INSTITUTO SUPERIOR TÉCNICO
Universidade Técnica de Lisboa

Performance Evaluation of a Wireless Mesh Network in a Residential Scenario

Emanuele Tido’

Dissertation submitted for obtaining the degree of

Master in Electrical and Computer Engineering

Jury

Supervisor:	Prof. Luís Manuel Correia
President:	Prof. António Topa
Members:	Prof. António Rodrigues
	Prof. Luís Manuel Correia

March 2008

To Mariarosaria, that continues to fight for her life.

Acknowledgements

This thesis is the final result of more than half year of hard work. It was only possible to complete this work as desired, with the support of some people to whom I would like to thank.

Starting by Prof. Luís M. Correia, I would like to thank him as the supervisor of this work, that gave a great and decisive contribution for the development of the work. He was always available when I needed to talk to him, when I needed some input or advice. To Instituto de Telecomunicações, I would like to thank for the support that it always provided me, starting by the working space and all conditions to finalise this work. Also, all colleagues that I have there, I give special thanks: Lúcio Ferreira, Martijn Kuipers , Mónica Antunes and Daniel Sebastiao.

To professor Fabio Graziosi that gave me the opportunity to conclude my study in Portugal, and to all my friends in Italy that have accompanied me for some years now with special thanks to Fabio, Piergiuseppe, Enrico, Claudio, Simone and Salvatore that have always supported me to go one step ahead.

I thank my parents that encourage me to finalise my academic goal in Lisbon. Finally, I thank Lisbon for her courtesy, and Leo for all the emotions that he can transmit playing, with one guitar and one leg he can fly.

Thank you all. Without your help, I would not be writing this.

Emanuele Tido'.

Abstract

Internet and multimedia services are widely-provided in All-IP based networks. There has been a growing demand for users to get these services. The Wireless Mesh Network (WMN) extends coverage range and provides Internet connectivity to users with low upfront investment. In WMN, the traffic goes through the gateway, making the common destination of traffic.

The goal of this work was to analyse capacity and performance for a WMN in a residential scenario The approach in this work was to start with an easy scenario, and going forward analyzing scenarios increasingly close to a real WMN for a residential scenario. The OPNET Modeler simulation tool was used.

The simulation results show that if the network is multi-hop, users that are until three hops far to the gateway can use with acceptable quality the real-time services and the users that are more far can not use the same services. So, using 802.11 in a multi-hop WMN, the network is exposed to the unfairness problem. The 802.11s group is studying a new protocol solution to solve this problem, the solution being expected in 2008.

Keywords

IEEE 802.11, WMNs, Multi-hop, Simulation.

Riassunto

Internet e servizi multimediali sono diffusi in tutte le reti basate su IP. Crescente è la domanda degli utenti per accedere a tali servizi. Wireless Mesh Network (WMN) estende la copertura e fornisce connessione a Internet per gli utenti con un basso investimento. In una WMN il gateway è la destinazione comune del traffico.

L'obiettivo di questo lavoro è quello di analizzare la capacità e le prestazioni di una WMN per uno scenario residenziale. L'approccio di questo lavoro è quello di iniziare con uno scenario molto semplice e andare avanti e analizzando scenari sempre più vicino a una reale WMN per uno scenario residenziale. OPNET Modeler è lo strumento di simulazione utilizzato.

I risultati della simulazione indicano che, se la rete è multi-hop gli utenti che sono fino a tre hops lontani dal gateway possono utilizzare con qualità accettabile i servizi real-time, gli utenti che sono più lontani invece non possono utilizzare gli stessi servizi. Quindi, utilizzando 802,11 in un multi-hop WMN si espone la rete al problema dell’unfairness. Il gruppo di lavoro dell’802.11s sta studiando un nuovo protocollo per risolvere questo problema, la soluzione è attesa per il 2008.

Parole-chiave

IEEE 802.11, WMNs, Multi-hop, Simulazione.

Table of Contents

List of Figures

List of Tables

List of Acronyms

AC	Access Category.
ACE	Application Characterisation Environment.
ACK	Acknowledgment.
AP	Access Point.
BSS	Basic Service Set
CA	Collision Avoidance.
CCA	Clear Channel Assessment.
CCK	Complementary Code Keying.
CD	Collision Detection.
CFP	Contention-Free Period.
CP	Contention Period.
CSMA	Carrier Sense Multiple Access.
CTS	Clear To Send.
CW	Contention Window.
DCF	Distributed Coordination Function.
DIFS	DCF IFS.
DSSS	Direct Sequence Spread Spectrum.
EDCA	Enhanced Distributed Access.
Email	Electronic mail.
FTP	File Transfer Protocol.
HTTP	Hyper Text Transfer Protocol.
HWMP	Hybrid Wireless Mesh Protocol.
IBSS	Independent BSS.
IEEE	Institute of Electrical and Electronics Engineers.
IFS	Interframe Space.
IP	Internet Protocol.
LAN	Local Area Network.
LoS	Line of Sight.
MAC	Medium Access Control.
MAP	Mesh Access Point.
MR-WMN	Multi-Radio WMN.
NAV	Network Allocation Vector.
NIC	Network Interface Card.
OFDM	Orthogonal Frequency Division Multiplexing.

OLPC	One Laptop For Child.
OPNET	Optimum Performance Network.
P2P	Peer to Peer.
PC	Point Coordinator.
PCF	Point Coordination Function.
PDF	Probability Density Function.
PHY	Physical.
PtMP	Point to Multi-Point.
QoS	Quality of Service.
RF	Radiofrequency.
RTP	Real-Time Transport Protocol.
RTS	Request to Send.
SIFS	Short IFS.
TCP	Transmission Control Protocol.
TDMA	Time Division Multiple Access.
TGs	802.11s Task Group.
UDP	User Datagram Protocol.
UP	User Priority.
VoIP	Voice over IP.
WDS	Wireless Distribution System.
WiMAX	Worldwide Interoperability for Microwave Access.
WLAN	Wireless LAN.
WMN	Wireless Mesh Network.

List of Symbols

C_{AP}	Coverage of a single access point.
C_{Total}	Total area to cover.
CW_{max}	Contention Window maximum value.
CW_{min}	Contention Window minimum value.
d	Distance between antennas.
d_0	Reference distance.
D_{COD}	The time necessary to encode binary data to form a two-level signal.
D_{COM}	The time necessary to compress a voice packet.
$D_{DE\text{-}COD}$	The time necessary to decode a two-level signal to form binary data.
$D_{DE\text{-}COM}$	The time necessary to de-compress a voice packet.
D_{E2E}	End-to-end delay.
D_{NET}	The time to transmit a packet.
G_{TX}	Gain of the transmitting antenna.
G_{RX}	Gain of the receiving antenna.
L	Path loss.
L_0	Path loss for free space propagation.
N	Number of users.
N_{AP}	Number of access points.
$N_{call/h}$	Number of calls per hour.
N_{CH}	Number of channel.
N_{VoIP}	Maximum number of simultaneous VoIP sessions.
N_{user}	Number of users per hour.
n	Path-loss exponent.
P_b	Probability of blocking.
P_{RX}	Power available at the receiving antenna.
P_{TX}	Power to the transmitting antenna.

R	Transmission Range.
R_I	Interference Range.
Rbss	BSS data rate.
T_{ACK}	The time to transmit an ACK packet.
T_{AIFS}	The AIFS time.
T_{BO}	The average time of backoff.
T_{call}	Time duration call.
T_{IP}	The time to transmit a header IP.
T_{MAC}	The time to transmit a header MAC.
TMT	Theoretical Maximum Throughput.
T_{PAY}	Time to transmit the payload.
T_{PHY}	The time to transmit a header PHY.
Tr	Total Erlang Traffic.
T_{SIFS}	The SIFS time.
Tu	User Erlang Traffic.
x	Payload expressed in byte.
ε	Coefficient of interference.

List of Programmes

OPNET Modeler	Discrete Event Simulator, implementing all the basic concepts of an objects programming language. Systems are described in terms of objects, which are instances of models (the OPNET equivalent to classes). There are a vast number of already implemented models addressing several technologies, protocols and commercially available equipment from various suppliers. They provide a user with all the necessary means to develop a complete description of a communication network or an information system.
Microsoft Visual C++ 6.0	Microsoft Visual C++ is a programming environment used to create computer applications for the Microsoft Windows family of operating systems.
Microsoft Excel 2002	It is an electronic spreadsheet program. It is useful to record, to analyse, and to show information. It is also helpful in computing formulas.
Microsoft Office Visio 2007	Visio is a tool for creating all manner of business diagrams, network layouts, storyboards and site flows, software entity relationship diagram, etc.

Chapter 1

Introduction

This chapter gives a short overview of the work, presenting the motivation, current state of the art, and novelty of the work. At the end of the chapter, the work structure is provided.

In recent years, there has been an explosive increase in technology. Computers, Internet, and cell phones have become common household words. With this increases in technology, wireless networks brought a drastical change to our world. Today, it is possible to have freedom to connect to the Internet almost anywhere and anytime, without the use of a wired link. Wireless networks, as the word implies, do not contain a physical medium of connect such as wired ones do. Many of the protocols in wireless networks have been taken straight from the protocols used in wired ones with some modifications to make them work with wireless networks.

More than half of the population in the world lives in rural isolated areas [AkWW04], without any type of terrestrial telecommunications network, which is the subject of Digital Divide. This term [WIKI03] refers to the gap between those people with effective access to digital and information technology, and those without access to it. It includes the imbalances in physical access to technology and hardware, as well as the imbalances in resources and skills needed to effectively participate as a digital citizen. In others words, it is the unequal access to some sectors of the community to information and communications technology, and the unequal acquisition of related skills.

Therefore, there is a need to implement the technology of cheap wireless networks. In fact the idea of scenarios as isolated villages, third world countries and municipalities, is to install equipment so that the network is self-organising, characterised by one or more intermediate nodes in a P2P (Peer to Peer)architecture with the possibility of redundant routes, and through a multi-hop to internet access.

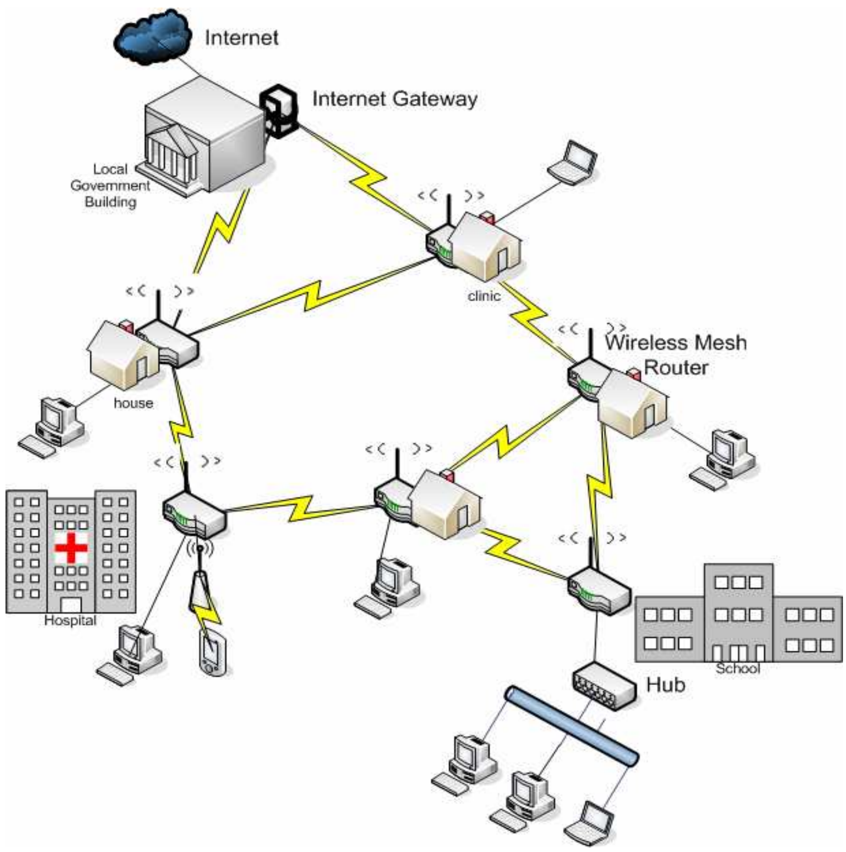


Figure 1.1 WMN Digital Divide solution [AkWW04].

A project to alleviate the problem of Digital Divide in the world is the OLPC (One Laptop For Child) [OLPC07]. It is a project for a laptop computer at low cost (100 $) in order to give to every child in the world access to knowledge and modern forms of education. One of the challenges of this project is to build a network and access to the Internet using these laptops. In fact, if only one laptop has an Internet connection, for example, the others can get online, too, via a the mesh network.

The rapid diffusion of wireless technologies in the recent years, particularly Wireless LANs (WLAN) 802.11, led to the development of new applications and usage scenarios [Whit00]. The Wireless Mesh Network (WMN) arises from a natural evolution of WLAN, completely replacing the system distribution among the nodes of the network by wireless technology. Unlike ad hoc multi-hop networks, which have found little practical interest in the consumer market, the WMN extension emerges as a low-cost flexible wire less infrastructure, finding application in a wide range of situations.

This architecture is regarded as a promising solution, and, although it is still not mature, the early stages of development, much academic work was born and various mesh products are already on the market. The technology on which it tends to do more testing is obviously IEEE 802.11, as well as a significant capillary market, a wide range of applications and tools of development.

The open source community has developed, over the years, many applications and tools solution based on IEEE 802.11 standards, which often represent a key resource for the development of prototypes of new solutions. Research in the field of WMN started from the review of existing protocols for the current IEEE 802.11 wireless networks, in particular ad-hoc multi-hop ones, trying to adapt to the needs of mesh networks. IEEE 802.11 provides for now the possibility of building WLAN networks with Wireless Distribution System (WDS), but missing all those services that make the WMN, like auto-configurability multi-hop and forwarding; it follows that, in most of the existing proposals, all these basic services are implemented at three stacks of ISO/OSI, which often makes the various architectures not interoperable.

The products existing on the market are islands of proprietary technologies, although they may be suitable to particular limited scenarios, having the disadvantage of not being interoperable with other solutions.

In order to allow an efficient development of WMNs based on IEEE 802.11, and ensure interoperability of the various solutions, IEEE began a process of standardisation that aims at establishing a basic framework for the development of WMNs, introducing the basic functionality as a logical extension of the 802.11 MAC. This allows the development of additional services, owners may be transparent to higher levels of the ISO/OSI stack, ensuring interoperability with other solutions.

WMNs have the following three distinct features compared to Ad Hoc Network:

- Unlike in pure Ad Hoc Networks, where nodes can have high mobility, the positions of nodes in a WMN are fixed.

- While traffic in a pure Ad Hoc Network can be between an arbitrary pairs of nodes, in a WMN all traffic is either to or from a designated gateway, which provides access to the Internet.

- WMN nodes are expected to be powered, and thus energy consumption is not a significant concern.

The unique topology of WMNs leads to many strong points that are hard to find in other access network technologies. WMNs provide a new way to access the Internet to potential users who are currently not serviced due to geographical, financial or technological restrictions.

The most popular application for WMNs is the Internet access service. Essentially, a WMN needs to provide Internet services for residential areas or businesses. In such a situation, though data services make primary service over a WMN, voice services such as Voice over Internet Protocol (VoIP) are also important.

Users and Internet service providers are expected to benefit from WMNs in many ways:

- **Scalability:** the network can grow as more and more customers are added. If capacity becomes an issue, additional gateways can be added.

- **Low upfront investment:** incremental expansion is possible in WMNs, which eliminates the challenge of huge initial investment required in deploying wired Internet access services, e.g. DSL or cable-based networks.

- **Reliability:** the mesh structure assures the availability of multiple paths for each node in the network, if a node fails, others will take over its traffic enabling uninterrupted service.

- **QoS:** Quality of Service (QoS) can be offered to users if the network is designed carefully, and enough gateways are placed at appropriate locations.

- **Market coverage:** direct Line of Light (LoS) is not required between a user node and the gateway, which leads to a larger number of potential users compared to an infrastructure-based wireless access.

- **Flexibility:** in mesh networks, nothing is imposed in terms of architecture, and an infinite number of configurations can be adapted to meet the requirements of a particular situation. For example, a high bandwidth backbone (802.11a) can be deployed at the core of the network, and medium bandwidth (802.11g) can be used at the edges.

- **Roaming service:** as customers travel, roaming contracts between companies can be agreed upon for added value services.

- **Geolocation:** using the communication constraints and received signal strength measurements, a GPS-less geo-location feature can be implemented within the coverage area.

The aim of this thesis is to study the performance of a WMN in a Residential Scenario, composed mostly of two floor residential houses, surrounded by streets and trees. This scenario extends itself over an half square kilometres covered by a backhaul network. Each Mesh Access Point (MAP) covers a specific area, where small communities of residential users use a set of services. The number of users is relatively low, and the usage of services is not intense, high traffic load levels not being expected. This WMN scenario will be characterised with real time and non real time services. Their performance is evaluated, considering several characteristics of the scenario (number of users per MAP, number of MAPs, number of gateways, and number of hops), using the Optimum Performance Network (OPNET) Modeler Wireless Suite simulation tool.

The motivation of this work is looking if Wireless Mesh Networks (WMNs) are a good Digital Divide solution. Today the challenge is connection to anyone anywhere with a cheap solution, so WMNs are particularly important because the people in general have only a little access to information and have so much need to information and is necessary to give them connectivity.

WMNs are an emerging two-tier architecture based on multi-hop transmission, with two fundamental objectives: to offer connectivity to end-users, and to form a self-organised wireless backbone. Each MAP covers a region, offering connectivity via Access Points (AP), building a WLAN sub-network that enables access to users that communicate by means of the wireless backbone.

The present study aims at evaluating the impact of capacity and performance at a global perspective in a WMN, specifically, the impact of the following parameters:

- Communication Range – To establish for each data rate how far a MAP can stay away of the other.
- WLAN throughput – To establish the maximum throughput available at MAC layer for two nodes network.
- Average node throughput at application level – To evaluate the decrease in throughput with the numbers of hops.
- Voice Packet End-to-End Delay – To evaluate how a call far can go,i.e., how many hops away to the gateway can a user stay to have an acceptable quality call.
- Number of available calls – To establish the maximum number of simultaneusly calls for each hop.

The thesis is organised into five chapters. In Chapter 2, one describes all the aspects of IEEE 802.11 standards that are relevant to the study, addressing two-tier mesh networks and what are the services used in a WMN. In Chapter 3, one describes the parameters that influence the capacity and performance in a WMN, how the WMN was implemented in the OPNET, and how was solved the drawback present in the simulator for this work and a brief description of the setting parameters of the applications used. In Chapter 4, the analysis of WMNs capacity is carried out for various topology models.

In Chapter 5, conclusions aree drawn, and some suggestions for future work are presented. Finally, annexes present some useful information, like the detailed description of all applications and scenarios needed to define them by using Modeler, and the definition of all performance metrics used in this work.

Chapter 2

Wireless Mesh Networks

This chapter provides an overview of WMNs, mainly focussing on capacity performance aspects.

2.1 The Wi-Fi system

2.1.1 IEEE 802.11 Wireless Local Area Network

The IEEE 802.11 standard [WaMB06] defines two types of networks:

- Independent Basic Service Set (IBSS).
- Basic Service Set (BSS).

IBSS includes a number of nodes or wireless stations that communicate directly with each other on an ad-hoc mode, in which no single node is required to function as a server or coordinator. Generally, an ad-hoc implementation covers a limited area and is not being connected to any larger network.

BSS consists of at least one AP connected to the wired network infrastructure, and a set of wireless end stations. Corporate WLANs require access to the wired LAN for services, they operate in infrastructure mode, and count on an AP that works as the logical server for a single WLAN or channel. In a communication between two nodes, *A* and *B*, flow from node *A* goes to the AP and then from the AP to the node *B*.

Original 802.11 wireless protocol supported 1-2 Mbps transfer rate in the 2.4 GHz frequency band. More extensions have since then been defined. The IEEE 802.11a standard [IEEE03], defines a series of new modulation methods that enable data transmission rates up to 54 Mbps. The higher data rates are obtained by the use of Orthogonal Frequency Division Multiplexing (OFDM), a technique in which the frequency band is divided into subchannels that are individually modulated. The IEEE 802.11a standard defines operations in the 5 GHz frequency band.

This means equipment supporting the standard is not backward-compatible with the basic 802.11 standard, because that standard defined operations in the 2.4 GHz frequency band. In addition, because high frequencies attenuate more rapidly than low ones, 802.11a stations have a shorter range than those operating in the 2.4 GHz band. This requires an organisation to deploy more APs to obtain a similar geographical area of coverage than would be required via the use of APs operating in the 2.4 GHz band, but provides a higher throughput.

The second extension to the basic IEEE 802.11 standard is the 802.11b standard [IEEE03]. Under IEEE 802.11b , DSSS is used with two new modulation methods to provide a data transfer rate of 11 Mbps and 5.5 Mbps. The 802.11b standard also provides compatibility with 802.11 DSSS equipment operating at 2 or 1 Mbps. To provide this compatibility, the IEEE 802.11b standard specifies the use of the 2.4 GHz frequency band, having available a bandwidth of 83 MHz. The 802.11a standard provides a higher data transfer rate, but since it is used for the 5 GHz frequency band results in a shorter transmission distance. Similarly, in a reverse manner, the IEEE 802.11b standard provides a greater transmission distance but lower data rate than obtainable from the use of 802.11a compatible equipment.

By combining the modulation method used in the 802.11a standard with the frequency band employed by the 802.11b standard, the IEEE provided a mechanism to extend both the data rate and transmission range of WLANs, resulting in the 802.11g standard [VaKo05]. To provide backward compatibility with the large base of 802.11b equipment, the 802.11g standard also supports DSSS operations at 11, 5.5, 2, and 1 Mbps and it uses the 2.4 GHz frequency band like 802.11b. Thus, the relatively new IEEE 802.11g standard can be considered to represent a dual standard because it provides 802.11b compatibility.

IEEE 802.11e is an approved amendment to the IEEE 802.11 standard that defines a set of Quality of Service (QoS) enhancements for WLAN applications, through modifications to the Media Access Control (MAC) layer. The standard is considered of critical importance for delay-sensitive applications, such as Voice over IP (VoIP) and Streaming Multimedia. It is describe more in detail in Section 2.1.2.

- The 802.11n standard [Xiao05] is a proposed amendment that builds on previous 802.11 standards by adding Multiple-Input Multiple-Output (MIMO). The expected typical throughput will be 200 Mbps, 10 times higher than that of 802.11g. It will offer better operating distance and full compatibility with current WLANs.

Table 2.1. PHY specifications (extracted from [WIKI03] and [CiSC03]).

Standard	**802.11b**	**802.11a**	**802.11g**	**802.11n**
Date	December 1999	January 2000	June 2003	September 2007
Modulation	DSSS,CCK	OFDM	DSSS,CKK,OFDM	DSSS,CKK,OFDM
RF band [GHz]	2.4	5	2.4	2.4 and 5
Channel Bandwidth[MHz]	20	20	20	20 or 40
Number of channels	11	52	11	/
Number of non-overlapping channels	3	19	3	12
Typical indoor Range [m]	38	35	38	70
Typical outdoor Range [m]	140	120	140	250
Max. Power [mW]	100	1000	100	50
Max Throughput [Mbps]	11	54	54	540
Typical Throughput [Mbps]	6.5	23	19	200

This standard is not expected to be approved before Sep. 2008, Table.2.1 shows the technical parameters, other existing extensions to IEEE 802.11 are listed below:

- IEEE 802.11k [MaBe05] is a proposed amendment for radio resource management. It defines and exposes radio and network information to facilitate the management and maintenance of a mobile WLAN.
- IEEE 802.11r [QSBB05] is a proposed amendment to allow connectivity aboard vehicles in motion, with fast handovers from one base station to another managed in a seamless manner.

IEEE 802.11s [HMZD07] is a draft amendment for mesh networking, defining how wireless devices can interconnect to create a mesh network, however, the standard is not ready and not considered in this thesis, since with the current standard it is possible to build a WMN, although less performing.

In the following section, one describes in detail the MAC contention access mechanism, of interest for this work.

2.1.2 Medium Access Control

In a WLAN, each terminal is unable to listen to all ongoing transmissions due to the limited coverage (hidden station problem). Thus, it is not possible to use Collision Detection (CD), but it is necessary to use Collision Avoidance (CA). Before the transmission of a frame, the MAC Layer entity has to get access to the medium. This access is administrated with a Distributed Coordination Function (DCF) or a Point Coordinated Function (PCF).

In DCF, the access to the medium is based on the Carrier Sense Multiple Access (CSMA) with CA technique. When the medium is sensed free, the transmitter station waits for a DIFS (Distributed Inter Frame Space), then, it starts the phase of contend for the medium usage (Contention Window), which at the start is set to a minimum value CWmin (15 in 802.11a). In particular, the station that senses the medium is free for the DIFS time (34 μs in 802.11a), waits for a casual time (backoff time), and at the end of it, if the medium is still free, the station can send the transmission request called Request to Send (RTS). The backoff time reduces the probability of collision when at the end of transmission there are many stations that are waiting to transmit. Each station determines the duration of this random time individually, as a multiple of a slot time (9 μs in 802.11a). Stations that deferred medium access, because of detecting the medium as busy, do not select a new random backoff time, but continue down-counting the time of the deferred backoff after sensing the medium as idle again. The receiving station picks up the RTS and, if it is available, answers with an authorisation to transmit called Clear to Send (CTS). To send the CTS, the station waits for a SIFS (Short Inter Frame Slot) shorter than DIFS, so there is no collision with the stations that are waiting. The transmitter station receives the CTS, and waits for a SIFS before sending its frame message. The receiving station picks up the message and sends an acknowledge (ACK) message, Figure 2.1.

Whenever the station fails its transmission (i.e., the transmitted data frame is not acknowledged), it increases its CW. After any unsuccessful transmission attempt, a new backoff procedure is performed with double-sized CW, up to a maximum value defined by CWmax (1023 in 802.11a). The CW is now larger than before to reduce the probability of repeated collisions if there are multiple stations attempting to access the medium.

Each station can listen to the common medium, and the Carrier Sense can be physical or virtual. The first is realised in the physical layer, by the Clear Channel Assessment (CCA) function, but it does not listen to the hidden stations. Thus, WLANs adopt the virtual carrier sense realised with the Network Allocation Vector (NAV), which is an indicator of the time in which transmission is not possible, even though the CCA function signals that the channel is "clean", being contained in the header of the packets RTS and CTS. With NAV, the transmitter station communicates to the other stations that the channel will be busy for a time interval, the other stations receiving the time NAV will count down from the NAV to zero before transmitting.

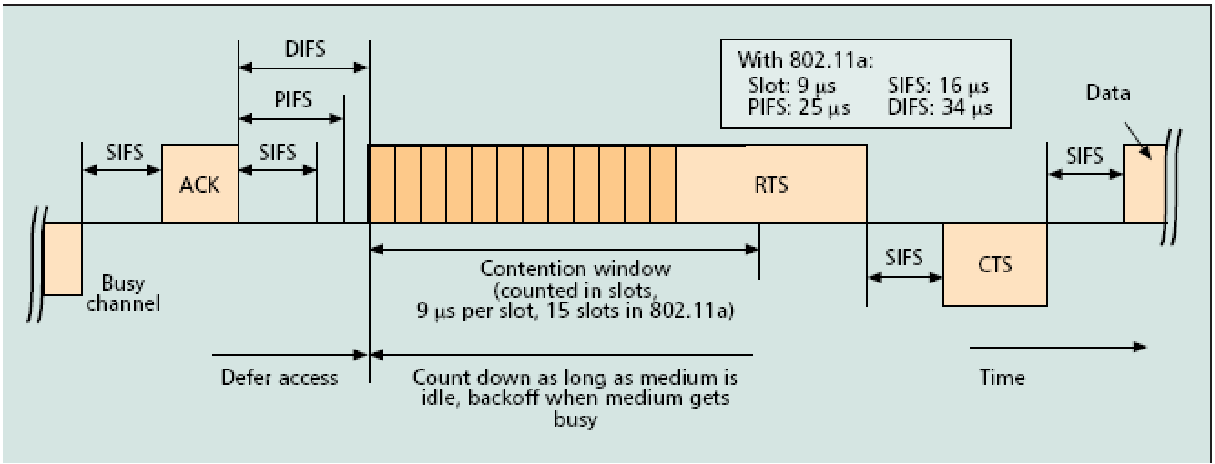


Figure 2.1 **-** Interframe spaces and backoff procedure with random contention window size (extracted from [MaSu03])**.**

In an attempt to support limited QoS, the PCF mode is also available. With PCF, the period after each beacon transmission is divided into two sections: the Contention-Free Period (CFP) and the Contention Period (CP), which together constitute a super frame. The Point Coordinator (PC) (generally assumed to be co-located at the AP) provides guaranteed access to the medium in the beginning of the CFP, by beginning the transmission before the expiration of the DIFS. During the CFP, the PC lets stations have priority access to the medium by polling the stations in a round-robin mode. The CFP is then followed by the CP, during which access to the medium is governed by DCF. In PCF, the PC has no knowledge of the offered load at each station. The PC simply round-robin polls all stations that have indicated the desire to transmit during the CFP. Any station can request to be added to the poll sequence by a special frame exchange sequence during the CP. In general, real-time services represent a problem to CSMA, because it does not guarantee QoS.

The 802.11 standard is characterised by a large overhead of headers, besides the data packet

[FCFN07]. The application level prepares data packet, very small for voice application, to which the layers below (IP, MAC and PHY) will attach their overhead.

A technique that allows for a guaranteed QoS, unlike CSMA, is Time Division Multiple Access (TDMA). In TDMA, the base station has the responsibility to coordinate the nodes of the network. The time on the channel is divided into time slots, which are generally of fixed size. Each node of the network is allocated a certain number of slots where it can transmit. Slots are usually organised in a frame, which is repeated on a regular basis, which is very good for real-time applications. Nevertheless, TDMA is not well suited for data networking applications, because it is very strict and inflexible. IP is connectionless and generates bursty traffic, which is very unpredictable by nature, while TDMA is connection oriented (so it has to suffer the overhead of creating connections for single IP packets). TDMA uses fixed size packets and usually a symmetrical link, which does not suit IP that well (variable size packets).

2.1.3 802.11e

QoS is used to indicate the parameters that characterise the quality of service offered by the network (i.e. packet loss, anddelay), or the means to achieve a desired quality of service. The traffic offered to the network and the involvement of mal-functions are stochastic processes, hence, the parameters used to characterise QoS are random variables.

The 802.11e standard [HiKW03] defines the MAC procedures to support applications (i.e., voice, audio, and video) that require specific QoS. It is a necessary mechanism to ensure priority access to latency sensitive multimedia applications, compared with data services as Email or web traffic. The medium itself and its characteristics hidden node problem, co-channel overlap, or even the fact that it is a half duplex medium, increase the difficulty to guarantee the QoS request. In order to indicate that the AP or the stations support QoS, their names change to QoS Access Point (QAP) and QoS Stations (QSTA), which together form a QoS Basic Service Set (QBSS).

The standard is compatible with the legacy 802.11 one, so that QSTA can connect to non-QoS Access Points (nQAP), without using the QoS enhancements. 802.11e, to introduce the QoS implements the Enhanced Distributed Channel Access (EDCA), which improves the DCF. EDCA implements an internal table of priority services. This classification is outlined in the IEEE 802.1D standard and defines 8 different types of traffic,Tab.2.2. The different types of traffic are associated with four different IEEE 802.11e Access Categories (ACs).

For each AC, specific medium access default parameters are given, in order to differentiate access categories, and so give better quality to certain services, compared to others, as shown in Table.2.3. Real-time traffic, such as video and voice, has more aggressive EDCA parameters, to ensure that QoS traffic has a better chance to acquire the medium than the best-effort or background traffics.

When data arrives, the IEEE 802.11e MAC classifies the data with the appropriate AC, and puts it in the corresponding AC transmit queue. Data packets from different AC queues first contend internally among themselves based on the queue of each Arbitration InterFrame Space Number (AIFSN), the contention window, and the random back-off time. The AC with the smallest back-off wins the internal contention.

Table 2.2. User Priority (UP) to AC mapping (extracted from[WaMB06]).

Priority	UP (802.1D)	AC (802.11e)	Traffic Type
Lower	1	0	Background
|	2	0	Standard
|	0	1	Best Effort
|	3	1	Exellent Load
|	4	2	Controlled Load
|	5	2	Less than 100ms latency and jitter
I I	6	3	Less than 10ms latency and jitter
V Higher	7	3	Lowest latency and jitter

The winning AC then contends externally for the wireless medium. The external contention is similar to the one in DCF. With the tuning of AC parameters provided by IEEE 802.11e, traffic performance from different ACs is optimised and prioritisation of traffic achieved.

Table 2.3. AC medium access default parameters (extracted from [WaMB06]).

AC	min CW	max CW	AIFSN	TXOP limit (802.11b)[ms]	TXOP limit (802.11a/g)[ms]
0	CW_{min}	CW_{max}	7	0.0	0.0
1	CW_{min}	CW_{max}	3	0.0	0.0
2	$((CW_{min}+1)/2)-1$	CW_{max}	2	6.0	3.0
3	$((CW_{min}+1)/4)-1$	$((CW_{min}+1)/2)-1$	2	3.0	1.5

2.2 Basic concepts of WMNs

Typically, a WMN consists of two types of tier architectures: wireless backhaul network and access network, Figure 2.2.

The name of this topology is Hierarchical Wireless Mesh Networks, and it consists of two types of nodes, mesh routers and mesh clients. Mesh routers that typically have a minimal mobility, are connected with each other by wireless, links forming a backbone infrastructure that is named Wireless

Distribution System (WDS), and give to users an access network that is named Basic Station System (BSS). They have all the functionality of routing, auto configuration of the network, and connectivity recovery in the event of failures. In addition, there is at least one mesh router with a gateway functionality, to allow clients access to external networks, such as the Internet, or networks of different technologies.

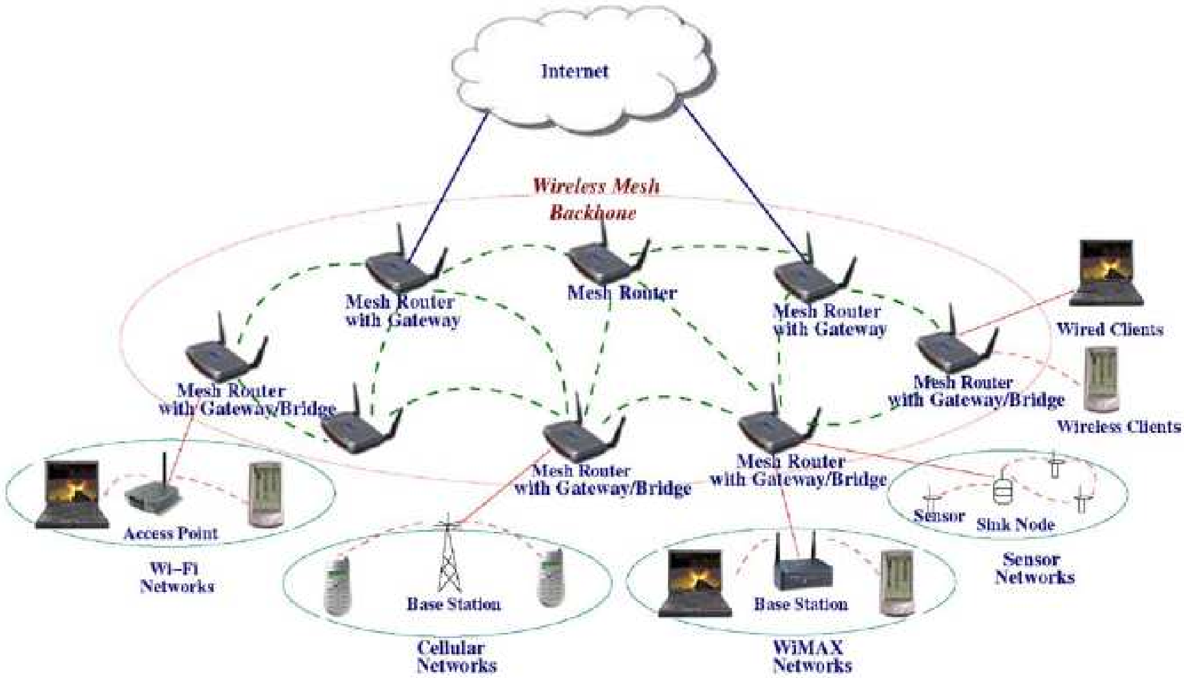


Figure 2.2 BSS and WDS in a WMN [AkWW04].

The backbone can be built by using heterogeneous technologies, such as IEEE 802.11 and IEEE 802.16, interfaced with each other through appropriate gateways and concentrated at the point of connection to cable network. There are two ways to build such architecture. One is to carefully design the layout of the network, such as the exact placement of mesh router and the type of antennas to be used. The other, known as "unplanned", is more spontaneous and random, in the sense that it is not designed as a topology of the network. Obviously in the formor, there are more study design and less flexibility, but it can produce more performance and reliability, because it is known before the links of the mesh network, and it can use additional techniques to boost performance, such as the use of directional antennas. This type of infrastructure is the most common and popular, for its convenience and advantages. In fact, in this scenario the client does not require specific features mesh and can interface directly with the same mesh router radio technology.

A WMN, unlike WLAN, allows to extend network coverage to an area wide enough (residential or campus) precisely because it creates a mesh network. The advantage of this network is that it has a good level of reliability due to the presence of redundant paths and a variety of possible destinations.

Due to the presence of partial mesh topology, a WMN uses multi-hop relaying similar to an ad hoc wireless network. Although ad hoc wireless networks are similar to WMNs, the protocols and architectures designed for the ad hoc wireless networks perform very poorly when applied in the WMNs. In fact, an ad hoc wireless network is generally designed for high mobility multi-hop environment; on the other hand, a WMN is designed for a static or limited mobility environment. In addition, factors such as the inefficiency of protocols, and the scarcity of electromagnetic spectrum further reduce the capacity of a single-radio WMN. WMNs are used for both military and civilian applications. Some of the popular civilian applications of WMNs include provisioning of low-cost Internet services to shopping malls, streets, and cities. In order to improve the capacity of WMNs, and for supporting the traffic demands raised by emerging applications for WMNs, multiradio WMNs (MR-WMNs) are under intense research. Traditional wireless ad hoc networks and WMNs were based on a single-channel or single-radio interface. WMNs, irrespective of its simplicity and high fault tolerance, face a significant limitation of limited network capacity. In fact, with increasing number of nodes in a network, the throughput capacity becomes unacceptably low. One approach to improve the throughput capacity of a WMN is to use multiple radio interfaces.

However, WMN is not the only way to cover vast areas, another technology capable of doing this being WIMAX defined by the standard 802.16. WIMAX, with the complexity of its system, can manage a mechanism reservation of resources that can offer a good QoS. In fact WIMAX supports four types of QoS: Unsolicited Grant Service (UGS) systems for real-time fixed size (eg VoIP), Real-Time Polling Service (rtPS) for real time systems of variable sizes (eg video applications), Not Real-Time Polling Service (nrtPS) flows data tolerant to delay, and Best Effort (BE) for data streams where it is not required minimum level of service. The maximum throughput for WIMAX is 70 Mbps, but it is more expensive than WMNs. It is possible to use together WIMAX and WMN technology [CMZW07], but is not considered in this thesis.

Although the process of standardisation of 802.11s for WMN is not yet finalised, WMNs are already a reality on the ground, which is possible through the use of proprietary networks solutions (e.g., Tropos, and Nortel).

802.11 [Abou06] proposes the introduction of a series of new mechanisms for the installation, configuration and operation of a WLAN mesh. These mechanisms are in the MAC level and can be implemented on the physical level of the whole family of standards, IEEE 802.11a/b/g/n. The proposal introduces an extension regarding the formation of topology, to make a WLAN mesh auto-configurable, dynamic and easy to use. The main feature that distinguishes the is the introduction of a mechanism for routing and forwarding completely at the MAC level, rather than at the network one, creating, in fact, a routing level 2. This mechanism is called Hybrid Wireless Mesh Protocol (HWMP).

The objectives of the specifics presented in the document are:

- Increase the radius of coverage and flexibility of use;
- Performance reliability;

- Security;
- Support for multimedia communications between devices;
- Energy efficiency for devices with consumption limits;
- Compatibility with previous standards;
- Ensure interoperability for networking;
- Opportunities to increase throughput.

However, these standards are not object of this thesis, since with the current standard it is possible to build a WMN, although less performing.

2.3 Services and scenarios

2.3.1 Services

The services [WIKI03] used in WMNs are (VoIP, Email, videoconference, video streaming, web browsing, FTP):

- **VoIP** (Voice over IP) is a technology that makes it possible to make a telephone conversation using an Internet connection or a dedicated network that uses the IP protocol, rather than go through the traditional telephone network.
- **Email** (Electronic mail) is an Internet service through which the user can send or receive messages. It is the most popular Internet application and used currently.
- **Videoconference** is a way to see the interlocutor, to interact with it, to have a control panel where there are the participants and a virtual work space and where all participants can share text, images, tables and other information.
- **Video streaming** identifies a streaming data stream audio / video sent from a source to one or more destinations through a network. These data are reproduced as they arrive at the destination. There are two types of streaming: streaming live and on-demand streaming.
- **Web browsing** is a service that enables a user to display and interact with text, images, videos, music and other information typically located on a Web page at a website on the World Wide Web.
- **FTP** stands for File Transfer Protocol providing the basic features for sharing files between remote hosts.

The services send data to transmit to the level below, the transport layer. The Transmission Control Protocol (TCP) can be roughly classified to the transport layer (OSI level 4) of the OSI reference model. The User Datagram Protocol (UDP) is also a transport protocol packet. TCP and UDP are usually used in combination with the IP protocol, but have different characteristics. Unlike TCP, UDP does not handle the adjustment package or retransmission of those lost. UDP has the characteristic of being an unreliable and connection-less transport protocol, but on the other hand, it is very quick and efficient for applications "light" or time-sensitive. It is often used for the transmission of audio or video information. Since real-time applications often require a minimum rate one, does not want to delay unduly the transmission of packets and it can tolerate some loss of data, the model of TCP service may not be particularly suited to the needs of these applications. UDP provides only the basic services of transport layer: the multiplexing of connections and error checking, while TCP ensures reliable transfer of data, control flow and congestion control. UDP is a stateless protocol: it does not maintain the state of the connection contrary to the TCP, therefore, it has less information to memorise.

802.11 can use both levels of transport, TCP or UDP. The choice depends on the application. TCP, which with each burst adds much overhead, is used for transmissions in which it is essential to receive the files correctly. UDP, on the other hand, is more suitable for real applications (VoIP) where it is more important to receive a continuous flow of data than data without errors.

2.3.2 Scenario

In this section, several examples of mesh scenarios are described that are already a reality for many countries. An interesting project is the WiFi Mesh network in Cuzco (Peru) [WCIT06] that provides to 12 rural health centres with Internet access and VoIP services. Figure 2.3 shows the design of the Cuzco network in a schematic way. There are seven Repeater Stations (actually they are wireless routers) that connect twelve End User Stations. The northern area is Quispicanchis Province (4 End User Stations and 3 Repeater Stations) and the southern one is Acomayo Province (7 End User Stations and 3 repeater).

There are two main radio links that connect these provinces to Cusco City: the first one is a 39 km radio link from Cusco Hospital Repeater to Josjojojahuarina1 Repeater, and the second one is a 42 km radio link from Josjojahuarina1 Repeater to Don Juan Repeater. These two radio links are the main part of the backbone. The Cusco Hospital Repeater is linked to a gateway that connects this network to the Internet and to the public network. Non overlapping channels (1, 6 and 11) are used in the 2.4GHz band, and Point to Multi-Point (PtMP) links involving distances longer than 6 km are limited to 4 stations. When necessary, two or more routers are installed in the same place and connected through an ethernet cable in order to guarantee this. The height of the antennas is calculated for assuring that the first Fresnel zone is clear.

In 2005, WMNs networks were a reality in 248 cities worldwide. Only in the US, for 2010, there will be 1500 cities that will have WMNs. In table 2.4 , one showes the parameters of two mesh projects.

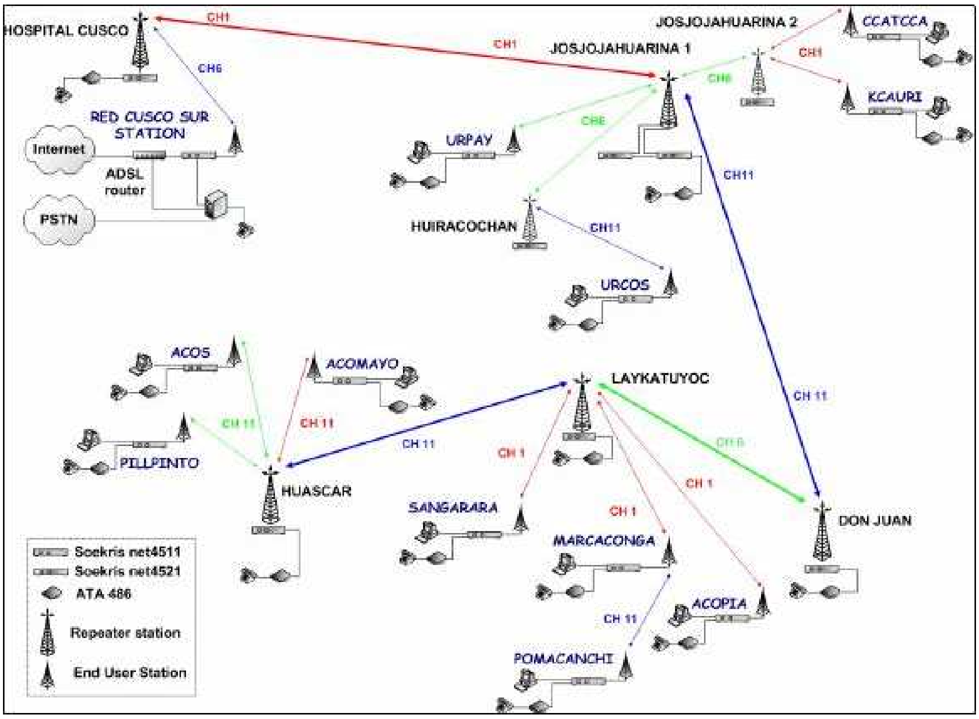


Figure 2.3. Cusco Sur WiFi Network [WCIT06].

Table 2.4. Parameters of network already deployed [Voni07].

City	Area of Network [Km^2]	# of mesh router	# of users	Access users	Access backhaul
Taipei	130	10,000	20,000	802.11b/g	802.11a
Huston	1000	15,000	1,000,000	802.11b/g	802.11a

2.4 State of the Art on WLAN and WMNs capacity estimation

A fundamental topic in WMNs is capacity estimation. Knowledge of network capacity as a function of various parameters is essential in properly provisioning the network.

There are a number of papers on capacity analysis of wireless networks. The most significant results are presented by Gupta and Kumar [GuKu00] and Li et al. [LiBi01]. Gupta and Kumar show for different traffic conditions the capacity behaviour of wireless networks with random located nodes that have a common range and random destinations. In their analysis, two types of networks are used: arbitrary and random networks, where node locations and traffic destinations are arbitrary and random respectively. For both types of networks, two types of wireless transmission reception models are used: protocol and physical models. In the protocol model, a successful transmission is determined based on the ratio of the distances. In the physical model, a transmission is successful when the Signal Interference Noise Ratio (SINR) is greater than a threshold. They used the product of bits and distances as an indicator of transport capacity. Although, they presented asymptotic capacity of wireless networks through rigorous theoretical analyses, their work lacks practical applicability.

Li et al. studied how traffic patterns affect the scalability of static wireless ad hoc networks. While Gupta and Kumar's work suggests that the throughput available to each node approaches zero as the number of nodes increases, Li et al. identified the criteria for the traffic pattern that provides scalable ad hoc capacity. Another contribution of Li et al.'s work is the practical analysis of scalability based on IEEE 802.11 networks. They analysed the capacity of specific topologies: chain and lattice. However, in their analysis of chain topologies, only one-source at the end generates traffic and other nodes forward it. They also considered multiple-source networks, but the destinations were chosen randomly. These scenarios are different from the traffic pattern in WMNs, where user nodes are assumed to be equally active, and user traffic is aggregated at the gateway; however, their definition of the ranges of communication and interference is useful.

Grossglauser and Tse [GrTs01] modified the model in [GuKu00] and included mobility to the capacity analysis. By allowing for unbounded delay and using only one-hop relaying, but taking advantage of the mobility.

Gastpar and Vetterli [GaVe02] worked on the case of relay capacity, while Gupta and Kumar considered only the end-to-end case. They considered the same physical model of a wireless network that Gupta and Kumar used, but under a different traffic pattern: relay traffic pattern.

Chapter 3

Modelling

In this chapter, the parameters that influence the capacity and the performance in a network are analysed. The chapter is closed by describing the OPNET Modeler simulator, together with its functionalities, and explaining how it is used and adapted for this work.

3.1 Capacity and Performance

3.1.1 Communication Ranges

In a residential scenario there are different types of propagation:

- AP installed outside that must give connectivity to users located inside or outside,
- AP installed inside that must give connectivity to users located inside or outside,

An important term in estimating the wireless-transmission range is path loss [CRSK06]. Generally the path loss is defined as:

$$L_{[\mathrm{dB}]} = P_{TX\,[\mathrm{dBm}]} + G_{TX\,[\mathrm{dBi}]} - P_{RX\,[\mathrm{dBm}]} + G_{RX\,[\mathrm{dBi}]} \tag{3.1}$$

where:

- P_{TX} is the power to the transmitting antenna,
- G_{TX} is the gain of the transmitting antenna,
- P_{RX} is the power available at the receiving antenna,
- G_{RX} is the gain of the receiving antenna.

The free-space-propagation model is used when the transmitter and the receiver have a clear, unobstructed, line-of-sight path between them.

$$L_{0[\mathrm{dB}]} = 20log\left(d_{[\mathrm{km}]}\right) + 20log\left(f_{[\mathrm{MHz}]}\right) + 32.44 \tag{3.2}$$

where:

- d is the distance between the antennas.
- f is the frequency of the signal.

This model applies only to clear lines of sight, and it is possible to use it only for initial estimates. The propagation model implemented in the simulator used for this work is:

$$L_{[\mathrm{dB}]} = L\left(d_{0[\mathrm{km}]}\right) + 10nlog\left(d_{[\mathrm{km}]} \middle/ d_{0[\mathrm{km}]} \right) \tag{3.3}$$

where:

- d_0 is a reference distance: for practical systems operating at 1 to 5 GHz is 100m for outdoor environments.

- n represents the path-loss exponent, n =2 in free space and n =3.3 in residential scenario.

In this thesis the assumption is to work in an outdoor environment, although it is possible to extend the study to indoor ones, but the implementation of the propagation model can be quite complex, because an internal building and external structures have a wide variety of partitions and obstacles. The partitions depend on whether the structure is a home or an office environment, [RaPa96].

The model used in this thesis is defined to have two ranges: the transmission, and the interference ranges similar to Li's work [LBCL01]. The transmission range is defined as a certain radius around the sender where good reception is possible. The interference range is defined as the radius where the sender may interfere with other transmissions by adding to the background noise. A node in a WMN can have multiple neighbours within the transmission range, but not all of them act as a next hop in the route path. If neighbours within the transmission range are not a direct next hop, they are regarded as interfering nodes. The interference range is typically over twice the transmission range. Thus, the range model can be written as:

$$R_I = \varepsilon \times R \tag{3.4}$$

where:

- R_I is the Interference Range.
- R the transmission Range.
- ε coefficient of interference typically is $2 < \varepsilon < 3$.

3.1.2 Delay

The following analysis [IEEE03] aims at providing an approximation of the end-to-end delay,(D_{E2E}). It is the most important parameter to determine if is possible to support VoIP calls in a network, so it is another fundamental parameter for this work.

The definition of the theoretical end-to-end delay that is taken as assumption is the total voice packet delay also called "analog-to-analog" or "mouth-to-ear" delay.

For the analysis, the following assumptions are taken:

- The wireless backbone network and the wireless access network operate on different channels. Therefore, it is possible to assume that there is no interference between the data transmissions among APs and the transmissions between users nodes and APs.
- Delay due to processing and propagation operations is negligible compared to transmission.

One of the components of the end-to-end delay is the Delay network, (D_{NET}), which is the time at which the sender node gave the packet to RTP to the time the receiver got it from RTP.

$$D_{NET[\mu s]} = T_{BO} + T_{AIFS} + (T_{PHY} + T_{MAC} + T_{IP} + T_{PAY}) + T_{SIFS} + (T_{PHY} + T_{ACK}) \quad (3.5)$$

As illustrated in Figure 3.1, D_{NET} is the amount of the AIFS time, the average backoff time, the overhead and data transmission time, the SIFS time and the time for the ACK procedure.

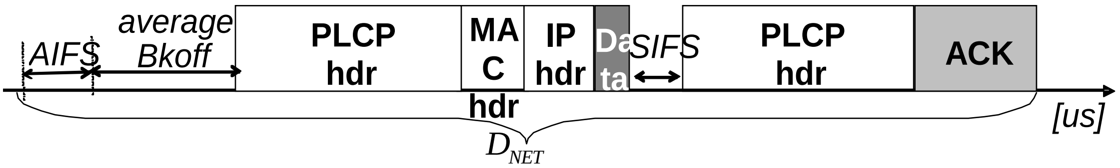


Figure 3.1 IEEE 802.11 overhead in the transmission of a voice data packet (extracted from [FCFN07]).

It is possible to do the same calculation with a more practical formula of D_{NET} using some coefficients as a function of payload size:

$$D_{NET}[\mu s] = a \times x_{[\text{bytes}]} + b \quad (3.6)$$

where:

- x is the payload.
- a and b are two coefficients that depend by the data rate used.

The end-to-end delay is given by:

$$D_{E2E[\text{ms}]} = D_{NET} + D_{COD} + D_{DE\text{-}COD} + D_{COM} + D_{DE\text{-}COM} \quad (3.7)$$

where:

- D_{COD} is the time necessary to encode binary data to form a two-level signal.
- $D_{DE\text{-}COD}$ is the time necessary to decode a two-level signal to form binary data.
- D_{COM} is the time necessary to compress a voice packet.
- $D_{DE\text{-}COM}$ is the time necessary to decompress a voice packet.

Taking as assumption that the time to encode a segment of voice is the same to decode ,and that the time to compress a voice packet is the same to decompress, it is possible to do the theoretical calculation of the end-to-end delay for an user at one hop away to the gateway, using a G.729 A encoder scheme, considering no collisions, and no silence suppression, Table 3.1.

As shown in Table 3.1 the payload in a voice packet generated by the G.729 A encoder is 20 bytes, and the delay for encoding and compression processes are respectively 10 and 20 ms. So, the most important contribution for the end-to-end delay in the one hop situation is given by the time spent for the processing voice.

Table 3.1. a , b coefficients and D_{E2E} calculation in IEEE 802.11a/g for a single-hop network.

Data Rate [Mbps]	a	b	$x_{[bytes]}$	D_{NET} [μs]	D_{COD} [ms]	D_{COM} [ms]	D_{E2E} [ms]
6	1.33	230.17	20	256	10	20	60.256
12	0.66	194.00	20	207	10	20	60.207
24	0.33	177.67	20	184	10	20	60.184
54	0.15	167.00	20	170	10	20	60.169

It is not easy to extend the theoretical analysis to a multi-hop context, because the fact that each receiver hears a subset of transmitters rather than all of them renders the analysis of multi-hop systems far more complex than that of single-hop networks.

Moreover in multi-hop WMNs, link failures are frequent and happen due to heavy contention, which is perceived as a link breakage on repeated failures to deliver a packet. These breakages lead to route failures, which then result in frequent route re-computations, that are very critical in terms of delay for real-time applications, [HoLe08]. So the theoretical study of the end-to-end delay is done only for the users that are one hop far to the gateway.

3.1.3 Throughput

The following analysis [JuPe03] aims at providing an approximation of the Theoretical Maximum Throughput , TMT i.e., the parameter to determine the amount of data that an user can receive in a second. The TMT of a node is synonymous to its capacity, being defined as the asymptotic throughput when the load (the amount of incoming data) is very large.

Before the theoretical analysis, it is important to say that the protocols for wireless networks usually have a much higher overhead than their wired counterpart :

- Each packet is followed by a short acknowledgment message to detect collisions and errors.
- The headers have to include more information due to the shared nature of the medium.

The overhead is fixed for each packet, for large packets; it represents only a few percent or less, but for small packets (real time application) its size may be of the same order as the payload itself. For these reasons, the TMT is not studied for applications that use small packets, like voice, but is the most important parameter to analyse the performance of non real-time application, like HTTP, FTP and Email. A very practical definition for the TMT is derived using the same approach of the previous section:

$$TMT_{[\text{Mbps}]} = \frac{8 \times x_{[\text{bytes}]}}{D_{NET[\mu s]}} \tag{3.8}$$

Unlike packet voice, the payload x in a data packet is big, and can be up to 536 bytes in a single packet. As shown in Table 3.2, the TMT value is 17.33 Mbps for the maximum data-rate. For the same reasons explained in the previous section, it is not easy to extend the theoretical study to more than one hop.

Table 3.2. a, b coefficient and TMT calculation in IEEE 802.11a/g for a single-hop network.

Data Rate [Mbps]	a	b	$x_{[\text{bytes}]}$	D_{NET} [μs]	TMT [Mbps]
6	1.33	230.17	536	943	4.55
12	0.66	194.00	536	548	7.83
24	0.33	177.67	536	355	12.09
54	0.15	167.00	536	247	17.33

3.1.4 Maximum number of users

The expanding interest in WMNs is creating a great deal of effort to support interactive voice services. The challenges in deploying VoIP over WMNs stem mainly from issues related to network congestion and delay. When several VoIP users are connected to the same network, delay and packet loss can be very significant, especially because of the large number of data packets VoIP communications send through the network. Thus, the efficiency of the system quickly deteriorates when the number of calls increases. A natural question to be answered is how many voice calls can be supported in a 802.11 environment. As shown in literature, this question does not have a unique answer. Rather, the capacity is strictly related to the channel bandwidth, voice codec and data traffic in the system [MGGK04].

The maximum number of simultaneous VoIP sessions per MAP, N_{VoIP}, for all the different data rates using again the same approach as in previous sections, considering no collisions and no silence suppression, is given by [FCFN07]:

$$N_{VoIP} = \left\lfloor \frac{D_{COD[\text{ms}]}}{2D_{NET[\mu s]}} \right\rfloor \tag{3.9}$$

As one can see in Table 3.3 , N_{VoIP} without collision, i.e., with minimum delay, is 29 for a particular encoder scheme and using the maximum data rate.

Table 3.3. a , b coefficient and N_{VoIP} calculation in IEEE 802.11a/g for a single-hop network.

Data Rate [Mbps]	a	b	D_{NET} [µs]	D_{COD} [ms]	N_{VoIP}
6	1.33	230.17	256	10	19
12	0.66	194.00	207	10	24
24	0.33	177.67	184	10	27
54	0.15	167.00	170	10	29

Before implementing a WMN, it is important to determine the number and placement of MAPs to ensure coverage. The number of access points required for coverage is estimated by the following simple equation:

$$N_{AP} = \frac{C_{Total}}{C_{AP}} \tag{3.10}$$

where :

- C_{Total} is the total area to be covered,
- C_{AP} is the coverage of a single AP based on maximum power.

3.2 OPNET Modeler

3.2.1 Modeler Architecture

OPNET is a software package supporting extensive features regarding network modelling and simulation means [OPNT07]. One of the most important OPNET features is its object-oriented approach. Moreover, OPNET hierarchical structure of models is well suited for representation of communication networks.

A highly developed graphical user interface integrated with graphical editors of models (e.g., project, node, process link editors) make the tool very user-friendly in supporting intuitive modelling. An additional feature is its openness towards customisation. Users can easily introduce their developed scripts that model communication protocols, algorithms or technologies. Furthermore, there are various application-specific statistics that can be automatically collected while running simulations. There is also an option for recording animations of the simulated system showing statistics changing over time. Another important OPNET feature is related to modelling environment, which is divided into hierarchical domains: network, node, process or external system. Network domain, Figure 3.2, focuses on modelling topologies within a geographical setting. It operates on objects such as subnetworks, nodes and links. Node domain refers to internal node elements and data flow design. When behaviour of protocols, algorithms or applications is modelled, then the process domain is applied.

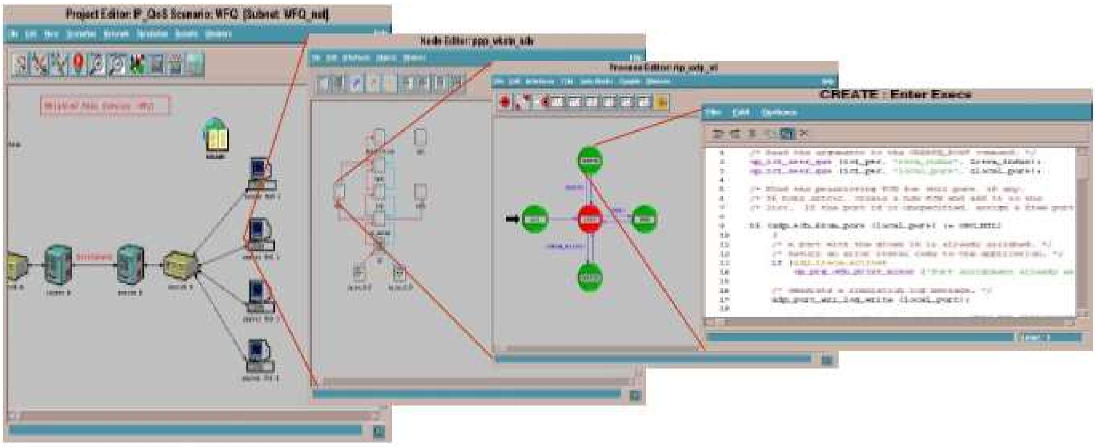


Figure 3.2 OPNET environments: Project editor, Node Editor and Process editor.

The first step in modelling network performance involves network setup that covers topology design and the equipment selection. Topology structure is generally the initial decision taken when developing a network model. It depends on the objectives of the analysis. Selected topologies differ for application deployment study, client server performance or core network analysis. OPNET provides the means for building sophisticated topology cases including complete or partial structures featured with pertinent aggregation. Relevant networks can be modelled exposing areas of interest and hiding complexity of elements that do not impact the study.

Traffic is completely decoupled from the topology, i.e., it is routed according to configuration and protocols operation. It is significant to note that regardless topology, changes the defined traffic is maintained. If topology changes, it is just routed the new paths. OPNET provides two traffic types model, i.e., explicit and background.

Explicit traffic is the most accurate traffic representation, as it imitates packet-by-packet transfer with all the discrete events, accounted. As the packet is generated explicitly, this traffic model covers all single packet requirements, e.g., memory allocation, queuing, etc.. Consequently, simulations with explicit traffic use more computing resources and generally take longer time.

Background traffic represents only an analytical model. Though it does not support so many discrete events, it influences explicit traffic performance in terms of delays as queues builds up at the intermediate elements. Even if faster and less computing demanding, it should be reminded that background traffic stays an approximate model. There are two means for introducing background traffic, i.e., on the basis of traffic flows or as static resource usage. OPNET supports various traffic flows models. This feature enables mapping the flows in source-to-destination fashion onto the network infrastructure. For IP flows background traffic is generated at the network layer, so any L3 layer-aware device, such as router, server or workstation, can generate or receive the IP flow. Traffic flows characteristics can be accurately specified with an explicitly defined pattern type reflecting assigned quality etc. Moreover, the traffic profile can be represented as a function of time with distinguished average bits per second and packets per second. Thus, the profile may change over simulated time.

Static resource usage relies on modelling of additional resource usage. It causes mathematically computed and analytically introduced delaying effects influencing explicit packet-based traffic. Resource usage could involve links as well as nodes or LANs and can be specified with regards to simulated time.

Traffic may be either configured manually or imported from the external files. Importing traffic relies on reading files e.g. with traffic captured by standard network tools. Such extracted real traffic data is usually kept in an archive format or ASCII files that can be read by OPNET. Regarding explicit application traffic, there can be Application Characterisation Environment (ACE) components employed. Another option is to introduce traffic manually. This approach can be applied to all types of traffic by configuring network elements editing their attributes. The background traffic flows can be dragged from source to destination nodes.

Traffic streams can have their characteristics edited. Static resource usage may be adjusted through element attributes. Explicit application parameters can be defined in the application configuration module. Moreover, the profile configuration unit can be used to define the user's manner of exploiting various applications during the simulation period. There are sets of standard application already defined within OPNET, e.g., FTP, Email, Voice, etc., as well as generic sample profiles such as Engineer, Researcher, Ecommerce Customer, Multimedia User, etc..

OPNET simulations are generated automatically as model specifications and are compiled into executable, discrete-event based C-programs. All statistics relevant for collection can be selected in the Simulation Configuration option by setting Choose Individual Statistics or Choose Statistics (advanced) preferences. Apart from the standard Node, Path, Link and Global statistics sets, the user can select attributes of values captured from objects attributes or output scalar values.

3.2.2 Implementation in OPNET

It is important to understand what are the limitations of the WLAN model implemented in OPNET and how to solve the problem. OPNET Modeler does not have a good propagation model for the scenario of interest. However, it is possible to develop, a custom propagation model using one of the standard propagation models. A new propagation model in OPNET requires the modification of the Parameter file, the Description file, the Source code file, and the rebuild of all the project and the links update, [CRSK06].

Another drawback is that in OPNET the transmission data rate used by a WLAN node is static through the entire simulation time. In other words, the model does not implement the rate adaptation feature specified in the standard, where, for an increment of distance, the system itself sets at lower data rate that can guarantee less transmissions errors.

It is important to understand how to change the data rate with distance so that knowing the distance between two nodes it correct set the data rate for the link. It is possible to do this by putting two nodes and change the distance for each simulation to check if there is connectivity or not, and repeat this operation for each data rate.The results are shown in Figure 3.3, where there is a comparison between theoretical and practical values of data rate for 802.11a/g vs. range transmission using 1 W for the power transmission of 802.11a and 100mW for 802.11g and the values recommended from Cisco for the sensitivity.

A real simple scenario is used as a basic project, to understand the behaviour of the most important parameters involved with only one application. Only one user that uses Voice application is taken. The most important values used to characterise the Voice and the others applications parameters are available in Annex B, and the applications profiles are presented in Annex C. All the various parameters that define the behaviour of the network nodes (stations, AP) can be accessed and changed directly through the main interface of the project editor. These include all WLAN parameters: Table 3.4 shows the WDS parameters, and Table 3.5 shows the BSS parameters.

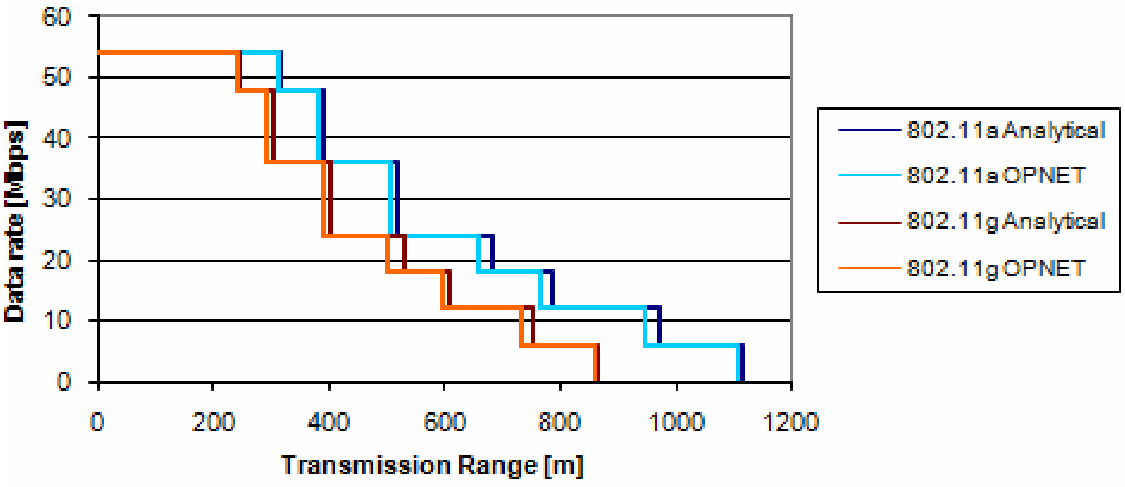


Figure 3.3 Data rate vs Transmission Range for 802.11a/g.

As one can see, the coverage done with 802.11a can be better that the one done with 802.11g, the data rate for 802.11a start to decrease when the two nodes are at 300 m, and at more than 1200 m there is no possibility to have any communication between the two nodes. These results show that, below certain conditions, and doing the analysis with the path loss implemented, it is possible to cover with a minimum data rate more or less 1 km. This means that it is possible to study a network with one side of 1 km, and guarantee coverage with a minimum number of MAPs, one for each extreme of the network.

Table 3.4. WDS parameters.

Attribute	Value
BSS Identifier	1
Access Point Functionality	Enabled
Physical Characteristics	OFDM (802.11a)
Data Rate [Mbps]	6;9;12;18;24;36;48;54
Transmit Power [W]	1
Packet Reception-Power Threshold [dBm]	-91;-89;-89:-86:-84:-80:-76:-73

Table 3.5. BSS parameters.

Attribute	Value
BSS Identifier	101
Access Point Functionality	Disabled
Physical Characteristics	Direct Sequence (802.11g)
Data Rate [Mbps]	6;9;12;18;24;36;48;54
Transmit Power [W]	0.100
Packet Reception-Power Threshold [dBm]	-91;-89;-89:-86:-84:-80:-76:-73

It is also important to define the outputs that the simulations have to show. Each of the modules at the node and link levels is a source of a significant number of available statistics. The selection of statistics is performed by specifying a list of probes, which perform the selection and control the flow of data from the selected statistics into output files.

All these statistics and performance metrics are clearly defined in Modeler. The definitions of the ones used in this work are:

- **Average node throughput at application level**: The average bytes per second forwarded to the application layer by the TCP layer in one node that is *n* hops far from the gateway.

- **Voice Packet End-to-End Delay**: The total voice packet delay (in seconds) called "analog-to-analog" or "mouth-to-ear" delay = network_delay + encoding_delay + decoding_delay + compression_delay + decompression_delay:

    - Network delay is the time at which the sender node gave the packet to Real-Time Transport Protocol (RTP) to the time the receiver got it from RTP.

- Encoding delay (on the sender node) is computed from the encoder scheme.
- Decoding delay (on the receiver node) is assumed to be equal to the encoding delay.
- Compression and Decompression delays come from the corresponding attributes in the Voice application configuration.

A complete description of all the metrics used in this thesis for the analysis of the results is available in Annex A.

It is important to define the main aspects of the simulations. In order to have some statistical relevance , in each scenario, the stabilisation period and the minimum number of simulations must be performed. The objective is to make 10 simulations with 10 random seed of 30 minutes for each scenario, as it is considered that 30 minutes is a large enough period to have meaningful and stable results in the network with any load traffic. Only for the first basic scenario was simulated only 10 minutes for each seed.

Chapter 4

Performance analysis

In this chapter, all simulated scenarios are presented, giving the definition of all the parameters of each scenario, i.e., services, their definition and usage profiles, and also the number of users. Finally, the results from all these simulations are presented and analysed.

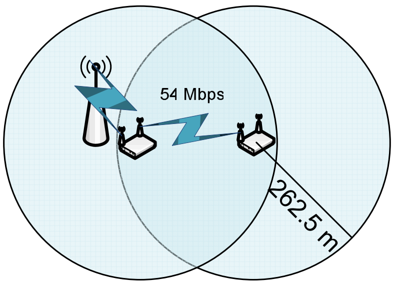
4.1 Scenario Description

The approach in this work is to start with an easy scenario, and going forward analysing scenarios increasingly close to a real WMN for a residential scenario. First of all, some consideration are done that are important to understand the analysis.

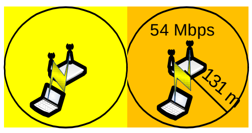
The density of MAPs is related to the data rates and coverage areas, and for a given data rate the maximum transmission power possible is used. To make the scenario more real a study is done on the position and the range of the nodes building the network, in order to have good coverage and minimum interference.

4.1.1 Basic Scenarios

A basic network topology with two MAPs and two users that have the same offered load sent to the gateway (GW) is shown in Figure 4.1. One need to analyse like [JuSi03], the maximum user throughput available for a multi-hop network, named in this work User_Th., each user downloads an FTP file. Increasing the file size increases the quantity of traffic in the network, so it is possible to show the limit of the network in term of throughput, and to compare it with the TMT found in the analytical study done in Section 3.1.3.



(a) WDS coverage.

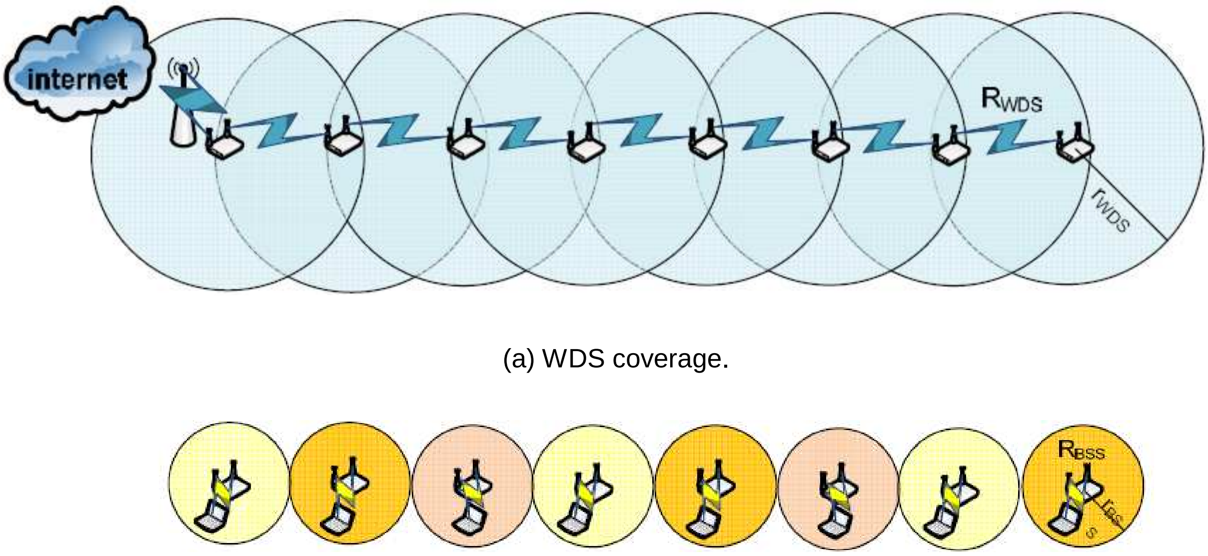


(b) BSS coverage.

Figure 4.1 Two-node network – WDS and BSS.

The second topology studied is a chain network formed by 8 MAPs and 8 users, one for each MAP, and can be seen as a study of an ideal street coverage, Figure 4.2. In the first set of simulations, taking only FTP traffic, one studies how the throughput decreases over a multi-hop WMNs. Three different traffic condition are simulated for three different file sizes: 0.5 MB, 5 MB, and 50 MB. Instead, in the second set of simulations, only voice traffic is taken being analysed the increase delay with the numbers of hops, and how many users per hop can do an acceptable quality call.

As described in Table 4.1 for each simulation there is one user per MAP. The first simulation is done putting the nodes as far is possible covering a large area but with a low data rate. For the others simulations is reduced the distance between the nodes consequently the data rate increases.



(a) WDS coverage.

(b) BSS coverage.

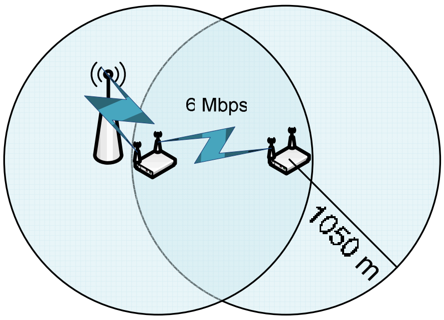
Figure 4.2.one-dimensional multi-hops network - WDS and BSS.

Table 4.1. Parameters set for each simulation in the chain network scenario.

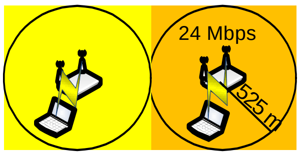
# simulation	# MAPs	# users per MAPs	R_{WDS} [Mbps]	r_{WDS} [m]	Prx_{WDS} [dBm]	R_{BSS} [Mbps]	r_{BSS} [m]	Prx_{BSS} [dBm]
1	8	1	6	1100	-91	18	610	-86
2	8	1	9	950	-89	24	530	-84
3	8	1	12	950	-89	24	530	-84
4	8	1	18	750	-86	36	400	-80
5	8	1	24	650	-84	36	400	-80
6	8	1	36	500	-80	48	300	-76
7	8	1	48	380	-76	54	240	-73
8	8	1	54	310	-73	54	240	-73

The next scenario takes as assumption that normally one street can be covered by using both low density MAPs with low data rates and several MAPs with high data rates. In addition, the scalability characteristics of a WMNs can build a network with the minimum number of MAP, and add others as more users joint the network. In the first set of simulations, the behaviour of the network with the variation of the number of MAPs in the same area is studied.

The study is repeated using only voice traffic, FTP and voice together, and analysing on the behaviour of calls in terms of delay. The first topology is composed of two MAPs that cover the region under study. The distance between the two MAPs is 1050 m and the possible data rate for 802.11a is 6 Mbps for the WDS and 24 Mbps for the BSS, which guarantees connection conditions with minimum interference, Figure 4.3.



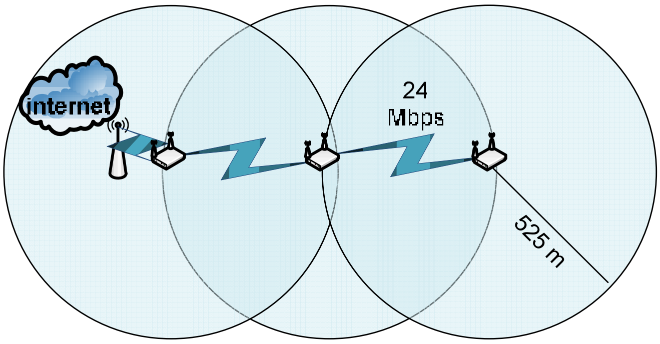
(a) WDS coverage.



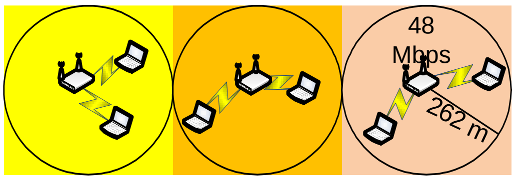
(b) BSS coverage.

Figure 4.3 Two-node network - WDS and BSS.

In the second topology, Figure 4.4, is put one more MAP in the middle. The distance between the three MAPs is decreased to 525 m, and the BSS range is 262 m. The possible data rate for 802.11a is 24 Mbps for the WDS and 48 Mbps for the BSS.



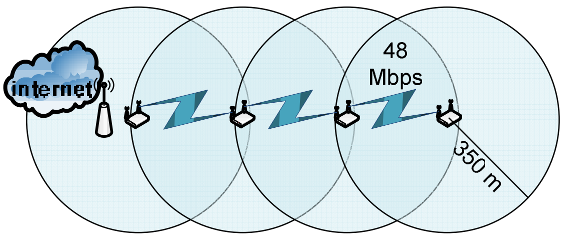
(a) WDS coverage.



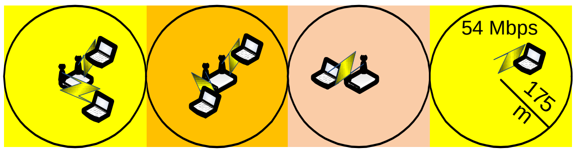
(b) BSS coverage.

Figure 4.4. Array of 3 MAPs – WDS and BSS.

As one can see in Figure 4.5, there is a regular chain composed of 4 MAPs. The distance between the MAPs is 262 m, and the possible data rate is 48 Mbps for the WDS and 54 Mbps for the BSS.



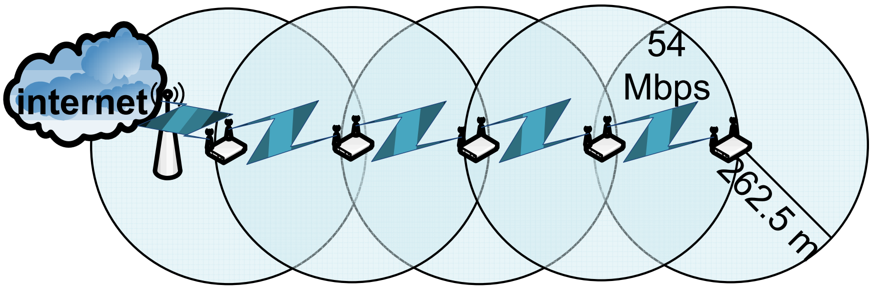
(a) WDS coverage.



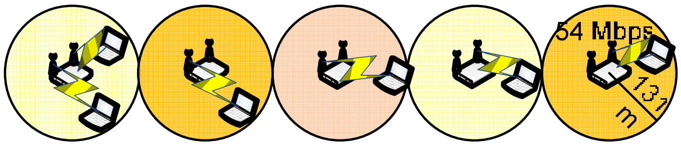
(b) BSS coverage.

Figure 4.5. Array of 4 MAPs – WDS and BSS.

To conclude, in Figure 4.6, there is a chain composed of 5 MAPs. The distance between the MAPs is 262 m, and the BSS range is 131 m. The possible data rate is 54 Mbps for both WDS and BSS.



(a) WDS coverage.



(b) BSS coverage.

Figure 4.6. Array of 5 MAPs – WDS and BSS.

As described in Table 4.2, for each simulation there is the same number of users in the network, but the number of users per MAPs is different.

Table 4.2. Parameters set for each simulation in the variable number MAPs scenario.

# simulation	# MAPs	# users per MAPs	R_{WDS} [Mbps]	r_{WDS} [m]	Prx_{WDS} [dBm]	R_{BSS} [Mbps]	r_{BSS} [m]	Prx_{BSS} [dBm]
1	2	3.0	6	1050	-91	24	525	-84
2	3	2.0	24	525	-84	48	262	-76
3	4	1.5	48	350	-76	54	175	-73
4	5	1.2	54	262	-73	54	131	-73

4.1.2 Residential Scenario

The Residential Scenario is drawn from the a chain network used to study a street in a residential scenario, by adding more then one chain of MAPs it is possible to draw a large area network that can cover a square residential zone of one km

In a Residential Scenario each MAP covers a specific area, where small communities of residential users use a set of services. The characteristics of each service are listed in Table 4.3, and the users profile is shown in Figure 4.7.

The number of users is relatively low, and the usage of services is not intense and high traffic load levels not being expected.

A two dimensional network is implemented, so that each configuration can give connectivity for all the users with minimum interference, Table 4.4.

Figure 4.8 shows the simulation scenario with the highest number of MAPs. The others drawing are not shown, but are easy to derive, reducing the number of MAPs and guaranteeing coverage like in the previous section.

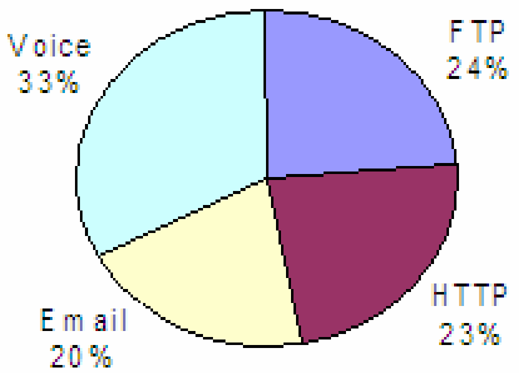


Figure 4.7. profile of services for a residential user.

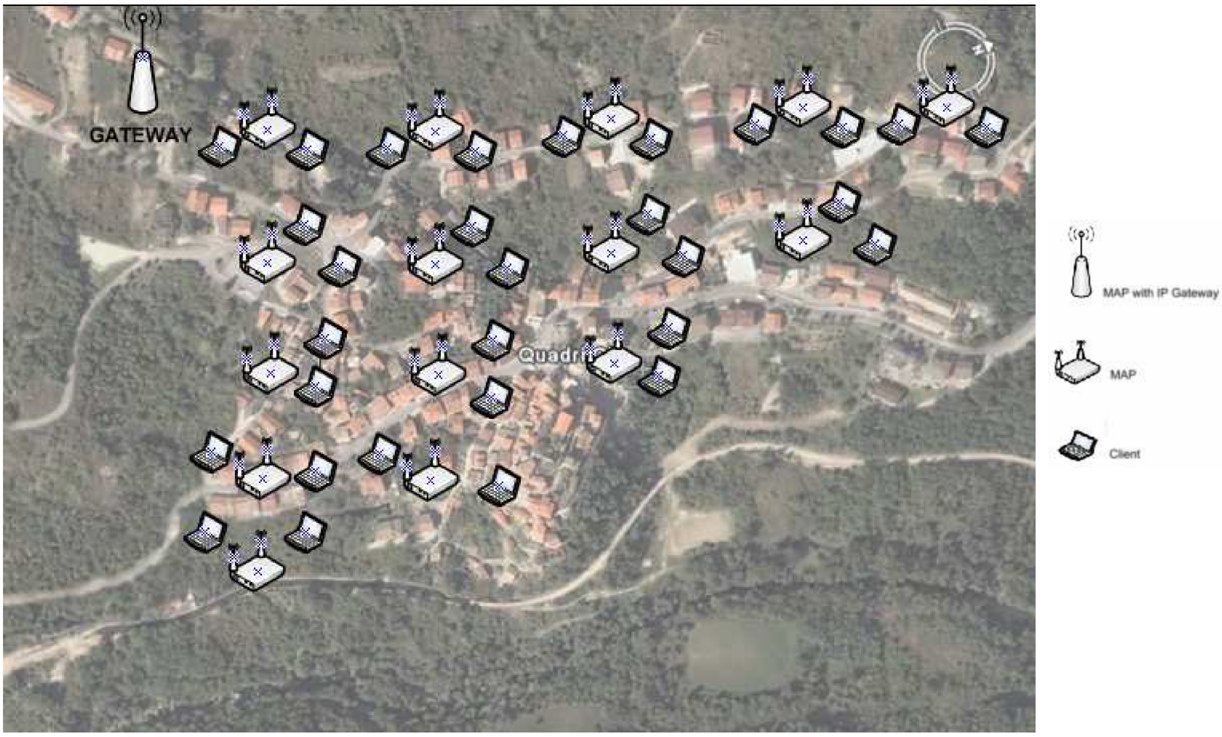


Figure 4.8**.** Residential scenario – maximum number of MAPs.

Table 4.3. Applications Characterisation.

Application	Mean Call Duration [s]	Bitrate [kbps]	Mean Time between Requests [s]	Mean Request Volume [kB]
VoIP	180	8 (max)	- -	- -
HTTP	- -	- -	39.5	200.0
FTP	- -	- -	360.0	5000.0
Email	- -	- -	600.0	100.0

Table 4.4. Parameters set for each simulation in the residential scenario.

# simulation	# MAPs	# users per MAPs	R_{WDS} [Mbps]	r_{WDS} [m]	Prx_{WDS} [dBm]	R_{BSS} [Mbps]	r_{BSS} [m]	Prx_{BSS} [dBm]
1	3	10	6	1050	-91	24	525	-84
2	6	5	24	525	-84	48	262	-76
3	10	3	48	350	-76	54	175	-73
4	15	2	54	262	-73	54	131	-73

As one can see in Table 4.4, the number of users is fixed to 30 for each simulation.

4.2 Analysis of the Basic Scenario

4.2.1 Two nodes network

The two nodes scenario was simulated using 14 different file sizes, with simulations of 10 minutes each, using 10 different and randomly chosen seeds, leading 140 simulations, and a running time about 11h 36 , using a HP Pavilion dv 1000 1.256 GB of RAM.

The main objective of the simulations performed in this section is to evaluate the User_Th available for two users that are respectively at one and two hops away from the gateway, the results being compared directly with the ones derivable by the analytical approach, Figure 4.9.

In the network there are two kinds of traffic internal and external. Internal traffic of a WMN node the traffic generated internally by the users, instead the external traffic of a node is the traffic that from a neighbouring node has to be forwarded to another neighbour.

WMN nodes that receive external traffic are expected to forward it to the appropriate neighbour after looking up the destination IP address, but the study of routing in WMNs is outside the scope of this thesis.

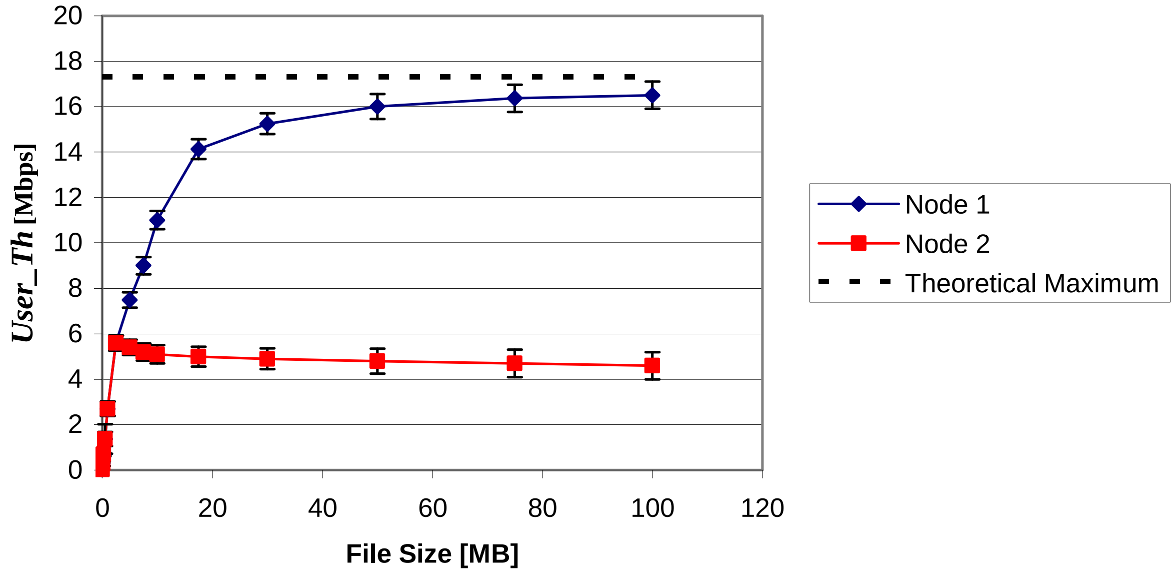


Figure 4.9 Throughput available at application level in a two-node network.

If the file size is short, there are no different behaviour between the two nodes, i.e. if the user is one or two hops away, the available network capacity is the same for both, because the network is not in heavy traffic conditions. When the file size increase, the throughput increases too and the network becomes more heavy. For a certain file size, the throughput for each node becomes different, since the network cannot forward all data enqueued to node close to the gateway, and the queue starts to overflow.

With a probability that increases with the offered load, the queue will be full when a new packet arrives to the node at one hop, and it will be dropped immediately after it is received. Putting a heavy network traffic, the throughput resulting is the maximum available User_Th. This study is for a link at 54 Mbps, and the maximum available User_Th for the user at one hop is less than the theoretical value TMT that is 17.33 Mbps like illustrated in Table 3.4.

4.2.2 Chain network

The chain network scenario was simulated using 48 different situation leading to the execution of 480 simulations, taking about 208 h to complete.

In what follows, one studies a one-dimensional multi-hops network, focusing on the throughput and delay aspect analysing the User_Th behaviour with only FTP traffic on the network and with only voice traffic, and the user end-to-end delay, User_E2E_D. To analyse the User_Th degradation with the number of hops, several simulations are done for a constant number of MAPs, changing the data-rate and the load of the network. The idea is to simulate the worst-case, where the internet access is not in the middle of the scenario, but to one side and all the users, one for each MAP, are trying to download a FTP file simultaneously.

In Figure 4.10, Figure 4.11, and Figure 4.12 the hop count of path versus the User_Th is plotted for a linear chain of nodes . There are eight plots, each corresponding to a different date rate. For each data rate, it is possible to observe that the throughput of traffic decreases with an increase in the hop count of path (i.e., the number of hops from source to destination).

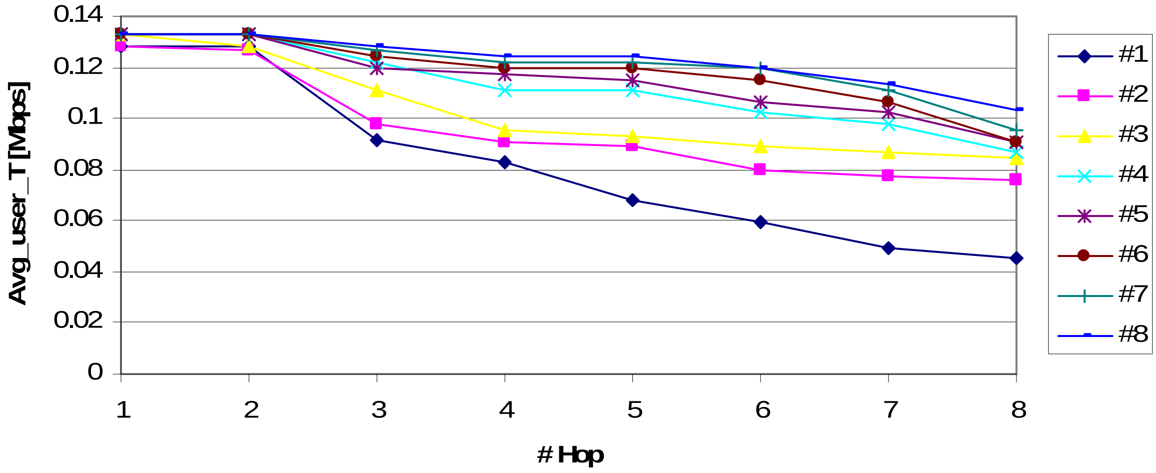


Figure 4.10. File size 0.5 MB - Throughput vs # hops for each WDS data-rate.

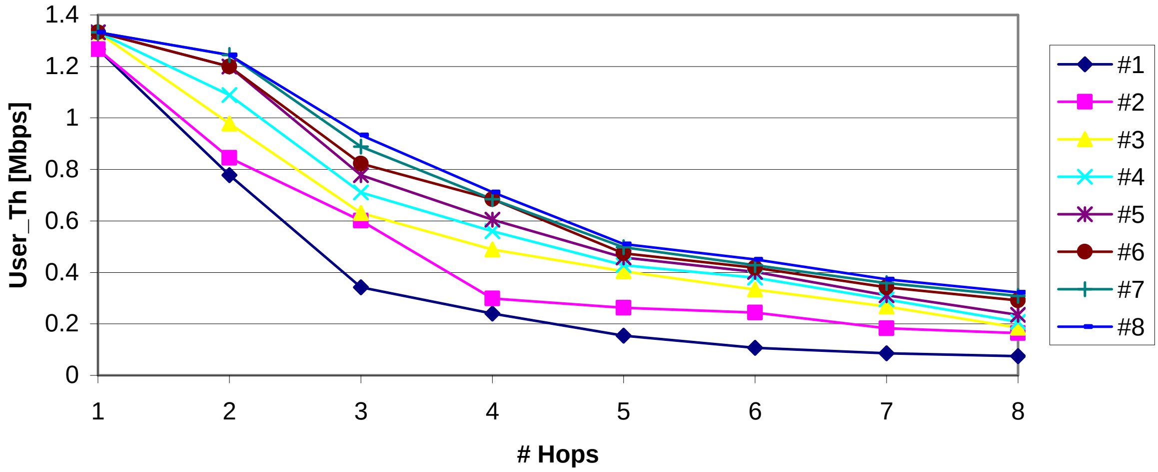


Figure 4.11. File size 5 MB - Throughput vs # hops for each WDS data-rate.

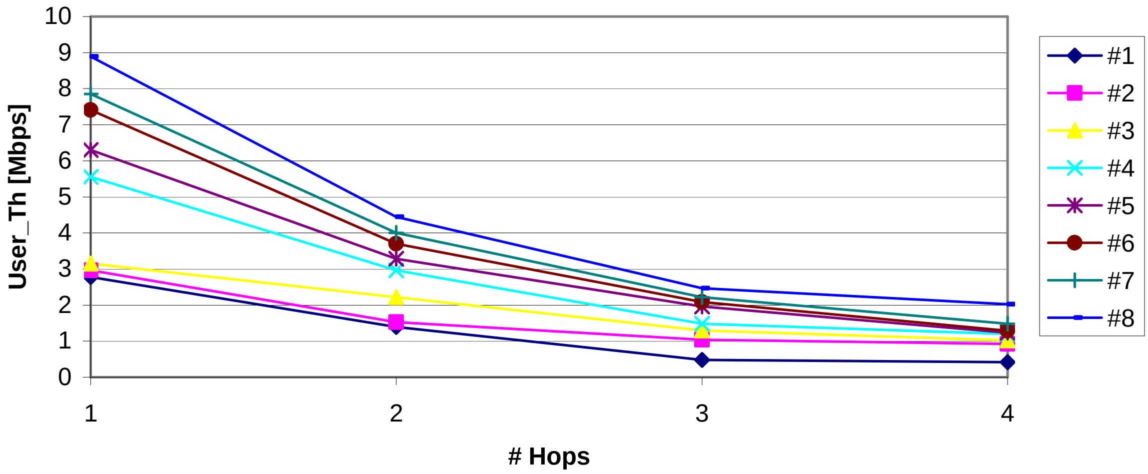


Figure 4.12. File size 50 MB - Throughput vs # hops for each WDS data-rate.

The packet error probability increases with an increase in the number of hops causing the throughput to decrease. For a fixed hop count, the throughput for a higher data-rate is higher than that for lower data-rate. However, the difference between the throughput decreases with an increase in the hop count; for nodes that are more far than 5 hops away from the gateway is the high number of hops that imposes a low throughput.

It is possible to see also that the gain in throughput at the application level is not in proportion to the increase in the link rate, because the actual packet transmission time has a fixed part, which does not depend on the link rate significantly. If the network is in low traffic conditions, the throughput degradation is not strong, i. e., one user at eight hop away from the gateway can download a small file with good performance, Figure 4.10.

It is possible to see the different behaviour of the throughput degradation: when increasing the file size, the network becomes busy and the degradation is more deep along the increasing number of hops. For the simulation with a file size of 50 MB, the network is very busy and the analysis is stopped at the fourth hop.

As mentioned in Section 3.1.3, it is not easy to do a theoretical analysis of TMT for a multi-hop environment, so it is necessary to find a behaviour of the throughput degradation over the number of hops, by analyzing the results, it is possible to see that they follow approximately the function *1 / n,* [GuKu00].

So, Figure 4.13 shows the normalised User_Th, for the simulation # 1 for a 5 MB file size. This result can be very useful in a project, because it gives directly an idea of what is the behaviour of the network below certain conditions, i.e. ,data rate and number of MAPs per path.

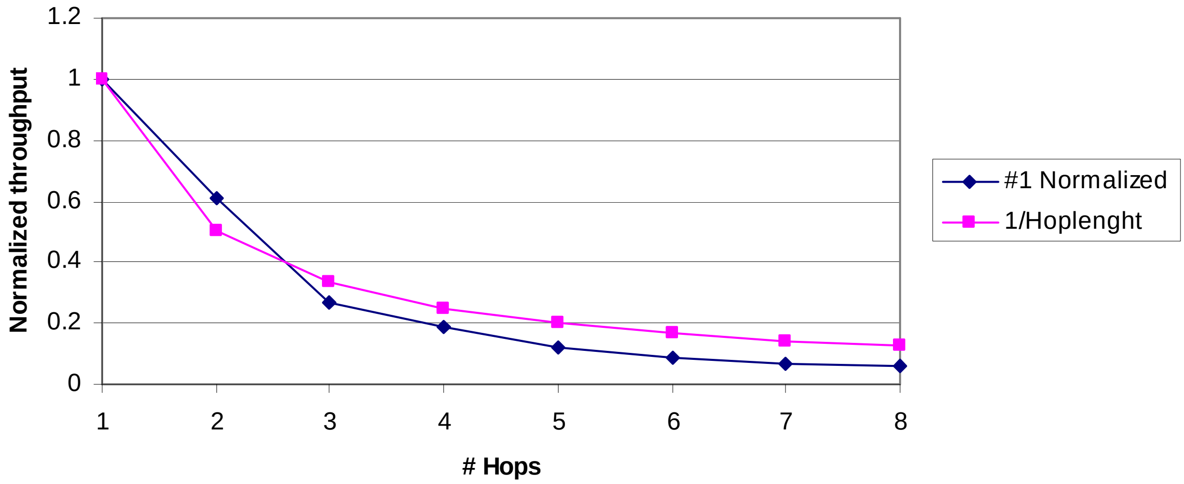


Figure 4.13. Throughput degradation in a WMN with chain topology.

When one node transmits to another, especially when CSMA/CA-based MAC protocols are employed, the neighbours cannot initiate another transmission, Figure 4.14. This exposed node problem contributes to the throughput degradation in WMNs over a relayed multi-hop path. For example, a two-hop flow has to share the bandwidth between the two, therefore the end-to-end throughput for a two-hop path is only 61% of the single-hop throughput.

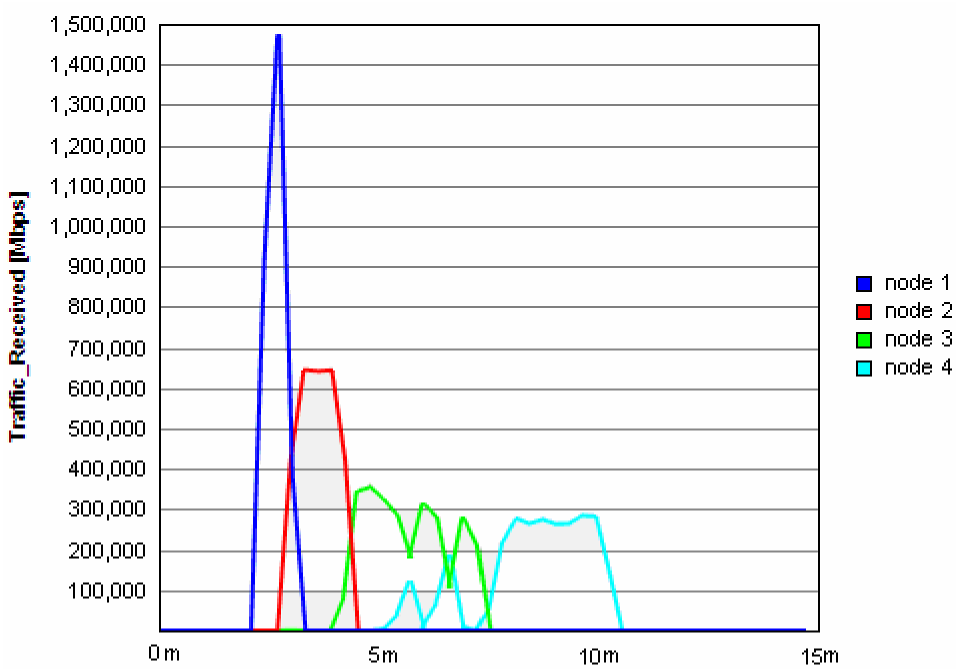


Figure 4.14. Simultaneous download of a 50 MB file in a 4 MAPs chain network.

To complete the study of this scenario, the system with the same characteristic but putting only voice traffic in the network is evaluated, so that all the users are trying to do a call simultaneously. One assumes that there is one user per MAP, and that during the peak hour of the day, each user at the same time makes one call of 3 minutes duration. Furthermore it is assumed that a medium quality call is a call where the coverage is less than 400 ms, which is the time that corresponds to the human ear sensitivity. The idea is to analyse the behaviour of the network when there is only voice traffic, the delay for each user, and maximum number of users per hop.

As one may see in Figure 4.15, one does not represent the User_E2E_D over all eight hops, but only until the fourth hop, because after it the values are very high. It is shown how critical the voice over WMN is, since when the data rate is low, only the user that is one hop away can do a call with acceptable end-to-end delay.

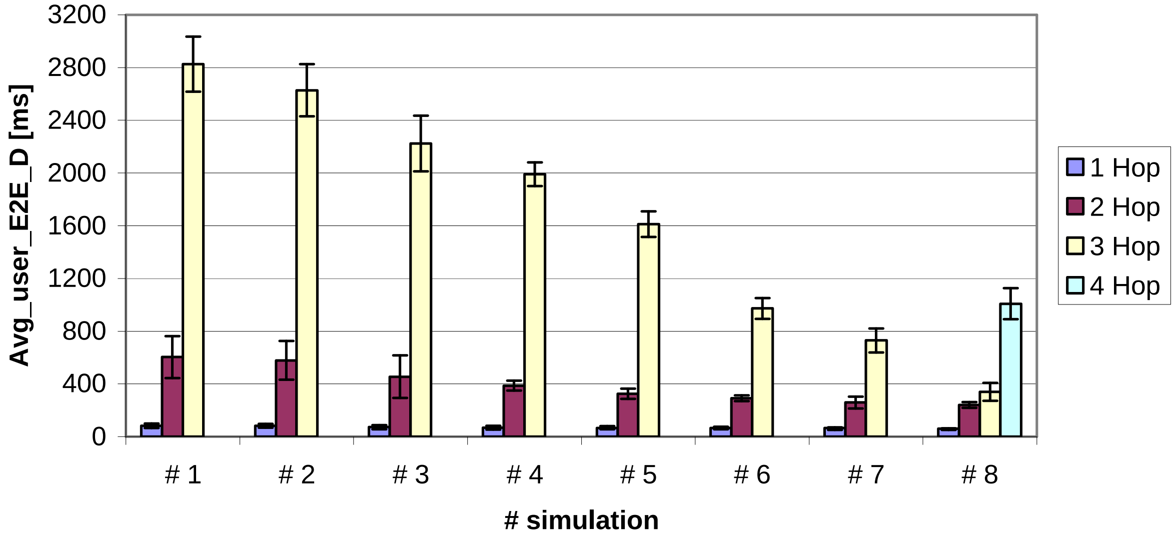


Figure 4.15. User end-to-end delay at different hops for each WDS data-rate.

When the data rate increases, it is possible for user farther away to do a call with acceptable quality, but for the maximum data rate only users that are 1, 2 and 3 hops can do a call with acceptable end-to-end delay. So when the numbers of hops increases User_E2E_D can easily go up to 400 ms, and can be as high as 2000 ms. This value is in according to the study done in Section 3.1.2. By analysing the delay in Figure 4.16, it is possible to understand the behaviour of the network in terms of User_E2E_D, and that it increases with quadratic and cubic law. This means that for low data rates User_E2E_D increases more quickly than for high data rates.

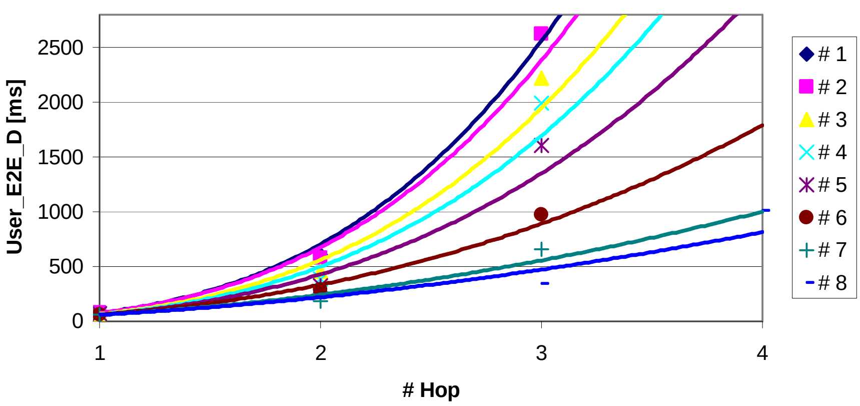


Figure 4.16. Trend of the user end-to-end delay in a chain of MAPs for different data rate.

Table 4.5. Trend Equations and correlations – end-to-end Delay for a chain network.

# simulation	Equation	R^2
# 1	$y = 77\ x^{3.18}$	0.994
# 2	$y = 78\ x^{3.11}$	0.994
# 3	$y = 67\ x^{3.07}$	0.989
# 4	$y = 62\ x^{3.01}$	0.983
# 5	$y = 60\ x^{2.83}$	0.977
# 6	$y = 62\ x^{2.42}$	0.992
# 7	$y = 59\ x^{2.04}$	0.963
# 8	$y = 60\ x^{1.89}$	0.960

Figure 4.17,shows how many users can have a good call staying at 1, 2 and 3 hops away.

This parameter is derived by doing several simulations, where the number of users per MAP is increased in the network until User_E2E_D is close to 400 ms, which means that #_hop_calls is achieved. The number of users was increased only for the MAPs that allowed an acceptable quality call.

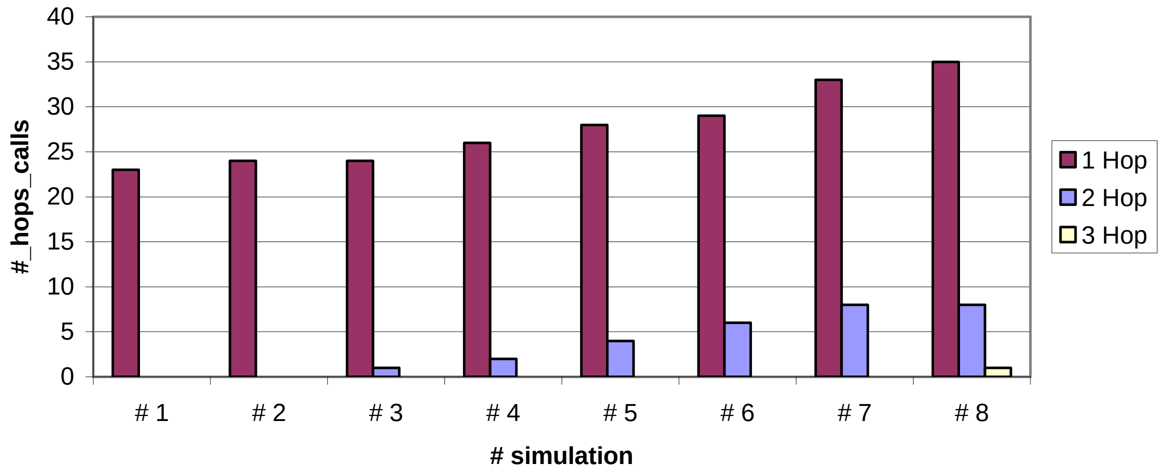


Figure 4.17. Maximum number of simultaneous calls available for each hop vs # simulations.

As shown in Figure 4.17, when the data rate increases the number of possible calls increase too. Moreover, with the increase of data rates in the nodes that are 2 or 3 hops away it is possible to have a certain number of acceptable quality calls. Generally, voice packets are small and they are sent very frequently so the probability of collision increases with the increase of the number of users.

It is not possible to compare directly these results for the maximum number of simultaneous calls with the theoretical study done in Section 3.1.4, which assumes for the calculation that there are no collisions.

Putting, in the MAPs one hop away to the gateway, some more users than N_{VoIP} the number of collisions becomes very high and as one can see in Figure 4.17 for simulation # 8 adding 6 users to N_{VoIP} delay for a certain call is higher than to 400 ms.

4.2.3 Variable number MAPs

Now that it is clear that in a network with more than 3 hops it is not possible to guarantee a medium quality call, it is not useful to study a topology with many hops, so the number of hops possible in this topology is not more than 5.

The scenario was simulated using 8 different situations, taking about 42 h 40 to complete.

The main objective of this section is to find an answer to the following question: which is a better path for VoIP or FTP traffic, e.g. a 4 hops path with 48 Mbps bandwidth links, or a 2 hop path with 6 Mbps bandwidth links? Even though a 4 hop path has a greater end-to-end packet error probability compared to a 2 hops ones, the link data rate in the 4 hops path is four times the link data rate in the 2 hops one.

Therefore any performance degradation of transport layer protocols due to an increase in the number of hops seems to be counter balanced by a higher link bandwidth. Hence, a need arises to perform simulations to understand this problem better.

So, another interesting study is to show how the average throughput in the network changes, simulating a more real one-dimensional multi-hops network, where at the start there are only 2 MAPs that are very far from each other at 6 Mbps for R_{wds} and at 24 Mbps for R_{BSS}, and as the density of MAPs between the two extreme MAPs increases, the range decreases and the data rate increases. In these simulations the file size is 5 MB and all the users are trying to download it simultaneously.

In Figure 4.18 and Figure 4.19, it is possible to see that there is a large increases in the throughput as the data rate increased from 2 MAPs to 3 MAPs, and from 3 MAPs to 4 MAPs. However, increasing more the number of MAPs does not show any increment in the throughput. Thus, increasing the link data rate only reduces the transmission of the packet but not the (almost) fixed MAC overhead. On the other hand, increasing link data rate causes more errors and, in addition, reduces the span of the network.

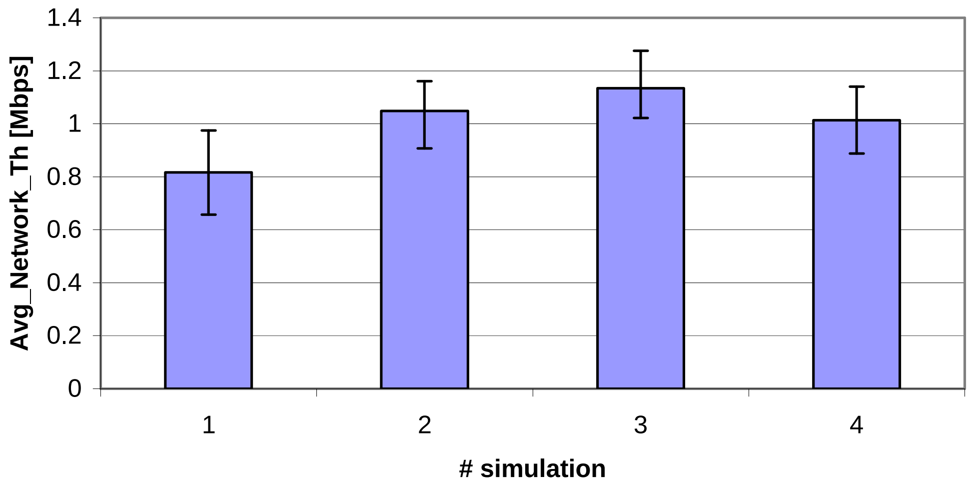


Figure 4.18. Average throughput network.

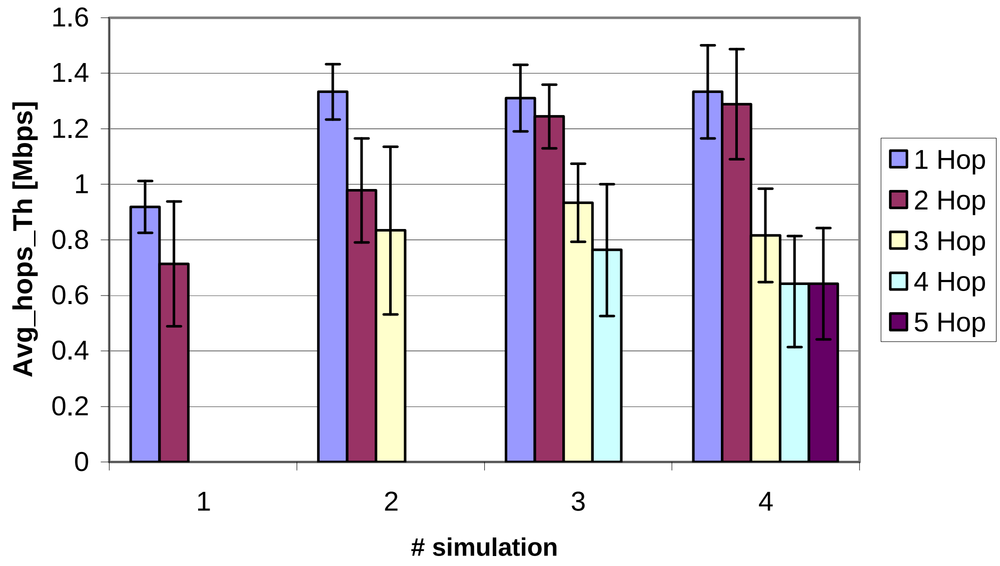


Figure 4.19. Average Throughput per hop vs # MAPs

There are other interesting results that show that if there are not heavy traffic conditions, it is important to evaluate the number of MAPs to put in the network: if it is too big relative to the expected traffic conditions, the performance in terms of throughput is not better that a network with less MAPs and the level of interference among MAPs increases. So, for each topology, it is important to find an ad-hoc solution that is a trade-off between performance, numbers of MAPs, and cost .

Dividing Avg_hops_Th by the Rbss it is possible to show Avg_hops_Th in relative terms. As one can see in Figure 4.20 the throughput in a chain network expressed in relative terms decreases with the increasing of data-rate. So it is confirmed that high data-rates improve the throughput in the network, but aldo that it is a low improvement compared with the increment of data-rate. To complete the study of this scenario, the behaviour of the network putting only voice traffic is evaluated, all users are trying to do a call simultaneously, and the parameters are described in Table 4.2

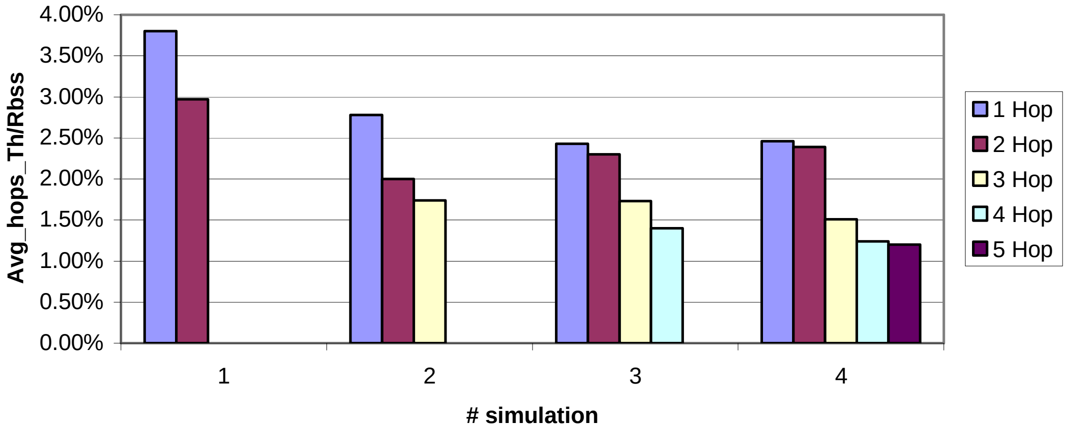


Figure 4.20. Average throughput in relative terms.

Comparing the results shown in Figure 4.21 with those of Figure 4.17 the behaviour is the same, but in Figure 4.21 there are better conditions in terms of acceptable quality call, which can be explained by the fact that the chain of MAPs is not formed by more than 5 ihops in the last study, instead of 8. To complete the study, how many users that can do an acceptable quality call when are using FTP and VoIP simultaneously is evaluated.

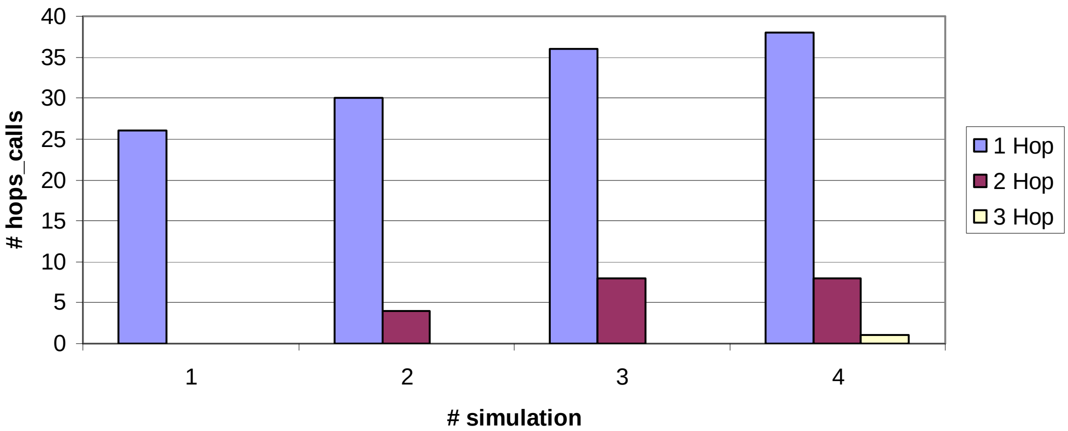


Figure 4.21.Maximum number of voice calls available for each Hop vs. # simulations.

In the case where users are simultaneously using FTP and VoIP, Figure 4.22 results in terms of performance are not very different from the results obtained in the case of only voice traffic, Figure 4.21, because the high priority given to voice relative to FTP and the low throughput of a call allows to handle a high number of calls, even in the presence of other types of traffic.

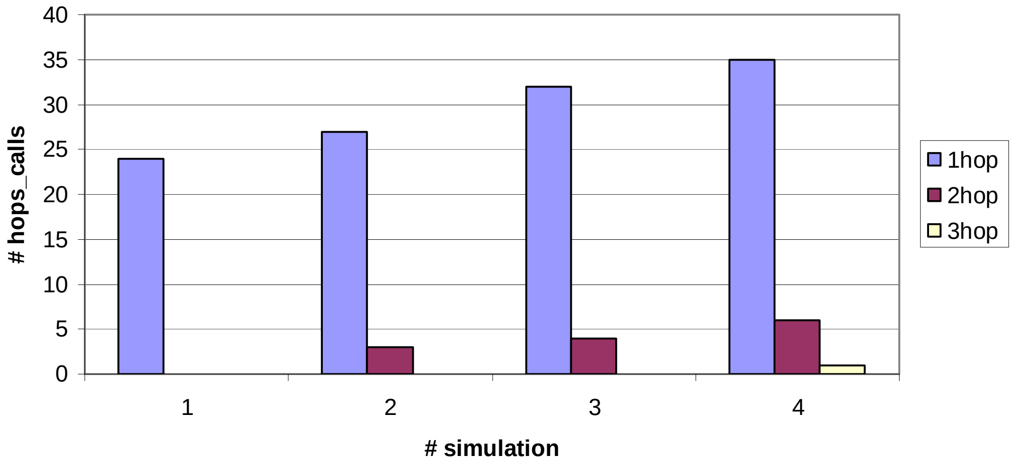


Figure 4.22. Maximum number of voice calls available for Hop vs. # simulations – FTP & voice.

Using #_hops_calls as the number of channels, N_{CH} one can estimate how many calls is possible to have in one hour with the assumption that each user can do a 3 minutes call per hour with a blocking probability of 1%. This estimation is only qualitative, because is not correct to use the Erlang B model in a Packet Switching (PS) system since it is to be used in a Circuit Switching (CS) network.

Table 4.6. Data traffic for the whole network.

# simulation	N_{CH}	P_b	$Tr_{[Erl]}$	$T_{call[min]}$	$N_{call/h}$	$Tu_{[Erl]}$	N_{user}
1	24	0.01	15.25	3	1	0.05	305
2	31	0.01	21.15	3	1	0.05	423
3	39	0.01	28.10	3	1	0.05	562
4	43	0.01	31.65	3	1	0.05	633

Table 4.6 shows how many channels are available considering the whole network, the probability of blocking P_b, the total traffic Tr, the duration of one call T_{call}, the number of calls per hour for each user $N_{call/h}$, the traffic of one user Tu and the total number of user calls N_{user} that the network can support in one hour with 1% of blocking probability.

To complete the estimation, it is interesting to repeat the calculation for the number of channel available at each hop. So the number of users that can do a call in one hour, # calls_per_h, is different for each hop: in the best data rate conditions, at one hop 492 users can do a call, but this number is reduced to 62 at 2 hops and to 3 at 3 hops, Figure 4.23.

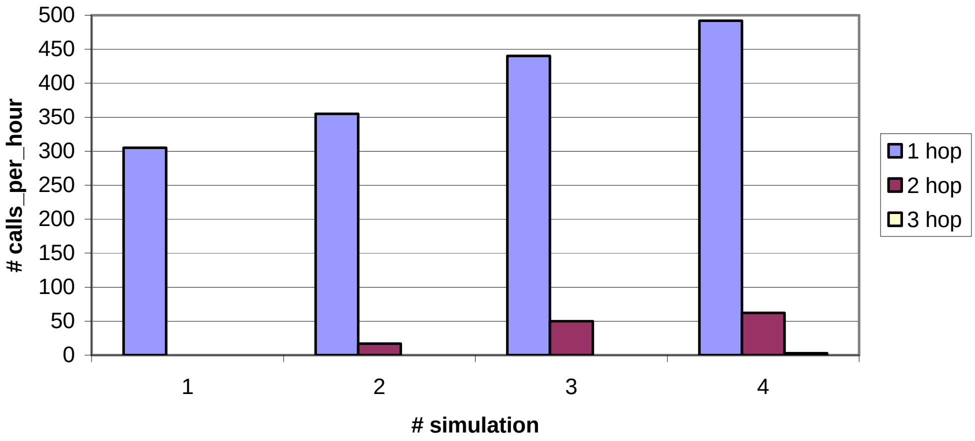


Figure 4.23. Total number of users that the system can support in one hour.

4.3 Analysis of the Residential Scenario

The Residential scenario was simulated using 8 different situations,taking about 100 h to complete. The objective in this set of simulations is to study the behaviour of the network in terms of end-to-end delay and user throughput while the users are using different types of services.

As one can see in Figure 4.24, when the network is composed of only a few MAPs that have to cover the whole area the quality of call is good, because the gateway it is not far away in terms of the number of hops.

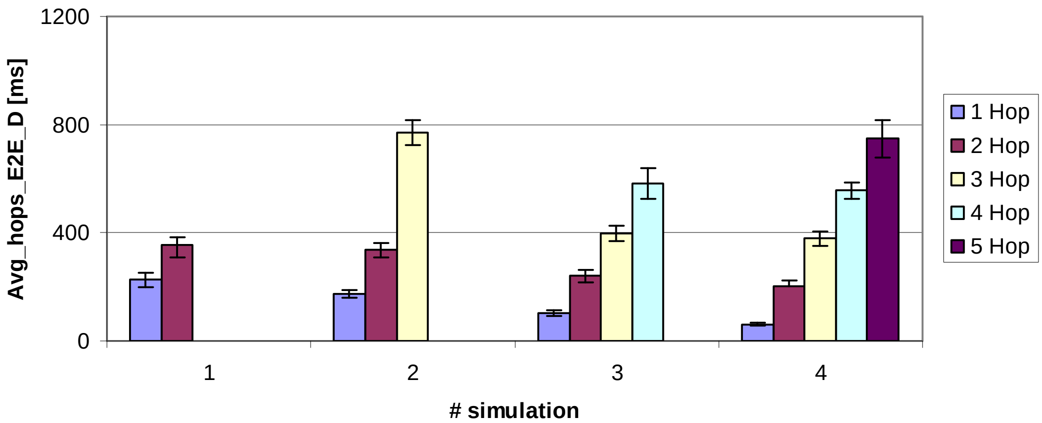


Figure 4.24. Average end-to-end delay for each hop at increase number of MAPs

As more MAPs are put in the network, the number of hop increases, and the quality of calls becomes unacceptable for a user that is more than 3 hops away from the gateway. Analysing only the end-to-end delay result it is possible to conclude that more multi-hop imply that less users can have good quality call, Figure 4.25.

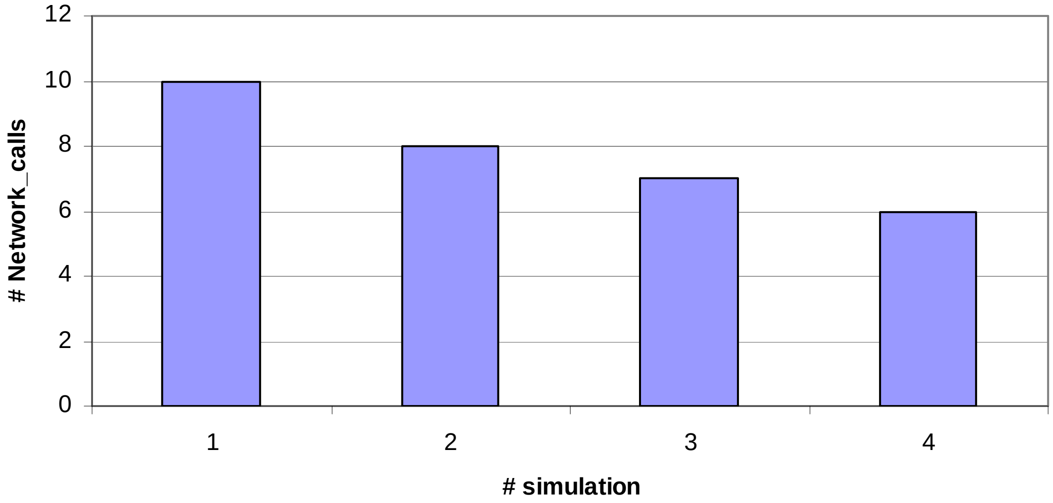


Figure 4.25**.** maximum number of voice calls available in the network vs. # simulations.

Analysing the trend for a chain with different numbers of MAPs, Figure 4.26, it is possible to note that the behaviour of the delay for a residential scenario is quite different relative to the one for a chain network, Figure 4.16.

In fact the delay for a chain network has a cubic or quadratic trend, Table 4.5, instead it is linear for the residential scenario, Table 4.7.

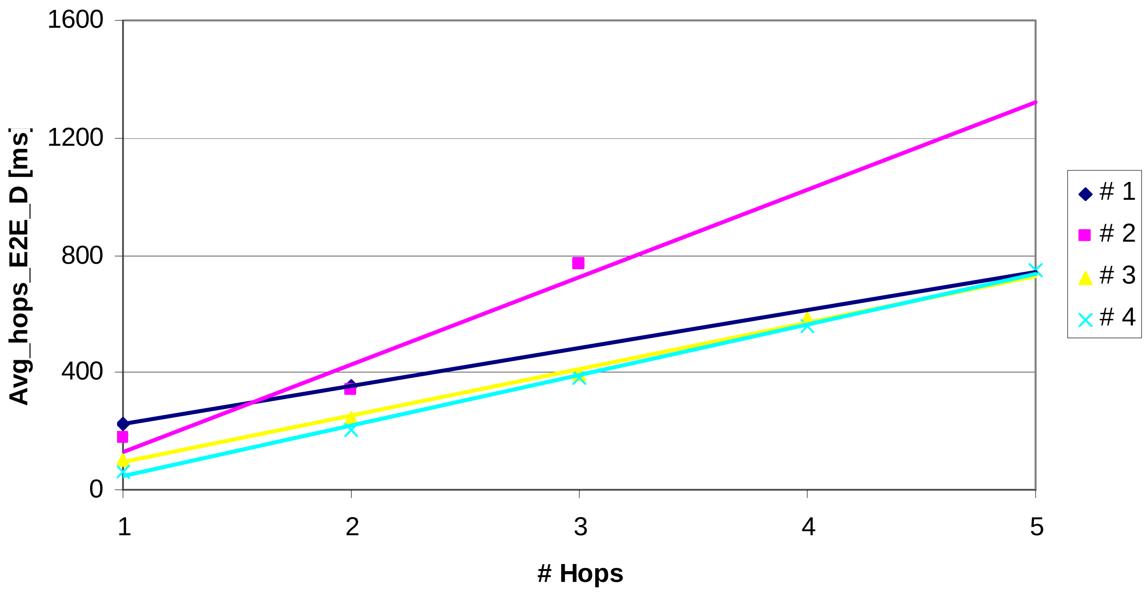


Figure 4.26. end-to-end delay trend in a residential scenario for different density of MAPs.

Table 4.7. Trend Equations and correlations – end-to-end Delay in a residential scenario.

# simulation	Equation	R^2
1	y = 129.07x + 97.26	1.000
2	y = 297.85x - 168.25	0.941
3	y = 159.2x - 66.5	0.995
4	y = 172.5x - 127.5	0.997

These different trends can be explained by the fact that in a chain network users have only one path to achieve the gateway, instead of the residential scenario, where the users have more than one possible route to arrive at the gateway, and so the increasing of delay over the number of hops is softer. However, there are other results that show how the network takes advantage of the increase of the numbers of hops, Figure 4.27.

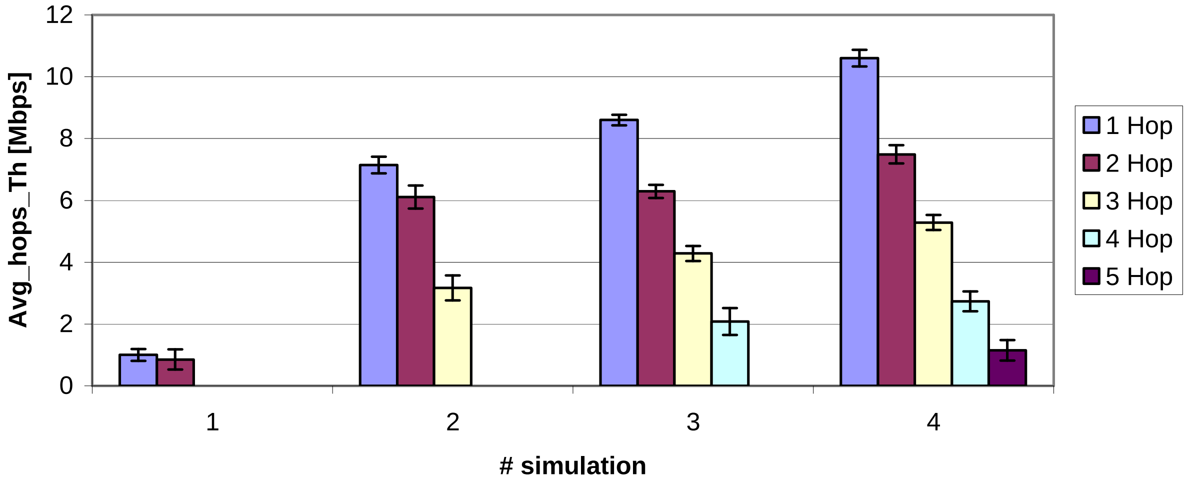


Figure 4.27. different number of Hops – Average Throughput vs # MAPs.

In the study of the average throughput, it is possible to see that when there are only 3 MAPs that have to cover all the whole residential area, the density of users per MAPs is higher, and the available bandwidth should be shared by many users, but by increasing the numbers of MAPs the data rate increases and the number of users per MAPs decreases which means that each user can have more bandwidth available.

In a residential scenario, it is not expected a high traffic load level, so a relevant bottleneck condition close to the gateway does not exist but if the traffic condition changes, it is useful to introduce more than one gateway in order not to have congestion problems near the gateway.

Dividing Avg_hops_Th by the BSS data rate used for each simulation, it is possible to show the throughput in relative terms, Figure 4.28.

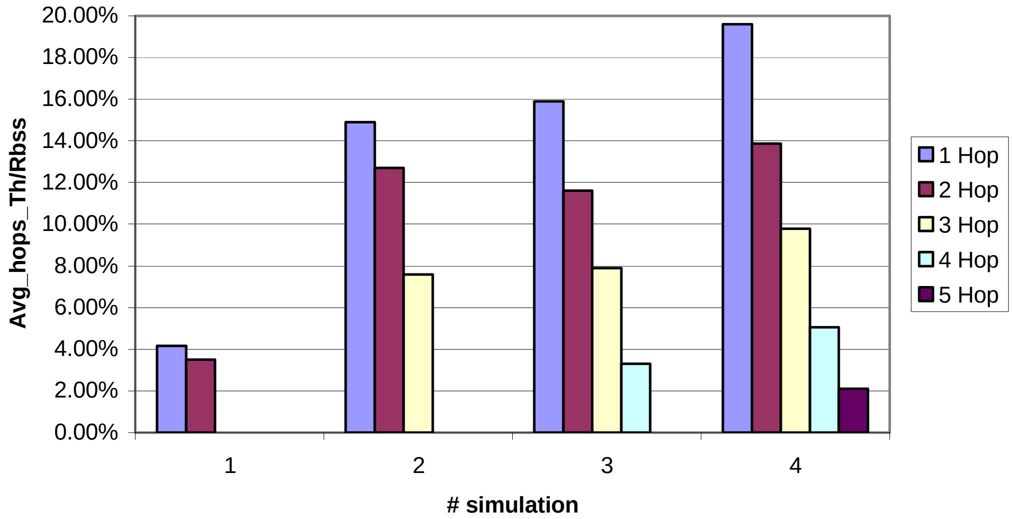


Figure 4.28. Average throughput in relative terms in a residential scenario.

In relative terms the throughput in a residential network increases with the increase of data rate. So as shown for the chain scenario, in a residential scenario a high data rate improves the throughput in the network in relative terms too. This mean that a two dimensional network gives more opportunity for users to send packets.

On the residential scenario, analysing on the different results for each density of MAPs, it is possible to conclude that the network with only 3 MAPs allows each user to do a call, but the performance in terms of throughput is very low, and the network with the maximum number of hops is the best in terms of throughput, but it can not guarantee a good quality call for each user, and moreover the economic cost to implement this network can be very high compared with the numbers of users subscribers. A good trade-off can be a network where the density of the MAPs can guarantee good throughput, and also the possibility to use real-time service for almost the total number of the users in the network.

Chapter 5

Conclusions

This chapter discusses what has been accomplished, and what remains to be addressed in the future.

In this thesis, an analysis of capacity and performance in WMNs is performed. This analysis is made for real time and no real times services, starting with the study of an easy scenario, and going forward analysing scenarios increasingly close to a real WMN for a residential scenario, which enables one to have a good know-how about capacity and performance in a multi-hop WMNs .

802.11 is studied more focused on the aspects that influence the performance of the network. The basic aspects of WLANs and WMNs are described, as well asthe services that can support and some example of mesh scenario that are already a reality in many countries. The OPNET Modeler simulation engine and its wireless module was used.

To make the implementation a WMN more real, a new propagation model was implemented, modifying the free-space model that is already in OPNET. This was done because the available propagation models are not useful for a residential scenario that works at 2.4 and 5 GHz. With OPNET, it is possible to develop, a custom propagation model, using one of the standard propagation models.

The first basic scenario is composed of only two MAPs that have the same offered load sent to the gateway, using 14 different file size values, the objective being to show the maximum average available throughput.

The second basic scenario is a chain network formed by 8 MAPs and 8 users, one for each MAP, which can be taken as a study of an ideal street coverage. Three different traffic condition are simulated for three different file size:0.5 MB, 5 MB, and 50 MB. Voice traffic is also studied, showing the increase in delay with the numbers of hops, and how many users per hop can do an acceptable quality call, for different data rates.

In the third basic scenario the behaviour of the network is studied in terms of throughput and delay ,with the variation of the number of MAPs in the same area.

A residential scenario is composed mostly of residential houses, surrounded by streets and trees. This scenario extends itself over a large square area of 1 km covered by a large backhaul network. Each MAP covers a specific area, where small communities of 30 residential users use a set of services. The number of users is relatively low, and the usage of services is not intense. Typically a residential user is in idle state for the most of the time, so the idea in this work was to evaluate the performance with the assumption that after a time without transmission all users start to use an application, so that in the network there are different data traffic with different priority.

The applications used in this thesis are: VoIP, HTTP, FTP and Email. The maximum distance for a wireless link at minimum data rate is 1050 m, this value being taken as a reference to draw the residential scenario analysed.

All these variations lead to the definition of 69 different scenarios, which amounts to a total of 690 simulations (10 simulations per scenario). Some of the scenarios are very computationally demanding and, the total simulation time reached more than 15 full days, using a HP Pavilion dv 1000 1.256 GB of RAM, with all these data occupying around 1 GB.

Results from the first basic scenario show that, for a certain file size, the throughput for each node is different, since the network cannot forward all data enqueued at a node close to the gateway, and the queue starts to overflow.

Results from the second scenario show that, for a fixed number of hops, increasing the data rate increases the throughput and decreases the delay but this behaviour is less evident for a user that is far away from the gateway. By putting more traffic in the network, the degradation of throughput is increasingly seen, because with heavy traffic conditions there are more possibilities for a user to find the channel busy. Moreover as shown by changing traffic conditions, the number of flows supported by the network is mostly influenced by the packet sending rate, and not by the data rate or packet size.

Results from the third scenario show that inserting more MAPs between extreme points of the network implies that less users can have acceptable quality calls but for each of them there is an increase in terms of throughput, derived by the higher data rates available and by the reduction of the distance between the MAPs.

Based on all these observations, one may add some conclusions regarding the implementation of a residential network with only one gateway at one side of the network: it is not useful to draw a network with more than 5 hops, considering the performance in terms of delay and throughput.

Finally, analysing the Residential Scenario, it is possible to see the improvements brought to the network in terms of performance by the implementation of a two dimensional topology network. The possibility for a packet to have more than one path to arrive to the gateway attenuates the delay of the packet that does not increase very quickly with the number of hop, but the impossibility to have acceptable quality call remains for users that is four hops away from the gateway.

The decision to simulate small scenarios, under 1 km^2, is derived from the fact that OPNET can solve a big network but it takes a lot of time, so the idea was to simulate a little residential village, although the work is useful also to understand the behaviour of a large area, because, e.g.,a large residential area formed by thousands of users and hundreds of MAPs can be made up with many replicas of the residential scenario analysed in this thesis.

The main conclusion for this work is that IEEE 802.11 was designed to work in a single-hop network. When several nodes are in the same contention area, the standard requires each node to pick a random backoff from the same window size. This translates into a good distribution of capacity for various nodes in a single-hop network.

In a WMN, due to its traffic pattern, all traffic is either to or from a designated gateway; each node along the path has to relay other traffic of nodes as well as transmitting its own traffic. This causes extra channel contention between a own traffic of node and its relayed traffic, besides the originally existing traffic with other nodes for the same gateway.

When farther away a node is from the gateway, more hops its data has to go through to reach the gateway. This node will have the higher chance its data will encounter collisions, queueing delay and loss, and the lower throughput. This causes the phenomenon that the nodes close to a gateway starve the nodes further away from the gateway in WMNs.

This phenomenon is known as unfairness, so there is not equal distribution of capacity among the users in a multi-hop WMN, so users that are one hop away from the gateway can have good quality real-time services, but the users that are farther away can not use the same services with the same performance. Although WMNs can be built up based on existing technologies, the simulations done prove that the performance of WMNs is still far below expectations.

New metrics are needed to adapt 802.11 to a multi-hop context, which is the work being developed by 802.11s group, and one of the objectives of the new standard is to ensure fairness in a WMN.

Regarding future work, one suggests the following:

- Redo the range study, putting directional antennas for the backhaul coverage.
- To study a very heavy network, and find if the bottleneck phenomenon is found close to the gateway, and to see if using more than 1 gateway can mitigate or solve the problem.
- To analyse the same scenario by putting more than 1 gateway in different locationa in the network repeat the analysis, and check the different results.
- To integrate the assumption that all user traffic go to and come from the gateway with the possibility to have some portion of traffic that is destined to another local user node instead of the gateway. Such localised traffic may occur when user nodes are somehow correlated, e.g.,different buildings in an area that frequently exchange data over the network.
- To include more real-time services, like Video Conference, and Video Streaming, with different priorities and to implement the peer to peer service, to have a more real and complete study of the services that are used in a residential scenario.

Annex A

Performance Metrics

In this annex, a complete overview of all metrics shown in the results is presented.

The metrics used to analyse the results shown in this work are listed below:

- User_Th: Represents the total number of bits (in Mbps) received at the application level when there is only one user below each MAP.

- Avg_hops_Th: Represents the average of the total number of bits (in Mbps) received at the application level by each user that is at same distance to the gateway in term of number of hops.

- Avg_Network_Th: Represents the average of the total number of bits (in Mbps) received at the application level by all the user that are in the same WMN.

- User_E2E_D: Represents the total voice packet delay (in seconds) called "analog-to-analog" or "mouth-to-ear" delay experimented when there is only one user below each MAP.

- Avg_hops_E2E_D: Represents the average of the total voice packet delay (in seconds) called "analog-to-analog" or "mouth-to-ear" delay experiment by each user that is at same distance to the gateway in term of number of hops.

- # hops_calls: Represents the maximum number of simultaneous calls with an acceptable quality at the same distance to the gateway in term of number of hops.

- # Network_calls: Represents the maximum number of simultaneous calls with an acceptable quality of all the user that are in the same WMN.

- # calls_per_h: Represents the total number of user calls that the system can support in one hour with 1% of blocking probability.

Annex B

Applications parameters

In this annex, a complete overview of applications parameters is presented. All 4 applications used in this work are presented, with all the values necessary to characterise them in Modeler.

VoIP is one of the applications that can be implemented in Modeler, and some parameters are used to characterise it, Table B.1.

Table B.1. VoIP attributes.

Attribute	Definition	Value
Incoming Silent Length [s]	Defines the time in seconds spent in silence mode by the called party.	exponential(0.456)
Outgoing Silent Length [s]	Defines the time in seconds spent in silence mode by the calling party.	exponential(0.456)
Incoming Talk Spurt Length [s]	Defines the time in seconds spent in speech mode by the called party.	exponential(0.854)
Outgoing Talk Spurt Length [s]	Defines the time in seconds spent in speech mode by the calling party.	exponential(0.854)
Encoder Scheme	Encoder Scheme to be used by the calling and called party.	G.729 A (silence)
Type of Service	Type of Service (ToS) assigned to packets sent from the client. It represents a session attribute which allows packets to be processed faster in IP queues. It is an integer between 0 - 252, 252 being the highest priority.	Interactive Voice (6)
Encoding Delay [s]	This attribute specifies the delay in encoding a voice packet.	0.01
Decoding Delay [s]	This attribute specifies the delay in decoding a voice packet.	0.01
Compression Delay [s]	This attribute specifies the delay in compressing a voice packet.	0.02
Decompression Delay [s]	This attribute specifies the delay in decompressing a voice packet. The total voice packet delay, called "analog-to-analog" or "mouth-to-ear", is given by: delay = network_delay + encoding_delay + decoding_delay + compression_delay + decompression_delay	0.02

Table B.2. HTTP attributes.

Attribute	Definition	Value
Mean Time between Requests [s]	Defines the time between the request of HTTP application.	39.5
Mean Request Volume [kB]	Defines the size in Kbytes of HTTP application.	200.0
HTTP Specification	Specifies HTTP parameters which are particular to the version of HTTP that is being used.	HTTP 1.1
Type of Services	Type of Service (ToS) assigned to packets sent from the client. It represents a session attribute which allows packets to be processed faster in IP queues. It is an integer between 0 - 252, 252 being the highest priority.	Best Effort (0)

Table B.3. Email attributes.

Attribute	Definition	Value
Mean Time between Requests [s]	Defines the time between the request of HTTP application.	600.0
Mean Request Volume [kB]	Defines the size in Kbytes of HTTP application.	100.0
Type of Services	Type of Service (ToS) assigned to packets sent from the client. It represents a session attribute which allows packets to be processed faster in IP queues. It is an integer between 0 - 252, 252 being the highest priority.	Background (1)

Table B.4. FTP attributes.

Attribute	Definition	Value
Command Mix (Get/Total)	Denotes the percentage of file "get" commands to the total FTP commands. The remaining percent of the commands are FTP file "put" transactions.	95%
Inter-Request Time [s]	Defines the amount of time between file transfers. The start time for a file transfer session is computed by adding the inter-request time to the time that the previous file transfer started.	constant(1000)
File Size [Kbytes]	Defines the size in bytes of a file transfer.	constant(5000)
Type of Service	Type of Service (ToS) assigned to packets sent from the client. It represents a session attribute which allows packets to be processed faster in IP queues. It is an integer between 0 - 252, 252 being the highest priority.	Best Effort (0)
Encoder Scheme	Encoder Scheme to be used by the calling and called party.	G.729 A (silence)

Annex C

Applications Profiles

In this annex, a complete overview of the applications profile is presented.

Each of the four applications was defined in terms of the size of downloaded files, codecs used, bit rates, and so on. In order to fully characterise an application, one must also define usage profiles that describe how applications behave through time, e.g., when they start, their repetition, or their duration. Each of the applications has its own usage profile defined, but all these values can be changed. Some of the parameters, like the application start time or the repetition of the application profile itself, were defined equally in all profiles. Others, like the applications duration or repetition in the profile, were defined accordingly in each application, Table C.1.

Table C.1. Application profiles.

	VoIP	***HTTP***	***Email***	***FTP***
Operation Mode	Serial (Ordered)	Serial (Ordered)	Serial (Ordered)	Serial (Ordered)
Start Time [s]	exponential (5)	exponential (5)	exponential (5)	exponential (5)
Profile Repetition	None	None	None	None
Duration [s]	Constant (0,180)	End of Profile	End of Profile	End of Profile

References

[Abou06] O. Aboul-Magd, *Joint SEE-Mesh /Wi-Mesh Proposal to 802.11 TGs*, Feb. 2006 (ftp://ftp.802wirelessworld.com/1/06/11-06-0328-00-000-s-joint-seemesh-wimesh-proposal-to-802.11-tgs.doc)

[AkWW04] I. F. Akyildiz, X. Wang and W. Wang, "Wireless mesh networks: a survey", *Elsevier Journal of Computer Networks 47*, Vol. 47,No 4, Mar 2005, pp.445-487.

[CMZW07] M. Cao, W. Ma, Q. Zhang and X. Wang,"Analysis of IEEE 802.16 Mesh Mode Scheduler Performance", IEEE *Transactions on Wireless Communications*, Vol. 6, No. 4, Apr. 2007, pp.1455-1464.

[CRSK06] J. Camp, J. Robinson, C. Steger and E. Knightly, " Measurement Driven Deployment of a Two-Tier Urban Mesh Access Network ", in *Proc of. ACM Mobisys*, Rice University, Houston, TX, USA, Dec. 2006.

[CISC03] http://www.cisco.com, Oct. 2007.

[FCFN07] L.S. Ferreira, L. Caeiro, M. Ferreira and M.S. Nunes, *Qos Performance Evaluation of a WLAN Mesh vs. A WIMAX Solution for an Isolated Village Scenario*, Instituto Superior Técnico/Instituto de Telecomunicações, Technical University of Lisbon, Lisbon, Portugal, 2007.

[GaVe02] M. Gastpar and M. Vetterli, "On the Capacity of Wireless Networks: The Relay Case", in *Proc of. IEEE INFOCOM*, NY, USA, June 2002.

[GrTs01] M. Grossglauser and D. Tse, "Mobility Can Increase the Capacity of Ad-hoc Wireless Networks," in *Proc of. IEEE INFOCOM*, Anchorage, Alaska, Apr. 2001.

[GuKu00] P. Gupta and P. R. Kumar, "The Capacity of Wireless Networks", *IEEE Transactions on Information Theory*, Vol. 46, No. 2, Mar. 2000, pp. 338-404 .

[HiKW03] G.R. Hiertz, O. Klein and B. Walke, "Analysis of IEEE 802.11e for QoS support in wireless LANs", *IEEE Wireless Communications*, Vol.10, No 6, Dec. 2003, pp.40-50.

[HMZD07] G.R. Hiertz, S. Max, R. Zhao, D. Denteneer and L. Berlemann, "Principles of IEEE 802.11s", in *Proc of. of ICCN: International Conference on Computer Networks, Hawaii*, USA, Aug. 2007.

[HoLe08] E. Hossain and Kin Leung, *Wireless Mesh Networks Architectures and Protocols*, Springer, NY, USA, 2008.

[IEEE03] IEEE, Wireless LAN Medium Access Control (MAC) and Physical Lay (PHY) specifications, IEEE Standard 802.11-99, 2003.(http://www.ieee.org)

[JuPe03] J. Jun and P. Peddabachagari, “Theoretical maximum throughput of IEEE 802.11 and its applications”, in *Proc. Network Computing and Applications*, Raleigh, NC, USA, Apr. 2003.

[JuSi03] J. Jun and M.L. Sichitiu, “The nominal capacity of wireless mesh networks”, *IEEE Wireless Communications*, Vol. 10, No 5, Oct 2003, pp. 8-14.

[LBCL01] J. Li, C. Blake, D.S.J. De Couto, H.I. Lee and R. Morris, “Capacity of Ad Hc wireless networks”, *Conference on Mobile Computing and Networking*, Rome, Italy, 2001, pp.61-69.

[MaBe05] S. Mangold, L. Berlemann, “IEEE 802.11k: Improving Confidence in Radio Resource Measurements”, *IEEE 16th International Symposium on Personal, Indoor and Mobile Radio Communications*, Sep. 2005.

[MaSu03] S. Mangold, and C. Sunghyun, “Analysis of IEEE 802.11e for QoS support in wireless LANs”, *IEEE Wireless Communications*, Vol. 10, No 6, Dic. 2003, pp. 40-50.

[MGGK04] M. Madepalli, P. Gopalakrishnan, D. Famolari and T. Kodama, ‘‘Voice capacity of IEEE 802.11b, 802.11a and 802.11g wireless LANs,’’ in *Proc of of the IEEE Global Telecommunications Conference (GLOBECOM)*, Dallas, Texas, USA, Dec. 2004.

[OLPC07] http://laptop.org Nov. 2007.

[OPNT07] OPNET Modeller Wireless Suite simulation tool, http://www.opnet.com, Nov.2007.

[QSBB05] E. Qi, K. Sood, S. Bangolae and C. Bell "Issues with real-time streaming applications roaming in QoS-based secure IEEE 802.11 WLANs", *Conference on Mobile Technology Applications and Systems*, May 2005.

[Rapa96] T.S. Rappaport, *Wireless Communications Principles and Practice*, Prentice Hall, New Jersey, USA 1996.

[VaKo05] D. Vassis, G. Kormentzas, A. Rouskas and I. Maglogiannis, ”The IEEE 802.11g standard for high data rate WLANs", *IEEE Network*, Vol. 19, No 3, June 2005, pp. 21-26.

[Voni07] http://www.vonitaly.com Oct 2007.

[WaMB06] B. Walke, S. Mangold, and L. Berlemann, *IEEE 802 Wireless System*, John Wiley & Sons, Ltd, 2006, West Sussex, UK, 2006.

[WCIT06] http://interno.ehas.org/publicaciones/congresos/wcit_2006.pdf Nov. 2007.

[Whit00] P. Whitehead, "Mesh networks: a new architecture for broadband wireless access systems", in *Proc of. IEEE Conference on Radio and Wireless (RAWCON)*, Denver, CO, USA, Sep. 2000.

[WIKI03] http://www.wikipedia.org Oct. 2007.

[Xiao05] Y. Xiao, "IEEE 802.11n: enhancements for higher throughput in wireless LANs", *IEEE Wireless Communications*, Vol. 19, No 6, Dec. 2005, pp. 82-91.