INSTITUTO SUPERIOR TÉCNICO
Universidade Técnica de Lisboa

Common Radio Resources Management in Heterogeneous Networks, with UMTS, Wi-Fi and WiMAX

Jorge Miguel Simões Venes

Dissertation submitted for obtaining the degree of
Master in Electrical and Computer Engineering

Jury

Supervisor: Prof. Luís Manuel de Jesus Sousa Correia

President: Prof. José Manuel Bioucas Dias

Member: Prof. António José Castelo Branco Rodrigues

October 2009

To *my family*

Acknowledgements

I would like to start by thanking Prof. Luís Correia for supervising this thesis and providing the ideal environment to develop it - for the second time in my student life, I was given the privilege of working under his supervision. His great professional and human skills can not be properly described in a few lines of text, so I would just like to stress how decisive his support and his words were in the most anguishing moments of the development of this thesis – and express him my deep gratitude for that.

I would also like to thank sincerely the decisive contribution of António Serrador, for sharing the product of many years of work with me without hesitating. My work was developed over a basis totally built by his hands, so this thesis wouldn't have been possible without his support. I also thank David Antunes for his important contribution, on which this work is also partially based.

This thesis demanded a lot of effort in terms of simulation time and resources. Family members, friends and colleagues generously provided precious help borrowing "CPU time" for running simulations. To Amílcar Pereira, Ana Gonçalves, Anabela Silva, Eduardo Santos, Filipe Fortunato, Nelson Rodrigues, Patricia Sacoto, Raquel Dias, and specially to my brother Nuno Venes and to my sister-in-law Paula, thank you very much for adding "statistical significance" to this work.

To GROW, for being a group of exceptional people, whether the subject is Research on Wireless or fun in Monsaraz.

To my family, specially my parents, for their unconditional love and support throughout my 31 years of living.

To my beloved wife Ana, for supporting my long "absence" devoted to this thesis and for making me wish for the moment of returning home at the end of every day of work.

To my wonderful 5 years old son André, for understanding so clearly the reason why his daddy had so little time to play with him during the last year.

Abstract

The present work addresses Common Radio Resources Management of a Heterogeneous Network (HN) integrating UMTS, Wi-Fi and WiMAX. A simulator was developed in order to evaluate the performance in scenarios like variation of density and geographic distribution of users, variation of average duration of voice calls and data packets volume, or variation of WiMAX channel bandwidth. The HN is managed by an algorithm based on a service/sub-system priority table. Blocking probability dependency with the number of users and with average duration of voice calls is well modelled by exponential and linear functions, respectively. The Average Delay for the reference scenario is around 20 ms and grows circa 15 ms per each 15 s added to the average duration of voice calls or per each 80 bytes added to the volume of data packets. Average delay is raised 100 times when WiMAX is switched-off. The joint overall network performance (all sub-systems active) is better than the one with "all-1" sub-systems active - switching-off WiMAX leads to 12 % Blocking Probability and 2 s Average Delay increase. Bitrates of UMTS, Wi-Fi and WiMAX vary around 1, 20 and 10 Mbps, respectively.

Keywords

Heterogeneous Network, CRRM, UMTS, Wi-Fi, WiMAX, Simulator.

Resumo

O presente trabalho visa Gestão Comum de Recursos Rádio de uma Rede Heterogénea (RH) com UMTS, Wi-Fi e WiMAX. Desenvolveu-se um simulador para aferir o desempenho em situações de variação da densidade e distribuição geográfica de utilizadores, da duração média das chamadas, do volume dos pacotes ou da largura de banda do WiMAX. A RH é gerida com base numa tabela de prioridades serviço/sub-sistema. A dependência da Probabilidade de Bloqueio com o número de utilizadores e com a duração média das chamadas de voz é bem modelada por funções exponencial e linear respectivamente. O Atraso Médio ronda os 20 ms no cenário de referência, e cresce cerca de 15 ms por cada 15 s acrescidos de duração média de chamadas de voz ou por cada 80 bytes adicionados ao volume dos pacotes. O Atraso Médio aumenta 100 vezes quando o WiMAX é desactivado. O desempenho conjunto da RH com todos os sub-sistemas activos é superior ao obtido com um dos sub-sistemas desligado – desligar o WiMAX faz aumentar 12 % a Probabilidade de Bloqueio e 2 s o Atraso Médio. Os débitos UMTS, Wi-Fi e WiMAX rondam 1, 20 e 10 Mbps, respectivamente.

Palavras-chave

Rede Heterogénea, Gestão Comum de Recursos Rádio, UMTS, Wi-Fi, WiMAX, Simulador.

Table of Contents

Acknowledgements v

Abstract vii

Resumo viii

Table of Contents ix

List of Tables xvi

List of Acronyms xvii

List of Symbols xxi

1 Introduction 1

2 Systems Overview 7

2.1 UMTS 8

2.1.1 System Description 8

2.1.2 RRM Aspects 10

2.2 Wi-Fi 12

2.2.1 System Description 12

2.2.2 RRM Aspects 13

2.3 WiMAX 14

2.3.1 System Description 14

2.3.2 Radio Interface 16

2.3.3 RRM Aspects 20

2.4 General Comparison of Systems 21

2.5 Services and Applications 22

2.6 Common Radio Resources Management 26

2.6.1 Levels of Integration 26
2.6.2 CRRM related work 29

3 Algorithms and Simulator Description 33

3.1 Algorithms and Models 34

3.1.1 CRRM 34
3.1.2 UMTS and Wi-Fi 35
3.1.3 WiMAX 36

3.2 Simulator 41

3.2.1 General Description 41
3.2.2 Inputs and Outputs 42
3.2.3 Assessment 46

4 Analysis of Results 51

4.1 Reference Scenario 52

4.2 System performance as a function of user-dependent parameters 55

4.2.1 User density 56
4.2.2 Geographical distribution of users 59
4.2.3 Average duration of voice calls 62
4.2.4 Packet volume of data services 65
4.2.5 Service Penetration 68

4.3 System performance as a function of network-dependent parameters.. 71

4.3.1 Priority table 71
4.3.2 Selective RAN switch-off 73
4.3.3 WiMAX channel bandwidth 76

5 Conclusions 81

Annex A – Traffic Source Models 87

A.1 Voice Models 88

A.2 Video Model 90

A.3 Non-Conversational Applications 91

A.4 Streaming Model 92

Annex B – Propagation Models 93

B.1 COST 231 Walfish-Ikegami Model 94

B.2 Double Breakpoint Model.. 96

Annex C – Link Budget for WiMAX...97

Annex D – HNP variation with the density of users 101

Annex E – HNP variation with the geographical distribution of users ..105

Annex F – HNP variation with the average duration of the Voice calls 109

Annex G – HNP variation with the packet volume of data services 113

Annex H – HNP variation with the service penetration 117

Annex I – HNP variation with the priority table 121

Annex J – HNP variation with selective RAN switch-off.......................125

Annex K – HNP variation with the bandwidth of the WiMAX channel ..129

References……………………………………………………………..139

List of Figures

Figure 1-1 – Example of a heterogeneous network (extracted from [RSAD05]).....3
Figure 2-1 - UMTS architecture (extracted from [HoTo04]). 8
Figure 2-2 – Architecture of a IEEE 802.11 WLAN in infrastructure mode (adapted from [RSAD05]).....12
Figure 2-3 - WiMAX network reference model (extracted from [Nuay07]).....14
Figure 2-4 – Nomadic or portable BWA (extracted from [Nuay07]). 16
Figure 2-5 - Comparison between multi-path frequency selective fading influence on SC and MCM signals (extracted from [Antu07]). 16
Figure 2-6 - TDD Frame Structure (extracted from [WiMA06a]). 18
Figure 2-7 - Low integration degree between CRRM and local RRM functions (extracted from [RSAD05]).....26
Figure 2-8 - High Integration degree between CRRM and local RRM functions (extracted from [RSAD05]).....27
Figure 2-9 - Very high integration degree between CRRM and local RRM functions (extracted from [RSAD05]).....28
Figure 2-10 – Simulation Topology (extracted from [LMSB08]).....29
Figure 2-11 – The simulation environment (extracted from [YNCF07]). 31
Figure 3-1 - CRRM algorithm. 35
Figure 3-2 - Congestion control function for voice and video calls (extracted from [Antu07]). 37
Figure 3-3 - Translation of 2-Dimensional Time and Frequency Domains to Frequency Domain (extracted from [Antu07]). 39
Figure 3-4 – CRRM simulator block diagram. 42
Figure 3-5 - User interface of the simulator.....43
Figure 3-6 - Average Global Bitrate for a 1h10m simulation period. 47
Figure 3-7 - Convergence of the HNP function. 49
Figure 4-1 - BS distribution in the REF and respective coverage areas. 52
Figure 4-2 – Blocking Probability variation with users density (real and adjusted). 56
Figure 4-3 – Average Delay variation with users density (real and adjusted).....57
Figure 4-4 – Number of handover variation with users density (real and adjusted).....58
Figure 4-5 – Average connected users per second variation with users density.....59
Figure 4-6 - Geographical distribution of users according to a bi-dimensional normal function with a standard deviation of 150 m. 59
Figure 4-7 – Blocking probability variation with the distribution of users. 60
Figure 4-8 – Average Delay variation with the distribution of users. 60
Figure 4-9 - Handover failure rate and drop rate variation with distribution of users. 61
Figure 4-10 - Handover variation with distribution of users.....61
Figure 4-11 - Average bitrate per service variation with distribution of users. 62
Figure 4-12 – Average number of users per service per second variation with average duration of Voice calls. 62
Figure 4-13 – Blocking Probability variation with average duration of the voice calls. 63
Figure 4-14- Average Delay variation with average duration of the voice calls. 63
Figure 4-15 - Vertical, horizontal handover failure rate and drop rate variation with average duration of the voice calls. 64

Figure 4-16 – Average bitrate per system variation with average duration of the voice calls. 65
Figure 4-17 - Blocking Probability variation with packet volume of data services. 65
Figure 4-18 – Average Delay variation with the packet volume of data services. 66
Figure 4-19 – Average number of users per second per system variation with packet volume of data services. 66
Figure 4-20 – Average bitrate per system variation with packet volume of data services. 67
Figure 4-21 – Vertical, horizontal handover failure rate and drop rate as a function of the packet volume of data services. 68
Figure 4-22 – Number of users per service per second variation with service penetration. 68
Figure 4-23 - Blocking Probability variation with service penetration. 69
Figure 4-24 – Average Delay variation with service penetration. 70
Figure 4-25 – Handover failure rate and drop rate variation with the service penetration. 70
Figure 4-26 – Average bitrate per system variation with the service penetration. 71
Figure 4-27 - Blocking Probability variation with priority table. 72
Figure 4-28 – Average Delay variation with priority table. 72
Figure 4-29 – Handover failure rate and drop rate variation with priority table. 72
Figure 4-30 – UMTS Average Bitrate variation with the priority table. 73
Figure 4-31 - Blocking Probability variation with selective RAN switch-off. 74
Figure 4-32 – Average Delay variation with selective RAN switch-off. 74
Figure 4-33 – Handover failure rate and drop rate variation with selective RAN switch-off. 74
Figure 4-34 – Handover variation with selective RAN switch-off. 75
Figure 4-35 – System bitrate variation with selective RAN switch-off. 75
Figure 4-36 – Average number of users per service per second variation with selective RAN switch-off. 76
Figure 4-37 - Blocking Probability variation with the bandwidth of the WiMAX channel. 77
Figure 4-38 – Average Duration variation with the bandwidth of the WiMAX channel. 77
Figure 4-39 – HO failure and Drop rate variation with the bandwidth of the WiMAX channel. 78
Figure 4-40 – Bitrate per system variation with the bandwidth of the WiMAX channel. 78
Figure A- 1 - Typical WWW session (adapted from [ETSI98]) ……………………………………….91
Figure D-1 – Blocking Probability variation with density of users. ……………………………………102
Figure D-2 – Average Delay variation with density of users. 103
Figure D-3 – Total number of handovers variation with density of users. 103
Figure D-4 – Vertical, horizontal handover failure rate and drop rate variation with density of users. 103
Figure D-5 – Average bitrate per service variation with density of users. 103
Figure D-6 – Average bitrate per system variation with density of users. 104
Figure D-7 – Average number of users per service per second variation with density of users. 104
Figure E-1 - Blocking Probability variation with distribution of users. ……………………………106
Figure E-2 - Average Delay variation with distribution of users. 107
Figure E-3 - Total number of handovers variation with distribution of users. 107
Figure E-4 - Vertical, horizontal handover failure rate and drop rate variation with distribution of users. 107
Figure E-5 - Average bitrate per service variation with distribution of users. 107
Figure E-6 - Average bitrate per system variation with distribution of users. 108
Figure E-7 - Average number of users per service per second variation with distribution of users. 108
Figure F-1 - Blocking Probability variation with average duration of the voice calls. …………110
Figure F-2 - Average Delay variation with average duration of the voice calls. 111

Figure F-3 - Total number of handovers variation with average duration of the voice calls. 111
Figure F-4 – Vertical, horizontal HO failure rate and drop rate variation with average duration of the voice calls. 111
Figure F-5 – Average bitrate per service variation with average duration of the voice calls...............111
Figure F-6 – Average bitrate per system variation with average duration of the voice calls...............112
Figure F-7 – Average number of users per service per second variation with average duration of the voice calls. 112
Figure G-1 - Blocking Probability variation with packet volume of data services. …………………..114
Figure G-2 - Average Delay variation with packet volume of data services.......................................115
Figure G-3 - Total number of handovers variation with packet volume of data services. 115
Figure G-4 – Vertical, horizontal HO failure rate and drop rate variation with packet volume of data services. 115
Figure G-5 – Average bitrate per service variation with packet volume of data services. 115
Figure G-6 – Average bitrate per system variation with packet volume of data services. 116
Figure G-7 – Average number of users per service per second variation with packet volume of data services. 116
Figure H-1 - Blocking Probability variation with service penetration. …………………………….118
Figure H-2 - Average Delay variation with service penetration. 119
Figure H-3 - Total number of handovers variation with service penetration. 119
Figure H-4 – Vertical, horizontal handover failure rate and drop rate variation with service penetration. 119
Figure H-5 – Average bitrate per service variation with service penetration.......................................119
Figure H-6 – Average bitrate per system variation with service penetration.......................................120
Figure H-7 – Average number of users per service per second variation with service penetration. 120
Figure I-1 - Blocking Probability variation with priority table. ……………………………………122
Figure I-2 - Average Delay variation with priority table. 123
Figure I-3 - Total number of handovers variation with priority table. 123
Figure I-4 – Vertical, horizontal handover failure rate and drop rate variation with priority table. 123
Figure I-5 – Average bitrate per service variation with priority table. 123
Figure I-6 – Average bitrate per system variation with priority table. 124
Figure I-7 – Average number of users per service per second variation with priority table. 124
Figure J-1- Blocking Probability variation with selective RAN switch-off. …………………………….126
Figure J-2 - Average Delay variation with selective RAN switch-off. 127
Figure J-3 - Total number of handovers variation with selective RAN switch-off. 127
Figure J-4 – Vertical, horizontal handover failure rate and drop rate variation with selective RAN switch-off. 127
Figure J-5 – Average bitrate per service variation with selective RAN switch-off. 127
Figure J-6 – Average bitrate per system variation with selective RAN switch-off. 128
Figure J-7 – Average number of users per service per second variation with selective RAN switch-off. 128
Figure K-1 - Blocking Probability variation with WiMAX channel bandwidth and density of users. ..131
Figure K-2 - Average Delay variation with WiMAX channel bandwidth and density of users. 131
Figure K-3 –UMTS Average Bitrate variation with WiMAX channel bandwidth and density of users. 131
Figure K-4 –Wi-Fi Average Bitrate variation with WiMAX channel bandwidth and density of users. 132
Figure K-5 –WiMAX Average Bitrate variation with WiMAX channel bandwidth and density of users. 132

Figure K-6 - Total number of handovers variation with WiMAX channel bandwidth and density of users. 132
Figure K-7 – Vertical, horizontal handover failure rate and drop rate variation with WiMAX channel bandwidth and density of users. 133
Figure K-8 – Average bitrate per service variation with WiMAX channel bandwidth and density of users. 133
Figure K-9– Average bitrate per system variation with WiMAX channel bandwidth and density of users. 134
Figure K-10 – Average number of users per service per second variation with WiMAX channel bandwidth and density of users. 134

List of Tables

Table 2-1 - Main SOFDMA Parameters. 17
Table 2-2 - DL PUSC Sub-carrier allocations....19
Table 2-3 - Receiver SNR assumptions (extracted from [IEEE06]). 20
Table 2-4 – Network architecture comparison between UMTS, Wi-Fi and WiMAX systems....22
Table 2-5 – General comparison between UMTS, Wi-Fi and WiMAX systems. 22
Table 2-6 - UMTS traffic classes (based on [3GPP09]). 23
Table 2-7 - Mapping between User Priority and traffic types (based on [IEEE04b])....24
Table 2-8 - WiMAX service classes (based on [WiMA05])....25
Table 2-9 – Service classification according to the system....25
Table 3-1 - Service priority. 34
Table 3-2 - AMC distribution and slot capacity per modulation type. 38
Table 3-3 - Available DL slots per radio frame 39
Table 3-4 – Maximum number of WiMAX slots assessment for a single BS 47
Table 4-1 – Terrain parameters....53
Table 4-2 – Base station main parameters. 53
Table 4-3 – Service parameters of the reference scenario. 54
Table 4-4 – HNP function for the reference scenario. 55
Table 4-5 – Blocking Probability numeric values and Correlation Coefficients. 57
Table 4-6 –Average Delay numeric values and Correlation Coefficients. 57
Table 4-7 – P_b numeric values and Correlation Coefficients. 63
Table 4-8 – Average delay numeric values and Correlation Coefficients. 63
Table 4-9 – Average Delay and exponential adjustment VS packet volume of data services. 66
Table A- 1 - Voice Source Model Parameters (partial extracted from [VRFo99]). 88
Table A- 2 – Typical VoIP codecs (extracted from [Nune02]). 89
Table A- 3 – Modelling of VoIP Traffic....89
Table B-1 - Valid parameters range. 95
Table C- 1 – Power classes (extracted from [IEEE04b])....98
Table D-1 - HNP variation with the density of users. 102
Table E-1 - HNP variation with the distribution of users....106
Table F-1 - HNP variation with the average duration of the voice calls. 110
Table G-1 - HNP variation with the packet volume of data services. 114
Table H-1 - HNP variation with the service penetration. 118
Table I-1 - HNP variation with the priority table. 122
Table J-1 - HNP variation with selective RAN switch-off....126
Table K-1 - HNP variation with the bandwidth of the WiMAX channel. 130

List of Acronyms

3GPP	Third Generation Partnership Project
AAA	Authentication, Authorisation and Accounting
ABC	Always Best Connected
AC	Access Category
AMC	Adaptive Modulation and Coding
AP	Access Point
ARQ	Automatic Repeat Request
ASN	Access Service Network
BCH	Broadcast Channel
BE	Best Effort
BoD	Bandwidth on Demand
BPSK	Binary Phase Shift Keying
BS	Base Station
BSS	Basic Service Set
BWA	Broadband Wireless Access
CAC	Call Admission Control
CC	Channelisation Code
CDMA	Code Division Multiple Access
CINR	Carrier-to-Interference-plus-Noise Ratio
CLPC	Closed Loop Power Control
CN	Core Network
CPCH	(UL) Common Packet Channel
CPE	Customer Premises Equipment
CQI	Channel Quality Indication
CQICH	Channel Quality Indicator Channel
CRRM	Common Radio Resources Management
CS	Client Station
CSMA-CA	Carrier Sense Multiple Access with Collision Avoidance
CSN	Connectivity Service Network
DAC	Distributed Admission Control
DCH	Dedicated Channel
DFS	Dynamic Frequency Selection
DL	Downlink
DS	Distributed System

DSCH	Downlink Shared Channel
E-DCH	Enhanced Uplink Dedicated Channel
EDCF	Enhanced Distributed Coordination Function
ERP	Extended Rate Physical
ESS	Extended Service Set
ETSI	European Telecommunications Standards Institute
FACH	Forward Access Channel
FCH	Frame Control Head
FDD	Frequency Division Duplex
FDMA	Frequency Division Multiple Access
FFT	Fast Fourier Transform
FUSC	Fully Used Sub-Channelisation
FWA	Fixed Wireless Access
GERAN	GSM Radio Access Network
GGSN	Gateway GPRS Support Node
GMSC	Gateway MSC
GPRS	General Packet Radio System
GSM	Global System for Mobile Communications
H-NSP	Home Network Service Provider
HARQ	Hybrid Automatic Repeat Request
HC	Hybrid Coordinator
HCF	Hybrid Coordination Function
HHO	Horizontal Handover
HLR	Home Location Register
HN	Heterogeneous Network
HNP	Heterogeneous Network Performance
HO	Handover
HS-DPCCH	High-Speed Dedicated Physical Communication Channel
HS-DSCH	High-Speed Downlink Shared Channel
HS-SCCH	High-Speed Synchronisation Control Channel
HSDPA	High-Speed Downlink Packet Access
HSPA	High-Speed Downlink/Uplink Packet Access
HSUPA	High-Speed Uplink Packet Access
HUMAN	High-Speed Unlicensed Metropolitan Area Network
IE	Information Element
IEEE	Institute of Electrical and Electronics Engineers
IFFT	Inverse Fast Fourier Transform
JRRM	Joint Radio Resources Management
KPI	Key Performance Indicator
LAN	Local Area Network

LB	Load Balancing
LoS	Line of Sight
LTE	Long-Term Evolution
MAC	Medium Access Control
MAP	Media Access Protocol
MAX	Maximum Throughput
MCM	Multi-Carrier Modulation
ME	Mobile Equipment
MS	Mobile Station
MSC	Mobile Services Switching Centre
MT	Mobile Terminal
NAP	Network Access Provider
NLoS	Non-Line-of-Sight
NRT	Non-Real-Time
NSP	Network Service Provider
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
OFUSC	Optional Fully Used Sub-Channelisation
OLPC	Open Loop Power Control
OPUSC	Optional Partially Used Sub-Channelisation
PAN	Personal Area Networks
PCH	Paging Channel
PF	Proportional Fair
PHY	Physical Layer
PLMN	Public Land Mobile Network
PMP	Point-to-Multipoint
PS	Packet-Switching / Packet-Switched
PUSC	Partially Used Sub-Channelisation
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
R5	Release 5 of UMTS
R99	Release 99 of UMTS
RACH	Random Access Channel
RAN	Radio Access Network
RAT	Radio Access Technology
REF	Reference Scenario
RF	Radio Frequency
RNC	Radio Network Controller
ROHC	RObust Header Compression

RR	Round-Robin
RRM	Radio Resources Management
RSSI	Received Signal Strength Indicator
RTP	Real Time Protocol
RTPS	Real-Time Polling Services
SC	Single Carrier
SCM	Single Carrier Modulation
SF	Spreading Factor
SGSN	Serving GPRS Support Node
SIR	Signal-to-Interference Ratio
SNR	Signal-to-Noise Ratio
SOFDMA	Scalable Orthogonal Frequency Division Multiple Access
ST	Subscriber Terminal
TC	Traffic Class
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
TTI	Transmission Time Interval
UDP	User Data Protocol
UE	User Equipment
UL	Uplink
UMTS	Universal Mobile Telecommunications System
USIM	UMTS Subscriber Identity Module
UTRA	UMTS Terrestrial Radio Access
UTRAN	UMTS Terrestrial Radio Access Network
VHO	Vertical Handover
VLR	Visitor Location Register
VoIP	Voice over IP
WCDMA	Wideband Code Division Multiple Access
WiMAX	Worldwide Interoperability for Microwave Access
WLAN	Wireless Local Area Networks
WMAN	Wireless Metropolitan Networks

List of Symbols

α	Pareto distribution parameter
α_i	Coefficient value for autoregressive Markov process modelling
β_{k1}	Shape parameter of the Weibull PDF for Voice model
γ	Distance exponent of free-space propagation model
Δ	Measure of convergence of simulation runs
Δf	Channel bandwidth
λ_{k1}	Scale parameter of the Weibull PDF for Voice model (inverted)
μD_d	Mean value of D_d
μD_{pc}	Mean value of D_{pc}
μN_d	Mean value of N_d
μN_{pc}	Mean value of N_{pc}
μ_x	Mean value of bi-dimensional Normal distribution (in *x*)
μ_y	Mean value of bi-dimensional Normal distribution (in *y*)
σ_x	Standard deviation of bi-dimensional Normal distribution (in *x*)
σ_y	Standard deviation of bi-dimensional Normal distribution (in *y*)
τ	Average Delay
$\overline{\tau^{\exp}}$	Estimation of Average Delay by Exponential function
τ_{k1}	Burst duration of Voice model with source in state *k1*
$\overline{\tau^{lin}}$	Estimation of Average Delay by Linear function
$\overline{\tau^{pol}}$	Estimation of Average Delay by Polynomial function
$\overline{\tau^{pow}}$	Estimation of Average Delay by Power function
τ^{sim}	Average Delay simulation values
Ψ	Street orientation angle
b	Building separation
BR_G	Average global bitrate
BR_{RAN}	Average bitrate per RAN

BR_{srv}	Average Bitrate per Service
d	Distance between BS and MT
D_d	Time interval between two consecutive packets inside a packed call
D_{pc}	Reading time between two consecutive packet call requests in a session
D_r	Drop rate
$EIRP$	Equivalent Isotropic Radiated Power
f	Frequency
Fc	Frame capacity
Fc_{MAX}	Maximum capacity of a frame
Fc_{MIN}	Minimum capacity of a frame
F_N	Noise figure of the receiver
g_i	Gaussian random variable
G_{ar}	Gain of the receiving antenna
G_{at}	Gain of the transmitter antenna
GDU	Geographic distribution of users
G_r	Additional gains, e.g. due to the use of receive diversity
G_t	Additional gains
h_{Base}	BS height
$h_{Building}$	Building height
h_{Mobile}	MT height
HNP	Heterogeneous Network Performance funtion
HNP^{cum_s}	Partial HNP cumulative mean at simulation run s.
HNP^{cum_S}	Total HNP cumulative mean
k	Pareto data volume parameter
$k1$	State variable
L_0	Free-space attenuation
L_{msd}	Multi-screen diffraction loss
L_{ori}	Attenuation caused by main street orientation with respect to the direct radio path
L_p	Path Loss
L_r	Attenuation between the receiving antenna and the receiving radio unit, due to cable loss, connector loss, etc.

L_{rts}	Roof-to-street diffraction and scatter loss
L_t	Attenuation between the transmitting radio unit and the transmitting antenna due to cable loss, connector loss, etc
m_i	Estimated average value of transmission speed (*i* state)
m_{k1}	Mean value of τ_{k1} random variable
N_{cb}	Number of blocked voice and video calls
N_{ct}	Total number of voice and video calls
N_d	Number of packets within a packet call
N_{DL-MAP}	Number of allocated slots for WiMAX DL-MAP
N_{fd}	Number of delayed frames
N_{FFT}	Size of the Fast Fourier Transform
N_{HHOa}	Number of Horizontal Handover Attempts
N_{HHOf}	Total number of failed HHO
$N_{MaxSlots}$	Maximum number of available WiMAX slots for DL transmission per radio frame
$N_{OHSlots}$	Number of slots consumed by WIMAX MAP overhead
N_{pc}	Number of packet call requests per session
N_{pt}	Total number of packets transmitted
N_{RANt}	Total number of RANs
N_{sd}	Total number of dropped sessions
N_{st}	Total number of sessions (ended normally and dropped)
$N_{sub-channels}$	Number of WiMAX sub-channels
$N_{symbols}$	Number of WiMAX symbols
N_{tot}	Total noise power
N_u	Number of users
N_{UDSM}	Maximum number of slots available for user data
N_{UL-MAP}	Number of allocated slots for WiMAX UL-MAP
N_{uMAX}	Maximum number of users
$N_{usedsub-carriers}$	Number of used sub-carriers
NU_{srv}	Average Number of Users per Service per second

N_{VHOa}	Number of Vertical Handover Attempts
N_{VHOf}	Total number of failed VHO
P	Probabilistic transitions matrix
P_b	CRRM Blocking Probability
$\overline{P_b^{\exp}}$	Estimation of CRRM Blocking Probability by Exponential function
$\overline{P_b^{lin}}$	Estimation of CRRM Blocking Probability by Linear function
$\overline{P_b^{pol}}$	Estimation of CRRM Blocking Probability by Polynomial function
$\overline{P_b^{pow}}$	Estimation of CRRM Blocking Probability by Power function
P_b^{sim}	Simulation values of CRRM Blocking Probability
P_{HHOf}	Probability of HHO failure
P_{k1}	Probability of having a packet with size S_{k1} in Voice source model
P_r	Power received at the receiving antenna
$P_{rx\min/subcarrier}$	Receiver sensitivity per subcarrier
$P_{rx\min}$	Composite receiver sensitivity
Ps	Packet Size
Ps_{MAX}	Maximum size of a packet
Ps_{MIN}	Minimum size of a packet
P_t	Power at the exit of the transmitting radio unit
$P_{t,\max}$	Maximum power at the exit of the transmitting radio unit
PT	Priority Table
P_{VHOf}	Probability of VHO failure
Q_{k1}	Probability of selecting a new state in Voice source model
R^2	Correlation Coefficient
$RANoff$	RAN switched-off in the Heterogeneous Network
R_i	Transmission speed of a I frame
s	Partial number of simulation runs
s_1^2	Log-Normal distribution parameter (total volume of data per session)
s_2^2	Log-Normal distribution parameter (inter-arrival time)
S	Total number of simulation runs

S_{k1}	Size of voice model packets for source in state *k1*
SNR_{req}	Required Signal-to-Noise Ratio
SP	Service Penetration
t_{OFF}	Mean Silent Phase of VoIP source model
t_{ON}	Mean Active Phase of VoIP source model
u_1	Log-Normal distribution parameter (total volume of data per session)
u_2	Log-Normal distribution parameter (inter-arrival time)
$VCAD$	Average Duration of Voice Calls
w	Street width;
WBW	WiMAX channel bandwidth

Chapter 1

Introduction

This chapter gives a brief overview of the work. The context in which this thesis fits in and the main motivations for it are brought-up, work targets and scope being established. At the end of the chapter, the work structure is provided.

The most recent developments of Universal Mobile Telecommunications System (UMTS) [HoTo04], High-Speed Downlink Packet Access (HSDPA) and High-Speed Uplink Packet Access (HSUPA) [HoTo06], represent the ultimate stage of evolution of wireless cellular systems in Europe, which started in the last decade of the 20th century with Global System for Mobile Communications (GSM) [Molis04]. The extension of GSM to General Packet Radio System (GPRS) was the first milestone in the evolutionary path of 2G cellular systems towards UMTS. Although its penetration probably has not been as successful as originally expected, GPRS played a key role in the transition from 2G to 3G systems, in the sense that initial UMTS releases used the same core network as GSM/GPRS. The coexistence and interaction between UMTS and GSM/GPRS technologies is considered as one of the key points for the success of 3G technologies.

In parallel with UMTS, other wireless access technologies have experienced significant growth and have arrived in the mass-market. This is the case with the Wireless Local Area Networks (WLAN) [STAL05] and Personal Area Networks (PAN), with the Institute of Electrical and Electronics Engineers' (IEEE) standard 802.11 (Wi-Fi) [IEEE09a] and Bluetooth [Blue09], respectively, being two of their most representative members. The use of non-licensed frequency bands together with the requirement of relatively simple network infrastructures, mainly relying on the existing IP core network, has allowed a fast rollout of WLAN technologies like Wi-Fi. In turn, these are able to provide much higher bandwidths than those currently available in the existing cellular systems like UMTS, though with reduced coverage areas. These aspects have contributed to an increase of the popularity of WLAN in recent years, which is expected to continue with the appearance of new multimedia services.

WiMAX (Worldwide Interoperability for Microwave Access) [Nuay07] is at the moment one of the most promising Broadband Wireless Access (BWA) solutions and has gained a lot of attention from the telecom industry in recent years. It was originally developed as a fixed wireless technology, based on IEEE 802.16-2004 standard for Wireless Metropolitan Networks (WMAN) [EMPS06]. Fixed WiMAX has the potential to bring broadband internet access to the millions of people worldwide who are not connected to a wired network infrastructure. In 2005, the amendment 802.16e added mobility to the original standard and created the basis for the development of Mobile WiMAX.

Compared to 3G Wideband Code Division Multiple Access (WCDMA) systems, like High-Speed Downlink/Uplink Packet Access (HSPA), Mobile WiMAX presents some advantages. HSPA has evolved from 3G WCDMA standards to provide data services over a network originally conceived for mobile voice services. These enhancements inherit both the advantages and limitations of legacy 3G systems. WiMAX, on the other hand, was initially developed for fixed broadband wireless access and is optimised for broadband data services. The physical layer of Mobile WiMAX is based on Scalable Orthogonal Frequency Division Multiple Access (SOFDMA) and the new technologies employed in this system result in lower equipment complexity due to the all-IP core network. Some more comparative advantages of WiMAX are scalable channel bandwidth (1.25 to 20 MHz), frequency-selective scheduling, tolerance to multipath and self-interference, and the fact that it is already an all-IP technology.

It is expected though that Third Generation Partnership Project's (3GPP) Long-Term Evolution (LTE) will bring to HSPA systems significant enhancements, able to rival with WiMAX (or *vice-versa*). The fact that for an UMTS/HSPA operator a smooth transition to LTE can sound more natural than a radical infrastructure swap needed to implement WiMAX can work as a counter factor for the widespread deployment of IEEE's broadband technology. Potentially, a significant part of Mobile WiMAX deployment cases is reserved to new-to-market mobile operators.

Even though HSPA constitutes an effective answer to enhance data throughput in UMTS networks, Mobile WiMAX can offer service providers a profitable model and an alternative to deploy high-throughput services while maximising user satisfaction. It is not beholden to a legacy infrastructure and it can be deployed in stand-alone mode, but Mobile WiMAX can also be deployed as overlay or as a complementary network. Operators can embed Mobile WiMAX within their existing networks to increase capacity and throughput as required, to deliver true wireless broadband services.

Regardless of what the most appropriate technology or system to implement in each context will be, at this point it turns clear that beyond 3G wireless systems will be based on a variety of coexisting Radio Access Technologies (RATs), partially or totally overlapped in coverage, with complementary technical characteristics that will physically coexist in a seamless integrated environment, Figure 1-1.

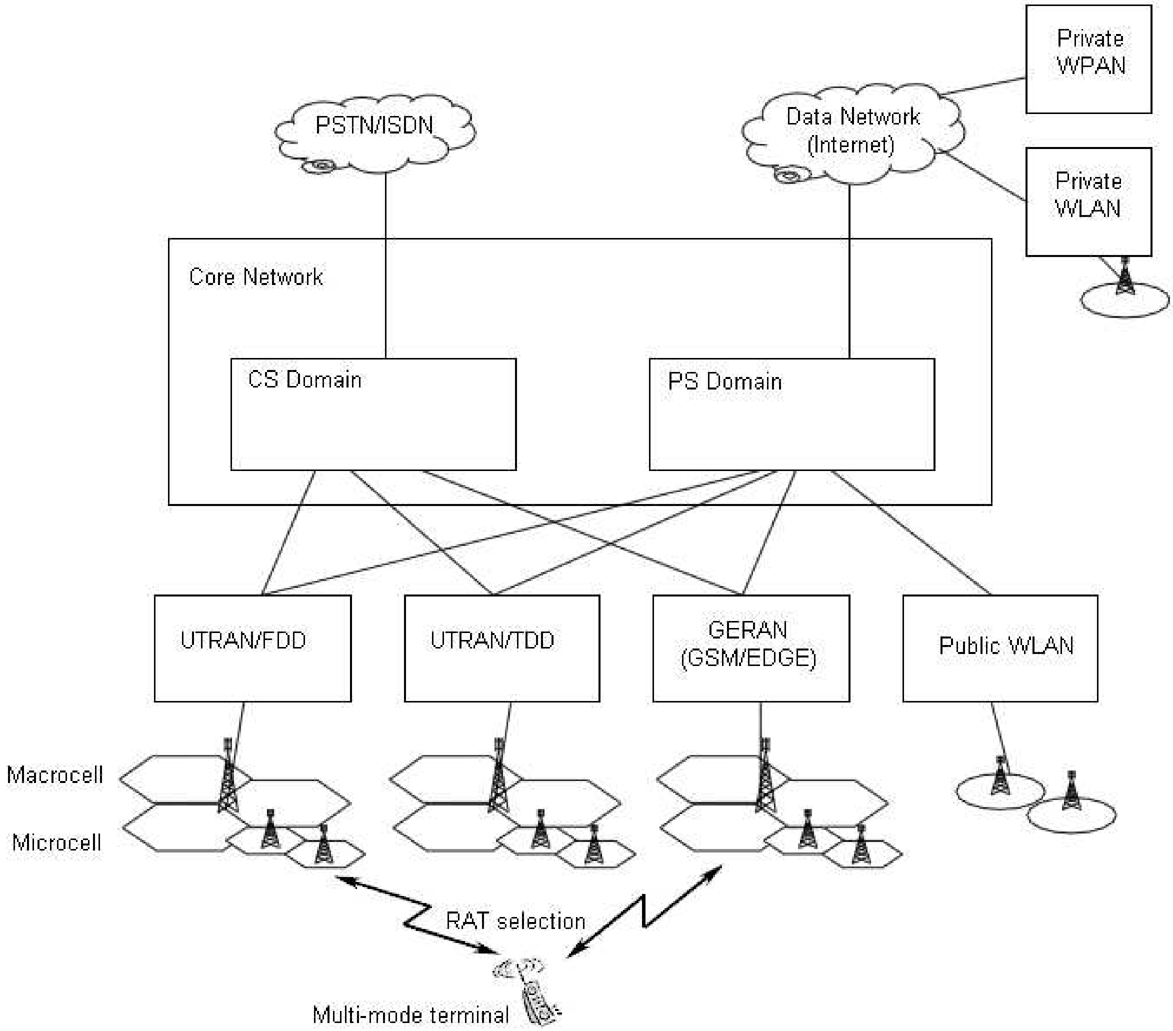


Figure 1-1 – Example of a heterogeneous network (extracted from [RSAD05]).

In this context, an important challenge is how to exploit in a coordinated manner the RATs radio resources to provide users with the required Quality of Service (QoS) levels, while maximising RAT systems revenues. Mobile communications will continue to be one of the most dynamic and profitable market sectors in current and future economics, but it is also one of the highest demanding economic sectors from the point of view of the required investments. In such a competitive and standard-centric industrial environment, the economic exploitation of the solutions directed towards the optimisation of the network performance is considered of key importance.

The notion of being always best connected is an extension for heterogeneous systems of the notion of being always connected. Now, users not only should be connected anywhere, anytime, but also they should be served with the best available connection, which can only be accomplished with the interworking of the different technologies. A common core network deals with all network functionalities and operates as a single network, while Radio Access Networks (RANs) handle only those specific functions of the corresponding RAT, mainly related to physical and link layer issues. Ideally, services must be delivered through the most efficient RAN according to the resource availability, the service QoS requirements, the Mobile Terminal (MT) capabilities and preferences and the operator's policies, by means of a common manager of the radio resources offered by the different RANs. Seamless handover between different RANs should also be assured.

In this way, the Heterogeneous Network (HN) becomes transparent to the final user and the so-called Always Best Connected (ABC) paradigm, which applies to the connection to the RAT that offers the most efficient radio access at each instant, can be achieved. Important research activity has been developed over the last years to define and optimise Common Radio Resources Management (CRRM) policies, also referred as Joint RRM (JRRM).

In the available literature, a great variety of algorithms, techniques, simulation results and other contributions are being proposed. CRRM applied to heterogeneous GPRS/UMTS, UMTS/Wi-Fi or WiMAX/Wi-Fi composed networks are the most common approaches. The contribution of the author to this effort, which is presented throughout this thesis, is to provide a perspective of CRRM applied to a HN composed by UMTS (R99 and R5-HSDPA), Wi-Fi and WiMAX RATs.

The current thesis is based on a previous work developed by Serrador [Serr02], a system level, time-based software simulator developed with the Visual Studio 2005 platform. In its original version, the software tool was able to simulate a HN composed by UMTS and Wi-Fi RANs only. Using the results from another work [Antu07], the author of the current thesis upgraded the original simulator by adding a WiMAX RAN functional module to the already existing UMTS and Wi-Fi ones, and produced the corresponding CRRM enhancements. The resulting simulator only considers the downlink channel.

After having gone through optimisation, testing and assessment, the newly developed version of the simulator was used to perform a series of simulation runs in order to produce a set of results with statistical significance. The objective was to evaluate the performance of a CRRM algorithm based on a service/RAN priority table, and simultaneously to characterise the performance of the heterogeneous UMTS/Wi-Fi/WiMAX network with the variation of mobile users' profiles and network parameters.

This thesis is composed of a main body with 5 chapters and an appendix formed by 11 annexes, identified from A to K. It is organised in the following way:

- In Chapter 2, the three systems in the scope of this work, UMTS, Wi-Fi and WiMAX, are separately described in detail in terms of network architecture, basic characteristics like bandwidth or channels, and Radio Resource Management (RRM) techniques. This chapter also describes the service classes defined in the context of each system, and closes with a general overview of the CRRM concept and the corresponding literature survey characterising the state-of-the-art.
- In Chapter 3, the RRM and CRRM algorithms on the basis of the thesis are described. This chapter also addresses the characterisation of the functional blocks and input/output variables of the simulator. The way that real systems features are adapted to a simulation environment is explained, together with the simplifications and approximations that are necessary in a work of this type.
- In Chapter 4, the results processed from the simulation sets are presented. Eight different situations are simulated, each of these consisting of characterising the response of the heterogeneous network to the variation of a single input parameter. The results are interpreted, explained according to the theory, and compared to what should be expected considering theoretical principles.
- In Chapter 5, the simplifications and approximations on the design and building of the simulator are listed. Final conclusions and future research topics are drawn.

Chapter 2

Systems Overview

This chapter provides an overview of UMTS, Wi-Fi and WiMAX systems. Network architectures, basic characteristics, like bandwidth or channels, and RRM techniques are described. This chapter also describes the service classes defined in the context of each system and closes with a general overview of the CRRM concept and the corresponding literature survey characterising the state-of-the-art.

2.1 UMTS

2.1.1 System Description

The UMTS [HoTo04] high-level network architecture is composed of three elementary elements: the User Equipment (UE), the UMTS Terrestrial Radio Access Network (UTRAN), and the Core Network (CN), Figure 2-1.

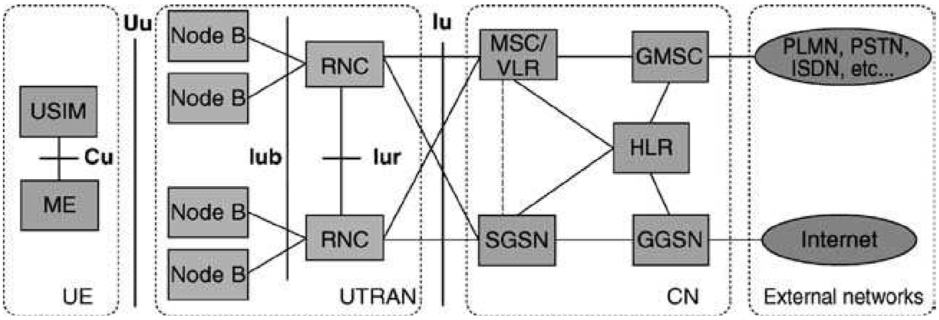


Figure 2-1 - UMTS architecture (extracted from [HoTo04]).

The UE (or MT), consists of two parts: the Mobile Equipment (ME) and the UMTS Subscriber Identity Module (USIM). The UTRAN is composed of two distinct elements: the Base Station (BS), designated by Node B in UMTS, and the Radio Network Controller (RNC), which owns and controls the radio resources achievable through the BSs. The cellular network base architecture is basically inherited from GSM/GPRS, being composed of the following elements:

- **HLR** (Home Location Register) - the database of user profiles;
- **MSC/VLR** (Mobile Services Switching Centre / Visitor Location Register) – the switch and database serving the UE for circuit-switched services;
- **GMSC** (Gateway MSC) - the switching point where UMTS Public Land Mobile Network (PLMN) connects to external circuit-switched networks;
- **SGSN** (Serving GPRS Support Node) - the equivalent MSC/VLR for Packet-Switched (PS) services;
- **GGSN** (Gateway GPRS Support Node) - the equivalent GMSC for PS services;

In Europe, four frequency bands have been allocated to the two possible UMTS duplex modes. For the Frequency Division Duplex (FDD) mode, two 60 MHz bands are available: [1 920; 1 980] MHz for Uplink (UL) and [2 110, 2 170] MHz for Downlink (DL). The FDD mode was initially thought to serve symmetric traffic services like Voice, leaving the support of asymmetric traffic services, like Web-browsing, to a more suited Time Division Duplex (TDD) mode. However, TDD has never reached commercial maturity, due to the significant investments involved in its implementation. Moreover, the concept of Bandwidth on Demand (BoD) is well supported by FDD, since highly variable user data rates are possible in this mode.

Multiple access in UMTS is based on WCDMA, where spreading codes are used to differentiate users. On top of spreading codes, also called Channelisation Codes (CC), Scrambling Codes are used to enhance the Signal-to-Noise Ratio (SNR) without changing the signal bandwidth, by separating MTs using the same CC in UL, and sector cells using the same CCs in DL. In versions prior to HSDPA, variable spreading factors and multi-code connections are used in order to support bit rates up to 2 Mbps. The data modulation scheme Quadrature Phase Shift Keying (QPSK) is used. The chip rate of 3.84 Mcps leads to 4.4 MHz bandwidth RF channels and a total of 12 channels for the FDD mode. Each frame is divided into 15 time-slots, with a duration of 10 ms. User data rates are kept constant during the frame period, but have the flexibility to change from frame to frame. This fast radio capacity allocation can be managed by the network in order to achieve optimum throughputs for packet data services.

In UMTS Terrestrial Radio Access (UTRA), data generated at higher layers are carried over the air with transport channels, which are mapped onto different physical channels. Two types of transport channels exist: dedicated and common. There are six different common transport channel types defined for UTRA in R99: the BCH (Broadcast Channel), the FACH (Forward Access Channel), the PCH (Paging Channel), the RACH (Random Access Channel), the CPCH (UL Common Packet Channel) and the DSCH (DL Shared Channel). The Dedicated Channel (DCH), which is the only dedicated transport channel, carries all the information intended for a given user coming from layers above the physical one, including both data for the actual service and higher layer control information.

HSDPA [HoTo06] is the key new feature included in 3GPP Rel. 5 specification. The idea of HSDPA is to achieve a higher data throughput by using techniques like Adaptive Modulation and Coding (AMC) and Hybrid Automatic Repeat Request (HARQ), combined with Fast Scheduling. AMC mechanisms are handled by the Node B, the main objective being to compensate for radio channel variations, basically by making use of radio channel measurements extracted by the MT and using the Channel Quality Indication (CQI) and retransmission procedure. Taking in addition traffic-related information, such as QoS and the state of radio and physical resources, AMC enables the network to select the most suitable modulation and coding methods. In Rel. 5, 16-QAM and QPSK modulation schemes are allowed, while a fixed Spreading Factor (SF) of 16 (15 codes) is used.

Due to radio channel variability, explicit radio measurements may not just by themselves form a reliable basis for AMC operation and, therefore, complementary mechanisms are needed. HARQ allows the receiving network element to detect errors and to request retransmissions, when necessary. Compared with the conventional ARQ, the added value brought by HARQ lies in its ability to combine the initial estimates or explicit information from both the original transmission and the corresponding retransmissions with the link adaptation process. In this way, it helps to reduce the number of required retransmissions, and also promotes errorless link adaptation regardless of radio channel variations.

In HSDPA, Packet Scheduling and HARQ are handled by the Node B instead of the RNC, as specified in Rel. 4. This change implies that the delay in the scheduling process is minimised, and a better estimation of the radio channel condition is achieved, thus, leading to more reliable scheduling decisions. In addition, the fixed code allocation strategy (15 codes) and the reduction of the Transmission Time Interval (TTI) from 10 ms in Rel. 4 to 2 ms in HSDPA, enable packet scheduling to undertake fast scheduling and frame formation.

There are also three new channel types in HSDPA: the HS-DSCH (High-Speed DSCH), the HS-SCCH (High-Speed Synchronisation Control Channel) and the HS-DPCCH (High-Speed Dedicated Physical Communication Channel).

The HSDPA peak data rate available in the terminals was initially 1.8 Mbps, and has increased to 3.6 and 7.2 Mbps during 2006 and 2007. Potentially, it will go beyond 10 Mbps.

After Rel. 5, 3GPP issued Rel. 6 in 2006, thus giving birth to HSUPA [HoTo06] - or more generally, to HSPA. In fact, HSUPA is not a standalone feature, in the sense that it reuses most of the basic features of Rel. 99. Basically, HSUPA consists of a new UL transport channel, the E-DCH (Enhanced Uplink Dedicated Channel), which brought some of the same features to UL as HSDPA with its new transport channel HS-DSCH provided for DL. The E-DCH transport channel supports fast Node B based scheduling, fast physical layer HARQ with incremental redundancy and, optionally, a shorter 2 ms TTI. Though - unlike HSDPA - HSUPA is not a shared channel but a dedicated one, by structure the E-DCH is more like the DCH of Rel. 99, but with fast scheduling and HARQ: that is, each UE has its own dedicated E-DCH data path to the Node B, which is continuous and independent from the DCHs and E-DCHs of other UEs.

Initial deployments of HSUPA are offering peak data rates from 1 to 2 Mbps. In a second phase, it is expected to reach more than 5 Mbps.

2.1.2 RRM Aspects

The family of RRM algorithms can be divided mainly into Power Control, Handover Control, Admission and Load Control, and Packet Scheduling functionalities.

The main reasons for implementing Power Control are the near-far problem, the interference-dependent capacity of CDMA, and the limited power source of the MT. Unlike TDMA or FDMA systems, where Power Control is applied to reduce inter-cell interference, its purpose in CDMA is mainly to reduce intra-cell interference. Determining the transmission power level is a very sophisticated task, due to unpredictable variations of the radio channel. It is employed in both UL and DL, even though Power Control is more vital in UL than in DL. The power level is adjusted at 1.5 kHz. To manage Power Control properly, two different mechanisms can be used: Open Loop Power Control (OLPC) and Closed Loop Power Control (CLPC), including Inner and Outer Loop mechanisms.

OLPC is used for UL power adjustment. In this process, the UE estimates the strength of the transmission signal by measuring the received power level of the pilot signal from the BS, and adjusts its transmission power level in a way that is inversely proportional to the pilot signal power level. Consequently, the stronger the received pilot signal the lower the MT power transmitted.

Although OPLC alone is useful, e.g., to determine the initial value of the transmitted power, it is insufficient to adjust UE’s transmission power, because the fading characteristics of the radio channel vary rapidly and independently between UL and DL. Therefore, a CLPC mechanism is also needed to adjust transmission power when the radio connection has already been established. Its main purpose is to compensate for the effect of rapid changes in radio signal strength, and hence, it needs to be fast enough to respond to them. The mechanism consists of Inner and Outer Loop variants. Inner loop is also called fast Power Control: in this variant, the BS commands the MT to increase or decrease its transmission power, by comparison to a pre-defined threshold, using a cycle of 1.5 kHz by 1, 2 or 3 dB step sizes. As for the other variant, Outer Loop, its main objective is to keep the target Signal-to-Interference Ratio (SIR) for the UL Inner Loop mechanism at a satisfactory quality level.

UMTS has the three types of handover: hard, soft and softer. Hard handover can be further divided into intra- and inter-frequency. The handover is a process typically handled by the RNC. However, when there is handover between two cells controlled by different RNCs, the MSC is also involved.

Hard handover happens when the old connection is released before making the new one. If the carrier frequency of the new radio access is different from the old one, the process is called *inter-frequency* hard handover. On the other hand, if the new carrier, through which the MT is accessed after the handover process, is the same as the original one, then, there is an *intra-frequency* handover. In multi-system scenarios, inter-system hard handover is also possible, taking place between UMTS and another system, for example GSM.

Unlike Hard handover, in a Soft handover event the MT is temporarily connected to two or more BSs at the same time, and all BSs have the same carrier frequency. Soft handover is the most frequent type of handover in UMTS.

In a Softer handover event, a MT is in the overlapping cell coverage area of two adjacent sectors of a BS, receiving signal from these sectors, i.e., the communication between MT and BS take place concurrently via two air interface channels, one for each sector separately. The two signals are received and combined by means of Rake processing.

If the air interface loading is allowed to increase excessively, the coverage area of the cell is reduced below the planned values, and the QoS of the existing connections cannot be guaranteed. Admission and Load Control are responsible for checking, before admitting a new MT, that this admittance will not sacrifice the planned coverage area or the quality of the existing connections. The Admission Control algorithm accepts or rejects a request to establish a radio access bearer in the radio access network, and is executed when a bearer is set up or modified. It is located in the RNC, where the load information from several cells can be obtained. The Admission Control algorithm also estimates the load increase that the establishment of the bearer would cause in the radio network. This has to be estimated separately for UL and DL. The requesting bearer can be admitted only if both UL and DL admission control admit it, otherwise it is rejected, because of the excessive interference that it would produce in the network. The limits for Admission Control are set by the radio network planning.

The Packet Scheduler, located in the RNC before Rel. 5, has been transferred to the BS in HSDPA. The BS provides the air interface load measurements and the MT provides UL traffic volume measurements to the Packet Scheduler. There is a user-specific Packet Scheduler, controlling the usage of Radio Resource Control states, transport channels and their bit rates according to the traffic, and a cell-specific Packet Scheduler, controlling the sharing of the radio resources among users.

2.2 Wi-Fi

2.2.1 System Description

Wi-Fi corresponds to the commercial implementation of the IEEE 802.11 standard [IEEE09a], the most representative technology of WLANs nowadays. IEEE 802.11 networks are flexible by design, being grouped into two types of WLAN architectures: infrastructure and *ad-hoc* [STAL05]. In *ad-hoc* networks, a group of 802.11 stations communicate directly with one another, usually on a temporary basis, to meet some temporary communication needs. Although *ad-hoc* networks are a possible WLAN configuration, the most usual case is to have infrastructured ones, Figure 2-2.

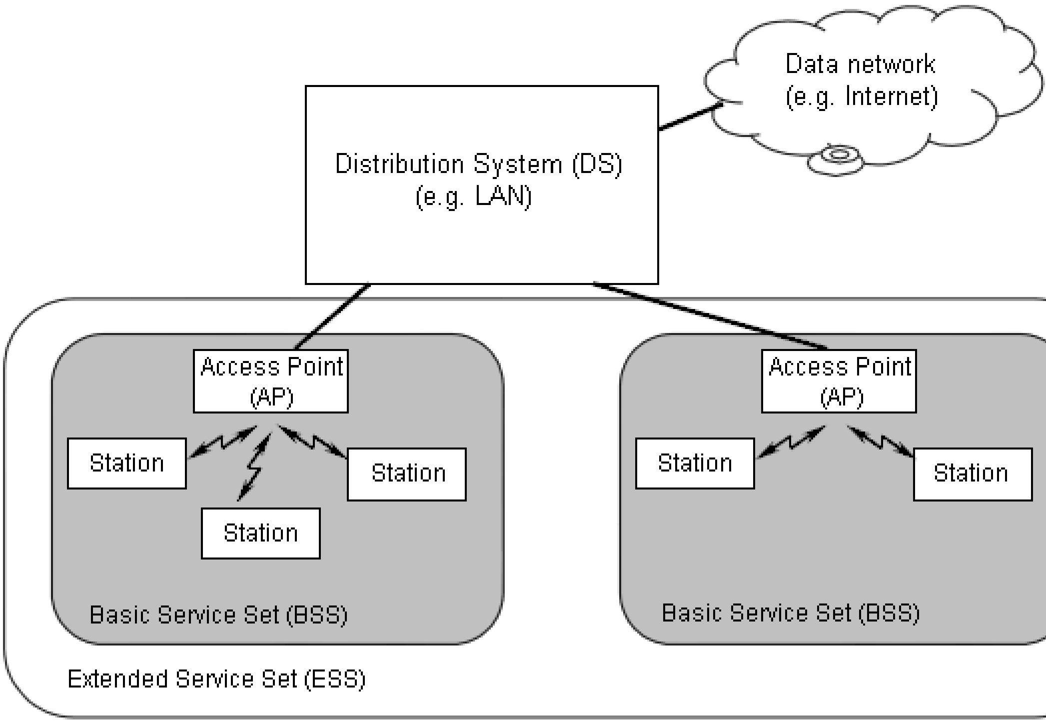


Figure 2-2 – Architecture of a IEEE 802.11 WLAN in infrastructure mode (adapted from [RSAD05]).

Infrastructure networks comprise two types of topologies: Basic Service Set (BSS) and Extended Service Set (ESS), where a Service Set is a logical grouping of devices. A BSS is a group of Client Stations communicating with one another through the Access Point (AP), no direct communication between CSs being allowed. In terms of access network architecture, the CS and the AP terminal equipments are the equivalent to the MT and the BS in UMTS.

The AP also interconnects the BSS with the Distributed System (DS), typically formed by a wired infrastructure (e.g., a wired LAN). The DS allows communication among stations belonging to different BSSs. The whole set of interconnected BSSs through the DS is called Extended Service Set, and is seen by the upper layers (i.e., the Logical Link Control) as a single LAN that can be connected to an external data network.

In terms of Medium Access Control (MAC), the Carrier Sense Multiple Access with Collision Avoidance (CSMA-CA) mechanism is used. CSMA-CA is a listen-before-talk mechanism, with the transmitting station sensing the medium for a carrier signal and waiting until the carrier channel is available before transmitting. This happens because, unlike wired Ethernet stations, wireless stations are not capable of detecting collisions, therefore, the access mechanism must make every effort to avoid collisions altogether.

The 802.11 standards body defined a number of different physical layers (PHY) technologies to be used with the 802.11 MAC: the 802.11 2.4 GHz frequency hopping (up to 2 Mbps), the 802.11 2.4 GHz direct sequencing (up to 2 Mbps), the 802.11b 2.4 GHz direct sequencing (up to 11 Mbps), the 802.11a 5 GHz OFDM (up to 54 Mbps) and the 802.11g 2.4 GHz extended rate physical (ERP) layer (up to 54 Mbps). In this work, 802.11a is implemented. 802.11a uses 20 MHz channels with four channels in each of the three unlicensed bands [5.15, 5.25] GHz, [5.25, 5.35] GHz and [5.725, 5.825] GHz. In terms of modulation, BPSK, QPSK, 16-QAM or 64-QAM can be used. The assigned throughput for a given connection depends on the channel conditions, and can assume a value of 6, 9, 12, 18, 24, 36, 48 and 54 Mbps.

2.2.2 RRM Aspects

802.11 networks work well for low-bandwidth, latency-insensitive data applications. In order to guarantee the same behaviour when it comes to latency-sensitive applications, like voice and video, the 802.11e [IEEE05] Task Group was assigned the responsibility of enhancing the 802.11 MAC to include bidirectional QoS. QoS mechanisms in 802.11 face different challenges compared to QoS in wired networks. 802.11 is a shared, half-duplex medium, subject to interference and performance-degrader phenomena, like co-channel overlap and hidden node problem. To overcome these issues, two solutions are proposed by 802.11e: Hybrid Coordination Function (HCF) with contention operation, more commonly known as Enhanced Distributed Coordination Function (EDCF), and HCF with polled access operation.

The purpose of QoS is to protect high-priority application traffic from low-priority one. Admission Control addresses this issue, by monitoring the available resources of the network and intelligently allowing or disallowing new application sessions. EDCF uses an Admission Control scheme known as Distributed Admission Control (DAC). DAC functions at a high level by monitoring and measuring the percentage of utilisation of the medium for each Access Category (AC). The unused percentage of the medium is referred to as the *available budget* for the AC, which is advertised to stations in the QoS parameter Information Element (IE) in the AP beacons. When the budget starts to approach 0, stations attempting to initiate new application streams avoid doing so, and existing nodes are not able to increase or extend the existing transmission opportunity that they are already using. This process protects the existing applications streams from being impacted by new ones.

In HCF with polled access operation, the AP contains a logical entity known as the Hybrid Coordinator (HC) that keeps track of HCF client stations and schedules the polling intervals. HCF operation, combined with HCF Admission Control, allows the HC to intelligently determine which resources are available on the wireless medium and accept or reject application traffic streams. What truly differentiates HCF-controlled access operation from EDCF is HCF's admission-control mechanism. EDCF's use of DAC relies on the stations to interpret and respect the transmission budget advertised in the QoS parameter set IE. HCF requires that the station request particular reservation parameters for the application traffic stream, such as VoIP, from the HC. The HC can evaluate and determine whether there is enough budget available on the wireless medium to facilitate the requested traffic; it can then accept, reject, or even offer an alternative set of parameters to the station. This mechanism is far more robust than DAC, but a strict schedule of the traffic streams must be kept by HC.

2.3 WiMAX

2.3.1 System Description

The WiMAX [Nuay07], [EMPS06] network reference model consists of three components interconnected by standardised interfaces or reference points R1 to R5, Figure 2-3. The three components are MS (Mobile Station), ASN (Access Service Network) and CSN (Connectivity Service Network).

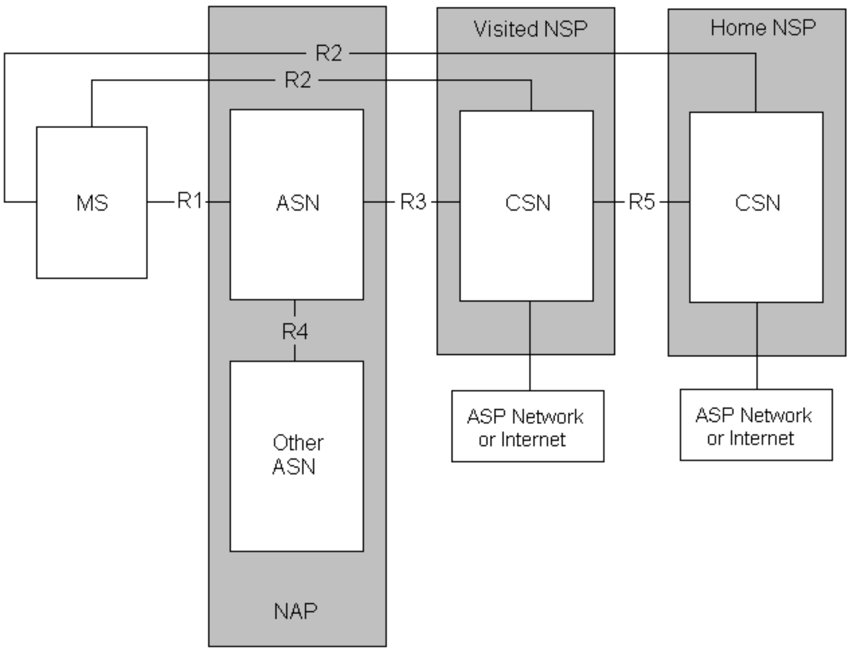


Figure 2-3 - WiMAX network reference model (extracted from [Nuay07]).

The ASN is the usual access network, comprising elements such as the BS, to which MSs - or Subscriber Terminals (ST) like sometimes named in the literature – connect. The ASN is defined as a complete set of network functions needed to provide radio access to a subscriber, providing some mandatory functions, such as connectivity with STs, transfer of Authentication, Authorisation and Accounting (AAA) messages to subscriber's Home Network Service Provider (H-NSP), network discovery and selection of the subscriber's preferred Network Service Provider (NSP), among others. The ASN is detained by a Network Access Provider (NAP), which provides radio access infrastructure to one or several NSPs. The NSP deploys the CSN, which comprises elements such as routers, AAA proxy/servers, user databases or interworking gateway devices. It is defined as a set of network functions that provide IP connectivity services to subscribers, enabling functions such as ST IP address and endpoint parameter allocation for user sessions, Internet access, AAA proxy or server, policy and admission control based on user subscription profiles, ASN-CSN tunnelling support, subscriber billing and inter-operator settlement, inter-CSN roaming and inter-ASN mobility.

WiMAX is a global telecommunications system based on the IEEE 802.16 standard for Wireless Metropolitan Networks (WMAN) with Broadband Wireless Access (BWA). The IEEE 802.16 BWA has two variants: IEEE 802.16-2004, which defines a fixed wireless access WMAN technology, and IEEE 802.16e, an amendment of the former approved in 2005, which introduces mobility and fast handover. The present work is focused in Mobile WiMAX, i.e., the 802.16e standard.

The fixed access is the first use of BWA, where an outdoor Customer Premises Equipment (CPE) realises the link between the BS and the MS. A Wi-Fi AP can be connected to the CPE to realise the so-called Wi-Fi network backhauling, Figure 2-4. A first evolution of the MS will be the case when it is no longer a CPE but a card installed in some laptop. A nomadic (or wireless) access is an access where the user may move in a limited area, e.g., in an apartment or a small campus. This area being covered by a BS, the communication session is interrupted whenever the user moves out of the zone, even if that zone is covered by a BS belonging to the same communications operator. The final expected step of WiMAX is a mobile access, where devices can move across all the cells in a seamless session. The cell-radius will be typically limited to 20 km.

Two types of network topology are supported: Point-to-Multipoint (PMP) and mesh. The first WiMAX network deployments are though planned to follow mainly PMP topology.

For WiMAX, the IEEE 802.16 standard considers the following typical frequencies: 2.5, 3.5 and 5.4 GHz. Both licensed and license-exempt bands can be used. Although the 2.5 GHz band is available in Europe since October 2007, it is expected that the 3.5 GHz band will be the most widely used, in particular a frequency band of 200 MHz between 3.4 GHz and 3.8 GHz.

Adaptive modulation is used, with four modulations supported: BPSK, QPSK, 16-QAM and 64-QAM. The more efficient 64-QAM modulation is used when the radio link is very good, the more robust BPSK being used with a poor radio link quality. Data rates between 10 and 100 Mbps (exceptional channel conditions) are to be obtained, but it is expected that typical values will be closer to the lower limit. Like in HSDPA, HARQ is used in WIMAX [EMPS06].

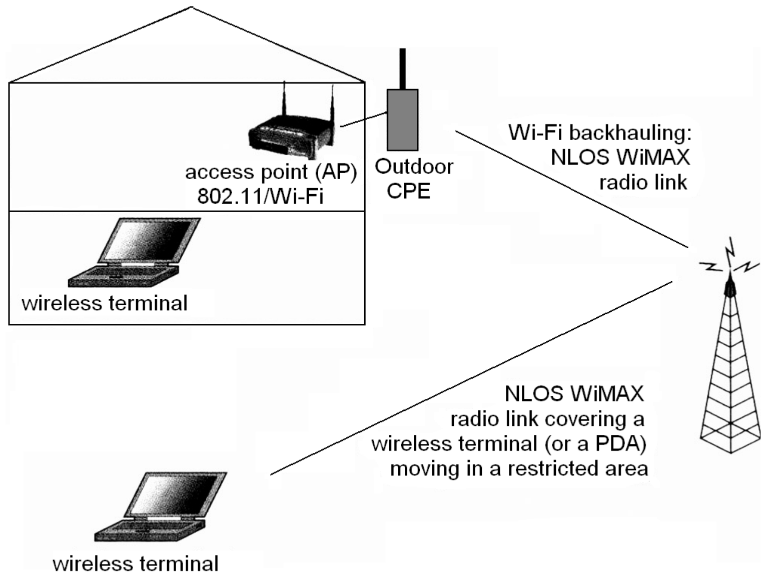


Figure 2-4 – Nomadic or portable BWA (extracted from [Nuay07]).

2.3.2 Radio Interface

The 802.16e standard specifies mainly 4 different air interface procedures, [IEEE04b]:

- WMAN Single Carrier (SC), operating in [10, 66] GHz in either FDD or TDD.
- WMAN Orthogonal Frequency Division Multiplexing (OFDM), operating below 11 GHz, also in either FDD or TDD.
- WMAN OFDMA, operating below 11GHz, in either FDD or TDD.
- Wireless High-Speed Unlicensed Metropolitan Area Network (HUMAN), operating below 11 GHz in license-exempt bands in TDD only.

OFDM is a special form of Multi-Carrier Modulation (MCM), and brings an obvious advantage compared to a traditional Single Carrier Modulation (SCM), which is the increased robustness against frequency selective fading and narrowband interference: in SCM, a single fade or interferer can cause the entire link to fail, while in MCM, only a small percentage of sub-carriers is affected, Figure 2-5.

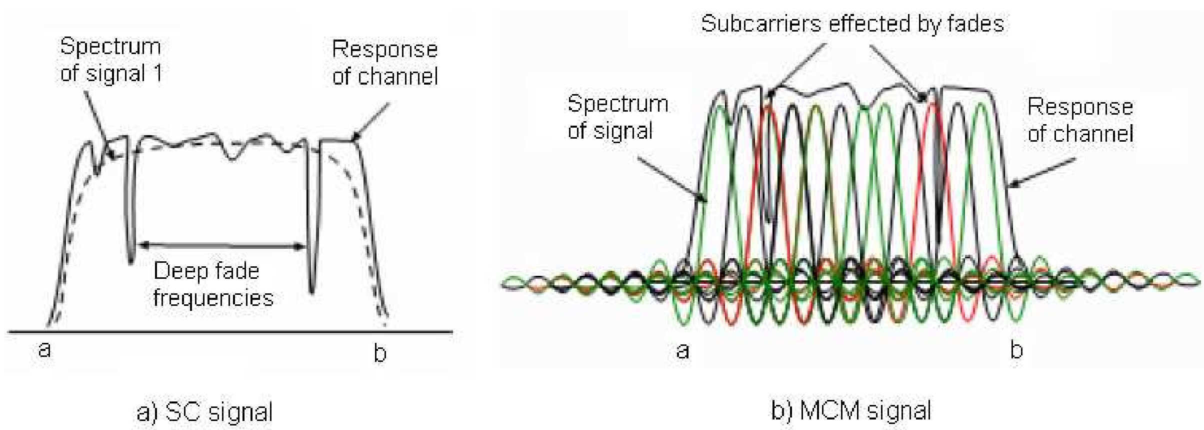


Figure 2-5 - Comparison between multi-path frequency selective fading influence on SC and MCM signals (extracted from [Antu07]).

OFDM differs from OFDMA in the sense that in OFDMA packets can be scheduled for different users on both frequency (sub-channels) and time (symbols) domains. With OFDMA, multiple users are allowed to transmit simultaneously on the different sub-carriers per time symbol, thus, the probability that all users experience a deep fade in a particular subcarrier is very low. OFDMA provides higher granularity in resource allocation, more degrees of freedom in scheduling, and improved fairness and QoS. In the present thesis, WiMAX takes OFDMA as the access and multiplexing mode for the PHY layer, since the 802.16e amendment was developed to cover mobile applications, which is boosted by OFDMA adoption instead of OFDM. OFDMA is capable of providing the flexibility to deal with diversified scenarios and challenges associated with rapidly moving mobile users in a NLoS environment, [Alva06].

OFDM can be realised via Inverse Fast Fourier Transform (IFFT), which enables a large number of sub-carriers with relatively low complexity. However, while in OFDM there is no user multiplexing across multiple sub-carriers within the same time slot, in OFDMA resources are available in both time and frequency domains, by means of OFDM symbols and sub-carriers, respectively. These time and frequency domain resources can be organised into sub-channels for allocation to individual users.

OFDMA scalability is supported by adjusting the Fast Fourier Transform (FFT) size, N_{FFT}, while keeping a constant subcarrier frequency spacing, which minimises the impact to higher layers [WiMA06a]. Table 2-1 summarises the main parameters that characterise SOFDMA (Scalable OFDMA), for some of the supported channel bandwidths. Although the standard refers to all channel bandwidths listed in Table 2-1, the system bandwidths in the initial planned profiles developed by the WiMAX Forum Technical Working Group for Release-1 are mainly 5 and 10 MHz.

Table 2-1 - Main SOFDMA Parameters.

System Bandwidth [MHz]	**1.25**	**5**	**10**	**20**
FFT Size (N_{FFT})	128	512	1024	2048
Subcarrier Frequency Spacing [kHz]	10.94			

Both TDD and FDD can be used. With FDD, different modulation types can be used for UL and DL carriers. With TDD, a fixed duration frame is used with different physical slots for UL and DL. The frame is not divided into two equal parts, in the sense that the split between UL and DL is adaptive and controlled at higher layers in the system. Therefore, it can be said that the TDD mode is more suitable for asymmetrical transmissions, like streaming ones.

When TDD is adopted, the WiMAX 802.16e frame is divided into DL and UL sub-frames. The frame structure is illustrated in Figure 2-6, being composed of:

- The Preamble, which is the first symbol of the frame, and is used for synchronisation.
- The Frame Control Head (FCH), which provides information related to the frame configuration, including length of Media Access Protocol (MAP) messages, allocated sub-channels and respective coding schemes.
- The DL-MAP, which provides additional information on DL sub-channel allocation, as well as

additional control information for the DL sub-frame.

- The UL-MAP, which has similar goals to DL-MAP, but related specifically to UL.
- The UL Ranging sub-channel, which is used by the ST to perform several adjustments, including closed-loop time, frequency and power adjustment, also being used for bandwidth requests.
- The UL Channel Quality Indicator Channel (CQICH) is used by the ST for feedback about the radio channel quality and reception, which is used by the BS to evaluate the specific conditions that each ST is experimenting, and based on that, scheduling options, proper modulation and coding schemes can be decided by the BS. The items available to report radio link quality include Physical Carrier-to-Interference-plus-Noise Ratio (CINR) and effective CINR.
- The UL ACK is used by STs to provide feedback on DL HARQ processes, particularly acknowledgement or not of the received data blocks.

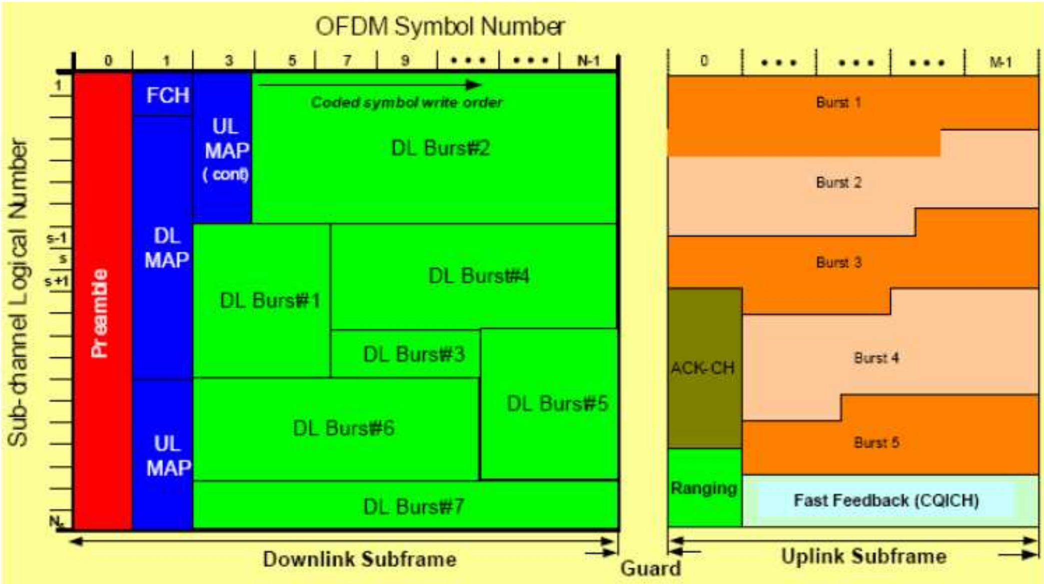


Figure 2-6 - TDD Frame Structure (extracted from [WiMA06a]).

Both DL-MAP and UL-MAP messages are critical for scheduling multiple users within the same radio frame: every ST addressed in the same TDD frame needs to be fully aware of which part of the DL subframe it must listen. The DL-MAP message contains, among others, special IEs for that purpose ([IEEE04b]), including used modulations, coding schemes, identification of target users, symbol and sub-channel offsets within the frame, etc.

As for each addressed ST in a single frame a specific DL-MAP IE is needed, if more users are to be simultaneously addressed in the same frame, more IEs are included in the frame, thus, more overhead exists, which needs to be accounted for in overall system performance. Additionally, the DL-MAP may be repeated in order to ensure higher decoding probability. The standard specifies a repetition factor of 1, 2, 4 or 6. The trade-off is simple: a higher repetition factor increases correct decoding probability at the expense of increased overhead, i.e., wasting resources; a lower repetition factor minimises MAP overhead, but decoding errors may happen. The present work considers a factor of 2, i.e., DL-MAP overhead is actually duplicated, which represents a fair trade-off between MAP overhead and reliability of correct decoding of MAP IEs content, [WiMA06a]. It is also assumed that DL-MAP is QPSK ½ modulate and coded, in order to increase its reliability. The MAP message size is variable, since it depends on the number of allocated users in a frame.

There are three types of OFDMA sub-carriers: data sub-carriers for data transmission, pilot sub-carriers for channel estimation and synchronisation purposes, and null sub-carriers, for guard bands. Among these sub-carriers, the active ones (data and pilot) are grouped into subsets called sub-channels. The minimum frequency-time resource unit of sub-channelisation is one slot, which is equal to 48 data tones.

The sub-carriers forming one sub-channel may be adjacent, although this is not absolutely necessary. Additionally, pilot allocation may be performed differently, depending on the subcarrier allocation mode used [Yagh04]:

- In DL Fully Used Sub-Channelisation (FUSC), pilot sub-carriers are allocated first, the remaining sub-carriers being divided into sub-channels. In this case, the pilot sub-carriers are used from a common set. Sub-carriers can be scattered throughout the frequency channel range.
- In DL Partially Used Sub-Channelisation (PUSC), the set of used sub-carriers, data and pilot, is first divided into sub-channels, with pilot sub-carriers being allocated within these. DL PUSC uses a cluster-based structure that spams over two OFDM symbols (in time domain) of fourteen sub-carriers, each one with a total of four pilot sub-carriers per cluster. Several scattered clusters of sub-carriers can thus be used to form a sub-channel.
- UL PUSC is similar to DL PUSC, but a tile structure is used instead, which spams over three symbols (in time) of four sub-carriers, each one with a total of four pilot sub-carriers.
- Optional DL FUSC (OFUSC) and Optional UL PUSC (OPUSC) can also be employed, being similar to “standard” DL FUSC and UL PUSC, but making use of a larger number of sub-carriers thanks to different pilot allocation mappings.
- In DL and UL AMC, adjacent sub-carriers are used to form sub-channels.

The IEEE 802.16 standard specifies that, for the DL, only PUSC is mandatory, due to which the present thesis only considers the support of PUSC. Table 2-2 summarises how the available sub-carriers are divided among data, pilot and guard sub-carriers for N_{FFT} of 512 and 1024.

Table 2-2 - DL PUSC Sub-carrier allocations.

N_{FFT}	**1024**	**512**
Number of guard sub-carriers	183	101
Number of used sub-carriers	841	421
Number of pilot sub-carriers	121	61
Number of data sub-carriers	720	360
Number of sub-channels	30	15
Number of data sub-carriers per sub-channel	24	

In DL PUSC, the minimum allocation unit is the slot, which spams over 2 symbols in the time domain and over 1 sub-channel in the frequency one, i.e., over 24 data sub-carriers. This is the minimum DL allocation unit that can be assigned to a user within a radio frame and every resource allocation within a frame is made in multiples of this unit.

Cell coverage in WiMAX depends naturally on the used modulation and coding rate. The less robust modulation and coding rates demand higher CINR, in the sense that a low interfered channel is required. On the contrary, lower CINR implies the use of more robust modulation and coding rate combinations, Table 2-3. This represents a trade-off between cell radius and throughput.

Table 2-3 - Receiver SNR assumptions (extracted from [IEEE06]).

Modulation	Coding Rate	Receiver SNR [dB]
QPSK	½	5.0
	¾	8.0
16-QAM	½	10.5
	¾	14.0
64-QAM	½	16.0
	2/3	18.0
	¾	20.0

2.3.3 RRM Aspects

Spectral efficiency and QoS are enhanced by RRM functions, like Admission and Congestion Control, Traffic Classification, Shaping and Policy and Traffic Scheduling. The WiMAX Forum [WiMF09] defines a framework to support and optimise the Admission Control, which algorithms are left to manufacturers. In particular, it must be ensured that enough radio resources are available at the BS to serve appropriately a new ST or connection or service flow (either at service request or after a handover). Congestion Control is also an important function, which acts similarly to Admission Control, but applied to on-going connections, discarding packets in case of overload.

Another relevant RRM function is Traffic Classification. In order for the network to be able to provide different treatment based on QoS requirements, it must be able to classify the incoming packets. Packets can be queued, delayed, discarded or re-classified in terms of priority by Shaping and Policy functions.

Finally, Traffic Scheduling is responsible for determining the transmission order of incoming packets, according to the defined QoS requirements, especially when fluctuations in the incoming rates result in traffic queuing. In [Antu07], the three generic traffic scheduling algorithms [RSAD05] are applied in order to treat differently the co-existence of a service mix and the different radio conditions felt by each user: Round-Robin (RR), Maximum Throughput (MAX) and Proportional Fair (PF).

RR is a simple and easy to implement algorithm that handles users alternatively, without concern for the particular radio conditions felt by each user. Thus, most likely, e.g., a VoIP user that suffered a packet lost in a frame will successfully be allocated system resources in the next frame. In the same way, a WWW user whose last packet was delayed will most likely have a successful allocation on the next frame. This algorithm is simple, but it does not maximise system capacity, as it does not give preference to users feeling the best radio conditions at each frame. Additionally, as in each frame the different users are prioritised in a sorted order, there is no QoS implemented, e.g., VoIP services are not prioritised over FTP ones.

MAX takes advantage of prioritising users that experiment a better AMC at that particular moment. The algorithm searches for those users and attempts to serve them in first hand, without concern for the different QoS priorities among services (for instance, a VoIP user suffering a bad AMC is served later than any WWW user). This algorithm maximises system capacity, but it can lead to extreme situations in which users that are constantly affected by bad radio conditions may have to wait a long time before successfully being allocated resources, resulting in unfair situations. Additionally, there are no QoS guarantees for any user, either Real-Time Polling Services (RTPS) or Best Effort (BE) based, meaning that again neither Admission Control nor Congestion Control is applied to RTPS calls.

PF, like MAX, looks for users experiencing the best radio conditions, but in order to avoid that users that experiment long-term worse radio conditions remain long periods without any assignment, the algorithm keeps track of the recent delays or packet losses felt by each user, and overrides the search for the better AMC in such situations, providing some fairness to the system.

In order for the BS to take appropriate decisions for the selection of the modulation and coding scheme, a set of channel quality indicators is defined. Two families of indicators are available: Received Signal Strength Indicator (RSSI), which gives information on the received power level, and CINR, which gives information on received carrier-to-interference levels. The support of Power Control is also mandatory in the UL. By measuring the received power at the BS, the BS can send power offset indications to the ST, which adjusts its transmit power level accordingly.

Dynamic Frequency Selection (DFS) mechanisms may also be required in the case of deployment of a system in a licence-exempt band (e.g., 5.8 GHz). In that case, the BS and the MT implement a set of mechanisms that allows: sounding the radio environment prior to the use of a channel; periodically detecting “specific spectrum users” (a specific spectrum user is a user that has been identified by the regulator as requiring strict protection from harmful interference); discontinuing operation on a channel after detection of a specific spectrum user; scheduling of periodic sounding testing periods (from the BS and the MT); selecting/changing to a new channel.

2.4 General Comparison of Systems

Many differences can be found between UMTS and WiMAX, the most significant one being the way the radio channel is multiplexed by the two systems. Despite of those differences, UMTS and WiMAX are very similar in terms of network architecture, both being based on a cellular structure. Mobility is therefore a key characteristic of the two systems, making them less competitive with others in terms of throughput, like Wi-Fi for example. In fact, Wi-Fi can be seen as an alternative to traditional cellular wireless systems when throughput is the key issue. The trade-off is naturally its very limited mobility, making it ideal to be used in scenarios like university campus, shopping centres and airports. Moreover, Wi-Fi can also be used as a complement of other systems, like WiMAX for example, where a Wi-Fi AP can make the network backhauling of an outdoor WiMAX CPE as depicted in Figure 2-4.

In terms of network architecture, UMTS presents a structural difference compared to Wi-Fi and WiMAX, since its access network is composed by two hierarchical layers: the Node B and the RNC. Table 2-4 compares the network elements of the three architectures, while a general comparison of the main characteristics of the three systems is presented in Table 2-5.

Table 2-4 – Network architecture comparison between UMTS, Wi-Fi and WiMAX systems.

System	**UMTS**	**Wi-Fi**	**WiMAX**
Mobile Terminal	UE	CS	ST
Access Network Interface Equipment	Node B	AP	ASN
Network Interface Controller	RNC	none	none
Connectivity Network Interface	CN	DS	CSN

Table 2-5 – General comparison between UMTS, Wi-Fi and WiMAX systems.

Main Characteristic	**UMTS**	**Wi-Fi**	**WiMAX**
System Architecture	cellular	hotspot-based	cellular
Services nature	Data	Data	Data
Mobility	High	Low	Medium
Maximum Throughput [Mbps]	14	54	100
Frequency Bands [GHz]	2	2.4, 5.2 and 5.8	2.5, 3.5 and 5.4
Spectrum Licence	Regulated	Free	Free/Regulated
Multiple Access	WCDMA	CSMA-CA	(O)FDMA/TDMA

2.5 Services and Applications

In this section, services and applications supported by the three systems are briefly characterised.

Second generation systems, like GSM, were originally designed for efficient delivery of voice services over switched circuits. UMTS networks are, on the contrary, designed from the beginning to support a wide range of packet switched data services. Distinct UMTS data services have different sensitivity to delay aspects, and are classified by 3GPP according to four QoS classes: Conversational, Streaming, Interactive and Background [HoTo04].

The Conversational class is characterised by low end-to-end delay and symmetric or nearly symmetric traffic between UL and DL in person-to-person communication services, and applications like voice over IP and video telephony, with low-emphasis on signal quality. The maximum end-to-end delay is given by the human perception of video and audio conversation. As an example, it is considered that a conversation only seems fluent when delays are lower than 400 ms.

In the Streaming class, information is transported in a continuous stream, allowing its processing by the end user (e.g., visualisation) before the reception of the entire file is finished. It is shaped for highly asymmetric traffic data applications, like video or audio streaming on demand. The usage of buffers allows this class to be more tolerant to delay, compared to the Conversational one. The human perception to the delay variation is the main requirement.

In the Interactive class, the user can ask for different kinds of information from a certain remote server. Services in this class are generally more tolerant to delays, therefore, with low round-trip delay, and generate asymmetric traffic. On the other hand, a high signal quality is demanded, since there is no tolerance to errors in the received information. The main service examples in this class are web browsing, data base retrieval, geographic location applications, and server access.

The Background class is characterised by the fact that the user does not have a limited time to receive the information, so the system does not need to process the information immediately, which allows delay to be high. Therefore, applications in this class use the network for the transmission of information when its resources are not being used by other applications from other service classes. Despite of the delays, the information transmitted should not contain errors. Examples of applications are E-mail, SMS, MMS and other non-real time asymmetric services that require high signal quality and are very sensitive to errors.

Table 2-6 summarises the characterisation of the four UMTS traffic classes.

Table 2-6 - UMTS traffic classes (based on [3GPP09]).

Class	Conversational	Streaming	Interactive	Background
General Characteristics	Symmetric traffic, delay sensitive, low-emphasis on signal quality	Asymmetric traffic, delay variation sensitive, not sensitive to transmission errors	Asymmetric “request-response” traffic, low round trip delay, high signal quality required	Asymmetric, non-real time traffic, high signal quality required
Transfer Delay	Minimum and fixed	Minimum and variable	Moderate and variable	Large and variable
Real Time	Yes	Yes	No	No
Bandwidth	Guaranteed bitrate	Guaranteed bitrate	No guaranteed bitrate	No guaranteed bitrate
Applications	Voice, video telephony, interactive games	Audio streaming, video on demand	Voice messaging, FTP, Web browsing	E-mail, SMS, MMS, FAX

Although the 3GPP classification in Table 2-6 is mainly directed to cellular networks, it can be used as a starting point to establish services differentiation in Wi-Fi systems. In fact, several proposed classifications exist, like the one in 802.1D standard [IEEE04a], which is suited for wired LANs, but also adopted by the 802.11e [IEEE05] standard for QoS in WLANs. Seven traffic types are defined in IEEE 802.1D, as follows:

- Network Control – Represents the traffic necessary to maintain and support the network infrastructure. Must have priority over the rest.
- Voice – Characterised by requiring very low delays.
- Video – Requires also low delays, but not as severe as with the Voice traffic type.
- Controlled Load – Represents applications with some sort of admission control and with controlled throughput.
- Excellent Effort – A best effort traffic with higher priority than the lower classes.
- Best Effort Traffic – The typical LAN traffic.
- Background – Traffic that is allowed on the network, but that should not interfere with the traffic from any of the other classes.

In some scenarios, e.g., 802.11e networks, a different number of Traffic Classes (TCs) can be available on a given network, and, sometimes, some of the defined traffic types must be grouped together into a single class. This problem is also addressed in the standard, some possible solutions being proposed, depending on the number of classes available within the network. A mapping between various TCs and the traffic types defined, including a spare type, is also identified in the standard, as Table 2-7 shows. According to the standard, this spare type is to be used for scenarios where it may be advantageous to have an additional traffic type, similar to best effort, to support bandwidth sharing management for bursty data applications.

Table 2-7 - Mapping between User Priority and traffic types (based on [IEEE04b]).

Class/User Priority	**1 (minimum priority)**	**2**	**0 (default)**	**3**	**4**	**5**	**6**	**7 (maximum priority)**
Traffic Type	Background	Spare	Best Effort	Excellent Effort	Controlled Load	Video	Voice	Network Control
Applications	Traditional applications of default IP networks			Bulk data, video streaming	Transaction data, interactive	Video Conference	VoIP	Signaling

Due to the fast radio interface capacity to handle balancing between UL and DL, as well as fast changing radio conditions, the 802.16e MAC layer is capable of providing flexible mechanisms that can meet the QoS required for a wide range of services and applications. The 802.16 standard has been developed to address a wide range of applications [WiMA05], defining classes as summarised in Table 2-8.

Comparing the three systems, one can observe that the QoS classification has been defined on a traffic basis for UMTS, while for WiMAX it is more based on the applications perspective. In a general way, there is an almost direct matching among the classes of each system, considering the resemblance between UGS and Conversational, rtPS and Streaming, nrtPS and Interactive and BE and Background. According to Table 2-9, where each service type is classified according to the system, there are some exceptions to this. While, for instance, Speech, VoIP and Video telephony are classified as Conversational and UGS, other examples, like FTP, are considered Interactive in UMTS and Best Effort in WiMAX.

Table 2-8 - WiMAX service classes (based on [WiMA05]).

Class	Unsolicited Grant Service (UGS)	Real-Time Polling Service (rtPS)	Non-Real-Time Polling Service (nrtPS)	Best Effort (BE)
General Characteristics	Real-time data streams, fixed-size data packets issued at periodic intervals	Real-time data streams, variable-sized data packets issued at periodic intervals	Delay-tolerant data streams, variable-sized data packets, minimum data rate required	Data streams, no minimum service level required, handled on a space-available basis
Transfer Delay	Minimum and fixed	Minimum and variable	Moderate and variable	Large and variable
Real Time	Yes	Yes	No	No
Applications (Bandwidth in kbps)	VoIP (4-64), Video telephony (32-384), interactive games (50-85)	Video clips (20-384) Music (5-128), Movies streaming (>2000)	Instant messaging (<250 byte messages), Web browsing and E-mail with attachments (> 500)	Bulk data, Movie download (>1000), FTP (>500)

Table 2-9 – Service classification according to the system.

Service \ System	UMTS	Wi-Fi	WiMAX
Speech/VoIP	Conversational	6 (Voice)	UGS
Video telephony	Conversational	5 (Video)	UGS
Video/Audio Streaming	Streaming	3 (Excellent Effort)	rtPS
Web Browsing	Interactive	0 (Best effort)	nrtPS
E-mail	Background	0 (Best effort) / 1 (background)	nrtPS
FTP	Interactive	0 (Best effort) / 1 (background)	BE

However, there are significant differences in terms of bandwidth assigned to each class. For UMTS, the difference between Conversational and Streaming classes is the delay sensitivity, the bandwidth being the same, whilst for WiMAX different bandwidths are assigned to different services, even if those services belong to the same service class. It can be stated as well that the all-IP approach is somehow more effective in WiMAX than in UMTS, the former assuming the support of VoIP.

Compared to Wi-Fi, the QoS classification for UMTS and WiMAX can be considered generic. Like in UMTS, the QoS classes of Wi-Fi are defined on a traffic basis, but the classification is far more specific since a wider range of traffic classes is considered.

2.6 Common Radio Resources Management

2.6.1 Levels of Integration

CRRM strategies [RSAD05] are intended to achieve an efficient usage of the radio resources in a HN (Heterogeneous Network), by means of a coordination of the available resources in the existing RATs. RRM entities carry out the management of the resources in a pool of a certain RAN, while a CRRM entity executes the coordinated management of the resource pools controlled by the RRM entities. From the functional point of view, different degrees of interaction exist between CRRM and local RRM entities. The first aspect to take into account is the specification of the master of the different decisions. One possibility is that the CRRM simply advises the RRM entity, so that the local RRM is mainly responsible for the decisions taken, and therefore, the master. Another possibility is that the CRRM decisions are binding on the local RRM entity, so that the CRRM is the master and responsible for these decisions. On the other hand, the split of functionalities between RRM and CRRM also depends on the degree of interaction between both entities, often referred as RRM-CRRM coupling. The lowest degree of interaction is shown schematically in Figure 2-7, where the CRRM only dictates policies for RRM operation.

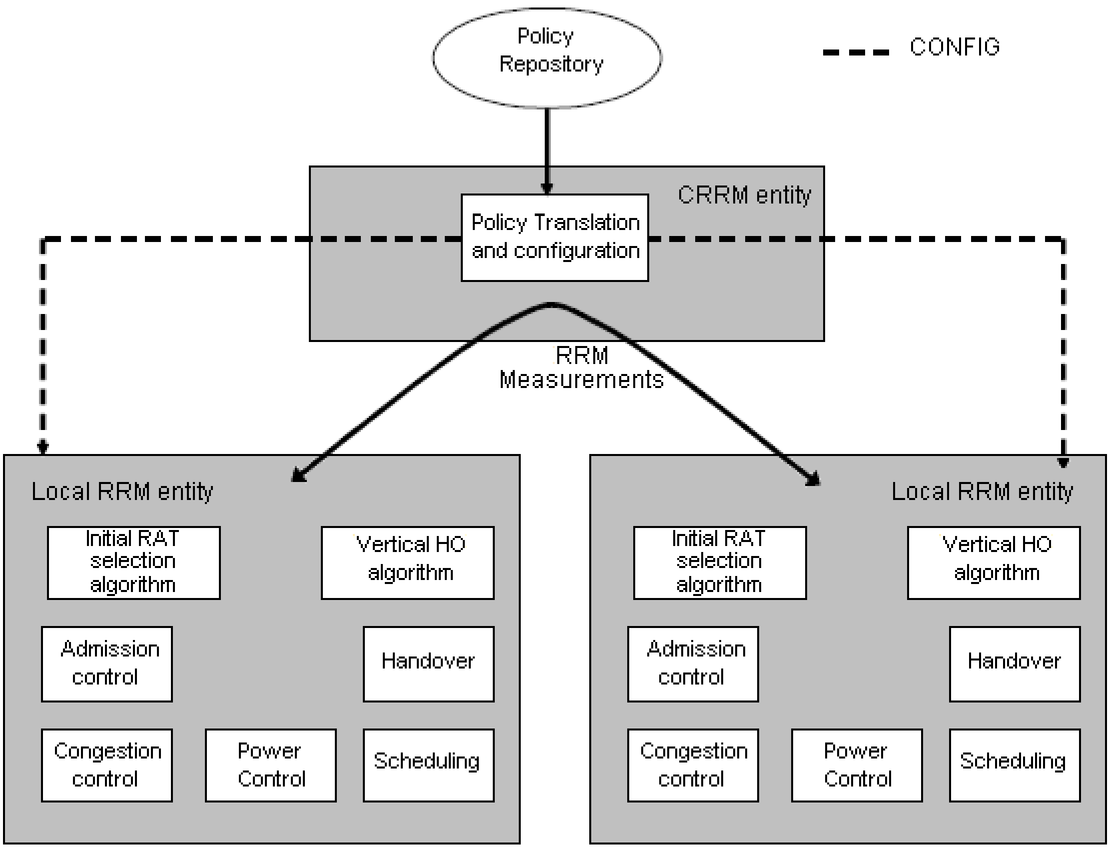


Figure 2-7 - Low integration degree between CRRM and local RRM functions (extracted from [RSAD05]).

A policy can be defined as a high-level declarative directive that specifies some criterion to guide the behaviour of a network responding to some network operator preferences. In this approach, the CRRM is considered simply as a policy consumer that translates the specific policies into an adequate configuration of the RRM algorithms, almost all functionalities residing in the RRM entity.

There are other approaches with a higher degree of integration between RRM and CRRM entities, like the intermediate interaction degree, where the CRRM not only provides the policies that configure the local RRM algorithms, but it is also involved in the RAT selection and vertical handover (inter-RAT handover) algorithms by deciding the appropriate RAT to be connected. The RAT selection, either during vertical handover or in the initial RAT selection case, will respond to the specific policies, e.g., establishing correspondences between RATs and different services or user types. Some examples would be to allocate Web browsing services to WiMAX and Voice services to UTRAN, or to allocate business users to WiMAX and consumer users to UTRAN. Similarly, other possible policies for selecting the adequate RAT could be related to having load balancing between the different access networks or could respond to congestion situations detected in any of the access technologies, for which status measurements are necessary.

Figure 2-8 shows an even stronger interaction between CRRM and RRM.

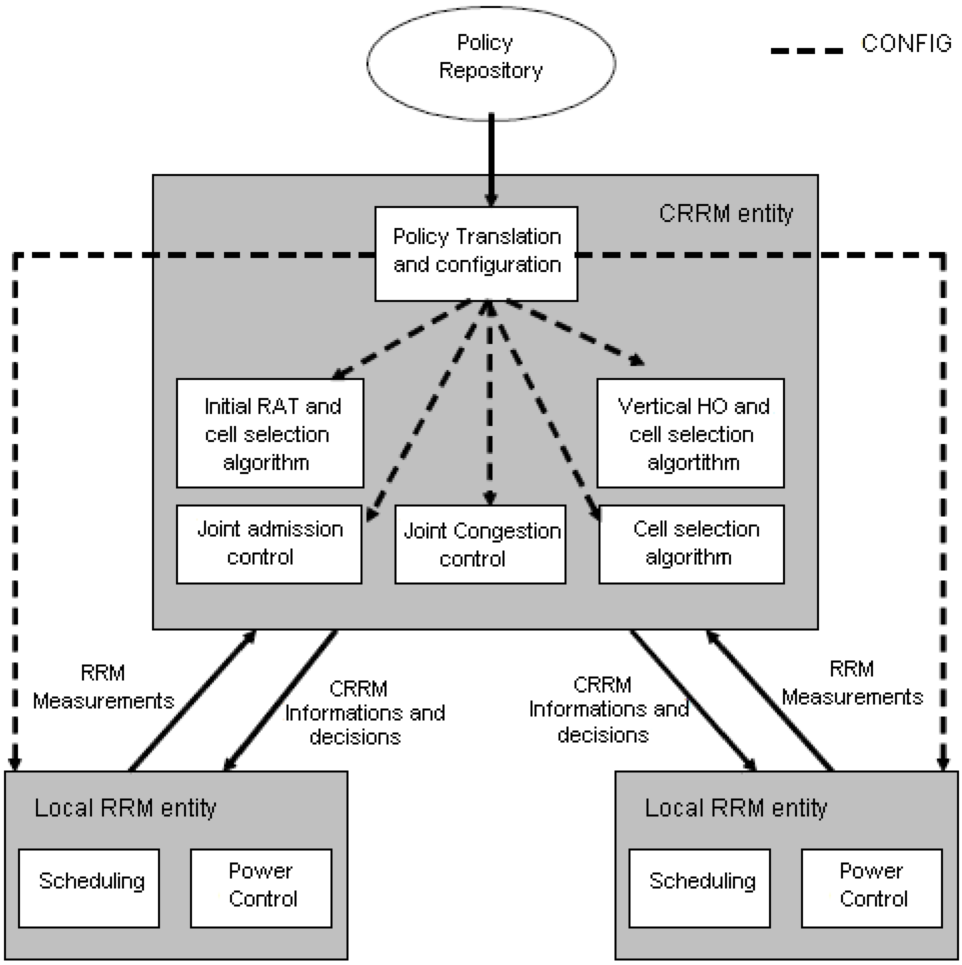


Figure 2-8 - High Integration degree between CRRM and local RRM functions (extracted from [RSAD05]).

In this case, CRRM is involved in most of the local RRM decisions, so interaction between both entities is expected to occur at a shorter-term scale than in previous cases. The CRRM is involved in each intra-system handover procedure and requires a more frequent measurement exchange. Similarly, joint congestion control mechanisms are envisaged to avoid overload situations in any of the underlying access networks.

The highest degree of interaction between CRRM and local RRM is the introduction of joint scheduling algorithms that take the status of the different networks into account to allocate resources to the most appropriate RAT. This solution is shown schematically in Figure 2-9, where the local RRM functionality remains at a minimum, limited to the transfer of adequate messages to CRRM and some specific technology dependent procedures that occur in very short periods of time (e.g., inner loop Power Control in the case of UTRAN, which occurs with periods below 1 ms).

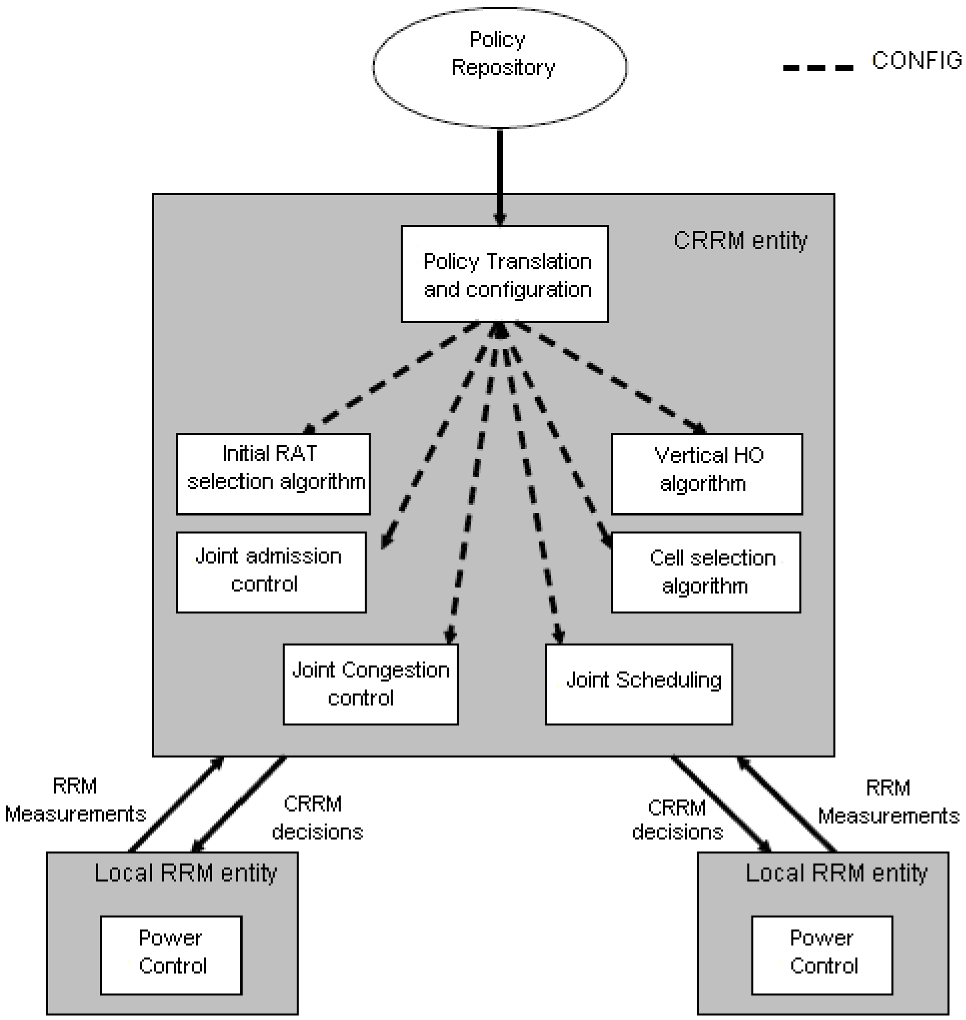


Figure 2-9 - Very high integration degree between CRRM and local RRM functions (extracted from [RSAD05]).

This solution requires CRRM decisions be taken at a very short time scale in the order of milliseconds, with the possibility of executing frequent RAT changes for a given MT. Consequently, this poses hard requirements to the reconfigurability capabilities of MTs, and can be regarded as a long-term implementation of CRRM functionalities.

The CRRM/RRM integration degree is a recent subject, born from the market circumstances of nowadays, in which telecom companies are at the same time challenged and compelled to integrate separate technologies, which were designed from the beginning to work on a stand-alone basis. The presented integration degrees probably represent the expected stages of evolution of HNs management, the final target being to achieve a very high integration degree, in which there is only room available for a generic, seamless, ubiquitous network, providing services to its customers regardless of the supporting technology.

2.6.2 CRRM related work

Over the last years, a large number of researchers have been developing work on the subject of CRRM and HNs. In order to evaluate the richness and diversity of approaches, methods and algorithms being proposed, a literature survey was performed. The main results of the survey are hereby briefly described.

A great part of the work in the CRRM field is devoted to handover in HNs, more particularly, inter-RAT handover, also defined by Vertical Handover (VHO). Tawil et al. [TaPS08] introduce a distributed VHO decision scheme, which combines a method called Multiple Attribute Decision Making method and Simple Additive Weighting in a distributed manner. The scenario is a typical HN, composed of UMTS and WLAN. In this context, the "best" network is selected heuristically using a network selection function based on HO criteria like bandwidth, cost and dropping probability.

Liu et al. [LMSB08] propose interworking and VHO mobility management schemes in order to realize a seamless VHO between UMTS and WiMAX, Figure 2-10. A tight coupling architecture is considered. A novel common interworking sub-layer is proposed, which is focused on eliminating packet loss and reducing HO latency that are common problems for most VHO scenarios. Simulations using an extended version of the NS2 simulator [NeSi08] are performed to analyse the performance of such common sub-layer.

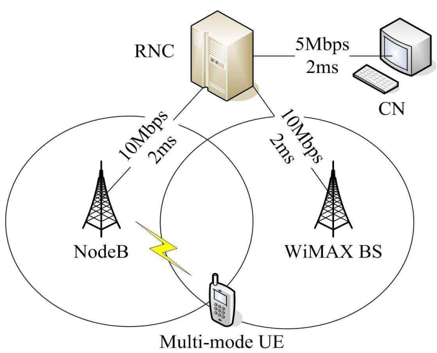


Figure 2-10 – Simulation Topology (extracted from [LMSB08]).

Baek et al. [BKSH08] propose a network initiated HO mechanism, based on the 802.21 framework [IEEE09b], which makes use of the resource information of the network to provide QoS service continuity in a UMTS/802.16e VHO. The HO procedure consists of three steps; QoS measurement, passive reservation, and activation.

In [YaGQ08], Yang et al. present an analytical framework for soft VHO, and a context-aware soft VHO algorithm for HN with UMTS and Wi-Fi. The algorithm considers system context parameters, such as user required bandwidth, user traffic cost, access network utilisation, and signal to interference-and-noise ratio.

Some other researchers developed work in HNs according to an economic perspective applied to CRRM. Zeng et al. [XZVY08] propose an auction mechanism based scheme for JRRM in a heterogeneous wireless multi-operator network. The idea is that the operator with spare resources is considered to be a seller in an auction while the operator in need of resources is regarded as a bidder. The objective is to increase the resources utilisation ratio and to lower the total session blocking probability. Network operators with spare resources would maximise their profits not only by striving for more and more users, but also by leasing the spare resources to other operators. The proposed auction mechanism is tested using a simulator with a scenario consisting of three operators, each of which running overlapped UMTS and Wi-Fi.

In the same line, Barbaresi and Sallent [BaSa08] present an analysis of the potential economic benefits of applying CRRM traffic steering strategies between GSM and UMTS. The CRRM algorithm under investigation was designed according to the “fittingness factor” framework [RoSA07], which helps in selecting the most suitable RAT according to the requested services and the instantaneous load conditions of the HN.

Serrador and Correia [SeCo07] present a method for cost estimation based CRRM. Heterogeneous cellular networks are analysed by taking several Key Performance Indicators (KPIs) into account, their proper balance being required in order to guarantee a desired QoS. An approach to integrate a set of KPIs into a single one is presented, by using a Cost Function that includes these KPIs, providing a single evaluation parameter as output, and reflecting network conditions and CRRM strategies performance. The proposed model enables the implementation of different CRRM policies by manipulation of KPIs according to the perspective of the HN operator. The method is simulated using a CRRM simulator integrating UMTS and Wi-Fi.

Coupechoux et al. [CoKG08] describe a joint RRM algorithm that assigns an MT to a given RAT, while taking into account the joint spatial distribution of already accepted MS, the current load of each RAT, the location of the newly accepted session and its influence on the global performance. In a study based on the Semi Markov Decision Process theory, it is shown how to obtain an optimal policy, both in the perspective of user satisfaction and minimisation of operator resources. The scenario is a micro- or femto-cell with two co-localised RATs, e.g. Wi-Fi and UMTS.

The list of contributions to multi-RAT integration is vast and diversified. In [YNCF07], Chen et al. propose a utility function-based access selection method, to determine the access network selection in heterogeneous UMTS/Wi-Fi networks. The utility function considers system QoS requirements, MT mobility and cells load balancing. The method is based on simple principles like assigning Wi-Fi to accommodate low mobility MTs with high transmission rate, and UMTS to accommodate high-mobility MTs requiring low transmission rates. A concatenated 19 cell environment is configured for simulating the heterogeneous network, Figure 2-11.

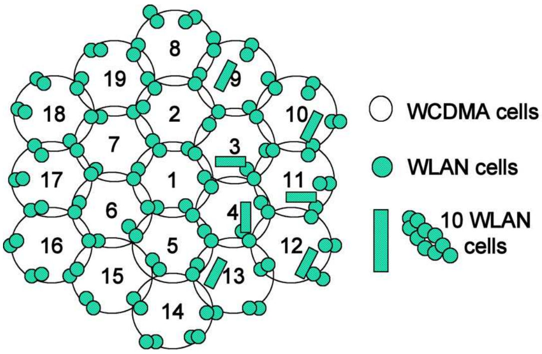


Figure 2-11 – The simulation environment (extracted from [YNCF07]).

Piao and David [PiDa07] developed a CRRM algorithm for non-real-time (NRT) services in a heterogeneous WiMAX/UMTS/GSM scenario. The algorithm uses a parameter, the NRT load information, which is mapped from the average user throughput onto the respective network. The performance level of the algorithm is assessed through a network, system-level simulator. The scenario is a WiMAX cell covering a hotspot area in the co-located GSM/EDGE and UMTS/HSDPA networks. The load redistribution, the mean data rate and the intersystem handover activity are investigated in three scenarios: a) completely separate systems, b) separate WiMAX and CRRM for GSM and UMTS, and c) CRRM for these three radio access networks. Simulation results have shown a promising gain of mean data rate in the last scenario.

Less common, eventually more exoteric contributions come from authors like Calabuig et al. [CMMC08], who propose a CRRM algorithm based on Hopfield Neural Networks to achieve “fast and sub-optimal solutions”. The generic formulation is particularized and evaluated through simulations with a UMTS/HSDPA and 802.11e WLAN coupled network scenario.

Bojan et al. [BaBB07] present an optimal RAN selection algorithm based on a TOPSIS method when solving the multi-criteria analysis. TOPSIS multi-criteria decision method is based on a concept that the chosen alternative should have the shortest Euclid distance from the ideal solution and the longest from the non-ideal solution. The RANs in this model represent the alternatives, while the network parameters (available bandwidth, QoS level, security level, data transfer cost) are considered as the criteria for determining the optimal network.

In [AnPi07], Antoniou and Pitsillides present a novel modelling approach for the network selection mechanism in 4G converged environments. The method is based on a game theoretic approach, specifically defining a game between the access networks participating in the 4G system themselves, competing in a non-cooperative manner to maximise their payoff. The game outcome is a decision of which subset of the set of service requests made to the 4G converged system should be admitted by each access network. The motivation for the decision of each access network is user satisfaction, fulfilling the user-centric paradigm on which 4G converged environments are based on.

Gozálvez et al. [GoES08] propose a set of CRRM policies designed to jointly manage the available RATs and their corresponding radio resources in HN. The proposed schemes are based on linear programming optimisation tools and QoS differentiation in multimedia traffic scenarios.

Finally, Ong and Khan [OnKh08], who present a dynamic access network selection algorithm. QoS parameters are in general non-stationary, however they can be considered stationary when observed over a short time. The motivation of the algorithm is to account for the effects of non-stationary components using sequential Bayesian estimation which relies on bootstrap approximation to estimate the *short-term* stationary data. A simulation model to evaluate the effectiveness of the algorithm was developed using OPNET Modeller, and simulations were performed considering a heterogeneous IP-based 4G architecture comprising UMTS, WLAN and WiMAX.

Most of the algorithms, techniques and methods hereby presented have a common topic: RRM centralisation. In [RSAN07], a different approach is presented. It addresses the implementation of decentralised RAT selection strategies at the MT in a heterogeneous wireless network scenario. This can be achieved if the network provides some information or guidelines to assist the MT in its decisions, making use of the IEEE P1900.4 protocol [IEEE09c] Simulation results show that the proposed decentralised strategy is able to outperform a reference centralised load-balancing strategy.

In fact, a greatly vast diversity of approaches can be found in literature dedicated to the CRRM subject. Several simulators of many types have been developed by authors to evaluate the effectiveness of their proposed models and algorithms. But one of the conclusions being taken from this survey is the reduced number of CRRM system-level simulators like the one developed for the current thesis. Moreover, the author of this thesis was able to find several HN simulators integrating combinations of UMTS R99, UMTS R5, Wi-Fi and WiMAX, usually addressing only two of those RAT, but a few number of system-level simulator integrating the four systems has been found in the survey. This fact brings additional value to the developed work for it addresses the CRRM subject from an almost unexplored and less common perspective, which is equally interesting.

Chapter 3

Algorithms and Simulator Description

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

3.1 Algorithms and Models

3.1.1 CRRM

The CRRM entity is, in general terms, responsible for the common radio resources allocation demanded by users. In the simulator, this entity is invoked whenever there is the need for allocating a BS to a MT. The resources allocation is performed according to a priority table, Table 3-1. The priorities for UMTS and Wi-Fi are based on the reference scenario of the IST-AROMA project [AROM07]. For WiMAX, it was not possible to reuse IST-AROMA results, as this system was not in the scope of the project, therefore, a single criterion was adopted to define WiMAX priorities: minimum priority (3) for “traditionally circuit-switched” services, medium priority (2) for “packet-switched” ones, except for Streaming, which is preferentially steered to a WiMAX BS (priority 1).

Table 3-1 - Service priority.

Service \ RAN	UMTS (R99)	UMTS (R5)	Wi-Fi	WiMAX
Voice	**1**	3	2	3
Video	**1**	2	3	3
Web browsing	3	2	**1**	2
Streaming	2	3	2	**1**
E-Mail	**1**	2	3	2
FTP	3	2	**1**	2

The priorities defined in Table 3-1 are the basis for the CRRM algorithm, Figure 3-1. Given a new call, the choice of the BS that will provide the necessary resources depends on the analysis of the information provided by three kinds of input: priority table, BS’s coverage, and BS’s load. Depending on the service, a call will be given resources from a BS of the system with the highest priority for that service, as long as that BS provides coverage and has enough resources. A BS from a system with priority n+1 is attempted should all BSs from the system with priority n be out of resources. In case no BS has available resources, that will result into a blocked call for Voice and Video services, or in a delayed connection (for a randomly defined number of time frames) in case of any other service.

Handover is another important mechanism in CRRM. In a HN environment, two types of HO can occur: the Horizontal HO (HHO) and the Vertical HO (VHO). HHO corresponds to the “classical” handover, processed at a cellular system level, being triggered by the MT’s movement and/or fading (signal strength level), traffic reasons (e.g., BS load balancing), or even in some cases by other

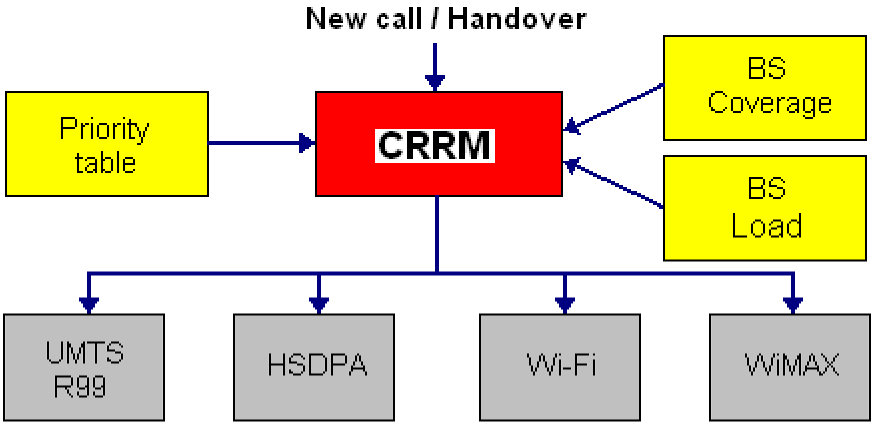


Figure 3-1 - CRRM algorithm.

particular operator policies. The VHO process is typically more complex, since decisions must consider parameters from more than one system, in order to optimise a given CRRM algorithm. In this thesis, HHO and VHO processes are handled by the CRRM entity, being treated as a unique process, where HHO can be seen as a particular case of VHO. As it happens in call admission, during a HO the CRRM entity also seeks for a “free” BS according to the priority table. In case it is not possible to allocate resources in a HO situation (out of coverage or full load) the system drops the call.

3.1.2 UMTS and Wi-Fi

The purpose of Section 3.1.1 was to describe the CRRM model in the base of the simulator. The current section is focused on the UMTS and Wi-Fi RRM models implemented in the simulator, which are hierarchically dependent of the CRRM entity.

The **Cell Breathing** process is a standard mechanism of UMTS R99. This RRM procedure has a preventive propose, in which the BS load is monitored. In the simulator, should the load level go beyond very high or very low levels, the power of the BS pilot channel will decrease or increase, respectively, inside a limited range. The result is a variation on the coverage area of the BS, which is computed after a pilot power change, the coverage matrix being updated accordingly.

A **Load Balancing** (LB) algorithm is also implemented. In this process, MTs are pushed away from highly loaded BSs by means of HO, until the BS reaches a desired load level. The LB algorithm may cause a ping pong effect on MTs, since they can be moved endless from one BS to another. A counter is implemented to limit and avoid this effect.

Call admission Control (CAC) is triggered when a given MT has a potential starting call/session. When this event is detected, the MT launches a request to the CRRM, this request being processed by the CRRM CAC algorithm. If the selected RAN is Wi-Fi, the MT will check the communication viability (UL and DL propagation). If there is available bandwidth in the BS or AP, the MT will be attached; otherwise it will be rejected (blocking or delay) depending on the service. If the selected RAN is UMTS, a similar process is executed: if the BS load is below a given limit and if there are orthogonal codes available, the attachment process is executed; otherwise, the call is blocked or delayed.

The **Transmission of Packets** in UMTS and Wi-Fi is processed according to a simple principle. The volume and scheduling of each packet are defined by traffic source models, described in Annex A. Given an active user with a packet to be transmitted, the number of time-slots to transmit the packet is calculated according to the instantaneous bandwidth (bitrate) assigned to the user at that moment. Therefore, the number of time slots needed to transmit a packet (the bitrate) is essentially dictated by the network.

For UMTS R99 and R5, the well known COST 231 - Walfisch-Ikegami model [DaCo99] is used to calculate propagation loss. For Wi-Fi, a model based on the free space propagation model is used (detailed information in Annex B). Usually, the free space propagation model is not applicable to many practical situations. However, due to its simplicity, it is common to use it for estimations. In this case, the distance exponent γ=2 is changed to better match practical situations. A breakpoint model can be applied in obstructed conditions, at which a distance exponent γ=2 is used for the first metres and a larger γ is used for distance above a certain breakpoint. For outdoor environments with antenna heights of a few metres and distances of a few hundred metres, a “double breakpoint” model gives a good characterisation of the path loss for urban environments in the presence of obstruction. The double breakpoint model [Pras01] has a first breakpoint at 1 m (reference distance for isotropic loss) and a second breakpoint at 100 m. Since the frequency range covered by this model matches with IEEE 802.11 family, the simulator uses this model for Wi-Fi.

3.1.3 WiMAX

The current version of the simulation tool results of an upgrade of a previous one, to which a WiMAX module was attached. The base simulation tool has a temporal resolution of 10 ms, whilst WiMAX works with 5 ms frames. An adaptation had to be introduced in order to keep coherence by harmonising the resolutions difference.

The solution to this problem demanded **scale equalisation** and some simplifications. Hence, the 10 ms resolution from the original simulator was preserved, while WiMAX 5 ms frames were “converted” to 10 ms. Bearing in mind that two WiMAX frames fit into a 10 ms frame, it was assumed as a simplification that the results of a WiMAX cycle were duplicated. Hence:

- each blocked call event was accounted as two blocked call events;
- each transmitted packet was accounted as two transmitted packets;
- the number of consumed channel elements in each frame per time unit was reduced to its half;

Considering that any network is limited in capacity, and consequently in the number of users handled simultaneously at a given instant, it is important to regulate which users can access the network and under which conditions. The **Traffic Control Algorithm** of the WiMAX module defines customisable margins of available resources that range from 0 (all available resources are reserved) to 1 (all available resources can be assigned). Thus, if the sum of the instantaneous system load plus the additional load needed to transmit a given packet is above the margin, the packet is discarded (lost or delayed). Figure 3-2 illustrates this mechanism. As the bitrates assigned to the users are variable in time, the traffic control algorithm is applied not only for admission (new calls) but also for congestion (on-going calls), thus, allowing the CRRM to control the load of the system at every instant.

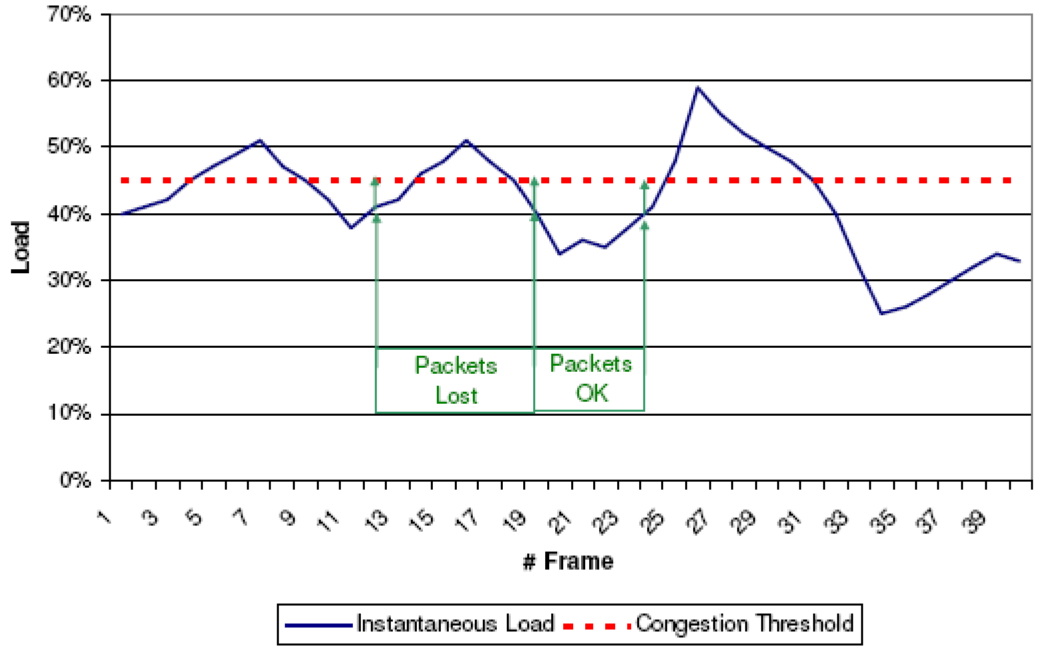


Figure 3-2 - Congestion control function for voice and video calls (extracted from [Antu07]).

When the system load gets close to the limit, the global QoS decreases, as a consequence of an excessive loss/delay of packets. The consequence in non-real-time services (Web browsing and E-mail) is a delay increase, while in real-time services (Voice and Video) the possibility of transmitting information becomes threatened. To cope with this, a maximum number of *n* lost frames is assumed as the limit of quality degradation, the call being dropped at the $(n+1)^{th}$ lost frame. In the present thesis, *n* is set to 30, meaning that calls are dropped if a maximum of 300 ms of “silence” is exceeded.

If, in a given cycle, a given call is active and it has a packet to be transmitted on that cycle, the BS Resources Computation Algorithm computes the necessary BS resources needed to transmit that packet. This depends on further inputs, particularly the packet volume (given by the Traffic Generation Module) and the AMC. Additionally, it interacts with the Overhead Estimation algorithm, which estimates the resources that are needed for MAP overhead purposes, and thus cannot be assigned for user data transmission.

In a real WiMAX system, the **AMC** is computed as a function of the radio channel conditions, namely the interference and the SNR. In a simulator, those mechanisms represent a relatively heavy CPU processing load. Moreover, other authors [Antu07] have simulated AMC evolution based on “realistic” channel conditions. This thesis takes advantage of those results to save significant CPU resources, retrieving AMC from a uniform distribution parameterised with the values in Table 3-2, and not from “realistic” propagation and interference data. The computation of the AMC leads directly to the slot capacity, Table 3-2 [Antu07].

Table 3-2 - AMC distribution and slot capacity per modulation type.

Modulation	Relative Weigth [%]	Slot Capacity [bits]
QPSK 1/2	39	48
QPSK 3/4	9	72
16QAM 1/2	14	96
16QAM 3/4	13	144
64QAM 2/3	5	192
64QAM 3/4	20	216

The useful capacity for DL of a single radio frame depends on the DL/UL TDD split, and on control fields per frame. Due to the TDD nature of the system, a subset of the available symbols is reserved for the UL sub-frame, and it cannot be used for DL bursts. Under the present work, a 2:1 ratio is considered for this split (typical current asymmetry factor for cellular operators), resulting in 32 symbols available for DL and 16 for UL. Other options would be, e.g., to consider a 3:1 split, assigning less weight to UL in favor of DL, more suited for networks dominated by WWW-like traffic, or, on the contrary, a 1:1 split, more suited for symmetric applications, like VoIP or Peer-to-Peer.

Concerning **allocation of slots** by multiple users, WiMAX presents a strong challenge of resolving a complex 2-dimension problem of addressing users in both frequency and time domains. It gets even more complex by considering the fact that, within the same radio frame, the same user can experience different radio channel conditions on the different sub-carriers, meaning that non-adjacent slots may provide the best allocation for a given user, at the cost of increased DL-MAP overhead. The challenge here is to provide that the BS is aware of the different channel conditions, experienced throughout the whole range of sub-carriers for the different users, and that complex calculations can be performed by the BS, in order to find the best commitment between overhead increase and non-adjacent slot allocation for the different users.

For simplicity, the present work only considers a 1-dimension problem [Antu07]. This is achieved by replicating the time domain on the single frequency domain, translating two-dimension allocation arrays of $[N_{sub-channels}, N_{symbols}]$ into single-dimension allocation arrays of $N_{sub-channels}$ x $N_{symbols}$, where $N_{symbols}$ stands for the number of symbols per frame, Figure 3-3.

With this approach, some additional assumptions are considered as well:

- It is assumed that the user experiments the same radio channel conditions for all the sub-carriers of the radio frame, and throughout the whole frame duration. This approach does not reflect the frequency selective nature of the radio channel, thus, it does not allow taking the full advantage of an eventual frequency selective scheduling, resulting in a system performance below the ideal one.
- As this approach makes it easier to fit more users in the same radio frame, the final results can be too optimistic. To cope with this, it is assumed that 1 sub-channel is always wasted, as well as 2 DL symbols.

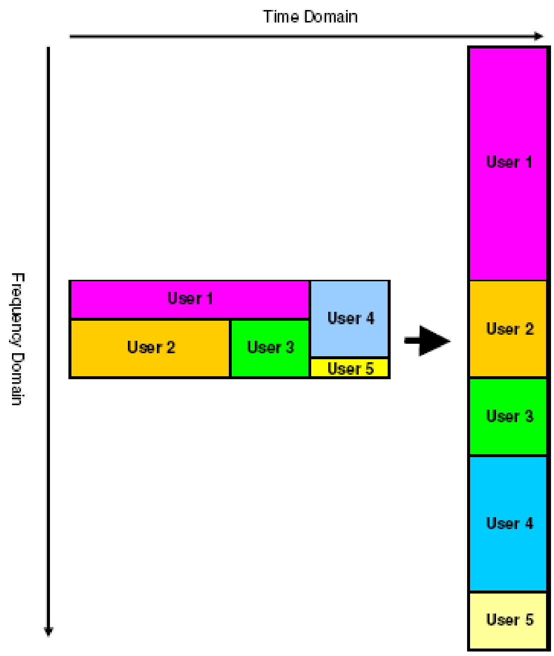


Figure 3-3 - Translation of 2-Dimensional Time and Frequency Domains to Frequency Domain (extracted from [Antu07]).

With this simplification, the present simulator considers:

- From the available 32 symbols for DL, only 30 can be used, meaning that up to 15 users can be multiplexed in time on the same sub-channel.
- From the available 15 (30) $N_{sub-channels}$ for N_{FFT} of 512 (1024), only 14 (29) can be used.

The combination of these into a 1-dimension problem results in the maximum number of available slots for DL transmission per radio frame, $N_{MaxSlots}$, Table 3-3.

Table 3-3 - Available DL slots per radio frame

Channel Bandwidth [MHz]	5	10
$N_{MaxSlots}$	210	435

The consumption of resources by **MAP overhead** is computed by an Overhead Estimation algorithm. The capacity per frame summarised in Table 3-3 still includes overhead, which must then be subtracted. Control fields like the preamble, FCH, DL and UL MAP, ACK-CH, Ranging and CQICH, as illustrated in Figure 2-6, are responsible for this overhead. Some of these fields have fixed size, while others have a variable one, like the case of the DL-MAP message. The simulator adopts a simplified approach for overhead considerations:

- All the sub-channels within the first symbol of the DL sub-frame are always assigned for the preamble, meaning that within the 30 available symbols per 5 ms frame, 1 is always used for the preamble (which includes all sub-carriers), thus, resulting in a reduction of 15 or 30 slots, depending on if N_{FFT} is 512 or 1024.
- The FCH always takes the first four sub-channels of the second and third DL sub-frame symbols,

occupying 8 slots.

- Each DL-MAP message contains fixed and single fields, occupying fixed 104 bits (208 bits with repetition factor of 2) plus additional 60 bits for each addressed ST (120 bits with repetition factor of 2). As DL-MAP is QPSK ½ coded, according to Table 3-2, the number of allocated slots for DL-MAP, N_{DL-MAP}, is given by:

$$N_{DL-MAP} = \left\lceil \frac{208 + N_u \times 120}{48} \right\rceil \tag{3.1}$$

  where N_u is the number of users addressed in the radio frame.

- Although the present thesis is only concerned with DL performance, the UL-MAP message also needs to be considered for overhead calculation in the DL sub-frame. Each UL-MAP takes a single 64 bits block (128 with repetition factor of 2, plus a variable block size per addressed ST, depending on the goal of the message. For the present work, 52 bits (104 with repetition factor of two) is considered for each assigned ST, as this is the value that covers most of the situations. The full details for UL-MAP message construction are available in [IEEE06], and the number of allocated slots for UL-MAP, N_{UL-MAP}, is given by::

$$N_{UL-MAP} = \left\lceil \frac{128 + N_u \times 104}{48} \right\rceil \tag{3.2}$$

- 1 symbol is also "wasted" because it is needed for the Transmit/Receive Transition Gap, but it is taken from UL allocation and thus does not affect available slots for DL

Summarising, the number of slots consumed by overhead, $N_{OHSlots}$, for a channel bandwidth of 10 MHz, is given by:

$$N_{OHSlots} = 38 + \left\lceil \frac{336 + N_u \times 224}{48} \right\rceil \tag{3.3}$$

As an example, if only 1 user is to be addressed in a given frame, and if N_{FFT} = 1024, 12 symbols are occupied by MAP overhead, meaning that 50 slots are consumed by overhead, thus, leaving only 385 slots free for user data.

The **transmission of packets** in WiMAX is processed according to a simple principle. The volume and scheduling of each packet are defined by traffic source models, described in Annex A. Given an active user with a packet to be transmitted, there will be assigned as many slots as necessary to transmit the packet in one single time frame (10 ms). In opposition to the method used for UMTS and Wi-Fi (see 3.1.2), the number of slots needed to transmit a packet (or the bitrate) is essentially dictated by the source model, so it can be said that the bitrate assigned to a user is essentially determined by the service he/she is using.

There are not well known **propagation models** to work in the 2.5 or 3.5 GHz bands, precisely the bands targeted for 802.16e. The 3.5 GHz band, for example, is mainly used for Fixed Wireless Access

(FWA) systems, in which typically LoS exists, thus, standard free-space formulas are used with additional losses due to obstruction of the Fresnel ellipsoids. But there are no specific models for propagation in typical cellular networks. For simplicity reasons, COST 231 – Walfisch-Ikegami model [DaCo9] is used for Mobile WiMAX. Even though the model is validated for the [0.8, 2] GHz band and for distances within [0.02, 5] km, it is acceptable to use a model that is subject to some lack of accuracy above the 2 GHz limit, provided that the main focus of this thesis is CRRM algorithms and not propagation issues. The propagation model allows the estimation of the average value for path loss, and, consequently, the average value for the received signal strength. The existence of fading results in variations of the received signal strength around its average value.

3.2 Simulator

3.2.1 General Description

The current thesis is based on the results produced by a CRRM software simulation tool. It is a system level, time based simulator, with a resolution of 10 ms, which has been developed over the Microsoft Visual Studio 2005 platform. The tool can produce simulation runs using combinations of three types of systems: UMTS (R99 and R5), Wi-Fi and Mobile WiMAX. In its original version [Serr02], the tool supported UMTS and Wi-Fi only. For the current work, an upgrade to the original version was performed, in order to attach a Mobile WiMAX dedicated module.

The simulator implements low-level system functionalities, like power control, link control, basic channel code management, radio bearer service, load control, access control, propagation estimation, interference estimation and generation. It is also possible to generate service traffic mixes through service source models parameterisation.

RRM functionalities are performed by each of the three systems, a CRRM entity being responsible for a common management of available resources and high-level decisions, like call admission, HHO and VHO. In its original version, the simulator was composed by three main functions, identified by the green, yellow and blue blocks of Figure 3-4, and supported UMTS and Wi-Fi RRM and CRRM simulations. The tool has been modified in order to include WiMAX RRM (red blocks) and to update CRRM policies accordingly.

The green blocks represent the following inputs:

- Scenarios Inputs: simulation area, services source models configuration data, services rates and duration, propagation models information, location of BS/APs, building and streets information, etc..
- MT Users Inputs: number of users, service penetration, etc..
- Multi-RRM Algorithms Inputs: parameters related with RRM issues, like service priority list, etc..

- CRRM Algorithms Policies Inputs: RAN selection for call admission and VHO based on service priorities.
- RAN # 1 (BS 1) up to RAN # n (BS n): input parameters for different RAN, like pilot power level, MT maximum power, total power, frequency, etc..

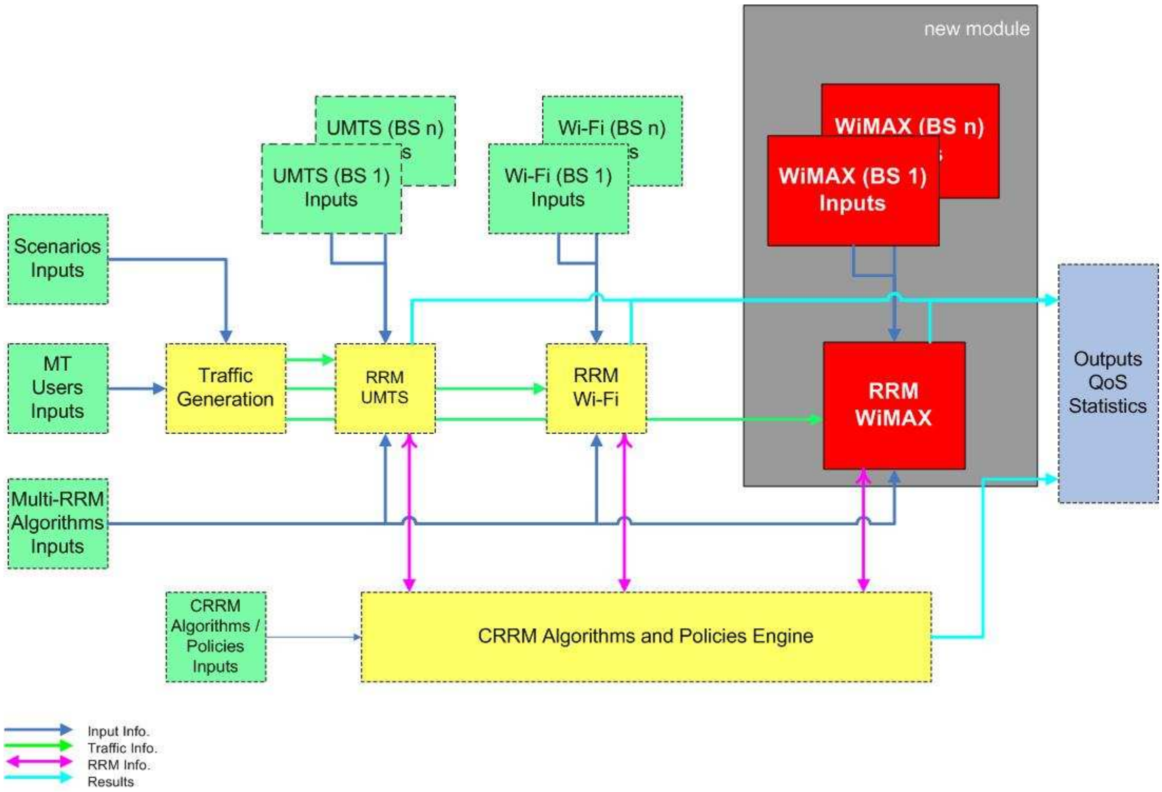


Figure 3-4 – CRRM simulator block diagram.

The yellow set of blocks performs most of the simulation computational effort:

- Traffic Generation: in this block all traffic information vectors of all MTs and services are built, usually within a time frame of one hour.
- RRM RAN #1 up to RAN #n: blocks performing the fundamental functionalities of a specific RAN, by running/managing and monitoring the radio links conditions and services attached.
- CRRM Algorithms and Policies Engine: blocks where major decisions are taken, like all types of handover and initial BS selection.

Finally, the blue block collects system statistics at RRM and CRRM levels, being responsible for the delivery of the simulator's output.

3.2.2 Inputs and Outputs

In its current version, the simulator presents a graphical MS-Windows based interface, Figure 3-5, which offers a significant level of flexibility, giving the tool user the possibility of setting up a relatively large set of parameters with great simplicity. The background is an urban scenario, composed basically by streets and buildings, over which BSs and MTs are located.

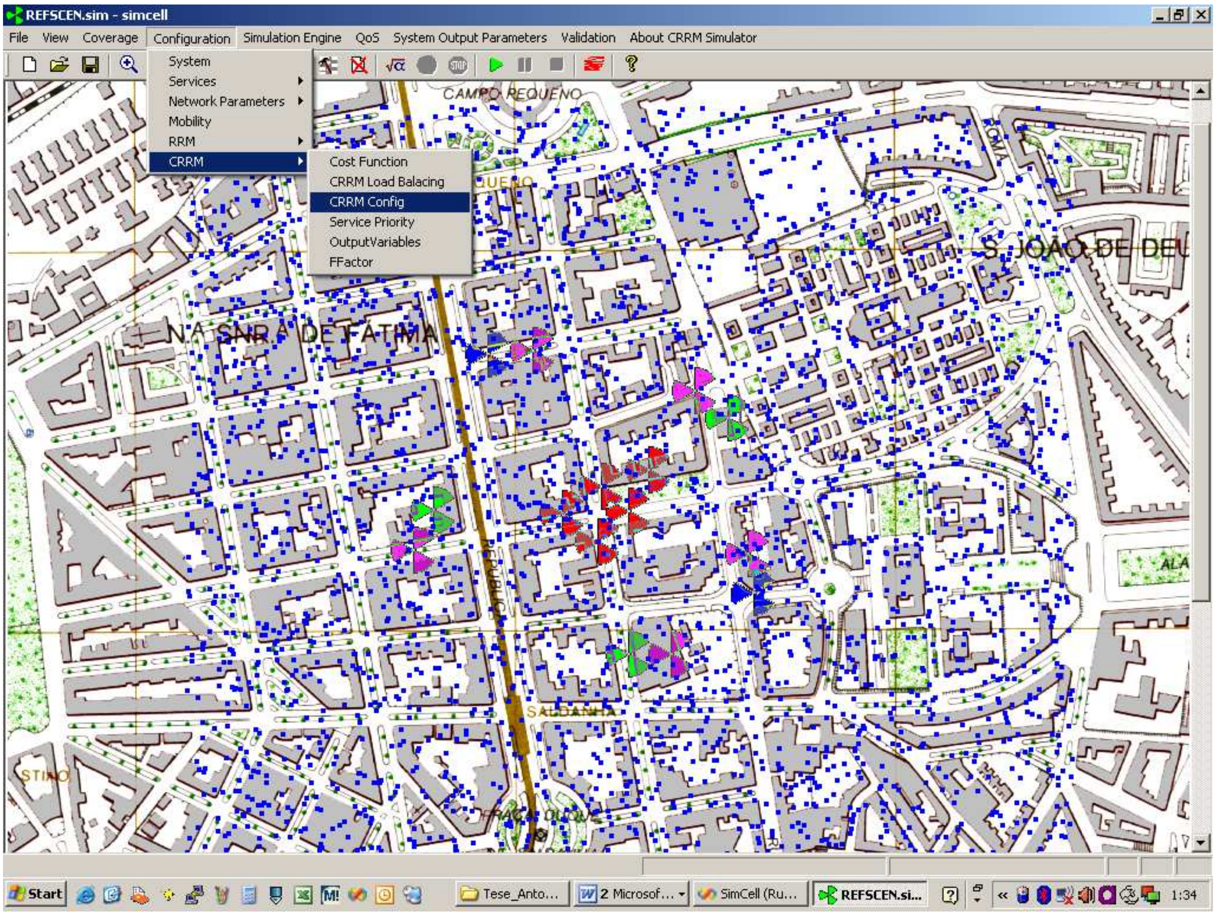


Figure 3-5 - User interface of the simulator.

The input parameters can be conceptually divided in two groups:

- Scenarios and MT users.
- Multi-RRM Algorithms.

The group “Scenarios and MT users” is composed by following input parameters:

- Number of BS and of MTs.
- Number of fast MTs and of slow MTs.
- Minimum speed for fast MTs and for slow MTs.
- Maximum speed for fast MTs and for slow MTs.
- Service mix (Voice, Video-telephony, Streaming, Web browsing, E-mail and FTP) and the corresponding individual source model parameters (described in Annex A).

The Multi-RRM Algorithms group defines the parameters required to define the BSs signal propagation conditions, as well as some RRM initial parameters like the service priority list:

- BS type (UMTS R99, UMTS R5, Wi-Fi or WiMAX), location, height and total power, total available channels, orthogonal factor (CDMA environments), receiver noise figure, cable loss, frequency and pilot Tx power.
- Number of channels used for signalling.
- Receiver noise density.
- Thermal noise density.

- Building loss (penetration additional factor).
- Body loss.
- MT maximum Tx power and height.
- Building height.
- Street width.
- Urban type.
- Street orientation.
- Building separation distance.
- Services priority table (services priorities mapped into RANs).

The CRRM simulator output block produces a set of results that are stored in a file. The simulator being structured in a modular architecture, one can define itself the statistics and output parameters that are interesting and may easily implement its extraction, by setting up virtual scopes. The following output parameters have been defined and implemented, in order to reflect the network's overall performance in a CRRM perspective.

The **CRRM Blocking Probability**, P_b, which is as a measure of blocked Voice and Video calls, is defined in the following way:

$$P_b = \frac{N_{cb}}{N_{ct}} \times 100 \quad [\%] \tag{3.4}$$

where:

- N_{cb} is the number of blocked Voice and Video calls
- N_{ct} is the total number of Voice and Video calls

The **Average Delay**, τ, is defined for Web browsing, Streaming, E-mail and FTP, as a measure of the delay affecting the transmission of packets:

$$\tau = \frac{N_{fd}}{N_{pt}} \times 10\,ms \quad \text{[ms]} \tag{3.5}$$

where:

- N_{fd} is the number of delayed frames.
- N_{pt} is the total number of packets transmitted.

The **Number of Vertical Handover Attempts**, N_{VHOa}, is defined as the number of attempts to perform a handover between BSs from different systems.

The **Number of Horizontal Handover Attempts**, N_{HHOa}, is defined as the number of attempts to perform a handover between BSs from the same system.

The **Probability of VHO failure**, P_{VHOf}, is defined as the percentage of failed vertical handovers:

$$P_{VHOf} = \frac{N_{VHOf}}{N_{VHOa}} \times 100 \quad [\%] \tag{3.6}$$

where N_{VHOf} is the total number of failed VHO.

The **Probability of HHO failure**, P_{HHOf}, is defined as the percentage of failed horizontal handovers:

$$P_{HHOf} = \frac{N_{HHOf}}{N_{HHOa}} \times 100 \quad [\%] \tag{3.7}$$

where N_{HHOf} is the total number of failed HHO.

In the current simulator, a session can end normally while reaching its end or it may be dropped due to one of the following three kinds of situation:

- Handover, if the MT is not covered by a BS with enough available resources.
- In WiMAX, when a maximum number of simulation steps without transmitting a packet is achieved for Conversational services (Voice and Video).
- In UMTS and Wi-Fi networks, when interference and/or load increases beyond acceptable limits.

The **Drop rate**, D_r, is defined in the following way:

$$D_r = \frac{N_{sd}}{N_{st}} \times 100 \quad [\%] \tag{3.8}$$

where:

- N_{sd} is the total number of dropped sessions.
- N_{st} is the total number of sessions (ended normally and dropped).

Three types of Average Bitrate are defined: per service, per system and global.

Average Bitrate per Service, BR_{srv}, is defined as the average value of the bitrate per service per second.

Average Bitrate per RAN, BR_{RAN}, is defined as the average value of the bitrate per RAN per second.

Average Global Bitrate, BR_G, is defined as the mean value of the Average bitrate per RAN:

$$BR_G = \frac{BR_{RAN=UMTS} + BR_{RAN=Wi-Fi} + BR_{RAN=WiMAX}}{3} \tag{3.9}$$

The **Average Number of Users per Service per Second,** NU_{srv}, is defined as the average value of the number of users per service per second.

3.2.3 Assessment

As stated before, the simulation tool used in this work consists of an upgrade of a previous version [Serr02], in which Mobile WiMAX RRM models extracted from another simulator [Antu07] where implemented. In this way, the models in which the resulting simulator is based have been properly validated in the respective sources. The resulting (upgraded) version of the simulator was also duly validated, and in particular a full debugging analysis of inputs and outputs of the new modules and functionalities was performed. The overall performance of the tool and its output were also validated by hours of test simulations, ran in a pre-production stage.

Let us consider an example involving Mobile WiMAX RRM. As seen before, the total number of slots per BS is 435 with a channel bandwidth of 10 MHz (Table 3-3). This leads us to:

$$435 = N_{OHSlots} + N_{UDSM} \tag{3.10}$$

where N_{UDSM} is the maximum number of slots available for user data. By applying (3.3) in (3.10), and assuming that the WiMAX frame capacity must be multiplied by 2 since a 10 ms frame contains two 5 ms WiMAX frames, (3.11) is obtained, which is valid for a single service scenario:

$$435 = 38 + \left\lceil \frac{336 + N_{uMAX} \times 224}{48} \right\rceil + \frac{Ps}{Fc \times 2} \times N_{uMAX} \tag{3.11}$$

where:

- N_{uMAX} is the maximum number of users.
- Ps is the packet size of the service.
- Fc is the instantaneous frame capacity.

By solving (3.11) in order to N_{uMAX}, one obtains upper and lower bounds for the maximum number of users considering a single service scenario:

$$\frac{390}{5 + \dfrac{Ps_{MAX}}{Fc_{MIN} \times 2}} \leq N_{u\,MAX} \leq \frac{390}{5 + \dfrac{Ps_{MIN}}{Fc_{MAX} \times 2}} \tag{3.12}$$

where Ps_{MAX} and Ps_{MIN} correspond to maximum and minimum volume that a packet can reach, and Fc_{MAX} and Fc_{MIN} correspond to maximum and minimum size of the WiMAX frame. The maximum number of users N_{uMAX} is defined as the number of users accounted in the last cycle of simulation before obtaining the first blocked (or delayed) call, which is the reflex of resources exhaustion. Regarding Ps, it is given by a source model according to the service in use (see Annex A). To verify (3.12), one uses a simulation scenario with one WiMAX BS. The results of the simulation are presented in Table 3-4, and as it can be seen, the simulated values fall inside the range theoretically defined by the source models and the frame capacity.

Table 3-4 – Maximum number of WiMAX slots assessment for a single BS

Service	Frame Capacity (bits)		Packet Size (bits)		N_{uMAX} theoretical		N_{uMAX} from simulation
	MIN	MAX	MIN	MAX	MIN	MAX	
Voice	48	216	56	56	70	76	71
Video			2432	2432	13	37	17
WWW			1920	∞	0	41	6
Streaming			1280	∞	0	49	9
E-Mail			640	∞	0	60	18
FTP			640	∞	0	60	15

The current simulator is a time-based, dynamic system, and as it happens with simulators of this type, an instability period is expected at the beginning of the simulation period. After this period, a convergence to stability is observed. It is necessary to assess a starting point from which the simulation parameters are expected to be stable, after an initial oscillation period. The results obtained before the starting point should be naturally discarded.

To assess the valid simulation period, a simulation of 1h10 was performed. The scenario for this simulation included the three systems (UMTS, Wi-Fi and WiMAX) and considered a user density of 10 000 users/km^2. Each user is active for a given service, which is characterised by a time variable bitrate, which value depends on the system and on the channel conditions. In Figure 3-6, the evolution of the Average Global Bitrate along 4200 s (1h10) of simulation is depicted.

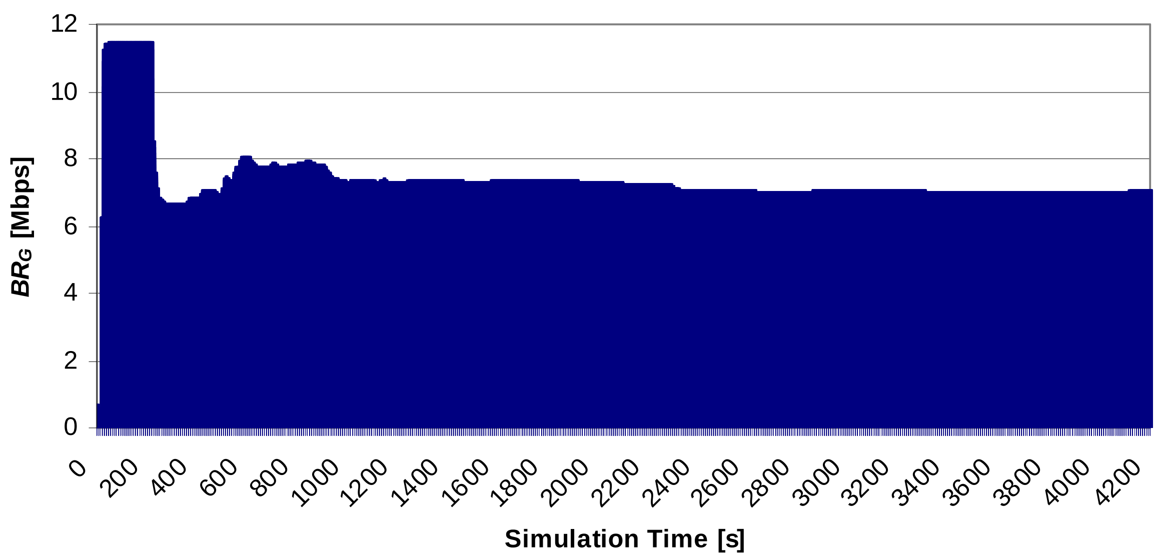


Figure 3-6 - Average Global Bitrate for a 1h10m simulation period.

A close observation to Figure 3-6 shows that stability is achieved around the 10th minute (600 s) of simulation. This fact is also observed in the previous version of this simulator [Serr02], so it can be concluded that the upgraded version inherits such threshold. Consequently, the first 10 minutes of all simulations performed in this work are discarded.

The proposed value of the current thesis resides basically in two main aspects. First, a simulation tool for a heterogeneous mobile network gathering three (four) RAN technologies was developed. And second, a heavy series of simulations was processed in order to characterise the HN performance. Like in any work involving heavy simulation workload, decisions had to be taken in order to conciliate statistically relevant results with a feasible (limited) time frame. The maximisation of the number of simulation results on the one hand, and the minimisation of time/CPU resources spent in simulation work on the other configures a non-trivial optimisation problem. In particular, an elementary question needed to be answered: given a specific scenario, how many simulations have to be run to characterise it with a minimum statistical error.

The answer to this question is given by the weighting of some facts, starting of course by simulation times. To run a 10 + 60 minutes process, a machine with a 2 GHz processor and 3 GB of RAM takes approximately 5 h. This time period raises up to 9 h should a 1 GHz processor / 1 GB RAM machine be used instead, which means a maximum production rate of less than 3 simulations per day time in that case.

In order to better describe the strategy to characterise the HN integrating three (four) RATs, let us define the HN Performance Function (HNP) as a column matrix, which entries correspond to the output parameters defined in Section 3.2.2:

$$HNP^T = \begin{bmatrix} P_b & \tau & N_{VHOa} & N_{HHOa} & P_{VHOf} & P_{HHOf} & D_r & BR_{srv} & BR_{RAN} & NU_{srv} \end{bmatrix} \qquad (3.13)$$

Consequently, it can be assumed that HNP is a function of the following inputs of the simulator:

$$HNP = f(N_u, GDU, VCAD, Ps, SP, PT, RANoff, WBW) \qquad (3.14)$$

where:

- N_u is the number of users.
- GDU is the geographic distribution of users.
- $VCAD$ is the average duration of the voice calls.
- Ps is the volume of the data packets.
- SP is the service penetration.
- PT is the priority table.
- $RANoff$ stands for the RAN that is switched-off in the HN
- WBW is the bandwidth of WiMAX channel

The planning of this thesis addressed the assessment of a reference scenario (REF), which can be described as a set of fixed values that are assigned to the input parameters of the simulator. The characterisation of the HNP function is given by the characterisation of its individual functions. In practical terms, this is achieved by defining a given number of scenarios, where all the parameters assume the fixed values defined for REF, except for one, that will float inside a given range of values.

So far, it has been seen that the choice of the optimum number of simulations must take simulation time/CPU resources and number of parameters of the HNP function into account. At this point, it is important to understand the convergence potential of the HNP function. This was done by running 15 simulations and defining a measure of convergence, *Δ*:

$$\Delta[\%] = \frac{\left| HNP^{cum\_s} - HNP^{cum\_S} \right|}{HNP^{cum\_S}} \times 100 \qquad (3.15)$$

where:

- s is the simulation run index.
- S is the total number of simulation runs conducted during the convergence study (15).
- HNP^{cum_s} is the partial HNP cumulative mean at simulation run s.
- HNP^{cum_S} is the total HNP cumulative mean (from all simulations).

In Figure 3-7, the convergence of the HNP function for REF is presented. It can be observed that about 5 to 7 simulations are needed to obtain a measure of convergence Δ below 15 %. 12 simulation will be needed instead to get a *Δ* under 10 %.

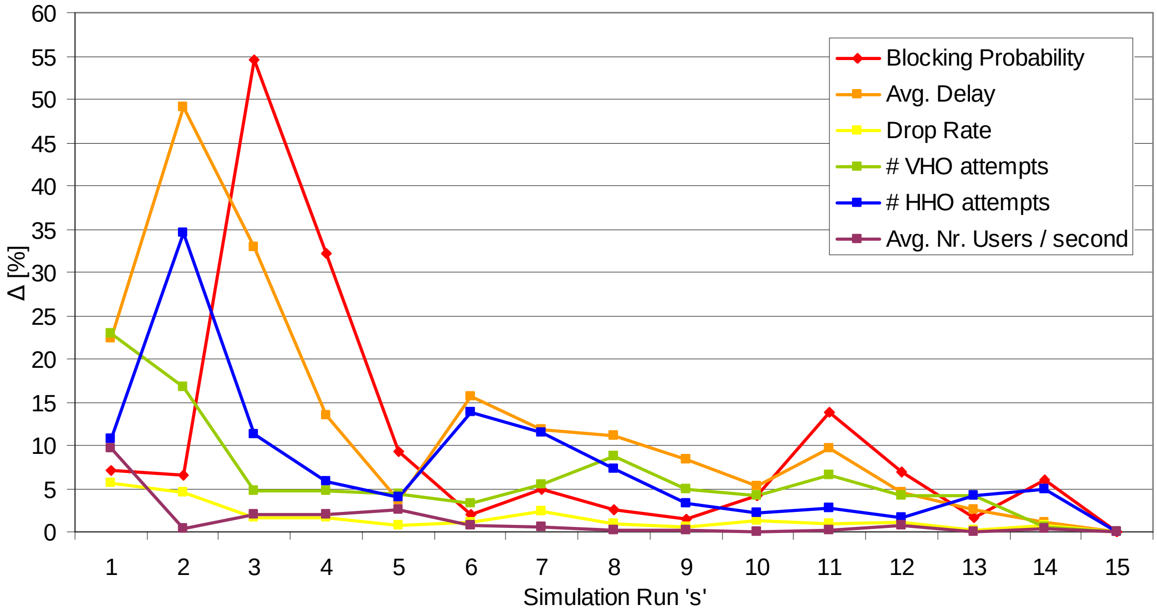


Figure 3-7 - Convergence of the HNP function.

Considering that the HNP function has 8 variables, and that the variation of its parameters can be characterised by 3 to 4 scenarios per variable, a total number of approximately 24 to 32 scenarios have to be run in the simulator. As it has been seen, a reasonable margin of 10 % is achieved with 12 simulations, which raises the total number of simulation runs up to 384. Assuming an average simulation time of 7 h per scenario, 112 days of simulations would be needed to accomplish this task, using only one PC. Given this huge logistic effort, it has been decided to adopt a number of 5 simulation runs per scenario, leading to 1120 h of simulation (47 days approximately). Despite of the uncertainty level associated to 5 simulations runs, it has to be taken into account that each simulation run represents one busy hour network traffic. Simulation results extracted and processed under these circumstances must be minimally reliable and worthy of trust.

Chapter 4

Analysis of Results

In this chapter, the results of the simulation runs for different scenarios are presented, analysed and discussed in detail. In the first section, the reference scenario is characterised. In following sections, the Heterogeneous Network Performance is described as a function of user-dependent parameters and as a function of system-level parameters.

4.1 Reference Scenario

By setting a Reference Scenario (REF), one seeks to reproduce a HN integrating UMTS (R99 and R5), Wi-Fi and Mobile WiMAX. The network of REF is composed of 2 WiMAX BSs, 2 UMTS R5 BSs and 1 UMTS R99 BS, geographically distributed around a hotspot of 5 Wi-Fi APs. REF is depicted in Figure 4-1, where WiMAX sites are represented in lilac, R99 site in blue, R5 sites in green and Wi-Fi hotspots in red; the blue dots represent the users. The coverage spots represent a minimum signal level of -102 dBm. The background is an urban environment, physically located between Campo Pequeno and Saldanha, in Lisbon, its main characteristics being summarised in Table 4-1. The main characteristics of the BSs of the different RANs are listed in Table 4-2. Users are characterised by two parameters: position and service. In REF, the position is generated according to a bi-dimensional uniform distribution in a 1000 x 1000 m^2 area. Users are assumed to be static, allowing significant CPU time savings, which are crucial to guarantee affordable simulation periods for a Master Thesis time scope. A mono-service policy is adopted for simplification reasons, in which only one type of service is allowed per user throughout the whole simulation period. In order to introduce service heterogeneity, six services are defined, Table 4-3. Services are associated to users in the beginning of the simulation according to the service penetration in Table 4-3. In the simulator, services are implemented according to the source models defined in Annex A.

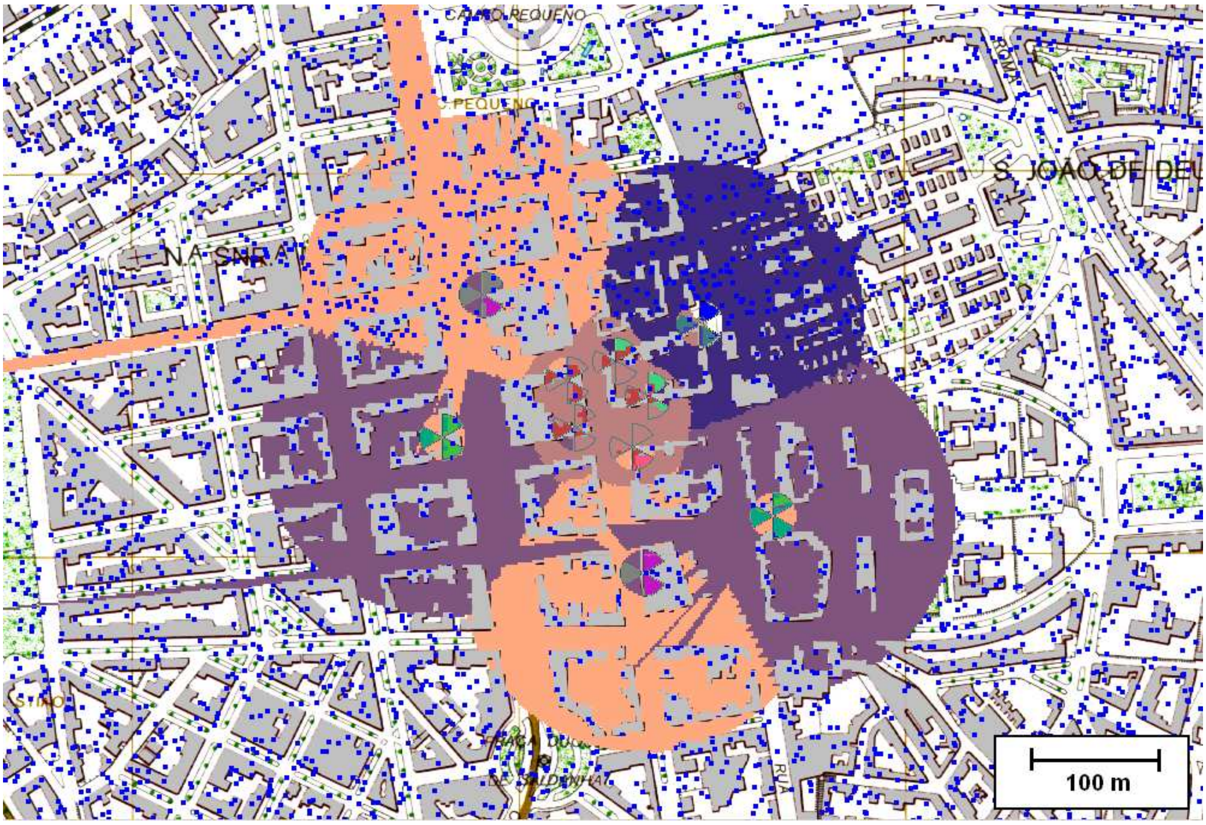


Figure 4-1 - BS distribution in the REF and respective coverage areas.

Table 4-1 – Terrain parameters.

Parameter	Value
Users density [km^{-2}]	10 000
Street width [m]	20
Building height [m]	25
Building separation [m]	40
MS height [m]	1.5

Table 4-2 – Base station main parameters.

Parameter \ RAN	UMTS R99	UMTS R5	Wi-Fi	WiMAX
Antenna type	omnidirectional			
BS height [m]	22	22	6	22
Maximum Tx Power [W]	20	20	0.1	20
Pilot Tx Power [mW]	100	150	90	150
Frequency Band [GHz]	2.1	2.1	5.0	3.5
Channel Bandwidth [MHz]	5	5	20	5

The assessment of the parameters of REF has a great impact in the simulation workload, because all the test scenarios are based on the reference one. As already seen, the real time spent with a 70 minutes simulation run has a direct dependency on the machine capabilities, like CPU and RAM, but it also depends from the values taken by the input parameters. For example, a simulation with 10 000 users takes less time than another with 20 000, naturally. Hence, a “good” REF can be assessed by following some elementary principles in terms of network load and simulation time. In particular, the network load has to be moderate enough so variations can easily be observed, while changing scenario’s input parameters. At the same time, the real simulation time has to be kept as reduced as possible, since a great part of the simulation scenarios are “heavier” than REF in terms of simulation effort, and may originate unaffordable resources consumption. Given these guidelines, REF assessment was generically based on the parameters defined in the AROMA project scenarios, Table 4-3, in which fine tunings were introduced in order to adapt it to the objectives of the present thesis.

The characterisation of the HNP function for the reference scenario is based on the results of 15 simulations. The error margin of the results is consequently lower, which is important, since they serve as the anchor for all the scenarios under analysis in this work. The results are presented in Table 4-4.

One of the first conclusions taken from Table 4-4 is the high level of dispersion that affects some of the parameters. This dispersion tendency is observed in a relevant part of the results throughout this thesis, putting in evidence the highly random nature of the simulator. However since the simulator is based on validated models whether traffic sources, users generation or propagation issues are concerned, the observed dispersion can be understood just as a reflection of the reality itself.

Table 4-3 – Service parameters of the reference scenario (extracted from [AROM07]).

Parameter \ Service	Voice call	Video call	Web browsing	Streaming	E-mail	FTP
Penetration [%]	50	4	12	10	14	10
Mean duration [s]	90	60	N/A	N/A	N/A	N/A
Session/Service rate (Poisson)	1	1	3	3	3	3
Packet calls in a session (Geometric)	N/A		5	4	N/A	N/A
Number of packets per call (Geometric)			25	200	N/A	N/A
Reading/Viewing Time [s] (Geometric)			10	40	N/A	N/A
Data Volume per packet (Pareto)			α = 1.1; k = 80	α = 1.1; k = 80	α = 1.1; k = 80	α = 1.1; k = 80
Total data volume per session (Log-Normal)			N/A	N/A	u_1=8.2; s_1^2=3.4	u_1=8.5; s_1^2=3.7
Inter-Arrival Time (Log Normal)			N/A	N/A	u_2 = -4.4; s_2^2 = 4.5	u_2 = -3.6; s_2^2 = 5.0

Considering P_b (Blocking Probability) and τ (Average Delay) values, it is evident that, in fact, REF is not exactly reflecting a typical busy hour load profile, should one consider typical P_b and τ close to 2 % or 200 ms for the busy period. But as already stated, the objective of fixing a reference scenario is not to reproduce the busy hour typical load profile. As it will be seen, the load profile increases significantly when other scenarios are considered.

Regarding Average Bitrate, an asymmetry between Wi-Fi and the remaining systems, mainly UMTS, is immediately detected. This is not surprising considering the high level of asymmetry in the performance / coverage trade-off of Wi-Fi, where the first is highly privileged compared to the second. As one knows, this does not happen in cellular systems, like UMTS, where the trade-off tends to be more balanced. WiMAX can work as a mid-term, with moderate bitrates and cellular coverage.

As far as handover is concerned, and since it is assumed that the users are static, it can only be due to load balancing or power control. It is observed that the number of VHOs is higher than the one of HHOs. One possible explanation is given by Figure 4-1, in which one can see that there is more area of coverage superposition of different systems than the ones covered by more than one BS of the same type.

Regarding NU_{srv}, it is not surprising that the biggest part corresponds to Voice users, which reflects not only the penetration values in Table 4-3 but also the networks reality nowadays in terms of services usage.

Table 4-4 – HNP function for the reference scenario.

		Mean val.	St.dev.
P_b [%]		1.00	0.87
τ [ms]		23.53	11.10
N_{VHOa}		237.53	76.83
N_{HHOa}		74.00	39.71
P_{VHOf} [%]		5.40	4.45
P_{HHOf} [%]		4.33	5.74
D_r [%]		24.60	2.29
BR_{srv} [Mbps]	Voice call	0.03	0.00
	Video call	0.51	0.01
	Web browsing	15.13	12.05
	Streaming	4.75	4.70
	E-mail	3.61	6.26
	FTP	2.87	3.75
BR_{RAN} [Mbps]	UMTS	0.15	0.05
	Wi-Fi	20.51	3.80
	WiMAX	7.99	3.52
NU_{srv}	Voice call	47.44	2.52
	Video call	0.98	0.38
	Web browsing	4.02	1.06
	Streaming	7.22	2.21
	E-mail	4.28	0.82
	FTP	4.25	1.56

4.2 System performance as a function of user-dependent parameters

The behaviour of users in a network is somehow unpredictable. Its density in a given area can be higher or lower, corresponding to a bigger or smaller number of users attached to the BSs on that area. In terms of geographic distribution, users can be uniformly scattered over the field or more concentrated around a Wi-Fi hotspot, for example. In a service perspective, there can be variations both on the average duration of Voice and Video calls and on the volume of “data” services. The service penetration is also an important parameter, since different services require different levels of resources. In order to study the response of a HN to the level of uncertainty affecting user-dependent parameters, several simulation scenarios were created and processed; in each one, only one parameter is varied, except in special cases that are addressed further on. This way, the cause-effect relationship between input parameter variation and output parameters is more easily observed.

4.2.1 User density

One of the most relevant parameters in the planning of a telecommunications network is the maximum number of users it can support, while maintaining a minimum QoS. In this case, 3 simulations scenarios were processed considering the following densities: 8 000, 12 000 and 14 000 users/km^2. Since the geographic area of the scenario is 1 km^2, the number of users corresponds exactly to 8 000, 12 000 and 14 000. Users being uniformly distributed over the area, a given number of them naturally falls out of coverage.

Figure 4-2 shows the variation of the average value of P_b with the number of users, where the standard deviation is represented by vertical black bars The results show that it grows as the number of users in the network increases, which is an expected conclusion. The figure also shows the adjustment by error minimisation (using MS Office Excel *Add Trendline* option) of four known functions to the simulation results: Linear, Polynomial, Power and Exponential, respectively (4.1), (4.2), (4.3) and (4.4):

$$\overline{P_b^{lin}} = 4.6131 \cdot \left(\frac{N_u - 6000}{2000} \right) - 5.777 \tag{4.1}$$

$$\overline{P_b^{pol}} = 1.1168 \cdot \left(\frac{N_u - 6000}{2000} \right)^2 - 0.971 \cdot \left(\frac{N_u - 6000}{2000} \right) - 0.1935 \tag{4.2}$$

$$\overline{P_b^{pow}} = 0.3044 \cdot \left(\frac{N_u - 6000}{2000} \right)^{2.6859} \tag{4.3}$$

$$\overline{P_b^{\exp}} = 0.1084 \cdot e^{1.2666 * \frac{N_u - 6000}{2000}} \tag{4.4}$$

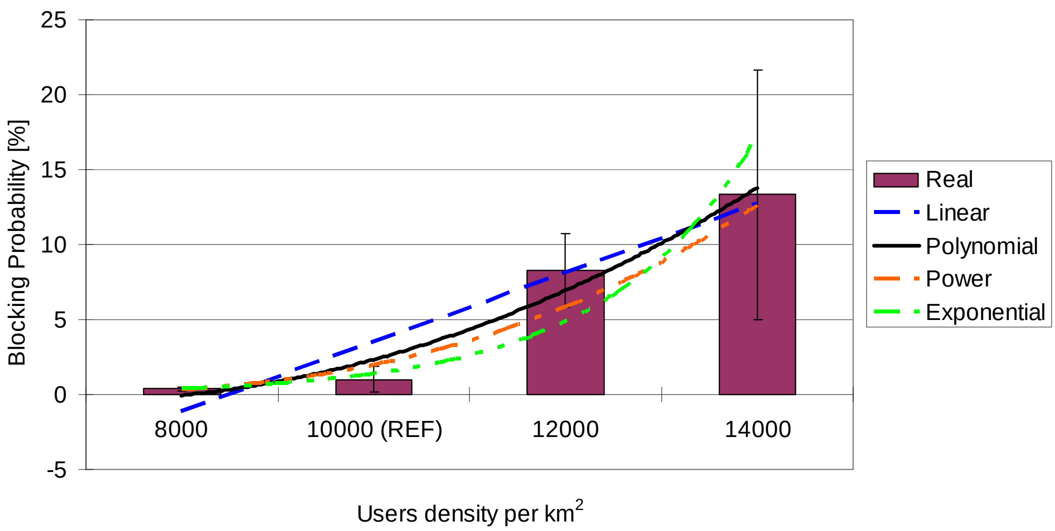


Figure 4-2 – Blocking Probability variation with users density (real and adjusted).

The values resulting from (4.1) to (4.4.) are represented in Table 4-5, together with the respective Correlation Coefficients, R^2. As the table shows, the best correlation with the simulation values (P_b^{sim}) is obtained through the Polynomial function, the worse being the Linear one.

Table 4-5 – Blocking Probability numeric values and Correlation Coefficients.

N_u	P_b^{sim} [%]	$\overline{P_b^{lin}}$ [%]	$\overline{P_b^{pol}}$ [%]	$\overline{P_b^{pow}}$ [%]	$\overline{P_b^{\exp}}$ [%]
8 000	0.40	-1.16(*)	-0.05(*)	0.30	0.38
10 000	1.00	3.45	2.33	1.96	1.37
12 000	8.28	8.06	6.94	5.82	4.84
14 000	13.35	12.67	13.79	12.60	17.19
		R^2=92.26%	R^2=96.59%	R^2=92.35%	R^2=94.71%

(*) negative P_b values are senseless and merely indicative

Figure 4-3 shows the variation of the average value of τ with the number of users, together with the adjustment by the four known functions. The values resulting from the functions are represented in Table 4-6, together with the respective Correlation Coefficients. The best correlation with the simulation values (τ^{sim}) is once again obtained through the Polynomial function, the worse being the Power one.

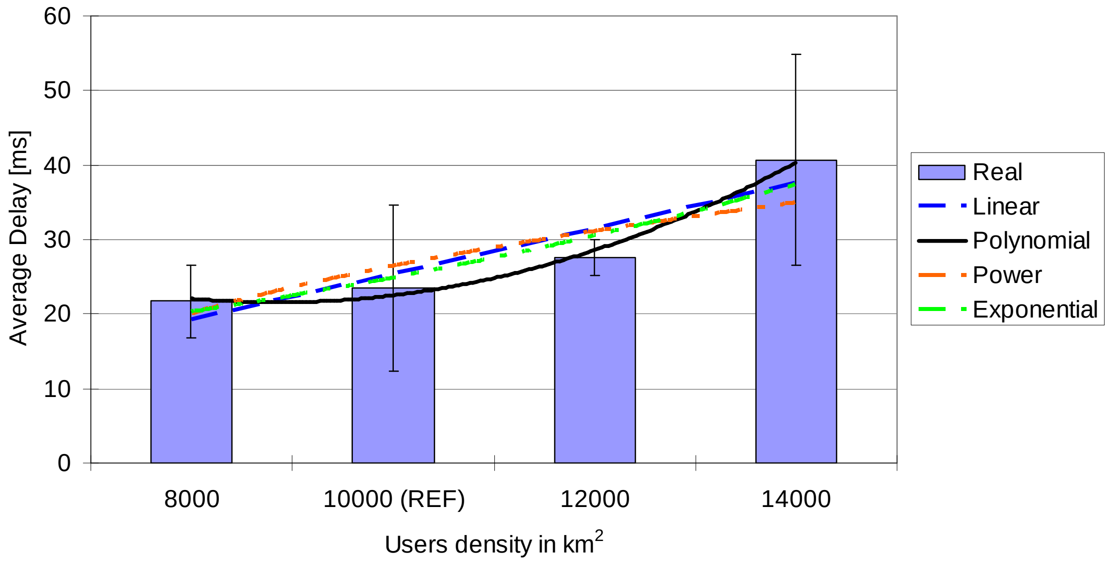


Figure 4-3 – Average Delay variation with users density (real and adjusted).

Table 4-6 –Average Delay numeric values and Correlation Coefficients.

N_u	τ^{sim} [ms]	$\overline{\tau^{lin}}$ [ms]	$\overline{\tau^{pol}}$ [ms]	$\overline{\tau^{pow}}$ [ms]	$\overline{\tau^{\exp}}$ [ms]
8 000	21.72	19.23	22.06	19.99	20.26
10 000	23.53	25.33	22.50	26.42	24.85
12 000	27.57	31.42	28.60	31.09	30.48
14 000	40.68	37.52	40.34	34.91	37.38
		R^2=84.42%	R^2=98.94%	R^2=75.07%	R^2=89.28%

The standard deviation of the simulation results reveals a significant dispersion, and consequently introduces a given level of uncertainty that should not be neglected. Despite of that, a growing tendency can be observed in both cases, which reinforces the confidence margin.

If a Blocking Probability of 2 % is considered as the limit for an acceptable level of QoS, the network would be over-dimensioned for 10 000 users/km^2, and under-dimensioned for 12 000. At the same time, the network has enough resources to deliver data services to 14 000 users/km^2 with an Average Delay under 50 ms, which represents an acceptable performance in terms of QoS.

The users density ranging from 8 000 to 14 000 corresponds to increasing the concentration of users in the area, and consequently more handovers, both vertical and horizontal. This effect can be observed in Figure 4-4. As the figure suggests, a linear curve adjusts to the simulation data with a Correlation Coefficient of 99.8 %. The linear adjustment is not so perfect in HHO, in which a coefficient of 70.3 % is obtained. In this case, an adjustment with a polynomial function leads to the best correlation coefficient: 88.5 %.

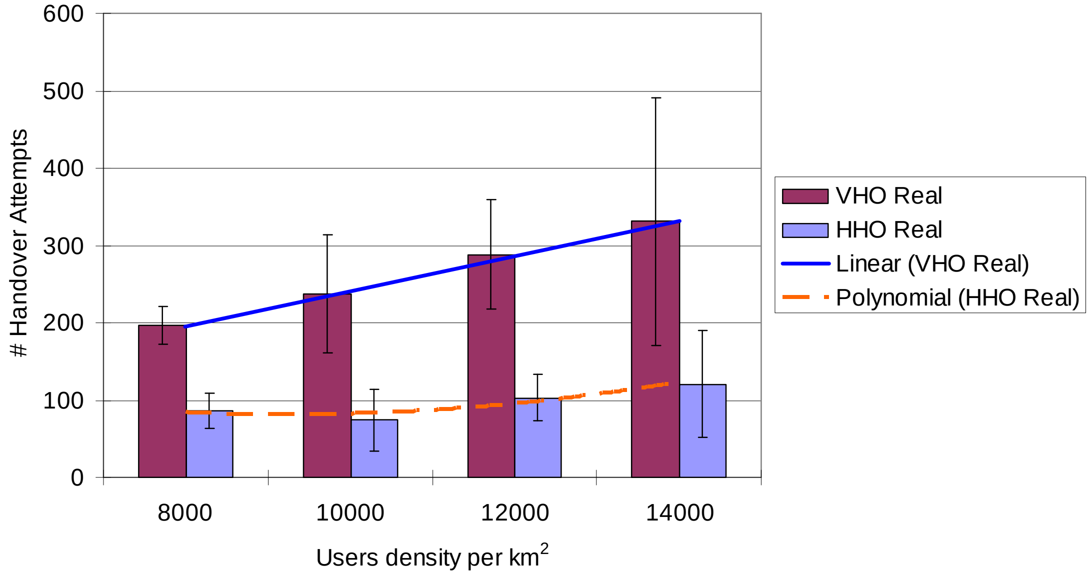


Figure 4-4 – Number of handover variation with users density (real and adjusted).

In what concerns the bitrate per service, Web browsing and Streaming are the most demanding services, which is directly explained by the source models in use. In terms of bitrate per system, Wi-Fi distinguishes itself from the cellular systems, with bitrates between 17 and 23 Mbps, more than 20 times higher than UMTS. WiMAX works as the mid-term, with bitrates not as high as Wi-Fi and not as modest as UMTS.

The increase of network users is well observed in Figure 4-5, where the average number of connected users per service per second is represented. One can observe that increasing the number of users in 2 000 leads to an increase of 5 to 10 active Voice users per second, and to about 50 more VHO attempts. A tiny increase of both the Web browsing and Streaming users is also visible, which can be responsible for a growing tendency of the Wi-Fi bitrate, given the fact that Wi-Fi is the preferred RAN for Web browsing and the second preferred RAN for Streaming (Table 3-1). UMTS and WiMAX bitrates do not exhibit a clear tendency to increase or decrease with the number of users. The whole data of the simulations runs for this case can be found in Annex D.

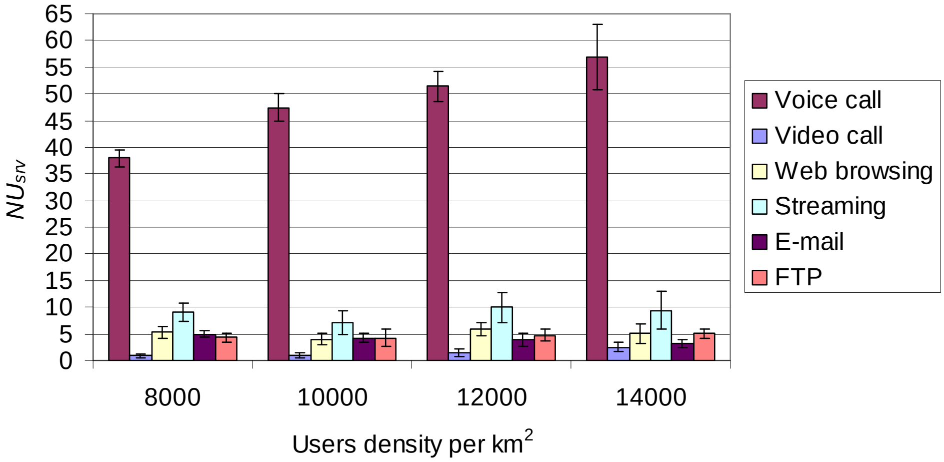


Figure 4-5 – Average connected users per second variation with users density.

4.2.2 Geographical distribution of users

In the reference scenario, users are assumed to be uniformly distributed in space. In the present case study, two additional scenarios were tested, in which users are spatially scattered according to a bi-dimensional Normal distribution. In the first scenario, users are more concentrated around a central point, with standard deviation parameters σ_x and σ_y equal to 150 m; in the second, users are less concentrated around the centre, the standard deviation being raised to 300 m. This type of distribution reflects a real distribution of users in a shopping centre area, where this density is greater inside the facility and gradually reduced in its vicinity, e.g., Figure 4-6.

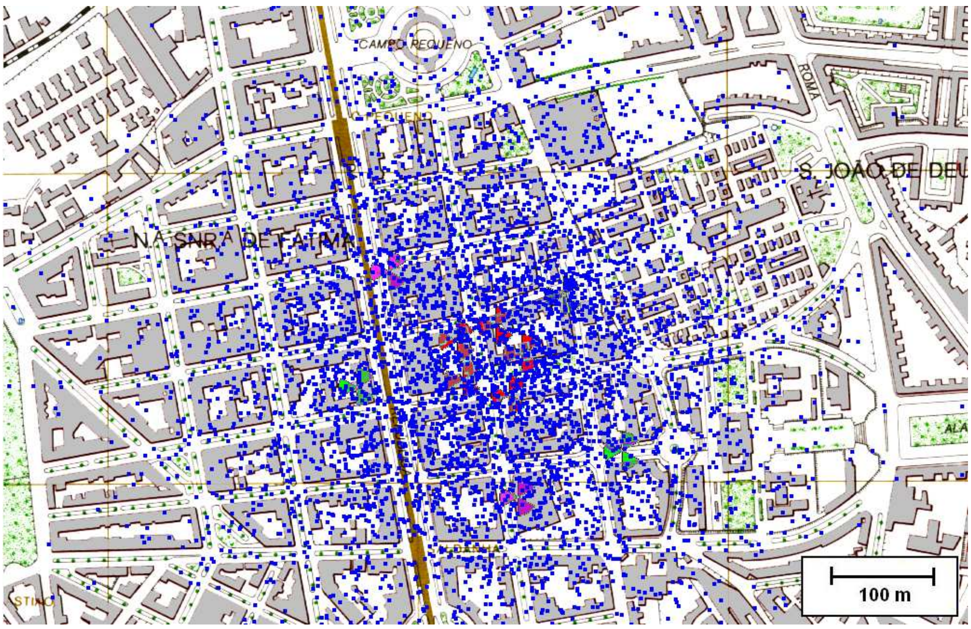


Figure 4-6 - Geographical distribution of users according to a bi-dimensional normal function with a standard deviation of 150 m.

In this type of scenario, interference increases, with a direct impact on the performance of UMTS. Despite of interference having also impact at the modulation level in real WiMAX, the user density increase has no influence in the simulator, since the modulation is given artificially by a statistical distribution (Table 3-2). Another consequence of shifting from a Uniform to a Normal distribution of users is the increase of covered users, which means an increase of available resources consumption. This is quite visible in P_b and τ, Figure 4-7 and Figure 4-8, respectively. As one can see, a reduction of the standard deviation from 300 m to its half is responsible for a 7 times-higher P_b, and for a 4 times-higher τ. These results show that the current network topology and/or dimensioning is clearly not well suited for this type of scenario, at least if the user distribution follows a normal distribution with a 150 m standard deviation. Nevertheless, if the standard deviation is doubled, P_b and τ reduce drastically back to quite acceptable values, practically equal to REF.

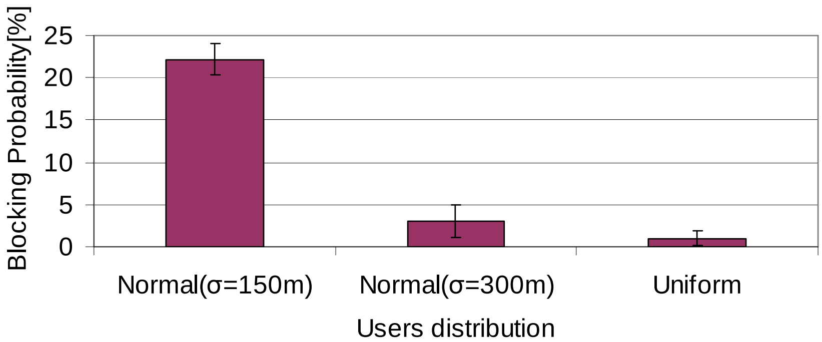


Figure 4-7 – Blocking probability variation with the distribution of users.

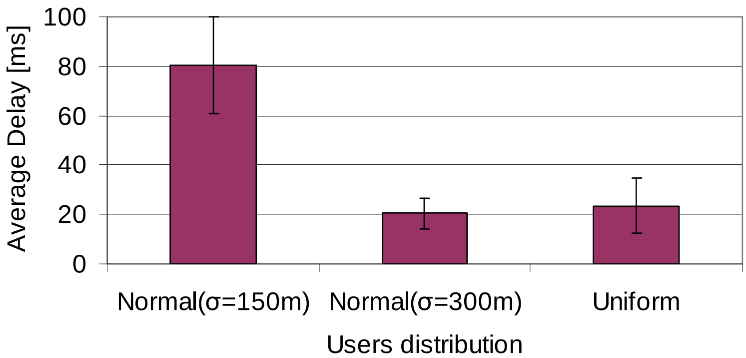


Figure 4-8 – Average Delay variation with the distribution of users.

Figure 4-9 shows some interesting results, which are worthy of a deeper analysis. In particular, one can see that P_{VHOf} (VHO failure) and D_r (Drop rate) present opposite tendencies, which is not a common behaviour in the tested scenarios. In fact, P_{VHOf} increases with the concentration of users near the centre. According to (3.6), this could only happen if either N_{VHOf} increased or N_{VHOa} decreased. Figure 4-10 shows that the latter is true, i.e., N_{VHOa} is decreasing together with σ_x and

σ_y, probably due to the fact that there are less users close to cell boarders, corresponding to less potential handover generators. In this case, it is fair assuming that N_{VHOf} also decreases, thus tending to neutralise the effect in the failure rate. But the fact that the network is overloaded can be responsible for an increase of handover failure, working as the "tie breaker", therefore, being responsible for the N_{VHOf} increase.

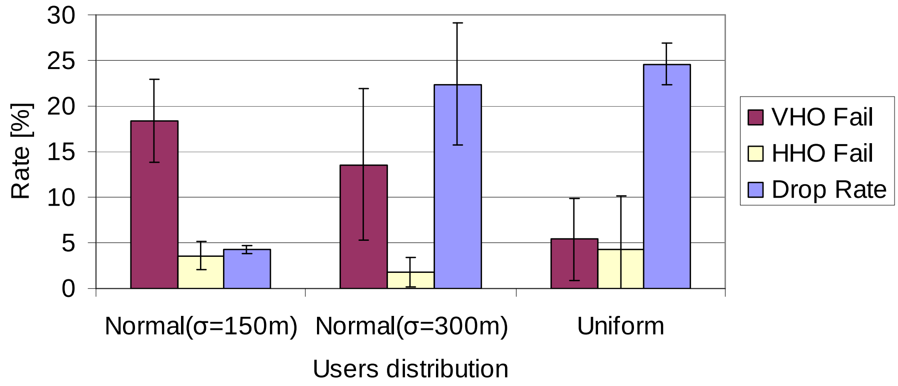


Figure 4-9 - Handover failure rate and drop rate variation with distribution of users.

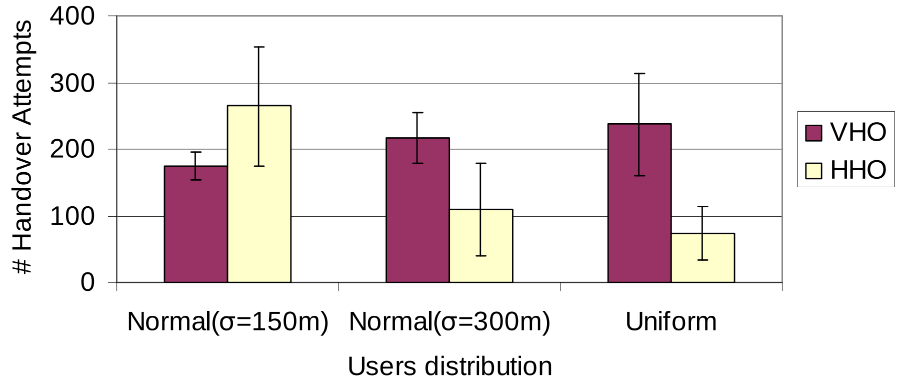


Figure 4-10 - Handover variation with distribution of users.

The same type of arguments must be considered to understand the decrease of D_r. Network overload should contribute not only for handover failure but also for drop calls increase, and the reduction of users in cell boarders should contribute to the opposite effect, i.e., the decrease of drop calls. Since there are no significant differences in the average number of calls per second, Figure E-7, the conclusion is that the reduction of "cell boarder" users is the dominant effect.

A last comment regarding bitrates per service, Figure 4-11. A higher concentration of users around the centre seems to produce an increase of Web browsing average bitrate. Despite of the high dispersion of the statistic, this result makes sense, bearing in mind that, according to Table 3-1, Web browsing users are preferentially assigned to Wi-Fi APs, which are only able to give coverage to a limited spot in the centre of the scenario.

The whole processed data of the simulations runs for this case can be found in Annex E.

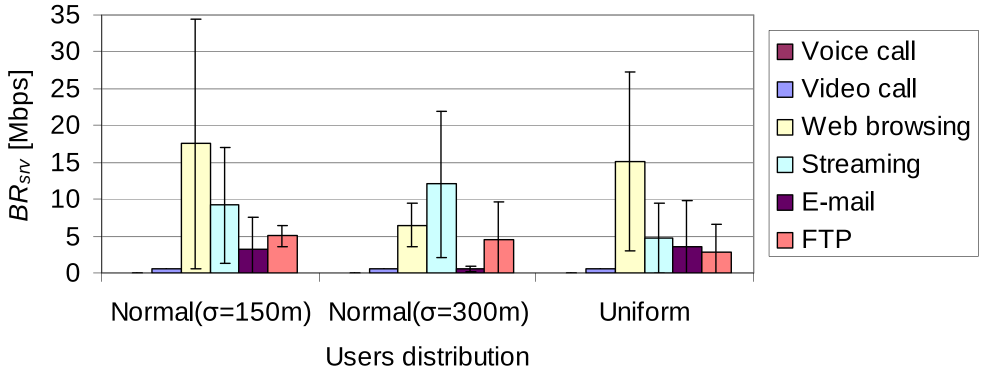


Figure 4-11 - Average bitrate per service variation with distribution of users.

4.2.3 Average duration of voice calls

As previously described, the duration of Voice calls in the simulator is computed according to an exponential distribution with a given mean value. In REF, the mean value is 90 s (Table 4-3); in the present case, two scenarios have been tested in order to evaluate HNP for an average duration of voice calls of 105 and 120 s. In practice, this change means that Voice calls are spending network resources during longer periods, which is equivalent to say that the average number of Voice users per second increases, Figure 4-12.

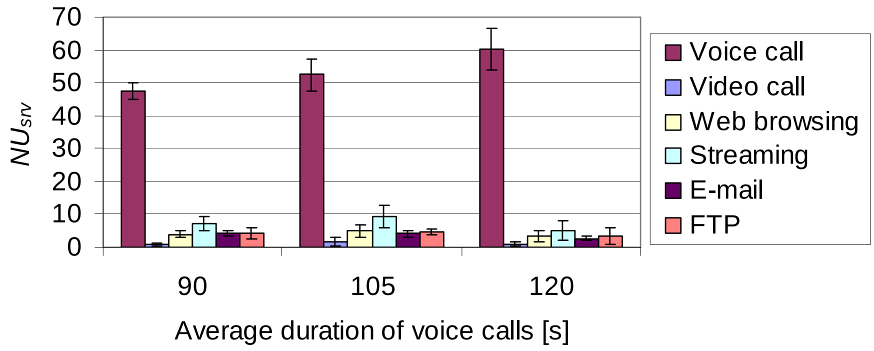


Figure 4-12 – Average number of users per service per second variation with average duration of Voice calls.

Consequences in performance indicators are expected from this increase. The Blocking Probability suffers direct impact, whilst Average Delay is indirectly affected. The impact is well perceived in P_b and τ , Figure 4-13 and Figure 4-14. In particular, an increase of 15 s in the average duration of Voice calls relatively to REF can raise P_b in 3 % and add about 10 ms to τ .

P_b variation with the duration of Voice calls is well modelled by any of the four functions, with correlation coefficients above 90 % in all the four cases (Linear, Polynomial, Power and Exponential). A perfect matching is obtained with a Polynomial adjustment, with R^2 =100 %, and the worse case is the Exponential adjustment, with R^2=91.81 %, Table 4-7.

Like P_b, a good adjustment is achieved for τ with any of the four functions, since Correlation Coefficients above 90 % are achieved in all cases. Once again, the adjustment with a Polynomial function is perfect, as shown in Table 4-8.

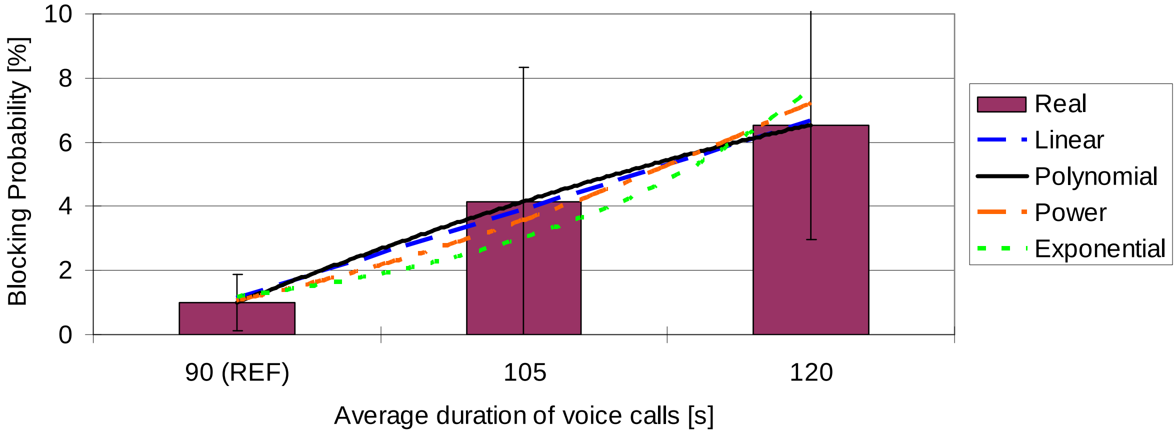


Figure 4-13 – Blocking Probability variation with average duration of the voice calls.

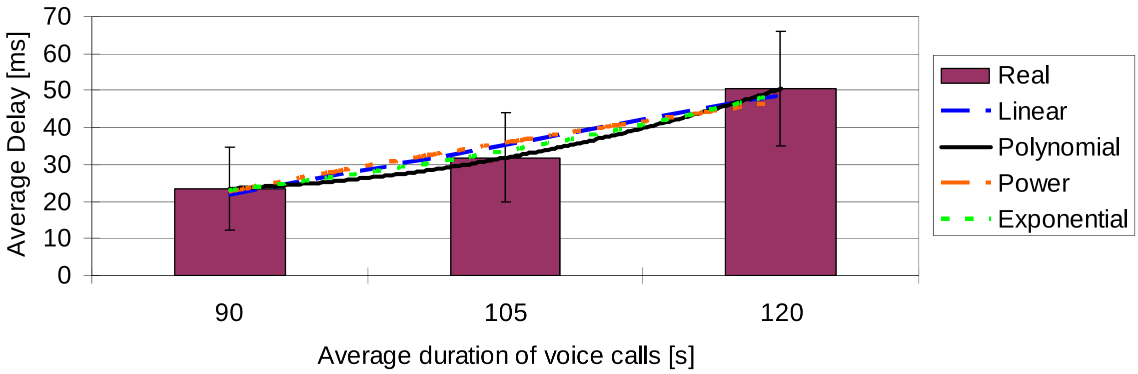


Figure 4-14- Average Delay variation with average duration of the voice calls.

Table 4-7 – P_b numeric values and Correlation Coefficients.

$VCAD$ [s]	P_b^{sim} [%]	$\overline{P_b^{lin}}$ [%]	$\overline{P_b^{pol}}$ [%]	$\overline{P_b^{pow}}$ [%]	$\overline{P_b^{exp}}$ [%]
90	1.00	1.13	1.00	1.06	1.18
105	4.16	3.90	4.16	3.56	3.01
120	6.54	6.67	6.54	7.22	7.69
		R^2=99.35%	R^2=100.00%	R^2=98.05%	R^2=91.81%

Table 4-8 – Average delay numeric values and Correlation Coefficients.

$VCAD[s]$	τ^{sim} [ms]	$\overline{\tau^{lin}}$ [ms]	$\overline{\tau^{pol}}$ [ms]	$\overline{\tau^{pow}}$ [ms]	$\overline{\tau^{exp}}$ [ms]
90	23.53	21.80	23.53	22.53	22.91
105	31.81	35.27	31.81	35.76	33.55
120	50.46	48.73	50.46	46.86	49.13
		R^2=95.30%	R^2=100.00%	R^2=92.87%	R^2=98.56%

Performance degradation is also perceived at the handover level, Figure 4-15. P_{VHOf} is increased in 10 % as the duration of Voice calls goes from 90 to 105 s, D_r growing 2 % in the same interval. No visible tendency can be identified in the P_{HHOf}.

Increasing the average duration of Voice calls means that, on average, each voice call user remains active during a longer period of time. The relative weight of the voice calls in the overall network traffic is therefore increased. As it is known, Voice is the less demanding service in terms of bandwidth, so it is expectable that the Average Bitrate of the preferred RAN for voice calls (UMTS) is decreased. Figure 4-16 represents the Average Bitrate of UMTS, and a reduction of 40 kbps is well visible when the average duration of Voice calls is raised from 90 to 105 s. The second RAN in the hierarchy for Voice calls assignment, Wi-Fi, is also affected, its Average Bitrate breaking some 200 kbps. The less preferential RAN for Voice, WiMAX, does not seem to feel this effect.

The processed data of the simulations runs for this case can be found in Annex F.

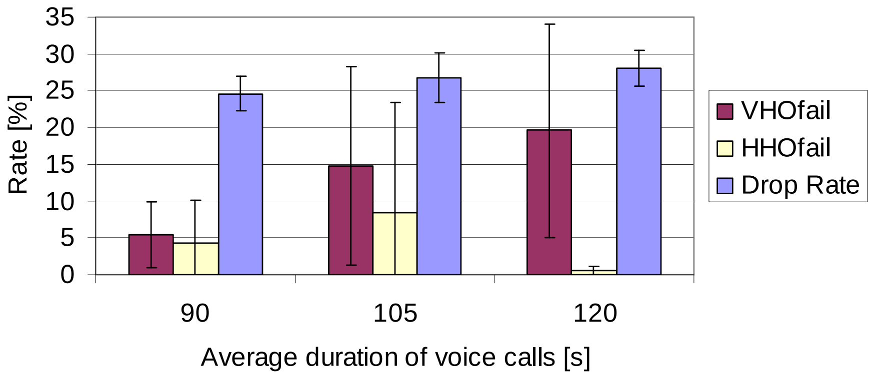
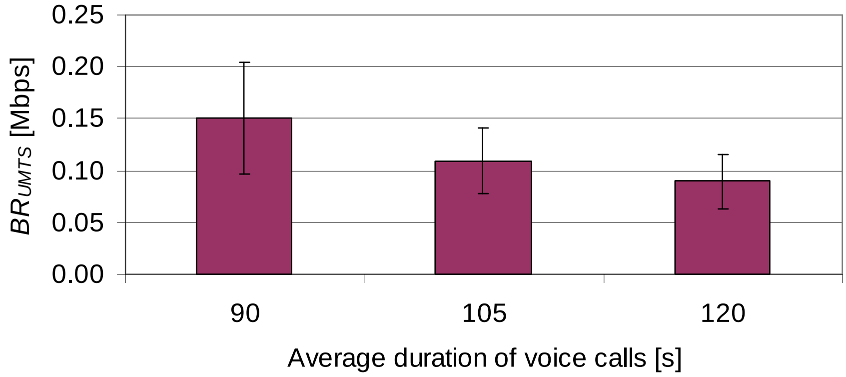
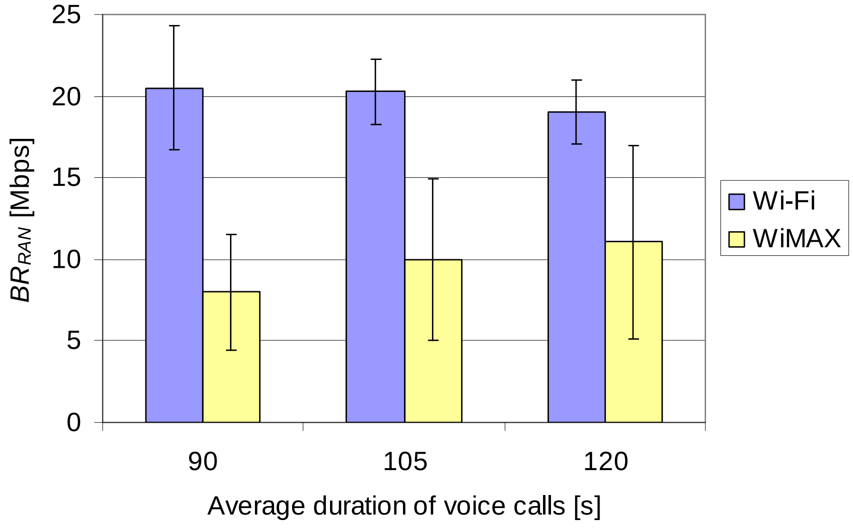


Figure 4-15 - Vertical, horizontal handover failure rate and drop rate variation with average duration of the voice calls.



a) UMTS



b) B) Wi-Fi and WiMAX

Figure 4-16 – Average bitrate per system variation with average duration of the voice calls.

4.2.4 Packet volume of data services

The volume of the packets of the “data services” (non-conversational) is generated according to a Pareto distribution with parameters α and k. In REF, the parameters values are 1.1 and 80 byte (Table 4-3); to create two additional scenarios for testing, the volume parameter (k) was multiplied by 2 and 3, to obtain k =160 byte and k =240 byte. The k parameter value is common for Web browsing, Streaming, E-mail and FTP services. The increase of the packets volume causes additional consumption of network resources, and this effect reflects in network performance. In particular, direct impact is expected in τ whilst P_b should be indirectly affected. The results show that there is a linear relationship between τ and k with a Correlation Coefficient above 96 %. Once again, perfect matching is achieved with a Polynomial curve. When k is doubled from 80 to 160 byte, P_b and τ follow the tendency and are also multiplied by 2, Figure 4-17 and Figure 4-18. The results for P_b present a high dispersion and are not so reliable as the ones obtained for τ, Table 4-9.

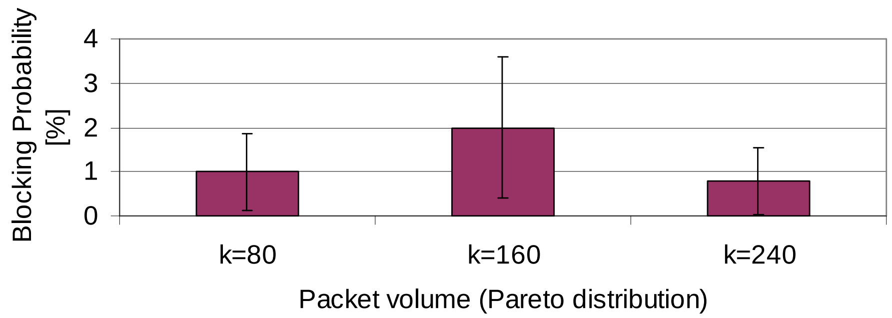


Figure 4-17 - Blocking Probability variation with packet volume of data services.

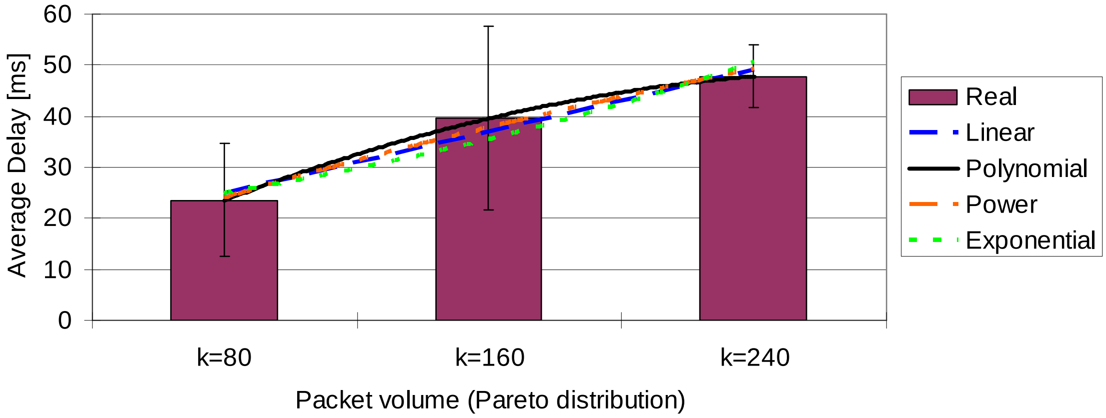


Figure 4-18 – Average Delay variation with the packet volume of data services.

Table 4-9 – Average Delay and exponential adjustment VS packet volume of data services.

k	τ^{sim} [ms]	$\overline{\tau}^{lin}$ [ms]	$\overline{\tau}^{pol}$ [ms]	$\overline{\tau}^{pow}$ [ms]	$\overline{\tau}^{exp}$ [ms]
80	23.53	24.83	23.53	23.94	24.86
160	39.59	36.98	39.59	37.76	35.45
240	47.84	49.14	47.84	49.29	50.55
		R^2=96.68%	R^2=100.00%	R^2=98.73%	R^2=93.24%

In WiMAX, bigger data packets mean more resources consumption, therefore, less available resources and fewer users served. In UMTS, a bigger packet takes more time to be transmitted, thus, keeping the user active for a longer period (assuming that throughput is constant). Due to this, the average number of users per time unit increases, causing an increase of the load factor. UMTS responds to this with cell breathing mechanism, trying to reduce the number of active users. It is therefore expected that the number of active users per second is lower as k increases. There is indeed a tiny reduction of the "non-conversational" users, as suggested by Figure 4-19, which can be caused by reduction of accepted sessions caused by cell breathing (shrinking).

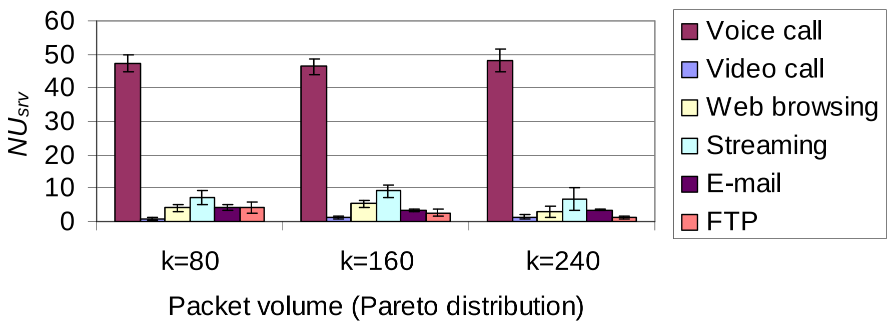


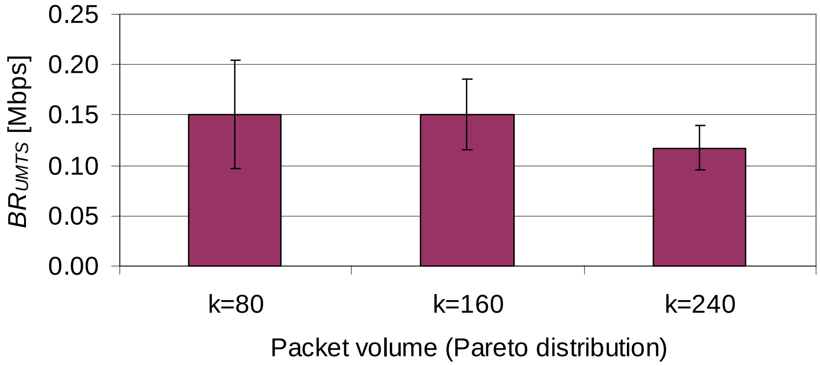
Figure 4-19 – Average number of users per second per system variation with packet volume of data services.

In UMTS, the system can also try to respond to a load factor increase by reducing the available bitrate per user. In WiMAX that is also true, since more interference will have direct impact in AMC, more robust modulations being privileged in that situation. In the simulator though, the bitrate is not sensitive to "real" interference since the bitrate is given by a statistic. Hence, bitrate behaves differently in UMTS and WiMAX, Figure 4-20: remaining constant or decreasing in UMTS, increasing to satisfy a higher resources demand in WiMAX, even if done at the expenses of a reduction of active users.

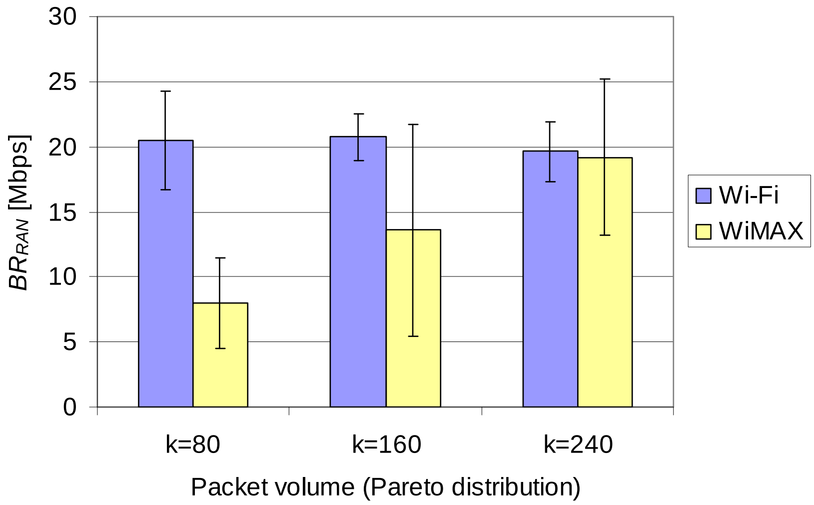
D_r is not affected by the data volume, and less can be concluded regarding handover failure, Figure 4-21, except that it is maximum for the middle value of the interval. No visible reason can explain this peak, except for the fact that N_{VHOa} is minimum for k =160 byte, or that N_{VHOf} is maximum. This is probably due to a punctual effect of the simulator.

One can finally conclude that, before a packet volume increase, the HN responds passively in terms of blocking, and passively and linearly in terms of delay. The network response is also passive for WiMAX bitrate, which grows together with the packet volume. The network, though, responds actively to the volume data increase as far as UMTS bitrate is concerned, by lowering down the available bitrate per user.

The whole processed data of the simulations runs for this case can be found in Annex G.



a) UMTS



b) Wi-Fi and WiMAX

Figure 4-20 – Average bitrate per system variation with packet volume of data services.

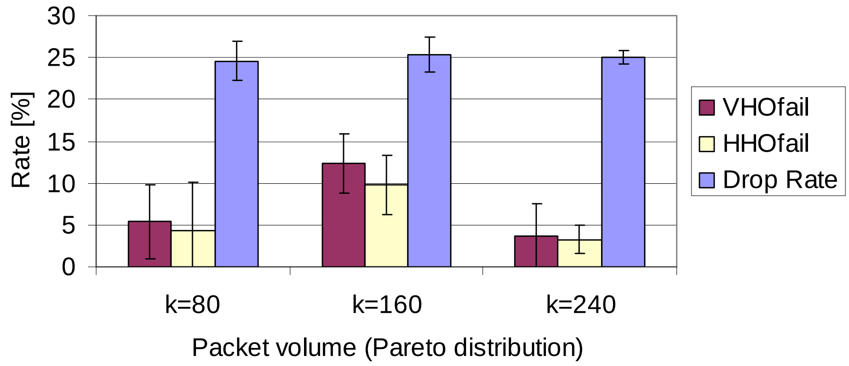


Figure 4-21 – Vertical, horizontal handover failure rate and drop rate as a function of the packet volume of data services.

4.2.5 Service Penetration

Nowadays, Voice is still the reference service both in terms of usage and revenue for mobile operators. This fact is reflected in the service penetration configuration of REF, where it is given a share of 50 %. Data services penetration, though, has been growing continuously over the last years, in result of the growing popularity of mobile Internet and the widespread of smart phones.

The objective of this case is to simulate in some way a reduction of Voice penetration, in benefit of non-conversational services. Three scenarios have been designed and tested. In the first, Voice penetration is reduced by 10 %, while Web browsing is raised 10 %. In the second scenario, Web service is raised 20 % at the cost of a 20 % reduction of Voice. Finally, Web browsing and E-mail are raised 10 % each at the cost of a 20 % reduction of Voice for the third scenario.

As depicted in Figure 4-22, a 10 % migration of users from Voice to Web causes an average reduction of 9 Voice users per second and an increase of 6 Web service users. If the migration is 20 % instead, there is less 19 Voice users and plus 8 Web users per second. One can also observe that the conversion of 10 % of Voice into E-mail is reflected in 6 more E-mail users per second.

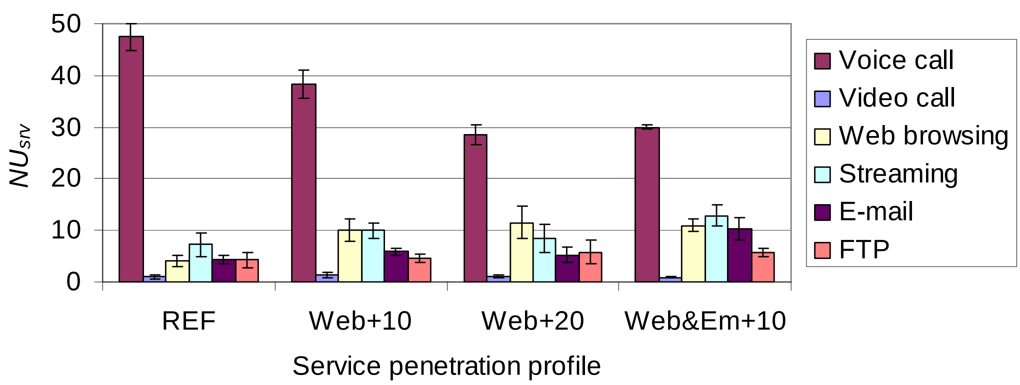


Figure 4-22 – Number of users per service per second variation with service penetration.

Compared to REF, all the three test scenarios represent a reduction of conversational traffic and a consequent increase on data traffic in the network. It is quite possible that the three alternative service profiles will raise the traffic load of the network, leaving fewer resources available. Given the fact that there are less Voice users in any of the alternative scenarios, a trade-off between available resources and number of Voice calls determines the behaviour of P_b. Hence, on the one hand, less Voice calls means also less blocked Voice calls; on the other, more resources consumption (if it is the case) means less available resources, which can lead to more blocked calls.

Figure 4-23 provides the answer, showing that the dominant effect for determining the evolution of call blocking is in fact the reduction of Voice calls. Moreover, the results show that a migration of 10 % of the users from Voice to Web service is responsible for a reduction of 0.27 % in P_b. It is curious to notice that the reduction is duplicated (0.54 %) if 20 % of users were migrated from Voice to Web instead. Unexpectedly though, P_b is almost the same whether 10 % of the Voice users are migrated to Web or 20 % of Voice users are migrated to Web and E-mail services instead. Nevertheless, one should not ignore the amplitude of the dispersion, showing that different results could have been obtained if the sample was larger.

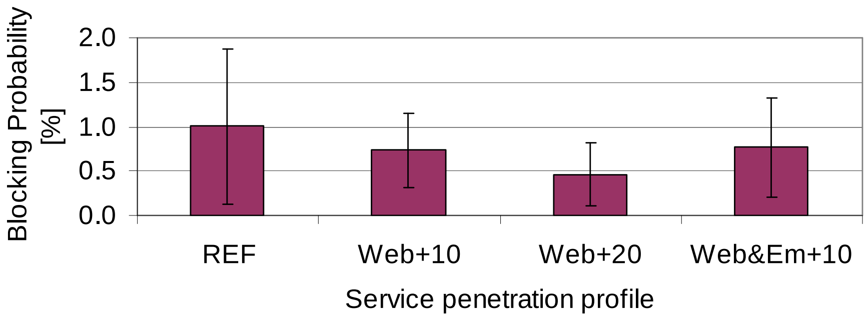


Figure 4-23 - Blocking Probability variation with service penetration.

A different behaviour should be observed for τ. Since Voice call users are not delay generators, the value of the parameter is expected to increase should the number of “non-conversational” users increase from 10 % to 20 %. However, the growing tendency in terms of average value is not evident, Figure 4-24. Anyway, the dispersion of the results “Web+20 %” and “Web&Email+10 %” is smaller, and if one compares the lower limit of the standard deviations instead of the average, those scenarios will have in fact higher τ compared to REF. To reinforce the confidence in any conclusion, the sample should be enlarged, by running a higher number of simulations.

Figure 4-25 depicts handover failure and D_r. It is visible that D_r is increased in 5 % when 20 % of Voice users become “non-conversational” services users. If one thinks that Voice calls are less tolerant by nature to resources exhaustion, and more vulnerable to a drop event than “non-conversational” ones, then the migration of Voice users to Web and E-mail should have resulted in a reduction instead. But at the same time, such migration increases the network load, which can lead to Drop Rate increase in turn. This is a trade-off in which the increase of the network load is more relevant, while dictating the behaviour of the parameter.

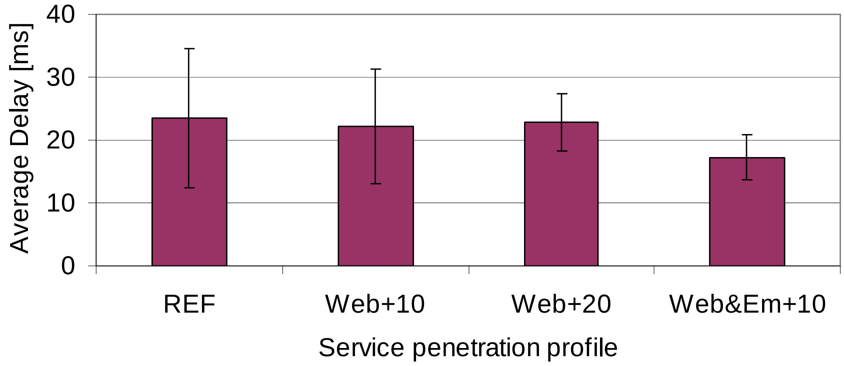


Figure 4-24 – Average Delay variation with service penetration.

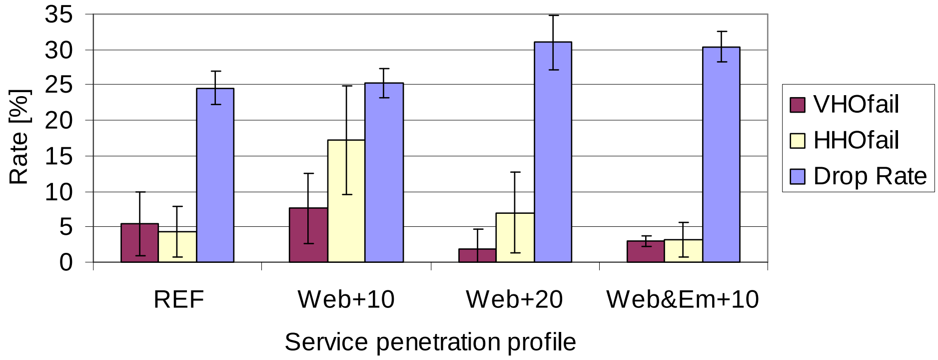
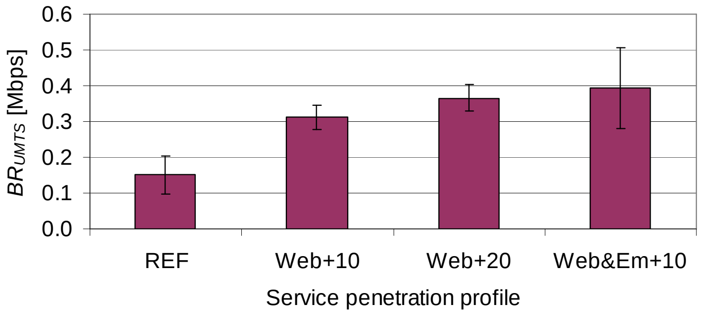


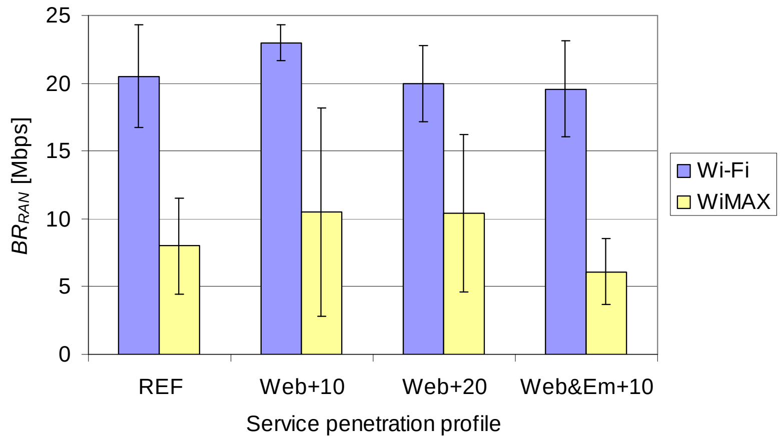
Figure 4-25 – Handover failure rate and drop rate variation with the service penetration.

Concerning system bitrates, Figure 4-26, one can observe that a conversion of 10 % of Voice users into Web service ones is responsible for the increase of 160 kbps in UMTS, 2 480 kbps in Wi-Fi and 2 490 kbps in WiMAX. In terms of simulator only, this can be directly related to the source models in use, and reflects the fact that a bigger number of high data rate services sessions is active. If the amount of converted Voice users raises to 20 % instead, UMTS bitrate is 220 kbps higher, while WiMAX is 2 410 kbps higher only; Wi-Fi bitrate is reduced in 530 kbps. Hence, a cumulative effect in the bitrates is only observed for UMTS, since both WiMAX and Wi-Fi bitrates are reduced from the 10 % to 20 %. In any case, compared to REF, UMTS and WiMAX bitrates are higher in almost all scenarios, reflecting directly and visibly the changes operated in the penetration of services.

The processed data of the simulations runs for this case can be found in Annex H.



a) UMTS



b) Wi-Fi and WiMAX

Figure 4-26 – Average bitrate per system variation with the service penetration.

4.3 System performance as a function of network-dependent parameters

Section 4.2 focuses on describing how the HNP function depends on the scenario and its parameters. The different scenarios are equal to REF, except in a given parameter, which is varied inside a certain interval. The variable parameters shared a common characteristic: they are essentially dependent on users. In this section, a new set of scenarios is created from a different perspective. They are also based on REF, but the variable parameter (or parameters, 2 at most) has a network dependency nature, instead of a user one. Hence, a first case addresses variations on the RAN priority table to handle sessions of each service; In a second one, RANs are selectively switched-off, while in the last case the bandwidth of WiMAX system is multiplied by 2.

4.3.1 Priority table

The service priority table for REF is based on the following assumption: Voice, Video calls and E-mail sessions are preferably steered to UMTS R99, Wi-Fi detains preference to handle Web browsing and FTP sessions, while Streaming is handled preferentially by WiMAX. Below maximum priority, 1, RANs are sub-prioritised from level 2 to 3. In the present case, 3 alternative priority profiles have been defined: the preference for handling non-conversational sessions (Web browsing, Streaming, E-mail and FTP) is given to WiMAX, then to UMTS R5 and finally to Wi-Fi, three test scenarios being created for the purpose of evaluating HNP.

The analysis of the P_b, τ, handover fail and D_r, represented in Figure 4-27, Figure 4-28 and Figure 4-29 respectively, reveals that the worse network performance is obtained when priority is given to

UMTS R5. On the opposite, the best performance is achieved when data services are allocated preferentially to WiMAX. One can also conclude that, from the three cases being tested, HNP improvement compared to REF is only achieved with WiMAX set as preferential: Blocking is reduced in 0.13 % and τ is reduced in 4 ms. HNP experiments degradation if Wi-Fi and UMTS are set as preferential. If one was to optimise HNP concerning the priority table, absolute preference to WiMAX could fit as a good first iteration in a convergence process. However, the optimisation of the priority table is a complex, challenging and interesting task to perform, being out of the scope of this thesis.

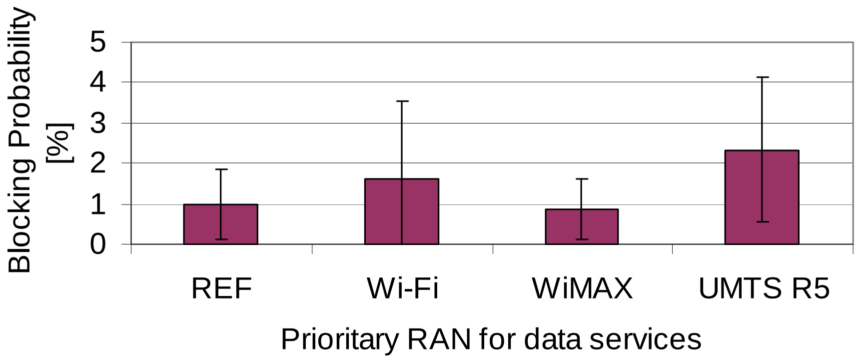


Figure 4-27 - Blocking Probability variation with priority table.

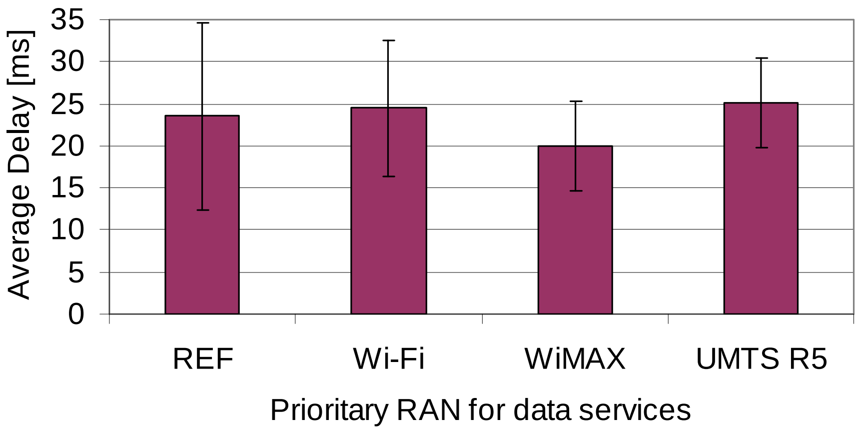


Figure 4-28 – Average Delay variation with priority table.

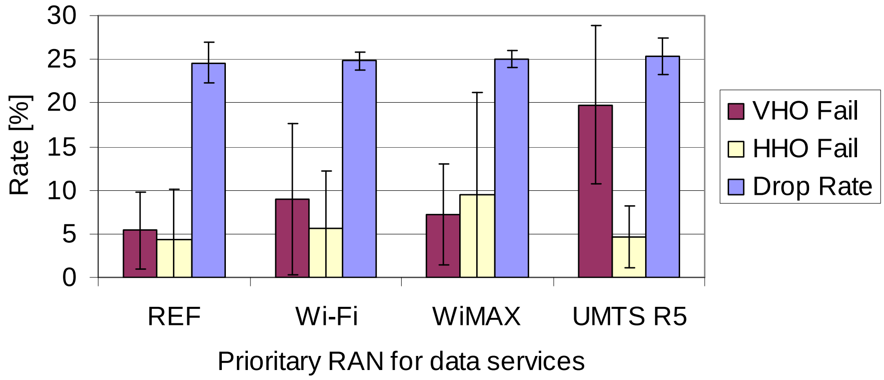


Figure 4-29 – Handover failure rate and drop rate variation with priority table.

In Figure 4-30, UMTS Average Bitrate is represented. The results show that the Bitrate is reduced in 30 kbps and 40 kbps when services are delivered preferentially to Wi-Fi and WiMAX respectively. This reduction can be motivated by a reduction of non-conversational service sessions assigned to UMTS. If one thinks of non-conversational services as more demanding services in terms of data volume, the fact that UMTS Bitrate is maximised when set as preferential for non-conversational traffic, jumping from 150 kbps to 490 kbps, is not surprising. But even if this was a somehow expected conclusion, it is not an obvious one. In fact, more demanding services being steered to UMTS could lead to quick resources exhaustion, and therefore to a load factor increase, which UMTS would try to compensate by reducing the global bitrate.

The processed data of the simulations runs for this case can be found in Annex I.

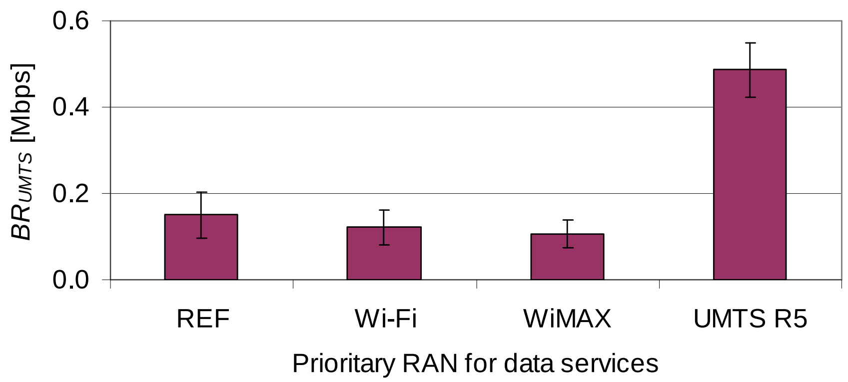


Figure 4-30 – UMTS Average Bitrate variation with the priority table.

4.3.2 Selective RAN switch-off

Given a HN integrating two or thee different types of RATs, it is interesting to reflect for a moment about how would network performance respond should all its sites be equipped with the same type of technology. That is to say, how would the equivalent homogeneous network perform. And if, hypothetically, the result was an improvement, with which type of technology would be recommended to build the homogeneous network. Provided that the amount of spectrum resources at stake was the same, the answer would be anything but trivial and probably worthy of a whole (or part of a) separate thesis. Such type of work is not performed in this thesis, but rather a similar one, which consists on evaluating HNP while selectively switching-off one of its sub-networks.

To perform the task, three additional scenarios were built, and in each of them UMTS R5, Wi-Fi and WiMAX BSs/APs respectively have been switched-off. Since neither the number nor the relative positioning of BSs/APs on the field is homogeneous for the three technologies, the comparison is not exactly fair, and one should take that into account while taking conclusions.

Figure 4-31 and Figure 4-32 show that the performance of the network is seriously affected in terms of blocking and delay when WiMAX is turned off, with P_b raising from 1 % to 12 % and τ raising strongly from a mere 24 ms to 2 s. Since WiMAX and UMTS are using the same channel bandwidth, 5 MHz, one can conclude that the spectral efficiency of WiMAX is higher in this particular case. D_r

and HO failure statistics however point WiMAX as the main contributor, not only for dropped sessions and HO failure rate, as Figure 4-33 demonstrates, but also for HO attempts, as shown in Figure 4-34. It is also interesting to observe that P_{VHOf} is minimum when the 5 Wi-Fi APs are off, which makes some sense. If one considers the users connected to UMTS or WiMAX BSs, that at some moment of time are forced to make handover, thus, becoming subject to VHO failure, it is easy to understand that eventual VHO failures turn on simple session drops when Wi-Fi signal is off. It is also visible that N_{VHOa} is minimised when WiMAX or UMTS R5 are off, which has probably something to do with the fact that areas with two available bearer signals are substantially reduced when one of the systems is off.

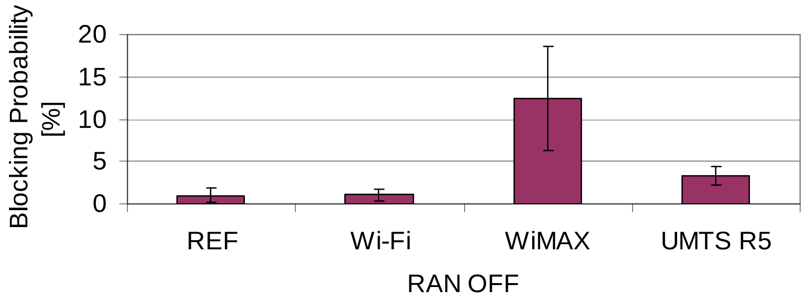


Figure 4-31 - Blocking Probability variation with selective RAN switch-off.

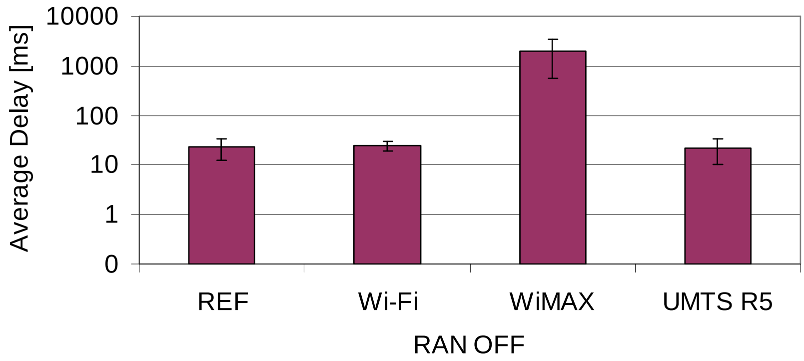


Figure 4-32 – Average Delay variation with selective RAN switch-off.

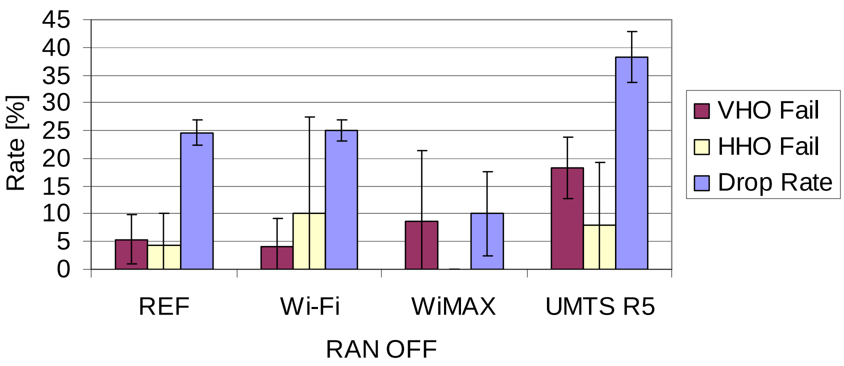


Figure 4-33 – Handover failure rate and drop rate variation with selective RAN switch-off.

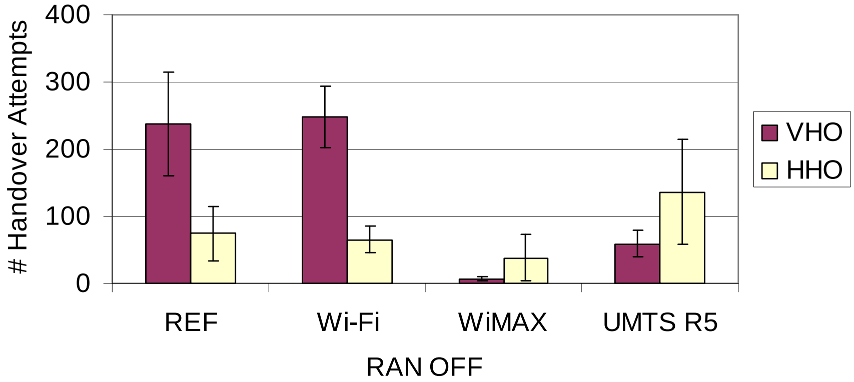
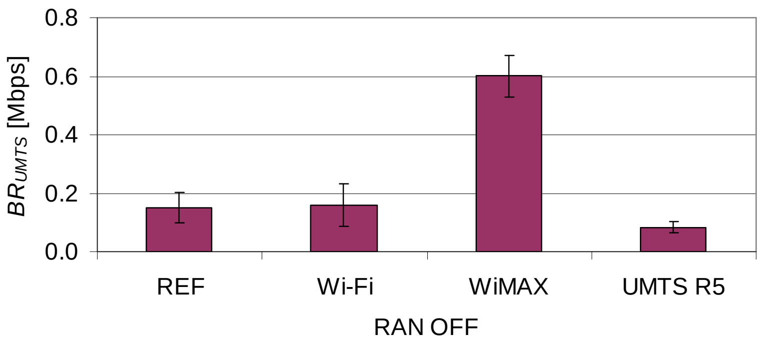
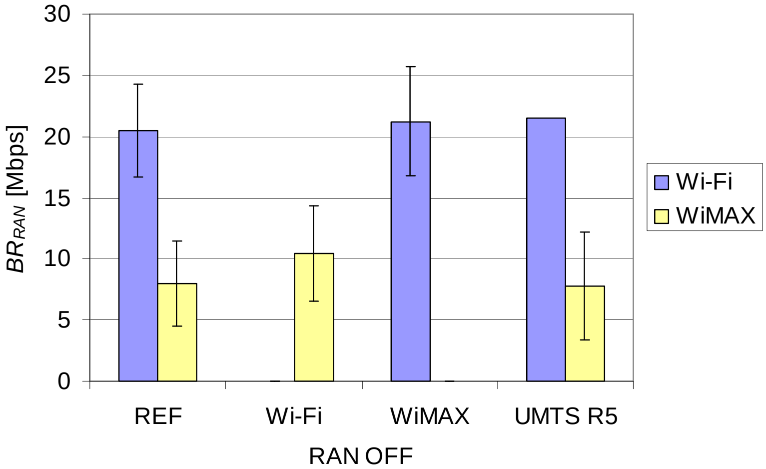


Figure 4-34 – Handover variation with selective RAN switch-off.

Regarding system bitrates, Figure 4-35, two aspects must be referred. First, UMTS bitrate is maximised when WiMAX is off, which is quite obvious, since WiMAX is a “competitor” for high data-rate services like Streaming (see Table 3-1). On the opposite side, UMTS bitrate is minimised when R5 BSs are switched-off: as UMTS R99 is preferential for Voice calls, the weight of Voice traffic in UMTS in general will increase, thus pushing the bitrate downwards.



a) UMTS



b) WiMAX and Wi-Fi

Figure 4-35 – System bitrate variation with selective RAN switch-off.

WiMAX bitrate is maximised when Wi-Fi is off exactly for the same reasons described previously, since Wi-Fi is a competitor for Web browsing and FTP, and Wi-Fi Bitrate is slightly increased with its "competitors" WiMAX and R5 switched-off.

A last comment on the reduction of Voice users when UMTS R5 is off, Figure 4-36. The reduction is explained by the resources allocation concept, which was set for each system. One can conclude that UMTS R5 is a more "democratic" system, with a "give less to reach more" policy, against the model defined for WiMAX resources allocation in the simulator, which is grounded on the idea of "reach less but give more". While switching-off WiMAX does not impact on the number of users significantly, switching R5 off causes a reduction of about 15 Voice users per second.

The processed data of the simulations runs for this case can be found in Annex J.

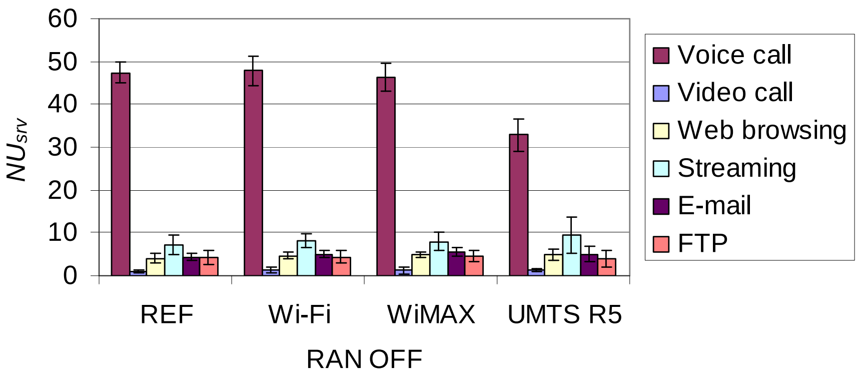


Figure 4-36 – Average number of users per service per second variation with selective RAN switch-off.

4.3.3 WiMAX channel bandwidth

WiMAX is prepared to work with channel bandwidths from 1.25 to 20 MHz. While running REF, in which a 5 MHz channel is used, performance indicators show that the network is working fairly under its maximum capacity in terms of QoS. Thus, raising the channel bandwidth to 10 MHz would probably be less more than unnoticed in terms of network performance. To better evaluate the impact of such capacity enhancement in network performance, the bandwith variation was tested with REF, but with 14 000 users instead of 10 000. The idea is to leverage the network load and introduce some degradation in the performance indicators, in order to enlarge the improvement margin of a duplication of WiMAX bandwidth.

Figure 4-37 shows the results of four sets of simulations, relative to two scenarios with 5 MHz WiMAX bandwidth (10 000 and 14 000 users) and two other scenarios using 10 MHz WiMAX bandwidth. As the figure shows, no improvement is detected in terms of P_b, but rather a tiny degradation from 5 MHz to 10 MHz bandwidth with 14 k users. This small variation though should be considered irrelevant due to the amplitude of error margin (see Table K-1). However, the improvement is clearly visible in τ, Figure 4-38, which is reduced in 20 % in the 10 000 users scenario, and in more than 30 % in the 14 000 users scenario.

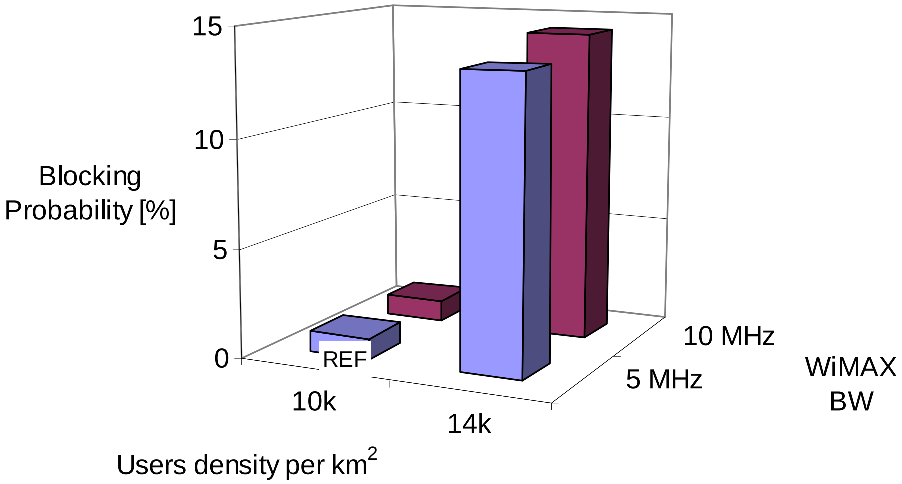


Figure 4-37 - Blocking Probability variation with the bandwidth of the WiMAX channel.

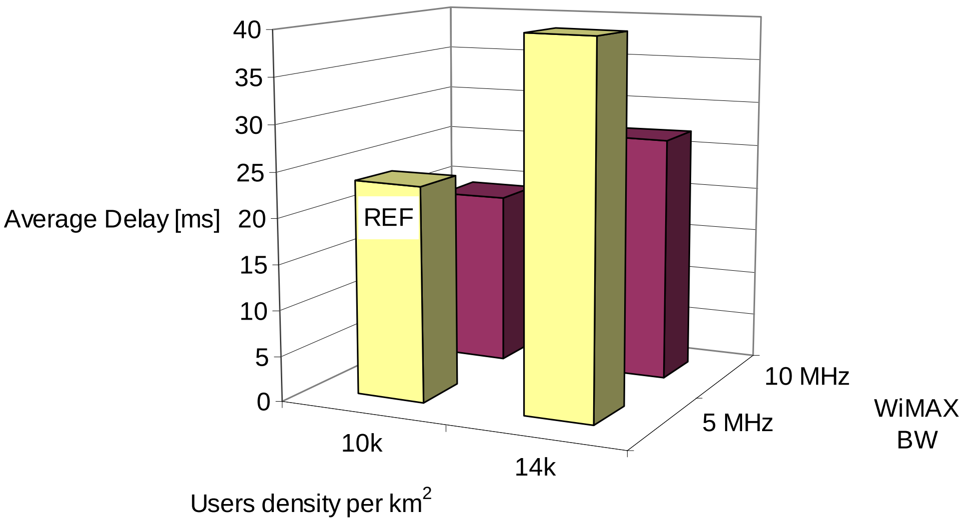


Figure 4-38 – Average Duration variation with the bandwidth of the WiMAX channel.

WiMAX bandwidth enhancement does not seem to be efficient to reduce P_{VHOf} . As Figure 4-39 shows, it is insensitive to system bandwidth duplication. In fact, the parameter depends only partially of capacity features, and it is not mandatory that it improves after a capacity enhancement.

Figure 4-40 depicts the variation of BR_{RAN} . Two interesting effects are visible in the graphics and are worthy of more detailed analysis. The figure shows an increase of UMTS bitrate when WiMAX bandwidth goes from 5 MHz to 10 MHz, showing a correlation between the two parameters. This is necessarily an indirect correlation, due to no cause-effect direct relationship exists between the parameters. A possible explanation for this is the following: the capacity enhancement of WiMAX might be causing a reduction of UMTS load factor, which in turn is reflected in less pressure to reduce the bitrate of the active sessions. Nevertheless, there may be other explanations as valid as this one.

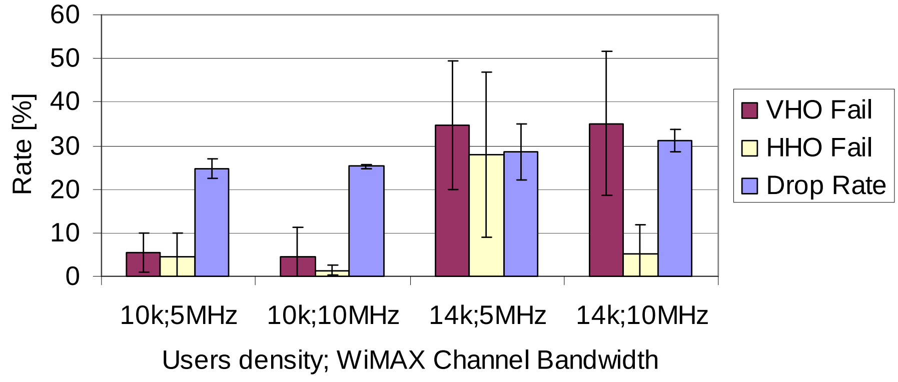
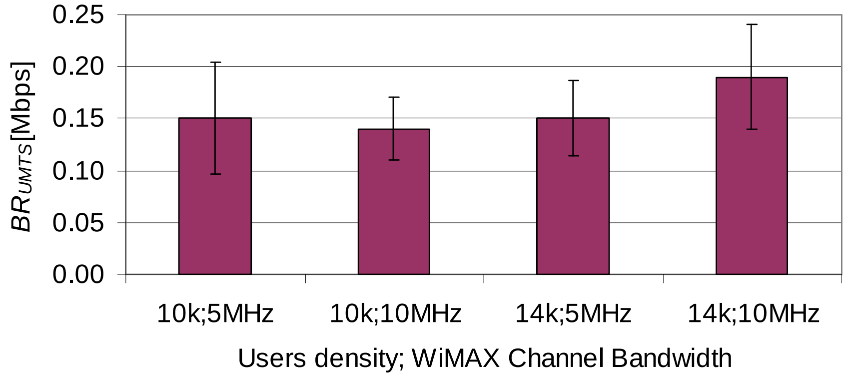
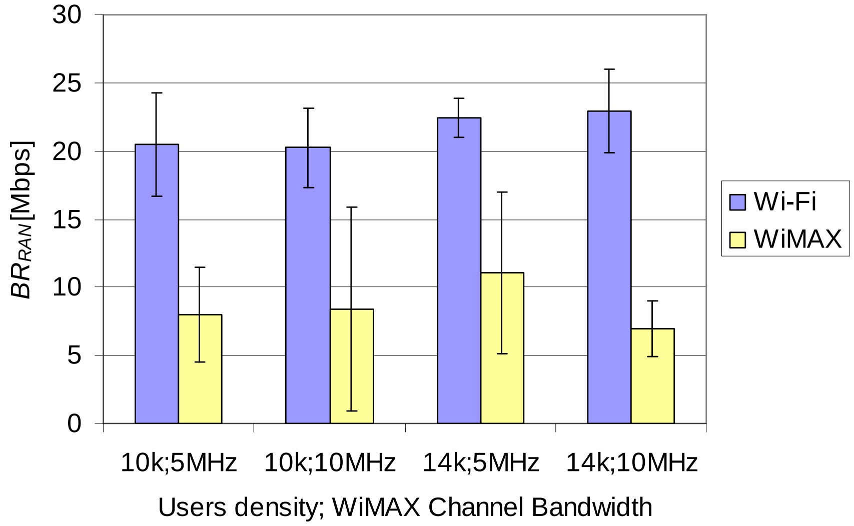


Figure 4-39 – HO failure and Drop rate variation with the bandwidth of the WiMAX channel.



a) UMTS



b) Wi-Fi and WiMAX

Figure 4-40 – Bitrate per system variation with the bandwidth of the WiMAX channel.

The other interesting effect is the reduction of WiMAX bitrate. According to the models defined for WiMAX in the simulator, this reduction can only be obtained through an increase of the relative weight of lower bitrate services like Voice, and/or a reduction of the relative weight of “heavier” services like Web browsing or Streaming. And, in fact, it can be happening that a large number of Voice calls (by far the most common service with a 50 % penetration) that would have been blocked by a system with a WiMAX 5 MHz channel, can be now being steered to the WiMAX additional capacity brought by a broader channel, thus, contributing to lower the Average Bitrate.

The processed data of the simulations runs for this case can be found in Annex K.

Chapter 5

Conclusions

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

Over the last years, several wireless technologies have experienced significant growth and commercial success in the mass-market. From GSM to UMTS/HSPA, through GSM/GPRS, WLAN, WiMAX, Personal Area Networks and others, the wireless world has become more heterogeneous than ever. Beyond 3G wireless systems will be based on a variety of coexisting Radio Access Technologies, partially or totally overlapped in coverage, with complementary technical characteristics that will physically coexist in a seamless integrated environment. Nowadays, there are already examples of operators owning distinct wireless networks, serving the different purposes of their specific clients, like it happens in Portugal, with the operator Portugal Telecom exploiting UMTS/HSPA and Wi-Fi countrywide commercial networks.

With the different RATs coexisting, overlapping in space, and owned by the same entity, the next challenge is how to exploit them in a coordinated manner, in order to constantly enhance users experience and satisfaction, by providing them ubiquitous service and higher bitrates. From the operators point of view, Mobile Communications is one of the most dynamic and profitable market sectors in current economics, but it is also one of the highest demanding sectors from the point of view of the required investment. The economic exploitation of the solutions directed towards the optimisation of heterogeneous networks performance is therefore a key issue. This can be summarised in a basic paradigm: the HN becomes transparent to the final user and he/she must be served with the most efficient radio access at each instant. The key to this paradigm, in which several investigators have been working in the last years, is CRRM.

The present work is devoted to CRRM and can be divided in two main stages: the first one was dedicated to build a software simulation tool. A system level, time based simulator has been developed over the Microsoft Visual Studio 2005 platform. The simulator is able to reproduce a heterogeneous environment composed by a combination of UMTS (R99 and HSDPA), Wi-Fi and WiMAX technologies, implementing low-level system functionalities like power control, load and access control and interference estimation. In the second phase, the simulator was used to run simulations with different scenarios, in order to extract results to evaluate the performance of a CRRM algorithm based on a service/RAN priority table. A liquid total of 47 days of CPU processing time has been spent to produce the results. The different scenarios are based on a Reference Scenario (REF), and in most cases they differ from it in a single parameter, which can be user- or system-dependent. The performance of the network is evaluated according to a Heterogeneous Network Performance function, defined as a column matrix with entries corresponding to the output parameters of the simulator: Blocking Probability, Average Delay, Handovers, Dropped Sessions, Bitrates, and Users.

The Reference Scenario is composed of 2 WiMAX BSs, 2 UMTS R5 BSs and 1 UMTS R99 BS, geographically distributed around a hotspot of 5 Wi-Fi APs. The background is an urban environment, physically located between Campo Pequeno and Saldanha, in Lisbon. Users are assumed to be static and uniformly distributed in a 1000 x 1000 m^2 area. A mono-service policy is adopted, in which only one type of service is allowed per user throughout the whole simulation period. Six services are defined: Voice call, Video call, Web browsing, Streaming, E-mail and FTP.

The simulator developed to serve as the basis for this thesis is quite a complex software system. Its main objective is to reproduce a real network environment in all its particular aspects, as close and faithfully as possible. In spite of the efforts to achieve that idealistic target, a software system of this type is only feasible (and runable) while applying approximations and models that help on recreating reality without introducing great distortion. The current simulator is not an exception, and makes use of several simplifications, some of which are hereby presented and discussed.

One of the major simplifications of the simulator is the fact that it accounts for the downlink channel only. This simplification would not be relevant if the services under consideration are highly asymmetric. But, as one knows, one assists nowadays to a global tendency towards symmetry, to which technologies like HSUPA seek to answer adequately. In order to keep synchronised with the reality, the simulator clearly needs an improvement in that particular aspect.

Other limitations can be pointed to the simulator. For example, the fact of the network structure of the Reference Scenario being a single cluster can be responsible for unrealistic values of Handover failure and Drop rate, mainly caused by users in the cells boarders, which are vulnerable victims to power control. One should also bear in mind that only the radio perspective of the network is being simulated, Transmission and Core networks aspects being ignored.

The simulations performed for this thesis consider static users. Despite of the user interface of the simulator offering that possibility, introducing mobility in simulations represents a huge additional processing load that can raise the processing times significantly. Optimisation work, perhaps based on sophisticated mathematical models is required in order to minimise that problem and produce more realistic results.

The software tool makes also use of several small approximations that may introduce a given amount of distortion. Most of them have been described throughout the thesis, like the harmonisation of the resolution UMTS/WiMAX, or using Walfisch-Ikegami model to estimate propagation loss in WiMAX. Simplifications like using a statistic to retrieve WiMAX AMC and other simplifications over WiMAX have allowed the adaptation of this system to a simulation environment, hopefully without a great distortion of the reality.

The simplicity of the handover algorithm is also one of the aspects that need more attention in a future release of the simulator. No intelligence beyond network choice according to the priority table is associated to a handover operation in the simulator, where the network swap is realised immediately after the instantaneous power of the received signal falls below a given threshold.

Regarding the most significant results, and starting by the Blocking Probability, its maximum value, 22.18 %, was obtained in the Normal distribution of users scenario with a standard deviation of 150 m. This huge figure is greatly above acceptable values (aprox. 2 %) and shows that the network configuration of REF together with the CRRM algorithm being used are clearly under-dimensioned for the number and geographical distribution of users in the covered area. It has also been observed that P_b variation with the number of users is well modelled by an exponential function (R^2=94.7 %), and even better by a polynomial function of 2nd degree (R^2=96.6 %) in the interval 8 000 to 14 000 users.

The variation of this parameter with the average duration of Voice calls is almost perfectly linear in the interval 90 s to 120 s. Being a parameter associated to conversational services, P_b exhibits indirect dependency of the packets volume variation: while duplicating packets volume (*k*) from 80 to 160 bytes, P_b is also duplicated from 1 % to 2 %. It was also observed that P_b can break more than 0.5 % when the relative penetration of the Voice service is reduced in 20 %.

Regarding P_b variation with system parameters, it is maximised when UMTS R5 is chosen as preferential for non-conversational services in terms of priority table, which demonstrates that the preference for that kind of services should not be given to UMTS in exclusive. P_b can go beyond 10 % when WiMAX RAN is switched-off. It is also practically insensitive to the duplication of the bandwidth of the WiMAX channel.

Average delay, τ, is associated to non-conversational services and is a measure of the delay affecting them. From all processed scenarios, τ is maximum when WiMAX is switched-off, reaching more than 2 s. Considering that τ never goes beyond 30 ms with Wi-Fi or HSDPA switched-off, the importance of WiMAX in the HN performance is in this way put in evidence.

The variation of τ with the number of users is well modelled by a Polynomial function. The results also show that τ is indirectly affected by the variation of the average duration of Voice calls, where a difference of 15 s can raise τ in 15 ms. Average Delay is also directly affected by the variation of the volume of data packets, where a difference of 80 bytes in packets volume is responsible for more than 15 ms of additional delay. The relationship between τ and packet volume is nearly linear in the interval 80 to 240 bytes. Regarding service penetration, and somehow against expectations, the variation of τ is not significant when the penetration of non-conversational services is raised 20 %. Regarding the variation with system parameters, τ is minimised when WiMAX is set as preferential in terms of priority table for non-conversation services, and does not vary significantly when other RANs are set as preferential. As already stated, the strong dependency from WiMAX is noticed when this system is switched-off, with the value of τ being raised about 100 times. Average delay has shown to be sensitive to WiMAX channel bandwidth in heavy-load conditions: with 14 000 users, a duplication of WiMAX bandwidth is responsible for a reduction of 13 ms.

Other output parameters, like the number of Handovers, Handover failure and Drop rates, help in characterising HN performance. Regarding the number of VHO attempts, it has been observed that a linear dependency with the density of users exists in the interval 8 000 to 14 000. The failure rate of VHO is maximum (35 %) with heavy network load, i.e., with 14 000 users. Its variation with the mean duration of voice calls is 10 % per 15 s. VHO failure rate figures also contribute to put in evidence the benefits of having WiMAX set as preferential for data services, since a minimum of 7.2 % is achieved in that condition. Switching off Wi-Fi leads to minimisation of P_{VHOf} , and it is insensitive to the duplication of WiMAX channel bandwidth. However, the results obtained for Handover must be taken with some reserve, since simulations are considering static users.

In terms of Drop rate, results show that a peak value of 38.2 % is obtained when UMTS R5 is switched-off. It is also shown that D_r grows 2 % when the mean duration of voice calls is 15 s longer, but it is nearly insensitive to packets volume variation. Concerning the dependency of the parameter with service penetration, Drop rate is 5 % higher when the penetration of Web service is increased in 20 %. In terms of user dependent parameters, D_r exhibits strong dependency of RANs switch-off. Like for P_{VHOf} , D_r is maximised when UMTS R5 is off, expressing the importance of this system for QoS. But, quite unexpectedly, the results show that WiMAX is responsible for a degradation of D_r , since a 15 % higher figure is obtained for the REF scenario compared to the one where WiMAX is off. D_r is also insensitive to the duplication of WiMAX bandwidth.

Bitrates are, by definition of the simulator, strongly dependent of the source models. In this way, Web browsing and Streaming are, generally speaking, the most demanding services in terms of service bitrate. At the system level, Wi-Fi bitrates floating around 20 Mbps are 20 times higher than UMTS ones, and WiMAX works as the mid-term, with moderate throughputs usually close to 10 Mbps. The three systems can be seen as complementary from the point of view of the bitrate, and when integrated in a HN structure, provide the operator an interesting diversity of QoS segments.

As the preferential sub-system for Voice service, BR_{UMTS} exhibits a visible dependency of the average duration of Voice calls, where a difference of 15 s can originate a reduction of 40 kbps. Wi-Fi is also sensitive to this parameter, suffering a 200 kbps break in the same interval. The variation of the packet volume causes a similar reaction in BR_{UMTS} , where an increase of 80 bytes is responsible for a 30 kbps break. WiMAX follows an opposite tendency, growing more than 5 Mbps in the same interval. This puts in evidence the reactive nature of the simulator regarding BR_{WiMAX} against its pro-active one as far as BR_{UMTS} is concerned. It turns visible that the HN offers resources on demand to WiMAX users, while manages UMTS according to a more “democratic” perspective, under a *give less to reach more* philosophy.

Probably, the most interesting aspect of the subject of this thesis in terms of future work is related to the development and testing of more sophisticated CRRM algorithms. The current thesis has been focused on a simple algorithm, based on a table of priority, but it can evolve towards interesting research targets, like for example, to optimise the priority table for a HN combining the 4 RANs – as already seen, establishing WiMAX preferential to handle non-conversational services is a good first iteration. Moreover, the ground is prepared to test other algorithms, and see how they can improve overall performance of a heterogeneous system.

Many other interesting examples of CRRM models can be pointed. If a mobility model was considered, one could implement a smart resources allocation algorithm taking decisions based on the direction and speed of MTs. In a different perspective, operators could assume themselves as sellers and bidders of resources, the former selling spare resources to the latter in real-time. According to another approach, different costs can be assigned to the resources of different RATs. In that case, resource allocation would be processed according to the minimisation of a cost function, thus, maximising network payoff. Network payoff maximisation can also be implemented through a more exotic game theoretic approach, where a game between the access networks, competing in a non-cooperative manner, is defined.

Despite of the simplifications and approximations used to build the simulator, it is undeniable that it represents a meaningful and rare effort to simulate a real heterogeneous network, and even more rare on its ability to congregate 3 (4) RATs. It is a fact that this software tool has not many competitors at his level, at least according to the results of the literature review presented in Chapter 2, and based on what is commonly known about worldwide work on the subject.

The subject of the present work, like Science in general, certainly does not end in a thesis. This Master Thesis represents the humble contribution of his author to go further towards the full knowledge on optimised management of heterogeneous networks of ours and future times.

Annex A

Traffic Source Models

In this annex, the traffic source models used throughout the present thesis are described.

A.1 Voice Models

For circuit-switched-based Voice service, a 4-state model is used [VRFo99], which is based on measurement and includes not only the ON-OFF behaviour but also the effect of the voice encoder, compression device and air interface.

The model described in [VRFo99] and [RaMe01] defines four-states: when the source is in state $k1$ it generates packets of size S_{k1} each 10 ms, for a burst duration of τ_{k1} , $k1$ = 1,…,4. In the long time average, the probability that a packet is of size S_{k1} is quantified as P_{k1} . The burst duration τ_{k1} is modelled as a random variable; with mean value m_{k1} , with the Weibull Probability Density Function (PDF) as follows:

$$f(x) = \beta_{k1} \cdot \lambda_{k1} \cdot (x \cdot \lambda_{k1})^{\beta_{k1}-1} \cdot e^{-(x \cdot \lambda_{k1})^{\beta_{k1}}} \tag{A.1}$$

where:

- $1/\lambda_{k1}$ is the scale parameter
- β_{k1} is the shape parameter

and, both taking the values defined in Table A-1.

Table A- 1 - Voice Source Model Parameters (partial extracted from [VRFo99]).

State $k1$	Packet size S_{k1} [Bytes]	Measured Probability P_{k1}	Measured mean burst duration m_{k1} [packet]	Weibull parameter λ_{k1}	Weibull parameter β_{k1}
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

After a $k1$ state, a new state is selected with probability Q_{k1} , which is defined as [RaMe01]:

$$Q_{k1} = \frac{\frac{P_{k1}}{m_{k1}}}{\sum_{j=1}^{4} \frac{P_j}{m_j}} \tag{A.2}$$

The voice calls generation process follows a Poisson process [Yaco93], and the duration of the calls is determined according to an exponential distribution.

VoIP services, in turn, present typically a symmetric or quasi-symmetric nature and require small end-to-end transmission delays. According to [Agui03], VoIP can be characterised through a traditional ON-OFF behaviour, in which sequences of speech-bursts are intercalated with silent bursts. Thus, a VoIP transmission can be modelled as a Markov model with two states of “silence” and “talk”: when in “silence”, no packets are generated, and when in “talk”, packets are generated at a constant rate. Particularly IP packets carrying the speech information are transmitted. Both activity and silent periods are generated by an exponential distributed random variable with mean values t_{ON} and t_{OFF}, respectively.

The payload size of the IP packets carrying speech bursts depends on the considered speech codec and the packet rate. Typical VoIP codecs are G711, G732.1 and G729.A, all of these with their specific frame duration and frame sizes, Table A- 2.

Table A- 2 – Typical VoIP codecs (extracted from [Nune02]).

Codec	Frame Duration [ms]	Frame Size [bytes]	Bitrate [kbps]
G711	10	80	64.0
G723.1	30	24	6.4
G729.A	20	20	8.0

As VoIP uses UDP (User Data Protocol) and RTP (Real Time Protocol) at the transport layer, the size of a full IPv6 (Internet Protocol version 6) header together with a RTP/UDP header is 60 bytes, and 40 bytes if IPv4 (Internet Protocol version 4) is used instead. As the size of a typical voice packet is 20 bytes if G729.A is used, the RTP/UDP/IP overhead figures illustrate the typical problem of the header overhead in VoIP: in this case, instead of an 8 kbps bitrate, a final bitrate of 32 kbps case IPv6 was in use would be generated (24 kbps if IPv4 is used instead). When operating in a bandwidth limited system such as WiMAX, it is important to use the radio band as effectively as possible, and header overhead up to 60 bytes can seriously degrade the spectral efficiency of a VoIP service over such link. Without header compression, two-thirds of the transmission would be just headers. To handle this purpose, protocols such as “RObust Header Compression” (ROHC) have been developed to tackle this problem [IETF01]. According to [Agui03], one can assume that header bytes can be compressed to 8 bytes. Additionally, and following European Telecommunications Standards Institute (ETSI) recommendations [ETSI98], speech calls should be generated according to a Poisson process, with mean call duration of 120 s. The resulting VoIP modelling is summarised in Table A- 3.

Table A- 3 – Modelling of VoIP Traffic.

Activity Factor [%]	50
Mean Active Phase, t_{ON} [s]	3
Mean Silent Phase, t_{OFF} [s]	3
Payload of IP Packets [bytes]	20
IP Overhead [bytes]	8
Transmission Time Interval [ms]	20
Mean Call Duration [s]	120

A.2 Video Model

In [ChRe98], a source model for variable bitrate video traffic model based on a finite-state Markov chain is presented, being demonstrated that it accurately models a one- and two-layers video. This model assumes two types of video frames generation, the *I* and *P* frames. *I* is driven by scene changes, depends on the video source, and as such can be considered independent of the encoder dynamics. However, in the time period characterisation by a sequence of *P* frames, one may expect that there are no significant changes in the information in successive frames. Therefore, the bit rates characterisation of successive *P* frames can be expected to be correlated, or more generally clustered around an average value. That fact allows a technique that uses correlation between successive frame bit rates to identify the *I* and *P* frames in the data. The frame rate ranges from 24 to 30 frames per second.

The *I* frame statistic is modelled by a Gaussian distribution function, which average and variance is adjusted as a function of measured data. *P* frames may be quite different, because they depend of video changes. In order to model these frames, a mechanism of $k1$ states was created, each state with its own average and variance.

The corresponding $k1$ state group to frame *P*, together with the only state that characterises frame *I*, typify the $K1$+1 Markov states that are represented by a probabilistic transitions matrix P. In this matrix, 90 % of the total of probabilistic transitions from one state to another are concentrated in {P_{ii-1}, P_{ii}, P_{ii+1}} ∀ *I* =1,…, $k1$. Considering *i* = *n* any state in the Markov chain and *j* = *n-1* state. The transmission speed of a *i* frame is given by [RaMe01], [ChRe98]:

$$\begin{cases} R_i(n) = m_i(1-\alpha_i) + \alpha_i R_j(n-1) + g_i(n) & i, j = 1,...,k(P-frames) \\ R_{k+1}(n) = m_k + g_{k+1}(n) & (I-frames) \end{cases} \tag{A.3}$$

where:

- m_i is the estimated average value of transmission speed (*i* state)
- α_i is a coefficient value that models the autoregressive process, which value is adjusted empirically (based on measurements over *i* state in P frame)
- g_i *(n)* is a Gaussian random variable, different for each state (average 0 and variance given by measurement data in *i* state)

A.3 Non-Conversational Applications

Non-real time applications typically present an asymmetrical nature, as they refer mostly to specific requests for information done by end users to remote machines. The most known applications are Web browsing, FTP and E-mail.

Several models are studied and proposed to characterise Web browsing but as the present work does not intend to focus specifically on traffic models, modelling of these applications is based on [ETSI98]. Figure A-1 illustrates a typical Internet surfing session, which consists of a sequence of packet calls. During a packet call several packets may be generated, which means that the packet call is composed by a bursty sequence of packets, Figure A-1.

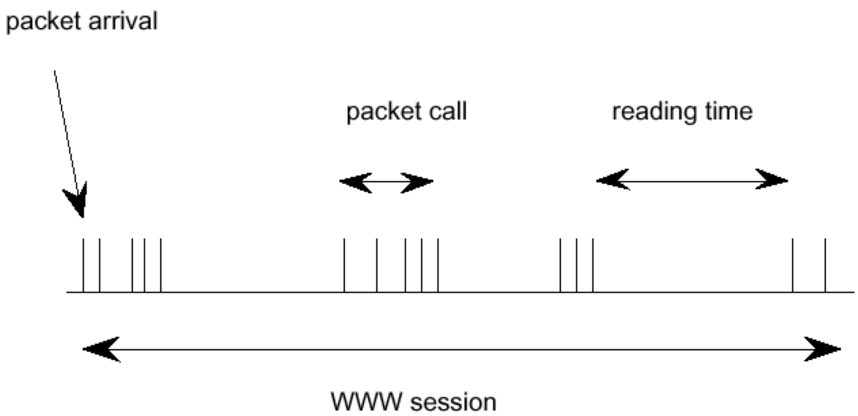


Figure A- 1 - Typical WWW session (adapted from [ETSI98]).

A packet service session is modelled as a Poisson process. It typically contains one or several packet calls, depending on the application. In a WWW browsing session for instance, a packet call corresponds to the downloading of a web page, and after the document has entirely arrived to the terminal, the user takes some time for analysing the information, which is often called the reading time. On the other hand, in a FTP session, it is likely that the session contains only one packet call.

In order to fully characterise a packet session, the following parameters must be modeled [ETSI98]:

- **The number of packet call requests per session**, N_{pc}: This is a geometrically distributed random variable with mean μN_{pc}
- **The reading time between two consecutive packet call requests in a session** , D_{pc}: This is a geometrically distributed random variable with a mean μD_{pc}. Note that the reading time starts when the last packet of the packet call is completely received by the user. The reading time ends when the user makes a request for the next packet call.
- **The number of packets within a packet call**, N_d: Although different statistical distributions can be used to generate the number of packets, it is assumed that N_d can be a geometrically distributed random variable, with mean μN_d .

- **The time interval between two consecutive packets inside a packed call**, D_d: This is a geometrically distributed random variable with a mean μD_d. Naturally, if there is only one packet in a packet call, this is not needed.
- **The Packet size**, Ps:The packet size distribution model is based on Pareto distribution that suits best for the traffic case under study; Pareto distribution with cut-off is used.

A.4 Streaming Model

Audio and Video Streaming services typically present an asymmetrical nature, as they refer mostly to the download of large files. Contrary to VoIP, Streaming is more flexible to end-to-end transmission delays and to its variations, although the implementation of buffering is required to accommodate those fluctuations. Although Video traffic can be transported either with a constant bitrate or with a variable bitrate, variable bitrate is the one expected to be the prime example, as it has several potential advantages over constant bitrate, particularly the possibility for implementing statistical multiplexing, allowing for improved channel allocation. IEEE has not standardised a specific codec for Video Streaming yet. On the contrary, 3GPP has specified the use of both MPEG-4 and H.263 codecs for Video Streaming services [3GPP02].

For simplification reasons, the present work uses of the non-conversational applications model presented in A.3 also for Streaming service, although with different parameters, namely the mean values μN_{pc}, μN_d and packet size (parameters of Pareto distribution).

Annex B

Propagation Models

The propagation models adopted and used in this thesis are described in this annex. These models are used for cellular and wireless local networks propagation estimation.

B.1 COST 231 Walfish-Ikegami Model

For a good estimation of the received average power, one may use the well know Walfisch-Ikegami propagation model adapted by COST 231 for microcell environment [DaCo99]. This model has the following input parameters:

- h_{Base}: BS height;
- $h_{Building}$: Building height;
- h_{Mobile}: MT height;
- w: Street width;
- f: Frequency;
- d: Distance between BS and MT;
- b: Building separation;
- Ψ: Street orientation angle.

The following default values are recommended:

- b : 20…50 m
- w : b /2
- $h_{Building}$: 3 m $\times$ [number of floors]+roof
- Ψ: 90 º

The path loss when in LoS is given by:

$$L_{p[\mathrm{dB}]} = 42.6 + 26 \cdot \log d_{[\mathrm{km}]} + 20 \cdot \log f_{[\mathrm{MHz}]} \tag{B.1}$$

The path loss in the case of NLoS of sight is given by (all path loss values are expressed in dB):

$$L_p = \begin{cases} L_0 + L_{rts} + L_{msd} & , L_{rts} + L_{msd} > 0 \\ L_0 & , L_{rts} + L_{msd} \le 0 \end{cases} \tag{B.2}$$

where:

$$L_{0[\mathrm{dB}]} = 32.4 + 20 \cdot \log d_{[\mathrm{km}]} + 20 \cdot \log f_{[\mathrm{MHz}]} \tag{B.3}$$

$$L_{rts[\mathrm{dB}]} = -16.9 - 10 \cdot \log w_{[\mathrm{m}]} + 10 \cdot \log f_{[\mathrm{MHz}]} + 20 \cdot \log \Delta h_{Mobile} + L_{ori[\mathrm{dB}]} \tag{B.4}$$

$$\Delta h_{Mobile} = h_{Mobile[\mathrm{m}]} - h_{Building\,[\mathrm{m}]} \tag{B.5}$$

$$L_{ori[\mathrm{dB}]} = \begin{cases} -10+0.34\cdot\Psi_{[^\circ]} & , \ 0^\circ \le \Psi < 35^\circ \\ 2.5+0.075\cdot\left(\Psi_{[^\circ]}-35\right) & , \ 35^\circ \le \Psi < 55^\circ \\ 4.0+0.114\cdot\left(\Psi_{[^\circ]}-55\right) & , \ 55^\circ \le \Psi \le 90^\circ \end{cases} \tag{B.6}$$

$$L_{msd[\mathrm{dB}]} = L_{bsh[\mathrm{dB}]} + K_{a[\mathrm{dB}]} + K_d \cdot \log d_{[\mathrm{km}]} + K_f \cdot \log f_{[\mathrm{MHz}]} - 9\cdot\log b_{[\mathrm{m}]} \tag{B.7}$$

where:

$$L_{bsh[\mathrm{dB}]} = \begin{cases} -18\cdot\log\left(1+\Delta h_{Base}\right) & , \ h_{Base} > h_{Building} \\ 0 & , \ h_{Base} \le h_{Building} \end{cases} \tag{B.8}$$

$$\Delta h_{Base} = h_{Base[\mathrm{m}]} - h_{Building\,[\mathrm{m}]} \tag{B.9}$$

$$K_{a[\mathrm{dB}]} = \begin{cases} 54 & , \ h_{Base} > h_{Building} \\ 54-0.8\cdot\Delta h_{Base} & , \ d \ge 0.5 \text{ km e } h_{Base} \le h_{Building} \\ 54-0.8\cdot\Delta h_{Base}\cdot d/0.5 & , \ d < 0.5 \text{ km e } h_{Base} \le h_{Building} \end{cases} \tag{B.10}$$

$$K_d = \begin{cases} 18 & , \ h_{Base} > h_{Building} \\ 18-15\cdot\Delta h_{Base}/h_{Building} & , \ h_{Base} \le h_{Building} \end{cases} \tag{B.11}$$

$$K_f = \begin{cases} -4+0.7\cdot\left(f/925-1\right) & \text{for medium size cities and suburban} \\ & \text{centres with moderate tree density} \\ & \text{densidade de árvores} \\ -4+1.5\cdot\left(f/925-1\right) & \text{for metropolitan centres} \end{cases} \tag{B.12}$$

L_0 is the free-space attenuation, L_{rts} is "roof-to-street diffraction and scatter loss", L_{ori} is the attenuation caused by main street orientation with respect to the direct radio path and L_{msd} is the "multi-screen diffraction loss". The output parameter of the model is L_p in dB. Some parameters have a validity range, Table B-1.

Table B-1 - Valid parameters range.

Frequency, f	800…2000 MHz
Distance NLoS, d	0.02…5 km
Distance LoS, d	0.02…0.2 km
BS antenna height, h_{Base}	4…50 m
MT antenna height, h_{Mobile}	1…3 m

B.2 Double Breakpoint Model

The free space propagation model is not applicable to many practical situations. However, due to its simplicity, it is common to use it for estimations, in this case the distance exponent γ=2 is changed to better match practical situations. A breakpoint model can be applied in relation to obstructed conditions, at which a distance exponent γ=2 is used for the first metres and for distance increase above a breakpoint to a larger γ.

For outdoor environments with an antenna height of a few metres and distance of a few hundred metres, a "double breakpoint" model gives a good characterisation of the path loss for urban environments in the presence of obstruction. The double breakpoint model [Pras01] has a first breakpoint at 1 m (reference distance for isotropic loss) and a second breakpoint at 100 m.

Notice that frequencies covered by this model match with IEEE 802.11 family, thus in this thesis this model is used for Wi-Fi propagation estimation. For 2.4 GHz the path loss is as follows:

$$L_p\left(f = 2.4GHz\right)_{[\mathrm{dB}]} = 40 + \gamma 20\log d_{[\mathrm{m}]} \qquad d \leq d_{break} \tag{B.13}$$

$$L_p\left(f = 2.4GHz\right)_{[\mathrm{dB}]} = 40 + \gamma_1 20\log d_{break[\mathrm{m}]} + 10\gamma_2 \log\left(\frac{d_{[\mathrm{m}]}}{d_{break[\mathrm{m}]}}\right) \qquad d > d_{break} \tag{B.14}$$

For the 5GHz band the path loss model is the following:

$$L_p\left(f = 5GHz\right)_{[\mathrm{dB}]} = 46.38 + \gamma 20\log d_{[\mathrm{m}]} \qquad d \leq d_{break} \tag{B.15}$$

$$L_p\left(f = 5GHz\right)_{[\mathrm{dB}]} = 46.38 + \gamma_1 20\log d_{break[\mathrm{m}]} + 10\gamma_2 \log\left(\frac{d_{[\mathrm{m}]}}{d_{break[\mathrm{m}]}}\right) \qquad d > d_{break} \tag{B.16}$$

It is considered that antenna gains are 0 dBi.

Annex C

Link Budget for WiMAX

In this annex, detailed link budget calculations for WiMAX system are presented and described.

The total propagation attenuation is calculated individually for both UL and DL using the following expression [Corr03]:

$$L_{p\,[\mathrm{dB}]} = EIRP_{[\mathrm{dBm}]} + G_{ar\,[\mathrm{dBi}]} - L_{r\,[\mathrm{dB}]} + G_{r\,[\mathrm{dB}]} - P_{r\,[\mathrm{dBm}]} \quad \text{(C.1)}$$

where:

- $EIRP$ is the equivalent isotropic radiated power, given by (C.2):

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

- P_t is the power at the exit of the transmitting radio unit. Values differ between BS and MS, being given by Table C- 1.

Table C- 1 – Power classes (extracted from [IEEE04b])

Class Identifier	**Transmit Power [dBm]**
1	17≤ $P_{t,\max}$ < 20
2	20≤ $P_{t,\max}$ < 23
3	23≤ $P_{t,\max}$ < 30
4	30≤ $P_{t,\max}$

- G_t represents the additional gains, e.g. due to the use of multiple transmitting antennas or use of power boosters.
- L_t is the attenuation between the transmitting radio unit and the transmitting antenna due to cable loss, connector loss, etc.. In WiMAX, typical commercial solutions allow the radio units to be installed quite close to the antennas, so minimum losses are caused by such causes. A value of 0.7 dBi is assumed.
- G_{at} is the gain of the transmitter antenna. Typical values are:
  1. For BS antennas, 17 dBi for 65º horizontal aperture antennas, plus additional 3 dB derived from considering an array of two transmit antennas in the BS, [WiMA06a].
  2. For MS antennas, 0 dBi.
- G_{ar} is the gain of the receiving antenna.
- L_r is the attenuation between the receiving antenna and the receiving radio unit, due to cable loss, connector loss, etc. For the DL, there are no cables at the receiving side, thus, this value is set to 0 dB.

- G_r represents the additional gains, e.g., due to the use of receive diversity. Due to received diversity, 3 dB gain is usually assumed.
- P_r is the power received at the receiving antenna.

In OFDM, the receiver's sensitivity per subcarrier can be calculated through:

$$P_{rx\,\mathrm{min}/\,subcarrier[\mathrm{dBm}]} = SNR_{req[\mathrm{dB}]} + N_{tot[\mathrm{dBm}]} \tag{C.3}$$

where SNR_{req} is the required Signal-to-Noise Ratio for the signal, given by Table 2-3, and N_{tot} is the total noise power, given by (C.4):

$$N_{tot[\mathrm{dBm}]} = F_{N[\mathrm{dB}]} - 174 + 10 \cdot \log\left(\Delta f_{[\mathrm{Hz}]}\right) \tag{C.4}$$

where:

- F_N us the receiver's noise figure:
  - For BS equipment, typical noise figure is around 4.0 dB [WiMA06a].
  - For ST equipment, typical noise figure is around 7.0 dB [WiMA06a].
- Δf is the channel bandwidth, in this case related to the subcarrier. According to Table 2-1, it equals 10.94 kHz.

The composite receiver sensitivity, i.e., considering all sub-carriers is given by:

$$P_{rx\,\mathrm{min}[\mathrm{dBm}]} = P_{rx\,\mathrm{min}/\,subcarrier[\mathrm{dBm}]} + 10 \cdot \log(N_{usedsubcarriers}) \tag{C.5}$$

where $N_{usedsub-carriers}$ stands for the number of used sub-carriers (Table 2-2).

Annex D

HNP variation with the density of users

In this annex, the HNP is evaluated as a function of the density of mobile users.

Table D-1 - HNP variation with the density of users.

Nº of users / HNP		8000		10000 (REF)		12000		14000	
		Mean val.	St.dev	Mean val.	St.dev	Mean val.	St. dev.	Mean val.	St.dev
P_b [%]		0.40	0.13	1.00	0.87	8.28	2.44	13.35	8.33
τ [ms]		21.72	4.93	23.53	11.10	27.57	2.41	40.68	14.13
N_{VHOa}		196.60	24.23	237.53	76.83	288.20	70.74	331.20	159.86
N_{HHOa}		86.40	22.28	74.00	39.71	103.00	30.49	120.40	69.16
P_{VHOf} [%]		1.60	1.67	5.40	4.45	28.00	6.96	34.60	14.74
P_{HHOf} [%]		4.00	6.78	4.33	5.74	2.60	0.89	28.00	18.92
D_r [%]		23.40	1.95	24.60	2.29	28.20	3.49	28.60	6.31
BR_{srv} [Mbps]	Voice call	0.03	0.00	0.03	0.00	0.03	0.00	0.03	0.00
	Video call	0.51	0.01	0.51	0.01	0.51	0.01	0.51	0.01
	Web browsing	13.67	7.58	15.13	12.05	9.90	5.27	16.10	13.19
	Streaming	4.82	3.23	4.75	4.70	8.17	7.98	12.36	10.27
	E-mail	0.49	0.28	3.61	6.26	2.29	3.19	0.62	0.27
	FTP	1.75	2.27	2.87	3.75	0.87	0.39	3.47	4.20
BR_{RAN} [Mbps]	UMTS	0.16	0.04	0.15	0.05	0.17	0.03	0.15	0.04
	Wi-Fi	17.70	1.20	20.51	3.80	22.06	2.96	22.42	1.46
	WiMAX	6.95	3.95	7.99	3.52	7.31	3.91	11.04	5.93
NU_{srv}	Voice call	37.92	1.54	47.44	2.52	51.41	2.89	56.91	6.25
	Video call	0.87	0.33	0.98	0.38	1.59	0.73	2.50	0.83
	Web browsing	5.31	1.10	4.02	1.06	5.94	1.24	5.08	1.88
	Streaming	9.06	1.78	7.22	2.21	9.95	2.91	9.38	3.52
	E-mail	4.94	0.63	4.28	0.82	3.91	1.26	3.17	0.83
	FTP	4.30	0.91	4.25	1.56	4.73	1.13	5.05	0.85

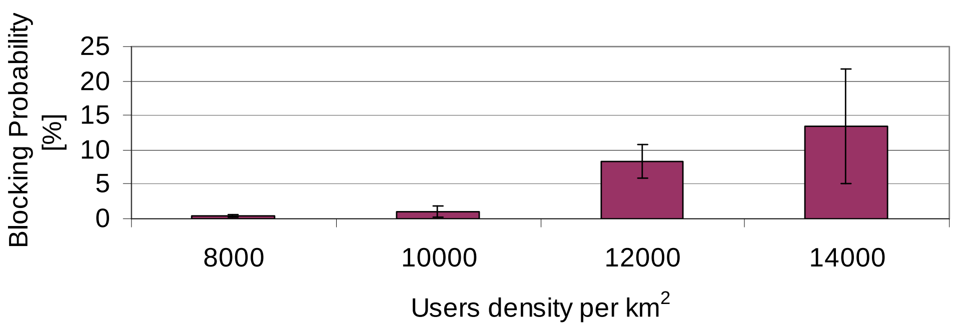


Figure D-1 – Blocking Probability variation with density of users.

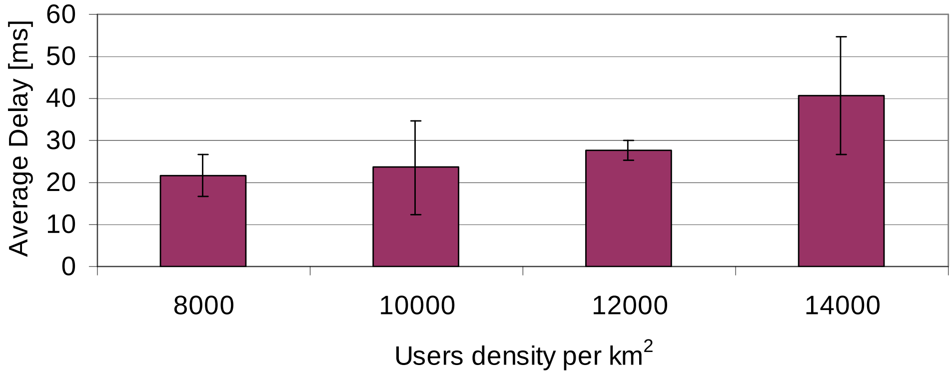


Figure D-2 – Average Delay variation with density of users.

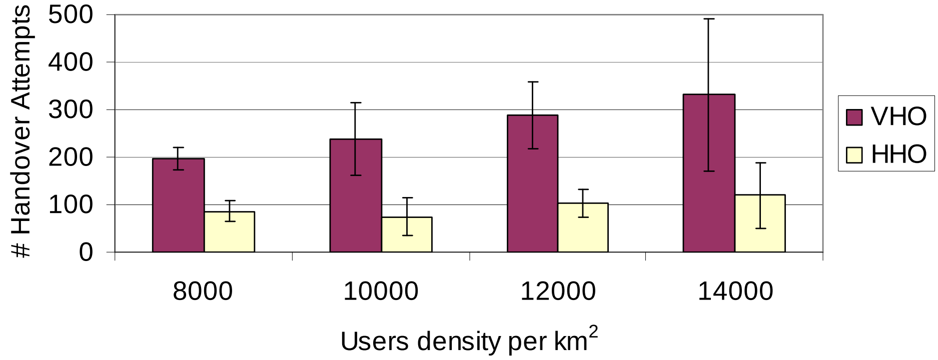


Figure D-3 – Total number of handovers variation with density of users.

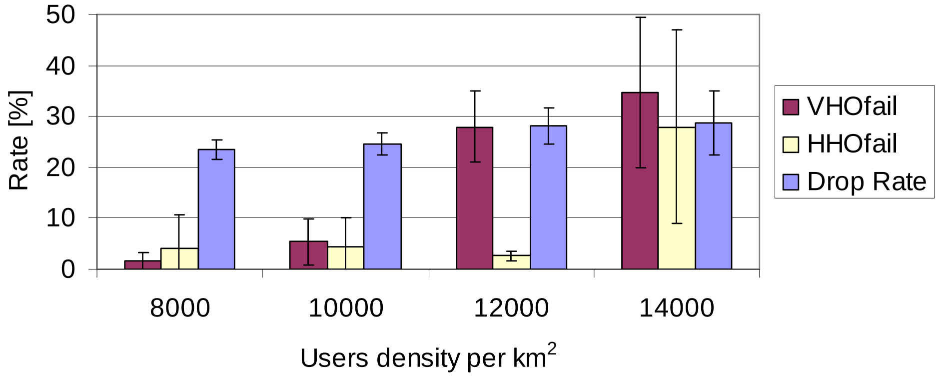


Figure D-4 – Vertical, horizontal handover failure rate and drop rate variation with density of users.

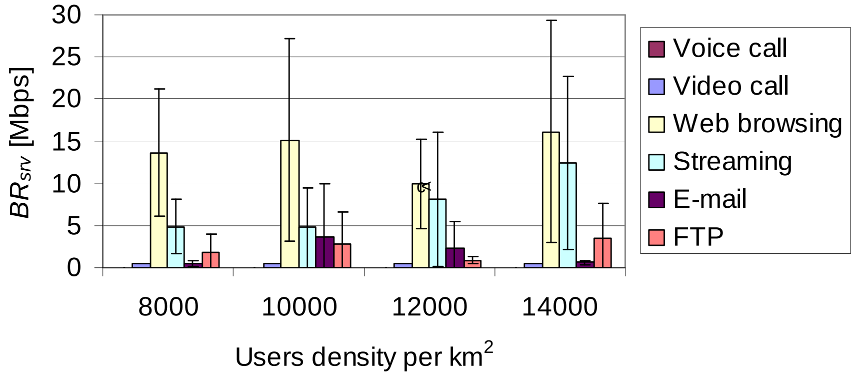
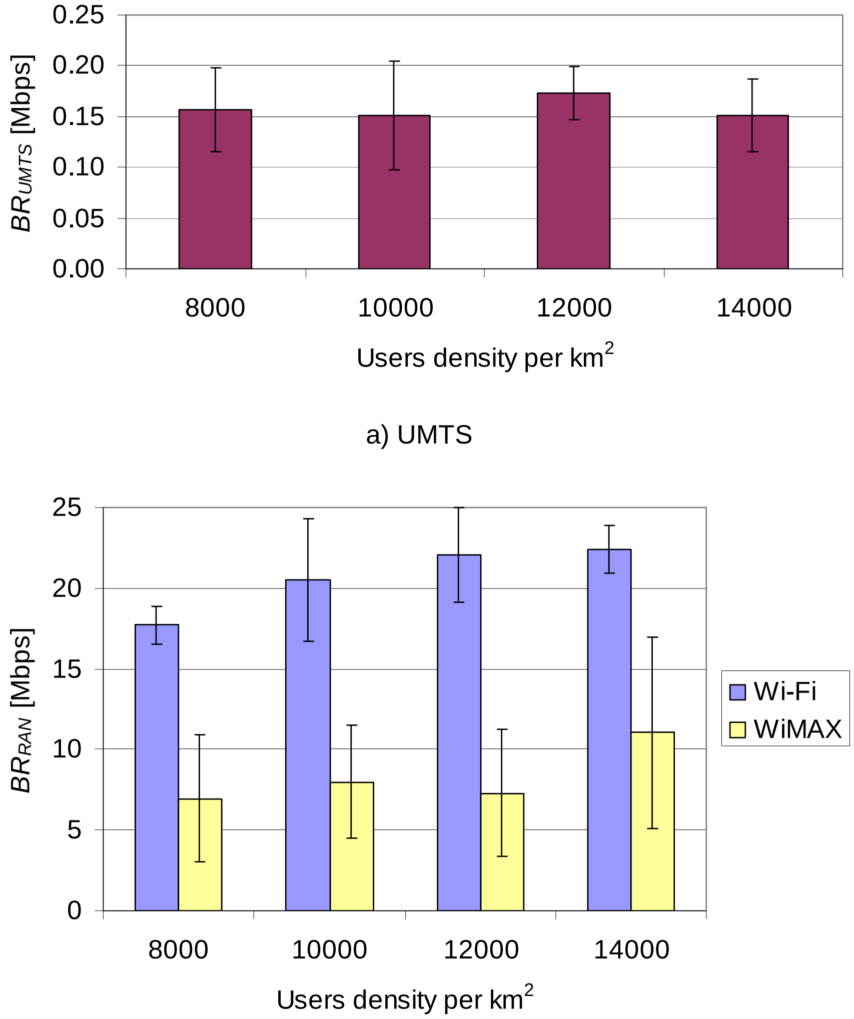


Figure D-5 – Average bitrate per service variation with density of users.



a) UMTS

b) Wi-Fi and WiMAX

Figure D-6 – Average bitrate per system variation with density of users.

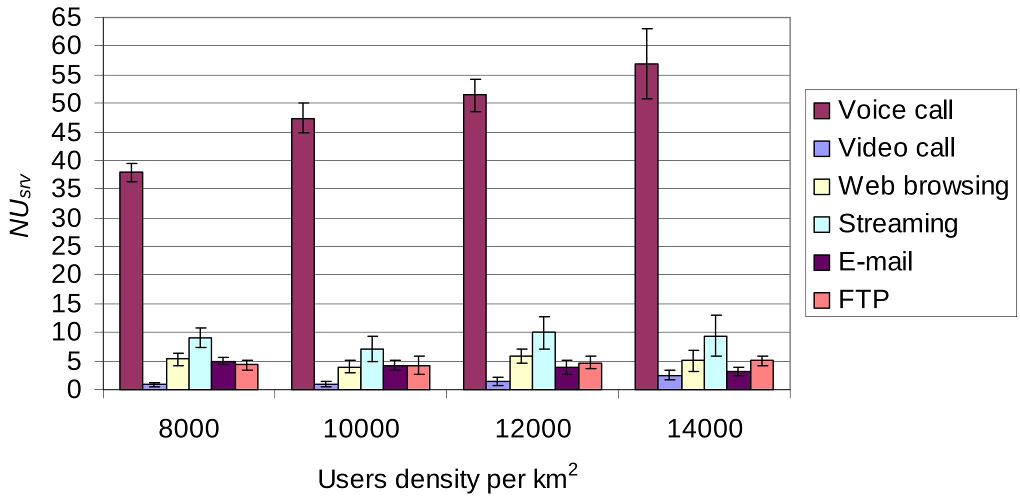


Figure D-7 – Average number of users per service per second variation with density of users.

Annex E

HNP variation with the geographical distribution of users

In this annex, the HNP is evaluated as a function of the geographical distribution of users.

Table E-1 - HNP variation with the distribution of users.

HNP \ Users Distribution		Normal σ_x, σ_y = 150 m μ_x=500 m, μ_y=366 m		Normal σ_x, σ_y = 300 m μ_x=500 m, μ_y=366 m		Uniform [0;1000m] (REF)	
		Mean val.	St.dev.	Mean	St.dev.	Mean	St. dev.
P_b [%]		22.18	1.85	3.07	1.89	1.00	0.87
τ [ms]		80.23	19.54	20.53	6.28	23.53	11.10
N_{VHOa}		174.20	21.37	217.60	37.84	237.53	76.83
N_{HHOa}		264.40	88.88	109.80	69.85	74.00	39.71
P_{VHOf} [%]		18.40	4.56	13.60	8.35	5.40	4.45
P_{HHOf} [%]		3.60	1.52	1.80	1.64	4.33	5.74
D_r [%]		4.20	0.45	22.40	6.66	24.60	2.29
BR_{srv} [Mbps]	Voice call	0.03	0.00	0.03	0.00	0.03	0.00
	Video call	0.50	0.01	0.51	0.01	0.51	0.01
	Web browsing	17.54	16.95	6.52	2.97	15.13	12.05
	Streaming	9.24	7.87	12.02	9.99	4.75	4.70
	E-mail	3.21	4.41	0.58	0.31	3.61	6.26
	FTP	5.04	1.48	4.51	5.15	2.87	3.75
BR_{RAN} [Mbps]	UMTS	0.16	0.03	0.20	0.05	0.15	0.05
	Wi-Fi	21.81	1.68	19.00	0.93	20.51	3.80
	WiMAX	9.35	3.78	8.58	4.45	7.99	3.52
NU_{srv}	Voice call	45.47	0.65	43.72	1.62	47.44	2.52
	Video call	1.75	0.21	1.66	0.40	0.98	0.38
	Web browsing	6.14	1.58	5.74	1.59	4.02	1.06
	Streaming	9.55	1.80	11.04	3.43	7.22	2.21
	E-mail	1.20	0.66	4.05	1.11	4.28	0.82
	FTP	6.05	1.23	5.96	2.03	4.25	1.56

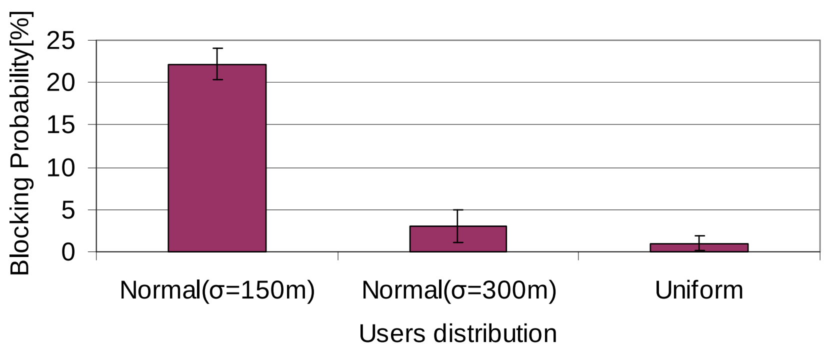


Figure E-1 - Blocking Probability variation with distribution of users.

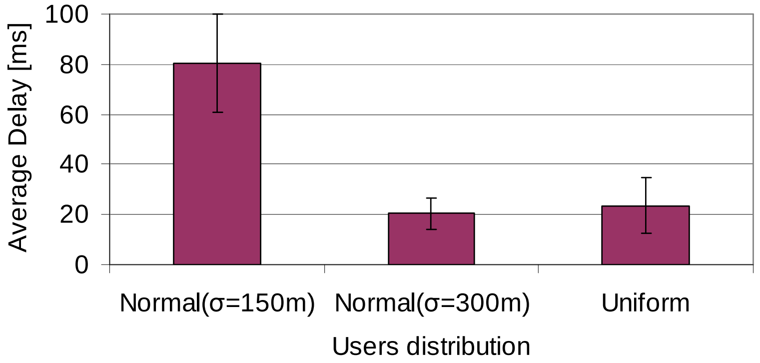


Figure E-2 - Average Delay variation with distribution of users.

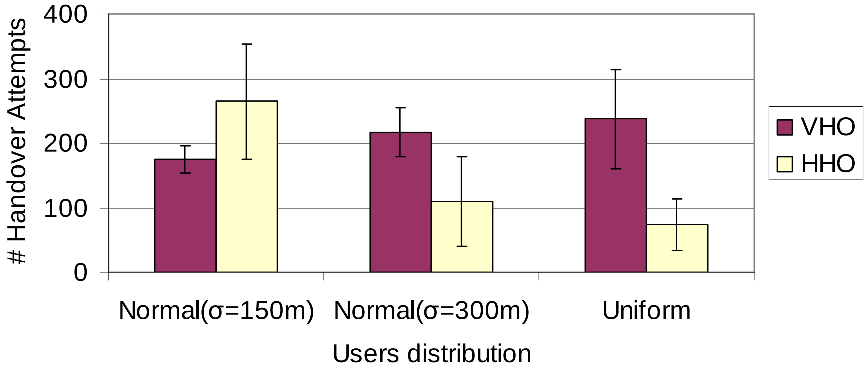


Figure E-3 - Total number of handovers variation with distribution of users.

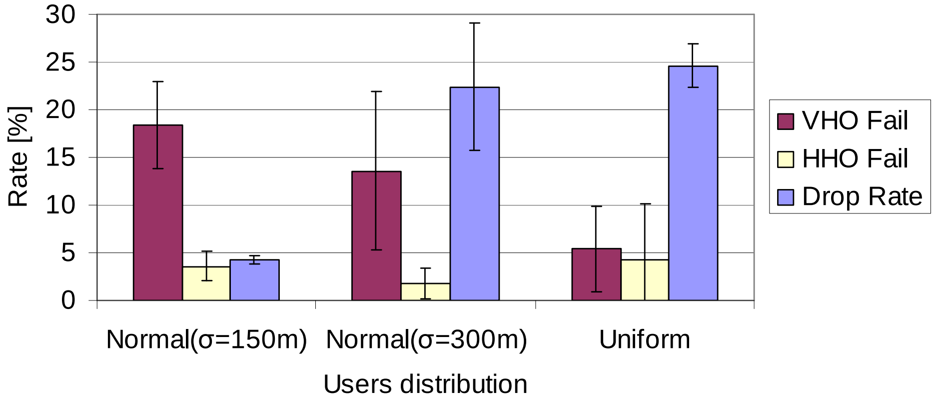


Figure E-4 - Vertical, horizontal handover failure rate and drop rate variation with distribution of users.

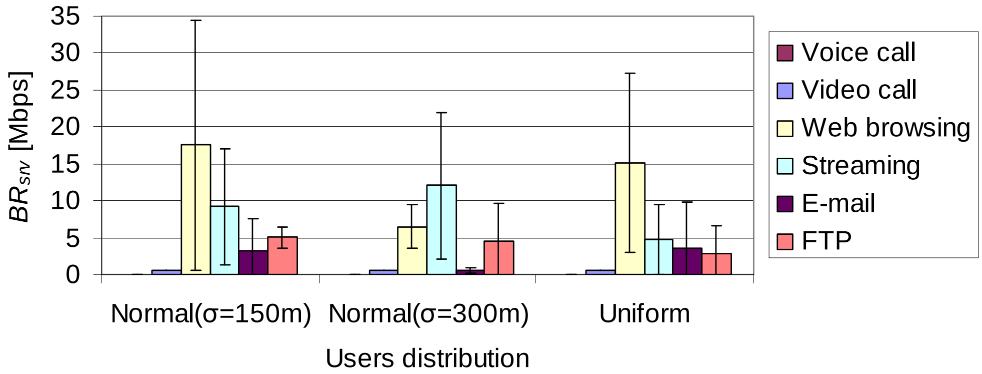
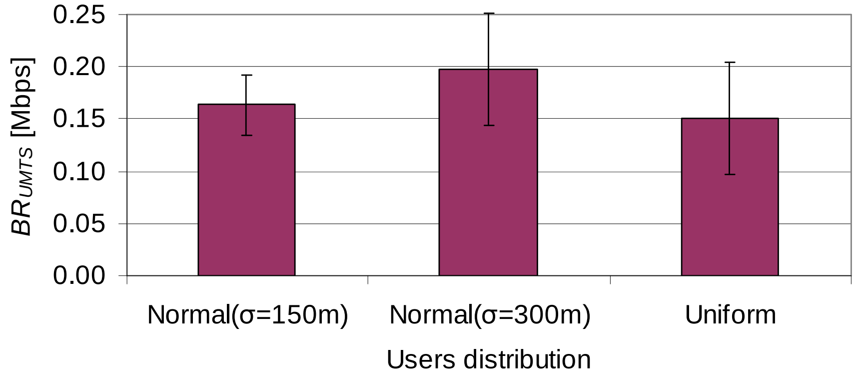
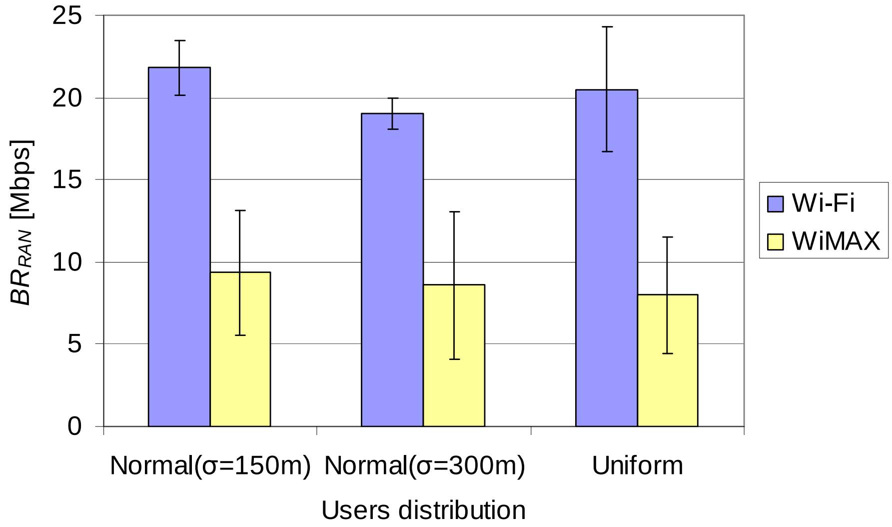


Figure E-5 - Average bitrate per service variation with distribution of users.



a) UMTS



b) Wi-Fi and WiMAX

Figure E-6 - Average bitrate per system variation with distribution of users.

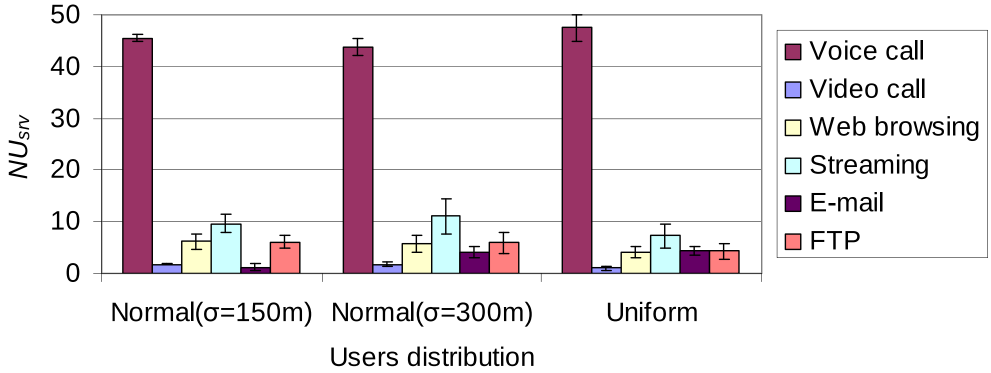


Figure E-7 - Average number of users per service per second variation with distribution of users.

Annex F

HNP variation with the average duration of the Voice calls

In this annex, the HNP is evaluated as a function of the average duration of the voice calls.

Table F-1 - HNP variation with the average duration of the voice calls.

HNP \ Avg. dur. voice calls		90 s (REF)		105 s		120 s	
		Mean val.	St.dev.	Mean	St.dev.	Mean	St. dev.
P_b [%]		1.00	0.87	4.16	4.18	6.54	3.57
τ [ms]		23.53	11.10	31.81	12.07	50.46	15.50
N_{VHOa}		237.53	76.83	253.80	81.74	232.40	51.21
N_{HHOa}		74.00	39.71	78.20	39.72	90.40	56.26
P_{VHOf} [%]		5.40	4.45	14.80	13.42	19.60	14.52
P_{HHOf} [%]		4.33	5.74	8.40	14.99	0.60	0.55
D_r [%]		24.60	2.29	26.80	3.35	28.00	2.45
BR_{srv} [Mbps]	Voice call	0.03	0.00	0.03	0.00	0.03	0.00
	Video call	0.51	0.01	0.51	0.00	0.52	0.01
	Web browsing	15.13	12.05	17.44	11.53	23.92	28.88
	Streaming	4.75	4.70	6.96	5.81	8.80	5.38
	E-mail	3.61	6.26	2.06	2.88	1.04	1.39
	FTP	2.87	3.75	6.03	11.74	3.91	2.78
BR_{RAN} [Mbps]	UMTS	0.15	0.05	0.11	0.03	0.09	0.03
	Wi-Fi	20.51	3.80	20.29	2.00	19.05	1.95
	WiMAX	7.99	3.52	9.98	4.93	11.08	5.93
NU_{srv}	Voice call	47.44	2.52	52.52	4.94	60.29	6.40
	Video call	0.98	0.38	1.69	1.07	1.06	0.69
	Web browsing	4.02	1.06	4.92	1.74	3.24	1.64
	Streaming	7.22	2.21	9.50	3.37	5.18	3.08
	E-mail	4.28	0.82	4.06	1.15	2.71	0.72
	FTP	4.25	1.56	4.69	0.89	3.43	2.40

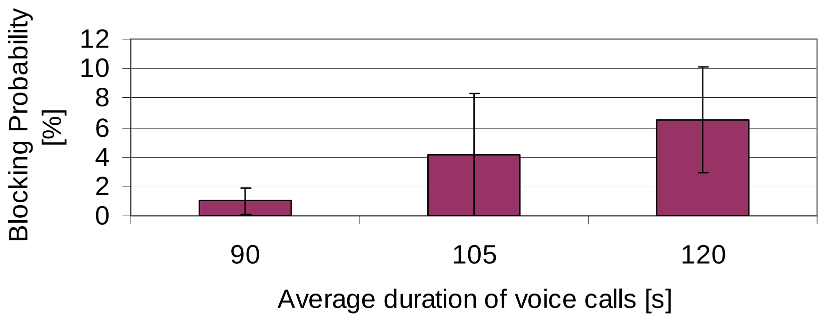


Figure F-1 - Blocking Probability variation with average duration of the voice calls.

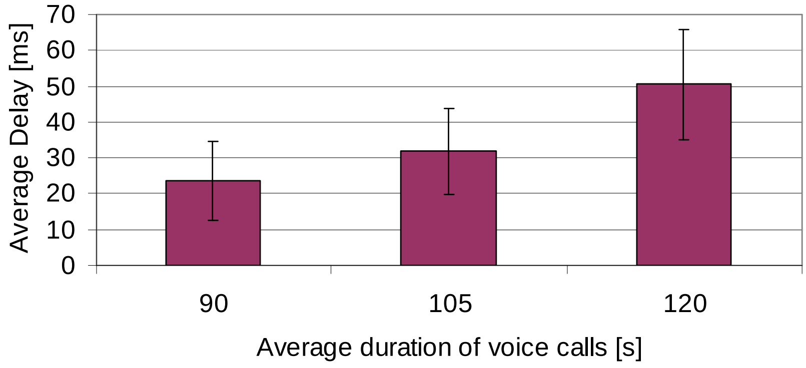


Figure F-2 - Average Delay variation with average duration of the voice calls.

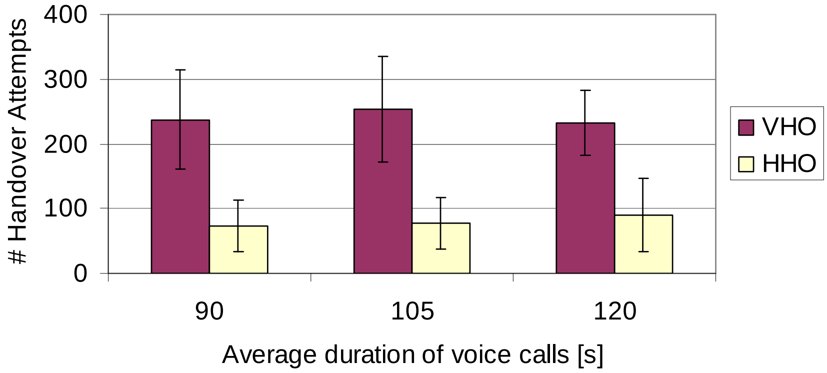


Figure F-3 - Total number of handovers variation with average duration of the voice calls.

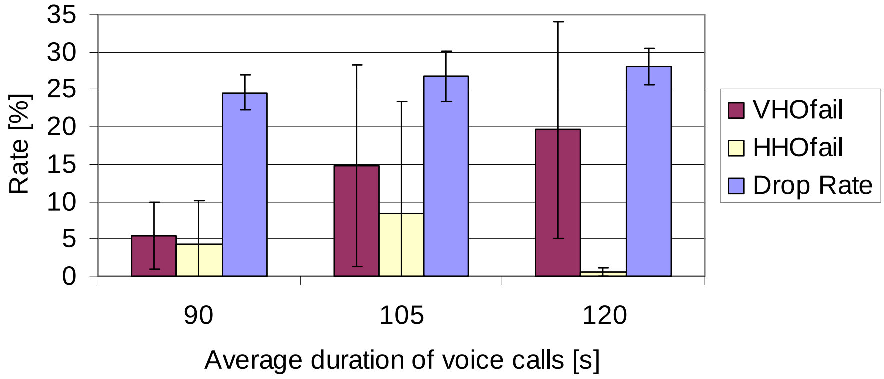


Figure F-4 – Vertical, horizontal HO failure rate and drop rate variation with average duration of the voice calls.

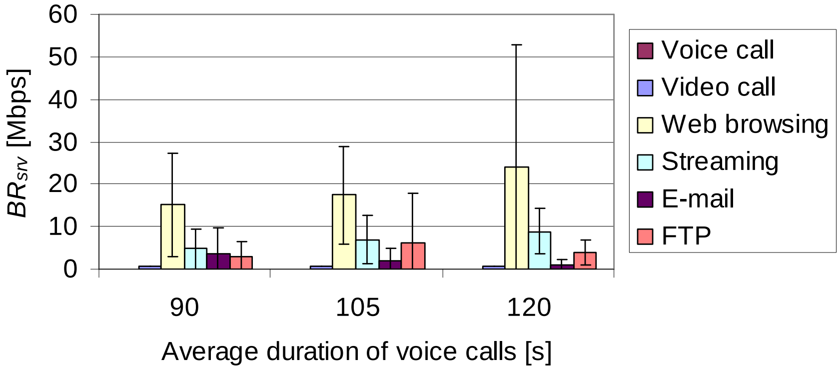
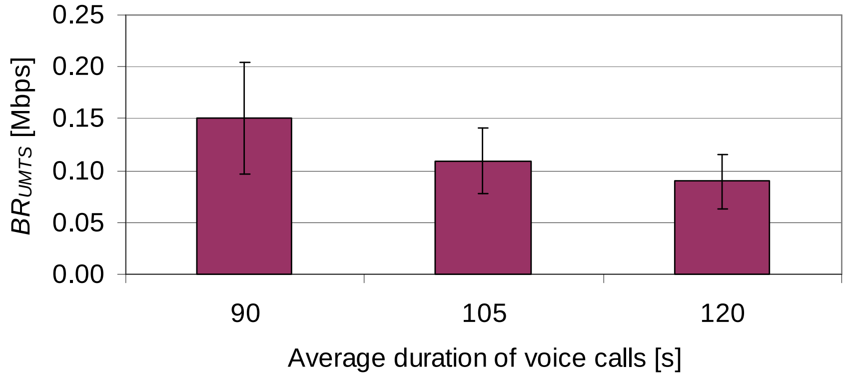
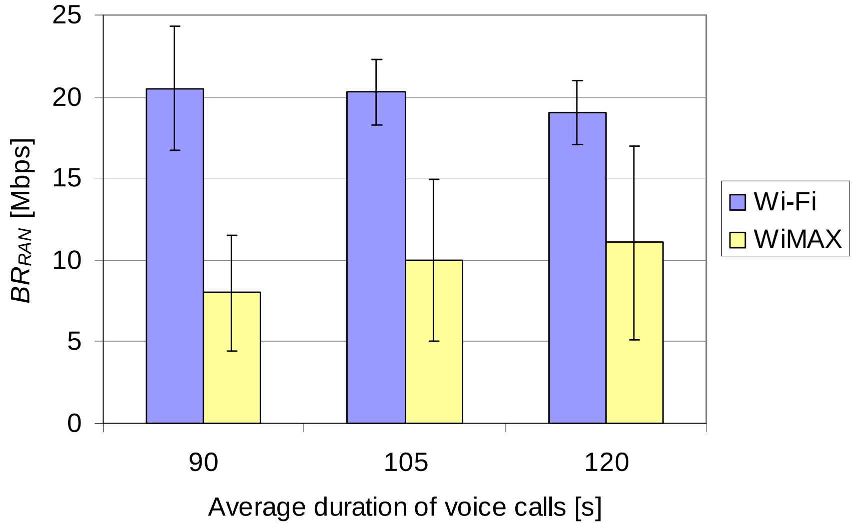


Figure F-5 – Average bitrate per service variation with average duration of the voice calls.



a) UMTS



b) Wi-Fi and WiMAX

Figure F-6 – Average bitrate per system variation with average duration of the voice calls.

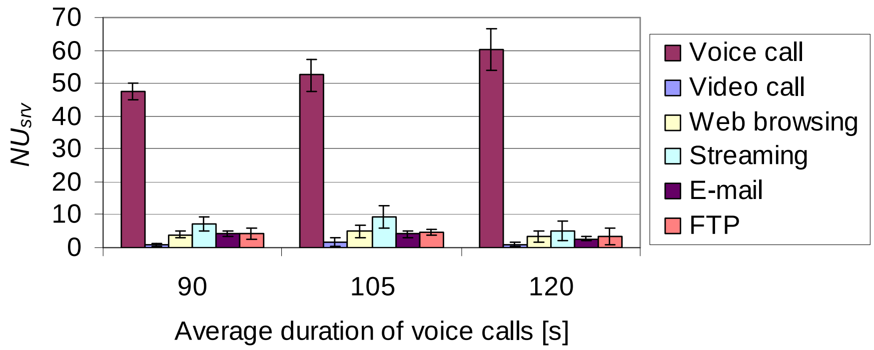


Figure F-7 – Average number of users per service per second variation with average duration of the voice calls.

Annex G

HNP variation with the packet volume of data services

In this annex, the HNP is evaluated as a function of the packet volume of data services.

Table G-1 - HNP variation with the packet volume of data services.

HNP \ Packet volume		K = 80 (REF)		K = 160		K = 240	
		Mean val.	St.dev.	Mean	St.dev.	Mean	St. dev.
P_b [%]		1.00	0.87	1.99	1.58	0.79	0.76
τ [ms]		23.53	11.10	39.59	18.00	47.84	6.17
N_{VHOa}		237.53	76.83	195.00	30.88	239.00	91.94
N_{HHOa}		74.00	39.71	108.00	70.95	66.75	13.91
P_{VHOf} [%]		5.40	4.45	12.40	3.51	3.75	3.86
P_{HHOf} [%]		4.33	5.74	9.80	3.56	3.25	1.71
D_r [%]		24.60	2.29	25.40	2.07	25.00	0.82
BR_{srv} [Mbps]	Voice call	0.03	0.00	0.03	0.00	0.03	0.00
	Video call	0.51	0.01	0.51	0.01	0.51	0.01
	Web browsing	15.13	12.05	15.27	9.61	5.16	1.78
	Streaming	4.75	4.70	15.79	11.07	23.89	14.02
	E-mail	3.61	6.26	1.13	0.35	5.60	5.93
	FTP	2.87	3.75	3.08	2.16	13.31	20.79
BR_{RAN} [Mbps]	UMTS	0.15	0.05	0.15	0.03	0.12	0.02
	Wi-Fi	20.51	3.80	20.76	1.77	19.64	2.29
	WiMAX	7.99	3.52	13.59	8.12	19.18	6.02
NU_{srv}	Voice call	47.44	2.52	46.33	2.30	48.16	3.22
	Video call	0.98	0.38	1.29	0.32	1.47	0.79
	Web browsing	4.02	1.06	5.31	0.88	3.02	1.56
	Streaming	7.22	2.21	9.09	1.86	6.76	3.36
	E-mail	4.28	0.82	3.33	0.58	3.42	0.20
	FTP	4.25	1.56	2.67	0.93	1.27	0.60

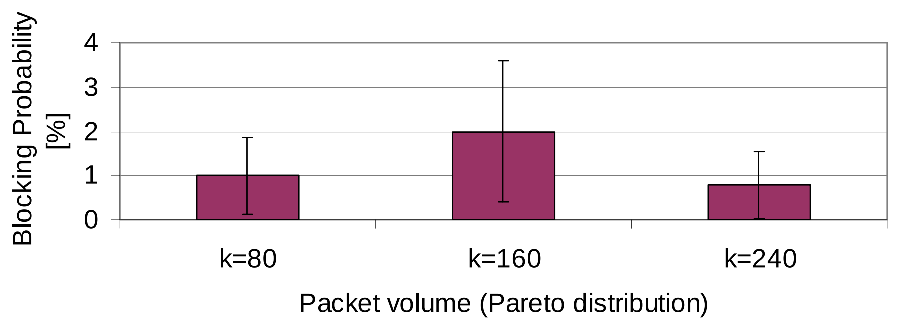


Figure G-1 - Blocking Probability variation with packet volume of data services.

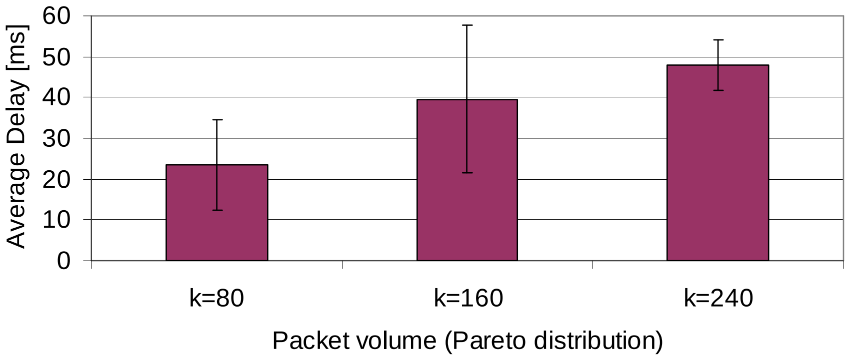


Figure G-2 - Average Delay variation with packet volume of data services.

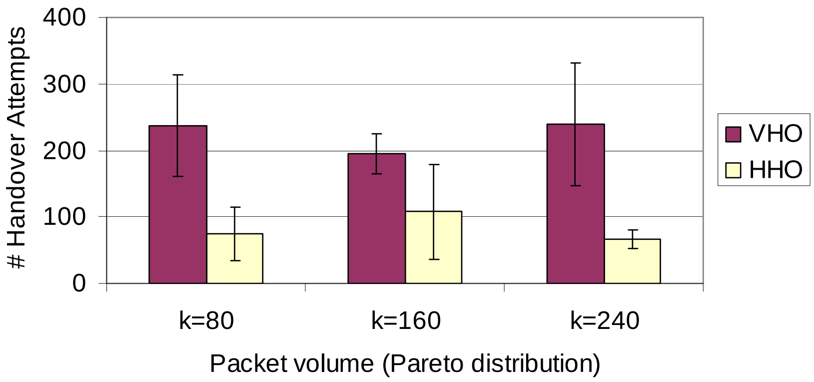


Figure G-3 - Total number of handovers variation with packet volume of data services.

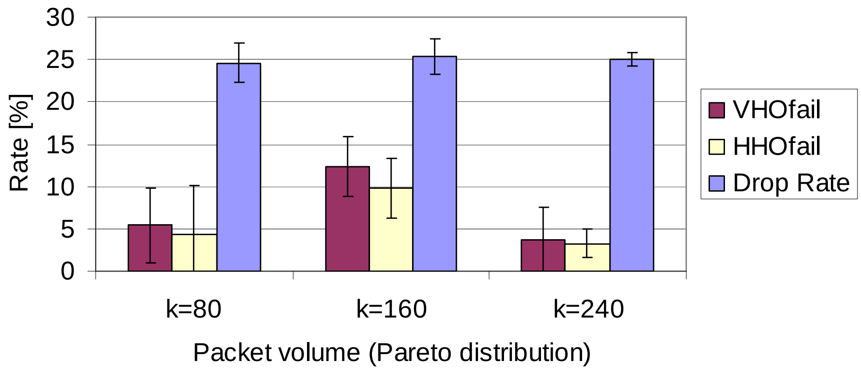


Figure G-4 – Vertical, horizontal HO failure rate and drop rate variation with packet volume of data services.

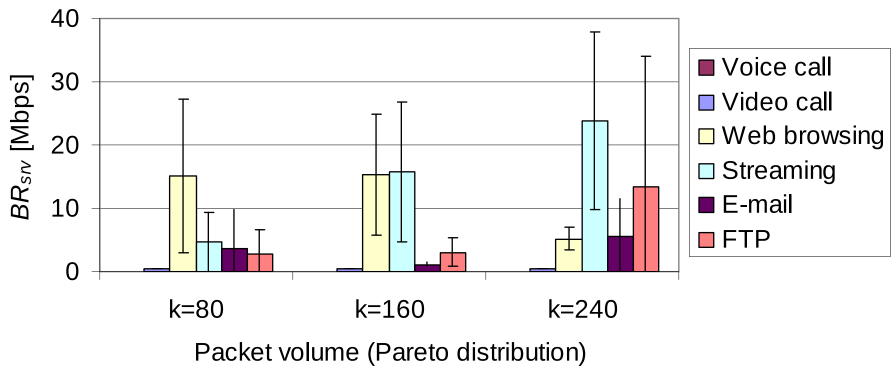
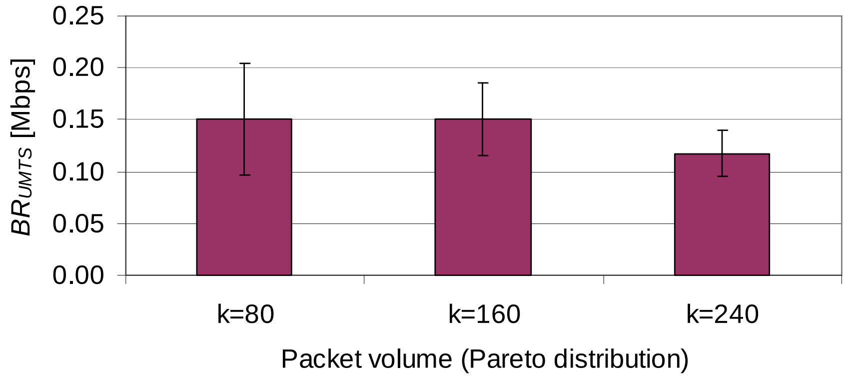
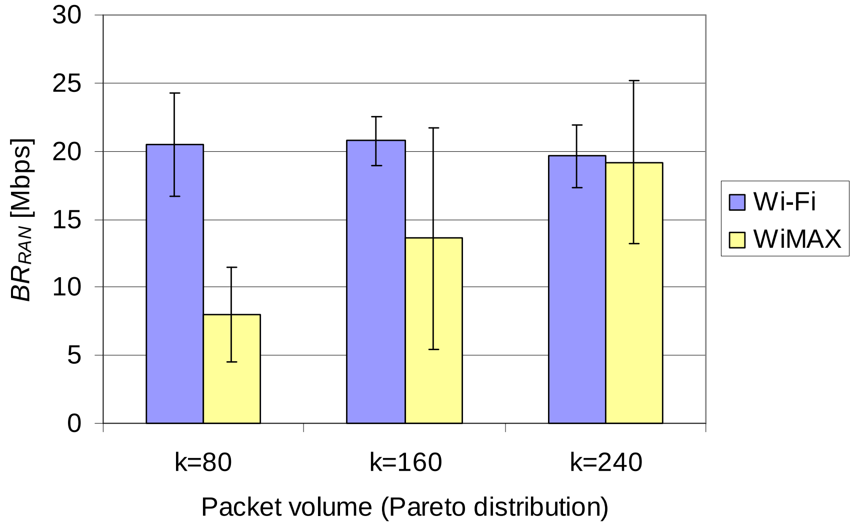


Figure G-5 – Average bitrate per service variation with packet volume of data services.



a) UMTS



b) Wi-Fi and WiMAX

Figure G-6 – Average bitrate per system variation with packet volume of data services.

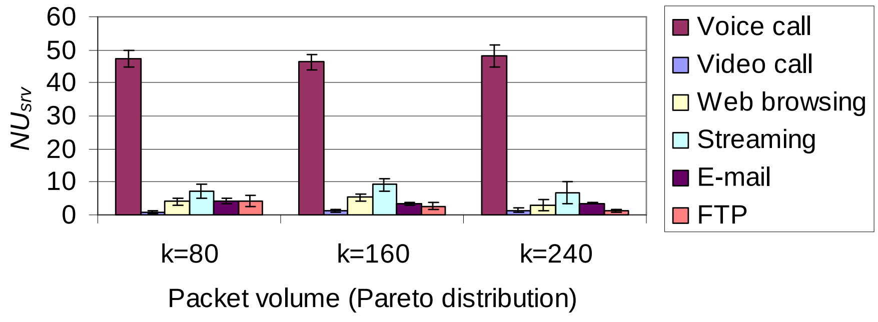


Figure G-7 – Average number of users per service per second variation with packet volume of data services.

Annex H

HNP variation with the service penetration

In this annex, the HNP is evaluated as a function of the service penetration.

Table H-1 - HNP variation with the service penetration.

HNP \ Serv. Pen. Profile		REF		www + 10%		www + 20%		www &E-mail + 10%	
		Mean val.	St.dev	Mean val.	St.dev	Mean val.	St. dev.	Mean val.	St.dev
P_b [%]		1.00	0.87	0.73	0.42	0.46	0.35	0.77	0.56
τ [ms]		23.53	11.10	22.21	9.17	22.78	4.60	17.27	3.54
N_{VHOa}		237.53	76.83	250.40	57.44	249.20	32.53	297.80	38.20
N_{HHOa}		74.00	39.71	128.20	23.48	125.00	27.60	187.40	83.49
P_{VHOf} [%]		5.40	4.45	7.60	4.98	1.80	2.95	3.00	0.71
P_{HHOf} [%]		4.33	5.74	17.20	12.32	7.00	4.12	3.20	1.48
D_r [%]		24.60	2.29	25.20	2.05	31.00	3.87	30.40	2.07
BR_{srv} [Mbps]	Voice call	0.03	0.00	0.03	0.00	0.03	0.00	0.03	0.00
	Video call	0.51	0.01	0.51	0.01	0.50	0.01	0.51	0.00
	Web browsing	15.13	12.05	17.15	15.74	10.24	4.95	7.59	5.54
	Streaming	4.75	4.70	3.51	2.29	11.73	8.79	5.56	4.83
	E-mail	3.61	6.26	2.65	5.06	0.29	0.04	0.95	1.20
	FTP	2.87	3.75	2.55	2.19	7.64	7.75	4.08	7.00
BR_{RAN} [Mbps]	UMTS	0.15	0.05	0.31	0.03	0.37	0.04	0.39	0.11
	Wi-Fi	20.51	3.80	22.99	1.35	19.98	2.84	19.56	3.53
	WiMAX	7.99	3.52	10.48	7.68	10.40	5.78	6.08	2.45
NU_{srv}	Voice call	47.44	2.52	38.37	2.65	28.65	1.91	29.99	0.50
	Video call	0.98	0.38	1.34	0.61	1.10	0.33	0.91	0.20
	Web browsing	4.02	1.06	10.08	2.17	11.49	3.18	10.92	1.20
	Streaming	7.22	2.21	9.99	1.46	8.36	2.67	12.89	2.03
	E-mail	4.28	0.82	5.90	0.75	5.29	1.42	10.31	2.13
	FTP	4.25	1.56	4.62	0.73	5.80	2.37	5.72	0.86

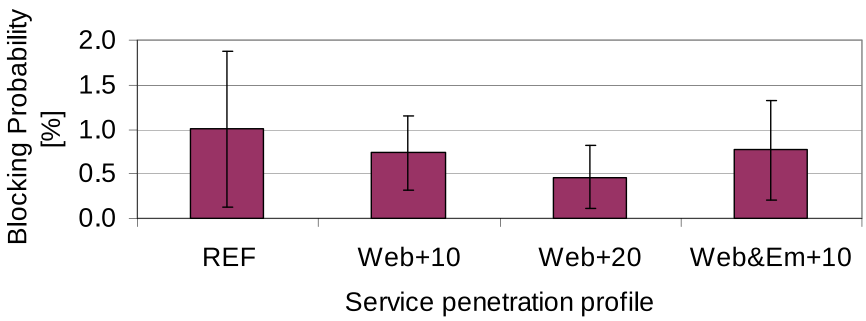


Figure H-1 - Blocking Probability variation with service penetration.

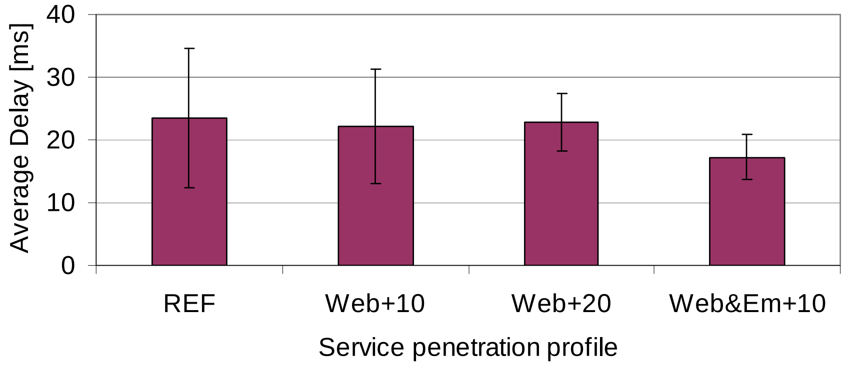


Figure H-2 - Average Delay variation with service penetration.

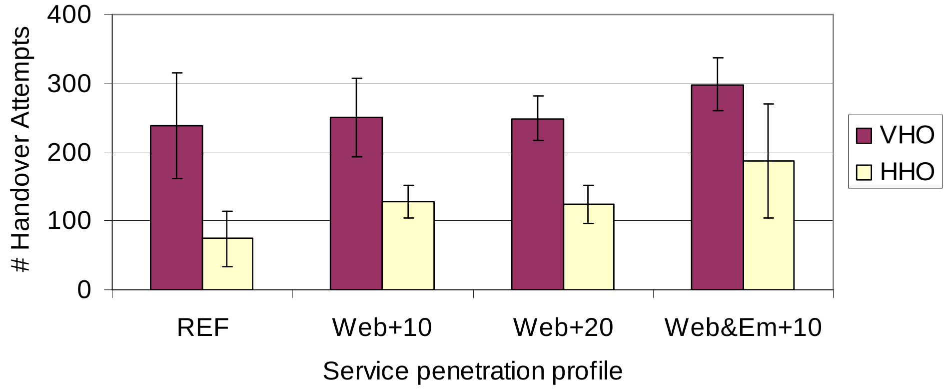


Figure H-3 - Total number of handovers variation with service penetration.

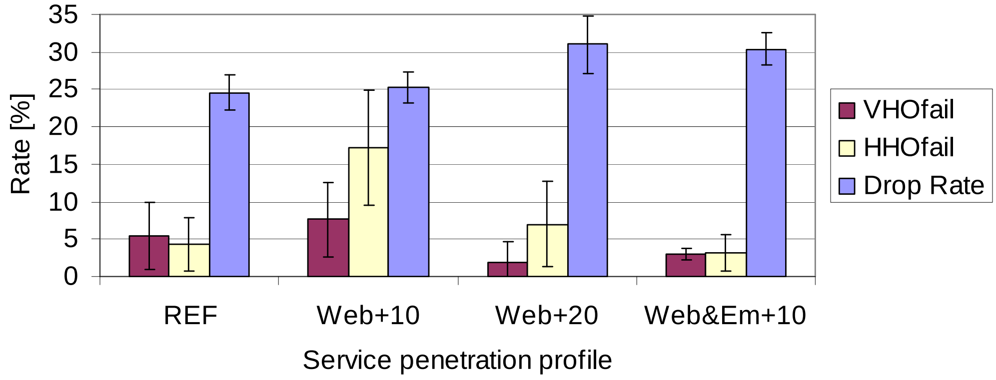


Figure H-4 – Vertical, horizontal handover failure rate and drop rate variation with service penetration.

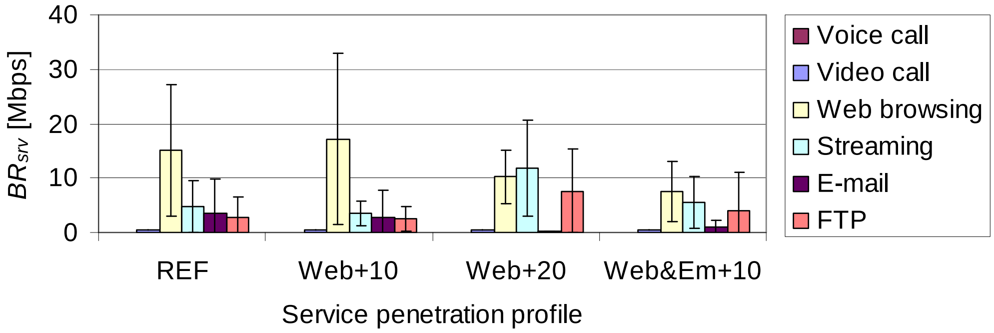
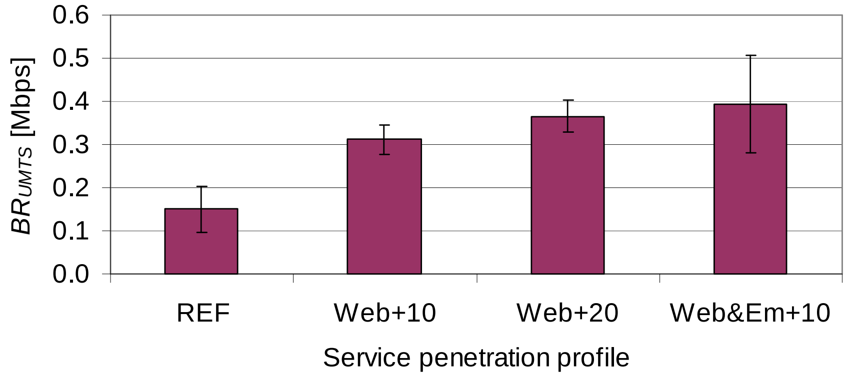
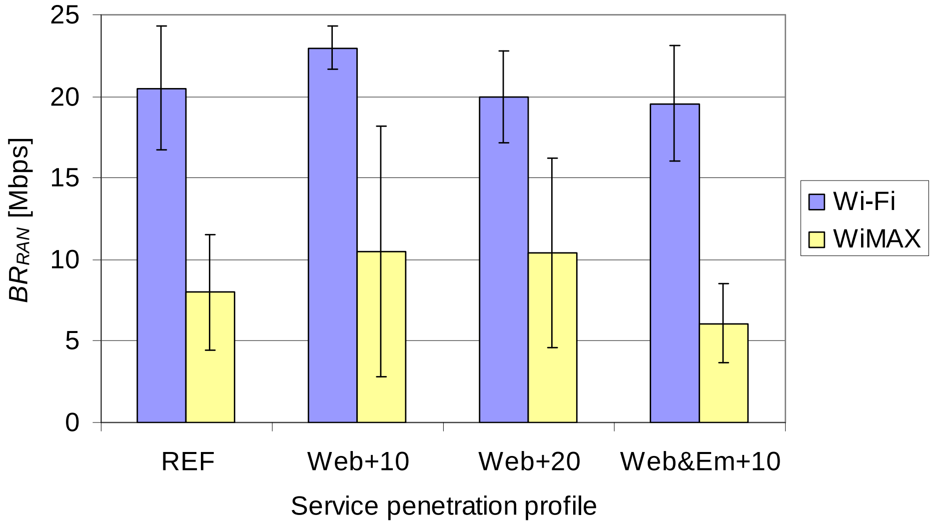


Figure H-5 – Average bitrate per service variation with service penetration.



a) UMTS



b) Wi-Fi and WiMAX

Figure H-6 – Average bitrate per system variation with service penetration.

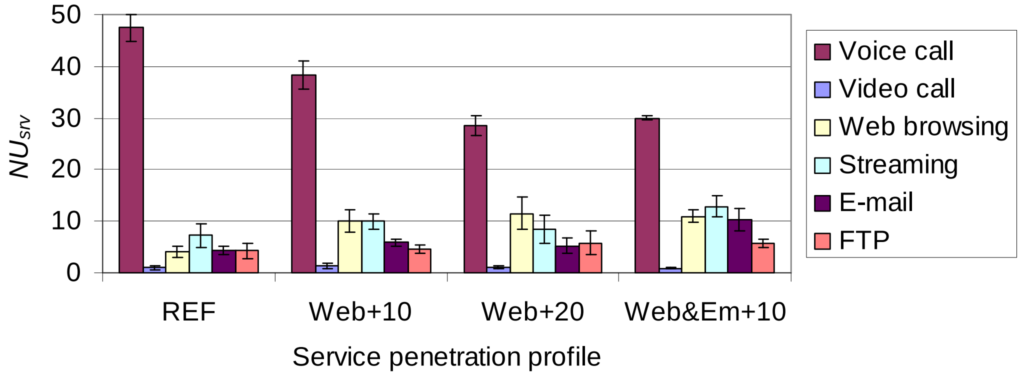


Figure H-7 – Average number of users per service per second variation with service penetration.

Annex I

HNP variation with the priority table

In this annex, the HNP is evaluated as a function of the priority table.

Table I-1 - HNP variation with the priority table.

HNP \ Prioritary RAN for data services		REF Mean val.	REF St.dev	Wi-Fi Mean val.	Wi-Fi St.dev	WiMAX Mean val.	WiMAX St. dev.	UMTS R5 Mean val.	UMTS R5 St.dev
P_b [%]		1.00	0.87	1.62	1.90	0.87	0.74	2.34	1.79
τ [ms]		23.53	11.10	24.52	8.07	19.89	5.34	25.09	5.28
N_{VHOa}		237.53	76.83	313.40	83.44	250.40	80.90	244.20	60.98
N_{HHOa}		74.00	39.71	145.60	56.50	131.00	63.30	49.00	11.49
P_{VHOf} [%]		5.40	4.45	9.00	8.60	7.20	5.72	19.80	9.04
P_{HHOf} [%]		4.33	5.74	5.60	6.54	9.40	11.84	4.60	3.51
D_r [%]		24.60	2.29	24.80	1.10	25.00	1.00	25.40	2.07
BR_{srv} [Mbps]	Voice call	0.03	0.00	0.03	0.00	0.03	0.00	0.03	0.00
	Video call	0.51	0.01	0.51	0.01	0.51	0.01	0.51	0.01
	Web browsing	15.13	12.05	16.58	24.03	8.34	7.34	12.23	4.67
	Streaming	4.75	4.70	2.65	1.91	4.97	5.99	1.29	0.61
	E-mail	3.61	6.26	6.75	6.71	6.35	6.92	2.36	2.08
	FTP	2.87	3.75	5.98	5.78	0.68	0.38	2.67	3.41
BR_{RAN} [Mbps]	UMTS	0.15	0.05	0.12	0.04	0.11	0.03	0.49	0.06
	Wi-Fi	20.51	3.80	9.85	3.32	0.00	0.00	0.00	0.00
	WiMAX	7.99	3.52	7.17	5.45	6.11	4.28	8.94	2.98
NU_{srv}	Voice call	47.44	2.52	47.68	3.99	47.48	3.20	44.58	0.72
	Video call	0.98	0.38	1.03	0.43	1.15	0.66	1.17	0.63
	Web browsing	4.02	1.06	4.16	2.09	4.21	0.78	5.99	1.89
	Streaming	7.22	2.21	8.98	3.67	9.37	1.91	12.89	1.18
	E-mail	4.28	0.82	4.56	1.81	4.83	1.24	7.29	0.49
	FTP	4.25	1.56	4.69	2.15	4.06	0.92	7.64	1.37

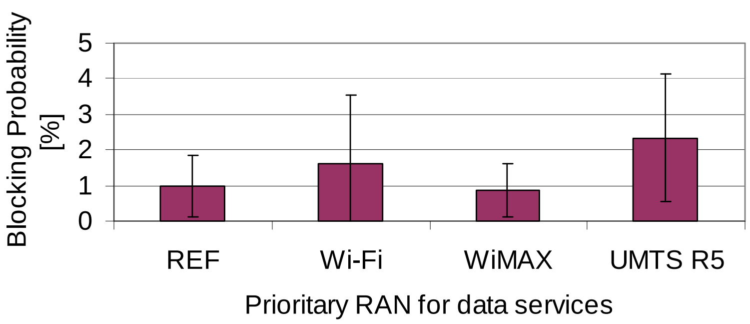


Figure I-1 - Blocking Probability variation with priority table.

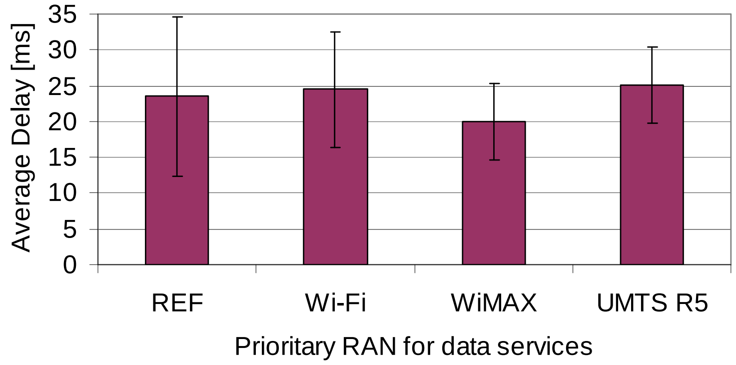


Figure I-2 - Average Delay variation with priority table.

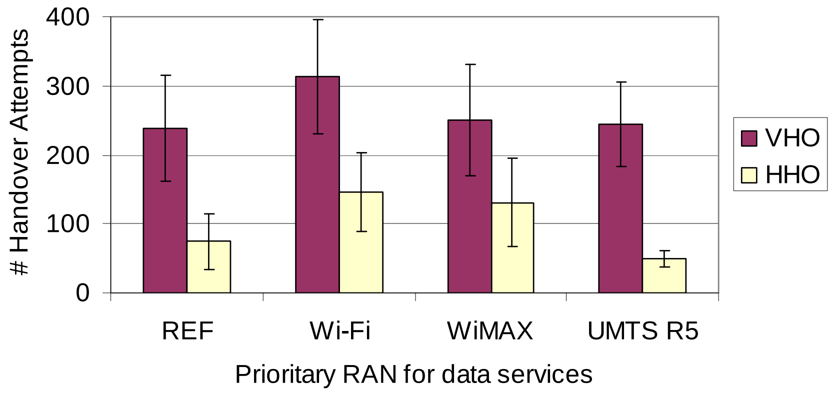


Figure I-3 - Total number of handovers variation with priority table.

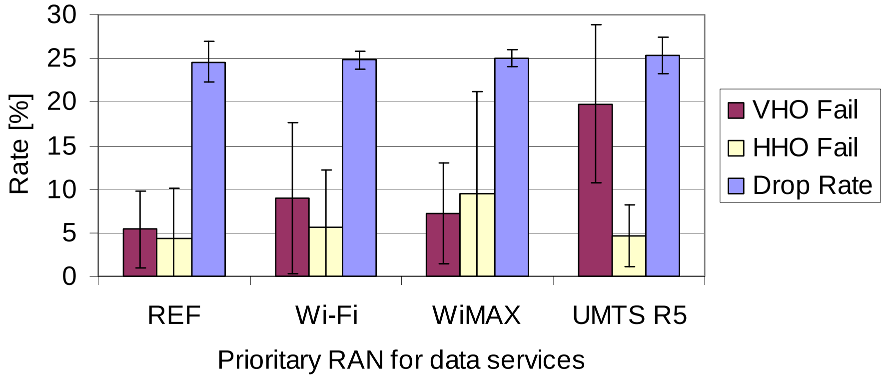


Figure I-4 – Vertical, horizontal handover failure rate and drop rate variation with priority table.

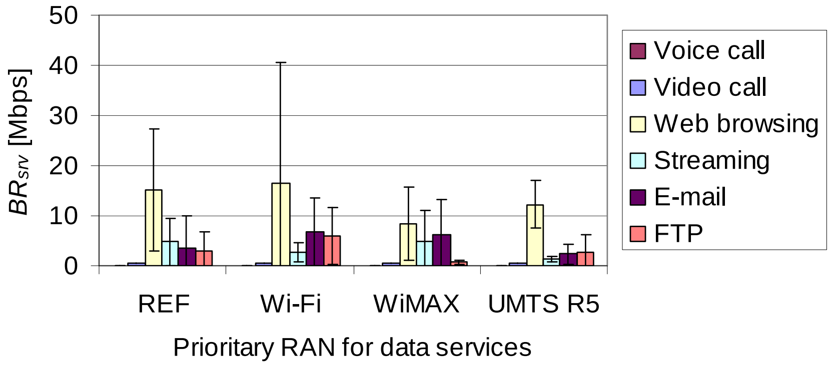
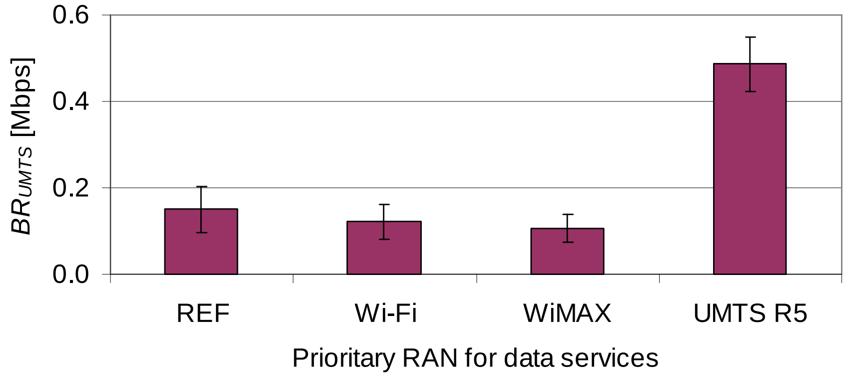
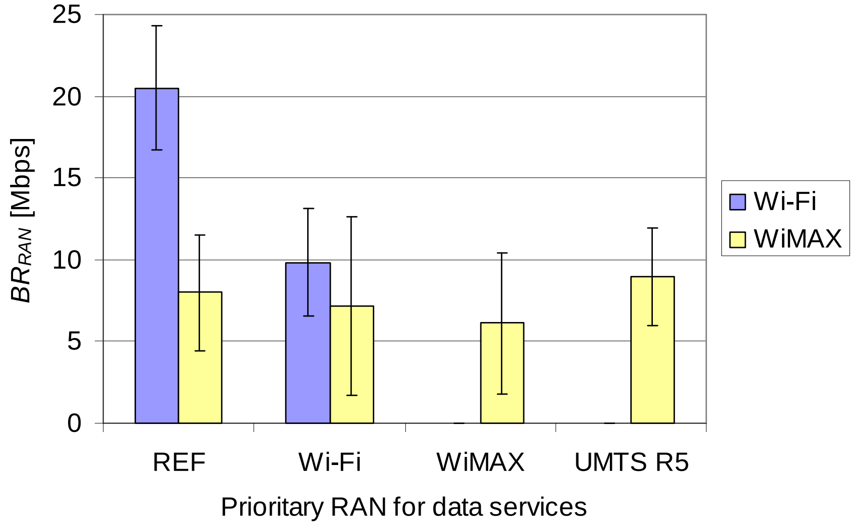


Figure I-5 – Average bitrate per service variation with priority table.



a) UMTS



b) Wi-Fi and WiMAX

Figure I-6 – Average bitrate per system variation with priority table.

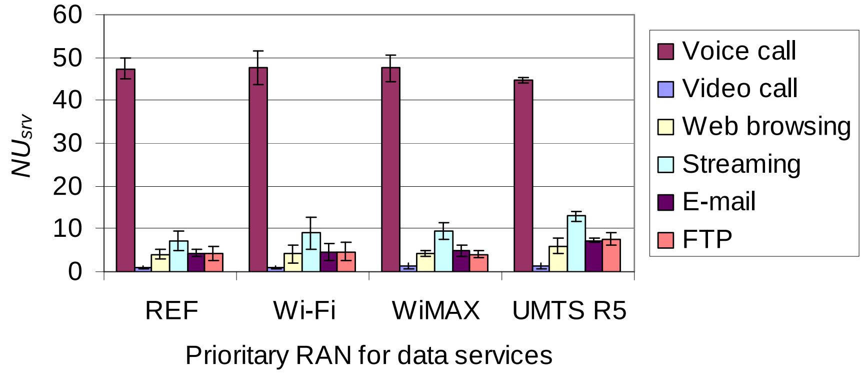


Figure I-7 – Average number of users per service per second variation with priority table.

Annex J

HNP variation with selective RAN switch-off

In this annex, the HNP is evaluated as a function of selective RAN switch-off.

Table J-1 - HNP variation with selective RAN switch-off.

HNP \ RAN OFF		REF		Wi-Fi		WiMAX		UMTS R5	
		Mean val.	St.dev	Mean val.	St.dev	Mean val.	St. dev.	Mean val.	St.dev
P_b [%]		1.00	0.87	1.06	0.70	12.45	6.19	3.26	1.09
τ [ms]		23.53	11.10	24.49	5.09	2011.06	1441.57	22.23	11.82
N_{VHOa}		237.53	76.83	248.80	45.83	7.00	3.74	59.20	20.46
N_{HHOa}		74.00	39.71	65.00	19.58	38.20	34.35	136.00	78.21
P_{VHOf} [%]		5.40	4.45	4.20	5.02	8.60	12.92	18.20	5.54
P_{HHOf} [%]		4.33	5.74	10.00	17.42	0.00	0.00	8.00	11.25
D_r [%]		24.60	2.29	25.00	2.00	10.00	7.48	38.20	4.60
BR_{srv} [Mbps]	Voice call	0.03	0.00	0.03	0.00	0.03	0.00	0.03	0.00
	Video call	0.51	0.01	0.51	0.01	0.52	0.00	0.50	0.01
	Web browsing	15.13	12.05	22.05	12.06	7.99	1.02	16.84	17.92
	Streaming	4.75	4.70	4.67	4.03	0.38	0.00	3.60	1.88
	E-mail	3.61	6.26	5.81	7.39	0.12	0.00	2.07	2.50
	FTP	2.87	3.75	1.51	1.08	1.20	1.45	4.42	5.04
BR_{RAN} [Mbps]	UMTS	0.15	0.05	0.16	0.07	0.60	0.07	0.08	0.02
	Wi-Fi	20.51	3.80	0.00	0.00	21.23	4.47	21.46	3.62
	WiMAX	7.99	3.52	10.43	3.91	0.00	0.00	7.79	4.37
NU_{srv}	Voice call	47.44	2.52	47.88	3.38	46.39	3.20	32.87	3.75
	Video call	0.98	0.38	1.23	0.72	1.21	0.72	1.27	0.39
	Web browsing	4.02	1.06	4.72	0.73	4.77	0.69	4.81	1.32
	Streaming	7.22	2.21	8.14	1.72	7.91	2.13	9.38	4.17
	E-mail	4.28	0.82	5.00	0.72	5.50	1.04	5.04	1.86
	FTP	4.25	1.56	4.40	1.38	4.60	1.41	3.90	1.81

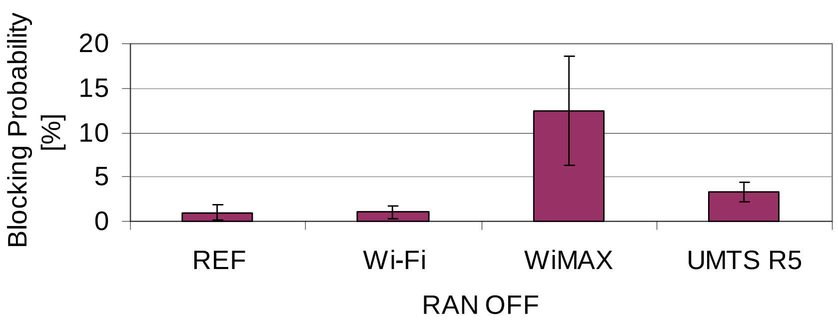


Figure J-1- Blocking Probability variation with selective RAN switch-off.

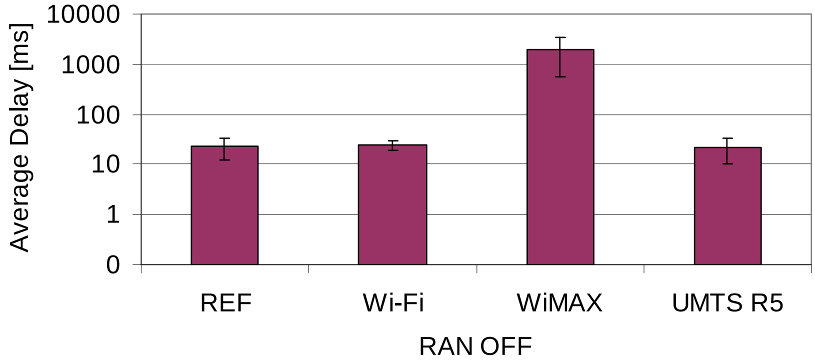


Figure J-2 - Average Delay variation with selective RAN switch-off.

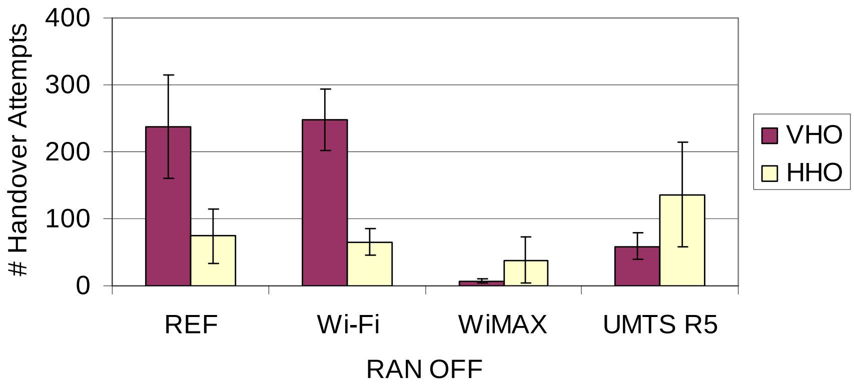


Figure J-3 - Total number of handovers variation with selective RAN switch-off.

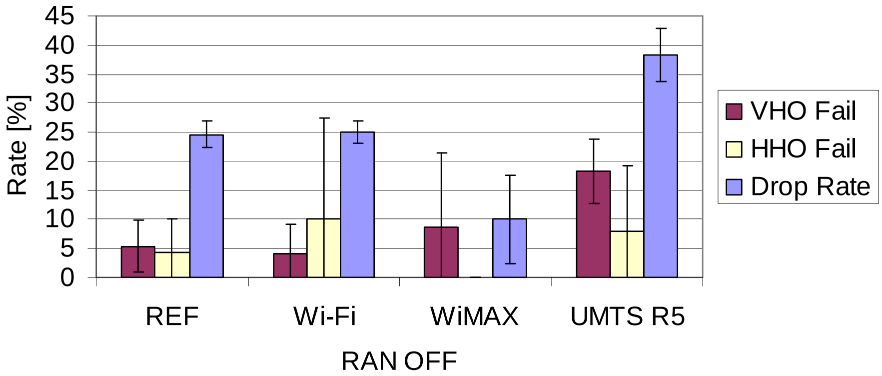


Figure J-4 – Vertical, horizontal handover failure rate and drop rate variation with selective RAN switch-off.

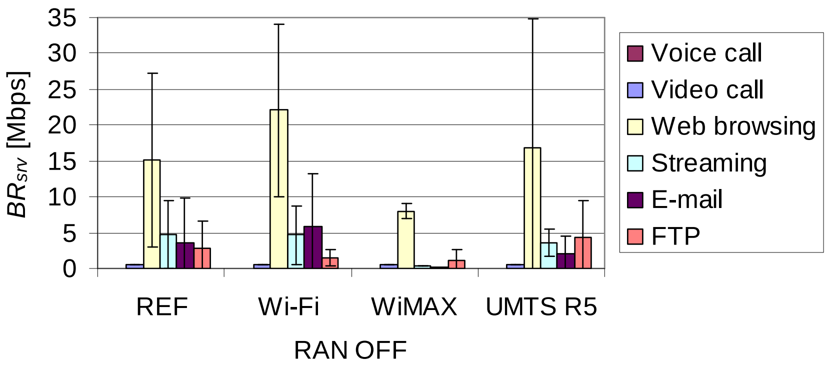
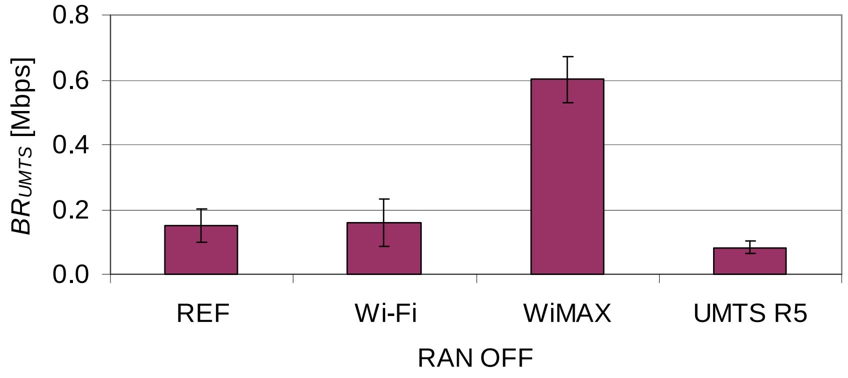
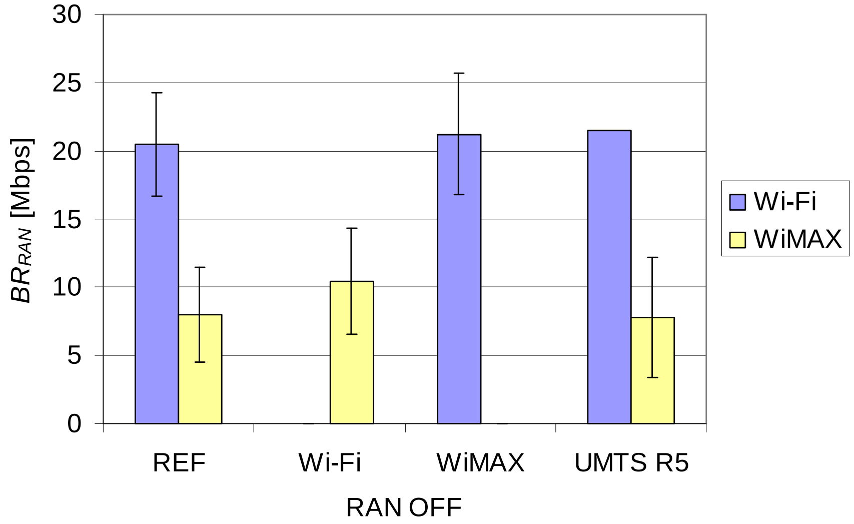


Figure J-5 – Average bitrate per service variation with selective RAN switch-off.



a) UMTS



b) Wi-Fi and WiMAX

Figure J-6 – Average bitrate per system variation with selective RAN switch-off.

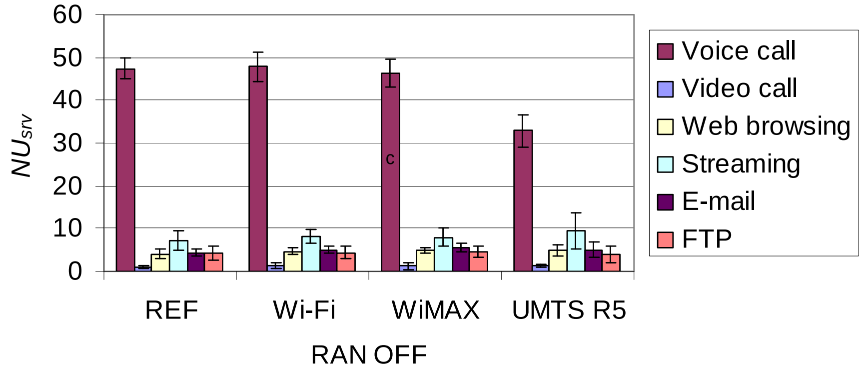


Figure J-7 – Average number of users per service per second variation with selective RAN switch-off.

Annex K

HNP variation with the bandwidth of the WiMAX channel

In this annex, the HNP is evaluated as a function of the bandwidth of the WiMAX channel and the density of users.

Table K-1 - HNP variation with the bandwidth of the WiMAX channel.

HNP \ WiMAX channel BW		10 000 users				14 000 users			
		5 MHz (REF)		10 MHz		5 MHz		10 MHz	
		Mean val.	St.dev.	Mean val.	St.dev.	Mean val.	St. dev.	Mean val.	St.dev.
P_b [%]		1.00	0.87	1.00	1.49	13.35	8.33	14.31	6.77
τ [ms]		23.53	11.10	19.22	13.58	40.68	14.13	27.08	11.18
N_{VHOa}		237.53	76.83	326.00	84.74	331.20	159.86	392.60	126.47
N_{HHOa}		74.00	39.71	118.40	101.72	120.40	69.16	121.00	27.08
P_{VHOf} [%]		5.40	4.45	4.60	6.54	34.60	14.74	35.00	16.52
P_{HHOf} [%]		4.33	5.74	1.40	1.14	28.00	18.92	5.00	6.96
D_r [%]		24.60	2.29	25.20	0.45	28.60	6.31	31.20	2.49
BR_{srv} [Mbps]	Voice call	0.03	0.00	0.03	0.00	0.03	0.00	0.03	0.00
	Video call	0.51	0.01	0.51	0.00	0.51	0.01	0.51	0.01
	Web browsing	15.13	12.05	22.33	19.07	16.10	13.19	9.15	7.09
	Streaming	4.75	4.70	3.94	2.60	12.36	10.27	5.59	6.84
	E-mail	3.61	6.26	0.34	0.09	0.62	0.27	9.37	6.61
	FTP	2.87	3.75	2.09	1.90	3.47	4.20	3.19	1.97
BR_{RAN} [Mbps]	UMTS	0.15	0.05	0.14	0.03	0.15	0.04	0.19	0.05
	Wi-Fi	20.51	3.80	20.24	2.92	22.42	1.46	22.92	3.04
	WiMAX	7.99	3.52	8.38	7.49	11.04	5.93	7.00	2.04
NU_{srv}	Voice call	47.44	2.52	49.47	1.30	56.91	6.25	55.44	4.66
	Video call	0.98	0.38	1.32	0.81	2.50	0.83	1.42	0.51
	Web browsing	4.02	1.06	4.44	1.61	5.08	1.88	6.28	1.00
	Streaming	7.22	2.21	7.42	3.41	9.38	3.52	11.33	2.06
	E-mail	4.28	0.82	4.50	0.97	3.17	0.83	4.55	0.76
	FTP	4.25	1.56	4.61	2.23	5.05	0.85	6.20	1.54

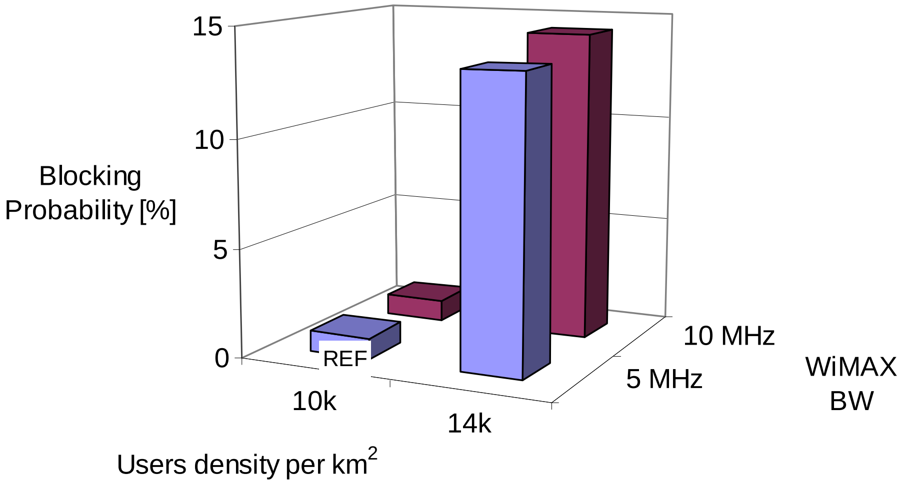


Figure K-1 - Blocking Probability variation with WiMAX channel bandwidth and density of users.

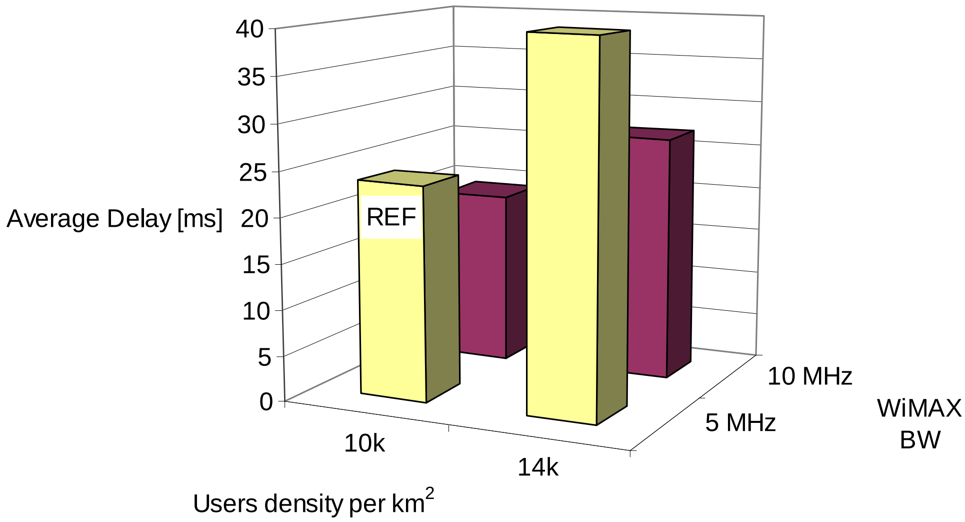


Figure K-2 - Average Delay variation with WiMAX channel bandwidth and density of users.

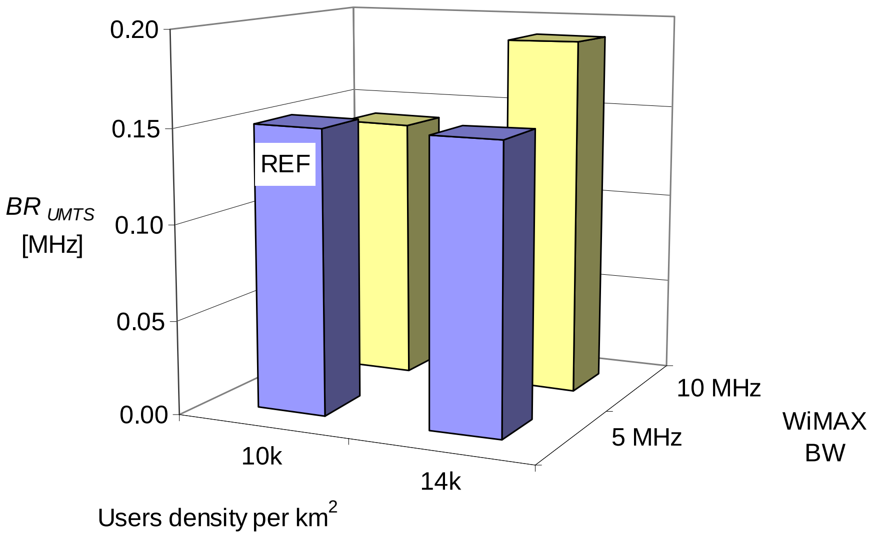


Figure K-3 –UMTS Average Bitrate variation with WiMAX channel bandwidth and density of users.

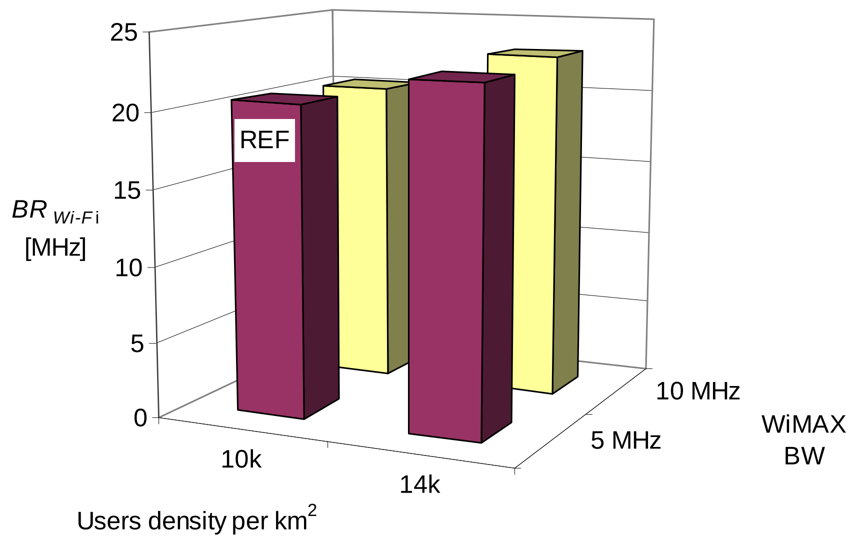


Figure K-4 –Wi-Fi Average Bitrate variation with WiMAX channel bandwidth and density of users.

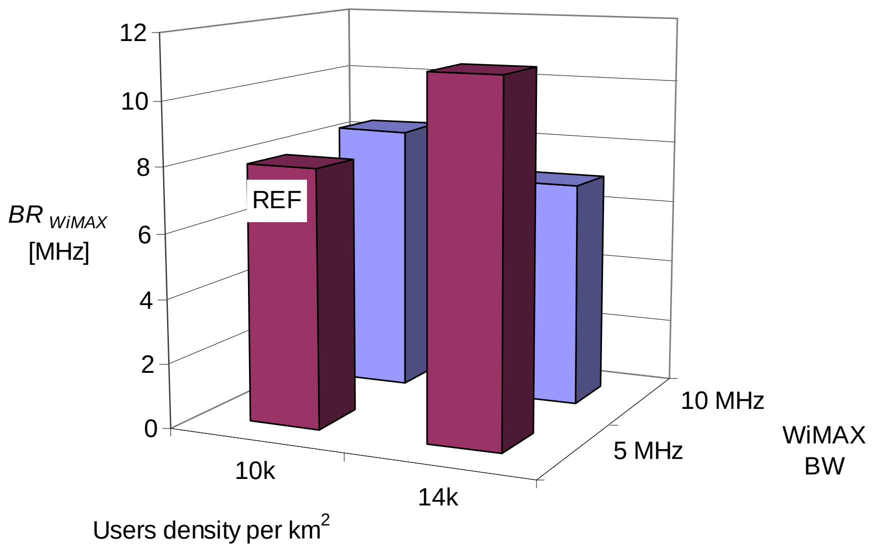


Figure K-5 –WiMAX Average Bitrate variation with WiMAX channel bandwidth and density of users.

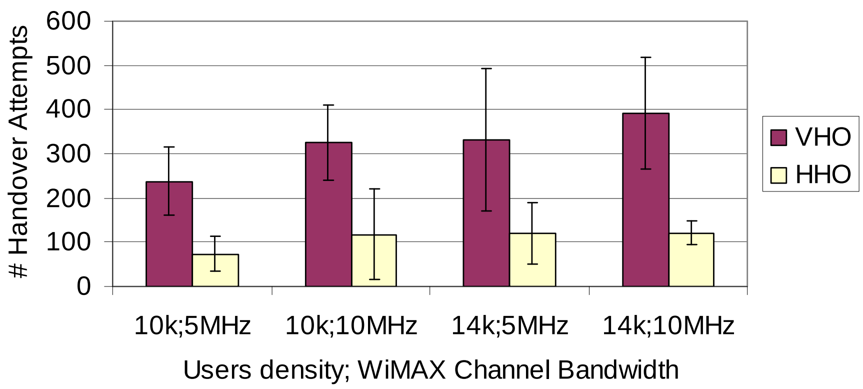


Figure K-6 - Total number of handovers variation with WiMAX channel bandwidth and density of users.

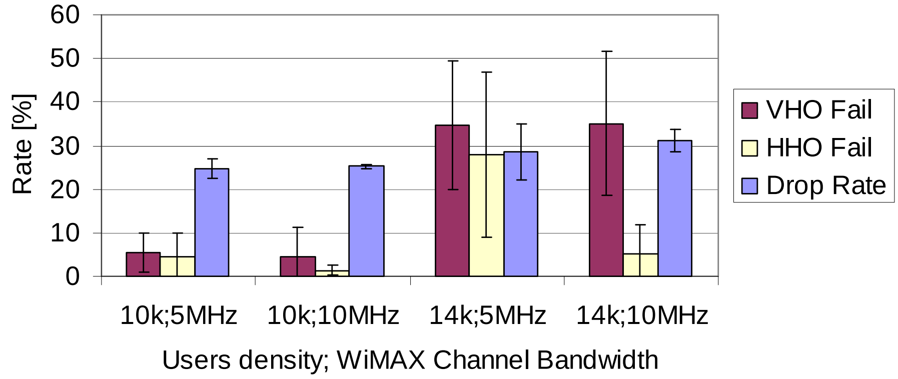


Figure K-7 – Vertical, horizontal handover failure rate and drop rate variation with WiMAX channel bandwidth and density of users.

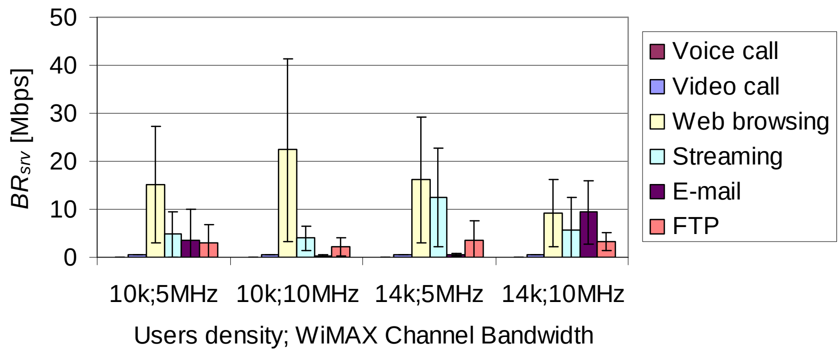
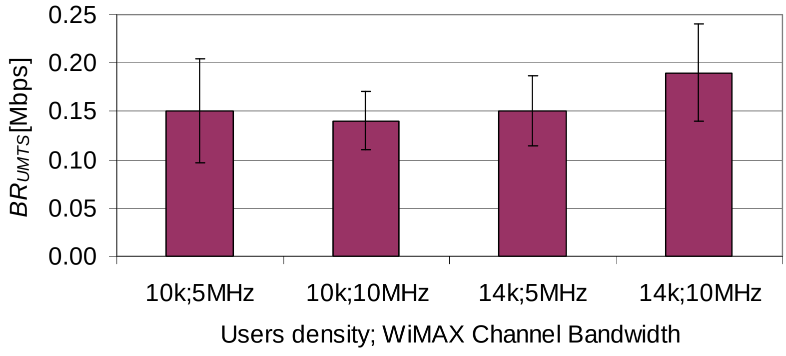
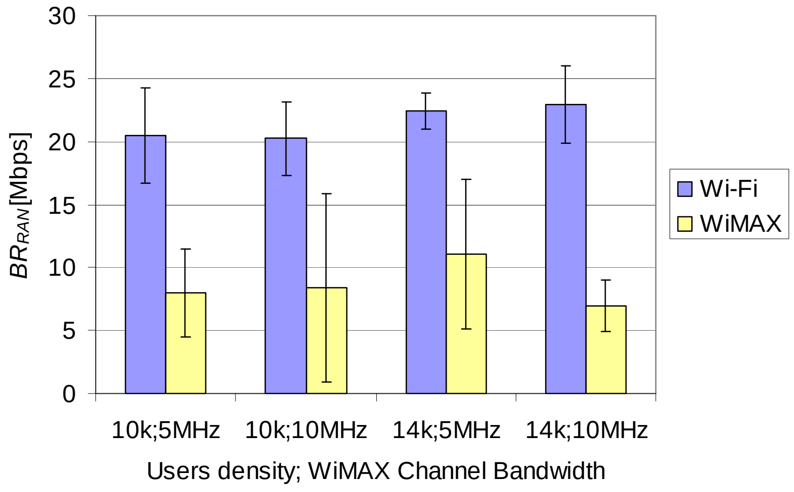


Figure K-8 – Average bitrate per service variation with WiMAX channel bandwidth and density of users.



a) UMTS



b) Wi-Fi and WiMAX

Figure K-9– Average bitrate per system variation with WiMAX channel bandwidth and density of users.

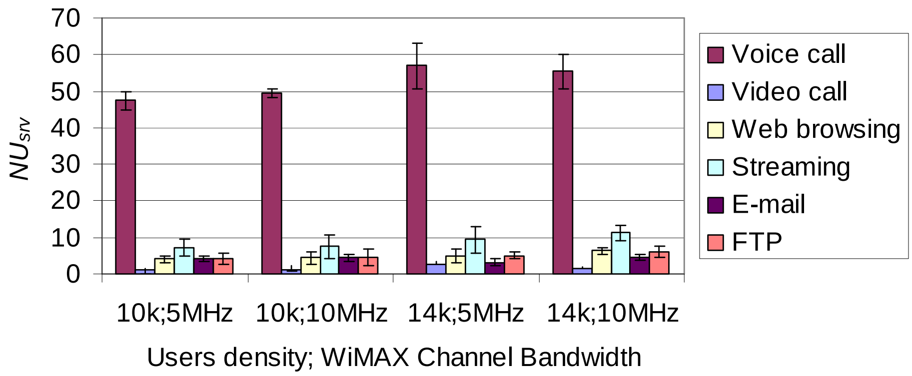


Figure K-10 – Average number of users per service per second variation with WiMAX channel bandwidth and density of users.

References

[3GPP02] 3GPP, *Transparent end-to-end packet switched streaming service (PSS); Protocols and codecs (Release 4)*, 3GPP Technical Specification No. 26.234 Ver. 4.3.0, Valbonne, France, Mar. 2002. (www.3gpp.org)

[3GPP09] http://www.3gpp.org, Sep. 2009

[Agui03] Aguiar,J., *Traffic Analysis at the Radio Interface in Converging Mobile and Wireless Communication Systems*, M.Sc. Thesis, Instituto Superior Técnico, Technical University of Lisbon, Lisbon, Portugal, 2003.

[Alva06] Alvarion, *Understanding the Radio Technologies of Mobile WiMAX*, Tel Aviv, Israel, 2006 (http://www.alvarion.com/upload/contents/291/Radio_Mobile_WiMAX_wp.pdf).

[AnPi07] Antoniou,J. and Pitsillides,J., "4G Converged Environment: Modeling Network Selection as a Game", in *Proc. of MWCS'07 – the 16th IST Mobile & Wireless Communications Summit*, Budapest, Hungary, July 2007.

[Antu07] Antunes,D., *WiMAX 802.16e Network Performance Analysis for Different Radio Resource Management Algorithms*, M.Sc. Thesis, Instituto Superior Técnico, Technical University of Lisbon, Lisbon, Portugal, Dec. 2007.

[AROM07] Sallent,O., *Intermediate report on AROMA algorithms and simulation results*, IST-AROMA Project, Deliverable 12, EC-IST Office, Brussels, Belgium, Apr. 2007.

[BaBB07] Bakmaz,B., Bojkovic,Z. and Bakmaz,M., "Network Selection Algorithm for Heterogeneous Wireless Environment", in *Proc. of PIMRC'07 - The 18th Annual IEEE International Symposium on Personal, Indoor and Mobile Radio Communications*, Athens, Greece, Sep. 2007.

[BaSa08] Barbaresi,A. and Sallent,O., "Techno-Economic Evaluation of a CRRM Traffic Steering Strategy Between GERAN and UTRAN", in *Proc. of WPMC'08 – The 11th International Symposium on Wireless Personal Multimedia Communications*, Lapland, Finland, Sep. 2008.

[BKSH08] Baek,J.-Y., Kim,D.-J., Suh,Y.-J., Hwang,E.-S. and Chung,Y.-D., "Network-initiated Handover Based on IEEE 802.21 Framework for QoS Service Continuity in UMTS/802.16e Networks", in *Proc. of VTC'08 – The 67th IEEE Vehicular Technology Conference*, Marina Bay, Singapore, May 2008.

[Blue09] http://www.bluetooth.com/bluetooth, Oct. 2009

[ChRe98] Chandra,K. and Reibman,A.R., "Modelling One and Two Layer Variable Bit Rate Video", *IEEE/ACM Trans. on Networking*, Vol. 7, No. 3, 1998, pp. 398-413.

[CMMC08] Calabuig,D., Monserrat,J.F., Martín-Sacristán,D. and Cardona,N., "Joint Dynamic Resource Allocation for Coupled Heterogeneous Wireless Networks based on Hopfield Neural Networks", in *Proc. of VTC'08 – The 67th IEEE Vehicular Technology Conference*, Marina Bay, Singapore, May 2008.

[CoKG08] Coupechoux,M., Kelif,J.-M. and Godlewski,P., "Network Controlled Joint Radio Resource Management for Heterogeneous Networks", in *Proc. of VTC'08 – The 67th IEEE Vehicular Technology Conference*, Marina Bay, Singapore, May 2008.

[Corr03] Correia,L.M., *Mobile Communication Systems*, *Course Notes*, Instituto Superior Técnico, Technical University of Lisbon, Lisbon, Portugal, 2003.

[DaCo99] Damosso,E. and Correia,L.M., *Digital Mobile Radio Towards Future Generation Systems*, COST 231 Final Report, COST Office, Brussels, Belgium, 1999.

[EMPS06] Eklund,C., Marks,R., Ponnuswamy,S., Stanwood,K. and Waes,N., *WirelessMAN Inside the IEEE 802.16 Standard for Wireless Metropolitan Networks*, IEEE Press, New York, NY, USA, 2006

[ETSI98] ETSI, *Universal Mobile Telecommunications System (UMTS); Selection procedures for the choice of radio transmission technologies of the UMTS*, ETSI Technical Report No. TR 101 112 Ver. 3.2.0, France, Apr. 1998.

[GoES08] Gozálvez,J., Lucas-Estañ,M.C. and Sánchez-Soriano,J., "Joint Radio Resource Management in Beyond 3G Heterogeneous Wireless Systems", in *Proc. of WPMC'08 – The 11th International Symposium on Wireless Personal Multimedia Communications*, Lapland, Finland, Sep. 2008.

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

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

[IEEE04a] IEEE 802.1 WG, *"802.1D – IEEE Standard for Local and metropolitan area networks , Media Access Control (MAC) Bridges"*, IEEE Std 802.1D, 2004

[IEEE04b] IEEE, *802.16 IEEE Standard for Local and metropolitan area networks – Part 16: Air Interface for Fixed Broadband Wireless Access Systems*, IEEE, New York, NY, USA, Oct. 2004 (http://standards.ieee.org/getieee802/802.16.html).

[IEEE05] IEEE 802.11 WG, *IEEE Standard for Information Technology – Telecommunications and information exchange between systems – Local and metropolitan area networks – specific requirements. Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications. Amendment 8: Medium Access Control (MAC) Quality of Service Enhancements*, IEEE Std 802.11e, Sep. 2005.

[IEEE06] IEEE, *802.16 IEEE Standard for Local and metropolitan area networks – Part 16: Air Interface for Fixed Broadband Wireless Access Systems – Amendment 2: Physical and Medium Access Control Layers for Combined Fixed and Mobile Operation in Licensed*

Bands and Corrigendum 1, IEEE, New York, USA, Feb. 2006 (http://standards.ieee.org/getieee802/802.16.html).

[IEEE09a] http://standards.ieee.org/getieee802/802.11.html, July 2009.

[IEEE09b] http://www.ieee802.org/21/, July 2009

[IEEE09c] http://grouper.ieee.org/groups/scc41/, July 2009

[IETF01] IETF, *RObust Header Compression (ROHC): Framework and four profiles*, IETF RFC 3095, Center Oak Plaza, VA, USA, July 2001 (http://www.ietf.org/rfc.html).

[LMSB08] Liu,B., Martins,P., Samhat,A.E. and Bertin,P., "A Layer 2 Scheme of Inter-RAT Handover between UMTS and WiMAX", in *Proc. of VTC'08 – The 68th IEEE Vehicular Technology Conference*, Calgary, Canada, Sep. 2008.

[Molis04] Molisch,A.F., *Wireless Communications*, John Wiley & Sons, Chichester, UK, 2005.

[NeSi08] http://www.isi.edu/nsnam/ns, July 2008

[Nuay07] Nuaymi,L., *WiMAX Technology for Broadband Wireless Access*, John Wiley, Chichester, UK, 2007.

[Nune02] Nunes.M.S., *Integrated Services Networks – 4th section –Voice over IP* (in Portuguese), AEIST, Lisbon, Portugal, 2002.

[OnKh08] Ong,E.H. and Khan,J.Y., "Dynamic Access Network Selection with QoS Parameters Estimation: A Step Closer to ABC", in *Proc. of VTC'08 – The 67th IEEE Vehicular Technology Conference*, Marina Bay, Singapore, May 2008.

[PiDa07] Piao,G. and David,K., "MXRRM for WIMAX Integrated to GSM and UMTS Heterogeneous Networks", in *Proc. of VTC'07 – The 65th IEEE Vehicular Technology Conference*, Dublin, Ireland, Apr. 2007.

[Pras01] Prasad,N. and Prasad,A., *WLAN System and Wireless IP for Next Generations Communications*, Artech House, London, UK, 2001

[RaMe01] Rábanos,J. and Mesquida,C., *Mobile Communication of Third Generation* (In Spanish), Telefónica Móviles España, Madrid, Spain, 2001.

[RoSA07] Pérez-Romero,J., Sallent,O. and Agustí,R., "A Generalized Framework for Multi-RAT Scenarios Characterisation", in *Proc. of VTC'07 – The 65th IEEE Vehicular Technology Conference*, Dublin, Ireland, Apr. 2007.

[RSAD05] Pérez-Romero,J., Sallent,O., Agustí,R. and Diáz-Guerra,M., *Radio Resource Management Strategies in UMTS*, John Wiley, Chichester, UK, 2005

[RSAN07] Pérez-Romero,J., Sallent,O., Agustí,R. and Nasreddine,J., "Radio Access Technology Selection enabled by IEEE P1900.4", in *Proc. of MWCS'07 – the 16th IST Mobile & Wireless Communications Summit*, Budapest, Hungary, July 2007.

[SeCo07] Serrador,A. and Correia,L.M., "Policies for a Cost Function for Heterogeneous Networks Performance Evaluation", in *Proc. of PIMRC'07 - The 18th Annual IEEE International Symposium on Personal, Indoor and Mobile Radio Communications*, Athens, Greece, Sep. 2007.

[Serr02] Serrador,A., *Optimisation of cell radius in UMTS-FDD networks*, M.Sc. Thesis, Instituto Superior Técnico, Technical University of Lisbon, Lisbon, Portugal, Dec. 2002.

[STAL05] Stallings,W., *Wireless Communications & Networks – Second Edition*, Pearson Prentice Hall, Upper Saddle River, NJ, USA, 2005.

[TaPS08] Tawil,R., Pujolle,G. and Salazar,O., "A Vertical Handoff Decision Scheme Heterogeneous Wireless Systems", in *Proc. of VTC'08 – The 67th IEEE Vehicular Technology Conference*, Marina Bay, Singapore, May 2008.

[VRFo99] 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

[WiMA05] WiMAX Forum, *Can WiMAX Address Your Applications*, Internal Report, Beaverton, OR, USA, 2005, (http://www.wimaxforum.org/technology/downloads/Can_WiMAX_Address_Your_Applications_final.pdf).

[WiMA06a] WiMAX Forum, *Mobile WiMAX – Part I: A Technical Overview and Performance Evaluation*, Internal Report, Beaverton, OR, USA, 2006, (http://www.wimaxforum.org/technology/downloads/Mobile_WiMAX_Part1_Overview_and_Performance.pdf).

[WiMF09] http://www.wimaxforum.org, July 2009

[XZVY08] Xian,Z., Zhiyong,F., Vanbien,L., Yuan,X. and Yuewei,L., "An Auction Based Joint Radio Resource Management Scheme and Architecture in a Multi-operator Scenario", in *Proc. of VTC'08 – The 67th IEEE Vehicular Technology Conference*, Marina Bay, Singapore, May 2008.

[Yaco93] Yacoub,M.D., *Foundations of Mobile Radio Engineering*, CRC Press, Boca Raton, FL, USA, 1993.

[Yagh04] Yaghoobi,H., "Scalable OFDMA Physical Layer in IEEE 802.16 WirelessMAN", *Intel Technology Journal*, Vol. 8, Issue 3, Aug. 2004

[YaGQ08] Yang,K., Gondal,I. and Qiu,B., "Context Aware Vertical Soft Handoff Algorithm for Heterogeneous Wireless Networks", in *Proc. of VTC'08 – The 68th IEEE Vehicular Technology Conference*, Calgary, Canada, Sep. 2008.

[YNCF07] Chen,Y.-H., Yang,N.-Y., Chang,C.-J. and Ren,F.-C., "A Utility Function-Based Access Selection Method For Heterogeneous WCDMA and WLAN Networks", in *Proc. of*

PIMRC'07 - The 18th Annual IEEE International Symposium on Personal, Indoor and Mobile Radio Communications, Athens, Greece, Sep. 2007.