Universidade de Aveiro Departamento de Electrónica, Telecomunicações e Informática
2007

Miguel Fernando Dias de Almeida

Soluções de Broadcast para redes 4G

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Soluções de Broadcast para redes 4G

Dissertação apresentada à Universidade de Aveiro para cumprimento dos requisitos necessários à obtenção do grau de Mestre em Engenharia Electrónica e Telecomunicações, realizada sob a orientação científica da Prof. Dra. Susana Sargento, Professora auxiliar convidada do Departamento de Electrónica, Telecomunicações e Informática da Universidade de Aveiro.

o júri

presidente

Prof. Dr. José Carlos Neves
Professor Catedrático do Departamento de Electrónica, Telecomunicações e Informática da Universidade de Aveiro

Prof. Dr. Susana Sargento
Professora Auxiliar convidada do Departamento de Electrónica, Telecomunicações e Informática da Universidade de Aveiro

Prof. Dr. Manuel Alberto Pereira Ricardo
Professor Associado do Departamento de Engenharia Electrónica e de Computadores da Faculdade de Engenharia da Universidade do Porto

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Se a amizade é um processo evolutivo e a aprendizagem também assim o é, então não poderia estar mais grato por ambas terem andado de mãos dadas no decurso deste trabalho.
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palavras-chave Broadcast Technologies, Mobilidade, QoS, DVB, 4G

resumo

A primeira difusão de conteúdos video e audio teve um forte impacto no quotidiano da população que assistiu a uma revolução nos modelos de transmissão de informação e de entretenimento. A evolução desde então foi significativa, e já na era digital, encontramo-nos face a uma nova sub-elevação da metodologia e do conceito subjacentes à transmissão de conteudos multimédia. O mundo actual apresenta, contudo, diferentes requisitos, de entre os quais se destacam a procura pela alta definição e mobilidade. A mobilidade tem sido um particular foco de atenção por parte dos operadores que exploram agora modelos para entregar uma vasta gama de serviços que sejam atractivos para os utilizadores.

Esta dissertação apresenta um sumário das tecnologias emergentes de broadcast que se distinguem nas várias partes do mundo com a sua particular incidência geográfica, características e cenários de aplicação.

É ainda apresentada uma arquitectura 4G abordando assuntos inerentes à mobilidade e qualidade de serviço com particular incidência nos aspectos relacionados com a integração de uma tecnologia de broadcast particular.

Para avaliação da arquitectura proposta foram efectuados estudos com base num equipamento de broadcast na sua versão comercial, permitindo desta forma obter uma análise que ilustra o que os operadores podem esperar do estado actual dos dispositivos. Os resultados permitiram retirar ilações sobre o comportamento de um equipamento considerado como um produto final a disponibilizar aos operadores, quando integrado num ambiente 4G com suporte de mobilidade e QoS. Nomeadamente é discutida a sua aplicabildiade tendo em linha de conta as desvantagens introduzidas pelas características inerentes à própria tecnologia.

keywords **Broadcast, Mobilidade, QoS, DVB, 4G**

abstract

Broadcast of video and audio through analogical television completely changed the paradigm of information and entertainment divulgation. Today, in the *"digital era",* the Analogue Switch Off revolution is being held. Manufacturers and operators already show concerns regarding the support of mobility, quality of experience and of service. Delivering competitive High Definition contents and providing solutions for the average *"on-the-move"* user are two of the most important issues to be dealt by the service providers, which are also within the analysis scope of this work.
This dissertation presents an overview on the most relevant broadcast technologies which are assumed to be of relative acceptance in their respective target market. It presents their main characteristics and applicability.

4G architectural concepts are also analyzed, closely dealing with mobility and quality of service provisioning, with particular focus on the seamless integration of broadcast technologies.

As a mean to evaluate the feasibility of integrating broadcast technologies with 4G architectures, a performance evaluation study was performed using commercial equipment. In this way a several set of considerations constructed illustrating the features and functionalities which operators can expect or disregard from professional commercial broadcasting devices. Results allow the withdrawing of conclusions concerning the integration of a final broadcasting solution when incorporated within a 4G environment with QoS and mobility support. Its applicability is evaluated having in mind the performance drawbacks introduced by the specific technology, and generalized towards the gathering of more general conclusions which consider the main characteristics of the commercial broadcasting devices.

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Acronyms

	Acronym	Description
	4G	Fourth Generation Networks
A	A4C	Authentication, Authorization, Auditing, Accounting and Charging
	AAA	Authentication, Authorization and Accounting
	ACM	Adaptive Coding and Modulation
	ADMR	Adaptive Demand-Driven Multicast Routing
	ADSL	Asymmetric Digital Subscriber Line
	ADT	Application Data Table
	AIMD	Additive Increase, Multiplicative Decrease
	AL-FEC	Application Layer FEC
	AN	Access Network
	AODV	Ad-hoc On-demand Distance Vector
	AP	Access Point
	APSK	Amplitude and Phase Shift Keying
	AR	Access Router
	ARP	Address Resolution Protocol
	ASM	Adaptive MIMO Switching
	ASO	Analogue Switch Off
	ATM	Asynchronous Transfer Mode
B	BER	Bit Error Rate
	BM-SC	Centre
	BPL	Broadband Over PowerLines
	BS	Base Station
	BWS	Broadcast WebSite
C	CARD	Candidate Access Router Discovery
	CBR	Constant Bit Rate
	CDMA	Code Division Multiple Access
	CID	Connection ID
	CIP	Cellular IP
	CIR	Committed Information Rate
	CN	Correspondent Node
	CoA	Care of Address
	COFDM	Coded OFDM
	CQoSB	Core QoS Broker
	CRC	Cyclic Redundancy Check
D	DAB	Digital Audio Broadcast
	DAD	Duplicate Address Detection
	DHCP	Dynamic Host Configuration Protocol
	Diffserv	Differentiated Service
	DL	Downlink
	DMB	Digital Multimedia Broadcast
	DMD	Digital Music Download

	DQPSK	Differential QPSK
	DSAP	Destiny Service Access Point
	DSR	Dynamic Source Routing
	DVB	Digital Video Broadcast
	DVB-C	DVB transmission via Cable
	DVB-H	DVB transmission to Handhelds
	DVBRC	DVB Return Channel
	DVB-S	DVB transmission via Satellite
	DVB-T	DVB transmission via terrestrial
	DVRG	Digital Video Recorder / Generator
E	eDAB	Enhanced DAB
	EDGE	Enhanced Data rates for GSM Evolution
	EIR	Excess Information Rate
	EMC	Electromagnetic Compatibility
	EPG	Electronic Program Guide
	erTPS	Extended Real Time Polling Service
	ES	Elementary Stream
	ESG	Electronic Service Guide
	ETSI	European Telecommunications Standard Institute
	FDD	Frequency Division Duplex
F	FEC	Forward Error Correction
	FER	Frame Error Rate
	FFT	Fast Fourier Transform
	FGNGN	Focus Group NGN
	FHO	Fast HandOvers
	FIC	Fast Information Channel
	FLO	Forward Link Only
	FMIP	Fast Mobile IP
G	GERAN	GSM/EDGE Radio Access Network
	GGSN	Gateway GPRS Support Node
	GIST	General Internet Signalling Transport
	GMP	Global Mobility Protocol
	GPS	Global Positioning System
	GSM	GSM/EDGE Radio Access Network
	GUI	Graphical User Interface
H	HA	Home Agent
	HDTV	High definition TV
	HIP	Host Identity Protocol
	HMIP	Hierarchical Mobile IP
	HoA	Home Address
	HSDPA	High-Speed Downlink Packet Access
I	IANA	Internet Assigned Numbers Authority
	IEEE	Institute of Electrical and Electronics Engineers
	IETF	Internet Engineering Task Force
	IIS	Intelligent Interface Selection
	IMS	IP Multimedia Subsystem

	INSIGNIA	In-band signaling support for QoS in Mobile Ad hoc Networks
	INT	MAC Notification Table
	IntServ	Integrated Services
	IP	Internet Protocol
	ISDB	Integrated Services Digital Broadcasting
	ITU-T	International Teleccomunication Union – Telecommunication Standardization Sector
L	LAN	Local Area Network
	LLC	Logical Link Control
	LLH	Low Latency Handoffs
	LMA	Local Mobility Anchor
	LMD	Local Mobility Domain
	LMP	Local Mobility Protocol
	LOS	Line of sight
M	MAG	Mobility Access gateway
	MANET	Mobile Ad-Hoc Networks
	MAODV	Multicast Ad-hoc On-demand Distance Vector
	MAP	Mobile Anchor Point
	MARQS	Mobility Management, AAA, Resource Management, QoS and Security
	MBMS	Multimedia Broadcast Multicast Service
	MBS	Multicast Broadcast System
	MICS	Media Independent Command Service
	MIES	Media Independent Event Service
	MIH	Media Independent Handovers
	MIHF	Media Independent Handovers Function
	MIHO	Mobile Terminal Initiated HandOvers
	MIIS	Media Independent Information Service
	MIMO	Multiple Input – Multiple Output
	MLC	Multimedia Logic Channel
	MM	Mobility Manager
	MMORPG	Massively Multiplayer Online Role-Playing Game
	MOLSR	Massively Multiplayer Online Role-Playing Game
	MPE	MultiProtocol Encapsulation
	MPEG-2 TS	MPEG-2 Transport Stream
	MPR	Multi Point Relay
	MRFP	Media Resource Function Processor
	MS	Mobile Station
	MSC	Main Service Channel
	MT	Mobile Terminal
N	NAI	Network Address Identifier
	NDP	Neighbour Discovery Protocol
	NetLMM	Network-based Localized Mobility Management
	NGN	Next Generation Networks
	NIHO	Network Initiated HandOvers
	NLOS	Non Line Of Sight

	NPA	Network point of attachment
	nrtPS	Non real time Polling service
	NSIS	Next Steps In Signalling
	NSLP	NSIS Signalling Layer Protocol
	NTLP	NSIS Transport Layer Protocol
	nwIIS	Network intelligent Interface Selection
O	ODMRP	On-Demand Multicast Routing Protocol
	OFDM	Orthogonal Frequency-Division Multiplexing
	OLSR	Optimized Link State Routing protocol
	OSI	Open Systems Interconnection
P	PACP	Polynomial-assisted Ad-hoc Charging Protocol
	PBNMS	Policy-based Network Management System
	PDUs	Protocol Data Units
	PID	Program Identifier
	PMA	Proxy Mobile Agent
	PMP	Point to MultiPoint
	PoA	Point of Attachment
	PP	Point to Point
	PSI	Program Specific Information
	PSK	Phase Shift Keying
Q	QAM	Quadrature Amplitude Modulation
	QOLSR	Quality of Service for OLSR
	QoS	Quality of Service
	QoSC	QoS Client
	QoSM	QoS Manager
	QPSK	Quadrature Phase Shift Keying
R	RAL	Radio Access Layer
	RS	Reed Solomon
	RSVP	ReSource reserVation Protocol
	RTT	Round Trip time
S	SAP	Service Access Point
	SCP	Secure Charging Protocol
	SDI	Serial Digital Interface
	S-DMB	Satellite Digital Multimedia Broadcast
	SFN	Single Frequency Network
	SI	Service Information
	SIB	Seamless Integration of Broadcast
	SIP	Session Initiation protocol
	SLS	Service Level Specifications
	SMS	Short Messaging System
	SNAP	SubNetwork Access Protocol
	SNDU	SubNetwork Data Unit
	SNMP	Simple Network Management Protocol
	SNR	Signal to Noise Ratio
	SOFDMA	Scalable Orthogonal Frequency Division Multiple Access
	SPAN	Services and Protocols for Advanced Networks

	SS	Subscriber Station
	SSAP	Source Specific Access Point
	SWAN	Service Differentiation in Stateless Wireless Ad Hoc Networks
T	TDD	Time Division Duplex
	TDMA	Time Division Multiple Access
	T-DMB	Terrestrial Digital Multimedia Broadcast
	TIPHON	Telecommunications and Internet Protocol Harmonization Over Networks
	TISPAN	Telecoms & Internet converged Services & Protocols for Advanced Networks
	TM-SSP	Technical Module for delivering Satellite Services to Portable Device
	TPS	Transmission Parameters Signalling
U	UE	User Equipment
	UGS	Unsolicited Grant Service
	UHF	Ultra High Frequency
	UL	Uplink
	ULE	Ultra Lightweight Encapsulation
	UMTS	Universal Mobile Telecommunications System
	USP	Ubiquitous and Seamless Pervasiveness
	UTRAN	UMTS Terrestrial Radio Access Network
V	VBR	Variable Bit Rate
	VCoA	Virtual Care of Address
	VHF	Very High Frequency
	VID	Virtual Identity
W	WiBro	Wireless Broadband
	WiMAX	Worldwide Interoperability for Microwave Access
Z	ZQoSB	Zone QoS Broker

Chapter 1 INTRODUCTION

1.1 Broadcasting – The Advent

The dawn of digital broadcast technologies has reached the market leaving behind the analogical broadcast transmissions first started in 1936. The evolution's first steps were taken with the creation of the European Broadcasting Union in 1950 which lead to the formation of the DVB Project in 1993, a consortium which is responsible for the creation of technology standards for the provisioning of digital television. Other standardization organizations and consortiums have also put efforts into presenting other similar solutions. With the new directives from the European parliament, and the Geneva Frequency plan which was held in 2006, all European broadcasters should shift their equipments from analogical to digital by the end of 2015. Some operators have already anticipated themselves, as shown in Table 1. Netherlands is ahead in this transition and has already finished the Analogue Switch Off (ASO) even before the initial estimation (2006-2008).

The success of mobile telecommunications has also brought an additional variable to broadcasters, which now study the feasibility of deploying the necessary infrastructure for the provisioning of digital mobile television. This concept results from the high success of the second generation of mobile communications, which lead to a high expansion of cellular terminals. The social usage of such devices has become a usual habit and people depend on them to access information, to communicate and to fight an urban psychological and social phenomenon – the urban solitude. The loneliness of each individual in a mobile tends to make him search for distractions. This is the reason which can lead to the sough of the services provided by digital mobile television, as long as it will be easily accessible.

	Country	DTT launch	ASO date	Estimated ASO
B Cable	Netherlands	2004	2006	2006 - 2008
	Germany	2004	2010	
C Nordic	Finland	2002	2007	
	Sweden	1999	2008	
	Denmark	2006	2009	2009 - 2012
	Norway	2007	2009	
B	Switzerland	2005	2009	
	Belgium	2004	2012	
	Austria	2006	2010	
A Terrestrial	France	2005	2011	2012 - 2015
	UK	1998	2012	
	Spain	2000	2010	
	Italy	2004	2012	

Table 1 – ASO transition

With the new digital technologies, several business models can be depicted and this is attracting both fixed and mobile operators. While analogical broadcasters tend to implement their own broadcasting systems and make use of them for direct commercial usage, some mobile operators admit to rent other broadcasters infrastructures to deploy their services. They can thus exploit their own infrastructures for the uplink traffic when using interactive services. These business models may lead to the co-operation between such companies with the objective of enhancing both of their profits. The line that separates mobile operators from broadcasters is becoming thinner which may lead to wider market coverage by the actual cellular companies. But is there a market for both mobile and fixed solutions?

While the fixed solutions will aim at delivering high quality digital TV to the users' homes being the direct transition from analogical TV, several market studies are being conducted in order to analyse the feasibility of deploying mobile TV solutions. Results confirm that most people are willing to pay up to 7.5 € for a mobile digital TV service as shown in Figure 1.

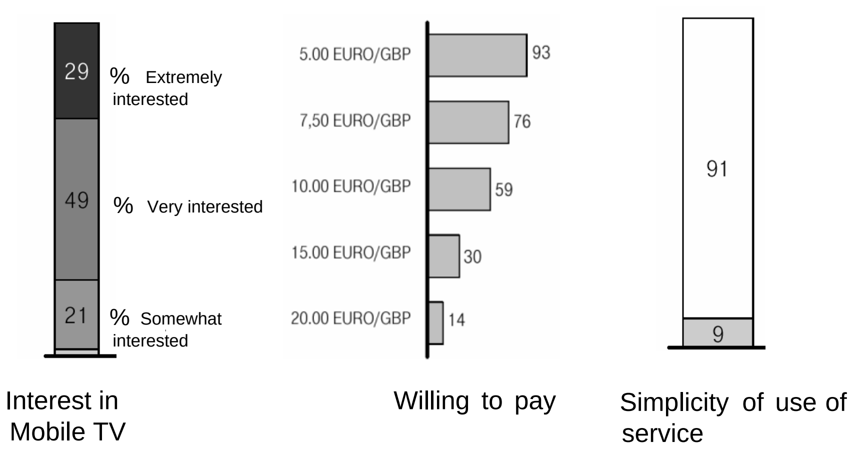

Figure 1 –Interest of users regarding Mobile TV

Figure 1 also indicates a concern on the user's side, regarding the complexity of the system in terms of usage simplicity. In order to fulfil such requirements ESGs (Electronic Service Guides) are being developed and enhanced. The end users should be only a button away from the contents, and be switched to a GUI containing all the available channels or programs on demand. Such software is being incorporated into mobile terminals, being that most of the current DVB handsets (DVB-H devices) only work when using such a feature.

As Digital TV walks towards its height, several solutions have been presented and are still, continuously appearing in the market for Digital Video Broadcasting, some already providing mechanisms for interactivity. The integration of these technologies with heterogeneous networks will however, catalyse the creation of new business models which better take advantage of such features. Multihomed terminals are more and more becoming a reality. Most of the nowadays' laptops are already equipped with WiFi and Ethernet cards, and many of its owners have chosen to buy 3/3.5G internet access cards. Although only few laptops are being sold with broadcast receivers, it is acceptable to consider that when such a concept is mass marketed, several technologies might be suitable to provide access for interactivity over unidirectional links, thus acting as the return channel.

With this enhancement the doors for other scenarios will open, and broadcast 2.0 will be born. Many TV shows already ask for the audience interactivity, via phone calls and SMS. When the shift from analogue to digital occurs, the simplified interactivity is an important issue to be taken into account, as if this was integrated into the Digital TV service. As an example, MTV alike channels, very popular amongst teenagers, could provide the possibility of direct download to the mobile terminal, of the music currently being present in video (buy what you see), thus providing a similar, but more modern concept to the iTunes. To support the validity of this idea, it is relevant to remark the success of current usage of cell phones as MP3 Players. Some of these devices are beginning to incorporate RSS feeds capabilities for news and podcasts, which can obviously be adapted for the transmission over broadcasting networks. If cell phones are to follow Moore’s Law regarding the exponential evolution of the memory, then such contents can also be stored in the device, so that the users select those they are more interested in receiving, with the device automatically storing them. These recording capabilities are also of interest when considering typical services provided according to an Electronic Program Guide (EPG).

Informational contents are also a possibility, providing local information related to the broadcast cell’s relevant cultural events, tourist attractions such as museums or parks, and even providing audio/visual tourist guides. Gaming experience may also benefit from a spreading of digital broadcasting. Several games already take profit of GPS information, but this technology was not conceived to provide high data transfers; it can still be of profit to acquire the positioning, to withdraw relevant information regarding that users location, and then by using a broadband solution the user can receive detailed information. MMORPGS are gaining more and more success and the transmission of scenario information can very well be a typical type of data to be broadcasted (not a type of information with intense delay/jitter requirements).

To support such features, these networks with specific characteristics need to be integrated with heterogeneous 4G networks, and be accounted for mobility and Quality of Service (QoS).

1.2 Motivation

In order to guarantee the success of futuristic broadcast networks, by assuring the deployment of the previously explained scenarios, several architectural issues must be taken into account by anticipating the technological constraints and adding support for 4G network’s features.

This dissertation presents considerations and results which were obtained and discussed in the scope of two projects: the IST-DAIDALOS and HAN4G.

DAIDALOS (**D**esigning **A**dvanced Network **I**nterfaces for the **D**elivery and **A**dministration of **L**ocation Independent, **O**ptimised Personal **S**ervices) is an integrated project funded by the European Commission, composed by 46 partners from industry and academia which focuses on creating an architecture integrating heterogeneous network technologies. DAIDALOS follows 5 key concepts:

- MARQS - Integrating end-to-end support of Mobility Management, AAA, Resource Management, QoS and Security across heterogeneous technologies;
- VID (Virtual Identities) – virtualization of identities for privacy and personalization;
- USP (Ubiquitous and Seamless Pervasiveness) – pervasiveness across personal and embedded devices allowing context transfer for mobility support and user request;
- SIB (Seamless Integration of Broadcast) - Both technological and at the service level;
- Federation – dynamic relations between operators, for a more competitive market.

These key concepts express the project's main concerns, and as it can be seen, the integration of broadcast technologies is a hot topic in discussion, especially motivated by the increase of interest on IPTV. Having broadcast technologies supporting IP connectivity and every service that the Internet is able to provide is surely a best add-on value, and may reason a generalized access to the contents. Since broadcasters need to perform the ASO until 2012 and IPTV is something that many users are already used to and use through their conventional internet accesses, the provisioning of the same contents may prove of great value. This is just an example of how the availability of IP contents over broadcast networks may prove to be advantageous; however, it requires a well defined architecture with good resources management.

DAIDALOS provides an operator driven architecture, by defining the core entities, the access network entities as well as the specific terminal entities required to support the desired features. Ad-Hoc networks integration is also a major concern of this project and these specific networks are seen as an extension of the operator's network, where terminals can access the same contents as when connected in infrastructure, and obviously be charged for them.

HAN4G project aims at the Integration of **H**eterogeneous **A**ccess **N**etworks in a **4G** **A**rchitecture. Hence, the main objective is to implement and test an integrated 4G architecture composed by diverse technologies, such as Ethernet, 802.11 (infrastructure and ad-hoc), 802.16 and DVB-T/H. The mobile terminals should be able to choose to connect to one (or more) of the available access technologies, depending on the services being requested, economical issues and specific user preferences. Special highlight is provided to broadcasting technologies DVB-T/H, at the technology level by: (1) deploying the necessary equipment, at the connectivity level; (2) by providing the means for service's delivery at the service level; and (3) by providing the desired services (either unicast, multicast or broadcast multimedia services), with a desired QoS to the end user. The specific goals are:

- Integration between infrastructure and ad-hoc networks;
- Integration between 802.16 networks and fixed wired (Ethernet) and wireless networks (802.11 ad-hoc and managed);
- Integration of unidirectional broadcast networks in operator networks;
- Seamless and fast mobility with QoS support between each access technology;
- The integration of Ad-Hoc networks is here seen as an extended mean to provide reachability to the operator's services, but not depending on it.

In the scope of this work, some conclusions are withdrawn, regarding the utility of such networks, especially regarding their usage within heterogeneous operator networks.

Both these projects deal with the integration of broadcast unidirectional technologies and mobility using DVB-T/H solution within 4G environments while gifting it of a heterogeneous return channel. Within these circumstances mobility support is a quest not easily solved, as it requires special interaction modules both on the network and terminal sides. Nowadays concerns already mind such a topic, and as we walk towards the future, more mobility requirements will have to be fulfilled. To disregard mobility support is to condemn the overall acceptance of the solution, which becomes the topic focus of this thesis.

1.3 Objectives

This dissertation aims at evaluating the feasibility of integrating unidirectional broadcast technologies with heterogeneous networks. For this purpose a 4G architecture was designed taking into account QoS provisioning and mobility support on both the down and uplink channels. Also a subset of this architecture was implemented to provide *"proof-*

of-concept" results regarding the seamless integration of the unidirectional link technology. This simplified architecture was implemented by envisioning the support of mobility with the creation of a terminal mobility controller interacting directly with the device's frontend, which reports performance metrics such as power levels, SNR (Signal to Noise Ratio) and BER (Bit Error Rate). Based on these values the controller is supposed to manage mobility on the terminal side and communicate the decisions to the network, which is supposed to perform control and admission.

Having such features in mind, a commercial set of DVB equipment was assembled and used to accomplish the objectives by withdrawing the necessary results. Some performance metrics are presented related to both technology usage scenarios: unidirectional usage for Digital contents' reception and bi-directional usage via the integration with a WiFi return channel.

The integration of DVB reception in scenarios with mobile terminals, which are also equipped with other technology devices such as WiFi, WiMAX or UMTS/HSDPA, is also within the scope of this study. Notice that most of current laptops are already equipped with WiFi transceivers; WiMAX transceivers are already being introduced in some laptops. It is then our understanding that these links can be used for the uplink channel, since multimedia service delivery usually requires reduced bit rates in the uplink. The presented metrics do also consider an analysis on the performance when presenting users with emulated bi-directional traffic for access to the internet when using the DVB as downlink. In short, this dissertation focuses on the study of the capabilities of this technology and possibilities to enhance its usage beyond simple broadcast of multimedia contents.

Furthermore this document provides a description of one of the most known Broadcasting technologies – DVB. The DVB standard provides a complete set of specifications dedicated to the System Information allowing a TV set to lock on a TV channel and retrieve the programs and associated services. This SI has been continuously enhanced to cover also datacast IP/Ethernet services, as well as mobility and interactive return channel.

1.4 Contributions

Concerning the activities expressed in the previous section several work has been presented in conference proceedings and journals.

The following work presents experimental study conducted on Self-Organization Mechanism, aiming to analyse the limits of an Ad-Hoc network:

- Miguel Almeida, Rafael Sarrô, João Paulo Barraca, Susana Sargento, Rui L. Aguiar, "Experimental Evaluation on the Usage of Ad-hoc Networks as Stubs for Multi-Service Networks", EURASIP Journal on Wireless Communications and Networking, 2007

- Miguel Almeida, Susana Sargento, João Paulo Barraca, Rafael Sarrô, Rui L. Aguiar, "On The Limits of Ad-Hoc Networks: Experimental Evaluation", Conftele 2007

- Susana Sargento; João Paulo Barraca; Miguel Almeida; Rafael Sarrô; Rui Aguiar; "Experimental Evaluation of an Integrated Ad-hoc Network 2006", Proc. IST - Mobile Summit, Mikonos, Jun 2006

The remaining publications refer to the provisioning of QoS and mobility support over 4G networks:

- Miguel Almeida, Pedro Neves, Daniel Corujo, Susana Sargento "Improving the Experience of Real Time Services in WiMAX Networks", Conftele 2007

- Miguel Almeida, Daniel Corujo, Susana Sagento, Rui Aguiar, "An End-to-End QoS Framework for 4G Mobile Heterogeneous Environments", OpenNET 2007

- Vitor Jesus, Susana Sargento, Daniel Corujo, Nuno Sénica, Miguel Almeida, Rui Aguiar, "Mobility with Quality-of-Service Support for Multi-Interface Terminals: Combined User and Network Approach", ISCC 2007

- Susana Sargento, Vitor Jesus, Filipe Sousa, Fabio Mitrano, Tina Strauf, Carsten Schmoll, Janusz Gozdecki, Gonçalo Lemos Miguel Almeida, Daniel Corujo, "Context-Aware End-to-End QoS Architecture in Multi-technology, Multi-interface Environments", Mobile Summit 2007

1.5 Document Outline

Chapter 2 presents an overview of the several existing broadcast technologies describing several technical specifications. DVB, DMB, 802.16 and MBMS are the highlighted technologies, due to their maturity and mobility concerns. Chapter 3 introduces architectural considerations regarding 4G Networks with mobility and QoS support. Also this chapter describes some generic concepts regarding self-organised networks and the necessary mechanisms to provide an extension to the operator's services. The last section provides an overview of a generic 4G architecture integrating the concepts previously referred as related work. Chapter 4 details the implemented architecture with the required modules to support the integration of broadcast technologies, expressing emphasis on Mobility and QoS Support. Chapter 5 presents the detailed specifications of the equipment composing the testbed, as well as the results for the performance evaluation of the integration of broadcast technologies in a 4G architecture. Finally chapter 6 presents the main conclusion withdrawn from this work and indicates future directions and possible trends.

Chapter 2 BROADCAST TECHNOLOGIES

Broadcast has always been a pertinent subject for research, development and commercialization. It is due to its profitability that several solutions were developed and are currently being studied and tested. This chapter introduces a set of the most significant broadcast proposals which aim to distribute multimedia contents over both, fixed and mobile environments. A short description is provided regarding for each technology with special emphasis on DVB, DMB, 802.16 and MBMS due to their maturity and mobility concerns. Several technical specifications are described introducing the scope of usage, scenarios of applicability and regional area of influence.

2.1 Digital Video Broadcast

DVB Project [1] represents a consortium of entities incorporating broadcasters, network operators, programmers and regulatory entities which aim to develop standards defining the physical and data layers for broadcast purposes. It is currently a forum with wider acceptance in Europe and it is one of the oldest in research terms, as it has been around since 1993 (in 1994 DVB-S and DVB-C were ratified). In 1995 the DVB-T standard was approved aiming at substituting the analogical television. Some trials were conducted over this technology to verify its feasibility to provide mobile digital television reception. The conclusions of such a performance evaluation conditioned the creation of a new standard in 2004. The DVB-H consists of a more modern standard taking into account mobility issues on the physical level as well as portability, but supports backwards compatibility as well. The following subsections provide more detail on each of the technological proposals regarding the DVB consortium and Table 2 shows a summary comparison of the main technical aspects for the three major solutions.

	DVB-S	DVB-T	DVB-H
Method	Single-Carrier	Muticarrier (OFDM)	Multicarrier (OFDM)
Carrier-frequency	10.7-12.7GHz	470-862MHz	VHF,UHF and L-Band 200MHz-About 1.7GHz
Channel bandwidth	26-54MHz	8MHz	5,6,7,8
Useful Data rate	45.4Mbit/s B:36MHz	24.7Mbit/s 64-QAM	2.49-31.67MHz(8MHz)
Modulation	QPSK	Up to 64QAM	Up to 64QAM
Channel-Property	Time-invariant	Time-variant	Time-variant

Table 2 – Comparison of basic characteristics for DVB-S/T/H

The DVB specifications are mainly covered by [3], [4], [5] and [6].

2.1.1. DVB-T

DVB-T is the standard for Terrestrial transmission and uses MPEG2 to stream video, audio and data contents. The MPEG2 Transport Streams (TS) refer to OSI's layer 2 and define a transmission method using MPEG2 to code audio and video, optimized for large bandwidths (bit rates bigger than 2Mbps). A TS is composed by multiple TSPackets with a fixed length of 188Bytes including a 4Bytes header with a well defined first byte which is used for synchronization. The remaining 13 bits refer to the Program Identifier (PID) with the purpose of indicating the logical channel (OSI level 2) to which the TSPacket belongs to. Multiple channels can be sent over one single physical channel using a TSMultiplexer. Each PID defines an unidirectional broadcast channel. When bidirectional transmissions are envisioned, there is also the possibility of using an indicator for the DVBRC. The TSPacket is shown in Figure 2.

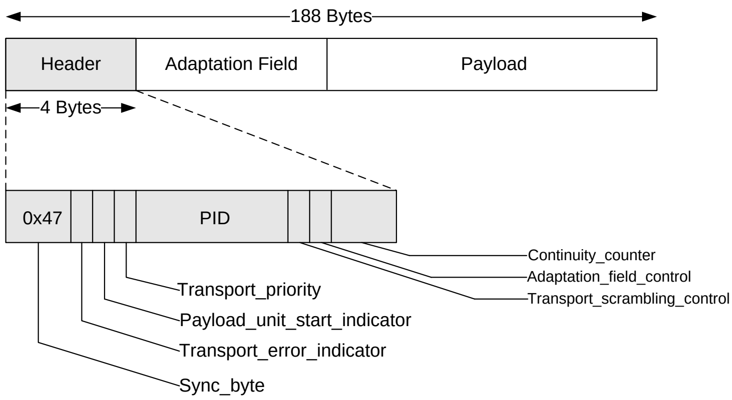

Figure 2 – Transport Stream Packet with special emphasis on its header

The needs for transmitting data make it necessary to assure compatibility of DVB with IP networks. In order to transmit IP or Ethernet packets, designed as Protocol Data Units (PDUs), it is necessary to perform an encapsulation process into a SubNetwork Data Unit (SNDU). SNDUs are fragmented into TS Packets forming a TS, in which each stream has the same PID. Program Specific Information (PSI) and Service Information (SI) are a major part in the DVB transmission system. PSI/SI form a set of tables which are transported in DVB frames. These tables provide the signalling mechanism related to the transported services and can be understood as service advertisement to the terminals. A terminal can in this way discover services that are provided to it by a certain operator. Each table is retransmitted in a certain time interval which is specified by DVB standards. In order to transmit IP data over DVBT/H networks two encapsulation protocols were created: MPE (MultiProtocol Encapsulation) and ULE (Ultra Lightweight Encapsulation). [56] presents a comparative study of both these encapsulation techniques by presenting the MPE's disadvantages and proposing ULE as a solution to them, by introducing results, namely, regarding the overhead.

DVB-T supports QPSK, 16-QAM and 64-QAM modulation schemes delivering up to 24 Mbps using 8MHz channel widths. It uses COFDM which enables either 1705 carriers (usually known as 2k), or 6817 carriers (8k). Also, at the link layer, it introduces a Reed-Solomon (RS) coding algorithm with 8% overhead and Interleaved convolutional coding.

2.1.2. *IPoDVB-T/H*

The demands for the transmission of data services make it necessary to assure compatibility with IP networks. In order to transmit IP or Ethernet packets, designed as Protocol Data Units (PDUs), it is necessary to perform an encapsulation process into a SubNetwork Data Unit (SNDU), as is shown in . SNDUs are fragmented into TS Packets forming a TS in which each stream has the same PID, as shown in . This process is described below.

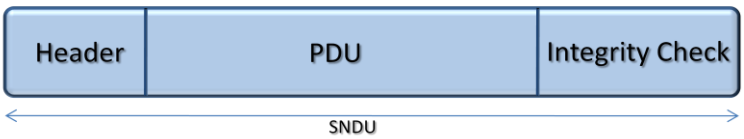

Figure 3 – PDU encapsulation into SNDU

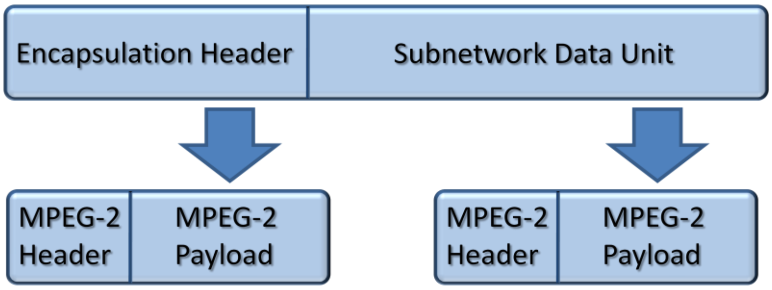

Figure 4 – PDU encapsulation into TS Packets

The encapsulation of PDUs into Ethernet is done using four different methods: Ethernet II, 802.3 Raw, 802.3 with 802.2 (LLC), 802.3 with 802.2 SubNetwork Access Protocol (SNAP). IP encapsulation only uses encapsulation of Ethernet II and 802.3 Snap. When using just Ethernet II, each frame transports more effective data. At a cost of some bits, we obtain a more complex and intelligent service connection oriented and with multicast and broadcast support. In this case we use SSAP (Source Specific Access Point) and DSAP (Destiny Service Access Point). 802.3 protocol with 802.2 Snap is used in cases when DSAP is the same as SSAP and is useful for protocols not standardized in SAP.

Upon the creation of the SNDUs, it is necessary to encapsulate them into the PDUs. MPE (Multi-Protocol Encapsulation) is a method that provides a mechanism for the transport of different protocols in a MPEG-2 TS. It can encapsulate any network protocol using LLC/SNAP and supports unicast, multicast and broadcast. It is a methodology which appeared long before ULE, and was optimized for transport of IP packets which were expected to be encapsulated in a Gateway of different protocol. ULE comes with the necessity to save bandwidth especially in satellite transmission systems like DVB-S, where bandwidth is especially expensive. Its usage becomes profitable because the header is adjustable, leading to situations where the reduction of overhead may be substantial.

0 1 2 3 4 5 6 7	8 9 0 1 2 3 4 5	6 7 8 9 0 1 2 3	4 5 6 7 8 9 0 1
Table ID	S | P | RES | Section Length (12 bits)		MAC 6
MAC 5	RES | PSC | ASC | L | C	Section No.	Last Section
MAC 4	MAC 3	MAC 2	MAC 1
IPv4 Datagram			
(CRC -32)			

Figure 5 – MPE header

0	1 2 3 4 5 6 7 8 9 0 1 2 3 4 5	6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
1	Length (15b)	Type = 0x800
IPv4 Datagram		
(CRC -32)		

Figure 6 – ULE header

There are several standardized encapsulation methods which will be presented bellow using ULE and MPE with considerations on the overhead each one introduces.

L2 Encapsulation	PDU overhead	L2 Frame Header Fields		
		Src MAC	Dst MAC	Type
ULE without dst MAC address	8	-	-	X
ULE with dst MAC address	14	-	X	X
MPE without LLC/SNAP	16	-	X	-
MPE with LLC/SNAP	24	-	X	X
ULE with bridging extension	22	X	X	X
ULE with bridging &NPA	28	X	X	X
MPE+LLC/SNAP+bridging	38	X	X	X

Table 3 – Encapsulation models comparison

2.1.3. DVB-H

When DVB-T tests started to be performed, it was concluded that this technology suffered from a major issue regarding power consumption. This is due to the fact that receivers are always active and always decoding information. In order to incorporate DVB technology with more portable devices such as Mobile Cellular phones or PDAs, it was urgent to come up with a solution which would minimize this problem. Having this in mind, several features were introduced in the DVB-H standard [7], which turned to be a logical evolution of the previous DVB-T standard. Time Slicing is an enhancement which enables the receivers only to turn into active mode when a certain PID, which the terminal is interested in receiving, is being broadcasted. Besides power consumption issues, special attention was devoted to the error protection, with the introduction of Reed-Solomon FEC on IP level (MPE-FEC), thus introducing three levels of error coding. For the advertisement of IP services, a new PID was defined for the creation of an INT table (IP MAC Notification Table). Also a 4k mode with time interleaving was introduced, as experimental and simulation results revealed that this transmission mode represents a good compromise between mobility and error tolerance demands. An adaptation for the 5, 6.7 and 8 MHz channel bandwidths was performed in order to make more feasible the usage of such a technology under the same frequency channels as, for instance, UMTS. Some reserved TPS (Transmission Parameters Signalling) bits have been used to express the existence of DVB-H, MPE-FEC and Time Slicing.

Time-Slicing aims at reducing the batteries usage, by dynamically switching the receiver on and off according to the intention on receiving a specific burst. In order to accomplish this, the terminals must synchronize to the burst of the desired services, and switch their receivers (front-ends) off when burst of other services are being broadcasted, thus entering in the "*off-time*". For streaming services, terminals play the information received in the last burst in such a way that users do not notice a discontinuous transmission. If one burst is lost, the service is interrupted until the next burst is received.

[51] concludes that the power saving values obtained may vart between 17% and 22%.. One interesting conclusion, also withdrawn in the same study, was that with the increase of the mean frame size, the power saving is reduced: for shorter frames the errors have less impact than for longer frames.

The MPE-FEC scheme was introduced to enhance the DVB-H capability to protect data against transmission errors on the link layer level. MPE-FEC is based on the Reed-Solomon parity data which is calculated from each burst of IP datagrams and then encapsulated along with the IP data into MPE-FEC sections. Then, they are transmitted after the MPE section of a burst in the same Elementary Stream, but with a different table ID value. This makes it easy for the receiver to differentiate two different types of sections. Each IP datagram is encapsulated into one MPE section. An Elementary Stream (ES) is formed out of MPE sections with a particular PID.

2.1.4. DVB-S/S2

DVB-S corresponds to the first version of the DVB's satellite standard [8], and is currently in use by several broadcasters which transmit their contents in MPEG-2 TSs. Since this standard was rectified in 1994, a new version of this standard was presented and ratified in 2005, proposing what is to be the second generation of DVB-S –DVB-S2 [10]. This new proposal uses Generic Streams which makes packet streams other than MPEG2 also valid, for instance, MPEG4 is now supported. More and higher modulation schemes are present besides QPSK (already defined in DVB-S), such as 8PSK, 16APSK and 32APSK. Error correction was also a concern as stronger FECs have been employed by the usage of BCH. Strong integration with return channels to support Adaptive Coding and Modulation (ACM) is required, aiming to lead to real time adaptability to the link characteristics while keeping support for Variable CM and Constant CM.

2.1.5. DVB-SH

From a merging of the DVB-H and DVB-S, a new standard was proposed trying to combine the strengths of both these technologies. With the portability features introduced by the handhelds and the worldwide coverage easily obtained by the deployment of satellite services, a strong solution may arise. However, some disadvantages can also arise, as for satellite purposes the line-of-sight is an important characteristic. This aspect is faded in the DVB-S by its use cases, which generically include home reception, and thus a fixed antenna on the top of a roof is acceptable. Nevertheless, DVB-SH offers a solution to this problem by combining an integrated system with direct satellite transmission to repeaters, which then broadcast the terrestrial signal using DVB-H. This is an architecture combining two technologies for a more centralized media distribution. The DVB Technical Module is performing studies on the impact of delivering Satellite Services to Portable Devices and in 2006 TM-SSP started to develop standards.

2.1.6. DVB-C

DVB-C is the DVB standard for cable transmission DVB-C [11], and is centred on the use of 64 QAM. For the European satellite and cable environment it can, if needed, convey a complete satellite channel multiplex on a cable channel. As the previously explained DVB flavours (except DVB-S2), it makes use of the MPEG2-TSs to transport media contents to the end users.

2.1.7. DVB-RC

The interactivity requirements have pushed the DVB consortium to produce a DVB return channel (DVB-RC). Such initiative was the start point for the definition of the three main flavours of DVB-RC (DVB-RCC for Cable, DVB-RCS for Satellite and DVB-RCT for Terrestrial). At a certain point, DVB-RCT [7] was considered to be promising in order to provide Home Movie interactivity. DVB-RCx is not currently being considered by the operators as a valid trend because of the power consumption compromises which the DVB technology already introduces. Also, typically the return channels are not used for transmitting video, especially MPEG-2, and terminals are already equipped with other networking solutions. Power consumption affects mainly the terminals: if the operators can reach higher areas coverage by consuming more power, they may see it as an advantage; however, higher distances between the terminal and the operator's antenna mean less autonomy for the terminal, when considering a return channel usage.

2.2 DxB

This subsection describes the evolution of the DxB technologies which are comprised of DAB, DAB+, DAB-IP, DMB; and T/S-DMB.

Digital Audio Broadcasting (DAB) (also known as the European Eureka 147 Project) is a technology for broadcasting of audio contents using digital radio transmission, and its standard was approved in 1993 with a wide acceptance. Currently over 50 countries use these specifications for radio broadcasting around 1.7MHz.

The natural evolution of DAB for media broadcast was defined as Digital Multimedia Broadcast (DMB). The DAB forum states that the shifting from DAB to any other Enhanced DAB/DMB technology is fast and economically appealing.

DMB was developed for terrestrial transmission (T-DMB) as well as for satellite S-DMB. T-DMB, however, has already been though with mobility concerns. It is a narrowband solution which makes it easier for a cellular operator to perform the deployment, as it may not need to acquire more frequency channel licences. Several studies have been conducted to obtain results considering the implementation of this technology and can be found in [59]. T-DMB has strong impact in South Korea, but some European countries like Germany, are also performing trials which can further make it a strong rival for DVB. S-DMB is the related technology for broadcast over satellites.

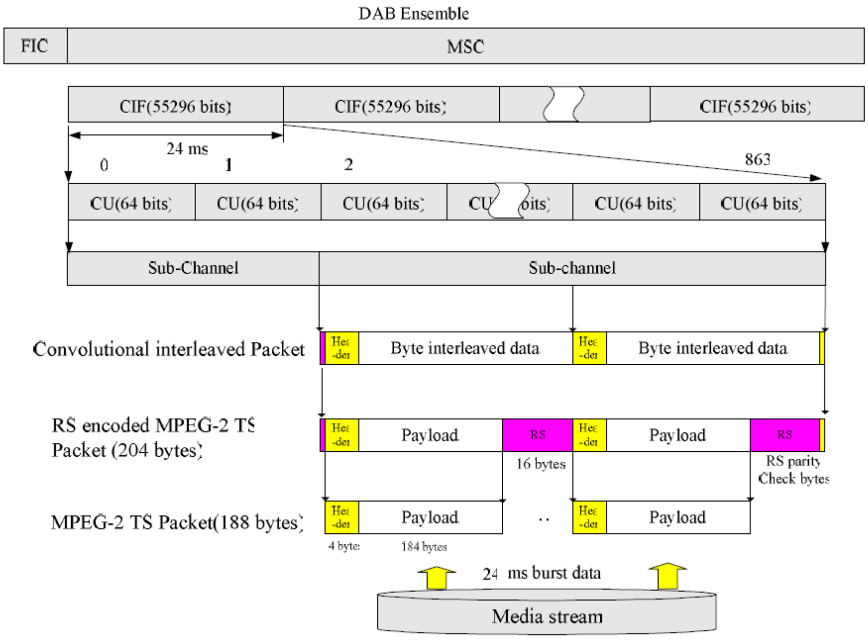

Figure 7 – Layer Structure of the T-DMB

T-DMB was initially based on MPEG-1 and MPEG-2 standards, but MPEG-4 was further included to perform the transport of video, and so far does not offer IP contents provisioning option. DMB states that the T-DMB receivers will be less complex than the DVB-H ones and more robust to fading. Time and frequency interleaving lead to the necessary robustness for mobile and portable reception, and power consumption can be reduced through macro time slicing as well as through power cycling. The DMB's Macro Slicing values can reach several minutes, which can be more adequate for transmission of information data, such as news or weather reports, which do not require a high refresh rate.

The low costs for deployment are also pointed as an advantage, due to the high proliferation of the DAB technology, which could be easily upgraded. On the other, and by using micro time slicing, the power saving is less efficient. Taking these considerations into account, the concept of enhanced DAB (eDAB) was created.

The focus was again turned to the DAB system with the incorporated IP stack which features two channels of special relevance, the Main Service Channel (MSC) for data transportation, and the Fast Information Channel for Service and Multiplex Configuration Information transport. MSC is time interleaved (in opposite to FIC) and each sub-channel can be separately error protected by a fixed data rate FEC.

DAB-IP was also developed in order to promote the usage of the existing infrastructure equipment and selling the idea that only few adjustments would be needed to support IP connectivity.

Seamless reconfiguration of services, e.g. changing data rates, error protection code rates, is enabled by the system and provides for a high degree of flexibility - including the removal or addition of services on the fly. DAB currently provides two variants of Mobile Television - DMB and DAB-IP based ones. Conditional Access as well as Digital Rights Management is enabled as well. In addition, DAB features an extensive set of multimedia and traffic information/navigation support applications such as Middleware/DAB Java, Digital Music Download (DMD), Voice Applications, Broadcast WebSite (BWS), SlideShow, TopNews or Dynamic Label.

2.3 IEEE 802.16

The IEEE 802.16 Working Group started in 1999 aiming at providing a solution for Wireless Metropolitan Area Networks; two years later, on December 2001, the IEEE 802.16-2001 standard [117], which specifies the air interface for fixed Line of Sight (LOS) Point-to-MultiPoint (PMP) broadband wireless access systems environments utilizing the 10-66 GHz frequency range, was approved by the IEEE. A unique, single carrier PHY layer is supported by this standard – the WirelessMAN-SC PHY layer. Several amendments followed, being the first one published in January 2003, the IEEE 802.16c-2002 standard [118], which defined the system profiles for the 10-66 GHz frequency range. The following amendment IEEE 802.16a-2003 standard [119] came out late in that same year. It defines the required enhancements and modifications for the MAC and PHY layers specifications to support the 2 - 11 GHz frequency band, supporting Non-LOS environments. In this standard, PMP and mesh topologies were also included. Multiple physical layers, single and multi-carrier, became supported, each suited to a particular operational environment. A single carrier PHY layer, named WirelessMAN-SCa air interface, and two multi-carrier PHY layers, WirelessMAN-OFDM (256 carriers) and WirelessMAN-OFDMA (2048 carriers), were defined. In 2004, the IEEE 802.16d-2004 [120] standard was published specifying the profiles for the IEEE 802.16a-2003 standard. This standard also specifies a Medium Access Control (MAC) layer and several Physical (PHY) layers, supporting both NLOS and LOS. The MAC layer is connection oriented and provides Quality of Service (QoS) assurances through the usage of service flows and uplink scheduling services. A set of convergence sublayers are defined to map the upper layer packets into the 802.16d-2004 system. The convergence sublayers support packet based protocols, such as support for IPv4 and IPv6, as well as cell based protocols, such as Asynchronous Transfer Mode (ATM). Both point-to-multipoint (PMP) and mesh modes of operation are supported by the standard, despite the mesh mode of operation is optional. Finally, in December 2005 a new revision – the IEEE 802.16e-2005 standard [121] – was introduced to allow mobility support for the Subscriber Stations (SS).

2.3.1. WiMAX

Worldwide Interoperability for Microwave Access (WiMAX) was defined by the WiMAX forum consortium [14], in order to improve conformance and interoperability of the IEEE 802.16 standard, officially known as WirelessMAN, with the manufactures, deploying companies and end-users, by certifying compliant equipment. Originally, the IEEE standard 802.16d was thought to provide wide internet access to the home. In this way the focus of the research was not mobility, as a Base Station (BS) could be located in the centre of a city and SSs should be spread throughout the buildings. The potential of this technology to provide multimedia contents distribution was realized in an early stage, and is still a strong research issue that is being exploited by several broadcasters.

In order to keep up with other emerging technologies which aim to more mobile scenarios, IEEE had to take measures and thus introduce an amendment to 802.16d. The 802.16e standard is technologically similar to other more recent solutions as it uses Scalable Orthogonal Frequency Division Multiple Access (OFDMA) in both the uplink and downlink channels to enable simultaneous multiple access (OFDMA). The disadvantage of this enhancement is that this feature alone introduces incompatibility between 802.16d and 802.16e. Even if there was a 256 tone option within 802.16e, the differences between the two MAC layers would prevent the fixed and mobile versions from working together. Nevertheless, SOFDMA brings an additional advantage over OFDMA. It scales the size of the Fast Fourier Transform (FFT) to the channel bandwidth in order to keep the carrier spacing constant across different channel bandwidths. Constant carrier spacing results in higher spectrum efficiency when considering wide channels, and a cost reduction in narrow channels. Additionally, 802.16e introduces extra FEC mechanisms, Multiple Input Multiple Output (MIMO) support, an extended real time polling scheduler (erTPS), and support for Multicast and Broadcast services in the MAC layer.

WiMAX is thus expected to provide a fixed, nomadic, portable and mobile wireless broadband connectivity without the need for direct LOS with a BS, although 802.16 offers both the options NLOS and LOS (Table 4). In typical cell radius deployment of 3 to 10 km, WiMAX Forum systems can be expected to deliver capacity up to 40 Mbps per channel, for fixed and portable access applications.

Mobile network deployments are expected to provide up to 15 Mbps of capacity within a typical cell radius deployment of up to 3 km. WiMAX defines Point-to-Point (PP), PMP and Mesh. In PMP mode traffic only occurs between the BS and SSs. The BS is connected to public networks and serves subscriber stations with first mile access to public networks. Several Cellular operators are considering WiMAX technology as an option for the broadcast of multimedia contents. There is a lot of work concerning QoS provisioning over WiMAX networks as well as mobility. WiMAX forum itself has provided an architecture for such purpose, which may not prove to be the most indicated when considering the integration with other heterogeneous technologies. At the same time, the Mesh possibility is also a research issue, which may bring light to many PMP applications.

Definition	Devices	Locations/ Speed	Handoffs	802.16-2004	802.16e
Fixed access	Outdoor and indoor CPEs	Single/ Stationary	No	Yes	Yes
Nomadic access	Indoor CPEs, PCMCIA cards	Multiple/ Stationary	No	Yes	Yes
Portability	Laptop PCMCIA or mini cards	Multiple/ Walking speed	Hard handoffs	No	Yes
Simple mobility	Laptop PCMCIA or mini cards, PDAs or smartphones	Multiple/ Low vehicular speed	Hard handoffs	No	Yes
Full mobility	Laptop PCMCIA or mini cards, PDAs or smartphones	Multiple/ High vehicular speed	Soft handoffs	No	Yes

Table 4 - Types of access to a WiMAX network

Despite a range of multicast connections being defined for polling purposes, the 802.16-2004 does not define a multicast/broadcast oriented connection dedicated for data (only for signalling). 802.16e, on the other hand, already defines Multicast Broadcast Services (MBS) for data transmission for Single BS access and multi-BS access. MBS aims at creating a solution to provide high data rate and coverage using Single Frequency Networks (SFN), flexible allocation of radio resources, low MS power consumption, support of data-casting in addition to audio and video streams, and low channel switching time. This clearly demonstrates a concern to add features in order to compete with other broadcasting solutions already conceived taking some of these issues into account.

When considering the Single BS access mode, the BS creates a multicast traffic connection with each MS, or a broadcast transport connection. Any available traffic Connection ID (CID) can be used for this purpose, and each multicast MAC SDU is only transmitted once per BS channel.

Regarding the Multi-BS Access mode, the MS can be registered via different BSs, hence each MS needs to register to the MBS at the network level (register within a MBS zone). The BSs are synchronized in the transmission of the broadcast/multicast frames and the CID for a specific MBS will be the same for all BSs distributing the service. This feature permits the terminals to receive signal from diverse BSs, enforcing the signal reception as well as the overall performance by decreasing the BER.

To support these features, the concept of MBS zones was created. Using SFN operation and flexible duration of MBS zones permits the scalable assignment of radio resources to MBS traffic.

2.3.2. WiBro

WiBro (Wireless Broadband) is a Korean wireless broadband Internet technology. It has been deployed with rather satisfactory results in terms of utilization by the end users. There is a lot of confusion regarding the relation between WiBro and WiMAX, even in the literature, which resulted from the anticipation on the implementation of the Korean solution regarding the proceeding publishing of the 802.16e amendment. Hence, WiBro can be best described as what it was expected for the WiMAX Mobile solution to be like. According to the WiMAX Forum, some more recent clarifications were made by the Korean Ministry of Internal Affairs and Communication, in order to shape the WiBro specifications under the IEEE 802.16e-2005 and to promote product certifications provided by the Forum. WiBro is a Mobile WiMAX service based on the same IEEE 802.16e-2005 standard as Mobile WiMAX. Both Mobile WiMAX and WiBro have similar PHY, MAC and Power Classes and use equipment to be certified by one of the Mobile WiMAX certification profiles. According to the WiBro website [122], it *"Demonstrates a leading*

Mobile WiMAX service that operators around the world are actively looking towards as a model that will ensure the commercial success of Mobile WiMAX technology".

WiBro and WiMAX are hence expected to walk along and cooperate, as WiBro is assuming a subset role of the forum, instead of the pioneering image it created upon its deployment.

WiBro uses Time Division Duplex (TDD), for duplexing, OFDMA for multiple access and 8.75MHz as a channel bandwidth at the 2.3GHz frequency band. It was developed to overcome the speed limitation of mobile phone (for example CDMA 1x), as well as to add mobility to broadband Internet (for example ADSL or Wireless LAN). Its main success reason has been the fast deployment, since WiMAX's 802.16e version is still only in testing stages. The entrance in the telecommunications market was also due to the good manufacturing of integrated Mobile Receivers, where power consumption compromises seem to be reasonably tolerated by the users. WiBro base stations offer an aggregate data throughput of 30 to 50 Mbps and cover a radius of typically 1 km with a maximum of 5 km, allowing for the use of portable internet usage. In detail, it will provide mobility for moving devices up to 120km/hr compared to Wireless LAN having mobility up to walking speed. The inclusion of QoS allows for WiBro to stream video content and other loss-sensitive data in a reliable way. Unlike the SS of the WiMAX, the end user receivers are denominated Personal Subscriber Stations, clearly to underline the mobility features.

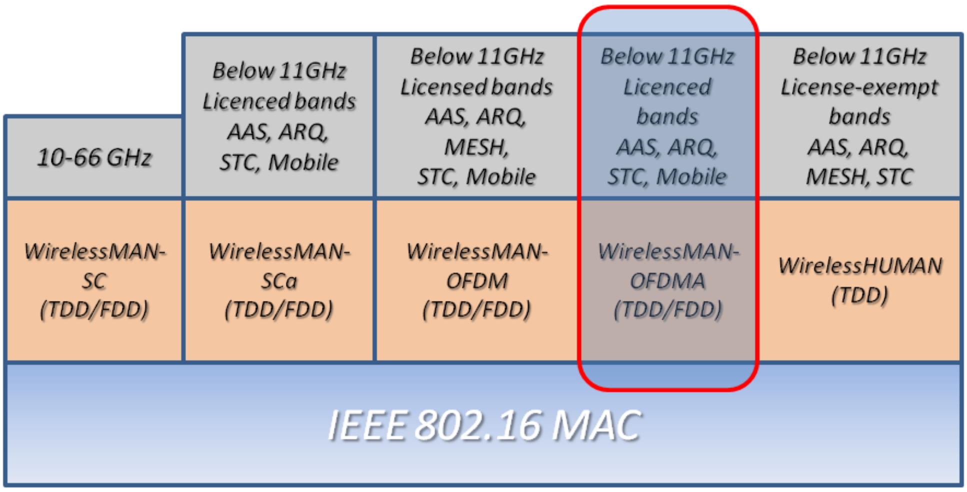

Figure 8 – WiBro's Role under the 802.16 specifications

Other aspects of WiBro are presented in Table 5

Frequency reuse factor	1
Mobility	<= 60 [Km/h]
Service Coverage	<= 1 [Km]
Spectral efficiency [bps/Hz/cell(sector)]	Maximum DL/UL= 6/2 Average DL/UL = 2/1
Handoff	<= 150 [ms]

Table 5 – Aspects of WiBro

2.4 MBMS

Several Cellular operators have expressed their interest in utilizing their already deployed infra-structures to distribute media contents. Especially for that purpose, 3GPP has been working on Multimedia Broadcast/Multicast Services (MBMS), a sub-system that allows the delivery of IP multicast datagrams to end-user terminals with specified QoS demands. The key motivation for integrating multicast and broadcast extensions into mobile communication systems, both UMTS Terrestrial Radio Access Network (UTRAN) and GSM/EDGE Radio Access Network (GERAN), is to enable efficient group related to one-to-many data distribution services.

On the control plane, MBMS manages bearer service activation status of the MNs, outsources authorization decisions to a newly introduced Broadcast Multicast Service Centre (BM-SC), provides control of session initiation/termination by the MBMS user service, and manages bearer resources for the distribution of MBMS data. As most UMTS equipments are IP enabled, IP also plays a key point role in MBMS, being used to identify the particular instance of the bearer service (which is composed by an IP multicast address and an access point name - network identifier) and to manage all MBMS multicast services. The Gateway GPRS Support Node (GGSN) serves as the entry point of IP multicast traffic as MBMS data. Upon notification from the BM-SC, the GGSN is responsible for setting up the required radio resources for the MBMS transmission inside the UTRAN/GERAN. The UTRAN decides on the appropriate radio bearer based on the number of users within a cell, prior to, and during a MBMS transmission. Mobility aspects are intrinsically supported in UTRAN/GERAN, but further mobility needs to be supported by the Serving GSN (SGSN), requiring the capability to store a user-specific MBMS context for each activated multicast MBMS bearer service (Figure 9).

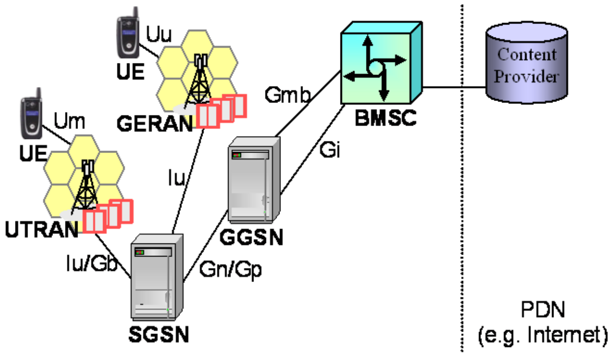

Figure 9 - MBMS Reference Architecture

The BM-SC functionalities acts as an IP Encapsulator, but is also responsible for setting up and controlling the MBMS transport bearers. It can also be used to schedule and deliver MBMS transmissions and is responsible for providing the announcement of services to the mobile terminals. These announcements contain all necessary information, such as multicast service identifier, IP multicast addresses, time of transmission, media descriptions, that a terminal needs in order to join an MBMS service. It is thus a crucial entity for the operator as it acts as a service gateway, both on the connectivity and service layers. Moreover, its meaning is of especial relevance considering that it can be used to manage security and keep charging records for the operators' specific contents.

MBMS is able to operate in two modes: broadcast and multicast (Figure 10). The broadcast mode works in a simplified manner, since it does not involve subscriptions management. It is composed by 5 phases: service announcement, session start, MBMS notification, data transfer and session stop. The service announcement is used to provide the MN with information on available MBMS services. The announced information includes parameters required for the service activation, such as service start time and content information, security parameters and associated delivery services. The session start phase is characterized by the trigger for bearer resource establishment for MBMS transfer. In the next phase - MBMS notification phase - the MNs are informed of forthcoming and ongoing MBMS broadcast data transfers. The following phase is the actual data transfer where MN receives the file or the streaming session announced. Finally, when the BM-SC has no more content to be delivered, the session stop phase releases the bearer resources.

In comparison to broadcast, the multicast mode adds three phases. There is a first subscription to the MBMS bearer. This allows reception of the service announcement, after which a Joining phase is required, consisting of the subscription to a specific multicast service, through the issuing of a Join message. The last additional phase corresponds to the termination of the service interest, by leaving the multicast group.

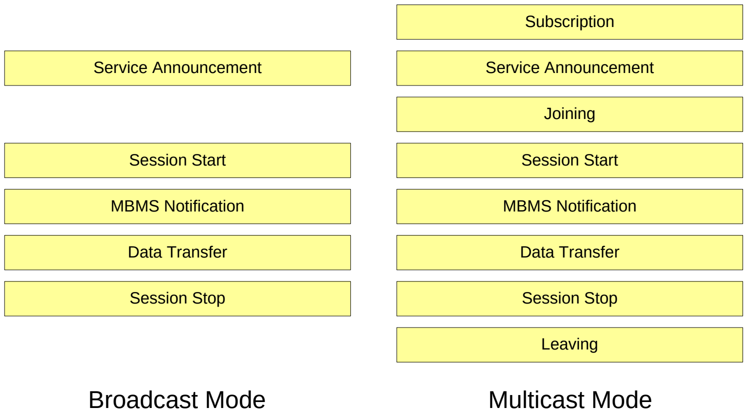

Figure 10 - MBMS Broadcast and Multicast phases

2.5 Other Broadcast technologies

2.5.1. ISDB

Integrated Services Digital Broadcasting's (ISDB) name was formed after its alikeness to ISDN, since multiple channels of data are multiplexed and then transmitted. This similarity is also present for the DMB case expressed above.

ISDB is divided into ISDB-S for satellite transmission, ISDB-T for terrestrial transmission, and ISDB-C for cable transmission. It operates at 2.6GHz band mobile broadcasting for high mobility terminals, which are all based on MPEG-2 video and audio coding (capable of high definition television - HDTV). This standard was developed in Japan by DiBEG [12], and its deployment is limited to that country; although some other countries such as Brasil are performing trials. ISDB-T [13] operates on unused TV channels, an approach taken by other countries for TV but never used for radio. ISDB also defines data connections (data broadcasting) for interactive interfaces like data broadcasting and electronic program guides (EPG).

ISDB-T (in VHF and/or UHF band) uses COFDM and supports QPSK, 16QAM, 64QAM and DQPSK modulation schemes. It divides the frequency band of one channel into thirteen segments. The combination of segments can be selected: this choice of segment structure allows for flexibility of services. For example, ISDB-T can transmit a LDTV and a HDTV using one TV channel or change to 3 SDTV, a switch that can be performed anytime. ISDB-T can also change the modulation scheme at the same time. The theoretical bitrates are shown in Table 6 and vary depending mostly on the type of constellation for each channel width. Field trials have shown that 21.47Mbps is an upper bound for stationary reception. [82] shows power consumption vs BER concerning both mobile and stationary cases; it shows that in order to achieve the same BER with approximate power consumption, the achieved bitrate is of 4.06 Mbps, since the DQPSK modulation needed to be used.

Channel Width	Information rate
6MHz	3.651~23.234Mbit/s
7MHz	4.259~27.107Mbit/s
8MHz	4.868~30.979Mbit/s

Table 6 – ISDB-T Bitrates

2.5.2. MediaFLO

MediaFLO (Forward Link Only) was developed by Qualcomm and is expected to be USA’s response to DVB-H, T-DMB or T-ISDB. It aims at providing media contents to cell phones, PDAs and laptops such as audio and video streams, individual video and audio "clips", as well as information such as stock market quotes, sports scores, and weather reports. It is specially designed to be adopted by other Mobile Operators, especially cellular ones; this was the reason to make MediaFLO systems to use frequency spectrum at approximately 700 MHz, which was previously allocated to UHF TV Channel 55. The MediaFLO System adds a Media Distribution System to the FLO Technology, which defines the Physical and Link layers.

FLO uses as the modulation technique and incorporates Turbo Codes for forward error correction techniques, as well as a Reed-Solomon erasure correcting code. FLO superframe has the duration of exactly 1 second, and consists of 4 frames of equal duration, each roughly 1/4th of second. These packets are first RS-encoded and then

Turbo-encoded. They are referred to as MAC layer packets. The packets with data and CRC bits are Turbo encoded and then transmitted.

Transmission and reception are based on using 4096 (4K) subcarriers and the QAM modulation symbols are chosen from a QPSK or 16-QAM alphabet. There are 4000 active subcarrieres per FLO OFDM symbol which are equally divided into eight interlaces. An interface thus consists of 500 subcarrires which are evenly spaced across the FLO signal bandwidth.

Although low to moderate data rates (i.e., tens of kbps),can be sufficient for voice streams, video streams on the other hand, may require bit rates ranging within 200 – 300 kbps. Regarding data connections, bitrates are the most valuable marketing item, as users nowadays always demand for more bandwidth, at the smallest price. MediaFlo can provide larger bit rates reaching up to 14.9Mbps, when using a channel width 8MHz and more complex modulation schemes such as QAM. The effective use of a statistical multiplexing system improves the spectral efficiency.

Each FLO service can be carried over one or more logical channels, Multimedia Logic Channel (MLCs). An MLC has the attribute that it contains one or more decodable subcomponents of a service that is of independent reception interest. Furthermore, an important aspect is that MLCs are distinguishable at the Physical layer.

2.5.3. Power Lines

Power Lines are inherently of a broadcast nature; several efforts have been made in order to provide internet connectivity “to the home” using the power grid which already has a high penetration index regarding the in human created structures, such as buildings, street illumination or power generating centrals. It may very well become an interesting solution for the provisioning of multimedia broadcast contents. IEEE is already working into this subject with the IEEE BPL workgroup: “Standardization of Broadband Over Power Line Technologies” which covers the following:

i) IEEE P1675, "Standard for Broadband over Power Line Hardware", which is a working group focused on hardware installation and safety issues;

ii) IEEE P1775, "Powerline Communication Equipment - Electromagnetic Compatibility (EMC) Requirements - Testing and Measurement Methods", which deals with PLC equipment, electromagnetic compatibility requirements, and testing and measurement method;

iii) IEEE P1901 "IEEE P1901 Draft Standard for Broadband over Power Line Networks: Medium Access Control and Physical Layer Specifications", which is a working group for delivering broadband over power lines aiming to define medium access control and physical layer specifications for all classes of BPL devices - from long distance connections to those within subscriber premises.

Also, “POWERNET: Broadband over Powerlines that works and meets the users expectations” [81] is an IST project which aims at developing a plug and play Cognitive Broadband Network over power lines by providing the equipment according to the regulatory requirements concerning electro-magnetic radiations.

One of the most severe problems is the weak quality of the existing power grid especially concerning older structures, which introduce a high noise variance and are the source for high interferences regarding the quality of the received multimedia contents. On the other hand, the possibility to have direct visualization without the need for extra components besides a Television itself along with the easy deployment of the solution indicating low costs, make this a hot subject for the operators. For simple broadcast it would only be needed (considering optimal conditions) a transmitter on the operator side, maybe within a power plant and a special receiver to withdraw the signal out of the power line. Multicast support would require the addition of extra hardware to enhance the efficiency of the contents’ distribution.

2.6 Summary

This chapter introduced the most significant broadcast proposals. DVB was one of the first unidirectional technologies which are intended to provide digital video contents (MPEG-2) via terrestrial and satellite antennas as well as through cable. More recently, and given mobility concerns, DVB-H was also proposed to cope with such a requirement. As a strong competitor, DMB urges as a more recent technology but nevertheless supported by a wide variety of manufacturers. These are the two strong unidirectional technologies which are competitors in Europe. MediaFlo seems to be the main broadcasting technology in North America while ISDB is making its way over the Asian market and some south American countries. Regarding Bi-directional links, WiMAX and WiBro are being developed and deployed with multicast/broadcast support in their most recent versions, with the detail that WiBro has already began commercial deployment over South Korea. However the

American and European continents seem to be very receptive to WiMAX proposals both fixed and mobile. MBMS is a strong candidate for provisioning of multimedia contents over cellular networks, as it represents few technological to the UMTS specifications. However many handhelds are expected to be equipped with either DVB-H or DMB solutions.

Chapter 3 4G ARCHITECTURAL CONSIDERATIONS

This chapter presents a set of considerations regarding architectural concepts related with the evolution of the most significant architectures developed for Telecommunications Networks. Particular interest is taken on QoS and Mobility with the objective of providing related work on both the topics. Still considering the concept of 4G networks, a group of proposals for multi-hop, auto-organized networks is presented, introducing a variety of mechanisms which aim at supporting operator's services and features. The last section presents the outcome architectural solution of the generic architecture which was designed taking into consideration the previously mentioned concepts.

3.1 4G Networks

The frameworks deployed by the mobile telecommunications' operators have been created upon a terminology according to the generation of the technology. The technological differences between 1G and 2G were substantial, as the first mobile devices operated on different frequencies, and with the introduction of 2G, several mobile terminals could share the same frequency in different time slots. The next step was to allow several users on the same time-slot and frequency, thus creating what is nowadays known as the 3G architecture, which already focuses on IP connectivity and enhanced mobility support.

The term 4G is an abstract concept that does not specify a well defined framework or an architecture itself. It results from the attempt to provide a feasible solution taking into account several issues such as QoS, mobility and pervasiveness, on futuristic environments, by solving futuristic problems, which are envisioned to exist and to be considered as requirements. 4G architectures have been mainly dealt at two levels: the radio level, which results from investigation efforts regarding the access to the medium, especially the access to the air-interface; and the network level, dealing with connectivity issues, provisioning of services with specific requirements as well as the deployment of the framework to support all bouquets of imaginable services, which are expected to be provisioned in the future. 3G technology already became a mixture of these two, as it integrates a new access to the medium (UMTS uses CDMA instead of 2G's TDMA) and some architectural changes (to support for instance, the IP layer).

Several Next Generation Network (NGN) [28][29] frameworks are now under study by standardization organizations, including the International Telecommunication Union – Telecommunication Standardization Sector (ITU-T) [30] Study Group 13 (SG13) and Focus Group NGN (FGNGN), and the European Telecommunications Standard Institute (ETSI) Telecoms & Internet converged Services & Protocols for Advanced Networks (TISPAN) [31] initiative.

The TISPAN is a standardization body of ETSI specialised in fixed networks and Internet convergence. It was formed in 2003 from the combination of the ETSI bodies Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) and Services and Protocols for Advanced Networks (SPAN). Both TISPAN and 3GPP consider IMS as the emerging technology for fixed and mobile convergence; hence, TISPAN acts as the fundamental architecture layout for the evolved service architecture for multimedia broadcast/multicast services.

IP Multimedia Subsystem (IMS) is a 3GPP [34] architecture for multimedia services based on the IP protocols defined by IETF (Internet Engineering Task Force). 3GPP adopted the SIP (Session Initiation Protocol) protocol as the session control protocol to manage multimedia sessions. As a consequence, SIP becomes the protocol used to setup, modify and tear down multimedia sessions over UMTS networks. It was designed to deal with point-to-point connections. For services involving groups of users receiving the same information, this limitation may be a drawback: since IMS only supports unicast transmission, resources may be consumed redundantly when data is to be delivered to a group of users. As an example, within a multiparty conference, a point-to-point connection between the media server and each user is created. A more efficient solution is to create a common bearer connecting the media server to all the participants. By using a multicast bearer, both radio and core network resources are saved, thus becoming available for other services.

IMS deals with multimedia content using SIP signalling for managing the real-time multimedia streams. Within the IMS architecture (Figure 11) the MRFP (Media Resource Function Processor) deals with stream delivery to different users using different codecs. However, it was not designed to handle non real-time streamed content and, as a result, does not provide proper support for delivery of files or other non real-time formats.

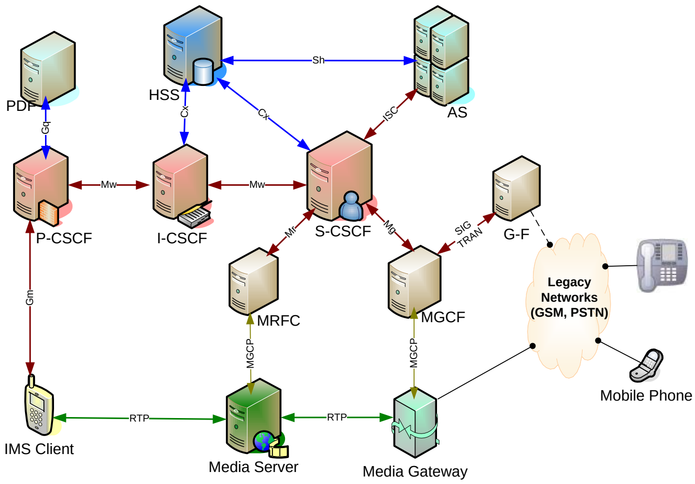

Figure 11 – IMS Architecture

In the first standards describing NGN, the ITU-T defines requirements and an overall framework of NGN including QoS [27][29].

The QoS architecture in NGN networks must be integrated with other subsystems including mobility protocols and authentication, authorization and accounting (AAA) mechanisms. Currently, NGN standards groups are becoming tightly coupled with mobile groups – SG13 and FGNGN with SG19, and TISPAN with 3GPP. E.g. in [33] it is addressed integration of 3GPP QoS architecture [36] in NGN architecture, including mapping between 3GPP QoS classes and QoS classes defined in ITU-T Y.1541.

Daidalos [35] is a project that presents a more flexible architecture with a more comprehensive set of characteristics, such as: a fully integrated approach to IP-based communication with different types of applications and protocols (e.g. legacy, Next Steps in Signalling (NSIS) and SIP-based applications are supported), including adaptive applications, multicast and broadcast; integration with global and local mobility protocols; integration of mobility, QoS, Authentication, Authorization, Accounting, Auditing and Charging (A4C) and multimedia services. It combines several technologies and integrates them within its own defined framework to provide QoS access to operators' services.

Administrative domains incorporate one or several Local Mobility Domains (LMD), each one connecting to access networks (AN). The Layer 3 connection between the AN and the user is performed through Mobility Access Gateways (MAGs); each Layer 2 connection is performed through Access Points (AP). From the point of view of mobility and connectivity, each LMD is managed by a Local Mobility Anchor (LMA). The LMA concept was borrowed from mobility architectures such as Hierarchical Mobile IP (HMIP) [65] or Network Local Mobility Management (NetLMM) [38], where an on-path network element provides stateful per-host forwarding. The most important feature resulting from this hierarchy is that the terminal is not required to reconfigure itself (e.g., obtain a new IP address) each time a handover occurs; consequently, only Layer 2 mobility schemes need to be in place inside this LMD and independent local mobility schemes are supported in different LMDs (for more details, please check [62]). Each LMD may contain different access networks with different technologies. Therefore, a homogeneous QoS support over heterogeneous technologies needs to be in place; and a major issue here is the provision of QoS-enabled mobility mechanisms to support seamless mobility across these heterogeneous technologies. Each technology specific entity or interface (in case of the MN) has a dedicated RAL (Radio Access Layer) which is responsible for monitoring and communicating performance metrics.

Other proposals for NGN architectures have been made, e.g. by the projects AQUILA [41], MESCAL [42], EuQoS [43].

AQUILA proposes architecture for QoS over the Internet using approaches such as Differentiated Services (DiffServ) [111], Integrated Services (IntServ) [109] and label switching technologies MultiProcototocol Label Switching (MPLS) [44], which have been exploited and significantly enhanced. The architecture has been designed to be cost-effective and scalable. Technical solutions have been verified by experiments and trials, including QoS-enhanced on-line multimedia services. The architecture aims at enabling dynamic end-to-end QoS provisioning in IP networks for QoS sensitive applications, analyze customer requirements and market situations and to create applicable business plans, to provide a toolkit for migration of end user applications to QoS, create tools for QoS monitoring and management, and develop and integrate a distributed QoS measurement infrastructure

According to the project's site, MESCAL's key objective is: "to propose and validate scalable, incremental solutions, enabling flexible deployment and delivery of inter-domain QoS across the Internet at large, with the following sub-objectives:

- To develop business models, based on current commercial practice and emerging business scenarios, describing the roles of and relationships between the stakeholders involved in providing QoS-based services across domains.
- To specify a generic, multi-domain, multi-service functional architecture for the flexible deployment and delivery of inter-domain QoS-based services.
- To develop templates, protocols and algorithms for the specification, negotiation, subscription and invocation of QoS-based IP services between customers and ISPs and between peer ISPs.
- To enhance existing inter-domain routing protocols and algorithms, and to investigate new approaches to convey QoS information to enable scalable inter-domain traffic engineering solutions.
- To examine the impact of:
    - IPv6 on inter-domain traffic engineering (TE) and to ensure that the TE solutions proposed by the project are applicable to both IPv4 and IPv6 infrastructures.
    - both unicast- and multicast-based services on inter-domain TE.
    - inter-domain aspects of Sevice Level Specifications (SLS) management and TE on corresponding intra-domain aspects, and vice versa, and to investigate the co-operation required between them.
- To adopt a policy-based approach to service provisioning and network operation and investigate policies for SLS negotiation, admission, and inter-domain TE.
- To evaluate and validate the devised algorithms and protocols through simulation and testbed prototypes.
- To contribute to international standardisation efforts, especially the IETF, and to participate in other consensus-forming activities in the IST program."

EuQoS is also a research project aiming at the integration, testing and validation of end-to-end QoS technologies to support their framework for advanced QoS-aware applications over multiple, distributed and heterogeneous network domains. EuQoS targets to develop solutions for various access networks (e.g. WIFI, Ethernet, UMTS) that communicate through the GEANT Core [45]. GEANT's principal purpose was to develop the GANT network - a multi-gigabit pan-European data communications network, reserved specifically for research and education use. The project also covered a number of other activities relating to research networking. These included network testing, development of new technologies and support for some research projects with specific networking requirements. The heterogeneous infrastructure that represents future pre-production networks requires the interoperability of dedicated QoS solutions concealed behind a common and well-defined interface. EuQoS aims at the development of new state-of-the-art methodologies that consist of the following modules: Monitoring and Measurements, Admission Control, Failure Management, Signaling & Service Negotiation, Security and AAA, Charging, and Traffic Engineering & Resource Optimization.

Main issues to be considered when planning a 4G architecture are seamless Mobility to the users and QoS assurances, which are discussed bellow.

3.2 Seamless mobility

Handovers can be mainly considered to be either initiated by the network (Network Initiated HandOver - NIHO) or by the mobile terminal (Mobile Initiated HandOver - MIHO). MIHO has been the approach taken inside IETF, since it embodies the end-to-end paradigm of the Internet. The simplest case is scanning the wireless medium for beacons, disconnecting from the current AP (Access Point) and connecting to a newly detected AP. After being connected, the mobile terminal must "redo the network" (e.g., send Mobile-IP binding updates). The performance of this process often depends on authorizations and reconfigurations at the network level, and its improvement is essential for seamless mobility. This is the main problem IETF has been tackling for a long time and some results have been obtained (e.g. Candidate Access Router Discovery - CARD [63] with Fast Mobile IP - FMIP [66] or localized mobility proposals, such as HMIP). However, this end-to-end paradigm is not adequate from a network operator view, since the network operator has little control over the terminal's mobility actions. IEEE is also working on 802.11f [116], an Inter AP protocol to provide fast transition between APs, concerning the necessary exchange of information securely. This standard specifies the necessary information to be exchanged between APs and defines higher layer entities to support the distributed functionalities of the system, enabling the inter-operability between APs of different vendors. It demonstrates a concern in providing a solution independent of the vender, and consequently, owner or operator managing the equipment.

On the other hand, cellular networks, (e.g., GSM and UMTS), typically use NIHO for mobility management. We also consider NIHO when the terminal aids, but does not control, the mobility process (e.g., the network can ask the terminal data, such as signal readings). NIHO is especially suitable for scenarios that include predictive handovers and load balancing.

Recently, efforts have been made inside IEEE to standardize these mobility mechanisms for media independency, the IEEE 802.21 [40] which is known as the Media Independent Handovers (MIH). MIH aims at optimizing and facilitating handovers between networks, through the usage of mechanisms providing link layer control and heterogeneous network information to upper layers. The standard defines a convergence layer, the Media Independent Function (MIHF), which abstracts the link layers to higher-level entities, the MIH-users. Link layer entities correspond to modules regarding the driver functionalities which are expected to report link discovery, association and parameters report (e.g.: report of signal level). The MIH proposal defines a high-level service access point (SAP), which allows high-level entities to interact with the MIHF, which in turn translate the interaction to the respective link layer technology, using media dependent SAPs. The standard defines three services within the MIHF. The Media Independent Command Service (MICS) supplies command primitives for media control and management. The Media Independent Event Service (MIES) allows collection and filtering of link layer events. Finally, the Media Independent Information Service (MIIS) gathers information about surrounding networks, in the form of information elements. These services not only affect local link layers, but can also be conveyed to remote entities through the 802.21 signalling protocol. These MIH services allow for information interchange and control of surrounding entities, thus enhancing the handover decision. It is worth noting that the focus of 802.21 is to supply mechanisms to enhance the handover, but the handover policy and decision making is outside the scope of the standard.

A trend has evolved leading to the division of the mobility architecture into two stages, concerning Local Mobility Protocols (LMP) to handle terminals at a local level with a higher granularity knowledge of the terminal’s status and independently of the Global Mobility Protocols (GMP), which manage the terminals at a higher mobility level and between different Local Mobility Domains (LMDs). With this hierarchy, the localized protocol can deal with more metrics regarding the terminal's network interfaces status and users' preferences, thus providing better chances for load balancing, flow optimization regarding QoS requirements and inter-technology management in order to achieve the perfect balance for "the right interface for the chosen service". Signalling across different Local Domains only occurs when concerning the GMP, and thus the signalling outside the corresponding domains is reduced. Although these two stages of protocols can simultaneously co-exist, their interaction is not mandatory and terminals can use only GMP and be always regulated on a global level. Multi-homing is thus also managed at two levels.

Mobile IPv4 [60] and Mobile IPv6 (MIPv6) [61] are the current IETF standards to provide global mobility management and to enable mobile nodes (MN) to roam across different networks, maintaining its global reachability. The MN can thus change its location and addresses while maintain the ongoing sessions. To assure this, the terminals use a specific address called Home Address (HoA), which is used in every connection (see Figure 12). The Care of Address (CoA) represents the mobile terminal's new location which advertises it to the correspondent node and to a mobility manager (Home Agent) in its home network. Even if MIPv6 potentially enables mobile Internet users to be always accessible regardless of the specific access network technology, increasing multimedia demands highlighted MIPv6 timing shortcomings. Real time audio/video applications underlined the need to have in place mechanisms minimizing the large handover latency and service degradation (eg. packet loss) usually associated with MIPv6.

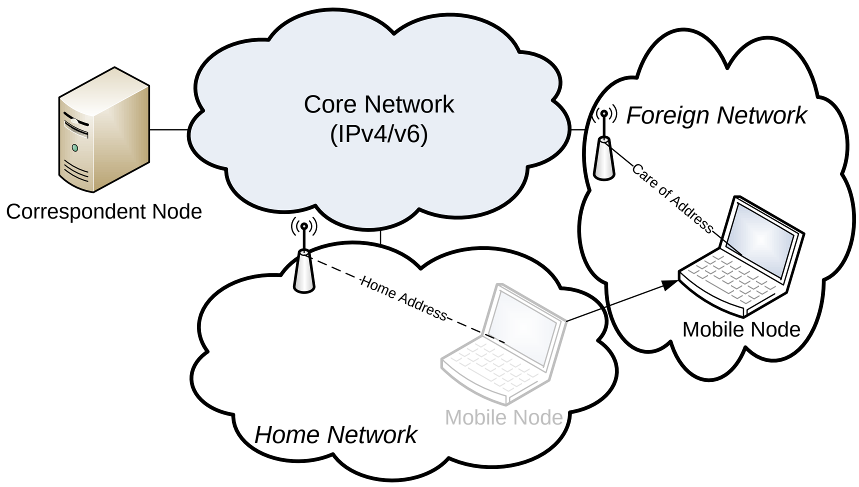

Figure 12 – Mobile IP operation

FMIPv6 [115] was introduced to enhance the handover latency of Mobile IPv6 procedures, by allowing a MN to configure a CoA prior to moving and connect at the new network. This allows the MN to use the new CoA as soon as it is attached to the new network. FMIPv6 provides a bidirectional tunnel between the new and the old access router which tries to eliminate latency while the Binding Update procedure is being performed. Compared to MIPv6, the FMIPv6 protocol claims to be more efficient by eliminating IPv6 configuration by means of Router Discovery, Address Configuration and Duplicate Address Detection

In [62] different localized mobility management schemes, such as Cellular IP [64] and Hierarchical Mobile IP [65], offering fast and seamless local mobility are discussed and compared. This comparison is purely based on the evaluation of local mobility schemes taking into account handover latency, packet loss and scalability issues without integration concerns.

Cellular IPv6 (CIPv6) [114] tries to combine the efficiency and scalability of IP with some of cellular networks' features such as seamless handoff support and passive connectivity. The efficiency is achieved by creating a hierarchical approach, through the usage of a Gateway router which connects the terminals' network to the Internet as well as a set of nodes that are responsible for routing packets to terminals. CIPv6 was designed taking into account the features of IPv6, such as the extension headers for control information, usage of IPv6 stateless address auto-configuration for CoA configuration, and authentication transactions based on the authentication headers. CIPv6 forwards packets on a per host basis. The terminal keeps a CoA within the Access Network (AN). To install routes, the terminal sends packets on the default route towards the gateway. Each node caches a binding between the CoA and the link on which the packet arrived. Downstream packets follow the reserve path. When the terminals move, a route update is sent to the gateway in order to install new entries and refresh existing ones.

Like CIPv6, HMIPv6 [113] aims at deploying a localized management of the handoffs in order to reduce the amount of signalling in the wireless network; this also increases the efficiency of MIPv6 in terms of handoff speed. Such purpose is achieved by running a local version of Mobile IP within the AN. The local mobility agent is the 'mobility anchor point' (MAP). In basic HMIP, the MH's CoA is the MAP's subnet prefix plus the MH's interface identifier (obtained via stateless auto-configuration). The MN registers unique virtual CoAs (VCoA) from the highest MAP to HA and Correspondent Nodes (CNs) outside the hierarchy. This binding does not change when MN moves inside the hierarchy. Also, the existence of the hierarchy is invisible to the correspondent host outside the hierarchy. The MN has a unique VCoA from every MAP in the path from the root of the hierarchy to the lowest MAP. In addition to the VCoA, the MN also has a physical CoA (PCoA) which it uses when communicating with hosts in the same hierarchy.

IETF is evolving to local mobility approaches, where the NetLMM working group [38] aims at relocating functionalities for mobility management from the MN to the network. In NetLMM the network learns through standard terminal operation (such as router- and neighbour discovery or by means of link-layer support) about a terminal's movement and coordinates routing state update without any mobility specific support from the terminal. NetLMM offers a hierarchical mobility approach for mobile terminal management. The objective of NetLMM is to manage the terminal's mobility at a hierarchical level. The network acquires terminal specific information regarding location and movement and coordinates routing state update with special requirements on the terminal side. Overall, one of the biggest objectives is to reduce the terminal's complexity requirements, which is usually highly demanding when concerning other proposals, sometimes not on a *"per-se"* basis, but when considering the cumulative deployment of software layers to support a complete bouquet of functionalities. Inside a LMD mobility is performed without any management support, but when terminals change from one LMD to another, they need to signal location updates to a global mobility anchor point. The LMA is the router which defines the boundary between the NetLMM domain and the core network. If a global mobility scheme is to be used, it is then the boundary between the GMD and the LMD. The MAG is the AR which is in charge of the MN. Intermediate routers are NetLMM unaware, which considerably reduces the signalling in the LMD and avoids the extensive use of resources (routing tables) in the intermediate nodes.

Proxy Mobile IPv6 (PMIP) [46] is another proposal for localized mobility protocol, which uses Mobile IPv6 concepts when possible. Like NetLMM, it introduces a local entity named Proxy Mobile Agent, which is located at the edge of the visited LMD. MNs are differentiated by a network address identifier (NAI), with an associated set of information stored in the network, such as a profile containing the home prefix. In this scenario, the MN configures its HoA on the network interface, even when roaming across foreign networks, transforming the visited LMD into a single link, from the node's point of view. Binding Updates are sent to the MN's Home Agent, informing it that the current CoA of the registered MN is the PMA's address. These procedures also lead to the establishment of tunnels between HA and PMA.

3.3 QoS

The support of Quality of service (QoS) running on top of the base IETF architectures, Integrated Services (IntServ) [109] [110] and Differentiated Services (Diffserv) [111][112], aims at being capable of controlling the following parameters: bandwidth, delay, jitter, packet loss and service availability. The combination of both strategies is still indicated as the success key for QoS Provisioning, with IntServ in the Access Network and DiffServ in the core.

As the world evolves towards high mobility scenarios, such issues are being dealt within several European projects and consortiums. Diffserv has the main disadvantage of not guaranteeing *per-se* the desired service accommodation in the whole time duration of a session. With an adequate strategy, it may be suitable for the usage on high mobility scenarios. Assuring QoS while a terminal is performing a handover is an achievement which demands some structural requirements on the network side, by introducing new software modules and signalling. There are also a large number of solutions for QoS and mobility, with some of them, [123] and [124], also integrating A4C functionalities through the usage of the combination of the mobility signalling with ReSource reserVation Protocol (RSVP) signalling. RSVP integration with MPLS is also a common approach. Given the RSVP's stateful nature, its usage on high mobility scenarios is however, questionable, as reservations must be performed on all nodes along the path. Some technologies may not respond well to this requirement as they may introduce high delays when performing a reservation. However, even in very optimistic conditions, a request/response mechanism is always necessary; with a high index of mobility, the signalling to guarantee QoS may become unbearable.

Next Steps In Signalling is a solution which aims at providing signalling across various network environments. It should be applicable in a wide range of scenarios, and at the same time be simple in implementation complexity in NSIS Entities. NSIS proposes a two-layer protocol model: a lower NTLP (NSIS Transport Layer Protocol) and a higher NSLP (NSIS Signalling Layer Protocol) layer. NSLP is composed of GIST (General

Internet Signalling Transport), Transport layers and security layers. GIST is soft state protocol that creates and maintains two states on a Per-flow message routing state basis. It uses message association state (bi-directional) to manage per-peer state associated with connection mode messaging to a peer (next peer address, protocol and port numbers, internal protocol configuration and state information). In NSIS the flow identification is achieved at the GIST layer in Message Routing Information object type which includes source and destination IP address.

IEEE is also expressing QoS concerns in their proposals, namely within 802.11e amendment for QoS Provisioning to WiFi technologies. The first standards concerning 802.16 already incorporated several QoS classes; Unsolicited Grant Service (UGS) supported real-time T1/E1 services and Constant Bit Rate (CBR) traffic. Real Time Polling Service (rtPS) supported real-time Variable Bit Rate (VBR) traffic. The third service, non-Real Time Polling Service (nrtPS) is used to carry non-real-time traffic. Best Effort (BE) traffic was also supported. The addition of an Extended Real Time service for 802.16e amendment is also a clear example of the workgroup's concern to provide sets of services with different requirements.

IEEE 802.17 [127], is a standard designed for the optimized transport of data traffic over fiber rings. All traffic on the ring is assigned a Class of Service (CoS) and the standard specifies three classes. Class A (or High) traffic is a pure CIR (Committed Information Rate) and is designed to support applications requiring low latency and jitter, such as voice and video. Class B (or Medium) traffic is a mix of both a CIR and an EIR (Excess Information Rate - which is subject to fairness queuing). Class C (or Low) is best effort traffic, utilizing whatever bandwidth is available. This is primarily used to support Internet access traffic.

One of the problems to be addressed is the delay penalty bonded with preparation of the network, which is dealt by many solutions by anticipating the movement of the terminals, thus assuring a proper pre-allocation of resources, so that when the terminal reaches the destination network everything is already set.

The operator's interest on provisioning multimedia contents is increasing and the possibility to provide multimedia services (through Session Initiation Protocol (SIP) [125] for heterogeneous mobile users while maintaining QoS requirements is nowadays a strong topic of Research and Development companies. Using a Diffserv architecture for QoS support, [126] introduces the possibility of integration of Fast Mobile IPv6 network layer protocol with SIP to reduce latencies and ensure QoS for the multimedia service. Before handoff, the terminal registers itself with the SIP server in the new domain and issues a re-invite message to the correspondent node through the SIP proxy. To support this, the proxy must interact with the QoS mechanism which the network is using. All these approaches make use of a SIP proxy to mediate the session and ensure the QoS needed for that multimedia session.

In next generation networks, fast mobility has to be considered along with QoS and authentication profiles. In [67], an enhanced fast handover stack was designed and implemented as an extension to MIPv6, exploiting the Fast Handover (FHO) basic ideas, while supporting QoS. [68] presents an integration effort and an evaluation of a 4G Network with a solution designed inside the IST-Daidalos project [69] framework, which bases the work on a modified approach of FHO with intrinsic QoS support.

There are, in the literature, several references about integration of mobility and QoS mechanisms. The approaches in [70][71][72] address seamless mobility on heterogeneous operator networks with QoS support, making use of DiffServ architectures with a central entity that performs access control and resource reservation.

As mentioned in the contributions section (Section 1.5), some work was also performed in order to extend the primitives of IEEE 802.21 to support QoS reservations, which enable the communication of decision entities with the receiving entities at the technologies. With such capabilities it can be assured that a session with certain QoS requirements can be handed off while keeping the same requirements.

3.4 Self-Organization Mechanisms

With the increasing of users' mobility and of connectivity desire, Ad-Hoc networks are appearing as a research topic of special interest, since they aim at providing multi-hop wireless connectivity. For such networks, collaboration plays a fundamental role, which allows the access to the networks services or simple establishment of local networks "on-the-fly". This collaborative behaviour can be promoted, as other sociological communitarian initiatives such as the Wikipedia, thus urging from the individual voluntarism to motivate the proliferation of knowledge, which in the Ad-Hoc particular case would represent the spreading of connectivity through the individual intentions of helping others to reach information more easily by providing connectivity. When considering Ad-Hoc networks as an extension to operators' networks, more efficient incentives with charging and rewarding mechanisms may provide better reliability and guarantees.

The concept of Mobile Ad-Hoc Networks (MANET), which include spontaneous grouping of nodes using wireless technologies and collaborating in order to provide communication facilities, gives an alternate path towards the provisioning of full connectivity requirements. The nodes in MANETs are typically PDAs, laptops or even sensors (with limited battery, reduced processing and wireless capabilities), sharing each other communication facilities in order to achieve overall system connectivity. One node by itself, with such limited characteristics, is not capable of a large communication range. When nodes collaborate helping each other in forwarding information from source to destination, the total value of the network is much higher than the sum of the communication span of each node. For such spontaneous networks to operate, address configuration mechanisms and routing protocols are the base mechanisms that need to be in place.

Bringing ad-hoc networks into a 4G scenario [89] implies interconnecting them with the infrastructure network and supporting basic mechanisms. These mechanisms ensure the creation of such a spontaneous network as a valid extension of the overall operator architecture. Thus, it is essential to evaluate performance on major functions: auto-configuration (including gateway awareness), routing, QoS and charging. Although not all of these functions are necessary in traditional ad-hoc networks, this basic set of mechanisms must exist for the operators to supply existing services (e.g. voice).

In order to effectively communicate in a given network, nodes must have valid and unique identifiers inside the network prefix they belong to. At physical and MAC layers, the wireless card must associate with the network, after which, at network layer, a routable IP address must be obtained. Although the infrastructure network already supports functionalities such as DHCPv6 [90], a node entering the ad-hoc network usually has several nodes around, and probably several independent networks to use, and needs to choose one of them (either by traffic or cost considerations). Several mechanisms have been proposed to the acquisition of valid addresses. In Perkins [91] auto-configuration proposal, nodes simply choose a random address and perform duplicate address detection based on a given network prefix. Wakikawa et al [92] propose a method to propagate the network prefix inside the network by means of an Internet Gateway Discovery process similar to the router discovery process of IPv6, and include the integration of MANET routing protocols with Mobile IPv6. Jelger et al [93] propose a method where the gateway providing connectivity to the Internet periodically broadcasts a message (GW_INFO). This message is then forwarded by all nodes in the ad-hoc network, with the support for multiple gateways in the same ad-hoc and the ability to choose one of them based on specific metrics (e.g. number of hops towards the gateway).

The routing protocol is the element responsible for determining the best route from a source to a given destination. Since the topology is expected to change during the sessions' lifetime, the routing protocol must react and update routes between end-points. These routing protocols should be highly dynamic and robust. Ad-hoc routing protocols are often classified regarding its method of finding and maintaining routes, namely: proactive, reactive or hybrid. Some of the most popular solutions providing routing in ad-hoc networks are AODV [86], OLSR [87] and DSR [88]. OLSR is a proactive protocol while AODV and DSR are reactive. The first keeps a Multi Point Relay (MPR) graph in the network, which is responsible for optimizing the routing messages flooding process. OLSR seems to be adequate to networks with high concentration of nodes, although its overhead increases directly with the number of nodes. AODV and DSR calculate routes on-demand and usually deliver better performance, especially in networks with stalled nodes. Overhead is not directly dependent on the number of nodes, making it more suitable to large scenarios where nodes have power limitations.

Streaming services, such as IP Television, require network conditions to be stable with low jitter and delay. Because consumption of these services is based on membership rules, and the same content is distributed to a large number of clients, multicast is an important method to consider. Multicast routing is able to deliver the same content to multiple clients upon proper service subscription. The cost to the network is some additional signalling required to maintain the distribution tree and client subscriptions. MAODV [94] and MOLSR [95] are, respectively, the multicast versions of AODV and OLSR. ODMRP [96] and ADMR [97] are multicast ad-hoc routing proposals that reduce the overhead of maintenance of the multicast tree in the ad-hoc network. However, none of these proposals is directly adapted to integration with an external infrastructure.

Network infrastructure is expensive and has very well known limitations in terms of bandwidth. As network load increases, QoS traffic parameters like delay, jitter or packet loss also increase, degrading network conditions. In order to provide the best possible service, while maximising profit, operators have a strict control over the QoS characteristics of their networks and keep their backhauls over provisioned.

When integrating an ad-hoc network with an existing commercial network, operators expect to apply the same QoS levels to users. Traditional hotspots can perform this easily by a set of rules at the access point. However, since the ad-hoc stub is a distributed and unstable environment, QoS has to be sustained in a distributed manner. Several protocols have already been proposed to support the delivery of adaptive services in mobile ad-hoc networks [99]-[102]. INSIGNA [99], one of the best known, uses a soft state resource management mechanism to enhance network usage. Packets transport an extra field for QoS information, which is used as an in-band signalling. The protocol supports Best Effort services and services requiring reservation with per-flow QoS support. QOLSR [100] is a QoS routing protocol defined to enhance OLSR. Each node gathers information related to QoS parameters such as available bandwidth, delay, jitter or loss probability. These parameters are reported to OLSR, based upon which, the MPRs create or change routes. However QOLSR is not able to limit the traffic in the network. SWAN [101] uses distributed control algorithms to handle two types of traffic, Best Effort and Real Time, through shaping. It performs rate control for Best Effort traffic, in which, traffic marked with less priority, can occupy up to the maximum bandwidth left by the Real Time traffic usage. The bandwidth usage by the Best Effort traffic raises according to an Additive Increase, Multiplicative Decrease (AIMD) rate control algorithm. SWAN uses source based regulation algorithms in which congested nodes send messages informing intermediate nodes to wait for a random amount of time before trying to re-establish connectivity. Dynamic regulation is also performed to deal with mobility and false admission issues. In [102] an extension of SWAN was proposed to make it interoperable with the infrastructure and to support four classes of traffic.

Operators need to be able to profit from the development of the network and services. Since infrastructure networks are driven by operator business models, it is mandatory to support for charging the users. The multi-hop and distributed nature (and dynamics) of ad-hoc networks requires the existence of distributed trust mechanisms, able to provide adequate information for charging and traffic authorization. Most important, these mechanisms need to be compatible and integrated with existing network authorization and charging architectures. Furthermore, ad-hoc networks also require incentives for users to participate in the forwarding process; otherwise, nodes may not forward others traffic without any benefit. Such incentives can be provided in many forms, like, for example, credit or service discounts.

Solutions like [103]-[108] envision scenarios where ad-hoc networks are integrated with an infrastructure supporting authentication, authorization and charging mechanisms. SPRITE [103] assumes that nodes have enough storage capacity to store traffic proofs. These proofs are later traded at a bank for credit when the node is connected to a high bandwidth medium. Salem [104] envisions ad-hoc extended cellular networks, where base stations are capable of charging, rewarding and enforcing profile policies on packets generated. In order to achieve this level of control, it proposes all traffic to cross the base station, independently of its origin and destination. SCP [105]-[107] proposes the creation of a distributed mechanism, actively marking packets with a proof that is updated at each forwarding node and then reported to the network operator, with intrinsic class differentiation. The proofs are built and updated using a defined set of rules and supported by cryptographic signing and verification primitives. PACP [108] improves many of SCP deficiencies (overhead, variable packet size) by encoding the route in a polynomial included in the packets, and securely updated at every node. Upon reception of the charging information on the infrastructure network, the appropriate charging and rewarding actions may be applied. These actions can take in consideration many individual parameters, like individual user profile, service description, QoS parameters, route length, time frame or data amount. Also, PACP supports distributed access control, allowing the operator to control which flows are allowed between each nodes, without sacrificing routing.

Most of these features create a complex stack of protocols which introduce a considerable performance penalty. The addition of such mechanisms must be well weighted according to the needs and requirements of the users/operators. For ad-hoc networks "on-the-fly", which may be intended to provide a fast local connection between a set of users, auto-configuration and routing features may suffice, while on the other hand, when accessing the operator's services, charging and rewarding are of special interest.

When considering the distribution of multimedia contents, a multicast solution is of outmost importance. However, 802.11b/g may not be the best choice to deliver such types of contents through ad-hoc multi-hop, as bandwidth becomes an issue at very small hop counts. Its usage can nevertheless be of special interest when considering a role of uplink channel for multicast contents to be delivered through broadcast technologies. Multicast Join messages and other signalling with no specific delay/jitter constraints can be used via these networks.

3.5 Proposed Architecture Overview

This subsection describes the envisioned architecture which aims to integrate the previously mentioned concepts. It consists on the formulation of the architecture fundamentals for the integration and support of broadcast technologies.

As mentioned before, when integrating unidirectional technologies within a heterogeneous environment, it is important to not condition its usage over the absence of a return channel. The true nature of broadcast systems should also be kept and supported by the network, which implies that the network is able to interpret the technology as unidirectional by gathering information from the terminal, which is responsible to provide the network with its available interfaces. To better understand such integration, Figure 13 introduces such an operator network's architecture.

Figure 13 shows an administrative domain, represented by the operator's core network and connected to two Local Mobility Domains (LMDs). Each LMD connects with one or multiple L2 Domains. These domains include switched technologies which can be concatenated. The Layer 3 connection between the L2 Domain and the user is performed through Mobility Access Gateways (MAGs); each Layer 2 connection is performed through an AP or a base station (in the figure it is also mentioned, as example DVB, WiMAX and UMTS). Regarding mobility and connectivity, each LMD is managed by a LMA. The most important feature resulting from this hierarchy is that the terminal is not required to reconfigure itself (e.g. obtain a new IP address) each time a handover occurs; consequently, only Layer 2 mobility schemes need to be in place inside this LMD, and independent local mobility schemes are supported in different LMDs (for more details, please refer to [62]). Each LMD may contain different L2 Domains with different technologies; therefore, QoS support over heterogeneous technologies needs to be in place, with an emphasis on the provision of QoS-enabled mobility mechanisms supporting seamless mobility across the heterogeneous links. Mobility support and its integration with QoS is handled by a mobility manager both in the MNs and MAGs.

Local mobility can, however, become an ambiguous term with the introduction of a hierarchy level related to the operator's administrative domain. It is thus required to underline the existence of several levels of mobility: 1) inter-administrative domains, where mobility is performed between two operator's domains, or between different domains of the same operator; 2) inter-LMDs, where mobility is performed inside the same administrative domain, but between two different LMAs; 3) intra-LMDs, when a terminal performs a handover between two MAGs connected to the same LMA. Within Intra LMDs' handovers, we can also consider the sub-case of intra-MAG, which is related to an handover between two base stations connected to the same MAG; and inter-MAG. To support all these mobility levels, a Mobility Manager is required: at the LMA for mobility support when considering inter-MAG handovers; and at the MAGs for mobility support when considering intra-MAG handovers.

QoS control is performed in a hierarchical framework, separating end-to-end QoS control at layer 3 from link-local QoS control at layer 2. Layer 3 QoS inside the LMDs is managed by Zone QoS-Brokers (ZQoSB). The ZQoSB can be collocated with the LMA, as the local mobility management between MAGs is performed at that entity by Local Mobility Manager. Upon mobility, the mobility manager at the LMA and the ZQoSB must exchange information to perform a handover, which considers the best target location of handoff with the capability to provide the desired QoS guarantees. The ZQoSB performs a wide range of functions such as per-flow admission control and resource management, handover authorization based on user profiles and available resources, and also participates with resource management optimization.

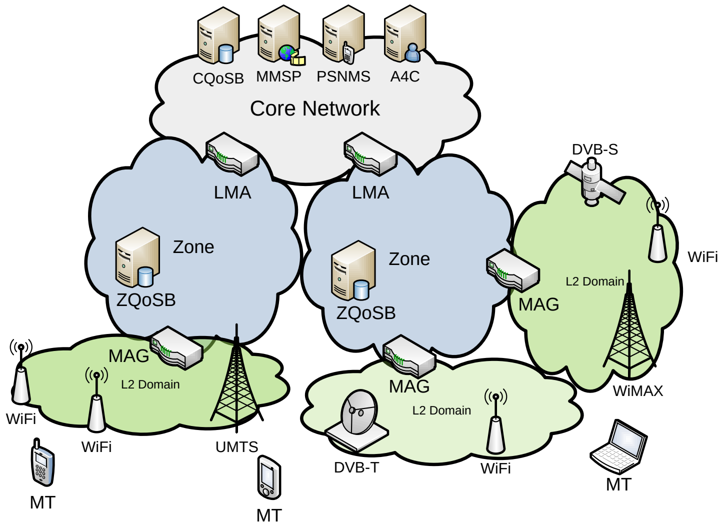

Figure 13 – Overall Architecture

Core QoS Brokers (CQoSB) manage the Core Network (CN) and inter-domain functions in a per-aggregate fashion. Using a hierarchy of QoS Brokers, it is then possible to handle scalability problems.

A Multimedia Service Proxy is also in place supporting multimedia services with QoS, and an A4C server manages user accounts and interfaces with QoS modules for authorization purposes as well as policies. User identities, service descriptions, contracts between users and service providers, roles and permissions, are examples of information stored in the A4C server database. A Policy-based Network Management System (PBNMS) is also present to contain the policies to manage the QoS network elements.

Both MNs and the access routers (AR), here denoted as MAGs-AR, contain elements that perform the enforcement of the QoS in the network and trigger the QoS process for admission control and resource reservation: QoS Client (QoSC) and QoS Manager (QoSM), respectively in the MNs and MAGs-AR. The specific characteristics and reservation handling of each technology are executed by technologies' specific drivers. The drivers deal with the specific characteristics of each radio technology by communicating APIs of the specific technology hardware, and providing a common interface to other modules. A proposal which already deals with specific technologies requirements and specificities is the IEEE 802.21 draft, which introduces a L2.5 layer providing primitives for the inter-communication of L3 entities with the L2 ones. It is a solution which can be responsible for the advertisement to upper layers of the discovery of new PoAs, signal measurements, QoS parameters, connection availability, etc. Such metrics will reach the MAG and will be stored there to be used by the Mobility Manager and the QoSM. The Drivers are also responsible for the reservations on their respective equipment, when such instruction is provided from the responsible network entities (LMA or MAG).

The MN contains a mobility decision module, the Intelligent Interface Selection (IIS), which checks the best interface and AN for each set of flows. A correspondent module in the network side, the network IIS (nwIIS), is available in the ZQoSB for resource allocation.

The integration of multi-hop Ad-Hoc networks is done between the interaction of the MAGs and of the first Ad-Hoc node, which becomes the gateway towards the operator's network. The provisioning of the services in such a network is ensured by the usage of the mechanisms referred in the previous section. The concepts explained in this section are also applicable to this case, with the exception that the gateway acts as the proxy for the network, extending the reachability by topological knowledge such like in concatenated technologies.

3.6 Summary

This Chapter presented several architectural considerations, first showing the evolution of the main mobile telecommunications' architectures which lead into the actual state of the art. The mobility section introduces several proposals and projects which deeply express mobility concerns and solutions. Also some standards and mechanisms are commented. Mobile IP is one of the most widely known protocols for global mobility support. HMIP and FMIP are natural evolutions dealing with performance enhancements. Local mobility management has recently begun to be of importance considering the dimensions of the mobility domains. With the increase of such domains, a hierarchical decision holder may be of benefit to reduce signalling, processing and instability of the network. Several QoS provisioning solutions and standards were also presented concerning the major models and also some direct support over specific technologies which were already designed taking into consideration such requests. Multi-hop extensions are presented in many cases as a chance to increase the revenue sources of the operators, taking the connectivity provisioning one or several hops beyond the direct connection to an operator's equipment. After the presentation of this state of the art, a conceptual architecture is presented, which aims at providing all the previously mentioned features assuring the provisioning of ubiquitous services over heterogeneous networks.

Chapter 4 IMPLEMENTED ARCHITECTURE

This chapter introduces the implemented architecture, by describing the details on the used mechanisms for signalling and features support. In this sense, a first section is presented, describing the main network entities and the way they cope with mobility and QoS. The mobility support section specifies the implemented modules ranging from the interaction with the technologies to the decision modules. The following section explains how the QoS is handled. To serve the necessity of explaining how broadcast technologies are supported and integrated, a small section was introduced, which mainly describes a necessary entity which is further detailed in the following chapter.

4.1 Deployed Architecture

This chapter aims at presenting an architecture which seamlessly integrates broadcast technologies. This architecture is a simplification from the one in the previous chapter, aiming at its development in a real testbed.

The scenario presented in Figure 14 consists of several access networks connected to a local network, which acts as a gateway towards a core network. This topology represents a hierarchical solution, with ARs connecting the ANs to a common local network, thus representing the functionalities of a MAG. The MAG is responsible for the local management of the entities within its AN. It is considered in this simplified architecture that each MAG supports only one technology, hence its notation contains the specific technology (DVB-AR and WiFi AR). All MAGs are connected to a proxy, the LMA, which is responsible for managing a certain network zone (or local network). The LMA, which is presented as an edge router (ER), has a central role, as it is responsible for managing mobility inside its LMD.

The represented ANs consider that a MAG is responsible for a single technology, which is not necessarily mandatory. It is simply an implementation simplification, since the PoA of the terminal can be co-located with the AR. For instance, the MAG machine can thus also become an AP. Regarding the uni-directional broadcast technologies, such an assumption also becomes much easier to perform the deployment, since usually the equipment responsible for the encapsulation of the IP/Ethernet of the specific broadcast frames, requires extra management.

Users must register and authenticate themselves in order to request for a certain service. Given the nature of the broadcasting technologies, it is considered that a single cell can provide coverage to wire areas (e.g. a city). It has been kept in mind that the user can, however, be interested in roaming beyond this physical location. In this architecture we consider that the terminal can also be connected via WiFi to an infrastructure AP, or to another computer using some ad-hoc routing protocol in order to extend the connectivity coverage. WiFi is here considered as the preferred technology for the terminal's uplink channel. One of our aims is to evaluate how WiFi in both ad-hoc and infrastructure modes fit to act as a last-resource alternative to the uni-directional link as well as its uplink.

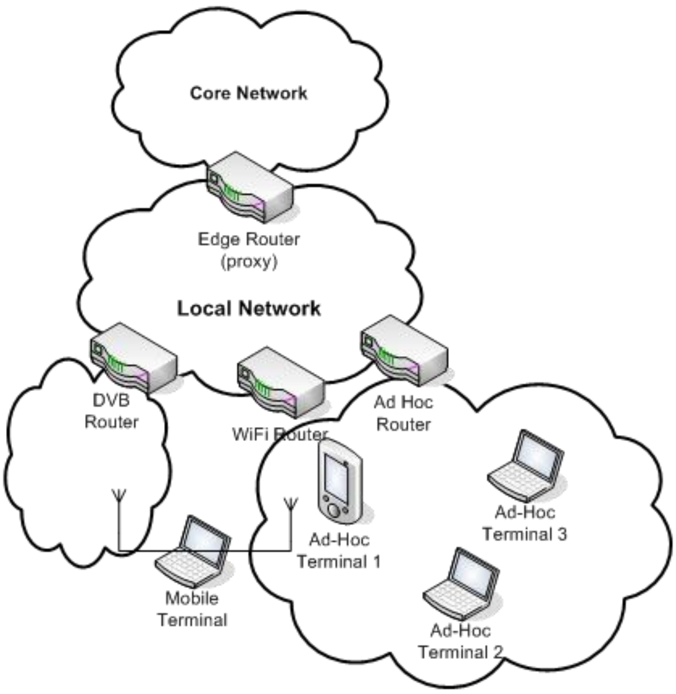

Figure 14 – Scenario overview

When the multi-homed MN presented in Figure 14 boots up, it must first configure its network interfaces: it will need to receive router advertisements in order to configure the interface's network addresses. For this purpose we propose using *radvd* [18] in the WiFi infrastructure link. The terminal then acquires the network prefix and sets up the addresses of its network interfaces. For ad-hoc connectivity, the usage of an auto-configuration protocol such as the GW_INFO [15] is required. GW_INFO is responsible for advertising the ad-hoc gateway which, in Figure 14, is collocated with the WiFi ad-hoc AR. The ER also advertises itself periodically; as soon as the node gets this advertisement, it tries to register at the LMA and request for a certain service. It does so by indicating its ID and the router which it wishes to use in order to receive such a service. This request is done via the uplink channel (in Figure 14 it is considered to be WiFi), which is chosen in the terminal. If the terminal is successfully authorized and the service request is granted, it will start receiving traffic on the desired interface. In order to accomplish this, the LMA needs to acquire knowledge of the connected MAGs (when they bootstrap, they need to be registered in the LMA).

The DVB router is in fact an AR coupled with an Encapsulator/Inserter. The functional purpose of such a device is to introduce the IP frames into the technology specific frames. DVB encapsulates IP into MPEG-2 as explained in 0. The equipment which supports this feature usually is enabled with some processing capabilities; nevertheless, this does not compromise networking features.

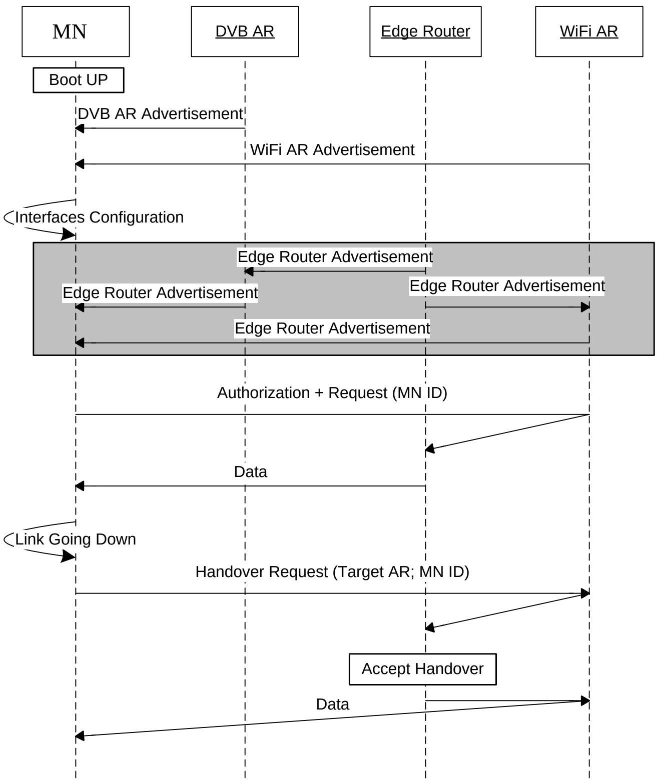

Figure 15 – Set up and Mobility Diagram

Figure 15 demonstrates the message sequence for the bootstrap and the simplified signalling for handover execution. When the terminal boots up, it starts by creating a DVB interface with a default PID, and by bringing up the WiFi interface in infrastructure mode. It receives a DVB AR and a WiFi AR advertisement in order to configure the IPv6 addresses in the respective interfaces, as well as to add routes to these networks (the route created via the DVB link is useless as it is an unidirectional link). Also, the terminal acquires the router's IP address of the interface connected to the zone. After this, the terminal receives the ER advertisement on both interfaces, since it is broadcasted periodically. As soon as the MN gets the ER's address, it tries to register itself and request a service. If this request is accepted, the terminal will start receiving the desired service.

The registration process is performed by nothing more than an IP message transporting an ID of the user, which the MM@ER verifies to be known or unknown. This ID is always performed upon handover. The used ID was the IP of the uplink interface.

Figure 15 also demonstrates a case where the signal level of a link is undergoing a certain pre-defined threshold. Suppose the terminal has requested to receive a video stream over the DVB link, and that the module which evaluates the signal strength and BER (Tzap Module), detects that one of these parameters has undergone the minimum required value (Link Going Down event in Figure 15). In this case the terminal issues a Handover Request message to the ER via the uplink channel which is in use (WiFi infrastructure), indicating that it wishes to receive such a service via another interface. This message is constituted by the target AR ID (which it has already acquired) and the terminal's own ID. These identifiers were considered to be the IP addresses of the corresponding entities. The ER then evaluates the feasibility of the handover and, if it chooses to accept the request, the terminal will receive the desired service via the requested interface. For the effect, no particular protocol was used, only the signalling described in Figure 15, as the aim of this work does not consist of a creation of a novel protocol, but rather study the impact of integrating heterogeneous networks which include unidirectional broadcast technologies.

4.2 Mobility Support

This section describes the necessary modules for the support of the mobility signalling as presented in Figure 15. Mobility management modules are present at the LMA, MAGs and MN for different purposes.

4.2.1. *Mobility Manager at the LMA*

The mobility manager at the LMA is responsible for the advertisement and for the individual management of the terminals. Each terminal must register itself at the LMA, which administrates the feasibility of handovers and service provisioning.

- Advertisement of the LMA
  - This is done using a message with a special ID and which is received on a particular port. This message is broadcasted and carries the LMA IP address.
- Users Authentication and handovers feasibility
  - Analyse the ID of the entity requesting the handover.
  - This module is expected to communicate with the QoS brokerage functionalities.

Advertisements are performed periodically, while the authentication is done upon service request (via a certain MN's network interface address). The mobility manager is prepared to receive a message from the terminal requesting to be reachable via a determined MAG. After that, it updates the LMA's routing table (using *RTNETLINK* linux libraries).

4.2.2. *Mobility Manager at the MAGs*

The purpose of this module is to advertise the MAG's ID, so that the terminal can request a handover based on its preferences when it has a technology abstraction. This is still work in progress, as the MAG should not advertise its zone's network interface address. Rather, the LMA should have this topological knowledge. The reasoning for this module's existence is related with the constraints that some broadcasting devices may possess. Typically, broadcasting equipment is not originally produced taking into account networking issues for such complex heterogeneous scenarios. In that sense, some additional software needs to be incorporated locally in the same equipment or, in case it is proprietary and cannot be changed, within a co-located machine. In the presented scenario such machine is the DVB MAG, which is co-located with the IP inserter.

4.2.3. *Mobility Manager at the MN*

This module is responsible for receiving information from the technologies' drivers. As indicated in Figure 16, the WiFi driver and TZAP are supported. TZAP acts as a driver, getting a lock on the desired channel's signal and reporting performance metrics such as link signal strength and signal error rate. These are the metrics that are evaluated. The mobility manager gets input from the user's defined preferences for the handover, which can consist of a second choice's interface. If the signal over the DVB link is beneath a certain threshold, it evaluates if the interface with highest priority in the preferences list gathers enough conditions to be used. The service information module should provide information regarding a specific service, e.g., the multicast IP address to which it should issue a *Join* message, QoS characteristics of a certain service, technologies through which the service is available. For testing purposes, currently this input is only characterized by the service's topple group and source, when considering multicast services.

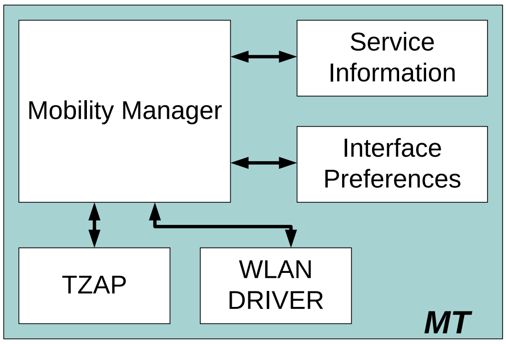

Figure 16 – MN's architecture

The service information can be static, but must consist of several mandatory parameters as well as some optional. In order to request a multicast stream, it is necessary for the terminal to know the multicast group address, which may suffice; but if the user is requesting a service which is only available via a broadcast technology such as DVB, a PID is also required to be specified. Although this results from a specificity of the technology, it is nevertheless a good example of an optional requirement, which arises from the fact that the creation of a DVB network interface is performed upon the indication of this identifier.

4.3 QoS Support

Section 3.3 introduced an overview of several protocols which were conceived to enhance the QoS by the operators. This subsection aims at describing the QoS support for the proposed mechanism in the antecedent section. From an L3 point of view, two stages are required: i) the gathering of the necessary QoS metrics from the network entities, ii) admission control upon request.

In order to achieve a better performance, the first stage is hierarchical, and is composed of the gathering of information from the specific technology devices, such as WiFi AP, WiMAX BS or Broadcast Encapsulator, and the communication of such parameters to the central QoS processing unit – the QoS Broker.

The report of the QoS metrics should be done by the devices, either on a query/response basis or when reaching thresholds', and stored at the Broker. For a higher performance index, and as explained above, when considering high populated mobility domains, it is best to store these resources at the MAGs which then communicate them to the QoS Broker. Figure 17 helps to explain this interaction. The Broadcasting equipment is responsible for encapsulating the IP frames into the technology specific transport frames (most commonly MPEG2). Uni-directional links do not get a feedback on the other edge of the link, in order to withdraw conclusions regarding metrics such as delays and jitters. These vary however according to the usage of the system, especially when considering low capacity usage levels. The bandwidth, on the other hand, is a metric which can be set according to the type of modulation in use, and then be communicated to the broker, which updates the available bandwidth according to the previous demands. Several problems regarding the implantation according to actual equipment is done in the next section, where constraints are mentioned, as well as their impact. When considering bi-directional technologies, such as WiFi, it is possible to evaluate performance metrics with more detail by introducing controllers on the technology devices. These parameters can then be reported to the MAG, which should have a database to store them and interact with the QoS Broker responsible for the respective LMD.

The Broker can be co-located with the LMA. In order to guarantee this process, several protocols can be used. It is proposed to use the 802.21, with specific extensions to support QoS as it introduces a topology-aware inter-layer mechanism, which is the ideal solution for this architecture since APs, BSs or Encapsulators/Inserters are usually L2 devices with low processing capabilities. Regarding the inter-communication between the MAGs and the LMAs, a core network protocol should be used such as DIAMETER, also with the proper extensions. Upon a session setup request, the broker analysis the feasibility of such an establishment, according to the desired metrics and the availability of resources (or quality of resources).

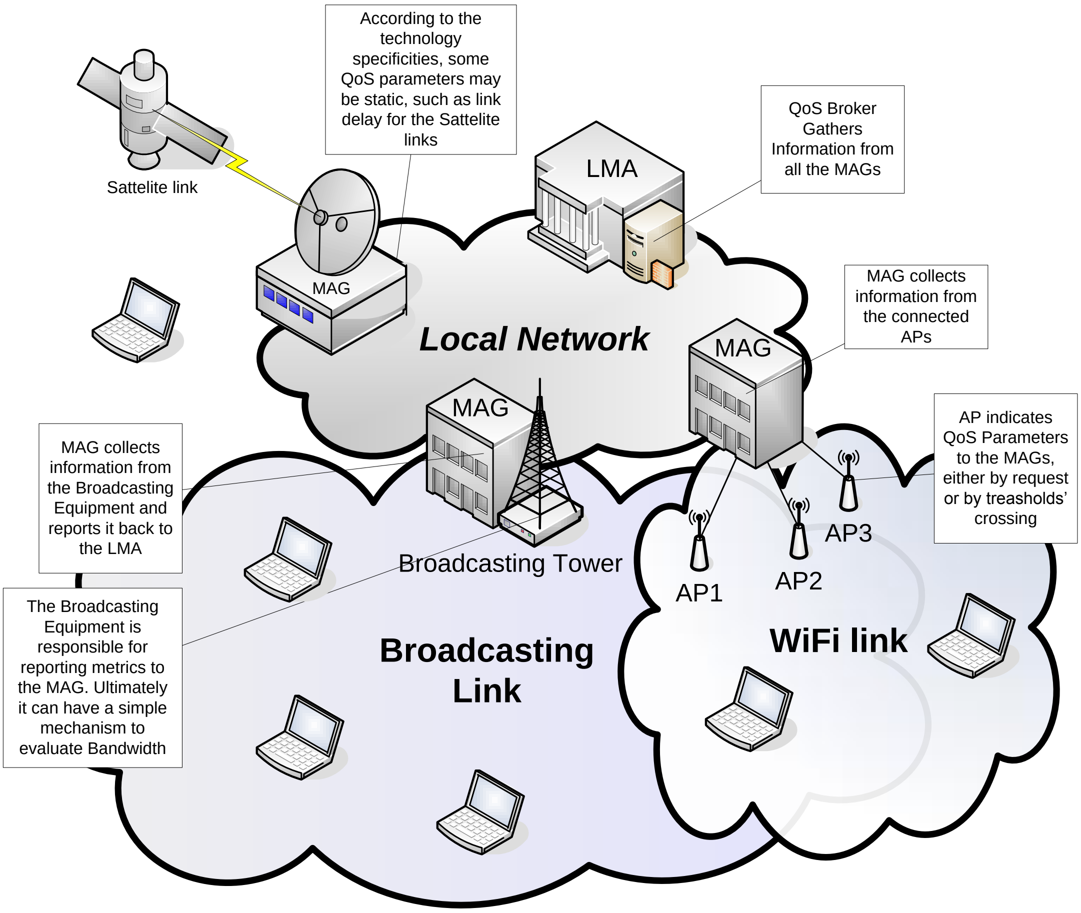

Figure 17 – Architecture with mapping of functionalities and entities

The described QoS mechanisms, regarding generic technologies' management were not implemented, and were only the focus of conceptual research. However some proposals for the extension of the necessary mechanisms have been introduced. Since this study reflects mainly in the broadcast management and the used device (as referred later on), it does not support any a query/response mechanism such as SNMP, and several difficulties have arisen. Measurements were done in order to provide a relation between delay time intervals and occupancy of the broadcast link, which can lead to the deployment of a QoS admission (acceptance/rejection) system. With the relationship between the percentage of occupancy and the variation of the delay, it is possible to set rules at the broker concerning the broadcast interfaces. Still it is necessary to configure the maximum available bandwidth at the link, which as said before, is strongly related to the channel width in usage, the chosen modulation, guard interval spacing, the usage of FEC, etc. When the equipment does not supply such data, it is only possible to set up a fixed bandwidth according to a set of parameters and assume it is not changed, which is not always true. In fact, WiMAX technologies tend to adjust these values according to power loop indications. Uni-directional technologies however, do not have a feedback, and thus do not change their modulation schemes according to the reception conditions, but rather according to bandwidth requirements.

4.4 Broadcast Support

Broadcast equipment usually requires special adaptations, as it was developed to be applied within a specific scenario, or regarding a pre-defined framework. Nevertheless, it encapsulates the IP or Ethernet frames within the broadcasting technology's restricted transport protocol. Several broadcast technologies use MPEG-2 Transport Streams as the predefined L2/L1.5 Layer. The reason why it is defined as 1.5 (in this thesis) is because, in some cases, it can also transport Layer 2 Protocols such as MAC, through the usage of an inserter. The inserter must also be capable of interconnecting with the network in order to provide some feedback: it is of extreme profit for the operator if such a device can provide it with performance metrics within a "*coherence window"* (one in which the conditions have not been considerably changed).

4.5 Summary

In this chapter a detailed functional structure of the implementation was shown, which suits as the progress of the previously explained architecture. It aims to be a proof of concept implementation focussing on assuring the integration of broadcast links to enable the heterogeneity of the technological links, in a way such that it becomes seamless to the user. With the increase of interfaces per terminal, it becomes harder to control each interface independently to choose the best one, for a certain application, given temporary and dynamic network status. The specified signalling explains how the results expressed in the following chapter could be obtained. It is of interest to underline that the resulting messages sequence here presented do not try to specify a new protocol, but rather provide a simple solution with the minimum overhead and maximum simplicity which, yet is able to provide metrics for the required conclusions. In a more consistent way, this chapter also describes the responsible entities for the support of the desired features, and how special network modules are required, as well as specific equipment, such as the broadcast interfaces introduced in the previous section.

Chapter 5 IMPLEMENTATION AND RESULTS

This chapter presents the implementation details by presenting the used equipment for the assembling of the used testbed. After the description of the network entities and their specifications, the withdrawn results are presented. The testbed description is performed by first introducing the DVB equipment mainly consisted by a generator, an inserter, a modulator and a receiver. The performance specifications of the mobile nodes and routers are then followed by the indication of the necessary specific modules used to support the DVB and Ad-Hoc components. Static performance metrics are shown in order to characterize the network in terms of required times to perform measurements or to perform horizontal handovers as well as throughput, delays and jitters. The presented mobility metrics concern the obtained handovers times when the previously described signalling is being used over heterogeneous networks, consisted of an unidirectional broadcast link and a return channel of WiFi in both structured and ad-hoc mode.

5.1 TestBed Description

This section details the used equipment for the deployment of the testbed, ranging from the description of the broadcasting equipment to the mobile nodes and access routers characterization in terms of specifications and software. The choice for the broadcast technology fell over DVB-T/H as it is mature and its equipment is easier to find as final commercial product. The integration with a return channel was made using WiFi and by using the protocol before explained for the mobility interaction between the network and the terminals.

5.1.1. DVB Equipment

The DVB equipment used for the deployment of the testbed consists of a full operator system capable of wide area's coverage. Figure 18 presents a basic DVB-T/H system, with the essential devices for a professional broadcasting set-up.

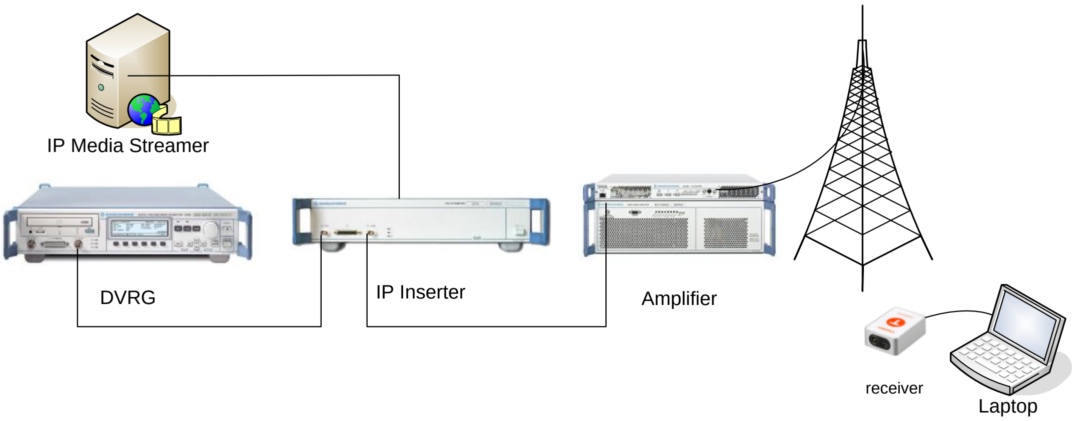

Figure 18 – Basic DVB-T/H system

The IP inserter acts as a multiplexer for the IP contents provided by an IP enabled device with an Ethernet connection. The DVB Recorder and Generator (DVRG) acts like an MPEG-2 carrousel. The multiplexed frames are then forwarded to the Amplifier, which broadcasts the signal to a DVB enabled Laptop that receives it and decapsulates the IP frames, or simply delivers to the end user the MPEG-2 contents.

5.1.1.1. DVRG

Although not a mandatory device, the DVRG can record and generate MPEG-2 contents which can be forwarded directly to the amplifier.

The DVRG can record and replay MPEG-2 TS and SDI signals. It allows seamless and endless generation of MPEG-2 TS, the integration of a HDTV stream library (optional) and data rates up to 214 Mbit/s (TS) or 270 Mbit/s (SDI). It is equipped with several TS-interfaces for input and output, (ASI, SPI, SMPTE-310M) and a hard disk memory up to 144 GByte (more than 8h at 38Mbit/sec, 1h at SDI). Also, it supports the following DVB-H features:

- DVB-H reference Stream library (DV-DVBH)
- DVB-H MPE
- Time Slicing
- Forward Error Correction
- A MPEG4, H264, WM9 (VC1) video formatsstream maker software tool allows creation of own DVB-H streams

5.1.1.2. *Inserter*

The Inserter (also known as Encapsulator) encapsulates IP data packets into the MPEG-2 TS, which are sent to the amplifier to be transmitted over the Air Interface, as described in 0. The used equipment was Rohde & Schwarz's DTV Data Inserter (R&S DIP 010), which supports MPE encapsulation [17] by receiving data traffic on the Ethernet interface and forwarding it to the modulator/amplifier. It allows the utilization of unused resources (null packets) when performing the real-time data insertion, allowing up to 15 Mbit/s.

This equipment can manage of up to 65000 IP data flows according to Bandwidth Management, Service Quality profiles. By supporting MPEG-2 insertion mode or MPEG-2 generation mode, the DVRG is unnecessary when using the last mode. An important feature to be remarked is the support of IP Multicast contents' forwarding via tunnelling.

This equipment is DVB-H ready. It supports:

- Time slicing
- Forward Error Correction
- Multiprotocol Encapsulation IPv6
- DVB-H specific PSI/SI signalling - INT table

When adding an unicast service, the destination IPv6/IPv4 address must be added to the corresponding stack table along with the interface's MAC address. Also, a PID needs to be specified. In case of a multicast address, the interface only needs to be provided with the multicast address and, as soon as the software detects packets belonging to this address, it will forward them as well as a PID for the specific service. Remote addition of such services is not possible, and would require direct interaction with the proprietary software.

The interface, depicted in Figure 19, shows the number of packets forwarded and the current activity (i.e., the bit rate that is being used to forward the services that have been preconfigured). After this configuration, the Media Router Service must be started in the Service tab (more information available in [132]).

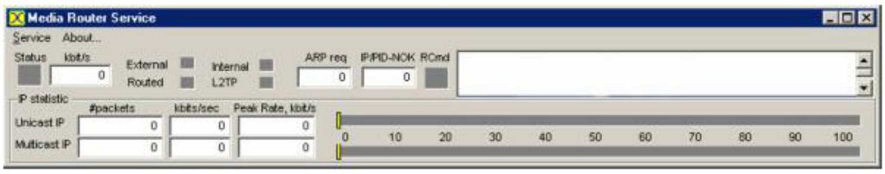

Figure 19 – DIP010 Media Router

The DTV Data Inserter (Figure 20) expects data packets prepared by the MediaRouter on a standard UDP/IP socket interface. After the reception of a packet, the DTV Data Inserter analyses it and, depending on the packet header, recognizes one of the following data types:

- MPEG-2 TS
- Packets for transparent insertion without data processing. This type is recognized if the payload starts with the byte 0x47 and its length is exactly 188 bytes.
- DataCarousel / ObjectCarousel sections for transparent insertion.
- IP packets for processing, preparation and insertion (i.e. for DVB-H).

The inserter software must be configured under the *Settings* tab and may operate in Insertion or Generation mode. This can be set under the *Basic* sub-tab, as shown in Figure 21, and for the first mode the application will act as a multiplexer. It receives MPEG2-TS on the input ASI port and inserts the IP frames on the remaining bandwidth or degrades the service that is already being provided. When working under this configuration, the device needs to get a lock on a signal provided by the ASI input port. On the other hand, it can generate MPEG2-TS frames on itself and insert directly the IP contents to be broadcasted. When using DVB-T these settings suffice.

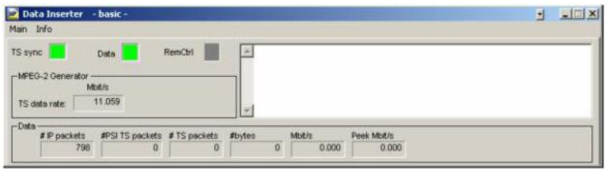

Figure 20 – Data inserter Module

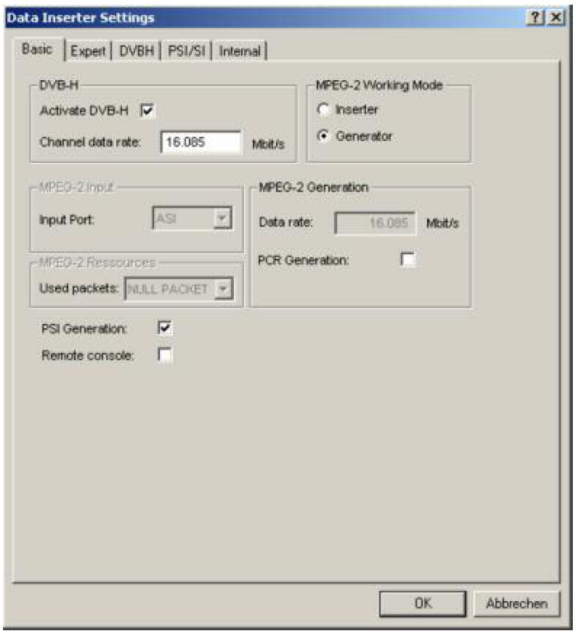

Figure 21 – Data Inserter Module Settings

By activating the DVB-H function in Figure 21, the user interface for the inserter becomes more detailed and shows the various defined PIDs in use.

The basic steps for the configuration of the Inserter must then be:

1. Configuration of the networking addresses for the Ethernet interface.
2. Configuration of the IPs to be broadcasted with specific MACs and PIDs. These parameters must be static, because the software does not allow dynamic access to change these parameters in real time.
3. Configuration of IP_1 to be broadcasted with MAC_1 using PID_1.

5.1.1.3. Modulator/Amplifier

The R&S DTV Exciter was used as the modulator, receiving base band signal from the inserter and broadcasting it to the air interface via the Rohde and Schwarz amplifier R&S SV8000, which is a BD IV/V category amplifier using an enhanced firmware to support also DVB-H besides DVB-T. Its general features comprise:

- Power classes ranging from 5W to 10kW
- VHF and UHF operation
- Advanced LDMOS technology for power amplifiers
- Remote access
- 4k rate mode support

- In-depth-Interleaver
- TPS Bits

The device was set to the frequency of 826 MHz. The R&S SFU Broadcasting Test System acted as another modulator which was used to create a second DVB-T cell. This equipment is prepared to support DVB-H testing as well. It was set at 839.25 MHz. In both cases, QPSK modulation was used with an 8MHz channel width, no FEC and a guard interval of 1/16. This equipment supports:

- Real-time and in-depth analysis of MPEG-2 TSs.
- It is enabled with a high resolution color display, controller, keys, hard disk, USB interfaces, a MPEG-2 Recorder and Generator.
- Data Broadcast Analysis.
- Self-contained unit with high resolution color display.
- MHP / Object Carousel.
- SSU / Data Carousel.
- IP over MPEG.
- Teletext, subtitle, VPS, WSS.
- Data streaming.
- DVB-H data broadcast analysis:
    - Detection and indication of DVB-H services.
    - INT monitoring and interpretation.
    - Data rate measurements.
    - Timing measurements.
    - DVB-H data de-encapsulation.
- Play and record of DVB-H TS.

The amplifier must be set with static parameters so that the receiver can be set according to the same values. The set of parameters used is detailed in 5.1.3 regarding the *Tzap* configuration list.

5.1.1.4. Receiver

The chosen receiver for the tests was the TerraTec Cinergy T2 DVB-T USB receiver connected which connects directly to MN indicated before. It was configured using the firmware available at LinuxTV [19]. The DVBNET tool was used in order to create the DVB network interface, which is responsible for the decapsulation of the IP frames from the MPEG-2 TS.

Currently only MPE is supported by the drivers.

Name	Cinergy T^2
Manufacturer	TERRATEC
Web site	www.terratec.com
Size:	74*49*23 mm
Power supply	USB
Connectivity	USB
Supported Modulations	QPSK, 16QAM, 64QAM
Carriers	2K, 4K, 8K
FEC	½, 2/3,3/4
Guard interval	1/32, 1/16, 1/8, ¼
Extras	External Antenna

Table 7 – DVB-T Receiver Detailed Information

5.1.2. Mobile Nodes and Routers

The testbed is composed by several types of PCs, ranging from the simplest laptop with a 1.2 GHz CPU and 256MB of RAM to the more powerful ER with an AMD Athlon(tm) 64 Processor 3000+ (1.8MHz) with 1GB of RAM. All PCs have plenty of storage space.

The lower limit in terms of hardware requirements was established by the ad-hoc nodes, since using PCs with (even) lower capabilities will cause too much slowness in the system altering the results. These specifications do not reflect typical, resource limited, (current) ad-hoc nodes, but are only suited to the extensive testing possible in a lab, or to yet-to-be-developed small form factor PDAs.

The software for the terminals and routers was developed on a Linux environment. Ubuntu 2.6.10 (kernel version 2.6.10) was selected as the distribution to be used in this testbed, as it provides a reasonably recent system and, at the same time, good compatibility for the software used in the tests.

All terminals (ad-hoc and DVB) are equipped with at least two network interfaces: one wireless and one wired. The wired interface is used to provide remote access during the tests and for administrative tasks. The communications performed for the tests are restrained to the wireless interface, whenever they are in ad-hoc or infrastructure mode.

The wireless interfaces used were Prism2.5 802.11b cards and were set in the WiFi channel 12. The bit-rate limitation was used to increase reliability, avoiding bit-rate changes and support of a channel with bit-rates easily handled by the mobile nodes. Channel 12 was selected for interference minimization.

Regarding the ad-hoc network topology, and since the tests were performed in the lab, the MACKILL [130] was used to create the desired (and emulated) topology

The infrastructure router was also a PC similar to the ad-hoc ones, but running none of the ad-hoc software, nor any other special software apart from the *RADVD* daemon. The MN PC consists of a Pentium M processor @ 1.73GHz with 2MB of L2 cache and 512 MB of RAM. The ER is an AMD Athlon(tm) 64 Processor 3000+ (1.8MHz) with 1GB of RAM and plenty of storage space.

5.1.3. DVB specific modules

One of the MNs has a DVB network adapter. A script was created to set up the network using *DVBNET* to create a network interface from the physical DVB device adapter. One interface is created by PID. After the configuration of this interface, with a valid MAC and IP, *TZAP* is initiated on a default channel and waits until it gets a lock on that channel's signal. This application accesses the tuner in the *frontend* of the card. It uses the list of channels constructed by the scan application. This list is obtained from a configuration file which we created with the following configurations for both the cells we used:

Tzap Configuration file:

DVB_cell_1:826000000:INVERSION_AUTO:BANDWIDTH_8_MHZ:FEC_1_2:FEC_NONE:QPSK:TRANSMISSION_MODE_8K:GUARD_INTERVAL_1_16:HIERARCHY_NONE:256:272:7

DVB_cell_2:839250000:INVERSION_AUTO:BANDWIDTH_8_MHZ:FEC_1_2:FEC_1_2 :QPSK:TRANSMISSION_MODE_8K:GUARD_INTERVAL_1_8:HIERARCHY_NONE:256 :272:7

Tzap typically gets a channel name as input, searches it in the channel configuration list, and then the appropriate parameters are used to tune the tuner into that channel. Some changes have been applied to this software in order to provide mobility between DVB cells and WiFi infrastructure. A state machine was added so that when the signal undergoes a certain threshold (or upon signal loss), measures can be taken in order to assure the delivery of the service the terminal is requesting. In this sense, when the terminal loses signal, it checks for a second option in the *Tzap* configuration list and then checks for signal on that DVB cell. If it is unsuccessful, it then requests the network to provide it with the service it was receiving on the DVB interface via another interface. The choice of an alternative interface is also an input for this module. In order to support predictive handovers, a threshold had to be set up on *Tzap*. Furthermore, the signal verification rate was changed so this module would be aware of the handover necessity as soon as it urges.

The basic steps are then:

1- Setting up the Inserter to forward packets with IP_1 as destination to the Broadcasting link with the MAC_1 and using PID_1.
2- Setting up the amplifier to broadcast on a specific channel according to the parameters on *Tzap's* configuration list.
3- Creation of a DVB network interface with a fixed IP_1, MAC_1 and PID_1.
4- Launching *Tzap* on the channel the amplifier is working.
5- Configuring the return channel according to the prioritization list fed as input.
6- Start the mobility modules on the terminal and routers.
7- Request for a service via an interface.

5.1.4. Ad-Hoc Specific Modules

Ad-hoc networks were studied according to two different setups. In a first approach, the focus was made on presenting a set of metrics reflecting the performance drawbacks of integrating a diversified set of features. An incremental study was performed to provide results for throughput, delay and jitter when introducing gateway advertisement and auto-configuration, unicast and multicast routing, QoS provisioning and charging.

The work presented on [129] withdraws conclusions based on the mapping shown in Table 8. The wireless interfaces used were Prism2.5 802.11b cards with the following configuration parameters: ad-hoc and promiscuous modes, channel 12, rate fixed to 2Mbits and RTS/CTS threshold of 1 byte.

Functionality	Protocols
Gateway Advertising and Auto-configuration	GW_INFO
Unicast Routing	AODV
Multicast Routing	MMARP
QoS	SWAN
Charging and Rewarding	PACP

Table 8 – Mapping of Ad-Hoc Functionalities and used Protocols

Given the conclusions extracted from this work, the second set-up regarded the usage of ad-hoc networks within a heterogeneous environment while reduced to a more restricted set of functionalities, gateway advertisement (with auto-configuration) and unicast routing. Furthermore, given the low throughput observed, special highlight is given on proving the concept of integrating ad-hoc networks as a return channel.

5.2 Static Performance metrics

This section presents an evaluation based on the performance metrics which do not require the interaction of the mobility modules. Such evaluation comprises the specific DVB equipment behaviour study, with the presentation of signal strength measurements, and the required times for the extraction of other performance indications such as switching times. The term static refers to the lack of interaction from the mobility protocol, albeit the results are directly influenced by the physical mobility of devices.

5.2.1. DVB Equipment Performance

5.2.1.1. Signal Strength Measurements

The goal of the results bellow presented is to provide measurement values to calibrate the signal strength value given by the receiver and find its sensitivity. To do so, the DVB signal level is varied at the Amplifier, and the signal strength value and IP error rate read at the terminal are registered. The presented measures were reported by *Tzap*. The sensitivity of the device is characterized by the needed power in order to reduce the Frame Error Rate (FER) to acceptable levels, 0.01%. As shown in Figure 22, -90 dBm is a good value to use as reference, when using QPSK, and -80dBm when using the 16QAM modulation scheme.

These values are concordant with measurements already performed in other studies like [58], with slight differences which are due to the outcome of the positioning of the antenna in relation to the receiver.

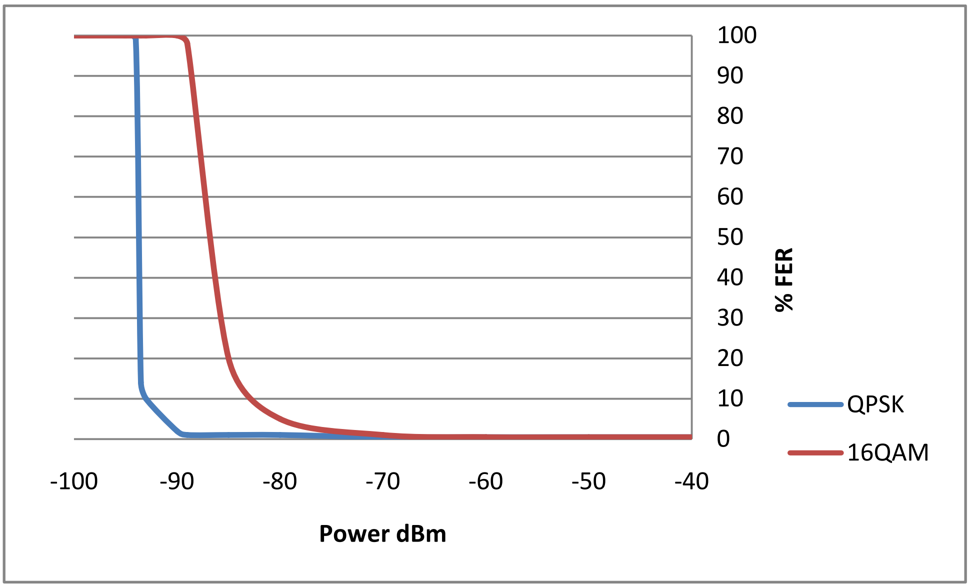

Figure 22 – Sensitivity with QPSK and 16 QAM

5.2.1.2. *Signal Strength Measurements Time*

To calculate the time necessary to perform a signal strength measurement, the DVB receiver was first locked to a DVB channel. The performance tool asked for a signal strength measurement and computed the time to process the *ioctrl* function. The average time to perform a signal strength measurement is of 15 µs. This time is small enough to ensure low losses on the reception of a certain service. When considering DVB-H technologies, this time is very low and appropriate to fit within an off-time, when considering the time-slicing feature in which the terminal is not receiving any frames. The least time it takes to perform measuring, the less battery is consumed.

5.2.1.3. SNR Measurements Time

The DVB receiver is locked on a DVB channel. The performance tool asks for a SNR measurement and computes the time to process the *ioctrl* function. The average time to perform a SNR measurement was measured as 5 µs. This value is sufficiently small. Problems arise on the necessity to scan the whole spectrum. For this case, the scan can take several minutes, depending on the capabilities of the receiver, specially the frequency range it can analyse. For the DVB receiver in use, the necessary time was always larger than 10 minutes. In this sense, it is desirable to use, for instance, a well known channel to advertise other existing channels, along with their configuration parameters, in order to save time acquiring the right bearer.

5.2.1.4. Switching Time (Horizontal Handover)

To calculate the time to switch on another channel, 2 modulators were set transmitting on 2 different frequencies. The DVB receiver was locked with one of these DVB channels. The performance tool asked for a channel switch to the second channel and computes the time to process the *ioctrl* function. We consider that the switching is done when we have a valid signal strength measurement. The average time to perform a switch and a measurement varied between 255 and 309 ms with a mean value of 280 ms.

To calculate the time to switch on another channel and start receiving again IP packets, the 2 modulators were, again, set on 2 different frequencies. The inserter is configured in DVB-T and the IP packet generator sends small IP packets at high data rate. The DVB receiver gets a lock signal on one of the DVB channels. The performance tool asks for a channel switch to the second channel. Ethereal was used to get the number and time of IP packet loss. The average time to perform a data switch varies between 130 and 140 ms.

To calculate the time to switch on another channel, perform signal strength and a C/N measurement, switch back to the current cell and start receiving again IP packets, the same procedure was followed. The 2 modulators transmit on 2 different frequencies but the second modulator is not sending data (not connected to the inserter). The inserter is configured in DVB-T and the IP packet generator sends small IP packets at high data rate.

The DVB receiver is locked on a DVB channel. The performance tool asks for a channel switch, a signal strength measurement, a SNR measurement and a channel switch back to the original channel. Ethereal is used to get the number and time of IP packet loss. The average time to perform a measurement on another channel varies from 510 ms to 617 ms.

5.2.2. DVB integration performance

The Round Trip Time (RTT) observed with DVB and WiFi set up as infrastructure for the uplink channel varies according to the bandwidth. This happens due to the buffers at the Inserter. This device only inserts the IP packets into the MPEG-2 TSs when the stack buffer reaches a certain degree of occupation. Table 9 shows the delay for the DVB link according to the bit rate in use. This delay was obtained after synchronizing both machines; the ER which is providing the service, and the MN with the DVB receiver. In order to synchronize the computers, each one was provided with an Ethernet card and special care was taken so that they both were at the same hop counts from the time server. Also for the conditions indicated above, it was shown that the maximum throughput was of 4216 Kbps.

Bit Rate (Kbps)	Delay (ms)	Jitter (ms)	Loss (%)
4216	23	1	0.1
3938	26	1	0
1939	27	2	0
100	40	4	0

Table 9 - DVB link delay

Delay and jitter increase with the decrease of the bit rate. Their variance also increases, as the system behaves according to bursts. For a 64B/s flow, the first packets would suffer a delay up to 2 seconds, but the latest packets would have a much reduced delay to complete the burst, as they would stay less time in the buffers of the Inserter.

For the remaining tests, 4Mbps of traffic flowing in the DVB link are considered, in order to meet the best system performance.

5.2.3. *WiFi Link*

In this section, some performance metrics are presented which are of interest to the tests regarding mobility procedures. As referred before, [129] walks through a wide range of incremental evaluations which are of special relevance especially to conclude that the ad-hoc wireless access is problematic due to a large variety of conditionings. The same study also concludes that the throughput rapidly decreases when increasing the hop count. A hop count of 3 still seems to offer a good compromise in terms of throughput, delay and overall overhead introduced by the multiple mechanisms. The incremental addition of such mechanisms lead to the conclusion that higher deployed services make the provisioning of contents a hard challenge, due to processing and signalling overhead. The limitations imposed by the throughput make these specific types of networks inadequate for streaming of high quality video (the one which DVB-T aims at broadcasting), but still seam reasonable for the transmitting of mobile video contents (which are the market target of DVB-H), which require less bitrate (384 Kbps).

From these premises, the heterogeneous mobility tests were performed for a hop count of 3 nodes, using auto-configuration and unicast routing features.

The first presented results concern the delay a join message would suffer, if sent from the MN to the LMA, when the MN is in infrastructure and ad-hoc mode. For the ad-hoc results, delay is shown as a function of the number of hops in Table 10. Also the throughput for the WiFi links is indicated with the purpose of introducing an idea on what kind of contents could be provided over such a link.

Hop Count	Delay (ms)
Infrastructure	1.12
1 Hop	1.27
2 Hop	3.27
3 Hop	7.24

Table 10 - Uplink channel delay

Hop Count	Throughput
Infrastructure	5520
1 Hop	1222
2 Hop	559
3 Hop	322

Table 11 - Throughput

The WiFi links (both in ad-hoc and infrastructure modes) are envisioned to be of a useful purpose for uplink purposes. For such a scenario the delay for the reception of multicast join messages is perfectly feasible. Also, this link could be suitable for downlink of a 384kbps up to the 2^{nd} hop.

5.3 Mobility performance metrics (vertical handovers)

5.3.1. Handover of the downlink channel between two DVB channels

In this section the handover between two DVB cells is analysed. When the terminal detects that it is no longer receiving signal on the current DVB cell, it looks for other channels on the *Tzap* configuration file. The change of channel alone takes an average of 150 ms. The process of changing channel and performing a signal measure on the other cell takes about twice that time (310 ms), considering the SNR is enough in the second cell option for a signal LOCK to be triggered. These values were obtained by switching off the power amplifier at the first cell, and correspond to the time between the event where the terminal stops receiving data and the event where the terminal detects new signal in the new cell.

These times are considered to be acceptable when receiving a digital multimedia content such as a movie, since 310 ms of non-connectivity corresponds to a single glitch. Nevertheless, if the configuration file is not complete and a scan is needed, the waiting process can take up to a few minutes. This is actually the reasoning for us to support the handover for WiFi technology, since the user can receive traffic through WiFi while performing a scan on the air-interface for available DVB cells.

The time for L3 handover, i.e., the time it takes for the receiver to start receiving traffic, could not be tested, since only one inserter is available. We thus leave this study for the following subsections.

5.3.2. *Handover of the downlink channel from DVB to WiFi Infrastructure*

Several tests were addressed in order to evaluate the performance of executing heterogeneous handovers. The results shown in this subsection regard handovers between DVB and WiFi links.

To force the handover, the antenna is unplugged from the DVB receiver, thus triggering *Tzap* to detect the loss of signal and react. Figure 23 shows two handovers; one nearly after 6 seconds, which represents the handover from WiFi to DVB, and the second at 16 seconds, which corresponds to a handover back to the WiFi link.

As can be seen in Figure 23, using *Tzap's* default configuration, upon handover from the DVB to WiFi, the system takes about 4 seconds to start receiving signal from the WiFi interface in infrastructure mode. These results were obtained by generating a 4Mbps traffic flow with *mgen* tool [133]. Since the WiFi router is set at 11 Mbps and has a throughput of 5.4Mbps; the reasoning for this particular choice of bitrates is to assure the minimum delay on the Inserter, when there are still no losses in both links (DVB and WiFi).

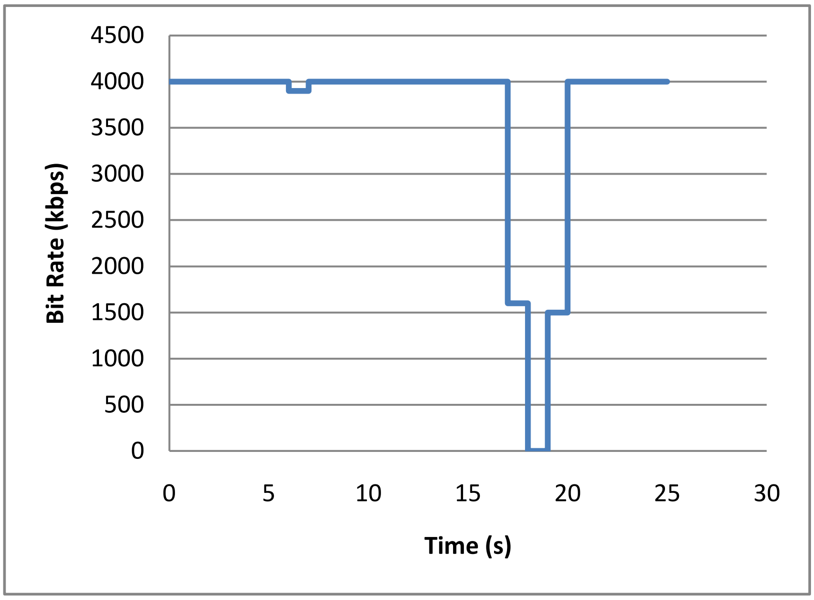

Figure 23 – Handover from DVB to WiFi

Having the previous data in mind the tzap configuration was changed so that it would perform measurements at a fastest rate using a 0.01 ms of interval between measurements (former period was of 1 second). Also the condition which triggers the handover was changed. Handovers are no longer triggered by signal loss or acquisition, but by a SNR value crossing. An initial value for the SNR was set with 0x8000 (The API [128] converts this value to dBs by multiplying by a value of 10^{-6}) which corresponds to 32 x 10^{-3} dB (1.0074). So a handover to DVB occurs when the SNR value overgoes the 0x8000 value, and the handover to WiFi occurs when the SNR value undergoes the 0x8000 value.

In the following test, the terminal is first receiving traffic via the WiFi interface and then performs a handover to DVB around 6 seconds and at the 11 seconds it re-starts receiving traffic on the WiFi interface. In this test, the *Tzap* configuration is tuned.

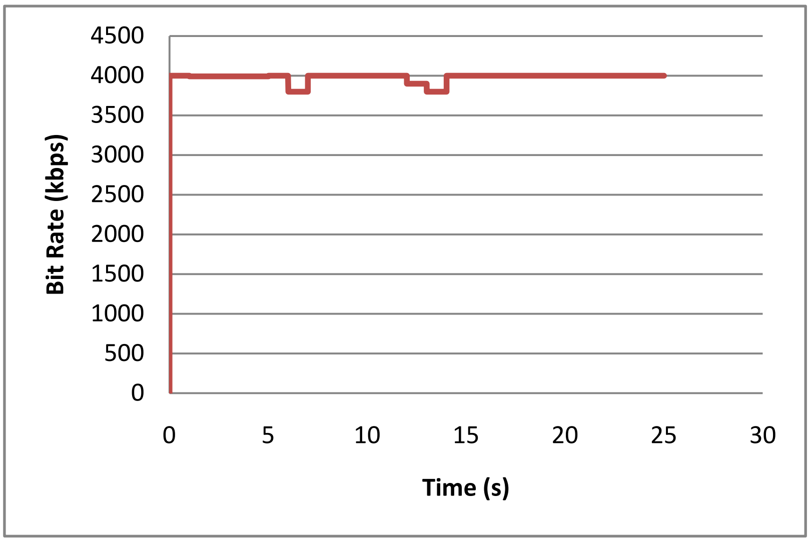

Figure 24 – Bit Rate variation during a Handover between DVB and WiFi after tuning Tzap's configuration

It can be seen that the transition from WiFi to DVB introduces nearly 5% of losses in a second which represents 50 ms of non-connectivity. The tool used to obtain the figures (*trpr* [134]) had a one second resolution. The handover is forced by plugging the equipment's antenna, thus raising the SNR to a level which triggers *tzap* to issue a signal detection message to the Mobility Manager. This entity, according to a preferences list, requests to receive the desired service via the DVB interface. The reported losses correspond to the time the handover requires to be performed. In fact this should consist of a soft handover, since when the handover is requested, the destination interface is already up and the signal is being received with sufficient SNR.

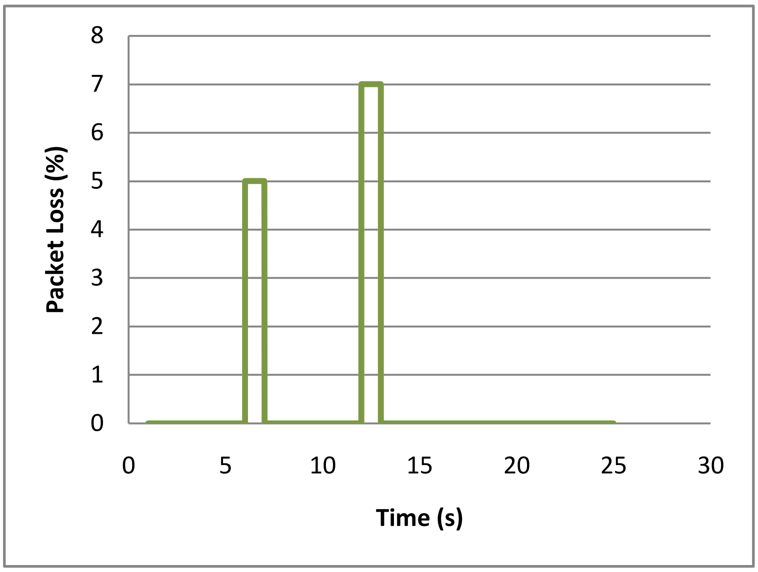

Figure 25 – Packet loss during a Handover between DVB and WiFi after tuning Tzap's configuration

The handover time from DVB to WiFi was minimized and now represents nearly 70 milliseconds of non connectivity, since reported losses are of 7%. The handover is forced by removing the antenna, thus forcing the terminal to overgrow a predefine level of SNR; *tzap* triggers a message towards the Mobility Manager which tries to find another possible candidate interface for the reception of the contents, again according to a preferences' list. Contrarily to the handover from WiFi to DVB, such a case no longer regards a soft handover, since there is a lost of connectivity.

# of run	% of Loss upon handover from DVB to WiFi	% of Loss upon handover WiFi to DVB
Experiment 1	8,0	6,0
Experiment 2	7,0	5,0
Experiment 3	8,0	5,0
Experiment 4	7,0	7,0
Experiment 5	6,0	6,0
Experiment 6	8,0	8,0
Experiment 7	7,0	7,0
Experiment 8	7,0	6,0
Experiment 9	8,0	5,0
Experiment 10	7,0	7,0
Average	7,3	6,2
Standard Deviation	0,7	1,0

Table 12 – Loss percentage upon handover between DVB and WiFi

Table 12 confirms the tendency of slightly higher losses upon handover from DVB to WiFi. This is because the DVB inserter incorporates buffers in order to perform a more efficient insertion on the MPEG-2 TSs, which the terminal does not receive after performing handoff thus introducing an extra loss factor.

For this same reason the handover from WiFi to DVB, albeit introducing similar losses, takes a longer time, since the buffers at the Inserter need to reach a certain level before performing the insertion of the IP frames in the TSs. Hence, the buffers at the equipment introduce two distinct situations: reduction of losses and of delay.

Consider an example application, which can be, for instance, video streaming. For this particular type of data the best way to solve the verified phenomenon when performing a handover from the WiFi to the DVB link is to add a buffer at the receiver, which is commonly used by applications. It was expected that this handover did not introduce any losses, since it was supposed to be a soft handover as signalling only occurs after the signal stability of the DVB link is achieved. Some tests were also conducted forcing the handover by software indication, thus assuring perfect link conditions for a soft handover. The results are shown in Figure 26 and Figure 27.

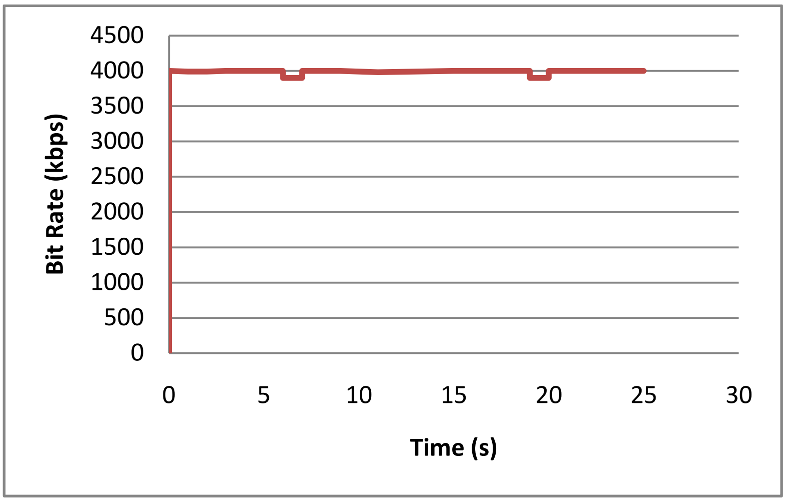

Figure 26 - Bit Rate variation during a Handover between DVB and WiFi without signal loss

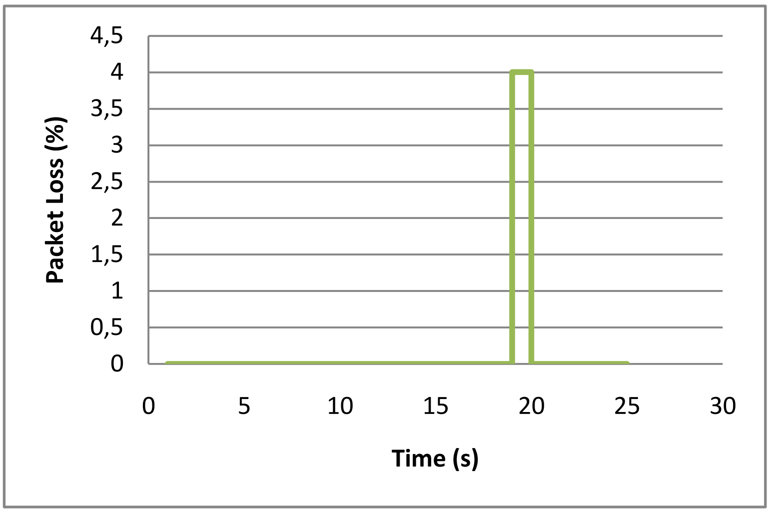

Figure 27 - Loss variation during a Handover between DVB and WiFi without signal loss

It can be seen that under these conditions, there are no losses reported regarding the handover from WiFi to DVB, but the terminal does not receive traffic during approximately 25ms. This can be seen by analysing the slight drop of bitrate, but without any loss report. This decrease of throughput is related to the link specific delay which was verified to be nearly 23 ms plus the time the equipment requires, in order to start broadcasting. When performing the handover back to the WiFi link, the percentage of loss packets is less then the one shown on the previous results. The switching procedure itself introduces delay less time than the DVB’s link delay time, and thus packets received on the WiFi link are more recent packets than the ones received on the DVB link. For this particular handover, the introduced losses regard mainly the packets retained at the inserter’s buffer. The reason why the loss percentage is lower than the one presented in Table 12 arises from the lack of awareness necessity from the *tzap* module, which is no longer required to verify link loss or acquisition.

The conclusion to withdraw from these considerations is that, for the results previously shown, although the software (*tzap*) is providing an indication that the SNR is high enough to assure a handover, the link conditions are still not as constant as desired, hence there are still losses associated, which obviously represents a more realistic scenario.

Considering the least optimistic scenario, when implementing a buffer at the terminal, it should, at least, accommodate 70 ms of traffic reception time, which represents a size related to the product between this time and the bit rate in use. This size will guarantee that the user will not be aware of the handover when it happens.

The handover from DVB to WiFi introduces a more complex problem to be solved, since information is lost at the buffers and cannot be recovered. Obviously, this problem can be either dealt on the network or on the terminal side.

Again, let us focus on a UDP multimedia stream to be delivered to a certain multihomed terminal, connected to a MAG with a DVB link and MAG with a WiMAX link. If the terminal is receiving this specific flow via the DVB interface, but verifies that it requires a handover to WiMAX, a simple solution would be having the LMA performing a bi-casting of the data to both the MAGs. However, as explained before, different delays would lead to the reception of two different time-spaced sequences of the same video. On the other hand, when performing a handover from a low to a high latency technology due to buffering, the terminal would receive duplicate packets on the destination network, but in different time instances. The outcome of this simple solution is a series of problems which the terminal itself must then solve. Other more complex schemes attempt to provide a solution by buffering information at a common-on-the-path network entity, which when applying to the described architecture, would be the LMA. This requires the implementation of a buffering system, with an *à posteriori* retrieval mechanism, resultant from the interaction of the terminal and the LMA, which means more processing and signalling as an outcome. Still, this would require the terminal to identify which information was lost upon handover and request the retransmission of these data packets, which are expected to be stored as close as possible to the terminal. Obviously, scalability issues arise as heavy constraints. The higher the number of terminals, the higher this buffer should become. Also, [83] discusses this subject explaining that although several mechanism already implement buffers to store packets during handover, such as by *Low Latency Handoffs in Mobile IPv4* (LLH) [8] and *A Distributed Buffer Management Approach Supporting IPv6 Mobility* [9], they do not look into problems dealing with multicast flows. Furthermore, such proposals do not envision scenarios where the technologies already introduce high losses due to internal operations with buffers.

In the end, special adjustments would always be needed on the terminal side, thus creating a solution which handles most of the problem locally seems more and more pertinent. If the terminal is receiving a multicast stream via the DVB interface and verifies that a handover is required, it should issue a *Join* via the WiFi interface. The data received on this last interface corresponds to time-advanced information, so it is necessary to store it until it is time to be used. Such time occurs when the same packet being received on the DVB interface reaches the first packet already stored. The packets from the DVB interface must now be discarded and the ones from the WiFi link must be transferred to the application's buffering system. Ultimately, this could be done simply by delaying the packets on the WiFi card, by an approximate value in comparison to the delay on the DVB link.

More intelligent buffering management systems could be implemented on the application side. They should however, constantly analyse a certain amount of data, and organize it on the fly.

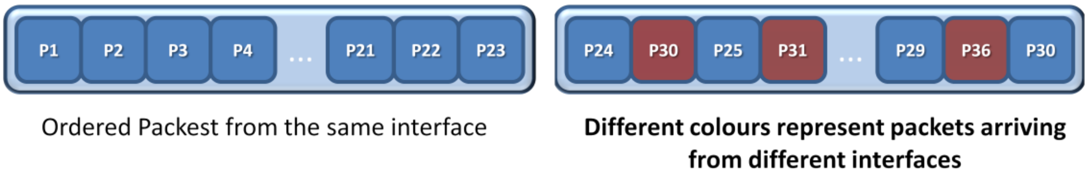

Figure 28 – Order or arrival to the buffers

The left block of Figure 28 shows packets arriving from a single interface and thereby ordered in the same sequence as generated. The right block of the same figure represents the time where a handoff is being performed on a multihomed terminal. The destination link of the handover introduces less delay than the one which the terminal was originally using, so packets arrive in two time-sequences. The red packets are arriving on the interface with less delay, so the algorithm needs to reorder the sequence, so that the application only reproduces these packets after the first repeated packet. This means that *P30* of the red sequence is only reproduced after the algorithm processes the *P30* of the blue sequence. The necessary processing power to achieve this is somewhat significant, especially when considering on-the-fly behaviours, but this can be smoothed considering window intervals. Again, the mention interval of 20ms is a good metric to use in order to withdraw conclusions; when an unordered packet is found, the algorithm must analyse the next 20ms, first evaluating if a lost has occurred, or if significant information will follow. It is obvious that, in order to assure the proper operation of this mechanism, the packets must be numbered with some kind of sequence number, which the algorithm must be capable of understanding.

Other concepts can be applied from Digital Satellite Links, which already deal with high intrinsic delays and link errors especially to support TCP services. Due to these issues, classical versions of the TCP protocol experience several performance problems, which origin several retransmission timeouts affecting the congestion window of the transfers. There are already several proposals [76], [77], [78], [79], which aim to improve the TCP connections over Terrestrial and Satellite communications. [80] evaluates the behaviour of four of the most popular TCP versions by analysing the throughput. The study is based on Selective Acknowledgements (SACK), two versions of NewReno and Westwood, and shows that the presented hybrid architecture introduces advantages for TCP connections in terms of the average goodput.

5.3.3. Handover of the uplink channel between WiFi Infrastructure to WiFi Ad-Hoc

Considering DVB cells to be of wider range than those of WiFi, it is expected that a mobile terminal performs a higher number of handovers of the uplink channel than of a downlink one. The integration of a DVB-T network with the WiFi technology (both infrastructure and ad-hoc) is studied in this section with the presentation of some performance metrics. The addition of WiFi ad-hoc networks as a mean to achieve uplink connectivity is an issue which is yet to be considered in the literature. This may be related to several performance issues introduced by this WiFi mode, but the concept may nevertheless be interesting, especially considering other types of ad-hoc networks provided by future technologies.

	1 Hop	2 Hops	3 Hops
With Traffic (sec)	2	2	3
Without Traffic (sec)	2	3	8

Table 13 – Handover from Infrastructure to Ad-Hoc

Table 13 presents times of handovers between infrastructure and ad-hoc modes in two situations: with no traffic in the ad-hoc link and also with traffic already flowing in the ad-hoc link. It can be seen that, in the first situation the handover takes longer, since the AODV protocol needs to construct the routes, for hop counts higher than 1. As expected, the time of handover also increases with the increase of the hop count due to the signalling between the terminal and the ER, but is only meaningfully on higher hop counts than 2.

	1 Hop	2 Hops	3 Hops
Time (sec)	1.5	1.5	1.5

Table 14 – Handover between Ad-Hoc and Infrastructure

The results regarding the handover from ad-hoc to infrastructure are shown in Table 14. As the terminal starts in the ad-hoc link, it is considered that it already has a route towards the gateway, hence the signalling delay is not significant when comparing to the hop count.

The current technology specifications only permit the realistic deployment of ad-hoc networks as uplink channels due to the presented metrics in Section 5.2.3 and more detailed in [135], a related work also mentioned in the contributions section of this document (Section 1.4). Low throughput, high delay, and inconstant jitter are the main causes for such reasoning. Also, devices with low processing capabilities and high power consumption constraints are unfit for high mobility scenarios (especially concerning that DVB-T receivers require some power to work). The used receiver is a USB device which can consume up to 2.4W, but typically DVB-T's receivers consume around 1W. Mobile handheld terminals power consumption of the RF and baseband processing may come down to 600 mW by the year 2007 according to the DVB project estimations [84]. DVB-H already takes these concerns into account and according to [85], through advanced TS mechanisms; it is possible to reduce the power consumption up to 95% when comparing to continuously operated conventional DVB-T receivers.

5.4 Conclusions

This chapter retains the outcome experimental evaluation of the signalling expressed in the previous chapter. One of the main characteristics of the broadcast links is the fact that they introduce high delays which are heavily correlated with the occupancy of the link, due to the buffer-based implementations which are motivated by the increase of efficiency. Given what was formerly said, this last sentence may seem controversy, but given the context of high mobility terminals, it is best to condense the information which is to be transmitted, and send it in burst, thus reducing the necessary time, which the receiver needs to be consuming power. From the point of view of the network's performance, however, the efficiency is reduced, since such behaviour introduces a significant delay increase, which has a strong impact on the handover's smoothness. Also when considering satellite transmissions where delays are high, the same problem arises, affecting not only the performance of Real Time applications but also TCP traffic, due to the strong variations of delay, which is inherently high. For this case there are already several proposals to adapt the TCP mechanisms over high delay/jitter links. [136] presents a study comparison on this particular scenario highlighting Hybla as a solution, which already expresses a concern in solving this issue and provides a good performance Hybla [137] is a TCP enhancement for heterogeneous networks where broadcast technologies which introduce high delays, are of great concern.

The integration of Ad-Hoc networks was also shown, but is was proven that with the current technology such specific multi-hoc links are not of great usage as a downlink for multimedia contents, however they are more than suitable to be used as the uplink channel for interactive multicast services. Nevertheless the designed signalling presented acceptable performance metrics, especially after some adaptations were made concerning the frontend configurations, and the terminal awareness.

Chapter 6 CONCLUSIONS

6.1 Summary

This dissertation presents an architectural proposal which supports mobility with QoS over heterogeneous networks, integrating several protocols and proposals. It uses several concepts to introduce a hierarchical management of the terminals, both at the entity level and at the signalling level. NetLMM alike proposals provide a hierarchical management by the entities, aiming at the low capabilities of the terminals, which should simplify its performance. This requires the support of multiple levels of management from the network side, since it can range from a handover between two APs connected to the same MAG, or two APs in different Local Mobility Domains or even Administrative Domains. For this last case, a global mobility protocol is required with a signalling which can react well to the characteristics of such movement of flows (not necessarily physical movement). The support of QoS is also taken into account, by the report of metrics to central hierarchical QoS databases, which are consulted by the decision holder units.

Also a series of performance metrics are shown which are used to withdraw considerations regarding the consequences of integrating a broadcast network, and especially a DVB-T network within a 4G network.

It can be seen that the buffers in the DVB inserter introduce a link delay which requires other techniques to be deployed in order to support fast and lossless handovers. Several proposals are being analysed in order to reflect an improvement based upon the caching of information on the ARs and its correct retrieval.

WiFi, when in infrastructure mode, support some broadcast services with a relative performance, but the drivers need to be tuned. Such a measure urges because broadcast support in WiFi, by the APs is provided according to the BSS' lowest rate, e.g., by default, APs lowest bit rate is 1Mbps, and hence that's what they set as the maximum bit rate for broadcast purposes. However it is acceptable to have broadcast dedicated APs providing the same services as a DVB, but in smaller bouquets or with media routers which downgrade the quality of videos. This application can be of particular interest to improve reachability over areas which are not covered by the DVB cells, such as indoor scenarios.

The usage of Ad-Hoc Networks for multicast services subscription signalling seems adequate, but due to its reduced throughput it should not be expected to deliver the same contents in the downlink as DVB.

Based on the mobility results, operators can withdraw practical results such as the timeout for the terminal de-registration at the ER, which should be always greater than 10 seconds considering ad-hoc connections. For this case it is considered that in the absence of a return channel the terminal cannot receive multimedia contents on the DVB interface, for at least a certain window. This window should be set according to the paying model, e.g. if a user has purchased a service for a certain amount of time, or if this user is paying according to the reception time.

Broadcast technologies introduce high delay witch becomes an issue when considering fast mobility and QoS requirements.

6.2 Future Directions

From the withdrawn results it is possible to conclude that the integration of broadcast technologies with 4G architectures is possible but still needs to be tuned from the available features provided by the devices. With upcoming versions of software some problems will be easily solved thus enabling the network to collect more information regarding the broadcast link conditions. This indeed is an aspect of interest, which is mainly conditioned by the interest of the fabricants, which in turn are conditioned by the demand (on the client side). The used inserter was able to report performance metrics to the network, and this study could focus more on the control and admission based on QoS demands.

With more intelligence given to the network, a more efficient module to choose the most profitable interface, on the client side, would also be of interest to implement. One that does not only take into account a list of options but which considers parameters and the users' preferences.

As a future direction on the non-personal pursuit level, one can help to fell intrigued on the unnecessary delay introduced by Broadcast technologies. This could be justified by the need to increase efficiency thus resulting in the delay caused by buffering, but even when the used bit rates are near to the maximum limit, delay is still a considerable constraint. Implemented 3G/HSDPA which will transport MBMS also introduce high and variable delays (with high jitter). It would be of interest that these issues are well measured, as they are crucial with the increase of interest on real-time applications.

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Annex 1

The bellow presented figure shows the DVB supported Tables as present in [54]

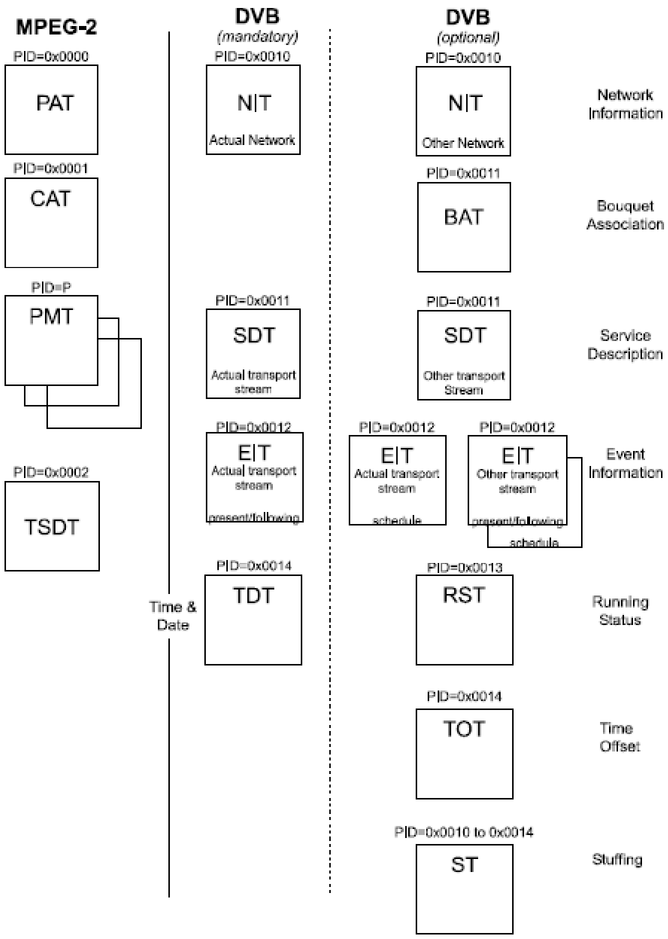

Figure 29 – General Organization of the Service Information (SI)

ANNEX 2

Example of IP/MAC streams with INT as presented in [55].

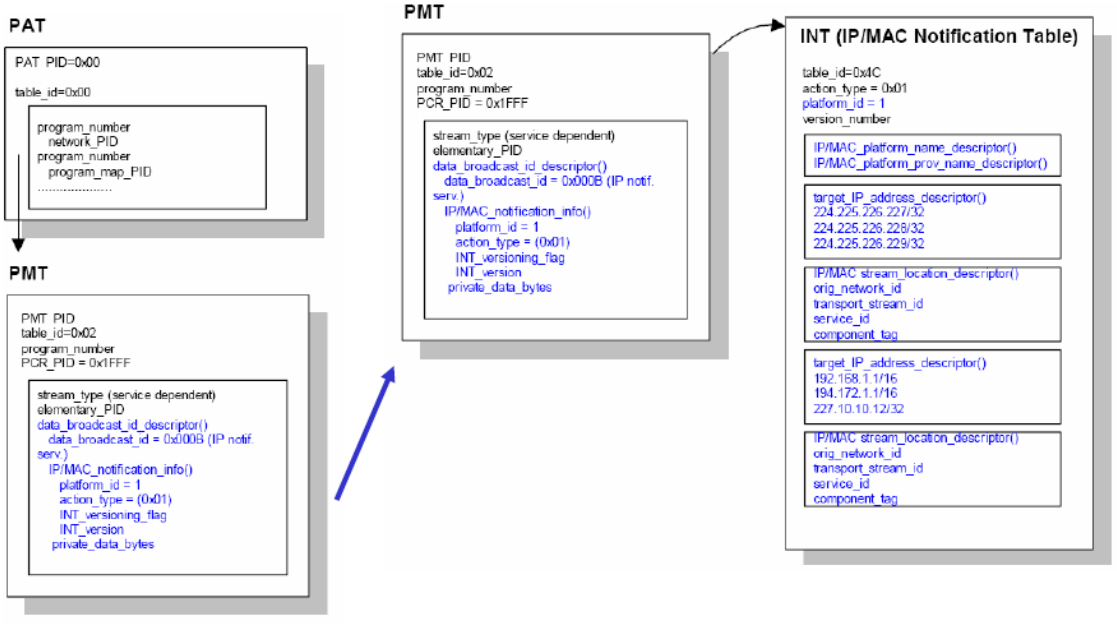

Figure 30 Example of IP/MAC streams with INT reference

ANNEX 3

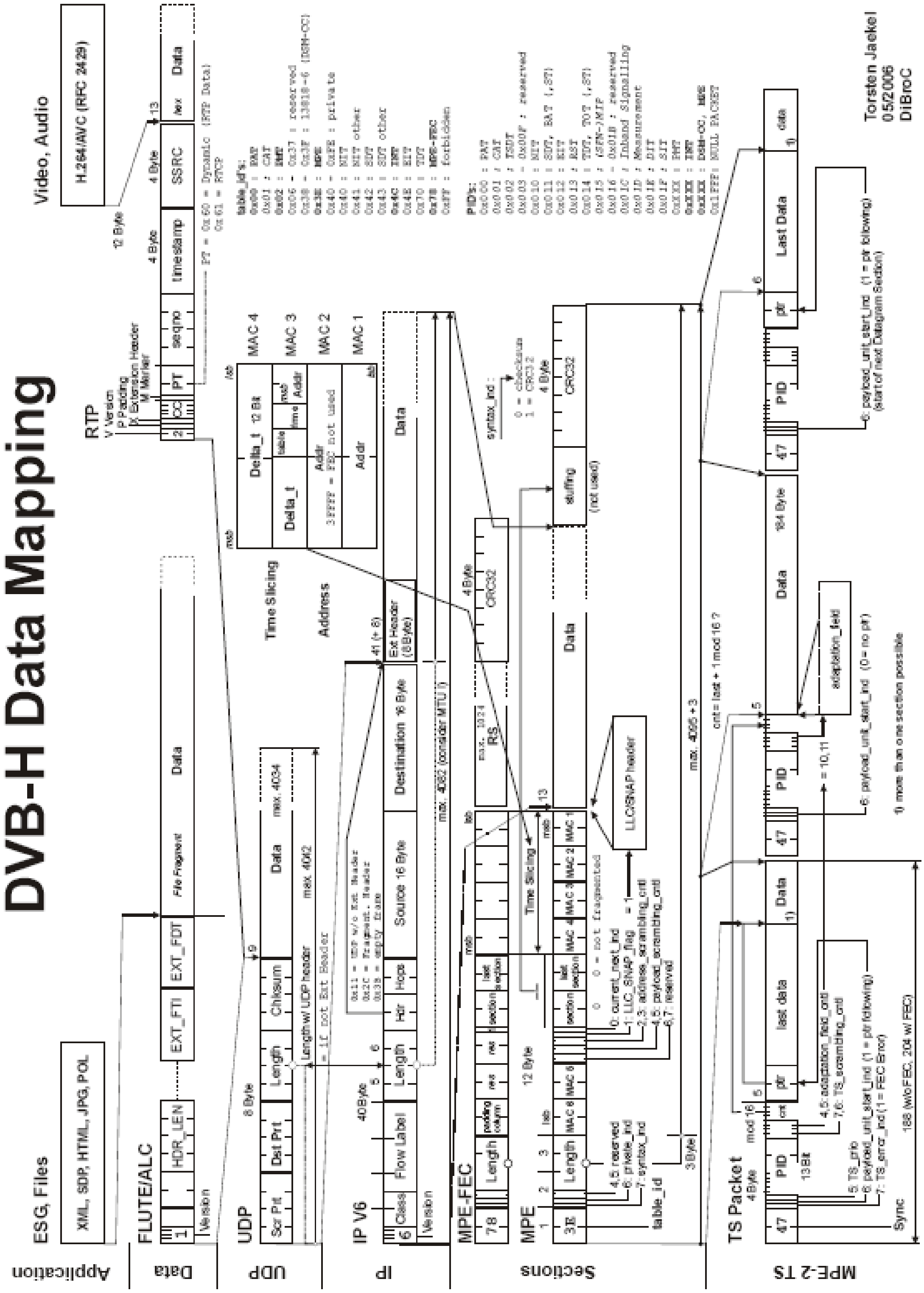

Figure 31 – DVB-H Data Mapping

ANNEX 4

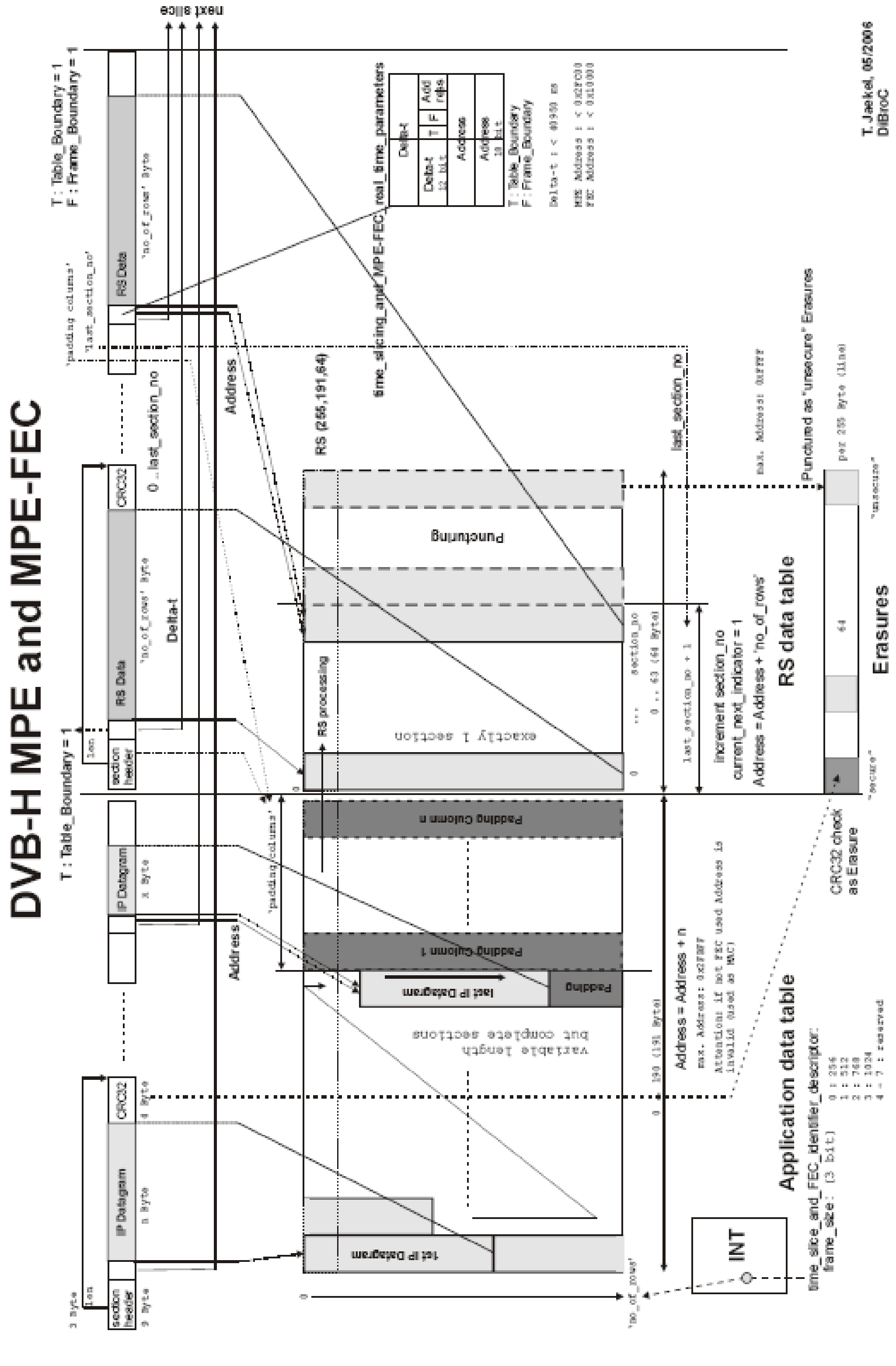

Figure 32 – DVB-H MPE and MPE-FEC