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Wireless technologies for isolated rural communities in developing countries based on cellular 3G femtocell deployments

D51

Technical requirements and evaluation of WiLD, WIMAX and VSAT for backhauling rural femtocells networks.

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Abstract:

The TUCAN3G project aims to provide rural users with 3G/4G coverage in remote regions. In order to do so, the access network must be connected to the operator's core network with a transport network. Three different wireless communications technologies are considered for rural transport networks in the TUCAN3G project: WiFi for Long-Distances (WiLD), WiMAX and satellite communications. WiLD is the adaptation to long distances of WiFi, which is a broadband technology in which stations contend to access the medium. Its interest is due to its low price and low power consumption, but several limitations must be visited before considering it for backhaul of the femtocell network: the link capacity drops with the distance, and the QoS support is limited to traffic differentiation. WiMAX may offer a parametric QoS support and a high capacity almost independent with the distance, but power consumption and price may be limiting factors. VSAT links have advantages as global coverage, short time of deployment, etc., and disadvantages related to delay and cost. A transport network may be based on either of the three technologies proposed, or even on a combination of two or the three of them. But the choice must be based on objective criteria, that is, one should be able to compare them in terms of performance and cost. Finally, it is clearly established under what conditions and limitations WiLD, WiMAX and VSAT can be used in the backhaul with femtocells.

Keyword list: transport network, rural communications, WiFi, WiLD, WiMAX, VSAT, satellite communications.

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15 Jul 2013	V0.2	María Gómez y Enrique Gil	New contents related to satellite communications.
25 Jul 2013	V0.3	Esteban Municio	Introduction of simulation results for sections regarding 802.11
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31 Jul 2013	V0.5	Javier Simó	Addition of theoretical contents about 802.11 and first results extracted from analytical models
26 Ago 2013	V0.6	Esteban Municio	Addition of WiMAX theoretical contents and experimental results with AirMAX and Mikrotik NV2
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23 Sep 2013	V0.12	Esteban Municio	New results about the comparison between WiLD and WiMAX
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Executive Summary

The TUCAN3G project aims to provide rural users with 3G/4G coverage in remote regions. In order to do so, the access network must be connected to the operator's core network with a transport network. Three different wireless communications technologies are considered for rural transport networks in the TUCAN3G project: WiFi for Long-Distances (WiLD), WiMAX and satellite communications.

WiLD is the adaptation to long distances of WiFi, which is a broadband technology in which stations contend to access the medium. Its interest is due to its low price and low power consumption, but several limitations must be visited before considering it for backhaul of the femtocell network: the link capacity drops with the distance, and the QoS support is limited to traffic differentiation. WiMAX may offer a parametric QoS support and a high capacity almost independent with the distance, but power consumption and price may be limiting factors. VSAT links have advantages as global coverage, short time of deployment, etc., and disadvantages related to delay and cost.

A transport network may be based on either of the three technologies proposed, or even on a combination of two or the three of them. But the choice must be based on objective criteria, that is, one should be able to compare them in terms of performance and cost.

IEEE 802.11 is a family of standards born in 1999. After several amendments, a second full revision was published in 2007 and a third revision was published in 2012. Several operation modes are described in the standard that differ from each other in the physical layer, in the medium access control, or in both: 802.11a (5 GHz band, OFDM PHY, DCF MAC), 802.11b (2.4 GHz band, DSSS PHY, DCF MAC), 802.11g (2.4 GHz band, DSSS/OFDM PHY, DCF MAC), 802.11e (permits to incorporate EDCA instead of DCF in the MAC, in order to support QoS) and 802.11n (several improvements in the PHY such as channel aggregation, frame aggregation, spatial diversity, etc. in both 2.4 GHz and 5 GHz). However, these standards are designed for short distances. This document analyses the performance that any of those operation modes may offer in long links.

The industry frequently offers long-distance WiFi systems that incorporate an alternate free-of-contention non-standard MAC (e.g. AirMAX, from Ubiquiti, that emulates the WiMAX TDMA on top of a WiFi PHY). A few of those alternatives are visited and experimentally compared with the standard solutions.

IEEE 802.16, often known as WiMAX, is a family of standards inaugurated in 2004, with a full revision in 2009. Some amendments also appeared during the last 4 years but there is nothing relevant to this document that is not contained in the IEEE 802.16-2009 standard. The WiMAX Forum certifies 802.16-compliant systems in several certification profiles including 2.3/2.5 GHz bands for mobile systems and 3.5/4.9 GHz bands for fix systems. The standard also considers and describes the use of the 5GHz non-licensed bands (WirelessHUMAN PHY) but there is not a certification profile for such option. However, within this document we will use the familiar term “WiMAX” for the systems designed for both licensed and non-licensed bands. WiMAX will is also evaluated and compared with standard and non-standard WiFi systems.

Finally, the alternative globally referred to as VSAT include several different technologies for satellite communications. These solutions cannot be compared with the previous ones in terms of performance or cost, but they become necessary when the distance from the access network to the operator's core network is excessive for the previous alternatives.

The technologies are firstly studied in order to detect distance limits based on the revision of the standards. The state of the art is reviewed in order to find any analytical models or methods that permit to find out what the performance of any of these technologies would be for a given scenario. Then, a simulation platform is prepared, validated and used (based on NS-3) for comparison with theoretical expectations. In some cases, laboratory tests are also performed when other alternatives are not available.

WiFi is found to be applicable to long distances, with good performance in PtP links, especially when 802.11n is used. There are also proprietary solutions based on WiFi hardware that are shown to give very high performance over long links. WiMAX is also proved to be a valid solution, better in terms of QoS support though more restricted in the maximum capacity. Finally, satellite systems are a fundamental complementary technology to be able to bring connectivity to fairly any point of the planet no matter what the distance to the closest urban area is.

DISCLAIMER

Table of Contents

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List of abbreviations & symbols

ARQ	Automatic Repeat Request
ATM	Asynchronous Transfer Mode
BPSK	Binary Phase-Shift Keying
BER	Bit Error Rate
CP	Cyclic Prefix
CSI	Channel State Information
DL	Downlink
FER	Frame error rate
GI	Guard Interval
GW	Gateway
H-ARQ	Hybrid-ARQ
ICT	Information and Communications Technologies
IP	Internet Protocol
ITM	Irregular Terrain Model
ITU	International Telecommunications Union
LOS	Line of Sight
LTE	Long Term Evolution
MAC	Medium Access Control
MIMO	Multiple Input Multiple Output
MS	Mobile Station
NTP	Network Time Protocol
OFDM	Orthogonal Frequency Division Multiplexing
OPEX	Operating Expenses
PER	Packet Error Rate
PoE	Power over Ethernet
PTN	Public Telecom Network
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature Phase-Shift Keying
RNC	Radio Network Controllers
RSSI	Received Signal Strength Indication
SINR	Signal to Interference plus Noise Ratio
SISO	Simple Input Simple Output
SNR	Signal to Noise Ratio
TTI	Transmission Time Interval
UE	User Equipment
UL	Uplink
VSAT	Very Small Aperture Terminals
WILD	WiFi Long Distance
WiMAX	Worldwide Interoperability for Microwave Access
XPD	Cross-Polar Discriminator

1 SCOPE OF THIS TECNICAL REPORT

1.1 The transport network in the TUCAN3G project

A telecommunication network is basically a set of communications devices that can communicate with each other by means of communication links. There is a huge variety of telecommunication networks depending on the types of nodes and links, and also depending on the network architecture. Most of the networks are a means for end users to access services or to communicate among them. Although there are networking strategies in which all the systems are owned by users (i.e. ad-hoc networks), most of telecommunications networks have three well defined parts that permit to provide services in an efficient and cost-effective fashion: access network, distribution network or backhaul, and core network. The access network is basically formed of the terminal devices and the base stations that permit them to access the network. The core network is the high performance infrastructure that permits the operator to efficiently connect any user to any other user or service. The backhaul is the infrastructure that permits to connect each base station to the core network. In most of places where an operator offers services, the backhaul for a base station can be a high capacity transmission line (coaxial or optical fiber), although sometimes physical infrastructure and wireless dedicated links are not available. Exceptionally, operators may install remote base stations in locations where neither a physical infrastructure is available nor the enormous distance permits to use conventional wireless links and satellite communications. However, satellite channels are very restrictive in terms of capacity and also very expensive, which makes this solution impractical for places sparsely populated in rural areas.

When a telecommunications operator considers the possibility of providing its services in a particular location, two cost figures are estimated in order to calculate the foreseen investment: CAPEX (capital expenditure) and OPEX (operational expenditure). On the other hand, the number of potential users and the estimated average service use per user permit to calculate the foreseen revenue. If and only if the foreseen revenue is clearly bigger than the required investment, then the operator might decide to deploy the infrastructure that will permit to provide services. That is mainly the reason why there are many places in rural areas where operators don't provide services: either the CAPEX would be exorbitant to deploy the required infrastructures for terrestrial backhaul, or the OPEX would be also exorbitant if high capacity satellite backhauls are regularly used.

In this project, other alternatives are going to be explored. If a satellite link is finally required because a set of base stations are extremely far from the core network, at least it will be much more cost-effective if a single satellite terminal is shared by several base stations, instead of installing a satellite station per base station. Other times may be that a terrestrial multi-hop network may be a practical solution that avoids the necessity of a satellite communications system. Hence, we are going to consider that a backhaul is no longer a dedicated link, but a transport network constituted by several links that can be shared. Furthermore, the use of non-licensed frequencies is generally impractical for backhauling because external interferences may seriously perturb the quality of communications, but in rural areas there may be many scenarios in which there are no possible interfering sources. Three technologies are going to be considered as alternatives: WiFi, WiMAX and satellite links. We are going to study how any of them, or a combination of them, can be an acceptable backhaul solution for rural areas. The final objective is to define a valid alternative for the backhaul that permits to go further and cheaper, thus allowing an operator to deploy services in regions where this was not possible with conventional solutions.

In summary, the technologies cited above are being studied as candidates for heterogeneous transport networks. Those networks may be made of the following elements:

- An optional satellite communication to connect a transport network to the operator's core network when the distance is excessive considering terrestrial solutions.
- A gateway node that connects the transport network to the operator's core network.
- Any number of wireless routers connected to neighbors through WiFi or WiMAX links. Some of these nodes may be edge nodes to which rural femtocells are connected.

1.2 Factors considered when comparing wireless technologies in the transport network.

1.2.1 Requirements and performance figures

The transport network under study takes the place of a backhaul link, an element that is usually a high capacity wired connection or a dedicated wireless point-to-point link operating in non-licensed frequencies. Hence, we need to clearly identify what performance parameters are relevant and what values those parameters must or must not achieve in order to perform the backhaul function correctly. This situation can be compared with the replacement of pure circuit switching in telephony by packet switching: the problem is not to obtain exactly the same quality that can define the dedicated circuit, but to define the minimum acceptable quality of service that the packet network must meet in order to be eligible to support the telephony service.

In the case of the backhaul of 3G/4G femtocells, the expected performance is related to the types of traffic to be transported and the intensity of each traffic flow:

- There is a signaling traffic flow in both directions between each femtocell and the femtocell controller that must be installed in the operator's core network.
- There is voice traffic in both directions for each active phone call. The characteristics of each traffic flow of this type depend on the codec used.
- Finally, there may be different kinds of data traffic flows depending on the services used.

In order to determine whether a given transport network is adequate as backhaul solution or not, two analyses must converge: on one hand, user and system generated traffic in the access network must be characterized and quantified. On the other hand, the performance of the transport network as perceived by the traffic exchanged between each femtocell and the core network must also be characterized. Finally, after comparing both results, a valid backhaul solution can be defined if the offered better that the required one.

Usually, any of the different traffic flows will have certain requirements in terms of several quality of service (QoS) parameters:

- Throughput: minimum (less cannot be accepted), average.
- Delay: maximum (longer delays are not accepted and packets must be then discarded), average (average delay that indicates an acceptable quality).
- Jitter: maximum and average.
- Packet loss (%): average.

The fact that each traffic flow may have different requirements; different characteristics should be defined to differentiate the traffic flows. Each packet has to receive a different treatment depending on the flow it belongs to.

As a consequence, we need to describe and quantify the capacity of each consideed technology to assure traffic class differentiation, and to guarentee the certain values of QoS parameters for each class. A future work will try to do a more complex analysis for the aggregation of several links used for different technologies in heterogeneous multihop networks.

1.2.2 Expected performance for backhauling one or more femtocells

Each femtocell generates flows for signaling, telephony and data communications; these are aggregated to other femtos' flows in the transport network. Hence, supported the traffic in the transport network is not a function of the number of femtos but to the number and nature of the active users at any time. The traffic offered by the active users connected to the different femtos equals the traffic generated by them. If an off-loading technique is being used, local traffic must be subtracted.

Thus, the performance requirements will be estimated with the number of users served and the intensity of traffic per user foreseen.

A general estimation provided by Telefónica del Perú for Peruvian 3G users based on their experience in urban areas is given in the following Tables 1 and 2:

- For voice traffic
  - unitary traffic in busy hour: 15 mEr
  - Acceptable blocking probability: 2%
  - throughput of a voice channel: 25 kbps
  - population with at least 1 cell phone: 53%
  - Itinerant population with a cell phone: 40% of permanent population
- For data traffic
  - Potential market: 5% of population
  - Peak throughput/user: 3 Mbps
  - Average throughput/user: 15 kbps (DL) + 5 kbps (UL)
  - Signalling: 20% of voice and data traffic

Population	Itinerancy	Cell Phones	Traffic (Erl)	Voice Ch.	Kbps
200	40	127.2	1.272	5	125
300	60	190.8	1.908	6	150
400	80	254.4	2.544	7	175
500	100	318	3.18	8	200
600	120	381.6	3.816	9	225
700	140	445.2	4.452	10	250
800	160	508.8	5.088	11	275
900	180	572.4	5.724	11	275
1000	200	636	6.36	12	300

Table 1: Voice traffic figures for Peruvian users.

		Download (Kbps)				Upload (Kbps)			
Population	Itinerancy	BW Peak	Users	BW Average	BW	BW Peak	Users	BW Average	BW
200	80	3072	14	210	3072	512	14	70	512
300	120	3072	21	315	3072	512	21	105	512
400	160	3072	28	420	3072	512	28	140	512
500	200	3072	35	525	3072	512	35	175	512
600	240	3072	42	630	3072	512	42	210	512
700	280	3072	49	735	3072	512	49	245	512
800	320	3072	56	840	3072	512	56	280	512
900	360	3072	63	945	3072	512	63	315	512
1000	400	3072	70	1050	3072	512	70	350	512

Table 2: Data traffic figures for Peruvian users.

Besides, the transport network may also be used for fixed Internet services of the institutions. For those cases, the following Table 3 will be considered per institution:

	Download (Kbps)			Upload (Kbps)		
Institutions	BW Peak	BW Average	BW	BW Peak	BW Average	BW
3	2048	600	2048	512	153.6	512
4	2048	800	2048	512	204.8	512
5	2048	1000	2048	512	256	512
6	2048	1200	2048	512	307.2	512
8	2048	1600	2048	512	409.6	512
10	2048	2000	2048	512	512	512
12	2048	2400	2458	512	614.4	614.4

Table 3: Traffic figures expected per institution.

In Tables 1, 2 and 3, the number of users is calculated as a percentage of the population. Then, the average bandwidth is calculated multiplying the average bandwidth per user times the number of users. Then, the offered bandwidth must be the maximum between the peak throughput per user and the average bandwidth.

Obviously, the quantitative adequacy of a transport network depends on the amount of generated traffic. If a link is used for backhaul of two or more femtocells, that link must offer enough capacity for the aggregation of their traffics. Regarding the delay, the end-to-end QoS experienced by communications between a femto and the outer world is the addition of several delays that happen all along the path. It is important to keep as low as possible the backhaul contribution to the overall delay. Our project proposes the use of multihop transport networks for the backhaul; hence, maximum one-hop delay that can be admitted depends on the specific topology (actually on the maximum number of hops).

From the qualitative perspective, the network must differentiate at least the signaling between voice and data traffic, and must assure different QoS parameters to each of those traffic classes. In the rest of the document, each of the proposed technologies will be separately studied and analyzed in order to clear out the expected performance and the achievement of certain QoS requirements.

Finally, it can be seen that the case for small villages in remote areas, targeted in TUCAN3G, may be serviced with almost the same infrastructure from one small village with less than 300 inhabitants up to two, three or four small villages totaling less than 1000 inhabitants. Even considering a few "institutions", a backhaul with 5 (Downlink) + 1 (Uplink) Mbps of capacity would be a typical minimal infrastructures for such scenarios. This reference will be taken later for comparing the different technologies.

2 STATE OF THE ART

2.1 IEEE 802.11: documentation from standardization bodies and scientific literature that may help to foresee the performance in long distance links

2.1.1 Brief descriptions of the standards

The first WiFi standards (IEEE 802.11a y IEEE 802.11b) [IEEE802.11-1999, IEEE802.11a-1999, IEEE802.11b-1999] were conceived for WLAN (Wireless Local Area Networks). The differential characteristics of these standards are:

- IEEE 802.11b: year 1999; frequency band: 2,4GHz; physical layer: DSSS (Direct Sequence Spread Spectrum). Bit rates can reach up to 11 Mbps with a real total capacity of less than 7.5 Mbps.
- IEEE 802.11a: year 1999; frequency band: 5GHz; physical layer: OFDM (Orthogonal Frequency Division Multiplexing). Bit rates can reach up to 54 Mbps with a real total capacity of less than 30 Mbps.
- IEEE 802.11g [IEEE802.11g-2003]: year 2003; frequency band: 2,4GHz; physical layer: OFDM; and DSSS compatible with 802.11b. Bit rates can reach up to 54 Mbps with a real total capacity of less than 32 Mbps.

Latter amendments improve the performance of these standards. The more relevants are:

- IEEE 802.11e: year 2005; this introduces traffic differentiation and prioritization.
- IEEE 802.11n: year 2010; this increases the real bit rate theoretically up to 300Mbps under optimal conditions.

All these standards in the 802.11 family were put together in a revised version of the IEEE 802.11 standard published in that year [IEEE 802.11-2012].

WiFi standards define two types of nodes: AP (Access Point) and STA (Stations); and two modes: infrastructure (the STAs only communicate through an AP that performs like a base station) and ad-hoc mode (a STAs can communicate directly with others STA). The optimal behavior is achieved in infrastructure mode. Many links in the transport network considered in this project are point-to-point links, in which WiFi can be used between an AP and a STA.

The WiFi standards are not going to be described here, the original standard documents are referenced for further reading. However the most relevant details are given in the following:

- In the 2.4 GHz band, channels are 22 MHz wide but channels are spaced 5 MHz. Hence, consecutive channels overlap. Three non-overlapping channels are: 1-6-11. On the contrary, on the 5 GHz band channels are 20 MHz wide and are spaced 20 MHz. Hence, adjacent channels don't overlap.
- The standard gives also 10 MHz and 5 MHz channels options. The consequence of reducing the bandwidth is twofold: (i) the sensitivity is improved because the received noise is reduced, hence improving the link budget, and (ii) the bit rate is reduce in the same proportion.
- The medium access control uses a mechanism named CSMA/CA (Carrier Sense Multiple Access with Collision Avoidance). This is a distributed mechanism in which each station uses a well-known strategy that minimizes the probability of transmitting at the same time with another station, but there is not a coordinator that determines when any given station may transmit. Collisions may occur and, in such cases, retransmissions are required. Hence, normal transmissions must be acknowledged so that the transmitting station verifies whether the transmission was successful, or it collided and a retransmission is required. This mechanism has

the great advantage of its simplicity but it is not efficient at all. That is why there is a big difference between the physical bit rate and the maximum available capacity in any WiFi network.

- There are several different coordination functions described in the standard (i.e. different ways the stations may coordinate with each other in order to reduce or eliminate the risk of collisions). However, some of these functions have never been implemented by the industry. Moreover, the first implemented coordination function that is widely available is DCF (Distributed Coordination Function) and it does not permit to differentiate traffic classes. Hence, only the EDCA (Enhanced Distributed Control Access) will be considered.
- EDCA (Enhanced Distributed Channel Access) defines four AC (Access Categories or traffic classes): Voice, Video, Best Effort and Background. This permits to give up to four different statistical priorities to up to four traffic classes. Other two mechanisms may be used to improve the performance and to give different opportunities to the AP: the TXOP (transmission opportunity), that permits to transmit more than one frame without contention when it is higher than zero, and Block-ACK, that permits to avoid a lot of inefficient acknowledgements.
- There is not any mechanism in WiFi that permits to ensure strict QoS support. This means that we cannot limit the maximum delay, the minimum throughput, or the maximum packet loss for a given traffic class. Only priorities can be established. Further QoS support needs complementary mechanisms at the IP layer and the assurance of being operating far from the saturation point. The network is said to be in saturation when a higher offered load does not drive to a higher throughput but to a higher number of collisions. Hence, the conditions to use WiFi systems in a network that guarantees QoS parameters are: (i) EDCA is available and correctly used, (ii) advanced traffic control may be performed at the IP layer, and (iii) the access network does not offer load to the transport network beyond the saturation point at any time.
- There is not any admission control mechanism in WiFi. Such a mechanism is foreseen but it is supposed to be implemented at a higher level.
- The traffic classification from higher layers is very important. The IP packets are marked by the generating application in accord to the DiffServ model (a 6 bits value in the Type of Service octet named DSCP). The 802.11e equipment must to establish the correspondence between the DiffServ values and the priorities defined in EDCA.
- IEEE 802.11n incorporates several improvements that increase the nominal bit rate to 600 Mbps under optimal conditions. The bandwidth may be increased to 40MHz, some parameters are optimized, and several mechanisms such as frame aggregation are introduced, etc. in order to foster the performance. But the most significant improvement of this standard is spatial diversity with MIMO (Multiple Input Multiple Output) transmission, that implies the possibility of spatial streams, beamforming, STBC (Space Time Block Coding) and spatial multiplexing; and the definition of 127 MCS (Modulation and Coding Schemes).

2.1.2 The PHY in the 802.11 family

Although there are several physical layer alternatives that have been defined in the 802.11 family and all of them are still in the 802.11-2012 release, some of them are obsolete. Only two kinds of alternative physical layers may be found in real products currently:

- Direct Sequence Spread Spectrum (DSSS): it was defined in 802.11b and adopted by 802.11g for compatibility reasons. The nominal bitrates are 1, 2, 5.5 and 11 Mbps.
- Orthogonal Frequency-Division Multiplexing (OFDM): it was defined in 802.11a for the 5 GHz band, adopted by 802.11g for the 2.4 GHz band, and extended by 802.11n for both frequency bands. The nominal bitrates are 6, 9, 12, 18, 24, 36, 48 and 54 Mbps in 802.11a and 802.11g, and higher bitrates are possible in 802.11n by applying several improvements like MIMO, or 40 MHz channels instead of 20 MHz channels in the former standards.

- 802.11n introduces the HT PHY, with is essentially a version of OFDM PHY with several improvements that permit to increase the maximum throughput up to ten times.

2.1.2.1 Direct Sequence Spread Spectrum (DSSS)

Direct Sequence Spread Spectrum (DSSS) is a Code Division Multiple Access technique also known as Direct Sequence CDMA (DS-CDMA). As a CDMA technique with multiple transmitters, it can communicate simulataneously with multiple transmitters over a single communication channel. To be able to do so without creating interference between the users, CDMA employs spread spectrum technology and a special coding scheme.

There are two approaches to spread spectrum modulation, DSSS and FHSS (Frequency Hopping Spread Spectrum). While in FHSS, also known as Frequency Hopping CDMA (FH-CDMA), a broad slice of the bandwidth spectrum is divided into many possible broadcast frequencies, DSSS spread the spectrum of data signal taking up much more bandwidth than the original data signal bandwidth. In Figure 1 a spread spectrum example is shown:

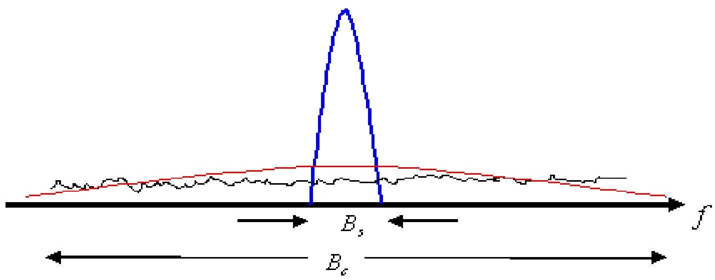

Figure 1: DSSS power spectral density scheme.

In DSSS the baseband signal bandwidth is intentionally spread over a larger bandwidth and the spectral density is reduced by injecting a higher frequency signal also known as chipping code. This chipping code is an orthogonal pseudo noise (PN) sequence with specific autocorrelation properties.

One of the most important properties of these PN sequences is the Cyclic Shift Property which define that the autocorrelation of any shift is always constant except when the shift is zero, and therefore, permitting each receiver distinguish the desired signal from the combined signal. PN sequences can be generated easily by Linear Feedback Shift Registers (LSFR), setting carefully the length of the sequence. The length of the sequence (N) not only sets the number of pseudorandom orthogonal codes and thus the number of users, but also establishes the autocorrelation parameters and thus the multiuser interference in each receiver. Long codes will assure stable and robust communications against noise and multiuser interference, but will overload the channel with redundant information.

As a direct consequence of this spreading process, energy used in sugnal transmission is spread over a wider bandwidth, and appears as noise. The ratio (in dB) between the spread baseband and the original signal is called processing gain. Typical spread-spectrum processing gains run from 10dB to 60dB.

Usually, additional modulation techniques such as Quadrature Phase-Shift Keying (QPSK), Binary Phase-Shift Keying (BPSK) or Gaussian Frequency-Shift Keying (GFSK) are also combined.

This processing involves some interesting features in DSSS:

- Due to the autocorrelation properties, this system has good protection against disruptive users (Jammers) that transmit with high power in order to block the channel, as all the power and the interference will be reduced significantly and equalized between all the users. Also, the communication media becomes difficult to intercept by non-authorized users with the existence noise-like signals

- The high protection against multiuser interference allow the frequencies reuse, which permits the frequency scheduling and therefore supporting a bigger amount of users.

- The correlation techniques permit to establish a Rake Receiver Channel, which extracts multipath diversity through coherent combination of the multipath components and suppresses the undesired components. Therefor improves the SNR of the desired signal

In general, FH-CDMA system uses less power and it's cheaper, but offers low bitrates. However the performance of DS-CDMA systems is usually better and more reliable, offering a much better interference response, since the correlation processes allow recover many chipping errors without result in bit errors. This is why DSSS is the most extended CDMA approach nowadays and the common solution for the major of the equipment manufacturer.

DSSS is widely implemented in the standard IEEE802.11b and also in IEEE802.11g for backward compatibility. Four bitrates are defined for both standards: 1Mbps, 2Mbps, 5.5Mbps and 11Mbps. Usually, all commercialized interfaces are compatibles with the four bitrates, even though 1Mbps and 2Mbps are defined in 802.11b and 5.5Mbps and 11Mbps in 802.11g

To distinguish it from the original direct sequence PHY, the high rate PHY that runs at 11 Mbps is abbreviated as HR/DSSS. Like its predecessor, it is split into a convergence procedure that prepares frames for radio transmission, and a medium-dependent layer that turns the bits into radio waves in the air. Some PHY/MAC parameters of DSSS and HR/DSSS are shown in Table 4:

Parameter	DSSS	HR/DSSS
Slot time	20 µs	20 µs
SIFS	10 µs	10 µs
Contention Window Size	31-1023 slots	31-1023 slots
Preamble	144 µs	144 µs
PLCP Header	48 µs	48 µs
Max Frame Size	8191 bytes	4095 bytes
Min Sensibility Receiver	-80dBm	-76dBm
Rejection Adyacent Channel	35dB	35dB

Table 4: PHY/MAC parameters for DSSS

The channels in DSSS are 5MHz width, where the first channel starts from 2.412GHz, however the signal energy spreads over a 22MHz band. In order to avoid the nearest channels interference, the main lobe is filtered at -30dB and the other lobes at -50dB.

The current frequency scheduling of 2.4 GHz is shown in Table 5:

Regulatory Domain	Channels Permitted
USA (FCC) / Canada (IC)	1 – 11 (2.412-2.462 GHz)
Europe (ETSI)	1 – 13 (2.412-2.472 GHz)
Japan (MIC)	1 – 13 (2.412-2.462 GHz) and 14 (2.484 GHz)

Table 5: Frequency policies for 2.4 GHz band

Aside from 802.11g, DSSS is also used in mobile communications technologies such as GSM and UMTS, satellite communication system like GPS, Galileo and GLONASS and Wireless Personal Area Network (WPAN) technologies from the IEEE 802.15.4 family.

2.1.2.2 Orthogonal Frequency-Division Multiplexing (OFDM)

Orthogonal Frequency-Division Multiplexing (OFDM) is a multicarrier modulation technique developed in 1960s that has recently found wide adoption in a wide variety of high data-rate communication systems, including IEEE 802.11a/g/n, Digital Subscriber Lines (DSL), IEEE 802.16 (WiMAX), Digital Video Broadcasting (DVB), or 4G LTE cellular systems.

The idea of multicarrier modulation consists in dividing the wireless channel in a number of sub-channels in which a different carrier is modulated in order to transmit data in a lower fraction of the total high rate.

This concept follows naturally from the desire of have an Inter Symbol Interference (ISI-free) channel so that the bit error rate can be tolerable. For wideband channels that provide high data rates, the symbol time is usually much smaller than the delay spread, but for have an ISI-free channel, the symbol period T_s has to be significantly larger than the channel delay spread τ.

In order to overcome this problem, multicarrier modulation divides the high rate transmit bit stream into L lower rate substreams, each of which has $T_s >> \tau$. These individual substreams can then be sent over L parallel orthogonal subchannels, maintaining the total desired data rate. The number of substreams is chosen to ensure that each subchannel has a bandwidth less than the coherence bandwidth of the channel $B/L << B_c$, experimenting a relatively flat fading.

OFDM, unlike a traditional frequency division multiplexing technique, do not waste the spectrum with guard bands, because the each sub-carrier's spectrum is a sinc function with zero crossing every f_o as shown in Figure 2. Since the sub-carrier spacing is identical to f_o, there is no inter-sub-carrier interference and OFDM spectrum is overlapping. Then the high frequency band usage is achieved placing the sub-carriers every f_o frequency interval. This makes that the total power spectrum shape is close to square, achieving high frequency usage efficiency.

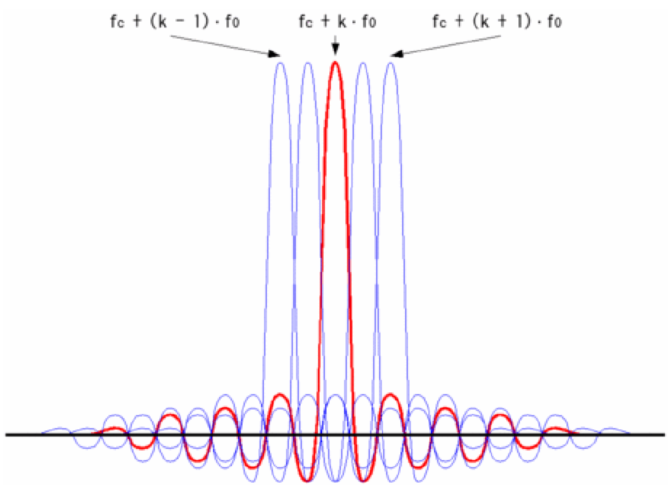

Figure 2: OFDM spectrum example

To carry out this modulation procedure, OFDM uses a highly efficient computational implementation of the Discrete Fourier Transform (DFT) commonly known as the Fast Fourier Transform (FFT) and its inverse, IFFT, enabling a multitude of orthogonal subcarriers using a single transmitter.

Beside the bandwidth division, OFDM incorporates other techniques to create an ISI-free channel in practice, like the Cyclic Prefix and the block transmission with Guard Intervals (GI). OFDM group L data symbols into a block known as an OFDM symbol. In order to keep each OFDM symbol independent of the others after going through a wireless channel, it is necessary to introduce a guard time GI between OFDM symbols. But if the GI is empty, the presence of spread delay can endanger the orthogonality.

For this reason each OFDM symbol is extended with a Cyclic Prefix filling the previous GI and thus the channel must appear to provide a circular convolution. Working with FFT, when an input data is sent through a linear time-invariant Finite Impulse Response (FIR) channel, the output is the circular convolution of the input and the channel response. This behavior it achieved adding this Cyclic Prefix onto the OFDM symbol, always the delay spread is smaller than the GI. These procedures are shown in Figure 3.

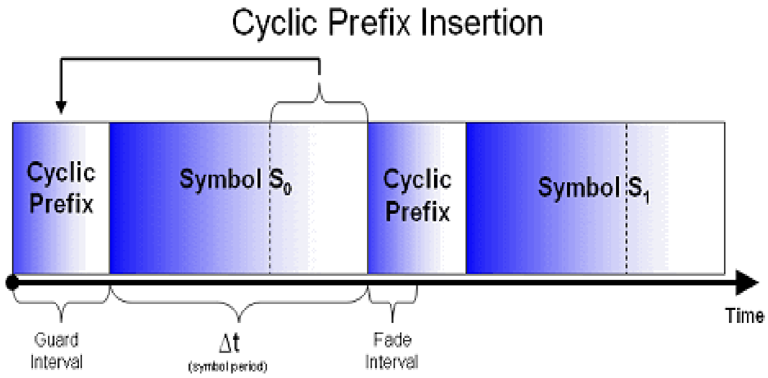

Figure 3: Cyclic Prefix example

IEEE 802.11a was the first version that used OFDM as PHY layer. 802.11a works in 5 GHz band with separation of 20 MHz, 10 MHz or 5 MHz between the channels where each of them has 52 subcarriers. Available bitrates are: 6 Mbps, 9 Mbps, 12 Mbps, 18 Mbps, 24 Mbps, 36 Mbps, 48 Mbps and 54 Mbps. The coding details of these bitrates are defined in Table 6.

Bitrate for 20MHz channel	Modulation	Code Rate	Bits per carrier	Bits per symbol
6 Mbps	BPSK	1/2	1	48
9 Mbps	BPSK	3/4	1	48
12 Mbps	QPSK	1/2	2	96
18 Mbps	QPSK	3/4	2	96
24 Mbps	16-QAM	1/2	4	192
36 Mbps	16-QAM	3/4	4	192
48 Mbps	64-QAM	2/3	6	288
54 Mbps	64-QAM	3/4	6	288

Table 6: Codification parameters in 802.11a

Like in 2.4 GHz band, the regulation of the 5 GHz band varies according to the regulatory domains. These are shown in Table 7.

Regulatory Domain	Channels Permitted
USA (FCC) / Canada (IC)	36, 40, 44, 48 (5.180, 5.200, 5.220 & 5.240 GHz) 52, 56, 60, 64 (5.260, 5.280, 5.300 & 5.320 GHz) 149, 153, 157, 161, 165 (5.745, 5.765, 5.785, 5.805 & 5.825 GHz)
Europe (ETSI)	100, 104, 108, 112, 116, 120, 124, 128, 132, 136, 140 (5.500, 5.520, 5.540, 5.560, 5.580, 5.600, 5.620, 5.640, 5.660, 5.680 & 5.700 GHz)
Japan (MIC)	8,12,16 (5.040, 5.060 & 5.080 GHz) 36,40,44,48 (5.180, 5.200, 5.220 & 5.240 GHz)

Table 7: Frequency policies for 5GHz band. In some of the channels, DFS and TPC mechanisms must be active, especially in Europe and Japan.

Later, IEEE 802.11g was launched to aggregate the former standards IEEE 802.11b and IEEE 802.11a but applying into the 2.4 GHz band. In this manner, 802.11g get all the advantages from 802.11a but maintaining backward compatibility with 802.11b, resulting in new available bitrates offered by OFDM through Extended-Rate PHY (ERP), and still maintains the DSSS rates.

Thereby, there are 12 bitrates available in 802.11g: 1Mbps, 2Mbps, 5.5Mbps and 11Mbps from DSSS and 6 Mbps, 9 Mbps, 12 Mbps, 18 Mbps, 24 Mbps, 36 Mbps, 48 Mbps and 54 Mbps from ERP-OFDM. The channels permitted in the 2.4 GHz band are the previously shown in Table 5.

ERP-OFDM and OFDM are defined by similar standards but implemented in different frequency bands; hence their operation modes are practically the same. The only differences reside in the backward compatibility is that ERP-OFDM has to be applied with 802.11b. In order to illustrate these differences, some PHY/MAC parameters of ERP-OFDM and OFDM are shown in Table 8.

Parameter	OFDM	ERP-OFDM
Slot time	9 μs	20 μs or 9 μs
SIFS	16 μs	10 μs
Contention Window Size	15-1023 slots	15 or 31-1023 slots
Preamble	16 μs	20 μs
PLCP Header	4 μs	4 μs
Max Frame Size	4095 bytes	4095 bytes
GI	0.8 μs	0.8 μs

Table 8: PHY/MAC parameters for ERP-OFDM.

2.1.2.3 High-Throughput OFDM (HT)

IEEE STD 802.11-2012, in its Clause 20, specifies the PHY entity for a high throughput (HT) OFDM system. This Clause 20 revises the former IEEE STD 802.11n-2009, which is now retired.

The HT PHY is based on the OFDM PHY defined previously, increasing the throughput with these extensions:

- Transmission using one to four spatial streams, with all equal modulation (EQM) or unequal modulation (UEQM).

- Transmission in 20 MHz (with 56 OFDM subcarriers) or 40 MHz (with 114 OFDM subcarriers).
- Optional 400 ns short guard interval (GI).
- Optional beamforming.
- Optional Space-Time Block Coding (STBC).
- Frame aggregation at the PHY level.

The HT PHY data subcarriers are modulated using binary phase shift keying (BPSK), quadrature phase shift keying (QPSK), 16-quadrature amplitude modulation (16-QAM), or 64-QAM. Forward error correction (FEC) coding (convolutional coding) is used with a coding rate of 1/2, 2/3, 3/4, or 5/6. LDPC codes are added as an optional feature. The maximum HT PSDU length is 65.535 octets.

These features are capable of supporting data rates up to 600 Mb/s (four spatial streams, 40 MHz bandwidth, 400 ns GI, EQM, 5/6 and 64-QAM).

A spatial stream is defined as one of several streams of bits or modulation symbols that might be transmitted over multiple spatial dimensions that are created by the use of multiple antennas at both ends of a communications link. Thus, to obtain more than one spatial stream (This implementation brings additional capabilities such as beamforming and STBC), it is necessary to implement Multiple Input Multiple Outputs (MIMO) links using STAs with more than one antenna or capable of transmit and receive on two orthogonal polarizations. The capability to transmit up to four spatial streams represents a direct increase in data rate. In this way and if the spatial dimensions are kept orthogonal to each other, the maximum data rate is up to four times the data rate achieved with a single spatial stream.

Multiple spatial stream usage allows exploiting the MIMO channel performing a link adaptation. This adaptation enables to adapt the data rate of each spatial stream in a highly flexibility and adaptability way which result in 76 available MCSs. There are three link adaptation modes defined:

- **Immediate**: An immediate response occurs when the MFB responder transmits the response in the TXOP obtained by the TXOP holder. This approach allows the MFB requester to obtain the benefit of link adaptation within the same TXOP.
- **Delayed**: A delayed response occurs when the MFB responder transmits the response in the role of a TXOP holder in response to an MRQ in a previous TXOP obtained by the MFB requester.
- **Unsolicited**: An unsolicited response occurs when a STA sends MFB independent of any preceding MRQ.

MIMO channels also allow beamforming: a spatial filtering mechanism is used at a transmitter to improve the received signal power or signal-to-noise ratio (SNR) at an intended receiver. In order for a beamformer to calculate an appropriate steering matrix to transmit spatial processing when transmitting to a specific beamformee, the beamformer needs to have an accurate estimate of the channel over which it is transmitting. Two estimation methods are defined:

- **Implicit feedback**: the beamformer receives long training symbols transmitted by the beamformee, which allow the MIMO channel to be estimated. If the channel is reciprocal, the beamformer can use the training symbols received from the beamformee to make a channel estimate suitable to compute the transmit steering matrix.
- **Explicit feedback**: the beamformee makes a direct estimate of the channel from training symbols sent by the beamformer. The beamformee may prepare CSI or steering feedback based on an observation of these training symbols. The beamformee quantizes the feedback and sends it to the beamformer. The beamformer can use the feedback to determine the transmit steering vectors.

Beamforming increases the SNR but does not achieve the maximum data rate. Additionally, beamforming can be used to realign a link if there is any slight problem of aiming of the antennas.

Implementing a MIMO system also enables STBC transmission, where the spatial streams are spreaded into space-time streams using STBC codes. This technique achieves a coding gain at the expense of decreasing the data rate.

MCS Index	Spatial streams	Modulation	Cod. Rate	Data rate			
				20 MHz channel		40 MHz channel	
				800 ns GI	400 ns GI	800 ns GI	400 ns GI
0	1	BPSK	1/2	6,5	7,2	13,5	15,0
1	1	QPSK	1/2	13,0	14,4	27,0	30,0
2	1	QPSK	3/4	19,5	21,6	40,5	45
3	1	16-QAM	1/2	26	28,8	54	60
4	1	16-QAM	3/4	39	43,2	81	90
5	1	64-QAM	2/3	52	57,6	108	120
6	1	64-QAM	3/4	58,5	64,8	121,5	135
7	1	64-QAM	5/6	65	72	135	150
8	2	BPSK	1/2	13	14,4	27	30
9	2	QPSK	1/2	26	28,8	54	60
10	2	QPSK	3/4	39	43,2	81	90
11	2	16-QAM	1/2	52	57,6	108	120
12	2	16-QAM	3/4	78	86,4	162	180
13	2	64-QAM	2/3	104	115,2	216	240
14	2	64-QAM	3/4	117	129,6	243	270
15	2	64-QAM	5/6	130	144	270	300

Table 9: MCS with one or two spacial streams in 802.11n. The four columns on the right show the physical bitrates for the corresponding transmission schemes, channel width and GI value.

2.1.3 802.11ac

A new amendment to the IEEE 802.11 standard is being prepared under the name 802.11ac for very high throughput in the 5 GHz band. The key improvements on top of 802.11n are:

- Channels up to 160 MHz width, not necessarily contiguous.
- MIMO with up to 8 spatial streams.
- New modulations up to 256QAM.

This standard is still under development, the final release being expected by the end of 2013.

2.1.4 The MAC in the 802.11 family

As it happens with the PHY, there are several options for the MAC in the 802.11 family but only two of them are "alive". The MAC layer uses Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA). A complementary mechanism named "Request-to-Send, Clear-to-Send" (RTS/CTS) is additionally introduced as a solution for the so called hidden node problem. Beyond this, two coordination functions were defined: the Point Coordination Function (PCF), centralized, in which the Access Point is given the power of opening contention-free periods that controls the access to the channel by polling the stations one by one, and the Distributed Coordination Function (DCF), in which all the stations (including the Access Point) use CSMA/CA all the time and there is no mechanism to open contention-free periods.

While the initial standard defined two coordination functions, the centralized one was never implemented in the industry because (i) it was optional, and (ii) it performed worse than the distributed one.

Later on, the 802.11e amendment proposed other coordination functions that introduced the possibility of a differentiated channel access priority for different access categories or traffic classes. Again, there was a centralized method called Hybrid-coordination-function Controlled Channel Access (HCCA) and a distributed one called Enhanced Distributed Channel Access (EDCA). And again, the first one has not been extensively been implemented because it is optional and additionally the Wi-Fi Alliance has not created a certification program for HCCA, being Wireless MultiMedia (WMM) the certification given to EDCA products.

Let's explain more in detail each coordination function defined in the 802.11 MAC.

Distributed Coordination Function (DCF)

DCF uses a Carrier Sense Multiple Access protocol with Collision Avoidance (CSMA/CA) that considers all stations as equals (client stations or access points).

The following parameters are the basics of DCF:

• SlotTime: Quantum of time to use as contention window time unit and to define other parameters.

• SIFS (Short Inter Frame Space): Time between the end of the reception of a frame and the start of the transmission of its ACK.

• DIFS (Distributed Inter Frame Space): A station needs to sense the channel idle for DIFS seconds before scheduling a new transmission or reactivating the contention window countdown if it was interrupted due to another station's transmission. Its duration is (2 × SlotTime + SIFS).

• EIFS (Extended Inter Frame Space): Replaces DIFS when the last transmission that kept the channel busy was corrupted.

A station willing to transmit has to sense the channel idleness during a DIFS time window. Then, it starts a contention window by calculating a random number of time slots to wait and starting a countdown. The duration of each slot depends on what is happening in the channel: Idle slots have a fixed duration SlotTime, but the countdown freezes when it is sensed that the channel is busy, hence producing a slot that contains a transmission or a collision followed by DIFS or EIFS. When the countdown finishes, the station transmits and starts a timer (n ACKTimeout) to wait for an ACK that confirms the correct reception. If the ACK is received, the transmission is considered as successful, and the station is ready to restart the whole process again with another frame. If ACKTimeout elapses and no ACK is received, a retransmission is started following the same process until the maximum retransmission limit is reached. The number of slots in the contention window is calculated as a uniform random variable in [0, CW_i] where CW_i + 1 grows exponentially each time the transmission is unsuccessful, starting at CW_{min} +1 and ending at CW_{max} +1.

The 802.11 MAC is strongly based in the carrier sense, but has a mechanism to deal with hidden nodes called RTS/CTS (Request to Send / Clear to Send). When activated, a station that is going to transmit a data frame longer than RTSThreshold sends firstly a very short RTS frame (only 14 bytes). If the receiver gets the RTS frame correctly, it sends back a CTS frame, giving the first station the right to transmit the data frame. Both the RTS and the CTS frames contain the information of the total duration foreseen for that transaction, including all the operations until the channel will be definitely idle. Stations listening to either the RTS or the CTS frame may initialize an internal timer called NAV (Network Allocation Vector) that causes the same effect to the MAC as the physical carrier detection.

Further details of the DCF operation can be found in the standard [IEEE 802.11-2012] and also [Bianchi2000, Bianchi2005] are recommended for a better understanding.

Enhanced Distributed Channel Access (EDCA)

EDCA uses four access categories (AC) to manage traffic priorities. Each access category has its own transmission queue characterized by a particular adjustment of several access parameters:

- $AIFSN_i$ (Arbitrary Inter-Frame Space Number), that defines $AIFS_i = SIFS+AIFSN_i \sigma$, which is the time that the channel must stay free after a transmission before the station opens the contention window for AC_i.
- CW_i (Contention Window), that determines the maximum possible size of the contention window for the first transmission of a packet in AC_i, and
- $TXOPlimit_i$ (Transmission Opportunity), that determines how long a station may retain the channel once it has won the contention.

A station willing to transmit in EDCA a packet that belongs to AC_i has to sense the channel and then

- If the channel is idle, it has to verify that the channel stays idle during a period $AIFS_i$, and then it may transmit.

- If the channel is busy, it has to wait for the channel to be idle and then to stay idle during a period $AIFS_i$. Then, the station opens a contention window with CW_i slots that each has *SlotTime* duration. CW_i size is calculated with a uniform random variable in [0, CW_i -1] interval for each i-th category. The upper limit of CW_i grows exponentially each time the transmission is unsuccessful, starting at CW_{min} and ending at CW_{max}. If at any time during the contention windows the channel is sensed busy, the countdown freezes, and continues after the channel becomes idle if it stays idle during an $AIFS_i$ time. When the countdown finishes the transmission happens and the stations waits for an ACK that confirm the correct reception. If the ACK arrives before a time *ACKTimeout* the transmission is considered successful, if not a retransmission is started following the same process until the maximum retransmission limit is reached.

A collision happens when two or more stations transmit simultaneously. If two AC try to transmit simultaneously in the same station, there is an internal collision mechanism that permits the class with highest priority to transmit and the other one is treated as if a real collision has happened. Finally, if a transmission is successful, the station may keep the channel busy for a time that is limited by the $TXOPlimit_i$ parameter.

EDCA also defines a mechanism named Block-ACK that permits to send several consecutive packets during the TXOP without individual confirmation of each one. The first packet is transmitted with explicit solicitation of Block-ACK scheme, then it receives the ACK that confirms the reception and the acceptance, and then the transmitting station may transmit several packets just spaced by SIFS without individual acknowledgments. When the transmitting station finishes with the block, then it sends a BAR (Block-ACK Request) and the receiving station sends the BA (Block-ACK). This mechanism is particularly interesting for long-distance point-to-point links because it dramatically improves the performance and reduces the impact of distance.

2.1.5 Literature on long-distance links using 802.11

[Baugh2003] were the first to propose the use of WiFi over long distances in the scientific literature, but only in 2010 [Simo2010] modeled and explained completely how to adapt WiFi for long distances and how to predict its performance.

The PHY layer and the MAC layer establish limits in the coverage distance. Also [Lopez2010] studies the impact of the distance in outdoor mid-range WiFi setups, especially on the delay.

In PHY layer, the highest nominal bit rates are achieved with powerful modulations and low redundancy coding schemes, but the high received power levels required are difficult to achieve in long-distance links. So, the achievable physical bitrate effectively decreases with the distance. On the other hand, lower nominal bitrates require lower received power levels and can be used over longer distances. Figure 4 shows the achievable distances for point to point, point to multipoint or mesh WiFi links. The gain is assumed to be 12dBi for omnidirectional antennas and 24dBi for directive antennas. Also, it shows that long distances can be reached only if high gain directive antennas are used. The

FCC regulations are taken as the references here regarding the maximum transmit power level and the maximum antenna gain.

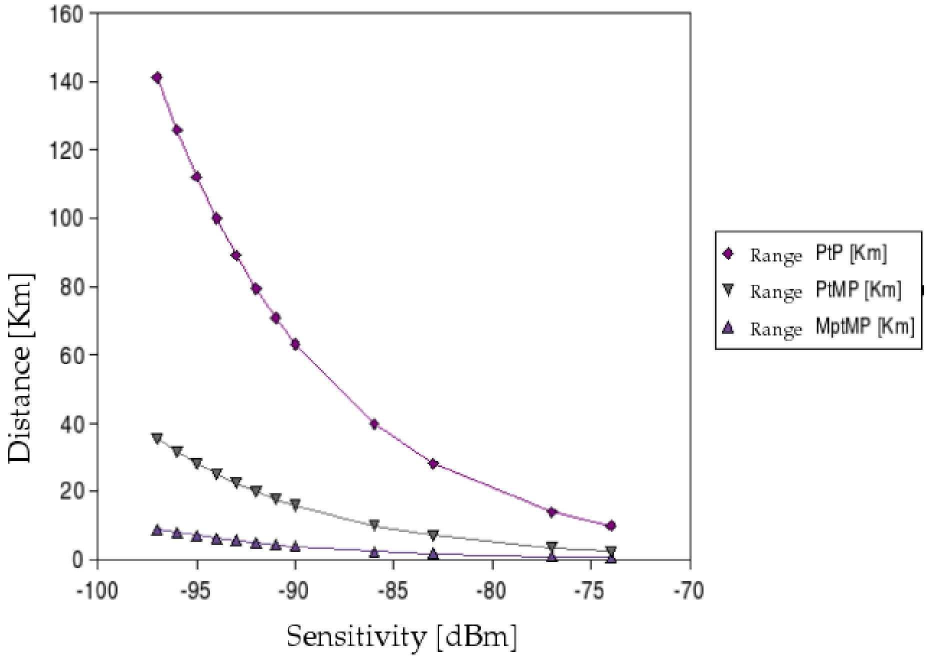

Figure 4: Range achieved according receiver sensitivity

Regarding the MAC layer, in principle CSMA/CA is unsuitable for long distances because the propagation delay affects the contention for distances longer than a few hundred meters. The longer the distance is, the worse the performance becomes [Simo2010, Lopez2010]. There are three different possible strategies for adaptation of WiFi systems to long distances:

- In the year 2007 WiFi is redefined as suitable for WMAN (Wireless Metropolitan Area Network). The standard introduces the Coverage Class, a numeric value that defines the MAC parameters to operate in diverse distances. The coverage distance reaches 15Km for the highest Coverage Class. If the distance is shorter than 15 Km, hardware systems may be configured this way to perform optimally. However, there is a certain loss of performance as the distance grows from a few hundred meters up to 15 Km.
- WiLD adaptations to CSMA/CA: The loose of performance with long distances may be reduced by adapting three MAC parameters in a non-standard fashion (that is possible with certain hardware systems); SlotTime, ACKTimeout and CTSTimeout. The loose of performance is a bit higher in this case than in the previous one, but the coverage distance may reach up to 105 km for 802.11b, or approximately 50 km for 802.11a/g with available hardware systems.
- Alternative MAC systems: modern WiFi chipsets are flexible enough to replace the medium access control mechanism. Some industrial and academics actors have proposed TDMA mechanisms that perform better for long links (or, at least, the performance does no longer depend on the distance beyond the power budget). Ubiquiti AirMAX and Mikrotik NV2 are some popular solutions of this kind. Moreover, products that implement these alternative MAC solutions also implement CSMA/CA (usually with EDCA) and one can chose what of the MAC alternatives must be used in a given scenario.

The adaptation of 802.11e for traffic differentiation over long distances in CSMA/CA (either standard or the WiLD adaptation) is straight-forward. Beyond that, 11e also offers the BlockACK facility that has been demonstrated useful to improve the performance, especially over long distances. The same thing happens with frame aggregation in 802.11n. [Salmer2008] shows the improvement of performance in long-distances WiFi links using Block-ACK.

The use of MIMO in 802.11n may also be of some use for long links, but the benefit is more limited here than in short distances and has not still been adequately measured or modelled.

2.1.6 Literature on DCF analytical models

There are hundreds of papers proposing analytical models of IEEE 802.11 DCF, most of them based on the seminal paper by Giuseppe Bianchi [Bianchi2000]. Bianchi proposed a model using a bi-dimensional Markov chain. The model was based on some strong simplifications, the limited number of retransmissions, the constant transmission probability and conditional collision probability for all stations and all stages, and the complete visibility among stations. For the rest, the model captured all the complexity of DCF, hence giving very good results.

After this model, many other authors published their own analysis on improving or extending Bianchi's bi-dimensional Markov chain. In [Chatzi2003], [Chatzi2002], [Wu2002], the authors took into account the finite number of retransmissions. Other models included variable packet sizes [Chen2004], or introduced the effects of non-ideal wireless channels [Dong2005], among other extensions. In 2005, Bianchi and Tinnirello [Bianchi2005] proposed an alternative model only based on conditional probability, incorporating the corrections and contributions from other authors to [Bianchi2000].

Recently [Bianchi2010] proposed a refined model that accurately considers all the details of DCF and proposes calculations for saturation throughput, delay and packet loss. The performance under saturation conditions gives an idea of the performance limits, but delay and packet loss in saturated conditions are not meaningful because they actually represent the size of packet buffers and the saturation level more than the behaviour of the MAC mechanisms. Moreover, in terms of delay and packet loss, the performance degrades as the network approaches the saturation point. That point needs to be known in order to define the working zone (i.e. the maximum offered load recommended), and delay and packet loss are better modelled for that working zone with an under-saturated model like the one proposed in [Malone2007]. Moreover, this last model succeeds to explain the long-term unfairness problems of DCF, making clear that DCF cannot be used in saturation conditions with real-time traffic. Although less accurate, [Zhao2009] is a very interesting approach because it defines an approximate model much less complex than [Malone2007] and still useful for admission control assessment.

There are also a number of authors that have proposed alternative models that are not based on Bianchi's approaches. None of them are referenced here because they are not as accurate as [Bianchi2010] nor they can incorporate the effect of long distances in the analysis.

2.1.7 Literature on EDCA analytical models

Analytical models that permit to predict the behaviour and performance of DCF are not applicable to EDCA, which is the currently implemented coordination function proposed by IEEE 802.11 as an alternative to DCF that permits relative QoS support. The reason is twofold:

- The traffic is no longer uncoloured. Different traffic classes (up to four) are differentiated and given different treatment, which makes the theoretical analysis much more complicated.
- There are slight differences in the way the contention window is used.

There are many papers regarding the performance analysis and optimization of IEEE 802.11e EDCA, but none of them considers explicitly the case for long-range rural communications, or even metropolitan area networks. Hence, we are aiming to identify in the literature those papers containing relevant contribution in (i) presenting and analyzing the effects of the different variable parameters in

EDCA, (ii) analyzing the performance with enough accuracy, and (iii) proposing an analysis in which the introduction of the distance is straight-forward.

The first important papers proposing analytical models for EDCA were published between 2004 and 2006, around the date of publication of the IEEE 802.11e standard [Kong2004, Robin2004, Tantra2005, Xiao2005, Tao2006, Tantra2006, and Chen2006]. These models were still far from the required accuracy in the representation of four ACs with several differentiated parameters concurrently contending for the medium. A more descriptive paper [Bianchi2005b] analyses the effect of using both CW_{min} and AIFSN for traffic differentiation, concluding that AIFSN is more useful for differentiating the average delay of the different ACs, while CW_{min} is the parameter to tune for throughput optimization. They also propose an optimal value of CW_{min} as a function of the biggest AIFSN value given to any AC. [Banchs2007] and [Banchs2010] extend this line of research.

Tinnirello and Choi [Tinnir2005] show how TXOP permits to increment the effective throughput in combination with the Block-Ack scheme. A similar analysis is done in [Tianji2005], but considering error-prone channels. [Lee2007] and [Lee2010] go deeper in this line.

In this period 2007-2010 many other papers study more in detail the performance analysis of EDCA in more realistic scenarios: channels with errors, finite traffic load, heterogeneous stations, etc. [Papapa2007, Sharkh2008, Liu2007].

Finally, in recent years more accurate models have been proposed including all the details of EDCA, all the possible variable parameters in each AC, under-saturated traffic and heterogeneous stations [Inan2009, Hu2012].

Some other papers with particular interest because of their consequences in practical scenarios are [Kim2009, Hu2011]. The first shows how to use the variable parameters in EDCA to obtain deterministic priorization of VoIP traffic, something that may be studied for the telephony traffic in backhaul networks in this project. The second analyses the effect of unbalanced traffic in the performance of access networks and proposes de asymmetrical use of TXOP for uplink/downlink tuning.

In the rest of the document, [Tinnir2010] and [Hu2012] will be specially taken as a reference for EDCA analysis, and [Lee2010] will also be taken into account for the study of TXOP and Block-ACK.

2.1.8 Literature on spatial diversity and MIMO

The use of multiple antennas at the transmitter and receiver in wireless systems provides potential powerful performance-enhancing capabilities [Biglieri2007]. The MIMO technology offers a number of benefits that help meet the challenges posed by both the impairments in the wireless channel (predominantly by multi-path fading) as well as resource constraints (limited power and scarce frequency bandwidth). The MIMO systems exploit the spatial dimension (provided by the multiple antennas) in addition to the time and frequency dimensions (that are also exploited in conventional SISO wireless systems).

The benefits of MIMO technology that help achieve significant performance gains are:

- **Array gain**: It is the increase in received SNR that results from a coherent combining effect of the wireless signals at a receiver. The coherent combining may be realized through spatial processing at the receive antenna array and/or spatial pre-processing at the transmit antenna array. Array gain improves resistance to noise, thereby improving the coverage and the range of a wireless network.

- **Spatial diversity gain**: It mitigates fading and is realized by providing the receiver with multiple (ideally independent) copies of the transmitted signal in space, frequency or time. The probability that at least one of the copies is not experiencing a deep fade increases, thereby improving the quality and reliability of reception. A MIMO channel with M_T transmit antennas and M_R receive antennas potentially offers M_TM_R independently fading links (spatial diversity order of M_TM_R).

- **Spatial multiplexing gain**: It offers a linear increase in capacity (data rate) by transmitting multiple, independent data streams within the bandwidth of operation. Under suitable channel conditions, such as rich scattering in the environment, the receiver can separate the data streams, effectively enhancing the capacity of the SISO system by a multiplicative factor equal to the number of streams. In general, the number of data streams that can be reliably supported by a MIMO channel is $\min\{M_T,M_R\}$.

- **Interference reduction and avoidance**: Interference may be mitigated in MIMO systems by exploiting the spatial dimension to increase the separation between users. For instance, array gain improves the signal-to-noise-plus-interference ratio (SINR); and the spatial dimension may avoid the interference (directing signal energy towards the intended user and minimizing interference to other users).

In general, it is not possible to exploit simultaneously all the benefits described due to conflicting demands on the spatial degrees of freedom. Two key performance metrics associated with any communication system are the transmission rate (R; defined as the data rate transmitted per unit bandwidth) and the frame-error rate (FER; defined as the probability with which the transmitted frame or packet is incorrectly decoded at the receiver, and is a function of the SNR and R). A fundamental trade-off exists in any communication system between R and FER and, in the context of MIMO systems, this trade-off is referred to as the diversity–multiplexing trade-off [Zheng2003] with diversity signifying the FER reduction and multiplexing signifying an increase in transmission rate. With the diversity gain (d) and multiplexing gain (r) defined [Biglieri2007], a flexible trade-off between d and r can be achieved: the optimal trade-off curve $d(r)$, is piecewise linear connecting (r, $d(r)$), with $r = 0$, 1... $\min\{M_T,M_R\}$, where $d(r) = (M_R - r)(M_T - r)$.

This trade-off is widely studied in [Lozano2010], where it is established that the transmission techniques which utilize all available spatial degrees of freedom to multiplex outperform techniques sacrifices multiplexing for diversity, especially in low-velocity regime scenarios in the context of most modern wireless systems and the operating point of interest. Features of modern MIMO systems include wideband channelization and OFDM; packet switching, complemented with time and frequency domain scheduling; powerful channel codes; link adaptation, and specifically rate control via variable modulation and coding; ARQ (Automatic Repeat Request) and H-ARQ (Hybrid-ARQ). These features have had a major impact on the operational conditions. For example, the fading (CSI; Channel State Information) of low-velocity users can be tracked and fed back to the transmitter thereby allowing for link adaptation to the supportable rate, scheduling on favorable time/frequency locations, and possibly beamforming and precoding. At low velocities, timely feedback regarding the current state of the channel becomes feasible and all uncertainty is removed except for the noise. With powerful coding, outages are essentially eliminated. The transmit diversity techniques are then beside the point and the rate maximization (multiplexing) becomes the overriding transmission design principle. This consideration posited perfect CSI at the transmitter, and it also extends to imperfect-CSI settings (caused by limited rate and/or delay in the link adaptation loop). This conclusion applies to indoor systems and to stationary and pedestrian users in outdoor systems.

Anyway, either to achieve diversity or multiplexing, it is necessary for the MIMO channel complies with specific characteristics. The MIMO systems transmit multiple spatial data streams in parallel and good performance is achieved when the MIMO system maintains separate these streams. The MIMO channel at a given time instant may be represented as an M_R x M_T matrix (**H**), and the best performance in the MIMO system can be achieved when this matrix is high-rank, i.e., when **H** consists is formed by uncorrelated channels. The degree of correlation between the individual channels comprising the MIMO channel is a complicated function of the scattering in the environment and antenna spacing at the transmitter and the receiver. The spacing between antennas and the presence of rich multipath leading to antenna decorrelation and full channel rank. In the ideal case, the elements of **H** are i.i.d. Rayleigh fading channels. In practice, the behavior of **H** can significantly deviate from the ideal due to a combination of inadequate antenna spacing and/or inadequate scattering leading to spatial fading correlation. Furthermore, the presence of a fixed (possibly line-of-sight or LOS) component in the channel will result in Ricean fading.

Because of the aforementioned importance of the MIMO channel in the performances obtained by the MIMO systems, there are multiple references dedicated to model the MIMO channel (especially in outdoor environments that are more challenging due to the presence of LOS) and their capacity performance [Gesbert2002, Abhuyupulu2003, Jensen2004, Molisch2004, Almers2006, Liu2007]. [Gesbert2002] proposes a general and realistic outdoor model and investigates the behavior of channel capacity as a function of the scattering radii, distance between the transmitter and receiver arrays, and antenna beam widths and spacing. [Molisch2004] derives a generic model for MIMO wireless channels in macro and micro cells. All these models predict the appearance of keyhole (or pinhole) channels: channels with presence of scatterers local to the transmitter and receiver but in which the multipaths travel from the area local to the transmitter to the area local to the receiver via one dominant path; thus the scattered components, despite potentially large angle spread, will all contain essentially the same information, leading to little or no capacity gain. This kind of channel is especially studied in [Almers2006].

The backhauling scenario in our project is an outdoor environment with long distance links and an important LOS component, which, in principle, seems to identify with keyhole channels. As described in [Sarris2007], if a LOS component is present, this is usually thought to limit the spectral efficiency of the MIMO system due to the high amounts of spatial correlation introduced. This can be attributed to the fact that transmit and receive arrays are in the far field, the LOS signals can be seen as plane waves, and as a result, the LOS response is rank one and the MIMO channel is rank deficient. Contrary to these observations, a number of studies have shown that the LOS response is not inherently correlated and that by using specifically designed antenna arrays, the orthogonality of the received signals can be preserved. A number of configurations that achieve high-rank MIMO channels are reported in [Bohagen2005, Sarris2005, Bohagen2007, and Sarris2007].

[Bohagen2007] proposes the optimal antenna separation for arbitrary uniform linear arrays orientation in LOS MIMO channels and how sensitive the performance of LOS MIMO systems are to deviations from the optimal design. This optimal antenna separation is calculated as

$$d_t \cdot d_r = \frac{\lambda \cdot R}{V \cdot \cos(\theta_t) \cdot \cos(\theta_r)} \tag{2.1}$$

where d_t and d_r are the element separation in the transmitter and receiver array respectively; λ is the wavelength; R represents the distance between the arrays; θ_t and θ_r are angles of the local spherical coordinate system at the transmitter and receiver respectively; and $V = \max\{M_T, M_R\}$.

For example, with $\theta_t = \theta_r = 0$, V = 2, frequency 2.4GHz (λ = 0.125m) and R = 10km, the previous equation results in $d_t\, d_r = 625$ (achievable with an element separation of $d_t = d_r = 25$m).

So, based on the previous references, we can deduce that it is theoretically possible to achieve long-distance high-rank LOS MIMO channels, but in practice this would mean transmission and receiving systems that are not viable (due to their excessive size or that are too expensive) because of the necessary separation between antennas. Therefore different authors propose cross-polarization as another way to obtain two orthogonal channels [Erceg2004, Oestges2004, Dong2005, Jiang2008, Rabe2008, Qin2010, and Vella2010]. [Oestges2004] describes and analyzes the propagation of MIMO multipolarized fixed wireless channels, and defines the XPD (cross-polar discriminator), the key parameter to keep separate channels. [Erceg2004] and [Jiang2008] extend the XPD analysis, finding that it is better in LOS scenarios and it slightly decreases with increasing distance. [Vella2010] specifically discusses the use of cross polarized antennas to mitigate the keyhole effect in long distance MIMO channels. In recent years, Valenzuela-Valdés et al. have proposed polarization schemes more complex than orthogonal polarization diversity for increased diversity gain and MIMO capacity [Valenzuela-Valdés2009a, Valenzuela-Valdés2009b, and Valenzuela-Valdés2012].

There are only few references that include measures of the performance of MIMO systems in real scenarios [Kadir2008, Garcia-Pardo2009, Binelo2010, Vella2010, Paul2011, and Guerra2012]. [Paul2011] presents an experimental performance evaluation study of WiFi links in an open-space outdoor environment, but the larger link is just 1800m long. [Guerra2012] is particularly significant because obtains measures of the performances in long distance LOS links, using 802.11n with dual-polarized antennas.

2.2 IEEE 802.16: documentation from standardization bodies and scientific literature relevant to foresee the performance for static setups

2.2.1 Brief descriptions of the standards

IEEE 802.16 is a set of standards based on Wireless Metropolitan Area Networking (WMAN) that has been commercialized under the name of Worldwide Interoperability for Microwave Access (WiMAX) by the WiMAX Forum. From 1998, the IEEE 802.16 group has developed several air-interface standards for wireless broadband, but was in 2004 when the first WiMAX solution for fixed applications, 802.16-2004, was published. In 2005, IEEE 802.16e-2005 was approved, establishing the basis for WiMAX solution for nomadic and mobile applications. Nowadays, the operative standard is 802.16-2009 that collects and revises the older versions, adding some functionality. However this document is focused in the static version of WiMAX.

2.2.1.1 Standard description

The standard is specially designed for point to multipoint (PtMP) architectures with or without line-of-sight, in which the Base Station (BS) and Subscriber Station (SS) are the main elements. It was developed to suit a variety of applications and deployments scenarios, and hence, offers a huge amount of design choices for system developers. The standard permits operate between the 2 GHz – 66 GHz micro and millimeter wave bands and there are multiple physical-layer choices: a single-carrier physical layer called Wireless-MAN-SCa, an OFDM-based physical layer called WirelessMAN-OFDM (with a version for unlicensed bands below 11GHz called WirelessHUMAN) and an orthogonal frequency division multiple access based physical layer called WirelessMAN-OFDMA. Similarly, there are multiples choices for MAC architecture, duplexing, etc.

For practical interoperability reasons, the scope of the standard is reduced defining a limited number of system profiles and certification profiles. Currently, the WiMAX forum has two different system profiles, the fixed system profile (with OFDM PHY) and the mobility system profile (with OFDMA PHY). A certification profile is defined as a particular implementation of a system profile where the operating frequency, channel bandwidth and duplexing mode are also specified. In order to summarize, in Table 10 some characteristics of IEEE 802.16 are shown.

WiMAX permits multiplexing using OFDMA or Time Division Multiple Access (TDMA) techniques. Both techniques permit have a detailed planning about the transmissions on the channel in each instant and hence avoid the interferences. The difference between them lies in that, while TDMA SSs transmit at different time at the same frequency, in OFDMA SSs transmit at different time at different frequencies, and can be roughly understood like a combination of OFDM and TDMA.

Due to the static requirements, this document is not focused in multiuser techniques, but in duplexing techniques. The duplexing (separation between uplink and downlink) between BS and SS can be realized by two duplexing techniques, Time Division Duplex (TDD) and Frequency Division Duplex (FDD), being only TDD the technique allowed for unlicensed bands.

Some of the interesting features of WiMAX are the scalable bandwidth and data rate, as well as an Adaptive Modulation and Coding (AMC) system. WiMAX has a scalable physical-layer architecture that allows for the data rate to scale easily with available channel bandwidth. WiMAX support a number of Modulation and Coding Schemes (MCS) together with Forward Error Correction (FEC) mechanisms that allows the scheme to be changed according to channel conditions.

AMC is an effective technique to maximize throughput in a time varying channel, and typically set the highest MCS that can be supported by the SNR and SIR at the receiver in order to provide the user the highest possible data rate.

	802.16 (2001)	**802.16-2004**	**802.16e-2005**
Frequency band	10 GHz – 66 GHz	2 GHz – 11 GHz	2 GHz – 11 GHz / 6 GHz
Application	Fixed LOS	Fixed NLOS	Fixed and Mobile NLOS
MAC architecture	PtP, PtMP, mesh	PtP, PtMP, mesh	PtMP, mesh
Tx Scheme	Single carrier	Single carrier, 256 /2048 OFDM	Single carrier, 128/256/512/1024/2048 OFDM
Modulation	QPSK, 16QAM, 64QAM	QPSK, 16QAM, 64QAM	QPSK, 16QAM, 64QAM
Data rate	32 - 134.4 Mbps	1 - 75 Mbps	1 - 75 Mbps
Multiplexing	Burst TDM/ TDMA	Burst TDM/ TDMA / OFDMA	Burst TDM/ TDMA / OFDMA
Duplexing	TDD and FDD	TDD and FDD	TDD and FDD
Channel bandwidths	20MHz, 25MHz, 28MHz	1.75 MHz, 3.5 , 7 MHz, 14 MHz, 1.25 MHz, 5 MHz, 10 MHz, 15 MHz, 8.75 MHz	1.75 MHz, 3.5 , 7 MHz, 14 MHz, 1.25 MHz, 5 MHz, 10 MHz, 15 MHz, 8.75 MHz
Air-interface designation	WirelessMAN-SCa	WirelessMAN-SCa WirelessMAN-OFDM WirelessMAN-OFDMA WirelessHUMAN	WirelessMAN-SCa WirelessMAN-OFDM WirelessMAN-OFDMA WirelessHUMAN
WiMAX implementation	None	Fixed WiMAX	Mobile WiMAX

Table 10: WiMAX features comparison

One of the strengths of WiMAX is its Quality-of-Service (QoS) support. This support is offer on one hand by the PHY layer through flexible and dynamic per user resource allocation (Grant/Request service) and support for advanced antenna techniques such as beamforming, space-time coding and spacial multiplexing through MIMO technologies. On the other hand, the connection-oriented WiMAX MAC layer is designed to bear a large number of users demanding a variety of applications, including voice and multimedia services. This is achieved offering support for constant and variable bit rate, real-time and best effort data traffic. Also, for connections that require enhanced reliability, WiMAX supports Automatic Retransmission Request (ARQ) as well as Hybrid ARQ (HARQ) at the link layer. HARQ is an effective hybrid technique between FEC and ARQ in which the error packets are not simply dropped but they are stored in order to try correct the error. There are two types of HARQ, Chase Combination, in which the retransmissions are identical copies of the original packet and Incremental redundancy, in which the packet is sent gradually in order to avoid the overload in the channel due to the retransmissions.

2.2.1.2 PHY Layer Parameters

Although WiMAX has a single carrier based physical layer, all the currents implementations are based in OFDM since is a transmission scheme for high data rate communications in a NLOS or multipath radio environment. In Chapter 2.1.2.2, the basis of OFDM have been explained.

Basically, the OFDM Layer used by WiMAX is similar to the used by IEEE 802a/g/n. It provides the same advantages like spectral efficiency, exploitation of frequency diversity, coherent demodulation, high protection against narrow band interference, reduced computational complexity, etc. but have a slightly different implementation. These differences reside in the OFDM related parameters. Table 11 shows these parameters for both Fixed WiMAX and Mobile WiMAX:

<table>
<tr><th colspan="2">OFDM Parameter</th><th>Fixed WiMAX</th></tr>
<tr><td colspan="2">FFT</td><td>256</td></tr>
<tr><td colspan="2">Nº of data subcarriers</td><td>192</td></tr>
<tr><td colspan="2">Nº of pilot subcarriers</td><td>8</td></tr>
<tr><td colspan="2">Nº of guardband subcarriers</td><td>56</td></tr>
<tr><td colspan="2">GI (Tguard/Tsymb)</td><td>1/4, 1/8, 1/16, 1/32</td></tr>
<tr><td rowspan="2">Channel bandwidth</td><td>For 3.5 GHz Band</td><td>3.5 Mhz and 7 MHz using TDD</td></tr>
<tr><td>For 5.8 GHz Band</td><td>10 MHz using TDD</td></tr>
<tr><td colspan="2">Oversampling rate</td><td>7/6</td></tr>
<tr><td colspan="2">Subcarrier frequency spacing</td><td>15.625 KHz</td></tr>
<tr><td colspan="2">Useful symbol time</td><td>64 μs</td></tr>
<tr><td colspan="2">OFDM symbol duration</td><td>72 μs</td></tr>
<tr><td colspan="2">Nº of OFDM symbol in 5 ms frame</td><td>69</td></tr>
</table>

Table 11: OFDM parameters for fixed WiMAX

Physical-layer data rate varies significantly depending on the PHY parameters mentioned in Table 11, specially the channel bandwidth, the MCS used, the number of subcarriers and the GI. For this reason Table 12 is attached in order to list the PHY-layer data rate in Kbps according these parameters. The rates shown are the aggregate PHY-layer data rate that is shared among through a TDD technique, assuming a 3:1 downlink-to-uplink bandwidth ratio, carriers spaced 45 KHz, 1/8 of GI and frame size of 5ms.

Channel Bandwidth	3.5 MHz		10 MHz	
PHY Mode MCR	256 OFDM PHY-Layer Data Rate (Mbps)			
	DL	UL	DL	UL
BPSK 1/2	0,946	0,326	2,560	1,270
QPSK 1/2	1,882	0,653	5,120	2,550
QPSK 3/4	2,822	0,979	7,680	3,830
16 QAM 1/2	3,763	1,306	10,240	5,110
16 QAM 3/4	5,645	1,958	15,360	7,670
64 QAM 2/3	7,526	2,611	20,480	10,230
64 QAM 3/4	8,467	2,938	23,040	11,510

Table 12: Physical-layer data rates for Fixed WiMAX

Finally, the WiMAX PHY layer is also responsible for slot allocation and framing over the air. In TDMA, the minimum time-frequency resource that can be allocated by a WiMAX system to a given link is called slot. Each slot consists of one subchannel over one, two, or three OFDM symbol depending of the subchannelization scheme used. Figure 5 shows an OFDM frame when operating in TDD mode in order to illustrate the frame structure.

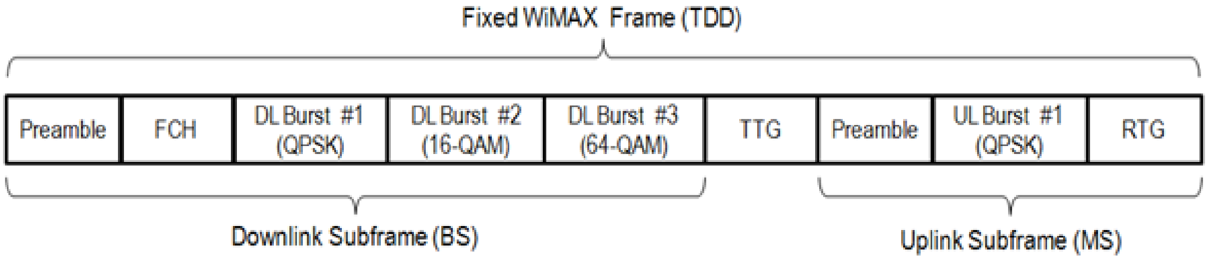

Figure 5: Fixed WiMAX frame structure example

2.2.2 Literature on long-distance links using 802.16

WiMAX has been identified by some authors as the technology that could bridge the digital divide and deploy telecommunication infrastructures profitably and efficiently, not only in urban areas but in the semi-urban and rural areas of the world. Some of the properties that make WiMAX an interesting alternative are its QoS support and its fast and cheap deployments. Low cost is a benefit intrinsic to wireless technologies, but is reinforced in WiMAX because is a standard technology, implying foster economies of scale and innovation among the different industry members besides being independent from a single vendor. Furthermore, IEEE 802.16 standard it is a very flexible standard which permit operate in wide range of scenarios, and in many different frequencies, both licensed and unlicensed.

But the main feature is the good behavior of WiMAX in long-distance links. Since its beginning, WiMAX was designed to provide broadband wireless access through long-distances links and hence, has no intrinsic constraints for long distances as said in [Crey2011], [Rashidah2010] and [Grønsund2007].

This is achieved using TDMA mechanism that can be easily configured in order to get the best performance for any distance. Furthermore, independence from the distance also is reached with adaptive modulations providing the best possible bit rates at each moment and defining multiplexing and duplexing techniques based both in time and frequency.

In this manner, the only restriction in WiMAX communication systems is the SNR/RSSI at the receiver. Therefore, out of the link budget, there are no intrinsic limits regarding the maximum distance that WiMAX can be effective. So in many cases, the distance limits are established for the equipment available or frequency regulation in each country.

In [Rashidah2010], the effects of the distance in WiMAX point-to-point links are shown with the NCTUns simulation tool and it is demonstrate that the average data rate remains constant regardless the distance.

[Grønsund2007] presents an experimental WiMAX testbed established in a real life environment. Its analysis is focused in the relation between the signal strength and signal to noise ratio, which is the only real WiMAX constraint. Figure 6 shows that the RSSI achieved for each distance follow the known formula $RSSI=n \cdot logD+A$, where n is the path loss exponent, D is the distance in meters and A is the RSSI in dBm at one meter.

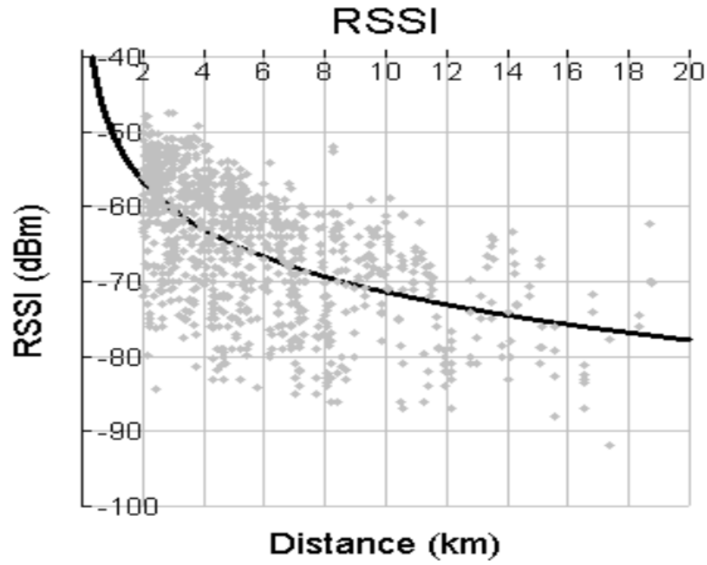

Figure 6: RSSI variation according the distance measured in real testbed, from [Grønsund 2007]

Another WiMAX performance analysis is done in [Yousaf2007], in which a real test bed is deployed. There is an interesting performance analysis for performance of high layer protocols over a fixed WiMAX link. It is demonstrated that the bit rates remains more or less constant regarding the distance, when the modulation level is low. However, the higher is the modulation level, the less stable is the bitrate.

[Yousaf2007] is a very clarifying study about the WiMAX performance. It shows that the same values of bitrates will be achieved from each distance always the SNR remains constant. However, in real links with a static hardware configuration the RSSI is different for each distance, and therefore the SNR is different. This causes bad performance in high level modulations like as 64 QAM 3/4, being advisable use a lower level modulation, such as BPSK ¾, which is more efficient for a lower SNR. This behavior is illustrated in Figure 7. This WiMAX performance analysis concludes that is necessary find a compromise between the bitrate supported and the stability of the link for each distance.

Finally, in [Trinchero2007] it is shown an experimental example which demonstrate the very long distance links can be deployed in WiMAX network, reaching links up to 295 Km. It is achieved stable TCP throughputs of 12 Mbps and 30 Mbps under the European and African regulatory respectively.

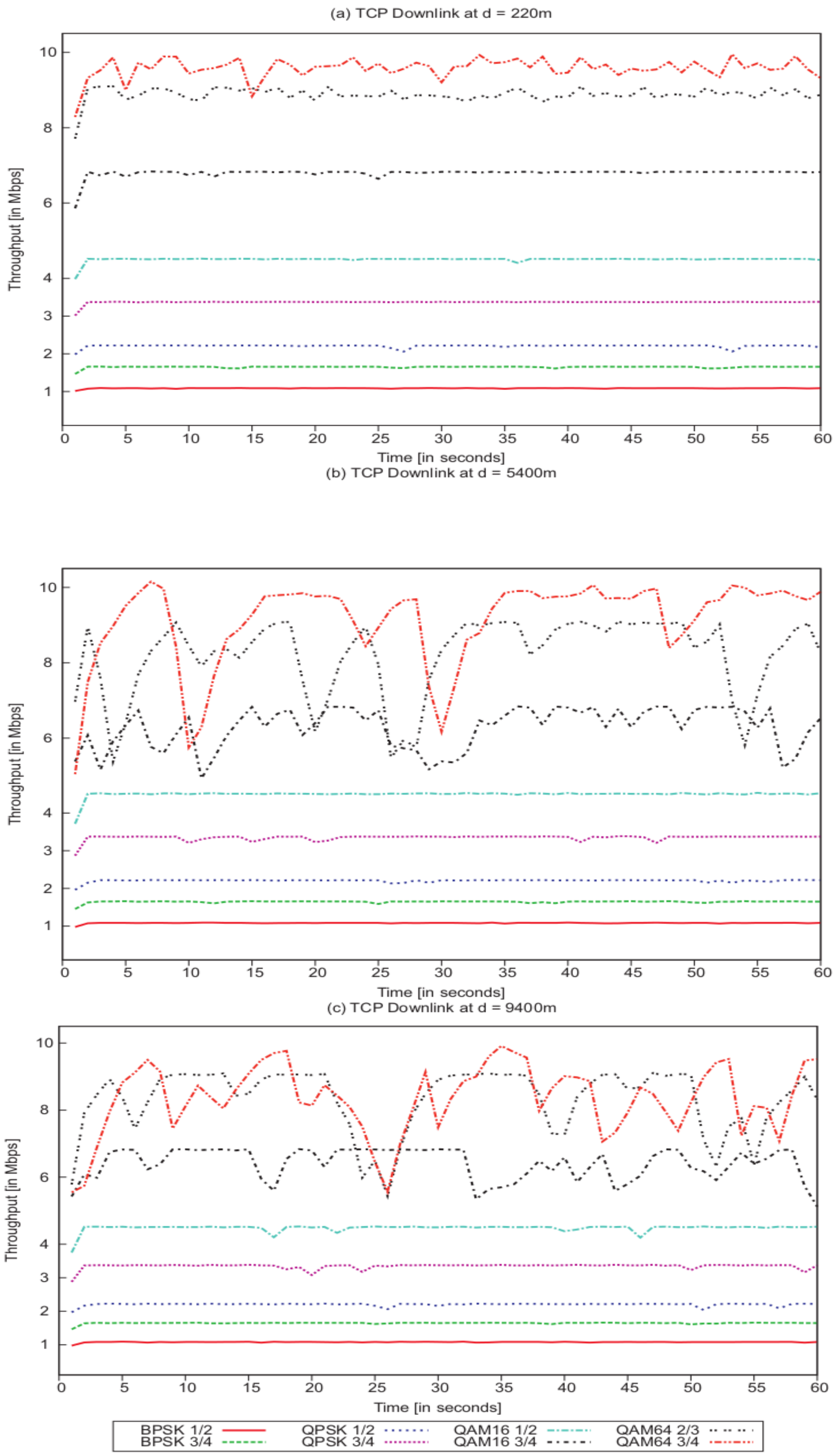

Figure 7: TCP throughput over a fixed WiMAX DL for several distances, from [Yousaf2007]

2.2.3 Literature on expected performance and QoS support in IEEE 802.16

One of the main features in WiMAX is its ability to assure resources for each application. Depending on how these resources are provided, applications will have a specific level of Quality of Service (QoS).

Support for QoS in WiMAX depends basically of the MAC layer and is based in the concept of a service flow and a connection. The connection is a unidirectional logical link between a BS and SS, and is identified by a connection identifier (CID). A service flow (SF) is a unidirectional flow of packets with a particular set of QoS parameters and is identified by a Service Flow Identifier (SFID). These QoS parameters could include traffic priority, maximum sustained rate (MSR), maximum burst rate (MBR), minimum tolerable rate, scheduling type, ARQ type, maximum delay, tolerated jitter, maximum of packet losses allowed, bandwidth request mechanism to be used, etc. The BS is responsible for assuring the SFID and mapping it to unique CIDs, and therefore, the policy access. In [Giambene2013] a theoretical analysis of the performance of WiMAX contention-based access is presented.

For the MAC-Layer scheduler, IEEE 802.16-2009 standard defines five classes in which a SP can be classified:

- **Unsolicited Grand Service (UGS):** In this class, UGS assure periodic frames with fixed size (constant bit rate), in order to avoid the overload and delay introduced by the SS requests. It is usually used for assure data flows which demand specific minimum throughput and maximum delay and tolerated jitter, like T1/E1 emulations and VoIP services without silence suppression.
- **Real-time Polling Service (rtPS):** This class is designed to support real time traffic with periodic frames with variable size (variable bit rate), such as MPEG video. In this case, the SS has a reserved a specific slot for request bandwidth periodically (unicast poll), specifying the desired frame size for the next transmission interval. This mechanism increases the overload regarding UGS, but also increases the efficiency of the channel. The QoS parameters defined are minimum throughput and maximum delay permitted.
- **Extended rtPS (ertPS):** This service is designed to support real-time applications such as VoIP with silence suppression that have variable data rates, but require guaranteed data rate and delay. ErtPS try to mix the advantages of both UGS and rtPS, assuring the resources in the transmission intervals, but also supporting mechanism for change the length of these transmission intervals. This class is not supported by fixed WiMAX.
- **Non-real-time Polling Service (nrtPS):** This class is designed to support data delay-tolerant data streams, such as FTP, assuring the resources even in a saturated network environment.
- **Best Effort service (BE):** This class is designed to support data streams, such as Web browsing, that do not require a minimum service-level guarantee.

Because of the IP-based architecture of WiMAX, each SF class must be mapped to upper-layer QoS mechanisms to enable end-to-end IP-based QoS. This can be done through a network management system or created dynamically through defined signaling mechanisms that will be explained below.

Until now, the air-interface aspect of WiMAX, such as physical-layer techniques to transport bits over the air, and media-access-layer techniques for sharing the available radio resources among multiple users and services have been discussed. However, in order to analyze the end-to-end QoS appreciate by final users, it is necessary to look at the overall network architecture, higher-layer protocols, and the interaction among several network elements beyond the SS and BS.

Since WiMAX is envisioned to provide end-to-end IP services and will likely be deployed using an IP core network, IP QoS and its interaction with the wireless link layer are what is most relevant to WiMAX network performance. Provide this end-to-end QoS requires mechanism in both the control plane and in data plane. While control plane mechanisms are needed to allow the users and the network negotiate and agree a specific QoS level for a specific application, data plane mechanisms are required to enforce the previously agreed QoS requirements by controlling the amount of network resources that each application can consume.

Control plane mechanisms include signaling, QoS policy management and admission control. Signaling is a control plane function which allows a user communicates QoS requirements to a network, may be dynamically or statically. QoS policy management is about defining and provisioning the different levels and types of QoS services. Admission control is the function in which the network has the ability to accept or reject new traffic based on resource availability, so that the performance of existing traffic will not be changed.

Data Plane mechanisms classify traffic inspecting the headers of incoming packets and allocating resources, using appropriate scheduling buffer-management techniques and algorithms.

From these QoS principles, it is possible describe some mechanisms for deliver end-to-end QoS in an IP network. Although transport protocols like TCP or RTP ensure reliable and ordered delivery, do not have any mechanisms for ensure latency and throughput. For this reason it is necessary use new architectures such as integrated services (IntServ), differentiated services (DiffServ) or multiprotocol label switching (MPLS). In each of them, the SP class must be mapped to specific set of parameters which define the QoS requirements. For example, in IntServ each SP class will be mapped to a traffic specification (Tspec) through the Resource Reservation Protocol (RSVP). In DiffServ, the QoS is provided without signaling. In order to treat each MAC-level flow differently, DiffServ map each SP class with a specific DSCP. In MPLS, each SP class is mapped in a classified in a forward equivalence class (FEC).

According to the QoS mechanisms defined by IEEE 802.16, the effects of the QoS parameters in End-to-End IP QoS mechanisms are analysed in several sources like [Cerutti2007], [Haseeb2011] [Chimin2010] or [Sunghyun 2011].

In [Cerutti2007], two solutions are proposed for implementing an integrated end-to-end QoS-guaranteed connection based on MPLS and RSVP. In [Haseeb2011] the integration of a WiMAX access system with DiffServ is investigated, measuring QoS parameters such as delay, jitter and packet drop rate in VoIP services.

Similarly, [Chimin2010] demonstrate that using DiffServ, VoIP services are not affected while users are simultaneously surfing the Internet in the same user profile, hence showing that VoIP services are feasible in WiMAX networks whenever there are multiple QoS classification rules which associate the SP class to the proper DSCP value in the IP network.

In [Sunghyun 2011] also, the end-to-end QoS variations in WiMAX system are analyzed according the DiffServ control functions, proposing examples of the QoS mapping between upper and MAC layers and also simulating the overload introduced by PHY, MAC, and upper layers in order to find improvements in the QoS performance.

Finally, in [Alinejad2011] a more specific QoS environment is described. It is addressed the preliminary performance analysis of WiMAX network for multiparametric telemedical scenarios, mapping and optimizing the medical QoS to typical WiMAX QoS parameters. The results conclude in that it is possible support correctly the medical QoS requirements using a proper mapping and well implemented SF.

Special attention must be paid for the implementation of QoS services in WiMAX networks deployed in developing countries. Although WiMAX has been presented like an interesting alternative for these environments, some fundamental initiatives that improve the deployments of WiMAX networks in rural areas have not been driven by the manufacturers.

One of these initiatives is the mesh topology implementation that offers an interesting solution for the remote regions with low user density and propagation problems due to irregular terrain. The mesh mode is able to provide a larger coverage, but the QoS support is no so easily achieved as in a PtMP topology. However there are numerous studies which address this issue successfully like [Zhang2008], [Sharma2008] or [Shukaili2010]. While in [Zhang2008] a theoretical framework to analyse the performance is shown, [Sharma2008] and [Awadh2010] describe scheduling algorithms and schemes in order to achieve the QoS support for mesh topologies. In spite of this, current distributors and suppliers of these equipments are scarce and expensive, making difficult the implementation of this kind of networks.

Other initiative is the use of relays or repeaters in a PtMP topology in order to extend the BSs coverage, and hence, the network coverage. Actually, the amendment IEEE 802.16j defines the use of relay stations for improve the performance and coverage of the WiMAX network, however, this amendment is only defined for licensed bands. This amendment is well studied in [Okuda2008] and [Peters2009], describing in detail the different operating modes of the relays and evaluating the improvements in terms of coverage and capacity in some typical scenarios. Also in [Schoenen2008] a real implementation is deployed in order to test the performance and the QoS support of the WiMAX system when relays are used.

In [Rey-Moreno2010], a modification in the amendment IEEE 802.16j for its implementation in no-licensed bands is proposed, concluding that is technically viable improve the network coverage keeping the same QoS requirements through the use of relays, but will not be implemented by the manufacturers because of the insufficient support by the ISPs and telephone companies.

However, the use of WiMAX in rural areas for deploys of QoS-support networks has been tested in numerous development projects successfully. In [Eliamani2010] a detailed evaluation of WiMAX QoS in a developing country environment is shown, addressing the technical challenges faced by developing areas by delivering low cost broadband connectivity with good QoS.

2.3 Satellite communications systems: documentation from standardization bodies and scientific literature that may help to foresee the performance

2.3.1 Brief descriptions of the standards

Satellite market for IP traffic transporting (private VPN communication, Internet access, cellular backhauling…) is known not to be dominated by open standards, but for industry standards, developed by the main satellite modem manufactures.

Performance of different solutions can be significantly different even using the same hardware and software for satellite equipment, as the performance for the end-to-end service also depends on the specific configuration, selected satellite and link budget, as well as on optional functionalities or enhancements made by the service operators.

These optional functionalities or enhancements can be provided by the same satellite modem manufacturer or provided by other technology manufacturer, and integrated in the solution as a third-party add-on. As an example, the use of external IP optimization/acceleration "boxes" is today widely used in satellite market by service providers, to enhance the IP performance and user experience of complex applications.

For application to TUCAN3G, there are two architectures to be considered for analysis, TDM/MF-TDMA (star-topology, where all the remote stations "talk" to a central station that also control them) and SCPC (dedicated link or leased line).

There is only one open standard for satellite broadband access under TDM/MF-TDMA architecture, named **DVB-RCS2**, and two other industry standards, that are not interoperable between equipment of different providers: **IPoS** and **S-DOCSIS**.

DVB-RCS2 should not be confused with DVB-S2. DVB-RCS2 uses DVB-S2 as the standard for Hub-to-Remote way of the transmission (forward channel), and then it adds the RCS-2 standard for the Remote-to-Hub way (returnlink channel). DVB-RCS2 takes the DVB-S2 standard, designed for broadcasting (one-way) digital TV, and adds the RCS-2 standard to obtain a bidirectional (two-way) and interactive broadband access.

For SCPC architecture, there are not standards, but the exception of **DVB-S2**, that can be used also for IP trunks. However, using DVB-S2 has some lacks in performance for low rate IP links.Standardization for TDM/MF-TDMA.

2.3.2 DVB-RCS2

The only real open standard (non-proprietary from manufacturers) for broadband services over satellite is **DVB-RCS**, formulated in 1999 by the DVB consortium, and regulated by ETSI EN 301 790 V1.5.1 (2009-05) for its 5th revision:

http://www.etsi.org/deliver/etsi_en/301700_301799/301790/01.05.01_60/en_301790v010501p.pdf

This updated revision of the standard is also known as "DVB-RCS+M" as it includes the "mobility extensions".

DVB-RCS has been updated in March 2011 to **DVB-RCS2**, and published in three parts, as TS 301 545-1 (OSL, overview), EN 301 545-2 (LLS, lower layers) and TS 301 545-3 (HLS, higher layers):

http://www.dvb.org/news_events/news/dvb-rcs2-published/index.xml

In 2012 mobility extensions were added to DVB-RCS2 and named DVB-RCS2+M, supporting mobile/nomadic terminals and direct terminal-to-terminal (mesh) connectivity. DVB-RCS2+M features include live handovers between satellite spot-beams, spread-spectrum features to meet regulatory constraints for mobile terminals, and continuous-carrier transmission for terminals with high traffic aggregation. It also includes link-layer forward error correction, used as a countermeasure against shadowing and blocking of the satellite link.

DVB membership is open to all companies and DVB-RCS/RCS2 was created in an open environment where any DVB member can participate.

There are several manufacturers of interoperable DVB-RCS/RCS2 hubs and terminals. Vendor independence is safeguarded by SatLabs, a non-profit organization of satellite operators, service providers and manufacturers devoted to the promotion of the DVB-RCS/RCS2 standard. SatLabs operates a qualification laboratory, where terminals can be tested to prove their operation in accordance with the standard.

Furthermore, SatLabs defines supplementary recommendations that build upon the solid foundation of DVB-RCS/RCS2 and offers conformance testing.

DVB-RCS/RCS2 is a mature open source satellite communication standard, with an efficient bandwidth management. It was created to be a cost-efficient base for manufacturers to create compatible systems and terminals that can interoperate between them, but sometimes the manufacturers, or the service providers, add other functionalities to their systems, making them incompatibles with the others.

The network architecture of DVB-RCS2 is Hub-spoke or star-topology, where all the remote terminals receive a shared forward channel with a DVB-S2 carrier and transmit using different return link channels (carriers) using MF-TDMA.

The core of DVB-RCS2 is a Multi-Frequency Time Division Multiple Access (MF-TDMA) transmission scheme for the return link, which provides high bandwidth efficiency for multiple users.

The demand-assignment scheme uses several mechanisms that allow optimization for different classes of applications, so that voice, video streaming, file transfers and web browsing can all be handled efficiently. DVB-RCS supports several access schemes making the system much more responsive, and thus more efficient than traditional demand-assigned satellite systems. These access schemes are combined with a flexible transmission scheme that includes state-of-the-art turbo coding, several burst size options and efficient IP encapsulation options. These tools allow systems to be fine-tuned for the best use of the power and bandwidth satellite resources.

In addition, DVB-RCS2 also includes Continuous Phase Modulation (CPM) for use with amplifiers in saturated mode.

DVB-S2 Forward Channel characteristics:

- Modulation: QPSK, 8-PSK, 16-APSK, 32-APSK.
- Codification based on LDPC (Low Density Parity Check) concatenated with BCH (Bose – Chaudhuri – Hocquenghem) codes.
- FEC coding rates from 1/2 to 9/10.
- ACM (Adaptive Coding and Modulation) with modulation/codification optimized frame by frame; VCM (Variable Coding and Modulation); and CCM (Constant Code and Modulation).

In Table 13 DVB-RCS/RCS2 Return Link Channel characteristics are shown.

Feature	DVB-RCS	DVB-RCS2
Harmonised management & control	None	Yes (optional)
Harmonised IP-level QoS	None	Yes
Multiple virtual network support	None	Yes
Security	Single solution	Support for multiple security systems, for applications with widely different requirements
Return link access scheme for traffic	TDMA, continuous carrier	TDMA, continuous carrier, random-access
Modulation schemes	QPSK	Linear: BPSK, QPSK, 8PSK, 16QAM. Constant-envelope: CPM
Channel coding	RS/convolutional, 8-state PCCC turbo code	16-state PCCC turbo code (linear modulation), SCCC (CPM)
Burst spread-spectrum	Burst repetition	Direct-sequence
Return link adaptivity	Limited support	Inherent in air interface (TDMA and continuous carrier)
Bandwidth efficiency	N/A	30% improvement over DVB-RCS

Table 13: DVB-RCS/RCS2 Return Link Channel characteristics

A module named NCC (Network Control Center), installed in the Hub, manages connection of the terminals to the network and assures timing synchronization and proper resource management for all the terminals.

The use of ACM in the forward channel is widely used to increase availability for services in areas of heavy rain, as it allows reducing the payload bits (less user traffic) in change of increasing the coding bits (more overhead), to deal with the attenuation. With the improvements of new RCS2, ACM is also possible to the return link channel.

It is important to remark that while most of the manufactures claim to be DVB-RCS/RCS2 compatible, it is difficult to find a commercial service working with different satellite modem manufacturers. The cause is that sometimes the manufacturers, or the service providers, develop proprietary functionalities for improving the performance of the standard and/or focused for specific applications. As these functionalities are not open neither standardized, if you used them you are linked to that provider. Also, there are times that preview versions of functionalities to be included in new revisions of DVB-RCS/RCS2 are used by some of the manufacturers, prior to be standardized.

These non-standard functionalities are intended to enhance performance in required Es/N_0, Bits/Hz ratio, minimization of delay and jitter, optimization at different levels, and performance in terms of TCP/UDP throughput. Even they are not standard, their use can be very useful for some special needs.

Examples of well-known VSAT manufacturers with compatible DVB-RCS/RCS2 solutions in their portfolios are Gilat SkyEdge-II DVB-RCS, Viasat LinkStar and STM SatLink. Well-known VSAT manufacturers that do not use DVB-RCS2 standard are Hughes Network Systems (HNS) and iDirect.

2.3.2.1 Changes and improvements of DVB-S2 vs. DVB-S

The main improvements and changes of the new DVB-S2 versus DVB-S are:

- New modulation schemes (8PSK, 16APSK and 32APSK).
- More roll-off factors (20, 25 and 35%).
- Introduction of ACM, CCM and VCM.
- LDPC (Low Density Parity Check) replaces Viterbi inner coding.
- BCH (Bose-Chaudhuri-Hocquenghem) replaces Reed Solomon outer coding.
- More inner code rates: 1/4, 1/3, 2/5, 1/2, 3/5, 2/3, 3/4, 4/5, 5/6, 8/9, 9/10.

Because of the new improvements, DVB-S2 has better spectral efficiency: up to 40% bandwidth saving (30% from better coding and 10% from smaller roll-off factor) or up to 2.5 dB margin gain (less than 1 dB away from the Shannon limit).

LDPC uses very big block size (16200 and 64800 bits). This means less fragmentation and then less overhead, but large frame size means more delay. Use of short frames is less performing than normal frames (about 0.3 dB) but with 1/4 of the delay.

APSK allows satellite link operation closer to saturation of the transponder. This is much better than QAM:

a) 16QAM in DVB has limited use in operation:
   - High carrier to noise levels required.
   - High demands on linearity: large back-off, huge HPAs and antenna sizes.
   - High demands on phase noise.

b) 16APSK in DVB-S2 is fully enabled:
   - Lower carrier to noise levels required.
   - Easier to decode by demodulator due to less different amplitude and phase levels.
   - More resistant to phase noise.
   - Availability of pilots.

DVB-S2 introduces **ACM, CCM and VCM**:

- **CCM (Constant Coding and Modulation):** All frames use the same parameters. It is mainly used in video broadcasting, as it is simple and requires a cheap demodolation ASIC chip.
- **VCM (Variable Coding and Modulation):** Different streams/services are coded with different parameters. It is used for IP trunking and video distribution.

- **ACM (Adaptive Coding and Modulation):** Each frame in a stream is coded with its own set of parameters. Parameters are modified dynamically according to the reception conditions for each receiver. It is used in broadband access for VPN or Internet services, especially when the shaping is made dynamical.

In DVB-S, the performance of a demodulator for a certain ModCod is expressed in Eb/N_0 (energy per bit over the normalised noise). In DVB-S2, $\mathbf{Es/N_0}$ is used (energy per symbol over the normalised noise).

Spectral efficiency is based on the ModCod, Frame Size, Pilot On/Off, and Padding On/Off.

As an example, Table 14 shows the comparison between two configurations in DVB-S and DVB-S2 with similar Es/N_0:

- DVB-S: QPSK 3/4, 29.27 Msps, 35% roll-off.
- DVB-S2: 8PSK 2/3, 30.0 Msps, 20% roll-off, pilots present.

DVB-S QPSK 3/4	DVB-S2 8PSK 2/3
Es/N_0=7.26 dB	Es/N_0=7.10 dB
C/N_0=82.03 dB-Hz	C/N_0=81.87 dB-Hz
OB=39.514 MHz	OB=36.000 MHz
Data Rate=40.4511 Mbps	Data Rate=58.0710 Mbps

Table 14: Comparison between DVB-S and DVB-S2

DVB-S2 data rate is 18.4199 Mbps more than DVB-S, which is 45.54% increment.

In DVB-S, the data format was exclusively the MPEG Transport Stream (TS). The size of the MPEG transport stream packet (188 bytes) was optimized for the Reed Solomon error correction code, which is no longer used by DVB-S2.

DVB-S2 introduces a new **Generic Mode** for Generic Continuous Stream and Packetized Stream.

The Generic Mode is compatible with any type of data (IP, ATM…). It does not need transport stream overhead (2%) and the efficiency gain for IP could be more than 4%.

DVB-S2 does not define an encapsulation mechanism for IP data such as MPE as in DVB-S. This issue was studied by the standardization group (TM-GBS) and two proposals were presented: EDGE (ESA) and GULE (IETF). They were merged and enhanced to create the **GSE (Generic Stream Encapsulation)** for the Generic Mode.

The GSE protocol has been devised as an adaptation layer to provide network layer packet encapsulation and fragmentation functions over the generic stream of the DVB-S2 standard:

- ETSI TS102 606 for protocol specification.
- ETSI TS 102 771 for implementation guidelines.

Performance of DVB-S2 near Shannon limit has a challenge: Maintain carrier recovery under phase noise of LNB and tuner. **Pilots** are introduced to deal with this challenge. Subject to the same thermal/phase noise as any other symbols, pilots help the demodulator in carrier recovery and receiver synchronization.

Each pilot shall be an un-modulated symbol, identified by $I = (1/\sqrt{2})$, $Q = (1/\sqrt{2})$.

Need for pilots increases with:

– Higher modulation schemes: 16APSK and 32APSK.
– Low code rates: 1/4, 1/3, 2/5, 1/2 and 3/5 for QPSK; 3/5, 2/3, 3/4 and 5/6 for 8PSK.
– Low symbol rates: <5 Mbaud for free-running DRO LNB or <3 Mbaud for PL-DRO LNB.

For operation in DVB-S2 under ACM/VCM, pilots must be on.

Pilots do not degrade Es/N_0 performance when used but not needed, but use of pilots adds around 2% of overhead.

DVB-S2 allows using two types of frames: **Normal Frames** and **Short Frames**. The comparison between the two philosophies is shown in Table 15:

	Short Frames (16200 bits)	**Normal frames (64800 bits)**
Performance	-	+ (average 0.3 dB better)
Spectral Efficiency	-	+
Delay (modulation+demodulation)	+ (only 25% of Normal Frames)	-
Broadcast	Not supported	Mandatory

Table 15: Comparison between short and normal frames in DVB-S2

Recommendations:

– Short frames are used only for time critical data applications.
– Broadcast is mandatory for normal frames.
– Do not mix short and normal frames for ACM/VCM.

While the performance of LDPC-BCH coding is very efficient, the main issue with this codification is the delay.

Modulation–Demodulation Delay Calculation:

Delay = (#frames x [frame-size] x spectral efficiency) / (m x bitrate)

Modulator delay is 2 frames, while Demodulator delay is 3 frames.

Example: 256kbit/s, 8PSK 2/3, Pilots on, Short frames

#frames=5,

Frame-size=16200 bits,

spectral efficiency=1.880672 bit/Hz,

m=3 (8PSK),

bitrate=256000 bit/s.

Then delay is calculated as 0.198s = 198 ms.

2.3.2.2 Disadvantages of DVB-S2

While the scheme defined by DVB-S2 is undoubtedly very effective for many broadcast and higher data rate applications, it is definitely not a "one size fits all" solution. Here are some of the disadvantages:

- *Excessive latency*. The so-called short blocks are too long for low latency IP applications at low data rates. This is exacerbated by the addition of interleaving. As an example:

  Latency for 64k block DVB-S2 ACM at 100 ksps = 1336 milliseconds

  Latency for 16k block DVB-S2 ACM at 100 ksps = 349 milliseconds

- *Complexity in its implementation*. The design of DVB-S2 dictates that all FEC blocks should be constant in bits. This means that for each ModCod, there are a varying number of symbols. This then makes the task of synchronization a much more demanding task. Also, because of the limitations of tracking the higher-order modulations in a very low SNR environment, so-called pilot symbols were added in order to aid tracking.

- Since the introduction of the original LDPC-BCH scheme, an enormous amount of research has been done on the design of LDPC codes. The concatenated BCH code was added to mitigate the problem of error rate 'flaring' and 'flooring'. This is no longer necessary. Most importantly, however, LDPC codes can now be designed that yield almost equivalent coding gain, but with considerably shorter block lengths.

- In an ACM mode, no overhead channel was defined by DVB-S2 for the purpose of reporting SNR metrics to the originating end. It has been left to individual equipment manufacturers to decide their own methods. This illustrates that all ACM systems, DVB-S2 or not, are proprietary. In addition, it implies that additional bandwidth needs to be consumed for the SNR reporting, and this is not accounted for in the code rate.

2.3.2.3 Changes and improvements of DVB-RCS2 vs. DVB-RCS

To appreciate the improvements of RCS2, it is necessary to understand the foundation first, RCS.

The DVB-RCS standard is all about implementing very fast, efficient and reliable Internet Protocol (IP) networks using Very Small Aperture Terminals (VSATs). DVB-RCS provides excellent support for fixed, transportable, and mobile VSATs with high-throughput communications. Communications-On-The-Move (COTM) is delivered for ships, off-road vehicles, high speed trains, and aircraft with roaming across beams and countermeasures against interference.

DVB-RCS works with any type of fixed, auto-pointing, electronic array, or stabilized antenna systems in any satellite band or beam configuration. DVB-RCS specifies use of the popular DVB-S2 standard for the Time Division Multiplexed (TDM) carriers in continuous mode, which can be broadcasted to hundreds or thousands of remote VSATs from a central hub, gateway, or master station.

It then also specifies the advanced TDMA control logic, precise burst synchronization, modulation and coding, plus fast frequency-hopping used on TDMA carriers. Timeslots may be allocated to a given VSAT statically and/or dynamically (i.e. as bandwidth-on-demand) in any mixture determined by policies set by network operators for different applications or traffic types.

Applications requiring any combination of VoIP, video conferencing, video feeds and/or video multicasting, plus any type of data application can be served with a single VSAT. Each capacity request from the VSAT for bandwidth-on-demand may distinguish the QoS treatment required for the service.

DVB-RCS is ideal for remote operations that need huge transmit bandwidths rapidly but unpredictably. A TDMA "carrier group" can have tenths of carriers creating a massive pool of shared bandwidth for return and mesh link communications. More than 36 MHz of shared TDMA capacity is feasible in commercial implementations today that use wide-band, multi-carrier burst demodulators, with more coming soon.

In the latest commercial burst demodulator systems, that 36 MHz can be digitized and processed for several concurrent bursts at the Hub or a number of gateway stations, providing even 50 Mbps of IP throughput on TDMA carriers.

Mesh networking capabilities, available in commercial DVB-RCS implementations today, over these same TDMA carrier groups, enable single-hop communications between small remote VSATs with capacity for demodulating several concurrent bursts. Mesh networking also supports bandwidth-on-demand with multiple QoS Class on every active mesh link, if desired.

In recent years, Adaptive Coding and Modulation (ACM) on DVB-S2 carriers operating at information rates well over 100 Mbps in TDM mode has proven to be a key advantage in many networks, often doubling the TDM carrier capacity and greatly improving TDM link availability. DVB-RCS, however, specifies how ACM on DVB-S2 is implemented optimally with rapid closed-loop control using the same signaling messages that handle closed-loop synchronization and capacity requests for TDMA timeslots.

In DVB-RCS, the modulation, coding, and symbol rates on TDMA bursts are also handled adaptively, in real-time, using intelligent carrier selection in the TDMA controller at the Hub. This completes increased link availabilities and average information rates obtainable for two-way communications, and delivers better gains than other proprietary adaptive methods, because of the larger dynamic ranges supported. This allows VSATs to perform fast frequency hopping across a wide range of TDMA carriers, operating, say, from 500 Ksps up to 8 Msps in a highly reliable manner.

Other features in the leading commercial DVB-RCS implementations at higher layers include built-in TCP and HTTP acceleration, Network Address Translation (NAT), and header compression of IP stack protocols. These are essential for high-performance, flexible networking. Many aspects of such features are now gradually being standardized by efforts in the DVB Higher Layers for Satellite (HLS) working group, in close coordination with the RCS working group.

Transmit capacities over 10 Mbps of IP content from a low-cost VSAT are feasible in today’s low-cost commercial implementations of DVB-RCS; and the receive capacities for total IP content over DVB-S2 to a single VSAT can reach 50 Mbps. Of course, throughput always depends on signal-to-noise ratios (SNR), which in turn depends on antennas, amplifiers, and many other factors including satellite characteristics and atmospheric fading, especially in higher frequency bands such as Ku and Ka.

This is where the DVB-RCS2, the Second Generation of DVB-RCS, becomes especially important.

The major advance in DVB-RCS2 specification over DVB-RCS is that DVB-RCS2 offers **8PSK and 16QAM modulation options for TDMA carriers**, as well as the usual QPSK. It also uses a new, powerful 2D 16-state Turbo Code FEC algorithm in TDMA bursts, giving up to 2 dB more gain. Selected alternative modem specifications (e.g., for non-linear modulation) are also allowed but yield lower efficiency. VSAT log-on signaling now handles an even wider-range of diverse terminal capabilities via auto-discovery, promoting simplicity of operation in networks with diverse terminal types, including some or all using encryption.

Particularly important is that **ACM is now also implemented on the TDMA carriers** for each timeslot (i.e., “per burst”). This is done in combination with the dynamic adaptive carrier selection already in use today.

TDMA carriers, therefore, have no defined modulation and coding until a timeslot is assigned on them to a given VSAT. Even the symbol rates and the number of carriers in the TDMA carrier group may be dynamically adjusted, given the flexible superframe and frame formats in the new standard. All this gives an additional dramatic improvement in average network capacity and link availability over TDMA carriers in high-frequency networks (e.g., Ka band), without increasing antenna sizes.

With these advances, the transmit capacities from a low-cost VSAT have reached 20 Mbps of IP content, with more in the years to come.

Another benefit comes from **new methods for encapsulating IP packets into TDMA bursts** and on DVB-S2 carriers, which reduce layer 2 overheads to the bare minimum and eliminate MPEG overheads, while retaining flexibility for diverse Layer 3 protocols, such as IPv6 and IPv4 running concurrently over the same carriers.

The upgrade to DVB-RCS2 could be a game-changer in the satellite communications industry, raising the bar for efficiency and speed while remaining a trusted, open standard. With improvements across several key aspects of the DVB-RCS specifications, RCS2 could be a valuable asset across many market sectors.

Upgraded or new RCS2 systems can provide solutions for high-speed Internet Access, TV/Video services, VoIP, and cellular backhaul used by the maritime industry, energy sector, utility companies, and global corporations. All of these services can operate on the same VSAT with excellent QoS.

DVB-RCS2 provides for higher order modulation and coding along with many other elements contributing to higher throughput, better bandwidth efficiency, and improved link availability relative to legacy VSAT systems. Modulation and coding from QPSK 1/3 up to 16QAM 5/6 is provided for TDMA burst mode carriers with Adaptive Coding and Modulation (ACM) controlled per burst for each VSAT. The other elements include a 16-state Turbo Code FEC and highly efficient Return Link Encapsulation (RLE) for IP packets imposing about 2% overhead.

Operators of both large and small networks can realize huge benefits with DVB-RCS2 implementation. In small networks with few VSATs and one or two TDMA carriers, the “ACM per burst” feature makes it so operators do not have to choose the “best” modulation and coding to use; it is determined automatically per VSAT with each burst. Large networks benefit in this same way, plus they get additional multiplexing efficiency gains by using larger pools of TDMA bandwidth, dynamically sharing tenths of Mbps across thousands of VSATs, if desired.

2.3.3 IPoS (Internet Protocol over satellite)

IPoS (Internet Protocol over Satellite) is a satellite Internet protocol that was developed by Hughes Network Systems (HNS), manufacturer of HN/HX broadband satellite systems (known in the past as Direcway).

HNS wanted to create a standard for broadband, applicable to the whole industry and based on IP-centric technology.

IPoS resides in the network layer - over physical and data link layers, which are specific to satellite technology. It features a well-defined interface between satellite dependent functions and the application layers, named SI-SAP (Satellite Independent-Service Access Point). As a result, SI-SAP creates a logical separation between the user interface and the satellite interface and allows applications designed to be compliant with SI-SAP to be easily ported. IPoS manages Layers 1 and 2, and adheres to SI-SAP when connecting to the open world. According to HNS, IPoS is scalable, supporting applications from a single home office to multinational VPN customers.

IPoS got endorsed by the Telecommunication Industry Association (TIA) of North America in December 2003. The European Telecommunications Standards Institute (ETSI) ratified the TIA 1008 IPoS air interface specification as ETSI TS 102 354.

HNS has started a multi-tiered licensing program available to industry applications and systems providers.

Although there is no other satellite broadband player on board this initiative yet, and IPoS has yet to achieve market acceptance outside of Hughes Network Systems, HNS foresees a significant interest in IPoS. HNS has lined up support for IPoS from several technology companies, including Microsoft, Intel, Texas Instruments and HP. It has also generated support from big-name players in the satellite sectors such as Intelsat and SES Americom.

Critics call IPoS a marketing gimmick from HNS. However, HNS tout IPoS technology as the first truly global broadband satellite standard and say an industry-standard open interface is essential for the future of the broadband satellite industry. Critics in the industry, however, have described IPoS as an HNS tactic to rob momentum from DVB-RCS/RCS2.

2.3.4 S-DOCSIS (Satellite - Data Over Cable System Interface Standard)

S-DOCSIS is a modification of the DOCSIS (Data Over Cable System Interface Standard) cable-modem protocol for transmission over satellite. DOCSIS is technically not a satellite standard, but a cable modem standard from CableLabs.

To work properly over satellite links, the DOCSIS protocol was modified to support, in addition to QAM, modulation schemes such as QPSK. DOCSIS over Satellite allows satellite broadband operators to leverage technologies already used in cable modem networks, as well as off-the-shelf back office and network management software, which gives it economies of scale. DOCSIS also provides QoS controls that allow deployment of tiered services, value-added services such as VoIP, security features and other configurations for broadband networks. In addition, variable modulation protocols allow support for a wide range of satellite transmission attenuation conditions.

S-DOCSIS has a lot of possibilities to be a winner among satellite Internet standards for broadband access services, especially for consumer, Small-Office Home-Office (SOHO) and Small Medium Business (SMB) markets, due to the enormous advantages to economy-of-scale made available through sharing development costs with all of the existing cable DOCSIS systems.

Surfbeam and Surfbeam-2 systems developed by ViaSat are DOCSIS 1.1 systems. They are the base for satellite broadband services *Exede* over Viasat-1 (USA) and *Tooway* over Eutelsat Ka-Sat (Europe). Both systems use Ka multi-beam satellites and can achieve traffic rates (download/upload) of 12Mbps/3Mbps (Surfbeam, Exede) and 20Mbps/6Mbps (Surfbeam-2, Tooway).

Critics say the DOCSIS satellite specification falls short because DOCSIS was designed for cable networks, not satellite. They pretend that DOCSIS does not account for attributes unique to a satellite environment, such as the variable link attenuation due to atmospheric conditions and long round-trip delay. Also, DOCSIS over satellite is comparatively less field-proven than its competitors.

2.3.5 Standardization for MCPC/SCPC

There is no official standard for transmitting IP traffic using SCPC links over satellite, but the exception of DVB-S2 used only for high data rate links.

Standards for SCPC links over satellite were developed in the past, for special applications like analog voice or fractional E1/T1 transportation, using G.703 interface, and while they are still in service, for legacy purposes, they are not valid for IP transport of broadband access in a cost-effective way.

Most of the satellite modem manufacturers have SCPC modems with integrated IP router/processor than can be used for establish dedicated links and transport IP traffic in a cost-effective way. For high data rates, DVB-S2 standard (or proprietary evolutions developed by some manufacturers) is used, while for low data rates most of the manufacturers have developed modifications of DVB-S2, focused on reducing the delay of LDPC-BCH coding.

Examples of satellite modem manufacturers than can be used for SCPC IP links:

a) **Comtech EF Data** (http://www.comtechefdata.com):

- CDM570/CDD564. It is a multi-purpose satellite modem. It can be used for pure SCPC links and also for MCPC/SCPC links in star-topology. It has been widely used for IP links and cellular backhaul, because of the optional “IP Module” (Comtech terminology) that acts like a basic IP router. It has also options for compressing IP header/payload and can use TPC (Turbo Product Code) and 8-PSK modulation. It is an old product, without development of new features, but it is still a cost-effective solution in same cases.

- CDM625. It includes two important (optional) features, both patented by Comtech EF Data, so they are not open: “VersaFEC” and “DoubleTalk Carrier-in-Carrier”. “VersaFEC” is an adaptation of LDPC-BCH (the FEC used on DVB-S2), for low delay at low traffic rates. “DoubleTalk Carrier-in-Carrier” is a functionality that allows the superposition of both carriers of the link in the same frequency spectrum. From the satellite point of view, the frequency spectrum appears to be an interference, as both carriers occupy the same spectrum, but on each side of the link, the modem can digitally “cancel” (“subtracts”) its own transmitted carrier, and obtain the clean carrier transmitted from the other side of the link. This modem cannot be used for MCPC/SCPC topology, only for pure SCPC point-to-point, and lacks of IP routing capabilities, only Layer 2 (bridging).
- CDM800/CDD880/CDM840 (also named “Advanced VSAT” by Comtech). It is an evolution and mix of the previous ones. It has the possibility of being configured as MCPC/SCPC in star-topology, as well as it includes IP router and VersaFEC. It adds ACM (Adaptive Code and Modulation). It has been developed to support, among others, connectivity for 3G cells using IP interface.
- CDM760. It is a modem designed for high speed IP trunking. It is compatible with DVB-S2 (EN 302 307 standard), but adds a proprietary “DVB-S2-EB1” (“Efficiency Boost”) patented by Comtech. Comtech-propietary “EB-1” extension adds new MODCODs (combinations of Modulation-Codification) and new ROFs (roll-off factors) to the DVB-S2 standard. CDM760 also has “DoubleTalk Carrier-in-Carrier”.

b) **Newtec** (http://www.newtec.eu):

Newtec is a well-known manufacturer of satellite modems, and a reference provider for DVB-S2 modems used for TV transmission over satellite. In addition to standard DVB-S2 compatible modems, Newtec has developed its own pack of enhancements for the standard, named “S2 Extensions”. This pack adds new ROFs and new MODCODs, as well as the possibility of using 64APSK modulation.

c) **Novelsat** (http://www.novelsat.com):

Novelsat is a relatively new manufacturer, and has developed as the rest of satellite modem manufacturers its own enhance of DVB-S2, named “NS3”, and their own functionality for superposition of both carriers of the link in the same frequency spectrum, named “Duet Echo Cancellation”.

3 EXPECTED PERFORMANCE IN 802.11 LONG-DISTANCE LINKS

3.1 Performance expected from IEEE 802.11-2012

In this section, each member of the 802.11 family will be examined in order to give a prediction as accurate as possible about the performance (essentially throughput and delay) that can be obtained in long distance links. Each case will be studied with theoretical analysis, with simulations and, when possible, with experiments using real equipment.

3.1.1 Performance parameters that may be obtained

802.11 EDCA does not permit to assure certain limits for the QoS parameters, due to the statistic nature of the CSMA/CA protocol. Although HCCA would be advantageous in this sense, we are going to focus on EDCA because any realistic solution currently implemented may only use this option.

Analytical models or experimental results (obtained from simulators or from real testbeds) permit to estimate the average, maximum or minimum values for throughput, delay, jitter and packet-loss under certain conditions. Beyond that, when EDCA is used, those results may vary from one access category (AC) to another, depending on their priority and the amount of traffic of each AC.

Hence, in the following we will try to obtain an approach to the performance parameters that can be obtained for each bitrate of the different versions of 802.11, firstly by using theoretical models, then by doing simulations or real tests. Then, some results will be obtained for EDCA. Finally, theoretical and practical results will be compared.

3.1.2 Methods and materials for simulation

Network simulators implemented in software are valuable tools in order to develop, test and diagnose networks without the need of carry out real experiments with actual hardware which can be expensive and often long lasting. Simulation results are easier to analyze than experimental results because important information at critical points can be easily logged and usually can be obtained quicker.

There are a substantial number of available network simulators, each one with its advantages and its disadvantages. Most of these network simulators are actually discrete events simulators which represent a system as an event chronological sequence. Each of these events, sending a packet for example, occurs in a specific instant and changes the state of the system. During the simulation, all this events are logged in a set of files which can be processed in order to get network parameters such as throughput, delay, jitter, packet-losses, etc.

Some of the most important features of a network simulator are its license, multi-platform support, real wireless standards support, multi-interface nodes support, available routing levels, available physical layers, channel and error models, community support and user interface. Furthermore, in the context of this work, the effect of the distance between nodes in the network performance is an essential requirement.

The network simulators considered are OMNeT++, NCTUns, NS-2 and NS-3. All of them have academic license or GNU/GLP license and are based in a C++ core.

OMNeT++ is a powerful simulation tool that uses its own programming language NED in order to simulate the network. Gives support both 802.11 and WiMAX, but its implementations are limited, since has not support for Mesh or PtMP topologies.

NCTUns works with C++ and has good support for both 802.11 and WiMAX, including all its possible topologies. However, it is only possible to use it through its graphical user interface and It doesn’t have a good community support as in other open source simulators.

NS-2 is one of the most famous network simulators. It based in C++/Tcl and there is available a big amount of source code and documentation, including for 802.11 and WiMAX. However, due to its big size, not all NS-2 implementations are compatible among them, and there are some design deficiencies recognized by the community which hinder the scalability of the project.

Finally, NS-3 is a new network simulator created in 2006 which was designed from its beginning in order to amend the NS-2 lacks. Although it is totally based in C++, it is possible to use a high-level Python interface. Its modularity and its object-oriented architecture makes NS-3 a very scalable and versatile tool which get rid out of the NS-2 tight dependencies. NS-3 has a good implementation of the standards 802.11 and WiMAX with an especially analogous MAC layer, and the effects of the channels and the distance between nodes are well reflected in the network simulation. Furthermore its community supports an extended documentation and detailed API, as well as a big amount of technical articles. For these reasons, NS-3 has been chosen in order to carry out the simulations.

NS-3 requires several C++ scripts which will be run by the NS-3 simulation core. These scripts will contain all the information about elements present in the network which will be simulated. This information is relative to nodes, devices, applications, topology, models, channels, stack protocols, etc. Furthermore is useful add some elements which permit log, trace and debug the events of the simulation. In order to ease the development of these scripts, often a tough issue, it is possible use an IDE, such as Eclipse.

3.1.3 Methods and materials for experimental tests

In order to give a realistic overview of both analytical models and simulations, the URJC laboratory has been prepared for experiment with real equipment. The aim of the laboratory is test numerous sets of experiments of heterogeneous and homogeneous networks using IEEE 802.11, WiMAX and VSAT technologies, as well as a variety of link topologies such as point-to-point, point-to-multipoint and even Ad-hoc. Only aspects related with the experiments designed for this D51 document. Later experiments will be accompanied with a description of the specific laboratory environment.

3.1.3.1 Equipment

The laboratory equipment for the required tests consists of:

- Four Mikrotik boards (RouterBOARD 493G) each one with a Mikrotik card (RouterBOARD R52Hn). Two boards are used for work with MIMO (2x2) links and the other two with SISO links.

- Two Ubiquiti Rocket M5s capable to work with both SISO and MIMO.

- 10 omnidirectional indoor antennas, eight of them used for work with two MIMO (2x2) links (Ubiquity and Mikrotik) and two for one SISO links (Mikrotik).

- Three Acer Extensa E264 computers with Ubuntu 12.04 (64bits) connected to the experimental network through Gigabit Ethernet. Two of them are used for test connectivity and performance between equipments. The other computer is used as log server.

- Ethernet UTP cabling category 5.

- Coaxial cable used for RF-shielded connections.

- Two RF-shielded boxes used for isolate the equipments from external interferences. Size 50cm x 35cm x 25cm. The RF-shielded boxes are used together with two shielded coaxial cables which unite them. In order to avoid the power saturation and damages in the equipments two 60dB attenuators are used.

- Two Alix boards (2d2) each one with one with a Wireless PCI card (Routerboard R52) and with an omnidirectional indoor antenna. They only support 802.11 a/b/g and neither MIMO nor spatial diversity technique can be used. For these reason, their use will be limited.

– Spectrum probe for detect interferences and choose the clearest wireless channels both in 2.4 GHz and 5 GHz. AirOS, the operating system of Ubiquiti equipment is equipped with a spectrum analyzer tool called AirView.

3.1.3.2 Topology

Bus topology is used in this laboratory environment to connect the equipment (routers, antennas, computers, etc.), where all the hardware share the same IP network through a Ethernet switch. While the management of the equipment such as SSH sessions, connectivity tests, route tests, etc. is made through the wired segment, the performance tests test use the wireless interfaces of both Mikrotik and Alix. This is achieved through IP-aliasing techniques and setting static routes for each task.

3.1.3.3 Software and traffic injectors

The software used consists basically in the traffic injectors and a synchronization tool. Two traffic injectors have been set up in the computers for performance tests:

– **Iperf:** It is a famous tool for traffic injecting. It is possible define unidirectional or bidirectional flows, transport protocol (TCP/UDP), packet size, bitrate, etc. It is necessary use both client and server version of the tool in different computers. Using Iperf allow us measure throughput, jitter and packet loss. The log process must be managed by the user.

– **DITG (Distributed Internet Traffic Generator):** It is a open-source platform capable to produce traffic at packet level accurately. It is a more complex tool than Iperf since allow the user define sending interval, packet size through a variety of stochastic processes and random variables such as exponential, uniform, Cauchy, normal, etc. As Iperf, is necessary use both client and server version of the tool in different computers. In order to get accurate parameters of delay, it is necessary synchronize the computers in which the software is installed, although the last version (2.8.1) launched in July 2013 can measure one-way or round-trip delays.

– NTP: It is a protocol for synchronizing the clocks of computer systems over packet-switched, variable-latency data networks. It is designed particularly to resist the effects of the jitter. There are open-source implementations for many operating systems including Linux. This synchronization will allow us measure delays precisely in the network.

3.1.3.4 Methodology

In order to carry out the tests systematically the next methodology has been defined:

- *Physical preparation*. It is necessary assure that the hardware is operating and the computers and the wireless boards have wired connection.

- *Set up the wireless link between the wireless equipment*. Depending of the hardware and software this can be done automatically, but this is the most important phase of the test, because is necessary a proper configuration for obtain valid measurements. A basic IP configuration is needed but the link level will determine the test results. For a good management of the wireless channel, at least the parameters shown in Table 16, 17, 18 and 19 must be configured. This configuration can be done through the web interface or the command line of the equipment. Also, it will be necessary assure a good line-of-sight between the hardware in order to avoid undesired additional interferences. For 2.4 GHz band, which is highly congested by undesired signals, will be used the RF-shielded boxed and the coaxial cables.

802.11 a/b/g	
SSID	The identifier of the wireless network. Usually has is a string value or a 48 bits identifier according if the network is in Ad-hoc or AP mode.
Channel/Freq	The frequency channel in which the link will be established within the 2.4 GHz band or 5GHz band. The less interfered channel will be selected. It is necessary check with a spectrum probe the level of interference in each channel in order to choose the less congested. This way, the measure will be the most accurate possible.
Mode	The operating mode will be always infrastructure mode or AP mode. Although also would be possible set the Ad-hoc mode, this mode do no support QoS mechanisms by default and is more simple. Within the AP mode, one station must be selected as AP (Access Point or Master) and the other as STA (Station or Slave)
RTS/CTS	This mechanism allows to operate in the network properly in the presence of hidden nodes. As the tests are designed for point-to-point links, this option will be disabled.
Power	The power level is important because transmit low power can push down the link, but high power can saturate the receptor. The better option for power non-relative test is set this option in Auto
Bitrate	This parameter will manage the physical layer mechanism (coding and modulation) Each standard have defined several possible bitrates. In order to test all the bitrates, this option must be changed usually. Is necessary bear in mind that the higher bitrate used, the higher SNR necessary.
Bandwidth	For 802.11 a/b/g three channel bandwidths have been defined: 5MHz, 10MHz and 20MHz. Usually 20MHz will be the bandwidth selected.

Table 16: Configurable parameters for 802.11 a/b/g.

NV2	
SSID	The identifier of the wireless network. Usually has is a string value or a 48 bits identifier according if the network is in Ad-hoc or AP mode.
Channel/Freq	The frequency channel in which the link will be established within the 2.4 GHz band or 5GHz band. The less interfered channel will be selected. It is necessary check with a spectrum probe the level of interference in each channel in order to choose the less congested. This way, the measure will be the most accurate possible.
Bitrate	This parameter will manage the physical layer mechanism (coding and modulation) Each standard have defined several possible bitrates. In order to test all the bitrates, this option must be changed usually. Is necessary bear in mind that the higher bitrate used, the higher SNR necessary.
Bandwidth	For 802.11 a/b/g three channel bandwidth have been defined: 5MHz, 10MHz, 20 MHz and 40 MHz. Usually only 20MHz and 40MHz will be the bandwidths selected.
Guard Interval	OFDM define a default Guard Interval value equal to 800 ns in order to avoid the multipath interference. In order to improve the performance it is possible reduce the Guard Interval to 400 ns for links with low multipath interference.

Table 17: Configurable parameters for NV2

802.11n	
SSID	The identifier of the wireless network. Usually has is a string value or a 48 bits identifier according if the network is in Ad-hoc or AP mode.
Channel/Freq	The frequency channel in which the link will be established within the 2.4 GHz band or 5GHz band. The less interfered channel will be selected. It is necessary check with a spectrum probe the level of interference in each channel in order to choose the less congested. This way, the measure will be the most accurate possible.
Mode	The operating mode will be always infrastructure mode or AP mode. Although also would be possible set the Ad-hoc mode, this mode do no support QoS mechanisms by default and is more simple. Within the AP mode, one station must be selected as AP (Access Point or Master) and the other as STA (Station or Slave)
RTS/CTS	This mechanism allow operate in the network properly in the presence of hidden nodes. As the test are designed for point-to-point links, this option will not be used.
Power	The power level is important because transmit low power can push down the link, but high power can saturate the receptor. The better option for power non-relative test is set this option in Auto
BitRate	This parameter will manage the physical layer mechanism (coding and modulation) 802.11n define this bitrates as MCS. There are for each bandwidth 32 MCS. From MCS 0-7 only SISO links can be tested. MCS 8-15, MCS 16-23 and MCS 24-31 will be used for MIMO links (2 spatial streams, 3 spatial streams and 4 spatial streams). It is necessary to bear in mind that the higher MCS is used, the higher SNR is required.
Bandwidth	For 802.11 a/b/g three channel bandwidth have been defined: 5MHz, 10MHz, 20 MHz and 40 MHz. Usually only 20MHz and 40MHz will be the bandwidths selected.
Guard Interval	OFDM define a default Guard Interval value equal to 800 ns in order to avoid the multipath interference. In order to improve the performance it is possible reduce the Guard Interval to 400 ns for links with low multipath interference.
BlockAck	802.11 provides this mechanism for improve the performance reducing the overhead due to the acknowledgements. Block-Ack permit confirm the reception of several frames using only one ACK frame. This can be enabled or disabled according to the test
Frame Aggregation	In order to reduce the overhead due to the headers, 802.11 permit frame aggregation. This is especially useful for low frame size. It is possible set a threshold to determine how many frames can be aggregated together. Some test will need change this parameter in order to compare the different effects in the performance.
Spatial Flows	The spatial flows are conditioned by the hardware, but is possible determine use SISO or MIMO mechanisms.

Table 18: Configurable parameters for 802.11n.

AirMax	
SSID	The identifier of the wireless network. Usually has is a string value or a 48 bits identifier according if the network is in Ad-hoc or AP mode.
Channel/Freq	The frequency channel in which the link will be established within the 2.4 GHz band or 5GHz band. The less interfered channel will be selected. It is necessary check with a spectrum probe the level of interference in each channel in order to choose the less congested. This way, the measure will be the most accurate possible.
Bitrate	This parameter will manage the physical layer mechanism (coding and modulation) Each standard have defined several possible bitrates. In order to test all the bitrates, this option must be changed usually. Is necessary bear in mind that the higher bitrate used, the higher SNR necessary.
Bandwidth	For 802.11 a/b/g three channel bandwidth have been defined: 5MHz, 10MHz, 20 MHz and 40 MHz. Usually only 20MHz and 40MHz will be the bandwidths selected.

Table 19: Configurable parameters for AirMax.

- *Traffic configuration*. Iperf or DITG must be configured in order to send the proper flow. The bitrate sent must be configured according the bitrate supported by the wireless link. If the test requires work in the saturation or on the saturation point, it is necessary set the bit rate equal or higher than the established in the wireless link. If it requires a non-saturation condition, a lower bitrate will be sent. Other important issue to bear in mind is the packet size. If the test does not require fragmentation, it is usually interesting send packets below the absolute fragmentation threshold of the protocols used. Furthermore, it is necessary set the correct size according the protocol headers UDP (+8 bytes), IP (+20 bytes), MAC (usually +24 bytes + 4 bytes from FCS).
- *Execute the test.* Once the proper link is established, it is possible launch the traffic injectors. Usually, due to its simplicity, Iperf will be used for this set of test, only using to DITG to validate weird measurements. Further, for more complex test, DITG will be used frequently. Iperf only measures throughput, jitter and packet loss. To measure the delay Ping utility will be used. Before injecting the traffic, ping test will be done to know the delay introduced by the empty traffic network. Simultaneously, the traffic will be injected and then, the real delay can be obtained. In order to get comparable results with the simulations, the load will be set in the saturating point for measuring the maximum throughput and in the optimum point in order to measure delay values, situated at approximately 80% of the maximum throughput for short distances.
- *Logging processes*. In order to have a tidy set of results, the test outputs will be stored in files which could be in the same computer or in the logging server, according the test duration.

3.1.3.5 Test scheduling

For these test which require long durations and a considerable amount of repetitions, an automatization can be done through scheduling tools like Cron and Unix-shell scripts. Due to the simplicity of the test designed for the D51 document, it has been considered not implement any scheduling mechanism.

3.1.4 Validation of the NS3 simulator

In order to validate the NS-3 simulator, several test has been designed, checking if the observable behavior matches with the expected results, so that long distance link can be simulated precisely and reliably. This will be done through 4 specific test from which it is know what would be the correct behavior: the effect in the throughput with the distance, the correct implementation of the AckTimeout, the throughput variation regarding the distance in the optimum SlotTime, and the analysis of the throughput vs. the offered load.

3.1.4.1 Distance dependence

The effect of the distance in a wireless link should be noticed mainly in the received signal power in the receiver and propagation delay. But while the received signal power can be fixed with a proper link budget, the simulation will be focused in the effects of the propagation delay. Assuming a constant SlotTime, it is expected that the throughput of the link will decrease with the distance. It is reflected in the Figure 8.

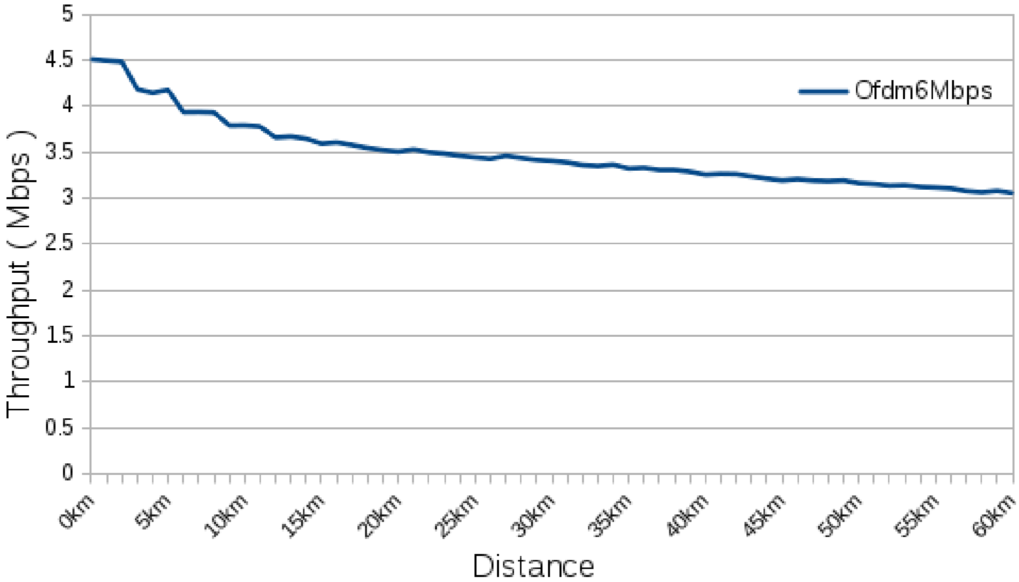

Figure 8: Throughput vs distance NS3 validation

In this simulation, a 802.11a point-to-point link with PHY mode OfdmRate6Mbps has been simulated. There is a bidirectional UDP flow with MAC-level frame size 1534 bytes. It is assumed that the RSSI is always 0 dBm and that negotiation between AP and STA has been already done, without RTS/CTS and without fragmentation. The SlotTime is 9 µs and the AckTimeout is large enough to influence in the test.

The throughput showed is the absolute throughput generated by the bidirectional UDP flow. Like expected, the throughput decreases as the distance increases due to the effects of the propagation delay in the DCF.

3.1.4.2 AckTimeout implementation

The next test tries to demonstrate the good implementation of the AckTimeout in NS-3. The Figure 9 shows the variation of the throughput between two stations, decreasing slowly as the distance increase. As expected, there is a sharp decrease when the distance passes the AckTimeout limit, adjusted for 28 Km. As is known, this the behavior viewed when ACK frames arrive systematically too late at the waiting station, causing unnecessary retransmissions.

For this test, a 802.11g point-to-point link has been tested, using four different PHY modes: DsssRate1Mbps, DsssRate2Mbps, DsssRate5_5Mbps and DsssRate11Mbps. There is one bidirectional UDP flow and the frame size has been set to 1088 byes. The SlotTime value has remained constant with a value of 9 µs, and the RTS/CTS mechanism and fragmentation has been disabled.

In Figure 10, the AckTimeout is changed in order to check different falls in the throughput occurs in the expected distances. The same 802.11g point-to-point link has been used, but using the PHY mode OfdmRate6Mbps. The AckTimeout tested 100 µs, 200 µs and 400 µs correspond with the throughput falls at the distances 6 Km, 21 Km and 51 Km respectively. The SlotTime value has remained constant with a value of 9 µs.

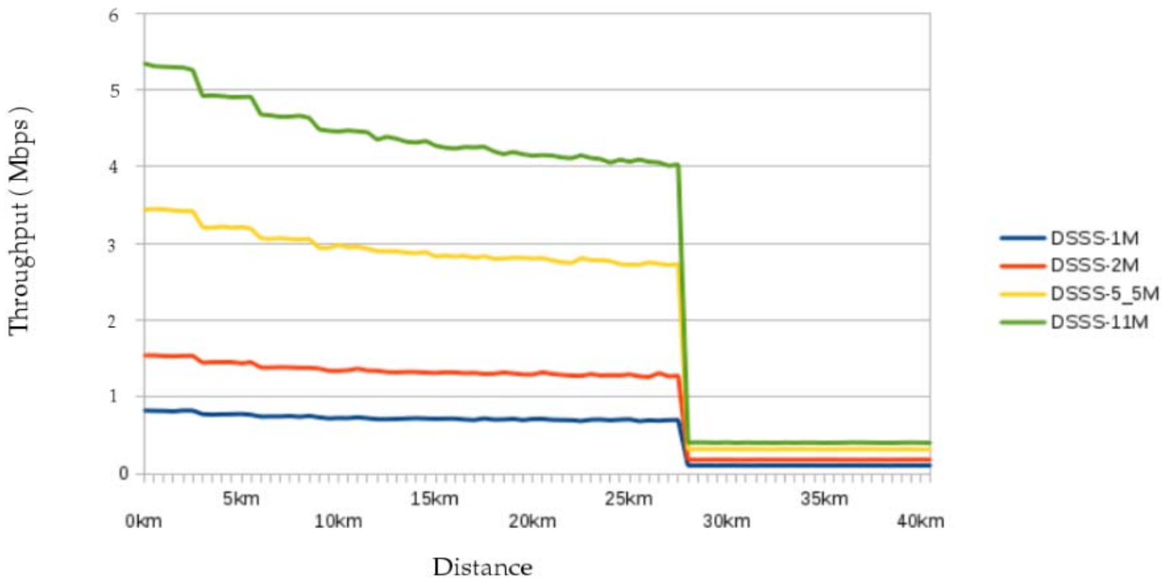

Figure 9: AckTimeout effect in NS3 validating example 1

These two tests demonstrate that the implementation of the AckTimeout is similar enough to the real behavior. In both Figure 9 and Figure 10, beyond the AckTimeout limit, the throughput falls to 10% of the maximum throughput approximately. For this reason, henceforth the AckTimeout will be adjusted large enough to assure the test will work without the AckTimeout problem.

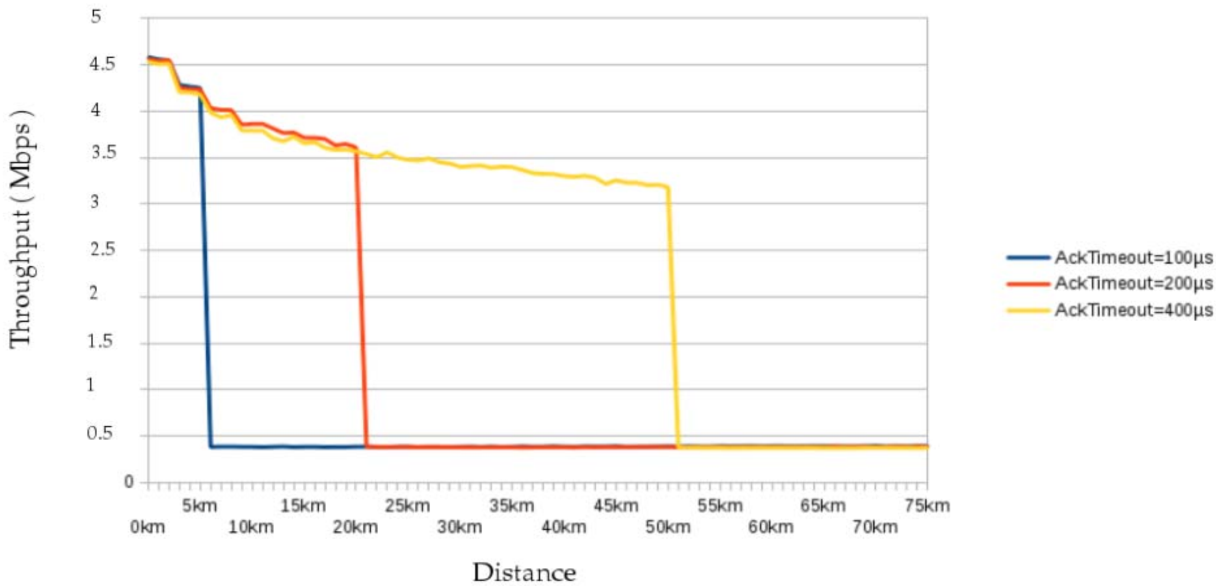

Figure 10: AckTimeout effect in NS3 validating example 2

3.1.4.3 *Similarity with the theoretical models*

Previously in this section, in Figure 8 it is possible perceive a slight irregularity in the shape of the curve caused by the constant value of the SlotTime, which is a exact multiple of the Vulnerability Interval (VI) at some distances. This effect is provided by the DCF model. In Figures 9 and 10 can be perceived more markedly, where the irregularities in the curves which look like a small staircase.

Ideally, in order to get the best throughput the value of the SlotTime must be set to an optimal value which depends of the distance. This is analyzed in detail in the theoretical model proposed by [Simo2010], and hence, each value of the SlotTime can be deduced by a specific expression.

In order to validate the NS-3 network simulator, a test in which the SlotTime value changes according the expression given the theoretical model has been done, obtaining for each distance, the higher throughput possible for particular parameters. Figure 11 shows the comparison between the theoretical model and the simulation in a 802.11g point-to-point link, with PHY mode OfdmRate6Mbps. In both, the frame size is 1500 bytes. As before, there is a bidirectional UDP flow, and the RTS/CTS mechanism and fragmentation has been disabled.

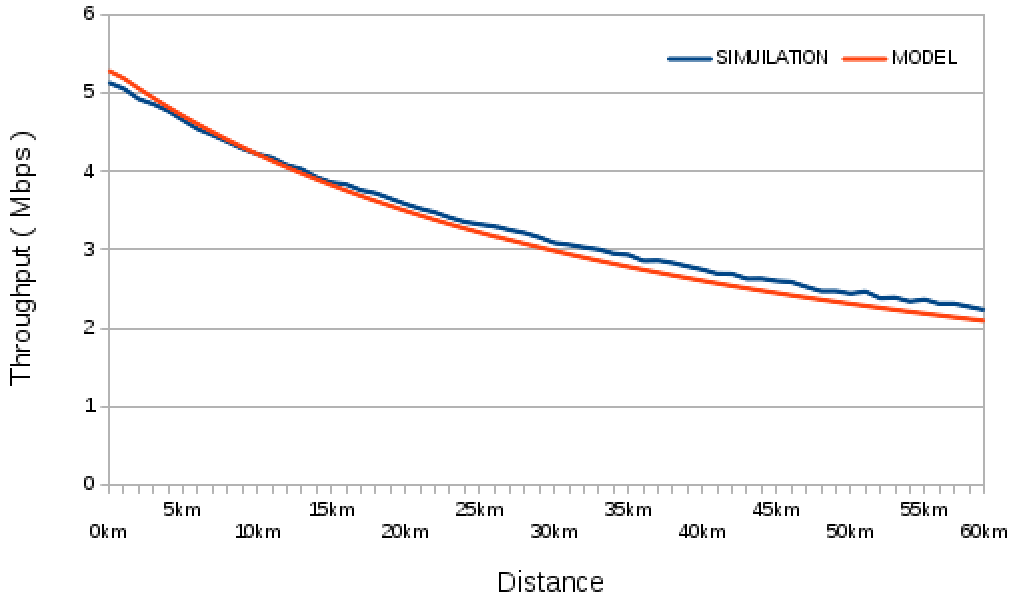

Figure 11: Theoretical model prediction compared to NS3 simulation

Although they have not the same behaviour, it is possible perceive that have high likeness. The model gives a slightly lower performance at distance below 10 km, but its behaviour is better than expected above this distance. This test shows that it is necessary bear in mind that the implementation is reliable enough but the results of the simulation will not be identically to the real test results using 802.11.

A similar simulation it is shown in Figure 12 showing how the throughput varies when the SlotTime is changed discretely for several distances. The obtained result is similar to the expected result.

In conclusion, it is reasonable to rely on NS-3 for simulations using 802.11 DCF, as it seems to be well implemented.

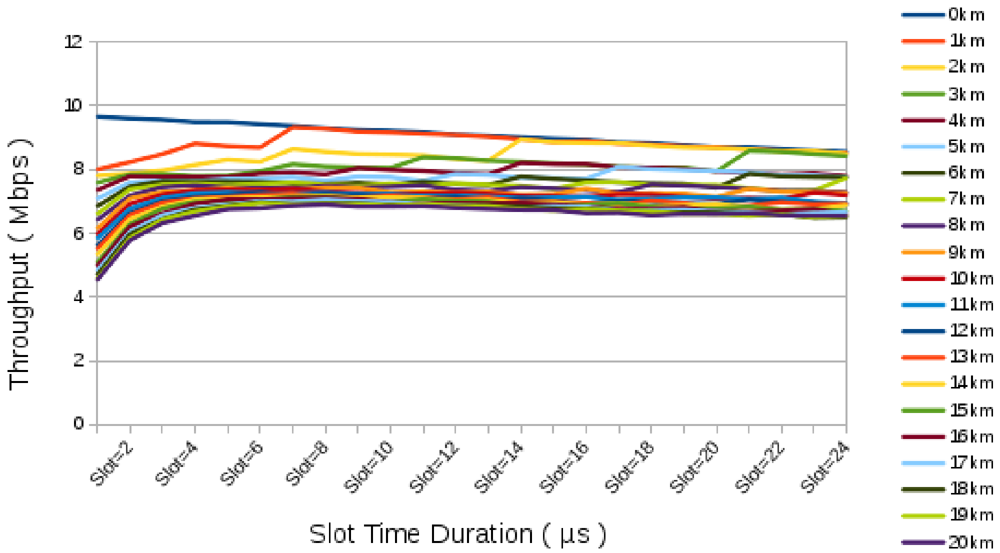

Figure 12: Througput vs. SlotTime discrete variation - NS3 Simulation

3.1.4.4 Saturation points

The last test made to validate the simulator consists of checking the performance around the saturation point of the link. The saturation point of a link is that in which the throughput begins to not change linearly with the offered load. When a increase in the offered load is translated in a linear increase of the throughput, the link is working in the unsaturated zone. However, if a increase of the offered load is not translated in a linear increase, the link is working in the saturated zone.

In order to illustrate this, a 802.11b point-to-point link, with PHY mode DsssRate5_5Mbps and a bidirectional UDP flow with frame size 1500 bytes has been simulated. The separation between stations is 50 meters. As before, the RTS/CTS mechanism and fragmentation has been disabled.

Figures 13, 14, 15 and 16 show for the offered load in each UDP flow, the absolute throughput, packets dropped, delay and jitter respectively. In the three figures is possible observe that the saturation point of this link is when the offered load is slightly above of 2000 Kbps per flow. This means that always will be interesting work below the saturation point, in order to reduce the delay and the packets dropped rate.

In Figure 13 the absolute throughput increase linearly with the offered load, reaching the maximum rate just before the saturation point, and then remains constant. Other tests made with a unidirectional flow do not show this maximum in the saturation point, but only remains at the maximum throughput constantly for each offered load.

Figure 14 shows that after the saturation point, losses are almost linearly with the offered load. This is the expected behaviour when more traffic than supported is introduced in a link. The losses begin after the saturation point due to the presence of a buffer which softens the losses.

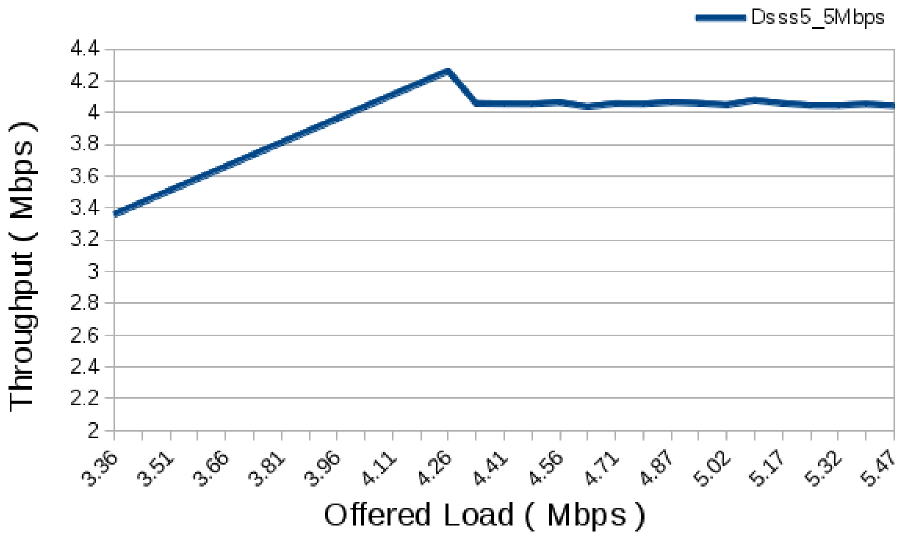

Figure 13: Throughput vs. offered load in NS3 simulations

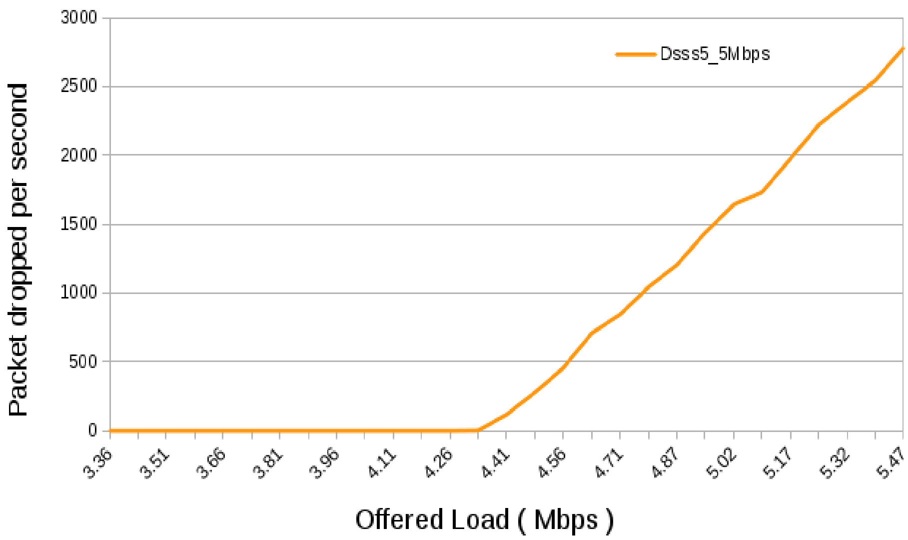

Figure 14: Packet loss saturation in NS3 validation

In Figures 15 and 16, the delay and the jitter have the expected value below the saturation point. After it they increase rapidly and from a certain value the increase is soften. The simulation shows that for high values of offered load, delay and jitter remains more or less constant. It is important to notice that work in the saturation point gives the highest throughput, but not the minimum delay. For this reason is advisable work below of saturation point. In order to get optimum values for throughput and delay, a optimum point of offered load will be defined. Although this point will not offer the highest throughput, it will be interesting work in this point since the delay is linear, controllable and relatively low.

Finally, it is possible assure that the NS-3 network simulator is a valid tool for simulate long-distance links for the IEEE 802.11 a/b/g standard. Full IEEE 802.11n support is not available yet, though NS-3.18 (released on 29th August 2013) supports most of its features.

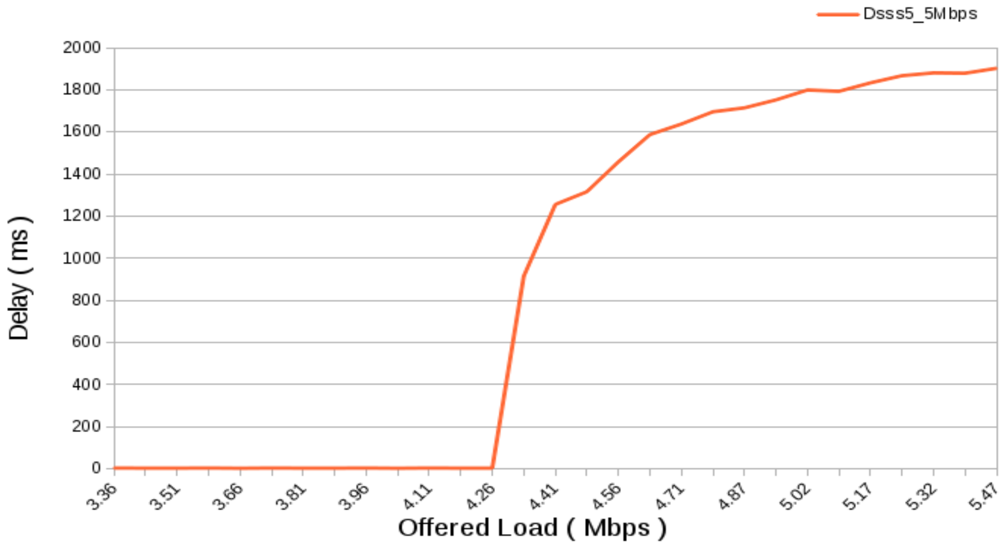

Figure 15: Delay saturation in NS3 validation

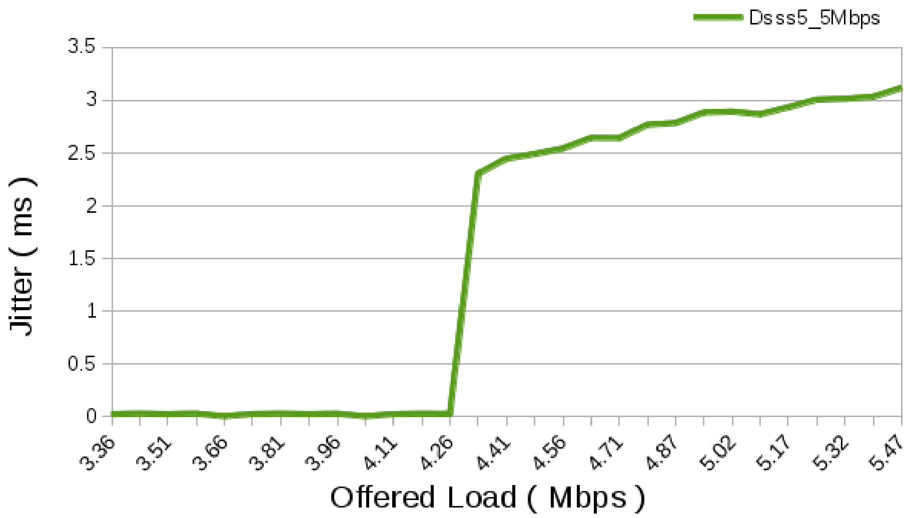

Figure 16: Jitter saturation in NS3 validation

3.1.5 General to 802.11

3.1.5.1 Theoretical Model

Data packets in 802.11 may be of two types: broadcast/multicast packets or unicast packets. The first ones are not acknowledged for obvious reasons. The second ones are acknowledged.

[Simo2010] demonstrated that any good analytical model including the SlotTime duration in the equations may be used either for short-range WiFi links or for long-range WiFi links, provided that the SlotTime is adapted to the distance as indicated in that paper. There are quite a few analytical models compliant with this requirement for DCF, and also a few for EDCA, that give formulas for throughput, delay and packet-loss. But the most interesting up to date is [Bianchi2010] because it takes into account all the minor details of CSMA/CA as it really operates for 802.11. In the model proposed by Bianchi et al. the authors model the MAC in an 802.11 station using two different strategies, one derived from [Bianchi2000] but correcting several inaccuracies, and the other derived from [Bianchi2005]. This model permits to estimate the saturation throughput for any 802.11 network using

DCF, no matter what the PHY is. The model considers a saturation situation in which all stations always have a packet to be transmitted, and that the conditional probability of colliding for a station that is transmitting does not depend on the number of retransmission. The saturation throughput is a very interesting result for our purposes because it indicates the level of offered load that should never be exceeded. Under the saturation throughput, the relationship throughput/load is known to be almost linear and the delay is acceptable for real-time communications, as it will be demonstrated in this document. On the contrary, if the saturation point is overtaken and the offer load is bigger that the saturation throughput, then the packet loss probability becomes important and the delay is high and almost proportional to the buffer size. Hence, this model is interesting in general but very specially as a means of discovering the saturation throughput for any physical bitrate in any PHY layer used in combination with DCF.

The result of the model proposed in [Bianchi2010] is the following set of equations Eqs.3.1 and 3.2:

$$\tau = \frac{1}{1+\frac{1-p}{2\left(1-p^{R+1}\right)}\sum p^{j}\left(2^{j}W-1\right)-\frac{\left(1-p\right)}{2}} \tag{3.1}$$

$$p = 1-\left(1-\tau\right)^{n-1} \tag{3.2}$$

where p and τ are the conditional collision probability and the probability on a station to transmit in any slot. W is a constant, corresponding with the highest possible value of the contention window for a packet that is being transmitted for the first time. R is the number or retransmissions (not considering the first transmission) before dropping a frame and trying with the next. This set of equations may be solved numerically. We have solved it for the most important values of W that can be considered here (the default values for the different PHYs), for the standard number of retransmissions (7 for basic mode, 4 for RTS/CTS mode) and for up to 15 stations. Actually, in this document we are going to concentrate on the case n=2 (point to point links) but the other values are also obtained and shown in Table 20.

N	W=16		W=32		W=8		W=4	
	p	tau	p	tau	p	tau	p	tau
2	0,1091	0,1091	0,0585	0,0585	0,1887	0,1887	0,2898	0,2898
3	0,1832	0,0962	0,1067	0,0549	0,2807	0,1519	0,3832	0,2147
4	0,2362	0,0859	0,1468	0,0516	0,3373	0,1282	0,4383	0,1749
5	0,2763	0,0777	0,1805	0,0486	0,3776	0,1118	0,4775	0,1498
6	0,3080	0,0710	0,2092	0,0459	0,4088	0,0998	0,5083	0,1324
7	0,3340	0,0655	0,2338	0,0434	0,4342	0,0906	0,5337	0,1194
8	0,3559	0,0609	0,2553	0,0412	0,4557	0,0832	0,5554	0,1094
9	0,3749	0,0570	0,2742	0,0393	0,4744	0,0773	0,5745	0,1013
10	0,3916	0,0537	0,2910	0,0375	0,4909	0,0723	0,5914	0,0947
11	0,4065	0,0508	0,3061	0,0359	0,5057	0,0680	0,6067	0,0891
12	0,4200	0,0483	0,3198	0,0344	0,5192	0,0644	0,6207	0,0844
13	0,4323	0,0461	0,3323	0,0331	0,5315	0,0612	0,6336	0,0803
14	0,4436	0,0441	0,3438	0,0319	0,5429	0,0584	0,6455	0,0767
15	0,4540	0,0423	0,3545	0,0308	0,5535	0,0560	0,6566	0,0735

Table 20: Values of transmission and conditional collision probability for different CW values and up to 15 stations

Based on those two variables, Bianchi et al. clearly establish how to calculate the saturation throughput, saturation average delay and packet drop probability. Before continuing, it is worth to mention that the model at [Bianchi2010] is valid for any distance as far as the SlotTime is adapted as indicated in [Simo2010]. In this case, several instrumental variables defined in that paper for calculating the performance figures must be redefined now.

The saturation throughput is calculated in Eq. 3.3.

$$S = \frac{P_s \cdot E[P] \cdot \frac{W}{W-1}}{(1-P_b)\delta_d + P_s \cdot E[T_s] + (P_b - P_s) \cdot E[T_c]} \tag{3.3}$$

where P_b denotes the probability of the channel to be busy in any slot, and P_s is the probability of any slot to contain a successful transmission. Both probabilities may still be defined as in [Biachi2010] no matter what the distance between stations is, under [Simo2010] conditions for δ_d (the SlotTime including twice the propagation time):

$$P_b = 1 - (1-\tau)^n \tag{3.4}$$

$$P_s = n\tau(1-\tau)^{n-1} \tag{3.5}$$

The other variables are E[P], the average packet payload size, E[Ts], the average time spent for a successful transmission, and E[Tc], the average time spent in a collision. These times are clearly affected by the propagation time. [Bianchi2010] does not even consider the propagation time because, under normal circumstances in WLAN, the standard specifies that $T_p \ll 1\,\mu s$, which is negligible. However, propagation times are much higher in our case and must be accounted as:

$$E[T_s] = \delta_d + \left(\frac{W}{W-1}\right)\left[T_{MPDU} + SIFS + T_{ACK} + DIFS + 2\mathrm{T}_p\right] \tag{3.6}$$

$$E[T_c] = \delta_d + \left[T_{MPDU} + SIFS + T_{ACK} + DIFS + T_p\right] \tag{3.7}$$

for basic mode, or:

$$E[T_s] = \delta_d + \left(\frac{W}{W-1}\right)\left[T_{RTS} + T_{CTS} + T_{MPDU} + 3 \cdot SIFS + T_{ACK} + DIFS + 4\mathrm{T}_p\right] \tag{3.8}$$

$$E[T_c] = \delta_d + \left[T_{RTS} + SIFS + T_{CTS} + DIFS + T_p\right] \tag{3.9}$$

for RTS/CTS mode.

The duration of MPDUs depends on the payload size and the bitrate, but also the MAC header must be considered at the same bitrate and the preamble and PLCP header must be considered as defined for each PHY. The duration of RTS and CTS packets, as well as SIFS and DIFS, are also PHY-dependent.

The delay that can be obtained with this model is the saturation delay which measures how long an average packet that is correctly received has spent since it was first issued to the MAC in the transmitter-side until it is correctly received in the receiver-side. This delay has mainly three components: the transmission time, that depends on the bitrate, the propagation time and the packet size, the MAC delay, depending on the collisions and the contention window size, and the queueing delay, that essentially depends on the size of the transmit queue. In saturation, the dominant component is the queueing delay. We are going to calculate the saturation delay, but the interest of these results is very limited. Further study is required to see how the delay can be modeled accurately under the saturation point.

[Bianchi2010] calculates firstly the packet-loss probability P_{LOSS} directly from the Markov chain, and then the delay as:

$$D = \frac{N \cdot W\left(1 - P\{\mathrm{LOSS}\}\right)}{S(W-1)} \tag{3.10}$$

For calculations on the unsaturated delay a different model is required. [Malone2007] is the reference in this sense for DCF. Fig. 4 in that paper is very useful because it shows how the MAC delay is related to the collision probability. A careful comparison of those results (obtained for 11Mbps, hence results will be general better for higher bitrates) with the values of conditional collision probability obtained above permits to see that, even for the lowest conceivable contention window, the MAC delay will be under 10 ms for point-to-point links. For normal values of W, the MAC delay will keep under 5 ms. The importance of the accuracy in the value of the MAC delay is not really worth of the complexity of a more detailed analysis, as the queuing delay is much more important. The approximation to the problem will be the opposite: a limit to the maximum average delay will impose the maximum desired average occupation for the transmission queue, which in turn will fix the desired ratio offered-load/saturation-throughput.

From the Little's Result we can derive:

$$\overline{T} = \left(\overline{N}_q + 1\right) E\left[S\right] \tag{3.11}$$

The MAC delay is the average service time; hence, the average number of users in the queue permits us to regulate the total time in the system. Once a time threshold is imposed in the system, a maximum number of users in the queue can be determined. Adjusting the ration offered-load/saturation-throughput one can obtain any value between the service time and the maximum.

Finally, for the packet-loss probability, the fact that the MAC incorporates the mechanism for retransmissions takes the probability of a HOL (head of line) packet to be lost close to zero. Hence, the packet-loss probability may be approximated by the probability of buffer overload. If we keep the link under the saturation point, this probability is close to zero.

3.1.6 802.11a (5GHz, OFDM, DCF)

3.1.6.1 Theoretical results

The 802.11a amendment to the IEEE 802.11 standard was issued in 1999. Currently it is included in the new revision [IEEE 802.11-2012] as the OFDM PHY. Table 21 contains the main physical parameters.

Frequency band [GHz]	5
Channel bandwidth [MHz]	20
Bitrates [Mbps]	6,9,12,18,24,36,48,54
PLCP overhead [us]	20
SIFS [us]	16
SlotTime [us]	9
DIFS [us]	34

Table 21: Physical parameters in 802.11a.

The following Figure 17 and Table 22 show how the maximum throughput evolves as the distance grows. Packets with an average size of 1500 bytes have been used to obtain these results.

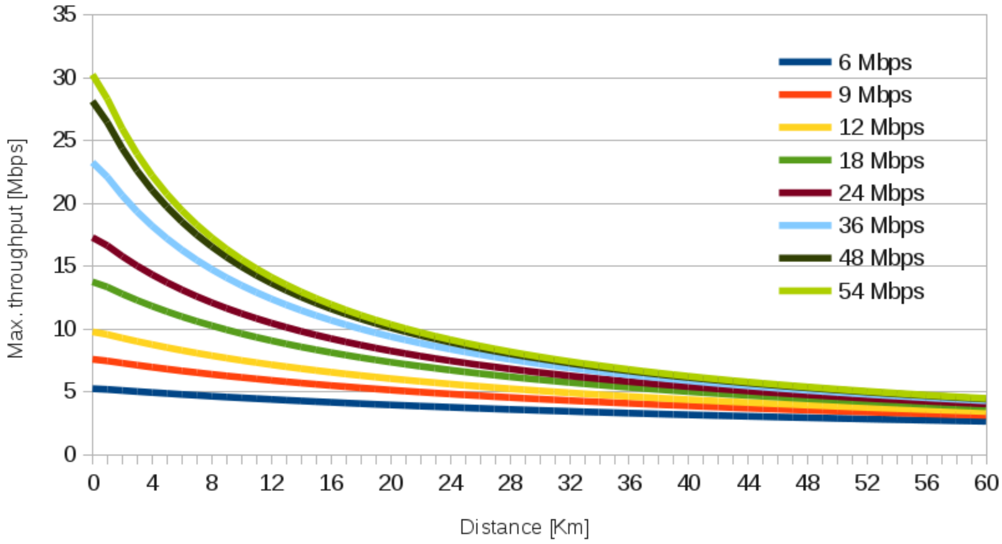

Figure 17: Throughput as a function of the distance for a point-to-point link in 802.11a for the different bitrates in basic mode

Basic mode	D [km]										
	0	5	10	15	20	25	30	35	40	45	50
6 Mbps	5,191	4,811	4,463	4,162	3,899	3,667	3,462	3,278	3,113	2,963	2,827
9 Mbps	7,529	6,754	6,088	5,542	5,085	4,698	4,366	4,078	3,825	3,602	3,403
12 Mbps	9,717	8,465	7,444	6,643	5,997	5,466	5,022	4,644	4,319	4,037	3,789
18 Mbps	13,699	11,334	9,576	8,290	7,309	6,535	5,909	5,393	4,960	4,591	4,273
24 Mbps	17,228	13,648	11,177	9,463	8,206	7,243	6,482	5,866	5,357	4,929	4,565
36 Mbps	23,208	17,148	13,420	11,024	9,353	8,123	7,178	6,430	5,824	5,322	4,899
48 Mbps	28,082	19,671	14,917	12,014	10,057	8,648	7,585	6,755	6,089	5,542	5,086
54 Mbps	30,196	20,685	15,494	12,385	10,316	8,839	7,732	6,871	6,183	5,620	5,151

Table 22: Saturation throughput for a point-to-point link in 802.11a, in basic mode

The next Figure 18 and Table 23 show equivalent results for the RTS/CTS case.

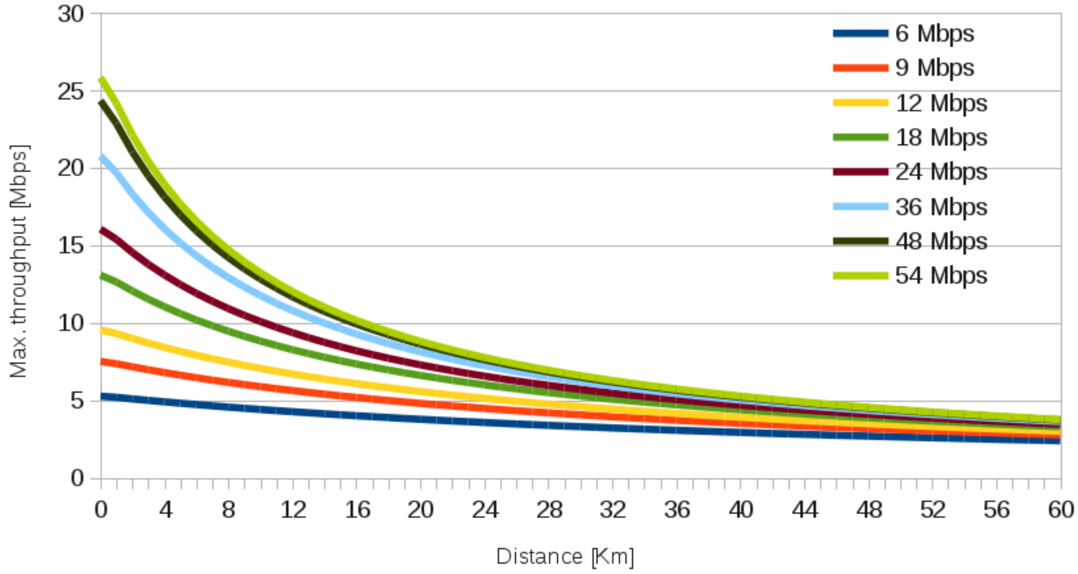

Figure 18: Throughput as a function of the distance for a point-to-point link in 802.11a for the different bitrates, in RTS/CTS mode

RTS/CTS mode	D [km]										
	0	5	10	15	20	25	30	35	40	45	50
6 Mbps	5,273	4,820	4,416	4,075	3,782	3,529	3,307	3,112	2,938	2,783	2,644
9 Mbps	7,516	6,628	5,887	5,296	4,812	4,409	4,069	3,777	3,525	3,304	3,109
12 Mbps	9,547	8,158	7,064	6,229	5,571	5,038	4,598	4,229	3,915	3,644	3,409
18 Mbps	13,080	10,606	8,829	7,562	6,613	5,876	5,286	4,804	4,403	4,063	3,772
24 Mbps	16,051	12,478	10,089	8,468	7,295	6,408	5,713	5,155	4,695	4,311	3,985
36 Mbps	20,767	15,153	11,769	9,620	8,135	7,047	6,216	5,560	5,029	4,591	4,223
48 Mbps	24,343	16,973	12,838	10,323	8,632	7,417	6,502	5,788	5,215	4,745	4,353
54 Mbps	25,825	17,680	13,239	10,580	8,811	7,549	6,603	5,868	5,280	4,799	4,398

Table 23: Saturation throughput for point to point links in 802.11a in RTS/CTS mode

There is not much advantage in point-to-point links for using the RTS/CTS mode. Firstly, nodes are mutually visible, and secondly, the collision probability is not as high as for a higher number of stations. We get a lower performance with no advantage at all.

We can also see that the distance penalizes much more the performance of the highest bitrates. At a few meters, the throughput at 54 Mbps is 5.8 times the one at 6 Mbps. At 40 km the improvement from the lowest to the highest bitrate is just 2 times. In real links, the EIRP limitation inherent to real transmitters and to regional regulations may force to use lower bitrates, but the penalty in that case is not as important as in shorter distances.

Last but not least, Table 24 shows the throughput results when packet size is as low as 200 bytes. This is the typical situation with Voice packets in telephony services. We can see that the throughput is reduced dramatically. It should be noted that when estimating the maximum throughput in a link, the packet size is determinant for the result in the case of WiFi.

Basic mode	D [km]										
	0	5	10	15	20	25	30	35	40	45	50
6 Mbps	3,337	2,416	1,868	1,522	1,285	1,111	0,979	0,875	0,791	0,722	0,663
9 Mbps	4,297	2,883	2,135	1,695	1,406	1,201	1,048	0,929	0,835	0,758	0,694
12 Mbps	5,019	3,190	2,299	1,797	1,475	1,251	1,086	0,959	0,859	0,778	0,711
18 Mbps	6,033	3,572	2,491	1,912	1,552	1,306	1,127	0,991	0,885	0,799	0,728
24 Mbps	6,710	3,799	2,599	1,975	1,593	1,335	1,149	1,008	0,898	0,810	0,737
36 Mbps	7,559	4,057	2,717	2,043	1,637	1,365	1,171	1,025	0,912	0,821	0,746
48 Mbps	8,070	4,199	2,781	2,079	1,659	1,381	1,183	1,034	0,919	0,826	0,751
54 Mbps	8,256	4,249	2,802	2,091	1,667	1,386	1,187	1,037	0,921	0,828	0,753

Table 24: Maximum throughput in 802.11, basic mode, when the average packet size is 200 bytes

The following Tables 25 and 26 and Figures 19 and 20 show the evolution of the saturation throughput for 20 km and 40 km respectively as the packet size grows. It can be seen that, having the values for 200 bytes and for much bigger packets (e.g. 1500 bytes) the linear interpolation gives a good predictor of the maximum throughput that can be obtained for any other packet size.

Basic mode	P [bytes]			
	200	600	1000	1400
6 Mbps	1,2848	2,6532	3,3713	3,8137
9 Mbps	1,4057	3,1702	4,2329	4,9430
12 Mbps	1,4751	3,5124	4,8530	5,8020
18 Mbps	1,5517	3,9375	5,6859	7,0223
24 Mbps	1,5931	4,1911	6,2197	7,8476
36 Mbps	1,6367	4,4796	6,8641	8,8927
48 Mbps	1,6595	4,6393	7,2391	9,5271
54 Mbps	1,6672	4,6951	7,3733	9,7592

Table 25: Maximum throughput in a 20 km link for different packet sizes

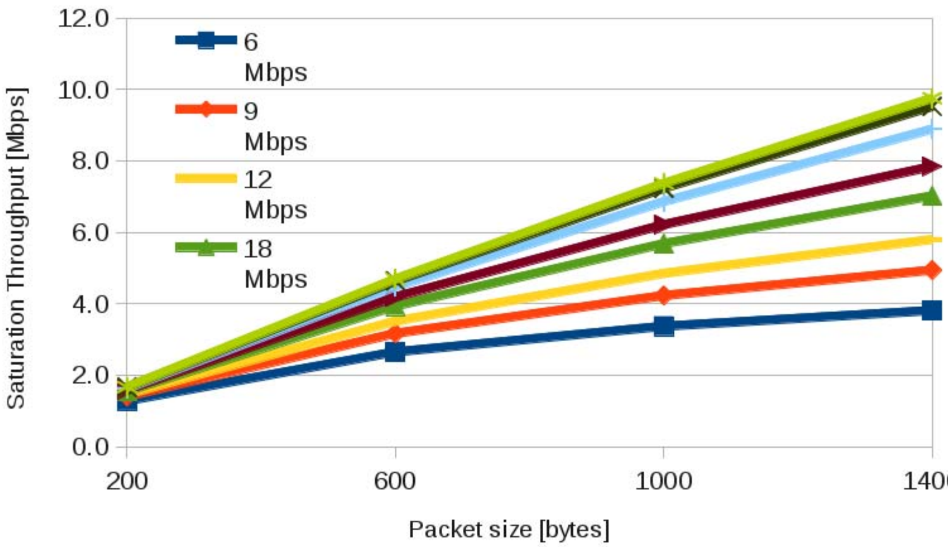

Figure 19: Maximum throughput vs. packet size for a 20 km link using 802.11a

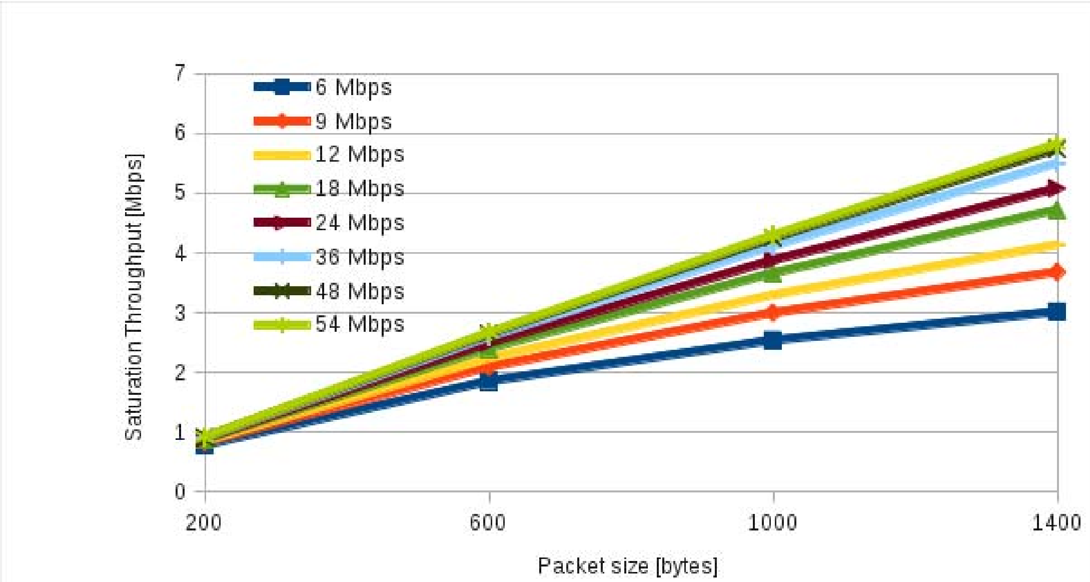

Figure 20: Maximum throughput vs. packet size for a 40 km link using 802.11a

The reason for showing results for all the bitrates in the precedent tables and figures is that any of them might be necessary depending on the link budget. The two frequency bands that can be used by IEEE 802.11 (2.4 GHz and 5 GHz) have transmission power restrictions that depend on the regional regulations. The FCC (Federal Communications Commission) is responsible for the radio-electric regulations in the USA, but many other countries, particularly in Latin America, have national regulations inspired in the FCC's. FCC permits up to 30dBm of transmitting power with omnidirectional antennas of gain up to 6dBi for omnidirectional communications. In the 5 GHz band, directional links may increase the antenna gain up to 27 dBi keeping the same maximum transmit power.

The transmitting power restriction is a limiting factor in terms of distance. We have calculated approximately the distances that can be achieved with typical values of transmitted power, sensitivity, antenna gain, and cable attenuation. The propagation loss has to be estimated for each specific environment, as irregular terrains condition has a considerable effect on the propagation.

However, a rough estimation of achievable distances obtained by using the free space model is presented in Table 27. A stability margin of 20dB has been preserved over the sensitivity for calculating the minimum acceptable received power level. We have calculated the maximum achievable distance for PtP links operating in the 5 GHz band (both ends using directional antennas with a gain of 27dBi). The results in the 2.4 GHz band (for 802.11b/g/n) would be approximately the same because the propagation loss is about 7 dB lower but antennas have lower gains and the regulations are more restrictive.

D		5	10	15	20	25	30	35	40	45	50	55	60
Path Loss		121.4968972	127.5174971	131.0393223	133.538097	135.4762973	137.0599222	138.398858	139.5586969	140.5817474	141.4968972	142.3247509	143.0805221
P_Rx		-47.4968972	-53.51749711	-57.03932229	-59.53809703	-61.47629729	-63.05992221	-64.398858	-65.55869694	-66.58174739	-67.4968972	-68.3247509	-69.08052212
Bitrate	Sensitivity												
54 Mbps	-77												
48 Mbps	-79												
36 Mbps	-82												
24 Mbps	-86												
18 Mbps	-88												
12 Mbps	-89												
9 Mbps	-90												
6 Mbps	-90												

Table 27: Feasibility of point-to-point links at the different 802.11a bitrates. Colour red marks links with fading margin under 15 dB, color yellow between 15 and 20 dB, and color green means a margin over 20 dB.

Table 27 has been done with sensitivity values obtained from the technical datasheet of a real product, but different 802.11a implementations may have better or worse sensitivities. We can see that the maximum bitrate that can be obtained is lower as the distance grows. In combination with throughput results given above, this can be an idea of the performance that one can expect from a specific long-distance 802.11a link.

3.1.6.2 802.11a Simulation

In a similar way, NS-3 simulations have been made in order to contrast the theoretical model detailed above. One 802.11a point-to-point link has been simulated for different distances and different bitrates, sending a bidirectional UDP flow, with frame size 1500 bytes. The SlotTime has been adjusted regarding the distance following the theoretical model.

The results of the simulations are shown next in Table 28 and Figure 21. As predicted in the validation section, both results are very similar.

	0km	5km	10km	15km	20km	25km	30km	35km	40km	45km	50km	55km	60km
Ofdm6	5.131	4.771	4.378	4.080	3.836	3.585	3.357	3.216	3.032	2.860	2.748	2.632	2.475
Ofdm9	7.331	6.604	5.894	5.411	4.938	4.554	4.24	3.932	3.692	3.458	3.298	3.149	2.932
Ofdm12	9.480	8.235	7.230	6.437	5.764	5.247	4.769	4.46	4.129	3.927	3.674	3.436	3.198
Ofdm18	14.07	10.91	9.143	7.904	7.009	6.219	5.642	5.240	4.802	4.391	4.08	3.816	3.595
Ofdm24	17.67	13.03	10.68	9.003	7.876	6.864	6.150	5.6	5.108	4.718	4.382	4.048	3.741
Ofdm36	23.18	16.2	12.67	10.39	8.863	7.719	6.872	6.164	5.525	5.117	4.687	4.304	4.010
Ofdm48	28.70	18.40	14.00	11.23	9.459	8.106	7.222	6.412	5.790	5.264	4.847	4.954	4.538
Ofdm54	29.80	19.30	14.42	11.72	9.668	8.348	7.261	6.511	5.849	5.331	5.376	5.001	4.616

Table 28: Simulation Maximum Throughput Basic mode

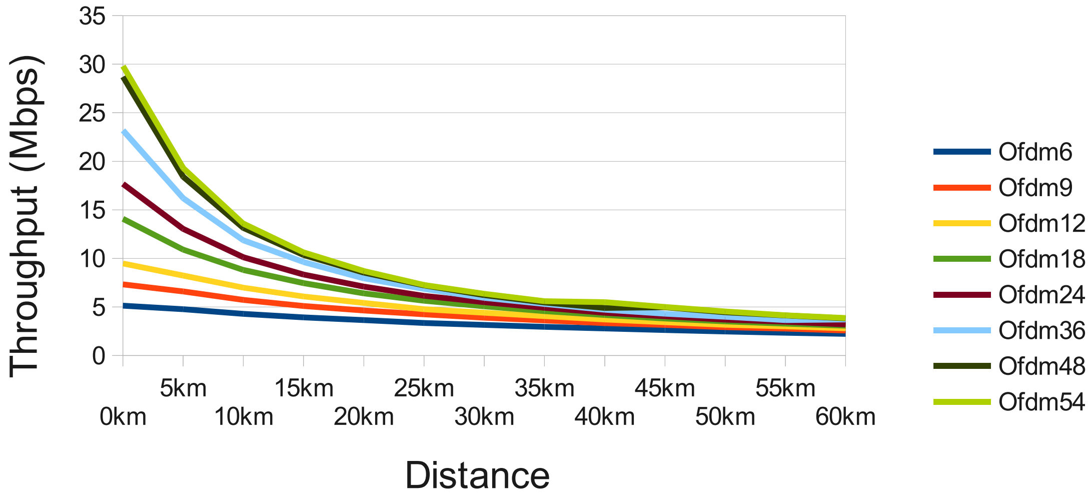

Figure 21: 802.11a Simulation Throughput vs. Distance with optimum SlotTime in Basic Mode

The results regarding the delay have been obtained through simulations. Figure 22 shows the delay per flow of each bitrate when is simulated just in its saturation point. The effects of the distance in the performance are not seen at first due to near saturation point, the delay is not linear, and the effects of the propagation delay are inappreciable. However, Figure 23 shows this effect of the propagation delay when simulating the link below its saturation point for each bitrate. Like expected, the higher bitrate, the lower is the delay.

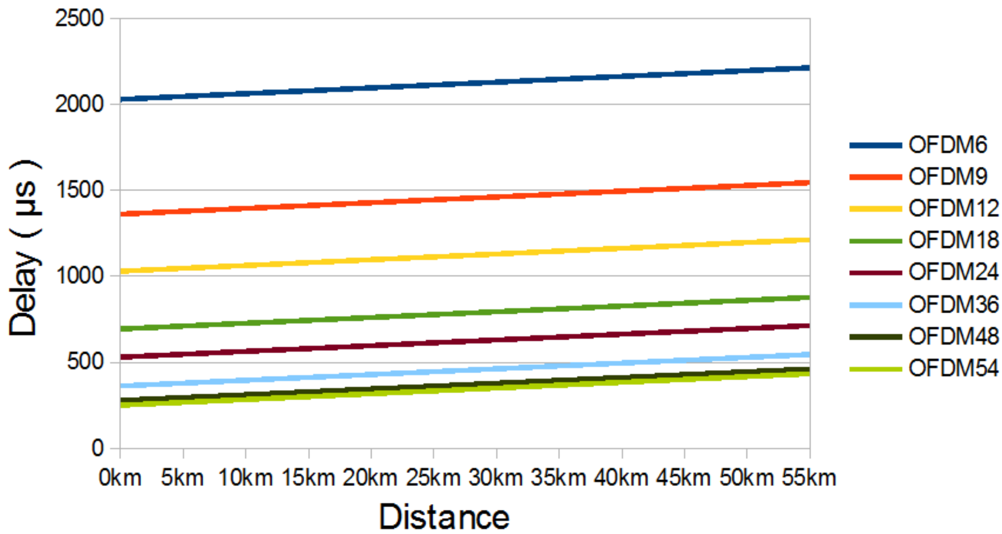

Figure 22: Delay below the saturation point - NS3 Simulation

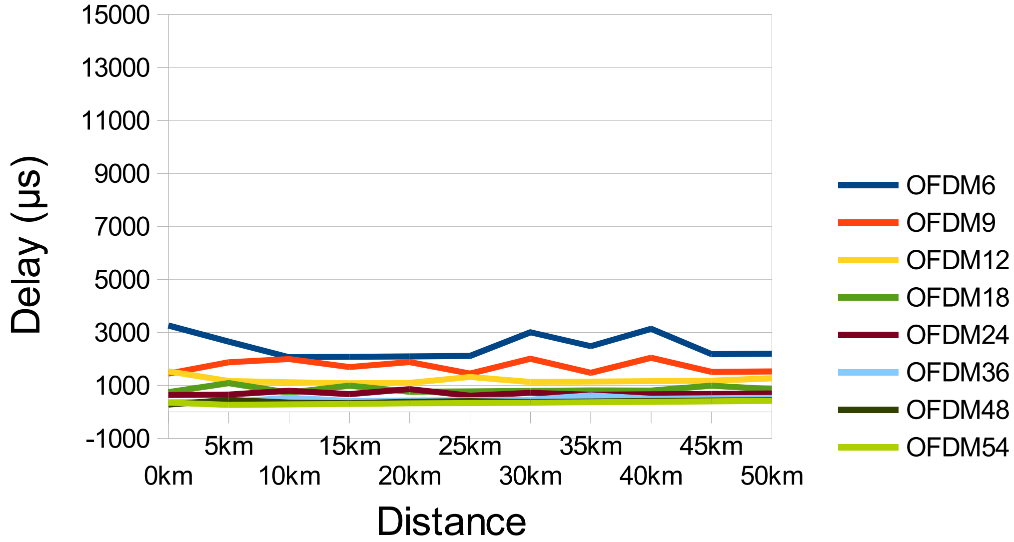

Figure 23: Delay in the saturation point - NS3 Simulation

Also values of jitter have been obtained. While non-negligible jitter values have been reached when simulating in the saturation point, almost zero values has been obtained when simulating below of the saturation point. Regarding the packet loss, the absolute obtained value is zero for each distance and each bitrate because of the non-saturated conditions of the link. If the link work below the saturation point do will not reach the highest throughput, but the delay, jitter and packet loss will be much better.

802.11a Real testbed

Last, in order to settle the results of both theoretical model and simulations, tests with real experiments have been made in the laboratory environment detailed previously. The indoor test were made with Mikrotik equipment separated 3 meters, injecting bidirectional UDP traffic with 1500 bytes frame size using with Iperf in a relatively clear-interference channel. The results, shown in Table 29, correspond to the throughput in the saturation zone and agree with both theoretical and simulation results.

802.11a	Throughput (Mbps)	Jitter (ms)
Ofdm6Mbps	4.97	3.031
Ofdm9Mbps	7.167	3.1965
Ofdm12Mbps	9.396	1.674
Ofdm18Mbps	13.606	1.235
Ofdm24Mbps	16.553	0.924
Ofdm36Mbps	22.434	0.668
Ofdm48Mbps	27.25	0.520
Ofdm54Mbps	30.4	0.525

Table 29: Real 802.11a testbed – Throughput

3.1.7 802.11b (2.4GHz, DSSS, DCF)

The 802.11b amendment to the IEEE 802.11 standard was issued in 1999. Currently it is included in the new revision [IEEE 802.11-2012] as the DSSS PHY. Table 30 contains the main physical parameters. The following Figure 24 and Table 31 show how the maximum throughput evolves as the distance grows. Packets with an average size of 1500 bytes have been used to obtain these results.

Frequency band [GHz]	2.4
Channel bandwidth [MHz]	22
Bitrates [Mbps]	1, 2, 5.5, 11
PLCP overhead [us]	96 (short) / 192 (long)
SIFS [us]	10
SlotTime [us]	20
DIFS [us]	50

Table 30: 802.11b PHY/MAC parameters

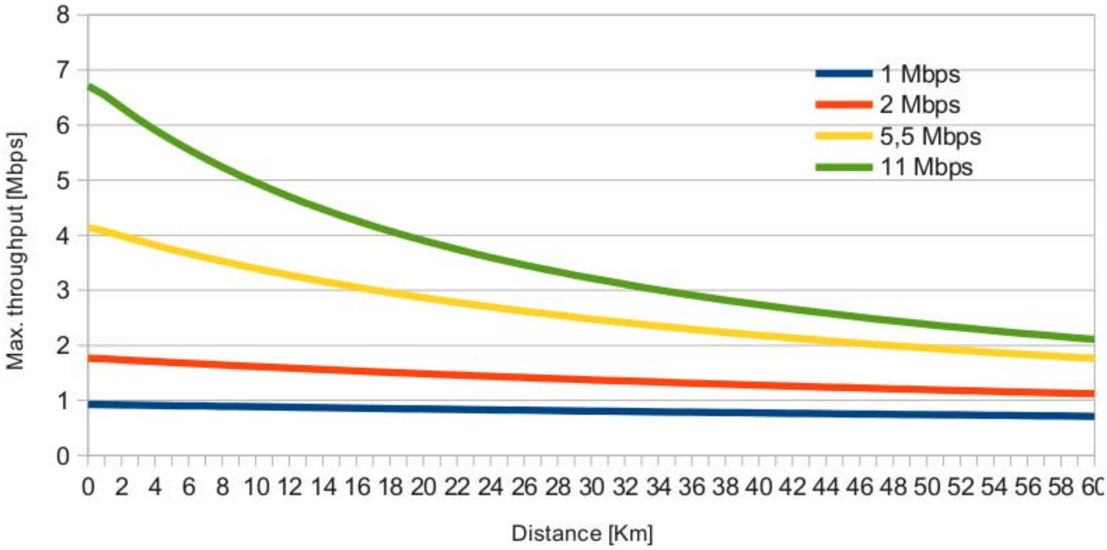

Figure 24: Evolution of the saturation throughput with the distance for 802.11b, basic mode

Basic mode	D [km]										
	0	5	10	15	20	25	30	35	40	45	50
1 Mbps	0,9302	0,9086	0,8866	0,8657	0,8458	0,8268	0,8086	0,7912	0,7745	0,7585	0,7432
2 Mbps	1,7676	1,6912	1,6167	1,5486	1,4860	1,4282	1,3748	1,3252	1,2790	1,2360	1,1958
5,5 Mbps	4,1390	3,7426	3,3967	3,1092	2,8666	2,6592	2,4797	2,3229	2,1848	2,0622	1,9526
11 Mbps	6,7114	5,7279	4,9554	4,3665	3,9027	3,5279	3,2189	2,9596	2,7389	2,5489	2,3836

Table 31: Saturation throughput at different distances for 802.11b in basic mode.

The next Figure 25 and Table 32 show equivalent results for the RTS/CTS case:

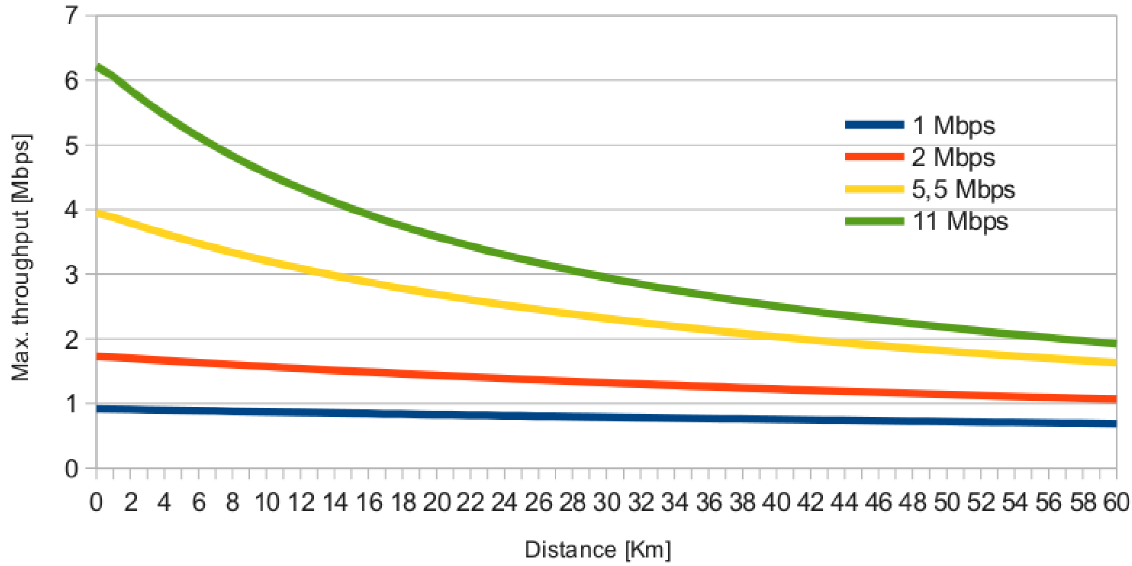

Figure 25: Evolution of the saturation throughput with the distance for 802.11b, RTS/CTS mode

RTS/CTS mode	D [km]										
	0	5	10	15	20	25	30	35	40	45	50
1 Mbps	0,92	0,8967	0,8733	0,851	0,8298	0,8096	0,7904	0,7721	0,7547	0,738	0,722
2 Mbps	1,7313	1,6507	1,5728	1,502	1,4372	1,3778	1,3231	1,2726	1,2258	1,1824	1,1419
5,5 Mbps	3,9449	3,5501	3,2084	2,9268	2,6906	2,4896	2,3166	2,1661	2,034	1,917	1,8128
11 Mbps	6,2157	5,2888	4,5647	4,015	3,5834	3,2356	2,9494	2,7097	2,506	2,3308	2,1785

Table 32: Saturation throughput at different distances for 802.11b in RTS/CTS mode.

Most of what has been said for 802.11a may be repeated here. RTS/CTS mode makes no sense in PtP links. We may also see here how the distance penalizes much more the performance of the highest bitrates. At a few meters, the throughput at 11 Mbps is about 7 times the one at 1 Mbps. At 40 km the improvement from the lowest to the highest bitrate is just 3.5 times.

The following Tables 33 and 34 and Figures 26 and 27 show the evolution of the saturation throughput for 20 km and 40 km respectively as the packet size grows. It can be seen that, having the values for 200 bytes and for much bigger packets (e.g. 1500 bytes) the linear interpolation gives a good predictor of the maximum throughput that can be obtained for any other packet size.

Basic mode	P [bytes]			
	200	600	1000	1400
1 Mbps	0,4406	0,6977	0,7899	0,8374
2 Mbps	0,5884	1,0991	1,3299	1,4615
5,5 Mbps	0,7481	1,7336	2,3539	2,7801
11 Mbps	0,8110	2,0761	3,0177	3,7457

Table 33: 802.11b saturation throughput for 20km

Basic mode	P [bytes]			
	200	600	1000	1400
1 Mbps	0,3240	0,5864	0,6997	0,7628
2 Mbps	0,3975	0,8460	1,0926	1,2486
5,5 Mbps	0,4644	1,1779	1,7003	2,0993
11 Mbps	0,4879	1,3265	2,0215	2,6068

Table 34: 802.11b saturation throughput for 40km

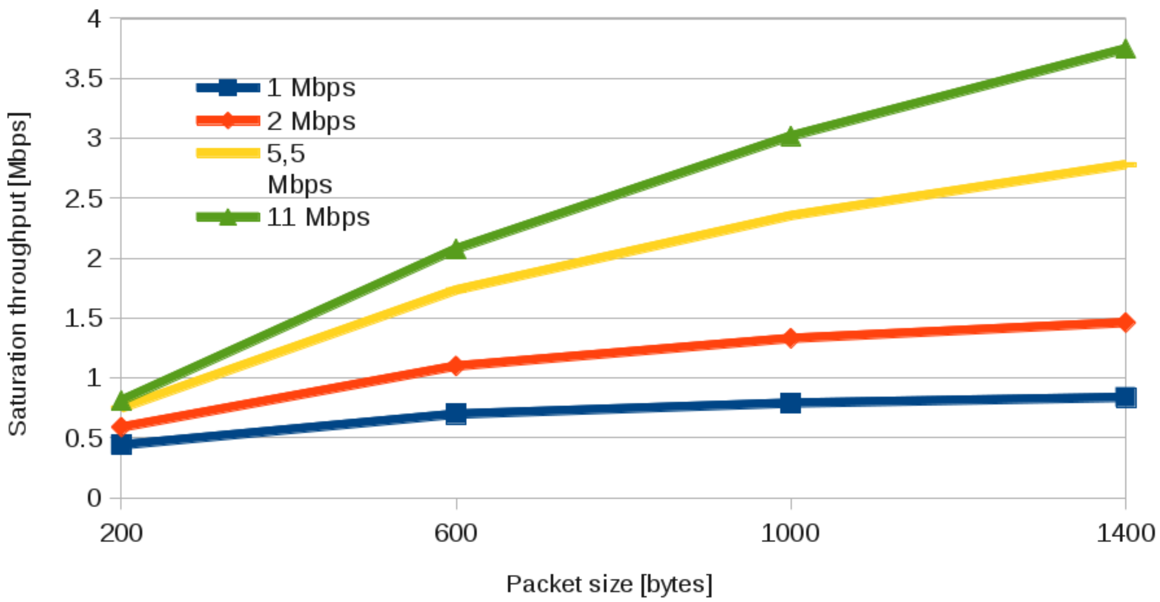

Figure 26: 802.11b graph saturation throughput for different packet size at 20km

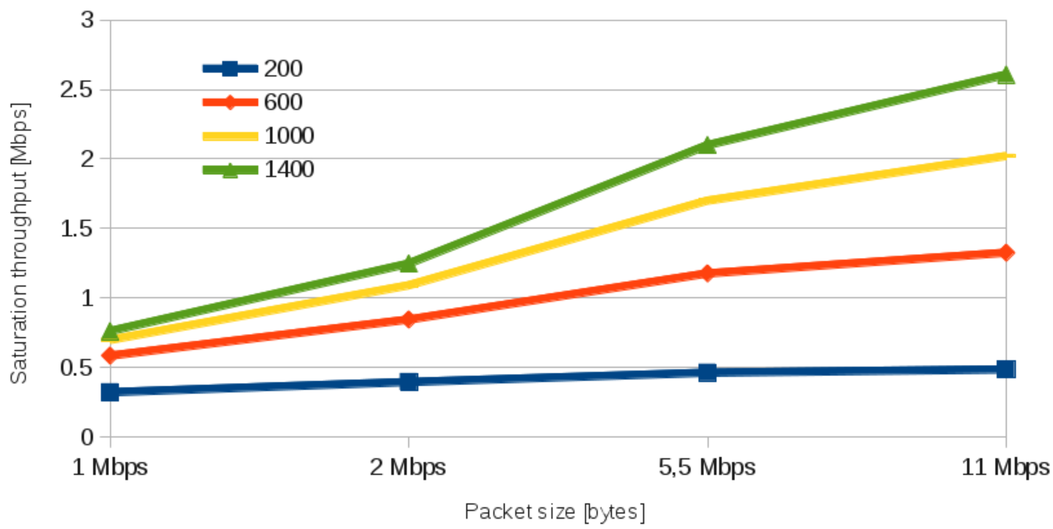

Figure 27: 802.11b graph saturation throughput for different packet size at 40km

Regarding the link budget and the usable bitrates, 802.11b works at the 2.4 GHz band where the FCC permits up to 30dBm of transmitting power with omnidirectional antennas of gain up to 6dBi for omnidirectional communications, and for every 3 dB of extra gain in the antenna for directive links the transmitted power must be reduced 1 dBm. A rough estimation of achievable distances obtained by using the free space model is presented in Table 35. A stability margin of 20dB has been preserved over the sensitivity for calculating the minimum acceptable received power level. We can see that almost any conceivable distance can be achieved with all the bitrates (distances up to 105 km are possible with many hardware products in 802.11b, but the performance is to low).

This subsection has been provided for completeness but no simulations are provided to complete these results as it does not make sense to use 802.11b nowadays. When DSSS is chosen as PHY (although this is usually senseless), 802.11g or 802.11n will be used instead.

D		5	10	15	20	25	30	35	40	45	50	55	60
Path Loss		114.1627218	120.1833217	123.7051469	126.2039216	128.1421219	129.7257468	131.0646826	132.2245215	133.247572	134.1627218	134.9905755	135.7463467
P_Rx		-46.16272177	-52.18332169	-55.70514687	-58.2039216	-60.14212186	-61.72574678	-63.06468257	-64.22452151	-65.24757196	-66.16272177	-66.99057548	-67.74634669
Bitrate	Sensitivity												
11 Mbps	-86												
5,5 Mbps	-89												
2 Mbps	-92												
1 Mbps	-95												

Table 35: Feasibility of point-to-point links at the different 802.11b bitrates. Color yellow between 15 and 20 dB, and color green means a margin over 20 dB

3.1.8 802.11g (2.4GHz, DSSS/OFDM, DCF)

The 802.11b amendment to the IEEE 802.11 standard was issued in 1999. Currently it is included in the new revision [IEEE 802.11-2012] as the DSSS PHY. Table 36 contains the main physical parameters.

Frequency band [GHz]	2.4
Channel bandwidth [MHz]	20
Bitrates [Mbps]	1, 2, 5.5, 6, 9, 11, 12, 18, 24, 36, 48, 54
PLCP overhead [us]	20 (OFDM), 96-192 (DSSS)
SIFS [us]	10
SlotTime [us]	9 (OFDM), 20 (DSSS)
DIFS [us]	28 (OFDM), 50 (DSSS)

Table 36: 802.11g PHY/MAC parameters

The following Figure 28 and Table 37 show how the maximum throughput evolves as the distance grows. Packets with an average size of 1500 bytes have been used to obtain these results. For the RTS/CTS mode, equivalent results are given in Figure 29 and Table 38.

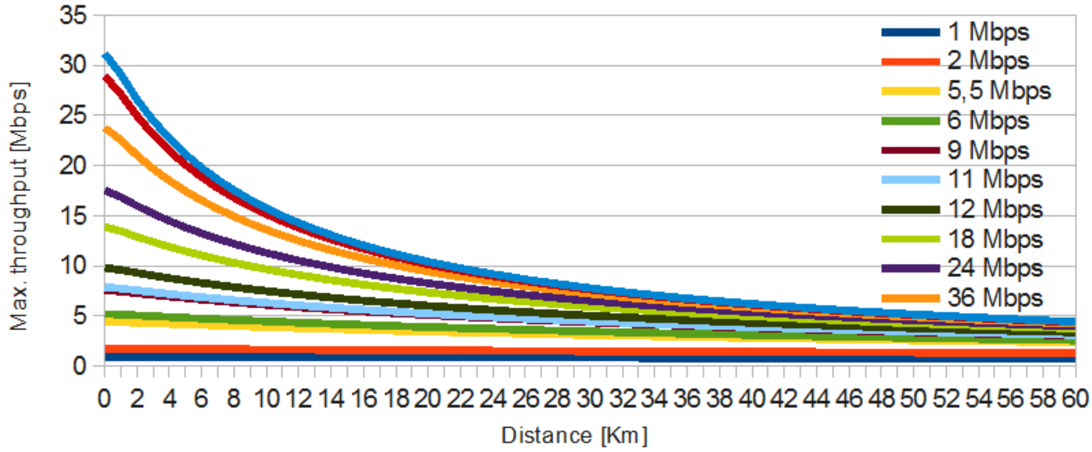

Figure 28: Evolution of the saturation throughput with the distance for 802.11g, basic mode.

Basic mode	D [km]									
	0	5	10	15	20	25	30	35	40	45	50
1 Mbps	0,904	0,892	0,879	0,867	0,855	0,843	0,832	0,821	0,810	0,800	0,789
2 Mbps	1,763	1,717	1,670	1,626	1,585	1,545	1,507	1,471	1,437	1,404	1,373
5,5 Mbps	4,453	4,170	3,906	3,674	3,467	3,283	3,117	2,967	2,831	2,707	2,593
6 Mbps	5,219	4,835	4,484	4,180	3,915	3,682	3,474	3,289	3,123	2,973	2,836
9 Mbps	7,589	6,803	6,128	5,574	5,113	4,722	4,386	4,095	3,840	3,615	3,415
11 Mbps	7,893	7,046	6,324	5,737	5,249	4,838	4,486	4,182	3,917	3,683	3,476
12 Mbps	9,818	8,541	7,503	6,690	6,036	5,498	5,049	4,667	4,339	4,054	3,804
18 Mbps	13,900	11,472	9,674	8,363	7,366	6,580	5,947	5,424	4,986	4,613	4,293
24 Mbps	17,548	13,848	11,311	9,559	8,277	7,299	6,527	5,903	5,388	4,955	4,587
36 Mbps	23,792	17,465	13,613	11,154	9,447	8,193	7,233	6,474	5,860	5,352	4,925
48 Mbps	28,942	20,089	15,156	12,169	10,165	8,728	7,647	6,804	6,128	5,575	5,113
54 Mbps	31,192	21,147	15,751	12,549	10,429	8,922	7,795	6,921	6,224	5,654	5,179

Table 37: Saturation throughput at different distances for 802.11g in basic mode

For the RTS/CTS mode, equivalent results are given in Figure 29 and Table 38.

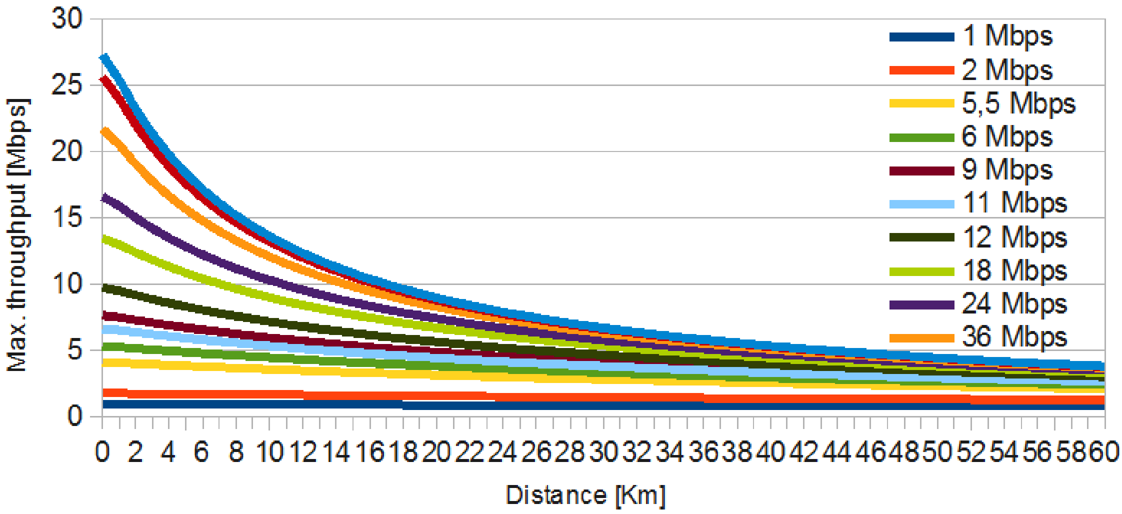

Figure 29: Evolution of the saturation throughput with the distance for 802.11g RTS/CTS mode

RTS/CTS mode	D [km]										
	0	5	10	15	20	25	30	35	40	45	50
1 Mbps	0,929	0,914	0,898	0,883	0,868	0,854	0,841	0,827	0,815	0,802	0,790
2 Mbps	1,762	1,708	1,655	1,604	1,557	1,512	1,470	1,430	1,392	1,357	1,323
5,5 Mbps	4,109	3,828	3,569	3,342	3,143	2,966	2,808	2,666	2,538	2,421	2,315
6 Mbps	5,331	4,868	4,456	4,109	3,812	3,555	3,330	3,132	2,956	2,799	2,658
9 Mbps	7,634	6,719	5,959	5,354	4,860	4,450	4,103	3,807	3,550	3,326	3,129
11 Mbps	6,632	5,930	5,331	4,841	4,434	4,089	3,795	3,540	3,317	3,121	2,946
12 Mbps	9,738	8,297	7,168	6,310	5,635	5,091	4,642	4,266	3,947	3,672	3,433
18 Mbps	13,442	10,842	8,992	7,681	6,704	5,947	5,344	4,852	4,443	4,098	3,802
24 Mbps	16,599	12,807	10,303	8,618	7,407	6,494	5,781	5,210	4,741	4,350	4,018
36 Mbps	21,693	15,641	12,061	9,815	8,274	7,151	6,296	5,624	5,082	4,635	4,260
48 Mbps	25,626	17,587	13,186	10,547	8,788	7,532	6,590	5,857	5,271	4,792	4,393
54 Mbps	27,274	18,348	13,609	10,816	8,974	7,668	6,694	5,939	5,338	4,847	4,439

Table 38: Saturation throughput at different distances for 802.11g, RTS/CTS mode

IEEE 802.11g is the almost the sum of the OFDM and the DSSS PHYs as defined in 802.11a and 802.11b. Hence, the results are very similar to the ones presented in the previous sections and the discussion is also similar.

The following Table 39 and 40 and Figures 30 and 31 show the evolution of the saturation throughput for 20 km and 40 km respectively as the packet size grows. It can be seen that, having the values for 200 bytes and for much bigger packets (e.g. 1500 bytes) the linear interpolation gives a good predictor of the maximum throughput that can be obtained for any other packet size.

Basic mode	P [bytes]			
	200	600	1000	1400
1 Mbps	0,526377	0,747381	0,815893	0,849257
2 Mbps	0,770879	1,274243	1,465649	1,566494
5,5 Mbps	1,094362	2,310925	2,971612	3,386558
6 Mbps	1,297989	2,671902	3,389443	3,830282
9 Mbps	1,421522	3,196976	4,261476	4,970822
11 Mbps	1,243444	3,010773	4,206540	5,069419
12 Mbps	1,492546	3,545334	4,890601	5,840362
18 Mbps	1,571041	3,978894	5,737656	7,078617
24 Mbps	1,613468	4,238029	6,281648	7,917990
36 Mbps	1,658251	4,533270	6,939598	8,983207
48 Mbps	1,681587	4,696872	7,323115	9,631045
54 Mbps	1,689513	4,754062	7,460551	9,868267

Table 39: Saturation throughput for 20 km as the packet size grows

Basic mode	P [bytes]			
	200	600	1000	1400
1 Mbps	0,419155	0,666668	0,755946	0,801973
2 Mbps	0,560791	1,056221	1,282896	1,412842
5,5 Mbps	0,714414	1,681468	2,305675	2,741905
6 Mbps	0,795927	1,864779	2,549534	3,025698
9 Mbps	0,840728	2,106208	3,013362	3,695507
11 Mbps	0,775078	2,023751	2,985789	3,749726
12 Mbps	0,865074	2,251987	3,314895	4,155462
18 Mbps	0,890873	2,419448	3,683485	4,746188
24 Mbps	0,904358	2,512878	3,900328	5,109352
36 Mbps	0,918257	2,613814	4,144298	5,532697
48 Mbps	0,925369	2,667385	4,278098	5,771814
54 Mbps	0,927764	2,685733	4,324639	5,856180

Table 40: Saturation throughput for 40 km as the packet size grows

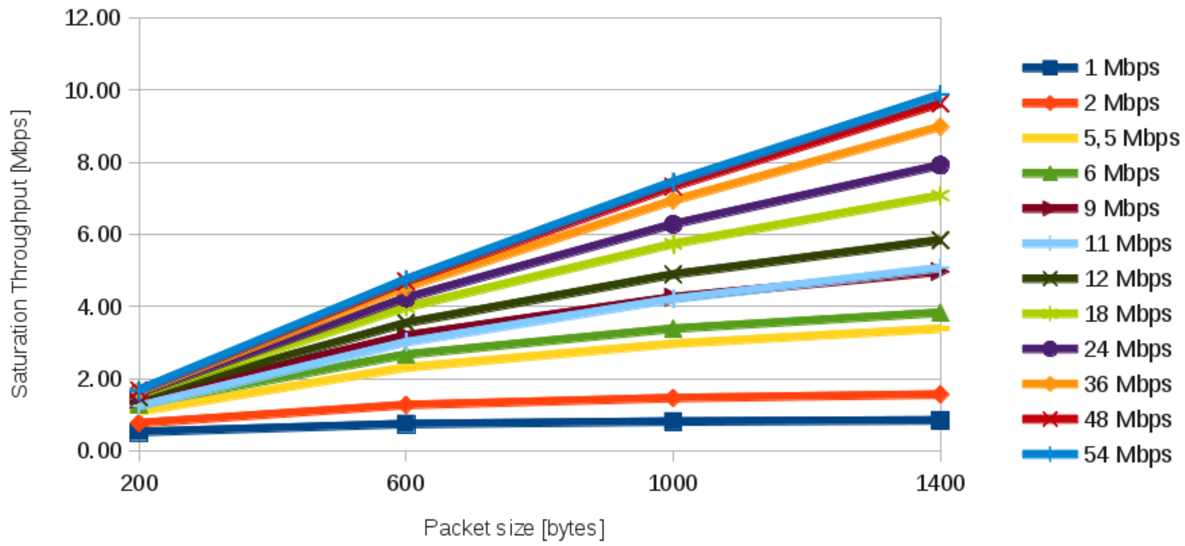

Figure 30: Evolution of the saturation throughput for 20 km as the packet size grows

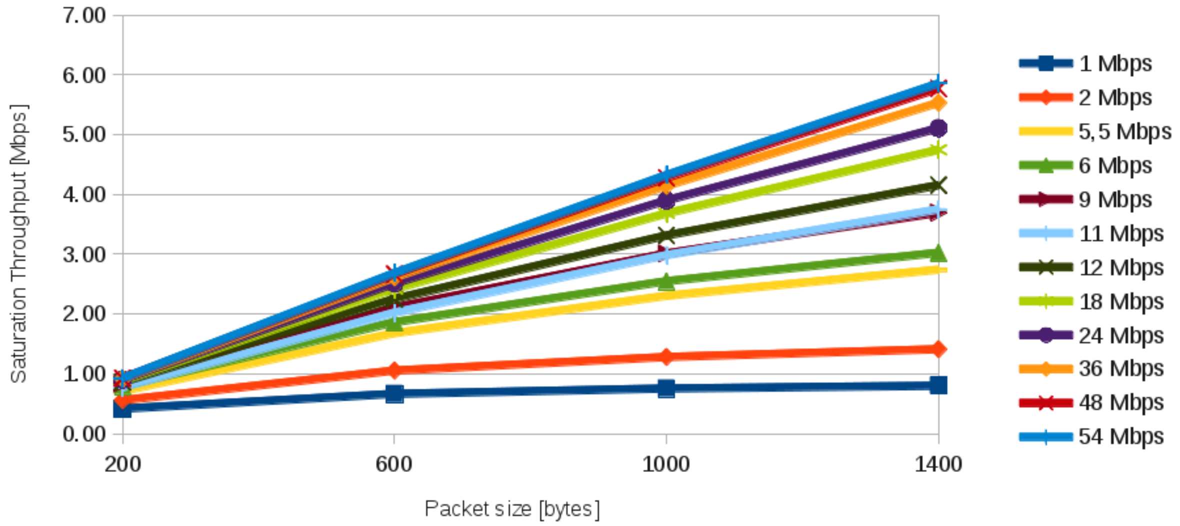

Figure 31: Evolution of the saturation throughput for 40 km as the packet size grows

Regarding the link budget and the usable bitrates, 802.11g works at the 2.4 GHz band and the same regulation is applied as for 802.11b. A rough estimation of achievable distances obtained by using the free space model is presented in Table 41. A stability margin of 20dB has been preserved over the sensitivity for calculating the minimum acceptable received power level.

D		5	10	15	20	25	30	35	40	45	50	55	60
Path Loss		114.1627218	120.1833217	123.7051469	126.2039216	128.1421219	129.7257468	131.0646826	132.2245215	133.247572	134.1627218	134.9905755	135.7463467
P_Rx		-46.16272177	-52.18332169	-55.70514687	-58.2039216	-60.14212186	-61.72574678	-63.06468257	-64.22452151	-65.24757196	-66.16272177	-66.99057548	-67.74634669
Bitrate	Sensitivity												
1 Mbps	-95												
2 Mbps	-92												
5,5 Mbps	-89												
6 Mbps	-91												
9 Mbps	-91												
11 Mbps	-86												
12 Mbps	-91												
18 Mbps	-90												
24 Mbps	-87												
36 Mbps	-84												
48 Mbps	-80												
54 Mbps	-77												

Table 41: Feasibility of point-to-point links at the different 802.11g bitrates. Colour red marks links with fading margin under 15 dB, colour yellow between 15 and 20 dB, and colour green means a margin over 20 dB.

As before, simulations have been made in order to contrast the theoretical model detailed above. One 802.11g point-to-point link has been simulated for different distances and different bitrates, sending a bidirectional UDP flow.

The results of the simulation of the 802.11g point-to-point link in Basic mode are shown next in Figure 32 and Table 42. As in previous simulations, a bidirectional UDP flow with 1500 frame size has been simulated. Although maybe redundant, Figure 33 shows the results of the same simulation, but in RTS/CTS mode. As predicted in the validation section, simulation values are very similar to the theoretical values

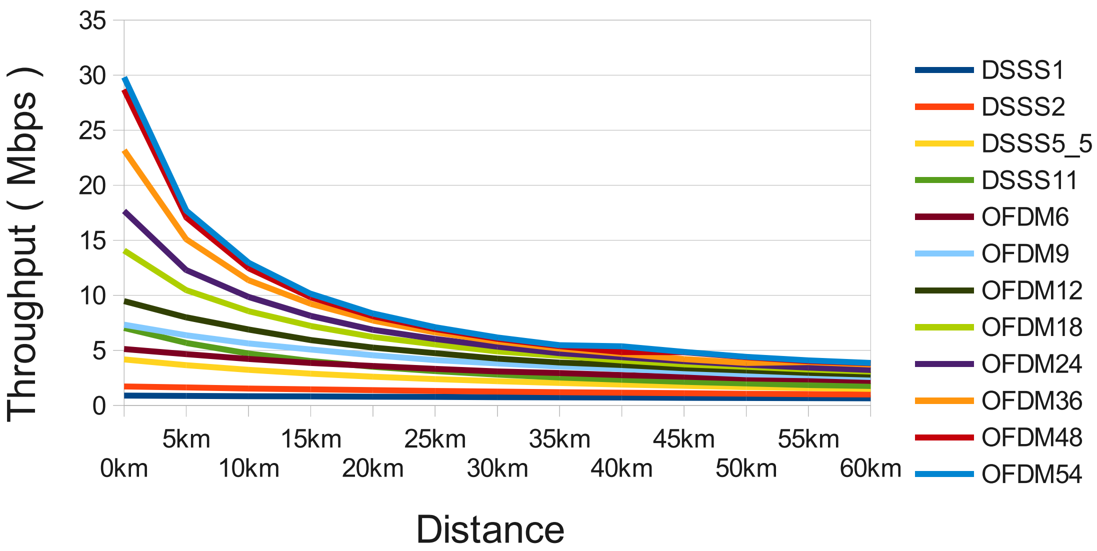

Figure 32: Simulation results - Bidirectional throughput for each bitrate in Basic Mode 802.11g

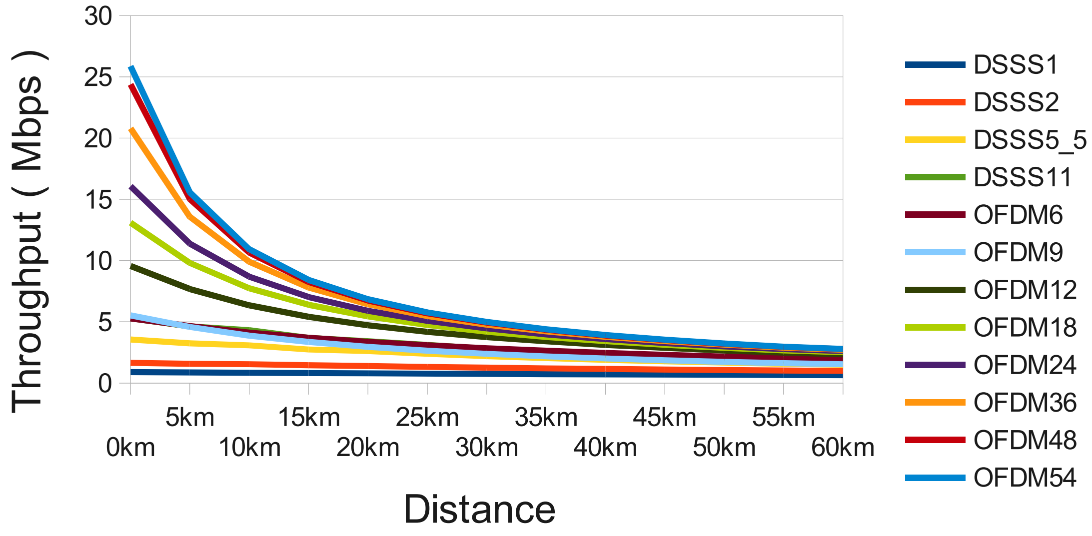

Figure 33: Throughput duplex for each bitrate in RTS/CTS - 802.11g

	0km	5km	10km	15km	20km	25km	30km	35km	40km	45km	50km	55km	60km
DsssRate1	0.89	0.87	0.85	0.84	0.82	0.81	0.81	0.79	0.77	0.76	0.75	0.74	0.74
DssRate2	1.70	1.65	1.59	1.54	1.50	1.44	1.41	1.37	1.32	1.29	1.22	1.21	1.20
DsssRate5_5	4.03	3.75	3.48	3.22	3.03	2.85	2.67	2.52	2.42	2.31	2.18	2.05	2.07
Ofdm6	5.13	4.65	4.22	3.86	3.58	3.32	3.08	2.93	2.74	2.60	2.44	2.36	2.31
DsssRate11	6.65	5.87	5.22	4.68	4.28	3.93	3.59	3.34	3.13	2.93	2.76	2.59	2.59
Ofdm9	7.33	6.37	5.64	5.07	4.55	4.12	3.81	3.54	3.29	3.07	2.83	2.73	2.70
Ofdm12	9.48	7.99	6.89	5.93	5.24	4.74	4.25	3.89	3.67	3.38	3.17	2.95	2.90
Ofdm18	14.0	10.4	8.56	7.21	6.21	5.55	4.89	4.49	4.08	3.73	3.51	3.26	3.18
Ofdm24	17.6	12.2	9.85	8.14	6.86	6.04	5.33	4.74	4.38	4.01	3.69	3.44	3.33
Ofdm36	23.1	15.0	11.3	9.23	7.71	6.62	5.81	5.16	4.68	4.21	3.85	3.93	3.87
Ofdm48	28.7	17.0	12.4	9.81	8.10	6.94	6.05	5.38	4.84	4.83	4.37	4.01	4.03
Ofdm54	29.8	17.7	12.9	10.1	8.34	7.10	6.16	5.47	5.37	4.86	4.41	4.09	4.03

Table 42: Values of the bidirectional throughput for each bitrate in Basic Mode - 802.11g

As before, delay values are shown in Figures 34 and 35 in the saturation point and below the saturation point of each bitrate respectively. Like expected, the higher bitrate, the lower is the delay.

Also values of jitter have been obtained. Maximum jitter values in the lowest bitrates have reached 2000 µs when simulating in the saturation point, but almost zero values has been obtained when simulating below of the saturation point. Regarding the packet loss, the absolute is zero for each distance and each bitrate because of the non-saturated conditions of the link.

For this reason, in order to get better performance, is desirable work below the saturation point, even if it means a throughput reduction.

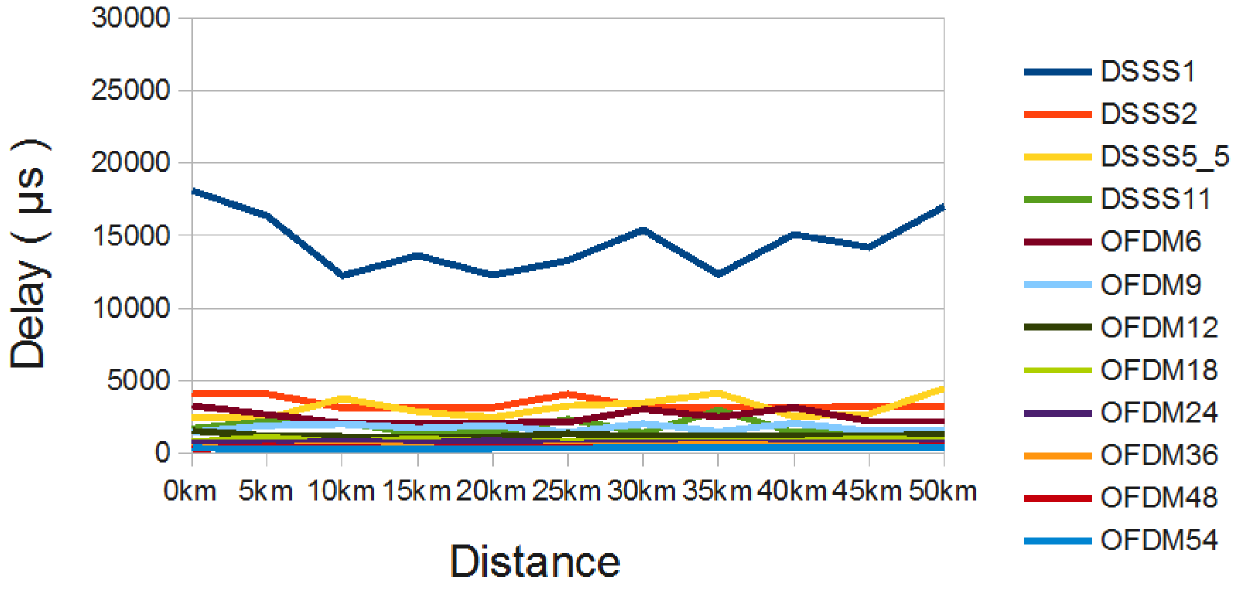

Figure 34: Delay obtained in the saturation point (maximum throughput)

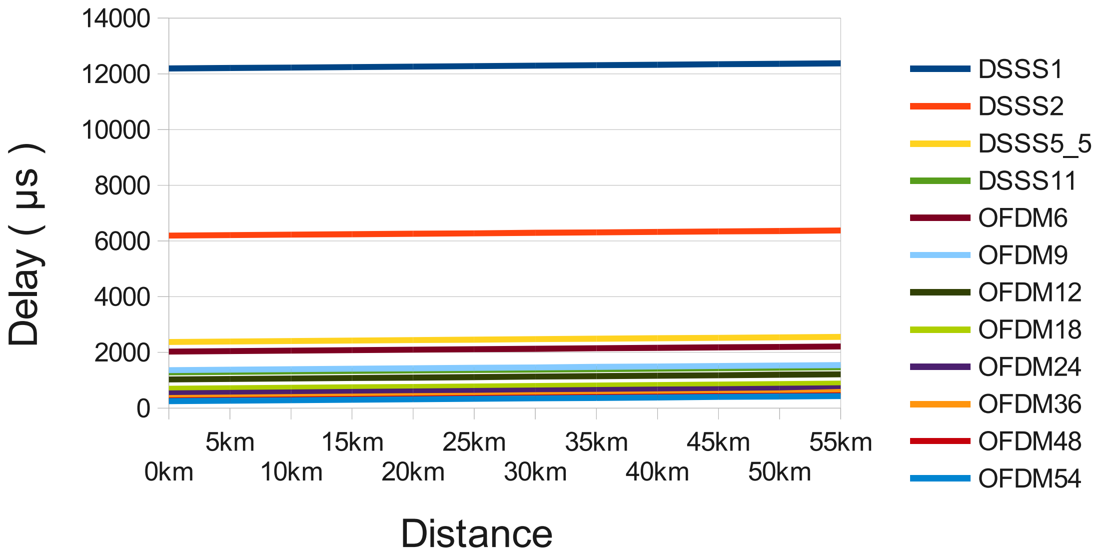

Figure 35: Delay obtained below the saturation point (more stable performance)

Last, as before, tests with real experiments have been made in the laboratory in order to settle the results of both theoretical model and simulations. The laboratory environment has been detailed previously: indoor test, Mikrotik equipments, distance 3 meters, bidirectional UDP traffic with 1500 bytes frame size, Iperf injecting software, clear-interference channel. The results, shown in Table 43, show the throughput and the jitter in the saturation point and coincide with both theoretical and simulation results.

	Throughput (Mbps)	Jitter (ms)
DSSS1Mbps	0,873	11.469
DSSS2Mbps	1,805	5.228
DSSS5_5Mbps	4.383	5.556
DSSS11Mbps	8,036	1.402
OFDM6Mbps	4,970	3.031
OFDM9Mbps	7.166	3.196
OFDM12Mbps	9.396	1.674
OFDM18Mbps	13.606	1.235
OFDM24Mbps	16.553	0.924
OFDM36Mbps	22.433	0.668
OFDM48Mbps	27.250	0.520
OFDM54Mbps	30.4	0.525

Table 43: Real experiments made in URJC Laboratory for 802.11g

3.1.9 802.11e (improvements when using EDCA instead of DCF)

The 802.11e amendment to the 802.11 standard permits to differentiate up to four ACs. AIFSN, CW_{min}, CW_{max} and TXOP may be given different values per AC. Saturation throughput, delay, jitter and packet loss for the different classes have been theoretically obtained using the implementation in Matlab of model proposed in [Hu2012].

[Biachi2005b] explains how the prioritization with AIFSN has a stronger impact on differentiation than acting on the contention window size. The main issue here is how much one must increase the AIFSN for less priority traffic to get the required effect, and how much that modifies the total throughput.

Distance[km]	Thr. 1 AIFSN=6 [Mbps]	Thr. 2 AIFSN=6 [Mbps]	Thr. 1 AIFSN=10 [Mbps]	Thr. 2 AIFSN=10 [Mbps]	Thr. 1 AIFSN=15 [Mbps]	Thr. 2 AIFSN=15 [Mbps]	Thr. 1 AIFSN=20 [Mbps]	Thr. 2 AIFSN=20 [Mbps]	Total Thr. AIFSN=6 [Mbps]	Total Thr. AIFSN=10 [Mbps]	Total Thr. AIFSN=15 [Mbps]	Total Thr. AIFSN=20 [Mbps]
2	4.4634	2.2832	5.2367	1.4523	5.5046	1.1390	5.5616	1.0110	6.7467	6.6890	6.6436	6.5726
24	2.1862	1.1409	2.4686	0.6631	2.6842	0.3305	2.8013	0.1537	3.3271	3.1317	3.0148	2.9549
48	1.4057	0.7266	1.5595	0.4189	1.6711	0.1957	1.7242	0.0968	2.1323	1.9784	1.8669	1.8209

Table 44: Saturation throughput per AC in 802.11g (11Mbps) with two traffic classes: VoIP and BE, at three distances

Distance[km]	D1 AIFSN=6 [ms]	D2 AIFSN=6 [ms]	D1 AIFSN=10 [ms]	D2 AIFSN=10 [ms]	D1 AIFSN=15 [ms]	D2 AIFSN=15 [ms]	D1 AIFSN=20 [ms]	D2 AIFSN=20 [ms]
2	95.1806	207.31845	58.24865	328.3494	16.4482	417.2404	8.3907	473.4324
24	217.4609	420.8345	192.28195	727.53995	175.43525	1454.20635	167.8082	3108.36545
48	341.8925	664.13845	307.00845	1150.07435	285.71375	2454.35605	273.7383	4880.9314

Table 45: Saturation delay per AC in 802.11g (11 Mbps) with two traffic classes: VoIP and BE, at three distances

The results of saturation throughput and saturation delay per AC shown in Tables 44 and 45 permit to see several important aspects of EDCA behavior over long distances. Firstly, the total throughput is penalized with traffic differentiation (about 15% less for AIFSN=20 as compared with IFSN=6). Secondly, we may effectively obtain a delay reduction for real time traffic by using EDCA. However, the saturation delay is still too high for real-time traffic. Figure 36 permits to see this more clearly.

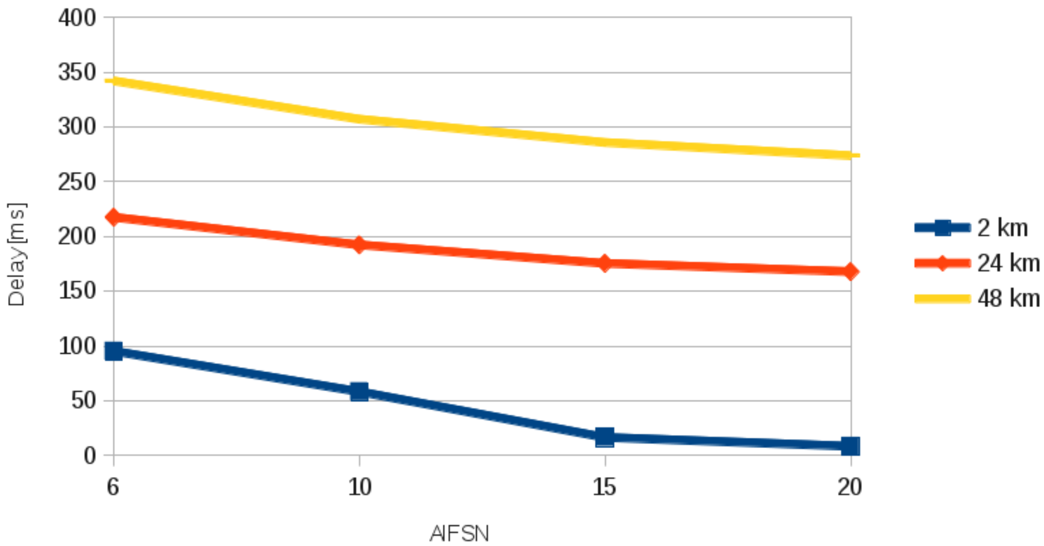

Figure 36: Saturation delay for VoIP traffic as a function of AIFSN used for BE traffic. The 11 Mbps bitrate in 802.11g is used

Only three distances have been represented, but the effect is pretty clear. While the AIFSN prioritization effectively gets a reduction in the delay for VoIP traffic (augmenting the AIFSN value for the BE traffic), for distances longer than 24 km get saturation delays higher than 150 ms, which may be considered as the limit for one-way end-to-end delay for voice traffic, following the ITU-T recommendations. And this is the average (not maximum) saturation delay for one single hop.

The obvious corollary is that we cannot work in saturation conditions, even using EDCA, when real-time traffic has to be transported. On the other hand, several papers studying the unsaturated delay [Malone2007, Zhao2009] show that the average delay in unsaturated conditions strictly depends on the collision probability and, for n=2 (point-to-point), we have already shown that this probability is very low and independent on the distance. It has been already shown that the collision probability depends on the contention window size.

For W=7, recommended value in the standard for VoIP traffic, p=0,128 and the average delay is about 4 ms. This is a more acceptable value for one-hop delay in a multihop network transporting real-time traffic. Hence, EDCA is a mean but not a guarantee to obtain a valid delay for priority traffic. EDCA can be used for a better result, but the link must be kept under the saturation point. This must be assured at higher layers, for example using traffic shaping and advanced queuing disciplines in order to control lower priority traffic, and then using access control for real-time communications. This will be developed for the case of the heterogeneous transport network in deliverable D52.

However, although delay depends on the collision probability, propagation delay must be borne in mind if this probability is low. According the simulations, for small size frames similar to VoIP traffic, such as 200 bytes, it is possible achieve very low delay and only distance dependent when the link is below enough of the saturation point. Figure 37 and Table 46 show that in non-saturated conditions, the delay is totally lineal with the distance.

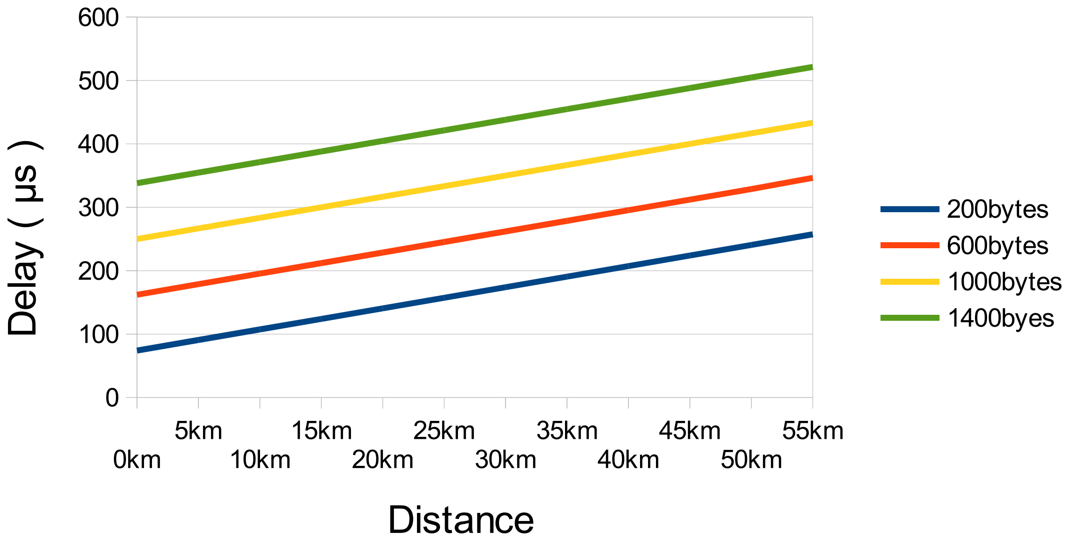

Figure 37: Delay for OFDM 12Mbps bitrate working below the saturation point for different frame sizes

	200bytes	600bytes	1000bytes	1400byes
0km	74.016	162.016	250.016	338.016
5km	90.683	178.683	266.683	354.683
10km	107.349	195.349	283.349	371.349
15km	124.016	212.016	300.016	388.016
20km	140.666	228.683	316.683	404.683
25km	157.333	245.349	333.349	421.349
30km	173.999	262.016	350.016	438.016
35km	190.666	278.683	366.683	454.683
40km	207.333	295.349	383.349	471.349
45km	223.999	312.016	400.016	488.016
50km	240.666	328.683	416.683	504.683
55km	257.333	346.5391594	433.349	521.349

Table 46: Delay Values in µs for OFDM 12Mbps bitrate working below the saturation point for different frame sizes

In case that a link must ensure absolute priority for the access category ACj, it is possible to do it giving this category the smallest AIFSN value and adjusting also the contention window limits so that AIFSNi > AIFSNj + CWmin,j, CWmax = CWmin, for i≠j.

Additionally to AIFSN or CWmin prioritisation, TXOP may also be used, possibly together with the Block-ACK mechanism, for traffic differentiation. [Salmer2008] showed that Block-ACK mechanism permits to foster the throughput and to keep the delay under control in long-distance WiFi links. The first improvement is obvious, but the second does not seem evident. The explanation to these results are in Figure 38 as well: if several packets may be sent together each time a station wins the contention for the channel, that proportionally reduces the collision probability; with the TXOP mechanism a packet never waits longer to be transmitted, as the pending packets are sent together once the station gets the channel, but that does not mean that the station has to wait for other packets to arrive to the queue. The negative effect in the delay is that, when the other station gets the channel, one has to wait longer until the channel gets free again. Both effects compensate to each other and the result seems to be advantageous for the use of TXOP.

On the other hand, TXOP may be used for another interesting purpose; if different values are given to both ends of the link for the same traffic class, which permits that end-1 gets more throughput than end-2 proportunal to TXOP1/TXOP2 under saturation conditions. When the traffic is not symmetric, TXOP will permit to tune the uplink/downlink ratio as desired.

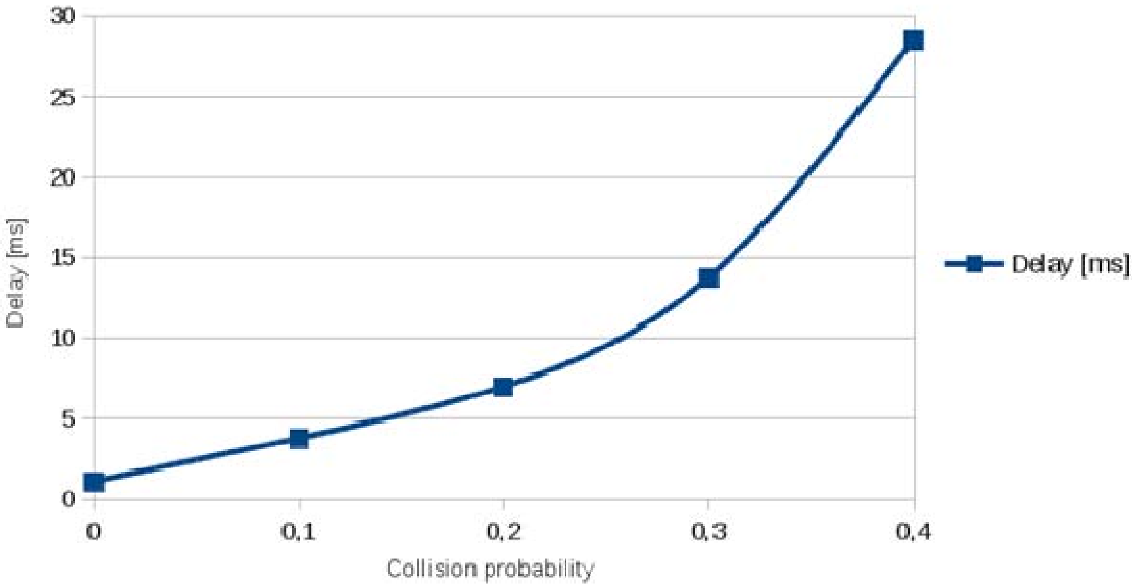

Figure 38: MAC delay as a function of the collision probability, from [Malone2007].

3.1.10 802.11n (2.4/5GHz, OFDM, EDCA)

802.11n introduces many mechanisms aiming to obtain hight throughput, as explained in Section 2. However, the protocol overhead is still very important and only through intensive frame aggregation at the physical layer the performance is dramatically improved.

Figures 39 and 40 and Tables 47 and 48 show the saturation throughput for all MCS with either one or two spatial streams. Results are only shown for 20 Mbps, for basic and RTS/CTS mode respectively.

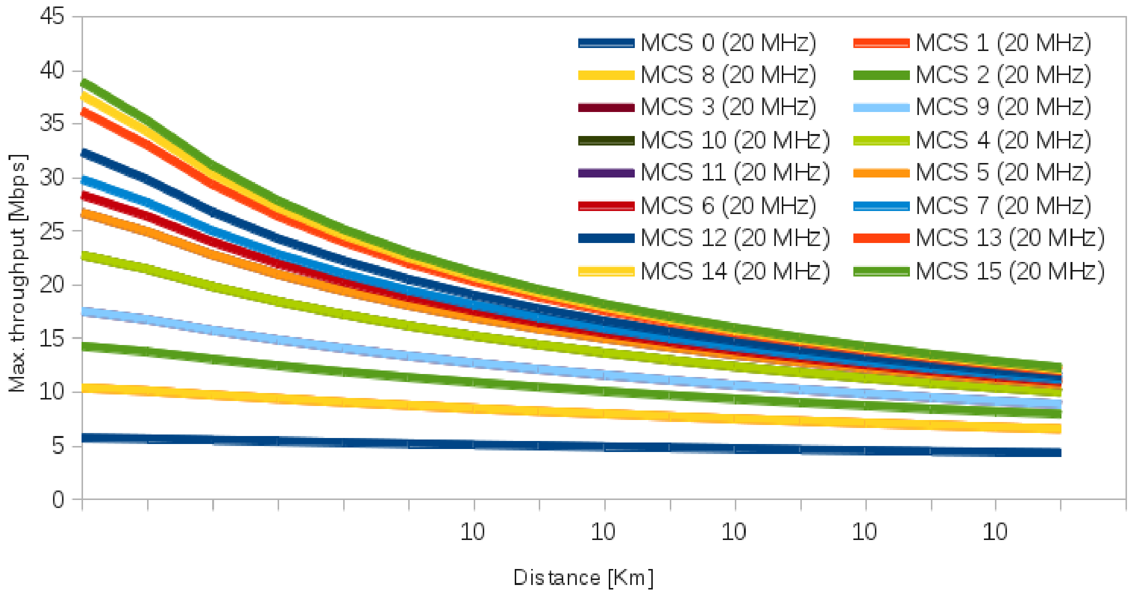

Figure 39: Saturation throughput for RTS/CTS mode in bitrates with one or two spatial streams in 802.11n, with 20 MHz channel. Packets are 1500 bytes long and no frame aggregation or Block-ACK is used

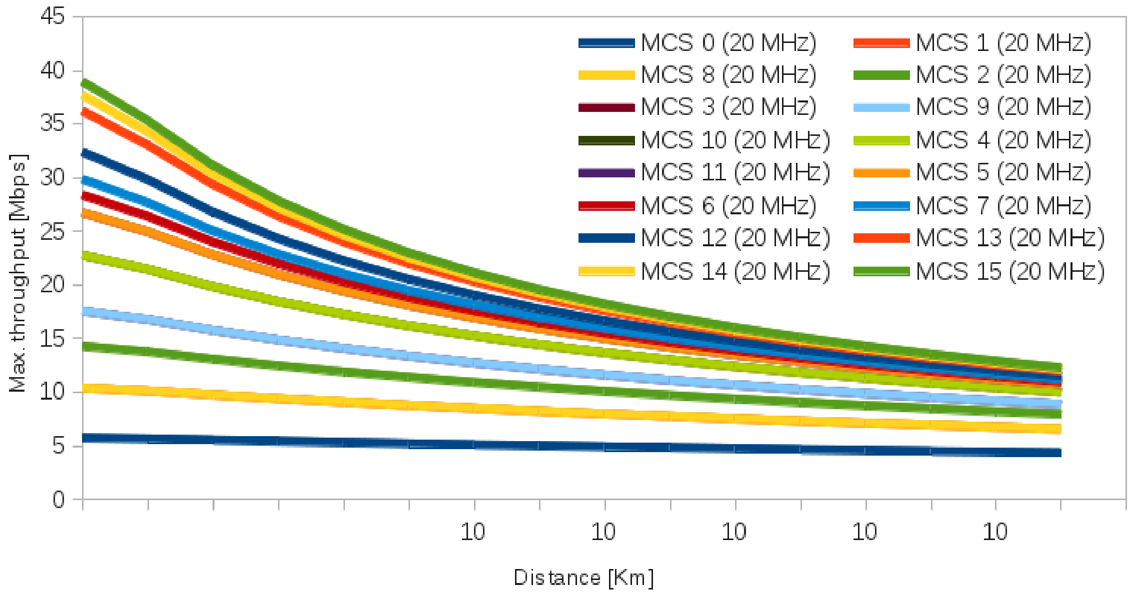

Figure 40: Saturation throughput for RTS/CTS mode in bitrates with one or two spatial streams in 802.11n, with 20 MHz channel. Packets are 1500 bytes long and no frame aggregation or Block-ACK is used

	0km	5km	10km	15km	20km	25km	30km	35km	40km	45km	50km
MCS 0 (20 MHz)	5,728	5,197	4,731	4,341	4,011	3,727	3,481	3,265	3,075	2,905	2,753
MCS 1 (20 MHz)	10,394	8,768	7,517	6,579	5,849	5,264	4,786	4,388	4,050	3,761	3,511
MCS 8 (20 MHz)	10,394	8,768	7,517	6,579	5,849	5,264	4,786	4,388	4,050	3,761	3,511
MCS 2 (20 MHz)	14,267	11,373	9,354	7,944	6,903	6,104	5,470	4,956	4,530	4,171	3,865
MCS 3 (20 MHz)	17,534	13,356	10,656	8,863	7,587	6,632	5,891	5,299	4,814	4,411	4,071
MCS 9 (20 MHz)	17,534	13,356	10,656	8,863	7,587	6,632	5,891	5,299	4,814	4,411	4,071
MCS 10 (20 MHz)	22,741	16,178	12,378	10,024	8,422	7,261	6,382	5,692	5,137	4,681	4,299
MCS 4 (20 MHz)	22,741	16,178	12,378	10,024	8,422	7,261	6,382	5,692	5,137	4,681	4,299
MCS 11 (20 MHz)	26,707	18,089	13,467	10,726	8,912	7,623	6,659	5,912	5,316	4,829	4,423
MCS 5 (20 MHz)	26,707	18,089	13,467	10,726	8,912	7,623	6,659	5,912	5,316	4,829	4,423
MCS 6 (20 MHz)	28,356	18,831	13,873	10,982	9,088	7,751	6,757	5,989	5,378	4,880	4,466
MCS 7 (20 MHz)	29,829	19,469	14,217	11,196	9,234	7,857	6,838	6,052	5,429	4,922	4,501
MCS 12 (20 MHz)	32,349	20,512	14,765	11,533	9,462	8,022	6,962	6,150	5,507	4,986	4,555
MCS 13 (20 MHz)	36,169	21,985	15,513	11,985	9,764	8,238	7,124	6,276	5,608	5,068	4,624
MCS 14 (20 MHz)	37,651	22,524	15,779	12,143	9,869	8,312	7,180	6,319	5,642	5,096	4,647
MCS 15 (20 MHz)	38,927	22,974	15,999	12,273	9,954	8,373	7,225	6,354	5,670	5,119	4,666

Table 47: Saturation throughput for 802.11n, RTS/CTS mode, with average packet size of 1500 bytes, no frame aggregation

	0km	5km	10km	15km	20km	25km	30km	35km	40km	45km	50km
MCS 0 (20 MHz)	5,728	5,197	4,731	4,341	4,011	3,727	3,481	3,265	3,075	2,905	2,753
MCS 1 (20 MHz)	10,394	8,768	7,517	6,579	5,849	5,264	4,786	4,388	4,050	3,761	3,511
MCS 8 (20 MHz)	10,394	8,768	7,517	6,579	5,849	5,264	4,786	4,388	4,050	3,761	3,511
MCS 2 (20 MHz)	14,267	11,373	9,354	7,944	6,903	6,104	5,470	4,956	4,530	4,171	3,865
MCS 3 (20 MHz)	17,534	13,356	10,656	8,863	7,587	6,632	5,891	5,299	4,814	4,411	4,071
MCS 9 (20 MHz)	17,534	13,356	10,656	8,863	7,587	6,632	5,891	5,299	4,814	4,411	4,071
MCS 10 (20 MHz)	22,741	16,178	12,378	10,024	8,422	7,261	6,382	5,692	5,137	4,681	4,299
MCS 4 (20 MHz)	22,741	16,178	12,378	10,024	8,422	7,261	6,382	5,692	5,137	4,681	4,299
MCS 11 (20 MHz)	26,707	18,089	13,467	10,726	8,912	7,623	6,659	5,912	5,316	4,829	4,423
MCS 5 (20 MHz)	26,707	18,089	13,467	10,726	8,912	7,623	6,659	5,912	5,316	4,829	4,423
MCS 6 (20 MHz)	28,356	18,831	13,873	10,982	9,088	7,751	6,757	5,989	5,378	4,880	4,466
MCS 7 (20 MHz)	29,829	19,469	14,217	11,196	9,234	7,857	6,838	6,052	5,429	4,922	4,501
MCS 12 (20 MHz)	32,349	20,512	14,765	11,533	9,462	8,022	6,962	6,150	5,507	4,986	4,555
MCS 13 (20 MHz)	36,169	21,985	15,513	11,985	9,764	8,238	7,124	6,276	5,608	5,068	4,624
MCS 14 (20 MHz)	37,651	22,524	15,779	12,143	9,869	8,312	7,180	6,319	5,642	5,096	4,647
MCS 15 (20 MHz)	38,927	22,974	15,999	12,273	9,954	8,373	7,225	6,354	5,670	5,119	4,666

Table 48: Saturation throughput for 802.11n, RTS/CTS mode, with average packet size of 1500 bytes, no frame aggregation.

This two figures show results with packets of 1500 bytes, without any frame aggregation or block-ACK techniques, for a 20 MHz channel. We can see how high throughput capacity is considerably lost in the first kilometers. However, in Figure 41 and Table 49 we show the same results for basic mode but using frame aggregation up to 64000 (the maximum aggregation size is 65535 bytes). Now we can see that the result is completely different as the high throughputs are achievable for long distances.

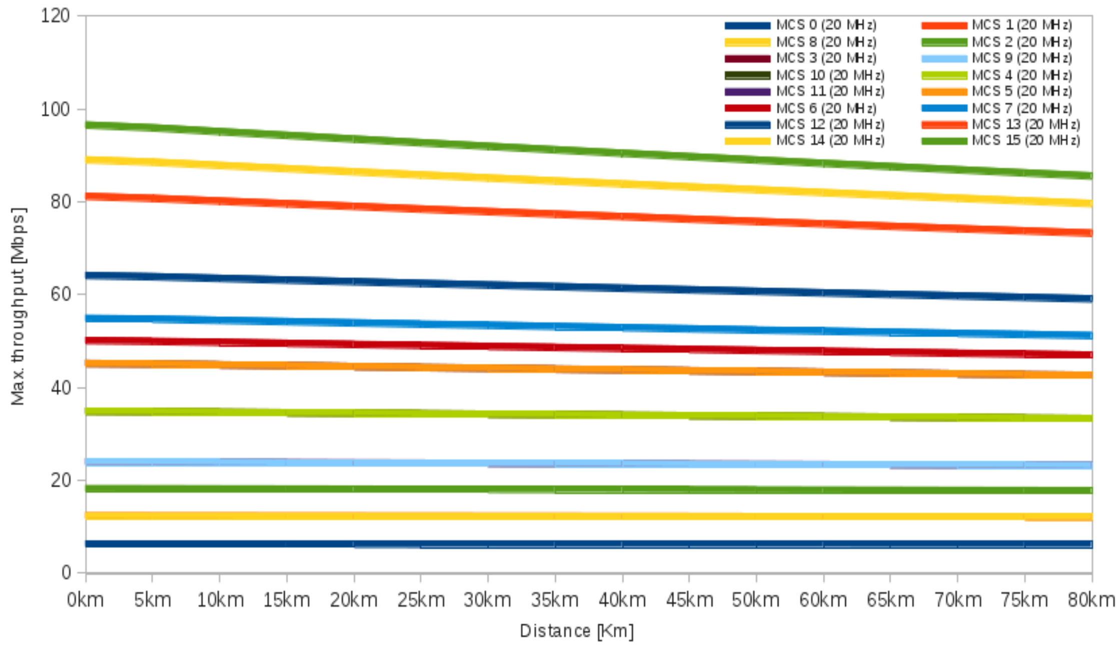

Figure 41: Saturation throughput for MCS 0.15 in 802.11n, basic mode, with aggregated payload of 64000 bytes

	0km	5km	10km	15km	20km	25km	30km	35km	40km	45km	50km
MCS 0 (20 MHz)	6,272	6,255	6,238	6,220	6,203	6,185	6,168	6,151	6,134	6,117	6,100
MCS 1 (20 MHz)	12,349	12,285	12,217	12,150	12,084	12,018	11,953	11,889	11,825	11,762	11,700
MCS 8 (20 MHz)	12,349	12,285	12,217	12,150	12,084	12,018	11,953	11,889	11,825	11,762	11,700
MCS 2 (20 MHz)	18,241	18,101	17,954	17,809	17,667	17,527	17,390	17,254	17,121	16,989	16,860
MCS 3 (20 MHz)	23,956	23,714	23,463	23,217	22,976	22,740	22,508	22,282	22,060	21,842	21,629
MCS 9 (20 MHz)	23,956	23,714	23,463	23,217	22,976	22,740	22,508	22,282	22,060	21,842	21,629
MCS 10 (20 MHz)	34,884	34,375	33,849	33,339	32,844	32,364	31,898	31,444	31,004	30,576	30,159
MCS 4 (20 MHz)	34,884	34,375	33,849	33,339	32,844	32,364	31,898	31,444	31,004	30,576	30,159
MCS 11 (20 MHz)	45,192	44,341	43,470	42,633	41,827	41,052	40,304	39,583	38,888	38,216	37,567
MCS 5 (20 MHz)	45,192	44,341	43,470	42,633	41,827	41,052	40,304	39,583	38,888	38,216	37,567
MCS 6 (20 MHz)	50,130	49,085	48,020	47,000	46,023	45,086	44,185	43,321	42,489	41,689	40,918
MCS 7 (20 MHz)	54,932	53,680	52,409	51,196	50,039	48,932	47,874	46,860	45,889	44,957	44,062
MCS 12 (20 MHz)	64,149	62,447	60,734	59,111	57,574	56,114	54,727	53,406	52,148	50,947	49,801
MCS 13 (20 MHz)	81,173	78,468	75,781	73,272	70,924	68,722	66,652	64,704	62,866	61,129	59,486
MCS 14 (20 MHz)	89,050	85,805	82,603	79,631	76,865	74,285	71,873	69,612	67,489	65,492	63,610
MCS 15 (20 MHz)	96,546	92,743	89,013	85,571	82,386	79,429	76,677	74,110	71,709	69,458	67,345

Table 49: Saturation throughput in basic mode for 802.11n using frame aggregation with 65535 bytes of aggregated size.

In order to see the importance of the frame aggregation quantitatively, we show in Figures 42 and 43 the evolution of the maximum throughput for MCS0 and MCS15 respectively for different aggregated payload sizes.

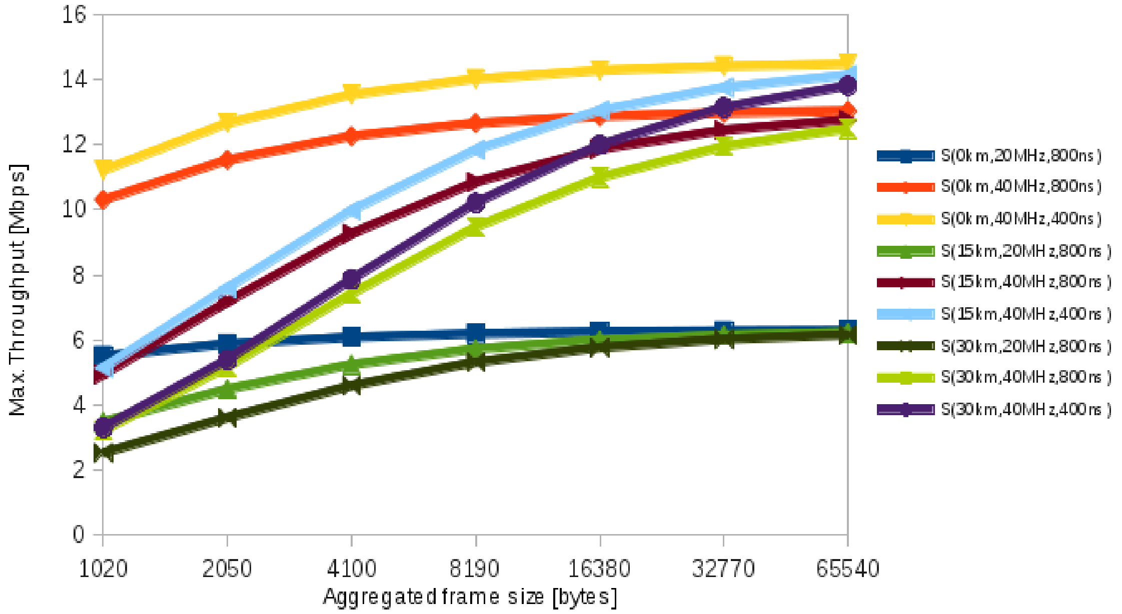

Figure 42: Saturation throughput for MCS0, aggregated frame payload size between 1023 and 65535 bytes, 20 MHz channels and 40 MHz channels, distances of 0 km, 15 km and 30 km. For each distance, the effect of the short GI is also shown.

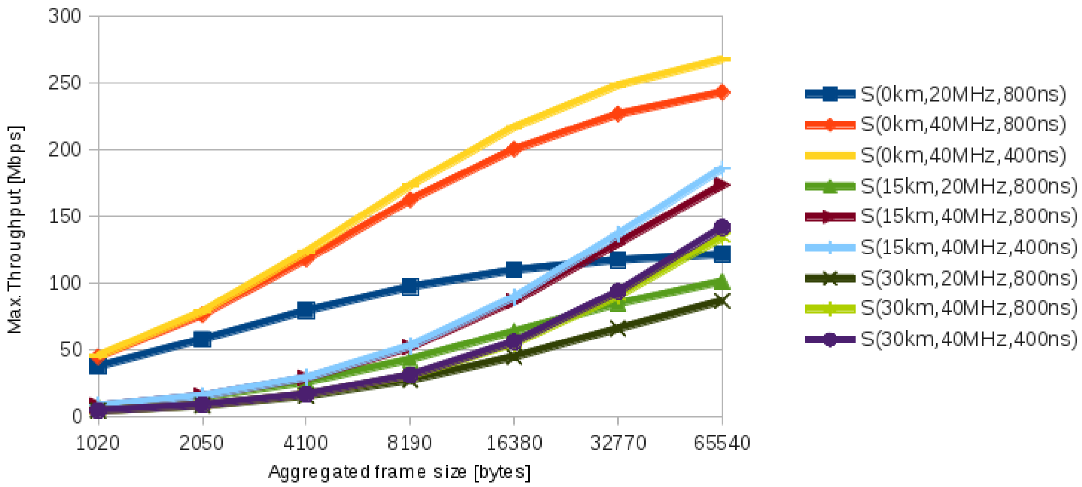

Figure 43: Saturation throughput for MCS15, aggregated frame payload size between 1023 and 65535 bytes, 20 MHz channels and 40 MHz channels, distances of 0 km, 15 km and 30 km. For each distance, the effect of the short GI is also shown.

We can see that, at long distances wide 40 MHz channels and short GI permit to obtain a saturation throughput as high as 6 Mbps almost for any average frame size and without aggregation. When frame aggregation is strongly used, the result achieved is the same with long GI. This permits to protect the link against multipath with long GI when necessary. Figure 43 shows equivalent results for MCS15.

The aggregation has a disadvantage related with the average delay. Figure 44 shows the impact of the aggregation on the transmission delay. When aggregated frames are too long, the transmission delay becomes significant (even tens of ms) if the bitrate is low. Note that the other components of the delay follow the same rules as for the other 802.11 versions; the delays when working in saturation conditions become unaffordable for real-time traffic, which makes advisable to limit the offered load in order to ensure that the link works in unsaturated conditions. In these conditions, delays are a bit lower than those of 802.11g, due to the higher physical bitrates and other optimized timings.

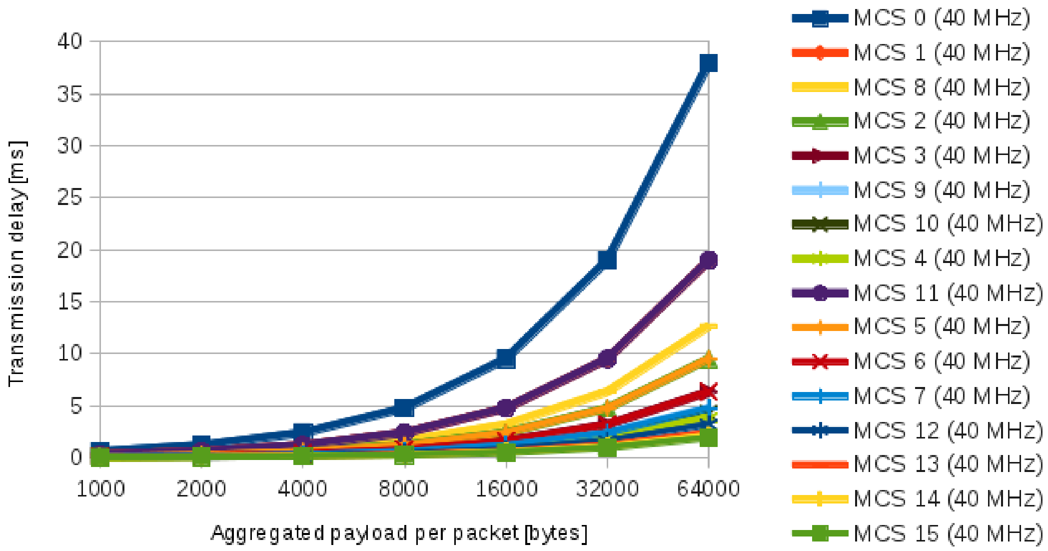

Figure 44: Delay for each modulation regarding the aggregation level

Due to the degradation of the delay and packet-loss figures that happen as the offered load approaches the saturation throughput, it is important to find out what is the highest load level (lower than the saturation throughput) that still corresponds to a low average delay and a low packet-loss. A set of tests have been simulated in NS-3 in order to find this points for different MCS at different distances. These points correspond to the maximum throughput v alues for which the delay is still low and controllable.

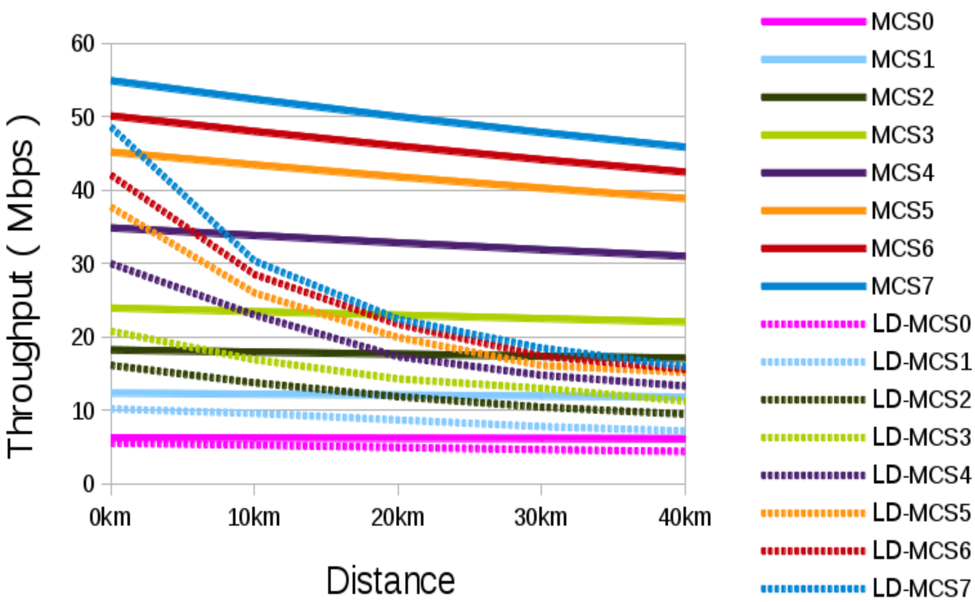

Figure 45: Comparison between maximum throughput values predicted by the model and the optimum points that keep the delay low.

Figure 45 shows a comparison between saturation throughput and highest Low-Delay throughput for each MCS at different distances. Only SISO modes have been simulated with 20MHz channel bandwidth, long GI, 1500 packet size and frame aggregation threshold of 8192 bytes. In continuum lines, the maximum theoretical throughput is shown and in dashed lines the optimal low-delay throughput is obtained through simulation with NS-3. Each modulation is represented in one different color in order to associate simulation and theoretical model.

It also illustrates that the optimal point decreases significantly with the distance for the highest MCS. However this behavior is not present in the lowest MCS. For short distances, the optimum throughput is around 80% of the maximum throughput for each modulation. However it behaves differntly for longer distances. The relationship between maximum and optimal throughput is shown in Table 50.

	0km		10km		20km		30km		40km	
	R	Delay (ms)	R	Delay (ms)	R	Delay (ms)	R	Delay (ms)	R	Delay (ms)
MCS0	88.49%	2.45	84.17%	25.05	79.81%	31.18	75.39%	31.21	71.73%	33.69
MCS1	82.60%	1.57	78.58%	13.22	72.00%	20.36	65.26%	20.73	60.89%	21.27
MCS2	88.46%	7.8	76.67%	18.59	67.19%	20.99	60.08%	25.58	55.44%	29.21
MCS3	86.83%	5.60	72.03%	11.74	62.24%	18.37	57.76%	0.59	51.06%	13.01
MCS4	86.00%	6.25	67.95%	17.68	52.77%	11.74	46.37%	7.19	43.00%	7.02
MCS5	83.42%	4.14	59.81%	11.50	47.65%	9.56	40.00%	12.71	39.09%	28.00
MCS6	83.88%	5.27	59.38%	9.50	47.26%	16.81	39.38%	16.89	36.72%	27.04
MCS7	88.47%	5.52	57.93%	10.14	44.85%	15.95	38.64%	20.33	34.87%	80.59

Table 50: Relationship between the theoretical and optimum throughputs with the distance in 802.11n. R is the fraction of the optimum throughput between the maximum throughput achievable. Also is shown the delay obtained in the optimum point of throughput.

As conclusion, Table 50 suggests that, in order to keep delays low, the offered load in 802.11n (and a/b/g) long distance links must be reduced drastically with the distance. Better throughput can be achieved for long distances but the delay will be also dramatically increased.

As in previously chapters, a rough estimation of achievable distances obtained by using the free space model is presented in Table 51. The same considerations regarding sensitivity, transmission power and antenna gains have been considered.

D		5	10	15	20	25	30	35	40	45	50	55	60
Path Loss		121.4968972	127.5174971	131.0393223	133.538097	135.4762973	137.0599222	138.398858	139.5586969	140.5817474	141.4968972	142.3247509	143.0805221
P_Rx		-47.4968972	-53.51749711	-57.03932229	-59.53809703	-61.47629729	-63.05992221	-64.398858	-65.55869694	-66.58174739	-67.4968972	-68.3247509	-69.08052212
Mode	Sensitivity												
MCS0	-96												
MCS1	-95												
MCS2	-92												
MCS3	-90												
MCS4	-86												
MCS5	-83												
MCS6	-77												
MCS7	-74												
MCS8	-95												
MCS9	-93												
MCS10	-90												
MCS11	-87												
MCS12	-84												
MCS13	-79												
MCS14	-78												
MCS15	-75												

Table 51: Feasibility of long distance links with 802.11n for the different MCS.

In order to contrast this analysis, a set of experimental results are shown. In the beginning of this chapter the laboratory environment and the methods for the test with real equipment have been detailed. Tables 52, 53 and 54 show the values of the throughput, jitter, delay in the saturation point of the link, and the delay measured below the saturation point. The experiments were done for each MCS 0-15, corresponding to SISO (0-7) and MIMO (8-15) with 2 spatial flows (2x2). One bidirectional UPD flow with 1528 MAC frame size was used to test the performance of the link. Frame aggregation technique was used with a threshold of 8192 bytes, equivalent to 5 MAC frames per A-MSDU. Also BlockAck technique was used with 5 block-acks.

Bidirectional GI = 800ns 20 MHz				
MCS	Throughput Saturation Point (Mbps)	Jitter Saturation Point (ms)	Delay Saturation Point (ms)	Delay No saturation (ms)
0	5.76	7.98	425.39	72.22
1	11.25	6.28	421.955	55.38
2	16.72	5.53	174.015	54.12
3	22.73	4.89	171.02	48.76
4	34.03	4.29	123.14	41.11
5	45.03	3.09	119.83	28.64
6	50.83	3.07	105.58	27.42
7	55.73	2.86	105.33	26.31
8	11.40	4.31	487.88	69.09
9	22.67	2.13	269.44	55.13
10	30.90	1.24	162.05	51.07
11	44.31	0.81	118.33	43.80
12	66.82	0.51	77.94	40.01
13	81.22	0.45	55.61	28.36
14	83.80	0.74	55.11	23.3
15	94.43	0.92	52.15	21.6

Table 52: Performance values obtained from real experiments in 802.11n / GI=800ns / BW=20MHz.

Bidirectional GI = 800ns 40 MHz				
MCS	Throughput Saturation Point (Mbps)	Jitter Saturation Point (ms)	Delay Saturation Point (ms)	Delay Below saturation point (ms)
0	11.88	4.238	412.70	115.32
1	23.63	1.662	201.66	77.11
2	35.06	1.645	163.01	36.21
3	47.30	1.105	123.83	31.18
4	69.74	0.864	80.44	27.42
5	92.83	0.692	54.77	25.25
6	103.64	0.495	50.80	19.04
7	115.97	0.463	44.11	17.6
8	24.03	1.69	237.66	109.11
9	45.70	0.90	115.27	70.94
10	68.86	0.51	80.77	53.35
11	86.23	0.643	65.01	42.72
12	136.9	0.32	42.21	32.20
13	178.5	0.30	32.63	24.90
14	189.16	0.22	27.82	17.66
15	196.02	0.26	24.30	16.97

Table 53: Performance values obtained from real experiments in 802.11n / GI=800ns / BW=40MHz

Bidirectional GI = 400ns 40 MHz				
MCS	Throughput Saturation Point (Mbps)	Jitter Saturation Point (ms)	Delay Saturation Point (ms)	Delay Below saturation point (ms)
0	10.75	4.28	364.11	75.05
1	26.26	1.97	202.61	62.10
2	39.30	1.37	144.72	54.10
3	52.46	1.29	93.01	43.07
4	76.57	0.83	71.83	25.46
5	101.93	0.70	57.01	21.20
6	115.24	0.49	45.21	20.17
7	126.93	0.34	46.18	16.25
8	24.10	1.16	198.88	68.15
9	51.16	0.60	105.88	59.60
10	76.76	0.46	70.27	51.01
11	101.66	0.32	48.53	38.27
12	143.76	0.29	34.59	30.21
13	195.20	0.38	30.87	22.27
14	219.67	0.28	26.60	15.76
15	240.33	0.24	23.80	13.72

Table 54: Performance values obtained from real experiments in 802.11n / GI=400ns / BW=40MHz

20 MHz channel GI 800ns Frames 1500 bytes		Without Frame Aggregation			Frame Aggregation: Threshold 8192 bytes		
		Thr. (Mbps)	Jitter (ms)	Delay (ms)	Thr. (Mbps)	Jitter (ms)	Delay (ms)
Experimental	MCS 4	23.22	0.81	123.14	31.93	2.29	165.77
Simulation	MCS 4	21.53	0.43	57.56	31.04	0.95	92.04

Table 55: Comparison between simulation and experimental test in frame aggregation performance

The effects of the frame aggregation have been also tested and simulated. Simulations were made using a 802.11n point-to-point link, sending a bidirectional UDP flow with frame size 1500 at a saturating bitrate. The PHY mode used MCS 4 (OfdmRate39MbpsBW20MHz). The same link was tested with same parameters in Mikrotik equipment. The results are shown in Table 55. As expected, both simulation and experimental test show that using frame aggregation techniques increase the throughput but also the delay. Also it is possible observe that the delay in saturating conditions is much bigger than in unsaturated conditions showed previously.

The throughput results are very close in simulations to those obtained in experimental tests. The differences in delay and jitter must be attributed to differences in buffering that is not easily estimated in real hardware products. In order to clarify the frame aggregation effect, the evolution of the throughput for different aggregation thresholds is shown in Figure 46. A 1526 byte packet size has been used and the same configuration as before: 20MHz channel bandwidth and 800ns of GI.

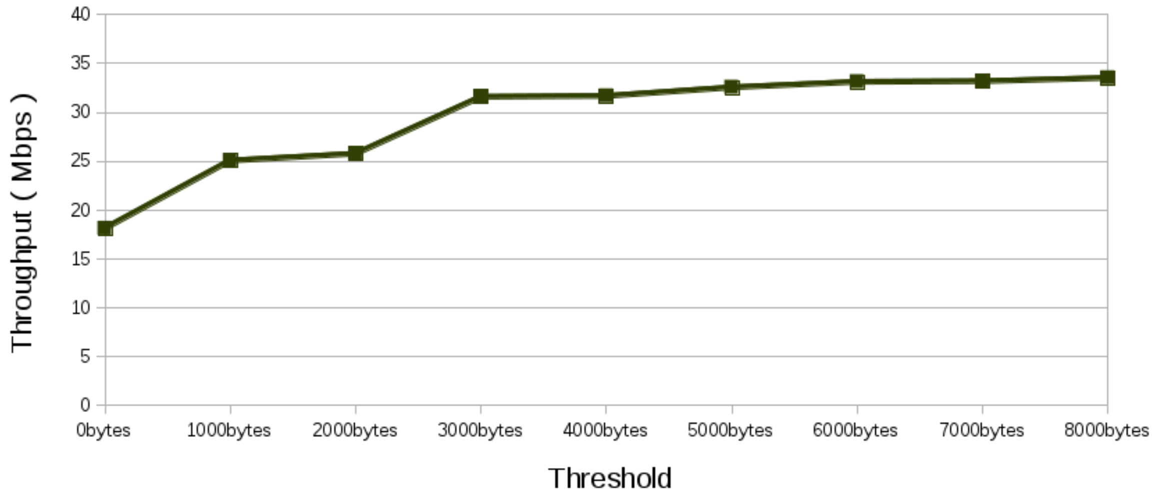

Figure 46: Evolution of the throughput for 802.11n with the aggregation threshold

Finally, real experiments in the real wireless network of Napo river have been done remotely. This network it is presented in the research project proposal of TUCAN3G. The link tested is a 29.6 Km distance link located between the cities Negro Urco and Tuta Pishcon and have a SISO configuration. A saturation test was made, in order to test its real performance. The link is configured with MCS4, GI=800ns, and 20MHz channel bandwidth to get the best reliability and stability. 1500 byte size packets were sent in a bidirectional UDP flow for different load values. Neither frame aggregation nor block-ack was used. Results are shown in Table 56. The saturation point is found around 6.4 Mbps, which is consistent with theoretical results.

Load per flow	Throughput (Mbps)	Jitter (ms)	Delay (ms)	Packet loss
1 Mbps	2	0.512	2.03	0 %
2 Mbps	4	0.82	1.87	0 %
3 Mbps	6	2.06	4.36	0 %
3.1Mbps	6.2	0.575	1.27	0 %
3.2 Mbps	6.4	0.75	15.22	0.1 %
3.5 Mbps	5.7	4.543	1743	14.5 %
4 Mbps	5.59	6.154	1913	25.9 (%)

Table 56: Performance results in a 29.6 Km distance link using 802.11n MCS4.

3.1.11 A comment about 802.11ac

As explained in Section 2, the new amendment 802.11ac will increase the performance of 802.11 based on the following contributions:

- Channels up to 160 MHz width, not necessarily contiguous.
- MIMO with up to 8 spatial streams.
- New modulations up to 256QAM.

However, its contribution for long distance solutions is fairly clear. New modulations with a higher number of symbols will require a much higher SNR, thus being exclusively useful for very short distances, even with high-gain antennas. MIMO with more than 2 spatial streams is not practical for long-distance shots with line of sight, as explained in this document. Finally, wide channels may effectively increase the performance, but it must be also understood that multi-hop networks require that contiguous links user non-overlapping channels. Hence, the available bandwidth for a single link in such circumstances might not be higher than 40 MHz in some scenarios, depending on the local regulations.

In general, the only clear improvement that 802.11ac may introduce is an increase in the capacity when 80 MHz or 160 MHz channels may be used, consequently multiplying the capacity two or four times respectively, or alternatively reducing the delay for equivalent throughputs. When available channels cannot be higher than 40 MHz, 802.11ac does not make any difference for long links compared to 802.11n.

3.1.12 Non-standard TDMA alternatives: the case for Ubiquiti Airmax and Mikrotik Nstreme

Ubiquiti AirOS systems using AirMAX or Mikrotik Routerboard Systems using NV2 are two cases of commercial proprietary solutions that replace the standard CSMA/CA for a TDMA MAC. Neither of them has published the technical details of the implementation of these TDMA solutions, but both seem to be very simple. QoS is left for the IP layer and most of the details of the TDMA are either fix (and hidden) or automatic. Enabling or disabling the TDMA replacement for the standard CSMA MAC is just a matter of ticking one option in the management interface. Hence, all we can do for the sake of comparison with standard 802.11 is a set of measurements.

The same tests as in 802.11n have been made for same MCSs in order to get a slight comparison with the IEEE 802.11n standard. The link whas tested with Iperf sending a saturating UDP flow with frame size 1526 bytes for both TDMA protocols, NV2 and AirMAX. The results are shown below in the Table 57, 58 and 59.

		Bidirectional GI = 800ns 20 MHz		
	MCS	Throughput (Mbps)	Jitter (ms)	Delay (ms)
NV2	0	4.93	4.76	13.22
	7	50.34	2.61	3.5
	8	10.06	2.31	8.76
	15	90.1	0.54	5.11
AirMAX	0	4.96	6.17	30
	7	40.1	2.72	25.57
	8	9.75	2.80	52.87
	15	79.63	0.83	19.1

Table 57: Performance comparison between Mikrotik NV2 and AirMAX for GI=800ns and BW=20MHz.

	MCS	Bidirectional GI = 800ns 40 MHz Throughput (Mbps)	Jitter (ms)	Delay (ms)
NV2	0	10.4	2.38	8.43
	7	99.86	0.50	3.79
	8	22.60	1.94	6.51
	15	174.96	0.46	5.02
AirMAX	0	12.92	3.2	32.03
	7	78.63	0.72	11.73
	8	22.22	1.30	27.9
	15	146.36	0.51	10.01

Table 58: Performance comparison between NV2 and AirMAX for GI=800ns and BW=40MHz

	MCS	Bidirectional GI = 400ns 40 MHz Throughput (Mbps)	Jitter (ms)	Delay (ms)
NV2	0	11.52	2.321	7.64
	7	112.46	0.522	3.91
	8	20.54	1.91	6.36
	15	196.63	0.60	3.89
AirMAX	0	X	X	X
	7	X	X	X
	8	X	X	X
	15	X	X	X

Table 59: Performance comparison between NV2 and AirMAX for GI=400ns and BW=40MHz

In Table 59, no results of AirMAX are shown because its operating system, AirOS, does not allow changing the GI to 400ns using TDMA techniques. The results casted by the previous tables show that the throughput is lower than in 802.11n, but the delay is lower too. It is an expected result due to TDMA techniques that reduce delay at expense of a higher protocol overhead. However, while this performance remains almost unchanged for each distance, the performance of the 802.11 technologies is dramatically affected by the distance despite of using the same techniques.

3.2 Discussion of results

3.2.1 Distances achieved, taking the regulatory environment into account

Results in [Simo2010] show that even the most permissive hardware products use to have restrictions that limit the achievable distance; this is a first restriction that must be taken into account. For example, AR5xxx Atheros chipsets limit to 105 km the use of 802.11b, 55 km the use of 802.11a, and to 49 km the use of 802.11g.

In terms of power link budget, Tables 27, 35, 41 and 51 show what the limits are for the different bitrates. However, those results must be taken as a rough approximation that depends on the regulatory restrictions (FCC's regulations have been taken as the reference), weather, environment and several other factors. Achievable distances have been calculated taking a minimum of 20dB of fading margin. This should not be considered as something optional, because the channel fluctuates with changes in the environmental conditions and fading happens. However, it can be necessary to foresee a higher margin for certain scenarios.

An additional consideration is related to auto-rate algorithms. The standard does not define how and when exactly a system must switch from one transmission scheme to another and each manufacturer implements its own strategy. The auto-rate algorithm for specific equipment must be carefully tested to ensure that the best performance is obtained for any conditions. If a system needs the received power to drop too deeply to perform auto-rate fallback, the link may be operating in a “high” bitrate but with a high frame error rate, thus offering poor results. In those cases, it might be advisable to fix the bitrate to a lower value that assures the stability.

Results show that long-distance links are feasible with WiFI, although the performance may be significantly degraded compared to short distances. If CSMA/CA is used, it is fundamental to adapt the SlotTime, ACKTimeout and CTSTimeout parameters as explained in [Simo2010] so that the logics under the contention system is respected. If ACKTimeout and CTSTimeout are increased in twice the propagation time, then ACKs and CTSs will be received normally. If SlotTime is increased in twice the propagation time, then the collision probability will stay in reasonable values. It must be recalled that the maximum throughput is not achieved for this adjustment (a shorter SlotTime value maximizes the throughput), but the fairness and a more stable delay require to do this way.

3.2.2 Saturation throughput and available capacity under optimal conditions

Results have been obtained for the saturation throughput as a function of the distance. It can be seen that the distance does not degrade the performance for all the bitrates equally. The faster is a transmission scheme, the faster it slows down with the distance.

Figures and tables with saturation throughput for different distances have been provided for the different versions of WiFi. It has been shown that 802.11b does not offer any advantage over 802.11g (unless distance limitations in the hardware avoid the use of the other versions, as it may happen with atheros chipsets beyond 49 km). In case of using 802.11g, it results obvious that OFDM modes are preferred unless the link budget forces the use of 1 or 2 Mbps. We may also see a great impact of the packet size on the saturation throughput. If the proportion of small packets in the total traffic is high, the maximum throughput will be importantly reduced. This effect may be partially attenuated by using Block-ACK and TXOP in EDCA.

There are also no advantages at using 802.11a OFDM or 802.11g OFDM if 802.11n is available. 802.11n offers similar profiles and sensitivities for GI=800ns and BW=20 MHz in the first 8 MCS, but it also offers physical frame aggregation (better than block-ACK), reduced GI, doubled BW and other MCS with two spatial streams, all of them permitting to increase the performance.

The following Figure 47 shows the comparison of saturation throughput between 802.11b, 802.11g, 802.11a and 802.11n for average packet size of 1500 bytes, without any frame aggregation or block-ACK technique. Figure 48 shows that comparison with big payload, obtained in 802.11n with frame aggregation (63000 bytes aggregated) and with Block-ACK combining 42 packets of 1500 bytes each in 802.11a/g (this is possible for bitrates higher than 18 Mbps, calculating that the total transaction enters in the maximum TXOP admitted in the standard). In the comparisons, 802.11n has been employed with BW=40 MHz and GI=800ns. In these figures, the link budget has been taken into account for choosing the highest realistic bitrate for each distance. In this sense, data in Tables 27, 35, 41 and 51 have been used.

Note that frame aggregation is used in 802.11n instead of Block-ACK. Whenever it is possible, physical frame aggregation is preferred due to its efficiency. Figure 49 compares both techniques and validates this affirmation.

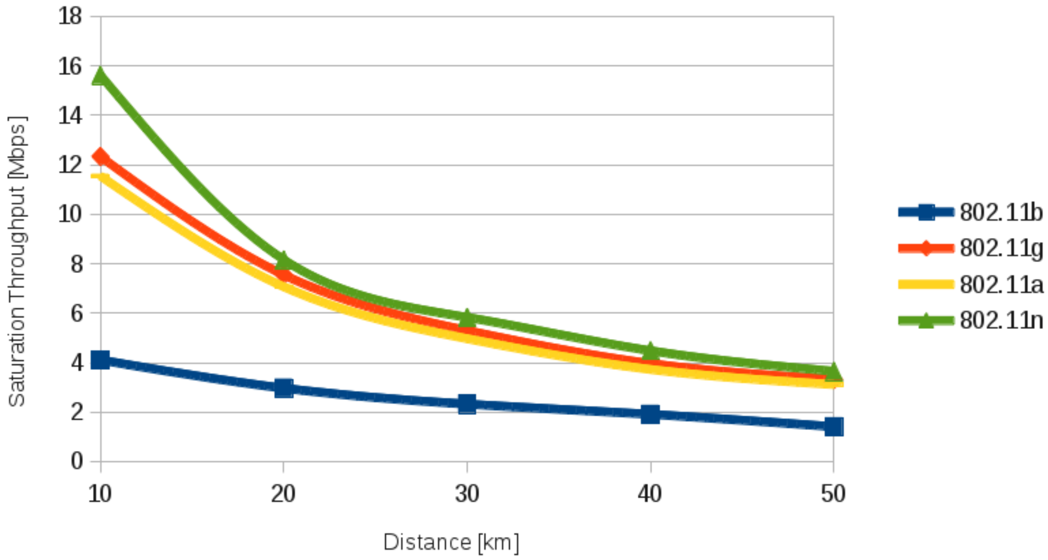

Figure 47: Comparison between 802.11b, 802.11g, 802.11a and 802.11n with 1500 bytes of packet size, without any aggregation or block-ACK technique. 11n is using 40 MHz channel.

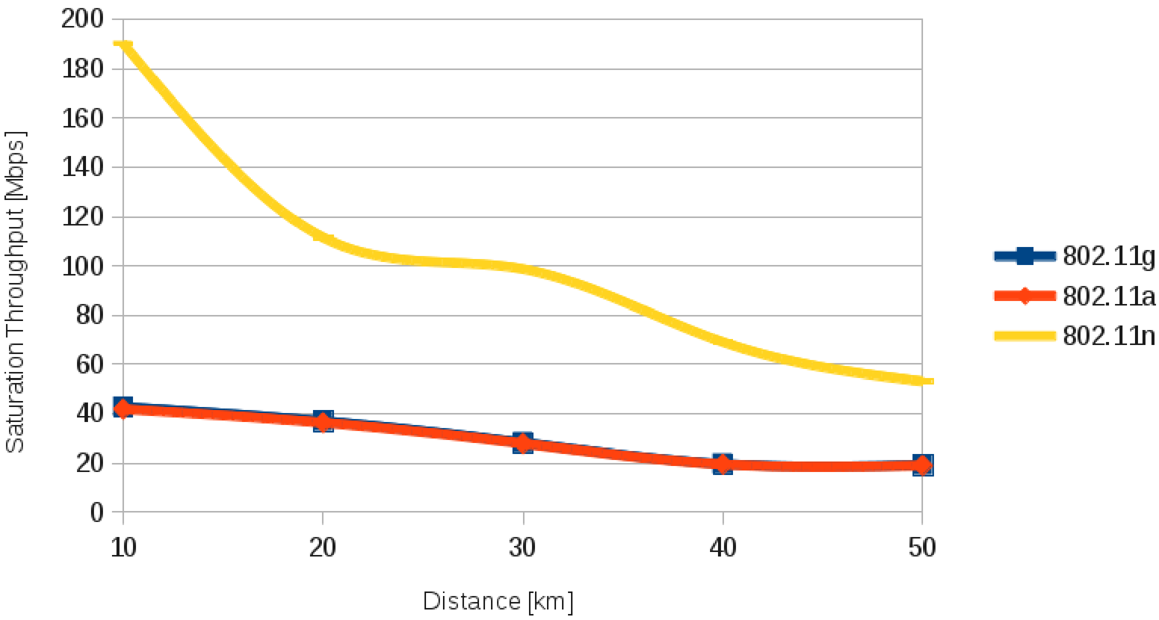

Figure 48: Comparison between 802.11b, 802.11g, 802.11a and 802.11n with 1500 bytes of packet size, using frame aggregation up to 63000 aggregated bytes in 11n, or 42 packets in a Block-ACK for the rest. 11n is using 40 MHz channel and GI=800ns

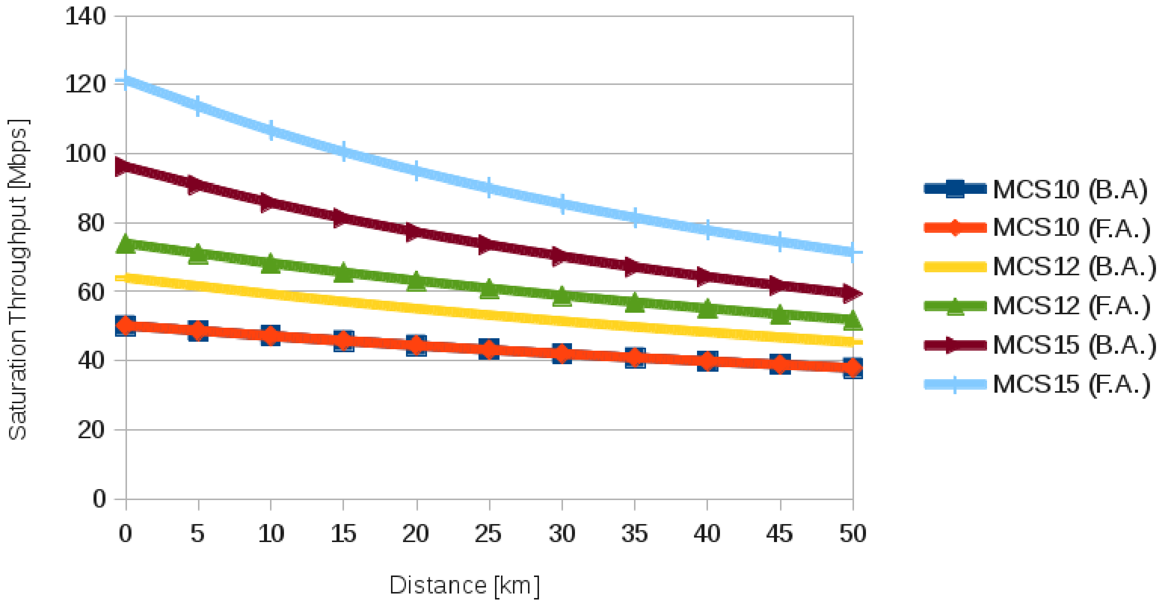

Figure 49: Comparison between frame aggregation (63000 bytes) and Block-ACK (42 x 1500 bytes) for MCS 10, 12 and 15.

The previous Figures show that 802.11n is always better and that frame aggregation permit to foster the saturation throughput.

Regarding the non-standard technologies, Figure 50 compares the performance in 802.11n with and NV2 and AirMAX obtained in the indoor test-beds. Better performance is obtained for 802.11n in all the cases, but these results will no be the same for longer distances, since 802.11n will perform under NV2 and AirMAX. Hence, NV2 is seen to be the best option for mid-range or long-range WiLD PtP links. Even AirMAX will be much faster than plain 802.11n over long distances.

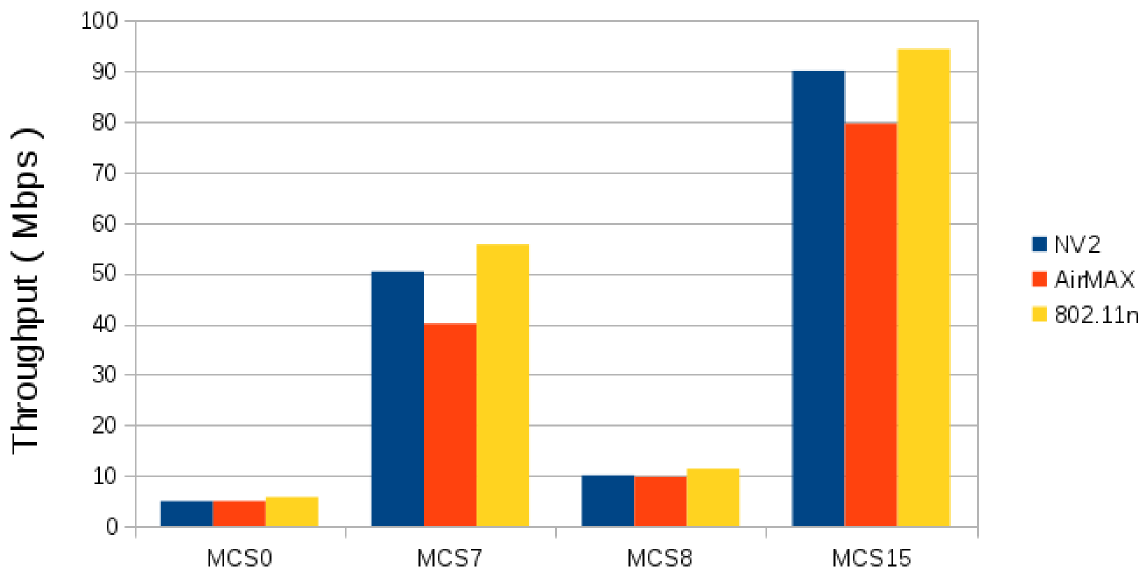

Figure 50: Throughput comparison between different technologies

3.2.3 One-way delay and jitter

As it has been shown, the delay is made of several components: transmission delay (the time to transmit the packet in the channel; it depends on the distance, on the packet size and on the bitrate), MAC delay (the time to start transmitting after the packet has been presented to the MAC for transmission, including unsuccessful retransmissions; in includes the transmission time; it mainly depends on the collision probability and the MAC parameters) and the queueing delay that depends on the average number of packets in the queue (and is limited to the queue size in saturation). The average MAC delay is the service time for the queue; hence the average queuing delay is that service time multiplied by the average number of packets in the queue.

The jitter is the standard deviation of the delay. Considering the stochastic nature of the contention process and the possibility of retransmissions that are transparent to higher layers, it is not obvious to obtain a probability density function for the delay, which in turn makes unaffordable to find an analytical way to obtain the jitter. In unsaturated conditions, with low collision probabilities, the main effect producing jitter is related to the variations in the number of packets in the queue, which in turn depends on the variability of both the service time and the inter-arrival time.

The use of the WiFi version and bitrate that offer the highest possible throughput at a given distance is also beneficial for the delay and jitter. The highest the bitrate is, the lowest the transmission time becomes. Also, for a given offered load, higher bitrates may operate under saturation with higher probability. Hence, there is no advantage at considering any other WiFi version but 802.11n.

The trade-off in 802.11n in terms of delay regards to the frame aggregation.

Figure 51 show the evolution of the near-saturation delay for a experimental test using 20MHz channel bandwith, 800ns of GI, 1426 bytes packet size and sending a bidirectional UDP flow not far below the saturation point.

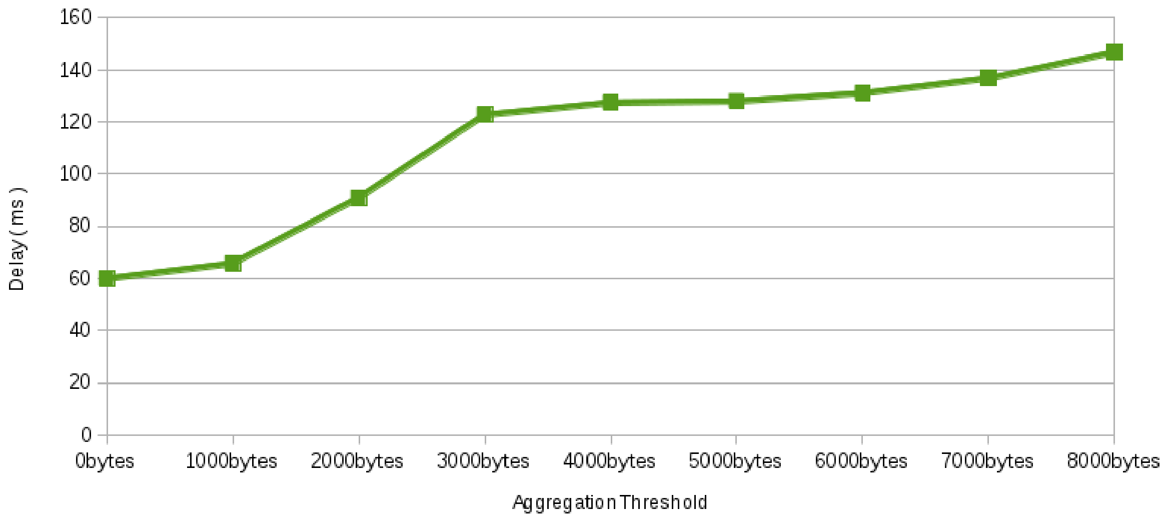

Figure 51: Evolution of the near-saturation delay for 802.11n regarding the frame aggregation

Same as before, Figure 52 and 53 compare the performance of 802.11n with and NV2 and AirMAX obtained in the indoor test-beds. The results are from test using 20MHz channel bandwidth and GI=800ns. As expected, lower jitter is obtained for NV2 and AirMAX in all the cases due its TDMA based technology. But better delay is obtained for non-standard technologies. Due to the unknown theoretical model in which both NV2 and AirMAX are based, it is not possible to know why the delay is lower than in 802.11n. However these results are useful to have an approximate view of the expected results from each technology. In particular, NV2 shows delays lower than 10 ms for realistic links under saturation conditions, which is very interesting for backhaul links.

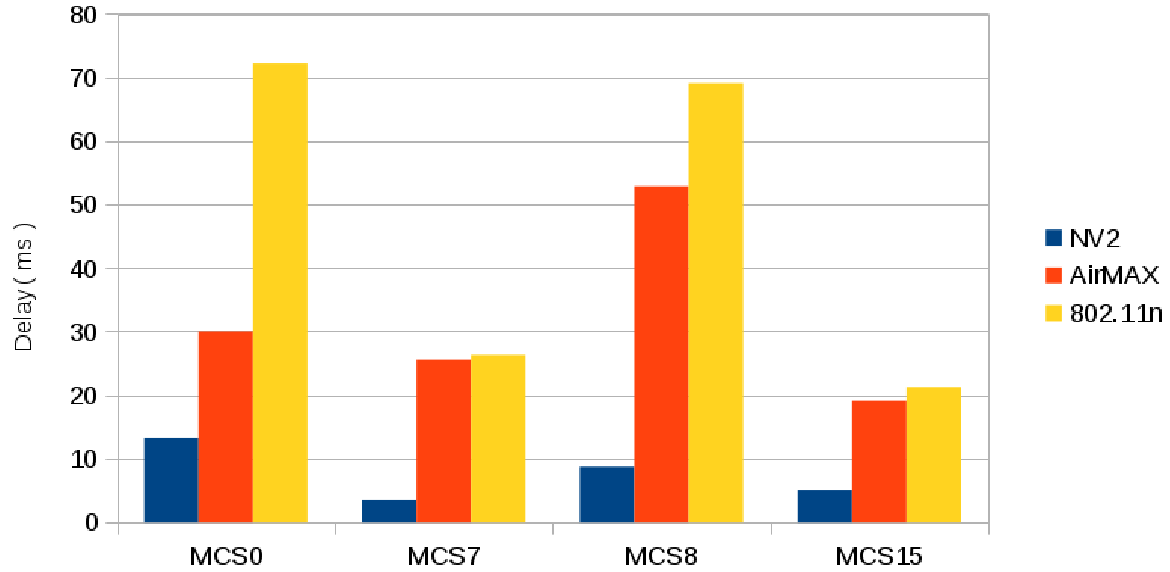

Figure 52: Delay comparison between different technologies

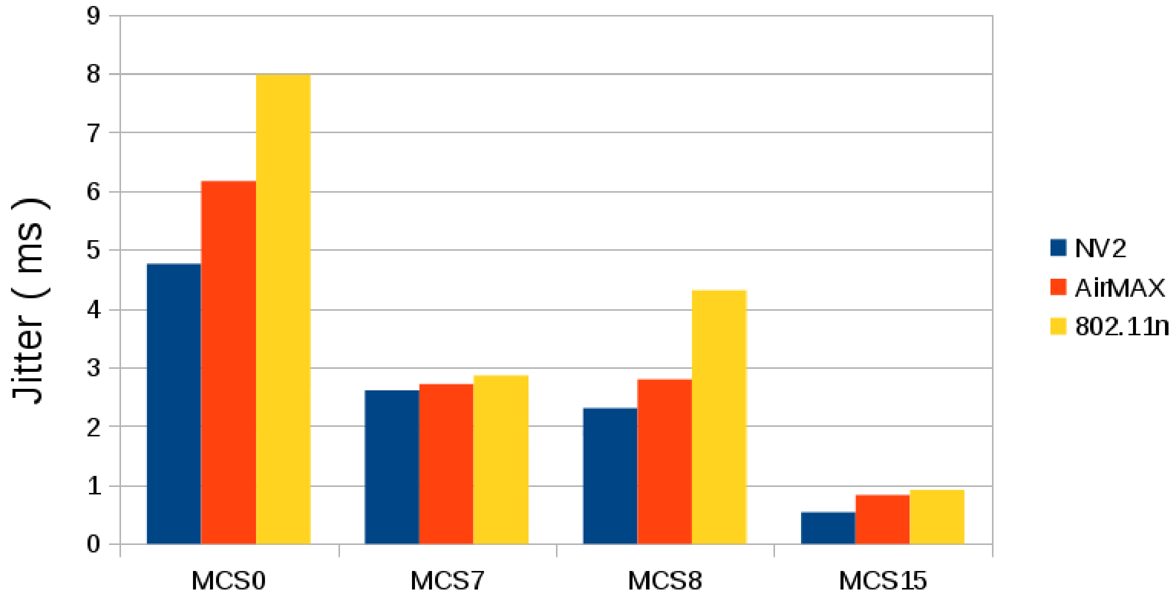

Figure 53: Jitter comparison between different technologies

3.2.4 Packet-loss probability

The results in the Section 3.1 have been obtained considering an ideal channel in which packet losses only occur by collision when two or more stations transmit simultaneously. However, in real conditions, the fading of the wireless channel causes packet losses because these packets are received with errors at the destination and thus the packets must be retransmitted. Thus, we must take into account the Packet Error Rate (PER) caused by impairments in the wireless channel. Furthermore, in non-saturation conditions, the packet loss depends primarily on the wireless channel condition because collisions are reduced to virtually zero, whereas the wireless channel fading may continue to produce errors and, therefore, packet losses and retransmissions.

There are several references that analyze the performance of IEEE 802.11 considering channel errors and PER [Samhat2006, Hyewon2007, Papapanagiotou2007, and Senthilkumar2010]. For a detailed and precise analysis, the PER should be calculated based on the BER performance for the bits of the packet header and payload. However, we can carry out a more simple and practical analysis considering that in the 802.11-2012 standard are marked sensitivity levels (minimum received power levels) for the different modulations providing a PER of 10% or less. Thus, when channel conditions worsen in real links reaching more than this 10% PER, the transmitter switches to a more robust modulation, thus reducing the PER at the cost of lower bit rate. In this way, we consider a 10% PER for calculating the maximum performance degradation in non-ideal channels.

To introduce the PER in the analytical model, we take into account that a transmitting station in 802.11 is unable to distinguish whether a transmitted packet has not reached its destination due to collision or the packet has arrived but with errors. In both cases, the transmitting station takes the same actions to retransmit the packet. Thus, we redefine p in the analytical model proposed in [Bianchi2010] as the probability that a packet should be retransmitted either by collision or errors in the channel, and the formula (3.2) becomes:

$$p = \left(1 - \left(1 - \tau\right)^{n-1}\right)\left(1 - PER\right) + PER \tag{3.12}$$

With this new definition of p, we can recalculate the values of p and τ previously presented in Table 20, and so we can calculate the saturation throughput performance, saturation average delay and packet drop probability in non-ideal channels with nonzero PER. For example, we obtain the values for PER = 10% in Table 60.

N	W=16		W=32		W=8		W=4	
	p	tau	p	tau	p	tau	p	tau
2	0,1861	0,0957	0,1464	0,0516	0,2486	0,1651	0,3307	0,25632
3	0,2451	0,0841	0,1846	0,0482	0,3244	0,1336	0,4131	0,19247
4	0,2881	0,0752	0,2165	0,0451	0,3732	0,1136	0,4635	0,15841
5	0,3214	0,0682	0,2434	0,0425	0,4089	0,0998	0,5002	0,13676
6	0,3483	0,0625	0,2664	0,0401	0,4371	0,0896	0,5293	0,12158
7	0,3707	0,0579	0,2865	0,0380	0,4604	0,0817	0,5535	0,11024
8	0,3900	0,0540	0,3042	0,0361	0,4803	0,0754	0,5742	0,10140
9	0,4068	0,0508	0,3199	0,0344	0,4977	0,0703	0,5924	0,09428
10	0,4218	0,0480	0,3341	0,0329	0,5132	0,0660	0,6087	0,08840
11	0,4353	0,0455	0,3469	0,0316	0,5271	0,0623	0,6235	0,08345
12	0,4476	0,0434	0,3587	0,0303	0,5399	0,0592	0,6370	0,07922
13	0,4588	0,0415	0,3695	0,0292	0,5516	0,0564	0,6494	0,07555
14	0,4692	0,0398	0,3796	0,0282	0,5624	0,0540	0,6609	0,07234
15	0,4789	0,0383	0,3889	0,0273	0,5726	0,0518	0,6717	0,06950

Table 60: Values of transmission (τ) and conditional collision or packet error (p) for different CW values and up to 15 stations.

Logically, as the PER increases, the system performance decreases. However, this decrease in performance is lower than we might imagine initially because a packet loss involves an artificial (not caused by a collision) increase of the CW, so that the probability of collision decreases in the following transmissions.

As an example, the following Table 61 presents the saturation throughput in 802.11 in basic mode, with different parameter settings and with different values of PER.

PER	802.11b	802.11a	802.11g	802.11n
0%	5,5714	4,2493	20,2979	92,743
0,1%	5,5707 (-0,01%)	4,2472 (-0,05%)	20,2916 (-0,03%)	92,7399 (-0,00%)
1%	5,5643 (-0,13%)	4,2285 (-0,49%)	20,2344 (-0,31%)	92,7117 (-0,03%)
5%	5,5316 (-0,71%)	4,1384 (-2,61%)	19,9557 (-1,69%)	92,5732 (-0,18%)
10%	5,4784 (-1,67%)	4,0084 (-5,67%)	19,542 (-3,72%)	92,3624 (-0,41%)

Table 61: Saturation throughput in basic mode with different values of PER.

As seen in Table 61, the performance decreases as the PER increases but, even for the maximum value of PER = 10%, the loss of performance is not overly significant.

3.2.5 QoS support and traffic differentiation

Traffic differentiation is possible at the MAC layer by using EDCA, and may be complemented at the IP layer. The most coherent way to perform QoS support at the IP layer when EDCA is underneath is DiffServ. The question to solve here is how important is the use of EDCA when DiffServ arquitecture is in operation.

Several tests were performed, injecting traffic to the network formed by an AP and two STAs in which the saturation point was measured at 12500 kbps. 1125 kbps of voice traffic were injected in each direction to/from each station, and 2000 kbps were injected in each sense for each station in the other traffic classes. Hence, we are ensuring that the WiFi network is saturated. During the experiments, the traffic differentiation was analyzed by measuring the bitrate, delay and jitter per class, per station and per sense (uplink -UL- and downlink -DL-).

The following different cases are tested:

- No QoS mechanism.
- With EDCA (MAC parameters adjusted with the values recommended in the standard).
- With DiffServ-like IP traffic management.
- With both EDCA+DiffServ.

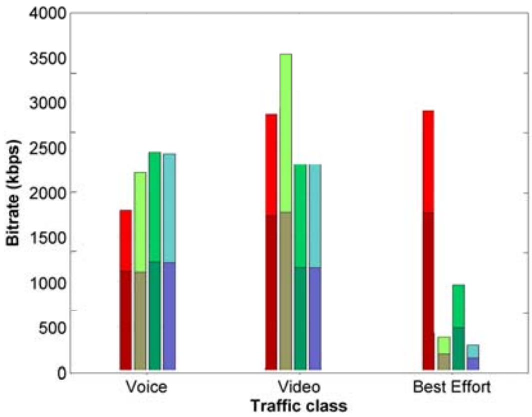

Figure 54: Throughput measured in the uplink per traffic class.

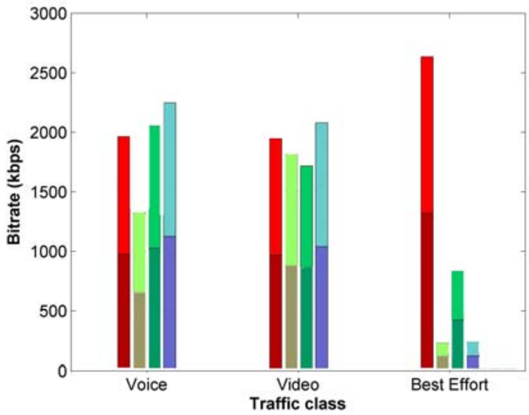

Figure 55: Throughput measured in the downlink per traffic class.

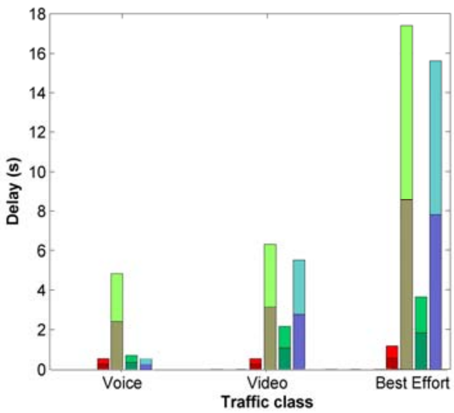

Figure 56: Delay measured in the downlink per traffic class.

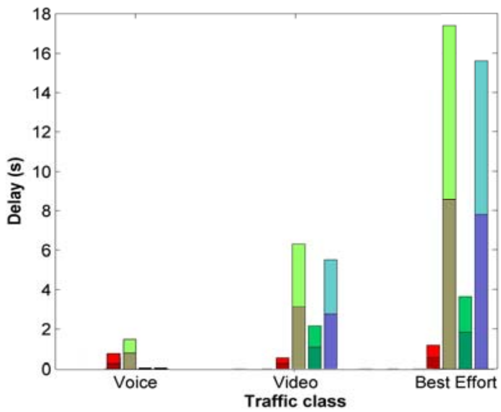

Figure 57: Delay measured in the uplink per traffic class.

The results show that EDCA itself does not obtain the best result. Even voice traffic, with the highest priority, cannot transmit successfully all the traffic, and the transmitted traffic is asymmetric. The delay results under saturation conditions are bad. When DiffServ is used, even though the IP flows are shaped to saturation, just by renforcing the priorities and avoiding offered loads beyond the saturation point permits to transmit almost all the voice traffic and much lower delays are obtained. However, the best result is found when combining both mechanisms. In these results the traffic shaping was adjusted to the saturation point, but ensuring that the system works in unsaturated conditions (through access control) will permit to transmit optimally the priority traffic with low delays.

Jitter results were not signficant and are not shown.

3.2.6 Costs: CAPEX and OPEX

The main elements to consider for the CAPEX are:

- Powering system (technology dependant)
- Supporting infrastructures

- Lightening protection
- Deployment and testing costs

WiFi is proposed as a transport communications technology for backhaul because it is supposed to be much less expensive than any other alternative, which would contribute to achieve the objectives of this project. Based on the recent experience in rural deployments done by the Rural Telecommunications Group (GTR) in the Pontificia Universidad Católica del Perú, some figures are given for a first approximation to the CAPEX.

On one hand, systems are low-cost. Taking an average professional solution like Mikrotik RouterBoard RB493G systems, a wireless router with high power 11n radios and external directive antennas may cost approximately $880 USD. This is far less expensive that other professional backhaul alternatives, but the other costs must also be considered.

Powering systems for rural telecommunications systems are often autonomous photovoltaic systems, whose cost is basically proportional to the power consumption. A WiFi Router has very low power consumption. Figure 58 shows the result of monitoring a WiFi router during one hour, firstly under heavy load conditions and later almost idle. We can appreciate that the average power consumption is not higher than 5.5 W. We can take 6W as a good approximation. The cost of the powering system for such consumption depends on the geographical situation, but may be around $1800 USD.

Lightning protections also depend on the height of the tower and the quality of the land for the grounding. Unattended installations in rural areas require durable grounding installations, $2100 USD should be foreseen as a minimum for this concept.

The supporting infrastructure is usually a tower, whose cost depends on the height, the geographical situation (and distance from the manufacturer). An average tower may cost around $15000 USD. This may be taken as an average value, real costs must be much cheaper (as low as $5000 USD) or more expensive (as high as $26000 USD) depending on the tower height and distance to the city. Transport to a remote location and installation may increase the cost in about $10000 USD (up to more than $20000 in a very isolated area).

Finally, the installation and testing costs may be estimated in $5000 in average.

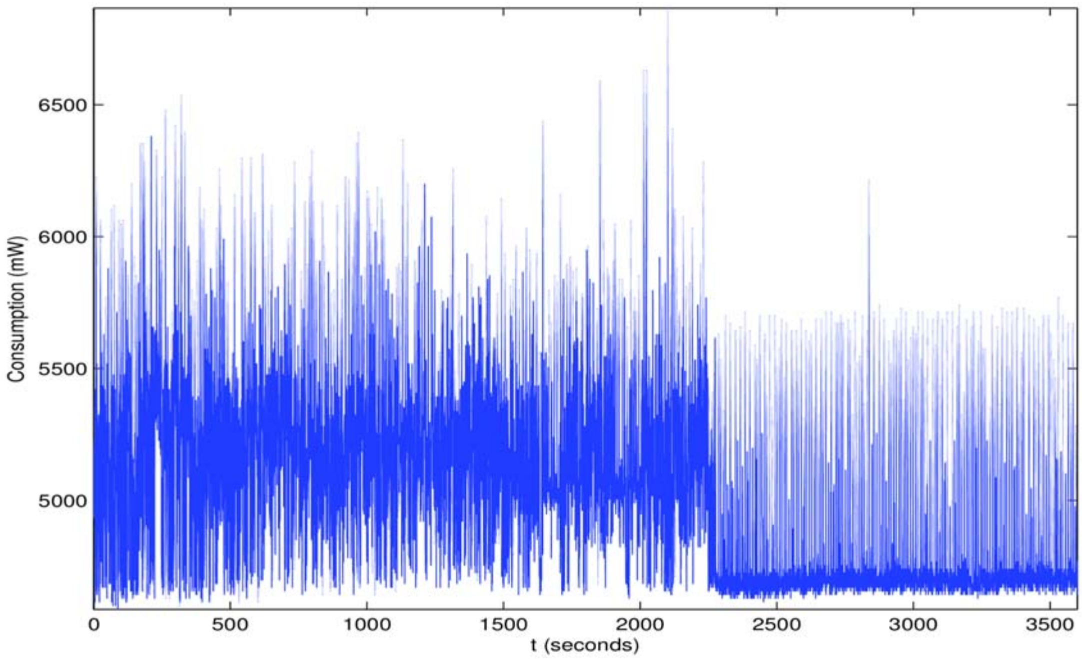

Figure 58: Measurements of power consumption in a wireless router with two high power 802.11n interfaces during 1 hour. A big file has been transferred through the router during the first part of the hour (intensive operation) and then the system has remained under-loaded for the rest of the time.

Hence, a first estimation for the capital expenditure per location gives:

$$CAPEX = (C + 300 \cdot S + O) \cdot (1 + n) \tag{3.13}$$

where n is the number of hops, S is the power consumption by one system, C is the cost of one telecommunication system, and O in the aggregated cost of tower, lightening protection and installation.

In the case of WiFi systems, with only one link, this results in

CAPEX = ($880 + $1800 + $32100) · 2 = $69560 USD

We may see that, in the case of WiFi routers, the cost of the communication system contributes to the total CAPEX in less than 3% of the total cost. It is also true that the importance of the technology may be higher if the alternatives have a high power consumption.

Let us now examine the OPEX. The main components of the OPEX are:

- Frequency-band licenses (technology dependant)
- Management (depends on the availability of a good network management system).
- Proactive maintenance
- Communications equipments (technology specific)
- Reactive maintenance (depends on the quality of the technology, the quality of the installation, the availability of a good network management system and the geographical context).
- Spare parts (depends on the quality of the technology and the cost of the technology).

The use of non-license bands permits to reduce the operation, but the other costs are present. It is difficult to estimate these costs, but based on the experience we will consider that we need about 5% of the CAPEX for spare parts, management and proactive and reactive maintenance per year. That makes around

$$OPEX \approx CAPEX \cdot 0{,}05\,[USD/Year] \tag{3.14}$$

These first approximations are just an orientation of the concepts that must be considered either for the CAPEX or for the OPEX, and for the sake of comparison. Values are just tentative. These costs are not different for the different versions of WiFi.

The example case proposed in the introductory section was a zone with one or more small villages being served with 5 Mbps (DL) + 1 Mbps (UL). The proposed costs will be enough for this example if only one link is needed.

3.2.7 Other considerations

Management has been mentioned in the previous subsection, but we must insist here about it. The final quality and stability of a rural network is very much related to the existance of a very consistent network management system (NMS) and good remote administration interfaces. This is sometimes the main difference between commercial products of very different costs. This comment is just for underlining the importance of choosing systems that have a powerful remote administration and that can be integrated in a NMS.

4 EXPECTED PERFORMANCE IN 802.16 LONG-DISTANCE LINKS

4.1 Theoretical expectations and simulation results from from IEEE 802.16-2009

4.1.1 Performance parameters that may be obtained

WiMAX support assured QoS services, since uses TDMA/OFDMA techniques. For static WiMAX, the common mechanism implemented is TDMA. Unlike the 802.11 medium access control, the simple nature of the time-division multiplexing used by WiMAX makes easy quantify its performance parameters, due to the medium access is assured by the base station.

Although the WiMAX performance is strongly-connected to the QoS-IP, only link-level performance will be analyze in this document. Analytical models or experimental results obtained from theoretical models and real testbeds permit to estimate the average, maximum or minimum values for throughput, delay, jitter and packet-loss under certain conditions. This document will be focused in the throughput and upper bound delay analysis for long distances.

Once these parameters are measured and quantified, it is possible map them to the WiMAX QoS classes (UGS, rtPS, nrtPS, BE, ERT-VR) which later will be translated to QoS-IP mechanisms such as DiffServ (DSCP) or IntServ (Traffic Specifications)

Hence, in the following we will try to obtain an approach to these performance parameters that can be obtained for each frame size and modulation/code rate of the different versions of WiMAX using analytical models and simulation, since it is well known that real WiMAX performance is well featured by these models. The models used are simply calculations made from the standard. In order to simulate WiMAX links, NS-3 network simulator is used, which has been previously exposed and validated in Chapter 3.1.

Finally, a comparison between performance parameters obtained and real testbed literature will be done.

4.1.2 General to 802.16

As said in Chapter 2.2.2, the only restriction for WiMAX long distance links is the link budget. The SNR perceived at the receiver, will determine the modulation and codification and hence, the link physical features. Also can be the medium access features like frame duration will affect performance parameters like throughput and delay.

Next these parameters will be analyzed for both WiMAX certification profiles: Wireless HUMAN and Wireless MAN.

4.1.3 WirelessHUMAN (static WiMAX on non-licensed bands)

WirelessHUMAN is a static WiMAX implementation based in the OFDM PHY system profile for LOS/NLOS links in non-licensed bands.

This implementation has the advantage of work in non-licensed bands, lowering the deployment expenses, but has the disadvantage that is subject to various rules and constraints, such as power limits. Also, using non-licensed bands imply have non-desired interferences from other technologies which use ISM bands.

These restrictions affect directly to WiMAX budget link, constraining the SNR perceived at the receiver and hence the link distance. This document will be focused in the 4.9 – 5.9 GHz band due the mayor of the WiMAX manufacturers produce equipment for WirelessHUMAN in this frequency band.

The FCC is responsible for the radio-electric regulations in USA. Many countries from Latin America, Africa and Asia have national regulations inspired in the FCC's.

The reason for showing results for all the bitrates in the precedent tables and figures is that any of them might be necessary depending on the link budget. The 5 GHz frequency band that can be used by

WirelessHUMAN have transmission power restrictions that depend on the regional regulations. The FCC (Federal Communications Commission) is responsible for the radio-electric regulations in the USA, but many other countries, particularly in Latin America, have national regulations inspired in the FCC's. In the 5 GHz band, FCC permits up to 30dBm of transmitting power and antenna gains up to 27dBi for directional links. Therefore, this will be the limiting factor in terms of distance.

4.1.3.1 Link budget

The next analytic model will calculate the throughput achieved in typical WiMAX long distance links using the 5GHz band. In order to calculate the throughput, the distance for each modulation and codification technique is calculated. This distance is obtained calculating the link budget parameters such as transmitted power sensitivity, antenna gain, cable attenuation, propagation loss, etc. The propagation model considered for give a general overview, is the free-space path loss, since it be can roughly approximate to the real conditions in a long distance link working in the millimetric band. However, for each specific environment the propagation model has to be chosen carefully. The distances achieved for each modulation shown in Table 64 have been calculated using typical parameters detailed in Table 62, obtained from real equipment and real WiMAX links. The standard IEEE 802.16-2004 defines for each modulation, the minimum sensitivity and SNR required. This relationship between modulation and SNR is shown in Table 63.

Transmit Power	24 dBm
Directional Antenna Gain	27 dB
Cable and connectors attenuation	2 dB
Sensitivity (Max. Modulation)	-71 dBm
Margin	20 dB

Table 62: Typical parameters of real WiMAX links.

	Minimum Sensitivity	Minimun SNR
BPSK 1/2	-89 dBm	6.4 dB
QPSK 1/2	-83.2 dBm	9.2 dB
QPSK 3/4	-83.2 dBm	11.4 dB
16-QAM 1/2	-76.2 dBm	16.4 dB
16-QAM 3/4	-76.2 dBm	18.2 dB
64-QAM 2/3	-70 dBm	22.4 dB
64-QAM 3/4	-70 dBm	24.4 dB

Table 63: Physical requirements defined for the standard for each modulation.

Distance Km	1	5	10	15	20	25	30	35	40	45	50	55	60	65	70	75	80	85	90	95	100
64 QAM 3/4																					
64 QAM 2/3																					
16 QAM 3/4																					
16 QAM 1/2																					
QPSK 3/4																					
QPSK 1/2																					
BPSK 1/2																					

Table 64: Distance achieved for each modulation. Red cells denote that the received power is below of the required sensitivity for the modulation. Yellow cells denote that the the SNR is in the 10 dB fading margin. Green cells denote that SNR is above the 10 dB fading margin.

As expected, the most powerful modulations can only work in short distances, while the most robust can set up long distance links up to 100km. These results are obtained taking last generation Albentia Systems WiMAX equipments as the reference. Different equipment can present better or worse results, but this can serve as a good approximation. Although this table can change for each environment and equipment, it is useful to give a broad idea of the distances achieved for each modulation.

4.1.3.2 Performance Analysis

Using Table 64 it is possible almost immediately to calculate the expected throughput for each distance. Following the theory explained in Chapter 2.2.1, the throughput offered by a WiMAX link is mainly dependent on its modulation and coding scheme. However other parameters can influence in higher or lower throughput, such as the cyclic prefix and the frame duration.

In one hand, cyclic prefix protect the wireless communication from the multipath interference, but at the expense of use redundant symbols. The duration of this cyclic prefix can be reduced in environments with low multipath interference in order to increase the throughput, but for stable communications in environments with high multipath interference, the cyclic prefix must be increased. The values of the cyclic prefix are discrete and for fixed WiMAX are: 1/4, 1/8, 1/16, 1/32.

In the other hand, the frame duration will define the overload introduced by the link level. Big frames will introduce low protocol overhead in the wireless link, and therefore the throughput offered for the upper levels will be optimized at expense of increase the average packet delay. On the contrary, small frames will introduce the lowest delay, but the overhead will be increased. For fixed WiMAX, there are established six specific frames duration values: 2.5 ms, 4 ms, 5 ms, 8 ms, 10 ms, 12.5 ms and 20 ms.

In Figure 59 and Figure 60 the throughput (uplink + downlink) at MAC level is shown for different WiMAX parameters. In Figure 59 a cyclic prefix of 1/4 of the OFDM symbol time and frame duration of 2.5 ms is used in order to model a wireless link with high multipath and low delay requirements. In Figure 60 a cyclic prefix of 1/32 of the OFDM symbol time and a frame duration of 20 ms is used in order to get the maximum throughput achievable. For both Figures 59 and 60, the throughput achieved is shown reaching 100km distance because, unlike 802.11, there is not any hard restriction for the maximum distance achievable.

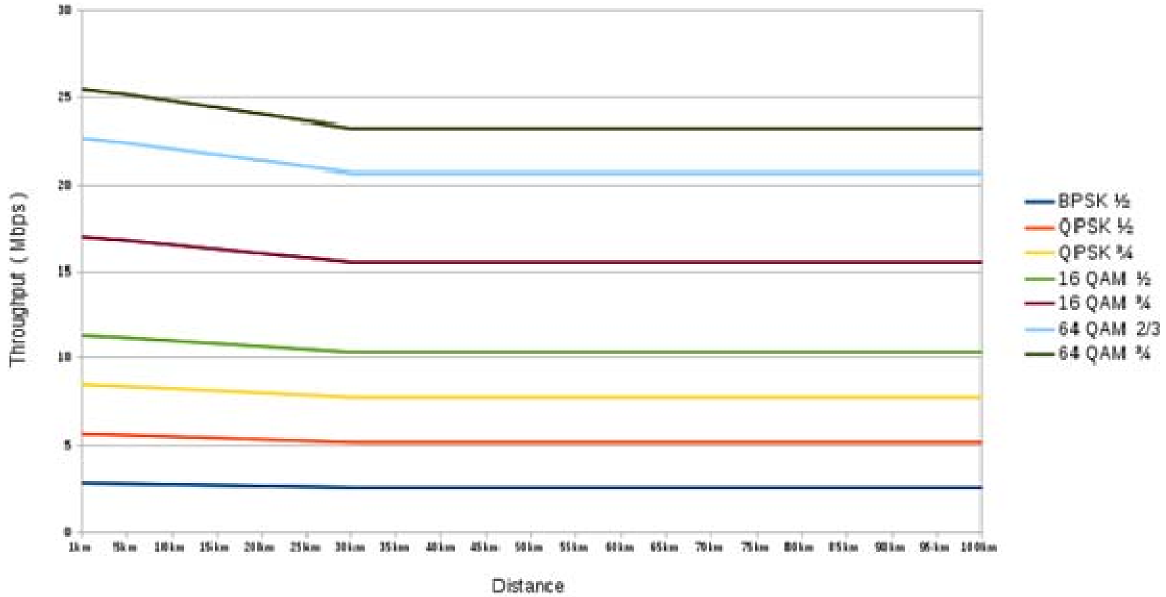

Figure 59: Expected throughput for each modulation using cyclic prefix 1/4 and frame duration 2.5 ms

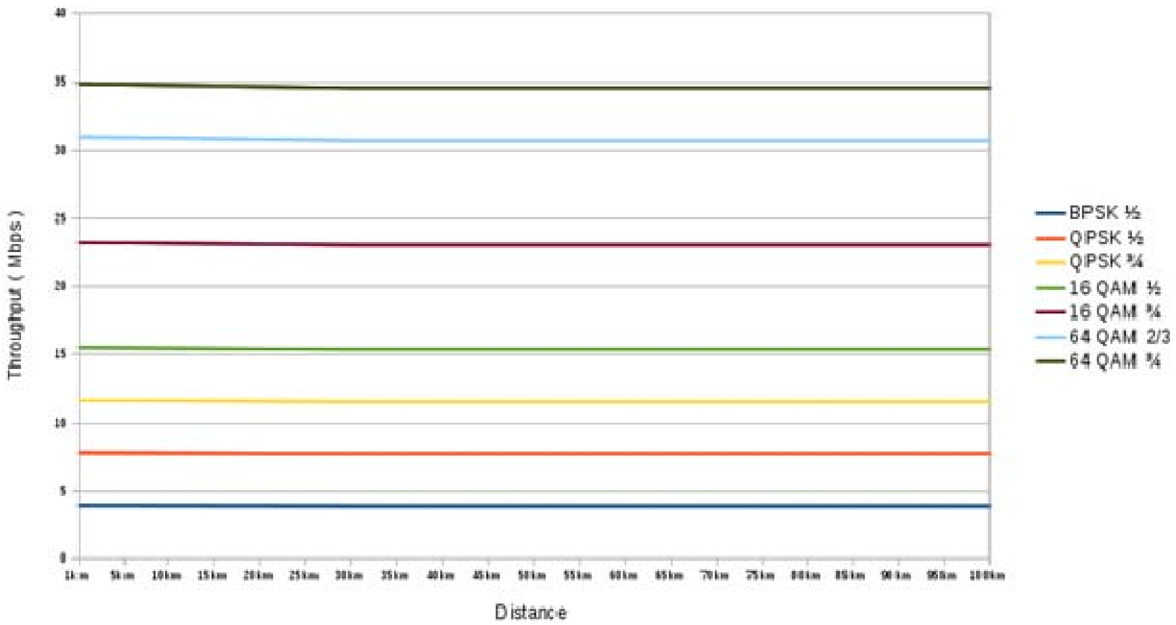

Figure 60: Expected throughput for each modulation using cyclic prefix 1/32 and frame duration 20 ms

	1km	5km	10km	15km	20km	25km	30km	35km	40km	45km	50km	55km	60km	65km	70km	75km	80km	85km	90km	95km	100km
BPSK ½	2.83	2.8	2.75	2.71	2.67	2.62	2.58	2.58	2.58	2.58	2.58	2.58	2.58	2.58	2.58	2.58	2.58	2.58	2.58	2.58	2.58
QPSK ½	5.66	5.6	5.51	5.42	5.34	5.25	5.16	5.16	5.16	5.16	5.16	5.16	5.16	5.16	5.16	5.16	5.16	5.16	5.16	5.16	5.16
QPSK ¾	8.5	8.4	8.27	8.14	8.01	7.88	7.75	7.75	7.75	7.75	7.75	7.75	7.75	7.75	7.75	7.75	7.75	7.75			
16 QAM ½	11.3	11.2	11	10.8	10.6	10.5															
16 QAM 3/4	17	16.8	16.5	16.2	16																
64 QAM 2/3	22.6	22.4	22																		
64 QAM 3/4	25.5	25.2																			

Table 65: Values of WiMAX throughput (UL+DL) for CP 1/4 and frame duration 2.5 ms

Distance	1km	5km	10km	15km	20km	25km	30km	35km	40km	45km	50km	55km	60km	65km	70km	75km	80km	85km	90km	95km	100km
BPSK ½	3.87	3.87	3.86	3.86	3.85	3.84	3.84	3.84	3.84	3.84	3.84	3.84	3.84	3.84	3.84	3.84	3.84	3.84	3.84	3.84	3.84
QPSK ½	7.75	7.74	7.73	7.72	7.71	7.69	7.68	7.68	7.68	7.68	7.68	7.68	7.68	7.68	7.68	7.68	7.68	7.68	7.68	7.68	7.68
QPSK ¾	11.6	11.6	11.5	11.5	11.5	11.5	11.5	11.5	11.5	11.5	11.5	11.5	11.5	11.5	11.5	11.5	11.5	11.5			
16 QAM ½	15.5	15.4	15.4	15.4	15.4	15.3															
16 QAM ¾	23.2	23.2	23.1	23.1	23.1																
64 QAM 2/3	31	30.9	30.9																		
64 QAM ¾	34.8	34.8																			

Table 66: Values of WiMAX throughput (UL+DL) for CP 1/32 and frame duration 20 ms.

It is possible to observe a small slope in the beginning of the graphs. Both Figures have the same effect, but in Figure 59 the effect is more notorious than in Figure 60. This is effect is produced by the RTG/TTG field in the WiMAX header. The duration of this field depends on the propagation time and hence the distance. This duration has a maximum value of 100 ms, for this reason the slope finishes at 30 km. The further is the SS, more OFDM symbols are used for the RTG/TTG field. For small frames, like 2.5 ms, the weights of these OFDM symbols are bigger than for long frames such as 20 ms. Values in Tables 65 and 66 correspond to Figures 59 and Figure 60.

Depending of the environment and the equipment used, the distances achieved for each modulation can variate slightly. However the throughput achieved for each modulation are accurate enough to the real WiMAX links.

	1km	5km	10km	15km	20km	25km	30km	35km	40km	45km	50km	55km	60km	65km	70km	75km	80km	85km	90km	95km	100km
BPSK ½	2.83	2.8	2.75	2.71	2.67	2.62	2.58	2.58	2.58	2.58	2.58	2.58	2.58	2.58	2.58	2.58	2.58	2.58	2.58	2.58	2.58
QPSK ½	5.66	5.6	5.51	5.42	5.34	5.25	5.16	5.16	5.16	5.16	5.16	5.16	5.16	5.16	5.16	5.16	5.16	5.16	5.16	5.16	5.16
QPSK ¾	8.5	8.4	8.27	8.14	8.01	7.88	7.75	7.75	7.75	7.75	7.75	7.75	7.75	7.75	7.75	7.75	7.75	7.75			
16 QAM ½	11.3	11.2	11	10.8	10.6	10.5															
16 QAM 3/4	17	16.8	16.5	16.2	16																
64 QAM 2/3	22.6	22.4	22																		
64 QAM 3/4	25.5	25.2																			

Table 67: Values of WiMAX throughput (UL+DL) for CP 1/4 and frame duration 2.5 ms.

Regarding the delay, other models have to be used due to its stochastic nature. The delay will be defined as the average time between the sending of a packet from a SS and and its arrival to the BS. The delay analysis requires know the transmission rate of the specific flow, therefore, this analysis will be applied after the resource allocation in the frame for the flow. For this reason, the analysis has to be divided in two steps: delay for packets which are transmitted in only one frame and for packets which are transmitted in several frames. In order to simplify the process it is supposed that the fragmented packets are similar to independent packets which are transmitted in only one frame. Therefore, the total delay of a big size frame is the sum of the delays from each fragment. After this supposition, the delay analysis for packets transmitted in only one frame will be present next.

In TDMA systems, the delay is analyzed by queue models where the packet arrivals can be considered as a independent processes and modeled by the Poisson distribution. But WiMAX must be analyzed carefully bearing in mind some features such as point-to-multipoint, resource allocation unicast polling based, and a specific uplink scheduling, such as round robin. The absolute delay can be discomposed in 5 terms specified in the next equation:

$$E\left[W\right]=E\left[W^r\right]+\alpha+E\left[W^s\right]+E\left[W^t\right]+\mu \qquad (4.1)$$

where:

- W^r is the time in the buffer in which packets wait for send a BW-Req and $E\left[W^r\right]$ is the expected reservation delay.
- α is the BW-Req transmission time, which can be approximate to a constant value, due to is always transmitted using BPSK ½.
- W^s is the scheduling delay, and it is defined as the time transcurred from the the BW-Req is sent until the bandwidth is granted. $E\left[W^s\right]$ is the expected value of this time.
- W^t is the waiting time in the SS buffer before a packet is sent, so $E\left[W^t\right]$ is the expected value of the transmission delay.
- μ is the packet transmission time and is equal to the packet length divided in the channel transmission rate.

Starting from this point, some models and analysis can be done depending of the imposed suppositions.

For point to point links, a accepted analysis is made in [Saffer2008], where the allocation delay is modeled by a M/D/1 queue with vacation, in with both the service time and the vacation time are deterministic and equal to $C=T_f/P$, where P is the polling parameter and T_f is the frame duration. The scheduling time is also modeled by a M/D/1 queue and the transmission delay is approximated by a wort-case delay which is the delay of a grant arriving into the empty BS grants buffer. This way, the system delay is bounded from above by M/D/1 waiting time and replacing factors, the total expected delay is modeled by the next equation:

$$E\left[W\right]\leq\frac{C}{2\cdot\left(1-\frac{\lambda\cdot T_f}{P}\right)}+\frac{P+1}{2}\cdot\alpha+T_f+\frac{\lambda\cdot T_f}{2\cdot\left(1-\lambda\cdot T_f\right)}+\mu \qquad (4.2)$$

Where λ is the expected packet arrival frequency modeled by a Poisson process. The model proposed by [Saffer2008] assumes infinite buffer lengths, error-free channel and negligible propagation delay. In [Prieto2010] this model is extended and completed with simulation and real test bed results, for different types of flows such as BE, UGS and nrTPS. A interesting conlusion in [Prieto2010] is that although the previously explained model proposed by [Saffer2008] depends largely of the specification imposed by equipment manufacturers, the total delay can be roughly approximated to 2 times T_f.

In order to check these studies about latency, a set of simulation test has been designed to compare the results. A simple WiMAX network is simulated in which a UDP flow mapped in the UGS class is sent in a non-saturating rate. The scheduler is a simple round robin implementation and the packet size is fixed to 1300 bytes at link level. Different modulations and different frame durations are used to analyze the delay. Due to the TDMA nature of the link, the distance does not influence in the results.

Figure 61 shows the results of the delay with different modulation and different frame durations for a simulation of a WiMAX link with cyclic prefix equal to 1/4. In Figure 62 the same simulation is shown, but with cyclic prefix equal to 1/32.

In both figures, it is easy to perceive that the bigger frame duration the bigger packet delay and that the smaller frame duration, the smaller packer delay. Also it is possible realize that obtained values of the delay correspond approximately to the expected values predicted in [Prieto2010], that is to say the values of the delay are 2 times the frame duration.

As expected, change the modulation scheme does not vary significantly the delay like happened in 802.11 technologies. This is due to the TDMA nature and its semi-static resource distribution.

Also, both figures show enough similarity for announce that the cyclic prefix duration has not a significant effect in the delay.

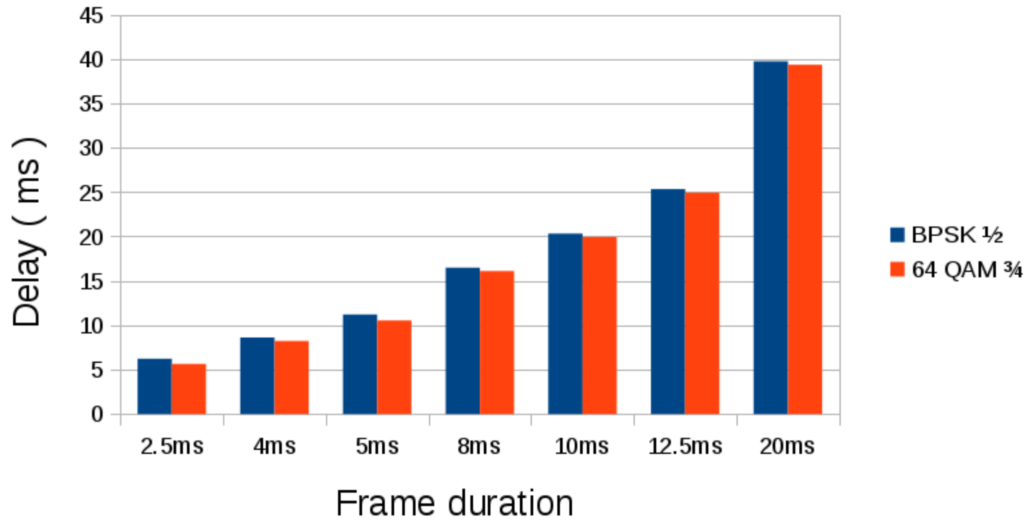

Figure 61: Different delays obtained for different frame durations using a 1/4 cyclic prefix.

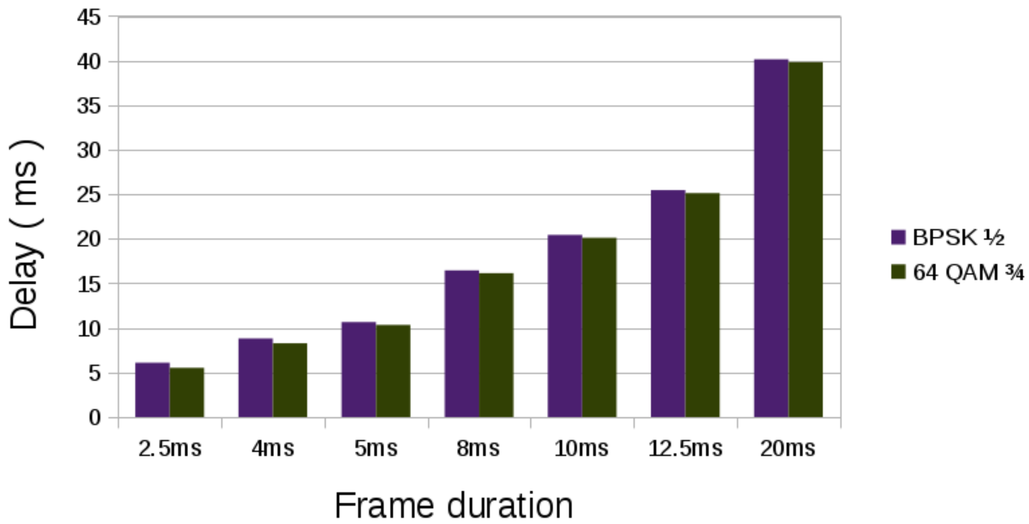

Figure 62: Different delays obtained for different frame durations using a 1/32 cyclic prefix.

Regarding the packet loss [Prieto2010] also gives a model which relates the packet loss with the frame duration. Figure 63 show the error according the theoretical model regarding the frame duration. It is, expressed as hundred per cent, due to the absolute error can be higher with longer frame duration than with smaller frame duration. According this model, the maximum error is associated to small frame duration and decrease for longer frame durations. This is due to for longer frame duration, the number of frames transmitted is lower, and therefore, the number of headers, but is not clear why the headers introduce higher error rate than data.

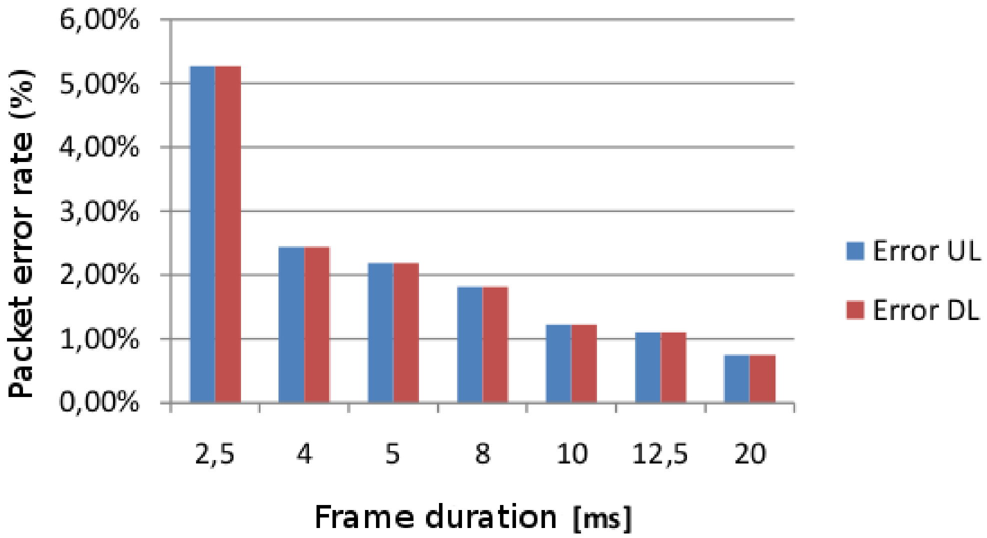

Figure 63: Packet error rate as a function of the frame duration according [Prieto2010].

Finally it is possible conclude that for static WiMAX links which work with a proper link budget, performance parameters such as throughput and delay can be obtained approximately with theoretical models and simulations with a good enough approximation to the real behaviour.

4.1.4 WirelessMAN (static WiMAX on licensed bands)

WirelessMAN is a static WiMAX implementation based in the OFDM PHY system profile for LOS/NLOS links in non-licensed bands. It is very similar to WirelessHUMAN but it is implemented on licensed bands.

This similarity makes no worth to do a detailed analysis of the performance since the results would be the same of WirelessHUMAN. Only aspects related with the link budget are different since on licensed bands, the frequency policies are different, allowing transmit more power and use directional antennas with bigger gains. For this reason, only the link budget and achievable distances will be discussed.

4.1.4.1 Link budget

To date, the only WirelessMAN implementation operate in the 3.5 GHz band over 3.5 MHz and 7 MHz channel and is based in IEEE 802.16-2004 OFDM physical layer, like the WirelessHUMAN implementation. In WirelessMAN, the FCC regulation is more permissive than in non-licensed bands and usually the power policies are not the main constraint in the link budget. In 3.5 GHz band, the more restrictive constraint is the equipment available, since is difficult find manufacturers which sell antennas with more than 30 dBi or equipment with 30 dBm power transmission. For this reason, next a typical case is studied, using the same typical transmission parameters shown previously in Table 62 with one exception: the transmission power is considered to be extended to up to 4W (36dBm), which actually permits to multiply times four the range of each modulation-coding schema. Also the same relationship between SNR and modulation scheme shown in Table 63 is used. Table 67 shows the new link budget for WiMAX in 3.5 GHz band.

Distance Km	1	5	10	15	20	25	30	35	40	45	50	55	60	65	70	75	80	85	90	95	100
64 QAM 3/4																					
64 QAM 2/3																					
64 QAM 3/4																					
16 QAM 1/2																					
QPSK 3/4																					
QPSK 1/2																					
BPSK 1/2																					

Table 67: Distance achieved for each modulation. Red cells denote that the received power is below of the required sensitivity for the modulation. Yellow cells denote that the the SNR is in the 10 dB fading margin. Green cells denote that SNR is above the 10 dB fading margin.

The main differences between Table 64 and 67 reside that in the lower attenuation of the 3.5 GHz band (approximately 5 dB less) and the extended range due to the higher transmission power. The first advantage is compensated by the lower gain antennas in the 3.5 GHz band, but the second advantage permits to achieve much longer distances with the same modulation scheme. The sensitivity for 7MHz channels might be lower than for 10 MHz channels in WirelessHUMAN, because of the reduction of the noise-equivalent bandwidth of receivers. However that has not been considered here as it is very product-dependent. For 3.5MHz channels the sensitivity could be up to 3dB better than for 7 MHz channels.

The rest of parameters are the same of WirelessHUMAN, so the previously WiMAX analysis can be applied also for WirelessMAN, but adapting the channel bandwidth to 3.5MHz or 7MHz. This is only an arithmetical operation that only depends of the number of subcarriers.

4.2 Discussion of results

4.2.1 Distances achieved, taking the regulatory environment into account

As said in Chapter 2.2.2, for IEEE 802.16-2004 there is no restriction for achieve a specific restriction. WiMAX hardware adapts their timers to the specific propagation delay, and the performance is practically the same regarding the distance.

In terms of power link budget, Tables 64 and 67 show what the limits are for the different modulations, and therefore the bitrates. The standard defines the minimum power requirements that a receiver must support for each modulation in terms of SNR and sensitivity. However, each manufacturer implements its own power requirements and auto-rate algorithm. For this reason, those results must be taken as a rough approximation. The most important factors that must be beared in mind are the the regulatory restrictions (FCC's regulations have been taken as the reference) and the equipment specifications, such as maximum transmission power and antenna gain. Furthermore, other environment specific factors like weather and topography can influence notably. As in previously link budget analysis, achievable distances have been calculated taking a minimum of 20dB of fading margin. But depending of the context, some times higher margins must be used and throughput must be sacrificed at expenses of stability.

4.2.2 Performance parameters and QoS under optimal conditions

In order to achieve the best performance for each environment it is necessary set the appropriate parameters in the physical and link layers.

Under optimum conditions and regarding the physical layer, the highest throughput will be achieved with the highest level of modulation and codification scheme. Furthermore, the wider the channel bandwidth the higher will be the number of subcarriers, and therefore, the symbol time will have less duration and the throughput will be increased. But the symbol time is also influenced by the cyclic prefix used in OFDM, so in order to achieve higher throughput, it is advisable reduce de cyclic prefix duration. However this will be the link more prone to suffer multipath interference, so a compromise between throughput and multipath interference protection must be defined.

The values of jitter and delay should no be influenced by the physical layer. As seen in Figures 59 and 60, both sets of values are very similar using different cyclic prefix duration and different modulation.

However, the packet loss will be influenced by both codification ratio and cyclic prefix. Lower codification ratios will achieve lower error rates, and the longer the cyclic prefix duration the lower the probability of suffer multipath interference.

Regarding the link layer, it is only relevant to discuss on the frame duration, as the rest of link layer parameters will be set and configured differently for each manufacturer. The frame duration will affect largely to the throughput and delay, and again, will be necessary reach a compromise between throughput and delay. For traffic with tight delay requirements lower delay will be achieved setting the lowest frame duration 2 ms, however the high WiMAX overload will decrease the total throughput. Longer frame duration such as 20 ms will foster higher throughput but will increase also the delay. As said in Chapter 4.1.3 regarding the delay, in optimal conditions the one-way delay can be approximated by 2 times the frame duration. The frame error rate, as said in this same chapter, grows when the frame duration decrease.

Bearing these concepts in mind, the traffic can be modeled by the scheduling services defined by WiMAX in order to support QoS traffic. However how this scheduling services work and how are implemented depends largely of the manufacturer. All aspects of QoS have to be managed together with some upper-level mechanism such as DiffSerf, IntServ or MPLS.

4.2.3 Costs: CAPEX and OPEX

The same elements considered for the CAPEX and OPEX in Section 3.2.6 for the case of WiFi are considered here.

For the CAPEX:

- Telecommunications systems may be either a point-to-point link or any base station with a single subscriber station. The total cost for the pair may be around $5700 USD.
- Powering system (technology dependant): modern stations and base stations designed for WirelessHUMAN (5 GHz non-licensed band) have a power consumption of up to 20W. Albentia Systems ARBA series have been taken as reference. The power consumption may double that figure for operation in licensed bands, as the transmitted power may be of several watts.
- Supporting infrastructures, lightening protection, deployment and testing costs: there is not any difference with the case of WiFi.

Hence, based on the same capital expenditure model used for WiFi,

$$\text{CAPEX} = (\$5700 + \$6000 + \$32100) \cdot 2 = \$87600 \text{ USD}$$

Although the main part of the CAPEX is common to any terrestrial broadband radio technology, the higher costs of equipements and the higher power consumption suppose an increment of 26% compared to WiFi.

In licensed bands this becomes

$$\text{CAPEX} = (\$5700 + \$12000 + \$32100) \cdot 2 = \$99600 \text{ USD}$$

Let us now examine the OPEX.

For non-licensed bands we may still consider $OPEX \approx CAPEX \cdot 0{,}05\,[USD/Year]$

For licensed bands, the OPEX will be more expensive due to the payment of licenses, but no figures can be given as this is extremely country-dependent.

4.2.4 Other considerations

As stated in Section 3.2.7 for 802.11, it is the important of choose systems that have a powerful remote administration and that can be integrated in a NMS.

5 EXPECTED PERFORMANCE IN SATELLITE COMMUNICATION SYSTEMS

5.1 Theoretical expectations

Theoretical performance for satellite links is usually and simply referenced as "*600-700 ms delay*", as the value of the RTT is the most different indicator comparing performance of a broadband service over satellite to other broadband technologies, like xDSL. This is true and enough if you plan to use a satellite link for basic broadband access and IP transporting, like web browsing, file transferring and emailing. However, when the satellite link is needed for VoIP transporting, cellular backhauling or intense interactive IP applications, other performance indicators are as important as RTT: jitter, packet loss, QoS …

Calculation of theoretical performance with accuracy depends not only on the technology but the implementation made by the service provider. While the performance of a satellite link with SCPC point-to-point configuration can be predicted with accuracy, performance of a satellite link with TDM/MF-TDMA star-topology configuration is more difficult to be predicted, as the existence of shared resources and other users have to be considered.

5.1.1 Theoretical RTT and Jitter

RTT for a communication over a GEO satellite is basically given by the propagation delay. The delay in sending a signal from a point on the equator beneath the satellite, 35838 km to the satellite and 35838 km back, is substantial.

The delay of communication between two locations on earth directly under the satellite is in fact (2 x 35838) / 300000 = 239 ms one-way, or 478 ms two-way (RTT). For other locations not directly under the satellite, the delay is even a little bit longer.

To calculate the expected delay in an IP communication over satellite, you have to take the propagation delay and add the delay in processing signal at ground stations (modems, routers …) for SCPC and TDM/MF-TDMA.

An SCPC provides less delay and jitter, comparing to a TDM/MF-TDMA link. This is because the high CPU power and simplicity of the processing load of the SCPC modems, versus a cheaper TDM/MF-TDMA modem and the complexity in tasks like resource management and timing synchronization of a TDM/MF-TDMA platform.

For GEO satellites, RTT for an SCPC link is expected to be in the range of 500-550 ms, while RTT for a TDMA link is expected to be in the range of 600-750 ms.

It is also important to remark that the delay in codification can have a big impact in the end-to-end delay, if a ModCod with high delay codification is selected and the traffic rate of the carrier is low. Selection of ModCod needs to be selected according to a balance of satellite link budget (availability), spectral efficiency (Bits/Hz) and requirements of IP applications.

In DVB-RCS/RCS2, delay in LDPC-BCH codification is high for DVB-S2, but as this TDM carrier type is used for the shared forward channel, the real impact is low with high traffic rates, of several (tenths) Mbps.

In SCPC, and as an example, since the development of the Turbo Product Codes, and its widely application in satellite modems, the use of different TPC rates provides a good choice to obtain a good spectral efficiency with low Eb/N_0 requirements. But some of the TPC rates, because of the codification packet size, add a significant delay in the codification process, so they are not recommended for low traffic rates, if the delay is very important.

Considering technical values provided by Comtech EF Data for its CDM570-IP modem, one of the most widely used modem for satellite IP links, impact in end-to-end delay for different FEC modes using Viterbi and TPC is as follows:

FEC Mode	End-to-end delay (ms) for 64 kbps data rate
Viterbi Rate 1/2	12
Viterbi Rate 1/2 + Reed Solomon	266
Turbo Product Coding, Rate 3/4	47
Turbo Product Coding, Rate 21/44	64
Turbo Product Coding, Rate 5/16	48
Turbo Product Coding, Rate 7/8	245
Turbo Product Coding, Rate 0.95	69

If you double the traffic rate, you divide the delay by two, so for medium traffic rates, greater than 2Mbps, the differences are minimal. Example:

FEC Mode	End-to-end delay (ms) for 2Mbps data rate
Viterbi Rate 1/2	0.4
Viterbi Rate 1/2 + Reed Solomon	8.3
Turbo Product Coding, Rate 3/4	1.5
Turbo Product Coding, Rate 21/44	2
Turbo Product Coding, Rate 5/16	1.5
Turbo Product Coding, Rate 7/8	7.7
Turbo Product Coding, Rate 0.95	2.2

With the standardization of LDPC-BCH codes for DVB-S2, most of the satellite modem manufacturers have developed their own proprietary parity check coding systems, trying to combine the good Eb/N_0 performance of LDPC-BCH but using less coding block length, to minimize delay.

As an example, Comtech EF Data have developed VersaFEC.

Eb/N_0 performance of VersaFEC is very close to DVB-S2 LDPC-BCH:

Coding Type	vs. Shannon Bound
Sequential or Viterbi	+ 4-8 dB
Turbo Product Codes	+ 2-3 dB
VersaFEC	+ 1-2 dB
DVB-S2 LDPC+BCH	+ 0.7 - 1.5 dB

Throughput and efficiency for VersaFEC are very close to DVB-S2 LDPC-BCH. Example:

	Viterbi + RS	TPC Coding	VersaFEC	DVB-S2
Fixed C/N	6.0 dB	6.0 dB	6.0 dB	6.0 dB
BW = SR	5.0 MHz	5.0 MHz	5.0 MHz	5.0 MHz
Best Modulation / Coding for Es/No	QPSK ½	QPSK ¾	QPSK .803	QPSK 5/6 (.827)
Spectral Efficiency	0.92 bits / Hz	1.5 bits / Hz	1.61 bits/ Hz	1.65 bits / Hz
User Data Rate	4.6 Mbps	7.5 Mbps	8.1 Mbps	8.3 Mbps

The main improvement of VersaFEC vs. LDPC-BCH is the latency:

Modulation	VersaFEC Rate	Latency at 64 kbps (ms)
BPSK	0.488	26
QPSK	0.533	53
QPSK	0.631	59
QPSK	0.706	62
QPSK	0.803	66
8-QAM	0.642	89
8-QAM	0.711	93
8-QAM	0.780	97
16-QAM	0.731	125
16-QAM	0.780	129
16-QAM	0.829	131
16-QAM	0.853	132

Solutions of this type can be found on different satellite modem manufacturers.

Jitter for a SCPC link over a GEO satellite is very stable, and it is expected to be in less than 30ms, if proper design and configuration have been done. This is valid also for MCPC/SCPC architecture, where a forward MCPC carrier is shared by several SCPC links.

Jitter for a TDM/MF-TDMA link over a GEO satellite depends on several factors, as sizing of shared resources and the different service profiles defined. Standard service profile for simple Internet access without QoS support can have up to 150ms of jitter, but any modern satellite TDMA platform should have the possibility to establish different prioritization, QoS levels and efficient resource management to achieve less than 50ms for jitter, for real-time applications like VoIP or multi-conferencing.

5.1.2 Theoretical packet-loss

Packet-loss probability for a communication over a GEO satellite is basically given by the availability used in the calculation of the link budget.

Availability for satellite communications link budget is calculated and given as a percentage over a year, for a Bit Error Rate (BER).

The link budget includes a power margin to deal with rain fade attenuation. The more accurate the rain fade pattern for the location is provided, the more accuracy of the desired availability could be achieved.

Link budgets are done by the service provider according to:

- Location of the earth stations.
- Band (C, Ku, Ka …) and satellite.
- Configuration of satellite transponder to be used: saturated EIRP, saturated flux density, bandwidth, output backoff, input backoff…
- Rain fade patterns for locations.
- G/T and EIRP values for locations under selected satellite/transponder beam.
- Traffic rates and limitations of antenna size and transmit power for earth stations.
- Satellite modem to be used, selected ModCod and required Eb/N_0 for specific Bit Error Rate (BER) performance.
- Desired availability.

Usually there is the need to make a balance between antenna size/power, ModCod and desired availability, as these ones are the variables it is possible to play with.

While increasing the size/power of the antennas can increase availability or Bit/Hz efficiency, costs of hardware and its transportation and installation need to be taken into consideration also, and included in the whole business plan.

Typical availability values than can be achieved in operational links are:

- MCPC/SCPC and C-Band: 99.95% availability in a year for BER $1x10^{-8}$.
- MCPC/SCPC and Ku-Band: 99.80% availability in a year for BER $1x10^{-8}$
- TDM/MF-TDMA and Ku-Band: 99.50 % availability in a year for BER $1x10^{-8}$

If using any type of ACM, availability can be increased at the cost of reducing traffic rate when rain fade attenuation affects the link.

5.1.3 Theoretical QoS support

All modern MCPC/SCPC and TDM/MF-TDMA platforms for IP transporting support traffic prioritization with different solutions.

QoS should be available by service providers in all platforms with service profiles adapted to professional enterprise market. It is difficult to find QoS support on large broadband platforms for Internet access, focused on residential market, but basic QoS to support one VoIP call.

Solution for providing QoS can be based on Diffserv or prioritization of IP addresses/ports/applications, with possibility of dynamic assignment of bandwidth according to the type of traffic or QoS level. Proper sizing of resources and proper QoS configuration guarantee fulfilment of throughput and jitter requirements for IP applications sensitive to these variables, like VoIP.

DiffServ QoS support is normative for DVB-RCS2 standard, as well as it is implemented in most of the not compatible with DVB-RCS2 VSAT platforms and most of the SCPC modems with integrated IP router/processor.

5.1.4 Theoretical throughput

IP throughput for a communication over a GEO satellite link is basically given by five factors:

- Overhead loss, result of the selected encapsulation for transporting IP packets over the satellite link.
- Transparent mechanisms to improve the performance of TCP communications, like "TCP Spoofing" or "Performance Enhance Proxies".
- Transparent optimization/compression for IP headers/payload.
- Efficiency (Bits/Hz), result of the calculated link budget for the selected ModCod.
- Use of dedicated bandwidth resources or shared ones (with guaranteed CIR and/or contention…).

For SCPC links, theoretical IP throughput is easy to be foreseen, as there is no shared bandwidth between users, and a leased carrier is assigned to that communication.

Sizing is done according to encapsulation overhead and efficiency, and a gain optimization can be considered if functionalities like IP header/payload compression are possible.

Real throughput can vary from theoretical:

- If average IP packet size is lower than expected, and the considered overhead need to be adjusted as result of more IP header bytes.
- Only if IP header/payload compression is used, if IP packet payload is not compressible in the way as it was considered, because of the type of the IP applications.
- If there is no mechanism for "TCP Spoofing", throughput of some TCP-IP applications can be affected by the limitations of TCP and the existing of the satellite delay.

For TDM/MF-TDMA links, theoretical throughput is more difficult to be foreseen and guarantee. Because of the satellite bandwidth is shared between users, the service profile is normally given as a nominal download/upload traffic rate (achievable if there are no other users using the bandwidth at that time), plus a contention ratio (or guaranteed CIR). Also, for low-cost Internet access on consumer market, there are service operators with service profiles based on monthly consumed volume.

Management of a TDM/MF-TDMA network can be very complex, as different service profiles with different nominal traffic rates and different contention ratios can coexist, using the same bandwidth resources.

Service operators need to have the required bandwidth resources to provide service to all their users with the required SLA. Also, service operators need to have implemented the required mechanisms to guarantee a fair access policy and traffic control, to ensure equitable service access for all users and avoid congestion. These mechanisms include the possibility of reducing the nominal traffic rate for users according to some criteria.

Example of service profiles that can be found on commercial TDM/MF-TDMA services are:

- Traffic rate (nominal download/upload speed):

From 256Kbps/64Kbps to 20Mbps/6Mbps.

- Contention:

From 1:1 (“simulation” of a SCPC link) to 1:5-1:20 (for VPN connectivity on professional market) and up to 1:50-1:200 (for low-cost Internet access on consumer market).

5.2 Simulation and experimental results

5.2.1 Saturation throughput and available capacity under optimal conditions

Following results are taken from three different networks/systems in commercial service, showing the performance of different solutions in day-to-day operation under real conditions:

- **Tooway (http://www.tooway.com)**. Tooway is an Internet broadband access service, operated by Eutelsat and with coverage mainly in Europe. The service uses satellite Eutelsat Ka-Sat (Ka-band) and the technology for remote terminals is Viasat Surfbeam-2, with TDM/MF-TDMA topology. The service is offered with two main groups of profiles: Consumer (designed for use at home) and Professional (designed for use at small offices). Resources of the network are shared by hundreds of thousands of terminals connected at the same time, across Europe. The service includes a proprietary web-accessed tool to measure locally the performance of a single connection (throughput, latency, jitter,…). Measures taken by this tool can be used to show performance for different profiles/sites at different times, and see differences according to network load.
- **TDM/MF-TDMA service based on Hughes HN**. Hughes HN is used by a lot of operators and service providers for VPN and Internet broadband access. The measures are taken from a Hughes HN platform that uses satellite Hispasat 1-D (Ku-Band) at 30ºW, with Hub station located in Spain and coverage in Europe.
- **MCPC/SCPC network based on CDM570/CDD564**. Comtech CDM570 is a satellite modem widely used for MCPC/SCPC solutions. Measures are taken from a CDM570/CDD564 platform that uses satellite Amazonas-3 (C-band), with MCPC station located in Quito (Equator) and several SCPC stations located across Equator, including Galapagos Islands.

5.2.2 Saturation throughput and available capacity under optimal conditions

Tooway TDM/MF-TDMA broadband access

Professional Data Services (with guaranteed CIR):

Measured with proprietary Tooway web tool, at different times (different network occupation levels), for different network locations (different coverage beams) and different services/profiles. User traffic at the terminal is not connected, so the measure shows the maximum throughput available at the remote station at that time.

Date	Location	Profile	Measured throughput
2013-02-04 17:00	Madrid	Gold 18/6 Mbps	17.92 Mbps / 4.49Mbps
2013-06-24 11:35	Cadiz	Bronze 18/6 Mbps	13.09 Mbps / 5.18 Mbps
2013-06-24 12:45	Malaga	Gold 18/6 Mbps	17.19 Mbps / 5.26 Mbps
2013-06-25 11:20	Valencia	Bronze 18/6 Mbps	17.73 Mbps / 5.22 Mbps
2013-06-25 12:00	Ciudad Real	Bronze 18/6 Mbps	17.11 Mbps / 5.24 Mbps
2013-07-01 11:05	Madrid	Bronze 18/6 Mbps	17.01 Mbps / 5.07 Mbps
2013-07-10 21:30	Madrid	Platinum 18/6 Mbps	16.38 Mbps / 5.39 Mbps

As it can be observed, real throughput is very close to nominal maximum of the profile.

Consumer Broadband:

Measured with proprietary Tooway web tool, at different times (different network occupation levels), for a single remote with real traffic, showing effect of heavy use and saturation on shared resources when there is no QoS and no CIR:

Date	Location	Profile	Measured throughput
2013-06-26 10:30	Mallorca	Absolute 20/6 Mbps	9.49 Mbps / 4.21 Mbps
2013-06-26 10:32	Mallorca	Absolute 20/6 Mbps	9.50 Mbps / 4.25 Mbps
2013-06-26 10:39	Mallorca	Absolute 20/6 Mbps	9.51 Mbps / 4.19 Mbps
2013-06-26 10:52	Mallorca	Absolute 20/6 Mbps	9.49 Mbps / 4.20 Mbps
2013-06-26 10:57	Mallorca	Absolute 20/6 Mbps	8.66 Mbps / 4.19 Mbps
2013-06-26 11:10	Mallorca	Absolute 20/6 Mbps	9.50 Mbps / 4.20 Mbps
2013-06-26 11:25	Mallorca	Absolute 20/6 Mbps	20.84 Mbps / 4.26 Mbps
2013-06-26 11:28	Mallorca	Absolute 20/6 Mbps	21.19 Mbps / 4.23 Mbps
2013-07-03 16:59	Mallorca	Absolute 20/6 Mbps	2.17 Mbps / 0.10 Mbps
2013-07-03 17:14	Mallorca	Absolute 20/6 Mbps	1.32 Mbps / 0.10 Mbps
2013-07-05 07:08	Mallorca	Absolute 20/6 Mbps	14.80 Mbps / 4.23 Mbps
2013-07-05 07:12	Mallorca	Absolute 20/6 Mbps	21.98 Mbps / 4.24 Mbps

As it can be observed, real throughput is very variable for different hours and dates. Consumer profiles of Tooway are designed to be used by residential users at home, not for professional use.

Hughes HN TDM/MF-TDMA broadband access

Location of remote site: Madrid.

Type of terminal: Hughes HN7000S.

Service/Profile: Internet 2Mbps/512Kbps (Down/Up) with 1:32 contention.

Measured with proprietary web tool at different times (different network occupation levels), for a single remote with no other traffic, showing effect of saturation on shared resources. There is no QoS.

Date	Location	Profile	Measured throughput
2013-07-15 10:00	Madrid	Internet 2M/512K 1:32	2.23 Mbps / 1.03 Mbps
2013-07-15 11:00	Madrid	Internet 2M/512K 1:32	2.25 Mbps / 0.81 Mbps
2013-07-15 12:00	Madrid	Internet 2M/512K 1:32	2.49 Mbps / 1.10 Mbps
2013-07-15 14:00	Madrid	Internet 2M/512K 1:32	1.88 Mbps / 0.68 Mbps
2013-07-15 16:00	Madrid	Internet 2M/512K 1:32	1.04 Mbps / 0.46 Mbps
2013-07-15 18:00	Madrid	Internet 2M/512K 1:32	2.10 Mbps / 0.73 Mbps

As it can be observed, real throughput is significantly variable for different hours, but even when a contention ratio of 1:32 in a 2Mbps/512Kbps only guarantees a CIR of 62.5/16 Kbps, the throughput is always greater than CIR and very close to maximum nominal, and even greater sometimes (when there are available/free resources).

5.2.3 RTT and Jitter

MCPC/SCPC link Quito - Cube

Quito (Pichincha, Equator) -> Long: -78.40 ºE. Lat: -0.32 ºN.

Cube (Esmeraldas, Equator) -> Long: -79.63 ºE. Lat: 0.58 ºN

Satellite: Amazonas-3 (61ºW)

Technology: CDM570L-IP modems

Shared MCPC Forward Carrier: Quito->Cube 8-PSK TPC 3/4 4103 Kbps

Leased SCPC Returnlink Carrier: Cube->Quito 8-PSK TPC 3/4 324 Kbps

Measured with ICMP ping, modem-to-modem, at medium link occupation (50%).

Packet Size	Packets	Loss	Average RTT	Min/Max RTT
32 bytes	2000	0%	523 ms	511/532 ms
100 bytes	2000	0%	523 ms	513/534 ms
500 bytes	2000	0%	523 ms	514/535 ms
1000 bytes	2000	0%	523 ms	514/535 ms

MCPC/SCPC link Quito - Galápagos

Quito (Pichincha, Equator) -> Long: -78.40 ºE. Lat: -0.32 ºN.

Puerto Ayora (Islas Galápagos, Equator) -> Long: -90.32 ºE. Lat: -0.74 ºN

Satellite: Amazonas-3 (61ºW)

Technology: CDM570L-IP modems

Shared MCPC Forward Carrier: Quito->Galápagos Q-PSK TPC 7/84200 Kbps

Leased SCPC Returnlink Carrier: Galápagos->Quito 8-PSK TPC 3/4 2100 Kbps

Measured with ICMP ping, modem-to-modem, at medium link occupation (50%).

Packet Size	Packets	Loss	Average RTT	Min/Max RTT
32 bytes	2000	0%	516 ms	503/522 ms
100 bytes	2000	0%	516 ms	505/522 ms
500 bytes	2000	0%	517 ms	505/522 ms
1000 bytes	2000	0%	523 ms	514/535 ms

Tooway TDM/MF-TDMA broadband access

Location: Madrid

Service/Profile: Tooway Professional Diamond

Measured with proprietary Tooway web tool, at very low network occupation (for reference):

	Test 0
Latency:	748 ms
Jitter:	19 ms

Measured with proprietary Tooway web tool, at medium network occupation, with three consecutive measures (no QoS):

	Test 1	Test 2	Test 3
Latency:	812 ms	784 ms	768 ms
Jitter:	103 ms	36 ms	14 ms

<u>Hughes HN TDM/MF-TDMA Broadband access</u>

Location of remote site: Madrid.

Type of terminal: Hughes HN7000S.

Service/Profile: Internet 2Mbps/512Kbps (Down/Up) with 1:32 contention.

Measured with VSAT-to-Hub ICMP ping, without any QoS, at medium link occupation (50%) and without entering Internet.

Packet Size	Packets	Loss	Average RTT	Min/Max RTT
32 bytes	2000	0%	632 ms	585/699 ms
128 bytes	2000	0%	646 ms	594/725 ms

5.2.4 Comments on results: latency and jitter

The main difference in performance between TDMA and SCPC solutions is latency and jitter. If low latency and jitter are needed to proper performance of the IP application to transported over the satellite, SCPC solution always offer better performance than TDMA. This is the reason why SCPC solution has been, and it still is, chosen by operators for satellite links carrying 2G/3G cellular traffic, as the performance of this traffic is very sensitive to these parameters.

5.2.5 QoS support and traffic differentiation

Here an example of configuration and performance of QoS is given for a CDM570L-IP SCPC link carrying GSM Abis traffic. GSM packets from an Abis link are optimized and encapsulated in IP packets, and transported using 3 levels of prioritization:

Internal signalling: CLS6, priority 2.

GSM HDLC signalling: CLS5, priority 3.

GSM Voice/GPRS: CLS4, priority 4.

<u>QoS Configuration:</u>

Class	DSCP	Priority
EXFD	101 110	3
CLS1	001 000	7
CLS2	010 000	6
CLS3	011 000	5
CLS4	100 000	4
CLS5	101 000	3
CLS6	110 000	2
CLS7	111 000	1
ASF1	001 xx0	7
ASF2	010 xx0	7
ASF3	011 xx0	7
ASF4	100 xx0	7

QoS monitor and statistics:

Class	Sent Pkts	Pkts Per Sec	Sent Bytes	Drop Pkts	Drop Bytes
DEF	56853	0	3046835	0	0
EXFD	0	0	0	0	0
CLS1	0	0	0	0	0
CLS2	0	0	0	0	0
CLS3	0	0	0	0	0
CLS4	171220875	101	4264101217	0	0
CLS5	51546298	32	1525918860	0	0
CLS6	3425344	2	89135382	0	0
CLS7	0	0	0	0	0
ASF1	0	0	0	0	0
ASF2	0	0	0	0	0
ASF3	0	0	0	0	0
ASF4	0	0	0	0	0

5.2.6 Costs: CAPEX and OPEX

Pure SCPC point-to-point links are only needed when trying to connect two different remote locations, locations where there is not any earth station infrastructure.

Usually, pure SCPC network evolve to an MCPC/SCPC network with star-topology, where there is a central MCPC station and several SCPC remote stations, and a shared carrier is transmitted from MCPC station and received for all the SCPC stations, while each SCPC station transmits its own carrier back to the MCPC station.

It does not have sense to compare CAPEX/OPEX of a pure SCPC point-to-point link with CAPEX/OPEX of a remote VSAT inside an existing TDM/MF-TDMA platform. Consider than for a pure SCPC link you need to use two SCPC stations, stations that are more expensive than a TDMA one.

Standard SCPC station cost can be around 8.000,00 USD, for an L-Band SCPC IP modem, a 1.8m Ku antenna, a Ku PLL LNB and a 4W Ku BUC; while a professional TDM/MF-TDMA station sold as a VSAT kit can be around 2.500 USD for a 1.2m Ku antenna, integrated RF unit with LNB/BUC-3W and professional VSAT terminal. Stations sold as a kit for residential market and simple Internet access can be found for less than 1.000,00 USD.

Costs can vary according to different manufacturers, optional functionalities and taxes for each country.

The comparison between TDMA and SCPC can be useful if comparing a remote station inside a TDM/MF-TDMA network vs. a remote station inside an MCPC/SCPC network.

To compare CAPEX/OPEX for both solutions, we will assume some consideration to make a fare comparison:

- The same frequency band, satellite and transponder are used for both solutions.
- Both remote stations are connected to the same shared earth station, with the same antenna size and power.
- Standard remote station is composed on the antenna, the RF unit and the satellite modem. No extra hardware is needed and all the functionalities are provided by the satellite modem.

- The required traffic rate is 1 or 2 Mbps, bi-directional, and as result of the link budget calculations, the remote antenna size and transmit power are the same one for both solutions. This can be true if using the same DVB-S2 scheme for forward carrier, and similar returlink configurations using TPC, for example.

With all these considerations, the main difference in CAPEX between the solutions is the satellite modem. An SCPC modem can cost more than 5 times the cost of a TDMA modem. This is because of the more possibilities, functionalities and flexibility that the SCPC modem have. SCPC has better performance than TDMA in terms of latency and jitter, so even it is an expensive solution it can be the only one that it is feasible for some applications.

Installation costs for very remote areas are basically impacted by the transportation of huge and heavy elements like the antenna, and the transportation of the technicians that have to install the remote station. Complexity for physical installation is today very similar for both TDMA and SCPC systems, but a higher cost for installing an SCPC system is usually applied, as auxiliary materials like cables and connectors are more expensive for SCPC systems and because of the better qualification and expertise usually found on technicians installing SCPC systems.

Regarding OPEX, the main components of a monthly service fee to be considered are:

- Use of shared resources (not bandwidth) at the central Hub.
- Bandwidth (shared or not), according to service profile.
- On-site maintenance service and spares.

For very remote areas, SLA and monthly service fee is very impacted by the on-site maintenance service, as the transportation of the maintenance technicians to the station is usually difficult and expensive.

Bandwidth cost is high. Reference prices for LatAm region are 3,500.00 USD MHz/month for Ku-Band and between 2.800,00 (hemispherical coverage) and 3,500.00 (global coverage) USD MHz/month for C-Band.

With such an expensive resource, optimization of Bits/Hz ratio is a key driver.

As a reference, considering 1 MHz dedicated to provide an SCPC link, the throughput it is possible to obtain with a widely used configuration for SCPC links, like CDM570 modem using 8-PSK TPC 3/4 and 1.25 roll-off factor, is 900 Kbps bidirectional:

900 Kbps / 3 / (3/4) x 1.25 = 500 kHz

500 kHz x 2 = 1 MHz

As the cost analysis given above may be complex, we give specific figures for the example case proposed in the introduction: 5 Mbps (DL) + 1 Mbps (UL). As the price may change very significantly depending on the chosen configuration and the contention rate, SCPC will be compared with two TDMA options: without contention or with 4:1 contention.

TDM/TDMA solution in Band-C:

Operation parameters:

– Forward DVB-S2 en 16APSK 3/4, roll-off 25%, Pilots On, Normal Frames. Overhead (ineficiencia): aprox. 3%.

– Retorno RCS2 en 8PSK 3/4 Turbo, roll-off 30%. Overhead (ineficiencia): aprox. 10%.

CAPEX:

Remote VSAT TDMA station operating in Band-C (1.8 m antenna, TDMA modem and radio channel) with 25-50W of power consumption (for a solar photovoltaic system, $6000 USD may be considered). Placed in the country: $1500 USD .

Transport and installation in remote location: 3500 USD.

OPEX:

Monthly cost for the use of the TDM/TDMA hub and licenses (not considering the channel): $3500 USD.

For the service itself, two options are calculated, depending on the contention rate:

- Option 1: Service for 5Mbps (DL) + 1 Mbps (UL) with no contention (100% guaranteed), using 3 MHz (2,3 MHz in forward channel plus 0,7 MHz in return channel): $ 3500 USD.
- Option 2: Service for 5Mbps (DL) + 1 Mbps (UL) with 4:1 contention (25% guaranteed), using 3 MHz (0,75 MHz): $ 8400 USD.

SCPC solution in Band-C:

The following configuration is considered:

- Forward channel (controller → femto): 8PSK 7/8 Turbo, Roll-Off 25%.
- Return channel (femto → controller): 8PSK 3/4 Turbo, Roll-Off 25%.

CAPEX:

Central station (controller location) VSAT SCPC in Band-C, 3.8m antenna + modem SCPC + radiofrequency. Placed in the country: $24500 USD.

Remote station (femto location) VSAT SCPC in Band-C, 2.4m antenna + modem SCPC + radiofrequency. Placed in the country: $11500 USD. Power consumption about 100-150W (for a solar photovoltaic system, $15000 USD may be considered).

Transport and installation in central point : $1500 USD.

Transport and installation in remote location: $3500 USD.

OPEX:

Monthly cost of licenses and operation (without the channel): $1000 USD.

Monthly cost for the bandwidth for 5Mbps (DL) + 1 Mbps (UL) using 3 MHz (2.4 MHz in the forward channel, 0.6 MHz in the return channel): $8400 USD.

5.3 Other considerations

Standard IP access through satellite can be used, and it is used, to transport 2G/3G traffic, but it is important to check the requirements of delay and jitter of 2G-3G small cells to select the proper solution.

In 2G deployments over satellite, QoS at the satellite modem is used to establish priority of HDLC signalling packets over voice packets, and priority of voice packets over GPRS/EDGE data packets, so congestion if occurs do not affect the stability of the BTS and the quality of voice communications. Also IP header/payload optimization/compression is used to reduce the overhead of IP encapsulation, compress the payload and minimize the use of satellite bandwidth.

It is important to remark that the performance of “TCP spoofing” and “Performance Enhance Proxies”, as well as the performance of QoS and IP header/payload compression/optimization features, are lost if the IP traffic is sent through an IPSEC tunnel over the satellite, as all the information is encrypted and not visible.

For this reason, for cellular backhauling over IP and satellite it is desirable to use a “clean” connection, based on a private VPN connection, instead of using an Internet broadband connection secured by an IPSEC tunnel.

Finally, and for best performance, an SCPC link will be a better solution, as this is the solution with less delay and jitter, and 3G data performance is very sensitive to these variables.

6 ADDED FUNCTIONALITY AT THE IP LAYER

6.1 Introduction

Many of the new applications implemented and intended for IP networks demand strict requirements on quality service (QoS) in order to function properly. On the other hand not all technologies and network structures have the resources to meet said QoS standards. That being said, currently the use of this kind of applications has massively increased through the Internet and the need of grant access to an increasing user population is inevitable. In order to solve this it requires a great economic effort from those who carry out the implementation of said infrastructure that supports and offers this service. Finding technologies that adapts to the conditions of each sector and have all the proper characteristics that offer a high quality service for a low price is of great interest.

In a basic IP network, the information packages are transferred using the best effort (BE) method. The complexity of the connection is located in the host in away the devices limit themselves to check the header of the package in order to determine the address of the next jump, before a jam the packages are delayed or dismissed.

This kind of issues don’t create any major inconvenience in applications like mail or ftp but for real time services like sound and video, loss of data or data being delayed is not tolerated. These kinds of communication don’t offer the appropriate conditions for a proper performance of the application. Each type of traffic have different types of delaying needs, jitter, bandwidth, loss of packages, availability which are the parameters for the QoS. An IP network should be able to resolve the QoS requirements in a great variety of applications.

At first it would be possible to suggest as a solution an oversizing of the network but with this option you will miss out on infrastructure and therefore have a disproportionate increase in costs. Furthermore with a rapidly increase in traffic volume an oversizing of the network would probably in a short time frame be maxed out.

Another suggestion to implement QoS is managing efficiently the resources of the network. First a classification of all the traffic types that pass through the network is needed in order to offer a service according to their requirements. To achieve this objective the IETF (Internet Engineering Task Force) have defined two models: Integrated services (IntServ) and differentiated services (DiffServ). IntServ is based on signalization (Signaled-QoS) in which the hosts are responsible of alerting the network about the QoS requirements. On the other hand DiffServ is based on planning where all the elements in the red are configured to process multiple traffic types with their respective QoS requirements.

IntServ presents inconveniences in terms of scalability when all the elements in the network have to save information about their state in all connections. DiffServ on the other hand solve this inconvenience since the packages themselves are responsible of transfer the information with what treatment they require. However the planning in DiffServ needs to be performed with extremely care since the network always need to receive a type of traffic it recognized and it’s prepare for, otherwise it won’t be able to offer an adequate service.

Therefore the architecture in a differentiated method needs to define a series of functions such as classify, mark, put on cue, empty etc. in order to perform the traffic process to each traffic type on the web and provide a good and individual treatment.

The implementation of this type or architecture is relative new so a study that gathers all the needed concepts is of high interest in order to have a clear and concise explication of what the differentiated services are.

This document intends to analyze the architecture of the DiffServ and explain each and one of the elements that form and show how it can be implemented to the current networks.

6.1.1 Quality Service (QoS.)

Talking about QoS [CiscoQoS2011] is referring to the capacity of a network to offer greater traffic services in which different applications with different technologies (Frame relay, ATM, Ethernet, 802.1X, etc.) connect to.

Within the services mentioned are of special interest:

- Dedicated bandwidth support
- Reduction of packet loss
- Traffic jam management
- Fixed priorities of traffic through the network
- Shaping (Establish space for traffic within the network)

QoS can be defined as the combination and function of the protocols and technologies that guarantee the delivering of data through the network in any given time [Mendiola2010] allowing the network administrators manage the available resources more efficient to realize a task.

It's also important with a network provides QoS from end to end, otherwise the result will not be the desired one. In order to meet this objective it's needed that each element and each layer of the network have the capacity to provide QoS for a proper performance of the application in question.

On the other hand it's actually very common for the data flow to pass from one network to another (for example heterogeneous networks) where the technology of each network determines how well the QoS is. Therefore it's very important to define universal components where the different technologies can complement each other and as best as possible maintain a good level of QoS to the different applications. In order to guarantee the QoS through different diverse networks three important components are needed.

d) Implementation of QoS on network elements, including: queuing mechanics, scheduling and traffic shaping.

e) Policy management functions in QoS that controls the traffic traveling from one end of the network to the other side.

f) Signaling tools that administrate and coordinate the QoS within the elements of the network.

6.2 Background

6.2.1 Best Effort

In the beginning, IP only supported one type of service known as Best Effort (BE) which handled each package in the network with the same priority. It would not bother from what application or where the destination is, it always tries to the best to deliver but it would not guarantee a good service since data loss and errors could occur from time to time. This method allows the applications send their data through the network whenever possible, without any notice and random amounts of data just in case it can be delivered. This method is obviously not reliable.

6.2.2 RFC 791

In September 1981 the first attempt to introduce QoS to the IP protocol by Request For Comments RFC 791 [Darpa1981], in which one bite of the header got reserved and given the name "Type of Service" (ToS) which was divided in three fields [Almquist1992] as you can see in Figure 64.

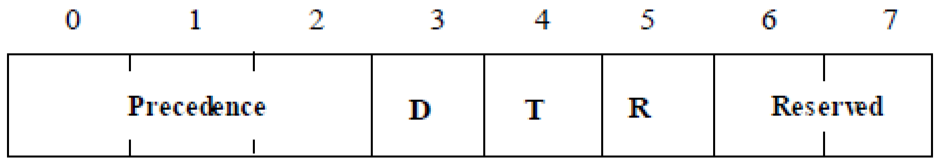

Figure 64: Byte ToS IPv4

The first part of the field is nominated "Precedence" and it's used to indicate the priority of the package where 7 is the highest and 0 lowest. The second part is named as Defined Type of Service (DTS) and it's formed by following bites: D-Delay, T-throughput and R-Reliability. The last part and bites where reserved for future use which most likely will be used to implement IPv6.

Even though this method implements a certain level of QoS and avoid the inconvenience in the network in terms of sacrifice scalability since the information about the QoS is found in the package and not in the nodes. This creates problems for the administrators and traffic managements since you are limited to eight possible traffic types and the limiting factor is the three bites that are part of the precedence. Another problem that might present itself is when there is two packages with the same value in "precedence" then the administrator will not have access to any mechanism to determine which package needs to be dismissed or continue. For these same reasons it very complicated to offer a good QoS from end to end using this method.

6.2.3 Integrated Services – IntServ

This is a model with multiple services which can contain a diverse QoS requirement and make it possible for one application to send a signal message through the network with help of the RSVP (Resource Reservation Protocol) protocol [Braden1997]. This protocol though a request reserves resources like bandwidth and maximum delay in order to send the data which will be sent once this same request have been confirmed.

Unfortunately for this scheme to work properly it requires that all the elements in the network including the hosts understand the protocol (RSVP) perfectly, which make the implementation really complex. Furthermore it's required to keep the flow in each node and its information updated; this can create a traffic jam due to the great number of messages that need to pass between the nodes and the network will be a bit scalable.

6.2.4 Differentiated Services – DiffServ

In 1998 the IETF proposed the differentiated service model – DiffServ [Blake1998] [Nichols1998] as the solution to offer QoS, based in the prioritization of traffic types on an IP level.

For this scheme the provision of the QoS it's done with through a reserve of resources, in this case the applications don't need to create a petition in the network. They just need in a proper manner indicate the amount of traffic it will generate, that way the network can organize the flow in the class they are part of.

As first step in this scheme the header byte ToS name in IPv4 is changed to DS [Nichols1998], and with this change new functions are defined for each byte (Figure 65).

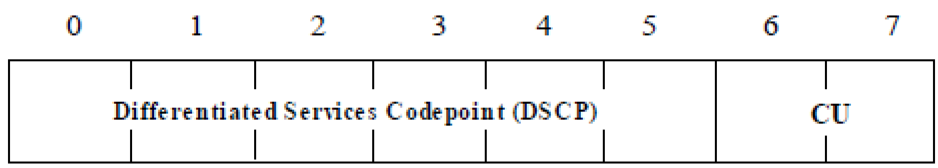

Figure 65: Byte DS IPv4

As you can see the DS byte is divided in two fields: The first 5 bites are known as Differentiated Services Codepoint (DSCP), this part will be used by the routers to identify what class each package belongs to and what they should do with it. The last two bites (6-7) are reserved for compatibility purpose with IPv4 ToS byte and its denominate Currently Unused (CU).

6.3 DiffServ architecture

The propose architecture for DiffServ [Blake1998] [Grossman2002] is based in classifying the traffic that enters the network in different types depending on their behavior, this way traffic groups are created where each one represents a similar behavior and they are named “Behavior Aggregates” (BA).

Then inside the network each BA is assigned a unique DSCP from the moment its specific treatment is identified or “Per-Hop Behavior” (PHB) that needs to be realized on each package in question.

6.3.1 Differentiated Services Domain (DiffServ Domain).

The group of nodes that implement DiffServ are defined as “DiffServ Domain” or DS domain. Within this group two element types are recognized: The border nodes (DS boundary nodes) and interior nodes (DS interior nodes).

A DS domain is characterized for having limits that are well defined and consist of the border nodes which classify the traffic that inters the domain. That way the packages that are within the domain are marked by a PHB inside this group that have previously defined for this domain.

The interior nods on the other hand are responsible of route packages and determined what to do with using the DSCP. In Figure 66 you can see the basic architecture of the DiffServ model.

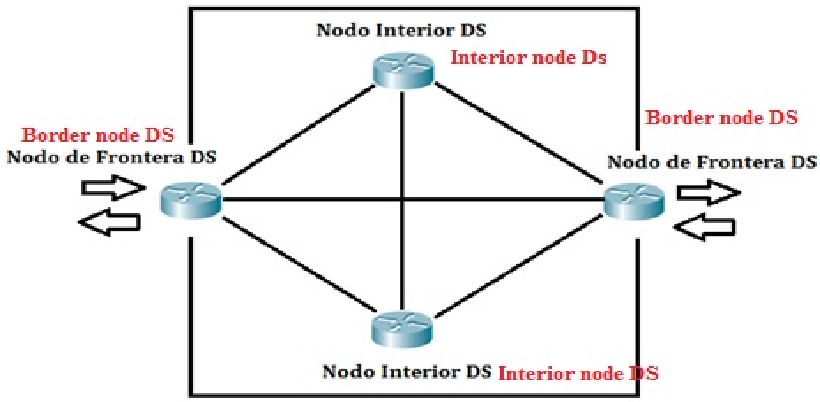

Figure 66: Byte DS IPv4

Occasionally a network can find a router which don’t recognize DiffServ, this are nominated “non – DS – complaint – nodes”. This situation represents a state of uncertainty when the performance results are not optimal and therefore the capacity of the network to satisfy the meet standards between the client and the provider are limited (SLA, Service Level Agreement).

The elements of he basic architecture of the DiffServ model are:

- **Boundary Nodes DS.** Boundary nodes work sometimes as “Gateways” between the domain DS and the external networks and are expected to mark the packages that enters with the proper PHB while DSCO are being used. Then in order to operate within domains DS the boundary nodes must act as exit and entry nodes, this means they are responsible of organizing the traffic that intend to enter/exit from the domain DS which the "Traffic Conditioning Agreement" (TCA) indicate and established between domains mentioned above.

- **Interior Nodes DS.** Interior nodes as boundary must be able to apply a correct PHB as indicated in DSCP but they are only connected to other interior nodes or a boundary node within the same domain DiffServ. For that same reason their traffic organizing functions are less limited and simple compared to the boundary nodes. The most important function for these nodes is to provide a queuing mechanism that is used as mean to prioritize of packages.
- **Differentiated Region Services (DS Region).** Its denominated DS Region when two or more DS domains are connected to each other. Each one of the domain can manage their own PHB in order to assure the technical support of the network and its service regardless what route the data flow takes. The connected domains must establish clear rules that specify how the packages pass form one domain to another, therefore a SLA between them is needed that implicit or explicitly restrain said terms of rules.

6.3.1.1 Classification and Traffic Conditioner

As it have been mentioned earlier one important part of this model is the implementation of the classification package polices which identifies the traffic subgroups that will be mapped or managed through a BA within the DS domain.

On the other hand the traffic management refers to the following space process, shaping, remarking, metering and policing (policy functions), used to assure the rules in the TCA are mee.

The classification process has the following stages:

- **Classifiers.** In the DiffServ model two types of classifiers are defined: The first is BA which realizes the classification of packages starting from the value in DSCP. The second is known as "Multi – Field" (MF), it realizes the classification of packages starting from the first value in the header IP package.
- **Traffic Profiles.** It's used to specify the temporary properties in a data flow selected by the classifier. It can determine the flow profile state (in – profile, out – of – profile). The in – profile means the package fulfills with all the norms set for a said profile, it would then be allow to enter the domain without any condition. Packages with out – of – profile status will be put in queue until it gets updated with a new codepoint (in – profile).
- **Traffic Conditioner.** After selecting a data flow the classifier send said flow towards a traffic conditioner. Figure 67 shows the diagram in blocks of the conditioner. Worth mentioning is that not always the elements show in this figure are fund, but all their requirements are implemented in each node.

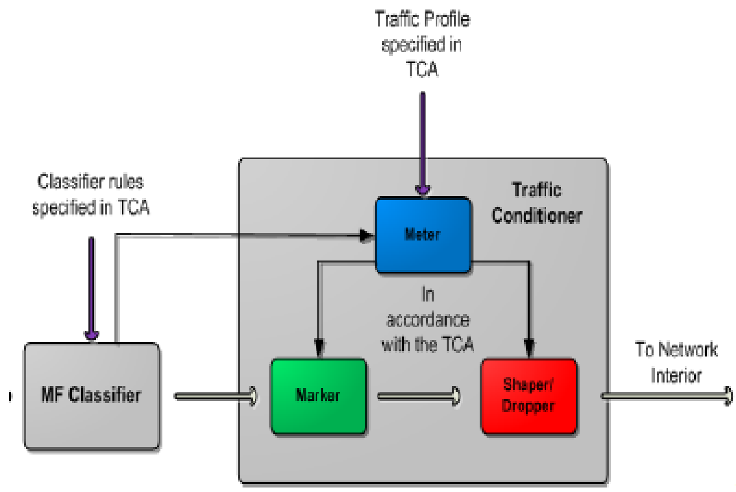

Figure 67: Traffic Conditioner

- **Meters.** Meters focus on the data flow selected by the classifier and measures their temporary proprieties to compares them with the requirements in a traffic profile defined in the TCA. This will help to identify the packages as in – profile or out – of – profile. Afterwards this information is passed to the other units of the traffic conditioner so they can perform their corresponding tasks.
- **Markers.** Markers are responsible of adding packages to a specific BA since they are who can modify the DSCP. In other words it selects on behalf of the PHB and based on the information received by the meter.
- **Shapers.** With the end of adjust the flow in an explicit traffic profile the traffic shapers are responsible of delaying some or all data flow packages. Usually a sharper have a small buffer and if the space is not enough to store the delayed packages it will dismiss of them.
- **Droppers.** In order to adjust the flow to a specific traffic profile the droppers will dismiss some or all the packages found within the traffic flow. An special case called Sharper is presented when the buffer have the size zero, known as one of the implementation forms of a dropper.

6.3.1.2 Localization of Traffic Conditioners and Multi-Field (MF) Classifiers

It's more common to find traffic conditioners and MF classifiers on the entry or exit of DS nodes, but you can also find them in interior nodes of a DS domain as well in a non-DS domain.

- Inside the source domain. The node where the data origins from is what denominate "source domain". The classify functions and traffic conditioner can be performed by the source that generates the traffic in the intermediate nodes within the source domain. Before exiting the source node, the traffic generated there will be marked directly and it's known as initial or pre-marking. When the packages are being marked close to the traffic source you gain some advantages. First, traffic sources manage to assume with a greater ease the precedence of the applications and the presence of the packages. Another advantage is the reduced amount of rules in the nodes thanks to the lower complicity in classification of the source packages, since the amount of different source types are reduced and needs to be classified.
- At the border of a DS Domain. Normally inside the SLA established between domains its need to specify which one will be responsible to map the traffic flow to the BA and at the same time condition the packages as indicated the corresponding TCA. Nevertheless an entry node can receive types of traffic that are not in line with the TCA or data flows from domains that

don't support DiffServ. For that same reason the node must have the capacity to apply the required conditions by the DS domain.

- Domains who don't implement DiffServ. In order to pre-mark the traffic before it reach the entry node of a DS domain traffic, conditioners can be applied in the source of traffic or the interior nods of a non-DS-capable domain. The classification and marking, local politics calls can be hidden in this particular case.

- In the interior DS nodes. The classification and conditioner of complex traffic are found in the exit and entry of a boundary node which is assumed by the basic architecture but on the inside of the network it's not possible to implement these functions.

6.3.1.3 Per – Hop Behaviors (PHB)

A PHB is the special treatment a node realizes to the content of the packages through a BA, in other words packages with the same value as DSCP. This treatment is known as scheduling and it have to monitor the node in terms of space and queuing of packages.

In [Black2001] there are four defined standards for PHBN: "The Default PHB" (PHB by default), "Class-selector PHBs" (PHB Class selector), "Assured Forwarding (AFxy) PHB" (PHB Secure routing) [Heinanen1999] and "Expedited Forwarding (EF) PHB" (PHB Fast routing) [Davie2002] [Charny2002].

- PHB per defect. The PHB per defect determines that packages with this mask (DSCP = 000000) are treated with a node that supports Best Effort (BE). On the other hand it's possible that packages with a DSCP value not mapped in any of the defined PHB can be assigned PHB per defect.

- Class selector PHB. With the objective to conserve the compatibility with the previous designs the ToS byte values in the DSCP field within the DS byte have the form "xxx000", whereas x represent a codepoint denominated Class – Selector Codepoint. This PHB counts basically with the same behavior as the IP Precedence which allows the DS nodes to coexist and cooperate with previous IP Precedence nodes.

- Expedited Forwarding PHB (EF PHB). The EF PHB has been defined to offer less loss, low latency and low jitter and this way assuring a proper bandwidth. It's intended for real time VoIP applications, video and electronic commerce programs. Applications like this require a robust treatment from the network, in other words the PHB must assure the rate of the exit node when the packages are leaving is the same on entry. The value of DSCP recommended for EF is '101110'.

- Assured Forwarding PHB (AF PHB). AF PHB is designed to process different types of traffic with requirements less demanding in comparison with the indicators for EF. It defines a method which the packages obtain different types of routing. Initially the traffic is divided in: gold, silver and bronze. Gold will dedicate 50% of the bandwidth capacity, while for silver its just 30%. Bronze gives just 20% of the capacity. The AF PHB defines four classes denominated as: AF1, AF2, AF3 and AF4. Each class is assigned an amount of resources (bandwidth and buffer). How much each class gets is determined in the SLA. Inside each class it's possible to define one of three discard levels in order to avoid traffic jams and in case it occurs then it will be easier to manage. For this reason is common to find the AFxy notation where "x" represents the class with a value between 1 and 4 while "y" represents the probability of a package being dismiss. The dismiss probability concept is used for example to penalize the traffic flow that break the rules within its own BA. Table 68 shows the values of DSCP for each dismiss probability. The class AF can de denominated by DSCP"xyzab0", where "xyz" is 001/010/011/100 and "ab" indicated the probability of dismiss.

Dismiss probability	Class 1	Class 2	Class 3	Class 4
Low	001010	010010	011010	100010
Medium	001100	010100	011100	100100
High	001110	010110	011110	100110

Table 68: DSCP for AF PHB

6.3.2 Recommendations

With the thought of implementing a DiffServ model that offers QoS guaranties from end to end it is recommended as first step to use nodes along the whole network that supports DiffServ, in other words the nodes have to be DS-complaints. Should this not be the case unpredictable behaviors will occur that most likely will cause delays and package loss.

On the other hand it must be taken in consideration when designing a DS domain that it may or may not communicated with other non-DS-complaint domains. Therefore it's important to establish agreements that are clear and well defined between the different domains on how they need to send the data flow back and forth.

The responsibility of creating said agreements is of the administrator of each network, which needs to perform this task before implementing the network.

6.3.3 Disadvantages

Basically the presentation of the QoS through the DiffServ is based on the reserve of resources that are created when PHB is implemented in all the nodes. Nevertheless if a type of Internet which the network doesn't recognize or encounter a non-DS-compliant the network will not have the capacity to assure a proper treatment that QoS offers.

6.3.4 Previous jobs

During the process of investigation that involved this document a search is realized of jobs that focus their efforts in perform guaranty on quality services at an IP level. Specially while implementing DiffServ which some are worth of mention.

This architecture have been discussed and investigated since their implementation in 1998 have not been popular due to the inconvenient presented for the way the reserve resource are realized. For this reason 2001 some jobs of investigation like [Trimintzios2001] in which a solution is presented to this inconvenient while the integration of "Multi – Protocol Label Switching" (MPLS) in DiffServ model. However just as information data is presented and supposed delay but more a clear implementation on how to realize the integration and the results of the simulation that confirm the efficiency the suppose implement.

The [Park2003] lay out the integration of DiffServ with wireless network. In this a study in how the traffic administration of the DiffServ with the standards IEEE 802.11e is performed.

Despite an interesting proposal there is no clear simulation that confirms the performance of the design in question of the traffic or of the interface in order to guarantee QoS.

Afterwards [Jiang2006] introduce some problems in order to offer QoS when using a wireless network like backbone. Here the actors investigate the effects of DiffServ on how wireless backbone in term of

QoS, routing and mechanics access to MAC. Nevertheless it just supplies the possible solution to the new lines of investigation but without any confirmation.

In 2009 [Myounghwan2009] presents a method an integration between IEEE 802.11e, DiffServ and the standard IEEE 802.16, in which a mapping is realizing between the classification method service of each technology. With this the authors can manage to create a network totally interoperable. Furthermore an algorithm is being implementing to the management of delays in the interior of the network with the end of adjusting the functions of DiffServ and assures with this the guaranties of QoS from end to end in the network.

[Segura2011] presents interesting data about the benefits in using DiffServ against other options of prioritizing in traffic which not bother about mayor improvements in respect to what have been writing in this document. Very similar it’s what presented in [Nageswara2013] where focus the work and demonstrate the benefits in DiffServ but also not contributing past to suggest an integration with MPLS as future work.

6.4 Virtual circuits: separating and protecting traffic

6.4.1 Definition

In connection orientated networks such as ATM, it’s necessary to perform an establish process of the connection before packages are being sent. Said process consist in sending a message through the network so each communicator create an entry in their tables and generate a path along the network and allowing resources to be reserved.

These types of connections are known as virtual circuits and offer a series of advantages. One of the main advantages is the communication between clients and access points with different speeds. On the other hand are attractive because they allow you to specify characteristics of quality of service (QoS) for each independent circuit.

Being a flexible connection format that can support a great quantity of virtual clients it have to deal with waste of time and resources, specially when the destination address have to be repeated on each package. The solution is to tag the package with a unique ID, that way each package being sent can be allocated to its corresponding circuit. Although permanent virtual circuits can exist (they require a manual configuration and can last for month or years), generally they get destroyed when the connection is shut down [Despres2010]. Permanent circuits are not force to take the same route, that being said in [Gyires1997] the algorithm shows how it’s possible to en-route permanent virtual circuits according to the needs of the network.

Modern networks which support the creation of dynamic virtual circuits are known as Dynamic Circuit Networks where the users can request a creation of personal virtual circuits. Which turn out to be the quality requirements in the QoS needed to perform a correct execution of the applications being used.

This type of networks use middleware systems that allows the users to request the creation of said services and achieve the goals set by the QoS, some example of this middleware are UCLP [Wu2006], OSCARS [Guok2006], AutoBAHN [Geant2013] and DRAGON [Yang2006]. As shown in [Santanna2012] to automatize the decision process when creating a new virtual circuit a number of rules must be established. That being said, it’s not always enough and sometimes it’s required a human intervention. In [Santanna2012] a system is presented, administration of process and business (BPM) in order to prevent the human intervention in the acceptance process of new virtual circuits in dynamic circuit networks.

6.4.2 Advantages and disadvantage

In terms of quality service, virtual circuits have some advantages in resources like (Buffer, CPU cycle, bandwidth which can be fixed or can vary depending on the condition of the network) which can be reserved beforehand the connection is established. Once the packages start arriving the necessary bandwidth and routing capacity will be ready.

On the other hand there is a processing cost to build the circuit that must be paid but this it's compensated with the performance of the memory in the routers due to the fact that when the packages are being sent the ID of the circuit it's needed and not the final destination [Aspnes1997].

6.4.3 Traffic jam control methods

In [Yang1995] a classification of all the algorithms responsible to control the dataflow is divided in two groups, open and close cycle. Basically the algorithms with an open cycle try to avoid a jam focusing on a proper design but when there is a jam, nothing is done to solve the problem. The ones with a close cycle are updated with information by the network, this way they make decision that will correct the jam.

In virtual circuits the default method to use in order to control the dataflow is the later one, closed cycle. Further on some of the existing methods will be explained more in detailed.

- Admission control. The basic idea is once the jam is detected the creation of new virtual circuits will be avoided which will prevent the network jam even worse. As consequence no connections will be permitted at the transport layer.
- Alternative Route. Involve allowing the creation of new circuits but considering the location of jam and avoiding new routes passing through them.
- Resource reservation. In order to reserve any resources there must be a communication between the host and the sub network where the volume and the type of traffic are specified. Only then bandwidth and memory along the whole route will be reserved beforehand. Having this sorted out the chance of getting a traffic jam is very slim.

Following you will find jobs related to the implementation of quality and service process while being used by the virtual circuit. [Lu2010] Presents a framework build to enhance the efficiency of the QoS, it's made of two layers, one that generates general nodes(Routers that are within the same domain) and the second layer locate super nodes(manage general nodes and represent routers that link different domains in the network) and the routing framework based on virtual circuits. [Garcia2003] presents a resource administration system using virtual networks. This job is focused in using the VQN (Virtual QoS Networks) which are dimensioned to provide the desire level of QoS. Normally it's expressed as a connection based on the anticipated demand for each category QoS, where the quality of the service is quantified in terms of performance matrix to a package level as the probably of package loss, delay or delay differences.

6.4.4 Implementation of virtual circuits

Virtual circuits as they are described in [Sato1990] are implemented while virtual routes are being used, cross connections, adding or removal of multiplexors. Said virtual routes are established or are released while the configurations of the routing tables are performed in each of the involved commutators in a dynamic manner and in agreement with long term service provision, the demand in short term and including alternative routes when the network is malfunction.

The initial configuration in an access node a route is identified in order to connect and if needed decide if it use said route or not.

If the route is accepted the resources required will be negotiated and an identifier is assigned to the circuit. This identifier will be needed to route the packages through the virtual network.

6.4.5 Assignation control of bandwidth in virtual circuits

As it has been brought up earlier bandwidth assigned to the virtual circuits can be fixed but in many occasions it can also be dynamic while an analysis is performed to find out available bandwidth in the network and the circuits. As described in [Ohta1992] the control of bandwidth it's an efficient method to reassign resources to each virtual circuit. Basically each circuit and their connections are monitored so that when they increase the capacity assigned will increase. A disadvantage with this method is a possible increment of process needed to maintain the network.

6.5 Perspectives from IPv6

6.5.1 Definition, tools, advantages and disadvantages in providing QoS

The Internet Protocol version 6 (IPv6) defined in RFC 2460 [RFC2460], is designed to replace Internet Protocol version 4 (IPv4) currently being used in most devices with access to the Internet. One advantage with IPv6 among many highlights the increase in length of the source and destination addresses to 128bits which allows a higher number of IP addresses, solving the routing limitation being presented in IPv4. It's estimated that with IPv6 theoretically we could have up to 2128 addresses. Also by extending the packet header, networks would have more security and would limit the misuse of accessing routers through superior layers to process packets when they should only be used a network layer [Parra2011]. Figure 68 shows the IPv6 header format.

```
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|Version| Traffic Class |           Flow Label                  |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|         Payload Length        |  Next Header  |   Hop Limit   |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                                                               |
+                                                               +
|                                                               |
+                         Source Address                        +
|                                                               |
+                                                               +
|                                                               |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                                                               |
+                                                               +
|                                                               |
+                      Destination Address                      +
|                                                               |
+                                                               +
|                                                               |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Figure 68: IPv6 Header Format [RFC2460]

Among the more important fields which provide Quality of Service (QoS), the IPV6 protocol offers:

- Version: 4-bit label of the signal IPv6 = 0110
- Traffic Class: 8 bits label determined for DiffServ classes of traffic [Mongkolluksamee2008]
- Flow Label: 20 bits label for special handling package. Experimental [RFC2460].

As In IPv4, there are two ways to implement QoS in networks using IPv6.

- Diffserv Differentiated Services: This technique is known as the Traffic Engineering (TE), routing guarantees are based on a pre-defined marked in class of packets before being sent by the router. Diffserv field in IPv4 is used as "Type of Service" (TOS), in IPv6 it's named Traffic Class (TC). However both have the same length of 8 bits.
- Integrated Services or Intserv: In this technique the guarantee of QoS is supported in the end-to-end reservation of resources and will be a function of the flow or link requirements. These parameters are set according to the reservation source and destination address, the interface for which information is received. In IPv6 the market using either CT or Flow field label will use different signaling protocols and reservation.

Within differentiated services and integrated services there are several techniques for control, traffic management, concepts and architectures that allow proper administration of user and customer requirements which will be explain in the following section after being tested in simulations and TestBed made in IPv6.

In [Marchese2006], the architecture is defined together with several definitions in a classical model for the deployment of QoS. Which are of great importance to our study, these include:

- Service Level Agreement (SLA): In overall SLA is the agreement between two networks to establish the minimum conditions of communication, such as an Internet Service Provider (ISP) and users with traditional parameters and QoS, fulfillment in correction and time between failures. In the case of QoS, establishing the terms of all connections End to End (e2e) is a priority.
- Autonomous Systems (AS): It's in general an autonomous system with own routing policies and must communicate with other AS, currently the numbering AS is assigned to the ISP, this numbering is administered by the ANSI. Today the AS are indicators of the implementation of IPv6 in the world, according to [Ripe2012], the country with most AS in Latin American is Brazil.
- Relay Point (RP): Are points that connect two AS, within its capabilities include establishing appropriate interfaces, transfer QoS, needs of each e2e through that interface in order to map performance [Marchese2006]. The requirements to pass a RP to another are done right through TC fields (IPv6) or TOS (IPv4) which defined the "The DiffServ codepoint" (DSCP) which it is a map of both protocols [Nichols1998]. Map handling and queuing priority depends on the equipment used, in the case of embedded systems with Linux kernel trees of classes, filters and queue disciplines are handled by TC iproute2 tools which have Iptables [Lee2013]. The use of these tools allows the use of different algorithms for queuing disciplines that have been previously mentioned as Class Based Queuing (CBQ), Priority, Hierarchical and Token Bucket (HTB) for prioritization of flows arrive with a specified brand.
- Multi Level Priority Pre-emption (MLPP) policy which can leave some connections already in progress because there are other higher priority flows which require access to the network, when there is enough bandwidth for all of them [Marchese2006]
- Flow Label (FL): 20 bits in size, offers a great opportunity for QoS experimentation, the field was designed to provide an environment support IntServ to identify and manage the flow more efficiently. It use is regulated by the recommendations in RFC 3697. If these fields are before the source and destination address you can label and flow classification them much faster than with IPv4 as [Parra2011], the most important advantages are:

  1. Decreases the average time of the processing load on routers in the network, thus decreasing the time delay in packet delivery systems End to End (e2e).

  2. Resource reservation with FL decreases the problems caused by frequent changes in the routers.

  3. With "flow label field", flow-based routing can easily be implemented.

  In some research like in [Parra2011] it was found the software used in CISCO routers did not support the classification of traffic classes by marking in FL, as a solution to this issue with labels they used Access Control Lists .

  In the commercial field, IPv4 provides the same QoS as IPV6, however higher levels are necessary as the level of implementation and other protocols that can use these fields can be exploited.

6.5.2 Problems

In both IPv4 and IPv6, DiffServ offers more scalability, QoS provisioning with IntServ networks due to the multiple problems posed secure a booking in a network flow. [Xiao1999] [Marchese2006], in the work of [Marchese2006] it is concluded that the use of this paradigm have IPv6 and IPv4, shows no difference in the quality of service. However the authors experiences in this document and their results shows a very successful guide of the behavior in a system, although these results are not trending away, but the value of the results are different in a real environment.

One disadvantage in using the IPv6 protocol to provide quality service today is the difficulty of adapting its interoperability with its predecessor protocol. While waiting for a faster update by many operators and manufacturers of the current state of technology which is predominated in almost all devices and networks. The methods used for the updates of the protocols are dual stack, tunneling and address translation. The double stack is the most popular mechanism in reviews. For this solution all equipment, design and network protocols must be configured to support IPv4 and IPv6 but its limited because not all hardware supports dual stack, it should be noted that embedded systems with Linux kernel, have the characteristic of being enabled with this option.

7 COMPARISON OF THE TECHNOLOGIES VISITED

7.1 On the regulatory environment, operation frequencies and other factors

The three families of technological communications solutions visited may be clearly separated in two sets from a regulatory point of view: satellite communications always work in licensed bands and WiMAX may work on a licensed band. On the other hand, WiFi works in non-licensed bands and 802.16 (accurately speaking, 802.16-compliant systems in non-licensed bands cannot be tagged as WiMAX) may work in a non-licensed band.

The decision of whether the network may operate in a non-licensed band (much less expensive) depends on several factors. However, it can be generally stated that the interference problem is the most radical factor that may incline the balance to one side or the other. Rural areas are less prone to interferences in ISM and UN-II bands than urban areas, but operators subject to strict service level agreements may not be free to use non-licensed bands. Whenever possible, non-licensed bands seem better.

The other critical factor for technology selection is distance between the city and the served rural area. If the distance is enormous and the required number of terrestrial links in cascade for the backhaul is too high, a satellite link may be required no matter what disadvantages it brings to the final solution. However, when a fully terrestrial solution is feasible, a rough comparison of the results presented in this document permits to affirm that any combination of WiMAX and/or WiLD links may provide the access network with better performance at a lower cost.

7.2 Comparing costs: CAPEX and OPEX

The CAPEX of any installation for terrestrial LOS links is very dependent on the topography and the prize to bring towers and other materials to the location. Average prices have been given in the analysis of WiFi (NV2 is similar to plain 802.11 in terms of cost), WiMAX and VSAT respectively. An example proposed in the introductory section has served in all the three sections to estimate prices for the same configuration (5 Mbps -DL- and 1 Mbps -UL-). In the following, Figures 69 and 70 show the result of this comparison.

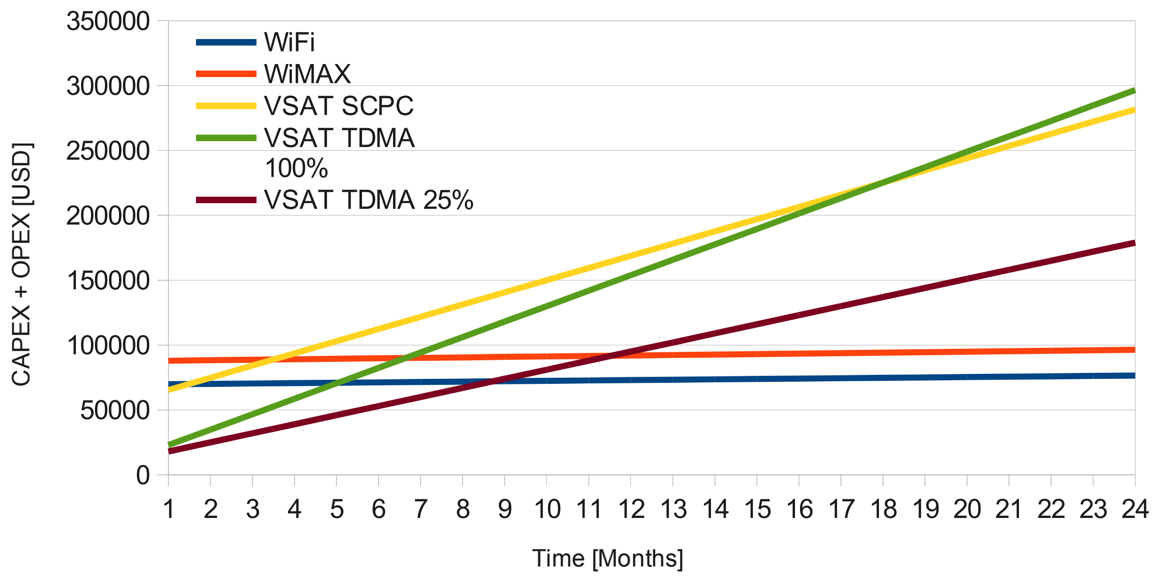

Figure 69: Total cost for one location being installed and serviced. The location is considered to be served through either one VSAT station or one hop of any of the terrestrial radio technologies. In the latter case, two towers are considered necessary for one link (one for each end).

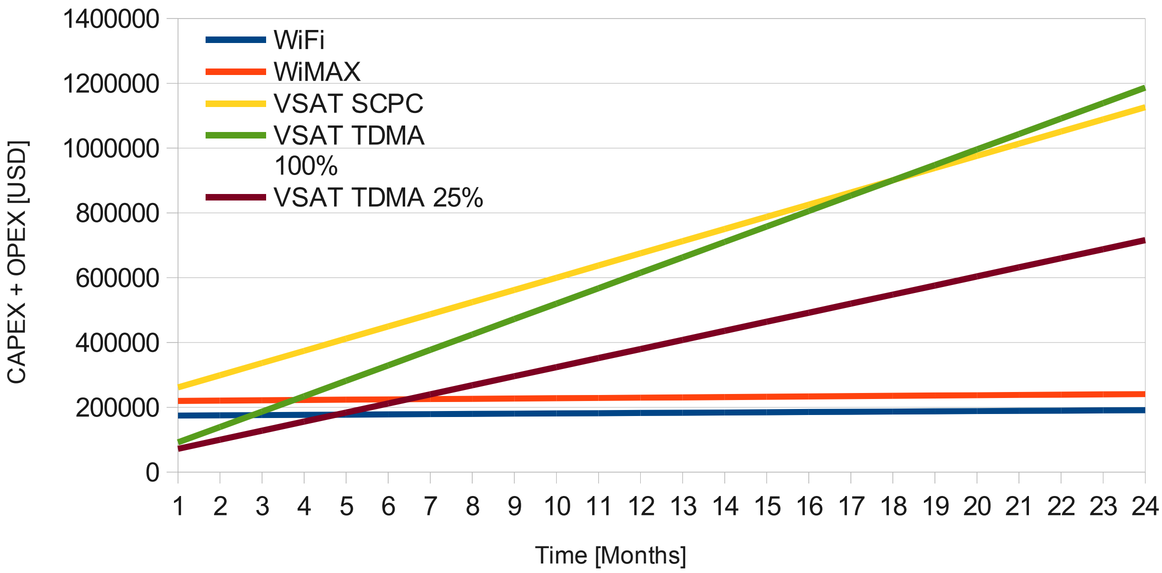

Figure 70: Total cost for 4 locations being installed and serviced. The locations are considered to be served through either one VSAT station per location or through 4 chained links of any of the terrestrial radio technologies. In the latter case, five towers are considered necessary for four links.

Although the CAPEX in WiFi or WiMAX systems is higher than VSAT alternatives (except for SCPC in Figure 70), it becomes obvious that terrestrial options become cheaper than any VSAT option in the first year for one site, and by the seventh month for four sites.

Regarding the satellite options, for one site it is not advisable to use TDMA with 4:1 contention because that makes impossible to give any guarantees to users. For 100% guarantees, SCPC becomes cheaper before the end of the second year. For four locations, the operator may decide to have 4:1 contention if the aggregated traffic of the four locations may be effectively been served by the 5+1 Mbps service, as it is in the WiF and in the WiMAX options. Even though that option becomes much cheaper than SCPC or VSAT TDMA with 100% guarantees, terrestrial options are cheaper after a few months.

WiMAX is always a bit more expensive than WiFi due to the price of equipments and the power consumption, that difference being permanent and stable in the time.

7.3 Comparing performance: throughput, delay, jitter and packet-loss

It is easy to see that 802.11n and proprietary TDMA solutions based on WiFi hardware have a much higher performance than any other alternative for short-range and mid-range links. Even for very long distances, solutions like NV2 give the best results and even plain 802.11n can give better results than WiMAX.

On the other hand, WiMAX performance is more stable and guaranteed in a per-flow basis, which is not the case for WiFi and WiFi-based technologies. Moreover, although 802.16-compliant systems under 11 GHz may not use channels beyond 10 MHz, there are commercial equipments that use 20 MHz in the 5 GHz non-licensed band and double the theoretical throughput. Regarding the delay, WiMAX may have higher delay than WiFi links (although that depends on the frame duration), but certain prioritized flows may obtain guarantees (maximum delay), which is not possible in WiFi.

Satellite communications are much more limited in terms of performance basically due to two reasons:

- Although it is physically possible to have high throughputs, the cost is a very limiting factor.

- There is a natural limitation in the minimum delay, specially important when using geostationary satellites. While WiFi has been tested in under-saturated conditions in order to find an operation point with an average delay of few milliseconds, satellite communications may not go under 500 ms in any case (if geostationary satellites are used).

Regarding only VSAT options, SCPC is the reference option for quality backhaul. However, when very small sites (villages with one undersaturated femtocell per village) may share the bandwidth, shared VSAT TDMA is cheaper and a good option if the contention ratio is carefully chosen.

7.3.1 Performance comparison in short distance links: 802.11n, NV2 and WiMAX

In order to clarify the comparison between terrestrial solutions, some detail results are given below.

In a first instance, Figures 71, 72, 73 and 74, and Figures 75, 76, 77 and 78 show the results of throughput and delay respectively for 802.11n (orange and red colors), WiMAX (blue colors) and NV2 (green colors) technologies in a short distance link (only 5 meters). Although the results are not representative of what happens at all distances, it give a good perspective about the maximum performance that each technology may provide. For 802.11n and NV2 the values were obtained making real tests in the URJC laboratory using Mikrotik equipment. This environment has been detailed previously. The values corresponding to WiMAX were obtained using the model for the throughput values and NS3 simulator for the delay values, and also validated with experimental tests using Albentia ARBA systems. All the results were obtained injecting bidirectional UDP flows with 1500 packet size. While WiMAX and NV2 were tested in an operation point close to saturation, 802.11n was tested in the highest offered load that still does not increase the delay over “a few ms” (2-5 ms). Hence, higher throughputs may be obtained in 802.11n but at the cost of worse delay, jitter and packet-loss values.

Figures 71 and 72 illustrate for the best and worst case the throughput achieved using SISO techniques. The worst case corresponds in 802.11n and NV2 to the use of 20MHz channels and long GI, while for WiMAX it corresponds to 2.5 ms frame duration and CP = 1/4. The best case corresponds to 40MHz channels and short GI for 802.11n and NV2, and to 20ms frame duration and CP = 1/32 for WiMAX. Notice that the frame duration in WiMAX that is better for the throughput is worse for the delay and viceversa.

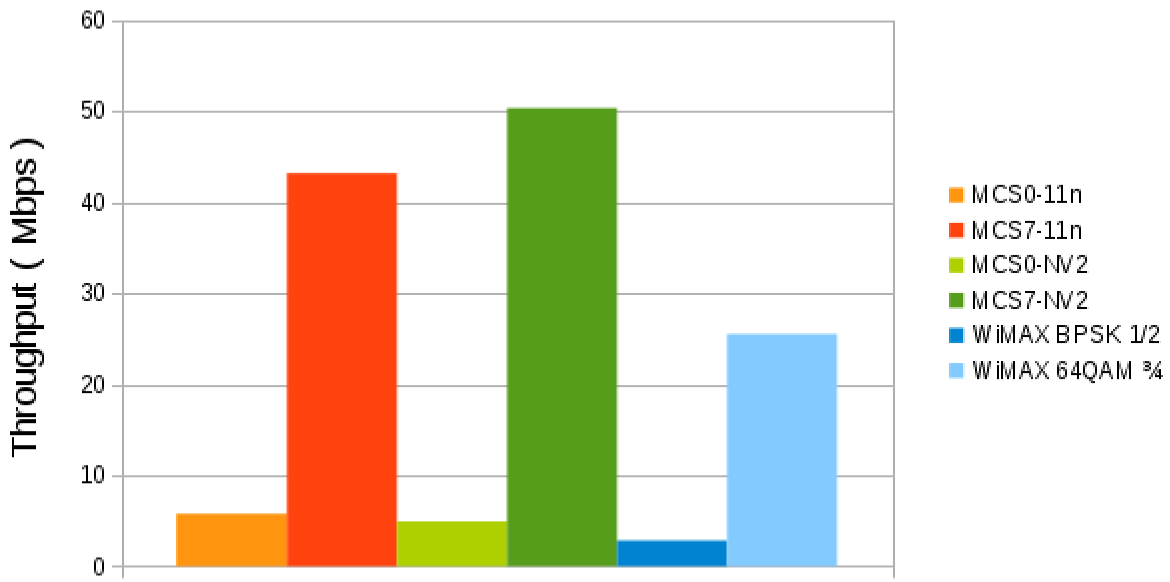

Figure 71: Maximum Throughput achieved for different technologies using SISO techniques. NV2 and 802.11n use 20MHz and Long GI. WiMAX uses 2.5 ms frame duration and CP=1/4.

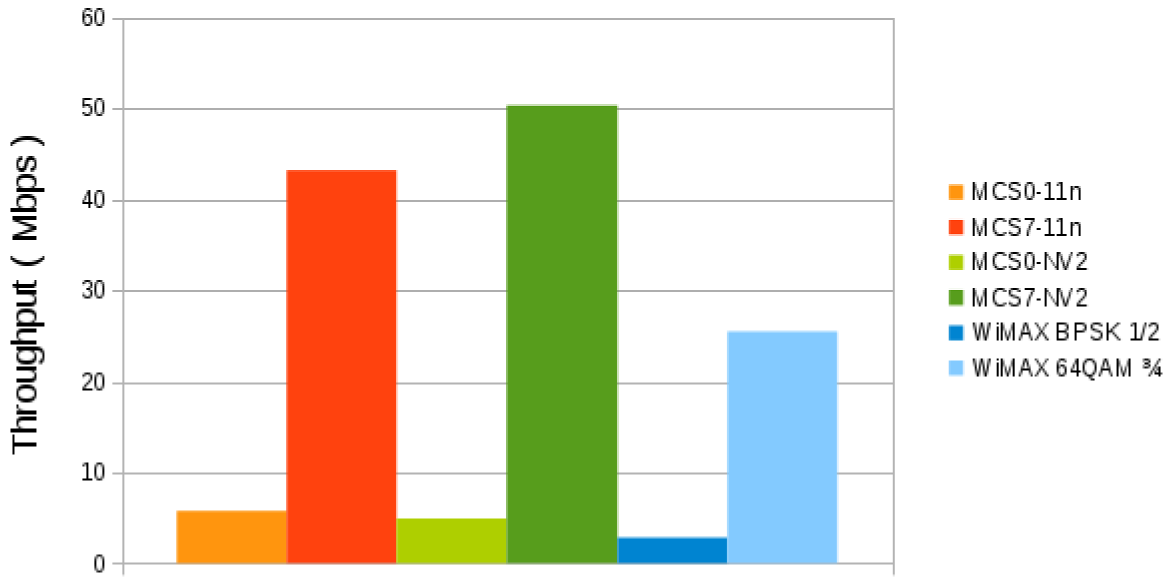

Figure 72: Maximum Throughput achieved for different technologies using SISO techniques. NV2 and 802.11n use 20MHz and Long GI. WiMAX uses 2.5 ms frame duration and CP=1/4

Figures 73 and 74 show the same configuration, but using MIMO techniques (2x2). For this reason, MCS8-15 are shown for 802.11n and NV2, and WiMAX is not present because there is not this option available neither in our laboratory nor in our simulator.

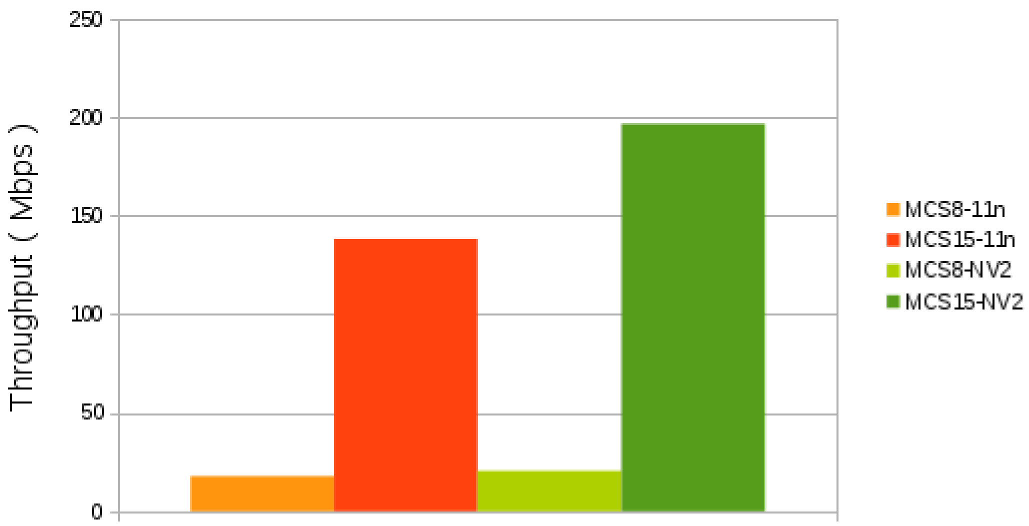

Figure 73: Maximum Throughput achieved for different technologies using MIMO techniques. NV2 and 802.11n use 40MHz and Short GI.

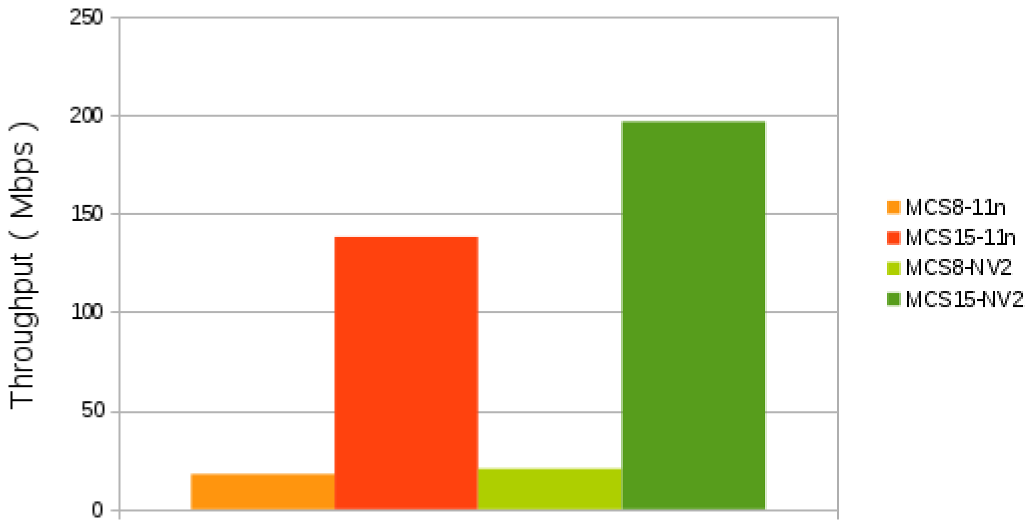

Figure 74: Maximum Throughput achieved for different technologies using MIMO techniques. NV2 and 802.11n use 40MHz and Short GI.

Figures 75-78 show the delay values corresponding to the throughput results shown in the previous 4 figures.

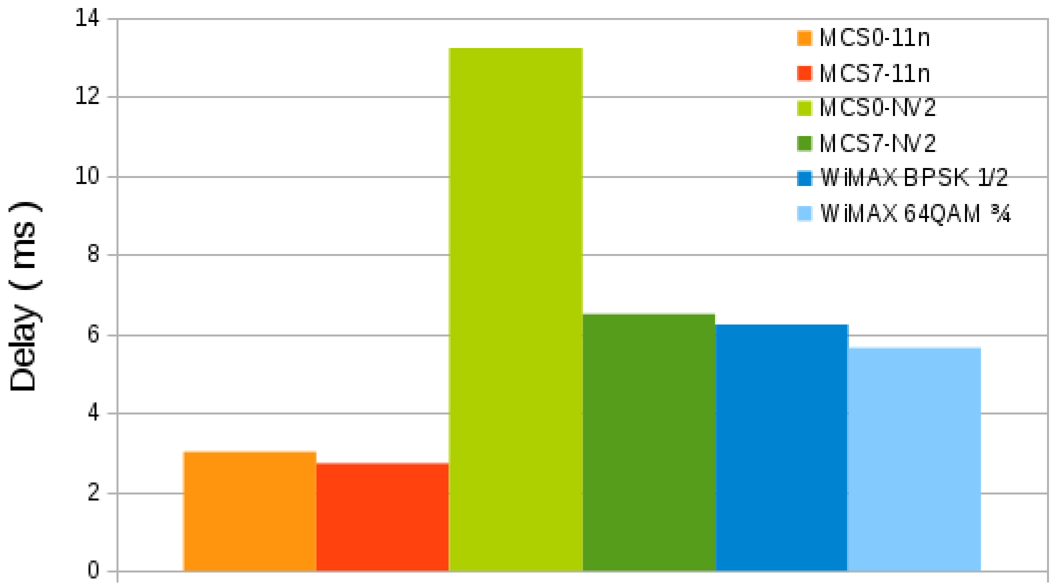

Figure 75: Delay obtained below the saturation point for different technologies using SISO techniques. NV2 and 802.11n use 20MHz and Long GI. WiMAX uses 2.5 ms frame duration and CP=1/4.

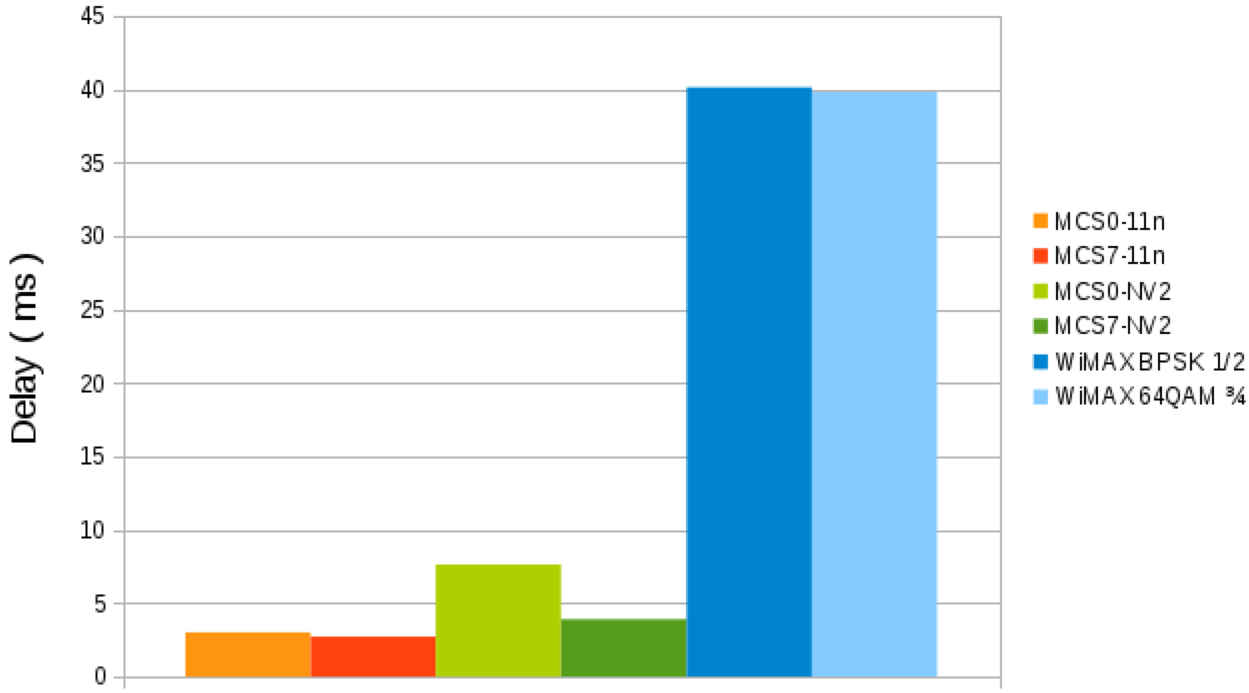

Figure 76: Delay obtained below the saturation point for different technologies using SISO techniques. NV2 and 802.11n use 40MHz and Short GI. WiMAX uses 20 ms frame duration and CP=1/32

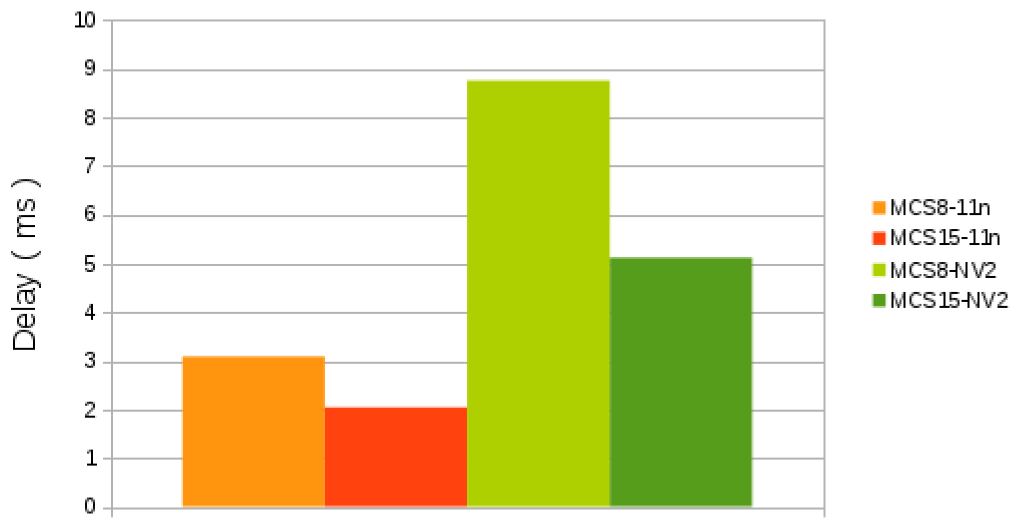

Figure 77: Maximum Throughput achieved for different technologies using MIMO techniques. NV2 and 802.11n use 20MHz and Long GI.

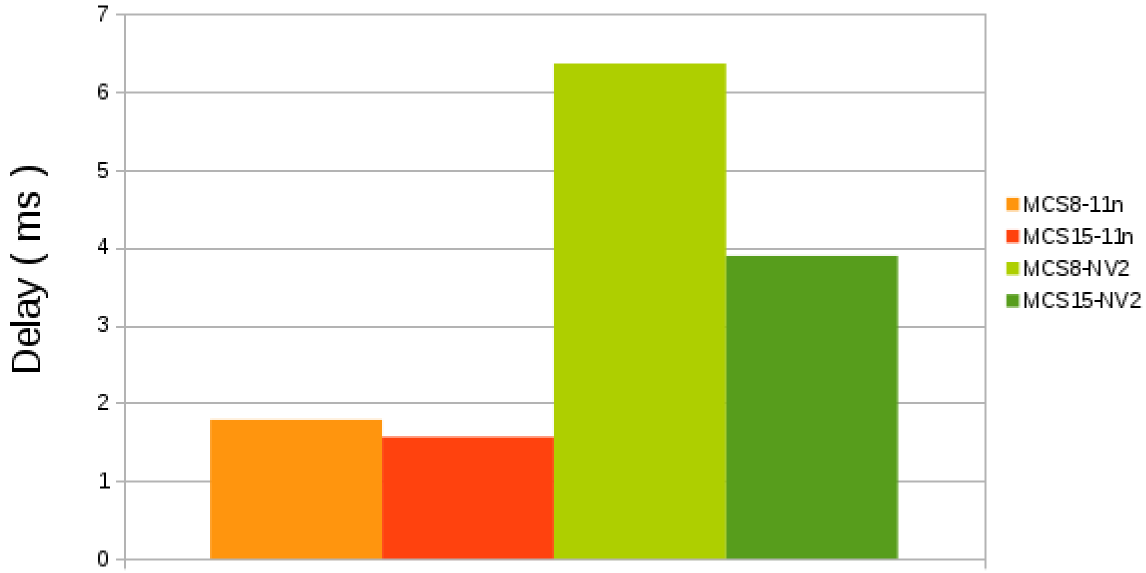

Figure 78: Maximum Throughput achieved for different technologies using MIMO techniques. NV2 and 802.11n use 40MHz and Short GI.

As predicted in the analysis chapters, in short distances 802.11n provides globally the best performance. NV2 shows better throughput but average delays are slightly higher. All these results can be extended for short links up to 5 km.

7.3.2 Performance comparison in long distance links: 802.11n and WiMAX

It has been made clear that 802.11n is the benchmarking solution in the 802.11 family. Satellite communications are clearly complementary to WiFi and WiMAX and it makes no sense to compare the achievable performance in satellite links with the other technologies. Hence, a comparison between 802.11n and WiMAX for long distance links follows.

Figure 79 shows the maximum throughput comparison between 802.11n and WiMAX regarding the distance. For 802.11n, a 20MHz channel bandwidth, long GI and SISO/MIMO techniques have been used. For WiMAX, 20 ms frame duration, CP=1/32 and only SISO techniques have been used (just due to unavailability). No frame aggregation has been used for 802.11n.

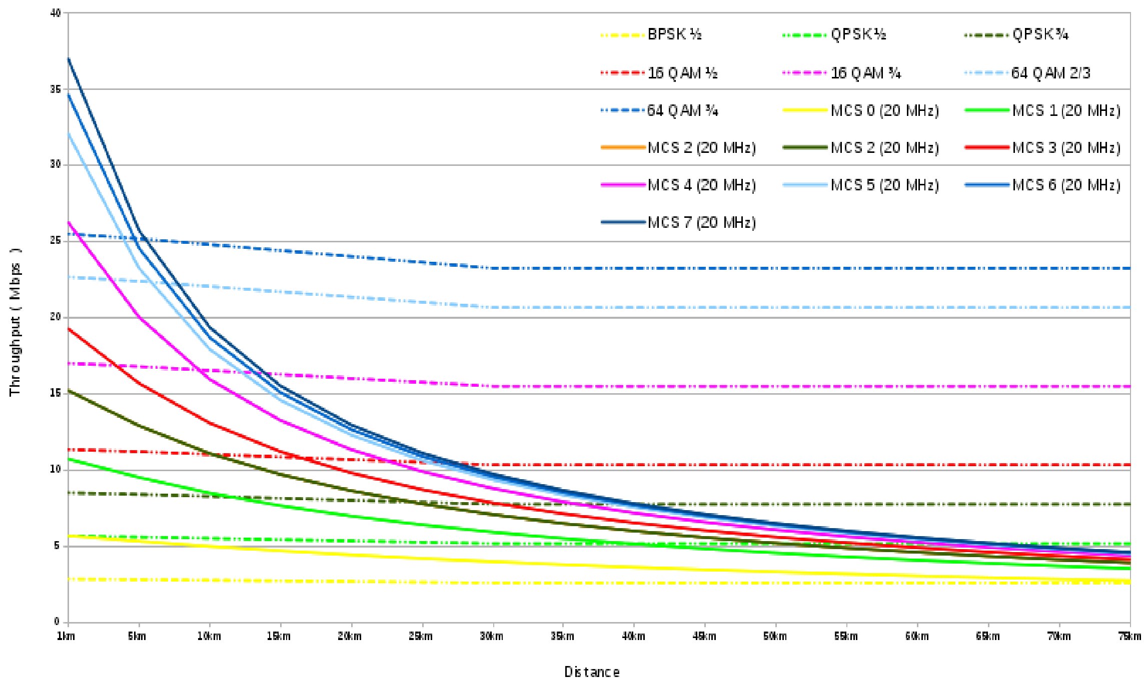

Figure 79: Throughput comparison between 802.11n (No Frame Aggregation, 20 MHz, Long GI) and WiMAX (2.5 ms frame duration, CP= ¼).

Figure 80 shows a similar graph in which frame aggregation is being used in 802.11n. A frame aggregation threshold of 8192 bytes is used (5 frames per A-MSDU). WiMAX now has been set to 20 ms frame duration and CP=1/32.

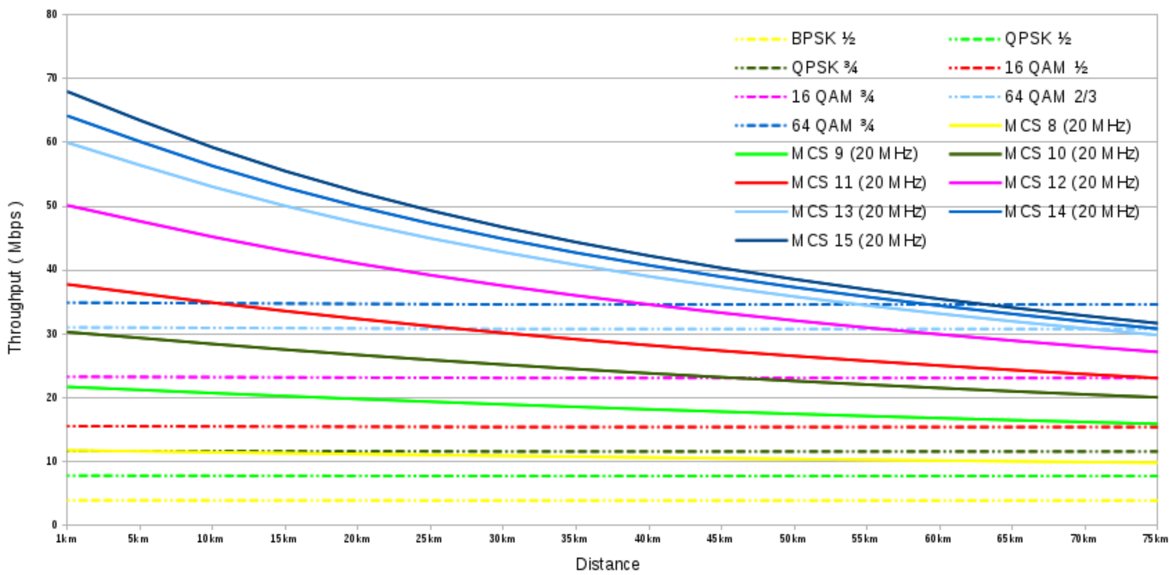

Figure 80: Throughput comparison between 802.11n (Frame Aggregation = 5, 20 MHz, Long GI) and WiMAX (20 ms frame duration, CP= 1/32).

While Figure 79 shows that WiMAX tends to be a better option for long-range links when no frame aggregation is used, Figure 80 shows that 11n overperforms WiMAX when frame aggregation is activated. However, it must been recalled once again that WiMAX is not using MIMO and channels are narrower.

7.3.3 Other considerations

Once again, satellite communications are a fundamental complementary technology when a fully terrestrial solution is not practical or even feasible. In any other case, WiMAX or WiFi will be preferred. WiMAX should be chosen when it maximum capacity for a given scenario is not a limiting factor, as the performance achieved will be eventually better. For the rest, 802.11n may be used as a default solution and NV2 or similar can be used instead when possible to improve the performance in mid-to-long-range links. For more detail information on the performance offered by either technology, the reader is encouraged to visit the specific chapter.

7.4 Comparing the QoS support

Either WiFi or satellite systems will be supporting QoS in a statistical fashion and strongly complemented by the advance traffic management at the IP layer. WiMAX provides the best native QoS support. This is basically the main clear advantage of WiMAX over the other alternatives.

In WiFi and satellite connexions, the use of a complementary admission control system is important to avoid the consequences of link saturation on the prioritized traffic.

8 CONCLUSIONS

Rural 3G-4G access neworks may use a combination of WiFi, WiMAX and satellite links as the backhaul. Capacities of several Mbps or tens of Mbps are possible for all the three technologies with different behaviours:

- Satellite systems may offer high bitrates but at very high prices. Delays may not be reduced under 500 ms (if geostationary satellites are used). Hence, VSAT links are the obvious option when the direct terrestrial link (in one or may be two hops) from the city to the first location to be served is not feasible. In case of using VSAT systems, SCPC systems are preferred in the long term unless a few small locations may share the bandwidth in the terms described in the introductory section.
- WiMAX may always offer quality of service an efficient high bandwidth to any distance for which the link budget admits the use of the best modulation-coding schemes. The effect of the distance on WiMAX performance is almost negligible except for the link budget. Another advantage of WiMAX is that the operator has the option of licensed bands if unlicensed bands may not be used for backhaul purposes. Although the increment of cost due to the licensed bands has not be specified in this document because it is very country-dependent, WiMAX is always a possible option for either one-hop or multihop backhaul of rural femtocells.
- WiFi hardware has also be proven to be an option. 802.11n may be used with frame aggregation for high throughputs and relatively low delays at long distances. Proprietaryalternate MAC solutions on top of WiFi hardware such as Mikrotik NV2 has been proven to be even better for very long distances. Moreover, 802.11n options are the cheapest and the most flexible. The only important conditions for considering 802.11n as the backhaul solution (or as part of it) are:
    - The complementary use of IP for traffic conditionning
    - The use of an access control system that limits the offered load and avoids that the link operates close to (or beyond) the saturation point, in order to keep the delay and the jitter low, and the packet-loss negligible.

Hence, these technologies are constructive elements that can be considered in a future work for a heterogeneous rural transport network that may act as the backhaul of rural access networks.