Final Year Project - EAB 4012 & EAB 4034

Electrical & Electronics Engineering Program

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Component Submitted - please tick (√) one in the appropriate box:

FYP I (First Semester FYP)

- ☐ 1. Preliminary Report (due Friday, 14/8)
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WIMAX 802.16 PHYSICAL LAYER IMPLEMENTATION

AND

WIMAX COVERAGE AND PLANNING.

By

AHMED EHAB MOHAMED ZAGHLOUL MOHAMED EL-RAFAH

FINAL PROJECT REPORT

Submitted to the Electrical & Electronics Engineering Programme
in Partial Fulfillment of the Requirements
for the Degree
Bachelor of Engineering (Hons)
(Electrical & Electronics Engineering)

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CERTIFICATION OF APPROVAL

WiMAX 802.16 Physical Layer implementation

and

Wimax Coverage and Planning.

by

Ahmed Ehab Mohamed Zaghloul

A project dissertation submitted to the
Electrical & Electronics Engineering Programme
Universiti Teknologi PETRONAS
in partial fulfilment of the requirement for the
Bachelor of Engineering (Hons)
(Electrical & Electronics Engineering)

Approved:

____

Associate Prof DR Nidal Kamal

UNIVERSITI TEKNOLOGI PETRONAS
TRONOH, PERAK

December 2009

CERTIFICATION OF ORIGINALITY

This is to certify that I am responsible for the work submitted in this project, that the original work is my own except as specified in the references and acknowledgements, and that the original work contained herein have not been undertaken or done by unspecified sources or persons.

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Ahmed Ehab Mohamed Zaghloul

ABSTRACT

Over the last decade, the impact of wireless communication on the way we live and carry out business has been surpassed only by impact of the internet. But wireless communications is still in its infancy and the next stage of its development will be supplementing or replacing network infrastructure that was traditionally wired.

The advent and adoption of the computer and the myriad software packages available for it offered the ability to generate a new wave of communication combining art, pictures, music and words into a targeted multimedia presentation. These presentations are large so that is requires higher bandwidth transmission facilities. Coupling this with the need for mobility, the solution would be wireless data delivery putting in consideration the bandwidth request.

WiMAX technology is based on the IEEE 802.16 standard, it was only recently when the first IEEE 802.16 based equipment broadband began to enter the market. The additional spectrum, bandwidth and throughout capabilities of 802.16 will remarkably improve wireless data delivery and should allows even more wireless data service areas to be deployed economically.

In this Final Year Project, a study about the IEEE 802.16 standard and mainly concentrate on the 802.16 PHY Layer behaviors was performed. A Simulink based model for the 802.16 PHY Layer was built for simulation and performance evaluation of WiMAX. MATLAB was used to study the 802.16 implementation to evaluate its performance.

ACKNOWLEDGEMENTS

To Allah I am grateful, always asking for his assistance and thanking him for my successes in accomplishing a distinguished project. Moreover words just cannot fill what I am trying to express towards my supervisor, Associate Prof Dr Nidal Kamal for his guidance, encouragement, patience and full support. He offered all help that made it possible; as a result I would like to express my gratitude, appreciation and sincere thanks for his great effort.

I dedicate my dissertation work to my family and many friends. A special feeling of gratitude to my loving parents whose words of encouragement and push led me to this success. I also dedicate my work to the special person who has always supported me toward such fulfilling life, my fiancée, Qamar.

I also dedicate this dissertation to my many friends in UTP especially my batch and my friends back home in Egypt who have supported me throughout the process and life.

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1.1 Background of Study

WiMAX, meaning Worldwide Interoperability for Microwave Access, is a telecommunications technology that provides wireless transmission of data using a variety of transmission modes, from point-to-point links to portable internet access. The technology provides up to 75 Mbit/s symmetric broadband speed without the need for cables. The technology is based on the IEEE 802.16 standard.

It is another all-in-one technological concept to serve user day-to-day demands all put together. As widely known WiMAX enables the delivery of last mile wireless broadband access as an alternative to ADSL and Cable broadband.

WiMAX also has every potential to replace a number of existing world communication infrastructures. In the fixed wireless region, it can replace the telephone copper wire networks, cable TV coaxial cable infrastructure and in cellular zone, WiMAX has the capacity to fill-in the place of existing cellular networks. [1]

1.2 Problem Statement

The advent and adoption of the computer and the myriad software packages available for it offered the ability to generate a new wave of communication combining art, pictures, music and words into a targeted multimedia presentation. These presentations are large so that is requires higher bandwidth transmission facilities. Such facilities are nowadays available only within a wired office LAN.

The growing volume of targeted multimedia presentation material requires bandwidth delivery facilities. Coupling this with the need for mobility, the solution would be wireless data delivery putting in consideration the bandwidth request.

WiMAX technology is based on the IEEE 802.16 standard, it was only recently when the first IEEE 802.16 based equipment broadband began to enter the market. The IEEE 802.16 standard is designed as a next generation broadband data delivery system for Metropolitan Area Networks (MANs).

The additional spectrum, bandwidth and throughput capabilities of 802.16 will remarkably improve wireless data delivery and should allows even more wireless data service areas to be deployed economically. [2]

1.3 Objectives

The objectives of this WiMAX study project are:

- To learn about the WiMAX as the last mile communication technology.
- To focus on the details of the 802.16 Physical layer and its implementation.
- To create Simulink based model for the 802.16 for performance evolution.
- To study the WiMAX coverage and capacity planning.

1.4 Scope of Work

The project scope of work will be covering data gathering and analyzing for WiMAX technologies including its expected uses, spectrum allocation, development stages, its advantages and a comparison between its features and the previous communication generations.

In addition the scope will also cover the WiMAX 802.16 physical layer and OFDM study as its generation and reception characteristics, its guide time and cyclic extension, advantages and disadvantages, modulation, synchronization and finally the channel estimation.

Moreover, the project will focus on the broadband wireless system, WiMAX coverage prediction, performance evolution as well as WiMAX capacity planning.

CHAPTER 2

LITERATURE REVIEW FOR WIMAX

2.1 Definition

WiMAX, meaning Worldwide Interoperability for Microwave Access, is a telecommunications technology that provides wireless transmission of data using a variety of transmission modes, from point-to-point links to portable internet access. The technology provides up to 75 Mbit/s symmetric broadband speed without need for cables. The technology is based on IEEE 802.16 standard.

The name "WiMAX" was created by the WiMAX Forum, which was formed in June 2001 to promote conformity and interoperability of the standard. The forum describes WiMAX as "a standards-based technology enabling the delivery of last mile wireless broadband access as alternative to cable & DSL". [1]

2.2 Expected Uses

The bandwidth and range of WiMAX make it suitable for the following potential applications:

1. Connecting Wi-Fi hotspots to the Internet.
2. Providing a wireless alternative to cable and DSL for broadband access.
3. Providing data and telecommunications services.
4. Providing portable connectivity with high data rate transfer.

2.3 Spectrum Allocation for WiMAX

- In the US: around 2.5GHz, and is already assigned.
- In the Asia: around 2.3/2.5 GHz.
- Elsewhere in the world: around 3.5GHz,2.3/2.5 GHz or GHz.
- In addition several companies have announced plans to utilize the WiMAX standard in the 1.7/2.1 GHz spectrum band.
- The actual bandwidth of the spectrum allocations is also likely to vary to provide channels of 5 MHz or 7MHz.in principle the larger the bandwidth allocation of the spectrum, the higher the bandwidth that WiMAX can support of user traffic. [3]

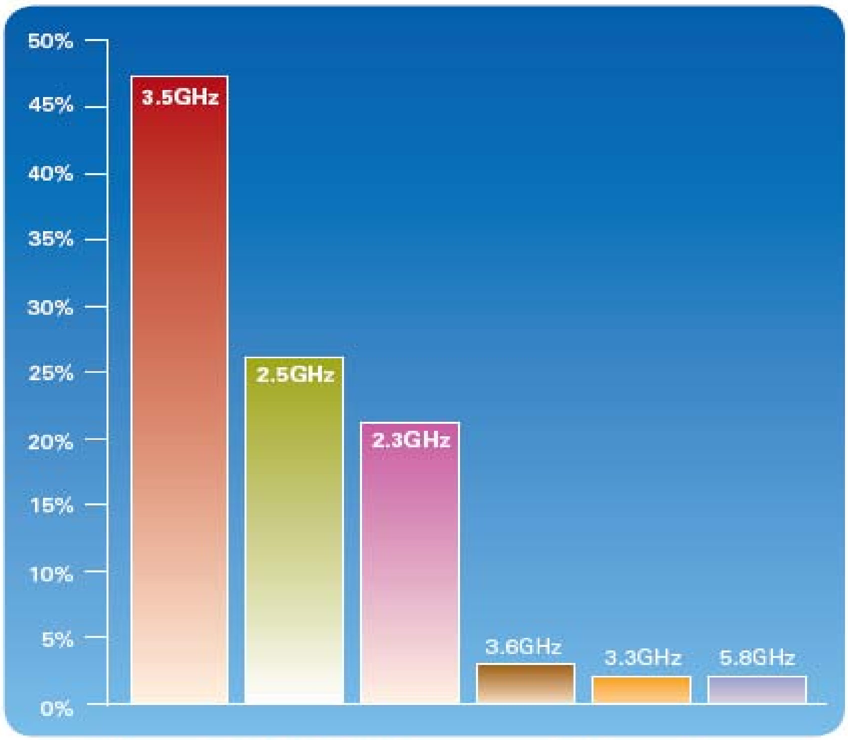

Figure 1: WiMAX Frequency Allocation around the world.

2.4 WiMAX IEEE 802.16 Development

- The original WiMAX standard (IEEE 802.16a) specified WiMAX in the 10 to 66 GHz range.
- Updated in 2004 to 802.16d which added support for 2 to 11 GHz range.
- 802.16d was updated to 802.16e in 2005 which uses scalable orthogonal frequency division multiplexing (SOFDM).This brings potential benefits in terms of:

1. Coverage.
2. Self installation.
3. Power consumption.
4. Frequency re-uses.
5. Bandwidth efficiency.
6. Full mobility support.
7. The non line of sight propagation.
8. Lower frequencies suffer less signal attenuation and so give improved range and in-building penetration and use of multipath signals. [4] [10] [2]

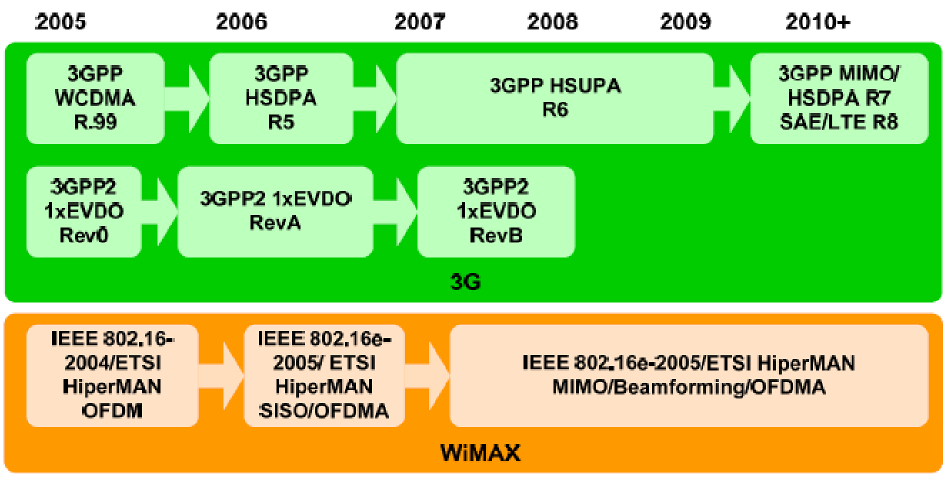

Figure 2: Evolution of 3G and WiMAX.

2.5 Comparison between Wi-Fi 802.11 and WiMAX 802.16

2.5.1 The Physical layer

2.5.1.1 WiMAX 802.16 PHY Layer

- Flexible RF channel bandwidths.
- Automatic transmit power control and channel quality measurements.
- Re-allocate spectrum reallocation trough sectoring and cell splitting as the number of subscribers grows.
- Frequency reuse for increasing capacity.
- Multiple channel bandwidths enable equipment makers to provide a means to address the unique government spectrum use and allocation regulations.
- Channels sizes ranging from 1.75MHz to 20MHz with many options in between.

2.5.1.2 Wi-Fi 802.11 PHY Layer

- Require at least 20MHz for each channel (22MHz in the 2.4 GHz band for 802.11b).
- Only the license exempts bands 2.4GHz ISM and 5GHz UNII for operation.

2.5.2 The Media Access Control (MAC) Layer

2.5.2.1 WiMAX 802.16 MAC Layer

- Relies on Grant/Request protocol for access to the medium.

- Supports differentiated service levels (e.g.: dedicated T1/E1 for business and best effort for residential).
- TDM data streams on the DL (downlink) and TDMA on the UL (uplink).
- Support delay sensitive services like voice and video (real time applications).
- Collision free data access to the channel.
- The 16 MAC improves total system throughput and bandwidth efficiency.

2.5.2.2 Wi-Fi 802.11 MAC Layer

- Contention based access techniques like the CSMA-CA protocol used in WLANs.
- CSMA-CA by contrast offers no guarantee on delay.
- WLANs in their current implementation will never be able to deliver the quality of service of 802.16 systems.

As a result: 802.16 systems users compete once to reserve a time slot which emerge or contract according to the usage but 802.11 systems users have to compete every time the log on to the network.

2.5.3 Coverage

2.5.3.1 WiMAX 802.16 coverage

- Optimal performance in all types of propagation environments including LOS, near LOS and NLOS environments.
- The robust OFDM waveform supports high spectral efficiency (bits per second per Hertz) over range from 2to 40 kilometres with up to 70 Mbps in single RF channel.

- Advanced topologies (mesh networks) and antenna improve coverage even further.
- The OFDM supports longer transmissions and the multi-path or reflection encountered.

2.5.3.2 Wi-Fi 802.11

- Basic CDMA approach or use OFDM with a much different design and have as a requirements low power consumption limiting the range.
- OFDM in WLAN covers tens to few hundreds of meters verses 802.16 which is designed for tens of kilometres.
- Due to all above WiMAX can be used for
  - Connecting Wi-Fi hotspots with each other and other parts of internet.
  - Providing a wireless alternative to cable and DSL for last mile (last km) broadband access.
  - Providing high speed mobile data and telecommunications services (

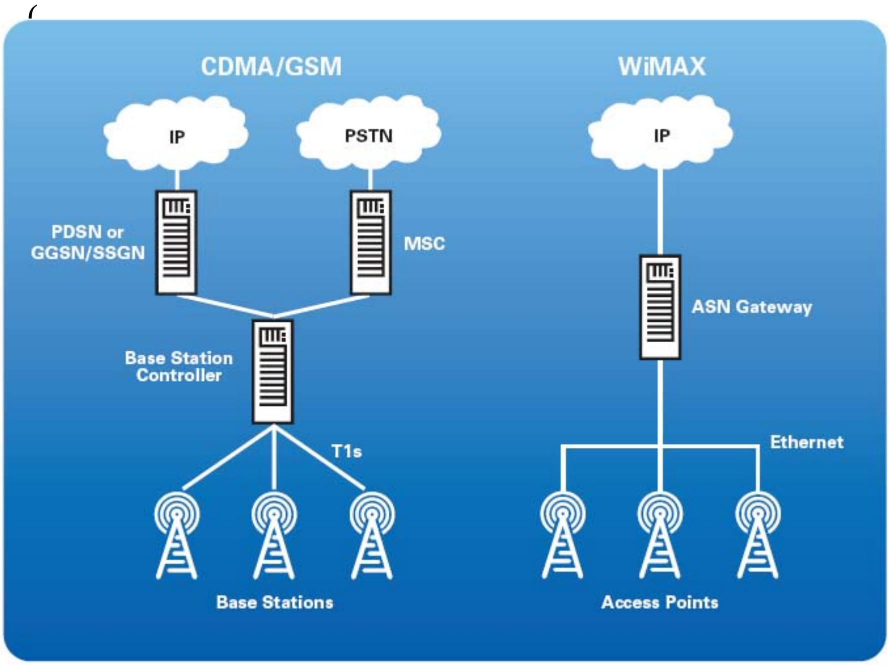

Figure 3: Comparison between GSM and WiMAX networks.

Traditional 2G network traffic must go through the equivalent of an MSC (Mobile Switching Center). The backhaul from the base stations to the MSC is either through low throughput frame relay or E1/T1 connection. Traffic is rerouted to a data network or a circuit network for voice. Furthermore, data traffic is directed to the SGSN/GGSN in a GPRS/EDGE network or through a PDSN in a CDMA network.

In contrast, the WiMAX network has a flat IP architecture with high throughput backhaul using Ethernet (10/100/1000 Base Ethernet) that is remarkably easy, efficient and cost-effective, significantly reducing CAPEX and OPEX. [5] [10]

2.6 Advantages of WiMAX

2.6.1 Superior performance

- Supports multiple handoff mechanisms ranging from hard handoffs (with break-before-make links) to soft handoffs (with make-before-break links)
- Power-saving mechanisms for mobile devices
- Advanced quality of services and low latency for improved support of real-time applications.
- Advanced authorization, Authentication and accounting (AAA) functionality.
- Use of OFDMA which suits multipath environments which gives :

  1. Higher throughout.
  2. Higher capacity.
  3. Greater flexibility in managing spectrum resources.
  4. Improved indoor coverage

- Supports both TDD (time Division Duplex) and FDD (frequency Division Duplex)

    1. FDD keeps the uplink and the downlink channels separate in frequency.
    2. TDD is ales complex, more efficient mechanism that uses a single frequency channel with uplink and downlink traffic separated by a guard time.

- Use of TDD for IP based services makes it less complex and more cost effective MIMO and beam forming.

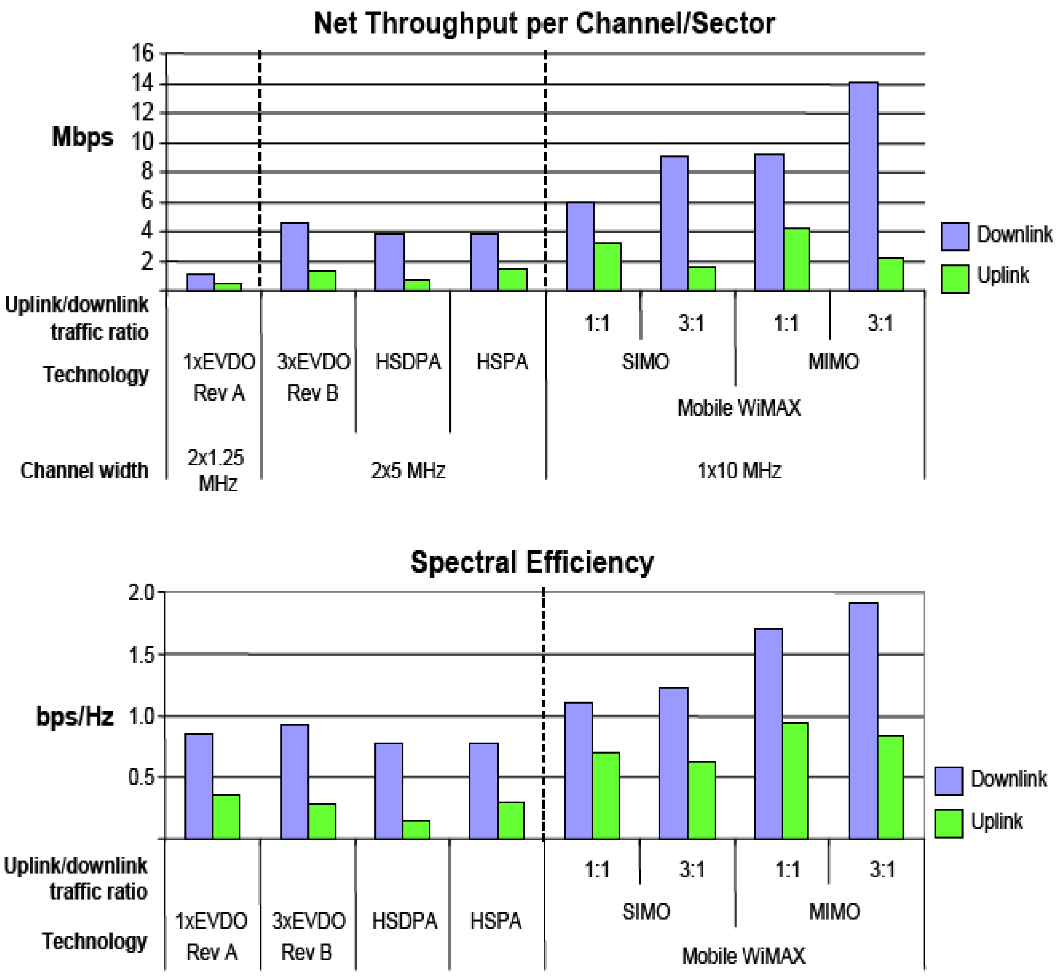

Figure 4: Performance comparison between 3G and WiMAX technologies.

2.6.2 Flexibility

- WiMAX was designed from the ground up to be all-IP technology that is optimized for high-throughput, real time applications and that is not beholden a legacy infrastructure.
- WiMAX can be deployed both in Greenfield or complementary networks.
- Global roaming among WiMAX networks using the same device and a single familiar interface, using a roaming agreements similar to those in place for cellular networks, service providers will be able to get the desired footprint in their market.
- Mobile WiMAX can be deployed in several licensed bands (2.3GHz, 2.5GHz, 3.3GHz, 3.4 - 3.8GHz) with channel sizes ranging from 3.5MHz to 10MHz.

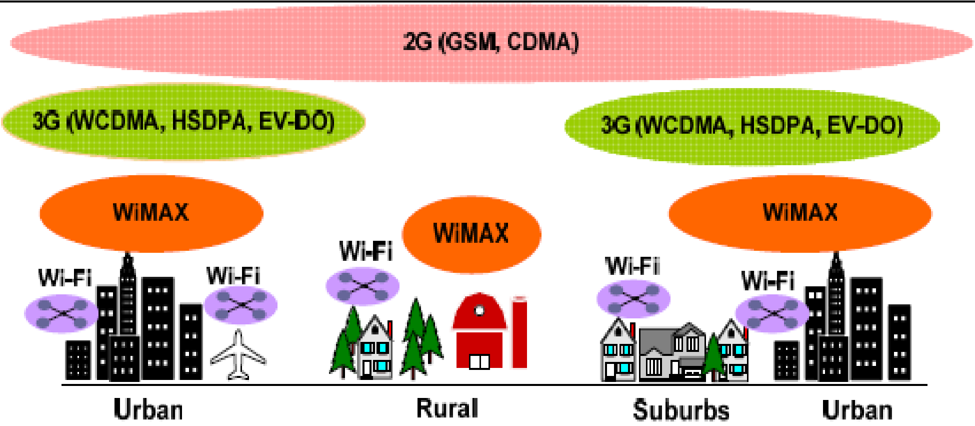

Figure 5: Different area coverage of Wi-Fi, WiMAX, 3G and 2G.

2.6.3 *Advanced IP-based architecture*

- WiMAX is a next generation technology that will facilitate the cellular operators' transition to all IP networks.
- WiMAX fully supports IMS2 (IP Multimedia subsystem) and Multimedia Domain (MMD) which give service providers the ability to:

    1. Introduce a wide range of rich voice and data applications rapidly and at a low marginal cost.
    2. With IMS and MMD, service providers can develop applications independently of the access technology within a flexible layered architecture.
    3. Application modules can be easily modified or reused.

- Support for IMS and MMD will further facilitate interworking and remove existing redundancies in the core network.

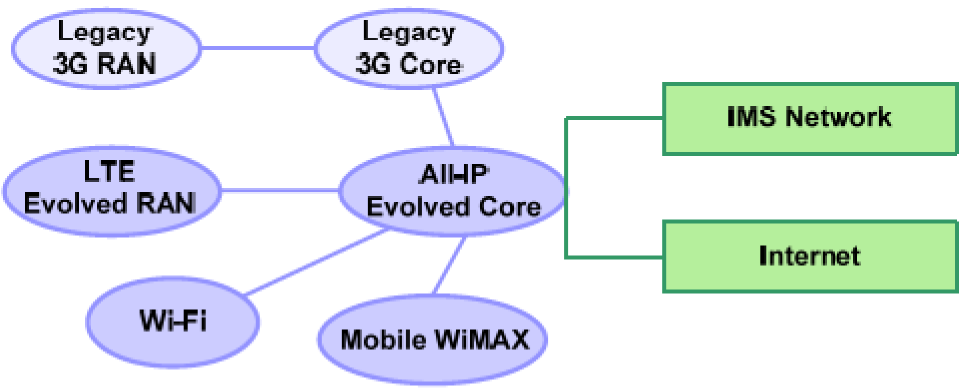

Figure 6: Role of IMS in a network with 3G, Wi-Fi and WiMAX.

2.6.4 Attractive economics

- The cost of open standards equipment tend to decrease rapidly with the increase in volume:

    1. Low cost subscribers units will further encourage adoption from subscribers.
    2. The presence of a large installed base will make deployment of the infrastructure more attractive to network operators.

- An attractive IPR structure (intellectual Property Rights): Royalties paid by manufactures on WCDMA phone are an average of 10% to 15% of the Average Selling Price of a handset, compared to a telecommunication industry norm of 2% to 5%. A less complex IPR model will lead to a significant reduction in equipment prices.
- Interoperability:

    1. The business case the cost of the equipment is kept low by combination of interoperable components based on open standards, mass adoption of subscribers units, attractive IPR structure and high base station capacity.
    2. For users: service providers will be offering personal broadband services at price that business and customer users will find attractive.

- Use of OFDMA, MIMO and beam forming increase the capacity of the users can be served by the same base station of any other system. [2] [3] [6]

2.7 WiMAX 802.16 standards

2.7.1 Table 1 showing the Physical Layer Features: [5]

Feature	Benefit
256 point FFT OFDM waveform	- Built in support for addressing multipath in outdoor LOS and NLOS environments.
Adaptive Modulation & variable error correction encoding per RF burst.	- Ensures a robust RF link while maximizing the number of bits / second for each subscriber.
TDD and FDD duplexing support.	- Address varying worldwide regulations.
Flexible Channel sizes (e.g. 3.5MHz, 5MHz, 10MHz, etc)	- Provides the flexibility necessary to operate in many different frequency bands with varying channel requirements around the world.
Designed to support smart antenna systems	- At nowadays affordable cost, they will become important to BWA deployments for their ability to suppress interference & raise system gain.

2.7.2 Table 2 showing the 802.16 MAC Layer Features: [5]

Feature	Benefit
TDM Scheduled UL & DL frames.	- Efficient bandwidth usage.
Scalable from 1 to hundreds of subscribers.	- Allows cost effective deployments by supporting enough subs to deliver a strong business case.
Connection-oriented	- Per Connection Quality of service. - Faster packet routing and forwarding.
Automatic Retransmission request (ARQ).	- Improves end-to-end performance by hiding RF layer induced errors from upper layer protocols.
Support for adaptive modulation.	- Enables highest data rates allowed by channel conditions, improving system capacity.
Security and encryption (Triple DES).	- Protects user privacy.
Automatic Power control.	- Enables cellular deployments by minimizing self interference.

2.8 Coverage in WIMAX

2.8.1 Introduction

This part of the FYP aims to discuss the coverage performance of WIMAX wireless metropolitan area networks based on the IEEE 802.16 standard of WIMAX technology.

2.8.2 Network topology and architecture

2.8.2.1 Point to Point Radio Systems

Point-to-point fixed wireless systems can be used effectively to carry very high-speed access lines or trunks from public telecommunication network operators to subscribers.

Higher frequencies (>20 GHz) are generally applicable only to PTP links. This is because at these frequencies, range is a limitation. The system is also plagued by other problems as the signal at higher frequencies is subject to attenuation in the atmosphere. Weather, particularly rain, leads to signal fading. The signal also suffers attenuation due to foliage.

In addition, the radio frequency (RF) bands allotted to PTP system usage (>20 GHz) are not able to propagate easily through obstacles or diffract around them. This makes LOS necessary between the transmitter and receiver. The need for a LOS system and the skill associated with verifying LOS during installation makes the system expensive. However, once deployed, the system is capable of realizing high bandwidth communications. [7]

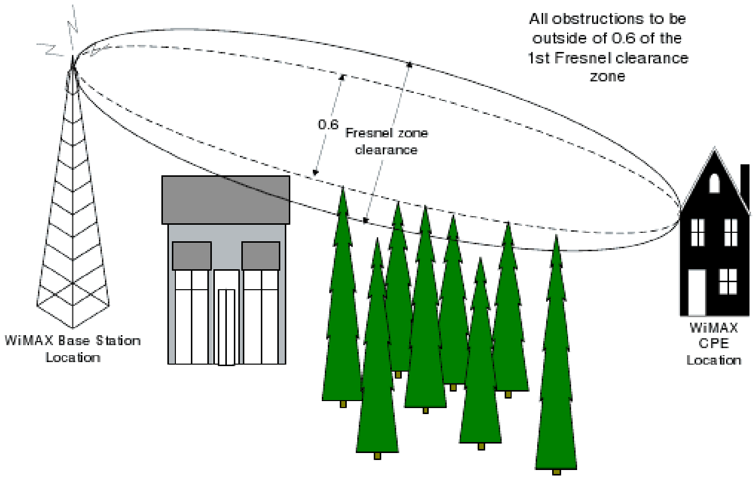

Figure 7: LOS Fresnel zone.

2.8.2.2 Point-to-Multipoint (PMP) Radio Systems

Point-to-multipoint radio systems are the focus of this thesis. These systems are more suitable for deployment of broadband wireless access, especially in an urban setting, where most of the time finding a LOS path from a transmitter to the receiver is improbable owing to the variation in terrain, building clutter, etc. Currently, PMP systems have broken the LOS barrier and can operate within a NLOS environment with the same fidelity as it would in a LOS environment. This has sparked a keen interest within the broadband wireless market to adopt such systems. [7]

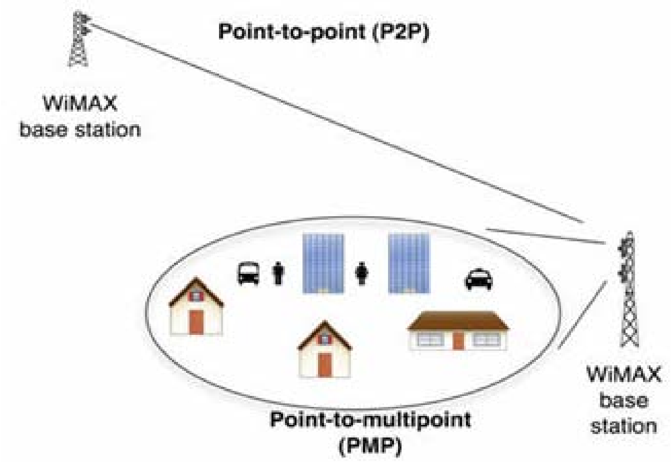

Figure 8: WIMAX PMP.

2.8.3 NLOS versus LOS Propagation

The radio channel of a wireless communication system is often described as being either LOS or NLOS. In a LOS link, a signal travels over a direct and unobstructed path from the transmitter to the receiver. A LOS link requires that most of the first Fresnel zone is free of any obstruction. If these criteria are not met then there is a significant reduction in signal strength.

The Fresnel clearance required depends on the operating frequency and the distance between the transmitter and receiver locations.

In a NLOS link, a signal reaches the receiver through reflections, scattering, and diffractions. The signals arriving at the receiver consists of components from the direct path, multiple reflected paths, scattered energy, and diffracted propagation paths. These signals have different delay spreads, attenuation, polarizations, and stability relative to the direct path. [2]

2.8.4 Cell sizes

Apart from high speeds for individual users and a high overall capacity of a cell, cell size is another important factor that decides if an 802.16 network can be operated economically. Ideally, a single cell should be as large as possible and should have a very high capacity in order to serve many users simultaneously. However, these purposes are mutually exclusive. The larger the area covered by a cell, the more difficult it is to serve remote subscribers.

As a consequence, discrete subscribers have to be served with a lower modulation and higher coding scheme, which reduces the overall capacity of the cell. A cell serving only users in close proximity can have a much higher capacity, as less time has to be spent sending data packets with lower modulation schemes, which requires more time than sending data packets of the same size with 16 and 64 QAM modulation.

In urban and suburban areas, cell sizes will be small because the number of users per square kilometer is high. In rural areas on the other hand, cell sizes need to be much larger in order to cover enough subscribers to make the operation of the network economically feasible. However, the capacity of the cell is reduced as the percentage of subscribers, which are quite distant from the cell, is higher than for the rural scenario. Also, the achievable data rates per user will be lower, especially for more distant subscribers. [8]

2.8.5 NLOS Operation of IEEE 802.16 Based Systems

Line of sight operation is often defined in terms of Fresnel zones. It is shown that the diffraction in radio propagation is minimized if there is no obstacle within the first Fresnel zone, which concentrates most part of wave energy. In a real world deployment scenario, this condition can be accomplished by increasing antenna height.

Since LOS operation imposes severe constraints on the deployment of any wireless network, acceptable system performance under NLOS propagation becomes a major requirement to enable fast network expansion. The first step to enable NLOS propagation is to reduce the carrier frequency below 11 GHz, in order to increase wavelength, thus enhancing radio signal propagation. Furthermore, multipath becomes significant in lower frequencies, which can increase reception performance if appropriate techniques are adopted.

Besides operating at lower frequencies, a set of key functionalities must be implemented at the MAC and PHY layers in order to support NLOS operation in real world scenarios. [9]

2.8.6 NLOS Technology Solutions

WiMAX technology, solves or mitigates the problems resulting from NLOS conditions by using:

- OFDM technology.
- Sub-Channelization.
- Directional antennas.
- Transmit and receive diversity.
- Adaptive modulation.
- Error correction techniques.
- Power control.

2.8.7 OFDM Technology

OFDM Technique: The Orthogonal Frequency Division Multiplexing (OFDM) is a key technique to enable NLOS operation of WiMAX technology, due to the higher multipath robustness achieved at reception. OFDM operation consists of multiplexing information on multiple narrowband subchannels, modulated by a set of orthogonal subcarriers.

The first benefit that arises from the transmission over narrowband subcarriers is the significant complexity reduction of channel equalization algorithms. Figure 9(a) illustrates the radio channel distortion over a wideband single-carrier transmission system. In Figure 9(b), a wideband transmission system is composed of multiple narrowband subcarriers, which are uniformly attenuated due to radio channel distortion. By comparing the effects of radio channel distortion in Figure 9, it becomes clear that equalization tends to be far less complex in radio transmission systems based on narrowband subcarriers, since it reduces to a simple gain recovery (amplification) procedure per subcarrier.

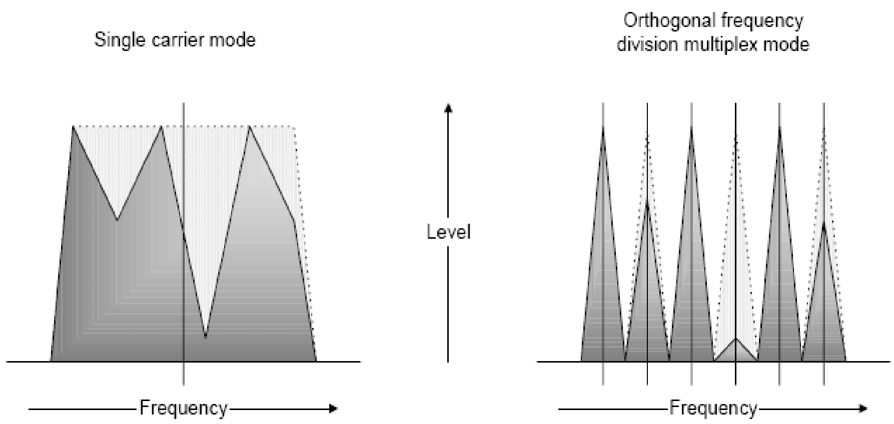

Figure 9: Radio channel distortion in wideband single-carrier and multi-carrier systems: (a) single-carrier transmission system, (b) multi-carrier transmission system.

The OFDM scheme specified in the IEEE 802.16 standard is shown in Figure 10. The symbol structure is composed of a guard interval (Tg) and the useful symbol interval (Tb), with the resulting symbol duration equal to Ts, as depicted in Figure 10. The last T_g portion of the useful symbol, named Cyclic Prefix (CP), in continuously copied on to the guard timePortion. The adoption of Cyclic Prefix increases robustness against multipath fading. [9]

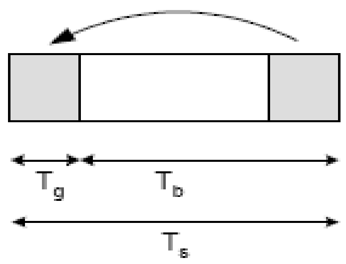

Figure 10: Cyclic Prefix.

2.8.8 Sub-channelization

Most wireless networks are subject to coverage unbalance between uplink and downlink. In fact, subscriber stations are often submitted to cost, physical and resource availability constraints (e.g., maximum antenna height, power consumption, maximum transmission power).

Depending on transmission power constraints of the subscriber station, the system coverage is limited by the uplink coverage, thus causing the link unbalance problem. In order to enhance uplink coverage, a subchannelization technique is specified in the IEEE 802.16 standard, for the OFDM version, illustrated in Figure 68. The SS transmission power is limited to 25 % of the maximum BS transmission power.

In order to increase uplink coverage, a subset of one forth of the available subchannels is selected for transmission, thus allowing the transmission power to be concentrated in a narrower frequency spectrum. By adopting this procedure, the resulting transmission power on the selected subchannels can be increased by a factor of 4, which corresponds to the link balance condition. The price to be paid for coverage enhancement, however, is the reduction of available uplink bandwidth by a factor of 4. [9]

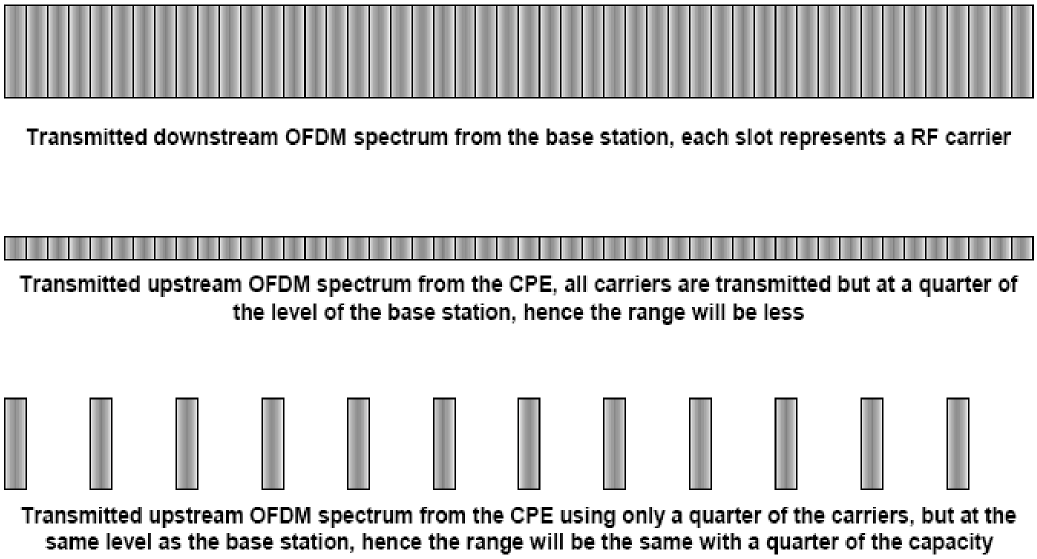

Figure 11: The effect of sub-channelization.

2.8.9 Adaptive Modulation

Adaptive modulation allows the WiMAX system to adjust the signal modulation scheme depending on the signal to noise ratio (SNR) condition of the radio link. When the radio link is high in quality, the highest modulation scheme is used, giving the system more capacity. During a signal fade, the WiMAX system can shift to a lower modulation scheme to maintain the connection quality and link stability. This feature allows the system to overcome time-selective fading.

The key feature of adaptive modulation is that it increases the range that a higher modulation scheme can be used over, since the system can flex to the actual fading conditions, as opposed to having a fixed scheme that is budgeted for the worst case conditions.

Depending on the signal-to-noise ratio (SNR) at the receiver, the SS and the BS negotiate the most appropriate modulation scheme, among the available options (BPSK, QPSK, 16 QAM and 64 QAM), as illustrated in figure 12.

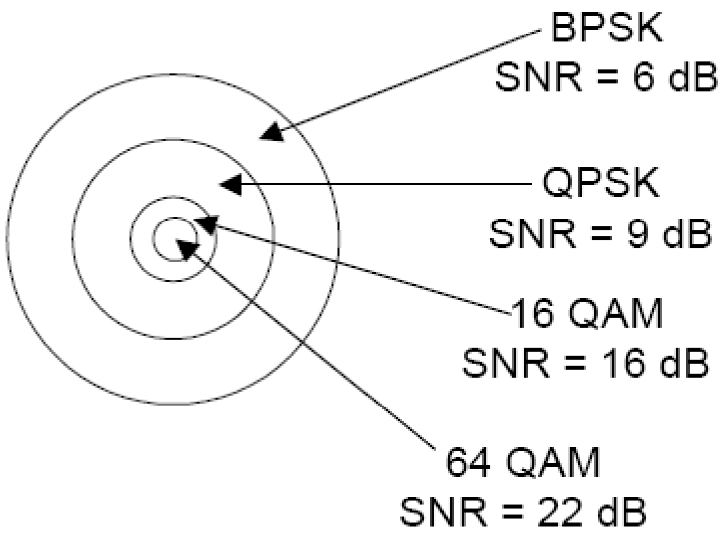

Figure 12: The Cell Radii.

This approach maximizes throughput and connectivity within a cell, as it allows the system to switch between high performance modulation scheme (64-QAM) and high robustness modulation scheme (BPSK) schemes, as the distance between the Base Station to the Subscriber Station varies. This approach has already been adopted in Wi-Fi technology. [9]

2.9 Capacity planning in WIMAX

Studies made had shown that the expected activities of different WIMAX users and the results are shown in the next chart.

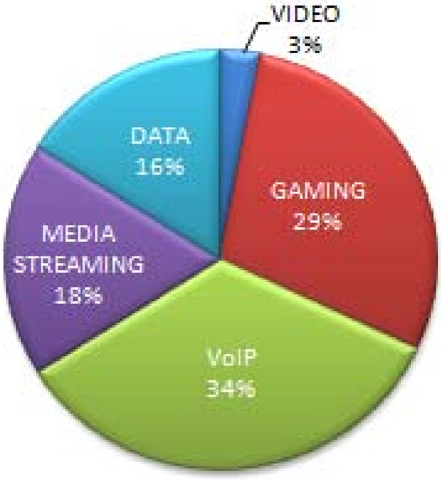

Figure 13: Pie Chart showing expected uses of Fixed WIMAX for different users.

The next chart shows the modulations schemes that will be used for transmissions between the previous users and the base station depending on their locations and the data they require.

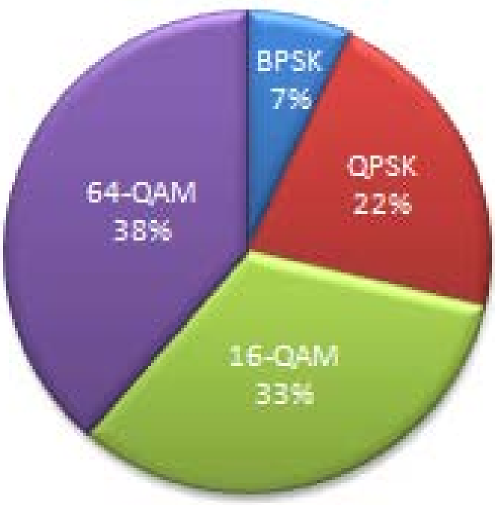

Figure 14: Pie Chart showing the different modulation schemes for the above users.

CHAPTER 3

METHODOLOGY

3.1 Procedure identification

The main task in this project is to fully study the WiMAX as a technology. Research on few aspects of WiMAX will be conducted as its physical layer, coverage and capacity planning. The first phase of the project is to learn about WiMAX uses, characteristics, features and advantages. The second phase will be about the WiMAX and OFDM physical layer description and modulation.

The third phase will be concerning the wireless broadband as a communication feature focusing on its challenges. The fourth stage will be on the WiMAX coverage prediction and performance evolution in addition to designing a computerized application for these purposes. The fifth phase will be covering the WiMAX capacity planning and its application method. The final phase will be to design a Simulink based model for WiMAX testing and evaluation.

3.2 The project flow

Researching and collecting information about WiMAX.

⇩

Researching on IEEE 802.16 PHY layer implementation.

⇩

Studying wireless Broadband.

⇩

Building a Simulink based model for WiMAX.

⇩

Analysis for the designed system.

⇩

Learning about WiMAX Capacity Planning.

⇩

Learning about Coverage Prediction.

3.3 Tools and equipment required

3.3.1 Software used:

MATLAB Simulink.

CHAPTER 4

RESULTS AND DISCUSSION

4.1 IEEE 802.16 PHYSICAL LAYER STUDY

4.1.1 Orthogonal Frequency Division Multiplexing

4.1.1.1 Introduction

Orthogonal Frequency Division Multiplexing (OFDM) is very similar to the well known and used technique of Frequency Division Multiplexing (FDM). OFDM uses the principles of FDM to allow multiple messages to be sent over a single radio channel.

In OFDM serial higher rate data sequence is converted to a parallel low rate data sequence which will be modulated on orthogonal sub-carriers , low rate streams have a narrow band transmission bandwidth which would be smaller than channel coherence bandwidth causing no frequency selective fades or distortions but only attenuation and minimal inter-symbol interference (ISI) . These attenuations that the whole sub-carriers of the signal might suffer can be equalized using channel estimation.

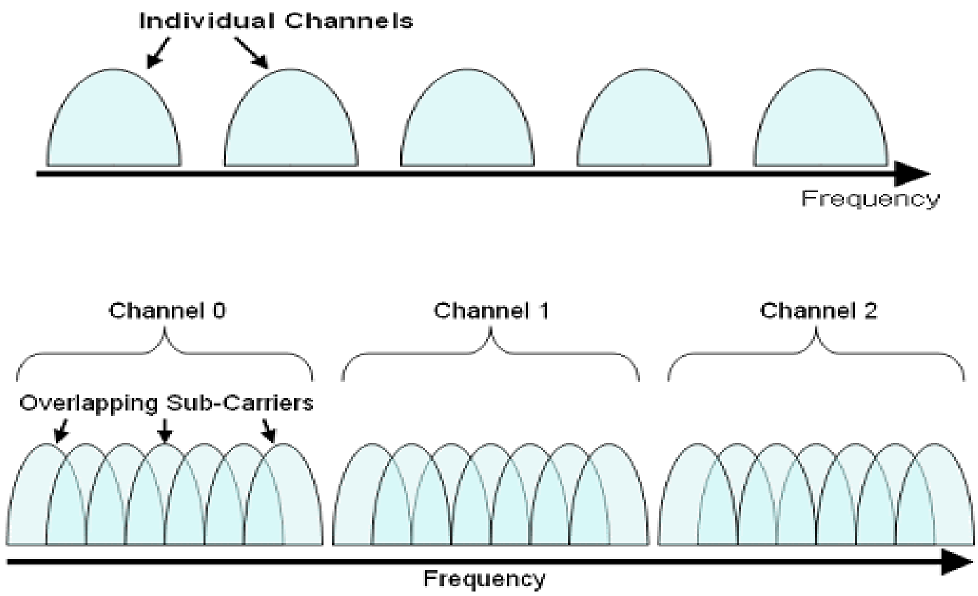

Figure 15: Single carrier vs. OFDM multi-carrier transmission.

Also spectral efficiency of orthogonal frequency multiplexed signals is better than single carrier transmitted signals as in OFDM the multiple sub-carriers used can overlap without having inter-carrier interference (ICI) due to the orthogonal nature of the multiple sub-carriers ,However in other techniques like FDM signals should have a sufficient guard band between each other to avoid interference. [11]

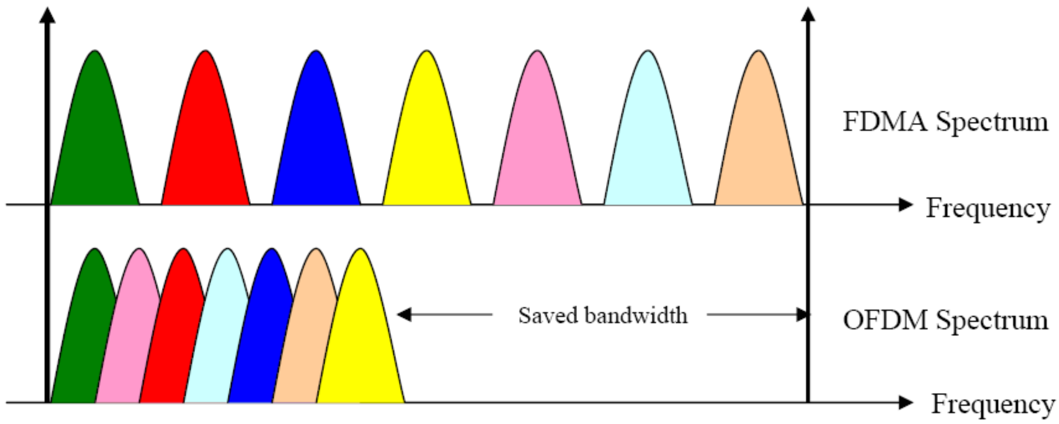

Figure 16: Spectral efficiency of OFDM.

4.1.1.1.1 Orthogonality

Signals are orthogonal if they are mutually independent of each other. Orthogonality is a property that allows multiple information signals to be transmitted perfectly over a common channel and detected, without interference. Loss of Orthogonality results in blurring between these information signals and degradation in communications. Many common multiplexing schemes are inherently orthogonal like:

- **Time Division Multiplexing (TDM):** each single source is assigned a different time slot to transmit its information to prevent interference, which makes TDM systems time orthogonal by nature.
- **Frequency Division Multiplexing (FDM):** each single source is assigned a different sub-band with a guard band between each allocated sub-band to prevent interference, which makes FDM orthogonal in frequency by nature**.**

In OFDM data is converted into lower rate parallel streams, each one is modulated on a different sub-carrier with no guard period allowed with no inter-carrier interference due to the orthogonal nature of these sub-carriers. The condition of Orthogonality would be as follows: [13] [14] [15] 16] [17]

- **Continuous in time:**

$$\int_0^T \cos(2\pi n f_0 t) \times \cos(2\pi m f_0 t) dt = 0 \quad (m \neq n)$$

- **Discrete in time:**

$$\sum_{k=0}^{N-1} \cos\left(\frac{2\pi kn}{N}\right) \times \cos\left(\frac{2\pi km}{N}\right) = 0 \quad (n \neq m)$$

4.1.1.1.2 Multi-carrier Transmission

The data stream is split into K parallel sub-stream, and each is modulated on its own sub-carrier at frequency f_k described by the complex baseband exponential exp ($j2\pi f_k t$).

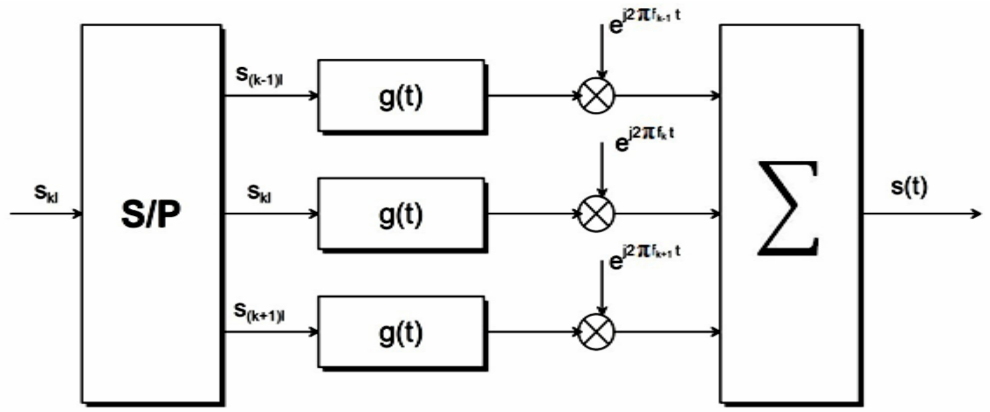

Figure 17: Multi-carrier transmission transmitter block diagram.

<u>Where:</u> k: frequency index

1: time index

S_{k1}: complex modulation symbols

g(t):pulse shaping filter

The complex baseband signal would be given by:

$$\sum_{\mathbf{k}} \mathbf{e}^{\mathbf{j2\pi f_k t}} \sum_{\mathbf{l}} \mathbf{s_{kl} g(t - 1T_s)}$$

Another approach of multi carrier generation is that a shifted bank of band-pass filters is excited by the spitted parallel data sub-streams, where the response of each filter will be:

$$\mathbf{g_k(t) = e^{J2\pi f_k t} g(t)}$$

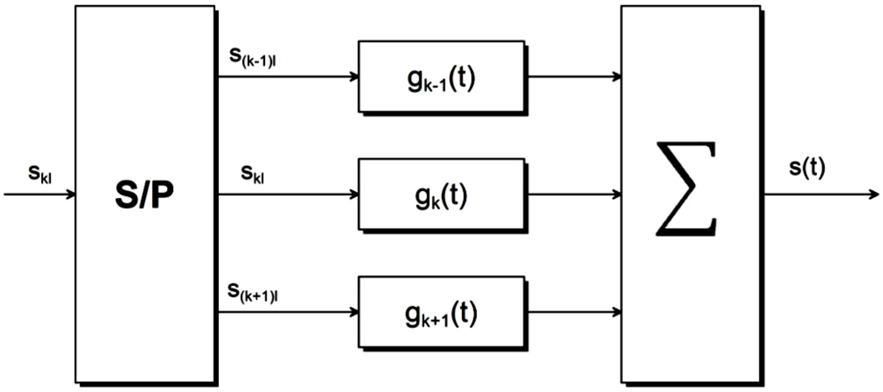

Figure 18: Another approach for multi-carrier transmission transmitter.

And the complex baseband signal would be given by:

$$\mathbf{s(t) = \sum_{l}\sum_{k} s_{kl} g_k(1 - lT_s)}$$

It can be noticed that if the modulation symbols Skl is replaced by Skl * exp(j2πfkt), it would return to the first approach .However the second approach is closer for implementation especially for OFDM where the bank of the band-pass filters would be proven later to be just a Fast Fourier Transform (FFT).

4.1.1.2 OFDM Generation and Reception:

After learning about Orthogonality and multi-carrier concept, An OFDM model will be constructed based on those two concepts besides fulfilling the Nyquist Criterion to avoid any inference keeping in mind that OFDM will offer overlapping narrow sub-bands each carrying the information of the low rate sub-stream.

From Orthogonality, each low rate sub-stream will be modulated on a frequency which is the integer multiple of a fundamental frequency.

From multi-carrier concept, the pulse shaping filter response g(t) that will fulfils the two other concepts.

From Nyquist criterion will gives us the conditions of ISI free transmission will be deducted by either two of the following approaches:

- **Band limited pulses**

1. The most famous example for it is the raised cosine pulses.
2. In frequency domain, Narrow sub-bands can’t overlap as it would result in having interference.

- **Time limited pulses**

1. In time domain, it would be a rectangular pulse shaping filter with the signals modulated at multiple integers of the fundamental frequency.
2. In frequency domain, narrow sub-bands would have a sine shape (the transform or rectangular pulses) of its peak corresponding to the nulls of other sub-bands resulting in interference free transmission.

The second approach will fulfill all conditions ,The choice will be a time limited pulse of time limit [-Ts/2,Ts/2] ,and in frequency domain it would result in a sine narrow sub-bands with the first null at 1/Ts which would gives a hint about the spacing between sub-carriers.

Then the OFDM symbol in time would be the sum of orthogonal sinusoidal each having an integer number of periods within the time limit [-Ts/2, Ts/2] and a multiple of the fundamental frequency f0=l/Ts.

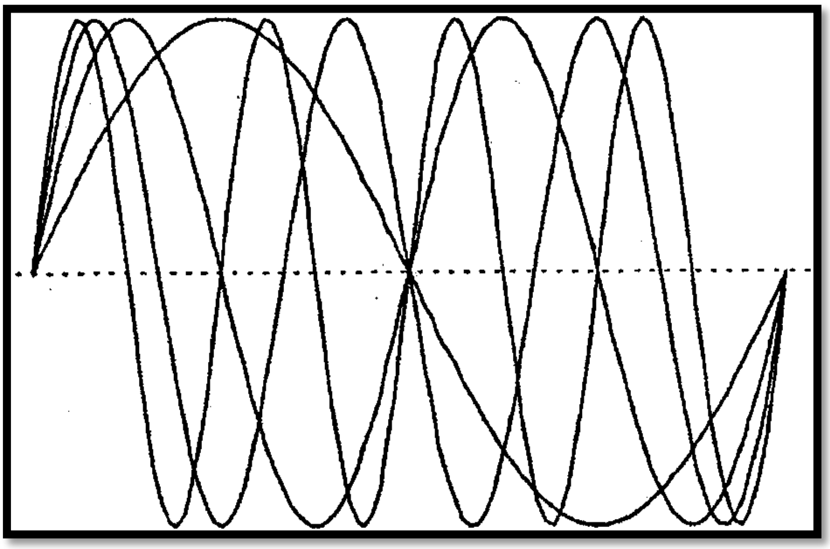

Figure 19: OFDM sub-carriers in time domain.

In frequency domain the OFDM symbol would be orthogonal sine functions with a spacing of 1/Ts with peak of each sine function corresponding to the nulls of the other sine functions.

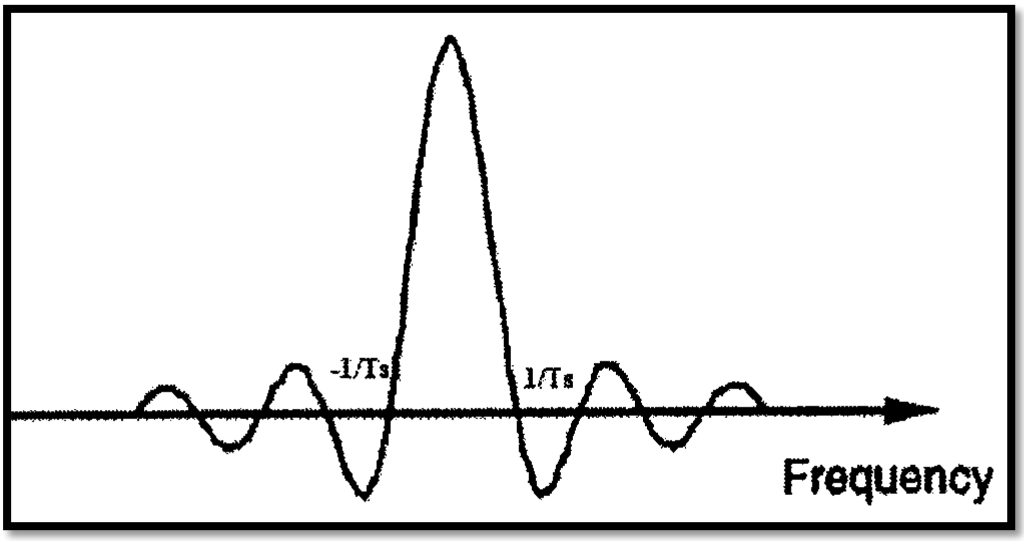

Figure 20: Spectra of OFDM sub-channel.

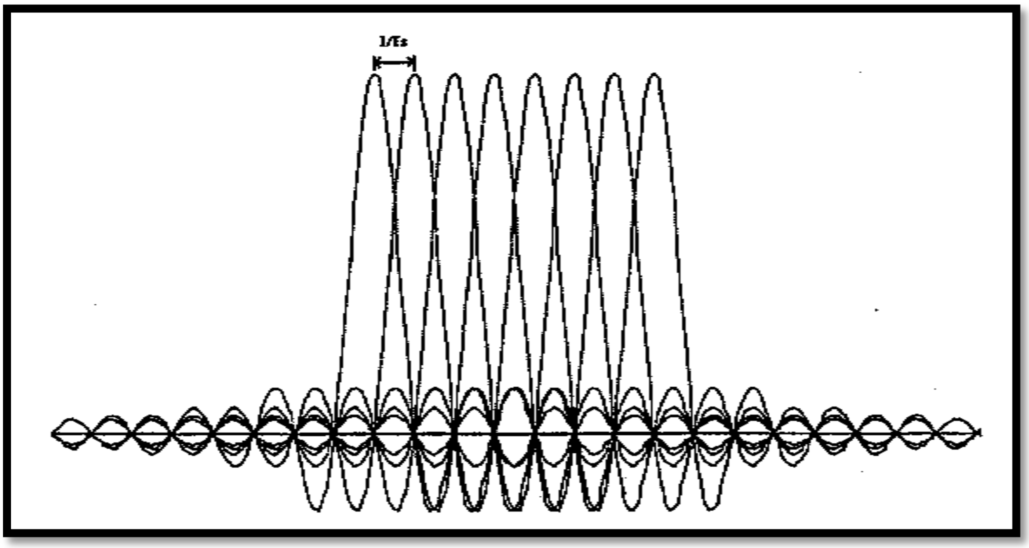

Figure 21: Spectra of OFDM signal.

The OFDM signals can be generated as follows:

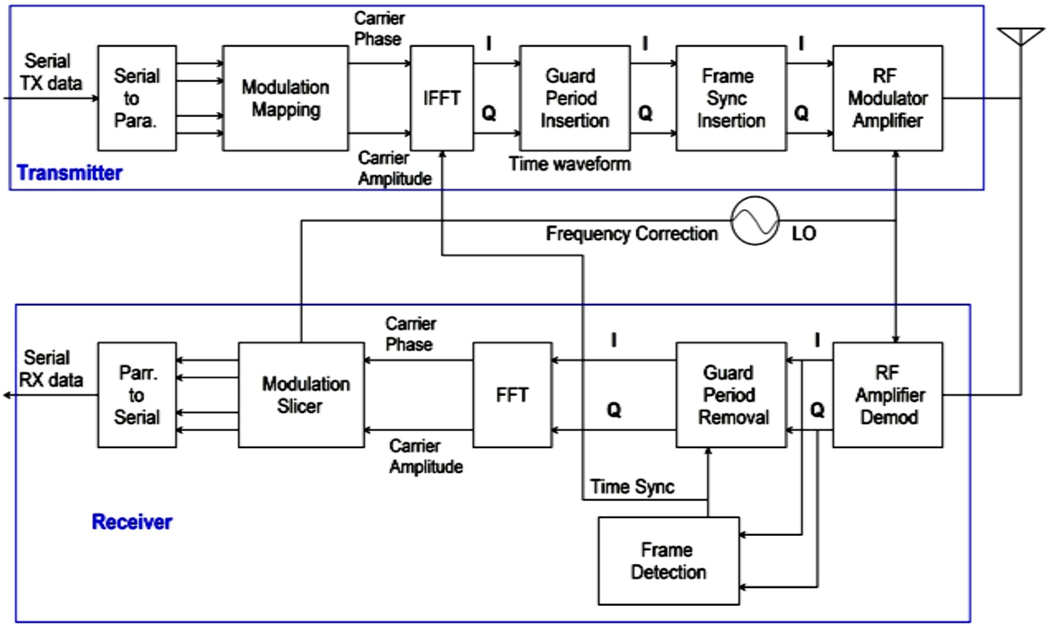

Figure 22: OFDM Transceiver.

The OFDM modulated signal can be expressed as:

$$\mathbf{s(t)} = \sum_{\mathbf{k=0}}^{\mathbf{N-1}} \mathbf{s_k e^{j2\pi f_k t}} = \sum_{\mathbf{k=0}}^{\mathbf{N-1}} \mathbf{s_k \varphi_k (t)}, \qquad \mathbf{0 \le t \le T_s}$$

And

$f_k = f_o + \Delta f$, $\Delta f * Ts = 1$ → can be considered as the Orthogonality condition.

Also;

$$\boldsymbol{\varphi}(\mathbf{t}) = \begin{cases} \mathbf{e^{j2\pi f_k t}}, & if\ 0 \le t \le \mathbf{T_s} \\ \mathbf{0}, & otherwise, \end{cases}$$

Satisfying the Orthogonality conditions as follows:

$$\frac{\mathbf{1}}{\mathbf{T_s}} \int_{\mathbf{0}}^{\mathbf{T_s}} \boldsymbol{\varphi}_{\mathbf{k}}(\mathbf{t}) \boldsymbol{\varphi}_{\mathbf{l}}(\mathbf{t}) \mathbf{dt} = \boldsymbol{\delta}[\mathbf{k - l}]$$

Where

$$\boldsymbol{\delta}[\mathbf{n}] = \begin{cases} \mathbf{1}, & if\ n = 0 \\ \mathbf{0}, & otherwise, \end{cases}$$

The OFDM modulated signal can be demodulated as follows:

$$\frac{\mathbf{1}}{\mathbf{T_s}} \int_{\mathbf{0}}^{\mathbf{T_s}} \mathbf{s(t)\, e^{-j2\pi f_k t} dt} = \frac{\mathbf{1}}{\mathbf{T_s}} \int_{\mathbf{0}}^{\mathbf{T_s}} \left(\sum_{\mathbf{l=0}}^{\mathbf{N-1}} \mathbf{s_l} \boldsymbol{\varphi}_{\mathbf{l}}(\mathbf{t}) \right) \boldsymbol{\varphi}_{\mathbf{k}}^{*}(\mathbf{t})\, \mathbf{dt} = \sum_{\mathbf{l=0}}^{\mathbf{N-1}} \mathbf{s_l} \boldsymbol{\delta}\, [\mathbf{l - k}] = \mathbf{s_k}$$

4.1.1.2.1 Generation of sub-carriers using FFT

Here is a description for the relationship between OFDM and discrete Fourier transform (DFT), which can be implemented by low complexity fast Fourier transform (FFT) From the previous discussion.

An OFDM signal can be expressed by:

$$\mathbf{s(t) = \sum_{k=0}^{N-1} s_k e^{j2\pi f_k t}}$$

Sampling s(t) at Tsampling=Ts/N ,where N is the number of sub-carriers.Then;

$$\mathbf{S_n = s(n\Delta_s) = \sum_{k=0}^{N-1} s_k e^{j2\pi f_k \frac{nT_s}{N}}}$$

Setting f_o=0 (dc sub-carrier) and then fk*Ts=k, then Sn becomes:

$$\mathbf{S_n = \sum_{k=0}^{N-1} s_k e^{j2\pi f_k \frac{2\pi kn}{N}} = IDFT\{s_k\}}$$

Where IDFT denotes the inverse discrete Fourier transform. Therefore, the OFDM transmitter can be implemented using the IDFT. For the same reason, the receiver can be also implemented using DFT.

FFT algorithm provides an efficient way to implement DFT and IDFT. It reduces the number of complex multiplications from N2 to N/2 log2N for an N-point DFT or IDFT. [13] [14] [15] [16] [17]

Then the OFDM signal can be implemented using Fast Fourier Transform as follows:

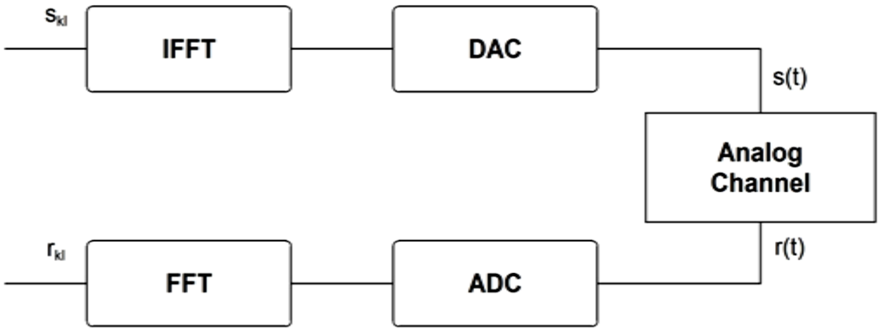

Figure 23: OFDM transmission using FFT.

4.1.1.2.2 Guard Time and Cyclic extension

As OFDM signal consists of the sum of a low rate data sequence which was originally split from higher rate sequence, it s clear that the new low rate data sequence have a long extended period by the factor N (FFT size) which would be larger than the channel delay spread, which makes OFDM an efficient way to deal with multi-path delay spread.

Previously a perfect synchronization between the transmitter and the receiver was assumed, however this is not the true real life case due to multi-path propagation caused by the radio transmission signal reflecting off objects in the propagation environment, such as walls, buildings, mountains, etc.

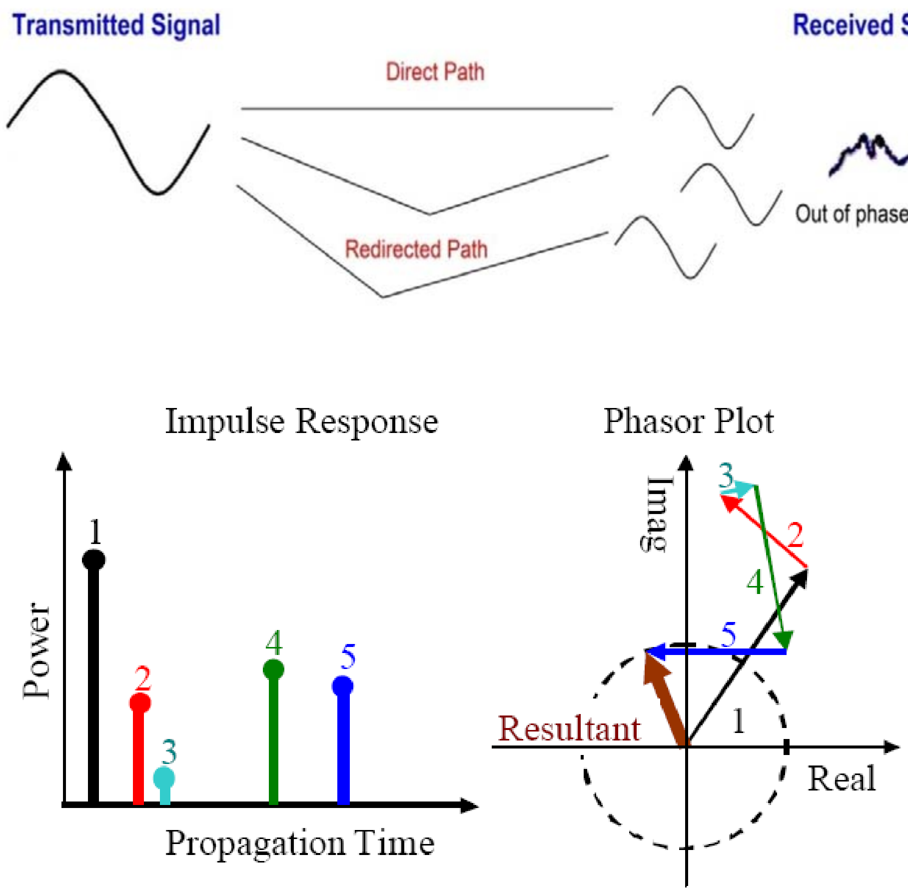

Figure 24: Effect of multi-path channel on a sample signal.

For OFDM signals propagation in a multi-path channel multi-path signals arrives as the sum of the direct path and the delayed path (OFDM signals with phase rotations according to the difference in path length), leading to the distortion of the orthogonal sinusoidals which now will not have an integer number of cycles within the OFDM symbol. Orthogonality is lost and severe ICI occurrence is the result.

In addition, Multi-path signals results in the spreading of the symbol boundaries causing energy leakage between different symbols leading to ISI.

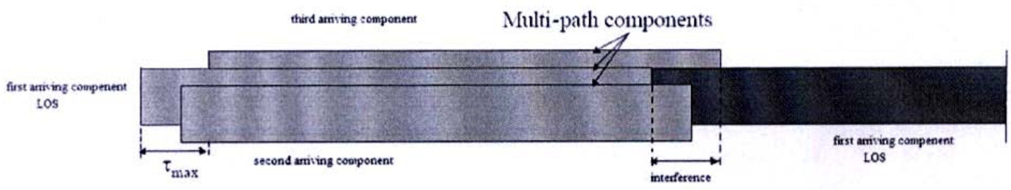

Figure 25: Effect of multi-path channel on OFDM.

This would disqualify OFDM as a useful technique in a multi-path channel. For that the following approaches were suggested to diminish the multi path effects:

I. Guard time: zero samples are introduced at the beginning of the OFDM symbol; it is chosen to be larger than the delay spread of the channel.

This approach would eliminate ISI between OFDM symbols as for the delayed symbol it would only interfere with the guard time of the next symbol, however, this approach would not solve the ICI problem as delayed subcarriers of same symbol would not have an integral number of cycles within the FFT symbol duration.

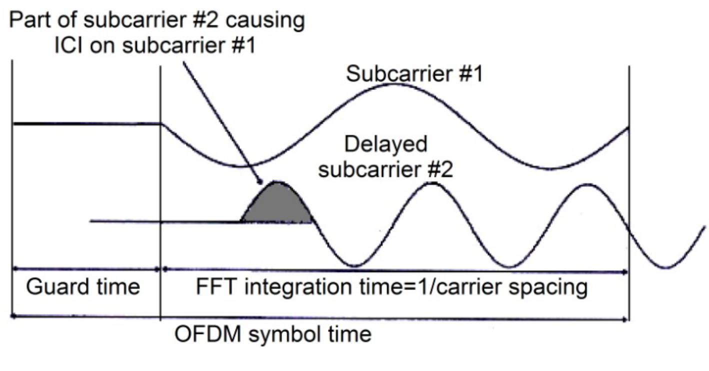

Figure 26: Using guard time technique.

II. **Cyclic prefix:** this is done by taking a number of samples from the end of symbol period appending them to the front of the period. The concept behind this and what it means comes from the nature of the IFFT/FFT process. When the IFFT is taken for a symbol period (during the OFDM modulation), the resulting time sample sequence is technically periodic.

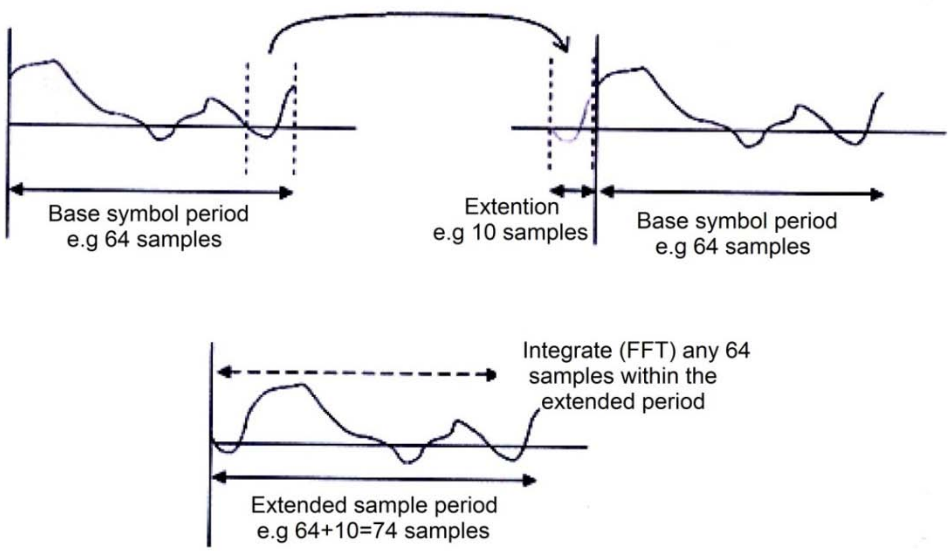

Figure 27: Using cyclic prefix technique.

This approach eliminates both ISI and ICI .OFDM signals propagating in a multi-path channel, its sub-carriers will always have an integer number of cycles within the FFT interval which will preserve orthogonality.

Cyclic prefix eliminates ISI as delayed OFDM signals of previous signals only will interfere with the cyclic extended part of the present symbol provided that the cyclic prefix duration is larger than the channel delay spread, at the receiver the cyclic prefix will be discarded and each block of N received samples is converted back to the frequency domain using an FFT free of ISI.

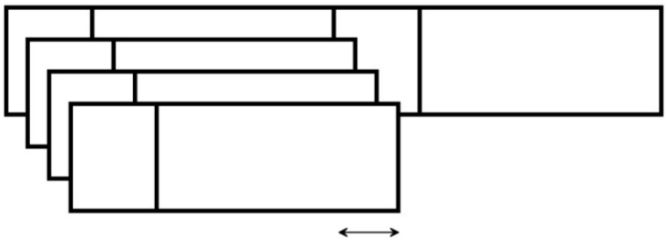

Figure 28: Using cyclic prefix.

Cyclic prefix would also eliminates the ICI as cyclically extending the OFDM symbol will ensure that delayed replicas of the OFDM symbol always have an integer of cycles within the FFT interval.

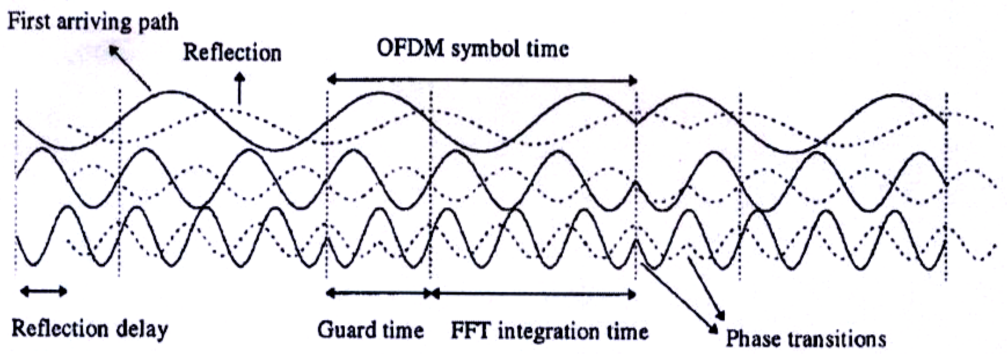

Figure 29: Phase of rotation of each sub-carrier n multi-path channels.

The last figure shows the elimination of ICI using cyclic prefix, it is clear that each sub-carrier within the FFT interval would be the summation of all paths, but it would still have an integer number of cycles within the FFT interval preserving orthogonality, However it would suffer from a phase rotation dependent on the sub-carrier frequency.

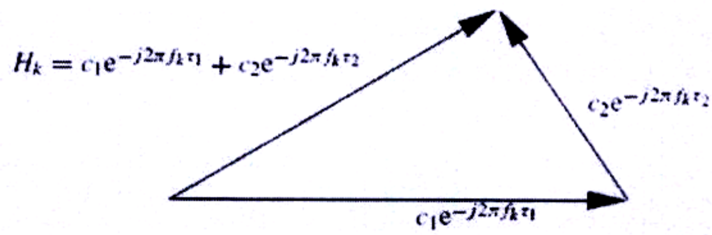

Figure 30: Received signal after propagation in a multi-path channel.

The N frequency domain samples (with phase rotations) are each processed with a simple one-tap Frequency Domain Equalizer (FDE) and applied to a decision device to recover the data symbols. The one-tap FDE simply multiplies each FFT coefficient by a complex scalar that eliminates the channel effect.

III.<u>Cyclic suffix:</u> This is done by taking a number of samples from the beginning of symbol period and appending them to the end of the period.

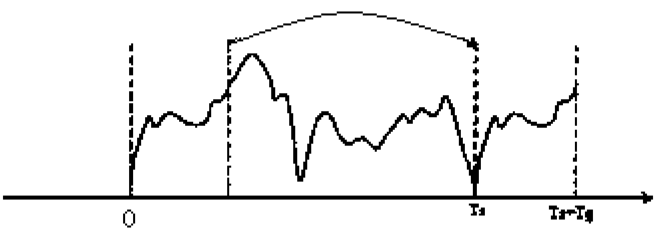

Figure 31: OFDM symbol with cyclic suffix.

At the receiver, A duration of Tg (guard period) will be discarded from the beginning of the symbol and restored from the end of the symbol which have the same information being discarded

IV. Summing up and important remarks

- The guard interval of length N_g, is an overhead that results in a power and bandwidth penalty, since it consists of redundant symbols.
- Elimination of ISI and ICI is done using cyclic extension of OFDM symbol, either cyclic prefix or cyclic suffix.
- Cyclic prefix is most likely to be used to eliminate ISI and ICI
- Cyclic prefix eliminates ISI and ICI, but phase rotation of sub-carriers would be treated by Cyclic FDE.
- Frequency domain equalizer FDE is used to eliminate those phase rotations, by just multiplying each FFT coefficient by an estimated complex scalar.
- Equalization in OFDM is not a complex operation as in single carrier transmission, in OFDM data would be restored to frequency domain by FFT which is a part of the receiver. However, in single carrier transmission a pair of IFFT/FFT will be used to transform time domain signal to frequency domain to be equalized and then retransform it to time domain.
- If the channel delay is large, the channel shortening technique is used which consists of a time domain equalizer (TEQ) placed in cascade with the channel to produce an effective impulse response that is shorter than the channel impulse response. [2] [13] [14] [15] [16] [17]

4.1.1.2.3 RF modulation

The output of the OFDM modulator generates a base band signal, which must be mixed up to the required transmission frequency. This can be implemented using analog techniques as shown in Figure 32 or using a Digital Up Converter as shown in Figure 33. Both techniques perform the same operation, however the performance of the digital modulation will tend to be more accurate due to improved matching between the processing of the I and Q channels, and the phase accuracy of the digital IQ modulator. [19]

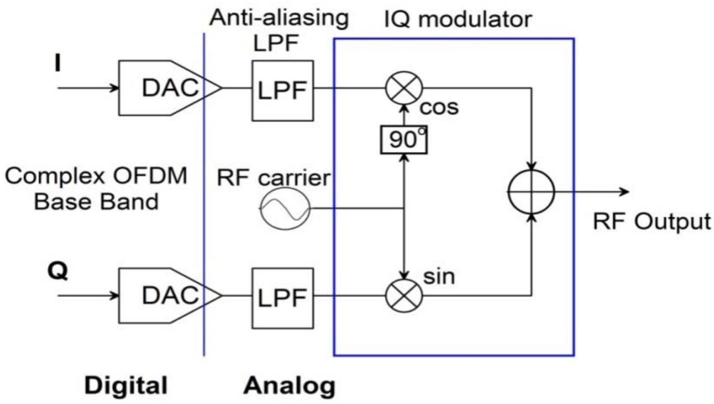

Figure 32: Analog RF modulation.

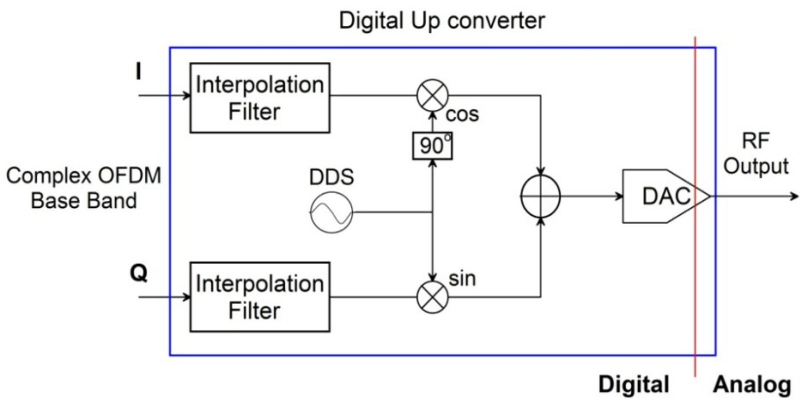

Figure 33: Digital up converter.

The pass-band signal can be represented as following: s(t): complex baseband signal

$$\mathbf{s(t) = s_I(t) + \; js_Q(t)}$$

Where the real part, $S_I(t)$, is called in-phase component of the baseband signal and the imaginary part, $S_Q(t)$ is called quadrature component [19]

$$\boldsymbol{s(t) = \sum_{k=0}^{N-1}(R\{s_k\}\cos(2\pi f_k t) - I\{s_k\}\sin(2\pi f_k t) + j\sum_{k=0}^{N-1}(I\{s_k\}\cos(2\pi f_k t) + R\{s_k\}\sin(2\pi f_k t)}$$

Then

$$\boldsymbol{s_I(t) = \sum_{k=0}^{N-1}(R\{s_k\}\cos(2\pi f_k t) - I\{s_k\}\sin(2\pi f_k t)}$$

And

$$\boldsymbol{s_Q(t) = \sum_{k=0}^{N-1}(I\{s_k\}\cos(2\pi f_k t) + R\{s_k\}\sin(2\pi f_k t)}$$

Then the pass-band signal can be expressed by:

$$\mathbf{s_p(t) = R\{s(t) \times e^{j2\pi f_c t}\} = s_I(t)\cos(2\pi f_c t) - s_Q(t)\sin(2\pi f_c t)}$$

For

$$\mathbf{S_k = d_k e^{j\theta k}}$$

Then

$$s_p(t) = \sum_{k=0}^{N-1} d_k \cos(2\pi(f_c + f_k) + \theta_k)$$

4.1.1.2.4 Band-limiting OFDM and windowing

OFDM in the time domain is equivalent to a sum of modulated sinusoidal carriers that are each windowed in time with a rectangular window function, also known as a boxcar window function. This window defines the boundary of each OFDM symbol and determines the frequency response of the generated OFDM signal.

The next figure shows an example time waveform for a single carrier OFDM transmission using Phase Shift Keying (PSK). The amplitude of the sub-carrier is fixed and the phase is varied from symbol to symbol to transmit the data information. The sub-carrier phase is constant for the entire symbol, resulting in a step in phase between symbols. These sharp transitions between symbols result in spreading in the frequency domain. [20]

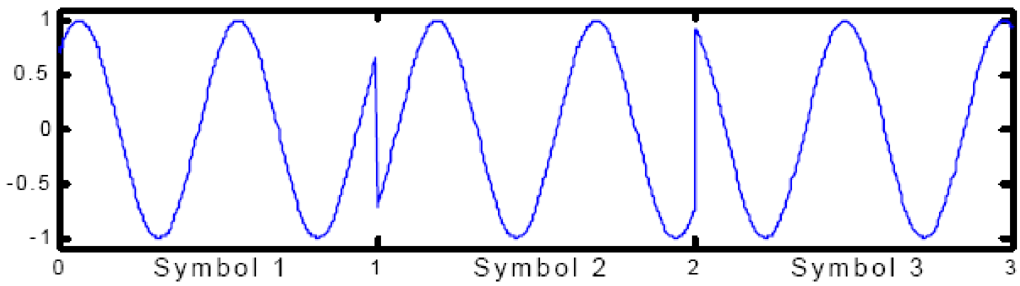

Figure 34: Single sub-carrier waveform.

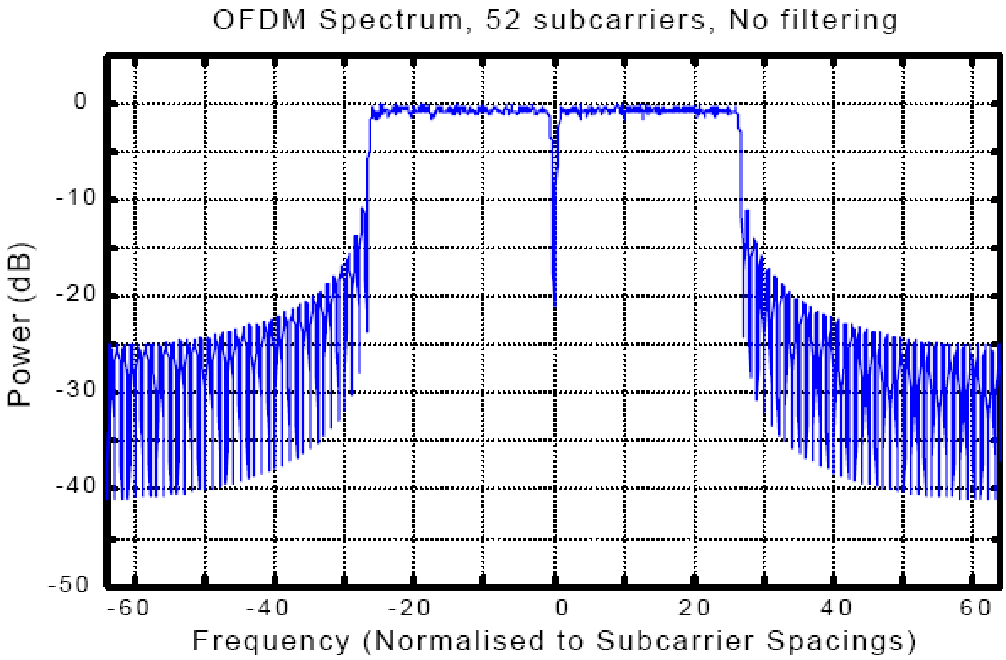

Figure 35: OFDM spectrum using a 52 subcarrier with no out band reduction.

The last figure shows the spectrum of a 52 sub-carrier OFDM signal (same as IEEE802.11a) with no band-pass limiting. The out of band components only fall off slowly due to the since roll off of each sub-carrier. If the number of sub-carriers is increased to a 1536 sub-carrier OFDM signal(same as type in DAB),it is noticed that sides-lobes roll off faster than the 52-subcarrier case. [20]

However the side-lobes are still significant (> -40 dBc) even far away from the edge of the OFDM main signal block. These side-lobes increase the effective bandwidth of the OFDM signal, degrading the spectral efficiency.

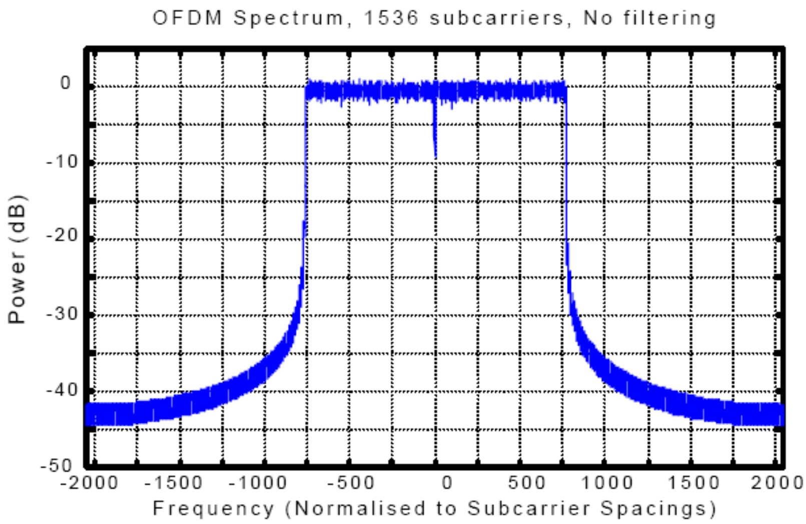

Figure 36: OFDM spectrum using a 1536 subcarrier with no out band reduction.

There are two common techniques for reducing the level of the side-lobes to acceptable limits: [20]

- Band pass filtering the signal.
- Adding a RC guard period.

4.1.1.2.4.1 Band Pass Filtering

Whenever signals are converted from the digital domain to an analog waveform for transmission, filtering is used to prevent aliasing occurring. This effectively band pass filters the signal, removing some of the OFDM side-lobes.

The amount of side-lobe removal depends on the sharpness of the filters used. In general digital filtering provides a much greater flexibility, accuracy and cut off rate than analog filters, making them especially useful for band limiting of an OFDM signal.

Finite Impulse Response (FIR) filters using the windowing method (Kaiser window) can be used as a digital filtering approach, the next figures will show us the effect of filtering on OFDM spectrum using various window width and various transition width. A low number of sub-carriers were used in these plots so that the roll off of the FIR filtering could be seen. [20]

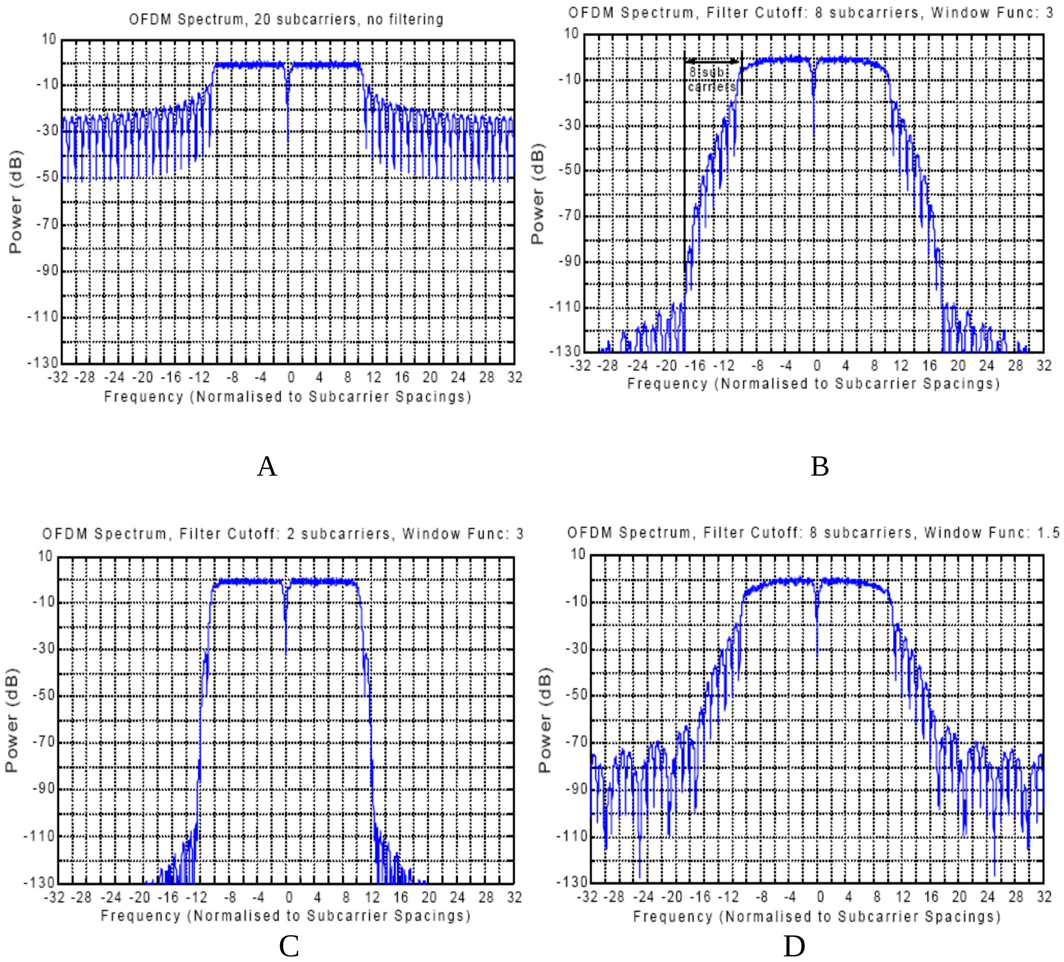

Figure 37: OFDM spectrum without band reduction using filtering.

Where:

A. OFDM spectrum with no band pass filtering.

B. Kaiser window width of 3 (Side lobe attenuation of 89 dB). The transition width of the filter was 8 sub-carrier spacing (24 tap FIR filter).

C. Kaiser window width of 3 (Side lobe attenuation of 89 dB). The transition width of the filter was 2 sub-carrier spacing (96 tap FIR filter).
D. Kaiser window width of 1.5 (Side lobe attenuation of 40 dB). The transition width of the filter was 8 sub-carrier spacing (12 tap FIR filter).

The filtering removes virtually all of the side lobes allowing separate blocks of OFDM signals to be packed very closely in the frequency domain improving the spectral efficiency, but does so at the cost of the computational expense of implementing the FIR filtering, and it reduces the effective SNR of the OFDM channel. Also filtering the OFDM signal, chops off significant energy from the outer sub-carriers, distorting their shape and causing ICI.

The computational overhead added by the FIR filters can be expressed by the number of tapes used to implement this filter, it is noticed in the last figures as the transition width decreases the number of tapes increases and filter becomes more complex, the number of tapes for a required FIR filter is given by:

$$\mathbf{N_{taps}} = \mathbf{ceil}\left(\frac{\mathbf{W_T.\,IFFT}}{\mathbf{F_T}}\right)$$

Where W_T is the transition bandwidth to generate the FIR filter, IFFT is the IFFT size, F_T is the transition width of the filter normalized for sub-carrier spacing.

In applications where the required number of taps in the filter is high (> 100), it is probably more efficient to implement it using an FFT implementation of an FIR filter. Another method for reducing the number of calculations is to implement the filtering using an IIR filter, however a review of the amount of ISI caused by the non-linear phase of the filter would need further investigation. [20]

4.1.1.2.4.2 Raised Cosine Guard Period

One of the simplest methods for suppressing the side-lobes of an OFDM signal is to round the guard period of the OFDM signal, tapering it smoothly to zero before the next symbol. This tapering smoothes the transition between symbols resulting in reduced side-lobe power. Figure 38 shows the makeup of a single OFDM symbol with a Raised Cosine (RC) guard period. This section of the guard period is windowed with a squared cosine shape ($\cos(\theta)^2$), hence the name raised cosine.

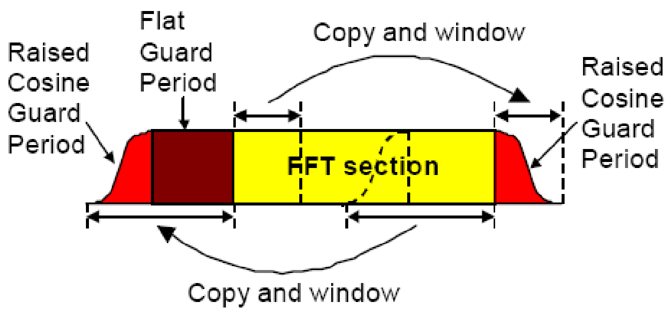

Figure 38: Using a raised cosine guard time.

The raised cosine section of a guard period can be overlapped with the previous and next symbol as this section of the guard period only provides minimal protection against multipath and timing errors, and is ignored at the receiver. Because this section tapers to zero it results in minimal additional ISI. The main advantage of overlapping is that the length of the raise cosine section can be made double in length without incurring additional time overhead.

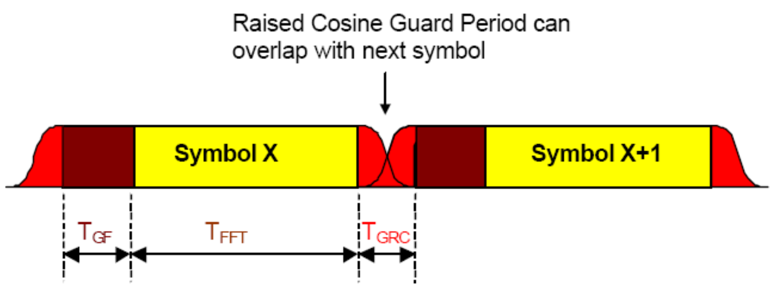

Figure 39: Overlapping raised cosine guard time.

The following figure shows the effect of adding a RC guard period to an OFDM signal for out of band side-lobes reduction: [20]

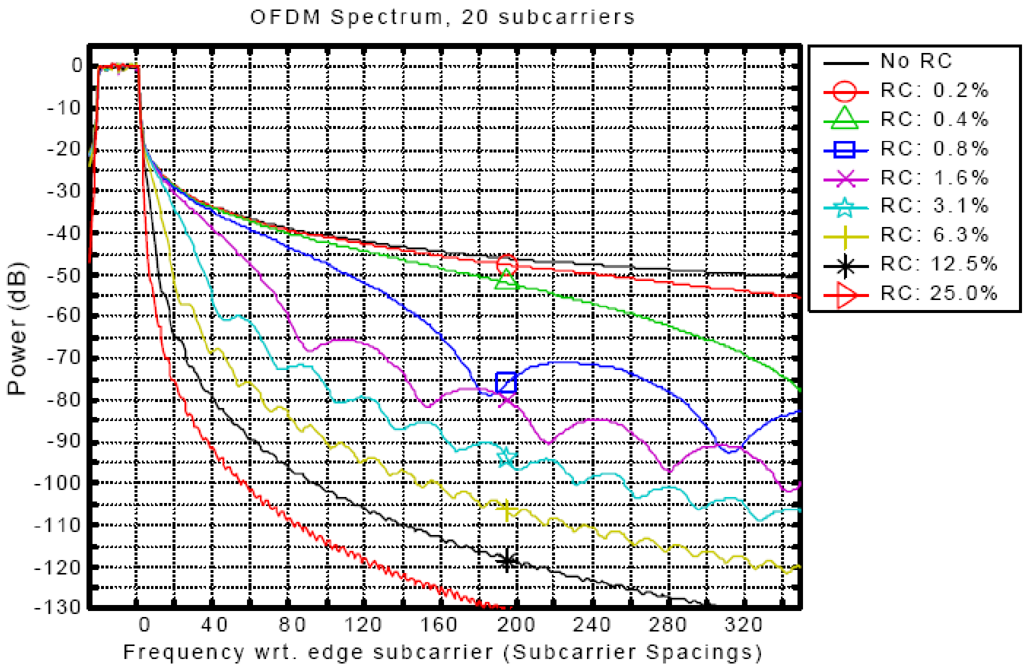

Figure 40: OFDM spectrum using various raised cosine durations.

The RC guard period length has been specified as a percentage of the flat section of the OFDM symbol, that is:

$$\mathbf{RC} = \frac{\mathbf{T_{GRC}}}{\mathbf{T_{FFT}\,T_{GF}}} \times \mathbf{100}\ \%$$

Where RC is the raised cosine percentage (RC/100: roll off factor), T_{GRC} is the length of the RC guard period, T_{FFT} is the length of the FFT section of the symbol and T_{GF} is the length of the flat guard period.

4.1.1.3 OFDM advantages and disadvantages

4.1.1.3.1 Bandwidth Efficiency

For an OFDM signal consisting of N sub-channels, the signal bandwidth is about (N+l)Δf. Since the transmission rate of each sub-channel is symbols/sec, the total transmission rate of OFDM signal is 1/T symbol/sec. Therefore, the bandwidth efficiency of the OFDM system is:

$$\boldsymbol{\eta} = \frac{\mathbf{N/T}}{(\mathbf{N}+\mathbf{1})\Delta\mathbf{f}} = \frac{\mathbf{N}/(\mathbf{T_s}/\mathbf{T_g})}{(\mathbf{N}+\mathbf{1})/\mathbf{T_s}} = \frac{\mathbf{1}}{\mathbf{1}+\frac{\mathbf{1}}{\mathbf{N}}} \times \frac{\mathbf{1}}{\mathbf{1}+\frac{\mathbf{T_g}}{\mathbf{T_s}}}$$

For most practical OFDM systems, N is much larger than 1 and the guard interval or cyclic extension is much smaller than the OFDM symbol duration, so η = 1 bit/sec/Hz. If each symbol carries k bit information, the bandwidth efficiency will be k bits/sec/Hz.

In QPSK the bandwidth efficiency is 1 bit/sec/Hz and in BPSK the bandwidth efficiency is 1/2 bit/sec/Hz, But OFDM provides a higher bandwidth efficiency which is k bits/sec/Hz.

4.1.1.3.2 Multi-path environment

Due to splitting of serial high rates into lower rates sequences, the transmission bandwidth is smaller than the coherence bandwidth of the channel and signal doesn't suffer from frequency selective fading anymore.

In addition, when cyclic prefix (Guard interval) duration is larger than the delay spread of the channel, the symbols doesn't suffer from ISI and orthogonality of the signal is not lost.

4.1.1.3.3 Synchronization

One of the main disadvantages of the OFDM is the sensitivity to synchronization; OFDM is very sensitive to frequency offset as it destroys the orthogonality and results in ICI, But OFDM is less sensitive to time offset as for time offset less than the cyclic prefix the result is only phase rotations of sub-carriers which can be easily fixed using a single tap frequency domain equalizers.

4.1.1.3.4 Peak Average Power Ratio PAPR

It can be easily checked that for an OFDM signal with N sub-channels and a normalized symbol power ,the peak power can be as large as N^2 while the average power is N consequently, the largest PAPR will be: PAPR = N.

It should be noted that the probability for an OFDM signal to have a large PAPR is very small even though the largest possible PAPR is very large. When an OFDM signal is passed through a nonlinear device, such as a transmitter power amplifier, it will suffer significant nonlinear distortion, which generates spectral spreading and in-band noise.

Practical power amplifiers are linear only over a finite range of input amplitudes. In order to prevent saturation and clipping of the OFDM signal peaks, the amplifiers must be operated with sufficient "back-off". The required back-off increases with the PAPR and, hence, the number of sub-carriers N. However, increased back-off reduces the efficiency of the power amplifier.

Several methods are proposed to solve the PAPR problem:

- Coding: code-words of low PAPR will be chosen.
- Clipping and filtering: clipping of large peaks and using error correcting codes to restore distorted bits, filtering is used here to reduce the amount of out of band components (Spectral spreading).
- Artificial signals: When the transmitter and receiver uses M-point FFT and IFFT, not all M frequencies carry data, only N < M number of carriers actually contain data. Thus, it has a few empty carriers per OFDM symbol. The basic premise of this technique is to add sine waves at these empty carrier frequencies in a way so that the composite OFDM symbol will have a lower PAPR. [21]

4.1.1.3.5 Table 3 showing OFDM advantages and disadvantages:

Advantages	***Disadvantages***
• Immunity to delay spread and multi-path & resistance to frequency selective fading. • Efficient implementation using FFT. • Simple Equalization. • High bandwidth efficiency. • Low sensitivity to time sync errors.	• High sensitivity to frequency sync errors. • High PAPR.

4.1.1.4 OFDM and WiMAX

4.1.1.4.1 Wireless MAN-OFDM PHY

The Wireless MAN-OFDM PHY is based on OFDM modulation and designed for NLOS operation in the frequency bands below 11 GHz.

4.1.1.4.1.1 OFDM Symbol Description in Time Domain

Inverse-Fourier-transforming creates the OFDM waveform; this time duration is referred to as the useful symbol time Tb. A copy of the last Tg of the useful symbol period, termed CP, is used to collect multi-path, while maintaining the orthogonality of the tones.

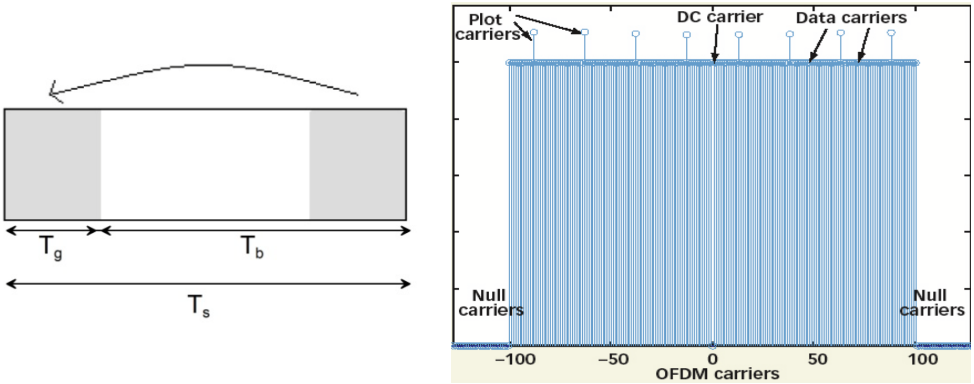

Figure 41: Time and Frequency domain OFDM symbol in WiMAX.

4.1.1.4.1.2 OFDM Symbol Description in Frequency Domain

The frequency domain description includes the basic structure of an OFDM symbol. An OFDM symbol (shown in the previous figure) is made up from sub-carriers, the number of which determines the FFT size used. There are three sub-carrier types:

- Data sub-carriers: For data transmission.
- Pilot sub-carriers: For various estimation purposes.
- Null sub-carriers: No transmission at all, for guard bands, non-active sub-carriers and the DC sub-carrier.

4.1.2 Modulation in Fixed WiMAX PHY

There are 4 types of digital modulations used in fixed WiMAX PHY, the type of chosen modulation type depends mainly on type of data being sent , and channel effect on the signal, that can be expressed as SNR.

4.1.2.1 BINARY PSK

Binary data are represented by two signals with different phases in BPSK. Typically these two phases are 0 and 1. The signals are:

$$\mathbf{s_1(t) = A\cos(2\pi f_c t), \qquad 0 \le t \le T. \qquad \text{for } 1}$$
$$\mathbf{s_2(t) = -A\cos(2\pi f_c t), \qquad 0 \le t \le T. \qquad \text{for } 0}$$

These signals are called antipodal. The reason that they are chosen is that they have a correlation coefficient of -1, which leads to the minimum error probability as it will be seen shortly. These two signals have the same frequency and energy. [25]

All PSK signals can be graphically represented by a signal constellation in a two dimensional coordinate system with

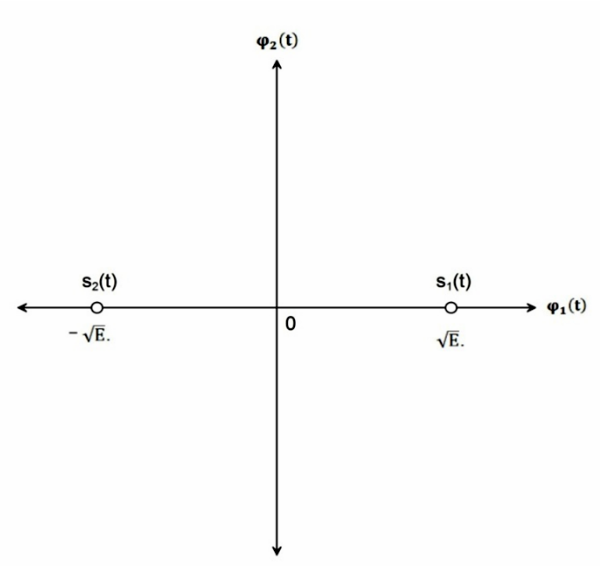

Figure 42: BPSK Signal Constellation.

$$\boldsymbol{\varphi_1(t) = \sqrt{\frac{2}{T}}\cos(2\pi f_c t),} \qquad \mathbf{0 \leq t \leq T.}$$

And

$$\boldsymbol{\varphi_2(t) = -\sqrt{\frac{2}{T}}\sin(2\pi f_c t),} \qquad \mathbf{0 \leq t \leq T.}$$

as its horizontal and vertical axis, respectively. Note that on purpose a minus sign is added to φz (t) so that PSK signal expressions will be a sum instead of a difference. Many other signals, especially QAM signals, can also be represented in the same way. Therefore the signal constellation is introduced to BPSK here as shown in figure where $s_1(t)$ and $s_2(t)$ are represented by two points on the horizontal axis, respectively, where

$$\mathbf{E = \frac{A^2 T}{2}}$$

The waveform of a BPSK signal generated by the modulator for a data stream {10110} is shown in figure. The waveform has a constant envelope like FSK. Its frequency is constant too. In general the phase is not continuous at bit boundaries. [25]

If the f = m×Rb = m/T, where m is an integer and Rb is the data bit rate, and the bit timing is synchronous with the carrier, then the initial phase at a bit boundary is either 0 or 180 corresponding to data bit 1 or 0.

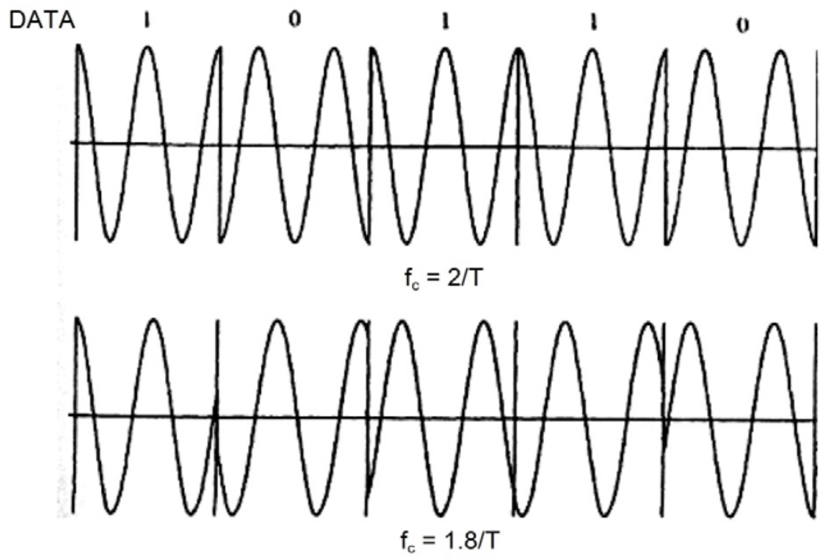

Figure 43: BPSK Waveforms.

However, if the f, is not an integer multiple of Rb, the initial phase at a bit boundary is neither 0 nor n (as in figure 43 where f_c = 1.8/T). In other words, the modulated signals are not the ones given in ordinary BPSK constellation.

The next discussion will demodulate that condition f = m×Rb is necessary to ensure minimum bit error probability. However, if f_c >> Rb, this condition can be relaxed and the resultant BER performance degradation is negligible. [25]

The bit error probability can be derived from the formula for general binary signals

$$\mathbf{P_b} = \sqrt{\frac{\mathbf{E_1 + E_2 - 2\rho_{12}\sqrt{E_2 E_2}}}{\mathbf{2N_o}}}$$

Where for BPSK ρ_{12} = -1 and $E_1 = E_2 = E_b$…..Thus

$$\mathbf{P_b} = \sqrt{\frac{\mathbf{2E_b}}{\mathbf{N_o}}}, (\textbf{coherent BPSK})$$

The curves of coherent and non coherent BFSK are also shown in the figure. Recall the expression for coherent BFSK is

$$\mathbf{P_b} = \mathbf{Q}\sqrt{\frac{\mathbf{E_b}}{\mathbf{N_o}}}, (\textbf{coherent BPSK})$$

which is 3dB inferior to coherent BPSK. However, coherent BPSK requires that the reference signal at the receiver to be synchronized in phase and frequency with the received signal. Noncoherent detection of BPSK is also possible. It is realized in the form of differential BPSK which will be studied in the next section. [25]

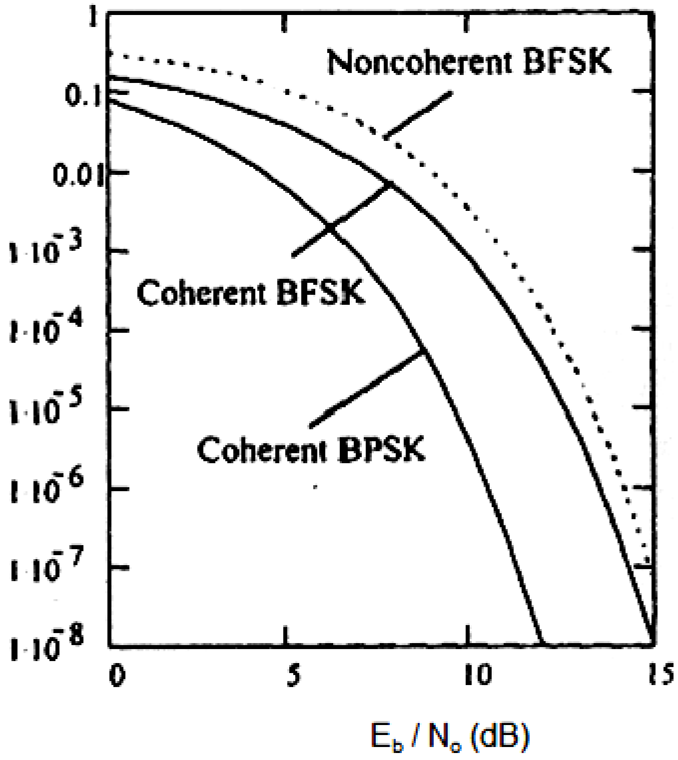

Figure 44: BPSK SNR.

4.1.2.2 Quadrature PSK

Among all MPSK schemes, QPSK is the most often used scheme since it does not suffer from BER degradation while the bandwidth efficiency is increased. Other MPSK schemes increase bandwidth efficiency at the expenses of BER performance. Since QPSK is a special case of MPSK, its signals are defined as

$$\mathbf{s_i(t) = A\cos(2\pi f_c t+\theta_i), \qquad 0 \le t \le T. \qquad i = 1,2,3,4}$$

Where

$$\boldsymbol{\theta_i = \frac{(2i-1)\pi}{4}}$$

The initial signal phases are $\frac{\pi}{4}, \frac{3\pi}{4}, \frac{5\pi}{4}, \frac{7\pi}{4}$.

The carrier frequency is chosen as integer multiple of the symbol rate, therefore in any symbol interval [kT: (k+l)T], the signal initial phase is also one of the four phases. [25]

The above expression can be rewritten as

$$\mathbf{s_i(t) = A\cos\theta_i\cos2\pi f_c t - A\cos\theta_i\cos2\pi f_c t}$$
$$\mathbf{= s_{i1}\,\varphi_1(t) + s_{i2}\,\varphi_2(t)}$$

Where

$$\boldsymbol{\varphi_1(t) = \sqrt{\frac{2}{T}}\cos(2\pi f_c t), \qquad 0 \le t \le T.}$$

$$\boldsymbol{\varphi_2(t) = -\sqrt{\frac{2}{T}}\sin(2\pi f_c t), \qquad 0 \le t \le T.}$$

And

$$\mathbf{s_{i1} = \sqrt{E}\cos\theta_i}$$

$$\mathbf{s_{i2} = \sqrt{E}\sin\theta_i}$$

And

$$\boldsymbol{\theta_i = \tan^{-1}\frac{s_{i1}}{s_{i2}}}$$

Where

$$\mathbf{E = A^2 T/2}$$

Observe that this signal is a linear combination of two orthonormal basis functions $\varphi_1(t)$ and $\varphi_2(t)$. On a coordinate system 0f $\varphi_1(t)$ and $\varphi_2(t)$, four points can be represented by these four signals or vectors:

$$\mathbf{s_i} = \begin{bmatrix} \mathbf{s_{i1}} \\ \mathbf{s_{i2}} \end{bmatrix}, \qquad \mathbf{i = 1,2,3,4}$$

The angle of vector s_i with respect to the horizontal axis is the signal initial phase θ_i. The length of the vectors is $\sqrt{E}$. [25]

In a QPSK system, data bits are divided into groups of two bits, called dibits. There are four possible dibits, 00, 01, 10, and 11. Each of the four QPSK signals is used to represent one of them. The mapping of the digital bits to the signals could be arbitrary as long as the mapping is one to one.

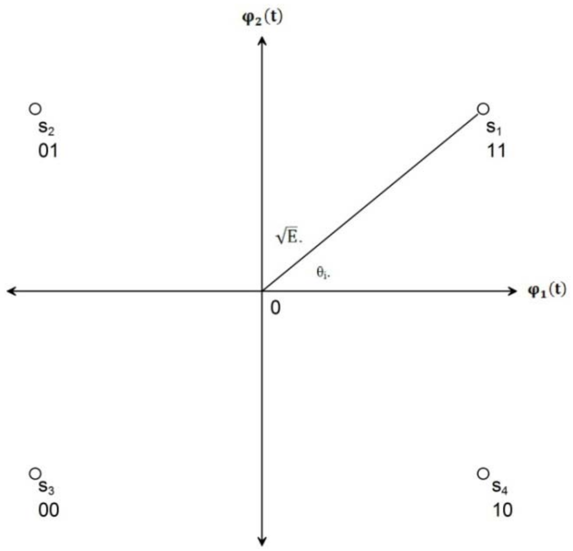

Figure 45: QPSK Signal Constellation.

Table 4 showing the QPSK bit mapping:

Dibit	Phase θ_i	s_{i1}	s_{i2}
11	$\pi/4$	$+\sqrt{E/2}$	$+\sqrt{E/2}$
01	$3\pi/4$	$-\sqrt{E/2}$	$+\sqrt{E/2}$
00	$-3\pi/4$	$-\sqrt{E/2}$	$-\sqrt{E/2}$
10	$-\pi/4$	$+\sqrt{E/2}$	$-\sqrt{E/2}$

Where the mapping for logic 1 is $+\sqrt{E/2}$ and logic 0 is $-\sqrt{E/2}$. Also mapping odd-numbered bits to s_{i1} and even numbered bits to s_{i2}.

The bit error probability for coherent QPSK can be represented by

$$\mathbf{P_b = Q\sqrt{\frac{2E_b}{N_o}}}$$

And as the symbol is composed of 2 bits, symbol error probability can be yet calculated by

$$\mathbf{P_s = 2Q\left(\sqrt{\frac{E_b}{N_o}}\right) - \left(Q\sqrt{\frac{E_b}{N_o}}\right)^2}$$

4.1.2.3 Quadrature Amplitude Modulation (QAM)

In MAM schemes, signals have the same phase but different amplitudes. In MPSK schemes, signals have the same amplitude but different phases. Naturally, the next step of development is to consider using both amplitude and phase modulations in a scheme (QAM). That is

$$\mathbf{s_i(t) = A\cos(2\pi f_c t+\theta_i),\qquad\qquad i = 1,2,\ldots M}$$

where A_i is the amplitude and θ_i is the phase of the ith signal in the M-ary signal set. Pulse shaping is usually used to improve the spectrum and for ISI control purpose in QAM. With pulse shaping, the QAM signal is

$$\mathbf{s_i(t) = A_i\, p(t)\cos(2\pi f_c t+\theta_i),\qquad\qquad i = 1,2,\ldots M}$$

where p(t) is a smooth pulse defined on [0, T]

$$\mathbf{s_i(t) = A_{i1}\, p(t)\cos(2\pi f_c t) - A_{i2}\, p(t)\sin(2\pi f_c t)\qquad\qquad i = 1,2,\ldots M}$$

Where $\mathbf{A_{i1} = A_i\cos\theta_i}$ and $\mathbf{A_{i2} = A_i\sin\theta_i}$

MPSK, QAM signal can be expressed as a linear combination of two similar to orthonormal functions

$$\boldsymbol{\varphi_1(t) = \sqrt{\frac{2}{E_b}}\, p(t)\cos 2\pi f_c t}$$

$$\boldsymbol{\varphi_2(t) = -\sqrt{\frac{2}{E_b}}\, p(t)\sin 2\pi f_c t}$$

And

$$\mathbf{s_{i1} = \sqrt{\frac{E_p}{2}}\, A_i\cos\theta_i}$$

$$\mathbf{s_{i2}} = \sqrt{\frac{\mathbf{E_p}}{\mathbf{2}}\ \mathbf{A_i}\ \mathbf{sin\theta_i}}$$

Where E_p is the energy in p(t) on [0,T], and $\sqrt{\frac{2}{E_p}}$ is to normalize the basis functions $\varphi_1(t)$ and $\varphi_2(t)$. [25]

There are three types of QAM constellations. The focus will mainly be on square type constellations used in WiMAX Fixed PHY.

$$\mathbf{s_i(t) = I_i\sqrt{\frac{E_o}{E_b}}p(t)\cos 2\pi f_c t - Q_i\sqrt{\frac{E_o}{E_b}}p(t)\sin 2\pi f_c t}$$

(I_i,Q_i) are pairs 0f independent integers which determine the location of signal

constellation , the maximum values for are (I_i,Q_i) are $(\pm1,\pm1)$

(I_i,Q_i) can be described as

$$\begin{bmatrix} \mathbf{(-L+1, L-1)} & \mathbf{(-L+3, L-1)} & \dots & \mathbf{(L-1, L-1)} \\ \mathbf{(-L+1, L-3)} & \mathbf{(-L+3, L-3)} & \dots & \mathbf{(L-1, L-3)} \\ \vdots & \vdots & \vdots & \vdots \\ \mathbf{(-L+1, -L+1)} & \mathbf{(-L+3, -L+1)} & \dots & \mathbf{(L+1, -L+1)} \end{bmatrix}$$

Where $L = \sqrt{M}, \quad M = 4^n, \quad n = 1,2,3, \dots..$

The bit error probability in QAM

$$\mathbf{E_{av}} = \mathbf{2}\left(\mathbf{1} - \frac{\mathbf{1}}{\sqrt{\mathbf{M}}}\right)\ \mathbf{erfc}\sqrt{\frac{\left(\mathbf{2}\left[\frac{\mathbf{2E_o}}{\mathbf{L}}\sum_{\mathbf{i=1}}^{\mathbf{L/2}}(\mathbf{2i-1})^{\mathbf{2}}\right]\right)}{\mathbf{N_o}}}$$

4.1.3 Channel Encoding

4.1.3.1 The concepts of channel coding

Channel coding is a common strategy to make digital transmission more reliable or equivalently to achieve the same required reliability for a given data rate at a lower power level at the receiver. This gain in power efficiency is called coding gain. [22] [23] [24]

The bit error rate in a Rayleigh fading channel decreses as $P_b \sim (E_b / N_o)^{-1}$ which would require an unacceptable high transmit power to achieve a sufficiently low bit error rate. It is clear that one possible solution is diversity and in the following section the channel coding can achieve the same gain as diversity with less redundancy.

This section gives a brief but self-contained overview over the channel coding techniques that are commonly applied in OFDM.

For simplicity, let's often speak of data bits as b_i and channel encoder output bits as c_i. some basic concepts and definitions are

- The output of the encoder is called a code word. The set of all possible code words is the code matrix.
- If the channel encoder always takes a data block b = $\{b_1....b_k\}$ of a certain length K and encodes it to a code word c = $\{c_1....c_k\}$ of a certain length N.
- If the encoder maps b = $\{b_1....b_k\}$ to the code word c = $\{c_1....c_k\}$, the ratio R_c = K/N is called the code rate.
- If two code words differ in d positions, then d is called the Hamming distance between the two code words. The minimum Hamming distance between any two code words is called the Hamming distance of the code and is usually denoted by d_H. For an (N,K) block code, the triple (N, K, d_H) to characterize the code.
- If the vector sum of any two code words is always a code word, the code is called linear.
- The Hamming distance of a linear code equals the minimum number of nonzero elements in a code word, which is called the weight of the code. A code can correct up to t errors if ($2t+1 \leq d_H$) holds.
- An encoder is called systematic if the data symbols are a subset of the code word. Obviously, it is convenient but not necessary that these systematic (i.e. data) bits (or symbols) are positioned at the beginning of the code word. In that case the encoder maps b = $\{b_1....b_k\}$ to the code word c = $\{c_1....c_k\}$ and b_i = c_i for1, 2, ..., K. The nonsystematic symbols of the code word are called parity check (PC) symbols.
- The channel decoder outputs c_i are the inputs of the modulator. Depending on these data, the modulator transmits one out of a set of possible signals s(t). For binary codes, there are 2^k code words and thus there are 2^k possible signals s(t).
- There is a one-to-one correspondence between the code word vectors c = $\{c_1....c_k\}$ and the transmit symbol vector s = $(s_i……s_L)^T$. Because each code

word c is uniquely determined by the corresponding data word b, the mapping b ⇒ c is uniquely defined and one may regard the channel encoder and the modulator as one single device that corresponds to the mapping b ⇒ c. This concept is called coded modulation.

- The modulated signal is corrupted by the noisy transmission channel. Thus, some of the demodulator output bits $\widehat{c_i}$ will be erroneously decided from the received signal r(t). The channel decoder then uses the redundancy of the code to correct the errors and delivers correct data $\widehat{b_i}$ identical to the source data b_i.

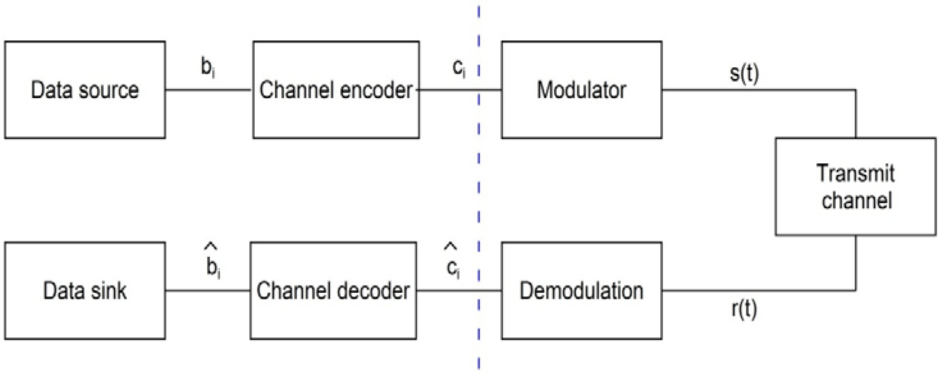

Figure 46: Modulation techniques.

4.1.3.2 Error probabilities

In this subsection, error rates for binary codes will be discussed with antipodal signaling, that is, BPSK. The same formulas apply for QPSK, which can be separated into antipodal signaling for both the in-phase and quadrature component. [22] [23] [24]

4.1.3.2.1 Error probabilities for the receiver and the AWGN channel

For the receiver, the probability that the receiver erroneously decides for the transmit vector s instead of the transmitted vector $\hat{s}$ is given by equation

$$P\left(s \to \hat{s}\right) = \frac{1}{2}\, erfc\left(\sqrt{\frac{1}{4N_o}\|s - \hat{s}\|^2}\right)$$

For BPSK transmission, s = $(s_i \ldots\ldots s_L)^T$ with s = $\sqrt{\pm E_S}$, where E_S is the symbol energy and assuming a binary code with rate R_c and Hamming distance d_H. Assume that s corresponds to a code word c and the receiver decides for a code word $\hat{c}$ corresponding to the signal vector $\hat{s}$, and the code words c and $\hat{c}$ have the Hamming distance d then;

$$\|s - \hat{s}\|^2 = 4dE_S$$

For each transmitted symbol, only R_c useful bits are transmitted, thus $E_s = R_c E_b$ and the error event probability P_b for an erroneous decision corresponding to a Hamming distance d is given by

$$P_b = \frac{1}{2}\, erfc\left(\sqrt{dR_c\frac{E_b}{N_o}}\right)$$

4.1.3.3 Linear binary block codes

In this subsection, some facts will be presented about linear binary block codes and some examples will be provided. Let C be a linear binary (N, K) block code and M = 2^K code words of length N as binary columns and join them together to a matrix

For M= 8, for example, the data matrix is given by

$$\mathbf{B} = \begin{pmatrix} \mathbf{0} & \mathbf{1} & \mathbf{0} & \mathbf{1} & \mathbf{0} & \mathbf{1} & \mathbf{0} & \mathbf{1} \\ \mathbf{0} & \mathbf{0} & \mathbf{1} & \mathbf{1} & \mathbf{0} & \mathbf{0} & \mathbf{1} & \mathbf{1} \\ \mathbf{0} & \mathbf{0} & \mathbf{0} & \mathbf{0} & \mathbf{1} & \mathbf{1} & \mathbf{1} & \mathbf{1} \end{pmatrix}$$

Between the vector spaces B and C. G is called the generator matrix. Using matrix notation, It may be written as P = GB. from linear algebra it is concluded that G is given by an N × K matrix.

Its (N-K)N generator matrix H is related to G by:

$$H^{T}G = 0.$$

Let the generator matrix for a code be

$$\mathbf{G} = \begin{bmatrix} \mathbf{1} & \mathbf{0} & \mathbf{0} & \mathbf{0}| & \mathbf{1} & \mathbf{1} & \mathbf{1} \\ \mathbf{0} & \mathbf{1} & \mathbf{0} & \mathbf{0}| & \mathbf{0} & \mathbf{1} & \mathbf{1} \\ \mathbf{0} & \mathbf{0} & \mathbf{0} & \mathbf{0}| & \mathbf{1} & \mathbf{0} & \mathbf{1} \\ \mathbf{0} & \mathbf{0} & \mathbf{1} & \mathbf{1}| & \mathbf{1} & \mathbf{1} & \mathbf{0} \end{bmatrix}$$

For information sequence [0, 1, 1, 0], the following transmitted codeword of length 7 appeared.

$$\mathbf{c} = [\mathbf{0} \quad \mathbf{1} \quad \mathbf{1} \quad \mathbf{0}] \begin{bmatrix} \mathbf{1} & \mathbf{0} & \mathbf{0} & \mathbf{0} & \mathbf{1} & \mathbf{1} & \mathbf{1} \\ \mathbf{0} & \mathbf{1} & \mathbf{0} & \mathbf{0} & \mathbf{0} & \mathbf{1} & \mathbf{1} \\ \mathbf{0} & \mathbf{0} & \mathbf{1} & \mathbf{0} & \mathbf{1} & \mathbf{0} & \mathbf{1} \\ \mathbf{0} & \mathbf{0} & \mathbf{0} & \mathbf{1} & \mathbf{1} & \mathbf{1} & \mathbf{0} \end{bmatrix} = [\mathbf{0} \quad \mathbf{1} \quad \mathbf{1} \quad \mathbf{0} \quad \mathbf{1} \quad \mathbf{1} \quad \mathbf{0}]$$

Let's take two examples for the linearly block codes.

4.1.3.3.1 Repetition (RP) codes

A very naive idea for coding is a simple repetition of the bits. An RP (N, 1, N) code has d_H = N & R_c = 1/N and thus the coding gain is zero. Obviously, RP coding is just another word for diversity, and, in a fading channel, it has a diversity gain if the fading amplitudes of the received coded symbols are sufficiently independent. The generator matrix of this code is the all-one column vector of length N. [22] [23] [24]

4.1.3.3.2 Hamming codes

Hamming codes are the dual codes of simplex codes. Hamming code can correct one error. Hamming codes are simple and weak codes, but they are popular to explain the concepts of algebraic coding.

4.1.3.4 Concatenated coding:

If an application requires very low bit error rates, concatenated coding is often the most efficient method to reach this goal. In such a setup, two codes are combined to a stronger overall concatenated code. At the transmitter, the source data will first be encoded by the outer code. 'The code words of this code will then serve as the input data for the inner encoder, the order of the symbols inside the stream of code words may be changed by a device that is called interleaver. The code words of the inner code are transmitted over the channel and then decoded by the inner decoder. [22] [23] [24]

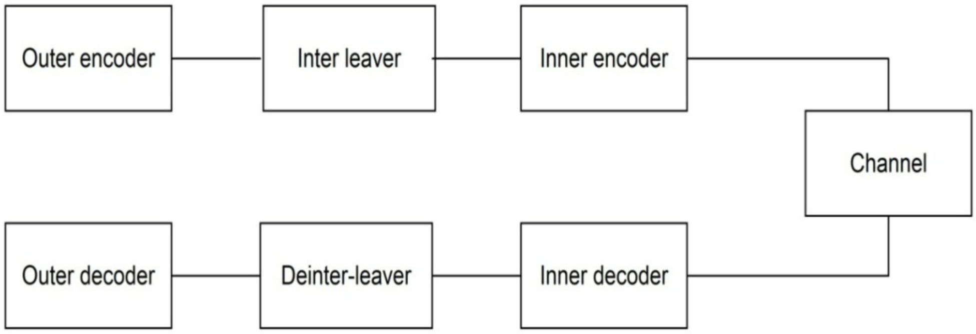

Figure 47: Modulation using concatenated coding.

The inner decoder has to be matched to the channel. At its output, the error rate will be low, but the errors are not uniformly distributed. The inner decoder will typically produce error bursts, For block codes, a burst error corresponds to an erroneously decided code word. For convolutional codes, an error burst corresponds to the sequence of states in the trellis. The deinterleaver inverts the interleaver. The interleaving scheme breaks up the error bursts and has to be matched to the error correction capabilities of the outer code, that is, to its Hamming distance and code word length. [22] [23] [24]

For a properly designed concatenated coding scheme, the output of the outer decoder will be nearly error free. Thus, one can visualize the inner decoder as a device that is suited for coarsely cleaning up the errors that are produced by a severely corrupted transmission channel. The outer decoder will then clean up the residual errors.

If QAM or PSK modulation are regarded as (nonbiliary) coding schemes, QAM or PSK with additional (convolutional) coding is a concatenated coding scheme. The outer QAM or PSK decoder will typically pass soft bits to the outer (convoiutional) decoder. The optimal soft bits are the LLRs (log-likelihood ratios) calculated by the receiver. Probably the most popular concatenated coding scheme is an inner convolutional code with an outer Reed-Solomon (RS) code.

Both types of codes are discussed in the following sections. Convolutional codes with soft decision decoding are well suited for channels that are severely corrupted by a high noise level and/or by multipath fading. However, because of their typically quite low Hamming distance, the BER curves show a poor decay. Thus, a high SNR is needed if very low BERs are required. The convolutional decoder produces bursts of erroneously decided bits.

RS codes can be designed as strong codes with high Hamming distances. They are based on byte arithmetics rather than bit arithmetics, that is, they correct byte errors rather than bit errors. The decoder works with hard decision input bytes. Thus a convolutional decoder together with a byte interleaving produces the favorite input for the RS decoder. [22] [23] [24]

Deep space communication was one of the first applications of such a scheme with a convolutional code and an RS code. Because of the power limitation, the physical channel is very noisy AWGN channels that make convolutional codes the best choice. If, for example, data compressed pictures have to be transmitted, the bit error rate has to be very low. Therefore an outer RS will give a considerable gain for this application, the outer codes give an additional coding gain of 2.5 dB at BER = 10^{-6} or a gain in data rate of 78%.

4.1.3.5 Convolutional Codes

4.1.3.5.1 General structure and encoder

In contrast to block codes, convolutional codes do not have a defined block structure. A continuously flowing data stream will be encoded into a continuously flowing code word.

Convolutional encoders are linear and time-invariant systems given by the convolution of a binary data stream with generator sequences. They can be implemented by shift registers and following is the output for the encoder

$$c_v = b(D)g_v(D)$$

$$\begin{bmatrix} c_1(D) \\ \vdots \\ c_n(D) \end{bmatrix} = b(D) \begin{bmatrix} g_1(D) \\ \vdots \\ g_n(D) \end{bmatrix}$$

Convolutional codes are linear codes. Thus, the Hamming distance of the code is the minimum weight. This is called the free distance and will be denoted by $d_{free.}$

4.1.3.5.2 Trellis diagrams

For any time instant i, one can characterize the encoding step by the actual state s = (s_1,…., s_m) that is, the content of the shift register.

Given a defined initial state of the shift register (usually the all zero state), each code word is characterized by sequence of certain transitions which is known as “a path in the trellis”. [22] [23] [24]

4.1.3.5.3 Punctured convolutional codes

Up to now, convolutional codes of R_C = 1/n is only considered. There are two possibilities to obtain R_c = k/n. The classical one is to use k parallel shift registers and combine their outputs. This, however, makes the implementation more complicated. A simpler and more flexible method called puncturing is usually preferred in practical communication systems.

The encoder produces two parallel encoded data streams $\{c_1, i\}_{i=0}^{\infty}$ and $\{c_2, i\}_{i=0}^{\infty}$. The first data stream will be left unchanged. From the other data stream every second bit will be discarded, that is, only the bits with even time index i will be multiplexed to the serial code word and then transmitted. Instead of the original code word, the punctured code will be transmitted.

At the receiver, the puncturing positions must be known. A soft decision receiver has metric values as inputs that correspond to the encoded bits. The absolute value of the metric values is an indicator for the reliability of the bit. Punctured bits can be regarded as bits with reliability zero. Thus, the receiver has to add dummy receive bits at the punctured positions of the code word and assign them the metric values zero. [22] [23] [24]

4.1.3.5.4 Performance of convolutional codes

From the previous sections it is clear that as the coding rate decreases it gives better performance. Also after comparing different performance for soft and hard bit decision decoding, it is clear that soft bit decision gives better decoding.

4.1.3.6 Reed-Solomon Codes

Reed-Solomon (RS) codes may be regarded as the most important block codes because of their extremely high relevance for many practical applications. They are nonbiliary cyclic codes with cede symbols from a Galois field. They were discovered in 1960 by I. Reed and G. Solomon.

Reed-Solomon codes are based on byte arithmetics rather than on bit arithmetics. Thus, RS codes correct byte errors instead of bit errors. As a consequence, RS codes are favorable for channels with bursts of bit errors as long as these bursts do not affect too many subsequent bytes, lids can be avoided by a proper interleaving scheme.

As another example, for a concatenated coding scheme with an inner convolutional code, the decoder produces burst errors. An inner convolutional code concatenated with an outer RS code is therefore a favorable setup. It is used in deep space communications.

For RS(N,K,D) code, K data bytes are encoded to a code word of N bytes. One of the most important features of RS codes is that the minimum distance of an (n, k) RS code is n-k+1. Codes of this kind are called "maximum-distance-separable codes. For odd values of D, the code can correct up to t byte errors with $D = 2t + 1$. For even values of D the code can correct up to t byte errors with $D = 2t + 2$, RS codes are linear codes.

If it is known that some received bytes are very unreliable (e.g. from an inner decoder that provides such reliability information), the decoder can make use of this fact in the decoding procedure. These bytes are called erasures.

Solomon algebraic decoding procedures can correct errors and erasures. An erasure occurs when the position of an erred symbol is known.

A decoder can correct t errors or up to 2t erasures. Erasure information can often be supplied by the demodulation in a digital communication system, i.e. the demodulator "flags" received that are likely to contain errors.

The advantage of using Reed-Solomun codes is that probability of an error remaining in the decoded data is (usually) much lower than the probability of an error if Reed-Solomon is not used. This is often described as coding gain.

Decoding of block codes is pretty easy. First compute something called a Syndrome of an incoming codeword by multiplying the incoming vector by the transposed parity matrix. [22] [23] [24]

4.1.3.7 WiMAX concatenated coding

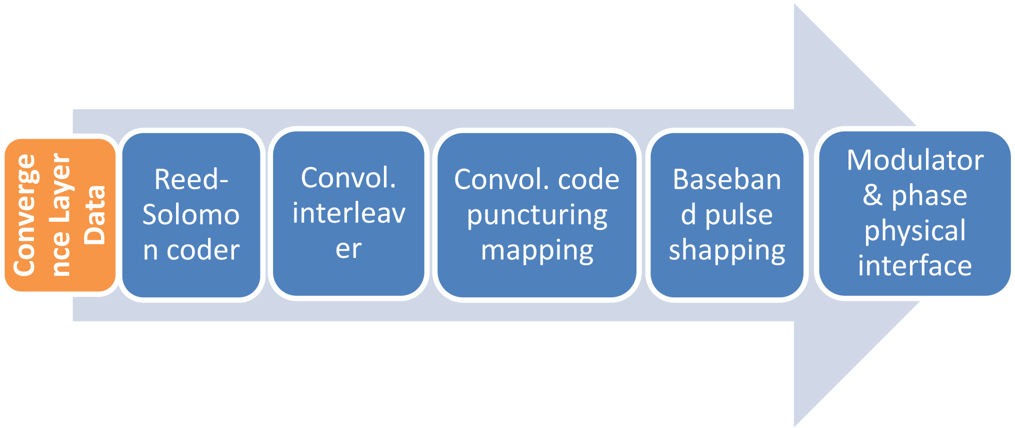

Figure 48: Concatenated coding in WiMAX. [26]

4.1.4 Synchronization

4.1.4.1 Introduction

OFDM, like any other digital communication system, requires synchronization. However, OFDM as a multicarrier system has a different structure than a single-carrier system and so have different requirements and different resources. [27]

For example in OFDM, one can tolerate larger errors in estimating the start of a symbol than in a single-carrier system. This is due to OFDM's longer symbol period and its cyclic prefix. On the other hand, frequency synchronization in OFDM must be tighter than that in single-carrier systems, due to the narrowness of the OFDM subcarriers. [27]

In terms of resources, OFDM has a structure that is not available in single-carrier systems that is useful for synchronization. For example, most OFDM systems have a cyclic prefix that can be used for synchronization. The cyclic prefix can act as pilot data. Often, an OFDM symbol itself is used as pilot data. In this case, the structure of the OFDM symbol can be exploited for time and frequency offset estimation. [27]

The choice of pilots versus no-pilots depends on many parameters: the operating SNR, the size of the cyclic prefix, coherent versus differential modulation. Whether to insert pilot data or use OFDM symbols as pilot data often depends on how much overhead the system can tolerate. [27]

Similar to other communication systems, carrier synchronization in OFDM is usually carried out in two places, namely, acquisition and tracking. Acquisition obtains an initial rough or coarse estimate of timing and/or frequency parameters. [27]

Tracking is an on-going process where this rough estimate is refined to get a better estimation. Acquisition parameter estimation schemes generally have a wide range, but low accuracy. Tracking algorithms have a narrower range and finer accuracy. [27]

4.1.4.2 Sensitivity to frequency offset:

At start-up, the local oscillator (LO) frequency at the receiver is typically different from the LO frequency at the transmitter.

Due to the carrier frequency difference of the transmitter and receiver, each signal sample at time t contains an unknown phase factor $e^{j2\pi\delta ft}$ Where δf is the unknown carrier frequency offset. [27]

This unknown phase factor must be estimated and compensated for each sample before FFT at the receiver since the LO offset results in a frequency shift of the received signal spectrum. This shift causes a condition called "loss of orthogonality" to occur. [27]

The frequency shift causes the OFDM sub-carriers to no longer be orthogonal. The orthogonality of the sub-carriers is lost because the bins of the FFT will no longer line up with the peaks of the received signals since pulses as in figure 48. The result is a distortion called inter-bin interference or IBI. [27]

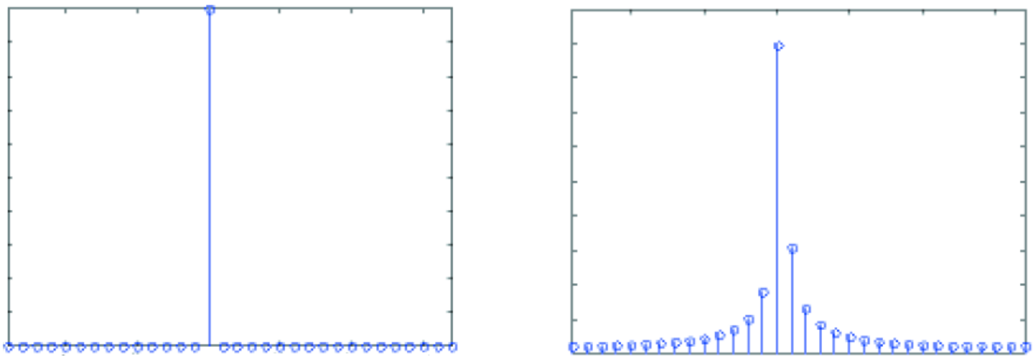

Figure 49: Left plot is for the case of no LO offset and right plot is for the presence of LO offset.

IBI occurs when energy from one bin spills over into adjacent bins and this energy distorts the affected sub-carriers. This interference power is inversely proportional to the frequency spacing. The amount of ICI for sub-carriers in the middle of the OFDM spectrum is approximately twice as large as that for sub-carriers at the band edges, because the sub-carriers in the middle have interfering sub-carriers on both sides, so there is more interference within a certain frequency distance.

The central limit theorem states that the sum of a large number of random processes will result in a signal that has a Gaussian distribution. Because of this property, the IBI will manifest itself as additive Gaussian noise, thus lowering the effective SNR of the system.

4.1.4.3 Sensitivity to timing errors

Another non-ideal effect that can occur in a real-world OFDM system is an FFT window location offset. An N-point FFT at the receiver processes data in blocks of N samples at a time. Ideally, the N samples taken in by the FFT will correspond to the N samples of a single transmitted OFDM symbol.

If a timing offset exists, the result is that the N samples sent to the FFT will not line up exactly with the corresponding OFDM symbol. If the offset is very large, part of the N samples will be from one OFDM symbol, and the rest of samples will be from another OFDM symbol. Such a situation would result in a severe distortion of the received sub-carrier's constellations.

The presence of the cyclic prefix gives enough headroom to enable a small offset to be present without taking samples from more than one OFDM symbol. But the sensitivity to delay spread will increase, so the system can handle less delay spread than the value it was designed for. [27]

The best region for starting the FFT is illustrated in following figure.

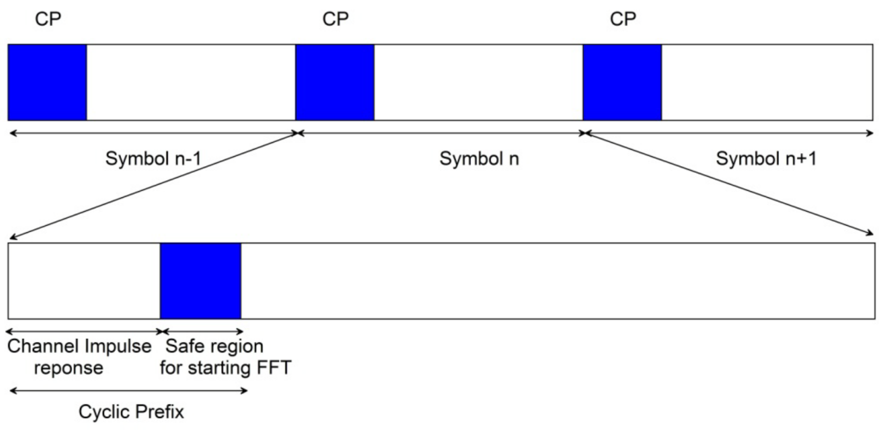

Figure 50: Region of timing synchronization.

As long as FFT window location offset does not go beyond an OFDM symbol boundary, this offset in time is equivalent to a linearly-increasing phase rotation in the frequency-domain. Looking at following figures, they show that the amount of rotation increases linearly as sub-carrier's FFT bin location increases.

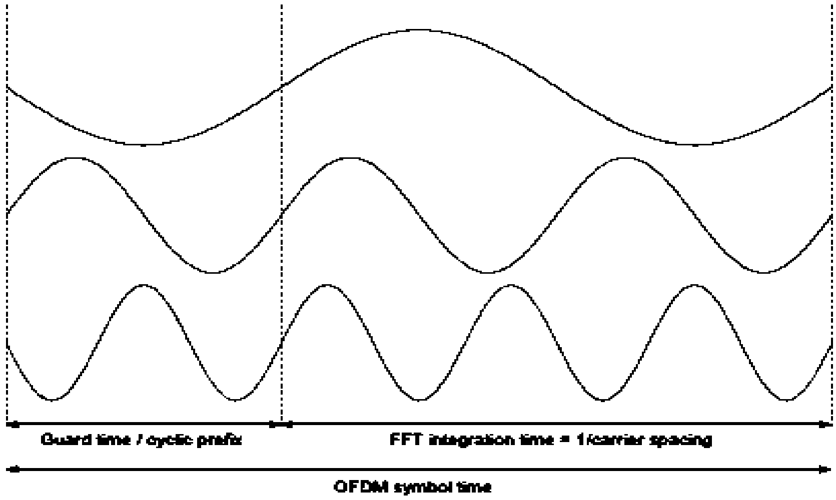

Figure 51: An OFDM signal with three sub-carriers.

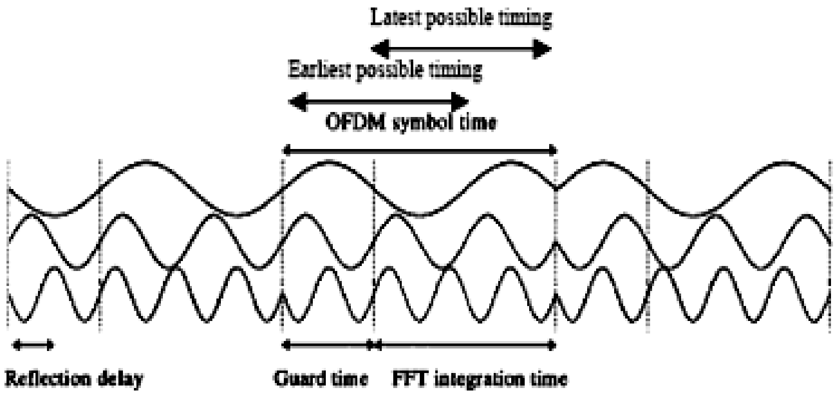

Figure 52: Three successive OFDM signals with three sub-carriers.

The relation between the phase φ_i of sub-carrier i and the timing offset £ is given by $\varphi_i = 2\pi f_i \zeta$ where ζ is the frequency of the ith sub-carrier. [27]

4.1.4.4 Timing and frequency offset estimation

There is many approaches to estimate timing and frequency offset in OFDM systems which operate in the time domain (before the FFT) and use the repeating pattern of the preamble or the cycle prefix, or both, to gain information about the symbol timing and frequency offset.

The timing is determined by noticing that the correlation of the signal with a delayed version of itself will reach a peak when the repeated pattern is located.

Using the repetition in the preamble proves more robust compared to methods that use the cycle prefix when this is short.

The frequency offset can be estimated by, for example, calculating the phase offset between one occurrence of a pattern and the next.

4.1.4.4.1 Frequency offset estimation

The normalized frequency offset can be written as

$$\frac{\delta f}{\Delta f} = k_o + \varepsilon$$

Where δf: is the frequency offset, Δf: is the sub-channel space, k_o: is an integer carrier frequency offset and ε: is a fractional carrier frequency offset, | ε |≤ l/2. [27]

Accordingly, this problem can be divided into two major parts that will be illustrated in the following sections.

- Fine frequency offset estimation of ε as shown in the next figure:

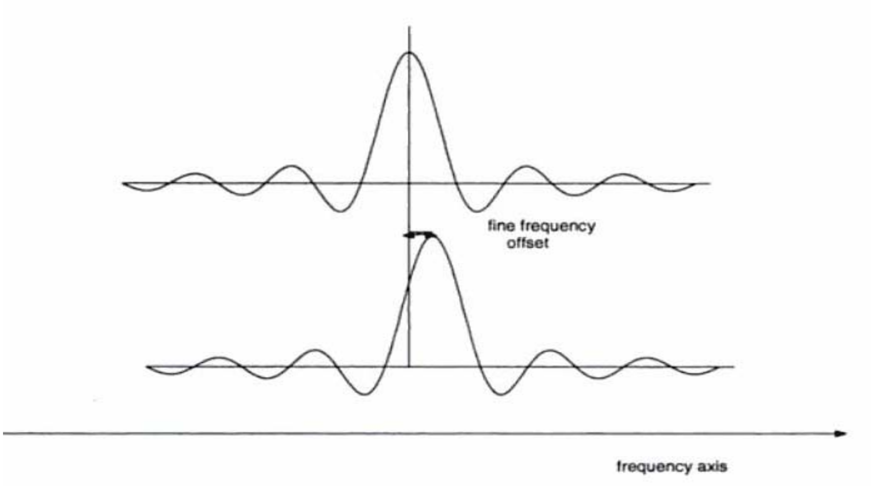

Figure 53: Fine frequency synchronization.

- Coarse frequency offset estimation of k_o as shown in the next figure:

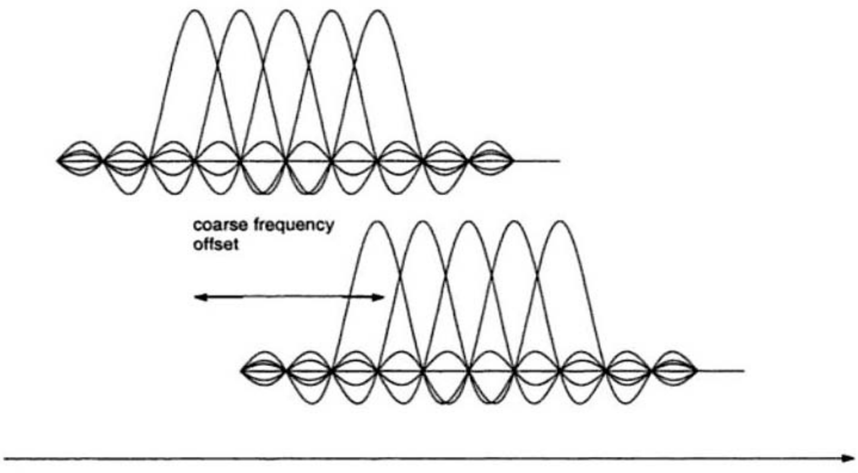

Figure 54: Coarse frequency synchronization.

A scheme based on intersecting repeated short sequences between OFDM symbols has been proposed. This scheme performs both coarse and fine frequency offset estimation. Coarse estimation involves comparing the phases of two symbols close together while fine frequency synchronization relies on comparing symbols further apart.

The requirement that symbols must be far apart for fine frequency synchronization puts some limitations on the system, especially if the channel is with fast fading. Any significant change in the channel phase will affect the fine frequency offset estimation. [27]

4.1.4.4.2 Timing Offset Estimation Sync Using The Cyclic Extension

Because of the cyclic prefix, the first T_G seconds part of the OFDM symbol is identical to the least part. This property can be exploited for both timing and frequency synchronization by using synchronization system like that illustrated in following figure. [27]

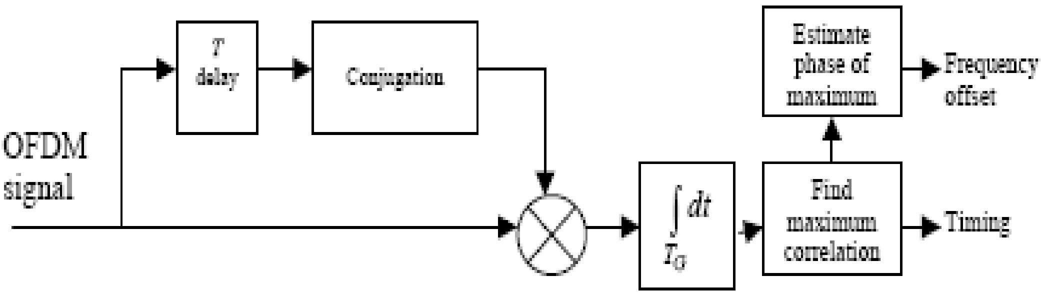

Figure 55: synchronization using the cyclic prefix.

An example of the correlation output is shown in the next figure which is for eight OFDM symbols with 192 sub-carriers and a 20% guard time.

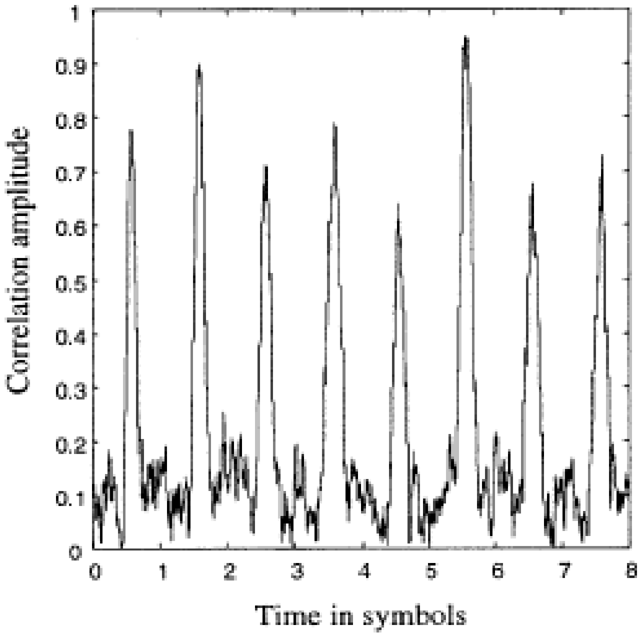

Figure 56: An example of the correlation output.

This figure illustrates a few characteristics of the cyclic extension correlation method. First, the figure clearly show eight peaks for eight different symbols, the sidelobes reflect the correlation between two pieces of the OFDM signal that belong partly or totally to two different OFDM symbols. [27]

The phase of the cyclic extension correlation output is equal to the phase drift between samples that are T seconds apart. Hence, the frequency offset can simply be found as the correlation phase divided by $2\pi T$. This method works up to a maximum absolute frequency offset of half the sub-carrier spacing.

4.1.4.4.3 Timing Offset Estimation Sync Using Special Training Symbols

The synchronization technique based on the cyclic extension is particularly suited to tracking or to blind synchronization when no special training signals are available.

Special OFDM training symbols can be used for which the data content is known to the recevier. In this way, the entire received training signal can be used to achieve synchronization, whereas the cyclic extension method only uses a fraction of each symbol. [27]

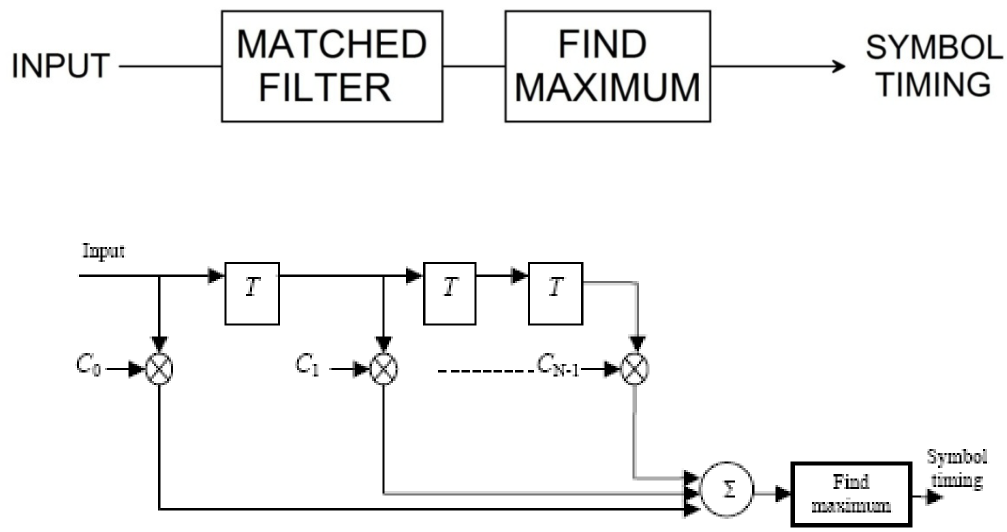

Figure 57: Synchronization using Special Training Symbols.

From the correlation peaks in the matched filter output signal, both symbol timing and frequency offset can be estimated.

The level of undesired correlation sidelobes can be minimized by a proper selection of the training symbols. [27]

4.1.5 Channel Estimation And Equalization

4.1.5.1 Channel Estimation

4.1.5.1.1 Introduction

For OFDM systems with multiple transmit and/or receive antennas for system capacity or performance improvement, channel information is essential to diversity combining, interference suppression, and signal detection. [27]

In summary, the accuracy of channel state information greatly influences the overall system performance. Therefore, in this section, channel parameter estimation in OFDM systems will be presented.

4.1.5.1.2 Pilot-Symbol-Aided Estimation

In pilot-symbol-aided estimation the parameters at the pilot tones are first estimated and then the parameters at the data tones are obtained by using interpolation and filtering approaches. [27]

To be able to interpolate the channel estimates both in time and frequency from the available pilots, the pilot spacing has to fulfill the Nyquist sampling theorem, which states that the sampling interval must be smaller than the inverse of the double-sided bandwidth of the sampled signal.

For the case of OFDM, this means that there exist both maximum sub-carrier spacing and maximum symbol spacing between pilots. By choosing the pilot spacing much smaller than these maximum requirements, good channel estimation can be made with a relatively easy algorithm. [27]

Increasing the pilot density increases the channel estimation performance but on the cost of resources. [27]

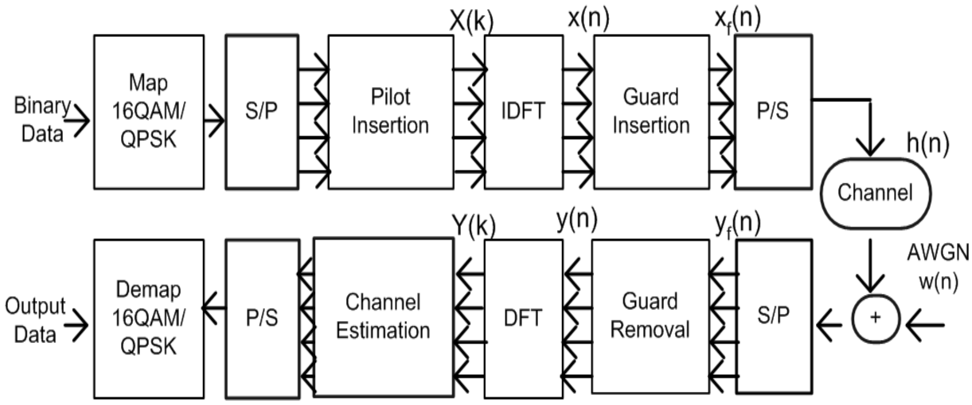

Figure 58: Baseband IFDM system.

4.1.5.2 Equalization

4.1.5.2.1 Introduction

One of the advantages of a multi-carrier system is its robustness against inter-symbol interference. The longer duration of OFDM symbols provides higher immunity against delay spread and ISI. As long as channel dispersion is not longer than the OFDM symbol guard interval, system performance does not degrade due to ISI and use of time domain equalization is not usually mandated. [28]

However, in case of higher data rates and channels with extensive time dispersion an equalizer is unavoidable. However, the structure of the equalizer is different from that of single carrier systems. The purpose of equalization is not complete removal but restriction of inter-symbol interference to a tolerable extent. [28]

Frequency domain equalization, in the absence of inter-channel interference (ICI) is used to compensate for channel complex gain at each sub-carrier frequency.

4.1.5.2.2 Time Domain Equalization (Channel Shortening)

In OFDM system, when the maximum delay is longer than the CP duration which is at most 1/4 symbol duration, the ISI effect is irreducible from the system. If a shortened impulse response filter (SIRF) w(n) with length W is applied before removing CP block in the receiver, which is indicated in Figure 59, the output of the SIRF can be expressed as

$$\mathbf{y(n) = (h(n) * w(n)) * x(n) = h_{eff}(n) * x(n)}$$

Where x(n) is the transmitted data, * denotes the convolution operator, and $h_{eff}(n)$ is the effective channel impulse response after shortening.

On a point of view of adding a FIR to obtain the effective channel, it is generally not possible to shorten the impulse response perfectly. On the contrary, the effective channel is always longer than the physical channel due to convolution. A measurement used to check the shortening performance is called shortening SNR (SSNR), which is defined as the ratio of the energy lies within the desired length D to the energy lies out of D. [28]

In OFDM application, the desired length is often set to be the length of CP. Instead, it is better to have D shorter than the CP.

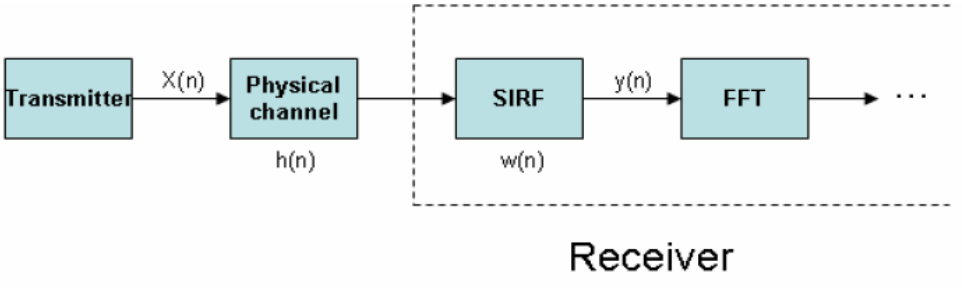

Figure 59: Simplified transceiver diagram with SIRF.

4.1.5.2.3 Frequency Domain equalization

Once it is assured that orthogonality of the sub-carriers is maintained, possibly through the use of the cyclic prefix and time domain equalization as previously discussed, then the final frequency domain equalization of an OFDM signal is an extremely simple process. This is certainly one of the key advantages of OFDM. OFDM systems work by resolving the frequency domain so that the width of the sub-carriers is much narrower than frequency selective fading of the radio channel. [28]

This makes the frequency response over the bandwidth of each sub-carrier effectively flat. Only simple equalization is required for each sub-carrier for data transmission as the flat fading on each sub-carrier only results in an amplitude scaling and a phase rotation. [28]

Frequency domain equalization therefore consists of separate adjustments for each sub-carrier gain and phase, or equivalently of adjusting the individual decision regions.

For the simple case of no noise, the ideal value of the equalizer's response is the inverse of the channel's frequency response. An example is shown in figure below. With such a setting, the frequency-domain equalizer would cancel out the multiplicative effect of the channel. [28]

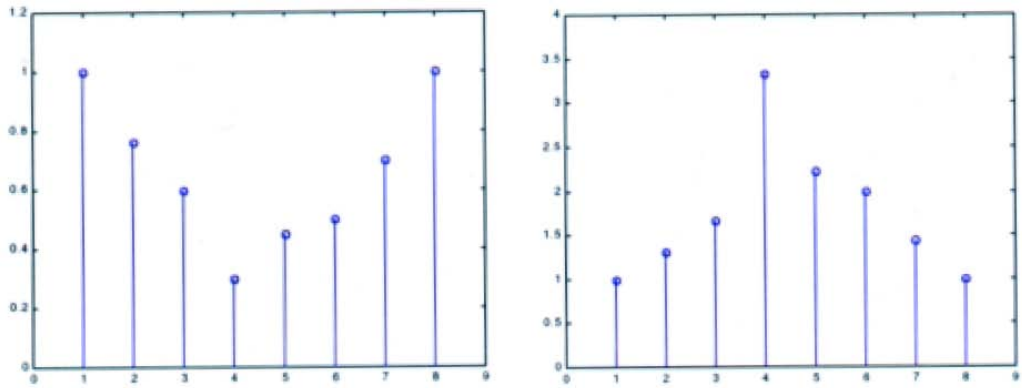

Figure 60: Left plot shows the frequency response of a channel and the right plot shows the corresponding frequency-domain equalizer response.

The FEQ includes a simple one-tap equalizer and an estimator for each sub-carrier. The estimated channel response is then

$$\widehat{H_n} = P_n^{-1} Z_n = \frac{Z_{n,1}}{P_{n,1}}, \frac{Z_{n,2}}{P_{n,2}}, \dots \dots, \frac{Z_{n,k}}{P_{n,k}}$$

Where n is the index of OFDM symbol, k is the index of sub-carrier, H_n is the whole channel response of the nth symbol, Z_n is the received preamble, and P_n is the known preamble. [28]

4.2 WiMAX Block Diagram Model

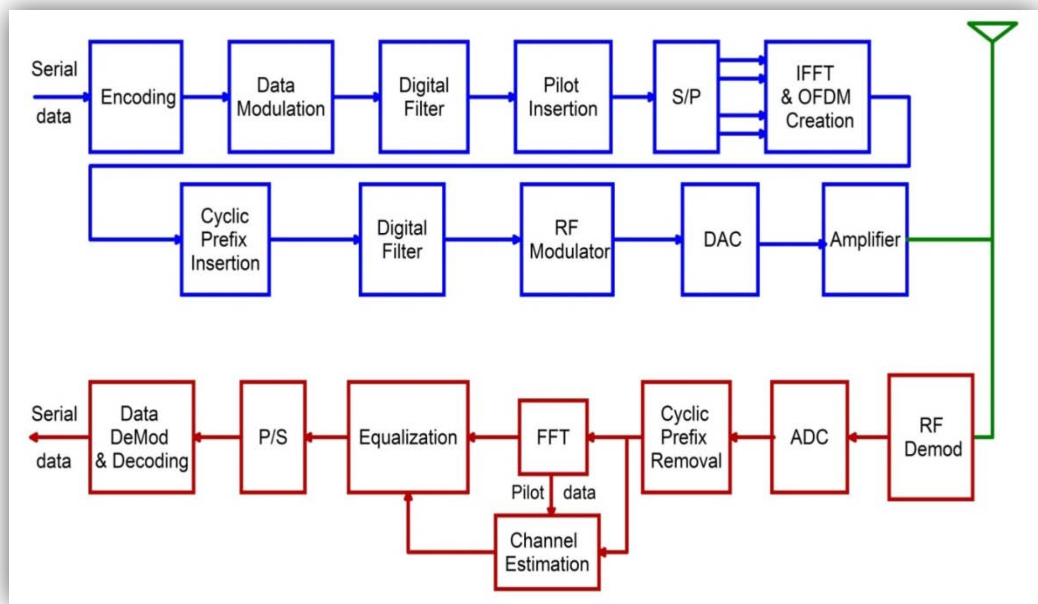

Figure 61: WiMAX block diagram model.

The figure above shows a design block for the suggested WiMAX system including all the important and basic features of the system.

4.2.1 Encoding Blocks

Reed-Solomon Codes will be used as it is based on byte arithmetics not bit arithmetics. It is also corrects Erasures (Marked unreliable bytes). RS decoder can correct up to t errors or 2t erasures, where t is linear with the minimum distance in the RS code. Advantages are that RS minimizes the no. of errors better than any other codes. [29]

4.2.2 Data Modulation Blocks

There are many types of digital modulations used in WiMAX. The type of chosen for modulation depends on type of data being sent and channel effect on the signal. QPSK is the most often used scheme as it uses four points on the constellation diagram and with these four phases, QPSK can encode two bits per symbol to minimize the Bit Error Ratio. The initial signal phases are 45^o, 135^o, -45^o & -135^o. [29]

4.2.3 Channel Estimation and Pilot-Symbol-Aided Insertion Blocks

In real life conditions, Channels cause time delay, phase and amplitude shift in the received signal. Time delay is a problem solved by introducing cyclic prefix while Phase and Amplitude shift could be eliminated using by adding Pilot carriers (or training). Channel estimates both in time and frequency are interpolated from the available pilots. Increasing the pilot density increases the channel estimation performance but on the cost of resources. This technique is called “Pilot-Symbol-Aided Estimation”. [29]

4.2.4 OFDM & IFFT/FFT

OFDM is the core of WiMAX. OFDM belongs to a family of transmission schemes called multicarrier modulation, which is based on the idea of dividing a given high-bit-rate data stream into several parallel lower bit-rate streams and modulating each stream on separate carriers.often called subcarriers.

Multicarrier modulation schemes eliminate or minimize intersymbol interference (ISI) by making the symbol time large enough so that the channel-induced delays. [29]

Therefore, in high-data-rate systems in which the symbol duration is small, being inversely proportional to the data rate, splitting the data stream into many parallel streams increases the symbol duration of each stream such that the delay spread is only a small fraction of the symbol duration. [29]

OFDM is a spectrally efficient version of multicarrier modulation, where the subcarriers are selected such that they are all orthogonal to one another over the symbol duration, thereby avoiding the need to have nonoverlapping subcarrier channels to eliminate intercarrier interference and that is the reason for using OFDM in WiMAX hogh data rate systems. [29]

Blocks for introducing the orthogonality and multi carrier transmission features are implemented in the model for OFDM creation.

4.2.5 Cyclic Prefix Insertion

Simply adding the end part of the transmitted symbol to its first part. Eliminates ISI as delayed OFDM signals of previous signals only will interfere with the cyclic extended part with is larger than the channel delay. [29]

Eliminates ICI as cyclically extending the OFDM symbol will ensure that delayed replicas of the OFDM symbol always have an integer of cycles within the FFT interval. [29]

4.2.6 Equalization Block

Time domain equalization, in case of high data rates and channels with extensive time spreading an equalizer is unavoidable. The purpose of equalization is not complete removal but restriction of ISI to a tolerable extent.

Frequency domain equalization, used to compensate for channel complex gain at each sub-carrier frequency. For the simple case of no noise, the ideal value of the equalizer's response is the inverse of the channel's frequency response. [29]

4.2.7 Finite Impulse Response Filters

Effective band pass filters the signal removing some of the OFDM side-lobes. Finite Impulse Response (FIR) filters using the windowing method (Kaiser window) can be used. The amount of side-lobe removal depends on the sharpness of the filters used. These filters are optional and can be adding or ignored depending on the results and channel effects. [29]

4.3 WiMAX Simulation on MATLAB using Simulink

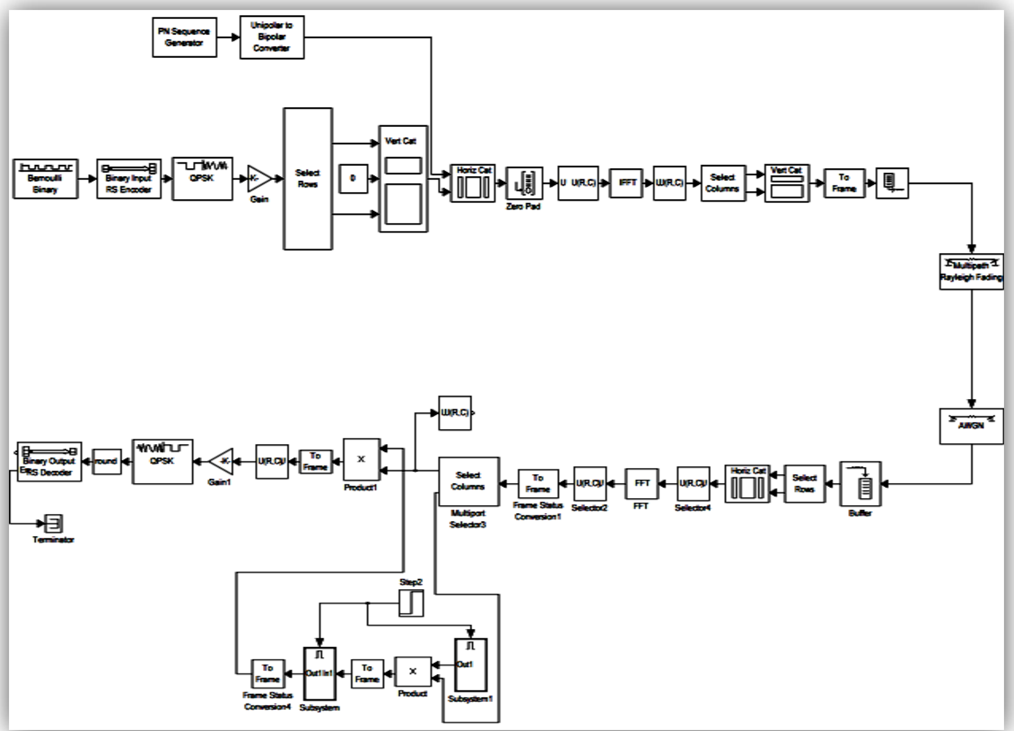

Figure 62: WiMAX model in simulink.

The above figure shows the design for WiMAX system on simulink. The model includes the basic blocks discussed before in additon to a bernolli binary block for input data simulation, two different channels which are:

AWGN channel: the additive white Gaussian noise channel model is one in which the only impairment is a linear addition of wideband or white noise with a constant spectral density (expressed as watts per hertz of bandwidth) and a Gaussian distribution of amplitude. [29]

The model does not account for the phenomena of fading, frequency selectivity, interference, nonlinearity or dispersion. However, it produces simple and tractable mathematical models which are useful for gaining insight into the underlying behavior of a system before these other phenomena are considered.

Wideband Gaussian noise comes from many natural sources, such as the thermal vibrations of atoms in antennas (referred to as thermal noise or Johnson-Nyquist noise), shot noise, black body radiation from the earth and other warm objects, and from celestial sources such as the Sun. [29]

The Multipath Rayleigh Fading Channel block: implements a baseband simulation of a multipath Rayleigh fading propagation channel. It is used to model mobile wireless communication systems. Relative motion between the transmitter and receiver causes Doppler shifts in the signal frequency. [29]

The block multiplies the input signal by samples of a Rayleigh-distributed complex random process. The scalar Initial seed parameter seeds the random number generator. Then, the block generates random numbers using the Ziggurat method. [29]

4.3.1 *Power spectrum of the transmitted and received signals*

The following figures shows the results of the simulation:

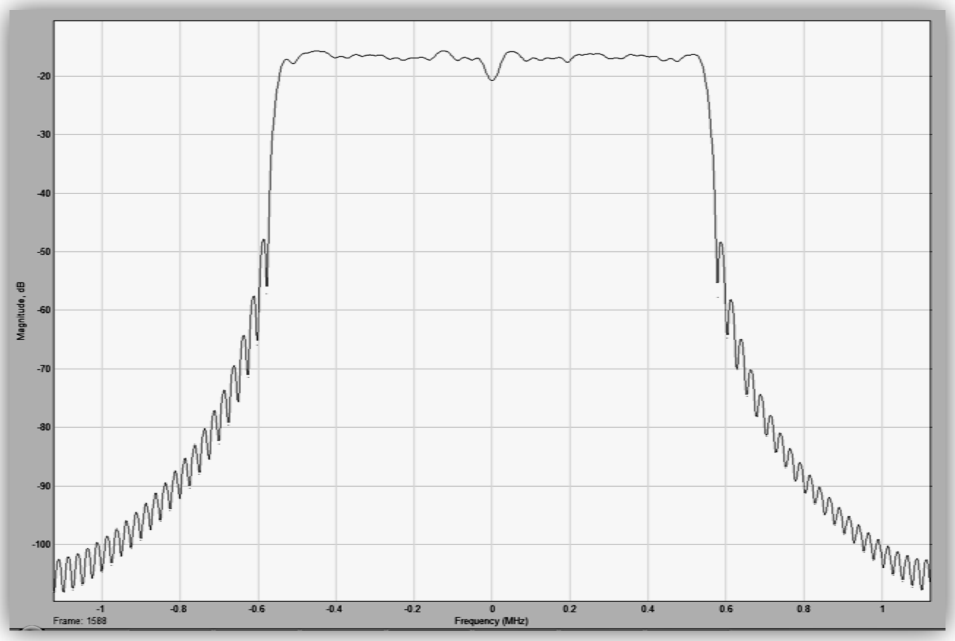

Figure 63: Power spectrum of the transmitted signal.

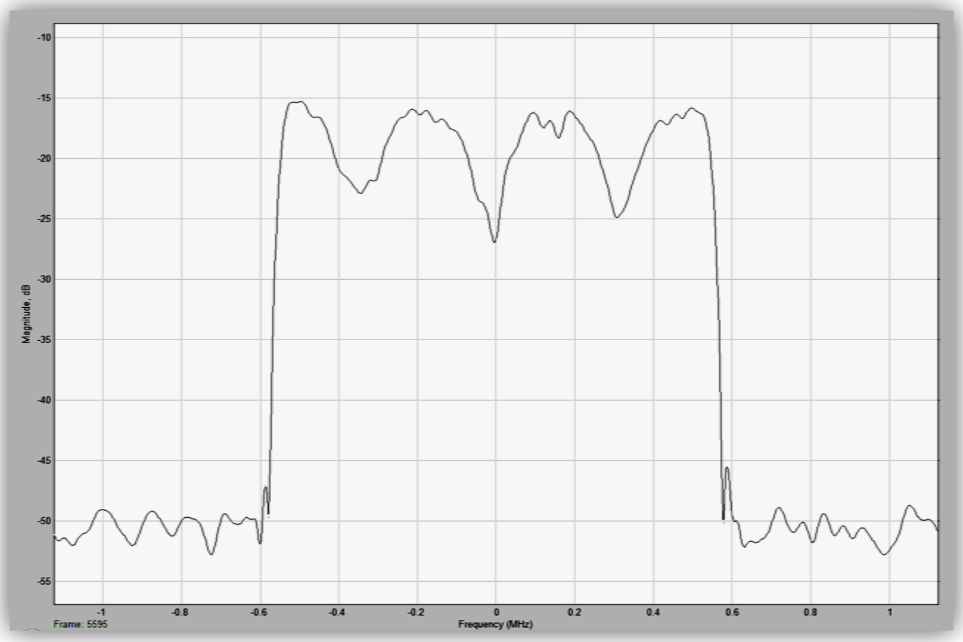

Figure 64: Power spectrum of the received signal.

4.3.2 Constellation diagrams of the model

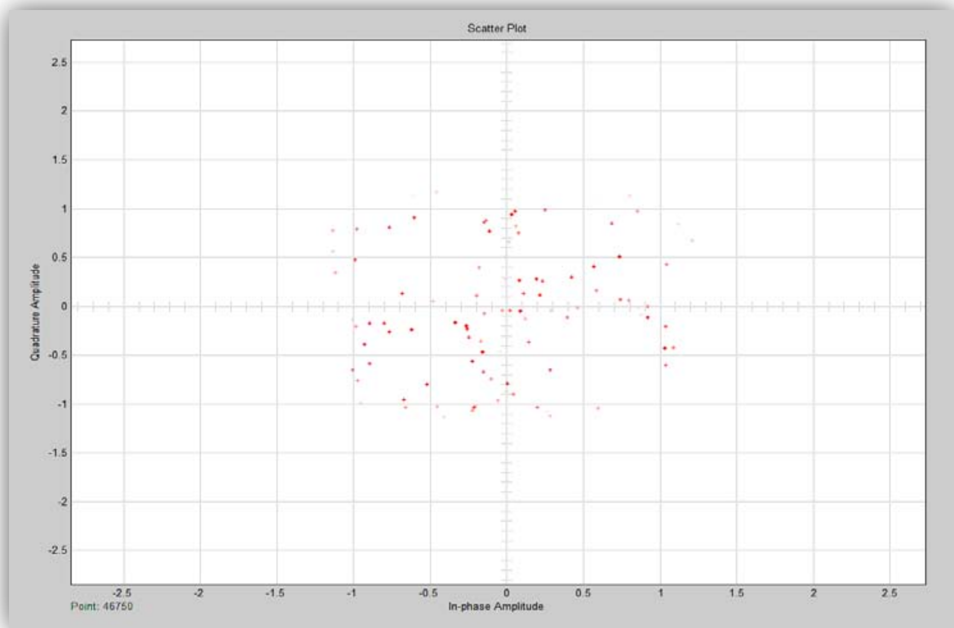

Figure 65: constellations before channel estimation and equalization.

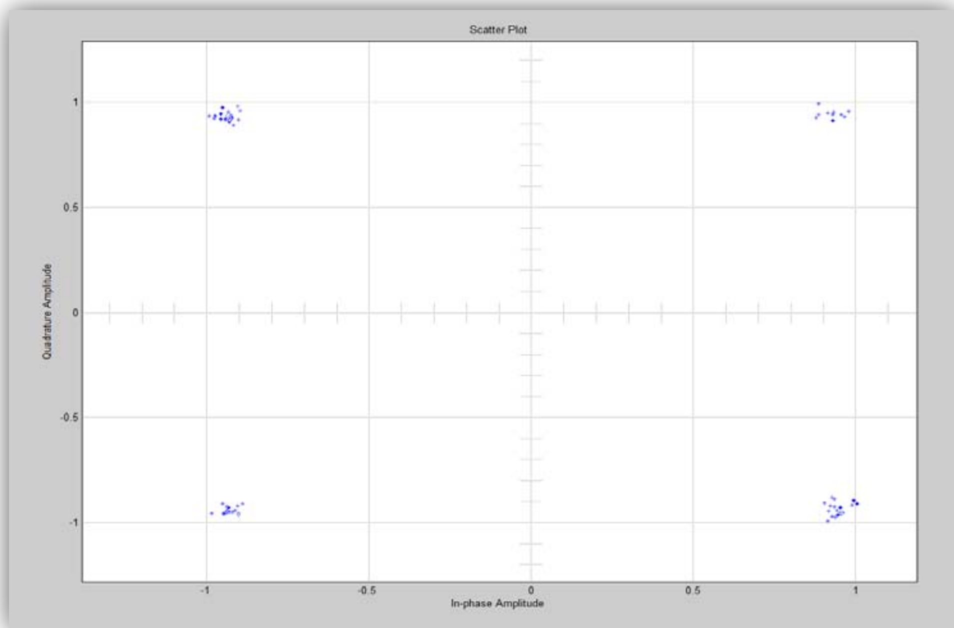

Figure 66: constellations after channel estimation and equalization.

From the constellations diagram for QPSK, this is clear that the equalizations have solved the channels effect in an acceptable manner and have retrieved the data with high accuracy.

4.3.3 Bit Error Rate (BER)

In telecommunications, an error ratio is the ratio of the number of bits, elements, characters, or blocks incorrectly received to the total number of bits, elements, characters, or blocks sent during a specified time interval.

The results are specially obtained from the model using the BERTool MATLAB application and this results are special for the severe channeling conditions that was applied.

4.3.3.1 BER for BPSK

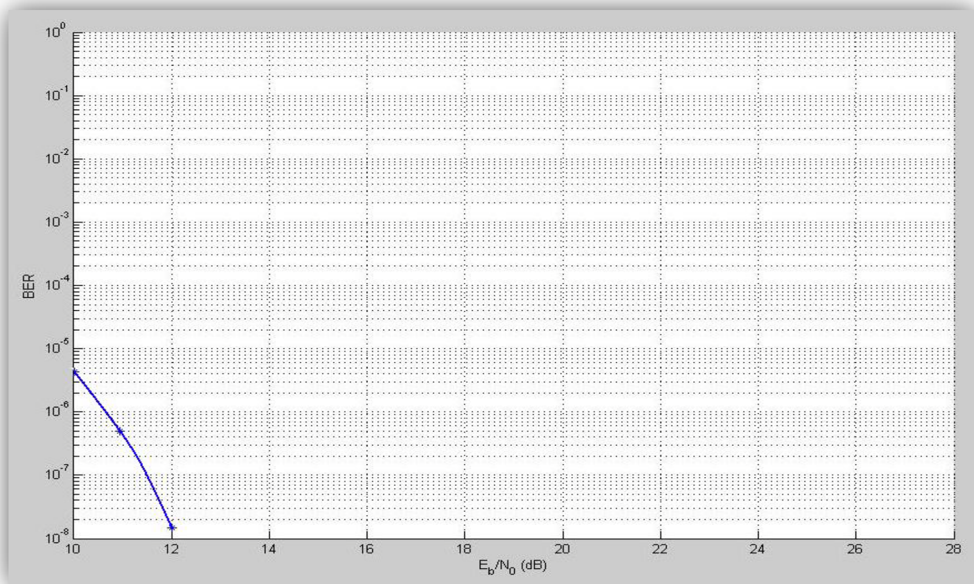

Figure 67: BER for BPSK for SNR ranging from 10 – 30 dB.

4.3.3.2 BER for QPSK

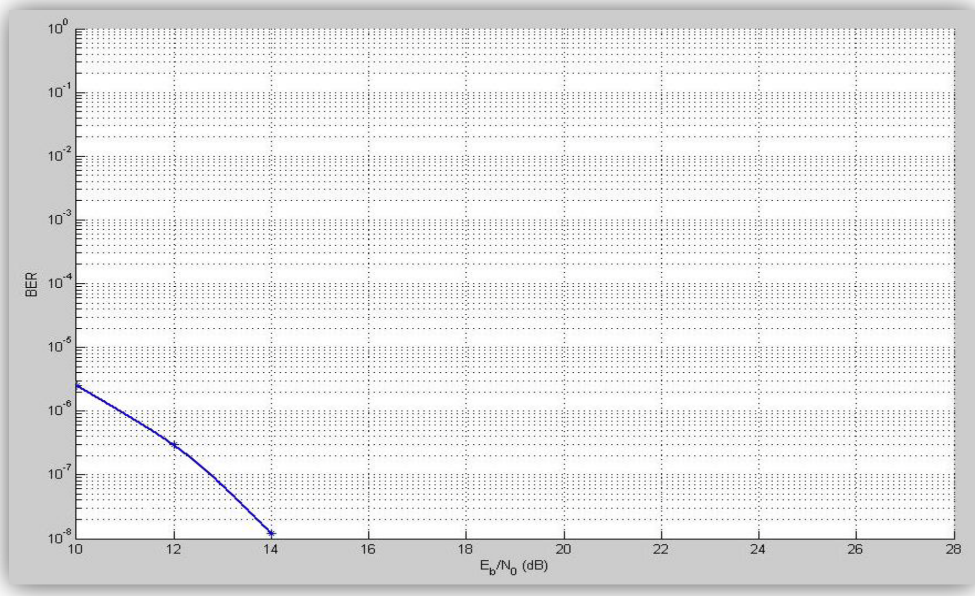

Figure 68: BER for QPSK for SNR ranging from 10 – 30 dB.

4.3.3.3 BER for 8PSK

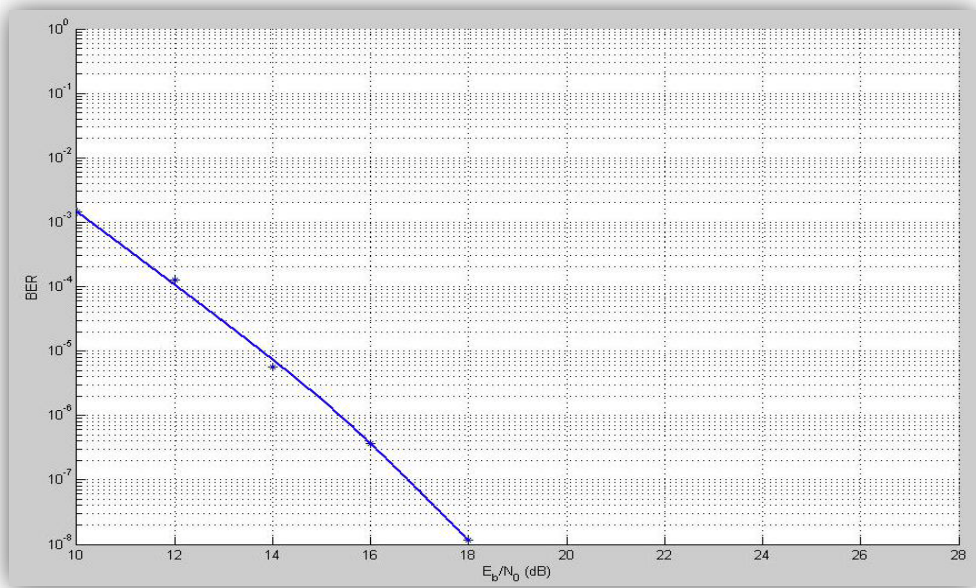

Figure 69: BER for 8PSK for SNR ranging from 10 – 30 dB.

4.3.3.4 BER for 16PSK

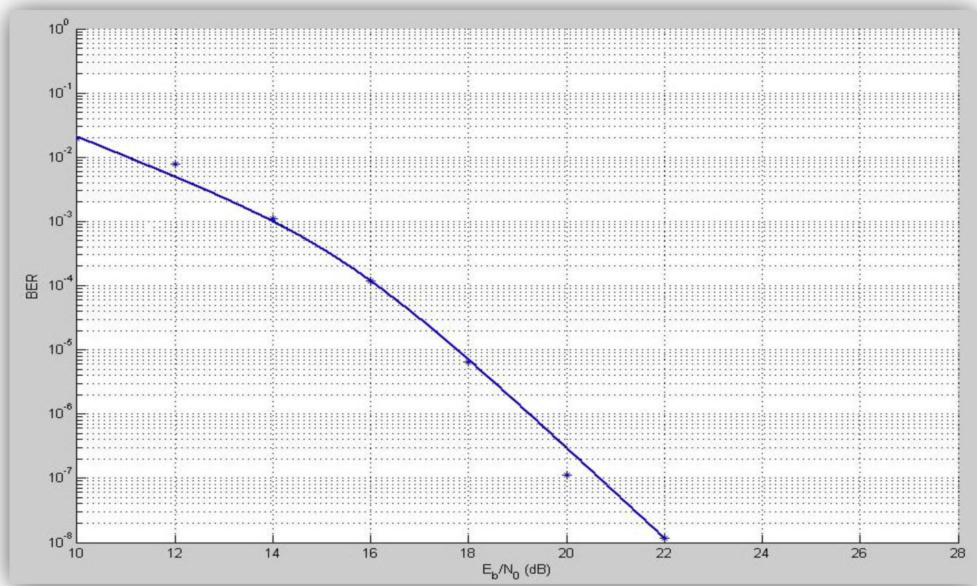

Figure 70: BER for 16PSK for SNR ranging from 10 – 30 dB.

4.3.3.5 BER for 32PSK

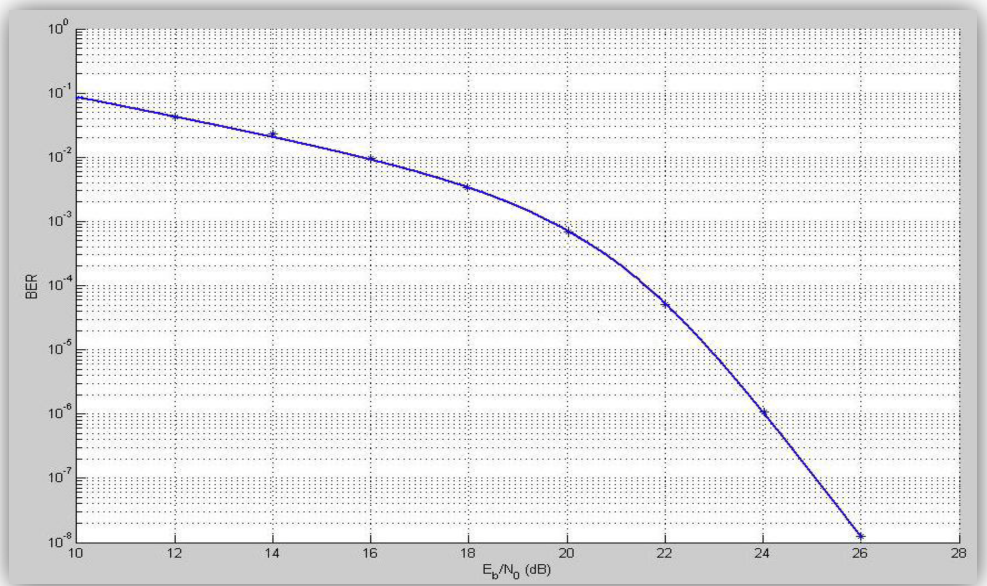

Figure 71: BER for 32PSK for SNR ranging from 10 – 30 dB.

The vertical axis shows the bit-error-rate and the horizontal axis the E_b/N_o (normalized SNR) values. From the previous obtained results two main conclusions were obvious. The first is that for low noise levels (i.e., SNR large), the BER is extremely small.

However, as noise increases beyond a certain threshold level, the BER rapidly becomes unacceptable. The second conclusion is that the larger values of M, i.e., more bits per symbol, require significantly higher SNR to provide comparable bit-error performance.

CHAPTER 5

CONCLUSIONS AND RECOMMENDATIONS

5.1 Conclusions

WiMAX is all-in-one technological solution that will serve users day-to-day demands all put together. As widely known WiMAX enables the delivery of last mile wireless broadband access as an alternative to ADSL and Cable broadband with great bandwidth, mobility and uses.

Orthogonal Frequency Division Multiplexing (OFDM), the physical layer of WiMAX, is very similar to the well known and used technique of Frequency Division Multiplexing (FDM). OFDM uses the principles of FDM to allow multiple messages to be sent over a single radio channel.

WiMAX all other basic and unique features as modulation, channel coding, synchronization, channel estimation and equalization systems have been discussed in details.

The simulation model for WiMAX was Successful and results obtained were very satisfying and very close to the standards. The BER test, power spectrum test and constellations diagrams are the perfect proves on that success.

The goals and objectives of a WiMAX application project that have been set are all satisfied and the research project on the field of WiMAX physical layer has been completed as well as the model design on simulink and WiMAX capacity and planning research.

5.2 Recommendations

The recommendations for this project are to perform more analyses for the system to test its performance completely. These analyses can be improved by adding more severe and real life parameters of channels that exists all around us. In addition more analyses can be done concerning the coverage and capacity planning for fixed WIMAX.

Furthermore a prototype for the 802.16 system will be very important. An advanced DSP kits can be used to perform this task efficiently. During my project time, this was not available but in the future it can be obtainable in the university.

The last recommendation is an economical analysis for the WIMAX system for the global market use.

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APPENDICES

APPENDIX A

FYP I GANTT CHART

No.	Detail/ Week	1	2	3	4	5	6	7	8	9	10	11	12	13	14
1	Selection of Project Topic														
	-Propose Topic														
	-Confirmation of Topic Selection														
2	Preliminary Research Work														
	-Data collection.														
	-Identifying software and tools required.														
	-Literature review on WiMAX.														
3	Submission of Preliminary Report														
4	Project Work														
	-Data Gathering														
5	Submission of Progress Report														
	-Literature Review														
	-Project Description														
6	Project work continue														
	-Start Discussion and Results														
7	Submission of Interim Report Final Draft														
8	Submission of Interim Report														

APPENDIX B

FYP II GANTT CHART

No.	Detail/ Week	1	2	3	4	5	6	7	8	9	10	11	12	13	14
1	Project Work Continue														
	- MATLAB Model Design														
	- MATLAB Simulink Results														
2	Submitting of Progress report														
3	Project work continue														
	-Coverage planning														
	-Capacity planning research														
4	Pre-EDX Poster Exhibition														
5	EDX (if available)														
6	Submission of First Draft														
7	Submission of Final Report (3 copies)														

APPENDIX C

WIMAX SIMULINK MODEL