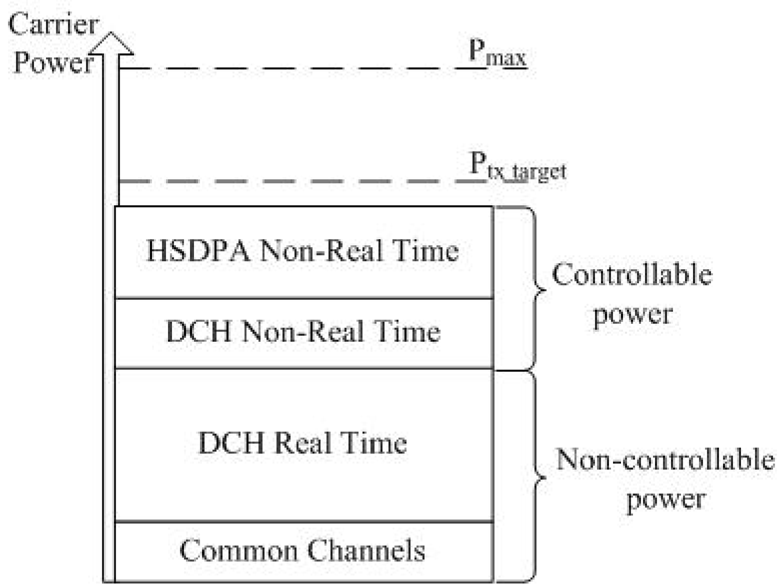

Figure 19 – Carrier power breakdown

Once the first equation is met, halted users will be again scheduled for transmission.

Chapter 4

Simulation Study

4.1 Problem Statement

As briefly presented in section 1.2, there are very few real commercial deployed HSDPA networks and very little HSDPA traffic. Hence, it is not possible to measure the performance and behavior of these networks nor the end-user performance. Next to this, HSDPA network performance and behavior will depend on the algorithms used to implement HSDPA link adaptation, scheduling, power control, and other radio resource control mechanisms. As mentioned in Chapter 3, these algorithms are vendor specific. No algorithm is standardized and hence HSDPA network performance and behavior will also be vendor specific.

There has been an intensive research on HSDPA and there are a large amount of research papers which present different algorithms to implement different HSDPA functionalities. Many of these papers have been already mentioned in Chapter 3, and others are mentioned in the following section. Yet, the problem remains. Those solutions are specific for the implemented algorithms and for the simulated environment. Nonetheless, these solutions are very useful for operators, providers, and manufacturers which can gain a deep knowledge about HSDPA in different situations.

Networks have to be dimensioned and planned according to the expected type of traffic which is supposed to be carried. In the first stages, HSDPA will be used for interactive and background traffic. Traffic patterns as well as user patterns are important to obtain reliable planning and dimensioning results. Obviously, a bad traffic/user pattern will lead to bad results. Next to that, a network cannot be planned or dimensioned without taking into account relevant network performance values, i.e. KPIs, and the end-user performance. On the top of this, networks have to be dimensioned and planned taking into account how the data will flow to the UEs. HSDPA uses a shared channel to send data to all UEs; hence, the amount of users in the network will affect the network performance and end-user performance.

How do the network performance and end-user performance vary when certain traffic parameters change or when the traffic pattern itself changes? How do the network performance and end-user performance vary when the number of users changes? Which is the number of multiple simultaneous users when a certain network performance and end-user performance is required? These are the questions which are not yet answered. This thesis will find some answers for these questions. These answers are needed by operators, providers, and manufacturers to plan and dimension their networks correctly.

4.2 Related Studies

The closest research paper to the purpose of this thesis is [Sha_03]. [Sha_03] presents a method to find the number of interactive data users which can fit in a wireless network. Furthermore, it presents the perceived average delay as the number of UEs increases. Finally, it also shows the perceived throughput as the equivalent throughput that a user would have obtained using a dedicated line instead of a shared channel. Though [Sha_03] is a good reference to base this thesis, it was difficult to reproduce the simulation model as well as the actual implementation of the simulation model. Thus, this paper was discarded as a base for this thesis.

[Sol_05] presents in his PhD dissertation a detailed QoS management description and introduces a simulator to achieve its purpose. This simulator is only for UMTS system and it is implemented with great detail. [Sol_05] provided with good ideas for this thesis; however, the implementation was far too complex.

[Lov_04] compares the performance of HSDPA and UMTS as described in Release 99. The last simulation case of this document is in the line of our research; however it fails to take into consideration KPIs or QoE parameters. Besides, it contemplates certain factors which this thesis is not dealing with.

Most of the research documents, such as [Boa_03], [Wan_04], [Nas_05], [Ped_04], [Ped_06] or any other previously mentioned HSDPA research paper, fail to take into account KPIs or the end-user performance. They do not aim, though, to find the end-user performance or the amount of multiple simultaneous users which a network can support providing certain QoS. Instead, these documents have other purposes and hence they use different methods.

A second issue is the traffic patterns to be used for browsing traffic. There is a vast amount of papers arguing about web traffic patterns, such as [Abr_00], [Cao_02], [Cho_99], [Kle_01], [Hal_06], [Her_03], [Tya_03], or [Vid_00] among others.

This thesis does not aim to discuss traffic/user patterns; it will rather select one traffic pattern. Though, [ETSI_112] specifies a traffic pattern used in [Sol_05] and [Tur_04]; the traffic pattern selected is the one specified by [3GPP_892]. This traffic pattern is used and supported by [3GPP2_02], [Gal_05], [Kha_04], [Nag_03], and [Van_04].

4.3 Research Solution: HSDPA Simulator

In order to answer the questions arisen in section 4.1, an HSDPA simulator will be built. As already seen in Chapter 3, HSDPA features implementation is vendor specific; hence, the difficulty of building an accurate simulator is far complicated and it would need further research to build it. Instead, the built simulator emulates in a simple way the network and the traffic. The HSDPA simulator works as described in Figure 21. Traffic is created according to [3GPP_892]. This pattern has been slightly modified with the patterns used in [Sol_05]. The modified pattern can be found in Appendix A in this thesis. The main traffic concepts are explained in the following paragraphs and they are depicted in Figure 20.

A session is defined as a set of pages that the user visits. The number of pages which the user will visit within a session is defined by the component 'session length'. Furthermore, 'session length' is calculated for every session using an inverse Gaussian distribution. Hence, 'session length' will be different for every session and for every user.

A page is made up of a main object and several embedded objects which can have different sizes. 'Main object size' as well as 'embedded object size' and 'number of embedded objects per page' will vary for each page and for every user. Sizes follow a truncated log-normal distribution while the number of embedded objects follows a truncated Pareto distribution.

Once the main object and the embedded objects have been downloaded, the user will spend certain amount of time reading the information before he/she moves to the next page. 'Reading time' follows an exponential distribution. This time is also calculated every time a new page is requested.

Finally, session inter-arrival time defines the time between two different sessions. 'Session inter-arrival time' follows an exponential distribution and it is calculated every time a session starts.

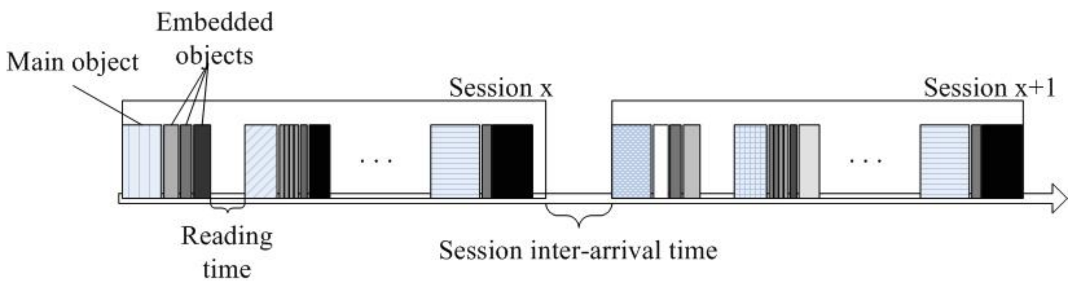

Figure 20 – Web browsing traffic parameters

UEs can have either ON or OFF status. A user has ON status when it is downloading information. On the other hand, OFF status refers to the situation when the user is not downloading any information, such as for example when the user is reading a web page or when the user is deciding for what to look next.

The available bit rate is apportioned among all ON users. Then, the downloaded information is calculated and the status for every UE is updated. This process continues for the simulating time.

This simulator has clear simplifications. To start with, the air interface and its effects are omitted since it is supposed that air conditions are similar for all users. Air conditions can be supposed similar when they are averaged during a long period of time. Obviously, instantaneous behavior will be different. Next to this, no link adaptation algorithm has been implemented. Node B, though, will fully use the assigned bandwidth, injecting as much information as possible. The simulator implements neither a real packet scheduler nor a hybrid automatic repeat request algorithm. These assumptions are reasonable if all the UEs have similar average radio conditions and similar capabilities. In this situation and due to the fact that HSDPA shares the physical channel among all the users, the available HSDPA bandwidth (and HSDPA resources) would be shared equally among all the users. Since it is considered that all UEs have similar radio conditions, different simulations will be performed for different mean cell throughput values. Mean cell throughput will be an indicator of the radio environment. Depending on the radio conditions, the UEs throughput will vary and hence the cell throughput. Good radio conditions will be tied to high mean cell throughput, while bad radio conditions will be tied to low mean cell throughput.

Next to this, admission and congestion control mechanisms are not necessary since it is supposed that the network can allocate enough resources for all ON HSDPA UEs. This means, for example, that the network has unlimited buffers' size, unlimited HSDPA power, or unlimited number of codes. Furthermore, no power control mechanisms are implemented either (see section 3.6.2).

Though more accurate results could be found if all HSDPA features would have been implemented, the simulator complexity would have increased heavily; thus, been out of reach of this thesis. This thesis is not looking for exact numbers (as it is extremely difficult because of the probabilistic nature of the user behavior, traffic patterns or, air propagation characteristics) but it roughly assesses the end-user performance and it let us know the amount of simultaneous users the network could support if there are no other hardware/software limitations, such as for example power limitations, or admission limitations.

These assumptions could lead to optimistic results; however, this point should be studied in real networks.

4.3.1 HSDPA Simulator: Parameters

First, traffic parameters need to be set. These parameters can be found in Appendix A. The mean session inter-arrival time will be set to 0 (i.e. one session starts after another) when the maximum number of multiple simultaneous users wants to be found. Nonetheless, this value will be modified to investigate how the session inter-arrival value affects the end-user performance. This will take place in the third main simulation in which the mean session inter-arrival time will have the following values: 0, 5, 10, 15, 30, 45, and 60 seconds. These values are mean values. Real values will be generated according to the distribution defined for the session inter-arrival time which can be found in Appendix A.

Each simulation will have a simulation time set to twenty hours (virtual time). This value has been chosen in order to collect enough samples and, as a result, achieve more accurate results. Simulations also vary the mean cell throughput value. Field measurements[1] have shown that typical mean cell throughputs are 600 Kbps, 800 Kbps, 1000 Kbps, 1200 Kbps, and 1400 Kbps when UEs with category 12 were used (category 12 only supports QPSK modulation, and a maximum number of 5 codes). These values will be used as an input for the simulations and they will set the maximum possible bit rate for a single user.

In order to find the maximum value of users who could fit, the simulator will iterate until certain QoS/QoE is achieved. For web browsing applications, the QoS/QoE is measured as the time used to download a web page. This is what the end-user cares about: time used to download a web page. Hence, the simulator will iterate until 90 % of the sessions of 90 % of the UEs have a "User's session download time" (see Equation 2, page 47) below four seconds. Four seconds is the maximum download time recommended in both [3GPP_105] and [ITU_010] for web browsing applications.

Once the maximum number of UEs has been found for the different mean cell throughputs, different values for the number of UEs will be used to perform other simulations.

4.3.2 Simulations

Three main simulations will be performed. First, the maximum number of multiple simultaneous users which the network can support will be found. "User's session download time" (see Equation 2) distribution, "User's session throughput" (see Equation 1) distribution, "Total mean session download time" (see Equation 6) and "Total mean session throughput" (see Equation 5) for different throughputs will be plotted. Different mean cell throughputs (600 Kbps, 800 Kbps, 1 Mbps, 1.2 Mbps, and 1.4 Mbps) will be simulated, and the maximum number of multiple simultaneous users for each mean cell throughput will be found. The benchmark performance to find the maximum number of users is, as mentioned in the previous section, that 90 % of the sessions of 90 % of the users experience a "User's session download time" (see Equation 2) below four seconds. Statistically, if the simulation is run long enough, all users will have similar amount of sessions. Then, the benchmark performance would be that over 81 % of the total amount of sessions has to have a "User's session download time" below 4 seconds. The maximum number of UEs can be also seen as the optimum value. This is the operator point of view: how to maximize the network usage, and at the same time offer to the end-user a fair enough QoS/QoE. Once the maximum number of users is known, a second main simulation will be performed. In this second simulation, the number of users will be modified. This simulation will be run using a mean cell throughput equal to 1 Mbps and different numbers of UE. The values used are 20, 40, 60, 81, 100, and 120 UEs. "Total mean session download time" and "Total mean session throughput" versus the number of UEs will be the main output.

[1] Nokia specific field measurements.

Finally, the third main simulation will take into account the mean session inter-arrival time. Simulated mean session inter-arrival values were mentioned in the previous section. The simulation will be run for two different numbers of UEs: 81 and 120 UEs.

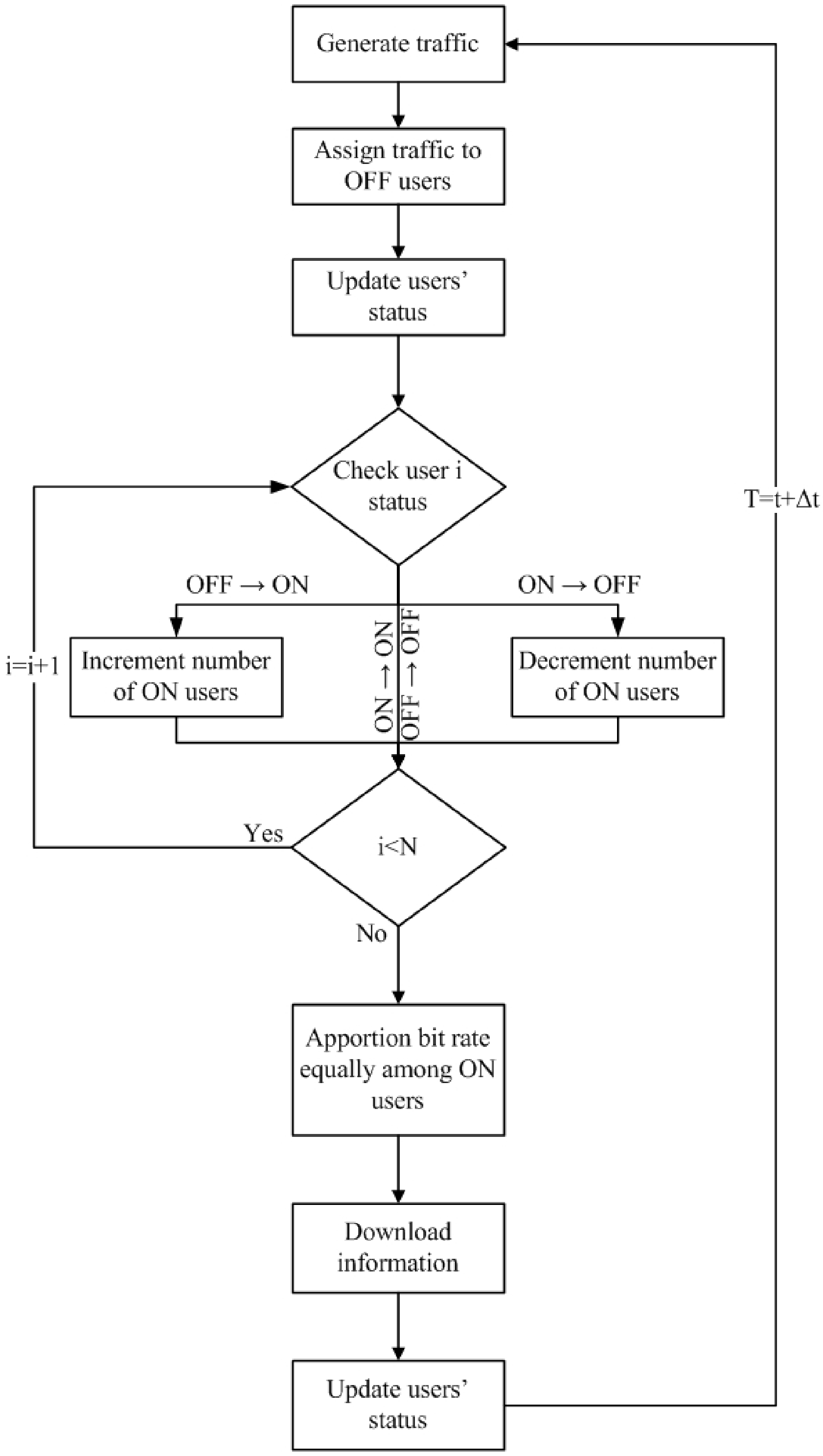

Figure 21 – HSDPA simulator flowchart

Chapter 5

Simulation Results

Before showing the results, some concepts need to be introduced and explained to understand the results.

"User's session throughput" is the mean of all page throughputs within session *j* of UE *k*.

$$\text{user's session throughput}_{j,k} = \frac{\sum_{i=1}^{n} page\_throughput_{i,j,k}}{n}$$

Equation 1 – "User's session throughput"

"User's session download time" is the mean of all page download times within session *j* of UE *k*.

$$\text{user's session download time}_{j,k} = \frac{\sum_{i=1}^{n} page\_download\_time_{i,j,k}}{n}$$

Equation 2 – "User's session download time"

"User's mean session throughput" is the mean of all "user's session throughputs" for UE *k*.

$$\text{user's mean session throughput}_{k} = \frac{\sum_{j=1}^{m_k} \frac{\sum_{i=1}^{n} page\_throughput_{i,j,k}}{n}}{m_k}$$

Equation 3 – "User's mean session throughput"

"User's mean session download time" is the mean of all "user's session download times" for UE *k*.

$$\text{user's mean session download time}_k = \frac{\sum_{j=1}^{m_k} \frac{\sum_{i=1}^{n} page\_download\_time_{i,j,k}}{n}}{m_k}$$

Equation 4 – "User's mean session download time"

"Total mean session throughput" is the mean of all "user's mean session throughputs" taking into consideration all UEs.

$$\text{total mean session throughput} = \frac{\sum_{k=1}^{p} \frac{\sum_{j=1}^{m_k} \frac{\sum_{i=1}^{n} page\_throughput_{i,j,k}}{n}}{m_k}}{p}$$

Equation 5 – "Total mean session throughput"

"Total mean session download time" is the mean of all "user's mean session download times" taking into consideration all UEs

$$\text{total mean session download time} = \frac{\sum_{k=1}^{p} \frac{\sum_{j=1}^{m_k} \frac{\sum_{i=1}^{n} page\_download\_time_{i,j,k}}{n}}{m_k}}{p}$$

$p \equiv \text{maximum number of UEs}$

$m_k \equiv \text{number of sessions of } \mathrm{UE}_k$

$n \equiv \text{number of pages in the session } m_j \text{ of the } \mathrm{UE}_k$

$page\_download\_time_{i,j,k} \equiv \text{download time of page i of session } \mathrm{m}_j \text{ of } \mathrm{UE}_k$

Equation 6 – "Total mean session download time"

5.1 Maximum Number of UEs

The first simulation used a mean cell throughput of 600 Kbps. In this situation, only 42 UEs could fit into the network. "User's mean session download time" varies between 1 and 2 seconds and “User’s mean session throughput” varies between 200 and 300 Kbps for all the users. “Total mean session throughput” is 286 Kbps, and the “Total mean session download time” is 1.70 seconds.

Additional information about the simulation results can be found in Appendix B. Table 4 contains the representative values for this section. The total number of completed sessions and downloaded pages are presented as well as the mean number of sessions per UE.

These values are the results of simulating until more than 90 % of the sessions of 90 % of the UEs “User’s session download time” less than 4 seconds. In other words, if there are 100 UEs, 90 UEs should have 90 % of the “User’s session download time” values below 4 seconds; i.e. 81 % of the total sessions should have a “User’s session download time” below 4 seconds.

Figure 22 and Figure 23 depicts the “User’s session download time” and “User’s session throughput” distribution.

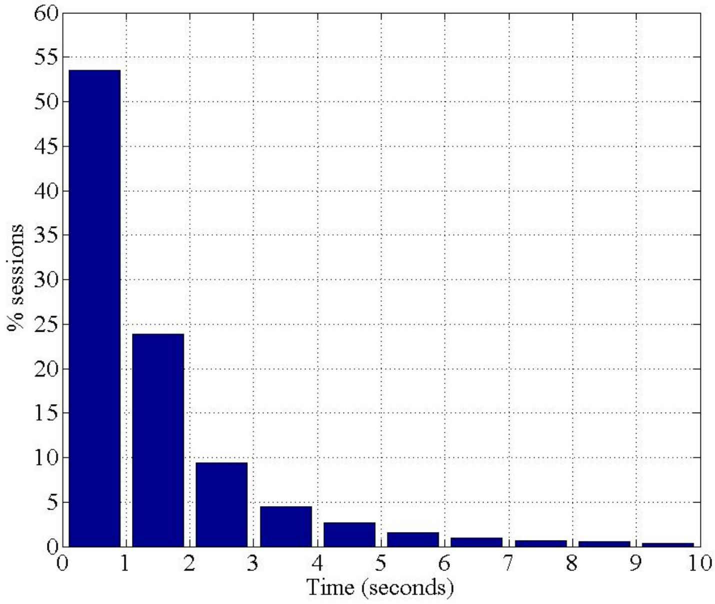

Figure 22 – “User’s session download time” distribution when cell mean throughput 600 Kbps

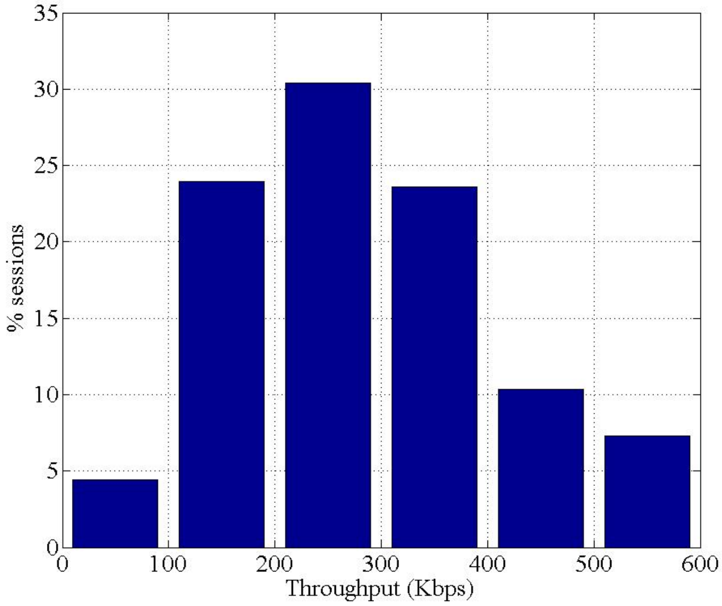

Figure 23 – "User's session throughput" distribution when cell mean throughput 600 Kbps

Mean cell throughput was increased to 800 Kbps. In this case, the maximum number of UEs that the network can support is 62. "User's mean session download time" varies between 1 and 2 seconds and "User's mean session throughput" between 300 and 400 Kbps for all the users. "Total mean session throughput" is 312 Kbps, and "Total mean session download time" is 1.71 seconds. Figure 24 and Figure 25 shows "User's session download time" and "User's session throughput" distributions.

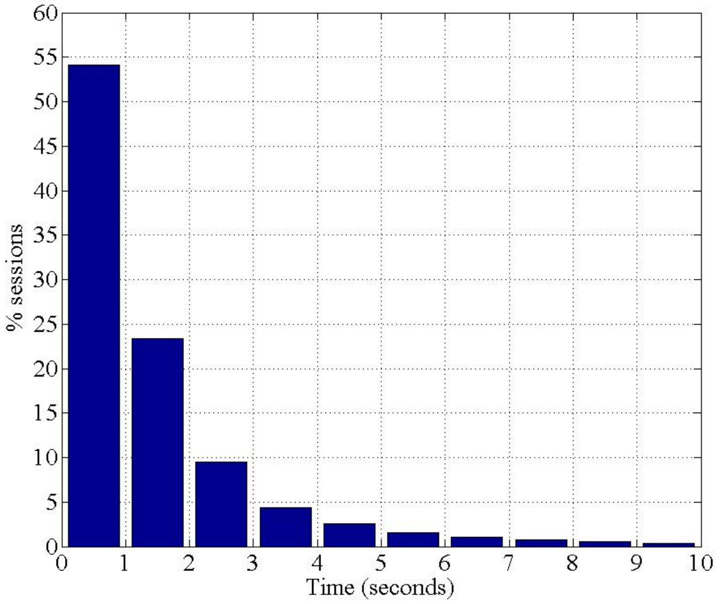

Figure 24 – "User's session download time" distribution when cell mean throughput 800 Kbps

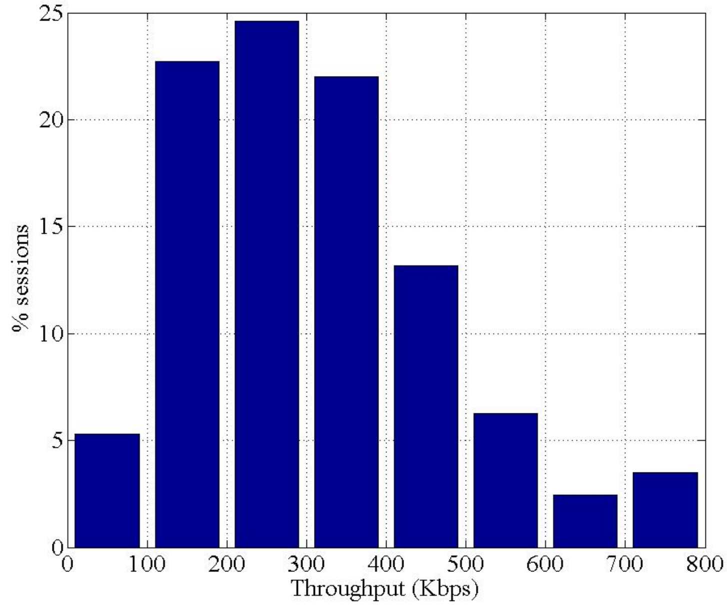

Figure 25 – "User's session throughput" distribution when cell mean throughput 800 Kbps

The third simulation took as a mean cell throughput 1 Mbps. The simulator obtained that the maximum number of UEs was 81. All the UEs experienced a "User's mean session download time" between 1 and 2 seconds. On the other hand, the experienced "User's mean session throughput" for all the users was between 300 and 400 Kbps. The obtained "Total mean session throughput" value was 338 Kbps and "Total mean session download time" was 1.68 seconds. Figure 26 and Figure 27 describes the "User's session download time" and "User's session throughput" distributions.

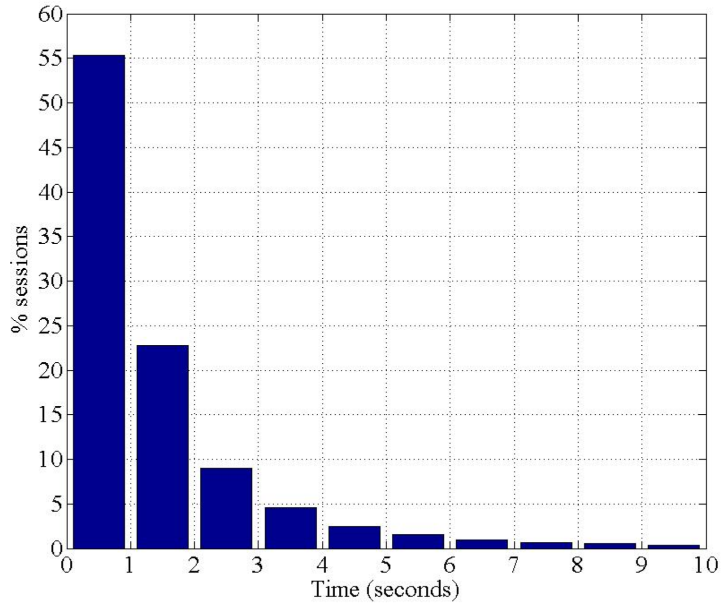

Figure 26 – "User's session download time" distribution when cell mean throughput 1 Mbps

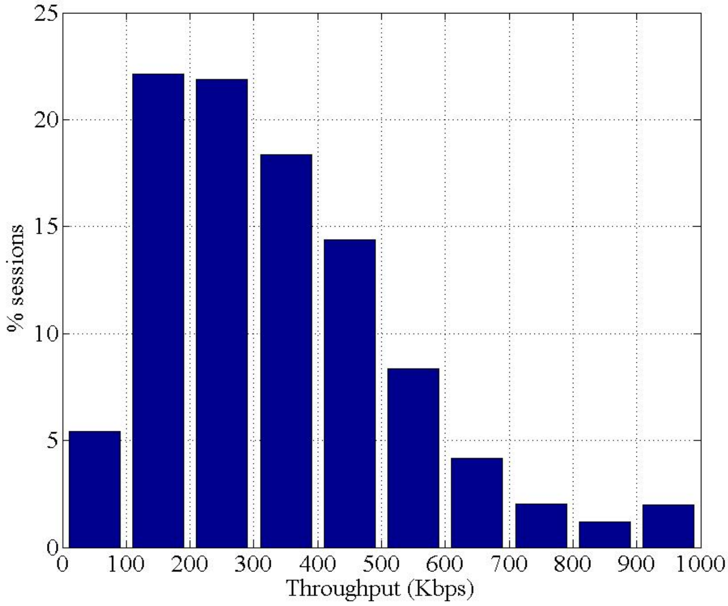

Figure 27 – "User's session throughput" distribution when cell mean throughput 1 Mbps

The following simulation in which the mean cell throughput was 1.2 Mbps returned 101 as the maximum number of users. "Total mean session throughput" was 349 Kbps, and "Total mean session download time" was 1.70 seconds. "User's mean session download time" continues being for all UEs between 1 and 2 seconds, and "User's mean session throughput" between 300 and 400 Kbps. "User's session download time" and "User's session throughput" distributions are depicted in Figure 28 and Figure 29 respectively.

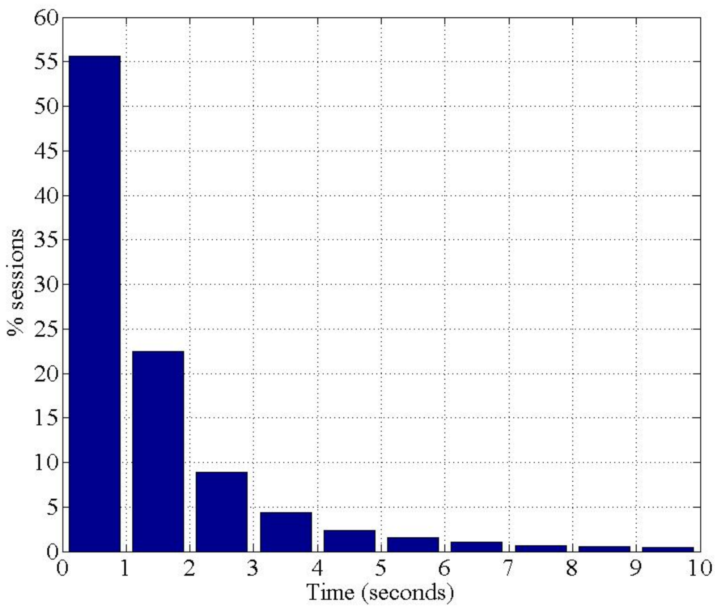

Figure 28 – "User's session download time" distribution when cell throughput 1.2 Mbps

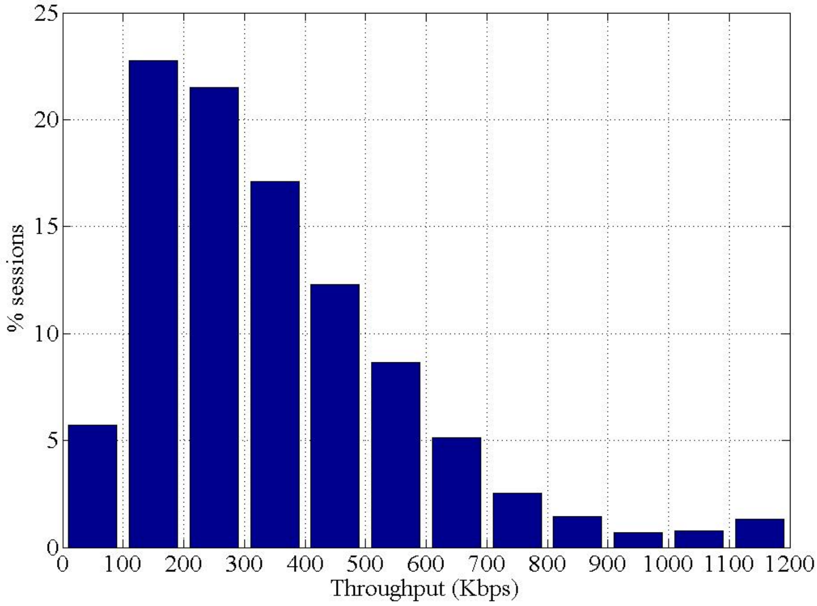

Figure 29 – "User's session throughput" distribution when cell mean throughput 1.2 Mbps

Finally, 1400 Kbps was used as a mean cell throughput. In this last scenario, the maximum number of multiple simultaneous users was 120. "Total mean session throughput" was 365 Kbps, and "Total mean session download time" was 1.70 seconds. As before, "User's mean session download time" is found between 1 and 2 seconds and "User's mean session throughput" between 300 and 400 Kbps. "User's session download time" and "User's session throughput" are shown in the next two figures.

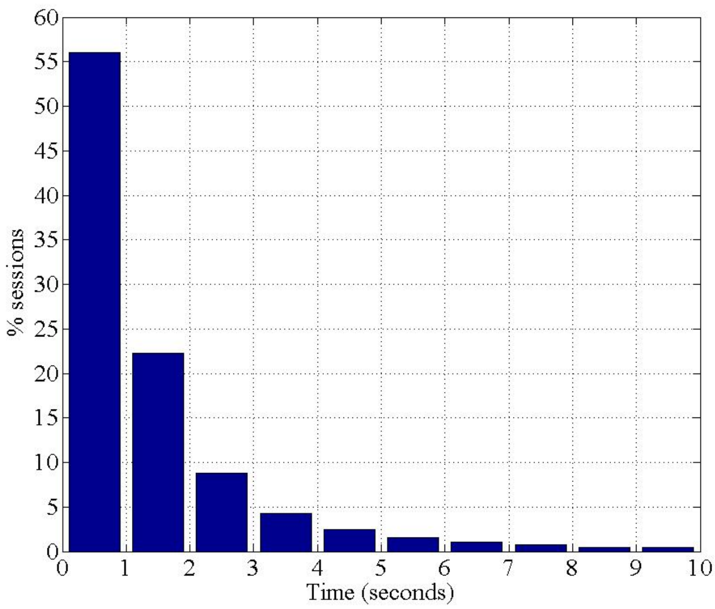

Figure 30 – "User's session download time" distribution when mean cell throughput 1.4 Mbps

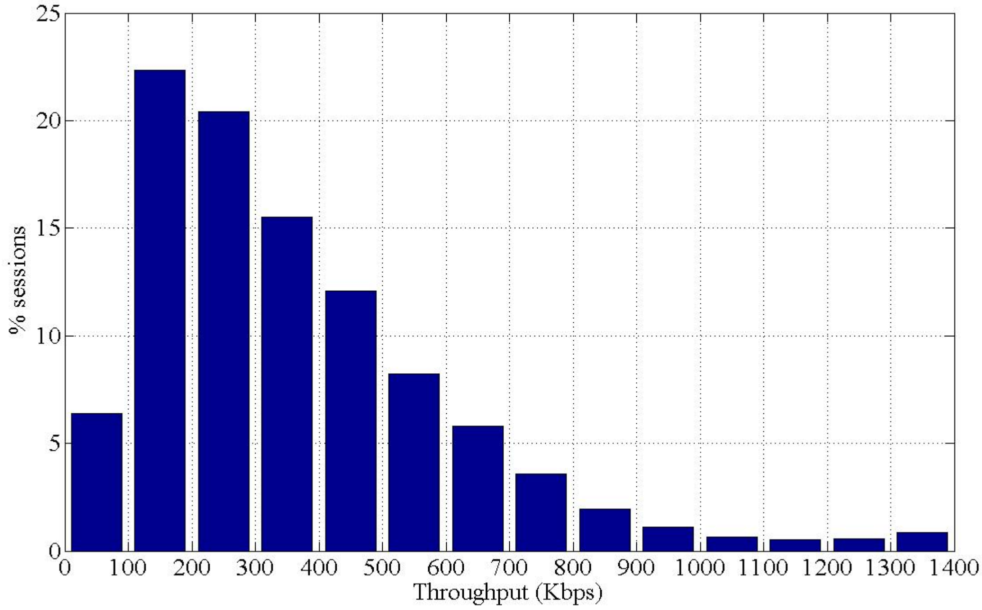

Figure 31 – "User's session throughput" distribution when cell mean throughput 1.4 Mbps

From the previous results, we can observe a slight increase in the "Total mean session throughput" value when the mean cell throughput changed. This is depicted in Figure 32. "Total mean session throughput" value experienced a slow and quasi-linear increased with the mean cell throughput. This means that the end-users' QoE is improved when the mean cell throughput increases even though the number of UEs also increases.

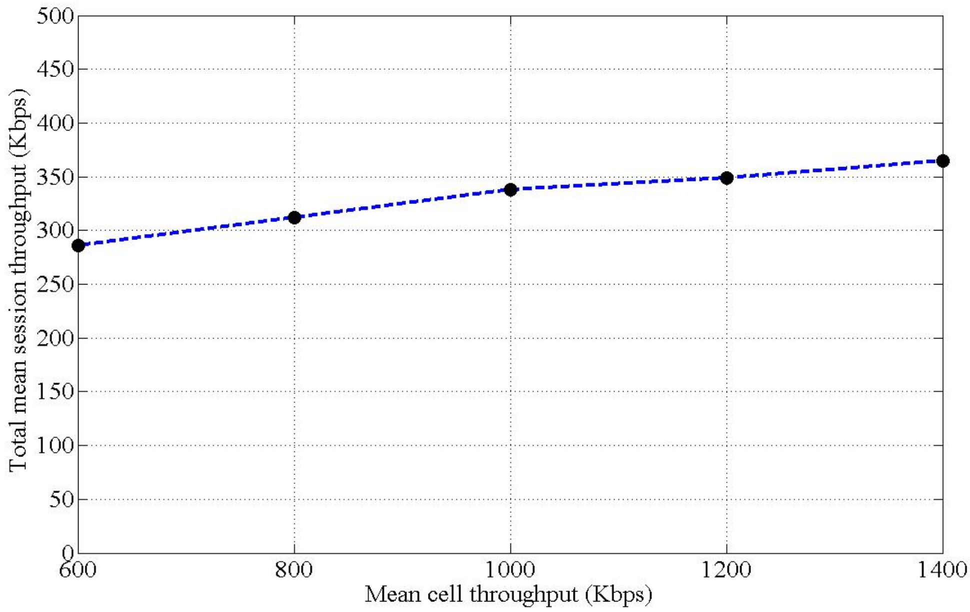

Figure 32 – Mean cell throughput vs. "Total mean session throughput"

Figure 33 shows how the number of users increases as the available mean cell throughput increases. We can observe from Figure 33 that the number of maximum multiple simultaneous users increases linearly.

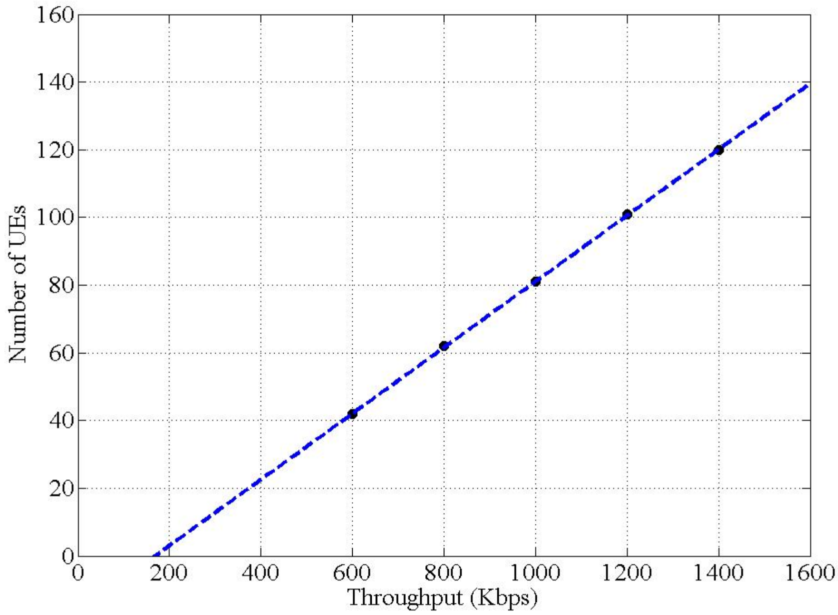

Figure 33 – Maximum number of simultaneous users versus mean cell throughput

5.2 Effect of Amount of UEs

This section shows how the number of UEs affects the “Total mean session download time” and “Total mean session throughput”. The simulated number of UEs was 20, 40 60, 81 (optimum, as defined in section 4.3.2), 100 and 120. Besides, the mean session inter-arrival time was set to 0, and the mean cell throughput was set to 1 Mbps. Results are depicted in Figure 34 and Figure 35.

When the network had 120 UEs, only 36 % of the sessions had a “user’s session download time” below 4 seconds. This value is far from the 81 % of the sessions set as a threshold in the previous section. When the number of UEs was reduced to 100, around 70 % of the sessions had a “user’s session download time” below 4 seconds. When the number of UEs was below 60, over 95 % of the sessions had a “user’s session download time” below four seconds.

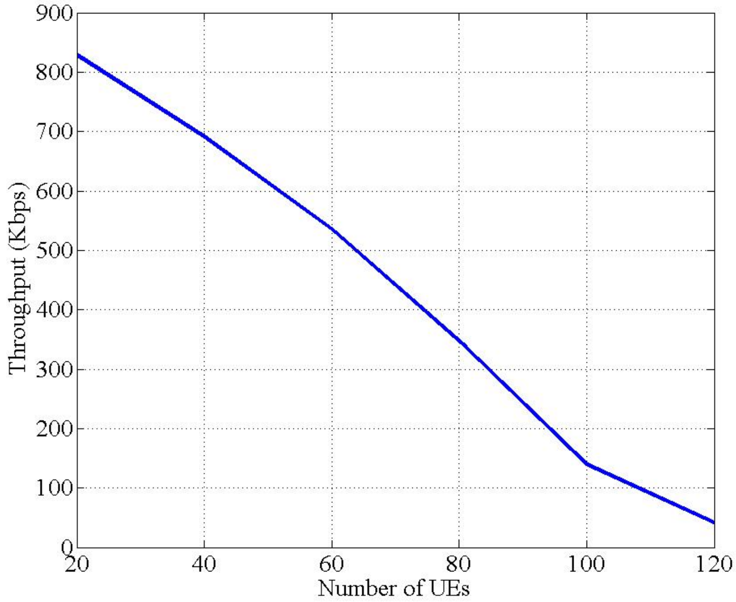

Figure 34 – "Total mean session throughput" vs. number of UEs

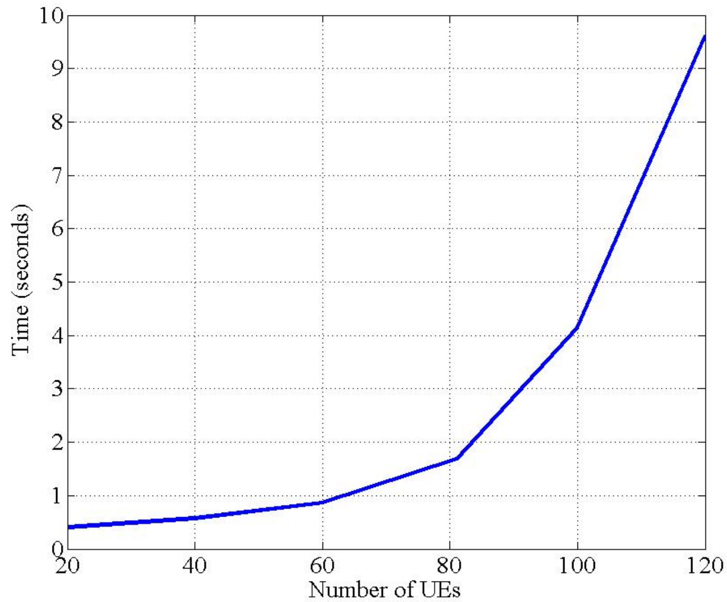

Figure 35 – "Total mean session download time" vs. number of UEs

We can clearly observe that "Total mean session throughput" decreases linearly until the optimum (as defined in section 4.3.2) number of UEs is reached. In the same way, the "total mean session download time" increases linearly. Once the optimum number of UEs is overcome, "Total mean session download time" increases rapidly. Table 5 in Appendix B presents the most representative values for this section.

5.3 Effect of Session Inter-arrival Time

In this section, the number of UEs was kept constant as well as the mean cell throughput, which was set to 1 Mbps. The mean session inter-arrival time (λ is the inverse of the mean session inter-arrival time) was modified to observe how this value affects the "User's mean session download time" and "User's mean session throughput" distribution values. The number of UEs used for the simulations was 81 (maximum number of UEs obtained in section 5.1), and a second arbitrary number of UEs set to 120.

"User's session download time" and "User's session throughput" distributions are represented in Figure 36 and Figure 37 when the optimum number of UEs was used. The same distributions are depicted in Figure 38 and Figure 39 when the number of UEs was 120.

From Figure 36, we can observe that 90 % of the "User's session throughputs" had a throughput below 600 Kbps when the mean session inter-arrival time was set to 0, i.e. one session starts immediately after another session. On the other hand, when the mean session inter-arrival time was set to 60 seconds, less than 50 % of the "User's session throughputs" had a throughput below 600 Kbps.

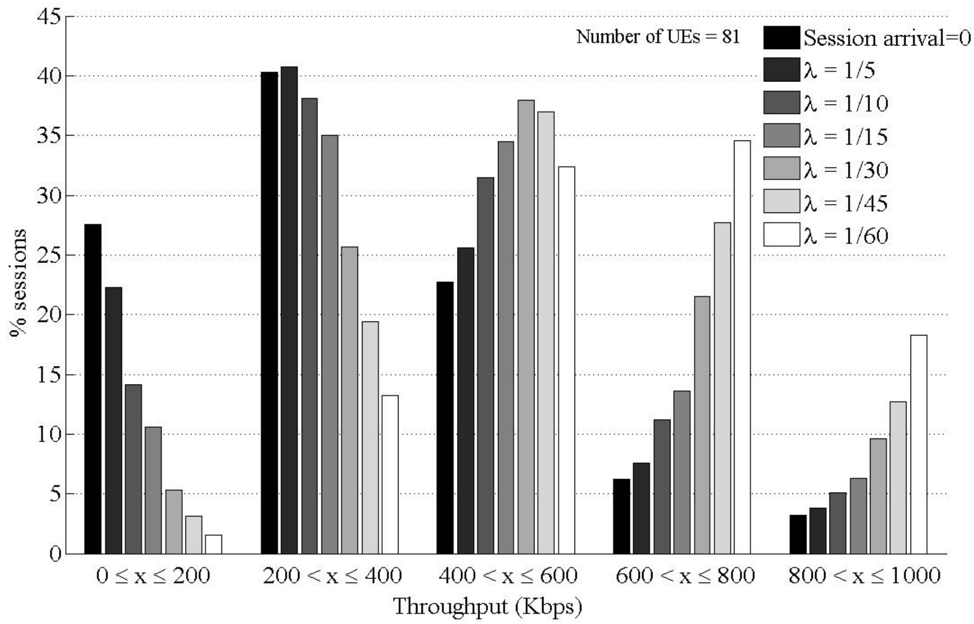

Figure 36 – "User's session throughput" distribution for different λ, 81 users

Figure 37 shows how the amount of sessions which have a "User's session download time" above 4 seconds decreases when the mean session inter-arrival time increases. When the number of UEs was set to 81 and the mean session inter-arrival time was 0, around 81 % of the sessions had a page download time below 4 seconds. However, when the mean session inter-arrival time was increased up to 60 seconds, 99 % of the sessions had a page download time below 4 seconds.

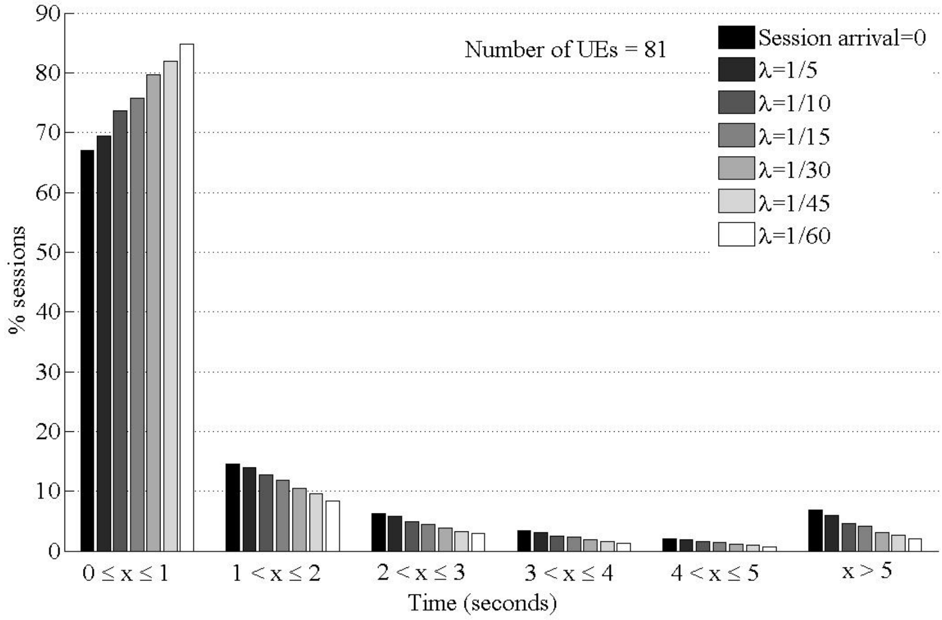

Figure 37 – "User's session download time" distribution for different λ, 81 users

Figure 38 shows the "User's session throughput" distribution when the number of UEs is increased up to 120. When the number of UEs was increased over the optimum value and the session inter-arrival time was 0, close to 100 % of the sessions had a "User's session throughput" below 200 Kbps. However, we can observe how the percentages of UEs move to higher throughput ranges when the mean session inter-arrival time increases. When the mean inter-arrival time was increased to 60 seconds, only 20 % of the sessions had a throughput below 200 Kbps.

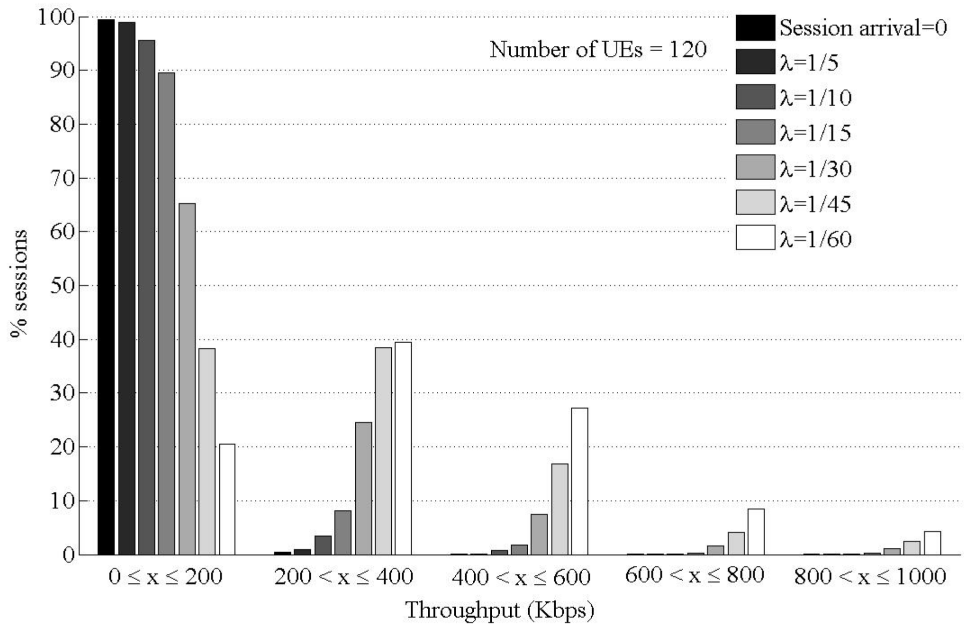

Figure 38 – "User's session throughput" distribution for different λ, 120 users

Figure 39 represents the "User's mean session download time" for different mean session inter-arrival times. We can observe how the distribution moves rapidly to values between 0 and 1 seconds when λ is smaller (i.e. higher mean session inter-arrival time). In this case, when the mean session inter-arrival time was set to 0, close to 50 % of the sessions had a page download time above 4 seconds. However, only 8 % of the sessions had a page download time above 4 seconds when the session inter-arrival time was set to 60 seconds.

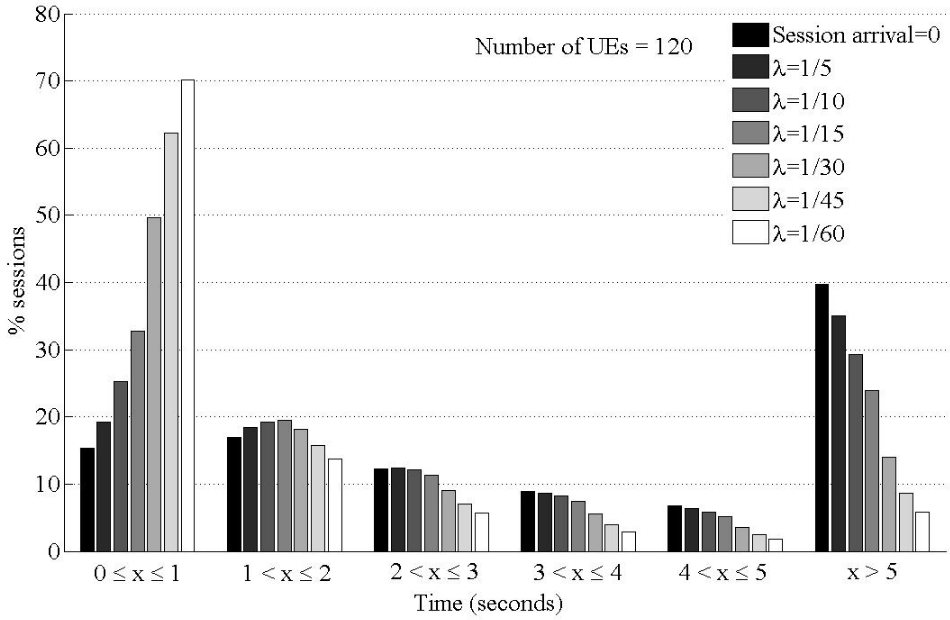

Figure 39 – "User's mean session download time" distribution for different λ, 120 users

Figure 40 and Figure 41 represents the "Total mean session download time" and "Total mean session throughput" values for different values of λ and for the simulated UE values. We can observe in Figure 40 how "Total mean session download time" decreases rapidly when the mean session inter-arrival time increases. This decreased becomes smoother when the "Total mean session download time" comes closer to the optimum value (which is around 1.70 seconds, value obtained in section 5.1). "Total mean session throughput" also increases rapidly and fairly linear when the mean session inter-arrival time increases.

We can conclude that the mean session inter-arrival time will be a rather important traffic parameter to optimize whether the number of UEs supported by the network is exceeded from the optimum value. It won't, though, affect so dramatically if the number of simultaneous UEs is below the maximum threshold; however, session inter-arrival time is a parameter to consider for dimensioning, planning, and optimizing HSDPA networks.

Table 6 and Table 7 in Appendix B contain the amount of samples (sessions and pages) for the simulated environments.

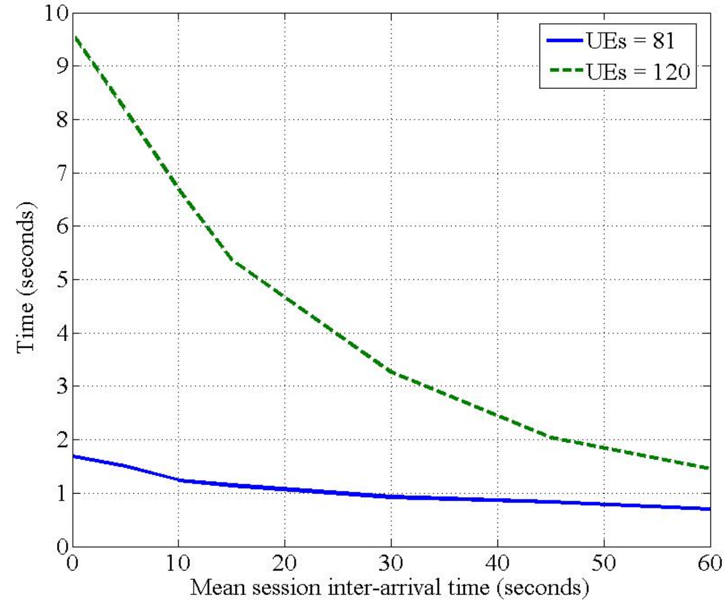

Figure 40 – "Total mean session download time" for different mean session inter-arrival times

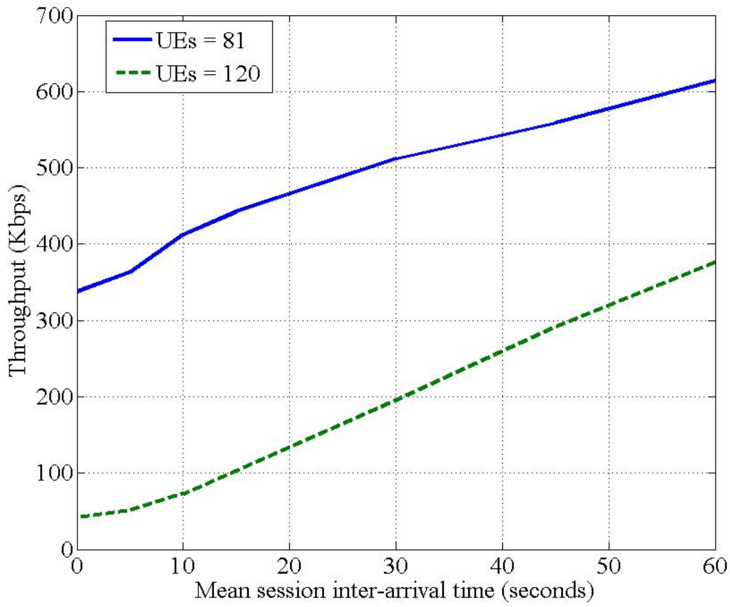

Figure 41 – "Total mean session throughput" for different mean session inter-arrival times

Chapter 6

Summary & Conclusions

6.1 Summary

This thesis has studied the effect of multiple simultaneous HSDPA users on the HSDPA end-user performance for non-real time services, such as web browsing, in a one cell system. First, the reader was put in the context. Then, the basic theory was explained. Chapter 2 and Chapter 3 went through UMTS technology, and HSDPA technology and novelty, respectively. Chapter 4 presented the research problem and the reasons for studying this problem. It was stated that one main reason for this research was due to the fact that HSDPA networks are not fully deployed and thus there is very little knowledge about how these networks behave for different types of traffic and which the effect on the end-user performance is. On the other hand, before a network is deployed, the information mentioned before is needed so the network can be planned and dimensioned efficiently. There is a real need to find a tool which helps to plan and dimension HSDPA networks before a HSDPA network is actually deployed.

Chapter 4 continued presenting supporting studies which could be very helpful for further research and for improving the solution presented in this thesis. Finally, the solution, a HSDPA simulator, was presented as well as the results which were achieved.

Chapter 5 showed the results of the different simulations. The maximum number of UEs which could fit in the network was the first output. This value depended on the mean cell throughput and, thus, several different mean cell throughputs were considered and simulated. Next to that, the effect of the amount of UEs on the end-user performance was also studied. All parameters were fixed except the number of UEs. The main output depicted the number of UEs versus the "User's session throughput" and "User's session download time". Finally, the effect of the mean session inter-arrival time was investigated. Results presented how the throughput as well as the download time varied when the mean session inter-arrival time changed.

6.2 Conclusions

Several conclusions can be achieved after studying the results presented in Chapter 5. The first finding is related to the amount of users versus the mean cell throughput. The number of UEs increases linearly when the mean cell throughput increases. The optimum number of UEs (from the operator point of view) was found for different typical mean cell throughputs. Results showed that for the optimum values, 100 % of the UEs achieved a session throughput ("User's mean session throughput") between 300 and 400 Kbps. Thus, any implemented scheduler should bring all UEs' sessions' values to that range. Next to that, we can say that, for the optimum number of UEs, the HSDPA shared channel is capable of performing close to a 512 Kbps ADSL dedicated line. At the session level, only around 25 % of the sessions will experience a "User's session throughput" under 200 Kbps. Furthermore, network performance and end-user experience slightly improve when the mean cell throughput increases even though the number of UEs increases too.

All these intermediate conclusions lead us to the first chunk of final conclusions.

- HSDPA seems to perform very well for web browsing applications.
- Hardware and software seems to be a bigger constraint than the air interface itself if the air interface can offer the average bit rates assumed in the study. Real implementations will have power limit constrains and admission control algorithms; timers and counters which can affect the network and user performance; and other hardware limitations such as the number of HSDPA users that hardware can support.

While the optimum number of UEs is not exceeded, the users will not experience high "Total mean session download time" value differences; however, once this number is exceeded, the end-user performance quality will worsen rapidly. The experienced "Total mean session download time" will increase close to an 'exponential' shape. We can conclude that:

- Until the optimum number of UEs, the network and user performance does not deteriorate significantly. Once the optimum number of UEs is exceeded, the QoE is not controllable.
- Thus, admission control algorithms will play an important role in the overall HSDPA network performance. These algorithms will cause a need to keep the amount of users below the maximum level.

Mean session inter-arrival time helps to increase the possible number of UEs which a cell can support or, if no more additional UEs come into the cell, the proportion of sessions with "User's session download time" below 1 second will increase. Nonetheless, the mean session inter-arrival time will not bring considerable changes in the end-user performance if the number of UEs is kept under the optimal value. Whether a non-optimal value is used, the mean session inter-arrival time will be decisive for the network performance and end-user performance. So,

- Mean session inter-arrival time will be critical for the network and end-user performance once the optimal number of UEs is exceeded, and it will considerably affect the network and end-user performance when the value of UEs is below the maximum. Besides, the mean session inter-arrival time will help to increase the amount of users that a cell can support. The session inter-arrival time is, hence, a rather critical parameter in order to increase the number of UEs that the network can support. Operators will need to fine-tune the web browsing traffic pattern they use according to the expected traffic characteristics in their networks.

6.3 Method Evaluation

Section 4.3 already mentioned some limitations of the implemented simulator. Here, limitations are seeing more accurately to provide input for further research.

The implemented simulator considers that the environment is ideal. No propagation models are implemented; thus, the radio propagation effects are omitted. In reality, radio propagation effects can deteriorate the quality of the signal significantly. As a result, UE throughput will vary according to the radio environment which also varies with the time in a probabilistic way. Whether good radio conditions are met, UEs will use a high level modulation; otherwise, lower level modulations will be used. Our simulator coped with the effect that UEs throughput varies with the time, in a simplistic way: it assumed a certain cell throughput which was shared equally by all UEs.

Our simulator considered that UEs are always connected to HSDPA; however, in real networks, it is likely that UEs will be dropped to DCH or FACH after some certain period of inactivity time. Whether the UE starts a new session or continues browsing after the thinking time period, the UE will need to request access to the HSDPA network again. This fact will lead to additional delays and hence the page download time will increase. Real implementations will drop to DCH those UEs to which HSDPA network access was not granted.

Next to that, the simulator does not implement any real TCP protocol and, thus, it does not take into account the TCP slow start process. Slow start will cause a delay in the page download time since the UE does not start downloading the first page at the maximum possible bit rate. Slow start depends on how fast the acknowledgements are sent and also on the RTT.

All these effects will have as a consequence a reduction in the maximum number of multiple simultaneous UEs which the network can support. It will also cause an increase in the page download times and, therefore, a decrease in the throughput. Consequently, the end-user performance will be deteriorated.

6.4 Further Research

There are two natural steps for further research. First, real measurements could be performed to validate the model and measure the effect of possible delays coming from state changes, TCP slow start, and others. The second step would imply to continue developing this simulator to bring it closer to a real implementation. It would, then, require removing step by step the limitations mentioned before (e.g., including the radio propagation effects or the HSDPA radio management algorithms). Next to that, the simulator could be improved modifying it to be a real time simulator instead of a virtual time simulator. Finally, results should be contrasted to the results of real networks to fine-tune the simulator.

A second step would imply to perform simulations for real time traffic such as streaming as well as the combination of streaming traffic and web browsing traffic.

Appendix A

Component	Distribution	Parameters	PDF
Session Inter-arrival time	Exponential	Mean session inter-arrival time	$f_x = \lambda e^{-\lambda x}, x \ge 0$ $\lambda = 1/mean\_session\_arrival$
Session length	Inverse Gaussian	Min.= 1 page Max.= 50 pages	$f_x = \sqrt{\lambda / 2 \cdot x^3 \cdot \pi} \cdot e^{-\lambda \cdot (x-\mu)^2 / 2 \cdot x \cdot \mu^2}$ $\lambda = 6.08, \mu = 3.86$
Main object size	Truncated Lognormal	Mean= 10710 B Std dev= 25032 B Min.= 100 B Max.= 2 MB	$f_x = \frac{1}{\sqrt{2\pi}\sigma x} \exp\left[\frac{-(\ln x - \mu)^2}{2\sigma^2}\right], x \ge 0$ $\sigma = 1.37, \mu = 8.35$
Embedded object size	Truncated Lognormal	Mean= 7758B Std dev= 126168 B Min.= 50 B Max.= 2 MB	$f_x = \frac{1}{\sqrt{2\pi}\sigma x} \exp\left[\frac{-(\ln x - \mu)^2}{2\sigma^2}\right], x \ge 0$ $\sigma = 2.36, \mu = 6.17$
Number of embedded objects per page	Truncated Pareto	Mean= 5.64 Max.= 53	$f_x = \frac{\alpha k^{\alpha}}{x^{\alpha+1}}, k \le x < m$ $f_x = \left(\frac{k}{m}\right)^{\alpha}, x = m$ $\alpha = 1.1, k = 2, m = 55$
Reading time	Exponential	Mean = 30 sec	$f_x = \lambda e^{-\lambda x}, x \ge 0$ $\lambda = 0.033$

Appendix B

Additional results for section 5.1.

Mean Cell Throughput	Pages DL time <4 s	Pages DL time >4 s	Sessions Mean E[2]<4 s	Sessions Mean E >4 s	Mean sessions/UE
600	84730	8537	25715	2439	687
800	127793	12829	38631	3665	682,2
1000	167758	16387	50631	4621	682,2
1200	208944	20510	63022	5966	683,05
1400	248274	24105	75157	7142	685,82

Table 4 – Additional results for section 5.1

Additional results for section 5.2.

Number of UEs	Pages DL time <4 s	Pages DL time >4 s	Sessions Mean E <4 s	Sessions Mean E >4 s	Mean sessions/UE
20	46960	497	14366	65	721,5
40	92624	1928	28045	341	709,65
60	134870	5401	41352	1176	708,8

[2] Mean E is equivalent to “User’s session download time” (Equation 2, page 47)

Number of UEs	Pages DL time <4 s	Pages DL time >4 s	Sessions Mean E <4 s	Sessions Mean E >4 s	Mean sessions/UE
100	162525	47964	45601	18068	636,69
120	116931	101416	24341	41671	550,10

Table 5 – Additional results for section 5.2

Additional results for section 5.3.

- Number of UEs = 81

1/ λ	Pages DL time <4 s	Pages DL time >4 s	Sessions Mean E <4 s	Sessions Mean E >4 s	Mean sessions/UE
5	163558	13856	49710	3786	660,44
10	159607	10516	48505	2666	631,74
15	154029	9061	46897	2244	606,68
30	139877	6037	42688	1364	543,85
45	126991	4765	38668	1039	490,21
60	115825	3228	35360	665	444,75

Table 6 – Additional results for section 5.3, number of UEs equal to 81

- Number of UEs = 120

1/ λ	Pages DL time <4 s	Pages DL time >4 s	Sessions Mean E <4 s	Sessions Mean E >4 s	Mean sessions/UE
0	116931	101416	24341	41671	550,10
5	127508	90012	28584	37011	546,62
10	141114	76646	34671	31241	549,26

1/ λ	Pages DL time <4 s	Pages DL time >4 s	Sessions Mean E <4 s	Sessions Mean E >4 s	Mean sessions/UE
15	153313	62974	40561	24882	545,36
30	168096	35994	48787	12697	512,36
45	168503	20921	50729	6515	477,03
60	161437	13208	49106	3597	439,19

Table 7 – Additional results for section 5.3, number of UEs equal to 120

References

[3GPP_002] 3GPP TS 23.002 v6.10.0. Network Architecture.

[3GPP_101] 3GPP TS 25.101 v6.12.0. UE radio transmission and reception (FDD). Page(s): 12 – 14.

[3GPP_105] 3GPP TS 22.105 v6.4.0. Service aspect; services and service capabilities. Page(s): 14 – 16, 28.

[3GPP_201] 3GPP TS 25.201 v6.2.0. Physical layer – General Description. Page(s): 7 – 10.

[3GPP_110] 3GPP TS 23.110 v6.0.0. UMTS access stratum: services and functions. Page(s): 13 – 16.

[3GPP_211] 3GPP TS 25.211 v6.7.0. Physical channels and mapping of transport channels onto physical channels. Page(s): 8, 10 – 38.

[3GPP_212] 3GPP TS 25.212 v6.7.0. Multiplexing and channel coding. Page(s): 54 – 66.

[3GPP_213] 3GPP TS 25.213 v6.4.0. Spreading and modulation. Page(s): 7 – 11, 22 – 23.

[3GPP_214] 3GPP TS 25.214 v6.7.1. Physical layer procedures. Page(s): 14 – 31, 33 – 41.

[3GPP_215] 3GPP TS 25.215 v6.4.0. Physical layer – Measurements. Page(s): 14 – 16.

[3GPP_301] 3GPP TS 25.301 v6.4.0. Radio Interface Protocol Architecture. Page(s): 11, 20 – 21, 29 – 31.

[3GPP_306] 3GPP TS 25.306 v6.8.0. UE Radio access capabilities. Page(s): 24 – 31.

[3GPP_308] 3GPP TS 25.308 v6.3.0. UTRA High Speed Downlink Packet Access (HSDPA); Overall description. Page(s): 12 – 18, 21 – 25.

[3GPP_321] 3GPP TS 25.321 v6.7.0. Medium Access Control (MAC) protocol specification. Page(s): 27

[3GPP_322] 3GPP TS 25.322 v6.6.0. Radio Link Control (RLC) protocol specification. Page(s): 17 – 18.

[3GPP_323] 3GPP TS 25.323 v6.4.0. Packet Data Convergence Protocol (PDCP) specification. Page(s): 8 – 9.

[3GPP_324] 3GPP TS 25.324 v6.4.0. Broadcast Multicast Control (BMC). Page(s): 7 – 9.

[3GPP_331] 3GPP TS 25.331 v6.8.0. RRC protocol specification. Page(s): 39 – 40.

[3GPP_401] 3GPP TS 25.401 v6.7.0. UTRAN overall description. Page(s): 24 – 25, 34 – 35.

[3GPP_858] 3GPP TR 25.858 v5.0.0. HSDPA. Physical layer aspects. Page(s): 7 – 13.

[3GPP_892] 3GPP TR 25.892 v6.0.0. Feasibility Study for Orthogonal Frequency Division Multiplexing (OFDM) for UTRAN enhancement. Page(s): 59 – 60.

[3GPP_896] 3GPP TR 25.896 v6.0.0. Feasibility study for enhanced uplink for UTRA FDD.

[3GPP_902] 3GPP TR 21.902 v6.0.0. Evolution of 3GPP system. Page(s): 6 – 7.

[3GPP_922] 3GPP TR 25.922 v.6.2.0. Radio resource management strategies. Page(s): 15 – 36.

[3GPP2_02] 3GPP2. C.R1002-0 v1.0. cdma2000 Evaluation Methodology. Page(s): 56 – 64.

[Abr_00] Abrahamsson, H.; Ahlgren, B. *Using Empirical Distributions to Characterize Web Client Traffic and to Generate Synthetic Traffic.* IEEE Global Telecommunications Conference, volume 1. December 2000. Page(s): 428 – 433.

[Ame_04] Ameigeiras, P.; Wigard, J.; Mogensen, P. *Performance of Packet Scheduling Methods with Different Degree of Fairness in HSDPA.* IEEE 60th Vehicular Technology Conference, volume 2. September 2004. Page(s): 860 – 864.

[Ani_04] Aniba, G.; Aïssa, S. *Fast Packet Scheduling Assuring Fairness and Quality of Service in HSDPA.* Canadian Conference on Electrical and Computer Engineering, volume 4. May 2004. Page(s): 2243 – 2246.

[Bau_03] Baum, K.L.; Kostas, T.A.; Sartori, P.J.; Classon, B.K. *Performance characteristics of cellular systems with different link adaptation strategies.* IEEE Transaction on Vehicular Technology, volume 52, issue 6. November 2003. Page(s): 1497-1507.

[Boa_03] Boaretto, M.; Angelini, D.; Zorzi, M. *Fast Scheduling Policies using WCDMA High Speed Downlink Packet Access.* IEEE 58th Vehicular Technology Conference, volume 4. October 2003. Page(s): 2650 – 2654.

[Cao_02] Cao J.; Cleveland, W.S.; Lin D.; Sun D.X. *Internet Traffic Tends Toward Poisson and Independent as the Load Increases.* Nonlinear Estimation and Classification eds. 2002.

[Cho_99] Choi, H.-K.; Limb, J.O. *A behavioral model of Web traffic*. Proceeding of the Seventh International Conference on Network Protocols. 1999. Page(s): 327 – 334.

[Don_04] Donna L. Hoffman; Thomas P. Novak, Alladi Venkatesh. *Has the Internet become indispensable?* Communications of the ACM, volume 47, issue 7. July 2004. Page(s): 37 – 42

[Dot_04] Dottling, M.; Raaf, B.; Michel, J. *Efficient channel quality feedback schemes for adaptive modulation and coding of packet data.* IEEE 60th Vehicular Technology Conference, volume 2. September 2004. Page(s): 1243-1247.

[ETSI_111] ETSI TS 121 111 v3.5.0. UMTS; USIM and IC cards requirements. Page(s): 10 – 12.

[ETSI_112] ETSI TR 101 112 v3.2.0. UMTS: Selection Procedures for the choice of radio transmission technologies of the UMTS. Page(s): 33 – 35.

[Eva_00] Evans, B.G.; Baughan, K. *Visions of 4G*. Electronics & Communication Engineering Journal, volume 12, issue 6. December 2000. Page(s): 293 – 303

[Fre_01] Frenger, P.; Parkvall, S.; Dahlman, E. *Performance Comparison of HARQ with Chase Combining and Incremental Redundancy for HSDPA.* IEEE 54th Vehicular Technology Conference, volume 3. October 2001. Page(s): 1829 – 1833.

[Fre_02] Frederiksen, F.; Kolding, T.E. *Performance and Modeling of WCDMA/HSDPA Transmission H-ARQ Schemes.* IEEE 56th Vehicular Technology Conference, volume 1. September 2002. Page(s): 472 – 476

[Gal_05] Gal, Dan. *IEEE 802.20 Evaluation Criteria (EC) – Simulation of Common*

Traffic Types. IEEE C802.20-05/25, May 2005. Page(s): 6 – 7.

[GSM_06] GSM subscriber statistics, GSMWorld. [Online PDF document]. Available at: http://www.gsmworld.com/news/statistics/ Last access: April 2006.

[Hal_06] Halvey, M.; Keane, M.T.; Smyth, B. *Mobile Web Surfing is the Same as Web Surfing.* Communications of the ACM, volume 49, issue 3. March 2006. Page(s): 76 – 81.

[Her_03] Hernandez-Campos, F.; Jeffay, K.; Smith, F.D. *Tracking the evolution of Web traffic: 1995-2003*. 11th IEEE/ACM International Symposium on Modeling, Analysis and Simulation of Computer Telecommunications Systems. October 2003. Page(s): 16 – 25.

[Hol_04] Holma, Harri; Toskala, Antti. *WCDMA for UMTS*. 3rd Edition. Wiley. Chichester, 2004. Page(s): 269 – 306.

[Hol_06] Holma, Harri; Toskala, Antti. *HSDPA/HSUPA for UMTS.* Wiley. Chichester, 2006. Page(s): 31 – 60, 95 – 121.

[ITU_010] ITU-T G.1010. End-user multimedia QoS categories. Page(s): 4 – 9.

[Kaa_05] Kaaranen, Heikki; Ahtiainen, Ari. *UMTS networks: architecture, mobility and services*. 2nd Edition. Wiley. Chichester, 2005. Page(s): 113 – 116.

[Kei_02] Keiji Tachikawa. *W-CDMA mobile communications system*. Wiley. Chichester, 2002.

[Kha_04] Khan, Farooq. *Traffic Models for IEEE 802.20 MBWA System Simulations*. Version 2. IEEE C802.20-04/22. January 2004. Page(s): 8 – 9.

[Kle_01] Klemm, A.; Lindemann, C.; Lohmann, M. *Traffic modeling and characterization for UMTS networks.* IEEE Global Telecommunications Conference, volume 3. November 2001. Page(s): 1741 – 1746

[Kol_03] Kolding, T. E.; Pedersen, K. I.; Wigard, J.; Frederiksen, F.; Mogensen, P. E. *High Speed Downlink Packet Access: WCDMA Evolution.* IEEE Vehicular Technology Society News, volume 50, issue 1. February 2003. Page(s): 4 – 10.

[Lai_06] Laiho, Jaana; Wacker, Achim; Novosad Tomáš. *Radio Network Planning and Optimization for UMTS*. 2nd Edition. Wiley. Chichister, 2006. Page(s): 250 – 254.

[Law_00] Lawrence G. Roberts. *Beyond Moore's Law: Internet Growth Trends* IEEE Computer Society, volume 33, issue 1. 2000. Page(s): 117-119.

[Lee_00] Lee, L.-N.; Hammons, A. R.; Feng-Wen Sun; Erozm M. *Application and standardization of turbo codes in third-generation high- speed wireless data services*. IEEE Transactions on Vehicular Technology, volume 49, issue 6. November 2000. Page(s): 2198-2207.

[Lov_04] Love, R.; Ghosh, A., Xiao, W.; Ratasuk, R. *Performance of 3GPP High Speed Downlink Packet Access*. IEEE 60th Vehicular Technology Conference, volume 1. September 2004. Page(s): 689 – 693.

[Mul_05] Müller A.; Tao Chen. *Improving HSDPA link adaptation by considering the age of channel quality feedback information.* IEEE 62nd Vehicular Technology Conference, volume 3. September 2005. Page(s): 1643-1647.

[Nag_03] Naguib, Ayman F.; Sutivong, Arak; Tiedemann, Ed; Tomcik, Jim. *802.20 Evaluation Methodology Strawman – 00*. IEEE C802.20-03/43. June 2003. Page(s): 11 – 15.

[Nak_02] Nakamura, M.; Awad, Y.; Vadgama, S. *Adaptative control of link adaptation for high speed downlink k packet access (HSDPA) in W-CDMA.* The 5th International Symposium on Wireless Personal Multimedia Communications, volume 2. October 2002. Page(s): 382-386.

[Nas_05] Nasser, N.; Al-Manthari, B.; Hassanein, H. *A Performance Comparison of Class-based Scheduling Algorithms in Future UMTS Access*. 24th IEEE Performance and Communication Conference. April 2005. Page(s): 437 – 441.

[Ped_04] Pedersen K.I.; Lootsma, T.F.; Støttrup, M.; Frederiksen, F.; Kolding T.E. Mogensen, P.E. *Network Performance of Mixed Traffic on High Speed Downlink Packet Access and Dedicated Channels in WCDMA.* IEEE 60th Vehicular Technology Conference, volume 6. September 2004. Page(s): 4496 – 4500.

[Ped_06] Pedersen K.I.; Michaelsen, P.H. *Algorithms and Performance Results for Dynamic HSDPA Resource Allocation*. IEEE 64th Vehicular Technology Conference. September 2006.

[Pok_05] Pokhariyal, A.; Kolding, T.E.; Frederiksen, F.; Olives, P.; Sorensen, T.B.; Mogensen, P.E. *Investigation of frequency-domain link adaptation for a 5-MHz OFDMA/HSDPA system.* IEEE 61st Vehicular Technology Conference, volume 3. June 2005. Page(s): 1463-1467.

[Ran_98] Randy H. Katz. *Beyond third generation telecommunications architectures: the convergence of internet technology and cellular telephony*. ACM SIGMOBILE Mobile Computing and Communications Review, volume 2, issue 2. April 1998. Page(s): 1 – 5

[San_03] Santhi, K.R.; Srivastava, V.K.; SenthilKumaran, G.; Butare, A. *Goals of true broad band's wireless next wave (4G-5G)*. IEEE 58th Vehicular Technology Conference, volume 4. October 2003. Page(s): 2317 – 2321

[Sha_03] Shankaranarayanan, N.K.; Jiang, Z; Mishra, P. *Performance of a Shared Packet Wireless Network with Interactive Users.* Mobile Networks and Applications, volume 8, issue 3. Kluwer Academic Publishers. 2003. Page(s): 279 – 293.

[Sha_05] Sharma, G.; Kumar, G.S. *Moving towards HSUPA (high speed uplink packet access): a complete 3.5G wireless system*. IEEE International Conference on Personal Wireless Communications. January 2005. Page(s): 174 – 177.

[Sol_05] Soldani, D. *QoS Management in UMTS Terrestrial Radio Access FDD Networks*. Thesis (PhD). University of Technology of Helsinki. 2005. Page(s): 186 – 187.

[Tya_03] Tyan, H.-Y.; Su, C.-F.; He, G.; Su, X. *Application-driven Internet traffic analysis - a simulation approach*. IEEE Global Telecommunications Conference, volume 7. December 2003. Page(s): 3857 – 3861.

[Van_04] Vanghi, Vieri; Damnjanovic, Aleksandar; Vojcic, Branimir. *Cdma2000 System for Mobile Communications, The: 3G Wireless Evolution.* Prentice Hall, 2004. Pages(s): 84 – 87.

[Vid_00] Vidács A.; Barta, J.; Kenesi, Z.; Éltetö, T. *Measurement-Based WWW User Traffic Model for Radio Access Networks.* Proc. of 7th International Workshop on Mobile Multimedia Communications. October 2000.

[Wan_04] Wang, L-C.; Chen, M-C. Comparison *of Link Adaptation Algorithms for the WCDMA System with High Speed Downlink Packet Access.* IEEE 59th Vehicular Technology Conference, volume 5. May 2004. Page(s): 2456 – 2460.

[Wan_05] Wang Hai; Wan Lei; Liao Min. *HSDPA link adaptation based on novel quality model.* IEEE 61st Vehicular Technology Conference, volume 1. June 2005. Page(s): 334 – 338.

[Tur_04] Türke, U.; Koonert, M.; Schelb, R.; Görg, C. *HSDPA Performance Analysis in UMTS Radio Network Planning Simulations.* IEEE 59th Vehicular Technology Conference, volume 5. May 2004. Page(s): 2555 – 2559.