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Analysis and Evaluation of a *Fixed* WiMAX System for Videoconferencing

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DESCRIPTION

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Abstract

This paper describes the structure and performance of a Fixed WiMAX System, as part of a network of networks that provides a wireless broadband access to the University of Applied Sciences of Mikkeli, Finland.

The aim of the study is to evaluate the performance of the whole system when it is working with videoconference traffic. There are two major blocks that describe the performance attending different levels of standardization. As a first approach, there is a testing work on the Network Layer that permits us to have preliminary conclusions and helps to predict the performance we may find at the Application level. In second term, a study on video conferencing is developed, as well as a second testing work focused on Videoconferencing Applications.

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Most of all, I thank my beloved partner, Bea, for the immeasurable love, support and generosity she has always demonstrated to me and, once again, during these long seven months, letting me go for my dream.

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* These images were taken from public platforms.

LIST OF ACRONYMS

3GPP	Third Generation Partnership Project
ADSL	Asymmetric Digital Subscriber Line
ADPCM	Adaptive Differential Pulse Code Modulation
AES	Advanced Encryption Standard
AMC	Adaptive Modulation and Coding
AMR	Adaptive Multi Rate
ARQ	Automatic Repeat reQuest
BE	Best Effort
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
BS	Base Station
CC	Convolutional Code
CDMA	Code Division Multiple Access
CP	Cyclic Prefix
CPE	Customer Premises Equipment
DL	Downlink
DVB	Digital Video Broadcast
EAP	Extensible Authentication Protocol
ErtVR	Extended Real-Time Variable Rate
ETSI	European Telecommunications Standards Institute
FDD	Frequency Division Duplex
FDM	Frequency Division Multiplexion
FEC	Forward Error Correction
FER	Frame Erasure Rate
FFT	Fast Fourier Transform
FPS	Frames Per Second
FTP	File Transfer Protocol
GSM	Groupe Special Mobile
HARQ	Hybrid Automatic Repeat reQuest
HSDPA	High-Speed Downlink Packet Access
ICMP	Internet Control Message Protocol
ICT	Information and Communication Technologies
IDFT	Inverse Discrete Fourier Transform
IEEE	Institute of Electrical and Electronic Engineers
IEFT	Internet Engineering Task Force
IFFT	Inverse Fast Fourier Transform
IPS	Inter-domain Performance and Simulation
ISDN	Integrated Services Digital Network
ISI	Inter-Symbol Interference
ISP	Internet Service Provider
ITU	International Telecommunication Union

ITU – R	ITU – Radiocommunication Sector
ITU – T	ITU – Telecommunication Standardization Sector
LOS	Line of Sight
MAC	Media Access Control
MAN	Metropolitan Area Network
MIMO	Multiple Input Multiple Output
MOS	Mean Opinion Score
MPLS	Multi-Protocol Label Switching
NAT	Network Address Translation
NLOS	Non Line-of-Sight
nrtPS	Non Real Time Polling Service
OF	Optical Fiber
OFDM	Orthogonal Frequency-Division Multiplexing
OFDMA	Orthogonal Frequency-Division Multiple Access
OSI	Open System Interconnection
PER	Packet Error Rate
PCM	Pulse Code Modulation
PHY	Physical layer
PLR	Packet Loss Rate
PMP	Point-toMultipoint
QAM	Quadrature Amplitude Modulation
QCIF	Quarter Common Intermediate Format
QPSK	Quadrature Phase Shift Keying
QoS	Quality of Service
QVGA	Quarter Video Graphics Array
rtPS	Real Time Polling Service
RS	Reed-Solomon code
RTP	Real-time Transport Protocol
RTCP	Real-time Transport Control Protocol
SNR	Signal to Noise Ratio
SS	Subscriber Station
TDD	Time Division Duplex
UDP	User Datagram Protocol
UE	User Equipment
UGS	Unsolicited Grant Service
UL	Uplink
UMTS	Universal Mobile Telephone System
VGA	Video Graphics Array
VoIP	Voice over Internet Protocol
VPN	Virtual Private Network
WiFi	Wireless Fidelity
WiMAX	Worldwide Interoperability for Microwave Access

FOREWORD

If we shall admit that new telecommunications services and applications are currently the strong drivers of progress, it would be fair to point out the major role that Internet has played for the last two decades, with the introduction of the World Wide Web. The Web, as we know it nowadays, has introduced a flow of communication at a global scale ever seen before, in the history of human being.

Since its appearance, the infrastructure of Internet was expanded worldwide to create the modern global network that we all know now; and it changed radically our lives affecting the way we communicate each others, the economy, or even the leisure. But it also contributed to the social division that has been growing between developed and emergent countries, creating what we know today as the Digital Divide.

In 1996, the International Telecommunications Union initiated a United Nations project for the "right to communicate", aimed at providing access to basic Information and Communication Technologies (ICT) for everyone, with motivation to reduce information poverty for developing countries. This goal is now at the heart of plans of the World Summit on the Information Society (WSIS). Thus, during the first WSIS held in Geneva in December 2003, the Digital Divide was defined as the unequal access to ICT. Although this unequal access usually applies to differences between countries comparing developed and developing regions, it can also be applied within countries if we consider the existing divide between rural and urban areas, or between poor and wealthy citizens, for instance.

In this sense, Finland is a pioneer. Currently, there is a huge network of networks covering most parts of the country, furthermore, the Government has recently announced to make 1Mb/s connection a legal right by July 2010. This initiative was announced on 14th October by the Ministry of Transport and Communications as an intermediary step to achieve a 100Mb/s broadband access, by 2015.

As a pioneer trying to overcome the Digital Divide, Finland has yielded significant deployments, as those developed by means of the *eSavo Project*, for instance, demonstrating the success of public – private partnerships for broadband deployments.

WiMAX is part of these deployments, yielding rapid development of the corresponding technologies in the broadband wireless access network domain. Broadband wireless sits at the confluence of two of the most remarkable growth stories of the telecommunications industry in recent years. Both wireless and broadband have on their own enjoyed a fast mass-market adoption. We could say that broadband wireless is about bringing the broadband experience to a wireless context, which offers users certain unique benefits and convenience.

In particular, WiMAX has been engineered to deliver the latest type of fixed and mobile services such as VoIP, Information Technology and Video at very low cost, which makes it a very appropriate technology for areas where the investment may not be profitable for other telecommunication technologies, such as the extended wired solution, ADSL.

But in its simplest form, WiMAX promises to deliver the Internet around the world, connecting the “last mile” of communications services for both developed and emerging nations, thus bridging some more the Digital Divide.

1. INTRODUCTION

This paper describes the potential role of WiMAX as a support for IP-based videoconferencing applications. In this sense, it was necessary to develop a work plan before facing the testing work and analysis on the system performance. First, a research on WiMAX Technology allowed us to have a general overview about the pros and cons, consistent with the requirements of a real-time communication, and its interoperability with other IP networks. Second, a deep study on IP-based videoconferencing applications was essential in order to figure out all the aspects related to the coding, compressing and the quality boundaries accepted by different institutions as a key balance between the system performance and the end-user's perception.

This case study is divided in two differentiated parts with a total of six chapters. The first part includes Chapters 2 and 3, and it represents the theoretical research on WiMAX Technology and the particularities of Voice and Video when delivered over data networks. Chapter 2 provides the background information necessary to have an overview about WiMAX, and the features of commercial devices that have already been released (which are also partially presented in Chapter 4, as a complement to the introduction about the Radio Access Network). It also presents a brief comparison between the recent *Mobile* WiMAX and the 3G Technology, as the strongest current alternatives for mobile broadband access.

On the other hand, in Chapter 3 we can find an introduction to "Voice over IP" applications, where the prior quality specifications about real-time communications are set. Second, an overview about videoconference over IP is presented, as well as an introduction to one of the latest real-time multimedia communications protocol: H.323 recommendation.

The second part includes Chapters 4 to 7, and it presents the practical work developed on the case network, as well as the analysis and evaluation of the results obtained. Chapter 4 is entirely a description on the case network. It includes an extended explanation about WiMAX certification profiles, as an approach to understand some key features of our radio link. In addition, we can find an exposition about the core network that was used to carry out the experiments, as well as the different mechanisms and theories used to figure out the network topology and its functional characteristics.

Chapter 5 is about the performance of the case network. It is presented in two parts. First, there is an introduction about the theoretical model used to evaluate the performance, according to the features of the core network. It is also explained the framework where we are going to develop the testing work, as well as a brief introduction into voice and video traffic, in order to attempt the tests on the case network with modeled traffic, based on the features of voice and video. All these aspects shape the first half part of the working plan. Second, a performance evaluation on layer 3 is developed according to the OSI (Open System Interconnection) network reference model.

Finally, in Chapter 6 we find the second half part of the working plan, which evaluates the performance of the WiMAX Network when applied for videoconferencing. Again, the chapter is organized in two parts, which consist of a theoretical introduction onto the monitoring tool used for videoconferencing and, second, the analysis and evaluation of the system performance when it works in a videoconference session.

In Chapter 7 there is a compilation of all the results obtained consistent with the theoretical information from the previous research work. A final conclusion about the potential of WiMAX Technology is presented, as well as a summary of what this working experience meant.

It is necessary to point out that we do not have the intention to extend the results and conclusions obtained from this case study to other systems or networks, as they are completely dependent on the context and particular features of a concrete case network. However, it is intended to be a useful document for engineers interested in the field, as it represents a study that has been developed over a *real* WiMAX installation, and therefore we believe it may help to have a general overview about the technology and its performance in contrast with other technologies already settled, like Optical Fiber Links or the ADSL.

2. WiMAX TECHNOLOGY

After years of development in the field of wireless broadband services, a final solution is emerging based on the idea of interoperability and standardization. A great group of companies are today part of what we know as the WiMAX Forum, which has started to certify wireless broadband products with the main goal of reaching the interoperability between them, working under a single standard: IEEE 802.16. This standard has been also adopted by the group ETSI HIPERMAN, which works in the development of an access wireless network for the metropolitan area (WMAN). Hiperman is a standard created by the European Telecommunications Standards Institute (ETSI) directed mainly to provide broadband wireless DSL, covering a wide geographical area. This standard has been developed in cooperation with the 802.16, and has the same PHY (Physical) and MAC (Medium Access Control) layers, so that both standards are completely interoperable between each other.

2.1 Technology Overview

WiMAX (Worldwide Interoperability for Microwave Access) is a standards-based wireless broadband technology, also known by the IEEE standard 802.16, offering high-speed wireless access over long distances.

The group IEEE 802.16 was founded in 1998. It was firstly focused for the development of a point-multipoint system with a line-of-sight link, in order to operate in the band of 10GHz - 66GHz. The final standard, presented by the end of 2001 specified a single carrier for the physical layer (PHY) and Time-Division Multiplexation (TDM) for the Medium Access Control layer (MAC). After this, a revision of the standard was developed and 802.16a was presented. This version included applications that did not need a line-of-sight link (NLOS), and operated in the band of 2 – 11 GHz. It specified a PHY layer based on Orthogonal Frequencies Division Multiplexation (OFDM) and added to the MAC layer the possibility of subchannelling through the Orthogonal Frequencies Division Multiple Access (OFDMA).

By 2004, further evolutions concluded in the standard 802.16d, which replaced the previous ones, and set a base for the first WiMAX solution: *Fixed* WiMAX. The standard was specified to allow nomadicity, where users could access the service from various locations covered by the network. However, in the absence of portable devices users could only access the service from their home location, where the CPE was installed.

In 2005 the concept of *mobility* acquired a greater dimension, and by December of that year the standard 802.16e was finally approved. This standard set a base for the future *Mobile* WiMAX, which added the feature of mobility as an advantage for the end user (PC data cards, mobile handsets and laptops with embedded WiMAX chips are being planned by vendors on this standard).

Key features of WiMAX:

a) **Physical layer based on OFDM:** PHY layer is based on the multiplexion of orthogonal frequencies, which makes it offer a nice resistance to interference caused by *multipath* reflection.

b) **High maximum transfer rates:** Up to 70Mbps using a bandwidth of 20MHz, in theory. In real systems, using 10MHz bandwidth, TDD (time-division duplexing) and a relation 3 to 1 between downlink and uplink, it is possible to offer about 25Mbps(DL) / 6,7Mbps(UL). These maximum rates are reached when using a 64QAM modulation.

c) **Variable bandwidth:** the architecture of the physical layer allows a “scalable” bandwidth. This scalability is a feature provided by the modulation OFDMA, where the size of the FFT (Fast Fourier Transform) can vary depending on the bandwidth available.

d) **Adaptive modulation and coding (AMC):** different types of modulation and coding are allowed for the purpose of *error correction* (FEC), and they can be modified for each end user, depending on channel conditions. It is an effective mechanism to maximize the transfer rate in a channel that varies through time. The adaptive algorithm starts usually with the fastest scheme of modulation and coding, and then it may vary according to the SNR present in the channel and the coefficient of interference of the receptor.

e) **Retransmissions in the data link layer:** Automatic Requests (ARQ) are supported by the data link layer, in order to enhance reliability. It can optionally support a hybrid architecture that relates FEC with ARQ.

f) **TDD and FDD applications:** both 802.16d and 802.16e admit duplexing in time and frequency, which reduces costs in implementation. This allows also flexibility to choose the relation between downlink and uplink [(1)].

g) **OFDMA:** Orthogonal Frequencies Multiple Access is used as an access technique for a group of end users, being possible to administrate subcarriers in order to enhance the capacity and spectral efficiency of the system.

h) **QoS:** the MAC layer is connection-oriented, and it was designed in order to support a variety of applications, including *voice* and *multimedia* services. The system is prepared for real-time traffic, variable bit rates and “best effort” traffic.

i) **Security:** encryption of data is supported using the standard AES (Advanced Encryption Standard). In addition, WiMAX has a very flexible authentication architecture based on the protocol EAP (Extensible Authentication Protocol), which allows a great variety of user credentials, including passwords, digital certifications and intelligent cards.

j) **Mobility:** the mobile alternative of WiMAX has mechanisms that ensure the “handover” between cells without interruptions, although it was designed for applications that tolerate some delay, such as VoIP. This alternative has also some mechanisms for saving power, with the aim of enlarging the life of batteries of the portable devices (handhelds, laptops, etc).

k) **Architecture based on IP:** WiMAX Forum defined a network architecture based on an IP platform. Every point-to-point service is held over an architecture that depends on IP transport protocols, QoS, management, security and mobility. This architecture enables convergence with other networks.

[(1)] However, TDD mode appears to be more resource-use efficient, being the preferred alternative for the WiMAX Forum as for *Mobile* WiMAX System Profiles. The main reason is that TDD mode allows adjustment of the downlink/uplink ratio to efficiently support asymmetric DL/UL traffic. Channel reciprocity guarantees better support of link adaptation, MIMO and other close-loop advanced antenna technologies. Reference: Loutfi Nuaymi, ”*WiMAX: technology for broadband wireless access*”, page 42. Wiley & Sons (2007).

2.2 Principles of OFDM

There are two major differences between wireless and wired systems which make the system design process significantly different at each case. The fading characteristics of wireless channel and the interference between users sharing the same wireless medium are perhaps the most differentiating factors. Frequency-selective fading is more prominent in wideband channels (where the channel's bandwidth is usually much greater than the coherence bandwidth) and it causes dispersion in time, which leads adjacent symbols to interfere with each other unless $T >> \tau_{max}$ (T= symbol time; τ_{max}= maximum delay spread). However, since the data rate *R* is proportional to 1/T, high-data-rate systems almost invariably have a substantial multipath *delay spread* [(2)], with $T << \tau_{max}$, and experience severe inter-symbol interference (ISI) as a result.

OFDM (Orthogonal Frequency-Division Multiplexing) is nowadays a popular technique chosen for solving ISI [(3)]. This technique is based on the "multicarrier concept", which we will explain briefly. The philosophy of multicarrier modulation sets a model where the diversity of the time-dispersive ISI channel is used, instead of being compensated. A large number of sub-carriers (L) are used in parallel, so that the symbol time for each goes from T→LT; this means that instead of sending a single signal with data rate R and bandwidth B, L signals are sent at the same time, each having a bandwidth B/L and data rate R/L. The result is that if $B/L << B_c$ (channel bandwidth), each signal will experience approximately flat fading, and the time dispersion for each signal will be negligible. Then, if L is large enough, the condition $B/L << B_c$ is possible to achieve. An appropriate choice of system parameters such as the number of subcarriers and the distance between them may reduce significantly, even totally, the inter-symbol interference (ISI). This idea is the basic principle of OFDM technique.

[(2)] The delay τ is a very important property of a wireless channel, specifing the duration of the channel impulse response $h(\tau, t)$. Intuitively, the *delay spread* is the amount of time that elapses between the first arriving path and the last arriving (non-negligible) path. Reference: Jeffrey G. Andrews, Arunabha Ghosh, Rias Muhamed, "*Fundamentals of WiMAX: Understanding Broadband Wireless Networking*", page 86. Prentice Hall (2007).

[(3)] Equalizers are the most logical alternative for ISI suppression to OFDM, since they don't require additional antennas or bandwidth and have moderate complexity. Equalizers are implemented at the receiver and attempt to reverse the distortion introduced by the channel. Reference: Jeffrey G. Andrews, Arunabha Ghosh, Rias Muhamed, "*Fundamentals of WiMAX: Understanding Broadband Wireless Networking*", page 109 . Prentice Hall (2007).

The number of substreams is chosen to ensure that each subchannel has a bandwidth less than the coherence bandwidth of the channel (Fixed WiMAX uses 256 subcarriers), so the subchannels experience relatively flat fading. Thus, the ISI on each subchannel is small. Moreover, in the digital implementation of OFDM, the ISI can be completely eliminated through the use of a Cyclic Prefix (CP).

The key to making OFDM realizable in practice is the use of the FFT algorithm, which has low complexity. In order for the IFFT/FFT to create an ISI-free channel, the channel must appear to provide a circular convolution(4). Adding cyclic prefix to the transmitted signal, creates a signal that appears to be a circular function(5). Figure 2.1 shows the mechanism to create the cyclic prefix.

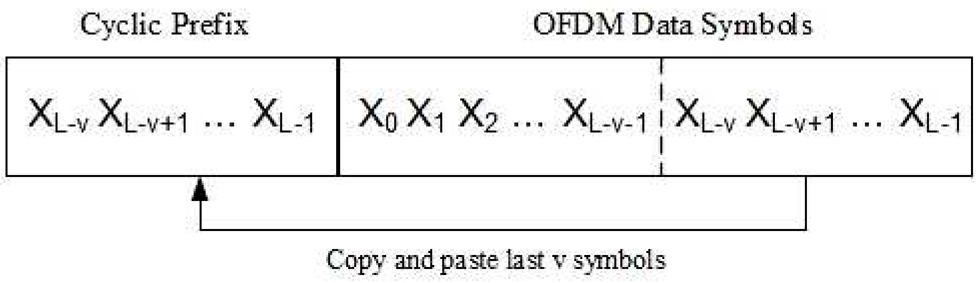

Figure 2.1 The OFDM cyclic prefix

If the maximum channel delay spread has a duration of v + 1 samples, adding a guard band of at least v samples between OFDM symbols makes each OFDM symbol independent of those coming before and after it, and so only a single OFDM symbol can be considered. Representing such an OFDM symbol in the time domain as a length L vector gives: $x = [x_1, x_2 \ldots x_L]$. After applying a cyclic prefix of length v, the transmitted signal is: $x_{cp} = [x_{L-v} x_{L-v+1} \ldots x_{L-1}\ x_0 x_1 \ldots x_{L-1}]$, which is the same structure shown in figure 2.1.

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(4) When an input data stream is sent through a linear time-invariant Finite Impulse Response (FIR) channel, the output is the linear convolution of the input and the channel: $y[n] = x[n]*h[n]$. If we create a *periodic version* of x[n] like: $x_L[n]=x[n \bmod L]$ with period L, the result now is a convolution of the extended input ($x_L[n]$) and the channel, and it is called *circular convolution* of x and h. Reference: Jeffrey G. Andrews, Arunabha Ghosh, Rias Muhamed, "*Fundamentals of WiMAX: Understanding Broadband Wireless Networking*", pages 117-119 . Prentice Hall (2007).

(5) The *circular function* is the extended periodic version $x_L[n]$. Reference: Jeffrey G. Andrews, Arunabha Ghosh, Rias Muhamed, "*Fundamentals of WiMAX: Understanding Broadband Wireless Networking*", pages 117-119. Prentice Hall (2007).

The output of the channel is by definition $y_{cp} = h * x_{cp}$, where h is a length vector describing the impulse response of the channel during the OFDM symbol. The output y_{cp} has $(L + v) + (v + 1) - 1 = L + 2v$ samples. The first samples contain interference from the preceding OFDM symbol and so are discarded. The last samples disperse into the subsequent OFDM symbol, so also are discarded. This leaves exactly L samples for the desired output y, which is precisely what is required to recover the L data symbols embedded in x.

In the PHY layer of 802.16-2004 standard, it is specified the WirelessMAN-OFDM, where it is established 256 subcarriers to be used. Among them, 192 are used for data, 8 for Pilot Subcarriers and 56 for Guard Bands and the DC subcarrier. We can see a description in Figure 2.2.

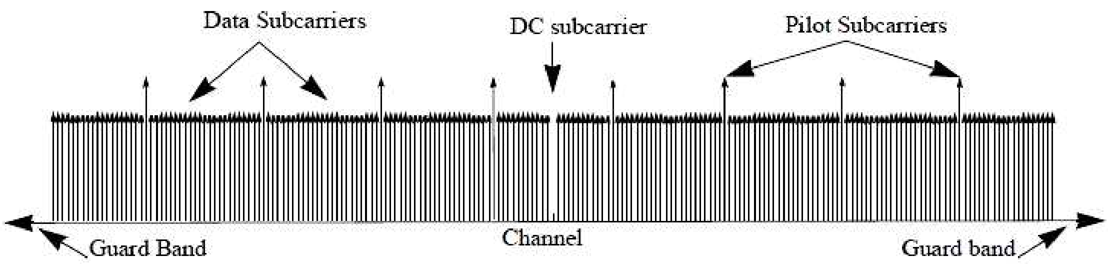

Figure 2.2 OFDM frequency description [6]

This is the distribution in the frequency domain. Inverse-Fourier-transforming creates the OFDM waveform. This time duration is referred to as the "useful" symbol time $T_{b.}$ The CP has a time duration called T_g , and both together form the OFDM symbol, with a duration time of $T_s = T_g + T_b$, as we can see in figure 2.3.

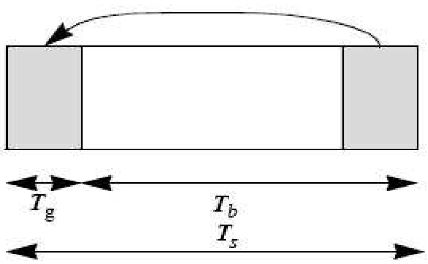

Figure 2.3 OFDM symbol time structure [7]

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[6] [7] Reference : IEEE 802.16TM - 2004. "*IEEE Standard for Local and metropolitan area networks. Part 16: Air Interface for Fixed Broadband Wireless Access Systems*", pages 427 – 428.

Now we can introduce briefly the structure of a passband OFDM modulation engine, looking at an example in Figure 2.4. The inputs to this figure are L independent QAM symbols (the vector **x**), and these L symbols are treated as separate subcarriers. These symbols can be created from a bit stream by a symbol mapper and serial-to-parallel converter (S/P). The L-point IFFT then creates a time-domain L-vector **X** that is cyclic extended to have length L(1+G), where G is the fractional overhead. This longer vector is then parallel-to-serial (P/S) converted into a wideband digital signal that can be amplitude modulated with a single radio at a carrier frequency of $f_c = \omega_c / 2\pi$.

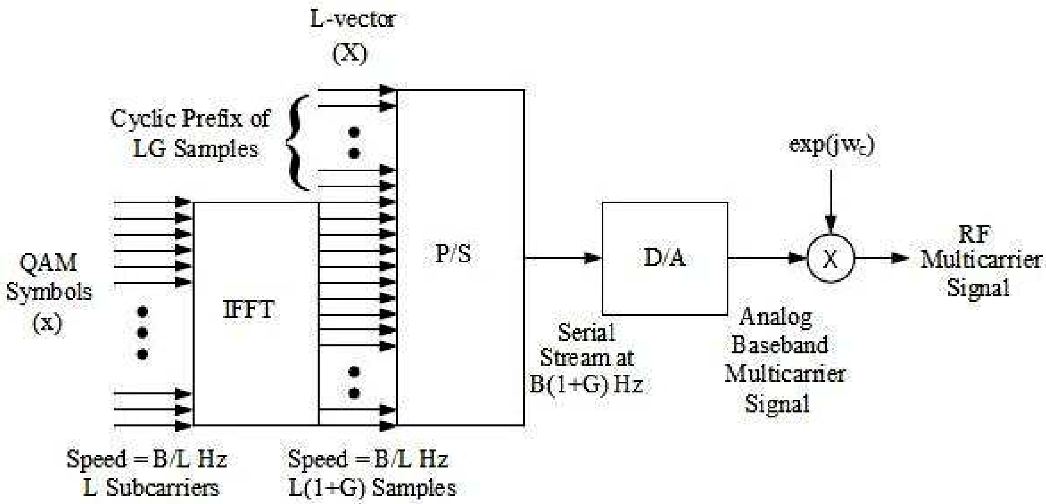

Figure 2.4 OFDM passband transmitter [(8)]

The key OFDM parameters are summarized in Table 2.1, along with some potential numerical values for them. As an example, if 16QAM modulation were used ($M = 16$), the raw (neglecting coding) data rate of this WiMAX system would be:

$$R = \frac{B}{L}\frac{L_d \log_2(M)}{1+G} = \frac{3.5MHz}{256}\frac{192\log_2(16)}{1+0.125} = 9.33Mbps$$

The numeric values of B, L, L_d and G have been chosen for this example according to Fixed WiMAX – specified parameters.

[(8)] Reference: Jeffrey G. Andrews, Arunabha Ghosh, Rias Muhamed, "*Fundamentals of WiMAX: Understanding Broadband Wireless Networking*", page 124. Prentice Hall (2007).

Table 2.1 Summary of OFDM parameters [9]

Symbol	Description	Relation	Fixed WiMAX value
B*	Nominal bandwidth	$B = 1/T_s$	3.5MHz
L*	Number of subcarriers	Size of IFFFT/FFT	256
G*	Guard fraction	$G = T_g/T_b$	1/4,1/8,1/16,1/32
L_d*	Data subcarriers	L-pilot/null subcarriers	192
T_s	Sample time	$T_s = 1/B$	0.285µs
N_g	Guard symbols	$N_g = G \cdot L$	64, 32, 16, 8
T_g	Guard time	$T_g = T_s \cdot N_g$	18.2µs - 2.2µs
T	OFDM symbol time	$T = T_s (L+ N_g)$	91.2µs - 75.2µs
B_{sc}	Subcarrier bandwidth	$B_{sc} = B/L$	13.67kHz

* Denotes WiMAX-specified parameters; the other OFDM parameters can all be derived from these values.

We can observe that OFDM symbol-time range is more than 100 times higher than the sample time, thanks to the 256 carriers it uses. This way, each signal will experience approximately flat fading, and the time dispersion for each signal will be negligible. Figure 2.5 shows an OFDM symbol in time (a) and frequency (b). In the time domain, the IFFT effectively modulates each data symbol onto a unique carrier frequency.

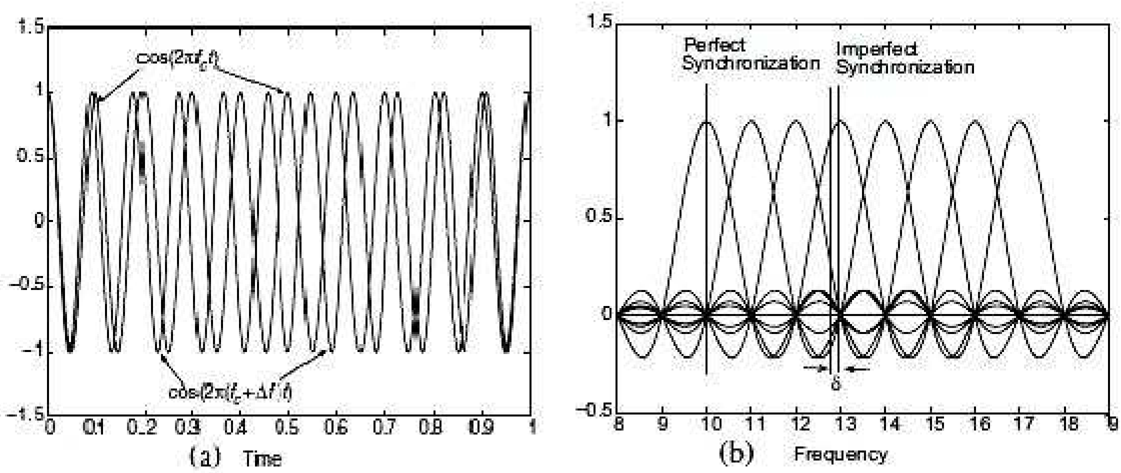

Figure 2.5 OFDM symbol in time (a) and frequency (b)

[9] Reference: Jeffrey G. Andrews, Arunabha Ghosh, Rias Muhamed, "*Fundamentals of WiMAX: Understanding Broadband Wireless Networking*", page 125. Prentice Hall (2007).

Although only two carriers are shown in graphic (a), we can observe that the transmitted signal is the superposition of all the individual carriers. Since the time window (sampling time) is normalized from 0 to 1, but 1 corresponds to 0.285µs, and a rectangular window is used, the frequency response of each subcarrier becomes a "sinc" function with zero crossings every 3.5MHz. This means that the carriers are disposed orthogonally, as we can see in picture (b), and demodulation is possible.

The orthogonal disposition of the subcarriers permits their spectrums to be overlapped without interference prejudice; hence the spectrum efficiency increases significantly as there is no need to use separating bands between subcarriers. This is also a very important advantage of OFDM-based systems.

2.3 Adaptive Modulation and Coding

WiMAX supports several types of modulations and codifications, which can be modified dynamically for each link, depending on the channel conditions. Using a "channel quality" indicator, it is possible to provide the base station with information about the state and quality of the downlink. For the uplink, the station can estimate the quality attending the level of the signal received. The *scheduler* [10] of the base station can assign a scheme of modulation and codification based on the quality of the channel, and maximize the throughput depending on the S/N relation. The adaptive configuration may increase the global capacity of the system, as well as the robustness, in real-time, for each channel. The adaptation is automatic and dynamic, attending the distance and the conditions of the channel, being used the QAM (Quadrature Amplitude Modulation) for short distances and greater bit rates, and PSK for longer and less stable links, which require a more robust modulation.

The best throughput is obtained using 64QAM (6 bits per symbol), and the most robust modulation is BPSK. The protocols used by WiMAX are highly adaptive, and they

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[10] The *scheduler* is an element situated in the base station that controls the assignation of bandwidth for both downlink and uplink, at each channel. The implementation of an effective scheduler is critical to the overall capacity and performance of a WiMAX system. Reference: Jeffrey G. Andrews, Arunabha Ghosh, Rias Muhamed, "*Fundamentals of WiMAX: Understanding Broadband Wireless Networking*", pages 50 - 51. Prentice Hall (2007).

allow the base station to adjust the operation parameters and power levels with the aim of giving an optimal level of signal to the CPE (Customer Premises Equipment).

Table 2.2 shows the different types of modulation and codification techniques supported by WiMAX. For *Fixed* WiMAX, the modulations BPSK, QPSK, 16QAM and 64 QAM are mandatory. As for the codification used by the standard, it is necessary that the FEC (Forward Error Correction) uses Convolutional Codes (11). In the downlink, these codes can be combined with external codes Reed-Solomon (12).

Table 2.2 Coding and modulations supported by WiMAX (13)

	Downlink	**Uplink**
Modulation	BPSK, QPSK, 16QAM, 64QAM; BPSK optional for OFDMA-PHY	BPSK, QPSK, 16QAM; 64QAM optional
Coding	Mandatory: convolutional codes at rate 1/2, 2/3, 3/4, 5/6 Optional: convolutional turbo codes at rate 1/2, 2/3, 3/4, 5/6; repetition codes at rate 1/2, 1/3, 1/6, LDPC, RS-Codes for OFDM-PHY	Mandatory: convolutional codes at rate 1/2, 2/3, 3/4, 5/6 Optional: convolutional turbo codes at rate 1/2, 2/3, 3/4, 5/6; repetition codes at rate 1/2, 1/3, 1/6, LDPC

Using adaptive modulation permits to provide a gradation of QoS depending on the distance between the BS and the SS. The longer the distance, the lower the guarantee of QoS. In good channel conditions, high-order modulations can be used to increase the throughput and spectral efficiency. Besides, when the radio channel conditions become worse (weather, signal interference, etc), lower-order modulations should be used to maintain certain quality in the communications.

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(11) *Convolutional code* is one of the main codification techniques used by the error correction mechanism, FEC. It usually operates on continuous data. The bits are codified as they arrive at the coder, and each codified bit has a relation with the previous ones. The output stream is the result of a generator function convolved with the input stream. Reference: Vijay Madisetti, Douglas Bennet Williams; "*The digital signal processing handbook*" 56/6.

(12) *Reed-Solomon* is a non-binary cyclic code, a subclass of the block codes with the purpose of error detection. It appends a number of redundant bytes to a block of data in order to achieve error correction.. This code belongs to the branch of FEC (Forward Error Correction) codes. Reference: Vijay Madisetti, Douglas Bennet Williams; "*The digital signal processing handbook*" 56/6.

(13) Reference: Jeffrey G. Andrews, Arunabha Ghosh, Rias Muhamed, "*Fundamentals of WiMAX: Understanding Broadband Wireless Networking*", page 47. Prentice Hall (2007).

2.4 Quality of Service

WiMAX provides Quality of Service at both PHY and MAC layers. It is important to notice the existence of real-time data prioritization, which lets the videoconference traffic experience a better performance. QoS is provided using the following techniques:

a) PHY Layer :

- **Frequency-division duplexing (FDD)** - The application of frequency-division to separate outward and incoming signals. The uplink and downlink sub-bands are separated by a "frequency offset". Highest efficiency in case of symmetric traffic.
- **Time-division duplexing (TDD)** - The application of time-division to separate outward and incoming signals, using only one carrier. It has a strong advantage in the case where the asymmetry of the uplink and downlink data speed is variable.
- **Forward Error Correction (FEC)** - The technique to allow the receiver to correct some errors without having to request a retransmission of data.
- **Orthogonal frequency-division multiplexing (OFDM)** - The frequencies of FDM are arranged to be orthogonal with each other. This eliminates most of the interference between channels.
- **Orthogonal Frequency Division Multiple Access (OFDMA)** - It works by assigning a subset of subcarriers to individual users.

b) MAC Layer :

- **Unsolicited Grant Service (UGS)** - Supports real time data streams, having fixed size packets issued at regular intervals (e.g. T1/E1 and VoIP [(14)]).
- **Real Time Polling Service (rtPS)** - Supports real time data streams, having variable size packets issued at regular intervals (e.g. VoIP, MPEG Video).
- **Non Real Time Polling Service (nrtPS)** – Supports delay tolerant data with variable packet sizes. A minimum data rate is specified (e.g. ftp).
- **Best Effort (BE)** - Supports data streams where no minimum service is required and packets are handled on a space-available basis (e.g. e-mail).

[(14)] Actually, there are two scheduling types for real-time services in IEEE 802.16, such as UGS and rtPS. However, neither of them is efficient in supporting VoIP as they do not consider ON/OFF property of voice traffic. The IEEE

Figure 2.6 shows how does the traffic flow between the subscriber and the base station.

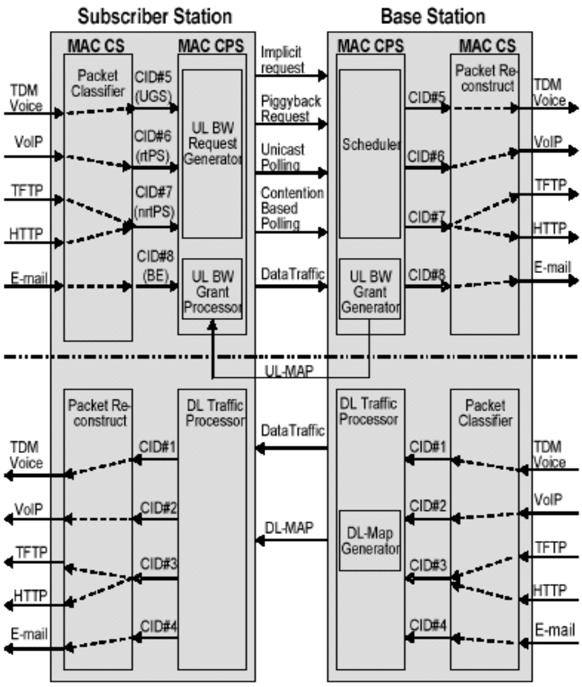

Figure 2.6 QoS mechanism

2.5 WiMAX vs. 3G

WiMAX is not the only solution in the distribution of wireless broadband services, actually, there are many proprietary solutions for mobile and fixed applications, which are already available in the market. For instance, the third generation of mobile telephony (3G) is getting more market share as time goes, as it has the possibility of transferring voice (phone call) and data from the terminal with good quality and high speed. The standards in 3G use CDMA[15] to share the spectrum between users, and although

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802.16 Broadband Wireless Access Working Group proposes an efficient uplink scheduling method considering this property, in the Contributed Document: IEEE C802.16e-04/351r1 "*Extended rtPS, for VoIP services*".

[15] *CDMA* is the Code Division Multiple Access, which defines a wireless interface based on the *spread-spectrum* technology, which allows several users to share the spectrum without interfering between each other. Spread-spectrum techniques are methods by which electromagnetic energy generated in a particular bandwidth is deliberately spread in the frequency domain, resulting in a signal with a wider bandwidth. These techniques are used for a variety of reasons, including the establishment of secure communications, increasing resistance to natural interference and jamming, to prevent detection, and to limit the power flux density on satellite downlinks.

initially it was specified a speed of 384Kbps, the continuous evolution of the technology allows to offer now a speed of downloading over 3Mbps.

The capacity in WiMAX depends on the bandwidth of the channel that is being used, unlike 3G systems, for instance, which have a fixed channel bandwidth. WiMAX defines a selectable bandwidth between 1,25MHz and 20Mhz. This allows very flexible developments, for instance, if we were using a 10MHz TDD channel, with a 3:1 downlink-to-uplink split and a 2x2 MIMO[16], WiMAX could offer a peak downlink throughput of about 46Mbps, and 7Mbps uplink.

WiMAX, as Wi-Fi, uses OFDM modulation, which allows it to support high bursts in the transfer rates. However, as 3G uses CDMA, which works with techniques of *spread spectrum*, and it is significantly more difficult to achieve these high bursts rates. But it is necessary to remark that it is more important to talk about the average of the data transfer rates.

From the point of view of the capacity, the most relevant issue about the performance of the system is the spectral efficiency. *Mobile* WiMAX takes advantage at this point, as it works with multiple antennas (MIMO), which is a feature mainly focused on spectral efficiency. The 802.16 defined MIMO configuration is negotiated dynamically between each individual base station and mobile station. The 802.16 specification supports the ability to support a mix of mobile stations with different MIMO capabilities. This helps to maximize the sector throughput by leveraging the different capabilities of a diverse set of vendor mobile stations. The physical layer OFDM used by WiMAX is more "friendly" in order to support MIMO, comparing to CDMA systems, since the complexity required for the same gain is smaller.

If we consider certain IP applications, such as voice, video or multimedia, we can observe differences in terms of prioritization of the traffic and control of quality.

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[16] *MIMO* is the acronym for Multiple-Input-Multiple-Output. It refers to the use of multiple antennas at the transmitter and receiver in a wireless system. This technology takes advantage from the multipath effect that takes place in a radio communication to enhance performance, by means of multiple antennas embedded in a single device. Without further qualification, MIMO is often assumed to mean specifically the spatial multiplexing approach, since spatial multiplexing transmits multiple independent data streams and hence has multiple inputs and outputs. Reference: Jeffrey G. Andrews, Arunabha Ghosh, Rias Muhamed, "*Fundamentals of WiMAX: Understanding Broadband Wireless Networking*", page 149. Prentice Hall (2007).

The MAC layer in WiMAX was developed in order to be able to support a significant variety of traffic, as it was already mentioned in the headland 2.4. On the other hand, 3G solutions such as HSDPA or 1xEV-DO have been designed for a variety of levels of Quality of Service. In Table 2.3 we can observe a summary about technical features of WiMAX, 3G and WiFi.

Table 2.3 Technical features of WiMAX, 3G and WiFi (17)

Parameter	Fixed WiMAX	Mobile WiMAX	HSPA	1x*EV-DO	WiFi
Standards	IEEE 802.16 - 2004	IEEE 802.16e - 2005	3GPP Release 6	3GPP2	IEEE 802.11a/g/n
Peak downlink data rate	9.4Mbps in 3.5MHz with 3:1 DL/UL ratio TDD; 6.1Mbps with 1:1 DL/UL	46Mbps in 10MHz with 3:1 DL/UL ratio TDD; 32Mbps with 1:1	14.4Mbps using all 15 codecs; 7.2Mbps with 10 cods	3.1Mbps Rev. A; 4.9Mbps Rev. B	54Mbps shared using 802.11a/g; More than 100Mbps using 802.11n
Peak uplink data rate	3.3Mbps in 3.5MHz with 3:1 DL/UL ratio; 6.5Mbps with 1:1	7Mbps in 10MHz using 3:1 DL/UL ratio; 4Mbps using 1:1	1.4Mbps initially; 5.8Mbps later	1.8Mbps	
Bandwidth	3.5MHz and 7MHz in 3.5GHz band; 10MHz in 5.8 GHz band	3.5MHz, 5MHz, 7MHz, 8.75MHz and 10MHz initially	5MHz	1.25MHz	20MHz for 802.11a/g; 20/40MHz for 802.11n
Multiplex.	TDM	TDM / OFDMA	TDM / CDMA	TDM / CDMA	CSMA
Duplexing	TDD, FDD	TDD initially	FDD	FDD	TDD
Frequency	3.5GHz and 5.8GHz bands initially	2.3GHz,2.5GHz and 3.5GHz initially	800/900/ 1800/1900/ 2100MHz	800/900/ 1800/1900 MHz	2.4GHz and 5GHz
Coverage	3 – 5 miles	< 2 miles	1 – 3 miles	1 – 3 miles	<100 ft indoors < 1000 ft outdoors

* 1x = single carrier

(17) Reference: Jeffrey G. Andrews, Arunabha Ghosh, Rias Muhamed, "*Fundamentals of WiMAX: Understanding Broadband Wireless Networking*", page 18. Prentice Hall (2007).

Another advantage about WiMAX is the possibility of supporting more efficiently symmetric links, and it also allows flexible and dynamic adjustment as for the relation between the download and the upload (3G has typically a fixed and asymmetric upload-download relation).

But maybe the major advantage of WiMAX comes from the potentially low-cost implementation that it has, thanks to its *IP architecture*. This architecture makes the core of the network more simple (3G has a complex core, and separates voice from data), which results directly in a reduction of the costs of implementation. Being "*IP native*" allows also an easier integration of the different applications, and a better convergence with other networks.

As a main advantage for 3G, we have the fact that it was conceived for mobility at high-speed, while WiMAX incorporated mobility capabilities as a later feature (initially it was conceived as a fixed system).

We could locate WiMAX between Wi-Fi and 3G technologies, if we compare it in terms of transfer rates, coverage, QoS, mobility and costs.

3. VOICE & VIDEO OVER DATA NETWORKS

Ever since its appearance, VoIP (Voice over IP) became a reference for domestic users as it opened doors for telephony communications bringing a huge bunch of new possibilities. The main point about it, and the reason for the general acceptance it has nowadays in our modern society, is a matter of costs: conversations through the Internet are extremely cheap. If we consider a fix cost for the Internet Service, which would be paid by the user independently of the Telephony Service, we could say that a conversation held using VoIP would be "free", as no additional costs are paid, comparing to a traditional telephone call. Then, if we can add real-time video and have a videoconference between the end users, the situation becomes even more interesting. We can easily admit how revolutionary have been these applications, taking advantage of the IP networks.

3.1 Voice over IP

"*Voice over IP*" uses Internet Protocol (IP) for transmission of sequences of voice as "packets" over IP networks. The process involves digitalization of voice, the isolation of unwanted noise signals and then a compression of the signal using specific algorithms and codecs. After the compression, the voice is packetized to be sent over an IP network, where each packet needs a destination address and a sequence number for error checking. The signaling protocols are added at this stage to achieve these requirements along with the other call management requirements.

When a voice packet arrives at the destination, the sequence number enables the packets to be placed in order and then the decompression algorithms are applied to recover the data from the packets. Here the synchronization and delay management need to be taken care of to make sure that there is a proper spacing. A jitter buffer is used to store the packets arriving out of order [(18)], so that the information is re-organized for the end-user.

[(18)] Some implementations (like G.729) enable the receiving station to wait for a period of time (per its jitter buffer) when a packet is lost, running afterwards a *concealment strategy* which replays the last packet received so that the listener does not receive a gap with silence. Reference: J.D., J.P, "*Voice over IP Fundamentals*", pages 126,127; Cisco Press (2000).

3.1.1 Voice Quality

Presently, the telephone companies are losing part of their business, as a consequence of the increase experimented in the use of VoIP applications. But still the speculation exists on the kind of voice quality that VoIP service providers may give to customers. The techniques used to measure the voice quality of a VoIP call are the Mean Opinion Score (MOS) and Perceptual Speech Quality Measurement (PSQM).

MOS follows the measurement techniques specified in ITU-T P.800, where different people are made to listen some voice signals and to rate factors like distortion, delay, echo, noise, etc. on a scale of 1 to 5 [(19)], being 1 the worst mark. After that, the mean opinion score is calculated, and it is important to remark that a value of MOS 4 is considered as the toll quality. For example, the conventional codec for fixed line telephones, G.711 PCM, has a MOS of 4.1 at 64kbps. The modified codecs mainly used for VoIP are G.729 and G.723.1. G.729 is widely used for VoIP because of its low bandwidth requirements, and it has a MOS value of 3.92 operating at 8kbps.

PSQM is based on ITU-T P.861 standard [(20)]. This technique uses artificial speech in order to provide numeric values of approximate speech intelligibility, taking into account effects such as noise, coding errors, packet reordering, phase jitter, and excessive bit error rate. A value of zero means no impairment, while a value of 6.5 indicates that the signal is totally unusable. Although PSQM values do not have direct correlation with MOS values, a PSQM value of 0 corresponds roughly to a value of 5 in the MOS scale, and a value of 6.5, to a MOS of 1.

Factors that affect the voice quality:

a) **Delay** .- Voice transmission over a wireless link brings along with it a problem related to packet delay, also known as *latency*. The end-to-end delay budget can be divided into six[(21)] major delays, differentiating between *fixed delays*, which would include: coding, packetization/depacketization, serialization, and propagation; and *variable* delays, which would include: queuing delay and the “dejitter” buffer.

[(19)(20)(21)] Reference: Johnathan Davidson, James Peters, “*Voice over IP Fundamentals*”, pages 124, 125 and 137 respectively; Cisco Press (2000).

Studies have proved that packet delay of 100ms does not do any harm, but if the delay increases over 150ms end-to-end, then the voice signal is practically unusable. The service providers have to ensure that the total delay caused does not exceed 100ms, if they want to be able offer reliable VoIP services. Table 3.1 shows some speech CODECs, along with their MOS and delay time.

Table 3.1 Speech codecs [22]

CODEC	Bit rate (Kbps)	MOS [23]	Sample size (ms)
G.711 PCM	64	4.1	0.125
G.723.1[24]	5.3 / 6.3	3.65 / 3.9	30
G.726 ADPCM	32	3.85	0.125
G.728 LD-CELP*	15	3.61	0.625
G.729	8	3.92	10

*Low Decay – Code Excited Linear Predictive

The Adaptive Multi Rate (AMR) codec is the getting relevance in wireless applications being the recent new vocoder for the existing GSM networks. It has been adopted as a mandatory speech codec processing function in UMTS.

b) **Packet Loss/ Dropped Packets** .- Packet loss causes an irreparable damage to the voice signal, as retransmission in not considered while transmitting voice. Loss of voiced frames at unvoiced/voiced transition causes a significant degradation of the signal, for this reason advanced error detection and correction algorithms are used to fill the gaps created by the dropped packets. A sample of the speaker's voice is stored and it is used to create a new sound from an algorithm which tries to approximate the contents of the dropped packets or lost packets. Despite these recovering techniques, it is assumed that the packet loss should not be higher than 1%, generally.

c) **Jitter** .- When the transmission channel makes the latency of the packets to vary as a result of different queuing times or different routes, it is referred to as *jitter*. Jitter can be taken care of by using an adaptive buffer (“dejitter” buffer) which adapts itself according to the delay encountered over the networks, in order to provide a smooth voice stream at the output.

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[22] Reference : Johnathan Davidson, James Peters, “*Voice over IP Fundamentals*”, page 124; Cisco Press (2000).

[23] ITU-T Codec MOS Scoring.

[24] Notice that G.723.1 is completely different from codec G.723.

3.2 Internet-based Videoconference Systems

Videoconferencing started over the ISDN (Integrated Services Digital Network) systems that had the ability to support bi-directional audio/video delivery in real time. These systems were successful in establishing the usefulness of videoconferencing. However, these systems were expensive to install, required special rooms and were complicated to use. This led the videoconferencing systems to be accessible only to a small section of the society.

With the advent of Internet-based Videoconferencing, the accessibility of the videoconferencing technology to the common user has significantly increased. The Internet based Videoconferencing systems need low installation cost, can be used from anywhere and are easy to use. Today, the videoconferencing technology has its application in “routine” conferencing (at home), distance learning, telemedicine and many other possibilities. With the Internet bandwidth becoming cheaper day-by-day, it looks like the future of videoconferencing over the Internet is brighter than ever before.

3.2.1 Videoconference over IP

When we are planning to implement a videoconference over IP, it is necessary to understand how videoconferences differ from other applications based on IP. There are three main key points:

a) Videoconference is a real time application
b) It uses more bandwidth than other applications
c) Video traffic through a firewall is problematic

Video conferencing has different bandwidth requirements for the audio and the video components. For example, the audio component may require between 16 and 64Kbps while the video component, between 32 Kbps and 1 Mbps. So the videoconference application throughput range is established approximately from a low rate of 32 Kbps to a high rate of 1 Mbps. A typical business-quality videoconference runs at 384 Kbps and can deliver TV-quality video at 25 / 30 frames per second.

It is very important to notice that approximately a 25% of packet overhead is necessary for routing videoconference traffic. Table 3.2 shows an example of the relation between the traffic delivered and the real bandwidth necessary in a videoconference communication held over an IP network.

Table 3.2 Bandwidth for videoconference over IP

Quality	Bandwidth	Real bandwidth used
15 frames/s	128 kb/s	128 kb/s + 25% (overhead)
30 frames/s	192 kb/s	192 kb/s + 25% (overhead)

The parameters that affect to the quality of the communication are basically the same as for VoIP applications. However, this time we have also the perception of the image received, in addition to the audio, which makes the applications even more sensible to network delays and packet loss, from the user's point of view (the applications have also more complex processes to deal with, than those used for VoIP). For instance, a packet loss of about 1% may cause a frozen image or a loss of the audio signal; losses over that level are not even acceptable for a business-quality communication.

Video data is packetized in frames with variable sizes, and for this reason, the *jitter* has more relevance than in VoIP applications (that use fix audio packet sizes). Jitter is introduced due to the internal operations of the components of the network. Queuing and buffering the data in the network, packet rerouting, packet loss, network multiplexing and other factors may cause jitter. It can also be introduced at the end-user system, which is the source of the traffic. This jitter is called the *insertion* jitter, and it is introduced when certain packets are delayed before they are placed in the transmission slots of a frame, because of the previous transmission being incomplete. We need it to be regulated, as the network tends to amplify it. The packet sizes also influence in the effect of the insertion jitter, as long packets increase the overall delay due to the overhead processing.

This is one reason why multimedia applications have characteristically small packet sizes (between 32 and 512 bytes usually). To reduce some of the sender side jitter, playback buffer devices can be used at the end points. Appropriate scheduling of video and audio traffic could also reduce sender side jitter. The tradeoff in the scheduling is usually to enhance audio data and sacrifice part of the video bandwidth in order to ensure a certain QoS. Packet drop can be caused by the variation in the inter-arrival times (jitter) of the audio and video packets, due to the intermediate router processing along the path. The packet drop value is negligible, or small, for smaller values of the jitter. At the receiver end, when buffers are used for reproducing the data units, buffer overflow or buffer refreshing frequency can cause packet drop. The impact of packet drop depends on the application; for multimedia information, for instance, dropping certain important frames might become significantly disturbing to the end user [25]. Selective discard of packets on the receiver end can help applications to maintain their QoS for the user.

As a last mile technology, WiMAX links are usually connected to IP networks. For that reason, the WiMAX Forum has identified several applications for 802.16 – based systems and has developed traffic and usage models for them. Applications can be broken down into five major classes which are summarized in the following table, for latency and jitter to assure a good-quality user experience.

Table 3.3 WiMAX Application Classes [26]

Class	Application	Bandwidth (kbps)		Latency (ms)		Jitter (ms)	
1	Multiplayer interactive gaming	Low	50 -85	Low	<150	Low	<100
2	VoIP & Videoconference	Low	4-384	Low	<150	Low	<50
3	Streaming media	Low to high	5-2000	N/A		Low	<100
4	Web browsing & instant messaging	Moderate	10-2000	N/A		N/A	
5	Media content downloads	High	>2000	N/A		N/A	

According to this recommendation, latencies over 150ms would be considered not appropriate for a good quality videoconference communication. As for the jitter, variations over 50ms are considered non-acceptable for business-class requirements.

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[25] According to Salah Aidarous and Thomas Plevyak, real-time applications do not tolerate variations in network performance. The basic characteristic of real-time applications is their stringent requirements for QoS parameters, such as packet loss and delay, and variations on the required QoS result in serious degradation of application behaviour, reducing consequently the end-user satisfaction. Reference: Salah Aidarous, Thomas Plevyak, "*Managing IP Networks: challenges and opportunities*", page 65; IEEE Press series on network management (2003).

[26] Reference: WiMAX Forum White Paper "*WiMAX System Evaluation Methodology - version 2.1*" (July 2008).

The 3GPP recommendation, however, considers the range 150 – 400ms still acceptable for the end-user performance expectations. In table 3.4 we can see the 3GPP (Third Generation Partnership Project) recommendation including boundaries for information loss. We will assume these values are valid for WiMAX as well.

Table 3.4 End-user Performance Expectations - Conversational / Real-time Services [27]

Medium	**Application**	**Degree of symmetry**	**Data rate**	**Key performance parameters and target values**	
				End-to-end One-way Delay	Information loss
Audio	Conversa-tional voice	Two-way	4-25 kb/s	<150 msec preferred <400 msec limit	< 3% FER
Video	Videophone	Two-way	32-384 kb/s	< 150 msec preferred <400 msec limit Lip-synch : < 100 msec	< 1% FER
Data	Telemetry - two-way control	Two-way	< 28.8 kb/s	< 250 msec	Zero
Data	Real-time games	Two-way	< 60 kb/s	< 75 msec preferred	< 3% FER preferred, < 5% FER limit

FER is the acronym for Frame Erasure Rate and it is a measure of the number of frames of data that contained errors and could not be processed. FER is usually expressed as a percentage or exponent, and we will consider it similar to the Packet Loss Rate (PLR), as both measure a percentage of information loss (3GPP [28] uses both terms in a bit ambiguous way). Later, in Chapter 6, we will see how the Application used to monitor the video conference traffic shows the parameter PLR instead of FER.

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[27] Reference: 3GPP (December, 2008). TS 22.105 - V9.0.0 "*Technical specification group services and system aspects; Service aspects; Services and service capabilities*", page 15.

[28] According to the 3GPP, requirements for information loss are influenced by the fact that the human perception of voice and image is tolerant to a certain amount of distortion. Some degree of packet loss is acceptable, depending on the specific video and audio coder and amount of error protection used. Reference: 3GPP TS 22.105 - V9.0.0; "*Technical specification group services and system aspects; Service aspects; Services and service capabilities*", page 29.

3.2.2 H.323 Protocol

Among the recent multimedia applications that have been developed to be used on the Internet, those using H.323 protocol have become significantly popular due to the good performance they present. H.323 is an umbrella standard recommended by the International Telecommunication Union (ITU-T) that defines how real-time multimedia communications, such as videoconferencing, can be exchanged on packet-switched networks. Several vendors have successfully implemented various H.323 related standards, and even though it was originally developed for multimedia conferencing over LANs, it was later modified for VoIP as well.

H.323 is basically a recommendation that sets standards for multimedia communications over Local Area Networks (LANs) that do not provide a guaranteed QoS. It includes parts of other protocols, such as H.225.0 (which follows Q.931 standard for message encoding), H.235, H.245, RTP and RTCP; and audio/video codecs, such as the audio codecs: G.711, G.722, G.723.1, G.728, G.729, etc; and the video codecs: H.261, H.263 and H.264, that compress and decompress media streams. Media streams are transported on RTP (Real-time Transport Protocol) and RTCP (RTP Control Protocol). RTP carries the actual media using UDP, and RTCP carries status and control information. The signaling is transported reliably over TCP; the following protocols deal with signaling:

- H.225.0 defines messages and procedures for:
    - RAS signaling function. It manages Registration, Admission, and Status.
    - Call signaling, based on ISDN recommendation Q.931
    - Media packetization.
- H.235 covers security and encryption.
- H.245 negotiates channel usage and capability exchange.

The H.323 protocol stack is based on the OSI (Open System Interconnection) model [29], which implements the functionality defined by the H.323 system.

[29] However, it has its own layer sub-divisions as we can observe from Figure 3.1 in next page.

The protocol stack would include an implementation of the basic protocol defined in ITU-T Recommendation H.225.0 and H.245, as well as RTP and other protocols described above. Figure 3.1 describes the structure of a typical H.323 over IP stack.

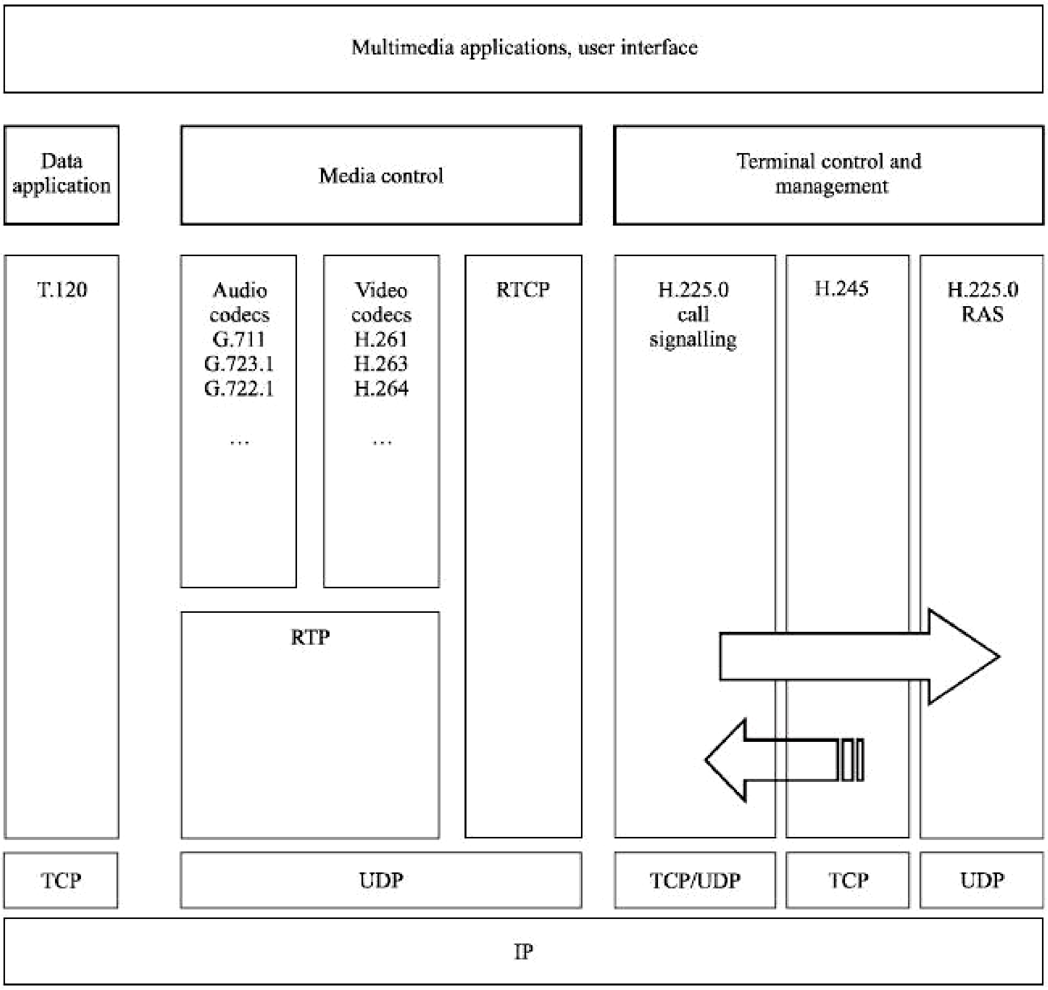

Figure 3.1 H.323 over IP stack [30]

It is important to remark that all the H.323 terminals providing video communications shall be capable of encoding and decoding video according to H.261 QCIF (Quarter Common Intermediate Format). Also, terminals shall be capable of encoding and decoding speech according to ITU-T Recommendation G.711 (that uses A-law and µ-law compression algorithms). Support for other audio and video codecs is optional.

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[30] Reference: ITU-T Rec. H.323 "*Packet-based multimedia communications systems*" (June 2006), page 289.

H.323 behind a Firewall

A firewall is part of a computer system or network, designed to regulate access and enforce an organization's security policy. It can be implemented in either software or hardware, or a combination of both, and it is possible to find it within a router, a personal computer, a host computer, or a collection of host computers. Without a firewall, a network is exposed to inherently insecure services such as common browser traffic (usually full of worms and malicious software), Telnet or FTP. Also, a firewall simplifies the management of network security by providing a single point of access to the network. All the traffic entering or leaving the network must pass through the firewall, who examines all data and blocks of data that do not match the specified security criteria.

H.323 uses TCP, as well as UDP, during phases of call setup and during audio/video transport. TCP and UDP use port numbers in their packets to identify individual connections or circuits. These port numbers are a key element used by firewalls to classify traffic so that security policy may be applied. Unfortunately, H.323 uses both statically and dynamically allocated port numbers. Generally, TCP ports 1718-1720 and 1731 are statically assigned for call setup and control. UDP ports in the range of 1024-65535 are dynamically assigned for audio/video data streams. A traditional firewall, with static policy definition, cannot predict which ports will be used for each call. The result is that a standard firewall must permit all possible port numbers to pass, which leaves the network open to a variety of hacker attacks. Also, messages sent with the H.323 protocol contain embedded transport addresses, which the firewall cannot access. Hence, traditional firewalls tend to block the passage of the H.323 traffic.

Many solutions have been proposed to overcome this issue. For instance, a non-scalable but feasible solution is to allow unrestricted ports for specific, known, external IP addresses. Another solution forces the videoconferencing clients to confine dynamic ports to a specific narrow range, which can be specified in the firewall policy. Some vendors proposed a solution where a H.323 application proxy can be used to relay H.323 calls to another H.323 endpoint. This concept is also known as "*software plug-boarding*" technique. However this approach leads to complex software, becoming a performance bottleneck due to the duplication process.

The best solution proposed so far uses a firewall that looks on the H.323 call set-up channels (static ports) and opens ports for audio/video (dynamic ports) as they are needed.

H.323 behind a NAT

Network Address Translation devices (NATs) are used to translate IP addresses so that users on a private network can “see” (referring to IP addresses) the external public network, but public network users are not able to “see” the private network user. Typically, on outgoing packets, a NAT device maps local private network addresses to one or more global public IP addresses. On incoming packets, the NAT device maps global IP addresses back into local IP addresses.

NATs affect the performance of H.323 in a way that can be understood by considering the following situation: if an H.323 endpoint A, which is inside a private network and, therefore, behind a NAT, sends a call setup message to another H.323 endpoint B on the outside, in the simplest case, H.323 endpoint B will extract the source IP address from the call setup message and send a response to this address. Because the call setup message came from H.323 endpoint A, behind the NAT, the source IP address is fictitious (maped IP), hence incorrect. The call setup will not succeed and the attempt to place a call will fail consequently.

The most extended solution to overcome this problem is to use H.323 protocol-aware NATs. These NATs maintain information about the H.323 calls originated from the private network, and map the source addresses to validate IP addresses before delivering the messages to the outside network. Later they can hand over the acquired responses to the appropriate H.323 endpoints within the private network.

4. STRUCTURE OF THE TESTING NETWORK

4.1 The Region of Mikkeli

Mikkeli is located in the region of Southern Savo, in Finnish named *Etelä-Savo*, which has a vast and beautiful extension of forests and lakes (more than 187.000 in the whole country), and belongs to the Lake District. This region has a population of 48,700 inhabitants (around 34,000 in the town itself) approximately and covers an area of 2,124.62 square kilometres, of which 422.68 km^2 is water. The population density is 28.62 inhabitants per square kilometre, so it appears to be a very interesting place to offer WiMAX broadband access, as this technology is primarily intended for rural areas. In fact, Finland has one of the largest commercial WiMAX networks nowadays, with more than 15.000 subscribers, and rising. In Figure 4.1 we can see the region where the testing network is located.

Figure 4.1 Region of Mikkeli

Mikkelin Puhelin Oyj is the company that provides this wireless broadband access in the region, and the whole network is part of a bigger one: *eSavo Network*, which covers also the eastern area (supported by Savonlinnan Puhelin Oy). It is important to note that WiMAX access network benefits from the previous networks developed, as it is providing broadband service only for the last mile, so we can find base stations nearby telephone network infrastructure most of the times. This is a very relevant point as it gives a solution not only for rural areas, but also for any place (even in big cities) where the cost of laying or upgrading landlines to broadband capacity may be prohibitively expensive.

FUNET is the Finnish University and Research Network. It is a backbone network providing Internet connections for Finnish universities and also research facilities, so it is the main network providing the service for Mikkeli University of Applied Sciences as well, and it is governed by the state-owned CSC – IT Center for Science Ltd. Figure 4.2 shows a connections map between FUNET and NORDUnet (Nordic Infrastructure for Research & Education).

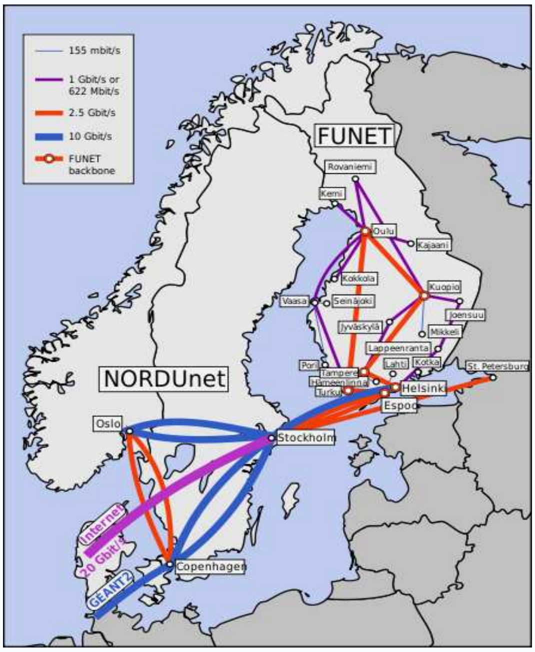

Figure 4.2 FUNET and NORDUnet.

The main backbone connections have been gradually upgraded to optical fiber since 2008. Subscriber connections are mainly 1000 or 100 Mbit/s Ethernet, but also dedicated lightpaths are offered. FUNET is connected to other research networks through NORDUnet, and to other Finnish ISPs via three FICIX points. The most important foreign connection from Helsinki to Stockholm has a capacity of 10 Gbit/s.

4.2 The Testing Network

The tests were carried out from Mikpoli building, which is situated in the main campus of Mikkelin Ammattikorkeakoulu (Mikkeli University of Applied Sciences), about 2km away from the MPY (Mikkelin Puhelin Oyj) base station and tower. Figures 4.3 and 4.4 show both WiMAX tower and Mikpoli building.

Figure 4.3 WiMAX tower

Figure 4.4 Mikpoli (Mikkeli Polytechnic)

In the following sections we will introduce a description on the structure of the case network, going through the Fixed WiMAX Network firstly, and the rest of the network afterwards.

4.2.1 The Radio Access Network.

Applications using a fixed wireless solution can be classified as *point-to-point* or *point-to-multipoint*. Point-to-point applications include inter-building connectivity within a campus and microwave backhaul. Point-to-multipoint (PMP) applications include broadband for residential areas and small to medium enterprise (SME) markets. Fixed wireless offers several advantages over traditional wired solution, which include: lower entry and deployment costs, faster and easier deployment and revenue realization, lower operational costs for network maintenance, management, and operation; and independence from the incumbent carriers.

From a subscriber station (SS) perspective, two types of deployment models can be used for fixed broadband services. One model requires the installation of an outdoor antenna at the customer premise; the other uses an all-in-one integrated radio modem that the customer can install indoors like traditional DSL or cable modems. Using outdoor antennas improves the radio link and hence the performance of the system. This model allows for greater coverage area per base station, which reduces the density of base stations required to provide broadband coverage, thereby reducing costs. An outdoor antenna, however, means that installation will require a truck roll with a trained professional.

In our case, an outdoor antenna points straight to the WiMAX tower from the rooftop, as part of a PMP fixed wireless service. Figure 4.5 shows a graphic of the WiMAX link between Mikpoli and MPY Base Station.

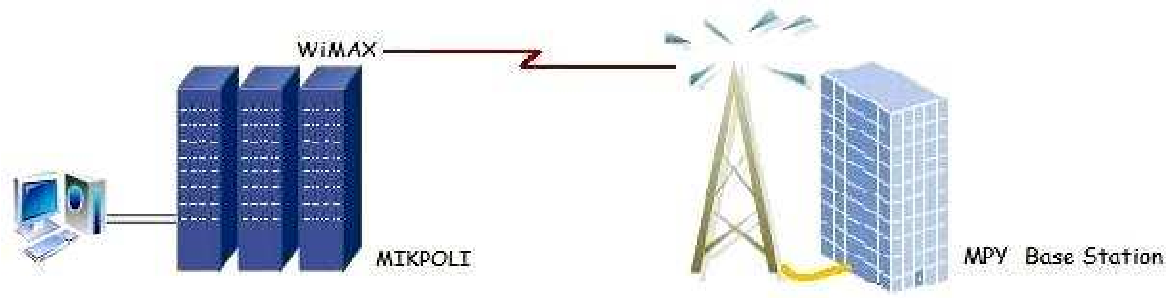

Figure 4.5 Fixed WiMAX Link.

In radio links using microwaves that are usually working with frequencies higher than 900MHz, it is a challenge to design a stable connection if there are obstacles between the transmitter and the receiver, in the 1st Fresnel zone [31].

[31] In radio communications a Fresnel zone is one of a number of concentric ellipsoids of revolution which define volumes in the radiation pattern of the electromagnetic *wavefront* (constant phase surface). To maximize receiver strength, it is necessary to remove obstacles (which may cause fading to the signal) from the radio frequency line-of-sight. The strongest signals are on the direct line between transmitter and receiver, in the 1st Fresnel Zone. Reference: Roger L. Freeman,"Radio system design for telecommunications" pages 14 – 16; Wiley & Sons (2007).

If there are objects (such as buildings or trees) along the path of the signal, some part of the transmitted signal may be lost through absorption, reflection, scattering, or diffraction. This effect is called *shadowing*, and causes attenuation to the signal. There is another effect caused by the reflection of the signal on the objects situated along the path, since each of these reflected signals travel through a different path and arrive to the receptor with different phase and amplitude, comparing to other signals. It is called *multipath* effect, and may result in fading, depending on the result of the interference between signals with different phase.

Generally, we can say that tolerance to the presence of obstacles in the line of sight between emitter and receiver decreases as the frequency used in the link gets higher. The radio channel of a wireless communication system is often described as being either LOS or NLOS. In a LOS link, a signal travels over a direct and unobstructed path from the transmitter to the receiver. A LOS link requires that most of the first Fresnel zone is free of any obstruction. If this criteria is not met then there is a significant reduction in signal strength. The Fresnel clearance required depends on the operating frequency and the distance between the transmitter and receiver locations. In our case, we know that the distance is about 2km, but we do not have access to the information about the WiMAX link configuration, so we are not sure about the frequency band or the modulation techniques the system is using. However we will try to figure out some of those details by means of our own research on the subject.

According to the WiMAX Forum [(32)] Certified™ Program for Fixed WiMAX™ released in January 2007, system profiles are based on versions of the IEEE 802.16 and ETSI HiperMAN standards and define the key mandatory and optional features that are tested in a WiMAX equipment. The list of features tested in system profiles is more stringent than the underlying standards, but does not include any new feature that is not included in the standards. For instance, the Fixed WiMAX profile is based on IEEE 802.16-2004 and only allows testing on equipment using point to multipoint operations up to 11 GHz, while IEEE 802.16-2004 equipment can operate up to 66 GHz.

[(32)] Reference: WiMAX Forum White Paper "*The WiMAX Forum Certified™ Program for Fixed WiMAX™*", pages 9-10 (January 2007).

Certification profiles limit the number of WiMAX implementations allowed to avoid market fragmentation, while meeting the demand from operators. The definition of certification profiles depends on market and vendor demand, which in turn are strictly linked to the availability of spectrum worldwide. Figure 4.6 shows the certification profiles that were available by 2007.

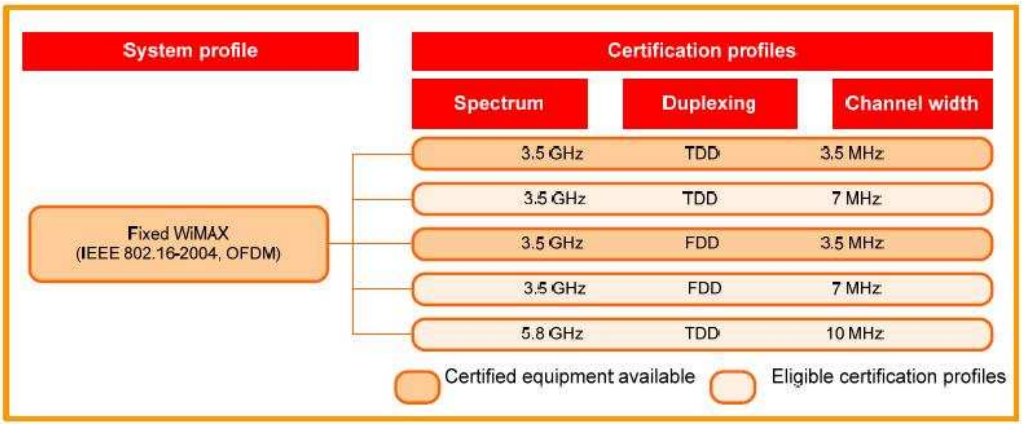

Figure 4.6 System and certification profiles (WiMAX Forum, January 2007).

We can observe that only 3.5GHz certified equipments were available by that time, although a 5.8GHz profile appears as "eligible certification profile". The Fixed WiMAX system in Mikpoli was installed in 2007 with either the first or second certification wave(33), hence, we will assume it works under the 3.5GHz profile, with a 3.5MHz bandwidth. This certification profile establishes OFDM - 256 subcarriers for Fixed WiMAX, but we do not have a lead about the duplexing technique that our system is using, however it makes sense a TDD (34) if we consider the low density of population in Mikkeli, which may suggest a dynamic allocation of spectrum based on traffic demand.

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(33) In the second certification wave it was included two optional modules: Quality of Service (QoS) and advanced security with Advanced Encryption Standard (AES), while the first wave (equipments certified in December 2006 under Release 1.0 Wave 1) only covered mandatory features and included testing for the air interface, network entry, dynamic services and bandwidth allocation. Reference: WiMAX Forum White Paper "*The WiMAX Forum Certified TM Program for Fixed WiMAX TM*", pages 11-12 (January 2007).

(34) TDD uses a single channel for the uplink and the downlink, allowing the operators to dynamically allocate spectrum for uplink or downlink transmission based on traffic demand. FDD is a simpler but less flexible mechanism that uses channels in separate frequency bands for the downlink and uplink. Regulators typically mandate the use of either TDD or FDD in licensed bands. Reference: WiMAX Forum White Paper "*The WiMAX Forum Certified TM Program for Fixed WiMAX TM*", page 9 (January 2007).

It is very important to pay attention at footnote (33) in the previous page, as it sets a possible significant lack in our WiMAX system: the QoS. In case it was not included in the profile, it can be a significant factor affecting the performance of a videoconference communication. We will discuss this in detail in Chapter 6.

Back to the Fresnel clearance required for a LOS link, we will try to develop an approximate calculation about the maximum high of the obstacles we should find in the path between the BS and the SS in order not to have a significant reduction in signal strength. Figure 4.7 illustrates the Fresnel zone, with a maximum radius “r”, at halfway.

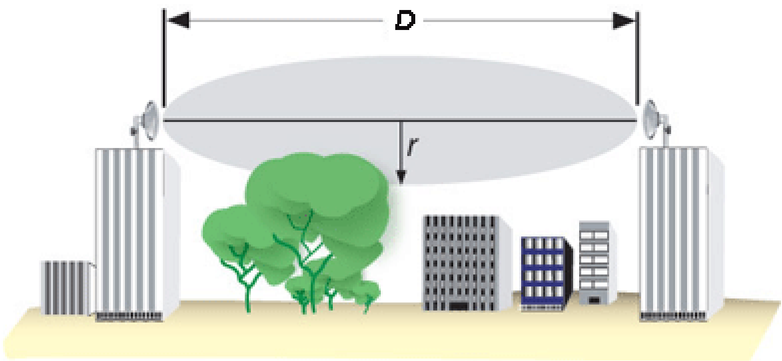


Figure 4.7 1st Fresnel Zone geometry.

To calculate the radius of the n-Fresnel zone, on a surface perpendicular to the propagation path, the following equation provides a good approximation [35]:

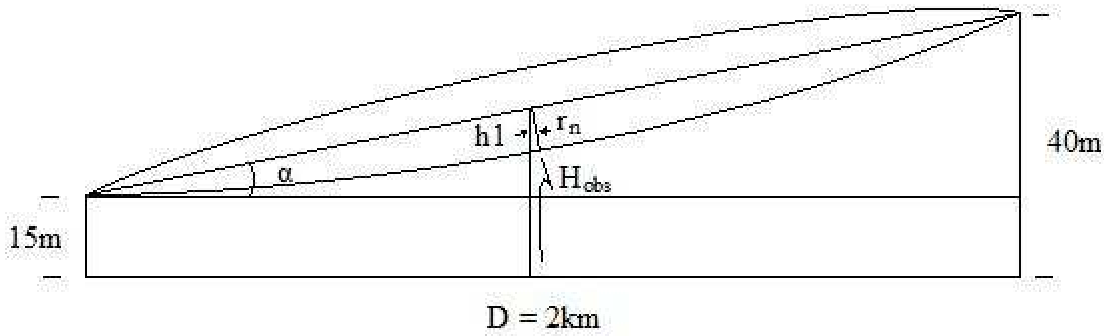
$$r_n \cong \sqrt{n\lambda\left(\frac{d_1 \cdot d_2}{d_1+d_2}\right)} \qquad (4.1)$$

where r_n and d are in the same units, and D is the sum of d_1 and d_2 (we can calculate r_n at any point in the path), which is approximately 2km in this case. The most critical point according to the maximum radius is the halfway between emitter and receiver, hence we will establish that point as a reference in our calculations.

[35] Reference: Roger L. Freeman, ”Radio system design for telecommunications” 3rd Edition, page 16. Wiley & Sons (2007). We find the same approximation (but different units) at Recommendation ITU-R P.530-12, page 4.

For a 3.5GHz frequency, the wavelength is approximately 85.71mm, if we consider the propagation speed of the wavefront, around $3 \cdot 10^8$ m/s. Then, for the 1st Fresnel zone (n = 1) we have an approximate radius of 6.53m in the halfway.

Now we can estimate the maximum high for the obstacles, H_{obs} [(36)], attending this reference point. We may consider that the antenna placed at Mikpoli building is about 15m high from the floor, and the WiMAX tower, about 40m, thanks to the location where it is placed (the only high hill in downtown). We will assume a flat floor approximation for the ground between the two stations.



where the approximate angle is: $\tan\alpha = \left(\frac{40-15}{2 \cdot 10^3}\right) \rightarrow \alpha \cong 0.716°$

We will also approximate h1≈ r_n, as the angle α is very small. Then we can calculate an approximate value for H_{obs} :

$$H_{obs} \cong 15 + \frac{D}{2}\tan\alpha - r_n = 21m$$

It is difficult to be certain about the clearance of the 1st Fresnel zone, as the whole city center has buildings higher than 20m. However, it seems to be a controllable margin.

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(36) Although it might be obvious, it is important to notice that H_{obs} is calculated as an approximate value, being possible to find lower obstacles in the first-half of the radio path that may cause degradation to the signal, as the path is not parallel to the ground floor, but oblique. In addition, it is necessary to remark that it is usual to permit an infringement of the 1st Fresnel Zone in a 40%, so that the *radius* r_n appears usually multiplied by 0.6 in the calculations. Reference: WiMAX Forum White Paper "*WiMAX's technology for LOS and NLOS environments*" (August 2004).

Radio signal *path loss* is a capital issue in the design of a wireless system, as it will determine many elements of the radio communications system like the power of the transmitter, the gain of the antennas and their location. It is important to notice that WiMAX is only a part of the end-to-end network that implements an access via radio, and for that reason, it is more likely to lose information, comparing to the core network which is usually a wired backbone implemented with high capacity optical links.

The sum of all the losses and gains of a communications system is called the *link budget*, and it is a fundamental concept in every radio link. The result of the link budget tells us the necessary power to achieve a certain level of signal at the receptor, according to a certain SNR given, at a certain BER (Bit Error Rate). Also depending on the distance, a certain modulation will be used, as we can see in Figure 4.8. In our case network, it is possible that a 64QAM modulation is being used when the channel conditions are good, as the SS is located about 2 km away from the BS.

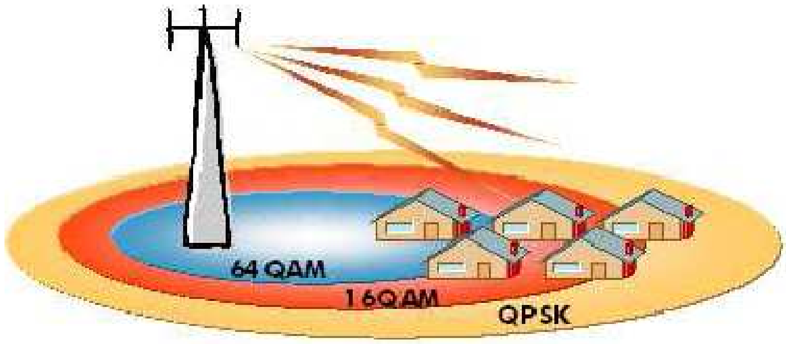

Figure 4.8 Point-multipoint scenario

In a fixed deployment with basic functionality, 802.16-2004 and 802.126e offer similar performance. Single sector maximum throughput for both versions of WiMAX is about 9.4 Mbps for a 3.5 MHz channel, or 35 Mbps for a 10 MHz channel. Base station range in densely populated areas can go up to a few kilometers depending on attributes such as CPE type, frequency band, morphology and so on. In networks that are capacity constrained, the number of base stations installed depends on throughput demand, rather than range.

4.2.2. The Backbone

It was a challenge to decide what kind of backbone (core network) would be the most suitable and useful according to the tests that were going to be developed, as the main purpose of this case study was to evaluate the behavior of the *Fixed* WiMAX link working with videoconference data, and it was necessary to be sure that the rest of the circuit involved would not interfere or distort the results.

As the experiments would be carried out from two computers placed in the same building, and there was only one access point available to the WiMAX link, it was not possible to have an end-to-end session with both end points transmitting through the same wireless access network. In this sense some options were considered, but all of them needed to use the Internet Network, so it seemed to be necessary to have a reliable access network for the second PC, such as an optical fiber link or an ADSL connection. Both wired, which made them reliable to help to have a videoconference lossless session. Figure 4.9 shows a scheme of the whole network used to carry out the tests.

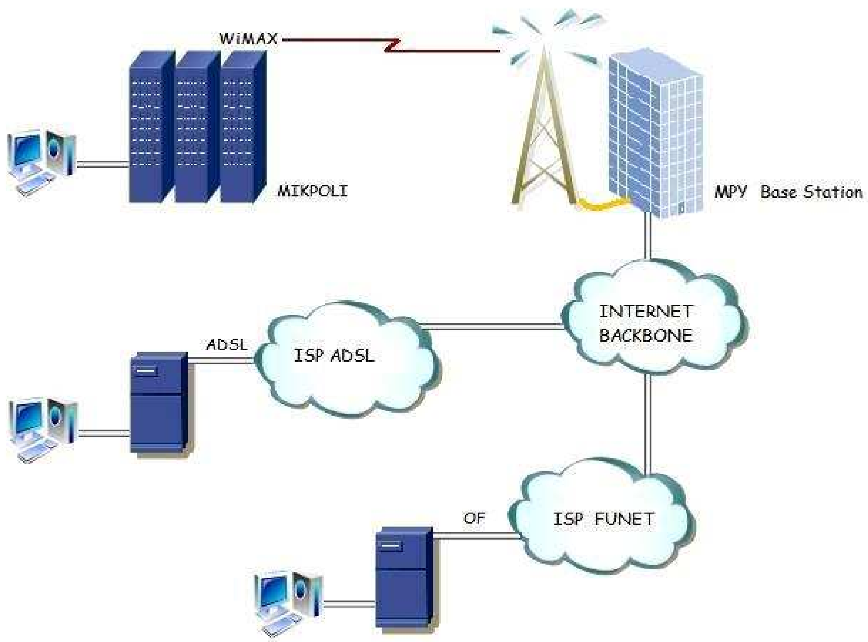

Figure 4.9 Scheme of the case network

As we can see, the endpoints used for the communication are situated in different ISP's networks, therefore, it was necessary to perform tests on different points of the core network, so that we could study the behavior of the whole system, section by section.

After the tests it would be necessary to separate and contrast the different results obtained, in order to have conclusions about the quality of the transmission, in a modular way.

A very important point is the throughput available, as we are going to study the behavior of the network in a videoconference session. It was already mentioned in Chapter 3 that the optimal rate for “business quality” videoconference is 384kbps, however, it doesn’t mean that it is not possible to work with lower rates. In fact, this rate is established according to the current standards used for video and audio sampling and compression, so it might happen that new standards are developed and the optimal data rates keep changing. Figure 4.10 shows the instant throughput provided by the ADSL connection.

Figure 4.10 ADSL – WiMAX throughput

The ADSL connection provided about 375kbps uplink, and therefore we could not test call rates over 350kbps at that end point for a videoconference session. Besides, the optical fiber link provided a 28Mbps uplink, which meant an unlimited speed for the purpose of the study; and the WiMAX connection provided around 480kbps uplink (which leads us to think about a probable 512k UL), enough to test the current optimal videoconference rates. We consider these possible nominal values (UL/DL) for the three technologies, according to the speed tests we carried out:

- ADSL → 384kbps/5Mbps
- OF → 30Mbps/60Mbps
- WiMAX → 512kbps/1Mbps

Internet represents a problem as for assurance of QoS for videoconference communications. For many years, the Internet was primarily used by scientists for networking research and for exchanging information between each other. Remote access, file transfer, and e-mail were among the most popular applications, and for these applications the datagram model works very well. But the World Wide Web, has fundamentally changed the Internet as it has become the world's largest public network. New applications such as video conferencing, Web searching, electronic media, discussion boards, and Internet telephony are being developed at an unprecedented speed.

Because the Internet treats all packets the same way, it can only offer a single level of service. The applications, however, have diverse requirements: interactive applications such as telephony or videoconference are extremely sensitive to latency and packet losses. When the latency or the loss rate exceed certain levels, these applications become literately unusable. In contrast, a file transfer can tolerate a fair amount of delay and losses without much degradation of perceived performance. Customer requirements also vary depending on what the Internet is used for.

Implementing these QoS capabilities in the Internet has been one of the toughest challenges in its evolution, touching on almost all aspects of Internet technologies and requiring changes to its basic architectur. For more than a decade the Internet community has made continuous efforts to address the issue and developed a number of new technologies for enhancing the Internet with QoS[37] capabilities (MPLS, traffic engineering, etc).

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[37] Salah Aidarous and Thomas Plevyak develop a detailed explanation on *Quality of Service Methods* and *Internet Technology Choices* in their book, "*Managing IP Networks: challenges and opportunities*", pages 167 - 171; IEEE Press series on network management (2003).

But it is still quite usual that ISPs are not able to assure higher levels of QoS, and therefore the packets are being transported through a public cloud with high variations in the traffic routed (the queuing that takes place in the nodes of the backbone affects directly to the jitter suffered by the packets routed). According to this, it was important to have the minimum possible hops between the end-points, so that the Internet Network did not affect considerably.

In addition, it was interesting to try to find some applications designed for videoconference that were provided with an option for QoS, including a special bit, or group of bits, in the IP header. But then it would be necessary to configure the network in order to indentify the packets and give priority according to the QoS assumed, but we did not have access to the inner routing devices, therefore we could not be sure at all about these capabilities, but only in the radio access network (WiMAX supports certain QoS capabilities). Later, we will see how the Internet backbone did not affect considerably on delays and jitter, so it was not a problem for the testing work.

As we can see in Figure 4.11, at least 12 hops separate the endpoints between ADSL and WiMAX (not all the inner devices of the backbone are capable of counting hops, usually). We obtained the same number of hops from the optical fiber link (FUNET).

Hop	IP	DNS	Ping	Last Ping	Max Pi...	Avg Ping	Packet Loss	Success	Failures	Updated
1	Timed out									
2	80.222.16.1		11 ms	11 ms	11 ms	11 ms	0 %	1	0	15:21:27
3	141.208.151.105		9 ms	9 ms	9 ms	9 ms	0 %	1	0	15:21:27
4	141.208.8.218	kvlcore1-s-2-1-1.datanet.tele.fi	12 ms	12 ms	12 ms	12 ms	0 %	1	0	15:21:27
5	141.208.204.129	hkicore1-e-1-0-0.datanet.tele.fi	16 ms	16 ms	16 ms	16 ms	0 %	1	0	15:21:28
6	141.208.8.10	hkiasbr1-s0-0-0.datanet.tele.fi	16 ms	16 ms	16 ms	16 ms	0 %	1	0	15:21:28
7	193.110.226.64	fne.ficix1-ge.ficix.fi	19 ms	19 ms	19 ms	19 ms	0 %	1	0	15:21:28
8	87.236.153.9		19 ms	19 ms	19 ms	19 ms	0 %	1	0	15:21:28
9	87.236.154.3		25 ms	25 ms	25 ms	25 ms	0 %	1	0	15:21:28
10	82.197.21.138	lt3-r1-ge3-10.mpynet.fi	24 ms	24 ms	24 ms	24 ms	0 %	1	0	15:21:29
11	82.197.21.241	lt1-r1-vl4.mpynet.fi	21 ms	21 ms	21 ms	21 ms	0 %	1	0	15:21:29
12	82.197.21.242	term-r2-ge0-0-4.mpynet.fi	24 ms	24 ms	24 ms	24 ms	0 %	1	0	15:21:30
13	88.85.146.162	a88-85-146-162.mpynet.fi	56 ms	56 ms	56 ms	56 ms	0 %	1	0	15:21:30

Figure 4.11 Hops between ADSL and WiMAX end-points. (pt360 Tool Suite)

In computer networking, a hop represents one portion of the path between source and destination. When communicating over the Internet, for example, data passes through a number of intermediate devices (like routers) rather than flowing directly over a single wire. Each such device causes data to "hop" between one point-to-point network connection and another. In networking, the hop count represents the total number of devices a given piece of data (packet) passes through. Generally speaking, the more hops data must traverse to reach their destination, the greater the transmission delay incurred.

Network utilities like *ping* can be used to determine the hop count to a specific destination. Ping generates packets that include a field reserved for the hop count. Each time a capable device receives these packets, that device modifies the packet, incrementing the hop count by one. In addition, the device compares the hop count against a predetermined limit and discards the packet if its hop count is too high. This prevents packets from endlessly bouncing around the network due to routing errors.

Both routers and bridges are capable of managing hop counts, but other types of intermediate devices (like hubs) are not. In this sense, it is necessary to remark that the number of hops, in this case 12, is just an approximate value (a minimum indeed) if we shall consider that not all the intermediate devices are capable of managing hop counts. This fact determined the way we focused the case study as for the network performance. We will go through this with more detail in the next chapter.

5. PERFORMANCE EVALUATION OF THE NETWORK

Experience over the years has demonstrated that the analysis and testing of large networks has significant scalability issues. For that reason, several simulation models have been progressively developed in order to simplify the network testing, with the main goal of looking for efficiency in the results. Among them, we could say that *Parallel Simulation* is the most prominent one, but there are others like *Fluid Flow Simulation*, or the *Cloud Model* (38). These models are used for simulation approaches but they also set a method for the analysis and troubleshooting of a real network, using a simplified view which may appear less accurate, but they are highly efficient.

In our case, the Cloud Model seems to fit quite well for the purpose of the study, as it is based in topology domains (clouds) that consider the delay and loss of information before and after the different domains, neglecting the behavior of the inner components of each cloud. This point of view is very helpful in our case for two reasons: firstly, the nature of Internet as a network of networks makes it difficult to manage as we do not have access to every single segment and device of it. We could say it is beyond our reach; for instance, if we want to know the topology of the network using the *traceroute* command, as we already mentioned at the end of Chapter 4, some devices are not capable of managing hop counts. Therefore, we cannot attempt hop-by-hop measurements in equal conditions for the whole backbone of our testing network.

In second term, as the aim of the study is to evaluate the performance of the radio access network, we only need certain details about how the information is degraded when it travels through the rest of the network, in order to have concluding results about the global trade of information. It is crucial to be able to separate what is caused by the WiMAX Network from what is caused by the rest of the network, and the Cloud Model establishes clear and simple schemes that permit to analyze the performances separately.

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(38) Reference: F. Baumgartner, M. Scheidegger, T. Braun, "*Enhancing Discrete Event Network Simulators with Analytical Network Cloud Models*", First international workshop on Inter-domain performance and simulation, IPS 2003.

5.1 Inter-domain Network: the Cloud Model

Using delay and loss models for network domains allows testing networks without exact knowledge of the network topology and without the need to test each single node within a network. A scalable approach is possible by replacing node-per-node network domains by analytical models for domains and inter-domain links. Figure 5.1 shows the simplification of a network node topology to a topology of domains.

In the graphic, we can assume each network as a cloud and replace the three domains by a single node providing the modeled behavior for the full network. This way, the new node assumes not only the routing work but also the delay and packet loss of the whole domain.

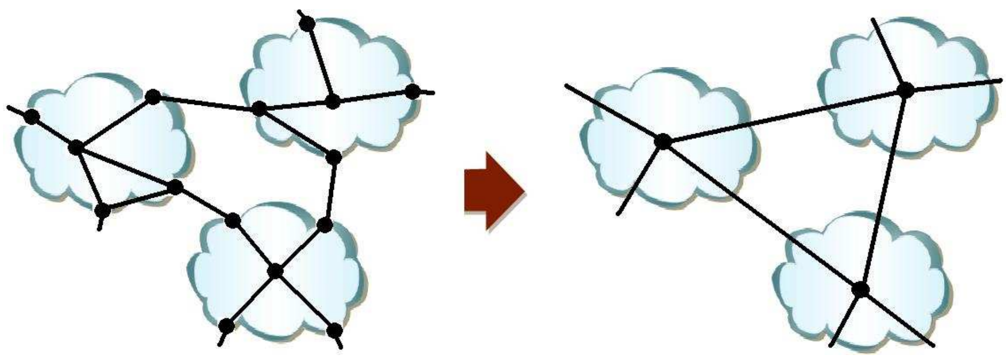

Figure 5.1 Topology of domains in the Cloud Model

In the case network of our study, there are three ISP's providing connection to three computers; therefore we can apply this inter-domain model in order to analyze the relation between the traffic loads at the inbound links and the distribution at the outbound links of each domain. This structure is based on the assumption that congestion is unlikely to occur inside ISP networks and the Internet backbone, since there are modern mechanisms nowadays that prevent it. However, as part of the study, in this chapter the main goal is to measure the jitter, delay and packet loss that take place between each domain, when we send a certain amount of traffic (different loads have been tested) between two endpoints, so that we can have a preliminary evaluation of the behavior of the network. This way we are able to predict, as a first approach, the limitations we may find when attempting a videoconference session. Later, on Chapter 6 we will see how the predictions matched quite well with the results obtained on a videoconference communication.

5.2 Testing Conditions

In radio communications the environment conditions have a capital relevance as for the quality of the data transference. There are many factors that can affect significantly causing fading to the signal, and therefore the designing process of a radio link is very important. As we introduced in Chapter 4, the radio signal path loss determines many elements of a radio communications system (power of the transmitter, gain of the antennas, etc), but there is a factor that must have special consideration in order to ensure a margin of availability in the service provided: the atmospheric conditions. In particular, precipitations have a major role in radio communications using frequencies over 1GHz.

Finland is located in a rainy and snowy area (practically 6 months of snow, and 6 months of irregular rain, but intense at intervals), and therefore it is always a challenge to make balance during the designing time, in order to ensure a future stable communication attending the hardest possible atmospheric conditions. In fact, the effect caused by the rain, snow and fog presence in a wireless connection for microwave frequencies may become considerable, depending very much on the modulation used.

Rain affects the transmission of an electromagnetic signal in three ways: It *attenuates* the signal, it changes the *system temperature* and it modifies the *polarization*. All three of these mechanisms cause degradation to the received signal quality and become increasingly significant as the carrier frequency increases.

The *attenuation* comes from the scattering and absorption of the electromagnetic waves caused by drops of liquid water. The scattering diffuses the signal, while absorption involves the resonance of the waves with individual molecules of water. Absorption increases the molecular energy, corresponding to a slight increase in temperature, and results in an equivalent loss of signal energy. Attenuation is negligible for snow or ice crystals, in which the molecules are tightly bound and do not interact with the waves.

The attenuation increases as the wavelength approaches the size of a typical raindrop, which may vary between 0.5mm and 10mm[39].We assume a range 1–5mm, since bigger raindrops shall break up into smaller pieces when colliding with other raindrops. Wavelength and frequency are related by the equation $c = \lambda f$, where λ is the wavelength, f *is* the frequency, and c is the speed of light (aprox. $3x10^8$m/s). For instance, at a link frequency of 4GHz, the wavelength is 75mm. The wavelength is thus from 15 to 75 times larger than a raindrop and the signal passes through the rain with relatively small attenuation [40]. For 12GHz the wavelength starts to be comparable to the size of a raindrop and attenuation becomes remarkable. Over 20 GHz, the wavelength is below 15mm, which causes a severe attenuation to the signal received. The standard method for representing the rain attenuation (A) is expressed in decibels, and depends on two key parameters:

$$A = \gamma_R d_{eff} \quad [\mathrm{dB}] \qquad (5.1)$$

d_{eff} [41] is the *effective* path length (km), and γ_R is the *rain specific attenuation* (dB/km), which depends on frequency, rain rate (mm/hour) and polarization. The design of a radio link includes margin to compensate the effects of rain at a given level of availability. The rain attenuation margin may vary between 5dB and 30dB [42] depending on availability requirements, range of the radio link, band of frequencies used, and rain statistics.

Changes in *polarization* are also important. Due to the resistance of the air, a falling raindrop assumes the shape of an oblate spheroid. Wind and other dynamic forces cause the raindrop to be rotated at a statistical distribution of angles and consequently, the

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[39] Reference: David R. Lide, "*Handbook of chemistry and physics*" 85th edition, page 15/41 (section: "Characteristics of particles and particle dispersoids"); CRC (2004).

[40] According to the ITU, the attenuation is negligible for frequencies below 5GHz. This assertion suggests that we should not find significant attenuation in our radio link, using a 3.5GHz signal. Reference: ITU-R P.530-12, page 15.

[41] Rain models differ mainly in the way the effective path length d_{eff} is calculated. Authoritative rain models widely used are the Crane models (*Global*, *Two-Component* and *Revised Two-Component model*) and the ITU terrestrial model. Crane models are popular for space-earth links, but also have terrestrial models; they are all deeply discussed in the reference book "*Electromagnetic wave propagation through rain*" by Robert K. Crane (Wiley, 1996). We follow here the ITU model, where d_{eff} is the product of the path length d and a "factor of distance" $r = 1/(1+ d/d_0)$, where d_0 comes to be $d_0 = 35\exp(-0{,}015R)$; being R the rain rate (mm/hour). Reference: ITU-R P.530-12, page 15.

[42] Reference: HNS-23885, "*Mitigación de Aspectos que Afectan la Propagación…*", Hughes Network Systems (2002).

transmission path length through the raindrop is different for different signal polarizations, and the polarization of the received signal is altered. For a radio communications system with dual linear polarization(43), the change in polarization has two effects: loss in the signal strength because of misalignment of the antenna, and additional interference noise which appears due to the admission of a portion of the signal in the opposite polarization.

The effect of the change in the *system temperature* is maybe the less noticeable in this case. The figure of merit of the station receiver antenna is the ratio of the antenna gain to the system temperature, *G*/*T*. If the system temperature increases, the figure of merit is reduced. The sum of the equivalent temperature of the receiver T_e and the antenna noise temperature T_a, result in the system temperature; relation represented in equation 5.2.

$$T = T_a + T_e \qquad (5.2)$$

The water from the rain has approximately the same temperature as the air, hence, T_a does not change, and the system temperature remains fairly constant. In satellite communications, the earth station receiver antenna is pointing to the clear sky, hence it has a temperature of about 25K. When it rains, the temperature of the liquid water is about 300K, thus the rain increases the sky temperature by an order of magnitude, so the noise admitted by the earth station antenna increases, causing further signal degradation.

We already introduced in Chapter 2 that WiMAX Technology presents an option for adaptive modulation which increases the availability of the channel by sacrifying bandwidth, as the system commutes to lower-order modulation when the channel conditions worsen. In LOS systems it is usual to work with high-order modulations due to the good behavior of the channel as for multipath attenuation. The only disadvantage about high-order modulations is the C/I relation, and the required sensitivity (44) of the receiver for demodulation. The typical difference in sensitivity required, between QPSK and

(43) Most deployments in 802.16-2004-based systems use both vertical and horizontal polarization. Reference: IEEE 802.16TM - 2004. "*IEEE Standard for Local and metropolitan area networks. Part 16: Air Interface...*", page 787.

(44) It is important to realize how much more robust is QPSK modulation comparing to 16QAM and 64QAM, which makes it the most reliable modulation in order to combat hard rain conditions, and therefore amplitude attenuation (QPSK does not use the amplitude of the signal to modulate the information, but the difference in phase. Therefore, attenuation affects less to this modulation, which uses four symbols with same amplitude but four different phases, separated 90º).

64QAM is about 14dB; between QPSK and 16QAM is about 7dB. This is directly related to the ranges provided, being 64QAM the worst in this sense.

As the tests were carried out between April and July, it was not possible to measure the effect of snow falling. Yet, we had the opportunity to compare the quality of the communication with different intensities of rain (up to 25 - 30 mm/hr), and we obtained interesting results in the different measurements. These results were later contrasted with the experiments that were carried out using videoconference applications, which were also tested under similar atmospheric conditions.

5.3 Traffic Modelling

A study of multimedia performance over a network requires accurate representation of the workload model. Measurements of modeled traffic loads on the network provide a snapshot of real videoconference workloads. In this chapter we present a characterization for video and audio traffic transported over the case network.

5.3.1 Audio

The performance of audio packets depends entirely on the application that is being used, as there are many umbrella recommendations available, and each one implements a different encoder. For example, G.723.1(45) is a dual-rate speech-coding standard, which operates in low bit rate while maintaining high perceptual quality. It is recommended as the preferred speech codec for the ITU H.323 conferencing standard over the Internet when the access link to the Internet has a limited bandwidth. It is important to notice that increasing the number of frames per audio packet improves the bandwidth utilization and decreases network packet overhead. However, it also introduces additional delay since a packet has to wait for all the audio frames to be accumulated before sending it across.

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(45) The G.723.1 encoder provides one frame of audio every 30 msec. The audio frame size is 20 bytes of PCM samples for the low rate (5.3 kb/s) and 24 bytes for the high rate (6.4 kb/s). Each application is set for a particular number of frames per audio packet before the conference starts. Both applications are forced to use this number as the maximum number of frames incorporated in an audio packet. Reference: ITU–T Recommendation G.723.1, "*Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s*" (May2006).

This delay can be even more harmful especially with the timeliness requirement of real-time audio telephony.

We can admit, therefore, that the choice of the number of frames per packet affects both local latency and protocol overhead. We are primarily concerned with the latency induced by the audio packet preparation manager that buffers the captured audio frames and synthesizes them into audio packets. This latency is computed before the packet is actually sent on the network, and it only accounts for local capture and packet preparation delay. The choice of the number of frames per audio packet should minimize both local latency and packet overhead, being the optimal point for the number of frames between *seven* and *eight*, for both G.723.1 audio bit rates. This value should be used by terminals that target packet efficiency (17–25 bytes packet overhead) as well as low latency.

A packet with a fixed header (RTP[46]+UDP+IP+PPP), eight high-rate audio frames (24 bytes/frame), and 19 bytes overhead, would have a total of 256 bytes:

$$(8+12+20+5) + 8*24 + 19 = 256 \text{ bytes}$$

5.3.2 Video

It is important to understand how the video source generates video data in order to analyze video-generated traffic. Video is different from audio in the sense that the generated bit rate can be variable while the audio bit rate is usually constant. The ITU-T standard left this issue up to the application as to whether to use a fixed or variable bit rate for video. In addition, video can be fragmented as a measure to reduce the size of the data, since video-generated frames can be quite large usually (audio is not fragmented). Fragmentation means that a video frame is divided into multiple pieces, and each piece forms a video packet that is sent across the network. There is an upper limit on the maximum fragment size used by the fragmentation process. The burstiness of the video source due to fragmentation may cause multiple fragments to cluster in a small interval of time, which can lead to packet loss, latency, or at least some amount of unfavorable jitter.

[46] Using RTP is important for real-time traffic, but a few drawbacks exist. The IP/RTP/UDP headers sum 40 bytes, which is twice as big as the payload present when using G.729 with two speech samples (20 ms). Though, it is possible to compress this to 2 or 4 bytes by using RTP Header Compression (CRTP). Reference: J. Davidson, James Peters, "*Voice over IP Fundamentals*", page 131; Cisco Press (2000).

The mean packet size of video traffic is usually less than 250 bytes and the majority of the video packet sizes lay in the range of 50-250 bytes, which suggests that a maximum fragment size of between 256 and 512 bytes is an appropriate choice for an efficient video transmission, in terms of latency, jitter and packet overhead. The jitter, in fact, is less significant as the maximum fragment size is increased, because a smaller number of packets with less delay variations is generated.

5.4 Performance Evaluation

According to the ranges used by the applications for audio and video packets, we decided to test packets between 50 and 1024 bytes in order to analyze the behavior of the network and obtain preliminary conclusions about hypothetical performance of videoconference communications. For the traffic modeling we used ICMP Echo Request and Echo Reply message streams (Figure 5.2 illustrates the structure of ICMP packets), so

ICMP packet

	Bit 0 - 7	Bit 8 - 15	Bit 16 - 23	Bit 24 - 31
IP Header (160 bits OR 20 Bytes)	Version/IHL	Type of service	Length	
	Identification		*flags* and *offset*	
	Time To Live(TTL)	Protocol	Checksum	
	Source IP address			
	Destination IP address			
ICMP Payload (64+ bits OR 8+ Bytes)	Type of message	Code	Checksum	
	Quench			
	Data (*optional*)			

Figure 5.2 ICMP Packet

that it was possible to measure the jitter, latency and packet loss at different points of the network dividing the values obtained by two, as the results referred to *round trip*[47] times.

[47] Round-trip time, in this case refers to the time that elapses since the Echo Request packet leaves an emitter ‘A’ until the Echo Reply message (acknowledge), sent by the destination ‘B’, arrives back to the emitter ‘A’.

For that purpose we used two different tools: MS-DOS commands (including ipconfig, ping, tracert, etc.), and the program PacketTrap pt360 Tool Suite, which was very useful, as we could find complementary testing and monitoring network tools to try. It was possible to use the basic *ping* command, but also some more sophisticated tools, such as the *Netflow Listener* (monitors network traffic) or the *Traffic Jam*, for instance, which is a traffic generator that permits to customize the size of packets, the bandwidth used (kbps), the percent of bandwidth dedicated to data (control over hypothetic overhead and payload), the amount of packets sent per second, and some other parameters. We tried different configurations in order to stress our case network in different ways, and we obtained satisfactory results even when we introduced traffic loads very close to the theoretical limit throughput of the network.

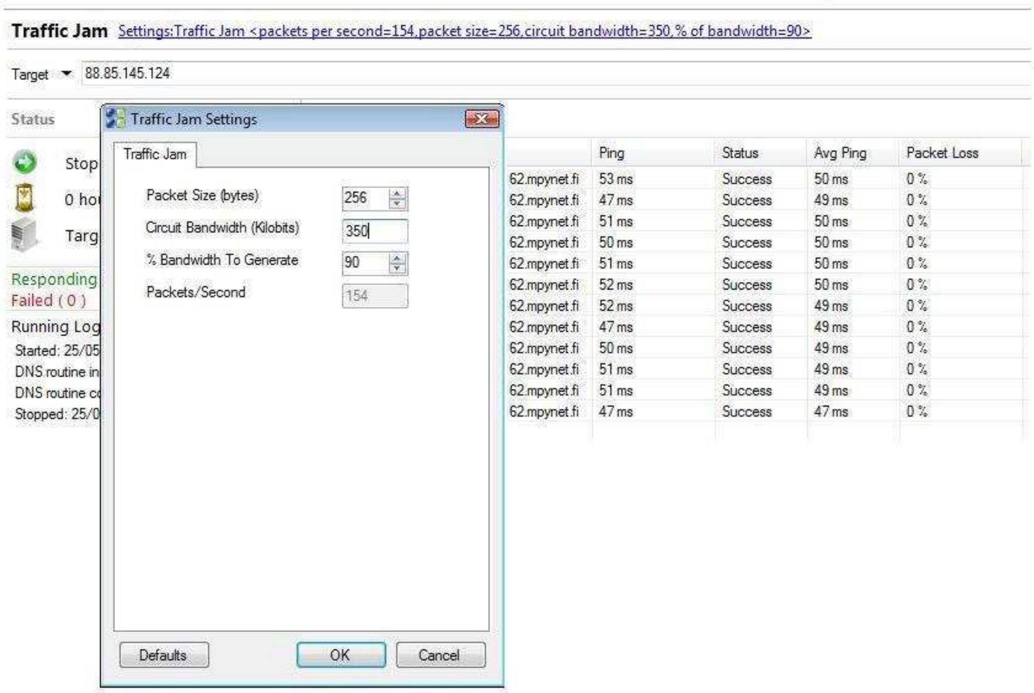

Figure 5.3 Traffic Jam tool (pt360 Tool Suite)

According to the Cloud Model, a network simplification is possible by replacing node-per-node network domains by analytical models for domains and inter-domain links. In this sense, as we already introduced in Chapter 4, we can see the end-to-end path as three interconnected clouds which include: a) WiMAX network b) Internet Backbone c) ISP Network (FUNET/ADSL). Figure 5.4 illustrates the model.

Figure 5.4 Three clouds represent the end-to-end path.

In the course of the tests, we could measure the end-to-end losses, delay and jitter, as well as the evolution of the data between the clouds, and it was clear how much more reliable were the wired sections comparing to the radio link, in every sense. The delays and jitter were quite stable most of the time within the three clouds (always higher through the wireless section), but we found remarkable differences attending the variations in the packet loss, as it was very stable (practically nil) through the Internet and the ISP networks, but not through the WiMAX link. In addition to the variations of the losses depending on the size and amount of the packets sent, we realized there was an increase of the percentage of packets lost when there was a rainfall heavier than 20 mm/hr. An example is showed in Figure 5.5, for a set of 300 packets of 128 bytes over the ADSL – WiMAX end-to-end connection.

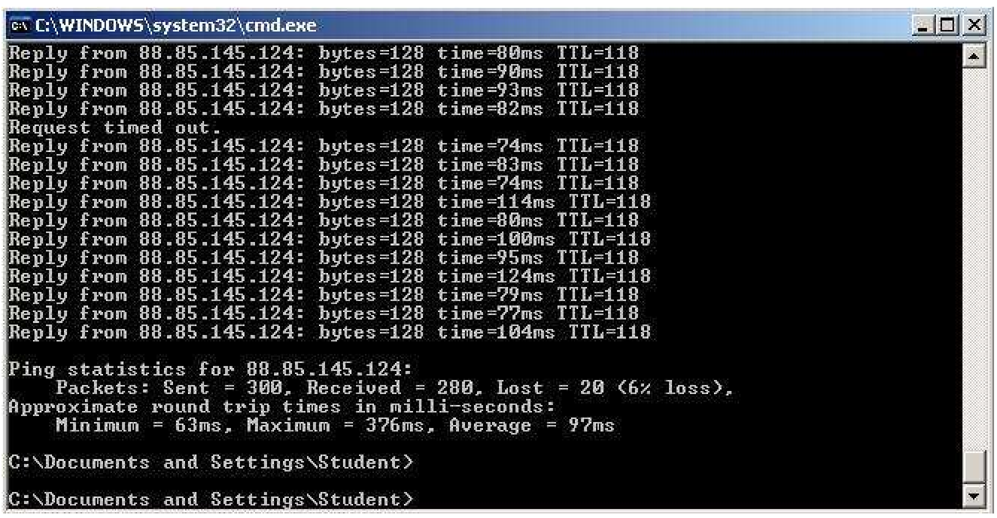

Figure 5.5 Ping statistics for ADSL – WiMAX

As we can observe from Figure 5.5, sometimes even bursts of losses of around 6% of the data were measured in a round-trip (so we may consider a 3% average for one way), during a short period of time. It is also remarkable the jitter experimented in these conditions, which increased even more than 40ms end-to-end, eventually (WiMAX Forum[48] considers 50ms a high-quality boundary). Also the average latency increased, comparing to the results obtained in normal conditions, that is, without rain.

An average of the evolution of packet loss through the WiMAX network is shown in Figure 5.6. We tested the link under different intensities of rain (these intensities were contrasted in different real-time weather websites) and packet sizes. It is important to point out that we are not able to assure the relation between the rain and the increment of the percentage of packets lost, as a deeper study on the physical layer of the WiMAX link should be done in order to obtain reliable conclusions about this issue, and we did not have the tools to develop such study. However, we found it interesting to remark this intriguing and, definitely, relevant coincidence between this two aspects that took place several times (different days) during the course of the experiments.

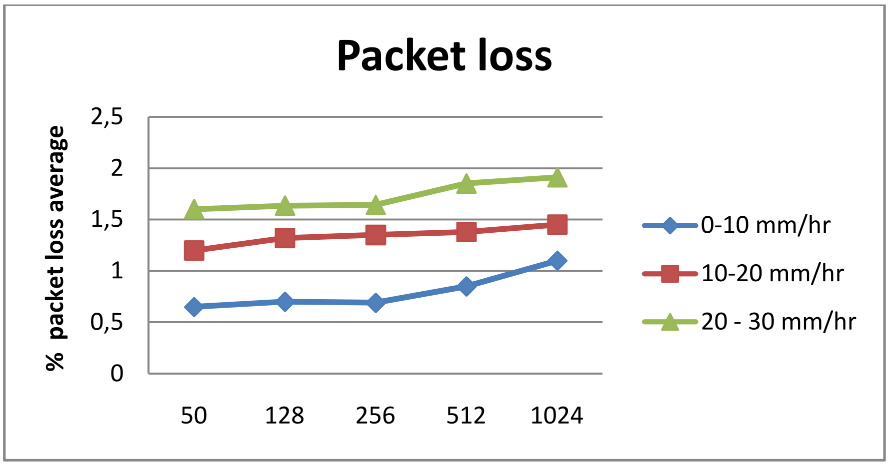

Figure 5.6 Average of packets loss through the WiMAX Network.

Between 1000 and 10000 packets were sent during each experiment, with different sizes in the range of 50 – 1024 bytes. We can neglect the losses along the Internet backbone and the ISP networks, as we obtained very low results comparing to the

[48] Explained in Chapter 3. Reference: WiMAX Forum White Paper "*WiMAX System Evaluation Methodology - version 2.1*" (July 2008).

WiMAX link ones. It is very important to remember that a packet loss over 1% may not be acceptable for a business-quality videoconference (videophone requirements, according to 3GPP [(49)] Recommendation).

The results obtained suggest that our case network worked around that limit sometimes, as a consequence of several factors related to the radio link configuration and the testing conditions, however, we cannot say what parameters were involved, as we already mentioned some lines ago (PHY and MAC layer information of the radio link was not available).

As for the delays and jitter, we can find the distributions in Figure 5.7 and Figure 5.9, for the three clouds (example WiMAX – ADSL end-to-end connection). Again, it is interesting to see how determinant was the WiMAX Network about these parameters, in comparison to the other sections.

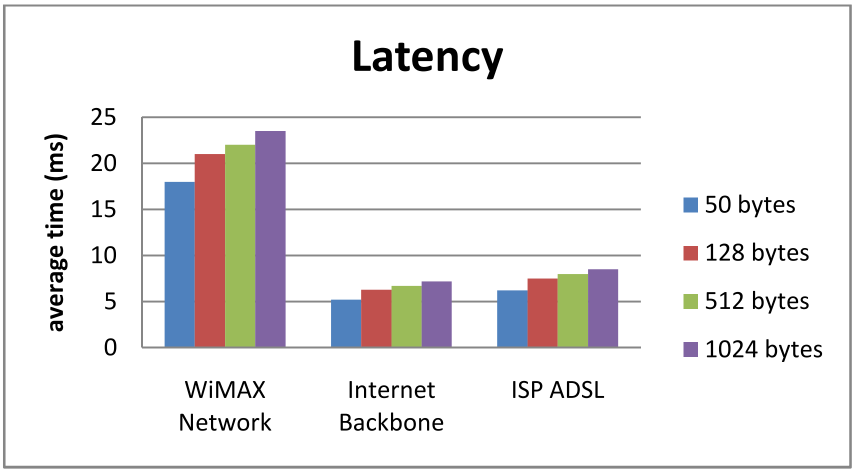

Figure 5.7 Latencies for WiMAX – ADSL (three clouds model)

As we can observe, latency is significantly higher in the WiMAX Network (which includes not only the radio link, but the path that starts at the customers PC and continues until the gateway provided by MPY BS). The end-to-end latency was usually between 25ms and 35ms in normal conditions, when using the ADSL connection, and a little bit smaller when using the optical fiber link, through FUNET (Figure 5.8).

[(49)] Explained in Chapter 3. Reference: 3GPP (December, 2008). TS 22.105 - V9.0.0 "*Technical specification group services and system aspects; Service aspects; Services and service capabilities*" (Release 9).

With hard rain conditions, we measured up to 65ms, but the average stayed below 45ms, usually. These ranges are acceptable for a videoconference communication, which can stand up to 150ms (WiMAX Forum[50] Recommendation) for a service in good-quality conditions from the user's point of view.

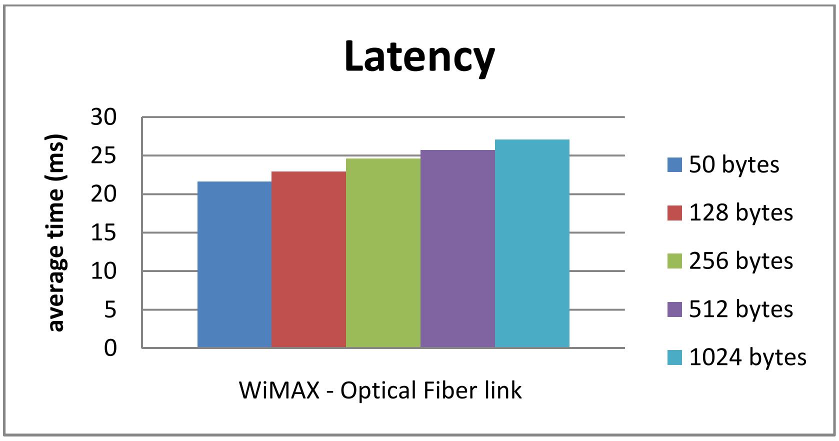

Figure. 5.8 Latencies for WiMAX – OF end-to-end connection

Once more, if we take a look at Figure 5.9 it is clear how much more stable was the wired section comparing to the wireless. We could predict higher jitter rates for the core network, based on the idea of possible congestion, or the packet switching effect. However the end-to-end path is not very long (about 12 hops), so we presume that the conditions of the core network are good enough to provide an acceptable QoS for the purpose of our study.

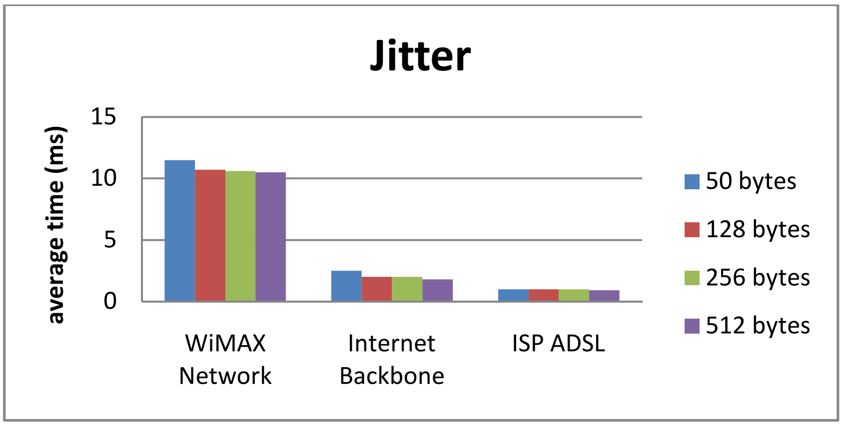

Figure 5.9 Jitter for WiMAX – ADSL (three clouds model)

[50] Explained in Chapter 3. Reference:WiMAX Forum,"*WiMAX System Evaluation Methodology - v2.1*" (July 2008).

In the graphic we can observe a slight difference in the jitter for packets of 50 bytes, however, we could measure generally very similar jitters for packets in the range 50 – 512 bytes, therefore we can predict similar results for the audio and video data used in a videoconference application, which use packets sizes in the same range.

Even though real-time communications are very sensitive as for delays and variations of inter-arrival times, it is possible for the applications to manage these issues using buffers. But there are always limits, and a jitter over 50ms end-to-end would make the communication very hard to handle in acceptable quality conditions.

5.5 Preliminary Conclusions

A purpose of this chapter was to understand what kind of information is going to be transmitted through the case network, and develop a testing plan according to the features of the network. After this, we could develop the tests on the Network Layer to evaluate the performance of the traffic that was modeled according to the features of videoconference traffic that we are supposed to transmit from the Application Layer.

If it is possible to analyze the behavior of this modeled traffic, then it is easier to predict what will happen when we apply WiMAX for a videoconference session, and assume the limits of what can be done and what cannot, according to certain parameters of quality in the communication. Due to the lack of information about the PHY and MAC layer of the WiMAX link, it is difficult to have definitive conclusions about the results obtained. However, we can say that the behavior of the link was good enough when it was tested in conditions of smooth rain, and absence of rain, as we obtained an average packet loss under 1% most of the time, which means it is possible to work under the specified recommended conditions for videoconference according to the Third Generation Partnership Project (3GPP, 2008) (51), but *quite much on the limit*, we could say, when we considered a rain over 10mm/hr(52).

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(51) Explained in Chapter 3. Reference: 3GPP (December, 2008). TS 22.105 - V9.0.0 "*Technical specification group services and system aspects; Service aspects; Services and service capabilities*" (Release 9).

(52) These results lead to a mandatory reflection about the boundaries for "negligible" signal attenuation levels, suggested by the ITU terrestrial model. Explained at footnote (40), page 54.

The same conclusion can be obtained attending the delays and variation of inter-arrival time of the packets. An average of 25 – 35ms delay was measured in normal conditions, which is far enough from the maximum recommended by the WiMAX Forum (150ms), but it could rise up to 65ms eventually. As for the jitter, we measured variations of less than 20ms the 99% of the time, but we found also bursts of more than 40ms sometimes (heavy rain conditions), during short periods of time.

These results suggest that the use of this WiMAX link for videoconference purposes is possible, but it might happen that the quality of the communication falls down to non-acceptable levels, when it is working under certain conditions of propagation. Therefore we can predict that it will be a challenge to ensure high performance for this access network the 100% of the time, with the current configuration, when it is working with videoconference class traffic, even though we do not have access to PHY and MAC parameters in order to be able to make a more reliable interpretation on the results obtained.

6. APPLYING WiMAX FOR VIDEOCONFERENCE

Currently, there are many applications we can use for videoconferencing, as they are deeply settled in the society. For instance, it is possible to have a video conference communication using the world wide distributed application *MSN Messenger*, or the so-known *Skype*, which is probably present in almost every personal computer nowadays (sometimes it is even pre-installed in the latest laptop distributions).

Skype uses a proprietary Internet telephony (VoIP) network. The protocol has not been made publicly available by Skype, and official applications using it are closed-source. The main difference between Skype and other VoIP networks is that Skype operates on a peer-to-peer model, rather than the more traditional server-client model. The Skype user directory is entirely decentralized and distributed among the nodes in the network, which means the network can scale very easily to large sizes (currently about 405 million users) without a complex and centralized infrastructure. Besides, it has the disadvantage of not being interoperable with other VoIP networks.

For the purpose of our study, however, none of these applications are useful as they do not offer monitoring possibilities, which we need to develop a deeper study of the quality of a video conference communication held over our case network. As we already introduced in Chapter 3, H.323 recommendation is widely extended among the vendors delivering video conferencing software. For that reason, after some research we selected the application *Policom PVX - trial version* - which is a quite complete monitoring tool using also H.323 recommendation. It is important to understand how it generates video and audio traffic, as well as what setup possibilities it offers, or the protocols it uses, therefore, we will develop a brief introduction about the application itself, and its features in the following section.

6.1 Polycom PVX: Videoconference Session Monitoring Tool

6.1.1 Introduction

Polycom is an American company founded in 1990, that established its headquarters in the state of California, USA. It provides products (hardware and software) specialized in remote audiovisual communications such as Voice and Video over IP, combined with data transferring. The Polycom PVX business-class video conferencing application is one of those products included in their software bunch, offering high-quality audio and video, based on H.323 recommendation. It is a remote visual communication tool intended for remote workers that do not have dedicated IT support, enabling them to have video conference communications over IP Networks.

6.1.2 Features

Polycom PVX is capable of delivering business-class video quality at VGA resolution, but it can be easily set up for optimal video quality resolution, depending on the user or network requirements. It conforms to International Telecommunications Union (ITU) H.264 video coding standard, which ensures industry-wide interoperability with other standard compliant manufacturers.

H.264 is a standard for video compression, also known as MPEG-4 part 10 or MPEG-4 AVC (Advanced Video Coding), developed by the ITU-T Video Coding Experts Group (VCEG) together with the ISO/IEC Moving Picture Experts Group (MPEG). It is a family of standards that covers all forms of digital compressed video from low bit-rate Internet streaming applications to HDTV broadcast and Digital Cinema applications with nearly lossless coding. It is a block-oriented codec, that uses the Block Motion Compensation (BMC) utility. This utility segments the frames into blocks of pixels, and each block is predicted using a block of equal size, from a *reference frame*. The blocks are not transformed in any way apart from being shifted to the position of the predicted block, and this shift is represented by a *motion vector*. The idea is to exploit the redundancy between neighboring block vectors, encoding only the difference between the current and the previous motion vector in the bit stream.

Initially, MPEG-4 AVC was focused towards low-quality video for videoconferencing and Internet-oriented applications, based on 8 bits per sample (pixel), with an orthogonal sampling of 4:2:0[53]. But this was not even enough to achieve a professional quality, which requires higher resolutions and therefore a sampling of at least 4:2:2. For that reason, it emerged the need to develop several extensions of the original codecs, supporting this demand. Among these extensions there were different possibilities supporting 4:2:2 and 4:4:4 sampling, and a range of 8 – 12 bits per pixel.

The blocks of pixels are not fixed-size, but variable. It is possible to find block sizes as large as 16x16 pixels, and as small as 4x4 pixels, enabling precise segmentation of moving regions (Variable Block-Size Motion Compensation). This way, for an Application using 12 bits per pixel, we can easily calculate a range of 24 to 384 bytes for video-generated traffic (which is quite within the range we used for the modeled traffic tested in Chapter 5). It is important to remark that with the use of H.264, bit rate savings of 50% or more are reported. This fact makes the standard very interesting if we are looking for optimization in the generation of traffic, consistent with bandwidth efficiency.

The audio codec used is G.722.1, a licensed loyalty-free ITU-T standard providing high quality, moderate bit rate *wideband audio* coding. It is a partial implementation of Siren 7 codec (providing 7 KHz audio, and bit rates of 16, 24, 32Kbps), developed by the PictureTel Corporation, now renamed as Polycom. Wideband audio is a technology used in telephony, which extends the frequency range of sound travelling over telephone lines, resulting in higher quality voice transmission. A new evolution used, Siren 14 codec, is capable of supporting 14KHz audio, but it is necessary to have a capable headset as well, in order to benefit from this feature. Polycom PVX supports calls up to 2Mbps, however, due to the features of the case network we never tested calls over 384kbps. It has a Video Error Concealment utility, which allows smoother video over IP networks, by concealing the deteriorating effects caused by the packet loss.

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[53] 4:2:0 refers to the sampling frequencies used for the format of color components YCbCr (luma, blue Chroma, red Chroma*)*, which is a variation of the classical image format RGB (Red,Green,Blue). The signal *Y* refers to the *luminance*, while Cb and *Cr,* referred to *chrominance,* represent differences B-Y and R-Y, respectively. The numbers indicate the relation between the frequencies used for sampling each component, yet the “0”, in this case, does not mean that there is no *Cr* signal, but that Cb and *Cr* samples do alternate. The highest resolution is achieved with a 4:4:4 sampling. Reference: Iain E. G. Richardson, “*H.264 and MPEG-4 video compression…*”, pages 16 – 20.

Regarding security matters, it conforms to ITU H.235 version 3 standard for embedded encryption. It supports 128-bit encryption with extended Diffie-Hellman [54] key distribution.

6.2 Evaluation of a Videoconference Communication

Measurements on the Network Layer have given us a lead about the performance of our case network. The results obtained permit us to have a draft about what we will find when a videoconference communication is held over the network, as the degradation experimented by the data at that layer should be proportionally transmitted to the upper layers and therefore, to the Application Layer, which is the one handled directly by the end user (video conferencing software).

Before we introduce ourselves in the analysis and evaluation of a video-conference communication, it may be useful to develop a brief introduction about the equipment used for the experiments, and the setup of the monitoring tool.

6.2.1 Testing Environment

As we already introduced at the beginning of this chapter, Polycom PVX was the application selected to test and monitor a videoconference communication over the case network. The computers used worked under Microsoft Windows XP - SP3, behind a NAT and a Firewall. It was necessary, therefore, to specify certain parameters of work at the setup of the application, in order to allow it to know the real IP's assigned to the equipments, as well as to permit the flow of data through certain UDP and TCP ports.

The processor for both PC's was an Intel Pentium 4, working at 3 GHz and supported by 2GB of RAM, which was enough to work fluently with real-time high quality videoconference. The devices in charge of capturing the sound and the image from

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[54] Diffie-Hellman (key agreement) is a cryptographic protocol that allows to establish a shared secret key between two parties that have not been in contact before, over an open channel. Reference: Alfred J. Menezes, Paul C. Van Oorschot, Scott A. Vanstone; *"Handbook of applied cryptography"*, page 515.

the room were two Logitech QuickCam webcams, provided with a high-quality VGA sensor, and an embedded microphone with Rightsound™ technology.In order to contrast the results obtained from the tests developed over the Network Layer (Chapter 5), we tried to develop the experiments applying for videoconference under similar atmospheric conditions. Sometimes the experiments were carried out during the same day (basically those ones measuring the possible influence of the rain), and sometimes on different days.

6.2.2. Setup

Polycom PVX is a very intuitive tool, and therefore fairly easy to set. The first thing we need to take care of is the NAT, as Polycom needs the real IP's in order to begin a communication between remote users. Figures 6.1 and 6.2 show the setup window where it is possible to specify the environment where each PC is working, as well as the ports we need to open in the firewall of our OS (Operating System).

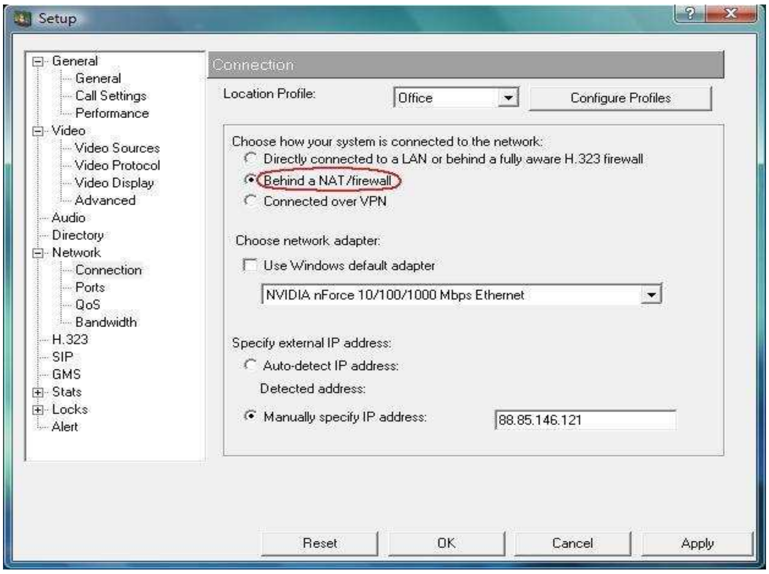

Figure 6.1 Setup to work behind a NAT and/or Firewall

As we can see in Figure 6.2, under *Ports* (in blue) there is an option for *QoS*. If we click on it, it pops-up an option to enable QoS specification within the data sent to the network. This means that in case the network has QoS capabilities, our videoconference traffic will be prioritized. But in case that some sections do not have QoS, it could have a remarkable effect on the videoconference communication performance.

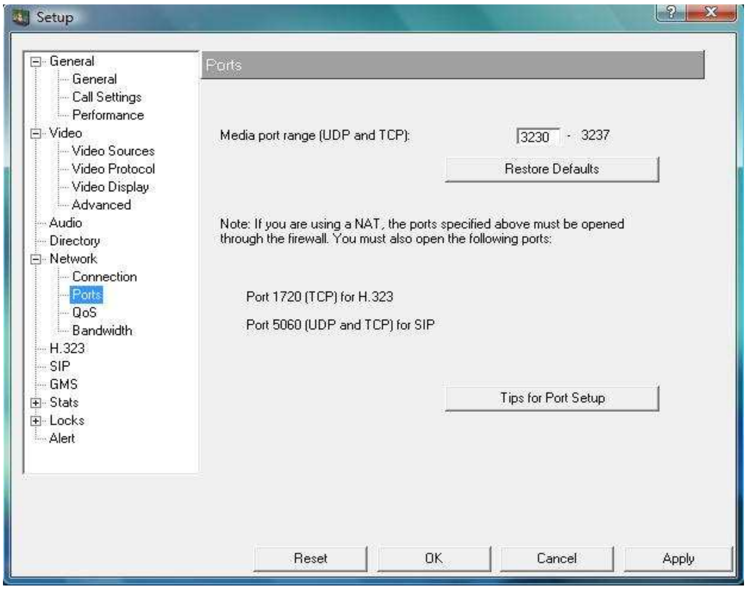

Figure 6.2 UDP and TCP Ports to open in the Firewall

If we continue downwards through the menu, we find the *Bandwidth* configuration, where we can select the range of work for the Call Rate, defining a maximum (the top is 2Mbps) and a default value. This range is later selectable in the main calling window, as we can see in Figure 6.3.

Figure 6.3 Call Rate selection

Regarding the video and the audio, it is possible to enable VGA resolution and define a video quality between *smoother motion* or *sharper image*, as well as different possibilities for *Echo cancellation*. If we click over *Video Protocol*, in the *Video* menu, we find three interesting options: a) Enable People+Content b) Enable H.239 content c) Enhanced video (H.264) status.

People+Content™ is a proprietary functionality of Polycom that captures the desktop at native resolution with *any* aspect ratio, and automatically converts it to an appropriate resolution for the remote participants. Besides, H.239 protocol describes the use of a double frame in videoconferencing, one of them usually for real-time video (audio & video multiplexed), and the other for presentation. A traditional video conference has an audio channel, a video channel and an optional data channel. H.239 defines rules and messages in order to establish an additional video/graphics channel, often to transmit a PC graphics presentation or video from a document camera, while still transmitting the videoconference traffic.

6.2.3 Analysis and Evaluation

According to the results obtained in Chapter 5, we have preliminary conclusions about the performance of our case network. We could observe an increasing degradation of the data transferred, as we tested the case network under heavier intensities of rain. The aim of this chapter may seem obvious to the reader, yet absolutely necessary for the completion of the study about the performance of this Fixed WiMAX system, in particular. It is necessary to remark, that we do not have the intention to extend the results and conclusions obtained from this case study, to other systems or networks, as they are completely dependent on particular and concrete conditions and circumstances. However it is intended to be a useful document, as it represents a study that has been developed over a *real* WiMAX installation.

At this point, we found it necessary to divide the analysis and evaluation of the communication in three differentiated parts:

I. Network performance
II. Human perception: quality of Image & Sound
III. The wired sections

This way we were able to compare the empirical results (part I) with our own direct perception in real time (part II), in order to corroborate (or not) the quality boundaries established by the different standard organizations (WiMAX Forum, 3GPP, etc). In addition, we could isolate the wireless network measuring the performance of the wired sections along the path ADSL – OF in part III, which helped us to have a better idea about the role of the WiMAX Network as for the data degradation.

I. Network Performance

In Chapter 4, we talked about the upload/download speeds for the three access networks, being the ADSL link the slowest one with an upload of approximately 375Kbps. This fact conditioned our experiments in a significant way, as we could not dial call rates higher than 350Kbps in order not to lose information due to the lack of bandwidth (we do not know the nominal value), rather than other factors. However, fortunately we could increase the call rates up to 384Kbps when we issued a call between the PC using the WiMAX access point and the PC using the Optical Fiber link.

The network factors are now quantified in terms of the overall end-to-end delay, jitter and packet loss. However, it is hard to clearly distinguish the device factors (55) from network factors even though we tried to set the PC's the best way in order not to have an additional bottleneck in our communications.

The calls lasted less than five minutes always, according to the *trial* nature of the Application used. However, it is not important as we are not looking for detailed averages here; at this stage, real-time monitoring is more valuable, if we consider reliable the results obtained in Chapter 5, where a detailed and modular study was made in order to clarify the performance of the different parts of the case network.

According to the BDTI Solution Benchmark™ for H.264 Decoders, a video device providing VGA (640x480) resolution at 30fps, may need a maximum bandwidth of 768kbps (56). Due to the limitations of our case network, we only tested QVGA video (320x240) at a maximum call rate of 384Kbps. Figures 6.4 and 6.5 show statistics of

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(55) Devices such as H.323 end-points, routers, firewalls, Network Address Translators (NATs), etc, may condition significantly the quality of the Videoconference.

(56) Parameters fully defined in BDTI's *proprietary* bit streams < http://www.bdti.com/products/sc_h264.htm>

a 384Kbps WiMAX – OF call in normal conditions, that is, without any kind of precipitation. The pictures were not taken exactly at the same time, but still very close.

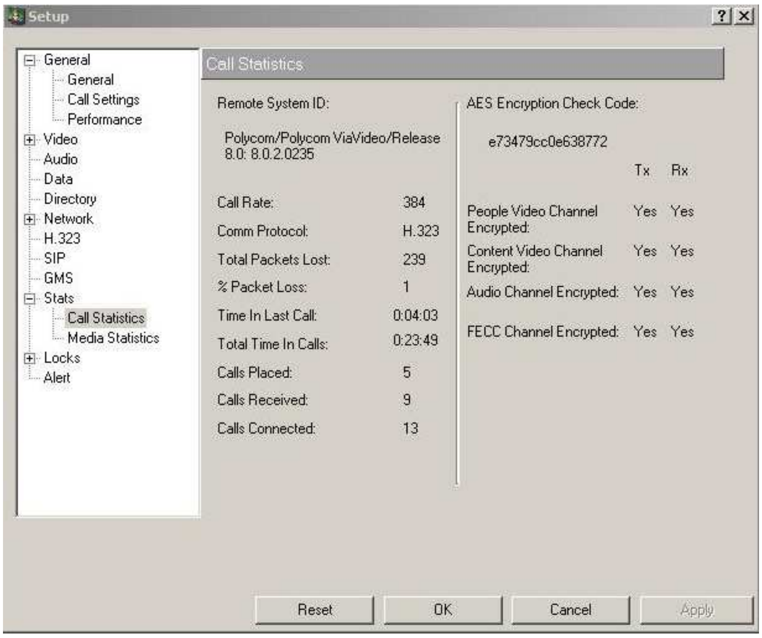

Figure 6.4 Call statistics for WiMAX – OF at 384Kbps

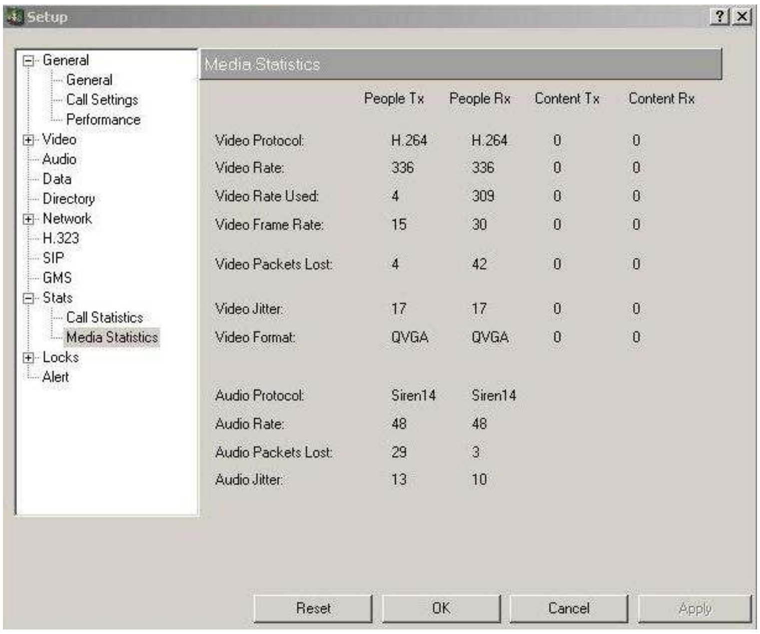

Figure 6.5 Media statistics for WiMAX – OF at 384Kbps

As we can observe from Figure 6.4, at that time there was a packet loss of around 1% which was not a fixed value, but alternating regularly with a value of 0%. That was an acceptable average, attending the particular requirements of a business-quality videoconference communication. In addition, it is remarkable the fact that the option for additional video quality provided by the People+Content™ utility was on, and therefore made it even more challenging to have an efficient communication.

With the current configuration, it was generated a QVGA resolution (320 x 240) video, with frame rates oscillating between 15 and 30 frames per second (Figure 6.5), and variable video rates up to a maximum of 336Kbps. As for the audio, it was delivered a fixed rate of 48Kbps using the Siren 14 codec, capable of supporting 14KHz audio, as we already mentioned. Both video and audio experienced a fairly constant jitter; about 17ms for the video and between 10 – 15 ms for the audio, which is quite within the range measured in Chapter 5, at the Network Layer.

Regarding the ADSL, we tried different call rates always below 350Kbps. We could observe a nice balance between the bit rate and the data degradation when we attempted a 256Kbps call rate. A reason for this could be the lack of stability at the speed provided by the ISP, despite the fact that it was supposed to deliver more than 300Kbps upload. In these conditions, we could observe similar results as for packet loss and jitter, comparing to the OF link.

Following the criteria developed in Chapter 5, we tried to capture instant values under conditions of light and heavy rain, and our expectations were widely covered, as we are going to explain now. First of all, it is necessary to remark that the values obtained for packet loss were slightly higher than those obtained after long sessions of traffic tradeoffs experienced during the experiments carried out for the Network Layer. This means that although the averages obtained for the Network Layer were, in general, acceptable, we found frequently bursts of data degradation when we attempted a real videoconference communication under rain conditions. For instance, with a moderate rain between 10 – 20mm/hr we found bursts of packet loss around 2% quite often. But it could even reach up to 3% easily (Figure 6.6 shows a snapshot), with a heavier rain, between 20 – 30mm/hr. These values were not constant at all but showed up frequently, and they revealed a significant weakness from our case network, as a fluent videoconference could not be guaranteed. We will describe it from the human perception's point of view in part II.

We measured even bursts of packet loss up to 14%, however, this value leads us to consider a possible occasional variability at the speed provided by the ISPs, or the Internet backbone, performing as a bottleneck from time to time.

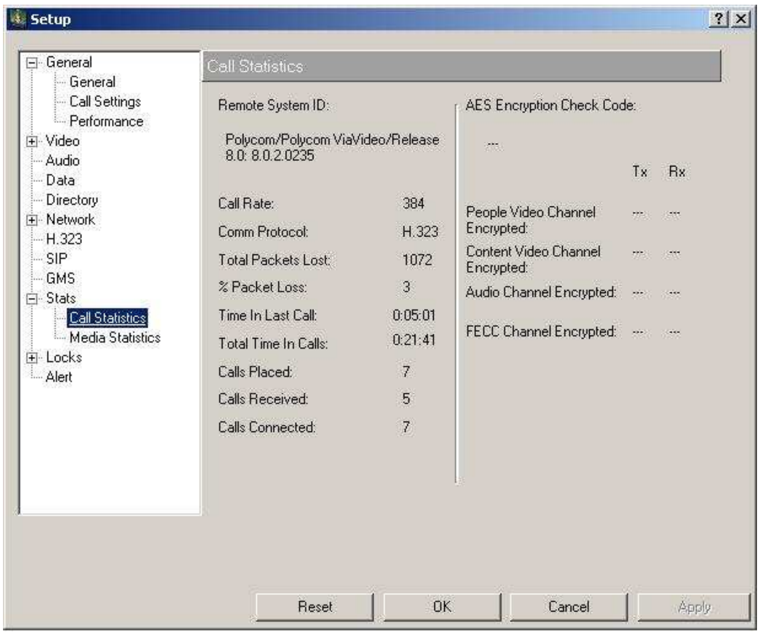

Figure 6.6 Call statistics for WiMAX – OF at 384Kbps in rain conditions

Once more, it is absolutely necessary to insist on the fact that we are not able to assure the effect caused by the rain, since we do not have access to the PHY layer of the radio link. However, it seems interesting, again, to observe the connection between the rain and the data degradation when we attempted experiments with different tools, on different layers, but under similar testing conditions.

II. Human Perception: Quality of Image & Sound

The aim of this second part is to establish a relation between the empirical results and the human perception’s point of view. Attending the recommendations of the WiMAX Forum and the 3GPP, a videoconference should not have a packet loss over 1% for an acceptable business-quality communication. Losses over this theoretical boundary may cause undesirable effects such as the loss of audio, freezing images or even the end of the communication.

Although delays, jitter and BER are all key factors affecting the quality of a communication, it is fair to admit the capital importance that PLR has in a real-time communication, where the user's information is not re-transmitted in case of loss. Even though ARQ is not used in this particular case, FEC [57] acts as a smoother for the BER received at the radio link, thanks to the RS code and the CC that WiMAX implements for the PHY layer.

After monitoring the videoconference traffic, we are about to analyze what happened to the quality of the video and the audio received, from the point of view of our own perception.

Figure 6.7 Quality of image for WiMAX – OF at 384Kbps

As we can observe in Figure 6.7, the quality of the image received when we used a call rate of 384Kbps, and QVGA (320x240) video-generated traffic at 30fps over the path WiMAX – OF, was acceptable. The testing conditions were normal (without rain).

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[57] The concatenated FEC is based on the serial concatenation of a Reed-Solomon outer code and a rate-compatible TCM inner code. Block interleaving between the outer and inner encoders is optional. The inner code is a rate-compatible pragmatic TCM code, derived from a rate 1/2 constraint length K = 7, binary convolutional code. Reference: IEEE 802.16™ - 2004. "*IEEE Standard for Local and metropolitan area networks. Part 16: Air Interface for Fixed Broadband Wireless Access Systems*", pages 357-358.

In addition to the quality of the video, frame by frame, we can confirm the quality of the motion image, without cuts or jumps, and fairly fluent. In the case of a communication using the ADSL end-point, we had very similar results, however, we assumed the path WiMAX – OF was the most suitable for our testing purposes as we were able to dial 384Kbps call rates, perfectly (considered as a typical rate for business-quality videoconference communications).

Regarding the audio, both paths (through OF/ADSL) presented satisfactory results, as the reception of the voice was clear and uninterrupted. Yet, we noticed a light noise signal constant in time, but presumably as a consequence of the equipment used at the end-points, rather than the network itself.

Figure 6.8 and 6.9 show the quality of the image under rain conditions, moderate to strong in the first case (15 – 25 mm/hr), and fairly strong in the second example.

Figure 6.8 Quality of image for WiMAX – OF in rain conditions

Now it is clear how pixelated the image is, and the loss of sharpness it is suffering. The image is not well-composed anymore, and there are parts we cannot distinguish well, as a consequence of the information lost along the path. As for the quality of the motion image, it was not acceptable for professional requirements, since the image was frozen at intervals, and it looked fragmented sometimes, mixing parts of different frames. On the contrary, the audio stayed in acceptable quality most of the time, which leads us to think about a possible influence of the variable packet size used in video-generated traffic, as it may have a more pronounced effect in the jitter, and therefore can suffer additional packet loss, comparing to the audio, which uses fixed size packets.

The snapshot shown in Figure 6.9 is just anecdotic, but still interesting. It was taken during one of the most rainy days (first half of July, 2009) among all we had in the course of our experiments.

Figure 6.9 Severe degradation of the videoconference.

This was not usual, yet we had the opportunity to experience the perception of a seriously damaged videoconference communication. In this case, the communication was hardly sustainable, the image was clearly damaged, in every sense, and the audio suffered cuts and losses that made a conversation not feasible. There are so many factors that can affect the overall end-to-end communication, that we do not dare to point at any section as responsible. As we already announced, this situation was anecdotic, hence we assume it was caused by a possible general failure, rather than a single, concrete part of the end-to-end path.

The experiments tested so far have a section present all the time: the radio access network. Unlike the tests carried out on the Network Layer, it was not possible to apply for videoconference separately, section by section, as we worked with the modeled traffic in Chapter 5. This was a great disadvantage considering that we were not able to analyze our case network in a modular way, in order to clarify the performance of every section, individually. However there was still something we could do in order to get a better view about the responsibility on the part of the wired sections, as for the data degradation: to evaluate the performance of a videoconference between the ADSL and the OF end-points.

Being the WiMAX Network the core of our study, it was very important to clarify, as much as possible, the differences in performance between this section and the rest of the network. For this reason, we decided to add a third part in the analysis and evaluation of our case network, which considers as well the points of view of parts I and II, but attending only wired sections. Figure 6.10 shows the scheme of the path tested.

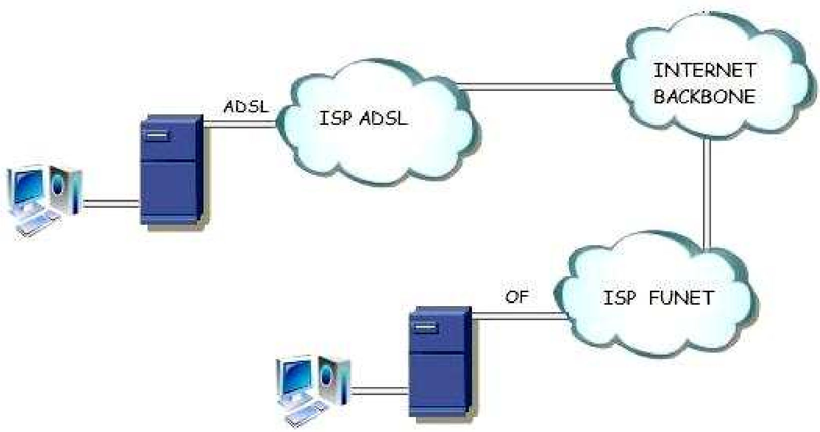

Figure 6.10 ADSL – OF path

In Chapter 5 we could already evaluate the performance of the wired sections, when we introduced a modeled traffic, according to the features of video and audio generated traffic. The wired sections showed then a nice reliability and a better performance comparing to the wireless section. For that reason, we decided not to separate the *network performance* (part I) and the *human perception* (part II) in part III, predicting a presumable acceptable behavior, according to the previous results obtained in Chapter 5.

III. The Wired Sections

The aim of this final experiment was to test the path shown in Figure 6.10 exactly at the same time, and same testing conditions as those experiments using the WiMAX Network that showed a severe degradation of the video conference, presenting a packet loss higher than 2%. It was the only way, with the evident lack of resources, to establish a division in responsibilities between the radio access network and the rest of the network. This way we had the opportunity to troubleshoot and point out to a possible origin of the degradation.

As the ADSL connection represented the main bottleneck in our case network, we dialed call rates between 128Kbps and 256Kbps, which are below the business-quality requirements, but allowed us to avoid packet loss due to the lack of bandwidth. As a consequence it showed more realistic results about packet loss and jitter, which was the aim of this experiment. In Figures 6.11 and 6.12, we can observe an example of the network performance transmitting at 256Kbps in the same testing conditions as the

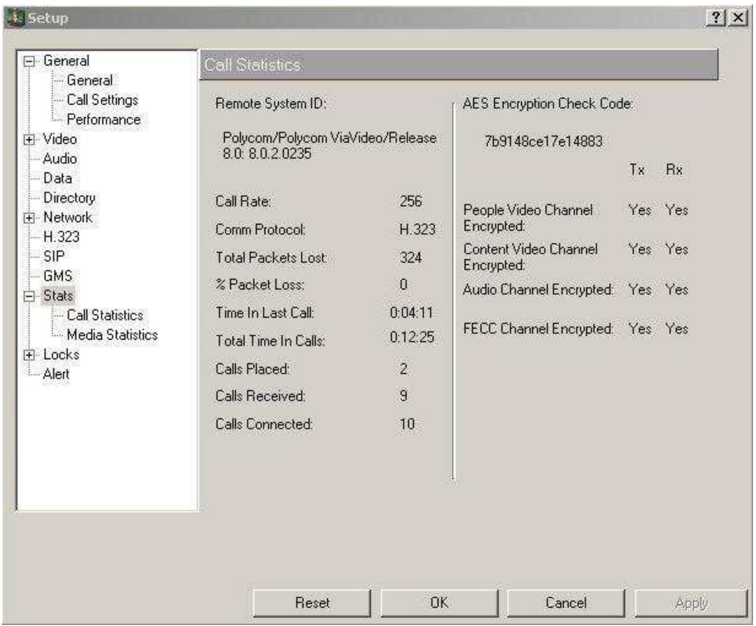

Figure 6.11 Call statistics for ADSL – OF at 256Kbps

experiment showed in Figure 6.6, which was experiencing a packet loss of 3%. In this case, the packet loss stayed at 0% all the time, and the jitter was fairly constant even though sometimes it increased up to 20ms. In addition, we could observe an acceptable quality of the image motion and sound at both end-points.

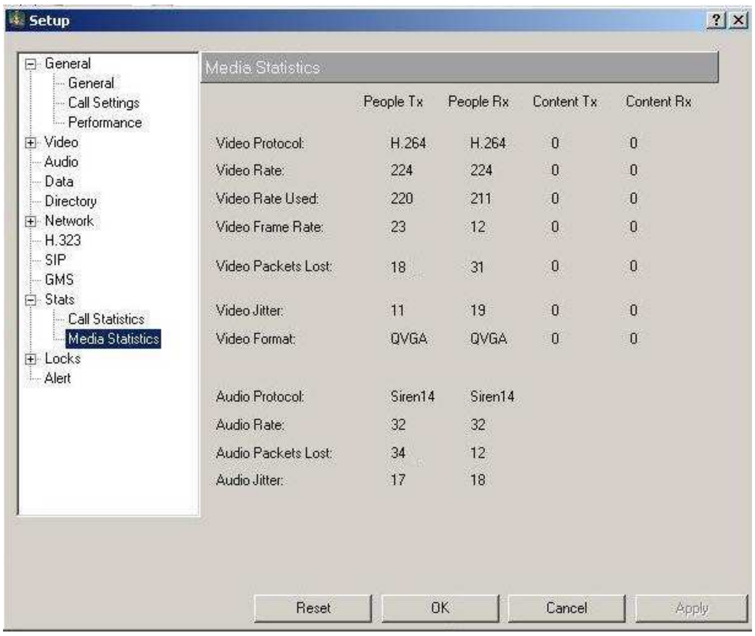

Figure 6.12 Media statistics for ADSL – OF at 256Kbps

We carried out this experiment several times, in different testing conditions, and we always obtained very similar results. Even though we could not test 384Kbps call rates, we could compare the quality of the communications to the cases where the WiMAX Network was involved, and the results pointed out to the WiMAX Network as the responsible for the lack of reliability [58].

With this experiment we are able to give more emphasis to some of the questions opened in Chapter 5, which suggested a main role of the WiMAX Network, as for the data degradation. However, it is still difficult to have definitive conclusions about the direct effect of the rain, even though we have experienced an undeniable connection with the degradation of the data transferred, along the different experiments we have carried out.

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[58] When we talk about the *lack of reliability* we refer to the concrete situation where the user information is being degraded in a level that is considered over the quality boundaries established by the WiMAX Forum and the 3GPP for a videoconference communication. But we do not intend to get the reader confused about the reliability of WiMAX Technology in a general way, as that is not the point of our discussion here.

6.3 Summary and Reflections

At this point, it is possible to have a better global view of the context where we are working. The purpose of separating the analysis and evaluation in three parts, is to link them afterwards and elaborate a final conclusion about the quality of our system, relating the results with the ones obtained in Chapter 5, but also comparing them to the theoretical quality boundaries established by different standards organizations (mainly the WiMAX Forum and 3GPP).

In part I, we talked about the network performance from an end-to-end point of view. Monitoring the calls allowed us to measure parameters like the jitter (separately for audio and video) or the percentage of packets lost. If we compare the results obtained here to those obtained in Chapter 5 (testing traffic modeled), we can easily observe that there is coherence between them, despite the fact that the values were slightly higher for videoconference traffic. In both cases we had satisfactory results when we tested the network in normal conditions, that is, without any kind of precipitation. We obtained acceptable values for the three main factors affecting the quality of the communication: jitter, latency and packet loss. If we compare these results with the quality viewed from our own perception (part II), that is, considering the quality of the frames, the motion image and the sound perceived by our own senses, we must admit that the global quality of the videoconference was within the boundaries established for business-class communications.

The same situation we had in rain conditions, as for the existing coherence between the different experiments developed. However, in this occasion we experienced higher quality degradation than what we expected, considering the preliminary results obtained from the tests carried out over the Network Layer. It is necessary to pay special attention at the quality boundaries, as there is a huge difference between having a packet loss of 1% and losing 3% of the information transmitted. If we compare the monitoring results to the quality of the image and sound received, we must admit that the global quality of the videoconference was not always within the boundaries of a business-quality communication, especially when the rain was fairly strong. It is fair, however, to remember that the quality of the sound received was good enough as to be accepted, most of the time. There were no cuts, no jumps, nor a significant amount of noise within the signal, but just some losses of synchronization at intervals.

Comparing the paths used for the different experiments (along the ADSL segment and the Optical Fiber Link) in parts I and II, we must say that in both cases we obtained similar results when we attempted appropriate call rates, attending the bandwidth available. It is very important to point out that we tried different call rates in order to discard possible bottleneck effects, which would distort the performance of our communications (packets dropped as a result of the lack of bandwidth).

The experiment based on the reliability of the wired sections (videoconference ADSL – OF) in part III supported the analysis developed on Chapter 5, where we suggested a major role by the WiMAX Network in terms of data degradation. However, here we have again the limitation imposed by the lack of information about the WiMAX link framework (frequency band, modulation, link budget).

The coincidences between the degradation of the communications and the existence of precipitations could be partially explained if the system were operating with 64QAM as the nominal modulation. We already explained in Chapter 5 that the effect of the rain may be particularly significant in a radio channel working with frequencies over 10GHz, however it is difficult to have a precise idea of the real overall effect it may cause when a frequency in the range 2 – 11GHz is working combined with 64QAM modulation, as for the packet loss in correlation with the bit error rate, in the particular case of a videoconference communication. In fact, we could say that the *real-time* nature of this kind of communications makes them the most difficult to keep in high quality performance if we consider, for instance, that they do not take benefit from ARQ mechanisms in this concrete case where the user's information is not sent again in case of loss, as it may happen with other kinds of traffic. Hence, the reliability is based mainly on the error correction mechanisms, that is, FEC in the case of WiMAX. This would also enforce somehow the idea of higher degradation for videoconference communications.

As we exposed in Chapter 2, WiMAX uses adaptive modulation managed by a scheduler, so it should change the nominal modulation to a lower-order modulation when the conditions of the channel are not optimal. This change could also explain the increment of packet loss, as a result of the decrease in the throughput (if it falls to bit rates lower than 384kbps, additional packets will be dropped in case we are using such a videoconference call rate). Unfortunately, we did not develop a deep study on the evolution of the throughput for the WiMAX Network, in the way we did for the traffic performance, so we cannot use this parameter in our considerations properly. However, there is something we can certify: the fact that the downlink channel was more likely to fail comparing to the uplink. We realized this by comparing the results (both percentages and quality of image & sound) obtained from the reception at ADSL/OF end-point (information sent by the uplink channel of WiMAX) and the results at the WiMAX end-point (information received by the downlink channel of WiMAX). This was very interesting as it led us to the idea that the uplink configuration was definitely more robust than the downlink. One reason for this could be the modulations used, for instance, if the uplink were using a low-order modulation comparing to the downlink. First, because it may not need to commute to a lower-order modulation in case of degradation of the channel conditions (in case of QPSK, there would be no chance to commute, anyway), being robust enough. Second, because in case that the downlink channel did not commute either, it would be clear that a higher-order modulation would suffer always more degradation than a lower-order one. This would also partially explain the difference in performances between the downlink and uplink channel, and we must admit that the possibility of different modulations used makes sense if we remember from Chapter 4 that WiMAX is using an asymmetric link 512k/1Mb.

The QoS could be also playing a major role as for the performance of the system. If we remember from Chapter 4, we suggested the possibility of having a certification profile Fixed WiMAX Release 1.0 – Wave 1, which would not include QoS. This could also affect the performance of the videoconference communication, but again, it is difficult for us to relate it with the results obtained from the tests.

Even though we are not able to assure the implication of the precipitations, at this point we may conclude that with the current configuration, our *Fixed* WiMAX System presents reliability problems in order to guarantee a *full-time* high quality communication. However, it is fair to remark that it is capable of managing a business-quality videoconference most of the time.

7. CONCLUSIONS

After seven months working in this Project, it is a satisfaction to achieve the final step. During this time, I have been able to deepen in one of the strongest emergent technologies matured in the last few years: WiMAX; but it was also an open door to enter the field of research in Telecommunication Engineering, as it was necessary to develop an extensive and meticulous research on various aspects related to the whole sense of this thesis, including issues about IP Networks, topologies, features and requirements of a radio link or the way in which videoconference traffic is generated.

It was necessary as well to develop a research on the latest software released, in order to perform the testing work in the best conditions possible, according to the intention of the Project, and the framework where we would carry out the tests. Now, we can say that the effort was worth and the results, satisfactory.

From Chapter 2 to 4, we can get an idea of the context and features of our particular case network; aspects related to WiMAX technology, and the Internet are presented and developed so that we have an introduction to the key aspects of our case network, always relating them to the purposes of our study, that is, to evaluate the performance of WiMAX when applied for videoconferencing. Chapters 5 and 6 conform the core of the Project, as they represent the application of the work plan engineered according to the information gathered regarding all the theoretical aspects related to the aim of the testing work.

We must say that the results obtained were satisfactory, and came to demonstrate the high reliability of WiMAX Technology, despite collateral issues it may have to deal with (configuration implemented, for instance). It is fair to point out that the data tested was probably the most delicate, considering that the human perception is involved in several ways, not only as for the speed in the transfer (resulting in a certain delay), but as a matter of quality, from a sensorial point of view, in the information received. WiMAX, itself, is not a videoconference solution, but it allows going forward and improving the way of developing it, as a powerful and reliable wireless service for global meetings, job interviews or promotional product broadcast, for example.

As a “data & voice” network, WiMAX is potentially more efficient than the current cellular infrastructure, and for that reason it is starting to be used as cellular backhaul. Some companies have already released *Mobile* WiMAX, which permits convergent wireless broadband services just having an IP address in the service network. We can say that among all industries included in the field of Telecommunications, mobile telephony service is very well positioned in order to take advantage of WiMAX networks, due to its competitive costs and the efficiency of the services delivered.

On the other hand, it is still to evaluate what effect and acceptance will have the next 4G network deployments, with the LTE (Long Term Evolution) technology on the front, promising data rates over 100Mbps DL. Next WiMAX– enabled mobile equipments will also support 4G and 3G connectivity, just as CDMA mobile phones, which currently switch between 1xRTT (2.5G) and 1xEV-DO (3G).

Some companies have stated their intention to deliver services at between 2 and 4 Mbps average to customers with *Mobile* WiMAX, similar to HSPA+ and LTE. So in the one metric that really matters, that is the end-user experience, all three technologies will be much of a muchness. The battle between WiMAX, HSPA+ and LTE is assured, we can say.

The importance of WiMAX as a global phenomenon in a medium term is still not guaranteed. Nevertheless, it is a fact that since 2006 until now the number of users has increased exponentially, mainly for accesses of less than 300kbps, which comes to announce a possible consolidation of this technology in the global market.

REFERENCES

[1] Advanced architecture for inter-domain quality of service monitoring, modelling and visualisation (INTERMON) < http://www.ist-intermon.org/ >

[2] Alfred J. Menezes, Paul C. Van Oorschot, Scott A.Vanstone, "*Handbook of applied cryptography"*, pages 515-521; CRC Press (1997).

[3] BDTI: products for benchmarking and analysis <http://www.bdti.com/products/sc_h264.htm>

[4] David R. Lide, "*Handbook of chemistry and physics* " 85^{th} Edition, page 15/41; CRC (2004).

[5] F. Baumgartner, M. Scheidegger, T. Braun, "*Enhancing Discrete Event Network Simulators with Analytical Network Cloud Models*", First international workshop on Inter-domain performance and simulation, IPS 2003, Salzburg 20-21 February. < http://www.ist-intermon.org/ overview/publications.html >

[6] F. Raspall, J, Quittek, M. Brunner, M. Martin, "*Path-Coupled Configuration of Passive Measurements*", Inter-domain Performance and Simulation Workshop, Budapest, 2004. http://www.ist-intermon.org/overview/publications.html

[7] H.323 Forum < http://www.h323forum.org/ >

[8] HNS-23885,"*Mitigación de Aspectos que Afectan la Propagación en Ondas de Frecuencia Milimétrica en Sistemas Inalámbricos Punto-a-Multipunto de Banda Ancha* ", Hughes Network Systems, Inc. (2002) < http://www.hughes.com/ >

[9] I. Miloucheva, A. Nassri, A. Anzaloni, "*Automated analysis of network QoS parameters for Voice over IP applications*", Inter-domain Performance and Simulation Workshop, Budapest, March 2004. < http://www.ist-intermon.org/ overview/publications.html >

[10] Iain E. G. Richardson, "*H.264 and MPEG-4 video compression: video coding for next-generation multimedia*", John Wiley & Sons (2003).

[11] Institute of Electrical and Electronics Engineers < http://www.ieee.org/portal/site >

[12] IEEE 802.16^{TM} - 2004."*IEEE Standard for Local and metropolitan area networks.Part 16: Air Interface for Fixed Broadband Wireless Access Systems*". < http://standards.ieee.org/getieee802/ 802.16.html >

[13] IEEE 802.16.2^{TM} - 2004. "*IEEE Recommended Practice for Local and metropolitan area networks. Coexistence of Fixed Broadband Wireless Access Systems*". < http://standards.ieee.org/ getieee802/802.16.html >

[14] IEEE 802.16cc-99/13, ”*Comparison of Propagation Models”*. < http://www.ieee802.org/16/tge/ >

[15] IEEE C802.16e-04/351r1, “*Extended rtPS, for VoIP services*”. < http://www.ieee802.org/16/tge/ >

[16] International Telecommunication Union < http://www.itu.int/en/pages/default.aspx>

[17] ITU–R P.530 – 12 , “*Propagation data and prediction methods required for the design of terrestrial line-of-sight systems*”, (June 2007).

[18] ITU–T G.722.1, “*Low-complexity coding at 24 and 32 kbit/s for hands-free operation in systems with low frame loss*” – Including Annex C: 14kHz mode at 24, 32, 48kbit/s (May 2005).

[19] ITU–T G.723.1, ”*Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s*” (May2006).

[20]ITU–T H.235, ”*Security and encryption for H-series (H.323 and other H.245-based) multimedia terminals*” (August 2005).

[21] ITU–T H.264,”*Advanced video coding for generic audiovisual services”* (May 2003).

[22] ITU–T H.323, “*Packet-based multimedia communications systems*” (June 2006).

[23] Jeffrey G. Andrews, Arunabha Ghosh, Rias Muhamed, “*Fundamentals of WiMAX: Understanding Broadband Wireless Networking*”, Prentice Hall (2007).

[24] Johnathan Davidson, James Peters, “*Voice over IP Fundamentals*”, pages 120-179; Cisco Press (2000).

[25] Jörn Seger, ”*Modelling approach for VoIP Traffic Aggregations to Transferring Tele-Traffic Trunks in an QoS enabled IP-Backbone Environment*”, First international workshop on Inter-domain performance and simulation, IPS 2003, Salzburg 20-21 February. < http://www.ist-intermon.org/overview/publications.html >

[26] Local weather report for Mikkeli, Finland < http://www.weathercity.com/fi/mikkeli/ >

[27] Loutfi Nuaymi, ”*WiMAX: technology for broadband wireless access*”, Wiley & Sons (2007).

[28] PacketTrap Networks, Managed Service Provider of Software for Network monitoring and management. < http://www.packettrap.com/ >

[29] Polycom solutions for Telepresence, Voice and Videoconference.
http://www.polycom.com/

[30] Robert K. Crane, "*Electromagnetic wave propagation through rain*", Wiley-Interscience (1996).

[31] Roger L. Freeman, "Radio system design for telecommunications" 3rd Edition, Wiley & Sons (2007).

[32] Salah Aidarous, Thomas Plevyak, "*Managing IP Networks: challenges and opportunities*", pages 1-10, 57-78, 167-175; IEEE Press series on network management (2003).

[33] Sied Ahson, Mohammad Ilyas, "*WiMAX: Technologies, Performance Analysis, and QoS*", pages 97-116; CRC Press (2008).

[34] Stanford Linear Accelerator Center, Network Monitoring Tools
http://www.slac.stanford.edu/xorg/nmtf/nmtf-tools.html

[35] The IEEE 802.16 Working Group on Broadband Wireless Access Standards
http://www.ieee802.org/16/

[36] Vijay Madisetti, Douglas Bennet Williams "*The digital signal processing handbook*",pages 40/1-40/25, 51/1-51/63, 55/1-56/8; CRC Press Handbook published in cooperation with IEEE Press (1998).

[37] WiMAX Forum < http://www.wimaxforum.org/ >

[38] WiMAX Forum White Paper "*WiMAX, an efficient tool to bridge the digital divide*" (November 2005).

[39] WiMAX Forum White Paper "*The WiMAX Forum Certified™ Program for Fixed WiMAX ™*" (January 2007).

[40] WiMAX Forum White Paper "*Fixed, nomadic, portable and mobile applications for 802.16-2004 and 802.16e WiMAX networks*" (November 2005).

[41] WiMAX Forum White Paper "*WiMAX's technology for LOS and NLOS environments*" (August 2004).

[42] WiMAX Forum White Paper "*WiMAX System Evaluation Methodology - version 2.1*" (July 2008).

[43] WiMAX Forum White Paper "*WiMAX Applications for Utilities*" (October 2008).

[44] WiMAX Forum White Paper "*WiMAX: The business case for Fixed Wireless Access in Emerging countries*" (June 2005).

[45] 3GPP TS 22.105 - V9.0.0 (December, 2008) "*Technical specification group services and system aspects; Service aspects; Services and service capabilities*" (Release 9); the Third Generation Partnership Project. < http://www.3gpp.org/ftp/specs/html-info/22105.htm >