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ABSTRACT

The purpose of this project is to analyse the Potential for Secondary Communication on existing wireless systems.

This is accomplished by reordering of users resources, within the frame, and without interfering in the primary communication. Permutations tables which satisfy the Hamming distance are designed and an error correction code is developed. Two wireless systems are chosen for investigation: WiFi and WiMAX. For WiFi different frames are studied and the most appropriate one is used to apply Secondary Communication. In WiMAX two scenarios are developed and the potential is analysed on each of them considering range and number of users.

Calculations and data analysis are done by using an analytical approach, while simulations are coded in Matlab. The thesis concludes with an overview of the project, as well as a summary of the results obtained to better emphasize the Secondary Communication benefits.

Preface

This report has been written by group 1200 in 2010 - 2011 at Aalborg University.

Literature references follow IEEE recommendations. Texts, figures, formulas and tables are referenced using number in brackets which indicates the position on the reference list:

Text [Reference Number]
Figure (number): Figure Description [Reference Number]
Table (number): Table Description [Reference Number]
[Reference Number]: Formula [units]

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Mobile Communication - Group 1200
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Abbreviation List

AAS	Adaptive Antenna System
ACK	Acknowledge
AMC	Adaptive Modulation and Coding
ASCA	Adjacent Sucarrier Allocation
ATM	Asynchronous Transfer Mode
AWGN	Additive White Gaussian Noise
BE	Best Effort
BER	Bit Error Rate
BER_{nd}	BER for n users and with Hamming distance d
BS	Base Station
BSS	Basic Service Set
BSSID	Basic Service Set ID
CID	Connection Identifier
CQI	Channel Quality Information
CRC	Cyclic Redundancy Check
CS	Convergence Sublayer
CTC	Convolutional Turbo Code
CTS	Clear to Send
d	Hamming distance
d’	Number of Errors
DA	Destination Address
dB	Decibel
DBPSK	Differential Binary Phase Shift Keying
DCD	Downlink Channel Descriptor
DIUC	Downlink Interval Usage Code
DL	Down-Link
DSL	Digital Subscriber Line
DSSS	Direct Sequence Spread Spectrum
EC	Encryption Control
ECRTP	Enhanced Compressed Real-Time Protocol
EIRP	Effective Isotropic Radiated Power
EKS	Encryption key sequence
ertPS	Extended rtPS
ESF	Extended Subheader field
Es/No	Energy per symbol per Noise power spectral density
FCS	Frame Check Sequence

FCH	Frame Control Header
FDD	Frequency Division Duplexing
FEC	Forward Error Correction
FFT	Fast Fourier Transform
FTP	File Transfer Protocol
FUSC	Full Usage of Subchannels
GHz	Gigahertz
GMH	Generic MPDU Header
GSM	Global System for Mobile Communications; originally from Groupe Special Mobile
HARQ	Hybrid Automatic Repeat Request
HCS	Header Check Sequence
H-FDD	Half-duplex Frequency Division Duplex
HT	Header Type
ID	Identifier
IE	Information Element
IEEE	Institute of Electrical and Electronics Engineers
IP	Internet Protocol
ITU	International Telecommunication Union
kbps	kilobits per second
km	kilometer
LB	Licensed Bands
LeB	Licensed-exempt Bands
LEN	Length
LOS	Line of Sight
LSB	Least Significant Bit
m	meter
MAC	Media Access Control
MAP	Map Message
Mbps	Megabits per second
MCS	Modulation Coding Scheme
MPDU	MAC Protocol Data Unit
MPEG	Moving Pictures Experts Group
MS	Mobile Station
MSB	Most Significant Bit
n	Number of Users
nrtPS	Non-Real-Time Polling Service
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
OSI	Open Systems Interconnection
PDU	Protocol Data Unit
PER	Packet Error Rate
PHY	Physical Layer
PHS	Payload Header Suppression
PHSF	Payload Header Suppression Field
PHSI	Payload Header Suppression Index
PHSM	Payload Header Suppression Mask
PHSS	Payload Header Suppression Size

PLCP	Physical Layer Convergence Procedure
PMD	Physical Medium Dependent
PMP	Point-to-Multipoint
PPDU	PHY Protocol Data Units
PUSC	Partial Usage of Subchannels
P(n,d)	Permutations Table for n users and d Hamming distance
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature Phase-Shift Keying
RA	Receiver Address
RTG	Receive Transition Gap
rtPS	Real-Time Polling Service
RTP	Real-Time Transport Protocol
RTS	Request to Send
ROHC	Robust Header Compression
SA	Sender Address
SAD	Speech Activity Detection
SAP	Service Access Point
SC	Single Carrier
SDU	Service Data Unit
SFID	Service Flow Identifiers
SS	Subscriber Station
STC	Space Time Coding
TA	Transmitter Address
TDD	Time Division Duplexing
TTG	Transmit Transition Gap
TUSC	Tile Usage of Subchannels
u_1	User 1
u_2	User 2
u_3	User 3
u_4	User 4
UCD	Uplink Channel Descriptor
UDP	User Datagram Protocol
UGS	Unsolicited Grant Service
UIUC	Uplink Interval Usage Code
UL	UP-Link
VAD	Voice Activity Detection
VoIP	Voice over IP
Wi-Fi	Wireless Fidelity
WiMAX	Worldwide Interoperability for Microwave Access
WirelessHUMAN	Wireless HIGH-speed Unlicensed Metropolitan Area Networks
WirelessMAN	Wireless Metropolitan Area Networks
WLAN	Wireless Local Area Network
WMAN	Wireless Metropolitan Area Network
WPAN	Wireless Personal Area Network
3G	3^{rd} Generation
4G	4^{th} Generation
ε	Error

Symbol List

$a_n a_{n-1} ... a_2 a_1 a_0$	Factoradic Number
A_0	The permutation with ascending ordering of 5 users
A_1	Random Permutation Sequence for 5 Users
b	Number of Extra Bits Available
B	Bytes per Frame
$bits$	number of extra bits available
$BytesperSlot$	Number of bytes per Slot
c	Path Loss Coefficient
$CodingRate$	Coding Rate
d	Distance/Range
D	Data Size per Frame
d_0	Reference Distance
$Data_Rate$	Data Rate
$DataSubcarriers$	Number of Data Subcarriers
$DLMAP$	The Fixed Size of the DL MAP = 11 bytes
$DLMAP(bytes)$	Downlink MAP Message length
$DLMAP(slots)$	Number of Slots Downlink MAP
$DLslots$	Number of Available slots for the DL Subframe
$DLusers$	Number of Users Schedule for DL in the Frame
$DR_{primary}$	Data Rate for Primary Communication
$DR_{secondary}$	Data Rate for Secondary Communication
$EmptySlots$	Number of Empty Slots
Es/No	Energy per symbol per Noise power spectral density
G_r	Received Antenna Gain
G_t	Transmitted Antenna Gain
$Header$	Header length
i	Index
MAC_{header}	6 bytes
$MPDU$	MAC Packet Data Units length
$MSDU$	MAC Service Data Units Length
n	Number of Users
$NumberofBitsperSymbol$	Number of bits per Symbol
P_{noise}	Noise Power
P_r	Received Power
P_t	Transmitted Power

$PLCP_{Header}$	PLCP Header length
$PLCP_{Preamble}$	PLCP Preamble length
r	Repetition Factor
R	Data Rate
RTS_{length}	RTS frame length
$R_{withheader}$	Data Rate for MAC frame plus header
S_i	Slot Size, given the i^{th} modulation and coding scheme
S_k	Size of the Slot Given the k^{th} MCS
SNR	Signal to Noise Ratio
SNR_0	Signal to Noise Ratio
$Subheaders$	Fragmentation and Packing Subheaders 2 bytes Each
$SymbolperSlot$	Number of Symbols per Slot
t_{perm}	Time of Transmitting a Permutation
t_{RTS}	Time to Transmit an RTS
$ULMAP$	The Fixed Size of the UL MAP = 6 bytes
$ULMAP(bytes)$	Uplink MAP Message length
$ULMAP(slots)$	Number of Slots for Uplink MAP
$ULslots$	Number of Available Slots for the UL Subframe
$ULusers$	Number of Users Schedule for UL in the Frame
λ	Wavelength
π	π Constant = 3.14

Chapter 1

INTRODUCTION

In the past decade the Internet has experienced a growth without precedent, in the number of services offered and the users that access it. Along with it, the demand for higher speed, easy and affordable access has been following a similar trend, forcing service providers and hardware manufacturers to develop and implement new technologies at a fast pace.

There are a number of associations that monitor this numbers and the most relevant one is International Telecommunication Union (ITU) which offers statistics and market research in the most relevant telecommunication related topics [?]. In Fig. 1.1 it can be seen that in a period of 10 years the number of users has grown from 500 million to 2 billion. As expected the demand for broadband access has also experienced a boost, the fixed broadband from approximately 2% in 2000 to 24.6% in 2010, while the mobile broadband from 0% to 51.1% in the same period.

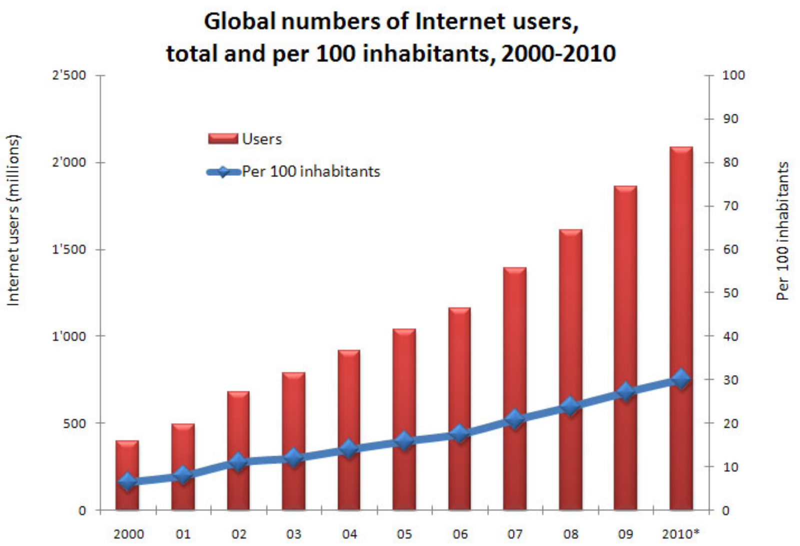

Figure 1.1: Internet users [?]

A similar trend has been reported also in the number of mobile cellular users, which in 2010 reached 5.3 billion subscribers most of them being in the developing world, although there are 116.1 subscriptions to 100 users in the developed world in 2010 according to ITU.

This statistics show a clear and growing interest of the general public in the new communication and information technologies and also open the way for new and innovative companies and ideas. The broadband access was clearly dominated by Digital Subscriber Line (DSL) and cable modem technologies, but in the recent years the interest grew in wireless solution that can provide broadband to the end user.

There are a number of wireless networks available at the moment and they can be classified in a number of ways, but the most commonly used is by the range they cover. Fig. 1.2 reflects this aspect, the smallest covered area corresponds to the Wireless Personal Area Network (WPAN) which holds the devices that are use for communications within the reach of one person, normally 10 meters, an example being bluetooth technology. Bigger areas can be covered by Wi-Fi and WiMAX systems which correspond to Local and Metropolitan Area Networks. Besides the concern about range, bigger data rates is the main characteristic that each wireless manufacturer wants to obtain, because it gives a wide range of possibilities to the end user: internet navigation, gaming, streaming etc., and also gives the service provider the possibility to have more subscribers using the same resources.

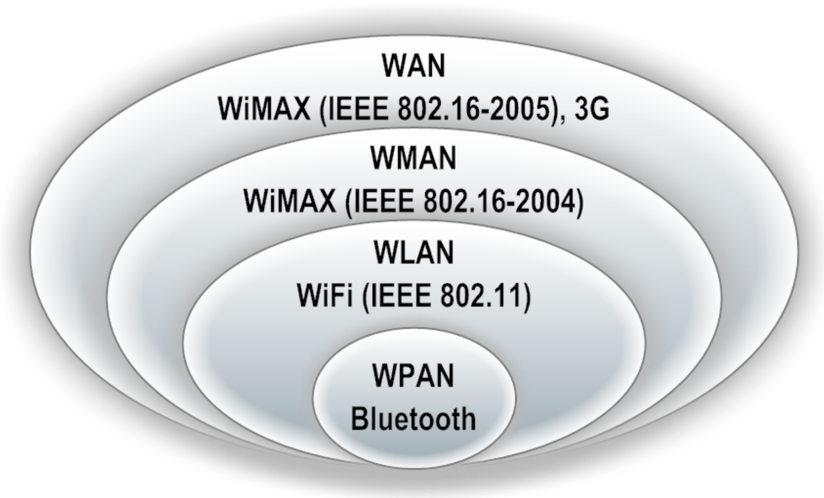

Figure 1.2: Wireless network types according to the range they cover, with common examples

As stated in the beginning of the Chapter, due to high demand new technologies are developed and implemented in a fast pace. Some of them make it and become huge success stories like WiFi and 3G but some of them do not receive the same popularity and ultimately discarded. Even the ones that are adopted eventually become obsolete, and improvements have to be made for them in order to survive in a competitive market. This improvements can be software related, which means that the current hardware can be kept and the manufacturer has to only update it, or hardware related which means that most of the equipment used to handle the old versions has to be modified or replaced.

In this thesis we will focus on the software modifications that can be brought to the existing wireless deployments due to the fact that they are much cheaper to implement and have the added bonus that neither the manufacturer or the end-user have to make changes to their terminals to receive a better service.

1.1 Project Definition

While researching for a way to improve the existing wireless systems, a new method that took advantage of the vast infrastructure already deployed without significant changes to it was presented in [4]. The authors of this paper propose sending additional bits by utilizing the protocol overhead information in existing systems which they name primary systems. By rearranging the packets that are going to be sent over the wireless channel additional information can be sent, receiving the name of Secondary Communication.

In [4] the authors explain the concept on all frame based communication without taken into consideration the restriction of a particular system and present some of the benefits of implementing it. In this thesis we are going to analyse the potential for Secondary Communication in close relation to a particular standard and its requirements. In consequence a choice has to be made between all existing wireless systems, its protocols analysed, in order to find the best place to make the modifications necessary to send additional information and then, having the imposed restrictions in mind, calculate the Secondary Communication data rate.

The most important aspects that we took into consideration when choosing the system that is going to be analysed were the number of users it has and the expected evolution in the following years. A high number of users is crucial, because it offers a big market in demand for higher data rates, reliable transmissions and extended coverage areas. Fulfilling this requirement are a number of systems like GSM which according to [5] has over 3.5 billion users in 2009, WiMAX with 620 million users according to [6] and WiFi for which a official number is not available. In order to decide between this three the second requirement has to be taken into consideration: expected evolution. Although by far GSM has the most users, it is an obsolete technology allowing only voice calls and short text messages, being replaced by 3G and soon 4G technologies which contain also WiMAX with the standard 802.16m which is still in the development phase. Regarding WiFi, since its first release it experienced a wide acceptance in the industry and successors have surpassed the market expectations. Its latest standard 802.11 n promises higher data rates, larger coverage areas and better quality, which leads to the conclusion that unlike GSM, WiFi has a sturdy future.

After this short analyses we have concluded that not one but two systems fulfil the requirements decided when choosing the system to investigate further. In consequence we will continue this thesis with the analyses of both systems in order to calculate the potential for Secondary Communication.

1.2 Project Outline

The present Master Thesis is based on an IEEE paper [4] which describes a new way to transmit extra information using the existing wireless systems, with minimum modifications and without interfering with the main communication.

Chapter 1 briefly describes the existing wireless networks, their growth in the last decade and their applicability and usability. The same chapter states the project definition and the goals of the investigation, concluding with the chosen wireless systems to be investigated for the implementation of Secondary Communication, using reordering of users resources.

Beginning with a short presentation of the main aspect of this concept - permutations - the construction of the optimum permutations table is described, as well as the proposed mapping algorithm and error correction code. All these can be found in Chapter 2.

The next two chapters (Chapter 3 and Chapter 4) present the Secondary Communication concept in specific wireless systems: Wi-Fi and WiMAX, starting with a brief description of each system and the most relevant aspects of it, that gives the possibility for implementing this new concept. The similarities and the differences between how extra information can be sent using Wi-Fi and WiMAX systems emphasize the flexibility and the adaptability of the method. Starting from permutation concept and analysing the structure of each system, the restrictions of the protocol and the degree of freedom, the Secondary Communication solutions take shape and the results for each wireless system are provided.

The thesis will end with a chapter dedicated to conclusions, which will briefly present the main results obtained, the benefits and the disadvantages of the solution proposed for each wireless system.

Chapter 2

SECONDARY COMMUNICATION THROUGH REORDERING OF USERS RESOURCES

In order to better understand the concept of Secondary Communication the authors of **??** offered an example that will be explained as a starting point of this chapter.

Considering a cellular mobile system that has 3 channels (C1, C2, C3) with 3 users (M1, M2, M3), and that the users are not obligated to use a particular channel and their allocation is announced on a per frame basis by the Base Station in a control header. Given a specific frame this allocation can be M1 $->$ C1, M2 $->$ C2, M3 $->$ C3, but if it is changed it will not have an impact on the primary communication. There are 3! ways in which this allocations can be changed but we are going to need only 4 of them to illustrate the concept. To each of this 4 allocation 2 bits can be associated in the following way:

Permutation	Coding
123	00
132	01
213	10
312	11

Table 2.1: Permutation coding

By rearranging the allocations at every frame 2 extra bits of data can be sent without interfering with the primary communication, thus creating a new channel. This is the basis of Secondary Communication.

The purpose of this thesis is to investigate the possibility of implementing Secondary Communication in existing wireless systems. Besides sending coded information simultaneously with the primary communication, without any hardware modifications and with easily implementation in the existing wireless systems, Secondary Communication can be used as the last mile solution for users that are out of the coverage range of a Base Station and can be easily implemented in smart houses (for example using this extra communication, opening the main door could turn on the light on the hallway).

How can Secondary Communication be accomplished? As we previously stated in Chapter 1, Secondary Communication can be established by scheduling the users to which we send data as sequence of permutations. The next section will give more information how permutations can be mapped and interpreted and how a permutation can be recovered when the address of one of the users was not correctly received.

2.1 Permutations

Any permutation can be interpreted as a message or part of a message. Let us imagine that we have 5 users in the coverage area. This means 5! possible permutation, if we suppose that all 5 users transmit/receive data. Having 5! possible permutations, means $\lfloor log_2(5!) \rfloor = 6$ extra bits transmitted (see [4]). Knowing all these, two big questions come:

- How can we map a permutation in 6 bits?
- All possible permutations with 5 users can be mapped in 6 bits?

To answer to these immediate questions the next Section 2.1.1 will describe the proposed mapping algorithm.

2.1.1 Permutation Mapping Algorithm

The proposed algorithm to map permutations to bits, follows the next steps:

1. Map permutation to factorial number (factoradic)
2. Convert factorial number to decimal number
3. Convert decimal number to binary number

Next we will describe each step and conclude with the advantages and disadvantaged of the proposed algorithm.

1. Map permutation to factoradic (see [7])

   Taking for example 5 users that can be served simultaneously in a WLAN network, results $5! = 120$ possible orderings of the users.

   The ascending order of the users $A_0 = \{1, 2, 3, 4, 5\}$ is associated with the factoradic 00000!, this being the initial state. Reordering the users in a random way, for example, $A_1 = \{3, 2, 5, 1, 4\}$ can be mapped to factoradic as follows:

(a) The first digit in A_1 - 3 - has been shifted to left with 2 positions with respect to the A_0 (the initial state), so number 2 will be associated with it (representing the first digit from the factoradic)

(b) The next step is to take out 3 from the initial state and then repeat 1a for the next digit from A_1 and so on; at the end resulting the factoradic 21200!, for this specific example.

Because at the last iteration remains only one digit in the vector, the rightmost digit of the factoradic is always 0.

2. Convert factoradic to decimal

   The factoradic number should be seen as a number in a factorial base, and the conversion to decimal can be done using the following formula:

$$a_n a_{n-1}...a_2 a_1 a_0 = \sum_{i=0}^{n} a_i \cdot i \tag{2.1}$$

3. Convert decimal to binary

   First of all we have to establish the number of bits available to send extra data:

$$b = \langle log_2 120 \rangle = 6 \Longrightarrow 111111_2 = 63_{10} \tag{2.2}$$

   possible permutations to send extra data.

   After this is done the decimal number will be represented in a binary base with b bits.

   The steps 1, 2, 3 present an algorithm for mapping permutations to binary. To do vice versa, map binary to permutations, the same steps are used, but in reverse order.

The proposed algorithm to map permutations to binary, even though seems complicate, it presents a simple way how to interpret any permutation as number (in factorial base) and than simply transform it in binary sequence. This is the optimal algorithm, firstly because it associates any permutation with only one binary number and secondly because it gives the possibility to do vice-versa when the number of users is known.

The algorithm has also disadvantages: not all the permutations can be mapped to the number of available bits, so the initial table of permutations with all the possible permutations for 5 users can not be used for Secondary Communication, but the available permutations number is satisfactory for implementing this new concept in the existing wireless systems; in this example 64 permutations from 120 can be mapped with the proposed algorithm.

2.1.2 Permutation Tables

Using the permutation to transmit information is not a new concept, the interest in permutation could being track with many years before. The most representative example is a common electric power line. As it is known, the main function of this is to deliver electric power and this can be achieved even though there could be small variations of the frequency, intentional or not. Creating a family of 'close' frequencies, the small variations of frequency can be decoded by the receiver as symbols. In this way can be transmitted information without interfering with the primary function of the electric power line (see [8]).

To the best of our knowledge the concept of transmitting information using the permutations in wireless systems was first introduced in [4].

The paper presents the conceptual idea of Secondary Communication and some systems where this can successfully be implemented, and the goal of our thesis is to analyze this capacity in wireless systems.

Lets take as example 4 users u_1, u_2, u_3 and u_4. With 4 users there can be 4! possible permutations which are presented in Table 2.2, which will be referred to as initial permutations table.
In the previous Section 2.1.1 we presented how to update the initial permutation table: removing the permutations that can not be mapped in the available bits. The update table can be seen in Table 2.3. So, the permutations that can be used to send extra information are the ones from update table. Let us consider the 3^{rd} permutation from the table: $u_3u_1u_2u_4$.

The question that comes now is: what happens if u_1's and u_3's data is not transmitted, or is transmitted with errors? In this case there are two possibilities. Comparing the permutation $\varepsilon\varepsilon u_2u_4$, where ε represents error, with the permutations from update table:

- it could be fully matched with only one of the permutation, from where we extract which users are instead ε;
- it could match with two or more permutations from the update table, case when a decision should be make, or an error correction code should be apply.

The second case can be solved by imposing a new condition for the update permutations table: the distance between two any permutation should be d$\geq$2 (Hamming distance, see [9]); in this way d-1 errors can be allowed to the received permutation, in order to be able to correct the permutation without any additional error correction code.

The next step is to create the permutations tables which fulfill Hamming distance. This update table will be further referred as P(n,d), where n is the number of users and d represents the Hamming distance. For easily construction of P(n,d) of any n and/or d, we developed a Matlab algorithm that does this automatically. The algorithm constructs the initial permutations table, where each line represents a permutation, deletes the permutations that can not be mapped into $\lfloor log_2(n) \rfloor$ bits, and from these permutations selects

Index	Permutation			
1	u_4	u_3	u_2	u_1
2	u_4	u_3	u_1	u_2
3	u_4	u_2	u_3	u_1
4	u_4	u_2	u_1	u_3
5	u_4	u_1	u_2	u_3
6	u_4	u_1	u_3	u_2
7	u_3	u_4	u_2	u_1
8	u_3	u_4	u_1	u_2
9	u_3	u_2	u_4	u_1
10	u_3	u_2	u_1	u_4
11	u_3	u_1	u_2	u_4
12	u_3	u_1	u_4	u_2
13	u_2	u_3	u_4	u_1
14	u_2	u_3	u_1	u_4
15	u_2	u_4	u_3	u_1
16	u_2	u_4	u_1	u_3
17	u_2	u_1	u_4	u_3
18	u_2	u_1	u_3	u_4
19	u_1	u_3	u_2	u_4
20	u_1	u_3	u_4	u_2
21	u_1	u_2	u_3	u_4
22	u_1	u_2	u_4	u_3
23	u_1	u_4	u_2	u_3
24	u_1	u_4	u_3	u_2

Table 2.2: Initial Permutations Table.

only the permutations that have at least distance d between them. In the same time the algorithm creates the mapping table, which contains on each line the mapping sequence of the corresponding permutation in P(n,d). The algorithm's running time tends to be very high for n>7.

The codes for creating P(n,d) are available on Appendix A.

For the previous example with 4 users, if we want to recover the received permutation, we can see from Table 2.3, that 2 permutations fit with it: permutation from line 3 and permutation from line 12. In this case we need an error correction code that could be able to choose from this 2 permutations.

If we would have use the table with hamming distance d=3 (P(4,3), see Table 2.4) for example, we would have be able to recover the received permutation without applying an error correction code.
Even though the number of permutations available for Secondary Communication is 3 times smaller than the initial one, it is preferable to use P(4,3) instead of update permu-

Index	Permutation			
1	u_3	u_2	u_4	u_1
2	u_3	u_2	u_1	u_4
3	u_3	u_1	u_2	u_4
4	u_3	u_1	u_4	u_2
5	u_2	u_3	u_4	u_1
6	u_2	u_3	u_1	u_4
7	u_2	u_4	u_3	u_1
8	u_2	u_4	u_1	u_3
9	u_2	u_1	u_4	u_3
10	u_2	u_1	u_3	u_4
11	u_1	u_3	u_2	u_4
12	u_1	u_3	u_4	u_2
13	u_1	u_2	u_3	u_4
14	u_1	u_2	u_4	u_3
15	u_1	u_4	u_2	u_3
16	u_1	u_4	u_3	u_2

Table 2.3: Update Permutations Table.

Index	Permutation			
1	u_3	u_2	u_4	u_1
2	u_3	u_1	u_2	u_4
3	u_2	u_3	u_1	u_4
4	u_2	u_4	u_3	u_1
5	u_2	u_1	u_4	u_3
6	u_1	u_3	u_4	u_2
7	u_1	u_2	u_3	u_4
8	u_1	u_4	u_2	u_3

Table 2.4: Permutations Table with Hamming Distance d=3, P(4,3).

tations table because the impose Hamming distance increases the possibility of permutation recovery, thus, Secondary Communication establishes the transmission of information from base station to the users that are far away from it.

2.2 Error Correction Code

In the previous Section 2.1.2 we specify that using permutations tables that satisfy Hamming distance, there is some self-correction - if d distance is imposed, d-1 errors are allowed, in order to be able to correct the received permutation. But, if the received sequence has d or more than d errors the permutation can not be corrected, in this case it is needed to apply an error correction code.

The proposed error correction code divides the algorithm in 3 sections:

1. When the received sequence has less then d errors:

   In this case the error correction code removes the columns where the error/errors appear and compares the received permutation (with errors deleted) with each line of the P(n,d). It will be only one line that matches with the received permutation, so the correction is 100% accurate.

2. When the received sequence has d' errors, where d≤d'<n:

   The process is similar: a new permutations table is constructed with the lines that matches with the received sequence. If the new permutations table has only one line it is full matching, otherwise a decision should be made: choose the first line from the new table. It is obvious that in this situation errors could arise.

3. When the received sequence has n error:

   When the received sequence contains only errors, a decision should be make: choose as permutation the one that has all bits 0 (the permutation that has all the users in ascending order).

Chapter 3

SECONDARY COMMUNICATION IN Wi-Fi SYSTEMS

This chapter offers an overview of IEEE 802.11 protocol, focusing on the relevant parts for possible implementation of Secondary Communication. Besides the general description of the protocol, the chapter will contain relevant data for implementation of Secondary Communication as well as the simulation results and it will end with a section dedicated to conclusions.

3.1 Background

Nowadays we are surrounded by a considerable number of wireless devices from different manufacturers and with different functionalities. The need to make a wireless device compatible with others and to assure that a mobile device (for example a mobile phone) works worldwide without interfering with other devices or with a minimum acceptable interference, constitute the basis of standards.

Standards have a big impact on how we select and use products. A typical example could be the International System of Units used all over the world, except some countries, for example Unites States - where the familiar measurement units are inch, pound, ounce.

Thinking at mobile wireless communication, a big challenge was to convey to a global agreement that will give, for example, to users of mobile phones , the possibility to go from Spain to America and then to Japan and use only one mobile phone.

802.11 Standard was introduced by Institute of Electrical and Electronics Engineers - IEEE - which is an organization comprised of both professionals and students. The area of work of IEEE includes technologies from computing and sustainable energy systems, to aerospace, communications, robotics etc. (for more information see [10]).

IEEE 802.11 Standard was first adopted in 1997, and it became the first WLAN standard. This protocol primarily controls the physical layer and the data-link layer (layers 1 and

2) of the OSI reference stack (see [11]).

After 1997 different amendments were added to the original standard and new standards were created - 802.11a, 802.11b, 802.11g and 802.11n. The most popular are the 802.11b and 802.11g protocols. 802.11b was the first widely accepted protocol, followed by 802.11g. The biggest advantages of 802.11g in comparison with 802.11b, with who is retro-compatible, are the reduction of manufacturing costs and higher data rates, up to 54 Mbps [12].

3.1.1 MAC Layer

The Media Access Control (MAC) is a set of rules which describes how to access the medium and how to send the data using Wi-Fi devices. The transmission and the reception of data is done by Physical Layer (PHY), which is the lowest OSI layer.

Each MAC frame consists of three basic components:

- *MAC header*, which contains frame control, duration, address/addresses and sequence control information;
- *Frame body*, of variable length;
- *FCS* - frame check sequence - which checks the integrity of received frames.

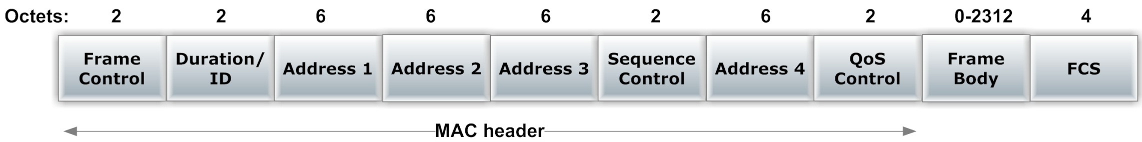

Figure 3.1: MAC Frame Format [1].

The most common name for MAC frame, used in literature, is MAC Protocol Data Unit (MPDU), hence from now one we will refer to it as MPDU. The general MAC frame format could be seen in Fig. 3.1, where the length of each field is expressed in bytes.

1. **Frame Control field.** With a length of 2 bytes, the Frame Control field indicates which version of 802.11 MAC is contained in the rest of the frame and it also identifies the type of frame used.
2. Duration/ID field. This field contains the time, in microseconds, that the medium is expected to remain busy for the currently transmission.
3. **Address fields.** Each wireless card has a 48-bit MAC address. Depending on the frame type, MPDU may contain the destination/source address, receiver/transmitter address or/and basic service set ID (BSSID).

4. Sequence Control field. The 16 bits of this field are split into frame number and sequence number field.

5. **QoS Control field** identifies the traffic category or traffic stream to which the frame belong and gives other QoS-related information about the frame that is transmitted, depending on the type of the frame.

6. **Frame Body field** contains the data which will be transmitted and has a variable length: 0-2312 bytes.

7. **FCS field**, sometimes referred as cyclic redundancy check (CRC), compares the received frame with the transmitted frame in order to detect errors.

The 802.11 protocol (see [1]) defines different types of MPDUs which could be grouped in three big categories: control frames, management frames and data frames. The next pages will provide an overview of different types of MPDUs in order to make a choice - which MPDU type should be further evaluated for Secondary Communication.

The most relevant frame types for our study are :

1. **Control frames** - assist the delivery of data frames between stations. The most used control frames are:

   - **Request to Send/Clear to Send (RTS/CTS) frames** (see Fig. 3.2). The usage of this frames is optional. They are used to avoid collisions: a station sends RTS in order to establish a connection, the RTS frame containing both receiver address and transmitter address and the expected time of transmission. The receiving station responds with CTS, which contains only the receiver address and gives access to the transmission line.

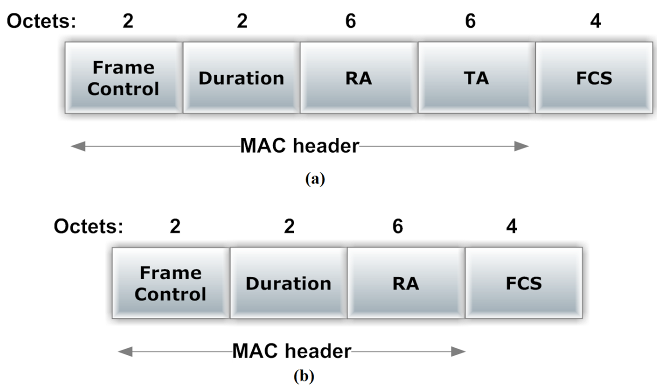

Figure 3.2: a) RTS Frame Format; b) CTS Frame Format [1].

   - **Acknowledge (ACK) frame.** In order to see if the sent data frame was transmitted correctly, the receiver, after checking for errors the received data, sends an ACK frame to confirm successful transmission. If the sending station does not receive ACK after a period of time, it retransmits the data. The frame format for ACK is similar to the one of CTS, shown in Fig. 3.2 (b).

2. **Management frames** have the general format shown in Fig. 3.3. Each type of management frame has this format, the field that differs being the body frame. Because of their specific purpose, these frames are out of our investigation and any detailed description of these is not necessary. For more information about management frames check [1].

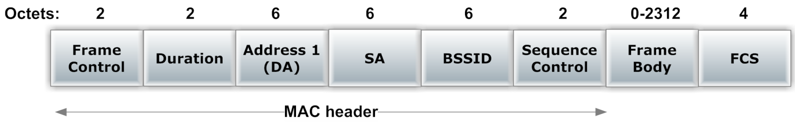

Figure 3.3: Management Frame Format [1].

3. **Data frame** - the data frame format is the general MPDU format (check Fig. 3.1); the length of this frame is variable, according to the frame body length.

3.1.2 PHY Layer

The PHY Layer contains two generic components:

- Physical Layer Convergence Procedure (PLCP) - to map the MAC frames onto de medium;
- Physical Medium Dependent (PMD) system - to transmit those frames [12].

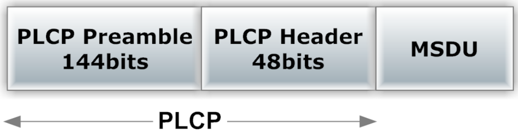

Figure 3.4: PHY Layer [1].

Our interest is in PLCP, which adds some fields to the MPDU, grouped into preamble and header, as can be seen in Fig. 3.4.

The standard defines two different preambles: the mandatory Long Preamble and a Short Preamble. For our investigation, the long preamble will be used. This preamble allows receiver to synchronize with the transmitter.

The PLCP header gives informations about modulation scheme, transmission rate and the expected duration of transmission for MPDU, expressed in microseconds.

3.1.3 Chosen Type of Frame for Evaluation

The main goal of the project is to implement Secondary Communication, using the permutation scheme, in existing wireless systems. Because the address field, which identifies the user, is in the MPDU frame, our biggest interest is the successfully transmission of

MAC frame. For a bigger range for a Wi-Fi device, the length of the packet is essential. From the list with different MAC frame types presented at page 25, ACK has the smallest length, but this frame is transmitted only after a data frame was successfully received, so we have to look at RTS/CTS (the frames with 20/14 bytes MPDU length) for implementing Secondary Communication.

Taking into account that CTS frame is transmitted as a response of an received RTS, we will study the possibility for Secondary Communication in Wi-Fi systems for RTS frame. RTS frame, as is defined in section 9.6 of IEEE 802.11b protocol from 1999 [13], shall be transmitted at one of the rates in the Basic Service Set (BSS) basic rate set. The authors of [14] present the assumptions made for implementing 802.11a and 802.11b in OPNET Simulation Model, establishing the BSS basic rate set for 802.11b at 1 and 2 Mbps.

Hence, higher data rates are not at the interest of our investigation, we will focus the research on 802.11b protocol, and we will study the RTS for both 1 Mbps and 2 Mbps. 802.11b, known as wireless fidelity (Wi-Fi), was introduced in 1999, and it operates in 2.4 GHz band. This standard uses a spread-spectrum-like modulation.

3.2 Implementation

This Subchapter will start with a brief presentation of the tools used to simulate a Wi-Fi connection, with the adjacent modifications, and it will continue with the evaluation of frame length impact on the range and the difference in transmission distance for some of the data rates available (1, 2, 5.5 and 11 Mbps). The Subchapter will also present a test scenario that will be used for obtaining relevant data, that will highlight the advantages of Secondary Communication.

3.2.1 Wireless Simulator

To simulate a simple wireless connection, we have used the IEEE 802.11b simulator offered by Matlab [2]. The simulator is an implementation of the direct sequence spread spectrum (DSSS) system with the payload data rates: 1, 2, 5.5 and 11Mbps. The Simulink scheme is illustrated in Fig. 3.5.

3.2.1.1 Description

The WLAN Physical Layer converts MPDUs to PPDUs (PHY protocol data units) by adding 2 components:

- PLCP preamble;
- PLCP header.

As presented previously in Section 3.1.2, the PLCP preamble can be choose to be of long format or short format, the first one being mandatory. As described in Section 18.2 of IEEE standard [1], the two components added to form the PPDU are DBPSK modulated,

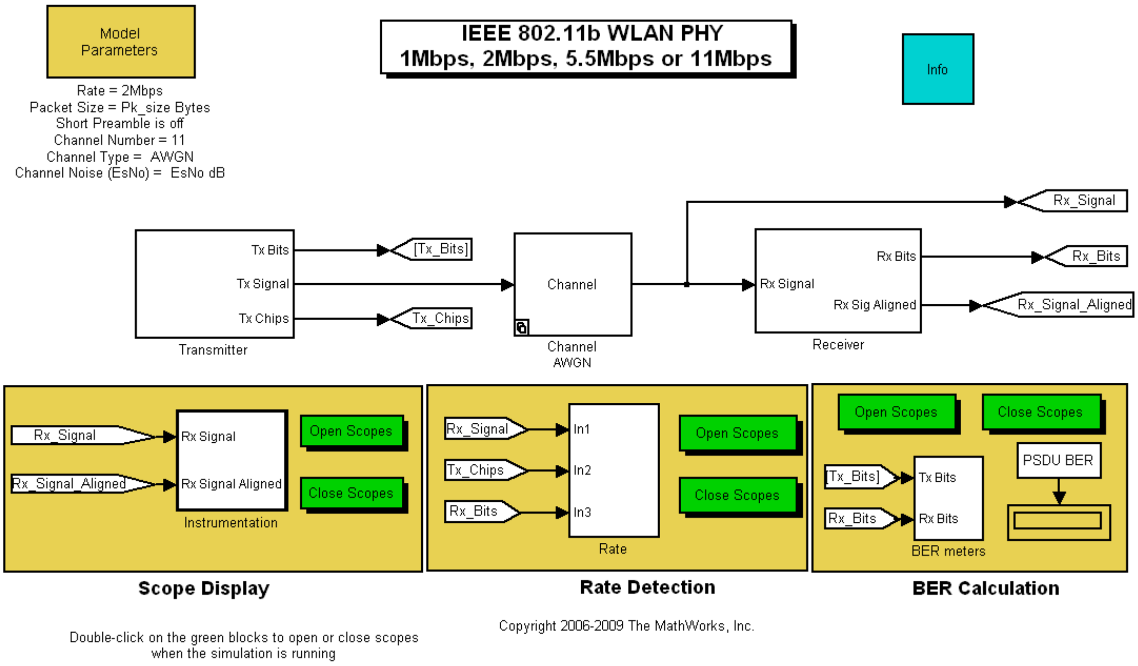

Figure 3.5: IEEE 802.11b WLAN PHY Layer [[2]].

while the MPDU modulation depends on the data rate used, and the simulator successfully implements this, having the possibility to change the modulation type by just modifying the data rate from Model Parameters block.

For the transmission channel is used the AWGN channel model, but the direct transmission could be also used.

At the receiver both PLCP and MPDU are demodulated and Error Rate Calculation block is used to compute the bit error rate of the input data. Since the PLCP preamble and header are added to the MPDU at the transmitter, and separately demodulated at the receiver, the simulator gives the possibility to calculate BER for the whole frame or for each component separately.

3.2.1.2 Additional Modifications

The main concern is how far away a frame can be received, for different frame lengths. So, we will investigate how far the RTS frame can be received compared to the average data frame. As an averaged length for data frame we used 1024 bytes.

For a better representation of a Wi-Fi communication we replaced the random sequence of MPDU with an explicit representation of the MAC layer (see Fig. 3.2) for an RTS frame.

The tested frame uses the long preamble, the generic header and an RTS MPDU of 20 bytes length. Because a packet is dismissed when the received packet has one or more errors (BER>0), the investigation will focus on packet error rate (PER). The PER results for RTS frame and for average data frame, with respect to energy per symbol per noise power spectral density (Es/No) are as we expected: the RTS can be received further away compared to the averaged data frame.

3.2.2 Scenario

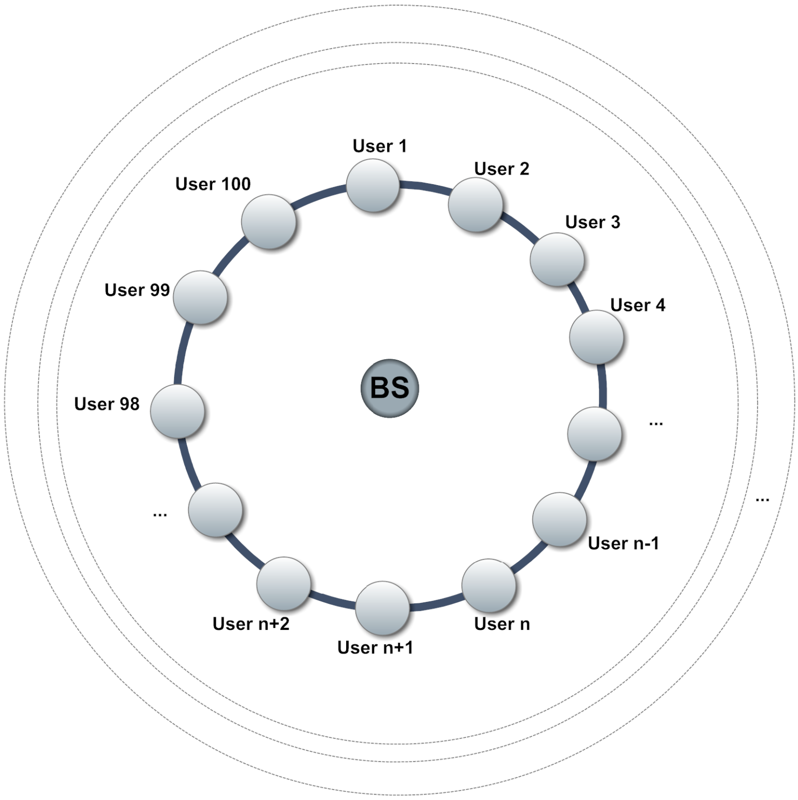

Figure 3.6: Test Scenario.

Once the MPDU was implemented in the simulator, the immediate step is to create a scenario in order to test the simulator for our desired frame format. The goal of our investigation is to find how far away from the transmitter the frame can be received, and by receiving a frame we mean having a BER for that frame equal to 0. Supposing that the access point has an omni-directional antenna the best test scenario is 100 users, positioned in such a way that forms a circle with radius d and center the access point, that receive the RTS/average data frame. Increasing d (going further away from the access point) we can see how far away RTS/average data goes. The scenario previously presented can be seen in Fig. 3.6.

Since the position of the user is irrelevant for the used Wi-Fi simulator, because the receiver address is the only field that changes, the length of the frame being the same, we will simulate the transmission of 1000 RTS frames, respectively 1000 average data frames for different Es/No. The results can be seen in Fig. 3.7. It can be seen that RTS can be received for smaller Es/No than average data, which can be concluded: the smaller the transmitted frame, the bigger the coverage distance.

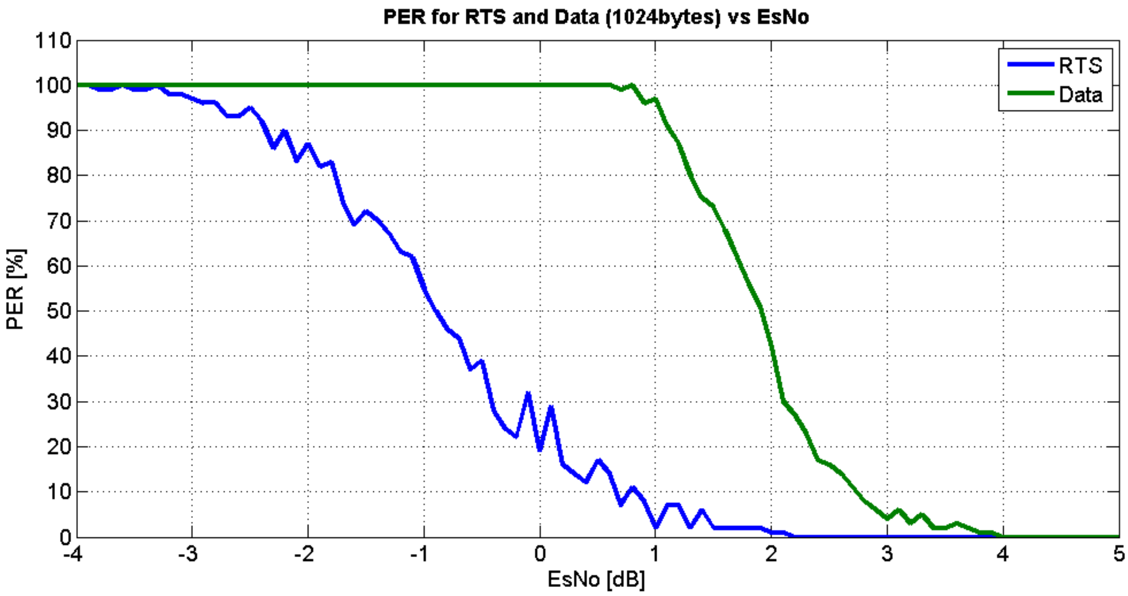

Figure 3.7: RTS vs. Average Data Frame for 1000 packets sent.

3.2.3 Wireless Range

As stated before one of the main goals of the project is to detect the range of an Wi-Fi device for different packet lengths. The most representative frame for investigating Secondary Communication in an 802.11b system is RTS; and as a comparison term for RTS frame we will use an average data frame (1024 bytes length). In Subsection 3.2.2 the packet lost (in percentage) vs. Es/No for both RTS and average data frame were presented. In order to make the link with transmission distance we will start with Friis transmission equation (see [15]):

$$P_r = P_t \cdot G_t \cdot G_r \cdot (\frac{\lambda}{4 \cdot \pi \cdot d})^c \tag{3.1}$$

Where:
P_r: received power.
P_t: transmitted power.
G_r: receiver's gain.
G_t: transmitter's gain.
λ: wavelength.
d: distance between transmitter and receiver.
c: path loss coefficient.

From Eq. 3.1 we want to extract the distance d, as a function of Es/No.

Since $E_s/N_0 \propto SNR$ and

$$P_r = SNR \cdot P_{noise} \tag{3.2}$$

Where:
P_{noise}: noise power.
SNR: signal to noise ratio.
E_s/N_0: energy per symbol per noise power spectral density.

, we will substitute P_r in Friis transmission equation:

$$SNR \cdot P_{noise} = P_t \cdot G_t \cdot G_r \cdot (\frac{\lambda}{4 \cdot \pi \cdot d})^c \tag{3.3}$$

For a reference distance (d_0), the formula becomes:

$$SNR_0 \cdot P_{noise} = P_t \cdot G_t \cdot G_r \cdot (\frac{\lambda}{4 \cdot \pi \cdot d_0})^c \tag{3.4}$$

Where:
d_0: Reference distance.
SNR_0: Corresponding SNR for distance d_0.

Dividing the last two formulas we get:

$$\frac{SNR}{SNR_0} = (\frac{d_0}{d})^c \tag{3.5}$$

Taking out d from this:

$$d = d_0 \cdot \sqrt[c]{\frac{SNR_0}{SNR}} \approx d = d_0 \cdot \sqrt[c]{\frac{E_s/N_{0_0}}{E_s/N_0}} \tag{3.6}$$

Where:
E_s/N_0: energy per symbol per noise power spectral density.
E_s/N_{0_0}: corresponding energy per symbol per noise power spectral density for reference distance, d_0.

Eq. 3.6 relates the transmission distance d to Es/No.

3.2.3.1 Comparison of RTS and Averaged Data Range in a WiFi System

Using the Eq. 3.6 and the results obtained with the Wi-Fi simulator for RTS and average data frame we can plot the Packet Error Rate (PER) over distance (see Fig. 3.8). In order to find the distance we need a reference distance value and a reference Es/No. As a reference Es/No we took the $Es/No = 4dB$, the maximum value where all the packets are transmitted without errors. The reference distance is the theoretical maximum range from literature, which varies with data rate (see Table 3.1). In Fig. 3.8 the PER was obtained transmitting RTS packets at 1Mbps, for different Es/No (from -4 to 5 with 0.1 step), using the AWGN channel model.

Hypothetical speed	Range (indoor) [m]	Range (outdoor) [m]
11Mbps	50	200
5.5Mbps	75	300
2Mbps	100	400
1Mbps	150	500

Table 3.1: 802.11b indoor/outdoor range for different data rates ([3]).

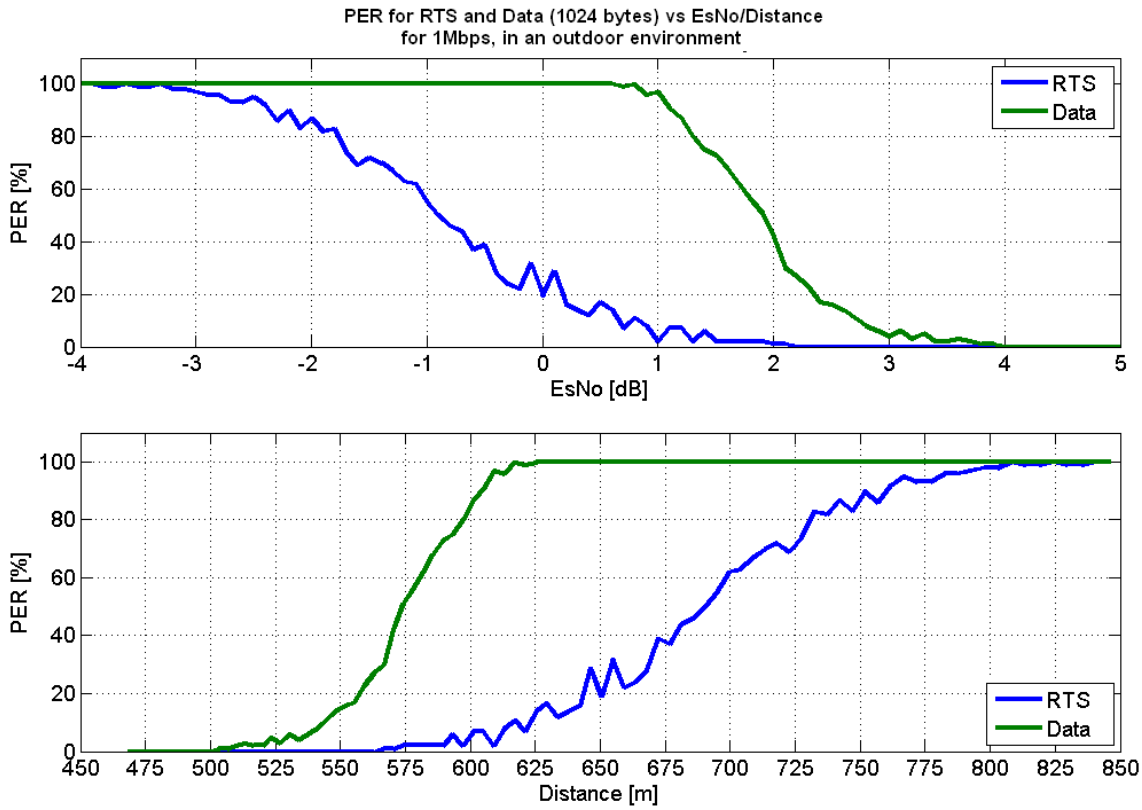

Figure 3.8: PER for RTS and Average Data vs EsNo/range at 1Mbps for outdoor.

As it was expected the range increases when the frame length decreases. In Fig. 3.8 the maximum distance where the RTS is received without any packet error is with app. 60m bigger than the distance where an average data frame is received without errors. Taking into account that the acceptable PER for 802.11b is around 8% (as presented in [16]), the maximum range for RTS is bigger than 550m.

Of course for an indoor environment the maximum range is much smaller than 550m, taking into account that the maximum theoretical range for 802.11b devices at 1Mbps in an indoor environment is 150m, but it adds almost 20m for RTS, compared to average data packet (see Fig. 3.9).

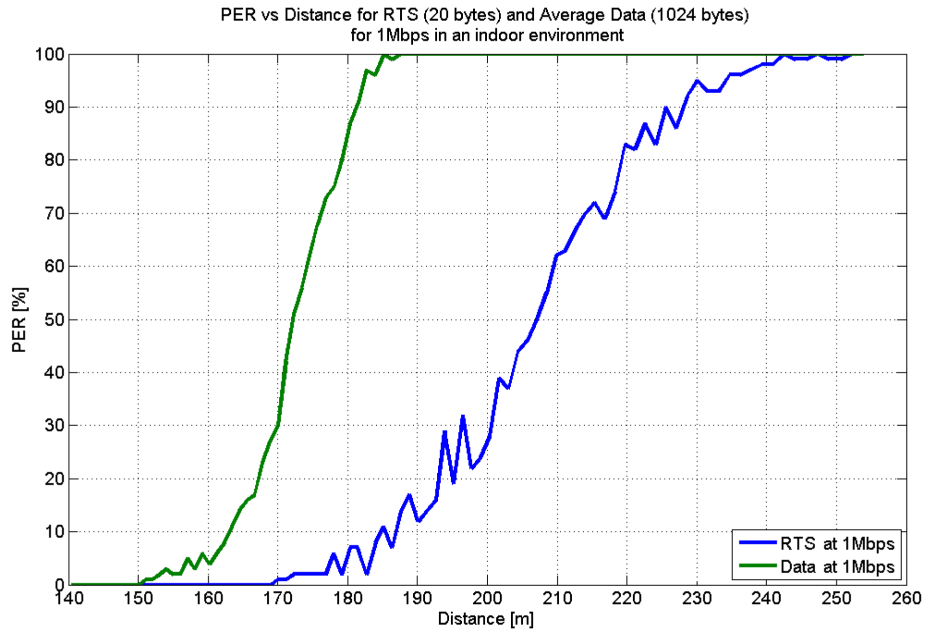

Figure 3.9: PER for RTS and Average Data vs range at 2Mbps for indoor environment.

3.2.3.2 Range vs. Data Rate

As we previously mentioned, the maximum range of an 802.11b WLAN device depends on data rate, which for this kind of device could be 1Mbps, 2Mbps, 5.5Mbps or 11Mbps in theory [1], in practice up to 6Mbps [3].

For illustration purposes we used different data rates for RTS, including 5.5 Mbps, and the conclusions showed can apply to different messages, not only control ones. Fig. 3.10 shows the PER results for RTS, at 1, 2 and 5.5 Mbps. The results of PER were obtained using 802.11b simulator with the settings from Table 3.2.

Data Rate	Nb. of transmitted pk.	Es/No [dB]	Step	$Es/No_0[dB]$	d_0 [m]
1Mbps	1000	-4 to 10	0.1	4	500
2Mbps				7	400
5.5Mbps				11	300

Table 3.2: Settings.

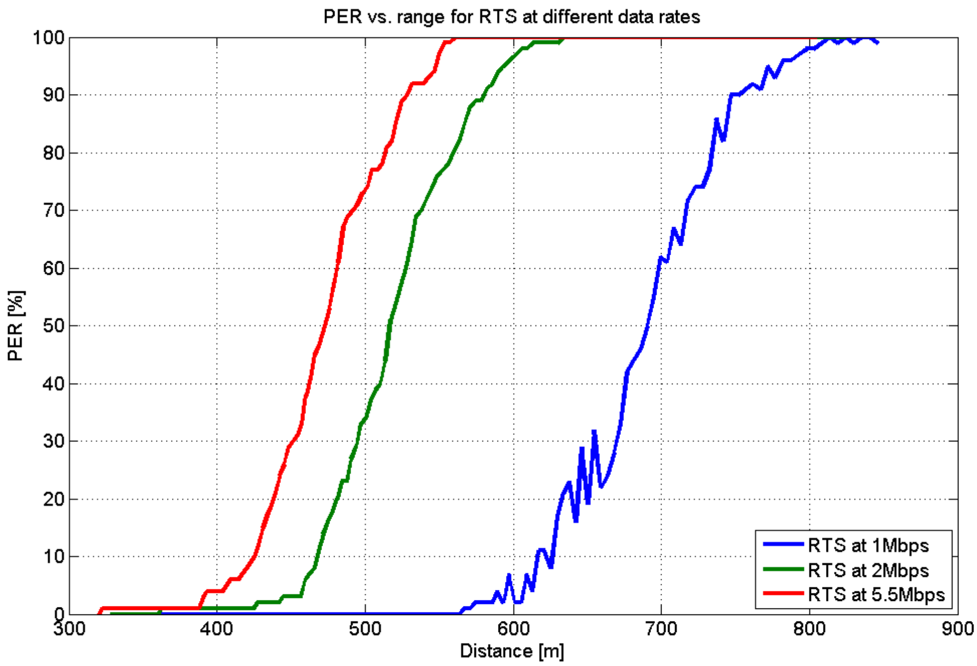

Figure 3.10: PER vs distance for 1000 RTS transmitted at different data rates for outdoor environments.

Lower data rate gives larger coverage area for Wi-Fi devices, and the difference in distance between packets sent with 1Mbps and packets sent with 5.5Mbps is more than 200m.

All the investigation until this point had as objective to show how far away RTS could be received without any packet lost compared with an averaged data packet, in order to further implement Secondary Communication in Wi-Fi devices, which could be seen as a last mile solution for users far away from the base station. Even though the user situated at 550m from a base station can not receive a data packet of 1024 bytes length at 1 Mbps data rate, because it is at the range boundary of the base station in cause, it can receive information using the solution proposed in [4] and closely investigated in Wi-Fi systems in this thesis - Secondary Communication using permutation concept.

3.3 Results

In Chapter 2 the main aspects of Secondary Communication were presented, as well as the assumptions and algorithms used to achieve this: using the existing wireless systems, and with minimum software modifications, extra information can be send using permutation concept. In the next chapter (Chapter 3) theoretical aspects of Wi-Fi systems were presented, focusing on the relevant parts for our investigation, and simulation results for range were provided.

If in Subchapter 3.2.3 the Wi-Fi primary communication was evaluated, this section will make the connection between this and the implemented secondary communication and it will end with the range and throughput results for the latter.

3.3.1 Range for Secondary Communication

The investigation, until this point, was drive to define how extra information is send without any hardware modifications, using permutations and to emphasize the frame length influence on Wi-Fi range.

Once we defined:

- the concept of Secondary Communication on Wi-Fi systems,
- the permutation tables construction,
- the mapping algorithm,
- and the error correction code;

a test should be performed in order to validate the algorithms. The test introduces probability of packet error, chooses a permutation from P(n,d), introduces errors according to the known probability of packet (user) error, and evaluates this permutation at the receiver: corrects the received permutation using the code previously presented 2.2, maps the permutation according to mapping algorithm (see 2.1.1) and compares the mapped received permutation with the mapped transmitted permutation in order to find the BER for Secondary Communication, as a function of PER for primary communication.

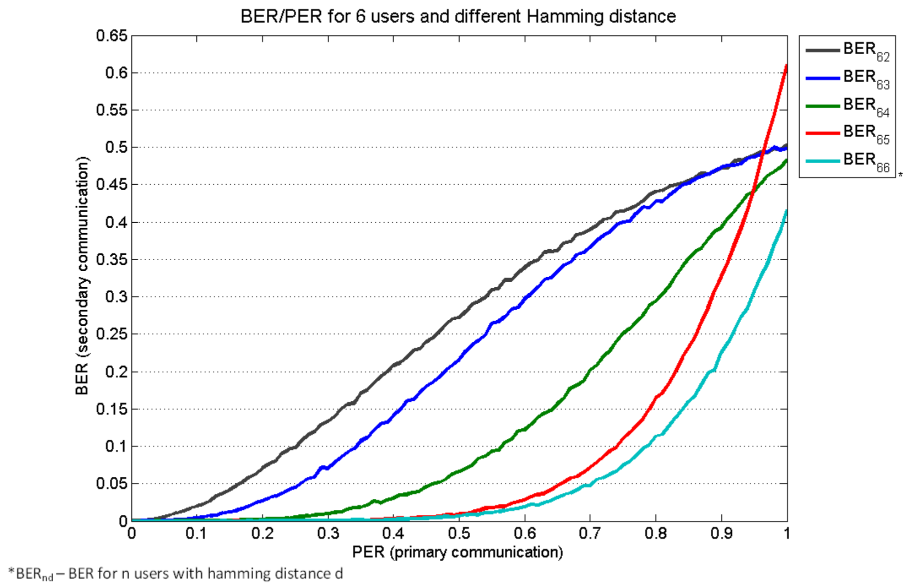

Figure 3.11: BER (Sec. Comm,) vs. PER (Prim. Comm.) for P(6,d), where d=2:6.

The results for P(n,d), with n=6 and d=2:6 can be seen in Fig. 3.11. Taking into account that error correction code associates the sequence with all bits 0 for PER (primary communication)=1, it is expected that the BER (Secondary Communication) is 0.5 or a value close to this. In the same time increasing d from 2 to 6, the probability to recover the received permutation increases, fact that it is confirmed by the obtained results (see Fig. 3.11).

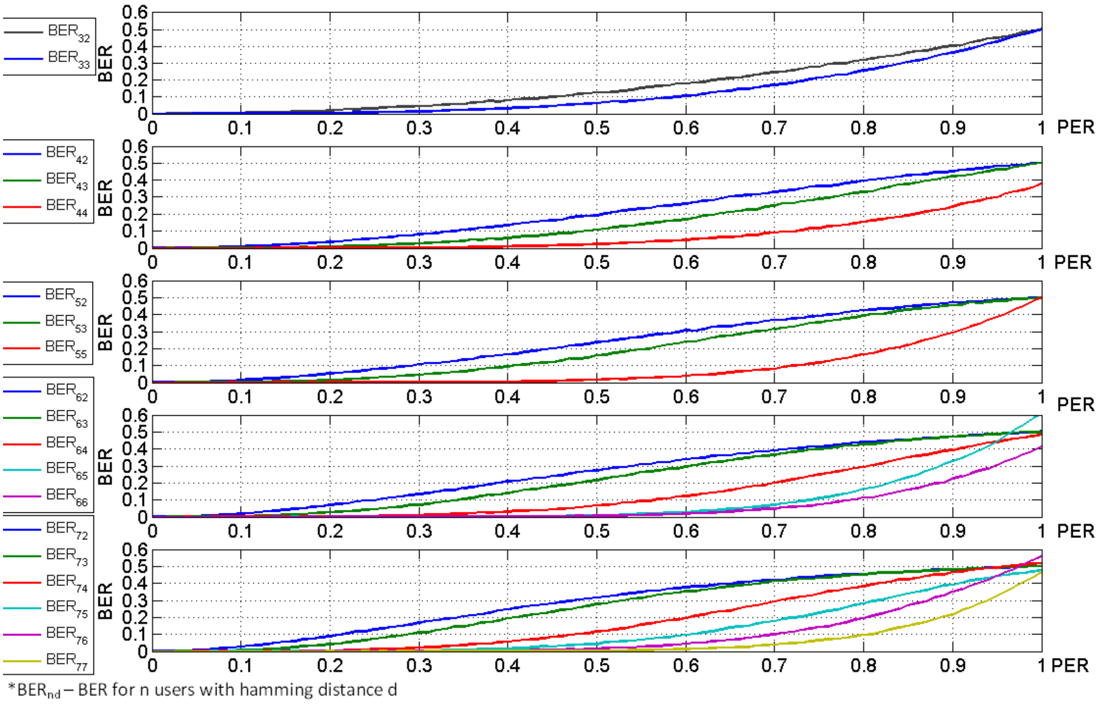

Figure 3.12: BER (Secondary Communication) vs. PER (Primary Communication) for P(n,d), where n=3:7 and d=2:n.

Fig. 3.12 presents the BER (Secondary Communication) vs. PER (primary communication) for different values of n and d. From this figure we can see that for d=3 and PER=0.5. increasing n, BER increases from 0.1 to 0.3, so, for a bigger n is preferable to use a permutation table with d close to n, for a bigger accuracy of error correction code and BER results smaller than 0.05 for 0.5 PER. All the other aspects presented previously are kept; at PER=1, the BER is 0.5 or a value close to 0.5 and increasing d from 2 to n the BER decreases.

One of our goals was to determine the Wi-Fi range (see Section 3.2.3) for Secondary Communication. Having the simulated range of a Wi-Fi system and the BER results for Secondary Communication, the link between these should be made, and Fig. 3.13 reflects this. The results were obtained by fitting a curve to PER (primary communication) vs. distance values for RTS (see Fig. 3.8) and evaluating the curve at PER from 0 to 1 with step of 0.01. After this, the connection between BER (sec. comm.) and range was established.

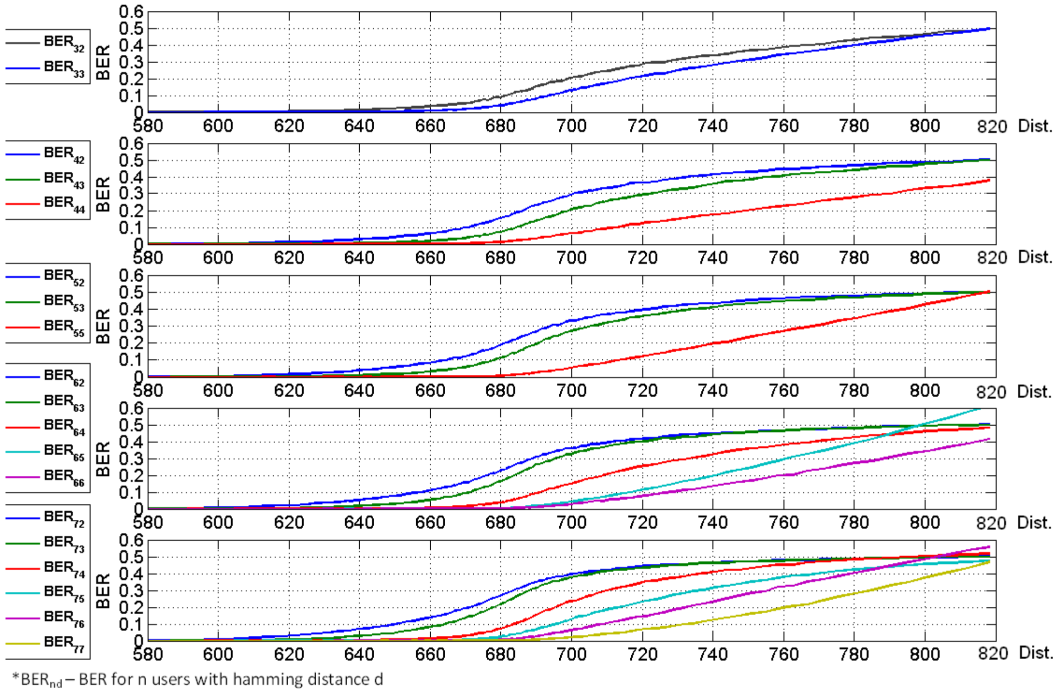

Figure 3.13: BER (Secondary Communication) vs. Distance [m] for P(n,d), where n=3:7 and d=2:n for 1Mbps data rate.

3.3.2 Throughput for Secondary Communication

After the range results for Secondary Communication for different permutations tables were provided (see Section 3.3.1), the next step is to calculate data rates for this as a function of data rate for primary communication. Even though RTS can be transmitted at 1 Mbps or 2 Mbps, we will calculate the data rate also for 5.5 Mbps and we will show the results of throughput for Secondary Communication using the results, only for illustrating purposes, the conclusions obtained applying to different messages, not only RTS packets. A very close example being the data frame with empty frame body which contains only two address fields - which has the same frame length as RTS.

In order to find the data rate for Secondary Communication, as a function of data rate for primary communication the next steps were followed:

- The RTS length is calculated: the frame has a PLCP long preamble (144 bits), a PLCP header of 48 bits and a 160 bits MPDU, with a total length of 352 bits.

$$RTS_{length} = PLCP_{Preamble} + PLCP_{Header} + MPDU = 352bits \tag{3.7}$$

- The time to transmit an RTS is calculated, using the data rate for primary communication $DR_{primary}$:

$$t_{RTS} = \frac{RTS_{length}}{DR_{primary}} \tag{3.8}$$

- Depending on n (number of users), the number of extra bits transmitted is calculated with $bits = \lfloor log_2(5!) \rfloor$.
- To find the time of transmitting a permutation:

$$t_{perm} = n \cdot t_{RTS} \tag{3.9}$$

- The data rate for Secondary Communication ($DR_{secondary}$) is obtained:

$$DR_{secondary} = bits/t_{perm} \tag{3.10}$$

The data rates for Secondary Communication can be seen in Table 3.3:

	Data Rate (primary comm.) [Mbps]	Data Rate (secondary comm.) [Kbps]
n=6	1 Mbps	4.26 Kbps
	2 Mbps	8.52 Kbps
	5.5 Mbps	23.43 Kbps

Table 3.3: Data rates (Secondary Communication) for 6 users.

Using these data rates, the throughput for Secondary Communication was obtained as a function of distance. The results can be seen in Fig. 3.14.

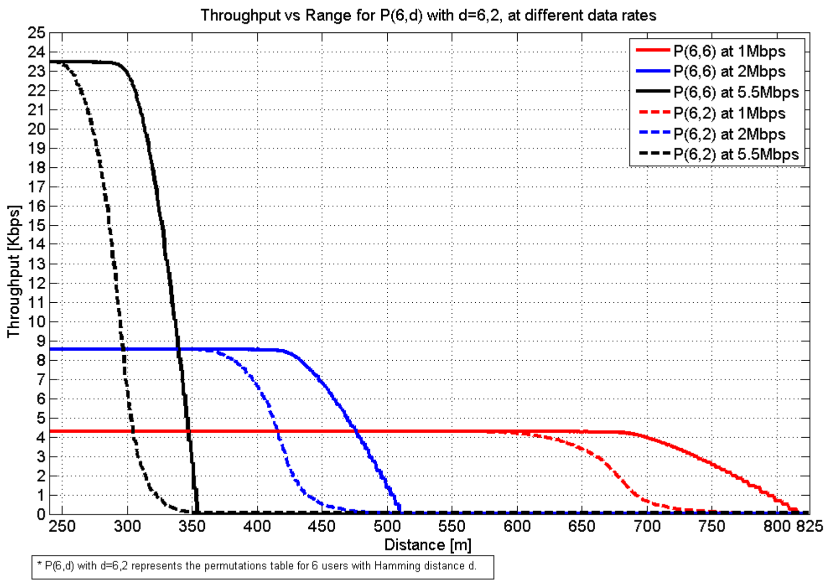

Figure 3.14: Throughput vs. range for RTS at different data rates.

For smaller data rates (primary comm.) higher ranges are obtained. From Fig. 3.14 it can be seen that the throughput for Secondary Communication for data rates of 1, 2 and 5.5 Mbps is up to 23.5 Kbps. If for 1Mbps data rate for primary communication, the throughput is around 4Kbps, for higher data rate the resulting throughput is significantly increased. As we were expecting, and we previously show on Fig. 3.13, the Hamming distance is proportional with the range, in this context. Hence, using P(6,2), the range is smaller than using P(6,6), for the same data rate, but the maximum throughput remains the same.

3.4 Conclusions

Chapter 1 presents the general concepts of Secondary Communication in existing wireless systems, using permutations and without interfering with primary communication and Chapter 3 briefly describes Wi-Fi systems, highlighting the possibility for this extra communication using RTS frames, introduces a mapping code and an error correction algorithm and evaluates the Secondary Communication with a 802.11b simulator, giving results for BER, PER, range and throughput.

In order to accomplish the transmission of extra information, permutations tables were used, each permutation being mapped in $\lfloor log_2(n!) \rfloor$ bits, where n is the number of users. The process to map permutations to bits was described in 2.1.1 and includes factoradic expression of a permutation, decimal representation of factoradic numbers and binary transformation of decimal numbers.

802.11b simulator was used to evaluate the PER for different Es/No, for RTS frames, and after the results were obtained the link between PER and range was established (see Fig. 3.8, at page 32).

Having the range results for a Wi-Fi system, the next step was to implement Secondary Communication using the permutations tables. The mapping algorithm imposed some restrictions: not all the permutation could be mapped, so from the initial permutations table (with n! permutations) were excluded the permutations that can not be mapped in the available number of bits.

For self correction, the Hamming distance was introduced: a permutation table which fulfill the Hamming distance (d) condition was constructed, P(n,d), allowing fully recovery of received sequence without using additional correction code for maximum d-1 occurred errors. Although, a correction code that covers all the possibilities (0-100% errors) was needed, and this was described in 2.2, at page 21.

Having all the parameters defined, the results of BER (Secondary Communication) and PER (primary communication) were obtained. Fig. 3.15 presents these results for P(n,2), where n=3:7, and makes the link between BER and range.

The chapter ends with the results for throughput (Secondary Communication), which is in range of Kbps, and has the maximum value 23.43 Kbps for 5.5 Mbps data rate (primary communication).

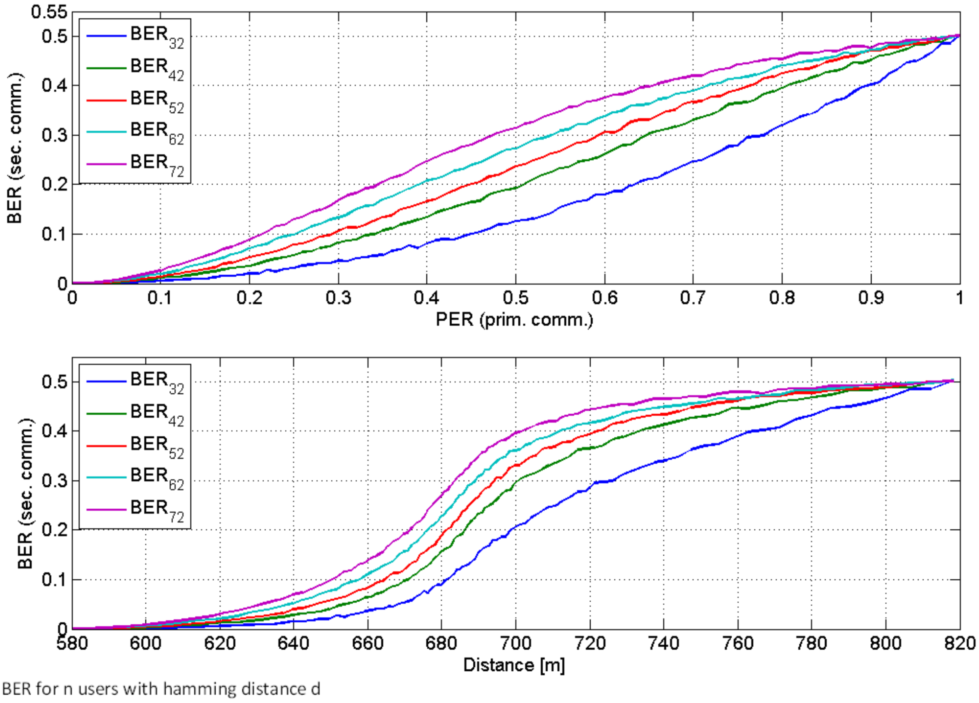

Figure 3.15: BER (secondary comm.) vs. PER (primary comm.) and BER (secondary comm.) vs. range for P(n,2), with n=3:7 at 1Mbps data rate.

Chapter 4

SECONDARY COMMUNICATION IN WiMAX SYSTEMS

The IEEE 802.16 group was formed in 1998 and had the main task to develop an air-interference standard for broadband wireless. In 2001 the first standard was completed, it was designed for LOS in 10-66 GHz based on a single carrier physical layer (PHY) with a burst time division multiplexing MAC layer. In the following years different revisions and amendments completed it until a new standard was developed IEEE 802.16-2004 which supports a point-to-multipoint (PMP) architecture with an optional mesh topology and the MAC is structured to support more PHY layers. Later this received the name of Fixed-WiMAX.

In 2005, IEEE 802.16-2005 was developed, known as Mobile-WiMAX, that had support for 2-11 GHz and a robust handover protocols defined, which allowed nomadic or mobile terminals [17]. The last standard developed and approved is IEEE 802.11-2009[1], that brings a lot of improvements to the ones previously stated, in spite of this, the most spread are the Fixed closely followed by the Mobile WiMAX standard.

To spread, develop, test and further standardize the IEEE papers the WiMAX forum was created. In order to insure interoperability between different vendors, the forum reduces the scope of the standard in smaller design and implementation choices which have the name of system and certification profiles, and it also test the different vendor specific hardware to maintain the interoperability promise, the ones that pass the tests achieve : 'WiMAX Forum Certified', which can be seen as a guaranty for the client.

Since its initial release WiMAX had an impressive development, being deployed in 149 countries totalling an impressive 620 million users, in November 2010, and expected to reach 1 billion users in 2011. So far the forum has certified 250 products, 62 of them being Base Stations (BS). The most used frequencies are 3.5 GHz with 307 deployments, 2.5 GHz with 115, 2.3 GHz with 53 frequencies, above 5 GHz with 21 and the last 3.3 GHz

[1] All the Standards mentioned can be found on http://standards.ieee.org/getieee802/802.16.html

with 9 deployments [6]

4.1 Background

In this sub-chapter we will study the PHY and MAC layer with the emphasise on potential for Secondary Communication on a WiMAX deployment. In order to apply it on the largest area possible we will study the most spread and available standard IEEE 802.16-2004 (see Fig. 4.1[2]) with its most common specifications and implementations. The focus will be on the lower layers (MAC and PHY) with an emphasis on the number of users, range, modulations and scheduling. After this the added value of Secondary Communication will be calculated, and its benefits analysed. In order to insure the compatibility with the newest standard the analyses will be general focusing on the common specifications of the standards.

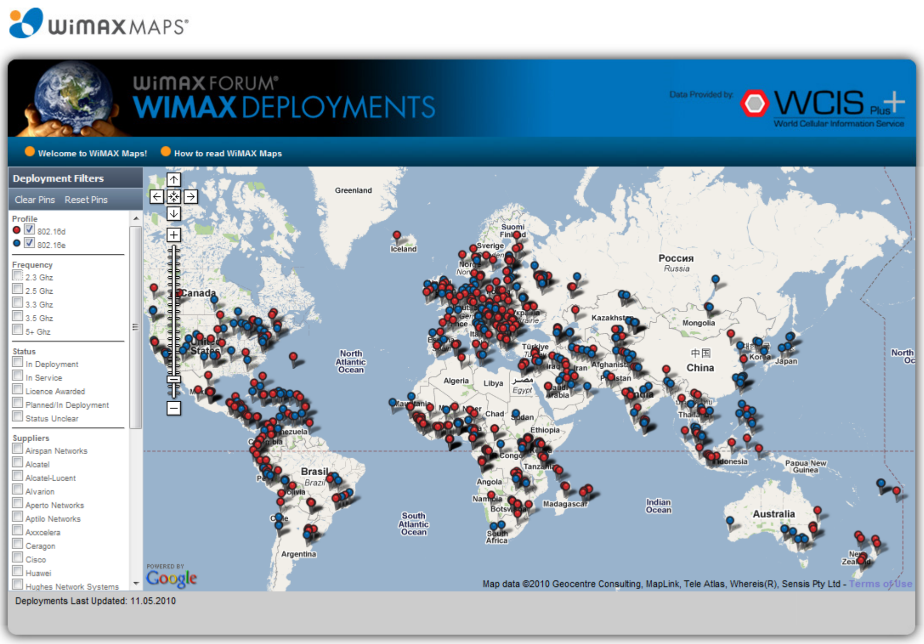

Figure 4.1: WiMAX deployment database

[2]The interactive map can be found at http://www.wimaxmaps.org/

4.1.1 Physical Layer

The Physical layer is the lowest layer and its main duty is to reliably deliver the information given by the upper layers from the transmitter to the receiver, using the physical medium: wire, air. The IEEE standards define different options of PHY layers that can be categorized based on the technology they use, the frequencies they are meant for and the standards they are used in as in Table 4.1

	802.11-2004	802.11-2005	802.11-2009	Frequencies
WirelessMAN-SC	x	x	x	10-66 GHz
WirelessMAN-SCa	x	x		< 11 GHz LB*
WirelessMAN-OFDM	x	x	x	<11 GHz LB
WirelessMAN-OFDMA	x	x	x	<11 GHz LB
WirelessHUMAN	x	x	x	<11 GHz LeB**

Table 4.1: PHY layers according to standards

* Licensed Bands
** Licensed-exempt Bands

The main focus in this thesis will be on the WirelessMAN-OFDMA, being the most used by the manufacturers, and also because it allows both duplexing schemes TDD, FDD and are based on the OFDM modulation. Another reason is that it supports multiple users in the same bandwidth, and it is the current direction of both WiMAX Forum and most WiMAX vendors as stated in [18]. The next step is to briefly explain it and the flow of information bits from entering this layer until transmission as shown in Fig. 4.2

Figure 4.2: Channel Coding steps

4.1.1.1 Channel Coding

Channel coding consists in four steps: randomization, FEC, interleaving and repetition. The latter was introduced only in IEEE 802.16-2009.

Randomization shall be used for each allocation independently (DL, UL) and when the transmitted data does not exactly match the allocation padding will be added at the end of the transmission block. Another important aspect is that FCH will not be randomized, and also the randomization resets at the beginning of the next frame.

Forward Error Correction (FEC) consists of an Reed-Solomon outer code and a rate compatible convolution inner code and it has to be supported in both UL and DL. The

Block Turbo Codes and Convolution Turbo Codes are optional throughout the standards.

Interleaving is done in order to reduce the probability of receiving bursts of errors and it is defined by a two step permutation. When CTC is used the interleaver is bypassed.

Repetition applies only to QPSK modulation and it is mostly used in bad channel condition to reliably transmit the DL and UL MAP messages.

Modulation is applied differently depending on what it is intended for:

Data After interleaving, bits are inserted serially into the constellation mapper. The mandatory modulations are QPSK, 16 QAM and optional for licensed-exempt bands is 64 QAM. The constellation has to be normalized with a factor c for equal average power.

Pilot The pilot sub carriers are inserted after each data burst and their modulation is dependent on their location in the OFDM symbol.

Preamble Because of the importance of the preamble it shall be modulated and coded with the most robust possible combination, theoretically QPSK 1/2.

4.1.1.2 Frame Structure

Both TDD and FDD are available in OFDMA-PHY in licensed bands, but in the exempted bands only TDD is mandatory so we will concentrate our analyses on it, an example of the TDD frame is showed in Fig. 4.3.

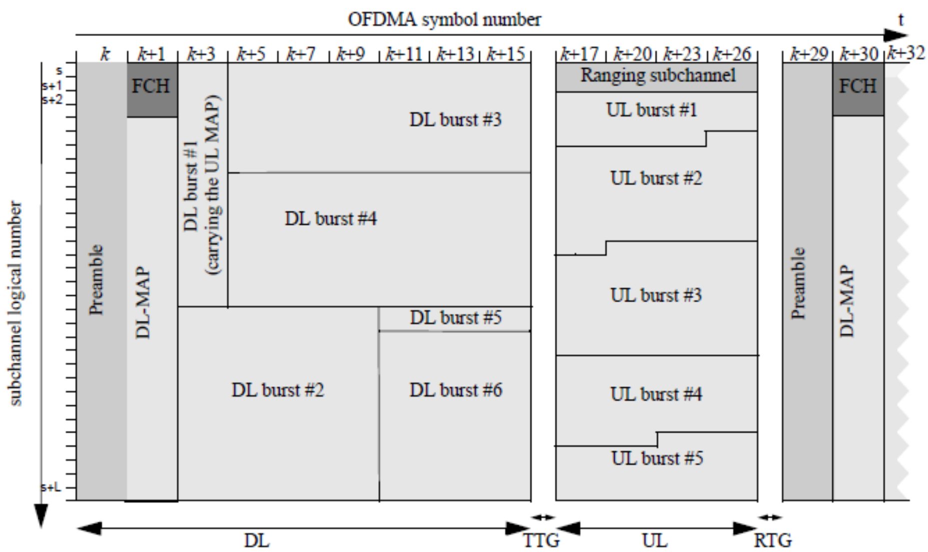

Figure 4.3: Example of OFDMA frame structure with TDD

A slot is the minimum allocation unit in the OFDMA PHY and its constructions depends on the permutation chosen and also the subframe (UL, DL) it is dedicated to, however the mandatory permutation is PUSC and the most used one based on [19], and so it will be the one studied further. A frame can have more then one permutation as shown in Fig. 4.4.

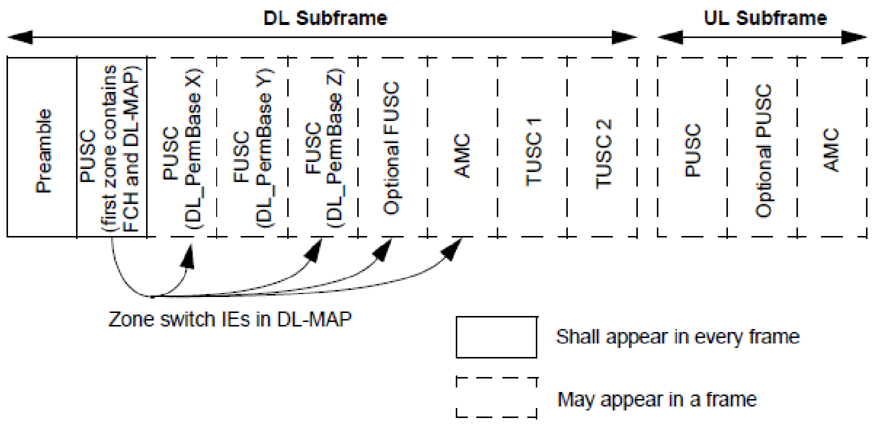

Figure 4.4: OFDMA TDD frame structure with multiple permuattion zones

In DL PUSC the symbols are first divided in 2 clusters, which are made of 2 symbols over 14 subcarriers, in UL PUSC the symbols are divided into 6 tiles which are 3 symbols over 4 subcarriers.

Every frame starts with a preamble, which is one OFDMA symbol, and it is used for different PHY layer procedures such as synchronization, initial channel estimation, noise and interference estimation. Following the preamble is the FCH (Frame Correction Header) which contains valuable information regarding the subcarriers used and the length of the DL MAP message, it is coded with the most robust MCS to ensure reliable transmission to the cell edge. After the FCH the DL MAP and the UL MAP follow with the information regarding the data regions allocated for different users in DL and the UL subframe respectively. By listening to this messages the MS can identify the slots assigned to it in the DL and UL subframe. The BS can transmit periodically also the Downlink Channel Descriptor (DCD) and the Uplink Channel Descriptor (UCD) which contain additional control information.

4.1.1.3 MAP Messages

In our study the crucial role is played by this two MAC management messages, which define the access to the DL and UL frame respectively. The DL-MAP indicates the burst start time and each burst is defined in a DL-MAP IE (Information Element) which is PHY dependent. The main parameters of an OFDMA DL-MAP IE are :

- DIUC (Downlink Interval Usage Code), which is 4 bits long, and it is an indicator

Extended DIUC	Usage
0x0	Channel Measurement IE
0x1	STC Zone IE
0x2	AAS DL IE
0x3	Data Location in Another BS IE
0x4	CID Switch IE
0x5	Reserved
0x6	Reserved
0x7	HARQ Map Pointer IE
0x8	PHYMOD DL IE
0x9	Reserved
0xA	Broadcast Control Pointer IE
0xB	DL PUSC Burst Allocation in Other Segment IE
0xC	PUSC ASCA ALLOC IE
0xD	H-FDD Group Switch IE
0xE	Extended Broadcast Control Pointer IE
0xF	UL Interference and Noise Level IE

Table 4.2: Extended DIUC code assignment

of the burst profile, specifying the PHY characteristics.

- CID (Connection Identifier), which states if the IE is meant for a unicast, multicast, broadcast transmission.
- OFDMA symbol offset, measured in OFDMA symbols, it indicates with what offset the burst starts.
- Subchannel offset, is the lowest OFDMA suchannel used for carrying the burst.
- Boosting, it indicates if power boosting was used.
- Number of OFDMA symbols, indicates how many OFDMA symbols were used to carry the PHY burst.
- Number of subchannels, indicates the number of subchannel used to carry the PHY burst, and their subsequent indexes.
- Repetition coding indication, is valid only if DIUC indicates QPSK modulation, and it states which repetition coding was used.

The standard also supports extended IEs, which will be shortly presented in Table 4.2, this extension is indicated in the DIUC (it has the value 15).

The UL-MAP IE has similar parameters with the option to use extended IE for UIUC value equal to 15, the tables and also the characteristics for are available in chapter 8.4.5.4 from the standard. A more comprehensive study is not the scope of the thesis, and the summary presented so far illustrates that the MAP messages contain the data necessary for Secondary Communication, pinpointing exactly were the burst are. Using this information is at the basis of implementing the Secondary Communication algorithms.

Also an obvious concern arises regarding all the fields that the MAP contain and the space it uses in a frame, leaving the impression that in a WiMAX frame there will be no slots available for data. In order to avoid a big part of the frame being used only for control and management purposes the compressed MAPs were introduced which use only a fraction of bits in comparison with the normal DL and UL MAP.

4.1.2 MAC Layer

The Media Access Control sublayer is a part of the Data Link Layer specified in the 7-layer OSI model (layer 2) and it has the main task of interfacing between the upper transport layers and the PHY layer. It does this by taking the MSDU received from the upper layers and transforms them in MPDU for transmission, and vice versa for reception. The process and also the components of the MAC layer are illustrated in Fig. 4.5.

The main attributes of the 3 sublayers are summarized below:

- **Service Specific Convergence Sublayer** has the task to map higher layer addresses into recognizable identifiers for the MAC and PHY layers having in mind that there is no visibility between higher layer addresses and the lower layers. This is done with Connection Identifiers (CID) that are different from UL and DL and their are also dependent on a number of factors like Service Flow Identifiers (SFID), destination and source address. Another task of the CS is the Header Suppression which will be presented in 4.1.2.1.
- **MAC common part sublayer** is at the core of MAC layer and describes the functionality of system access, bandwith allocation, connections and also the QoS and the scheduling over the PHY layer.
- **Security sublayer** has the task of protecting the subscribers privacy and it is design for robust security. Another task is to perform the user authentication and has support for fast handover.

4.1.2.1 Payload Header Suppression

Sometimes the packets received from the upper layers have large headers, but the information in them is often redundant and unnecessary to send via the scarce medium, at the cost of overall throughput, in order to avoid this, PHS was introduced. Although it is optional, it is likely that it will be implemented by WiMAX manufacturers because it improves considerably the efficiency of the systems, especially in VoIP where not only the sender and receiver IP addresses can be suppressed but also the length field.

The PHS operation is based on a PHS rule, which can be dependent on the type of service, but clear specifications are not presented in the standard, leaving the decision to upper

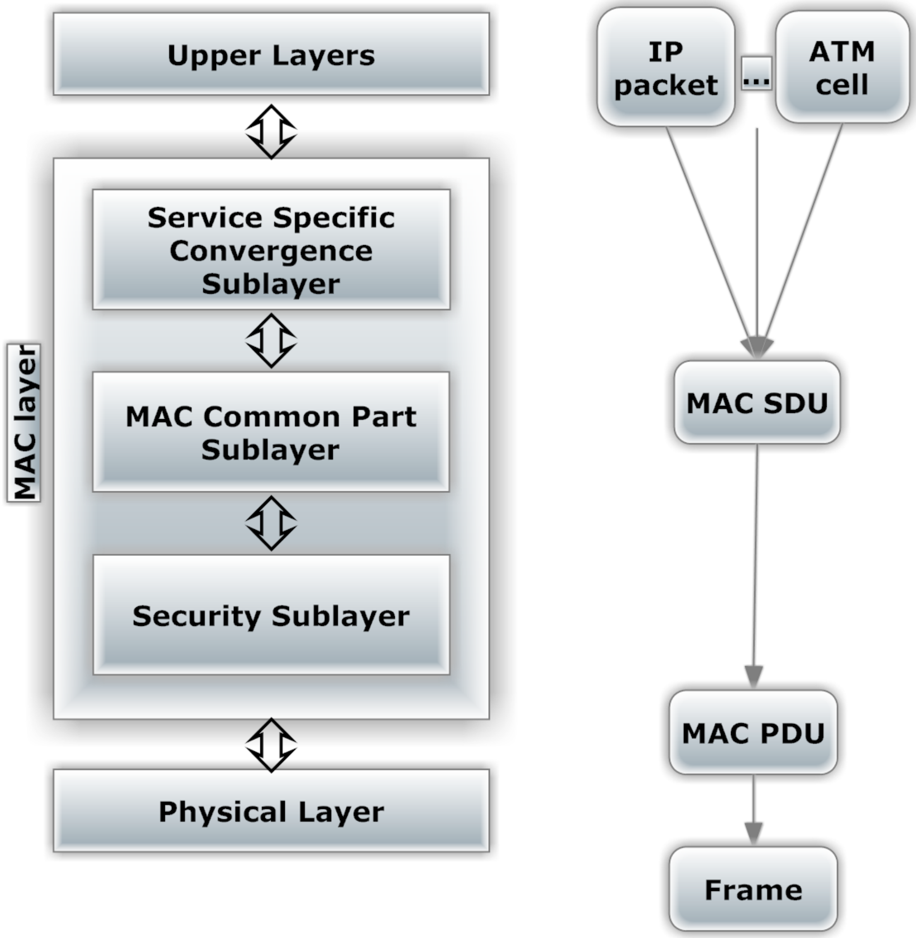

Figure 4.5: MAC layer

layers. When a packet arrives in the CS, and the optional PHS is implemented it will go through the steps highlighted in Fig. 4.6.

Based on the figure we will present the steps of the PHS operation based on a DL packet. When the packet arrives in the CS it is compared with the classification rules resulting in a service flow and a PHS rule. Associated with this rule are the PHS Field, PHS Index, PHS Mask, PHS Size and PHS Verify (set or not present). The portion of the header that will not be suppressed is masked with the PHSM, the part that will be suppressed is the PHSF which will be compared with the suppression fields from the packet and deleted. After this steps the PDU is appended with the PHSI and sent to the MAP SAP. When receiving a packet the CS identifies the PHSI and based on it and the CID, looks up the PHSF, PHSM and PHSS reconstructing the header, and sending it to the upper layers.

Besides this PHS the standard also refers to compression algorithms, defining them as IP header compressions and they are RObust Header Compression (ROHC) and Enhanced Compressed Real-Time Protocol (ECRTP) which can have similar, if not better results

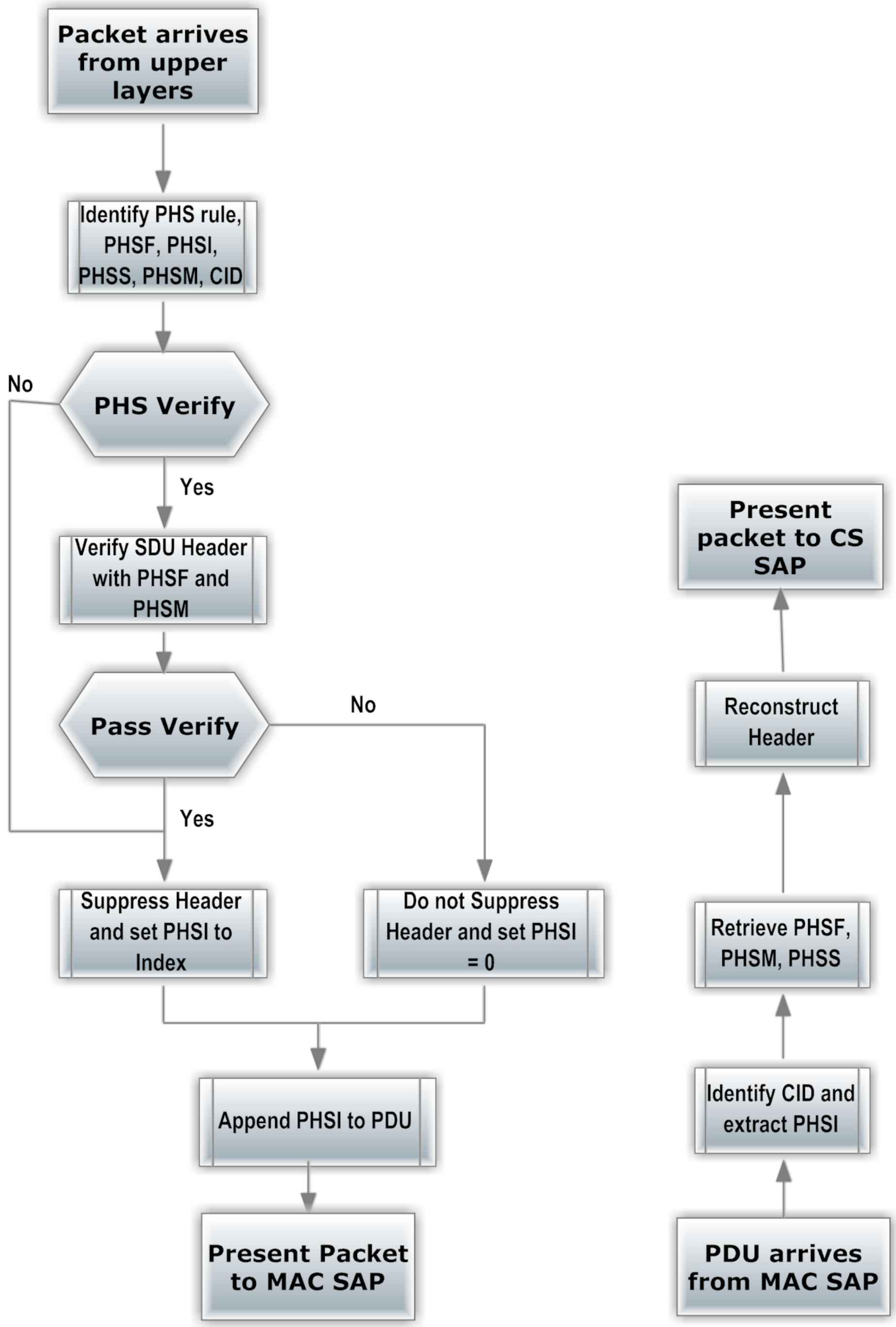

Figure 4.6: PHS operation

then the PHS.

4.1.2.2 MAC PDU Construction and Transmission

As stated before the SDU arrives from the upper layer and within the MAC Common Part Sublayer it is shaped in the MPDU. The format of a generic MAC frame is presented in Fig. 4.7 and the general characteristics are presented below.

Figure 4.7: General MAC frame

The MAC frame starts with a generic MAC header, 4 bytes long, after it the payload, which is optional and can contain subheaders, zero or more MSDU or fragments of MSDU. The CRC is 32 bits long and it is calculated over the entire frame including the header and the payload. The CRC is optional in some PHY but mandatory in OFDMA PHY, were also our area of interest lays.

There are two header formats defined in the standard and they are dependent of the MPDU that has to be sent.

- The Generic MPDU header (GMH) is the only one used in DL. It is part of the frame used to carry data and MAC management messages. Its format is illustrated in Fig. 4.8 (a).
- The bandwith request header was used in the IEEE 802.16-2004, but replaced in the IEEE 802.16-2005,-2009 with the MAC header without payload, that can be used only in the UL subframe. This header is divided into two categories Type I which is used for bandwidth requests and Type II which is used for feedbacks, being specific to OFDMA. This header formats are illustrated in Fig. 4.8 (b1 and b2)

The meaning of the notations in the previous figure are summarized in Table 4.3.

In order for the MAC layer to identify that a header without payload was transmitted the HT field will be set to 1 and the EC field to 0 for Type I and 1 for Type II. The extra parameter available in this type of headers is the Header Content which is Type dependent. More details about this headers can be found in Chapter 6.3.2.1.2 from the standard.

Besides this MAC headers the standard allows the use of MAC subheaders and special payloads. The ones relevant to our study and also their processes are summarized below.

- **Fragmentation** is the process by which a large SDU is fragmented in smaller MPDU. This process is mandatory in WiMAX systems and it is announced by a fragmentation subheader.

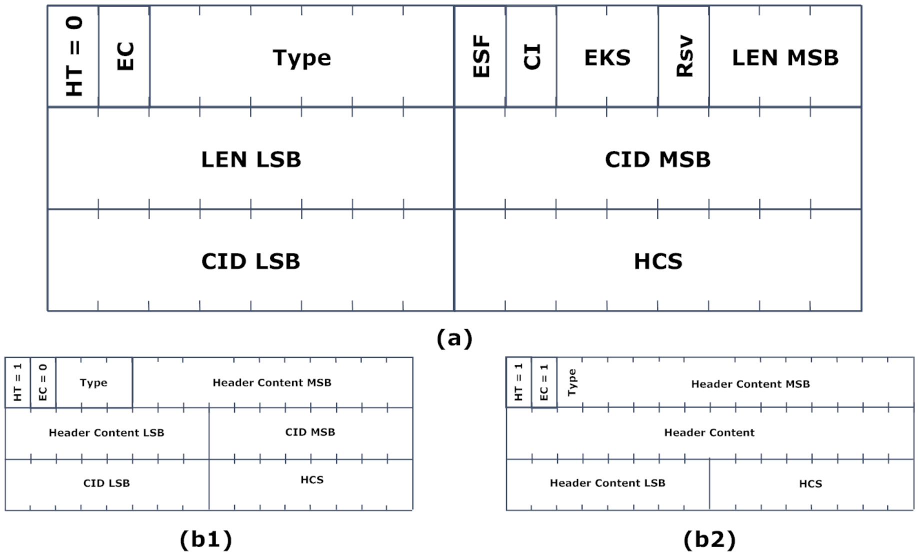

Figure 4.8: MAC frame headers (a)generic (b1)without payload Type I (b2)Type II

- **Packing** is the process by which smaller PDU are packed in a MPDU. This is done for small packets, or when the radio channel is good in order to better use the available resources. When it is applied a Packing subheader is added to the MPDU before each MAC SDU if the lengths are different or only once after the MAC header if the packets have the same length.

Field	Description	Generic MAC header	Length
HT	Header type	set to 0	1
EC	Encryption control	0 = payload not encrypted 1 = payload encrypted	1
Type	Type	Table 6 from [20]	6
ESF	Extended Subheader Field	0 = not present 1 = present	1
CI	CRC indicator	0 = CRC present 1 = CRC not present	1
EKS	Encryption key sequence		1
LEN	Length	The length of the MAC PDU	11
CID	Connection Identifier		16
HCS	Header Check Sequence		8

Table 4.3: Generic MAC header fields

4.1.2.3 Quality of Service

One of the most important tasks of the MAC layer is to ensure the QoS requirements, and it is also one of the key features of WiMAX when comparing to other broadband wireless technologies. It uses a scheduling service to handle the SDU with different QoS requirements (latency, data rate, packet error rate) and categorize them into four (IEEE 802.16-2004), five (IEEE 802.16-2005,-2009) QoS classes or scheduling services.

- **Unsolicited Grant Service (UGS)** designed for real time, it is used for fixed size packets on periodic basis. Primarily used for T1/E1 and VoIP without Silence Suppression. Having in mind the nature of the data transmitted and received the BS using the UGS service takes advantage by offering fixed-size data grants at periodic intervals, in this way minimizing the overhead created by the SS requests.
- **Real-Time Polling Service (rtPS)** similar to UGS, being also intended for real time data streams, and packets generated with periodic intervals, with the exception that it services packets with variable sizes. It is mainly used for MPEG video transmission. This service has more overhead then UGS because it allows the SS to specify the size of the grant.
- **Extended rtPS (ertPS)** has similarities to services presented, it allows variable size packets like rtPS, but the data grants are done like in UGS with unicast grants.
- **Non-Real-Time Polling Service (nrtPS)** is designed for non real time data streams with variable size packets, that have a minimum data rate requirement. An example of such a case is FTP transmissions.
- **Best Effort (BE)** can be considered the class with the lowest restrictions and serves data with minimum or no QoS requirements, packets being handled only when resources are available.

4.1.3 Scheduling

As stated before the WiMAX standard has defined 5 QoS classes, but the scheduling algorithms are out of the scope of the standard and they are left to the WiMAX manufactures. As a result different scheduling architectures were developed some of them not available for study due to copyright, some conventional algorithms used in wireless networks and some developed by researchers maximizing the efficiency of the conventional ones. There are a number of papers available in the literature that propose and compare different scheduling algorithms but in this study we are going to focus on two of them.([21], [18]).

The authors of [18] dedicate their research in surveying different scheduling algorithms comparing them and classifying them into two categories channel aware and channel unaware. It can be clearly stated that channel aware algorithm will provide a better solution, but as always the complexity and the cost have to be taken into account when searching for the most appropriate solution to a WiMAX deployment. In [21] a survey is done between the most commonly used algorithms and a new one is developed and proposed. The authors divide the algorithms into 2 categories DL and UL Scheduling Algorithms, that will also be the case in this subchapter.

An important conclusion to be considered from [18] is that most of the mapping algorithm don't take into account the variable part of the DL, and UL-MAP, and as showed in 4.1.1.3 they constitute a big part of the frame, reducing considerably the maximum number of users that can be serviced in a frame. In this thesis we try to avoid this pitfall with the cost of the scheduling algorithm complexity.

In order to have a better emphasize on the schedulers involved, Fig. 4.9 was designed based on the references cited so far. There are three schedulers involved in a communication, two in the BS and one in the MS or SS, but the latter is not the main focus of this thesis and in consequence is not drawn in the figure.

Uplink Scheduling Algorithms

A MS request bandwidth from the BS for the UL subframe. The BS grants the MS the bandwidth it requires for all it is connections and then the MS uses the resources given to schedule the users/packets it has to send according to it is internal scheduling algorithm based on QoS requirements and the service class of the user/packet.

There are a number of scheduling algorithms that can be used but the simplest and well known one is Round Robin and its variants. More complex ones were developed, giving the choice to the manufacturer that has to pick the best one or its needs, and also have in mind that the more complex they are the longer it takes to implement, the limit for the decision is very small (a WiMAX frame) and the more expensive it is to develop.

Downlink Scheduling Algorithms

The Downlink Scheduler has the task to efficiently use the resources at its disposal to send the data to the MS and also the control and management packets.

As stated before the scheduling algorithms are vendor specific, but the simplest one that can be used is Round Robin. It has the advantage of being the simplest to implement but it assigns slots to each queue with the same priority, not being able to efficiently serve the QoS requirements. It is not channel dependent which results in a low channel capacity.

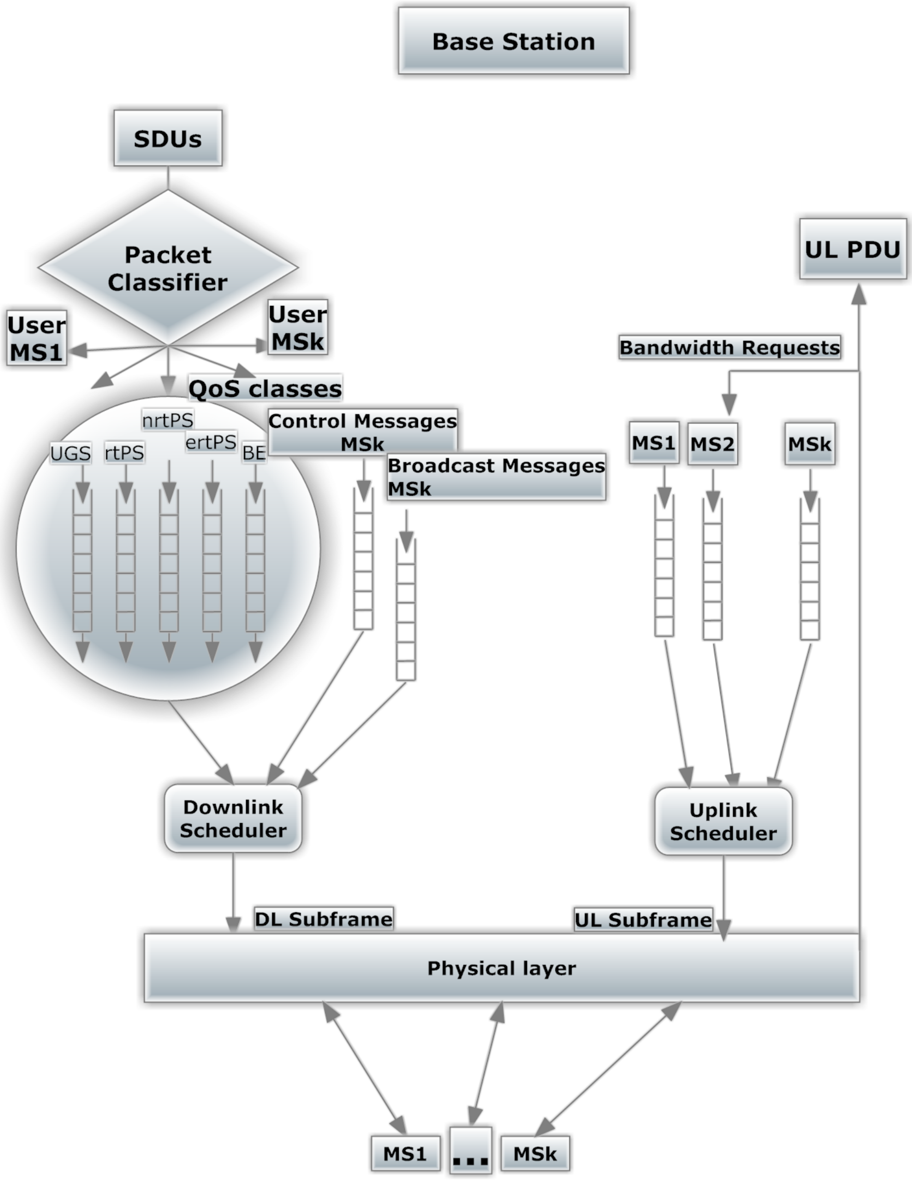

Figure 4.9: Schedulers in the BS

4.2 Implementation

In order to better understand and analyse the potential for Secondary Communication in WiMAX we briefly presented the MAC and the PHY layer before, but to have a more accurate estimation, we need to take into account also the above layers. Unfortunately due to time restrictions a complete study of those can not be done so we just take into account the most important influences they have on the lower layers in according to the potential studied. In our opinion this are represented by the overheads added by different upper layers.

In order to better emphasize the importance of overheads as well as too have a close to practice system we designed two examples that will be taken through all the steps, and at the end the improvements brought by the new system will be shown. The examples are:

- Only VoIP users. This system is based on the assumption that an WiMAX operator will offer only voice calls, for example in a rural environment as a last mile solution.
- Mixed Scenario. This system is based on all the service classes for which WiMAX is addressed to, summarized in [22] and also illustrated in Table 4.4 with the addition of the scheduling services and percentage of usage. For calculating the latter, detailed user behaviour in a WiMAX deployment should be taken into account, but to the best of our knowledge a concrete study was not concluded to this day, and even if, the user behaviour will be highly dependent on the particular deployment and the purpose of it, a few examples defined in [22] are: cellular backhaul, banking networks, education networks, public safety, temporary construction communications. Thus we take the liberty to use the same percentages as in [23]with the mention that the algorithms will be provided in order for the reader to apply them on his particular scenario.

Class Description	Real time?	Application Type	Bandwidth	Scheduling Services	Usage [%]
Interactive gaming	Yes	Interactive gaming	50-85 kbps	UGS	25.0%
VoIP, Video Conference	Yes	VoIP Video Phone	4-64 kbps 32-384 kbps	UGS or ertPS	10.0%
Streaming Media	Yes	Music/Speech Video Clips Movies streaming	5-128 kbps 20-384 kbps >2 Mbps	rtPS	12.5%
Information Technology	No	Instant Messaging Web Browsing Email(attachments)	<250 kbps >500 kbps >500 kbps	BE	32.5%
Media Content Download	No	Bulk Data, Movie Download Peer-to-Peer	>1 Mbps >500 kbps	nrtPS	20.0%

Table 4.4: WiMAX service classes

The two main advantages brought by the Secondary Communications are further distance and more users available with the same resources without expensive upgrades to the existing system. In order to highlight them we will first present the normal levels and then show the improvements.

4.2.1 Distance

It is a known fact that the advertised range for a WiMAX Base Station is 50 km in perfect channel conditions under LOS, but no experiments could actually reach close to this distance, and furthermore most of the WiMAX deployments have a cell range of 1 - 2 km depending on the area they are positioned in (rural, urban), which leads to the conclusion that in practice the range expected by the WiMAX operators is half of that.

A link budget for QPSK 1/2, with a 3:1 DL:UL ratio is presented in Table 4.5. A set of predefined parameters are used in order to calculate the link margin. With the results from the indoor and outdoor scenarios and the modified COST231 Okumura-Hata for urban and suburban environments, the estimated range is calculated in Table 4.6, for different frequencies, according to the world usage, presented at the beginning of this Chapter. This tables were constructed based on [24] and can also be found in [17].

Parameter	Mobile handheld in outdoor scenario		Fixed desktop in indoor scenario	
	DL	UL	DL	UL
Output amplifier power	43	27	43	27
Number of TX antennas	2	1	2	1
Power amplifier backoff	0	0	0	0
Transmited antenna gain	18	0	18	6
Transmiter losses	3	0	3	0
EIRP	61.01	27	61.01	33
Channel bandwith	10	10	10	10
Number of subchannels	16	16	16	16
Reciver noise level	-104	-104	-104	-104
Reciver noise figure	8	4	8	4
Required SNR	0.8	1.8	0.8	1.8
Macro diversity gain	0	0	0	0
Subchanalization gain	0	12	0	12
Data rate per subchannel (kpbs)	151.2	34.6	151.2	34.6
Reciver sensitivity	-95.2	-110.2	-95.2	-110.2
Reciver antenna gain	0	18	6	18
System gain	156.21	155.2	162.21	161.2
Shadow-fade margin	10	10	10	10
Building penetration loss	0	0	10	10
Link margin	**146.21**	**145.2**	**142.21**	**141.2**

Table 4.5: Sample Link Budget for QPSK 1/2

Modulation	QPSK 1/2		64QAM 3/4	
Link margin	146.21		120	
BS height	30 m			
MS height	1.5 m			
	Distances(km)			
Frequencies (GHz)	Urban	Suburban	Urban	Suburban
3.5	0.84	1.02	0.15	0.18
2.5	1.16	1.41	0.21	0.25
2.3	1.25	1.53	0.23	0.28

Table 4.6: Range calculations based on COST231 Okumura-Hata

As showed in the Table 4.6, the expected range is not even close in a real application, the numbers may vary depending on a large number of variables, this being just an example. Significant improvements can be accomplished by changing the antenna height or its gain.

The most important conclusion regarding our system is that the lowest modulation and coding have the largest range, as expected, which means that the Secondary Communication which will be applied on the MAP messages will have the highest probability to be accurately received at the largest distances, no matter the modulation chosen for transmitting the data.

In the best case, if all the MS receive data modulated with 64QAM for the 3.5 GHz frequency in an urban environment, the users capable of Secondary Communication will have a larger expected range with ≈700m, without any added cost to the system. In the worst case for our system will be that all the data sent is with QPSK 1/2, and this will result in the same distance for the normal users and for the users capable of Secondary Communication, but the added value here is that there are more users, further explain in the next subchapter.

Another important aspect that has to be taken into account is the repetition factor which was introduced in the PHY-OFDMA starting with IEEE 802.16-2005, and the impact it has on the Receiver Sensibility is defined in IEEE 802.16-2009, for example for a repetition of 2 will provide a 3 dB improvement, while 4 a 6 dB and 6 a 7.78 dB one. As we will show in the next part this will also have an impact on the maximum number of users in a frame, because it takes the available slots from being allocated to users into using them for repeating the MAPs. Repetition is a good feature to be used in a bad channel environment, and if it is used it will give the Secondary Communication a boost in it's range even if the rest of the data is send with the same modulation like explained in the previous paragraph. The improvements in distance are showed in the following table with the mention that further investigation has to be done in this area, with different channel models, and different parameters, in order to have a better understating of the impact of repetition coding.

Modulation	QPSK 1/2					
Repetition factor	2		4		6	
Link margin	149.22		152.23		153.99	
BS height	30 m					
MS height	1.5 m					
	Improvement(km)					
Frequencies (GHz)	Urban	Suburban	Urban	Suburban	Urban	Suburban
3.5	0.18	0.22	0.4	0.49	0.55	0.68
2.5	0.25	0.3	0.56	0.25	0.77	0.93
2.3	0.28	0.33	0.61	0.28	0.84	1.01

Table 4.7: Range improvements with repetition coding based on COST231 Okumura-Hata

4.2.2 Number of Users

In order to calculate the maximum number of users that can be associated with a BS a lot of variables have to be taken into account as well as the restriction imposed by the protocols, and the vendor specific services.

As presented in Subchapter 4.1.1.2 the frame is split in burst, clusters, tiles and slots. The slots are the lowest unit of space in the allocation algorithm, and an user receives an integer number of slots. An important aspect to remember is that the users from the DL subframe can be different from the ones in the UL subframe, and so the opportunity to apply Secondary Communication in a frame doubles. The data relevant to our system is in the DL and UL-MAP, both present at the beginning of the Downlink Subframe, and send with the most reliable MCS. Unfortunately, when studying the number of users, this control headers have also a negative impact by adding overhead for every user.

When researching for a way to calculate the maximum allowed users, a lot of conflicting studies were found, mostly because the authors presumed a fixed DL, UL-MAP but as seen in [25] these fields can go up to 50% of the frame, which leads to a lower number of users then expected. Another common assumption is a perfect scheduler which in real life is improbable and very expensive to implement.

From all the methods found two seamed the most adequate for our purposes. In the first one [23] the author dedicates his master thesis in finding this number for different scenarios, and provides a Matlab code for his method. Applying it to the case in hand it results in:

- 339 simultaneous subscribers for **Example 1**
- 314 simultaneous subscribers for **Example 2**

We will use this as a reference for the rest of the analysis, and with the help of the second method [26] we will better show the maximum number of users taking into account more parameters and calculating the exact number of users on a per frame basis.

As a permutation we will use PUSC, the characteristics of which are explain in the previous subchapter, and DL:UL ratio of 2:1 it will result in 29 symbols for DL and 18 for UL, which in turn will have 420 slots for DL and 210 slots for UL. As a first glance this will result in a maximum of 630 users, but all of the overheads have to be taken into account and also the possibility that some DL users are the same as UL for example in **Example 1** this will result in a maximum of only 210 users. Because of this reasons we will make the calculations separately for both examples.

- **Example 1**

In VoIP the workload is symmetric in that the UL data rate is equal to the DL data rate. The size of the packets as well as their periodicity depend on the vocoder used, a list of them an their characteristics is presented in Table 4.8. For our example we will use G723.1 Annex A which will result in a bit rate of 5.3 Kbps, a 20 bytes voice packet every 30 ms.

Vocoder	Bit-rate(kbps)	Frame duration	Payload(Active, Inactive)
AMR	4.5 to 12.2	20	33 , 7
G.729A	8	10	20 , 0
G.711	64	10	21 , 0
G.723.1A	5.3	30	22 , 0
G.723.1B	6.3	30	23 , 0

Table 4.8: Common vocoders and their characteristics [26]

4.2.2.1 Overheads

Upper layer overhead consist in the type of transport protocol chosen, in this case a Real Time Transport Protocol has to be used and it will result in a 40 bytes per packet overhead (RTP over UDP over IP = 12 + 8 + 20). This will significantly decrease the capacity of the system resulting a lower number of accepted users. In order to reduce the overhead several techniques can be used: Payload Header Suppression (PHS), which is a (optional) Mobile WiMAX feature and Robust Header Compression (ROHC). PHS can reduce the overhead to 3 bytes and ROHC between 1 to 3 bytes.

When considering VoIP traffic another valuable tool is Silence Suppression. In a normal conversation, one user will speak at a time which results in 50% usage of the bandwidth, for example if the user assigned to the DL subframe talks, in the UL subframe the packets transmitted will be only background noise, and this leads to inefficient use of resources because no relevant data is transmitted from one user. Using an algorithm like voice/speech activity detection (VAD/SAD) which monitors the background noise level dynamically and sets speech detection thresholds the number of users will be double then without this feature enabled.

Lower layer overhead consist in the overheads assigned by the two lowest layers MAC and PHY. In the MAC layer there are numerous possible headers and optional subheaders explained in Subchapter 6.3.2.1 of the IEEE 802.16(d,e,-2009) and an optional CRC. Every PDU will have at a minimal a 6 bytes overhead with the optional subheaders of 2 bytes each. Packing multiple SDU is possible depending on the scheduler.

When taking into account a enhanced scheduler, different deadlines can be associated with the generated data, and packing will be possible. For VoIP calls we will consider a deadline of 60 ms, but different ones can be chosen depending on the specific QoS requirements.

In the PHY layer the overheads are added in the form of preamble, that is typically 1 slot for the short preamble and 2 for the long one, the FCH, the DL-MAP and the UL-MAP. WiMAX forum asks for the implementation of the compressed MAP, which reduces the fixed part to 11 bytes including the 4 bytes CRC, and the UL-MAP to 6 bytes. The variable part of the DL-MAP has a 60 bits entry per burst and the UL-MAP a 52 bits one. Using equation 4.1 we can calculate the sizes of the compressed MAPs.

$$\begin{aligned} ULMAP(bytes) &= \frac{48 + 52 \times ULusers}{8} \\ DLMAP(bytes) &= \frac{88 + 60 \times DLusers}{8} \\ ULMAP(slots) &= \frac{ULMAP}{S_i} \times r \\ DLMAP(slots) &= \frac{DLMAP}{S_i} \times r \end{aligned} \tag{4.1}$$

Where:

r: repetition factor;
S_i: slot size, given the $_i$th modulation and coding scheme(usually QPSK 1/2)

The UL subframe also has both fixed and variable parts, like the ranging, CQI and ACK, these being defined by the network administrator. As a result we will use 3 symbols with the mention that more can be used, but in order to have a large number of users in the UL this number should be as low as possible.

- **Example 2**

In order to construct a reliable traffic model and accurately calculate the number of users in this scenario, further investigations have to be done in each class showed in Table 4.4 and correctly choose the restrictions and bandwidth demands according to the state of the art for each class. The choices are motivated for each one in the following paragraphs :

Interactive gaming In [19] the Internet Game traffic is analysed and for 3 different games Quake II, Halo 2 and Toon Town, and the traffic model is generated. Having in mind that the demands for bandwidth and delay are game dependent, and newer games will probably need bigger packets and lower delays, intense study and analyses is required to make a valid assumption. A multitude of papers tackle this subject ([27], [28] , [29] and many more) and as expected the traffic is highly dependent on a large set of variables, but this not being the main focus of this thesis we will work with the values recommended by the WiMAX forum, from which we will use 50 kbps in or investigations.

VoIP and Video Conference The VoIP part is already disused in Example 1, and we will use the values mentioned there. Regarding the Video Conference, it can be divided in 2, the audio and the video component. The audio part ranges between 16 and 64 Kbps and the video part 320 Kbps and 1 Mbps, depending on the video compression used. As an example a business-quality video conference has a 384 kbps [19]. Regarding the compression there are a lot of options, the popular ones are MPEG-2, MPEG-4 with the newest video compression of H.264 In our example we will use the mean of 384 kbps with the mention that this value can drastically change with depending on the resolution needed and the codec used.

Streaming Media This class is divided in 3 parts : Music, Video Clips and Movies streaming. Music streaming became an important bandwith consumer in recent years and according to a French study [30] people are streaming more music that they are downloading. The data rate and the packet rate are dependent on the desired quality, the most common ones are 64 kbps and 128 kbps. Video clips and Movie streaming are almost the same, the difference being in the length of the video, so we will approach them as one. As mentioned before in order to approximate the necessary bandwith the quality of the desired video as well as the resolution have to be taken into account, as well as the codec(H.264). The mean data rate is from 0.5 Mbps [19] to 5 Mbps [31] depending on the variables mentioned before, we will choose 2 Mbps in our simulations.

Information Tehnology This is one of the most difficult classes to predict and to create a traffic model because, for instance in web browsing 2 pages are not the same. There can be small web pages for example www.google.com that is only a few kilobytes and bigger ones like www.youtube.com that can take up to a few megabytes. Also the behaviour on a page can be very different depending on the user and the page itself, once on a page a user can stay between a few seconds and a few hours. Also some pages store a part of the data on the users computers, and so if the page has been visited before it will load faster then if it's the first time the user accesses it. In order to have a good estimate we will use the parameters showed in [26] that have packets between 498 bytes and 1422 bytes with a data rate of 14.5 Kbps.

Media content download For this class we will consider different packet sizes and provide the users the best data rate based on the free space available after serving the other classes, but also try to have a minimum data rate of 20 kbps.

4.2.2.2 Algorithms Used

Example 1

For calculating the available number of slots we assume a PUSC permutation as pointed out before, with a 1:1 UL:DL ratio, because of the VoIP reciprocity in a WiMAX frame. We will use a 10 MHz channel which in turn will lead to a 1024 FFT size with 841 used subcarriers, different parameters can be chosen from Table 4.9. In a 5 ms frame this will result in 48,6 OFDMA symbols, considering 1,6 for guard (TTG, RTG) we'll have 47 symbols left.

Parameter	Value			
DL				
System Bandwidth(MHz)	1.25	5	**10**	20
FFT size	128	512	**1024**	2048
Number of guard subcarriers	43	91	**183**	367
Number of used subcarriers	85	421	**841**	1681
Number of data subcarriers	72	360	**720**	1440
UL				
Number of guard subcarriers	31	103	**183**	367
Number of data subcarriers	97	409	**841**	1681

Table 4.9: OFDMA parameters

In DL each slot consists of 2 clusters, a cluster is made up by 2 symbols over 14 subcarriers and in UL a slot consist of 6 tiles, each tile made up by 3 symbols over 4 subcarriers, so the DL subframe will have 23 symbols while the UL one 24, in order to respect the 1:1 ratio. This will result in 11× 30 (subchanneles) or 330 slots for the DL subframe and 8× 35 (subchanneles) or 280 slots in the UL subframe. Each slot has a number of bits associated to it depending on the modulation and coding range, that can be calculated by using Equation 4.2

$$BytesperSlot = \frac{NumberofBitsperSymbol \times CodingRate \times DataSubcarriers \times SymbolperSlot}{8bits} \quad (4.2)$$

For example using this equation for QPSK 1/2 will result in 6 bytes per slot, while when applied to QAM 5/6 30 bytes per slot will be available.

The data rate R for VoIP communications using the values from Table 4.8, for G723.1 Annex A is 5.3 Kbps, and the packet will be 20 Bytes, but adding the header it will change according to Equation 4.3:

$$R_{withheader} = R \times \frac{MSDU + Header}{MSDU} \quad (4.3)$$

Using the ROHC 1 byte of overhead will be added resulting in $R_{withheader} = 5.565$ Kbps ≈ 5.6 Kbps, and for a 5 ms frame in 3.5 bytes per frame per user, both in UL and DL. Using PHS 3 bytes of overhead will be added and $R_{withheader} = 6$ Kbps and 3.8 bytes per frame per user. The results are summarized in Table 4.10.

Parameter	VoIP		
	Upper layers		
Header used	None	PHS	ROHC
MAC SDU size	20 bytes		
with header	40	23	21
Data rate	5.3 kbps		
with header	10.6	6	5.6
Bytes/frame per user	6.7	3.8	3.5
MAC header	6 bytes		
Bytes/frame per user	12.7	9.8	9.5
Size with fragmentation header	14.7	11.8	11.5
Number of data slots occupied (QPSK 1/2)	3	2	2

Table 4.10: Number of slots occupied depending on the overheads

After calculating the impact of the different layer overheads on the data slots, in order to calculate the maximum number of users in a frame the DL-MAP and the UL-MAP have to be taken into account using 4.1. In order to avoid recursion we will use the following equations:

$$\begin{aligned} DLslots = & \left\lceil \frac{DLMAP + DLusers \times 60/8}{S_i} \right\rceil \times r+ \\ & + \left\lceil \frac{ULMAP + ULusers \times 52/8}{S_i} \right\rceil \times r+ \\ & + DLusers \times \left\lceil \frac{D}{S_k} \right\rceil \\ ULslots = & ULusers \times \left\lceil \frac{D}{S_k} \right\rceil \\ D = & B + MAC_{header} + Subheaders \end{aligned} \tag{4.4}$$

Where:

DL slots = number of available slots for the DL subframe
UL slots = number of available slots for the UL subframe
DL MAP = the fixed size of the DL MAP = 11 bytes
UL MAP = the fixed size of the UL MAP = 6 bytes
DL users = number of users schedule for DL in the frame
UL users = number of users schedule for UL in the frame

r = repetition factor
S_i = size of the slot given the $_i$th MCS, usually QPSK 1/2
S_k = size of the slot given the $_k$th MCS
D = data size per frame
B = bytes per frame
MAC_{header} = 6 bytes
Subheaders = fragmentation and packing subheaders 2 bytes each

For example for QPSK 1/2 with a repetition of 4, DL slots = 330, UL slots = 245, and ROHC will result in :

$$\begin{aligned}
ULslots = 245 = ULusers \times \left\lceil \frac{3.5+6+2}{6} \right\rceil \Rightarrow \\
ULusers = 122 \\
DLslots = 330 = \left\lceil \frac{11 + DLusers \times 60/8}{6} \right\rceil \times 4+ \\
+ \left\lceil \frac{6 + ULusers \times 52/8}{6} \right\rceil \times 4+ \\
+ DLusers \times \left\lceil \frac{11.5}{6} \right\rceil \Rightarrow \\
DLusers = 27
\end{aligned} \tag{4.5}$$

From this results we can clearly conclude that a 1:1 DL:UL ratio is ineffective in VoIP communications because the DL frame is filled up with the UL and the DL MAP leaving room only for 27 users, and because the number of DL users is equal to the one uf UL the final result will be the minimum between this 2, which is 27, even with Silence Suppression the number will be only 54 users in the frame.

In order to increase the number we will use a 2:1, 3:1, 4:1, 26:21 DL:UL ratio and pick the most appropriate one, and the results will be summarized in the Table 4.11. Using this table it is clear that the best ratio for **Example 1** for QPSK 1/2 is 3:1, because it uses most of the frame and has the largest amount of users from all studied. Proceeding the same with the rest of modulations we will calculate the number of users according to each modulation, the results being showed in Table 4.12.

Considering QPSK 1/2 which results in 43 users in DL subframe with 8 empty slots and 43 users in UL with 19 slots. In consequence they can be arranged in a frame in :

Parameter	VoIP				
	1:1	2:1	3:1	4:1	26:21
DL symbols	23	32	35	38	26
UL symbols	24	15	12	9	21
DL slots	330	480	510	570	390
UL slots	280	175	140	105	245
Users/DL subframe	27	40	43	48	33
Users/UL subframe	122	70	52	35	105
Number of users	27	40	43	35	33
With Silence Suppresion	54	80	86	70	66
Empty slots in DL	8	12	8	160	0
Empty slots in UL	191	60	19	0	144

Table 4.11: Different DL:UL ratios

MCS	Number of users
QPSK 1/8	13
QPSK 1/4	16
QPSK 1/2	43
QPSK 3/4	43
16QAM 1/2	48
16QAM 2/3	48
16QAM 3/4	48
64QAM 1/2	48
64QAM 2/3	48
64QAM 3/4	48
64QAM 5/6	48

Table 4.12: Number of users depending on the MCS

$$\begin{aligned} DLsubframe &= \frac{(DLusers + EmptySlots)!}{EmptySlots!} = \frac{(43+8)!}{8!} \\ ULsubframe &= \frac{(ULusers + EmptySlots)!}{EmptySlots!} = \frac{(43+19)!}{19!} \end{aligned} \tag{4.6}$$

The same equations can be applied to all the MCS and after based on [4] a $\lfloor \log_2 DLsubframe \rfloor$ and $\lfloor \log_2 ULsubframe \rfloor$ to see how many bits of data can be sent with Secondary Communication and after that the data rate can be calculated. As an example, this is calculated for the previous case :

$$
\begin{aligned}
\log_2 \frac{(43+8)!}{8!} &= 204 Bits \\
\log_2 \frac{(43+19)!}{19!} &= 227 Bits \\
DataRate &= \frac{(227+204) \times \frac{1000}{5}}{1000} = 86 Kbps
\end{aligned} \tag{4.7}
$$

In other words this added value can be interpreted as 5 (ROHC + MAC overhead), 5 (PHS + MAC overhead), 4 (no upper layer compression + MAC overhead) extra users if this is the desired improvement to the system or the additional bits can be used according to the systems needs.

Example 2

For Example 2 the Matlab program developed to calculate the number of users in the VoIP scenario has to be modified. The equation used in Example 1 will be the same for this scenario, with the modification on packet sizes. In order to avoid developing an complex algorithm that takes into account all the parameters and constraints of a real deployment, only a frame will be studied from a primary communication, and the Secondary Communication will be studied on it. In order to better emphasize the program created in Figure 4.10 the general steps will be highlighted. As stated before a complex scheduler can increase the number of users, but this not being the goal of this thesis, a simple one will suffice for illustrative purposes.

In order to calculate the bits sent with Secondary Communication a frame is chosen. The selected frame has 50 DL users with 3 empty slots and 13 UL users with 49 free UL slots. This will results in 303 bits sent with this particular frame. It is obtained using the following equation:

$$
\begin{aligned}
\log_2 \frac{(50+3)!}{3!} &= 228 Bits \\
\log_2 \frac{(13+49)!}{49!} &= 75 Bits \\
TotalSent &= 303 bits
\end{aligned} \tag{4.8}
$$

In order to have a better estimate and to calculate an average data rate, 12000 frames were selected which corresponds to a minute of primary communication. In this minute an average of 135 bits are transmitted every frame with an average data rate of 27 kbps. During this interval a total of 1.6 Mbits were sent using Secondary Communications. This results correspond to a QPSK 1/2 MCS, if an 64 QAM 5/6 is used for the DL subframe and a 64QAM 2/3 is used for the UL subframe then the data rate averaged over a minute of primary communication is 52.8 Kbps.

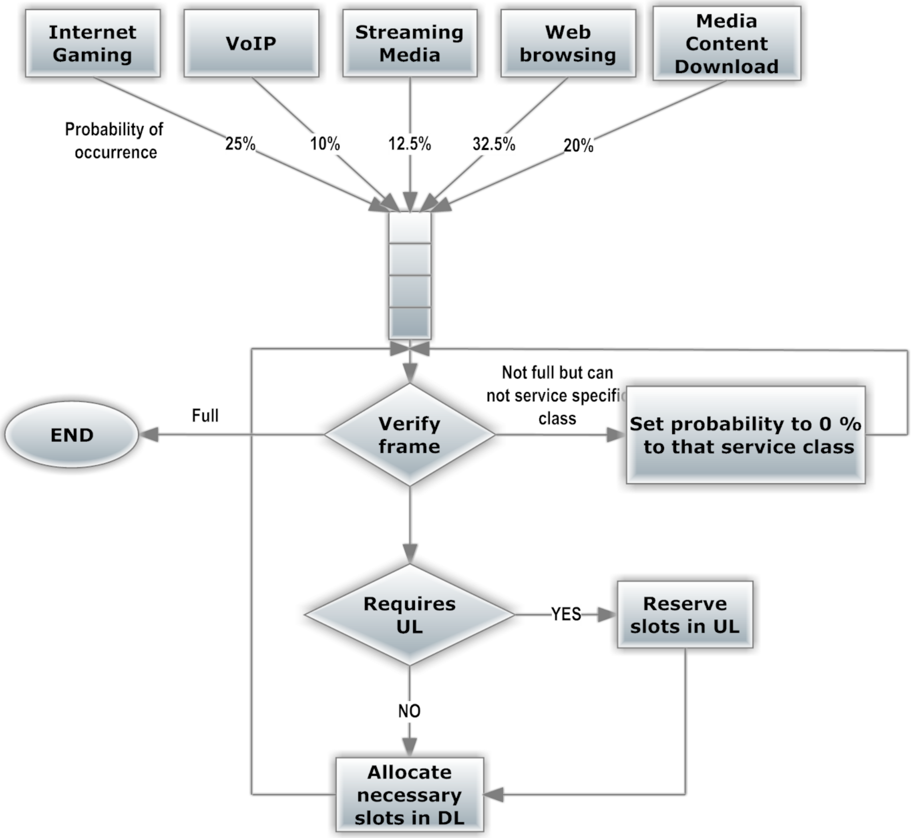

Figure 4.10: Matlab flow diagram

4.2.3 Results

In this section the results obtain so far will be summarized. The improvement in distance for both scenarios is the same, an average of 700 m in the best case, and if repetition coding is used, the range is further improved by an average of 400 m. This is a very important aspect of Secondary Communication and was discussed in [4]. The improvement is depicted in Figure 4.11.
Aside for the improvement in range the Secondary Communication can also increase the number of users, it has been showed that in a VoIP scenario 4 or 5 extra users can be added to the primary communication without any modification to the standard.

The extra number of bits obtained using this algorithm can be also used in other purposes for example: additional control information that can be sent through a reliable channel, coded or secret information, can be used for military purposes, or it can be used to aid the primary communication in bad channel conditions.

The data rate for Secondary communication was calculated for different scenarios in the previous sub chapters and it will be summarized in Table 4.13.

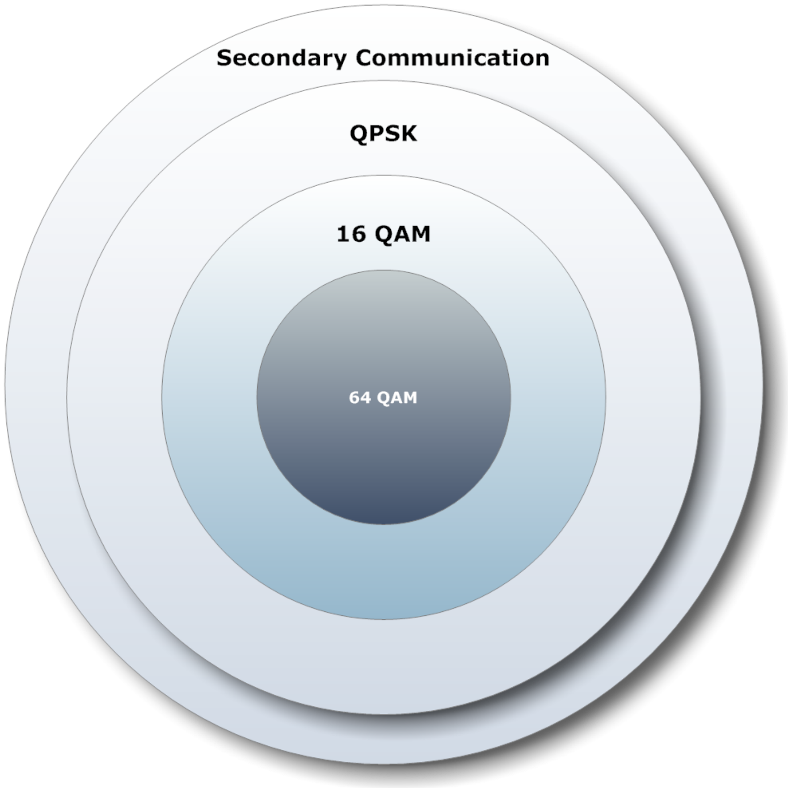

Figure 4.11: Secondary Communication Range

Example	Data rate (QPSK 1/2)	Data rate (highest MCS)
1	86 Kbps	103.2 Kbps
2	27 Kbps	52.8 Kbps

Table 4.13: Data rates

The results stated so far work on the assumption that all the permutations can be mapped, but as stated 2.1.1 not all of them can. Having in mind that a factorial table for 20 users will contain 20! elements it will be time consuming for a BS to calculate at every frame which permutation is adequate and for the MS to decode the information and to properly perform error correction. Instead of using 20+ ! tables, it is highly indicated to use smaller tables that can be processed in real time, or to use a high Hamming distance that will reduce the number of elements in the permutation table and increase the recovery possibility at the receiver as stated in 2.1.2. For example if the user wants to send ASCII code a 20! elements table is not adequate for the task, because any ASCII character can be mapped in 8 bits, so a 6! table will suffice.

Chapter 5

CONCLUSIONS

In the last decade, the interest in wireless systems grew simultaneously with the number and complexity of users requirements and needs. The fast spread of wireless devices and systems brought new challenges: what can we add to the existing systems without any hardware modifications and without disturbing the main communication. This question founds its answer in one of the IEEE papers [4] that evaluates new ways to transmit extra data using the existing systems. The authors of this paper call this new concept secondary communication, and the concept is the usage of permutations. The main idea of it is to change the order in which the users are served and these permutations to represent coded information. They present the conceptual idea of secondary communication without going into details in any wireless system. The purpose of this thesis is to evaluate the possibility of Secondary Communication in two of the most representative wireless systems for local and metropolitan areas: Wi-Fi and WiMAX.

Chapter 1 briefly presents the concept of Secondary Communication, without going into details of permutation concept, permutations table or error correction code which are treated in Chapter 2. Here, the concern of sending extra information using existing resources is backtrack to small frequency variations in electric power lines, which can be decoded by the receiver as extra information. Using permutations, and adequately mapping each of them, extra information could be sent with a simple software update. The proposed mapping scheme is describe in detail in Section 2.1.1, the process including factoradic representation of permutations, decimal calculation of factoradic numbers and binary transformation of resulting decimal number. This algorithm is preferable especially because each permutation is associated with a binary number and each binary number can be expressed in only one permutation. It has also disadvantages reducing the permutations table from n! permutations, where n is the number of users, to the maximum binary number that can be expressed in $\lfloor log_2(5!) \rfloor = 6$ bits. The error correction code presented in the same chapter introduces Hamming distance, requiring a new permutations table update: only the permutations that satisfy the Hamming distance impose are included.

After all the parameters are presented, Chapter 3 and Chapter 4 briefly present the two wireless system, chosen to be investigated in order to successfully implement Secondary Communication.

The system characteristic of most interest for our investigation in

1. Wi-Fi systems are:
   - Frame type that contains the data necessary for Secondary Communication.
   - Range, for both primary and Secondary Communication.
   - Throughput for Secondary Communication.
2. WiMAX systems are:
   - The part of WiMAX frame that contains the information necessary for implementing Secondary Communication.
   - Maximum number of users.
   - Range.

Studying different types of frames for Wi-Fi systems, RTS frame seem to be the smallest frame that contains information about destination address. All our investigation in these systems had as purpose to illustrate the range dependence on the frame length and also on data rate using as comparison element the RTS frame. These results can be seen in Table 5.1.

Data Rate [Mbps]	Wi-Fi range [m]			
	Outdoor		Indoor	
	RTS	Average data	RTS	Average data
1	563	500	169	150
2	411	400	103	100
5.5	320	300	80	75

Table 5.1: Wi-Fi range results.

If for outdoor environments the RTS is transmitted at 563m for 1 Mbps, the indoors will limit the range at 169m.

After checking RTS range in Wi-Fi systems, the results for Secondary Communication should be found. In order to simulate the secondary communication, for a certain number of users n, the permutations tables are constructed, for different Hamming distance. For packets sent with error probability from 0 to 1 with steps of 0.01, a permutation is choose to be evaluated (transmitted), the mapping algorithm and the error correction code are applied and after, the BER for secondary communication is calculated at the receiver (the mapped transmitted permutation is compared with the mapped received permutation and the packet is discarded if BER>0, in this way being also calculated the PER for secondary communication).

Checking this for 6 users and d=2:6, for PER (primary communication) of 0.1, the BER (secondary communication) is found to be close to 0 for d bigger than 3, and approximately 0.02 for d=2. For d values close to the number of users, the probability to recover the permutation increases, so it is preferable to use permutations table with d very high, even though this means smaller number of permutation available for Secondary Communication.

In Section 3.3.2 the data rates for secondary communication are calculated and the throughput results are presented. The resulted data rates are in range of kbits, and go up to 23 kbps for 5.5 Mbps primary communication data rate.

In WiMAX systems 3 standards have been at the center of the research, IEEE 802.16-2004, IEEE 802.16-2005, IEEE 802.16-2009. The latter, although being the newest standard with the best results according to range and data rate, hasn't been widely implemented and accepted, the largest deployments being accredited to the Fixed WiMAX (IEEE 802.16 - 2004), in consequence the study continued with an emphasis on it but key differences were highlighted during the thesis.

In order to apply the Secondary Communication first the standard had to be analyzed in order to find were the necessary information could be found and also the restrictions imposed by it. This was done structured on the lower layers MAC and PHY, and the most common features were found, both in the WiMAX forum as in vendor specific papers by means of white papers. The study of the lower layers was focused only on data relevant to our goal and the general characteristics studied, for example AMC, FUSC permutations were avoided due to the non mandatory nature, and due to the fact that most manufacturers implement only PUSC. An optional feature was also studied PHS, because of its relevance to out goal and also because it is highly likely that it will be implemented in practice because of its many benefits.

As a result of the analyses we have concluded that the data needed to apply Secondary Communication to a WiMAX deployment is in the MAC management messages, DL and UL - MAP. Also it was found that this messages have a negative impact on the number of users, because they take a lot of space in the frame, sometimes up to 50% . This is mainly because they are coded with the most robust MCS and they are repeated 2, 4 or 6 times within the same frame. This proved to be an advantage for Secondary Communication because being applied to it, subsequently inherits its reliability and receives a boost in the coverage area up to 1 km. After the range improvements have been quantified, the number of users was calculated. In order to have a better estimate, 2 scenarios were developed and eventually their data rate calculated.

After this study it can be clearly stated that the work done by the authors in [4] can be applied to both WiFi and WiMAX with minor adjustments to the standard. The improvements in coverage are significant in both technologies and the data rate ranges from 4 to 100 Kbps.

Future Work

The present thesis investigate the possibility for Secondary Communication in only two wireless system, and only using simulations. As future work we propose:

- Investigation for implementing this new concept on other wireless systems of interest: for example LTE.

- Practical implementation of permutation schemes in real networks. Comparison of data results with the simulation results obtained and presented in this thesis.
- More complex error models calculations.
- Matlab implementation of WiMAX system for a better comparison with analythical and practical results.

Appendix A

PERMUTATION TABLES P(n,d). MAPPING AND ERROR CORRECTION CODES

A.1 Permutations Table Construction

In order to construct the optimal permutations table for secondary communication, Hamming distance was introduced. Hamming distance d between two vectors with the same length states that the two vectors differ from each other with at least d elements. P(n,d) - permutations table for n users with d Hamming distance - was created from the initial permutations table, with n! permutations, deleting the permutations that can not be mapped into $\lfloor log_2(n!) \rfloor$ bits with the proposed mapping algorithm (see the code at page 75) and after that keeping only the permutations that fulfill Hamming distance condition. The code used to create these permutations table can be seen below:

```
clear all;
clc;
% Input arguments
n=7;
d=7;
A=1:n;
k=2;
% create the initial permutations table
table=perms(A);
% impose Hamming distance
if d==2
    new_table=table;
else
    new_table(1,1:n)=table(1,:);
    for i=2:factorial(n)
        line_table=table(i,:);
        size_new_table=size(new_table);
        for j=1:size_new_table(1)
```

```
19              distance(j)=pdist2(new_table(j,:),line_table,"Hamming");
20          end
21          d_test=distance*n;
22          if d_test≥d
23              new_table(k,1:n)=line_table(1,:);
24              k=k+1;
25          end
26      end
27  end
28  save("table_7_7.mat","n","d","new_table");
29
30  % create the mapped permutations table
31  clear all;
32  clc;
33  load("table_7_7.mat");
34
35  X=size(new_table);
36  W=X(1);
37  L=X(2);
38  % calculate the number of bits available
39  bits=floor(log2(factorial(n)));
40  table=[;];
41  counter=1;
42
43  if W==1
44      table=new_table;
45      binary=map_code(table);
46      x=bits-length(binary);
47      binary_table(1,1:x)="0";
48      binary_table(1,x+1:bits)=binary(1,1:bits-x);
49  else
50      for i=1:W
51          perm=new_table(i,:);
52          binary=map_code(perm);
53          if length(binary)>bits
54              table=table;
55          else
56              for j=1:length(binary)
57                  result(counter,j)=binary(j);
58                  % table updated
59                  table(counter,:)=new_table(i,:);
60              end
61              x=bits-length(binary);
62              binary_table(counter,1:x)="0";
63              binary_table(counter,x+1:bits)=result(counter,1:bits-x);
64              counter=counter+1;
65          end
66      end
67  end
68  save("table_10_9.mat","n","d","table","binary_table");
```

[1]

[1] symbol " used on lines 19, 28, 33, 47, 62, 68 is ' in the original Matlab code.

A.2 Mapping Code

The mapping code has three parts:

- Mapping permutations to factoradic (numbers in factorial base),
- Transform factoradic to decimal number,
- Express decimal number in binary base.

The mapping of permutations to binary numbers was accomplish with the next Matlab code:

```
% This function has as input a permutation and gives as output
% the binary representation of the permutation.

function binary=map_code(permutation)

%% permutation to factoradic
factoradic=zeros(1,length(permutation));

    for i=1:length(permutation)-1
        m=0;
        for k=i+1:length(permutation)
            if permutation(i)>permutation(k)
                m=m+1;
            end
        end
        factoradic(i)=m;
    end

    % the most right bit of factoradic is always 0
    factoradic(length(permutation))=0;

%% factoradic to decimal
    sequence=fliplr(factoradic);
    dec=0;
    for i = 1:length(sequence)
        dec = dec + sequence(i)*factorial(i-1);
    end

%% decimal to binary
    binary=dec2bin(dec);

end
```

A.3 Error Correction Code

The proposed error correction code is divided into three big parts:

1. If the number of errors is less than Hamming distance, the self-correction applies, so there will be 100% matching.

2. If the number of errors is bigger than Hamming distance, but less than the number of users, we can have:

   - Full matching - one solution,
   - Two or more solutions, and the first solution from the permutations table is chosen.

3. The permutation is received 100% with errors,case when this is associated with the permutations which contains the ascending order of users.

```
function [final_table ] = correction_code(table, d, n, error, permutation)

% output
final_table=[;];
% save initial table of permutations
TABLE1=table;
X=size(table);
L=X(2);
W=X(1);
index=[];
j=1;
% save in a vector "index" the positions where errors occur
for i=1:L
    if permutation(i)==error
        index(j)=i;
        j=j+1;
    end
end
% if the permutation was transmitted without errors:
if isempty(index)
    final_table=permutation;
else
% take out the errors from received permutation
    new_perm=strrep(permutation,error,'');
% case with 100% errors
    if isempty(new_perm)
        final_table=TABLE1(1,:);
   else
        % check for 0 at the beginning of new_perm
        a=1;
        zeros_begin=0;
        while(new_perm(a) == 0)
            zeros_begin=zeros_begin+1;
            a=a+1;
            if a>length(new_perm)
            % stop if a> new_perm dimension
                break
```

```
            end
        end
        % convert new_perm to vector
        if str2num(new_perm)==0
            new_perm1=[0];
        else
            i=1;
            new_perm=str2num(new_perm);
            while new_perm>0
                x(i)=mod(new_perm,10);
                new_perm=floor(new_perm/10);
                i=i+1;
            end
            number=fliplr(x);
            if zeros_begin==0
                new_perm1=number;
            else
                new_perm1=zeros(1,zeros_begin);
                new_perm1=cat(2,new_perm1,number);
            end
        end
        % remove the columns where e appears from table
        for k=1:length(index)
            x=index(k)-k+1;
            if x==1
                table=table(1:W,x+1:L-k+1);
            else
                new_table1=table(1:W,1:x-1);
                new_table2=table(1:W,x+1:L-k+1);
                table=cat(2,new_table1,new_table2);
            end
        end
    % Select the permutation(s) that fit
    l=1;
    m=1;
    while l≤W
        vector=table(l,:);
        difference=vector-new_perm1;
        if  difference == zeros(1,length(new_perm));
            final_table(m,1:L)=TABLE1(l,1:L);
            m=m+1;
        end
        l=l+1;
    end
    % output
    final_table_size=size(final_table);
   % Non ideal case - number of errors >d
    if final_table_size(1)≠1
        final_table=final_table(1,:);
    end
    end
end
end
```

Appendix B

WiMAX ALGORITHM

In order to calculate the number of users per frame, for both examples: only VoIP users and mixed scenario, the following Matlab code was used:

```
DL_symbols = 35;
UL_symbols = 12;
DL_slots = floor(DL_symbols / 2) * 30;
UL_slots = floor(UL_symbols / 3) * 35 - 35 ;

Si = 6;   % the slot size for QPSK 1/2 modulation used for MAP calculation
Sk = 6;   % the slot size for the modulation needed to send the packets
Sk_UL = 6;% can be diffrent with diffrent modulations
Mac = 6;  % the size of the MAC header always needed in calculations

Subheaders = 2;
UL_users = 0;
DL_users = 0;
DL_MAP = 0;
UL_MAP = 0;
used_DLslots = 0;
used_ULslots = 0;
set_prob = [0.25,0.35,0.475,0.8,1];

while (DL_MAP+UL_MAP+used_DLslots<DL_slots)&&(used_ULslots<UL_slots)
    prob = rand(1); % generate traffic
    % prob = 0.3;    % set for VoIP traffic only
    if prob < set_prob(1)
        B = 32;
        appl = "IG";
        i = 1;
    else if prob < set_prob(2);
            B = 3.5;
            appl = "VoIP";
            i = 2;
        else if prob < set_prob(3);
                B = 1250;
                appl = "Media Streaming";
                i = 3;
            else if prob < set_prob(4);
                    B = 9.1;
                    appl = "Web browsing";
```

```
38                  i = 4;
39              else if prob < set_prob(5);
40                      B = 12.5;
41                      appl = "Media download";
42                      i = 5;
43                  else disp("The frame is full");
44                      break;
45                  end;
46              end;
47          end;
48      end;
49  end;
50
51  D = B + Mac + Subheaders;
52
53  if strcmp(appl,"IG") || strcmp(appl,"VoIP")
54      ns_UL = ceil(D/Sk_UL);
55      used_ULslots = used_ULslots + ns_UL;
56      if used_ULslots > UL_slots
57          used_ULslots = used_ULslots - ns_UL;
58          set_prob(i)= 0;
59      end;
60      UL_users = UL_users + 1;
61      ns = ceil(D/Sk);
62      used_DLslots = used_DLslots + ns;
63      DL_users = DL_users + 1;
64      DL_MAP = ceil((11 + DL_users * 7.5) / 6) * 4;
65      UL_MAP = ceil(1 + 1.083 * UL_users) * 4;
66
67      if DL_MAP + UL_MAP + used_DLslots > DL_slots
68          used_DLslots = used_DLslots - ns;
69          used_ULslots = used_ULslots - ns_UL;
70          DL_users = DL_users - 1;
71          UL_users = UL_users - 1;
72          set_prob(i) = 0;
73      end;
74  else
75      ns = ceil(D/Sk);
76      used_DLslots = used_DLslots + ns;
77      DL_users = DL_users + 1;
78      DL_MAP = ceil((11 + DL_users * 7.5) / 6) * 4;
79      UL_MAP = ceil(1 + 1.083 * UL_users) * 4;
80      if DL_MAP + UL_MAP + used_DLslots > DL_slots
81          used_DLslots = used_DLslots - ns;
82          DL_users = DL_users - 1;
83          set_prob(i) = 0;
84      end;
85  end;
86 end;
87
88 DL_MAP = ceil((11 + DL_users * 7.5) / 6) * 4;
89 UL_MAP = ceil(1 + 1.083 * UL_users) * 4;
90 free_DLslots = DL_slots - DL_MAP - UL_MAP - used_DLslots;
91 free_ULslots = UL_slots - used_ULslots;
```

[1]

[1]symbol ” used on lines25, 29, 33, 37, 41, 43 and 53 is ′ in the original Matlab code.

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