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Délivrance de services médias ubiquitaires adaptés selon le contexte au sein de réseaux de nouvelles générations
-
Context-awareness for ubiquitous media service delivery in Next Generation Networks

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ABSTRACT

The latest advances in technology have already defied Moore's law. Thanks to research and industry, hand-held devices are composed of high processing embedded systems enabling the consumption of high quality services. Furthermore, recent trends in communication drive users to consume media *Anytime*, *Anywhere* on *Any Device* via multiple wired and wireless network interfaces. This creates new demands for ubiquitous and high quality service provision management.

However, defining and developing the next generation of ubiquitous and converged networks raise a number of challenges. Currently, telecommunication standards do not consider context-awareness aspects for network management and service provisioning. The experience felt by the end-user consuming for instance Voice over IP (VoIP) or Internet Protocol TeleVision (IPTV) services varies depending mainly on user preferences, device context and network resources. It is commonly held that Next Generation Network (NGN) should deliver personalized and effective ubiquitous services to the end user's Mobile Node (MN) while optimizing the network resources at the network operator side.

IP Multimedia Subsystem (IMS) is a standardized NGN framework that unifies service access and allows fixed/mobile network convergence. Nevertheless IMS technology still suffers from a number of confining factors that are addressed in this thesis; amongst them are two main issues :

The lack of context-awareness and Perceived-QoS (PQoS) : The existing IMS infrastructure does not take into account the environment of the user , his preferences , and does not provide any PQoS aware management mechanism within its service provisioning control system. In order to ensure that the service satisfies the consumer, this information need to be sent to the core network for analysis.

In order to maximize the end-user satisfaction while optimizing network resources, the combination of a user-centric network management and adaptive services according to the user's environment and network conditions are considered. Moreover, video content dynamics are also considered as they significantly impact on the deduced perceptual quality of IPTV services.

The lack of efficient mobility mechanism for conversational services like VoIP : The latest releases of Third Generation Partnership Project (3GPP) provide two types of mobility solutions. Long-Term Evolution (LTE) uses Mobile IP (MIP) and IMS uses Session Initiation Protocol (SIP) mobility. These standards are focusing on signaling but none of them define how the media should be scheduled in multi-homed devices. The second section introduces a detailed study of existing mobility solutions in NGNs.

Our first contribution is the specification of the global context-aware IMS architecture proposed within the European project ADAptative Management of mediA distributioN based on saTisfaction orIented User Modeling (ADAMANTIUM). We introduce the innovative Multimedia Content Management System (MCMS) located in the application layer of IMS. This server combines the collected monitoring information from different network equipments with the data of the user profile and takes adaptation actions if necessary Then, we introduce the User Profile (UP) management within the User Equipment (UE) describing the end-user's context and facilitating the diffusion of the end-user environment towards the IMS core network. In order to optimize the network usage, a PQoS prediction mechanism gives the optimal video bit-rate according to the video content dynamics.

Our second contribution in this thesis is an efficient mobility solution for VoIP service within IMS using and taking advantage of user context. Our solution uses packet duplication on both active interfaces during handover process. In order to leverage this mechanism, a new jitter buffer algorithm is proposed at MN side to improve the user's quality of experience. Furthermore, our mobility solution integrates easily to the existing IMS platform.

Keywords - Context-awareness, Quality of Experience, Voice over IP, IPTV, IP Multimedia Subsystem, Next Generation Network.

RÉSUMÉ

Les dernières avancées technologiques défient la loi de Moore. Grâce aux avancées de la recherche et de l'industrie, les terminaux mobiles sont dotés de systèmes embarqués performants rendant accessible les services multimédias les plus exigeants. De plus, le nouveau modèle de consommation de médias se résume par le concept "Anytime, Anywhere, Any Device". Cela impose donc de nouvelles exigences en termes de déploiement de services ubiquitaires.

La conception et le développement de réseaux ubiquitaires et convergents de nouvelles générations soulèvent cependant un certain nombre de défis techniques. Les standards actuels ne considèrent pas les aspects de sensibilité au contexte pour la gestion du réseau.

Le ressenti de l'utilisateur concernant certains services multimédias tels que la VoIP et l'IPTV dépend fortement des capacités du terminal et des conditions du réseau d'accès. Cela incite les réseaux de nouvelles générations à fournir des services ubiquitaires adaptés à l'environnement de l'utilisateur optimisant par la même occasion ses ressources.

L'IMS est une architecture de nouvelle génération qui centralise l'accès aux services et permet la convergence des réseaux fixe/mobile. Néanmoins, l'évolution de l'IMS est nécessaire sur les points suivants :

- **l'introduction de la sensibilité au contexte utilisateur et de la PQoS**:
l'architecture IMS ne prend pas en compte l'environnement de l'utilisateur , ses préférences et ne dispose pas d'un mécanisme de gestion de PQoS. Pour s'assurer de la qualité fournie à l'utilisateur final, ces informations doivent transiter en cœur de réseau afin d'y être analysées. Ce traitement aboutit au lancement du service qui sera adapté et optimisé selon les conditions observées. De plus pour le service d'IPTV, les caractéristiques spatio-temporelles de la vidéo influent de manière importante sur la PQoS observée côté utilisateur. L'adaptation des services multimédias en fonction de l'évolution du contexte utilisateur et de la nature de la vidéo diffusée assure une qualité d'expérience à l'utilisateur et optimise par la même occasion l'utilisation des ressources en cœur de réseau.

- **une solution de mobilité efficace** pour les services conversationnels tels que la VoIP:
Les dernières publications 3GPP fournissent deux solutions de mobilité: le LTE propose MIP comme solution de mobilité alors que l'IMS définit une mobilité basée sur le protocole applicatif SIP. Ces standards définissent le système de signalisation mais ne s'avancent pas

sur la gestion du flux média lors du changement d'interface réseau. La deuxième section introduit une étude comparative détaillée des solutions de mobilité dans les NGNs.

Notre première contribution est la spécification de l'architecture globale de notre plateforme IMS sensible au contexte utilisateur réalisée au sein du projet Européen ADAMANTIUM. Nous détaillons tout d'abord le serveur MCMS intelligent placé dans la couche application de l'IMS. Cet élément récolte les informations de qualité de service à différents équipements réseaux et prend la décision d'une action sur l'un de ces équipements. Ensuite nous définissons un profil utilisateur permettant de décrire l'environnement de l'utilisateur et de le diffuser en cœur de réseau. Une étude sur la prédiction de satisfaction utilisateur en fonction des paramètres spatio-temporels de la vidéo a été réalisée afin de connaître le débit idéal pour une PQoS désirée.

Notre deuxième contribution est l'introduction d'une solution de mobilité adaptée aux services conversationnels (VoIP) tenant compte du contexte utilisateur. Notre solution s'intègre à l'architecture IMS existante de façon transparente et permet de réduire le temps de latence du handover. Notre solution duplique les paquets de VoIP sur les deux interfaces actives pendant le temps de la transition. Parallèlement, un nouvel algorithme de gestion de mémoire tampon améliore la qualité d'expérience pour le service de VoIP.

Keywords - Sensibilité au contexte, QoE, VoIP, IPTV, IMS, NGN.

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List of Abbreviations

16CIF 16xCIF

3G Third Generation

3GPP Third Generation Partnership Project

3SQM Single Side Speech Quality Measure

4CIF 4xCIF

9CIF 9xCIF

AAA Authentication Authorization and Accounting

ACD Automatic Call Distribution

ACR Absolute Category Rating

ADAMANTIUM ADAptative Management of mediA distributioN based on saTisfaction orIented User Modeling

AEM Action Engine Module

AES Advanced Encryption Standard

AMG Adaptation Media Gateway

AMR Adaptive Multi-Rate

AS Application Server

ANAM Access Network Adaptation Module

ANFIS Adaptive Neural-Fuzzy Inference System

ANMM Access Network Monitoring Module

AP Access Point

AS Application Server

BGCF Breakout Gateway Control Function

B2BUA Back-to-Back User Agent

BER Bit Error Rate

BLER Block Error Rate

CC/PP Composite Capabilities/Preference Profiles

CCR Comparison Category Rating

CIF Common Interface Format

CN Correspondent Node

CNR Carrier-to-Noise Ration

CoA Care-of Address

CSCF Call State Control Function

CLI Command Line Interface

CLIPS C Language Integrated Production System

CRM Customer Relationship Management

CSCP Comprehensive Context Profiles

CT Content Type

CTI Computer Telephony Integration

CU Corporate User

DAR Dynamic Address Reconfiguration

DCR Degradation Category Rating

DDL Description Definition Language

DI Digital Item

DSCQS Double Stimulus Continuous Quality Scale

DSIS Double Stimulus Impairment Scale

EPC Evolved Packet Core

EPS Evolved Packet System

EMM External Monitoring Module

FE Front End

FEC Forwarding Error Correction

FN Foreign Network

FR Frame Rate

GGSN Gateway GPRS Support Node

GOP Group Of Pictures

GSM Global System for Mobile communications

GUP 3GPP Generic User Profile

HA Home Agent

HN Home Network

HBM host-based Mobility

HLR Home Location Register

HPLMN Home Public Land Mobile Network

HSS Home Subscriber Server

HVS Human Visual System

I-CSCF Interrogating-CSCF

IEEE Institute of Electrical and Electronics Engineers

IETF Internet Engineering Task Force

iFC initial Filter Criteria

IMM Internal Marking Module

IMS IP Multimedia Subsystem

IP Internet Protocol

IPC Inter Process Communication

IP-PBX Internet Protocol Public Branch eXchange

IPMS IP Mobility Selection

IPTV Internet Protocol TeleVision

ISC IMS Service Control

ISUP ISDN User Part

ITU International Telecommunication Union

IVR Interactive Voice Response

LAN Local Area Network

LMA Local Mobility Anchor

LTE Long-Term Evolution

MAC Medium Access Control

MAG Mobile Access Gateway

MA-PCSCF Mobility-Aware P-CSCF

MA-ICSCF Mobility-Aware I-CSCF

MBL Mean Burst Length

MCF Media Control Function

MCMS Multimedia Content Management System

MDF Media Delivery Function

MGCF Media Gateway Control Function

MGW Media Gateway

MIH Media-Independent Handover

MIP Mobile IP

MIPv6 Mobile IPv6

MMIF MCMS Interface

MMSC Multi-Media Session Continuity

MN Mobile Node

MOS Mean Opinion Square

MPEG Moving Picture Expert Group

MPLS Multi-Protocol Label Switching

MRF Media Resource Function

MRFC Media Resource Function Controller

MRFP Media Resource Function Processor

mSCTP mobile-SCTP

MSAM Multimedia Service Adaptation Module

MSMM Multimedia Service Monitoring Module

MSRF Media Server Resource Function

NASS Network Attachment SubSystem

NBM network-based Mobility

NEMO NEtwork MObility

NGN Next Generation Network

NN Neural Network

NP KPI Network Performance Key Performance Indicator

NQoS Network-QoS

OFDMA Orthogonal Frequency-Division Multiple Access

OSI Open System Interconnection

PAMS Perceptual Analysis Measurement System

P-CSCF Proxy-CSCF

PLMN Public Land Mobile Network

PCRF Policy and Charging Rules Function

PBX Public Branch eXchange

PC Pair Comparison

P-CSCF Proxy-CSCF

PCI Peripheral Component Interconnect

PDF Policy Decision Function

PESQ Perceptual Evaluation of Speech Quality

PIS Proxy Information Server

PLC Packet Loss Concealment

PoS Point of Service

PMIP Proxy-MIP

PMIPv6 Proxy-MIPv6

PQoS Perceived-QoS

PQH Perceptual Quality Level

PSNR Peak Signal to Noise Ratio

PSTN Public Switched Telephone Network

PSQA Pseudo-Subjective Quality Assessment

PSQM Perceptual Speech Quality Measure

QCIF Quarter Common Interface Format

QoE Quality of Experience

QoS Quality of Service

qPSNR quasi-PSNR

RACS Resource and Admission Control Subsystem

RACF Resource and Admission Control Function

RAF Repository Access Function

RDF Resource Description Framework

RNN Random Neural Network

RTSP Real-Time Streaming Protocol

RTCP Real-time Transfer Control Protocol

RTP Real-Time Protocol

S-CSCF Serving-CSCF

SCTP Stream Control Transmission Protocol

SBR Send Bit Rate

SC Stimulus Comparison

SCF Service Control Function

SCTP Stream Control Transmission Protocol

SDSCE Simultaneous Double Stimulus for Continuous Evaluation

SDF Service Delivery Function

SDP Session Description Protocol

SDR Software Defined Radio

SER SIP Express Router

SGW Signalling Gateway

SLA Service Level Agreement

SIP Session Initiation Protocol

SS Single Stimulus

SSCQE Single Stimulus Continuous Quality Evaluation

SSF Service Selection Function

SSIM Structure SIMilarity

SVGA Super Video Graphic Array

SVC Scalable Video Coding

TCP Transmission Control Protocol

TiSPAN Telecoms Internet converged Services Protocols for Advanced Networks

TNAM Transport Network Adaptation Module

TNMM Transport Network Monitoring Module

UA User Agent

UCD Universal Constraints Description

UDC User Data Convergence

UDR User Data Recovery

UDP User Datagram Protocol

UA User Agent

UE User Equipment

UED Usage Environment Description

UOP University of Plymouth

UP User Profile

UPM User Profile Manager

UMTS Universal Mobile Telecommunications System

VGA Video Graphic Array

VPN Virtual Private Network

VoD Video on Demand

VoIP Voice over IP

VoLTE Voice over LTE

VQEG Video Quality Experts Group

VQM Video Quality Measurement

VPLMN Visited Public Land Mobile Network

WAN Wide Area Network

WPA2 Wi-Fi Proctected Access II

Wi-Fi Wireless Fidelity

WLAN Wireless Local Area Network

XML eXtensible Markup Language

Chapter 1

Introduction

1.1 Motivation

Computing is becoming more and more embedded and ubiquitous. With the compelling proliferation of mobile devices (i.e. smartphones, tablets, laptops, etc.) and the rapid growth in wireless networking technologies (according to [1], by 2016, Wi-Fi and mobile devices will account, for 61 percent of IP traffic whereas wired devices will account for 39 percent), end-users are willing to access high quality services (i.e. Video-on-Demand, IPTV, VoIP, etc.) anywhere, anytime and through any device or network.

These new trends of consuming high quality multimedia services over ubiquitous computing impose stringent requirements on the current operational networks in terms of scalability, reliability and efficiency. This is especially true for video services (IPTV, Interactive Video, etc.) which generate huge traffic that heavily loads core and access networks and are known to be time-critical and loss-sensitive. These requirements become even more critical when service delivery has to migrate from one access network to another.

Furthermore, consumers always demand high quality multimedia services with better user experience. Consumers also expect efficient and well integrated services into their environment (i.e. device capacity, network condition) but also services delivered according to their preferences (i.e. security options, language, willing to pay or not for a service, network selection, etc.).

One of the main challenges in the telecommunication domain is to provide consumers with personalized and effective services according to the current operational environment. In order to optimize multimedia consumption experience, QoE targets should be defined for each service depending on the initial consumer's environment. If these targets are not fulfilled throughout the consumption, service adaptation or MN's handover towards better network conditions should be triggered. Service provider should then monitor the estimated quality of experience such as PQoS[2] observed at the user's terminal.

1.2 Problem statement

IMS[3] is a standardized architecture designed by 3GPP for controlling and delivering multimedia services that employ IP for transport and SIP for service signaling. This Next Generation Network (NGN) accomplishes fixed/mobile convergence but does not include standardized services. As stated above, consumers require high quality services on disparate devices delivered with a satisfying QoE. Better user experience can be achieved by utilizing *context* information from different sources (i.e device, network, service). By standardizing and unifying the access to media services, IMS has a total control of the active media sessions and can combine service logic (i.e. service prioritization, user preferences) for seamless end-user experience. IMS also has a QoS provisioning system[4] which could be reused and extended for context-awareness mechanism and adaptation enforcement. The most accurate parameter that reflect the perceived quality is at the end user's Mobile Node (MN). Objective QoS evaluation exists in the literature and tends to have significant correlation with subjective QoS evaluation. Nevertheless, most of the objective evaluation mechanisms are resource consuming and require the original data to process the comparison with degraded media received by the end user. Lightweight PQoS prediction and models trained on subjective QoS evaluation emerged recently and provide good correlation with VoIP and IPTV subjective tests.

Nevertheless, the existing IMS infrastructure [5; 6] does not provide any perceived quality mechanism that could tailor the media service to the user's context and thus improve the user experience.

In this context, this thesis focuses on defining a novel IMS compatible user-centric network management solution that employs a **user profiling mechanism** and **adaptive techniques** based on context-awareness. Within the European ADAMANTIUM project, we propose a solution that monitors and gathers network quality parameters at different locations along the delivery chain in order to be analyzed in an innovative management system called Multimedia Content Management System (MCMS). Therefore, we target the integration of a flexible User Profile Manager (UPM) [7] within the UE that dynamically conveys the user's environment information and the PQoS estimation of the IPTV service. The integration of the MCMS within the IMS application layer triggers service adaption dependent on the collected PQoS values. More specifically, we focus on the user's perceived quality of IPTV service by enabling PQoS awareness and adaptation enforcement within IMS. Furthermore, spatiotemporal activity in videos has a huge impact on the deduced perceptual quality as shown in [8]. We therefore propose a satisfying video quality threshold[9] that derives the right encoding parameters and optimizes the network usage while maximizing the end-user satisfaction.

Furthermore, this thesis defines a **seamless mobility management** that can be applied in any mobility use-cases. LTE proposes different mobility mechanisms depending on the nature of the mobility (i.e. macro, global mobility). IMS controls SIP media sessions independently of the underlying transport layer. IMS supports SIP mobility but involves the Correspondent Node (CN) in the signaling which adds further delay in the handover process. Interrogating-CSCF (I-CSCF) and Proxy-CSCF (P-CSCF) are the entry points of the IMS architecture handling signaling and media traffic for any transport layer. In our solution, this strategic element is chosen as the anchor mobility point within IMS architecture.

1.3 Thesis outline

The reminder of this thesis is organized as follows:

Chapter 2 "State of the art" presents the state of the art of the two NGN challenges targeted by this thesis. We first present the NGN architecture and specification. In this part, we introduce 3GPP IMS, Telecoms Internet converged Services Protocols for Advanced Networks (TiSPAN) IMS, LTE specification and the corporate networks which play a role in coverage extension. The second part is dedicated to context-awareness, User Profile (UP) modeling and QoE definition followed by the PQoS existing solution for VoIP and IPTV services. Then we introduce the mobility definition, explain the different types of handover and finish with a complete study on existing mobility solutions in NGNs.

In **Chapter 3 "A novel IMS-Based architecture achieving adaptation for IPTV"**, we propose a solution around a novel IMS-based architecture which has been commonly conceived through a FP7 European Project, called ADAMANTIUM [10]. Within this architecture, we introduce a management system towards innovative features of monitoring and adaptation to be integrated part of the IMS environment. This is made possible by a breakthrough element called Multimedia Content Management System (MCMS), associated with a new IMS client empowered by a flexible UP and a MSRF element which streams the video content. We focused on defining context-aware mechanism within IMS architecture in order to adapt dynamically video streams according to the PQoS estimated at the end user side. To achieve this, we define a new UP middle-ware integrated in the UE that allows efficient transmission of user environment characteristics. We integrate the MCMS modules that interact with the UE and the MSRF using SIP signaling. Content video dynamics have a great impact on the PQoS. A PQoS prediction model is then introduced to improve user satisfaction.

Chapter 4 "Seamless handovers in the IMS environment based on End- User's context" introduces a new mobility solution within IMS architecture improving the perceived quality of the user for the VoIP service. We define a Mobility-Aware P-CSCF (MA-PCSCF) which enforces the media switching from one network to another. A new jitter buffer algorithm is defined in the MN to leverage our new MA-PCSCF mobility mechanism. A live test-bed validates our mobility solution integrated to a smartphone roaming from WLAN to Universal Mobile Telecommunications System (UMTS) networks.

Finally, the conclusion summarizes the proposed solutions in this thesis and discusses ongoing and future works.

Chapter 2

State-of-the-Art&Discussions

2.1 Introduction

The increasing demand for high quality media services, anytime, anywhere, and the development of new technologies are driving the evolution of mobile communications. Traditional services offered only by the Internet are now possible with guaranteed QoS over technologies linked to a NGN platform. Additionally to that, consumers have high expectations concerning the quality and usability of the service. Recent enhanced services like interactive multimedia services tends to require QoE-enabled NGN in order to ensure user satisfaction for the provided service. Although QoE is very subjective in nature, it is very important that a strategy is devised to measure it as realistically as possible. The ability to predict and control the delivered QoE is facilitated by enabling context-awareness in NGN and by estimating PQoS along the delivery chain. The PQoS monitoring will give the provider some sense of the contribution of the network's performance. The service provider's main objective is to provide service to users without disruption. Decision taking can be carried out and enforced in order to maintain the service above acceptable PQoS thresholds. Service disruption or service unavailability leave the user frustrated. Another way to ensure user satisfaction within NGN is to allow MN to roam seamlessly through available access networks.

In the following chapter, we first present the NGNs and more specifically the IMS architecture and its integration within the following standard architectures : TiSPAN, LTE and corporate networks. Then we introduce the related work to context-awareness. Context and context-awareness definitions are given. Then different User Profile (UP) standards are described and more specifically the 3GPP Generic User Profile (GUP) and the User Data Convergence (UDC) concept. Then we introduce the QoE concept and the PQoS estimation for VoIP and IPTV. Finally, the vertical mobility issue in NGN is treated and summarized in a deep analysis of the different mobility solutions.

2.2 Next Generation Network

"NGN is a packet-based network able to provide telecommunication services and able to make use of multiple broadband, QoS-enabled or even QoE-enabled transport technologies and in which service-related functions are independent from underlying transport-related technologies. It enables unfettered access for users to networks and to competing service providers and/or services of their choice. It supports generalized mobility which will allow consistent and ubiquitous provision of services to users." [11]

2.2.1 3GPP IMS

IMS network aims to merge two of the most successful paradigms in telecommunications: cellular networks and Internet [5]. Besides, IMS delivers innovative multimedia services over fixed and mobile networks using open standards as depicted in Figure 2.1. It addresses key issues such as convergence, service creation and delivery, service interconnection and open standards. IMS can allow an operator to retain its existing business models, or evolve towards new ones.

The role of IMS is to provide a secure and reliable means for terminals and applications to reach, negotiate and communicate with each other. The architecture of IMS consists of a set of functions linked by standardized interfaces called "Referenced Points". 3GPP does not standardize the nodes, but the functions, this allow operators to have some freedom regarding hardware implementations. The IMS architecture is evolving across multiple 3GPP releases. For example, Release 6 includes WLAN access mechanisms; and Release 7 includes broadband/wire access capabilities. There are three distinct operational planes within IMS architecture : the application plane, control plane and the transport plane as depicted Figure 2.2.

The **application plane** contains a number of application server types. These

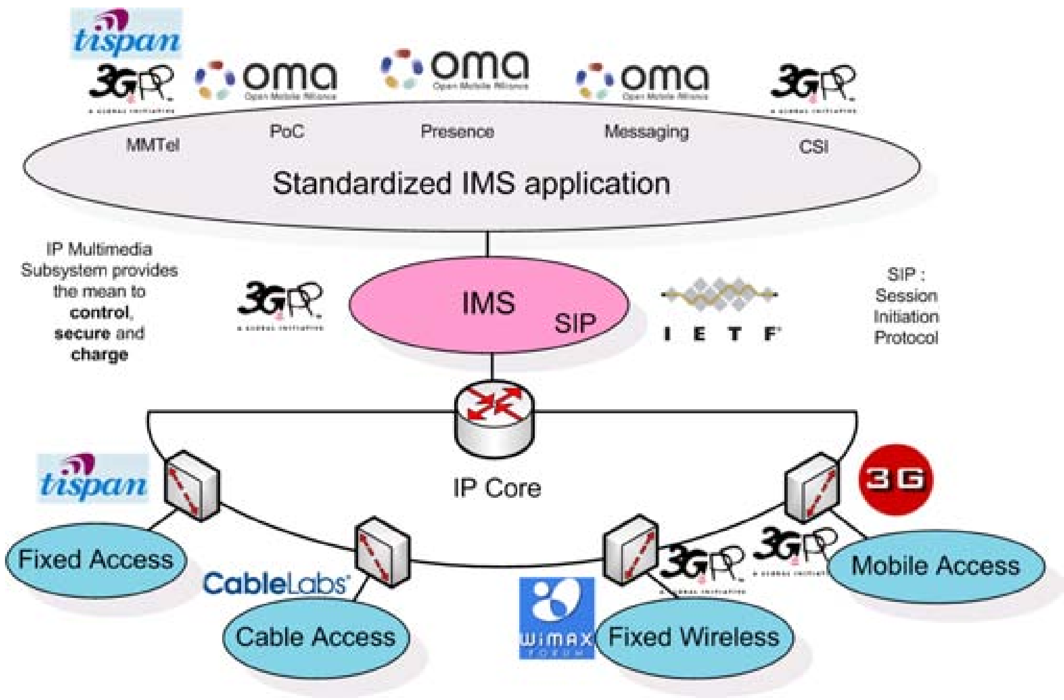


Figure 2.1: IMS Seamless Service Provision

are all SIP entities as expected within the IMS architecture. These servers host and execute services and can operate in a number of SIP functional modes i.e. SIP User Agent (UA), a terminating function; SIP Back-to-Back User Agent (B2BUA) which acts like two SIP UA or as a SIP proxy server.

The **control plane** also called the **signaling plane** deals with session signaling and comprises a number of distinct functions to process the signaling traffic flow such as the IMS core. In the IMS specification [3], the "Core" of IMS comprises two main nodes : the Home Subscriber Server (HSS) and the Call State Control Functions (CSCFs) which are the P-CSCF, the P-CSCF and S-CSCF. The Public Switched Telephone Network (PSTN) interworking functions are composed of the Media Gateway Control Function (MGCF), the Signalling Gateway (SGW) and the Breakout Gateway Control Function (BGCF). The resource element is mainly constituted of Media Resource Function Controller (MRFC).

The **user plane** also known as the **transport plane** or **media plane** transports the media streams directly between subscribers; and between subscribers and IMS media generating functions such as the Media Resource Function Processor (MRFP) acting as a media announcement server. The Media Gateway (MGW) is the media interworking element for IP/TDM media streams.

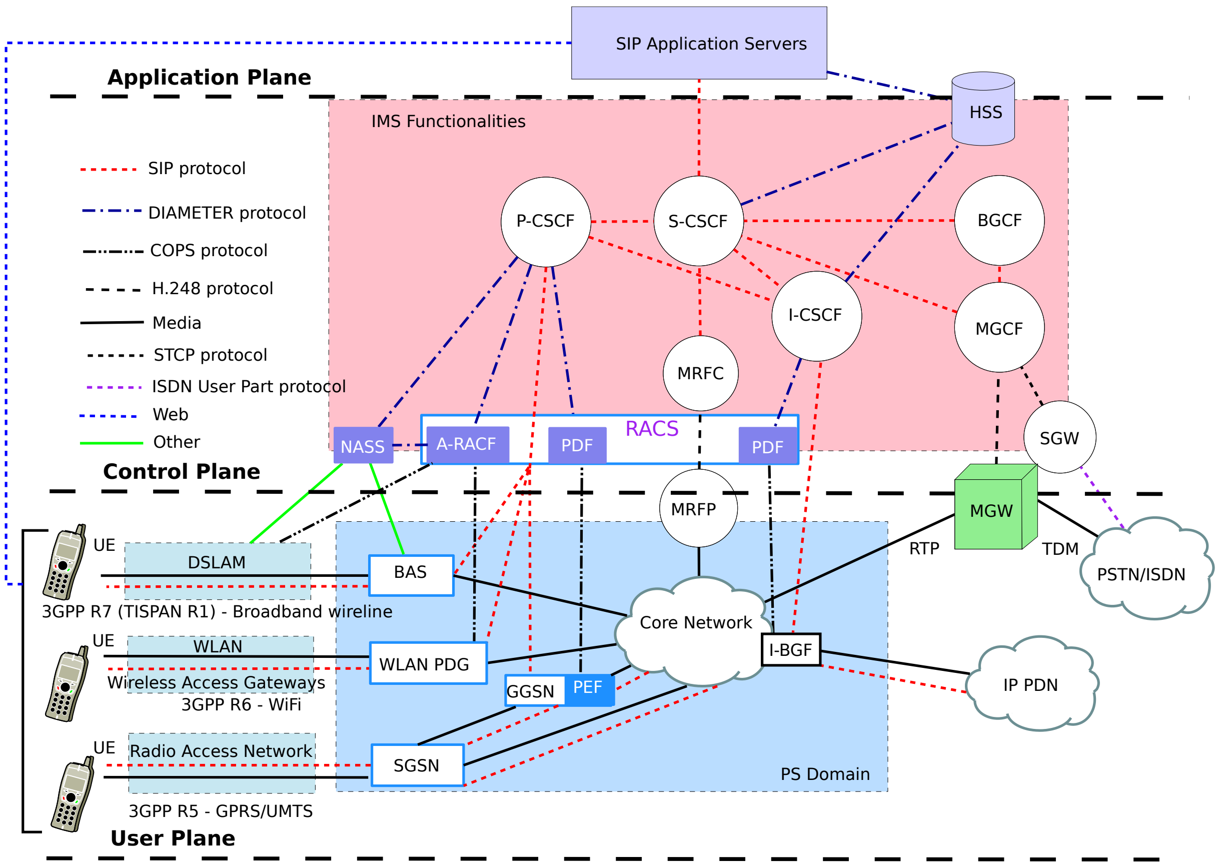


Figure 2.2: IMS Detailed Architecture with Gateways

The IMS control elements, composed of Call State Control Function (CSCF)s, are the heart of the IMS architecture and are used to process SIP signaling. The main function of the CSCF is to provide session control for terminals and applications using the IMS network. Session control includes the secure routing of the SIP messages, subsequent monitoring of the SIP sessions and communication with the policy architecture to support media authorization. The control plane controls the user plane traffic through the Resource and Admission Control Subsystem (RACS). This consists of the Policy Decision Function (PDF), which implements local policy on resource usage, for example to prevent overload of particular access links, Access-Resource and Admission Control Function (RACF), which control QoS within the access network, and Network Attachment SubSystem (NASS) which sends network attachment information and the subscriber access profile information to the A-RACF (reference point e4 in [12]). A description of each of CSCF functions is detailed below:

- The **Proxy-CSCF (P-CSCF)** is the first point of contact between the IMS terminal and the IMS network in the signaling plane. All the requests initiated by the IMS terminal or destined for the IMS terminal go across the P-CSCF. The P-CSCFis responsible for the security of messages between the network and the user and allocating resources for the media flow. It can be located in the visited network or in the home network. The P-CSCF acts as a SIP proxy server;

- The **Serving-CSCF (S-CSCF)** is the central node for the provisioning and the central brain of the IMS system. It is responsible for processing registrations to record the location of each user, user authentication, and call processing (including routing of calls to applications). The operation of S-CSCF is controlled by policy stored in the HSS;

- The **Interrogating-CSCF (I-CSCF)** is the contact point for all connections going to or coming from an external IMS network. It is responsible for querying the HSS to determine the S-CSCF allocated for the user and may also hide the operator's topology from peer IMS networks.

  The main **databases element** is the Home Subscriber Server (HSS) which contains user and subscriber information to support entities handling calls and sessions. This component provides Authentication Authorization and Accounting (AAA) functionality and unique service profile for each user handling mobility management (keeping track of which session control entity is serving the user). It is an evolution of the Home Location Register (HLR) in a Global System for Mobile communications (GSM) system. The HSS contains all the data subscriptions required to handle multimedia sessions. These data include, location information, security information, user profile information and the S-CSCF allocated to the user. When a user registers in the IMS domain, the user profile (relevant information related to the services to be provided to the user) is downloaded from the HSS to the CSCF. For session establishment, HSS provides information on which CSCF currently serves the user.

The **control plane interworking elements** are defined below :

- The **Media Gateway Control Function (MGCF)** controls the MGW media flow and the SGW using a Stream Control Transmission Protocol (SCTP).

MGCF and SGW collectively represent equipment that provides interworking with the PSTN;

- The **Breakout Gateway Control Function (BGCF)** identifies if a session terminates on the PSTN and determines which MGCF should handle it;
- The **Signalling Gateway (SGW)** acts as a signaling gateway, which performs protocol conversion between ISDN User Part and SIP.

The resource elements such as Media Resource Function (MRF) provide media services in the home network and implement functionality to manage and process media streams such as voice announcements, voice mixing (for conferencing), video, text-to-speech, and real-time transcoding of multimedia data. Each MRF can be divided into a Media Resource Function Controller (MRFC) and Media Resource Function Processor (MRFP) respectively control and process media stream resources.

Media Gateway (MGW) performs the actual switching for user data and so is responsible for providing the interworking of media flows between different networks. It provides interworking between the different media transport formats, Real-Time Protocol (RTP)/User Datagram Protocol (UDP)/Internet Protocol (IP), TDM flow from circuit-switched network, as well as media transcoding of voice and video, if required. It interacts with MGCF for resource control.

An **Application Server (AS)** hosts and executes services and can run in a number of classical SIP operational modes. AS accepts SIP requests and responses and ables to control, finish or initiate a new SIP-transaction such as SIP UA or B2BUA. It can route the session towards another user or network (Proxy-SIP mode) and interact with other service platforms for the support of services. The AS are attached to the Serving-CSCF (S-CSCF)s to host and serve IMS services and communicates with HSS in order to obtain information about subscriptions and services.

Even though IMS is sometimes considered too expensive and complex, it is the only standardized architecture that enables convergence between different kinds of access networks. In fact, many IMS elements are gateways interfacing non-IP based networks or systems. For this reason, telecommunication manufacturers have introduced the “light IMS” approach to provide operators with small scaled or simplified IMS architecture.

2.2.2 TiSPAN IMS

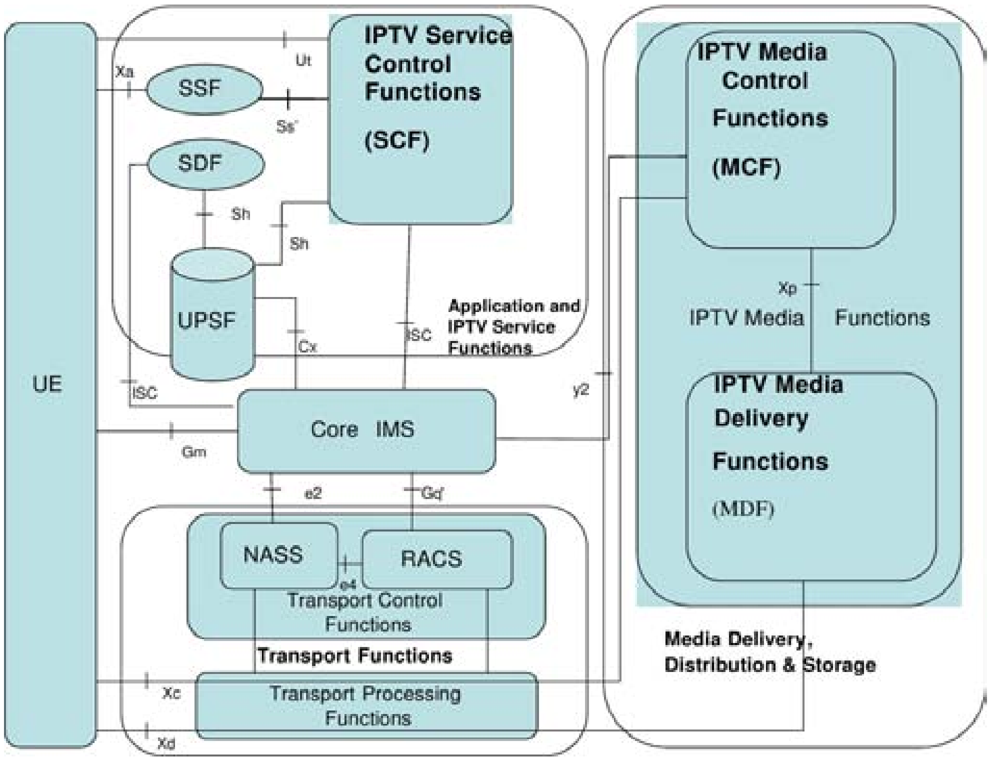


Figure 2.3: IPTV TiSPAN architecture

Telecoms Internet converged Services Protocols for Advanced Networks (TiSPAN) IMS is the standardization response to the NGN evolution of fixed access. TiSPAN [13] reuses IMS core network and extends 3GPP standardization with IPTV services.

Several specifications of TiSPAN NGN Release 2 and 3 address the integration of IPTV in NGN. The specifications consider IPTV in terms of service requirements, functional architecture, including definition of functions, communication protocols, reference points and implemented procedures and communication flows. The specification in [13] is more generic and considers NGN subsystems in general and the one in [14] is IMS-specific and addresses the IMS-based IPTV architecture that relies on IMS for the session control. Figure 2.3 depicts the different functional entities which compose the IMS-based IPTV architecture. Most of them fit the ones defined in NGN integrated IPTV subsystems. IPTV services execution involves the IPTV Media function that includes the Media Control Function (MCF) and the Media Delivery Function (MDF), as well as the Service Control Function (SCF) that is in charge of service management. The main function of the latter consists in service authorization during session initiation and session modification, including checking IPTV users' profiles in order to allow or deny access to the service. The IMS user profile and the IPTV specific profile data are held by the UPSF. The Service Delivery Function (SDF) and Service Selection Function (SSF) are functions for providing information necessary to the UE to select an IPTV service. The communication between the UE and the SCF for session management purposes is transferred via the Core IMS. The Ut reference point can also be used for the purpose of service profile configuration. Media Control messages are exchanged between the UE and the MCF via the Xc reference point in order to control the media flow, and Media Data is exchanged between UE and MDF via the Xd reference point to deliver it. Figure 2.4 is the simplified IMS TiSPAN architecture for UMTS access on which our first contribution is based in chapter 3.

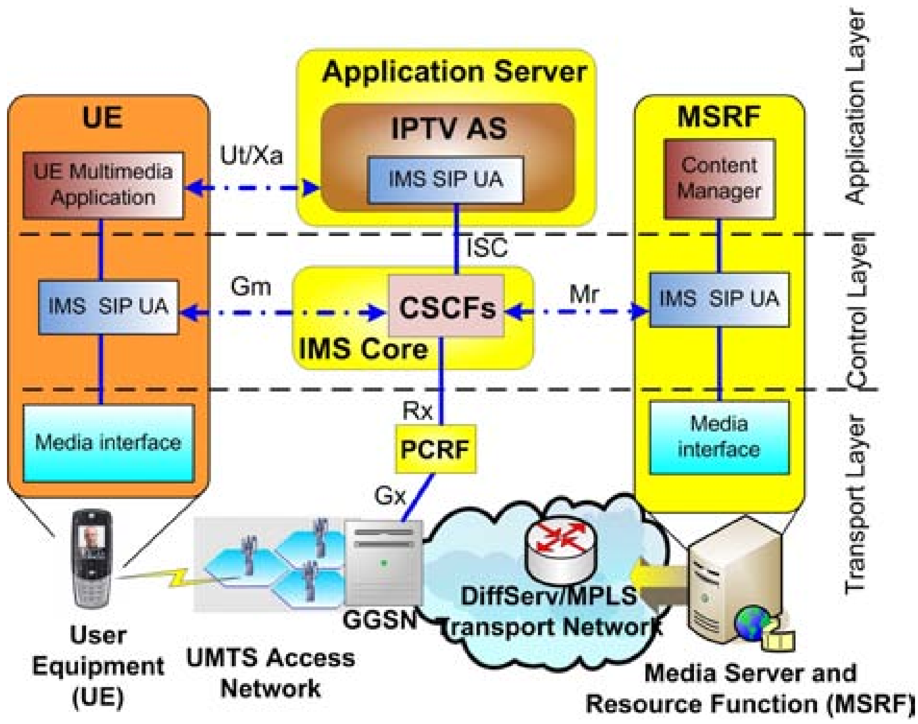


Figure 2.4: simplified IMS TiSPAN Architecture

2.2.3 LTE Interfaces

LTE is a wireless communication framework designed by the 3GPP to substitute existing UMTS and GSM technologies and provide high-speed data for mobile phone and terminals.

The LTE [15] architecture is called EPS and his composed of E-UTRAN [16] and an Evolved Packet Core (EPC) as depicted in Figure 2.5. The EPS specification provides support for non-3GPP access technologies (i.e. WiMAX, Corporate networks) making these technologies able to interwork with the 3GPP specified EPC as discussed in subsection 2.4.4.1. LTE uses Orthogonal Frequency-Division Multiple

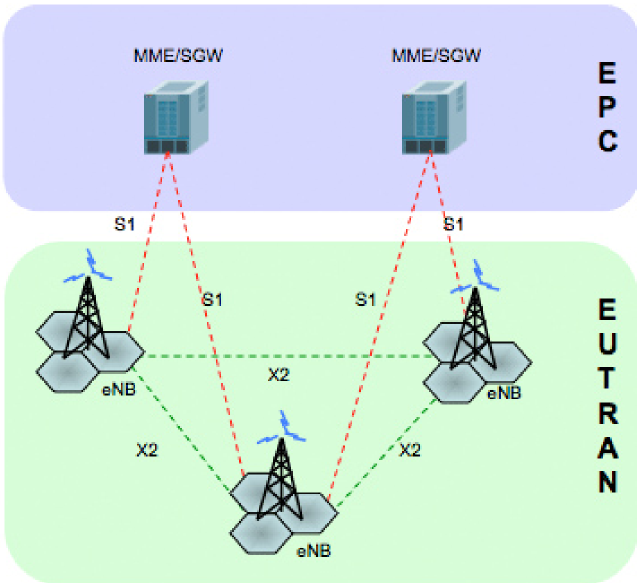


Figure 2.5: Evolved Packet System (EPS) Architecture

Access (OFDMA) radio technology to meet LTE requirements for spectrum flexibility and enables cost-efficient solutions for wide carriers with high peak rates. LTE requirements are designed and achieved only for Packet Switched Networks. Thus, technical challenges arise, particularly for voices services. VoIP services are both delay and packet loss sensitive. Voice over LTE (VoLTE) requires strong evolution in voice delivery, abandoning well established and reliable legacy voice services.

The latest version of LTE specifies interconnection with non 3GPP networks. Nowadays, Corporate Networks and Residential Networks play an important role in telecommunication. According to [1], *31 percent of smartphone traffic was offloaded onto the fixed network through dual-mode or femtocell in 2010"*. We strongly believe that Corporate Networks have a huge impact on business today. Effectiveness of business communication within or outside the corporation drastically decreases the costs of the company and optimizes revenues. Corporate Networks are evolving towards "all IP" also enabling interconnection with NGNs. Residential Networks with broadband connection are considered as a basic Corporate Network. Home automation also represent a huge interest in industry enabling a centralized control of energy, security and communication within the home.

2.2.4 Corporate Networks and WLANs

2.2.4.1 Introduction

A corporate network is a closed and private network known as an "Enterprise Network" that provides facilities for seamless communications, processing and storage resources within a single corporation. In order to carry out business functions, corporations have traditionally employed a dedicated network to provide business services like telephony and faxing [17].

2.2.4.2 Corporate Network evolution

The evolution of corporate networks has been driven by the necessity of having cost-effective and reliable telecommunication networks anywhere, anytime. With the globalization of business, working style has changed increasing the mobility of business users. The need for communication between different countries from several locations has involved expensive dedicated resources. In order to face communication expenditure, an attractive solution is to use the corporate network combined with new technologies such as VoIP. Indeed, the evolution from circuit switched to packet switched communications in the corporate network has been one of the major steps that allows that use of the same packet switched circuit for both voice and data. This evolution unifies business services and Computer Telephony Integration (CTI) application. CTI is the technology that integrates or co-ordinates telephony and computer networks. This integration enables enterprises to combine strength of telephony and data to gain a competitive edge. Advanced CTI applications can offer powerful business enhancing solutions such as Interactive Voice Response (IVR) and Customer Relationship Management (CRM).

IVR system is a phone technology that allows a computer to detect voice and touch tones using a normal phone. An IVR can play announcements and request an input from the caller. This information can be used to route the call to a particular

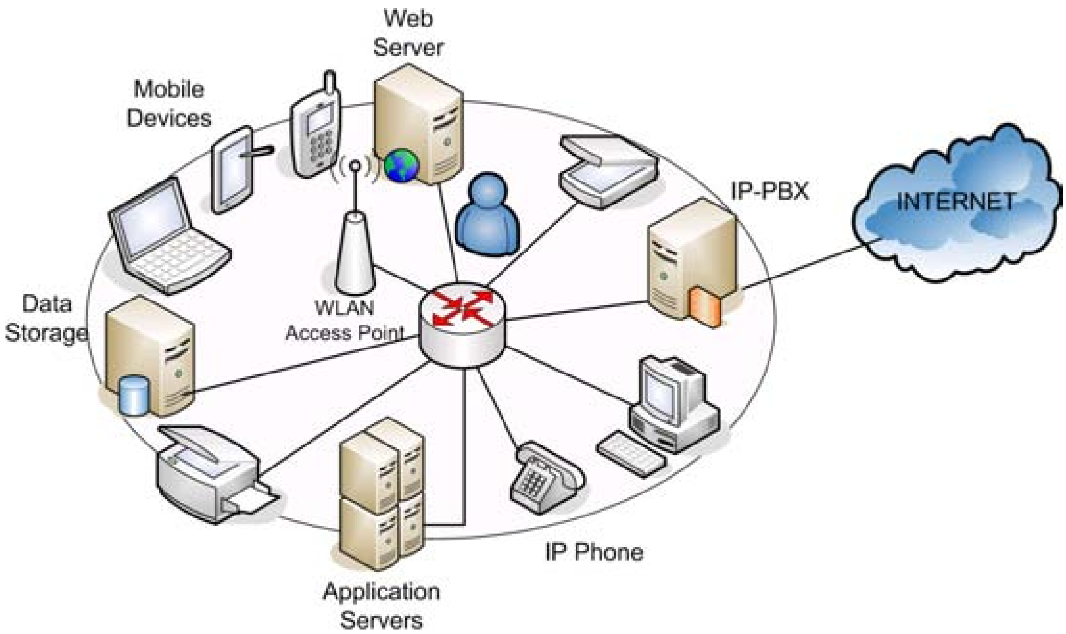


Figure 2.6: Corporate Network Overview

skill-set which is function applied to a group of agents with a particular skill. IVR systems typically used to service high call volumes, reduce cost, extend the business hours of operation and improve customer experience.

CRM is a term used by companies to manage their relationship with their customers, including the capture, storage and analysis of customer information. Each company has a different approach in dealing with its customers concerning marketing, sales and service departments. CTI application can trace and store information about their customer's phone calls and analyze its data to plot surveys.

Figure 2.6 shows an example of a corporate network overview. A user in the corporate network accesses a list of services defined according to the policies in the corporation. Services offered can be multimedia and data such as : voice mail, conference, IVR, Automatic Call Distribution (ACD), data transfer, mail, fax, web-based applications, database access, application access and others.

2.2.4.3 Internet Protocol Public Branch eXchange (IP-PBX)

An IP-PBX is a Public Branch eXchange working over the IP layer. A Public Branch eXchange (PBX) is a telephone exchange that serves a particular business or corporation as opposed to the common carrier or telephone company that operates in the public domain. The aim of the traditional PBX is to give privacy to the corporate telephony network by switching calls between the private organization to the Public Switched Telephone Network via trunk lines. The main advantages of PBX are cost savings on internal phone calls and customization of telephony functionalities. It performs call processing functions like call answering, dialing, and transaction features such as hold transfer, call forward, conference, call parking. The understanding of packet switching in telephony networks reduces hardware costs and energy consumption. An IP-PBX can be installed on a reliable existing server with a stable Internet connection. IP-PBX can also interface the existing traditional telephones through Peripheral Component Interconnect (PCI) cards.

2.2.4.4 Mobility in corporate networks

In order to support mobility in corporate networks, a common trend is the implementation of wireless access with low cost technologies like Institute of Electrical and Electronics Engineers (IEEE) 802.11 WLAN having mobility. VoIP has become a common application enabling users to make phone calls in data networks with a significant reduction in costs. Wireless networks in a convergence scenario contain a mixture of heterogeneous access networks. The increasing business advantages of having connectivity and service availability anytime, anywhere, imply that mobile devices in corporate networks should be capable to attach these different access technologies. The support of multiple radio access technologies is achieved through multi-mode devices and Software Defined Radio (SDR). The first solution includes several network interface cards and a mechanism to switch them, whereas the SDR solution implements the radio functions as software in a common hardware platform.

2.2.4.5 WLANs accesses

A WLAN is a flexible data communication system design to support wireless connectivity to the local network. WLANs use electromagnetic airwaves (radio and infrared) to communicate information from one node to another. WLANs are increasing becoming an attractive alternative to licensed spectrum : *"Today, WLANs have increased and become more widely recognized as a general-purpose connectivity alternative for a broad range of business customers"* [18]. The range of wireless technologies available in the 802.11 standard vary between one to hundred meters. In a typical WLAN configuration, a transmitter/receiver (transceiver) device, called an

Table 2.1: Wireless Access Technologies

Standard name	*Access type*	*Status*	*Data rate*	*Cell radius*	*User type allowed*	*HO capability*	*Frequency band*
IEEE 802.11g/WiFi	WLAN	Available	54 Mbps	50-60m	Fixed (LOS and NLOS), nomadic	No	2.4GHz
IEEE 802.11n/WiFi	WLAN	Available	540 Mbps	60-100m	Local/Global	No	2.4GHz
IEEE HiperLAN/2	WLAN	Available	54 Mbps	50-60m	Fixed (LOS and NLOS), nomadic	No	5GHz
IEEE 802.16 WiMAX	WMAN	Available	36-135 Mbps (LOS), 75 MBps (NLOS)	Up to 70-80Km	Fixed (LOS and NLOS) nomadic	No	2-66GHz
IEEE 802.16e WiMAX	WMAN	Available	30 Mbps	Up to 70-80Km	Fixed (LOS and NLOS), nomadic, mobile	Yes	2-6GHz
IEEE HiperACCESS	WMAN	Available	25-100 Mbps	1.8-2.5Km	Fixed (LOS)	No	11-43.5GHz
IEEE HiperMAN	WMAN	Available	25 Mbps	2-4Km	Fixed (LOS and NLOS)	No	<11GHz
WiBro	WMAN	Available	18 Mbps	1Km	Fixed (LOS and NLOS), Nomadic, mobile	Yes	2.3-2.4GHz
HAP	WMAN WWAN	Available	Varies	Varies	Varies	Varies	28-31GHz 42-43 GHz
IEEE 802.20	WWAN	Available	16 Mbps	>15Km	Fixed (LOS and NLOS) nomadic, mobile, highly mobile	Yes	3.5GHz
IEEE 802.22	WMAN WWAN	Available	18 Mbps	40Km	Fixed (LOS and NLOS)	No	54-862MHz
Satellite(GEO)	WWAN	Available	Up to few Gbps	Four Satellites Global coverage	Fixed (LOS) Nomadic, mobile	Varies	4-8GHz (C Band), 10-18GHz (Ku Band), 18-31GHz (Ka Band), 37-50GHz (Q/V Band)
Satellite(MEO)	WWAN	Available	Up to few Mbps	11 Satellites Global Coverage	Fixed (LOS), Nomadic, mobile	Varies	As GSO satellites
Satellite(LEO)	WWAN	Available	Up to few Mbps	Varies	Fixed (LOS), Nomadic, mobile	Varies	As GSO satellites

Access Point (AP), connects to the wired network from a fixed location providing services available on the wired network. Table 2.1 from [19] enumerates different wireless technologies.

In 1997, the IEEE created the first WLAN standard called 802.11. Unfortunately, 802.11 only supported a maximum bandwidth of 2 Mbps which is too slow for most multimedia applications. IEEE expanded on the original 802.11 standard in July 1999, creating the 802.11b specification. 802.11b supports bandwidth up to 11 Mbps, comparable to traditional Ethernet.

802.11b works around the same radio signaling frequency (2.4 GHz) as the original 802.11 standard. Being an unregulated frequency, 802.11b node can incur interference from microwave ovens, cordless phones, and other appliances using the same 2.4 GHz range. However, by installing 802.11b antenna at a reasonable distance from other appliances, interference can easily be avoided. In the same time, the 802.11a standard has been developed to improve the 802.11b low data rate and to curb its inefficiency multi-user connectivity. 802.11a supports bandwidth up to 54 Mbps and signals in a regulated frequency spectrum around 5 GHz. Due to the higher signaling frequency, the range of 802.11a is shorter.

In 2002 and 2003, a new WLAN product appeared : the standard 802.11g takes advantage of both previous standards. It supports fastest maximum speed (up to 540 Mbps), more simultaneous users and has finally a bigger range. The main difference between 802.11a/b/g/n is the data and the physical layer.

Without strong QoS, the existing version of the 802.11 standard does not optimize the transmission of voice and video. As a result, in 2005, the 802.11e task group began to refine the 802.11 Medium Access Control (MAC) to improve QoS for better support of audio and video (such as VoIP) applications. Because 802.11e falls within the MAC Layer, it will be common to all 802.11 physical layers and be backward compatible with existing 802.11 WLANs. In addition, a simple firmware upgrade will enable existing access point to comply with the 802.11e standard.

In regards to security, IEEE proposes in 2004 802.11i also known as Wi-Fi Proctected Access II (WPA2) adding Advanced Encryption Standard (AES) key in the data transmission.

In WLANs infrastructure, multiple access points allow end-user to efficiently share local network resources linking WLAN to wired and long distance Wide Area Network (WAN)s and the Internet.

2.3 Context and Context-awareness

2.3.1 Context and Context-awareness Definition

2.3.1.1 Context Definition

The context in ubiquitous computing area is a generic term describing the environment of the user. In 1994, [20] proposes concepts of location, identity of people and objects surroundings the user and changes that may impact on him. [21] also identifies context information such as location, orientation, time, season, temperature, etc. [22] defines the context such as the user's emotional state, his interests, location and orientation, date and time, and the surrounding objects and people.

Another approach consists in defining the context by using synonyms such as context as the environment or the situation. [21] defines the context as all the components of the user's environment that are known by the user terminal. [23] sees the context as the environment state of an application. [24] describes the context as aspects of the current situations.

Actually, these definitions are very difficult to apply. Based on the first approach, the definitions are very specific. Thus, it is difficult to say, using this definition, whether a new information should be considered as part of the context or not. For the second approach, it is too broad and vague. By defining the context, for example environment, some studies identify it with the user environment, whereas others assimilate it more to an application environment.

[25] identifies three categories of contexts: computing context (resources used in physical access to the service), the user context (location, profile, etc.) and the physical context (temperature, light, noise, etc.). [26] extends the classification by including the "time" feature (hour, date, season, etc.), allowing to capture the temporal aspect of the first three categories and get a historical past that can be very useful for applications. [27] identifies, as relevant, context information associated with the user terminals (i.e. memory, CPU power, peripheral I/O, etc), with user's device network connectivity (i.e. bandwidth, delay, bit error rate, etc.) and with the user himself (i.e. profiles and preferences).

Even these extensions do not satisfy the whole situation relevant to an application and its set of users. In some cases, the physical environment may be important, while in others it may be completely immaterial.

Another generic definition expressed by [28] and widely adopted by the research community is that following sentence:

> *"Context is any information that can be used to characterize the situation of an entity. An entity is a person, place, or object that is considered relevant to the interaction between a user and an application, including the user and applications themselves."*

[28] derives the notion of situation and defines it as the description of states of relevant entities. Indeed, applications are often interested in the aggregation states of relevant entities for their implementation. The notion of situation has been taken in most systems sensitive to context.

This thesis relies on this last definition and tries to identify the relevant entities for the various actors involved of a large scale multimedia system. Our understanding of the context and the information that we consider relevant and useful in multimedia domain is presented in chapter 3.

2.3.1.2 Context Categories

As context covers a wide range of heterogeneous information that continues to grow with the advent of new services and technologies, it is useful to classify this information. Categories provided by [25] (where you are, who you are with and what objects are around you) only include location and identity information. To characterize a situation, activity or status and time information are necessary.

A very popular approach in the classification of context information is the one that distinguishes internal and external information about the user as described by [29].

Other studies have also targeted physical and logical context, measured and conceptual context [30] or physical and social context [21]. The physical (external) dimension refers to the context information measured by sensors such as location, brightness, temperature, etc. The logical dimension (internal), on the other hand, refers to information derived from monitoring of user interactions such as the user goal, its activity, its emotional states, etc.

Another widely used simplification in context-aware-systems consists in separating the dynamic context information that changes over time, from static context information. This does not necessarily mean that the latter ones never change but they do not change during the service consumption. This classification is mainly used in multimedia systems where time is critical.

[31] distinguishes three main entities:

- Places (region or geographical location i.e. office, building, street, etc.)
- Persons (individuals or groups of individuals being in the same location or different locations);
- Objects (physical object or software component, i.e. an application, a document, etc.)

Each of these entities is described by a set of attributes that, in turn, can be classified into four different categories:

- Identity: each entity has an unique identifier in the space of names used by the application;

- Location: in addition to the position, the location includes information such as orientation, or altitude as well as any information that can be used to infer spatial relationships between entities as co-location, the capacity or proximity;

- Status (or Activity): it is any intrinsic characteristic of an entity that can be captured. For the entity "places", these features represent the temperature or noise that can be measured. For the entity "person", they correspond to its state or its business; Objects (physical object or software component, i.e. an application, a document, etc.)

- Time: it is used to timestamps situations. This context information provides track-records that are often very useful to applications. The information "time" is often used with other context information such as stamp or to set the time interval in which they remain valid.

2.3.1.3 Context-awareness definition

Once the context information is identified, an important task is to define how information will be useful for applications. The notion of Context-Awareness, has also been given a number of definitions in the literature. In [20], Schils and Theimer define the notion of context-awareness as the ability for an application to discover the changes that may occur in the environment in which and to react accordingly. From the perspective of adaptation to the context, the above definition is included in [25], defining context-aware applications as the examination of the environment and adaptation according to the changes that may occur, such as the location of use or nearby people or resources. The authors identify four categories of applications as sensitive to the context:

- Selection depending on the proximity: it is a user interface technique, exploiting the location information to highlight the closest resources thereby facilitating their selection;

- Automatic reconfiguration based on the context: it is the process of adding a new component, removing existing components or modification of relations between different components;

- Information and contextual commands: they consist in producing different results depending on the context in which they are issued;
- Actions triggered at the context change: they are simple IF THEN rules, which role is to guide the adaptation process.

On the same approach, in [32], Dey and Abowd propose to classify the functionalities of a context-aware application into three categories. These are:

- The presentation of information: this category refers to applications that either present the context information to users or uses the context to offer to users actions according to it;
- The automatic execution of services: it describes applications that trigger commands or reconfigure the system, transparently to the user, depending on the context information;
- The storage and enrichment of context information: it refers to applications that enrich the context information with metadata and which perpetuate it for future use.

According to Razzaque et al., the context-awareness is a term that comes from computing to designate terminals having the knowledge of the circumstances in which they are used and, consequently, reacting accordingly. The authors add that the sensitivity to context implies the development of applications able to acquire the context and a dynamic behavior depending on it.

In [31], Dey summarizes the above definitions in a simple and generic definition, adopted by the research community, which states that a system is context-aware if it uses context to produce information and / or services relevant to the user. The relevance depends on the tasks requested by the user.

2.3.2 User Profile

User Profile Modeling is based on markup scheme-based models which are characterized by a hierarchical data structure. The context information is organized into elements identified by their tags, which are associated with attributes and contents. Recursively, an element can itself contain other elements. These models are often used as the standard for user profiling. Below are some examples of user profiles defined in different networking areas:

1. GUP : The GUP [33; 34; 35] has been defined by the 3GPP for harmonizing the usage of user-related information coming from different entities. The standard does not impose any classification of the information to be included in the profile but recommendations consist in the following:

    - Information on the user subscription such as service offering and accessibility ;
    - General information on the user himself, including his name, address, preferences and currently active profile;
    - Information related to Public Land Mobile Network (PLMN) such as the GPRS parameters and the favorite access technology;
    - Security policies for some services, for example the localization service;
    - User information specific to a given service, such as relevant information for service personalization, and service access (key, certificate, password, etc.);
    - Information related to the terminal device such as its hardware characteristics, interfaces and supported services;
    - Other information, for example accounting.

    A profile consisting of several components is defined for each user. The components can be managed and stored in different network nodes to which the user is subscribed, as well as in external service providers. All profiles must have the same structure, defined in a W3C XML Schema [36]. The component can be defined in external schemas, referenced by their namespaces in the element `<xsd: schema>`, and then imported into the global schema using element `<xsd: import>`.

2. Moving Picture Expert Group (MPEG) standards : In order to ensure multimedia interoperability, two standards are defined by the MPEG, namely the MPEG-7 [37] also known as Multimedia Content Description Interface for MPEG and the MPEG-21 [38]. The structure of these standards is based on

the W3C XML Schema. MPEG-7 provides tools for describing the multimedia resources. The MPEG-7 description includes various information on the multimedia content such as its classification, creation (title, creators, etc..), usage (history of use, copyright, etc..), storage (format, encoding, etc..), as well as structural aspects (spatial components, temporal or spatio-temporal content), conceptual aspect (objects, events, etc..) or some low-level characteristics (colors, textures, etc.). Thanks to these descriptions, MPEG-7 allows the indexation of these multimedia resources strongly based on the content, and, consequently, more effective search and discovery based on criteria such as the content type and the involved person/object. From a structural point of view, a MPEG-7 description is composed of basic elements called descriptors (D) used to describe the characteristics of multimedia content. The relationships between these descriptors are then described by Description Scheme (DS). The syntax of description tools is defined by the Description Definition Language (DDL), which is an extension of the W3C XML Schema. In order to support efficient storage, exchange and processing of MPEG-7 descriptions, the standard also offers several tools for encoding these descriptions in binary form or for synchronizing the descriptions with the content they describe, etc.

MPEG-21 aims to allow users to access, consume, share or more generally to manipulate multimedia content in an efficient, transparent and interoperable way. For this, the standard defines an open architecture that covers the entire distribution and consumption chain of multimedia contents. MPEG-21 relies on two innovative concepts: the Digital Item (DI) which is the basic unit involved in a transaction (audio, video, image, etc.) and the User that interacts (publishes, issues, etc.) with the MPEG-21 architecture. The MPEG-21 standard covers several independent aspects such as the identification, declaration, adaptation and streaming of a Digital Item, the protection of intellectual properties, session mobility, etc. The Part 7 of the standard, called Digital Item Adaptation (DIA), provides the necessary tools to adapt the DI and enable their universal access. These tools are classified into eight categories, namely Usage Environment Description (UED), BSD (Bitstream Syntax Description), BSD Link, Terminal and Network Quality of Service, Universal Constraints

Description (UCD), Metadata Adaptability, Session Mobility and DIA (DI Adaptation) configuration.

3. The Composite Capabilities / Preference Profiles (CC/PP) [39] is a standard proposed by the W3C for the representation of user profiles. The Composite Capabilities/Preference Profiles (CC/PP) profiles allow the description of user preferences and terminals, especially mobile devices that are used to access the service. For this, the standard defines necessary tools for modeling the context of service consumption in order to enable the service adaptation. The CC/PP profiles are described by Resource Description Framework (RDF) triples [40], serialized as XML documents for communication purposes. A profile is then represented as a two levels-tree structure in: one or more components associated with one or more attributes, with the possibility for each component to reference by default a set of external attributes. These components are represented by resources ccpp:Component and are connected to the root node of the profile using the property ccpp:component. At the second level, each component is then described by a labeled sub-tree. The labels represent the attributes describing the characteristics of the component and the leaves represent the values of these attributes. The default attributes (usually defined in external RDF documents and identified by their URIs) are referenced using the property ccpp:default. For the extensibility purpose, the components and attributes contained in the CC/PP profiles are not defined by the standard. The vocabularies that are specific to different areas are defined in separate standards based on the RDF Schema [40]. An example of the vocabulary for describing the WAP terminal profiles is the WAG UAProf [41] proposed by the WAP Forum. As shown in several studies [42; 43], the CC/PP profiles have several limitations due to their structure defined in the standard, and due to some missing features in RDF such as the cardinalities. Different profiles are constructed based on CC/PP in order to overcome these limitations and to complete the vocabulary defined by CC/PP and UAProf, for example, Comprehensive Context Profiles (CSCP) [44] or CC/PP Context Extension [42]. Markup scheme-based model are mostly adopted to represent context. There exist many standards and tools (parsers, validation tools, etc.) that are based on it especially with the generalization of the use

of XML within web services tools and standards. In addition, the hierarchical structure on which this model is built fits well the decomposition nature of content information. However, existing standards are always targeting a specific type of applications and context domain. For example, MPEG is specific to User and Content, CC/PP is user and device oriented, etc. None has reached a full generic solution capable of meeting all requirements for a generic User Profile. Another limit of these models is that they only model the syntax of context information and lack for semantic. There is no way to represent meta-information or to model the relationship that may exist between the context information. Consequently, there are no possibilities to reason on context data.

2.3.2.1 GUP

The 3GPP Generic User Profile (GUP) [33] aims to provide a conceptual description to enable harmonized usage of the user-related information located in different entities. Technically, the GUP provides an architecture, data description and interface with mechanisms to handle the data. In this section, we focus on the GUP architecture and interfaces (the GUP content and structure is presented in section above).

The GUP architecture [34] consists of the following functional entities, as shown in Figure 2.7:

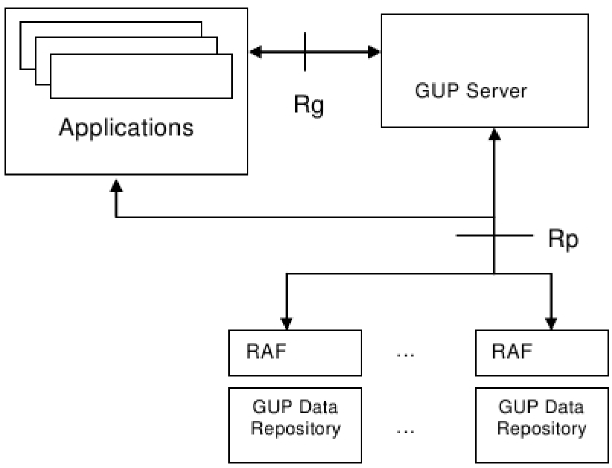


Figure 2.7: GUP architecture

A. GUP Server : The GUP Server is a functional entity providing a single point of access to the Generic User Profile data of a particular subscriber. The GUP Server includes the following main functionalities:

- Single point of access for reading and managing generic user profile data of a particular subscriber.
- Location of Profile Components.
- Authentication of profile requests.
- Authorization of profile requests.
- Synchronization of Profile Components.

In addition to proxying the requests (or handling them by itself), the GUP Server may also apply the redirect mode of operation for applications that support it. This implies that the GUP Server responds to the request with the redirection information such as redirection address and authorization assertions. Redirection can be made with Create, Delete, Modify, Query and Subscribe procedures.

B. Repository Access Function (RAF) : The RAF realizes the harmonized access interface. It hides the implementation details of the data repositories from the GUP infrastructure. In addition, the RAF may take part in the authorization of access to such GUP information, under the control of the RAF.

C. GUP Data Repositories : Each GUP Data Repository stores the primary master copy of one or several profile components. It is assumed that the RAF and the GUP Data Repositories are usually co-located in the same network element. The GUP Data Repository may also contain the authorization data depending on the authorization model and architecture.

D. Rg and Rp reference points : The Rg reference point allows applications to create, read, modify and delete any user profile data using the harmonized access interface. This reference point supports also third party profile access. There are means to authorize all requests and protect the user's privacy in all operations. The defined procedures applied in the Rg reference point between the applications and the GUP Server are: Create, Delete, Modify, List, Query, Subscribe, Unsubscribe and Notify.

The Rp reference point shall allow the GUP Server or operator's own applications to create, read, modify and delete user profile data using the harmonized access interface. Rp is an intra-operator reference point. External applications and third party GUP data repositories shall be connected to the GUP Server only using the Rg reference point. The Rg and Rp reference points carry user related data, and therefore shall be protected by security mechanisms to protect the user's privacy.

E. Applications These are the Application Servers and applications that need access to GUP data components. They may host some GUP data components themselves and may act as RAFs. Third party applications belonging to external security domains shall use a discovery service in a secured way to discover the GUP Server.

2.3.2.2 IMS - User Data Convergence

UDC [45] concept supports a layered architecture separating the data from the application logic, so that user data converge from where it belonged to a logically unique User Data Recovery (UDR), managed and accessed to in a common way. Convergence in data model avoids data duplication and inconsistency, overcomes the data capacity bottleneck of a single entry point, simplifies the overall network topology and interfaces and consequently simplifies the development and deployment of new integrated services through a common and unified set of user data that are up until now, scattered in several domains (i.e. PS, CS, IMS) and different network entities (i.e. HLR, HSS, Application Servers) of the current 3GPP system.

The UDR is the functional entity that acts as a single logical repository of user data and is unique from application Front End's perspective. Applications Front End (FE) are functional entities such as HLR/HSS/AUC, Application Servers, Access Network Discovery and Selection Functions in the Home Network (H-ANDSF), etc. that access the user data stored in UDR according to the UDC Information Model detailed. Application FEs are only able to access user data after authentication and authorization. Application FEs should then support common security algorithms and keys.

UDR provides unique reference point called Ud that allows different FEs to create, modify and delete user data. Ud should also support the subscription/notification functionality which allows a relevant FE to be notified about a specific event which may occur on a specific user data in UDR and transaction. More information on the Ud's procedures and flow can be found in [46].

2.3.3 Quality of Experience

QoE is the level of satisfaction of a consumer after experiencing a service. QoE is a wide expression concerning the usage of the service such as service customization according to user preferences, perceived quality and simplicity of service use.

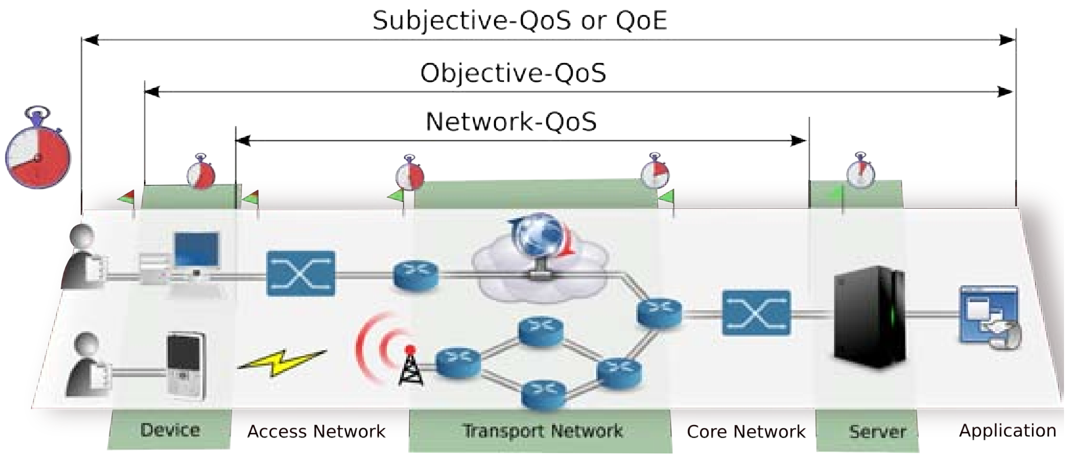


Figure 2.8: Different definition of QoS

Although the term QoS has been widely used in both traditional and newer IT services, there are several definitions of QoS available. Most of them agree on considering Quality of Service (QoS) as "a set of characteristics related to the performance of the elements that provide the services that have an effect into final end users perception". Thus, from a customer perspective, the concept of subjective-QoS also called QoE should be clearly related to the user satisfaction relative to the service consumption, beyond classical technical parameters such as Network-QoS also called Network Performance Key Performance Indicator (NP KPI) as depicted in Figure A.1. The effect of performance into satisfaction is usually described as PQoS. This concept of PQoS, although basically customer-centric, is significantly differentiated by the type of delivered service, since different performance situations will have different impacts on satisfaction. Then, multiple models to map NP KPIs into PQoS related satisfaction levels have to be developed for each service. Furthermore, the way that a user tolerates a specific level of PQoS is usually referenced in the bibliography by the term QoE. The QoE is a subjective measure from the user's perspective of the overall value of the services provided. Apart from its user-dependent nature, it is influenced by the user's terminal device, his environment, his expectations, the nature of the content and its importance. Although QoE is perceived as subjective, it is the only measure that counts for customers of a service and should therefore be the target of any management system. Whereas, PQoS values are derived from objective measurements emulating subjective satisfaction measurements. PQoS methods are being trained and tuned from subjective tests. Mean Opinion Square (MOS) is the most widespread used metric for subjective tests. The MOS method is based on the analysis of the customers' perceptual opinion about one service. Since subjective evaluations directly depict the quality perceived by the user, they have been used for protocols, transmission lines and coding algorithm evaluation as well.

2.3.3.1 Subjective Quality Evaluation Methods for Audio

Subjective voice quality tests are carried out by asking people to grade the quality of speech samples under controlled conditions as set out in the ITU-T P.800. This includes listening-opinion tests and conversation-opinion tests. The recommended method for listening-opinion tests is Absolute Category Rating (ACR) in which subjects only listen to a test speech sample and grade the voice quality on a five-point scale from 5 to 1 (excellent, good, fair, poor, and bad). Other listening-opinion tests include Degradation Category Rating (DCR) and Comparison Category Rating (CCR). In CCR tests, a pair of speech samples including reference and degraded samples without a particular order are presented to subjects who will judge the quality of the second sample to the first one at a seven-point scale ranging from -3 (much worse), -2 (worse), -1 (slightly worse), 0 (about the same), 1 (slightly better), 2(better), to 3 (much better). Listening-opinion tests are normally carried out in a controlled environment (i.e. in a soundproof room). The MOS is obtained by averaging individual opinion scores for a number of listeners (i.e., from 32-100). The suggested speech samples for testing (International Telecommunication Union (ITU), 1998) are normally 10-30 seconds consisting of several short sentences spoken by male and female speakers. Conversation-opinion tests required two subjects seated in two separate soundproof rooms/cabinets to carry out a conversation test on selected topics. Subjective tests that are based on the MOS reflects an overall speech quality which is an opinion score normally given by a subject at the end of a tested speech sample. A MOS score reflect the average as voted to different listeners of a voice sample.

2.3.3.2 Subjective Quality Evaluation Methods for Video

The subjective test methods, which have mainly been proposed by ITU and Video Quality Experts Group (VQEG), involve an audience who watches a video sequence and scores its quality as perceived by them, under specific and controlled watching conditions. Afterwards, the statistical analysis of the collected data is used for the evaluation of the perceived quality. The MOS is regarded as the most reliable method of quality measurement and has been applied on the most known subjective techniques. Subjective test methods are described in ITU-R Rec. T.500-11 (2002), ITU-T Rec. P.911 (1999), and ITU-T P.920 (2000) suggesting specific viewing conditions, criteria for observers and test material selection, assessment procedure descriptions and statistical analysis methods. ITU-R Rec. BT.500-11 described subjective methods that are specialized for television applications, whereas ITU-T Rec. P.910 is intended for multimedia applications and ITU-T P.920 for interactive applications.

The most known and widely used subjective methods for grading videos are:

- **Double Stimulus Impairment Scale (DSIS)** or **DCR** (ITU-T P.911) : observers are shown multiple references and degraded paired samples on one screen. The reference signal is always shown first. Scoring is on an overall impression scale of impairment: imperceptible, perceptible but not annoying, slightly annoying, annoying, and very annoying. This scale is commonly known as the 5-point scale with 5 being imperceptible and 1 being very annoying.

- **Single Stimulus (SS)** or **ACR** (ITU-T P.911) : multiple degraded separate samples are shown with no repetition or with multiple repetition. Three different scoring methods are used:

  - Numerical: an 11-grade numerical scale, useful if a reference is not available.

  - Adjectival: the aforementioned 5-grade impairment scale, however half-grades may be allowed.
  - Non-categorical: a continuous scale with no numbers or a large range, i.e. 0 - 100.

- **Stimulus Comparison (SC)** or **Pair Comparison (PC)**: Usually accomplished with two well matched monitors, where the differences between scene pairs are scored in one of two ways:

  - Adjectival: a 7-grade, +3 to -3 scale labelled: much better, better, slightly better, the same, slightly worse, worse, and much worse.
  - Non-categorical: a continuous scale with no numbers or a relation number either in absolute terms or related to a standard pair.

- **Single Stimulus Continuous Quality Evaluation (SSCQE):** the viewers watch the test video of typically 20-30 minutes without viewing the reference video. The viewer using a slider continuously rates the instantaneously perceived quality on scale from a scale of 0 to 100 ('bad' to 'excellent').

- **Double Stimulus Continuous Quality Scale (DSCQS):** this method is an extension of SSCQE but the viewers watch multiple pairs of quiet short (10s) reference and test sequences. Viewers are asked to assess the quality of both videos contrary to the DCR method where the impaired is evaluated according to the reference one. The viewers are not aware of the reference/test order and they are asked to rate each of the two separately on a continuous quality scale namely ranging from "bad" to "excellent", which corresponds to an equivalent numerical scale of 0 to 100. This method is usually used for evaluating slight quality differences between the test and the reference sequence.

- **Simultaneous Double Stimulus for Continuous Evaluation (SDSCE):** observers are shown reference and degraded sequences at the same time for a longer period on one screen. This method is used to evaluate the effect of sparse impairments such as transmission errors. Observers know which sequence is the reference and tested one. Viewers are requested to check the differences

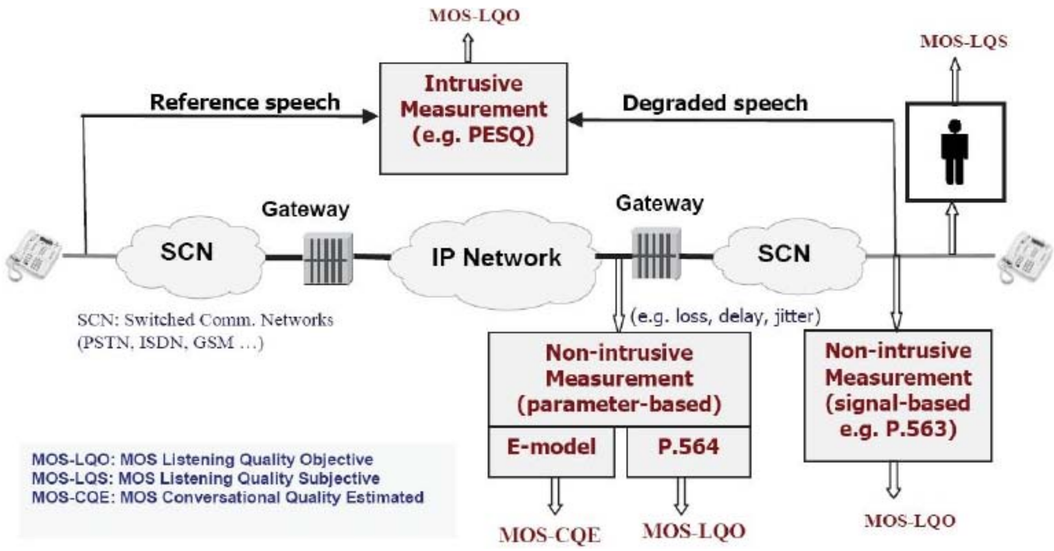


Figure 2.9: Voice Quality Assessment methods for VoIP

between two sequences grading them according to the reference video from 0, score for null fidelity to 100 for perfect fidelity.

These methods are expensive and time consuming as they require lots of human beings and material. Furthermore, MOS estimation need to be automated for real time systems. Mathematical models for emulating results of subjective procedures have been a hot topic during the last decade.

2.3.3.3 Objective Quality Evaluation for VoIP

Several objective algorithms have been developed in recent years, such as the ITU-T standard recommendations adopted for measuring speech quality in telephone networks Perceptual Analysis Measurement System (PAMS), Perceptual Speech Quality Measure (PSQM), Perceptual Evaluation of Speech Quality (PESQ) and Single Side Speech Quality Measure (3SQM). The objective of algorithms is to mimic the human listening system using models in order to score voice quality that will be similar, on average, to that produced by a human listener. Objective algorithms can be intrusive, non-intrusive or hybrid as depicted in Figure 2.9.

- intrusive method uses an algorithm which compares original and degraded signals.
  - PAMS [47] is an intrusive method developped by British Telecom in 1998.

  - PSQM [48] was developed by PTT Research in 1994 and is not anymore recommended by ITU-T P.861 as it is not suitable for constant monitoring and not designed to VoIP even though it was giving good performance. It objectively evaluates and quantifies voice quality of speech codecs. PSQM was revised to overcome new problems such as burst errors, and varying delay which resulted in the developement of other versions like PSQM+ and PSQM/IP.
  - PESQ also known as ITU-T recommendation P.862 is the ITU-T's current state-of-the-art intrusive mechanism. PESQ is designed to predict perceptual quality of a degraded video by analyzing specific parameters such as noise, errors, coding distortions, delay, delay jitter, wrapping time and transcoding. PESQ measures only the listening quality and combines the psycho-acoustic from PSQM+ with a time alignment algorithm from PAMS.

- non-intrusive methods:
  - The ITU E-model [49] is a computationally simple algorithm that can be used in real-time application but does not give a good correlation with subjective quality tests. It takes into account the encoding distortion, delay and delay jitter, echo... The purpose is to predict the subjective effect of combinations of impairments.
  - 3SQM is based on the ITU-T recommendation P.563 (2004).
  - P.564 (2006) predicts voice quality analyzing the header of IP/UDP/RTP VoIP call.

- hybrid method is a combination of methods from the other two categories or based on Neural Network (NN).
  - Combination of E-model and PESQ [50] done by Lingfen and Ifeachor in 2006. This work predicts perceived speech quality for modern codecs such as G.729, G.723.1, Adaptive Multi-Rate (AMR) and iLBC.
  - Ding and Goubran [51] in 2003 extended the E-model using PAMS to measure MOS score for G.723.1 and G.729 codecs. They added new parameters such as packet loss, delay jitter and buffer size.

  - NN[52] models have recently showed high prediction accuracy. Random Neural Network (RNN) have the advantage to learn in order to adapt according to dynamic environment. It is shown that there is no significant correlation between packet size and QoE for a given packet loss rate[53].

2.3.3.4 Objective Quality Evaluation for IPTV

Objective evaluation methods are based on and make use of multiple metrics, which are related to the content's artefacts (i.e. tiling, blurriness, jerkiness, etc.). Tiling is the presentation of image blocks at incorrect locations or times resulting from an error block. Tiling commonly occurs during burst errors. Blurriness is the reduction in detail of an image or objects contained within images. Jerkiness is the skipping or retaining of images displays. We distinguish three types of Full Reference, Reduced Reference and No Reference metrics.

- Full Reference methods is a frame by frame comparison between a reference video and the video under test. This method is accurate but requires the source video which is not always available.

  - The most widely used primitive methods and quality metrics that used an Error Sensitivity framework are the Peak Signal to Noise Ratio (PSNR) and MSE.

    $$PSNR = 10log_{10}\frac{L^2}{MSE} \tag{2.1}$$

    where L expresses the dynamic pixel value (equal to 255 for 8 bits/pixel monotonic signal).

    $$MSE = \frac{1}{N}\sum_{i=1}^{N}(x_i - y_i)^2 \tag{2.2}$$

    where N expresses the total pixels, x_i and y_i denotes respectively the i^th pixel value in the original and distorted signal.

  - Structure SIMilarity (SSIM) [54] is a full reference method using Human Visual System (HVS) parameters. SSIM compares two image structures : SSIM metric is calculated on various windows (NxN pixels : usually 8x8) of an image. SSIM is applied on luma(brightness in an image) and SSIM index is between -1 and 1.

$$MSE = \frac{1}{K.M.N} \sum_{k=1}^{K} \sum_{m=1}^{M} \sum_{n=1}^{N} [o_k(m,n) - d_k(m,n)] \tag{2.3}$$

where x:degraded pixels, y:original pixels, μ_x : average of x, σ_x^2 : variance of x, σ_{xy} : covariance of x and y, c1 and c2 two variables to stabilize the division.

- Reduced Reference method is using part of the reference video and compares it with the degraded signal. It then requires access to source video. Video Quality Measurement (VQM)[55] [56] from Institute for Telecommunication Sciences (ITS) extracts information from original and distorted sequences and compares them. VQM defines 7 independent parameters.
  - 4 parameters from spatial gradients of the Y luminance component (*si_loss*, *hv_loss*,*hv_gain*,*si_gain*)
  - 2 parameters from vector formed by the two (Cb, Cr) chrominance components (*chroma_spread*, *chroma_extreme*)
  - 1 parameter measure contrast and motion both extracted from the Y luminance component (*ct_ati_gain*)

$$\begin{aligned} VQM = -0.2097 * si\_loss + 0.5969 * hv\_loss + 0.2483 * hv\_gain \\ +0.0192 * chroma\_spread - 2.3416 * si\_gain \\ +0.0431 * ct\_ati\_gain + 0.0076 * chroma\_extreme \end{aligned} \tag{2.4}$$

- No Reference models : Pseudo-Subjective Quality Assessment (PSQA)[53] is based on RNN[52] which is a stochastic dynamic system behaving as a network of interconnected nodes or neurons. PSQA method takes QoS parameters of Wireless Fidelity (Wi-Fi) networks as inputs (packet loss, delay, jitter and more recently to frame loss) and outputs a MOS estimation. Adaptive Neural-Fuzzy Inference System (ANFIS)[57] combines the advantages of a neural network and a fuzzy system. [8] trains three neural networks for three distinct types to predict video quality based on a set of objective parameters. MOS score is the result of video quality prediction from both network (i.e. packet error rate and link bandwidth) and application (i.e. frame-rate, Send Bit Rate (SBR)) parameters for IPTV over wireless network application.

2.3.3.5 QoS Requirements for Media Services of Packet Switched Networks

Media services are considered a special service in terms of format and performance requirements for their delivery. In this section, an overview of the diverse multimedia streaming applications and their common requirements on the delivery of multimedia streams is given. Compared with traditional web content, media streaming-based applications introduce complex process through the following stages before they display to the end-user:

- Capturing: The audio or video stream must be captured and converted to a digital form (but still the content remains uncompressed i.e. raw digital data).
- Encoding: An encoder converts the raw digital data into a particular audio or video format, following a specific standard.
- Storing: A server may store the encoded stream for future transmission or in case of live streaming the storing stage can be skipped.
- Delivering: The stream is transmitted to one or more recipients. A live stream may be transmitted as it is captured and encoded, whereas a pre-recorded stream is transmitted by a server.
- Decoding: The receiver decodes and displays the data upon their arrival. Alternatively, the receiver may store the entire stream before initiating playback (i.e. progressive download), which is mainly used for video flash applications.

The receiver plays the samples or frames in a manner that preserves their temporal spacing, which means that the data rate of the video transmission is quite high, depending also on the frame size. Thus, in order to minimize the demands for data transmission, video compression techniques are applied, which reduce the video size by a significant factor (with the respecting reduction in the perceptual quality of the video). The encoding of media streams introduces a fundamental trade-off between the amount of data and the quality as perceived by the user. In a similar way, the audio data can be compressed, but in contrast to video signals, users are more sensitive to degradation in the quality of audio content. In addition, audio streams require much less bandwidth than video streams, making the requirements to reduce the size in a exchange for a significant quality degradation less important. Considering the special nature of media services, it is clear that strict requirements are applied on the delivery of data, in terms of QoS requirements. The performance QoS requirements of the multimedia streams delivery are summarized in the following parameters:

- Delay: For traditional services, like web services, the delay does not dramatically affect the user satisfaction, if it is limited to few tens or hundreds of milliseconds, given that the content is downloaded completely without further issues. This is not the case of multimedia streaming. Once the playback of an audio or video stream begins, the successive audio samples or frames must arrive in a timely manner. Otherwise, the media player must compensate for the missing data. Latency for audio streams causes even more performance degradation.

- Loss: Due to the time-dependent requirements that are applied in the successful delivery of the applications, a retransmitted packet is of limited use if it arrives after the media player has displayed the associated frame or reproduced the specific audio sample. For these reasons UDP is preferred for transport of multimedia services relative compared to Transmission Control Protocol (TCP) for multimedia streams. However, the packet loss is considered as the most important QoS parameter, which affects the perceived quality dramatically, given that error propagation is occurred in a media service.

- Throughput: Streaming audio and video data requires sustained and specific throughput in order to perform flawlessly. Although voice applications have low throughput requirements, the bandwidth requirements for video streams vary widely, depending on the encoding bit rate, the frame rate and the spatial resolution.

Following this general section, the following subsections summarize in terms of QoS parameters the requirements of IPTV and VoIP service respectively.

2.3.3.6 QoS requirements for VoIP

Delay is considered as the most disruptive impairment in VoIP service. The degradation that delay introduces in VoIP service appeared as echo and talker overlap.

Talker overlap happens when the end-to-end delay is so high that user A cuts off the speech of user B. According to ITU-T G.114 the following delay shown in Table 2.2 values provide the respective perceptual impact on the VoIP quality.

Table 2.2: Guidelines of impact of delay on voice quality

Delay	Quality
0-150ms	Acceptable for most calls
150-400ms	Acceptable if callers are aware of impairment
>400ms	Not acceptable

The bandwidth as shown in Table 2.3 that VoIP streams consume (in bits per second) is calculated by adding the VoIP sample payload (in bytes) to the 40-byte IP, UDP, and RTP headers (assuming that compressed RTP is not in use), multiplying this value by 8 (to convert it to bits), and then multiplying again by the packetisation rate (default of 50 packets per second).

Table 2.3: Bit-rate requirement for different codecs

Codec	Payload Size (ms)	Payload Size (bytes)	RTP/UDP/IP	Packets Per Second	Bandwidth (Kbits/s)	Voice Quality
G.711 64 kbps	20	160	40	50	80	4.1
G.726 32 kbps	20	80	40	50	48	3.85
G.728 16 kbps	30	60	40	34	37.2	3.61
G.729 8 kbps	20	20	40	50	24	3.92
G.723.1 6.3 kbps	24	30	40	34	19.04	3.9

Jitter is often considered as another aspect of the network delay parameter, since it refers to the delay variation of packet arrival between consecutive packets. Although the mobile terminal generate constant rate of packets, the network condition may cause the network (i.e. routers and gateways) to be unable to process the packets in real-time. This means that buffers must be employed by the network elements to temporarily store a packet while it is processing other traffic. The less traffic present, the faster the VoIP packet can be processed. Jitter will therefore result in the clumping and gaps of the incoming voice stream. Since jitter is calculated as the magnitude of the delay variation, it will always be a positive number, with zero indicating that no jitter is present. The generalized way to minimize jitter is to use a buffer at the user side that will hold all incoming packets for a period of time so that the slowest packet arrive in time to be played in the correct sequence. The jitter buffer will add to the overall delay of the network and so once jitter exceeds a certain level, the jitter buffer will begin to impair the call through excessive delay. Adaptive jitter buffers are usually employed in managed VoIP networks. These adaptive buffers increase in size only as needed when the jitter increases. Managed adaptive buffers will intentionally drop packets in order to maintain low enough delay to facilitate an acceptable level of call performance.

It is not uncommon in VoIP services, for the bandwidth requirement to exceed the available network bandwidth. This is especially common in wireless VoIP systems such as in 3G networks where the wireless channel is bandwidth limited, which is the case of this thesis as well. The excess data due to bandwidth mismatch is usually buffered and transmitted according to the available rate. A advantage of buffering is the reduction in jitter but at a cost of increased delay and eventual packet loss due to buffer overflows. **Packet loss** leads to the loss of speech/video samples which impact the perceptual quality. The relationship between packet loss rate and the amount of information lost depends on the codec in use and packetization scheme in place. The packetization interval determines the size of samples contained within a single packet, which determines the amount of data lost with the loss of a single packet. For example, for a default 20ms packetization interval, the loss of two or more consecutive packets results in a noticeable degradation of voice quality due to the amount of lost information. Loss causes voice clipping and skips and to ameliorate the impact of loss on quality, Packet Loss Concealment (PLC) are employed to mask the effects of lost or discarded VoIP packets on perceptual voice quality. The method of PLC used depends upon the type of codec and application (voice or video call). A simple method used by waveform codecs such as G.711 (PLC for G.711 is defined in G.711 Appendix I) is to replay the last received sample with increasing attenuation at each repeat with the waveform not changing much from one sample to the next. This technique can be effective at concealing loss of up to 20 ms of samples.

2.3.3.7 QoS requirements for IPTV

Video applications are delay-sensitive and loss-sensitive by nature. Therefore, in order to reassure the perceptual accuracy of the video communication, the underlying transport network has to satisfy specific network-level QoS characteristics in terms of end-to-end delay, delay variation (i.e. jitter) and loss ratio. When addressing the QoS needs of IPTV service, the following guidelines are considered as requirements for acceptable and efficient provision of the service:

- Loss should be no more than 5 percent.
- Latency should be no more than 4 to 5 seconds (depending on the video application's buffering capabilities).
- Jitter does not affect substantially the service, subject to buffering techniques.
- Guaranteed bandwidth requirements depend on the encoding format and rate of the video stream.

Streaming-Video applications have more soft QoS requirements because they are not delay sensitive (the video can take several seconds to initially start) and are largely not jitter sensitive (because of application buffering). In video, PLC may include the use of adjacent information or the use of previously received frames to cover the lost information with increasing intelligence.

Latency is defined as the elapsed time (delay) between the generation of the IPTV service and the delivery of it at the end-user, who has requested to receive the specific service. In general, for one-way (i.e. unidirectional) services, the parameter of delay is not as important as it is in bidirectional services like video conferencing or VoIP telephony, because in the case of unidirectional service the delay happens only at the beginning of the service. However, if the delay is high enough to be perceptible, then a latency problem is raised, which can cause various issues like session interruption, buffer underflow etc. Latency is capable to be introduced into a media delivery chain due to various parameters that can be related to encoder delay, network delay, jitter buffer delay, and the delay introduced by the decoder.

Video signal transmitted through a network is sensitive to errors. At the receiver side, each bit or symbol is compared with one or more threshold levels. If the instantaneous amplitude of the noise is high enough at the sampling instant, the received bit or symbol may be incorrectly recognized, which leads to bit errors. The Bit Error Ratio or **Bit Error Rate** describes the mean rate at which bit errors occur. Bit Error Rate (BER) depends on the characteristics of the transmission channel. The error rate of a radio link (i.e., in a mobile radio receiver) is variable depending on the receiver's location, on atmospheric conditions, and also on other interference sources. Coded video data is highly sensitive to transmission errors due to the inter-dependent nature of the encoding process, making the error-propagation possible.

IP-based networks do not guarantee that transmitted data from the one end will finally reach its destination at the other end. Delays through the network depend on a number of factors such as the route taken, the processing speed and capacity of each node along the route and the amount of other data traffic in transit concurrently. If the capacity of a node within the network is exceeded due to congestion, then the delay factor increases and then true **packet losses** may also occur, which in turn will result in frame loss. Moreover, even if the delay increases to levels that will prevent real packet loss to take place, then if data packets are delayed enough and finally reach the end-user at a time point that the display moment has passed then in practice they will be treated as lost packets. So the packet loss effect is either directly or indirectly caused by the increase of the delay.

2.3.3.8 Perceived Quality of VoIP

Voice non-intrusive quality prediction modeling is the process of describing the PQoS in terms of MOS as a function of codec settings and network Network-QoS (NQoS).

$MOS \leftarrow f(Loss, Delay, Jitter, Codec, Bitrate...)$

A user's perception of quality is not limited to the physical aspect of the network and the environment or the context in which the user listens to the voice has an impact on perception of quality. The environment can have an impact on voice quality even without network impairments.

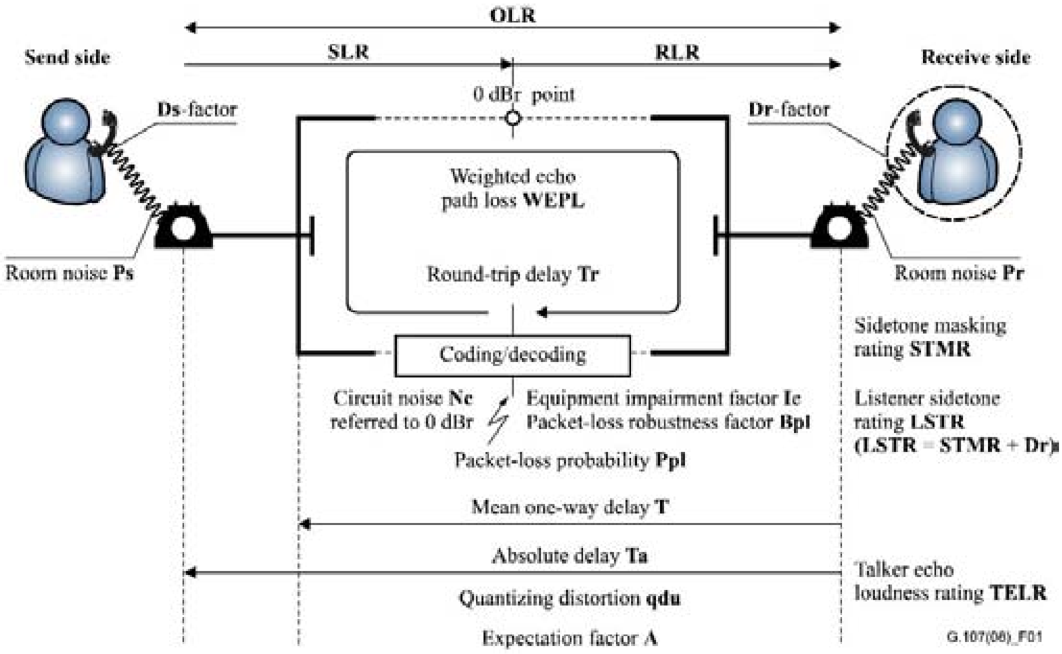


Figure 2.10: E-model voice quality assessment parameters

The ITU E-model[49] computationally assesses voice quality from network quality parameters (i.e. loss, delay and jitter) and other quality affecting parameters such as those depicted in Figure 2.10. This quality assessment tools uses a function to map an R-scale to a MOS score given by :

$$\begin{aligned} For R < 0 : MOS = 1 \\ For 0 < R < 100 : MOS = 1 + 0.035R + R(R-60)(100-R) \times 7.10^{-6} \\ For R > 100 : MOS = 4.5 \end{aligned} \tag{2.5}$$

where R is the Rating factor, which has a range from 0 to 100, given by :

$$R = R_0 - I_s - I_d - I_e + A \tag{2.6}$$

where R_0 is the quality with no distortion, I_s is the impairment of the speech signal itself, I_d corresponds to impairment level caused by the delay and jitter, I_e is the impairments caused by the encoding artifacts and A is the expectation given by :

$$A = (95 - Ie)\frac{Ppl}{Ppl + Bpl} \tag{2.7}$$

where Bpl is a packet loss robustness factor and Ppl is the packet loss rate.

The E-Model has a limitation which makes it erroneous when applied to VoIP networks which are non-additive parameters [58]. New PQoS quality predictions are being developed, such as the PSQA[53] and the AMR perceptual model developed at the University of Plymouth (UOP), due to the limitation of the E-Model. UOP developed a PQoS quality prediction model [50] for AMR encoded voice frames from NQoS parameters. MOS is following Equation2.5 but R is given by :

$$R = 93.2 - I_d - I_e \tag{2.8}$$

where

$$I_d = 0.024d + 0.11(d - 177.3)H(d - 177.3) \tag{2.9}$$

where $\begin{cases} H(x) = 0 & \text{if} x < 0 \\ H(x) = 1 & \text{if} x > 0 \end{cases}$ where d is the one-way delay measurement and I_e given by :

$$I_e = aln(1 + bp) + c \tag{2.10}$$

where p is the packet loss rate and a, b and c are constants depending on the AMR mode. This PQoS prediction model was extensively tested under different network scenarios and conditions and the results are reported in [59]. The presented results showed a high prediction of 0.987 when tested using real Internet VoIP trace data.

2.3.3.9 Perceived Quality of IPTV

quasi-PSNR (qPSNR)[60] is a PQoS tool developed by Rohde&Schwarz and used in their protocol analyser to provide an estimate of the PSNR of H.264-encoded video sequences that are received by the IPTV test terminal. The algorithm is based on the statistical analysis of encoded transformation coefficients. It provides a non-reference analysis of the image quality of compressed video sequences. This estimation uses the encoded video data stream as input. The completely decoded video stream is not required. The PSNR shows a good correlation to the perceived image quality but there are cases where the correlation does not suffice. This is due to several reasons. One reason is that the PSNR measures the difference between single pictures. This means that temporal masking effect caused by swift movements of the picture content from one frame to the next, are not reflected in the PSNR. Another reason is that the perceived quality of a single frame does not only depend on the

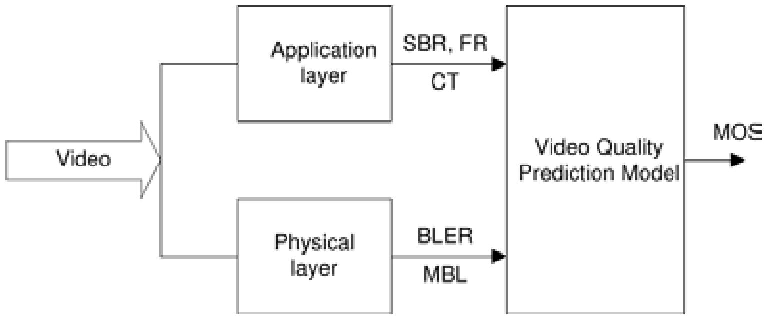


Figure 2.11: Functional block of proposed video quality prediction model

difference between the encoded picture and the original but on the spatial resolution of error components and the signal itself, i.e. on a spatial masking effect. Temporal masking is the expression used for the inability of the human eye to notice fast changes in the image. The exact amount of masking depends on the individual viewer because the way the viewer's eye follows the moving objects in the image influences their visibility. Therefore, temporal masking can only be estimated roughly from the motion in a video sequence. Spatial masking can best be described by introducing visibility thresholds which depend on the spatial resolution (frequency) of details in the image. [61] shows how the PSNR can be calculated in the transform domain if certain assumptions for the transform coefficients hold. These assumptions concern the probability density function of the transform coefficients. Another assumption is that the quantisation of the image should not too coarse. This can occur in transmission systems which operate on a low data rate even when the image content is critical. In such a case the PSNR would be underestimated.

UOP developed a perceptual video model based on ANFIS and a model based on non-linear regression analysis, used in the prediction of video quality in terms of MOS. ANFIS is well suited for video quality prediction over error prone and bandwidth restricted UMTS or WLAN as it combines the advantages of neural networks and fuzzy systems. Link losses were modeled using a 2-state Markov models with variable Mean Burst Lengths (MBLs) depicting the various UMTS scenarios and WLAN[8]. Both models were trained with a combination of physical and application layer parameters and validated with unseen data sets as seen in Figure2.11. Results showed that good prediction accuracy was obtained from both models.

Figure 2.12 depicts the methodology used for the non-intrusive prediction of

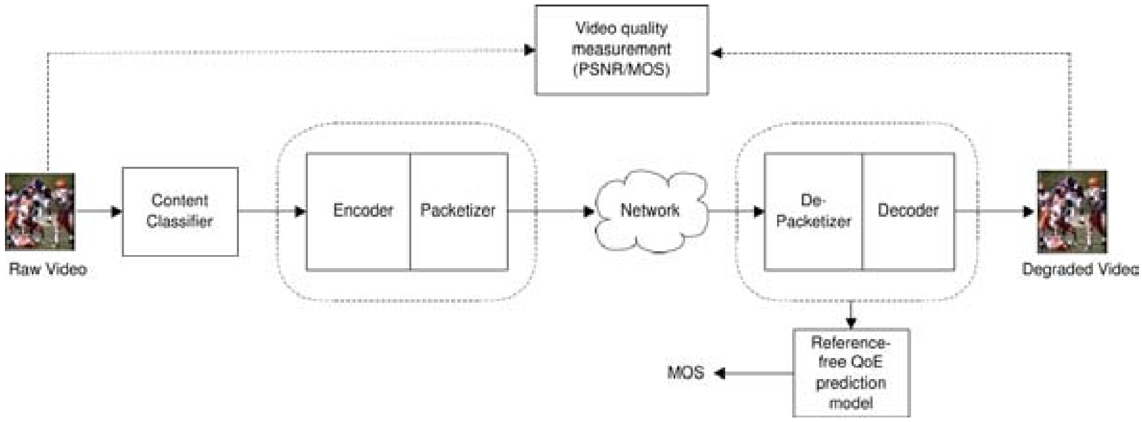


Figure 2.12: Conceptual diagram to illustrate video quality prediction

video quality. At the top of of the figure, an intrusive video quality measurement block is used to measure video quality at different network conditions (e.g. different packet loss, jitter and delay) or for different application settings (e.g. different codec type, content type, sender bitrate, frame rate, resolution). The measurement is based on comparing reference and the degraded video signals. PSNR is used for measuring video quality to prove the concept. MOS values are obtained from a PSNR to MOS conversion scheme [62].

Regression-Based model used in UOP model establishes a relationship between MOS and Block Error Rate (BLER) for three different content types. It is observed that the quality is highly dependent on the type of content. "Suzie" video with low spatio-temporal dynamics has an acceptable quality up to BLER of 40% whereas for "Football" video, a BLER greater than 20% reduces quality drastically. UOP ran some test for UMTS networks and observed that at higher SBRs, the video quality degrades rapidly due to network congestion as UMTS has drastic bandwidth restriction. The overall impact of Frame Rate (FR) is less obvious compared to that of Content Type (CT), SBR and BLER. The relationship of the individual QoS parameters on MOS was established. UOP obtained the following nonlinear equation given in Equation 2.11.

$$MOS = \alpha + \frac{\beta e^{FR} + \gamma log(SBR) + CT(\delta + \epsilon(SBR))}{1 + (\zeta BLER + \eta BLER^2) + \mu MBL} \tag{2.11}$$

with a reasonable goodness of fit given in Table 2.4.

[63] proposed a PQoS estimation over 802.11g in ad-hoc systems. Their solution estimated the quality with a curve fitting technique. This curse is a function of three

Table 2.4: Coefficients for regression based model for UMTS

α	β	γ	δ	ϵ	ζ	η	μ
4.2694	-1.4826x10^{-9}	0.0656	-0.9559	-0.0261	-2.4767	-5.3168	0.3327
R^2	83.52%			RMSE	0.2778		

parameters: the video bit-rate, the background traffic and signal power level over the wireless link. Nevertheless, the three parameters are considered correlated to each other and limited to 802.11g networks within ad-hoc architecture. The estimation does not take into account the delay, lost packet or BLER which may have a huge impact on the video. In [64], a study and an analysis on context-aware IPTV system are presented. The outcome analysis states that no existing contribution could satisfy service personalization in a complete and adequate manner. Besides, the PQoS [2] of the IPTV services may also suffer from wireless access network impairments and is not discussed in IMS architectures. Papers [65] and [66] respectively introduce an IPTV client for IPTV over IMS services and a complete next generation IPTV over IMS platform. The client and the global architecture are handling presence, contact list, user generated content, remote control functionalities and media content service discovery for TV and Video on Demand (VoD) services. Dynamic adaptation of the Audio/Video (AV) content based on access network condition or user's environment is not treated in these two papers.

Perceived quality for VoIP and IPTV services depends mainly of the codec type and network conditions. One advantage of NGNs is that one core network handles many access networks. Then when non satisfying PQoS is observed at the end-user's device, MN can switch on its other network interfaces and scan for other network attachement possibilities. MN or core network triggers then the handover towards the chosen access network expecting PQoS improvement.

2.4 Mobility

As stated in [1], *31 percent of smartphone traffic was offloaded onto the fixed network through dual-mode or femtocell in 2010"*. More devices are connecting to private networks to get lower cost, higher network bandwidth, better performance for applications [67]. Mobility between heterogeneous wireless networks are required.

Many mobility definitions exist but one gives a definition of generalized mobility [11]: "Generalized mobility is the ability for the user or other mobile entities to communicate and access services irrespective of changes of the location or technical environment. The degree of service availability may depend on several factors including the Access Network capabilities, service level agreements between the user's home network and the visited network (if applicable), etc. Mobility includes the ability of telecommunication with or without service continuity."

2.4.1 Handover Definition

More specifically, a handover is the process for a MN that allows a MN to migrate its air-interface, service flow, and network attachment from a serving AP to a target AP. Several criteria distinguish the nature of the handover (i.e. handover initiated and controlled by the MN or the network, backward and forward handover, proactive and reactive handover, seamless handover, soft and hard handover, horizontal and vertical handover). A handover begins with a decision for a MN to handover from one AP to another AP. This change of AP may be required by the movement of the MN or caused by spectrum, capacity or network management issues.

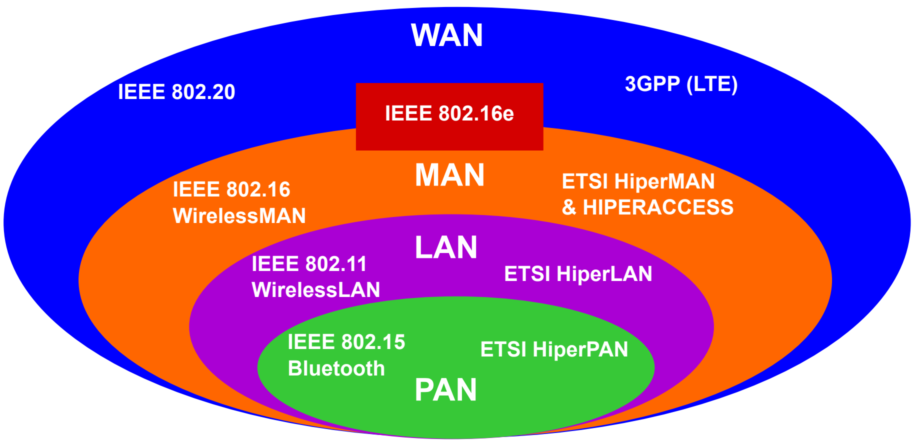


Figure 2.13: Overlay Wireless Standards

Figure 2.13 shows existent heterogeneous wireless networks standards and its overlapping in terms of coverage. According to the Corporate User (CU) perspective, the convergence of these heterogeneous wireless networks increases its connectivity.

2.4.2 Horizontal vs Vertical Handovers

When the MN accesses a new AP using the same access technology than its previous AP, the handover is named horizontal handover or intra-system handovers. A example could be a user moving between two cells in a cellular system. Vertical handover or inter-handovers refers to a change of access technology. Figure 2.14 shows both cases. In this thesis, the study refers to the implementation of a vertical handover between a WLAN Access Point and an UMTS access.

In term of mobility, there exists an other type of classification focused more on the network topology :

- Pico mobility which is a movement within the same radio cell;
- Micro mobility which is a movement within the same sub-net;
- Macro mobility which is a movement across different sub-nets but within same administrative domain;
- Global mobility which is a movement across different administrative domains;

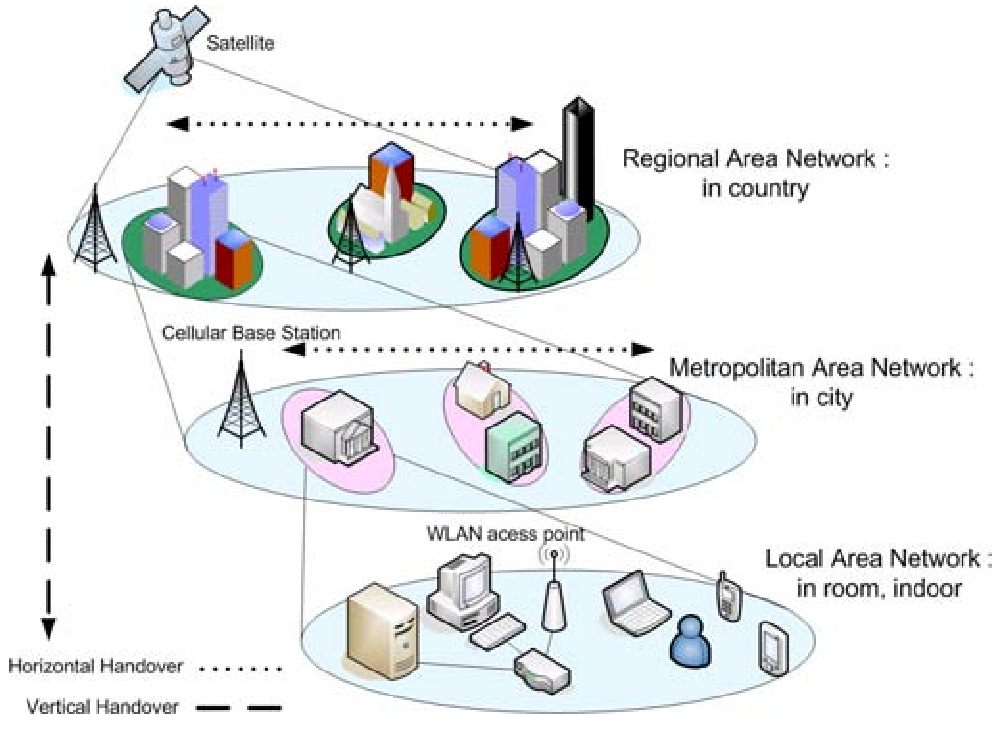


Figure 2.14: vertical and horizontal Handover

2.4.3 Hard vs Soft Handovers

The difference between hard handover and soft handover is the number of MN's simultaneous connections. If a MN can access a unique AP, a "hard" handover happens when the MN loses connectivity due to the change of Access Point. Since communication is lost for a short period, this introduces a service interruption for the user. A MN accessing multiple access points can manage a soft handover. Soft handover may be referred to as "make before break" handover : When the new access point is ready to let the terminal access in, the terminal will disconnect with the old access link. The soft handover will maintain several links with different AP at the same time. The same information can be sent from different AP and be received at the terminal and be combined during the handover execution. In the process of horizontal soft handover in UMTS, the MN has several active links with different AP, the packet can be transmitted correctly as long as one link is efficient. The MN can disconnect with the old AP only when the new AP is available.

2.4.4 Existing Mobility Solutions

2.4.4.1 Classification

Many different mobility solutions based on different signaling protocols raised up. One classification is based on the Open System Interconnection (OSI) network layer on which these protocols are acting. We extended the survey [68] with transport and application layer solutions.

- Link layer solutions
- Network layer solutions
- Transport layer solutions (extension of the survey)
- Application layer solutions (extension of the survey)
- Cross-Layer solutions

Pure link layer mobility in cellular networks does not actually allow an extensive mobility solution as mobility management requires higher layer for the signaling between different equipments, handover decision computation, location management and routing functionalities. Nevertheless in some network infrastructures, link layer solution for horizontal handover gives promising results [69] and [70].

Thanks to MIP[71], network layer mobility enforces vertical handover by redirecting traffic of the MN to its current IP address. MIP is an Internet Engineering Task Force (IETF) standard that introduces the concepts of home address, Home Agent (HA) and Care-of Address (CoA). The home address is the permanent address of an end-user device. The CoA is a temporary address assigned to a device that leaves its home network. The HA is a router in the home network aware of CoAs. It relays all packets addressed to the home address towards the newly-acquired CoA. MIP's extension HMIP, HAWAII, CIP, MIP-RR, IDMP and DMA, TeleMIP are defined and analysed in these surveys [68] and [72].

NEtwork MObility (NEMO) Basic Support Protocol (BSP) [73] originally extends Mobile IPv6 (MIPv6) and has been adapted for MIPv4 in [74]. This standard aims to give global connectivity for a group of MN such as devices in ships, aircraft and trains. NEMO adds Mobile Routers (MRs) redirecting all traffic through a bidirectional tunnel with its HA. These MRs act as intelligent gateways performing registration proxy and auto-configuration for access network connectivity.

Based on MIPv6 and NEMO [73] technologies, DSMIPv6 [75] enables mobility for dual-stack MNs supporting IPv4 and IPv6. The HA also has to be dual stack. 3GPP adopted DSMIPv6 and Proxy-MIPv6 (PMIPv6) to handle host-based mobility and network-based mobility functionalities in its LTE architecture [76]. As network controlled mobility, Proxy-MIP (PMIP) avoid any UE actions in the mobility management. The proxy mobility agent in the network ensures the mobility management on behalf of the MN. The standards define two core functional elements : the Local Mobility Anchor (LMA) and the Mobile Access Gateway (MAG). When MN changes its network attachment, the new MAG (nMAG) should detect the attachment and initiate the necessary procedures to authenticate and authorize the MN. PMIP lowers the probability for a signaling message to be dropped as it does not travel on the air interface. Nevertheless, handover interruption tends to be higher even more for slow access technology as PMIP is a reactive solution. A proactive solution predicting the MN's movement thanks to a Proxy Information Server (PIS) is presented in [77] and reduces the handover interruption. Another hard handover solution [78] buffers packets in New-MAG while the MN is doing handover. Buffering avoids any packet loss but does not satisfy real-time services requirements. J. Kim et al in [79] proposes soft handover for PMIPv6 through a bicasting system. This solution further reduces media disruption as bicasting optimizes the packet switching on the MN side. Nevertheless, no MN algorithm is given in order to know how applications should play packets from each interface. Besides, extra and non standardized signaling between Previous-MAG and New-MAG is required.

L. Magagula et al introduce a Handover Coordinator (HC) coupled with Media-Independent Handover (MIH) functions described in [80] to perform bicasting during PMIPv6 handover. This solution implies modification in the PMIPv6 stack as the new-MAG is advertised by the LMA/HC to a new attachement with a PBA without having previously sent any PBU. So the new-MAG should allow and manage PBA even if the MN may not have (yet/anymore) access to the network without having received any attachment request. Furthermore, bicasting solutions in PMIPv6 do not treat how the MN is handling both streams. In general, PMIP, as network-based mobility, raises more MN computation for correctly triggering the handover process. For instance, a Wi-Fi to 3G handover demands accurate triggering as Wi-Fi access can vanish rapidly. In the host-based mobility case, even if the Wi-Fi connection is lost just before the handover has been triggered, the session could be reestablished and not lost.

Figure 2.15 depicts the LTE architecture and its interconnecting interfaces as standardized by 3GPP [76]. Mobility protocols (MIPv6, DSMIPv6, PMIPv6 or MIPv4 in FA mode) between 3GPP and non 3GPP IP accesses vary depending on whether the mobility is host-based or network-based and in roaming mode or not. host-based Mobility (HBM) means that the MN intervenes in the mobility process whereas in network-based Mobility (NBM), the MN actions are not needed. In HBM, DSMIPv6 is used in non-roaming mode (S2c interface) and MIPv6 in roaming mode. In NBM, PMIPv6 or MIPv4 in FA mode are used in roaming and non-roaming scenarios.

S2c reference point in P-GW defines host-based mobility and S2a/S2b define network-based mobility. Evolved Packet Data Gateway (ePDG) is similar to a Virtual Private Network (VPN) concentrator enabling (de-)encapsulation of packets for IPSec and PMIPv6 tunnels. ePDG also ensures lawful interception and

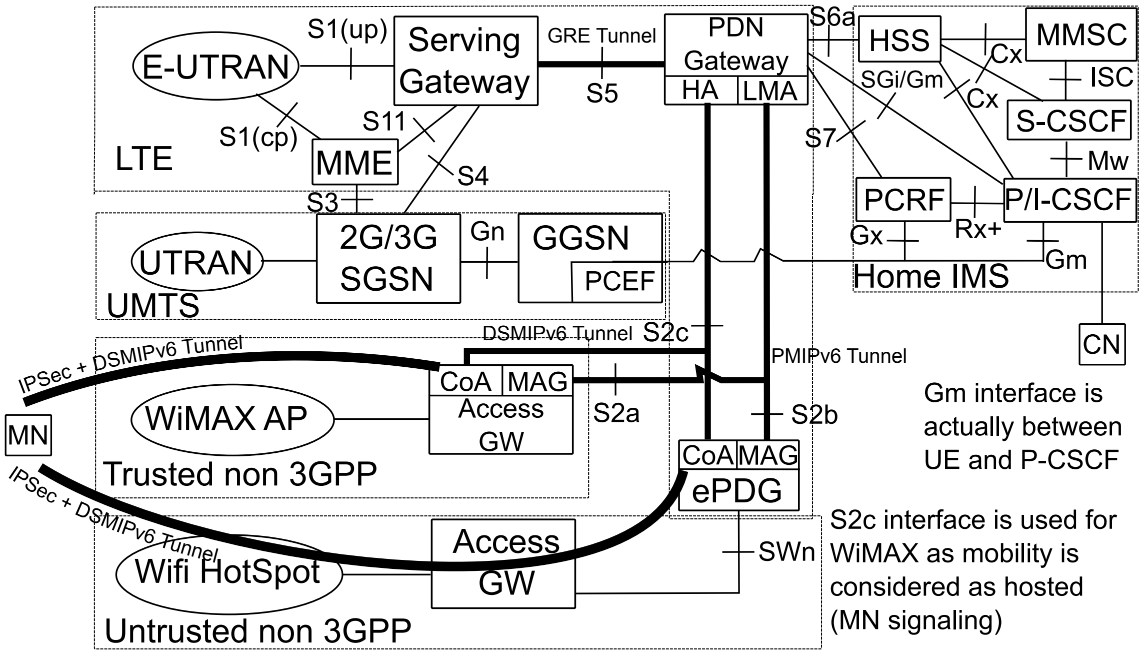


Figure 2.15: LTE Interfaces

MAG/CoA functionalities for mobility purposes. This figure describes the mobility anchor points. The HA for MIPv6/v4 and DSMIPv6 protocols and LMA for PMIP protocols are located in P-GW. CoA and MAG functionalities are located in either Access GateWay or ePDG whether the non 3GPP access is trusted or not. In the host-based mobility mode, IPsec tunnel is established between the MN and its corresponding CoA and then a DSMIPv6 is established over this IPSec tunnel. In the network-based mobility mode, no tunnel is required between MN and its MAG. Nevertheless, a PMIP tunnel is created between MAG and LMA. Figure 2.16 de-

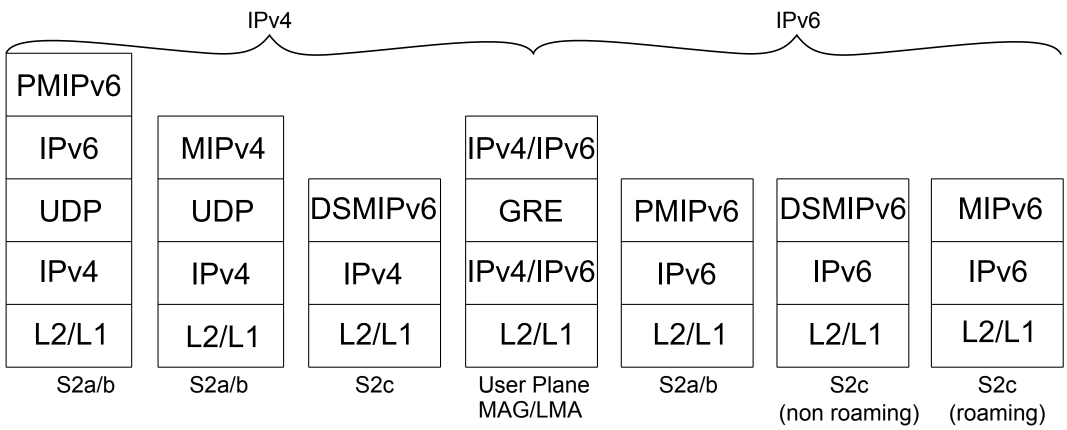


Figure 2.16: Protocols Stack for S2a, S2b and S2c

scribes the protocol stack of mobility protocols used in LTE. As described in [81], IPv4 based network encapsulates PMIPv6 and MIPv4 over IPv4/UDP, DSMIPv6 supports IPv4 and IPv6 networks and user plane is for the media encapsulated over a IPv4/GRE or IPv6/GRE. As described in [71], MIPv4 is encapsulated in UDP packets over IPv4.

With the NEMO Basic Support protocol, all traffic to and from the mobile network must go through the bi-directional tunnel resulting in a longer route. This kind of sub-optimal routing leads to transmission delay, packet overhead and bottleneck of the HA. Applications, i.e. real-time streaming, may be unable to tolerate such sub-optimality.

With MIP, users can seamlessly move without having to reconnect/reconfigure at every point of attachment. However, it is based on an asymmetric routing, which can add unpredictable delay to the traffic. Added delay might thus be an important issue in real-time communications. Because MIP is not appropriated, cellular IP was designed to take into account cellular principles and IP design principles. Within Cellular IP network model, mobility between gateways is still managed by MIP, whereas mobility within access networks is managed by Cellular IP. Cellular IP improves MIP by providing fast handoff control at the price of potentially some packets loss. Notwithstanding, triangular routing, as in MIP, still exists and is likely to add undesirable delay to real-time communications.

Transport layer mobility is an alternative at network layer mobility. TCP-migrate [82] implements host mobility without network layer support and uses DNS servers for location repository. 802.11f implements mobility based on proprietary Inter-Access Point Protocol (IAPP) which is working on UDP. SCTP and its mobile extension in IMS are studied in [83] and discussed in the following section. One interesting approach of mobile-SCTP (mSCTP) is the Dynamic Address Reconfiguration (DAR) function, which is mobility through different interfaces. A mSCTP solution in IMS is proposed in [83]. This paper is discussed in the following section 2.3.4.2.

Application layer solutions are mainly focused on SIP protocol. [84] introduces mobility support using SIP. This study proposes to use MIP mobility when the UE has TCP ongoing connection and SIP mobility only when UDP connection is used. [85] implemented this solution and results show that hybrid SIP/MIP solution gives better performance than pure SIP or MIP solutions. A comparison from 2002 in [86] between SIP and MIP explains that MIP mobility gives smaller disruption for handoff than SIP mobility since MIP triggers mobility to HA whereas SIP Re-INVITE message travels all the way to CN. Vertical Handover with adaptation in IMS is analysed in [87]. This solution uses SIP mobility and introduces a dedicated AS to handle media adaptation during the mobility process. Even if this solution uses dual homing to optimize new link configuration and SIP registration, it inserts a Adaptation Media Gateway (AMG) in the new target network to proceed the data flow rebind and adaptation to the new link capacity. This AMG element is directly controlled by AS and introduces extra signaling in the IMS. SIP B2BUA hiding MN mobility from CN and RTP translator multicasting RTP packets during handover is presented in [88]. These two elements optimize handoff delay and reduce packet loss. In [89], SIP B2BUAs and RTP media gateway are embedded in base stations or access points in order to duplicate and re-route the RTP traffic as soon as the new interface is up. A "joint" header is introduced in SIP Re-Invite request in order to retrieve the media session and start duplicating RTP packets on both interfaces. The re-Invite is then sent to the CN to change the IP address. This solution implements extra signaling on the client side by sending two re-Invite message (one for B2BUA and one for CN) and needs special parsing for "joint" header. This solution is not applicable in NGN networks. In IMS, a SIP-based solution [90] transferring context information about one SIP session from one "old" P-CSCF to a "new" P-CSCF optimizes signaling messaging during the re-Invite process such as QoS resource reservation. Nevertheless, this solution does not benefit from the re-registration optimization than can be proceed during an ongoing multimedia session in a dual homing scenario. This optimization could further reduce by 60% the handover time.

Cross-Layer solutions are the most commonly used in live system as low layers information improves the overall handover mechanism. GSM network triggers a handover based on signal strength measurement reports sent by the MN. An interesting research [91] provides the means to detect whether or not the MN is moving by comparing the difference of overhead cellular tower's ID and the signal. Leveraging the mobile GSM module and saving most of energy used to detect Wi-Fi APs, this detection algorithm triggers Wi-Fi scans only when the MN has moved at a given distance. Seamless-MIP [92] based on HMIP introduces handoff algorithm based on movement tracking techniques. TIMIP [93] derived from CIP and HAWAII architectures focuses on micro mobility. TIMIP allows legacy phones not equipped with mobility aware IP stack to wander through access networks thanks to layer 2 mechanism between MN and BS. NETworking Context Aware Policy Environment (NetCAPE) [94] enriches the mobility management protocols with a Policy Engine collecting information at the different layers and enforcing decision derived from them. SIP-NEMO [95] achieves NEMO route optimization at the application layer. Interesting work [96] extends SIP-NEMO with IEEE 802.21 MIH standard [97] using multihoming functionalities. MIH main functions focuses on the initiation and preparation of cross-layer handover. MIH Functions in the MN facilitate handover process exchanging data (link quality, link ID, offered QoS, target point of attachement) with the MIHF in Point of Service (PoS). The latter communicates with Media-Independent Information Service (MIIS) server hosted in the core network. The MIIS server collects information related to the access networks close to the MN. The PoS computes the handover decision and requests MN to proceed a handover. The communication between two MIHFs are done through MIH_NET_SAP primitives which are MIHF abstraction layers above specific transport layer (layer2 taking existing MIH_LINK_SAP primitives, layer 3 or above). A cross layer solution with SIP and MIH functionalities is presented in [98] and shows the interest of using interface handler and mobility manager inside the MN. Nevertheless, this solution requires MIH support in IMS. Besides, the SIP signaling and the media stream are not optimized for live system as the CN is involved in the mobility process. Indeed, the renewal of IP address at the CN side might change the media path resulting in extra delay and unacceptable jitter. Another architecture in [99] extends the same idea by inserting more detailed information about user and device in order to perform the handover with adaptation if necessary. [100] is a mobility solution combining SIP and mSCTP. New SIP methods ANNEXIP and SWITCH-SESSION are used to update the core with new IP connectivity. Nevertheless, these two new SIP methods have the same functionalities as a simple Re-INVITE without the ACK message which plays an important role in SIP session establishments with Session Description Protocol (SDP) negotiation. Summary of 802.21 MIH standard for inter-RAT handovers is done in [101].

2.4.4.2 Discussion and analysis

Latest release of LTE uses MIP, DSMIP or PMIP as mobility protocols to maintain communication during a handover. Nevertheless xMIP hides location update from SIP layer and then from application layer. MIP functions hide the mobility from the CN. The element Home Address assigns a permanent IP address to the MN when it wanders from its Home Network (HN) through HA to a Foreign Network (FN) through a CoA. With MIP, users can seamlessly move without having neither to reconnect nor to reconfigure at every point of attachment. However, it is based on an asymmetric routing, which can add unpredictable delay to the traffic. Added delay might thus be an important issue in real-time communications. Furthermore, MIP handles mobility on IP level bypassing any SIP session update or SDP re-negotiation according to the new link capacity. In a context of user-centric networking, real-time information is very sensitive information in order to bring advanced and profiled services to the end-user. We understand global mobility as a transparent vertical handover between two different access technologies managed in two separate domains during an ongoing multimedia session. The end-user's device wanders through different networks while consuming multimedia services. Literature often classifies mobility in micro, macro and global mobility corresponding respectively to a roaming between cells, subnets and domains. Another classification [102] brings the notion of terminal mobility, service mobility and personal mobility. The terminal mobility corresponds to the classification of literature. The service mobility allows the consumer to keep control of the application while he is wandering through networks. Then, personal mobility ensures that the end-user is able to consume services in any location on any terminal while he moves across administrative domains. The latter is also referred to as Location Update. Seamless macro and global mobilities are challenging for interactive and delay sensitive services like VoIP as this type of mobility requires a new IP address and network configuration. Seamless handover implies at least two simultaneous access to the core network in order to minimize the handover interruption. Multihoming in mobility management becomes then a necessary issue. IETF standardizes multihoming within network mobility support [103] and its motivations are expressed in [104]. Furthermore, global mobility requires that operators from both domains support the same mobility management. Vertical handover may be mobile controlled such as DECT needs reference of DECT and GSM mobility Vertical handover may be mobile controlled, network controlled, or mobile assisted such as in GSM. One advantage of host-based mobility is that the MN has the best perspective of what alternative links it can wander to. User profile management helps to collect this context information and the corresponding QoS/QoE information on each interface [99]. The mobility management should also be informed about the targeted access network in order to duplicate/reroute the traffic towards the new connection.

In this thesis, we focus our attention on mobility between 3GPP and non 3GPP access networks. LTE standard needs an IP Mobility Selection (IPMS) mechanism in order to choose the type of handover. This handover selection within 3GPP networks is done during the attach procedure over the source access network. On handover between non 3GPP accesses, the selection depends on whether the UE and the network are capable of handling network-based mobility. Our solution avoids this mechanism as the same signaling works for macro and global handover.

The mSCTP solution in IMS proposed in [83] requires several modifications and adaptations to comply with the standards. Firstly, modification in HSS is needed to handle the different IP primary/secondary addresses. Secondly, the REGISTER SIP messages are extended with 2 new optional headers "add-location", "set-primary" and a new "SWITCH" method is created to handle handover signaling and only understood by an extra component in the IMS architecture : the mSCTP-based proxy. As SCTP stack is not yet supported as a transport protocol in IMS, this proxy converts all mSCTP messages towards SIP messages. Some extra signaling like a redirect message "305 use proxy" in order to reroute the traffic to the mSCTP based proxy and a REFER sip request to establish a session from this proxy to the AS are needed. In this paper, the handover mechanism is launched on the old interface which is not efficient in case of abrupt disconnection from this access attachment. Some integration and design issues still need to be overcome.

The Dynamic Address Reconfiguration function defined in mSCTP can also be implemented directly at the client side through a UP management. [99] expresses the interest of using UP gaining in user satisfaction for IPTV services over different terminals and access networks. User context information is gathered in the SIP extended XML body and lets the IMS core aware about the UE environment. SIP with an XML body using a Re-INVITE handles the basic function of DAR and allows the mobility management in the core network to accept or not the mobility request according to the collected UE information and the service which is consumed.

IMS on top of a LTE [76] architecture as depicted in Figure 2.15 is part of NGN and both are standardized by 3GPP. Vertical handover in IMS is still under study whereas session continuity is addressed in 3GPP with [105]. This technical report introduces the Multi-Media Session Continuity (MMSC) Application Server acting as a SIP B2BUA and handling the mobility management at a SIP layer. The operator is controlling the session mobility within its home network and keeps a life trace about where the MN is located. Nevertheless, no media re-routing is discussed in this specification. In this thesis, we propose a full solution. One of the major issues of mobility in general is the complexity of the signaling and the cost of extra resources. [106] gives a comparison of different mobility schemes during the registration process. IMS SIP registration time lasts up to five times longer than MIPv6 as HSS database and S-CSCF are involved in the process. Nevertheless, registration time is not considered as an important parameter in our solution, as dual-homing allows registration execution while consuming the media service on the previous interface. Recent concept of Session Mobility [107] through different devices brings complex session handoff using REFER and NOTIFY methods and "replaces" header in SIP invite. It has not yet been mapped to IMS standard. As an intermediate solution, we propose a 'simple' session mobility.

Table 2.5 summarizes the existing mobility solution for NGNs. It gathers description of the protocols used in LTE standard, of mSCTP including the DAR functionality and two SIP-based solutions. It first gives nature of the mobility solution (i.e. operating layer, mobility scope, mobility management type and handover control). It then indicates the required elements for the corresponding mobility solution and the requirements on the MN side (MN Modification and Tunneling over Wireless). The complexity of the signaling and the number of elements involved in the mobility management encompasses the "complex to deploy", "scalability" and "overhead" criteria. Finally, no existing solution is fully IMS/NGN oriented and QoE enabled as session continuity is not addressed for MIP, mSCTP solutions and none of the existing solution considers QoE.

Table 2.5: Summary of existing Mobility Solution

Protocol criteria	*(DS)MIP*[71; 75]	*PMIP*[108]	*mSCTP*[83]	*SIP*[109]	*B-SIP*[89]
Operating Layer	Network	Network	Transport	Application	Application
Mobility Scope	Global	Local	Local/Global	Local/Global	Local/Global
Mobility Managing	Host-based	Network-based	Host-based	Host-based	Host-based
Handover Control	Hard	Hard, soft[79]	Soft	Soft	Soft
Required Infrastr.	HA	LMA, MAG	mSCTP proxy	MIH-based	B2BUA in BS
MN Modification	Yes	No	Yes	Yes	Yes
Tunneling over WL	Required	No	No	No	No
Handover Latency	Bad	Bad, good[77]	Good	Bad	Good
Complex to deploy	No	No	Yes	Yes	Yes
Scalability	No	No	No	Yes	No
Overhead	No	No	Yes	No	No
IMS/NGN oriented	No	No	No	Yes	Yes
QoE management	No	No	No	No	No

2.5 Conclusion

In this chapter, we have presented NGNs. 3GPP and TiSPAN IMS have been described along with the LTE architecture. Corporate Networks are playing an important role in the telecommunication area as Fixed/Mobile convergence gives more credits to WLANs. Context-awareness definition and existing architecture within NGN have also been introduced. We have depicted the main context models used to describe context information. QoS requirements and PQoS model specifications for VoIP and IPTV have also been presented. Finally mobility aspects in NGN have been raised up and discussed. Table 2.5 summarizes the different existing solutions.

The following chapter will present context-awareness in IMS platform retrieving PQoS information from the UE side and enforcing service adaptation at the core side. This chapter will introduce smart video encoding depending on video saptiotemporal dynamics.

Chapter 3

A novel IMS-Based architecture achieving adaptation for IPTV services

3.1 Introduction

The increasing proliferation of mobile devices combined with the increasing expectation of users in term of ubiquity, personalization and better experience create stringent demands and requirements on the NGNs.

Many different hand-held devices appeared last decade with different characteristics and capabilities. Most of them have several network interfaces (i.e. bluetooth, Wi-Fi, 3G, LTE, WiMax, cable). In terms of ubiquitous networking, rich media services should be able to be played on all of these devices. As described in Chapter 2, VoIP and IPTV services have different QoS requirements to satisfy the end-user. Each media stream should then be correctly adapted to the device and to the network conditions. For that, we define a User Profile (UP) and its modeling facilitating the introduction of context-awareness in the novel context-aware NGN proposed by the European project ADAMANTIUM. The ADAMANTIUM consortium presents an IMS-based adaptive IPTV service with a PQoS evaluation mechanism to control the user satisfaction. We participated in the definition of signaling for the dynamic cross layer adaptation. We introduce the innovative MCMS element that decides whether to proceed adaptation or not interworking with the Media Server Resource Function (MSRF) that enforces adaptation request. Finally, a PQoS prediction model according to video dynamics quality level is introduced in the ADAMANTIUM architecture and indicates QoE threshold in order to encode videos with an optimal bit-rate.

3.2 ADAMANTIUM architecture

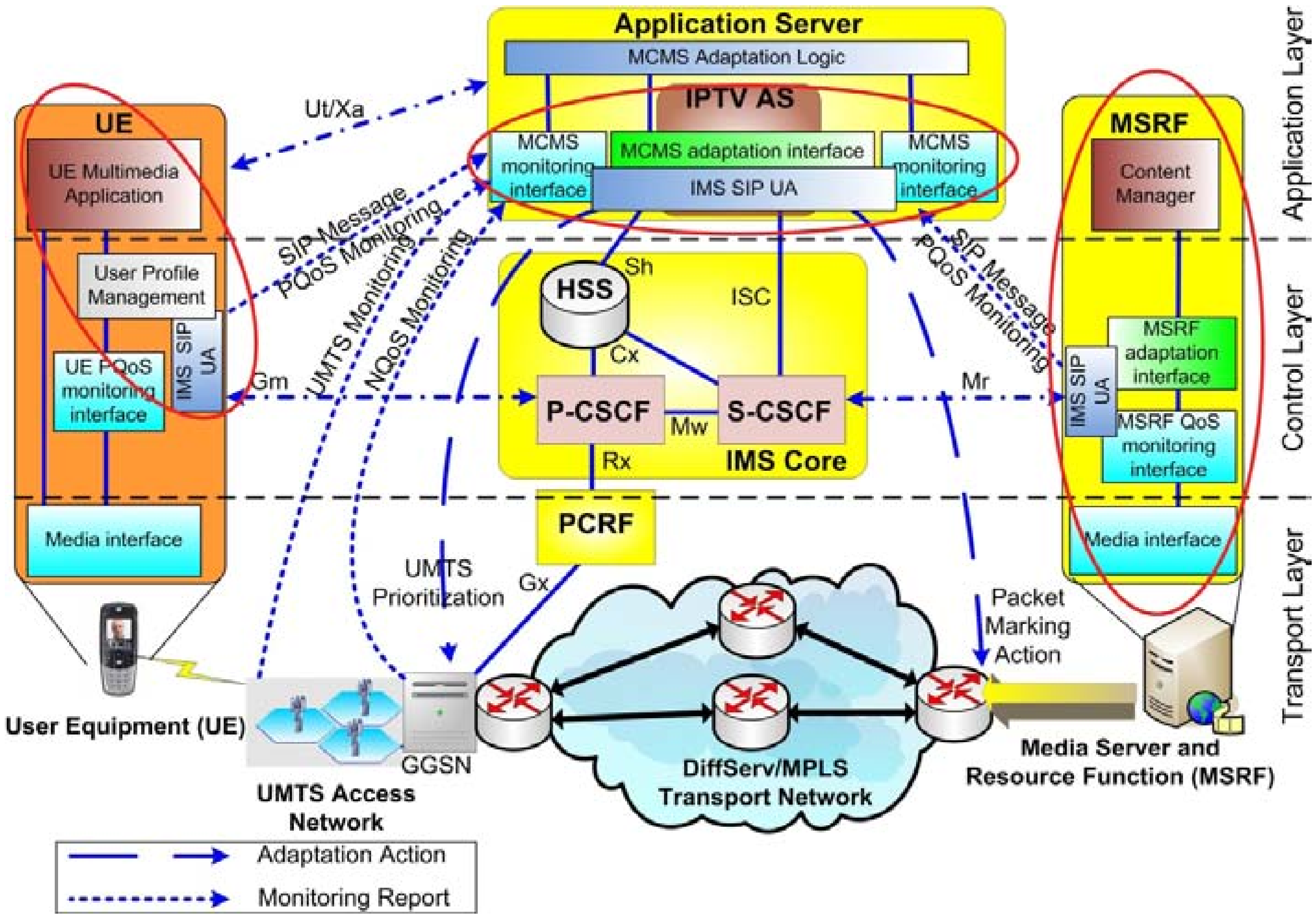


Figure 3.1: IMS Architecture with Context-Awareness

ADAMANTIUM is a FP7 project sponsored by the European commission. In this project depicted in Figure 3.1, we propose a novel IMS-compatible user-centric network management solution that employs UP management and adaptive techniques for IPTV services in order to (a) compensate network impairments (NQoS) according to the time varying conditions of the network delivery chain, (b) perform a content dependent optimization of the encoding and/or streaming parameters, and to (c) improve the end user experience/satisfaction by maximizing the delivered PQoS level and delivering content adapted to the end-user environment. Cross-layer adaptive techniques include service layer adaptation (e.g. source/terminal coding parameters and Forwarding Error Correction (FEC)), network layer adaptation (e.g. traffic policies) and link layer adaptation (e.g. service classification). This provides an efficient solution for future networked multimedia making it possible to maintain the quality of the media at every step of the media stream lifecycle from delivery to consumption. The novelty of the IMS-based architecture is mainly located at the IMS AS layer with the integration of a Multimedia Content Management System (MCMS), at the end-user terminal or the UE, at the Media Server Resource Function (MSRF) where the IPTV stream is created and at the edges of IP core transport and access networks. In this thesis, we design and implement the modules highlighted in red in Figure 3.1.

3.2.1 Multimedia Content Management System (MCMS) description

One of the essential components of the proposed solution is the MCMS, enhancing the existing IMS architecture as described in Figure3.1. This entity is in charge of intervening in the media delivery process when PQoS degradation occurs at the end-user side by performing dynamic adaptation, aiming at enhancing and optimizing the delivered PQoS level of the degraded service session. When PQoS degradation occurs at the user terminal, a PQoS alarm is triggered by the MN. Then the MCMS monitoring interfaces are initiated. A decision is then taken to answer the following question:

What are the necessary adaptation enforcement to improve the delivered PQoS?

The MCMS modules first focus on treating PQoS alarms coming from the MN. Then they trigger monitoring actions, gather the monitoring information until either alarm ceases or degradation persists/increases and finally launch the actual PQoS alarm. They collect the network statistics (i.e. core, access, terminal) and the service delivery information (i.e. MSRF) in order to define and apply an optimal cross layer adaptation action across the delivery network chain and media delivery lifecycle (i.e. service generation node, core network, access network and end-user terminal). The final objective is to maximize the user satisfaction. As shown in Figure 3.1, MCMS modules are located in the AS layer of the IMS architecture. MCMS modules widen the IPTV AS scope with real- time monitoring important information, including impairment information from source (e.g. content dynamics), core/access network (e.g. network/link QoS) and terminal (e.g. delivered PQoS). IPTV AS is mainly responsible for SCF as defined in TiSPAN v2 standardization [13], IPTV service provisioning, IPTV service personalization and other IPTV specific services such as voting or advertisement functionalities. The SCF has a threefold responsibility:

- Service Authorization during session initiation and session modification, which includes checking IPTV users' profiles through the 'Sh' interface in order to allow or deny access to the service;
- Credit and limit control which collects charging information and sends it towards the charging system for consolidation with other charging information collected in other strategic network elements;
- Selection of the relevant IPTV media functions. The SCF is choosing one of the three different IPTV services available which are Content on Demand, Broadcast and Network Personal Video Recorder.

The SCF then manages the IPTV session management, constantly aware of the UE status. Therefore, IPTV AS can also be extended with the MCMS Adaptation Interface in order to trigger cross-layer adaptation including service layer adaptation (e.g. source/terminal coding parameters and FEC), network layer adaptation (e.g. traffic policies) and link layer adaptation (e.g. service classification). This MCMS Adaptation Module is controlled by some intelligent decision logic called MCMS Adaptation Logic within the MCMS system. The integration of the MCMS modules in the Application Layer makes the PQoS aware management available for any kind of access or transport network. Figure3.1 only shows UMTS Access network, but this solution can also work for fixed access, WLAN or even cable access with corresponding policy and rule functions.

3.2.1.1 MCMS Framework

The MCMS has several components in order to achieve cross layer adaptation techniques across multimedia delivery chain. They are separated in three categories :

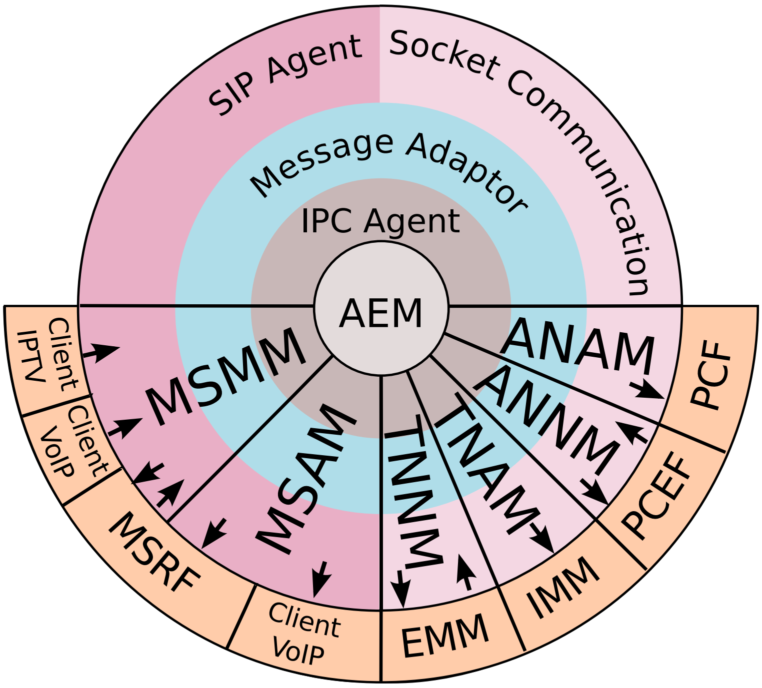


Figure 3.2: MCMS detailed architecture with interactions

monitoring modules, adaptation modules and logic central node called Action Engine Module (AEM).

For monitoring purposes, the following modules are considered :

- The Multimedia Service Monitoring Module (MSMM) monitors media session receiving SIP requests from S-CSCFs through the IMS Service Control (ISC) interface. It also monitors the PQoS and CNR at the end-user mobile terminal device. Finally it receives the video content dynamics and encoding parameters of the delivered video stream sent by MSRF;

- The Transport Network Monitoring Module (TNMM) is used during the dynamic cross layer adaptation procedure for monitoring at the DiffServ/Multi-Protocol Label Switching (MPLS) core transport network. Towards this, the appropriate External Monitoring Module (EMM) interface is integrated at the egress edge router of the core transport network for enabling interaction and communication with the TNMM of MCMS;

- The Access Network Monitoring Module (ANMM) monitors the UMTS access network statistics.

For adaptation purposes, the following modules enforce adaptation orders chosen by AEM:

- The Multimedia Service Adaptation Module (MSAM) performs adaptation actions at the end-user terminal device and at the service generation entity;
- The Transport Network Adaptation Module (TNAM) applies the adaptation actions to the DiffServ/MPLS-enabled core transport network through the Internal Marking Module (IMM) at the ingress router of the core network;
- The Access Network Adaptation Module (ANAM) applies the adaptation actions as decided by the AEM to the UMTS access network through the IMS PDF module. The PDF in turn applies them at the Gateway GPRS Support Node (GGSN) by performing service bearer classification in order to improve its QoS characteristics and therefore enhance the delivered PQoS level.

The AEM is responsible for making optimal adaptation decisions based on the monitoring of network and perceptual statistics, gathered by IMS-based monitoring and adaptation modules. After obtaining the monitoring data, AEM, by exploiting theoretical mapping frameworks between NQoS and PQoS, processes all the collected statistics and defines a perceptually optimal cross layer adaptation action.

Figure 3.2 gives an overview of the MCMS modules with their interactions with the different nodes in the network and the central AEM. It is divided into a series of components, among them:

- Outgoing interfaces, which cover the specification of the interfaces for communication from the AEM towards the monitoring and adaptation modules. Note that these outgoing interfaces are specified as part of the AEM module, but implemented by the corresponding MCMS module. They guarantee that AEM logic is not affected by changes in the monitoring and adaptation modules implementation.
- Incoming interfaces, which cover the specification of the communication interfaces from the MSMM module towards the AEM. It is the communication API with the AEM. The most important events to communicate include IMS session related events and PQoS alarms. These events can be notified

to the AEM asynchronously, but in this new version of the prototype they are sequentially processed by the AEM. This behaviour is provided by the communication interface which must ensure that no event is lost. Note that the incoming interfaces are not only specified as part of the AEM module, but also implemented by it, in the "Adapter" components.

- As previously mentioned, adapters for the incoming interfaces provide the implementation of those interfaces. They distribute the incoming events towards the appropriate AEM logic module. From the MSMM, the adapter receives IMS session related events, PQoS alarms and monitored VoIP or video (IPTV or VoD) client and server parameters. From the ANMM, the adapter receives monitored access network QoS parameters. And from the TNMM, monitored transport network QoS parameters are received. All monitored parameters are delivered to the Monitoring Info Collector, IMS Session Events to the IMS Session Handler, and PQoS alarms to the PQoS Alarm Handler.

  The adapters hide the details of the internal communication mechanism used between the different MCMS monitoring modules and the AEM, and conceal deployment issues such as whether the MCMS system is distributed in one or multiple processes or if it is deployed into a single host or multiple hosts. Although the communication between the AEM and the MCMS Monitoring/Adaptation modules is very simple in principle, the adapters provide a flexible enough architecture to support more complex communication mechanisms in the future.

The different MCMS modules are composed of a communication layer (SIP or Socket Communication), a message adaptor in order to prepare the information to dialog with the AEM and finally a Inter Process Communication (IPC) Agent to enable the interaction between different processes.

The MSAM and MSMM modules interface with the IMS core network. They are acting then as SIP Application Servers generating and/or terminating SIP dialogs. SIP messages are decoded, formatted and sent to the AEM through an IPC agent to enable the interaction between different processes.

3.2.1.2 MCMS functionalities

The adaptation and monitoring modules are in idle state waiting for orders from AEM, except for the MSMM module which is also listening for any session event or alarm triggered by the monitoring interface from the MN. The initialization of a session of the MN starts a passive monitoring of the session in each layer:

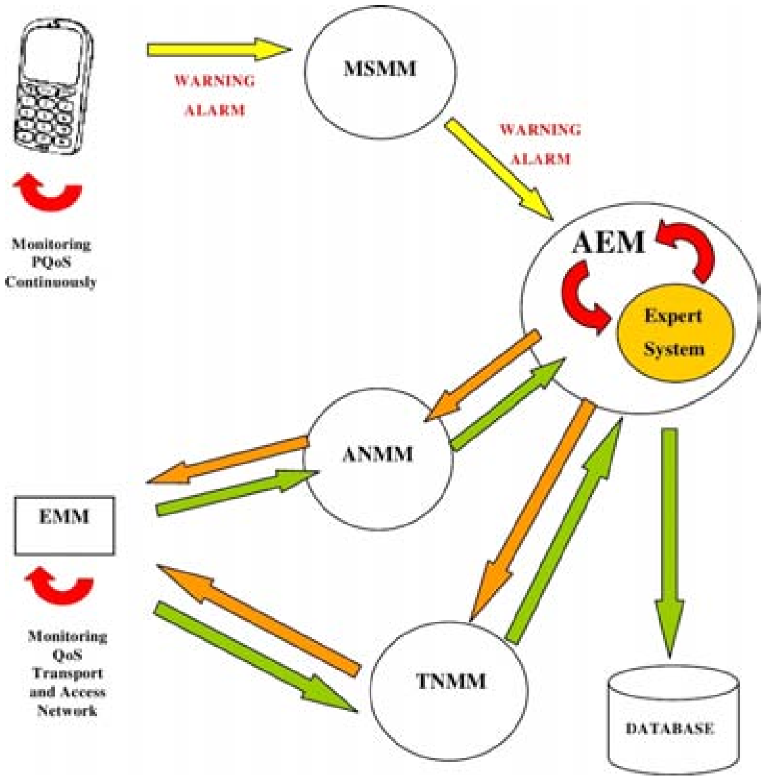


Figure 3.3: Example of Warning Alarm Reception

- Service layer: MSMM is responsible for the monitoring of the service layer interfacing with the MN's monitoring module. The monitoring module within the MN collects QoS session statistics of the incoming RTP streams and monitors the defined threshold of some parameters. If a PQoS degradation below one of these thresholds is observed for a given duration, the MN's monitoring module triggers one alarm back to the MSMM. The MSMM forwards the alarm to the AEM which requests specific actions. Sometimes the AEM orders one or several modules to start monitoring actions. This means that the MSMM must be ready to receive monitoring data and to ask for monitoring data to the MSRF monitoring module. The MSMM internally is divided into two processes to ensure these two tasks, one process to wait for alarms from the MN and another process to wait monitoring data requests from the AEM. This process is shown in Figure 3.3 and Figure 3.4.

- Network layer: The monitoring of the network layer is responsibility of the TNMM through the EMM. The EMM collects QoS statistics from the Transport Network continuously and then sends them to the AEM across the TNMM module when it would be requested. In this case, the EMM does not send alarms to AEM, the EMM only sends monitoring data of the network layer when it is requested by the AEM. So, the TNMM is not listening continuously to the EMM. This process is shown in Figure 3.3 and Figure 3.4.

- Link layer: The monitoring of the link layer is responsibility of the ANMM through the EMM. The Access EMM collects continuously QoS statistics from the Access Network that are sent to the AEM across the ANMM module when it is requested. As in the network layer, the EMM does not send alarms so the ANMM is not either listening continuously the EMM.

While the passive monitoring occurs, the MCMS waits in an idle mode which changes to active as soon as the MN triggers an Alarm event to the MSMM. A two alarm method is chosen for more flexibility in experimenting and minimizing the amount of data exchanged. The alarms are either:

- a warning alarm which is triggered through the MN's monitoring interface of the terminal when the PQoS degradation is below the first defined threshold with longer duration than a specified time. This prevents false alarm activations. The alarm captured by the MSMM is delivered to the AEM, which in turn starts the monitoring stage, interrogating those monitoring modules whose QoS statistics are useful for selecting the adaptations if a red alarm is triggered. These modules are selected by the expert system depending on the rules used and the alarm data provided. The monitoring data is stored in the database for further use. A possible example of a warning alarm reception is shown in Figure 3.3.

  In the case of Figure 3.3, the MN's monitoring module is collecting continuously QoS session statistics in the terminal if the PQoS is below the first

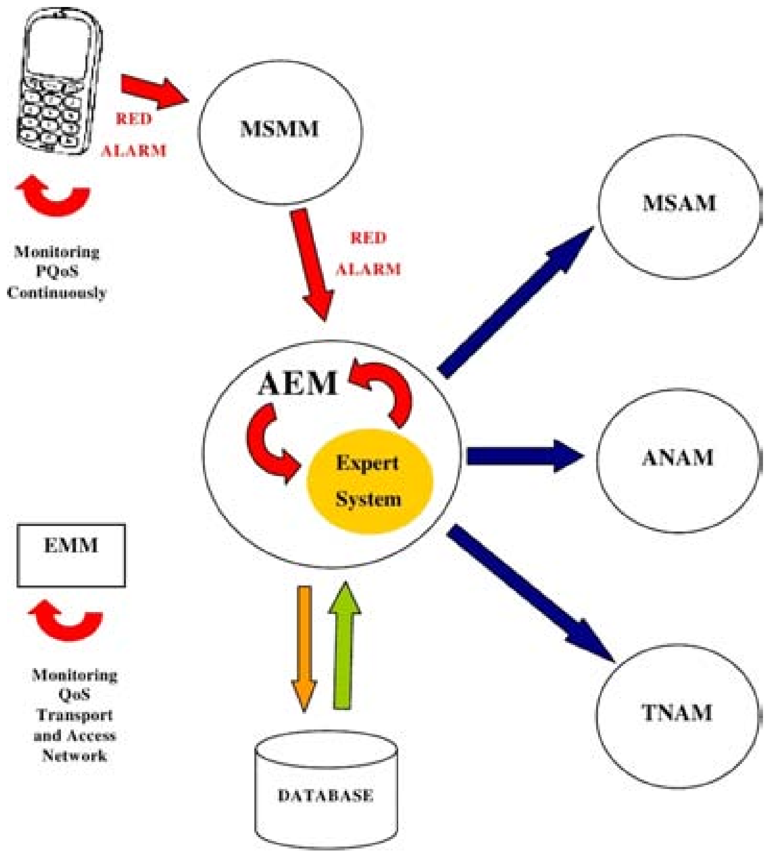


Figure 3.4: Example of Red Alarm Reception

defined threshold for enough time. The MN sends a warning alarm to the MSMM which forwards it to the AEM. The AEM asks the Expert System for orders. The Expert System suggests asking the ANMM and the TNMM for monitoring data. Then the ANMM and the TNMM ask their respective EMM for monitoring data and forward it to the AEM which stores that monitoring data into the database. In Figure 3.3, the yellow arrows symbolize the warning alarms sent by the MN's monitoring module, while the orange arrows symbolize the monitoring data requests and the green arrows symbolize the monitoring data returned.

- a red alarm which is triggered through the MN monitoring module of the terminal when the PQoS level at the user terminal decreases enough and gradually reaches the second defined threshold. Once the red alarm reaches the AEM, the previous stored data and the alarm data is evaluated by the Expert System and some adaptations are selected in order to improve the deteriorated

PQoS. An example of warning alarm reception is shown in Figure 3.4.

In the case of Figure 3.4, the MN's monitoring module is collecting continuously QoS session statistics in the terminal if the PQoS is below the second defined threshold for enough time. The MN sends a red alarm to the MSMM which forwards it to the AEM. The AEM asks the Expert System for orders. The Expert System suggests asking the database for the stored monitoring data. When the AEM gets the stored monitoring data, it asks the Expert System for orders. Then the Expert System suggests ordering adaptations to the respective adaptation modules, ANAM, TNAM and/or MSAM. In Figure 3.4, the red arrows symbolize the red alarms sent by MN, while the orange arrow symbolizes the monitoring data requests, the green arrow symbolizes the monitoring data returned from database and the blue arrows symbolize the adaptation requests. Once the MCMS has been switched to active mode because of a warning alarm or red alarm, the current QoS relative statistics across the network delivery chain (i.e. from the core and access network) are reported to the AEM via the ANMM and the TNMM modules. Also, reports from the encoder about the coding parameters, the content dynamics of the multimedia service, the current FEC scheme (if applied) and the decoding parameters at the terminal (i.e. buffer scheme) are reported to the AEM via the MSMM. If a VoIP session is considered, then the session monitoring takes place at both end-user terminals that participate in the session, where the voice is encoded and decoded at the sender and receiver terminal respectively. All the collected statistics are further exploited by the MCMS, through a sophisticated processing procedure and a decision algorithm. This sophisticated algorithm is implemented in the Expert System, which decides the appropriate actions and adaptations across the network delivery chain, in order to optimize the delivered PQoS, but maintaining the total service traffic constant. After the monitoring phase of the network statistics across the delivery chain has been completed, the adaptation phase is following. This is performed through the adaptation action modules of the MCMS (i.e. MSAM, TNAM and ANAM). Towards this, the AEM processes all the received statistics from the ANMM, the TNMM and the MSMM in order to define the adaptation actions across the network delivery chain, aiming at the optimization of the NQoS, which leads to maximization of the user satisfaction and the delivered PQoS level, without altering the total service traffic of the bearer. The System Expert executes a sophisticated processing procedure and a decision algorithm, and it decides the appropriate actions and adaptations across the network delivery chain, in order to optimize the delivered PQoS. The AEM needs to have some intelligence to process the information coming from the different Monitoring Modules, and make decisions about the best solution to improve the PQoS, sending instructions to the Adaptation Modules. As described above, input data is collected from the different monitoring modules, and it is used by the AEM to make some decisions to be sent as adaptation commands to the adaptation modules. In the following section, the input data coming from the different monitoring modules are listed. All the MCMS modules communicate with the AEM using a communication library, which implements a tail of POSIX messages. This tail is created and destroyed by the MSMM and only the remaining modules use it.

3.2.1.3 AEM framework

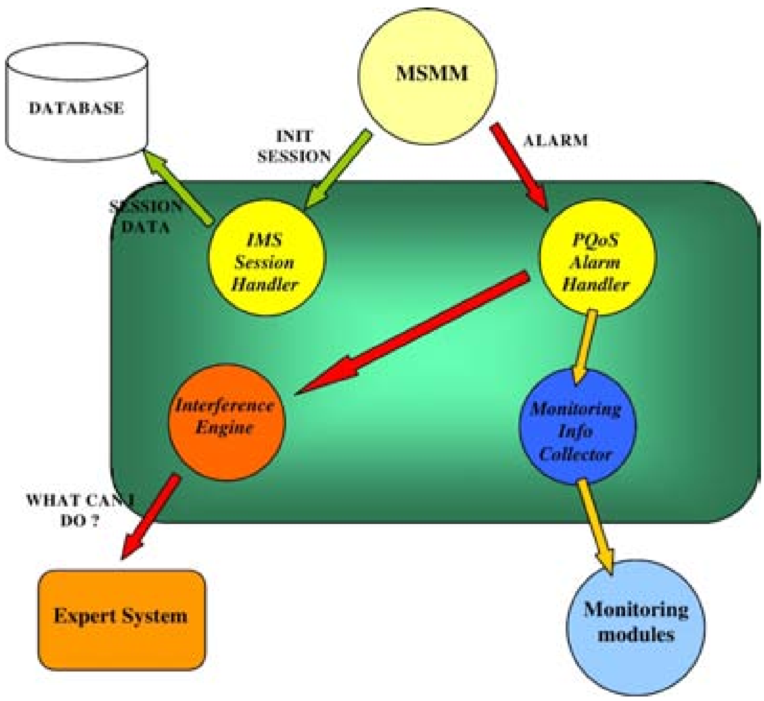


Figure 3.5: AEM internal logic architecture

The AEM depicted in 3.5 is composed of :

- an Expert System which selects the best cross layer adaptation in order to optimize the delivered PQoS level based on CLIPS [110] and provides a complete environment for the construction of rule and/or object based expert systems. The rules are inserted by the operator. The forward chaining is chosen as the method of reasoning so the inference rules and knowledge representation will use session, monitoring and adaptation data. Figure 3.6 shows the data sources used by the Expert System for making decisions. The Expert System can suggest to generate a monitoring request when a warning alarm is received. These monitoring data are stored into the database and reused by the Expert System to decide whether to proceed through an adaptation action or not.

- a Database decomposed into the following elements:

  – knowledge base for the expert system. The rules stored in this database will be used by the rule-based expert system implemented in AEM.

  – Subscriber information to store AEM behaviour for each user.

  – Provisioning server provides external interface for remote provisioning of the static data required by AEM to work, and stores such data for future retrievals.

- an Internal Logic Manager receives alarms, communicates with the Expert System and asks monitoring data to the monitoring modules and sends adaptation orders to the adaptation modules. The Internal Logic Manager is composed of :

  – an IMS Session Handler managing the IMS session related and storing them in the database. It is important to receive IMS session termination notifications from the network for liberation of internal AEM resources.

  – PQoS Alarm Handler processes the PQoS alarms triggered from the terminal equipments. The two threshold PQoS alarm approach has been chosen, distinguishing between "warning" alarm and "red alarm". Besides, the duration that a threshold is violated is also considered when deciding which actions to take.

  – Monitoring Info Collector requests the monitoring information to the service delivery chain components, through the usage of the appropriate

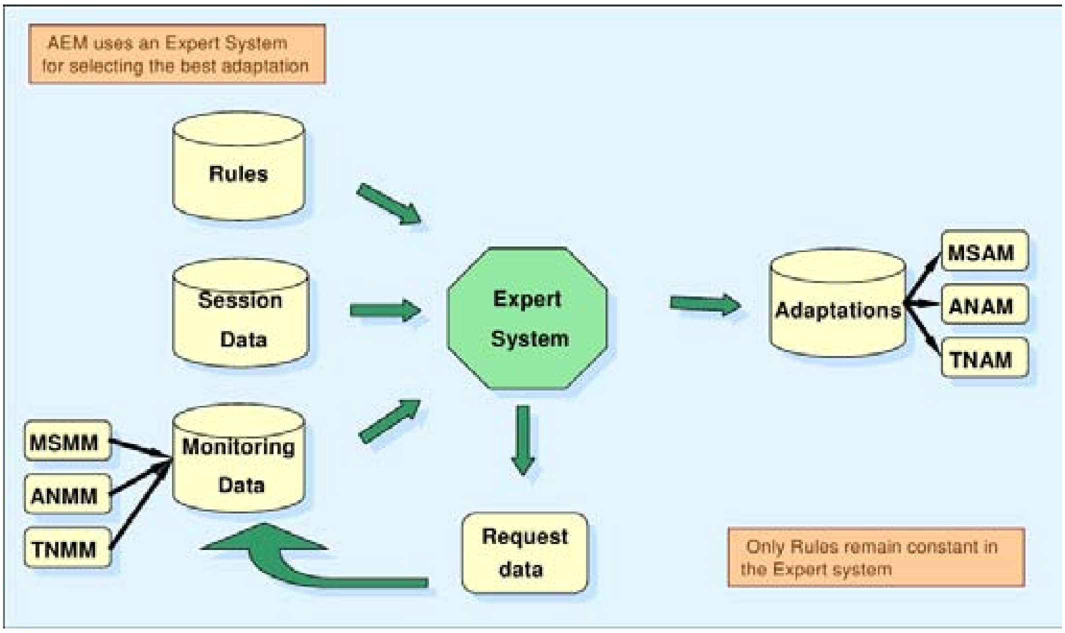


Figure 3.6: AEM expert system data sources

outgoing interfaces. It also takes care of receiving the monitored parameters and storing them for further processing.

The following subsection presents solutions to achieve context-awareness within IMS. Along the different UP frameworks, we have chosen the markup scheme-based framework in order to provide efficient and transparent to existing GUP.

3.2.2 User Profile & Mobile Node design

This contribution is built on the Markup scheme-based languages and tools. The aim is to propose a context-aware framework with high responsiveness and performance in order to ensure the real-time requirements of multimedia services. In the following, we present the proposed context model (UP model), its management and the context-awareness middle-ware implemented and deployed in the end-user's equipments.

3.2.2.1 User Profile Modeling

The first step of our work identifies the pertinent context information that will constitute our user profile. Thus, we have considered the definition of a context as given in [111] that assimilates the context to any information that can be used to characterize the situation of an entity. We then started by identifying the entities which we believe to be necessary parts of the context. Then, we divided our user profile into the following five components according to the identified entities:

A. General user profile

It contains basic information about a user such as name, address, phone number, gender, age, profession, etc. It also includes user disabilities such as color perception or hard-to- hear, and all the user preferences. Those information are service independent and can thus be applied to all kinds of services.

B. Device profile

It collects the descriptions of all user devices. For each device, it gives the user's hardware and software information and characteristics like brand, serial number, CPU speed, memory capacity, battery life, OS name, version and vendor, related peripherals with their characteristics, etc.

C. Network profile

It describes networks that the user can access. For each network it gives, the type, characteristics, the medium used for transmission is provided along with charging and billing information, etc. In operator's networks, the Network profile includes the information and clauses defined in the Service Level Agreement (SLA).

D. Service profile

It records all the information about services (name, version, related protocols and ports, etc.) and the content that those services manipulate (multimedia content, databases, files, etc.). It also contains information that can serve in the publishing, discovering and invoking, presenting and billing phases.

E. Context profile

It contains information about the user environment factors such as time, date and location information, the serving device and network, a set of applications that are running, the perceived values of QoS parameters at a given moment like available bandwidth, loss and error rate, etc. This constitutes the dynamic part of the user profile, mainly composed of volatile data. Therefore, it must be filled automatically by sensing the environment.

It is obvious that the user profile will be extended whenever a user wants to access a new service or acquires a new device or a new network subscription. The user profile must therefore be as extensible as possible. Besides, the fact that the user profile is

organized into sub-profiles, it should facilitate the insertion of new information.
To be efficient and shared between different services, the user profile must be as interoperable, extensible and laconic as possible. For this, XML and its dialects seem to be the best candidates to describe and format it. We thus chose to structure our user profile in an XML document. Although XML does not have the semantic power of anthologies, it is especially suited to the description of a user profile. Its tree structure presents a good way to express a decomposition of the user profile in one or more nodes which, in turn, can be decomposed and this on several levels. It also allows an easy access to each node by using XPath. The expression of information according to others can be done using attributes.
For the XML user profile validation, we define its format, its tree structure and the types of data in an XML Schema [36]. XML Schema offers all the primitive types that are usually used and many features to create other types.

3.2.2.2 User Profile Management Middleware Architecture

In order to cope with the heterogeneity of their operational environment, the applications must benefit from user profile management services. These services must be as transparent as possible for the applications. For this, we propose a separate middleware that performs the tasks of the management and the transport of the user profile between the consumer and the provider application. The architecture of the middleware is shown in Figure 3.7. The different modules that compose the middleware are described in the following:

1. User Profile Manager (UPM): it is the central module of the middleware. It interacts with all other modules in order to construct the adequate user profile according to the needs of the application that performs the request. When an application starts to run, the UPM retrieves the request of the application from the API, enables the correspondent context functions, asks the query manager for the correspondent query model, then performs the retrieved query on the user profile and finally transmits the new constructed user profile to the transport manager.

2. Context manager (CM): it is composed of a set of functions that have the role of monitoring the operational environment and collecting information about it.

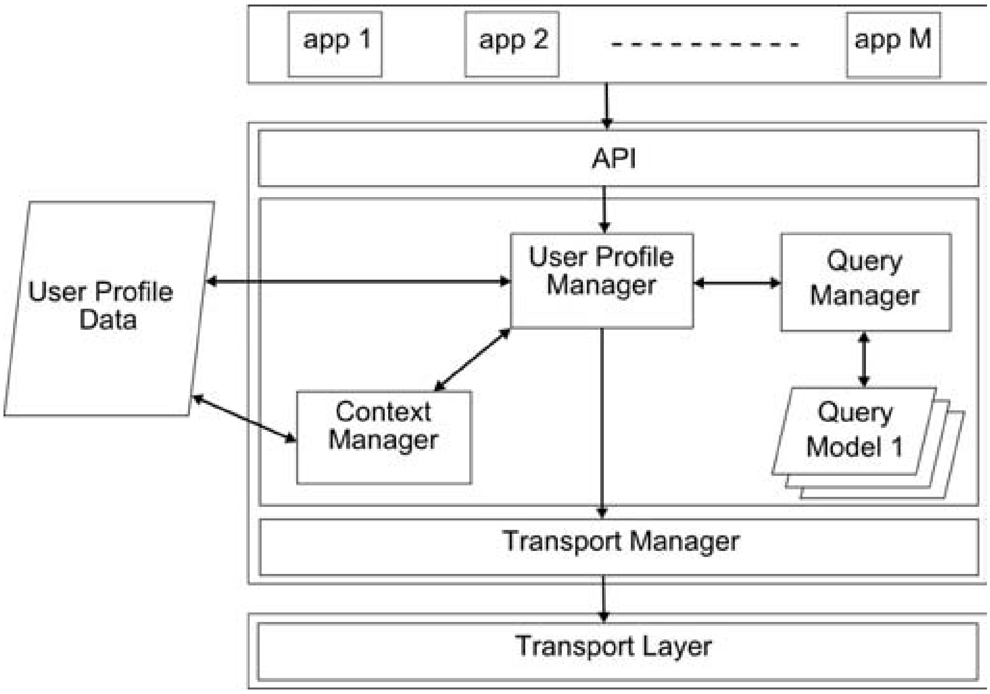


Figure 3.7: User Profile management

Thanks to it, the dynamic part of the user profile that is composed of volatile data representing a context, is regularly updated. This update is made in both a reactive manner when receiving events like the start of an application or in switching from one network to another, and a proactive manner according to a prefixed period.

3. Query manager (QM): it is in charge of validating and saving a query model for each application. A query model is an XQuery [112] file that expresses all the needs of the application in term of user's information. It is constructed once at the application installation and used whenever this application requests the middleware. However, it can evolve according to the needs of the application. In addition to the multiple facilities offered by XQuery in the presentation of the queries results, we have chosen this language for its ability to express conditions thanks to its Where clauses. It becomes, then, easy to specify the information that should be returned according to the context.

4. Transport manager (TM): it ensures the transport of the user profile from the consumer application to the correspondent provider application through a signaling protocol. The user profiles related to different applications are

encapsulated into signaling messages.

5. API: it constitutes the only interface point between the applications and the middleware. Its main function is to provide a set of primitives that allow the retrieval of the application's requests in the consumer side and the provision of a user profile in the provider side.

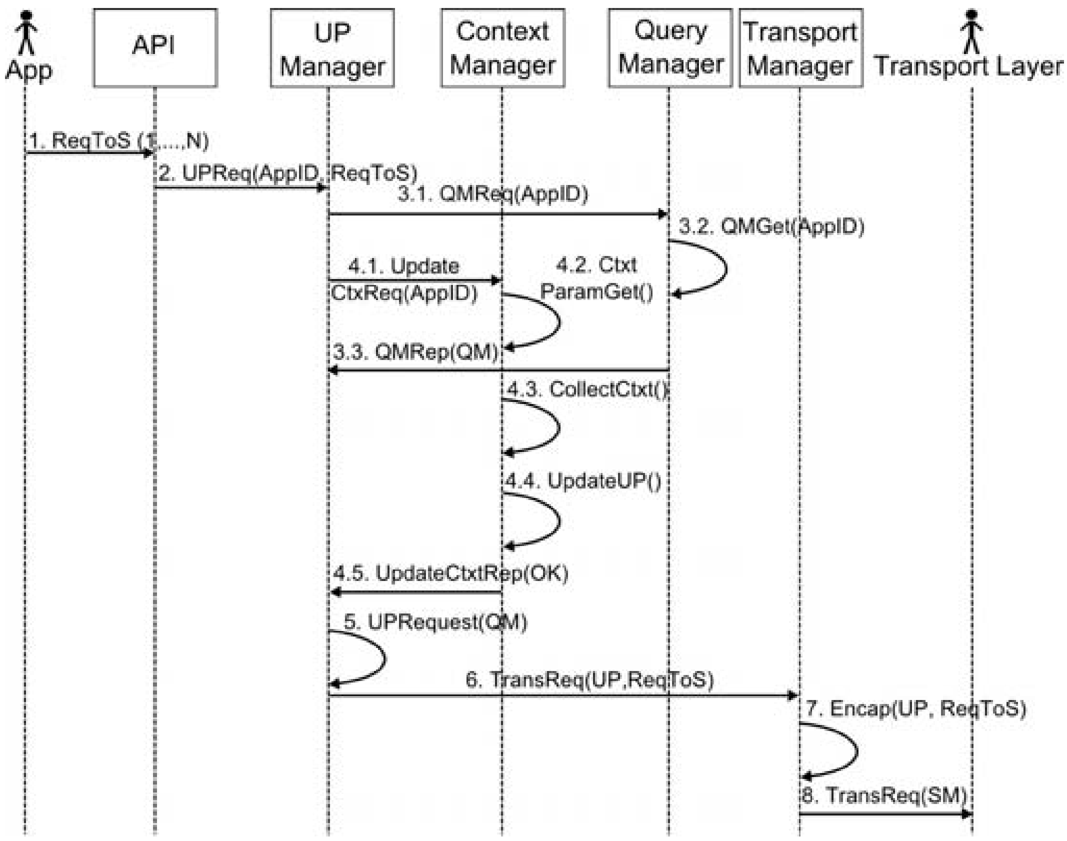


Figure 3.8: UP construction diagram

3.2.2.3 User Profile Construction Process

The different interactions between the components of the middleware that lead to the construction of an instance of the user profile related to a specific application are illustrated in the sequence diagram of Figure 3.8. These interactions are detailed in the following:

1. The consumer application transmits its request to the middleware API.

2. The API transfers the request to the UPM that retrieves the application identifier for which it will shape a new user profile.

3.1. The UPM asks the QM for the Query Model of the application. It includes in its request the application identifier.

3.2. The QM retrieves the Query Model of the corresponding application identifier, received from the UPM.

3.3. The QM replies to the UPM with the Query Model in its response.

4.1. In parallel to step 3, the UPM notifies the CM that the AppID application starts to run and asks it to update the User Profile with the current context information.

4.2. The CM retrieves the context parameters to monitor from the User Profile.

4.3. The CM enables the monitoring functions that collect the context information.

4.4. The CM updates the corresponding elements of the User Profile with the collected information.

4.5. It then notifies the UPM that the UP was updated.

5. The UPM executes the Query retrieved from the QM and constructs the new shaped UP that responds to the application's needs.

6. The UPM transfers the new shaped UP to the TM with the request ReqToS received at the beginning.

7. The TM encapsulates the received parameters in a Signaling Message.

8. Finally, this module transmits the Signaling Message to the transport layer which will ensure its transport to the server application.

The proposed middleware of UP is integrated within the IPTV UE as depicted in Figure 3.9 that enables the context-awareness within IMS-based ADAMANTIUM architecture. The UP management enables the terminal's interaction with the appropriate interfaces/modules of the MCMS. The UP management is in charge of collecting information of the user's environment and sending them to the decision logic modules. Based on the monitoring data gathered by the UP management, the UE initiates an IPTV media session with all information needed to adapt the request service to its current environment. As soon as the IPTV session is established, monitoring module in each media session is created in initialized. Whenever the quality lowers a warning threshold, UP management sends estimated PQoS values to the MSMM in the MCMS. When the quality suffers from quality impairments, UP orders to send a SIP message to initiates a red alarm status which implies adaptation somewhere in the delivery chain. Dynamic adaptation procedure is launch according to the monitored network statistics. The MCMS modules are implemented on a real

IMS platform comprising an UMTS access network where IPTV over IMS services are integrated.

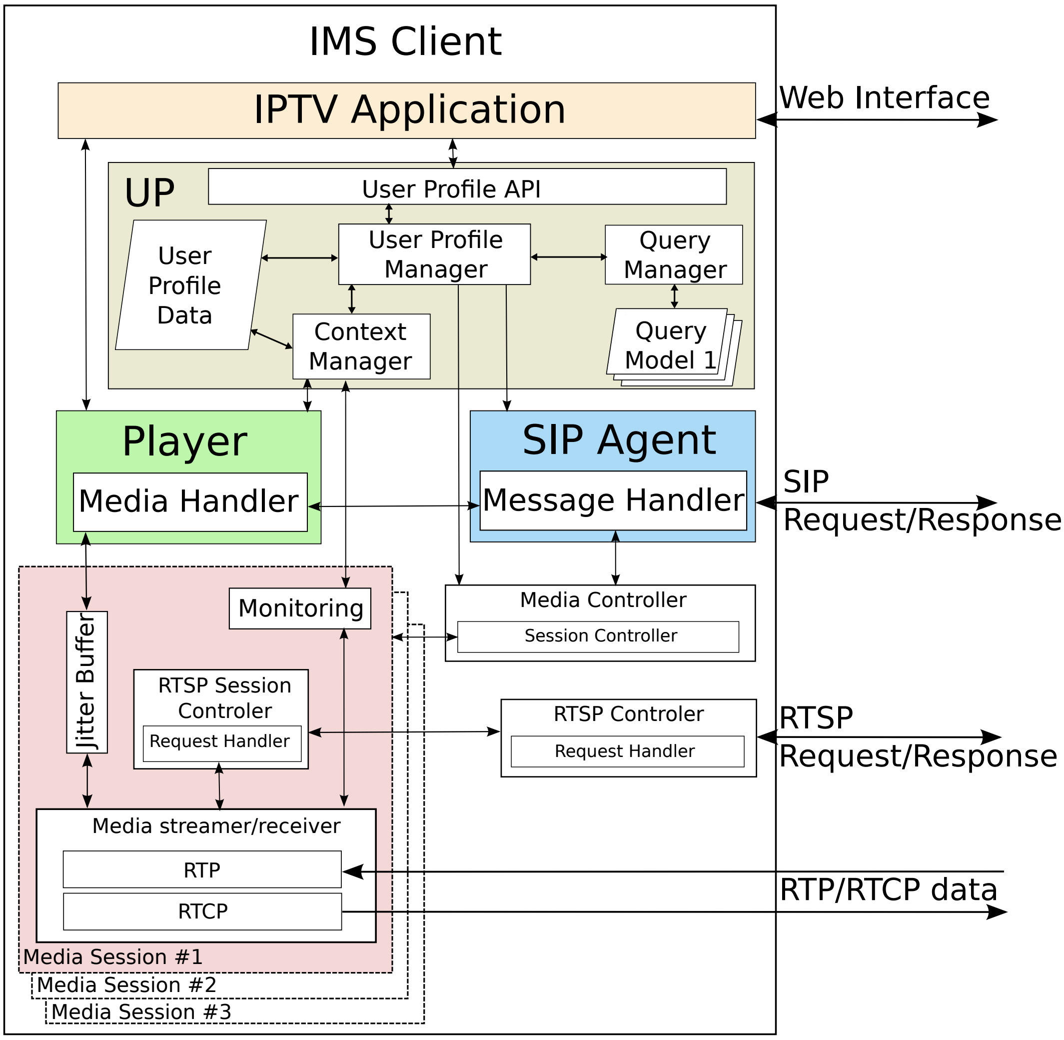


Figure 3.9: PQoS aware IMS client architecture

3.2.3 MSRF functionalities

The MSRF is an IMS based MRF module with an additional threefold responsibility:

- IPTV service generation, session management and streaming,
- The monitoring of the spatial and/or temporal content dynamics [2] along with the selected encoding/streaming parameters,
- The adaptation, according to the MCMS commands, of the IPTV encoding parameters and/or the respective FEC value.

The MSRF terminates the SIP session initiated by the end-user. It creates and manages IPTV media sessions taking into account the spatiotemporal content dynamics [9] in order to minimize the bandwidth usage while providing a satisfying user experience. When the MCMS PQoS alarm is triggered, it receives from the MCMS a request to monitor the corresponding Real-time Transfer Control Protocol (RTCP) information and sends it back to the MCMS. Then if the adaptation process is triggered, the MSRF receives the adaptation request and modifies the IPTV stream parameters.

Within this subsection, we defined and specified the different ADAMANTIUM modules on which we contributed. The following subsection describes how the signaling is processed between these modules in order to perform dynamic cross layer adaptation.

3.3 Context-Aware Adaptation for IPTV

The ADAMANTIUM architecture depicted in Figure 3.1 defines the modules involved in the context-aware IPTV service delivery. In order to improve the end-user satisfaction, we propose an adaptive IPTV service based on PQoS evaluation analyzed by the MCMS. This subsection describes the signaling between MCMS and MSRF to enforce the adaptation ordered by the AEM engine.

First, we present the adaptation done during the IPTV service initiation based on the UP information. Then we describe the signaling to process dynamic adaptation and the MSRF functionalities that adapt the IPTV streams.

3.3.1 Adaptation enforcement

The Figure 3.10 describes sequence diagram of the messages that are exchanged when an IPTV session is established and torn down between a UE and the MSRF through IMS. An example of adaptation mechanism with SDP [113] re-negotiation is also shown. For sake of simplicity in the diagram, we neither represent messages flowing through the CSCF elements nor do we represent the Rx interface between P-CSCF and Policy and Charging Rules Function (PCRF). The Ut/Xa reference points, used for service profile configuration are also not shown.

The successfully registered UE initiates the IPTV service sending a SIP INVITE

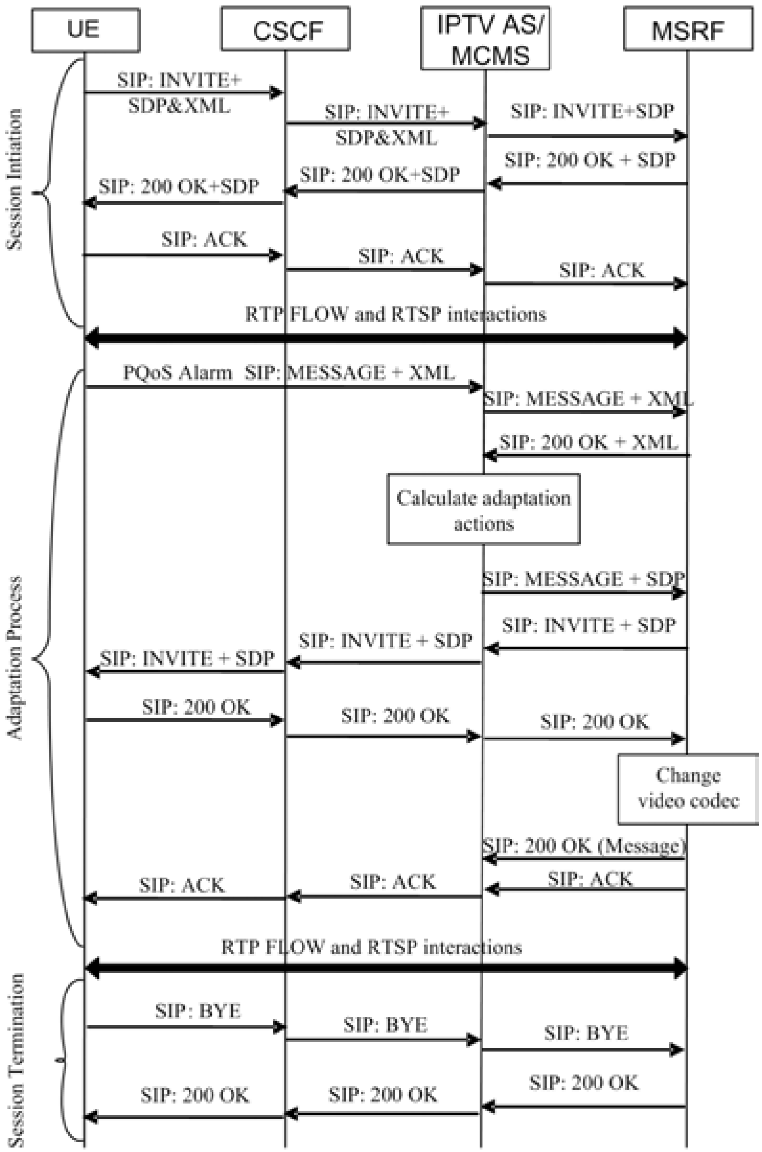


Figure 3.10: IPTV session handling for adaptation with SDP re-negotiation

request with two bodies. A first SDP body describes the media session (codec, video to play, framesize, framerate, bitrate) as depicted in Figure 3.12 and the second XML body describes user environment as depicted in Figure 3.11. The first SIP contact entry point for the UE is the P-CSCF. The latter forwards the INVITE request to S-CSCF. The latter detects an IPTV service initiation thanks to the service-triggering information presented in the form of initial Filter Criteria (iFC) downloaded from HSS during the MN's Registration process. S-CSCF forwards the request to a specific IPTV AS through the ISC interface. IPTV AS treats the request by parsing the SDP and XML bodies in order to retrieve the media

Figure 3.11: XML body in SIP messages sent by the UE before and after adaptation

filename to play and the user's environment information. The IPTV AS may also use the IPTV user profile to customize the user experience according to the user's preferences. Then, the IPTV AS forwards the "customized" INVITE request to a corresponding MSRF which terminates the SIP dialog. The MSRF sends back a 200 OK message with the final negotiated SDP body and the CSCF elements forward it to the UE. Then, the MCMS adaptation interface is informed about the active service sessions since it belongs to IPTV AS. The MCMS modules do not perform any monitoring or adaptation actions (idle mode) but simply wait for the reception of a PQoS-degradation alarm from the end-user terminal, indicating poor perceptual quality and therefore bad user experience. The idle mode is essential for the MCMS scalability abilities, because during this passive operating mode resources are not consumed. From this point, we distinguish two main use cases:

1. The MCMS adaptation interface triggers an adaptation that requires a renegotiation of the SIP session. In that case, the MCMS adaptation interface will send a SIP Message to the MSRF. The latter will renegotiate the session with the user terminal, in order to inform the terminal of the new video parameters such as changing the video codec. A Re-INVITE is used to renegotiate the SDP parameters because the INVITE process acknowledges the renegotiated SDP (request, response, acknowledgment) contrary to an UPDATE process [114] (request, response).

2. The MCMS adapts the required parameter without informing the user terminal. The user terminal adapts itself to the changed video and audio streams such as a new bitrate of the video. A simple SIP UPDATE is sent from IPTV AS on the behalf of the MSRF. Only the first case is depicted in Figure 3.10 as the adaptation process involving SDP negotiation is more complex than the second case.

The adaptation policy degradation action in terms of temporal degradation (i.e. Frame Rate) and Encoding Bit rate in order to set the appropriate IPTV parameters such that the perceived video quality is kept as high as possible, according to the available bandwidth.

3.3.2 MSRF adaptation process

The MSRF is driven by an IMS SIP UA. It instantiates a streaming session when it receives a SIP INVITE message from the IPTV. Each streaming session is composed of:

- A Media Streamer that streams media resources to the requesting client,

```
v=0
o=aude 13460 13460 IN IP4 192.168.1.153
s=ADAMANTIUM streaming session
t=0 0
a=tool:GStreamer
a=type:broadcast
a=media:a.avi
m=video 5000 RTP/AVP 96 34 32
c=IN IP4 192.168.1.153
a=rtpmap:96 H264/90000
a=rtpmap:34 H263/90000
a=rtpmap:32MPV/90000
a=framerate:25.0
a=framerate:CIF
m=audio 5002 RTP/AVP 8
c=IN IP4 192.168.1.153
a=rtpmap:8 PCMA/8000
```

Figure 3.12: Example of SDP body

- a Real-Time Streaming Protocol (RTSP) [115] session controller that controls the Media Streamer with respect to the commands received from the client
- a MSRF/MCMS Interface (MMIF), which sends monitoring data to the MCMS monitoring interface and enforces adaptation commands received from the MCMS.

Only the first case is depicted in Figure3.10 as the adaptation process involving SDP renegotiation is more complex than the second case.

3.3.2.1 MSRF components involved in the adaptation process

The different components of the MSRF are the following:

- The IMS SIP UA Component handles all SIP incoming requests destined for MSRF. During the session initialization, it communicates with the Media Controller for media resource availability and the streaming session creation (session streamer instantiation to serve one user). Then, during the streaming, the IMS SIP UA listens to monitoring and adaptation requests and forwards them to the MMIF. It then sends back then SIP requests/responses depending on the MMIF status. Finally, the IMS SIP UA also listens to session release by communicating with the Media Controller for the session teardown.
- The MMIF collects monitoring data from the media streamer and sends them back to the MCMS monitoring interface. MMIF also receives adaptation orders

from the MCMS adaptation interface through a SIP Message communication. With respect to the state of the RTSP session, the MMIF will then forward these orders to the Media Streamer. The Media Streamer will finally enforce the adaptation actions. The RTSP Session Controller is responsible for particular user session streaming control using the RTSP protocol. The session itself is initialized by the IMS SIP UA. The RTSP Server bloc listens to the RTSP request messages and forwards them to the given RTSP Session Controller, based on the request URL.

- The Media Streamer component is responsible for streaming multimedia resources, using RTP [116]. The Session Controller subcomponent is responsible for effective session streaming creation or removal based on IMS SIP UA request, respectively after successful SIP INVITE or SIP BYE methods.

3.3.2.2 Streaming and adaptation use case elementary tasks

This section describes the use case for the media resource streaming and the adaptation with renegotiation focusing on the message exchanges that take place in each elementary task. We distinguish five elementary tasks carried out by the MSRF in this use case chain as described in Figure3.13:

1. The IMS-SIP compliant session initiation phase, started from the IPTV client towards CSCF with a SDP body and a XML body. The XML body depicted in Figure 3.11 describes the user context. The dynamic part of the UP is inserted in the initial SIP INVITE sent by the UE. The AS/MCMS is able to parse and decode the XML body and adjusts the SDP according to the user environment. The INVITE message then contains the terminal's SDP offer corresponding to its capabilities (video size, codecs, etc.) and the name of the requested media resource. On reception of an INVITE + SDP message, the IMS SIP UA in the MSRF asks the Media Controller for the media resource availability and for a new session creation. The Media Controller instantiates a new session and, based on the resource availability the MMIF generates a SDP answer (new adapted SDP). Then, a response is sent back to the terminal using SIP OK + SDP message. If the terminal accepts the MSRF SDP answer, it acknowledges with a SIP ACK. Finally, the Media Streamer and the related RTSP session

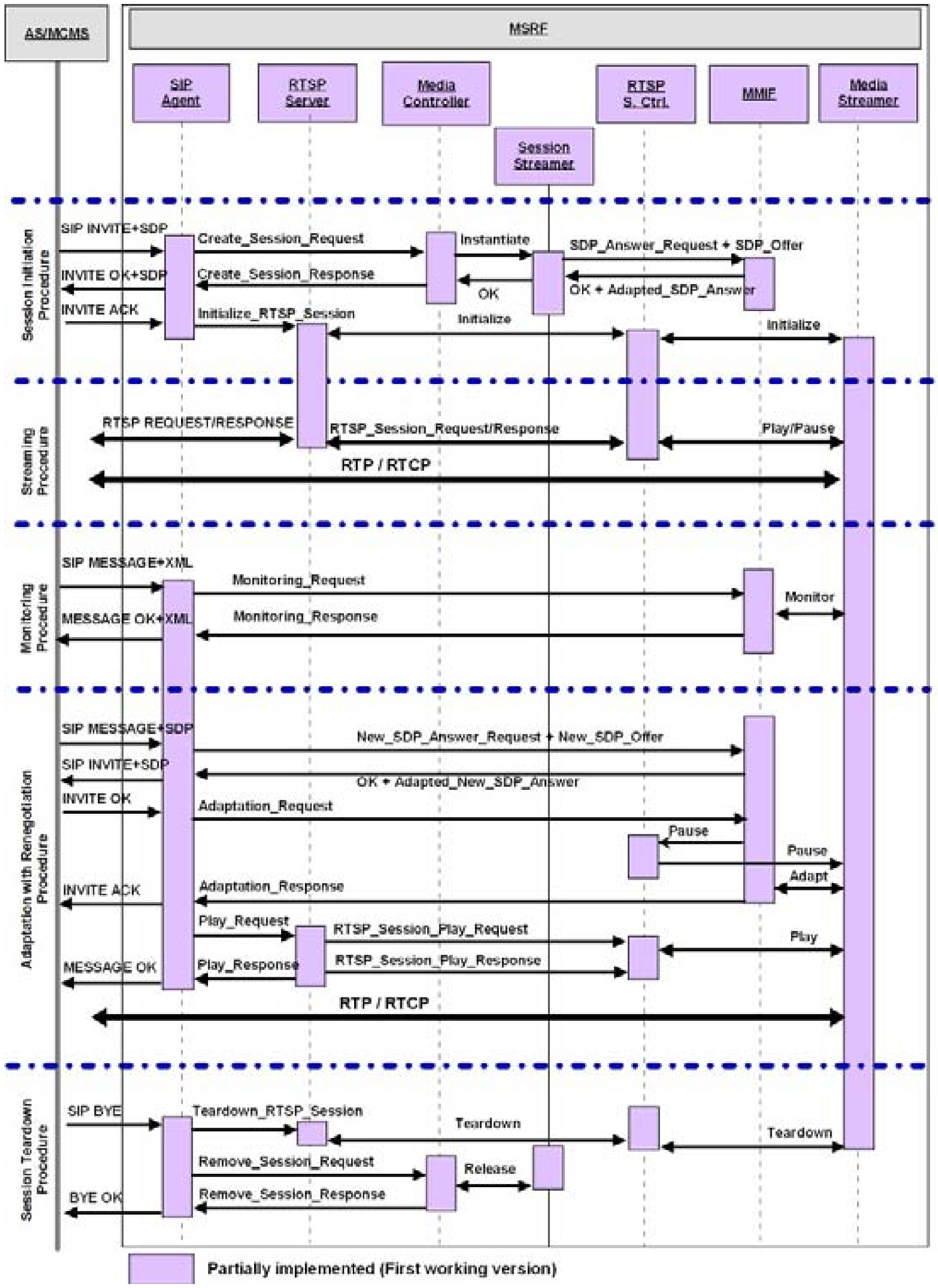


Figure 3.13: MSRF adaptation procedure

at the RTSP Server and the RTSP Session Controller are initialized.

2. The media resource streaming phase, based on RTP/RTCP/RTSP protocols. More specifically, only the PLAY and PAUSE RTSP methods are used to start

and pause the current media resource streaming. In fact, the terminal requests are received by the RTSP Server and forwarded to the RTSP Session Controller in order to play or pause the media resource at the Media Streamer. The responses are reported to the terminal through the reverse path. The media resource is streamed to the terminal using the classic RTP/RTCP protocol.

3. The session monitoring phase, triggered by the MCMS on PQoS degradation detection at the terminal. The MSRF monitoring procedure uses the SIP MESSAGE with a XML body to send the QoS information of the RTCP session back to the MCMS. On monitoring request, the MCMS contacts the monitoring subcomponent of the MMIF. The latter retrieves the monitoring data from the Media Streamer, aggregates them and sends the results back to the MCMS.

4. The session adaptation phase, also triggered using SIP signaling, by the MCMS to enforce the adaption decision based on the previous monitoring task. This adaptation phase is MCMS driven on PQoS degradation and after a successful monitoring procedure. It aims at adapting the media streaming parameters to the new network conditions for better media resource consumption at the terminal. In this task, MCMS sends a new SDP offer to the MMIF based on the MCMS adaptation decision available in SIP MESSAGE + SDP request. The generated SDP offer by the MMIF is sent to the terminal using SIP INVITE + SDP. If the terminal accepts this new SDP offer, it replies with a SIP INVITE OK. Then, the MMIF stops (PAUSES) the streaming process through the RTSP Session Controller and adapts the streaming parameters at the Media Streamer. After a successful adaptation, the IMS SIP UA acknowledges to the terminal and requests the RTSP Session Controller to restart (PLAY) the streaming process. Finally, a SIP MESSAGE OK is sent back to the MCMS. Figure 3.11 depicts end-user's context at its initial state, at the middle of the session and after the adaptation process. We can see that the IPTV service is degraded when FTP traffic is added while the end user is consuming the IPTV service. The packet loss is 10.49%. After adaptation, the service has a reduced throughput compared to the left XML body but the packet loss is 2.44% .

5. The IMS-SIP compliant session teardown phase, requested by the terminal to free resources allocated in the aforementioned phases. This session clean-up procedure is started by the terminal which sends an SIP BYE message through the CSCFs, the IPTV AS and is treated by the MSRF. Then, the MSRF's SIP UA sends a teardown request through the RTSP Server, the RTSP Session Controller, and the Media Streamer. The SIP UA also releases the reserved resources at the Media Controller and Session Streamer. Finally, the SIP BYE OK is reported back to the terminal.

We have defined the complete signaling and MCMS functionalities that trigger adaptation for IPTV service according to the collected PQoS information from UP. In order to optimize service delivery from the initiation phase, PQoS prediction according to video dynamics is studied in the following subsection.

3.3.3 PQoS Prediction according to Video Dynamics

ITU and VQEG proposed subjective test methods involving an audience watching a video sequence under specific conditions. As introduced in chapter 2, objective evaluation methods provide faster quality assessment. Nevertheless, the majority methods require the undistorted video source as a reference entity in the quality evaluation process. Due to this requirement, they are characterized as Full Reference Methods. The recent research is focused also on developing methods that can evaluate the video quality level based on metrics, which use only some extracted structural features from the original signal (Reduced Reference Methods) or do not require any reference video signal (No Reference Methods).

Thus, the aim of the current methods is the quantification of the user experience in terms of satisfaction. However, from a service provider aspect, which is interested to provide its contents free of charge, there is a need in term of more efficient bandwidth management for specifying i)the threshold up to which the user considers the quality of the encoded service as acceptable or below which considers it as unacceptable ii) the maximum perceived quality level that each video content can reach upon encoding and iii) the pattern of the video quality level vs. the encoding bit rate (which will provide to the user the capacity to offer a video at various quality levels). Apart from the various encoding parameters that play significant role in the deduced perceived quality level (e.g. bit rate, spatial and temporal resolution), the dynamics of the content (i.e. spatial and temporal activity of the content) are critical for the final perceptual outcome. Although a lot of research is focused on developing techniques and methods estimating the video quality of a compressed/encoded video signal, the impact of the video spatiotemporal dynamics on the video quality after encoding is not well addressed by the research community and hence explains the motivation of our work.

The main contribution of this section is an experimental approach of the spatiotemporal content dynamics impacting i) the video quality acceptance threshold (i.e. the perceptual quality level below a certain quality which the user considers as unacceptable), ii) the highest achievable video quality level and iii) the pattern of video quality vs. encoding bit rate. More specifically, we present a study on the perceptual quality of the spatiotemporal dynamics of the content in correlation with the encoding bit rate. We consider that the other encoding parameters (e.g. spatial and temporal resolution, encoding scheme, GOP pattern etc.) remain constant. Towards this, we provide results, depicting the actual perceived efficiency for various activity levels. We consider not only the engineering effectiveness such as simple error-based metrics is considered but based also as videos are actually perceived by the human visual system through a respective objective assessment metric. In this section, we uses reference video clips, which are representative of different spatial and temporal activity levels, covering by this way all the range of the spatiotemporal scale. Afterward, for each clip the relative PQoS vs. Bit rate curve for MPEG-4 encoding is drawn, showing how the differentiation in 4 the content affects the deduced video quality.

3.3.3.1 Spatiotemporal Content Plane: A two-dimensional classification of the content dynamics

The content of each video clip may differ substantially depending substantially on its dynamics (i.e. the spatial complexity and/or the temporal activity of the depicted visual signal). The quantification of this diversity is of high interest to the video coding experts, because the spatiotemporal content dynamics of a video signal specify and determine the efficiency of a coding procedure. From the perceptual aspect, the quality of a video sequence is dependent on the spatiotemporal dynamics of the content. More specifically, it is known from the fundamental principles of the video coding theory that action clips with high dynamic content are perceived as degraded in comparison to the sequences with slow-moving clips, subject to identical encoding procedures. Thus the classification of the various video signals according to their spatiotemporal characteristics will provide to the video research community the ability to quantify the perceptual impact of the various content dynamics on the perceptual efficiency of the modern encoding standards. Towards this classification, in[117] it is proposed a spatiotemporal plane, where each video signal (subject to short duration and homogeneous content) is depicted as Cartesian point in the spatiotemporal plane, where the horizontal axis refers to the spatial component of its content dynamics and the vertical axis refers to the temporal ones. The respective plane is depicted on Figure 3.14.

Therefore according to this approach, each video clip can be classified to four categories depending on its content dynamics, namely:

- Low Spatial Activity – Low Temporal Activity (upper left)
- High Spatial Activity – Low Temporal Activity (upper right)
- Low Spatial Activity – High Temporal Activity (lower left)
- High Spatial Activity – High Temporal Activity (lower right)

The accuracy of the proposed spatiotemporal content plane is subject to the duration of the video signal and the homogeneity of the content. For short duration and homogeneous content video clips, the classification is representative and efficient. However, for video clips of longer duration and heterogeneous content, their spatiotemporal classification is becoming difficult.

3.3.3.2 Objective Metrics for the Spatiotemporal Classification of Video Content

We propose to use two discrete metrics, one for the spatial metrics component and one for the temporal one in order to cover the spatiotemporal plane. The averaged frame variance is proposed for the spatial of the video signal. This objective metric permits the quantification of the spatial dynamics of a video signal short in duration and homogeneous. Considering that a frame y is y is composed of N pixels x_i, then

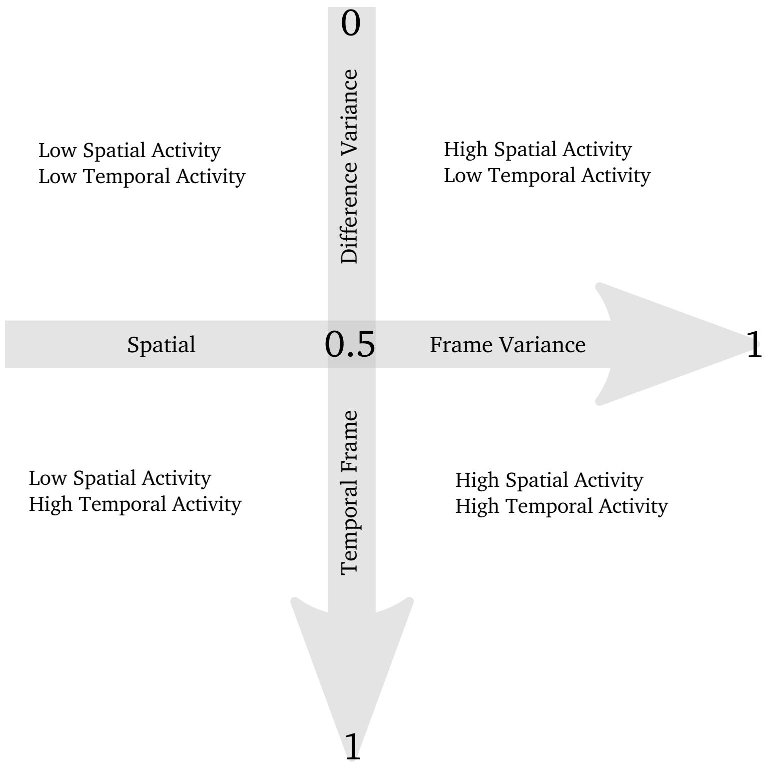


Figure 3.14: The Spatiotemporal grid used for classifying a video sequence according to its content dynamics

the variance of a frame is defined in equation

$$\sigma^2_{frame_y} = \frac{1}{N}\sum_{i=1}^{N}(x_i - \bar{x})^2 \tag{3.1}$$

Derived from equation3.1, equation3.2 presents the averaged frame variance for the whole video duration. K represents the number of frames in the video.

$$\frac{1}{N}\sum_{k=1}^{K}\sigma^2_{frame_y} = \frac{1}{K}\frac{1}{N}\sum_{k=1}^{K}\sum_{i=1}^{N}(x_{k,i} - \bar{x_k})^2 \tag{3.2}$$

The averaged variance of the successive y frame luminance difference is proposed as a metric for the quantification of the temporal dynamics of a video sequence. Considering that a frame contains N pixels x_i and K the number of frames in the video, then the averaged frame difference of the successive frame pairs is defined in equation3.3.

$$\frac{1}{K-1}\sum_{k=2}^{K}\frac{1}{N}\sum_{k=1}^{K}\sum_{i=1}^{N}(x_{k,i} - x_{k-1,i}) \tag{3.3}$$

Therefore, the averaged variance for the overall duration of the test signal is defined in equation 3.4.

$$\frac{1}{K-1}\sum_{k=2}^{K}(\frac{1}{N}\sum_{i=1}^{N}(x_{k,i}-x_{k-1,i})-\frac{1}{K-1}\sum_{k=2}^{K}\frac{1}{N}\sum_{i=1}^{N}(x_{k,i}-x_{k-1,i}))^2 \tag{3.4}$$

The scale in both axes refers to the normalized measurements (considering a scale from 0 up to 1) of the spatial and temporal component, according to the aforementioned metrics. The normalization procedure sets the test signal with the highest spatiotemporal content to the lower right quarter and specifically to the Cartesian (Spatial, Temporal) values (0.75, 0.75). This hypothesis, without any loss of generality, allows to our classification grid the possibility to consider also the signals test that may have higher spatiotemporal content in comparison to the tested ones.

3.3.3.3 Classification of Test Signals to the Spatiotemporal Content Plane

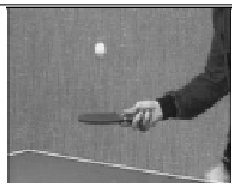
For the needs of this study five short reference sequences are used. These sequences are depicted in Table3.1. Applying the described spatial and temporal metrics on the reference signals of Table 3.1, their classification on the proposed spatiotemporal grid is depicted on Figure3.15.

According to Figure, it can be observed that the spatiotemporal dynamics of the selected reference signals are distributed to all the four quarters of the spatiotemporal grid, indicating their diverse nature of the content dynamics. Moreover, the validity of the proposed metrics is certified by these experimental results, showing that they provide adequate differentiation among the dynamics of the signals under test.

Based on the experimental results of Figure 3.15 and Table 3.1, it can be observed that the selected video signals are representatives of the whole range of the spatiotemporal activity range of the content dynamics and the spatiotemporal content plane.

In the next section, we discuss the spatiotemporal content dynamics impact on i) the video quality acceptance threshold (i.e. the perceptual quality level below which the user considers that an encoded video is of unacceptable quality), ii) the highest achievable video quality level and iii) the pattern of video quality vs. encoding bit rate.

Table 3.1: The five reference test signals

Suzie	
Cactus	
Flower Garden	
Table Tennis	
Mobile&Calendar	

3.3.3.4 Spatiotemporal Activity and Video Quality

This section focuses on the impact of the spatiotemporal activity of the content on the video quality. The encoding bit-rate needs to be adjusted according to the spatiotemporal impact in order to provide a satisfying video quality to the end-user. It must be noted that the used sequences in these experiments are reference signals with limited duration and therefore with practically homogeneous content (i.e. constant spatial and temporal activity level). The study with longer videos is out of the scope in this work. Each test video clip of 3.1, is encoded from its original uncompressed format to ISO MPEG-4 Visual Simple Profile MPEG format, at different constant bit rates (spanning a range from 50kbps to 1.5Mbps for Common Interface Format (CIF) with key-frame period equal to 100 frames in both cases). For each corresponding bit-rate, a different ISO MPEG-4 compliant file is created. The frame rate is set at 25 frames per second for the whole encoding process.

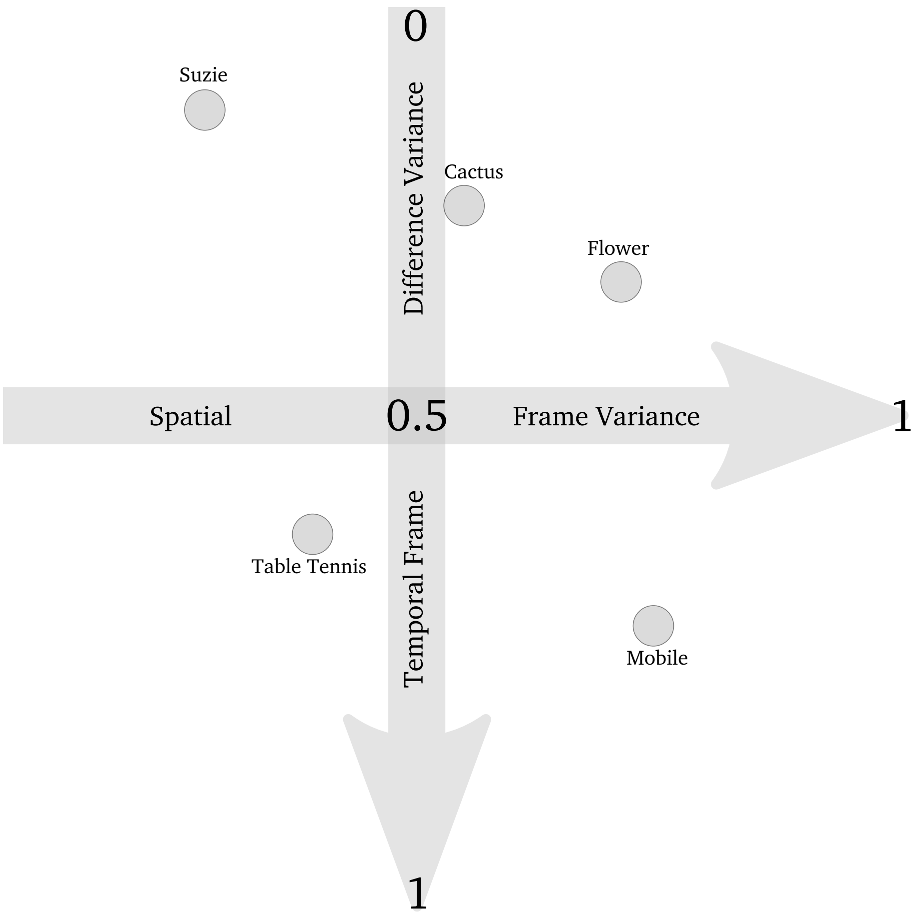


Figure 3.15: The Spatiotemporal classification of the test signals

Therefore, in the following section, we present our test-bed to test the impact of video dynamics on the PQoS.

3.4 Test-Bed and Results

In order to validate our novel user-centric IMS platform with the seamless integration of the UP in the MN, the MCMS and the MSRF in the IMS core network, we simplified the architecture of ADAMANTIUM project and configured the test-bed as depicted in Figure 3.16. Table 3.4 gathers the testbed information. We use the UOP[8] PQoS model at the MN's side. We first present the results of the PQoS prediction defining the quality threshold and the adequate Send Bit Rate (SBR) for MPEG-4 videos of our test-bed. Then subjective tests show the benefit of using UPM and dynamic adaptation based on PQoS estimation. Finally, we present the response time results of the overall adaptation process depending if the UPM and

PQoS mechanism for dynamic adaptation are used or not.

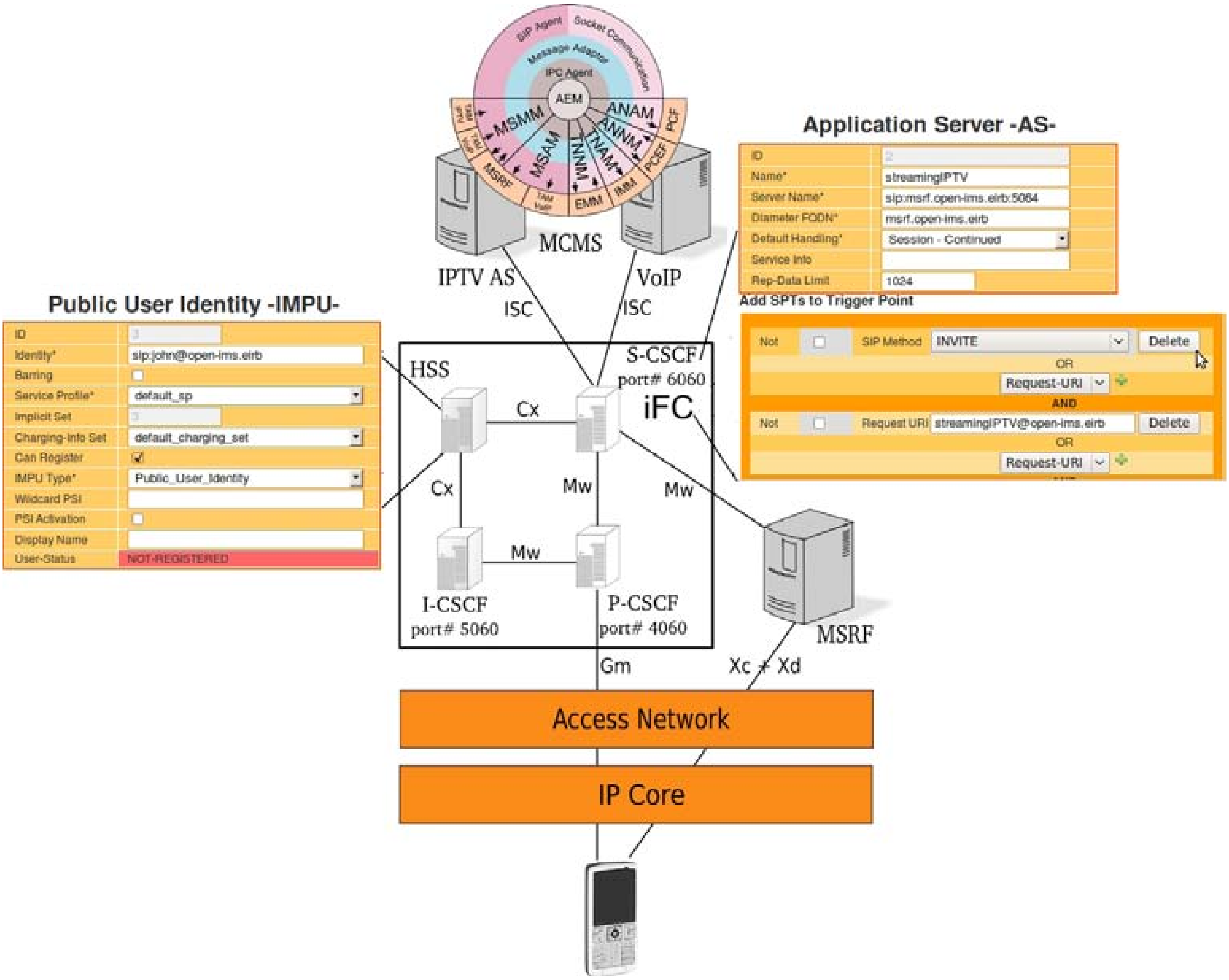


Figure 3.16: Core Network Configuration

3.4.1 Impact of content dynamics

Each ISO MPEG-4 video clip is used as input in a no-reference objective quality measurement tool [118]. From the measurement resulting quality per frame measurements, the average quality for the whole clip is calculated.

Table 3.2: System configuration

Component	Hardware	Software	Access network
TV HD connected to a desktop	Screen : Dell P2310H (1920x1080) Desktop: Dell Optiplex 745	Ubuntu 9.04, Python, Twisted, VLC	Wired
Notebook	Dell XPS 1530m (1440x900)	Ubuntu 9.04, Python, Twisted, VLC	Wi-Fi
Netbook	Acer Eeepc (800x600)	Ubuntu 9.04, Python, Twisted, VLC	Wi-Fi
CSCFs	Dell Optiplex 745	Ubuntu 9.04, OpenIMSCore	-
AS/MCMS	Dell Optiplex 745	Ubuntu 9.04, Twisted,	-
MSRF	Dell Optiplex 745	Ubuntu 9.04, Twisted, VLC	-

3.4.1.1 Video quality vs bit-rate

This experimental procedure is repeated for each tested video clip and the respective curves representing the video quality vs. encoding bit rate is depicted in Figure 3.17. The curves are following a general exponential pattern and present a significant leeway between the various spatiotemporal dynamics.

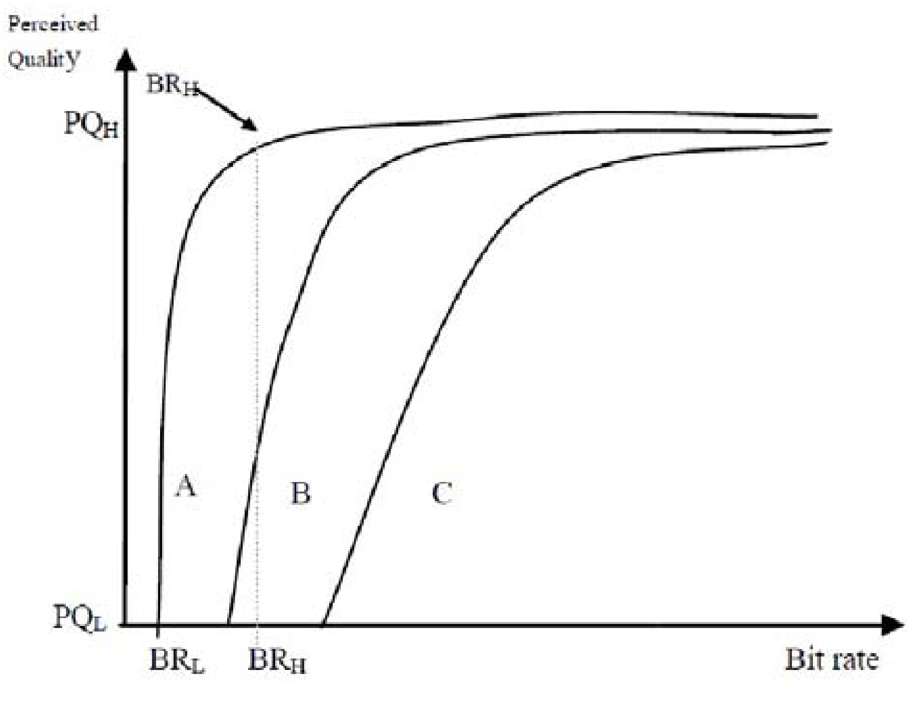


Figure 3.17: Impact of dynamics on the video quality vs bit-rate curves

More specifically, it can be observed that curve A represents video clip with low temporal and spatial dynamics, i.e. video content with “poor” movements and low picture complexity such as a talk show scene. Curve C represents video clip with high dynamics, such as a football match. Curve B represents an intermediate case. Practically, it can be observed that in low bitrates curve A reaches a higher perceptual level compared to aches curve B depicting a sequence with higher spatiotemporal content. On the other hand, the curve C) requires higher bit rate in order to reach a satisfactory PQoS level. Nevertheless, curve(C) reaches its maximum PQoS value more smoothly than in the low activity case.

Moreover, each curve -and therefore each video clip- can be characterized by: (a) a low bit rate (BRL), which corresponds to the lower value of the accepted PQoS (PQL) by the audience, (b) the high bit rate (BRH), which corresponds to the minimum value of the bit rate for which the PQoS reaches its maximum Perceptual Quality Level (PQH) value (see BRH for curve (A) in figure 3) and (c) the mean) inclination of the curve, which can be defined as : $ME = (PQH–PQL)/(BRH–BRL)$. From the curves of Figure3.14, it can be deduced that video clips with low dynamics have lower BRL and higher ME than clips with high dynamics.

Following the general pattern, the respective experimental data for the reference signals that have been tested be are depicted in Figure 3.18. As it can be observed, the impact of the spatiotemporal activity on the content is depicted very clear. It also shows two more important outcomes: i) For video signals with low spatiotemporal activity, a saturation point appears, above which the perceptual enhancement is negligible even for very high encoding bit rates. ii) As the spatiotemporal activity of the content becomes higher, the respective perceptual saturation point (i.e. the highest perceptual quality level) becomes lower, which practically means that video of high dynamics never reach a very high perceptual level. Based on these observations, the next sub section examines in more details the impact of the content dynamics on the perceptual saturation point (i.e. the highest perceptual quality level).

3.4.1.2 Highest perceptual quality level and acceptance threshold

Focusing more on the impact of the spatiotemporal content dynamics on the perceptual saturation point (i.e. the highest perceptual quality level that each video signal can achieve), it can be observed directly from both Figures 3.17 and 3.18 that video signals with relatively low spatiotemporal content achieve higher perceptual levels than video signals that contain content of high dynamics. In this framework,

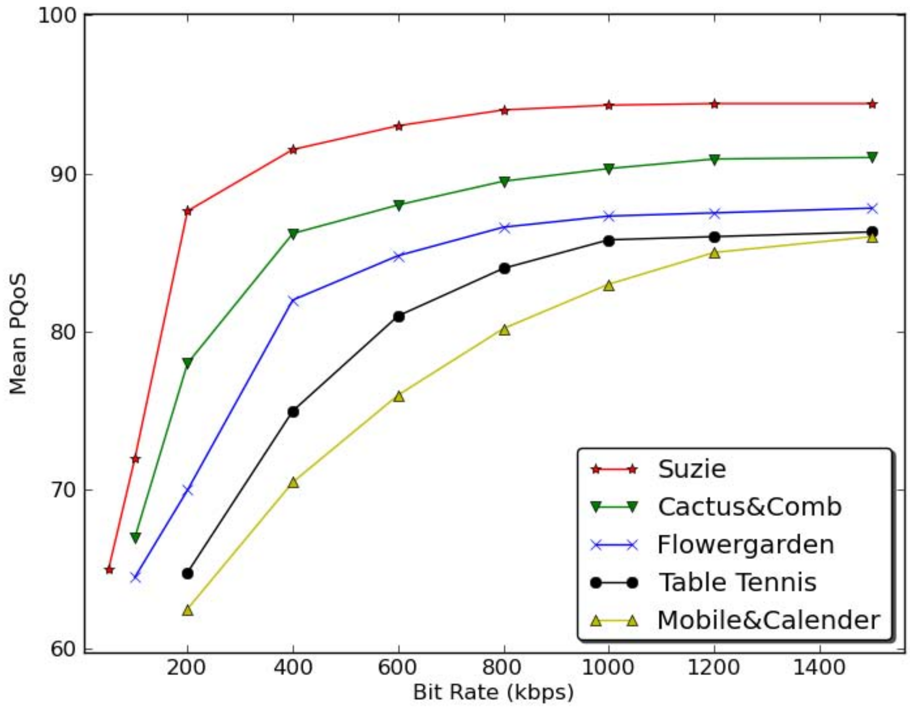


Figure 3.18: The Video Quality vs. Bit Rate curves

Figure 3.20 depicts the experimental results for the test signals with the highest PQH for both CIF and Quarter Common Interface Format (QCIF) spatial resolution. It can be observed that for both CIF and QCIF spatial resolution, the impact of the spatiotemporal activity is significant making especially the signals of low content dynamics less demanding in terms of encoding bit-rate for a certain perceived threshold. Figure 3.19 examines the impact of the spatiotemporal activity of the content on the perceptual acceptance threshold for various test signals located in MSRF. The lowest acceptable perceptual level is fixed to 3.5 in the MOS scale. Based on these experimental results, it is shown that for both CIF and QCIF spatial resolution need higher bit rate in order to achieve the perceptual acceptance threshold when the spatiotemporal activity becomes more complex. The main objective is to choose the right bit-rate for adapted video streams. More specifically in the case of CIF, the demand in terms of bit rate becomes higher than for the case of QCIF. We use these results in our test-bed in order to optimize the video bitrate sent by the MSRF.

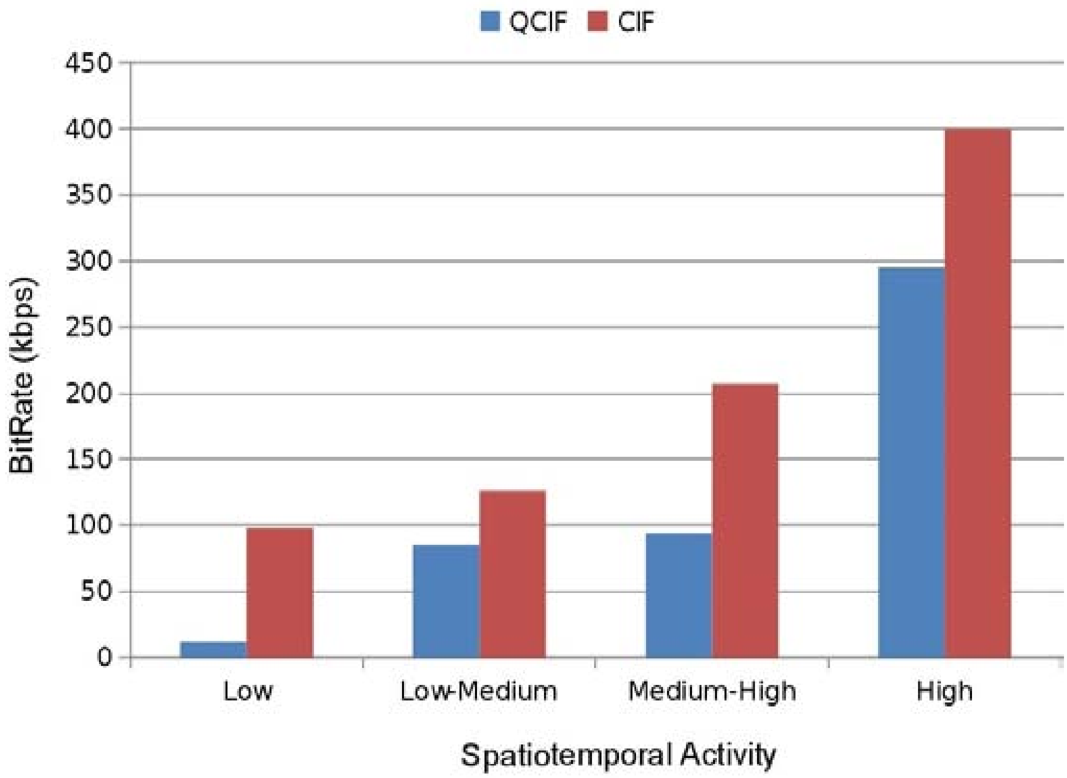


Figure 3.19: Impact of dynamics on the acceptance PQoS threshold

Table 3.3: Video resolution

Abbreviations	Resolution (width x height pixels)
Quarter Common Interface Format (QCIF)	176x144
Common Interface Format (CIF)	352x288
Video Graphic Array (VGA)	640x480
4xCIF (4CIF)	704x576
Super Video Graphic Array (SVGA)	800x600
9xCIF (9CIF)	1056x864
HDTV (720p)	1280x720
16xCIF (16CIF)	1408x1152
HDTV (1080p)	1920x1080

3.4.2 QoE evaluation for adaptive IPTV service

This section presents the scenario used to validate the benefits of the UP management, the MCMS and MSRF modules and the adaptation process according to the video spatiotemporal activity also called content dynamics. The test-bed is

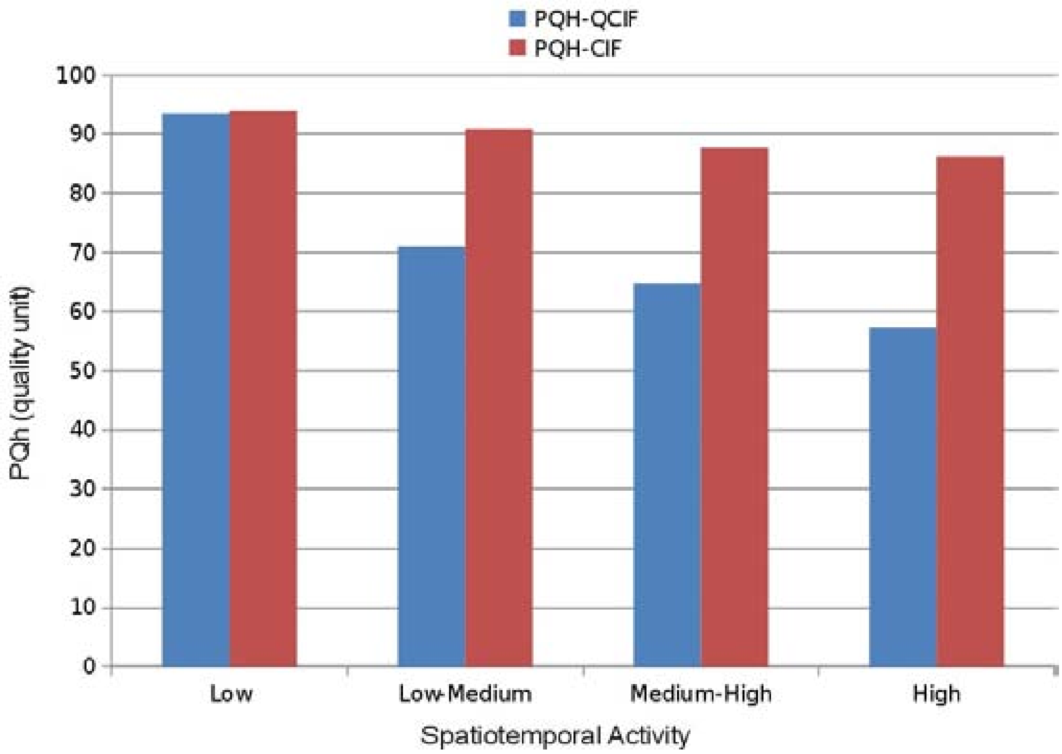


Figure 3.20: Impact of dynamics on the PQh video quality

composed of a MN, an OpenIMS Core, an AS/MCMS and a MSRF as depicted in Figure 3.16. The purpose of this test-bed is to estimate the QoE based on subjective tests in a live system. We use MOS estimations following the SSCQE method as described in Table 3.4. 30 people estimated an HD IPTV (1920x1080) stream quality on an HDTV ready, a notebook and a netbook. We could not test the benefit of the PQoS aware IMS platform on a mobile handset as no open source phone supports VLC [119].

Figure 3.21 depicts the average MOS estimation for the different MNs. We distinguish 2 types of adaptation, the UPM adaptation starting when the end-user initiates the IPTV service and the PQoS adaptation when the PQoS alarm is triggered and the MSRF performs dynamic adaptation. We have four use-cases depending on whether the UP management and the PQoS mechanism are activated or not. The HD TV Desktop has no problem decoding an HD Video Stream and does not suffer from network impairment as it possesses a wired access. The addition of UPM and PQoS mechanism do not improve the MOS values. The Dell XPS 1530 notebook benefits from using PQoS mechanism as it suffers from network impairments.

Table 3.4: System configuration

Aspect	Short tests	Long tests	References
Test method	ACR/SS	SSCQE	
Scene characteristics	Duration: < 20s	Duration : 2min approx.	
Replications	Implicit	Implicit	
Presentation order	Random	Random	
Viewers	30 non-expert people	30 non-expert people	
Viewing Conditions	Device: Dell XPS 1530m, Acer Eeeepc Place : Laboratory	Device: Dell XPS 1530m, Acer Eeeepc Place : Laboratory	
Training session	Trial videos presented	Trial videos presented	
Evaluation	MOS scale	continuous MOS scale	
Test general structure	Initial phase: Instruction&training	Initial phase: Instruction&training	

Finally, the Netbook suffers mainly from its hardware limitations. Indeed, the Netbook is not able to decode an HD IPTV stream. The UP management notifies the Netbook hardware limitation to the MSRF from the initiation of the media stream. Thus, the user is consuming an adapted stream from the start of the video content.

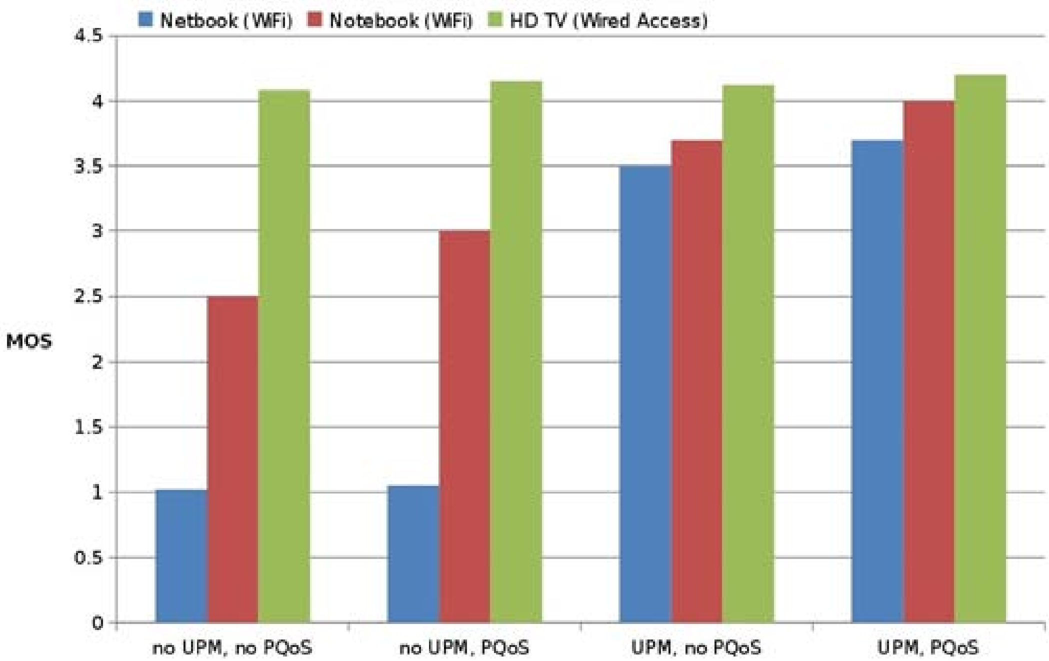


Figure 3.21: MOS estimation with or without UPM and PQoS adaptations

Figure 3.22 presents an overview of the response time evaluation of the overall adaptation process. The measured time starts from the service initiation for UPM adaptation process and the first alarm sent in the PQoS adaptation to the adaptation

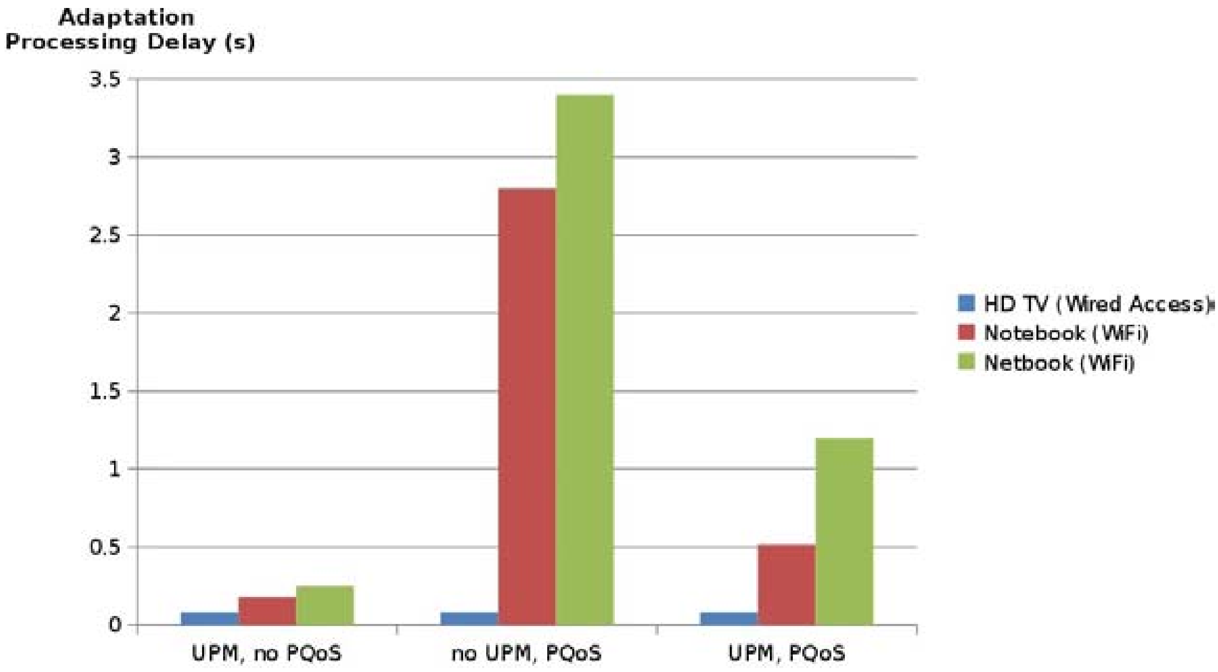


Figure 3.22: Adaptation delay depending with or without UPM and PQoS adaptations

enforcement. We use the same four use-cases with and without UP and PQoS management. We consider that the MSRF is always able to deliver the requested IPTV stream. UPM adaptation processing delay corresponds to the time elapsed for the IPTV service to initialize with customized parameters. PQoS adaptation processing delay represents the time elapsed to adapt the content to the terminal and the access network condition of the end user. PQoS adaptation without UPM needs more time to deliver an acceptable IPTV stream as it requires a monitoring phase at the UE and MSRF sides. The HD TV Desktop does not need to trigger PQoS adaptation as the HD IPTV stream suits to its environment. As far as the Notebook and the Netbook are concerned, the time of UPM without PQoS adaptation is the sum of packets transmission delay and video transcoding/transrating. When the PQoS adaptation is enabled without UPM, PQoS adaptation mechanism requires few seconds to enforce the adaptation due to the monitoring and end-to-end session renegotiation delay (transcoding). However, in our test-bed configuration, when a PQoS adaptation is triggered with the UPM enabled, no end-to-end media session renegotiation is required: only transrating and/or transsizing is processed. Thanks to these enhanced adaptation mechanism, end user satisfaction is guaranteed with a non significant overhead. For a 10 min video streaming with an average bitrate of

800 KB/s (considering that the PQoS monitoring SIP messages is sent every second), the overhead of the adaptation mechanism is 0,2% of the total traffic. Furthermore, adaptation is triggered by the UE only when the PQoS goes below an adjusted threshold. So the 0,2% overhead is thought to be the worse case in these use-cases.

3.5 Conclusion

This chapter introduces context-awareness in IMS that aim to provide end-users with adapted multimedia services according to their environment and preferences. We participated on the design and development of the ADAMANTIUM architecture by i) defining all signaling between the MN and the IMS core, ii) inserting the UPM inside the MN architecture, iii) defining and implementing MCMS modules enforcing the adaptation process in the MSRF and iv) implementing a live test bed with adaptive IPTV service.

The complete signaling process between the MN and IMS core network and the dialogs between MCMS and MSRF are proposed and implemented in the test-bed. The UPM mechanism in the MN enables the initiation of adapted IPTV media streams according to the static information of the user. When the user is consuming the service and experience low PQoS estimation, warning or red alarms are sent towards the MCMS thanks to the dynamic part of the UP. MCMS receives PQoS information and triggers an adaption request within the MSRF if this is necessary. The combination of UPM and PQoS mechanisms give the user better satisfaction of ubiquitous services for devices with high resource limitation. PQoS estimation on the spatiotemporal nature of the video optimizes the used bit-rate with a given PQoS threshold.

When the observed quality at the end-user side does not be improve even after adaptation process, UE has the possibility to switch on different network interface and proceed to a vertical mobility.

Chapter 4

Seamless handovers in IMS environment based on End-User's context

4.1 Introduction

IMS as depicted in Figure 2.1 aims to unify service interfaces through a common IP-based core network hiding access technology characteristics. IMS enables the fixed/mobile network convergence. This convergence remove the distinctions between fixed and mobile networks. End user benefits of seamless services using the WLAN of their home or office combined with a fixed broadband access. Corporate networks have been introduced in Chapter 2. The evolution of this business place into IP technologies enhances the way of communication. Business-oriented and personalized services can be delivered to employees. More bandwidth is usually available within corporate buildings using Local Area Network (LAN) or WLAN. UMTS and LTE wireless technologies have good coverage outside the buildings but signal strength drops drastically within corporate premises. Therefore IMS can leverage broadband accesses of houses and companies in order to provide session continuity and better perceived quality for media services.

Nevertheless, mobility between two different wireless network technologies also called **vertical mobility** raises technical issues. Chapter 2 of this thesis compares existing vertical mobility solutions for VoIP and are summarized in Table 2.5. We are focusing on mobility for VoIP services as they have strong QoS requirements in order to reach correct user satisfaction (i.e. MOS > 3.5). VoIP is a conversational service highly sensitive to delay variation. For IPTV services, network impairments caused by handover process can be solved by increasing the size of jitter buffer.

In this chapter, we introduce our vertical mobility solution for VoIP services based on SIP mobility. Our proposal is transparent to IMS architecture. Only two slight modifications are necessary in P-CSCF and MN for mobility support. We use our UP management developed in the previous chapter in order to inform the core network with our new environment capabilities. If the targeted access network will not improve the service PQoS, the core network rejects the mobility request. We implemented our mobility solution on a live test-bed with 3G and WLAN connections.

4.2 Mobility mechanism in NGN

IMS is defined to control media session over a wide range of access network and domains. Only handovers requiring a change of IP address on dual-homed devices will be considered in our solution. As discussed in Chapter 2, xMIPv4/xMIPv6 and mSCTP lack of integration in IMS systems. mSCTP is not yet standardized and implemented in IMS. xMIPv4/xMIPv6 support mobility funtionalities in LTE architecture but hide IP modification from SIP layer of IMS. Existing SIP mobility exists but involves the CN in the signaling path. If the CN is located far from the Home Network of the user, handover process may be delayed and satisfaction of the user decreased. IMS standard defines the MMSC-AS in order to overcome this issue. Nevertheless, IMS does not provide efficient media switching in the case of vertical mobility. Indeed, the handover from one access network to another may suffer from delay due to the different network condition.

The proposed mobility solution faces session continuity issues in IMS networks with control of the corresponding media flow. Our SIP-based solution merges intra-system (within Home Public Land Mobile Network (HPLMN) or domain A) and inter-system (between HPLMN/Domain A and Visited Public Land Mobile Network (VPLMN)/Domain B) handovers. Our concept is to proceed the media switching as close from the MN as possible. As P/I-CSCF are the first entry point in IMS, we targeted these network elements to handle media switching. Mobility Awareness can be introduced in P-CSCF/I-CSCFs. Mobility Awareness in P/I-CSCFs (MA-P/I-CSCF) comprises mobility detection and automatic RTP duplication and switching. Packet duplication is useful during the first seconds of the new path establishment. Some packets may be dropped during the new path connection.

These functionalities require two MN modifications that will leverage the P/I-CSCF evolution. During the SIP re-establishment on the new path, MN adds a Route Header in the SIP Re-INVITE message targeting to the current P/I-CSCF involved. When handover request is accepted by the MMSC-AS server, the MA-P/I-CSCF enforces first the packet duplication on 'old' and new path and then switches definitely the media to the 'new' path. In order to leverage the packet duplication, a new jitter buffer algorithm is introduced in the MN to smoothen the RTP stream switching.

4.2.1 Architecture

As discussed in Chapter 2, our solution requires to handle handovers in roaming and non roaming use-cases. Figures 4.1, 4.2, 4.3 and 4.4 show the corresponding architectures. In the non roaming use-cases, the Wi-Fi access gateway could be connected to the P-GW through the S2c interface if the operator has a LTE access network as depicted in figure 4.1 or through a separate P-CSCF 2 if the operator does not fully comply to LTE or provides only 3G access as shown in figure 4.2. Figure 4.4 depicts the roaming use case where a MN registered in the IMS domain "A" switches from its LTE or 3G connection towards a Wi-Fi hotspot managed by another IMS domain "B". Each IMS domain uses I-CSCF to protect and hide its IMS core network towards other operators. In figure 4.1 and 4.3, P-CSCF is assumed to be MA-PCSCFs.

In figure 4.2, both P-CSCF1 and P-CSCF2 are MA-PCSCFs as MN is assumed to support mobility in both direction. Finally in Figure 4.4, I-CSCF A is assumed to be Mobility-Aware I-CSCF (MA-ICSCF) as the latter is linked to the CN. It is also assumed that Mobility Awareness functionalities are enforced in either a P-CSCF or a I-CSCF located in the MN's HPLMN.

Our solution depicted by one scenario in Figure 4.5 supports all these 4 showcases but the following section is focusing on one scenario.

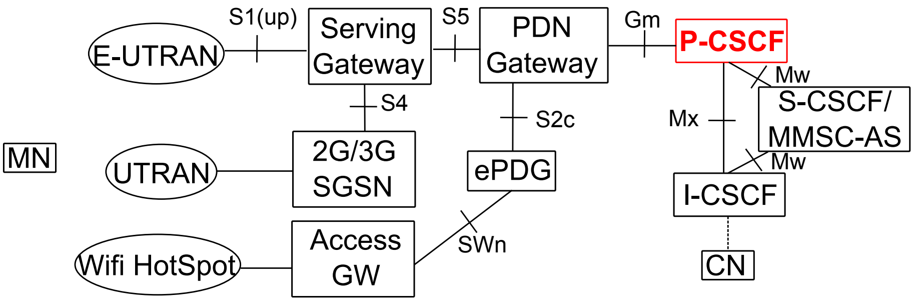


Figure 4.1: Architecture for non roaming-LTE

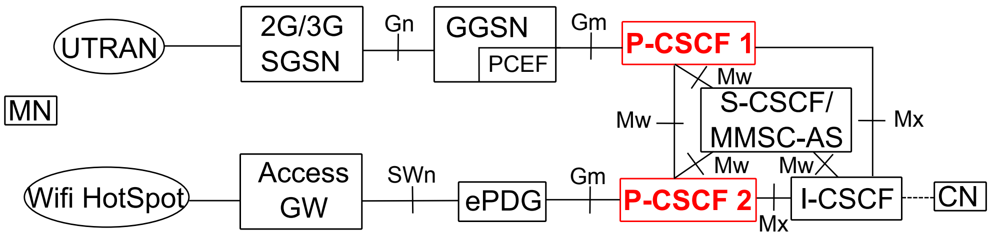


Figure 4.2: Architecture for non roaming-IMS only

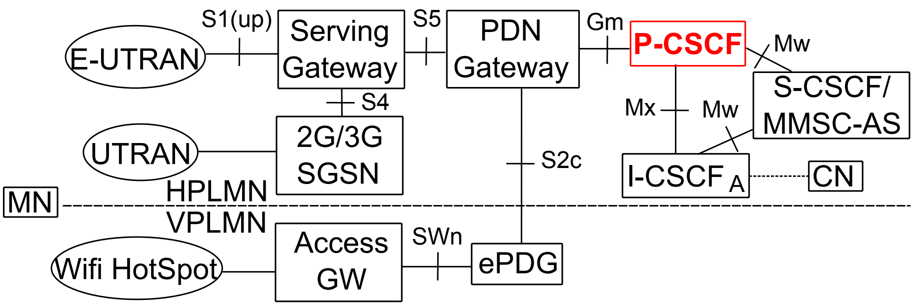


Figure 4.3: Architecture for roaming-LTE

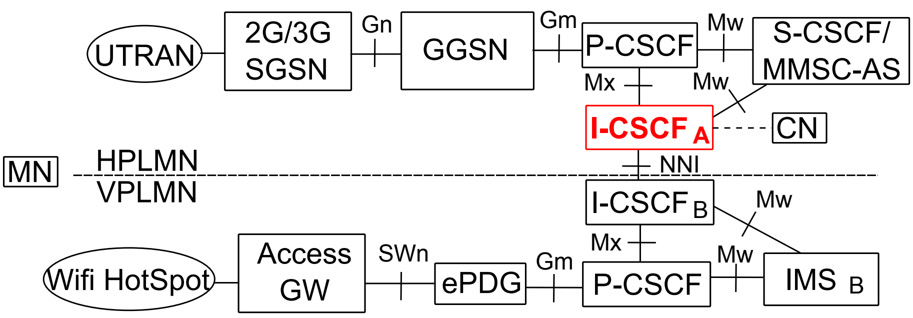


Figure 4.4: Architecture for roaming-IMS

4.2.2 Mobility Scenario

In order to validate our mobility solution, we consider that the MN is roaming from WLAN towards UMTS access network. The MN is considered in communication

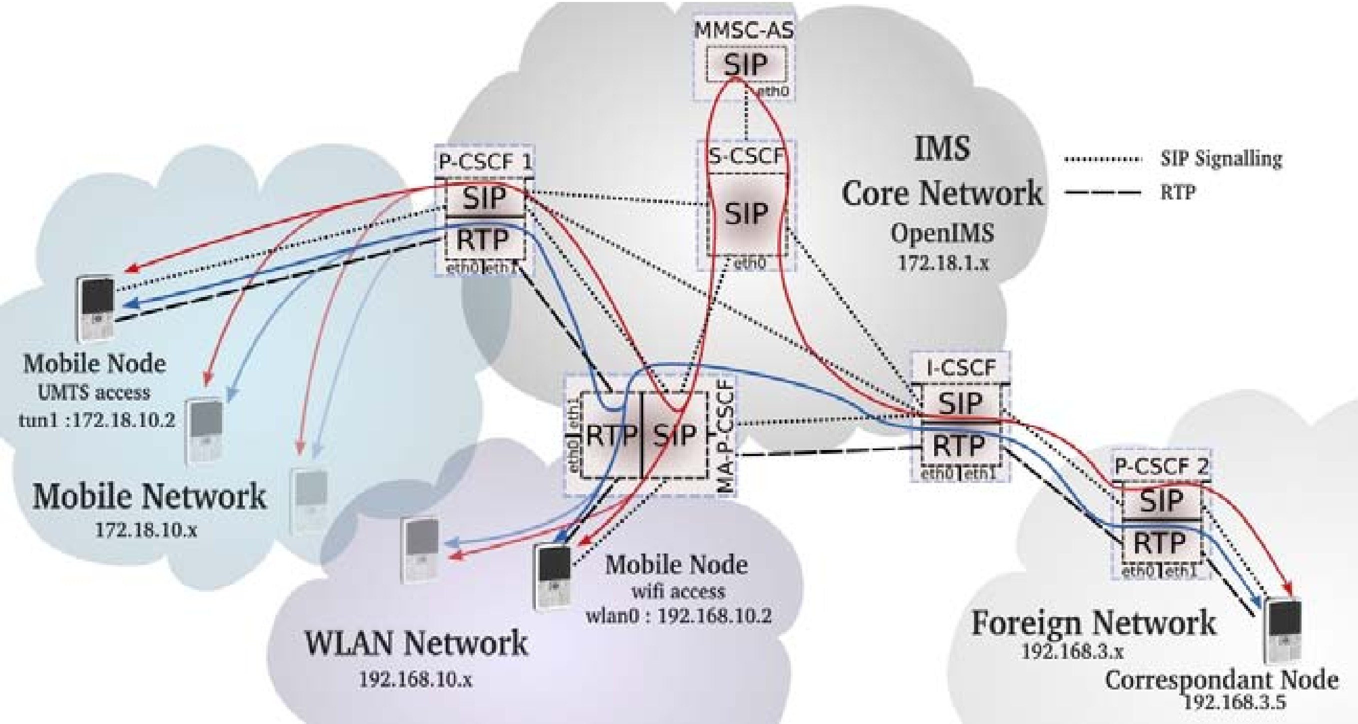


Figure 4.5: Solution Architecture

with a CN located in a foreign network using its Wi-Fi access. Then the MN is moving out of the coverage of WLAN network and registers its public UMTS interface at IMS. Figure 4.5 describes the roaming of the MN from its WLAN interface towards its public UMTS access. The handover from WLAN to UMTS network is triggered. The MN sends a Re-INVITE SIP message through its UMTS interface towards the P-CSCF 1 which forwards it to MMSC-AS. The latter decides whether to accept or refuse the handover. The complete signaling path is shown in red color in the Figure 4.5 whereas the media path is represented by the blue color. Please notify that IMS core (i.e. S-CSCF and MMSC-AS) are forwarding only SIP signaling.

Our solution is based on the integration of mobility-awareness within the P-CSCF therefore named MA-PCSCF. The following subsection introduces the new Mobility-Aware P/I-CSCFs and the modification of the multi-homed MN.

4.2.3 Mobility-Aware P/I-CSCF

P/I-CSCFs are composed of a SIP proxy and a RTP proxy as depicted in Figure 4.6. One or several SIP proxies control one RTP proxy through a socket communication.

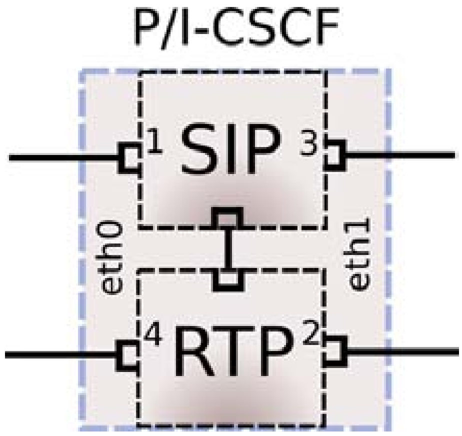


Figure 4.6: Proxy and Interrogating-CSCF architecture

Whenever SIP messages arrive on the SIP proxy (slot 1), the latter dialogs with the RTP proxy (update request in Figure 4.7) in order to open a media port for incoming media flow (slot 2). When the session is granted, a SIP 200 OK response is coming back (slot 3), the SIP proxy sends a "LookUp" request to the RTP proxy which opens the last port (slot 4). If the media session initiation is not granted by MMSC-AS, then RTP proxy deletes the opened port (slot 2).

To achieve mobility enforcement, we empowered the SIP and RTP proxies with two new functionalities. The first one is the mobility detection. The SIP proxy has to detect if the incoming INVITE request corresponds to a new session establishment or a Re-INVITE message (Update of the current session with new SDP/XML bodies). The second functionality is the packet duplication. The RTP proxy has to duplicate incoming media packets on the both paths and ensure the media switching when the handover process is acknowledge. The complete handover signaling of the MN roaming from WLAN network towards 3G network is described in Figure 4.7.

The handover is triggered by the MN sending a SIP Re-INVITE request to P-CSCF 1 which forwards it towards the MA-PCSCF. Figure 4.8 depicts the architecture of the MA-PCSCF. Each time an INVITE Request reaches the Mobility-Aware P/I-CSCF, the latter contacts the RTP Proxy. The latter compares Call-ID and SIP-URI of Caller/Callee for all media streams that the proxy is in charge with. If the 2-tuple Call-ID, Caller/Callee is matching with an existing media stream then the MA-P/I-CSCF switches to mobility mode.

Then the SIP Re-INVITE is forwarded to the S-CSCF and the MMSC-AS. The

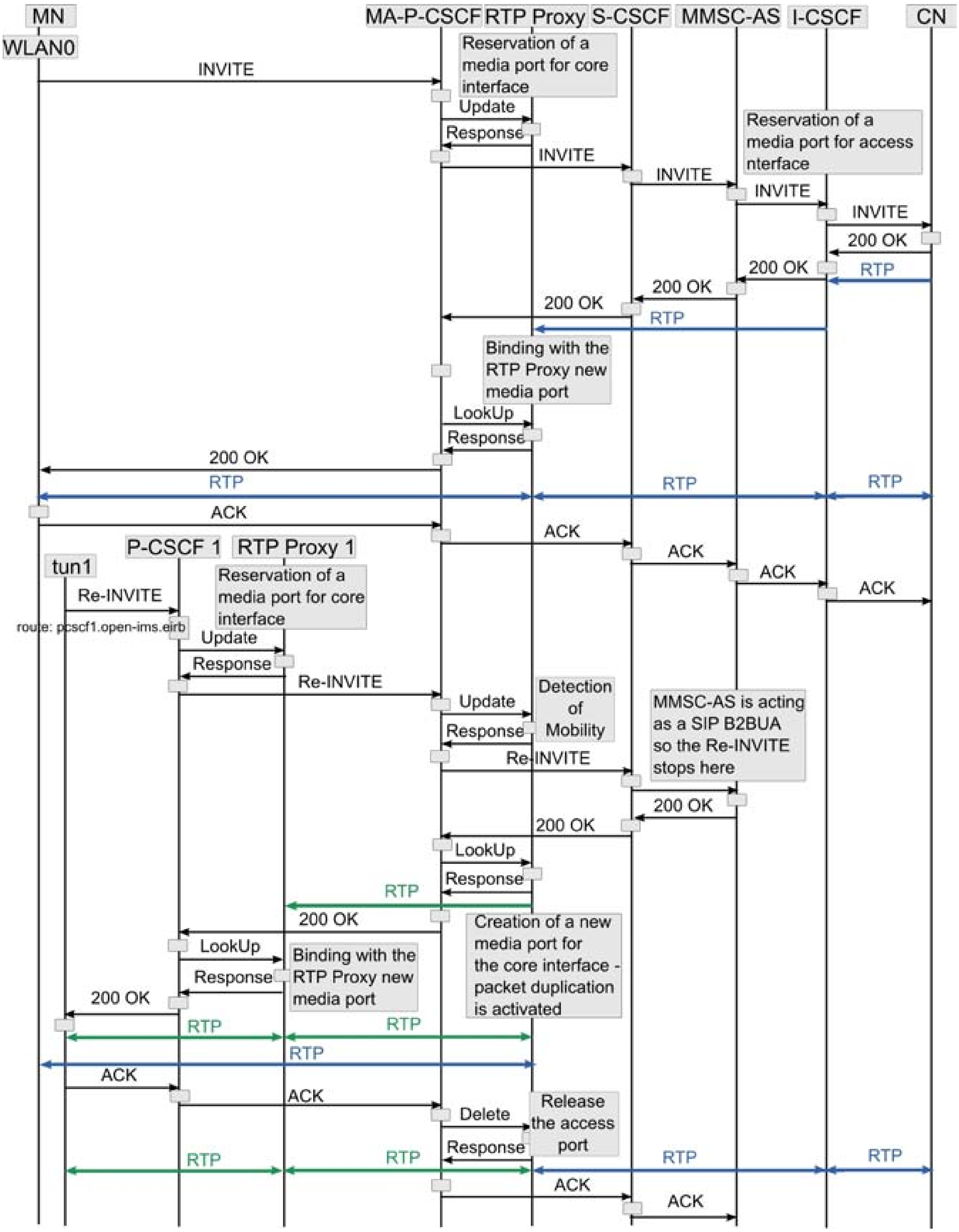


Figure 4.7: Handover Signaling

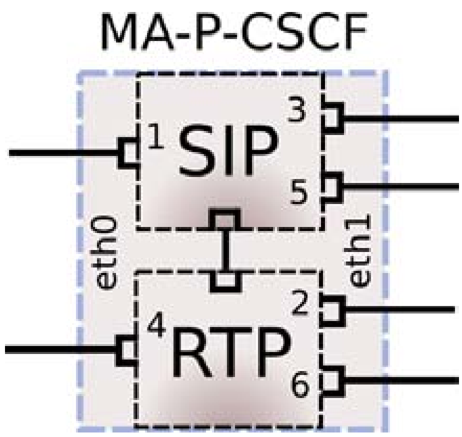


Figure 4.8: MA-P-CSCF architecture

latter accepts the handover request by responding with a 200 OK which is forwarded back to the MN. When the MA-P/I-CSCF is in mobility mode and receives a 200 OK from the MMSC-AS, the RTP proxy opens the new port (slot 6) on the corresponding interfaces and starts to duplicate RTP packets on both link (i.e. WLAN and Third Generation (3G) paths) at the same time. Due to network characteristics disparity, RTP packets reach the MN at different time and sometimes with burst of 3-4 packets. The MA-PCSCF forwards definitely packets to the new path and releases unused resources (slot 4) when the MN sends the SIP ACK.

This SIP Re-INVITE method is considered as a session update and not as a separate session. Thus, there is no need to terminate the *old* session by sending a SIP BYE.

4.2.4 Centralized RTP proxy

Our mobility solution based on new functionalities implemented within P/I-CSCFs has several advantages. Among them, the possibility of using a centralized RTP proxy within the core network as depicted in Figure 4.9 reduces the number of hopes in the media delivery path and thus reduces the media switching latency during the handover process. The SIP proxies interface the media server through socket communication thanks to a low latency dedicated network. This solution does not imply any integration effort as the dialog between the SIP and RTP proxies is already based on socket communication.

In order to leverage the mobility functionalities implemented in the MA-PCSCF,

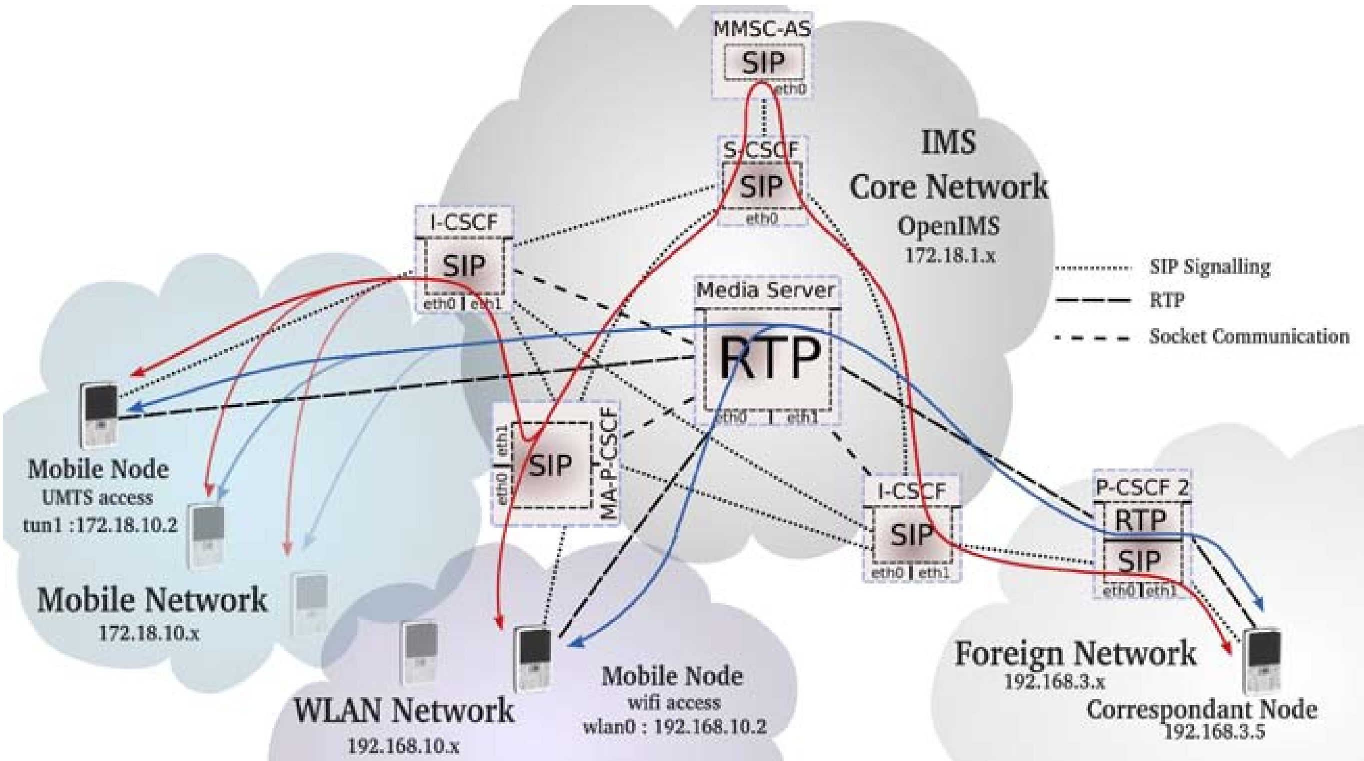


Figure 4.9: Solution Architecture with centralized RTP Media Server

the MN also needs slight modifications to enable seamless mobility.

4.3 Media Handling at Multi-homed Devices

MN has two network interfaces : one Wi-Fi interface and one UMTS interface. No existing solution is focusing on how the media is scheduled from one interface to the other. We then propose a fast media switching at the core network side. To achieve that, we suggest to force the routing through the old P-CSCF. For that, we reuse the first SIP INVITE message (with the same Call-ID) sent to IMS and insert the route header of the new P-CSCF (P-CSCF1). The INVITE message arriving at P-CSCF 1 will then be forwarded to MA-PCSCF as depicted in Figures 4.5 and 4.7. Then, the SIP INVITE follows the same path as the first INVITE (towards S-CSCF and MMSC-AS).

4.3.1 Route header in the SIP Re-INVITE

Wi-Fi signal strength can vary rapidly when the user is leaving the WLAN coverage. Therefore, our vertical mobility solution is mobile controlled. The MN triggers the handover depending on the WLAN degradation. Therefore, the SIP Re-INVITE is sent directly through the new P/I-CSCF. Nevertheless, in order to perform fast switching, our solution requires the old P/I-CSCF in the path of signaling. SIP Re-INVITE request keeps the same route header as the SIP INVITE but add the new entry at the top position as shown in red in Figure 4.10. This mechanism The added route forces the request to be forwarded to the 'new' P/I-CSCF and not the old path as before.

```
INVITE sip:test@open-ims.eirb:5060;ob SIP/2.0
Route: <sip:pcscf1.open-ims.eirb:4060;lr>
Route: <sip:mo@pcscf.open-ims.eirb:4060;lr>
Route: <sip:mo@scscf.open-ims.eirb:6060;lr>
Route: <sip:mmsc.open-ims.eirb:5060;lr>
Route: <sip:mo@scscf.open-ims.eirb:6060;lr>
Route: <sip:mt@scscf.open-ims.eirb:6060;lr>
Route: <sip:mmsc.open-ims.eirb:5060;lr>
Route: <sip:mt@scscf.open-ims.eirb:6060;lr>
Route: <sip:mt@icscf.open-ims.eirb:5060;lr>
Via: SIP/2.0/UDP 172.18.10.2:5060;rport=5060;branch=z
Max-Forwards: 70
From: sip:test2@open-ims.eirb;tag=iE25fWDJ648g1He35uU
To: sip:test@open-ims.eirb
Contact: <sip:test2@172.18.10.2:5060>
Call-ID: dqE4wvcot60AmdECES8yUybBeOpddx1F
CSeq: 22406 INVITE
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSC
Supported: replaces, 100rel, timer, norefersub
Session-Expires: 1800
Min-SE: 90
User-Agent: iPJSUA v1.8/iphone-iOS
Content-Type: application/sdp
Content-Length:   451
```

Figure 4.10: Extra route header in the SIP ReInvite

4.3.2 Theory and QoE estimation for VoIP

Jitter buffers (also known as playout buffers) are used to change asynchronous packet arrivals into a synchronous stream by turning variable network delays into constant delays at the destination end systems. The role of the jitter buffer is to trade off between delay and the probability of interrupted playout because of late packets.

Late or out-of-order packets are discarded. If the jitter buffer is set either arbitrarily large or arbitrarily small, it imposes unnecessary constraints on the characteristics of the network. A jitter buffer set too large adds to the end-to-end delay, meaning that less delay budget is available for the network; hence, the network needs to support a tighter delay target than practically necessary. If a jitter buffer is too small to accommodate the network jitter, buffer underflows or overflows can occur. In an underflow, the buffer is empty when the codec needs to play out a sample. In an overflow, the jitter buffer is already full and another packet arrives; that next packet cannot be enqueued in the jitter buffer. Jitter buffer underflows and overflows cause voice quality degradation. Adaptive jitter buffers aim to overcome these issues by dynamically tuning the jitter buffer size to the lowest acceptable value. Well-designed adaptive jitter buffer algorithms should not impose any unnecessary constraints on the network design by doing the following:

- Instantly increasing the jitter buffer size to the current measured jitter value following a jitter buffer overflow
- Slowly decreasing the jitter buffer size when the measured jitter is less than the current jitter buffer size
- Using PLC to interpolate for the loss of a packet on a jitter buffer underflow

When such adaptive jitter buffers are used in theory, one can "engineer out" explicit considerations of jitter by accounting for worst-case per-hop delays. Advanced formulas can be used to arrive at network-specific design recommendations for jitter (based on maximum and minimum per-hop delays). Alternatively, because extensive lab testing has shown that voice quality degrades significantly when jitter consistently exceeds 30 ms, this 30 ms value can be used as a jitter target.

4.3.3 Jbuf scheduling for multi-homed devices

In our scenario, the MN communicates with the CN through its WLAN interface. The MN triggers mobility request and switches the current traffic towards its UMTS interface. In our test-bed, we use live 3G connection that is actually best-effort Internet traffic. Thus, our UMTS connection is really poor in terms of QoS and bandwidth due to limitation by the operator. We experience high delay during the establishment of the UMTS data connection and burst effect. Our WLAN network is attached to a fast broadband connection provided by the university. Therefore, our jitter buffer algorithm becomes even more important facing the scenario requirements.

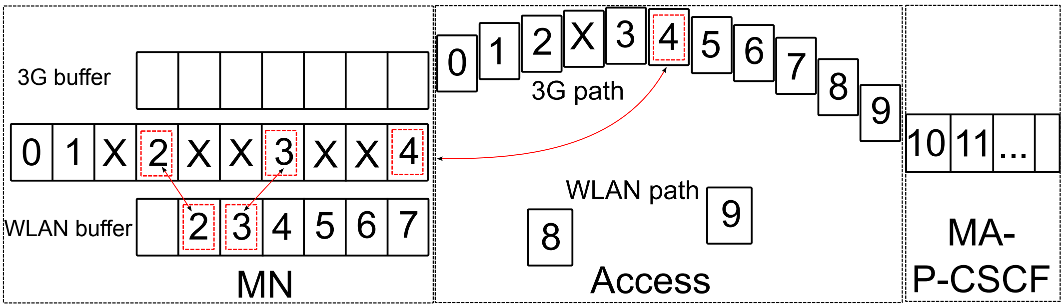


Figure 4.11: Jbuf Scheduling

Figure 4.11 shows the new approach to curb the increase of delay and jitter during the mobility between a WLAN to a 3G access network on the downlink side. Indeed, this algorithm tries to overcome the delay variation created by the new path by scheduling packets received from the WLAN buffer with some virtual delay while waiting 3G buffer to be filled with RTP packets. It is supposed that duplicated data packets travel faster through WLAN access than 3G access. For instance, in the Figure 4.11, each slot represent a data packet generated (i.e. usually every 20ms for VoIP packet). The duplicated packet starts from packet 0 and ends at number 10. The WLAN buffer is full with the duplicated packet whereas 3G buffer still waits for incoming duplicated packets. The algorithm principle described below is to drift audio packet scheduling with empty packets in order to overcome the delay variation. The Algorithm 1 detects whether both network interface buffers have frames or not and schedule them according to what frame the playback needs.

4.4 Test-Bed

The test-bed implemented for validation is a live test-bed focusing on the following scenario depicted in Figure 4.5. An *Apple Iphone 4* smartphone in communication with a Correspondent Node (CN) through its WLAN connection roams towards its public UMTS attachment without any disrupt. The WLAN access is provided by a NetGear WNR3500L router connected to the broadband connection

Algorithm 1 Jitter Buffer algorithm

```
if 3g_buffer is empty then
  if WLAN_packet_number_min > last_scheduled_packet &&
  WLAN_packet[last_scheduled_packet+1] exists then
    schedule WLAN_packet[last_scheduled_packet+1]
  else
    schedule empty frame
  end if
else
  schedule one empty frame
  if 3g_packet_number_min < last_scheduled_packet &&
  WLAN_packet[last_scheduled_packet+1] exists then
    schedule WLAN_packet[last_scheduled_packet+1]
    schedule two empty frames
  else
    if 3g_packet_number_min ≥ last_scheduled_packet &&
    3g_packet[last_scheduled_packet+1] exists then
      schedule 3g_packet[last_scheduled_packet+1]
    end if
    schedule one empty frame
  end if
end if
```

of ENSEIRB-MATMECA. The 3G connection is supported by Orange network. A SSH tunneling is established between the smartphone and the IMS core hosted in ENSEIRB-MATMECA premises as depicted in Figure 4.12.

4.4.1 IMS platform

The Open IMS Core [120] is an Open Source implementation of IMS acCSCF and a lightweight HSS, which together form the core elements of all IMS/NGN architectures as specified today within 3GPP, 3GPP2, ETSI TISPAN and the PacketCable intiative. The four components are all based upon Open Source software (i.e. the SIP Express Router (SER) or MySQL).

4.4.2 Mobile Node (MN)

The IMS client is a PJSUA [121] open source softphone originally developed on x86 architecture. A ported version on iOS system called iPJSUA has recently been released with version 1.8. Nevertheless, the effort concerning the modification of this client is twofold. First, the current version of PJSUA does not allow to trigger

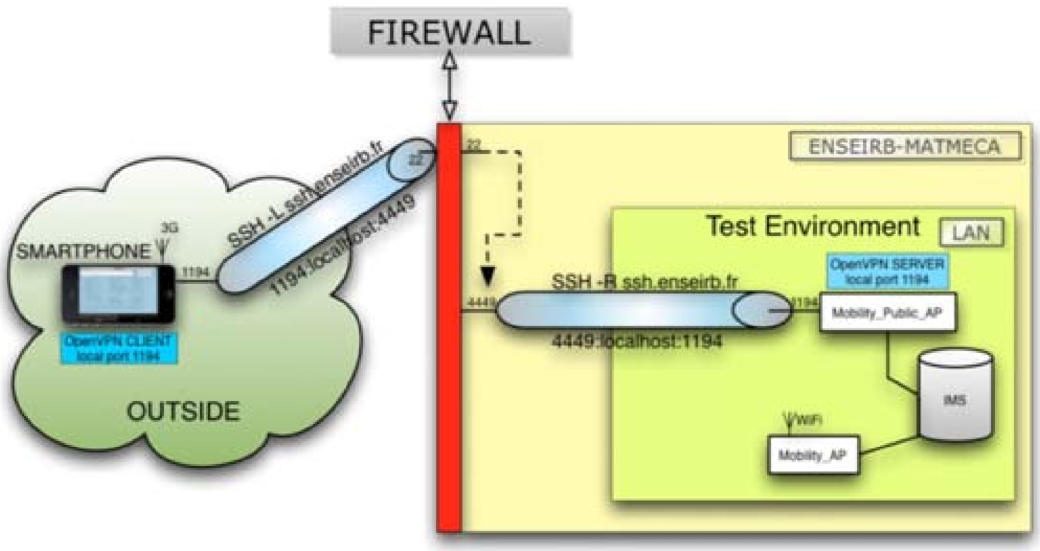


Figure 4.12: Iphone 3G connection towards IMS Core Network

Table 4.1: IMS configuration

	P-CSCF 1	MA-PCSCF
Operating System	Linux Ubuntu 9.10	Linux Ubuntu 9.10
Software Version	Open IMS Core	Open IMS Core
Network Interfaces	eth0 (UMTS) : 172.18.10.22	eth0 (WLAN) : 192.168.10.5
	eth1 (Core) : 172.18.1.22	eth1 (Core) : 172.18.1.5

handover through different network interfaces. Second, the PJSUA modifications require extra effort to port them into iPJSUA. Unfortunately, Android Operating System does not accept to have two data connections enabled at the same time. Many hacks were tried to enable both interfaces, without success.

The main functionality added to iPSJUA is the implementation of the jitter buffer scheduler following the Algorithm 1. The mobility functionalities added to the Command Line Interface (CLI) of iPJSUA program are shown in Figure 4.13. GSM and G711 codecs are used as voice codec but no adaptation mechanism according to wireless network condition has been yet developed.

Table 4.2 summarizes the configuration of the MN and CN.

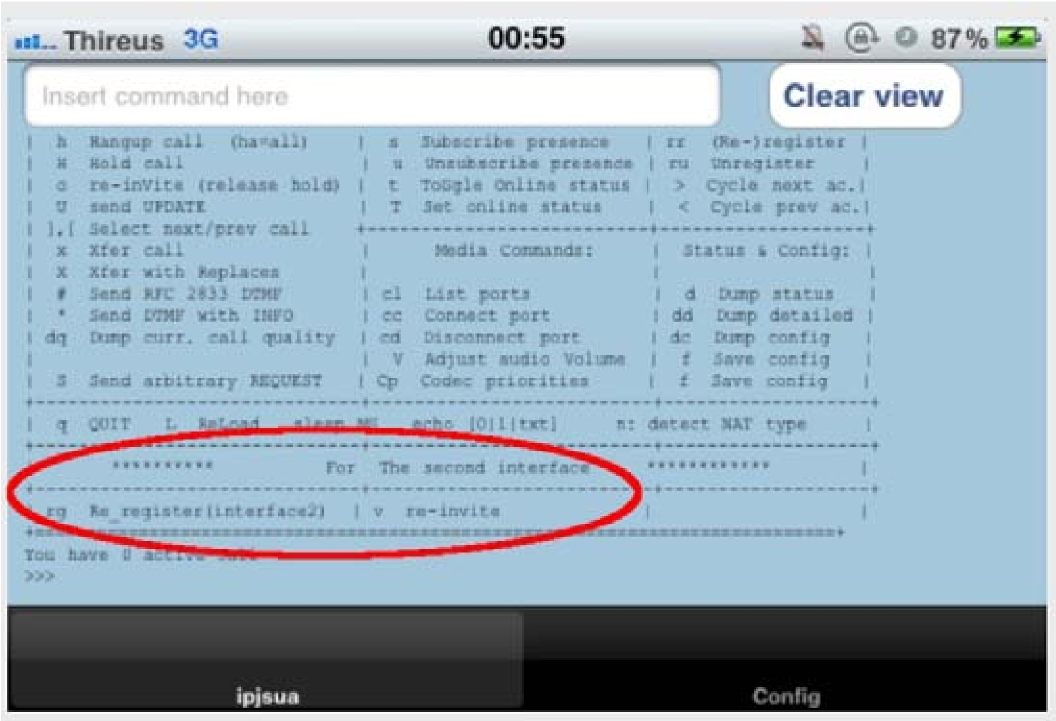


Figure 4.13: modified PJSUA IMS client on Iphone

Table 4.2: MN and CN configuration

	MN	CN
Operating System	iOS 5.1 (root)	Linux Ubuntu 10.04
Software Version	modified iPJSUA 1.8.10	PJSUA 1.8.10
Network Interfaces	WLAN : 192.168.10.2	One LAN interface
	UMTS : 172.18.10.2	IP : 192.168.3.5

4.4.3 Performance Evaluation

Table A.2 summarizes the total signaling cost of the processes for a IMS network hiding its topology which means that all signaling messages are going through the I-CSCF for security reasons. Our SIP handover solution is compared against results given by [83] which is treating mobility between two visited network. In order to compare the signaling cost of our solution, we consider that all SIP signaling goes through an I-CSCF and that the mobility is done in this central equipment. This condition is not changing our results. The normal SIP handover is the solution based on SIP Re-INVITE through IMS until the CN. mSCTP-based solution for IMS presented in [83] needs 58 messages for the full handover. Our solution requires

19 messages as depicted in Figure 4.7. Note that 100 Trying and ACK messages are not shown for the simplicity of the figure.

Table 4.3: Signaling Messages

Operation	Number of signaling messages during HO		
	mSCTP [83]	SIP [98]	MA-PCSCF
Registration/LU	24	24	24
Session Establishment	66	46	46
Full Handover Process	58	46	19

Table A.1 compares the existing mobility solutions with the proposed MA-PCSCF solution. The MA-PCSCF solution is transparent to existing architecture and requires less signaling compared to other solutions. Only small modifications in MN and P/I-CSCF are required. Another strong aspect is that the user interest is involved in the handover process using efficiently several network interfaces available. In our test-bed, only two network interfaces were available but our implementation works with any access technologies having an entry point to IMS core. Finally, a new algorithm fully described in this chapter, schedules media packets from heterogeneous network interfaces and tends to overcome delay variation and packet loss during vertical handovers. The full solution was tested in a live test-bed using corporate WLAN of ENSEIRB-MATMECA premises and public UMTS connection. More qualitative results such as PQoS[50] or [53] using embedded PQoS estimation curves are foreseen in the near future. VoIP adaptation using the *novel IMS for adaptive media services* of Chapter 3 is also foreseen.

4.5 Conclusion

This chapter describes a vertical mobility solution for conversational services such as VoIP. Vertical mobility usually implies complex signaling and costly integration. Our Mobility-Aware P/I-CSCF (MA P/I-CSCF) solution does not require any integration effort within existing IMS architecture and supports any kind of handover (i.e. pico, macro, global handover).

Our mobility solution is NGN oriented as the session control is aware about the mobility and participates in the handover process. VoIP coding parameters can then

Table 4.4: Summary of Mobility Solution

Protocol criteria	*(DS)MIP*[71; 75]	*PMIP*[108]	*mSCTP*[83]	*SIP*[98]	*B-SIP*[89]	*MA-PCSCF*
Operating Layer	Network	Network	Transport	Application	Application	Application
Mobility Scope	Global	Local	Local/Global	Local/Global	Local/Global	Local/Global
Mobility Managing	Host-based	Network-based	Host-based	Host-based	Host-based	Host-based
Handover Control	Hard	Hard, soft[79]	Soft	Soft	Soft	Soft
Required Infrastr.	HA	LMA, MAG	mSCTP proxy	MIH-based	B2BUA in BS	MA-P/I-CSCF
MN Modification	Yes	No	Yes	Yes	Yes	Yes
Tunneling over WL	Required	No	No	No	No	No
Handover Latency	Bad	Bad, good[77]	Good	Bad	Good	Good
Complex to deploy	No	No	Yes	Yes	Yes	No
Scalability	No	No	No	Yes	No	Yes
Overhead	No	No	Yes	No	No	No
IMS/NGN oriented	No	No	No	Yes	Yes	Yes
QoE management	No	No	No	No	No	Yes

be adjusted to the new access network conditions and improve then the perceived quality. The design of the mobility functionalities within MA-PCSCF allows to centralize the RTP switching reducing the hopes in the media delivery chain.

We implemented our mobility solution on a live test-bed with WLAN and 3G connections. The IMS client supports our new algorithm handling media from two separate network interfaces. This algorithm reduces the overall delay variation of the two access networks. Even if the new interface is not filled with current packet, our algorithm triggers packet arrived on the previous network interface.

A complete mobility solution for conversational services is defined and implemented focusing on the perceived quality.

The IMS client has been successfully tested on a iOS smartphone connected to a real 3G network.

Chapter 5

Conclusion & Future Works

The exponential growth of media consumers demanding high quality media services on huge diversity of devices forces service providers to consider the end user's context in their delivery system. This raises stringent constraints in NGN design even more when end-users have high expectations for better experience.

Four main contributions have been proposed in this thesis to satisfy these requirements. Three of them are introduced in the novel IMS-based architecture achieving adaptation for IPTV services and the last one proposes a seamless mobility in IMS environment based on end-user's context :

- In order to address the lack of context-awareness and PQoS management within NGNs, we proposed an evolved IMS-based solution delivering adaptive IPTV service based on the user's context. Our proposal employs a **flexible User Profile mechanism** within the MN to convey dynamic user's environment information to the IMS core network. Then, the innovative element **MCMS was seamlessly integrated** into the IMS application layer to perform dynamic cross layer adaptations according to the PQoS information retrieved from strategic network elements in the delivery chain. When PQoS degradation occurs at end-user side, MCMS triggers service adaptation enforced by the MSRF. Finally, a **PQoS prediction model based on the content video dynamics** is defined and integrated in the MSRF. This mechanism enables to select the right video encoding parameters for a targeted PQoS.

- To achieve better user satisfaction during vertical handover, we introduced a **Mobility-Aware P/I-CSCF interworking with a flow scheduling technique in multi-homed devices**. This solution extends the VoIP service coverage leveraging private networks connected to broadband access (i.e Corporate Networks, Home Networks). The proposed service continuity solution enables seamless vertical handover between two disparate wireless networks : WLAN and UMTS. Many proposals already exist but none of them focus specifically on media handling and user satisfaction. In this thesis, we address this issue by defining the Mobility Aware-P/I-CSCF. This solution is transparent to existing IMS architecture and requires less signaling compared to other studied solutions. In order to curb delay variation during vertical handovers, a new jitter buffer algorithm is introduced at the End-User device leveraging the packets duplication mechanism of the MA-P/ICSCF.

The work achieved in this thesis brings many concepts but need further experiments to be completely approved. Several perspectives have then been identified for ongoing and further work. For assessing the context-awareness within IMS, we have identified the following directions :

- More extensive objective PQoS estimation is needed to validate our mobility solution. How many packet do we need to duplicate within MA P-CSCF? What is the overall PQoS estimation during the handover process.
- Concerning the PQoS prediction model based on content video dynamics, future works will lead to study variable bit-rate encoding for longer video. Until now, only short video with uniform sequences are considered. "On-the-fly" encoding parameters could be chosen according to the forthcoming video dynamics and will lead to a better bandwidth optimization in the core network.
- In this thesis, we are focusing on providing adaptation for services requiring high bit-rate such as IPTV and mobility for conversational service such as VoIP. Services combining both aspects such as video calls or interactive IPTV impose even more stringent requirements on our solution. Interactive video services (i.e. games executed on distant servers) require evolution of existing PQoS models discussed in this thesis. No accurate objective PQoS model define the user satisfaction concerning his action on the service. Video calls raise

additional issues for vertical mobility mechanism. Different buffers for voice and video have to be used and managed. As video requires more bandwidth than voice, dynamic adaptation would be required when the MN switches access networks (i.e. WLAN to 3G). Nevertheless, the following bullet suggest a method to fulfill the mobility requirements.

- The MSRF server is only supporting MPEG-4 video adaptation. The evolution towards Scalable Video Coding (SVC) will facilitate the adaptation process by streaming only required layers. This solution will enable seamless handover for video calls.

- The mobility mechanism introduced in chapter 4 requires a handover triggering system that launches the mobility process efficiently. We need to study the balance between changing access network or keeping the current connection. We have to compare the minimum PQoS acceptable in both networks and make a decision whether to accept or not the mobility request. The mobility management could take advantage of the MCMS decision taking.

Appendix A

Résumé en Francais

A.1 Introduction

Les services médias font partie intégrante de notre vie quotidienne. Les récentes évolutions technologiques nous permettent de consommer ces services de façon ubiquitaire et transparente. La mobilité connait une forte croissance ces dernières années avec l'apparition de nouveaux terminaux dit "légers" ou "mobiles". D'après Cisco VNI[1], le trafic IP sera routé à plus de 61% vers les équipements mobiles dotés d'interface Wifi ou 3G/4G. De plus, ce fort engouement des utilisateurs à consommer ces services médias (Vidéo à la demande, IPTV, Téléphonie IP, Jeux en ligne, etc.) sur leur terminaux mobiles impose de lourdes contraintes en termes de montée à l'échelle, de fiabilité et de performance.

De plus, l'attente des consommateurs de ces services ubiquitaires sont très élevée. En effet, du point de vue de l'utilisateur, seul le rendu final est apprécié par l'utilisateur peu importe le terminal ou le réseau utilisé pour y accéder. L'utilisateur s'attend à recevoir la qualité maximale par rapport à son environnement. Les services médias consommés étant gourmands en bande passante et sensibles à la qualité du réseau, les fournisseurs de services ou de contenu doivent prendre en compte ces nouveaux paramètres liés à l'environnement de l'utilisateur (capacité du terminal, conditions réseaux, langue choisie, options de sécurité...).

Cependant, les standards actuels définissant les réseaux de nouvelle génération ne prennent pas en compte le contexte utilisateur et ne s'assurent pas de la satisfaction de l'utilisateur. Nous proposons donc dans cette thèse les deux évolutions suivantes : (1) l'introduction de la sensibilité au contexte utilisateur et de la PQoS ; et (2) une solution de mobilité efficace pour les services conversationnels tels que la VoIP.

La première contribution permet de diffuser les services multimédias en fonction de l'évolution du contexte utilisateur et de la nature de la source média. Elle assure aussi une qualité d'expérience à l'utilisateur et optimise par la même occasion l'utilisation des ressources des éléments réseaux impliqués. Pour cela, nous avons spécifié un nouvel élément central au sein de l'IMS : le MCMS. Cet élément intelligent récolte les informations de qualité de service à différents équipements réseaux et prend la décision d'une action au niveau d'un de ces équipements. Nous introduisons aussi un profil utilisateur permettant de décrire l'environnement de l'utilisateur qui sera diffusé en coeur de réseau. Afin d'optimiser l'utilisation des capacités des réseaux et du terminal mobile de l'utilisateur tout en assurant une qualité perçue par l'utilisateur, une étude sur la prédiction de la satisfaction utilisateur en fonction des paramètres spatio-temporels de la vidéo a été réalisée afin de connaître le débit idéal pour une PQoS désirée.

La deuxième contribution définit une solution de mobilité adaptée aux services conversationnels tels que la VoIP ou services interactifs tout en tenant compte du contexte utilisateur. Notre solution s'intègre à l'architecture IMS standardisée par 3GPP et permet de réduire le temps de latence du handover. Afin d'assurer la transparence de la mobilité à l'utilisateur, un nouvel algorithme de gestion de la mémoire tampon améliore la qualité d'expérience pour le service de VoIP.

A.2 Evolution des réseaux de nouvelle génération

Les réseaux de nouvelle génération sont apparus avec la migration des réseaux vers le "tout IP" permettant la convergence des réseaux et des services. Ces réseaux comportent une gestion de qualité de service au niveau réseau comme illustré dans la figure A.1. Les nouveaux services tendent à devenir "User Centric" qui signifie centré autour de l'utilisateur. Pour cela, nous devons tenir compte la qualité percue par l'utilisateur aussi appelée qualité d'expérience.

L'IMS est une architecture de nouvelle génération définie par 3GPP permettant la gestion de bout en bout de services médias à travers des réseaux hétérogènes. Ce standard ne prend malheureusement pas en compte l'évolution de l'environnement de l'utilisateur. Pour cela, nous définissons la gestion du profil utilisateur illustré par

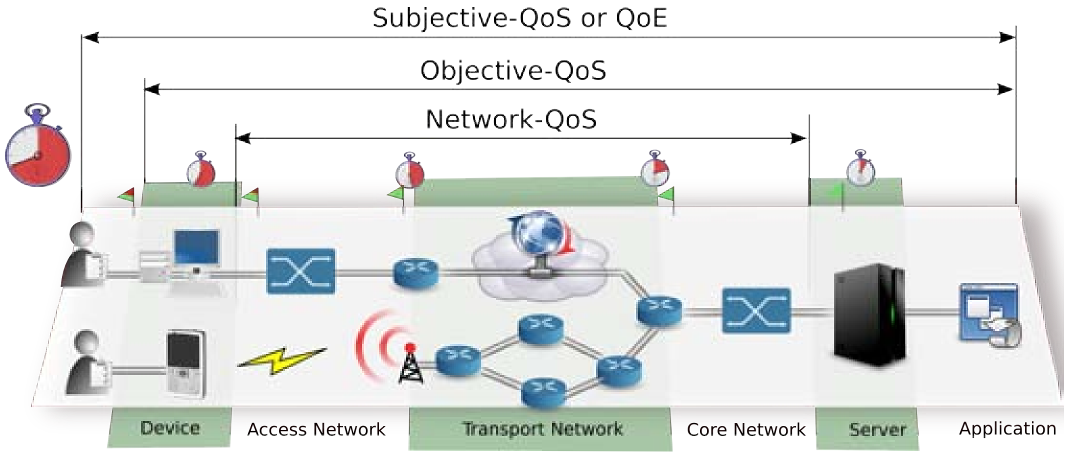


Figure A.1: Different definition of QoS

la figure A.2. Ce profil utilisateur va interagir avec les applications et la couche transport afin de compléter les requêtes multimédias avec des informations sur l'environnement de l'utilisateur. Il va permettre de récolter des informations statiques (informations personnelles, préférences utilisateurs...) et dynamiques comme l'état du terminal (charge de la batterie, bande passante utilisée, charge processeur, mémoire disponible...).

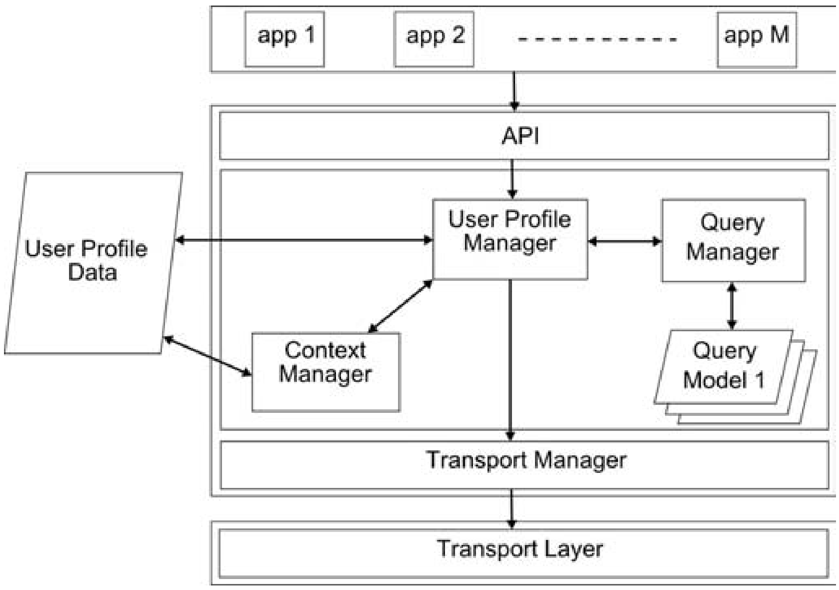


Figure A.2: Gestion du Profil Utilisateur

Ce profil utilisateur [7] se base sur un format XML et contient cinq sections: le profil utilisateur général, le profil du terminal, le profil réseau, le profil du service et le profile du contexte contenant toutes les données dynamiques et volatiles. Ce format XML a été choisi car il peut facilement être intégré dans des messages de signalisation HTML ou SIP.

L'architecture IMS au sein du projet ADAMANTIUM va pouvoir traiter ces informations et prendre une décision à différents endroits dans la chaîne de diffusion comme illustré dans la figure A.3.

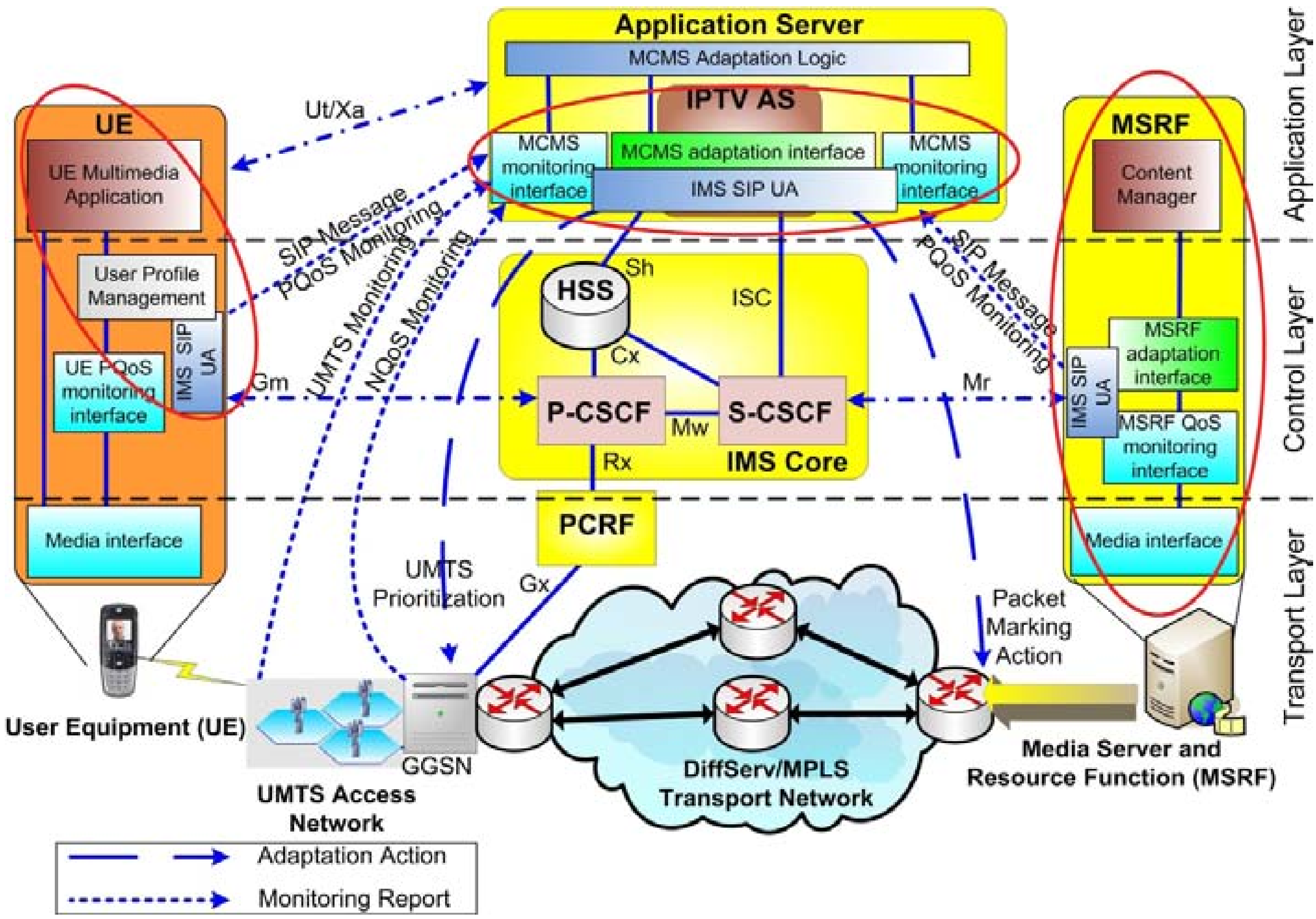


Figure A.3: Architecture de l'IMS avec sensibilité au contexte utilisateur

Le module central de ce mécanisme d'adaptation sensible au contexte du réseau et de l'utilisateur est intitulé MCMS. Ce module assemble et traite les informations de qualité de service à différent endroits stratégiques. Suite à ce traitement, le MCMS prend des décisions d'actions afin d'améliorer un problème de QoE détecté au niveau de l'utilisateur. La figure A.4 décrit un exemple de cas d'utilisation d'adaptation aux niveaux des trois modules d'adaptation.

La signalisation complète d'une renégociation de codec vidéo au sein d'une ses-

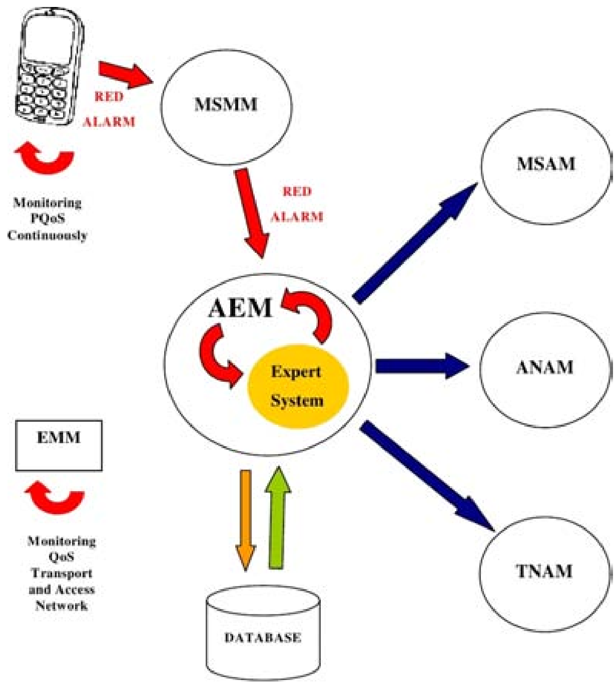


Figure A.4: Prise de décision à l'intérieur du MCMS suite à une alerte du terminal

sion IPTV est illustrée dans la figure A.5.

Grâce au User Profile Manager, l'architecture globale permet de diffuser les flux médias adaptés aux conditions réelles de l'utilisateur. L'ajout du module MCMS dans le plan de signalisation permet d'adapter dynamiquement le flux média à travers différents équipements stratégiques (MSRF, routeur en bordure de coeur de réseau, GGSN, UTRAN).

En plus de pouvoir adapter le flux médias de façon dynamique, le MSRF, serveur de contenu et de diffusion, contient un outil de prédiction de PQoS basé sur la dynamique de la vidéo. Pour une qualité perçue équivalente, les caractéristiques spatio-temporelles d'un contenu vidéo influent grandement sur les paramètres d'encodage et de transport lors de la diffusion. Un contenu dynamique devra être encodé avec un débit bien plus important [9] qu'un contenu présentant peu de mouvements. Pour cela, nous avons défini des seuils de débit A.6 d'un flux vidéo de résolution fixe afin d'obtenir une qualité exigée par l'utilisateur.

Afin de valider nos propositions, nous avons défini un banc de test pour des services IPTV diffusés à travers des réseaux hétérogènes et consommés sur des terminaux différents en terme de capacités. La figure A.7 met en évidence l'utilité du User Profile lors du démarrage du service IPTV. Les résultats de qualité perçue par

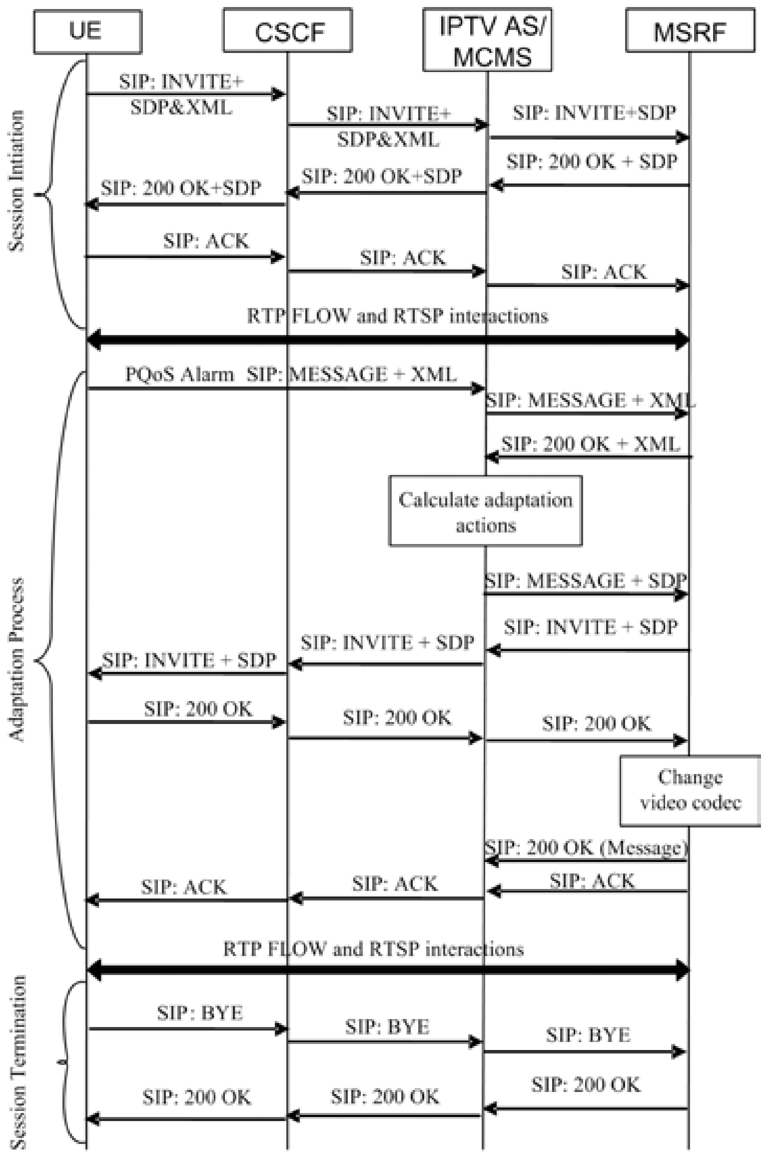


Figure A.5: Diagramme des messages pour le service d'IPTV avec adaptation avec renégociation SDP

les utilisateurs sont supérieurs lorsqu'ils sont directement adaptés aux conditions du réseau et du terminal notamment pour les terminaux à faible capacité. L'adaptation dynamique en fonction de la PQoS perçue par l'utilisateur permet d'améliorer de 30% environ la satisfaction de l'utilisateur.

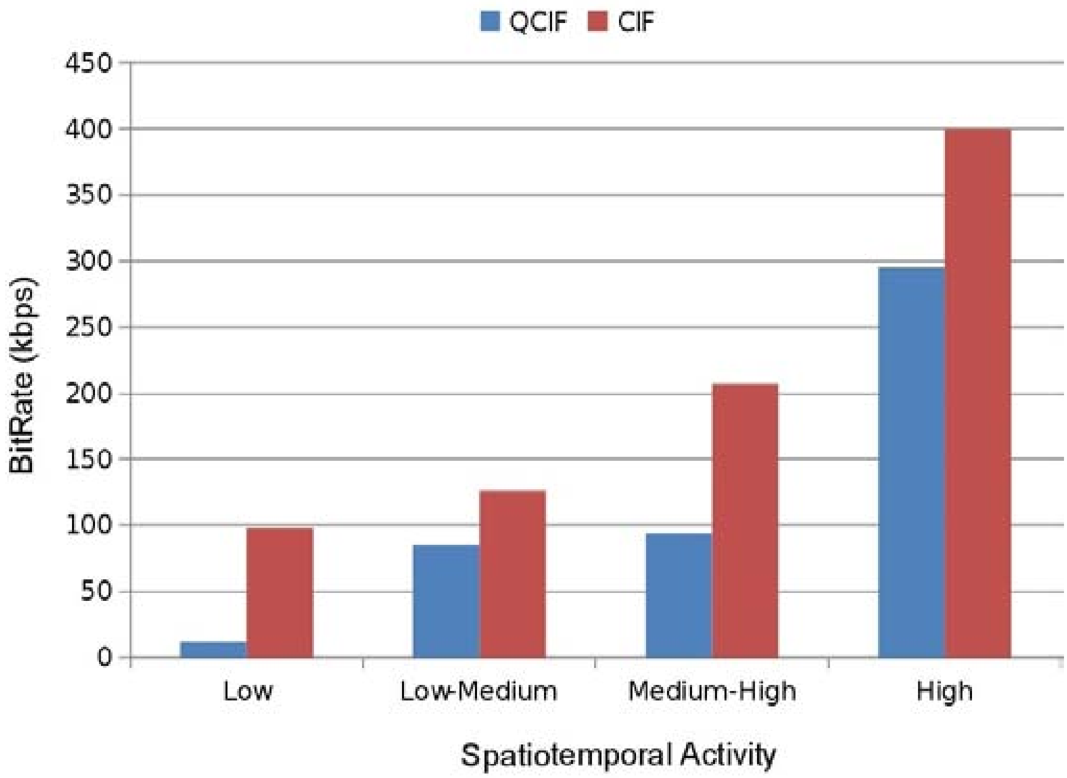


Figure A.6: Influence de la dynamique video pour un seuil de PQoS donné

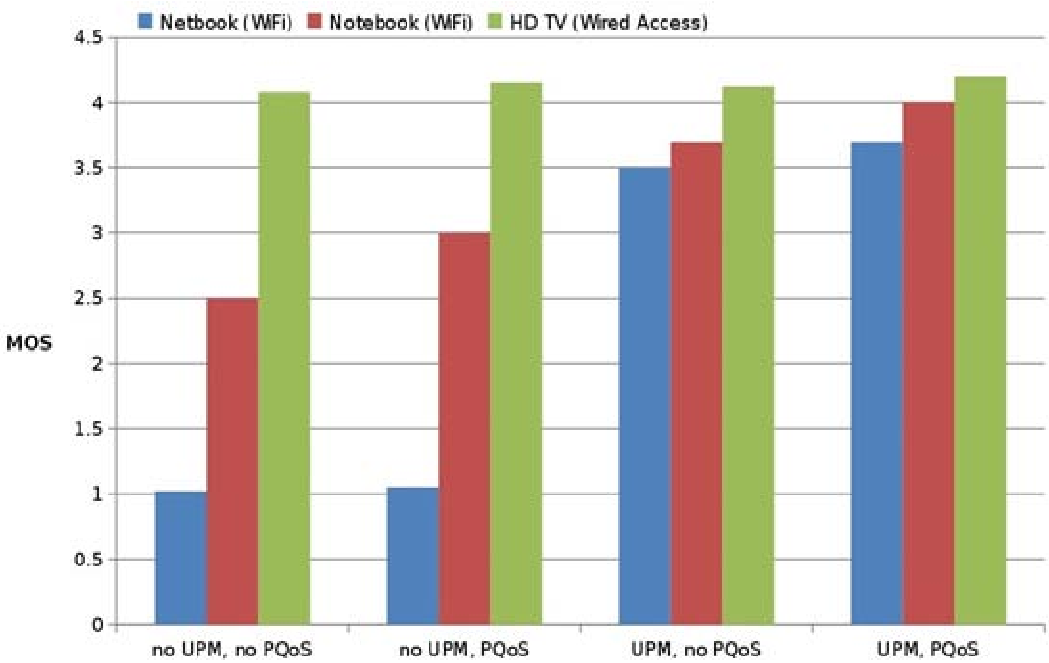


Figure A.7: Estimation du MOS avec ou sans utilisation du UPM and et de l'adaptation dynamique en fonction de la PQoS

A.3 Solution de mobilité au sein d'une architecture IMS tenant compte du contexte utilisateur

Les dernières évolutions du standard IMS introduisent plusieurs niveaux de gestion de mobilité. L'introduction des technologies LTE dans ĺ'architecture IMS permet de gérer la mobilité au niveau IP A.8 dans le PDN Gateway alors que le standard IMS définit une mobilité au niveau du protocole SIP. Nous avons résumé les différentes solutions de mobilité dans le tableau 2.5.

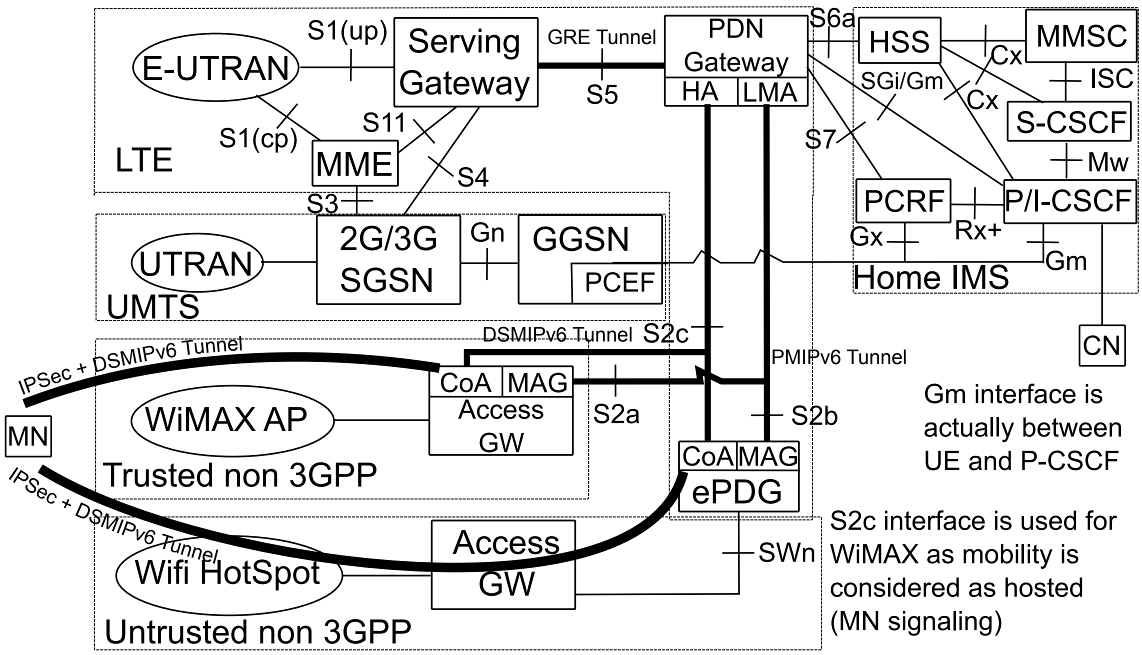


Figure A.8: Interfaces LTE

xMIP est une famille de mécanismes de mobilité au niveau IP. Le principal problème de cette solution est la non prise en compte du profil utilisateur et sa complexité pour gérer tous les scénarios de mobilité (micro, macro, globale). La mobilité au niveau du protocole SIP est bien connue mais se base uniquement sur la signalisation et non pas sur le flux média essentiel à l'utilisateur. Nous proposons donc d'intégrer notre solution de mobilité dans l'équipement P-CSCF qui gère aussi bien la signalisation que le transport média. La figure A.9 décrit le scénario de mobilité d'un utilisateur se déplaçant d'un réseau Wi-fi à un réseau 3G. Nous pouvons observer le nouvel élément MA-PCSCF qui joue le rôle stratégique de la mobilité. Notre solution est bien sûr comparée dans le tableau A.1.

Table A.1: Summary of Mobility Solution

Protocol criteria	*(DS)MIP*[71; 75]	*PMIP*[108]	*mSCTP*[83]	*SIP*[98]	*B-SIP*[89]	*MA-PCSCF*
Operating Layer	Network	Network	Transport	Application	Application	Application
Mobility Scope	Global	Local	Local/Global	Local/Global	Local/Global	Local/Global
Mobility Managing	Host-based	Network-based	Host-based	Host-based	Host-based	Host-based
Handover Control	Hard	Hard, soft[79]	Soft	Soft	Soft	Soft
Required Infrastr.	HA	LMA, MAG	mSCTP proxy	MIH-based	B2BUA in BS	MA-P/I-CSCF
MN Modification	Yes	No	Yes	Yes	Yes	Yes
Tunneling over WL	Required	No	No	No	No	No
Handover Latency	Bad	Bad, good[77]	Good	Bad	Good	Good
Complex to deploy	No	No	Yes	Yes	Yes	No
Scalability	No	No	No	Yes	No	Yes
Overhead	No	No	Yes	No	No	No
IMS/NGN oriented	No	No	No	Yes	Yes	Yes
QoE management	No	No	No	No	No	Yes

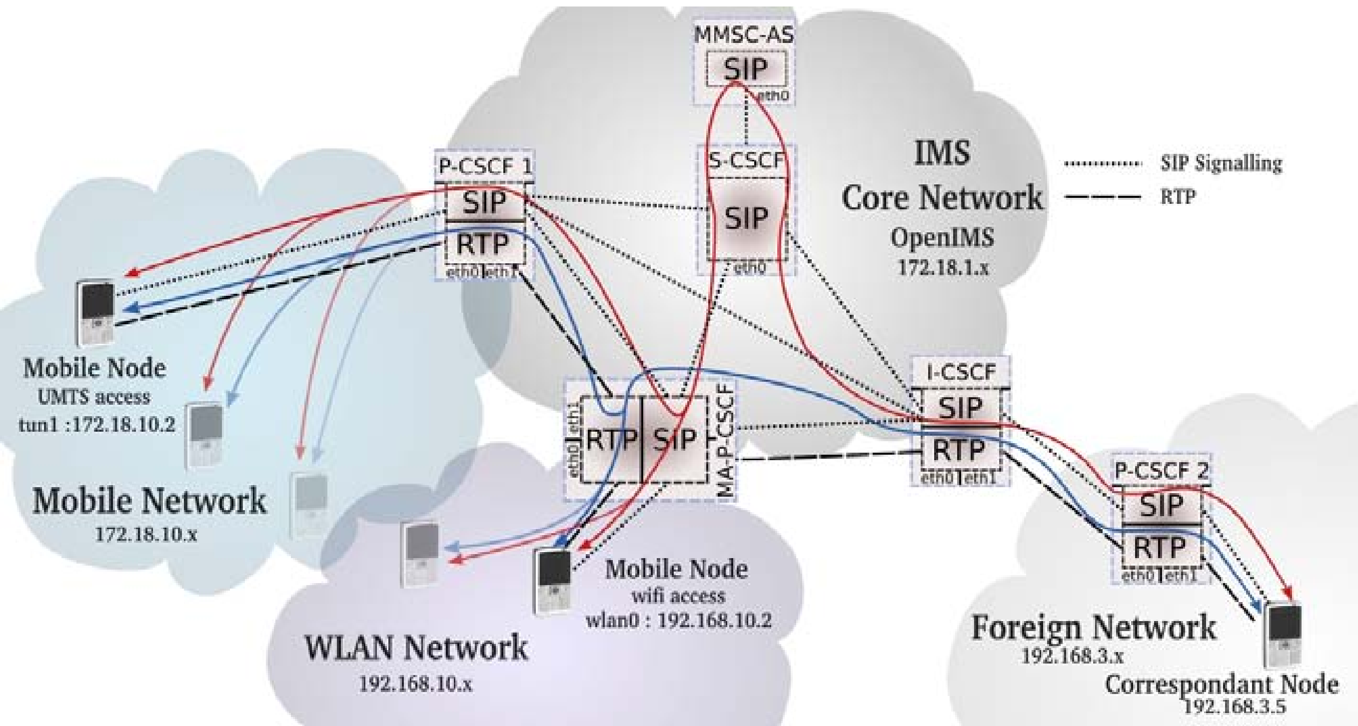


Figure A.9: Solution de mobilité

Le tableau A.2 présente le nombre de messages nécessaires pour la gestion complète du handover. Le fait d'introduire la mobilité dans le P-CSCF apporte une simplification au niveau de la signalisation mais aussi au niveau de l'efficacité du changement de chemin du flux média.

Table A.2: Signaling Messages

Operation	Number of signaling messages during HO		
	mSCTP [83]	SIP [98]	MA-PCSCF
Registration/LU	24	24	24
Session Establishment	66	46	46
Full Handover Process	58	46	19

Afin de s'assurer de la qualité perçue par l'utilisateur, nous définissons aussi un algorithme de gestion de flux RTP. Le flux médias est dupliqué au niveau du MA-PCSCF pendant l'exécution du handover comme illustré dans la figure A.10.

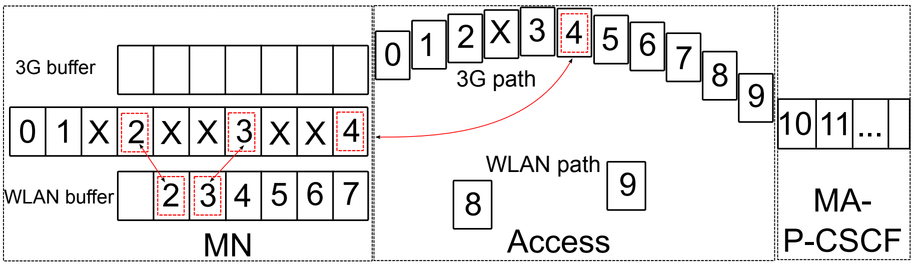


Figure A.10: L'ordonnanceur de la mémoire tampon

L'établissement du flux de données à travers le réseau 3G ajoute de la latence et des pertes de paquets. Un ordonnanceur de mémoire tampon au niveau du terminal utilise les données dupliquées arrivant par les deux chemins afin de pouvoir rendre la mobilité transparente à l'utilisateur final. Notre algorithme prévoit de rajouter des paquets vides entre les paquets de données (comme le PLC) afin de combler la latence et les pertes de paquets induit par le handover.

Des tests réels sont réalisés pour valider nos propositions. Aucun effet indésirable dû à la mobilité n'a été perçu par nos testeurs.

A.4 Conclusion

La prolifération de nouveaux équipements hétérogènes pour la consommation de services médias tout aussi diversifiés ont donné naissance à de nombreuses problématiques de recherche. Dans cette thèse, nous nous sommes particulièrement intéressés à l'introduction de la sensibilité au contexte de consommation au sein des réseaux de nouvelle génération. Ces réseaux permettent la convergence des réseaux et des services en unifiant l'accès aux services médias. L'utilisateur pourra donc s'authentifier auprès d'une seule plateforme et consommer les services médias sur un panel de terminaux. Afin de s'assurer de la qualité perçue par l'utilisateur final, nous introduisons la gestion d'un profil utilisateur combiné avec un mécanisme d'adaptation dynamique au sein d'une architecture IMS. Ce profil utilisateur permet d'adapter le flux directement lors du démarrage du service média. Un mécanisme de prédiction de PQoS optimise les paramètres d'encodage des flux médias en fonction de la nature du média pour une qualité d'expérience garantie réduisant ainsi l'utilisation des resources réseaux. L'introduction d'un estimateur de PQoS informe le coeur de réseau d'un problème de diffusion au niveau réseau. Le MCMS prend ensuite la décision d'agir à un endroit précis le long de la chaîne de diffusion.

Cette thèse introduit aussi une solution de mobilité transparente à l'architecture IMS car elle est basée sur le protocole SIP. Notre solution intègre les fonctionnalités de mobilités dans le MA-PCSCF afin de pouvoir faciliter la gestion de la signalisation et du flux média. Afin de contrer les effets de délais et perte de paquets pendant le changement d'interface, nous présentons un algorithme de gestion de mémoire tampon optimisant l'ordonnancement des paquets médias.

Appendix B

Publications

Journal Papers :

"Adaptive IPTV services based on a novel IP Multimedia Subsystem", Julien Arnaud, Daniel Négru, Mamadou Sidibe, Julien Pauty, Harilaos Koumaras, *In Multimedia Tools and Applications*, Springer Netherlands, pp. 1-20, 2010.

"IPTV QoS adaptation for multi-homed mobile terminals in a new IMS based architecture", Sinda Boussen, Julien Arnaud, Francine Krief, Nabil Tabbane, Sami Tabbane, *Telecommunication System Journal*,

Conference Papers :

"Flow Scheduling at Multihomed devices during vertical handover in NGN", J. Arnaud, D. Négru, *In International Wireless Communications and Mobile Computing Conference (IWCMC), 2012 8th IEEE*, 2012.

"Mobility-Aware P/I-CSCF : a solution for achieving seamless handover in IMS", J. Arnaud, D. Négru, *In IEEE Symposium on Computer and Communications (ISCC), 2012 17th IEEE*, 2012.

"Using PR-SCTP for IPTV QoS Adaptation Over IMS Network", Sinda Boussen, Julien Arnaud, Francine Krief, Nabil Tabbane, *WMCNC2011*, 2011

"Flexible User Profile Management for Context-Aware Ubiquitous Environments", S. Ait Chellouche, J. Arnaud, D. Négru, *In Consumer Communications and Networking Conference (CCNC), 2010 7th IEEE*, pp. 1-5, 2010.

"An experimental approach of video quality level dependence on video

content dynamics", Harilaos Koumaras, Julien Arnaud, Daniel Négru, Anastasios Kourtis, *In Proceedings of the 5th International ICST Mobile Multimedia Communications Conference*, ICST (Institute for Computer Sciences, Social-Informatics and Telecommunications Engineering), pp. 44-48, 2009.

"Adapted IPTV service within novel IMS architecture", Julien Arnaud, Daniel Négru, Mamadou Sidibé, Julien Pauty, Harilaos G. Koumaras, *In Proceedings of the 5th International ICST Mobile Multimedia Communications Conference*, ICST (Institute for Computer Sciences, Social-Informatics and Telecommunications Engineering), pp. 43:1-43:5, 2009.

Reports :

"D3.1 : Design and Development of the MCMS Action Engine Module",Harilaos Koumaras, Lemonia Boula, Julien Pauty, Mamadou Sidibé, Fidel Liberal, José Óscar Fajardo, Julien Arnaud, Thierry Filoche, Mónica Gorricho, Alejandro Bascuñana, Miguel Ángel García. *ADAMANTIUM) deliverable*, 2010.

"D3.2f : Development of MCMS monitoring and adaptation modules", Harilaos Koumaras, Lemonia Boula, George Xilouris, Nikolaos Zotos, Julien Pauty, Mamadou Sidibé, Ioanna Gkika, Julien Arnaud, Filoche Thierry, Richard Finkenzeller, Prof. Emmanuel Ifeachor, Lingfen Sun, Is-Haka Mkwawa, Emmanuel Jammeh. *ADAMANTIUM) deliverable*, 2010.

"D3.3f : IPTV and VoIP Services generation and adaptation", Harilaos Koumaras, Lemonia Boula, George Xilouris, Fidel Liberal, Jose Oscar Fajardo, Prof. Emmanuel Ifeachor, Lingfen Sun, Is-Haka Mkwawa, Emmanuel Jammeh, Julien Pauty, Mamadou Sidibé, Ioanna Gkika, Thierry Filoche, Julien Arnaud, Ana Sanz, Antoine de Poorter, Monica Gorricho. *ADAMANTIUM) deliverable*, 2010.

"D5.2 : Integration of IPTV and VoIP applications", Harilaos Koumaras, Lemonia Boula, George Xilouris, Fidel Liberal, Jose Oscar Fajardo, Prof. Emmanuel Ifeachor, Lingfen Sun, Is-Haka Mkwawa, Emmanuel Jammeh, Julien Pauty, Mamadou Sidibé, Ioanna Gkika, Thierry Filoche, Julien Arnaud, Ana Sanz, Antoine de Poorter, Monica Gorricho. *ADAMANTIUM) deliverable*, 2010.

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