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Final Report

WiMAX Technology Assessment in the Context of the Fourth Generation Networks

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Abstract

With the rising of mobile devices and applications, and the need for constant connection to the network and to the Internet, wireless technologies are constantly changing and being updated. The Worldwide Interoperability for Microwave Access (WiMAX) and the Long Term Evolution (LTE) technologies have emerged in this context. WiMAX is a Broadband Wireless Access (BWA) network, based on the IEEE 802.16 standard, aiming to provide network access with an extensive coverage offering good levels of availability and quality of service. The LTE, based on the 3GPP release 8, is also a BWA network, aiming to provide a smooth transition between the latest 3GPP releases, offering better Quality of Service (QoS) support, coverage and throughput. Both technologies are considered as a fourth generation networks (4G), with the capability to support the existing and emerging multimedia applications.

This work presents an analysis of the WiMAX and LTE technologies, with emphasis on the assessment of multimedia applications performance over a WiMAX testbed in real scenarios and through simulations. This analysis comprises the evaluation of generic applications, representing general traffic patterns, through network and connection quality parameters in different scenarios, namely with and without line of sight, as they are key conditions for the analysis of wireless networks. Also, a specific analysis of Voice over IP (VoIP) is conducted on the same scenarios, assessing the capacity of the WiMAX native QoS mechanisms, through QoS and Quality of Experience (QoE) metrics. This work is conducted in the context of the Energias De Portugal (EDP)-WiMAX project, a project that consisted of building a WiMAX Testbed in order to test multimedia and EDP-specific applications in real scenarios. A simulation evaluation, comparing WiMAX and LTE technologies, is presented, assessing the capabilities of each technology concerning the multi-users support in realistic access conditions, evaluating the end-user perceived voice quality.

This study has shown the ability of Mobile WiMAX and LTE in the support of multiple multimedia applications and with several simultaneous users in different scenarios, and also, evaluated the ability of native WiMAX and LTE QoS mechanisms. Both WiMAX and LTE technologies have proven to be capable of supporting such applications in all the tested scenarios, maintaining good QoS and QoE values, however, with some differences related to the QoS assurance.

Keywords: IEEE 802.16, Mobile WiMAX, WiMAX assessment, LTE, LTE assessment, Quality of Service, Quality of Experience, Testbed, Simulation

Table of Contents

List of Figures

List of Tables

Acronyms

3G Third Generation of mobile phone standards and technology.

3GPP 3rd Generation Partnership Project.

4G Fourth Generation of mobile phone standards and technology.

AAA Authentication, Authorization and Accounting.

AAS Advanced Antenna Systems.

ADPCM Adaptive Differential Pulse Code Modulation.

ADSL Asymmetric Digital Subscriber Line.

AK Authorization Key.

AKA Authentication and Key Agreement.

AMBR Aggregate MBR.

AMC Adaptive Modulation and Coding.

ARP Allocation and Retention Priority.

ARQ Automatic Repeat Request.

ASN Access Service Network.

ASN-GW Access Service Network - Gateway.

ASP Application Service Provider.

BE Best Effort.

BPSK Binary Phase Shift Keying.

BS Base Station.

BTS Base Transceiver Station.

BWA Broadband Wireless Access.

C-RNTI Cell Radio- Network Temporary Identifier.

CAPEX Capital Expenditure.

CBR Constant Bit Ratio.

CID Connection Identifier.

CINR Carrier Interference-plus-Noise Ratio.

CoA Care of Address.

CPE Customer Permises Equipment.

CPS Common Part Sublayer.

CQI Channel Quality Indicator.

CS Convergence Sublayer.

CS-ACELP Conjugate-Structure Algebraic Code Excited Linear Prediction.

CSN Connectivity Service Network.

CSTD Cyclic Shift Transmit Diversity.

DCCP Datagram Congestion Control Protocol.

DL Downlink.

DSLAM Digital Subscriber Line Access Multiplexer.

DTC Distribution Transformer Controller.

DwPTS Downlink Part.

E-UTRAN Evolved UMTS Terrestrial Radio Access Network.

EAP Extensible Authentication Protocol.

EDGE Enhanced Data Rates for GSM Evolution.

eNB evolved Node B.

EPC Evolved Packet Core.

EPS Evolved Packet System.

ertPS Extended Real Time Polling Service.

FBSS Fast Base Station Switching.

FDD Frequency Division Duplexing.

FEC Forward Error Correction.

FFT Fast Fourier Transform.

FTP File Transfer Protocol.

GBR Guaranteed Bit Rate.

GERAN GSM EDGE Radio Access Network.

GP Guard Period.

GPRS General Packet Radio Service.

GPS Global Positioning System.

GSA Global mobile Suppliers Association.

GSM Global System for Mobile Communications.

GTP GPRS Tunneling Protocol.

GW Gateway.

HARQ Hybrid Automatic Repeat Request.

HHO Hard Handover.

HSPA High Speed Packet Access.

HSPA+ Evolved HSPA.

HSS Home Subscriber Server.

HTTP Hypertext Tramsfer Protocol.

IEEE Institute of Electrical and Electronics Engineers.

iLBC Internet Low Bit Rate Codec.

IMOS intermediary Mean Opinion Score.

IMS IP Multimedia Subsystem.

ISDN Integrated Services Digital Network.

KEK Key Encryption Key.

LAN Local Area Network.

LOS Line of Sight.

LTE Long Term Evolution.

MAC Medium Access Control.

MAN Metropolitan Area Network.

MBR Maximum Bit Rate.

MCS Modulation and Coding Scheme.

MDHO Macro Diversity Handover.

MIH Media Independent Handover.

MIMO Multiple Input Multiple Output.

MME Mobility Management Entity.

MOS Mean Opinion Score.

MS Mobile Station.

NAP Network Access Provider.

NAS Non Access Stratum.

NAT Network Address Translation.

NLOS Non Line of Sight.

Non-GBR Non-Guaranteed Bit Rate.

NRM Network Reference Model.

nrtPS Non-Real Time Polling Service.

NSP Network Service Provider.

NWG Network Working Group.

OFDM Orthogonal Frequency Division Multiplexing.

OFDMA Orthogonal Frequency Division Multiplexing Access.

OPEX Operational Expenditure.

OWD One Way Delay.

P-GW Packet Data Network Gateway.

PAPR Peak to Average Power Ratio.

PCC Policy and Charging Control.

PCRF Policy and Charging Resource Function.

PDA Personal Digital Assistant.

PDCP Packet Data Convergence Protocol.

PDH Plesiochronous Digital Hierarchy.

PER Packet Error Rate.

PHY Physical.

PKM Privacy Key Management.

PLMN Public Land Mobile Network.

PSNR Peak Signal Noise Ratio.

PSTN Public Switched Telephone Network.

QAM Quadrature Amplitude Modulation.

QCI QoS Class Identifier.

QoE Quality of Experience.

QoS Quality of Service.

QPSK Quadrature Phase Shift Keying.

RLC Radio Link Control.

RRC Radio Resource Control.

RSSI Received Signal Strength Indication.

RTP Real-time Transport Protocol.

rtPS Real Time Polling Service.

RTT Round Trip Time.

RTU Remote Terminal Unit.

S-GW Serving Gateway.

SAE System Architecture Evolution.

SAID Security Association Identity.

SAP Service Access Point interface.

SC-FDMA Single-carrier FDMA.

SDF Service Data Flows.

SDU Service Data Unit.

SFID Service Flow ID.

SGNS Service GPRS Support Node.

SINR Signal-to-Interference plus Noise Ratio.

SMS Short Message Service.

SNR Signal-to-Noise Ratio.

SOFDMA Scalable Orthogonal Frequency Division Multiplexing Access.

SS Subscriber Station.

STBC Space Time Block Coding.

TCP Transmission Control Protocol.

TD-CDMA Time Division Code Division Multiple Access.

TD-SCDMA Time Division Synchronous Code Division Multiple Access.

TDD Time Division Duplexing.

TEK Traffic Encryption Key.

TLS Transport Layer Security.

TTLS Tunneled Transport Layer Security.

UDP User Datagram Protocol.

UE User Equipment.

UGI Unsolicited Grant Interval.

UGS Unsolicited Grant Service.

UL Uplink.

UMTS Universal Mobile Telecommunications System.

UPI Unsolicited Polling Interval.

UpPTS Uplink Part.

USIM Universal Subscriber Identity Module.

UTRAN UMTS Terrestrial Radio Access Network.

VAD Voice Activity Detection.

VBR Variable Bit Ratio.

VoD Video on Demand.

VoIP Voice over IP.

VS Video Streaming.

WAP Wireless Application Protocol.

WCDMA Wide-Band Code Division Multiple Access.

WCI WiMAX 2 Collaboration Initiative.

WiMAX Worldwide Interoperability for Microwave Access.

1. Introduction

This work presents an overview of the Worldwide Interoperability for Microwave Access (WiMAX) and Long Term Evolution (LTE) technologies, through an insight analysis of the state of the art of both technologies, followed by several testbed and simulation tests emulating real multimedia applications (e.g., Voice over IP (VoIP)). These tests enabled an extensive evaluation and assessment of both technologies in different conditions and with several applications. Since this work is carried out within the EDP-WiMAX project, some of the tests performed are related to real EDP applications and deployment.

1.1 Motivation

The increasing use of Internet and mobile devices leads to the need for better access to the network, with higher throughput and availability. The current technologies, namely Global System for Mobile Communications (GSM), Universal Mobile Telecommunications System (UMTS) and High Speed Packet Access (HSPA), do not have the ability to support such developments, and therefore, the so-called 4G technologies have arisen, such as WiMAX and LTE.

It is therefore important to conduct an evaluation and assessment of these technologies, to observe the capabilities of supporting multiple users with different types of applications and environments.

In terms of the EDP-WiMAX project, the increased necessity for wireless network coverage in urban and rural scenarios with high availability, led EDP to start a project with a technology that would allow to test these scenarios, namely WiMAX, allowing an analysis of it behavior and capability to support the EDP specific applications.

1.2 Objectives

The main objective of this work is to evaluate the capabilities of the WiMAX technology in the support of multimedia and EDP specific applications, in both testbed and simulation scenarios. This evaluation is conducted through the analysis of Quality of Service (QoS) and Quality of Experience (QoE) parameters allowing an estimation of maximum supported users.

The work is complemented with the assessment of the LTE technology through simulation, allowing a comparison between both technologies.

In both semesters, the work has been performed in parallel on two parts. In the first semester the main objectives were:

- Evaluation of the WiMAX technology from an applicability point of view, testing real applications, such as VoIP. The assessment was conducted in a real testbed, through several tests emulating real applications traffic patterns.

- Deployment of real applications and equipment in the context of the EDP-WiMAX project, using the WiMAX testbed pilot. These applications are better described in Section 1.3.

In the second semester, the main objectives were:

- Finalization of the EDP-WiMAX project: several new tests, such as link monitoring, coverage assessment and traffic characterization, were conducted, as well as the execution of the final report, including the technology final evaluation and the main conclusions.

- WiMAX and LTE assessment: several new test scenarios were considered and the previous tests were complemented with new test conditions, namely VoIP tests using different WiMAX QoS mechanisms. Also, two different simulation studies were performed, comparing the real WiMAX testbed with a similar simulation scenario and comparing the LTE and WiMAX technologies through simulation.

1.3 EDP-WiMAX Project

This project aims to implement and deploy a WiMAX Network in the city of Coimbra, in order to test and evaluate its performance in the EDP context. This evaluation is conducted through a set of tests with the main goal of demonstrating the feasibility of the WiMAX Network and determining if it is interesting and capable of supporting the EDP needs, such as multi-users support and connectivity under Non Line of Sight (NLOS) and Line of Sight (LOS) conditions.

It should be noted that the EDP-WiMAX Project was intended to be completed in February 2012. However, due to some time constrains and delays in the project tasks, caused by the late arrival of the WiMAX equipment, various constraints in the conciliation of the EDP teams availability, as well as the need for adaptation and re-configuration of some applications, the project only finished in May 2012. Despite this delay, all the proposed tasks were successfully completed.

The applications involved in this project are:

- **Work Force Management and Mobile Teams Management (EDP: WFM / GME)**:

  This application consists of mobile nodes connecting to the network to access to EDP-specific applications for teams managament. The application traffic is generated by EDP servers at LOGICA[1] and transmitted to the nodes, located anywhere within the WiMAX network range. A good coverage and link avalaibility are needed to provide a good Quality of Service and Experience to the end-user. The equipment used consists of a tablet equipped with a USB Customer Permises Equipment (CPE).

- **Medium Voltage Remote Operations (EDP: Telecomando)**:

  This application consists of transmitting data over the network to control remote equipments. The application traffic is generated at Alto de S. João, in Coimbra, and it is transmitted from the EDP network to the WiMAX network in EDP - Rua do Brasil (interconnecting the EDP Plesiochronous Digital Hierarchy (PDH) network through a RS232 (Electronic Industries Association, 2011) to the WiMAX network through an *indoor* CPE). Thus, it is possible to communicate with both field Remote Terminal Unit (RTU)s, located in Quinta da Várzea and Vila Nova.

- **Transformer station interconnectivity for control, monitoring and telemetry (EDP: INOVGRID)**:

  This application consists of transmitting data over the network to control, monitoring and telemetry operations over the network. The application traffic is generated by LOGICA servers and transmitted to the field Distribution Transformer Controller (DTC)s, located in Quinta da Várzea and Estádio Universitário.

- **Transformer station interconnectivity for telemetry with legacy equipments (EDP: Telecontagem)**:

  This application consists of transmitting telemetry data from the EDP legacy equipments and clients. The application traffic is generated by EDP servers at LOGICA and transmitted to the field equipment, located in Quinta da Várzea and Estádio Universitário.

1.4 Contributions

The author has performed important contributions in the assessment of the Worldwide Interoperability for Microwave Access (WiMAX) and Long Term Evolution (LTE) technologies, as described below.

[1] Company that outsources some of the EDP servers and services.

The current state of the art of the WiMAX technology, based on the Institute of Electrical and Electronics Engineers (IEEE) 802.16 standard, is evaluated through a deep analysis and description, detailing some of the most important features of this technology. With the same level of detail and analysis, the state of the art of the 3rd Generation Partnership Project (3GPP) release 8, the LTE technology, is also described, enumerating the most important features. Based on the analyzed features, a comparison between WiMAX and LTE is performed, explaining the most relevant differences of each technology.

In the context of the EDP-WiMAX project, in which the author was involved, a Mobile WiMAX testbed was deployed in the city of Coimbra, consisting on the deployment and adaptation of several EDP applications and the WiMAX equipment. The author was involved in the testbed deployment, management and monitoring, assuring the interoperability with the EDP applications in a constant collaboration work with the EDP teams and personnel. The author obtained the *Alvarion System Certified Specialist* (2011-2012) certificate during this work, which allowed a better understanding of WiMAX hardware and management software.

An extensive analysis of the WiMAX technology in a real testbed environment was performed, evaluating the maximum number of supported users in different scenarios and with different applications. The first evaluation studies consisted in the variation of traffic patterns, number of simultaneous flows and line of sight conditions, in order to estimate and analyze the WiMAX network capabilities. After this overview evaluation, an analysis of Quality of Service (QoS) and Quality of Experience (QoE) parameters while sending Voice over IP (VoIP) data was performed. The VoIP evaluation was performed in two phases. The first phase, evaluated the impact of background traffic in the VoIP flows, in multi-user scenario and within different line of sight conditions. The second phase, analyzed different WiMAX QoS mechanisms (namely Real Time Polling Service (rtPS) and Best Effort (BE)), by sending VoIP data over different WiMAX channels with and without background traffic, in the same line of sight conditions as the first phase.

The assessment of VoIP applications provided an evaluation of the maximum number of supported users and the end-user perceived voice quality (QoE) in a real Mobile WiMAX testbed, which results were included in one work that was accepted for publication in an international conference. It is included in Appendix A - VoIP performance over Mobile WiMAX: An Urban Deployment Analysis

João Henriques, Vitor Bernardo, Paulo Simões, Marília Curado, "VoIP performance over Mobile WiMAX: An Urban Deployment Analysis", 2nd IEEE Baltic Conference on Future Internet Communications, 25-27 April 2012, Vilnius, Lithuania

Several availability and coverage tests were performed in the context of EDP-WiMAX project, as well as the EDP- applications specific tests and traffic patterns characterization. These tests, along with the VoIP tests performed and the WiMAX state of the art, were included in the EDP-WiMAX final report, included in Appendix B - EDP - WiMAX Pilot - Final Report.

The author performed an extensive assessment through simulation using both WiMAX and LTE Technologies. One of the contributions was an assessment comparing WiMAX performance on a testbed and by simulation in similar VoIP scenarios, which provided a comparison evaluation between real and simulation scenarios encompassing QoS and QoE parameters. In the same context, an analysis and comparison between WiMAX and LTE in multi-user VoIP scenarios was performed, assessing the capabilities of the maximum supported VoIP users by each technology, using different codecs and different QoS mechanisms, employing a QoE analysis. The assessment of WiMAX and LTE through simulation will be submitted to the Telecommunication Systems - Modeling, Analysis, Design and Management journal, with the title of "4G Technologies for VoIP: Quality of Experience Assessment".

1.5 Thesis Structure

In Chapter 2 the WiMAX technology and the IEEE 802.16 standard are detailed and explained. In Chapter 3 the LTE technology (release 8) is described, explaining some of the most important features of this technology. Further on Chapter 3, namely in Section 3.6, an overview of the most relevant differences and similarities between WiMAX and LTE is given, illustrating them through a comparison table. The relevant related work in the assessment of WiMAX and LTE, as well as the analysis of work methodologies, is explained in Chapter 4.

All the tests performed in the Mobile WiMAX testbed, including the preliminary and reference tests, and the assessment of VoIP scenarios are depicted in Chapter 5. Moreover, in Chapter 6, an evaluation of WiMAX via simulation is performed, allowing a comparison between the simulation and the testbed scenarios. Further in this chapter, a comparison between WiMAX and LTE technologies is performed via simulation, encompassing a multi-user VoIP scenario.

Chapter 7 gives an overview of the performed tasks during both semesters, accompanied with a description of each task. In Chapter 8, the most relevant conclusions of all the work performed are described.

2. WiMAX Technology

This chapter describes Worldwide Interoperability for Microwave Access (WiMAX), a Broadband Wireless Access (BWA) Technology based on the Institute of Electrical and Electronics Engineers (IEEE) 802.16 Standard. The first section describes WiMAX main features, while the following sections detail the WiMAX characteristics. Then, in Section 2.4 the WiMAX deployment evolution is demonstrated, and in the Section 2.5 the main conclusions of the WiMAX features are described.

2.1 Worldwide Interoperability for Microwave Access

WiMAX is an "all-IP" Broadband Wireless Access (BWA) Technology, based on the IEEE 802.16 standard, aiming to provide wireless access on Line of Sight and Non Line of Sight conditions at mobile, fixed and nomadic scenarios, allowing large areas of coverage while assuring link security and Quality of Service (QoS) parameters.

As referred, the WiMAX technology is based on the IEEE 802.16 standards, and the most relevant versions are the Fixed WiMAX, based on IEEE 802.16d (IEEE, 2004), and the Mobile WiMAX, defined by IEEE 802.16e (IEEE, 2005b). The latter is the main scope of this work, and it has significant improvements in the support of multiple users, as well as new mechanisms for Quality of Service (QoS), mobility support and also energy efficiency.

This technology is planned to achieve long ranges of coverage with high availability and throughput, allowing rural or urban wireless access in diverse deployment environments, namely with and without line of sight.

The QoS support is provided by different service classes, allowing the traffic flows differentiation by setting the applicable network parameters, namely the maximum and minimum reserved traffic rate, the maximum allowed delay, or the maximum tolerable jitter. Each QoS service class is designed and most suitable to different types of applications. Another relevant feature of Mobile WiMAX is the support of energy saving mechanisms, such as the idle and sleep modes. These energy efficiency mechanisms are particularly relevant in the context of mobile networks, since the majority of the connected devices are battery based. The WiMAX technology supports full mobility, allowing the access to the network at moving speeds, supporting vertical seamless handovers, where the users do not notice the attachment point change.

This section describes the Institute of Electrical and Electronics Engineers (IEEE) 802.16 standard and gives an overview of WiMAX main features.

2.1.1 IEEE 802.16 Standard

IEEE 802.16 is a standard that specifies the Medium Access Control (MAC) layer, Physical (PHY) layer and also the air interface of Broadband Wireless Access (BWA) Networks. The main purpose of the standard is to enable a rapid worldwide development and deployment of BWA technologies. The IEEE 802.16 Working group on Broadband Wireless Access Standards is responsible for the development of the IEEE 802.16 Standard for Wireless Metropolitan Area Networks and it is known as IEEE WirelessMAN®Standard (IEEE, 2011b).

The standard has been developed through successive revisions and amendments. The most important releases are described below:

- **IEEE 802.16d - 2004 - Fixed WiMAX**

  With the amendments and the improvements conducted since the first release of IEEE 802.16, the IEEE 802.16d-2004 (IEEE, 2004) standard was released with the following features:

  - Support of Non Line of Sight (NLOS) and Line of Sight (LOS) environments.

  - Low frequency bands (between 2-11GHz).

  - Orthogonal Frequency Division Multiplexing (OFDM), Orthogonal Frequency Division Multiplexing Access (OFDMA) and Frequency Division Duplexing (FDD). OFDM improves the multi-path by coding and interleaving the information across the different subcarriers. OFDMA is a flexible multiple-access technique which enables the use of more robust time and frequency domain scheduling algorithms. Also, OFDMA reduces the Peak to Average Power Ratio (PAPR) by splitting the entire bandwidth among the Mobile Station (MS), where each MS uses only a subset of subcarriers.

- **IEEE 802.16e - 2005 - Mobile WiMAX**

  Since the IEEE 802.16d was developed to work at fixed scenarios, the IEEE 802.16e (IEEE, 2005b) was later released with the following improvements:

  - Full Mobility Support.

  - Scalable Orthogonal Frequency Division Multiplexing Access (SOFDMA) access technology. SOFDMA allows different Fast Fourier Transform (FFT) sizes, scaling the FFT size to the channel bandwidth, keeping the sub-carrier frequency constant, reducing the system complexity to smaller channels and improving the performance of wider channels.

- **IEEE 802.16m - 2011 - WiMAX 2**

  With the evolution of mobile communications, the IEEE 802.16e standard was not enough to support the needs of these communications. The IEEE 802.16m standard (IEEE, 2011a), also known as *WiMAX 2*, was developed and released to overcome some of limitations of IEEE 802.16e. This standard supports the following improvements:

  - Higher throughput supported by multi-carrier aggregation.

  - Higher coverage supported by Advanced Antenna Systems (AAS) improvements.

  - Relay functions.

  However, this standard is not yet implemented and deployed, since it was released this year (2011). The IEEE 802.16m standard was released with the objective of improving the previous standard, allowing better coverage, throughput and improved mobility.

  The throughput improvement is achieved by Multicarrier aggregation and support, and Superframes. Superframes are frames comprising four radio frames of 5ms. The new multi-hop relay architecture and the advanced SU/MU-MIMO techniques guarantee a large coverage. The improved mobility is achieved by supporting Network Controlled and MS-Assisted handovers.

2.1.2 IEEE 802.16 Communication Layers

The IEEE 802.16 Standard specifies the air interface of BWA Networks, including the MAC and Physical (PHY) Layer. These layers are explained in the next subsections.

Figure 2.1 shows the PHY and MAC layers, and the interfaces between each sublayer.

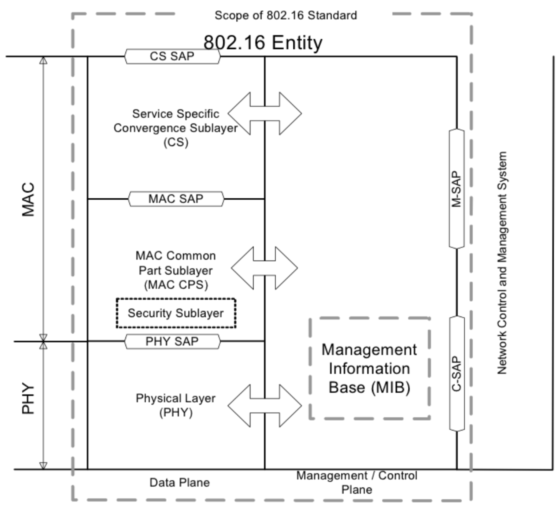


Figure 2.1: WiMAX Network Layers
Source: Fundamentals of WiMAX (Andrews et al., 2007)

Physical Layer

IEEE 802.16 Standard specifies different Physical layers that can be used with the MAC Layer, as described below:

- WirelessMAN-SC: This physical layer is part of the first 802.16 specification, and is a Single Carrier PHY for operating in LOS conditions, with frequencies beyond 11GHz. Both Time Division Duplexing (TDD) and FDD mechanisms are supported.

- WirelessMAN-SCa: This is a Single Carrier PHY Layer operating in licensed bands, between 2GHz and 11GHz, under NLOS conditions.

- WirelessMAN-OFDM: This PHY Layer introduces point-to-multipoint operations under NLOS conditions and operating frequencies between 2GHz and 11GHz. Also, some mechanisms were improved, such as Forward Error Correction (FEC) and Modulation Schemes.

  It introduces the OFDM multi-carrier modulation with a FFT size of 256, meaning that each OFDM symbol is made of 256 sub-carriers. This PHY layer was finalized in IEEE 802.16d-2004 (IEEE, 2004) specifications.

- WirelessMAN-OFDMA: This PHY layer is based on OFDM, and it allowed a FFT size of 2048 for point-to-multipoint operations under NLOS conditions at frequencies between 2GHz and 11GHz. This layer was modified in IEEE 802.16e-2005 IEEE (2005b) to SOFDMA, in order to support better results in fixed/mobile scenarios, by allowing different and variable FFT sizes, such as 128, 512, 1024 and 2048.

- WirelessHUMAN: This layer, High-speed Unlicensed Metropolitan Area Network, was designed to work under license-except bands, between 5GHz and 5GHz.

The Physical layer is very important in radio communications, since it has direct impact on the radio transmission efficiency. The PHY layer has a set of functions and states, as shown in Figure 2.2.

The first states include channel encoding, symbol mapper and space time coder. The channel encoding represents a set of encoding and error correction functions, such as Randomization, Forward Error Correction and Interleaving. The symbol mapper represents the modulations of the bits to symbols. The Space Time Coder is responsible for mapping the data into the correct subcarriers and the construction of OFDM/OFDMA symbols in the frequency domain.

The last step is responsible for converting the symbols from the frequency domain to the time domain or to the analog signal, so it can be transmitted over the air.

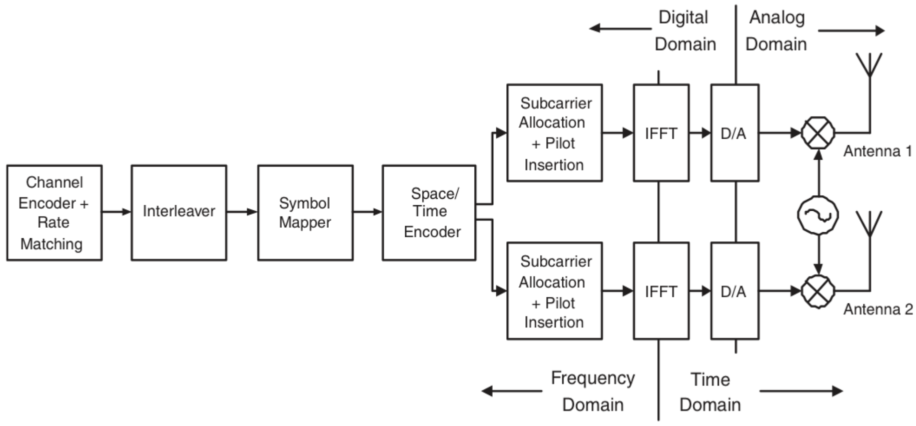


Figure 2.2: WiMAX Physical Layer States
Source: Fundamentals of WiMAX (Andrews et al., 2007)

MAC Layer

The IEEE 802.16 Standard defines the PHY and MAC layers. The MAC layer is a crucial layer, since it supports the most important features of WiMAX, such as mobility and power management, QoS assurance and security mechanisms. This layer consists of three sublayers: The Convergence Sublayer, the Common Part Sublayer and the Security Sublayer.

The *Convergence Sublayer (CS)* is responsible for mapping external data received from the higher layers, through the CS Service Access Point interface (SAP) interface, into MAC Service Data Unit (SDU)s, associating and classifying the external data to the proper MAC Service Flow ID and Connection ID to be sent to the Common Part Sublayer (CPS) through the MAC SAP interface. Also, the CS layer is responsible for performing higher layer protocol operations, such as payload header suppression and compression and address mapping.

The *Common Part Sublayer (CPS)* is responsible for the control of the medium and it performs all packet operations that are independent from the upper layers. This layer is responsible for most of the QoS support, such as connection establishment, scheduling functions, power saving and mobility functions.

The *Security Sublayer* is responsible for assuring all the security mechanisms necessary to provide a secure connection, such as authorization, secure key exchange and data encryption. This layer also uses the Privacy Key Management (PKM) Protocol (IEEE, 2001).

2.1.3 WiMAX Features Overview

This section describes some of the main features of Worldwide Interoperability for Microwave Access (WiMAX) Technology, giving an overview of which mechanisms support those features. These mechanisms are described and detailed in the further sections.

Throughput

One of the most promising WiMAX features is the throughputs achieved. WiMAX supports throughputs up to 63Mbps on the downlink and 28Mbps on the uplink, assuming a 10MHz bandwidth channel with TDD frames and with 64QAM5/6 as modulation scheme (Andrews et al., 2007). Some of the mechanisms that support this feature are:

- Multiple Input Multiple Output (MIMO)(B) antenna techniques - Two antennas sending and two antennas receiving, increasing the throughput, up to twice of the bandwidth capacity from single antenna techniques.
- Adaptive Modulation and Coding (AMC) - based on Hybrid Automatic Repeat Request (HARQ) and Channel Quality Indicator (CQI) - AMC allows an adaptive modulation scheme according to the link conditions.

Coverage

The WiMAX network coverage can reach up to 30 miles, supporting mobile, fixed and nomadic nodes in rural and urban environments and allowing the access to the network at NLOS and LOS conditions (Andrews et al., 2007). This coverage is supported by the following mechanisms:

- MIMO(A) antenna techniques - Two antennas sending and receiving the same data, allowing a more efficient signal spreading.
- Space Time Block Coding (STBC) - Provides better transmit diversity by sending different symbols in different antennas for the data stream.
- Cyclic Shift Transmit Diversity (CSTD) - Also provides better transmit diversity by sending the same OFDM symbol in each antenna but with different circularly shifted versions (circularly shifted by time domain).
- Beamforming - This mechanism focuses the transmitted beam in the direction of the Subscriber Station (SS), by weighting the signal between the two antennas. This provides better Received Signal Strength Indication (RSSI) and Signal-to-Noise Ratio (SNR), by avoiding interferences, and also improving the coverage.
- OFDMA - This mechanism provides better multi-path performance in NLOS scenarios, by concentrating more Transmission Power(TX Power) only on certain sub-channels.

Mobility

The IEEE 802.16e-2005 standard supports full mobility, which means it can support users moving from one point to another without loosing the connection, through seamless handover processes.

WiMAX supports Access Service Network (ASN)-Anchored mobility and Connectivity Service Network (CSN)-Anchored mobility. The first means changing from one Base Station (BS) to another within the same ASN, being invisible to the CSN. The CSN-Anchored mobility is referred to mobility across different ASNs (multiple foreign agents), also meaning *roaming*. This feature is described in section 2.2.3.

Quality of Service

The IEEE 802.16e standard also defines one of the most important features of this technology, the QoS support. For that, five different classes of service are defined, each one suited to different applications:

- Unsolicited Grant Service (UGS) - Interactive and real time applications, such as Voice over IP (VoIP).
- Extended Real Time Polling Service (ertPS) - Interactive and real time applications, and VoIP with Voice Activity Detection (VAD).
- Real Time Polling Service (rtPS) - Streaming applications, such as Video on Demand (VoD).
- Non-Real Time Polling Service (nrtPS) - Reliable file transfers, such as File Transfer Protocol (FTP).
- Best Effort (BE) - General traffic, such as Hypertext Tramsfer Protocol (HTTP).

These mechanisms are explained in section 2.2.1.

Energy Efficiency

One of the biggest concerns in wireless systems and mobile devices is the battery life and the power efficiency. The IEEE 802.16 Standard defines two different power saving modes, the sleep and the idle mode. Moreover it has specific PHY layer mechanisms to improve the power efficiency. These features are better described in section 2.2.2.

Security

Security is one of the most important features in wireless systems. The security normally involves three main objectives, which are authentication, authorization and encryption. These functions are assured by the PKM Protocol. The security mechanisms are explained in section 2.2.4.

2.2 Insight on WiMAX Features

In this section some of WiMAX features previously referred are described and explained, such as Quality of Service (QoS), Security and Power Saving mechanisms, and Mobility support.

2.2.1 Quality of Service Assurance

One of the most important features of WiMAX Technology is the Quality of Service (QoS) support. This feature is mostly supported by the WiMAX Medium Access Control (MAC) layer through different service classes, where each one defines a set of network parameters and different uplink scheduling mechanisms.

The MS can have one or more Service Flow, each one with a different set of QoS parameters. A Service Flow is an unidirectional flow, identified by Service Flow ID (SFID) with one connection associated, identified by the Connection Identifier (CID). Each connection is specified with different QoS parameters and different Service Classes. The different Service Classes with the corresponding uplink scheduling and QoS parameters are described below, as defined in the IEEE 802.16e standard.

- **Unsolicited Grant Service (UGS):**

  UGS is most suited for real-time and interactive applications with fixed sized packets at constant periodic intervals, such as Voice over IP (VoIP) traffic, with Constant Bit Ratio (CBR). UGS defines the following QoS parameters:

  - Reserved Traffic Rate

  - Maximum Tolerated Latency

  - Maximum Tolerated Jitter

  The uplink scheduling for UGS is the Unsolicited Grant Interval (UGI), as shown in Figure 2.3. It uses fixed size transmission opportunities at regular time intervals without the need of requests or polls. This service class maintains the requested bandwidth in use, even if there is no traffic, which is very costly in terms of air resources. The

Extended Real Time Polling Service (ertPS) was developed also to overcome this limitation, as described below.

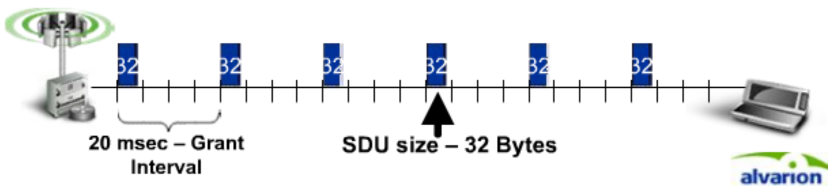


Figure 2.3: Uplink scheduling for UGS - UGI
Source: Alvarion (Alvarion, 2011)

- **Extended Real Time Polling Service (ertPS):**

  ertPS is most suited for real-time and interactive applications with variable sized packets at constant periodic intervals, such as VoIP with Silence suppression. The ertPS overcomes the UGS reserved traffic rate limitation. ertPS defines the following QoS parameters:

  - Minimum Reserved Traffic Rate
  - Maximum Sustained Traffic Rate
  - Maximum Tolerated Latency
  - Maximum Tolerated Jitter

  The uplink scheduling for ertPS is the UGI. But here, in accordance with the variable downlink data sized packets, it uses uplink dynamic bandwidth allocation at regular time intervals, also without the need of requests or polls.

- **Real Time Polling Service (rtPS):**

  rtPS is most suited for real-time applications which are not sensible to delay variations (jitter), with variable sized packets at periodic intervals, as shown in Figure 2.4, such as Video on Demand (VoD) and video streaming.

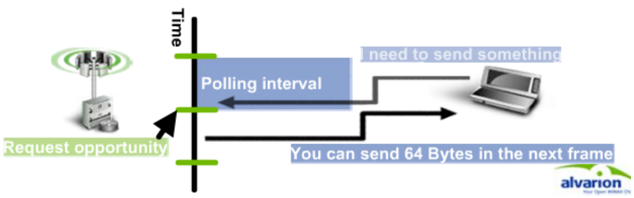


Figure 2.4: Uplink scheduling for rtPS - UPI
Source: Alvarion (Alvarion, 2011)

rtPS defines the following QoS parameters:

- Minimum Reserved Traffic Rate

- Maximum Sustained Traffic Rate

- Maximum Tolerated Latency

The uplink scheduling for rtPS is the Unsolicited Polling Interval (UPI). The Base Station (BS) offers periodic request opportunities to the Mobile Station (MS), and the MS indicates the necessary bandwidth for reservation.

- **Non Real Time Polling Service (nrtPS):**

  nrtPS is most suited for delay tolerant applications which the only requisite is the bandwidth, such as File Transfer Protocol (FTP), allowing reliable file transfers. nrtPS defines the following QoS parameters:

  - Minimum Reserved Traffic Rate

  - Maximum Sustained Traffic Rate

  For the uplink scheduling, the BS regularly sends unicast request opportunities and the MS indicates the necessary bandwidth (Contention Request Opportunities (CRO)). The BS then responds indicating when the MS shall communicate. This mechanism is shown in Figure 2.5.

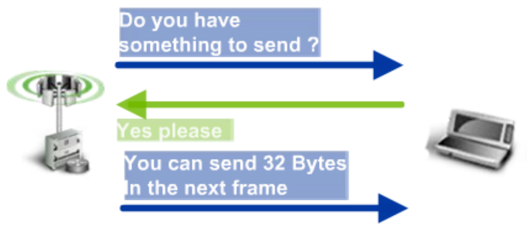


Figure 2.5: Uplink scheduling for nrtPS - CRO
Source: Alvarion (Alvarion, 2011)

- **Best Effort (BE):**

  Best effort offers no QoS guarantees and is typically used for HTTP traffic. BE defines only the Maximum Sustained Rate.

  For the uplink scheduling, the BS sends unicast request opportunities and the MS indicates the necessary bandwidth. The BS then responds indicating when the MS shall communicate, but it does not guarantee the requested bandwidth.

2.2.2 Power Saving Mechanisms

The development of power efficient mechanisms is an important goal in all wireless technologies. This goal can be attained through different approaches, such as: Efficient Power

Amplifiers, PHY layer improvements and MAC layer improvements. The IEEE 802.16e Standard improves the power management by introducing new power saving mechanisms at PHY and MAC layers, as described below.

PHY Layer improvements

WiMAX technology provides PHY and MAC layer mechanisms in order to improve the MSs power efficiency. Some of the mechanisms of the PHY layer that allow power efficiency are: OFDMA and uplink sub-channelization, ranging channel and Channel Quality Indicator (CQI), and the Adaptive Modulation and Coding (AMC).

By using only a subset of all sub-channels and splitting the bandwidth among the MSs, each MS can concentrate more transmission power on those sub-channels, allowing it to reduce the Peak to Average Power Ratio (PAPR). Also, accompanying these mechanisms, ODFMA allows a better Signal-to-Noise Ratio (SNR), because the MS knows the exact time and frequency that it shall communicate with the BS.

The Ranging channel allows periodic adjustments for power control with the help of Channel Quality Indicator (CQI), which indicates to the BS some MS signal parameters, such as Received Signal Strength Indication (RSSI) and Signal-to-Interference plus Noise Ratio (SINR). With Channel Quality Indicator (CQI) information the BS can adjust the modulation scheme and transmission power needed. The adjustment of the modulation scheme is handled by the Adaptive Modulation and Coding (AMC), which allows the node to vary the modulation scheme in order to maintain the best connectivity parameters.

Usually, more transmission power represents higher rates and better SNR, but there can be a tradeoff between throughput, distance and battery usage. Lowering the Transmission Power (TX Power) can improve the battery usage, but it will decrease the distance allowed between the MS and the BS and it will also decrease the link quality.

MAC Layer improvements

At the MAC Layer, WiMAX provides two power efficiency mechanisms: Sleep and Idle Mode.

Sleep Mode

The Sleep Mode is a mechanism where the MS remains registered with the BS, performing normal handoff procedures. While in sleep mode, the MS scans neighbor Base Stations in order to collect information to facilitate the handoff procedure.

In sleep mode, each connection (CID) can be disrupted individual. Since each MS can have several connections, each one has its own sleep time. The period when all the connections are in sleep mode is called the *unavailable interval*, as shown in Figure 2.6.

During the *unavailable interval* the BS does not schedule any DL transmissions to the BS. For Unicast Requests, the BS buffers the arriving Service Data Unit (SDU)s until the MS becomes available. For multicast requests, the BS buffers all the requests until the available time common to all MSs in the multicast group.

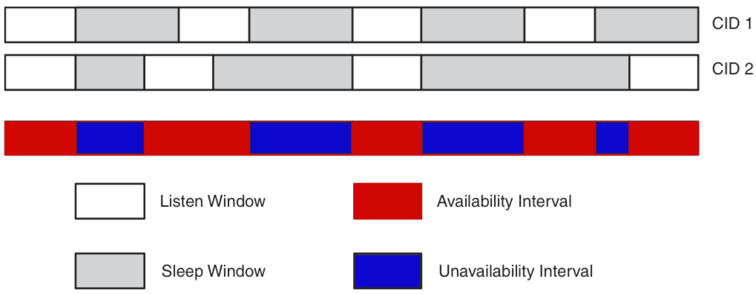


Figure 2.6: Sleep Mode - Available and unavailable periods
Source: Fundamentals of WiMAX (Andrews et al., 2007)

The sleep mode is represented by a Sleep and a Listen Window, negotiated between the MS and the BS, based on the Power Saving Class, also defined by the BS. There are three different power saving classes, as described below.

- **Power Saving Class 1**:

  In this power saving class, the BS indicates to the MS the listen window size, the initial and the final sleep window size.

  The listen window has a fixed length and it is followed by a sleep window, which size will be twice the size of the previous sleep window until it reaches the final sleep window size. At that moment, the sleep window will maintain the sleep window size until the sleep mode is reset.

  If the MS listen window is not sufficient to receive all traffic, the BS can reset the window sizes. This power saving class is most appropriated for Best Effort traffic and Non-real-Time traffic.

- **Power Saving Class 2**:

  The power saving class 2 defines a fixed length for listen and sleep window sizes, and it is most appropriated for Unsolicited Grant Service (UGS), typically VoIP constant bit rate.

- **Power Saving Class 3**:

  Power saving class 3 defines a sleep start time and a sleep window size, consisting of a single sleep window. After the sleep window, the power saving becomes inactive.

This power saving class is most appropriated for multicast traffic and MAC management traffic, since the BS may guess when next data will appear, and it can allocate sleep window for periods without data. The BS can decide to reinitiate power saving operation.

Idle Mode

The idle mode is a different power management mechanism. It allows the MS to be deregistered from the BSs and to be completely turned off for certain periods. This mechanism allows greater power-savings to the MS and also benefits the network and the BS by avoiding unnecessary handover traffic from inactive MSs and without the need of performing hand-off related procedures.

The idle mode works through Paging Groups (PG) which are formed by several BSs. The MS constantly monitors the DL network transmissions to determine its current location, so it can perform a paging group update when changing to a different PG, in order to inform the network about its new location and PG.

With paging groups, when downlink traffic is pending, the network only pages the BS belonging to the MSs Paging Group (PG). Paging is the method used to alert idle MSs, and avoids paging all the network because it only pages one Paging Group. Therefore, the paging area should be large enough to avoid frequent Paging Group Updates, and small enough to avoid the overhead of sending data to a large number of BSs. Paging groups are illustrated in Figure 2.7.

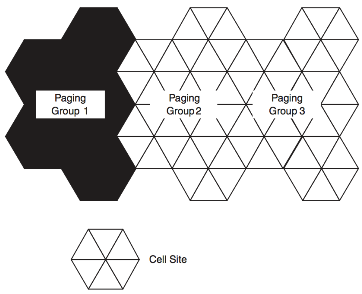


Figure 2.7: Idle Mode - Paging Groups
Source: Fundamentals of WiMAX (Andrews et al., 2007)

There are three different entities at paging groups, as represented at Figure 2.8:

- Paging Agent (PA):

  The paging agent represents a BS functional entity and it handles the iteration with the Paging Controller (PC). One or more PAs form a Paging group and a PA can belong to one or more Paging Groups.

- Paging Controller (PC):

  The paging controller is responsible for controlling the activity of idle MSs on the network. Each PC is associated to a Location Register.

- Location Register (LR):

  The Location Register is a distributed database with information about idle MSs, and it is updated with the MSs Location Updates.

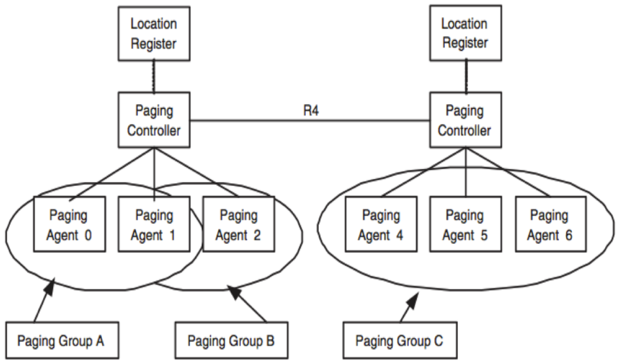


Figure 2.8: Idle Mode - Paging Groups Entities
Source: Fundamentals of WiMAX, (Andrews et al., 2007)

During Idle Mode the MS can be in *Paging Unavailable Interval* or *Paging Listen Interval*. During the *Paging Unavailable Interval* the MS is not available for paging and it can power down. The MS only has to scan and conduct periodical ranging with neighboring BSs.

During the *Paging Listen Interval* the MS listens to DL MAP messages from the BS to determine when the broadcast paging message is scheduled. If the MS is paged in broadcast paging message it responds and terminates the idle mode. Otherwise, if the MS is not paged, it will enter the next paging unavailable interval.

Through these PHY and MAC Layer mechanisms WiMAX can increase its power efficiency. This efficiency does not always represent less battery usage, but it means that the power is used more efficiently by delivering better rates, coverage and link quality and availability.

2.2.3 Mobility Support

The IEEE 802.16e Standard, also known as *Mobile WiMAX*, introduces new mechanisms for mobility support. The WiMAX technology supports full mobility, allowing the access to the network at moving speeds, supporting vertical seamless handovers, where the users do not notice the attachment point change. Three different handover modes are specified by the standard:

- **Hard Handover (HHO)** - This is the only mandatory handover process. Here, the MS abruptly disconnects from one base station and connects to another BS previously scanned. The decision of this change is made by the BS.
- **Fast Base Station Switching (FBSS)** - In this method, the MS maintains a valid connection with two or more BSs, belonging to an *active set*. Although the MS only communicates with one BS, called anchor BS, it continuously monitors the active set of BSs, doing periodic raging. When a change of BS is required, the connection is switched without disrupting the connection (which happens in HHO).
- **Macro Diversity Handover (MDHO)** - This mechanism also keeps a set of active BSs, as in FBSS, but instead of communicating with a single BS, the MS communicates with multiple BSs, called *Diversity Set*, receiving and sending the same data from and to different BSs.

Although FBSS and MDHO modes are superior, allowing a better Quality of Service to the MS, they require the BSs to be synchronized and share information, and also to be in the same carrier frequency.

Also, two types of mobility are considered by the mobile management, addressed by the WiMAX Forum Network Working Group (NWG):

- **Access Service Network (ASN)-Anchored Mobility** - This type of mobility represent the mobility within the same BS without the need of changing the Care of Address (CoA).
- **Connectivity Service Network (CSN)-Anchored Mobility** - This type of mobility is referred to mobility across different ASNs, involving communication with different providers. The CSN-Anchored Mobility is also referred as Roaming.

2.2.4 WiMAX Security

Security is one of the biggest concerns in wireless communication systems. The security involves three main steps: authentication, authorization and encryption. In authentication and authorization, it must be assured that only the legitimate users access the network. Accompanying these steps, the confidentiality and privacy of data transmission must be assured by data encryption methods.

All security mechanisms are at the Security Sublayer, handled by the Privacy Key Management (PKM) Protocol. Figure 2.9 illustrates the Security Sublayer.

In WiMAX, the BS acts as a bridge between the client and the network Authentication, Authorization and Accounting (AAA) server. At the network entry process, the MS sends it

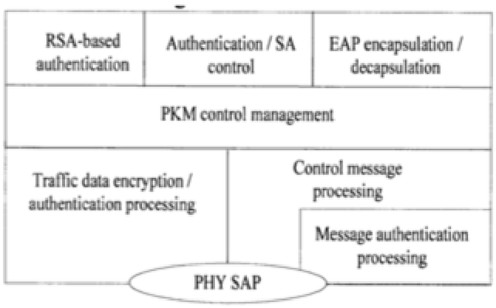


Figure 2.9: WiMAX Security Sublayer
Source: Fundamentals of WiMAX, (Andrews et al., 2007)

credentials to the BS, which will then forward the request to the AAA Network Server. This server contains the MS security informations and it will respond to the BS, informing if the MS is, or not, allowed to access to the network.

Authentication

WiMAX supports three different authentication methods:

RSA Based - X.509

In this method the user is identified by its unique X.509 digital certificate. This certificate, which is issued by the MS manufacturer, contains the MS public key and MAC Address, and it is sent from the MS to the BS, which is also used to generate the authorization key.

EAP

In Extensible Authentication Protocol (EAP) (IETF, 2004a) the client is identified by an unique operator issued credential, which can be a X.509 digital certificate, a SIM card or a user/password. To support these credentials, there are three different EAP types:

- **EAP - Authentication and Key Agreement (AKA) (IETF, 2006)** - For SIM based authentication;
- **EAP - Transport Layer Security (TLS) (IETF, 2008b)** - For X.509 digital certificate authentication;
- **EAP - Tunneled Transport Layer Security (TTLS) (IETF, 2008a)** - For SS-CHAPv2 (user/password) based authentication;

RSA based and EAP

This method combines the RSA-based (client digital certificates) method and the EAP (operators issued credential).

Authorization

The Authorization process consists on the verification of the MS by the BS/network, by requesting the Authorization Key (AK) and the Security Association Identity (SAID). Figure 2.10 illustrates this procedure.

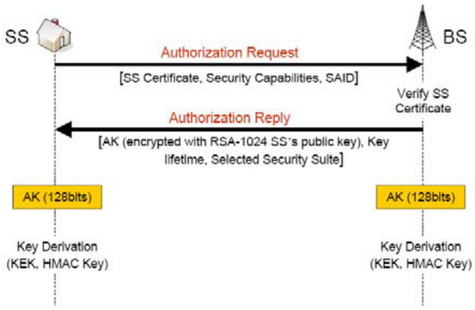


Figure 2.10: Authorization Key Request process
Source: (Prasad, 2009)

The first step is the Authorization Request message sent by the MS, containing the MS credentials and the security capabilities supported, such as encryption algorithms and cryptography ID. Next, the BS verifies the MS credentials by interacting with an AAA server in the network, which can be RADIUS (IETF, 2000) / DIAMETER (IETF, 2003). When the credentials are validated, the BS responds with an Authorization Reply, containing the requested AK (encrypted with the MS public key) and the SAID. This process is illustrated by Figure 2.10.

Encryption

The encryption is obtained through the Traffic Encryption Key (TEK) request and it will be used to encrypt all traffic between the MS and the BS. Figure 2.11 illustrates this procedure.

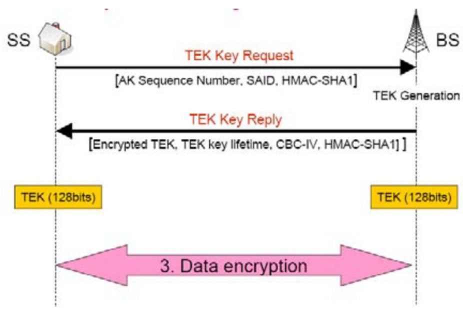


Figure 2.11: Traffic Encryption Key Request process
Source: (Prasad, 2009)

The process is started by the MS by requiring the TEK from the BS, through a TEK

Key Request. The TEK is then generated by the BS TEK encryption algorithm, using the Key Encryption Key (KEK). This KEK is a key derived from the Authorization Key, which is previously generated at the authorization process. The BS then responds to the MS with a TEK Key Reply, containing the encrypted TEK and the TEK lifetime. This process is demonstrated at Figure 2.11.

2.3 Network Architecture

In this section the WiMAX architecture is described, referring the WiMAX Reference Model defined by WiMAX Forum, which is also described in the next section.

2.3.1 WiMAX Forum and WiMAX Collaborative Initiative

WiMAX Forum Forum (2011) is a non-profit organization that promotes the WiMAX Technology, with the objective of accelerating the deployment and expansion of this Broadband Wireless Access (BWA) technology, based on IEEE 802.16. Also, the WiMAX forum certifies the WiMAX products in order to promote the interoperability and compatibility between vendors and equipments.

The WiMAX 2 Collaboration Initiative (WCI) is an initiative between the WiMAX vendors, suppliers and research organizations with the objective of accelerate the WiMAX 2 deployment and interoperability.

This group works directly with the WiMAX Forum in order to collaborate with the technology testing and performance benchmarking, accelerate the interoperability tests between WiMAX 2 products and also testing Fourth Generation of mobile phone standards and technology (4G) applications over WiMAX 2 systems.

2.3.2 Network Reference Model

In order to enable the interoperability between vendors and manufactures and to create higher layer specifications for WiMAX systems beyond what is defined the IEEE 802.16 Standard, the WiMAX Forum's Network Working Group (NWG) was formed, (ForumTM, 2006). An End-to-end All-IP network architecture was defined, represented by the Network Reference Model (NRM).

The NRM is a logical representation of the network architecture, including the Logical Entities and the Network Reference Points that interconnect those entities, in order to provide interconnectivity and interoperability. The NRM is illustrated at Figure 2.12, followed by an overview of each reference point functions.

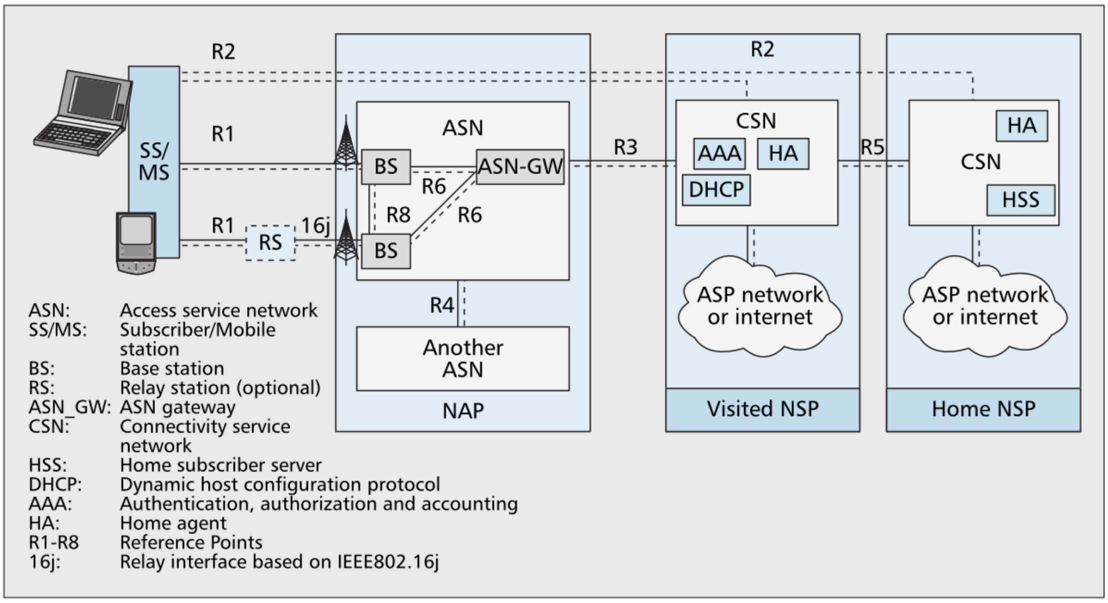


Figure 2.12: WiMAX Forum Network Reference Model
Source: (Etemad, 2008)

WiMAX Entities

WiMAX has several layers comprising Business, Functional and Logical entities, as described below.

Business Entities

There are three Business entities.

- **Network Access Provider (NAP)** - This entity is responsible for providing the radio access infrastructure to one or more Network Service Provider (NSP). NAP is constituted by one or more Access Service Network (ASN).
- **NSP** - This entity is responsible for providing the connectivity and services to the WiMAX subscribers.
- **Application Service Provider (ASP)** - This business entity is responsible for providing the applications and services via NSP.

  Usually, the NAP, the NSP and the ASP are the same business company.

Functional Entities

There are two functional entities, the Connectivity Service Network (CSN) and the ASN:

- **ASN** - This functional entity has a set of network functions for providing radio access to the WiMAX subscribers. These functions include: provide WiMAX layer-2 connectivity to the subscribers; provide Authentication, Authorization and Accounting (AAA) services and transfer these messages between the subscribers and the Home Network Service Provider; network discovery and relay functions to establish layer 3 connectivity with the mobile station; also, the ASN must support ASN and CSN Anchored mobility.
- **CSN** - This functional entity has the main function of providing IP connectivity to the WiMAX subscribers. It has AAA server/proxy allowing the clients to access the network based on policy and admission control (authorization, authentication and accounting). Also, it must support ASN-CSN and inter-CSN tunneling (roaming).

Logical Entities

There are three logical entities:

- **Base Station (BS)** - The base station provides radio access to the network for the mobile stations. It also has scheduling functions for uplink and downlink.
- **Mobile Station (MS)** - The mobile station represents the Customer Permises Equipment (CPE) equipment.
- **Access Service Network - Gateway (ASN-GW)** - This logical entity provides routing and bridging functions between the base stations and the ASN and between the ASN and the CSN.

WiMAX NRM Reference Points

As demonstrated by Network Reference Model, at Figure 2.12, some reference points are defined, as described below.

- **R1** - Reference point 1 consists of the protocols and procedures between MS and the ASN. It also may include additional protocols related to management plane.
- **R2** - This reference point includes the protocols and procedures between the MS and the CSN, such as AAA authentication and host IP configuration. This is a logical reference point and does not represent a direct protocol interface between MS and CSN.
- **R3** - Reference point 3 consists of Bearer and Control Plane procedures between ASN and CSN to support AAA, mobility management and policy enforcement features. Also, this reference point includes bearer plane procedures, like data tunneling between the ASN and CSN to transfer user data.

- **R4** - This reference point consists of the bearer and control plane procedures between ASNs and ASN-Gateways in order to coordinate MS mobility.
- **R5** - Reference point 5 consists of the bearer and the control plane procedures between the Home NSP and the Visited NSP for supporting roaming capabilities.
- **R6** - This reference point represents the bearer and control plane communications between the BS and the ASN-Gateway (GW).
- **R8** - Reference point 8 consists of the communication between Base Stations in order to provide seamless handovers.

2.3.3 Network Entry Process

In this section the Network Entry Process for the IEEE 802.16e is described.

Downlink and Uplink scan/Synchronization

The first step is the Downlink and Uplink synchronization. In this step the MS scans for channels from a pre-defined list or performs a full band scan. After the PHY level synchronization, which is obtained by reading the Frame Control Header of the Preamble, the MS tries to listen for DL and UL-Map messages in order to complete the synchronization (MAC level), by obtaining information about ranging opportunities.

Initial Ranging

In this step, the MS tries to send ranging request messages to the BS in order to perform the initial ranging, so it can obtain the relative timing and adjust the power levels to communicate with the BS.

This step also includes the allocation of the basic and primary management connections to the MS. After this initial ranging, the MS should do periodic ranging to maintain the connection with the BS, by adjusting the track timing and the power-level. This periodic ranging is needed because of the fluctuations that can arise, caused by factors such as mobility and multi-path.

Capabilities Negotiation

After obtaining the UL/DL synchronization, the MS must announce its capabilities to the BS. These capabilities announcements include parameters such as the Modulation levels and coding schemes supported, rates, and duplexing modes.

Authentication and Authorization

The authentication and authorization includes several steps, such as the exchange of authorization and Traffic Encryption Key (TEK), which are better explained in section 2.2.4.

These procedures are controlled and assured by the Privacy Key Management (PKM) Protocol.

Registration

After successfully authentication and authorization and the exchange of the TEK, the MS is registered at the network and it receives the secondary connections.

Setup IP Connectivity and establish Service Flows

After the successfully registration, the MS can perform a request to obtain IP address and also establish the provisioned service flows. The establishment of service flows can be initiated either by the BS or the MS.

2.4 Deployments

Since the first release of WiMAX, it has had an increase in the deployments and usage, as noted in Figure 2.13.

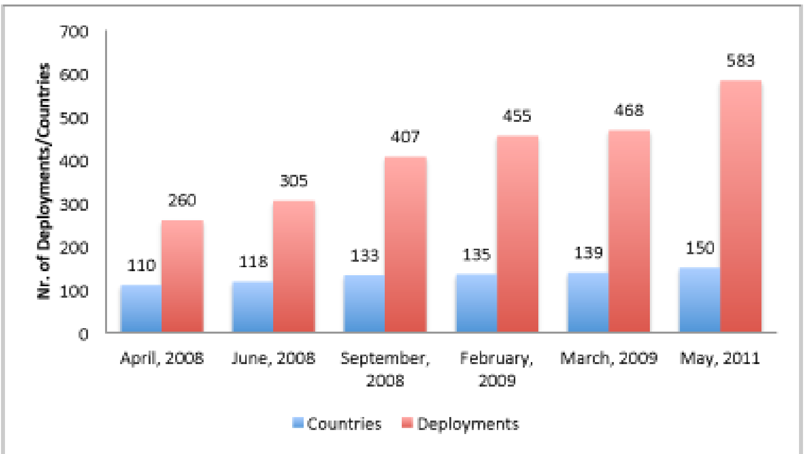


Figure 2.13: WiMAX Deployments Evolution (May 2011)
Source: WiMAX Forum

Observing the data from May 2011 (latest informations), the WiMAX was deployed over 150 countries, with 583 deployments. As shown in Figure 2.14, WiMAX has a large number of deployments in Latin and Central America, Africa, Asia-Pacific and Eastern Europe.

Although WiMAX has competitive characteristics, most service providers that have previously chosen WiMAX, have abandoned the technology in favor of LTE for their next-generation mobile broadband deployments. Since most of the operators must guarantee backward compatibility with the 3GPP technologies, such as GSM and GPRS, the LTE was the main choice. Also, comparing LTE to Mobile WiMAX, the first allows higher data rates in

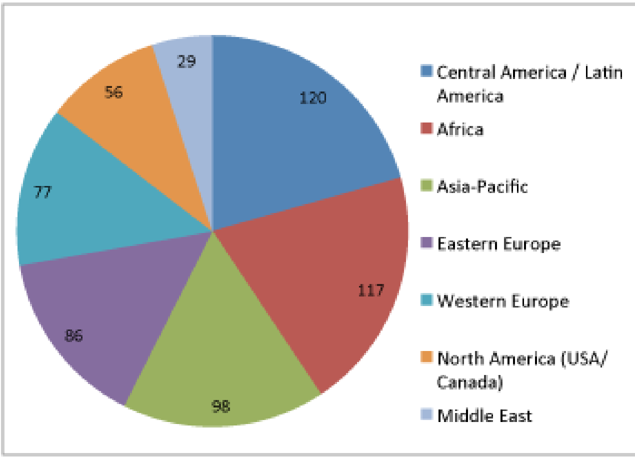


Figure 2.14: WiMAX Deployments (May 2011)
Source: WiMAX Forum

urban environments, which represents a major commercial advantage (Dahlman et al., 2011) (Andrews et al., 2007). WiMAX is being repositioned for vertical applications and other niche markets, such as industrial markets and smart grids, and also, last mile broadband access on rural environments.

Some good examples of WiMAX deployments are the Portuguese municipalities pilots. Working as Metropolitan Area Network (MAN) and last mile broadband, there are at least 3 examples:

- Municipal WiMAX Network at Águeda - Águeda municipality has installed a WiMAX pilot with the objective of providing free internet access to its residents. The WiMAX version installed is Mobile WiMAX (IEEE 802.16e), operating in 2.6GHz frequency band and with bandwidths of 10MHz.
- Municipal WiMAX Network at S. João da Madeira - With the objective of providing wireless access network to its residents, the city has implemented a WiMAX network in town. This network operates at 3.5GHz frequency band, along a WiFi-Mesh network in 2.4GHz band.
- Municipal WiMAX Network at Abrantes - To allow the internet access to the merchants from the historical city center, the Abrantes municipality installed a WiMAX network. The network details are not published.

2.5 Summary

In summary, the WiMAX technology has various technological advantages. The *All-IP* network provides better service and hardware integration. It supports native Quality of Service

(QoS) assurance, which has mechanisms to differentiate the clients and applications, providing the indicated QoS parameters to each application/user. Also, it supports seamless handovers and full mobility at moving speeds.

WiMAX proves to be a good solution for *Last Mile Broadband*, providing high speed internet access to remote areas, where there is not either fiber optics or good Asymmetric Digital Subscriber Line (ADSL) connection. Also, WiMAX can be a good alternative for new operators that do not need to ensure the interoperability with legacy or 3G/GSM equipments. Other possible implementation of a WiMAX network, is the Smart Grid / Industrial sector. The WiMAX Technology provides high rates and high coverage with low Operational Expenditure (OPEX) and relatively low Capital Expenditure (CAPEX), allowing a company to have its own network, without the need of third party contracts (Sweeney, 2006).

The integration of the author in a pilot project for a real WiMAX testbed (i.e., EDP-WiMAX Project) allowed the acquisition of a deeper knowledge of this technology, as well as the understanding and perception of a real pilot maintenance. Also, the achievement of the *Alvarion Certified System Specialist Certificate* (2011-2012) contributed to such deep understanding and learning.

3. LTE Technology

This chapter describes the Long Term Evolution (LTE) technology and the 3rd Generation Partnership Project (3GPP) access networks evolution, and explains some of the most relevant LTE features.

3.1 3GPP Long Term Evolution (LTE)

Since 1992, the Global System for Mobile Communications (GSM) networks launch date, the wireless communication systems have suffered a large increase of users, as consequence of the growth of these type of networks. After the GSM appearance and the success of wireless voice calls, the General Packet Radio Service (GPRS) has emerged in 2001. Previously in 2000, the 3GPP was formed with the main responsibility of developing the future 3GPP standards.

The GPRS offered better transfer rates and propelled the Internet services appearance, such as Wireless Application Protocol (WAP), Short Message Service (SMS) and e-mail access. However, because of the GPRS limited data rate and cell establishment times, the *internet services* never became popular. The Enhanced Data Rates for GSM Evolution (EDGE) was later released, representing an evolution of GPRS, with higher data rates, a result obtained from the different modulation schemes used.

The year 2005 marks the raise of the well know Third Generation of mobile phone standards and technology (3G), such as Universal Mobile Telecommunications System (UMTS) (3GPP release 99). The first UMTS release supported data rates up to 384 Kb/s, allowing Internet accesses with similar speeds than the fixed networks of that time, such as Asymmetric Digital Subscriber Line (ADSL). The UMTS aimed to provide higher data rates, seamless mobility and interoperability with GSM and GPRS networks. UMTS used Wide-Band Code Division Multiple Access (WCDMA) and High Speed Packet Access (HSPA) protocols, and, through several releases the UMTS data rates were improved. In release 4 the data rates were improved to about 2 Mb/s, in release 5 these were improved to 14.4 Mb/s, and, in release 7, it the data rates were improved to 28.8 Mb/s (Sauter, 2009). The HSPA, used in UMTS communications, has evolved through the several 3GPP releases, and, the release 7, known as Evolved HSPA (HSPA+), offered peak data rates up to 42 Mb/s. This release also changed the system architecture, in order to provide a smooth transition to LTE.

The large increasing of mobile users led to new challenges and targets of the novel mobile access networks. The main drivers for the LTE development were a reduced latency, higher data rates to the users, higher coverage and improved system capacity, Capital Expenditure (CAPEX) and Operational Expenditure (OPEX) reduction.

LTE is an evolved 3GPP technology - Evolved UMTS Terrestrial Radio Access Network (E-UTRAN), namely release 8. It is developed to support higher number of users with higher data rates and coverage, while maintaining backward compatibility with other legacy 3GPP access networks and equipment, such as UMTS Terrestrial Radio Access Network (UTRAN) and GSM EDGE Radio Access Network (GERAN).

3.1.1 3rd Generation Partnership Project

3GPP is a partnership project formed by standards-developing organizations from all the world. It is responsible for preparing, approve and maintain technical specifications and reports for the 3rd generation and beyond Mobile systems, based on the evolved 3GPP core networks, in order to transpose those specifications into the Standards.

3.1.2 LTE Features Overview

LTE technology is designed to operate in paired and unpaired spectrum allocations, allowing system bandwidths from 1.4 to 20 MHz. In paired bands, the uplink and downlink are conducted in different frequency bands, while in unpaired bands, the transmissions share the same frequency band. LTE supports two duplex schemes, Time Division Duplexing (TDD) and Frequency Division Duplexing (FDD):

- In the TDD scheme, there is only one carrier frequency, where the uplink and downlink are separated in time. Seven Uplink(UL)/Downlink(DL) configurations are supported, allowing different UL/DL ratios by splitting the time (sub-frames) for the uplink and downlink. This feature is better explained in the TDD description, section 3.1.3.
- The FDD duplexing scheme is based on the usage of two different frequencies, one for the uplink and one for the downlink, where the data transmission can be done simultaneously.

The Orthogonal Frequency Division Multiplexing (OFDM) is the multiplexing technique employed for the downlink, since it is more robust to time dispersion on the radio channel and it allows a simplified baseband processing on the receiver side, a lower power consumption and reduced terminal cost. In the uplink, the multiplexing scheme is the Single-carrier FDMA (SC-FDMA), which allows a lower Peak to Average Power Ratio (PAPR) at the node, and consequently higher power efficiency.

Each sub-frame has duration of 1 ms, composed by 12 or 14 OFDM symbols, depending on the cyclic prefix used. The radio frame is formed by 10 sub-frames, which allows small delays for the user and control data, such as the Hybrid Automatic Repeat Request (HARQ) and the channel quality feedback from the nodes, providing feedback to the transmitter at each sub-frame. Also, the sub-frames short time duration allows fast channel variations by the scheduler, located in the base station, deciding which users shall transmit and which data rate can be used. Therefore, the scheduler has an important role in the QoS assurance, especially in highly loaded networks.

Several modulation schemes are supported in the transmissions, namely, Quadrature Phase Shift Keying (QPSK), 16 and 64 Quadrature Amplitude Modulation (QAM), dynamically adapted to each node depending on channel conditions. Each node also informs the base station about the receiving signal quality, through *Sounding Reference Signals*.

Two layers of retransmission schemes are supported, HARQ located in the Medium Access Control (MAC) layer and Automatic Repeat Request (ARQ) in the Radio Link Control (RLC) layer. These multi-layer schemes allow a fast hybrid retransmission protocol with low overhead feedback.

LTE supports Advanced Antenna Systems (AAS) such as Multiple Input Multiple Output (MIMO) technology, allowing higher data rates, better signal quality (Signal-to-Noise Ratio (SNR)) and better coverage. Three modes of downlink MIMO are available:

- Transmit Diversity: The transmit diversity consists of sending the same data stream through all the transmit antennas. This mechanism allows a higher SNR at the receiver side an also a better coverage to the nodes.
- Spatial Multiplexing: In spatial multiplexing different streams are sent over each antenna, and the receiver will receive the data also in different antennas. This mechanism allows higher data rates, although it introduces some limitations, since the signal can reach the node by different paths causing problems in the path correlation procedure.
- Beamforming: Beamforming consists of forming a beam in the direction of the receiver, allowing a better coverage and signal quality (SNR).

For the uplink, the LTE provides two antenna techniques:

- Uplink transmit antenna selection, where the User Equipment (UE) can send data with both antennas (although this support by the UE it is optional).
- Multi-user MIMO/Collaborative MIMO: Two UEs are allowed to transmit on the same time/frequency resource.

3.1.3 TDD Specification

The TDD duplexing scheme is a time division scheme, which shares the channel with several users, multiplexing the channel in time and frequency. The frames are divided for the uplink and downlink, containing special sub-frames, composed by the Downlink Part (DwPTS), Uplink Part (UpPTS) and Guard Period (GP). The frames are configured according to the desirable Uplink/Downlink ratio, with seven different configurations, adapting the network configuration to different scenarios.

The Downlink Part (DwPTS) is commonly used for downlink data transmission. The Uplink Part (UpPTS) is used for the transmission of sounding-reference signals and for random access. The Guard Period (GP) is used to allow the different nodes and base stations to switch from transmitting to receiving with no overlap with the signals to be transmitted or received. The Guard Period (GP) must be large enough to handle the propagation delays of the cells, leaving time for the node to start the transmission only when it already ended the data reception, and small enough to avoid high delays in the communications.

The special sub-frames also provide a solution to the backward compatibility issue, with the Time Division Synchronous Code Division Multiple Access (TD-SCDMA) and Time Division Code Division Multiple Access (TD-CDMA) - both used in UMTS TDD, which have lower sub-frame time durations. By selecting a proper DwPTS, UpPTS and GP, it is possible to achieve a switch-point alignment between the different radio-access schemes.

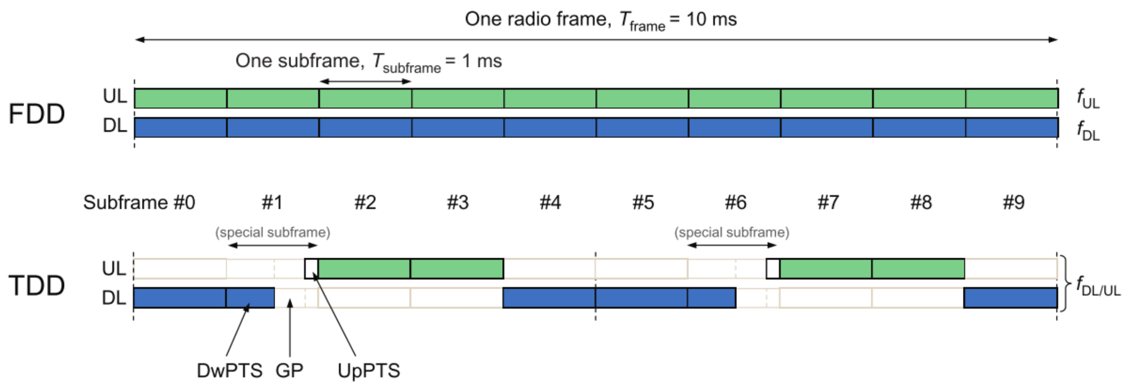


Figure 3.1: LTE Frames
Source: 4G LTE/LTE Advanced (Dahlman et al., 2011)

Figure 3.1 demonstrates the FDD and TDD frames, where one can observe the difference between FDD and TDD and also the DwPTS, GP and UpPTS.

As the TDD frames are divided into uplink and downlink (allowing different ratios), there are several configurations allowed, as shown in Figure 3.2.

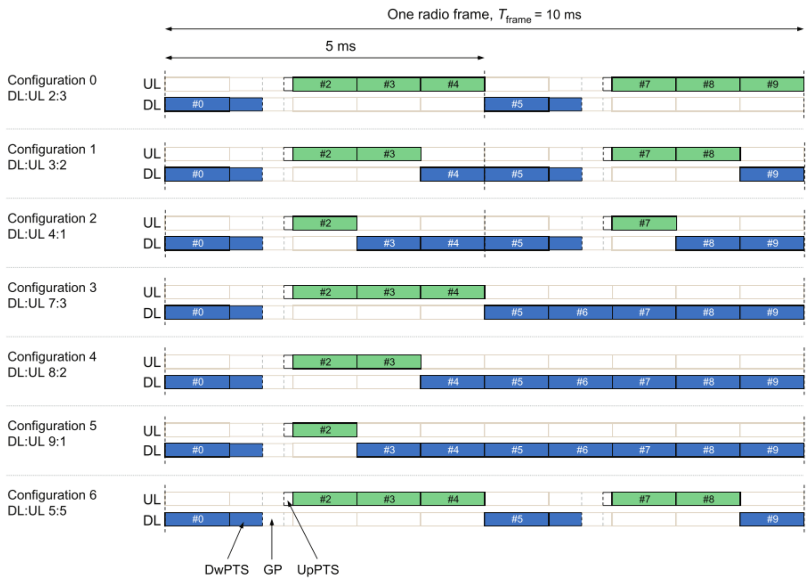


Figure 3.2: LTE TDD Frames
Source: 4G LTE/LTE Advanced (Dahlman et al., 2011)

3.2 Insight on LTE Features

This section gives an insight on the main LTE features.

3.2.1 Quality of Service Assurance

In previous 3rd Generation Partnership Project (3GPP) releases, such as Global System for Mobile Communications (GSM) and Universal Mobile Telecommunications System (UMTS), it was possible to define dedicated channels with pre-defined allocated resources. In 3GPP release 8, Long Term Evolution (LTE), these dedicated channels are not possible. Since there are only two channels in LTE, uplink and downlink, the QoS must be handled with scheduler mechanisms. The scheduling process is handled by the evolved Node B (eNB).

In LTE the Quality of Service (QoS) support is based on *bearers*. A *bearer* is a packet flow established between the Packet Data Network Gateway (P-GW) and the User Equipment (UE), having different QoS attributes. When the UE connects to the network, a default bearer is assigned, allowing the node to be authenticated and provided with IP connectivity. Additional bearers will be set up on demand, depending on the QoS nodes policy. The request for bearers can be initiated by the UE or the eNB, but they must be authorized by the Policy and Charging Resource Function (PCRF).

Two types of Bearers are available, Guaranteed Bit Rate (GBR) and Non-Guaranteed Bit Rate (Non-GBR). In guaranteed bit rate bearer (GBR), the network parameters related to the bearer definition are permanently allocated when the bearer becomes established. This allocation allows the GBR to be more resistant to network congestion conducting to lower packet losses. Moreover, the Non-GBR does not guarantee bit rates and it is more susceptible to the factors previously mentioned, such as network congestion and packet loss. This kind of bearer is also referred as default bearer, which is used to establish the IP connectivity to the nodes. For each bearer, a number of Service Data Flows (SDF) are mapped, receiving the same QoS treatment.

The bearers are assigned with a QoS Class Identifier (QCI), represented by a scalar value, defining a set of packet forwarding treatments, such as scheduling weights, queue management thresholds and link layer protocol configuration. Also, other QoS attributes are set to each bearer, as described below:

- Allocation and Retention Priority (ARP): The ARP is used in the admission control, to decided whether to accept or reject the bearer establishments and modifications. Also, the ARP is used to decide which bearer is released in overload situations.
- Maximum Bit Rate (MBR) and GBR: Valid for GBR bearers, the MBR defines the maximum bit rate allowed, that can not be exceeded, and the Guaranteed Bit Rate defines the minimum reserved bit rate.
- Aggregate MBR (AMBR): Since several IP flows can be mapped to the same bearer, this parameter indicates the maximum bit rate allowed to a UE for all the bearers in the same P-GW connection.

The QoS parameters for CQI as described in Table 3.3:

QCI	Resource type	Priority	Delay budget	Loss rate	Example application
1	GBR	2	100 ms	1e-2	VoIP
2	GBR	4	150 ms	1e-3	Video call
3	GBR	5	300 ms	1e-6	Streaming
4	GBR	3	50 ms	1e-3	Real time gaming
5	Non-GBR	1	100 ms	1e-6	IMS signalling
6	Non-GBR	7	100 ms	1e-3	Interactive gaming
7	Non-GBR	6	300 ms	1e-6	Application with TCP:
8	Non-GBR	8			browsing, email, file
9	Non-GBR	9			download, etc.

Figure 3.3: LTE QCI Table
Source: LTE for UMTS (Holma and Toskala, 2009)

The resource type indicates the bearer class associated, GBR or Non-GBR. The priority field defines the packet priority, to be used by the eNB scheduling of radio interface. The Delay Budget indicates which is the maximum delay allowed, that the scheduler must take into account when scheduling the packets to the nodes. The loss rate defines the maximum loss rate allowed.

3.2.2 Power Saving

Since the majority of the connected UEs are mobile devices, the power saving mechanisms are very important and object of research. The first step of LTE in order to provide power efficiency was the choice of Single-carrier FDMA (SC-FDMA) as the uplink-multiplexing theme. The SC-FDMA allows a lower Peak to Average Power Ratio (PAPR) from the UE, and so, it allows lower power consumption when sending data. Also, through idle mode and paging, the LTE can save resources in the UEs. The idle mode time and paging group is obtained from the network, as well as the wake up paging periods (paging cycles), where the UE is informed if there is data to be received or not.

3.2.3 Mobility Support

One of the main targets of LTE is to support seamless and low delay handovers, allowing the users to remain connected to the network while moving between access points (i.e. eNB). Also, since LTE was designed to allow backward compatibility, it supports mobility between different 3GPP access networks.

There are two types of mobility in LTE, the idle mode and connected mode. In idle mode, the handover decisions are made by the UE, re-selecting the cells based on pre-determined thresholds, defined by the network. The connected mode represents the handover while transmitting data to the network, and it is controlled by the network.

Idle mode

By constantly monitoring its reception quality, the UE chooses the cell that most suits the re-selection criteria. After receiving the broadcast channels of the target cell, the UE knows if that cell is suitable for camping (i.e. register on that network). This cell selection process can be affected by the Public Land Mobile Network (PLMN) priority value, which prioritizes the networks to which the UE will connect, representing the preferred networks. These values can be stored in the Universal Subscriber Identity Module (USIM) card.

After selecting and camping in a suitable cell, the UE must register itself in the network. If the UE performs this registration from a different tracking area, a location registration must be conducted in order to update the bearers and connections of the UE.

The tracking area consists of several cells, to which the UEs belong, that are paged when some communication to the belonging UEs arrives. The tracking areas must be large enough to avoid constant location registrations, and small enough to avoid the overload of paging to a large number of cells.

Connected mode

The connected mode handover procedure between eNBs consists of four main phases. When the eNB receives the channel feedback information from the UE, it decides whether the UE shall change to another eNB. The decision is made accordingly to the UE radio parameters and to the defined signal quality thresholds at the eNB. If the handover procedure is chosen, the source eNB preforms one handover request with the target eNB. In case of acceptance, the source eNB establishes a GPRS Tunneling Protocol (GTP) tunnel with the target eNB, through the X2 interface, forwarding all the incoming downlink packets to the target eNB. At this point, the UE has already established the UL connections with the target eNB.

After this resource allocation is ready, the source eNB instructs the UE to proceed with the handover to the target eNB, and the Late Path switching is performed. Late path switching consists of changing the network bearers path to the target eNB, by informing the network (i.e. the MME and the Serving Gateway (S-GW)) about the handover. The last step consists of releasing the source eNB resources, which happens when the lane path switching is concluded.

This process is illustrated in Figure 3.4.

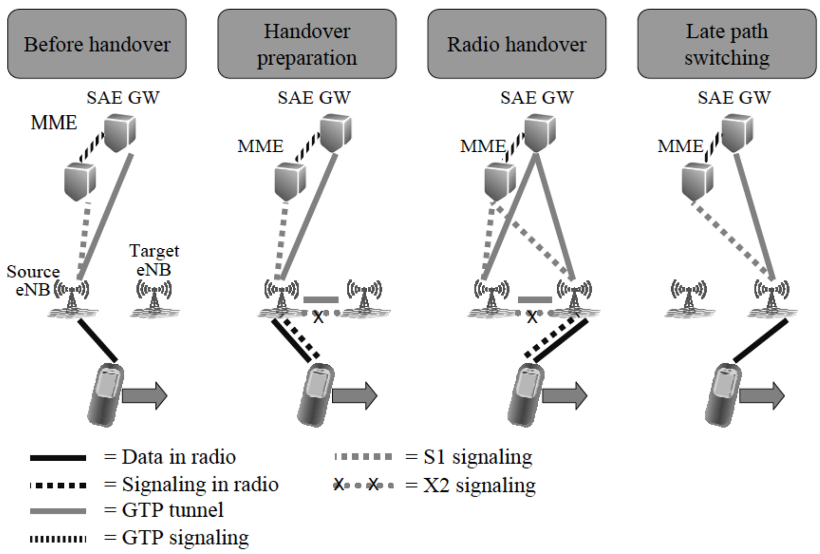


Figure 3.4: LTE Handover Procedure
Source: LTE for UMTS (Holma and Toskala, 2009)

Since the Packet Data Convergence Protocol (PDCP) layer is responsible for integrity control, it ensures that the packets are delivered in sequence and without duplicates during the handover process.

The handover decisions for other 3GPP systems, represented in Figure 3.5, are handled by the source access system. The GSM EDGE Radio Access Network (GERAN) and UMTS

Terrestrial Radio Access Network (UTRAN) Access Networks, are connected to the Evolved Packet Core (EPC) through two components: the Service GPRS Support Node (SGNS) which acts as intermediary between the cells and the Mobility Management Entity (MME), and through the S-GW, which acts as mobility anchor.

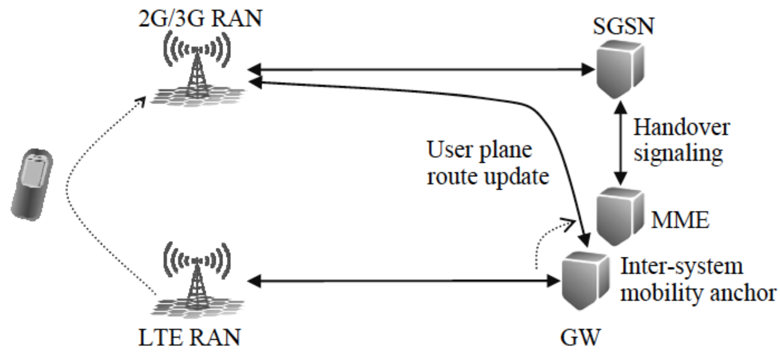


Figure 3.5: LTE Intra Handover
Source: LTE for UMTS (Holma and Toskala, 2009)

3.2.4 Security

Security is one of the biggest concerns of wireless technologies. Security not only involves the authentication, but also authorization, integrity, confidentiality and non-repudiation of the messages. To accomplish these objectives, the network must provide strong and reliable security mechanisms. In LTE, the main requirement to the security is to be at the same level, or better, than the UMTS, and the usage USIM must be continued.

The network security procedures are initiated when the UE firstly contacts the eNB, to perform the UE authentication. The eNB will then contact the MME, and the latter will handle the interaction with the Home Subscriber Server (HSS), Serving Gateway (S-GW) and evolved Node B (eNB) in order to perform the UE authentication and authorization.

The protocol layer in charge of these communications is NAS, which is part of the MME protocol stack. Below this protocol is the Radio Resource Control (RRC), also with an active role in the authentication procedure, by handling the communication with the Packet Data Convergence Protocol (PDCP). The PDCP layer is below the RRC and above the Radio Link Control (RLC), and it will handle the upper layers data protection and confidentiality.

3.3 Network Architecture

This section describes the main characteristics of LTE network architecture, as well as some important procedures, such as, network entry.

3.3.1 LTE System Architecture Evolution (SAE)

Figure 3.6 illustrates the LTE System Architecture Evolution (SAE).

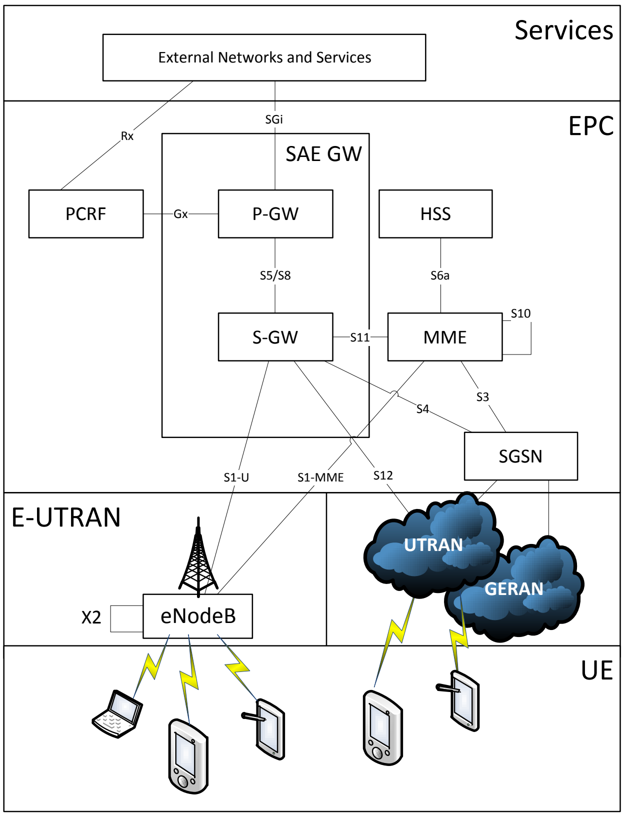


Figure 3.6: LTE SAE Architecture
Adapted from LTE for UMTS (Holma and Toskala, 2009)

The IP connectivity layer in Long Term Evolution (LTE) System Architecture Evolution (SAE) is represented by the User Equipment (UE), Evolved UMTS Terrestrial Radio Access Network (E-UTRAN) and Evolved Packet Core (EPC), and it is known as the Evolved Packet System (EPS). This network architecture is the result of 3rd Generation Partnership Project (3GPP) standardization process in order to achieve a more simple and effective network, being compatible with older 3GPP releases.

In the upper layer, along the External Networks, is the service layer. This layer provides services to several networks, such as the IP Multimedia Subsystem (IMS), which allows the interconnectivity of the legacy circuit switched networks, such as Public Switched Telephone Network (PSTN) and Integrated Services Digital Network (ISDN), through the Media Gateways.

As depicted in figure 3.6, in order to provide interconnectivity between 3GPP access networks, the GSM EDGE Radio Access Network (GERAN) and UMTS Terrestrial Radio Access Network (UTRAN) networks are connected to Service GPRS Support Node (SGNS) and to the Serving Gateway (S-GW).

The S-GW has a very similar role as the Mobility Management Entity (MME), by controlling the interconnectivity between the nodes of the different access networks. For that, it is connected to the MME through the S3 interface and to the S-GW through the S4 interface.

The EPS components are described in the next sub-sections.

User Equipment (UE)

The User Equipment represents, as the name indicates, the equipment used by the users to connect to the network, containing the Universal Subscriber Identity Module (Universal Subscriber Identity Module (USIM)), used to identify and authenticate the user in the network. This equipment is connected to the network through the evolved Node B (eNB).

Evolved Packet Core (EPC)

The EPC consists of different entities, such as the MME, Packet Data Network Gateway (P-GW) and S-GW. Also, the EPC contains the Policy and Charging Resource Function (PCRF) and the Home Subscriber Server (HSS).

Mobility Management Entity (MME)

The MME is a crucial component in the EPC, and its main functions are:

- Mobility Management: When a UE connects to the network, the MME registers the node and its location in the HSS, assigning the correct resources with the HSS information and selecting the S-GW, which will serve the node. Also, the MME will keep tracking the UEs, managing the idle states and handover processes.
- Authentication and Security: By accessing to the Home Subscriber Server (HSS), the Mobility Management Entity (MME) authorizes and identifies the UE that wants to connect to the network. The MME assigns a default bearer to the node, in order to provide basic IP connectivity, and is responsible for the coordination of the authentication procedures. The authentication is conducted in the first time that the UE registers in the network and also periodically.

One Mobility Management Entity (MME) can support multiple UEs, although each User Equipment (UE) is only connected to one MME.

As mentioned, the MME is connected to the HSS through the S6 interface, providing Authentication and Security, Location Management and access to the user profiles. Through the S11 interface, the MME controls the User Plane tunnels. The control connection with the eNB is made through the S1-MME to provide the coordination functions the referred. The MMEs are interconnected through the S10 interface to execute the handovers between them and to manage the eNBs idle states. The connection to the SGNS is used to perform interconnectivity between 3GPP access networks

Packet Data Network Gateway (P-GW)

The P-GW, referred as Packet Data Network Gateway, is the highest level of mobility anchor, and it connects the network to external networks and services. P-GW is responsible for assigning an IP address to the UEs and mapping the packets to the correct GTP tunnels (representing bearers) to the S-GW, through S5/S8 interface. This functions also represent forwarding and filtering packets based on policies, which are obtained through the Gx interface from the PCRF. The information about policy and charging control as well as PCC rules are also obtained from the PCRF. The bearers are created by the P-GW, based on the PCRF information or from S-GW (which relays the MME information), meaning that when a S-GW handover occurs, the bearers must be modified in the P-GW.

Serving Gateway (S-GW)

The main function of the serving gateway is to act as a local mobility anchor during the mobility between eNB, and to provide tunneling management and switching. This component provides the resources allocation based on the MME, P-GW and PCRF decisions, and tunnels the data between the UEs and the Packed Data Network Gateway (P-GW). This component will also be responsible for law related interceptions, providing data to the authorities. The S-GW is connected to the MMEs through S11 and to the P-GW through S5/S8, receiving control messages and establishing the GPRS Tunneling Protocol (GTP) tunnels and IP data flows. It is also connected to other S-GWs to forward DL data during handovers, when inter-eNB connection is unavailable. The connection to SGSN and UTRAN is used to perform interconnectivity between 3GPP Access Networks.

Policy and Charging Resource Function (PCRF)

The PCRF is responsible for the decisions about the QoS assurance and policies, by transmitting Policy and Charging Control (PCC) information to the P-GW about the authorized bearers to be created. The PCRF is connected to the external networks through Rx interface, to provide the PCC rules when a roaming request occurs, and to the P-GW through the Gx interface, in order to provide QoS and policies information for the bearers setup.

Home Subscriber Server (HSS)

The HSS stores the registered UE profiles, keys and location, and also information about roaming authorizations. Basically, the HSS stores all the user information that is needed and transmitted to the MME, to provide mobility and interconnectivity between 3GPP access networks. The HSS is connected to the MMEs through the S6a interface. In order to provide mobility between MMEs, the HSS needs to be able to access to all MMEs in the network.

3.3.2 Evolved UMTS Terrestrial Radio Access Network (E-UTRAN)

The E-UTRAN is constituted by the several evolved Node B (eNB)s of the network, meaning that the eNBs are responsible for most of the procedures and functions of radio access. This component provides radio access to the network, which the users are connected to, acting as a layer 2 bridge between the clients (User Equipment (UE)) and the Evolved Packet Core (EPC).

The eNB protocol stack consists of:

- Physical Layer
- Medium Access Control Layer
- Radio Link Control Layer
- Packet Data Convergence Protocol Layer
- Radio Resource Control Layer

The main eNB functions are:

- Radio Resource Management, such as scheduling functions and QoS assurance, traffic prioritization and radio resources monitoring.
- Mobility Management: eNB constantly receives information from the UEs and communicates with the MMEs and other eNBs in order to exchange handover signaling messages. After a handover, eNB is responsible for setting the correct routes between the UE and the new MME.
- Radio Bearer control: By acting as a bridge between the UE and the MME, it transfers the bearer control messages, in order to provide service data flows (bearers) to the UE.

As mentioned, the eNB acts as a layer 2 bridge between the UE and the EPC. eNB is connected to the MMEs through the S1-MME interface, to provide Mobility management, bearer handling and security settings, and connected to the S-GW through the S1-U interface, in order to tunnel the data between the UE and EPC to provide UL/DL data delivery. The eNBs are also connected to each other through the X2 interface, in order to prepare handover situations and to forward packets between them to execute the handover process.

3.3.3 Protocol Stack

This section gives an overview of user plane and control plane protocol stacks.

User Plane

The user plane protocol stack is represented between the UE and the eNB. This protocol stack is represented below.

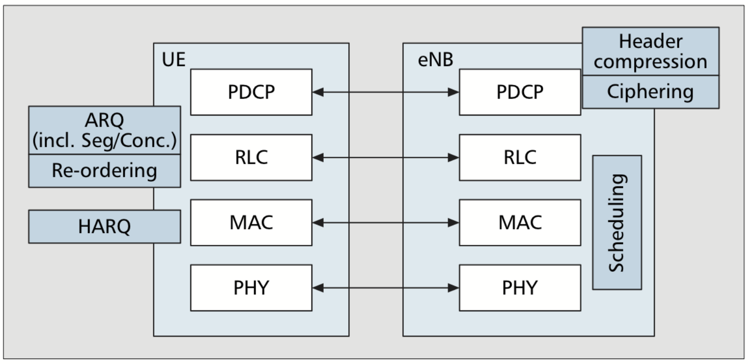


Figure 3.7: LTE User Plane Protocol Stack
Source: LTE Link Layer Design (Larmo et al., 2009)

Three sub-layers constitute the LTE link layer design:

- Packet Data Convergence Protocol (PDCP): The PDCP is responsible for several functions, such as:

  - Header compression, such as compression of large IP packet headers to improve the efficiency of the transmission;

  - Encryption and integrity control, such as in-sequence deliver upper layer Service Data Unit (SDU) and elimination of lower layer SDUs, being most relevant during handover procedures, where the data transmission is more unstable.

- Radio Link Control (RLC): The RLC layer performs tasks such as error correction with Automatic Repeat Request (ARQ), data sequence delivery and duplicate detection, and data segmentation and concatenation.

- Medium Access Control (MAC): MAC layer is responsible for mapping between the logical and transport channels, packet priority and QoS scheduling, multiplexing and de-multiplexing data streams, and it also processes Hybrid Automatic Repeat Request (HARQ) protocol acknowledge ACK messages.

The physical layer is the lower layer, acting as a bit pipe, providing data reliability, security and integrity, protected by turbo-coding and Cyclic Redundancy Check (CRC).

Control Plane

The user plane and control plane protocol structures are similar, although the control plane main functions are related to control procedures, such as connection setup, security and mobility. Figure 3.8 illustrates the control plane protocol stack.

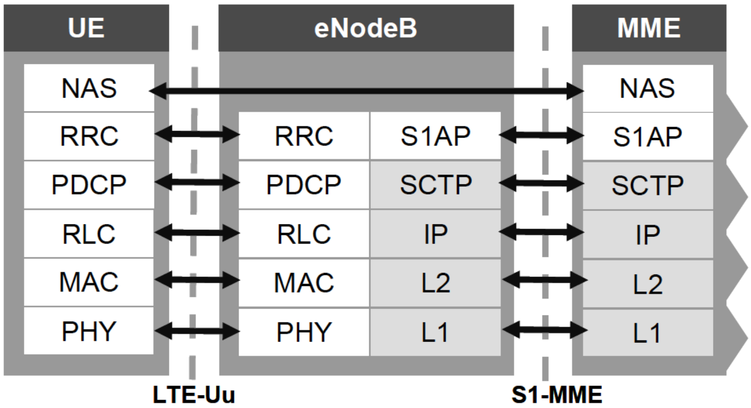


Figure 3.8: LTE Control Plane Protocol Stack
Source: LTE for UMTS (Holma and Toskala, 2009)

In addition to the previously referred layers, the control plane consists of:

- Non Access Stratum (NAS): NAS is responsible for Evolved Packet System (EPS) session management, through bearer management and authentication, and EPS mobility management, handling the mobility and paging messages related to UEs.
- Radio Resource Control (RRC): RRC has an important role in mobility, since it is responsible for paging and broadcast messages, as well as other mobility functions. The RRC is also responsible for radio bearer control.

3.3.4 Network Entry

The UE network entry process is composed by several steps, as described below.

Firstly, the UE sets up the radio access to the eNB, by conduction the random access procedure. In this step, the messages transmitted for the channel setup, such RRC and NAS, are not yet encrypted. This step is detailed in Random Access Procedure Random Access procedure.

The second step consists of an attachment request, made by the UE. This is known as registration request, and, it implies that the UE sends all the measured radio details, and, also its key, so it can be checked by the network. The UE keys are stored at the USIM card.

The third step is the authentication and key agreement, consisting of mutual authentication and key exchange processes, resulting in a secured communication. The authentication and key agreement is started by the UEs USIM identification, which is received by the MME and forwarded to the HSS, which will confirm or reject the UE network entry authorization. The HSS response is then sent back to the UE and, if the authorization succeeds, the UE will generate a top-level mutual key, which is sent back to the MME.

After the authentication procedures are completed, the MME will register the UE in the network. This step consists of informing the HSS about the UE location and contacting the SGW and PGW for the creation of the UEs corresponding bearers and IP address allocation.

Finally, the MME informs the eNB and the UE about the session setup, and the network entry is completed.

Random Access procedure

In order to perform the network entry and the connection setup, the UE must perform the random access, with the main objective of acquiring the uplink timing and synchronization. The random access is also used for other purposes, such as connection re-establishment and for the uplink synchronization used in handover, allowing lower delays in the procedure.

Two Random Access modes are used: the contention based and non-contention based. The first is used for the initial network entry and it is composed by four steps, while the second is only used for handover, uplink synchronization re-establishment and position, and it is composed by the two first steps of the contention based random access. These steps are explained as below, and represented in Figure 3.9.

The UE initiates the procedure by sending the random access preamble to the eNB, which will tell to the eNB the UEs transmission timing. The preamble response will allow the UE to attain the uplink synchronization with the network, namely the eNB, which is a requirement for the data transmission to the eNB.

In case of contention based random access, the previous step will also be used to assign uplink resources in the terminal to allow the next random access step. The next random access step, happening only in the contention based random access, consists of sending the UEs identity to the network using the Uplink shared channel resources assigned in the previous step, so the UE can be assigned with a Cell Radio- Network Temporary Identifier (C-RNTI), which uniquely identifies the node in the network.

The last step consists of the contention resolution. Since multiple nodes can execute the random access simultaneously, the same random access resources can be used by several UEs. However, to accomplish the random access procedure to the correct terminal, the temporary C-RNTI sent in step 3 is compared to the received C-RNTI sent by the network in this

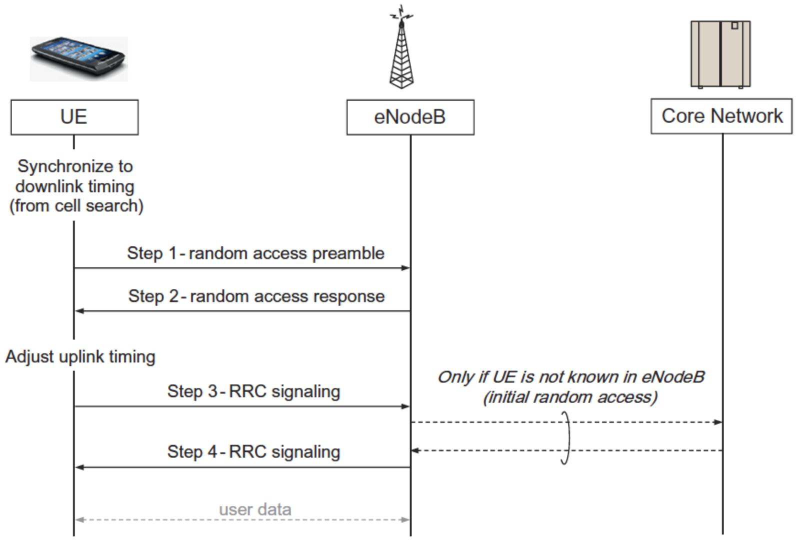


Figure 3.9: LTE Random Access Procedure
Source: 4G LTE/LTE Advanced (Dahlman et al., 2011)

step. This comparison is made by the UE. If they match, the random access is concluded, otherwise, the UE must restart the random access procedure from the first step.

3.4 Deployments

As LTE is an evolved version of 3G, and also a 3GPP release, it is designed to support and guarantee interoperability between legacy 3GPP versions. Because of that, it is possible for the operators to evolve their networks will lower Capital Expenditure (CAPEX), since they must guarantee full interoperability between 2G/3G equipment and the new 4G infra-structure. The cost of maintaining two parallel networks (e.g. 3G/2G and WiMAX) (Operational Expenditure (OPEX)) would also be higher. Thus, the WiMAX does not represent a solution for these kind of operators.

As the main operators, globally, currently have 3GPP deployed networks, the natural evolution will be to the latest 3GPP releases, such as LTE and LTE Advanced. This natural evolution will also allow an economy of scale for the LTE equipment, allowing even less costs for the LTE providers and consumers.

The LTE is currently being deployed in several countries around the world, including Portugal. These deployments can be observed in Figure 3.10.

Accordingly to the recent reports from Global mobile Suppliers Association (GSA) - A non-profit trade association representing suppliers - (GSA, 2012) there are already 72 deployed

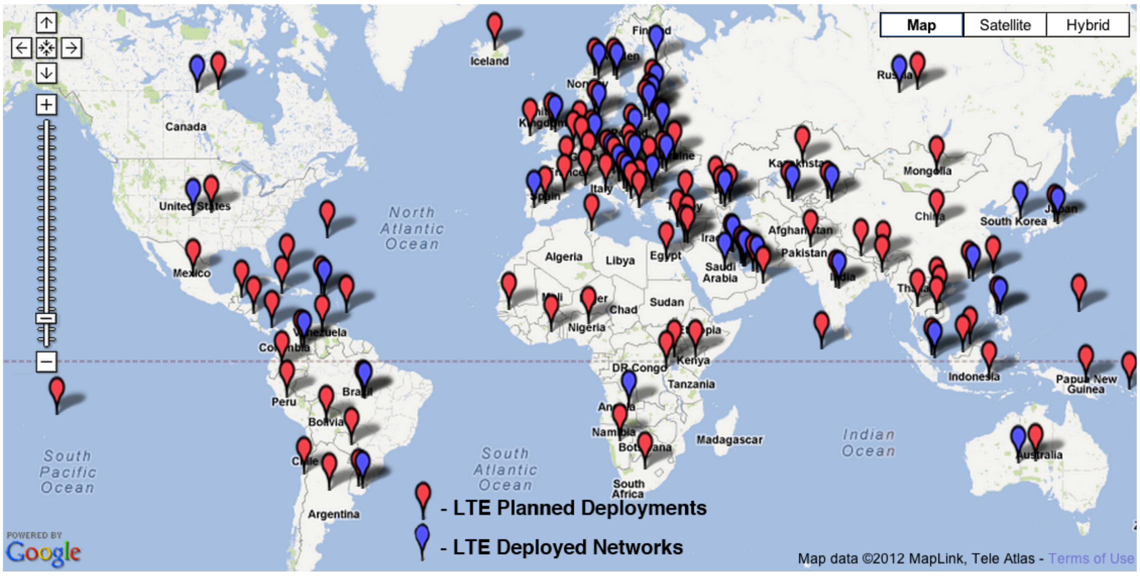


Figure 3.10: LTE Deployments

networks in 37 countries, and 319 operators in 97 countries are currently investing in LTE. This report is from May 23, 2012.

3.5 Summary

LTE is currently experiencing a large growth worldwide. Because of its features, especially the supported throughput, and also because of the backward compatibility with previous 3GPP releases, both operators and costumers are showing a lot of interest in this technology. The *all-IP* network provides a good support for new multimedia applications and also a simplified network architecture.

As seen in the LTE deployments figure, this technology will be available in several countries in the next few years, contributing to a scale economy not only for the operators equipment and also for the users equipment. This large growth will allow a cheaper technological spread and migration, where several mobile equipments (i.e. computers, mobile phones) are already being sold with LTE support.

3.6 Comparison between LTE and WiMAX

From the analysis of both technologies, it was possible to attain some knowledge of the similarities and differences between them, resulting in the elaboration of a comparative table, Table 3.1, in which one can observe each technology characteristics.

	LTE	WiMAX
Standard	3GPP Release 8	IEEE 802.16e
Duplexing Modes	FDD and TDD	Mainly TDD
Downlink multiplexing	OFDM	SOFDMA
Uplink multiplexing	SC-FDMA	SOFDMA
QoS mechanisms support	Limited Bearer Support, with 9 pre-defined QCI with pre-specified network parameters and thresholds.	QoS mechanisms with high granularity (Specify several network parameters).
Mobility	Seamless handovers and full mobility support	
Antenna Systems	Adaptive Antenna Systems (MIMO and beamforming)	
Modulation techniques	Adaptive Modulation and Coding	
Modulation schemes supported	QPSK, 16 and 64 QAM	BPSK, QPSK, 16 QAM, 64 QAM;
Backward compatibility and interoperability	With the previous 3GPP releases	With the previous WiMAX versions
Commercial Release Date	2012	2008
Downlink maximum theoretical data rates	100 Mb/s	75 Mb/s
Uplink maximum theoretical data rates	50 Mb/s	25Mb/s

Table 3.1: Comparison - LTE and WiMAX

From the table it is possible to observe that both WiMAX and LTE aim to provide wireless access with high bandwidth and with several levels of Quality of Service (QoS) assurance. In this point, WiMAX has some advantages over LTE, since it allows a better definition of service flows and network parameters. The LTE allows the definition of different types of bearers, but with pre-defined network parameters.

The duplexing schemes used in the two technologies are slightly different. WiMAX uses Scalable Orthogonal Frequency Division Multiplexing Access (SOFDMA) for both uplink and downlink, allowing the network to scale the bandwidth and network parameters accordingly to the radio conditions. In the case of LTE, it uses Orthogonal Frequency Division Multiplexing (OFDM) in the downlink, to allow better signal diversity, especially in Non Line of Sight (NLOS) scenarios, and it uses Single-carrier FDMA (SC-FDMA) in the uplink. SC-FDMA allows lower values of Peak to Average Power Ratio (PAPR), which means it can achieve higher energy efficiency than WiMAX.

The Frequency Division Duplexing (FDD) duplexing scheme used in LTE is the most preferred by the operators, since it allows paired spectrum, which means that the uplink and downlink data can be transmitted simultaneously. The WiMAX also supports FDD, although the latest versions are most suited for Time Division Duplexing (TDD), being the preferred duplexing mode for WiMAX.

Both WiMAX and LTE support Advanced Antenna Systems, such as MIMO and Beamforming, and also Adaptive Modulation and Coding (AMC). The AMC allows the adaptation of the modulation accordingly to the radio conditions, measured by signal quality indicators.

Seamless handovers and full mobility are assured for WiMAX and LTE, allowing moving speeds up to 350 km/h (Sauter, 2009).

The WiMAX technology had some time advantage, since it was released earlier. However, most of the operators have waited for the release of LTE, since it can guarantee interoperability with the previous 3GPP Access Networks. Nevertheless, some operators have emerged with WiMAX-only solutions, especially in developing countries, aiming to provide Internet to their subscribers, with similar speeds and availabilities of cable networks, such as Asymmetric Digital Subscriber Line (ADSL).

4. WiMAX and LTE Assessment

In order to correctly assess the WiMAX and LTE technologies it is important to investigate previous and related works. This investigation allows a better analysis and a better understanding of assessment metrics and guidelines.

In this chapter the related work is analyzed and described. Also, some guidelines are learned from the works mentioned and some conclusions are drawn as described in section 4.3

4.1 Performance Metrics

This section describes the main performance metrics according to three categories: Signal quality, Quality of Service and Quality of Experience.

4.1.1 Signal Quality

In order to analyze the signal quality, several metrics are used:

- Received Signal Strength Indication (RSSI): The RSSI indicates the strength of the signal. It only evaluates the strength of the signal, not taking into consideration it quality or the interferences. RSSI is measured in dBm, with values between -50 and -80 dBm, where the stronger signal is -50dBm and the weaker is -80dBm.
- Carrier Interference-plus-Noise Ratio (CINR): The CINR provides the information about signal quality to interferences/noise ratio. CINR is measured in dB, where 30dB represents the better signal quality, while 10 dB represents the lower quality signal.

4.1.2 Quality of Service

The quantitative measures of Quality of Service (QoS) are usually defined by the following parameters:

- Bandwidth: This is a very important parameter since it demonstrate the real bandwidth availability on the link. It is most relevant for data transfers or applications with high bandwidth usage.

- Delay: Delay, either Round Trip Time (RTT) or One Way Delay (OWD), are very important to real-time applications, such as voice, video, online gaming or interactive applications.

- Packet Loss: Packet loss is an important measure in every application, but most important in data transfers. Although some applications have mechanisms to decrease the packet loss impact at the end user, when the packet loss increases it has a negative impact on applications performance.

- Jitter: This measure represent the delay variation between the packets sent. It is most important in applications which require network stability in order to receive the packets with exact intervals.

Also, ITU-T Y1541 Recommendation (ITU-T, 2003) defines the performance objectives for the different classes of network Quality of Service.

4.1.3 Quality of Experience

The Quality of Experience perceived by the end user can be measured through several metrics. However, for the tests performed only MOS (ITU-T, 1996b) and E-Model (ITU-T, 2009a) were used. Both are described below:

- Mean Opinion Score (MOS): MOS consists of several users evaluating one service, giving a score of one to five, where one is the worst value and five is the best value. MOS is a subjective metric commonly used in Voice applications. However, the E-Model overcomes the need of real users and the subjectiveness associated to this evaluation, calculating the R-Factor and associating it value to the MOS Scale. The E-Model evaluates the VoIP Quality of Experience through network parameters, such as One Way Delay (OWD) and Packet Loss. The end-user perceived voice quality for each MOS values is represented in Table 4.1.

MOS	Quality
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

Table 4.1: MOS Values and End-User QoE

4.2 Related Work on WiMAX and LTE assessment

An overview and analysis of the previous Worldwide Interoperability for Microwave Access (WiMAX) and Long Term Evolution (LTE) works is performed in this section. The main focus of this analysis are the methodologies used and the results obtained. This section is divided in three sub sections, namely Fixed and Mobile WiMAX, and LTE.

4.2.1 Fixed WiMAX

This subsection describes previous work on the assessment of Fixed WiMAX.

Measurements and Analysis of Fixed WiMAX with LAN in Microgrid

In Siow et al., a combination of fixed WiMAX and Local Area Network (LAN) network is proposed for monitoring and control power meters data in a micro-grid, where the fixed WiMAX operates as the main communication network for information exchange between buildings, and the LAN serves as a secondary network within the buildings with power meters attached to the distributed energy resources (Siow et al., 2011).

The main target is to develop a set of field tests, measuring some parameters such as RSSI and Signal-to-Noise Ratio (SNR). The performance analysis of these parameters will determine the feasibility of WiMAX being used in micro-grid. These parameters are also compared with some propagation models, in order to find a proper model to be implemented for future analysis and simulation.

The measurements were taken at various locations, to allow a comparison between the real measures and the theoretical propagation models. These comparisons between real and simulated results, such as RSSI over Distance and SNR over RSSI, are shown and analyzed. The results showed which simulation model was the most suitable for this scenarios and that RSSI decreased along the distance. These comparisons also allowed to observe the increase in SNR along the increase in RSSI.

The authors found some problems when using the COST-231-Hata-propagation Model, since it only supports frequency of 1.5 GHz to 2GHz, and the frequency used was 5.8GHz. To solve this issue, the model was modified by the authors using mathematical functions to support the used frequency.

Through the results and measurements shown in this work, it was concluded that the WiMAX network is suitable to be deployed in a micro grid.

Experimental Evaluation of IEEE 802.16 WiMAX Performances at 2.5 GHz Band

Durantini et al. aim to provide data, performance and capabilities results of a 2.5GHz WiMAX deployed test-bed (Durantini et al., 2008). It is essentially focused on link throughput and quality analysis, with different parameters, such as bandwidth and delay.

In order to cover different types of conditions, various test scenarios are considered, such as dense urban, urban, suburban and rural, by testing in several locations with different types of urbanizations degree and orographic characteristics. The parameters taken in consideration are: bandwidth, end-to-end delay, jitter and packet loss. These are measured with different generated traffic, such as Video Streaming (VS) and Video on Demand (VoD), using IPerf (IPerf, 2011), for bandwidth tests and ixCharriot (IxChariot, 2011), for generating realistic traffic conditions, varying the QoS, number and characteristics of streams.

Beyond the parameters already presented, the performance is evaluated in Line of Sight (LOS) and Non Line of Sight (NLOS) conditions, with different distances from the Base Station (BS) and with different configurations of the system parameters, such as the duplexing mode, the transmit power, the multi-rate support, the Downlink (DL)/Uplink (UL) partition and modulation scheme.

A large set of performance tests are conducted, showing real performance differences when changing modulation schemes, DL/UL partition schemes, LOS/NLOS conditions, number of flows, etc. It gives a clear idea of what these changes cause. Also it is demonstrated the impact of different QoS classes in the link stability, such as Real Time Polling Service (rtPS) and Non-Real Time Polling Service (nrtPS), for VS and VoD, showing that rtPS is much more stable and suitable for real time video streaming.

This paper is focused on evaluating WiMAX performance and potential in supporting real-world applications. From the results obtained, it is clear that WiMAX has the potential to support various types of real-world applications. However, it is shown that in different conditions, these applications can be somehow conditioned. The variation of numerous conditions can improve or worsen the performance of the WiMAX network.

From the test scenarios it is also possible to obtain info about the different changes: Changing the modulation scheme from basic to complex modulation, clearly improves the link throughput, although it becomes more variable. When the UL/DL ratio is changed from 50/50 to 65/35, the DL throughput also increases. It is also shown the decrease in throughput when having more than one client/ Customer Permises Equipment (CPE). Finally, the difference between rtPS, nrtPS and differentiated QoS classes is demonstrated, showing that rtPS is much more stable in terms of throughput.

Performance Evaluation of IEEE 802.16 WiMAX Link With Respect to Higher Layer Protocols

The main objective of Yousaf et al. is to evaluate, analyze and compare the performance of a WiMAX link over different load and traffic conditions, considering data throughput and link stability over distance (Yousaf et al., 2007). For this purposes, Institute of Electrical and Electronics Engineers (IEEE) 802.16d compliant equipment operating in 3.5GHz band was used.

The tests were conducted in two different categories:

- Link capacity/throughput: testing both UL and DL performance with both Transmission Control Protocol (TCP) and User Datagram Protocol (UDP) traffic for various modulation schemes (Quadrature Amplitude Modulation (QAM), Binary Phase Shift Keying (BPSK), Quadrature Phase Shift Keying (QPSK)).

- QoS testing: testing WiMAX QoS support feature, where one client is sending UDP data (emulating Voice over IP (VoIP) using rtPS), while other client is generating TCP traffic (emulating normal Hypertext Tramsfer Protocol (HTTP), using Best Effort (BE)). In QoS tests, the modulation scheme was set at QAM64 3/4. Multi-rate capability was disabled in order to correctly measure throughput and link quality of each performance test. Modulation schemes were manually defined, and the distances varied from 220 to 9400 meters, with minimum transmission power defined in the BS. A second set of tests were conducted at 9400meters with more transmission power (20dBm, instead of 13dBm), in order to evaluate link stability and throughput differences.

It was not possible to test NLOS. Some conclusions about worsen link quality during adverse weather conditions should be, however difficult, demonstrated.

The main differences in the throughput when changing modulation schemes are shown. Through the graphics it is possible to observe the consequences of a bigger distance to the BS, and also the changes in the modulation scheme. Higher modulation, results in less link consistency. However, higher modulation, also results in higher throughput, even if far away from the BS. Other interesting result is the behavior of the transmitted flows when using QoS, where we can see the decrease of data transmission rate in TCP flows (BE) in order to provide a good transmission rate to UDP (rtPS) flows. We can also observe the effect of the transmission power in link consistency/throughput. The second set of tests, increasing transmission power, demonstrated that, in longer distances, the link is more stable and consistent with higher transmission power from the BS.

This paper has shown very interesting results in a real test-bed scenario. It can help to understand the effects of various conditions, parameters and configurations in the WiMAX link.

VoIP over WiMAX: Quality of Experience Evaluation

Bernardo et al. is essentially focused on evaluating the performance of UDP/Real-time Transport Protocol (RTP) and Datagram Congestion Control Protocol (DCCP) and the Quality of Experience (QoE) in voice application scenarios, using a real Fixed WiMAX testbed (Bernardo et al., 2009b).

To compare the different protocols, the transmission quality will be analyzed, as well as the voice quality experienced by the end users. This will allow to measure the performance between different protocols in terms of OWD and Packet loss and in terms of QoS, where the MOS is measured through E-Model calculations.

The tests are conducted using rtPS scheduling type, and using two types of bandwidth reservation: Overestimated (9000 Kb/s, higher than required) and Underestimated (dependent on flows number, more limited). The authors used a traffic generator (D-ITG) in order to emulate G.711 Voice Codec traffic, between a server in Connectivity Service Network (CSN) and three machines connected through a Subscriber Station (SS), which is connected to the BS.

This work demonstrated the differences between over and underestimated bandwidth conditions over Fixed WiMAX and also the differences between two different protocols (UDP and DCCP). These differences were shown by MOS measurements, based on One Way Delay (OWD) and Packet loss results.

From the results obtained, it is shown that in overestimated scenario the OWD and Packet loss have higher values with DCCP than UDP, also visible in the decrease of MOS, which is related to the OWD and packet loss. On underestimated scenarios, the behavior of DCCP flows is better than before, but however, worst than UDP, with an overall of worst MOS in all protocols.

To conclude, based on the results demonstrated, the UDP is the most appropriated protocol to voice transmission, and, as expected, the voice quality perceived (MOS) in overestimated scenarios is better than the underestimated scenario.

Summary

A summary of the previous works is depicted in Table 4.2. In this table, the main characteristics of each work are shown, such as frequency bands, communication schemes used, parameters measured, tests scenarios and other relevant comments.

IEEE 802.16d - *Fixed WiMAX*					
Work	Frequency	Measurements and parameters	Modulation Schemes	Test conditions	Additional comments
Measurements and Analysis of Fixed WiMAX with LAN in Microgrid (Siow et al., 2011)	5.8 GHz	RSSI SNR	BPSK	LOS Type: Testbed	COST-231-Hata propagation model
Experimental Evaluation of IEEE 802.16 WiMAX Performances at 2.5 GHz Band (Durantini et al., 2008)	2.5GHz	Throughput, packet loss, jitter, delay	QAM, BPSK, QPSK	LOS and, NLOS(indoor), Full Duplex, Half Duplex Type: Testbed	Variation of traffic flows and scenario conditions
Performance Evaluation of IEEE 802.16 WiMAX Link With Respect to Higher Layer Protocols (Yousaf et al., 2007)	3.5GHz	Throughput	QAM, BPSK, QPSK	LOS, Near-NLOS Type: Testbed	Consistency of WiMAX link with various distances, QoS effects in data transmission.
VoIP over WiMAX: Quality of Experience Evaluation (Bernardo et al., 2009b)	3.5HHz	MOS, Packet Loss, and OWD	QAM	LOS Type: Testbed	MOS calculated with E-Model

Table 4.2: Fixed WiMAX previous works summary

4.2.2 Mobile WiMAX

This subsection describes previous work on the assessment of Mobile WiMAX.

Mobile WiMAX: Performance analysis and comparison with experimental results

Tran et al. aim to provide a performance analysis of a mobile WiMAX system operating on all link-speeds in an urban microcell (Tran et al., 2008). This analysis is obtained from a fully compliant 802.16e simulator, covering such aspects as: link adaptation, packet error rate and throughput. These theoretical results are supported by experimental data collected in an urban microcell environment.

In this paper the Packet Error Rate (PER) and the throughput performance of mobile WiMAX as function of SNR are analyzed. As mentioned before, this results are then compared against real measurements collected from a various number of drive tests in an urban microcell. These real measurements are performed in a Global Positioning System (GPS) enabled vehicle, driving at speeds of up to 35 km/h, receiving a H.264 streaming video and logging various parameters (PER, Throughput, signal level, GPS location).

It is interesting to see the capabilities (throughput) of each modulation scheme varying the distance from the BS, as well as the evolution of PER when the SNR increases.

The main contribution of this work was to show the agreement between the real and predicted measures and results, indicating that the simulator can be used to predict WiMAX performance for a range of environments and conditions. Also, it is shown that Mobile WiMAX was able to achieve street-level range of 300-2100 meters with speeds up to 35km/h, while maintaining a reasonable data rate and latency.

Mobile WiMAX Throughput and Delay Measurements in Railroad Environment

The main objective of this work, Mahasukhon et al., is to study and analyze the behavior of throughput and link stability over distance in high speed moving trains, over a Mobile WiMAX network (Mahasukhon et al., 2011). The study is also accompanied with laboratory tests results.

Firstly, to determine some parameters, like maximum supported communication distance and system configuration parameters, a laboratory emulation experiment was conducted. Thereafter, the field test was performed, by driving a car at approximately 60 Mph at the highway, covered by the WiMAX network. Parameters like throughput, latency and velocity are measured over distance, for both UL and DL.

This work demonstrate the potential of a WiMAX network to allow good transmission rates over long distances, and also supporting mobility.

During the laboratory experiments, it is shown the throughput capabilities of each modulation scheme over path loss, using SISO Butler Model, for both UL and DL. It was found that, using BE scheduling, it is possible to achieve a downlink throughput of nearly 2.8 Mb/s and 2 Mb/s for the uplink, using a modulation scheme of QPSK 1/2, achieving a maximum distance of 7.5 km and allowing a moving speed of 60 mph. It was also found that the latency is much more higher and unstable over longer distances.

Evaluation of Multimedia Services in Mobile WiMAX

Sousa et al. focus on analyzing the mobile WiMAX performance using different kinds of real time multimedia applications, such as voice and video, at handover and moving speeds scenarios, and also measuring the potential of IEEE 802.21/Media Independent Handover (MIH) (IEEE, 2005a), by analyzing different parameters, such as packet loss and delay, employing the MIH information in three different modes (Sousa et al., 2008a). It also gives an overview of 802.16, 802.21, and briefly describes WiMAX technology.

The evaluation process of this work is based on simulations. For that, NS-2 and WiMAX mobility package from NIST was used. Two different moving speeds and different cell sizes are configured for the test scenarios, which are 30 and 120 km/h and urban micro and macro cells, according to the ITU vehicular A profile. In order to evaluate the use of MIH information, three modes are considered: No Events (No MIH), Events (LinkDown trigger is used to trigger the handover process), Predictive triggers (Link Going Down trigger is used to start the handover process, with 60 and 80% of confidence levels). Voice traffic is simulated with Constant Bit Ratio (CBR) streams, with the same characteristics as the G.711, G.726 and G.729 codecs. Video stream quality is measured by calculating the Peak Signal Noise Ratio (PSNR), which compares the original frames with the received frames, and relates it to the MOS 5-point scale, in order to categorize the quality of the video received. These measures are obtained with Evalvid tool.

This work demonstrate the effects of the handover processes (high levels of packet loss and delay) and shows how to increase the performance of this procedure by using MIH information.

With this study, it is proven that without MIH information, the real time multimedia applications performance is very poor when handovers occur, leading to high packet loss and delays. However, the use of cross layer information, provided by MIH, demonstrated that the handover process is improved, leading to lower packet loss and lower delays.

Multi-client Video Streming over WirelessMAN-OFDMA

In Bernardo et al. the main target of this work is to evaluate the performance of video streaming over a deployed 802.16e WiMAX Testbed, operating on 3.5Ghz band (Bernardo et al., 2009a). These tests will allow to observe the capabilities of WiMAX network in terms of video streaming supporting multiple clients/flows in limit conditions, with no QoS differentiation and with high bandwidth usage.

To analyze the behavior of the network, four CPE's are configured and connected to the BS and the tool Evalvid was used in all CPE. One raw video was compressed with MPEG-4 in three different rates: CBR 256 Kb/s, CBR 512 Kb/s and Variable Bit Ratio (VBR) with peak rate of 540 Kb/s. To avoid high packet losses at sending side due buffer overflows, the authors changed the sending methodology to allow the sending of higher number of simultaneous flows.

From the tests executed it was possible to measure the OWD, Packet Loss and the MOS, obtained through Evalvid tool.

This work demonstrate the capabilities of WiMAX, in particular WirelessMAN-OFDM, of supporting multi-user/streaming at the same time and sharing the same link, with high bandwidth usage, even exceeding the theoretical measured throughput.

Also it shows the differences in supporting different types of traffic and which is the most appropriate in terms of better QoE and lower operator costs.

From this work we can observe that when sending video streams in CBR256 Kb/s it is possible to attain a good MOS, meaning a relatively good QoE and allowing a large number of concurrent clients and flows, keeping OWD under 25ms until 43 simultaneous flows. With CBR512 Kb/s stream the number of concurrent flows is decreased by half, limited by high bandwidth usage and higher delays and packet loss. However, the MOS in this codification is excellent. The VBR behavior is similar to CBR512 Kb/s, since the bandwidth usage is quite similar.

Summary

A summary of the previous works is depicted in Table 4.3. As in the previous Fixed WiMAX summary, the main characteristics of each work are shown in this table.

IEEE 802.16e - *Mobile WiMAX*					
Work	Frequency	Measurements and parameters	Modulation Schemes	Test conditions	Additional comments
Mobile WiMAX: Performance analysis and comparison with experimental results (Tran et al., 2008)	2.3GHz	PER, Throughput, SNR	QPSK, QAM	LOS and NLOS Type: Simulation and Testbed	80/20 DL/UL ratio in experiment tests. Throughput over SNR; Modulation scheme and PER over SNR
Mobile WiMAX Throughput and Delay Measurements in Railroad Environment (Mahasukhon et al., 2011)	2.5GHz	Delay, Throughput	Laboratory: AMC; Testbed: QPSK1/2	LOS Type: Testbed	Laboratory: 31:15 DL/UL rate. Testbed: 25:21 DL/UL rate
Evaluation of Multimedia Services in Mobile WiMAX (Sousa et al., 2008a)	3.5Ghz	Throughput, packet loss, delay	16QAM	LOS Type: Simulation	Three handover MIH modes; Moving speeds of 30 and 120 km/h; different cell sizes: Urban Micro and Urban Macro cells
Multi-client Video Streaming over WirelessMAN-OFDMA (Bernardo et al., 2009a)	3.5Ghz	MOS, packet loss, and OWD	UL 16QAM3/4; DL: 64QAM1/2 and 5/6	LOS Type: Testbed	Video Streaming with Evalvid, encoding raw video with CBR 256 Kb/s, CBR 512 Kb/s and VBR with peak rate of 540 Kb/s. BE QoS class was used.

Table 4.3: Mobile WiMAX previous works summary

4.2.3 LTE

In this section, an analysis of the current related work in LTE assessment is conducted.

Performance Analysis of VoIP Services on the LTE Network

Asheralieva et al. have performed an extensive analysis of LTE in a simulation environment (Asheralieva et al., 2011). This simulation consists of the emulation of several VoIP users simultaneously, employing different LTE parameters, with optimal and realistic test conditions.

By comparing the performance of different scheduling techniques, namely fully dynamic and semi-persistent packet scheduling, it is possible to analyze the LTE voice service capacity and the Medium Access Control (MAC) layer functionalities. Also, the authors change some LTE network parameters, such as channel bandwidth and modulation schemes to analyze their impact on the network behavior. In order to study the link adaptation and the impact of Hybrid Automatic Repeat Request (HARQ) in the network QoS values, these parameters are also varied and analyzed.

All the simulations consist of bi-directional calls with two different codecs, the G.723.1 and the G.711. The packet loss and end-to-end delay are the network parameters measured. This work comprises a LTE simulated network working in 2GHz frequency band, with Frequency Division Duplexing (FDD) duplexing mode. This work demonstrates the results through several charts, where the effects of the different parameters are shown, such as the usage of HARQ and different scheduling mechanisms, followed by a clearly explanation of those results.

From this work it was possible to observe that the semi-persistent packet scheduling has some advantages in the QoS parameters assurance over the fully dynamic scheduling. Also, the usage of the HARQ mechanism demonstrated some advantages in terms of QoS assurance (delay and packet loss). From the several tested scenarios, the best support for VoIP simultaneous users is set at about 150 users, by guaranteeing an end-to-end delay below 200ms.

Mobile VoIP User Experience in LTE

Andersson et al. aim to demonstrate the end-user perceived VoIP quality through the measurement of MOS in different LTE scenarios and with different codecs, in a simulation environment (Andersson et al., 2011).

This work analyzes the end-user VoIP QoE in two different LTE scenarios (i.e., with different bandwidths) and with several codecs, giving an overview of the usage of different bandwidth, as well as the QoE allowed by each codec. This work is performed in OPNET modeler 16 (OPNET, 2012), using a pre-defined scenario with 7 cells, each with 5 users, using FDD as duplexing mode. Each user starts a bi-directional call with another randomly selected user. This methodology is conducted with 1.4 and 20 MHz bandwidths, using four different codecs: GSM FR (Radio-Electronics, 2011), G.729.A (ITU-T, 2007), G.723.1 5.3K (ITU-T, 2006) and G.711 (ITU-T, 2009b).

It was possible to observe that the employed bandwidth slightly affects the results, and, as expected, the codecs that use higher bandwidths can guarantee higher levels of MOS.

VoIP Quality Monitoring in LTE Femtocells

Olariu et al. analyzed the employment of LTE femtocells, using the end-users xDSL connections as backhaul, in terms of quality of experience assurance (Olariu et al., 2011). This evaluation takes into account several possible scenarios of these implementations, e.g, in situations where the femtocell operator does not have control of the Digital Subscriber Line Access Multiplexer (DSLAM) traffic prioritization. Also, the authors propose a new method of mapping the measured MOS in a way that the QoE degradation could be measured at intermediary points.

This study is conducted in the Qualnet (Qualnet, 2012) network simulator, in a VoIP multi-user scenario, with and without background traffic, where the users and the background traffic are incrementally increased until the measured MOS decreases below a pre-determined limit. Each VoIP flow is a full-duplex call (i.e. bi-directional call) employing the G.711 codec. Several scenarios are considered, where the callee and caller change from different network conditions (e.g., changing network service provider, how the nodes are connected to the network, etc). This methodology has the objective of analyzing the different proposed scenarios, as well as observing if the DSLAM can become the bottleneck in the communications. The reason for this behavior is caused by the background traffic generated from other users sharing the same DSLAM, and the lack of prioritization mechanisms.

With the growth of mobile users and with the lack of indoor coverage in some locations, the femtocells are, in some occasions, considered. This work demonstrates some possible problems of the usage of those femtocells in commercial deployments, where the femtocell connection is not controlled by the Internet service provider, leading to some problems in the guarantee of the service level agreements.

The results shown that the DLSAM traffic prioritization should be employed, in order to guarantee good levels of QoE to the end-users. The employment of intermediary Mean Opinion Score (IMOS) at several intermediary points helps the analysis of the MOS, since this concept isolates some problems in the network, by partitioning the link between the callee and the caller.

Comparison of VoIP capacity between 3G-LTE and IEEE 802.16m

Wang et al. focus on the evaluation of WiMAX IEEE802.16m and LTE in terms of voice users capacity, using different types of scheduling, link adaptation and modulation schemes (Wang et al., 2009b). For both WiMAX and LTE, numerous characteristics, such as frame structure, physical signals and overheads, resource mapping for uplink and downlink, scheduling and link adaptation, and, HARQ process, are explained and presented.

This work is performed through simulation, employing TDD as duplexing scheme for both LTE and WiMAX, using a 10 MHz bandwidth, and employs different values of the parameters referred above. To quantify the voice users capacity per sector, a VoIP capacity criterion is used, assuming a 12.2K b/s codec with 50% of activity factor, where the maximum capacity per sector is reached with 5% of users in outage.

The results shown that LTE allows more granularity in terms of modulation coding schemes used. This way, shows advantages over WiMAX, which cannot effectively use the frequency resources and get benefits from the semi-persistent scheduling in the downlink, that allowed a higher voice user capacity. In case of dynamic scheduling, the LTE can also support a higher number of users, since the coarse Modulation and Coding Scheme (MCS) granularity allows a better trace the link variations.

From the results obtained, it was possible to conclude that LTE has advantages over WiMAX in terms of dynamic scheduling due to its Modulation and Coding Scheme (MCS) granularity, since the lower WiMAX MCS does not allow effectively tracing the link changes and adapting the dynamic scheduling.

VoIP performance in multi radio mobile devices

Iwayemi and Zhou demonstrate the effects of the transmission of VoIP data through different communication technologies and equipment, such as WiFi and High Speed Packet Access (HSPA), along with laptops and mobile phones (Iwayemi and Zhou, 2009) . This evaluation analyzes network QoS and QoE parameters, such as packet loss, delay, jitter, and MOS.

This study is conducted in a testbed within different connection scenarios, namely mobile phone to PC (via WiFi), mobile phone to mobile phone (via WiFi) and HSPA phone to WiFi PC. This work is made by executing VoIP calls between the nodes, where the VoIP codecs used are the G.711 (ITU-T, 2009b), which is less complex but high bandwidth consumer, and with Internet Low Bit Rate Codec (iLBC) (IETF, 2004b), which is a lower bandwidth codec, although more complex. The choice of the codecs was due to the worldwide usage of both codecs - G.711 is widely used in wired networks (such as Public Switched Telephone Network (PSTN) and Integrated Services Digital Network (ISDN)), where the bandwidth availability is higher, whether the iLBC is most popular in low bandwidth scenarios, such as Skype and other Internet messengers, as well as wireless networks. In order to measure the network parameters, the authors use a sound card scope oscilloscope for the end-to-end delay and the jitter measurements, while packet loss and the throughput are measured with a packet trace tool Wireshark (Wireshark, 2012).

The results demonstrated the impact of the usage of different codecs, with different bandwidths and complexities, in the network parameters values and the achieved MOS values. The simpler codecs allow lower end-to-end delays, caused by the lower processing overhead. Also, it is seen that the WiFi provides better QoE values, as well as lower delays and jitter values, mainly because the smaller WiFi protocol overheads and the lack of HARQ mechanism overhead. Despite the good QoE values attained with the G.711 codec, this is an unattractive codec for bandwidth-constrained networks, since it requires high bandwidth, when compared to other voice codecs, such as iLBC. Also, it is observable that the mobile phones are more sensitive to codec complexity than the PCs, mostly due to their limited processing capacity.

This work allowed to understand the different VoIP codec behaviors in different network conditions. Through the analysis of the QoS parameters within different technologies, it was possible to observe the effects of the codec complexity and bandwidth usage in the network and nodes performance. It was concluded that it is possible to attain better performances over a WiFi network than in HSPA, mostly due to the different technological overhead and network architecture. Although the G.711 codec demonstrated better MOS results, a lower bandwidth codec must be considered to the bandwidth-constrained networks.

Summary

A summary of the previous works is depicted in Table 4.4. As in the previous summaries, the main characteristics of each work described in this subsection are shown in this table.

LTE - *Long Term Evolution*				
Work	Technology and frequency	Measurements and parameters	Test conditions	Additional comments
Performance Analysis of VoIP Services on the LTE Network (Asheralieva et al., 2011)	LTE; FDD at 2 GHz	Packet loss and end-to-end delay	Ideal and realistic test conditions Type: Simulation	Analysis of semi-persistent scheduling and fully dynamic scheduling. Usage of several MCS. With and without HARQ. Usage of G.711 and G.723.1 voice Codecs
Mobile VoIP User Experience in LTE (Andersson et al., 2011)	LTE; FDD at 2 GHz	MOS	LOS Type: Simulation	Allowed MOS for different system bandwidths (1.4MHz and 20 MHz) and with different codecs (G.711, G.723.1 5.3K,G.729A, GSMFR)
VoIP Quality Monitoring in LTE Femtocells (Olariu et al., 2011)	LTE and ADSL FDD	MOS and delay	LOS Type: Simulation	Analysis of LTE Femtocells using users xDSL connection as backhaul. Proposal of intermediary MOS (IMOS)
Comparison of VoIP capacity between 3G-LTE and IEEE 802.16m (Wang et al., 2009b)	LTE and WiMAX TDD at 2GHz	Voice users capacity - through calculations	LOS. Type: Simulation	Analysis of semi-persistent scheduling and fully dynamic scheduling. Maximum voice users capacity. Usage of different MCS.
VoIP performance in multi radio mobile devices (Iwayemi and Zhou, 2009)	HSPA and WiFi	Packet loss, jitter, delay and MOS	LOS. Type: Testbed	Analysis of different codecs (G.711, iLBC, G.729 and AMR). Several scenarios considered (WiFi to HSPA communication, and WiFi to WiFi communication)

Table 4.4: LTE previous works summary

4.3 Conclusions

From the work described above, it was possible to understand and learn new testing guidelines that can be used in real and simulation evaluation scenarios.

It is observable that none of the previously analyzed works has an analysis of the end-user perceived voice quality (QoE) while comparing different WiMAX QoS service classes, using a Mobile WiMAX real deployment. Bernardo et al. evaluate the VoIP QoE in a Fixed WiMAX testbed without assessing the impact of different WiMAX QoS mechanisms and the background traffic (Bernardo et al., 2009b). Durantini et al. perform an evaluation of a fixed WiMAX network within different scenarios and traffic flows, but only considering the network QoS parameters (Durantini et al., 2008). The mobile WiMAX works studied are much focused on the evaluation of network QoS parameters, and the one that is focused on QoE evaluation, in Bernardo et al., performs an evaluation of video streaming (Bernardo et al., 2009a).

This work aims to fill this gap, by conducting an evaluation of VoIP multi-users scenarios in a real Mobile WiMAX testbed deployed in an urban environment. This assessment will also evaluate the differences between some of the available QoS service classes and within LOS and NLOS scenarios, using real end-user equipment.

In terms of simulation works, none of the analyzed studies has an analysis of end-user perceived voice quality while comparing Mobile WiMAX and LTE through simulation. Mainly, the related works perform an analysis of the impact of different codecs (Andersson et al., 2011) (Iwayemi and Zhou, 2009), different technologies (Olariu et al., 2011) (Wang et al., 2009b) and different access conditions and modulation schemes (Asheralieva et al., 2011). However, none aggregates the analysis of the impact of background traffic with the usage of different codecs within different technologies, such as WiMAX and LTE.

In terms of comparisons between simulation and testbed, Tran et al. (Tran et al., 2008) conducted an analysis of the WiMAX technology, in simulation and testbed, however, the end-user VoIP quality in a multi-user scenario involving background traffic is not analyzed, as performed in this study.

The simulation studies performed aimed to fill these gaps, by analyzing the different scenarios and technology's capabilities, namely WiMAX testbed and WiMAX simulation, as well as WiMAX and LTE, in terms of multi-users VoIP capacity and support, and its QoS assurance capabilities, by sending the VoIP data with and without background traffic.

5. WiMAX Pilot Assessment

During this work several tasks were performed, not only related to the assessment of generic and multimedia applications in a real WiMAX Testbed, but also the evaluation and characterization of specific EDP applications, related to the EDP-WiMAX project.

The possibility of conducting tests in a real testbed was very important and represented an added value to this work. In order to seize this advantage, several test conditions were used and different applications were considered. By performing tests in different access locations and scenarios, such as with and without line of sight, emulating real traffic patterns with and without background traffic, and, access to the network in fixed and mobile scenarios (i.e., fixed - outdoor and indoor - and mobile - usb - Customer Permises Equipment (CPE)s), it was possible to evaluate the technology in numerous real world scenarios.

Since this work is related to EDP-WiMAX project, other tasks related to this project were also conducted. The EDP-WiMAX project involved several tasks, from the equipment deployment to traffic pattern characterization. Also, through the constant link and equipment monitoring, it was possible to study and analyze the WiMAX Pilot availability and behavior in different weather conditions.

This chapter describes the different tests conducted for the WiMAX technology assessment, as well as the EDP-WiMAX project tasks. The first section explains the testing guidelines and methodologies used in the tests. The second section describes the results obtained, and the last section, which is related to the EDP-WiMAX Project, describes the several tasks and results related to the project.

During the second semester several new test scenarios were considered and the previous tests were complemented with new test conditions, namely VoIP tests using different WiMAX Quality of Service (QoS) mechanisms. Also, the main EDP applications assessment and traffic characterization tasks were performed during the last semester.

5.1 Testing guidelines and methodologies

In this section the different testing guidelines and methodologies are presented. The testbed configuration, scenarios and equipment are also described.

5.1.1 WiMAX Testbed Configuration

All the testbed equipment used for the tests belongs to the EDP-WiMAX Project Testbed. In fact, this project testbed was set up to assess the WiMAX technology capacity for the support of multi-users environments in different orographic conditions, such as rural and urban scenarios, with and without line of sight, as demonstrated in this document.

The WiMAX IEEE 802.16e (IEEE, 2005b) Project testbed consists of two *Base Transceiver Station (BTS) - Alvarion BreeMAX Macro Outdoor* connected to the same *Access Service Network (ASN) Gateway - Alvarion BreezeMax ASN-GW Mini-Centralized-* through a centralized architecture, in accordance with the WiMAX specifications and WiMAX Forum (Forum, 2011) Network Reference Model (NRM).

As mentioned, two BTSs were deployed: One in Polo 1, in order to provide connectivity in urban conditions and the other in Polo 2, to provide network access in rural conditions.

This testbed is also connected to the EDP network and to the internet, allowing the connection from EDP applications to EDP field equipments through the WiMAX link. The network architecture is represented in Figure 5.1.

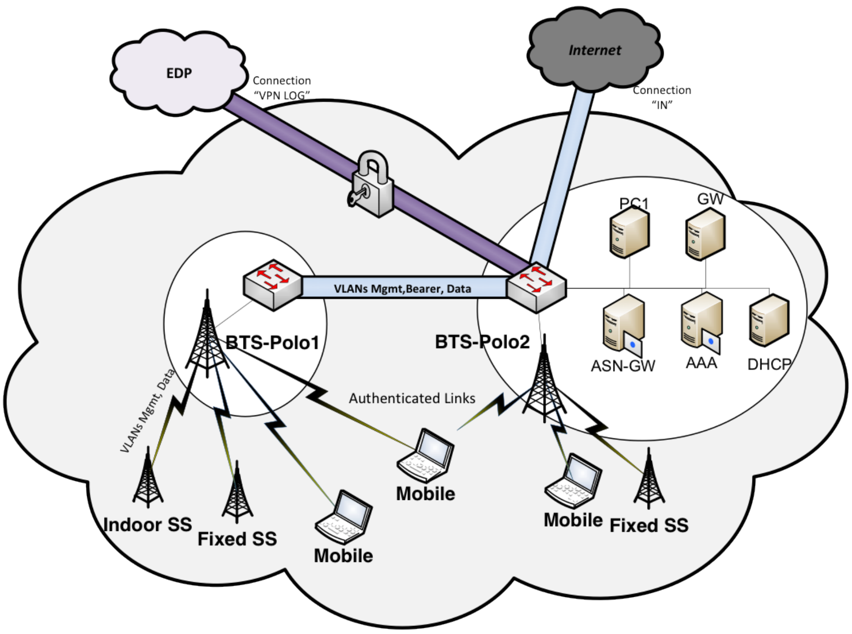


Figure 5.1: Testbed Network Architecture

Each BTS is configured with 4x2 Multiple Input Multiple Output (MIMO) Matrix A (allowing better coverage - both antennas send the same data stream), operating in 2.6 GHz frequency band, with a 10MHz channel bandwidth. This channel is configured with Adaptive Modulation and Coding (AMC), allowing different modulation schemes for the clients - which can change due to factors such as weather conditions, line of sight and distance to the BTS. Both BTSs are equipped with two Dual- Slant antennas (65° for each sector, with dual polarization). The frame configuration is 60/40, where 60 is for downlink and 40 is for uplink. These values were chosen to allow a connection as symmetrical as possible, but still slightly favoring the downlink, since it is the direction where most traffic is generated. The testbed configuration was maintained in all the tests performed.

5.1.2 Testing Scenarios and Locations

The testing scenarios described were chosen in the context of the EDP-WiMAX Project, in order to demonstrate the WiMAX technology behavior in different orography conditions, such as urban/rural and Line of Sight (LOS) and Non Line of Sight (NLOS).

The locations defined were the following:

- **Quinta da Várzea**

  This location is considered an urban location, without Line of Sight (NLOS) with the BTS. The distance to the BTS in straight line is 1.7km. This location is used in the EDP context for telemetry, monitor and control of the transformer station, namely *TCMT*, *DTC/ Inovgrid* and *Telecontagem*. It represents a good testing location for NLOS urban conditions.

- **Estádio Universitário**

  Estádio Universitário is a location with clear LOS with the BTS (in mobile scenarios), also in urban conditions. It allows near-optimal conditions for the nodes (close to the BTS, clear line of sight). The distance to the BTS in straight line is 0.9 km. However, the fixed deployment location (for EDP-WiMAX project purpose) is not the same testing location as the mobile tests, since the EDP transformer station has some obstructions, representing an indoor NLOS location. In the EDP context, this location supports legacy telemetry applications and the *DTC/Inovgrid* application.

- **EDP- Rua do Brasil**

  EDP - Rua do Brasil represents an indoor location, with partial LOS with the BTS. This location is mainly used for the connection of the *TCMT* application field equipment with the EDP *Plesiochronous Digital Hierarchy (PDH)* Network. This location is connected to EDP PDH Network through an RS232-IP converter, accessible to the WiMAX Network. This converter is configured to act as server, allowing the client connections from the field RS232-IP converters.

- **Vila Nova**

  Vila Nova is a rural location with clear Line of Sight to the BTS. This testing scenario allows the analysis of WiMAX capabilities over long distances (20km) with line of sight. In the EDP context, this scenario is used to remote control and monitor the transformer station (*TCMT* Application).

The urban locations, represented by Quinta da Várzea, Estádio Universitário and EDP - Rua do Brasil, as well as the Polo 1 BTS, are shown in Figure 5.2 at the left side. The rural locations, represented by Vila Nova, as well as the Polo 2 BTS, are depicted in the same figure, at the right side.

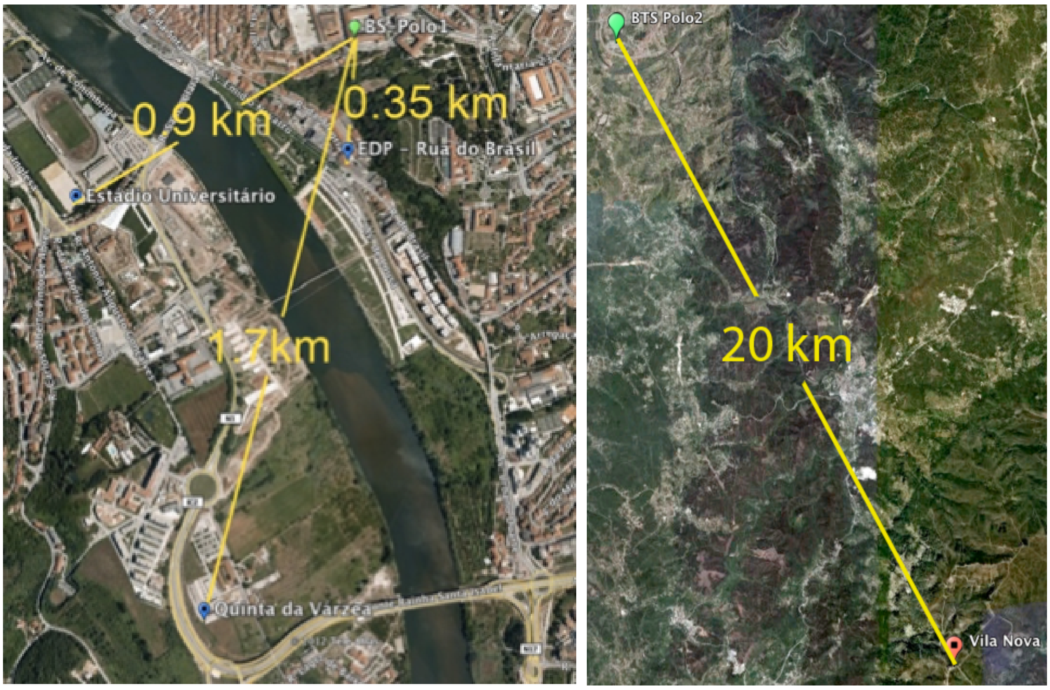


Figure 5.2: Testbed - Urban (left) and Rural (right) Locations

The tests were made with mobile (i.e., USB CPEs) and fixed (i.e., Indoor and Outdoor fixed CPEs) nodes. The mobile tests were made with mobile nodes, using two netbooks, each one equipped with an USB CPE. Although the mobile nodes used, these tests were conducted in the same locations, with no movement. The fixed nodes vary from outdoor and indoor CPEs supported by a fixed computer in each scenario. At the core network side, a fixed node was also used, connected to the *core* network via ethernet. All the nodes were running Debian Linux, where the mobile nodes were running a specific kernel version - 2.6.32-21, in order to operate correctly with the USB CPE equipment. The equipment used by the nodes is explained in Table 5.1.

Location	CPE	Processor	RAM	Disk
Parque Eólico Vila Nova	Outdoor	Pentium 4 @3.2 GHz	2 GB DDR2	250GB 7200RPM
EDP - Rua do Brasil	Indoor	Pentium 4 @3.2 GHz	2 GB DDR2	250GB 7200RPM
Quinta da Várzea (Mobile)	USB	Atom N450 @1.66 GHz	1 GB DDR2	160GB 5400RPM
Quinta da Várzea (Fixed)	Outdoor	Pentium 4 @3.2 GHz	2 GB DDR2	250GB 7200RPM
Estádio Universitário (Mobile)	USB	Atom N450 @1.66 GHz	1 GB DDR2	160GB 5400RPM
Estádio Universitário (Fixed)	Indoor	Pentium 4 @3.2 GHz	2 GB DDR2	250GB 7200RPM
Core Network	–	Pentium 4 @3.0 GHz	1 GB DDR2	120GB 7200RPM

Table 5.1: Nodes characteristics and locations

5.1.3 Test characteristics and specifications

To assess the WiMAX technology capabilities, a set of scripts and applications were needed. There are several network testing tools, however, each one has strong and weak points, depending on the objectives. Based on the output network parameters and related work, the D-ITG (Botta and Dainotti, 2007) tool was used, since it allows different types of emulated traffic, good precision in traffic generation, and also provides network parameters such as jitter, delay (Round Trip Time (RTT) and One Way Delay (OWD)), bitrate and packet loss. In order to measure the maximum throughput allowed in each scenario, as well as to generate background traffic, the tool Iperf (IPerf, 2011) was used.

All the tests were repeated between 5 and 10 times, each with a duration of 60 seconds, in order to correctly measure the network parameters, since there are many fluctuations caused by factors such as weather conditions and multi-path propagation. For all the results obtained the standard deviation was calculated with a confidence interval of 95%.

- Preliminary Tests

In order to do a pre-evaluation of the WiMAX network, a preliminary set of tests were conducted. These tests were made using three packet sizes: 128, 512 and 1024 Bytes. These packet sizes were chosen in order to validate the pilot deployment, and to analyze if the network behavior was as expected and known from previous works. The TCP and UDP protocols were used, either individual either at the same time (where 10 packets/s belong to each protocol). Only 1 flow at a time was tested.

Since these tests were used only for basic reference test, the results are not included because they are indirectly used in the reference tests.

P.Size	Pkts p/sec	Time	Nr.Runs	Transport Prot.	Nr. Flows p/ run
128, 512 and	20	60 sec	10	TCP, UDP	1
1024 Bytes	10 + 10	60 sec	10	TCP + UDP	1

Table 5.2: Preliminary Tests Parameters

For these tests, exceptionally, the nodes used were the USB and Indoor CPE, both located at the laboratory.

- Reference Tests

The second set of tests was the *reference tests*. The main goal of these tests was to assess the WiMAX network capabilities in multi-user/flows generic traffic conditions, where all traffic was transmitted over a Best Effort (BE) channel, without background traffic, in order to estimate the WiMAX network link availability and radio quality.

These tests had a packet rate of 20 packets per second, where each packet had a size of 512 bytes. The number of flows used was 1, 5, 10, 15 and 20 simultaneous flows, using TCP and UDP transport protocols.

The location of these tests was in Quinta da Várzea (NLOS) and Estádio Universitário (LOS).

The summary of these test conditions is shown in Table 5.3.

P.Size	Pkts p/sec	Time	Nr.Runs	Transport Prot.	Nr. Flows p/ run
512 Bytes	20	60 sec	10	TCP, UDP	1, 5, 10, 15 e 20

Table 5.3: Reference Test Parameters

The results were obtained through D-ITG flow sets, which worked as explained: Until 15 simultaneous flows only one flow set was used, with a maximum of 15 simultaneous connections (flows). For 20 simultaneous flows, four flow sets were used, each one with 5 connections. These approach was used to avoid the buffer overflow D-ITG application at the sender side.

- VoiP Tests

The synthetic tests, representing real applications, were essentially focused on Voice over IP (VoIP) traffic. Traffic generation was conducted to evaluate multi-users VoIP traffic over WiMAX. These flows emulated CODEC G.711.1 in a bi-directional call. The packet rate was 100 packets per second, with a compression rate of 96 Kb/s, as defined by CODEC G.711.1 (IEEE, 2008). The number of simultaneous flows was: 1, 5, 10, 20, 30, 40, 50 and 60 flows. For all the tests the same traffic generator seed was used, in order to generate similar traffic patterns across the tests.

In order to avoid the packet loss at the sender side, caused by application buffer overflow, each flow set had a maximum of 10 simultaneous flows, and, at each step, one more flow sets were used (e.g 1, 5 and 10 flows used only one flow set. 40 flows used four, and 60 used six simultaneous flow sets). The flow sets were started incrementally, and the measured period was when all flow sets were active, as shown in Figure 5.3.

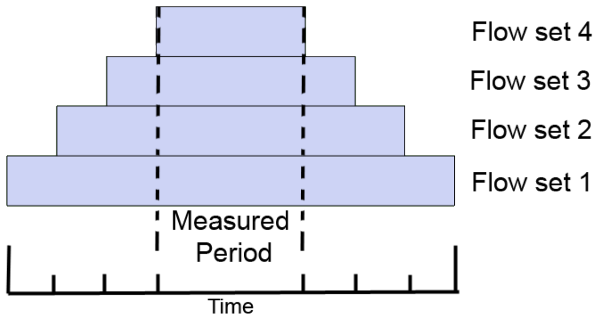


Figure 5.3: Measurement methodology

The VoIP traffic was transmitted over a Real Time Polling Service (rtPS) and BE channel, depending on the tests. Initially, this communication was planned to be over an Extended Real Time Polling Service (ertPS) or Unsolicited Grant Service (UGS) channel, which are best suited for VoIP traffic (since they guarantee the maximum allowed jitter and delay), but because of Alvarion hardware/software limitations[1], it was not possible to use these service classes.

The usage of rtPS mechanism was due to its guarantees, such as maximum tolerated OWD, reserved rate and traffic prioritization. The maximum tolerated OWD was set to 150 ms, as defined in ITU-T Y.1541 (ITU-T, 2003) and WiMAX Forum (Forum, 2008). The reserved rate was overestimated, in order to guarantee an higher rate than needed by all the flows, defined as 15 Mbps.

The VoIP tests were conducted in two phases. The first phase of VoIP tests was conducted on mobile nodes (i.e., USB CPE, with no movement) in LOS and NLOS access scenarios. The main target of this first evaluation was to observe the impact of the background traffic in the end-user Quality of Experience (QoE), while sending VoIP traffic over a prioritized rtPS channel and the background traffic over a non-prioritized BE channel.

The second phase of VoIP tests (i.e., USB CPE, with no movement), performed in the second semester, had the main objective of evaluating the efficiency of different WiMAX native QoS mechanisms and their impact on end-user QoE, with and without background traffic. For that, the VoIP traffic was sent on both prioritized rtPS and non-prioritized BE channels, using the D-ITG Tool, and the random background traffic was emulated using the

[1]These limitations are caused when setting the maximum reserved rate to high values (in order to allow a large number of flows), and, since this service classes are only prepared for low bandwidth usage (voice data), it was not possible to emulate more than 10 simultaneous voice calls.

tool IPerf (IPerf, 2011), simulating a channel obstructed with data from other users. The background traffic was generated in a non-prioritized BE channel, using all the bandwidth allowed by the BTS. Therefore, as referred, the main goal was to evaluate the efficiency of native QoS mechanisms of WiMAX, observing the behavior of VoIP traffic when the channel is obstructed, and the maximum supported users while maintaining good levels of QoE.

In summary, the VoIP tests to be performed (first and second phase) are as shown in Table 5.4.

First Phase
→ VoIP multi user scenarios → With and without background traffic → Using rtPS for VoIP traffic → Using BE for background traffic
Second Phase
→ VoIP multi user scenarios → With and without background traffic → Comparion between rtPS and BE (both used for VoIP traffic transmission) → Using BE for background traffic

Table 5.4: VoIP Tests - Summary

5.2 Results and Analysis

The results obtained in the different tests are described and analyzed in this section.

5.2.1 Reference Tests

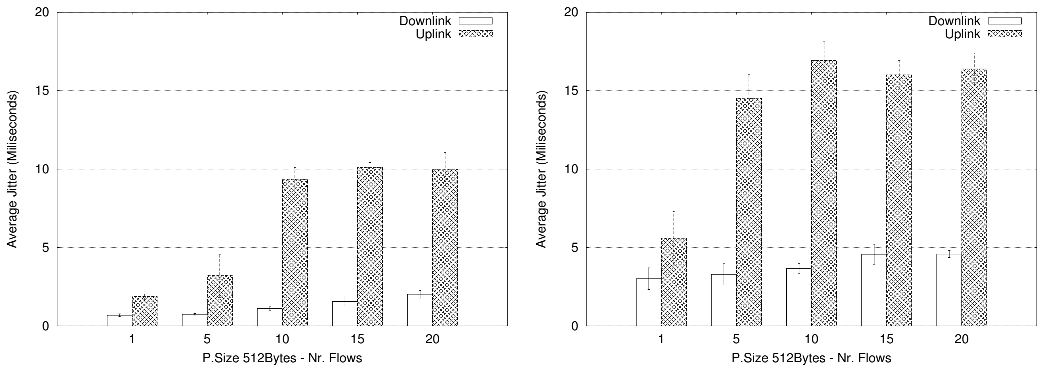
In order to assess the WiMAX technology in terms of multi-user support and link quality, a set of tests was conducted, with a generic traffic pattern, as explained previously. The main goal of these tests was to estimate the WiMAX capabilities in terms of multi-users environment and different line of sight conditions.

The description of the results is focused on QoS network parameters, such as jitter, round trip time and packet loss.

- Jitter

As shown on Figure 5.4, the jitter values vary between line of sight conditions, number of flows and uplink/downlink transmission.

In the uplink and in NLOS conditions, it is possible to observe slightly higher jitter values than in LOS. In the downlink, although with smaller differences, this behavior is also observable. It follows then that the line of sight is crucial to maintain a good signal quality.



(a) LOS - Estádio Universitário

(b) NLOS - Quinta da Várzea

Figure 5.4: Reference Tests - Jitter

However, given the NLOS conditions, where the signal is received through reflexions, the WiMAX technology proved to be reliable, ensuring average jitter values within the recommendations (ITU-T Y.1541 (ITU-T, 2003)), which defines 50 ms as maximum jitter value for real time applications.

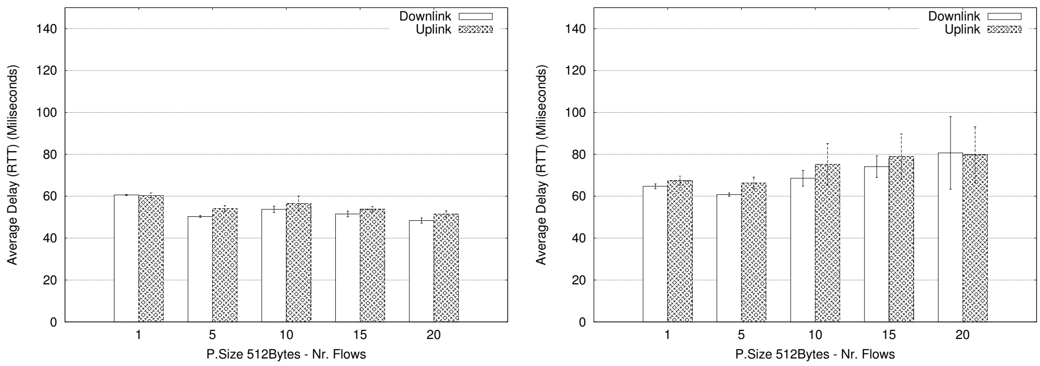
In all the conditions, either uplink or downlink and LOS or NLOS, the jitter values increase with the utilization of the channel, i.e., with the number of simultaneous flows.

One point to note, comparing the LOS and NLOS conditions, is the fluctuations of the values, represented by standard deviation (i.e., vertical lines on the graphs). It can be observed that in NLOS conditions the variations are greater, as well as the standard deviation. This means that, in line of sight conditions, the values are more constant and consistent along the several tests, unlike the non line of sight conditions.

- Round Trip Time

Figure 5.5 depicts a slight increase in the values of the delay in the NLOS conditions, when compared to LOS conditions. It is also noticeable that in NLOS conditions, as the utilization rate of the channel becomes higher, it will increase the fluctuations of RTT delay, both in downlink and uplink.

This behavior is due to the lower bandwidth available in NLOS, caused by the factors outlined above, such as multi-path and diffraction. The higher fluctuations in the uplink are also caused by the lower modulation schemes used. The fluctuations are quite small in LOS, as would be expected. The effect of the increase of simultaneous flows is only noted in NLOS conditions, since the bandwidth available is lower, and so, it becomes occupied with fewer flows.



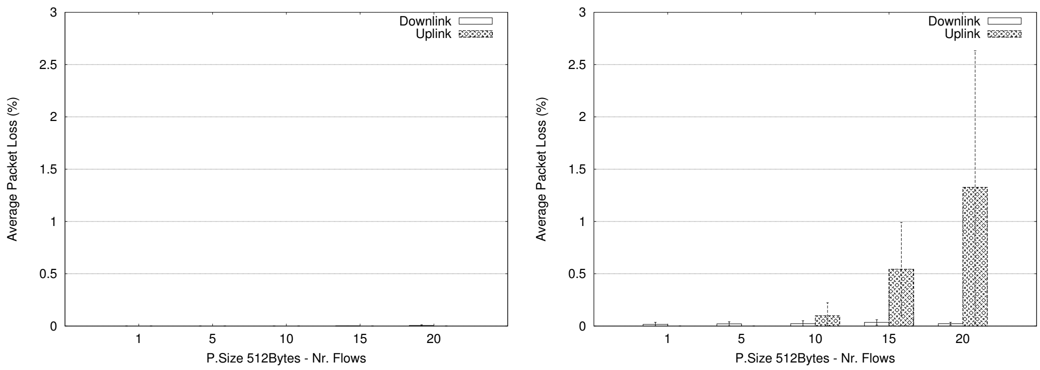
(a) Estádio Univ. (b) Qta. da Várzea

Figure 5.5: Reference Tests - RTT

In both cases, the average RTT delay does not exceed 90 ms, which are therefore very satisfactory results. Despite the recommendation (ITU-T Y.1541 (ITU-T, 2003)) only refers the OWD, the maximum value obtained, 90 ms, is even within the recommended values, which set as maximum OWD the value of 150 ms for real time applications.

- Packet Loss

The packet loss results are shown in Figure 5.6.



(a) Estádio Univ. (b) Qta. da Várzea

Figure 5.6: Reference Tests - Packet Loss

As expected, in LOS conditions there was no packet loss, even with 20 simultaneous streams. However, under NLOS, one can observe that the higher the utilization rate of the channel, the higher packet loss. Even with 10 simultaneous flows there is already some packets loss, and with 20 simultaneous flows this rate is already outside the ITU-T Y.1541 (ITU-T, 2003) recommendations, with a mean value of 1.5 %. The ITU-TY.1541 defines 1% as maximum packet loss rate for real time applications.

In the downlink, the same behavior is observed: greater losses when the use of the channel is higher. Nonetheless, with very low values, it is almost negligible. In the uplink, one can observe wide fluctuations in the values (represented by standard deviation - vertical lines), caused by the changes in the link quality and the factors referred previously, such as modulation scheme, link interferences and lower TX power (leading to worse Signal-to-Noise Ratio (SNR)).

Still, even in the worst conditions (i.e. NLOS and uplink), the packet loss rate did not exceed the 3% value.

Reference Tests Summary

Given the results and according to ITU-T Y.1541 (ITU-T, 2003), one can conclude that:

Jitter: Based on the maximum allowed jitter for sensitive applications, such as voice (50 ms), all the applications will be supported within all scenarios, since the maximum registered jitter was 20 ms.

Round Trip Time UDP: For the UDP Round Trip Time, since the maximum delay was 100 ms (where OWD would be about half (Cole and Rosenbubluth, 2001) - 50 ms), all scenarios enable the use of any type of applications.

Packet Loss: Given the recommendations, the maximum packet loss for classes 0-4 (i.e., delay sensitive applications) is around 1%. As such, except for the worst case (NLOS, Uplink), all applications can be used in the number of streams tested. Applications that use the TCP protocol (or similar) can be used in all scenarios, since there is no packet loss (also because TCP implements re-transmissions procedures to avoid the packet loss).

In the worst case, with 20 streams simultaneously in uplink and without line of sight, the packet loss rate of UDP packets is about 1.5%, which despite being a little above the recommendation is an acceptable value.

Conclusion: From the testes conducted, it can be concluded that the WiMAX link can support up to 20 simultaneous flows of generic traffic patterns in all the tested scenarios. This conclusion is based on the previously analyzed values, where the ITU-T recommendation for packet loss is exceeded in the uplink at NLOS scenario. The stop condition in the increasing of flows was based on the ITU-T recommendations (i.e., 1% of packet loss exceeded in the NLOS uplink scenario).

5.2.2 VoIP Tests - First Phase

The first phase of VoIP tests was conducted with two main goals. One was to evaluate the capacity of the WiMAX technology in supporting multi-users VoIP traffic in LOS and NLOS conditions. The other was to evaluate the WiMAX QoS native mechanisms when supporting the same environment as before, but with background traffic, emulating other users competing for network resources. This phase was conducted on mobile nodes (i.e., USB CPE, with no movement) and the employed methodology was to send VoIP traffic over a prioritized rtPS channel and the background traffic over a non-prioritized BE channel. The stop condition in the increasing of simultaneous flows for these tests was when the QoE levels decreased below MOS values of 2 in some of the scenarios.

This subsection is divided into three parts. The first analyzes multi-user VoIP traffic in NLOS and LOS conditions, while the second analyzes the same conditions but with background traffic. The last part depicts a summary of the first phase of VoIP tests.

LOS and NLOS - Without background traffic

This subsection includes the network and user evaluation results, such as OWD, Packet loss and Mean Opinion Score (MOS), related to LOS and NLOS scenarios without background traffic.

- One Way Delay

As Figure 5.7 shows, one can observe that in both conditions, NLOS and LOS, the OWD increases with the number of users, resulting from the increased occupancy rate of the channel. It is also possible to observe that the OWD for the NLOS scenario is always higher than the LOS scenario, since the signal only reaches the node through reflections (multi-path) and it is also in a shadow zone, caused by the buildings that location.

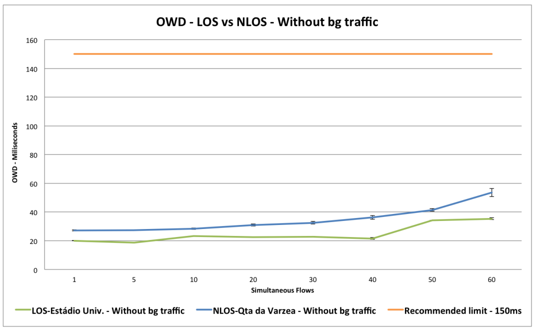


Figure 5.7: OWD - LOS and NLOS - Without background traffic

Still, in both cases, the OWD values obtained are well below the recommended limit

(ITU-T, 2003), which defines 150 ms as maximum OWD for real time applications, such as VoIP.

- Packet Loss

Figure 5.8 represents the packet loss (in percentage), either in uplink or downlink, for NLOS and LOS scenarios without background traffic.

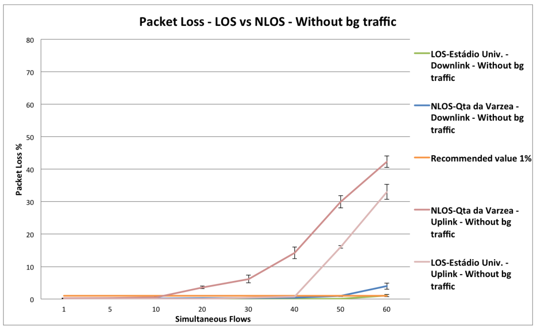


Figure 5.8: Packet Loss - LOS and NLOS - Without background traffic

One can observe that the packet loss is much higher in the uplink than in the downlink (in both scenarios), due to the lower transmission power (TX power) of the mobile node and to the UL/DL frame ratio (i.e., 60% of the symbols for the downlink and 40% for the uplink). In the worst scenario (NLOS and uplink), the packet loss rate exceeds the recommended limit (1%) with 10 simultaneous flows. In the LOS scenario, the uplink packet loss rate remains below the limit up to 40 users.

In the downlink, in both scenarios, the packet loss rate is lower than 1% until 50 simultaneous flows. With 60 flows, the values of NLOS are slightly higher than the recommendation.

- Mean Opinion Score

Figure 5.9 compares the MOS values obtained in the downlink, in LOS and NLOS, without background traffic.

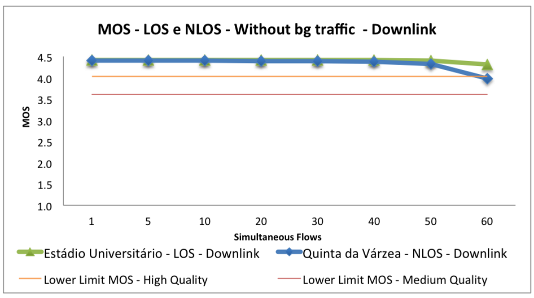


Figure 5.9: MOS - LOS and NLOS - Without background traffic - Downlink

In order to obtain the MOS values though network parameters, namely delay and packet loss, the calculations proposed in (Cole and Rosenbubluth, 2001) were used.

Through its analysis, one can consider that the voice quality achieved in communication, either LOS and NLOS, is excellent (MOS $\simeq$ 4.4) and remains constant up to 50 simultaneous users.

When the number of concurrent users varies between 50 and 60, there is a decrease in voice quality. This decrease is more pronounced in the NLOS scenario, given the worst conditions of receiving and sending the signal. Still, these values correspond to a high/medium voice quality (MOS $\simeq$ 4.0).

Figure 5.10 represents the MOS values obtained in LOS and NLOS, without background traffic, in the uplink.

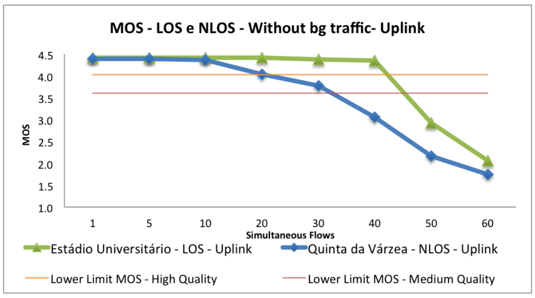


Figure 5.10: MOS - LOS and NLOS - Without background traffic - Uplink

From Figure 5.10, one can observe that the relation of voice quality obtained in the uplink, in LOS and NLOS, is not as constant as for the downlink traffic (Figure 5.9). While the voice quality in LOS downlink remained constant up to 60 simultaneous flows, in the LOS uplink the voice quality starts decreasing at 40 simultaneous flows.

For the uplink traffic in NLOS, there is also a major difference when compared to the results obtained in downlink ($\simeq$ 60 simultaneous flows). In this case, the MOS values obtained only represent acceptable voice quality up to 30 users.

In summary, one can conclude that, without background traffic, only 30 users are acceptable to the NLOS scenario and 40 users for the LOS scenario.

LOS and NLOS - With background traffic

This subsection includes the network and user evaluation results, such as OWD, Packet loss and MOS, related to LOS and NLOS scenarios with background traffic.

- One Way Delay

Figure 5.11 shows the OWD results obtained for LOS and NLOS scenarios with background traffic. In this Figure, it can be seen that the OWD results obtained in NLOS increase even with a low number of flows. Given the poor signal reception conditions and the high channel utilization (although in different flows, frames are shared between the nodes, since the mechanism used is Time Division Duplexing (TDD)), the QoS native mechanisms do not prove sufficient to address this load.

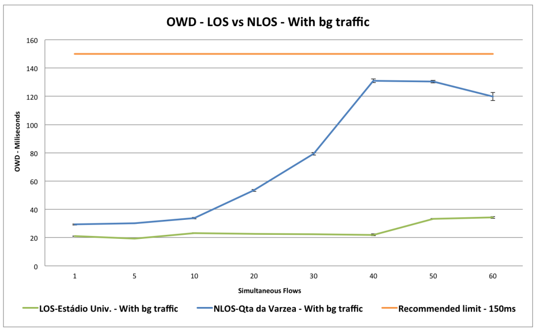


Figure 5.11: OWD - LOS and NLOS - With background traffic

Still, in both tests (LOS and NLOS) the results are below the recommended limit (150 ms). However, the packet loss rate values are quite high, as shown in Figure 5.12.

- Packet Loss

The Figure 5.12 represents the packet loss for uplink and downlink, in LOS and NLOS scenarios, with background traffic. As stated above, the poor signal reception and the higher usage of the channel lead to a higher packet loss rate and also to an increased delay.

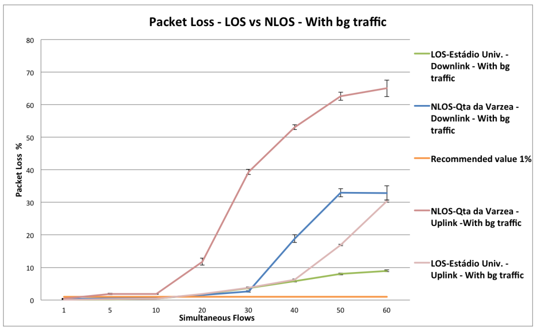


Figure 5.12: Packet Loss - LOS and NLOS - With background traffic

- Mean Opinion Score

As noted in the previous results, the packet loss rate and delay worsened with the use of background traffic. As the MOS calculation is related to these two parameters, there is an evident decrease in the MOS obtained with background traffic.

Figure 5.13 shows the MOS values obtained in the downlink, with and without line of sight and with background traffic.

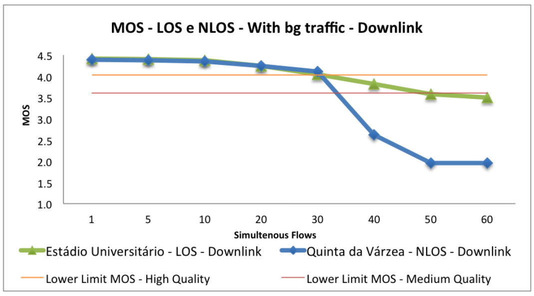


Figure 5.13: MOS - LOS and NLOS - With background traffic - Downlink

Compared with tests without background traffic, (in Figure 5.10) it can be seen that there is a greater deterioration of voice quality in the NLOS scenario. This deterioration is reflected in the ability to support concurrent users, about 50% less than without background traffic.

For the LOS scenario, despite the more stable results, they also suffer a deterioration of voice quality when there is background traffic. Despite the deterioration of voice quality, the results average a good quality (MOS $\simeq$ 3,6) up to 50 users.

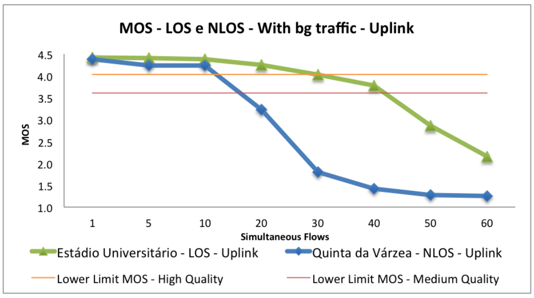


Figure 5.14: MOS - LOS and NLOS - With background traffic - Uplink

For the uplink traffic, which can be observed in the Figure 5.14, the degradation of voice quality was lower. Yet, in NLOS scenario, it reduced the number of supported users with medium/high quality from 30 to about 15. In LOS scenario, while maintaining the same number of users, the MOS values went down from excellent to medium/high.

In summary, one can conclude that, with background traffic, only 10 users are acceptable to the NLOS scenario and the same 40 users for the LOS scenario, but with lower voice quality than in the previous tests - without background traffic.

VoIP Tests - First Phase - Summary

After performing the tests and analyzing the results obtained, the native QoS mechanisms of WiMAX do not show the desired efficiency in a congested channel in NLOS scenarios, although in the LOS scenario they have shown a notable efficiency. Therefore, to complement this analysis, the second phase of VoIP tests was conducted, as described in the next section.

Regarding the number of simultaneous users, it is concluded that, without background traffic (i.e. without the link being congested) 40 simultaneous users are supported, both in LOS and NLOS. When the link is congested (i.e. with background traffic), it is concluded that in LOS scenario about 30 simultaneous users are possible, while in the NLOS scenario this number is reduced to 10 simultaneous users.

As these tests represent bi-directional calls, both uplink and downlink values have been considered to evaluate the WiMAX capacity.

5.2.3 VoIP Tests - Second Phase

The second phase of VoIP Tests aimed to achieve three main goals. The first consisted on the evaluation of the native WiMAX QoS mechanisms and their impact on the end-user QoE, using different WiMAX service classes, such as rtPS and BE. The second objective was to observe the impact of different line of sight conditions in both QoS and QoE metrics. Finally, this work aimed to address the effect of background traffic in the end-user perceived quality, as well as in the WiMAX network global performance.

In this phase, the VoIP traffic was sent on rtPS and BE channels using the D-ITG Tool, and at the same time, random background traffic was emulated using the tool IPerf (IPerf, 2011), simulating a channel obstructed with data from other users. The background traffic was generated in a BE channel, using all the bandwidth allowed by the BTS. Therefore, as referred, the main goal was to evaluate the efficiency of native QoS mechanisms of WiMAX, observing the behavior of VoIP traffic and the end-user perceived QoE when obstructing the channel with background traffic. The stop condition in the increasing of simultaneous flows for these tests was when the QoE levels decreased below MOS values of 2 in some of the scenarios.

This work led to the realization and submission of one paper to the 2nd IEEE Baltic Conference on Future Internet Communications 25-27 April 2012, in Vilnius, Lithuania. This paper was accepted and presented, and is included in Appendix A - VoIP performance over Mobile WiMAX: An Urban Deployment Analysis.

The results are presented and discussed in the following subsections.

LOS Scenarios

In the next subsections the results obtained in LOS scenarios are shown and explained.

Packet Loss without BG traffic

Figure 5.15 depicts the packet loss without background traffic in LOS conditions. The packet loss values are comparable in both service classes.

In the uplink, the first noticeable impact of the packet loss is observed when transmitting more than 30 simultaneous flows. With 30 or less simultaneous flows the packet loss remains below 1%, as advised in ITU-T Y.1541 (ITU-T, 2003) recommendation. When transmitting 40 simultaneous flows, the uplink packet loss is 1% for BE and 0.8% for rtPS. With 50 simultaneous flows, these values are, respectively, 9% and 9.8% for BE and rtPS. In the downlink, the packet loss values remain below 1% with all simultaneous flows, for both service classes. From these results, it is possible to observe that the packet loss increases with the number of simultaneous flows, since the bandwidth usage becomes higher. The differences

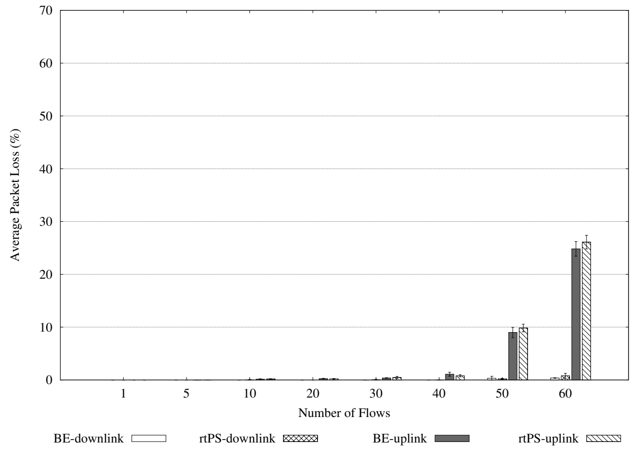


Figure 5.15: Packet Loss without background traffic in LOS conditions

between uplink and downlink are due to USB CPE transmission power limitations and the uplink/downlink testbed frame ratio configuration.

Packet Loss with BG traffic

Figure 5.16 depicts the packet loss with background traffic in LOS conditions. In this case, the maximum feasible packet loss limit (i.e., less than 1%) is achieved with 20 simultaneous flows. For 30 simultaneous flows, packet loss rate is already 4%, which is 3% beyond the acceptable limit.

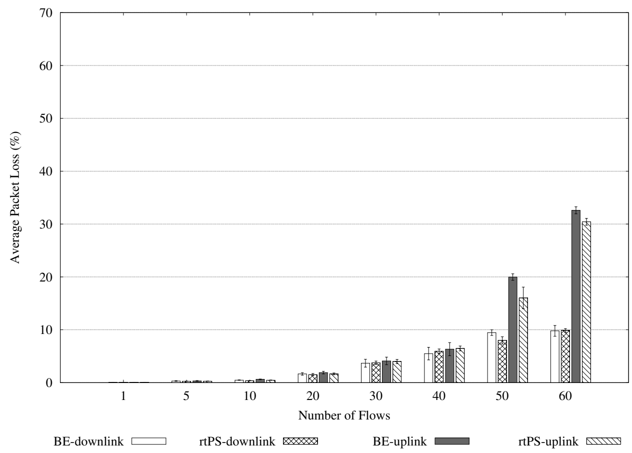


Figure 5.16: Packet Loss with background traffic in LOS conditions

The packet loss in the presence of background traffic is higher than when just VoIP traffic is being transmitted (Figure 5.15). These values are explained due to the high traffic load on the link, caused by background traffic. The rtPS service class effectiveness is perceived, but only marginally.

MOS without BG traffic

Figure 5.17 shows the MOS without background traffic in LOS conditions.

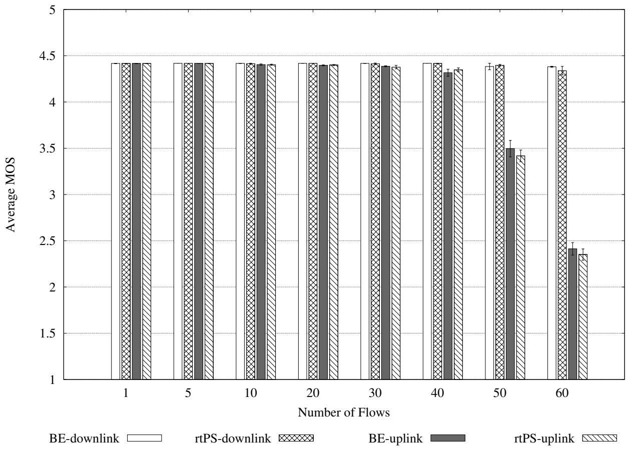


Figure 5.17: MOS without background traffic in LOS conditions

It demonstrates that without background traffic in LOS conditions it is possible to support up to 50 flows with *good* quality (i.e., higher than 3). In this scenario, rtPS is slightly worse than BE, which can be caused by the QoS scheduling mechanisms overhead.

MOS with BG traffic

The MOS with background traffic in LOS conditions is depicted in Figure 5.18. It is noticeable that the rtPS service class has some advantages when compared with BE. It guarantees *good* quality up to the same 50 flows as without background traffic (Figure 5.17), while the BE service class only supports up to 40 simultaneous flows.

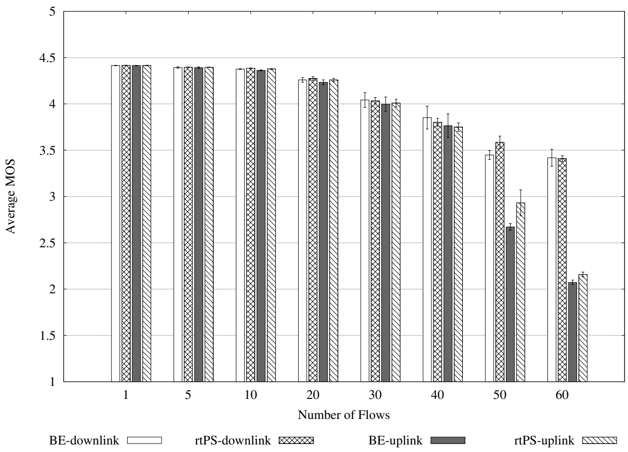


Figure 5.18: MOS with background traffic in LOS conditions

In Line of Sight (LOS) conditions with background traffic, the differences between downlink and uplink are not so noticeable, since both communication directions are congested.

NLOS Scenario

This subsection presents the results concerning assessment performed in Non-Line of Sight conditions.

Packet Loss without BG traffic

Figure 5.19 shows the packet loss rate without background traffic in NLOS conditions.

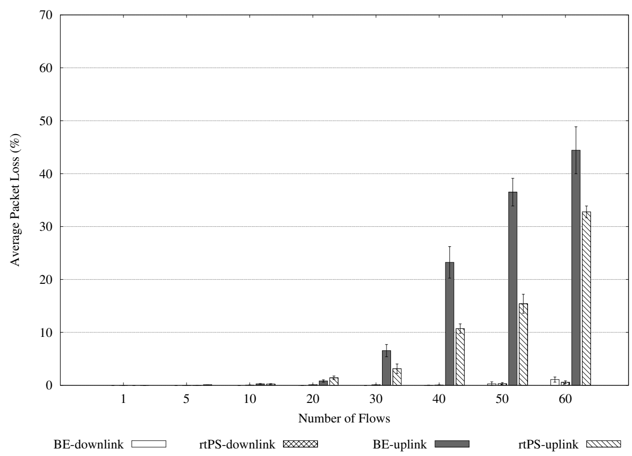


Figure 5.19: Packet Loss without background traffic in NLOS conditions

This figure shows that WiMAX supports 30 simultaneous flows under 10% for BE service class and 40 simultaneous flows under 10% for rtPS. In this scenario, for the same number of simultaneous flows, the packet loss rate is higher than in LOS scenario (Figure 5.15). This fact is caused by the lower bandwidth available due to the worst signal quality and the low CPE transmission power. Also, in this scenario, the differences between BE and rtPS are more noticeable.

Packet Loss with BG traffic

Figure 5.20 shows the packet loss percentage with background traffic in NLOS conditions. The differences between rtPS and BE are visible, but with minor differences.

Figure 5.20 shows that in NLOS conditions with background traffic it is possible to support, under 10% of packet loss, up to 20 simultaneous flows for BE service class and 30 simultaneous flows for rtPS. This represents a lower capacity for supporting several flows when compared to Figure 4, where the packet loss values, for both uplink and downlink and rtPS and BE, were below 10% until 40 flows. The worst obtained values are in this scenario, as expected, since it represents the NLOS access conditions with a fully loaded channel.

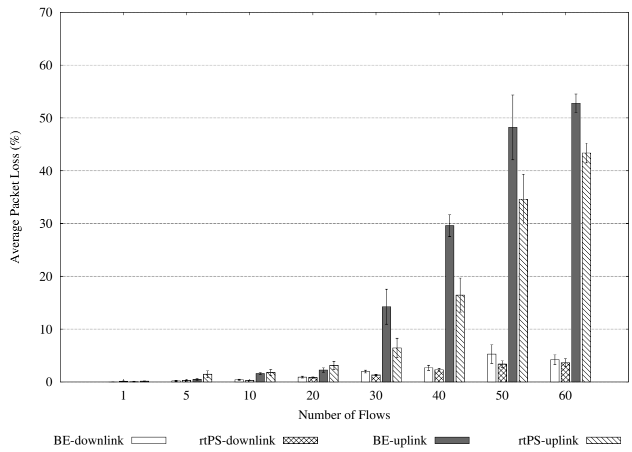


Figure 5.20: Packet Loss with background traffic in NLOS conditions

MOS without BG traffic

The Mean Opinion Score metric with background traffic in NLOS access condition is presented in Figure 5.21.

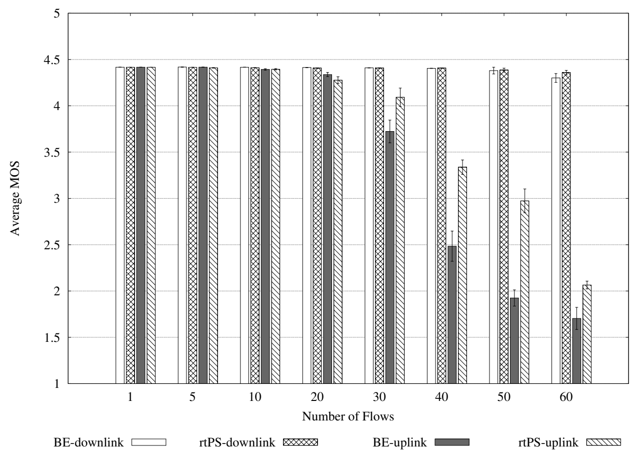


Figure 5.21: MOS without background traffic in NLOS conditions

In this scenario, the WiMAX technology was able to support up to 50 simultaneous flows using the rtPS service class, while with BE only 30 simultaneous flows are sustained. This well sustained number assumes the expected *good* conditions to the end-user perceived quality (i.e., QoE). When comparing the NLOS and LOS scenarios without background traffic, (Figure 5.21 and Figure 5.17) the MOS values start decreasing at 40 simultaneous flows in LOS, while in NLOS, it starts decreasing at 20 flows. In short, the rtPS service class supports, in both NLOS and LOS, up to 50 simultaneous flows with *good* quality. The BE class supports up to 50 flows in LOS and, in NLOS, supports a maximum of 30 flows.

MOS with BG traffic

Figure 5.22 depicts MOS values with background traffic in NLOS conditions. In this scenario, WiMAX supports up to 40 flows in the rtPS service class, while in BE it supports up to 30 simultaneous flows (with *good* conditions). This represents a decrease of 10 flows for rtPS, when compared to the previous scenario, without background traffic. Also, the differences between rtPS and BE are clearly shown, where rtPS supports more 10 flows than BE.

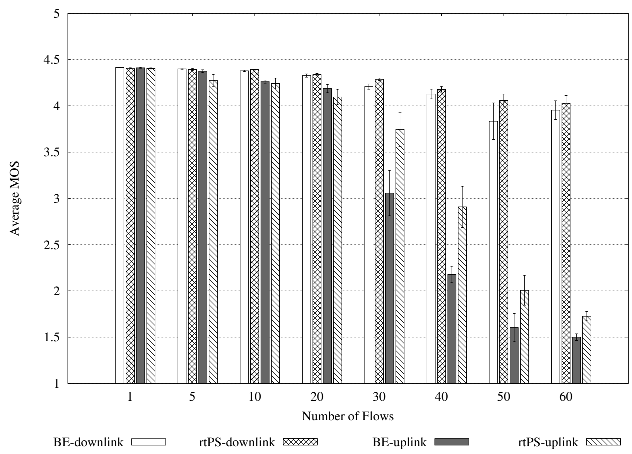


Figure 5.22: MOS with background traffic in NLOS conditions

Comparing the NLOS and LOS scenarios with background traffic, Figure 5.21 and Figure 5.22, the MOS values start decreasing at 20 simultaneous flows in LOS, while in NLOS it start deceasing at 5 flows. When comparing rtPS service class with BE, rtPS supports up to 40 simultaneous flows in NLOS and 50 in LOS, while the BE in LOS supports up to 40 flows and in NLOS supports a maximum of 30 flows.

VoIP Tests - Second Phase - Summary

This analysis demonstrated the efficiency of the WiMAX QoS mechanisms. It has been shown that the rtPS service class always has a better performance than BE, although in LOS conditions this difference is barely noted. The differences between rtPS and BE can result in less 20 simultaneous flows well sustained.

The results also show the benefits of accessing to the network in LOS when compared with NLOS conditions. This is an expected result, since the NLOS scenario offers much more interferences than the LOS scenario. Mostly due to the low USB CPE transmission power, leading to worse Signal-to-Noise-Ratio (SNR) values, and the factors that arise in NLOS scenarios, such as multi-path fading and shadowing, the NLOS values are worse than in LOS.

The rtPS service class, without background traffic, supports the same amount of flows either in LOS or NLOS (50 simultaneous flows), while with background traffic it decreases from 50 simultaneous flows in LOS to 40 flows in NLOS. The BE class without background can support up to 50 flows in LOS and 40 flows in NLOS. With background traffic, this number decreases to 40 flows in LOS and to 30 flows in NLOS. These values are assuming a *good* voice quality.

These results clearly depict the differences between LOS and NLOS, with and without background traffic and the importance of the correct service classes employment. The impact of background traffic in the end-user QoE is noticeable in both LOS and NLOS access scenarios, resulting in the decreasing of well supported simultaneous flows.

5.3 An Overview of EDP-WiMAX Project Tasks

During the project progress many tasks have been conducted in order to test the different EDP Applications. These tasks are described in this section. The Appendix B - EDP - WiMAX Pilot - Final Report represents the final project report delivered to EDP, containing all the main information and results related to the project.

- Project Testbed Deployment

The testbed deployment was a constant activity, since the equipments were not deployed at the same time. It firstly involved the Base Transceiver Station (BTS) deployment at Polo 1 - Department of Physics - and Polo 2 - Informatics Engineering Department - and also the deployment of the different CPEs - Vila Nova, Quinta da Várzea EDP - Rua do Brasil and Estádio Universitário. Besides the deployment of WiMAX network equipment, it was necessary to deploy EDP equipment, which involved numerous site visits with the EDP teams.

Involved team members: João Henriques,Vitor Bernardo.

- D-ITG Application and Scripts

In order to evaluate the different applications over the WiMAX network, it was necessary to emulate those applications with a traffic generation tool. For that, D-ITG (Botta and Dainotti, 2007) was used, since it supports several types of traffic and it is possible to specify the traffic generation conditions, such as application types, packet size, inter departure time, packet rate. It also provides several QoS parameters, such as delay, jitter and packet loss used to estimate the end user Quality of Experience (QoE). These tests and scripts were referred in section 5.1.3. The configuration referred also involved the preparation of the nodes, such as operative system install and configuration.

Involved team members: João Henriques.

During this work there were some problems with the D-ITG tool and the hardware/software WiMAX, as explained below:

- Traffic Signaling Protocol (TSP) problem:

  The TSP encapsulated the packets with the source and destination address. When the receiver, behind Network Address Translation (NAT), received these packets, it tried to create a socket back to source address but it was binded at destination address. Since the machine was behind NAT, the destination address was different than the real machine address (i.e., destination address: 10.10.2.100; Real machine IP: 192.168.1.5), and so, an exception was thrown. To fix this problem, it was necessary to change the application source code and re-compile it, and perform a set of tests in order to validate such change. During this process the authors of D-ITG were contacted and informed, and, a new version of D-ITG was released at the same time, which corrected this limitation.

- Packet Size problem:

  When testing different packet sizes we found a limitation of the WiMAX hardware. It happens that WiMAX could not fragment and send to the CPE User Datagram Protocol (UDP) packets with a packet size greater than 1436 Bytes. This problem had no solution since it was a hardware specific problem.

- Signal Quality Measurement

In order to evaluate the WiMAX Signal Quality two scripts were made. These scripts were created in collaboration with Vitor Fonseca, with the objective of retrieving signal parameters (i.e Received Signal Strength Indication (RSSI) and Signal-to-Interference plus Noise Ratio (SINR)) from the different CPEs. The programming language used for these scripts was Python and Bash scripting.

Involved team members: João Henriques, Vitor Fonseca.

- Remote Control - RS232 Tasks

To achieve interoperability with some EDP legacy equipments, it was necessary to use RS232-IP converters (i.e MOXA NPORT 5100A(Nport, 2011)).

Since the EDP equipment had particular specifications, the default operation mode of converters and null modem cables were not a solution. It was necessary to create a set of RS232 cables with specific pinouts in order to connect the equipments to the converters, and therefore to the WiMAX Network. Several meetings were conducted to specify and analyze the converters behavior and a set of documents and cable specifications were made. The cable specification can be found in Appendix B - EDP - WiMAX Pilot - Final Report

Involved team members: João Henriques, Vitor Bernardo.

- Link monitorization - Availability and Delays

In order to analyze the WiMAX links availability and delays during several months, some scripts were created to be constantly running. These scripts executed a *ping* command each 5 minutes. This configuration was set up because of its similarity with the EDP applications (random single communications), and to avoid to flood the network with ping requests and replies. The network parameters obtained from these scripts were the RTT and the packet loss values.

Involved team members: João Henriques.

5.3.1 EDP Applications

With the main goal of assessing the WiMAX technology capabilities in the support of EDP Applications, each application was analyzed in terms of traffic patterns, bandwidth usage and sustained delays. The specification of each application was one of the requirements of the EDP-WiMAX project. This analysis was conducted with a packet analyzer tool, namely TCPDUMP (TCPDUMP, 2011), by capturing all the packets transmitted by the network to the field equipments.

Each EDP application traffic and link specification is described below, accompanied by the traffic patterns analysis.

The applications characterization was conducted during the second semester, caused by the delay in the EDP applications deployment. This delay in the deployment was due to various constraints in the conciliation of the various EDP teams availability, as well as the need for adaptation and re-configuration of some applications in order to make them work properly over IP.

It should be noted that it is the first time that these applications work over IP, through the RS232-IP converters, which represents a general solution for any IP network, not being specific for the WiMAX technology.

All the work related to EDP is also described in Appendix B - EDP - WiMAX Pilot - Final Report, which represents the final report delivered to EDP, containing all the informations and tests related to the EDP-WiMAX project. All the tasks described below are included and deeply analyzed in the referred appendix. The referenced document was restructured and rewritten during the second semester, including the new applicational, availability and simultaneous tests results.

- Medium Voltage Remote Operations (EDP: Telecomando):

  The real application traffic is generated in Alto de S. João, in Coimbra, to the field Remote Terminal Unit (RTU), located in Quinta da Várzea and Vila Nova. The traffic is transmitted over an ertPS channel, since there are high delay requirements. The generated traffic is bi-directional.

  Through the analysis of the transmitted traffic, namely the communication between the RS232-IP converters, it was possible to enumerate the traffic characteristics, as described below.

  Encapsulated packets communication between converters:
  - Average packet size and rate: 20 packets of 60 Bytes per second.
  - Periodicity: Constant
  - Transport protocol: TCP
  - Delay requirements: High (below 500 ms)

  The described traffic pattern represents an average of 9600 transmitted bits per second. In terms of bandwidth support, the WiMAX technology could sustain between 500 and 800 simultaneous applications, and so, the scalability does not represent a concern for this specific application.

- Transformer station interconnectivity for control, monitoring and telemetry (EDP: DTC/Inovgrid):

  In the project context, the real application traffic is generated by LOGICA server and transmitted to the field DTCs, located in Quinta da Várzea and Estádio Universitário. The traffic is also transmitted over a Non-Real Time Polling Service (nrtPS) channel, since there are no delay and bandwidth requirements. The generated traffic is bi-directional.

  Through the analysis of the captured traffic, it was possible to characterize the DTCs traffic patterns, as described below.

  Gathering DTC identifier:
  - Average packet size and rate: 1200 Bytes per request.
  - Periodicity: Hourly
  - Transport protocol: TCP
  - Type of request: Web-service call (Remote Procedure Call)
  - Delay requirements: Low
  - Observation: Non critical data. The equipment supports re-transmission and it is natively designed for IP communication.

  Gathering DTC load diagrams:
  - Average packet size and rate: Variable, depending on number of records

- Periodicity: Daily
- Transport protocol: TCP
- Type of request: Web-service call (Remote Procedure Call)
- Delay requirements: Low
- Observation: Non critical data. The equipment supports re-transmission and it is natively designed for IP communication.

The scalability support for this application is variable, depending on the number of records that each equipment reports. However, based on the analyzed traffic patterns, the scalability is not a concern, due to the low bandwidth and delay requirements.

- Work Force Management and Mobile Teams Management (EDP: WFM / GME):

  The real application traffic is generated between the WFM EDP servers and the field equipments (i.e., tablets and Personal Digital Assistant (PDA)), at any place within the WiMAX network range. The packet capture is conducted between the core network and the field locations. The traffic is transmitted over an nrtPS channel.

  Because of the relevant importance of this application for EDP, a deeper analysis was conducted. This analysis involved the enumeration of the different steps to access the application, as well as the traffic description and specification of each of those steps. This specification can be found in Appendix B - EDP - WiMAX Pilot - Final Report.

- Transformer station interconnectivity for telemetry with legacy equipments (EDP: Telecontagem):

  The real application traffic is generated by EDP servers at LOGICA and transmitted to the field equipment, located in Quinta da Várzea and Estádio Universitário. The packet capture is conducted between the core network and the field locations. The traffic will be transmitted over an rtPS channel, since the application has low delay requirements.

  Due to EDP constrains, this application was not completely deployed. The field equipments and WiMAX configuration were fully configured, however, the EDP access to the WiMAX network was missing, preventing the possibility of sending data for the field equipments. Thus, it was not possible to analyze the application traffic patterns.

5.4 Summary

After the completion of the previously described tests, it was possible to get a better insight into the behavior of WiMAX technology, as shown below.

Through the different tests it is possible to observe that the line of sight conditions are very important in WiMAX communications. The wireless propagated signals (i.e., waves) can be affected by factors such as:

- Multi-path fading: This term means the fading and the various paths of the signal, caused by the reflection and deflexion of the signal through its way to the node. These reflections and deflections are usually caused by obstacles (e.g., buildings).
- Shadowing: Shadowing means the effect caused by the orographic conditions (i.e., buildings, land characteristics) that puts the node into a *shadow* zone, without line of sight with the BTS. This effect leads to poor signal reception.
- Diffraction: The change in the signal, caused by obstacles in its path to the node, is called diffraction. When the propagated signals crosses the obstacles, the propagated waves can suffer diffractions.

These factors can lead to signal instability and fluctuations, which will also deteriorate the network access quality, which is observable through the differences between LOS and NLOS results. Several network parameters are affected, such as RSSI, SINR, jitter, delay, bit rate and packet loss.

Also, the difference from downlink to uplink is commonly visible in the graphs. This difference is usually caused by:

- Frame Structure: In testbed configuration, the ratio between uplink and downlink for the frame construction is defined as 40% for uplink and 60% for downlink (TDD).
- Modulation Scheme: The modulation scheme used by the CPEs must be considered. Since the BTS is configured with AMC, it will adapt the downlink and uplink modulation schemes according to the link conditions. A more complex scheme (e.g 64 QAM) is usually used in good channel conditions.
- Transmission Power: The transmission power of the CPEs is also a concern. The mobile CPEs, such as USB, have a lower TX power than the Outdoor or Indoor CPE, since it has no power source and it is designed to have a good ratio between energy consumption and signal quality obtained.
- MIMO: The testbed is configured with MIMO Matrix A, meaning that both BTS antennas send the same data stream, increasing the downlink conditions for the CPEs. However, the CPEs only send data with one antenna. This also leads to better performance on the downlink than on the uplink.

5.5 Conclusion

This chapter has shown the results obtained in different tests for assessing WiMAX technology. This technology proved to be efficient in supporting multi-user access in different conditions (LOS and NLOS), both with and without background traffic.

These tests allowed to evaluate the WiMAX technology in the support of generic and voice applications through the emulation of real traffic patterns, such as VoIP, with and without background traffic. It was possible to fill the gap of the previously analyzed works, such as: Bernardo et al. (Bernardo et al., 2009b), which evaluated the VoIP transmissions in a Fixed WiMAX testbed without assessing the impact of different WiMAX QoS mechanisms and background traffic; Durantini et al. (Durantini et al., 2008), which evaluated different access scenarios and traffic flows, but considering only the QoS parameters.

This work performed an analysis of the end-user perceived voice quality (QoE) while comparing different WiMAX QoS service classes with and without background traffic, assessing the maximum supported users in a real Mobile WiMAX testbed within different levels of QoE. Different access conditions were considered, such as LOS and NLOS, using real end-user equipment, such as USB CPEs and a real deployed WiMAX network.

This chapter also demonstrated the different tasks carried out under the EDP-WiMAX project. Not only the testing tasks, but also the field tasks, were very important in the understanding, knowledge and applicability of WiMAX equipment with the usage of real applications. These tasks allowed a deeper understanding of WiMAX technology, EDP equipments and applications and network testing tools, and, at the same time, allowed to work with teams and people in a real scenario and in a real project.

6. 4G Technologies Assessment by Simulation

In order to complement the previous study of a real Worldwide Interoperability for Microwave Access (WiMAX) testbed with a simulation analysis, the OPNET (OPNET, 2012) simulator was used during the second semester.

In terms of technological comparison, several identical scenarios were created with the two Fourth Generation of mobile phone standards and technology (4G) competing technologies, namely WiMAX and LTE, allowing a comparison study between both technologies by simulating several multi-users Voice over IP (VoIP) scenarios.

Such comparisons are very important to allow a performance evaluation between the different technologies, simulating real usage scenarios, in particular, multi-users VoIP scenarios with and without background traffic. By analyzing the resulting network Quality of Service (QoS) and Quality of Experience (QoE) parameters, it is possible to attain the knowledge of the network capabilities in the support of this kind of applications.

The scenarios and tests conducted are described in subsequent sections.

As mentioned above, the simulator chosen for the tests was the OPNET. The choice of this simulator was based on its potential and versatility in the creation and adaptation of scenarios, as well as its international reputation, observable in several published works.

The OPNET simulator allows the differentiation of the granularity of the simulation, through the specification of the simulation mode. In the case of the WiMAX technology, the OPNET allows to specify four simulation modes: *Efficiency Enabled*, which only considers the Medium Access Control (MAC) layer, and it is suitable for capacity planning; *Framing Module Enabled*, considering the MAC layer frame-by-frame, being most suitable for the analysis of QoS mechanisms for network planning and deployment in terms of network capacity; *Physical Layer Enabled*, considers the previously MAC layer frame-by-frame and also the PHY layer, used when the main objective is to analyze the physical layer effects, such as pathloss, multipath and interference, but also comprising the network and QoS planning. The last simulation mode is the *Mobility and Ranging Enabled*, which has the same features as the *Physical Layer Enabled* mode, with the ability to model and simulate mobility scenarios.

LTE features two modes of simulation (called efficiency modes): *Physical Layer Enabled*, which allows the simulation considering only the PHY layer, and therefore considering the interferences, pathloss and multi-path. *Efficiency Enabled*, similar to the framing module enabled of the WiMAX, which considers the delivery of packets directly to the destinations without considering the PHY layer and the adjacent effects (e.g., interference between nodes, multipath, etc.). The latter mode allows the assessment of the technology in terms of scheduling and prioritization, avoiding the considerably heavier calculations of the PHY mode enabled.

6.1 WiMAX Simulation

This section explains and demonstrates the WiMAX scenarios and results, obtained through simulation. This assessment aims to achieve the same goals as the previously conducted study in a testbed, but this time in a simulation scenario. Through the comparison of testbed and simulation evaluations it will be possible to perform an analysis and comparison of the different test scenarios.

6.1.1 Scenarios Setup

In order to replicate the testbed scenario in a simulation environment, several simulation tests have been specified with similar configurations. Although it is very difficult to replicate the scenarios as they really are in the testbed, the most important parameters were set, such as the simulation mode, the antenna gain and transmission power (Base Station (BS) and the nodes), the location of the nodes in the scenario, the generated traffic, among others.

Figure 6.1: WiMAX Simulation - Tesbed scenario to replicate

As in the real testbed, two WiMAX nodes were considered: One node that sustains all the VoIP flows and other node that sustains the background traffic. The arrangement of these nodes was considered as the Line of Sight (LOS) and Non Line of Sight (NLOS) scenarios in real testbed (as in Figure 6.1), and it can be seen in Figure 6.2.

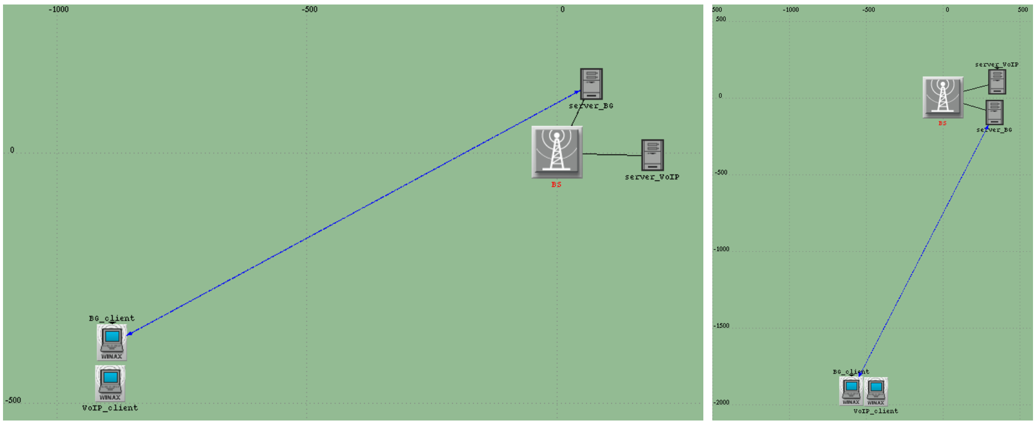


Figure 6.2: WiMAX Simulation - LOS Scenario (left) and NLOS Scenario (Right)

In the LOS scenario, the nodes were placed at 0.9 km from the BS, with the *freespace* pathloss model and without *multi-path*, representing ideal line of sight conditions, like the real testbed scenario. In the case of NLOS scenario, the nodes were placed at 1.7km from the BS, using the *EREC Fixed Suburban* pathloss model and the *ITU-T Pedestrian B* multi-path model. The choice of the *EREC Fixed Suburban* model with *Terrain Type B* was due to the *WiMAX System Evaluation Methodology* (Forum, 2008) recommendation for suburban scenarios with the pretended orographic conditions. The choice of the *ITU-T Pedestrian B* multi-path model was also due to the *WiMAX System Evaluation Methodology* recommendations, which recommends this multi-path model for low mobility pedestrian scenarios.

The simulation model used was the *Physical Layer Enabled*, in order to consider the variations in the signal quality, interferences and multi-path, as well as the variations related to MAC layer scheduling and traffic prioritization (i.e., WiMAX QoS mechanisms).

The frequency band used in the simulation tests was also similar to the real scenario. The center frequency set was 2615 MHz, using TDD with 10 MHz of bandwidth, using a frame configuration with 60% of symbols for the downlink and 40% in the uplink. It should be noted that the simulator Multiple Input Multiple Output (MIMO) settings only affect the signal diversity, and, therefore, it allows higher levels of SNR and it does not impact the ability to receive or transmit data (similar to MIMO Matrix A, used in the testbed).

The detailed configuration of the simulation environment is represented in Table 6.1.

As the generated traffic in testbed scenario, the voice codec utilized was the G.711.1, with a packet rate of 100 packets per second and a compression rate of 96Kb/s, as defined by CODEC G.711.1 (IEEE, 2008). Several simultaneous flows were simulated, where each flow is intended to emulate one real bi-direction call (i.e., one user).

	Parameter	WiMAX – simulation and testbed
Simulator Parameters	Frequency Band	2.615 GHz
	System Bandwidth	10MHz
	FFT Size (Subcarriers)	1024
	Duplex Mode	TDD
	MIMO technique	MIMO – Matrix A (4x2)
	Coding Scheme	Convolutional Turbo code (CTC)
	Modulation Format	64 QAM
	LOS pathloss / multi-path model - Simulation	Freespace / none
	NLOS pathloss / multi-path model - Simulation	Erceg Suburban Fixed, terrain type B / ITU-T Pedestrian B
	Frame Size	5 ms
	Subcarrier Spacing	10.94 kHz
	DL / UL sub-frame	29 symbols (60%) / 18 symbols (40%)
Application Parameters	Simulation time (per run)	120 seconds
	Call time (per run)	60 seconds
	Call start time	Uniform distribution (10 to 30 seconds)
	Run repetition	10 repetitions
	VoIP CODEC	G.711.1 with 96Kb/s compress rate

Table 6.1: WiMAX Simulation Parameters

Each test is repeated through 10 runs to avoid the outlier results caused by signal fluctuations and BS scheduling mechanisms variations, where each test used a different seed. The different seeds were used to generate different traffic patterns across the tests, as well as different PHY signal fluctuations, emulating a real scenario. The standard deviation is represented by vertical lines in graphs, calculated with a confidence interval of 95%.

All the VoIP traffic was transmitted over an rtPS or a BE channel just like the testbed scenario. The reasons of the choice of rtPS and BE are explained in Section 5.1.3.

The MOS values were obtained as in the real testbed, through the calculations in (Cole and Rosenbubluth, 2001), already including the packet loss and delay values. The MOS values considered include both uplink and downlink. The background traffic generated is also represented, in order to provide a better understanding of the WiMAX QoS assurance to priority traffic classes.

6.1.2 Results and Analysis

In this subsection, the WiMAX simulation results are demonstrated and explained.

LOS Scenario

Figure 6.3 depicts the Mean Opinion Score in LOS conditions. The left graph represents the scenario with background traffic and the right graph represents the scenario without background traffic.

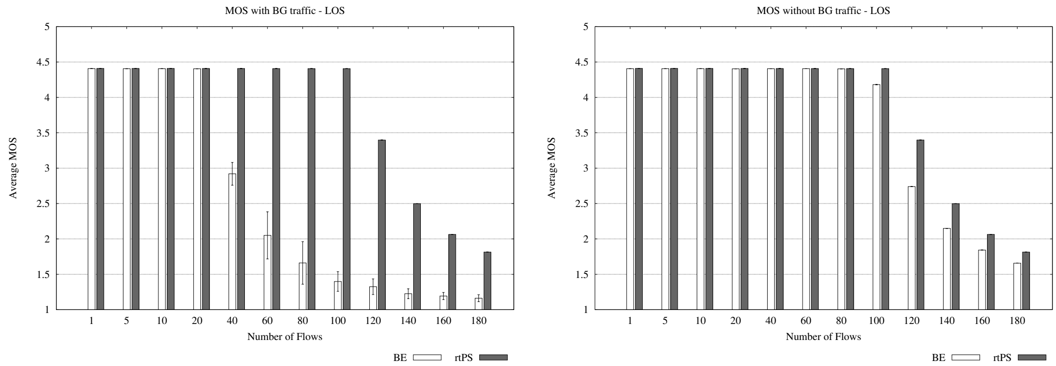


Figure 6.3: WiMAX Simulation - MOS - LOS

Several behaviors can be observed in the graphs included in Figure 6.3. The first is the impact of the constant increase of simultaneous users. As the number of users increases, the MOS tends to decrease. This behavior is justified by the decrease of available bandwidth as the number of users increase, leading to high values of packet loss, and, consequently, having a high impact in user perceived QoE.

In this context, one can observe that the MOS starts decreasing earlier in the BE service class than in Real Time Polling Service (rtPS), which is caused by the lower traffic prioritization imposed by the BE class. This fact can be seen in both graphs, either with and without background traffic, although with background traffic is more obvious.

In the background traffic scenario, the MOS starts decreasing at 40 users for the BE service class, while for the rtPS class it only starts decreasing at 120 simultaneous flows. This result clearly shows the advantage of the WiMAX QoS mechanisms, also observable in the testbed results (Section 5.1.3).

The next analysis considers that values between 4 and 5 of MOS represent an *excelent* voice quality, the values between 3 and 4 of MOS represent a *good* voice quality and the values between 2 and 3 represent a *fair* voice quality.

With the above values as reference, the BE service class allows a maximum of 100 and 20 simultaneous flows with *excelent* voice quality, without and with background traffic, respectively. Assuring a *good* voice quality, the BE service class allows a maximum of about 120 and 40 simultaneous flows, without and with background traffic, respectively. In case of *fair* voice quality, the BE allows a maximum of 140 and 60 simultaneous flows with *fair* voice quality, without and with background traffic, respectively.

As referred, the advantages of rtPS mechanism are clear. The rtPS service class guarantees, for both scenarios - with and without background traffic, an *excelent* voice quality for a maximum of 100 simultaneous flows, a *good* voice quality for a maximum of 120 simultaneous flows and a *fair* voice quality for a maximum of 160 simultaneous flows. These results demonstrate that the background traffic has no impact on the end-user perceived voice quality, when sending the voice data over a rtPS prioritized channel. The testbed results also demonstrated this advantage, although it was not so noted. Many variables can interfere in a real world testbed scenario, such as signal interferences and multi-path. These factors can lead to several differences between testbed and simulation scenarios, even though both tests characteristics are the same.

From the graphs, it can be seen that the standard deviation values (vertical lines within the columns) are very low or nonexistent, due to the optimal LOS conditions. These conditions lead to very stabilized channel conditions, without affecting the results.

The maximum simultaneous flows supported described above are summarized in Table 6.2.

LOS Scenarios - Simulation		
Maximum Simultaneous Flows - Fair Voice Quality		
	BE	RTPS
Without Background Traffic	140	160
With Background Traffic	60	160
Maximum Simultaneous Flows - Good Voice Quality		
	BE	RTPS
Without Background Traffic	120	120
With Background Traffic	40	120
Maximum Simultaneous Flows - Excelent Voice Quality		
	BE	RTPS
Without Background Traffic	100	100
With Background Traffic	20	100

Table 6.2: WiMAX Simulation - Summary of maximum allowed simultaneous flows

The MOS values described in the background traffic scenarios can also be complemented with the graphs in Figure 6.4. These graphs illustrate the background traffic sent and received by the background traffic node during the voice calls (affecting the perceived end-user QoE).

From the graphs in Figure 6.4 the decrease of the background traffic with the increase of simultaneous users is noticeable. This behavior is due the higher usage of bandwidth by both voice and background traffic, and, in the case of rtPS, the prioritization of voice traffic leads to the higher decrease of background traffic.

Until about 20 simultaneous voice users, both BE and rtPS can guarantee above 10 Mb/s of background traffic. However, as soon as the bandwidth begins to run low, the BS scheduler prioritizes the traffic for the rtPS flows, decreasing the background traffic.

In the BE, the decrease of background traffic only happens due to the lower bandwidth available and the resources competition with voice flows, maintaining the background traffic

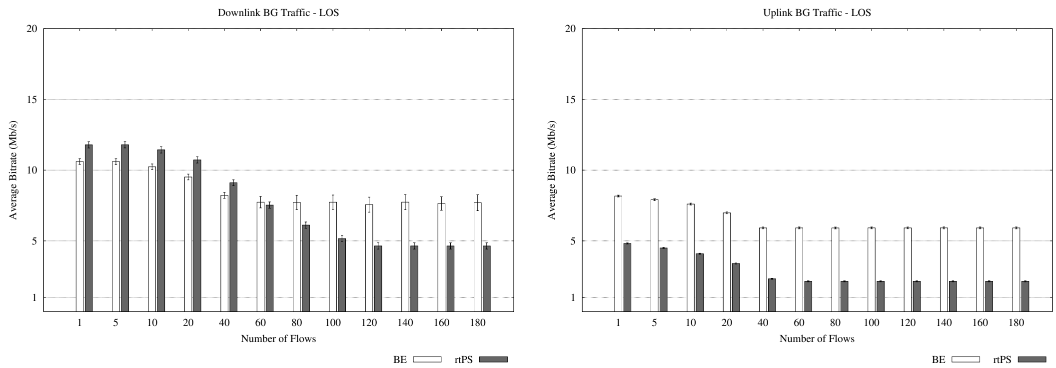


Figure 6.4: WiMAX Simulation - DL/UL background traffic - LOS

along several voice flows (resulting in a high impact in the end user QoE of VoIP flows). This behavior is both observable in downlink and uplink.

NLOS Scenario

Figure 6.5 depicts the Mean Opinion Score in NLOS conditions. The left graph represents the scenario with background traffic and the right graph represents the scenario without background traffic.

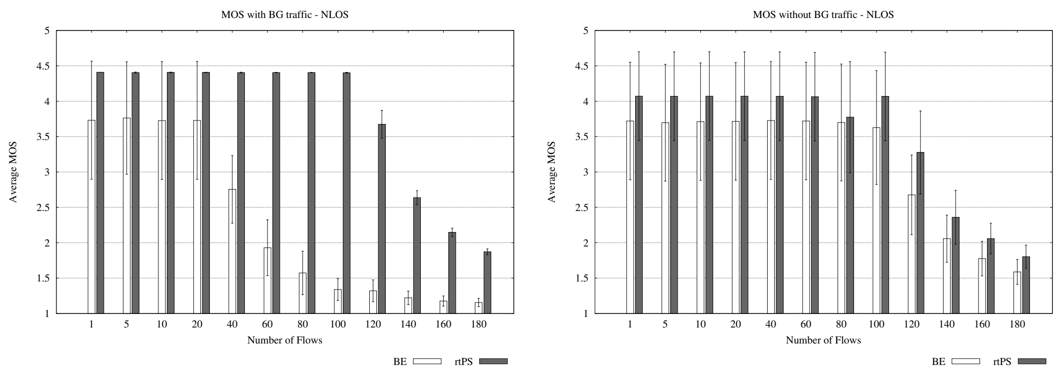


Figure 6.5: WiMAX Simulation - MOS - NLOS

The simulator NLOS conditions lead to a poor signal reception, and sometimes with losses of signal. This kind of interferences imposes high variations, not only in signal quality, but also in the quality of the data transmission.

These variations are shown in the graphs, through the standard deviation lines. However, these signal fluctuations do not have the same impact as in a real testbed. As previously referred, the real testbed is subjected to numerous interferences that are very difficult to consider and replicate in the simulation, such as multi-path fading, pathloss, signal fluctuations and signal reflections.

By analyzing the MOS values obtained, it is possible to observe that the maximum supported users are about the same as in LOS conditions (Table 6.2), however, with higher signal quality fluctuations, resulting in higher variations.

Figure 6.6 also serves as a complement to the previously shown graphs. Figure 6.6 illustrates the background traffic sent and received by the background traffic node during the voice calls.

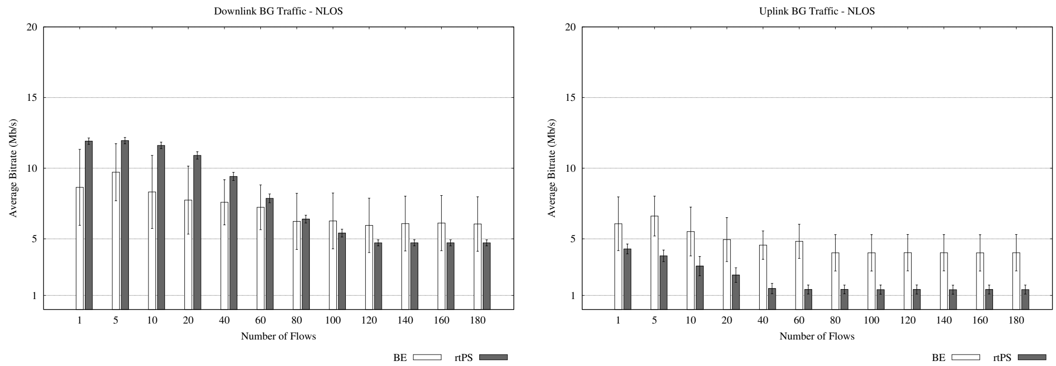


Figure 6.6: WiMAX Simulation - DL/UL background traffic - NLOS

As in the LOS scenario, the same behavior is observed - the decreasing of background traffic with the increase of simultaneous users. Due to the signal fluctuations, the received and sent background traffic is also affected and it is reflected in the variations (standard deviation).

6.1.3 Conclusion

This work allowed a better understanding of the WiMAX technology behavior in a real testbed and in simulation, observing the differences between scenarios and also the effectiveness of the QoS mechanisms to allow high levels of QoE.

From the attained results, it was proven that the WiMAX QoS mechanisms are very efficient in maintaining high values of end-user QoE, even with high background traffic. This conclusion was also obtained through the testbed results, however, not so noted as in the simulation.

It was observed that, even with the specification of all the parameters of the simulation, the simulation behavior is not exactly as in the real testbed. Although the efficiency of QoS mechanisms and the impact of background traffic is seen in both testbed and simulation results, the maximum users supported is very different. In the real testbed, it was possible to support about 50 users with rtPS with background traffic with *good* voice quality, while in the simulation scenario, this number increased to 120. Although the supported uplink and background traffic is about the same in both the simulation and the testbed, the numerous variables affecting the real testbed have a large impact on the maximum supported users.

The behavior of the WiMAX network in NLOS condition was proven to be worse than in LOS, however, the network behavior was not as in the real testbed. Despite the variations among the results and the specification of realistic settings, it is not possible to replicate all the real testbed parameters and characteristics, such as the interferences, pathloss and multi-path, resulting in different network behaviors. Although these parameters are considered, the models do not always correspond to the observed behaviors in real scenarios.

Even though some results were not as expected, this work demonstrated that there are some differences between a real testbed and a simulation scenario, but, despite those differences, it validated the WiMAX QoS mechanisms efficiency in the assurance of good VoIP QoE values with and without background traffic.

From the previous works, only Tran et al. (Tran et al., 2008) conducted an analysis of WiMAX technology in simulation and testbed, however, the end-user VoIP quality in a multi-user scenario involving background traffic is not analyzed, as performed in this study.

6.2 WiMAX and LTE

This section explains and demonstrates the results obtained in WiMAX and LTE simulation scenarios, describing the main differences between the technologies and the specified characteristics.

6.2.1 Scenarios Setup

In order to conduct a simulation study of the two competing 4G technologies, within the previously used WiMAX Work guidelines, several simulations were performed to analyze the different network capabilities in the support of multi-user VoIP, with and without traffic background.

In this setup, unlike the real testbed scenario, each user represents a network node, which implies a higher overhead at the network, not only each node overhead, but also a higher number of scheduling requests for traffic prioritization.

In the context of previous work, the tests herein performed are focused on the evaluation of multi-user VoIP calls over WiMAX and LTE, with and without background traffic. However, with the main objective of evaluating the technologies with a similar setup to a real world scenario, two different codecs were used, with higher compressing rates than the G.711.1 codec (which was used in previous testbed tests). Thus, it is possible to evaluate the network behavior with codecs that are most suited for wireless communications, with lower bandwidth utilizations and better packet loss robustness.

Figure 6.7 demonstrate, respectively, the WiMAX and LTE simulation scenarios.

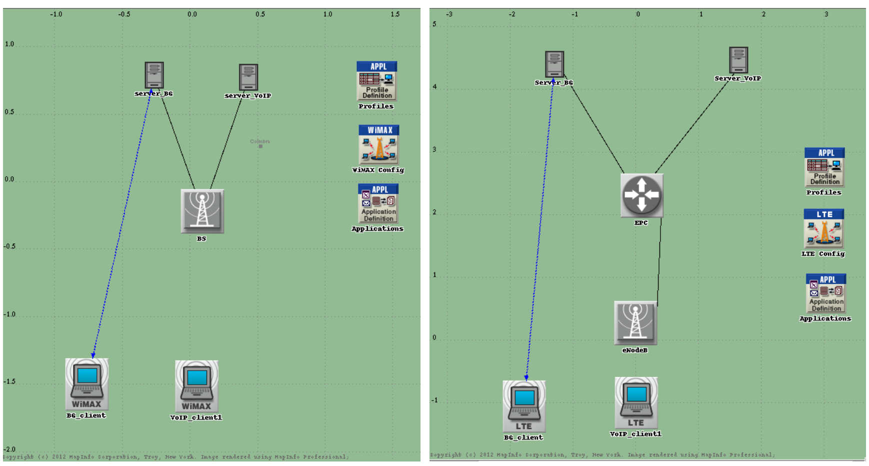


Figure 6.7: WiMAX (left) and LTE (right) Simulation Scenarios

The codecs chosen for the tests were G.726 (ITU-T, 1990) and G.729A (ITU-T, 2007).

The choice of the G.726 codec was by the fact that it provides high levels of QoE (MOS 4.3), with lower bandwidth usage, being a strong competitor to the G.711.1 codec. The G.726 is an Adaptive Differential Pulse Code Modulation (ADPCM) codec and it allows different bitrates, namely 16, 24, 32, 40Kb/s, with a frame size of 10ms. The delay introduced by the codec is 0.125ms, and it has no look-ahead delay. The chosen bitrate for the G.726 codec was 16Kb/s.

The choice of the G.729A codec was due to the fact that it supports high levels of QoE (MOS 4.0), using about a half of the bandwidth of the 16kb/s G.726 (ADPCM). This codec operates with a bandwidth of 8kb/s, with a frame size of 10ms and 5ms of look-ahead time. The A version of this codec represents an improvement on the complexity of the codec, allowing it to have a high ratio value of quality/complexity compared to other codecs. This is a Conjugate-Structure Algebraic Code Excited Linear Prediction (CS-ACELP) codec type.

From the analyzed characteristics, Table 6.3 was created to summarize the mentioned codecs.

	G.726	G.729A
Coding Type	ADPCM	CS-ACELP
Bitrate (used)	16 Kb/s	8Kb/s
Frame Size	10 ms	
Look ahead time	0	5ms
Complexity (MIPS)	1.25	10
Maximum MOS	4.3	4.0

Table 6.3: WiMAX and LTE Simulation Codecs Characteristics

Since this simulation is not intended to provide an assessment of the scenarios line of sight and interferences, but instead, it is intended to provide an evaluation of multi-user support and QoS mechanisms efficiency, the simulation model chosen for WiMAX was *Framing Module Enabled* and for LTE was the *Efficiency Enabled*. Thus, the arrangement of the nodes on the map will have no relevance in the final results.

In order to maintain equal characteristics between both technologies, the center frequency of 2.5GHz was used with TDD duplex mode, with a channel bandwidth of 10MHz. As in previous tests, the frame configuration (for uplink / downlink) was set up to 60% to the downlink data and 40% to the uplink data. This setup represents the Time Division Duplexing (TDD) channel configuration index 1 (UL/DL 2:3) at LTE and 18 OFDMA symbols for uplink and 29 for the downlink at WiMAX.

In this study, the chosen types of service flows and EPS Bearers were based on the most suitable types for VoIP applications. For WiMAX, the Extended Real Time Polling Service (ertPS) was used, since it allows the definition of minimum and maximum allowed rate, as well ass maximum allowed delay and jitter. For LTE, it was used a Guaranteed Bit Rate (GBR) Bearer with a QoS Class Identifier (QCI) of 1. This type of QCI defines the guaranteed bit rate, the packet delay budget (maximum allowed delay) and the packet error loss rate (maximum packet loss allowed).

The best effort background traffic was sent over a Best Effort (BE) channel in WiMAX and over a Non-GBR channel with QCI of 9 in LTE. This background traffic setup represents the lowest priority configuration of both technologies.

It should be noted that the minimum guaranteed rate was set to 32Kb/s (which is the minimum allowed value by the both technologies in the simulator), and the maximum guaranteed rate was set to 64 Kb/s (which is the minimum value after 32Kb/s).

Each test was repeated through 10 runs, to avoid the outlier results caused by network changes and BS scheduling mechanisms variations, where each test used a different seed. The different seeds were used to generate different network behaviors in terms of QoS scheduling and traffic prioritization, emulating a real scenario. The standard deviation is represented by vertical lines in graphs, calculated with a confidence interval of 95%.

The detailed configuration of the simulation environment is described in Table 6.4.

<table>
<tr><th></th><th>Parameter</th><th>WiMAX</th><th>LTE</th></tr>
<tr><td rowspan="9">Simulator Parameters</td><td>Frequency Band</td><td colspan="2">2.615 GHz</td></tr>
<tr><td>System Bandwidth</td><td colspan="2">10MHz</td></tr>
<tr><td>FFT Size (Subcarriers)</td><td colspan="2">1024</td></tr>
<tr><td>Duplex Mode</td><td colspan="2">TDD</td></tr>
<tr><td>MIMO technique</td><td colspan="2">SISO (1x1)</td></tr>
<tr><td>Modulation Format</td><td>64 QAM</td><td>Index 28 (64 QAM)</td></tr>
<tr><td>Frame Size</td><td>5 ms</td><td>10 ms</td></tr>
<tr><td>DL / UL sub-frame</td><td>29 symbols (60%) / 18 symbols (40%)</td><td>TDD channel configuration index 1 (UL/DL 2:3 – 40%/60%)</td></tr>
<tr><td>QoS definitions</td><td>VoIP Traffic: QCI 1 GBR
BG Traffic: QCI 9 Non-GBR</td><td>VoIP Traffic: ertPS
BG Traffic: Best Effort</td></tr>
<tr><td rowspan="5">Application Parameters</td><td>Simulation time (per run)</td><td colspan="2">300 seconds</td></tr>
<tr><td>Call time (per run)</td><td colspan="2">60 seconds</td></tr>
<tr><td>Call start time</td><td colspan="2">Uniform distribution (60 to 90 seconds)</td></tr>
<tr><td>Run repetition</td><td colspan="2">10 repetitions</td></tr>
<tr><td>VoIP CODEC</td><td colspan="2">G.729A with 8Kb/s compress rate and G.726 with 16Kb/s compress rate</td></tr>
</table>

Table 6.4: WiMAX and LTE Simulation Parameters

6.2.2 Results and Analysis

In this subsection, the WiMAX and LTE simulation results are demonstrated and explained, consisting of two subsections, namely G.726 results and G.729A results.

G.726 Results

Figure 6.8 depicts the Mean Opinion Score obtained in WiMAX and LTE simulations, for the scenarios with and without background traffic. The left graph represents the scenario without background traffic and the right graph represents the scenario with background traffic.

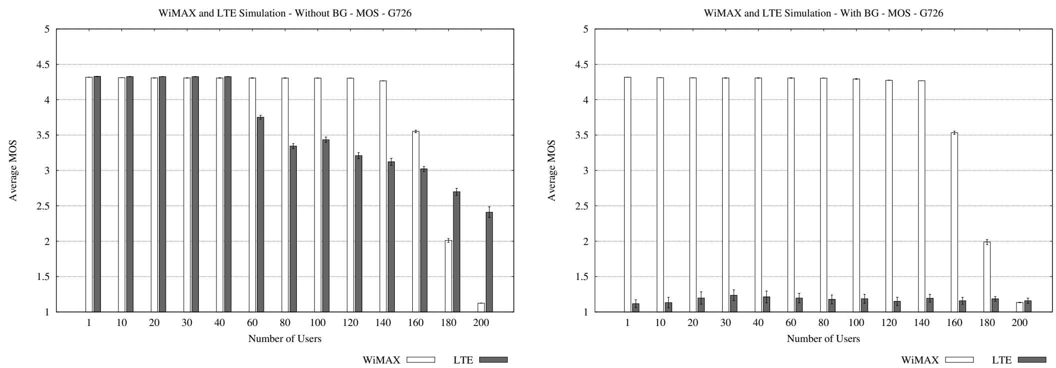


Figure 6.8: WiMAX and LTE Simulation - G.726 - MOS

The graphs shown in Figure 6.8 show several network behaviors, not just the efficiency of the QoS mechanisms of both technologies, but also their behavior with several simultaneous voice users.

This analysis considers that values between 4 and 5 of MOS represent an *excelent* voice quality, the values between 3 and 4 of MOS represent a *good* voice quality and the values between 2 and 3 represent a *fair* voice quality.

By conducting an analysis of the user support in both technologies, it is possible to observe that without the background traffic, LTE supports a higher number of simultaneous users, although the reduction of speech quality (QoE) is earlier than with WiMAX.

Ensuring an *excellent* voice quality up to 40 users, a *good* voice quality to 160 and a *fair* voice quality up to 200 users, LTE proves to be capable of supporting a large number of simultaneous voice users with high levels of Quality of Experience (QoE). The decrease in the QoE is gradual with the increase of simultaneous users, since the bandwidth is being more used, and thereby, causing more delays and packet losses (such network parameter values are implicit in the calculation of the MOS, provided by OPNET).

In case of WiMAX, it is observable through the graphs that this technology ensures the highest quality allowed by this codec up to 140 users, meaning that it provides an *excellent* voice quality until this number of users. This quality is assured due to the WiMAX scheduling mechanisms, which gives priority to the voice streams until it reaches the maximum link capacity. Once this capacity is reached, with 140 users, there is a noted decrease in the voice quality. Yet, the technology allows a *good* voice quality up to 160 users, while the *fair* quality is ensured up to 180 simultaneous users.

Thus, in the scenario without background traffic, there is a slight advantage of the LTE over the WiMAX, ensuring about 20 simultaneous users more with a *fair* voice quality than the WiMAX.

Regarding the background traffic scenarios, the results are rather different, in particular in the case of LTE, as described below.

In WiMAX, the guarantee of high levels of quality of experience is ensured even with background traffic, with identical results to the scenario without background traffic. This behavior is due to the efficiency of the WiMAX scheduling mechanisms and quality assurance, which discards the lowest priority traffic in favor of higher priority traffic - in this case, VoIP traffic. Figure 6.9 illustrates the referred behavior, which represents the background traffic sent on the uplink by WiMAX. It can be seen that as the simultaneous voice users are increasing, the background traffic decreases, in order to provide the sufficient bandwidth for the voice streams until it reaches the limit of the bandwidth and it stops sending background traffic (when the WiMAX capacity is reached).

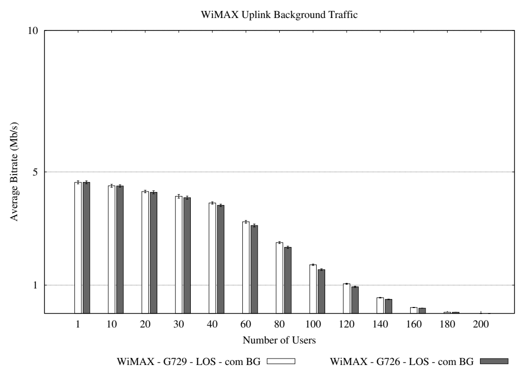


Figure 6.9: WiMAX and LTE Simulation - G.726 and G.729A - Uplink background traffic

In the case of LTE, the results show that the QoS mechanisms do not have the desired effectiveness. Even with the background traffic being sent in a flow that does not guarantee any bandwidth and with the lowest level of prioritization, the background traffic has a great impact on the end-user perceived voice quality, preventing this technology to guarantee good quality for the voice users.

Since these results were somewhat unexpected, several LTE and WiMAX tests were performed with different background traffic loads, in order to analyze its impact on the end-user perceived voice quality. Therefore, several tests were performed with the G.726 codec and with 10 voice users, employing different loads of background traffic: no background traffic, 2.5Mb/s of background traffic (sending and receiving), 5Mb/s, 7.5 Mb/s and 10Mb/s (used value in previously tests when considering *background traffic*). What was observed through these tests, whose results are shown in Figure 6.10, is that with the gradual increase of background traffic, the voice quality in LTE is highly affected. In LTE, as the background traffic increases, the MOS value decreases, until it reaches very low levels (with 7.5Mb/s and 10Mb/s of background traffic), as previously observed. In WiMAX, the background traffic has no impact on the MOS values.

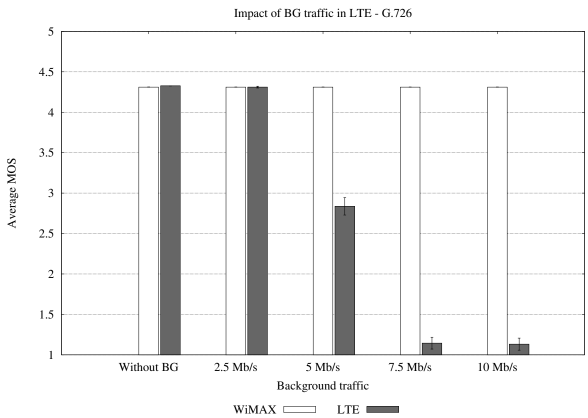


Figure 6.10: WiMAX and LTE Simulation - Impact of background traffic (G.726)

In summary, through the results it is observed that the support for multiple users of the voice codec G.726, without background traffic, is better supported by LTE, demonstrating some advantages in terms of maximum number of simultaneous users supported, ensuring at least *fair* voice quality up to 200 users. The WiMAX technology shows, however, some advantages in the supported users with *excellent* voice quality quality, ensuring an *excellent* voice quality up to 140 users. Overall, the LTE can support about 20 simultaneous users more than WiMAX, even though with a lower voice quality.

In the background traffic scenario, the efficiency of the WiMAX scheduling mechanisms are evident, rather of LTE, which shows very low levels of quality of experience (QoE) and an inefficient scheduling mechanism.

G.729A Results

Figure 6.11 illustrates the results obtained with the codec G.729A, which, although having a higher complexity compared to G.726, uses about half of the bandwidth of the codec G.726 (16Kb/s of G.726 compared with the 8kb/s of G.729A).

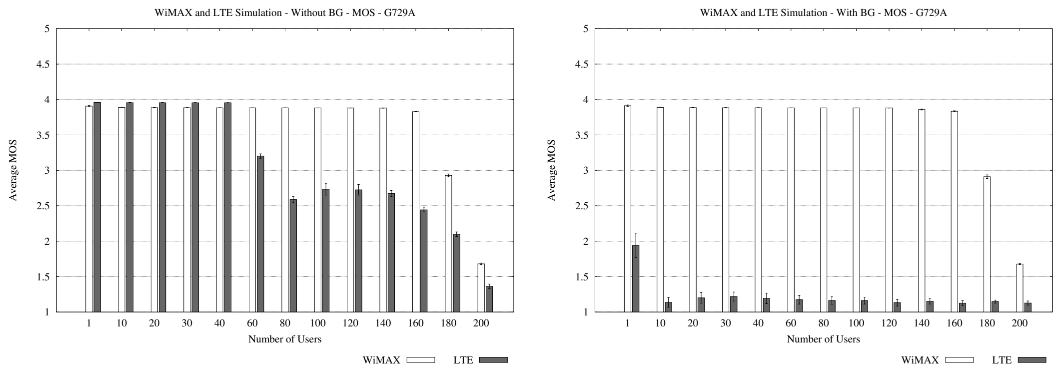


Figure 6.11: WiMAX and LTE Simulation - G.729A - MOS

By using a codec with a lower bandwidth, the expected result is that more users may be supported, although with a lower voice quality (up to 4 MOS). This expected result is seen in the graphs, namely without background traffic, where the multi-users support is increased, however, with a lower MOS than previously observed with the G.726 codec. This lower MOS values are explained by the maximum supported MOS value by the codec G.729A, which is 4, making it impossible to achieve *excelent* voice quality.

In case of LTE, this technology provides a *good* voice quality up to 60 simultaneous users, and a *fair* quality up to 180 simultaneous users. This way, this codec allows about 20 more users than the previously used codec, G.726, although the average voice quality is lower.

Regarding the WiMAX technology, with the G.729A codec, it supports up to 180 users with a *good* voice quality. Beyond this simultaneous users, there is an abrupt drop in the perceived voice quality, decreasing to MOS values below 2.

The maximum number of supported voice users without background is, for both technologies, 180 simultaneous users. Nevertheless, the WIMAX technology supports these users with a *good* voice quality, while the LTE supports from 60 to 180 simultaneous users with *fair* voice quality.

The reasons for the decrease of voice quality with the increase of voice users are the same as mentioned above, namely the reduction of the available bandwidth and the higher scheduling overhead. In the case of WiMAX, it ensures the best voice quality up to the maximum limit of its capacity, while in LTE, due to it lower QoS scheduling mechanisms efficiency, the packet losses and delays become higher with the increasing use of bandwidth, causing a significant impact on the end-user perceived voice quality - represented by MOS.

For the scenario with background traffic, WiMAX provides the same number of simultaneous users as without background traffic, discarding the background traffic, while the LTE does not manage this traffic correctly, causing a great impact on voice quality. This quality is thus degraded to very low levels (MOS values below 2).

As performed in the previous tests, several LTE and WiMAX tests were conducted with different background traffic loads, employing the G.729A codec, in order to analyze its impact on the end-user perceived voice quality. The results are depicted in Figure 6.12.

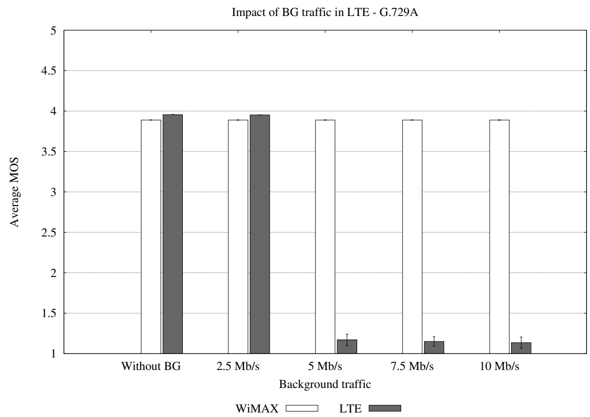


Figure 6.12: WiMAX and LTE Simulation - Impact of background traffic (G.729A)

Through these tests, whose results are shown in Figure 6.12, the same previous behavior was observed. With the gradual increase of background traffic, the voice quality in LTE is highly affected, and, as the background traffic increases, the MOS value decreases, until it reaches very low levels. In WiMAX, the background traffic has no impact on the MOS values.

6.2.3 Conclusion

Through the tests, there are several distinct behaviors, as explained below.

Without background traffic, both the LTE and WiMAX have similar behaviors and similar multi-users support. However, it is possible to observe that the traffic prioritization made by WiMAX has some advantages, always ensuring the best possible quality of experience within the available bandwidth. In the case of LTE, for both codecs, it is noted that the MOS gradually decreases with the increasing of simultaneous users. Still, the LTE can support more users (with *fair* quality) than WiMAX, although with lower MOS values.

In scenarios with background traffic, the efficiency of traffic prioritization is more evident. In this scenario, WiMAX allows the same number of users as without background traffic, while in case of LTE, the performance and the MOS values are so low that makes it impossible to transmit the voice streams with an acceptable voice quality.

Comparing both codecs, and despite the fact that the codec with less bandwidth (G.729A) allows a higher user support (i.e., about more 20 users in both technologies), the levels of MOS are lower when compared to those observed for the G.726 codec.

Through the analysis of the previous works, it was observable that none had an analysis of end-user perceived voice quality while comparing Mobile WiMAX and LTE. This study aimed to fill this gap, by complementing those works with new conditions and parameters. The most relevant related work for this study were the Andersson et al. (Andersson et al., 2011) and Iwayemi et al. (Iwayemi and Zhou, 2009) which performed an analysis of different codecs, and Olariu et al. (Olariu et al., 2011) and Wang et al. (Wang et al., 2009b) which investigated the usage of different technologies. This current study takes these works into consideration, and complements them with an analysis of a multi-user VoIP scenario employing the QoS assurance mechanisms from both technologies (i.e., WiMAX and LTE), by measuring the end-user VoIP Quality of Experience (QoE) and assessing the maximum supported VoIP users with the usage of different codecs

7. Project Plan

This chapter describes the project and work plan for the first and second semesters of the MSc thesis presented on this document.

The first section demonstrates the work performed in the first semester, while the second section demonstrates the planned and effective schedule for the second semester. In the third section, the project risk analysis is conducted.

7.1 First Semester

The effective project plan for the first semester is represented in Figure 7.1.

The tasks performed are explained below:

- Alvarion Training

One of the first tasks of the project was the Alvarion training. This training was focused on Alvarion WiMAX Equipment and software, with the objective of learning how the equipment was configured and managed. The WiMAX Pilot for this project consists of two Base Transceiver Station (BTS) - *Alvarion BreezeMAX Macro Outdoor* - and one Access Service Network (ASN) gateway - *Alvarion BreezeMAX ASNGW - Mini Centralized*. The CPEs are also *Alvarion* equipment, such as USB (mobile), Indoor and Outdoor fixed WiMAX access points.

- Reading WiMAX/Alvarion Documentation

After the Alvarion training, and to consolidate the knowledge, the reading of WiMAX related work and Alvarion documentation was needed. This task represents the reading of these materials.

- WiMAX - State of the Art

This task included all the work related to the analysis of WiMAX state of the art, including the reading of several WiMAX documentation and writing the corresponding state of the art.

- Testbed Deployment

This task represents the testbed deployment. It involved the BTS deployment at Polo 1 and Polo 2 and also the deployment of the different CPEs. Although the deployment of WiMAX network equipment, it was also necessary to deploy EDP equipment. This deployment included the deployment of the WiMAX and EDP equipments in all the testing scenarios, namely PT- Quinta da Várzea, PT- Estádio Universitário, EDP-Rua do Brasil, Eolic Park at Vila Nova and also the configuration of mobile nodes for EDP WFM team.

Besides the deployment of the testbed, it was necessary to constantly monitor the EDP and WiMAX equipments to analyze it behavior and, if necessary, re-deploy and re-configure some of the field nodes.

- Writing Technical Documentation

This task represents the written of documentation for EDP, including tests description and analysis.

- RS232 Tasks

RS232 tasks represent the MOXA N5100A converters and cable specification configuration, testing and documentation. This task involved a coordinated and co-work with the EDP *Telecomando* team, in order to correctly specify the cables accordingly to the EDP specifications.

- Pre-set of Tests (preliminary tests)

The work performed in this task consisted on specification and execution of the preliminary tests in order to estimate the WiMAX capabilities, to allow the correct specification of the reference tests.

- T1 - Reference Tests

The first set of tests, represented by this task, had the objective of demonstrating the WiMAX capabilities of supporting multi-user applications in different LOS and NLOS conditions. The traffic emulated was generic, only to serve as reference for the next tests.

- T2 - Application Tests

This task includes the execution, analysis and description of Voice over IP (VoIP) evaluation over WiMAX, in the same conditions as the previous reference tests.

- T3 - EDP-Application Tests

The main goal of this task was to evaluate the WiMAX capabilities in the EDP-WiMAX context, emulating traffic conditions similar to the real EDP-Applications.

Due to some delays in the implementation of real applications, caused by the unavailability of EDP teams and testbed hardware/software problems, only part of this task was executed in the first semester. Since the deployment of EDP applications was not finished in the first semester, and it was necessary to obtain traffic patterns of real applications to enable the emulation of these tests, these test set was delayed to the second semester.

- Writing Report

This task consisted on writing of the MSc intermediary report.

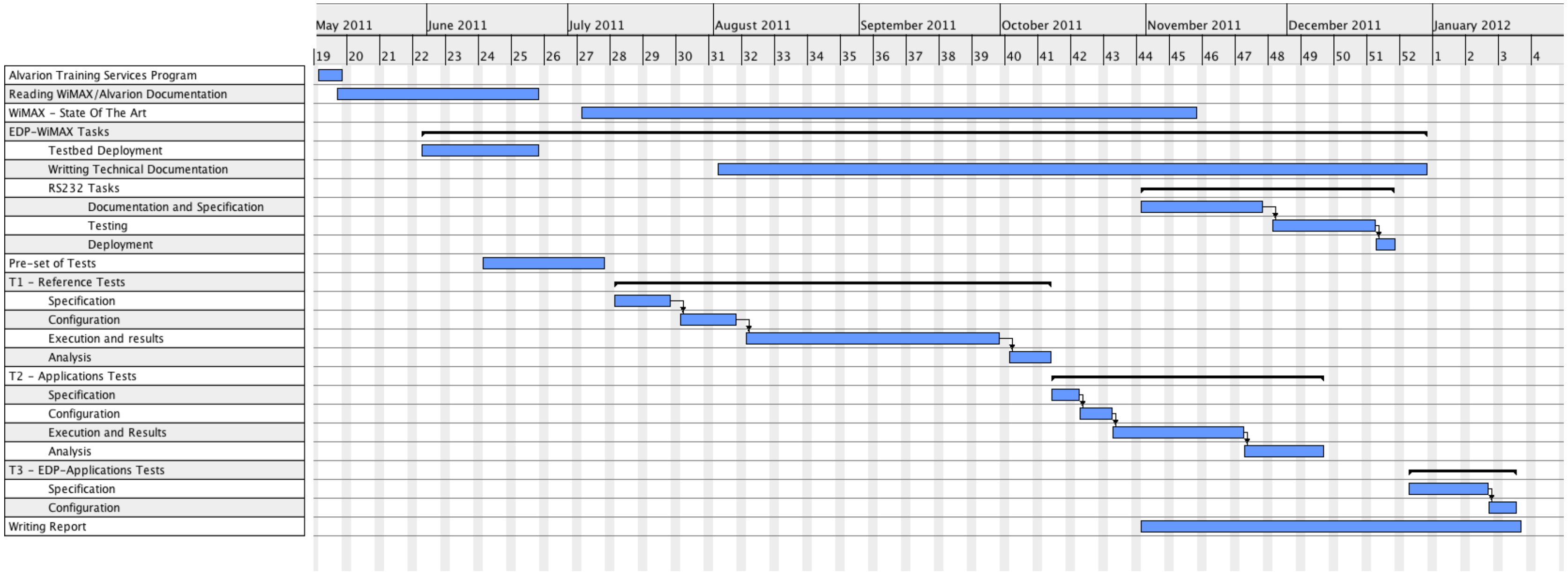


Figure 7.1: Project Plan First Semester - Effective schedule gantt chart

7.2 Second Semester

The initial project plan for the second semester is demonstrated in Figure 7.2, while the effective project plan is demonstrated in Figure 7.3. The tasks depicted in the gantt charts are described below.

Although the scheduled final date for the project was in February 2012, the project was only completed in May 2012. This delay was due to several factors, such as: Difficulty in synchronizing all EDP teams to deploy the EDP applications; EDP delays in the delivery of EDP applications reports, to be included in the final report; Deployment issues that delayed the full operability of the applications (as the case of the specification of RS232/IP cables and converters); Problems that arose in the testbed, such as nodes failures (abnormal shutdowns, equipment crashes, etc.).

All these problems caused a three month delay in the project, which affected the execution of simulation evaluation and the predicted dates for the submission of one paper, which would include the simulation tests. However, the second semester objectives were achieved.

- VoIP over WiMAX: QoE Evaluation

This task is related to the writing of the paper "VoIP performance over Mobile WiMAX: An Urban Deployment Analysis", as well as the realization of additional tests to include in this paper. This paper was submitted to the 2nd IEEE Baltic Conference on Future Internet Communications, 25-27 April 2012, in Vilnius, Lithuania. This paper was accepted and presented, and it is included in Appendix A - VoIP performance over Mobile WiMAX: An Urban Deployment Analysis.

- EDP - WiMAX

This task is divided in several sub-tasks. As described above, the project was delayed and some tasks were only performed in the second semester.

The deployment and monitoring sub-task consisted of the link monitoring and the deployment of the remaining applications, which also included some site visits to verify and re-configure the equipments.

The applications tests sub-task is related to the analysis of all the EDP applications traffic patterns, through a network packet analyzer. Also, it involved the coordination and execution of EDP applications simultaneous tests, with the main objective of assessing the WiMAX network behavior with all EDP applications working together and at the same time.

The application specific tests consisted in the characterization of the WFM application, in terms of traffic patterns and procedures. This sub-task involved one site visit to the EDP-Rua da Arregaa, to specify the application in collaboration with EDP WFM team.

After the execution of all the application tests, the EDP pre-report was completed (which was started in the first semester). This pre-report was then restructured to the final report, in a collaboration work with Professor Paulo Simões. The final report was also changed during several interactions with the EDP project coordinators. The final delivered report is included in Appendix B - EDP - WiMAX Pilot - Final Report, and it was presented in 12 July 2012 at EDP- Distribuição Coimbra headquarters.

- State of the Art - LTE

This task includes all the work related to the analysis of Long Term Evolution (LTE) state of the art, including the reading of several LTE documentation and writing the corresponding state of the art.

- Simulator

In order to conduct an analysis of WiMAX and LTE in a simulation environment, it was necessary to acquire experience with a network simulator. Thus, this task includes learning and configuring a network simulator, namely OPNET (OPNET, 2012), in order to perform those tests. This task was performed during several weeks, due to the simulator complexity and the numerous parameters of the simulator, as well as the duration of each simulation.

- Simulator Study and Comparisons

After the learning phase of the OPNET simulator, the effective WiMAX and LTE simulations were conducted. These simulations were performed in two phases with different objectives.

The first phase consisted on the replication of the testbed scenarios into simulation. It involved the creation of several simulation scenarios with the same specifications as the real testbed, as well as the comparison and the analysis of the results.

The second phase of the simulation study consisted on the comparison of WiMAX and LTE technologies, simulating several VoIP users with and without background traffic. This way it was possible to evaluate the impact of several users sharing the same channel and also the impact of background traffic on the end-user Quality of Experience (QoE), for both technologies.

During the execution of these tests, and with the objective of complementing this study with the analysis of different codecs, the study was performed with two different voice codecs. All the simulation studies involved the specification and configuration of each simulation.

- 4G Technologies for VoIP: Quality of Experience Assessment

After the performed assessment of the WiMAX and LTE technologies by simulation, the results were included in one work which will be submitted to Telecommunication Systems -

Modeling, Analysis, Design and Management journal, with the title of “4G Technologies for VoIP: Quality of Experience Assessment”.

- Writing Report

This task consisted on writing of the MSc final report.

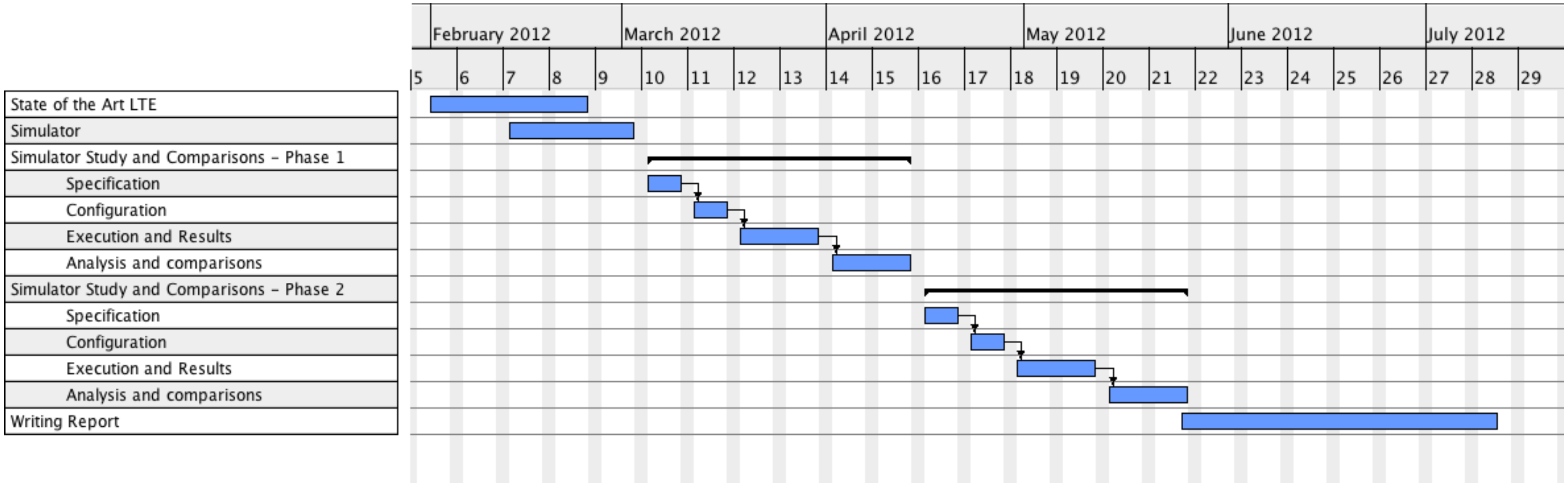


Figure 7.2: Project Plan Second Semester - Initially proposed schedule gantt chart

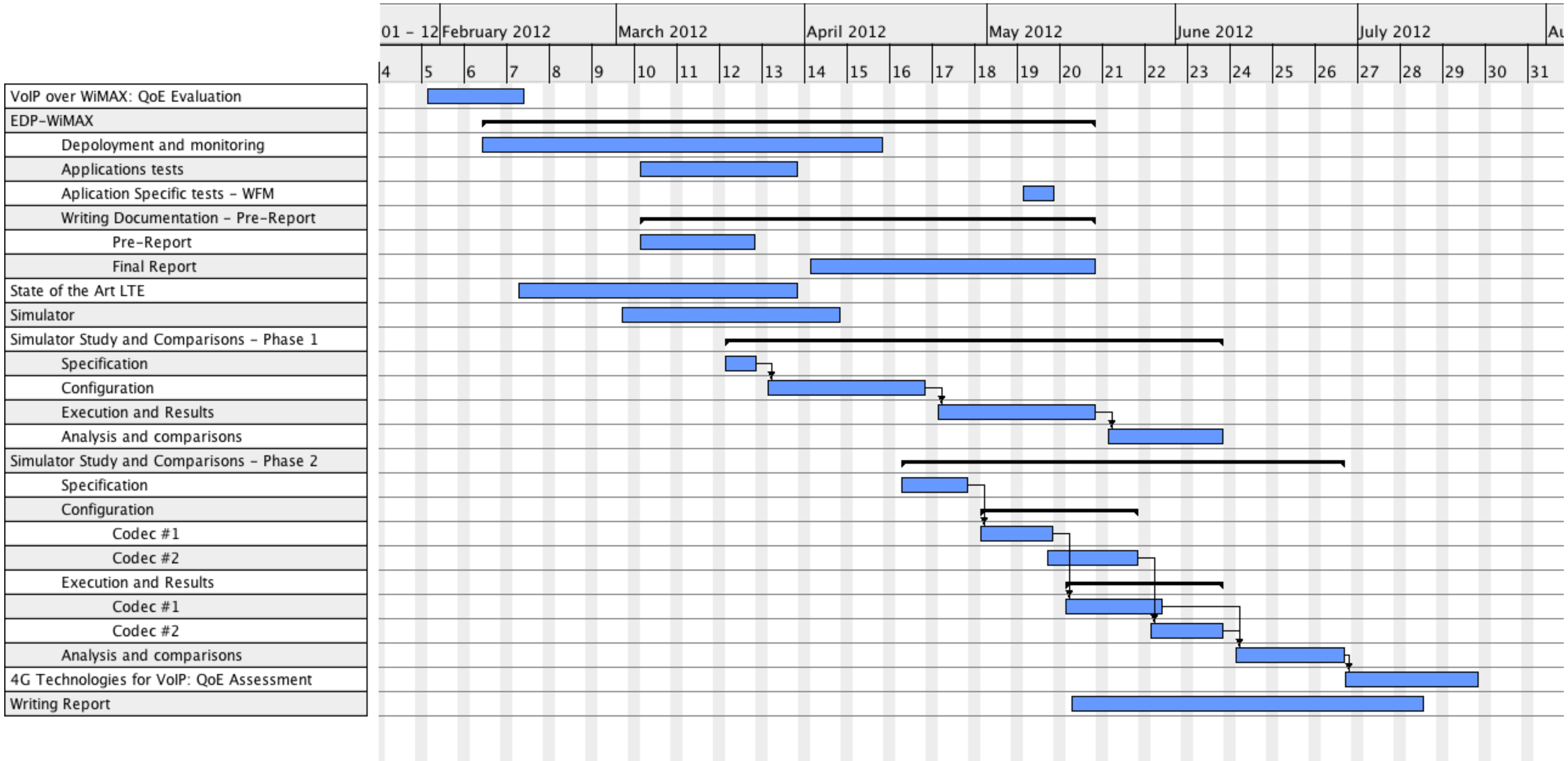


Figure 7.3: Project Plan Second Semester - Effective schedule gantt chart

7.3 Risk Analysis

Since the EDP-WiMAX project has involved different entities, personnel and different kinds of legacy hardware, many risks and problems have arisen. Some of those risks, that became problems, caused some delays in the project tasks. To avoid these risks to become problems, some contingency measures were planned and enforced.

Risk	Risk level	Contingency Measures
WiMAX Spectrum License Period	Moderate	- Schedule all tests to be done; - Evaluation through simulation;
EDP staff not available	Moderate	- Schedule meetings with more advance
Lack of communication with other entities	Low	- Schedule regular meetings; - Identify the stakeholders and the hierarchy of the organization;
Applications hardware not available	Moderate	- Change points of evaluation;
Applications IP integration problems	Moderate	- Evaluate with synthetic traffic;
Node failures and problems	High	- Constantly monitor the connections and links

Table 7.1: Risk Analysis

During this work some contingency measures were taken in order to prevent the risks to become problems. In particular, the simulation tests were planned in the second semester and some of the synthetic tests were re-scheduled, thus to avoid the risk of WiMAX spectrum license period expiration. Although the communication with the several entities was assured by regular meetings, the unavailability of EDP staff was a risk that sometimes became a problem and which the contingency measures were ineffective, causing some delays in the project plan. The risks related to the applications hardware and integrations were quite present, being attenuated by the execution of parallel synthetic tests and by changing the type of application's evaluation, and this way, avoiding the delays in the ongoing tasks.

The constantly monitoring of the links and the nodes did not resolve all risks, but allowed a fast response and the resolution of the risks that became problems within a short period of unavailability.

Therefore, the risk and the contingency measures planning in this work was helpful and allowed to avoid some problems and excessive delays.

8. Conclusion

This work was essentially focused on the analysis of two similar technologies, which are direct competitors in the Fourth Generation of mobile phone standards and technology (4G) networks, namely Worldwide Interoperability for Microwave Access (WiMAX) and LTE. Both technologies are considered Broadband Wireless Access (BWA) networks, allowing the network access with high coverage and throughputs. Some of the most important improvements against the legacy BWAs (such as Global System for Mobile Communications (GSM) and High Speed Packet Access (HSPA)) are the Quality of Service (QoS) assurance mechanisms and the fact that these are *All-IP* networks. These mechanisms come along with the higher QoS requirements of the existing and future multimedia applications, as well as with the growing usage of these networks for common data traffic (e.g., Internet usage).

This work starts by detailing and explaining the current state of the art of LTE and WiMAX technologies, describing their main characteristics. These characteristics are also explained with more detail, allowing a perception of the mechanisms behind the improvements and features of both technologies. The comparison between these technologies, carried out in this work, provides a comparative analysis of the mechanisms of each technology and the differences between them. After this analysis, and before the execution of the assessment tests, several works related to WiMAX and LTE assessment in the context of this work were studied, in order to observe and analyze the most used and correct methodologies to be followed in the real pilot and simulations assessments.

The interaction with a real WiMAX testbed, deployed in the context of the EDP-WiMAX project, enabled a deeper understanding of this technology, implicit in the WiMAX state of the art and also enabled the acquisition of knowledge of how to manage and maintain a real WiMAX deployment. The fact that this work required a constant interaction with the EDP teams, related to EDP-WiMAX project, enhanced the development of author's teamwork and interoperability skills between different teams. Also, the acquisition of *Certified Alvarion System Specialist* promoted a deeper and richest knowledge of the WIMAX technology.

Dealing with a real WiMAX testbed was an asset to this work, both in the context of technology assessment, and in the learning of the technology. It was possible to develop a fairly comprehensive study, with different types of traffic scenarios and conditions of access. The study was performed in an academic aspect, namely the evaluation of various traffic types and applications, but also in an enterprise aspect, with the evaluation and specification of several EDP applications, including their monitoring, adaptation and management. This analysis showed that WiMAX has good coverage and quality o signal assurance capabilities, allowing the access to the network up to about 20 km of distance from the Base Station (BS), with an average throughput of 10 Mb/s. Regarding to applications and multi-users support, the WiMAX technology is able to support several simultaneous users maintaing good QoS and QoE levels.

The work performed within the WiMAX testbed allowed the execution of an assessment paper, with the title of "VoIP performance over Mobile WiMAX: An Urban Deployment Analysis", submitted and accepted at the 2nd Baltic Conference on Future Internet Communications, which took place between 25 and 27 April 2012 in Vilnius, Lithuania. In the context of the EDP-WiMAX project, all work related to EDP applications and WiMAX pilot assessment was included in a final report, in order to provide to EDP a deeper knowledge of its applications and its suitability for wireless networks, such as WiMAX, as well as some conclusions and considerations about their applications and a comparison between WiMAX and LTE technologies.

Since the EDP-WiMAX project was planned to finish in February 2012, several simulation studies were planned for the second semester. However, although the project was extended until May 2012, because of the previously referred reasons, it was still possible to meet these objectives. Therefore, two simulation studies were conducted: evaluation of WiMAX via simulation, comparing it behavior with the testbed results; assessment of LTE and WiMAX via simulation in a multi-users VoIP scenario. The first evaluation showed that, comparing WiMAX via simulation and via testbed, the testbed results are usually worst than the simulation results, since there are several factors that can randomly affect the results, such as multi-pathing, fading and shadowing. Concerning the comparison study between WiMAX and LTE via simulation, it was observed that both technologies are able to support multi-users scenarios, allowing several dozens of simultaneous users, although that WiMAX QoS assurance mechanisms demonstrated clear advantages over the LTE QoS assurance mechanisms.

After the performed studies it was possible to observe that both WiMAX and LTE have good capabilities in support of multi-users scenarios, maintaining between them similar levels of performance in the tested scenarios (except for scenarios with background traffic). The assessment of WiMAX and LTE through simulation will be submitted to the Telecommunication Systems - Modeling, Analysis, Design and Management journal, with the title of "4G Technologies for VoIP: Quality of Experience Assessment".

Regarding to the real usage of these technologies in commercial deployments, WiMAX is falling behind when compared to LTE, since the latter allows a smoother transition from the previous versions of 3GPP, namely the HSPA and Evolved HSPA (HSPA+), which is preferred by the operators. Despite the good WiMAX capabilities, it could not compete with the LTE advantages for the current network operators, being left for the niche markets, where it is not necessary to provide interoperability with legacy 3GPP networks.

Thus, it is the opinion of the author that both technologies show great improvements and potential when compared to previously technologies, and both are suitable for different markets. LTE, as seen in the present days, is having a great growth and it is increasingly being used by the operators that seek to evolve their networks, while WiMAX is having more impact in the developing countries, where there is no need for interoperability between legacy networks. However, WiMAX is still a strong candidate for the usage in corporate and enterprise networks, where the companies intend to have wider network coverage for their employees and deployed equipment, allowing an installation and maintenance costs relatively low when compared to LTE.

This work allowed the author to understand and learn with great detail both of the technologies and their characteristics, by performing several assessment studies accompanied by the detailed state of the art and related works analysis. This work also contributed to the personal development of the author, given the numerous interactions with the EDP teams and the teamwork with laboratory colleagues.

Bibliography

3GPP (2007). 3GPP Scope and Objectives. Technical Report August.

3GPP (2011). TS 21.101 - Technical Specifications and Technical Reports for a UTRAN-based 3GPP system (Release 8). Technical Report Release 8.

Ahmadi, S. (2009). An overview of next-generation mobile WiMAX technology. *Communications Magazine, IEEE*, 47(6):84–98.

Ahson, S. and Ilyas, M. (2008). *WiMAX - Applications*.

Ahson, S. and Mohammad, I. (2008). *WiMAX - Technologies, Performance Analysis, and QoS*.

Alasti, M., Neekzad, B., and Hui, J. (2010). Quality of service in WiMAX and LTE networks. *IEEE Communications Magazine*, (May):104–111.

Ali, A. a., Vassilaras, S., and Ntagkounakis, K. (2009). A Comparative Study of Bandwidth Requirements of VoIP Codecs over WiMAX Access Networks. *2009 Third International Conference on Next Generation Mobile Applications, Services and Technologies*, pages 197–203.

Ali, F. and Brown, T. (2007). Delivering on the Promise of WiMAX : A White Paper from WireIE s Office of the CTO. Technical report.

Alvarion (2007). The Disruptive Approach of Open WiMAX. Technical report.

Alvarion (2011). Alvarion Training Center. 15, 16

Andersson, K., Mostafa, S. A. M., and Ui-Islam, R. (2011). Mobile VoIP user experience in LTE. *2011 IEEE 36th Conference on Local Computer Networks*, pages 785–788. 62, 66, 67, 116

Andrews, J., Ghosh, A., and Muhamed, R. (2007). *Fundamentals of WiMAX: understanding broadband wireless networking*. Prentice Hall PTR. 9, 11, 12, 18, 19, 20, 22, 29

Asheralieva, A., Khan, J. Y., and Mahata, K. (2011). Performance analysis of VoIP services on the LTE network. *2011 Australasian Telecommunication Networks and Applications Conference (ATNAC)*, pages 1–6. 62, 66, 67

Bernardo, V. (2009). *Multimedia Application Support on WiMAX : Cost / Benefit Analysis.* Msc thesis.

Bernardo, V., Pentikousis, K., Jarno, P., Piri, E., and Curado, M. (2009a). Multi-client video streaming over wirelessMAN-OFDMA. *Proceedings of the 4th ACM workshop on Performance monitoring and measurement of heterogeneous wireless and wired networks*, pages 46–53. 60, 61, 67

Bernardo, V., Sousa, B., and Curado, M. (2009b). VoIP over WiMAX: Quality of experience evaluation. *2009 IEEE Symposium on Computers and Communications*, pages 42–47. 56, 57, 67, 98

Besacier, L., Bergamini, C., Vaufreydaz, D., and Castelli, E. (2001). THE EFFECT OF SPEECH AND AUDIO COMPRESSION ON SPEECH RECOGNITION PERFORMANCE. pages 2–7.

Botta, A. and Dainotti, A. (2007). Multi-protocol and multi-platform traffic generation and measurement. *INFOCOM 2007 DEMO Session*, pages 4–5. 73, 92

Botta, A., Dainotti, A., and Pescapé, A. (2010). Do You Trust Your Software-Based Traffic Generator? *IEEE Communications Magazine*, pages 158–165.

Cao, J. and Gregory, M. (2008). Performance Evaluation of VoIP Services using Different CODECs over a UMTS Network. *2008 Australasian Telecommunication Networks and Applications Conference*, pages 67–71.

Cole, R. and Rosenbubluth, J. (2001). Voice Over IP Performance Monitoring. *ACM SIGCOMM Computer Communication Review*, 31(2):9–24. 79, 82, 102

D-ITG (2011). Distributed Internet Traffic Generator.

Dahlman, E., Parkvall, S., and Skolds, J. (2011). *4G LTE/LTE-Advanced for Mobile Broadband*. ELSEVIER. 29, 34, 35, 47

Dunaytsev, R. (2010). Network Simulators : OPNET Overview and Examples.

Durantini, A., Petracca, M., and Ananasso, F. (2008). Experimental Evaluation of IEEE 802.16 WiMAX Performances at 2.5 GHz Band. *2008 International Wireless Communications and Mobile Computing Conference*, pages 338–343. 54, 57, 67, 98

Electronic Industries Association (2011). RS-232 INTERFACE. Technical report. 3

Eren, E. (2007). WiMAX Security Architecture - Analysis and Assessment. *2007 4th IEEE Workshop on Intelligent Data Acquisition and Advanced Computing Systems: Technology and Applications*, (September):673–677.

Ericsson (2010). Voice Over LTE. (December).

Etemad, K. (2008). Overview of mobile WiMAX technology and evolution. *IEEE Communications Magazine*, 46(10):31–40. 25

Farajidana, A., Chen, W., Damnjanovic, A., Yoo, T., Malladi, D., Link, A., and Interface, S. (2009). 3GPP LTE Downlink System Performance.

Forum, W. (2008). WiMAX system evaluation methodology v2.1. Technical report. 75, 101

Forum, W. (2010). The WiMAX guide.

Forum, W. (2011). WiMAX Forum. 24, 70

ForumTM, W. (2006). Mobile WiMAX Part I : A Technical Overview and Performance Evaluation. Technical report. 24

Fu, I.-k., Chen, Y.-s., and Cheng, P. (2010). Multicarrier Technology for 4G WiMAX System. *IEEE Communications Magazine*, (August):50–58.

Garroppo, R. G., Giordano, S., and Iacono, D. (2009). Experimental and Simulation Study of a WiMAX System in the Sea Port Scenario. *2009 IEEE International Conference on Communications*, pages 1–5.

Ghosh, A., Zhang, J., and Muhamed, R. (2010). *Fundamentals of LTE*. Prentice Hall PTR.

GSA (2012). LTE Developments Worldwide - GSA Global mobile Suppliers Association - May 23, 2012. Technical report. 47

Han, M., Lee, Y., Moon, S., and Jang, K. (2008a). Evaluation of VoIP quality over WiBro. pages 51–60.

Han, T., Zhang, N., Liu, K., Tang, B., and Liu, Y. (2008b). Analysis of mobile WiMAX security: Vulnerabilities and solutions. *2008 5th IEEE International Conference on Mobile Ad Hoc and Sensor Systems*, pages 828–833.

Hasan, M. I. and Ullah, O. (2009). Performance Evaluation of IEEE 802 . 16e (Mobile WiMAX) in OFDM Physical Layer. pages 1–93.

Himayat, N., Talwar, S., Rao, A., and Soni, R. (2010). Interference Management for 4G Cellular Standards. *IEEE Communications Magazine*, pages 86–92.

Holma, H. and Toskala, A. (2009). *LTE for UMTS OFDMA and SC-FDMA Based*. 36, 38, 39, 40, 45

Hossain, E. (2007). IEEE802.16/WiMAX-Based Broadband Wireless Networks: Protocol Engineering, Applications, and Services. *Fifth Annual Conference on Communication Networks and Services Research(CNSR'07)*, pages 20–21.

IEEE (2001). IEEE Standard for Local and metropolitan area networks - Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems. Technical Report April. 11

IEEE (2004). IEEE Standard for Local and metropolitan area networks - Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems. Technical Report 2. 7, 8, 10

IEEE (2005a). 802.21 - 2008 - Part 21: Media Independent Handover Services. Technical Report January. 59

IEEE (2005b). IEEE Standard for Local and metropolitan area networks - Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems. Technical Report February. 7, 8, 10, 70

IEEE (2008). ITU-T Rec. G.711.1 -. Technical report. 74, 101

IEEE (2009). IEEE Standard for Local and metropolitan area networks - Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems. Technical Report May.

IEEE (2011a). 802.16 - Part 16 : Air Interface for Broadband Wireless Access Systems Amendment 3 : Advanced Air Interface. Technical Report May. 9

IEEE (2011b). The IEEE 802.16 Working Group on Broadband Wireless Access Standards. 8

IEEE (2012). IEEE Communications Magazine - February 2012.

IETF (1999). A One-way Delay Metric for IPPM. Technical report.

IETF (2000). Remote Authentication Dial In User Service (RADIUS). Technical Report 1645. 23

IETF (2003). Diameter Base Protocol. Technical report. 23

IETF (2004a). Extensible Authentication Protocol (EAP). Technical report. 22

IETF (2004b). Internet Low Bit Rate Codec (iLBC) specification - rfc3951. Technical report. 64

IETF (2006). Extensible Authentication Protocol Method for 3rd Generation Authentication and Key Agreement (EAP-AKA). Technical report. 22

IETF (2008a). Extensible Authentication Protocol Tunneled Transport Layer Security Authenticated Protocol Version 0 (EAP-TTLSv0). Technical report. 22

IETF (2008b). The EAP-TLS Authentication Protocol. Technical report. 22

Ikuno, J. C., Wrulich, M., and Rupp, M. (2010). System Level Simulation of LTE Networks. *2010 IEEE 71st Vehicular Technology Conference*, (i):1–5.

IPerf (2011). IPerf. 54, 73, 76, 86

ITU-T (1990). ITU-T Rec. G.726: GENERAL ASPECTS OF DIGITAL TRANSMISSION SYSTEMS; TERMINAL EQUIPMENTS - 40, 32, 24, 16 kbit/s ADAPTIVE DIFFERENTIAL PULSE CODE MODULATION (ADPCM) -. Technical report. 108

ITU-T (1996a). ITU-T Rec. G.114, One-Way transmission time. Technical report.

ITU-T (1996b). SERIES P: TELEPHONE TRANSMISSION QUALITY - Methods for objective and subjective assessment of quality. Technical report. 52

ITU-T (2001a). ITU-R BS . 1387-1 - Method for objective measurements of perceived audio quality (PEAQ).

ITU-T (2001b). P.862 - Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs. Technical report.

ITU-T (2003). ITU-T Rec. Y. 1541-Network Performance Objectives for IP-Based Services. 52, 75, 77, 78, 79, 81, 86

ITU-T (2006). ITU-T Rec. G.723.1 - Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s. Technical report. 63

ITU-T (2007). ITU-T Rec. G.729: Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP). Technical report. 63, 108

ITU-T (2009a). ITU-T Rec. G.107 - The E-model: a computational model for use in transmission planning. 52

ITU-T (2009b). ITU-T Rec. G.711 - Lossless compression of G.711 pulse code modulation. Technical report. 63, 64

Iwayemi, A. and Zhou, C. (2009). VoIP Performance in Multi-radio Mobile Devices. *2009 IEEE International Symposium on Parallel and Distributed Processing with Applications*, pages 354–360. 64, 66, 67, 116

IxChariot (2011). IxChariot. 54

Jadhav, S., Zhang, H., and Huang, Z. (2011). Performance Evaluation of Quality of VoIP in WiMAX and UMTS. *2011 12th International Conference on Parallel and Distributed Computing, Applications and Technologies*, pages 375–380.

Jou, Y.-C., Attar, R., Ma, J., and Zhang, X. (2010). Comparison of CDMA and OFDMA Reverse Link for Cellular Voice Communications. *2010 IEEE International Conference on Communications*, pages 1–6.

Katz, M. and Fitzek, F. (2009). *WiMAX evolution: emerging technologies and applications*. Wiley.

Kim, R. (2010). Advanced power management techniques in next-generation wireless networks. *IEEE Communications Magazine*, (May):94–102.

Larmo, A., Lindström, M., Meyer, M., Pelletier, G., and Torsner, J. (2009). The LTE Link-Layer Design. *IEEE Communications Magazine*, (April):52–59. 44

Lee, H., Kim, H.-d., and Chol, D.-h. (2008). Extended-rtPS+ Considering Characteristics of VoIP Codecs in Mobile WiMAX. pages 0–4.

Lee, S., Cho, K., Kim, K., Yoon, D., and Kim, T. (2009). Performance analysis of LTE-Advanced system in the downlink spatial channel model. *2009 IEEE International Conference on Network Infrastructure and Digital Content*, pages 169–171.

Li, B. and Kong, H. (2007). A Survey on Mobile WiMAX. *IEEE Communications Magazine*, (December):70–75.

Mahasukhon, P., Sharif, H., Hempel, M., Ma, T., and Lal Shrestha, P. (2011). Mobile WiMAX throughput and delay measurements in railroad environment. 58, 61

Mun, H., Han, K., and Kim, K. (2009). 3G-WLAN interworking: security analysis and new authentication and key agreement based on EAP-AKA. *2009 Wireless Telecommunications Symposium*, pages 1–8.

Myung, H. G. (2008). Technical Overview of 3GPP LTE. Technical report.

Nasreldin, M., Aslan, H., El-Hennawy, M., and El-Hennawy, A. (2008). WiMax Security. *22nd International Conference on Advanced Information Networking and Applications - Workshops (aina workshops 2008)*, pages 1335–1340.

Nport (2011). MOXA 5100A. 93

Oh, S. and Kim, J. (2010). User-perceived QoS performance enhancement for VoIP services in IEEE 802.16 systems. *Pervasive Computing and Communications Workshops (PERCOM Workshops), 2010 8th IEEE International Conference on*, pages 570–574.

Olariu, C., Foghlu, M. O., Perry, P., and Murphy, L. (2011). VoIP quality monitoring in LTE femtocells. *12th IFIP/IEEE International Symposium on Integrated Network Management (IM 2011) and Workshops*, pages 501–508. 63, 66, 67, 116

OPNET (2012). OPNET - Modeler 17.1. 63, 99, 122

Oyman, O., Foerster, J., and Corporation, I. (2010). Toward Enhanced Mobile Video Services over WiMAX and LTE. *IEEE Communications Magazine*, (August):68–76.

Pareek, D. and Group, F. (2006). *WiMax: Taking Wireless to the MAX*.

Park, Y., Leem, H., and Sung, D. K. (2010). Power Saving Mechanism in IEEE 802 . 16m.

Parkvall, S. and Dahlman, E. (2011). Evolution of LTE toward IMT-Advanced. *IEEE Communications Magazine*, (February):84–91.

Prasad, A. R. (2011). 3GPP SAE / LTE Security.

Prasad, N. R. (2009). State of the Art of the Wireless Security in OFDM(A)-Based Systems. *2009 IEEE Mobile WiMAX Symposium*, pages 107–110. 23

Qualnet (2012). Qualnet. 63

Radio-Electronics (2011). GSM Audio Codec / Vocoder. 63

Rengaraju, P., Lung, C.-h., and Srinivasan, A. (2011). Measuring and Analyzing WiMAX Security and QoS in Testbed Experiments. *Communications (ICC), 2011 IEEE International Conference on*, pages 1–5.

Rottinghaus, A., Hariharan, R., and Guo, X. (2011). Supporting High Speed on WiMAX: From Theory to Practice. (Dl).

Sauter, M. (2009). *Beyond 3G: bringing networks, terminals and the Web together*. 31, 50

Sawahashi, M., Kishiyama, Y., Taoka, H., Tanno, M., and Nakamura, T. (2009). Broadband radio access: LTE and LTE-advanced. *2009 International Symposium on Intelligent Signal Processing and Communication Systems (ISPACS)*, (Ispacs):224–227.

Seok-Yee, T., Muller, P., and Sharif, H. R. (2010). *WiMAX - Security and Quality of Service - An End-to-End Perspective*. Wiley.

Siow, L. K., Hui, L., Angie, L., So, P. L., and Member, S. (2011). Measurements and Analysis of Fixed Wimax with Lan in Microgrid. *Electronic Engineering*, pages 162–165. 53, 57

So-In, C., Jain, R., and Tamimi, A.-K. (2010). Capacity Evaluation for IEEE 802.16e Mobile WiMAX. *Journal of Computer Systems, Networks, and Communications*, 2010(1):1–12.

Sousa, B., Curado, M., and Pentikousis, K. (2008a). Evaluation of Multimedia Services in Mobile WiMAX. pages 64–70. 59, 61

Sousa, B., Monteiro, E., and Curado, M. (2008b). IP and WiMAX coalition. pages 1–6.

Sweeney, D. (2006). *WiMax Operators Manual - Building 802.16 Wireless Networks (Second edition)*. 30

TCPDUMP (2011). TCPDUMP. 94

Tiwari, B. P. (2010). WiMAX 2.0 for Operators. Technical report.

Torad, M., El Qassas, A., and AL henawi, H. (2011). Comparison between L TE and WiMAX based on System Level Simulation Using OPNET modeler (release 16). *28th NATIONAL RADIO SCIENCE CONFERENCE*, (NRSC).

Tran, M., Zaggoulos, G., Nix, A., and Doufexi, A. (2008). Mobile WiMAX: Performance Analysis and Comparison with Experimental Results. *2008 IEEE 68th Vehicular Technology Conference*, pages 1–5. 58, 61, 67, 107

Ustafsson, E. V. A. G., Onsson, A. N. J., and Esearch, E. R. R. (2003). Always Best Connected. *IEEE Wireless Communications*, (February):49–55.

Wang, F., Ghosh, A., Sankaran, C., Fleming, P., Hsieh, F., and Benes, S. (2008). Mobile WiMAX systems: performance and evolution. *IEEE Communications Magazine*, 46(10):41–49.

Wang, H. and Lte, A. V. (2007). Uplink Capacity of VoIP on LTE System. In *Proceedings of Asia-Pacific Conference on Communications 2007*, number 11, pages 397–400.

Wang, Z., Wang, Y., Jiang, D., Tian, C., and Yang, D. (2009a). Scheduling and Link Adaptations for VoIP in TDD-LTE Uplink. (2007).

Wang, Z., Wang, Y., and Wang, F. (2009b). Comparison of VoIP Capacity between 3G-LTE and IEEE 802.16m. pages 2192–2196. 64, 66, 67, 116

WeDo (2012). The unstoppable LTE and its impact on CSPs operations.

Westall, J. and Martin, J. (2004). Performance Characteristics of an Operational WiMAX Network. (99):1–1.

Wireshark (2012). Wireshark- Network Trace Tool. 65

Xiehua, L. and Yongjun, W. (2011). Security Enhanced Authentication and Key Agreement Protocol for LTE/SAE Network. pages 0–3.

Yousaf, F. Z., Daniel, K., and Wietfeld, C. (2007). Performance Evaluation of IEEE 802.16 WiMAX Link With Respect to Higher Layer Protocols. *2007 4th International Symposium on Wireless Communication Systems*, pages 180–184. 55, 57

Zhang, H., Boutabia, M., and Nguyen, H. (2011a). Field performance evaluation of VoIP in 4G trials. pages 2–5.

Zhang, H., Gu, Z., and Tian, Z. (2011b). QoS Evaluation Based on Extend E-Model in VoIP. *ICACT 2011*, (2):852–854.

. Appendices

Appendix A - VoIP performance over Mobile WiMAX: An Urban Deployment Analysis

This Appendix shows the submitted and accepted paper to the 2nd IEEE Baltic Conference on Future internet Communications, 25-27 April 2012 in Vilnius, Lithuania.

The results and the first version of the paper was written by the author. The later versions were made in collaboration with Vitor Bernardo, and revised by Professor Marília Curado and Professor Paulo Simões.

VoIP performance over Mobile WiMAX: An Urban Deployment Analysis

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Abstract—**The significant growing of Voice over IP (VoIP) ready mobile devices raises new challenges in the deployment of novel broadband wireless access networks (BWA), such as WiMAX or Long Term Evolution (LTE). Due to voice service importance for the mobile market, the empirical assessment of voice traffic performance and quality is crucial for a successful deployment. In this work, the Mobile WiMAX (IEEE 802.16e) capabilities to support VoIP traffic under different scenarios and employing distinct Quality of Service (QoS) service classes were performed. Additionally, the paper characterizes the heterogeneity access conditions within a city, by analyzing both Line of Sight (LOS) and Non-Line of Sight (NLOS) conditions. By examining the end-user perceived quality (Quality of Experience) and the network QoS related parameters, the attained results shown the impact of the correct WiMAX QoS service classes management on the number of well served VoIP users.**

Keywords-Mobile WiMAX; VoIP; Testbed; Quality of Service; Quality of Experience; IEEE 802.16e

I. INTRODUCTION

Future Internet will encompass an increasing number of users, aiming to be always best connected [1] anytime and anywhere. Besides network availability, application performance should also be taken into account, particularly when using multimedia applications, such as Voice over IP (VoIP) or Video Streaming. Novel broadband wireless access technologies, such as Worldwide Interoperability for Microwave Access (WiMAX) or Long Term Evolution (LTE), are being developed aiming to accomplish these goals. The exhaustive usage of the 4G and beyond "all-IP" broadband wireless access technologies, where all the traffic will be managed as an IP packet, raise new issues related with end-user quality perception.

By using an urban real Mobile WiMAX deployment in the city of Coimbra, Portugal, this work performs an empirical evaluation of VoIP performance over Mobile WiMAX, performing experiments in different access scenarios, namely in Line of Sight (LOS) and Non-Line of Sight (NLOS) conditions. The assessment includes also the study of Mobile WiMAX native Quality of Service (QoS) mechanisms through the usage of different QoS service classes, and a scalability / performance tradeoff analysis when supporting several simultaneous VoIP calls. Besides analyzing the network typical QoS parameters, such as packet loss rate or jitter, this work also performs a Quality of Experience (QoE) assessment of each VoIP call, aiming to comprehend the impact of the studied network conditions and scenarios in the end-user perceived quality.

The rest of the paper is organized as follows: Section II gives a brief description of the WiMAX technology, and presents the main motivations of this work. Section III discusses the most significant related work on WiMAX assessment. In Section IV the testing methodology and the testbed configuration for the VoIP evaluation are described. The results obtained in the Mobile WiMAX testbed are presented in Section V. Finally, Section VI presents the conclusions and future work.

II. BACKGROUND AND MOTIVATION

WiMAX and LTE are both broadband wireless access (BWA) network technologies, aiming to play a crucial role in 4G and beyond "all-IP" broadband wireless access technologies progress. LTE is an evolved 3GPP technology [2], and as WiMAX, was developed to support higher number of users with higher data rates, coverage and availability. However, LTE is not in the scope of this work, which focuses on the WiMAX technology. This technology is based on the IEEE 802.16 standards [3], and the most relevant versions are the Fixed WiMAX, based on IEEE 802.16d [4], and the Mobile WiMAX, defined by IEEE 802.16e [5]. The latter has significant improvements in the support of multiple users, as well as new mechanisms for Quality of Service, mobility support and also energy efficiency. This technology is planned to achieve long ranges of coverage with high availability and throughput, allowing rural or urban wireless access in diverse deployment environments, namely with and without line of sight.

The QoS support is provided by different service classes, allowing the traffic flows differentiation by setting the applicable network parameters, namely the maximum and minimum reserved traffic rate, the maximum allowed delay, or the maximum tolerable jitter. The service classes ordered by traffic prioritization (higher to lower) are the following: Unsolicited Grant Service (UGS), Extended Real Time Polling Service (ertPS), Real Time Polling Service (rtPS), Non-Real Time Polling Service (nrtPS), and Best Effort (BE). Each QoS service class is suitable to different applications (e.g. rtPS is befitting for real-time applications and nrtPS performs better for bulk file transfers).

The WiMAX QoS model is based on *service flows*. Each *service flow* is a unidirectional flow, with QoS parameters defined to fulfill the application requirements, to which is assigned a connection, identified by a *connection identifier* (CID). Each node can have multiple connections and *service flows*. The Base Station (BS) always manages the traffic scheduling for the different service classes, whether in the uplink or downlink. Another relevant feature of Mobile WiMAX is the support of energy saving mechanisms, such as the idle and sleep modes. The idle mode allows the node to be completely turned off and deregistered from the base stations for certain periods, consuming less energy, while the sleep mode disrupts the nodes connections individually, needing normal handoff procedures, and so, consuming more energy than the idle mode. These energy efficiency mechanisms are particularly relevant in the context of mobile networks, since the majority of connected devices are battery based.

It should be noted that the technology supports different access scenarios, namely fixed, nomadic and mobile. WiMAX technology supports full mobility, allowing the access to the network at moving speeds, supporting vertical *seamless handovers*, where the users do not notice the attachment point change.

The main goal of this work is to assess the WiMAX technology capabilities to support VoIP traffic in a real deployment urban testbed, within LOS and NLOS scenarios, using different native WiMAX QoS mechanisms. This empirical study is focused on the assessment of the end-user perceived quality, also known as QoE, in a multi-user environment performing real VoIP calls, emulated by simultaneous bi-directional VoIP traffic flows.

III. RELATED WORK

This section presents the most relevant related work on WiMAX assessment.

Oh et al. [6] evaluate distinct WiMAX QoS mechanisms using Automatic Repeat Request (ARQ) in a simulation environment using OPNET [7]. In this work the VoIP traffic assessment was performed with special emphasis on the uplink scheduling mechanisms of WiMAX. The study encompasses the usage of three different WiMAX QoS classes, namely ertPS, rtPS and UGS. The results demonstrate that the usage of ARQ in ertPS can save wireless resources, while the usage of this mechanism in UGS causes deterioration in the perceived VoIP quality.

A study of voice traffic over Mobile WiMAX using both LOS and NLOS conditions was performed by Zhang et al. [8]. The evaluation procedure was performed during the handover (Hard Hand Over (HHO)) in order to assess the impact of this procedure during a voice call. The network parameters evaluated are the jitter, the packet loss and the delay. The Quality of Experience, assessed using the Perceptual Evaluation of Speech Quality (PESQ), was obtained by sending an audio file from the sender to receiver. Then, an analysis of the audio file is conducted, resulting in a value between 1 and 4.5, representing the end user perceived quality. The tests are performed without specifying a channel with QoS (i.e., using a BE channel). This work verified that when the node is at the cell-center with LOS the PESQ values are higher, when compared to the cell-edge NLOS values. Furthermore, it was also showed that performing HHO has a direct impact in the packet loss, and consequently, in the end-user perceived quality.

Jadhav et al. [9] evaluate, using the OPNET simulator, the differences between WiMAX and Universal Mobile Telecommunications System (UMTS) when transmitting voice data to multiple users simultaneously, employing the G.711 codec. The channel configuration used is BE for both technologies. This work evaluates network QoS parameters such as jitter, delay and packet loss, but also the QoE perceived by the end user, through the MOS. This evaluation is limited to a low number of simultaneous users and the main focus is on the comparison between WiMAX and UMTS, leaving behind the comparison between different QoS service classes. WiMAX has proven to have better capabilities while supporting VoIP applications, allowing a higher number of simultaneous users within a good perceived quality.

Durantini et al. [10] evaluated the WiMAX performance and capabilities in a real Fixed WiMAX testbed when transmitting Video on Demand (VoD), video streaming and web traffic with different QoS service classes, within different LOS and NLOS conditions in fixed and nomadic scenarios. Additionally, it also assesses the difference between different modulation schemes. These measurements are focused only in QoS metrics, such as delay and throughput. The results demonstrated that higher throughputs were achieved with more complex modulation schemes and that the rtPS is the most suited service class for video transmissions.

Bernardo et al. [11] present VoIP traffic evaluation over a real WiMAX tested using different transport protocols, such as User Datagram Protocol (UDP) / Real-time Transport Protocol (RTP) and Datagram Congestion Control Protocol (DCPP) in overestimated and underestimated scenarios. From the results obtained, it is shown that in overestimated scenario the One Way Delay (OWD) and packet loss have higher values when using DCCP protocol than with UDP. MOS also decreases, since it is related to the OWD and packet loss. On underestimated scenarios, the behavior when using DCCP is better than in the overestimate scenario. Nonetheless, it is always worst than when using UDP, which support higher MOS values. Although this evaluation also assesses the end-to-end delay, packet loss and user perceived MOS, it is only performed over an rtPS configured channel.

Table I summarizes the discussed related work. The “Type” indicates the environment used in the tests, and the WiMAX related fields, “Version” and “QoS Service Classes”, show the used WiMAX version and the native WiMAX QoS reservation channel employment during the tests. Finally, the “Assessment” related fields depict the used metrics during for the performed evaluations.

TABLE I. RELATED WORK SUMMARY

Work	Type	WiMAX		Assessment Metrics	
		Version	***QoS Service Classes***	***QoS***	***QoE***
Oh et al. [6]	Simulation	Mobile	Yes	Yes	Yes
Zhang et al. [8]	Testbed	Mobile	No	Yes	Yes
Jadhav et al. [9]	Simulation	Mobile	No	Yes	Yes
Durantini et al. [9]	Testbed	Fixed	Yes	Yes	No
Bernardo et al. [11]	Testbed	Fixed	Yes	Yes	Yes
This Work	Testbed	Mobile	Yes	Yes	Yes

From the works described, none has an analysis of the end-user perceived voice quality while comparing different WiMAX QoS service classes, using a Mobile WiMAX real deployment. This work aims to fill this gap, in a real Mobile WiMAX testbed deployed in a city environment. This assessment will also evaluate the differences between some of the available QoS service classes, and both LOS and NLOS scenarios, using real end-user equipment.

IV. METHODOLOGY

In this section the several aspects of the tests specification and methodologies, as well as the testbed configuration and equipment used are presented.

A. *Testbed Configuration*

This work was performed on a Mobile WiMAX (IEEE 802.16e) real deployment testbed, consisting of two BSs - *Alvarion BreezeMax Macro Outdoor* connected to the same Access Service Network (ASN)-Gateway - *Alvarion BreezeMax ASNGW Mini-Centralized* - through a centralized architecture, making the testbed fully compliant with the WiMAX Forum Network Reference Model (NRM) [12]. The testbed comprehends two Base Stations, but only the one providing connectivity in urban conditions was used in this assessment. Each BS is equipped with two Dual-Slant antennas (65º for each sector, with dual polarization).
The BS is configured with 4x2 Multiple Input Multiple Output (MIMO) Matrix A (allowing better coverage, i.e., both antennas send the same data stream), operating in 2.615GHz, with a 10MHz channel bandwidth. This channel is configured with Adaptive Modulation and Coding (AMC), allowing different modulation schemes for different clients. Each client may modify the modulation scheme due to weather conditions, line of sight constrains or distance to the BS. As the communication mechanism is Time Division Duplex (TDD), the frames are divided to the uplink and downlink communications. In this testbed, the frame configuration is 60%/40%, where 60% is for downlink and 40% is for uplink. The equipment configuration was the same in all the tests performed. In order to assess LOS and NLOS scenarios, two distinct locations were used and named correspondingly, LOS and NLOS. These locations are shown in Figure 1.

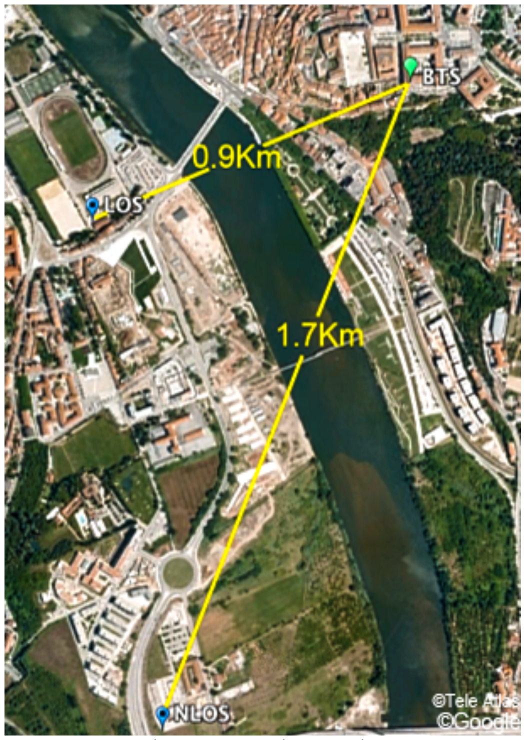

Figure 1. Scenarios Location

Both locations are within an urban scenario. The one without line of sight (NLOS) is located 1.7Km away from the BS. This location was used since it represents a good testing location for NLOS urban conditions, where the node is connected to the network behind buildings and where the signal is only obtained through reflections (i.e., line of sight to the BS is not actually possible). The one with line of sight (LOS) is located 0.9Km away from the BS, allowing near-optimal conditions for the nodes (close to the BS, clear line of sight).
All the tests were performed with mobile nodes, using two netbooks equipped with an USB Customer Premises Equipment (CPE). In the core network, a fixed node was used, connected to the network via Ethernet. All the nodes were running *Debian Linux*. The testbed architecture is illustrated in Figure 2.

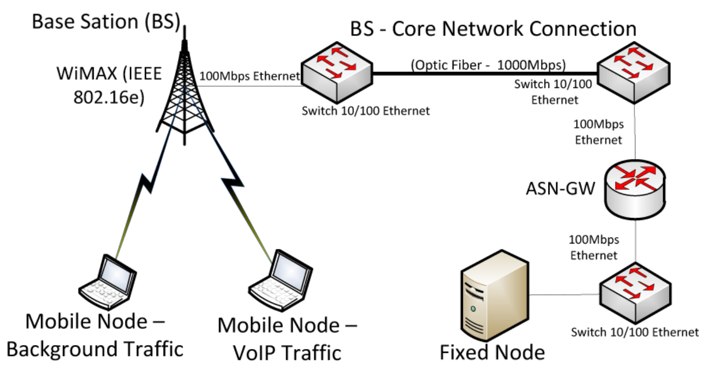

Figure 2. IEEE 802.16e Testbed Architecture

B. Tests Specification

To perform the WiMAX technology capabilities assessment, a set of scripts and applications were needed. Based on the output network parameters and in the related work, the D-ITG [13] tool was used, since it allows different types of emulated traffic (e.g., VoIP with G.711.1 CODEC). D-ITG is able to generate realistic traffic patterns, and also provides an analysis of the most common network parameters, such as jitter, delay, and packet loss. The packet rate of voice data was 100 packets per second, with a compression rate of 96Kbps, as defined by CODEC G.711.1 [14]. The number of simultaneous flows was: 1, 5, 10, 20, 30, 40, 50 and 60 flows, where each flow is intended to emulate one real bi-direction call (i.e., one user). For all the tests the same traffic generator seed was used, in order to generate similar traffic patterns across the tests. Each test is repeated through 5 runs to avoid the outlier results caused by signal fluctuations and unpredictable changes in the environment, since it is a real deployed testbed. The standard deviation is represented by vertical lines in graphs, calculated with a confidence interval of 95%. The flows were started incrementally, and the measured period was when all flows were active. This methodology was used to avoid packet loss at the sender side, caused by buffer overflows.

All the VoIP traffic was transmitted over an rtPS or a BE channel. Initially, this communication was planned to be over an ertPS or UGS channel, which are best suited for VoIP traffic, but due to Alvarion hardware/software limitations, it was not possible to use these mechanisms. These limitations are caused when setting the maximum reserved rate to high values (in order to allow a large number of flows), and, since this service class is only prepared for low bandwidth usage (voice data), it was not possible to emulate more than 10 simultaneous voice calls.

The usage of rtPS mechanism was due to its guarantees, such as maximum tolerated OWD, reserved rate and traffic prioritization. The maximum tolerated OWD was set to 150ms, as recommended in ITU-T Y.1541 [15] and by WiMAX Forum [16]. The reserved rate was overestimated, in order to guarantee a higher rate than needed by all the flows, defined as 15Mbps. BE was used with the objective of evaluating the real differences between both service classes, where the reserved rate for this class was also 15Mbps. In order to complement this analysis, these tests were also carried out with background traffic. Using the tool IPerf [17], random traffic was emulated to represent a channel obstructed with data from other users. This traffic was generated in a BE channel, using all the bandwidth allowed by the BS. Therefore, the main goal was to evaluate the efficiency of native QoS mechanisms of WiMAX, observing the behavior of VoIP traffic when the channel is obstructed.

C. Evaluation Metrics

This section describes the main evaluation metrics used in the following tests, describing each one by categories: Quality of Service and Quality of Experience.

1) Quality of Service

The network QoS metrics assessed during this work are the delay and packet loss. The delay is measured through the Round Trip Time (RTT). As the mobile nodes are connected to the network only through WiMAX and they are not equipped with Global Positioning System (GPS) cards, it was not possible to accurately synchronize them to enable the correct OWD measurement.

The packet loss rate assessment enables the comparison of the obtained results with the ITU-T Y.1541 Recommendation, where the acceptable ranges for the different network Quality of Service are defined. For instance, for VoIP applications the ITU-T Y.1541 defines 150ms as the maximum acceptable OWD and a maximum packet loss of 1%. These QoS metrics allow the evaluation of network conditions, however, they do not allow assessing the impact of those conditions in the end-users perceived quality. To overcome this limitation, Quality of Experience metrics were also employed, as explained in the next subsection.

2) Quality of Experience

The Quality of Experience perceived by the end user is the main focus of this work, and it can be measured through several metrics. The most commonly used metrics are the MOS, E-Model, PESQ [18] and Perceptual Evaluation of Audio Quality (PEAQ) [19]. Since the tests were intended to be non-intrusive, the MOS and E-Model [20] were employed. MOS consists of several users evaluating one service, giving a score of one to five, where one is the worst value and five is the best value. MOS is a subjective metric commonly used in voice applications. However, the E-Model overcomes the need of real users and the subjectiveness associated to this evaluation, calculating the R-Factor and associating it value to the MOS Scale. The E-Model evaluates the VoIP Quality of Experience through network parameters, such as OWD and packet loss rate. In Cole et al. [21] the authors have reduced the R-Factor calculation to an equation based on OWD and packet loss values. Those equations were used to perform the MOS calculations in this work.

It should be noted that the MOS values depend on several factors, not only the network parameters such as delay and packet loss, but also on the codec used. For the G.711.1 codec used in this work, based on the calculations and parameters provided, the maximum MOS value in optimal conditions is 4.43 [21]. Table II depicts the relationship between the end-user perceived quality and MOS values.

TABLE II. MOS VALUES

MOS	Quality
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

V. Experimental Assessment

In this section the main goals and objectives are explained, as well as the analysis of the results obtained and their description.

A. Objectives

This assessment aims to achieve three main goals. The first consists on the native WiMAX QoS mechanisms evaluation and their impact on the end-user QoE, using different WiMAX service classes, such as rtPS and BE. The second objective measure is the impact of different line of sight conditions in both QoS and QoE metrics. Finally, this empirical works aims to address the effect of background traffic in the end-user perceived quality, as well as in the WiMAX network global performance. The results are presented and discussed in the following subsections.

B. LOS Scenario

In this subsection the results obtained in LOS scenarios are shown and explained.

1) Packet Loss without BG traffic

Figure 3 depicts the packet loss without background traffic in LOS conditions. The packet loss values are comparable in both service classes.

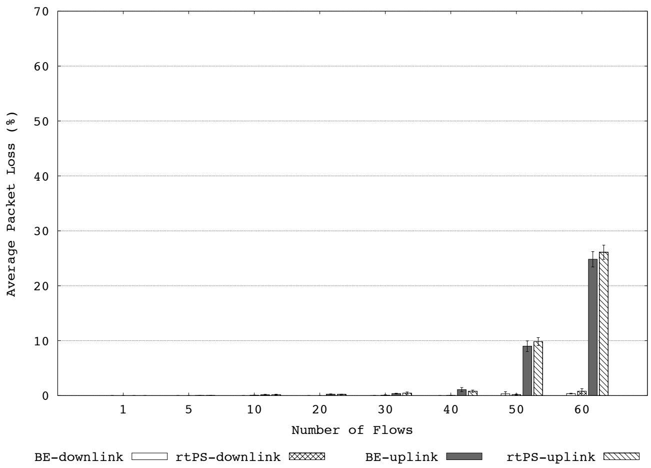

Figure 3. Packet Loss without background traffic in LOS conditions

In the uplink, the first noticeable impact of the packet loss is caused when transmitting more than 30 simultaneous flows (i.e., < 1%, as advise in ITU-T Y.1541 recommendation). For instance, when transmitting 40 simultaneous flows, the uplink packet loss is 1% for BE and 0.8% for rtPS. With 50 simultaneous flows, these values are, respectively, 9% and 9.8% for BE and rtPS. In the downlink, the packet loss values remain below 1% with all simultaneous flows, for both service classes. From these results, it is possible to observe that the packet loss increases with the growth of the number of flows, since the bandwidth usage becomes higher.

The differences between uplink and downlink are due to USB stick transmission power limitations and the uplink/downlink testbed frame ratio configuration.

2) Packet Loss with BG traffic

Figure 4 depicts the packet loss with background traffic in LOS conditions. In this case, the maximum feasible packet loss limit (i.e., < 1%) is achieved with 20 simultaneous flows. For 30 simultaneous flows, packet loss rate is already 4%, which is 3% beyond the acceptable limit.

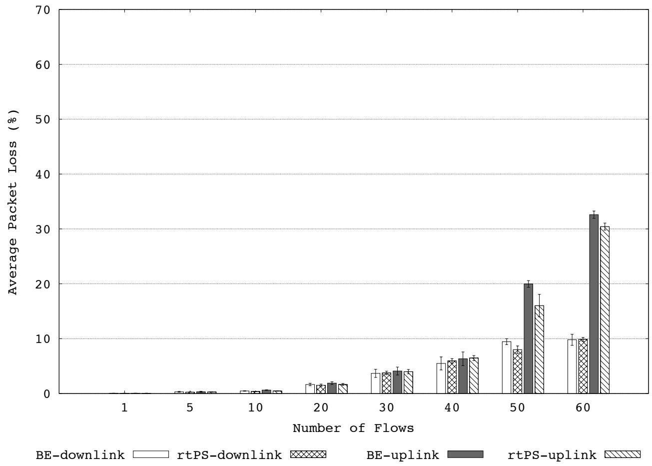

Figure 4. Packet Loss with background traffic in LOS conditions

The packet loss in the presence of background traffic is higher than when just VoIP traffic is being transmitted (Figure 3). These values are explained due to the high traffic load on the link, due the background traffic usage. The rtPS service class effectiveness is perceived, but only marginally.

3) MOS without BG traffic

Figure 5 shows the MOS without background traffic in LOS conditions.

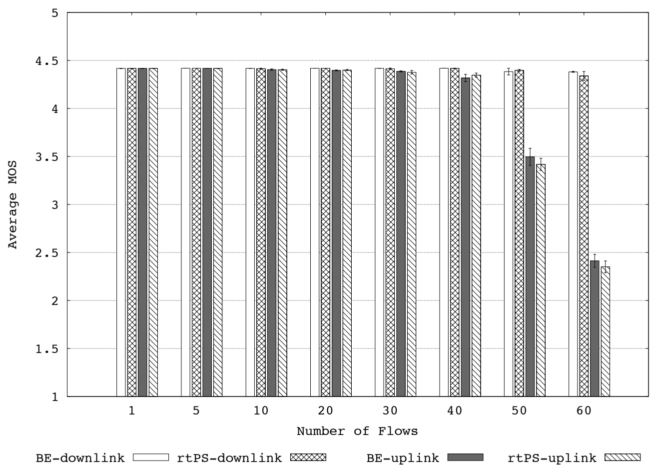

Figure 5. MOS without background traffic in LOS conditions

It demonstrates that without background traffic in LOS conditions it is possible to support up to 50 flows with "fair" quality (i.e., higher than 3). In this scenario, rtPS is slightly worse than BE, which can be caused by the QoS scheduling mechanisms overhead.

4) MOS with BG traffic

The MOS with background traffic in LOS conditions is depicted in Figure 6. It is noticeable that the rtPS service class has some advantages when compared with BE. It guarantees "fair" quality up to the same 50 flows as without background traffic (Figure 5), while the BE service class only supports up to 40 simultaneous flows.

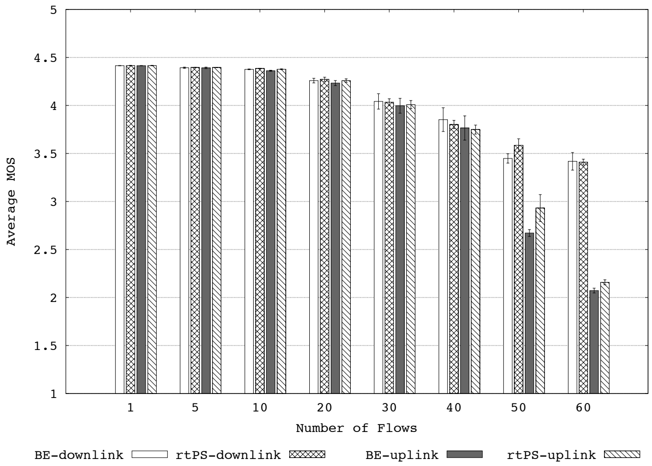

Figure 6. MOS with background traffic in LOS conditions

In Line of Sight conditions with background traffic, the differences between downlink and uplink are not so noticeable, since both communication directions are congested.

C. NLOS Scenario

This subsection presents the results concerning assessment performed in Non-Line of Sight conditions.

1) Packet Loss without BG traffic

Figure 7 shows the packet loss rate without background traffic in NLOS conditions.

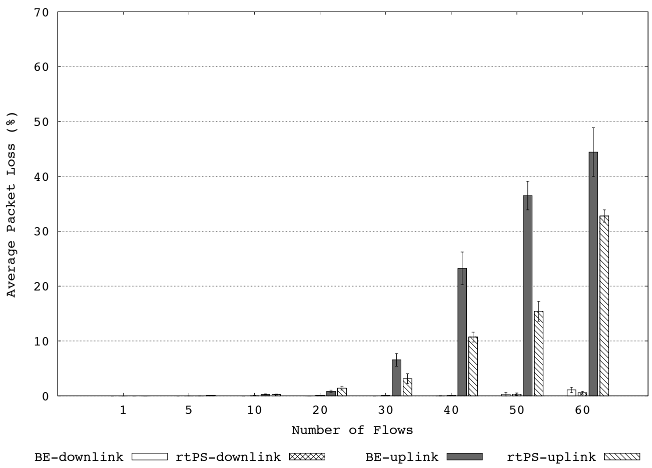

Figure 7. Packet Loss without background traffic in NLOS conditions

This figure shows that WiMAX supports 30 simultaneous flows under 10% for BE service class and 40 simultaneous flows under 10% for rtPS. In this scenario, for the same number of simultaneous flows, the packet loss rate is higher than in LOS scenario (Figure 3). This fact is caused by the lower bandwidth available due to the worst signal quality and the low CPE transmission power. Also, in this scenario, the differences between BE and rtPS are more noticeable.

2) Packet Loss with BG traffic

Figure 8 shows the packet loss percentage with background traffic in NLOS conditions. The differences between rtPS and BE are visible, but with minor differences.

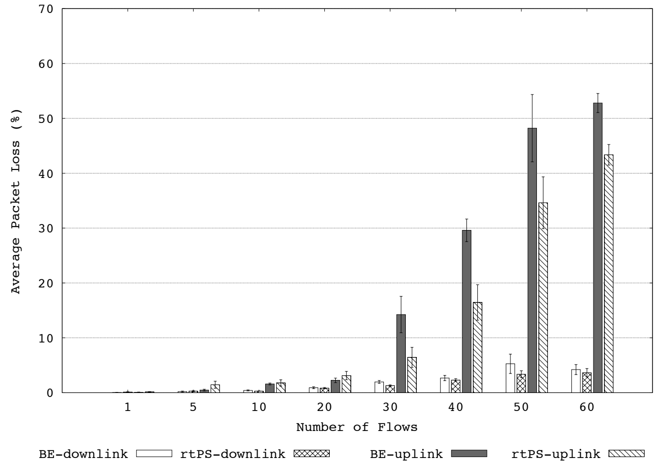

Figure 8. Packet Loss with background traffic in NLOS conditions

Figure 8 shows that in NLOS conditions with background traffic it is possible to support, under 10% of packet loss, up to 20 simultaneous flows for BE service class and 30 simultaneous flows for rtPS. This represents a lower capacity for supporting several flows when compared to Figure 4, where the packet loss values, for both uplink and downlink and rtPS and BE, were bellow 10% until 40 flows. The worst obtained values are in this scenario, as expected, since it represents the NLOS access conditions with a full load channel.

3) MOS without BG traffic

The Mean Opinion Score metric with background traffic in NLOS access condition is presented in Figure 9.

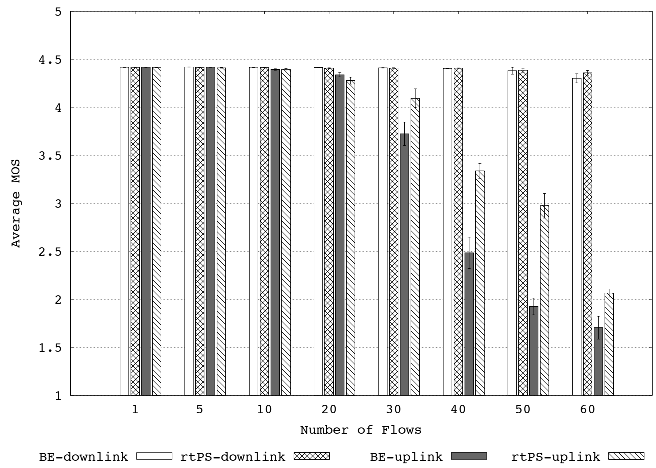

Figure 9. MOS without background traffic in NLOS conditions

In this scenario, the WiMAX technology was able to support up to 50 simultaneous flows using the rtPS service class, while with BE only 30 simultaneous flows are sustained. This well sustained number assumes the expected “fair” conditions to the end-user perceived quality (i.e., QoE). When comparing the NLOS and LOS scenarios without background traffic, (Figure 5 and Figure 9) the MOS values start decreasing at 40 simultaneous flows in LOS, while in NLOS, it starts decreasing at 20 flows. In short, the rtPS service class supports, in both NLOS and LOS, up to 50 simultaneous flows with “fair” quality. The BE class supports up to 50 flows in LOS and, in NLOS, supports a maximum of 30 flows.

4) MOS with BG traffic

Figure 10 depicts MOS values with background traffic in NLOS conditions. In this scenario, WiMAX supports up to 40 flows in the rtPS service class, while in BE it supports up to 30 simultaneous flows (with “fair” conditions). This represents a decrease of 10 flows for rtPS, when compared to the previous scenario, without background traffic. Also, the differences between rtPS and BE are clearly shown, where rtPS supports more 10 flows than BE.

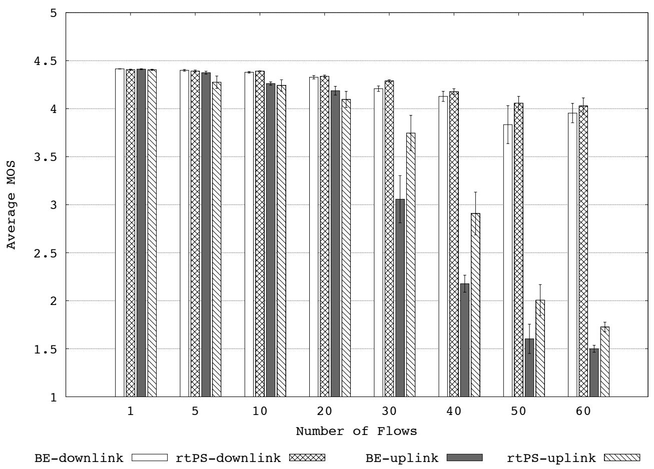

Figure 10. MOS with background traffic in NLOS conditions

Comparing the NLOS and LOS scenarios with background traffic, Figure 6 and Figure 10, the MOS values start decreasing at 20 simultaneous flows in LOS, while in NLOS it start deceasing at 5 flows. When comparing rtPS service class with BE, the rtPS supports up to 40 simultaneous flows in NLOS and 50 in LOS, while the BE in LOS supports up to 40 flows and in NLOS supports a maximum of 30 flows.

D. Evaluation Summary

This analysis demonstrated the efficiency of the WiMAX QoS mechanisms. It has been shown that the rtPS service class always has a better performance than BE, although in LOS conditions this difference is barely noted. The differences between rtPS and BE can result in less 20 simultaneous flows well sustained.

The results also show the benefits from access to the network in LOS when compared with NLOS conditions. This is an expected result, since the NLOS scenario offers much more interferences than the LOS scenario. Mostly due to the low USB CPE transmission power, leading to worse Signal-to-Noise-Ratio (SNR) values, and the factors that arise in NLOS scenarios, such as multi-path fading and shadowing, the NLOS values are worse than in LOS.

The rtPS service class, without background traffic, supports the same amount of flows either in LOS or NLOS (50 simultaneous flows), while with background traffic it decreases from 50 simultaneous flows in LOS to 40 flows in NLOS. The BE class without background can support up to 50 flows in LOS and 40 flows in NLOS. With background traffic, this number decreases to 40 flows in LOS and to 30 flows in NLOS.

These results clearly depict the differences between LOS and NLOS, with and without background traffic and the importance of the correct service classes employment.

The impact of background traffic in the end-user QoE is noticeable in both LOS and NLOS access scenarios, resulting in the decreasing of entire simultaneous flows well supported.

VI. CONCLUSION

The empirical assessment of the 4G and beyond broadband wireless networks, such as WiMAX, is very important since it allows the real evaluation of these technologies in real deployment conditions. Together with the assess of the network related parameters, such as delay or packet loss, also the end-user perceived quality should be considered, especially when transmitting multimedia application, such as VoIP or Video Streaming. This work provides an evaluation, in a real urban deployment, of the WiMAX technology capabilities of supporting multi-user VoIP calls in real access locations and environments, such as LOS and NLOS. Since Base Station equipment is able to support several simultaneous users, it is also relevant to observe the impact of other random background traffic in the QoE of the established VoIP calls.

This performed analysis showed that the changes in the Line of Sight could cause a large impact on the number of supported simultaneous flows, decreasing the average end-user perceived quality. In NLOS conditions, mostly because of the impact of multi-path fading and also due to the USB CPE limited transmission power, the support for multiple users is lower. The CPE transmission power affects the signal-to-noise ratio, causing a lower signal quality, which leads to a lower available bandwidth than in LOS scenario.

The background traffic has a large impact on the end-user VoIP QoE, since it will obstruct the channel with a random traffic, requiring the need of prioritization in the BS scheduling mechanisms. The different WiMAX QoS service classes, as evaluated in this work, support this traffic prioritization, allowing more simultaneous users with higher MOS values. The QoS service classes used were rtPS and BE, as the rtPS class guarantees the maximum OWD and the reserved rate, while the BE class does not offer any QoS guarantee. Because of these guarantees, it is shown that the rtPS has some advantages of maintain higher QoE levels, as well as the support for more simultaneous users. This service class can achieve higher MOS values with background traffic than BE. The BE service class allowed to maintain good QoE levels in LOS without background traffic, where the BS scheduling algorithm cause some overhead to the rtPS traffic, allowing the BE to achieve similar results. Although this VoIP tests were conducted in rtPS class, due to software and hardware limitations, the most suited mechanisms for VoIP are ertPS and UGS, as they guarantee the maximum allowed jitter, delay and the reserved rate.

By conducting this analysis in a city environment with a real testbed and with similar end-user equipment as the network providers (i.e., USB CPEs), it was possible to provide a good and realistic assessment of a real world communications scenario.

ACKNOWLEDGMENT

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REFERENCES

[1] Ustafsson, E. V. A. G., Onsson, A. N. J., & Esearch, E. R. R. (2003). Always Best Connected. *IEEE Wireless Communications*, (February), 49-55.

[2] 3GPP, “TS 21.101 - Technical Specifications and Technical Reports for a UTRAN-based 3GPP system (Release 8),” 2011.

[3] IEEE, “IEEE Standard for Local and metropolitan area networks - Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems,” *IEEE Standard*, no. May, 2009.

[4] IEEE, “IEEE Standard for Local and metropolitan area networks - Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems,”, *IEEE Standard,* 2004.

[5] IEEE, IEEE Standard for Local and metropolitan area networks - Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems, vol. 2005, no. February. 2005.

[6] Oh, S.; Kim, J.; , "User-perceived QoS performance enhancement for VoIP services in IEEE 802.16 systems," *Pervasive Computing and Communications Workshops (PERCOM Workshops), 2010 8th IEEE International Conference on* , vol., no., pp.570-574, March 29 2010 - April 2 2010 doi: 10.1109 / PERCOMW . 2010.5470502.

[7] OPNET. Last Visit: 10/02/2012. Available: http://www.opnet.com/

[8] Zhang, H.; Mohammed, B.; Hang, N.; Xia, L.; , "Field performance evaluation of VoIP in 4G trials," *Multimedia and Expo (ICME), 2011 IEEE International Conference on* , vol., no., pp.1-4, 11-15 July 2011 doi: 10.1109 / ICME.2011.6011962

[9] Jadhav, S., Zhang, H., & Huang, Z. (2011). Performance Evaluation of Quality of VoIP in WiMAX and UMTS. *2011 12th International Conference on Parallel and Distributed Computing, Applications and Technologies*, 375-380. Ieee. doi:10.1109/PDCAT.2011.67.

[10] Durantini, A., Petracca, M., & Ananasso, F. (2008). Experimental Evaluation of IEEE 802.16 WiMAX Performances at 2.5 GHz Band. *2008 International Wireless Communications and Mobile Computing Conference*, 338-343. Ieee. doi:10.1109/IWCMC.2008.59

[11] Bernardo, V., Sousa, B., & Curado, M. (2009). VoIP over WiMAX: Quality of experience evaluation. *2009 IEEE Symposium on Computers and Communications*, 42-47. Ieee. doi:10.1109/ISCC.2009.5202392

[12] Etemad, K. (2008). Overview of mobile WiMAX technology and evolution. *IEEE Communications Magazine*, *46*(10), 31-40. doi:10.1109/MCOM.2008.4644117

[13] Alessio Botta, Alberto Dainotti, Antonio Pescapè, "Multi-protocol and multi-platform traffic generation and measurement", INFOCOM 2007 DEMO Session, May 2007, Anchorage (Alaska, USA)

[14] IEEE. (2008). RTP Payload Format for ITU-T Recommendation G.711.1 (pp. 1-15).

[15] ITU, T. S. S. of. (2003). Recommendation Y. 1541-Network Performance Objectives for IP-Based Services. *Network*.

[16] WiMAX Forum (2008). WiMAX system evaluation methodology v2.1. WiMAX Forum, Jan (pp. 1-209).

[17] IPerf. Last Visit: 10/02/2012. Available: http://dast.nlanr.net/Projects/Iperf

[18] ITU-T, “P.862 - Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs,” 2001.

[19] ITU-T, “ITU-R BS . 1387-1 - Method for objective measurements of perceived audio quality (PEAQ)” 2001.

[20] ITU, T. S. S. of. (2009). ITU-T Rec. G.107 - The E-model: a computational model for use in transmission planning. Networks

[21] Cole, R. G. and Rosenbluth, J. H. 2001. Voice over IP performance monitoring. *SIGCOMM Comput. Commun. Rev.* 31, 2 (April 2001), 9-24.DOI=10.1145/505666.505669 .

[22] Zhang, H.; Gu, Z.; Tian, Z.; , "QoS evaluation based on extend E-Model in VoIP," *Advanced Communication Technology (ICACT), 2011 13th International Conference on* , vol., no., pp.852-854, 13-16 Feb. 2011.

[23] Bernardo, V., Pentikousis, K., Jarno, P., Piri, E., & Curado, M. (2009). Multi-client video streaming over wirelessMAN-OFDMA. *Proceedings of the 4th ACM workshop on Performance monitoring and measurement of heterogeneous wireless and wired networks* (pp. 46–53). ACM.

Appendix B - EDP - WiMAX Pilot - Final Report

This Appendix shows the final report delivered EDP, in the context of EDP-WiMAX Project.

The first version of this document was written by the author. The further versions were deeply revised and complemented by Professor Paulo Simões in a collaboration work with the author.

All the test results presented were conducted by the author. Also, all the scripts used were created by the author, with the exception of signal quality measurement scripts (created in collaboration with Vitor Fonseca) and the mobile throughput script tests (created by Vitor Fonseca).

The testbed deployment, also described in the document, was conducted by Vitor Bernardo and the author.

This report is confidential, but it is included in the materials to be provided to the juri.