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Evaluation of data applications quality of service over UMTS

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To my loved ones

“Wisdom is not a product of schooling but of the lifelong attempt to acquire it.”

- *Albert Einstein -Scientist*

“Too many people go through life waiting for things to happen instead of making them happen! “

- *Sasha Azevedo – Actress*

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Abstract

The main objective of this thesis was to study the assessment of data applications, quality of service in UMTS/HSDPA, considering various applications and environments, at the cellular level. These objectives were achieved through the development of a model and its implementation in a simulator in Matlab. The main scripts are one for each data service, and the auxiliary are the script for instantaneous bit rate, the script to evaluate the available codes and the script for the network load. With different input parameters, like environment, SNR variation, channels, number of other users, service penetration, indoor/outdoor reference user location and data service characteristics one has obtained as output parameters average bit rate, the initial delay and average delay. By varying the network load up to 60 other users, there is a decrease of the average bit rate of about 80% in the web browsing and file download services, for video streaming the ratio between the video duration and the average video streaming delay reaches 200%. The influence of the reference user environment, signal to noise ratio, is reflected in the average bit rate for web browsing and file download services, being 17 Mbps in the urban outdoor environment and 6 Mbps in the rural indoor one. The server location may be crucial especially for short videos transmitted via satellite the ratio between the video duration and the average video streaming delay being almost 250% for a rural indoor environment.

Keywords

UMTS, HSDPA, QoS, Services and Applications, Data.

Resumo

O objetivo principal desta tese foi estudar a avaliação das aplicações de dados, qualidade de serviço UMTS/HSDPA, considerando várias aplicações e ambientes, a nível celular. Estes objectivos foram conseguidos com o desenvolvimento de um modelo e sua implementação num simulador em *Matlab* Para cada serviço de dados foram feitos *scripts* principais, para o ritmo de transmissão, para o número de códigos disponíveis e para a carga da rede foram realizados *scripts* auxiliares. Com diferentes parâmetros de entrada, como o ambiente, variação da relação sinal ruído, número de canais, outros utilizadores, penetração do serviço, utilizador referência dentro ou fora dos edifícios e características dos serviços obtiveram-se os parâmetros de saída ritmo de transmissão médio, atraso inicial e atraso médio. Variando a carga da rede até 60 outros utilizadores, observa-se um decréscimo de ritmo de transmissão médio de cerca de 80% nos serviços de navegação na internet e *download* de ficheiros, no serviço de transmissão vídeo a razão entre a duração do vídeo e o atraso médio atinge os 200%. A influência do ambiente do utilizador referência, relação sinal ruído, reflete-se no ritmo de transmissão médio para os serviços de navegação na internet e *download* de ficheiros, sendo no ambiente urbano exterior 17 Mbps e no ambiente rural interior 6 Mbps. A localização do servidor tem crucial importância principalmente para vídeos de curta duração e para transmissões via satélite sendo a razão entre a duração do vídeo e o atraso médio de quase 250% para um ambiente rural interior.

Palavras-chave

UMTS, HSDPA, QoS, Serviços e Aplicações, Dados.

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List of Acronyms

3GPP	3rd Generation Partnership Project
AMC	Adaptation Modulation and Coding
AoD	Audio On Demand
ARP	Allocation/Retention Priority
ARQ	Automatic Repeat Request
ATM	Asynchronous Transfer Mode
BER	Bit Error Ratio
BLER	Block Error Rate
BMC	Broadcast/Multicast Protocol
BS	Base Station
CDMA	Code Division Multiple Access
CN	Core Network
CQI	Channel Quality Indicator
CQM	Customer QoS Management
CS	Circuit Switch
DL	Downlink
DPCCH	Dedicated Physical Control Channel
DS-CDMA	Direct-Sequence CDMA
EML	Element Management Layer
ETSI	European Telecommunications Standards Institute
FDD	Frequency Division Duplex
FDMA	Frequency Division Multiple Access
FP	Frame Protocol
FTP	File Transfer Protocol
GBR	Guaranteed Bit Rate
GGSN	Gateway GPRS Support Node
GMSC	Gateway Mobile Switching Centre
GPRS	General Packet Radio Service
GTP	GPRS Tunnelling Protocol
GoS	Grade of service
HLR	Home Location Register
HSDPA	High Speed Downlink Packet Access
HSPA	High Speed Packet Access
HSPA+	HSPA Evolution
HSUPA	High Speed Uplink Packet Access

HTTP	Hypertext Transfer Protocol
IP	Internet Protocol
ISCP	Interference Signal Code Power
ISDN	Integrated Services Digital Network
ITU	International Telecommunications Union
KPI	Key Performance Indicators
MAC	Medium Access Control
MCS	Modulation and Coding Scheme
ME	Mobile Equipment
MIMO	Multiple Input Multiple Output
MMS	Multimedia Messaging Service
MSC	Mobile Switching Centre
MT	Mobile Terminal
NE	Network elements
NML	Network Management Layer
OVSF	Orthogonal Variable Spreading Factor
PDCP	Packet Data Convergence Protocol
PDF	Probability Distribution Function
PLMN	Public Land Mobile Network
PS	Packet Switch
PSTN	Public Switched Telephone Network
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
RAB	Radio Access Bearer
RAN	Radio Access Network
RLC	Radio Link Control
RNC	Radio Network Controller
RNS	Radio Network Sub-system
RRC	Radio Resource Control
RSCP	Received Signal Code Power
RT	Real Time
RTT	Round Trip Time
RU	Reference User
SAP	Service Access Point
SDU	Service Data Unit
SF	Spreading Factor
SGSN	Serving GPRS Support Node
SIMO	Single Input Multiple Output
SINR	Signal-to-Interference-plus-Noise Ratio
SIR	Signal-to-Interference Ratio
SISO	Single Input Single Output

SMS	Short Messaging Service
SQM	Service Quality Management
TC	Traffic Class
TCP	Transmission Control Protocol
TDD	Time Division Duplex
THP	Traffic Handling Priority
TPC	Transmit Power Control
TTI	Transmission Time Interval
UDP	User Datagram Protocol
UE	User Equipment
UL	Uplink
UMTS	Universal Mobile Telecommunications System
USIM	UMTS Subscriber Identity Module
UTRAN	UMTS Terrestrial Radio Access Network
VLR	Visitor Location Register
VoD	Video on Demand
WCDMA	Wideband CDMA

List of Symbols

α_j	Code orthogonality factor of user *j*
Δ	Active Connections
β	Block Error Rate BLER
ψ	Interference signal code power
$\overline{\chi}$	Mean Throughput of connection
ε	Received signal code power of a single UE
ρ	RLC Transport Blocks with CRC Error
σ	Signal to Interference Ratio (SIR)
$\chi_{(i,j)}$	Throughput *(i,j)*
μ_{Dd}	Mean inter arrival time between two consecutive packets inside a packet call
$\overline{\eta_{DL}}$	Average DL load factor value across the cell
η_{DL}	DL factor
ρ_{Nj}	Signal-to-noise ratio (SNR) of user *j*
ρ_{IN}	Signal-to-Interference-plus-Noise Ratio
ρ_N	Signal to Noise Ratio
ε_{RLC}	RLC Transport Blocks
η_{UL}	UL factor
μ_{Dpc}	Mean reading time between two consecutive packet calls
μ_{Npc}	Mean number of packet calls requests per session
μN_d	Mean number of datagrams within a packet call
a_{pd}	Average power decay
C_R	Coding rate
D_d	Inter arrival time between two consecutive packets inside a packet call
D_{pc}	Reading time between two consecutive packet calls
E_0	Noise spectral density
E_b	Energy per bit
$F_{a\,j}$	Activity factor of user *j*
G_r	Gain of the receiving antenna
G_t	Gain of the transmitting antenna
$I_{inter\,n}$	Normalised inter-cell interference
$\overline{L_{pj}}$	Average path loss between the BS and user *j*
L_{ref}	Propagation losses
M_I	Interference margin
N_0	Noise spectral density at the MT

N_C	Number of codes
N_d	Number of datagrams within a packet call
N_{dc}	Number of codes for data services
N_{du}	Number of users connected in a data services
N_{mc}	Maximum number of codes for a channelisation system type
N_{pc}	Number of packet calls requests per session
$n_{oc}[n]$	Number of occurrences within interval n
N_{pc}	Number of packet calls requests per session
N_U	Number of active users
N_{vc}	Number of codes for voice service
N_{vu}	Number of users connected in voice service
P_C^{BS}	BS transmission power allocated to common channels
P_r	Power available at the receiving antenna
P_t	Power fed to the transmitting antenna
P_{Tot}^{BS}	Total BS transmission power
P_{TX}^{BS}	BS transmission power allocated to traffic channel
$R_{b\,j}$	Data rate associate to service of user j
Rc	Chip rate of WCDMA
s	Sample period
S	Measurement period
S_d	Packet size
SF_n	Spreading Factor ranges
T_C	Coherence Time

List of Software

Mathworks Matlab R2011a	Numerical computing software
Microsoft Excel 2010	Calculation and graphical chart tool
Microsoft Visio 2010	Scheme design software
Microsoft Word 2010	Text editor software
Microsoft Power Point 2010	Graphical presentation software

Chapter 1

Introduction

The present chapter introduces the theme of this dissertation, in particular over a contextual and motivational perspective, while simultaneously providing an overview of the assumptions established for the work development. Furthermore, it establishes the scope for the work performed, together with its main contributions, followed by the detailed presentation of the work's structure.

1.1 Overview

The first successful commercial offerings of mobile wireless services emerged in the 70s. In the following decade, the first major development of mobile telephone systems took place, and several analogue systems were launched on the market, which had a very good acceptance and a rapid growth. These mobile communication systems have become known as first generation (1G) ones [HoTo04].

At this stage, specific characteristics of each system led to an incompatible scenario with others, namely in the aspects of operation, such as the use of terminal equipment, which made it impossible for the interconnection of networks, by confining the use of services to the borders of each country.

In 1982, the *Conférence Européenne des Administrations des Postes et des Télécommunications* (CEPT) established a working group - *Groupe Spéciale Mobile* (GSM) - to develop a system of pan-European mobile communications.

In 1989, the newly created European Telecommunications Standards Institute (ETSI), assumed the coordination group GSM, with the first published specifications in 1990/1991, opening the door to a second generation (2G) in mobile communications.

GSM has changed the traditional telephone service, adding mobility, becoming a commercial success for technology manufacturers and operators who have obtained exponential growth until the end of the 90s. At this time, the number of users of the mobile phone services surpassed, in many countries, the traditional fixed ones.

GSM technology became the mobile communications system with greater worldwide success; the International Telecommunications Union (ITU) launched an ambitious project: a federation of mobile communications systems for third generation (3G) that would allow continued access to the global infrastructure of telecommunications, by anyone, anytime, anywhere.

The project, called IMT-2000 (International Mobile Telecommunications-2000), identified the key factors for the success of this new generation of mobile communications:

- high access bit rate, to support broadband services, such as Internet access and multimedia applications;
- flexibility, allowing new types of service, such as universal personal number and satellite phone;
- compatibility with existing mobile networks constituting an evolution of these.

Universal Mobile Telecommunications System (UMTS) belongs to the IMT-2000 family, being an evolution of GSM/GPRS(General Packet Radio Service), which uses Wideband Code Division Multiple Access (WCDMA) as radio access technology and is currently the leading 3G technology worldwide. Seamless interoperability of UMTS with GSM makes possible the interoperability of services.
In addition to the services supported by GSM (voice, SMS - Short Messaging Service, MMS -

Multimedia Messaging Service, etc.), UMTS adds mobile multimedia services (video telephony, video streaming, video conferencing, mobile TV, etc.) and allows access to the Internet and other data services with transmission rates higher than those allowed by GSM/GPRS. 3rd Generation Partnership Project (3GPP) is responsible for important evolution steps on top of WCDMA: High Speed Packet Access (HSPA) for downlink (DL) in Release 5 and uplink (UL) in Release 6. The DL solution, High Speed Downlink Packet Access (HSDPA) was commercially deployed in 2005 and the UL counterpart, High Speed Uplink Packet Access (HSUPA), during 2007. The combination of HSDPA and HSUPA is referred to as HSPA. The initial peak data rate of HSDPA was 1.4 Mbps but, by the end of 2007, 7.2 Mbps were available, with the peak data rate of 41.2 Mbps being currently available, starting the mobile Internet Protocol (IP) revolution [HoTo06]. HSUPA started to be deployed at the end of 2007, with a maximum peak data rate around 5.7 Mbps [HoTo06].

Performance and usability of mobile handsets has improved, together with a wide adoption of 3G dongles. As a consequence, mobile data traffic has been increasing, Figure 1.1. There is also a change in the used technology: UMTS/HSPA grows and GSM tends to decrease Figure 1.2.

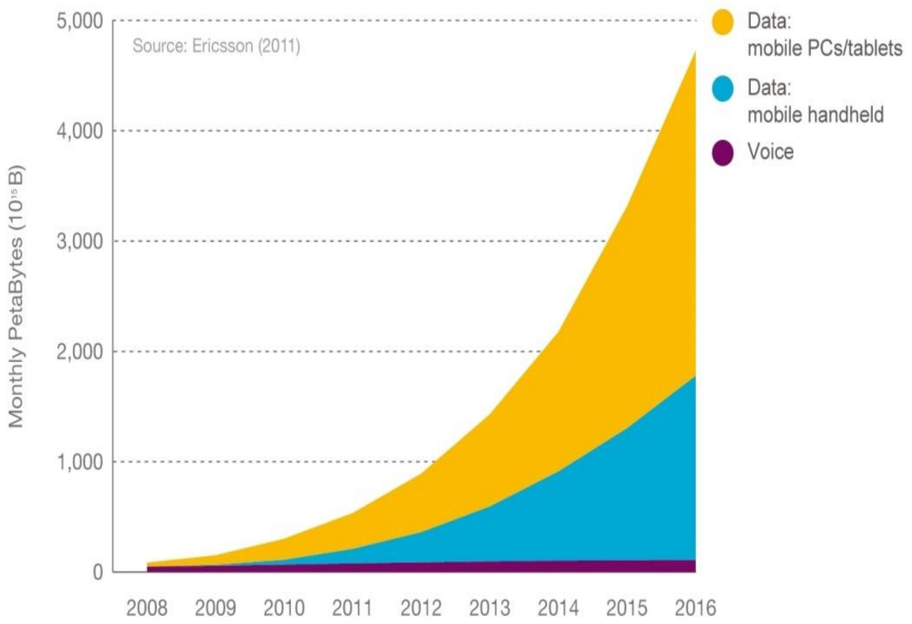

Figure 1.1. - Mobile traffic: voice and data, 2008-2016, (extracted from [Eric11]).

In GSM and UMTS, the Quality of Service (QoS) from the user perspective is of crucial importance. It is of vital importance for the management and performance optimisation of a mobile network, allowing operators to provide quality services without overbuilding their infrastructure. From the user point of view, quality indicators give important information about network performance, and concomitantly allow to choose the best configuration for the network, according to the operator’s needs and objectives.

From the end user perspective, mobile operators that provide superior service have a competitive advantage over competitors who despise this aspect. Market studies [SoLC06] show that the frustration of consumers in relation to the quality of service is responsible for about 82% of the operator changes. There is also a chain effect: an unhappy customer will tell, on average, to 13 people their poor experience.

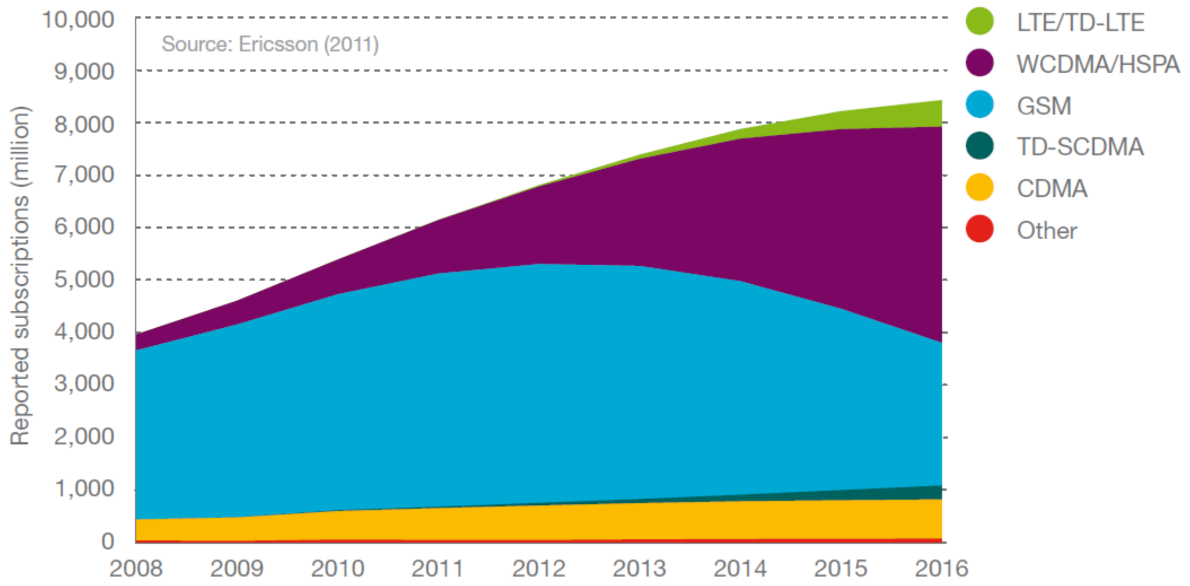

Figure 1.2. - Mobile subscriptions by technology, 2008-2016, (extracted from [Eric11]).

Also, operators cannot wait for customer's complaints, in order to take corrective actions, since for every customer who complains there are 29 others who never do, and simply change the operator when they feel dissatisfied. These dropouts affect the operators' revenue and image. So, to prevent these situations, a strategy including the constant monitoring should be adopted, in order to gather key indicators to be used in QoS improvement, also taking the way as it is perceived by users into account.

It is becoming increasingly evident that data is equivalent to 'gold' on balance sheets, especially with the data growth. As the requirements for different applications vary, this growth of non-voice services has posed a new challenge to managing their performance in more effective ways. This is essential in order to provide best-of-class services to the end-user, without over dimensioning precious network resources.

Wireless connectivity worldwide is becoming a real need in advanced societies, not only for business, but also for entertainment, communities, or even security purposes. In that sense, a huge variety of services is nowadays coexisting in a very complex and heterogeneous network infrastructure, which is additionally managed by different parties. The complexity of monitoring and optimisation processes of data service performance is evident, considering not only the quick diversification of emerging services and quality requirements associated to them, but also the coexistence of a huge variety of access technologies and the wide coverage through which such technologies are offered. The goal of a mobile network operator is to offer the customer an assured end-to-end QoS, with a variety of service levels and predictable service response.

The first step to define a QoS technique is the creation of a set of parameters associated to each data service. Delay, bandwidth and reliability are attributes that are commonly used for that purpose. This is especially important for supporting the new generation of Internet and mobile applications, such as video streaming and other real-time services. The final goal of QoS is to be able to classify the packets, and treat them according to the application requirements, so that the maximum number of users is satisfied.

QoS differentiation gives more preference to the packets of more demanding service (video) in order to guarantee a minimum delay, while less bandwidth is given to the e-mail service, where user's perception of delay is not so critical. This is why some operators offer to their customers' different payment plans, according to their services and applications.

Other factors that have direct influence on QoS, and are not considered in this thesis, are mobility with the influence of handover and user velocity. Another aspect with influence is the category of the reference user terminal; equipment's with lower category have lower modulation and consequently reach lower throughput levels. For example, if the user terminal is an equipment of category 14, just by changing it to category 13 the maximum theoretical peak data rate decreases 16%, and if it is changed to category 1 it could decrease 94%.

1.2 Motivation and Contents

The main scope of this thesis is to evaluate data applications QoS over UMTS, considering various applications and environments, at the cellular level. These objectives were accomplished through the development of a model, and its implementation in a simulator.

The main contribution of this thesis is the development of a model for the downlink of three different data services in three different environments, considering in all of them an indoor and outdoor location, a service penetration, being evaluated by the initial delay, the average delay and the average bit rate. Another component of the study is to evaluate the influence of the server location in the delay and the influence of the network load in the data service in study.

This work is composed of 5 Chapters, including the present one, followed by a set of annexes. Chapter 2 introduces mainly UMTS and its latest Release (HSPA+), radio interface, interference coverage and capacity. At the end, services and applications, quality of service and quality of experience are presented.

Chapter 3 introduces the developed models used for simulation. The three environments, their coherence time and the respective Signal-to-Noise-Ratio (SNR) are characterised, as well the three data services in study. The influence of the sever location and the network load are parameters considered in the model. The simulator assessment is presented at the end.

Chapter 4 begins with a description of the analysed scenarios. After that, a measurement overview is given, followed by its results, where the influence of the reference user service, environment, network load, and server location are considered.

Chapter 5 concludes the present dissertation, where a critical analysis is drawn followed by the main work conclusions. Furthermore suggestions for future work are outlined, and paths for further research on next generation mobile communications are proposed. Finally, a set of annexes closes the present document, with supplementary information, when the need for the global comprehension of the problem exists.

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Chapter 2

Basic Concepts

This chapter provides an overview of UMTS, HSDPA and HSPA+. Network architectures, basic characteristics, like bandwidth and channels, are described. This chapter also describes the service classes defined in the context of each system, and closes with a general overview of the QoS concept, and the corresponding literature survey characterising the state-of-the-art.

2.1 Network Architecture

This section, based on [HoTo04], UMTS Releases 5, 6 and 7, presents basic concepts, network architecture, air interface and performance aspects. UMTS is formed by a number of logical network elements, each having a defined functionality. The network elements, for both UMTS and GSM, presented in Figure 2.1, are grouped into:

- Radio Access Network (RAN, UMTS Terrestrial Radio Access Network - UTRAN) that handles all radio related functionality;
- Core Network (CN), which is responsible for switching and routing call and data connections to external networks;
- User Equipment (UE) that interfaces with the user and the radio interface.

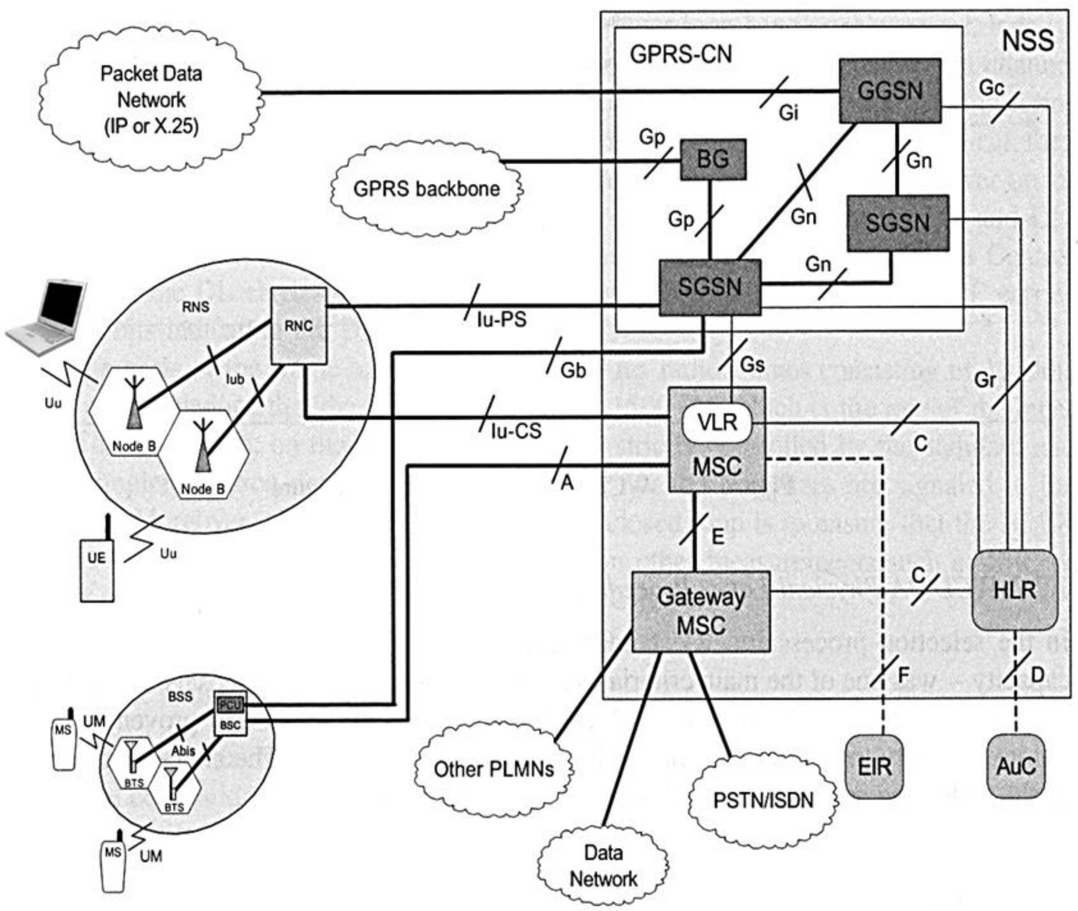


Figure 2.1. - UMTS network architecture (including GSM) (adapted from [CBGM06]).

The UE is formed by the Mobile Equipment (ME), Mobile Terminal (MT), and the UMTS Subscriber Identity Module (USIM). The first one is the radio terminal, used for radio communication over the *Uu* interface, and the third one is a smartcard that holds the subscriber identity, performs authentication algorithms, and stores information. The ME and USIM communicate over the *Cu* interface.

UTRAN is formed by sub-networks consisting of one or more Radio Network Sub-system (RNS), which in turn are formed by one Radio Network Controller (RNC) and one or more Node Bs, commonly called Base Station (BS). The Node B converts the data flow between the *Iub* and *Uu* Interfaces, and takes part in radio resource management. The RNC is responsible for the control of the radio resources of UTRAN, and may be connected to each other via an *Iur* Interface.

The CN is adapted from the GSM one, the main elements being:

- Home Location Register (HLR) is a database located in the user's home network;
- Mobile Switching Centre/Visitor Location Register (MSC/VLR) is the switch and the database that serves the UE in its current location for Circuit-Switched (CS) services;
- Gateway MSC (GMSC) is the switch at the point where UMTS Public Land Mobile Network (PLMN) is connected to external CS networks; it is where all incoming and outgoing CS connections carry by;
- Serving GPRS Support Node (SGSN) that has similar functionalities to MSC/VLR, but is used for Packet-Switched (PS) services;
- Gateway GPRS Support Node (GGSN) which functionality is close to that of GMSC, but relative to PS services.

The main interfaces between the logical network are:

- *Cu* Interface, reference point between USIM and ME;
- *Uu* Interface, reference point between User Equipment and Infrastructure domains, UMTS radio interface;
- *Iu* Interface, reference point between Access and Serving Network domains;
- *Iur* interface, support soft handover between RNCs connecting multiple RNCs within the same UTRAN;
- *Iub* interface. The Iub connects a Node B and an RNC. UMTS is the first commercial mobile telephony system where the Controller–Base Station interface is standardised as a fully open interface.

At a higher level, the migration from Release 99 to Releases 4, 5 and 6, does not change the structure of the network. However, the details do differ: for example, the transport for the interfaces changes from Asynchronous Transfer Mode (ATM) in Release 99 to all Internet Protocol (IP) in Release 5. In addition, the layering changes in Release 5, to support HSDPA and Node B scheduling [CBGM06].

The protocols over *Uu* and *Iu* interfaces are divided into two structures: User Plane (UP) protocols, i.e., the protocols implementing the actual Radio Access Bearer (RAB) service; and Control Plane (CP) protocols, i.e., the protocols for controlling radio access bearers and the connection between the UE and CN. Both Radio and Iu protocols provide transparent transfer of Non-Access Stratum (NAS) messages between the UE and CN.

The biggest QoS-related difference between the *Gb* and *Iu* interface architecture is the timing of Real

Time (RT) application support by the Guaranteed Bit Rate (GBR) or RT bearer services [SoLC06].

Radio interface protocols are needed to set up, reconfigure and release the Radio Bearer services. The radio interface consists of three protocol layers – the physical layer (L1); the data link layer (L2); and the network layer (L3). Layer 2 contains the following sub layers – Medium Access Control (MAC), Radio Link Control (RLC), Packet Data Convergence Protocol (PDCP) and Broadcast/Multicast Control (BMC). RLC is divided into Control (C) and User (U) planes, whilst PDCP and BMC exist only in the U-plane. Layer 3 consists of one protocol, denoted Radio Resource Control (RRC), which belongs to the C-plane.

The radio interface protocol architecture and the connections between protocols are shown in Figure 2.2. Each block represents an instance of the corresponding protocol. The dashed lines represent the control interfaces through which the RRC protocol controls and configure the lower layers. The service access points between MAC and physical layer and between RLC and MAC sub layers provide the transport channels (TrCHs) and the logical channels (LoCHs), respectively. The TrCHs are specified for data transport between physical layer and Layer 2 peer entities, whereas logical channels define the transfer of a specific type of information over the radio interface.

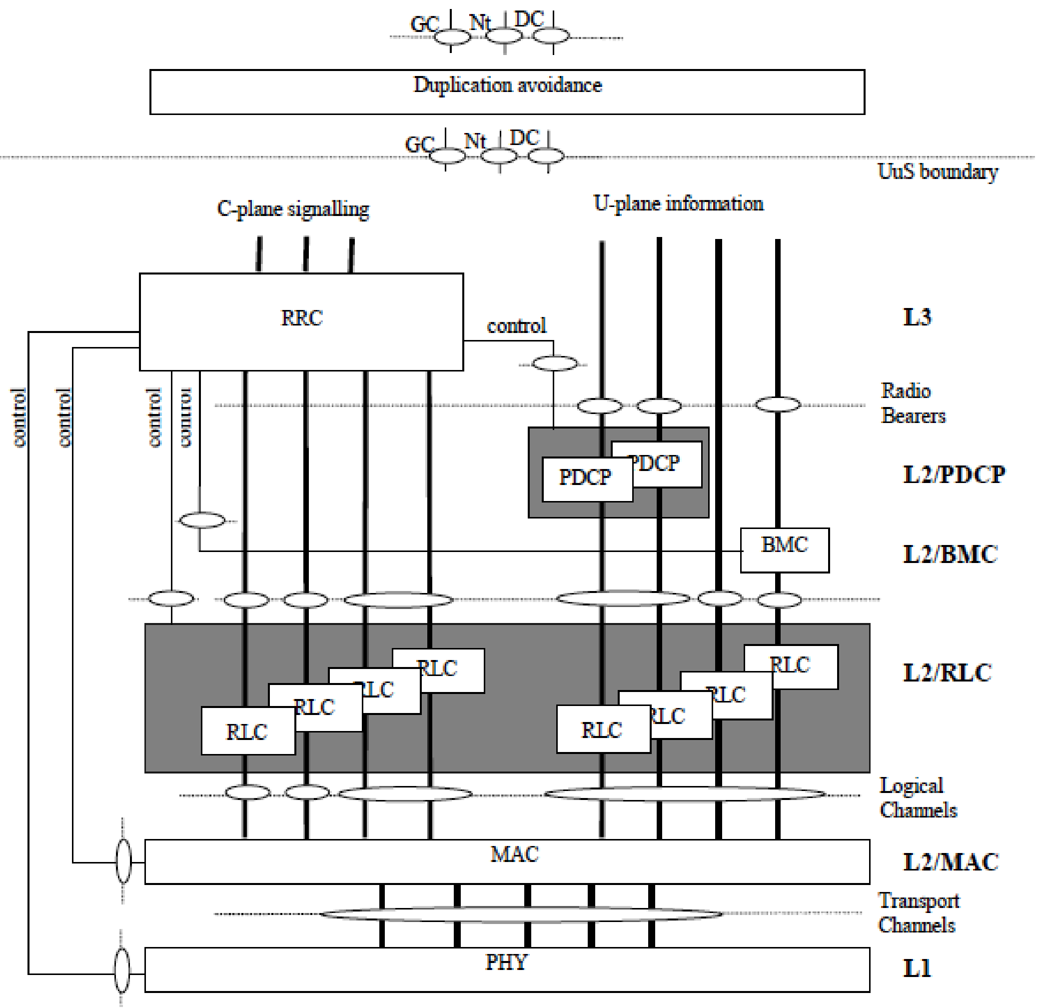


Figure 2.2. - Radio Interface protocol architecture (extracted from [ETSI04]).

The functionality of the Medium Access Control Protocol MAC layer includes: priority handling between data flows of one connection (selection of a TFC for which high priority data is mapped onto L1 with a “high bit rate” TF); priority handling between UE(s) by means of dynamic scheduling (MAC realises priority handling on common and shared TrCHs); identification of UEs on common transport channels (in band identification). MAC provides multiplexing and de-multiplexing functions of RLC PDUs into and from TBs delivered to and from the physical layer on both common and dedicated channels (services or better LoCHs multiplexing for CCHs and DCHs, respectively). MAC is further responsible for: traffic volume monitoring (measurement of traffic volume on logical channels and reporting to RRC, based on which the RRC performs TrCH switching decisions); dynamic TrCH type switching (execution of the switching between common and dedicated transport channels); ciphering (for transparent RLC mode); Access Service Class (ASC) selection for UL common channels transmission; and HARQ functionality for HS-DSCH transmission, in-sequence delivery and assembly-disassembly of higher layer PDUs on HS-DSCH

Each RLC instance is configured by RRC to operate in one of the three modes: Transparent Mode, where no protocol overhead is added to higher layer data; Unacknowledged Mode (UM), where no retransmission protocol is in use and data delivery is not guaranteed; Acknowledged Mode (AM), where the Automatic Repeat request (ARQ) mechanism is used for error correction. For all RLC modes, the CRC error detection is performed on the physical layer, and the results of the CRC are delivered to the RLC together with the actual data. Some of the most important functions of the RLC protocol are segmentation and re-assembly of variable length higher layer PDUs into/from smaller RLC Payload Units; error correction, by means of retransmission in the acknowledge data transfer mode; in-sequence delivery of higher layer PDUs; flow control, i.e., rate control at which the peer RLC transmitting entity may send information; protocol error detection and recovery; and ciphering.

Air interface data channels are carried over I_{ub} by the Frame Protocol (FP). Frame Protocols are applied to transfer user data and the necessary control information between the serving RNC (SRNC) and the BS. They are characterised by a frame structure consisting of the payload, containing the data and the header with control information. When data pass through each of the layers, an overhead is added to it, which increases the required capacity.

The $I_{u\text{-}ps}$ interface carries the PS traffic and connects the RNC with the SGSN. The data are routed to the SGSN through the tunnelling protocol (GTP). Protocols GPT and UDP contain headers that are12 bytes and 8 bytes long, respectively. The IP layer also adds a header of variable lengths with the minimum being 20 bytes long.

2.2 Radio Interface

UMTS in Europe uses only FDD in the band [1920, 1980] MHz for the UL and [2110, 2170] MHz for the DL [Moli05].

The WCDMA (Wideband Code Division Multiple Access) air interface used in UMTS is a wideband Direct-Sequence Code Division Multiple Access (DS-CDMA) system. User information bits are spread over a wide bandwidth by multiplying the user data with quasi-random bits (called chips) derived from CDMA (Code Division Multiple Access) spreading codes.

The chip rate of 3.84 Mcps leads to a carrier bandwidth of approximately 5 MHz. The inherently wide carrier bandwidth of WCDMA supports high user data rates, and also has some performance advantages, such as increased multipath diversity.

WCDMA supports highly variable user data rates. Each user is allocated 10 ms frames, during which the user data rate is kept constant, data capacity being able to change from frame to frame. This fast radio capacity allocation will typically be controlled by the network, to achieve optimum throughput for packet data services. Table 2.1 summarises the main parameters related to the WCDMA air interface.

Table 2.1. - Main WCDMA parameters (adapted from [HoTo04]).

<table>
<tr><th>Parameter</th><th colspan="2">Value</th></tr>
<tr><td>Multiple access method</td><td colspan="2">DS-CDMA</td></tr>
<tr><td>Duplexing method</td><td colspan="2">FDD</td></tr>
<tr><td>Base station synchronisation</td><td colspan="2">Asynchronous</td></tr>
<tr><td>Chip rate</td><td colspan="2">3.84 Mcps</td></tr>
<tr><td>Service multiplexing</td><td colspan="2">Multiple services with different quality of service requirements multiplexed on one connection</td></tr>
<tr><td>Multi rate concept</td><td colspan="2">Variable spreading factor and multicode</td></tr>
<tr><td>Detection</td><td colspan="2">Coherent using pilot symbols</td></tr>
<tr><td rowspan="5">Channelisation</td><td>Use</td><td>DL: MT separation
UL: Channel separation</td></tr>
<tr><td>Duration</td><td>DL: 4 – 512 chip
UL: 4 – 256 chip</td></tr>
<tr><td>Number</td><td>Spreading factor</td></tr>
<tr><td>Family</td><td>OVSF</td></tr>
<tr><td>Spreading</td><td>Yes</td></tr>
<tr><td rowspan="5">Scrambling</td><td>Use</td><td>DL: Sector separation
UL: MT separation</td></tr>
<tr><td>Duration</td><td>38 400 chip</td></tr>
<tr><td>Number</td><td>DL: 512
UL: >1 000 000</td></tr>
<tr><td>Family</td><td>Gold or S(2)</td></tr>
<tr><td>Spreading</td><td>No</td></tr>
</table>

WCDMA uses two types of codes for spreading and multiple access: channelisation and scrambling.

The former spreads the signal by extending the occupied bandwidth in accordance to the basic principle of CDMA. The latter do not lead to bandwidth expansion, but help to distinguish among cells and users. These codes are Orthogonal Variable Spreading Factor (OVSF) ones, which allow the Spreading Factor (SF) to be changed orthogonality between codes to be maintained.

Spreading and modulation of data and control channels are different for the DLand UL. In DL, data and control channels are time-multiplexed and the resulting single bit stream is then Quadrature Phase Shift Keying (QPSK) – modulated. The resulting I- and Q-signals are spread separately using the same channelisation code. The complex scrambling code is then applied to the resulting complex signal. Finally, the scrambled complex signal is fed into the complex modulator. With the Rel-99, a number of schemes are available to extract maximal functionality from the system, one of them being transmission diversity, which is mainly in order to improve the reception quality in DL (increasing the downlink capacity). Two antennas are used at the BS and the UE combines the received signals. Another scheme is soft handover, which is a form of macro-diversity. Two sectors from two different BSs communicate simultaneously with the UE (two radio interface links are used, using two power control loops). Both signals are received and used by the UE.

In UL, the RNC selects the best frame, at each interleaving period (every 10-80 ms). Softer handover is used when the two sectors belong to the same BS. The two signals (using the same power control loop) can be combined within the receivers of the UE and the BS. Two separate codes are used in DL, so that the UE can separate the signals. The difference is that in UL, combining is performed within the same BS. This combining can be performed in the baseband receiver of the BS, in comparison to the more drastic selection in the RNC performed by the soft handover. Compressed mode is also used when parallel measurements to another UTRAN frequency has to be performed, a pure parallel measurements would require dual receiver. The Compressed Mode avoids this complexity, by operating the receiver in a slotted mode leaving some time for the UE to perform measurement on another frequency.

The aspects mentioned so far are common to Releases 99, 5 and 6. However, Releases 5 and 6 introduced changes in order to implement HSDPA and HSUPA, respectively. Some changes that are worth mentioning concern features such as SF, modulation, power control and the support of soft handover, which are summarised in Table 2.2. In HSDPA, the modulation is no longer fixed. In cellular communication systems, the quality of a signal received by a UE depends on a number of factors: the distance between the desired and interfering BSs, path loss exponent, log-normal shadowing and short term Rayleigh fading, and noise. In order to improve system capacity, peak data rate and coverage reliability, the signal transmitted to and by a particular user is modified to account for the signal quality variation through a process commonly referred to as link adaptation. Adaptation Modulation and Coding (AMC) provides the flexibility to match the modulation-coding scheme to the average channel conditions for each user. With AMC, the power of the transmitted signal is held constant over a frame interval, and the modulation and coding format is changed to match the current received signal quality or channel conditions. In a system with AMC, users close to the Node B are typically assigned higher order modulations with higher code rates (e.g., 64 QAM with R=3/4 turbo codes), but the modulation-order and/or code rate will decrease as the distance from Node B increases, Figure 2.3.

Table 2.2. - Compilation of fundamental properties (Releases 99, 5, 6 and 7), (adapted from [Arau09]).

	Release 99	*Release 5 (HSDPA)*	*Release 6 (HSUPA)*	*Release 7 (HSPA +)*
SF	Variable	Fixed and equal to 16	Variable	Variable
Modulation	Fixed (BPSK for the UL and QPSK for the DL)	Variable (16-QAM or QPSK)	Fixed (BPSK)	QPSK (UL); 16QAM (UL/DL), 64QAM (UL/DL)
Power control	Yes	No	Yes	No
Soft handover	Yes	No	Yes	No
Maximum data rates [Mbps]	0.384 (urban, outdoor), 2 (indoor)	14.4 (TTI=2ms)	5.7 (TTI variable)	21.1 DL 42 DL (MIMO 2×2, 64QAM)

The benefits of AMC are the increased average cell throughput, the reduced interference variation, and the higher data rates for users in favourable positions.

Automatic Repeat Request (ARQ) is an error detection mechanism used in the link layer. Basically, the receiver informs the transmitter that a block has been received incorrectly and the transmitter resends it. It can be done with a Stop And Wait (SAW) procedure, where the transmitter sends a block and waits for the receiver response, before sending a new block or resending the incorrect one. This is not very efficient, since the transmitter is inactive until it gets a response. It can be improved with the dual channel SAW, where two SAW instances work alternatively in the same channel. The solution used for HSDPA is N-channel SAW, which is a generalised version of the dual channel and can be used by multiple users.

Hybrid ARQ is a combination of ARQ and Forward Error Correction (FEC). The erroneous blocks are kept and are used for a combined detection with the retransmissions. There are various types: Code Combining, Incremental Redundancy (IR), and Chase combining.

HSDPA uses IR and Chase combining. In the former, successive retransmissions of an erroneous block are sent with additional redundancy that is increased with each retransmission; with the latter, the retransmissions are identical to the original, but when combined for detection they are weighted with their SNR.

In Release 5, the new functionality of hybrid ARQ and HSDPA scheduling are included in the MAC layer. In UTRAN, these functions are included in a new entity called MAC-hs, which is terminated in the Node B.

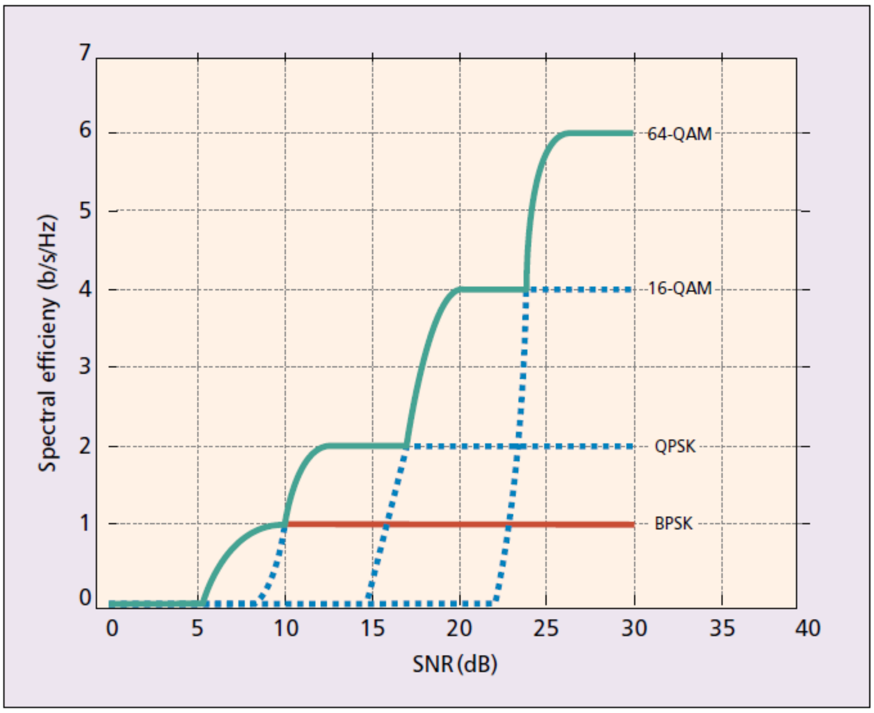


Figure 2.3. - Spectral efficiency variation, according to modulation behaviour, (extracted from [CEGH02]).

HSUPA bit rates depend upon the protocol stack layer from where they are measured. They also depend upon the spreading factor, number of E-DPDCH channelisation codes, coding rate and modulation scheme. The set of HSDPA UE categories specified by 3GPP is presented in Table A.2.

2.3 Interference, Coverage and Capacity

The trade-off between capacity and interference is of key importance in cellular networks. In UMTS, capacity depends strongly on the number of users, and on their type of service, via the interference margin, the sharing of transmitting power at the BS, and the number of available OVSF codes [Corr10].

The interference margin is given by:

$$M_{\mathrm{I}\,[dB]} = -10\log(1-\eta) \tag{2-1}$$

where:

- η is the load factor.

The load factor is not the same for UL and DL, because codes are different in UL and DL; it should not be higher than 50% in UL and 70% in DL. These factors, for a user j, are given by [Corr10]:

$$\eta_{UL} = \left(1 + I_{inter\ n} \sum_{j=1}^{N_U} \frac{1}{1+\frac{R_C/R_{b\ j}}{\rho_{N\ j} . F_{a\ j}}}\right) \tag{2-2}$$

$$\eta_{DL} = (1 + I_{inter\ n}) \sum_{j=1}^{N_U} F_{a\ j} \times \frac{\rho_{N\ j}}{R_C/R_{b\ j}} \left[\left(1-\alpha_j\right) + I_{inter\ n\ j}\right] \tag{2-3}$$

where:

- η_{UL} is the UL load factor;
- $\eta_{DL}$is the DL load factor;
- $I_{inter\ n}$ is the normalised inter-cell interference (between [40,60]% in UL and 0% in DL);
- N_U is the number of active users;
- R_C is the chip rate of WCDMA (3.84 Mcps);
- $R_{b\ j}$ is the data rate associated to service of user *j*;
- $\rho_{N\ j}$ is the SNR of user *j*;
- $F_{a\ j}$ is the activity factor of user *j* (50% for voice and 100% for data);
- α_j is the code orthogonality factor of user *j* (typically in [0.5,0.9]).

When η_{UL} or η_{DL} approach unit, the network reaches its pole capacity, and the noise rise reaches infinite. The main difference between the two load factors has to do with, in DL, the fact that the transmission power does not vary with the number of users, but it is shared among them, whereas in UL, each MT has its own power. Therefore, coverage depends more on the load in DL; even with a low load in DL, coverage decreases as a function of the number of users.

The radius of a given cell can be estimated using the definition of the path loss and the model of the average power decay with distance. The radius of a cell is given by [Corr10]:

$$R_{[km]} = 10^{\frac{P_{t[dBm]} + G_{t[dBi]} - P_{r[dBm]} + G_{r[dBi]} - L_{ref[dB]}}{10 . a_{pd}}} \tag{2-4}$$

where:

- P_t is the power fed to the transmitting antenna;
- G_t is the gain of the transmitting antenna;
- P_r is the power available at the receiving antenna;
- G_r is the gain of the receiving antenna;
- L_{ref} are propagation losses;
- a_{pd} is the average power decay.

The DL transmission power also limits cell capacity. Therefore, one should be able to calculate the total BS transmission power, which can go up to 20W (43 dBm) in Release 99, and to 40 W (46 dBm) in Release 5 [HoTo07]. Part of that power has to be allocated to the common channels that are transmitted independently of traffic ones. Hence, the total BS transmission power can be separated into two components:

$$P_{Tot}^{BS} = P_{T_X}^{BS} + P_C^{BS} \tag{2-5}$$

where:

- P_C^{BS} is BS transmission power allocated to common channels;
- $P_{T_X}^{BS}$ is BS transmission power allocated to traffic channel, given by:

$$P_{T_X}^{BS} = \frac{N_0 \times R_C \times \sum_{j=1}^{N_U} \upsilon_j \frac{\rho_{Nj}}{G_{Pj}} \times \overline{L_{pJ}}}{1-\overline{\eta_{DL}}} \tag{2-6}$$

  - N_0 is the noise spectral density at the MT;
  - R_C is WCDMA chip rate (3.84 Mcps);
  - $\overline{L_{pJ}}$ is the average path loss between the BS and user *j;*
  - $\overline{\eta_{DL}}$ is the average DL load factor value across the cell.

The power allocated to each channel group is variable according to the operating Release. In Release 99, the transmission power is shared with a ratio of 75% for data channels, and 25% for signalling and control. In HSPA and in HSPA+, the enhancements done over the network signalling result in a greater portion of power to execute these tasks properly, therefore, 40% of transmission power is delivered to the common channels, it being the sum of all common channels from Release 99 and on top of those the addition of the specific ones from HSPA. The remaining 60% is taken for the dedicated user's channels in DL.

HSPA pushes more functionalities to the BS and allows a flat architecture, which improves the efficiency and the QoS capabilities for packet services [HoTo07].

Release 99 typically uses E_b/N_0 as an approximation to SNR:

$$\rho_N \approx E_b/N_0 \tag{2-7}$$

where:

- E_b is energy per bit;
- N_0 is noise spectral density.

Depending on the service, and on the bit rate, the associated SNR values are different, e.g., in voice (12.2 kbps) the SNR should be within [4.8, 8.8] dB, while for data (384 kbps) it should be in [0.4, 3.2] dB, [Corr10]. However, to evaluate HSDPA performance, the E_b/N_0 metric is not used.

The number of users, their location and handling services define the interference on the radio interface, thus, coverage and available capacity. Three main parameters are limiting capacity: the number of available codes (maximum15 codes) in DL, the BS transmission power, and the system load [HoTo07]. The number of available codes in DL defines the maximum number of simultaneous active services handled by users. The number of codes employed for each service type, voice or data, depends on the number of active users for a particular service, and has to be higher than the total amount of codes for a channelisation system type, otherwise the system is congested:

$$N_{mc} \geq N_{vc} \times N_{vu} + N_{dc} \times N_{du} \tag{2-8}$$

where:

- N_{mc} is the maximum number of codes for a channelisation system type;
- N_{vc} is the number of codes for voice service;
- N_{dc} is the number of codes for data services;
- N_{vu} is the number of users connected in voice service;
- N_{du} is the number of users connected in a data services.

Each service is characterised by a specific channelisation type, according to the demanded radio resources, with direct impact in the final bit rate delivered by the network to the user; that is why for HSDPA one has a fixed SF of 16.

There is a direct relation between the bit rate per service and the number of codes made available by the network, Table 2.3.

Table 2.3. - Bit rate and number of codes relation, (adapted from [Lope10]).

Services	***Bit Rate [kbps]***		***Applicational Codes assigned per user***	
	Min	Max	Min	Max
Voice	15		1	
Streaming	300	5 600	5	15
WWW	64	384	1	5
FTP	384	14 400	10	15
e-mail	64	384	1	5

A correspondence between codes, services and allowed number of users can be a total of 15 channel codes for a 16 SF channelisation code type, Table 2.4.

Table 2.4. - Correspondence between services and code capacity request (adapted from [Lope10]).

Service	***Codes per user***	***Number of available codes***	***Number of users/service per SC or carrier***
HSDPA 3.6 Mbps	1 SF 16	5	3
HSDPA 7.2 Mbps		10	1
HSDPA 14.4 Mbps		15	1

From a user point of view, the most important output, when a data service is being requested, is bit rate, which means satisfaction in cases where there is a high demand and a high value is reached. Secondly, it is important to have low interference, representing low retransmission rate, higher integrity (low packet drops and error correction on bits transmission), low transmission delay and a better efficiency on employed modulation. Under such conditions, a better QoS can be reached, meaning that the service is provided with no interruptions, and keeping the thresholds of integrity and delays inside reasonable values. Another important parameter is latency (service response time, end to end), relevant mainly for interactive services, but not directly dependent on RRM algorithms, because it is influenced by other interfaces (not just the radio interface). Other important parameters that influence the final grade of service (GoS) are: signalling efficiency, correction efficiency, modulation and rate of retransmissions.

2.4 Services and applications

To manage access to different services and optimise system capacity, 3GPP defined four classes of services based on their QoS requirements: Conversional, Streaming, Interactive and Background. There are some important factors to distinguish them, such as transfer delay, guaranteed bit rate, and the different priorities. When defining the UMTS QoS classes, also referred to as traffic classes, the restrictions and limitations of the air interface have to be taken into account.

The Conversational class is mainly intended for speech services (e.g., CS or VoIP). The conversational class is characterised by low end-to-end delay and symmetric or nearly symmetric traffic between UL and DL in person-to-person communications. The maximum end-to-end delay is given by the human perception of video and audio conversation: subjective evaluations have shown that the end-to-end delay has to be less than 400 ms [HoTo07]. Video telephony has ever tighter Bit Error Ratio (BER) requirements than voice, due to video compression. When it works on the PS domain, and in order to guarantee an efficient VoIP, service IP header compression and QoS differentiation are needed. This class has priority over others, because voice is a primary service and the one that is most required. Traffic is assumed to be symmetric in this case.

The Streaming class represents the audio and the video streaming. This type of service enables the end user to access the data before the transfer is complete, which is possible with the use of buffers in the final applications and a continuous stream transmission. In this class, traffic is not symmetric, thus, DL traffic is the most significant. The streaming class requires bandwidth to be maintained like the conversational class, but the streaming class tolerates some delay variations that are hidden by a dejitter buffer in the receiver. Fundamental characteristics for QoS in real time conversation are preserve time relation between information entities of the stream and conversational pattern. Streaming class traffic is asymmetric.

The Interactive class has a very asymmetric traffic and is tolerant to delay. This class includes Web browsing, online multiplayer games and push-to-talk applications, services that are based on PS connections. These are characterised by requesting response patters and preservation of payload contents. Nevertheless, there are upper limits to the tolerable delay, such as the time between choosing a certain website and its actual appearance on the screen, which should not exceed a few seconds. For online multiplayer games, Round Trip Time (RTT) is a very important parameter, especially in the real time action games, where the end-to-end delay should be below 100 ms [Carv09]. The interactive class is characterised by the request response pattern of the end user. At the message destination there is an entity expecting the message (response) within a certain time. Traffic is assumed to be asymmetric in this class.

The Background class covers services where transmissions delays are not critical (e.g., Short Messaging Service (SMS), Multimedia Messaging Service (MMS), E-mail), as opposed to the interactive one, where the end user is not waiting for a response within a certain time; however, this class is intolerant to transmissions errors. Applications in this traffic class only use resource transmissions when none of the other classes are active. Background class traffic is asymmetric.

The main distinguishing factor among the four traffic classes is how delay sensitive the traffic is: the Conversational class is meant for very delay-sensitive traffic, while the Background one is the most delay insensitive. Conversational and Streaming classes are mainly intended to be used to carry real-time traffic flows. Conversational real-time services, like video telephony, are the most delay sensitive applications and those data streams should be carried in Conversational class.

UMTS QoS classes are not mandatory for the introduction of any low-delay service. It is possible to support streaming video or conversational VoIP from an end-to-end performance point of view by using just the background QoS class. QoS differentiation becomes useful for the network efficiency during high load when there are services with different delay requirements. If the radio network has knowledge about the delay requirements of the different services, then it would be able to prioritise the services accordingly and improve the efficiency of the network utilisation

It is also important to know the usual bit rates and typical file dimensions associated to the different services. Table 2.5 shows some examples, which also includes the QoS priority list. If reduction strategies are applied, the first services to be reduced are the ones with the lower QoS priority (that corresponds to a higher priority value), according to the traffic classes shown in Table 2.6. The delay presented is the start-up service delay, and does not take the transport delay variation possible to occur in the transfer process into account.

Table 2.5. - Bit rates and applications of different services (adapted from [Arau09]).

Service	***Bit rate [kbps]***		***QoS priority***	***Delay [s]***	***Characteristics***		
	DL	**UL**			**Average volume [MB]**	**DL**	**UL**
Web	[512,1536]	[128, 512]	3	< 4 /page	Page size	0.3	0.02
Streaming	[512, 1024]	[64, 384]	4	< 10	Video size	9.6	0.2
FTP	[384, 2048]	[128, 512]	6	< 10	File size	10	

The scope of this thesis are the Streaming (VoD) and Interactive (FTP and Browsing) classes, however, all services will be covered in a generic way.

For all real time test services, calls are assumed to be generated according to a Poisson process, with a mean call duration of 120 s for speech.

Table 2.6. - UMTS Service Classes (adapted from [ETSI03]).

Traffic class	***Conversational***	***Streaming***	***Interactive***	***Background***
Fundamental characteristics	Preserve time relation (variation) between information entities of the stream Conversational pattern (stringent and low delay)	Preserve time relation (variation) between information entities of the stream	Request response pattern Preserve payload content	Destination is not expecting the data within a certain time Preserve payload content
Transfer delay [s]	Minimum Fixed <<1	Minimum Variable ~1	Moderate Variable <10	Large Variable >10
Symmetric	Yes	No	No	No
Bandwidth	Guaranteed bit rate	Guaranteed bit rate	No guaranteed bit rate	No guaranteed bit rate
Real time	Yes	Yes	No	No
Buffer	No	Yes	Yes	Yes
Switching Type	CS	CS	PS	PS
Highly delay sensitive	Yes	Yes	No	No
Preserves payload content	No	No	Yes	Yes
Priority	High	Medium-high	Medium	Low
Example of the application	Speech VoIP Video Telephony	VoD AoD Media Broadcast	WWW Location Based Interactive Games FTP	SMS Email

For non-real time services, the geometrical distribution is used (discrete representation of the exponential distribution), since the simulations are using discrete time scale. Figure 2.4 depicts a typical WWW browsing session, which consists of a sequence of packet calls. One only considers the packets from a source that may be at either end of the link, but not simultaneously. The user initiates a packet call when requesting an information entity. During a packet call, several packets may be generated, which means that the packet call is composed of bursty sequence of packets, which is taken into account in the traffic model. The burstyness during the packet call is a characteristic feature of packet transmission in the fixed network.

A packet service session contains one or several packet calls depending on the application. For example in a www browsing session a packet call corresponds the downloading of a www document.

After the document has entirely arrived to the terminal, the user will consume certain amount of time for studying the information. This time interval is called reading time. It is also possible that the session contains only one packet call. In fact, this is the case for a file transfer (FTP).

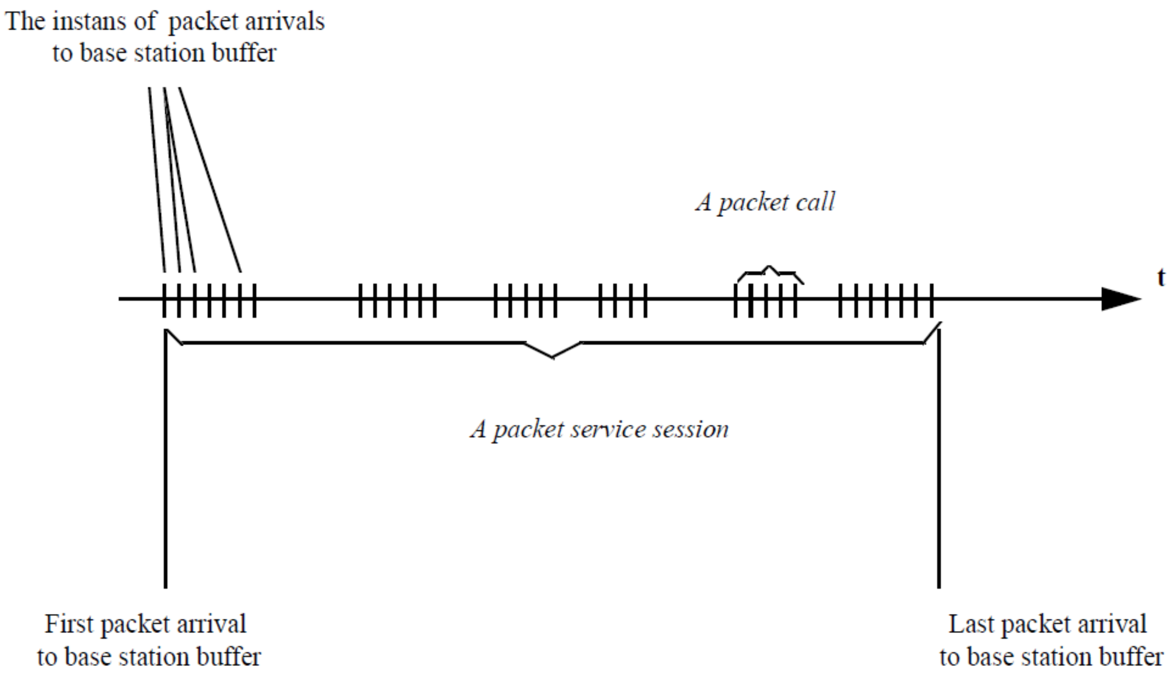


Figure 2.4. - Typical characteristic of a packet service session (extracted from [ETSI01]).

Parameters are:

- **Session arrival process** - The arrival of session set-ups to the network is modelled as a Poisson process. For each service there is a separate process.
- **Number of packet calls requests per session,** N_{pc} - is a geometrically distributed random variable (Geom) with a mean μ_{Npc} [packet calls],

$$N_{pc} \in Geom(\mu_{Npc}) \tag{2-9}$$

- **Reading time between two consecutive packet calls,** Dpc **-** This is a geometrically distributed random variable with a mean μD_{pc} [model time steps],

$$D_{pc} \in Geom(\mu_{Dpc}) \tag{2-10}$$

Note that the reading time starts when the last packet of the packet call is completely received by the user. The reading time ends when the user makes a request for the next packet call.

- **Number of datagrams within a packet call,** N_d - The traffic model should be able to catch the various characteristic features possible in the future UMTS traffic. For this reason different statistical distributions can be used to generate the number of packets. For example N_d can be geometrically distributed random variable with a mean μN_d [packet];

$$N_d \in Geom(\mu_{Nd}) \tag{2-11}$$

- **Inter arrival time between two consecutive packets inside a packet call,** D_d - This is a geometrically distributed random variable with a mean μD_d [model time steps],

$$D_d \in Geom(\mu_{Dd}) \tag{2-12}$$

- **Packet size,** S_d - The traffic model can use such packet size distribution that suits best for the traffic case under study. Pareto distribution with cut-off is used.

The normal Pareto distribution (without cut-off) is defined by:

$$\begin{cases} f_x(x) = \frac{\alpha\times k^{\alpha}}{x^{\alpha+1}},\ x \geq k \\ F_x(x) = 1 - \left(\frac{k}{x}\right)^{\alpha},\ x \geq k \\ \mu = \frac{k\times\alpha}{\alpha-1},\ \alpha > 1 \\ \sigma^2 = \frac{K^2\times\alpha}{(\alpha-2)\times(\alpha-1)^2}, \alpha > 2 \end{cases} \tag{2-13}$$

Packet Size is defined by the following formula:

$$\text{Packet Size} = \min_{(P,\, m)} \tag{2-14}$$

where:

- P is the normal Pareto distributed random variable (α=1.1, k=81.5 bytes),
- m is maximum allowed packet size, m=66 666bytes. The Probability Distribution Function (PDF) of the Packet Size becomes:

$$f_n(x) = \begin{cases} \frac{\alpha\times k^{\alpha}}{x^{\alpha+1}},\ k \leq x < m \\ \beta, \qquad\qquad x = m \end{cases} \tag{2-15}$$

where:

- β is the probability that $x>m$. It can easily be calculated as:

$$\beta = \left(\frac{k}{m}\right)^{\alpha},\ \alpha > 1 \tag{2-16}$$

Then it can be calculated as

$$\mu_n = \frac{\alpha\times k - m\times\left(\frac{k}{m}\right)^{\alpha}}{\alpha-1} \tag{2-17}$$

With the parameters above the average size μ_n = 480 bytes indicates that according to the values for α and k in the Pareto distribution, the average packet size m is 480 bytes. The average requested file size is $\mu_{Nd}\times\mu$= 25×480 bytes≈12kbytes. The inter-arrival time is adjusted in order to get different average bit rates at the source level.

2.5 Quality of Service and Quality of Experience

QoS is the ability of the network to provide a service at an assured service level. In order to provide the best Quality of Experience (QoE) to users in a cost-effective, competitive and efficient manner, network and service providers must manage network QoS and service provisioning efficiently and effectively. QoE is the term used to describe user perceptions of the performance of a service. Service integrity concerns throughput, delay, delay variation (or jitter) and data loss during user data transmission; service accessibility relates to unavailability, security (authentication, authorisation and accounting), activation, access, coverage, blocking, and setup time of the related bearer service; service retainability, in general, characterises connection losses [SoLC06].

The QoS of a wireless network is affected by attenuation, multi-path interference, spectrum interference (for example spread-spectrum interferences from neighbouring cells), noise (noise sources can be natural and man-made such as radio, TV and other radio-frequency transmission), mobility (affects handover and resource utilisation, management) and limited capacity (resources are costly).

QoS and QoE are so interdependent, that they have to be studied and managed with a common understanding, Figure 2.5. The aim of the network and services should be to achieve the maximum user rating (QoE), while network quality (QoS) is the main building block for reaching that goal effectively. QoE, however, is not just limited to the technical performance of the network, there are also non-technical aspects, which influence the overall user perception.

- QoS encompasses all functions, mechanisms and procedures in the cellular network and terminal that ensure the provision of the negotiated service quality between the UE and the core network (CN).
- QoE is how a user perceives the usability of a service when in use – how satisfied they are with a service in terms of, for example, usability, accessibility, retainability and integrity of the service. Service accessibility relates to unavailability, security, activation, access, coverage, blocking, and setup time of the related bearer service; service retainability, in general, characterises connection losses.

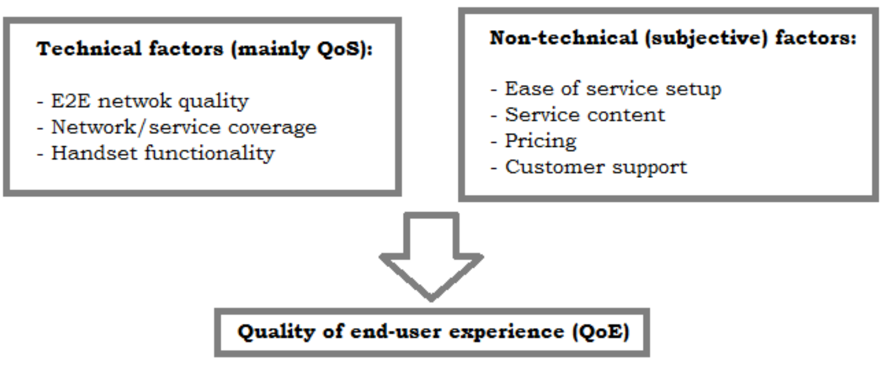


Figure 2.5. - QoE is affected by technical (QoS) and non-technical aspects of service (extracted from [SoLC06]).

QoE and QoS concepts are shown in Figure 2.6, QoE is expressed in “feelings” rather than metrics. QoS relates to all mechanisms, functions and procedures in the network and terminal that implement the quality attributes (bearer service) negotiated between the UE and the CN.

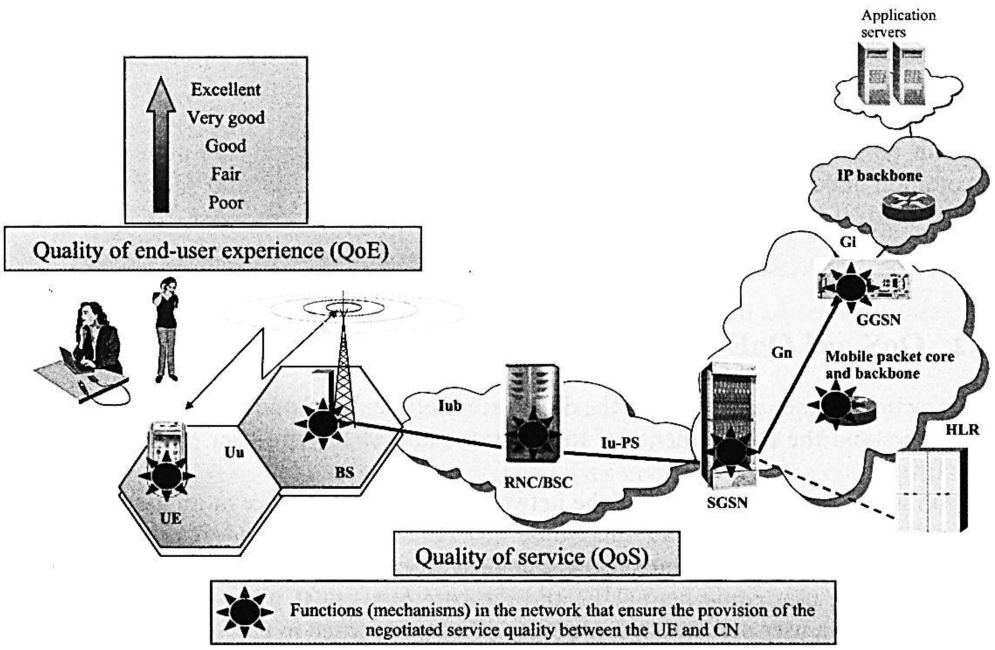


Figure 2.6. - QoE and QoS concepts (extracted from [SoLC06]).

QoE is a function of multiple protocol layers and network elements. Radio interface is the bottleneck, with its restrictions in bandwidth and coverage.

Class priority is thus enforced at call admission control and during scheduling of non-real time (NRT) data. New connection requests from the service with higher priority (conversational RT service, on Conversational class) are admitted Into the system if there is enough power remaining in the BS power budget to compensate for the estimated path loss by the MT on an open loop basis, and the individual dedicated traffic channel power limit is not violated, Figure 2.7. The second priority service (streaming RT service, on Streaming class) must meet the same constraints but the notion of link quality is added in the admission process.

Although QoE is subjective in nature, at the same time it is very important. The international bodies ETSI, ANSI and ITU (European, American and International respectively) have different views. ETSI has put a lot of effort into explaining QoS metrics from an end-user point of view, defined quality of service parameters, and their computations are based on field measurements. An ANSI view specifies the classes of QoS sufficient to support business multimedia conferencing on IP networks, defined as being equivalent to legacy conference system performance. The ITU view defines a model for multimedia QoS categories from an end-user viewpoint, by considering user expectations for a range of multimedia applications, identifying eight distinct categories, based on tolerance to information loss and delay.

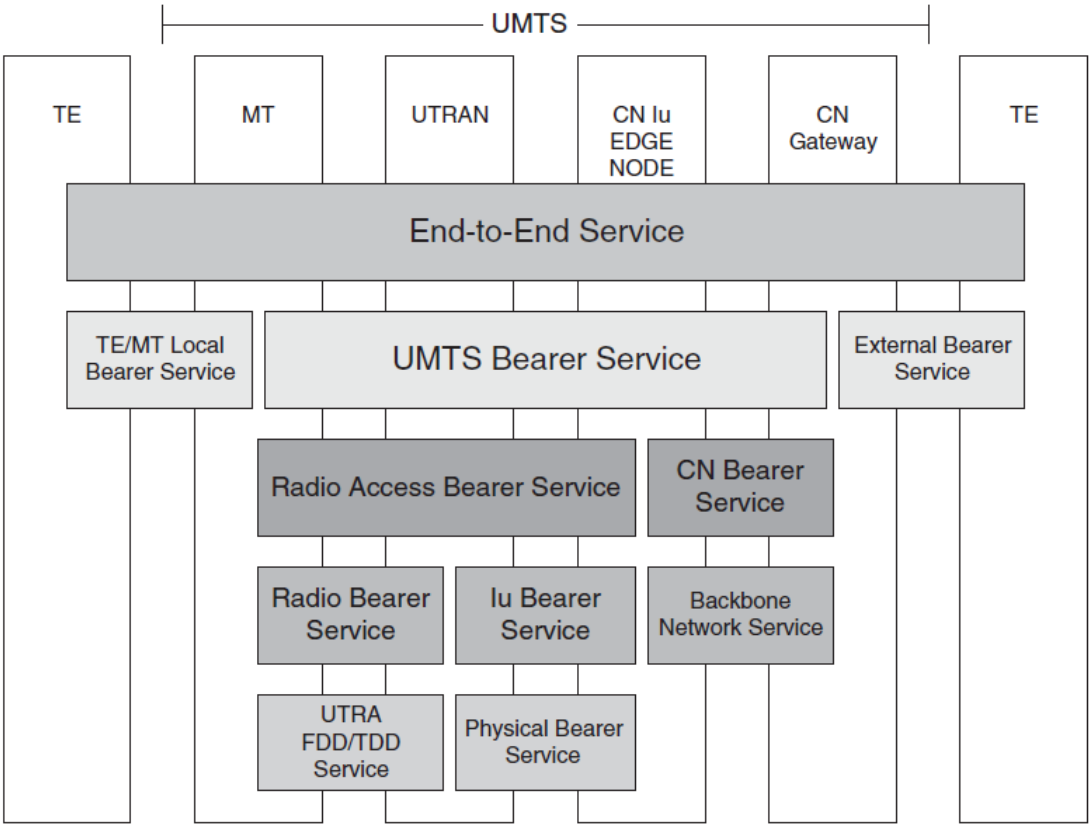


Figure 2.7. - Architecture of a UMTS bearer service (adapted from [HoTo07]).

QoS monitoring consists of collecting/processing performance statistics, usage data and QoS related faults. In order to obtain end-to-end QoS monitoring, the Network Elements (NE), the Element Management Layer (EML) and Network Management Layer (NML) must all be involve with the QoS monitoring process. Alarm and performance collection is done at the Network Element layer and alarm/performance aggregation, report generation, and analysis is done at the Element Management and Network Management Layers [ETSI05], Figure 2.8.

The functions of the QoS monitoring process are:

- Manage QoS fault conditions received from NE.
- Retrieve QoS Performance data from NE.
- Collect and process usage data.
- Generate QoS Reports – trend analysis of key QoS parameters.
- Audit/Analyse collected QoS parameters against expected values.

The Network Element is responsible for collecting performance measurements, usage data and generating alarms. It supports the following functions:

- Collects performance data according to the definition of the measurements and to return results to the EML.

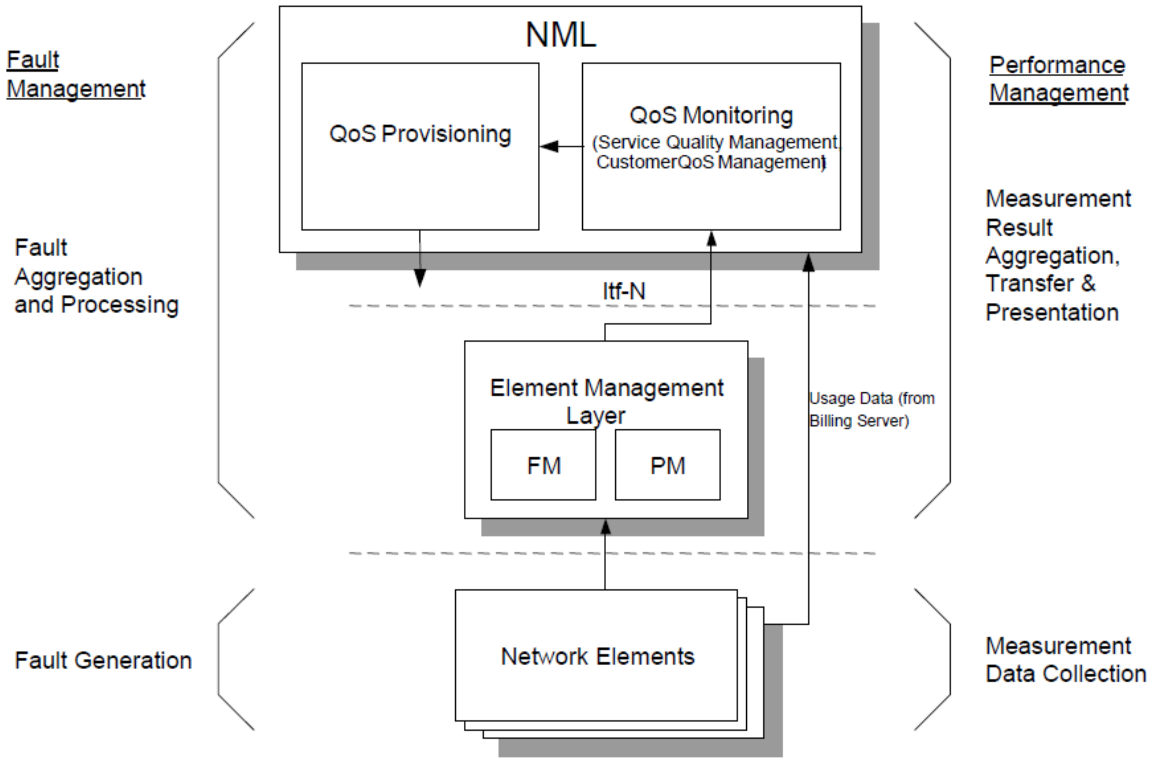


Figure 2.8. - QoE and QoS Monitoring (extracted from [ETSI06]).

- Collects usage data and forward the data to mediation.
- Performs the following fault management functions: fault detection, generation of alarms, clearing of alarms, alarm forwarding and filtering, storage and retrieval of alarms in/from the NE, fault recovery, configuration of alarms.

The Element Management Layer is responsible for aggregating and transferring the collected performance measurements and generated alarms/events. The Element Management Layer provides the following functions:

- Performance Management (PM);
- Fault Management (FM).

From a QoS monitoring perspective, the NML is responsible for the collection and processing of performance, fault, and usage data. The NML QoS monitoring layer provides the following functions:

- Service Quality Management (SQM) is responsible for the overall QoS as it interacts with other functional areas to access monitored information, processes that information to determine quality metrics, and initiates corrective action when quality level is considered unsatisfactory. Inputs to SQM include both performance and fault data.
- Customer QoS Management (CQM) – includes monitoring, managing, and reporting QoS customers received against what has been promised to the customer in Service Level Agreements and any other service related documents. Inputs to CQM include data from SQM and usage data.

One can identify subsets of QoS attributes that allow an adequate number of priority “pipes”:

- UMTS Traffic Class (TC);
- Traffic Handling Priority (THP);
- Allocation/retention priority (ARP);
- Maximum bit rate;
- Guaranteed bit rate.

The QoS profile consists of the following attributes [ETSI03] and [Sold05]:

- Traffic Class - Application type for which the UMTS bearer service is optimised. The main distinguishing factor between these QoS classes is how delay sensitive the traffic is; Conversational class is meant for traffic which is very delay sensitive, while Background class is the most delay insensitive traffic class. Conversational and Streaming classes are mainly intended for carrying real-time traffic flows, such as video telephony and audio/video streams. Interactive class and Background are mainly meant for carrying traditional Internet applications like WWW, Email, Telnet, FTP and News. Due to delay requirements, compared to Conversational and Streaming classes, both provide better error rate by means of channel coding and retransmission. Traffic in the Interactive class has higher priority in scheduling than Background class traffic, so background applications use transmission resources only when interactive applications do not need them. However, these are only typical examples of usage of the traffic classes.
- Maximum Bit Rate (kbps) – maximum number of bits delivered by UMTS and to UMTS at a SAP within a period of time, divided by the duration of the period. The traffic is conformant with Maximum Bit Rate as long as it follows a token bucket algorithm where token rate equals Maximum Bit Rate and bucket size equals Maximum SDU size. The Maximum bitrate is the upper limit a user or application can accept or provide. All UMTS bearer service attributes may be fulfilled for traffic up to the Maximum Bit Rate depending on the network conditions.
- Guaranteed Bit Rate (kb/s) – guaranteed number of bits delivered by UMTS at a SAP within a period of time (provided that there is data to deliver), divided by the duration of the period. The traffic is conformant with the guaranteed bit rate as long as it follows a token bucket algorithm where token rate equals guaranteed bit rate and bucket size equals Maximum SDU size. The Guaranteed bit rate may be used to facilitate admission control based on available resources, and for resource allocation within UMTS.
- Delivery Order (y/n) - indicates whether the UMTS bearer shall provide in-sequence SDU delivery or not. Derived from the user protocol and specifies if out-of sequence SDUs are acceptable or not. Whether out-of-sequence SDUs are dropped or re-ordered depends on the specified reliability.
- Maximum SDU Size (octets) - Maximum SDU size for which the network must satisfy the negotiated QoS. The maximum SDU size is used for admission control and policing and/or optimising transport (optimised transport in for example the RAN may be dependent on the size of the packets).

- SDU Format Information (bits) - List of possible exact sizes of SDUs. RAN needs SDU size information to be able to operate in transparent RLC protocol mode, which is beneficial to spectral efficiency and delay when RLC re-transmission is not used.
- SDU Error Ratio - Fraction of SDUs lost or detected as erroneous. It is defined only for conforming traffic. It is used to configure the protocols, algorithms and error detection schemes, primarily within RAN.
- Residual Bit Error Ratio - Indicates the undetected bit error ratio in the delivered SDUs. If no error detection is requested, Residual bit error ratio indicates the bit error ratio in the delivered SDUs. As above, it is used to configure radio interface protocols, algorithms and error detection coding.
- Delivery of Erroneous SDUs (y/n/-) - Indicates whether SDUs detected as erroneous must be delivered or discarded.
- Transfer Delay (ms) - Indicates maximum delay for 95th percentile of the distribution of delay for all delivered SDUs during the lifetime of a bearer service, where delay for an SDU is defined as the time from a request to transfer an SDU at one SAP to its delivery at the other SAP. This attribute allows RAN to set transport formats and ARQ parameters. Transfer delay of an arbitrary SDU is not meaningful for a bursty source, since the last SDUs of a burst may have long delay due to queuing, whereas the meaningful response delay perceived by the user is the delay of the first SDU of the burst.
- Traffic Handling Priority - Specifies the relative importance for handling of all SDUs belonging to the UMTS bearer compared to the SDUs of other bearers. Within the Interactive class, there is a definite need to differentiate between bearer qualities. This is handled by using the traffic handling priority attribute, to allow UMTS to schedule traffic accordingly. By definition, priority is an alternative to absolute guarantees, and thus these two attribute types cannot be used together for a single bearer.
- Allocation/Retention Priority - It specifies the relative importance compared to other UMTS bearers for allocation/retention of the UMTS bearer. The Allocation/Retention Priority attribute is a subscription attribute that is not negotiated from the MT. In situations where resources are scarce, the relevant network elements can use the ARP to prioritise bearers with a high Allocation/Retention Priority over bearers with a low ARP when performing admission control.
- Source Statistics Descriptor (‘speech’/’unknown’) - It specifies characteristics of the source of submitted SDUs.
- Signalling Indication (Yes/No) - Indicates the signalling nature of the submitted SDUs. This attribute is additional to the other QoS attributes and does not over-ride them. This attribute is only defined for the Interactive traffic class. If signalling indication is set to ‘Yes’, the UE should set the traffic handling priority to ‘1’. This attribute permits enhancing the RAN operation accordingly. An example use of the Signalling Indication is for IMS signalling traffic.
- Evolved Allocation/Retention Priority - Enhances the Allocation/Retention Priority attribute with an increased value range of the priority level and additional information about the pre-emption capability and the pre-emption vulnerability of the bearer. The pre-emption capability information defines whether a bearer with a lower priority level should be dropped to free up the required

resources. The pre-emption vulnerability information whether a bearer is applicable for such dropping by a pre-emption capable bearer with a higher priority value, Table 2.7.

Table 2.7. - Value ranges for Radio Access Bearer Service Attributes for UTRAN (adapted from [ETSI03]).

Traffic class	***Conversational***	***Streaming***	***Interactive***	***Background***
Maximum bitrate (kbps)	≤16000	≤16000	≤16000-overhead	≤16000-overhead
Delivery order	Yes/No	Yes/No	Yes/No	Yes/No
Maximum SDU size (octets)	≤1 502	≤1 502	≤1 502	≤1 502
Delivery of erroneous SDUs	Yes/No/-	Yes/No/-	Yes/No/-	Yes/No/-
Residual BER	$5*10^{-2}$ - 10^{-6}	$5*10^{-2}$ - 10^{-6}	$4*10^{-3}$ - $6*10^{-8}$	$4*10^{-3}$ - $6*10^{-8}$
Transfer delay (ms)	<80	<80	-	-
Guaranteed bit rate (kbps)	≤16000	≤16000	-	-
Traffic handling priority	-	-	1,2,3	-
Allocation/Retention priority	1, 2, 3	1, 2, 3	1, 2, 3	1, 2, 3

These attributes are used to define the level of quality of the UMTS Bearer Service, but all parameters are not used for all quality classes.

The Node B scheduler determines the sequence in which HSDPA UE are assigned access to the 2ms Transmission Time Interval (TTI). The simplest form of scheduler is a round robin scheduler which assigns the 2ms TTI in rotation without accounting for the channel conditions experienced by each UE. A round robin scheduler has a linear impact upon the throughput experienced by each UE. Increasing the number of UE by a factor of 2 causes the individual UE throughput to decrease by a factor of 2. This results directly from the UE being allocated half the original number of TTIs.

A proportional fair scheduler accounts for the RF channel conditions when allocating the 2ms TTI. This type of scheduler can increase both the individual connection throughputs and the total cell throughput. A proportional fair scheduler does not offer any benefit over a round robin scheduler if the quantity of HSDPA traffic is relatively low and there is only a single HSDPA connection within the cell, i.e., the single connection is allocated all TTIs in both cases. The benefits of a proportional fair scheduler increase as the number of HSDPA UE increases. A proportional fair scheduler attempts to allocate a TTI to the UE that is experiencing the best short-term channel conditions relative to its average channel conditions. This approach helps to avoid TTIs being scheduled during deep fades.

The number of TTIs allocated over a longer period of time does not change relative to a round robin scheduler, i.e., each UE is allocated an equal share of the TTI. The benefit of using a proportional fair scheduler results from allocating the TTI at the best possible instants. This requires the scheduler to track the short-term channel conditions experienced by each UE. Locating the scheduler within the Node B rather than within the RNC reduces delay and increases the availability of RF channel condition information. Nevertheless, it remains difficult to track the conditions unless they are changing relatively slowly. The benefit of a proportional fair scheduler increases when UEs have relatively low speeds. Higher UE speeds cause the RF channel conditions to change prior to scheduling the next TTI.

From a theoretical perspective, the maximum HSUPA achievable throughputs without considering the limitations imposed by the UE transmit power capability. In practice, the maximum bit rates are only achieved when the UE has sufficient transmit power, i.e., the link loss is not large, and the Node B packet scheduler has allocated a relatively high power ratio. The total HSUPA cell throughput is obtained by summing the throughputs achieved by each connection. However, the interference margin at the Node B receiver represents a shared resource. If the allocated power ratios cause the planned maximum interference margin to be reached, then increasing the number of connections will decrease both the allocated power ratio per connection and the corresponding throughput per connection.

The throughput figures presented in Table A.3. can be derived using the equation below:

$$\chi = \sum_{n=1}^{\#codes} \frac{R_C}{SF_n} \times Bps \times C_R \qquad (2\text{-}18)$$

where:

- χ is Throughput;
- R_C is Chip rate is 3.84Mcps;
- SF_n is the spreading factor, ranges from 256 to 2;
- *Bps* is bits per symbol (BPSK maps 1, 4PAM maps 4 bits onto each symbol);
- C_R is coding rate references the throughput result to the top of the Physical layer.

A low coding rate indicates that the physical layer introduces a large quantity of redundancy to help protect the information bits as they are transferred across the air-interface. A coding rate of 0.5, means that the throughput at the top of the physical layer is half of the throughput at the bottom of the physical layer. The resulting throughput at the top of the physical layer is applicable to a single E-DPDCH channelisation code so the total throughput is obtained by summing the throughputs generated by each code.

The throughput figures presented in Table A.4 do not necessarily reflect the maximum throughput that can be achieved in practice. Not all HSUPA UEs are capable of achieving the complete set of bit rates and UEs are categorised according to their ability.

2.6 Metrics

Key Performance Indicators (KPIs) are a minimum set of metrics for tracking system progress towards a performance target. Because web browsing is very much an Interactive service, the total download time of a webpage should be lower than 4-10 s. The acceptable level depends on if the click-to-content time refers to the time when the first parts of the webpage are shown (text), or the total page (text and pictures).

A common function in Streaming applications is adaptation of the content bit rate, because end-user experience for streaming depends a lot on the bit rate, which varies between networks.

What end-users experience is the average applications level bit rate, which is often lower than the maximum bit rate provided by the radio technology. One explanation for this is that in browsing, HTTP and TCP reside in between the application and the radio protocols. The packet round trip time (RTT-time it takes to send a small packet from a computer to a server and back again) is one fundamental property that affects the efficiency of HTTP and TCP. If the packet RTT is large, it takes a long time before there is a response back from the server. The packet RTT determines how fast TCP can establish a connection and, in some cases, also the maximum sustainable bit rate. The end user can hence experience the network as slow even if the network offers high radio bit rates.

Some services need a constant minimum bit rate without interruptions throughout the service delivery. If the constant minimum bit rate is not delivered, the user experiences irritating breaks or distortions. A stable bit rate is achieved by keeping the radio connection active, by providing minimal mobility interruption times and by controlling network resources with QoS procedures.

Table 2.8 provides some end-user KPIs and network requirements.

Table 2.8. - Main KPI per service.

Application	***KPIs***
Video streaming	Initial delay and average delays video average delay
Web browsing (HTTP)	Initial delay and average throughput during Web download
File download	Initial delay and average delay

In basic connected stat us following a successful call origination, or termination mode, UL an DL quality can be estimated by means of several metrics:

- **Block Error Rate** – BLER should be close to the signalled BLER target during a call. Increased DL BLER indicates that the required E_b/N_t cannot be fulfilled. This could indicate incorrect outer loop power control or that the required Dedicated Physical Channel (DPCH) power is not available, either due to the load on the network, or because the maximum allowed power has been reached. BLER is an analysis of transmission errors on the radio interface. It is based on analysis of cyclic redundancy check (CRC) results for radio link control (RLC) transport blocks and computed by defining the relation between the number of RLC transport blocks with CRC error indication and the total number of transmitted transport blocks as expressed below:

$$\beta = \frac{\sum \rho}{\sum \varepsilon_{RLC}} \times 100\% \tag{2-19}$$

where:

- β is the Block Error Rate (BLER)
- ρ is the RLC Transport Blocks with CRC Error
- ε_{RLC} is the RLC Transport Blocks

BLER is measured separately on the UL and DL direction.

- **Downlink Signal-to-Interference (DL SIR).** DL SIR indicates DL quality. It is the ratio between during a call; the SIR varies between the implemented lower and upper boundaries. Lower SIR denotes favourable RF conditions.

$$SIR_{DL} = \frac{RSCP}{ISCP} \times SF \tag{2-20}$$

where:

- $RSCP$ is the received Signal Code Power, unbiased measurement of the received power on one code;
- $ISCP$ is the interference Signal Code Power, the interference on the received signal;
- *SF* is the spreading factor.

- **Uplink Signal-to-Interference Ratio (UL SIR).** UL SIR can be used to estimate the RF conditions of the UL channel in the same way as DL SIR. As with the UL BLER, extracting this information requires access to UTRAN vendor-specific tools for logging and for parsing logs. UL SIR is the ratio between the measured received signal code power (RSCP) of a single UE’s UL DPCCH’s signal and the interference signal code power (ISCP) multiplied with the spreading factor of the DPCCH, which has a constant value of 256. ISCP is measured based on sophisticated proprietary algorithms implemented in Node B software.

$$\sigma = \frac{\varepsilon}{\psi} \times 256 \tag{2-21}$$

where:

- σ is the Signal to Interference Ratio (SIR);
- ε is the received signal code power of a single UE’s;
- ψ is the Interference Signal Code Power (ISCP).

- **Transmit Power Control (TPC).** During a call, the TPC should present a random arrangement of “0” and “1” based on the power control algorithm. In either direction, a higher occurrence of “1” (power up) indicates degradation of the radio link. Power control tries to compensate by requesting additional transmit power. For the UL TPC, the sequence of bits has specific meaning in out-of-sync conditions.

- **UE Transmit Power**. Can also indicate UL RF conditions, if power control works perfectly. An increase of User Equipment transmits power correlates with high Up Link BLER or high UL interference. It is easier to examine UE transmit power than the UL quality metrics BLER or SIR because the UE transmit power information can be extracted directly from UE log and does not require additional post-processing or trace synchronisation.

- **Mean throughput.** Specifies the average rate at which data are expected to be transferred across the GPRS network for the PDP context. In addition to various exact values, a "best-effort" mean throughput class may be negotiated. This means that throughput shall be made available to the UE on a per need and availability basis.

$$\bar{\chi}(i) = \frac{s}{S} \sum_{j=1}^{\frac{S}{s}} \frac{\sum_{i=1}^{\Delta(j)} \chi(i,j)}{\Delta(j)} \qquad (2\text{-}22)$$

where:

- $\bar{\chi}$ is Mean Throughput of connection *i*;
- *S* is the measurement period;
- *s* is the ample period;
- Δ is the active Connections j is the number of connections active during sampling interval j;
- $\chi(i,j)$ is the throughput i,j is the throughput of connection i measured during sample interval j.

HSDPA bit rates depend upon the protocol stack layer from where they are measured. They also depend upon the allocated modulation scheme, number of HS-PDSCH channelisation codes and coding rate. The throughput capability of HSDPA is often quoted using the figures present in Table A.2. These figures can be derived using the equation below:

$$\chi = \frac{R_C}{SF} \times B_{ps} \times C_R \times N_C \qquad (2\text{-}23)$$

where:

- χ is the throughput
- R_c is the chip rate 3.84Mcps;
- SF is the spreading Factor fixed at 16;
- B_{ps} i is the bits per symbol (QPSK maps 2, 16QAM maps 4 and 64QAM maps 6 bits onto each symbol);
- C_R is the coding rate references the throughput result to the top of the Physical layer;
- N_C is the number of codes - 5, 10 or 15.

In the scope of this thesis, the UE category chosen for HSDPA is category 14. Because this UE category 14 can receive data during every TTI, then it must be able to support 6 parallel HARQ processes.

2.7 State of the art

A brief overview of the state of the art is presented in this section, in order to show what has been done in this field up to now, thus, emphasising the importance of this work.

Several works present statistical studies of QoS and QoE for data services in UMTS and HSPA networks. Khirman and Henriksen [KhHe02] showed that to successfully resolve the Internet infrastructure's most pressing problems, the Internet industry needs to improve the service quality. So, they concluded of their analysis based on objective metrics for web user satisfaction that there is a non-linear relationship between QoS and QoE. Based on this analysis, they conclude that effective network bandwidth plays a crucial role in end-user satisfaction and that there is no gain in web browsing satisfaction for connection speeds above 200kBps, although the studied only reflect the data service HTTP. They also find that network latency plays a less significant role on the level of user satisfaction, especially in the range of 50-500 ms where its influence is negligible.

In [VaSo05], K. Valkealahti and D. Soldani want to evaluate the sensitivity of QoS to a number of parameters that determine the resource allocation and release in packet scheduler. The QoS was found highly sensitive to the inactivity timer, the minimum allowed bit rate, and the maximum allowed bit rate. The most significant result that they observed was the increase in the spectral efficiency that was obtained with the priority-based differentiation of the explored parameters. The satisfaction ratio of the lowest performing service increased from 79% with the best undifferentiated configuration to 97% with the best differentiated settings. In terms of spectral efficiency, the gain was 50%. The parameter analysis procedure yielded optimised parameters that showed more stable operation with improved throughputs in high-loaded cells and with smaller cell-to-cell variation. However, generalising the results to other cases, or real networks, is not straightforward because the results were likely highly specific to the satisfaction criteria, the traffic models, and the traffic mix.

Other studied performed by Hyun-Jong Kim *et al.* [HDJK08] propose the approach method for the objective QoE measurement through the QoS parameters. By using the QoS information measured at a network-level, they explain the proposed QoS-QoE correlation model for the objective QoE, and describe the example of the service QoE evaluation. However, no reference to the network environment is made.

[ZiWa04] studies QoS attributes for PS services, in order to meet end users expectations, and to overcome the limitation of system resources and the air interface capacity it is necessary to properly define QoS. They concluded there is a necessity to implement and control the quality of air interface traffic, as well as traffic between UTRAN and CN elements, using the advanced routing protocols and QoS management mechanisms.

[WeKO10] develops a systematic approach for evaluating QoS in cellular networks based on multiple criteria. Data collected from drive testing was used as the means to evaluate different drive routes of varying quality. They concluded it provides a technology agnostic evaluation methodology that can be applied to multiple cellular technologies. This provides researchers with a common tool for evaluating and comparing QoS in a wide variety of communication environments. In addition, by identifying the regions of coverage that provide lower quality indexes, network providers can take corrective actions to improve their capabilities. This studies use drive tests and this methodology can be easily modified to work with different networks.

Chapter 3

Models

In this chapter, the models and algorithms in which the software simulation tool is based are presented. The simulator is formed by different blocks, inputs and outputs, which are described. The chapter ends with the assessment of the simulator.

3.1 Models

In the past, the concern of the operators was the average throughput of the users. With the development of the wireless systems, the concerned changed to the QoS and QoE. The traffic classes for this thesis was chosen to be streaming and interactive with the web browsing, file download and video streaming data services, hence, providing a good sample. The parameters performed to evaluate the QoS are the initial delay, time elapsing since the application is asked to be performed, the average throughput for web browsing and file download, and the average delay for video streaming. A real case approach considers a static user in Portugal, who could be in three different scenarios, urban, sub-urban and rural, and indoor or outdoor. This Reference User (RU) is performing DL data services with an equipment category 14, using HSDPA codes. The servers can be in Lisbon, London and Houston, hence, the time delay may vary.

The environment has impact on the coherence time, which is the time that the channel can be considered as invariant, in log normal fading, and in building penetration if the user is inside.

The instantaneous bit rate will change at each coherence time in accordance with the environment. One has assumed that the instantaneous bit rate changes when:

$$\frac{t_{[\mathrm{ms}]}}{T_c[\mathrm{ms}]} = integer \tag{3-1}$$

where:

- t is time,
- T_C is the coherence time.

This variation is related to the variation of SNR and the receiver modulation, Figure D.7. The variation of the SNR will change according to the environment where the reference user is, and if it is indoor one has to consider an extra attenuation *(L_{extra}).* In this model the body loss is not taken to account.

The throughput for a SISO configuration with 64QAM is given in Annex B, for which one has made an approximation:

$$R_b[\mathrm{Mbps}] = \begin{cases} 0, & -11 \leq \rho_{N[\mathrm{dB}]} \\ -2\times10^{-6}\times\rho_N{}^5 - 2\times10^{-5}\times\rho_N{}^4 + 0.0006\times\rho_N{}^3 \\ \quad +0.0267\times\rho_N{}^2 + 0.6031\times\rho_N + 4.229, & -11 \leq \rho_{N[\mathrm{dB}]} < 20 \\ 22.2, & \rho_{N[\mathrm{dB}]} \geq 20 \end{cases} \tag{3-2}$$

This instantaneous bit rate is 80% of the one given by (3-2), due to protocols headers. The instantaneous bit rate is distributed for all codes. In this model, the user's distance to the BTS is not taken into account, but SNR is statistically change which introduces differences in the user’s instantaneous bit rate.

In the flowcharts Figure 3.1 one has used colours to identify the input, output, and intermediate parameters:

Figure 3.1. - Algorithms legend.

In this model, a mixed system with HSDPA and R99 channels one considers an exclusive channel HSDPA and a shared channel R99/HSDPA. The HSDPA codes has SF=16, and R99 codes has SF=128. Codes for signalling and control were taken into account.

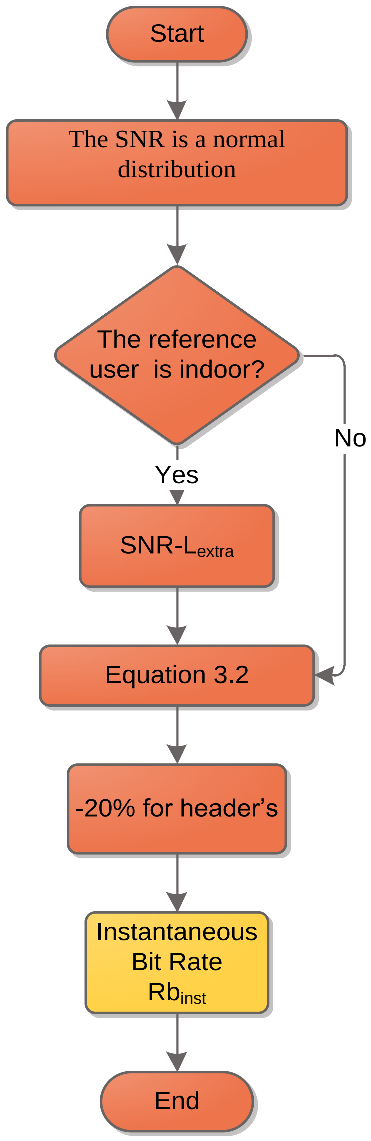


Figure 3.2. - Instantaneous bit rate algorithm.

For the network load, one has considered a number of other users, uniformly distributed, which have been distributed according to a service penetration. It was considered that users of voice, audio streaming and web browsing are distributed primarily by R99 codes, and then to HSDPA ones. Video streaming users (reference user and other users) only use HSDPA codes.

At each time frame, the available codes are distributed by the different users of the services. Primarily, R99 codes are assigned to voice users, and then to audio streaming, file download and web browsing. If the R99 codes are not sufficient for these services, are allocated HSDPA codes, with a different spreading factor. For that, the SF was divided by eight, to go from SF128 to SF16.

At the end of this distribution, HSDPA codes are available or not for the service user reference that change each time period (2 ms). The algorithm for the HSDPA available codes are:

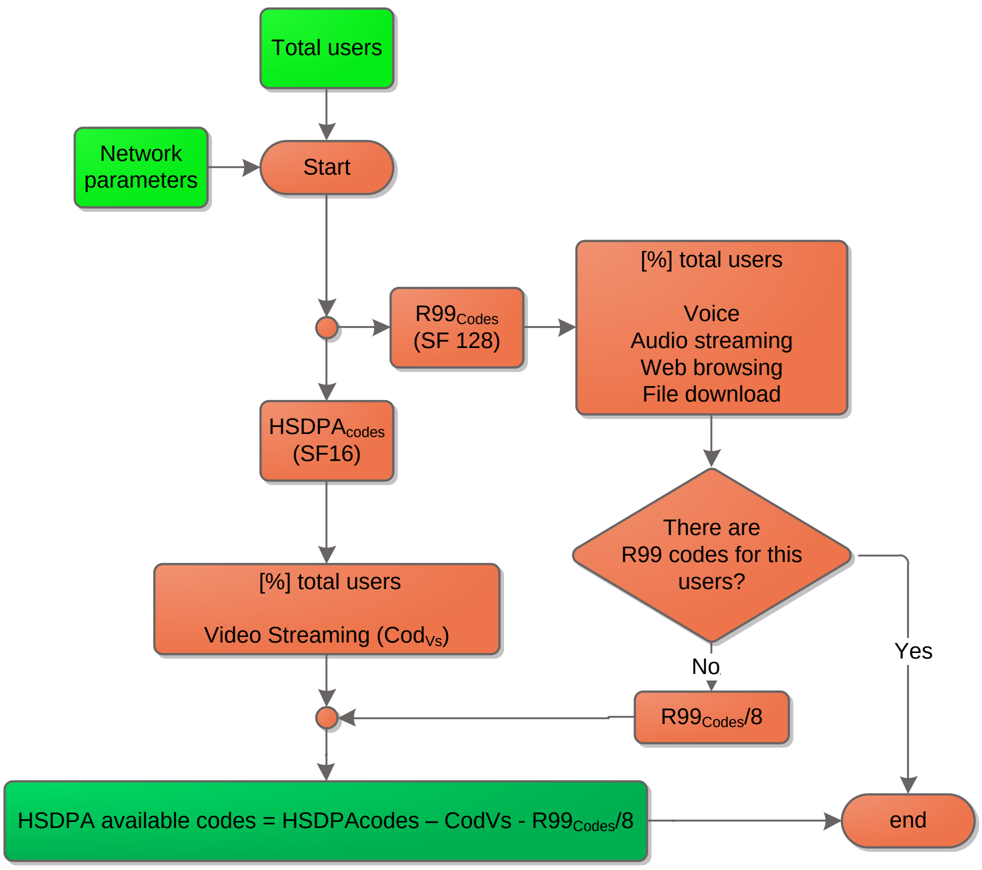


Figure 3.3. - HSDPA available codes algorithm.

The reference user is the last one to be served, and all HSDPA available codes are assigned to the reference user (except for the video streaming service), but in fact this never happens, because this is not the way operators work.

If when there are no codes HSDPA available in the time frame at which the reference user starts the service, there is an initial time delay *(t_D)*. Only when there are codes available the reference user service starts.

For the video streaming service, one considers a buffer with a specific size. After filling in the buffer, the video begins. When the video buffer is empty, the video stops until the buffer refills.

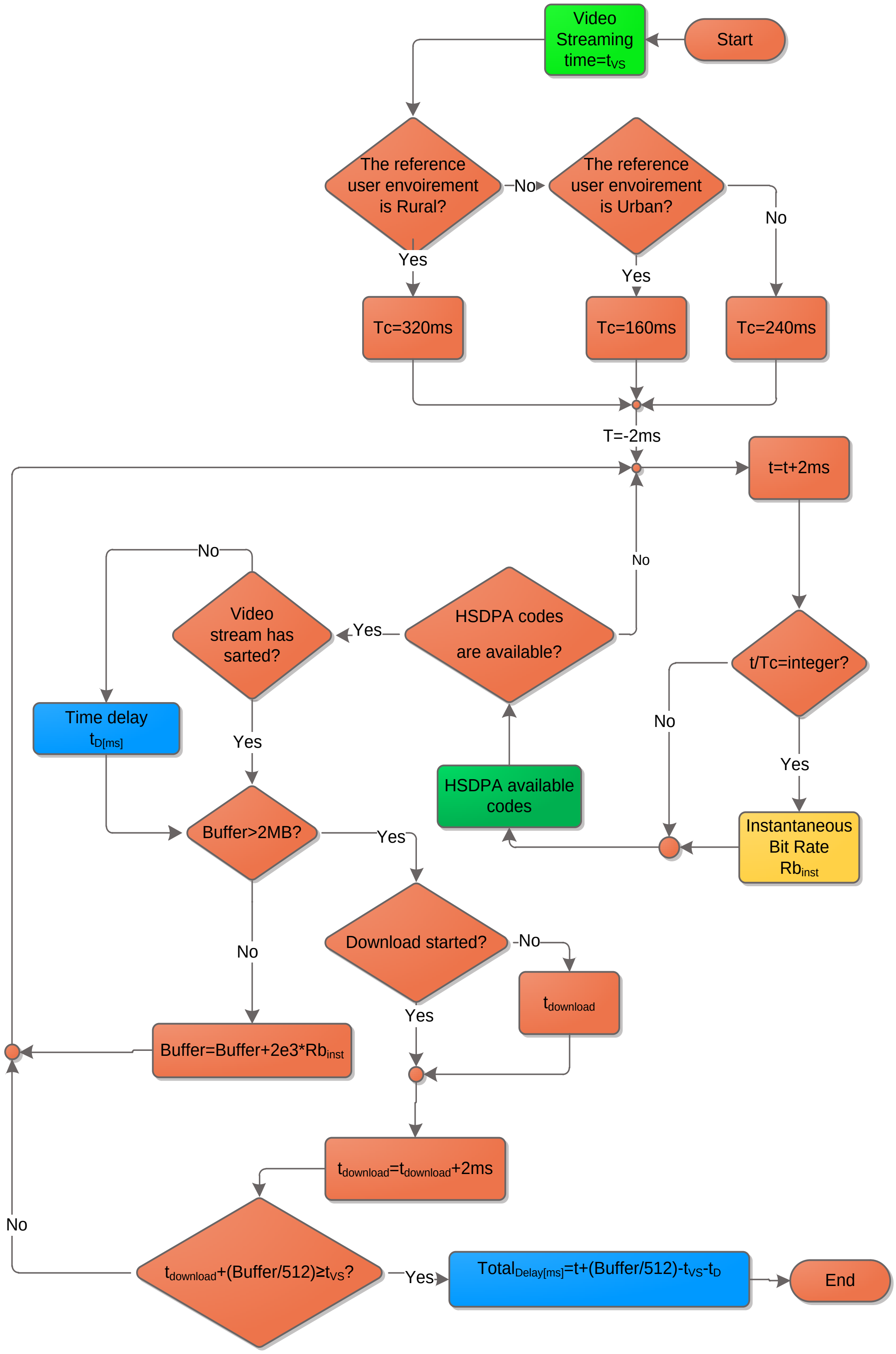


Figure 3.4 - Video Streaming algorithm.

At each frame time, the buffer has an increment of:

$$B_{(t+2ms)[\text{bytes}]} = B_{t[\text{bytes}]} + t_{frame[\text{s}]} \times R_{b_{inst[bps]}}/8 \tag{3-3}$$

where:

- B_t is the Buffer size in the previous time frame,
- $\text{B}_{(\text{t+2ms})}$ is the Buffer size in the actual time frame,
- t_{frame} is the time frame,
- $R_{b_{inst}}$ is Instantaneous Bit Rate at every coherence time.

When:

$$t_{download\ [\text{s}]} + 8 \times \frac{B_{[bytes]} \times 10^3}{V_{BR\ [\text{bps}]}} \geq t_{Vs\ [\text{s}]} \tag{3-4}$$

where:

- $t_{download}$ is the download time,
- B is the Buffer size,
- V_{BR} is the video bit rate (512 000),
- t_{Vs} is the video duration.

The final delay T of the video streaming service can be calculated by:

$$T_{Delay\ [\text{ms}]} = t_{[\text{s}]} + 8 \times \frac{B_{\ [bytes]} \times 10^3}{V_{BR\ [\text{bps}]}} - t_{Vs\ [\text{s}]} - t_{D\ [\text{s}]} \tag{3-5}$$

where:

- t_D is initial time delay.

The file download and web browsing services are presented in the same flowchart. The services are different, especially in the size of the file. In the web browsing data service, the reading time of web pages and the number of accesses made by the reference user were not considered, because it is a statistical model. As input parameter, one has considered a specific file download volume V_{IN}, which is equal to V_{Wb}, for the web browsing service and to V_{DL}, for the file download service.

In these two services, there is an initial time delay *(t_D),* like in video streaming, when there are no HSDPA codes available at the beginning of the service.

At each time frame the V_{IN} will decrease with the form:

$$V_{IN\ [\text{bytes}]} = V_{IN\ [\text{Bytes}]} - \frac{Rb_{inst\ [\text{bps}]}}{8} \times t_{frame[\text{s}]} \times Cod \tag{3-6}$$

where,

- V_{IN} is the file volume,
- Cod is the number of HSDPA codes available.

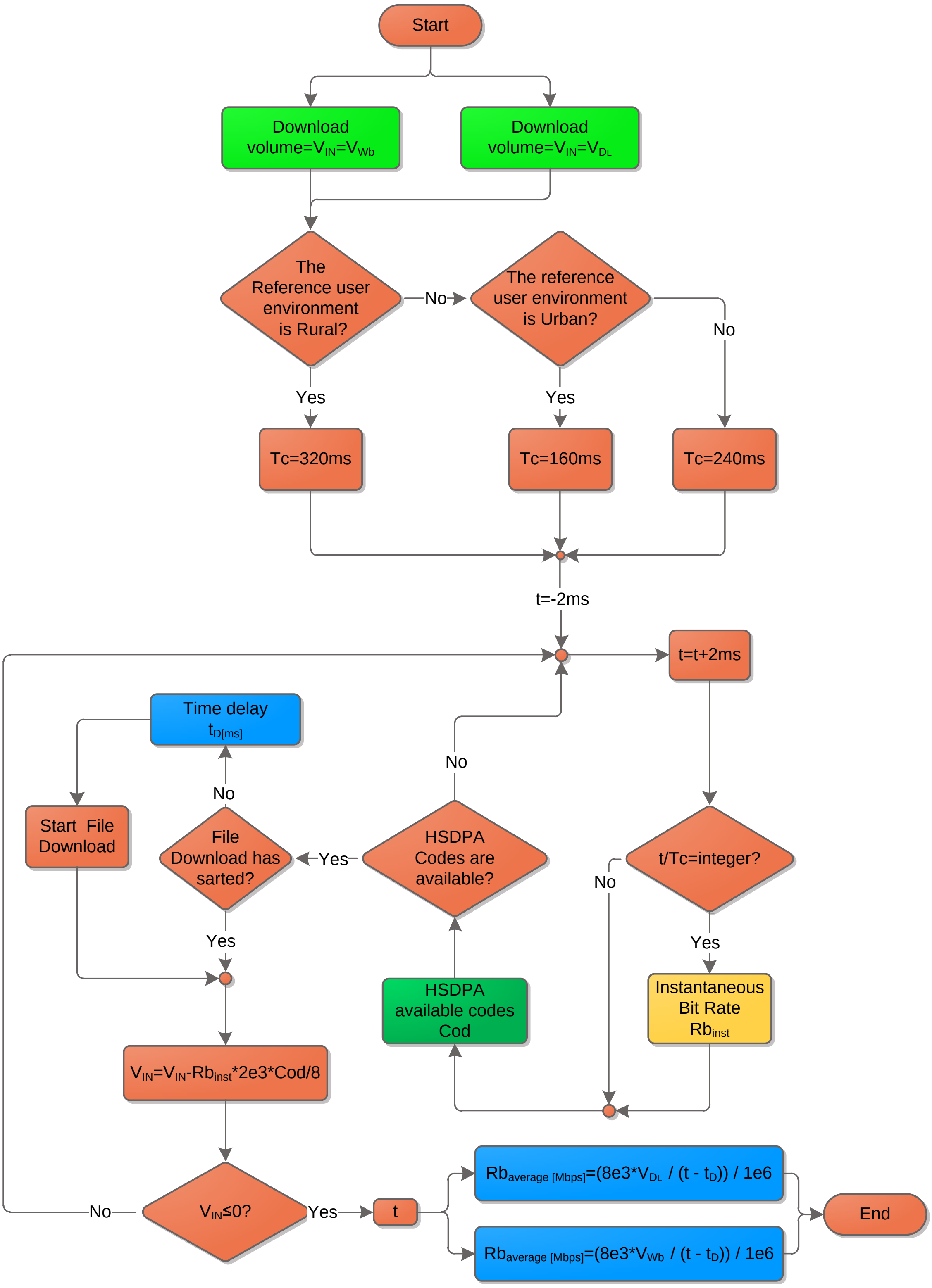


Figure 3.5 - File Download and web browsing algorithm.

For the file download and when $V_{IN} \leq 0$, the output average bit rate can be calculated as:

$$R_{b_{average\,[\mathrm{Mbps}]}} = \left(\frac{8 \times V_{DL\,[\mathrm{Bytes}]}}{(t - t_D)_{[\mathrm{s}]}}\right) / 10^6 \qquad (3\text{-}7)$$

where:

- V_{DL} is the file volume for download,
- t is the time download,

For the web browsing service and when $V_{IN} \leq 0$, the output average bit rate is:

$$R_{b_{average}}\,[\text{Mbps}] = \left(\frac{8 \times V_{Wb\,[\text{Bytes}]}}{(t - t_D)_{[\text{ms}]}}\right)/10^6 \tag{3-8}$$

where:

- V_{Wb} is web volume for download,

To assess the influence of the distance of the server in the delay it was need to calculate the latency induced by the distance:

$$L_{\,[\text{ms}]} = \frac{1000 \times d_{\,[\text{km}]}}{c_{[\text{km/s}]}} \tag{3-9}$$

where:

- L is the latency,
- d is the distance between the server and the RU,
- c is the light velocity (299 792.458 km/s).

This equates to a latency of 3.33 μs/km of path length. It was considered that this length was performed in fiber optic and by satellite. Light travels slower in fiber due to the fiber refractive index and this increases the latency to approximately 5 μs/km.

3.2 Implementation in Matlab

In this section, one presents the main structure of the developed simulator. To implement the simulator in Matlab (Numerical computing software), main scripts and auxiliary scripts were taken. The main scripts are one for each data service, and the auxiliary ones are the script for instantaneous bit rate, the script to evaluate the available codes, and the script for the network load.

In Figure 3.6, one presents the overview for the script of web browsing and file downloads data services. The input parameters, identified with the green colour, were used as auxiliary scripts. The input parameters for the auxiliary script instantaneous bit rate, used in the two data service scripts, are the environment, the SNR variation, and the reference user localisation indoor or outdoor. The input parameters used in the auxiliary script, available codes, also used in three data service scripts, are the R99 and HSDPA channels. For auxiliary script network load, the input parameters are the number of other users and service penetration. For the web browsing data service, it was necessary to define the web size used in each simulation. For the file download data service it was necessary to define the file size used in each simulation. The output parameters for these data services were initial delay and average bit rate.

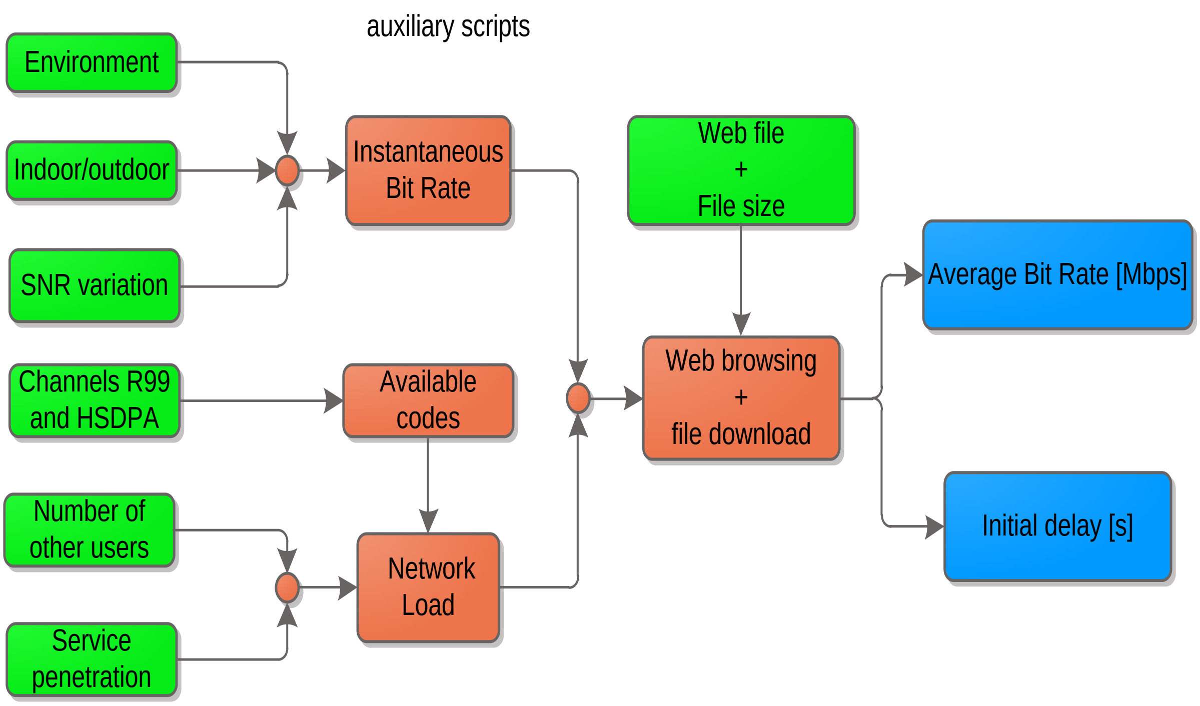


Figure 3.6 - Simulator overview for web browsing and file download data service.

For the video streaming data service, Figure 3.7, the auxiliary scripts are equal to web browsing and file download data services, with the same input parameters. However, it is necessary, for this case, to define the video duration in each simulation. The output parameters for these data services were initial delay and average video delay.

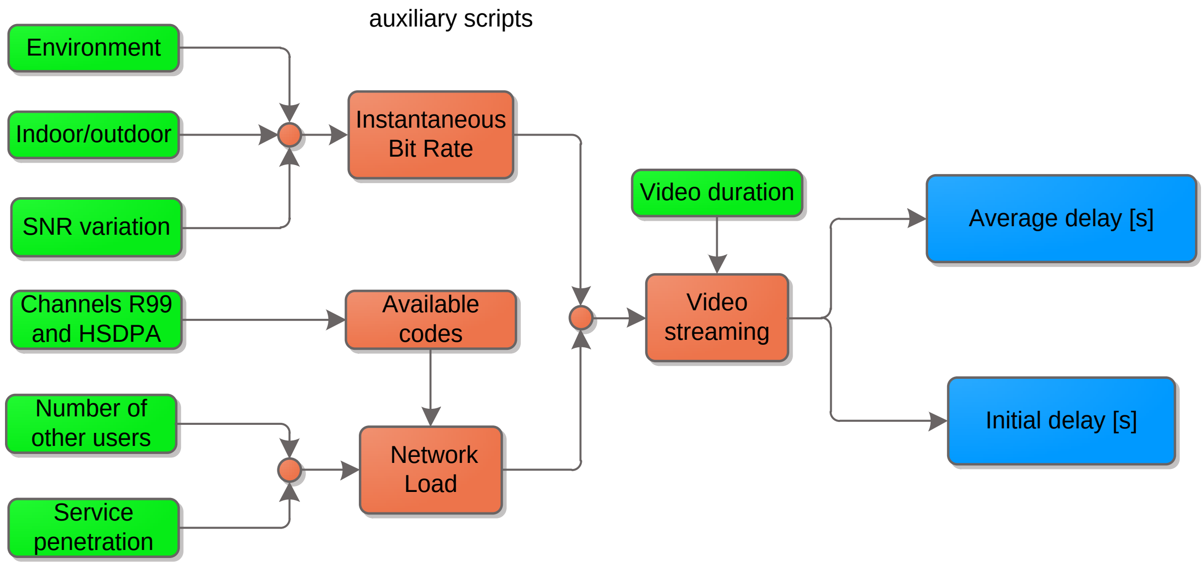


Figure 3.7 - Simulator overview for video streaming data service.

The objective of this simulation is to assess the initial delay, the average bit rate and the average video delay of a reference user, which could be in three different environments, indoor or outdoor performing three different services. The simulator has three scripts for the three different services of the reference user. Those scripts are linked to the auxiliary scripts. The auxiliary script “Instantaneous bit rate” gives, according to the environment, at coherence time, the instantaneous bit rate, taking 20% for headers into account. For this script, the input parameters are the SNR according to the environment and the location, indoor or outdoor of the reference user.

Depending on the environment of the reference user, the coherence time will change accordingly:

Table 3.1. - Coherence Time in the urban and rural environment (adapted from [CaKn10]).

Environment	***Urban***	***Suburban***	***Rural***
Coherence time $T_{C[ms]}$	160	240	320

If the reference user is in indoors, the SNR will have an extra attenuation of L_{extra}=10 dB.

It was defined that the SNR has a normal distribution with different values according to the environment. So, one has considered for outdoor μ=18 dB and σ=6 dB for urban environment, [Carr11], for sub urban environment μ=16 dB and σ=6 dB and for rural environment μ=13 dB and σ=6 dB (Figure 3.8).

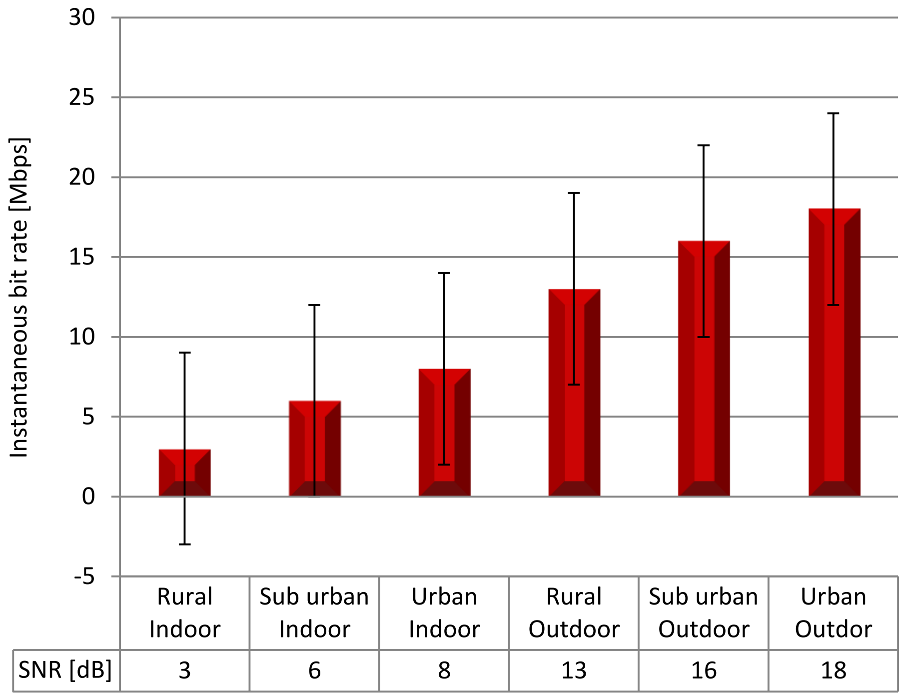


Figure 3.8 – SNR versus environment.

In each coherence time loop, the instantaneous bit rate was the same and only changed in the next *(t+Tc)*.

The auxiliary script “Available codes” gives at each time frame the HSDPA codes available for the mixed system with HSDPA and R99 channels; for this script the input parameters are the number of channels R99 and HSDPA.

In the auxiliary script “Network load”, the codes obtained in the script “Available codes” are distributed by the other users and by the reference user. The input parameters for this script are the number of other users and the service penetration. The number of users is changed uniformly between 0 and 40; however, along the day, and in working days and weekends, the distribution is another with implications in the results.

This distribution of service penetration is typical for a given environment. The distribution of service penetration is different for the same place, in working days or weekends. The data services are different along the day and have a peak at night in the happy hour.

The loop time in the “Network load” script is one time frame of 2 ms, and the HSDPA codes available change at each time frame. If there are no HSDPA codes available when the reference user starts the service, the elapsed time until the codes are available is considered as initial delay. If the same occurs during the video streaming and the buffer is empty, the video stream stops.

3.3 Assessment

Before performing simulations and the results analysis, the simulator was assessed, namely the validity of the output and the necessary number of simulations that ensure statistical relevance.

To validate the variation of the coherence time, one has tested that the bit rate changes at every coherence time according to the environment. It was necessary to check that the instantaneous bit rate keeps constant during the coherence time. It can happen, two or more coherences time with the same instantaneous bit rate, due to statistical reasons. In Figure 3.9, one presents the Matlab output, where it is possible confirm the changing of the instantaneous bit rate at each coherence time for the urban environment. The simulation results for the others environments are in Annex D.

One has evaluated the mean variation of HSDPA codes available with the network load, to know how many other users the network would support. For that, the number of other users was fixed with intervals of 5 users, between 0 and 45. In Figure 3.10 one presents an example of the Matlab output, with the HSDPA codes available for a 10 MB File

It is visible, for the same data service with the same file size, that when are no users, only 900 frames are necessary whilst for a network with 40 users it is necessary almost 4000 frames. This is expected, because with more network load there are no HSDPA codes available and the reference user needs to wait. From this output, one has removed the HSDPA codes available for the ten examples of network load, the Figure 3.11 giving the mean variation of HSDPA codes available with the network load.

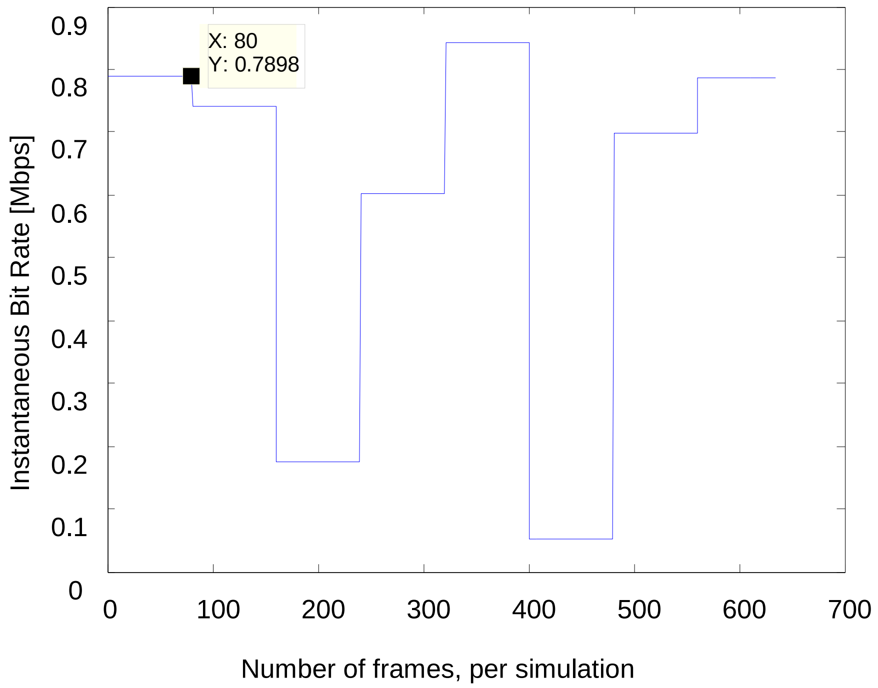
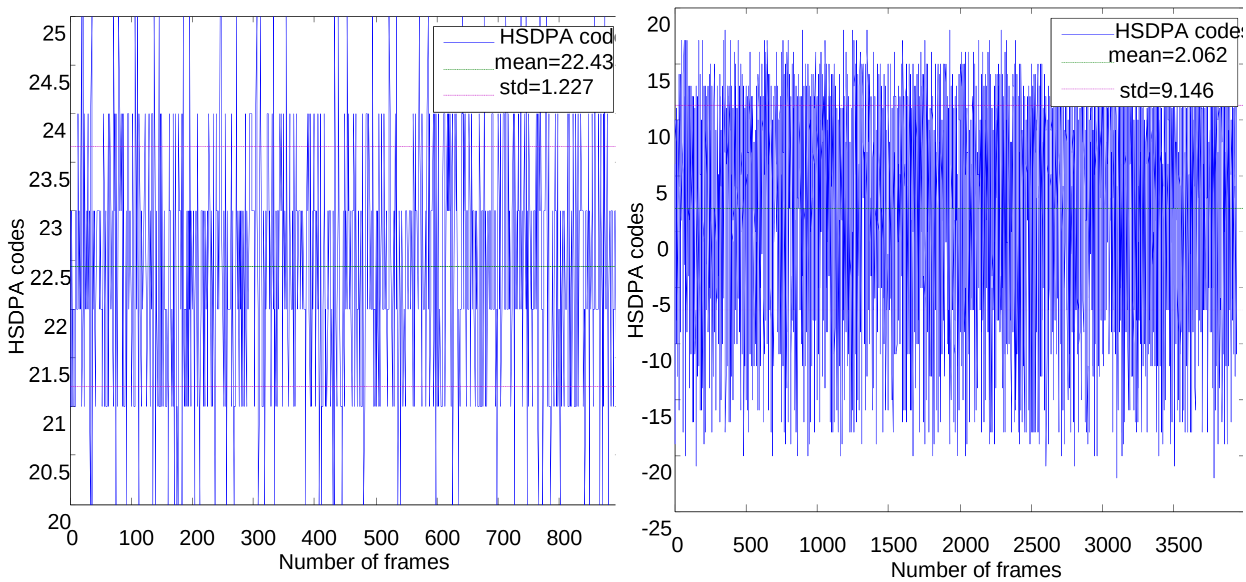


Figure 3.9 - Bit Rate at each frame of 2ms, *Tc*=160 [ms] – Urban environment.



a) 0 users

b) 40 users

Figure 3.10 - HSDPA available codes for 10 MB File download.

It was concluded that with 45 other users, the mean HSDPA available codes are negative, so, a variation between 0 and 40 other users was defined.

The mean HSDPA available codes are negative means that there are no available codes to the RU.

The assessment of the math code was made using a calculator and fixing some variables separately or simultaneously, like the number of other users, HSDPA available codes, instantaneous bit rate, etc.

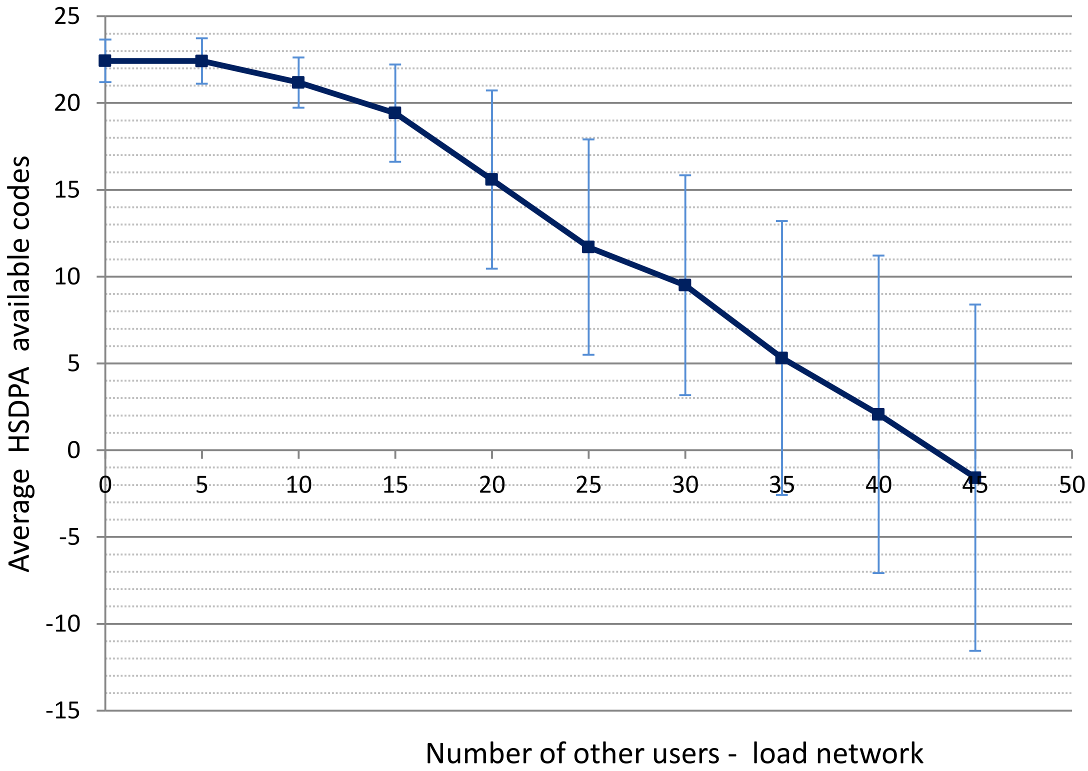


Figure 3.11 - Mean variation of HSDPA available codes with the network load.

For instance, the instantaneous bit rate was forced to be 2 Mbps and to one HSDPA available code. It was observed that there was no initial delay, which was expected, because there is always one HSDPA available code and the average bit rate was 2 Mbps.

At this point, it is important to understand the convergence potential of the mean bit rate and mean delay functions. This was done by running 200 simulations for the 3 services and defining a measure of convergence, Δ:

$$\Delta[\%] = \frac{|\Phi^{cum\_s} - \Phi^{cum\_S}|}{\Phi^{cum\_S}} \times 100 \qquad (3\text{-}10)$$

where:

- s is the simulation run index;
- S is the total number of simulation runs conducted during the convergence study (200).
- Φ^{cum_s} is the partial delay cumulative mean at simulation s (for video streaming service) and partial bit rate cumulative mean at simulation s for other services;
- Φ^{cum_S} is the total delay cumulative mean or bit rate cumulative mean (from all simulations).

In Figure 3.12, the convergence of the mean bit rate and the mean delay functions are presented. Because web browsing service presents a bigger fluctuation, it was performed three times. Thirty simulations have a convergence potential better than 4%, corresponding to thirty minutes per simulation, maximum.

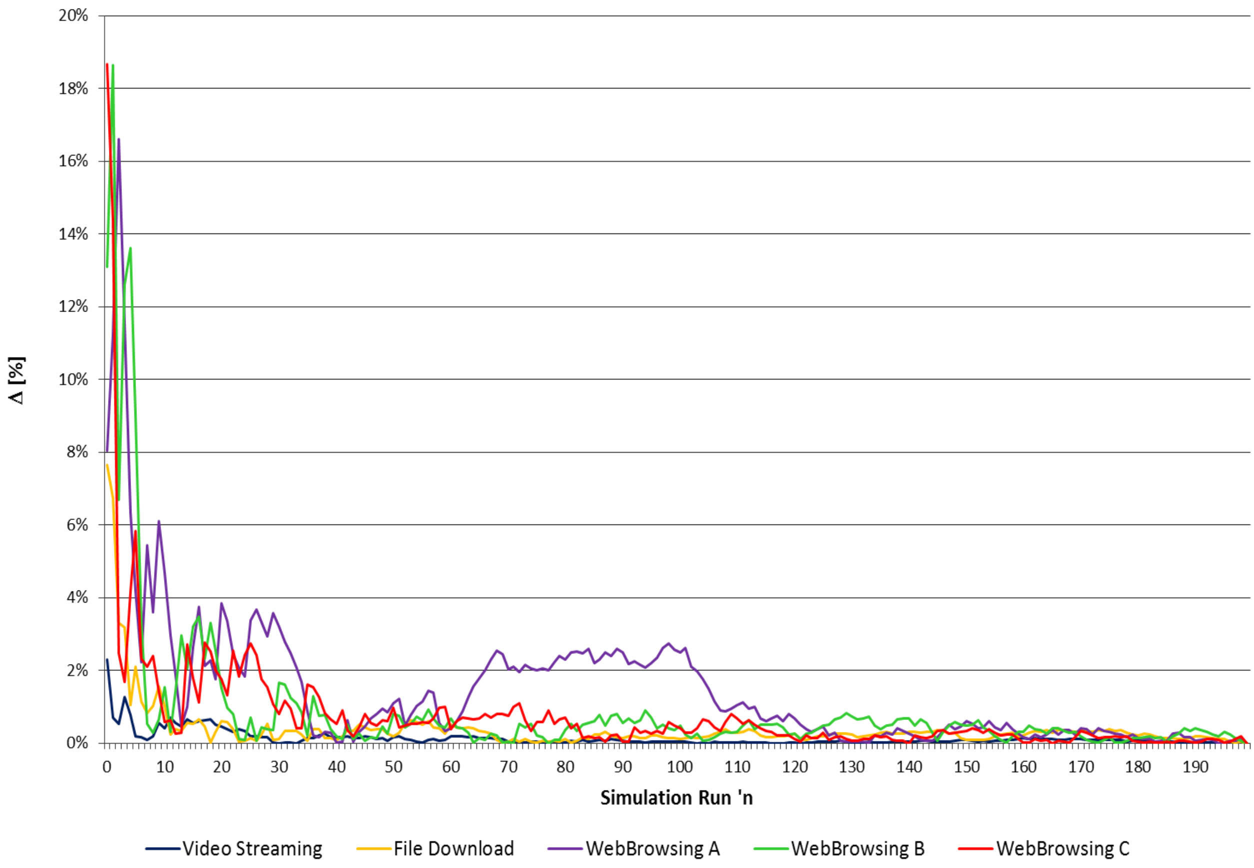


Figure 3.12 - Convergence of mean bit rate and mean delay functions.

The assessment of instantaneous bit rate for each environment indoor and outdoor was achieved by simulating the instantaneous bit rate algorithm. The output for urban environment is in Figure 3.13. The outputs to the other environments are present in Annex D.

The shapes of the instantaneous bit rate histograms (except rural indoor) have a peak at 1.26 Mbps, because for values of SNR above 20 dB the instantaneous bit rate is constant and equal to 22.2 Mbps, Figure D.4, Figure D.5 and Figure D.6. For rural indoor, the SNR is very low so the value 1.26 Mbps is not reached often.

a) SNR

b) Indoor Bit Rate

c) Outdoor Bit Rate

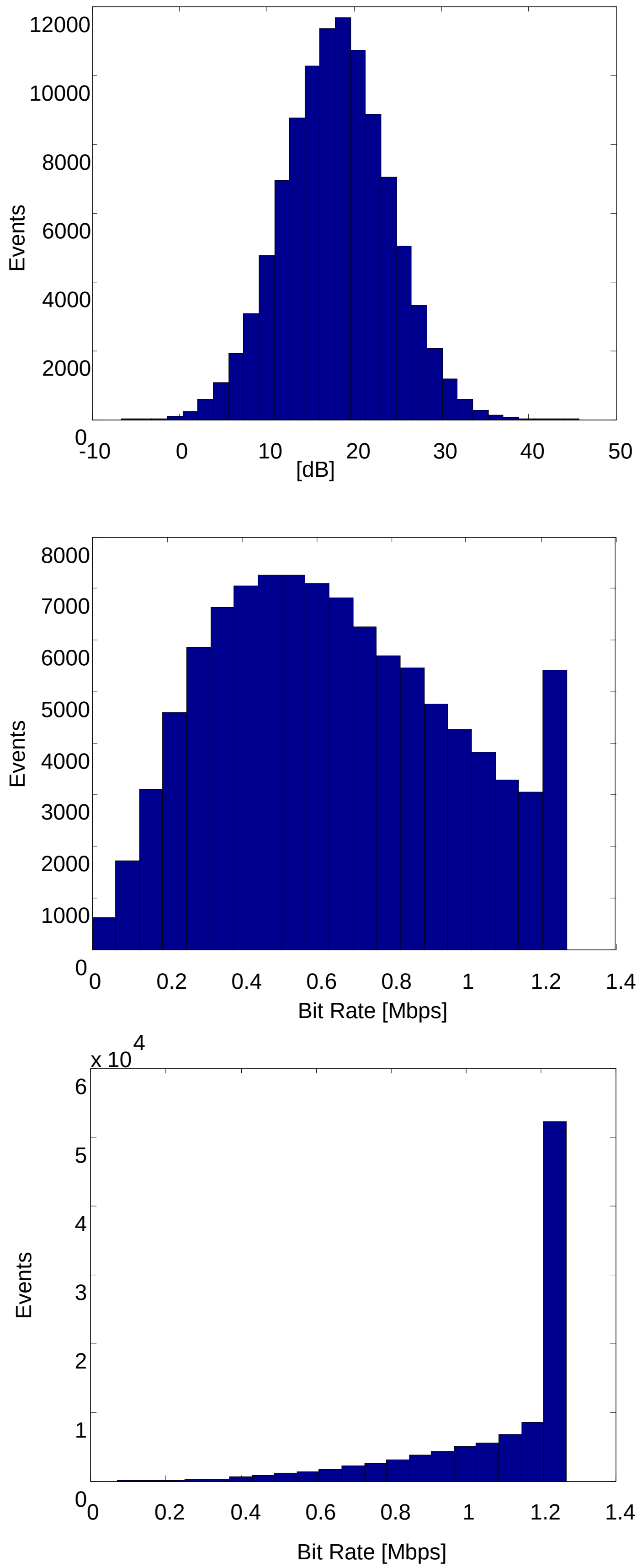


Figure 3.13 - SNR, indoor and outdoor Bit Rate variation in urban environment.

One presents in Figure 3.14, Figure 3.15 and Figure 3.16 the Matlab output, for the 30 simulations, of the average bit rate for the outdoor situation in the three environments.

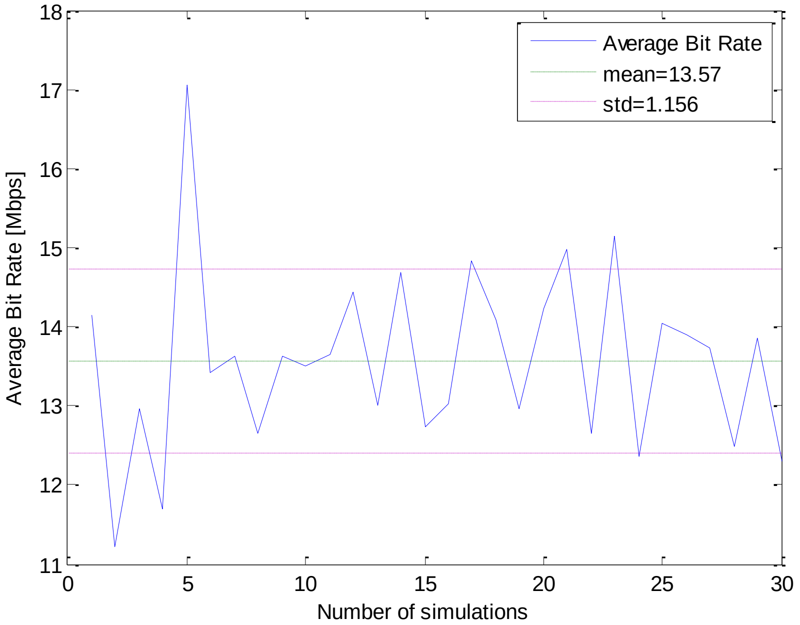


Figure 3.14 - Average bit rate for 10MB File Download, Rural, Outdoor.

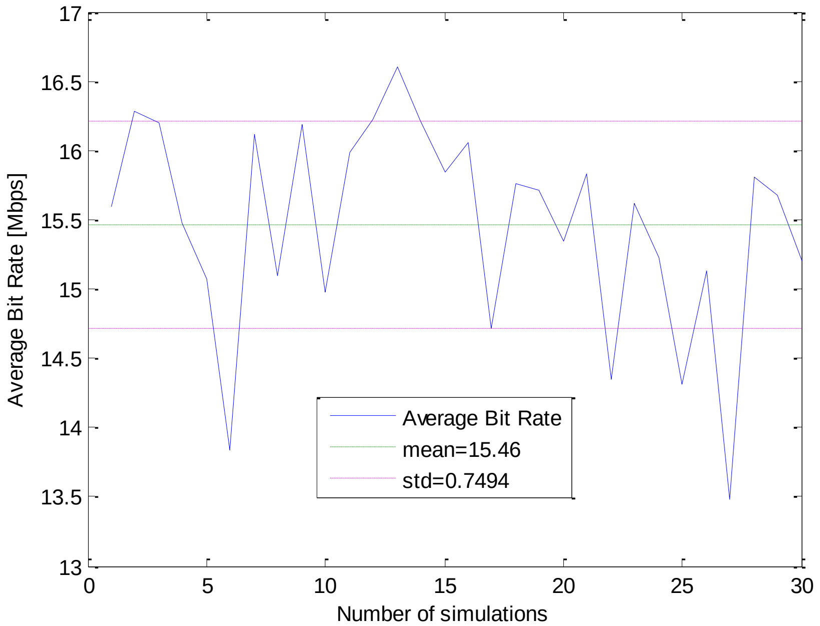


Figure 3.15 - Average bit rate for 10MB File Download, Suburban, Outdoor.

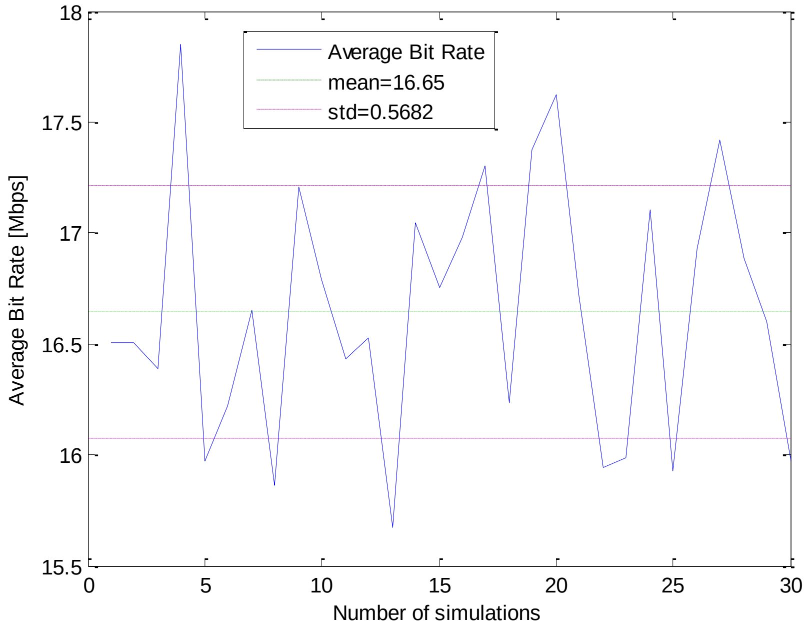


Figure 3.16 - Average bit rate for 10MB File Download, Urban, Outdoor.

It is possible to note that both the average bit rate and the SNR grow with the change of environment from rural to urban.

Chapter 4

Results

In this chapter, the results of the simulations for different environments are presented, analysed and discussed in detail. The scenarios development is characterised, in the first section.

4.1 Scenarios development

The scenarios under study in this thesis are for a reference user, indoor or outdoor of a rural, suburban and urban environment. As an example a city like Lisbon can be used, where in a few kilometres radius these three different environments can be found. The choice of these environments is to understand, for the reference user point of view, the differences between performing the same data services in the three different environments.

The RU will perform three data services: web browsing, file download and video streaming. The selection of these data services is to have a sampling of the principal data service and compare the differences between data services.

For web browsing one considered the minimum, average and maximum typical web size pages [Rama10]. For FTP, three sizes of files were analysed. In the video streaming with a bit rate of 512 kbps, streams with three different times were analysed. The summary of this is presented in Table 4.1. The choice of these parameters intended to illustrate the reality of users in our days.

Table 4.1. – Scenarios on study.

Environment		***Data services***								
Indoor	***Outdoor***	***Web browsing [kB]***			***File download [MB]***			***Video streaming [s]***		
Rural		20	320	800	1	10	100	120	300	600
Suburban										
Urban										

The services for network load are voice, video and audio streaming, file download and web browsing with this service penetration distribution:

Table 4.2. - Service penetration (adapted from [SeCo10]).

	Voice	***Audio Streaming***	***Video Streaming***	***File Download***	***Web Browsing***
Service Penetration [%]	58	10	10	14	8

The selection of the number of other users was based on a study to analyse how the mean HSDPA codes available change with the network load (Annex C). As expected, when the number of other users grows, the number of HSDPA codes available decreases, Figure 3.11, so the maximum number of other users for network load, was 40.

To analyse the influence of network load, in the performance of the initial delay, average delay and average bit rate, it was simulated to the three different data services the network load of 0, 20, 40 and 60 users. So, for rural outdoor environment, the network load was simulated for web browsing of 20 kB, file downloads of 1MB and video streaming of 120 s.

Table 4.3 summarises the cases that were analysed in this thesis.

Table 4.3. - Applications and real cases studied for DL.

Destination	Source		
RU - Lisbon. Portugal	***Web Browsing***	***Streaming***	***FTP***
	Server FCCN Lisbon. PT		
	Server London. UK [~ 1000km]		
	Server Houston. USA [~ 9000km]		

For the simulation plan, one has considered the reference user placed indoor in a rural environment performing the least demanding situation condition for each data service, and this as compared with the other places and environments, for each data service. Table 4.4 presents the simulation plan.

Table 4.4. – Simulation plan.

Data Service \ Environment		Rural	Suburban	Urban	Rural	Suburban	Urban
		Indoor			Outdoor		
Web browsing [KB]	20	**A1**$_{RU}$	A2	A3	A4	A5	A6
	320	B1	B2	B3	B4	B5	B6
	800	C1	C2	C3	C4	C5	C6
File download [MB]	1	**D1**$_{RU}$	D2	D3	D4	D5	D6
	10	E1	E2	E3	E4	E5	E6
	100	F1	F2	F3	F4	F5	F6
Video streaming [s]	120	**G1**$_{RU}$	G2	G3	G4	G5	G6
	300	H1	H2	H3	H4	H5	H6
	600	I1	I2	I3	I4	I5	I6

The simulations were made for each input parameter of the selected data service. There was fix input and variable input parameters. The simulations were made for all variable parameters, data service, environment, indoor/outdoor, 18 simulations per data service, making a total of 54 simulations.

For this evaluation it was taken as example a RU performing a video streaming with 120 s, 300 s and 600 s in the Rural indoor and urban outdoor environments, which are the ones with extreme characteristics because it is where the SNR has the smallest value (rural indoor) and the biggest value (urban outdoor).

For the same environment one considers indoor and outdoor, as well the coherence time and corresponding SNR.

To analyse the influence of network load, one forced the number of other users to be 0, 20, 40 and 60 when the RU is doing browsing with files of 20 kB, downloading a file with 1 MB, and doing streaming of 120 s in outdoor rural environment.

The influence of the server location was simulated for the data service video streaming, 120 s, in the extremes SNR, which corresponds to the rural indoor and urban outdoor environments. For the simulation, it was considered that the server locations were in Lisbon, London and Houston and the means used to data transfer would be submarine communications cable or satellite.

4.2 Reference user service

The analyses of the results are done by varying one input parameter and fixing the others. The analysis was performed in order to know how the average bit rate, average delay and initial delay are affected by the change of reference user data service, by the influence of the environment (SNR and T_C), by the network load, and by the server location.

Considering the RU doing web browsing with 20 kB per page, indoor in a rural environment, one changed the page size from 20 kB, to 320 kB and 800 kB. For each environment and for the three sizes of the web pages, Figure 4.1, one can observe that the average bit rate has a growing envelope with SNR of indoor and outdoor environments. It is possible to conclude that the average bit rate for web browsing data service is proportional to the environment SNR.

It can be seen that the standard deviation values are close to 50 % for indoor environments and close to 30% in outdoor environments. This value tends to be lower when the file size increases. Due to the higher standard deviation, it is not possible to draw conclusions about the average bit rate; the average value of one environment is included in the standard deviation of the next.

The reason for the existence of no regular differences in the environment for the same web size pages is because in each simulation the number of changes in T_C is less than 5 for 800 kB being only one T_C for pages of 20 kB. The conclusion that the average bit rate is not regular, can be obtained by the difference of the maximum and minimum average bit rate, in percentage, for the three sizes of web pages comparing the evolution indoor and outdoor. For indoor, it is 25%, 18% and 2%, and for outdoor it is 7%, 15% and 3%, for rural suburban and urban respectively.

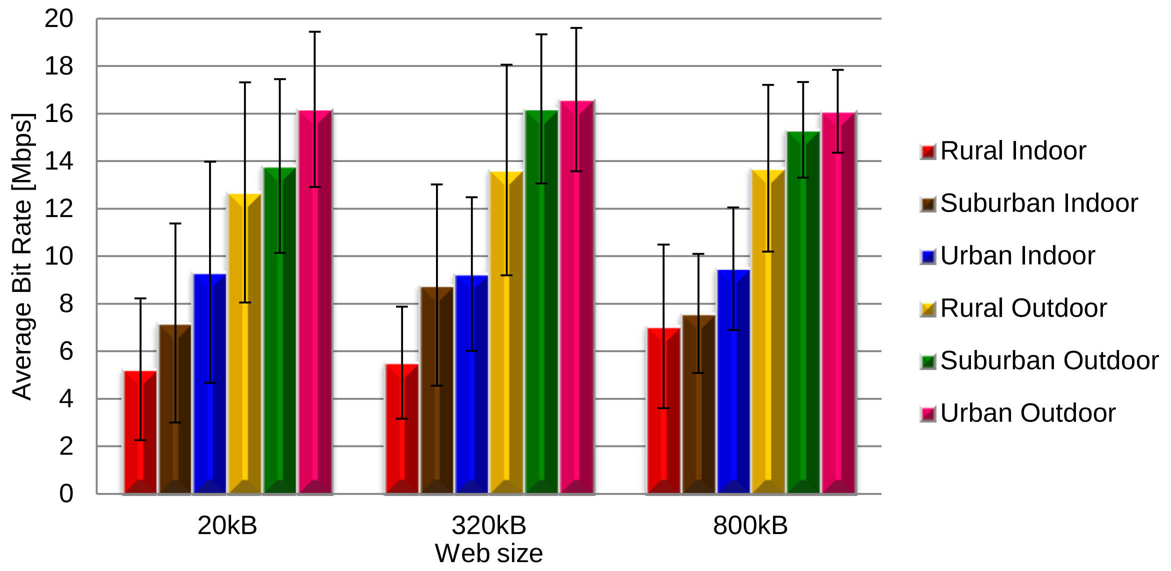


Figure 4.1 – Average bit rate RU – web browsing.

Taking the RU doing file download with a 1 MB file size, indoor in a rural environment the size of the file was changed from 1 MB, to 10 MB and 100 MB, Figure 4.2. One can observe that the standard deviation is decreasing when the file sizes grows, in indoor environments the standard deviation being nearly 30% of the average bit rate, and with files of 100 MB it is around 3%. For outdoor environments the standard deviation is around 20% of the average bit rate, and with files of 100 MB it is next to 2%. That is expected because when the file size grows, it is necessary to have more time for download, and consequently more variations on T_C, namely more changes in the instantaneous bit rate.

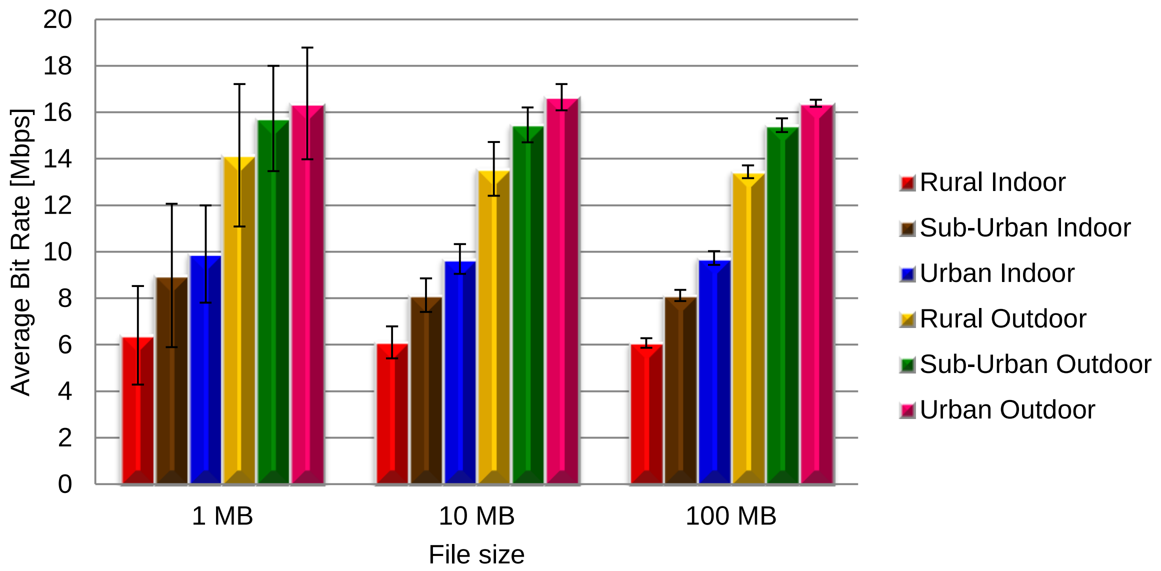


Figure 4.2 – Average bit rate RU – File Download.

For both web browsing and file download services, one can observe a grow evolving for each size of the file. Comparing these values of average bit rate with the service metrics, one can conclude that, for the file download, they are better than the metrics [512, 1536] kbps.

Considering the RU doing 120s of video streaming in a rural indoor environment one changed the video size from 120 s, to 300 s and 600 s. As shown in Figure 4.3, when the video duration increases, the average delay also increases and this happens in every environment, indoor and outdoor. This average delay is expected because with more time for video duration, more variations happen in SNR and consequently, because it is a statistical parameter it can have lower bit rates.

These differences are more pronounced in rural indoor, due to the variation in SNR in this environment as it can be seen in Figure D.4, Figure D.5 and Figure D.6.

It is necessary to understand if this delay is significant compared to the video duration. In Figure 4.4, it is shown that for the same environment, the ratio between the video duration and the average video streaming delay is significantly the same, which means that the average delay is proportional to the video duration.

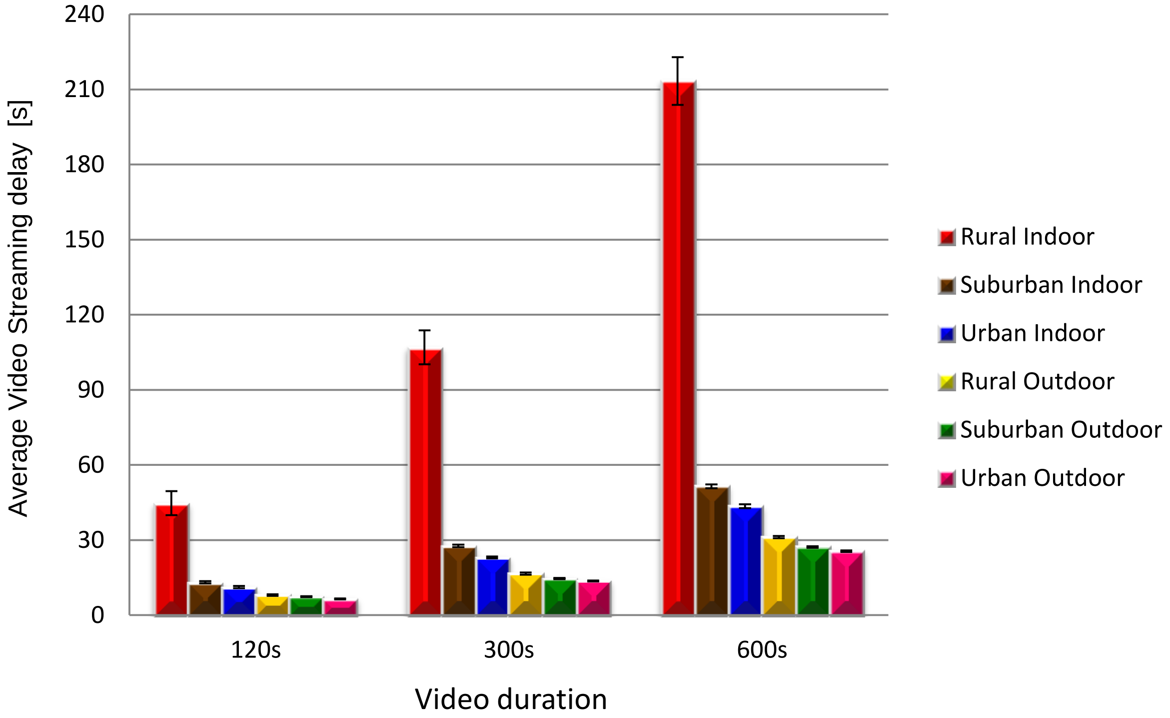


Figure 4.3 - Average delay RU – Video streaming.

For all environments except rural indoor, the ratio video duration vs. average delay is between 4% and 11%. For rural indoor environment, that ratio is higher, 37%, due to the same reason presented earlier for the variation of SNR in rural indoor. From the viewpoint of the reference user, a ratio video duration vs. average delay up to 10% is acceptable but long waiting times can be frustrating for the user.

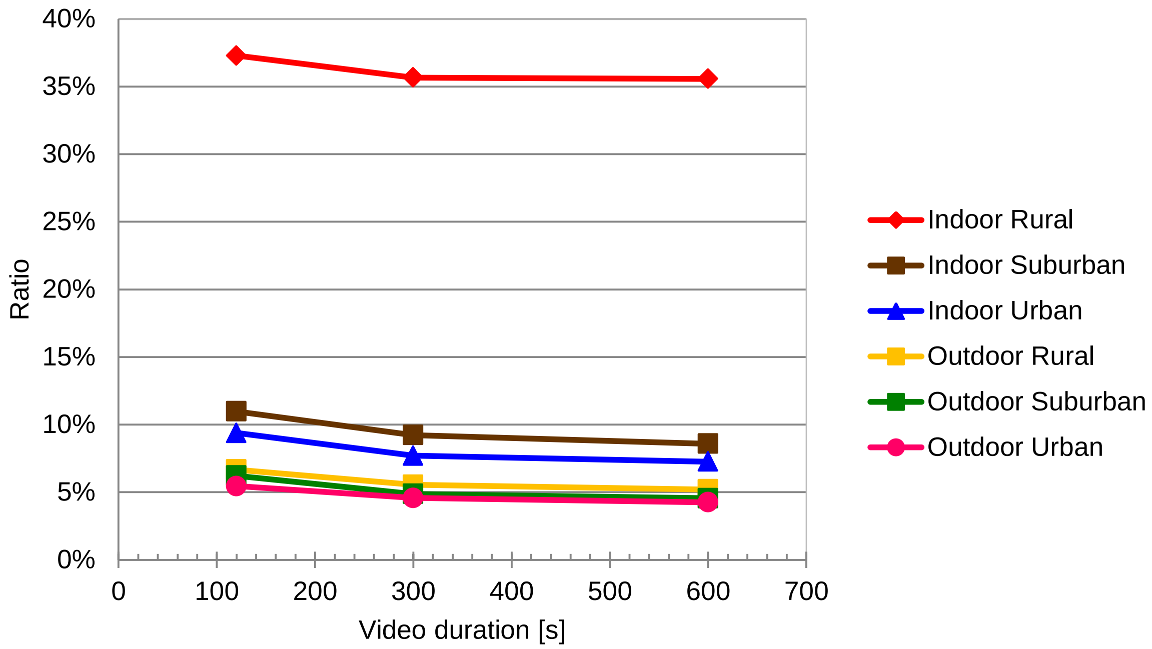


Figure 4.4 – Ratio average delay vs. video duration.

The other output parameter is initial delay, presented in Figure 4.5 for web browsing service, in Figure 4.6 for file download service and in Figure 4.7 for video streaming.

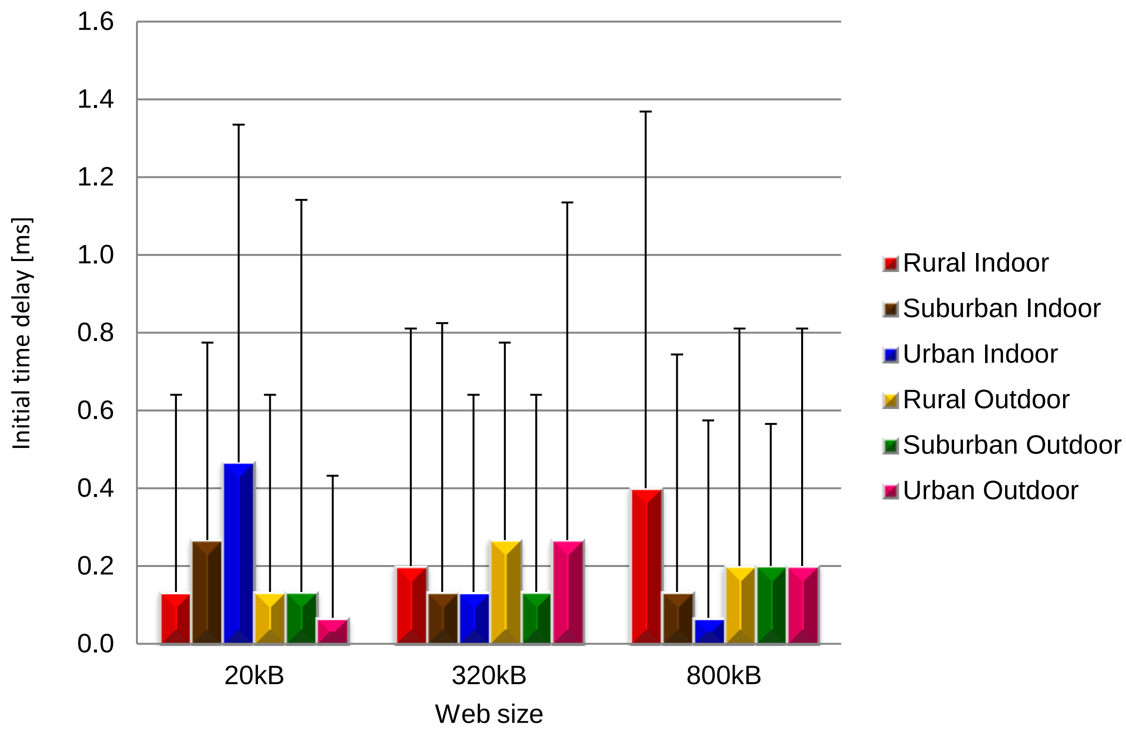


Figure 4.5 - Initial delay – web browsing.

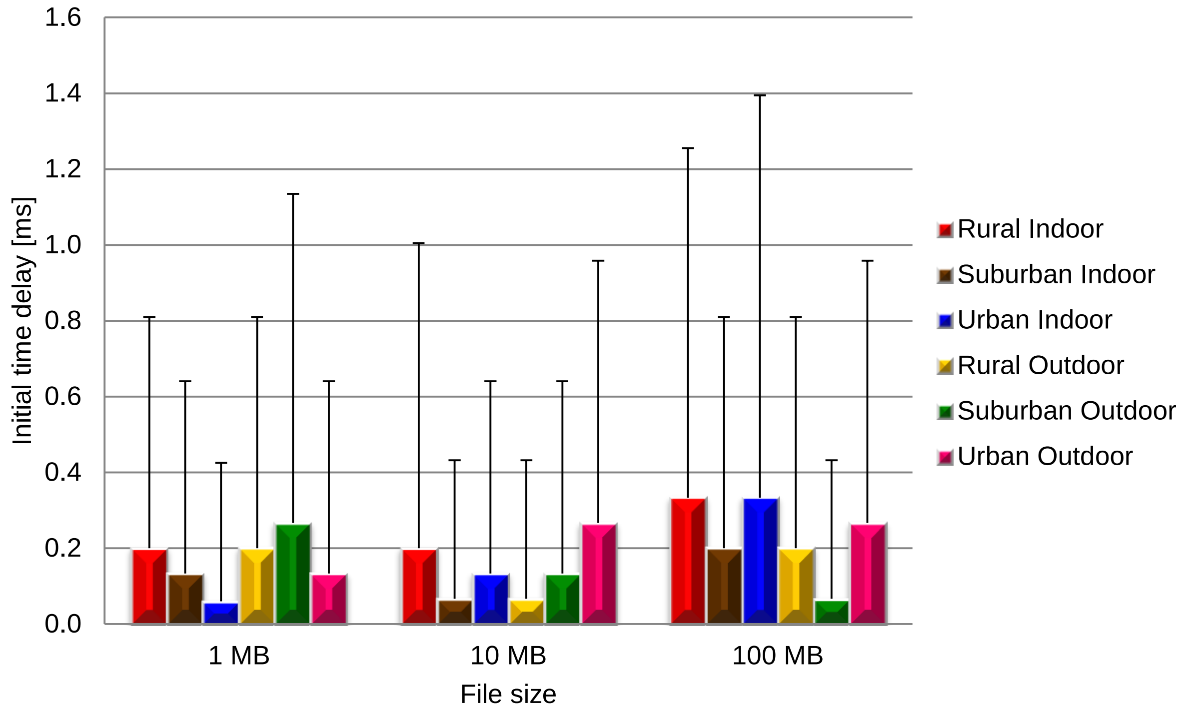


Figure 4.6 - Initial delay RU – File Download.

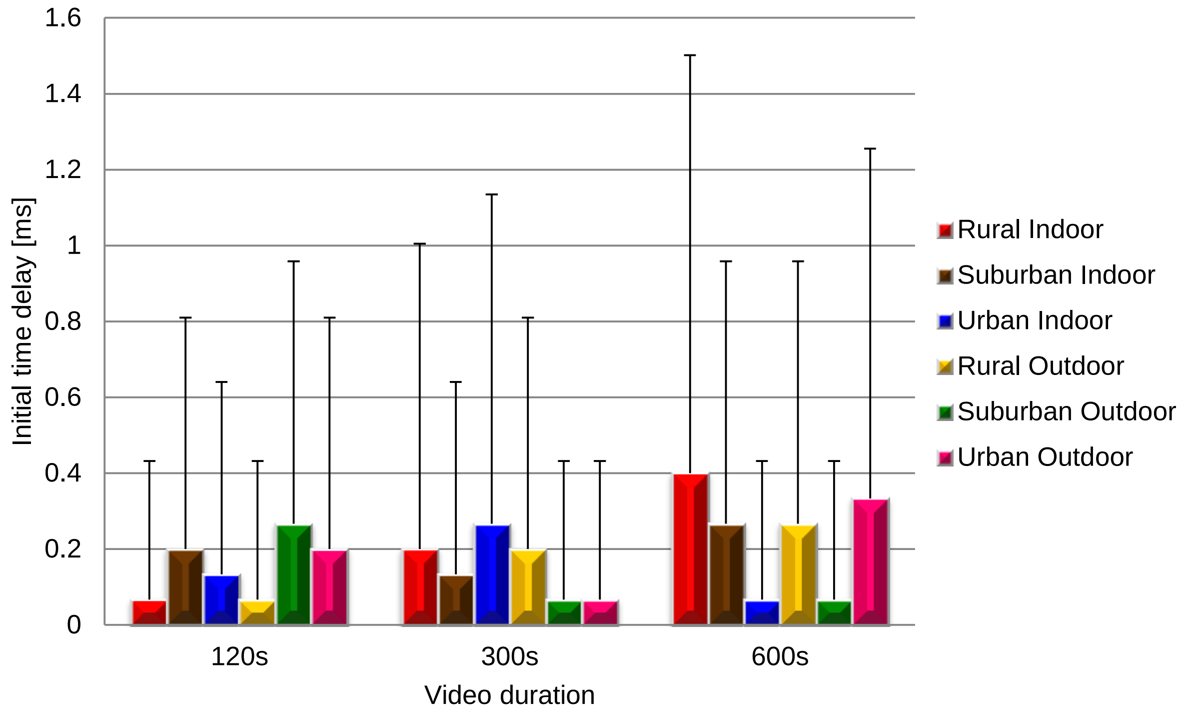


Figure 4.7 - Initial delay RU – Video Streaming.

It can be concluded that, for these scenarios, the initial delay is data, size or duration service independent. Because the values of the standard deviation are so high when compared with the initial delay and because the average values are within the standard deviation, one can say that there are no differences between the data services and the environments. The input parameter that affects the initial delay is the network load, and for this scenario the network load is a uniform distribution between 0 and 40 users. It can be concluded that, for these conditions, the reference user is not sensitive to the initial delay. The influence of network load in the initial delay is presented in Section 4.4.

4.3 Environment Rural, Suburban and Urban

In this sub section one evaluates the impact of the rural, suburban and rural environment, in indoor and outdoor situations. For the web browsing data service of 20 kB, 320 kB and 800 kB, the environment is changed from rural, suburban and urban in an indoor situation, and the same environments is used for an outdoor situation. As shown in Figure 4.8, for each web size the average bit rate increases when the environment changes from rural, to suburban, to urban and from indoor to outdoor. These differences are due to SNR in the different environments, as it can be seen in Figure D.4, Figure D.5 and Figure D.6. The average bit rate variation from rural indoor to urban outdoor does not have a progressive variation. Like it was previously shown, for lower web page sizes the download occurs in a few T_C, which is also the reason for a large standard deviation, which is reduced when the web file increases.

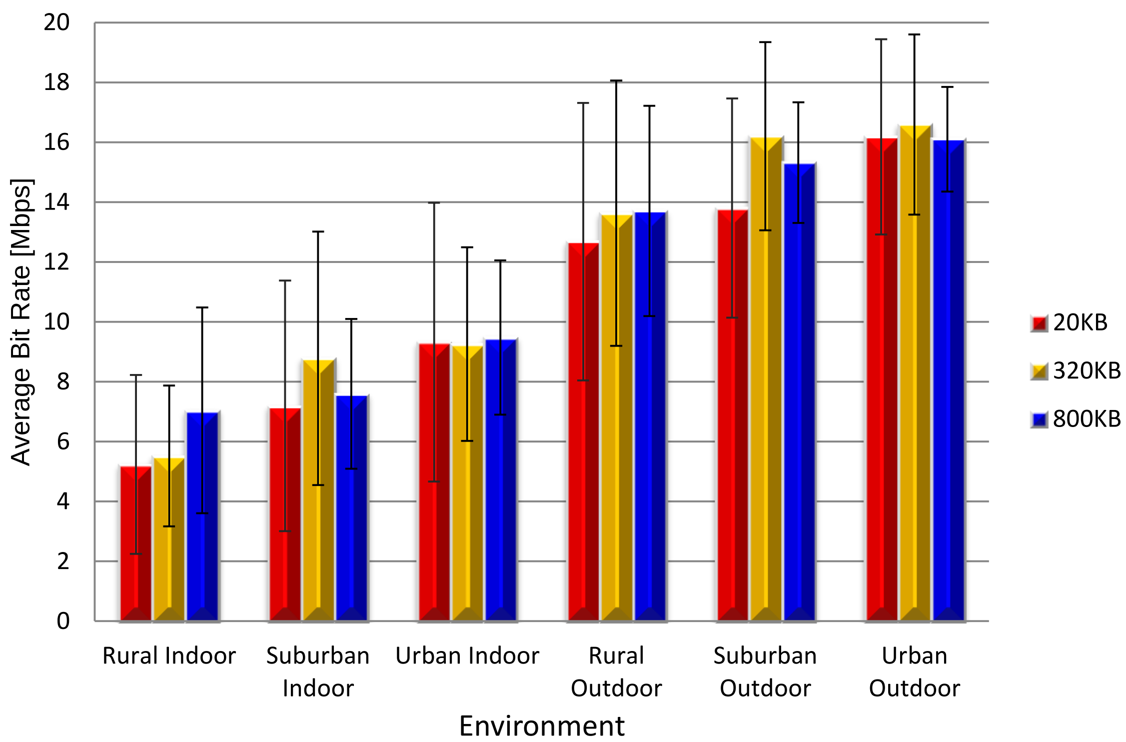


Figure 4.8 - Environment - Web Browsing.

The standard deviation for 20 kB web pages is, for the indoor environments, about 50% of the average bit rate value, decreasing to around 30% for web pages with 800 kB. For the outdoor environment, the standard deviation starts at 30% for web pages with 20 kB, and it is near to 10% with web pages of 800 kB. It can be observed that for a RU doing web browsing in an indoor rural environment, that he/she would have plus 6 Mbps if in an urban outdoor environment.

For the data service file download, the RU is downloading 1 MB in a rural indoor environment. Like in web browsing, the average bit rate increases when there is a change from rural, suburban and urban environments in indoor situations, and the same environments in outdoor situations. As shown in Figure 4.9, the average bit rate is almost higher for files of 1 MB and lower for files of 100 MB, however, the standard deviation is, for files of 1 MB, around 30% in the worst environment and 1% for urban outdoor. This means that the larger files have the most consistent results. It can be assumed, for the web browsing and file download data services, that the average bit rate is proportional to the SNR of the environment.

For the same environment and the same file size, one has evaluated the ratio, between the average bit rate for indoor and outdoor situations, which is about 50%. This means that just by the mere fact of being in an indoor situation the reference user has less 50% of the average bit rate.

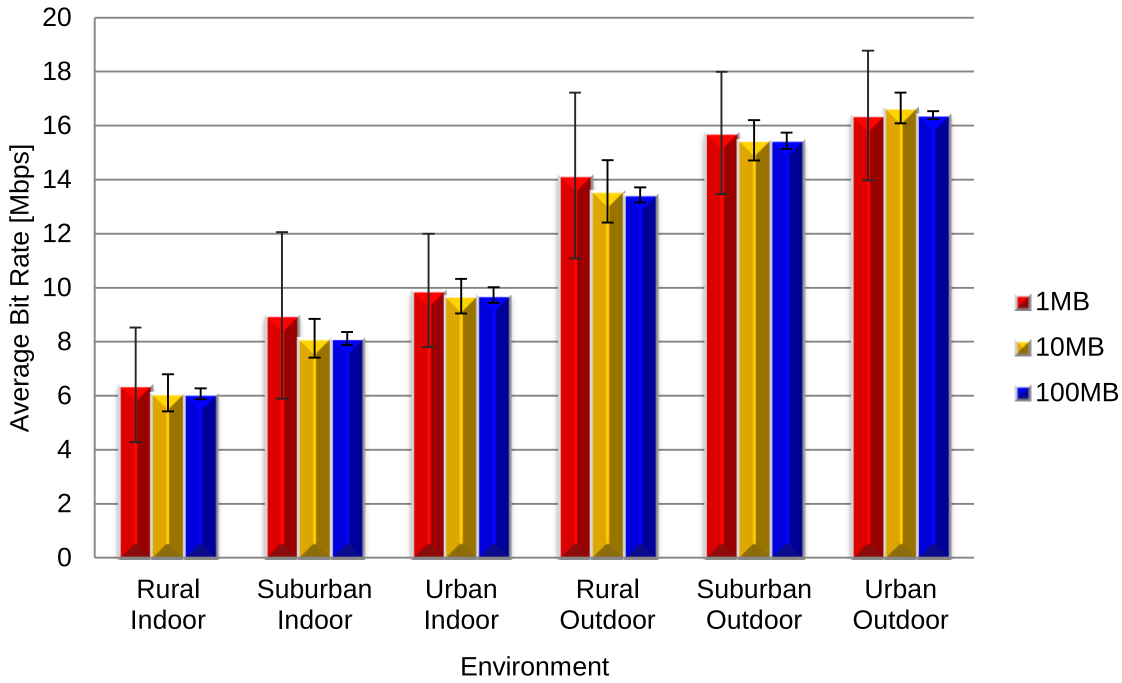


Figure 4.9 – Environment – File download.

Considering the RU doing 120 s of video streaming, in a rural indoor environment, the environment is changed from rural indoor, to suburban also indoor and after that to the same outdoor situations, Figure 4.10.

The first impact of the video streaming in the rural indoor environment is that the RU spends 35% of the time in stand-by, waiting for the video. Once again, this leads to the conclusion on impact on the SNR of the average bit rate and consequently in the average delay.

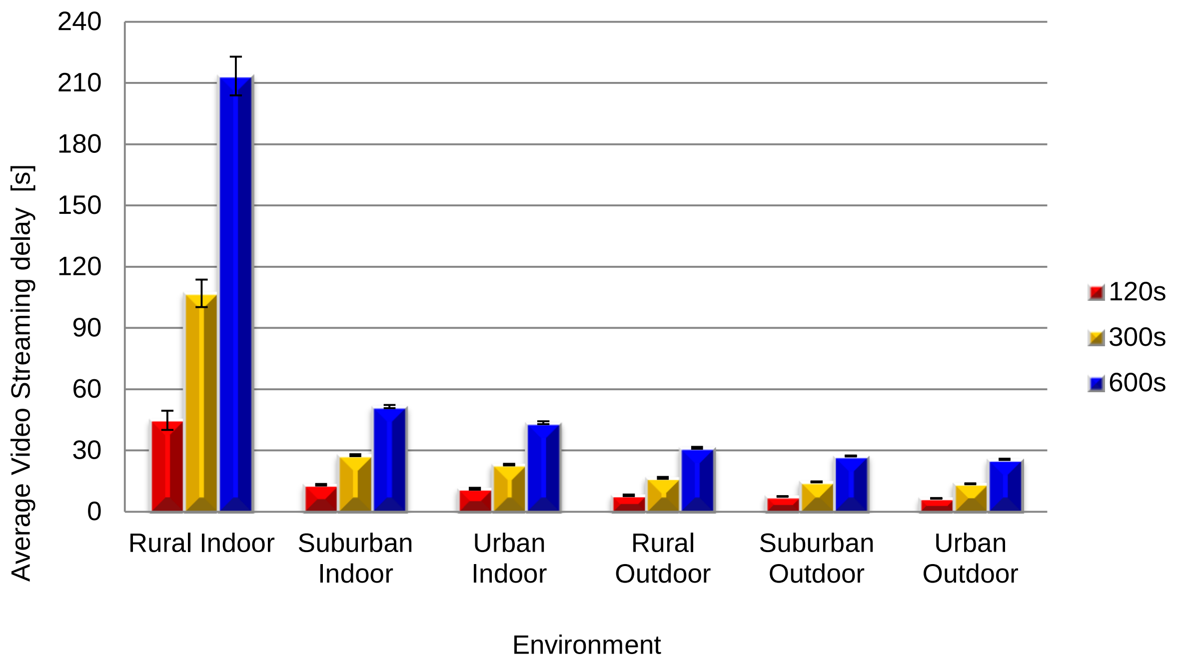


Figure 4.10 - Reference User service – video streaming.

The reason for data service video streaming to have a greater average delay, in an indoor rural environment, is due to the SNR in this environment as it was said before. Another reason is the video streaming buffer: if during a T_C the SNR is low the instantaneous bit rate is low, the buffer empties and the video stops. With the particularity that for the rural environment the T_C is bigger than in the others environments, so when in indoor rural environment the instantaneous bit rate is low, it is for longer periods than for the other environment situations. In Figure 4.11 one presents the ratio video duration versus average video stream delay.

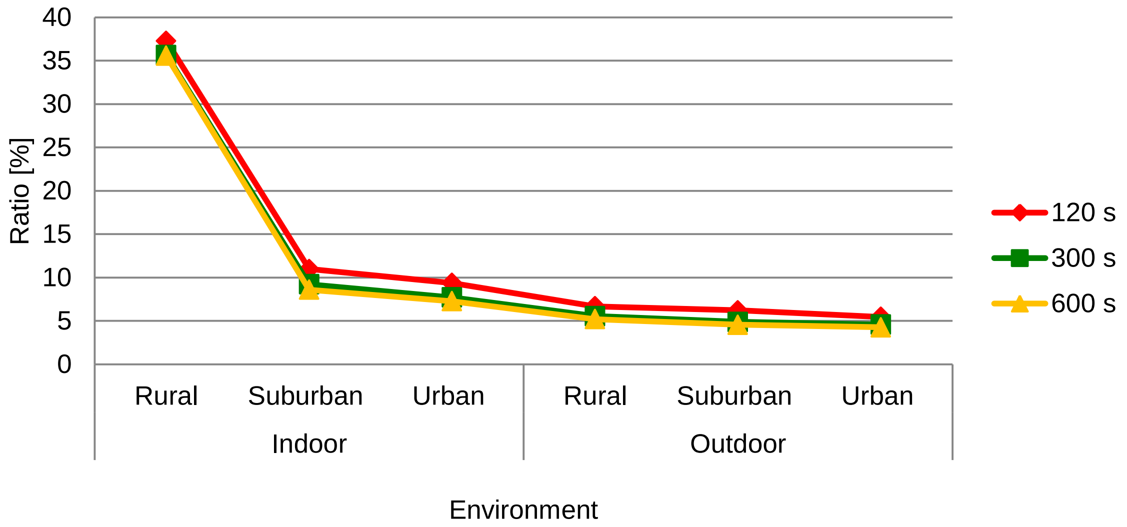


Figure 4.11 - Ratio - Video Duration versus average video stream delay.

For each environment, the ratio of the average delay between the indoor and the outdoor situations is around 50%, except for the rural environment, where this ratio is around 80%. Once again, one can conclude that long waiting times, plus than 10%, can be frustrating for the user. The behaviour of the ratio video duration versus average video stream delay for the different video durations is similar in all environments, which means that it is video duration independent.

4.4 Network load

In this subsection, one presents the influence of the network load and how data services are influenced by the number of other users. The initial delay for the web browsing, file downloading and video streaming are presented in Figure 4.12 for a rural outdoor environment. When the network has a number of users between 0 and 40, there is no influence in the initial delay. When the number of users reaches at 40, data services begin to show some initial delay. For web browsing and file download this delay is around 0.9 ms and 0.8 ms respectively, which are very similar values. When the number of other users increases to 60, the initial delay is 8 ms and 11 ms for web browsing and file download respectively. This difference in the values is due to the fact that the file download has greater file size than the web browsing and consequently more time frames. In video streaming, like in the other data services, a number of users between 0 and 40 has no influence in the initial delay. For 40 users the initial delay is 2 ms and for 60 users there is around 6 ms of initial delay. The standard deviation of the initial delay in these data services has a higher a value compared to the initial delay; one can say that the average initial delay is within the standard deviation. For this reason, it can be assume that the average initial delay is equal for all data services and for the same network load, and that it is below the human sensibility about 200 ms; it can be assumed that the reference user is not sensitive to the initial delay. This evaluation was done for a rural outdoor environment, but if it was done for an indoor environment the initial delay would be higher and the user would be sensitive to it, once again due to the SNR.

Concerning the average bit rate for web browsing and file download data services, the variation of average bit rate with the network load is in Figure 4.13. It is clear that the average bit rate decreases with the increase in the number of users. However, the standard deviation is more than 50% of the average bit rate for web browsing, and for file download changes from 31% for 0 other users to 8% for 60 other users. These differences in the standard deviation are due to the fact that the size of the web page is 20 kB, so only needs a T_C to download the page in each simulation the instantaneous bit rate is different, for file download with files of 1MB, needs more than one T_C and consequently the standard deviation is lower.

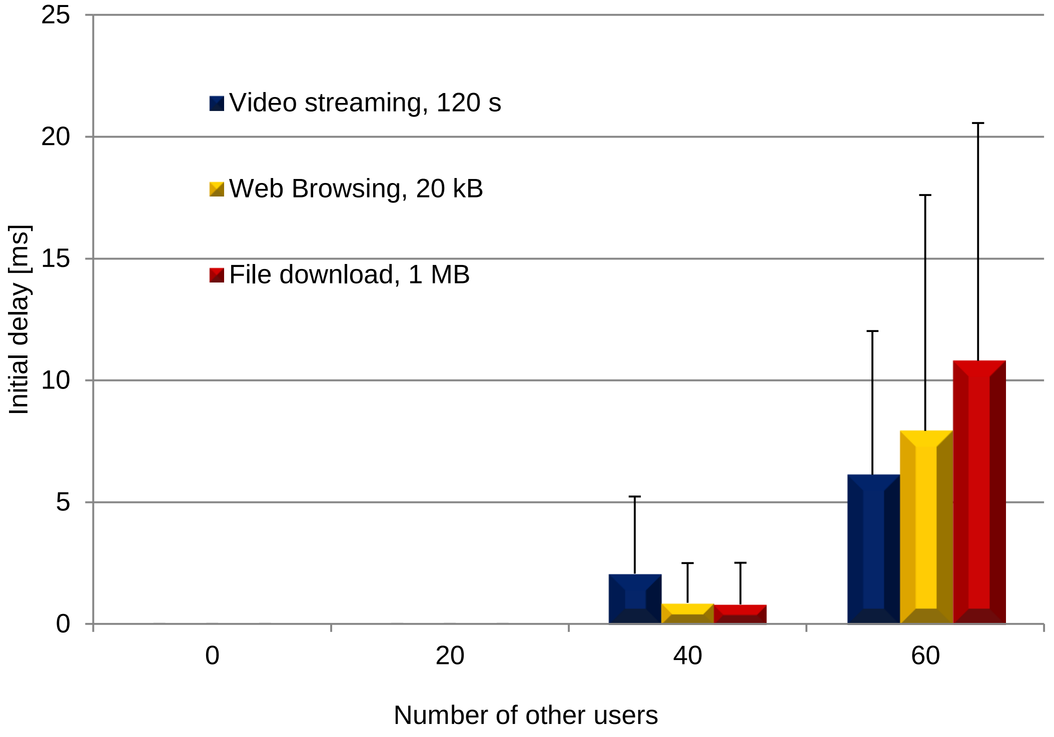
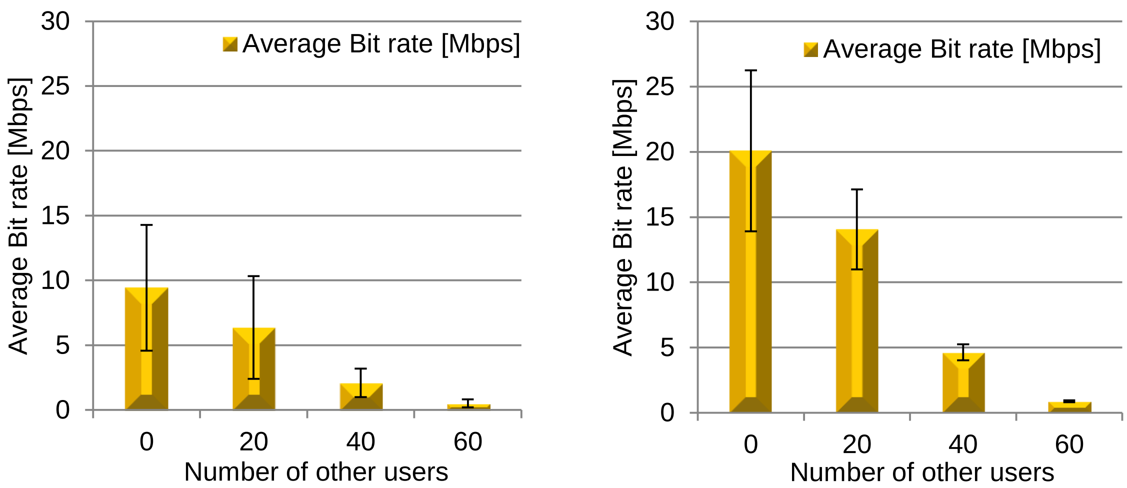


Figure 4.12 – Variation of initial delay with the network load, in rural outdoor environment.



a) Web browsing, 20kB

b) File download, 1MB

Figure 4.13 – Variation of average bit rate with the network load, in rural outdoor environment.

When the number of users changes from 0 to 20, the average bit rate decays around 30%, for web browsing and file download, When this number changes from 20 to 40 users, the average bit rate decays about 67%, and when it changes to 60 it is almost 80%.

It is clear that the average bit rate is affected by the number of users for the web browsing and file download data services. When users are 60, the average bit rate is in the threshold of the service metrics.

The average delay of the video streaming data service is evaluated in Figure 4.14. For this data service, when the number of users is 0 or 20 the ratio video duration-average delay is 2%, which is very low, but when the network load reaches 40, the ratio video duration-average delay grows. For a video streaming of 120 s, the RU spends more 35% of the time waiting. In the extreme case of 60 users the average delay reaches 233 s, which means that the RU spends waiting twice as much time as the video length. In this case, the ratio video duration-average delay is almost 200%.

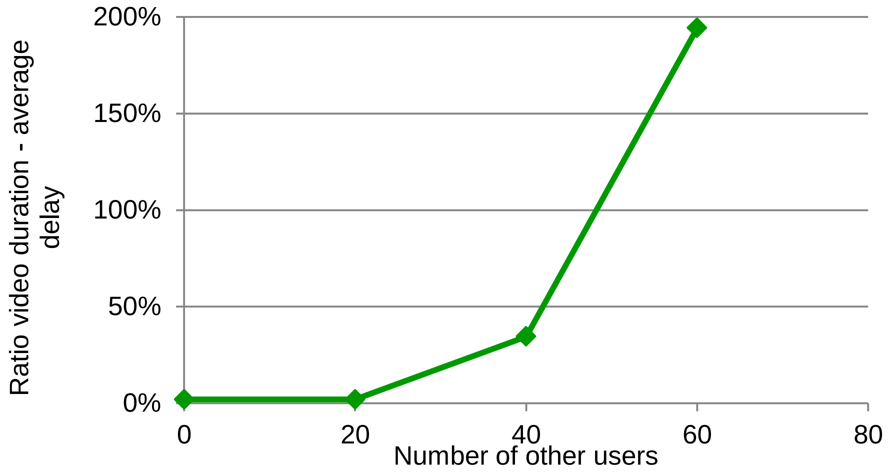


Figure 4.14 – Variation of average delay with the network load.

This change in the network load can happen mostly in the happy hours, promoted by operators in times of lower traffic at dawn. It can be concluded that the ratio video duration vs. average video stream delay is mostly sensitive to the number of other users.

4.5 Server location

When travelling by fibre, the server in London has an additional delay of 8 ms and the server in Houston has an additional delay of 48 ms. When the travel is done using the satellite, the geostationary satellites are at a minimum of 36 000 km from earth, so the distances Lisbon-London and Lisbon-Houston can be neglected when compared with the geostationary distance, and the additional delay for travelling by satellite is 240 ms.

To analyse the impact of server location in the output parameters, one considered the streaming video data service, with 120 s, 300 s and 600 s in a rural indoor environment and urban outdoor environment. Thus, for this data service, and for the three server locations (Lisbon, London and Houston), the initial delay is presented in Figure 4.15. The ratio Video Duration - Average video stream delay is for rural indoor environment also analysed in Figure 4.16 and for urban outdoor environment in Figure 4.17. The initial delay when the server is in Lisbon was analysed in Section 4.1

and is neglected from the viewpoint of reference user. When the RU is performing a video streaming data service, he/she does not know where the server is, or what support is used to bring him/her the data he has requested. If the server is in London this can be done by one of two ways: submarine communications cable or satellite. The initial delay is about 8 ms for submarine communications cable and 30 times more, about 240 ms, for satellite. On the other hand, if the server is in Houston, the initial delay is about 48 ms for submarine communications cable and 5 times more, about 240 ms, for satellite.

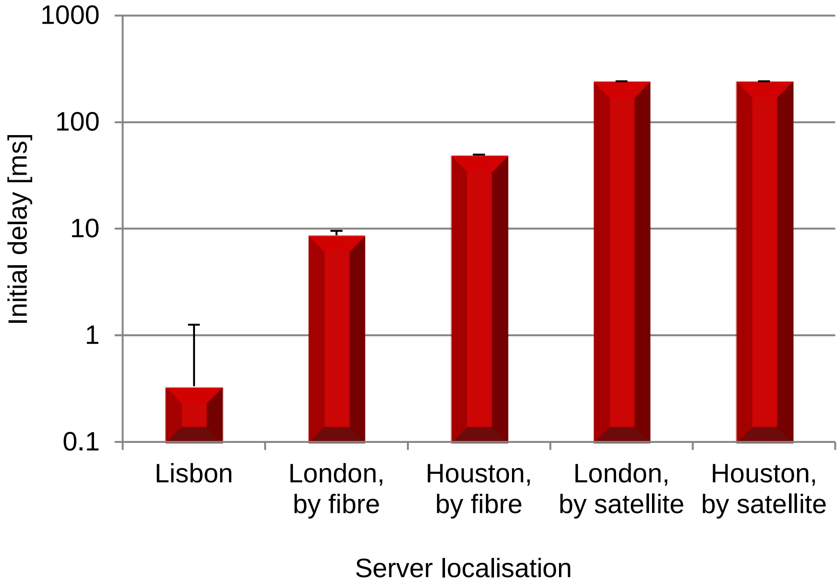


Figure 4.15 – Initial delay for Video streaming service, with server in different localisations.

This means, according to the location of the server, that the RU could have initial delays up to 240 ms without any intervention from the operator.

For the average delay, when the RU is placed in a rural indoor environment and when the server is in Lisbon, the ratio Video Duration - Average video stream delay, Figure 4.16, is almost 40%. For a server located in Lisbon (LIS), the ratio Video Duration - Average stream delay stream delay has the same value irrespective of the length of the video in the same environment, which does not happen if the server is in London, the difference being more aggravated if the server is in the USA. If the server is in London (LON-f) instead of Lisbon, and data arrives via submarine communications cable, the ratio in rural environment grows and is about 44% for a 120 s video duration and 37% for a video of 600 s. If the environment is urban outdoor the ratio is about 10%.

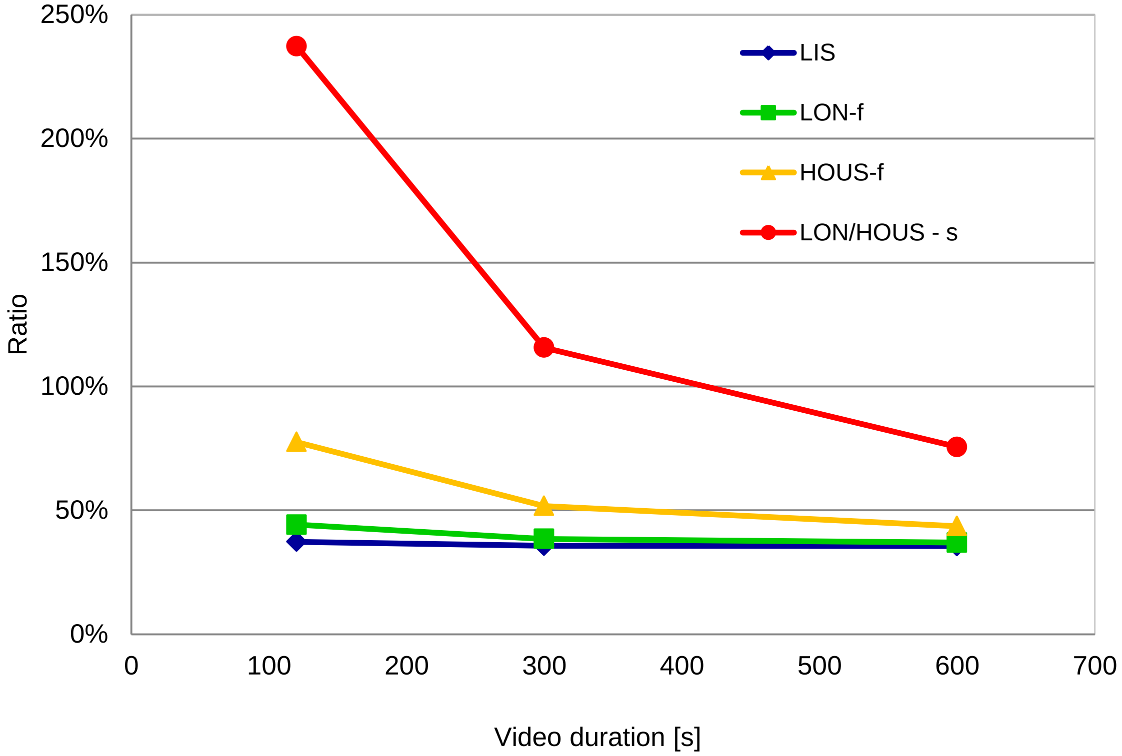


Figure 4.16 – Ratio Average video stream delay - Video Duration, with server in different locations RU in Rural indoor environment.

When the server location changes to Houston (HOUS-f), data arriving via submarine communications cable starts to show differences, especially for short videos. The ratio Video Duration - Average video stream delay, for a 120 s video duration is almost 80% for rural indoor environment and almost 50% for urban outdoor. For larger videos, 600 s, in rural indoor the ratio Video Duration - Average video stream delay is above 40% for rural and above 10% for urban.

If the server is in London or Houston (LON/HOUS-s) and the transmission is via satellite, the ratio is flat different. The differences between the environments are not so evident among them, however, for a video of 120 s, the ratio Video Duration - Average video stream delay is in the worst case almost 240%. The average delay has greater impact on short videos, mainly when the average delay is larger.

It can be concluded, Figure 4.17, that when the reference user is the rural indoor environment the ratio Video Duration - Average video stream delay is always above 10%, which means long waiting times that are frustrating for the user. In the urban outdoor environment only the videos with a duration greater than 300 s meet this specification below 10%.

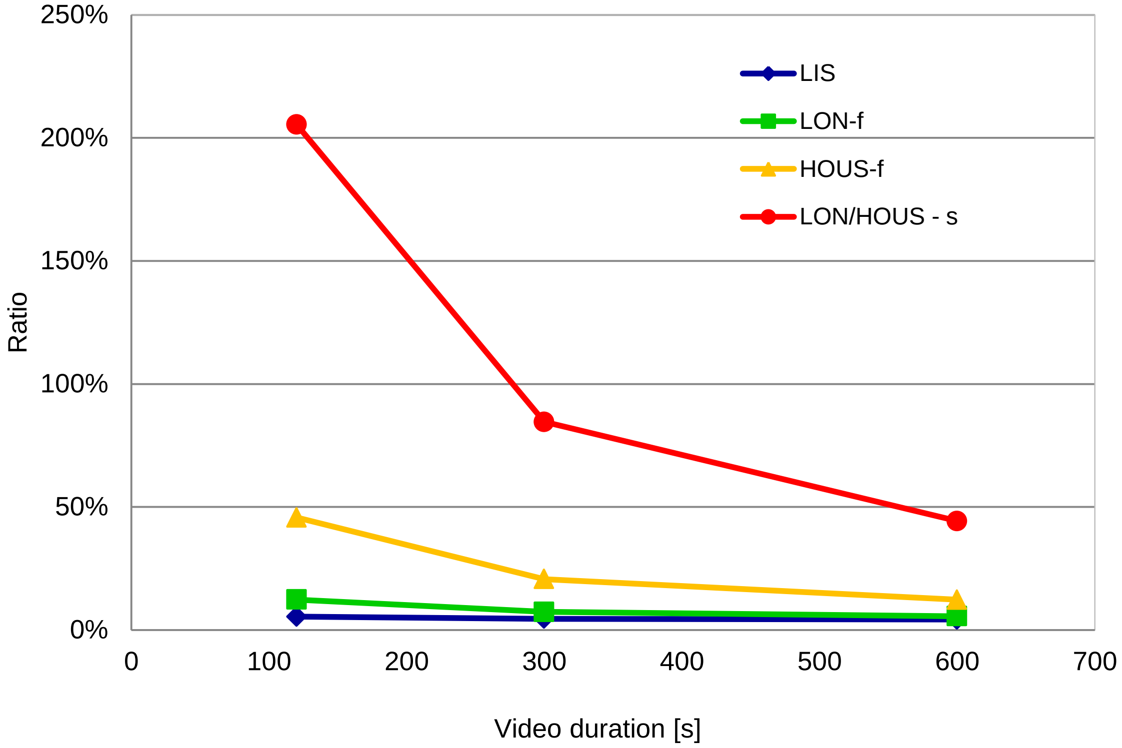


Figure 4.17 – Ratio Average video stream delay - Video Duration, with server in different locations RU in Urban outdoor environment.

These differences are explained by the differences in SNR between the rural indoor environment, and urban outdoor, 3 and 18 dB respectively. For short videos durations these differences are more evident because the video needs, in most cases, a unique coherence time to do the video stream.

Chapter 5

Conclusions

This chapter summarises the main conclusions of the present work. Major results are highlighted and future work guidelines are drawn.

The main objective of this thesis was to evaluate UMTS/HSPA Quality of Service and Quality of Experience in terms of initial delay, average delay and average throughput for the chosen data service, in different scenarios and with a specific network load.

Chapter 2 focuses on the network architecture with the radio access network, the core network and the user equipment. The section of the radio interface includes the main WCDMA parameters as channelisation and modulation. The interference section includes a brief description of the interference phenomenon, and shows the interference models available in literature. Afterwards, traffic classes are considered with the fundamental characteristics, priority and examples of application. The metrics section describes the type of metrics, and shows several approaches for their calculation. Finally, the state-of-the-art is overviewed, considering the most relevant works.

Chapter 3 presents the developed models. Three environments were chosen (rural, sub urban and urban) and for each of these environments an indoor and outdoor situation was defined. One considers the variation of SNR with each environment, and the corresponding instantaneous bit rate, within each coherence time. The coherence time of the channel is not always the same, changing more quickly in the urban environment. One can observe the direct relationship between the SNR and the instantaneous bit rate. The changes in the SNR values are very random, even for the same environment. If the RU is indoor, he/she could have an instantaneous bit rate change of almost 60% for example for a SNR of 20 dB outdoor and 10 dB indoor, and this corresponds to 13.1 Mbps and 21.8 Mbps, respectively.

The data services in study are streaming, web browsing and file download. For each service one defined three video durations, 120 s, 300 s and 600 s, three web sizes 20 kB, 320 kB and 800 kB and three download files of 1 MB, 10 MB and 100 MB. These data service options were tested in all environments and, as output, one considered, the initial delay for all data services, the average delay for video streaming, and average throughput to web browsing and file download. One has performed 56 simulations.

The implementation of the model was performed in Matlab, where the input are: available R99 and HSDPA channels, number of other users, RU environment, RU location indoor or outdoor, service penetration, SNR variation, data sizes, and video durations. Auxiliary scripts were also used and the output parameters are: initial delay, average delay and average bit rate.

The assessment of the algorithm was made to validate the code as well the convergence potential of the output parameters. With 30 simulations, it had a convergence potential better than 4%.

Another influence in the bit rate is the network load, the network load was simulated with a variable number of users and according to a service penetration. The services for the network load are voice, which has priority over other services, audio streaming, file download and web browsing. For the network, one used a mixed system UMTS and HSDPA, with a shared channel and with an exclusive HSDPA channel. With these resources, one evaluated the number of users that the network supports. One defined the other users varing between 0 and 40 with a uniform distribution.

In this study, the reference user is performing three different data services, and it is possible to conclude that the initial delay is independent either from the data service or from the environment of the RU. It is also inconclusive about the average initial delay, because the average values for all environments are within the standard deviation. It can be concluded that for these conditions the reference user is not sensitive to initial delay. For both web browsing and file download services, one can observe a growth evolving for each size of web and file. Comparing these values of average bit rate with the service metrics, one can conclude that for, web browsing and file download, are better than the metrics [512, 1536] kbps.

For the same environment and the same file size, the ratio between the average bit rate for indoor and outdoor situations was evaluated, and one can conclude that it is about 50%. This means that just by the mere fact of being in an indoor situation the reference user has less 50% of the average bit rate.

For the video streaming data service at each environment, the average delay ratio between the indoor and the outdoor situations is around 50%, except for rural environment where it is around 80%. Once again, one can conclude that long waiting times, more than 10%, can be frustrating for the user. The behaviour of the ratio of video duration versus average video stream delay for the different video durations is similar in all environments; which means that it is video duration independent.

However, when the server of the data service is not in Lisbon the initial delay starts to have higher values, this initial delay being affected by the means used for transmission: submarine communications cable or satellite. For a server located in London, the RU can have differences in the initial delay of 8 ms or 240 ms according to the transmission being made by submarine cable or by satellite. For a server located in Houston, the difference is higher for a transmission using submarine cable 48 ms, being also 240 ms for satellite transmission. The RU initial delay can be affected in about plus 30% by the location of the server and by the means of transmission: submarine cable or satellite.

When studying the impact of network load in the initial delay, results are as expected. When the number of users grows, the initial delay also grows. Up to 20 users in the network have no impact on the initial delay, when is number grows to 40 and to 60 the initial delay grows but without impact in the RU because the value is about 10 ms for 60 users. This evaluation was done for a rural outdoor environment, but if it was done for an rural indoor environment the initial delay will be higher and the user would be sensible to it, once again due to the SNR.

For the video streaming data service, the output in study is the average delay. For this data service, one can conclude that the RU location has direct influence on the average delay for a video, which means this is related to the SNR of the RU environment, the indoor rural environment being the most prominent case. The ratio video duration - average delay for rural indoor environment is about 35%. For a server located in Lisbon, the average delay is independent of the video duration in the same environment.

When the location of the server changes, the ratio video duration - average delay, greatly increases and for the same RU environment the video duration has implications. So, for the rural indoor environment and for a server located in Lisbon (LIS) the RU has a ratio video duration - average delay of 35%, for a submarine communications cable and a server located in London (LON-f) and Houston

(HOUS-f) the RU has a ratio of about 40% and within 40 – 80%, respectively. The range 40 – 80% is for a video duration of 120 s and 600 s respectively. However if the transmission is made by satellite, for both servers (LON/HOUS-s), the ratio video duration - average delay will increase almost until 250% for a video with a duration of 120 s, for a video with 600 s the ratio is about 75%. For the most favourable environment, urban outdoor, the ratio video duration - average delay is not so pronounced, for a server located in London (LON-f) and Houston (HOUS-f) and transmission using submarine communications cable, the ratio is about 10% and between 10% and 44% according to the video duration, respectively. Whereas, for transmission by satellite from both servers, London and Houston (LON/HOUS-s), the ratio video duration - average delay, increases up to 200% for a video with a duration of 120 s, and for a video with 600 s the ratio is almost 50%.

One can conclude, when the server is not in Lisbon, that the video duration has implication on the ratio video duration - average delay, being more pronounced when the video is short. Once again the location of the server and the means used for transmission affects the ratio video duration – average delay.

For the web browsing and file download data service, it can be assumed they are similar services, because this is a statistical process and the reading time was not taken in to account, the difference being on the size of the file. One can conclude that the average bit rate is related to the SNR of the environment, and is independent of the file sizes.

For the smallest file sizes, the standard deviation is larger, being possible to conclude that the average bit rate for smallest file sizes varies more. The file is downloaded in a few, or even one coherence time, which means unique instantaneous bit rate, which can be small.

The impact of the network load on the initial delay is the expected: when the number of users grows the initial delay increases. The standard deviation is of the same order of magnitude of the initial delay. However, there is no impact, because the values are below 200 ms.

For the web browsing and file download data services, the impact of the network load is the decrease of the average bit rate with the increase of the other users. For both services, when the number of users grows from 0 to 20, the average bit rate decreases about 30%, when it grows to 40 users decreases 67%, and it when grows to 60 users around 80%. If the data service is video streaming the impact is in the average delay. For this data service, when the number of users is up to 20 the ratio video duration-average delay is very low, 2%, when the network load arrives at 40 the ratio video duration-average delay grows, for a video streaming of 120 s the RU spends more than 35% of the time waiting. For the extreme case of 60 users, the average delay reaches 233 s, which means that the RU spends waiting twice as much time as the length of the video because the ratio video duration-average delay is almost 200%.

For future work, one suggests an improvement in the characterisation of data services including, in particular, the possibility of file sizes and the videos duration can be statistical. Another possibility of study is to evaluate RU mobility. It would be important to evaluate the impact of velocity as a consequence of motion. Motion and velocity have an impact on the handover, in the initial delay and in average bit rate. The study could be extend to other services, like voice using voice over IP and games online, which are services with high priority on the mobile systems. The deployment of the next generation systems, LTE, could be of great interest to evaluate, since the latency is much lower and the bit rate is very high, potentially providing QoS and the QoE equally excellent.

Annex A

HSPA Categories and Throughput Capabilities

This annex presents user equipment categories, FDD E-DCH physical layer categories, HSDPA and HSUPA throughput capability – theoretical perspective.

Table A.1. – User Equipment categories (Adapted from [Jaci09]).

UE Category	***Modulation schemes***	***Maximum number of codes***	***Minimum inter-TTI interval***	***Maximum transport block size***	***Maximum theoretical peak data rate [Mbps]***
1	QPSK/16QAM	5	3	7298	1.22
2	QPSK/16QAM	5	3	7298	1.22
3	QPSK/16QAM	5	2	7298	1.82
4	QPSK/16QAM	5	2	7298	1.82
5	QPSK/16QAM	5	1	7298	3.65
6	QPSK/16QAM		1	7298	3.65
7	QPSK/16QAM	10	1	14411	7.21
8	QPSK/16QAM	10	1	14411	7.21
9	QPSK/16QAM	15	1	20251	10.20
10	QPSK/16QAM	15	1	27952	14.40
11	QPSK	5	2	3630	0.91
12	QPSK	5	1	3630	1.82
*13**	QPSK/16QAM/64QAM	15	1	35280	17.64
*14**	QPSK/16QAM/64QAM	15	1	42192	21.10

* Introduced by release 7 of the 3GPP specifications.

Nowadays, one has HS-DSCH categories up to 32 that allow supported modulations MIMO operation with and without aggregate cell operation without MIMO operation with aggregate cell operation and with MIMO operation and aggregate cell operation which are out of this thesis scope.

Table A.2. – FDD E-DCH physical layer categories (Adapted from [ETSI05]).

UE Category	***Modulation schemes***	***Maximum number of codes***	***Maximum transport block size [bits]***		***Peak rate [kbps]***	
			10 ms TTI	2 ms TTI	10 ms TTI	2 ms TTI
1	QPSK	1 × SF4	7110	-----	711	-----
2	QPSK	2 × SF4	14484	2798	1448	1448
3	QPSK	2 × SF4	14484	-----	1448	-----
4	QPSK	2 × SF2	20000	5772	2000	2886
5	QPSK	2 × SF2	20000	-----	2000	-----
6	QPSK	2 × SF4 + 2 × SF2	20000	11484	2000	5742
7*	QPSK/16QAM	2 × SF4 + 2 × SF2	20000	22996	2000	11498

* Introduced by release 7 of the 3GPP specifications.

Currently, one has categories 8 and 9 that support QPSK, 16QAM and QPSK, respectively in Dual Cell E-DCH operation, which are out of this thesis scope.

In Table A.3 and Table A.4 are present the HSDPA and HSUPA throughput capability, respectively.

Table A.3. – HSDPA throughput capability – theoretical perspective (adapted from [John08]).

MCS	***Modulation***	***Coding Rate***	***5 HS-PDSCH codes [Mbps]***	***10 HS-PDSCH codes***	***15 HS-PDSCH codes***
1	QPSK	0.25	0.6	1.2	1.8
2		0.50	1.2	2.4	3.6
3		0.75	1.8	3.6	5.4
4	16QAM	0.50	2.4	4.8	7.2
5		0.75	3.6	7.2	10.8
6		1.00	4.8	9.6	14.4
7	64QAM*	0.50	3.6	7.2	10.8
8		0.75	5.4	10.8	16.2
9		1.00	7.2	14.4	21.6

* Introduced by release 7 of the 3GPP specifications.

The throughput figures presented in Table A.3 do not necessarily reflect the maximum throughput which can be achieved in practice. Not all HSDPA UE are capable of achieving the complete set of bit rates and UE are categorised according to their ability. This UE category information is signalled to the network during connection establishment to ensure that both the RNC and Node B are aware of the UE capability, e.g. to ensure that 16QAM is not used for a UE which only supports QPSK. The set of HSDPA UE categories specified by 3GPP is presented in Table A.1.

Table A.4. – HSUPA throughput capability – theoretical perspective (adapted from [John08]).

Modulation	***Coding Rate***	***1 E-DPDCH 1 × SF4 Code [kbps]***	***2 E-DPDCH 2 × SF4 Codes [kbps]***	***2 E-DPDCH 2 × SF2 Codes [Mbps]***	***4 E-DPDCH 2 × SF4 Codes + [Mbps]***
BPSK (QPSK)	0.50	480	960	1.92	2.88
	0.75	720	1.44	2.88	4.33
	0.75	960	1.92	3.84	5.76
4PAM (16QAM)	0.50	960	1.92	3.84	5.76
	0.75	1.44	2.88	5.76	8.64
	1.00	1.92	3.84	7.68	11.52

The throughput figures presented in Table A.4 do not necessarily reflect the maximum throughput which can be achieved in practice. Not all HSUPA UE are capable of achieving the complete set of bit rates and UE are categorised according to their ability. This UE category information is signalled to the network during connection establishment to ensure that both the RNC and Node B are aware of the UE capability, e.g. to ensure that the resources corresponding to 2×SF4 + 2×SF2 are not allocated to a UE which only supports 2×SF4.

Annex B

HSPA Throughput Models

The present Annex, presents the HSPA models, which shows the SINR and throughput of each considered system for several system configurations.

Previous works, such as [Jaci09], have produced interpolated experimental expressions to precise SINR, ρ_{IN}, as a function of throughput, R_b, and vice-versa for several types of antenna configurations and modulations. As stated before, all throughputs obtained are valid for the physical layer, and the authors ensure the extrapolations do not have relative mean errors greater than 5%, which are acceptable for this kind of approximations. Some minor errors present in the equations were corrected, such as discontinuities between curves and the absence of saturation curves.

Nevertheless, the importance to extend the measurements among different radio channels connoted by different characteristics is significant. Estimations for a Vehicular A channel were not properly done due to the lack of simulations for HSPA+ with the necessary assumptions, however, the HSDPA curve of SINR as a function of physical throughput for a Vehicular A channel in [HoTo04] is extrapolated to HSPA+, shifting down the Pedestrian channel A curve in 1 dB.

Considering a SISO configuration with QPSK, for DL, one has:

$$\rho_{IN}[dB] = \begin{cases} -0.0271 \times {R_b}^6 + 0.47465 \times {R_b}^5 - 3.36 \times {R_b}^4 + 12.33 \times {R_b}^3 \\ -24.86 \times {R_b}^6 + 27.77 \times R_b - 14.74, \qquad 0.7 \leq R_{b[Mbps]} < 4.5 \\ \\ -0.0649 \times {R_b}^4 + 1.581 \times {R_b}^3 - 13.81 \times {R_b}^2 + 53.78 \times R_b \\ -76.46, \qquad 4.5 \leq R_{b[Mbps]} \leq 7.2 \end{cases} \tag{B.1}$$

Considering a SISO configuration with 16 QAM, for DL, one has:

$$\rho_{IN}[dB] = \begin{cases} -0.0541 \times {R_b}^6 + 0.9496 \times {R_b}^5 - 6.7214 \times {R_b}^4 + 24.6466 \times {R_b}^3 \\ -49.805{R_b}^2 + 55.0299 \times R_b - 31.1894, \qquad 0.7 \leq R_{b\,[Mbps]} < 4.5 \\ \\ -0.0319 \times {R_b}^2 + 1.7534 \times R_b - 6.9882, \qquad 4.5 \leq R_{b[Mbps]} < 9.7 \\ \\ 0.1529 \times {R_b}^3 - 5.1218 \times {R_b}^2 + 57.816 \times R_b - 211.471, \qquad 9.7 \leq R_{b[Mbps]} \leq 14.4 \end{cases} \tag{B.2}$$

For a SISO configuration with 64 QAM, for DL, the SNR is given by:

$$\rho_{IN}[dB] = \begin{cases} -0.0541 \times {R_b}^6 + 0.9496 \times {R_b}^5 - 6.7214 \times {R_b}^4 + 24.6466 \times {R_b}^3 \\ -49.805 \times {R_b}^2 + 55.0299 \times R_b - 31.1894, \qquad 0.7 \leq R_{b\,[Mbps]} < 3.7 \\ \\ 1.3691 \times R_b - 5.8516, \qquad 3.7 \leq R_{b[Mbps]} < 8.7 \\ \\ 0.9565 \times R_b - 2.3371, \qquad 8.7 \leq R_{b[Mbps]} < 20.0 \\ \\ 0.0396 \times {R_b}^2 + 0.0799 \times R_b + 1.9286, \qquad 20.0 \leq R_{b[Mbps]} \leq 21.5 \end{cases} \tag{B.3}$$

In a SIMO 1×2 configuration with QPSK modulation, for DL, the SINR can be calculated by:

$$\rho_{IN}[dB] = \begin{cases} 0.373 \times {R_b}^6 - 4.59 \times {R_b}^5 + 22.68 \times {R_b}^4 - 57.03 \times {R_b}^3 \\ +75.55 \times {R_b}^2 - 45.01 \times R_b - 0.8666, \quad 1.0 \leq R_{b\,[Mbps]} < 3.1 \\ \\ 0.0671 \times {R_b}^4 - 1.2777 \times {R_b}^3 + 8.947 \times {R_b}^2 - 24.65 \times R_b \\ +20.22, \quad 3.1 \leq R_{b[Mbps]} < 5.8 \\ \\ -0.1809 \times {R_b}^3 + 4.233 \times {R_b}^2 - 28.42 \times R_b + 62.05, \quad 5.8 \leq R_{b[Mbps]} < 7.2 \end{cases} \tag{B.4}$$

In a SIMO 1×2 configuration with 16QAM modulation, for DL, the SINR can be calculated by:

$$\rho_{IN}[dB] = \begin{cases} -0.0012 \times {R_b}^6 - 0.0171 \times {R_b}^5 + 0.0476 \times {R_b}^4 + 0.4255 \times {R_b}^3 \\ -3.251 \times {R_b}^2 + 10.0299 \times R_b - 17.1838, \quad 1.0 \leq R_{b\,[Mbps]} < 1.8 \\ \\ -0.4437 \times {R_b}^2 + 4.3888 \times R_b - 13.5340, \quad 1.8 \leq R_{b[Mbps]} < 3.2 \\ \\ 0.0661 \times {R_b}^4 - 1.2758 \times {R_b}^3 + 8.8721 \times {R_b}^2 \\ -24.7943 \times R_b + 19.3601, \quad 3.2 \leq R_{b[Mbps]} < 5.9 \\ \\ -0.1323 \times {R_b}^3 + 2.7646 \times {R_b}^2 - 17.8122 \times R_b + 36.0243, \quad 5.9 \leq R_{b[Mbps]} < 8.3 \\ \\ 0.0208 \times {R_b}^3 - 0.6278 \times {R_b}^2 + 7.276 \times R_b - 26.0464, \quad 8.3 \leq R_{b[Mbps]} < 13.5 \\ \\ 3.3333 \times {R_b}^2 - 87.6667 \times R_b + 585.0, \quad 13.5 \leq R_{b[Mbps]} \leq 14.4 \end{cases} \tag{B.5}$$

Considering SIMO 1×2 configuration with 64 QAM, for DL, one has:

$$\rho_{IN}[dB] = \begin{cases} -0.0012 \times {R_b}^6 - 0.0171 \times {R_b}^5 + 0.0476 \times {R_b}^4 + 0.4255 \times {R_b}^3 \\ -3.251 \times {R_b}^2 + 10.0299 \times R_b - 17.1838, \quad 1.0 \leq R_{b\,[Mbps]} < 2.2 \\ \\ -0.1349 \times {R_b}^2 + 2.7519 \times R_b - 11.4313, \quad 2.2 \leq R_{b[Mbps]} < 5.9 \\ \\ -0.0148 \times {R_b}^4 + 0.2876 \times {R_b}^3 - 1.6684 \times {R_b}^2 \\ +2.8789 \times R_b - 0.07, \quad 5.9 \leq R_{b[Mbps]} < 7.4 \\ \\ -0.0381 \times {R_b}^2 + 1.7802 \times R_b - 9.1641, \quad 7.4 \leq R_{b[Mbps]} < 12.4 \\ \\ -0.0158 \times {R_b}^2 + 1.4815 \times R_b - 9.0373, \quad 12.4 \leq R_{b[Mbps]} < 18.5 \\ \\ 0.6466 \times {R_b}^2 - 23.7609 \times R_b + 230.2882, \quad 18.5 \leq R_{b[Mbps]} \leq 21.5 \end{cases} \tag{B.6}$$

In a MIMO 2×2 configuration, using QPSK, for DL, the SINR is given by:

$$\rho_{IN}[dB] = \begin{cases} 0.0699 \times {R_b}^5 - 1.229 \times {R_b}^4 + 8.639 \times {R_b}^3 \\ -30.48 \times {R_b}^2 + 56.039 \times R_b - 49.1, \qquad 1.7 \leq R_{b\,[Mbps]} < 3.4 \\ -0.0276 \times {R_b}^2 + 1.988 \times R_b - 9.967, \qquad 3.4 \leq R_{b[Mbps]} < 5.6 \\ -0.579 \times {R_b}^2 + 1.9799 \times R_b - 9, \qquad 5.6 \leq R_{b[Mbps]} < 7.0 \\ 0.0003 \times {R_b}^4 - 0.0059 \times {R_b}^3 + 0.01781 \times {R_b}^2 \\ +2.144 \times R_b - 12.6, \qquad 7.0 \leq R_{b[Mbps]} < 12.0 \\ 0.0205 \times {R_b}^3 - 0.6132 \times {R_b}^2 + 8.363 \times R_b - 35.7, \qquad 12.0 \leq R_{b[Mbps]} < 14.4 \end{cases} \tag{B.7}$$

For a MIMO 2×2 configuration, with 16QAM, for DL, the SINR is given by:

$$\rho_{IN}[dB] = \begin{cases} -0.0052 \times {R_b}^6 + 0.1479 \times {R_b}^5 - 1.7114 \times {R_b}^4 + 10.2135 \times {R_b}^3 \\ -33.3531 \times {R_b}^2 + 58.6222 \times R_b - 50.9322, \qquad 1.7 \leq R_{b\,[Mbps]} < 3.4 \\ -0.0.642 \times {R_b}^2 + 1.9468 \times R_b - 10.8835, \qquad 3.4 \leq R_{b[Mbps]} < 5.6 \\ -0.0579 \times {R_b}^2 + 2.1091 \times R_b - 12.0231, \qquad 5.6 \leq R_{b[Mbps]} < 7.0 \\ -0.0704 \times {R_b}^2 + 2.3595 \times R_b - 13.1371, \qquad 7.0 \leq R_{b[Mbps]} < 12.0 \\ -0.0043 \times {R_b}^3 + 0.1489 \times {R_b}^2 - 0.8793 \times R_b + 1.6067, \qquad 12.0 \leq R_{b[Mbps]} < 14.2 \\ -0.0170 \times {R_b}^2 + 1.1714 \times R_b - 6.3410, \qquad 14.2 \leq R_{b[Mbps]} < 19.3 \\ -0.0016 \times {R_b}^3 + 0.1082 \times {R_b}^2 - 1.67553 \times R_b + 13.4935, \qquad 19.3 \leq R_{b[Mbps]} < 25.8 \\ 0.5533 \times {R_b}^2 - 28.4577 \times R_b + 381.012, \qquad 25.8 \leq R_{b[Mbps]} < 28.8 \end{cases} \tag{B.8}$$

In a MIMO 2×2 configuration with 64QAM, for DL, the SINR can be calculated by:

$$\rho_{IN}[dB] = \begin{cases} -0.0673 \times {R_b}^6 + 1.5397 \times {R_b}^5 - 14.3404 \times {R_b}^4 + 69.4089 \times {R_b}^3 \\ -184.0043 \times {R_b}^2 + 255.3831 \times R_b - 154.7503, \quad 1.7 \leq R_{b\,[Mbps]} < 3.5 \\ -0.0202 \times {R_b}^4 + 0.5189 \times {R_b}^3 \\ -4.7933 \times {R_b}^2 + 20.2255 \times R_b - 37.2841, \quad 3.5 \leq R_{b[Mbps]} < 6.4 \\ -0.0202 \times {R_b}^4 + 0.5189 \times {R_b}^3 \\ -0.0579 \times {R_b}^2 + 2.1091 \times R_b - 14.0231, \quad 6.4 \leq R_{b[Mbps]} < 7.0 \\ -0.0817 \times {R_b}^2 + 2.4592 \times R_b - 13.2108, \quad 7.0 \leq R_{b[Mbps]} < 7.8 \\ -0.0933 \times {R_b}^3 + 2.5064 \times {R_b}^2 - 21.18938 \times R_b + 57.9987, \quad 7.8 \leq R_{b[Mbps]} < 9.5 \\ 0.8613 \times R_b - 5.1806, \quad 9.5 \leq R_{b[Mbps]} < 14.1 \\ -0.0042 \times {R_b}^2 + 0.7262 \times R_b - 2.4267, \quad 14.1 \leq R_{b[Mbps]} < 34.5 \\ 0.0482 \times {R_b}^2 - 2.879 \times R_b + 60.0064, \quad 34.5 \leq R_{b[Mbps]} < 42.5 \\ 0.2984 \times {R_b}^2 - 21.9131 \times R_b + 417.3976, \quad 42.5 \leq R_{b[Mbps]} \leq 43.2 \end{cases} \tag{B.9}$$

For QPSK, in UL direction, one has:

$$(E_c/N_0)[dB] = \begin{cases} -3.33 \times R_b - 10.0, \quad 0 \leq R_{b\,[Mbps]} < 1.5 \\ -0.5998 \times {R_b}^2 + 5.0194 \times R_b - 11.1447, \quad 1.5 \leq R_{b[Mbps]} < 3.4 \\ -5.2083 \times {R_b}^3 + 62.5 \times {R_b}^2 - 244.7917 \times R_b + 313.5, \quad 3.4 \leq R_{b[Mbps]} < 4.2 \\ -4.3821 \times {R_b}^3 + 65.5602 \times {R_b}^2 \\ -321.7734 \times R_b + 521.6365, \quad 4.2 \leq R_{b[Mbps]} < 5.5 \end{cases} \tag{B.10}$$

Considering 16QAM for UL, the E_c /N_0 is given by (C.8), a modified version of the equation is presented.

$$(E_c/N_0)[dB] = \begin{cases} -1.5432 \times {R_b}^3 + 6.9444 \times {R_b}^2 - 6.9444 \times R_b - 3.0, \quad 0.6 \leq R_{b[Mbps]} < 1.6 \\ 2.0 \times R_b - 6.0, \quad 1.6 \leq R_{b[Mbps]} < 3.8 \\ 0.1307 \times {R_b}^4 - 3.041 \times {R_b}^3 \\ +26.0522 \times {R_b}^2 - 95.8265 \times R_b + 129.0191, \quad 3.8 \leq R_{b[Mbps]} < 7.7 \\ 0.1386 \times {R_b}^3 - 3.5025 \times {R_b}^2 \\ +30.979 \times R_b - 87.2192, \quad 7.7 \leq R_{b[Mbps]} \leq 11.0 \end{cases} \tag{B.11}$$

It is also important to present the inverted expressions, i.e., the throughput as a function of SINR for DL and as a function of E_c/N_0 for UL. Regarding the SISO configuration with QPSK, for DL, the expression is defined by:

$$R_b[Mbps] = \begin{cases} 0.0072 \times {\rho_N}^2 + 0.1743 \times \rho_N + 1.383, & -10 \le \rho_{N[dB]} < -6 \\ 0.025 \times {\rho_N}^2 + 0.425 \times \rho_N + 2.25, & -6 \le \rho_{N[dB]} < -1 \\ 0.00779 \times {\rho_N}^2 + 0.368 \times \rho_N + 2.193, & -1 \le \rho_{N[dB]} < 10 \\ -0.0148 \times {\rho_N}^2 + 0.499 \times \rho_N + 3.0, & 10 \le \rho_{N[dB]} < 16 \\ 7.2, & \rho_{N[dB]} \ge 16 \end{cases} \tag{B.12}$$

For a SISO configuration with 16QAM, one has:

$$R_b[Mbps] = \begin{cases} 0.0143 \times {\rho_N}^2 + 0.3486 \times \rho_N + 2.7657, & -10 \le \rho_{N[dB]} < -6 \\ 0.05 \times {\rho_N}^2 + 0.85 \times \rho_N + 4.5, & -6 \le \rho_{N[dB]} < -1 \\ 0.02239 \times {\rho_N}^2 + 0.631 \times \rho_N + 4.3203, & -1 \le \rho_{N[dB]} < 10 \\ -0.05 \times {\rho_N}^2 + 1.57579 \times \rho_N + 1.9286, & 10 \le \rho_{N[dB]} < 16 \\ 14.4, & \rho_{N[dB]} \ge 16 \end{cases} \tag{B.13}$$

For a SISO configuration with 64QAM, the throughput is given by:

$$R_b[Mbps] = \begin{cases} 0.0143 \times {\rho_N}^2 + 0.3586 \times \rho_N + 2.7657, & -10 \le \rho_{N[dB]} < -6 \\ 0.0005 \times {\rho_N}^3 + 0.0208 \times {\rho_N}^2 + 0.6167 \times \rho_N + 4.3131, & -6 \le \rho_{N[dB]} < 11 \\ -0.0652 \times {\rho_N}^2 + 2.879 \times \rho_N - 9.7048, & 11 \le \rho_{N[dB]} < 20 \\ 21.6, & \rho_{N[dB]} \ge 20 \end{cases} \tag{B.14}$$

In a SIMO 1x 2 schemes with QPSK, the expression is defined by:

$$R_b[Mbps] = \begin{cases} 0.017 \times {\rho_N}^2 + 0.416 \times \rho_N + 2.973, & -10 \le \rho_{N[dB]} < 3 \\ -0.0404 \times {\rho_N}^2 + 0.9318 \times \rho_N + 1.824, & 3 \le \rho_{N[dB]} < 13 \\ 7.2, & \rho_{N[dB]} \ge 13 \end{cases} \tag{B.15}$$

For a SIMO 1×2 configuration with 16QAM, the throughput can be computed by:

$$R_b[Mbps] = \begin{cases} 0.03 \times {\rho_N}^2 + 0.7823 \times \rho_N + 5.8266, & -10 \le \rho_{N[dB]} < 3 \\ -0.0626 \times {\rho_N}^2 + 1.6255 \times \rho_N + 3.813, & 3 \le \rho_{N[dB]} < 13 \\ 14.4, & \rho_{N[dB]} \ge 13 \end{cases} \tag{B.16}$$

Considering a SIMO 1×2 configuration with 64QAM, one has:

$$R_b[Mbps] = \begin{cases} 0.0255 \times {\rho_N}^2 + 0.7265 \times \rho_N + 5.6914, & -10 \le \rho_{N[dB]} < -1 \\ 0.0105 \times {\rho_N}^2 + 0.8517 \times \rho_N + 5.783, & -1 \le \rho_{N[dB]} < 13 \\ -0.0542 \times {\rho_N}^2 + 2.2154 \times \rho_N - 0.9696, & 13 \le \rho_{N[dB]} < 19 \\ 21.6, & \rho_{N[dB]} \ge 19 \end{cases} \tag{B.17}$$

For a MIMO 2×2 scheme with QPSK, the throughput is defined by:

$$R_b[Mbps] = \begin{cases} -0.00695 \times {\rho_N}^3 - 0.1357 \times {\rho_N}^2 - 0.6502 \times \rho_N + 0.9762, & -10 \le \rho_{N[dB]} < -5 \\ -0.00042 \times {\rho_N}^3 + 0.0151 \times {\rho_N}^2 + 0.435 \times \rho_N + 3.592, & -5 \le \rho_{N[dB]} < 10 \\ -0.0523 \times {\rho_N}^2 + 2.009 \times \rho_N - 4.904, & 10 \le \rho_{N[dB]} < 20 \\ 14.4, & \rho_{N[dB]} \ge 20 \end{cases} \tag{B.18}$$

For a MIMO 2×2 configuration with 16QAM, one has:

$$R_b[Mbps] = \begin{cases} -0.0139 \times {\rho_N}^3 - 0.2714 \times {\rho_N}^2 - 1.3004 \times \rho_N + 1.95242, & -10 \le \rho_{N[dB]} < -5 \\ 0.0021 \times {\rho_N}^3 + 0.0209 \times {\rho_N}^2 + 0.7905 \times \rho_N + 7.0537, & -5 \le \rho_{N[dB]} < 10 \\ -0.0722 \times {\rho_N}^2 + 3.1463 \times \rho_N - 5.2526, & 10 \le \rho_{N[dB]} < 20 \\ 28.8, & \rho_{N[dB]} \ge 20 \end{cases} \tag{B.19}$$

In a MIMO 2×2 configuration with 64QAM, the physical throughput is given by:

$$R_b[Mbps] = \begin{cases} -0.0083 \times {\rho_N}^3 - 0.1357 \times {\rho_N}^2 - 0.2131 \times \rho_N + 4.8057, & -10 \le \rho_{N[dB]} < -6 \\ 0.0005 \times {\rho_N}^4 + 0.0018 \times {\rho_N}^3 \\ +0.0089 \times {\rho_N}^2 + 0.7812 \times \rho_N + 7.0784, & -6 \le \rho_{N[dB]} < 1 \\ -0.001 \times {\rho_N}^3 + 0.0657 \times {\rho_N}^2 + 0.5792 \times \rho_N + 7.211, & 1 \le \rho_{N[dB]} < 4 \\ 0.008 \times {\rho_N}^3 - 0.0593 \times {\rho_N}^2 + 0.8046 \times \rho_N + 6.0472, & 4 \le \rho_{N[dB]} < 13.5 \\ -0.0757 \times {\rho_N}^2 + 4.3661 \times \rho_N - 19.392, & 13.5 \le \rho_{N[dB]} < 26.5 \\ 43.2, & \rho_{N[dB]} \ge 26.5 \end{cases} \tag{B.20}$$

For UL, for QPSK configuration, the physical throughput as a function of E_c/N_0 is given by:

$$R_b[Mbps] = \begin{cases} 0.0643 \times (E_c/N_0)^2 + 0.8557 \times (E_c/N_0) + 4.18, & -5 \le (E_c/N_0)_{[dB]} < -1 \\ -0.05 \times (E_c/N_0)^2 + 0.31 \times (E_c/N_0) + 3.77, & -1 \le (E_c/N_0)_{[dB]} < 2 \\ 0.0417 \times (E_c/N_0)^3 - 0.5429 \times (E_c/N_0)^2 \\ +2.5012 \times (E_c/N_0) + 1.04, & 2 \le (E_c/N_0)_{[dB]} < 6 \\ 5.5, & (E_c/N_0)_{[dB]} \ge 6 \end{cases} \tag{B.21}$$

For UL, for 16QAM configuration, the physical throughput can be computed by:

$$R_b[Mbps] = \begin{cases} -0.0087 \times (E_c/N_0)^4 - 0.0669 \times (E_c/N_0)^3 - 0.0936 \times (E_c/N_0)^2 \\ +0.6056 \times (E_c/N_0) + 3.0522, & -5 \le (E_c/N_0)_{[dB]} < -3 \\ 0.0333 \times (E_c/N_0)^4 + 0.1 \times (E_c/N_0)^3 - 0.0333 \times (E_c/N_0)^2 \\ +0.4 \times (E_c/N_0) + 3.0, & -5 \le (E_c/N_0)_{[dB]} < 1 \\ 0.0583 \times (E_c/N_0)^3 - 0.575 \times (E_c/N_0)^2 \\ +2.3667 \times (E_c/N_0) + 1.66, & 1 \le (E_c/N_0)_{[dB]} < 5 \\ -0.0003 \times (E_c/N_0)^3 - 0.0195 \times (E_c/N_0)^2 \\ +0.9558 \times (E_c/N_0) + 2.1899, & 5 \le (E_c/N_0)_{[dB]} < 14 \\ 11.0, & (E_c/N_0)_{[dB]} \ge 14 \end{cases} \tag{B.22}$$

Annex C

HSDPA available codes variation with network load

The present Annex, presents the throughput variation with the number of other users (network load) for a fixed bit rate of 2Mbps.

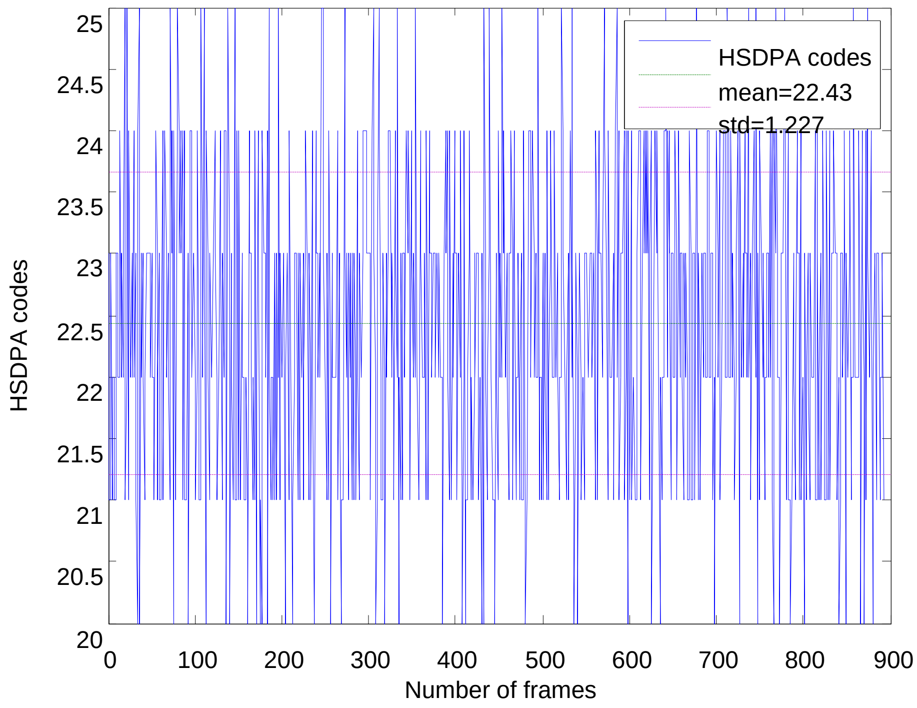


Figure C.1 - HSDPA available codes for 10 MB File download - Network = 0 user´s.

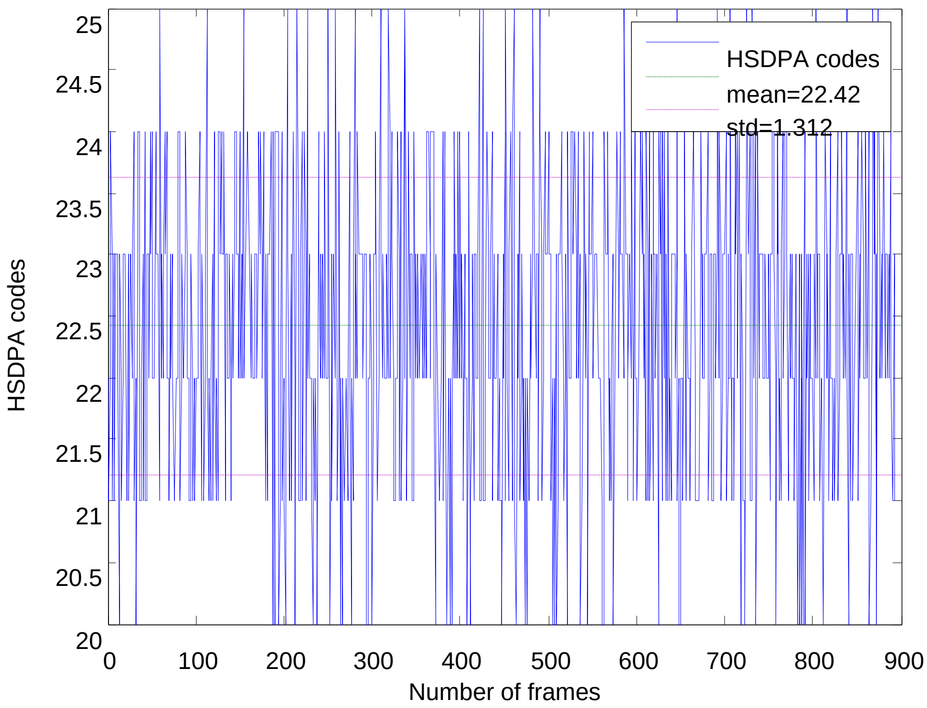


Figure C.2 - HSDPA available codes for 10 MB File download - Network = 5 user´s.

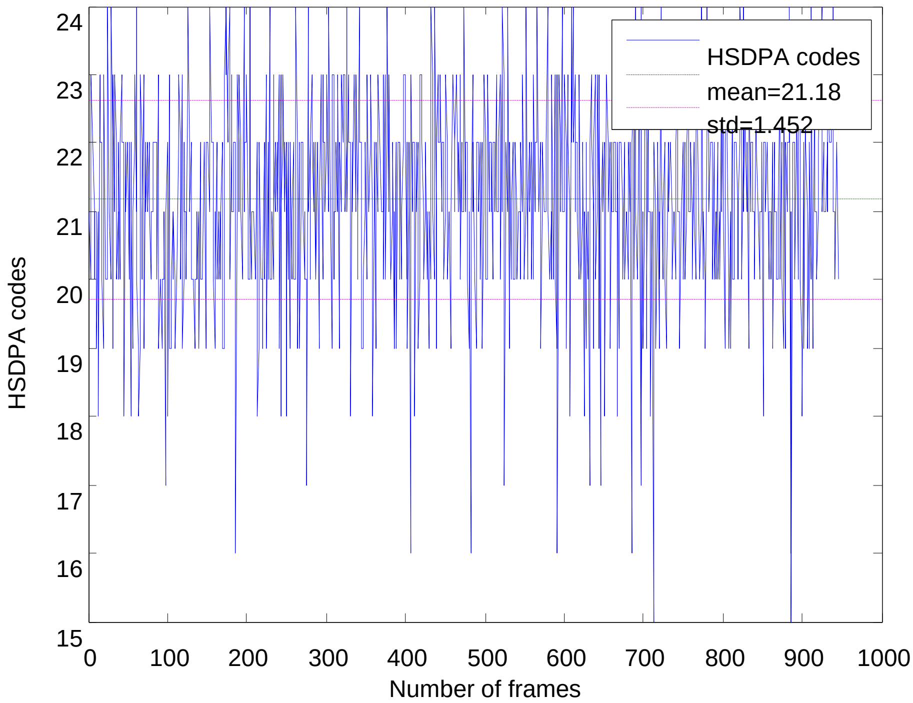


Figure C.3 – HSDPA available codes for 10 MB File download - Network = 10 user´s.

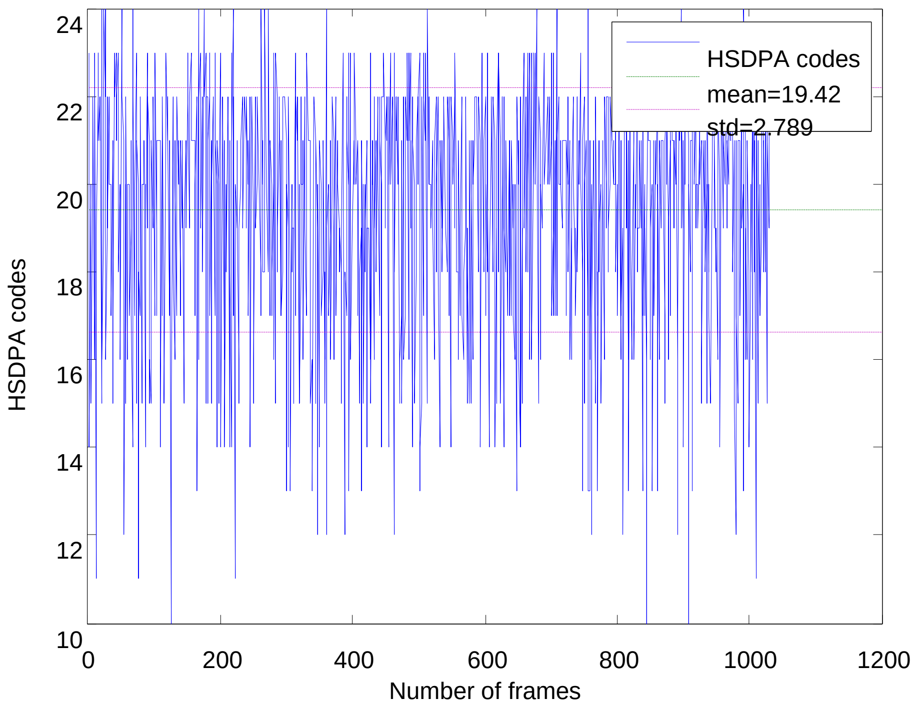


Figure C.4 – HSDPA available codes for 10 MB File download - Network = 15 user´s.

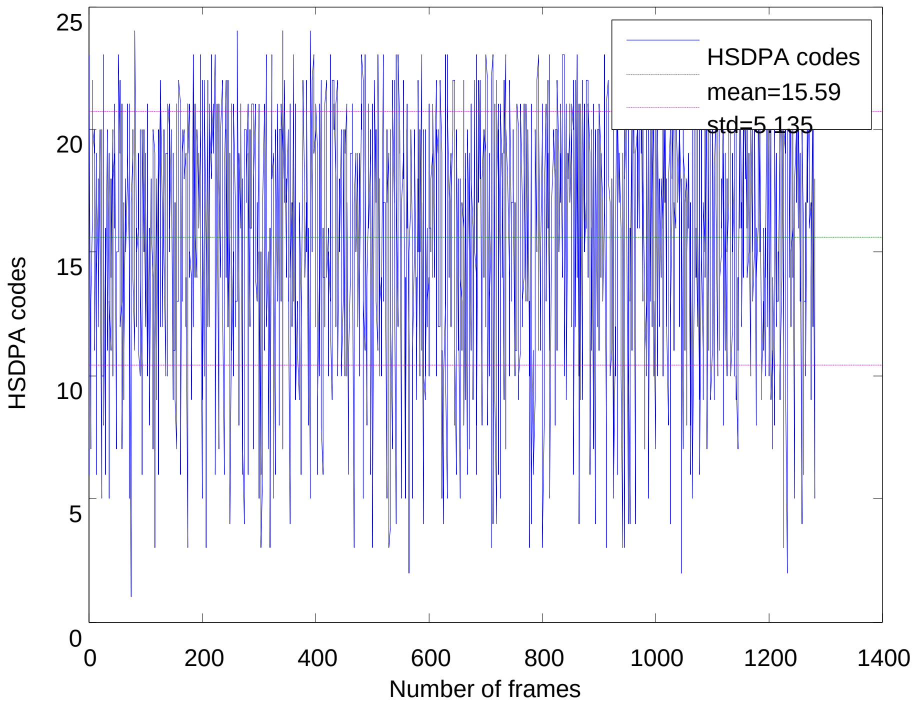


Figure C.5 – HSDPA available codes for 10 MB File download - Network = 20 user´s.

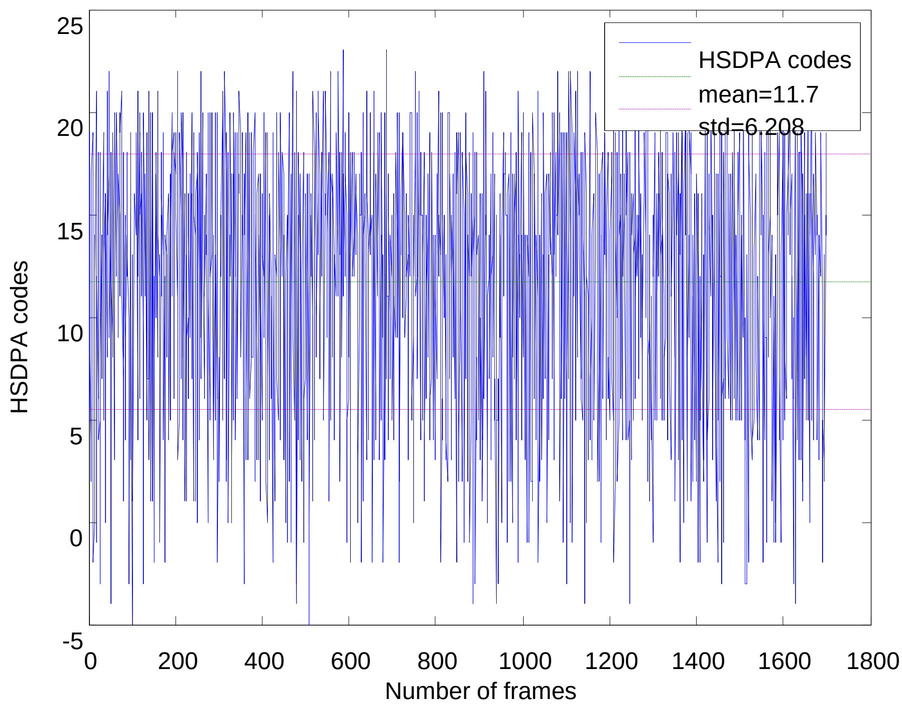


Figure C.6 – HSDPA available codes for 10 MB File download - Network = 25 user´s.

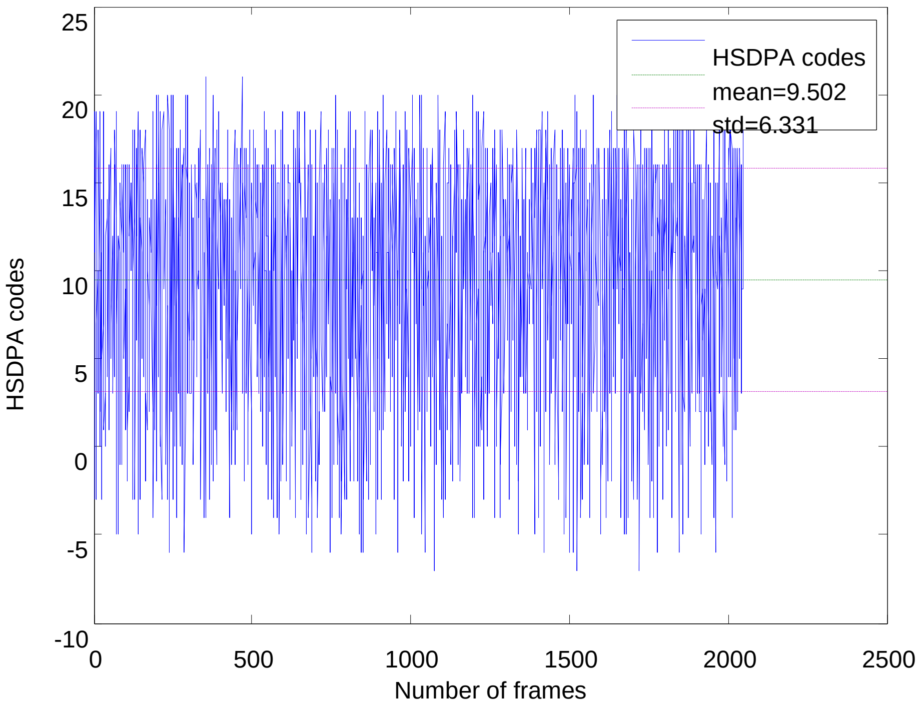


Figure C.7 – HSDPA available codes for 10 MB File download - Network = 30 user´s.

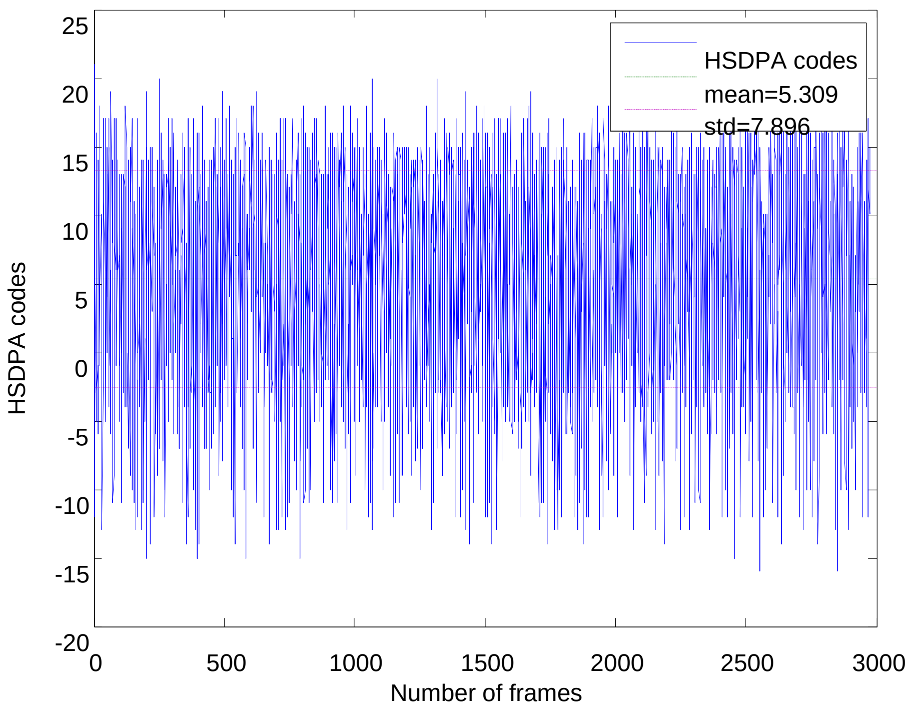


Figure C.8 – HSDPA available codes for 10 MB File download - Network = 35 user´s.

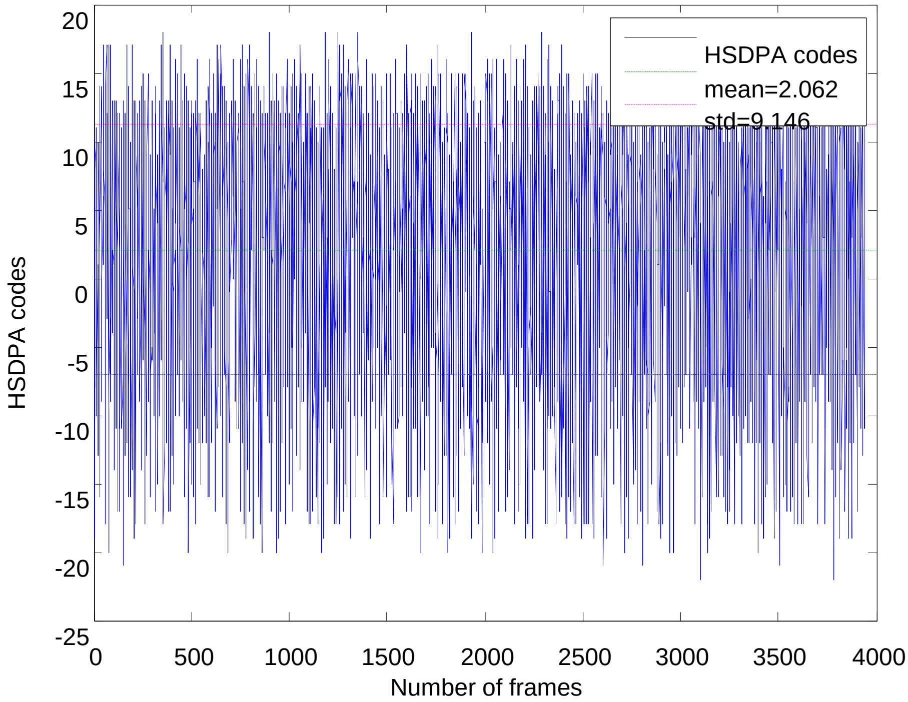


Figure C.9 – HSDPA available codes for 10 MB File download - Network = 40 user´s.

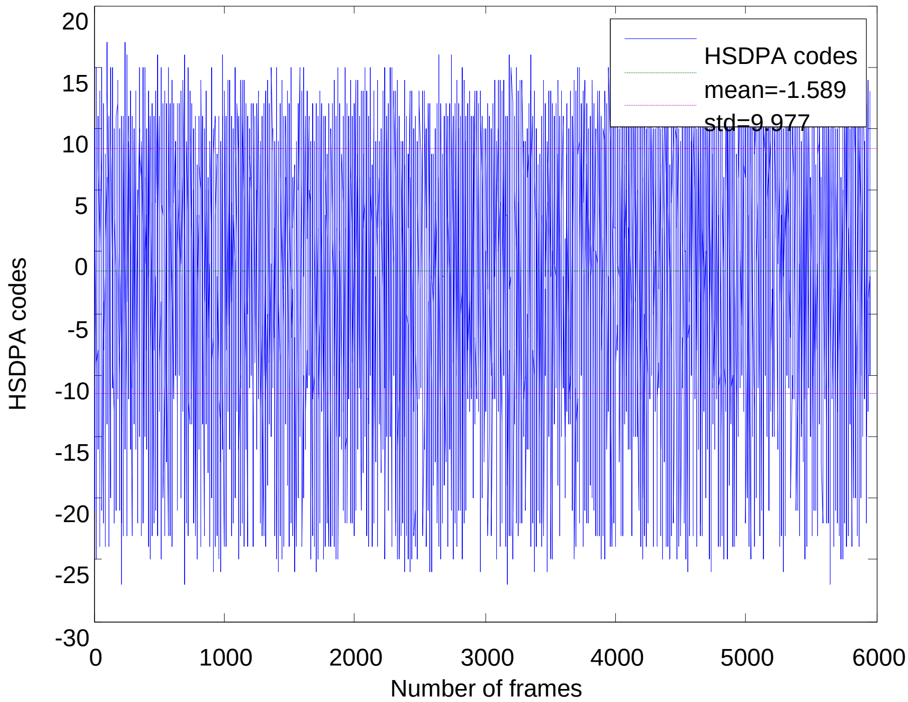


Figure C.10 – HSDPA available codes for 10 MB File download - Network = 45 user´s.

Annex D

Simulator assessment

The present Annex presents the assessment of coherence time with the environment, convergence potential of the mean bit rate and mean delay functions.

Figure D.1, Figure D.2 and Figure D.3 are the result for the coherence time assessment.

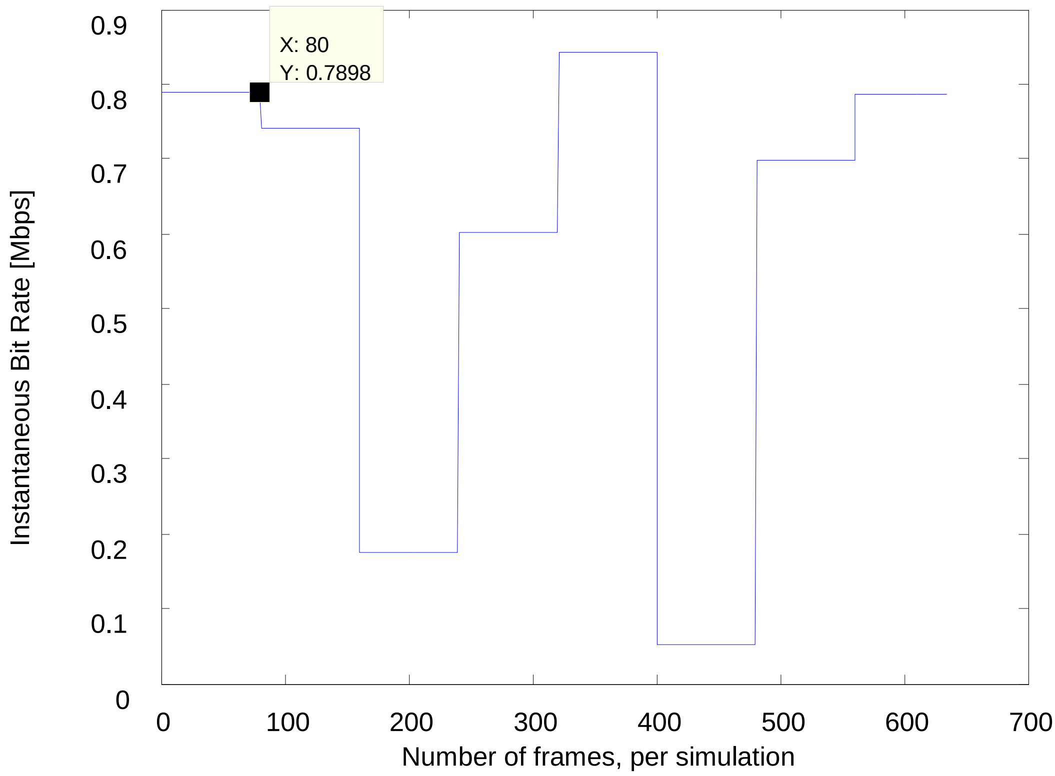


Figure D.1 - Bit Rate at each frame of 2ms, Tc=160 [ms] – Urban environment.

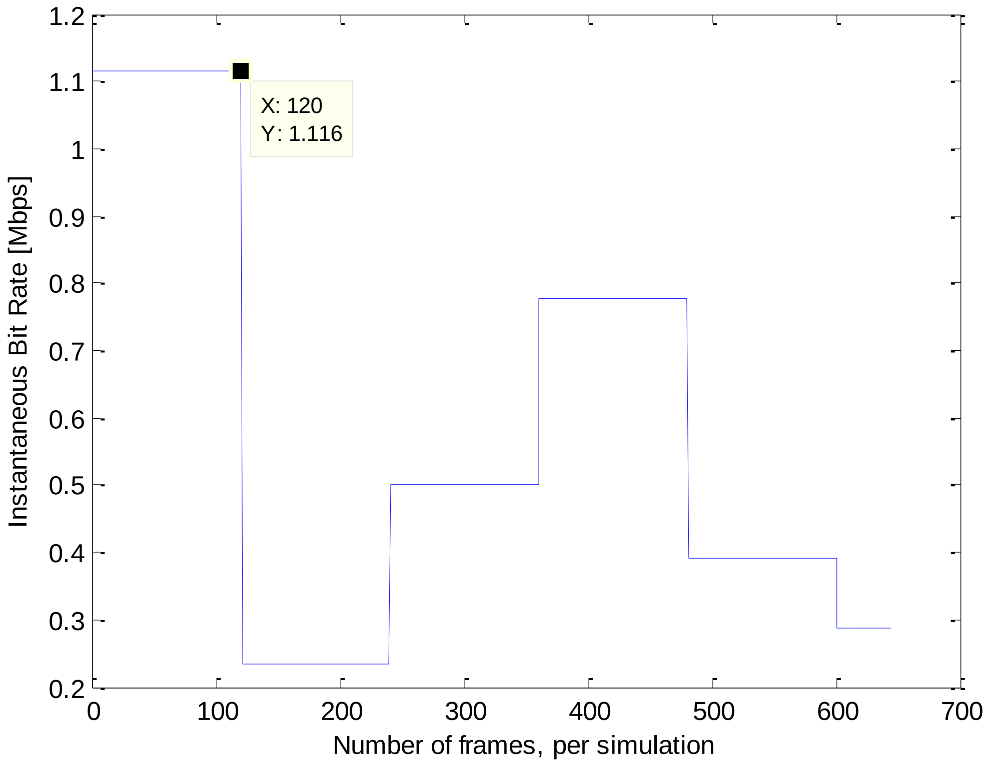


Figure D.2 - Bit Rate at each frame of 2ms, Tc=240 [ms] – Suburban environment.

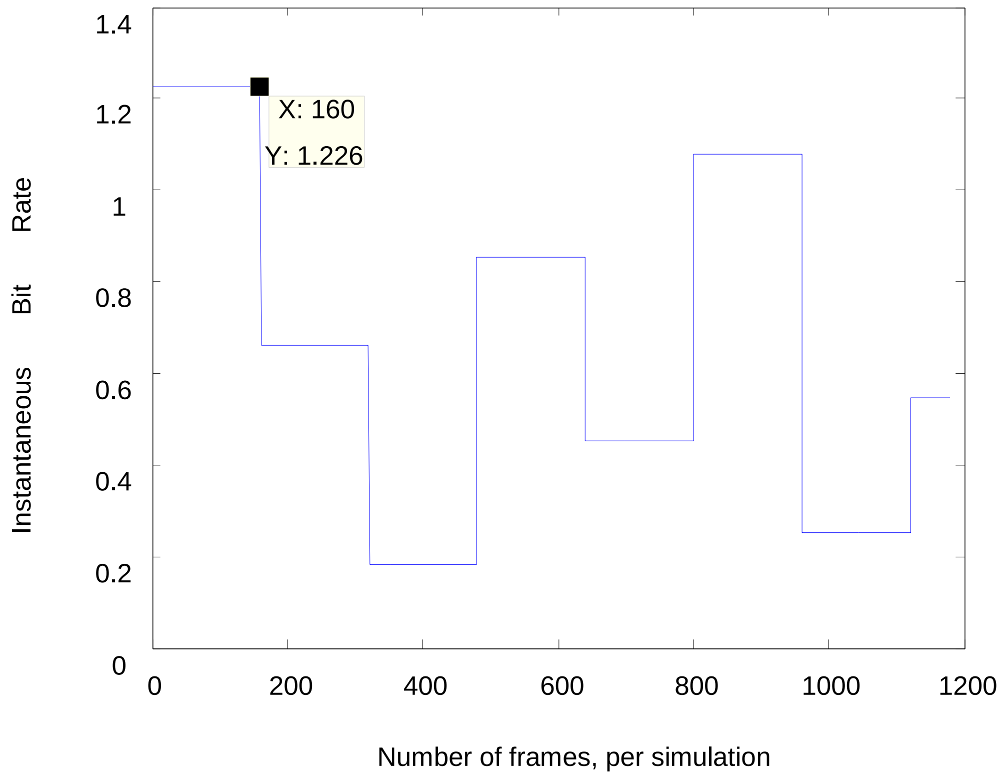


Figure D.3 - Bit Rate at each frame of 2ms, Tc=320 [ms] – Rural environment.

Figure D.4, Figure D.5 and Figure D.6 present for the same environment, the relationship between, SNR, and average throughput for indoor and outdoor environment.

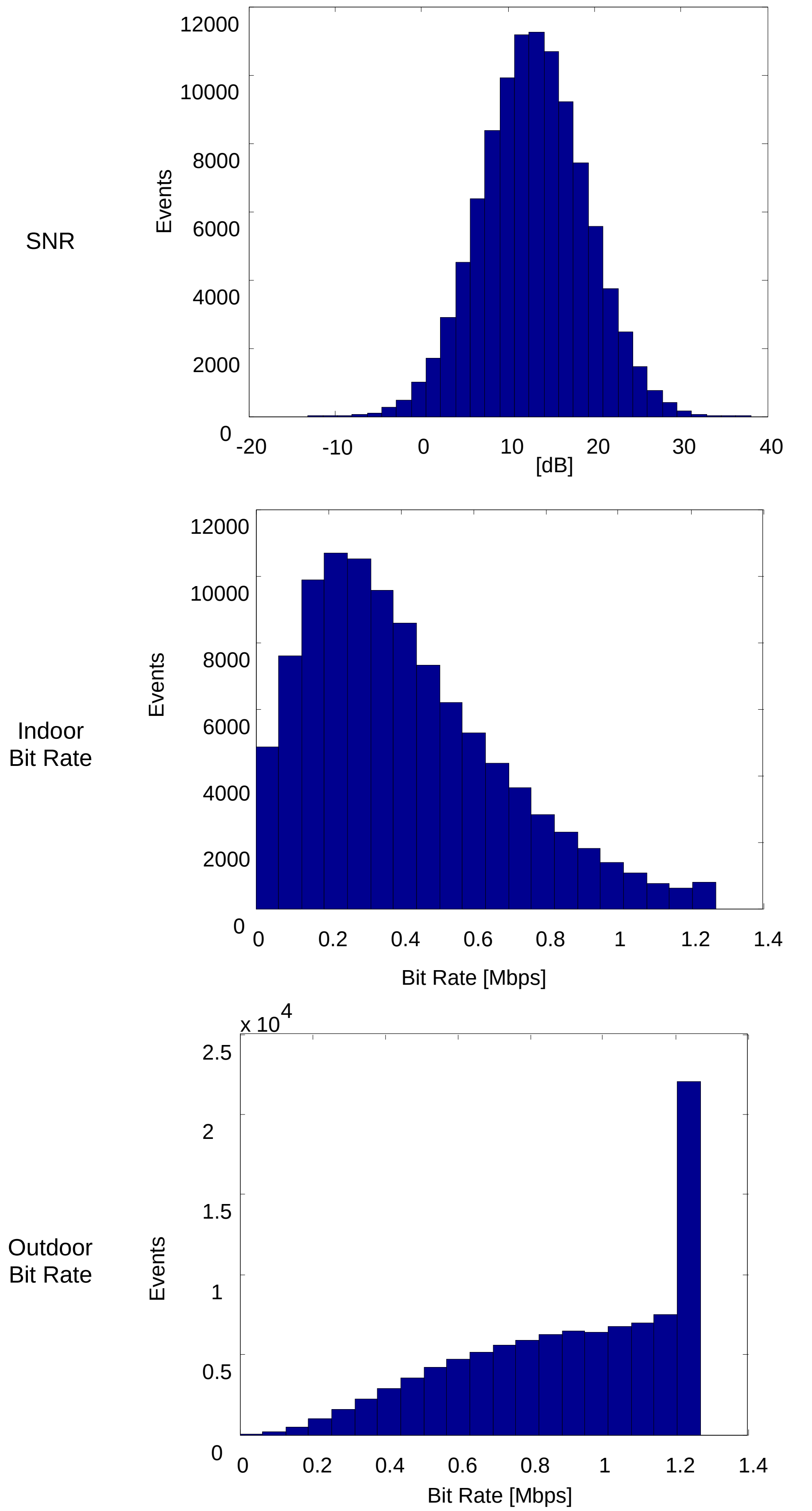


Figure D.4 - SNR, indoor and outdoor Bit Rate variation in rural environment.

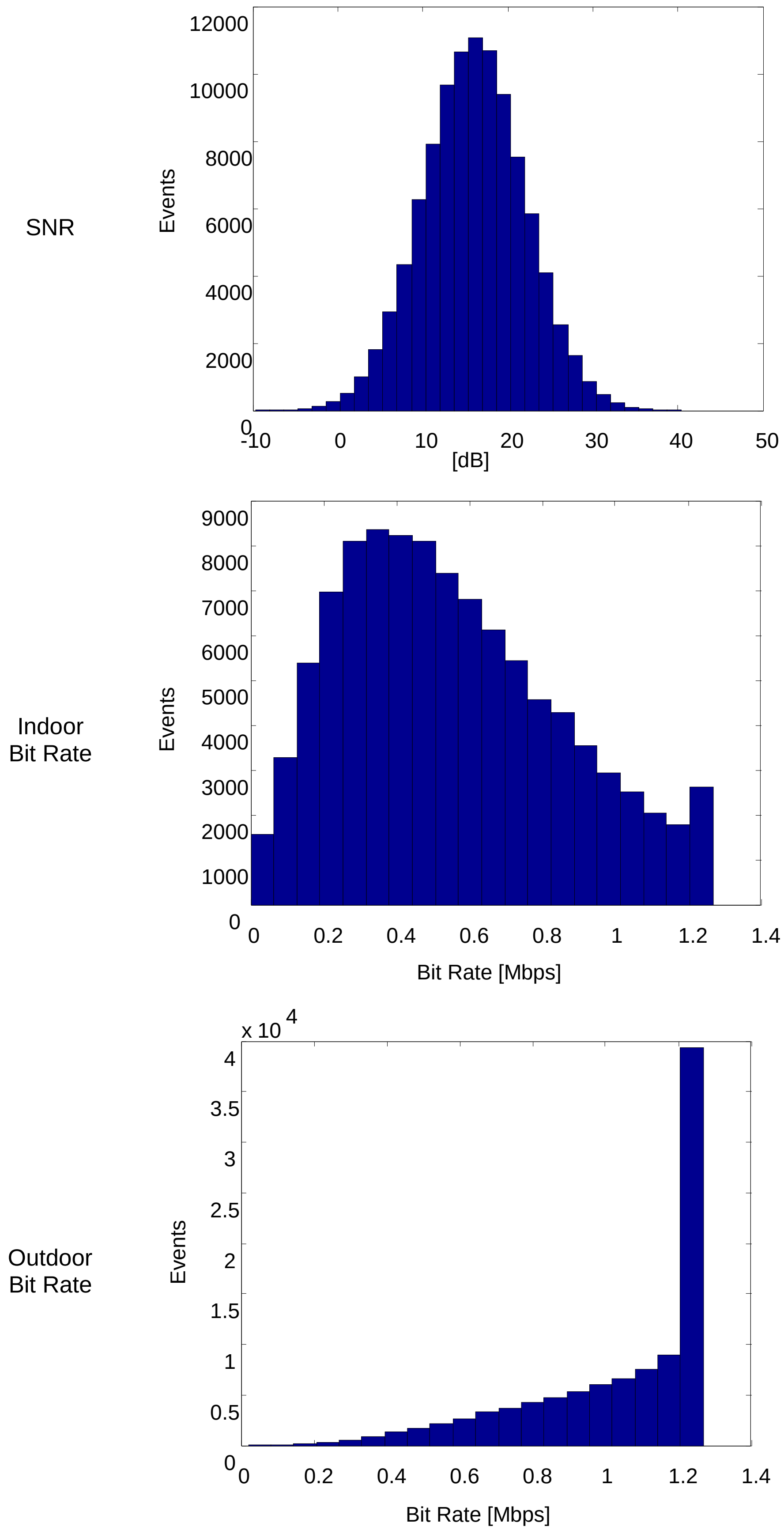


Figure D.5 - SNR, indoor and outdoor Bit Rate variation in suburban environment.

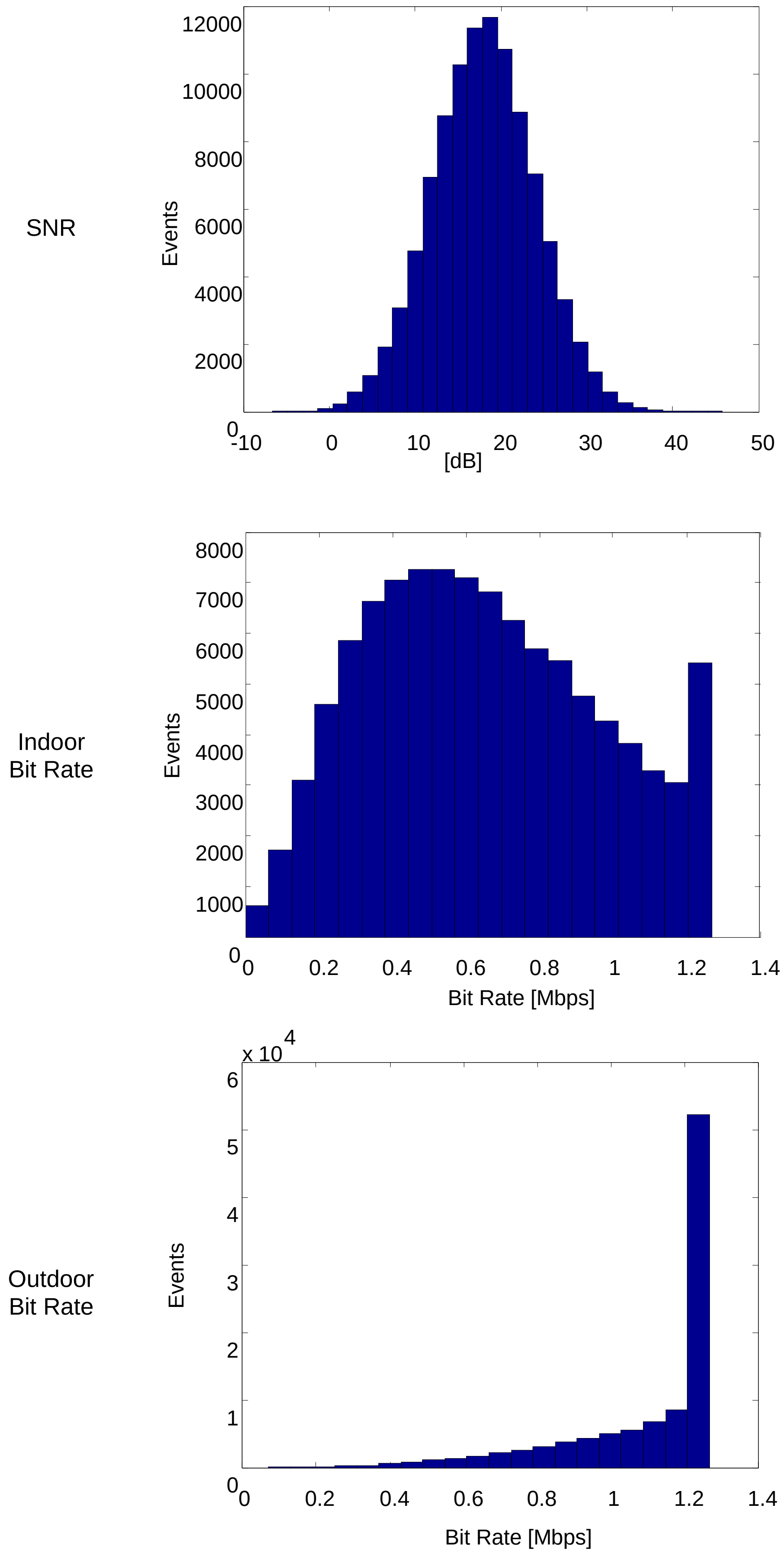


Figure D.6 - SNR, indoor and outdoor Bit Rate variation in urban environment.

Figure D.7, shows the relationship between bit rate and the SNR.

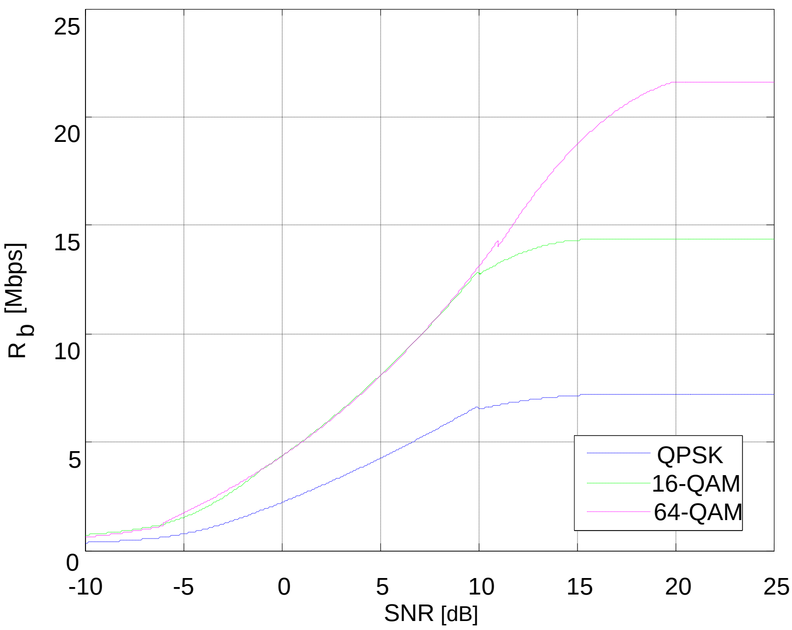


Figure D.7 - Bit rate versus SNR.

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