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Traffic Analysis at the Radio Interface in UMTS FDD

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To my parents

"With ordinary talent and extraordinary perseverance all things are attainable"

Sir Thomas Foxwell Buxton

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ABSTRACT

This work analyses the overall performance at the radio interface of UMTS networks, via a simulation tool. The interference among users in a given cell, the propagation model and the detailed service characterisation enables the analysis of the blocking probability for circuit switched services and delay probability for packet switched ones in a given cell. Eight service types are implemented: Speech, Video Telephony, Streaming Multimedia, Web Browsing, Location Based Services, MMS, Email and File Download. The traffic source models, call/session generation and duration process for each service have a statistical approach. For traffic analysis purposes, three implementation scenarios, have been considered, Business, SOHO and Mass-Market, corresponding to a different service characterisation and utilisation. Two services implementation are considered, one where all the services are available and used and another where only four services are required (Speech, Location Based Services, MMS and Email). For each scenario, the population density is varied, resulting in a blocking probability increase. The most limiting scenario corresponds to the Business one, with all eight services available; the Mass-Market scenario with only four services active corresponds to the less limiting one. It is verified that in a single cell scenario with all services available, the Business and the SOHO scenarios are more adapted for micro-cell environments, a population density of 300 pop/km^2 and 900 pop/km^2 being achieved. For Mass-Market a maximum of 1200 pop/km^2 is obtained. Considering only four services available a higher population density is obtained 750, 1000 and 10000 pop/km^2 for Business, SOHO and Mass-Market scenarios respectively.

KEYWORDS

UMTS, Radio Interface, Traffic, Multi-Services, Simulation.

RESUMO

Este trabalho analisa a desempenho global de uma rede UMTS considerando a interface rádio, através de um simulador. A interferência entre os utilizadores numa dada célula, o modelo de propagação e uma pormenorizada caracterização dos serviços permite a análise da probabilidade bloqueio para comutação de circuitos e a probabilidade de atraso para comutação de pacotes numa dada célula. Foram implementados oitos tipos de serviços: Voz, Vídeo Telefonia, *Streaming* Multimedia, Navegação na *Internet*, Serviços de Localização, MMS, Correio Electrónico e *Download* de Ficheiros. Os modelos das fontes de tráfego, a geração das chamadas/sessões e a respectiva duração segue para cada serviço uma distribuição estatística específica. Para a análise do tráfego, foram considerados três cenários de implementação, *Business*, SOHO e Mass-Market, correspondendo cada um deles a diferentes caracterizações dos serviços e a diferentes percentagens de utilização. São consideradas duas implementações dos serviços, uma em que todos os serviços estão disponíveis e outra em que apenas quatro são requeridos (Voz, Serviços de Localização, MMS e Correio Electrónico). Para cada cenário, a densidade de utilizados é variada resultando num aumento da probabilidade de bloqueio. O cenário mais limitativo corresponde ao *Business* com todos os serviços disponíveis e o cenário mais favorável ao *Mass-Market* com apenas quatro serviços activos. Foi verificado que considerando o cenário com apenas uma célula em que todos os serviços estão activos, os cenários *Business* e SOHO estão mais adaptados a ambientes tipicamente micro-celulares tendo sido obtidas densidades de utilizadores de 300 pop/km^2 e 900 pop/km^2 respectivamente. Para o cenário *Mass-Market* foi obtida uma densidade de 1200 pop/km^2. Considerando apenas quatro serviços disponíveis foram obtidas densidades de utilizadores superiores de 750, 1000 e 10000 pop/km^2 *Business*, SOHO e *Mass-Market* respectivamente.

PALAVRAS-CHAVE

UMTS, Interface Radio, Tráfego, Multi Serviços, Simulação

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LIST OF ACRONYMS

3GPP	3^{rd} Generation Partnership Project
AICH	Acquisition Indication Channel
AP-AICH	Access Preamble Acquisition Channel
BCH	Broadcast Channel
BER	Bit Error Rate
BHCA	Busy Hour Call Attempt
BLER	Block Error Rate
BS	Base Station
CA-ICH	Channel Assignment Indication Channel
CBD	Central Business District
CD	Collision Detection
CDF	Cumulative Density Function
CDMA	Code Division Multiple Access
CPCH	Common Packet Channel
CPICH	Common Pilot Channel
CS	Circuit Switch
CSICH	CPCH Status Indication Channel
DAB	Digital Audio Broadcast
DCH	Dedicated Channel
DL	Downlink
DPCCH	Dedicated Physical Control Channel
DPCH	Dedicated Physical Channel
DPDCH	Dedicated Physical Data Channel
DS-CDMA	Direct Sequence Code Division Multiple Access

DSCH	Downlink Shared Channel
DVB	Digital Video Broadcast
ETSI	European Telecommunication Standard Institute
FACH	Forward Access Channel
FDD	Frequency Division Duplex
FER	Frame Erasure Rate
FTP	File Transfer Protocol
GGSN	Gateway GPRS Support Node
GMSC	Gateway MSC
GPRS	General Packet Radio Service
GSM	Global System for Mobile Communications
HLR	Home Location Register
HSDPA	High Speed Data Packet Access
IMS	IP Multimedia Subsystem
IP	Internet Protocol
ITU-T	International Telecommunications Union – Telecommunication sector
LAN	Local Area Network
LCD	Long Constrained Delay
MM	Multimedia
MMS	Multimedia Messaging Service
MPEG	Moving Picture Expert Group
MSC	Mobile Services Switching Centre
MSE	Mean Square Error
MT	Mobile Terminal
NF	Noise Figure
NGN	Next Generation Networks
NRT	Non Real Time

NTB	Non Time Based
OVSF	Orthogonal Variable Spreading Factor
P-CCPCH	Primary Common Control Physical Channel
PCH	Paging Channel
PDF	Probability Density Function
PDSCH	Physical Downlink Shared Channel
PICH	Paging Indication Channel
PNBSCH	Physical Node B Synchronisation Channel
PRACH	Physical Random Access Channel
PS	Packet Switched
PUSCH	Physical Uplink Shared Channel
QPSK	Quaternary Phase Shift Keying
RACH	Random Access Channel
RG	Random Generator
RNC	Radio Network Controller
RT	Real Time
S-CCPCH	Secondary Common Control Physical Channel
SCH	Synchronization Channel
SDU	Service Data Unit
SF	Spreading Factor
SGSN	Serving GPRS Suport Node
SIR	Signal-to-Interference Ratio
SMS	Short Message Service
SNR	Signal-to-Noise Ratio
SOHO	Small Office Home Office
TB	Time Based
TD/CDMA	Time Division/Code Division Multiple Access

TDD	Time Division Duplex
TDMA	Time Division Multiple Access
TFCI	Transport Format Combination Indicator
TFI	Transport Format Indicator
TS	Time Slot
UE	User Equipment
UL	Uplink
UMTS	Universal Mobile Telecommunication System
USCH	Uplink Shared Channel
USIM	UMTS Subscriber Identity Module
UTRA	UMTS Terrestrial Radio Access
UTRAN	UMTS Terrestrial Radio Access Network
VLR	Visitor Location Register
VoIP	Voice over IP
WCDMA	Wideband CDMA
WLAN	Wireless Local Area Network
WWW	World Wide Web

LIST OF SYMBOLS

$\overline{\alpha}$	Average orthogonality factor in the cell
α_j	Orthogonality of channel of user *j*
Φ	Incidence angle
η_{DL}	Downlink load factor
η_{UL}	Uplink load factor
λ	Arrival rate
μ	Mean
ν_j	Activity factor of user *j* at physical level
σ	Standard deviation
τ	Average call duration
d	Distance
E_b	Signal energy per bit
f	Frequency
G_p	Processing gain
h_{base}	Node B height
h_{mobile}	Mobile height
h_{roof}	Building height
i	Other cell to own cell interference ratio seen by the base station receiver
$\bar{i}$	Average ratio of the other cell to own cell base station power received by user
i_j	Ratio of other cell to own cell base station power, received by the user *j*
k	Boltzmann constant
k_a	Increase in path loss floor the base station antennas below the roof tops
k_d	Dependence of multiscreen diffraction loss versus distance

k_f	Dependence of multiscreen diffraction loss versus radio frequency
L_0	Free space loss
L_C	Cable losses
L_{msd}	"multi-screen loss"
L_p	Propagation loss
L_{rst}	"roof-to-to-street deffraction and scater loss"
L_u	Losses due to the presence of the user
M_I	Interference Margin
N	Number of users per cell
N_C	Number of channels per cell
NF	Mobile station receiver noise figure
N_O	Noise spectral density
N_{rf}	Spectral noise density of the mobile receiver front-end
N_T	Total noise plus interference
P_r	Power at the antenna terminals
P_{rx}	Receiver input power
R_b	Service bit rate
R_j	Bit rate of user j
T	Temperature
t	Time
W	Chip rate
w	Building separation
w_b	Building separation plus buildings width

1 INTRODUCTION

More than a telecommunication system, UMTS will play an important role in the availability of innovative services, and although there are many different classifications of service classes that are based on different approaches, they are complementary. UMTS offers business users and consumers an evolution to their current mobile experience adding video and exciting new services. Data services like Short Message Service (SMS), Multimedia Messaging Service (MMS), downloadable ring-tones, images and games, news and information sources, mobile chats and Internet-style portals are already a reality in driving operators new revenues. The service evolution, like photo messaging, as part of multimedia messaging service, and also including e-postcards, audio clips, and logos will absolutely hit the market, and the video service messaging will evolve from the transmission of still pictures. The information services will include timetables, location information, local guides, news reports, movie/theatre guides and sport results (including video clips), music and games download and multimedia broadcast. The M-commerce, including payment and purchase, will be also a driving factor on the UMTS evolution.

The UMTS key success characteristics in shaping future mobile services correspond to mobility, interactivity, convenience, any time/anywhere, ubiquity, easy access, immediacy, personalisation and multimedia.

The coexistence of two operation modes in UMTS, Time Division Duplex (TDD) and Frequency Division Duplex (FDD), will give flexibility to the system to handle both symmetric and asymmetric traffics, with a wide area coverage and high-density traffic. FDD is more efficient for the transmission of symmetric traffic in wide areas coverage, while TDD is more adequate in the transmission of asymmetric traffic due to the high flexibility in the bandwidth allocation.

The evolution beyond 3G is already being considered, and WCDMA will support High Speed Data Packet Access (HSDPA), enabling the transmission at rates up to 14.2 Mbit/s.

Another enhancement is the IP Multimedia Subsystem (IMS), which enables real-time, person-to-person services, such as voice or video telephony, to be provided by means of packet switched technology simultaneously with non-real-time information and data services. Further advantages come from the ability to interwork with other networks such as HIPERLAN, Digital Video Broadcast (DVB) and Digital Audio Broadcast (DAB), and to take advantages of the contents that can be efficiently delivered to small terminals.

The current thesis is motivated by the analysis of the performance of UMTS network with a diversity of services and applications that have different requirements in terms of network resources and quality of service. The main objective of this work is to study the capacity of a single UMTS cell, considering a diversity of service, different service penetration and distinct resource needs. The network aspects related with the UMTS capacity are complex, and include a large range of variables that are included in the network link budget calculations, propagation model estimation, and also in the interference analysis.

This thesis was developed based on the results obtained in [1] which considers the network performance characterisation in terms of link budget estimation, propagation model implementation, the objective being the cell radius optimisation. In this thesis a different approach is considered, a fix cell radius being implemented, and the analysis of the network performance being done by varying the population density, the multi services characterisation and implementation correspond to the main focus of the current thesis.

Service characterisation corresponds nowadays to an important research topic, considering that few models are available, as a result of the limited knowledge of the users behaviour for the new services availability in the different scenarios. The scenario characterisation corresponds also to an important topic, it being fundamental for cell planning in the various phases of UMTS implementation. Much research has been devoted to these subjects, in particular in:

- IST-MOMENTUM [2] – where an important service characterisation in terms of scenario, simulation models and user mobility have been done.
- European Telecommunication Standard Institute (ETSI) [3].
- 3rd Generation Partnership Project (3GPP) [4].
- UMTS Forum [5] – traffic demand and service characterisation.

An accurate radio network planning is naturally associated to very precise service characterisation. This is the reason why in this thesis an extensive analysis of services is presented particularly in terms of:

- Non uniform traffic distribution
- Services traffic models
- Network resources requirement (bit rate, signal-to-noise ratio, sensitivity to delay)
- Service penetration
- Population density
- Percentage of service utilisation

Although planning is confined to only one cell, a fix amount of resources is reserved for intercell interference and soft handover. One important step towards the progression of this work is to analyse the influence of intercell interference and user mobility in the system capacity. The inter-working with other networks, particularly GSM and HIPERLAN, as a Wireless LAN (WLANs), is also an important future development of this work, to give more precise results for the planning of UMTS.

This document is composed of five chapters, besides the current one, and one annex. The following chapter introduces the general principles of UMTS, and describes the main characteristics of the services that are going to be supported. Chapter 3 describes the fundamental traffic aspects in UMTS, particularly radio network planning aspects and traffic model considerations. In Chapter 4, a functional description of the simulator is presented. In Chapter 5, the results of the simulations are presented and analysed considering different user scenarios. In the closing chapter, conclusions and future development suggestions are presented. The Annex presents the statistical models validation.

2 GENERAL PRINCIPLES

2.1 Network Architecture

The terrestrial bands available for UMTS are [1900 – 1980] MHz, [2010 – 2025] MHz, and [2110 – 2170] MHz [6]. A total of 155 MHz is available; the channel bandwidth being 5MHz, which corresponds to 31 channels. The spectrum distribution in use in some countries can be seen in Figure 2.1.

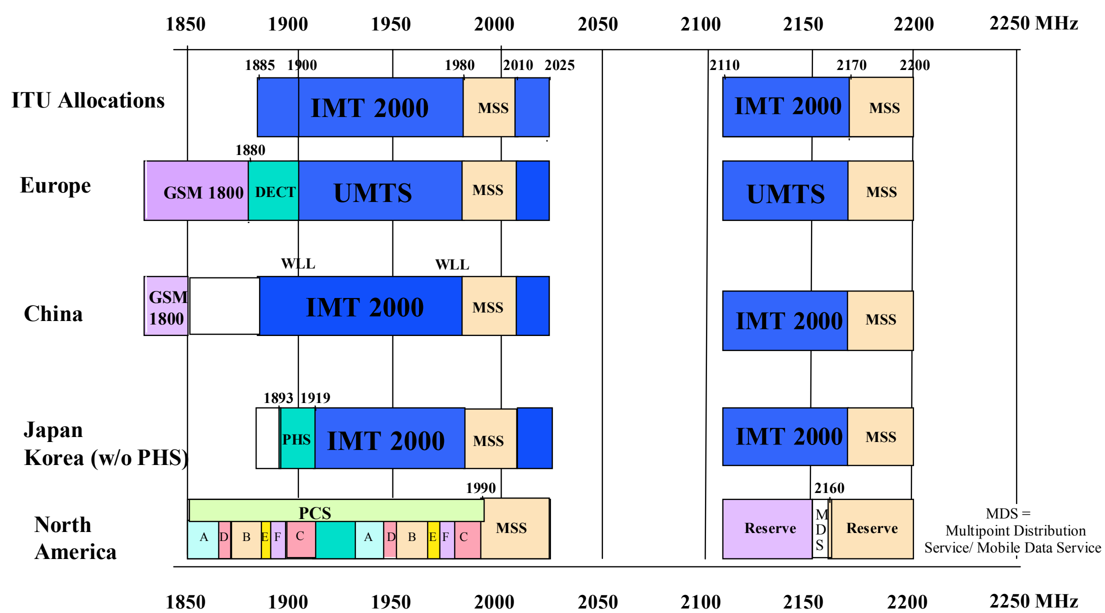

Figure 2.1 - Frequency bands for IMT 2000 (extracted from [6]).

Two operation modes are possible in UMTS, Time Division Duplex (TDD) and Frequency Division Duplex (FDD), the separation between the Upink (UL) and the Downlink (DL), being 190 MHz in the FDD mode. The spectrum allocation for the TDD and FDD modes can be seen in Figure 2.2.

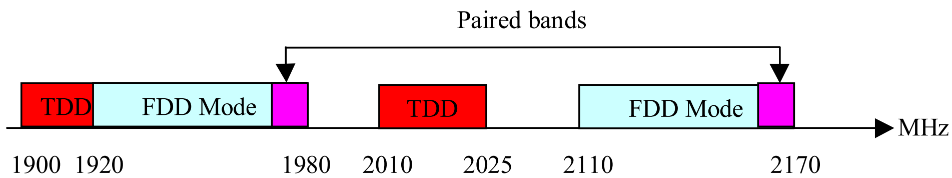

Figure 2.2 – Frequency allocation for the FDD and TDD modes.

UMTS consists of three main components: the user equipment, the radio access network (UTRAN) and the core network. The user equipment is the interface between the subscriber and the radio network, the radio access network handles all the radio-related functionality, and the core network is responsible for switching and routing calls and data connections to external networks. The interfaces between the user equipment and the access network, and between the access network and the core network are Uu and Iu respectively. The user equipment has two main components: the mobile equipment used for radio communication, and the subscriber identity named UMTS Subscriber Identity Module (USIM) that performs authentication algorithms [7].

The main elements of the UTRAN are Node B, the Base Station (BS) that converts the data flow between the Uu and the Iu interfaces, and the Radio Network Controller (RNC), which controls the Node B, Figure 2.3.

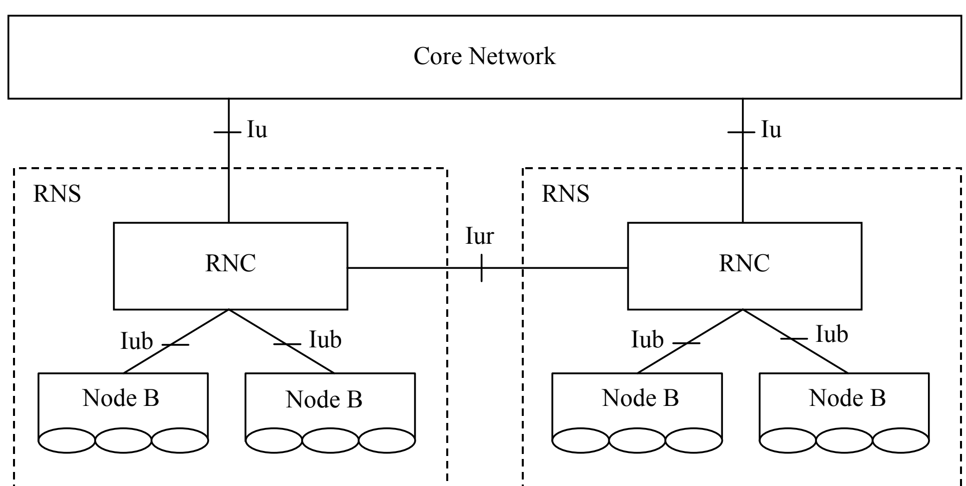

Figure 2.3 – UTRAN Architecture (extracted from [8]).

The main elements of the core network are the Home Location Register (HLR), Mobile Services Switching Centre/Visitor Location Register (MSC/VLR), Gateway MSC (GMSC), Serving GPRS Suport Node (SGSN) and Gateway GPRS Support Node (GGSN) [7]. In the implementation phase, the traffic in the radio network will still be Circuit Switch (CS) based, but it is expected that the evolution to the Next Generation Networks (NGN) will be based in Packet Switch (PS).

UMTS will handle a mixed range of traffic, and the cell architecture being considered is a layout, that will consist of macro-cells overlaid on micro-cells and pico-cells. The cell size, the bit rate and the deployment environment are some of the main characteristics; Table 2.1 shows the characteristic parameters for each type of cell [9].

Table 2.1 – Assumed values for cell classification (extracted from [9]).

Cell Type	Distance [km]	Cell Area [km^2]	Mobility Class	Max. bit rate [kbps]
Macro	1	0.288	high	384
Micro	0.4	0.138	high	384
	0.4	0.138	low	2000
Pico	0.075	0.005	low	2000

Macro-cells provide wide area coverage, and are also used for high-speed mobiles. Micro-cells are used at street level, providing extra capacity where macro-cells are not able to cope with the traffic demand. Pico-cells will be deployed mainly indoors, in areas where there is a demand for high data rate services. Figure 2.4 shows an example of the hierarchical cell structure implementation, F1 and F2 being the frequencies used in macro-cells, f3 dealing with micro-cell layers, and f1 with pico-cells ones [9].

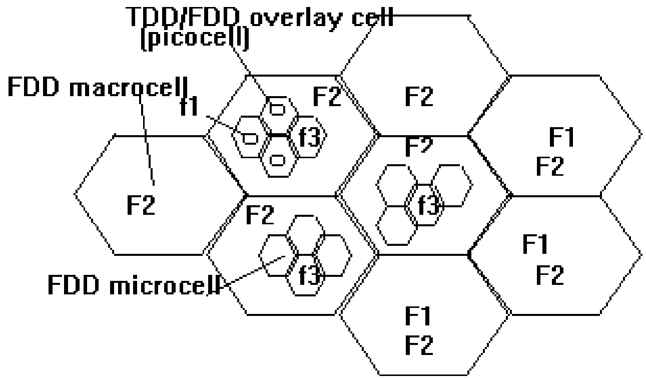

Figure 2.4 – Example of hierarchical cell structure (extracted from [9])

2.2 Modes of Operation

2.2.1 UTRA FDD

In the FDD mode, two frequencies are used at the same time; one for the UL transmission and the other for DL. The main characteristics of the FDD mode are related to the symmetric traffic and large coverage. The FDD mode is used with Wideband Code Division Multiple Access (WCDMA), which is a Direct Sequence Code Division Multiple Access (DS-CDMA) technique that handles higher bit rates, and supports a highly variable bit rate. The user bit rate can be adjusted each 10 ms frame and during the transmission of the frames the data rate is kept constant [7]. The main characteristics of the WCDMA can be seen in Table 2.2. For compatibility reasons, particularly with GSM, the FDD super frame length is 720 ms = 6×120 ms, which is an integer multiple of the corresponding GSM super frame.

In UL, data and control channels are I/Q multiplexed, while in DL, data and control channels are time multiplexed [10]. The frame structures for both UL and DL can be seen in Figure 2.5 and Figure 2.6.

Table 2.2 – WCDMA Parameters.

Multiple Access Method	DS-CDMA
Duplex Technique	FDD
Chip Rate	3.84 Mchip/s
Carrier Spacing	5 MHz
Frame Duration	10 ms
Spreading Technique	Variable-spreading factor + multi-code
Channel Coding	1/2-1/3 rate convolutional code (Turbo Coding)
Modulation	QPSK (roll-off factor 0.22)

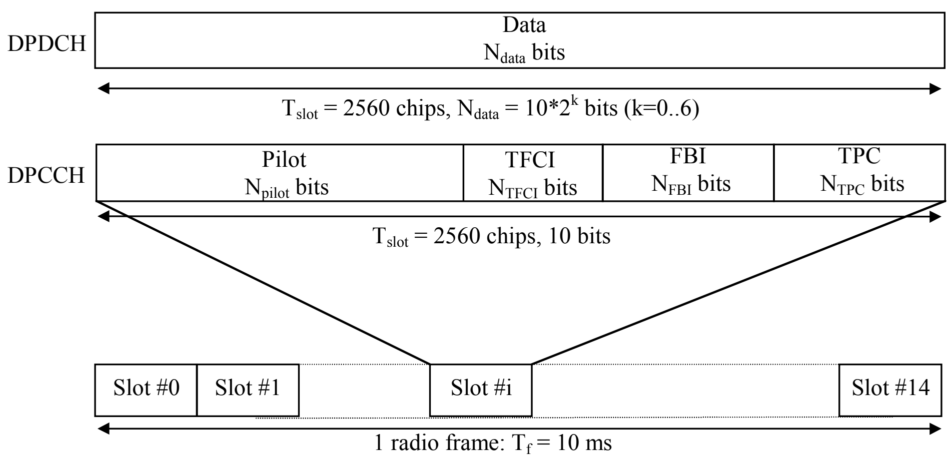

Figure 2.5 – Frame structure for FDD uplink (extracted from [10]).

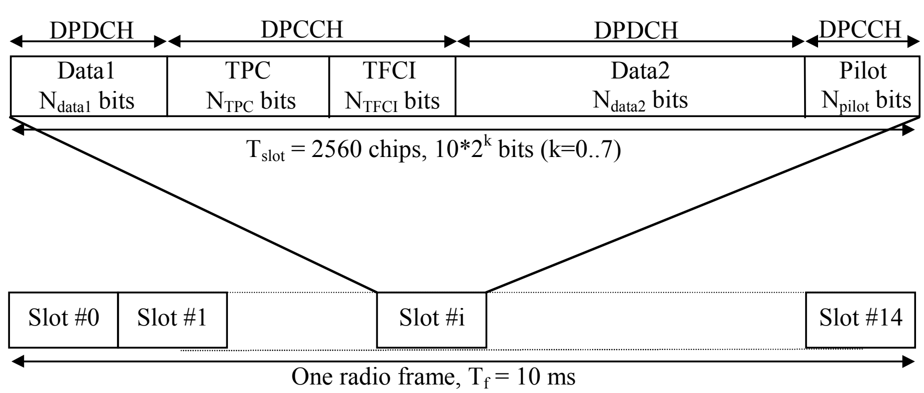

Figure 2.6 – Frame structure for FDD downlink (extracted from [10]).

Spreading is applied to the transmitted signal in order to increase the signal bandwidth and to separate Mobile Terminals (MTs) and Base Stations (BSs) from each other. It consists of two operations, channelisation and scrambling as one can see in Figure 2.7. Channelisation transforms every data symbol into a number of chips, thus increasing the bandwidth of the signal; scrambling does not change the signal bandwidth, and it only separates signals from different sources. The number of chips per data symbol is called Spreading Factor (SF). The use of the Orthogonal Variable Spreading Factor (OVSF) technique allows the SF to be changed and orthogonality among different spreading codes of different lengths to be maintained; the OVSF codes can be defined using the code tree of Figure 2.8. A physical channel may use a certain code in the tree if no other physical channel to be transmitted using the same code tree is using a higher spreading factor code generated from the intended spreading code to be used. The DL orthogonal codes within each BS are managed by the RNC in the network.

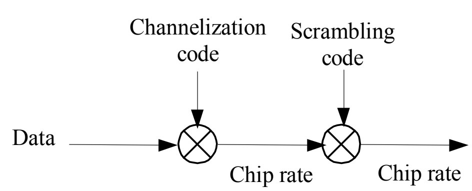

Figure 2.7 – Relation between spreading and scrambling (extracted from [7]).

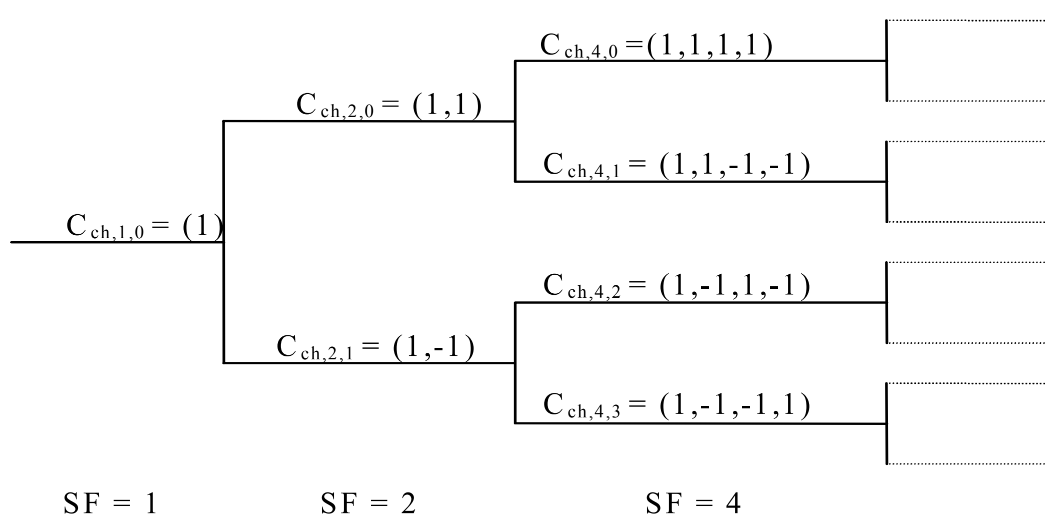

Figure 2.8 – Code tree for generation of OVSF codes (extracted from [10]).

In UL, Dedicated Physical Control Channel/ Dedicated Physical Data Channel (DPCCH/DPDCH) may be scrambled by either long or short scrambling codes. The long scrambling codes came from a set of Gold sequences of 38400 chips and the short ones are derived from a sequence of the family extended S (2) codes. In DL, the scrambling codes are segments of a different set of Gold sequences and are divided into 512 sets of a primary code and 15 secondary scrambling codes. The scrambling codes are repeated for every 10 ms of the radio frame. A total of 2^{18}-1 scrambling codes can be generated, however only a relative small set is used.

In UTRA, the data generated at higher levels is mapped onto transport channels, and is then mapped onto the corresponding physical channels. There are two types of transport channels: the common and the dedicated ones. The main difference between them is that the common channels are a resource that can be shared among all or a group in a cell, while the dedicated channels, identified by a certain code on a certain frequency, are reserved for a single user. The dedicated channels are characterised by fast power control and fast data rate change [7]. There are six different common transport channels types: Broadcast Channel (BCH), Forward Access Channel (FACH), Paging Channel (PCH), Random Access Channel (RACH), Common Packet Channel (CPCH), and Downlink Shared Channel (DSCH) [10]. In order to give bandwidth on demand, variable bit rate, and to be able to multiplex several services to one connection, each transport channel has a Transport Format Indicator (TFI). The physical channel combines the TFI information from the different transport channels to the Transport Format Combination Indicator (TFCI). Figure 2.9 shows the transport channels and their mapping onto the physical channels.

In addition to the transport channels, there are the SCH (Synchronisation Channel), the CPICH (Common Pilot Channel), and the AICH (Acquisition Indication Channel), that only carry relevant information to the physical layer and are of key importance. The CSICH (CPCH Status Indication Channel) and the CD/CA-ICH (Collision Detection/Channel Assignment Indication Channel) are only needed if CPCH is used [7].

Transport Channels	Physical Channels
DCH	Dedicated Physical Data Channel (DPDCH)
	Dedicated Physical Control Channel (DPCCH)
RACH	Physical Random Access Channel (PRACH)
CPCH	Physical Common Packet Channel (PCPCH)
	Common Pilot Channel (CPICH)
BCH	Primary Common Control Physical Channel (P-CCPCH)
FACH	Secondary Common Control Physical Channel (S-CCPCH)
PCH	
	Synchronisation Channel (SCH)
DSCH	Physical Downlink Shared Channel (PDSCH)
	Acquisition Indicator Channel (AICH)
	Access Preamble Acquisition Indicator Channel (AP-AICH)
	Paging Indicator Channel (PICH)
	CPCH Status Indicator Channel (CSICH)
	Collision-Detection/Channel-Assignment Indicator Channel (CD/CA-ICH)

Figure 2.9 – Transport channel onto physical channel mapping (extracted from [11]).

Power control is a very important method for controlling the interference and the quality of system deployment. Power control is implemented in UL to avoid the near far problem, and in DL to give a supplementary power margin to the MTs that are in the cell limit, hence, receiving higher interference from neighbouring cells. The methods for power control in FDD are open-loop and fast closed-loop power control. The open-loop power control estimates the path loss from the received signal; this method is inaccurate due to the fast fading uncorrelation between UL and DL, resulting from the large separation between UL and DL bands in FDD mode. The closed-loop power control, in the UL, is implemented in the BS comparing the target SIR (Signal-to-Interference Ratio) with measured values: if the measured SIR is higher than the target SIR, the BS will command the MT to lower power; if it is too low, it will command the MT to increase power. In order to obtain a constant quality, usually defined by BER (bit error rate) or BLER (block error rate), the RNC by using the outer loop power control adjusts the corresponding SIR levels.

FDD supports intra-frequency handover, inter-frequency handover and intersystem handover. Inter-frequency handover is the most common in WCDMA, since the operator starts with one frequency and then upgrades the system with more one or two frequencies. The intra-frequency handover, i.e., soft handover, handles a MT that is connected to two or more BSs providing an additional diversity gain against interference. Intersystem handover will be very important in the WCDMA implementation phase handling global coverage through the handover between WCDMA and GSM. For inter-frequency handover, dedicated CS channels use soft handover. Dedicated PS channels can use soft or hard handover, and the common channels use hard handover. In the case of inter-frequency handover, hard handover is applied where the MT measurements on the other frequencies are performed in slotted mode DL transmission, or with a dual receiver. Intersystem handover is a hard handover procedure [12].

2.2.2 UTRA TDD

The TDD mode uses the same frequency band for both UL and DL transmission. Each frame structure allows efficient bandwidth allocation in asymmetric bit rate transmission, enabling different bit rates in UL and DL.

TDD is based on TD/CDMA, which is a combination of TDMA (Time Division Multiple Access) and CDMA (Code Division Multiple Access). The TDMA frame has a duration of 10 ms and is subdivided into 15 Time Slots (TS) of 2560 chips duration each, so that the TS duration is 666 μs. Each TS of each frame can be allocated to the UL or DL (at least one in each direction). With such flexibility, the TDD mode can be adapted to different environments and deployment scenarios [13].

Figure 2.10 shows examples for multiple and single switching point configurations as well as symmetric and asymmetric UL/DL allocations of the TDMA frame structure. Each TS comprises several (maximum 16) orthogonal spreading codes. Different bit rates are supported by code and/or TS pooling [12].

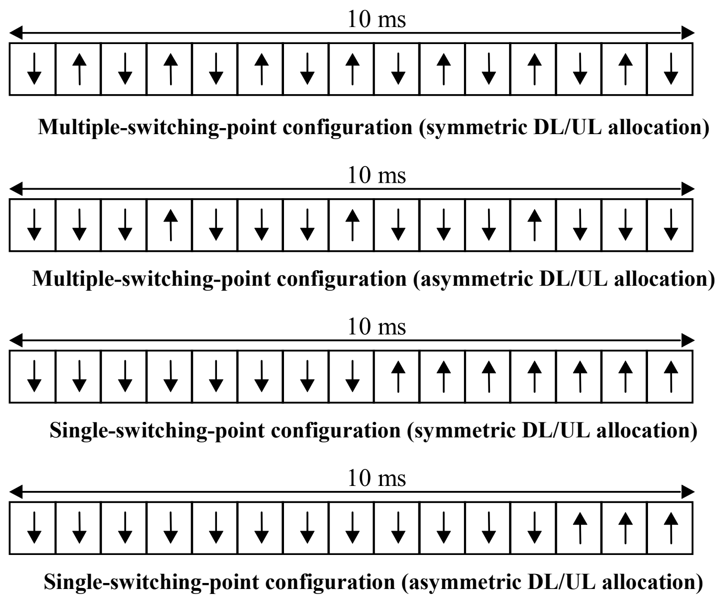

Figure 2.10 – TDD frame structure examples (extracted from [13]).

The basic frame structure of TDD is similar to FDD, although the former has 15 TS. The modulated data symbols are spread with specific channelisation codes of length 16, and then scrambled by a pseudorandom sequence of length 16. The same type of orthogonal channelisation codes is used in the FDD mode and the scrambling process is chip-by-chip multiplication. As FDD, there are two types of transport channels: common and dedicated ones. In TDD there are six different common transport channels types: Broadcast Channel (BCH), Forward Access Channel (FACH), Paging Channel (PCH), Random Access Channel (RACH), Uplink Shared Channel (USCH) and Downlink Shared Channel (DSCH) [7]. Figure 2.11 shows transport channel onto physical channel mapping.

The main goal of power control is to minimise the interference of separated radio links in both UL and DL, DPCH and PRACH being power controlled: in DL, closed-loop power control is used after initial transmission; in UL, the reciprocity of the channel is used for open-loop power control. Based on interference level at the BS and on the attenuation in the DL, the MT estimates the transmitter power [7].

Transport Channels	Physical Channels
DCH	Dedicated Physical Channel (DPCH)
BCH	Primary Common Control Physical Channel (P-CCPCH)
FACH	Secondary Common Control Physical Channel (S-CCPCH)
PCH	
RACH	Physical Random Access Channel (PRACH)
USCH	Physical Uplink Shared Channel (PUSCH)
DSCH	Physical Downlink Shared Channel (PDSCH)
	Paging Indicator Channel (PICH)
	Synchronisation Channel (SCH)
	Physical Node B Synchronisation Channel (PNBSCH)

Figure 2.11 – Transport channel onto physical channel mapping (extracted from [13]).

UTRA TDD supports inter-system handover and intra-system (with UTRA FDD and with GSM) based on hard handover mechanism. The main difference between FDD and TDD is that TDD does not use soft handover. Although the TDD protocol structure has followed the same architecture as FDD for termination points, there are specific parameters for each one [7].

2.3 Services and Applications

2.3.1 Service Classes

The classification of four service classes in UMTS (conversational, streaming, interactive and background) [14] by 3GPP has the main purpose of defining the quality requirements of different applications, taking into account the limitations of the radio interface, in particular the connection delay:

- *Conversational Class* – is related with the voice transmission. The traditional voice applications, Internet and multimedia applications, like voice over IP (VoIP) and video conferencing tools, are conversational applications.
- *Streaming Class* – characterises real time video transmission. The human perception to the delay variation is the main requirement to design the system.

- <u>*Interactive Class*</u> – is manly used in interactive applications like web browsing, data base retrieval and server access.
- <u>*Background Class*</u> – is characterised by a destination that is not expecting data within a certain time. Examples of applications are e-mails, SMS, download of databases, and reception of measured records.

The main quality requirements for each service class can be seen in Table 2.3. 3GPP also defined most relevant parameters that classify the several services and applications in UMTS, and their relevance for each class, as it can be seen in Table 2.4.

Table 2.3 – Characteristics of the different UMTS services classes [14].

	Conversational	**Streaming**	**Interactive**	**Background**
Fundamental Characteristics	• Preserve time relation between information entities of the stream. • Conversational pattern (stringent and low delay)	• Preserve time relation between information entities of the stream.	• Request response pattern • Preserves payload content (low bit error rate)	• Destination is not expecting the data within a certain time. • Preserve payload content (low bit error rate)
Connection Delay	Minimum fixed	Minimum variable	Moderated variable	Big variable
Buffering	Not Allowed	Allowed	Allowed	Allowed
Symmetry	Symmetric	Asymmetric	Asymmetric	Asymmetric
Bandwidth	Guaranteed bit rate	Guaranteed bit rate	No guaranteed bit rate	No guaranteed bit rate
Applications examples	• Speech service • VoIP • Video telephony	• Video on-demand • Web broadcast	• Web browsing • Sever access • Play Interactive computer games	• SMS • E-mail • Electronic postcard • Downloading of databases

Table 2.4 – UMTS bearer attributes defined for each bearer traffic class [15].

Traffic class	Conversational	Streaming	Interactive	Background
Maximum bit rate	X	X	X	X
Delivery order	X	X	X	X
Maximum SDU size	X	X	X	X
SDU format information	X	X		
SDU error ratio	X	X	X	X
Residual bit error ratio	X	X	X	X
Delivery of erroneous SDUs	X	X	X	X
Transfer delay	X	X		
Guaranteed bit rate	X	X		
Traffic handling priority			X	
Allocation/Retention priority	X	X	X	X
Source statistics descriptor	X	X		

UMTS Forum and ITU-T have also defined service classes that are derived from other factors, but are complementary to the ones presented by 3GPP.

The classification of service classes by UMTS Forum [6] is mainly attributed to the marketing requirements. The six UMTS Forum service classes are High Interactive Multimedia, High Multimedia, Medium Multimedia, Switched Data, Simple Messaging and Speech.

The ITU-T classification [16] is a more general approach, which is not directly related to UMTS and is mainly based on the network. ITU-T defines two main service classes: interactive services, and distribution ones. Interactive services are those in which there is a two-way exchange of information between two subscribers or between a subscriber and a service provider. The interactive services are subdivided into three classes of service; conversational, messaging and retrieval. The conversational is characterised by providing a bi-directional communication with real time end-to-end information transfer, user-to-user, or between user and host. Messaging services offer user-to-user communication between individual users via storage units with store-and-forward. The user of retrieval services can retrieve information stored in information centres provided for public use. The distribution services characterise the information distribution from one service provider to the subscriber, where the user has no control over the presentation of the information, and is sub-divided into one that allows user individual presentation control and in another which does not allow it.

2.3.2 Services and Applications Characterisation

For the traffic classes described in Section 2.3.1, eight service were identified in [17] as described in Table 2. 5.

- *Speech-telephony* - Traditional speech-telephony.
- *Video-telephony* - Communication for the transfer of voice and video between two locations.
- *Streaming Multimedia* - Service that allows the visualisation of multimedia documents on a streaming basis, e.g., video, music, or slide show.
- *Web Browsing* - This is an interactive exchange of data between a user and a web server. It allows the access to web pages. This information may contain text, extensive graphics, video and audio sequences.
- *Location Based Service* - Interactive service that enables users to find location-based information, such as the location of the nearest gas stations, hotels, restaurants, and so on.
- *Multimedia Messaging Service (MMS*) - A messaging service that allows the transfer of text, image and video.
- *E-mail* - A process of sending messages in electronic format. These messages are usually in text form, but can also include images and video clips.
- *File Download* - Download a file from a database.

Table 2. 5 presents the characterisation parameters for each service and details the following characteristics:

- *Information type* - sound, video, text, data, still image.
- *Intrinsic time dependency* - time-based (TB, where data blocks must be displayed consecutively at predetermined time instants), or non-time-based (NTB).
- *Delivery requirements* - real-time (RT, for immediate consumption), or non-real-time (NRT, stored for later consumption).
- *Directionality of Connection* - unidirectional (Uni), or bidirectional (Bid).

- *Symmetry of Connection* (for Bid connections) - symmetric (Sym), or asymmetric (Asy).
- *Number of Parties* - one-to-one (O-O), or one-to-many (O-M).
- *Switching mode* - Packet Switched (PS), or Circuit Switched (CS).
- *SF* - Spreading factor.
- *Average Duration* - average duration and DL session volume are directly related by the DL average source bit rate.
- *Maximum transfer delay* - This is the maximum time used to transmit information through the air interface and the UMTS network.
- *Burstiness* - ratio between peak and average bit rates.
- *Bit Error Ratio (BER)* - fraction of lost or erroneous bits.
- *Frame Erasure Ratio (FER)* - fraction of lost or erroneous frames.

The four 3GPP service classes described in Section 2.3.1, grouping services according to specific characteristics and performance requirements, are well represented in the chosen service set, Figure 2. 12.

- From the conversational class (characterised by symmetric and real-time conversational pattern services, with low emphasis on signal quality), *Speech-telephony* and *Video-telephony* services are chosen. The bit rate and session volume strongly differs between these two services, being important to handle this diversity in simulations.
- From the streaming class (characterised by real-time almost unidirectional data flow applications with low delay variation, which can be processed as a steady continuous stream) *Streaming Multimedia* is chosen. This service covers both audio and video streaming.
- From the interactive class (characterised by 'request-response' pattern services, highly asymmetric, with low round trip delay and high signal quality) *Web Browsing* and *Location Based* services are chosen. The average DL session volume differentiates these two services.
- From the background class (non real-time asymmetric services, with high signal quality), *File Download*, *E-Mail* and *MMS* services are chosen. *File Download* is a bi-directional service but highly asymmetric, most of the traffic being DL. The remaining services are differentiated by their average bit rate and DL session volume.

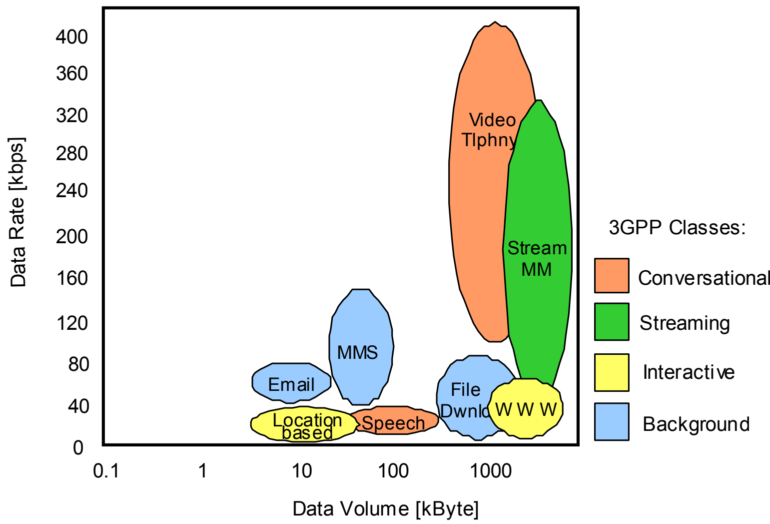

Figure 2. 12 - Service set bit rate range and DL session volume (extracted from [17])

Table 2. 5 – Service Characteristics. (extracted from [17])

Class	Service	Info. Type	TB/ NTB	RT/ NRT	Uni/ Bid	Sym/ Asy	Par-ties	CS/ PS	Source Bit rate Range [kbps]	Av. Source Bit rate [kbps] UL	Av. Source Bit rate [kbps] DL	DL session volume [kByte]	Av. Duration [s]	Max. Tansf. Delay [s]	Burst-iness	BER	FER
Convers	Speech-telephony	Sound	TB	RT	Bid	Sym	O-O	CS	4 - 25	12.2	12.2	91.5[1]	120	0.15	1 - 5	10^{-4}	< 3%
	Video- telephony	Sound Video	TB	RT	Bid	Sym	O-O	CS	32 - 384	100	100	1 500	120	0.15	1 - 5	10^{-3}	<1%
Stream..	Stream. MM	MM	TB/ NTB	RT	Bid	Asy	O-O	PS[2]	32 - 384	3	60	2 250	300	10	1	10^{-6}	<1%
Interactve	Web-browsing	MM	TB	RT	Bid	Asy	O-O	PS	< 2000	1	30	1 125	300	4/ page	1 - 20	10^{-6}	<1%
	LocationBased	MM	TB/ NTB	RT	Bid	Asy	O-O	PS	< 64	1	10	22.5	180	0.2	1 - 20	10^{-6}	<1%
Background	MMS	MM	TB	NRT	Uni[3]	Asy	O-O	PS	< 128	30	30	60	16.2	300	1 - 20	10^{-6}	-
	E-Mail	Data	NTB	NRT	Uni	Asy	O-O	PS		30	30	10	2.4	4[4]	1	10^{-6}	-
	File Dwnld.	Data	NTB	NRT	Bid	Asy	O-M	PS	64 - 400	1	60	1 000	132	0.5	1 - 50	10^{-6}	-

[1]: For the calculation of the equivalent Speech-telephony call volume, an activity factor of 50% was considered.

[2]: Streaming Multimedia can also be CS for the case of video streaming.

[3]: MMS and E-Mail are unidirectional services, existing in a session as an UL or DL transmission, but never both.

[4]: For the E-Mail maximum transfer delay, a server access of 4 seconds was considered.

3 TRAFFIC FUNDAMENTALS

3.1 Radio Network Planning

3.1.1 Initial Considerations

The three main factors of radio network planning in UMTS are coverage, capacity and quality of service. The main parameters for coverage planning are related to the coverage regions, the area type information and propagation conditions. For capacity planning is important to take into account the spectrum available, the subscriber growth forecast and the density of traffic. The quality of service is directly dependent on the blocking probability, end user throughput and area location probability.

In addition to the general parameters used in the link budget estimation, there are some WCDMA-specific parameters that have to be taken in account [7]:

- *Interference margin* – the interference margin is needed in the link budget because the loading of the cell affects coverage. The more loading that is allowed in the system, the larger is the interference margin needed in the UL, and the smaller is the coverage area.
- *Fast fading margin* (power control headroom) – some headroom is needed in the MT transmission power for maintaining adequate closed-loop power control. This applies especially to slow-moving pedestrian mobiles MTs fast power control is able to compensate fast fading.
- *Soft handover gain* – Handovers (soft and hard) give a gain against slow fading by reducing the required log-normal fading margin. This is because the slow fading is partly uncorrelated between the BSs, and by making handover, the MT can select a better BS. Soft handover gives an additional macro diversity gain against fast fading by reducing the required Signal-to-Noise Ratio (SNR).

This chapter focuses on WCDMA, and the main differences between the UL and the DL planning in terms of load factor, coverage and capacity. The influence of admission control mechanisms and soft handover capacity is also analysed.

Multi-operator interference is also mentioned, and a short description of the fundamental characteristics that a planning tool must have.

3.1.2 Coverage and Capacity

In DL, the coverage is more dependent on the system load than in the UL. This is due to the BS power being shared by all the users in the cell, and having the same value regardless of the number of users. In UL, each new user has its own power amplification, therefore, even in low load the coverage in DL decreases in proportion to the number of users. On the other hand, coverage is more limited in UL than in the DL because the MT power is limited. One example of this limitation corresponds to the high transmission bit rate when there are few users in the cell. In this situation the BS can deliver high power to the user with a high bit rate in the DL, but the MT is not able to increase its power to improve the UL coverage [7].

When in the downlink the power is increased by 3 dB, the maximum path loss admitted in the system also increases by 3 dB for a given load. The capacity improvement is smaller than the coverage improvement and for the same increase in power, the capacity can only be increased by 10%. Thus, increasing downlink transmission power is an inefficient method to increase the downlink capacity. However, splitting the downlink power between two frequencies would increase downlink capacity by approximately 80 %. This is an efficient method to increase the downlink capacity without extra investment in power amplifiers. In order to implement this method it is necessary that the operator has two carriers.

WCDMA allows a trade-off between capacity and coverage. If there are few users in the cell, more power can be allocated for one user allowing a higher path loss. The power amplification also allows the addition of a second carrier without adding a power amplifier.

For capacity upgrade, there are many options. In UMTS implementation phase, sectorisation allows more capacity and better coverage. WCDMA allows sectorisation with only one power amplifier that is shared by all the sectors in the cell. This is a low cost solution that provides 28% of the capacity compared with the real sectorisation solution [7]. Another method of providing the link upgrade, when the operator allocation plan allows, is the use of two or three carriers. Solutions with diversity with 2^{nd} power amplification per sector are also possible for link upgrade.

Another aspect to consider is the asymmetric behaviour that allow different load in UL and DL, which must be taken into account in the radio network planning. In general, DL coverage of higher bit rates is better than UL coverage, since more power can be allocated per connection in DL than in UL. If the cell is planned to provide a higher bit rate in UL, the cell coverage is reduced and the DL coverage is better than the UL one.

The MT speed also has an impact on coverage and capacity. When MTs are moving at low speed, it is possible to obtain a better capacity. The influence of MT speed in the coverage is different. For higher speeds the required fast fading margin is low, thus, the coverage probability is improved when the MT speed increases. At very high MT speeds (higher than 100 km/h), the fast power control cannot follow the fast fading, and a higher power level is needed to obtain the required quality. However, the diversity helps to keep the received power level constant and lower the average receiver power level enough to provide the same quality of service.

The SNR requirements are a function of the bit rate, class of service, multipath diversity, MT speed, reception algorithm and BS antennas structure. For lower MT speeds the SNR requirements are low, but it is necessary to have a higher margin against fast fading; the main limiting factor is coverage for low MT speeds.

3.1.3 Load factor

In interference limited systems, the estimation of the amount of supported traffic per BS is the main issue in dimensioning the system.

The UL load factor is expressed by [7]:

$$\eta_{UL} = \sum_{j=1}^{N} \frac{(E_b/N_0)_j}{W/R_j} \cdot \nu_j \cdot (1+i) \qquad \textbf{(3.1)}$$

- N is the number of users per cell
- ν_j the activity factor of user j at physical level
- E_b/N_0 is the energy per bit and the noise spectral density ratio that is required to meet a predefined quality of service
- W is the chip rate
- R_j is the bit rate of user j
- i corresponds to other cell to own cell interference ratio seen by the BS receiver.

The noise rise is derived from the ratio of the total wideband power to the noise power, and is related to the load factor by the equation $-10\cdot\log(1-\eta_{UL})$. The interference margin in the link budget must be equal to the maximum planned noise rise.

Using a similar principle as in the UL, the DL load factor can be expressed by:

$$\eta_{DL} = \sum_{j=1}^{N} \nu_j \cdot \frac{(E_b/N_0)}{W/R_j} \cdot \left[(1-\alpha_j) + i_j \right] \qquad \textbf{(3. 2)}$$

- N is the number of connection per cell. The number of connection per cell is obtained by the number of user per cell×(1+soft handover overhead)
- ν_j is the activity factor of user j at physical level
- R_j is the bit rate of user j
- α_j is the orthogonality of channel of user j
- i_j is the ratio of other cell to own cell BS power, received by user j.

As in the UL, the noise rise is related to the load factor by $-10\cdot\log(1-\eta_{DL})$.

The main difference between UL and DL transmission is the use of orthogonal codes in DL to separate users. However, the delay spread in the radio channel can cause an imperfection in the orthogonality, causing the MT to see part of the BS signal as multiple access interference. The α_j parameter takes into account this factor, and has as typical values 0.4 e 0.9 for multipath channels (1 corresponds to perfectly orthogonal users). Another parameter that differentiates UL and DL transmission is related to *i*, the other cell to own cell interference ratio: in UL, *i* has the same value for all the users because this value relates to the BS measurements; in DL, *i* relates to the MS measurements, so this value is different for each user and is dependent on the user location. The soft handover transmission is modulated taking into account additional links in the cell.

The most important parameter in DL dimensioning is the BS transmission power required, and the number of users in the cell. In the cell borders users need more power, while users that are close to the BS need much less power. As a result, for dimensioning the BS transmission power, the average value for all the users should be used. The difference between the average power and the power calculated by the link budget is called trunking gain, and is typically 6 dB. The total BS transmission power can be expressed by:

$$BS\_TxP = \frac{N_{rf} \cdot W \cdot \overline{L} \cdot \sum_{j=1}^{N} \nu_j \cdot \frac{\left(E_b / N_0 \right)_j}{W / R_j}}{1 - \overline{\eta}_{DL}} \qquad \textbf{(3. 3)}$$

where N_{rf} is the spectral noise density of the MT receiver front-end that can be obtained from:

$$N_{rf} = k \cdot T + NF \qquad \textbf{(3. 4)}$$

where *k* is Boltzmann constant in J/K, *T* the temperature in Kelvin and *NF* is the MT receiver noise figure with typical values of 5-9dB.

After taken into account the trunking gain, the load factor can be approximated by its average value across the cell:

$$\overline{\eta}_{DL} = \sum_{j=1}^{N} \nu_j \cdot \frac{(E_b/N_0)}{W/R_j} \cdot \left[(1-\overline{\alpha}) + \bar{i}\right] \qquad \textbf{(3. 5)}$$

- $\overline{\alpha}$ is the average orthogonality factor in the cell
- $\bar{i}$ is the average ratio of other cell to own cell BS power received by user, the own cell interference is here wideband.

If the air interface loading is allowed to increase excessively, the coverage area of the cell is reduced below planned values, and the quality of service of the existing connection cannot be guaranteed. For this reason, it is very important that a admission mechanism controls the access and rejects a new connection if the cell is overloaded. The requesting bearer can be admitted only if both UL and DL control allows it. The radio network planning sets the limits for admission control.

There are different methods for admission control implementation, the use of the total power received by the BS is used as a primary UL admission control decision criteria. In the DL the admission control algorithm is based on the total DL power transmitted by the BS.

3.1.4 Handover

In UMTS, three types of handover are allowed, softer handover, soft handover and hard handover (inter-frequency or intersystem). In softer handover, the communication between a MT in the overlapping of two adjacent sectors and the BS take place via two air interface channels. The way to distinguish the two signals in the DL is the use of two separate codes, which are received in the MT by means of RAKE receiver processing. In UL, a similar process takes place in the BS, the code channel of the MT is received in each sector and then routed to the same baseband RAKE receiver. During softer handover only one power control loop per connection is active.

In soft handover the communication of a MT in the overlapping cell coverage area of two sectors belonging to different BS and the corresponding BS occur via the air interfaces from each BS separately. The main difference between the softer and the soft handover corresponds to the data routing and combining by the RNC from the signals upcoming from the two BS. During soft handover two power control loops per connection are active, one for each BS.

Hard handover can be used to hand a MT over from one frequency carrier to another. This kind of handover is needed to balance the loading between carriers if there are several carriers in the same BS. It also enables handover between different cell layers of the multi-layered cellular network, when the cell layers use different carrier frequencies (for example handover between macro-cells and micro-cells). Since there is need for compatibility between third-generation and second-generation systems, to provide a continuous coverage, hard handovers may take place between FDD mode and TDD or even another system like GSM.

If the capacity is hard blocking, the Erlang capacity can be obtained from the Erlang B model. For interference limited systems, there is no single fixed value for the maximum capacity and the Erlang capacity cannot be calculated from the Erlang B formula because it would give too pessimistic results.

The maximum capacity is larger than the average number of channels per cell, therefore, the adjacent channels share part of the same interference, and more traffic can be served with the same blocking probability. In fact, if there is less interference from the neighbouring cell, more channels are available to the cell. On the other hand, if average loading is low, there will be additional capacity in the neighbouring cells that can be shared with soft handover.

To estimate the increase in capacity due to soft handover, it is assumed that the number of users is the same in all cells, but the start and end of connection is independent from each one.

To estimate the soft handover capacity increase, it is necessary to calculate the number of channels per cell, N_c, in the equally loaded case, based in the UL load factor. Then by multiplying that number of channels by $1+i$ (i is the ratio of other cell to own cell BS power, received by user) the total channel in soft handover blocking case is obtained. The CS capacity estimation is obtained by calculating the offered traffic from the Erlang B model and then dividing the Erlang capacity by i+1. For PS a model that contemplate the queues behaviour like for example the Pollaczek-Khinchin one can be used.

3.1.5 Link Budget Calculations

The estimation of the receiver power in UL is by:

$$P_{Rx[\mathrm{dBm}]} = P_{r[\mathrm{dBm}]} - L_{c[\mathrm{dB}]} \quad \textbf{(3. 6)}$$

where P_{Rx} corresponds to the receiver-input power, P_r to the power at the antenna terminals, and L_c to the cable losses.

In the DL one has:

$$P_{Rx[\mathrm{dBm}]} = P_{r[\mathrm{dBm}]} - L_{u[\mathrm{dB}]} \quad \textbf{(3. 7)}$$

where L_u is associated to the losses due to the presence of the user. The receiver sensitivity depends on the service:

$$P_{Rx\,\min[\mathrm{dBm}]} = E_b / N_{0[\mathrm{dB}]} - G_{p[\mathrm{dB}]} + N_{T[\mathrm{dBm}]} \quad \textbf{(3. 8)}$$

where the processing gain G_p is given by:

$$G_p = 10 \cdot \log\left(\frac{W}{R_b}\right) \quad \textbf{(3. 9)}$$

where R_b is the corresponding service bit rate.

The total noise plus interference is given by:

$$N_{T[\mathrm{dBm}]} = N_{R[\mathrm{dBm}]} + M_{I[\mathrm{dB}]} \qquad \textbf{(3. 10)}$$

where:

$$N_{R[\mathrm{dBm}]} = -174 + 10 \cdot \log(W) + NF_{[\mathrm{dB}]} \qquad \textbf{(3. 11)}$$

For the BS the noise figure is assumed equal to 5dB and

$$M_I = -10 \cdot \log(1 - \eta) \qquad \textbf{(3. 12)}$$

where η corresponds to the load factor and depends on the utilisation as detailed previously.

3.1.6 Propagation Model

The COST 231 Walfish-Ikegami Model [18] is adequate to urban and micro-cell environments taking into account the major urban parameters, like street and building dimensions. For the scenarios characterised in this work, this model satisfies the environment and system parameters conditions. In the following, the major parameters used for path loss calculations are presented.

For distance d ≥ 20m in light of sight the propagation loss L_p is given by:

$$L_{p[\mathrm{dB}]} = 42.6 + 20 \cdot \log\left(d_{[\mathrm{km}]}\right) + 20 \cdot \log\left(f_{[\mathrm{MHz}]}\right) \qquad d \geq 20m \qquad \textbf{(3. 13)}$$

where *d* correspond to the distance and *f* to the frequency.

In the others cases the model is composed by three terms:

$$L_{p[\mathrm{dB}]} = \begin{cases} L_{0[\mathrm{dB}]} + L_{rst[\mathrm{dB}]} + L_{msd[\mathrm{dB}]}, & L_{tt} + L_{tm} > 0 \\ L_{0[\mathrm{dB}]}, & L_{tt} + L_{tm} \leq 0 \end{cases} \qquad \textbf{(3. 14)}$$

the first term represents the free-space loss L_0, the second one the "roof-top-to-street diffraction and scatter loss" L_{rts} and the third one the "multi-screen loss" L_{msd}.

The free-space loss is given by:

$$L_{0[\mathrm{dB}]} = 32.4 + 20 \cdot \log\left(d_{[\mathrm{km}]}\right) + 20 \cdot \log\left(f_{[\mathrm{MHz}]}\right) \qquad \textbf{(3. 15)}$$

The roof-top-to-street diffraction and scatter loss is given by:

$$L_{rst[\mathrm{dB}]} = -16.9 - 10 \cdot \log\left(w_{[\mathrm{m}]}\right) + 10 \cdot \log\left(f_{[\mathrm{MHz}]}\right) + 20 \cdot \log\left(\Delta h_{Mobile[\mathrm{m}]}\right) + L_{ori[\mathrm{dB}]} \qquad \textbf{(3. 16)}$$

where

$$L_{ori} = \begin{cases} -10 + 0.374 \cdot \Phi_{[^\circ]} & 0^\circ \leq \Phi < 35^\circ \\ 2.5 + 0.075 \cdot \left(\Phi_{[^\circ]} - 35\right) & 35^\circ \leq \Phi < 55^\circ \\ 4.0 + 0.114 \cdot \left(\Phi_{[^\circ]} - 55\right) & 55^\circ \leq \Phi \leq 90^\circ \end{cases} \qquad \textbf{(3. 17)}$$

and w corresponds to the building separation and Φ to the incidence angle.

$$\Delta h_{Mobile[\mathrm{m}]} = h_{Roof} - h_{Mobile} \qquad \textbf{(3. 18)}$$

The multi-screen diffraction loss is given by:

$$L_{msd[\mathrm{dB}]} = L_{bsh[\mathrm{dB}]} + k_a + k_d \cdot \log\left(d_{[\mathrm{km}]}\right) + k_f \cdot \log\left(f_{[\mathrm{MHz}]}\right) - 9 \cdot \log\left(w_{b[\mathrm{m}]}\right) \qquad \textbf{(3. 19)}$$

where w_b corresponds to building separation plus buildings width.

$$L_{bsh} = \begin{cases} -18 \cdot \log\left(1 + \Delta h_{Base[\mathrm{m}]}\right) & h_{Base} > h_{Roof} \\ 0 & h_{Base} \le h_{Roof} \end{cases} \qquad \textbf{(3. 20)}$$

$$\Delta h_{Base[\mathrm{m}]} = h_{Base} - h_{Roof} \qquad \textbf{(3. 21)}$$

$$k_a = \begin{cases} 54 & h_{Base} > h_{Roof} \\ 54 - 0.8 \cdot \Delta h_{Base[\mathrm{m}]} & d \ge 0.5\mathrm{km} \quad \textit{and} \quad h_{Base} \le h_{Roof} \\ 54 - 0.8 \cdot \Delta h_{Base[\mathrm{m}]} \cdot \left(d_{[\mathrm{km}]} \big/ 0.5 \right) & d < 0.5\mathrm{km} \quad \textit{and} \quad h_{Base} \le h_{Roof} \end{cases} \qquad \textbf{(3. 22)}$$

k_a represents the increase in path loss floor the BS antennas below the roof tops of the adjacent buildings.

$$k_d = \begin{cases} 18 & h_{Base} > h_{Roof} \\ 18 - 15 \cdot \dfrac{\Delta h_{Base[\mathrm{m}]}}{h_{Roof}} & h_{Base} \le h_{Roof} \end{cases} \qquad \textbf{(3. 23)}$$

$$k_f = -4 + \begin{cases} 0.7 \cdot \left(\dfrac{f_{[\mathrm{MHz}]}}{925} - 1 \right) & \textit{medium sized cities and suburban centres} \\ 1.5 \cdot \left(\dfrac{f_{[\mathrm{MHz}]}}{925} - 1 \right) & \textit{metropolitan centers} \end{cases} \qquad \textbf{(3. 24)}$$

k_d and k_f control the dependence of the multi-screen diffraction loss versus distance and radio frequency, respectively.

The COST-231-Walfish-Ikegami-Model is restricted to the parameters indicated in Table 3. 1.

Table 3. 1 – Limits for COST 231 Walfish-Ikegami Model.

Parameter	Limits Values
f [MHz]	800-2000
h_{Base}[m]	4-50
h_{Mobile}[m]	1-3
d [km]	0.02-5

Table 3. 2 shows the recommended values for utilisation in the cases where the data concerning the structure of buildings and roads is unknown.

Table 3. 2 – Recommended values for unknown parameters.

Parameter	Limits Values
w_b[m]	20-50
w[m]	$w_b/2$
h_{Roof}[m]	3.(nº of floors)+roof
$roof$ [m]	3(pitched); 0 (flat)
Φ[º]	90

3.1.7 Final Considerations

The coexistence of different operation modes and the small guard bands between the carriers can provide interference between multi-operator systems. This interference can occur between two FDD systems, two TDD systems or between a TDD and a FDD system.

The interference between two FDD systems can occur when there is a small guard band between adjacent channels. A limit situation occurs when a MT from the operator A is transmitting on full power very close to a BS from operator B that is receiving on the adjacent carrier.

The interference between two TDD systems can occur, since UL and DL share the same frequency band. TDD is a synchronous system, and this interference occurs if the base station is not synchronised. Another factor that can produce interference between UL and DL is the asymmetry between UL and DL in adjacent cells. UL and DL interference can also occur between adjacent carriers and between different operators. The interference between operators depends on the BS location and increases if the attenuation between the BS is low.

Although the TDD and FDD coexistence is a good solution for global coverage, interference problems can occur as a result of the proximity between the frequency bands. A UTRA TDD MT operating in 1920-1980 MHz can interfere with the reception of a UTRA TDD BS operating in 1900-1920 MHz. Additionally, a UTRA FDD MT operating in 1920-1980 MHz can also interfere with the reception of UTRA TDD MT operating in 1900-1920 MHz.

With WCDMA, all the users share the same interference source and for that reason they cannot be independently analysed. Each user is influenced by the others, and power changes as a consequence. Thus, the planning process in WCDMA must be iterative until the power stabilises. The fast power control, the soft and hard handover and the orthogonal channels in DL are very important in system performance, and have to be taken into account.

The planning tools in WCDMA are very important, since they perform automatic optimisation of the BS configuration, antennas selection and respective height, and also site localisation in order to provide the required levels of quality and coverage at low cost. The rigorous control of the available power, admission control, load control and handovers are the main parameters in providing the required quality of service and coverage with maximum capacity at low costs.

3.2 Traffic Source Models

3.2.1 Mobile Communication Voice Model

In the classical voice model, voice calls and switched data services are generated according to a Poisson process [19]:

$$P_n(t) = \frac{(\lambda \cdot t)^n \cdot e^{-\lambda \cdot t}}{n!} \tag{3.25}$$

where $P_n(t)$ is the n message probability, λ is the mean arrival rate (calls per second) and t is the time.

For voice, according to [19], the traffic model should be an on-off one, with activity and silent periods generated by an exponential distribution. The mean value for active and silent periods is 3 s, independent of UL and DL, and both exponential distributed. The exponential Probability Density Function (PDF) is as follows:

$$f(t) = \frac{1}{\tau} \cdot e^{-\frac{t}{\tau}},\ t \geq 0 \tag{3.26}$$

where τ is the average voice call duration, and $\mu = 1/\tau$ is the service rate. The exponential Cumulative Density Function (CDF) is as follows:

$$F(t) = 1 - e^{-\frac{1}{\tau} \cdot t},\ t \geq 0 \tag{3.27}$$

In [19] a new model for voice is described, which includes not only the on-off behaviour but also the effects of voice encoder, compression devices and air interfaces characteristics. Most encoders used in WCDMA produce voice packets periodically, with a specific packet size distribution in the range between 20-40 Bytes. The model described in [19] uses a four state model, generating packets of size s_k each 10 ms, for a burst duration of τ_k, k=1,…,4. The burst duration τ_k is modelled as a random variable Weibul distributed with a mean value m_k and CDF:

$$F_W(x) = 1 - e^{(-\lambda_k \cdot x)^{\beta_k}} \qquad if\ x \geq 0 \tag{3.28}$$

In long time average, the probability that a packet is of size s_k is given by P_k. After the burst of state k a new state is selected with probability Q_k:

$$Q_k = \frac{P_k / m_k}{\sum_{j=1}^{4} P_j / m_j} \quad k = 1,\ldots,4 \tag{3.29}$$

The model parameters are presented in Table 3. 3.

Table 3. 3 – Voice Source Model Parameters (extracted from [19])

State k	Packet size s_k [Bytes]	Measured probability P_k	Measured mean burst duration m_k [packet]	Weibull parameter λ_k	Weibull parameter β_k
1	2	0,5978	29,8	0,03	0,75
2	3	0,0723	2,5	0,45	0,80
3	10	0,0388	1,8	0,80	0,70
4	22	0,2911	38,8	0,05	0,90

3.2.2 Video Traffic Model

Traffic streams generated by video compression exhibit complicated patterns, which vary from one scheme to another. In [20] a model with three types of Moving Picture Expert Group (MPEG) frames is proposed: Intra-coded (I), Prediction (P), and Bidirectional (B). Frames of type I are compressed, while B and P frames involve motion compensation (prediction and interpolation) as well. The compression pattern used to encode the video stream is presented in Figure 3. 1.

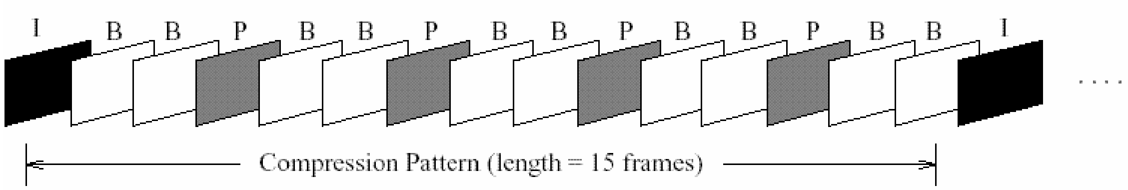

Figure 3. 1 – Compression pattern used to generate the video stream.

The model assumes that the compression pattern presented in Figure 3. 1 is continuously repeated until the end of the stream. The number of cells in a frame is assumed to follow a lognormal distribution whose parameters are presented in Table 3. 4.

Table 3. 4 – Parameters of fitting distributions. (extracted from [20]).

Frame type	Lognormal	
	μ	σ
I	5.1968	0.2016
P	3.7380	0.5961
B	2.8687	0.2675

where the lognormal PDF is given by:

$$f(t) = \frac{1}{\sqrt{2 \cdot \pi} \cdot \sigma \cdot t} \cdot \exp\left[-\frac{[\ln t - \mu]^2}{2 \cdot \sigma^2} \right] \quad \textbf{(3. 30)}$$

where σ is the log of the standard deviation and μ is the log of the mean.

The effect of scene change is incorporated in the model as follows: each stream is assumed to consist of several scenes. Scene lengths constitute a sequence of iid random variables with a geometric distribution with a mean = 10.5 frames (in units of I frames). The geometric PDF is given by:

$$f(k) = \begin{cases} p \cdot (1-p)^k & k \in \{0,1,...\} \\ 0 & otherwise \end{cases} \quad \textbf{(3. 31)}$$

where p is related with the mean μ by:

$$p = \frac{1}{1+\mu} \quad \textbf{(3. 32)}$$

The size of the first I frame in each scene is sampled from a lognormal PDF. Consecutive I frames in the same scene have exactly the same size of the first I frame.

3.2.3 General WWW or Interactive Traffic Model

For the World Wide Web (WWW) browsing sessions modelling, one could start by modelling a typical WWW browsing session, which consist of a sequence of packet calls as presented in Figure 3. 2 [21].

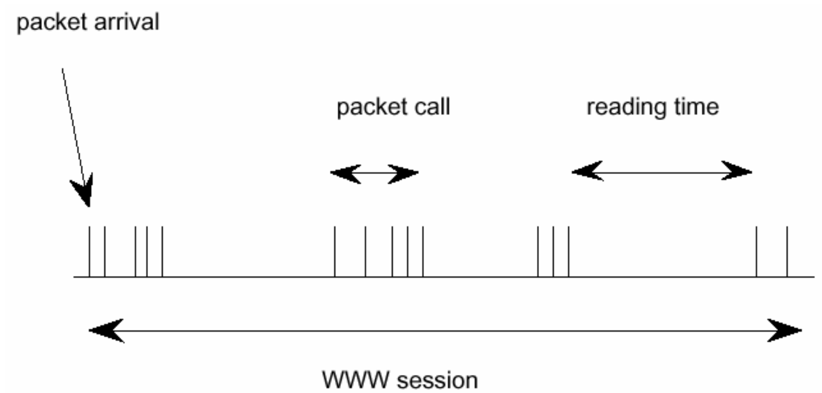

Figure 3. 2 – Typical WWW session (adapted from [21]).

Typically WWW services are classified as Long Constrained Delay (LCD), because most of them are tolerant to delay. This traffic presents asymmetric behaviour with a high volume in DL.

In a WWW session, a packet call corresponds to the downloading of a WWW document, a reading time being also modelled. During a packet call, several packets may be generated, a packet service session containing one or several packet calls, depending on the application. After the document is downloaded, the user spends a certain amount of time reading the information, this time interval is being called the reading time. It is also possible that the session contains only one packet call, this being the case for the File Transfer Protocol (FTP).

For the streaming multimedia source model the ETSI model is assumed, which consists of session and packets calls. For modelling a WWW session the following parameters are defined:

- Session arrival process - for each service only the time instant when service call begins is generated.
- N_{PC} - Number of packet call requests per session.
- D_{PC} - Reading time between two consecutive packet call requests in a session. Note that the reading time starts when the last packet of a packet call is completely received by the user and ends when the user makes a request for the next packet call.
- N_d - Number of packets in a packet call.
- D_d - Time interval between two consecutive packets inside a packet call.
- S_d - Packet size.

The statistical distribution associated with the main parameters for the ETSI session modelling are presented in Table 3. 5.

Table 3. 5 – Statistical Distributions

	Session Arrival [s]	N_{PC}	D_{PC} [s]	N_d	D_d [s]	S_d [Bytes]
Statistical Distribution	Poisson	Geometrical	Geometrical	Geometrical	Geometrical	Pareto
Parameters	Mean	μ_{NPC} - mean	μ_{DP} - mean	μ_{Nd} – mean	μ_{Dd} - mean	α and K

The Pareto PDF is given by:

$$f(x) = \begin{cases} 0 & x < k \\ k^{\alpha} \cdot \alpha \cdot x^{-\alpha-1} & x \geq k \end{cases} \tag{3. 33}$$

where k corresponds to the minimum packet size and α is related with the mean μ by:

$$\alpha = \frac{\mu}{\mu - 1} \tag{3. 34}$$

The streaming multimedia service, web browsing and location based service are modulated by the ETSI general WWW model with the default mean values for the distribution presented in Table 3. 6.

Table 3. 6 – ETSI Model based services.

	Session Arrival [s]	N_{PC}	D_{PC} [s]	N_d	D_d [s]	S_d [Bytes]
Streaming Multimedia 64 kbit/s	P(BHCA)	G(1)	G(0)	G(3280)	G(0.0625)	P(81.5,1.1)
Web Browsing 32 kbit/s	P(BHCA)	G(15)	G(20)	G(156)	G(0.125)	P(81.5,1.1)
Location Service 8 kbit/s	P(BHCA)	G(6)	G(30)	G(8)	G(0.5)	P(81.5,1.1)

For the services that follow the ETSI model, as can be seen in Table 3. 6, the arrival process follows a Poisson distribution and the number of packets call, the reading time between two packet call, the number of packets in a packet call and the time interval between two consecutive packets inside a packet call follows a Geometric distribution. For packet size modelling, as described in Table 3. 6, the Pareto distribution is considered limited by a maximum packet size of 66 666 bytes.

3.2.4 Specific WWW Traffic Model

A user that runs Non-Real-Time (NRT) applications follows a characteristic usage pattern [22]. A single user can run different applications. Each application is completely described by its statistical properties, which consists of an alternating ON-OFF process with some application specific length or data volume distribution. Within each ON period, the packet arrival process is completely defined by the packet interarrival-times and the corresponding packet sizes.

Considering the NRT user, the single user traffic model is employed in three different levels:

- <u>*Session level*</u> **-** it describes the dial-in behaviour of the individual users, characterised by the session interarrival-time distribution and the session data-volume distribution.
- <u>*Connection-level*</u> **-** it describes for each individual application the corresponding distribution of connection interarrival-times and connection data volume.
- <u>*Packet-level*</u> **-** it characterises the packet interarrival-time distribution and the packet size distribution within the application specific connections.

Email service and file download follows the Specific WWW Traffic Model with the main parameters described in Tables 3.7 to 3.11 [22].

Table 3. 7 – Distribution of Session arrival interarrival-times.

Distribution	Parameters
Lognormal (μ; σ^2)	0.6061; 7.5330

Table 3. 8 – Distribution of Session volume.

Distribution	64 kbps	144 kbps	384 kbps
UMTS Users	50%	30%	20%
Lognormal (μ; σ^2)	5.1613; 13.8210	5.2943; 14.1839	5.4197; 14.4979

Table 3. 9 – Statistical properties at connection level.

		Distribution	64 kbps	144 kbps	384 kbps
Email	Interarrrival Time [s]	Pareto (k; α)	14.4360; 2.1345	15.1334; 2.1254	16.0229; 2.1223
	Volume [bits]	Lognormal (μ; σ^2)	3.2677; 13.2369	3.2799; 13.5579	3.3084; 3.8518
File Download	Interarrrival Time [s]	Lognormal (μ; σ^2)	na	na	na
	Volume [bits]	Lognormal (μ; σ^2)	3.3433; 13.9698	3.3020; 14.7826	3.3435; 15.3229

Table 3. 10 – Fractions of different packet sizes in overall traffic.

	Fractions of packets in overall traffic [%]			
	Packet size 40 Bytes	Packet size 576 Bytes	Packet size 1500 Bytes	Other packet size
Email	38.25	25.98	9.51	26.26
File Download	40.43	18.08	9.33	32.16

Table 3. 11 – Parameters of packet interarrival-times.

	Distribution	64 kbps	144 kbps	384 kbps
Email	Lognormal (μ; σ^2)	-0.3678; 2.4220	-3.7152; 2.6410	-4.1058; 2.8902
File Download	Lognormal (μ; σ^2)	-0.0691; 3.8055	-3.2656; 3.9347	-3.4160; 4.0337

The MMS source model may be modelled by simple ON-OFF model [23], where a Poisson distribution models the service with an information volume (60 kBytes).

4 SIMULATOR DESCRIPTION

4.1 Simulator Overview

The planning tool performs the radio network planning of a UMTS network based on three main items: traffic, propagation and services. The mayor functional blocks that are used to optimise this parameters are the followings:

- User Traffic Generation
- Propagation Model
- Link Budget Calculations

To simulate the radio network, a real time traffic and propagation simulator [1] was used; as described in section 3.1, the estimation of the number of channels in UMTS is non-linear due to the radio interface used (CDMA). It depends of several factors, being difficult to find an analytical approach that computes and predicts some of these major features.

The planning tool simulates the transmission in one cell, but a fixed percentage of codes (configurable, 30% choose for simulations) for soft handover is reserved.

4.2 User Traffic Generation

For the user generation, the demographic density (pop/km^2) must be configured in the tool, and in addition, the penetrations factor for each service. After this it is possible to uniformly distribute the users over the simulation area.

For each service, it is possible to configure the BHCA (Busy Hour Call Attempt), the penetration and the service bit rate. For CS, the average duration must be also configured. For PS, the models described in section 3.2 give the duration.

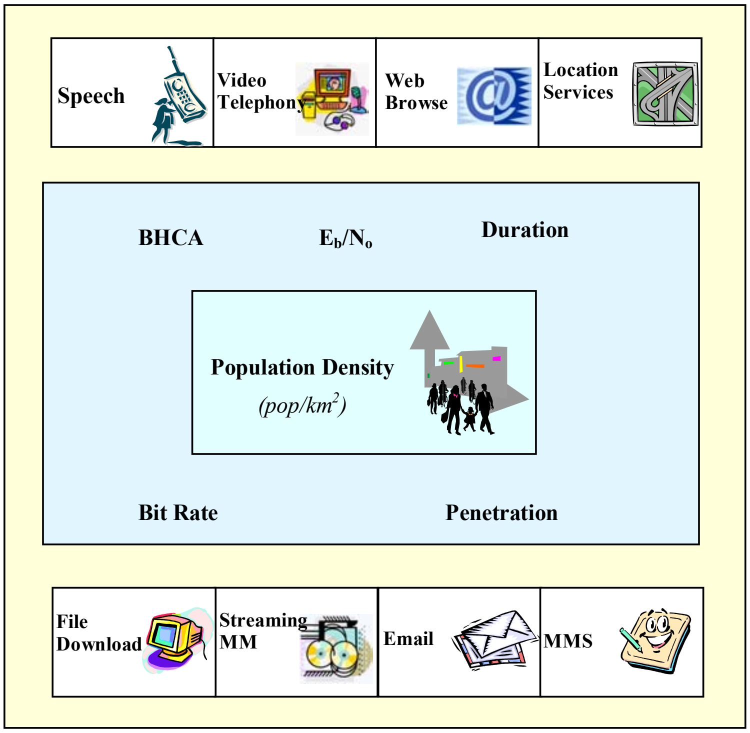

Figure 4. 1 – User traffic Generation.

For Streaming Multimedia, Web Browsing, Location Based Services and MMS it is also possible to configure the Bit Rate, Figure 4. 1. Four scenarios are considered, with different characteristics, Table 4. 1 [17].

Table 4. 1 – Simulator configuration options for Bit Rate (kbps)

	Video	Streaming	Web Browse	MMS
Scenario I	960	960	240	480
Scenario II	240	120	120	120
Scenario III	480	240	120	240
Scenario IV	480	480	240	240

The users profile generation follows a statistical approach, and a MT may use several services.

The voice model assumes that in a given call, the allocated resource commutes between four states, whose parameters can be seen in Table 4. 2.

Table 4. 2 – Speech Telephony parameters associated with each state [17].

Parameters		State I	State II	State III	State IV
Bit Rate [kbps]	UL	1.6	4.0	8.0	17.6
	DL	1.6	4.0	8.0	17.6
Radio Channel Bit Rate [kbps]	UL	15.0	15.0	30.0	60.0
	DL	15.0	15.0	30.0	60.0
E_b/N_o [dB]	UL	7.5	7.5	7.0	6.5
	DL	9.5	9.5	9.0	8.5

The video transmission assumes three quality levels as indicated in Table 4. 3.

Table 4. 3 – Video Telephony parameters associated with the quality of service [17].

Parameters		Type I	Type II	Type III
Bit Rate [kbps]	UL	64	128	384
	DL	64	128	384
Radio Channel Bit Rate [kbps]	UL	240	480	960
	DL	240	480	960
E_b/N_o [dB]	UL	4.0	3.5	4.0
	DL	4.5	4.5	4.5

For the Streaming MM service four quality levels are assumed as indicated in Table 4. 4.

Table 4. 4 – Streaming Multimedia parameters associated with the quality of service [17].

Parameters		Type I	Type II	Type III	Type IV
Bit Rate [kbps]	UL	8	8	8	8
	DL	32	64	128	384
Radio Channel Bit Rate [kbps]	UL	30	30	30	30
	DL	120	240	480	960
E_b/N_o [dB]	UL	7.0	7.0	7.0	7.0
	DL	7.5	7.5	7.5	7.5

For the Web Browsing service two quality levels are assumed as indicated in Table 4. 5

Table 4. 5 – Web Browsing parameters associated with the quality of service [17].

Parameters		Type I	Type II
Bit Rate [kbps]	UL	8	8
	DL	32	64
Radio Channel Bit Rate [kbps]	UL	30	30
	DL	120	240
E_b/N_o [dB]	UL	7.0	7.0
	DL	7.5	7.5

For the location Based Services the assumed parameters are indicated in Table 4. 6

Table 4. 6 – Location Based Services parameters [17].

Parameters		
Bit Rate [kbps]	UL	8
	DL	16
Radio Channel Bit Rate [kbps]	UL	30
	DL	60
E_b/N_o [dB]	UL	7.0
	DL	8.5

For the MMS service transmission three quality levels are assumed, as indicated in Table 4. 7.

Table 4. 7 – MMS service parameters associated with the quality of service [17].

Parameters		Type I	Type II	Type III
Bit Rate [kbps]	UL	32	64	128
	DL	32	64	128
Radio Channel Bit Rate [kbps]	UL	120	240	480
	DL	120	240	480
E_b/N_o [dB]	UL	8.5	7.0	6.5
	DL	7.5	7.5	7.5

The Email and File Download model assumes that in a given call, the allocated resource commutes among three states, whose parameters can be seen in Table 4. 8.

Table 4. 8 – Email and File Download parameters associated with each state (adapted from [17] and [22]).

Parameters		Email			File Download		
		State I	State II	State III	State I	State II	State III
Bit Rate [kbps]	UL	32	32	32	8	8	8
	DL	64	144	384	64	144	384
Radio Channel Bit Rate [kbps]	UL	120	120	120	30	30	30
	DL	240	480	960	240	480	960
E_b/N_o [dB]	UL	8.5	8.5	8.5	7.0	7.0	7.0
	DL	7.5	7.5	7.5	7.5	7.5	7.5

4.3 Link Budget and Propagation Model

The propagation model that the simulator uses to predict path loss is the COST 231 Walfish-Ikegami [18] as described in Section 3.1.6. It includes some major urban parameters, like street and buildings dimensions, and it is adapted to the urban and micro-cell environment analysed in this work. The main input parameters of the propagation model are the scenario type (urban or suburban), build height, street width, frequency, UE-NodeB distance, building separation, Node B location, geographical information and UE height, Figure 4. 2.

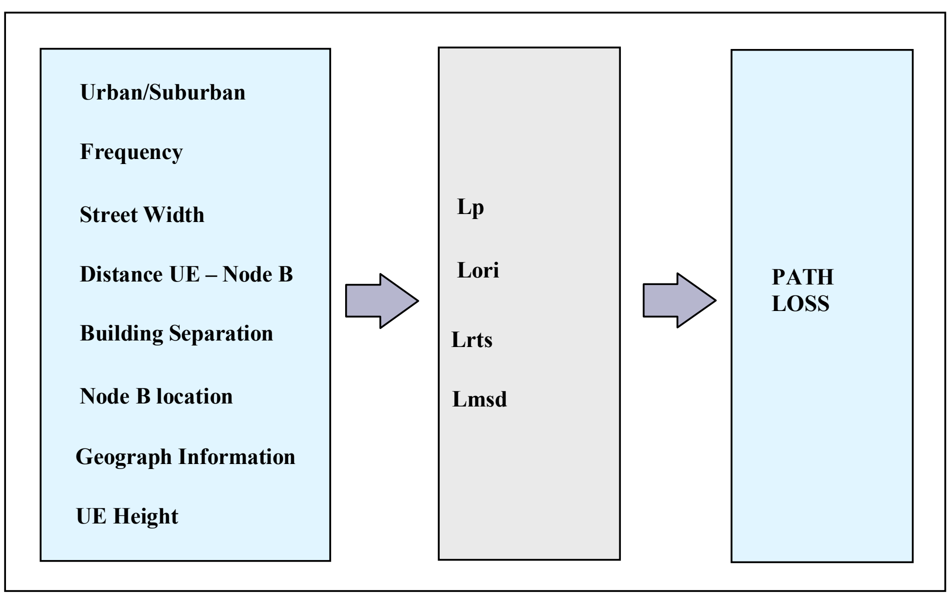

Figure 4. 2 – COST 231-Walfish-Ikegami Model calculations

The MT location and the distance between the BS and the MT are variable, and are a result of the users uniform distribution over the coverage area.

For the link budget, the system calculates the MT receiver power based on the BS transmission power and in the path loss estimation given by the propagation model algorithm. In other to have for each user an accurate power control, the simulator performs power control calculations in the DL in a predefined frequency, Figure 4. 3.

The parameters used for power control implementation are the received power compared with the corresponding service sensitivity and the SIR compared with the respective service E_b/N_o threshold.

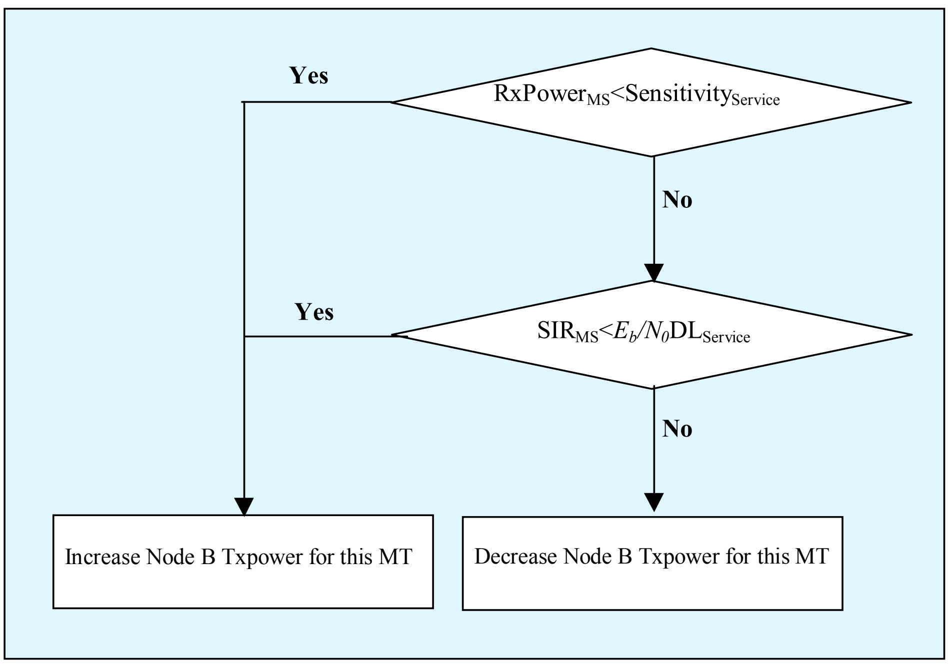

Figure 4. 3 – Power Control Algorithm.

The power control implementation in the simulator considers two levels of power adjustment, configurable, deciding is each situation which adjustment to use based on the difference between the MT received power and the corresponding service sensitivity. If the difference is higher than the threshold, the system performs an adjustment in the MT transmission power with the maximum step configured, whether if it is lower the system controls the power with the lower step configured in the simulator, Figure 4. 4.

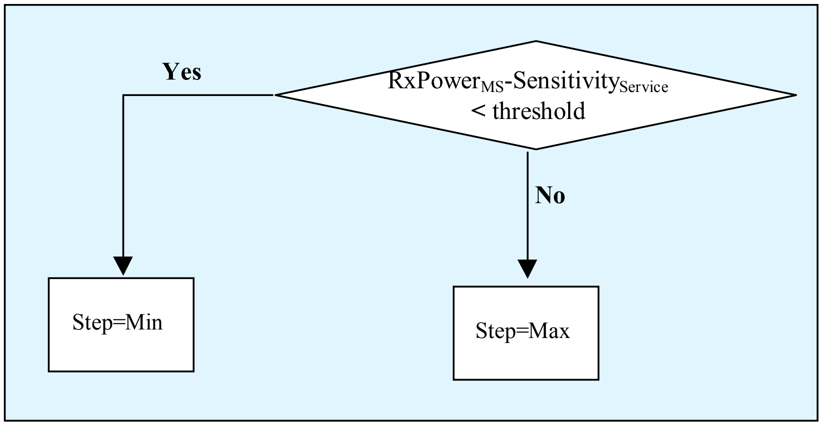

Figure 4. 4 – Power Control Step Algorithm

The simulator performs the link budget calculations using the parameters described in Section 3.1.5 for DL transmission. The simulator main input parameters for the link budget are the E_b/N_o, channel/system chip rate, user bit rate and total effective noise to perform the receiver power estimation calculation. Additionally, the system has also the possibility to configured transmission power, antenna gain in a given direction, receiver antenna gain, additional attenuation (cable, body and car loss) and the corresponding path loss given by the propagation model calculations.

The Node B configuration allows up to three sectors, being also possible to configure the corresponding orientation. The percentage of codes reserved for soft handover and the percentage of inter-cell interference coming from others cells are also included in the simulation tool. Additional Node B parameters correspond to codes reserved for signalling, height, frequency, maximum transmission power, maximum allowed load factor (the system controls this parameter considering blocking or delay respectively for CS and PS connections analysis).

4.4 Traffic Processing Engine

The simulator was implemented in the C++ language and the main output parameters are obtained in files for further analysis of results. The simulator has a simulation step of to 10 ms within which the new calls arrival, departure, and transmission conditions in the system are analysed.

The main functional blocks in the simulator, described in Figure 4. 5, correspond to the user profile generation, network analysis, random traffic generator and network performance analysis:

- *user profile generation* - the combination of the population density, user service types and penetration gives the input to the user distribution. Users are uniformly distributed over the coverage area according to the respective scenario considered.
- *network analysis* - calculates for each user/service the corresponding network traffic and propagation parameters. For each user location, there is a corresponding path loss (given by the propagation model) and consequently a different link budget. The link budget depends also on the active service, because each user can have more than one service associated. Although the simulator only considers a single cell, a percentage of resources for soft handover and inter-cell interference are also included in the tool.
- *random traffic generator* - generates for each services the call/packet start, its duration and finishing time. This traffic generation is specific for each service and is given by the corresponding statistical model. For CS, the parameters correspond to the call start, duration and end. For PS, there are more parameters involved that are related with the information volume transmitted, the number of sessions used, the number of packets in each session, the interarrival time between packets and packet size. Depending on the service type, the transmission rate can also vary during the transmission.
- *network performance analysis* - this block is responsible for the presentation of results and gives the main parameters for the planing decision. The most important results are related with blocking for CS or delay for PS. The system considers blocking/delay when the load factor is higher than a maximum threshold

configured in the simulator, by either lack of codes or power. The results also include the traffic associated to each active service, the corresponding service blocking percentage, the service packets/call duration and the evolution of the network resources during the evaluation process. The number of active sessions for PS and CS is also available.

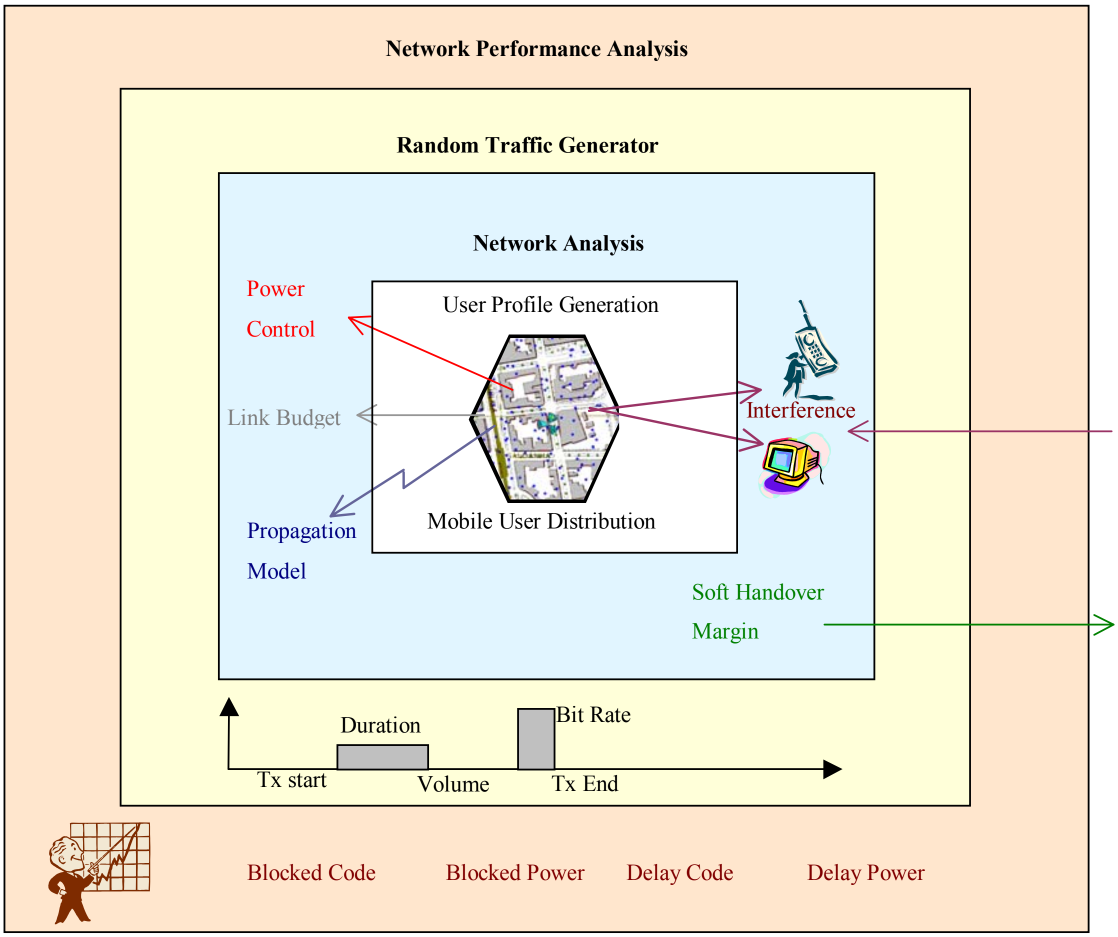

Figure 4. 5 – Simulator Overview

Each 10 ms (simulation step), the simulator analyses for all the users and active services if there is a new call to start or packets to be transmitted. If there is a new transmission the system analyses the resources require for this user/service and compares then with the available resources. If there are no resources, the system increments the blocking in the system due to power or codes and refuses the call if it is a CS situation, or delays the packets if it is a PS transmission. The delay is randomly generated within minimum and maximum values configured. Each active call (CS or PS) has a predefined duration time that is checked every 10 ms. When the end of the call is reached, the system release the resources allocated to the user/service.

For the voice transmission, the simulator considers that the call commutes among the four states, each one with a different bit rate. In a congestion condition, it may happen that the call is dropped because the network has resources for a given state but an increase in the bit rate is not support by the available resources. The Email and File Download service consider a variable bit rate during the transmission, however the bit rate is assumed constant in each transmission session. For the other services, the bit rate is constant along the simulation time, Figure 4. 6.

The system has 230 channels available for DL transmission. The remaining channels are used for soft handover (30% of the total number of channels in the DL) and for signalling (25% of the total number of channels in the DL).

The number of channels allocated for each service is given by the following equation where *SF* corresponds to the spreading factor.

$$ChannelAllocation_{per\ service} = \frac{ChannelsDownlink}{SF} \qquad \textbf{(4. 1)}$$

The *SF* for each service corresponds to the relation between 7680 (512 codes x 15 secondary codes) and the respective radio channel bit rate.

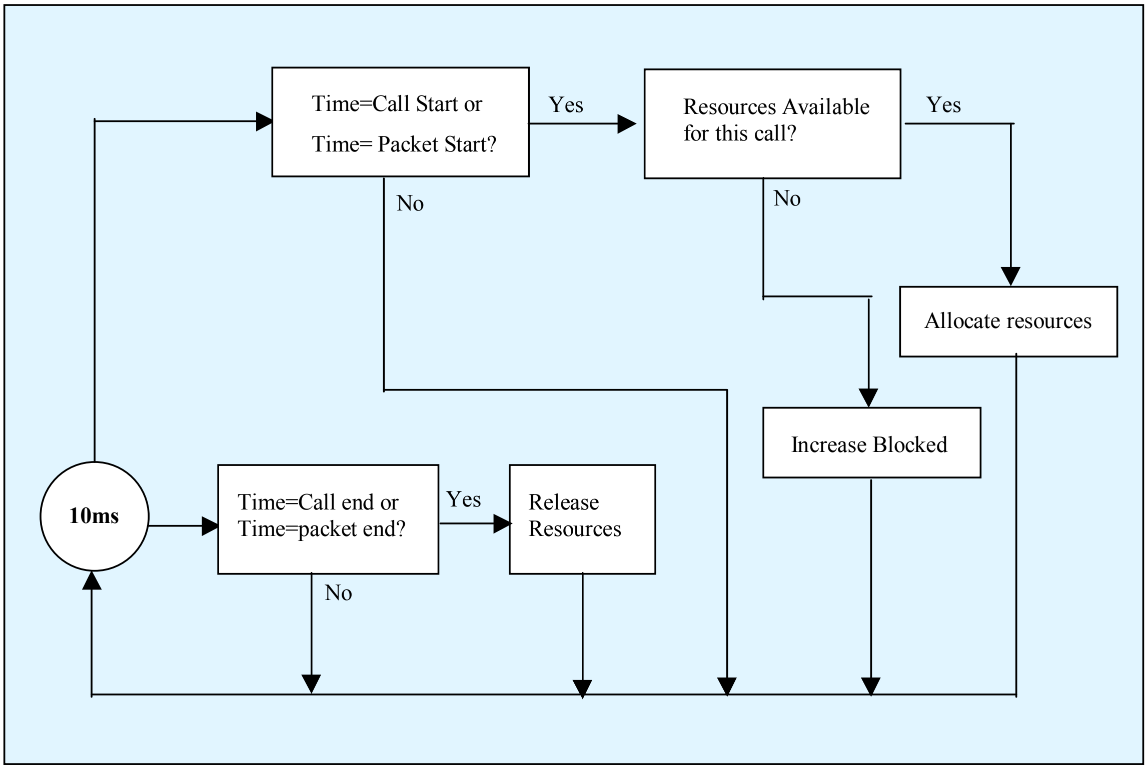

Figure 4. 6 – Simulator Resources Allocation.

4.5 Input Parameters

One of the most important functionalities of the simulator is the possibility to set a large number of parameters for each simulation, hence being possible to manage a significant number of variables that have implications in the system behaviour. The following tables' present the main input parameters of the simulator and the blocks that are related to them, as well as a brief description of each one.

Table 4. 9 describe various services and the input parameters used for their characterisation.

Table 4. 9 – Service Configuration Parameters.

Parameter		Description
Service Type	Speech	Traditional speech-telephony
	Video Streaming	Communication with voice and video
	Streaming Multimedia	Visualisation of multimedia documents (ex: video, music)
	Web Browsing	Exchange of data between a user and a web server.
	Location Based Services	Find location based information
	MMS	Messaging that allows text, image and video
	Email	Usually in text form but can included image and video
	File Download	Download a file from a database.
Bit Rate [kbps]		Downlink Transmission Rate for each service
E_b/N_o [dB]		Signal to Noise ratio in the downlink for each service
BHCA		Busy Hour Call Attempt for each service
Penetration [%]		Percentage of penetration for each service
Duration [s]		Average duration for each CS call

For the propagation model the parameters that are used for path loss estimation are configurable as described in Table 4. 10.

Table 4. 10 - Propagation Model Configuration Parameters.

Parameter		Description
Urban Type	Urban; SubUrban	Walfish Ikegami model classification
Building	Height, Separation [m]	Typical values
Street	Width [m]	Typical values
Frequency	Freq. [GHz]	UMTS frequency range
MT height	UE Height[m]	Typical value human height
Average Street orientation	Angle [º]	Average street orientation
Additional Loss	Penetration Loss [dB]	Typical values
	In Body Loss [dB]	Typical values
	BS Cable Loss [dB]	Typical values

To perform the receiver power estimation algorithm, several input parameters can be configured as described in Table 4. 11.

Table 4. 11 - Link Budget Configuration Parameters.

Parameter	Description
Thermal Noise Density [dB/Hz]	Typically –174 =$k.T$ where k is the Boltzmann constant $1.381 \cdot 10^{-23}$ J/K and T the temperature in Kelvin
Receiver Noise Figure [dB]	Typical values between 5-9 dB
Receiver Noise Density [dBm/Hz]	Thermal Noise Density [dB/Hz] + Mobile Station Receiver Noise Figure [dB]
Receiver Noise Power [dBm]	Receiver Noise Density [dBm/Hz]+10.log(384000)
Interference Margin [dB]	Margin between noise and interference
Receiver Interference Power [dBm]	Estimated interference power
Total Effective Noise+ Interf. [dBm]	Sum of interference and noise power in the receiver
MT Antenna Gain [dBi]	Antenna Gain of the MT
Node B Noise Figure [dB]	Characteristic of the equipment, internal noise, attenuators
Eb/No [dB]	Dependent of each service
Node B Antenna Gain [dBi]	In a given direction
Fading Margin [dB]	Typical values
Soft Handover Margin [%]	Codes reserved for soft handover
Additional Loss [dB]	Penetration, in body loss, BS cable loss
Path Loss [dB]	Given by the propagation model

Table 4. 12 indicate the parameters to be established for the MT.

Table 4. 12 – Mobile Configuration Parameters.

Parameter	Description
Population density	People/km, Uniform Distribution
Mobile Antenna Gain [dBi]	Typical values

For the Node B configuration, and the respective sectors, the configuring parameters are indicated in Table 4. 13.

Table 4. 13 – Node B Configuration Parameters.

Parameter	Description
Nº of Sectors	[1, 2, 3]
Ortogonality	α [0…1]
Sinalisation Codes [%]	Number
Soft Handover Codes [%]	Percentage of codes for soft handover
Node B Height [m]	Typical values
Frequency [GHz]	System operating frequency
Node B Location	Node B position in the cell
Tx Power Max [W]	Maximum Node B transmission power
Tx Power Max/user [W]	Maximum Node B transmission power for a single user
Inter Interference [%]	Percentage of codes reserved for intracell interference
Max Load Factor	Value over than blocking or delay start to occur
Antenna Height [m]	Typical values
Antenna Radius [m]	Typical values
Power Control [pc/s]	Number of times that the power control is executed per 10ms.
Power Control Step Min [%]	Lower power step adjustment
Power Control Step Max [%]	Higher power step adjustment
Power Control Step Threshold [dB]	Power control decision margin.
Minimum random packet delay [s]	Minimum delay for PS transmission
Maximum random packet delay [s]	Maximum delay for PS transmission

4.6 Output Parameters

The simulator generates text files that are analysed via a Matlab program to generate the plots. The analysis contemplates the general performance as well as a one per service. The simulator user has the possibility to choose the time interval to write in the text output files. In Figure 4. 7, the number of available codes in the system and the system load are presented in a time interval of 1 s. A description of the outputs of the simulator are given in what follows.

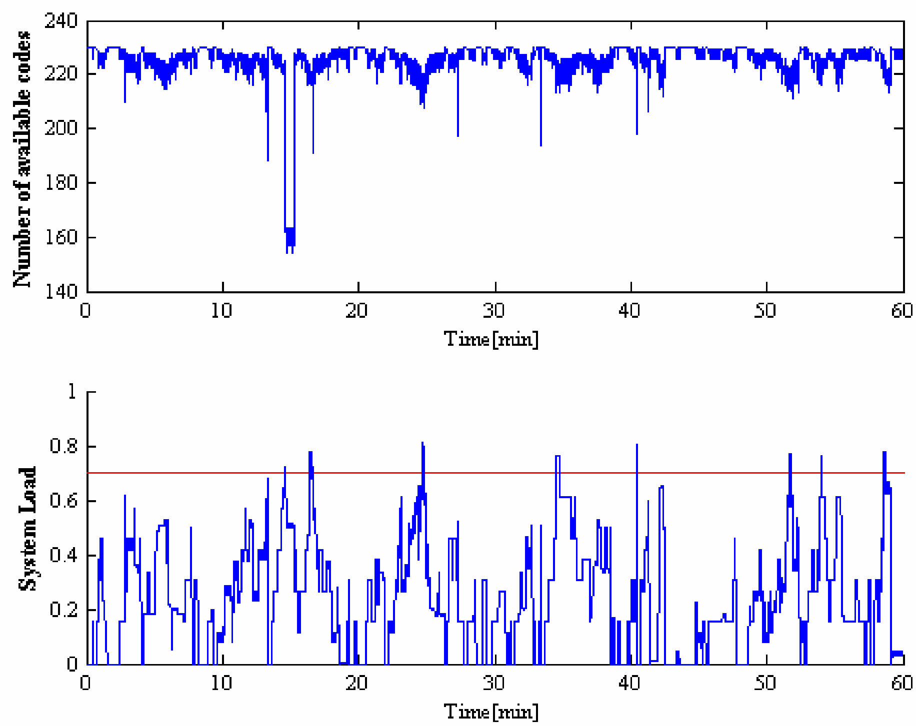

Figure 4. 7 - Number of Available Codes and System Load

The simulator evaluates for the call admission the blocking probability for CS and the delay probability for PS due to lack of power, capacity or code as presented in Figure 4. 8. The relation between the number of blocked code calls and the total number of calls gives the blocking code probability while between the number of blocked power calls and the total number of calls gives the blocking power probability. For PS admission, the delay code probability corresponds to the relation between the packets delay due to the lack of codes and the total of packets calls; the delay power probability corresponds to the relation between the packet delay due to the lack of power and the total of packet calls.

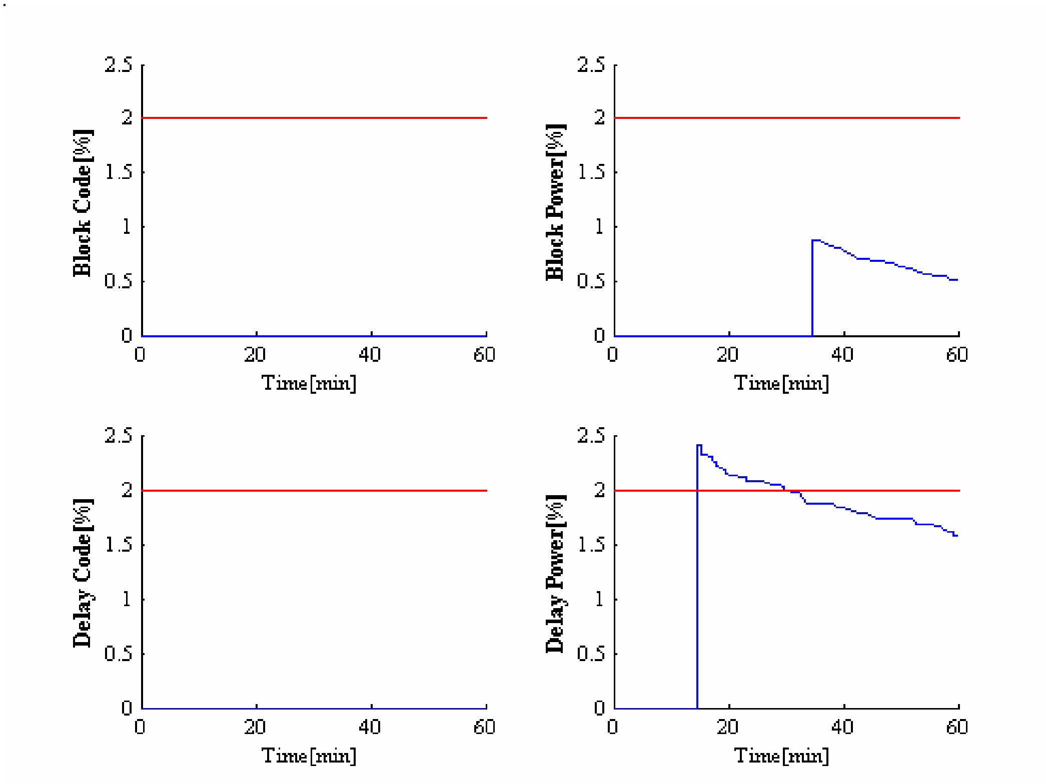

Figure 4. 8 – Block (CS) and Delay (PS) in System Arrival.

For each service type, the simulator gives the number of total calls corresponding to the sum of transmitted calls, the blocked power calls and the blocked code call for CS, while for PS the system gives the total number of packets that corresponds to the sum of total packets transmitted, total packets delayed due to the lack of codes and packets delayed due to the lack of power. Figure 4. 9 represents the total number of calls for CS services and the total number of packets for PS ones are represented in Figure 4. 10.

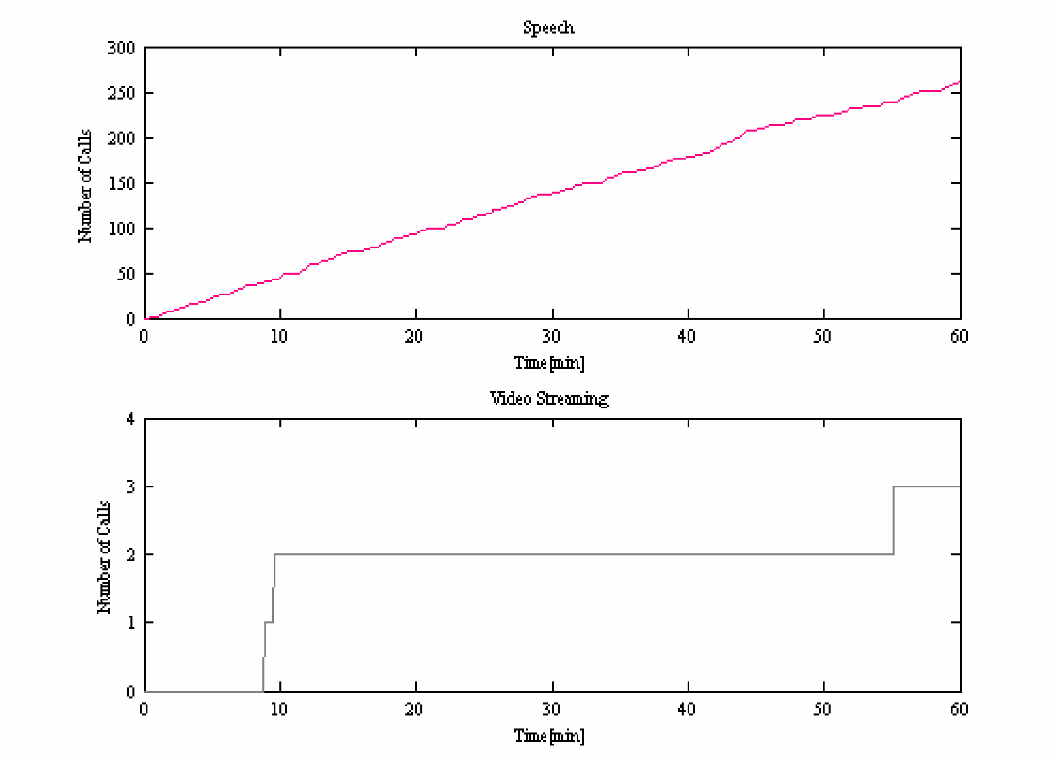

Figure 4. 9 – Total number of calls

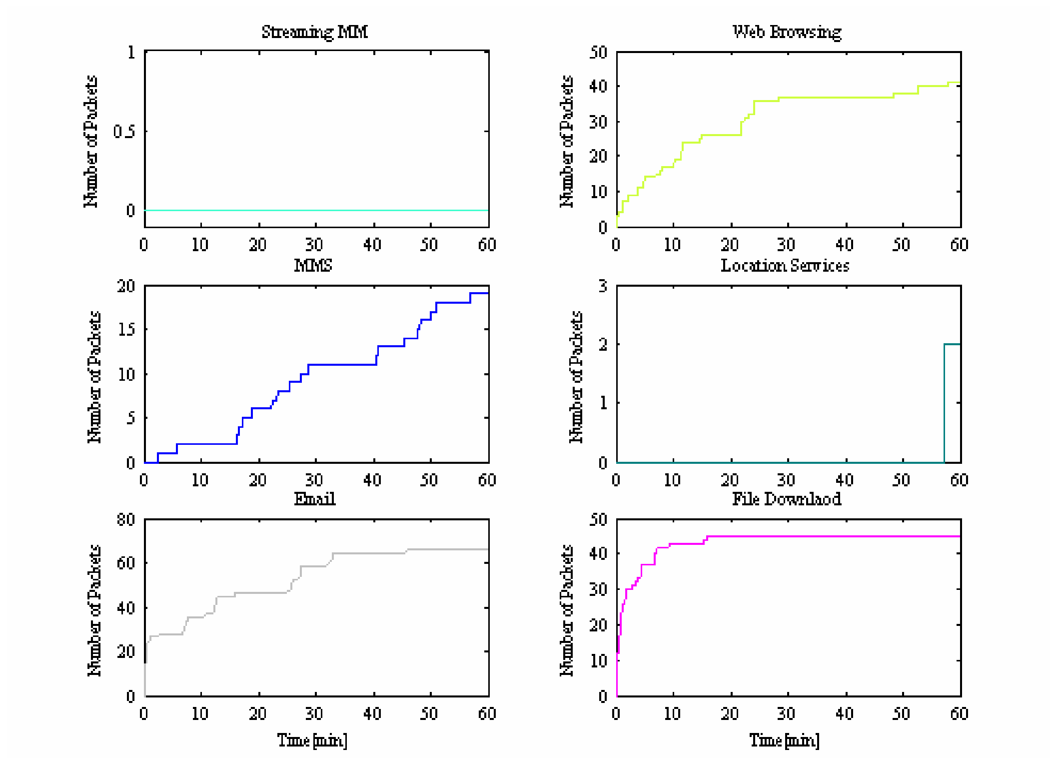

Figure 4. 10 – Total number of packets transmitted per service.

The total number of calls blocked in the system due to lack of power or code for each service is presented in Figure 4. 11 and Figure 4. 13. The total number of packets delayed are presented in Figure 4. 12 and Figure 4. 14.

Figure 4. 15 shows the contribution to the total system load coming from each service, for CS services and Figure 4. 16 for PS. The statistical behaviour of the traffic generator for PS can be can be analysed by the packet duration as represented in Figure 4. 17.

Figure 4. 18 and Figure 4. 19 present the comparison between the total connection transmitted until a given time and the corresponding blocked/delay in the system arrival and the blocked/delay for active connections.

The number of active sessions in a given instant, PS or CS, is presented in Figure 4. 20.

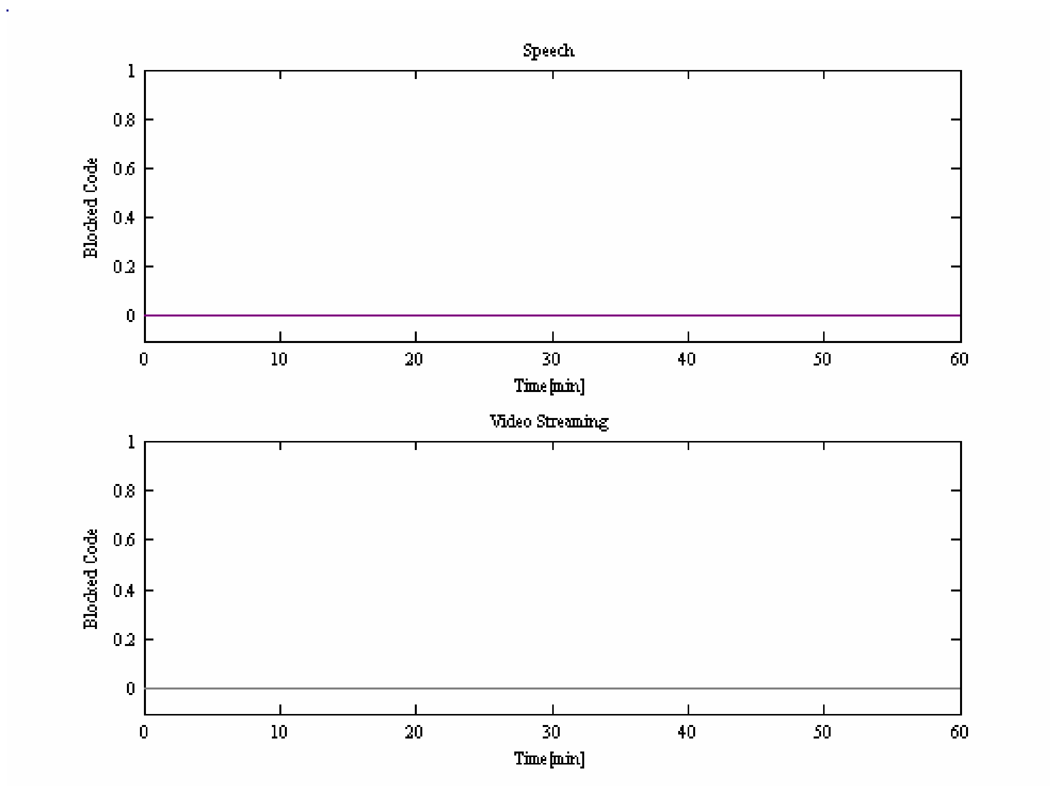

Figure 4. 11 – CS Blocked Code Calls

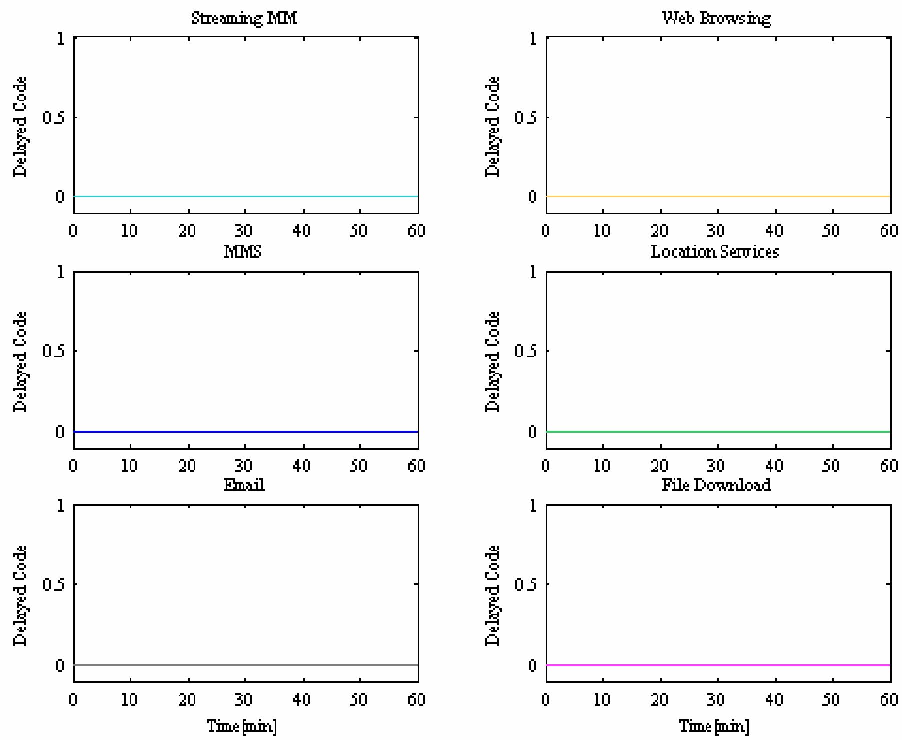

Figure 4. 12 – PS Delayed Code Calls

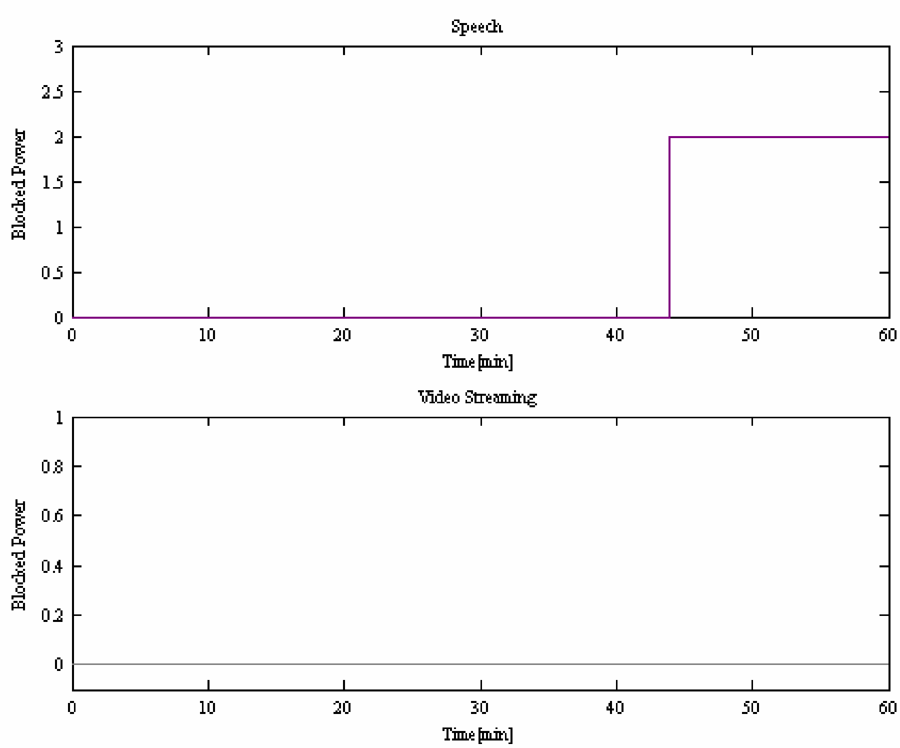

Figure 4. 13 – CS Blocked Power Calls

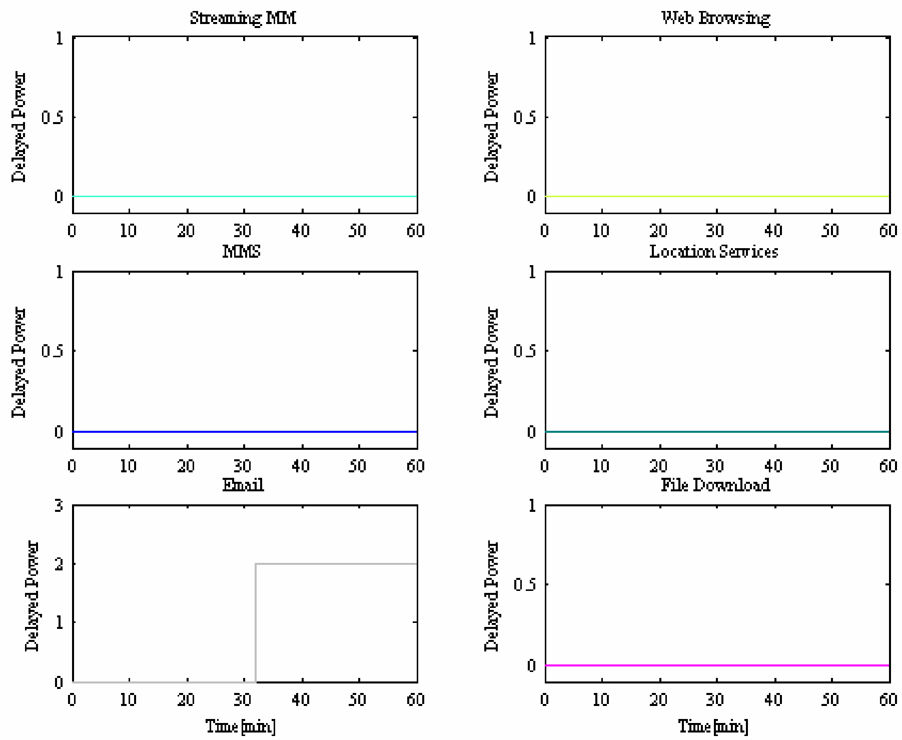

Figure 4. 14 – PS Delayed Power Calls

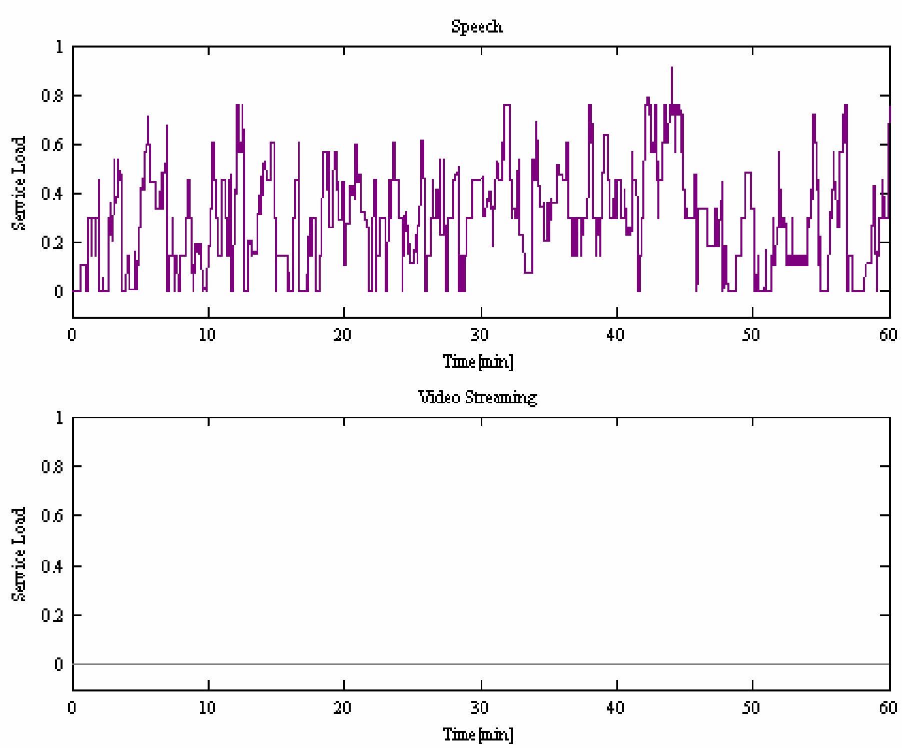

Figure 4. 15 – CS Service Load

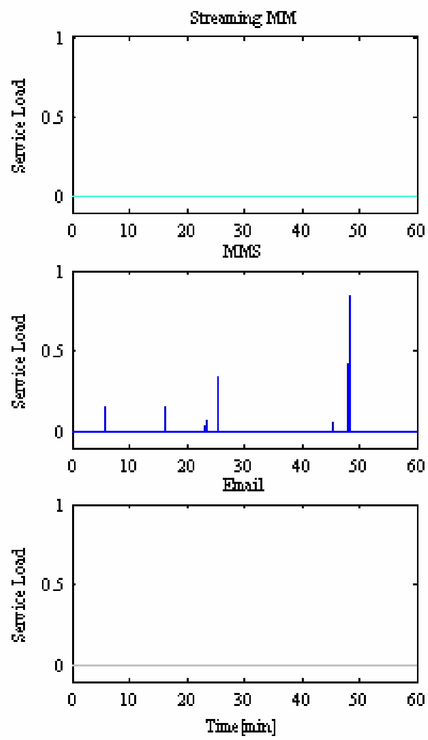

Figure 4. 16 – PS Service Load

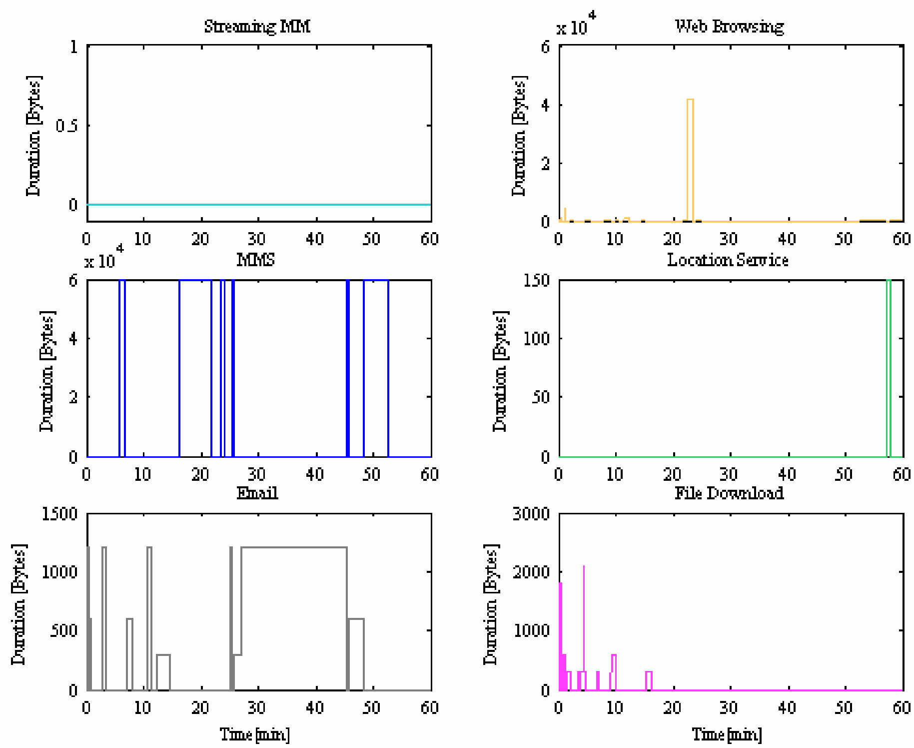

Figure 4. 17 – Packet Duration for each service

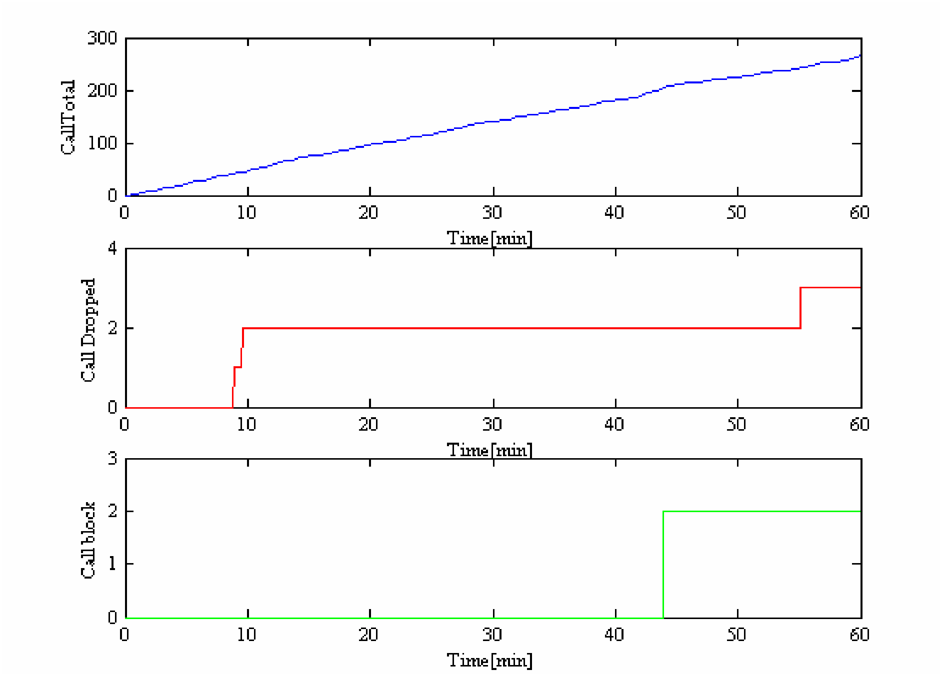

Figure 4. 18 – Call Block and Drop for CS.

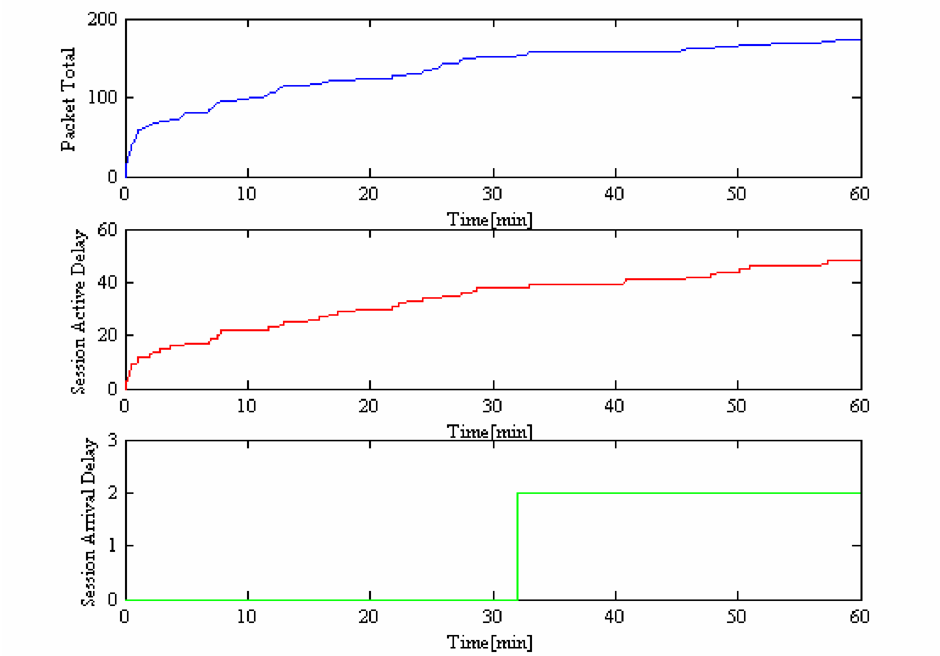

Figure 4. 19 – PS, Packet Total, Delay session active and delay session arrival.

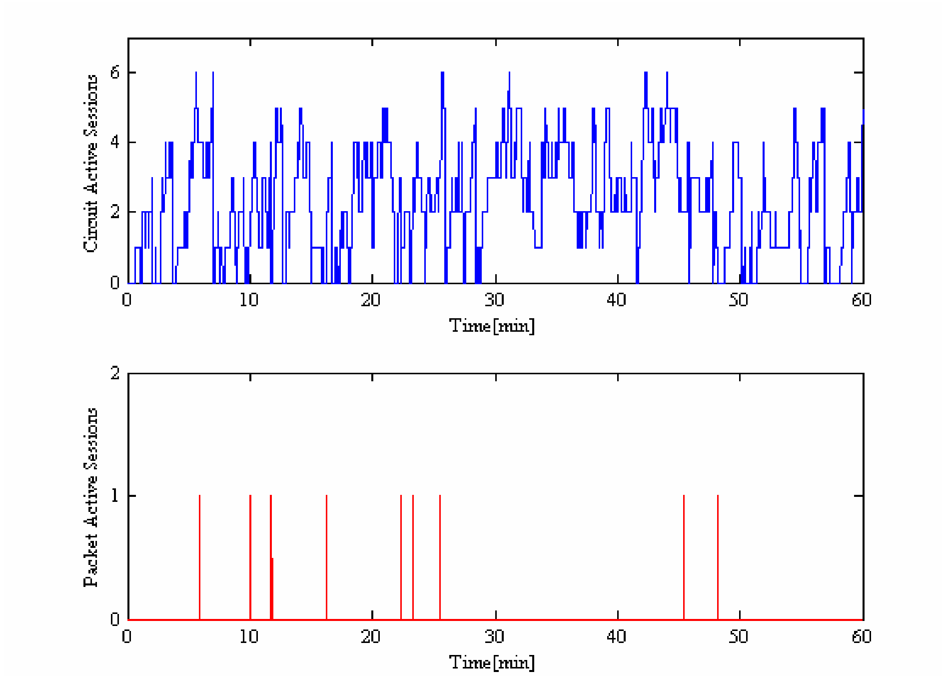

Figure 4. 20 – Active connection in each second.

5 ANALYSIS OF RESULTS

5.1 Scenarios Description

To characterise the diversity of usage pattern, three customer segments are considered as indicated in Table 5. 1 – Business, SOHO (Small Office/Home Office) and Mass-Market.

Table 5. 1 – Customer Segmentation [24].

Segment	**Description**
Business	Early adopters, with intensive and almost entirely professional use, primarily during office hours.
SOHO	Followers, with both professional and private use, during the day and in the evening.
Mass-Market	With low use, with flat traffic levels.

For the three customer segments considered, Business, SOHO and Mass-Market, Table 5. 2 shows the BHCA for each customer segment per service. The respective penetration percentages can be seen in Table 5. 3.

Table 5. 2 – BHCA per service and customer segment [24].

Service	Businnes	SOHO	Mass-Marke
Speech-Telephony	2.40	1.80	0.56
Video-Telephony	0.80	0.30	0.07
Streaming Multimedia	0.40	0.15	0.14
Web Browsing	1.80	1.35	0.63
Location Services	0.20	0.15	0.07
MMS	0.40	0.30	0.35
Email	2.00	1.20	0.21
File Download	1.00	0.75	0.07

Table 5. 3 – Percentage of call per day and per customer segment [24].

Service	Business [%]	SOHO [%]	Mass-Market [%]
Speech-Telephony	27	30	27
Video-Telephony	9	5	3
Streaming Multimedia	4	3	7
Web Browsing	20	23	30
Location Services	2	3	3
MMS	4	5	17
Email	22	20	10
File Download	11	13	3

Ten operational environment classes were characterised: railway, highway traffic jam, highway, main road, street, open, rural, sub-urban, urban and CBD (Central Business District) as described in Table 5. 4.

Table 5. 4 - Operational Environmental Classes [24].

Class	Environment
Railway	Railway.
Highway	Highway.
Highway with traffic jam	Traffic jam in a highway, corresponding to a lot of cars stopped, or moving at a very low speed.
Road	Main road of relatively high-speed users, typically inserted in suburban and rural areas.
Street	Street of low-speed users, typically inserted in an urban area.
Rural	Rural area, with low building and high vegetation density; Area with low population density, mainly of residential and primary sector population; Little commerce.
Sub-urban	Sub-urban area with medium building and vegetation densities; Area with medium population density, mainly of residential and secondary sector population; Little commerce.
Open	Small pedestrian land area (square, open area, park, large pedestrian areas along streets) surrounded by mean urban, dense urban, or residential areas.
Urban	Area with high building density and low vegetation density; Area with high population density, mainly of tertiary sector with some residential population.
Central Business District (CBD)	Area with very high building density, very high buildings, with almost no vegetation. Area with very high population density, with tertiary sector population.

Table 5. 5 characterises the two possible service combinations, where Model II characterises the active services in highway and railways while Model I is associated to the remaining operational environment classes.

Table 5. 5 – Active Services for Simulation [24].

	Speech	Video	Streaming	Web Browse	Location Services	MMS	Email	File Download
Model I	✓	✓	✓	✓	✓	✓	✓	✓
Model II	✓				✓	✓	✓	

5.2 Default Input Parameters

Although it would be interesting to analyse the influence of the various input parameters in system performance, the limited time frame to develop this work implied the characterisation of each one considering only the typical values. Only the scenario (that characterises the user density per service and the respective utilisation), the population density and the cell radius in some simulations was varied. Table 5. 6 presents the default values used for the propagation model characterisation.

Table 5. 6– Propagation Model Default Parameters [18].

Description	Parameter	Default Values
Urban Type	Urban, Suburban	Urban
Building	Height [m]	20
	Separation [m]	60
Street	Width [m]	30
Frequency	Freq. [GHz]	2
Mobile height	UE Height [m]	1.5
Average Street orientation	Angle [º]	10
Additional Loss	Penetration Loss [dB]	10
	In Body Loss [dB]	3
	BS Cable Loss [dB]	0

Table 5. 7 presents the link budget default values used for the link budget calculation based on the link budget calculation presented in [7]. As previously mentioned a fixed percentage of the capacity for soft handover is used corresponding to 30% of the total capacity.

Table 5. 7 – Link Budget Default Parameters [7].

Description	Unit	Default Values
Thermal Noise Density	[dB/Hz]	-174
Receiver Noise Figure	[dB]	5
Interference Margin	[dB]	3
Max Mobile Tx Power	[W]	0.125
MT Antenna Gain	[dBi]	0
Node B Noise Figure	[dB]	5
Interference Margin	[%]	30
Eb/No	[dB]	see services
Node B Antenna Gain	[dBi]	5
Fading Margin	[dB]	0
Soft Handover Margin	[%]	30
Additional Loss	Pen., Body, Cable [dB]	10+3+2
Path Loss	[dB]	user location

In Table 5. 8, the Node B characterisation parameters are presented, considering the default UMTS values. For system analysis purposes, a cell radius of 900 m is considered.

The MT characterisation depends greatly on the user distribution over the simulation area, the selected service types, and the propagation conditions associated to the respective location. The population density, in (pop/km^2), is varied according to the scenarios.

Table 5. 8 – Node B Default Parameters

Description	Unit	Default Values
Nº of Sectors	-	3
Ortogonality	-	0.5
Sinalisation Codes	-	128
Soft Handover Codes	[%]	30
Node B Height	[m]	22
Frequency	[GHz]	2
Node B Location	[m]	[511, 509]
Tx Power Max	[W]	10
Tx Power Max/user	[W]	5
Inter Interference	[%]	30
Max Load Factor	-	0.7
Antenna Height	[m]	22
Antenna Radius	[m]	900
Power Control	[/s]	500
Power Control Step Min	[%]	10
Power Control Step Max	[%]	50
Power Control Step Threshold	[dB]	5
Minimum random packet delay	[s]	3
Maximum random packet delay	[s]	10

5.3 Scenario Analysis Considerations

One of the most limiting factors in the scenario analysis is the simulation time required to obtain the results. Although being aware that more accurate results would be obtained with more samples, the simulation time for each simulation, 1hour in average in a PIV 2.66GHz computer with 768Mbyte of RAM, corresponds to a limiting factor for the analysis. For this reason a compromise solution of 10 simulations was chosen.

The main performance parameter that was analysed for the scenario validation was the blocking or delay probability in the system access. Which has two meanings depending on the call type: CS or PS. For the CS calls (speech and video) the system tests if there are any resources available, and if not it blocks or drops the connection. For PS calls (Streaming

Multimedia, Web Browsing, Location Based Services, MMS, Email and File Download), the system also analysis the resources requirements, but in this case if it has no possibility to accept the connection, it delays the packets according to the profile set up in the system.

For the call admission, the blocking probability and the delay probability are equivalent. The system does not implement different policies for the active connection or for the admitted ones, the available resources being equally shared by already the active users and by the newly admitted ones in a given frame (10 ms). For the new admitted calls, CS or PS, the system allocates a fix value of power that is adjusted afterwards by the power control algorithm according to the user needs. For PS, the system never drop a connection, rather delays the call until the system has the available resources to comply with the requirements. Considering this implementation, it is expected that the percentage of packets for an active connection that have to be delayed is greatly higher than the percentage of delay in the call arrival situation. For a given packet call, it is expected in a real system that the number of failed attempt is limited to a maximum number. This configuration avoids that the user continually tries to conclude the transmission without success. This is a point that could be interesting to implement in a future work.

The simulator perform calculations for the blocking calls as well as for the delayed ones both for the system arrival and for the active connections (drop for CS and delay for the PS). The available resources, power, capacity and codes, are analysed separately and for each system the limiting factor is presented.

Power control implementation is one of the most important features in UMTS due to the limitation by the power sharing among the users. The power control implemented in the simulator assumes for call admission a fixed power level equal for all the users. Then, for a predefined frequency, the system performs power control adjusting the DL power level according to the service type, localisation and propagation conditions. Certain users with higher power level requirements and/or far from the Node B never have the possibility to transmit, due to the system limitation condition, although they are admitted in the system related with the initial power level allocation (initial power level minor than the effective needs).

The sensitivity depends on the service type, and for that reason it is expected that for the same propagation conditions, received power and interference, the system can comply with a service requirement and can not fulfil others. Figure 5. 1 presents this relation for the various services. As it can be observed, the speech service has the less limiting sensitivity requirement (given by the respective Bit Rate and E_b/N_0) being expected that this service could reach higher limits (distance, interference).

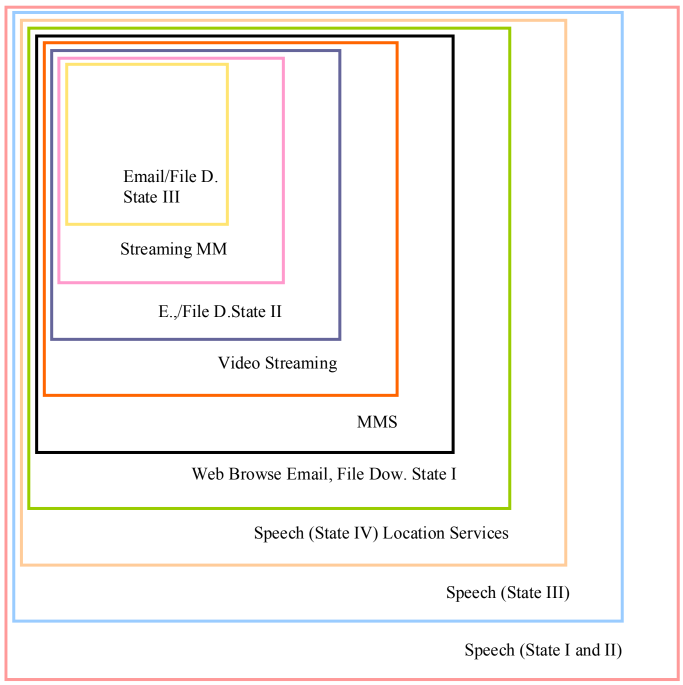

Figure 5. 1 – Relation between the sensitivity of the various services.

5.4 Model I Scenarios

In this Section the population density variation combined with Model I (all services active) is analysed. Table 5. 9 present typical values for the users distribution in the system for each service considering the Business scenario. Note that the user distribution in the system comes from a statistical, hence, these values vary according to the simulation.

Table 5. 9 – Model I, Business Scenario. Number of Users per Service.

Pop Dens [pop/km^2]	Speech [users]	Video S. [users]	Stream. [users]	Web Br. [users]	Location [users]	MMS [users]	Email [users]	File Dw. [users]
100	10	6	2	13	1	1	15	7
200	24	13	5	18	1	6	17	19
300	57	12	10	34	1	5	27	14
400	59	11	7	46	4	10	56	20
500	81	26	10	58	3	14	71	38
600	95	33	14	74	8	12	76	50
700	111	60	17	94	12	12	89	43
800	121	37	26	100	6	13	85	45
900	136	48	30	110	8	23	124	65
1000	155	44	18	116	16	26	126	62
1100	158	47	23	104	8	22	124	68
1200	162	69	33	121	9	28	147	76
1300	195	64	30	147	9	29	162	74

This scenario, as described in Section 5.1, characterises applications where high values of service penetration (given by the BHCA values) are expected.

Figure 5. 2 present Blocking/Delay probabilities due to new call admission into the system. The results correspond to a mean value, and the standard deviation; the confidence interval of plus or minus the standard deviation over the average is show in this figure; as well as in those that follows. As it can be observed, CS blocking probability and PS delay probability take approximated values. It is also possible to observe that the standard deviation value is proportionally high. Note that these values could be significantly improved by increasing the number of simulations.

As seen in Figure 5. 2 for the scenario considered, the blocking/delay probability reaches values higher than the acceptable 2% for a density of users higher than 1000 pop/km^2. Although these results can give the idea that the system has a good performance until this mentioned number of users is reached, by looking to the number of drop calls, presented in Figure 5. 3, one can see that this parameter is already significant for a density of users higher than 300 pop/km^2.

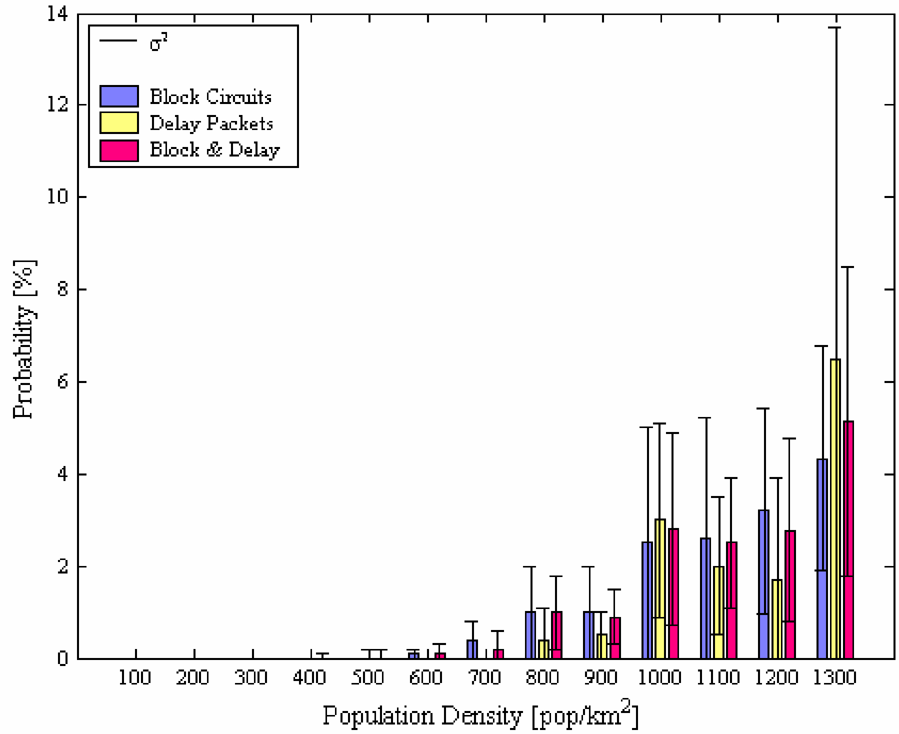

Figure 5. 2 – Model I Business Scenario, blocking and delay.

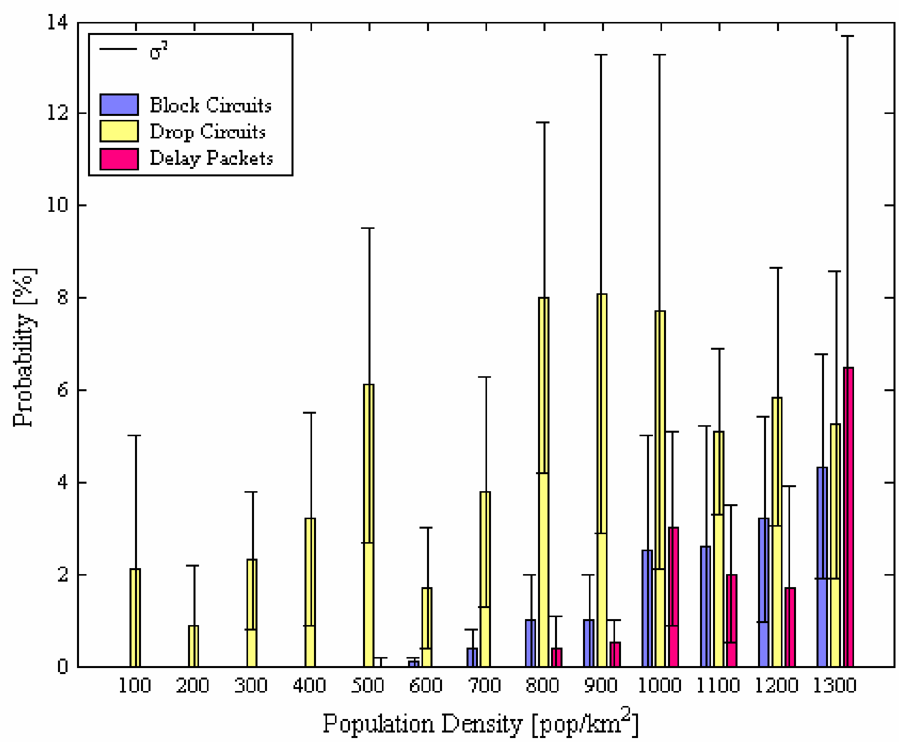

Figure 5. 3 – Model I Business Scenario, drop calls

The analysis of the limiting resources that are associated with this high blocking probabilities leads to the conclusion that the transmitted power is the limiting resource (for a given maximum load). In addition, this scenario is characterised by a high percentage of video streaming transmission that has associated a strict power requirement, as described in Section 5.3. For this reason, the allocated power at the call admission is significantly lower than the effective power needed for the service, resulting in a great number of power increase demand that has obviously implications in system performance, and results in a large number of dropped calls.

For PS, it can be observed in Figure 5. 4 the comparison between admitting a new packet call and a packet delay for the active sessions. Note that the active session have associated a great amount of packet, increasing significantly the delay probability if the propagation conditions do not change significantly. In other hand, the delay probability in the call admission is associated to only one packet being for that reason this probability significantly lower.

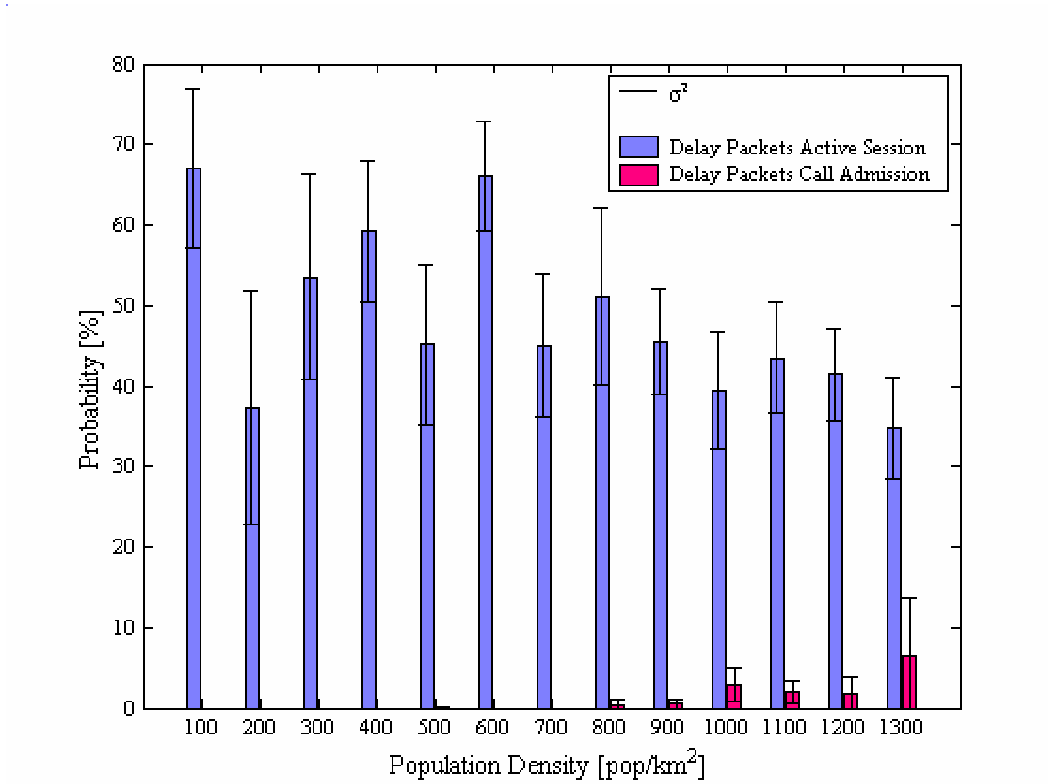

Figure 5. 4 - Delayed packets for call admission and for active sessions.

Taking into account that this scenario corresponds to the most demanding one in terms of resources, it must be expected that for a lower cell radius the results would improved. In fact, from the results obtained, it can be concluded that there are many users transmitting in the cell limits, and that better results are obtained for a small cell.

Maintaining all parameters but considering lower cell range, 500 m for a population density between 700 and 1300 pop/km^2, it can be confirmed from Figure 5. 5 and Figure 5. 6 that the performance has improved, comparing with previous results where a 900 m cell radius has been considered. In this situation (cell radius equal to 500 m) the system is considered to have an acceptable quality until 1000 pop/km^2. This scenario is a typical micro-cell environment, where high bit rate and traffic demand are expected, being important to have for that reason a relatively small cell.

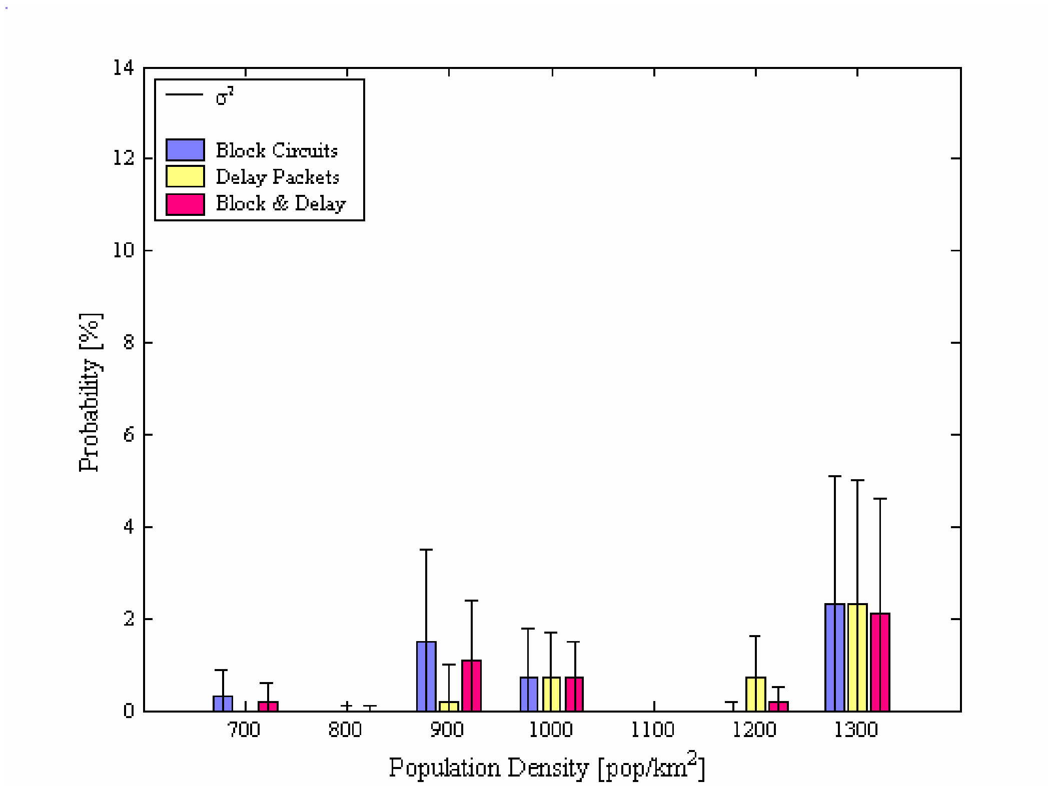

Figure 5. 5 – Model I Business Scenario, blocking and delay for a cell radius of 500 m.

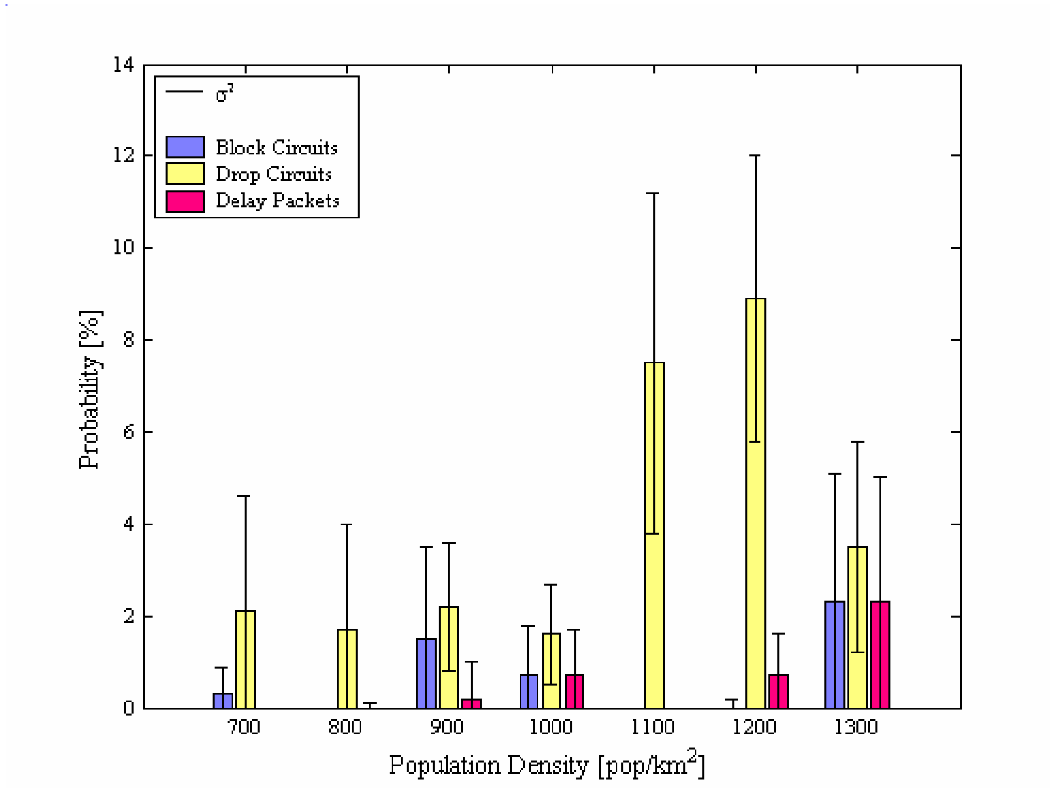

Figure 5. 6 – Model I Business Scenario, drop calls for a cell radius of 500 m.

As expected, by increasing the cell radius to 1300 m, the blocking/delay probabilities increase significantly, as it can be seen in Figure 5. 7 and Figure 5. 8. These results lead us to conclude that this scenario corresponds to a typical micro-cell environment, where even for a low population density, the power requirement for the various services is so demanding that it is necessary to have strict cell coverage.

Comparing the blocking probability obtained for the call admission with the dropping probability for the active connection, it can be concluded that the latter is higher, due to the initial allocated power being lower than the effective user needs.

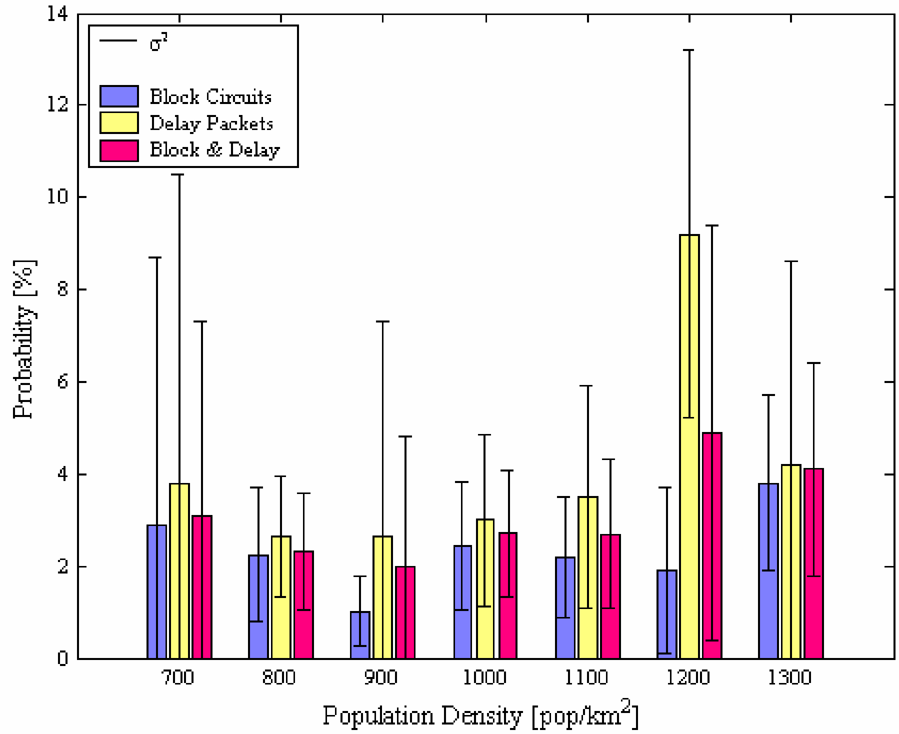

Figure 5. 7 – Model I Business Scenario, blocking and delay for a cell radius of 1300m

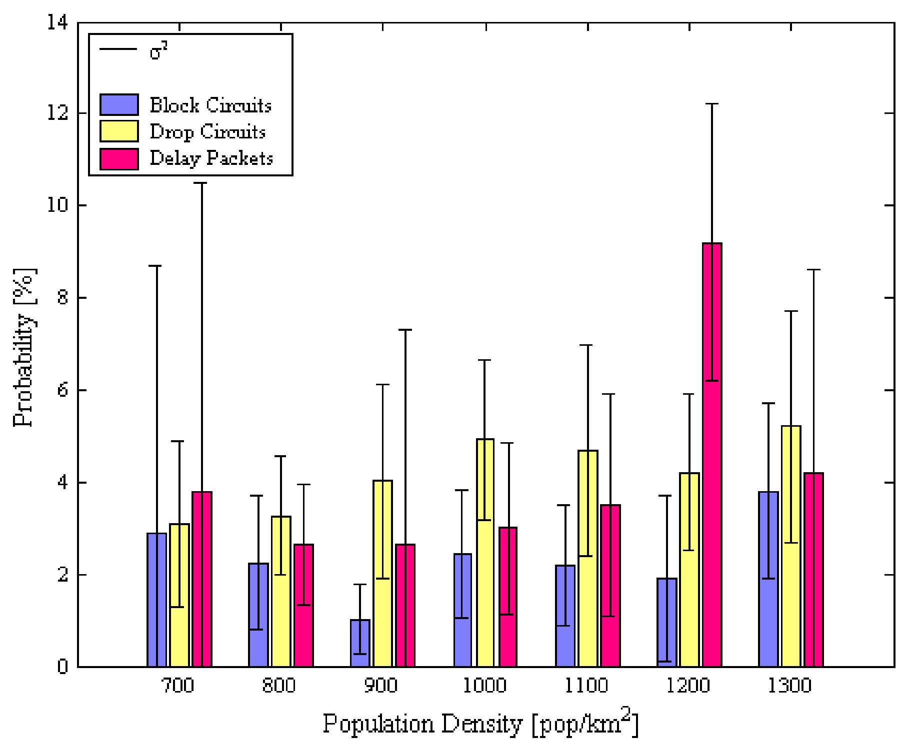

Figure 5. 8 – Model I Business Scenario, drop calls for a cell radius of 1300m

The Mass-Market scenario characterises a lower service utilisation and penetration, being expected that higher population densities can be allowed in the network.

Table 5. 10 – Model I, Mass-Market Scenario. Number of Users per Service.

Pop Dens [pop/km^2]	Speech [users]	Video S. [users]	Stream. [users]	Web Br. [users]	Location [users]	MMS [users]	Email [users]	File Dw. [users]
400	69	6	22	66	8	42	19	10
800	149	14	26	127	8	75	46	15
1200	195	27	40	187	22	101	76	27
1600	249	31	61	268	29	164	98	19
2000	286	32	82	338	34	208	115	47
2400	365	45	91	373	43	243	147	51
2800	432	54	115	508	42	275	155	38

In order to have an overall overview of the performance a higher simulation step for the population density was used. In Figure 5. 9 the blocking probability and delay probabilities in the call admission are presented. For the obtained results, it is possible to conclude that it is possible to guarantee the performance until 1200 pop/km^2.

Comparing the results obtained in the call admission with the values measured for the active drop calls Figure 5. 10, it is confirmed that 1200 pop/km^2 can be used for dimensioning. The cell radius used for this calculation (900 m) is more adjusted to this scenario than for the Business one. In fact, for this scenario high Node B coverage is expected, typical of a macro-cell environment.

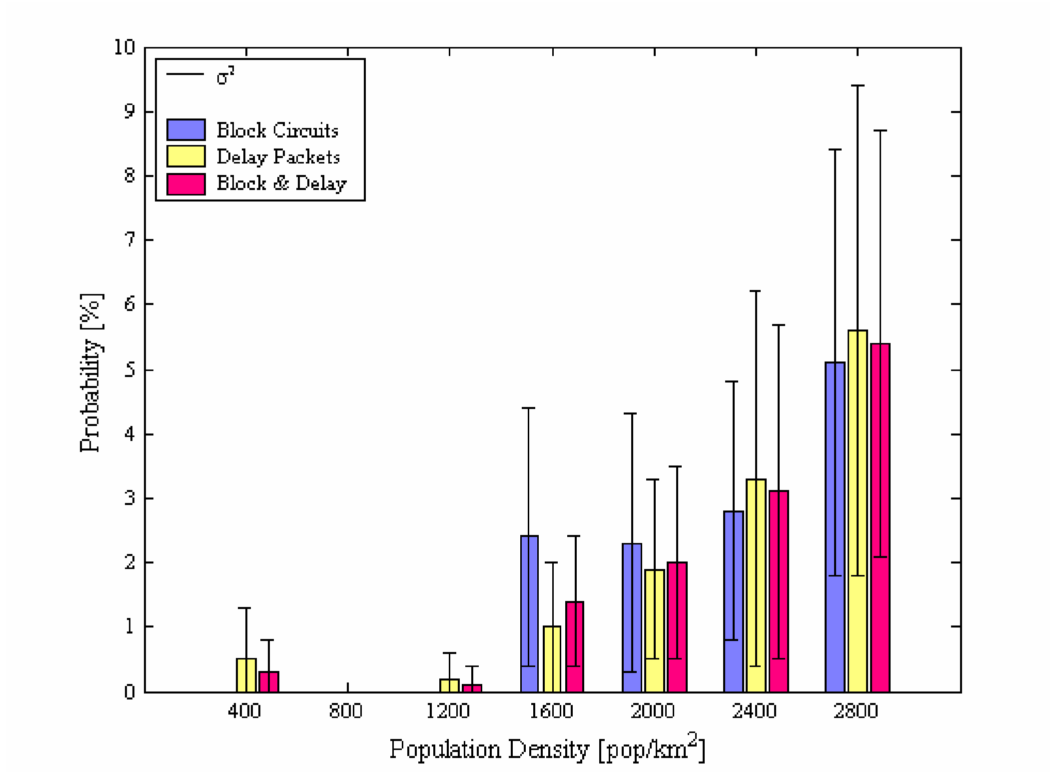

Figure 5. 9 – Model I Mass-Market Scenario, blocking and delay.

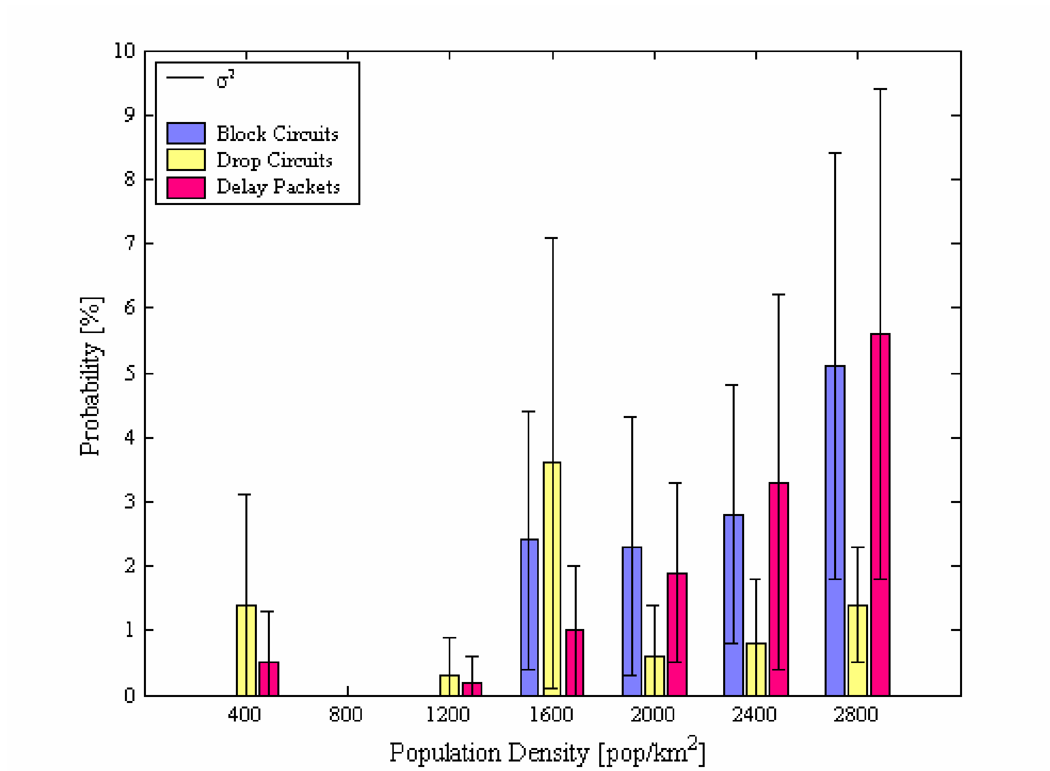

Figure 5. 10 – Model I Mass-Market Scenario, drop calls

The SOHO scenario corresponds to a medium step between the Business and the Mass-Market ones. In Table 5. 11 the respective numbers of user per service for the simulated population density are presented. Figure 5. 11 shows the results for blocking and delay, which are as expected.

Table 5. 11 – Model I, SOHO Scenario. Number of Users per Service

Pop Dens [pop/km^2]	Speech [users]	Video S. [users]	Stream. [users]	Web Br. [users]	Location [users]	MMS [users]	Email [users]	File Dw. [users]
100	17	6	1	12	1	2	14	5
300	54	5	6	40	5	13	47	28
500	91	12	11	61	13	12	64	38
700	122	27	13	92	12	21	73	46
900	151	34	15	119	17	31	90	63
1100	191	39	18	130	13	35	122	77
1300	206	28	31	177	25	31	155	79
1500	230	45	24	180	19	35	158	111

As it can be observed in Figure 5. 8, considering a blocking percentage higher than 2% is obtained for population density higher than 900 pop/km^2. However, considering the results presented in Figure 5. 12, a higher drop call probability is obtained. This indicator may lead to conclude that for this scenario dimensioning a small cell radius could improved significantly the results.

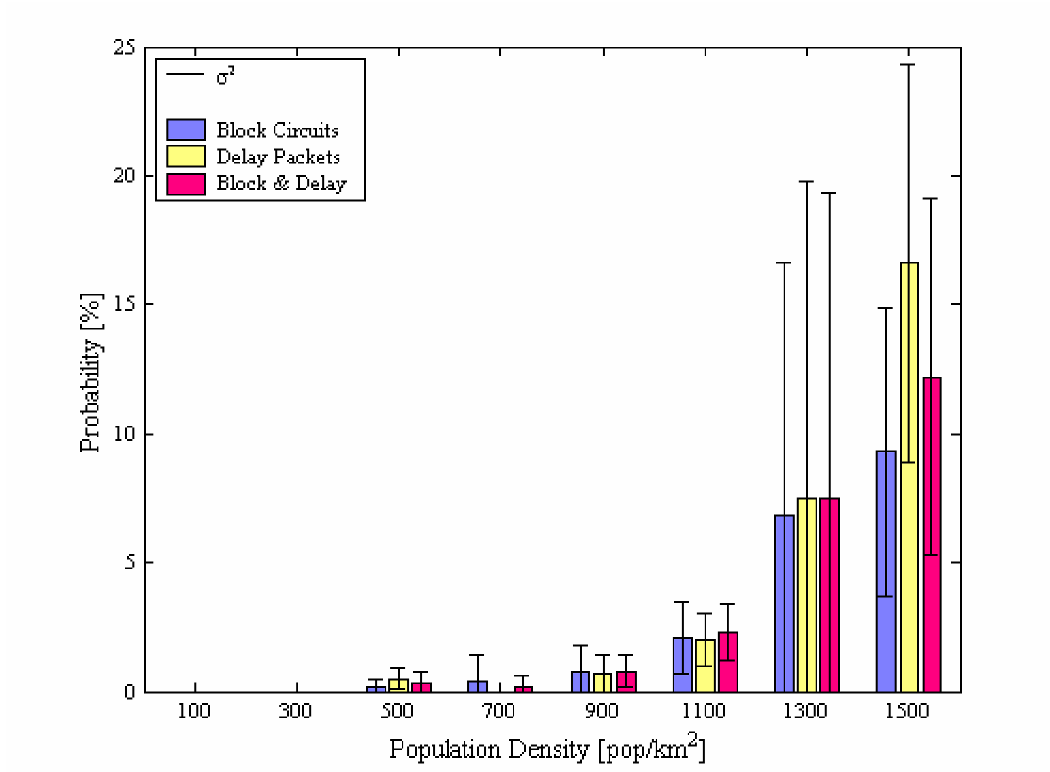

Figure 5. 11 – Model I SOHO Scenario, blocking and delay.

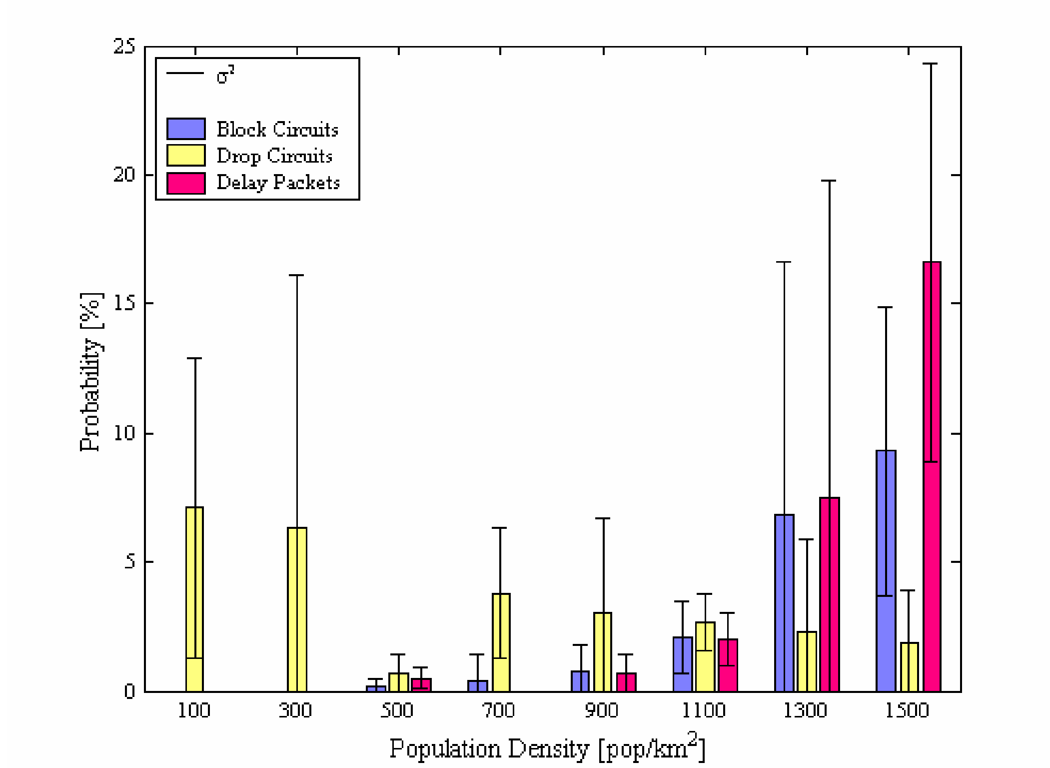

Figure 5. 12 – Model I SOHO Scenario, drop calls

5.5 Model II Scenarios

The Model II characterises a restricted number of services that have additionally associated to them a defferent sensitivity compared to the ones described in Model I. Table 5. 12 details the number of users per service in the simulated area considering the Model II Business scenario.

Table 5. 12 – Model II, Business Scenario. Number of Users per Service.

Pop Dens [pop/km^2]	Speech [users]	Location [users]	MMS [users]	Email [users]
500	73	4	8	67
750	108	8	12	90
1000	168	17	22	106
1250	192	6	21	151
1500	216	20	33	188
1750	278	16	37	228
2000	285	22	43	279

As it can be observed in Figure 5. 13 until a number of users equal to 750 pop/km^2 the predefined level of quality (2%) is obtained, this population density corresponding to a reference for this scenario implementation.

From Figure 5. 14, one can see that there are no drop calls in all the simulations. This is related with speech being the only CS service active, and as explained before, this service has, compared with the other services (particularly video), a better sensitivity. The admission control process, where a fixed amount of power is allocated for all services assumes that the necessary transmission conditions are meet to guarantee that the power adjustment is not significant for this service. These results lead to conclude that a higher coverage area, particularly in the implementation phase, could be attainable as a result of the higher services density considered for this scenario.

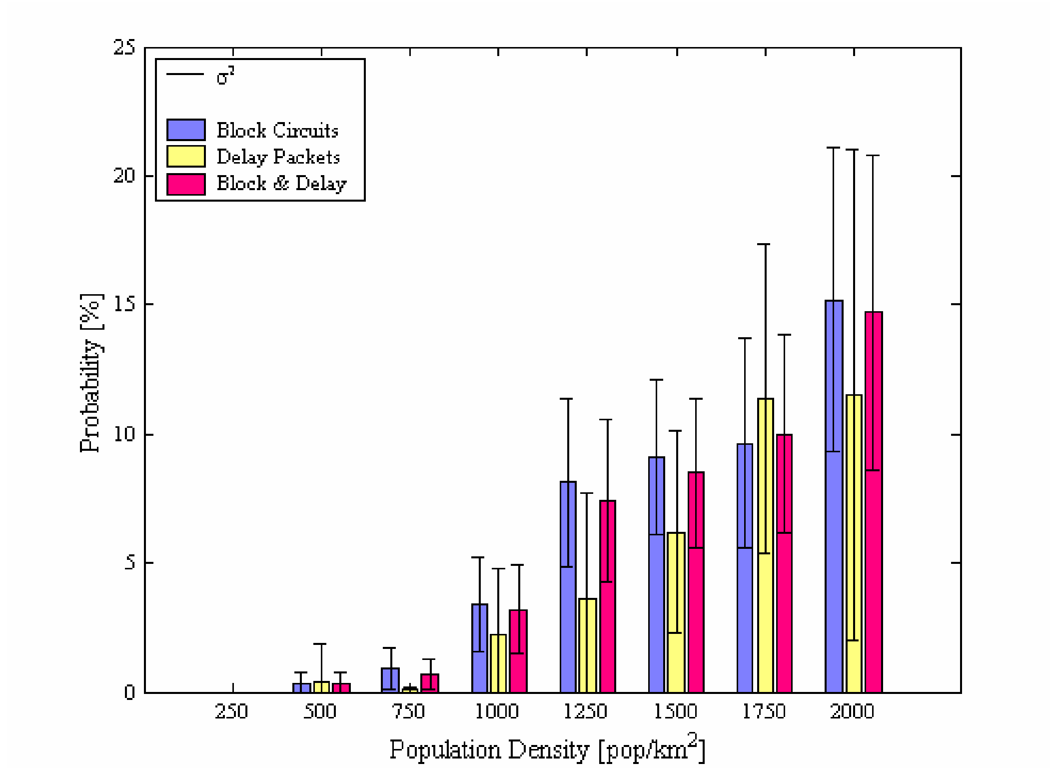

Figure 5. 13 – Model II Business Scenario, blocking and delay.

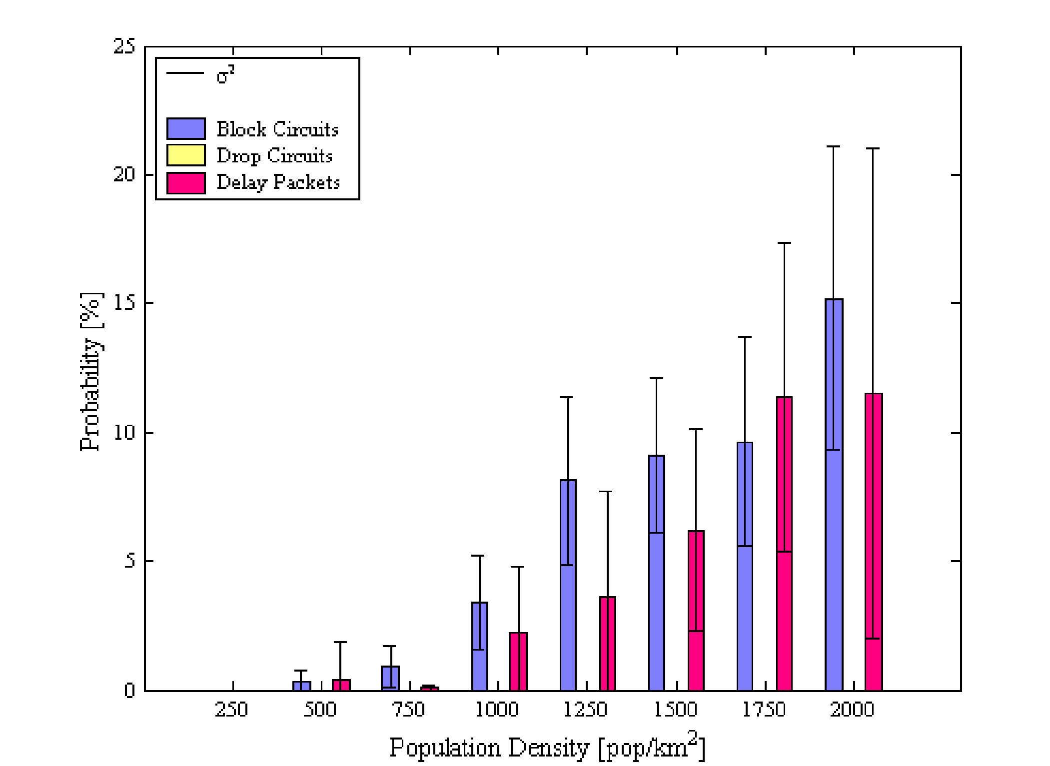

Figure 5. 14 – Model II Business Scenario, drop calls

The Model II Mass-Market scenario corresponds to the lower limiting one, being difficult to predict the maximum number of user that could be obtained. As can be seen in Table 5. 13 a step of 1000 pop/km^2 was considered for the analysis of system performance.

Table 5. 13 – Model II, Mass-Market Scenario. Number of Users per Service.

Pop. Dens. [pop/km^2]	Speech [users]	Location [users]	MMS [users]	Email [users]
1000	143	19	105	52
2000	302	30	202	111
3000	460	62	328	153
4000	622	57	370	222
5000	768	81	481	283
6000	908	100	557	344
7000	1046	122	678	374
8000	1260	146	767	448
9000	1361	147	899	545
10000	1503	165	944	590

From the results obtained for call admission, block calls and delay packets for PS, Figure 5. 15 and Figure 5. 16 the threshold probabilities are never achieved.

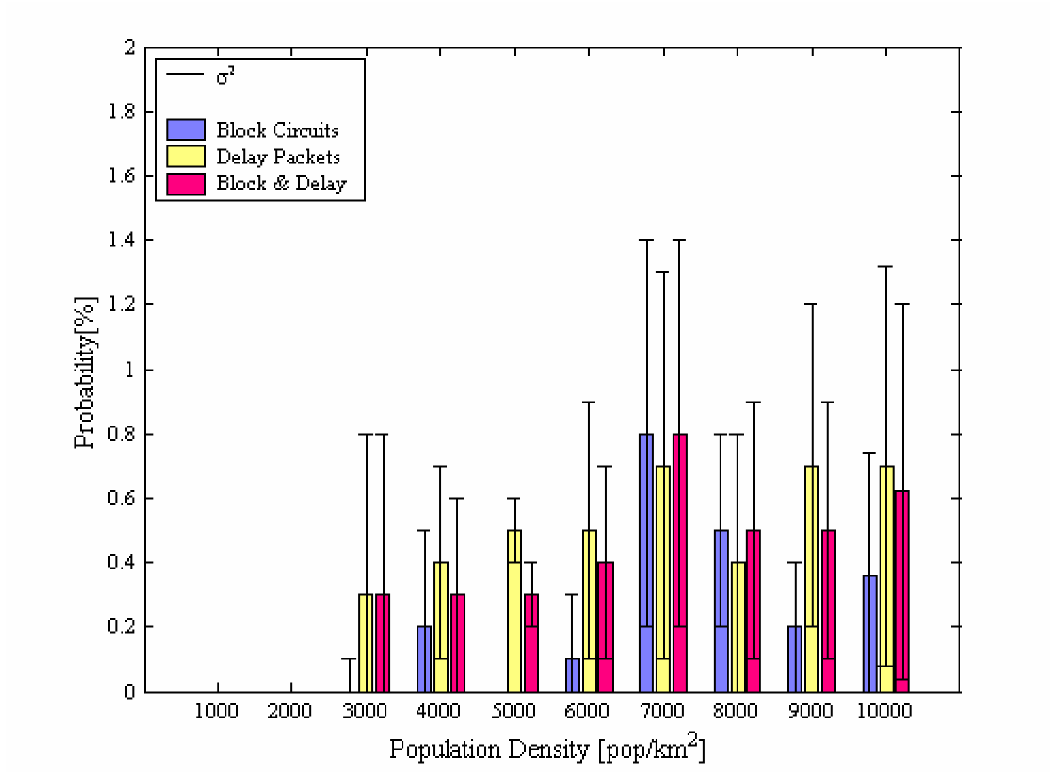

Figure 5. 15 – Model II Mass-Market Scenario, blocking and delay.

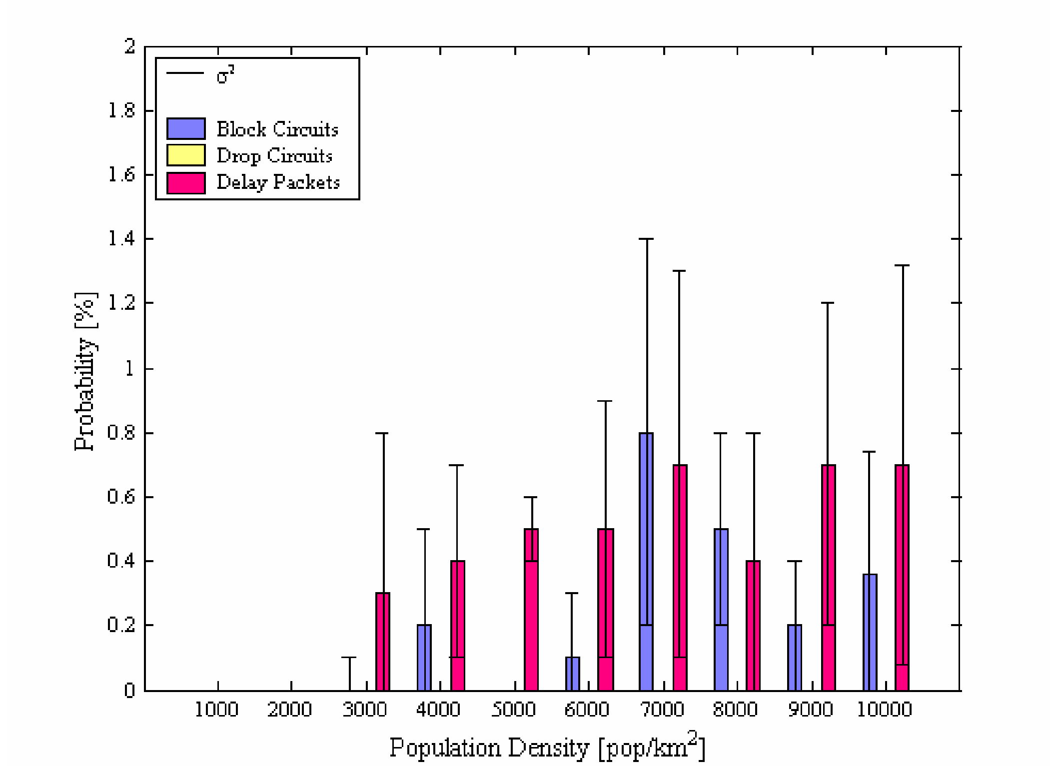

Figure 5. 16 – Model II Mass-Market Scenario, drop calls.

By analogy to the Model II Business scenario, there was no dropped calls in all the simulations.

For the Model II SOHO scenario, the respective users per service in the simulation area is presented in Table 5. 14.

Table 5. 14- Model II, SOHO Scenario, Number of users per service

Pop. Dens. [pop/km^2]	Speech [users]	Location [users]	MMS [users]	Email [users]
500	79	6	11	46
1000	170	17	31	116
1500	237	24	35	159
2000	329	30	53	208
2500	417	46	67	279
3000	514	49	87	303

From the results obtained, Figure 5. 17 and Figure 5. 18, it can be concluded that until 1000 pop/km^2 the system meats the quality requirements. Additionally, it can be expected that a higher area coverage could be attainable.

The number of dropped call is zero in all the simulations related with the preferential speech service transmission conditions, as it can be seen in Figure 5. 18.

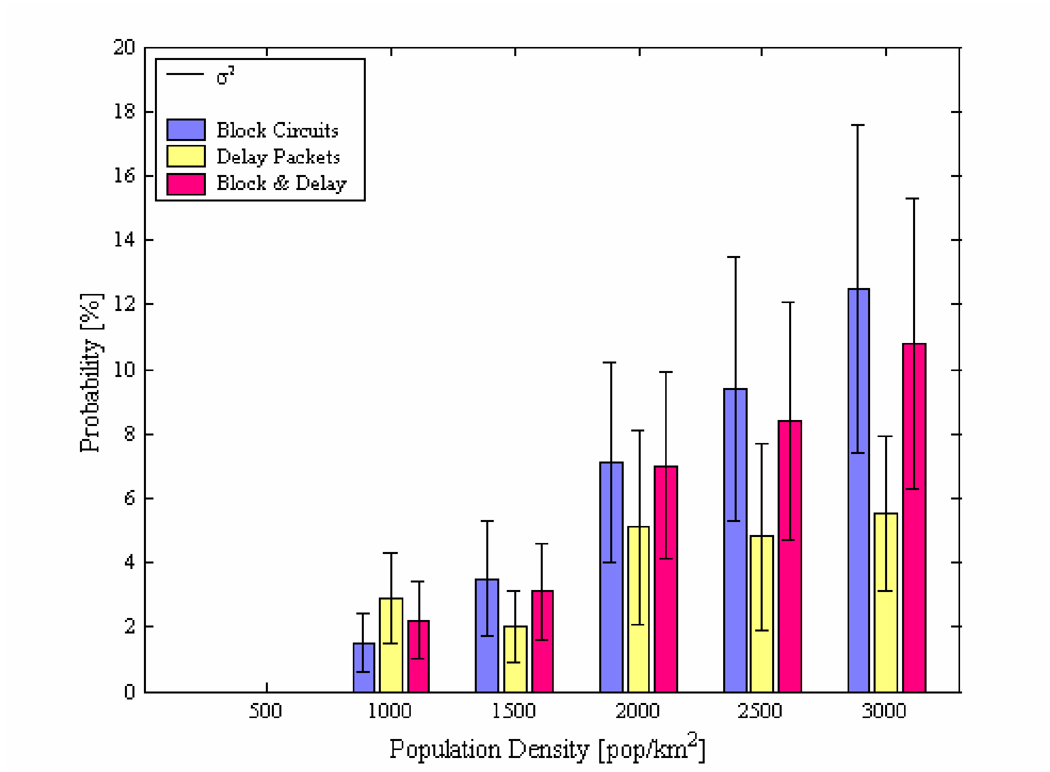

Figure 5. 17 – Model II SOHO Scenario, blocking and delay

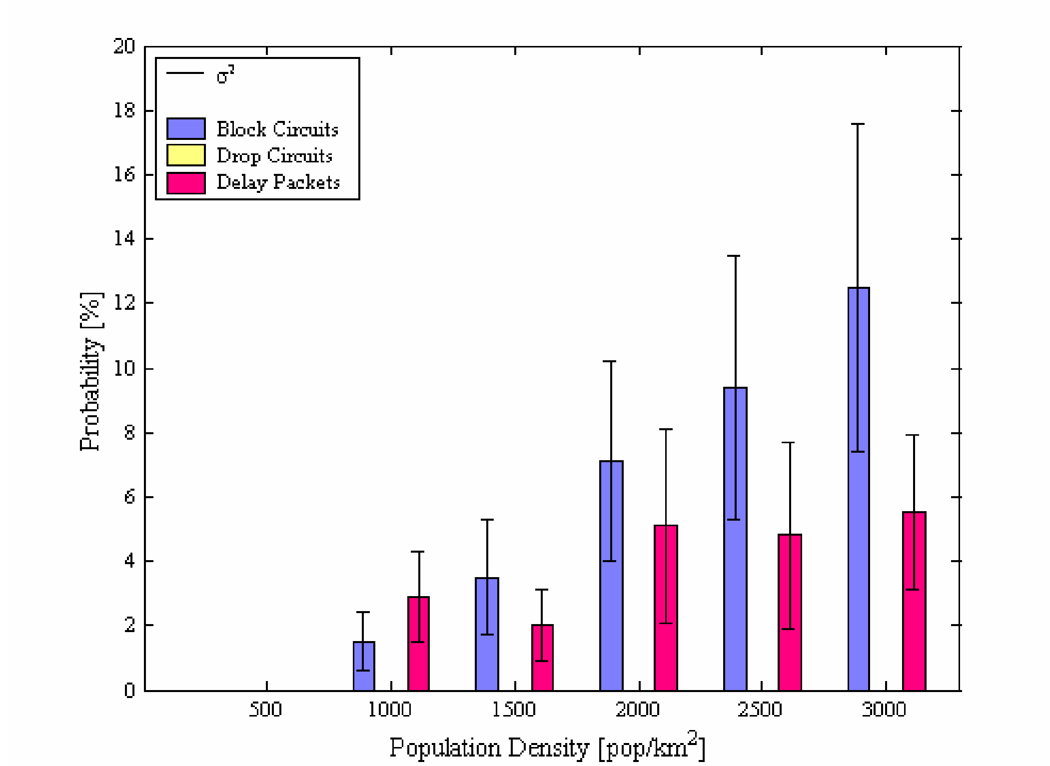

Figure 5. 18 – Model II SOHO Scenario, drop calls.

6 CONCLUSIONS

This thesis presents an analysis of the overall performance of a UMTS network, focusing on the traffic limitation at the radio interface, by simulating the aspects that influence the overall system performance:

- Service characterisation, which includes source models generation for each service in the call/session arrival, traffic generation, duration and interarrival time between call/packets.
- Link budget calculation, including the interference limitations, propagation model and load factor control.
- Different scenario implementations, which consider different service penetration, utilisation and service mixing.

As a result of this work a given cell can be analysed with a great range of input parameters, resulting in an important tool for simulation analysis.

An important system approach corresponds to estimate the traffic only in the DL transmission where high bit rates and capacity limitations are expected. Another approximation corresponds to neglecting the user mobility in the overall system performance analysis, considering that the users are fixed during the simulation time.

Eight service types are considered: Speech, Video Telephony, Streaming Multimedia, Web Browsing, Location Based Services, MMS, Email and File Download. The different services characterisation allow the identification of three implementation scenario, Business, SOHO and Mass-Market, with all the services active and with only some requested services (Speech, Location Services, MMS and Email).

This analysis gives an overview of the initial phase of system deployment, as well as characterises different user needs, like for example the load in a traffic jam and in Business areas with a high building density.

The simulator was implemented in the C++ language based in four main functional blocks:

- User traffic generation, where the traffic characteristics, the population density and the penetration for each service is configured.
- Network analysis, which performs the network traffic and propagation calculation for each user in the system.
- Random traffic generation, which implements the statistical service models for each service resulting in the traffic generation.
- Network performance analysis, which outputs the general performance parameters for the general system performance and for each service.

The simulator provides several output indicators for call admission blocking probability for CS and delay probability for PS. For active connections, the simulator calculates the drop probability for CS and the delay probability for PS.

From the statistical analysis of different user traffic scenario, it is the Business scenario with all the services actives that corresponds to the most limiting situation, and the Mass-Market scenario with only four services active to the most favourable one. For a reasonable system operation, it is considered that the population density for the Business scenario should be around 300 pop/km^2 for a cell radius of 900 m. For the SOHO scenario, with a 900 m cell radius, a population density of 900 pop/km^2 is achieved, being however concluded that an improvement could be obtained by reducing the cell radius. The Mass-Market scenario is the most adapted to macro-cell environments, a population density of 1200 pop/km^2 being achievable. Considering only four services available, Speech, Location Based Service, MMS and Email, a significant improvement in the maximum population density that can be covered with the predefined quality requirements, i.e., 750 pop/km^2, 1000 pop/km^2 and 10000 pop/km^2 for the Business, SOHO and Mass-Market scenarios respectively. In this situation it is also observed that generally the services with a lower sensitivity lead to higher coverage, this corresponding to an important service combination for the UTMS implementation phase.

The subject that is considered in this thesis is complex and influenced by a great number of parameters, and although many of them are contemplated in the simulator many others can influence the general results. An important contribution as future work, to have more accuracy should include:

- Implementation of user mobility.
- Implementation of a multi-cell network, for the analysis of the influence of intra-cell interference and soft handover capacity.
- Implementation in the PS component of a limitation of the maximum number of retries, after which the call is considered dropped.
- Queues implementation, analysing the network performance when different quality of service policies are applied to each service.
- Non uniform distributions of user over the cell.
- Interworking with other networks, namely GSM, HIPERLAN and others WLANs.

ANNEX – SOURCE MODELS VALIDATION

The simulator uses a variety of statistical traffic models, for which it is important, a Random Generator (RG) validation. Table A. 1 lists the distributions used in the simulator: Poisson, Exponential, Lognormal, Geometric, Pareto, Uniform and Weibull.

In this Annex the comparison between the theoretical distributions and the equivalent histograms generating a considerable amount of samples (5000) is presented, with the corresponding Mean Square Error (MSE).

For MSE calculations, the following equation was used:

$$MSE = E \cdot \left[\left(p_X \left[x_N \right] - p_N \left[n \right] \right)^2 \right] \quad \textbf{(A. 1)}$$

where E correspond to the expected value of x, $p_x[x_N]$ to the theoretical distribution and $p_N[n]$ is the equivalent modified histograms obtained with the intervals and bar width described in Table A. 1.

Table A. 1 – Random Generator Analysis

Distribution	x_{min}	x_{max}	w	MSE
Poisson (μ=2.4)	0	20	1.0	3.9×10^{-8}
Exponential (μ=60)	0	500	9.8	3.2×10^{-10}
Lognormal (μ=5.1968, σ=0.2016)	0	500	6.9	1.3×10^{-9}
Geometric((μ=15)	0	150	5.0	5.3×10^{-11}
Pareto (k=81.5,α=1.1)	0	1200	13.3	2.3×10^{-6}
Uniform (*min*=0,*max*=1)	0	1	0.16	1.9×10^{-3}
Weibull (λ=0.03, β=0.75)	0	300	9.4	1.7×10^{-11}

From the analysis of the results, it can be concluded that the simulated results are approximated to the theoretical ones.

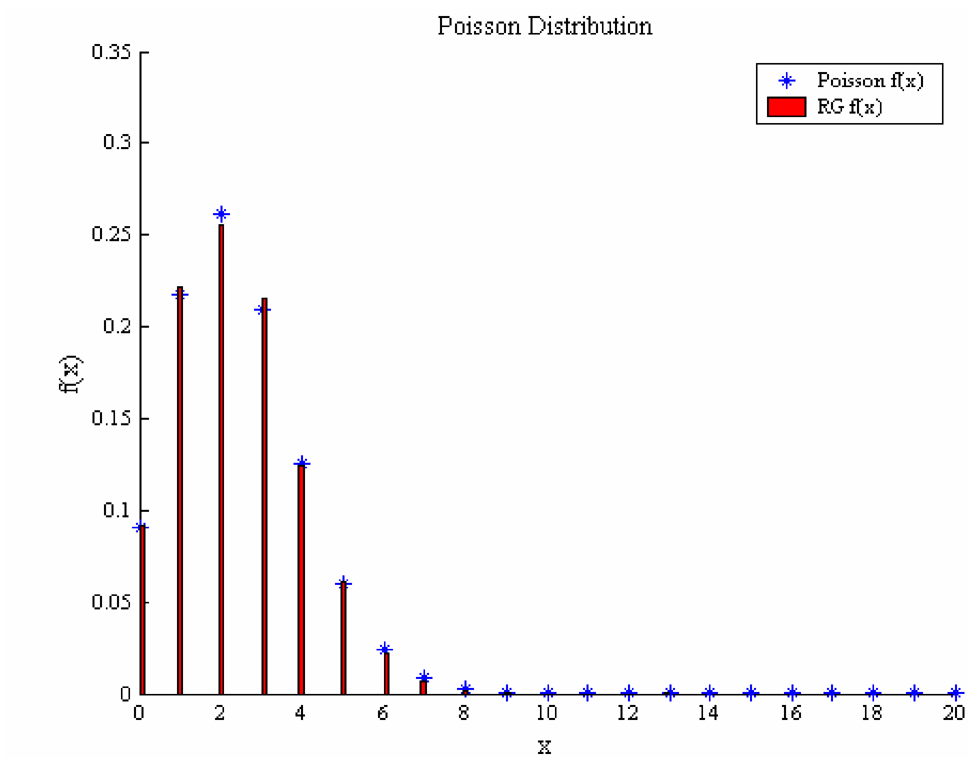

Figure A. 1– Comparison for the PDF of the Poisson distribution (μ=2.4).

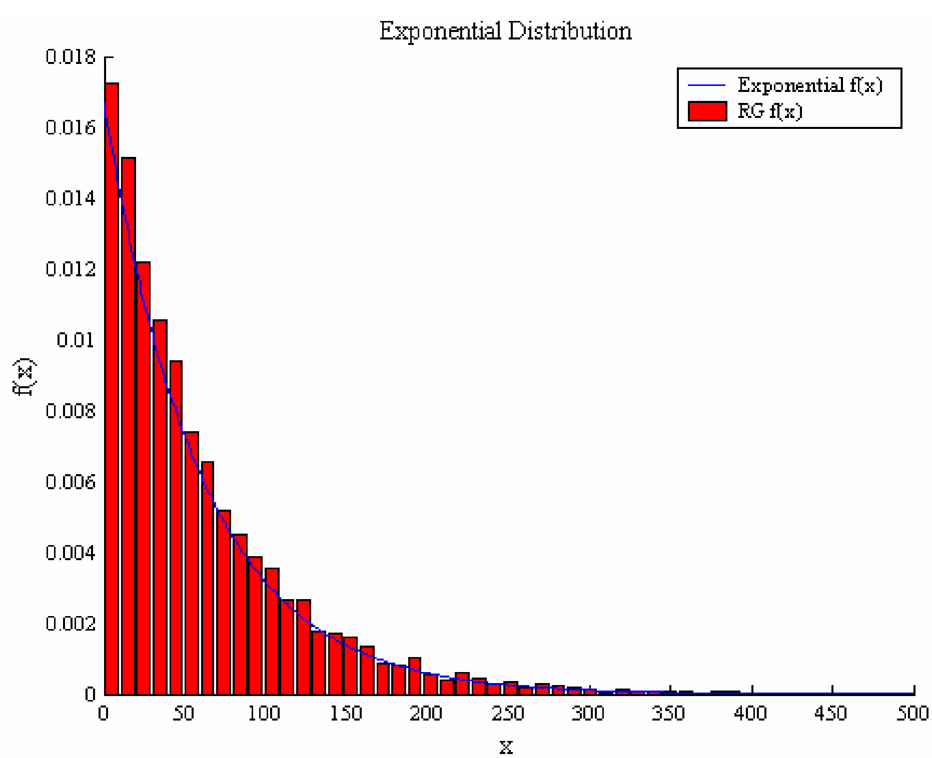

Figure A. 2 – Comparison for the PDF of the Exponential distribution (μ=60).

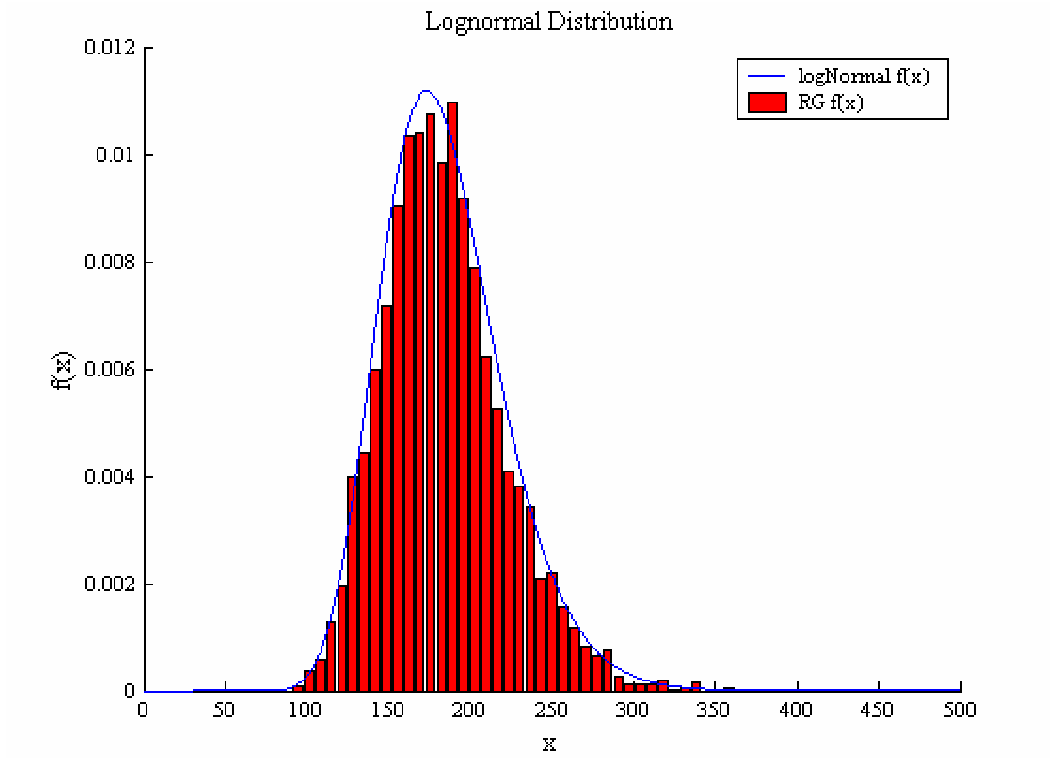

Figure A. 3 – Comparison for the PDF of the Lognormal distribution (μ=5.1968, σ=0.2016).

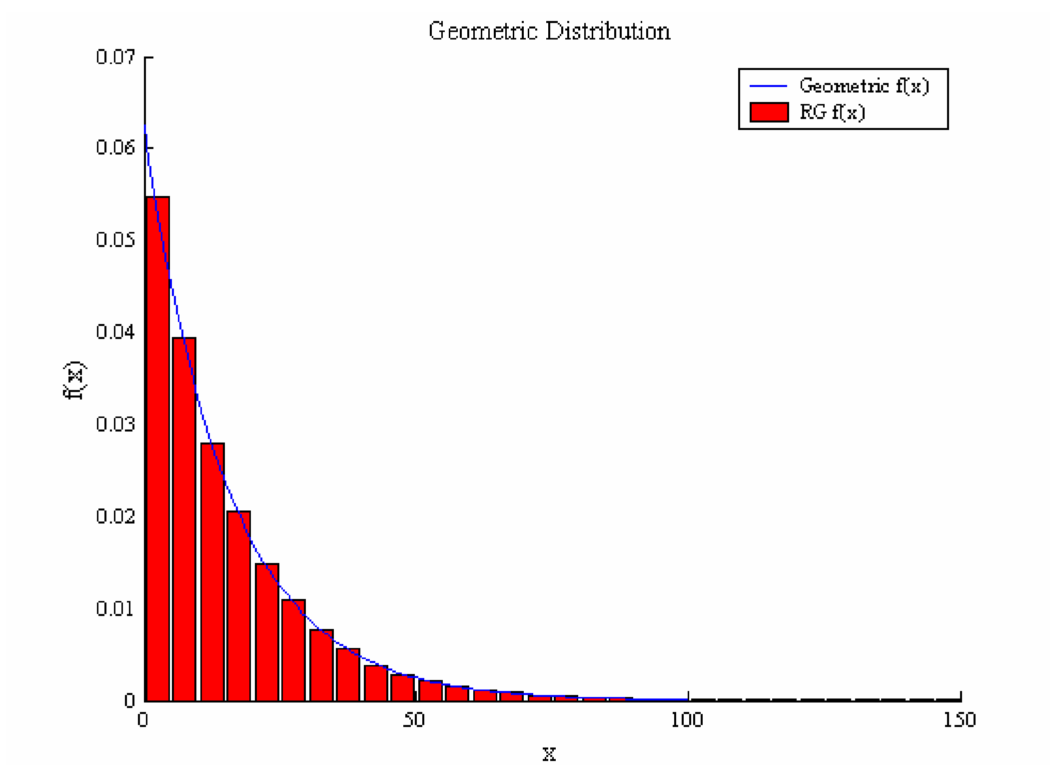

Figure A. 4 – Comparison for the PDF of the Geometric distribution (μ=15)

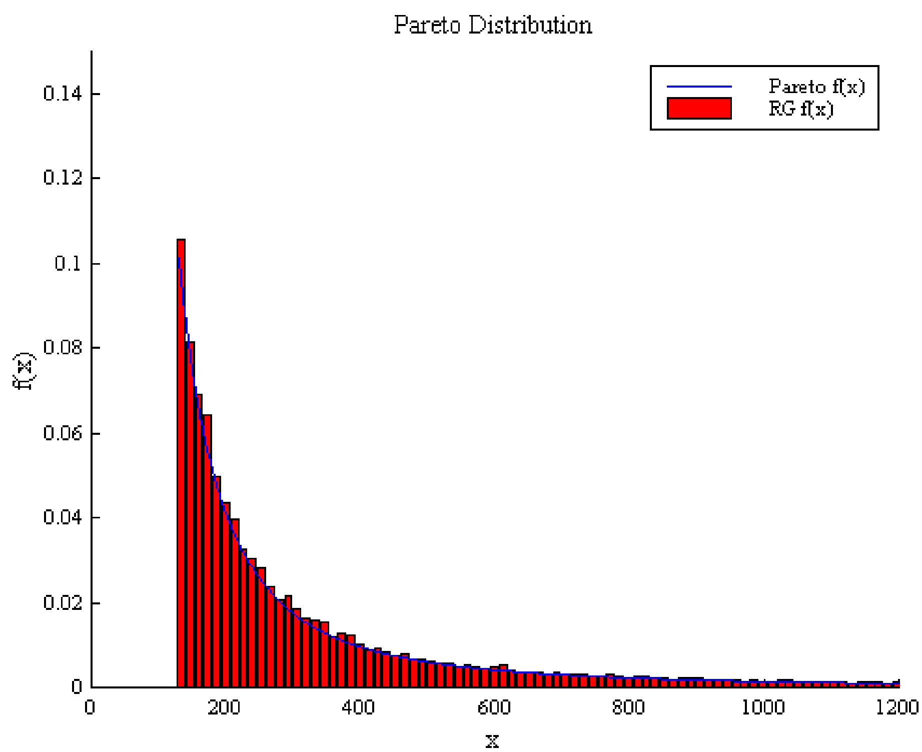

Figure A. 5 – Comparison for the PDF of the Pareto distribution (k=81.5, α=1.1)

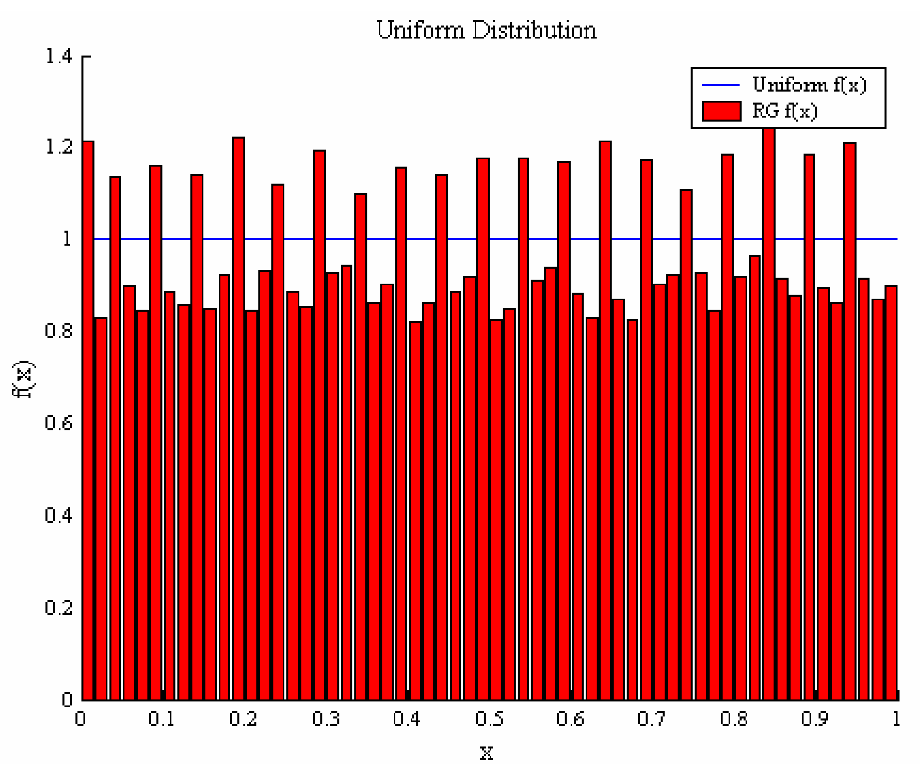

Figure A. 6 – Comparison for the PDF of the Uniform distribution (0,1)

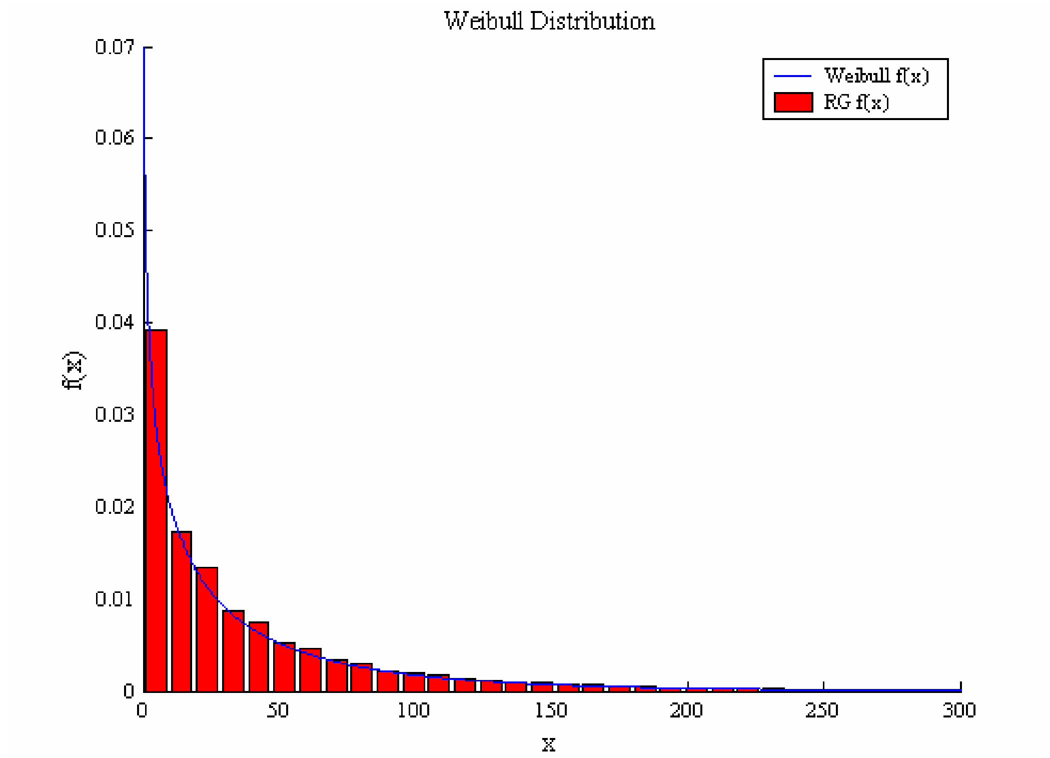

Figure A. 7 – Comparison for the PDF of the Weibul distribution (λ=0.03, β=0.75)

Table A. 2 – Voice ON/OFF Model simulation

State	Probability	Simulation 1	Simulation2
1	0.5978	0.5554	0.5688
2	0.0723	0.1011	0.0935
3	0.0388	0.0543	0.0509
4	0.2911	0.2893	0.2867

Table A. 3 – Specific WWW States simulation

States	Probability	Simulation 1	Simulation 2
0	0.5048	0.4870	0.5000
1	0.3023	0.3132	0.3000
2	0.1923	0.1998	0.2000

REFERENCES

[1] Serrador,A., *Optimisation of Cell Radius in UMTS-FDD Networks*, Master Thesis, Instituto Superior Técnico, Lisbon, Portugal, Dec.2002.

[2] IST-MOMENTUM Project (https://momentum.zib.de).

[3] ETSI (http://www.etsi.org).

[4] 3GPP (http://www.3gpp.org).

[5] UMTS Forum (http://www.umtsforum.org).

[6] UMTS Forum, *UMTS/IMT-2000 Spectrum*, Report No. 6, London, UK, June 1999.

[7] Holma,H. and Toskala,A., *WDCDMA for UMTS*, John Wiley & Sons, Chichester, UK, 1999.

[8] 3GPP, *UTRAN Overall Description,* Technical Specification Group Radio Access Network Report No. 25.401 version 4.0.0, Sophia Antipolis, France, Mar. 2001.

[9] UMTS Forum, *Minimum spectrum demand per public terrestrial UMTS operator in the initial phase*, Report No. 5, London, UK, Dec. 1998.

[10] 3GPP, *Spreading and modulation (FDD),* 3GPP Technical Specification Group Radio Access Network Report Nº 25.213 version 4.0.0, Sophia Antipolis, France, Mar. 2001.

[11] 3GPP, *Physical channels and mapping of transport channels onto physical channels (FDD),* Technical Specification Group Radio Access Network Report Nº 25.211 version 4.0.0, Sophia Antipolis, France, Mar. 2001.

[12] Chaudhury,P., Mohr,W. and Onoe,S. “The 3GPP Proposal for IMT-2000”, *IEEE Communications Magazine*, Vol.37, No. 12, Dec. 1999, pp. 72-81.

[13] 3GPP, *Physical channels and mapping of transport channels onto physical channels (TDD),* Technical Specification Group Radio Access Network Report Nº 25.221 version 4.0.0, Sophia Antipolis, France, Mar 2001.

[14] 3GPP, *QoS Concept,* Technical Specification Group Services and System Aspects 3G TR Nº 23.907 version 1.2.0, Sophia Antipolis, France, Mar 2001.

[15] 3GPP, *QoS Concept and Architecture,* Technical Specification Group Services and System Aspects 3G TR Nº 23.107 version 5.3.0, Sophia Antipolis, France, Jan. 2002

[16] ITU-T, *B-ISDN Services Aspects,* ITU-T Recommendation I.211, Geneva, Switzerland, Mar. 1993.

[17] Ferreira,L., Correia,L. , Xavier,D. and Vasconcelos,A., *Final report on traffic estimation and services characterisation*, IST-Momentum Project, Deliverable D1.2, IST-TUL, Lisbon, Portugal, Apr. 2003.

[18] Damosso,E. and Correia,L., *COST 231 Final Report*, COST Office, Brussels, Belgium, 1999.

[19] Valko,A.G., Racz,A. and Fodor,G., “Voice QoS in Third-Generation Mobile Systems”, *IEEE Journal on Selected Areas in Communications*, Vol. 17, No. 1, Jan. 1999, pp 109-123

[20] Krunz,M. and Hughes,H., “*A Traffic Model for MPEG-Coded VBR Streams*”, Department of Electrical Engineering, Department of Computer Science, Michigan State University USA.

[21] ETSI, *Selection procedures for the choice of radio transmission technologies of the UMTS* (UMTS 30.03 version 3.2.0), TR 101 112 v3.2.0, Sophia Antipollis, France, April 1998.

[22] Klemm,A., Lindemann,C. and Lohmann,M., “Traffic Modelling and Characterisation for UMTS Networks”, *Proc. of the Globecom, Internet Performance Symposium*, San Antonio, TX, USA, Nov 2001.

[23] Serrador,A., Ferreira,L. and Correia,L., *Contribution to D2.2*, IST-Momentum Project, IST-TUL, Deliverable D2.2, IST-TUL, Lisbon, Portugal, Jan 2003.

[24] Ferreira,L., Correia,L., Serrador,A., Carvalho,G., Fledderus,E. and Perera,R. *UMTS Deployment and Mobility Scenarios*, IST-Momentum Project, Deliverable D1.3, IST-TUL, Lisbon, Portugal, Nov. 2002.