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Traffic Analysis at the Radio Interface in Converging Mobile and Wireless Communication Systems

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To Gabriela and my parents

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Abstract

This work analyses the overall performance at the radio interface of a converged multi-technology network, with the aid of a newly developed simulation platform. Three technologies are considered: GSM/GPRS, UMTS and HIPERLAN/2. Three major performance indicators are used for the analysis: CS calls blocking rate, % of PS discarded packets/frames and mean bitrate obtained per PS application.

A key and novel concept adopted in this work is the Convergence Manager. This new element, which is present at both terminal and network sides, is responsible for the implementation of all convergence functionalities in the overall network.

Six different applications were considered and characterised for user traffic generation: Voice Call, Video Call, Video on Demand, Web Browsing, Email and FTP. Traffic source models, as well as call/session generation and duration processes, have been implemented for each of these applications, based on an extensive literature survey.

From the statistical analysis of user traffic variations performed in this work, it has been verified that results obtained from single simulation runs allow a reasonable degree of confidence, reducing the number of needed simulation runs per scenario.

As expected, the performance indicators degrade with the increase of the number of users for a specific technology scenario, and better overall performances are achieved when more technologies are available for a specific user scenario (specially if HIPERLAN/2 is available).

It has been verified that, for instance, a single cell scenario with all three technologies available supports up to 700 users with acceptable performance margins.

Keywords

Traffic, Radio Interface, Convergence, GSM/GPRS, UMTS, HIPERLAN/2, Simulation.

Resumo

Este trabalho analisa o desempenho global da interface rádio de uma rede multi-tecnológica com convergência, com o auxílio de um novo simulador desenvolvido para tal. São consideradas três tecnologias: GSM/GPRS, UMTS e HIPERLAN/2. Na análise são utilizados três indicadores principais de desempenho: taxa de bloqueio de chamadas, % de pacotes/tramas descartados e débito binário médio.

Um conceito chave e novo utilizado neste trabalho é o Gestor de Convergência. Este novo elemento, que se localiza tanto do lado do terminal como do lado da rede, é responsável por todas as funcionalidades de convergência.

Para geração de tráfego dos utilizadores, foram consideradas e caracterizadas seis aplicações: Chamada de Voz, Chamada de Video, *Video on Demand*, *Web Browsing*, Email e FTP. Para cada uma destas aplicações foram implementados modelos de fonte, assim como processos de geração e duração de chamadas/sessões, baseado numa extensa pesquisa bibliográfica.

Da análise estatística de variações do tráfego de utilizador efectuada neste trabalho, foi verificado que os resultados obtidos de simulações únicas fornecem um grau de confiança razoável, permitindo a redução do número de simulações necessárias por cenário.

Como esperado, os indicadores de desempenho degradam-se com o aumento do número de utilizadores para cada cenário tecnológico, e melhores desempenhos globais são obtidos à medida que maior número de tecnologias são disponibilizadas para um determinado cenário de utilizadores (especialmente quando HIPERLAN/2 está disponível).

Foi verificado que, por exemplo, um cenário unicelular com todas as três tecnologias disponíveis suporta até 700 utilizadores com margens de desempenho aceitáveis.

Palavras-chave

Tráfego, Interface Rádio, Convergência, GSM/GPRS, UMTS, HIPERLAN/2, Simulação.

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List of Abbreviations

2G	2nd Generation
3G	3rd Generation
3GPP	3rd Generation Partnership Project
4G	4th Generation
AAA	Authentication, Authorisation and Accounting
AAC	Advanced Audio Coding
ACF	Association Control Function
ACH	Access feedback CHannel
AI	Air Interface
AMR	Adaptive Multi-Rate
AP	Access Point
ARQ	Automatic Repeat reQuest
ASCH	Association Control CHannel
ATM	Asynchronous Transfer Mode
ATTPM	Access Technology Traffic Processing Modules
AuC	Authentication Centre
BCCH	Broadcast Control Channel
BCH	Broadcast CHannel
BER	Bit Error Rate
BHCA	Busy Hour Call Attempts
Bid	Bi-directional
BLER	BLock Error Rate
BN	Backbone Network
BoD	Bandwidth-on-Demand
BPSK	Binary Phase Shift Keying
BRAN	Broadband Radio Access Networks
BS	Base Station
BSC	Base Station Controller

BSS	Base Station Subsystem
BTS	Base Transceiver Station
CBCH	Cell Broadcast traffic CHannel
CBR	Constant BitRate
CCCH	Common Control CHannel
CCH	Control CHannel
CDF	Cumulative Distribution Function
CDMA	Code Division Multiple Access
CEPT	European Conference of Postal and Telecommunications Administrations
CL	Convergence Layer
CM	Convergence Manager
CMM	Convergence Manager Module
CN	Core Network
CPCH	Uplink Common Packet CHannel
C-PDU	Control PDU
CRC	Cyclic Redundancy Check
CS	Circuit Switched
CS-*i*	Coding Scheme *i*
DAB	Digital Audio Broadcasting
DCC	DLC Connection Control
DCCH	Dedicated Control CHannel
DCH	Dedicated CHannel
DFS	Dynamic Frequency Selection
DL	DownLink
DLC	Data Link Control layer
DS-CDMA	Direct Sequence CDMA
DSCH	Downlink Shared CHannel
DTX	Discontinuous Transmission
DVBP	Digital Video Broadcasting Project
DVB-T	Digital Video Broadcast – Terrestrial
EC	Error Control
EDGE	Enhanced Data Rate for GSM Evolution
E-GSM	Extended GSM 900 band

EIR	Equipment Identity Register
ETSI	European Telecommunications Standards Institute
FACH	Forward Access CHannel
FCCH	Frame Control CHannel
FCH	Frame CHannel
FDM	Frequency-Division Multiplexing
FDMA	Frequency Division Multiple Access
FER	Frame Erasure Rate
FTP	File Transfer Protocol
GBAR	Gamma Beta Auto-Regressive
GERAN	GSM/EDGE Radio Access Network
GGSN	Gateway GPRS Support Node
GMSC	Gateway MSC
GOP	Group Of Pictures
GPRS	General Packet Radio Service
GPS	Global Positioning System
GSM	Global System for Mobile Communications
HDTV	High Definition TeleVision
HIPERLAN/1	HIgh PErformance Radio Local Area Network Type 1
HIPERLAN/2	HIgh PErformance Radio Local Area Network Type 2
HLR	Home Location Register
HSCSD	High Speed Circuit Switched Data
HTTP	Hyper Text Transfer Protocol
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
IMT-2000	International Mobile Telecommunications-2000
IP	Internet Protocol
ISDN	Integrated Services Digital Network
IST	Information Society Technologies
ITU	International Telecommunications Union
ITU-T	ITU – Telecommunication standardisation sector
LA	Link Adaptation
LAN	Local Area Network
LCCH	Link Control CHannel

LCH	Long CHannel
LF	Load Factor
LRD	Long-Range Dependence
MAC	Medium Access Control
ME	Mobile Equipment
M-M	Many-to-Many
M-O	Many-to-One
MoU	Memorandum of Understanding
MP3	Moving Picture Experts Group Layer-3 Audio
MPEG	Moving Picture Expert Group
MS	Mobile Station
MSC	Mobile Services Switching Centre
MSE	Mean Square Error
MT	Mobile Terminal
NG	Next Generation
NRT	Non-Real-Time
NSS	Network and Switching Subsystem
NTB	Non Time Based
OFDM	Orthogonal Frequency Division Multiplex
O-M	One-to-Many
OMC	Operation and Maintenance Centre
O-O	One-to-One
OSS	Operation SubSystem
OVSF	Orthogonal Variable Spreading Factor
PC	Power Control
PCH	Paging Channel
PCMCIA	Personal Computer Memory Card International Association
PDA	Personal Digital Assistant
PDCH	Packet Data Channel
PDER	PDU Error Rate
PDF	Probability Density Function
PDTCH/F	Full rate Packet Data TCH
PDTCH/H	Half rate Packet Data TCH
PDU	Packet Data Unit

PER	Packet Error Rate
PF	Probability Function
P-GSM	Primary GSM 900 band
PHY	PHYsical layer
PLMN	Public Land Mobile Network
PS	Packet Switched
PSTN	Public Switched Telephone Network
PT	Priority Table
QCIF	Quarter Common Intermediate Format
QoS	Quality of Service
QPSK	Quaternary Phase Shift Keying
RACH	Random Access CHannel
RBCH	RLC Broadcast CHannel
RF	Radio Frequency
RFCH	Random access Feedback CHannel
R-GSM	Railways GSM 900 band
RLC	Radio Link Control
RNC	Radio Network Controller
RNG	Random Number Generator
RNS	Radio Network Subsystem
RR	Resource Request
RRC	Radio Resource Control
RRM	Radio Resource Management
RSS	Radio Subsystem
RT	Real-Time
SCH	Short CHannel
SDU	Service Data Unit
SF	Spreading Factor
SGSN	Serving GPRS Support Node
SIM	Subscriber Identity Module
SIP	Session Initiated Protocol
SIR	Signal to Interference Ratio
SIRT	Target SIR
SMG	Special Mobile Group

SMS	Short-Message Service
SRD	Short-Range Dependence
TB	Time Based
TBR	Target BitRate
TCH	Traffic CHannel
TCH/F	Full-rate TCH
TCH/F2.4	2.4 kbps Full-rate TCH for data
TCH/F4.8	4.8 kbps Full-rate TCH for data
TCH/F9.6	9.6 kbps Full-rate TCH for data
TCH/FS	Full-rate TCH for Speech
TCH/H	Half-rate TCH
TCH/H2.4	≤2.4 kbps Half-rate TCH for data
TCH/H4.8	4.8 kbps Half-rate TCH for data
TCH/HS	Half-rate TCH for Speech
TDD	Time Division Duplex
TDM	Time-Division Multiplexing
TDMA	Time Division Multiple Access
TFCI	Transport Format Combination Indicator
TPE	Traffic Processing Engine
TR	Technology Router
TRX	Transceiver
TSM	Traffic Source Model
TSMIM	Traffic Source Model Implementation Modules
TTI	Transmission Time Interval
UBCH	User Broadcast CHannel
UDCH	User Data CHannel
UDD	Unconstrained Delayed Data
UE	User Equipment
UL	UpLink
UMCH	User Multicast CHannel
UMTS	Universal Mobile Telecommunication System
Uni	Unidirectional
U-PDU	User PDU
U-SAP	User Service Access Point

USIM	UMTS SIM
UTGM	User Traffic Generation Module
UTRA	Universal Terrestrial Radio Access
UTRAN	UMTS Terrestrial Radio Access Network
UTV	User Traffic Vector
VAD	Voice Activity Detector
VBR	Variable BitRate
VLR	Visitor Location Register
VO	Video Object
VoIP	Voice over IP
VOL	Video Object Layer
VOP	Video Object Plane
WCDMA	Wideband CDMA
Wi-Fi	Wireless Fidelity
WLAN	Wireless Local Area Network
WSI	Wireless Strategic Initiative
WWW	World Wide Web

List of Symbols

α	Parameter for GBAR model
$\bar{\alpha}$	Average orthogonality factor in the cell
α_I	Parameter of the GOP GBAR model
α_p	Parameter of Pareto distribution
β	Shape parameter of Ga(β,λ)
β_k	Weibul CDF shape parameter
γ	Probability of transition from "OFF" state to "ON" state
$\Gamma(x)$	Gamma function
η_{1PD}	CMM 1st Priority Decisions Percentage
η_{1PD-T}	CMM 1st Priority Decisions Percentage on Technology
η_{1PD-B}	CMM 1st Priority Decisions Percentage on Bitrate/Quality
η_{BD}	CMM Bitrate/Quality Decision Percentage
η_D	Percentage of Discarded Packets/Frames
η_{DL}	Downlink load factor
η_{DL}'	Downlink power based load estimation
η_{DLmax}	Maximum allowed value for η_{DL}
η_{TD}	CMM Technology Decision Percentage
η_U	Normalised Capacity Usage
η_{UL}	Uplink load factor
η_{UL}'	Uplink power based load estimation
λ	Scale parameter of Ga(β,λ)
λ_c	Mean arrival rate of calls
λ_i	Parameter of the GOP GBAR model
λ_k	Inverse of Weibul CDF scale parameter
μ	Mean of the sample data

μ_B	Mean of the B-frame sample sequence
μ_{CD}	Inverse of the mean call duration time
μ_{Dd}	Mean value of D_d
μ_{Dpc}	Mean value of D_{pc}
μ_I	Mean of the I-frame sample sequence
μ_j	Expected frame size in state j
μ_k	Mean burst duration in state k
μ_{LN}	Natural log of the mean of the Lognormal PDF
μ_N	Mean of the Normal PDF
μ_{Nd}	Mean value of N_d
μ_{Npc}	Mean value of N_{PC}
μ_P	Mean of the P-frame sample sequence
μ_{Sd}	Mean packet size
ρ	Lag 1 autocorrelation coefficient of the sample data
ρ_B	Lag 1 autocorrelation coefficient of B-frame sizes
ρ_I	Lag 1 autocorrelation coefficient of I-frame sizes
ρ_i	Parameter of the GOP GBAR model
ρ_P	Lag 1 autocorrelation coefficient of P-frame sizes
σ	Probability of transition from "ON" state to "OFF" state
σ_B	Standard deviation of B-frame sample sequence
σ_I	Standard deviation of I-frame sample sequence
σ_{LN}	Natural log of the standard deviation of the Lognormal PDF
σ_N	Standard deviation of the Normal PDF
σ_P	Standard deviation of P-frame sample sequence
σ_{wj}	Standard deviation of white noise in state j
τ	Time slot duration
$\overline{\tau}_C$	Average Call Duration
τ_k	Burst duration in state k
A	Generated Traffic
A_n	Random variable associated to a stochastic process $\{X_n\}$ with marginal $Ga(\beta,\lambda)$ distribution
Be(p,q)	Beta distributed random variable

b_j	Feedback parameter in state j
B_n	Random variable associated to a stochastic process $\{X_n\}$ with marginal Ga(β,λ) distribution
D_{η_D}	Overall Discarded Packets/Frames Degradation
D_d	Interarrival time between packets within a packet call
$D_{\overline{R}_b}$	Overall Mean Bitrate Degradation
$D_{R_{CB}}$	Overall Call Blocking Rate Degradation
D_{pc}	Reading time between packet calls
E_b	Signal energy per bit
E[x]	Expected value of x
f_0	Initial frame size
$f_{max,j}$	Upper bound to adjust frame sizes in state j
$f_{min,j}$	Lower bound to adjust frame sizes in state j
Ga(β,λ)	Gamma distributed random variable
GBAR(β,λ)	GBAR process
I	Width of interval
i	Other cell to own cell interference ratio seen by the BS receiver
$\bar{i}$	Average ratio of other cell to own cell base station power received by user
I_{oth}	Received wideband interference power from inter-cell users
I_{own}	Received wideband interference power from intra-cell users
I_{total}	Received wideband interference power
$J(k)$	State at time step k
K	Parameter of Pareto distribution
M	Spacing between successive anchor VOPs
$M(k)$	Sum of the expected frame sizes up to frame k
M_{ps}	Maximum allowed packet size
N	Spacing between successive I VOPs
N_0	Noise spectral density
N_{1SB}	Total number of sessions using the 1st priority bitrate/quality
N_{1ST}	Total number of sessions routed through the 1st priority access technology
N_{AM}	Number of active MTs
N_{AS}	Number of Active Sessions

N_{BC}	Number of Blocked Calls
N_C	Total number of CS calls
N_{Co_U}	Equivalent number of used codes assuming SF=512
N_{Co}	Equivalent total number of available codes assuming SF=512
N_{CoSH}	Number of codes reserved for soft handover
N_d	Number of packets within a packet call
N_{DL}	Number of connections per cell
N_{DP}	Number of Discarded Packets/Frames
N_I	Number of intervals
n_{init}	Number of initial frames
N_{LCH}	Total number of available LCHs
N_{LCH_U}	Number of used LCHs
N_{oc}	Total number of occurrences
$n_{oc}[n]$	Number of occurrences within interval n
N_P	Total number of packets/frames
N_{PC}	Number of packet calls per session
N_S	Total number of sessions
N_s	Number of states
N_{SB_j}	Total number of sessions using bitrate/quality j
N_{ST_i}	Total number of sessions routed through a given access technology i (GSM/GPRS, UMTS or HIPERLAN/2)
N_{TS}	Total number of available TSs
N_{TS_U}	Number of used TSs
N_{UL}	Number of users per cell
P	State transition probability matrix
p	Parameter of $Be(p,q)$
$P(i,j)$	Transition probability from state i to state j
$p(x)$	PDF of x
$p[x]$	PF of x
P_a	Pareto distributed random variable
$p_B(x)$	Beta PDF
$P_E(t)$	Exponential CDF
$p_E(t)$	Exponential PDF

$p_G(x)$	Gamma PDF
$p_{Ge}[n]$	Geometric PF for the generation of n packet calls during a session
P_k	Probability that a packet is of size s_k
$p_{LN}(x)$	Lognormal PDF
P_{max}	Maximum BS transmission power
P_N	Background and receiver noise
$p_N(x)$	Normal PDF
$p_P(x)$	Pareto PDF
P_{PC}	Probability of generating one packet call
$p_{Po}[n,t]$	Poisson PF for generating n calls/sessions in a certain time interval t
P_{total}	Total downlink transmission power
$p_U(x)$	Uniform PDF
$P_W(x)$	Weibul CDF
$p_x[n]$	PDF/PF equivalent modified histogram
$p_x[x_n]$	Discretised theoretical PDF / Theoretical PF
q	Parameter of $Be(p,q)$
Q_k	Probability of selecting a new state s_k
$\overline{R}_b$	Mean Bitrate
$\overline{R}_{b_{max}}$	Maximum target mean bitrate for a specific application
R_{CB}	Call Blocking Rate
R_j	Bitrate of user j
R_{SM}	Ratio of Active Sessions per Active MTs
S_0	Initial state
S_d	Size of a packet
$S(k)$	Sum of the previous frame sizes up to frame k
s_k	Packet size generated in state k
t_{OFF}	Mean duration of silence period ("OFF" state)
t_{ON}	Mean duration of voice activity period ("ON" state)
v	Variance of the sample data
v_j	Activity factor of user j at physical layer
W	WCDMA chip rate
$W(k)$	Innovations for frame k size calculation
X_k	Size of the k^{th} frame

$X(k)$	Preliminary size of frame k
$X'(k)$	Adjusted size of frame k
x_{max}	Upper limit of the observation window
x_{min}	Lower limit of the observation window
X_n	Sample function from $\{X_n\}$
$\{X_n\}$	Stationary stochastic process
Z_{ik}	Sample function of a GBAR(α_i,ρ_i) process

Chapter 1

Introduction

This chapter gives a brief overview of the work. Before establishing work targets and original contributions, the scope and motivations are brought up. The current State-of-the-Art in relation to the scope of the work is also presented. At the end of the chapter, the work structure is provided.

1.1 Overview

During May 2003, the total number of mobile users has broken the 1.3 billion mark [CELL03]. This revolution has been implemented through a continuous evolution of standards and products, accommodating people's basic needs for communication and mobility. It started in the early '90s with the replacement of the analogue mobile network by the digital one, with the introduction of second generation (2G) mobile systems, and is continuing today with the deployment of the third generation (3G). From circuit-driven networks, we now enter the packet world through intermediate overlay networks, followed in years to come by all-IP networks.

3G mobile networks promise to support, besides the existing mobile telephony service, data services with higher bitrates and efficiency than 2G ones. However, driven by the enormous success of the Internet within the last ten years, with increasing data rates made available to the (wired) subscribers and the inherent deployment of new services, certain expectations emerged. Not only businessmen, who used to be the first customers in the beginning of mobile communications, but also private people demand for a wired-service like connection while on the move. This challenge will obviously not be coped with by sole 3G deployments.

An auspicious solution to this problem is seen in Wireless Local Area Networks (WLANs) and other wireless technologies arriving in the market, featuring higher data rates and different service offerings compared to 2G and 3G mobile networks. Due to the high bitrates and the free frequency licensing, WLAN technology is partly seen as a competitive technology to the 3G mobile network one. However, due to the rather small cell area, it seems utopian to implement a nationwide coverage based solely on WLAN deployment. This is why currently WLAN and 3G mobile networks are seen as complementary technologies. In fact, WLAN volunteers to be used in indoor environments or hot-spots, and 3G (and existing 2G) systems in broad outdoor environments. Convergence between these technologies will ensure the continuity of the service to the mobile users following the "anywhere, anytime" dogma.

Therefore, the convergence of mobile and wireless systems, combining the advantages of each specific access network technology, is foreseen as the solution for providing the expected (and new) services to users, *e.g.*, [FLOW03]. The aspired convergence of systems is currently referred to as systems 'beyond 3G' or '4G' (4^{th} Generation), *e.g.*, [SB3G01], [4GMF03]. 4G systems intend to provide an interconnection and interoperability between different technologies in order to ensure (seamless) handovers from one technology to another, thus, enabling a continuous service to the users. This can be, for example, based on choices made by the applications taking into account both application and user requirements, together with the capability of the different networks. It will require higher intelligence in the terminals to make best use of the different applications locally available. At the price of this complexity, the vision brought out by the Wireless Strategic Initiative (WSI) [WSI03] may become real, bridging all access technologies from fixed to satellite and from person to person to customised broadcast, Figure 1.1.

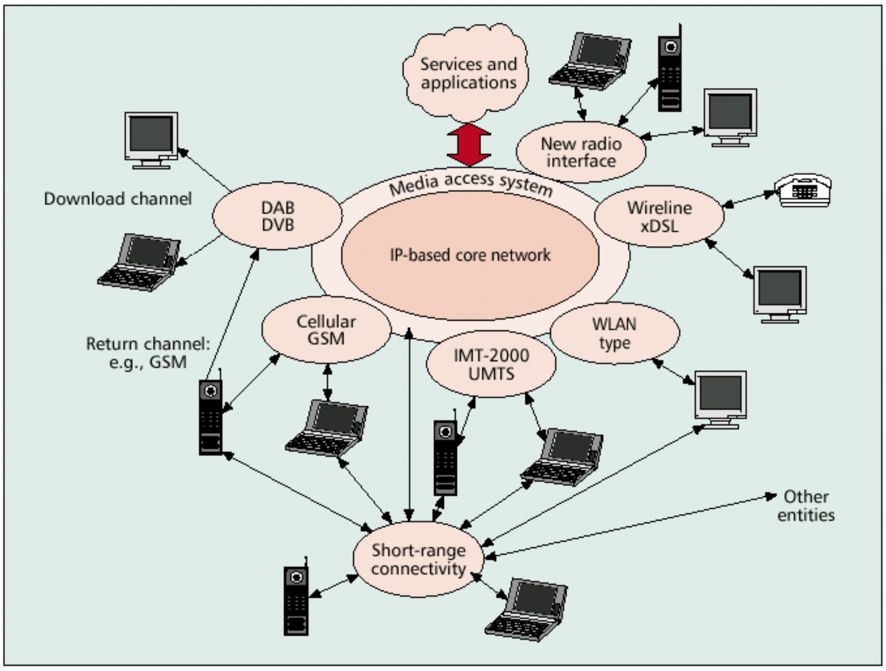

Figure 1.1. The multi-technology access network (extracted from [VLLX02]).

The convergence of systems will therefore provide mobile users with easy access to communications of any sort anywhere. This will have direct benefits in terms of greater accessibility and reduced location dependence to radio communications. In doing so, it will also provide the possibility for service providers to move beyond their traditional areas of operation. As well as greater choice, this will result in a higher quality of service and the integration of services around user needs.

The following two illustrative usage scenarios show possible examples of how the convergence of wireless systems can present clear benefits to the end users:

- Two teenagers are watching video clips on their mobiles while walking into and then through a very crowded airport with UMTS (Universal Mobile Telecommunication System) coverage. They sit down and carry on watching their video clips, but are now in a HIPERLAN/2 (HIgh PErformance Radio Local Area Network Type 2) hotspot, and because of traffic congestion, their devices switch automatically and seamlessly to HIPERLAN/2. One of the teenagers wants to compare prices for a skateboard, so he starts web browsing.
- A PDA (Personal Digital Assistant) user is uploading a report onto the company's Intranet, while walking slowly through a busy city centre which has UMTS coverage. Simultaneously, the user needs to make an important call on the PDA, but to make optimal use of the bearer services available, the call is made seamlessly over a GSM (Global System for Mobile Communications) network as he enters a museum.

Besides end users, operators will also benefit from the convergence of wireless systems (although in most cases a benefit to an operator will also ultimately be a benefit to a user and vice versa). Benefits primarily to the operator include:

- Increased capacity - the use of multiple standards removes the constraint that an operator must provide ubiquitous coverage using a single standard. For instance, introducing extra base stations into a network in order to provide full coverage may excessively increase interference levels, thereby reducing capacity. Additionally, in hot-spot areas, it may not be practical to increase the capacity, because there are insufficient spectrum or infrastructure sites. Convergence will allow the most appropriate standards to be used in the most appropriate locations.
- Increased coverage - the ability to offer the same service over different standards means that in an area where one standard could not normally be effectively deployed, coverage can still be achieved by use of an alternative standard.
- Reduced infrastructure costs – convergence will allow for a reduction in the number of different standards that "must" be served in a particular location.
- New services – the use of converged standards may offer totally new services, which can be used to provide new applications. Technological developments in recent years have already provided more flexibility for operators to offer services outside their traditional business, as a result of the convergence of different types of

communications media. Recent examples include: Internet services delivered to TV sets via digital TV decoders; Email and World Wide Web (WWW) access via mobile phones; Webcasting of radio and TV programs on the Internet; using the Internet for voice telephony via Voice over IP (VoIP).

- Increased numbers of users/usage - because the increased capacity and coverage mean services are more widely available, users are more likely to try to use them, leading to increased numbers of users and increased usage per user (increased profits).

Several research groups and projects are (or have been recently) focussed on system convergence aspects (at different levels), namely:

- ETSI (European Telecommunications Standards Institute) BRAN (Broadband Radio Access Networks) [BRAN03] and 3GPP (3rd Generation Partnership Project) [3GPP03], on HIPERLAN/2 and UMTS inter-working.
- Digital Video Broadcasting Project (DVBP) [DVBP03], on a "multiplatform" approach that effectively combines digital TV with UMTS and GPRS (General Packet Radio Service) cellular technologies.
- IST (Information Society Technologies)- WINE GLASS project [WINE03] (concluded 02/2002), on IP-based wireless mobile multimedia networking with UMTS and WLANs.
- IST-SUITED project [SUIT03] (concluded 06/2002), on the development of an integrated/converged system consisting of both satellite and terrestrial (UMTS, GPRS, WLAN) technologies.
- IST-BRAIN project [BRAI03] (concluded 04/2001), followed up by the IST-MIND project [MIND03] (concluded 11/2002), on broadband wireless multimedia services for IP over heterogeneous access networks, with a common IP core.
- IST-FLOWS project [FLOW03], on the convergence and simultaneous use of a variety of wireless systems/standards (*e.g.*, GSM, UMTS, HIPERLAN/2), based on the connection of different wireless access points (from different standards) to a common access network based on IP.
- IST-TRUST project [TRUS03] (concluded 03/2002), followed up by the IST-SCOUT project [SCOU03], on reconfigurability in radio systems and networks, with particular attention on the terminal (users' requirements), and considering IP

based radio access and core networks.

- IST-MOBIVAS project [MOBI03] (concluded 06/2002), on the development of a middleware architecture that enables the creation of an open, dynamic market environment for mobile services, provided over reconfigurable networks and systems.
- IST-DRIVE project [DRIV03] (concluded 03/2003), followed up by the IST-OVERDRIVE project [OVER03], addressing the convergence of cellular and broadcast networks (GSM, GPRS, UMTS, DAB (Digital Audio Broadcasting), DVB-T (Digital Video Broadcast – Terrestrial)) to lay the foundations for innovative IP-based multimedia services. The convergence between radio access networks is made possible by the application of an intelligent backbone network, a novel dynamic spectrum allocation mechanism, and flexible and adaptive applications.
- IST-MobyDICK project [MOBY03], defining an architecture for wireless Internet access by developing new mechanisms for seamless handover, QoS (Quality of Service) support after and during handover, AAA (Authentication, Authorisation and Accounting), and charging, over a heterogeneous network infrastructure.
- IST-CISMUNDUS project [CISM03], on the convergence of IP-based services for mobile users and networks in DVB-T and UMTS systems.
- IST-ANWIRE project [ANWI03], a thematic network that organises and coordinates parallel actions in key research areas of Wireless Internet and Reconfigurability, in order to encompass research activities towards the design of a fully integrated system.

There are already a few initial developments (by telecom and terminal manufacturers) of systems following a convergence approach. For example, Nomadix [NOMA03] and ipUnplugged [IPUN03] have announced recently complete interoperability between their solutions for operators to enable secure seamless roaming across Wi-Fi (Wireless Fidelity - IEEE 802.11b or 802.11a WLAN networks), GPRS, and UMTS while maintaining a single billing solution; Cisco Systems [CISC03] has recently presented a wireless phone that operates on Wi-Fi (802.11b) networks; Nokia [NOKI03] has launched multimode radio card for laptop computers that enables network access through GPRS, HSCSD (High Speed Circuit Switched Data), and WLAN networks; Texas Instruments [TEXA03] has developed a device that offers simultaneous networking over 802.11b and Bluetooth wireless protocols and voice and data calls over a GPRS-enhanced GSM radio. Besides these examples, several terminal manufacturers have recently presented multi-band and multi-mode terminal devices (*e.g.*, PCMCIA cards), including for example GPRS, UMTS and WLAN capabilities. In these terminals, the system typically operates using only one standard at a time, with the objective of offering the consumer access to existing mobile services, but having an improved roaming capability.

1.2 Motivation and Contents

To date, convergence is mainly considered at the core network level, with the use of IP in this context. However, not much attention has been given to how the various wireless standards can be converged at the consumer terminal, if multiple standards are available to the user from a common core network. Under this context, important issues arise, like determining how services map onto certain standards (or even split between standards), and how can this be made to be seamless and appear as a simultaneous connection to the user. Another important issue, still to be analysed, is the impact of convergence on the several access networks performances (and vice-versa).

The current thesis is precisely motivated by this vision of future wireless communication systems where there will be a convergence of wireless standards and an enhancement of services. In particular, the objective is to focus on the capacity of the radio interface of such converged multi-standard systems, and study (by means of a simulation platform developed for this purpose) the overall performance obtained for different scenarios (convergence benefits), using different approaches, namely:

- Analysis of the impact of users traffic variations;
- Analysis of the impact of the number of users in specific scenarios corresponding to different available access technologies;
- Analysis of the impact of the availability of different access technologies for a particular number of users.

A key concept adopted in the present work is a common core network based on IP, to which a variety of wireless access points/technologies will be connected. The selected access technologies that will be considered throughout this thesis are GSM/GPRS, UMTS or HIPERLAN/2. The combination of these three access technologies provides a broad data rate support, a wide range of mobility, and from the application point of view, voice, data and multimedia support, Figure 1.2. Through convergence at the air interface of these wireless standards, a user with a single Mobile Terminal (MT) can be connected simultaneously (and efficiently) to the three kinds of access points. The overall network structure under consideration for the development of the present work is depicted in Figure 1.3.

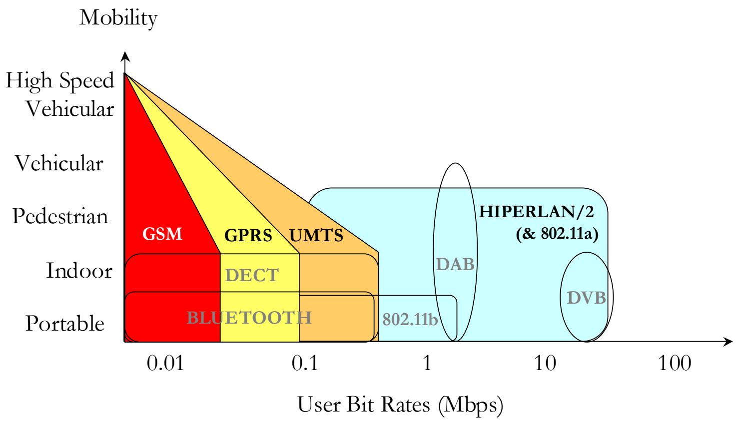

Figure 1.2. Support for user data rate versus mobility as provided by the GSM/GPRS, UMTS and HIPERLAN/2 (adapted from [ACGN02]).

A very important step towards the development of the present work is the selection of a representative and diversified (in characteristics) set of services and applications, which can be supported by the considered systems (alone or combined), and their detailed characterisation. For simulation purposes, traffic source models, as well as the definition of the associated generation and duration processes, will be necessary for each of the considered applications. Another crucial task is to establish a mapping of the considered services and applications onto the available access technologies (GSM/GPRS, UMTS and HIPERLAN/2).

In the development of the present work, a new entity has been introduced – the Convergence Manager (CM) (Figure 1.3). This element is basically responsible for the implementation of all convergence functionalities in the overall system. The CM is represented by a functional entity in the Access (or Core) Network and a corresponding one in each associated MT. The placement of the CM is a crucial issue for the architectural design of a converged multi-technology network. By locating the CM functionality in different places of the network, different functions and different layers of optimisation may be achieved. Each possibility presents advantages and disadvantages, when different aspects are taken into consideration, *e.g.*, architecture, complexity of implementation, end users and network operator's points of view.

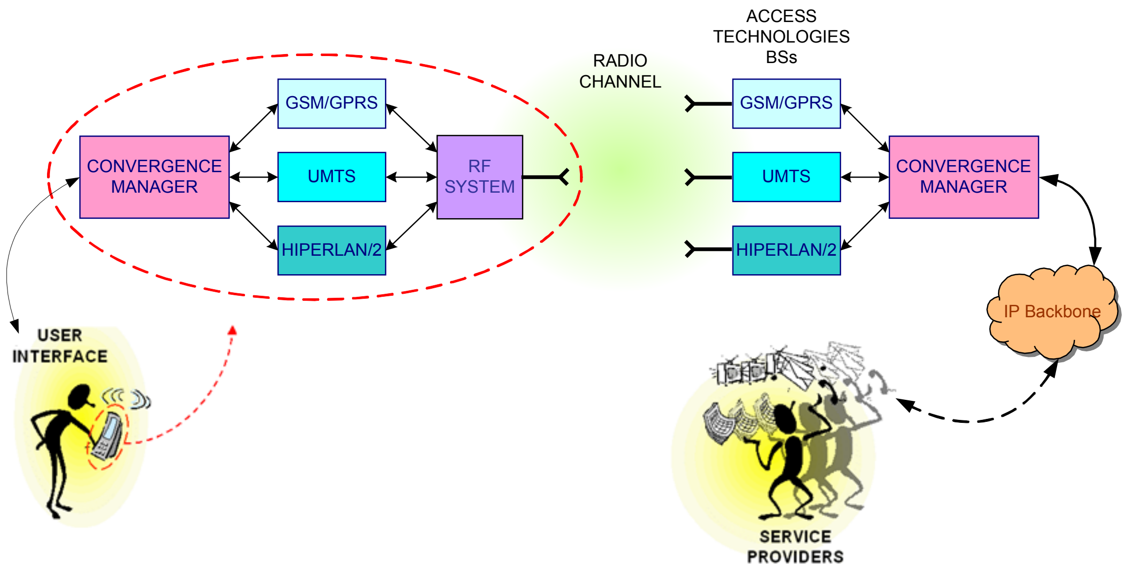

Figure 1.3. Network structure under consideration.

For instance, an apparently optimum solution from a radio access technology performance point of view, would be to locate the network CM functionality on a multi mode BS. This would allow the MTs to access different technologies simultaneously while optimising the radio access technology choice, the network route (BS used) and the radio interface physical layer parameters (*i.a.*, modulation schemes and power control). However one can easily identify serious limitations in this solution: it places some significant new requirements on the network architecture, since the different systems/technologies have different cell sizes (*e.g.*, in the case of a simultaneous usage of UMTS and WLAN, either the UMTS cell areas have to overlap dramatically to provide full WLAN coverage or the WLAN areas can not be designed continuously); another drawback is that in cases of handovers from one BS to another the state of the convergence manager has also to be transferred, and therefore, standardisation effort for the interworking protocol would have to be considered. Besides these technical limitations, another limiting factor for this approach is that only one operator can be involved, thus, a cooperation of different business entities is implicitly impossible in this approach.

On the other hand, locating the CM functionality deeper into the network, for instance behind the gateways of each access technology, would require minor changes or enhancements at the network nodes. In this case, the CM would split the sessions into media streams or direct sessions of specific characteristics onto the best-suited access network. For example, for a file download the CM could select a WLAN access network, while for a speech call the CM could direct it to a 2G network. In all cases the network entities of the various access networks will not need extra functionalities. However, in this case a lower level of convergence benefits would be achieved.

Another possible solution could include, for example, a hierarchical convergence manager functionality, *i.e.*, the CM functionality could be split and located in several places of the network. For instance, there could be a CM in the core network splitting streams onto different standards, in addition to a CM outside the core network splitting sessions onto the networks from different operators. This combination may be possible, and in some cases it may be business profitable and beneficial for the users. However, this type of approach would demand more upgrade work than in a single CM location scenario.

The CM concept adopted in the development of the present work considers a session based access technology decision process, which goes in line with the second CM location possibility previously discussed (*i.e.*, behind the gateways of each access technology). Another important issue to consider is the intrinsic architectural design of such an element as the CM, strongly dependent of its location in the network and MT. Although this is a serious factor to take into account for the implementation of such converged networks, it falls outside the scope of the current Thesis, which focuses mainly on the radio interface capacity performance.

The idea of the CM is to obtain the maximum flexibility among users and networks. The CM allocates the packets coming from different users using various services to different networks/access technologies. The CM may use different perspectives/strategies by considering parameters such as traffic loads on different networks, channel status between the BS and MTs, different user's priorities and service requirements, etc.. By applying different perspectives/strategies, the different parameters are weighted differently and different benefits are obtained. Examples of different perspectives/strategies of the CM can be:

- QoS oriented - the user can achieve better quality of service because the communications link will always operate with the most appropriate standards or combinations between different standards. Parameters regarding different QoS can include data rate, PER (Packet Error Rate) or FER (Frame Erasure Rate), and system delay when setting up connections.
- Capacity oriented - this perspective/strategy may be important for operators since

it minimises the amount of network infrastructure that needs to be deployed. Further, the use of multiple standards, removes the constraint that an operator must provide ubiquitous coverage using a single standard. The parameters affecting the capacity oriented policy can be the average session capacity occupation per standard, available capacity per standard per unit of time, outage probability, etc..

- Price/cost oriented - there will be different standards or combinations of standards offered to the user. Since different standards have different prices when carrying data packets, the user will consider the price when choosing standards. This may then be implemented in the CM based on users' preferences.
- Fairness/Priority oriented - in most cases, all users are served fairly. Nevertheless, it is possible to implement priority schemes in the CM to serve special users with higher priority.

In the present work, one will consider a unique CM perspective/strategy based on a highest throughput / best system approach. This approach can be seen as a combination of the above-mentioned ones (excluding the price/cost oriented strategy). The highest throughput concept means that for data and video applications, the highest available/possible transmission bitrate is selected based on the current networks' loads. In respect to the best system approach, priority is given to the most appropriate application-oriented access technology (*e.g.*, data applications first priority – WLAN; Voice Calls first priority – GSM).

This new functionality, the CM, which allows the convergence of wireless standards and an enhancement of the service offering performance, is the main original contribution of the present work. This concept has been implemented in a newly developed simulation platform, together with specific traffic source models for several applications and radio interface (capacity) models for GSM/GPRS, UMTS and HIPERLAN/2. The developed simulator, which presents high flexibility and ease of upgrade, will enable the evaluation of the benefits of convergence of mobile and wireless communication systems from a capacity/traffic point of view, taking into account the possibility of analysing different convergence perspectives/strategies.

Of course, some approximations were introduced in the development of the simulator, in order to enable to perform this analysis in the context of a Master Thesis. Nevertheless, there was always the concern that the taken approximations keep the essential characteristics of each standard under consideration. These approximations will be identified along the text.

This document is composed of five chapters, besides the current one, and three annexes. The following chapter is dedicated to the overview of the GSM/GPRS, UMTS and HIPERLAN/2 systems, mainly focussing on the capacity aspects of the radio interfaces. Chapter 3 provides a description of the services and applications that are considered in the present work. A detailed characterisation and parameterisation of traffic source models, as well as of generation and duration processes, for each considered application is also presented. At the end of the chapter, a mapping of the access technologies (GSM/GPRS, UMTS and HIPERLAN/2) versus the services/applications and traffic source models is performed. Chapter 4 presents a functional description of the simulator platform developed within the work scope of this thesis, with the main objective of allowing the evaluation of the benefits of convergence of mobile and wireless systems from a capacity/traffic point of view. In Chapter 5, the results of simulations for several different scenarios are analysed following three different approaches; the simulated scenarios are characterised according to three main components: technologies, users and applications. Conclusions and further suggestions of work to be done are drawn in the final chapter. Annex 1 includes a statistical validation of the random number generators used in the simulator platform, while Annex 2 and Annex 3 present, as a complement to Chapter 5, some additional results from simulations.

Chapter 2

Technologies Overview

This chapter provides an overview of the GSM/GPRS, UMTS and HIPERLAN/2 systems, mainly focussing on the capacity aspects of the radio interfaces.

2.1 The GSM/GPRS system

2.1.1 Introduction

In June 1982, CEPT (European Conference of Postal and Telecommunications Administrations) decided to develop and standardise a Pan-European cellular mobile radio network [Walk00]. The aim was for the new system to operate in the 900 MHz frequency band allocated to land mobile radio. A working group, called *Group Spéciale Mobile* (GSM), was set up under the direction of CEPT, to develop and standardise the so-called GSM mobile radio system. The GSM objectives for its Public Land Mobile Network (PLMN) included offering [Walk00]: a broad set of speech and data services, cross-border system access for all mobile phone users, automatic roaming and handover, highly efficient use of frequency spectrum, supplier-independence, etc..

By 1987, comprehensive guidelines for the new digital mobile radio system had already been established by the GSM group. By signing the *Memorandum of Understanding* (MoU) *on the Introduction of the Pan-European Digital Mobile Communication Service* on 7 September 1987, the 13 participating countries confirmed their commitment to introducing mobile radio based on the recommendations of the GSM. Later, in March 1989, the GSM working party was taken over by ETSI, and since 1991 it has been called the *Special Mobile Group* (SMG). Today the abbreviation GSM stands for *Global System for Mobile Communications*, thereby underlining its claim as a worldwide standard. In the meantime, all the European countries, as well as a large number of other countries in the world, have signed the GSM-MoU and have developed or will be developing mobile radio systems in their countries based on the GSM recommendations.

GSM is regarded as an important advance compared with predecessor systems, and is considered to be representative of so-called 2nd Generation (2G) systems. Along with important technological advances (particularly the introduction of digital transmission technology), the standardisation of the interfaces between subsystems in GSM has provided manufacturers and network operators flexibility in their development work and configurations.

In GSM, besides the normal voice service, data transfer is made possible using Circuit-Switched Data (CSD), which offers throughput up to 14.4 kbps. This limitation led to the standardisation of the High Speed Circuit Switched Data (HSCSD) and General Packet Radio Service (GPRS). HSCSD enables higher rates (up to 57.6 kbps), but like CSD is circuit-based; therefore, it is inherently inefficient for bursty traffic, continuously using several radio channels (up to four). Only around 30 operators have introduced HSCSD so far [VLLX02]; most operators use GPRS instead. This technology increases the data rates of existing GSM networks, allowing transport of packet-based data. New GPRS handsets are able to transfer data at rates much higher than the 9.6 or 14.4 kbps currently available to mobile-phone users, and under ideal circumstances, GPRS will support rates of 171.2 kbps, surpassing ISDN (Integrated Services Digital Network) access rates. However, a more realistic data rate for early network deployments is probably around 40 kbps using one Uplink (UL) and three Downlink (DL) timeslots. Unlike Circuit-Switched (CS) 2G technologies, GPRS is an "always-on" service. It will allow GSM operators (and others) to provide high speed Internet access at a reasonable cost, by billing mobile-phone users for the amount of data they transfer, rather than for the length of time they are connected to the network.

In GPRS, the connection to the PLMN is based on IP, and on the air interface, the resources being assigned to the terminal are on a per-IP packet basis [Yaco02]. GPRS supports PS (Packet Switched) services and allows services based on QoS requirements, such as priority, reliability and bitrate.

GSM networks, either in the original GSM conception or as an evolution of it, are currently spread worldwide and are unanimously considered a very successful project.

2.1.2 System architecture

The GSM/GPRS system can be divided into the following three subsystems [Walk00]: Radio SubSystem (RSS), Network and Switching SubSystem (NSS) and Operation SubSystem (OSS).

These subsystems and their components are represented in the simplified version of the functional architecture in Figure 2.1.

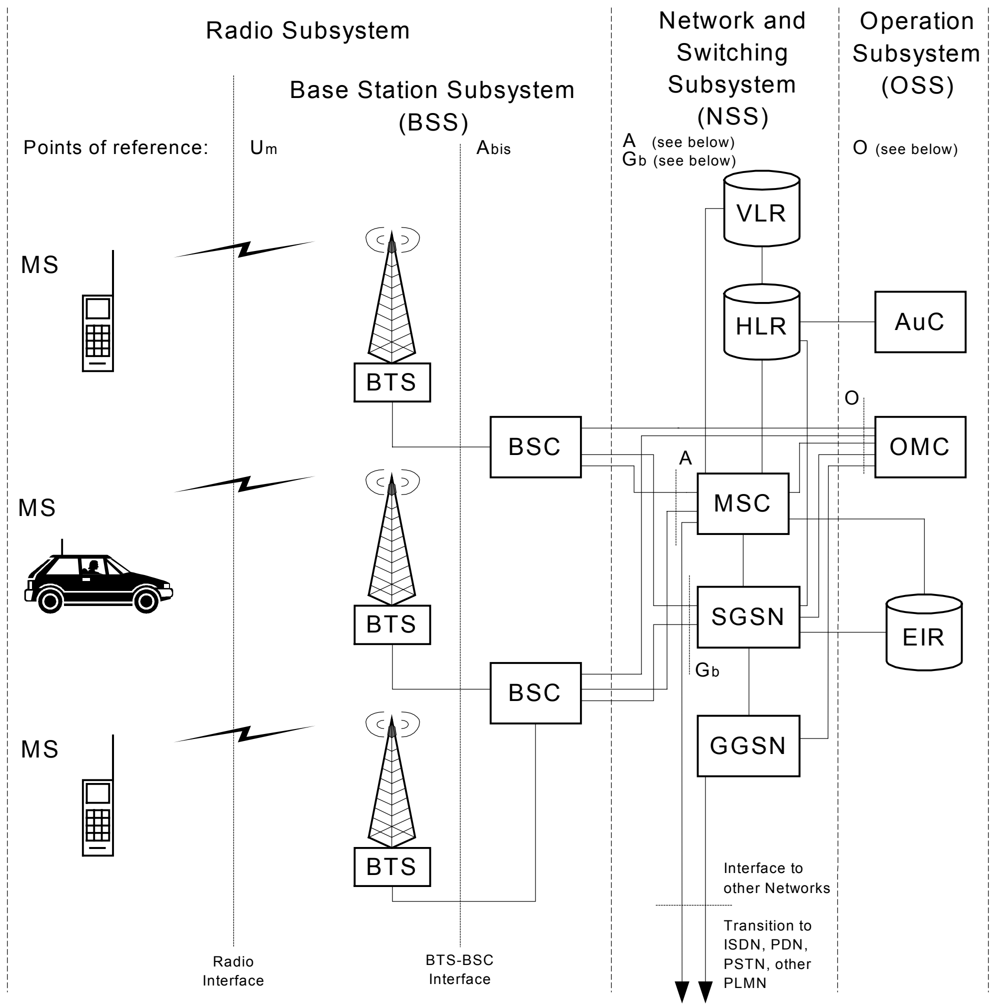

Figure 2.1. Functional architecture of the GSM/GPRS mobile radio network (based on [Walk00]).

The RSS is constituted by Mobile Stations (MS) and the Base Station Subsystem (BSS). The term MS refers to all the physical equipment of a PLMN user. Both CS and PS facilities may be provided within the MS.

The BSS comprises all the radio-related functions of the GSM network. The BSS is functionally and physically separated into two components: the Base Transceiver Station (BTS) and the Base Station Controller (BSC). The BTS comprises the transmitting and receiving facilities, including antennas and all the signalling related to the radio interface. The BSC is responsible for the management of the radio interface trough the BTS, namely for the reservation and the release of radio channels (radio resource management), as well as handover management and frequency hopping control.

The NSS carries out the switching procedures, packet routing and transfer within the PLMN, and the manipulation of the databases for mobility management of the subscribers. Its functions include coordination of call setup, paging, resource allocation, location registration, encryption, interfacing with other networks, handover control, billing, synchronisation, and others [Yaco02]. The NSS components include the Mobile Services Switching Centre (MSC), the Gateway GPRS Support Node (GGSN), the Serving GPRS Support Node (SGSN), the Home Location Register (HLR) and the Visitor Location Register (VLR).

The OSS performs operation and maintenance functions, namely: subscription management, network operation and maintenance, and mobile equipment management [Walk00]. The following network elements are part of the OSS: Operation and Maintenance Centre (OMC), Authentication Centre (AuC) and Equipment Identity Register (EIR).

2.1.3 Radio interface capacity aspects

The radio interface is located between the MS and the rest of the GSM/GPRS network. Physically, the information flow takes place between the MS and the BTS. But, viewed logically, the MSs are communicating with the BSC, MSC and the SGSN. The gross transmission rate over the radio interface is 270.833 kbps.

Currently there are several frequency bands defined for the operation of GSM, Table 2.1. Operators may implement networks that operate on a combination of these frequency bands to support multi band MSs.

GSM 900 is the original GSM cellular network initially conceived to serve large areas (macro cells) and to operate with high power terminals. GSM 1800 and GSM 1900 incorporate the personal communication service concept. GSM 1800 is designed to operate in Europe and GSM 1900 is designed to operate in America, and both comprise low power terminals and serve small areas (micro cells) [Yaco02]. Other operating bands have also been specified, like E-GSM and R-GSM. In E-GSM, the original GSM 900 operating band is extended and lower power terminals and smaller serving areas (micro cells) are specified [Yaco02].

Band	Uplink [MHz]	Downlink [MHz]	Duplex interval [MHz]
GSM 450	450.4-457.6	460.4-467.6	10
GSM 480	478.8-486	488.8-496	10
GSM 850	824-849	869-894	45
Primary GSM 900 or P-GSM	890-915	935-960	45
Extended GSM 900 or E-GSM	880-915	925-960	45
Railways GSM 900 or R-GSM	876-915	921-960	45
GSM 1800	1710-1785	1805-1880	95
GSM 1900	1850-1910	1930-1990	80

Table 2.1. GSM frequency bands (based on [3GPP02b]).

The GSM frequency bands are divided into 200 kHz bandwidth channels, providing a certain amount of radio channels, each for transmitting and receiving operations. Refer to [3GPP02b] for a detailed overview of the GSM channel arrangements in each frequency band.

In the GSM recommendations, a combination of Frequency-Division Multiplexing (FDM) and Time-Division Multiplexing (TDM) has been standardised, providing multiple access by MSs to theses systems, *i.e.*, Frequency Division Multiple Access (FDMA) and Time Division Multiple Access (TDMA).

Figure 2.2 shows how a physical channel is produced through a combination of FDMA and TDMA (see channel 0 on frequency $F_{(n+1)}$). With the TDMA method, a carrier frequency is divided into eight physical channels, *i.e.*, Time Slots (TSs), Figure 2.3.

A physical channel is therefore characterised by its carrier frequency and the TS available to it, which recurs every frame with a duration of 4.615 ms. Each TS has a length corresponding to the duration of 156.25 bit or 0.577 ms. A slot is used by a burst with a length of 148 bit, which, corresponding to the guard time, is 8.25 bit shorter in duration than the slots to avoid overlapping with other bursts. Data is transmitted in bursts, and if messages are longer than a burst, they are split up among several bursts and then transmitted.

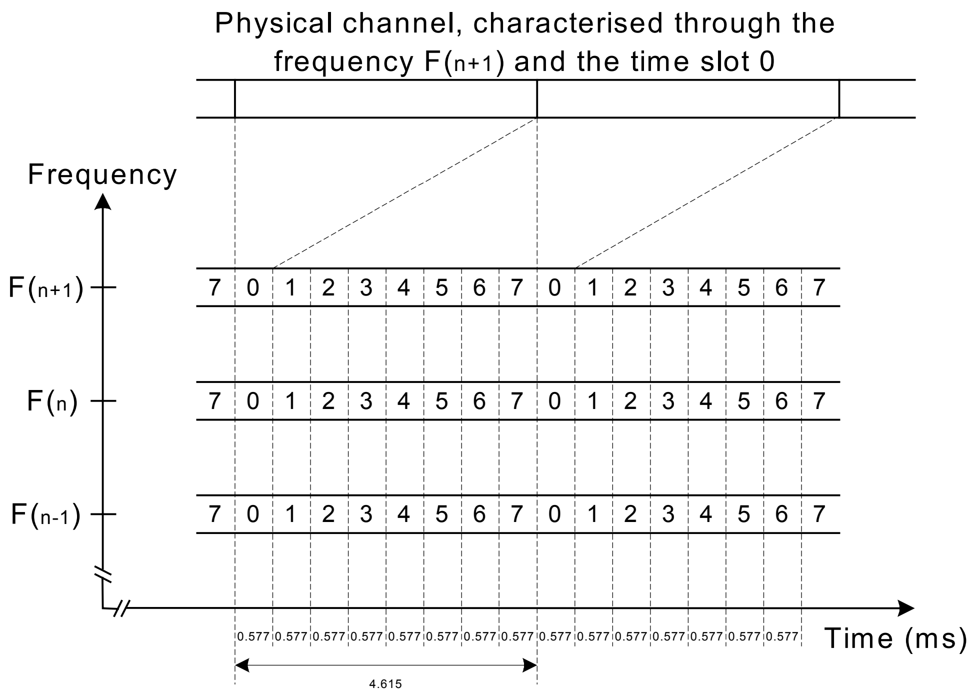

Figure 2.2. Physical channels using FDMA and TDMA (extracted from [Walk00]).

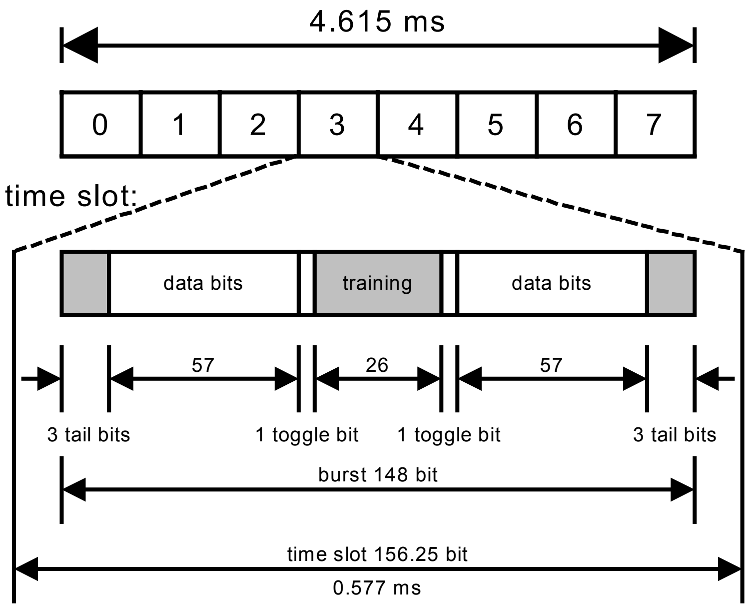

Figure 2.3. Structure of a TDMA frame (extracted from [Walk00]).

In order for the MSs not to transmit and receive at the same time, the UL TDMA frames are transmitted with a delay of 3 time slots in relation to the DL ones.

The tasks performed in a GSM/GPRS platform are supported by a number of functional channels, the logical channels. Logical channels occur through the allocation of TSs to physical channels. Consequently the data of a logical channel is transmitted in the corresponding TSs of the physical channel. During this process, logical channels can occupy a part of the physical channel or even the entire channel.

GSM recommendations [3GPP01a] define several logical channels, dividing them into two main groups: Traffic Channels (TCHs), which convey payload information (speech, data), and Control Channels (CCHs), which carry overhead (control) information.

The TCHs are intended to carry two types of user information streams: encoded speech and data, therefore, TCHs can be assigned either to CS calls or to PS data. Different transmission capacities are required depending on the type of service used (*e.g.*, voice transmission, Short-Message Service (SMS), data transfer, facsimile). In Table 2.2, a list of GSM traffic channels is presented.

Traffic Channel	**Abbreviation**
Full-rate TCH for speech	TCH/FS
Half-rate TCH for speech	TCH/HS
9.6 kbps full-rate TCH for data	TCH/F9.6
4.8 kbps full-rate TCH for data	TCH/F4.8
4.8 kbps half-rate TCH for data	TCH/H4.8
≤2.4 kbps full-rate TCH for data	TCH/F2.4
≤2.4 kbps half-rate TCH for data	TCH/H2.4
Full rate packet data TCH	PDTCH/F
Half rate packet data TCH	PDTCH/H

Table 2.2. GSM traffic channels (based on [Walk00] and [3GPP01a]).

Other traffic channels, such as Cell Broadcast Traffic CHannels (CBCH) have also been recently specified [3GPP01a].

Control information, which is used for signalling and for system control (not passed to the subscribers), is transmitted over CCHs. The CCHs offer the MSs a packet-oriented continuous signalling service enabling them within the PLMN to receive messages from the BSs and to send messages to the BSs at any time.

The logical channels are mapped onto physical channels using the technique of "multiframing". A multiframe is a set of some fixed number of TDMA frames that are together assigned a functionality. A distinction can be made between multiframes of three different lengths

[3GPP01a]: 26-frame, 51-frame and 52-frame multiframes.

Fifty-one of the 26-frame multiframes and 26 of the 51-frame multiframes are combined into a superframe, and 2048 superframes produce a hyperframe (see Figure 2.4). It takes almost 3.5 hours to transmit a hyperframe.

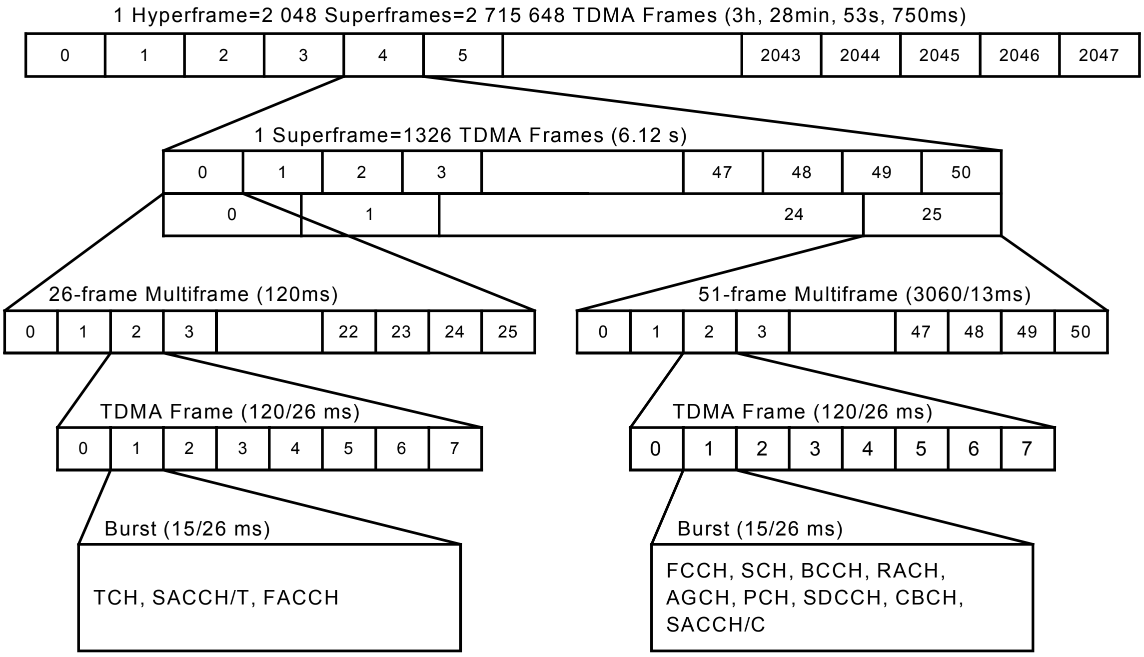

Figure 2.4. Structure of the GSM TDMA frames, multiframes, superframes and hyperframes (extracted from [Walk00]).

The BSS provides a cell with a set of logical channels occupying several physical channels. On the basis of the anticipated traffic load of a cell, the network operator establishes a particular channel configuration. Each individual transceiver (TRX) can offer different channel combinations in each TS. The TS is identified by a Time slot Number (TN). Three common combinations for conventional GSM are [Walk00]:

- Low-capacity cell with one TRX:
  - TN 0: FCCH + SCH + BCCH + CCCH + SDCCH/4(0,1,2,3) + SACCH/C4(0,1,2,3)
  - TN 1 to 7: TCH/F + FACCH/F + SACCH/TF
- Medium-capacity cell with four TRXs:
  - Once on TN 0: FCCH + SCH + BCCH + CCCH
  - Twice (on TN 2 and TN 4): SDCCH/8 + SACCH/8

– 29 times: TCH/f + FACCH/F + SACCH/TF

- High-capacity cell with 12 TRXs:

  – Once on TN 0: FCCH + SCH + BCCH + CCCH

  – Once on TN 2: BCCH + CCCH

  – Once on TN 4: BCCH + CCCH

  – Once on TN 6: BCCH + CCCH

  – 5 times: SDCCCH/8 + SACCH/8

  – 87 times: TCH/F + FACCH/F + SACCH/TF

As for conventional GSM, GPRS must allocate resources for signalling and traffic control. Some of the GPRS specific control channels can be multiplexed with the conventional GSM channels by using different possible channel configurations.

After successful synchronisation, the MS is informed through the system information of the BCCH (Broadcast Control Channel), which channel combinations on which physical channels are being offered to it by the BSS. Depending on its current operating state (idle state or dedicated state), it uses a particular subset from this offering of channels.

In conventional GSM, a channel is permanently allocated for a particular user during the entire call period (whether data is transmitted or not). In contrast to this, in GPRS channels are only allocated when data packets are sent or received, and they are released after the transmission. For bursty traffic this results in a much more efficient usage of the scarce radio resources. With this principle, multiple users can share one physical channel.

The channel allocation in GPRS is different from the original GSM. GPRS allows a single MS to transmit on multiple time slots of the same TDMA frame (multislot operation)[1]. This results in a very flexible channel allocation: one to eight time slots per TDMA frame can be allocated to one MS. Moreover, UL and DL are allocated separately, which efficiently supports asymmetric data traffic (*e.g.*, Web browsing).

A cell supporting GPRS may allocate physical channels for GPRS traffic, such a physical channel being denoted as Packet Data CHannel (PDCH). PDCHs are taken from the common pool of all

[1] Multislot operation is also supported in GSM for circuit switched data – HSCSD service.

channels available in the cell, thus, the radio resources of a cell are shared by all GPRS and non-GPRS MSs located in this cell. The mapping of physical channels onto either PS (GPRS) or CS (conventional GSM) services can be performed dynamically (capacity on demand principle [3GPP02a]), depending on the current traffic load, the priority of the service, and the multislot class. A network operator can decide to dedicate permanently or temporarily some physical resources (*i.e.*, PDCHs) for GPRS traffic. A load supervision procedure monitors the load of the PDCHs in the cell and, according to the current demand, the number of channels allocated for GPRS (*i.e.*, the number of PDCHs) can be changed. Physical channels not currently in use by conventional GSM can be allocated as PDCHs to increase the QoS for GPRS. When there is a resource demand for services with higher priority, PDCHs can be de-allocated.

Various radio channel coding schemes are specified for GPRS, depending on the quality of the channel, allowing bitrates from 9 to more than 150 kbps per user, Table 2.3. Under very bad channel conditions, CS-1 may be used, obtaining a data rate of only 9.05 kbps per time slot, but with a very reliable coding. Under good channel conditions, transmission is made without convolutional coding and a data rate of 21.4 kbps per time slot is achieved. With eight time slots, a maximum data rate of 171.2 kbps can be achieved; in practice, multiple users share the time slots, thus, a much lower bitrate is available to the individual user. For example, approximately 40 kbps per user will be achieved, if three users share eight time slots and CS-3 is employed [BeVE99]. CS-1 is used for the coding of the signalling channels. Nevertheless, if the overhead of all protocols and the phenomena involved in the transmission are accounted for, then the actual throughputs will be substantially smaller than those presented in Table 2.3 (less than 70 % [Yaco02]).

Coding scheme	**Code rate**	**Data rate per TS [kbps]**
CS-1	½	9.05
CS-2	~2/3	13.4
CS-3	~3/4	15.6
CS-4	1	21.4

Table 2.3. Channel coding schemes in GPRS (extracted from [BeVE99]).

2.2 The UMTS system (FDD mode)

2.2.1 Introduction

The conception of 3G wireless systems has been embodied by the International Mobile Telecommunications-2000 (IMT-2000). IMT-2000 standards and specifications have been developed by various standards organisations worldwide under the auspices of the International Telecommunications Union (ITU). 3G telecommunication services target both mobile and fixed users, with the access provided via a wireless link. A wide and ambitious range of user sectors, radio technology, radio coverage, and user equipment is covered by IMT-2000 [Yaco02].

IMT-2000 provides access to a wide range of telecommunications services via several radio interfaces, including both terrestrial and satellite component radio interfaces. The IMT-2000 CDMA Direct Spread and IMT-2000 CDMA TDD (Time Division Duplex) radio interfaces, both using Direct-Sequence Code Division Multiple Access (DS-CDMA), have been developed with the aim of interconnecting with the evolved GSM core network.

The IMT-2000 radio interface for DS-CDMA is referred to as Universal Terrestrial Radio Access (UTRA) or Wideband CDMA (WCDMA). Within the IMT-2000 framework, the original target of the 3G process was a single common global air interface. 3G systems are closer to this target than were 2G systems: the same air interface – WCDMA – is to be used in Europe and Asia, including Japan and Korea, using frequency bands around 2 GHz.

WCDMA supports two basic modes of operation: FDD and TDD. In the FDD mode, separate 5 MHz carrier frequencies are used for the UL and DL respectively, whereas in TDD only one 5 MHz is time-shared between UL and DL. The FDD mode is the main mode to be used in UTRA, while the TDD one was added in order to leverage the basic WCDMA system also for the unpaired spectrum allocations of the ITU for the IMT-2000 systems.

In UTRA, the radio interfaces are defined in such a way that a wide range of services including speech, data, and multimedia can be simultaneously used by a subscriber and multiplexed on a single carrier. Therefore, CS and PS services are efficiently supported, and real-time and non-real-time operations employing transparent or non transparent data transport are specified. The QoS is an important feature in UTRA, and it is specified to be adjusted in terms of parameters such as delay, bit error rate, frame error rate, and others.

The integration of user equipment, UTRA and a core network results in a 3G system known as the Universal Mobile Telecommunications System (UMTS).

In the next sections, an overview of the UMTS system and respective air interface is provided, focusing mainly on UTRA FDD mode.

2.2.2 System architecture

UMTS consists of a number of logical network elements, each having a defined functionality. In the standards, network elements are defined at the logical level, but this quite often results in a similar physical implementation, especially since there are a number of standardised open interfaces [HoTo00]. Functionally, the network elements are grouped into UMTS Terrestrial Radio Access Network (UTRAN), which handles all radio-related functionalities, and the Core Network (CN), which is responsible for switching and routing calls and data connections to external networks. To complete the system, the User Equipment (UE) that interfaces with the user and the radio interface is defined. Figure 2.5 shows elements in an UMTS PLMN and, in order to illustrate the connections, also external networks.

From a specification and standardisation point of view, both the UE and the UTRAN consist of completely new protocols, the design of which is based on the needs of the new WCDMA radio technology. On the contrary, the definition of CN is adopted from GSM/GPRS. This gives the system with new radio technology a global base of known CN technology that accelerates and facilitates its introduction, and enables such competitive advantages as global roaming.

The UE consists of the Mobile Equipment (ME) and the UMTS Subscriber Identity Module (USIM).

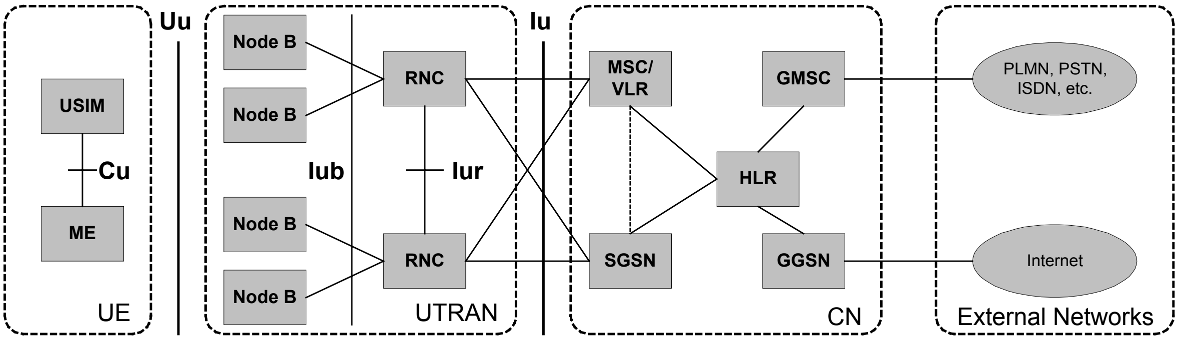

Figure 2.5. Network Elements in an UMTS PLMN (extracted from [HoTo00]).

The UTRAN architecture is shown in more detail in Figure 2.6. UTRAN consists of one or more Radio Network Subsystems (RNS). An RNS is a sub-network within UTRAN and consists of one Radio Network Controller (RNC) and one or more Node Bs. RNCs may be connected to each other via an Iur interface. RNCs and Node Bs are connected with an Iub interface. The Iu, Iur and Iub are logical interfaces, which may be provided via any suitable transport network.

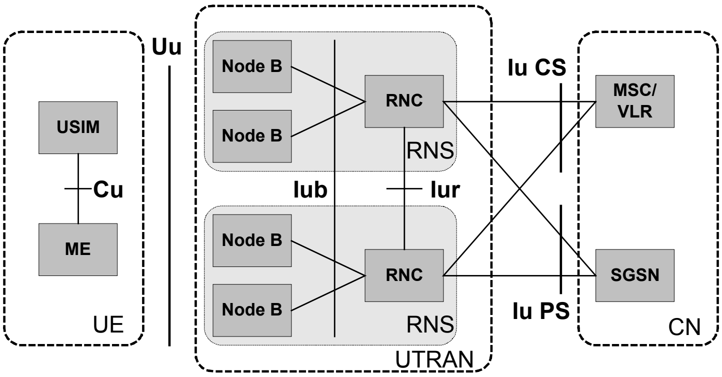

Figure 2.6. UTRAN architecture (extracted from [HoTo00]).

The RNC is the network element responsible for the control of the radio resources of the UTRAN. It interfaces the CN (normally to one MSC and one SGSN) and also terminates the RRC (Radio Resource Control) protocol that defines the messages and procedures between the mobile and UTRAN.

The main function of the Node B is to perform the air interface Layer 1 processing (channel coding and interleaving, rate adaptation, spreading, etc.). It also performs some basic Radio Resource Management (RRM) operations, such as the inner loop power control.

The main elements of CN, whose definition was adopted from GSM/GPRS, are HLR, MSC/VLR, GMSC (Gateway MSC), SGSN, and GGSN.

2.2.3 Radio interface capacity aspects

The IMT-2000 spectrum allocation in Europe, Japan, Korea and the USA is shown in Figure 2.7. In Europe and in most of Asia the IMT-2000 bands of 2×60 MHz (1920-1980 MHz plus 2110-2170 MHz) will be available for UMTS FDD. The availability of the TDD spectrum varies: in Europe it is expected that 25 MHz will be available for licensed TDD use in the 1900-1920 MHz and 2020-2025 MHz bands. The rest of the unpaired spectrum is expected to be used by unlicensed TDD applications in the 2010-2020 MHz band.

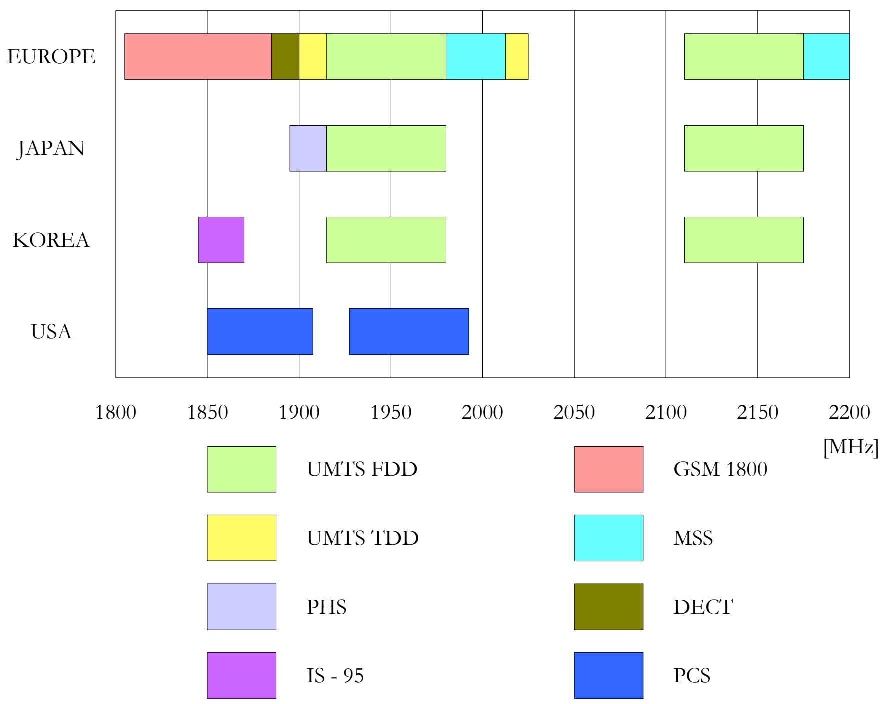

Figure 2.7. IMT-2000 spectrum allocation in Europe, Japan, Korea and USA (based on [HoTo00]).

WCDMA refers to a wideband DS-CDMA system, meaning that user information bits are spread over a wide bandwidth by multiplying the user data with quasi-random bits (called chips) derived from CDMA spreading codes. In order to support very high bitrates (up to 2 Mbps), the use of a variable spreading factor and multicode connections is supported.

The chip rate of 3.84 Mchip/s used leads to a carrier bandwidth of approximately 5 MHz. Subject to his operating licence, the network operator can deploy multiple 5 MHz carriers to increase capacity, possibly in the form of hierarchical cell layers.

As already referred, WCDMA supports highly variable user data rates, in other words the concept of obtaining Bandwidth-on-Demand (BoD). A frame of 10 ms duration is allocated to each user, during which the user data rate is kept constant. However, the data capacity among users can change from frame to frame.

There is no need for a global time reference, such as Global Positioning System (GPS), since WCDMA supports the operation of asynchronous BSs.

The WCDMA air interface has been crafted in such a way that advanced CDMA receiver concepts, such as multiuser detection and smart adaptive antennas, can be deployed by the network operator as system options to increase capacity and/or coverage. Table 2.4 summarises the main parameters related to the WCDMA air interface.

Multiple access method	DS-CDMA
Duplexing method	FDD / TDD
Base station synchronisation	Asynchronous operation
Chip rate	3.84 Mcps
Carrier spacing	5 MHz
Frame length	10 ms
Service multiplexing	Multiple services with different QoS requirements multiplexed on one connection
Multirate concept	Variable spreading factor and multicode
Detection	Coherent using pilot symbols or common pilot
Multiuser detection, smart antennas	Supported by the standard, optional in the implementation

Table 2.4. Main WCDMA parameters (extracted from [HoTo00]).

In UTRA, the data generated at higher layers is carried over the air interface in transport channels, which are mapped onto the physical layer to different physical channels. The physical layer is required to support variable bitrate transport channels to offer BoD services, and to be able to multiplex several services on one connection.

There are two types of transport channels: dedicated and common ones. The main difference between them is that a common channel is a resource divided between all or a group of users in a cell, whereas a dedicated channel resource, identified by a certain code on a certain frequency, is reserved for a single user only. A list of the transport channels defined for UTRA is presented in Table 2.5. In the “direction” column, the possible transmission links are indicated for each channel (UL, DL or both).

Channel	Type	Links	Channel identification
DCH	Dedicated	MS↔BS	Dedicated Channel
BCH	Common	MS←BS	Broadcast Channel
FACH	Common	MS←BS	Forward Access Channel
PCH	Common	MS←BS	Paging Channel
RACH	Common	MS→BS	Random Access Channel
CPCH	Common	MS→BS	Uplink Common Packet Channel
DSCH	Common	MS←BS	Downlink Shared Channel

Table 2.5. Transport channels in UMTS (based on [HoTo00]).

The only dedicated transport channel is the dedicated channel, for which the term DCH is used. The dedicated transport channel carries all the information intended for the given user coming from layers above the physical layer, including data for the actual service as well as higher layer control information.

The common transport channels needed for the basic network operation are RACH, FACH and PCH, while the use of DSCH and CPCH is optional and can be decided by the network operator.

The different transport channels are mapped onto different physical channels, though some of the transport channels are carried by identical (or even the same) physical channel.

The DCH is mapped onto two physical channels. The Dedicated Physical Data Channel (DPDCH) carries higher layer information, including user data, while the Dedicated Physical Control Channel (DPCCH) carries the necessary physical layer control information. These two dedicated physical channels are needed to support efficiently the variable bitrate of DPDCH, which can change from frame to frame.

Once the physical channels are formatted, they are ready for transmission. The transmission of the physical channels consists of two processes: spreading and modulation, Figure 2.8. Spreading consists of the following two operations: channelisation and scrambling. The channelisation operation transforms each symbol into a number of chips, given by the Spreading Factor (SF), thus, increasing the signal bandwidth. In the scrambling operation, the resulting spread signal is further multiplied by a scrambling code, not changing the signal bandwidth and allowing the signals from different sources to be separable. Finally, the chip sequence generated by the spreading process is modulated (QPSK in DL and BPSK in UL) and transmitted over the radio interface.

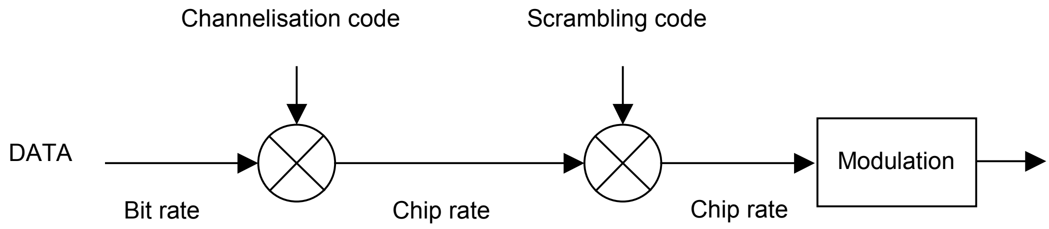

Figure 2.8. UTRA transmission: spreading and modulation.

Transmissions from a single source are separated by channelisation codes, *i.e.*, DL connections within one sector and the dedicated physical channel in the uplink from one terminal. The channelisation codes of UTRA are based on the Orthogonal Variable Spreading Factor (OVSF) technique [HoTo00]. The use of OVSF codes allows the spreading factor to be changed and orthogonality between different spreading codes of different lengths to be maintained.

The transmission from different sources are separated by the scrambling codes. The functionalities and characteristics of the spreading and channelisation codes are summarised in Table 2.6.

	Channelisation code	**Scrambling code**
Usage	UL: Separation of physical data (DPDCH) and control channels (DPCCH) from same terminal DL: Separation of downlink connections to different users within one cell	UL: separation of terminal DL: Separation of sectors (cells)
Length	4-256 chips (1.0-66.7μs) DL additionally 512 chips	UL: 10ms = 38400 chips or 66.7 μs = 256 chips DL: 10ms = 38400 chips
Number of codes	Number of codes under one channelisation code = spreading factor	UL: several millions DL: 512
Code family	Orthogonal Variable Spreading Factor	Long code: Gold code Short code: Extended S(2) code family
Spreading	Increases transmission bandwidth	Does not affect transmission bandwidth

Table 2.6. Functionalities of the channelisation and scrambling codes (adapted from [HoTo00]).

Managing power is of essential importance in WCDMA. In WCDMA based radio networks, users are sharing the same bandwidth at the same time, thus, interfering with each other. Power Control (PC) is applied in order to set the SIR (Signal to Interference Ratio) for each connection to its target level, SIRT, providing the desired link quality while at the same time maximally reducing the interference. PC not only decreases interference, and consequently improves system capacity, but it is also crucial in order to overcome effects like the near-far effect.

In the UTRA FDD mode, open- and closed-loop PC can be distinguished. Open loop PC is used if no direct feedback loop between UE and Node B is available, *e.g.*, during connection set-up phase on the random access channel. Thus, the transmit power has to be adjusted autonomously by the respective transmitting network element. Open-loop PC is mainly applied with common channels (*e.g.*, random access channels). With closed-loop PC a feedback loop exists between UE and Node B.

Closed-loop power control is used to adapt the transmit power on dedicated as well as shared channels to the time varying radio link conditions, *i.e.*, due to fading. Feedback from the receiving station is directly used to set the power levels at the transmitting station for both, the data channel and the corresponding control channel, which have a fixed relative power offset. The closed-loop PC can be characterised by a two-staged process, comprising two feedback loops:

- The inner loop control loop, also referred to as fast power control, is performed between UE and Node B. Its aim is to adapt the transmit power to the fast varying radio link conditions mainly caused by fast fading. The SIR value is estimated at the receiving station and a power up or down command is sent at a rate of 1500 Hz depending on the fact if the value is lower or greater than SIRT.
- The outer loop aims to control the target level SIRT of the inner loop. For this purpose the RNC measures the BLock Error Rate (BLER) of the received data, and sets SIRT in order to match the desired BLER.

In the UL, the physical layer control information is carried by the DPCCH with a fixed spreading factor of 256. The higher layer information, including user data, is carried on one or more DPDCHs, with a possible spreading factor ranging from 256 down to 4. The UL transmission may consist of one or more DPDCHs with a variable spreading factor, and a single DPCCH with a fixed spreading factor.

The DPDCH data rate may vary on a frame-by-frame basis. Typically with a variable rate service the DPDCH data rate is informed on the DPCCH. The DPCCH is transmitted continuously and rate information is sent via the Transport Format Combination Indicator (TFCI).

The maximum user data rate on a single code is derived from the maximum channel bitrate, which is 960 kbps without channel coding and a spreading factor of 4. With channel coding, the practical maximum user data rate for the single code case is in the order of 400-500 kbps. When higher data rates are needed, parallel code channels are used. Up to six parallel codes can be used, raising the channel bitrate for data transmission up to 5740 kbps, which can accommodate 2 Mbps user data or an even higher data rate if the coding rate is ½. The achievable data rates with different spreading factors are presented in Table 2.7. The rates given assume ½-rate coding and do not include bits taken for coder tail bits or the Cyclic Redundancy Check (CRC). The relative overhead due to tail bits and CRC bits has significance only with low data rates [HoTo00].

DPDCH spreading factor	**DPDCH channel bitrate [kbps]**	**Maximum user data rate with ½-rate coding (approx.) [kbps]**
256	15	7.5
128	30	15
64	60	30
32	120	60
16	240	120
8	480	240
4	960	480
4, with 6 parallel codes	5740	2300

Table 2.7. Uplink DPDCH data rates (extracted from [HoTo00]).

In addition to the UL dedicated channel, user data (low rate) can be sent on both the RACH and CPCH. In the UL, services are multiplexed dynamically so that the data stream is continuous with the exception of zero rate.

In DL, the spreading factor for the highest transmission rate determines the channelisation code to be reserved from the given code tree. As the spreading factor is also always fixed in the DL DPCH, the lower rates are implemented with Discontinuous Transmission (DTX), by gating the transmission on/off or with repetition of the bits.

In the DL the spreading factors range from 4 to 512, with some restrictions on the use of spreading factor 512 in the case of soft handover.

Modulation causes some differences between the UL and DL data rates. While the UL DPDCH consists of BPSK symbols, the DL DPDCH consists of QPSK symbols. The DL data rates are given in Table 2.8 with raw bitrates calculated from the QPSK-valued symbols.

When higher data rates are needed, parallel code channels are used as in the case of UL transmission. Up to three parallel codes can be used, raising the channel bitrate for data transmission up to 5760 kbps, which can accommodate 2 Mbps user data or an even higher data rate if the coding rate is ½.

Spreading factor	**Channel bitrate [kbps]**	**DPDCH channel bitrate range [kbps]**	**Maximum user data rate with ½-rate coding(approx.) [kbps]**
512	15	3-6	1-3
256	30	12-24	6-12
128	60	42-51	20-24
64	120	90	45
32	240	210	105
16	480	432	215
8	960	912	456
4	1920	1872	936
4, with 3 parallel codes	5760	5616	2300

Table 2.8. Downlink dedicated channel bitrates (based on [HoTo00]).

In addition to the DL dedicated channel, user data (low rate) can be sent on both the DSCH and FACH. In the DL, services are multiplexed dynamically in a similar way than in UL.

In UTRA, two channel coding methods have been defined. Half-rate and 1/3-rate convolutional coding are intended to be used with relatively low data rates, equivalent to the data rates provided by 2G cellular networks today, though an upper limit has not been specified [HoTo00]. For higher data rates 1/3-rate turbo coding can be applied, which typically brings performance benefits when large enough block sizes are achieved.

The coding methods usable by different channels are summarised in Table 2.9. Although the FACH has two options, the cell access use of FACH is based on convolutional coding, as not all terminals support turbo coding.

DCH	Turbo coding or convolutional coding
CPCH	Turbo coding or convolutional coding
DSCH	Turbo coding or convolutional coding
FACH	Turbo coding or convolutional coding
Other common channels	½-rate convolutional coding

Table 2.9. Channel coding options for different channels (extracted from [HoTo00]).

When dimensioning an UMTS network (or more generally, a WCDMA network), it is very important to estimate the air interface load. For this purpose, two load measures can be considered [HoTo00]: load estimation based on throughput and based on power.

For the throughput based load estimation, the following two load parameters are considered: UL Load Factor (LF), η_{UL}, and DL LF, η_{DL}. These two parameters are commonly used for the purpose of predicting cell capacity and planning noise rise in the dimensioning process of WCDMA networks, and are given by the following two equations [HoTo00]:

$$\eta_{UL} = (1+i)\cdot\sum_{j=1}^{N_{UL}} \frac{1}{1+\dfrac{W}{\left(E_b/N_0\right)_j \cdot R_j \cdot v_j}}; \tag{2.1}$$

$$\eta_{DL} = \sum_{j=1}^{N_{DL}} v_j \cdot \frac{\left(E_b/N_0\right)_j}{W/R_j} \cdot \left[\left(1-\overline{\alpha}\right)+\overline{i}\right]. \tag{2.2}$$

The parameters of these equations are explained in Table 2.10. These load equations allow for the prediction of the amount of noise rise over thermal noise due to interference. The noise rise in the UL is equal to $-10\cdot\log_{10}(1-\eta_{UL})$. Similarly, the noise rise in the DL is given by $-10\cdot\log_{10}(1-\eta_{DL})$. When the load factors become close to 1, the corresponding noise rise approaches to infinity and the system has reached its pole capacity.

The load estimation based on power for UL, η_{UL}', and DL, η_{DL}', are given by the following two equations [HoTo00]:

$$\eta_{UL}' = 1-\frac{P_N}{I_{total}} \tag{2.3}$$

$$\eta_{DL}' = \frac{P_{total}}{P_{\max}} \tag{2.4}$$

where P_N is the background and receiver noise, I_{total} is the received wideband interference power (at the BS), P_{total} is the total DL transmission power and P_{max} is the maximum BS transmission power. I_{total} can be divided into the powers of own-cell (= intra-cell) users, I_{own}, other-cell (=inter-cell) users, I_{oth}, and P_N:

$$I_{total} = I_{own} + I_{oth} + P_N \ . \tag{2.5}$$

Parameter	Definition
N_{UL}	Number of users per cell
N_{DL}	Number of connections per cell=number of users per cell·(1+soft handover overhead)
v_j	Activity factor of user j at physical layer
E_b/N_0	Signal energy per bit divided by noise spectral density that is required to meet a predefined QoS (*e.g.*, bit error rate). Noise includes both thermal noise and interference
W	WCDMA chip rate (3.84 Mcps)
R_j	Bitrate of user j (dependent on service)
i	Other cell to own cell interference ratio seen by the BS receiver
$\overline{\alpha}$	Average orthogonality factor in the cell
$\bar{i}$	Average ratio of other cell to own cell base station power received by user

Table 2.10. Parameters used in UL LF and DL LF calculation (based on [HoTo00]).

The two load estimation methods (throughput and power) provide different tools for load measuring. For instance:

- In the power based approach for DL, the total base station transmission power P_{total} does not provide accurate information concerning how close to the downlink air interface pole capacity the system is operating, in opposition to the throughput based one. For example, in a small cell the same P_{total} corresponds to a higher air interface loading than in a large cell.
- In the power based approach for UL, interference from adjacent cells is directly included in the load estimation, meaning that if the loading in adjacent cells is low, this can be seen in the power based load measurement, and a higher load can be allowed in this cell, *i.e.*, soft capacity can be obtained. In opposition, UL throughput based load estimation does not take interference from adjacent cells or adjacent carriers directly into account.

2.3 The HIPERLAN/2 system

2.3.1 Introduction

The increasing demand for "anywhere, anytime" communications and the merging of voice, video and data communications has created a demand for broadband wireless networks. Under this scope, ETSI created the BRAN [ETSI00a] project to develop standards and specifications for broadband radio access networks that cover a wide range of applications and are intended for different frequency bands (licensed and license exempt use). The BRAN project covers the following systems: HIPERLAN/1, HIPERLAN/2, HIPERACCESS and HIPERLINK [ETSI00a]. In the present report, one will concentrate on the HIPERLAN/2 system, which basically acts as a WLAN system offering high bitrates.

HIPERLAN/2 is a standard for a high speed radio communication system with typical data rates from 6 Mps to 54 Mbps. It connects portable devices with broadband networks that are based on IP, ATM (Asynchronous Transfer Mode) and other technologies. Centralised mode is used to operate HIPERLAN/2 as an access network via a fixed access point. In addition, a capability for direct link communication is provided. This mode is used to operate HIPERLAN/2 as an ad-hoc network without relying on a cellular network infrastructure.

HIPERLAN/2 is capable of supporting multimedia applications by providing mechanisms to handle QoS. Restricted user mobility is supported within the local service area; wide area mobility (*e.g.*, roaming) may be supported by standards outside the scope of the BRAN project. HIPERLAN/2 systems are intended to be operated in the 5 GHz band.

2.3.2 System architecture

A HIPERLAN/2 network typically has a topology as depicted in Figure 2.9. MTs communicate with the Access Points (AP) over a radio interface as defined by the HIPERLAN/2 standards. There is also a direct mode of communication between two MTs (as previously referred), which is still in its early phase of development and is not further considered in this report. The user of the MT may move around freely in the HIPERLAN/2 network, which will ensure that the user gets the best possible transmission performance. An MT, after association has been performed (can be viewed as a login), only communicates with one AP in each point in time. APs see to that the radio network is automatically configured, taking into account changes in the radio network topology, *i.e.*, there is no need for manual frequency planning.

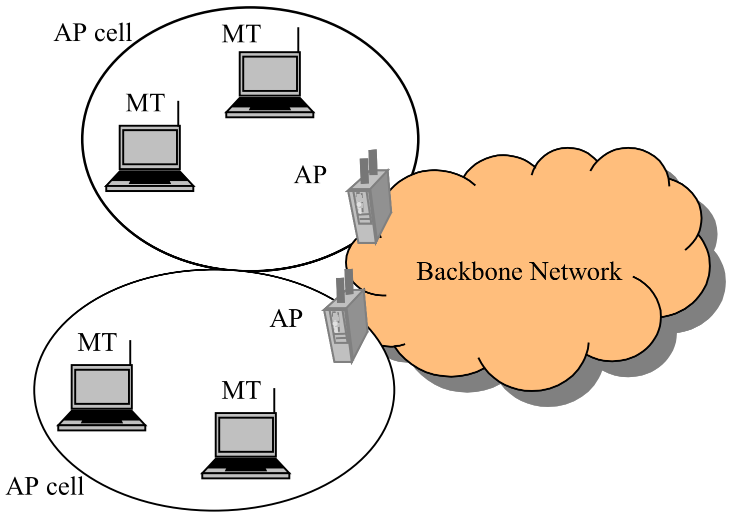

Figure 2.9. HIPERLAN/2 network structure (extracted from [KrSa01]).

The geographical area covered by an AP cell depends on the environment; a radio range of about 30 m in an indoor environment and about 100 to 150 m in an outdoor environment is expected [KrSa01]. The AP cell is connected to a Backbone Network (BN) through the AP. The BN can for example be a wired Ethernet LAN (Local Area Network). Through the BN, different AP cells are interconnected and an increased coverage is accomplished. The BN also provides specific services for MTs, *e.g.*, access to Internet. Different AP cells may partially overlap and an MT, which is dynamically associated to an AP cell, can therefore with a handover or a reassociation mechanism move between two different cells.

2.3.3 Radio interface capacity aspects

Different parts of the 5 GHz band, which are allocated for HIPERLAN/2 (and HIPERLAN/1) in Europe, have different operational conditions to allow the co-existence with other systems, *e.g.*, mobile radar systems [ETSI00a], [KrSa01].

The allocated spectrum shown in Table 2.11 is a license exempt band, which means that the frequency bands are free to be used as long as the spectrum rules are fulfilled. In addition to the spectrum rules shown in Table 2.11, there are requirements on implementing PC and Dynamic Frequency Selection (DFS), which are RRM functions.

Frequency band [MHz]	Nominal carrier frequencies [MHz]	Number of channels	RF power limit	Comment
5 150 - 5 350	5 180 + $n \cdot 20$ (n=0..7)	8	200 mW mean EIRP	Indoor use only
5 470 - 5 725	5 200 + $n \cdot 20$ (n=0..10)	11	1 W mean EIRP (200 mW for the highest channel)	Indoor and outdoor use

Table 2.11. Spectrum allocation and rules for HIPERLAN/2 in Europe (based on [ETSI00a] and [KrSa01]).

For HIPERLAN/2, a channel spacing of 20 MHz is used, allowing high bitrates with a reasonable number of channels.

In Figure 2.10, the protocol reference model for the HIPERLAN/2 radio interface is depicted. The protocol stack is divided into a control plane part and a user plane part following the semantics of ISDN functional partitioning, *i.e.*, user plane includes functions for transmission of traffic over established connections, and the control plane includes functions for the control of connection establishment, release, and supervision.

The HIPERLAN/2 protocol has three basic layers; PHYsical layer (PHY), Data Link Control layer (DLC), and the Convergence Layer (CL).

The PHY layer delivers a basic data transport function by providing means of a baseband modem and an RF part [ETSI00a]. The baseband modem also contains a forward error correction function.

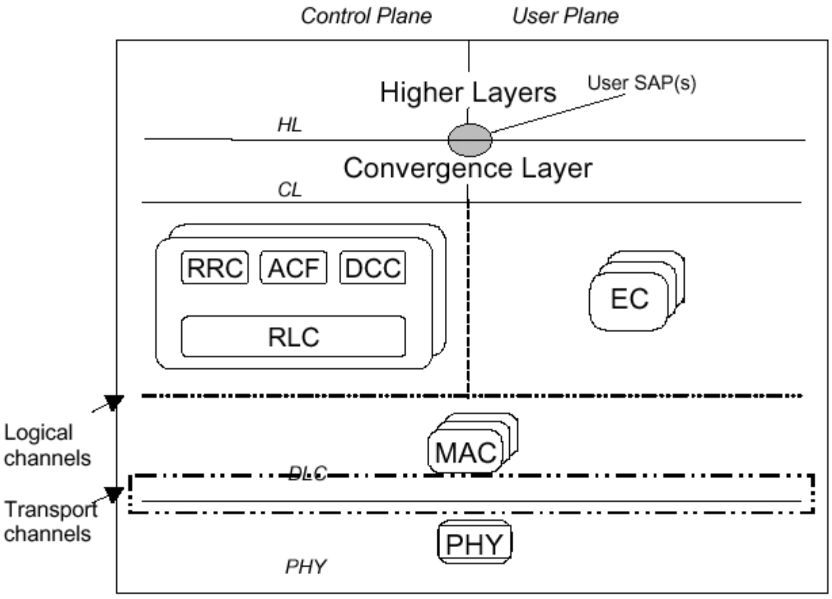

Figure 2.10. HIPERLAN/2 protocol reference model (extracted from [John99]).

The DLC layer consists of the Error Control (EC) function, the Medium Access Control (MAC) function and the Radio Link Control (RLC) function. It is divided into the user data transport functions and the control functions. The user data transport function is fed with user data packets from higher layers via the User Service Access Point (U-SAP). This part contains the EC, which performs an ARQ (Automatic Repeat Request) protocol. The RLC Sublayer delivers a transport service to the DLC Connection Control (DCC), the RRC and the Association Control Function (ACF).

The CL has two main functions: it adapts service requests from higher layers to the service offered by the DLC, and it converts higher-layer packets of fixed or variable length into a fixed length Service Data Unit (SDU) that is used within the DLC [KMST00].

The air interface is based on TDD and TDMA, *i.e.*, the time-slotted structure of the medium allows for simultaneous communication in both DL and UL within the same time frame, called MAC frame in HIPERLAN/2. Time slots for DL and UL communication are allocated dynamically depending on the need for transmission resources. The basic MAC frame structure on the air interface has a fixed duration of 2 ms and comprises transport channels for broadcast control, frame control, access control, DL and UL data transmission and random access (see Figure 2.11). All data from both AP and MTs is transmitted in dedicated time slots, except for the random access channel (RCH) where contention for the same time slot is allowed. The duration of broadcast control is fixed, whereas the duration of other fields is dynamically adapted to the current traffic situation. The MAC frame and the transport channels form the interface between DLC and the PHY layer.

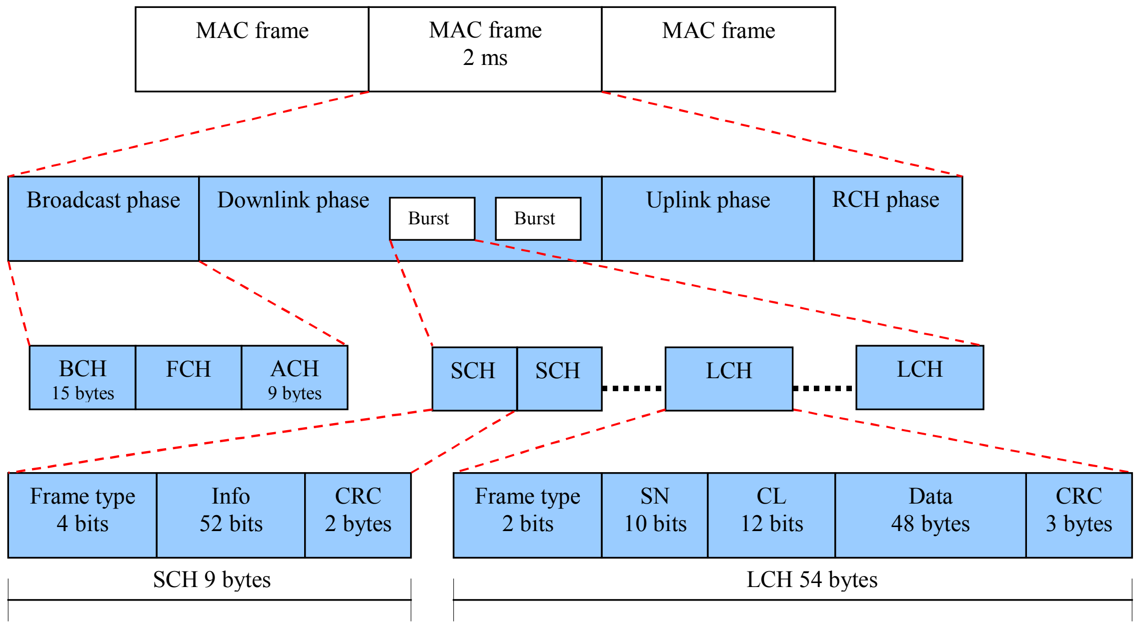

Figure 2.11. MAC frame structure in HIPERLAN/2 (extracted from [KrSa01]).

A MAC frame comprises a broadcast phase, downlink phase, uplink phase and a random access phase. The broadcast phase comprises fields for broadcast control (BCH), frame control (FCH) and access feedback control (ACH). The uplink and downlink phases carry traffic from/to an MT in bursts of packets called Long CHannels (LCH) and Short CHannels (SCH). A LCH contains control or user data and a SCH contains only control data (*e.g.*, acknowledgements and Resource Requests – RR - in the uplink) and is always generated by the DLC layer.

Both logical and transport channels were introduced in the HIPERLAN/2 standards, in order to improve readability and to define exact terms for the structures that are used to construct MAC frames [ETSI00b]. The logical channels are mapped onto different transport channels. The transport channels are the basic elements to construct PDU (Packet Data Unit) trains. The PDU trains (that consist of a sequence of transport channels) represent the interface between the DLC protocol and the PHY layer [ETSI01]. Transport channels describe the basic message format. The message contents and their interpretation, however, are subject to the logical channels.

Table 2.12 contains a list of the HIPERLAN/2 transport and logical channels (according to [ETSI00b]), and in the "direction" column indicates the possible transmission links on each channel (UL, DL or both).

Type	Links	Channel	Channel identification
Transport Channels	MS←BS	BCH	Broadcast Channel
	MS←BS	FCH	Frame Channel
	MS←BS	ACH	Access Feedback Channel
	MS↔BS	SCH	Short Transport Channel
	MS↔BS	LCH	Long Transport Channel
	MS→BS	RCH	Random Access Channel
Logical Channels	MS←BS	BCCH	Broadcast Control Channel
	MS←BS	FCCH	Frame Control Channel
	MS←BS	RFCH	Random Access Feedback Channel
	MS↔BS	RBCH	RLC Broadcast Channel
	MS↔BS	DCCH	Dedicated Control Channel
	MS↔BS	UBCH	User Broadcast Channel
	MS↔BS	UMCH	User Multicast Channel
	MS↔BS	UDCH	User Data Channel
	MS↔BS	LCCH	Link Control Channel
	MS→BS	ASCH	Association Control Channel

Table 2.12. HIPERLAN/2 transport and logical channels (based on [ETSI00b]).

Downlink or uplink traffic (DL- and UL-phase, bidirectional) consists of PDU trains to and from MTs. A PDU train comprises DLC user PDUs (U-PDUs of 54 bytes with 48 bytes of payload) and DLC control PDUs (C-PDUs of 9 bytes) to be transmitted or received by one MT. There is one PDU train per MT (if resources have been granted in the FCH). The C-PDUs are referred to as the SCH, and the U-PDUs are referred to as the LCH.

The PHY layer breaks down the packets from the MAC layer into Orthogonal Frequency Division Multiplex (OFDM) symbols and transmits them [KrSa01]. In OFDM, the available bandwidth is divided into several sub channels where each sub channel is modulated and transmitted in parallel. The sub-carrier frequencies are chosen so that the different signals carried by these frequencies are orthogonal in time, thereby the name. Four of the 52 sub-carriers are pilots, which facilitate phase tracking for coherent demodulation.

The PHY layer provides several transmission modes with different coding rates and modulation schemes. The benefit of using different schemes is that a scheme supporting a high transmission rate can be used when transmission conditions are favourable. The function of choosing modulation schemes depending on transmission conditions is called Link Adaptation (LA). LA can be based on estimated C/I values at the receiver and this of course requires protocols for initiating and sending measurements. Another technique is more simple and selects the mode depending on PDER (PDU Error Rate). In Table 2.13 the different transmission modes are presented together with requirements on receiver sensitivity. The data rate ranging from 6 to 54 Mbps can be varied by using the several transmission modes for each OFDM sub-carrier.

Mode	**Modulation and coding rate**	**Code rate**	**PHY layer transmission rate [Mbps]**	**Receiver sensitivity [dBm]**
1	BPSK	½	6	-85
2	BPSK	¾	9	-83
3	QPSK	½	12	-81
4	QPSK	¾	18	-79
5	16-QAM	9/16	27	-75
6	16-QAM	¾	36	-73
7	64-QAM	¾	54	-68

Table 2.13. Transmission modes and requirements for HIPERLAN/2 (based on [ETSI01] and [KrSa01]).

The PHY layer supports other RRM functionalities, besides LA, such as DFS, PC and handover. The main purpose of these functions is to increase the overall performance of the system

HIPERLAN/2 uses a centralised control mechanism, and the AP is responsible for assigning radio resources within the MAC frame.

Any MT that wants initial access to the radio network uses the RCH for sending a RR to the AP. If a collision occurs during a resource request, the AP will inform the requesting MT in the following ACH and a backoff procedure in the RCH will take place. After a successful RR, which is indicated in the ACH, the MT goes into a contention free mode where the AP schedules the MTs transmissions. A RR contains the number of pending LCHs in the MT and the AP uses this information when assigning radio resources. The FCH sent by the AP contains an exact description of the resource allocation within a MAC frame, *i.e.*, the time slots during which an MT shall transmit or receive. The length of the FCH depends on the number of scheduled sessions. The BCH is used by the AP to broadcast cell information such as the identity of the AP, the AP transmission power and the expected received power level at the AP. The transmission mode used in the broadcast phase and the RCH phase is the most robust (mode 1) since this information is supposed to be received by all MTs in the AP cell.

HIPERLAN/2 supports QoS by allowing the access point to set up and manage different radio bearers during transmission [KMST00]. The AP selects the appropriate error control mode (acknowledged, unacknowledged and repetition) including detailed protocol settings (*e.g.*, ARQ window size, number of retransmissions, discarding). Scheduling is performed at the MAC level, where the AP determines how much data and control signalling will be sent in the current MAC frame. For example, by regularly polling a mobile terminal for its traffic status (pending data to be transmitted), the AP provides the terminal's radio bearer with short access delay. The polling mechanism provides rapid access for real-time services. Additional QoS support includes LA and internal functions (admission, congestion, and dropping mechanisms) for avoiding overload situations.

Chapter 3

Services Characterisation and Traffic Source Models

In this chapter, the services and applications that are considered in the present work are described. A detailed characterisation and parameterisation of traffic source models, as well as of generation and duration processes, for each considered application is also presented. At the end of the chapter, a mapping of the technologies (GSM/GPRS, UMTS and HIPERLAN/2) versus the services/applications and traffic source models is performed.

3.1 Services characterisation

In order to start the characterisation of the services and applications that will be considered in the present work, a definition of the terms services, applications, and service capabilities will be initially presented, followed by a detailed definition of each service capability and service class. At the end of this section, a description, classification, and characterisation of each service and application that will be used in the current work is presented.

3.1.1 Definitions

This section provides an overview of the definitions of service, application, and service capabilities, in order to have a clear understanding of these terms throughout the present report.

The definition of **services** given by standardisation (ITU-T) [EURE00] is:

- This term represents telecommunication capabilities that the customer buys or leases from a service provider. Service is an abstraction of the network-element-oriented or equipment-oriented view. Identical services can be provided by different network elements, and different services can be provided by the same network elements.
- A set of functions and facilities offered to a user by a provider.

Services are not directly accessible by users; they have to be utilised by other services and/or applications. Services can contain rather complex program logic (potentially implemented by modules), or they can be simple wrappers to other services and/or modules. One service can serve several service consumers at the same time. The execution environment of a service is a server.

The definition of **applications** given by standardisation (ITU-T) [EURE00] is:

- The word **application** is used as the generic term to represent the set of features, combining communication and document processing, on which end users may perform operations. Applications may depend on working methods, and on allowed processing of documents. Examples of "Applications" are: open interchange of processable documents, co-operative working, etc.

- A set of activities performed to respond to the needs of users in a given situation, for purposes such as business, education, personal communication or entertainment. It implies software and hardware utilisation, which could be performed in a fully or partially automatic way, and could be accessed locally or remotely. In the last case, it requests use of telecommunication services.

Regarding **service capabilities,** the following definition is adopted:

- **Service capabilities** are the characteristic parameters of a service.

Examples of Service Capabilities include information type, intrinsic time dependency, directionality of connection, transfer delay and bit error rate.

3.1.2 Service capabilities

The following are the service capabilities that will be considered for service characterisation within the present work [Kwok95]:

- **Information type**: Information to be communicated can be classified as one of the following type (or combination of types): sound, video and sound, text, data, still image, graphics. Multimedia representation designates a mixture of text, data, graphics, sound, still image or video information types.
- **Intrinsic time dependency**: Information can be classified as Time-Based (TB) or Non-Time-Based (NTB). Time-based information is user data that has an intrinsic time component. Video, audio and animation are time-based information, because they generate a continuous sequence of data blocks that must be displayed or played back consecutively at predetermined time instants. For example, video can be displayed at 15 or 30 video frames/s. On the other hand, text and images are non-time-based information.
- **Delivery requirements**: An application can be classified according to its

information delivery requirements as a Real-Time (RT) or a Non-Real-Time (NRT) one. Real-time is one that requires information delivery for immediate consumption. In contrast, non-real-time application information is stored (perhaps temporarily) at the receiving points for later consumption.

Users who communicate via a real-time application must be present at the same time, whereas those who communicate via a non-real-time application can participate at different times. The former requires sufficient bandwidth, while the latter requires sufficient storage (and potentially bandwidth as well if delivered at high speed). For example a telephone conversation is considered real-time, while sending Email is non-real-time.

- **Directionality of connection**: Communications can be Unidirectional (Uni) or Bi-directional (Bid). For example, an usual telephone call is a bi-directional application, while radio broadcasting is an unidirectional one.
- **Symmetry of connection**: In general, a communication is bi-directional, and bi-directional applications can be classified as either symmetric or asymmetric. A broadcast application is an extreme example of an asymmetric application that is one-way only, while telephony is a symmetric application.
- **Number of parties**: Depending on whether there are two or more parties involved, an application can be classified as either point-to-point (two parties) or multipoint (more than two), respectively. Obviously, point-to-point (One-to-One, O-O) applications require only point-to-point connections, while multipoint applications require multiparty connection types, such as point-to-multipoint (One-to-Many, O-M), multipoint-to-point (Many-to-One, M-O), or multipoint-to-multipoint (Many-to-Many, M-M) connections.
- **Switching mode**: There are two possible transmission modes: PS or CS. In the CS type of connection, a dedicated path is established between the transmitting and receiving devices before transmission is begun. PS is a mode of data transmission in which messages are broken into packets, each of which can be routed separately from a source, then reassembled in the proper order at the destination.
- **Average duration**: Average duration quantifies the average duration of a connection/session.
- **Transfer delay**: This is the time used to transmit information through the air

interface and the network. It includes channel coding/decoding, interleaving, propagation delay, etc. Delay variation within a call may also occur due to changes in the channel delay, compared to the average delay. In CS type of connections, the jitter can be thought to be insignificant; consequently, the delay is a constant.

- **Bitrate range**: Bitrate range designates the foreseen range of bitrate values a certain application can support.
- **Burstiness**: Burstiness is the ratio between peak and average bitrates. A communication with low burstiness is for example a streaming application (*e.g.*, audio on-demand), where a continuous stream of information is transferred during the connection/session. Interactive applications (*e.g.*, web browsing) are typically applications with high burstiness, because in short periods of time high amounts of data are transferred (when a certain web page is downloaded), while communication does not exist during long periods of time (the user is reading the page).
- **Bit Error Rate and Frame Erasure Rate**: Bit Error Rate (BER) is the ratio of lost or erroneous bits to total transferred information bits. Frame Erasure Rate (FER) is the ratio of erased frames to total frames[2]. These parameters refer to the end user / application QoS requirements.

3.1.3 Service classes

Different types of applications have different characteristics and performance requirements. The large set of possible applications has been grouped into four main categories of service classes, by 3GPP [3GPP02d], according to different QoS requirements. Although this classification is envisaged for UMTS, it can also be adopted for other wireless systems, since these classes are general enough to cover most of the foreseen services for NG (Next Generation) wireless networks. The four services classes are [3GPP02d]:

---

[2] Care should be used to distinguish between frame erasure rate (FER) and frame error rate. They are similar, but not identical. Erased frames usually are counted as errored frames, but not all errored frames are erased, that is, some may be undetected by the receiver.

- Conversational class
- Streaming class
- Interactive class
- Background class.

In the next paragraphs each of these service classes is described in detail, as presented in [3GPP02d]. As it will be seen in this description, some of the parameters (service capabilities) presented in the previous section are key attributes for the distinction among classes.

- Conversational class

The most well known use of conversational class is telephony speech (*e.g.*, GSM). But with Internet and multimedia, a number of new applications will require this scheme, for example VoIP and video conferencing tools. Real-time conversation is always performed between peers (or groups) of live (human) end-users. This is the only scheme where the required characteristics are strictly given by human perception.

The real-time conversation scheme is characterised by a low transfer time, because of the conversational nature of the scheme, and at the same time the time relation (variation) among information entities of the stream must be preserved in the same way as for real-time streams. The maximum transfer delay is given by the human perception of video and audio conversation. Therefore, the limit for acceptable transfer delay is very strict (less than 200 ms), as failure to provide low enough transfer delay will result in unacceptable lack of quality. The transfer delay requirement is both significantly lower and more stringent than the round trip delay of the interactive traffic case. Traffic is nearly symmetric in this service class.

- Streaming class

When the user is looking at (listening to) real-time video (audio) the scheme of real-time streams applies. Multimedia streaming is a technique for transferring data such that it can be processed as a steady continuous stream (*e.g.*, a client browser can start displaying data before the entire file has been received). The real-time data flow is always aiming at a live (human) destination. Streaming services are mostly unidirectional (applications are very asymmetric.). At the receiver, streaming data is played by a suitable independent media player application (downloadable from the Web or readily bundled to a browser) or a browser plug-in.

This scheme is one of the newcomers in data communication, raising a number of new requirements in both telecommunication and data communication systems.

Another characteristic of this service class is that time relations (variation) among information entities (*i.e.*, samples, packets) within a flow must be preserved, although it does not have strict requirements on low transfer delay.

The delay variation of the end-to-end flow must be limited, to preserve the time relation (variation) between information entities of the stream. But as the stream normally is time aligned at the receiving end (in the user equipment), the highest acceptable delay variation over the transmission media is given by the capability of the time alignment function of the application. Thus, acceptable delay variation is much greater than the delay variation given by the limits of human perception.

- Interactive class

When the end-user, that is either a machine or a human, is on line requesting data from remote equipment (*e.g.*, a server), the interactive scheme applies. Examples of human interaction with the remote equipment are: web browsing, data base retrieval, and server access. Examples of machines interaction with remote equipment are: polling for measurement records, and automatic data base enquiries (tele-machines).

Interactive traffic is a classical data communication scheme that on an overall level is characterised by the request response pattern of the end-user. At the message destination there is an entity expecting the message (response) within a certain time. Round trip delay time is therefore one of the key attributes. Another characteristic is that the content of the packets must be transparently transferred (with low bit error rate).

- Background class

When the end-user, which is typically a computer, sends and receives data-files in the background, this scheme applies. Examples are background delivery of Emails, SMS, download of databases and reception of measurement records.

Background traffic is one of the classical data communication schemes that on an overall level is characterised by that the destination is not expecting the data within a certain time (they do not require immediate action). Thus, the scheme is more or less delivery time insensitive. Delay may vary from seconds to minutes or even hours. Another characteristic is that the content of the packets must be transparently transferred (with low bit error rate).

Table 3.1 resumes and compares the main characteristics of the different service classes described previously.

Class	**Conversational**	**Streaming**	**Interactive**	**Background**
Fundamental characteristics/ requirements	Preserve time relation (variation) among information entities of the stream. Conversational pattern (stringent and low delay).	Preserve time relation (variation) among information entities of the stream.	Request response pattern within a certain time; round-trip delay time is therefore a key attribute. Preserve payload content: packets must be transparently transferred with low bit error rate.	Destination is not expecting the data within a certain time. Preserve payload content: packets must be transparently transferred with low bit error rate.
Transfer delay	Minimum Fixed	Minimum Variable	Moderate Variable	Large Variable
Buffering	No	Allowed	Allowed	Allowed
Nature of traffic	Symmetric	Asymmetric	Asymmetric	Asymmetric
Bandwidth	Guaranteed bitrate	Guaranteed bitrate	No guaranteed bitrate	No guaranteed bitrate

Table 3.1. Main characteristics of the different service classes.

The main distinguishing factor among these classes is how delay sensitive traffic is. Conversational is meant for traffic that is very delay sensitive, while Background is the most delay insensitive traffic class.

Conversational and Streaming classes are mainly intended to be used to carry real-time traffic flows. The main divider between them is how delay sensitive traffic is. Conversational real-time services, like video telephony, are the most delay sensitive applications and those data streams should be carried in Conversational class.

Interactive and Background classes are mainly meant to be used by traditional Internet applications, like WWW, Email, Telnet, FTP (File Transfer Protocol) and News. Due to less stringent delay requirements, compared to Conversational and Streaming classes, both provide better error rate by means of channel coding and retransmission.

Traffic in the Interactive class has higher priority in scheduling than Background class traffic, so background applications use transmission resources only when interactive applications do not need them. This is very important in a wireless environment, where the bandwidth is low compared to fixed networks.

3.1.4 Selected services and applications description and characterisation

Within the present work, a limited and representative set of services and applications will be considered for the activities developed in the following chapters. This set should be heterogeneous enough, in order to translate the multiplicity of services, applications and traffic patterns that the systems under consideration in the present work will bear.

The selected set of services and applications is presented in Table 3.2. As it can be seen in the table, services and applications covering all four service classes, as well as all the main information types, are considered.

Service Class	Information Type	Selected Services	Selected Applications
Conversational	Sound	Speech-telephony	Voice Call
	Video, Sound	Video-telephony	Video Call
Streaming	Video, Sound	Video Streaming	Video on Demand
Interactive	Multimedia	Multimedia Communication Service	Web Browsing (WWW)
Background	Data, Sound	Messaging	Email
	Multimedia	Unrestricted Data Retrieval Service	FTP

Table 3.2. Selected services and applications.

In the following paragraphs, a description, classification and characterisation of the selected services and applications (Table 3.2) is presented.

In the following tables, services are grouped into the 3GPP service classes, described in section 3.1.3. In each class, services are also grouped according to the type of information.

Type of Information	Services	Description	Reference
Sound	Speech telephony	Provides a traditional speech quality two-way conversational service between one or many user devices. In the near future, as IPv6 based networks evolve, speech telephony will tend to become VoIP.	[UMTS01]
Video and Sound	Video-telephony	Communication for the transfer of voice (sound) and video between two or more locations (person-to-person). There is a wide range of quality of service levels for video telephony.	[ITUT93]

Table 3.3. Conversational services description.

Type of Information	Services	Description	Reference
Video and Sound	Video streaming	One-way video (and sound) streaming. Represents a pull model of content delivery on a one-to-one or one-to-many multicast basis.	[HoTo00]

Table 3.4. Streaming services description.

Type of Information	Services	Description	Reference
Multimedia	Multimedia Communication Service	Exchange between users and workstations or user-to-user transfer of a range of multimedia documents with minimal delay.	[ITUT93]

Table 3.5. Interactive services description.

Type of Information	Services	Description	Reference
Data, Sound	Messaging	Point-to-point one-way connectionless transmission of data or sound or multimedia messages between users or machine and users.	[ITUT93]
Data	Unrestricted Data Retrieval Service	Transfer of unrestricted data, without stringent constraints on delay or transfer times.	[ITUT93]

Table 3.6. Background services description.

In Table 3.7, the selected applications are presented and described.

Application	Description	Reference
Voice Call	One to one voice communication with at least GSM voice quality, and connection expectations (at all mobility levels).	[SPWW02]
Video Call	One to one voice and video communication. This application can be used in a conversational way, but also to transmit video of the users' surroundings in a wide variety of activities, from shopping to telemedicine to police work. Face to face calls will usually be made while the user is stationary, but moving in vehicles at any speed.	[SPWW02]
Video on Demand	Browsing for a short video clip (*e.g.*, news, advertisement or sports) of an event, and then watching it, stationary or moving. This application can be for entertainment, business, educational or training purposes.	[SPWW02]
Web Browsing	Downloading of mixed media. It is one of the nowadays most popular multimedia applications, also with large future foreseen acceptance (*e.g.*, Mobile Internet). This information contains text, extensive graphics, video and audio sequences. It gives access to a wide number of areas: Communications and community (Email, calendar and chat), information (news, weather, directories), lifestyle (events, restaurants, movies, games), travel (hotel listings, direction assistance, timetables), transaction (banking, stock trading, purchasing and auctions), other (information about personalisation, location-based services, device type, advertising).	[UMTS01]
Email	A process of sending messages in electronic form. These messages are usually in text form. However, they can also include images and video clips.	[UMTS01]
FTP	Application with FTP functionalities, allowing interactive browsing of directories and transfer of files. It allows the transfer of any type of data file between different types of computers or networks.	[UMTS01]

Table 3.7. Applications description.

A detailed characterisation of the services and applications previously described, using the parameters (service capabilities) identified in section 3.1.2, is presented in what follows.

In Table 3.8 to Table 3.11, Conversational, Streaming, Interactive and Background services and applications are characterised, mainly based on [FCXV03] (which is focussed on UMTS oriented services and applications).

Type of Info.	**Service**	**Applications**	**Service Capabilities**										
			TB/ NTB	**RT/ NRT**	**Uni / Bid**	**Sym/ Asy**	**Part.**	**CS/ PS**	**Average Duration [s]**	**Transfer Delay [s]**	**Bitrate Range [kbps]**	**Burstiness**	**BER/ FER**
Sound	Speech-telephony	Voice Call	TB	RT	Bid	Sym	O-O	CS/ PS	120	<0.15	4-25	1-5	$<10^{-4}$ BER [HoTo00] < 3 %FER [3GPP01c]
Video, Sound	Video-telephony	Video Call	TB	RT	Bid	Sym	O-O	CS/ PS	120	<0.15	32-384	1-5	< 1 % FER [3GPP01c]

Table 3.8. Conversational services and applications characterisation.

Type of Info.	**Service**	**Applications**	**Service Capabilities**										
			TB/ NTB	**RT/ NRT**	**Uni / Bid**	**Sym/ Asy**	**Part.**	**CS/ PS**	**Average Duration [s]**	**Transfer Delay [s]**	**Bitrate Range [kbps]**	**Burstiness**	**BER/ FER**
Video, Sound	Video streaming	Video on Demand	TB	RT	Bid	Asy.	O-O O-M	CS/ PS	300	<10	32-384	1	$<10^{-6}$ BER <1 % FER [3GPP01c]

Table 3.9. Streaming services and applications characterisation.

Type of Info.	**Service**	**Applications**	**Service Capabilities**										
			TB/ NTB	**RT/ NRT**	**Uni / Bid**	**Sym/ Asy**	**Part.**	**CS/ PS**	**Average Duration [s]**	**Transfer Delay [s]**	**Bitrate Range [kbps]**	**Burstiness**	**BER**
Multi-media	Multimedia Communication Service	Web Browsing	TB/ NTB	RT	Bid	Asy	O-O	PS	300	<4/pag	< 2000	1-20	$<10^{-6}$ BER [PCMF99]

Table 3.10. Interactive services and applications characterisation.

Type of Info.	Service	Applications	Service Capabilities										
			TB/ NTB	**RT/ NRT**	**Uni / Bid**	**Sym/ Asy**	**Part.**	**CS/ PS**	**Average Duration [s]**	**Transfer Delay [s]**	**Bitrate Range [kbps]**	**Burstiness**	**BER**
Data	Messaging	Email	NTB	NRT	Uni	Asy	O-O	PS	2.4	<4	<128	1	$<10^{-6}$ BER [HoTo00]
	Unrestricted Data Retrieval Service	FTP	NTB	NRT	Bid	Asy	O-M	PS	132	<0.5	64-400	1-50	$<10^{-6}$ BER [HoTo00]

Table 3.11. Background services and applications characterisation.

3.2 Traffic source models

3.2.1 Initial Considerations

Traffic density differs among services, and therefore must be modelled separately for each service. The central idea of traffic modelling lies in constructing models that capture the important statistical properties of the underlying measured trace data. The more accurate traffic is estimated, the more realistic results are achieved. A review of generic modelling techniques is presented in [FrMe94].

Many traffic source models are proposed in the literature for different types of services. However, traffic source models for advanced NG mobile wireless networks services are vary scarce and little information (mainly on parameterisation settings) can be found. In the particular case of data services, nearly all existing traffic models are presented for wireline data networks such as ATM or LANs. Another particular case of scarce literature is for video source models (*e.g.*, video telephony and video streaming) for NG mobile wireless networks. Again, in this case, the existing models are usually made for broadband wireline networks (*e.g.*, ATM) and using coding schemes (*e.g.*, MPEG-1, MPEG-2) different than the ones foreseen for NG mobile wireless networks (*e.g.*, MPEG-4, ITU-T H.263). In the present work, and in the particular case of video source models, very recent MPEG-4 and ITU-T H.263 trace data [FiRe01] will be processed and used to parameterise some existing traffic models.

The following subsections will provide an overview of some selected traffic source models (from the literature) for the applications described in 3.1.4. The selection criteria was based on the source models that presented analytical tractability and suitable example parameters (for the considered mobile wireless systems). Other source models than the ones presented are referenced during the following sections.

The traffic source models are divided into the four service classes. A description and parameterisation of the generation and duration processes for each application is also presented.

3.2.2 Conversational type traffic models

Conversational type services typically present a symmetric or quasi-symmetric nature and require small end-to-end transmission delays. Examples of this type of traffic are: voice, VoIP and video telephony.

3.2.2.1 Classical ON-OFF voice model

A voice source generates a pattern of voice activity and silence periods, which can be classified by a Voice Activity Detector (VAD), usually used in mobile terminals supporting voice (*e.g.*, GSM). These periods correspond mainly to talking, pausing and listening patterns of a conversation [NaGT91]. Figure 3.1 illustrates these periods for the example of a circuit switched voice call.

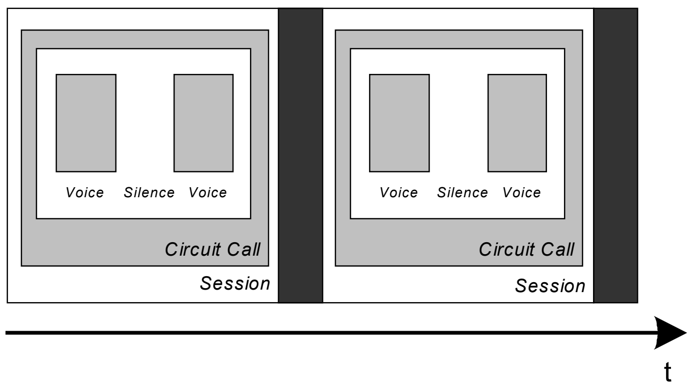

Figure 3.1. Voice activity and silence periods in a circuit switched voice call (extracted from [Vlah01]).

The voice activity detector is usually modelled by the classical ON-OFF voice model [RaMe01], [MWIF01], [ETSI98], [KoPP01a], [KoPP01b], [NaGT91], which is a two-state Markov model, Figure 3.2. The probability that a voice activity period ("ON" state) with mean duration t_{ON} ends in a time slot of duration τ follows an exponential distribution, given by:

$$\gamma = 1 - e^{-\tau / t_{ON}} . \tag{3.1}$$

This is the probability of a transition from the "ON" state, to the silence state ("OFF" state). Correspondingly, the probability that a silence period, of mean duration t_{OFF} ends in a time slot of duration τ also follows an exponential distribution, given by

$$\sigma = 1 - e^{-\tau / t_{OFF}} . \tag{3.2}$$

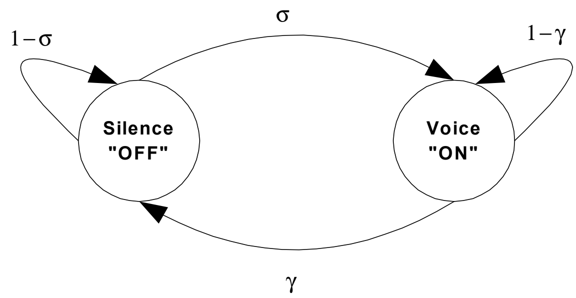

Figure 3.2. Voice activity model (ON-OFF model) (extracted from [NaGT91]).

In these models, the two directions of communication (UL and DL) are modelled independently. Typical values for the mean durations of the “ON” and “OFF” state, t_{ON} and t_{OFF} respectively, are 3 s for both [RaMe01], [ETSI98].

3.2.2.2 CDMA mobile systems voice model

In [VaRF99], a measurement-based voice traffic model is described, which captures not only the ON-OFF behaviour, previously described, but also the effect of voice encoder, compression device and air interface (AI) characteristics in CDMA mobile communication systems. Most encoders used in CDMA systems produce voice packets periodically, with a specific packet size distribution in a range between 20-40 octets. The model described in [VaRF99] is a four-state Markov model: when the source is in state k it generates packets of size s_k each 10 ms, for a burst duration of τ_k, k=1,…,4. In the long time average, the probabilities that a packet is of size s_k are measured to be P_k. The burst duration τ_k is modelled as a random variable with mean value μ_k with the Weibul CDF (Cumulative Distribution Function):

$$P_W\left(x\right) = 1 - e^{(-\lambda_k \cdot x)^{\beta_k}}, \; x \geq 0, \tag{3.3}$$

where $1/\lambda_k$ and β_k are the scale and shape parameters, respectively. In [VaRF99] (and references therein) it is observed that with the proper selection of these parameters, this distribution function is a more accurate model of packetised encoded voice than exponential burst length approximations.

After the burst of state k, a new state is selected with probability Q_k given by:

$$Q_k = \frac{P_k / \mu_k}{\sum_{j=1}^{4} P_j / \mu_j}, \quad k = 1, \ldots, 4. \tag{3.4}$$

The parameters for this model are taken from [VaRF99] and are summarised in Table 3.12.

State	Packet size [octet]	Measured probability	Measured mean burst duration [packet]	Weibul parameters	
k	s_k	P_k	μ_k	λ_k	β_k
1	2	0.5978	29.8	0.03	0.75
2	5	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

Table 3.12. Voice source model parameters (extracted from [VaRF99]).

Figure 3.3 summarises the voice source model, and also illustrates that all voice sources have the same repetition rate but, depending on AI settings, they might have different phase offsets. In the figure, the first and second sources are of equal phase, the n^{th} source has offset.

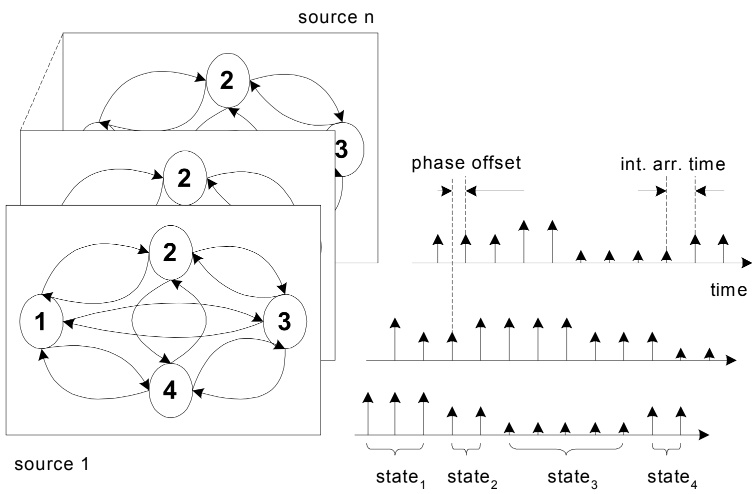

Figure 3.3. Voice source model (extracted from [VaRF99]).

3.2.2.3 IP telephony (VoIP) model

For IP based voice communications, [DiLS01], [Mand01], and [VaRF99] describe possible traffic models. As for the previous voice models, these models describe voice as a birth-death process, with a Poisson distributed arrival process and an exponential distributed call duration.

Both [DiLS01] and [Mand01] present a similar traffic model for VoIP, based on the fact that speech can be characterised with a sequence of talkspurts (service times, messages) separated by silent spurts (idle times), Figure 3.4. In speech, each party is usually silent for more that 60 % of the time [Mand01]. Therefore, a PS voice transmission can be modelled as a Markov model with two states of "silence" and "talkspurt". When in "silence" no packets are generated, and when in "talkspurt" packets at a constant rate are generated [Mand01]. During the activity phase, IP packets carrying the speech information are transmitted. This type of traffic can therefore be simulated as an ON-OFF model with activity and silent periods generated by an exponential distributed random variable with mean values t_{ON} and t_{OFF} respectively, as for the classical ON-OFF model presented in Section 3.2.2.1.

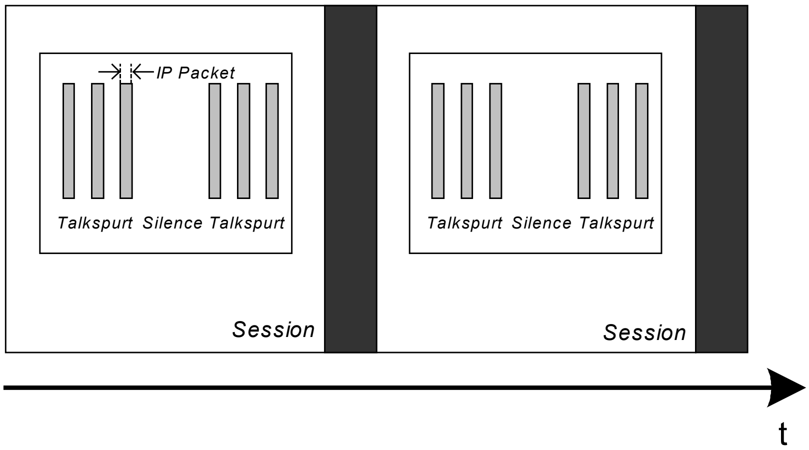

Figure 3.4. IP telephony transmission session.

The payload size depends on the considered speech codec and the packet rate. A typical speech codec used in second and third generation mobile communications is the adaptive multi-rate speech codec (AMR), which is a multi-mode codec with bitrates between 4.75 and 12.2 kbps. In the AMR, processing is done on 20 ms frames taking 160 samples per frame. It offers high speech quality under a wide range of transmission conditions. With a packet rate of 12.2 kbps, the packet size has a payload of 32 bytes. Furthermore, the header comprising of RTP, UDP and IP has 40 bytes using IPv4 and up to even 60 bytes using IPv6. This shows the problem of the header overhead typical for VoIP. However, header size can be reduced considering that there is significant redundancy between header fields, both within the same packet header, but in particular between consecutive packets belonging to the same packet stream.

One can assume that header bytes can be compressed to 8 bytes [Mand01] and that the Transmission Time Interval (TTI) of these (32+8) byte packets is 20 ms. For instance within an active period of 3 s 150 IP packets have to be transmitted.

Table 3.13 summarises the parameters for the above described traffic model, based on [Mand01].

Activity	50 %
Mean active phase, t_{ON}	3 s
Mean silent phase, t_{OFF}	3 s
Payload of IP packet	32 bytes
IP framing overhead	8 bytes
Transmission time interval	0.02 s
Voice packet generation	0.02 s
IP packet rate	0.02 packet/s (50 Hz)

Table 3.13. Parameters of the VoIP traffic model (extracted from [Mand01]).

3.2.2.4 Video telephony/conference models

3GPP has specified that the transmission of video telephony in third generation mobile systems, using CS connections, shall follow ITU-T H.324M while for PS connections, ITU-T H.323 and IETF (Internet Engineering Task Force) SIP (Session Initiated Protocol) are two main candidates [HoTo00]. In the case of ITU-T H.323 the video signal is generated according to the recommendation ITU-T H.261 or optionally according to ITU-T H.263. However, 3GPP has specified that the use of the video codec ITU-T H.263 is mandatory for mobile terminals supporting packet switched conversational multimedia applications (MPEG-4 is considered as an option) [3GPP01b]. In fact, ITU-T H.263 supports efficient compression that enables low-bitrate video. 3GPP has specified that for associated audio signals the packet switched multimedia terminals shall support (mandatory) AMR narrowband speech codec [3GPP01b]. 3GPP also predicts the possibility of transmitting real time text, according to ITU-T T.140 [3GPP01b].

In [Heym97] a video source model based on observed video traffic statistical features for video conference services has been proposed, represented as the Gamma Beta Auto-Regressive (GBAR) Model. The GBAR model [Heym97] is a first order auto-regressive process that relies on two statistical features observed in H.261 and H.263 [LaGD00] Variable BitRate (VBR) video conferencing traffic:

- the marginal distribution follows a Gamma distribution
- the auto-correlation function is geometric.

Towards defining the GBAR model, let Ga(β,λ) denote a random variable with a Gamma distribution with shape parameter β and scale parameter λ; its PDF (Probability Density

Function) is given by:

$$p_G(x) = \frac{\lambda(\lambda x)^{\beta}}{\Gamma(\beta+1)} e^{-\lambda x}, \quad x > 0, \tag{3.5}$$

where $\Gamma(\cdot)$ is the Gamma function. Similarly, let Be(p,q) denote a random variable with a Beta distribution with parameters p and q, then its PDF is given by:

$$p_B(x) = \frac{\Gamma(p+q)}{\Gamma(p+1)\Gamma(q+1)} x^{p-1}(1-x)^{q-1}, \quad 0 < x < 1, \tag{3.6}$$

where p and q are both larger than –1. The GBAR model is based on two well known results: the sum of independent Ga(α,λ) and Ga(β,λ) random variables is a Ga($\alpha+\beta$,λ) random variable, and the product of independent Be(α,β-α) and Ga(β,λ) random variables is a Ga(α,λ) random variable. Thus, if X_{n-1} is a Ga(β,λ), A_n is Be(α,β-α), and B_n is Ga(β-α,λ), and these three are mutually independent, then

$$X_n = A_n X_{n-1} + B_n \tag{3.7}$$

defines a stationary stochastic process $\{X_n\}$ with a marginal Ga(β,λ) distribution.

Simulating the GBAR process (GBAR(β,λ)) only requires the ability to simulate independent and identically distributed Gamma and Beta random variables. Parameters β and λ can be estimated from the mean and variance of marginal distribution of sample data as follows. The mean and variance of a Ga(β,λ) distribution are β/λ and β/λ^2, respectively. Let μ and v be the mean and variance of the sample data, then parameters β and λ are given by:

$$\lambda = \frac{\mu}{v} \quad \text{and} \quad \beta = \frac{\mu^2}{v}. \tag{3.8}$$

The parameter α can be calculated by [Heym97]:

$$\alpha = \rho \cdot \beta, \tag{3.9}$$

where ρ is the lag 1 autocorrelation coefficient of the sample data (estimated directly from the data statistics).

The GBAR process is used as a source model by generating non-integer values from (3.7), and then rounded to the nearest integer.

Being an autoregressive model that only requires a previous sample for generating the next one, the GBAR model does not consider the co-existence of LRD (Long-Range Dependence) and SRD (Short-Range Dependence) [LaGD00]. However, it accounts for observed Gamma PDF and the geometric decay in the auto-correlation function at large lags (*i.e.*, autocorrelation function intervals) when applied to video conferencing traffic.

In order to obtain possible realistic values for the parameters of the GBAR model (for simulation purposes), statistical parameters where calculated based on a real trace from a VBR video sequence (Office Cam: office camera observing person in front of a terminal) of 45 min encoded in H.263 format. The video sequence data that was used is described in [FiRe01] and the corresponding trace library is available at http://www.eas.asu.edu/trace. The digital frame format image adopted in the trace is QCIF (Quarter Common Intermediate Format) with 176×144 pixels, since it is expected that handheld wireless devices of next-generation wireless systems will typically have a screen size that corresponds to the QCIF video format [FrNg00]. A fixed reference frame rate of 25 frames/s was used. No target bitrate was specified for the encoding resulting in a VBR sequence. This VBR sequence can be used, for example, for simulation of videoconferences for HIPERLAN/2 type terminals.

The calculated values for the parameters of the GBAR source model from the analysed VBR video sequence are presented in Table 3.14.

VBR Video Sequence	μ **[byte]**	$\sqrt{v}$ **[byte]**	ρ
Office Cam	903.8	327.3	0.943

Table 3.14. Parameter values for the GBAR source model.

In [NyJO01], a video traffic model that mimics an H.263 codec with constant frame rate and specified target bitrate is proposed. The considered target bitrates are assumed to be suitable for mobile access (*e.g.*, 32 kbps) in, *e.g.*, the GSM/EDGE Radio Access Network (GERAN) and the UTRAN. The proposed model is based on a simple linear model with just a few parameters. The parameters are state dependent and the states are controlled by a Markov chain.

The traffic model generates video frames at a certain specified rate. The size of the video frame after coding is dependent on the target bitrate of the data connection.

Frames are generated at equidistant time steps given by the codec frame rate. The first video frame is large (standalone picture or *I*-frame) and its size is given by the parameter f_0. It might be followed by one or more time steps without any frames ("empty frames"). The remaining sequence of frames (predicted inter-frames or *P*-frames), denoted k=1,2,…, is essentially given by a linear time series with state dependent parameters. States of the frames are governed by a Markov chain, specified by the number of states N_s and the state transition probabilities in the transition probability matrix

$$P = \left(p(i,j)\right)_{i,j=1}^{N_s} \tag{3.10}$$

that governs transitions from frame to frame. The state at time step k is denoted by $J(k)$. The initial state $J(1)$ is fixed and represented by the parameter s_0.

The frame sizes after the initial frames are generated by a state dependent time series model with three components: the expected frame size; a feedback term taking cumulative deviations from expected frame sizes into account; a white noise term that induces random variation. The feedback term, which probably captures the impact of target bitrate control during encoding, will force the sequence of cumulative frame sizes not to deviate too much from the expected cumulative sum. Correspondingly, the time series model has three state dependent parameters:

- μ_j: the expected frame size in state j
- b_j: feedback parameter
- σ_{wj}: standard deviation of white noise ("innovations").

Given that the state for frame k, $J(k)$, equals j, the preliminary (to be adjusted below) frame size is given by

$$X(k) = \mu_j - b_j \cdot \left(S(k-1) - M(k-1)\right) + \sigma_{wj} \cdot W(k),\ k = 1,2,\dots. \tag{3.11}$$

The quantity $S(k)$ is the sum of the previous frame sizes (including the initial frame) and $M(k)$ is the sum of the expected frame sizes up to frame k. $W(k)$ is white noise that induces random deviations. The distribution of the innovations $W(k)$ is chosen to be the Normal one with mean zero and variance one. The Normal PDF is given by [Ross87]:

$$p_N(x) = \frac{1}{\sqrt{2\pi}\sigma_N} e^{-\frac{1}{2}\left(\frac{x-\mu_N}{\sigma_N}\right)^2} \tag{3.12}$$

where σ_N is the standard deviation, and μ_N is the mean.

Initial values are determined by the initial state $J(1)=s_0$, the first frame size f_0 and the number of initial frames n_{init} (the first large frame and empty frames), *i.e.*,

$$S(0) = f_0 \tag{3.13}$$

and

$$M(0) = n_{init} \cdot \mu_{s_0} . \tag{3.14}$$

In order to keep frame sizes within certain bounds, two state dependent variables are introduced: $f_{\min,j}$ and $f_{\max,j}$ which are used as the lower and upper bound respectively to adjust frame sizes when they are generated using (3.11). The adjusted frame size $X'(k)$ is then given by

$$X'(k) = \min\left(\max\left(f_{\min,j}, X(k)\right), f_{\max,j}\right). \tag{3.15}$$

Equation (3.15) is applied after (3.11) for every time step before calculating the next frame size. Note that $S(k)$ should be calculated using the adjusted frame sizes, *i.e.*,

$$S(k) = S(k-1) + X'(k). \tag{3.16}$$

Correspondingly

$$M(k) = M(k-1) + m_{J(k)} . \tag{3.17}$$

The following tables provide an example of parameter settings for the described model [NyJO01]. The parameters are fitted to an H.263 encoded video with target bitrate of 32 kbps. Parameter values correspond to frame sizes and packet sizes in bytes. Table 3.15 gives basic parameters while Table 3.16 gives the state dependent parameters. The first column of Table 3.16 shows the states and the corresponding steady-state probabilities to be in these states. The two last columns show lower and upper bounds of frame sizes. The state transition probability matrix P, is given in Table 3.17.

Although the model proposed in [NyJO01] is aimed for low bitrate streaming video applications, one can (as an approximation) consider it also applicable for video telephony applications in mobile wireless systems, by using the model for both transmission ways (uplink and downlink).

Frame rate, r_{frame}	7.5 frames/s
Target bitrate	32 kbps
Duration	120 s
Initial frame size, f_0	2433 bytes
Number of initial frames (first and empty), n_{init}	3
Initial state, s_0	5
Number of states, N_s	10
Innovation distribution	Normal (0,1)

Table 3.15. Basic parameters, 32 kbps video model (extracted from [NyJO01]).

State	Prob.	μ_j	$(\sigma_{wj})^2$	b_j	$f_{\min}$	$f_{\max}$
1	1.1 %	104	179.4	1.06	24	745
2	0.3 %	198	91.53	1.67	24	545
3	1.1 %	550	21.57	1.67	461	634
4	13.9 %	566	66.71	1.67	220	1424
5	9.7 %	533	132.6	1.07	105	1097
6	3.3 %	548	212.9	1.02	84	1213
7	7.5 %	574	20.19	1.02	468	696
8	44.8 %	576	63.99	1.14	264	854
9	17.2 %	576	115.4	1.19	115	1075
10	1.1 %	534	283.5	0.94	176	1576

Table 3.16. State dependent parameters, 32 kbps video model (extracted from [NyJO01]).

$$
\begin{bmatrix}
0.90 & 0.10 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\
0 & 0.67 & 0 & 0 & 0.33 & 0 & 0 & 0 & 0 & 0 \\
0 & 0 & 0.90 & 0 & 0.10 & 0 & 0 & 0 & 0 & 0 \\
0 & 0 & 0 & 0.915 & 0 & 0 & 0.023 & 0.054 & 0.008 & 0 \\
0 & 0 & 0 & 0.011 & 0.901 & 0.011 & 0.011 & 0.033 & 0.0033 & 0 \\
0 & 0 & 0 & 0 & 0 & 0.900 & 0 & 0.033 & 0.067 & 0 \\
0 & 0 & 0 & 0.014 & 0 & 0 & 0.914 & 0.043 & 0.029 & 0 \\
0 & 0 & 0 & 0.019 & 0.007 & 0 & 0 & 0.955 & 0.017 & 0.002 \\
0.006 & 0 & 0.006 & 0.006 & 0.025 & 0.013 & 0.013 & 0.025 & 0.906 & 0 \\
0 & 0 & 0 & 0 & 0 & 0 & 0 & 0.10 & 0 & 0.90
\end{bmatrix}
$$

Table 3.17. State transition probability matrix (extracted from [NyJO01]).

In [RaMe01] and [LaOR98], a general traffic model for the video signal in a packet switched videoconference is described, which is also applicable to digital video signals generated according to ITU-T H.261.

3.2.3 Streaming type traffic models

Streaming type services are characterised by a continuous flow of data capable of supporting substantially higher end-to-end transmission delays and fluctuation (jitter) than for conversational type services. The nature of the traffic in streaming services is clearly non-symmetric, since it refers to the download and simultaneous processing of huge data files, such as video. Furthermore, a larger fluctuation of packet arrival time is allowed although the use of intermediate buffers at the reception allows a nearly full cancellation of this fluctuation, assuring a constant data rate to the applications. Examples of this type of traffic are: Video on Demand and MP3 streaming.

Video traffic can be transported either with a Constant BitRate (CBR) or with a VBR. In fact, video traffic, in compressed form, is expected to be the prime example of VBR traffic type. VBR video has several potential advantages over traditional CBR video: improved image quality and shorter delay [ChRe99]. In addition, through statistical multiplexing, improved channel allocation may be obtained compared to CBR transport.

MPEG-4 and H.263 encoded video is expected to account for large portions of traffic in future wireline and wireless networks [FiRe01]. In particular, 3GPP has specified the use of MPEG-4 and H.263 codecs for video streaming services in UMTS [3GPP02c]. For the associated speech and/or audio signals, 3GPP has specified the mandatory use of AMR narrowband speech codecs and/or MPEG-4 AAC (Advanced Audio Coding) Low Complexity codecs, respectively [3GPP02c]. In the present work a model for MPEG-4 VBR video streaming will be considered, as described below.

Numerous traffic models incorporating the characteristics of VBR (compressed) video have been proposed, like those described in [RaMe01], [ChRe99], [KoPP01b], [MASK88], [LaOR98], [Mand01], [FrNg00], [ILDK96], [HaCS01] and [KrTr97]. A survey of statistical source models for VBR video can be found in [IzRe99].

In the present work, one will describe the source model presented in [FrNg00] – the GOP GBAR Model – for characterisation of VBR MPEG video sources. Firstly a brief overview of MPEG-4 coding will be presented, followed by the source model description. At the end of this section, estimated parameters for the GOP GBAR model will be provided based on real MPEG-4 trace statistics.

MPEG-4 provides very efficient video coding covering the range from the very low bitrates of wireless communications to bitrates and quality levels beyond High-Definition TeleVision (HDTV). In contrast to the frame-based video coding of H.263 (and MPEG-1), MPEG-4 is object based. Each scene is composed of Video Objects (VOs) that are coded individually (in particular, the entire scene can be composed by only one VO). Each VO may have several scalability layers, which are referred to as Video Object Layers (VOLs). Each VOL in turn consists of an ordered sequence of snapshots in time, referred to as Video Object Planes (VOPs). For each VOP the encoder processes the shape, motion, and texture characteristics. There are 3 types of VOPs: intracoded (*I*), forward predicted (*P*), and bidirectionally predicted (*B*). The *I* and *P* VOPs are also referred to as anchor VOPs. The *I*, *P*, and *B* VOPs are arranged in a periodic pattern referred to as a Group Of Pictures (GOP). This periodic pattern is usually described as a (*N*,*M*) cyclic GOP, where *N* is the spacing between successive *I* VOPs, and *M* is the spacing between successive anchor VOPs. For example, a typical GOP structure, for a 25 Hz frame rate video on-demand, is *IBBPBBPBBPBB* ((12,3) cyclic GOP). When considering a single VOP per snapshot, one can use the traditional concept of frame instead of VOP (*e.g.*, *I*-, *P*-, and *B*-frames) – this approach will be used in the description of the GOP GBAR model. To combat the frequent transmission errors typical in wireless communications, MPEG-4 provides a number of error resilience and error concealment features.

The GOP GBAR model [FrNg00], is a generalisation of the GBAR model for video conferencing presented in Section 3.2.2.4. The GBAR model is only intended to apply to video conferencing and is not suited to more general MPEG video sources, since it does not account for the GOP cyclicity present in typical MPEG video. The GOP GBAR model constitutes a MPEG video source model with the features of the GBAR model, while explicitly accounting for the presence of GOP cyclicity.

According to the GOP GBAR model [FrNg00], the size X_k of the k^{th} frame in an MPEG-encoded video sequence starting with an I-frame and using a (*N*,*M*) cyclic GOP is given by

$$X_k = \begin{cases} \lambda_1 Z_{1k} + \lambda_2 Z_{2k} + \lambda_3 Z_{3k}, & \text{if } k = 1 \bmod N, \\ \lambda_1 Z_{1k} + \lambda_2 Z_{2k}, & \text{if } k \neq 1 \bmod N \text{ but } k = 1 \bmod M, \\ \lambda_1 Z_{1k,} & \text{otherwise} \end{cases} \tag{3.18}$$

where $\{Z_{1k} = 0,1,...\} \sim \text{GBAR}(\alpha 1, \rho 1)$, $\{Z_{2k} = 0,1,...\} \sim \text{GBAR}(\alpha 2, \rho 2)$, and $\{Z_{3k} = 0,1,...\} \sim \text{GBAR}(\alpha 3, \rho 3)$ are independent stationary GBAR processes. The GOP GBAR model (3.18) has a composite geometric correlation structure made up of the geometric autocorrelations of the three component GBAR processes $\{Z_{1k}\}$, $\{Z_{2k}\}$, and $\{Z_{3k}\}$. The GOP GBAR model (3.18) has therefore nine parameters, easily estimated from sample statistics, given by:

$$
\begin{aligned}
&\lambda_1 = \frac{\sigma_B^{\,2}}{\mu_B}, \quad \lambda_2 = \frac{\sigma_P^{\,2} - \sigma_B^{\,2}}{\mu_P - \mu_B}, \quad \lambda_3 = \frac{\sigma_I^{\,2} - \sigma_P^{\,2}}{\mu_I - \mu_P} \\
&\alpha_1 = \frac{\mu_B^{\,2}}{\sigma_B^{\,2}}, \quad \alpha_2 = \frac{(\mu_P - \mu_B)^2}{\sigma_P^{\,2} - \sigma_B^{\,2}}, \quad \alpha_3 = \frac{(\mu_I - \mu_P)^2}{\sigma_I^{\,2} - \sigma_P^{\,2}} \\
&\rho_1 = \rho_B, \quad \rho_2 = \left(\frac{\sigma_P^{\,2} \rho_P^{\,M} - \sigma_B^{\,2} \rho_B^{\,M}}{\sigma_P^{\,2} - \sigma_B^{\,2}} \right)^{1/M} \\
&\rho_3 = \left(\frac{\sigma_I^{\,2} \rho_I^{\,N} \left(\sigma_P^{\,2} - \sigma_B^{\,2}\right) \rho_2^{\,N} - \sigma_B^{\,2} \rho_B^{\,N}}{\sigma_I^{\,2} - \sigma_P^{\,2}} \right)^{1/M}
\end{aligned}
\tag{3.19}
$$

where μ_B, μ_P, and μ_I are the sample means of the *B*-, *P*-, and *I*-frame sequences, respectively, and σ_B^2, σ_P^2, and σ_I^2 are the corresponding sample variances. ρ_B, ρ_P, and ρ_I are the estimated lag 1 autocorrelation coefficients of the *B*-, *P*-, and *I*- frame sizes, respectively.

The GOP GBAR model presents several advantages: it is analytically tractable, it has a small number of easily estimated parameters, and it is implemented through simple algebraic expressions involving independent standard Gamma variables.

In order to obtain possible realistic parameters for simulation purposes, statistic parameters where calculated based on real traces from two video sequences (the well known "Jurassic Park I" and "Silence of the Lambs") of 60 min encoded in MPEG-4 format. The video sequences data that was used is described in [FiRe01] and the corresponding trace library is available at http://www.eas.asu.edu/trace. The digital frame format image adopted in the traces is QCIF with 176×144 pixels, for the same reasons presented in 3.2.2.4. For the MPEG-4 encoding, the number of video objects was set to one (*i.e.*, the entire scene is one video object). The single video object was encoded into a single video object layer, and the object layer frame rate (*i.e.*, the rate at which video object planes are generated) was set to 25 frames/s. The GOP pattern was set to *IBBPBBPBBPBB*. For the present work purposes, two different encoding quality levels of each video sequence are considered: low quality, corresponding to quantisation parameters of 10 for *I* frames (VOPs), 14 for *P* frames, and 18 for *B* frames; and high quality, corresponding to quantisation parameters of 4 for all three frame types. As an example, the low quality can be foreseen for UMTS type terminals and the high quality for HIPERLAN/2 type terminals.

The calculated values for the parameters of the GOP GBAR model for the four analysed sequences are presented in Table 3.18.

Film	Quality	μ_B [byte]	μ_P [byte]	μ_I [byte]	σ_B [byte]	σ_P [byte]	σ_I [byte]	ρ_B	ρ_P	ρ_I
Jurassic Park I	Low	334.6	966.8	3645.8	297.1	666.1	1311.7	0.936	0.825	0.940
	High	3176.7	4461.3	7179.9	1793.4	2240.6	2147.9	0.955	0.891	0.935
Silence of the Lambs	Low	246.9	641.6	2435.5	338.5	709.9	1573.7	0.960	0.888	0.943
	High	2429.4	3203.2	5471.3	1929.7	2406.4	2678.2	0.979	0.937	0.942

Table 3.18. Parameter values for the GOP GBAR model.

3.2.4 Interactive type traffic models

Interactive type traffic is generated when a user performs an information request to a remote equipment, typically a server. A typical example of this type of traffic is the access to a web server (Web Browsing).

Several models have been proposed for Web browsing traffic, like those described in [RaMe01], [DiLS01], [MWIF01], [JeJe01], [ShRJ02], [ETSI98], [KlLL01], [FäBC98], [Vica97] and [ZSMH00], usually based on measured trace data. In the present report, one will describe the model adopted by 3GPP for UMTS systems as presented in [RaMe01] and [ETSI98].

The model described in [RaMe01] and [ETSI98] considers a sequence of packet calls during a WWW browsing session, Figure 3.5. The user initiates a packet call when requesting an information entity. During a packet call, several packets may be generated, constituting a bursty sequence of packets.

WWW browsing sessions are basically unidirectional (DL). In these types of sessions, a packet call corresponds to the downloading of a WWW document (*e.g.*, web page with images, text, applets, etc.). After the document is downloaded, a reading time interval (by the user) is experienced.

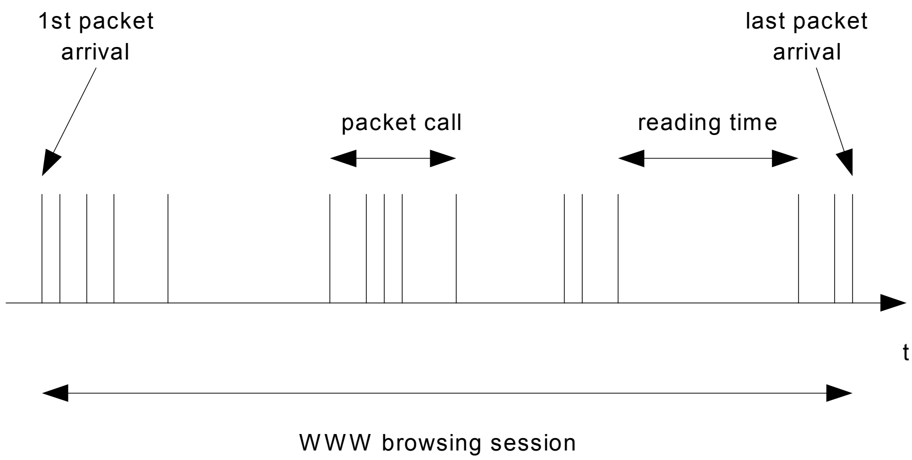

Figure 3.5. Typical WWW session (adapted from [ETSI98]).

The following parameters must be modelled in order to catch the typical behaviour described in Figure 3.5:

- Session arrival process
- Number of packet calls per session, N_{pc}
- Reading time between packet calls, D_{pc}
- Number of packets within a packet call, N_d
- Interarrival time between packets (within a packet call), D_d
- Size of a packet, S_d.

Note that the number of events during the session models the session length implicitly.

These six different events are modelled in [ETSI98] and described in the following paragraphs. The geometrical distribution is used (discrete representation of the exponential distribution), for discrete time scale simulation purposes.

- **Session arrival process:** The arrival of session set-ups to the network is modelled as a Poisson process (refer to Section 3.2.7). For each service there is a separate process. It is important to note that this process for each service only generates the time instants when service calls begin, and is not related in any way to call termination.
- **Number of packet call requests per session**, N_{pc}: This is a geometrically distributed random variable, with mean μ_{Npc}, where the probability of generating one packet call, P_{PC} is given by:

$$P_{PC} = \frac{1}{\mu_{Npc}} . \tag{3.20}$$

The probability, $p_{Ge}[n]$, of generating n packet call requests during a session is therefore given by the following Probability Function (PF):

$$p_{Ge}[n] = P_{PC} \cdot (1 - P_{PC})^{n-1} . \tag{3.21}$$

- **Reading time between two consecutive packet call requests in a session**, D_{pc}: This is a geometrically distributed random variable, with mean μ_{Dpc}. Note that the reading time starts when the last packet of the packet call is completely received by the user. The reading time ends when the user makes a request for the next packet call.
- **Number of packets in a packet call**, N_d: Different statistical distributions can be used to generate the number of packets. For example, N_d can be a geometrically distributed random variable, with mean μ_{Nd}. It must be possible to select the statistical distributions that describe best the traffic case under study. An extreme case would be that the packet call contains a single large packet.
- **Time interval between two consecutive packets inside a packet call**, D_d: This is a geometrically distributed random variable, with mean μ_{Dd}. Naturally, if there is only one packet in a packet call, this variable is not needed.
- **Packet size**, S_d: The packet size is defined as $S_d = \min(P_a, M_{ps})$, where P_a is a Pareto distributed random variable (based on Table 3.19), and M_{ps} is the maximum allowed packet size (66 666 bytes). The packet size distribution model is therefore based on a Pareto distribution with cut-off. The normal Pareto PDF (without cut-off) is given by [ETSI98] (k corresponds to the minimum packet size):

$$p_P(x) = \begin{cases} 0, x < k \\ \dfrac{\alpha_p \cdot k^{\alpha_p}}{x^{\alpha_p + 1}}, x \geq k \end{cases} . \tag{3.22}$$

Table 3.19 gives default mean values for the distributions of typical WWW browsing services [ETSI98]. According to the values for α_p and k in the Pareto distribution, the average packet size μ_{Sd} is 480 bytes. Therefore, the average requested file size is $\mu_{Nd} \cdot \mu_{Sd}$ = 25×480 bytes =12 kbytes. The interarrival time is adjusted in order to get different average bitrates at the source level.

Packet based information types (WWW browsing)	μ_{Npc}	μ_{Dpc} **[s]**	μ_{Nd}	μ_{Dd} **[s]**	**Parameters for S_d distribution**
UDD 8 kbps	5	4-12	25	0.5	$k = 81.5$ $\alpha_p = 1.1$
UDD 32 kbps				0.125	
UDD 64 kbps				0.0625	
UDD 144 kbps				0.0277	
UDD 384 kbps				0.0104	
UDD 2048 kbps				0.00195	

Table 3.19. Parameter values for web browsing model (extracted from [ETSI98]).

3.2.5 Background type traffic models

Background type traffic corresponds typically to data transfer applications that do not demand an immediate response/reaction from the user. Typical examples of applications that generate this type of traffic are: Email, FTP, music download (*e.g.*, Napster) and Telnet.

Several traffic models for this type of traffic have been proposed [RaMe01], [KoPP01a], [KlLL01], [KoPP01b]. In the present report, the models proposed in [KlLL01], which are synthetic traffic models for UMTS based on measured trace data from the University of Dortmund ISP, will be described.

The traffic model proposed in [KlLL01] utilizes the notion that a user who runs non real-time applications (*e.g.*, HTTP, Napster, Email), follows a characteristic usage pattern and that a single user may run different applications that may be concurrently active, *e.g.*, WWW browsing while downloading Napster music files.

Each application is completely described by its statistical properties, which comprise of an alternating process of ON and OFF periods with some application specific length or data volume distribution, respectively, Figure 3.6. Moreover, within each ON-period the packet arrival process is completely captured by the packet interarrival times and the corresponding packet sizes.

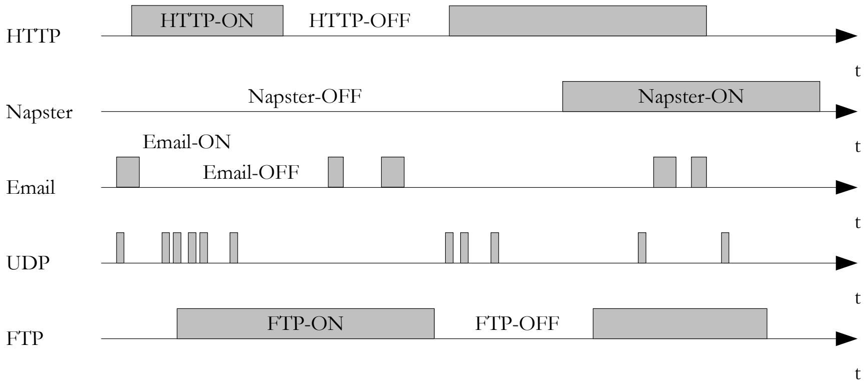

Figure 3.6. Classification of non-real time IP traffic streams (extracted from [KlLL01]).

In this model, the single user traffic model is employed on three different levels:

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

The following tables, Table 3.20 to Table 3.24, summarise the results presented in [KlLL01], for Email and FTP services.

The Lognormal PDF is given by [Ross87]:

$$p_{LN}(x) = \frac{1}{\sqrt{2\pi}\sigma_{LN} x} e^{-\frac{1}{2}\left(\frac{\ln(x)-\mu_{LN}}{\sigma_{LN}}\right)^2}, x > 0, \tag{3.23}$$

where σ_{LN} is the natural log of the standard deviation and μ_{LN} is the natural log of the mean.

Distribution	
Lognormal (μ_{LN}; σ_{LN}^2)	(0.9681; 4.3846)

Table 3.20. Distribution of session interarrival times (extracted from [KlLL01]).

Distribution	64 kbps	144 kbps	384 kbps
Lognormal (μ_{LN}; σ_{LN}^2)	(11.1170;1.9095)	(11.4107;1.9509)	(11.6795;14.4979)

Table 3.21. Distribution of session volume (extracted from [KlLL01]).

		Distribution	64 kbps	144 kbps	384 kbps
Email	Interarrival time	Pareto (k; a_p)	(14.4360;2.1345)	(15.1334;2.1254)	(16.0229;2.1223)
	Data Volume	Lognormal (μ_{LN}; σ_{LN}^2)	(8.1934;3.3852)	(8.2944;3.5288)	(8.4124;3.6439)
FTP	Interarrival time	single connection within a session			
	Data Volume	Lognormal (μ_{LN}; σ_{LN}^2)	(8.4944;3.6674)	(8.6403;4.1059)	(8.8409;4.3343)

Table 3.22. Statistical properties at connection level (extracted from [KlLL01]).

	Distribution	64 kbps	144 kbps	384 kbps
Email	Lognormal (μ_{LN}; σ_{LN}^2)	(-4.4052;4.4970)	(-4.8790;4.9687)	(-5.4096;5.4978)
FTP	Lognormal (μ_{LN}; σ_{LN}^2)	(-3.2770;5.2887)	(-3.7830;5.6710)	(-4.3020;6.0997)

Table 3.23. Parameters of packet interarrival times (extracted from [KlLL01]).

	Fractions of packets in overall traffic [%]			
	Packet size 40 byte	**Packet size 576 byte**	**Packet size 1500 byte**	**Other packet sizes[3]**
Email	38.25	25.98	9.51	26.26
FTP	40.43	18.08	9.33	32.16

Table 3.24. Fractions of different packet sizes in overall traffic (extracted from [KlLL01]).

3.2.6 Summary table of traffic source models

Table 3.25 presents a summary of the traffic source models described in the previous sections. Input and output parameters are outlined and a short name, for referencing in the following sections/chapters, is provided. Another parameter included in the table is the connection link applicability within a session, *i.e.*, the link to which the source model can be applied within each session (DL or UL). In this parameters' column, some approximations that will be assumed for the following chapters are also presented, regarding the neglecting of traffic in a particular connection direction for some applications (*e.g.*, UL traffic during a WWW session).

[3] The remaining packet sizes are distributed uniformly between 40 bytes and 1500 bytes.

Service Class	Traffic Source Models	Short Name	Input Parameters	Output Parameters	Connection Link Applicability within a Session	Description
Conversational	Classical ON-OFF Voice Model	Classical ON-OFF	Mean "ON" and "OFF" durations.	"ON" and "OFF" states duration.	UL and DL (simultaneously)	Section 3.2.2.1
	CDMA Mobile Systems Voice Model	CDMA MSVM	Packet sizes, probability and mean burst duration for each state. Weibul parameters for each state. (All values based on real data).	Voice packet sizes at constant packet rate.	UL and DL (simultaneously)	Section 3.2.2.2
	IP Telephony (VoIP) Model	VoIP	Mean "ON" and "OFF" durations. IP packet payload and overhead. Transmission time interval. Voice packet generation. IP packet rate.	"ON" and "OFF" states duration. During "ON" state, voice packets are generated according to input parameters.	UL and DL (simultaneously)	Section 3.2.2.3
	VBR H.263 Video Telephony GBAR Model	GBAR H.263 VBR	Sample mean, variance and lag 1 autocorrelation coefficient of frame sizes.	Frame sizes at constant frame rate.	UL and DL (simultaneously)	Section 3.2.2.4
	H.263 Video Telephony with target bitrate	H.263 TBR *n* kbps (*n*=32kbps)	Target bitrate. Frame rate. Initial frame size. Number of initial frames. Initial state. Number of states. Innovation distribution. State dependent parameters. State transition probabilities matrix.	Frame sizes at constant frame rate.	UL and DL (simultaneously)	Section 3.2.2.4

Table 3.25. List of traffic source models.

Service Class	Traffic Source Models	Short Name	Input Parameters	Output Parameters	Connection Link Applicability within a Session	Description
Streaming	VBR MPEG-4 Video GOP GBAR Video Streaming Model	GOP GBAR MPEG-4 *n* Quality (*n*=LOW, HIGH)	Sample means, variances and lag 1 autocorrelation coefficients of B-, P-, and I-frame sizes.	Frame sizes at constant frame rate.	Only DL (no UL traffic considered)	Section 3.2.3
Interactive	Web Browsing Model	WWW *n* kbps (*n*=8, 32, 64, 144, 384, 2048)	Target bitrate and statistical parameters from real trace data.	Number of packet calls per session. Reading time between packet calls. Number of packets within a packet call. Interarrival time between packets within a packet call. Size of packets.	Only DL (no UL traffic considered)	Section 3.2.4
Background	Email Model	Email *n* kbps (*n*=64, 144, 384)	Target bitrate and statistical parameters from real trace data.	Session interarrival times. Session data volume. Connection interarrival times. Connection data volume. Packet Interarrival times. Packet sizes.	UL or DL (depending on connection direction)	Section 3.2.5
	FTP Model	FTP *n* kbps (*n*=64, 144, 384)	Target bitrate and statistical parameters from real trace data.	Session interarrival times. Session data volume. Connection interarrival times. Connection data volume. Packet Interarrival times. Packet sizes.	UL or DL (depending on file transfer direction. Only one file transfer direction considered per session. No traffic considered in opposite direction.)	Section 3.2.5

Table 3.25 (continued). List of traffic source models.

3.2.7 Call/session generation and duration processes

For a complete modelling of traffic sources, one must also characterise call/session arrivals and durations.

Generally, voice calls and data sessions, are considered to be generated according to a Poisson process [ETSI98], [Yaco93], [JeJe01], [ZSMH00], where the probability, $p_{Po}[n,t]$, of generating n calls/session in a certain time interval t is given by:

$$p_{Po}[n,t] = \frac{(\lambda_c \cdot t)^n}{n!} e^{-\lambda_c \cdot t} , \tag{3.24}$$

where λ_c is the mean arrival rate of calls (*e.g.*, calls per second). $p_{Po}[n,t]$ represents the Poisson PF.

In general, it is also assumed that durations for voice calls and data sessions follow an exponential distribution [Yaco93], [MWIF01], [ETSI98]. The corresponding PDF and CDF are given by (3.25) and (3.26), respectively:

$$P_{\mathrm{E}}(t) = 1 - e^{-\mu_{CD} \cdot t} , \tag{3.25}$$

$$p_{\mathrm{E}}(t) = \mu_{CD} \cdot e^{-\mu_{CD} \cdot t} , \tag{3.26}$$

where $1/\mu_{CD}$ is the mean call duration time [Yaco93]. Equation (3.26) is usually referred to as the negative exponential distribution [Yaco93].

In Table 3.26, the generation and duration processes that will be used in the current work for each application are presented. Again, as with the traffic source models, scarce information can be found in the literature for parameterisation of these processes, mainly for advanced services and applications. Therefore, in this work some of these values will be arbitrated, with a reasonable practical sense.

Service Class	Services	Applications	Arrival Rate in Busy Hour	Session/Call duration
Conversational	Speech telephony	Voice Call	Poisson	Exponential
	Video telephony	Video Call	Poisson	Exponential
Streaming	Streaming Video	Video on Demand	Poisson	Exponential
Interactive	Multimedia Communication Service	Web Browsing	Poisson	Given by source model
Background	Messaging Service	Email	Poisson	Given by source model
	Unrestricted Data Retrieval Service	FTP	Poisson	Given by source model

Table 3.26. Session/call generation and duration processes.

3.3 Mapping of technologies versus services and traffic source models

In the previous sections, a complete description and characterisation of services/applications and traffic source models has been presented. In order to complete this study, a full mapping of services/applications, traffic source models and supporting technologies (GSM/GPRS, UMTS, and HIPERLAN/2) must be performed. For this purpose, Table 3.27 has been built, where traffic source models and supporting technologies are selected based on the information type of each service/application.

As it can be seen in Table 3.27, in the present work, the speech and audio components of the Video on Demand applications are not considered. The main reason for using this approach is that no literature was found on source models of audio for these types of applications. Moreover, the amount of data generated by these components can somehow be considered negligible compared to the video one.

Service Class	Services	Applications	Information Type	Traffic Source Model	Technology		
					GSM/GPRS	UMTS	HIPERLAN/2
Conversational	Speech telephony	Voice Call	Sound	Classical ON-OFF	✓	✗	✗
				CDMA MSVM	✗	✓	✗
				VoIP	✗	✗	✓
	Video telephony	Video Call	Sound	Classical ON-OFF	✓	✗	✗
				CDMA MSVM	✗	✓	✗
				VoIP	✗	✗	✓
			Video	H.263 TBR 32 kbps	✗	✓	✓
				GBAR H.263 VBR	✗	✗	✓
Streaming	Video Streaming	Video on-demand	Video	H.263 TBR 64 kbps	✗	✓	✓
				GOP GBAR MPEG-4 (Low Quality)	✗	✗	✓
				GOP GBAR MPEG-4 (High Quality)	✗	✗	✓

Table 3.27. Mapping of technologies versus services and traffic source models.

Service Class	Services	Applications	Information Type	Traffic Source Model	Technology		
					GSM/GPRS	UMTS	HIPERLAN/2
Interactive	Multimedia Communication Service	Web browsing	Multimedia	WWW 8 kbps	✓	✓	✓
				WWW 32 kbps	✓	✓	✓
				WWW 64 kbps	✓	✓	✓
				WWW 144 kbps	✗	✓	✓
				WWW 384 kbps	✗	✓	✓
				WWW 2048 kbps	✗	✗	✓
Background	Messaging Service	Email	Data/ Multimedia	Email 64 kbps	✓	✓	✓
				Email 144 kbps	✗	✓	✓
				Email 384 kbps	✗	✓	✓
	Unrestricted Data Retrieval Service	File Transfer – FTP	Data	FTP 64 kbps	✓	✓	✓
				FTP 144 kbps	✗	✓	✓
				FTP 384 kbps	✗	✓	✓

Table 3.27 (continued). Mapping of technologies versus services and traffic source models.

Chapter 4

Simulator Description

The present chapter provides a functional description of the simulator platform developed within the scope of this thesis, with the main objective of allowing the evaluation of the benefits of convergence of mobile and wireless systems from a capacity/traffic point of view.

4.1 Simulator Overview

For the purpose of evaluating the benefits of convergence of wireless systems, mainly from a capacity/traffic point of view, a complete simulator was implemented. The main goal underlying the development of the simulator was to build a platform enabling the following three main tasks:

- Analysis of the impact of convergence on the radio network's overall performance;
- Evaluation of different convergence perspectives/strategies;
- Analysis of the impact of different traffic distributions in a convergence scenario.

The selected programming language was Visual C++, using object-oriented programming (*e.g.*, each MT is considered as an object). The overall program has approximately 12 000 code lines, and constitutes a flexible and easily upgradeable platform for the simulation of different convergence scenarios.

The simulator is constituted by the following two main functional blocks, Figure 4.1:

- User Traffic Generation Module (UTGM),
- Traffic Processing Engine (TPE).

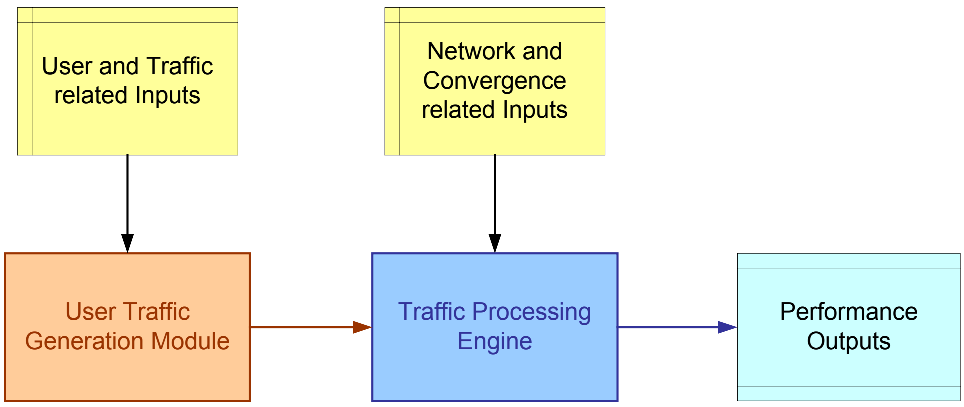

Figure 4.1. General structure of the simulator.

The UTGM is responsible for generating the user traffic for all active users and applications. For each user (or MT), the traffic is generated in the form of vectors, User Traffic Vectors (UTVs), including the following information:

- For CS applications – the beginning instant (in ms) and duration of each session;
- For PS applications – the beginning instant (in ms) of each session, and the beginning instant (in ms) and size (in bytes) of each packet (or frame, for video applications) within a session.

The TPE is the "brain" of the simulator, receiving and processing the UTVs, according to different convergence perspectives/strategies. For this purpose, Access Technology Traffic Processing Modules (ATTPMs) are implemented inside it, for simulating several different access technologies available to the users. The TPE's outputs, corresponding to the simulator outputs, include the overall performance indicators that allow the evaluation of convergence benefits.

Both the UTGM and TPE receive several input parameters from the simulator's user, represented in Figure 4.1 as yellow boxes, allowing the simulation of different user, applications and technological scenarios.

For simplicity reasons, the simulator only considers downlink user traffic processing, disregarding the uplink components. From a capacity/traffic analysis point of view, this approach is quite realistic since:

- nearly all of the considered applications are asymmetric, with a higher capacity demand on the downlink direction,
- in general, the downlink component is capacity limited, in opposition to the uplink components which are usually interference/power limited.

Also for simplicity reasons (and feasibility reasons within a Masters Thesis timeframe), the developed simulation tool does not currently take into account propagation issues, although the simulator already considers the geometry of the simulated scenarios (exact locations and distances between MTs and BSs). This is however an important issue that should be taken into account for the analysis of convergence of systems, since, based on the channel conditions between the MTs and the different access technology BSs, different technology routing decisions can be taken. Therefore, a further improvement of the simulator should address this issue.

Each simulation run corresponds to a complete Busy Hour (BH) simulation, and uses an internal clock with a period of 1ms. The choice of the internal clock period is based on the MAC frame durations of each access technology, presented in Section 4.3.

In the following sections, the UTGM and TPE are described in more detail.

4.2 User Traffic Generation

As mentioned in the previous section, users and their traffic are generated by the UTGM. The UTGM's functional process is depicted in Figure 4.2, and described in the following paragraphs.

The UTGM initially generates a user-defined (simulator input) number of users (MTs). Each user is then given a user type classification. A user type corresponds to a particular user profile (simulator input), with specific traffic usage characteristics, namely: Busy Hour Call/session Attempts (BHCA) and average call/session durations per application. In the present simulator implementation, three different user types were considered (User Type 1, 2 and 3), although other user types can easily be added. The allocation of a specific user type to each user is randomly performed, and depends on the penetration of each user type (simulator input).

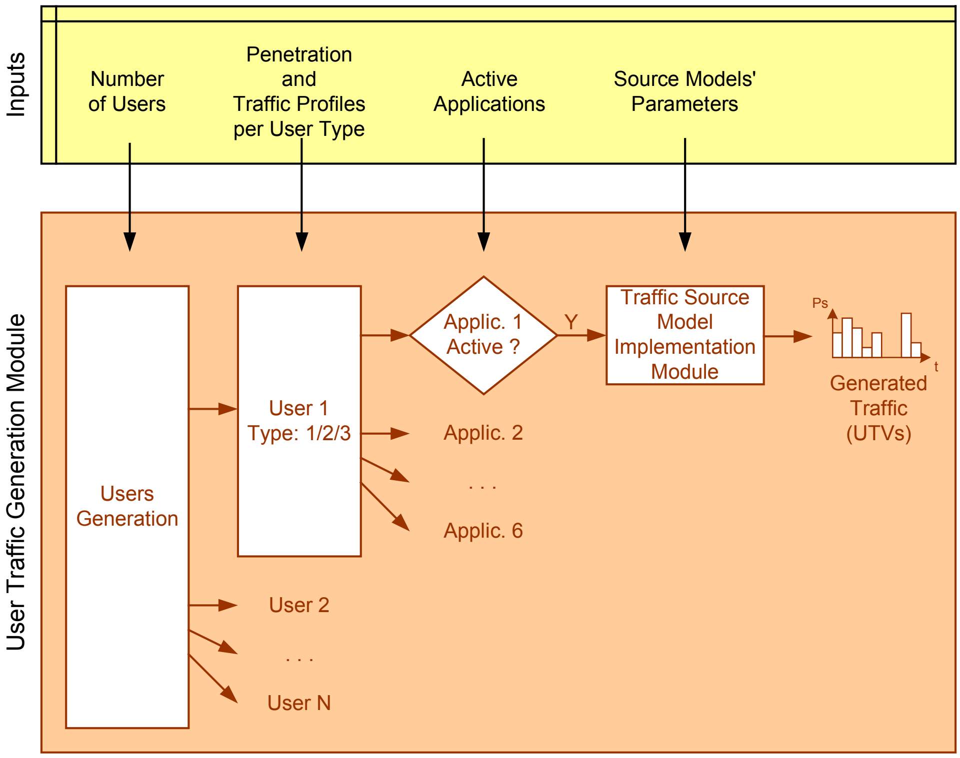

Figure 4.2. UTGM's functional process.

Six different applications (listed in Table 4.1) are supported by the simulator, corresponding to those presented and described in Chapter 3, which can be individually enabled or disabled for analysis of the impact of different traffic mixtures. This is a global setting for all generated users, and therefore, the corresponding UTVs of a given application (per user) will only be generated if that application is enabled.

Applications	**Services**	**Service Class**
Voice Call	Speech Telephony	Conversational
Video Call	Video Telephony	Conversational
Video on Demand	Video Streaming	Streaming
Web Browsing	Multimedia Communication Service	Interactive
Email	Messaging Service	Background
FTP	Unrestricted Data Transfer Service	Background

Table 4.1. Applications supported by the simulator.

The last and most elaborate task of the UTGM is to generate the UTVs for each active application of each user. This task is performed by the Traffic Source Model Implementation Modules (TSMIMs). As previously mentioned, each UTV includes the following information:

- For CS applications – the beginning (in ms) and duration of each session;
- For PS applications – the beginning (in ms) of each session, and the beginning (in ms) and size (in bytes) of each packet (or frame, for video applications) within a session.

The mapping of the supported applications onto the CS and PS types (as a function of the access technologies) that was adopted in the simulator implementation is presented in Table 4.2. As one can observe, only the Voice Call and Video Call (voice component), when transmitted over GSM or UMTS, were considered to be of the CS type, requiring a constant capacity usage during the complete call transmission. In the remaining cases, the applications traffic is processed as of a PS type.

It should be noted that the resulting UTVs generated by the UTGM (for each active application of each user) depend on the users' type, since, as previously mentioned, the user type defines the corresponding BHCA and average call/session durations per application.

TSMIMs have been implemented for each Traffic Source Model (TSM) presented and described in Chapter 3. As a basis for the implementation of the TSMIMs, the following 10 different Random Number Generators (RNGs) had to be initially programmed: Uniform, Beta, Gamma, Exponential, Geometric, Normal, LogNormal, Pareto, Poisson and Weibul. A validation of each of these RNGs is presented in ANNEX I.

Application		**Access Technology**		
		GSM/GPRS	**UMTS**	**HIPERLAN/2**
Voice Call		CS	CS	PS
Video Call	**Voice component**	CS	CS	PS
	Video component	PS	PS	PS
Video on Demand	**Video component**	PS	PS	PS
Web Browsing		PS	PS	PS
Email		PS	PS	PS
FTP		PS	PS	PS

Table 4.2. Classification of Applications (CS and PS) per Access Technology.

In the following paragraphs, example outputs for each of the implemented TSMIMs are presented.

In Figure 4.3, example outputs for two different voice call TSMIMs are presented: CDMA MSVM and VoIP TSMIMs. The CDMA MSVM TSMIM generates 4 different packet sizes (2, 5, 10 and 22 bytes) every 10 ms, while the VoIP TSMIM generates constant sized packets (40 bytes) every 20 ms during ON periods and no packets during OFF periods. For both TSMIMs example outputs, the ON-OFF aspects of a voice call are apparent.

In Figure 4.4, two different example outputs for the video component of video telephony TSMIMs are depicted: H.263 TBR 32 kbps and GBAR H.263 VBR TSMIMs. The speech components are also represented (using the same TSMIMs as for the voice call), in order to illustrate the synchronisation of the beginning and ending of each video call (speech and video components). The H.263 TBR 32 kbps TSMIM supplies the size of each video frame generated at a constant frame rate of 7.5 frames/s. In this TSMIM, the first generated frame is large (standalone picture or I-frame), which is followed by 3 "empty" frames. The GBAR H.263 VBR TSMIM implements a VBR H.263 QCIF video source, generating frames of variable size at a constant frame rate of 25 frames/s.

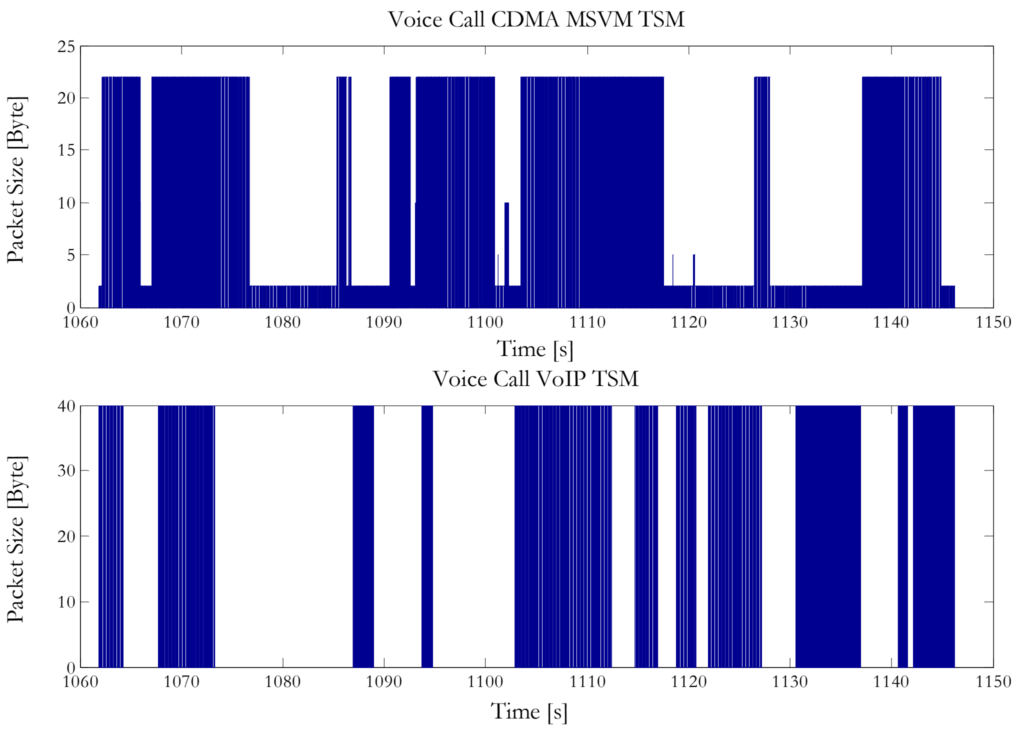

Figure 4.3. Voice Call TSMIMs' Outputs.

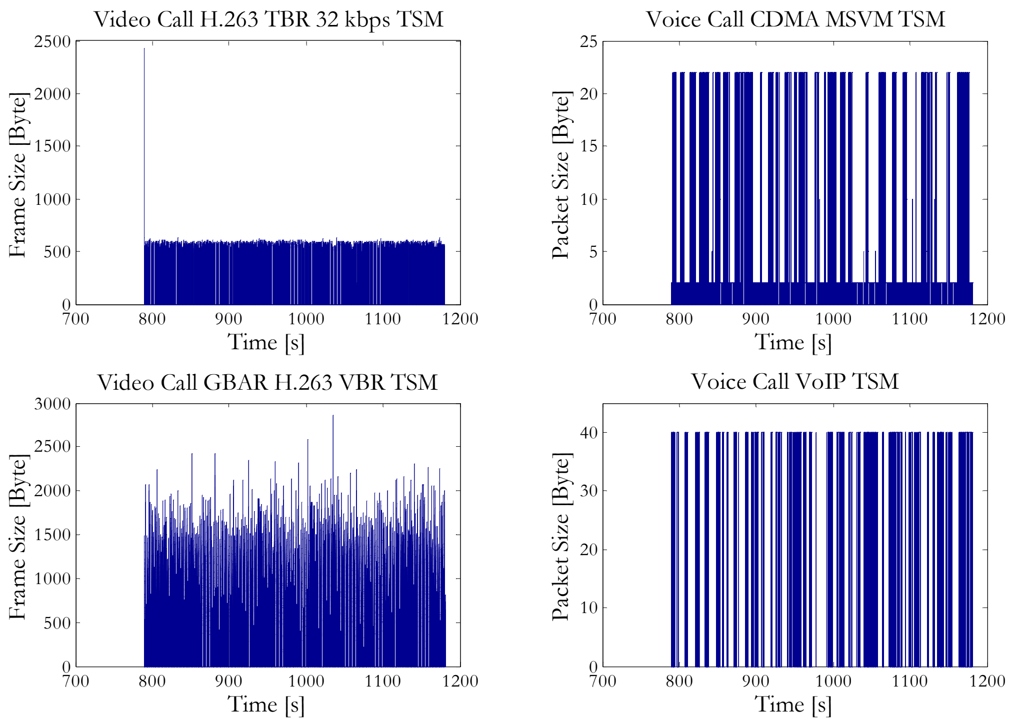

Figure 4.4. Video Call TSMIMs' Outputs.

Two different TSMIMs were implemented for the Video on Demand application, corresponding to the H.263 TBR 64 kbps and the GOP GBAR MPEG-4 TSMs. Figure 4.5 shows an output example of the H.263 TBR 64 kbps TSMIM and two output examples (low quality and high quality) of the GOP GBAR MPEG-4 TSMIM. All three outputs are synchronised (beginning and ending instants). The H.263 TBR 64 kbps TSMIM is based on the H.263 TBR 32 kbps video telephony TSMIM, but with a higher frame rate (15 frames/s). The GOP GBAR MPEG-4 TSMIM generates a VBR MPEG-4 QCIF video streaming source at a constant frame rate of 25 frames/s, based on the parameter settings obtained in Chapter 3 for the film "Silence of the Lambs". One can observe clearly the GOP cyclicity (GOP pattern – IBBPBBPBBPBB) in the GOP GBAR MPEG-4 TSMIM outputs.

For the Web Browsing (WWW) application, six different TSMIMs were implemented, corresponding to six different transmission bitrates (8 kbps, 32 kbps, 64 kbps, 144 kbps, 384 kbps and 2 Mbps). Figure 4.6 shows an output of a synchronised session (beginning and session volume) for all six modules. As expected, one can observe that the session duration depends on the transmission bitrate (lower duration for higher bitrates, since web pages are available faster) and identical reading time intervals between consecutive packet calls (web page requests).

Three TSMIMs were developed for each of the Email and FTP applications (corresponding to a total of 6 different TSMIMs). For each application, the three different TSMIMs correspond to three different transmission bitrates (64 kbps, 144 kbps and 384 kbps). For the implementation of these TSMIMs, slight changes were introduced to the TSMs described in Chapter 3 (for both applications): the same data volume distribution was considered for all three transmission bitrates (corresponding to the data volume presented for the 384 kbps TSMs) and a constant packet interarrival time was considered for each TSM, based on the average packet size and transmission bitrate (63.75 ms for the 64 kbps TSMs, 28.3 ms for the 144 kbps TSMs and 10.625 ms for the 384 kbps TSMs). Figure 4.7 and Figure 4.8, corresponding to the Email and FTP applications respectively, show outputs of a synchronised session (beginning and data volume) for all three TSMIMs of each application. As expected, one can observe that for higher bitrates, the session duration is shorter (for the same data volume).

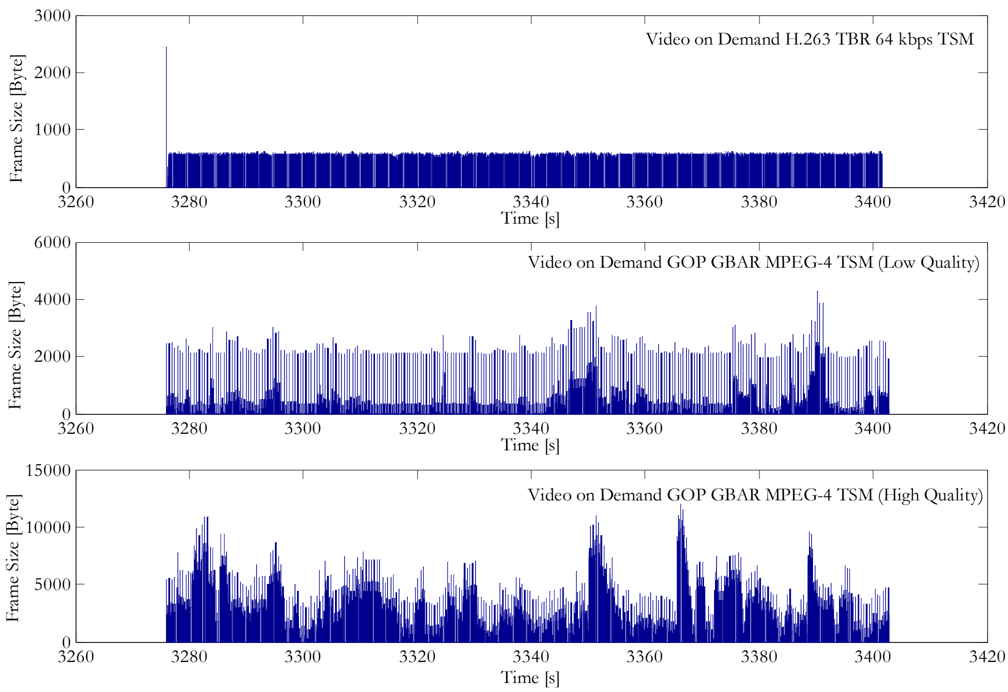

Figure 4.5. Video on Demand TSMIMs' Outputs.

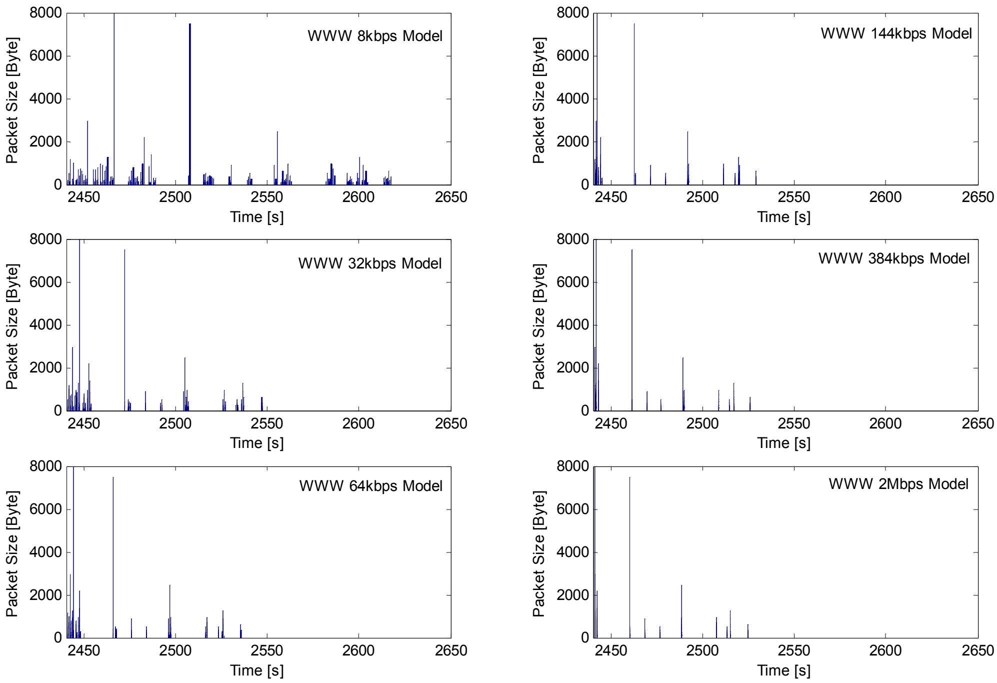

Figure 4.6. Web Browsing (WWW) TSMIMs' Outputs.

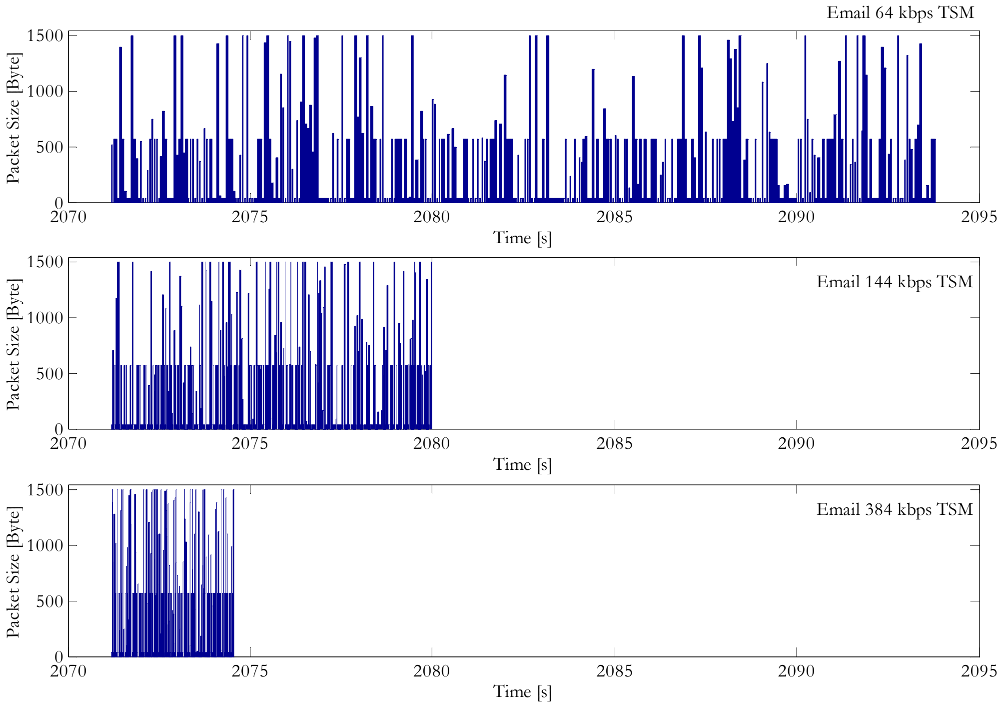

Figure 4.7. Email TSMIMs' Outputs.

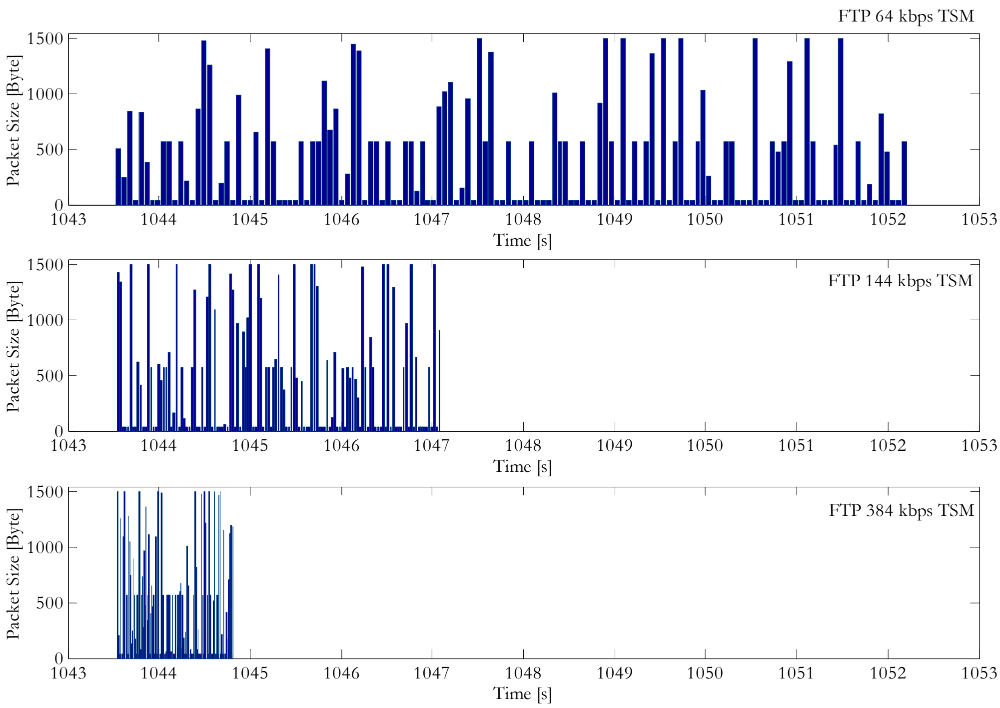

Figure 4.8. FTP TSMIMs' Outputs.

4.3 Traffic Processing Engine

The TPE is the main functional entity of the simulator as it performs all the "convergence" and traffic processing tasks, associated to a real implementation of a converged network. As previously addressed in Chapter 1, one assumes that this converged network is supported by a common IP backbone, offering access to different services via several mobile and wireless technologies.

From a functional point of view, the TPE receives the UTVs generated by the UTGM and processes them according to different convergence perspectives/strategies. The traffic is then routed to the MTs through different access technologies. In the present work, the considered access technologies are: GSM/GPRS, UMTS (FDD Mode), and HIPERLAN/2, which were described in Chapter 2. Each access technology is implemented in the simulator as an ATTPM (Access Technology Traffic Processing Modules). Other ATTPMs can be easily added into the current simulation platform for analysis of convergence of/with other systems (*e.g.*, broadcasting systems, such as DVB-T). All MTs are considered to be "convergence" capable, *i.e.*, they are capable of receiving traffic through all available access technology networks. The main functional building blocks of the TPE and their interactions are depicted in Figure 4.9.

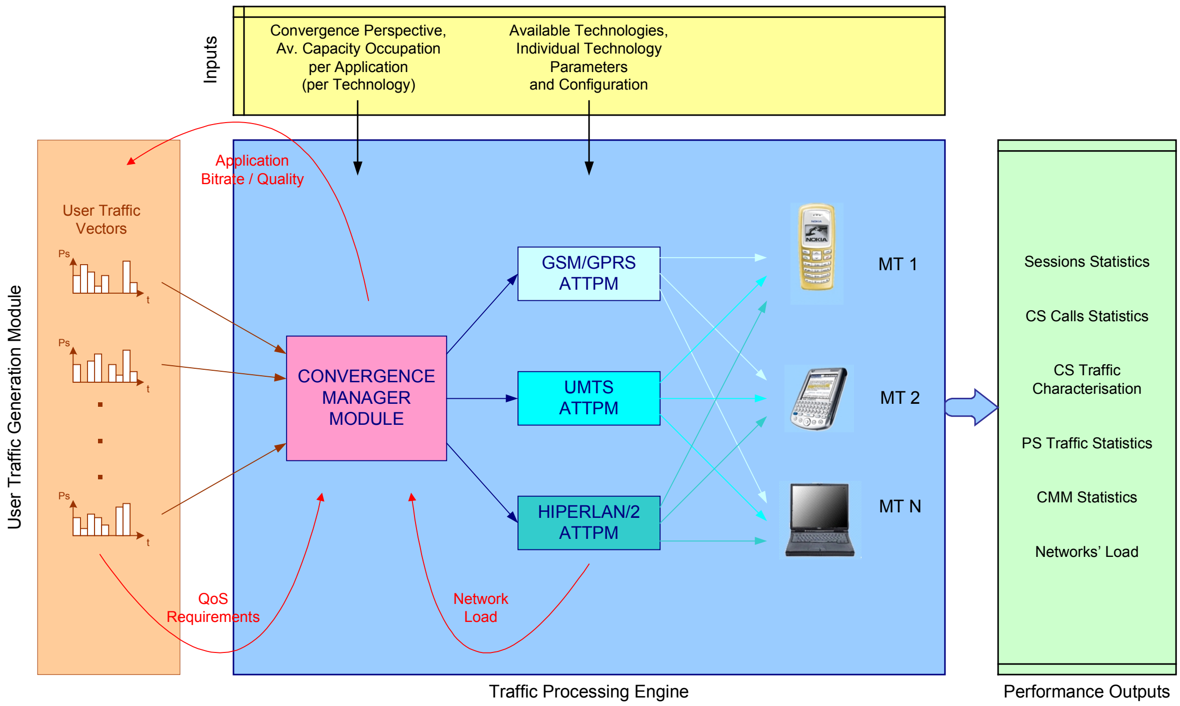

Figure 4.9. Traffic Processing Engine.

The UTVs generated by the UTGM are firstly analysed by a new functional entity here referred to as the Convergence Manager Module (CMM). The main functions of the CMM are to detect new sessions and to route the associated traffic to different ATTPMs according to a user-defined (simulator input) convergence perspective/strategy. For this purpose, the CMM considers variable parameters such as the loads of the different access technology networks and the different application's priorities/QoS requirements. By applying different convergence perspectives/strategies, the different parameters are weighted differently and different benefits are taken into the overall system. The idea behind the CMM is therefore to allow the maximum flexibility among users and networks.

The CMM's functional process is depicted in Figure 4.10. The UTVs are all processed by the CMM before being routed to the different ATTPMs. The CMM constantly monitors the incoming UTVs, performing the following process:

- If a request for beginning of session is detected, the CMM decides to which access technology the traffic must be routed (by the Technology Router - TR), and at what bitrate/quality must the application be delivered (interaction with the application source). The basis for these decisions are user-defined (simulator input) and are represented as a Priority Table (PT). A description of the PT functionalities is presented below. In a real system, all interactions between the CMM and other entities/network elements will have to be performed via new signalling protocols/channels.
- During a session (after CMM decisions) the traffic is constantly routed by the TR to the selected ATTPM and at the selected quality/bitrate.

An alternative for the implementation of the CMM would be, instead of a session based routing, a packet based routing. This approach would however require, in a real system, much more sophisticated signalling and synchronisation protocols.

The PT currently implemented in the simulator is represented in Table 4.3. For each application, the PT defines the priority selection of access technologies and of application bitrate/quality (given by the TSM column). This PT is based on a highest throughput / best system approach. The highest throughput concept means that for data and video applications, the highest available/possible transmission bitrate is selected based on the current networks' loads. In respect to the best system approach, priority is given to the most appropriate application-oriented access technology (*e.g.*, data applications first priority – WLAN; Voice Calls first priority – GSM). These approaches are technical oriented approaches and do not take into account other important/relevant issues such as applications usage costs, user preferences/QoS requirements, network ownerships, etc. Other PTs can be easily tested in the current simulation platform, enabling the evaluation of other convergence perspectives/strategies.

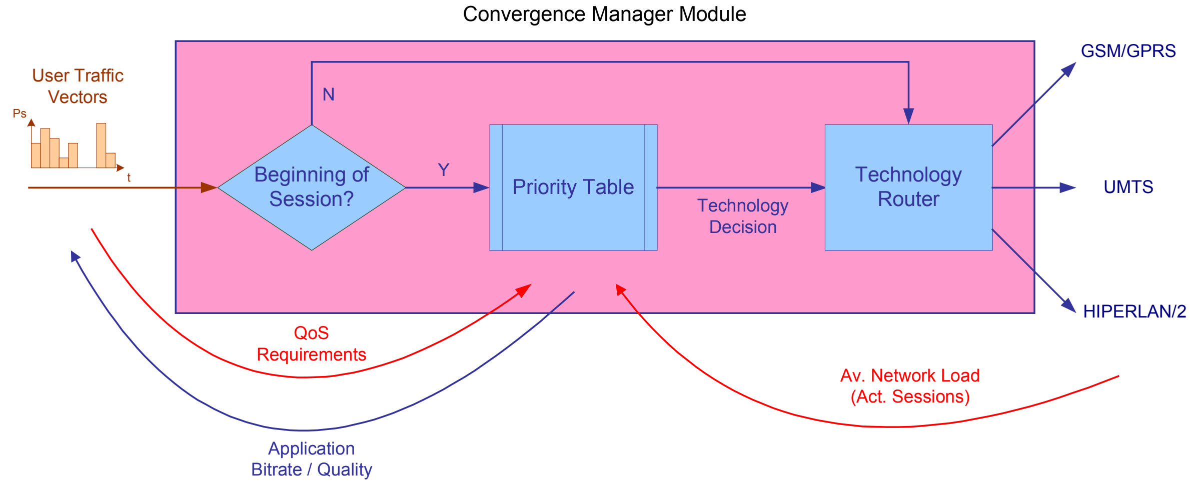

Figure 4.10. Convergence Manager Module.

As it can also be seen in Table 4.3, the voice and video components of the Video Call application are dealt with separately, and even present different technology priorities, in order to optimise the highest throughput / best system concepts for each individual component in a "convergence" scenario.

Other alternative approaches to a single and static PT, as implemented in the simulator, could be:

- a different PT (static or dynamic) per user type, *e.g.*, associated to different monthly price plans per user type, influencing the technology and quality priorities definition;
- a different PT (static or dynamic) per user/MT, addressing a more elaborate scenario where each user can define his own criteria/preferences, based on, for example, cost reasons or quality reasons (*e.g.*, my sons calls always come through and at the best quality).

Application	Priority	Access Technology	Traffic Source Model
Voice Call Video Call (Voice Component)	1	GSM/GPRS	Classical ON-OFF
	2	UMTS	CDMA MSVM
	3	HIPERLAN/2	VoIP
Video Call (Video Component)	1	HIPERLAN/2	GBAR H.263 VBR
	2	UMTS	H.263 TBR 32 kbps
	3	GSM/GPRS	H.263 TBR 32 kbps
Video on Demand (Video Component)	1	HIPERLAN/2	GOP GBAR MPEG-4 HIGH Quality
	2	HIPERLAN/2	GOP GBAR MPEG-4 LOW Quality
	3	HIPERLAN/2	H.263 TBR 64 kbps
	4	UMTS	H.263 TBR 64 kbps
Web Browsing	1	HIPERLAN/2	WWW 2 Mbps
	2	HIPERLAN/2	WWW 384 kbps
	3	HIPERLAN/2	WWW 144 kbps
	4	UMTS	WWW 384 kbps
	5	UMTS	WWW 144 kbps
	6	UMTS	WWW 64 kbps
	7	UMTS	WWW 32 kbps
	8	UMTS	WWW 8 kbps
	9	GSM/GPRS	WWW 64 kbps
	10	GSM/GPRS	WWW 32 kbps
	11	GSM/GPRS	WWW 8 kbps
Email	1	HIPERLAN/2	Email 384 kbps
	2	HIPERLAN/2	Email 144 kbps
	3	UMTS	Email 384 kbps
	4	UMTS	Email 144 kbps
	5	UMTS	Email 64 kbps
	6	GSM/GPRS	Email 64 kbps
FTP	1	HIPERLAN/2	FTP 384 kbps
	2	HIPERLAN/2	FTP 144 kbps
	3	UMTS	FTP 384 kbps
	4	UMTS	FTP 144 kbps
	5	UMTS	FTP 64 kbps
	6	GSM/GPRS	FTP 64 kbps

Table 4.3. Convergence Manager Module - Priority Table.

The main input of the CMM is the current load of each access technology network. These loads are calculated as the sum of the mean capacity occupation of all active sessions associated to each access technology. The session mean capacity occupations (per application and access technology) are fixed internal reference values of the simulator. These values were obtained based on the generation and further analysis of 300 sessions of each application (all TSMs).

The first step in order to calculate the session mean capacity occupations was to obtain the mean session bitrates per application, Table 4.4. Each session bitrate is calculated as the total amount of bytes generated within the session over the total session duration. The mean session bitrates per application are therefore the session bitrates averaged over the considered 300 sessions. As it can be noticed in Table 4.4, the mean session bitrates obtained for applications with a target bitrate do not correspond exactly to their targets (*e.g.*, H.263 TBR 32 kbps). This is due to the fact that the associated TSMs are statistically defined. Another fact that can be noticed, is the mean session bitrates obtained for the WWW application, which are lower than the corresponding target bitrate values. This is an expected result, since the reading times are not accounted for within the target bitrates, *i.e.*, these target bitrates only consider the information transmitting periods (page downloads).

Based on the values listed in Table 4.4, the mean capacity occupation per session of each application was calculated for each considered access technology. Table 4.5 presents the resulting values for each access technology and application, based on:

- GSM/GPRS: average number of TSs occupied per session;
- UMTS: average SF used per session and average DL LF per session;
- HIPERLAN/2: average number of LCHs occupied in average per session.

For each new session, the CMM decision process takes into account the overall load (capacity occupation) of each access technology network, based on the current active sessions (using the values presented in Table 4.5), and verifies the existence of free capacity for the additional session (again considering its mean capacity occupation). This process respects the priorities defined in the PT, and tries to "fit" the new session to the most prioritised available possibility. If there is no capacity for the new session in all technology priorities defined in the PT, the session is routed to the highest priority technology (that is available) and selects the lowest application quality/bitrate.

Application	Traffic Source Model	Mean Session Bitrate [kbps]
Voice Call	CDMA MSVM	7.62
Video Call (Voice Component)	VoIP	8.69
Video Call (Video Component)	H.263 TBR 32 kbps	34.02
	GBAR H.263 VBR	180.88
Video on Demand (Video Component)	H.263 TBR 64 kbps	68.02
	GOP GBAR MPEG-4 LOW Quality	105.95
	GOP GBAR MPEG-4 HIGH Quality	573.39
Web Browsing	WWW 8 kbps	5.39
	WWW 32 kbps	14.97
	WWW 64 kbps	24.62
	WWW 144 kbps	45.63
	WWW 384 kbps	103.35
	WWW 2 Mbps	517.79
Email	Email 64 kbps	59.78
	Email 144 kbps	137.00
	Email 384 kbps	365.44
FTP	FTP 64 kbps	58.81
	FTP 144 kbps	136.65
	FTP 384 kbps	356.50

Table 4.4. Mean session bitrates.

The routed traffic from the CMM is processed by each ATTPM. As already mentioned, three different ATTPMs were implemented in the simulator, corresponding to the following systems: GSM/GPRS, UMTS and HIPERLAN/2. The available ATTPMs per simulation are also user-defined (simulation input), in order to simulate different scenarios with different available access technologies.

As a basis for the implementation of the ATTPMs, simple capacity models were defined for each individual access technology. The considered parameters for the definition of these models are listed in Table 4.6. In summary, the simulator capacity measurement for each access technology depends on:

- GSM/GPRS: number of TSs;
- UMTS: SF (related to the available number of codes) and DL LF[4];
- HIPERLAN/2: number of LCHs.

The traffic processing algorithms implemented in the ATTPMs for CS and PS applications are represented in Figure 4.11 and Figure 4.12, respectively. These processes work as follows:

- For CS applications, each ATTPM monitors the routed traffic from the CMM in order to detect new calls. If a new call is detected, the ATTPM verifies if there is enough capacity in the respective network (TSs for GSM/GPRS, SF and DL LF for UMTS) for its transmission. If there is enough capacity, the ATTPM reserves capacity in the network for the whole duration of the call (corresponding to N MAC Frames). If there is not enough capacity for the call transmission, it is blocked.
- For PS applications, a similar process to the CS case is applied, but on a packet basis rather than a session basis. After detecting an incoming session, the ATTPM checks for available capacity for each packet transmission. Capacity is reserved during each packet complete transmission or if there is no available capacity, the packet is discarded.

For the Video Call application case, since it presents both CS (when the voice component is transmitted over GSM or UMTS) and PS components, an additional mechanism is included in its traffic processing algorithm: if the voice component is blocked due to lack of capacity, the overall system blocks the complete Video Call session (therefore, no video transmission is considered).

[4] The UMTS power based load estimation is not considered due to two main reasons: 1) the developed simulation tool does not take into account propagation issues; 2) the target scenarios to be simulated (Chapter 5) assume a small physical area (and consequently small cells), and therefore, in this case, the limiting load measurement is the DL LF.

Application	Traffic Source Model	Mean Capacity Occupation per Session			
		GSM/GPRS [TSs]	UMTS SF	UMTS DL LF	HIPERLAN/2 [LCHs]
Voice Call Video Call (Voice Component)	Classical ON-OFF	1.00	-	-	-
	CDMA MSVM	-	128	0.007	-
	VoIP	-	-	-	0.05
Video Call (Video Component)	H.263 TBR 32 kbps	2.54	64	0.038	0.36*
	GBAR H.263 VBR	-	-	-	1.90*
Video on Demand (Video Component)	H.263 TBR 64 kbps	-	32	0.074	0.36
	GOP GBAR MPEG-4 LOW Quality	-	-	-	0.56
	GOP GBAR MPEG-4 HIGH Quality	-	-	-	2.99
Web Browsing	WWW 8 kbps	0.41	256	0.006	0.03
	WWW 32 kbps	1.12	128	0.017	0.08
	WWW 64 kbps	1.84	64	0.027	0.13
	WWW 144 kbps	-	32	0.046	0.24
	WWW 384 kbps	-	16	0.074	0.54
	WWW 2 Mbps	-	-	-	2.70
Email	Email 64 kbps	4.47	32	0.065	0.32
	Email 144 kbps	-	16	0.139	0.72
	Email 384 kbps	-	4	0.263	1.91
FTP	FTP 64 kbps	4.39	32	0.064	0.31
	FTP 144 kbps	-	16	0.139	0.72
	FTP 384 kbps	-	4	0.256	1.86

* Double capacity occupation per session due to service symmetry (TDD system).

Table 4.5. Mean Capacity Occupation per Session.

Access Technology	Parameter	Comment
GSM/GPRS	# BTSs	Configurable.
	# TRX Frequencies per BTS	Configurable.
	# TSs for User Traffic per BTS	Configurable. Remaining TSs used for signalling and control.
	GSM/GPRS TS Allocation Scheme	Current simulator implementation considers free TS allocation for voice and data applications.
	GPRS Average Coding Scheme	Configurable (CS-1 to CS-4). Provides the average throughput in kbps per TS.
	Max TSs per GPRS MT	Configurable. Defines the maximum supported bitrate per GPRS MT.
UMTS	# Node Bs	Configurable.
	# Frequency Channels per Node B	Configurable (n×5 MHz).
	Duplex Mode	Current simulator implementation considers only FDD mode.
	Target *Eb/No* Values per Application	Configurable.
	Power Control	Current simulator implementation considers Ideal Power Control.
	% codes reserved for Soft Handover	Configurable.
	Activity Factor	Configurable for voice and for data applications.
	Ratio Other Cell to Own Cell Interference	Configurable.
	Orthogonality Factor	Configurable.
	Downlink Load Factor	Configurable (max value).
	Channel Coding Rates	Configurable per Application.
	Code Allocation Scheme	Current simulator implementation considers as ideal/optimised allocation.
HL/2	# APs	Configurable.
	Average PHY Mode	Configurable (PHY Mode 1 to 7).
	# LCHs / 2 ms MAC frame / AP	Configurable (depends on the selected average PHY Mode). Throughput per LCH = 192 kbps.

Table 4.6. Access Technologies Capacity Models' Parameters.

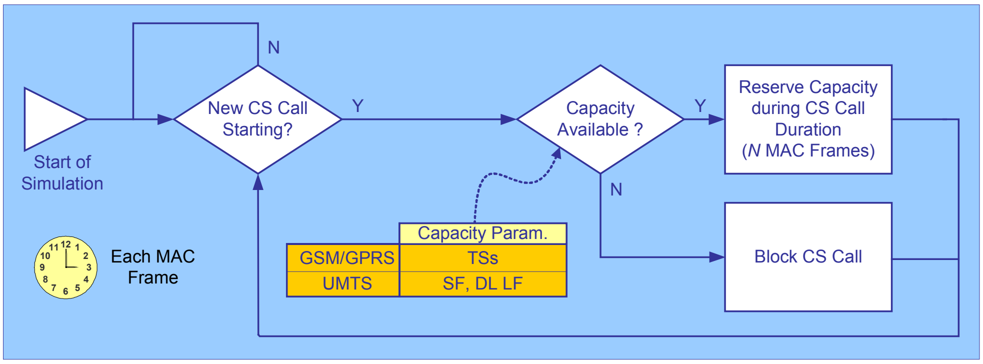

Figure 4.11. Traffic Processing Algorithm – Circuit Switched Applications.

The implemented traffic processing algorithms are realistic for both CS applications and PS video applications. This is not the case for PS data applications (*e.g.*, Web Browsing), since the buffering mechanisms (traffic shaping) and scheduling policies existing in real data traffic processing systems were not considered for simplicity reasons. The major drawback of the current implementation is that the packet discard rate (number of discarded packets / total number of packets), considered here for performance evaluation of the data traffic transmission, is in fact a measure of the amount of packets that would be buffered and delayed in a real system, and not a real measure of discarded packets (*e.g.*, due to transmission timeouts). In order to avoid over capacity requests during packet transmissions, maximum allowed peak bitrates for packet transmission was considered in the ATTPM implementations, Table 4.7, introducing a pseudo-shaping effect on the data traffic processing algorithms.

In a more realistic implementation, the performance measures of packet data transmission would be in terms of average (or total) packet delay and packet retransmission rate. These enhancements are proposed for future work in Chapter 6.

Each ATTPM process run corresponds to the MAC Frame duration of the associated access technology. Since the MAC frame durations of the different considered access technologies do not have a common divider, an approximation was used for the GSM/GPRS system. For CS calls, this approximation has a very reduced impact, since it only affects the total call duration, which will be a multiple of 5 ms (instead of 4.615 ms). For PS sessions, this approximation is compensated by a proportional increase in the capacity (in bytes) per GSM/GPRS TS, minimising its impact in the overall performance.

Technology	Max. Peak Bitrate for Packet Transmission	Comment
GSM/GPRS	Function of Coding Scheme and Max. number of TSs per GPRS MT, *e.g.*, 67 kbps (user traffic bitrate)	Corresponds to 5 TS with CS-2
UMTS	1 920 kbps (channel bitrate) ~640 kbps (user traffic bitrate with turbo coding 1/3)	Corresponds to SF=4
HIPERLAN/2	3 840 kbps (user traffic bitrate)	Corresponds to 20 LCHs

Table 4.7. Maximum Peak Bitrate for Packet Transmission.

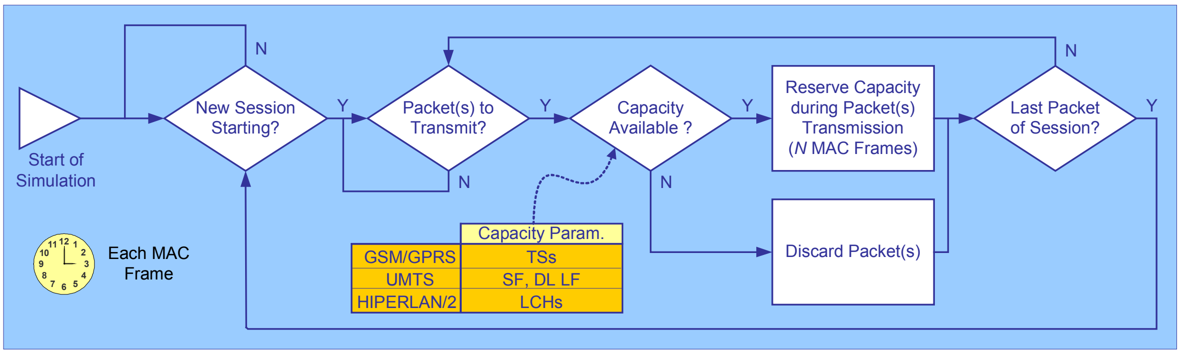

Figure 4.12. Traffic Processing Algorithm – Packet Switched Applications.

System	Real MAC Frame Duration [ms]	Simulator MAC Frame Duration [ms]
GSM	4.615	5
GSM/GPRS	18.46*	20
UMTS	10	10
HIPERLAN/2	2	2

* corresponds to a radio block (4 consecutive TSs).

Table 4.8. MAC Frame Durations in the Simulator.

The fact that the MAC frames do not have a common divider can cause implementation and synchronisation problems in a real system design, especially if packet based routing approached are used by the CMM.

As mentioned previously, the TPE takes into account the different application's priorities/QoS requirements. These requirements are implemented in the simulator in two ways, using the same priorities listed in Table 4.9:

- the CMM decisions are made in the same order as the applications priorities;

- the traffic processing of the ATTPMs also takes into account the priorities of each application (*e.g.*, if two sessions of different priorities require capacity at the same MAC frame, capacity is first reserved for the highest priority application).

Application	**Processing Priority**
Voice Call	1
Video Call (Voice Component)	2
Video Call (Video Component)	3
Video on Demand (Video Component)	4
Web Browsing	5
Email	6
FTP	7

Table 4.9. Application Processing Priorities.

An enhancement of the simulator, could be to consider different priority schemes (dynamic or static) per user type or even per MT, enabling the analysis of more sophisticated scenarios.

The TPE's outputs, corresponding to the simulator outputs, include the overall performance indicators that allow the evaluation of convergence benefits. These outputs will be described in the following section.

4.4 Output Parameters

The several inputs of the simulator have been presented during its description in the previous sections. In this section, the output parameters supplied by the simulator and some output examples will be presented. The simulator allows the selection of the sampling rate of each output parameter (*e.g.*, every 100 ms), for further analysis. A list of the available output parameters is presented in Table 4.10, organised into seven different output types.

<table>
<tr><th rowspan="2">Output Type</th><th rowspan="2" colspan="2">Output Parameter</th><th colspan="2">Available per</th></tr>
<tr><th>Application</th><th>Access Technology</th></tr>
<tr><td rowspan="2">Session Statistics</td><td colspan="2">Number of Active Sessions (N_{AS})</td><td>✓(CS+PS)</td><td>✓</td></tr>
<tr><td colspan="2">Ratio of Active Sessions per Active MTs (R_{SM})</td><td colspan="2">Global</td></tr>
<tr><td rowspan="2">CS Calls Statistics</td><td colspan="2">Number of Blocked Calls (N_{BC})</td><td>✓ (CS)</td><td>✓</td></tr>
<tr><td colspan="2">Call Blocking Rate (R_{CB})</td><td>✓ (CS)</td><td>✓</td></tr>
<tr><td rowspan="2">CS Traffic Characterisation</td><td colspan="2">Average Call Duration ($\overline{\tau}_C$)</td><td>✓ (CS)</td><td>-</td></tr>
<tr><td colspan="2">Generated Traffic (A) [Erl]</td><td>✓(CS)</td><td>-</td></tr>
<tr><td rowspan="3">PS Traffic Statistics</td><td colspan="2">Number of Discarded Packets/Frames (N_{DP})</td><td>✓(PS)</td><td>✓</td></tr>
<tr><td colspan="2">Percentage of Discarded Packets/Frames (over all Packets/Frames) (η_D)</td><td>✓(PS)</td><td>✓</td></tr>
<tr><td colspan="2">Mean Bitrate ($\overline{R}_b$)</td><td>✓(PS)</td><td>-</td></tr>
<tr><td rowspan="4">CMM Statistics</td><td colspan="2">CMM Technology Decision Percentage (η_{TD})</td><td>✓(CS+PS)</td><td>-</td></tr>
<tr><td colspan="2">CMM Bitrate/Quality Decision Percentage (η_{BD})</td><td>✓(CS+PS)</td><td>-</td></tr>
<tr><td rowspan="2">CMM 1st Priority Decisions Percentage (η_{1PD})</td><td>on Technology ($\eta_{1PD\text{-}T}$)</td><td rowspan="2" colspan="2">Global</td></tr>
<tr><td>on Bitrate/Quality ($\eta_{1PD\text{-}B}$)</td></tr>
<tr><td>Network's Load</td><td colspan="2">Normalised Capacity Usage (η_U)</td><td>-</td><td>✓</td></tr>
<tr><td rowspan="3">Overall Performance Degradation</td><td colspan="2">Call Blocking Rate Degradation ($D_{R_{CB}}$)</td><td colspan="2">Global</td></tr>
<tr><td colspan="2">Overall Discarded Packets/Frames Degradation (D_{η_D})</td><td colspan="2">Global</td></tr>
<tr><td colspan="2">Overall Mean Bitrate Degradation ($D_{\overline{R}_b}$)</td><td colspan="2">Global</td></tr>
</table>

Table 4.10. Simulator Output Parameters.

The parameters R_{SM}, R_{CB}, η_D, η_{TD}, η_{BD}, $\eta_{1PD\text{-}T}$, $\eta_{1PD\text{-}B}$, and η_U, are given by the following expressions:

$$R_{SM} = \frac{N_{AS}}{N_{AM}}, \tag{4.1}$$

where N_{AM} is the number of active MTs;

$$R_{CB} = \frac{N_{BC}}{N_C}, \tag{4.2}$$

where N_C is the total number of CS calls;

$$\eta_D = \frac{N_{DP}}{N_P}, \tag{4.3}$$

where N_P is the total number of packets/frames;

$$\eta_{TD} = \frac{N_{ST_i}}{N_S}, \tag{4.4}$$

where N_{ST_i} is the total number of sessions routed through access technology i (GSM/GPRS, UMTS or HIPERLAN/2) and N_S is the total number of sessions;

$$\eta_{BD} = \frac{N_{SB_j}}{N_S}, \tag{4.5}$$

where N_{SB_j} is the total number of sessions using bitrate/quality j;

$$\eta_{1PD-T} = \frac{N_{1ST}}{N_S}, \tag{4.6}$$

where N_{1ST} is the total number of sessions routed through the 1st priority access technology (according to the PT);

$$\eta_{1PD-B} = \frac{N_{1SB}}{N_S}, \tag{4.7}$$

where N_{1SB} is the total number of sessions using the 1st priority bitrate/quality (according to the PT);

$$\eta_U = \begin{cases} \dfrac{N_{TS_U}}{N_{TS}}, & \text{for GSM/GPRS} \\ \left. \begin{array}{c} \dfrac{N_{Co_U}}{N_{Co}} = \dfrac{N_{Co_U}}{512 - N_{CoSH}} \\ \dfrac{\eta_{DL}}{\eta_{DLmax}} \end{array} \right\}, & \text{for UMTS} \\ \dfrac{N_{LCH_U}}{N_{LCH}}, & \text{for HIPERLAN/2} \end{cases}, \tag{4.8}$$

where N_{TS_U} is the number of used TSs, N_{TS} is the total number of available TSs, N_{Co_U} is the equivalent number of used codes assuming SF=512, N_C is the equivalent total number of available codes assuming SF=512 excluding the amount of codes reserved for soft handover (N_{CoSH}), η_{DLmax} is the maximum allowed value for the DL LF (η_{DL}), N_{LCH_U} is the number of used LCHs, and N_{LCH} is the total number of available LCHs.

In order to have a general and simple overview of the networks' overall performance, three overall performance degradation indicators are available (as presented in Table 4.10), considering global results from all applications and technologies. These parameters are given by the following expressions:

$$D_{R_{CB}} = \left. \frac{N_{BC}}{N_C} \right|_{(all\ CS\ applications)} ; \qquad (4.9)$$

$$D_{\eta_D} = \left. \frac{N_{DP}}{N_P} \right|_{(all\ PS\ applications)} ; \qquad (4.10)$$

$$D_{\overline{R}_b} = \frac{\displaystyle\sum_{i=1}^{5} \left. N_P \cdot \frac{\overline{R}_b}{\overline{R}_{b_{max}}} \right|_{(PS\ application\ i)}}{\left. N_P \right|_{(all\ PS\ applications)}} , \qquad (4.11)$$

where $\overline{R}_{b_{max}}$ is the maximum target mean bitrate for a specific application (taking into account the PT).

The BR degradation (4.9) is calculated as the total blocked CS calls over the total number of CS calls (for both Voice Call and Video Call applications) and the Discarded Packets/Frames degradation (4.10) is obtained by dividing all the discarded packets/frames by the total amount of packets/frames to transmit considering all PS applications. The Mean Bitrate Degradation (4.11) is obtained by normalising the mean bitrate of each PS application to its maximum target mean bitrate (taking into account the PT) and weighing each application with the total number of transmitted packets/frames.

The outputs from the simulator are generated in the form of text files (.txt) and further processed with Matlab for obtaining graphical plots. In the remainder of this section, some examples of the output parameters obtained from simulations are shown. Merely for information purposes, all output graphics were obtained from a 1000 users scenario with all access technologies available (except for the examples presented in Figure 4.24, which was obtained with only GSM/GPRS and HIPERLAN/2 technologies active, and Figure 4.26 and Figure 4.27, which were obtained with only GSM/GPRS and UMTS active) with a sampling rate of 10 s during one complete BH. The users profiles, network configuration and CMM parameters correspond to the ones used in Chapter 5.

The first set of 4 graphics corresponds to examples of the session statistics. Figure 4.13 shows the evolution of the total active sessions/calls and the ratio between the total active sessions and the total active MTs. The latter provides an indication of the evolution of the average number of active applications per active MT. Figure 4.14 and Figure 4.15 provide the evolution of the number of active sessions per application, while Figure 4.16 provides the evolution of the number of active sessions processed per access technology.

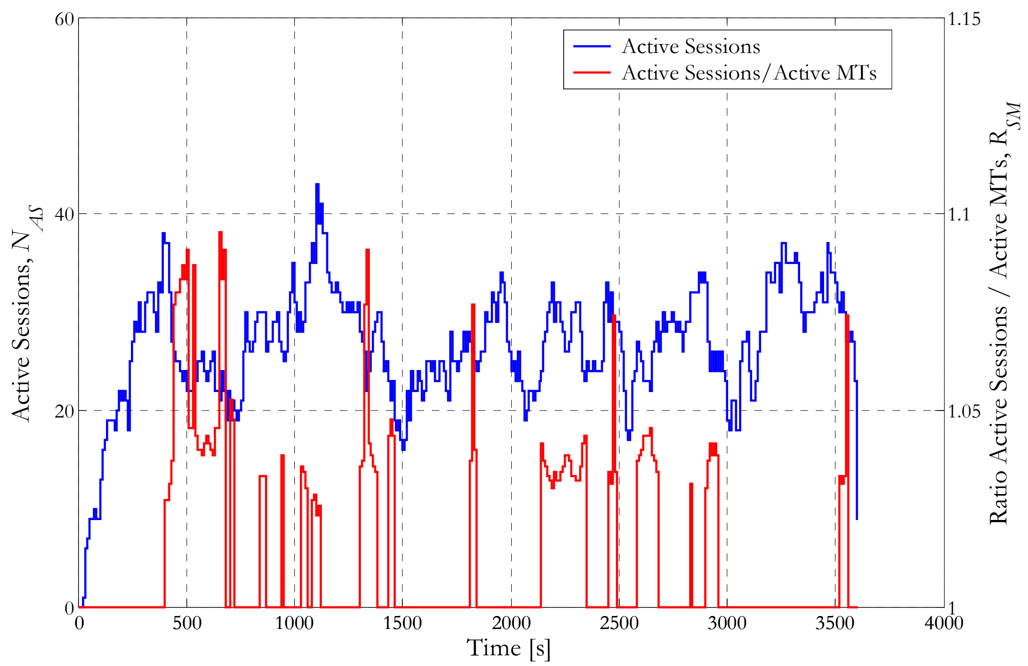

Figure 4.13. Active Sessions and Ratio Active Sessions/Active MTs.

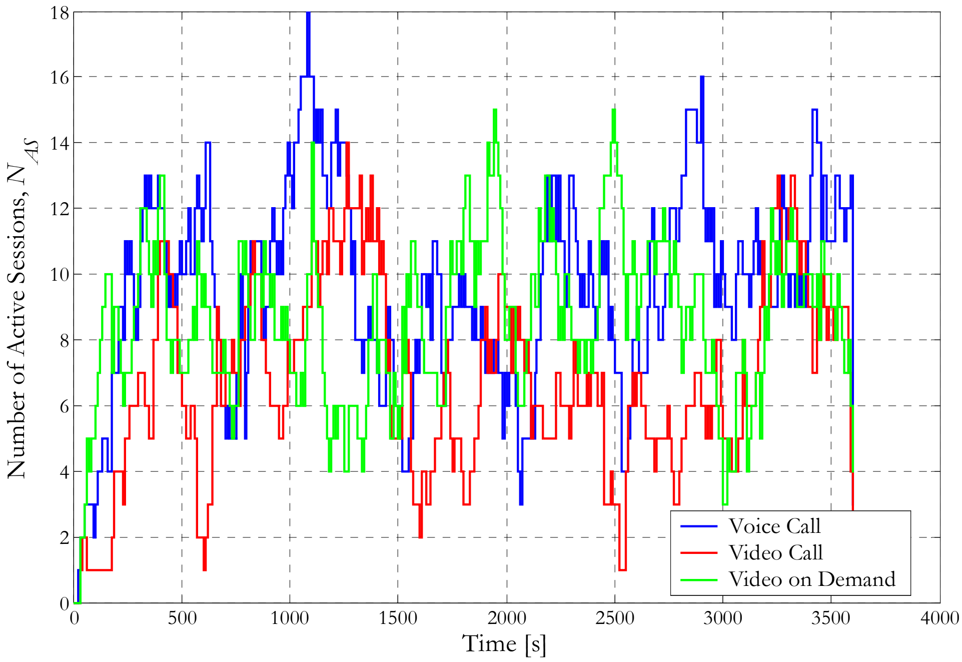

Figure 4.14. Active Sessions per Application (Voice Call, Video Call and Video on Demand).

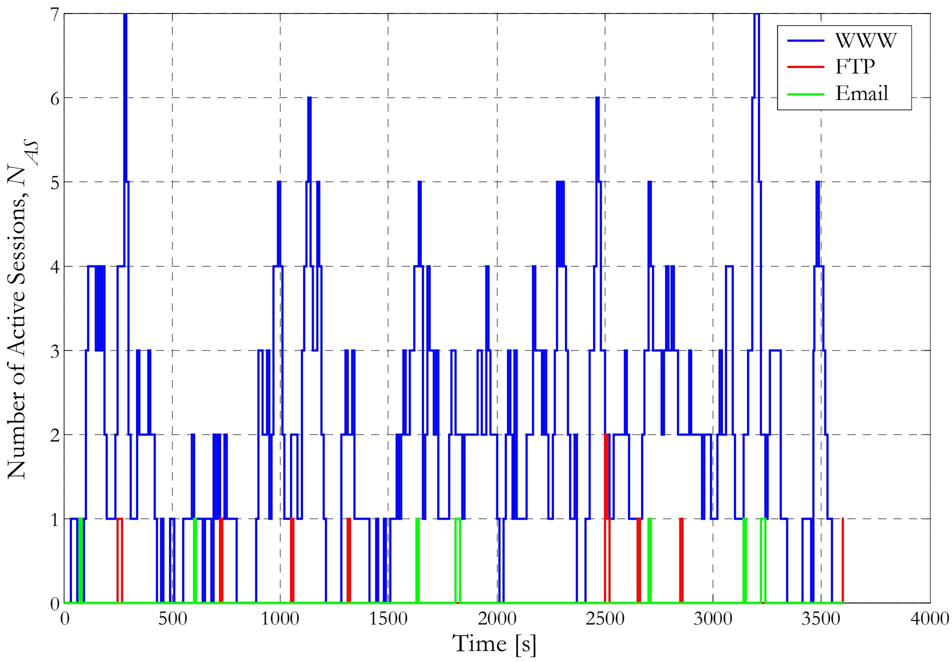

Figure 4.15. Active Sessions per Application (WWW, FTP and Email).

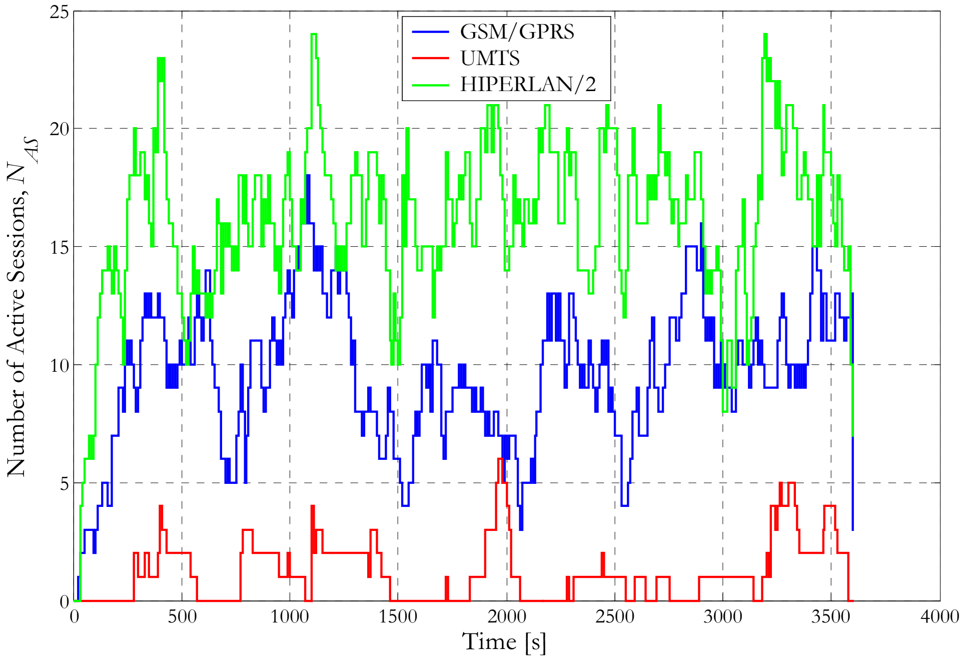

Figure 4.16. Active Sessions per Access Technology.

The following set of three figures corresponds to examples of the CS calls statistics. Figure 4.17 presents the evolution of the cumulative number of CS calls and of CS blocked calls per CS Application (voice call and video call (voice component)). Figure 4.18 and Figure 4.19 depict the evolution of the call BR per CS application and per access technology, respectively. Analysing Figure 4.19 one observes that all blocked calls were performed by GSM in the associated simulation. The simulator also provides the blocking rate due to lack of available codes or excess of DL LF for the UMTS system.

Examples of the CS traffic characterisation outputs are presented in Figure 4.20 and Figure 4.21. Figure 4.20 provides the average call duration for the CS applications. The Video on Demand average session duration is also included in this figure. One can observe that the average call/session duration converges approximately to the parameterised values (in this case, 120 s for all three applications), although the number of calls/sessions is relatively small (few hundreds). Figure 4.21 provides the overall generated traffic (in Erlang) per CS application. As expected, these values also converge to specific values.

In what regards to PS traffic statistics, Figure 4.22 presents an example of the evolution of the total number of transmitted packets/frames (for video applications) and of the total number of discarded packets/frames per PS application. The voice sessions that are transmitted over HIPERLAN/2 (VoIP) are also included in these statistics, providing the number of discarded packets an indicator of the quality of the voice transmission.

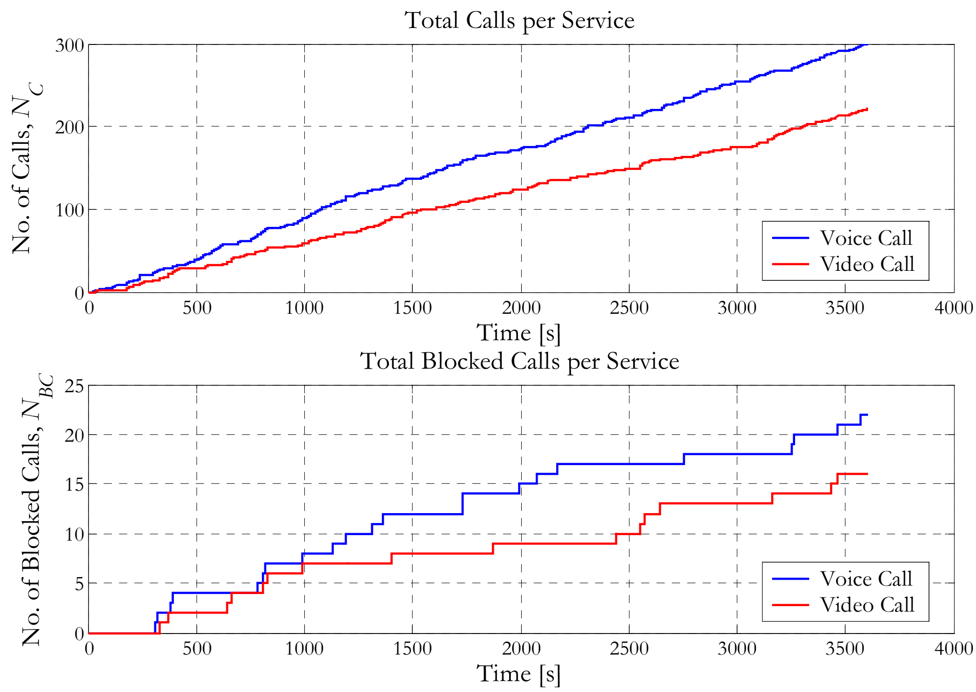

Figure 4.17. Number of CS Calls and CS Blocked Calls.

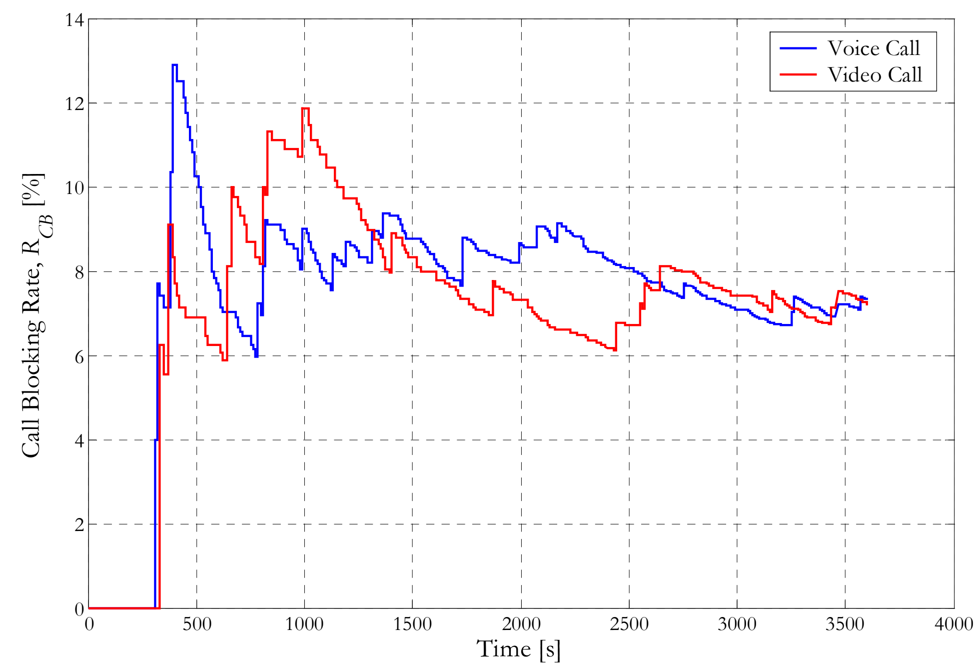

Figure 4.18. Blocking Rate per CS Application.

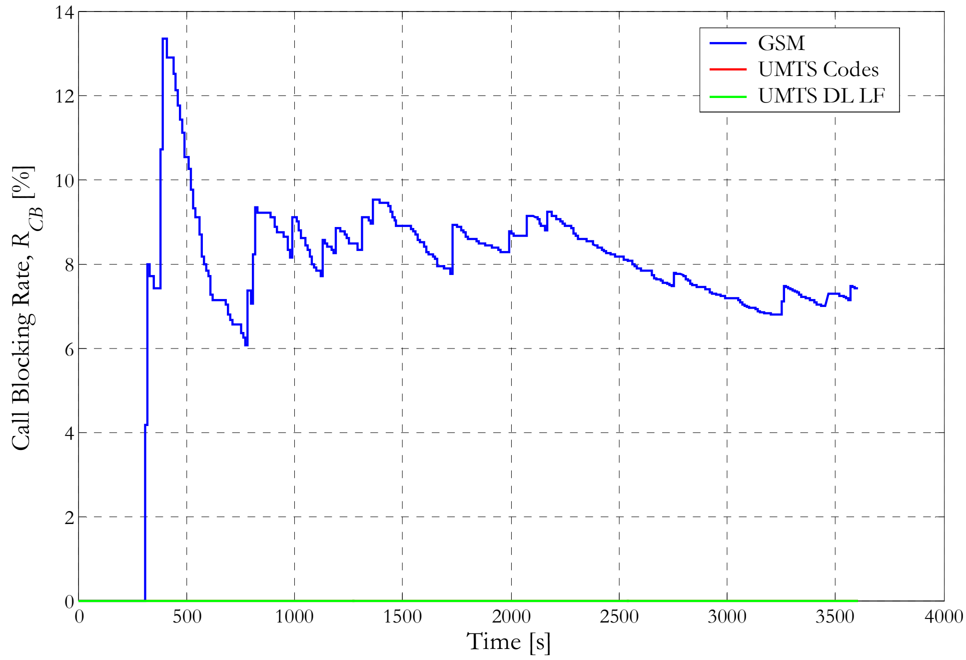

Figure 4.19. CS Calls Blocking Rate per Access Technology.

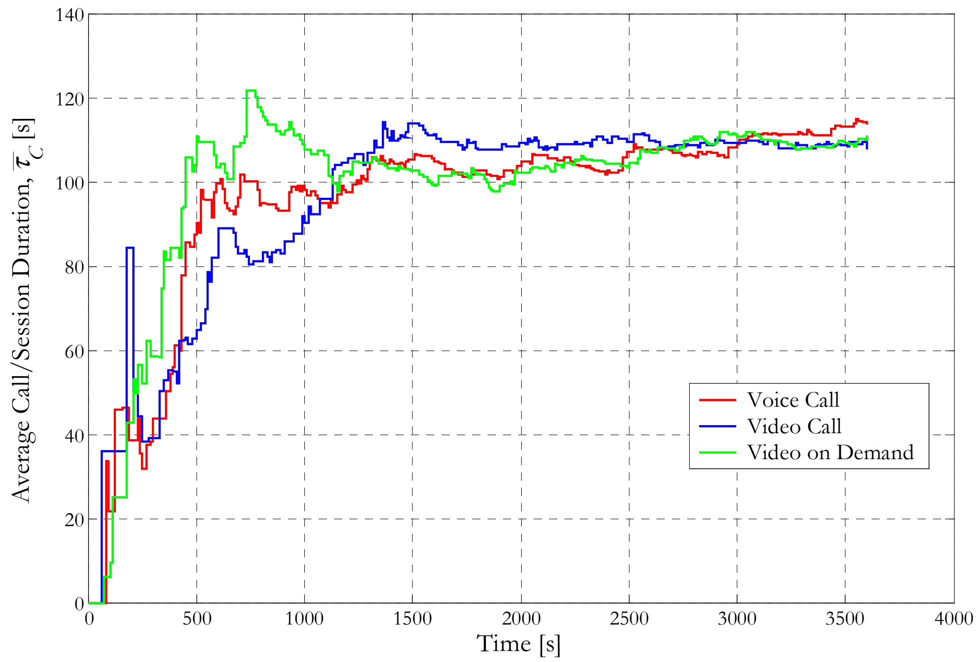

Figure 4.20. Average Call/Session Duration per Application.

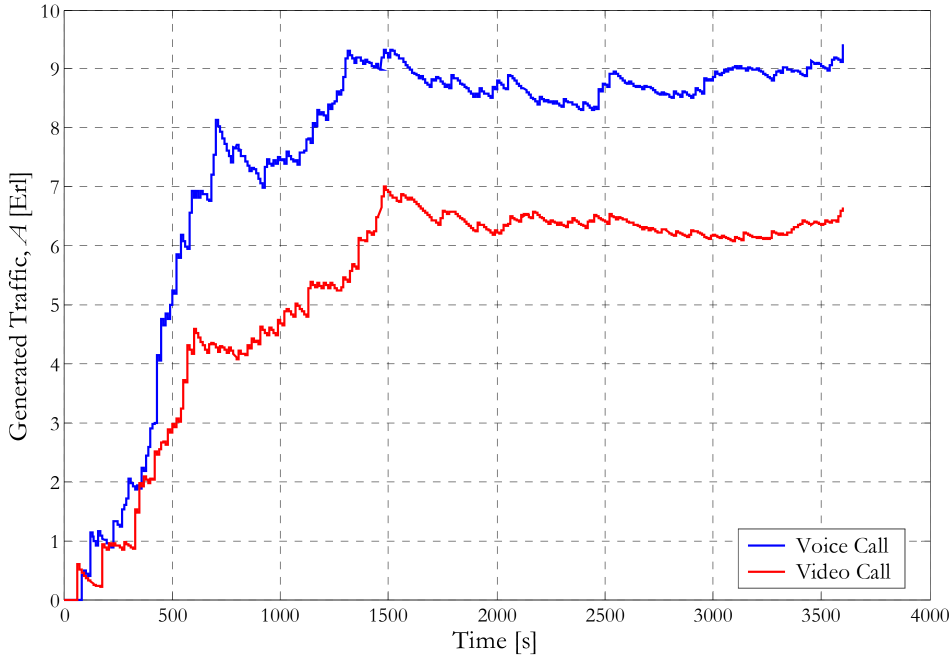

Figure 4.21. Generated Traffic per CS Application.

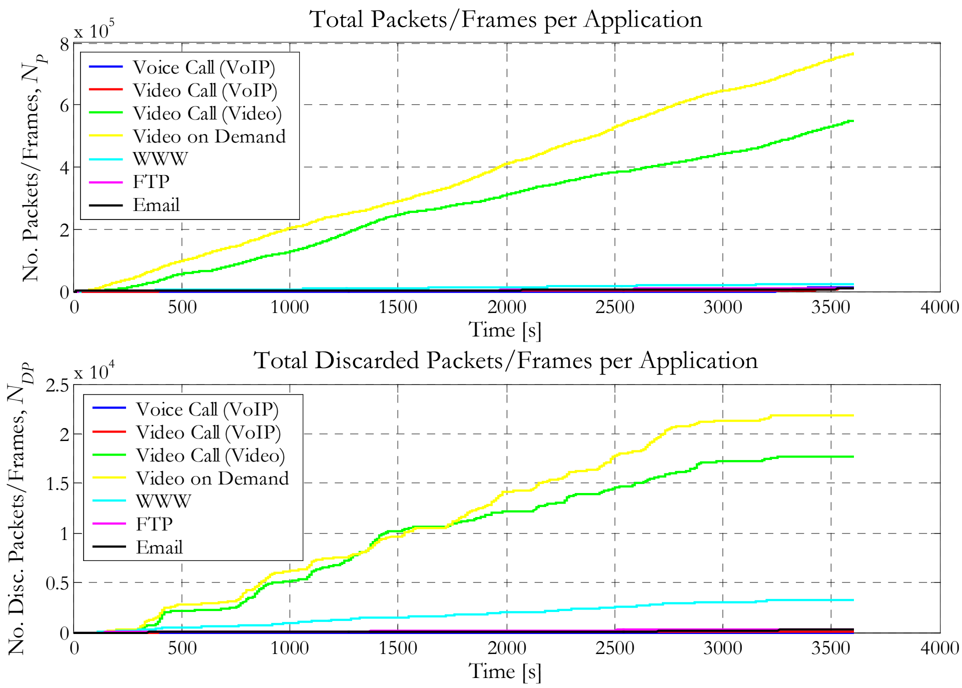

Figure 4.22. Number of Transmitted Packets/Frames and Discarded Packets/Frames per PS Application.

Figure 4.23 and Figure 4.24 depict the evolution of the discarded packets/frames (in %) per PS application and per access technology for the Web Browsing application, respectively. In both graphics, as expected, a convergence tendency is observed in all curves. As one can observe in Figure 4.24, the simulator provides the percentage of discarded packets/frames within each access technology: GSM, UMTS (due to lack of available codes or excess of DL LF) and HIPERLAN/2.

Another important output of the simulator relatively to the PS traffic statistics is the mean bitrate per PS Application. Figure 4.25 presents an example of the evolution of these values. Again, as expected, the values converge to specific values. The variations in the mean bitrate per application are due to the CMM decisions on the bitrate/quality of each new PS session.

The CMM statistics provide information on the decision percentages relatively to the different technology routing choices per application and the respective transmission quality/bitrate. Examples of the evolution of these decisions percentages can be observed in Figure 4.26 and Figure 4.27. Figure 4.26 shows the evolution of the CMM technology decision for the WWW and FTP applications, while Figure 4.27 shows the evolution of the CMM bitrate/quality decision for the Email application. The decisions percentages in both cases also present a converged behaviour. As mentioned previously, both these figures were obtained for a scenario with only GSM/GPRS and UMTS.

The simulator also provides statistics on the ATTPMs, namely the capacity usage of each access technology network. As mentioned in Section 4.3, the capacity measurement parameters used in the simulator are the number of TSs for GSM/GPRS, the available number of codes and DL LF for UMTS and the number of LCHs for HIPERLAN/2. An example of the evolution of these four parameters normalised to their maximum allowed values is presented in Figure 4.28.

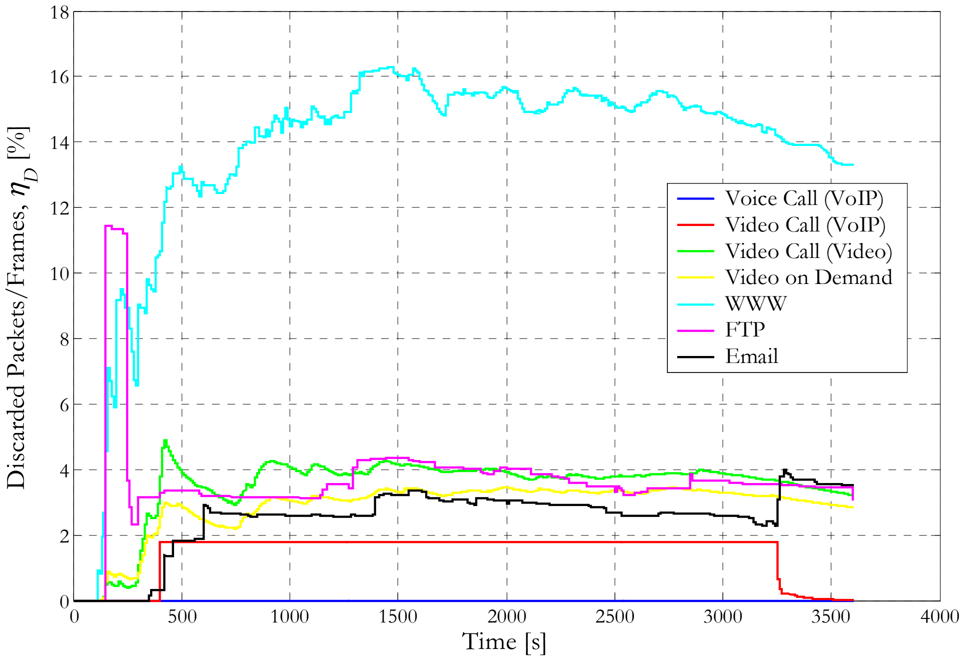

Figure 4.23. Percentage of Discarded Packets/Frames per PS Application.

Figure 4.24. WWW Application Discarded Packets per Access Technology.

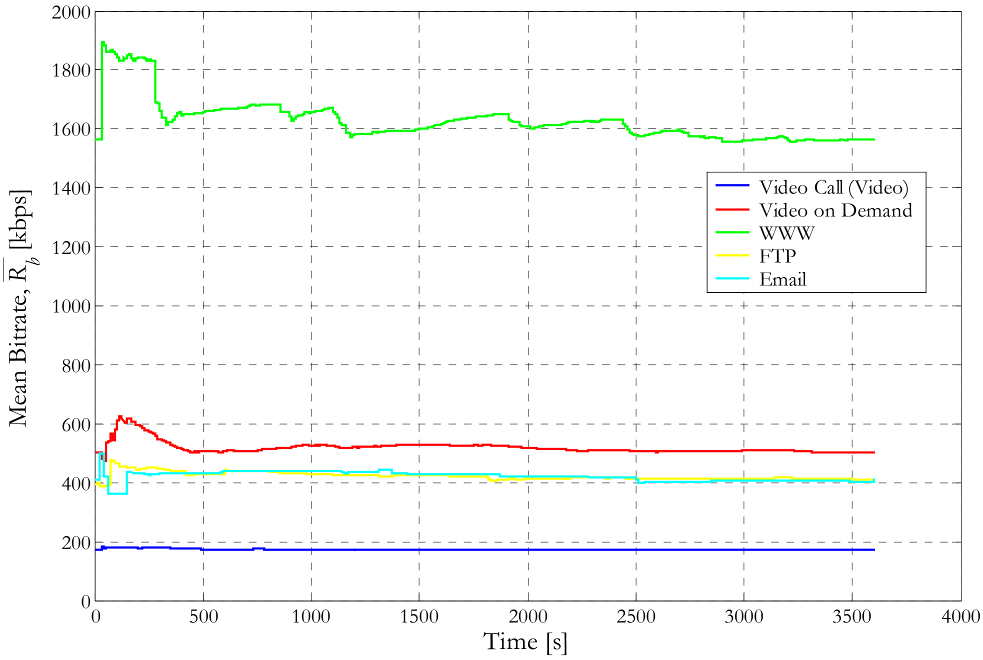

Figure 4.25. Mean Bitrate per PS Application.

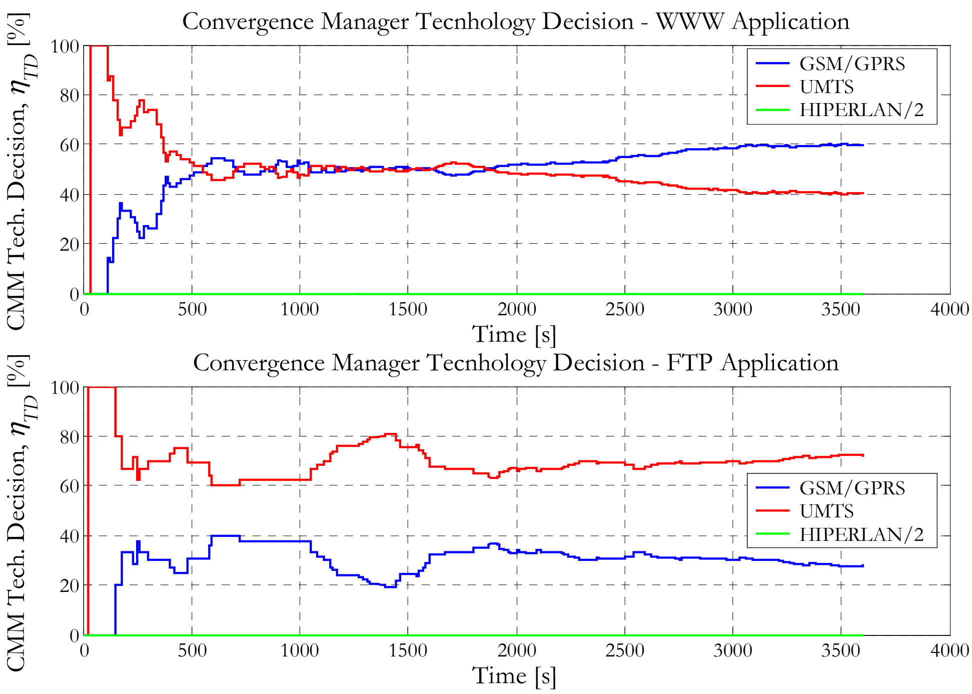

Figure 4.26. CMM Technology Decision Percentage – WWW and FTP applications.

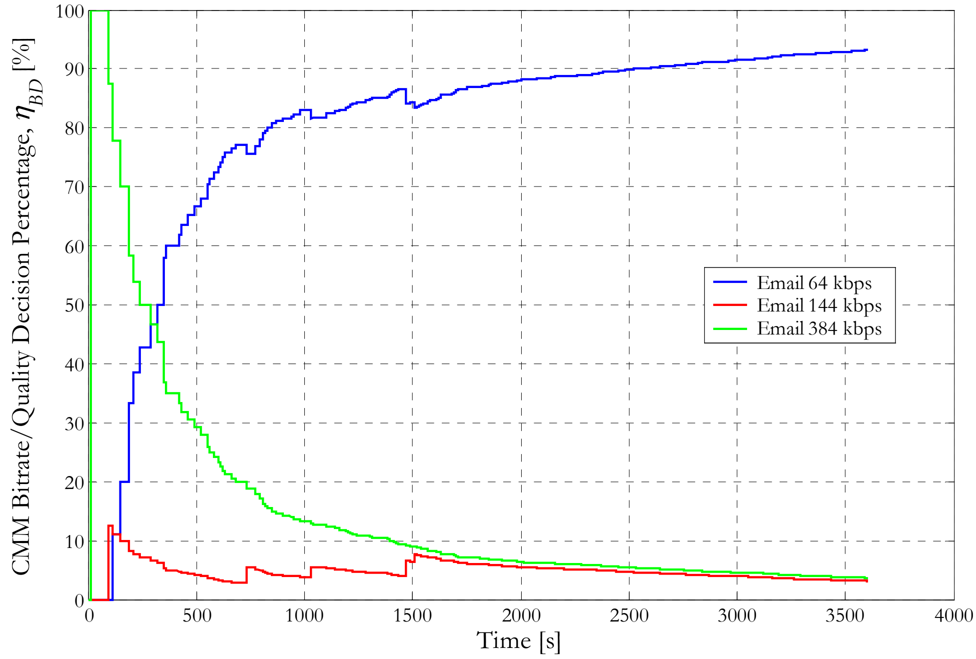

Figure 4.27. CMM Bitrate/Quality Decision Percentage - Email Application.

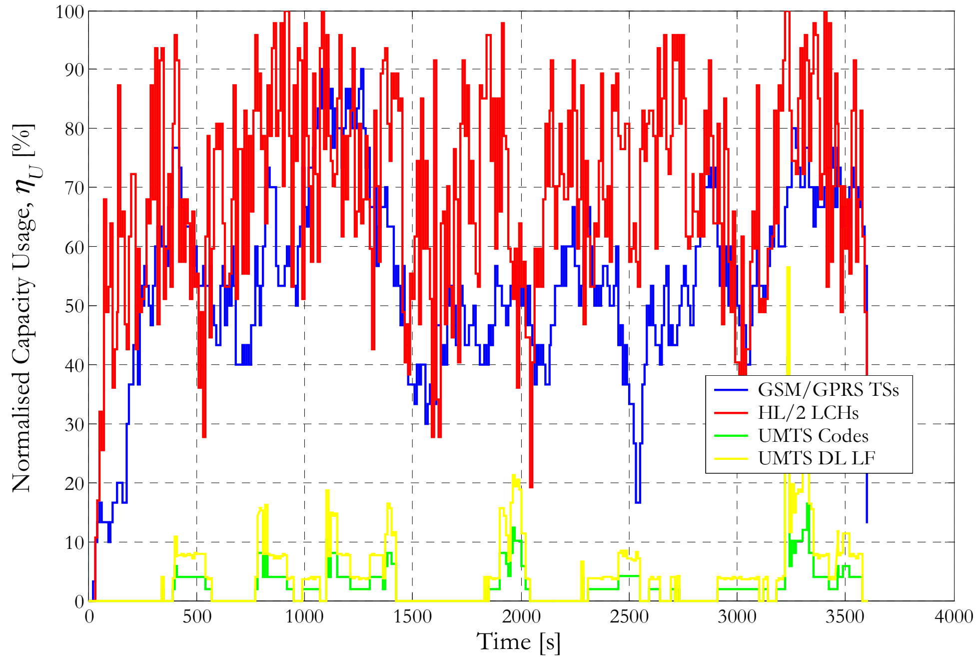

Figure 4.28. Normalised Capacity Usage per Access Technology.

Chapter 5

Analysis of Results

In this chapter, one analyses the results of simulations for several different scenarios and following three different approaches. The simulated scenarios are characterised according to three components: technologies, users and applications. The major performance indicators that are considered for the overall performance analysis of the simulated scenarios are: CS calls BR, percentage of PS discarded packets/frames and mean bitrate per PS application.

5.1 Scenarios Description

In order to characterise in a simple way the scenarios to be simulated in the present chapter, referred from here onwards as Convergence Scenarios, three main components have been identified, Figure 5.1:

- Access Technology Scenario, defined by the available access technologies (to all users/MTs) and their specific characterisation;
- User Scenario, characterised by the number of users/MTs and user type penetrations (market segmentation);
- Applications Scenario, identifying the active applications (for all users/MTs), their characterisation, and their specific traffic characteristics (per user type).

The overall convergence technique/strategy associated to each Convergence Scenario interacts with these three components, depending on their individual characteristics/requirements.

As mentioned in the previous chapter, the overall convergence technique/strategy is defined by means of a PT in the CMM of the simulator. The construction of a specific PT is influenced by the three components above identified, *i.e.*, it contemplates access technology aspects (*e.g.*, available technologies with convergence capabilities and their application specific performances), users preferences (*e.g.*, user defined applications priority/quality requirements, applications costs issues), and applications' requirements (*e.g.*, maximum delay, bitrate). As stated in Chapter 4, a fixed PT was implemented in the current simulation platform, given by Table 4.3, based on a highest throughput / best system approach. The characterisation of this PT has also been discussed in Chapter 4.

Each of the components of a Convergence Scenario will be characterised in the following paragraphs.

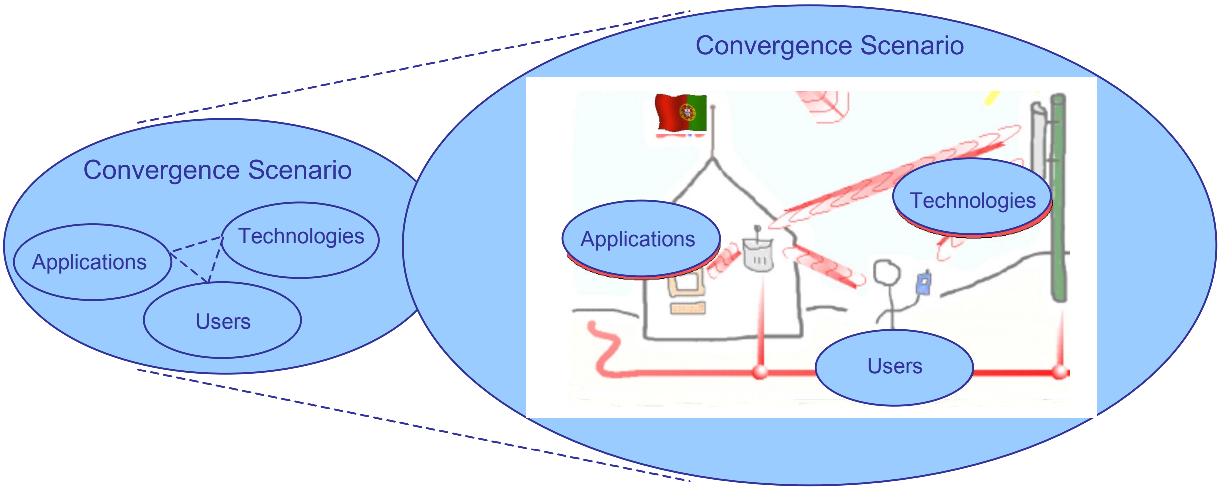

Figure 5.1. Main components for the characterisation of a Convergence Scenario.

As mentioned in Chapter 4, three different access technologies were considered in the development of the simulation platform: GSM/GPRS, UMTS and HIPERLAN/2. An Access Technology Scenario characterises/parameterises these three access technologies and defines which of them are available to the users/MTs. For all Access Technology Scenarios adopted throughout the present chapter, a common technology parameterisation for the different access technologies was used, Table 5.1. As a basis for the overall network configuration, a "Public Square" type scenario (*e.g.*, with 100×100 m^2) was considered, assuming collocated base stations (*e.g.*, in the centre of the square), one per access technology, Figure 5.2.

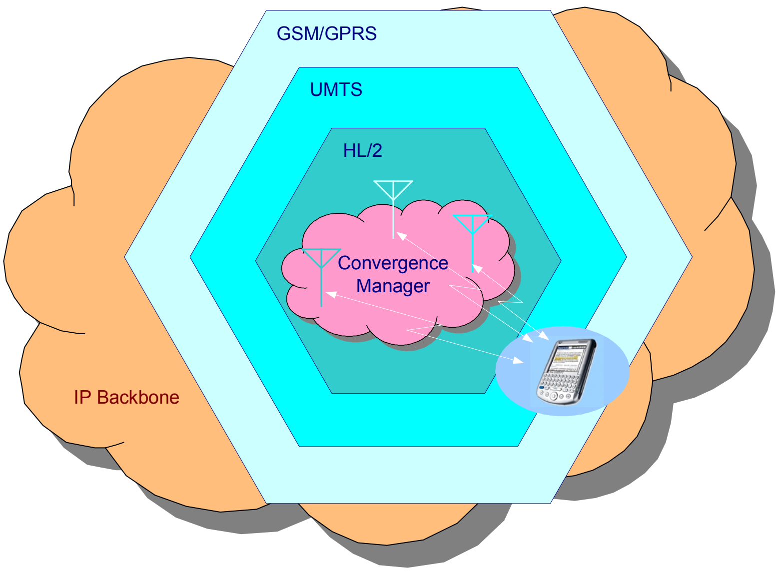

Figure 5.2. Network Configuration of Simulated Scenarios.

For the GSM/GPRS system, 30 TSs were considered available for user traffic (in a similar configuration as presented in Chapter 2 for a medium-capacity cell with four TRXs, with 2 TSs reserved for signalling and control). For data applications (via GPRS), a maximum user bitrate of 67 kbps is adopted, corresponding to an allocation of 5 TSs with CS-2 (on average). A free TS allocation scheme between CS and PS applications was considered, although assuming higher priority for the CS traffic, as stated in Chapter 4.

Access Technology	Parameter	Comment
GSM/GPRS	# BTSs	1
	# TRX Frequencies per BTS	4
	# TSs for User Traffic per BTS	30 (remaining 2 TSs used for signalling and control).
	GSM/GPRS TS Allocation Scheme	Free (all TSs usable for voice and data application).
	GPRS Average Coding Scheme	CS-2 (13.4 kbps throughput per TS).
	Max TSs per GPRS MT	5 (maximum supported bitrate per GPRS MT: 5×13.4 kbps=67 kbps).
UMTS	# Node Bs	1
	# Frequency Channels per Node B	1×5 MHz
	Duplex Mode	FDD mode.
	Target DL Eb/No Values per Application	See Table 5.2.
	Power Control	Ideal Power Control.
	% codes reserved for Soft Handover	30 %
	Activity Factor	0.55 for voice and 1.0 for data applications.
	Ratio Other Cell to Own Cell Interference	0.65
	Orthogonality Factor	0.5
	Downlink Load Factor	Max. allowed value: 0.7
	Channel Coding Rates	1/2 rate for voice applications and 1/3 rate for data applications.
	Code Allocation Scheme	Ideal/optimised allocation.
HL/2	# APs	1
	Average PHY Mode	PHY Mode 3
	# LCHs / 2ms MAC frame / AP	47 (192 kbps / LCH -> Tot ~9 Mbps/AP).

Table 5.1. Parameterisation of the Access Technology Scenarios.

In what regards the UMTS system, one 5 MHz channel was considered available at the Node B for FDD mode. The DL target *Eb/No* values per application used in the simulations are the ones presented in Table 5.2. These values were obtained from [FCXV03] for the particular case of a 3 km/h mobility scenario (low mobility). The maximum allowed value for the DL LF was set to 0.7, and a 1/2 rate channel coding rate was considered for voice applications and a 1/3 rate for video and data applications. The values for these parameters and for the remaining ones, listed in Table 5.2 (for UMTS), were selected from the common value ranges presented in literature (*e.g.*, [HoTo00]).

For HIPERLAN/2, PHY Mode 3 was considered as the average PHY Mode in use by the system. A total of 47 LCHs were considered available per MAC frame (each 2 ms) for user traffic, providing an overall throughput (DL + UL) of approximately 9 Mbps at the AP. These values are based on the throughput calculations and simulation results presented in [KrSa01].

Seven different Access Technology Scenarios were considered in the present chapter for simulation purposes. These scenarios correspond to all possible combinations of available access technologies, as listed in Table 5.3.

Application		**DL Target Eb/No [dB]**
Voice Call		7.5
Video Call (Video Component)		5.7
Video on Demand (Video Component)		5.6
Data Applications (Web Browsing, Email and FTP)	8 kbps	5.7
	32 kbps	5.7
	64 kbps	5.6
	144 kbps	5.3
	384 kbps	3.8

Table 5.2. DL Target Eb/No values adopted for the UMTS ATTPM.

Ten different User Scenarios are considered in the present chapter, corresponding to different number of users/MTs, ranging from 100 to 1 000 users in steps of 100 users, Table 5.4. These User Scenarios will enable to evaluate the impact of the variation of the number of users for different Access Technology and Applications Scenarios.

Access Technology Scenario	**Available Access Technologies**		
	GSM/GPRS	**UMTS**	**HIPERLAN/2**
G	√		
U		√	
H			√
G+U	√	√	
G+H	√		√
U+H		√	√
G+U+H	√	√	√

Table 5.3. Access Technology Scenarios.

User Scenario	1	2	3	4	5	6	7	8	9	10
Number of Users	100	200	300	400	500	600	700	800	900	1 000

Table 5.4. User Scenarios.

For all considered User Scenarios, fixed user type penetration values were adopted, as presented in Table 5.5. These different user types allow for a more realistic representation of different market segments with different applications requirements and usage profiles.

	User Type 1	**User Type 2**	**User Type 3**
Penetration [%]	25	40	35

Table 5.5. User Scenario: User Type Penetration Values.

A fixed Applications Scenario, with all six implemented applications activated, was considered throughout all the different simulations, Table 5.6. The precise values that were adopted for the BHCAs and average call/session durations, were arbitrated (with reasonable values), since scarce literature exists on user traffic profiles for NG networks and applications.

Different BHCA values (providing the average number of CS call attempts / PS sessions per application during the BH) were considered for each user type. The User Type 1 users are the most traffic demanding, representing a market segment such as business users, while User Type 3 users are the less traffic demanding, representing, *e.g.*, a mass market segment. The User Type 2 traffic demands represent an intermediate market segment, in which users, for example, use their MTs for both personal and professional communications, but with lower overall demand than users of User Type 1.

Application	Active	BHCA [Calls(/Sessions)/Hr]			Average Call/Session Duration [s]
		User Type 1	User Type 2	User Type 3	
Voice Call	√	0.5	0.3	0.1	120
Video Call	√	0.4	0.2	0.1	120
Video on Demand	√	0.3	0.3	0.3	120
Web Browsing	√	0.4	0.3	0.2	Given by Traffic Source Models
Email	√	0.3	0.2	0.1	
FTP	√	0.2	0.1	0.05	

Table 5.6. Applications Scenario Characterisation.

For the Voice Call, Video Call and Video on Demand applications, average session durations of 120 s were considered for all three user types. For the remaining applications, the session duration is provided by the traffic source models in function of the session volumes (randomly generated according to the respective TSMs).

In what regards the simulation methodology followed in this work, the analyses of results are to be based on simulation sets, composed by a set of simulation runs. A simulation run corresponds to a complete BH simulation (as previously defined in Section 4.1) of a given Convergence Scenario concretisation. Each concretisation implies, from the Applications Scenario point of view, a statistical concretisation of its associated UTVs; this statistical concretisation will be from here onwards referred to as "instance" of a given Applications Scenario.

In the following sections, analyses of the results of simulations for several different Convergence Scenarios will be presented, following three different approaches:

- Statistical analysis of user traffic variations (for several instances of the same Applications Scenario) in the overall performance indicators, for each Access Technology Scenario and for a fixed User Scenario (Section 5.2);
- For each Access Technology Scenario, analysis of the impact of the number of users (User Scenario variation) in the overall performance indicators (Section 5.3);
- For a specific User Scenario, analysis of the existence of different Access Technology Scenarios in the overall performance indicators, given a particular instance of the Applications Scenario (*i.e.*, several simulation runs maintaining the same UTVs while varying the Access Technology Scenario) (Section 5.4).

Three major performance indicators are considered for the purpose of analysing the results: CS calls BR, percentage of PS discarded packets/frames and mean bitrate per PS application. The combination of these three indicators allows for a general understanding of the overall performance of the simulated scenarios.

When interpreting the performance results of the PS applications, it is important to recall that the Discarded Packet/Frames indicator provides information on the percentage of packets that would be buffered and delayed in a real system, and not a real measure of discarded packets.

On average, each simulation run (corresponding to one Busy Hour) lasted about 1 hour in a PIV 2.66 GHz computer with 768 Mbyte of RAM.

5.2 Statistical Analysis

Since user traffic is statistically generated (call/session beginning instants, call/session durations, session volumes, packet sizes and inter-arrival times, etc.), one needs to analyse the impact of its variation in the overall performance indicators, maintaining all other parameters constant. In the present work, this analysis was the first to be done in order to understand if the results of a single simulation run (corresponding to a single instance of an Application Scenario) would be representative or not of a more in-depth analysis (*i.e.*, considering several simulation runs), avoiding excessively long simulation times for obtaining reasonable results.

The adopted procedure was to analyse the mean and the standard deviation values of each performance indicator for a specific User Scenario (700 users) and for all the different (7) Access Technology Scenarios. Ten different instances of the Application Scenario were considered in this analysis, generating ten simulation runs per Access Technology Scenario. Although this is a quite short set of values in terms of statistical meaning, it presents already a good compromise in respect to simulation durations within a reasonable timeframe for obtaining indicative results.

In Figure 5.3, the statistical results for the BR performance indicator are presented. The mean values for each scenario are given by the coloured bars (different coloured bars correspond to different applications) and the respective standard deviation boundaries are given by the black error bars centred at the mean values (upper and lower limits corresponding to one standard deviation). The same type of results is presented for the Discarded Packets/Frames and Mean Bitrate performance indicators in Figure 5.4 and Figure 5.5, respectively.

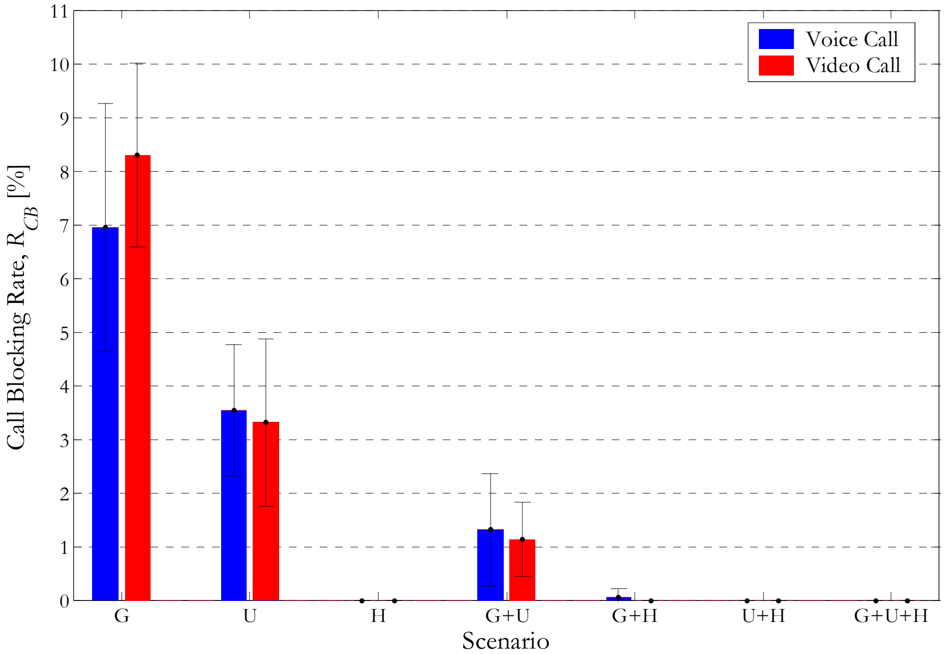

Figure 5.3. BR Statistical Analysis per Access Technology Scenario (700 users, 10 simulation runs).

Analysing the obtained results, one may conclude that the impact of user traffic variation in the Mean Bitrate indicator is in general relatively low (nearly negligible), while for the BR and Discarded Packets/Frames indicators a more significant impact is observed.

The standard deviation of the BR results (corresponding to a 68 % confidence interval under the assumption of a Gaussian process) assumes values between 0.5 % and 2.5 %. Although in mathematically terms, these values appear to be quite large in comparison to the mean, from a network performance analysis point of view, these are quite reasonable margins.

In relation to the Discarded Packets/Frames results, the observed standard deviations generally assumed values between 1 % and 5 %, which again from a network performance analysis point of view, are quite reasonable.

As previously mentioned, the Mean Bitrate variations are generally very low and can therefore be considered negligible in most cases. More specifically, the observed standard deviations of the Mean Bitrate indicator are always below 9 % (in relation to the average values), except for the G and G+U scenarios, where higher values are observed (for the data applications).

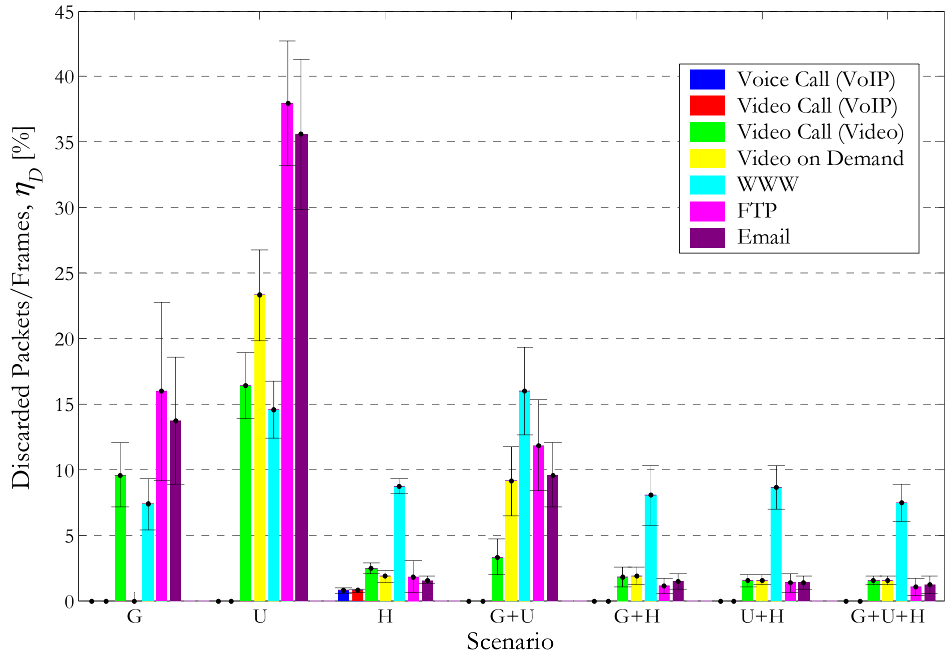

Figure 5.4. Discarded Packets/Frames Statistical Analysis per Access Technology Scenario (700 users, 10 simulation runs).

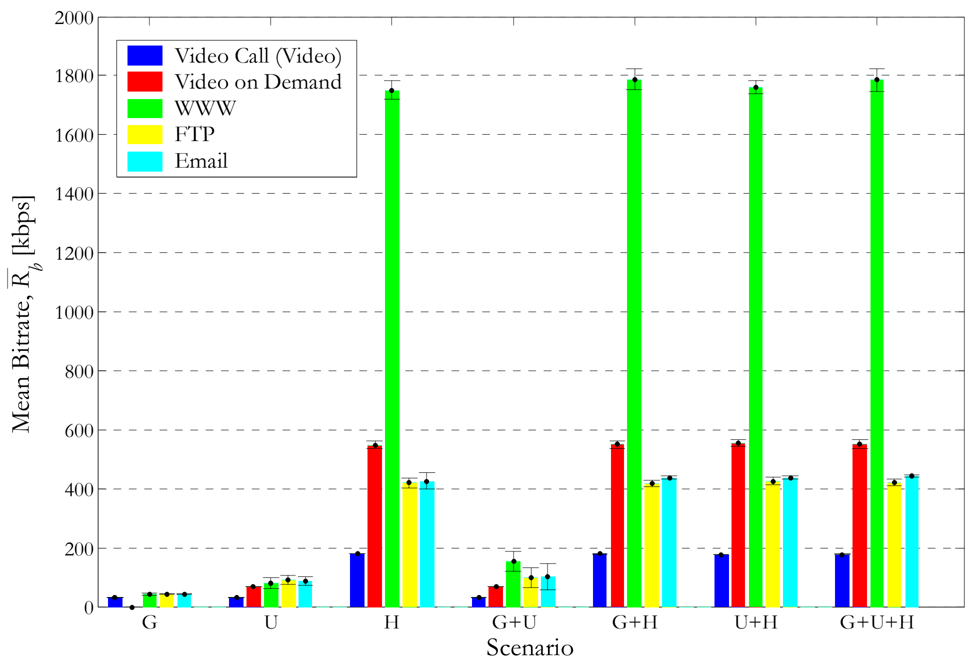

Figure 5.5. Mean Bitrate Statistical Analysis per Access Technology Scenario (700 users, 10 simulation runs).

Although the above results already show a relatively good statistical behaviour for a set of 10 simulation runs, due to the reasonably low standard deviation boundaries observed, a set of 100 simulation runs was performed for the particular case of the Access Technology Scenario with GSM and HIPERLAN/2, in order to check the reliability of the results obtained previously for 10 simulation runs. It should be noted that once again due to time constraints, this test was restricted to a specific Convergence Scenario and therefore its results have a limited validity. The obtained results for the three performance indicators are presented in Figure 5.6 to Figure 5.8.

As it can be observed, the obtained results for 10 and 100 simulation runs are quite similar for all performance indicators, which confirms the previous statements regarding the reliability of results based on only 10 simulation runs.

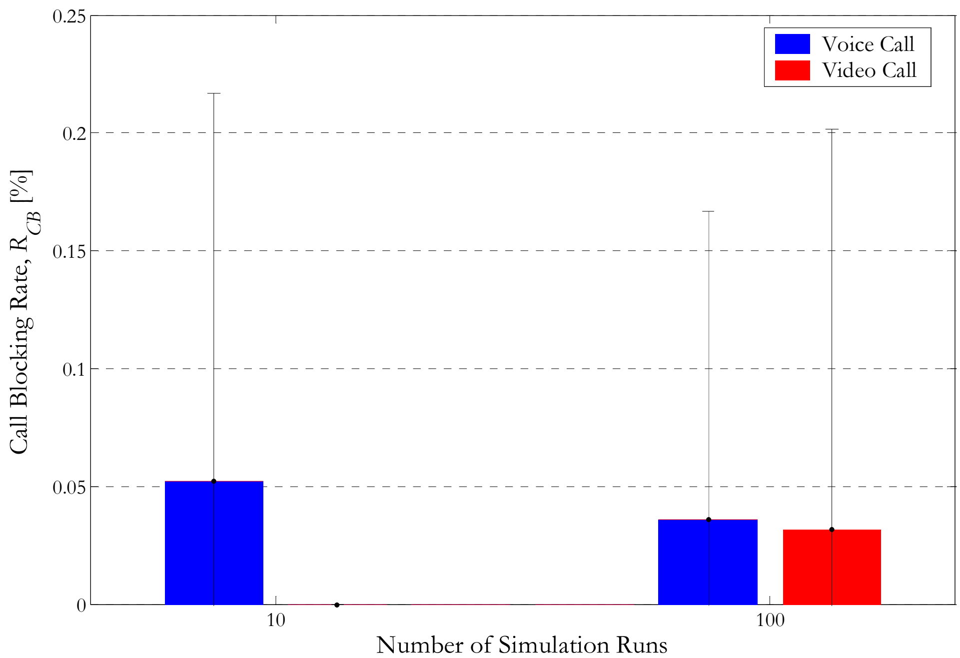

Figure 5.6. BR Statistical Analysis for 10 and 100 simulation runs (700 users, G+H).

As a general conclusion of the statistical analysis presented in this section, one can consider that ten simulation runs per Convergence Scenario provide an already reasonable statistical behaviour, and therefore one can extrapolate from them indicative error margins for results later obtained from a single simulation run. This will enable to reduce time consumption by decreasing the number of simulation runs needed while yet maintaining a reasonable degree of confidence on the obtained results.

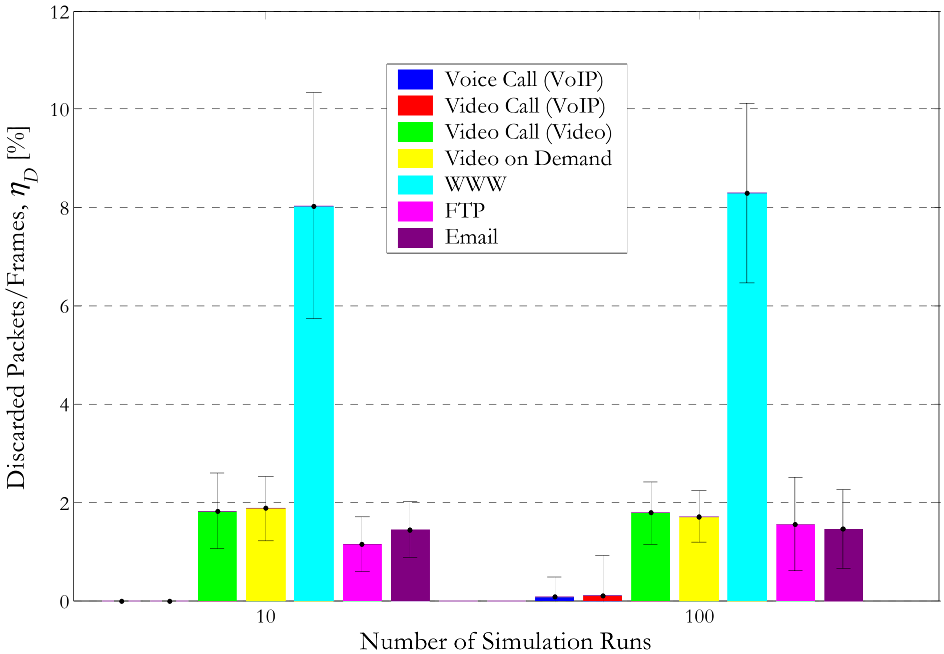

Figure 5.7. Discarded Packets/Frames Statistical Analysis for 10 and 100 simulation runs (700 users, G+H).

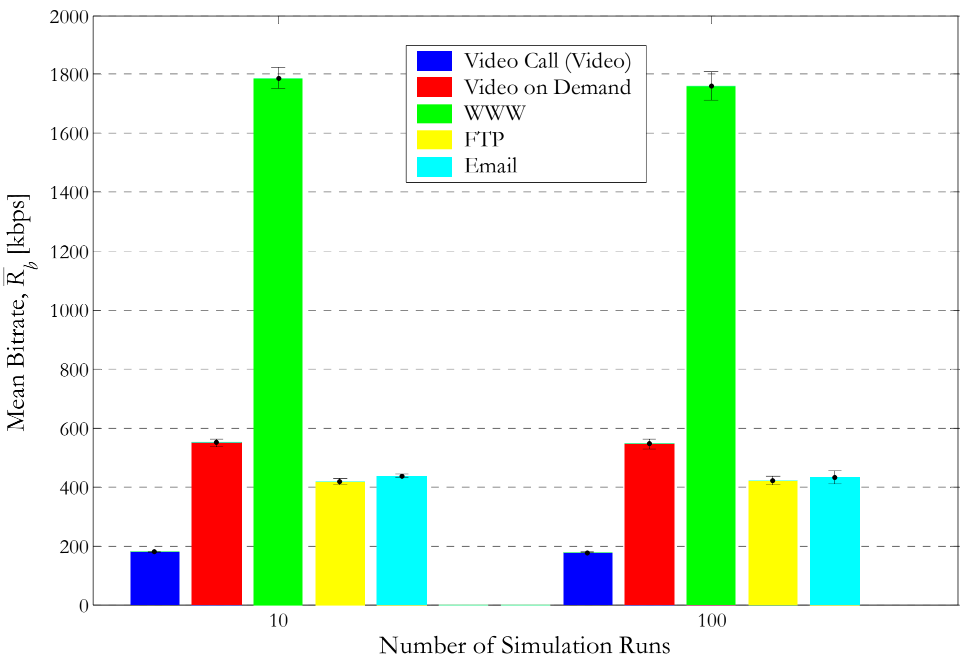

Figure 5.8. Mean Bitrate Statistical Analysis for 10 and 100 simulations (700 users, G+H).

5.3 User Scenarios Variation

In this section, one analyses the behaviour of the main performance indicators in function of the variation of the number of users, *i.e.*, varying the User Scenario, for each Access Technology Scenario. This type of analysis is useful, for example, for obtaining the number of supported users for a specific Access Technology Scenario and for certain user profiles in a converged network scenario.

The qualitative behaviour of the performance indicators are similar for the several Access Technology Scenarios, and therefore only the results for two particular cases are analysed here, as a representative set: G+U and G+U+H scenarios. The results of the remaining five Access Technology Scenarios are presented in Annex 2.

The results presented in this section (and in Annex 2) were obtained from single simulation runs, and therefore when analysing them one must take into account the associated error margins (explained in the previous section).

The first set of results corresponds to the G+U Access Technology Scenario. The associated BR, Discarded Packets/Frames and Mean Bitrate indicators are presented in Figure 5.9 to Figure 5.11, respectively. Although all performance indicators are completely interrelated, the present analysis will look at the CS applications performance indicator (BR) and the PS applications performance indicators (Discarded Packets/Frames and Mean Bitrate) separately.

In respect to the BR results, one can observe that it assumes values in the range of 0 % to 9 % (for 1 000 users), only assuming values greater that zero from 600 users onwards. As expected, the BR increases with the number of users, due to the increase of the overall user traffic demand. From a point of view of network performance, these results should be compared with typical target BR values (commonly between 1 % and 2 % per CS application). Under this aim, and taking into consideration the error margins analysed in the previous section, the current network could support between 500 and 600 users (if looking only at the BR indicator), for the given user profiles.

For the PS applications performance indicators, one can observe distinct behaviours for the video applications (Video Call and Video on Demand) and for the data ones (WWW, FTP and Email). These two cases will be analysed separately in the following paragraphs. As a general overview of the two performance indicators, one observes that packets/frames start to be discarded at 200 users and from thereon presenting an overall increasing tendency, while the Mean Bitrate indicator presents a general decreasing behaviour, as expected.

As expected, with the increase of the number of users, the percentage of discarded frames for each video application gradually increases (from 0 % to 30 %), since the mean bitrate of these applications remains constant (due to the TSMs in use: TBR 32 kbps for Video Call and TBR 64 kbps for Video on Demand). The percentage of discarded frames is higher for the Video on Demand application, since it requires the double of the capacity than the Video Call one (BHCA values are similar) and the Video on Demand application is only supported by UMTS.

In respect to the data applications, one observes a decreasing tendency of the mean bitrate values per application, gradually approaching the minimum values supported by the simulator (8 kbps for WWW and 64 kbps for FTP and Email) with the increase of the number of users. This behaviour results in a nearly constant percentage of discarded packets (averaging between 10 % and 15 %) from 300 up to 800 users. Above 900 users the percentage of discarded packets starts to increase due to the fact that the mean bitrates of the applications have started to stabilise, resulting in an effect similar to the case of video applications.

In general one can also observe the effects of prioritisation decisions, implemented in the traffic processing and in the CMM decision process, on the overall performance results, namely:

- Packets start to be discarded before (in number of users) CS calls are dropped;
- Although in a quite subjective way, the video applications present a more clear (and dominant) behaviour in relation to % of discarded frames than the data ones;
- The highest priority PS applications present generally lower percentage of discarded packets/frames than the lowest priority ones (*e.g.*, video call compared to WWW);
- Higher mean bitrates are generally achieved for higher priority data applications (*e.g.*, higher mean bitrate for WWW than for FTP and Email[5]).

[5] Note that for the current simulated Access Technology Scenario (G+U), the maximum allowed bitrate (in the PT) for all data applications (WWW, FTP and Email) is 384 kbps.

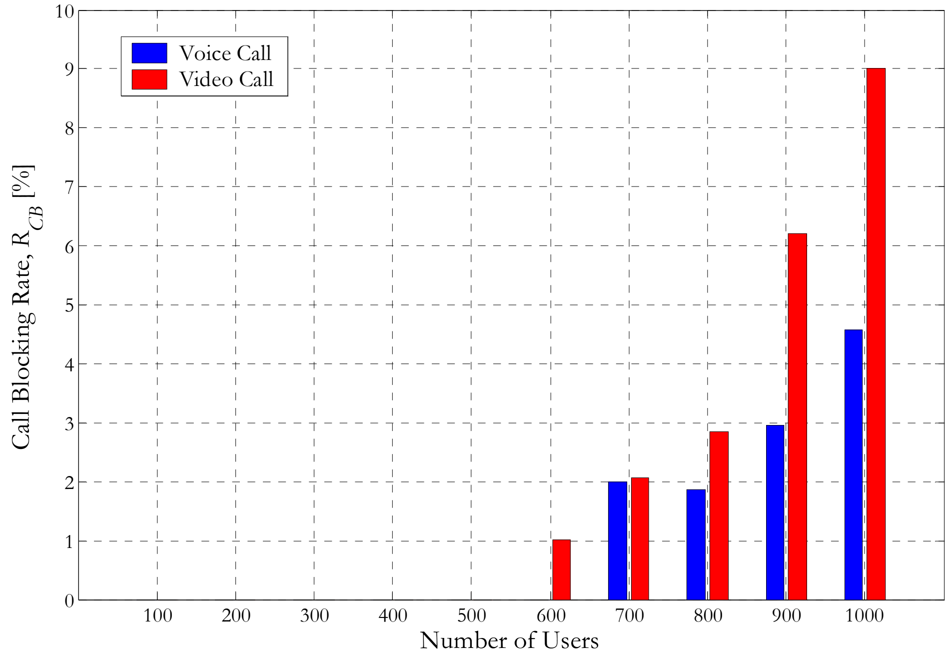

Figure 5.9. BR versus User Scenario (G+U).

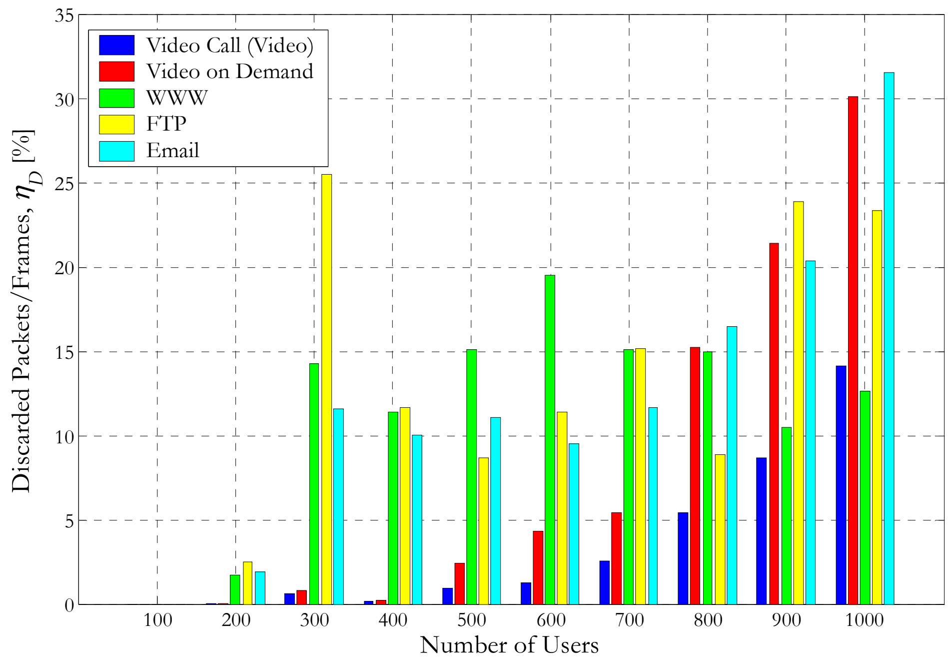

Figure 5.10. Discarded Packets/Frames versus User Scenario (G+U).

If one considers a target maximum percentage of discarded frames between 1 % and 2 % for video applications (typical values) and a packet discard rate up to 10 % (meaning up to 10 % of the packets will be delayed, potentially dropped), the overall network could support up to approximately 500 users. For this number of users, the mean bitrate of the data applications would still be acceptable (>100 kbps). As previously analysed, from a purely BR perspective, a number of users between 500 and 600 could be supported.

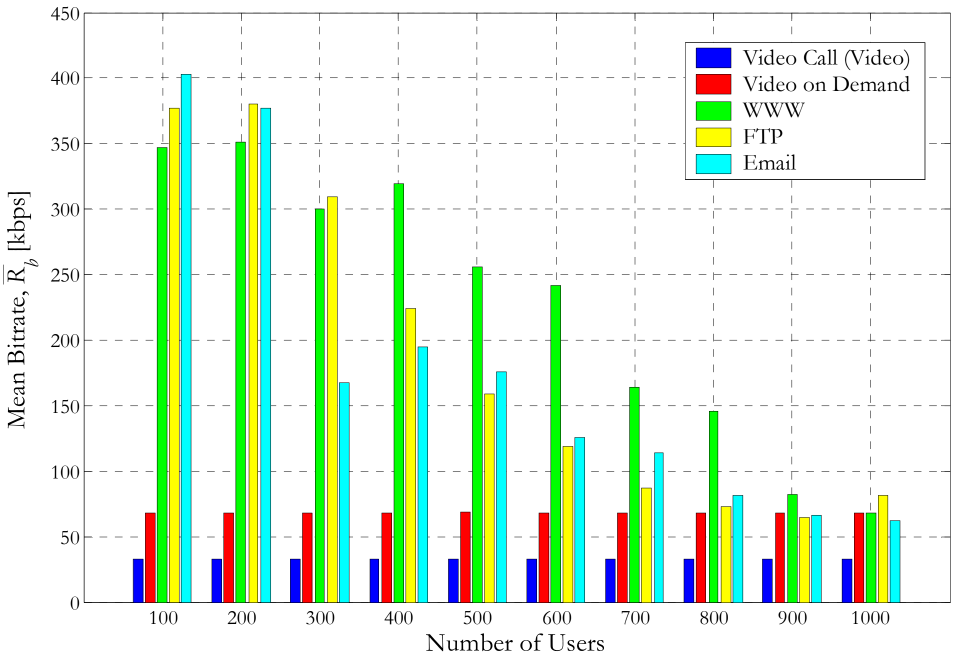

Figure 5.11. Mean Bitrate versus User Scenario (G+U).

In order to have a general and simple overview of the networks' overall performance, one can calculate the overall performance degradation parameters presented in Table 4.10 ($D_{R_{CB}}$, D_{η_D} and $D_{\overline{R}_b}$). These parameters are represented in Figure 5.12 for the Convergence Scenarios under analysis. The considered maximum target mean bitrate per PS application are given in Table 5.7.

PS Application	Maximum Target Mean Bitrate for GSM+UMTS scenario [kbps]
Video Call (video component)	34.02 (from Table 4.4, H.263 TBR 32 kbps model)
Video on Demand	68.02 (from Table 4.4, H.263 TBR 64 kbps model)
Web Browsing	384
FTP	384
EMAIL	384

Table 5.7. Maximum Target Mean Bitrate per PS Application (G+U).

As expected from the previous analysis, all three parameters have an increasing tendency, meaning an increasing overall performance degradation, with the number of users. This degradation starts to be more relevant from 500 users onwards. One can also notice that the mean bitrate degradation tends to stabilise for more than 800 users, due to the fact that the applications are already transmitting at approximately the lowest bitrate (as previously mentioned). This effect produces a greater increase in the percentage of discarded packets/frames and in the overall CS BR.

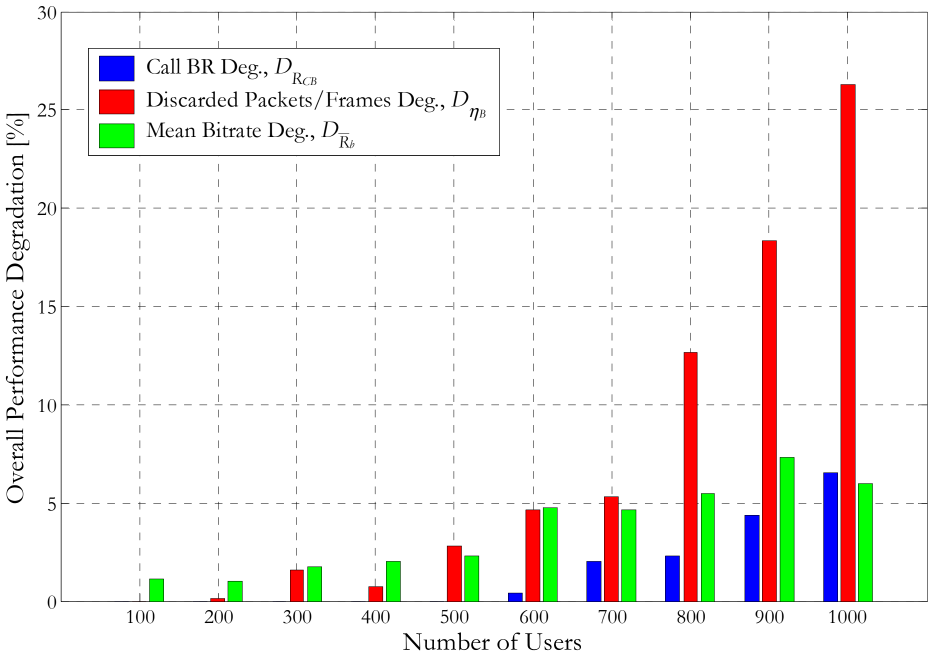

Figure 5.12. Overall Performance Degradation versus User Scenario (G+U).

Other interesting parameters to consider, and which help to understand the previous analysis, are the percentage of 1st priority decisions for both access technology ($\eta_{1PD\text{-}T}$) and application bitrate/quality ($\eta_{1PD\text{-}B}$) selections performed by the CMM, taking into account the PT in use and the available access technologies. These values are represented in Figure 5.13 for the Convergence Scenarios under consideration. One can observe a remarkable reduction in the amount of CMM 1st priority decisions starting from 500 users onwards, resulting in the previously analysed overall performance degradation.

The second Access Technology Scenario that will be analysed in this section is the G+U+H scenario, with all access technologies available. For this purpose, a similar analysis to the previous case will be performed.

For the considered number of users (up to 1000), no BR values above zero were obtained (and therefore no graphic is presented for this parameter), meaning that GSM and UMTS handled all their respective CS calls with success. In what regards to the performance indicators of the PS applications, Figure 5.14 provides the percentage of discarded packets/frames per PS application, while Figure 5.15 depicts the mean bitrates obtained per PS application.

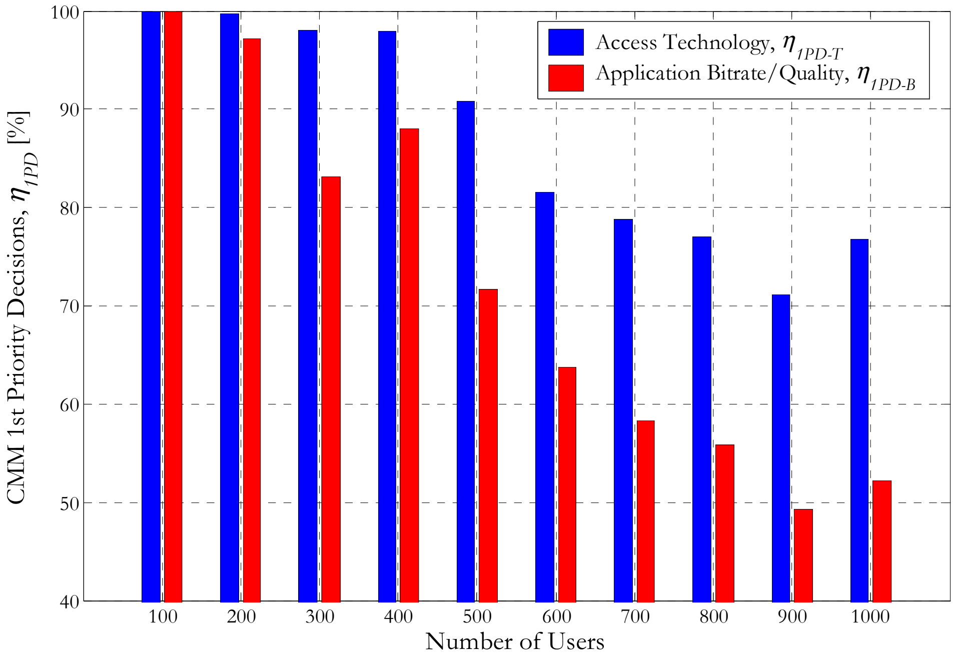

Figure 5.13. Percentage of CMM 1st Priority Decisions versus User Scenario (G+U).

As expected, the percentage of discarded packets/frames generally increases with the number of users, while the mean bitrate per PS application generally decreases. Since the three access technologies provide a higher capacity than in the previous case (G+U), the performance degradation is therefore lower. The mean bitrate of the applications only starts to decrease from their maximum allowed values when more than 600 users are present, although the decrease is considerably low. Even for 1 000 users, the mean bitrates obtained for all PS applications are still very high. This can also be observed in Figure 5.16, where the overall mean bitrate degradation for 1 000 users is approximately 10 % (in comparison to the maximum allowed values). The considered maximum target mean bitrate per PS application used in Figure 5.16 is given in Table 5.8.

PS Application	Maximum Target Mean Bitrate for G+U+H scenario [kbps]
VoIP (Voice Call and voice component of Video Call)	8.69 (from Table 4.4)
Video Call (video component)	180.88 (from Table 4.4, GBAR H.263 VBR model)
Video on Demand	573.39 (from Table 4.4, GOP GBAR MPEG-4 HQ model)
Web Browsing	2000
FTP	384
EMAIL	384

Table 5.8. Maximum Target Mean Bitrate per PS Application (G+U+H).

The percentage of discarded packets/frames increases until the number of users reaches approximately 800 users and from thereon tends to stabilise (up to 1 000 users). This can be more clearly observed in Figure 5.16, where the overall discarded packets/frames degradation stabilises between 3 % and 4 %. This effect is due to the compensation of the increase of the number of users with the mean bitrate reduction observed in Figure 5.15. Considering again a target maximum percentage of discarded frames between 1 % and 2 % for video applications and a packet discard rate of up to 10 %, the overall network could support up to approximately 700 users. For this number of users, the mean bitrates of the data applications are still close to their maximum allowed values.

Observing now the percentage of 1st priority decisions performed by the CMM, Figure 5.17, it is quite clear that there starts to exist a considerable reduction in the amount of CMM 1st priority decisions between 700 and 800 users, resulting in the previously analysed overall performance degradation.

As previously mentioned, the performance results of the remaining five Access Technology Scenarios (G, U, H, G+H and U+H) are presented in Annex 2. These results present in general a similar behaviour to the ones considered in this section, and are therefore not analysed in detail.

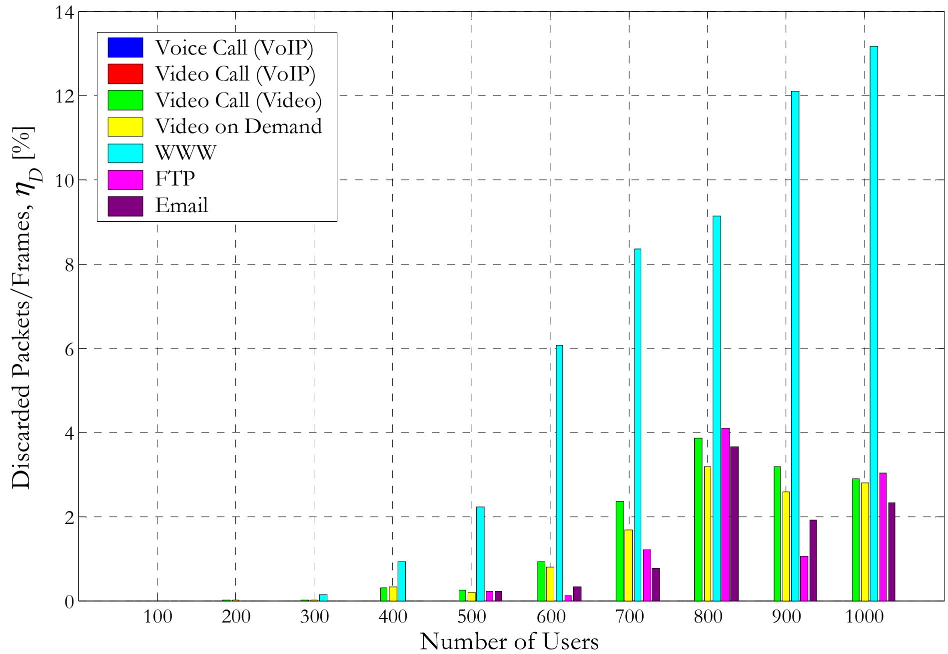

Figure 5.14. Discarded Packets/Frames versus User Scenario (G+U+H).

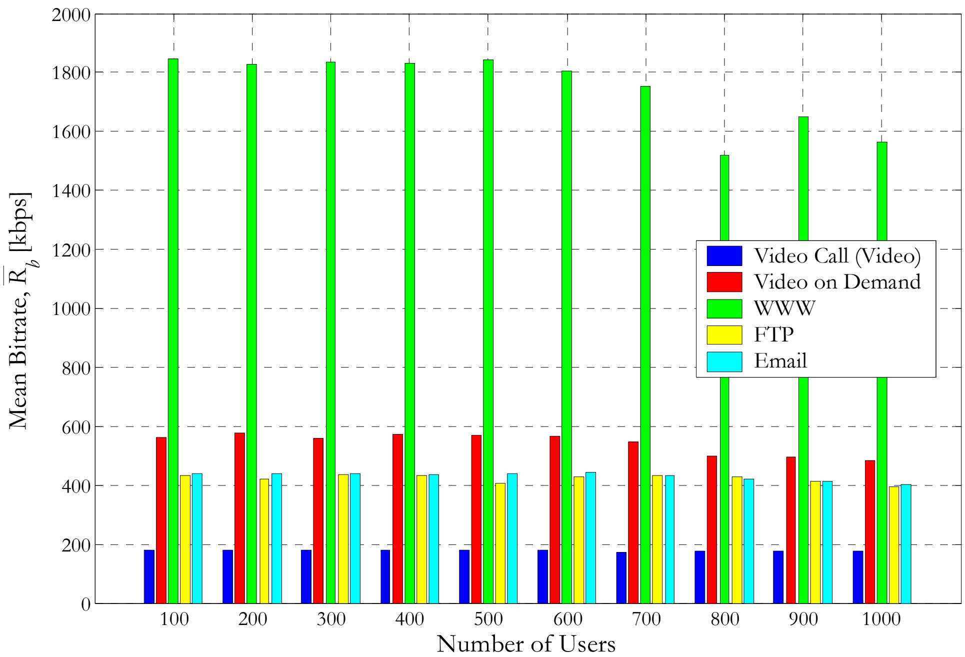

Figure 5.15. Mean Bitrate versus User Scenario (G+U+H).

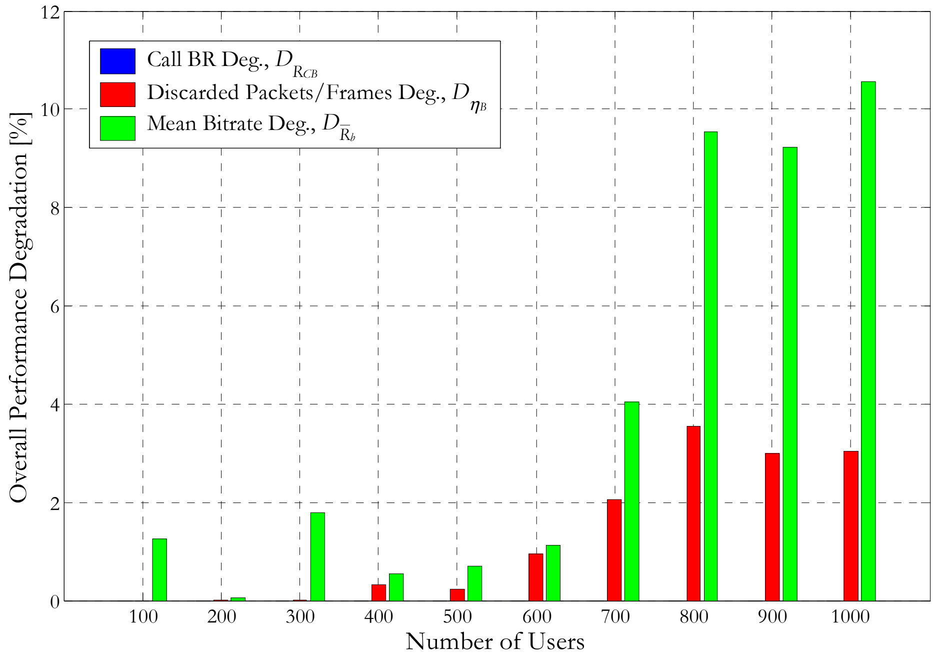

Figure 5.16. Overall Performance Degradation versus User Scenario (G+U+H).

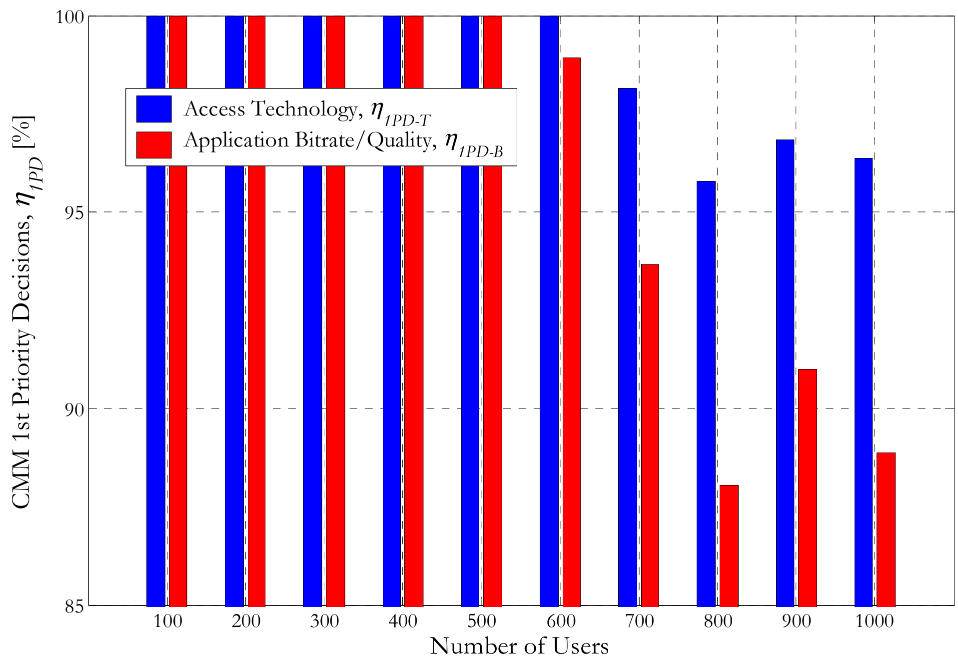

Figure 5.17. Percentage of CMM 1st Priority Decisions versus User Scenario (G+U+H).

5.4 Access Technology Scenarios Variation

The third and last approach adopted for analysing the simulator results is presented in this section. This approach will focus on the analysis of the impact of different Access Technology Scenarios (*i.e.*, different available access technologies) in the overall performance indicators, for a specific User Scenario (*i.e.*, number of users) and for a particular instance of the Applications Scenario (*i.e.*, maintaining the same UTVs for all the different Access Technology Scenarios). By maintaining the same instance of the Applications Scenario, a more precise comparative analysis can be performed between the different outputs for the several Access Technology Scenarios. This type of analysis is useful, for example, for the evaluation of which technology or combination of technologies are ideal for supporting a given number of users with specific traffic profiles in a converged network scenario.

Only one User Scenario will be considered in the present analysis: 700 users. The results for other User Scenarios, namely 500, 600, 800, 900 and 1 000 users are presented in Annex 3, which can be analysed in a similar way as the one presented in the following paragraphs.

Figure 5.18 presents the obtained results for the CS applications BR, while the percentage of discarded packets/frames and the mean bitrate values for the PS applications are presented in Figure 5.19 and Figure 5.20, respectively. As one can expect, when more technologies are available better overall performances are achieved. This fact is quite clear for all performance indicators (for instance, the BR parameter only presents values greater than zero when HIPERLAN/2 is not available).

Looking firstly at the individual technology scenario performances (GSM, UMTS and HIPERLAN/2), one can observe the following results for each performance indicator:

- BR (only applicable to GSM and UMTS): UMTS presents a lower BR in comparison to GSM;
- % of Discarded Packets/Frames: HIPERLAN/2 presents the overall lowest values while UMTS presents the highest ones;
- Mean Bitrate: HIPERLAN/2 obtained again the overall best results, followed by UMTS, that presents slightly higher values than GSM.

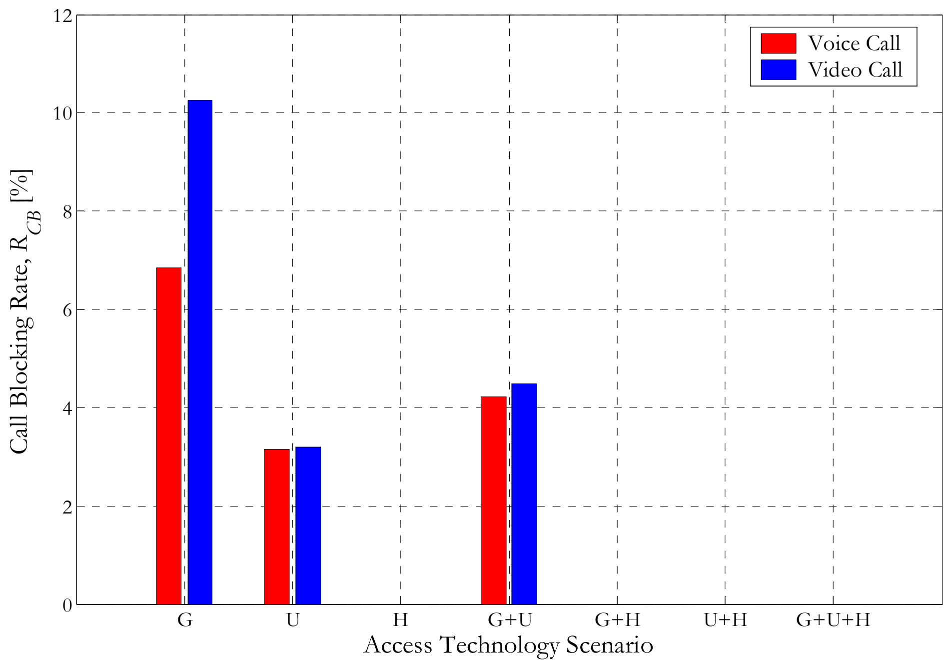

Figure 5.18. BR versus Access Technology Scenario (700 users).

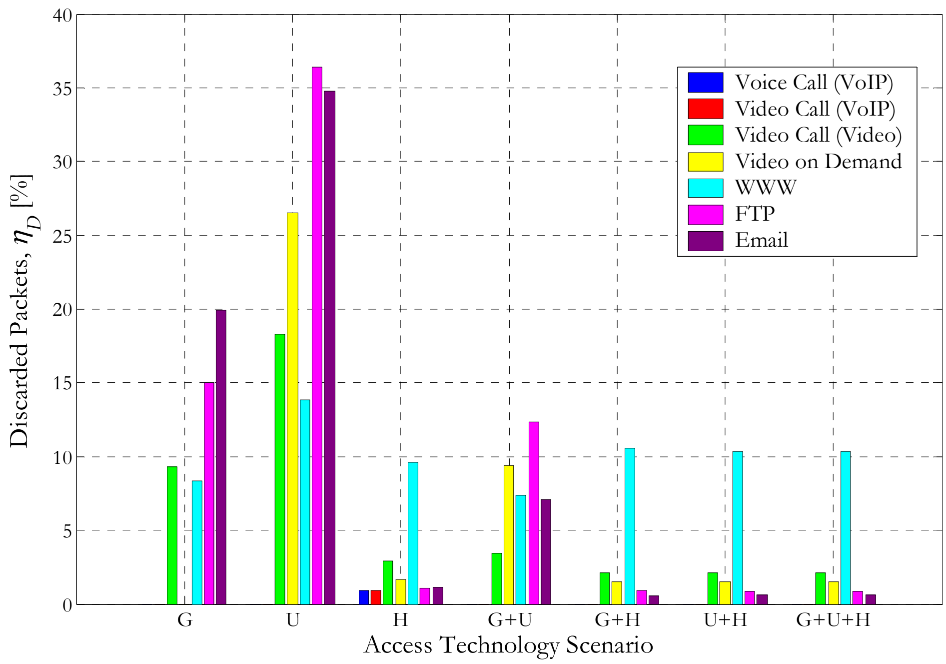

Figure 5.19. Discarded Packets/Frames versus Access Technology Scenario (700 users).

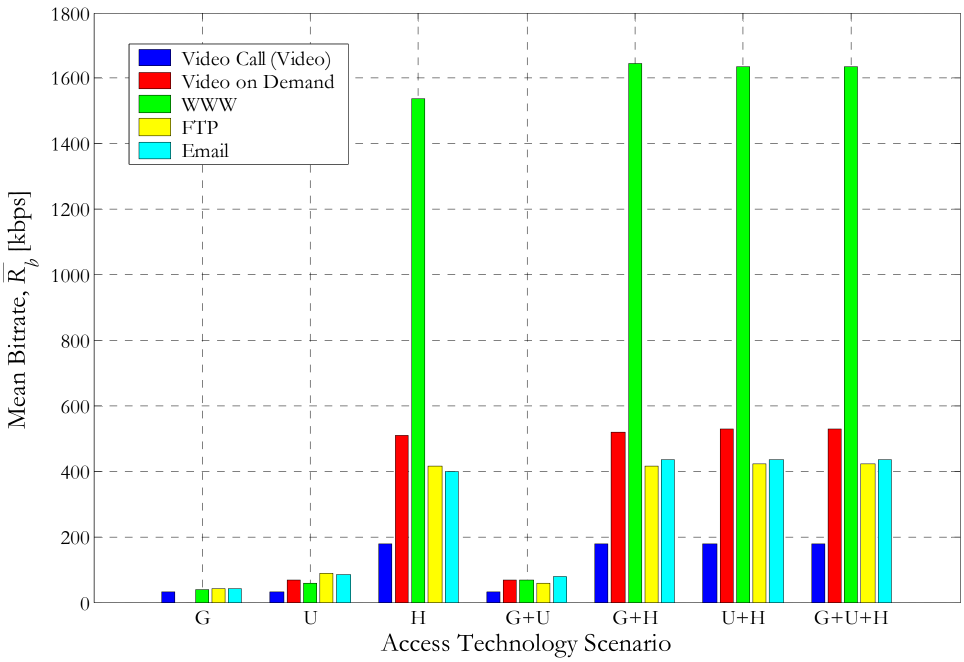

Figure 5.20. Mean Bitrate versus Access Technology Scenario (700 users).

In order to help on the analysis of these results, one can again report to the overall performance degradation parameters and the percentage of 1st priority decisions performed by the CMM. These parameters are represented in Figure 5.21 and Figure 5.22, respectively. In order to have the same comparative basis between the different Access Technology Scenarios, these figures were obtained using exactly the same target objectives for all scenarios. In particular, the maximum target mean bitrate that is used in Figure 5.21 for each PS application (for all Access Technology Scenarios) is given by Table 5.8 (corresponding to the maximum target values of the PT). The values presented in Figure 5.22 also assume, independently of the access technology scenario, the first priority decision (for both access technology and application quality/bitrate selection) that is considered in the overall PT.

As expected, the HIPERLAN/2 system, due to its much higher capacity, presents better performances than the other individual systems. The worst performance of the UMTS system in terms of discarded packets/frames in comparison to the GSM system is due to the fact that the Video on Demand application is not supported by the latter system, and therefore less capacity is required. In an equivalent situation (*i.e.*, deactivating the Video on Demand application), the UMTS system (taking into account the parameters of Table 5.1) would present better results than the GSM system (for all three performance indicators).

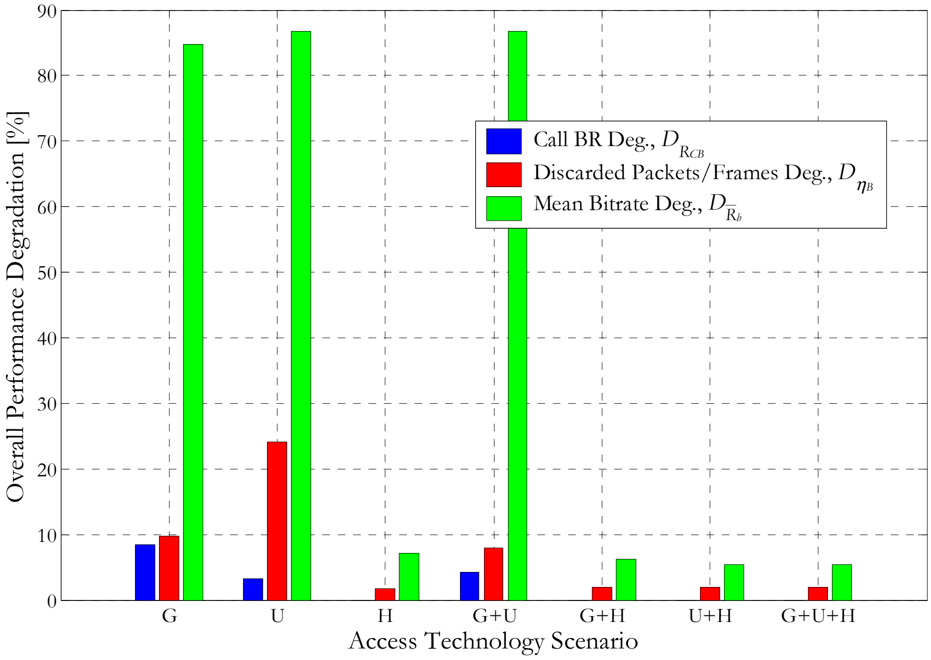

Figure 5.21. Overall Performance Degradation versus Access Technology Scenario (700 users).

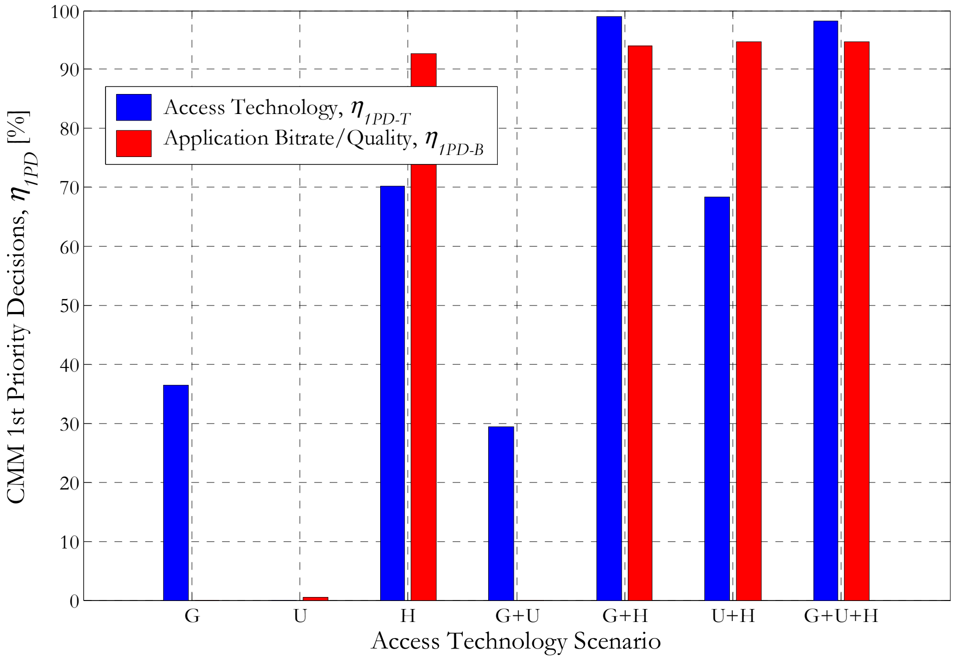

Figure 5.22. Percentage of CMM 1st Priority Decisions versus Access Technology Scenario (700 users).

The overall performance degradation parameters, as well as the percentage of 1st priority decisions performed by the CMM confirm the better performance of the HIPERLAN/2 system and relatively low performance of the GSM and UMTS systems.

Analysing the results for the cases where more than one technology is available, one can observe that, for the number of users under consideration, the performance obtained is approximately the same (close to its maximum values) for the cases where the HIPERLAN/2 system is available. This is due to the high capacity of the HIPERLAN/2 system, which handles nearly all the PS traffic, while the other system(s) handle without any blocking the CS calls. More specifically one can observe the following facts for the multi-technology scenarios where HIPERLAN/2 is present:

- zero BR is always obtained;
- the % of discarded packets/frames as well as the mean bitrate per PS application present approximately the same values for all scenarios.

These results can be further confirmed and understood by observing the overall performance degradation parameters, with values lower than 6 %, and the percentage of CMM 1st priority decisions, generally above 90 %.

For the User Scenario under analysis (700 users), one can conclude that there is no significant benefit of combining the three systems in relation to the results obtained for the 2 technology scenarios where one system is HIPERLAN/2, since the same overall results are approximately obtained. In fact, the standalone HIPERLAN/2 scenario already provides nearly the same overall performance than the multi-technology scenarios where HIPERLAN/2 is available.

The G+U scenario, although with much lower performance than the scenarios with HIPERLAN/2, obtained better results from the point of view of the percentage of discarded packets/frames, than the UMTS standalone scenario, maintaining similar values for the other parameters. More specifically, the G+U scenario presented similar overall results for the CS calls BR and mean bitrate indicators in comparison to the UMTS standalone scenario, but providing a much lower percentage of discarded packets/frames for the PS applications.

Chapter 6

Conclusions

This chapter finalises this work, summarising conclusions and pointing out aspects to be developed in future work.

The current thesis intended to analyse the overall performance of a converged multi-technology system (convergence benefits), mainly focussing on capacity aspects of the radio interface. Three technologies have been considered for this purpose: GSM/GPRS, UMTS and HIPERLAN/2. The analysis was performed with the aid of a newly developed simulation platform, considering several scenarios. Different approaches were followed for the analysis of the overall performance results, namely:

- Analysis of the impact of users traffic variations (in order to analyse the impact of the statistical behaviour of user generated traffic in the overall performance);
- Analysis of the impact of the number of users in specific scenarios corresponding to different available access technologies (*e.g.*, for obtaining the number of supported users for a specific Access Technology Scenario and for certain user profiles in a converged network scenario);
- Analysis of the impact of the availability of different access technologies for a particular number of users (*e.g.*, for evaluation of what technology or combination of technologies are ideal for supporting a given number of users with specific traffic profiles in a converged network scenario).

A key and novel concept adopted in the development of the present work is the Convergence Manager. This new element, which is present at both the terminal and network sides, is basically responsible for the implementation of all convergence functionalities in the overall system. Basically, the CM allocates the packets (on a session basis) coming from different users using various applications to different access technologies (and vice-versa). In the implemented simulation platform, a single convergence perspective/strategy was implemented based on a highest throughput (highest available/possible transmission bitrate per application) / best system (priority is given to the most appropriate application-oriented access technology) approach.

As a first step for the development of the present work, the radio interfaces of the three considered technologies, GSM/GPRS, UMTS and HIPERLAN/2, have been studied with the aim of establishing simple capacity models per individual technology to be further implemented in the simulation platform. In the implementation of these capacity models, several approximations were considered for each technology in order to enable the desired analysis in the context of a Master Thesis. Nevertheless, there was always the concern that the taken approximations kept the essential characteristics of each technology standard under consideration. An example of the adopted approximations can be the considered air interface load estimation in DL for UMTS. The simulator considers only the throughput based approach (DL LF) and does not take into account the power based one. This approximation was taken since, besides the developed simulation tool not supporting propagation aspects, the target scenarios to be simulated assume a small physical area (and consequently small cell sizes), and therefore, in this case, the limiting UMTS load measurement is the DL LF. Other approximations include, for example, "rounded" MAC frame durations for GSM and GSM/GPRS, ideal power control for UMTS and average PHY mode consideration for HIPERLAN/2.

Six different applications have been considered for user traffic generation: Voice Call, Video Call, Video on Demand, Web Browsing, Email and FTP, representing all four service classes (with different QoS requirements) defined by 3GPP. These applications and the supporting services have been characterised in detail. In order to implement the user traffic generation mechanisms in the simulation platform, specific traffic source models, as well as call/session generation and duration processes, were necessary for each considered application. For this purpose, an extensive literature survey has been performed to obtain such traffic source models and respective parameterisation values, taking into consideration the three supporting access technologies. A complete mapping between the considered technologies (GSM/GPRS, UMTS and HIPERLAN/2), applications and traffic source models has been performed, forming a basis for the simulator development. In particular, the voice and video components of the Video Call application (with separate traffic source models per component) can be transmitted via separate access technologies (although synchronisation issues can be a limiting factor in a real network scenario).

The developed simulation platform, implemented in Visual C++ programming language, is based on two main functional blocks:

- the User Traffic Generation Module, which is responsible for generating the users and their associated traffic (by means of the implemented traffic source models). The outputs of this block are User Traffic Vectors per active application per user, providing complete information on the beginning instants and durations of CS calls, and on the beginning instants and sizes of each packet belonging to PS sessions;
- the Traffic Processing Engine, which is the main functional entity of the simulator

as it performs all the "convergence" and traffic processing tasks. It receives the User Traffic Vectors generated by the User Traffic Generation Module and processes them according to different convergence perspectives/strategies (by means of a Convergence Manager Module and three Access Technology Traffic Processing Modules).

For simplicity reasons, the simulator only considers downlink user traffic processing, disregarding the uplink components, due to the high asymmetry of the majority of the considered applications, with a higher capacity demand on the downlink direction, and also because, in general, the downlink component is capacity limited, in opposition to the uplink components which are usually interference/power limited. All Mobile Terminals have been considered to be capable of receiving traffic through all available access technology networks. Several user-defined inputs parameterise the simulator, enabling the simulation of different user, applications and technology scenarios. The simulator provides several types of output indicators for overall performance evaluation, namely CS (call blocking rate) and PS (% discarded packets/frames and mean bitrate per application) traffic statistics.

The developed simulator constitutes a flexible and upgradeable platform, allowing, for example, an easy inclusion of different convergence perspectives/strategies. Hence, this simulation platform provides a complete functional framework for the evaluation of the convergence of mobile and wireless communication systems.

All different scenarios (Convergence Scenarios) that were analysed in the present work have been characterised according to their three main components: access technologies, users and applications. Three major performance indicators were considered for the overall performance analysis of these simulated Convergence Scenarios: CS calls BR, percentage of PS discarded packets/frames and mean bitrate obtained per PS application. Auxiliary parameters, helping in the results analysis, have also been considered, namely the overall performance degradation parameters (joining the results for all applications in unique parameters) and the percentage of 1st priority decisions performed by the CMM in each simulation. On average, each simulation run (corresponding to one Busy Hour simulation) lasted approximately 1 hour in a PIV 2.66 GHz computer with 768 Mbyte of RAM.

From the statistical analysis of user traffic variations performed in this work, it was verified that ten simulation runs per Convergence Scenario provide an already reasonable statistical behaviour, therefore allowing the extrapolation of indicative error margins for results obtained from single simulation runs. From a network performance point of view, the error margins (*i.e.*, variability) observed for the performance indicators were quite acceptable, mainly taking into account the fact that the increase of the parameters for some applications are compensated with their decrease for the others, maintaining the overall performance with a relatively low variation. These results were obtained analysing the results for all possible access technology scenarios, considering a fixed User Scenario of 700 users. In particular, considering 10 simulation runs per scenario, the obtained BR standard deviations assumed (absolute) values between 0.5 % and 2.5 %, while the percentage of discarded packets/frames standard deviations were generally between 1 % and 5 % (absolute values). The obtained mean bitrate standard deviations were also quite low (always below 9 % relatively to the average values, except for the GSM and GSM+UMTS scenarios, where higher values were observed for the data applications). These facts allow for a reasonable degree of confidence on results obtained with single simulation runs, reducing the number of needed simulation runs and consequent time consumption.

The impact of the number of users has been analysed for two specific Access Technology Scenarios: GSM+UMTS and GSM+UMTS+HIPERLAN/2, maintaining the same Applications Scenario. The evolution of the performance indicators with the increase of the number of users has behaved as expected, *i.e.*, the CS calls BR and the percentage of discarded packets/frames increases while the mean bitrate obtained per PS application decreases. The effects of the prioritisation schemes implemented in the simulator were also noticed (*e.g.*, higher mean bitrates were generally achieved for higher priority data applications and higher priority PS applications presented generally lower percentage of discarded packets/frames than the lowest priority ones). Considering a target maximum percentage of discarded packets/frames of 10 % for data applications and between 1 % and 2 % for video applications and a target maximum BR for CS calls also between 1 % and 2 %, one has observed that the GSM+UMTS Access Technology Scenario supports up to approximately 500 users, while the GSM+UMTS+HIPERLAN/2 one supports up to approximately 700 users. Although the difference in amount of supported users between these two scenarios is not very significant, the achieved mean bitrates per PS application is much higher for the GSM+UMTS+HIPERLAN/2 scenario, allowing the support of a broader range of high bitrate services.

From the analysis of the impact of the availability of different access technologies, maintaining the same User and Applications Scenarios, one has observed that, as expected, better overall performances are achieved when more technologies are available. Considering only the standalone Access Technology Scenarios (*i.e.*, GSM, UMTS and HIPERLAN/2 scenarios), the HIPERLAN/2 system obtained the best performance results, due to its much higher capacity. The results obtained for the cases where more than one access technology was available, show that, for 700 users (User Scenario that was analysed), the overall performance is approximately the same (and close to its maximum values) for the cases when HIPERLAN/2 is available. This is again due to the high capacity of the HIPERLAN/2 system that handles nearly all the PS traffic, while the other system(s) handle without any blocking the CS calls. Hence, for the analysed User Scenario, one can conclude that there is no significant benefit of combining the three systems in relation to the results obtained for the 2 technology scenarios where one system is HIPERLAN/2. In fact, the standalone HIPERLAN/2 scenario already provides nearly the same overall performance than the multi-technology scenarios where HIPERLAN/2 is available.

The GSM+UMTS scenario, although with much lower performance than the scenarios with HIPERLAN/2, presented better results from the point of view of the percentage of discarded packets/frames, than the standalone UMTS scenario, maintaining similar performances in relation to the other parameters.

The use of several HIPERLAN/2 APs (more specifically 4, for coverage enhancement) was initially considered, instead of only one, but all the user traffic (for the User Scenarios considered) would be perfectly supported (at maximum bitrates) by this system itself due to the high offered capacity. This would in fact mean that in hot-spots with HIPERLAN/2 coverage (with several APs), and if all users had convergence capable terminals (supporting HIPERLAN/2 connectivity), the deployment of other systems would not present considerable advantages. Although HIPERLAN/2 (and other WLAN systems) offers a high capacity on the radio interface, in a real network deployment one must also take into account the backbone transmission capacities, which can create a considerable bottleneck on the overall supported traffic. Summarising, and considering a convergence approach, HIPERLAN/2 (or other WLAN systems) can be foreseen as optimum candidates for coverage of hot-spot areas (in or even not in conjunction with 2G and 3G systems), providing high throughputs for the user applications, and the other systems (GSM/GPRS and UMTS) as the providers of broader coverage and still supporting all applications (although with less performance in terms of bitrate).

This is not at all a finalised work, mainly due to the complexity of the subject underlying this

Thesis. A wide range of issues can be improved, leading to the evaluation in a more realistic and detailed manner of the possible benefits of convergence of mobile and wireless systems. Some of these issues are left as suggestions for work continuation:

- implementation of additional access technologies, namely broadcasting systems, in order to study the convergence benefits of such systems with other cellular technologies;
- inclusion of propagation aspects into each implemented access technology, allowing a better and more detailed analysis of convergence between systems, since, based on the channel conditions between the MTs and the BSs, different technology routing decisions can be performed (*e.g.*, in response to fast fading);
- in depth analysis of other application scenarios, namely with different traffic profiles per user type, and their impact in the overall performance of a converged network;
- enhancement of the session based routing of the current convergence manager implementation to a packet based routing; this, however, requires a much more detailed analysis of the protocols, namely for signalling and synchronisation;
- implementation of other convergence perspectives/strategies (*i.e.*, different priority tables), *e.g.*, different (dynamic or static) priority tables per user type (or even per user) allowing a more realistic evaluation of a converged network where the user selects (pre-configures) his priorities based on price, location, type of call (*e.g.*, family calls routed with highest quality independent of price);
- improvement of the traffic processing algorithms, namely with the implementation of the involved protocol stacks and of (new) scheduling and traffic shaping techniques (namely for packet based routing). In this case, the performance measures of packet data transmission will also include: average (or total) packet delay and packet retransmission rate;
- support of different types of MTs (*e.g.*, some with access to one or two types of access technologies and others with access to all available technologies); this will also allow a more realistic evaluation of a converged network, since it is expected that in such networks, only a part of the users (namely in an initial stage) would have fully convergence capable terminals, while others would continue with their existing single technology terminals (but still using the same network).

Annex 1

Validation of Random Number Generators

In the present annex, a simple validation of each implemented RNG is presented, based on a comparison with the respective theoretical PDFs/PFs.

For the implementation of statistical traffic source models, Random Number Generators (RNG) are essential. In the case of the traffic source models considered in the present report, 10 different RNGs were necessary for their implementation. More specifically, RNGs for the following statistical distributions were used: Uniform, Gamma, Beta, Exponential, Geometrical, Normal, LogNormal, Pareto, Poisson and Weibul.

The selected Uniform RNG algorithm is the function "ran1()" presented in [PTVF92], since, according to the authors, no statistical test is known to fail with it. This function guarantees uncorrelation between consecutive samples for a number of samples up to 10^8. The remaining RNG algorithms are based on transformations of this Uniform RNG.

In the present annex, a simple validation of each implemented RNG is presented, based on:

- a comparison between the theoretical PDFs (for the continuous random variables) or PFs (for the discrete random variables: Geometric and Poisson) and the PDF/PF equivalent modified histograms obtained from data generated by the RNGs (Figures A1.1 to A1.10);
- the calculation of the respective Mean Square Error, MSE (Table A1.1).

For this purpose, a set of data containing a total number of occurrences N_{oc}=5×10^4 for each RNG was analysed.

Histograms within an observation window of [x_{min}, x_{max}] sub-divided into N_I intervals of width I were built. The PDF/PF equivalent modified histograms, $p_x[n]$, were obtained normalising the histograms by N_{oc} and I:

$$p_x[n] = \frac{1}{N_{oc} \cdot I} \cdot n_{oc}[n], \; n = 1, 2, ..., N_I , \tag{A1.1}$$

where $n_{oc}[n]$ corresponds to the number of occurrences within interval n.

The MSE obtained for the data of each RNG is presented in Table A1.1 and was calculated as:

$$\mathrm{MSE} = \mathrm{E}\left[\left(p_x[x_n] - p_x[n] \right)^2 \right] \tag{A1.2}$$

where E[x] is the expected value of x and $p_x[x_n]$ is the discretised theoretical PDF or the theoretical PF of the statistical distribution under consideration. The x_n values of the discretised theoretical PDF correspond to the centre values of each of the N_I intervals under consideration.

RNG	x_{min}	x_{max}	I	MSE
Uniform (min=0, max=1)	0	1	0.01	1.90×10^{-3}
Gamma (β=9, λ=100)	0	2500	25	3.58×10^{-10}
Beta (p=1, q=8)	0	1	0.01	1.40×10^{-3}
Exponential ($1/\mu_{CD}$ =3)	0	25	0.25	3.35×10^{-6}
Geometric (P_{PC}=0.2)	0	35	1	3.84×10^{-7}
Normal (μ_N=0, σ_N=1)	-4	4	0.1	3.28×10^{-5}
LogNormal (μ_{LN}=0.97, σ_{LN}^2=4.38)	0	100	1	1.90×10^{-5}
Pareto (α_p=1.1, k=81.5)	0	1200	10	3.90×10^{-8}
Poisson (λ_c=60)	30	90	1	4.16×10^{-7}
Weibul (λ_k=0.45, β_k=0.80)	0	25	0.25	2.49×10^{-4}

Table A1.1. Observation parameters and MSE of each RNG.

From the analysis of the graphs, one can conclude that for such a set of data, the PDF/PF equivalent modified histograms present an acceptable approximation to the respective theoretical PDFs/PFs. The obtained values for the MSE are reasonably low in comparison to the values assumed by the theoretical PDFs/PFs. Therefore, one can assume the liability of the above RNGs for the intended simulation purposes, which encompass the generation of a number of samples of each statistical distribution in the order of magnitude of 10^4.

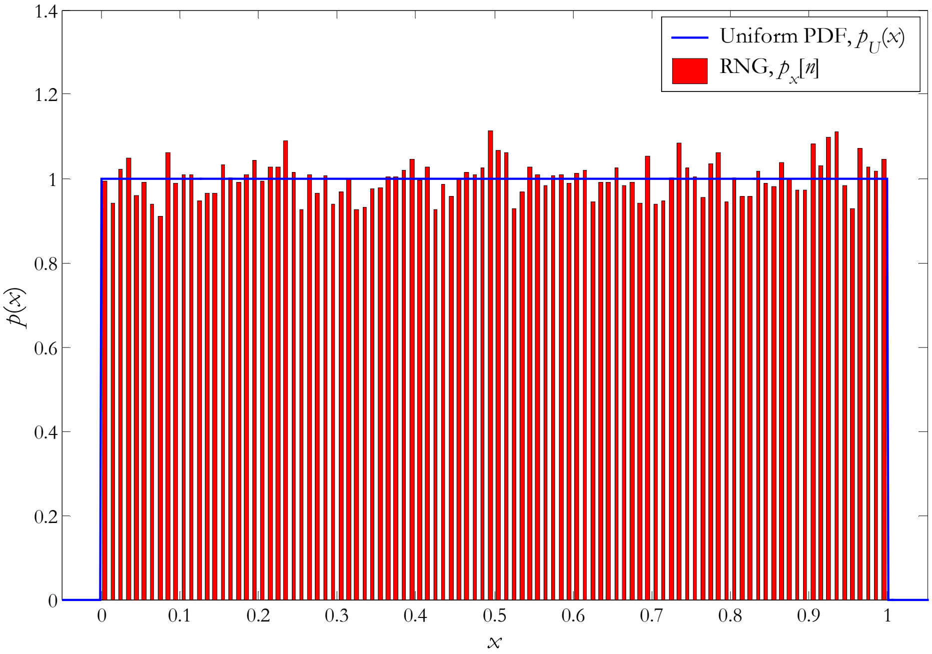

Figure A1.1. Comparison between the Uniform PDF and the PDF equivalent modified histogram obtained from the Uniform RNG (min=0, max=1).

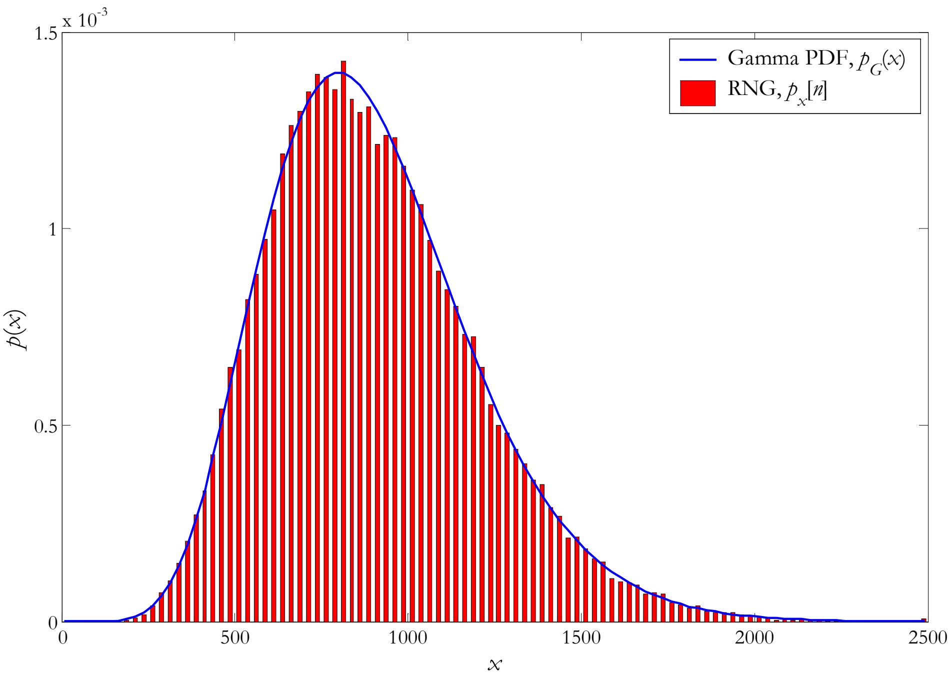

Figure A1.2. Comparison between the Gamma PDF and the PDF equivalent modified histogram obtained from the Gamma RNG (β=9, λ=100).

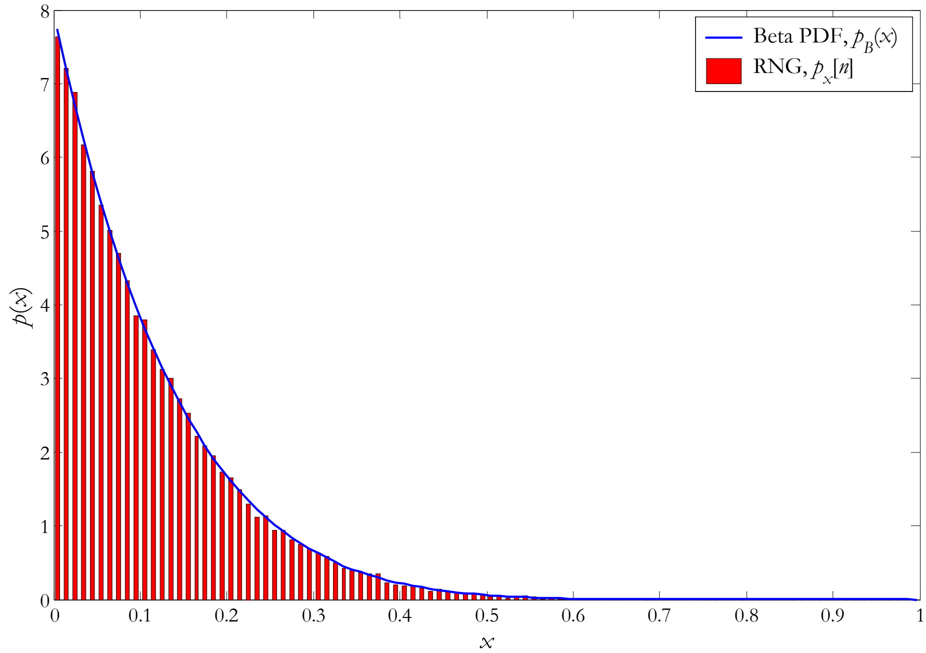

Figure A1.3. Comparison between the Beta PDF and the PDF equivalent modified histogram obtained from the Beta RNG (p=1, q=8).

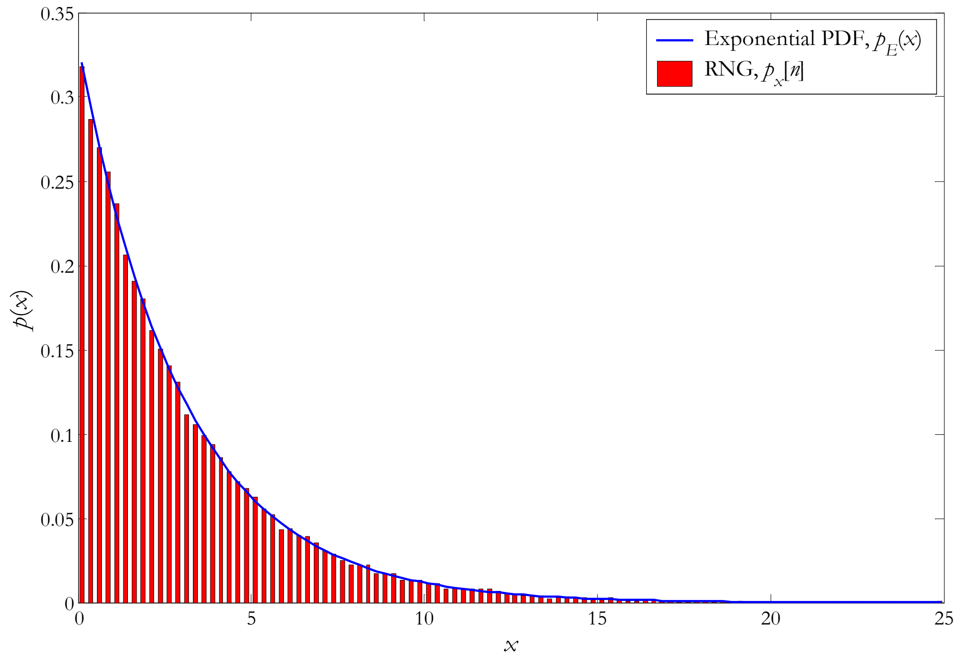

Figure A1.4. Comparison between the Exponential PDF and the PDF equivalent modified histogram obtained from the Exponential RNG ($1/\mu_{CD}$ =3).

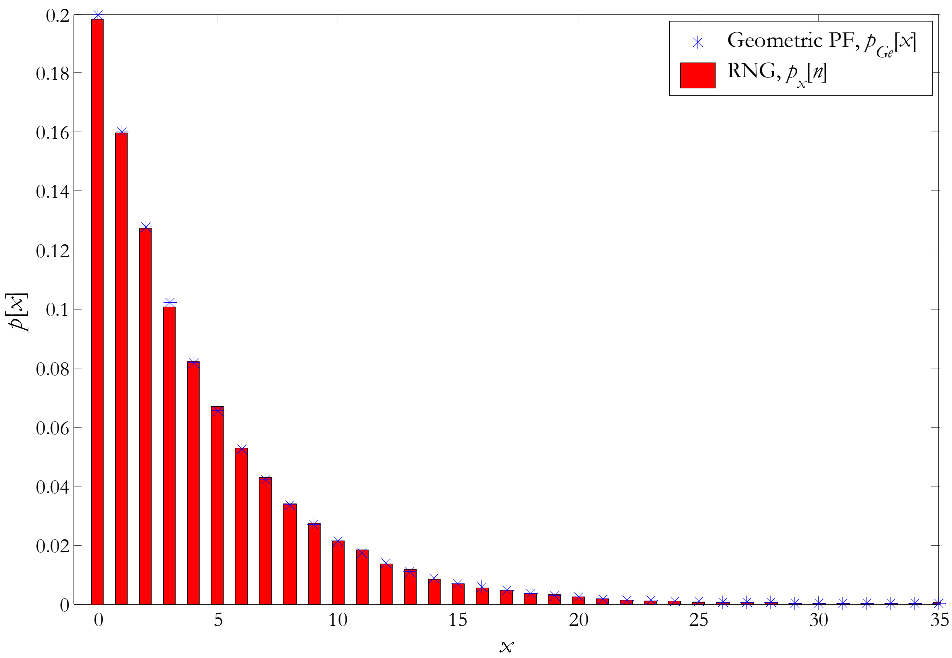

Figure A1.5. Comparison between the Geometric PF and the PF equivalent modified histogram obtained from the Geometrical RNG (P_{PC}=0.2).

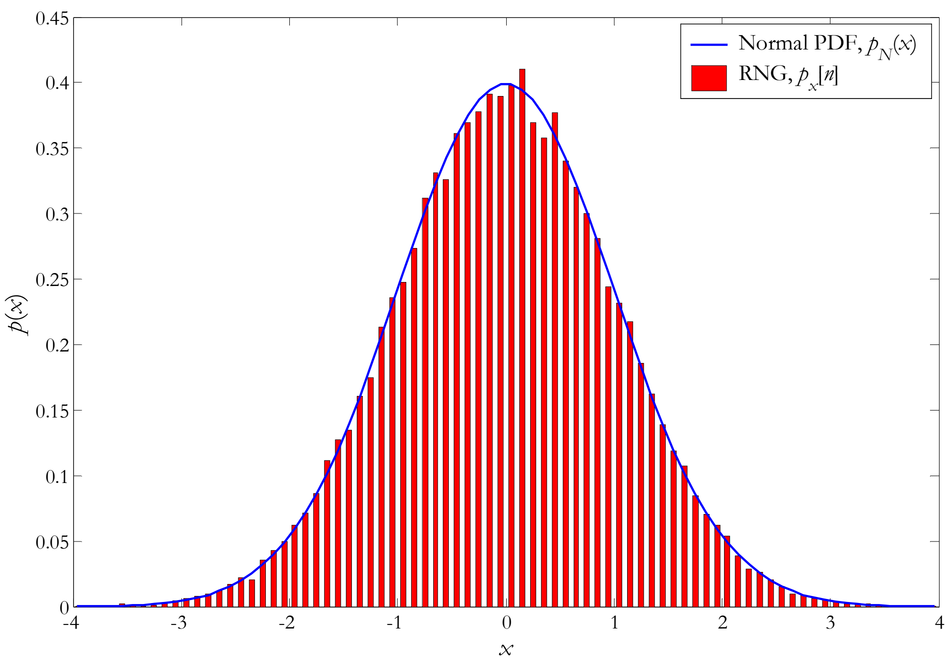

Figure A1.6. Comparison between the Normal PF and the PF equivalent modified histogram obtained from the Normal RNG (μ_N=0, σ_N=1).

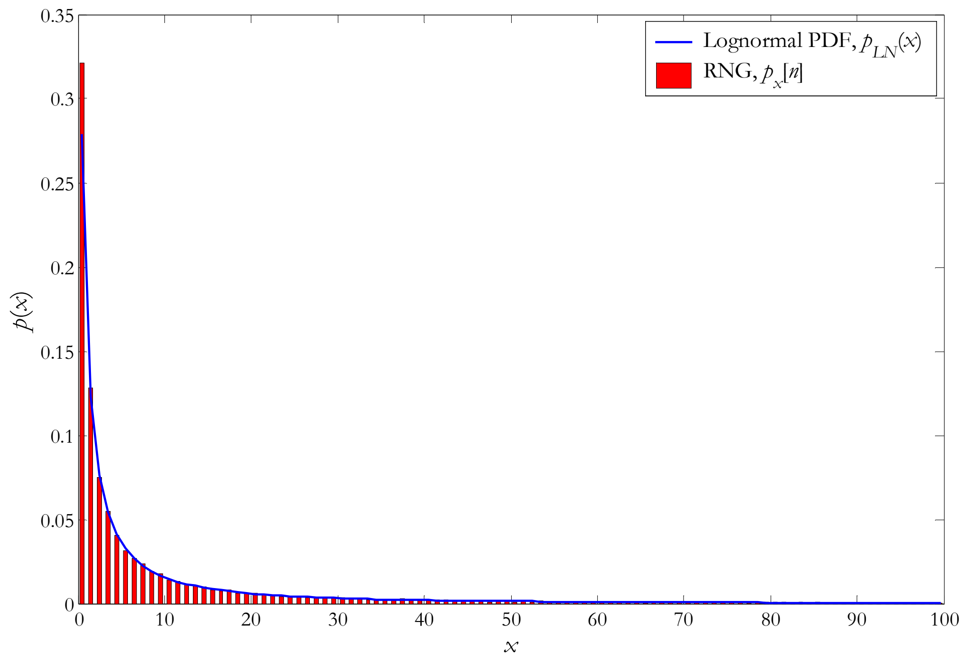

Figure A1.7. Comparison between the Lognormal PDF and the PDF equivalent modified histogram obtained from the Lognormal RNG (μ_{LN}=0.97, σ_{LN}^2=4.38).

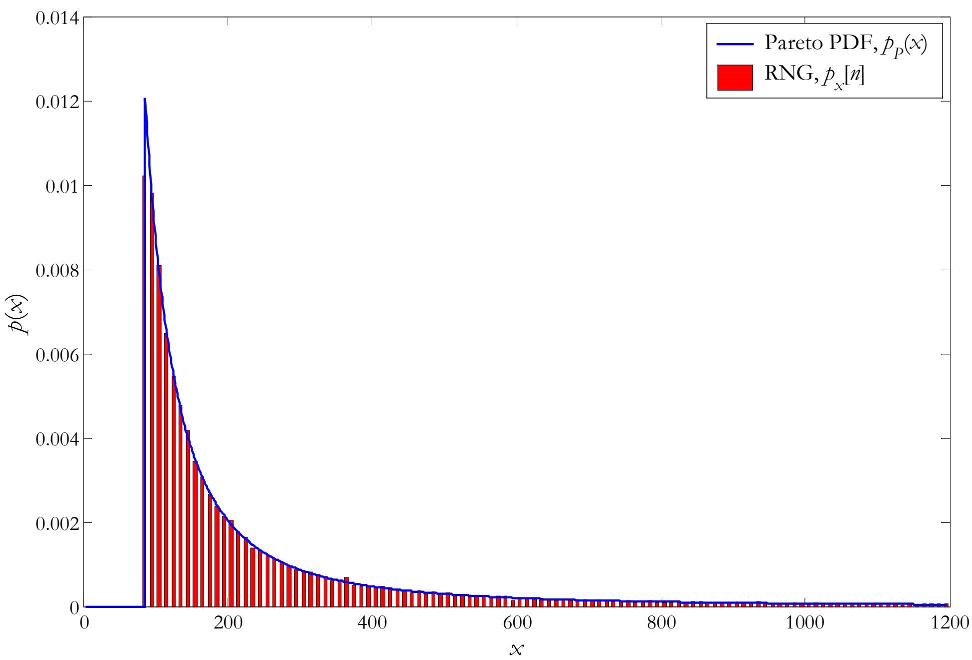

Figure A1.8. Comparison between the Pareto PDF and the PDF equivalent modified histogram obtained from the Pareto RNG (α_p=1.1, k=81.5).

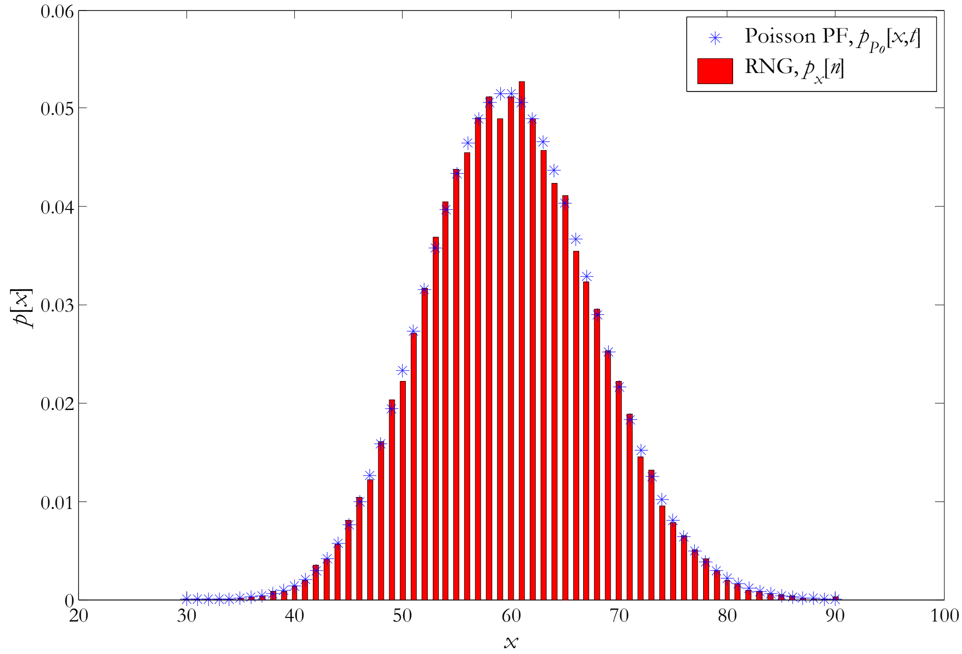

Figure A1.9. Comparison between the Poisson PF and the PF equivalent modified histogram obtained from the Poisson RNG (λ_c=60).

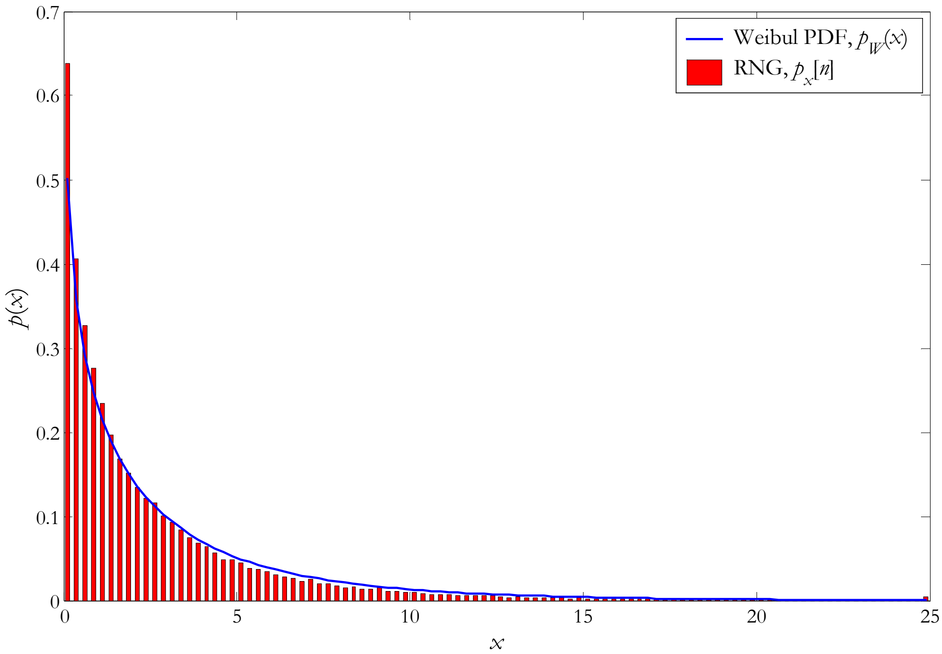

Figure A1.10. Comparison between the Weibul PDF and the PDF equivalent modified histogram obtained from the Weibul RNG (λ_k=0.45, β_k=0.80).

Annex 2

User Scenarios Variation Results

This annex presents results for the main performance indicators (BR, Discarded Packets/Frames and Mean Bitrate) as a function of the User Scenarios (*i.e.*, of the number of users), for five different Access Technology Scenarios (G, U, H, G+H and U+H). These results are a complement to those presented in Section 5.3 and were also obtained from single simulation runs.

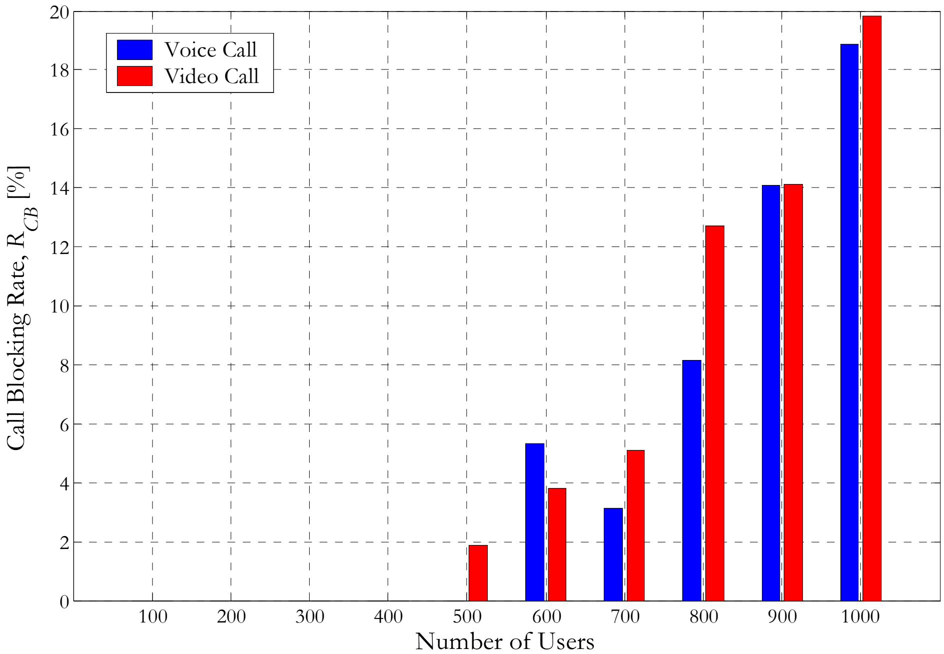

Figure A2.1. BR versus User Scenario (G).

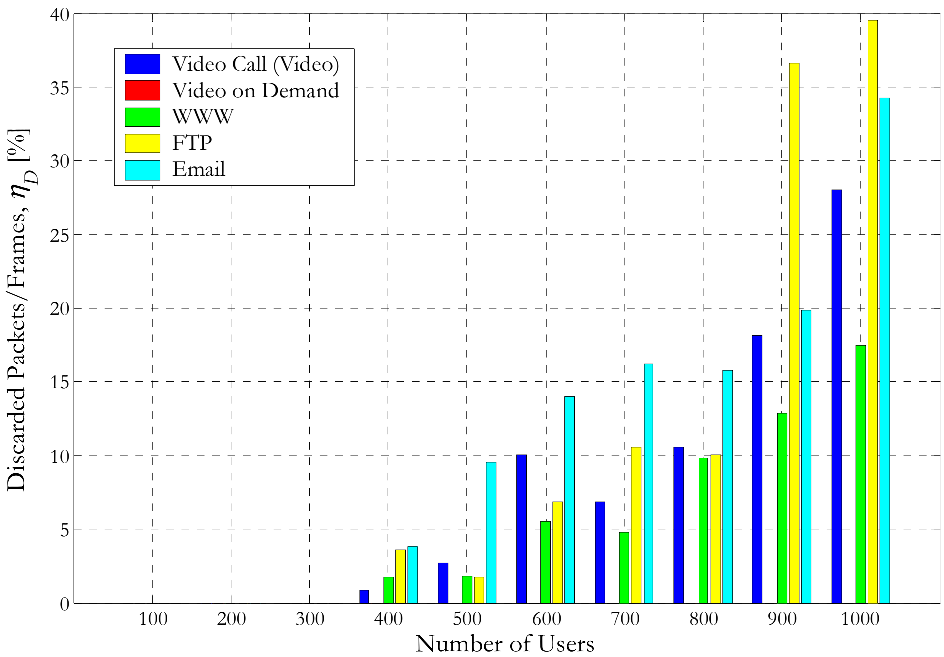

Figure A2.2. Discarded Packets/Frames versus User Scenario (G).

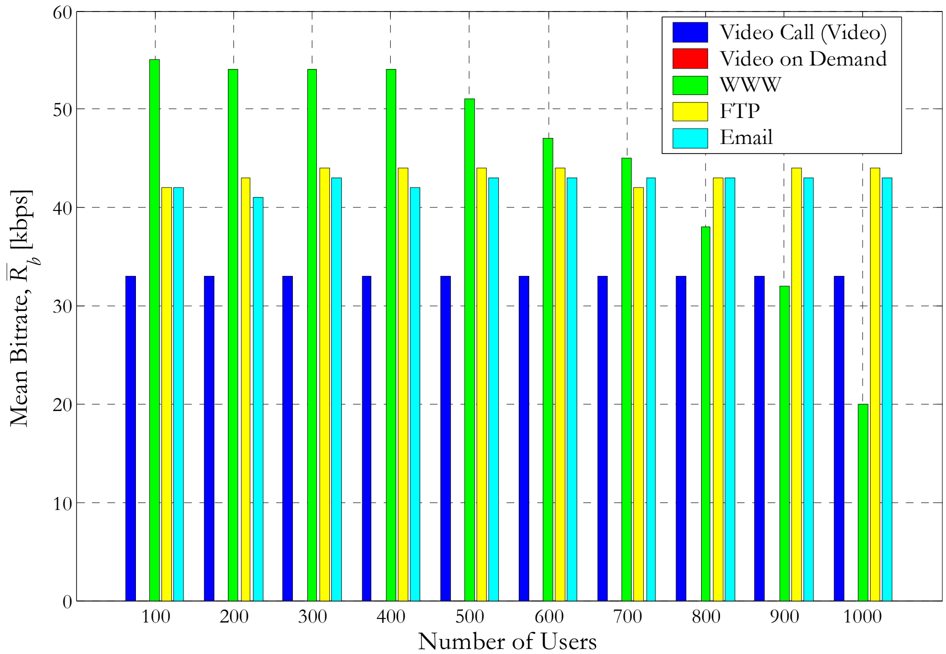

Figure A2.3. Mean Bitrate versus User Scenario (G).

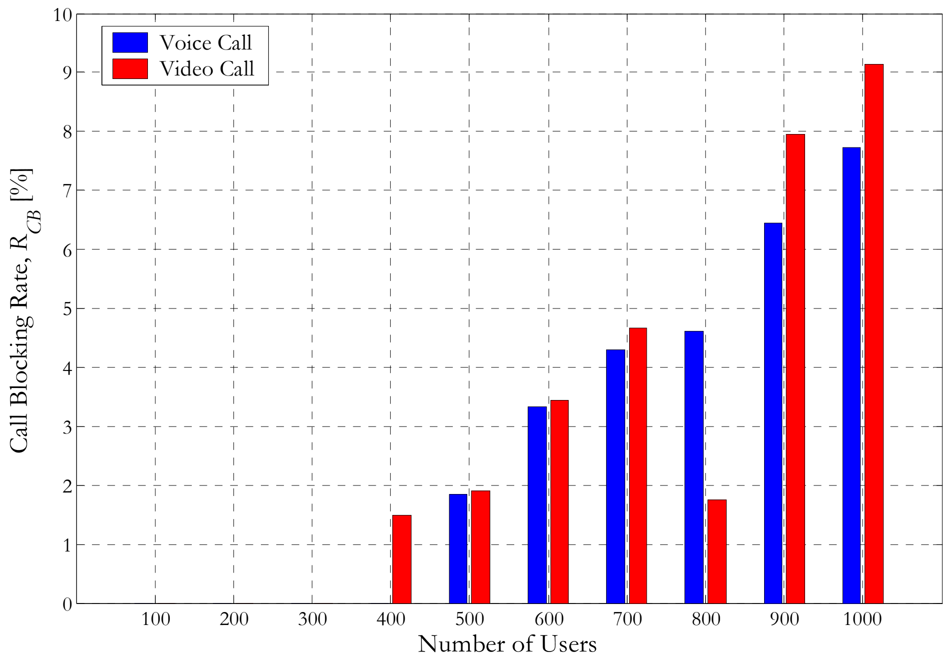

Figure A2.4. BR versus User Scenario (U).

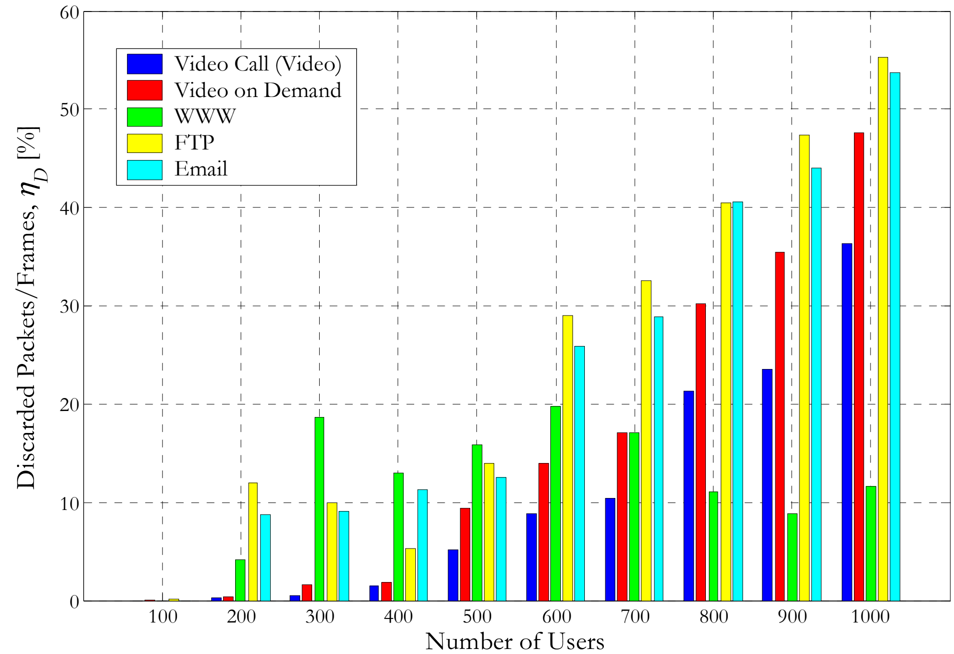

Figure A2.5. Discarded Packets/Frames versus User Scenario (U).

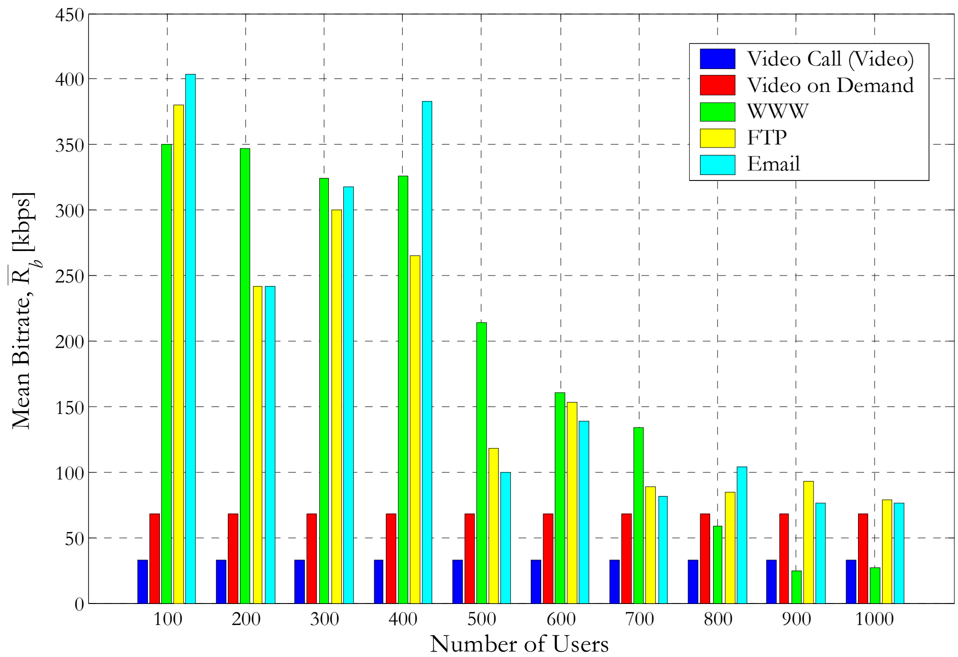

Figure A2.6. Mean Bitrate versus User Scenario (U).

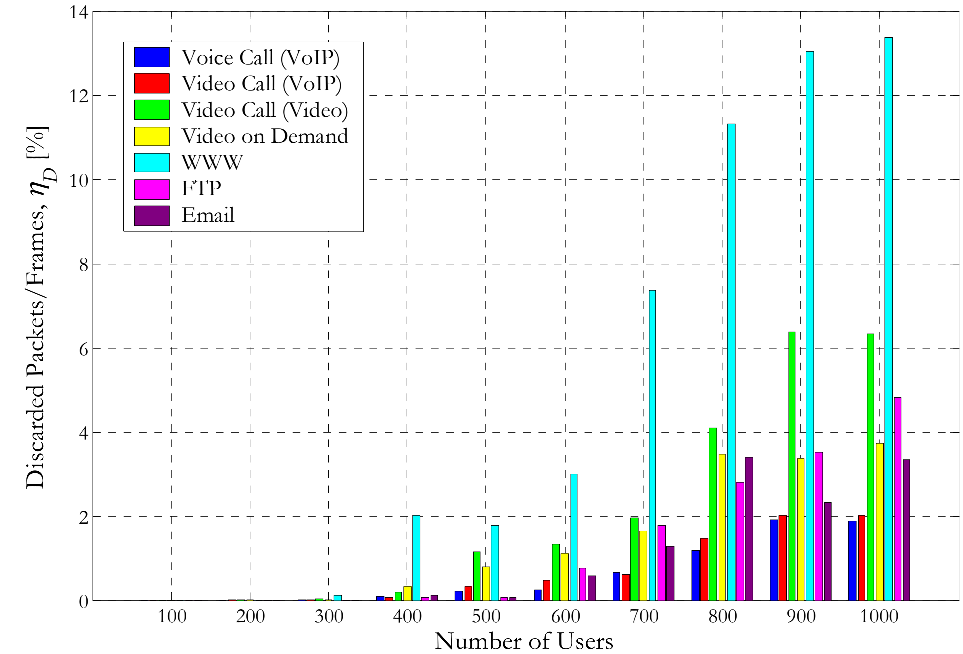

Figure A2.7. Discarded Packets/Frames versus User Scenario (H).

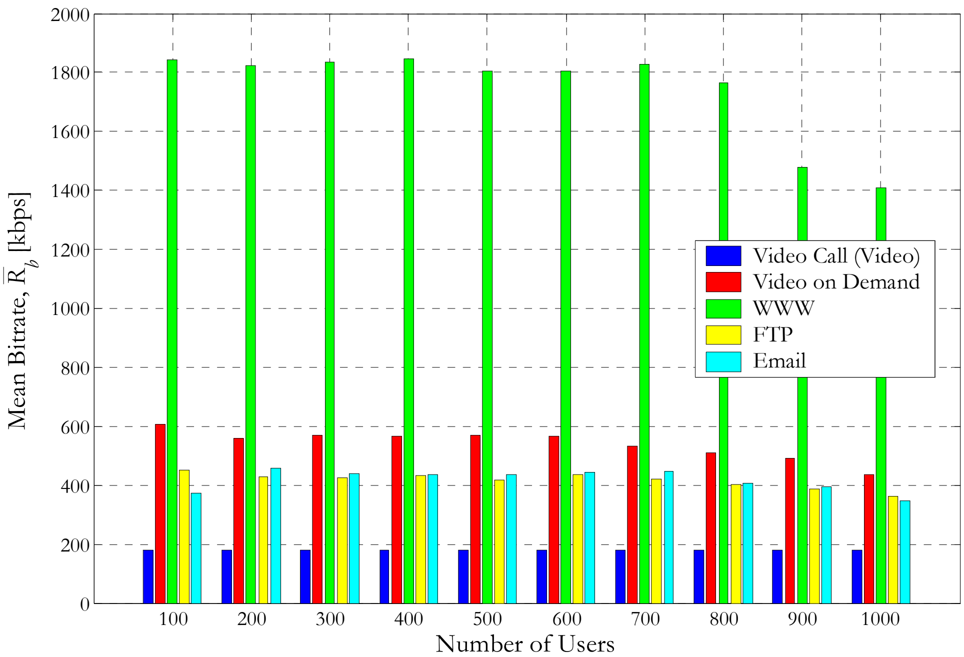

Figure A2.8. Mean Bitrate versus User Scenario (H).

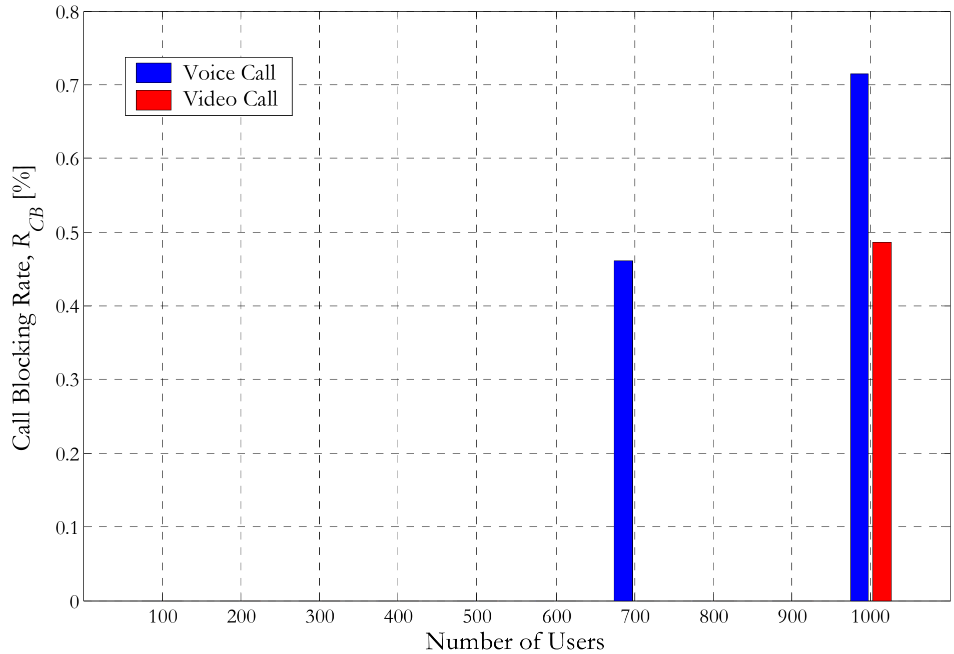

Figure A2.9. BR versus User Scenario (G+H).

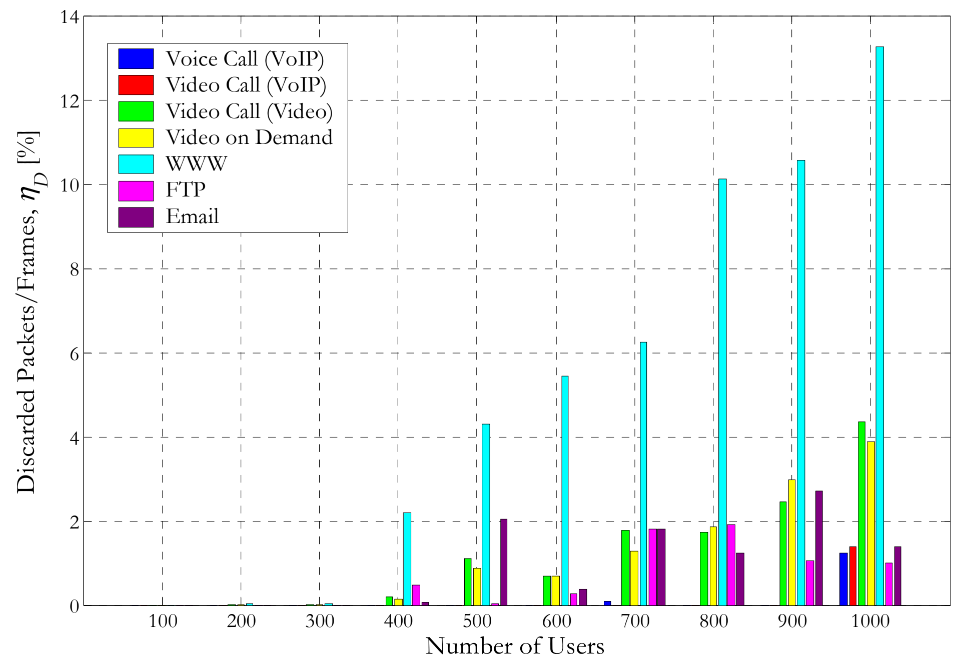

Figure A2.10. Discarded Packets/Frames versus User Scenario (G+H).

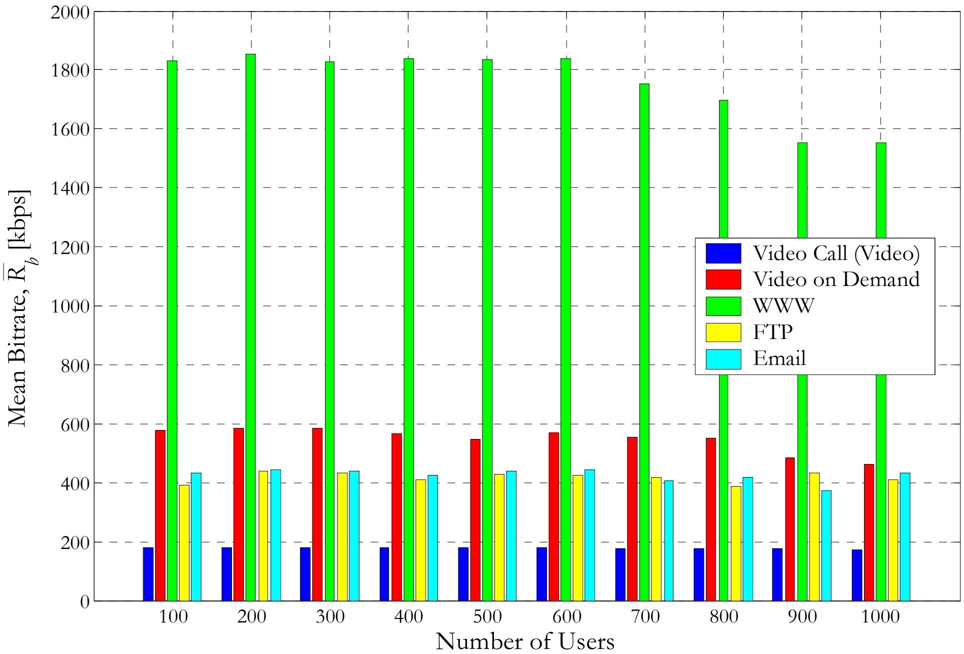

Figure A2.11. Mean Bitrate versus User Scenario (G+H).

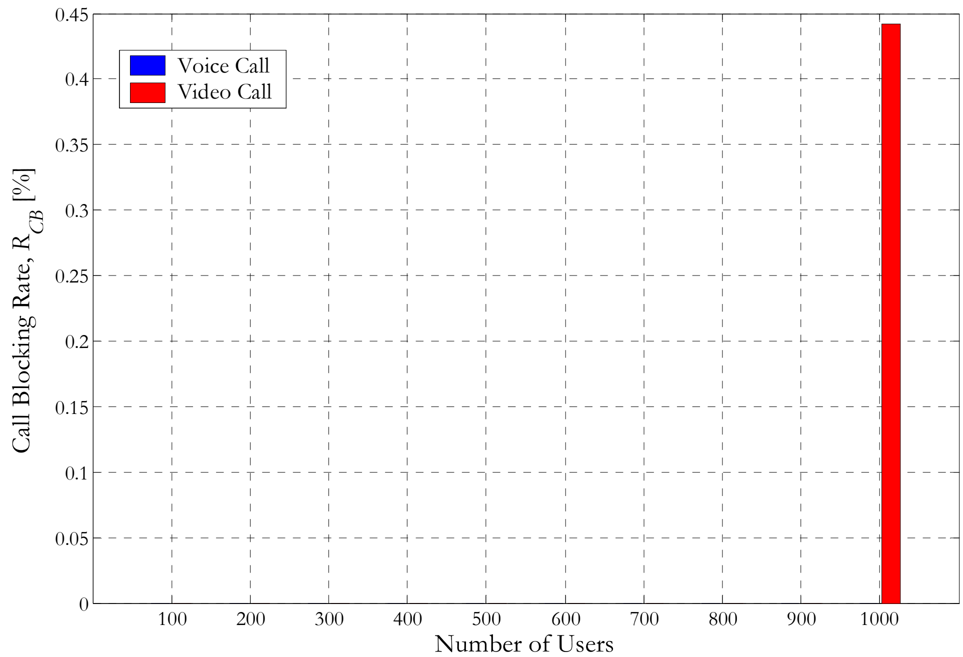

Figure A2.12. BR versus User Scenario (U+H).

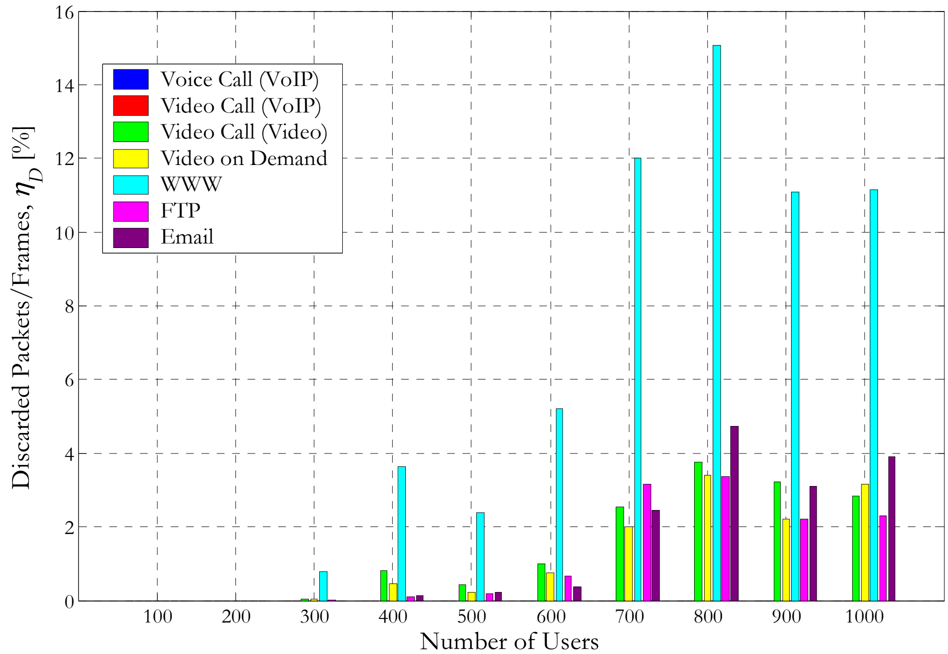

Figure A2.13. Discarded Packets/Frames versus User Scenario (U+H).

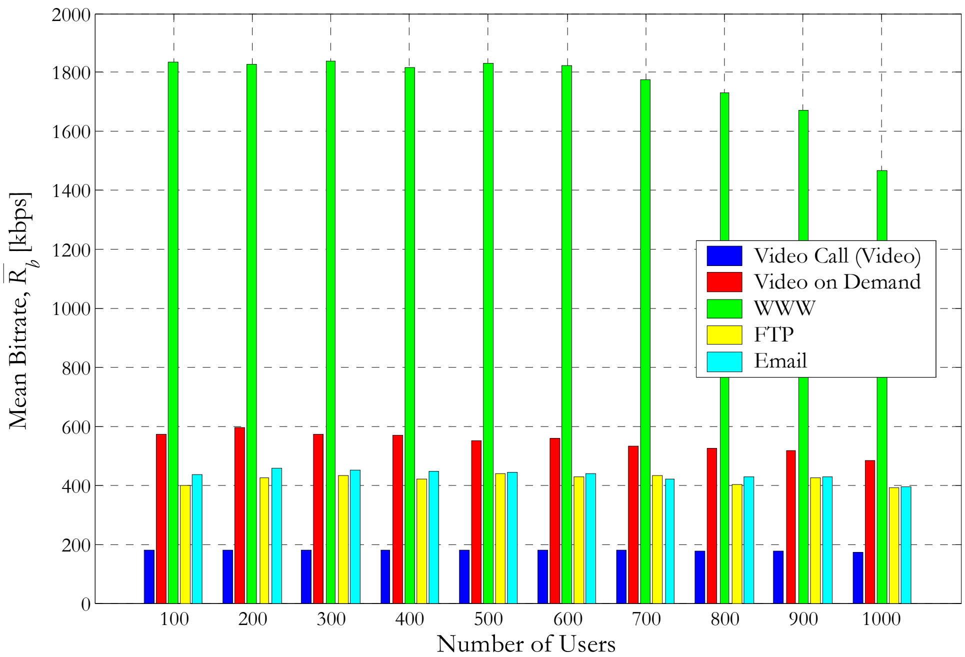

Figure A2.14. Mean Bitrate versus User Scenario (U+H).

Annex 3

Access Technology Scenarios Variation Results

This annex presents results for the main performance indicators (BR, Discarded Packets/Frames and Mean Bitrate) as a function of the Access Technology Scenarios (*i.e.*, of the available access technologies), for five different User Scenarios (500, 600, 800, 900 and 1 000 users). These results are a complement to those presented in Section 5.4 and were also obtained from single simulation runs. For each User Scenario, the results are obtained maintaining the same Applications Scenario instance (*i.e.*, the same UTVs).

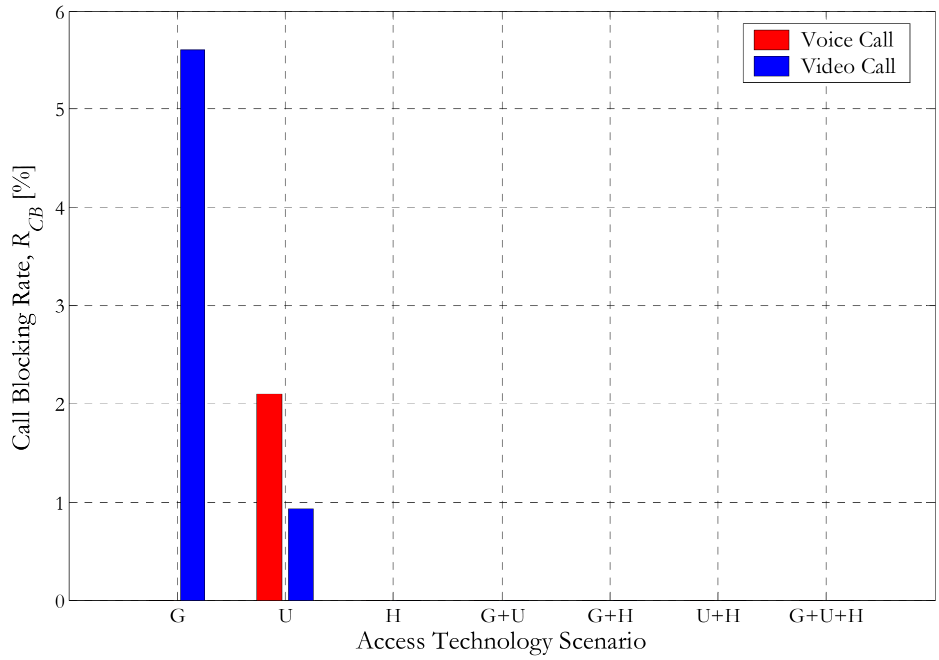

Figure A3.1. BR versus Access Technology Scenario (500 users).

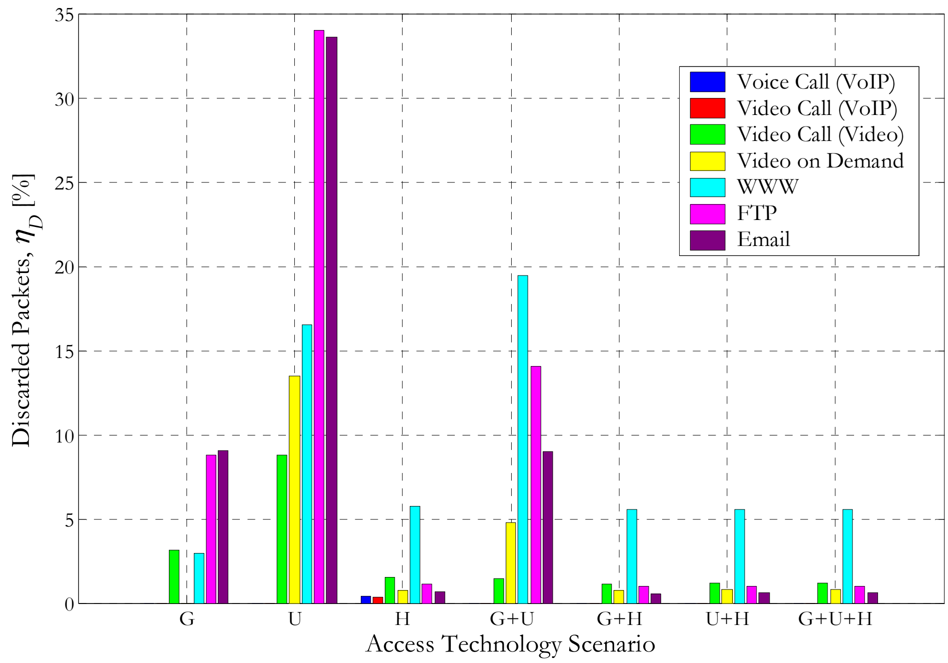

Figure A3.2. Discarded Packets/Frames versus Access Technology Scenario (500 users).

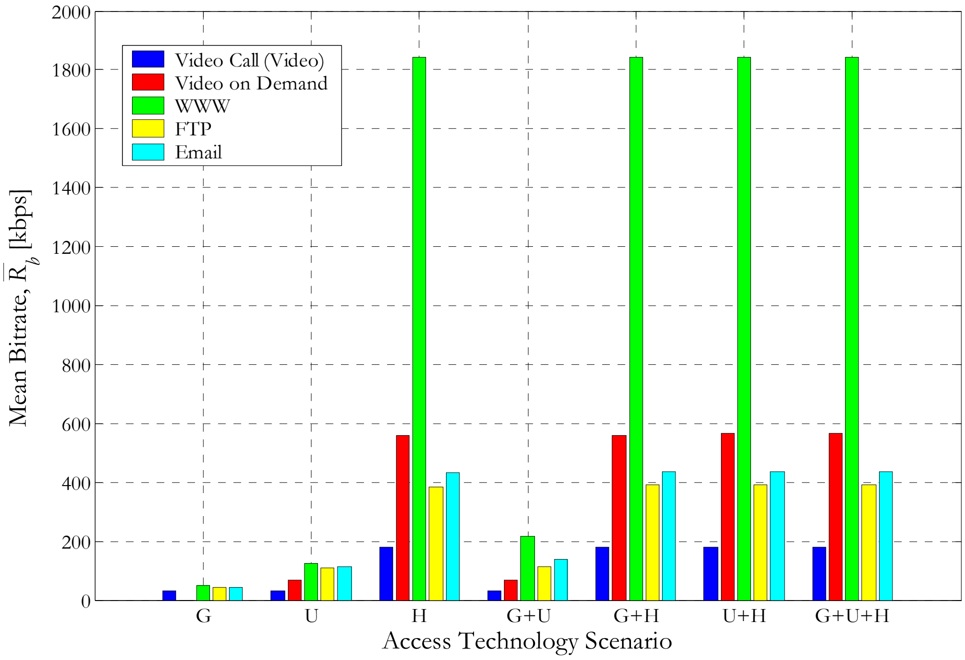

Figure A3.3. Mean Bitrate versus Access Technology Scenario (500 users).

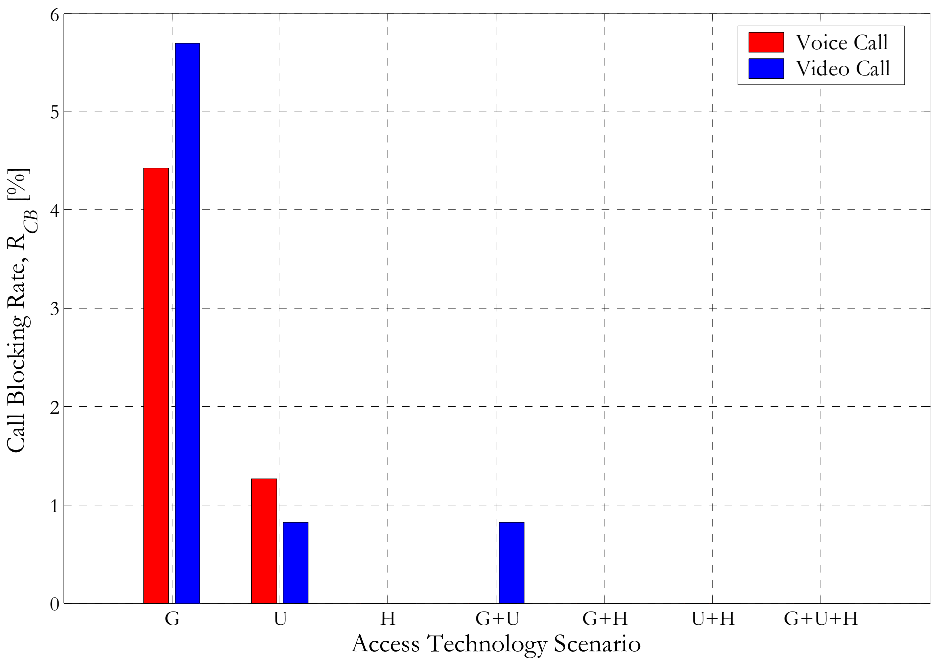

Figure A3.4. BR versus Access Technology Scenario (600 users).

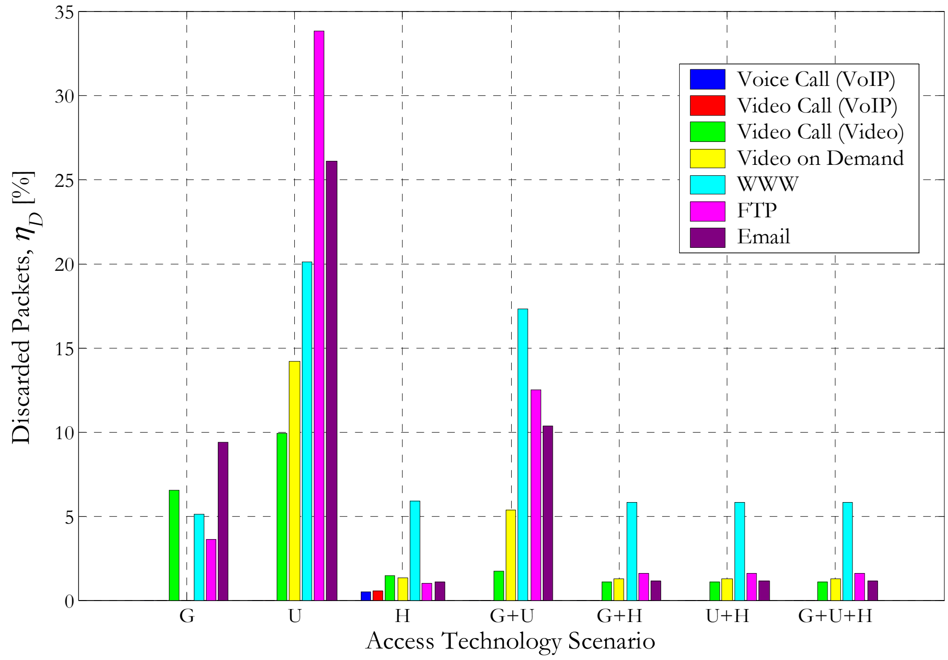

Figure A3.5. Discarded Packets/Frames versus Access Technology Scenario (600 users).

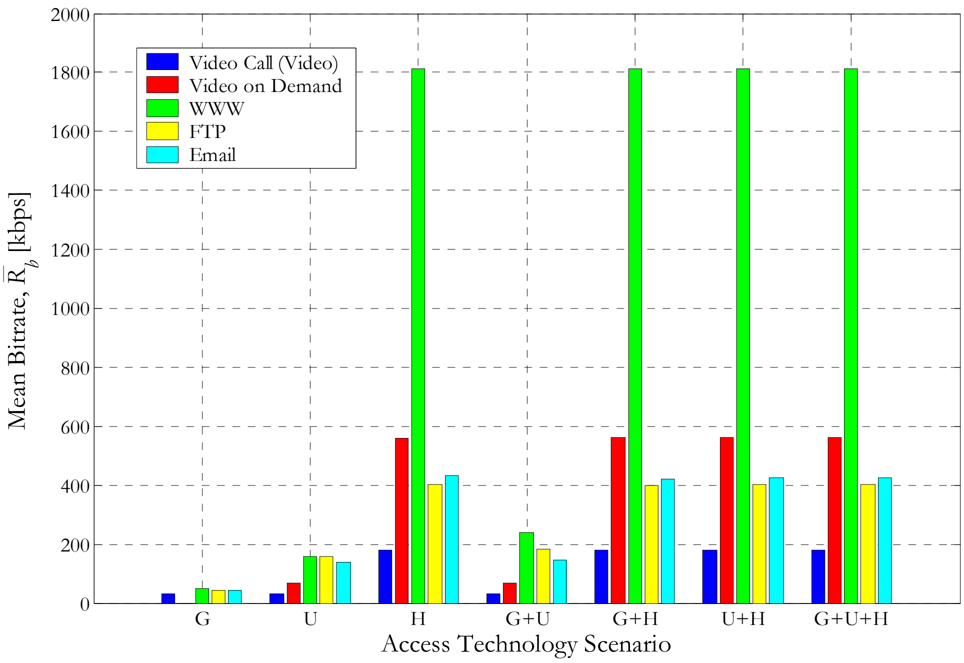

Figure A3.6. Mean Bitrate versus Access Technology Scenario (600 users).

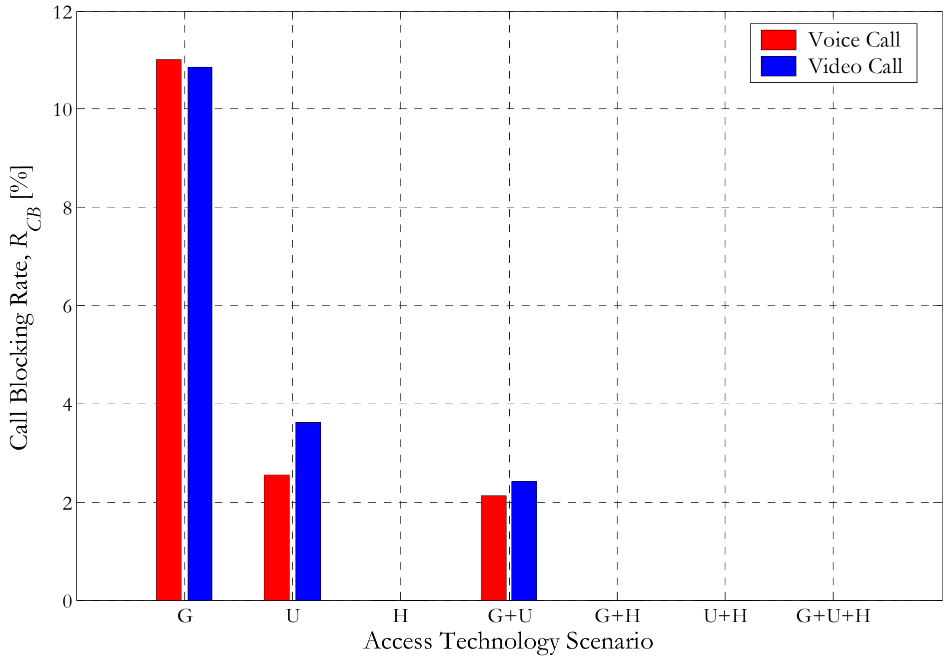

Figure A3.7. BR versus Access Technology Scenario (800 users).

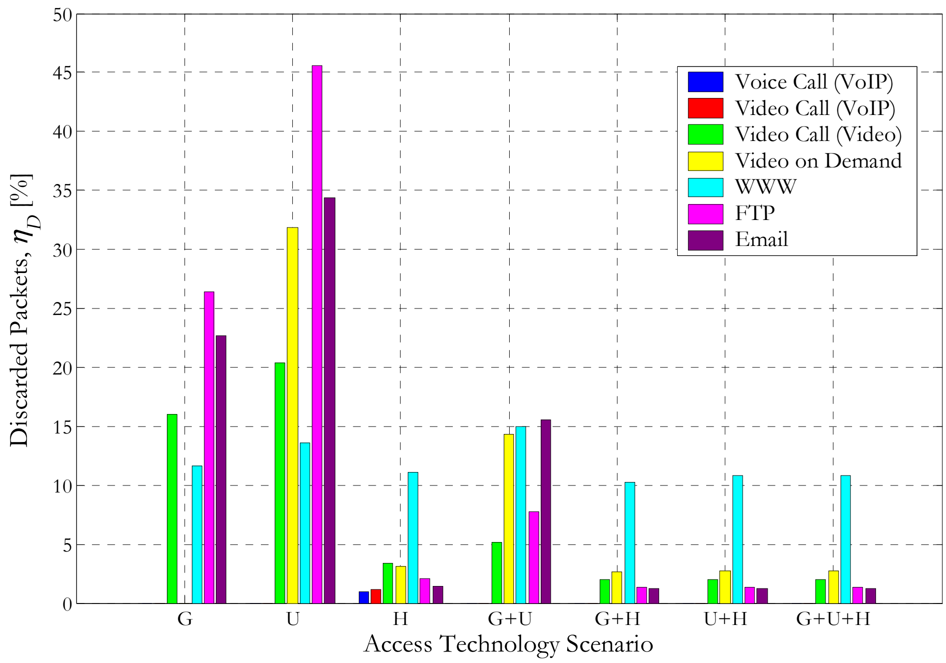

Figure A3.8. Discarded Packets/Frames versus Access Technology Scenario (800 users).

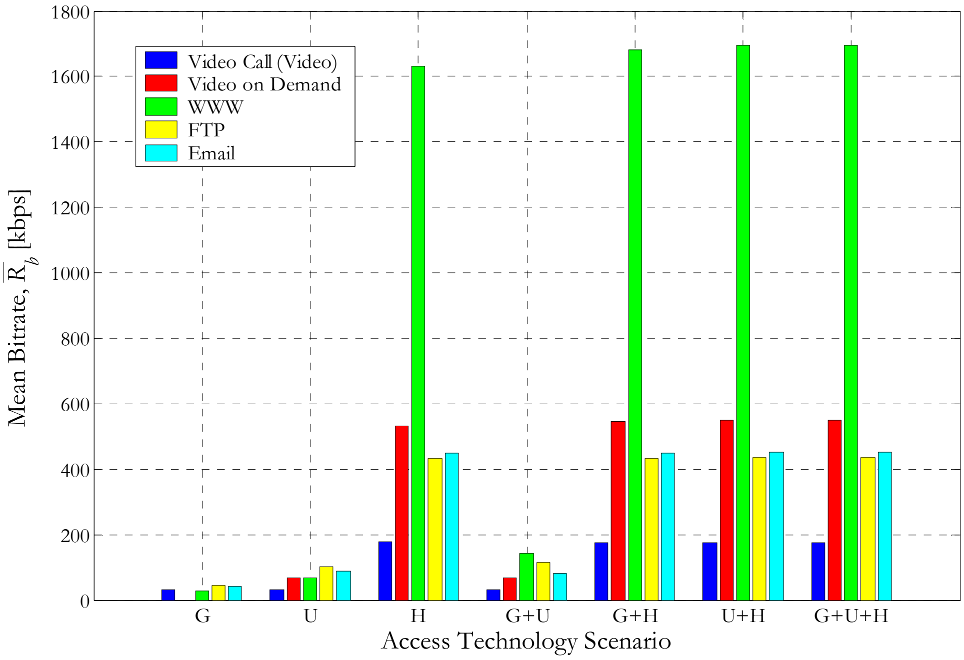

Figure A3.9. Mean Bitrate versus Access Technology Scenario (800 users).

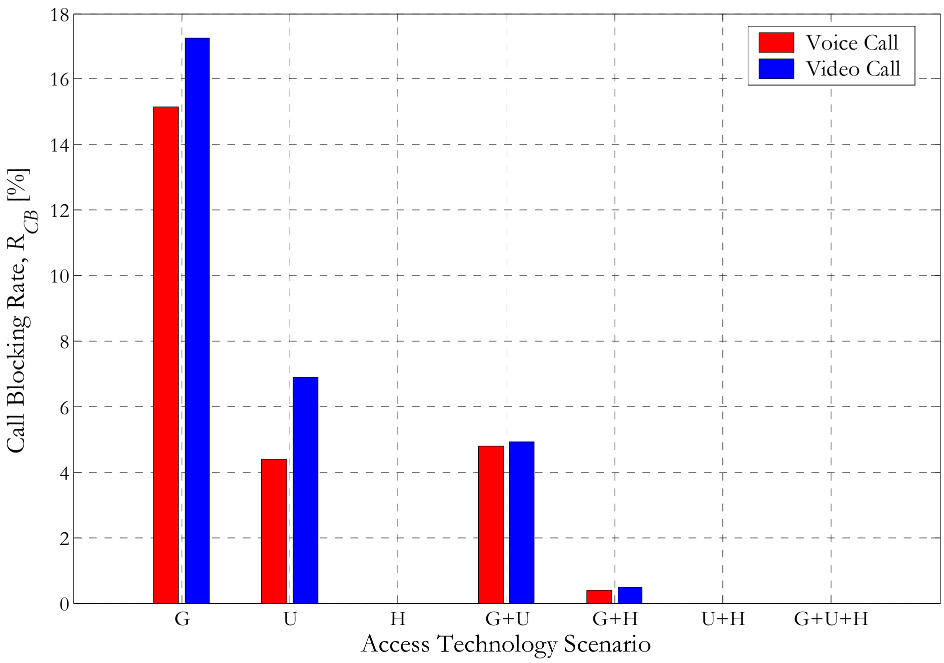

Figure A3.10. BR versus Access Technology Scenario (900 users).

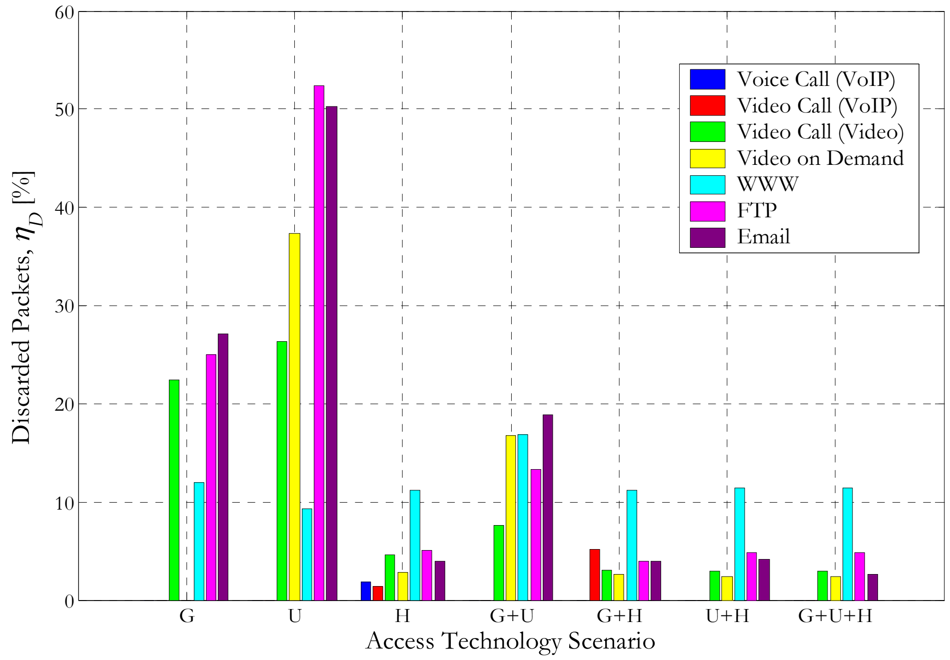

Figure A3.11. Discarded Packets/Frames versus Access Technology Scenario (900 users).

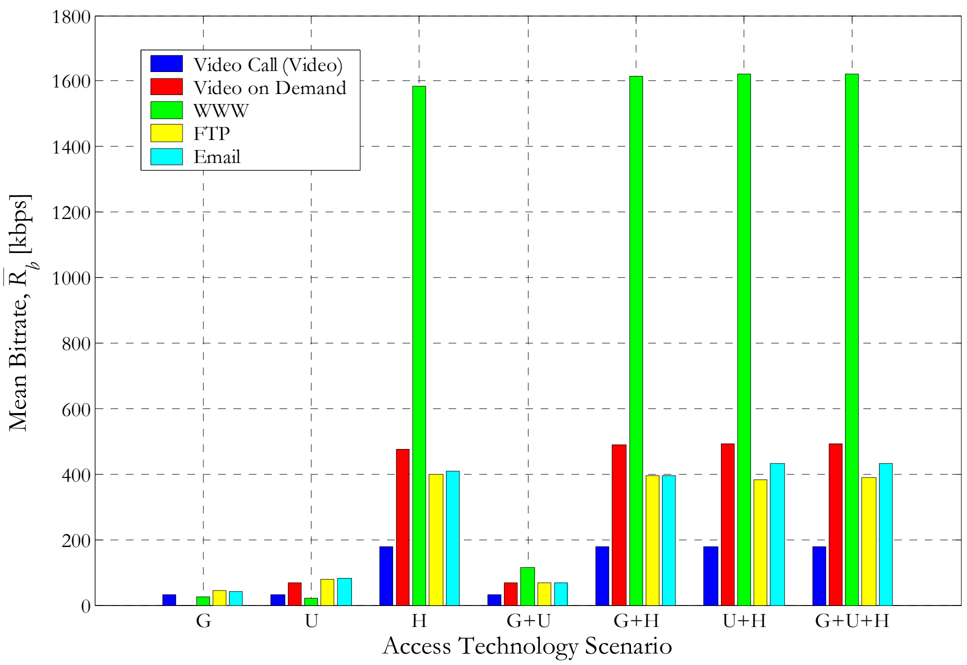

Figure A3.12. Mean Bitrate versus Access Technology Scenario (900 users).

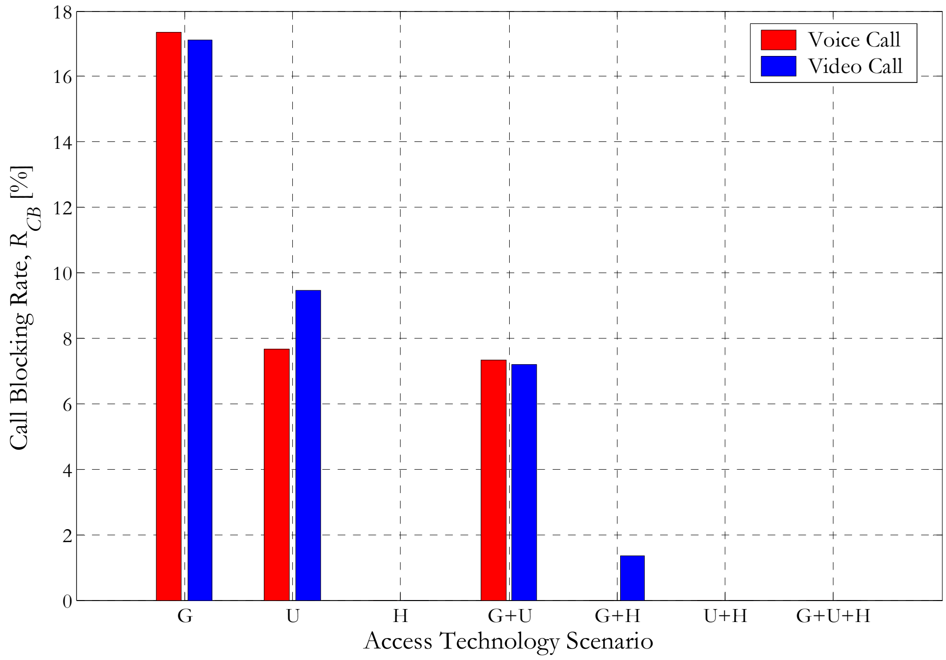

Figure A3.13. BR versus Access Technology Scenario (1 000 users).

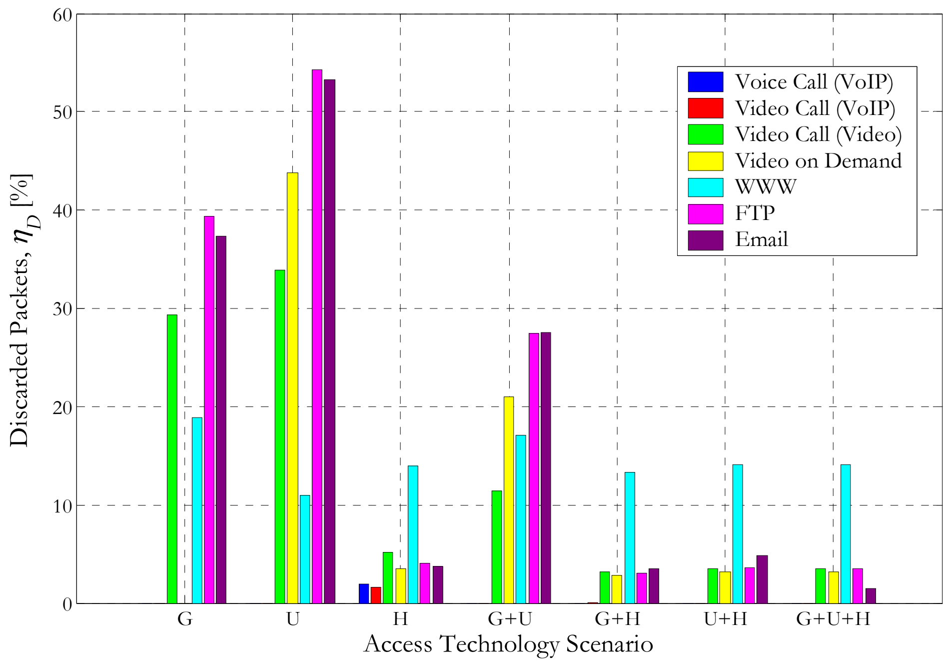

Figure A3.14. Discarded Packets/Frames versus Access Technology Scenario (1 000 users).

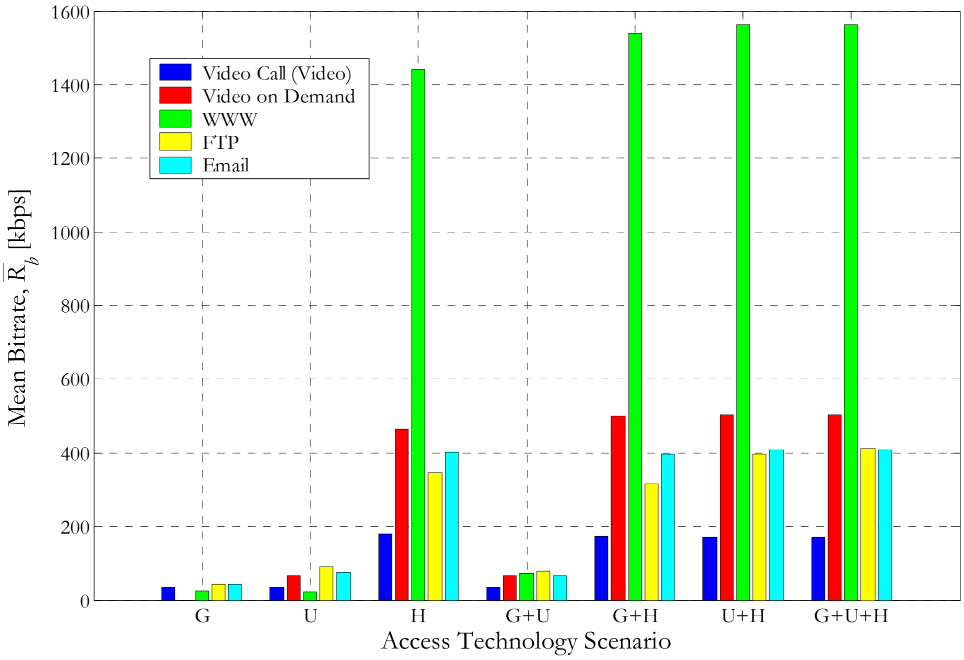

Figure A3.15. Mean Bitrate versus Access Technology Scenario (1 000 users).

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