Sudan University of Science and Technology

College of Graduate Studies

Performance Compration of Equalization Techniques in Downlink Long Term Evolution

مقارنة اداء تقنيات المساواة في الوصلة الهابطة للتطور طويل الامد

A Thesis Submitted in Partial Fulfilment for the Requirements of the Degree of M.Sc. in Electronics Engineering (Communications)

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الآيـــــــة

قَالَ تَعَالَىٰ:

﴿ قَالُواْ سُبۡحَٰنَكَ لَا عِلۡمَ لَنَآ إِلَّا مَا عَلَّمۡتَنَآۖ إِنَّكَ أَنتَ ٱلۡعَلِيمُ ٱلۡحَكِيمُ ٣٢ ﴾

البقرة الاية(32)

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الإهـــــداء

إلى من تعهدتني بالتربية في الصغر، وكانت لي نبراساً يضيء فكري

بالنصح و التوجيه في الكبر

أمي حفظها الله

إلى من شملوني بالعطف ، وأمدوني بالعون ، وحفزوني للتقدم

زوجى الذي دعمني وكان نعم السند

اختي و أخواني رعاهما الله

الى ابنائى فلذات كبدي

إلى اختي روان التى كانت نعم العون

إلى كل من علمني حرفاً ، وأخذ بيدي في سبيل تحصيل العلم والمعرفة

إليهم جميعاً أهدي ثمرة جهدي ، ونتاج بحثي المتواضع .

الشكــر والتقـديــر

ومن حق النعمة الذكر، وأقل جزاء للمعروف الشكر

فبعد شـــكر المــولى عز وجل ، المتـفضل بجليل النعم ، وعـظيم الجزاء

يجدر بي أن أتـقـدم بـبالغ الامتـنان ، وجزيـل العرفان إلى كل من وجهني ، وعلمني ، وأخذ بيدي في سبيل إنجاز هذا البحث .. وأخص بذلك مشرفي الدكتـور: محمد حسين و الدكتور إبراهيم خضر اللذان قوما ، وتابعا ، وصوبا ، بحسن إرشادهم لي في كل مراحل البحث ، والذي وجدت في توجيهاتهم حرص المعلمين ، التي تـؤتي ثمارها الطيـبة بإذن الله.

ABSTRACT

In this research the equalization concepts in Long Term Evolution (LTE) has been discussed in theoretical and practical manners as an analysis and implementation study of channel equalization in LTE downlink. As channel equalization is achieved on the receiver side after the OFDM modulation , we proposed a complete scenario for generation of OFDM frame from random bits which symbolized and QPSK modulated to generate the OFDM frame which consist of 10 sub frame, then the frame is passed through a fading channel with the additive of noise and received to reverse the OFDM operation and the equalization in order to return the original transmitted frame, then due to the different types of equalizer according to different receiver implementations we have to examine the effects of two of the familiar linear equalizers which are the Zero Forcing (ZF) and the Minimum Mean Square Error (MMSE), then calculate the RMS percentage of Error Vector Magnitude for pre-equalized and post-equalized signal for each equalizer. The simulation environment used is MATLAB R2014a, which offers very helpful built-in commands from the communication and LTE toolboxes that designed according to the 3GPP LTE technical specification and communication engineering standards latest releases.

المستخـلص

هذه الأطروحة ناقشت المفاهيم النظرية والعملية للمساواة في نظام التطور طويل الأمد ، باعتبارها دراسة تحليل وتنفيذ معادلة القناة الهابطة في نظام التطور طويل الأمد. كما تحققت معادلة القناة على جانب المستلم بعد تشكيل (او اف دي ام) ، اقترحنا سيناريو كامل للجيل (اواف دي ام) إطار من البتات العشوائية التي تم ترميزها و(كيو بي اس كي) المضمن لتوليد إطار (او اف دي ام) والتي تتكون من ١٠ إطارات فرعية، ثم يتم تمرير إطار من خلال قناة متلاشية مع اضافة الضوضاء والمتلقى لعكس عملية (او اف دي ام) وتحقيق تكافؤ لكي يعود الإطار الأصلي المرسل ، ثم استنادا على أنواع مختلفة من التعادل ووفقا لمختلف تطبيقات الاستقبال لدينا لدراسة الآثار المترتبة على اثنين من المعادلات الخطية المألوفة التي هي الصفر الإجبارى(زد اف) وخطأ الحد الأدنى ساحة المتوسط (ام ام اس اي)، ثم حساب النسبة المئوية (ار ام اس) من خطأ ناقل حجم لما قبل هدف التعادل وبعد هدف التعادل، إشارة لكل هدف التعادل.

بيئة المحاكاة المستخدمة هى ماتلاب ٢٠١٤، والتي تقدم أوامر مفيدة جدا مضمنة من الاتصال وأدوات عمل نظام التطور طويل الأمد التي صممت وفقا للمعايير والمواصفات لمشروع شراكة الجيل الثالث لنظام التطور طويل الأمد و أحدث إصدارات الاتصالات التقنية الهندسية.

TABLE OF CONTENTS

LIST OF FIGURES

LIST OF TABLES

LIST OF ABBREVIATIONS

3GPP	Third Generation Partnership Project
ACI	Adjacent Channel Interference
ADSL	Asymmetric Digital Subscriber Line
APN	Access Point Name
BRANs	Broadband Radio Access Networks
BS	Base Station
CCI	Co-channel Interference
COFDM	Coded OFDM
CS	Packet Switched
DAB	Digital Audio Broadcasting
DL	Downlink
DMT	Discrete Multi-Tone
DSP	Digital Signal Processing
DVB-H	Broadcasting For Handheld Terminals
DVB-T	Digital Video Broadcasting For Terrestrial Television
eNB	Evolved Node B
EPC	Evolved Packet Core
EPS	Evolved Packet System
ETSI	European Telecommunications Standards Institute
E-UTRAN	Evolved UMTS Terrestrial Radio Access Network
EVM	Error Vector Magnitude
GERAN	GSM EDGE Radio Access Network
GSM	Global System for Mobile Communications
HDSL	High-bit-rate Digital Subscriber Line
HSDPA	High-Speed Downlink Packet Access
HSS	Home Subscriber Server
ICI	Inter-Carrier Interference

IEEE	Institute of Electrical and Electronics Engineers
ISI	Inter Symbol Interference
MAP	Maximum A Posteriori
ME	Mobile Equipment
MIMO	Multiple Input Multiple Output
ML	maximum likelihood
MMSE	Minimum Mean Square Error
MRC	Maximum Ratio Combining
MSE	Mean Squared Error
MVUE	Minimum-Variance Unbiased Estimator
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency-Division Multiple Access
PAPR	Peak-to-Average Power Ratio
PDN	Packet Data Network
PHY	Physical Layer
PMI	Pre-coding Matrix Indicator
PS	Packet Switched
QOS	Quality of Service
RI	Rank Indicator
SC-FDMA	Single Carrier Frequency Division Multiple Access
SIM	Subscriber Identity Module
TM	Transmission Modes
UE	User Equipment
UICC	Universal Integrated Circuit Card
UL	Uplink
UMTS	Universal Mobile Telecommunication System
USIM	Universal Subscriber Identity Module
UTRAN	UMTS terrestrial radio access network
V-BLAST	Vertical Bell Labs Layered Space-Time

VDSL	Very-high-speed Digital Subscriber Line
WLANs	Wireless Local Area Networks
ZF	Zero Forcing

CHAPTER ONE

INTRODUCTION

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INTRODUCTION

1.1 Preface

With the increase of demand on the multimedia services, 3GPP LTE is the evolution of the UMTS in response to ever increasing demands for High quality multimedia services according to users' expectations.Since downlink is always an important factor in coverage and capacity aspects, special attention has been given in selecting technologies for LTE downlink.

Novel technologies such as orthogonal frequency division multiplexing (OFDM) and multiple input, multiple output (MIMO), can enhance the performance of the current wireless communication systems. The high data rates and the high capacity can be attained by using the advantages of the two technologies. These technologies have been selected for LTE downlink [1].

Accordingly, exact channel estimation technology is the key to the high transmission performance and ensures the superiority of the system. So this thesis is engaged in the research on the channel equalization technology for LTE downlink.

1.2 Problem Statement

- In wireless communication, a signal suffers with an Inter Symbol Interference (ISI) which occurs when the data transmitted through the channel is dispersive, in which each received pulse is affected somewhat by adjacent pulses and due to which interference occurs in the transmitted signals.

- Multiple paths of propagation are equivalent to transmitting the same signal through a number of separate channels, each having a Deferent attenuation and delay.
- Co-channel Interference (CCI) and Adjacent Channel Interference (ACI) occur in communication systems due to multiple access techniques using space, frequency or time.

1.3 Proposed Solution

- Introduce the concept of equalization which meant by adjusting or rendering the balance between frequency components within signal.
- Overview of techniques used to Reduce inter-symbol interference to allow recovery of the transmit symbols.
- Background of equalizer in wireless communications.
- Simulation of Linear detection and interference cancelation in LTE by evaluation of two receiver equalizers; the Zero Forcing (ZF) and the Minimum Mean Square Error (MMSE)

1.4 Aim and Objectives

In OFDM, as well as SC-FDMA, equalization is achieved on the receiver side, after the FFT calculation, by multiplying each Fourier coefficient by a complex number.

Thus, frequency selective fading and phase distortion can be easily combated.

The advantage is that frequency domain equalization using FFTs requires less computation than conventional time domain equalization.

Equalization is one of the two procedures used to reduce the ISI phenomenon by using a certain mathematical algorithm that predesigned and prototyped in the receiver physical implementation for this mission, the other way is by using the error correcting codes which known as turbo codes.

The objectives of this these are:

- Evaluate the performance and summarize the results of ZF and MMSE equalizer in case of SISO.
- Evaluate the performance and summarize the results of ZF and MMSE equalizer in case of 2x2 MIMO without channel MIMO correlation.
- Evaluate the performance and summarize the results of ZF and MMSE equalizer in case of 2x2 MIMO with channel MIMO correlation.

1.5 Methodology

In this research which focuses in LTE technology, after reviewing the LTE network architecture in order to understand the major components we dealing with in this research, the equalization concepts in LTE will be discussed in theoretical and practical manners as an analysis and implementation study of channel equalization in LTE downlink.

As channel equalization is achieved on the receiver side after the OFDM modulation and the channel estimation processes, we will propose a complete scenario for generation of OFDM frame from random bits which symbolized and QPSK modulated to generate the OFDM frame which consist of 10 sub frame.

Then the frame is passed through a fading channel with the additive of noise and received to reverse the OFDM operation and establish the channel estimation process and the equalization in order to return the original transmitted frame, Then due to the different types of equalizer according to different receiver implementations we have to examine the effects of two of the familiar linear equalizers which are the Zero Forcing (ZF) and the Minimum Mean Square Error (MMSE),

Then calculate the RMS percentage of Error Vector Magnitude for pre-equalized and post-equalized signal for each equalizer.

The simulation environment used is MATLAB R2014a, which offers very helpful built-in commands from the communication and LTE toolboxes that designed according to the 3GPP LTE technical specification and communication engineering standards latest releases.

1.6 Thesis Outlines

Chapter one contains introduction and general description of the research topics. Chapter two gives theoretical background of the research topics and equalization techniques. Chapter three describes methodology and system design and the implementation of the study in MATLAB. Chapter four includes discussion of the simulation returned results. Chapter five contains conclusion and summary acknowledge after the study.

CHAPTER TWO

LITERATURE REVIEW

CHAPTER TWO

LITERATURE REVIEW

2.1 Background

Queue length of each CC is taken into account to balance the load among all CCs. A set of system-level simulations have been performed to support the proposed method. The obtained results demonstrated that the proposed method significantly improves the user throughput up to 39.40% compared to the well-known method of previous studies.

2.1.1 Introduction to LTE

LTE stands for Long Term Evolution, and is a registered trademark owned by ETSI (European Telecommunications Standards Institute) for the wireless data communications technology and a development of the GSM/UMTS standards. However other nations and companies do play an active role in the LTE project. The goal of LTE was to increase the capacity and speed of wireless data networks using new DSP (Digital Signal Processing) techniques and modulations that were developed around the turn of the millennium.

A further goal was the redesign and simplification of the network architecture to an IP-based system with significantly reduced transfer latency compared to the 3G architecture. The LTE wireless interface is incompatible with 2G and 3G networks, so that it must be operated on a separate radio spectrum [6].

LTE was designed by a collaboration of national and regional telecommunications standards bodies known as the Third Generation Partnership Project (3GPP) [3], and is known in full as 3GPP Long Term Evolution. LTE evolved from an earlier 3GPP system known as the Universal Mobile Telecommunication System (UMTS), which in turn evolved from the Global System for Mobile Communications (GSM). To put LTE into context, we will begin by reviewing the architectures of UMTS and GSM and by introducing some of the important terminology.

2.1.1.1 Architectural Review of UMTS and GSM

A mobile phone network is officially known as a public land mobile network (PLMN), and is run by a network operator such as Zain, Sudani, or MTN. UMTS and GSM share a common network architecture, which is shown in Figure 2.1.

There are three main components, namely the core network, the radio access network and the mobile phone.

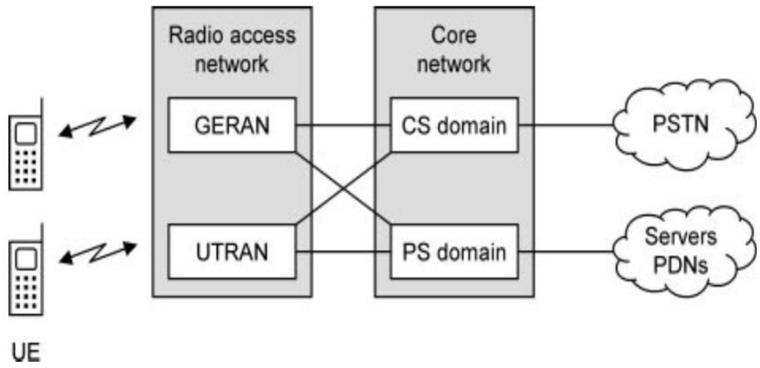

Figure 2.1 High Level Architecture of UMTS and GSM.

The core network contains two domains. The circuit switched (CS) domain transports phone calls across the geographical region that the network operator is covering, in the same way as a traditional fixed-line telecommunication system.

It communicates with the public switched telephone network (PSTN) so that users can make calls to land lines and with the circuit switched domains of other network operators. The packet switched (PS) domain transports data streams, such as web pages and emails, between the user and external packet data networks (PDNs) such as the internet [3].

The radio access network handles the core network's radio communications with the user. In Figure 2.1, there are actually two separate radio access networks, namely the GSM EDGE radio access network (GERAN) and the UMTS terrestrial radio access network (UTRAN). These use the different radio communication techniques of GSM and UMTS, but share a common core network between them.

The user's device is known officially as the user equipment (UE) and colloquially as the mobile. It communicates with the radio access network over the air interface, also known as the radio interface.

The direction from network to mobile is known as the downlink (DL) or forward link and the direction from mobile to network is known as the uplink (UL) or reverse link.

A mobile can work outside the coverage area of its network operator by using the resources from two public land mobile networks: the visited network, where the mobile is located, and the operator's home network. This situation is known as roaming.

2.1.2 Architecture of LTE

Figure 2.2 reviews the high-level architecture of the evolved packet system (EPS). There are three main components, namely the user equipment (UE), the evolved UMTS terrestrial radio access network (E-UTRAN) and the evolved packet core (EPC).

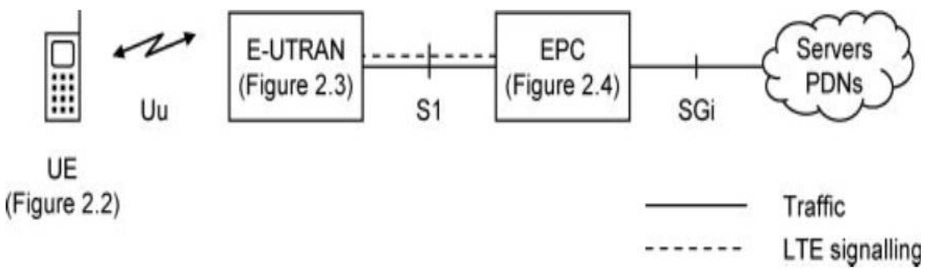

Figure 2.2: High Level Architecture of LTE

In turn, the evolved packet core communicates with packet data networks in the outside world such as the internet, private corporate networks or the IP multimedia subsystem.

The interfaces between the different parts of the system are denoted Uu, S1 and SGi. The UE, E-UTRAN and EPC each have their own internal architectures and we will now discuss these one by one.

2.1.2.1 User Equipment

Figure 2.3 shows the internal architecture of the user equipment [5]. The architecture is identical to the one used by UMTS and GSM.

The actual communication device is known as the mobile equipment (ME). In the case of a voice mobile or a smartphone, this is just a single device. However, the mobile

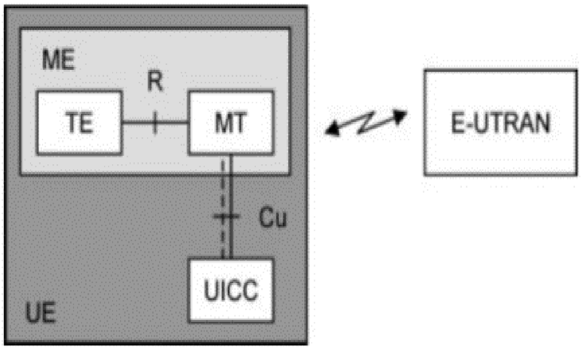

Figure 2.3: Internal Architecture of The UE.

The universal integrated circuit card (UICC) is a smart card, colloquially known as the SIM card. It runs an application known as the universal subscriber identity module (USIM) [5], which stores user-specific data such as the user’s phone number and home network identity.

The USIM also carries out various security related calculations, using secure keys that the smart card stores. LTE supports mobiles that are using a USIM from Release 99 or later, but it does not support the subscriber identity module (SIM) that was used by earlier releases of GSM.

2.1.2.2 Evolved UMTS Terrestrial Radio Access Network

The evolved UMTS terrestrial radio access network (E-UTRAN) is illustrated in Figure 2.4. The E-UTRAN handles the radio communications between the mobile and the evolved packet core and just has one component, the evolved Node B (eNB). Each eNB is a base station that controls the mobiles in one or more cells [6].

A mobile communicates with just one base station and one cell at a time, so there is no equivalent of the soft handover state from UMTS. The base station that is communicating with a mobile is known as its serving eNB.

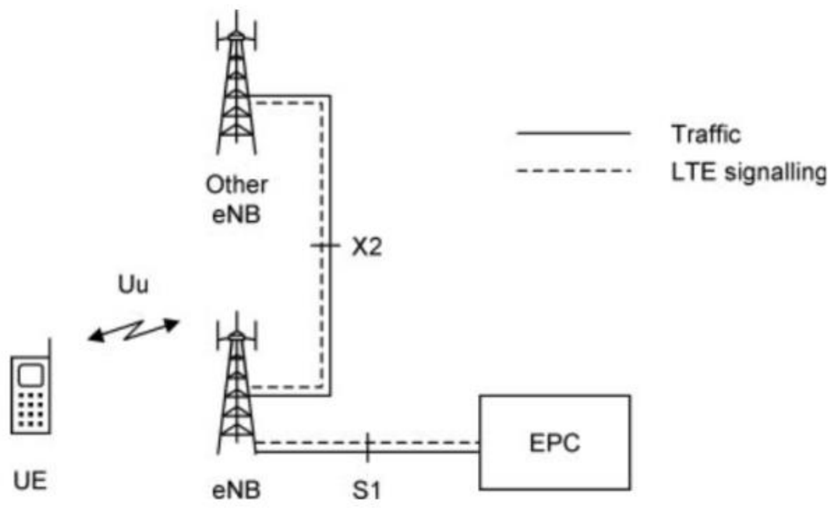

Figure 2.4: Architecture of the evolved UMTS terrestrial radio access network.

The eNB has two main functions. Firstly, the eNB sends radio transmissions to all its mobiles on the downlink and receives transmissions from them on the uplink, using the analogue and digital signal processing functions of the LTE air interface.

Secondly, the eNB controls the low level operation of all its mobiles, by sending them signalling messages such as handover commands that relate to those radio transmissions. In carrying out these functions, the eNB combines the earlier functions of the Node B and the radio network controller, to reduce the latency that arises when the mobile exchanges information with the network.

2.1.2.3 Evolved Packet Core

Figure 2.5 shows the main components of the evolved packet core. We have already seen as one component, the home subscriber server (HSS), which is a central database that contains information about all the network operator's subscribers.

This is one of the few components of LTE that has been carried forward from UMTS and GSM [6], [7].

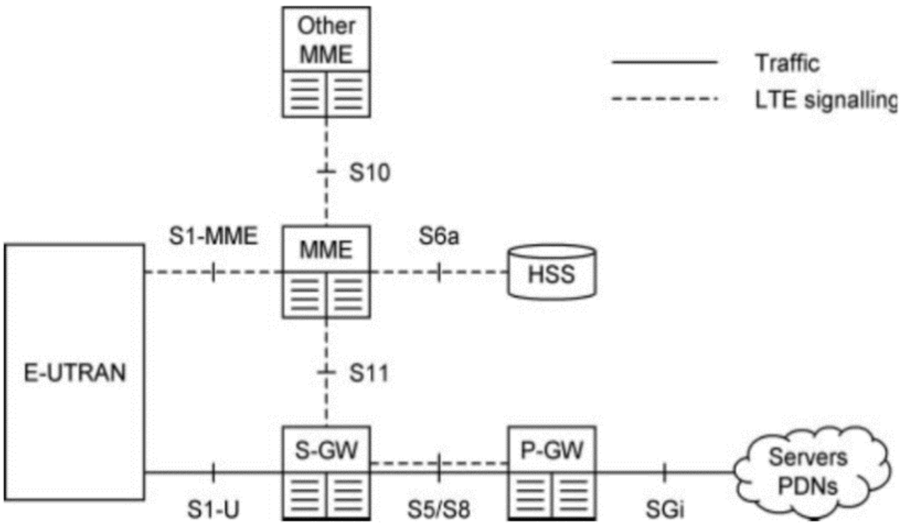

Figure 2.5 Main Components of the Evolved Packet Core

The packet data network (PDN) gateway (P-GW) is the EPC's point of contact with the outside world. Through the SGi interface, each PDN gateway exchanges data with one or more external devices or packet data networks, such as the network operator's servers, the internet or the IP multimedia subsystem.

Each packet data network is identified by an access point name (APN) [8]. A network operator typically uses a handful of different APNs, for example one for its own servers and one for the internet.

2.1.3 Orthogonal Frequency-Division Multiplexing

In recent years Orthogonal Frequency-Division Multiplexing (OFDM) has emerged as a successful air-interface technique [9], [10]. In the context of wired environments, OFDM techniques are also known as Discrete Multi-Tone (DMT) transmissions and employed within Asymmetric Digital Subscriber Line (ADSL), High-bit-rate Digital Subscriber Line (HDSL), and Very-high-speed Digital Subscriber Line (VDSL) [11].

OFDM has been advocated by many European standards, such as Digital Audio Broadcasting (DAB), Digital Video Broadcasting for Terrestrial television (DVB-T), Digital Video Broadcasting for Handheld terminals (DVB-H), Wireless Local Area Networks (WLANs) and Broadband Radio Access Networks (BRANs) [11].

Furthermore, OFDM has been ratified as a standard or has been considered as a candidate standard by a number of standardization groups of the Institute of Electrical and Electronics Engineers (IEEE), such as the IEEE 802.11 and the IEEE 802.16 standard families [12].

The main merit of OFDM is the fact that the radio channel is divided into many narrowband, low-rate, frequency-non-selective sub channels or subcarriers, so that multiple symbols can be transmitted in parallel, while maintaining a high spectral efficiency.

Each subcarrier may deliver information for a different user, resulting in a simple multiple-access scheme known as Orthogonal Frequency-Division Multiple Access (OFDMA).

This enables different media such as video, graphics, speech, text or other data to be transmitted within the same radio link, depending on the specific types of services and their Quality-of-Service (QOS) requirements [11].

Furthermore, in OFDM systems different modulation schemes can be employed for different subcarriers or even for different users.

For example, the users close to the Base Station (BS) may have a relatively good channel quality, thus they can use high-order modulation schemes to increase their data rates.

By contrast, for those users where far from the BS or are serviced in highly loaded urban areas, where the subcarriers' quality is expected to be poor, low-order modulation schemes can be invoked.

2.1.3.1 OFDM Advantages

Besides its implementation flexibility, the low complexity required in transmission and reception as well as the attainable high performance render, OFDM is a highly attractive candidate for high data rate communications over time-varying frequency selective radio channels.

For example, in classic single carrier systems, complex equalizers have to be employed at the receiver for the sake of mitigating the Inter-Symbol Interference (ISI) introduced by multi-path propagation.

By contrast, when using a cyclic prefix, OFDM exhibits a high resilience against the ISI. Incorporating channel coding techniques into OFDM systems, which results in Coded OFDM (COFDM) [16], allows us to maintain robustness against frequency-selective fading channels, where busty errors are encountered at specific subcarriers in the FD.

2.1.3.1 OFDM Disadvantages

Besides its significant advantages, OFDM also has a few disadvantages. One problem is the associated increased Peak-to-Average Power Ratio (PAPR) in comparison with single-carrier systems [10], requiring a large linear range for the OFDM transmitter's output amplifier. In addition, OFDM is sensitive to carrier frequency offset, resulting in Inter-Carrier Interference (ICI) [13].

2.1.4 Multiple Access Schemes

Employment of multiple antennas at transmitter and receiver is on of attracted significant interests of using high data rate wireless communications, which referred to as the Multiple Input Multiple Output technique to realize the concept of effective cost to throughput approach. As an important element for building next generation wireless communication systems, MIMOs are capable of supporting significantly higher data rates than the Universal Mobile Telecommunications System (UMTS) and the High-Speed Downlink Packet Access (HSDPA) based 3G networks [14].

As indicated by the terminology, a MIMO system employs multiple transmitter and receiver antennas for delivering parallel data streams, as illustrated in Figure 2.6. Since the information is transmitted through different paths, a MIMO system is capable of exploiting transmitter and receiver diversity, hence maintaining reliable communications.

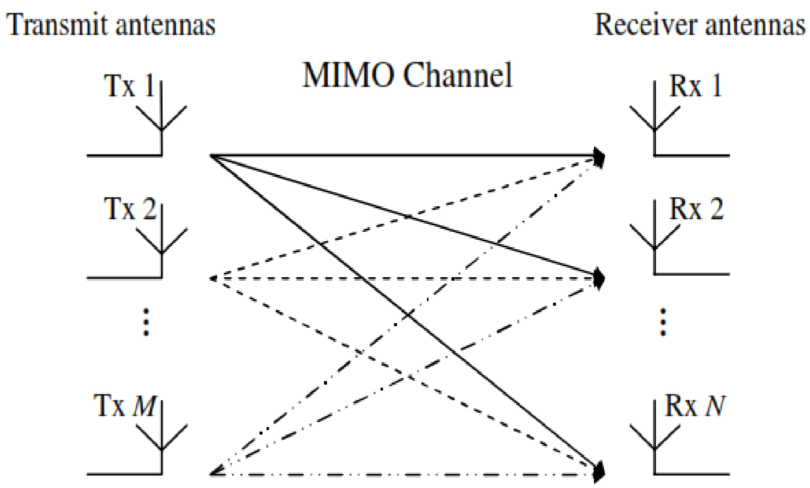

Figure 2.6: Multiple Input Multiple Output Technique

Briefly compared with Single Input Single Output (SISO) systems, the two most significant advantages of MIMO systems are:

A significant increase of both the system's capacity and spectral efficiency. The capacity of a wireless link increases linearly with the minimum of the number of transmitter or receiver antennas [14].

The data rate can be increased by spatial multiplexing without consuming more frequency resources and without increasing the total transmit power.

Dramatic reduction of the effects of fading due to the increased diversity.

This is particularly beneficial when the different channels fade independently.

Multiple input multiple output (MIMO) technologies introduced in LTE such as spatial multiplexing, transmit diversity, and beam-forming are key components for providing higher peak rate at a better system efficiency, which are essential for supporting future broadband data service over wireless links.

2.1.4.1 Multiple Transmission Modes in LTE

There are seven Transmission Modes (TM) defined for LTE, which are defined in LTE Release 8, and they expressed as follow:

- TM1 - Single-antenna transmission: In this mode the data is transmitted only by one antenna.
- TM2 - Transmit diversity: In this mode, the same information is transmitted on multiple antennas using Space-Frequency Block Codes (SFBC) which is an open-loop diversity technique. Only Channel Quality Indicator (CQI) information is required from the UE side.
- TM3 - Large delay CDD (Open-loop spatial multiplexing): Pre-coded transmission is used in this mode over two or more transmit antennas. As multiple code-words are used, this scheme provides

better peak throughput than transmit diversity. This mode requires the UE to transmit only the (transmit rank indicator) to assign the number of code words.

- TM4 - Closed-loop spatial multiplexing: In this mode the UE feeds back the Pre-coding Matrix Indicator (PMI) and Transmit Rank Indicator (RI) obtained from CRS reference signals. The closed-loop operation allows the transmitter to pre-code the data into orthogonal streams (maximum 4). The used pre-coder matrix is signalled to the UE in the PDCCH.
- TM5 - Multi-user MIMO: This is a Rank 1 MU-MIMO transmission mode which is based on the same pre-coders and feedback information as TM4.
- TM6 - Closed-loop Rank 1 with pre-coding: This mode is similar to TM4 except that only one transmission stream is used.
- TM7 - Single antenna transmission: This mode is suitable for UE-specific beam-forming which makes use of the angle of arrival information (not closed-loop PMI feedback). The CQI is fed back with the time of arrival assumption.

LTE transmission mode No.1 is usually regarded as the SIMO transmission mode. In this mode, the signal processing chain is very similar to the SISO case, with the exception that it employs multiple receive antennas.

Using multiple antennas at the receiver allows us to take advantage of receive diversity. Receive diversity with Maximum Ratio Combining (MRC) results in a system with better BER performance than its SISO counterpart.

The multiple-access scheme in LTE downlink uses Orthogonal Frequency Division Multiple Access (OFDMA) and uplink uses Single Carrier Frequency Division Multiple Access (SC-FDMA). These multiple access solutions provide orthogonality between the users, reducing the interference and improving the network capacity.

The resource allocation in the frequency domain takes place with a resolution of 180 kHz resource blocks both in uplink and in downlink. The frequency dimension in the packet scheduling is one reason for the high LTE capacity.

The uplink user specific allocation is continuous to enable single carrier transmission while the downlink can use resource blocks freely from different parts of the spectrum.

The uplink single carrier solution is also designed to allow efficient terminal power amplifier design, which is relevant for the terminal battery life. The LTE solution enables spectrum flexibility where the transmission bandwidth can be selected between 1.4 MHz and 20 MHz depending on the available spectrum. The 20 MHz bandwidth can provide up to 150 Mbps downlink user data rate with 2 × 2 MIMO, and 300 Mbps with 4 × 4 MIMO. The uplink peak data rate is 75 Mbps [4].

The multiple access schemes in LTE are illustrated in Figure 2.7.

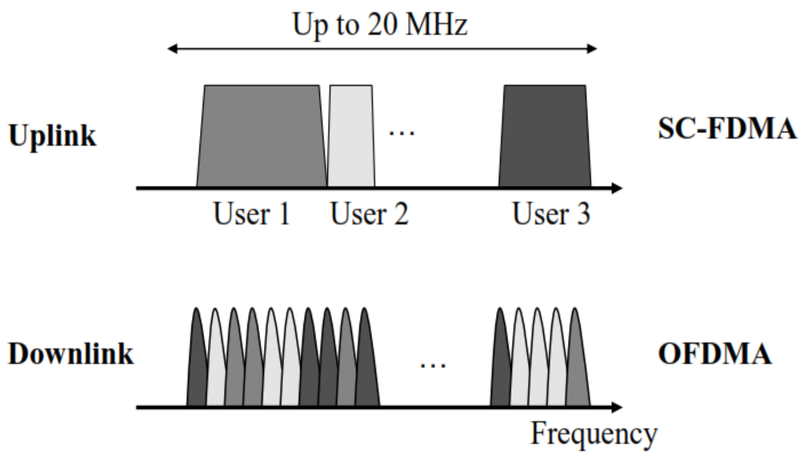

Figure 2.7: LTE Multiple Access Schemes

2.1.5 Inter-Symbol Interference

In telecommunication, Inter Symbol Interference (ISI) is a form of distortion of a signal in which one symbol interferes with subsequent symbols.

This is an unwanted phenomenon as the previous symbols have similar effect as noise, thus making the communication less reliable. ISI is usually caused by multipath propagation or the inherent non-linear frequency response of a channel causing successive symbols to "blur" together.

In addition, if the modulation bandwidth exceeds the coherence bandwidth of the channel, ISI will be introduced and the consecutive transmitted symbols are distorted, since the past and current symbols of the signals overlap. Hence, at the receiver, channel equalizers have to be employed to remove the effects of ISI [9].

The presence of ISI in the system introduces errors in the decision device at the receiver output. Therefore, in the design of the transmitting and receiving filters, the objective is to minimize the effects of ISI, and thereby deliver the digital data to its destination with the smallest error rate possible [8].

2.1.6 Channel Estimation and Equalization

Channel estimation and equalization is the process of adjusting the balance between frequency components within a signal depending on previously mandated constrains and conditions that are defined for each transmission mode in the technical specification to maintain the radio link reliability and spectral efficiency.

The most well-known use of equalization is in sound recording and reproduction but there are many other applications in electronics and telecommunications.

The circuit or equipment used to achieve equalization is called an equalizer. These devices strengthen (boost) or weaken (cut) the energy of specific frequency bands.

In telecommunications, equalizers are used to render the frequency response for instance of a telephone line flat from end to end. When a channel has been "equalized" the frequency domain attributes of the signal at the input are faithfully reproduced at the output. Telephones, DSL lines and television cables use equalizers to prepare data signals for transmission.

Modern digital telephone systems have less trouble in the voice frequency range as only the local line to the subscriber now remains in analog format, but DSL circuits operating in the MHz range on those same wires may suffer severe attenuation distortion, which is dealt with by automatic equalization or by abandoning the worst frequencies.

In digital communications, its purpose is to reduce Inter Symbol Interference to allow recovery of the transmit symbols. It may be a simple linear filter or a complex algorithm. The following equalizer types are commonly used in digital communications:

- ❖ Linear Equalizer

  processes the incoming signal with a linear filter

- ▪ MMSE equalizer

  Designs the filter to minimize $E[|e|^2]$, where e is the error signal, which is the filter output minus the transmitted signal.[1]

- ▪ Zero Forcing Equalizer

  Approximates the inverse of the channel with a linear filter.

- ❖ Decision Feedback Equalizer

Augments a linear equalizer by adding a filtered version of previous symbol estimates to the original filter output [2].

- Blind Equalizer

Estimates the transmitted signal without knowledge of the channel statistics, using only knowledge of the transmitted signal's statistics.

- Adaptive Equalizer

Is typically a linear equalizer or a DFE. It updates the equalizer parameters (such as the filter coefficients) as it processes the data. Typically, it uses the MSE cost function; it assumes that it makes the correct symbol decisions, and uses its estimate of the symbols to compute e, which is defined above.

- Viterbi Equalizer

Finds the maximum likelihood (ML) optimal solution to the equalization problem. Its goal is to minimize the probability of making an error over the entire sequence.

- BCJR Equalizer

Uses the BCJR algorithm (also called the Forward-backward algorithm) to find the maximum a posteriori (MAP) solution. Its goal is to minimize the probability that a given bit was incorrectly estimated.

- Turbo Equalizer

applies turbo decoding while treating the channel as a Convolutional code.

The effect of an equalization system is to compensate for transmission channel impairments such as frequency dependent phase and amplitude distortion.

Besides correcting for channel frequency-response anomalies, the equalizer can cancel the effects of multipath signal components, which can manifest themselves in the form of voice echoes, video ghosts or Rayleigh fading conditions in mobile communications channels.

Equalizers specifically designed for multipath correction are often termed echo cancelers or de-ghosters. They may require significantly longer filter spans than simple spectral equalizers, but the principles of operation are essentially the same.

DSP-based equalizer systems have become ubiquitous in many diverse applications including voice, data, and video communications via various transmission media.

Typical applications range from acoustic echo cancelers for full-duplex speakerphones to video de-ghosting systems for terrestrial television broadcasts to signal conditioners for wireline modems and wireless telephony.

Multiple input multiple output (MIMO) technologies introduced in LTE such as spatial multiplexing, transmit diversity, and beam-forming are key components for providing higher peak rate at a better system efficiency, which are essential for supporting future broadband data service over wireless links.

2.1.6.1 Minimum Mean Square Error (MMSE)

In MIMO wireless communication, an equalizer is employed which is a network that makes an attempt to recover a signal that has suffers with an Inter Symbol Interference (ISI) and proves the BER characteristics and maintains a good SNR.

A Minimum Mean Square Error (MMSE) estimator is a method in which it minimizes the mean square error (MSE), which is a common measure of estimator quality.

Minimum mean-square error equalizer, which does not usually eliminate ISI completely but instead, minimizes the total power of the noise and ISI components in the output. The MMSE estimator is then defined as the estimator achieving minimal MSE. Generally, it very difficult to determine a closed form for the MMSE estimator; in these cases, one possibility is to seek the technique minimizing the MSE within a particular class, such as the class of linear estimators.

The linear MMSE estimator is the estimator achieving minimum MSE among all estimators of the form AY + b. If the measurement Y is a random vector, A is a matrix and b is a vector.

In statistics and signal processing, a minimum mean square error (MMSE) estimator is an estimation method which minimizes the mean square error (MSE) of the fitted values of a dependent variable, which is a common measure of estimator quality.

In the Bayesian setting, the term MMSE more specifically refers to estimation in a Bayesian setting with quadratic cost function. In such case, the MMSE estimator is given by the posterior mean of the parameter to be estimated.

Since the posterior mean is cumbersome to calculate, the form of the MMSE estimator is usually constrained to be within a certain class of functions. Linear MMSE estimators are a popular choice since they are easy to use, calculate, and very versatile.

It has given rise to many popular estimators such as the Wiener-Kolmogorov filter and Kalman filter

The term MMSE more specifically refers to estimation in a Bayesian setting with quadratic cost function. The basic idea behind the Bayesian approach to estimation stems from practical situations where we often have some prior information about the parameter to be estimated.

For instance, we may have prior information about the range that the parameter can assume; or we may have an old estimate of the parameter that we want to modify when a new observation is made available; or the statistics of an actual random signal such as speech.

This is in contrast to the non-Bayesian approach like minimum-variance unbiased estimator (MVUE) where absolutely nothing is assumed to be known about the parameter in advance and which does not account for such situations. In the Bayesian approach, such prior information is captured by the prior probability density function of the parameters; and based directly on Bayes theorem, it allows us to make better posterior estimates as more observations become available.

Thus unlike non-Bayesian approach where parameters of interest are assumed to be deterministic, but unknown constants, the Bayesian estimator seeks to estimate a parameter that is itself a random variable. Furthermore, Bayesian estimation can also deal with situations where the sequence of observations is not necessarily independent. Thus Bayesian estimation provides yet another alternative to the MVUE.

This is useful when the MVUE does not exist or cannot be found Let x be a $n \times 1$ hidden random vector variable, and let Y be a $m \times 1$ known random vector variable (the measurement or observation), both of them not necessarily of the same dimension.

An estimator $\hat{x}\ (Y)$ of x is any function of the measurement Y. The estimation error vector is given by $e = \hat{x} - x$ and its mean squared error (MSE) is given by the trace of error covariance matrix.

$$MSE = tr\,\{\,E\,\{(\hat{x} - x\,)(\hat{x} - x\,)^T\}\,\} \tag{2.1}$$

Where the expectation E is taken over both x and . When x is a scalar variable, the MSE expression simplifies to $E\,\{(\hat{x} - x)^2\}$.

Note that MSE can equivalently be defined in other ways, since

$$tr\ \{E\ \{ee^T\}\} = E\ \{tr\ \{ee^T\}\} = E\ \{e^T e\ \} = \sum_{i=1}^{n} \quad E\ \{e_i^2\} \qquad (2.2)$$

The MMSE estimator is then defined as the estimator achieving minimal MSE:

$$\hat{x}_{MMSE}(Y) = arg\ arg\ MSE \qquad (2.3)$$

2.1.6. 2 Zero forcing

1. Zero Forcing Equalizer refers to a form of linear equalization algorithm used in communication systems which applies the inverse of the frequency response of the channel. This form of equalizer was first proposed by Robert Lucky.

2. The Zero-Forcing Equalizer applies the inverse of the channel frequency response to the received signal, to restore the signal after the channel [15].

3. It has many useful applications. For example, it is studied heavily for IEEE 802.11n (MIMO) where knowing the channel allows recovery of the two or more streams which will be received on top of each other on each antenna. The name Zero Forcing corresponds to bringing down the inter symbol interference (ISI) to zero in a noise free case.

4. This will be useful when ISI is significant compared to noise. For a channel with frequency response $F(f)$ the zero forcing equalizer $C(f)$ is constructed by $C(f) = 1/F(f)$.

Thus the combination of channel and equalizer gives a flat frequency response and linear phase $F(f)C(f) = 1$.

5. In reality, zero-forcing equalization does not work in most applications, for the following reasons:

1) Even though the channel impulse response has finite length, the impulse response of the equalizer needs to be infinitely long

2) At some frequencies the received signal may be weak. To compensate, the magnitude of the zero-forcing filter ("gain") grows very large. As a consequence, any noise added after the channel gets boosted by a large factor and destroys the overall signal-to-noise ratio.

Furthermore, the channel may have zeroes in its frequency response that cannot be inverted at all. (Gain * 0 still equals 0).

This second item is often the more limiting condition.

These problems are addressed in the linear MMSE equalizer [16] by making a small modification to the denominator of

$C(f)$: $(f) = 1/F(f) + k$, Where(k) is related to the channel response and the signal SNR .

Algorithm

If the channel response (or channel transfer function) for a particular channel is H(s) then the input signal is multiplied by the reciprocal of it.

This is intended to remove the effect of channel from the received signal, in particular the inter symbol interference (ISI).

The zero-forcing equalizer removes all ISI, and is ideal when the channel is noiseless. However, when the channel is noisy, the zero-forcing equalizer will amplify the noise greatly at frequencies f where the channel response H (j2πf) has a small magnitude (i.e. near zeroes of the channel) in the attempt to invert the channel completely.

A more balanced linear equalizer in this case is the minimum mean-square error equalizer, which does not usually eliminate ISI completely but instead minimizes the total power of the noise and ISI components in the output.

2.2 Related Works

As have been a major and complex issue in wireless communication, channel estimation and error detection and equalization at receiver's decoder in LTE is studied and fully defined by the 3GPP technical specification. The effect of fading and interference effects can be combated with equalizer for a MIMO system as mentioned in[2] and The BER characteristics for the various transmitting and receiving antennas simulated in MATLAB tool box and many advantages and disadvantages the system is described.

An improved vertical Bell Labs Layered Space-Time (V-BLAST) with spatial multiplexing technique used in MIMO systems is proposed in [3] with a Simulation results shows that the proposed V-BLAST, which employs SIC scheme, offers the performance very close to the optimal turbo MIMO approach, while providing stupendous improvements in computational complexity.

In [4] a developed algorithms and architectures to meet the high data rate, low complexity requirements of the future mobile communication systems are proposed.

With algorithms, architectures, and implementations for detection, channel estimation and interference mitigation in the multiple input multiple output (MIMO) orthogonal frequency division multiplexing (OFDM) receivers.

In [5] the channel estimation is performed for analysing effect of channel on signal by either inserting pilot tones into all of the subcarriers of OFDM symbols with a specific period or inserting pilot tones into each OFDM symbol, with an improved various channel performance measures based on the comparison of various channel estimation algorithms and suggest a new technique which provides better performance.

Beside the related issues mentioned in this section, we introduce the calculation of the Error Vector Magnitude (EVM) to evaluate the performance of the frame recover in RMS percentage value for each processed frame, with the visualization of each effect in the resource grid of the OFDM frame in SISO and in 2x2 MIM

CHAPTER THREE

Methodology

CHAPTER THREE

Methodology

In this chapter we introduce a detailed description of the simulation study environment we used, which includes tools and methods from MATLAB and 3GPP technical specifications together to deploy the simulation script to output the visualizations and theoretical analytics that help us in this study concerns for better understanding.

3.1 System Model

The model implementation of this study uses transmitter channel receiver simulation may be created using functions from the LTE System Toolbox and Communications System Toolbox from MATLAB R2014a.

Communications System Toolbox™ provides algorithms for designing, simulating, and analysing communications systems. These capabilities are provided as MATLAB® functions; the system toolbox enables source coding, channel coding, interleaving, modulation, equalization, synchronization, and channel modelling. You can also analyse bit error rates, generate eye and constellation diagrams, and visualize channel characteristics.

Using adaptive algorithms, you can model dynamic communications systems that use OFDM, OFDMA, and MIMO techniques. Algorithms support fixed-point data arithmetic and C or HDL code generation.

LTE System Toolbox™ provides standard-compliant functions and tools for the design, simulation, and verification of LTE and LTE-Advanced communications systems. The system toolbox accelerates LTE algorithm and physical layer (PHY) development, supports golden reference verification and conformance testing, and enables test waveform generation.

With the system toolbox, you can configure, simulate, measure, and analyse end-to-end communications links. You can also create and reuse a conformance test bench to verify that your designs, prototypes, and implementations comply with the LTE standard.

LTE downlink uses orthogonal frequency division multiplexing (OFDM) as its digital multicarrier modulation scheme. Channel estimation plays an important part in an OFDM system.

It is used for increasing the capacity of orthogonal frequency division multiple access (OFDMA) systems by improving the system performance in terms of bit error rate.

LTE assigns each antenna port a unique set of locations within a subframe to which to map reference signals. This means that no other antenna transmits data at these locations in time and frequency.

This allows channel estimation for multi-antenna configurations to be performed. These unique set of locations values are called pilot symbols.

A very detailed description of the implementation of equalization in LTE downlink is provided in this chapter, which can be represented by flowchart in the figure 3.1:

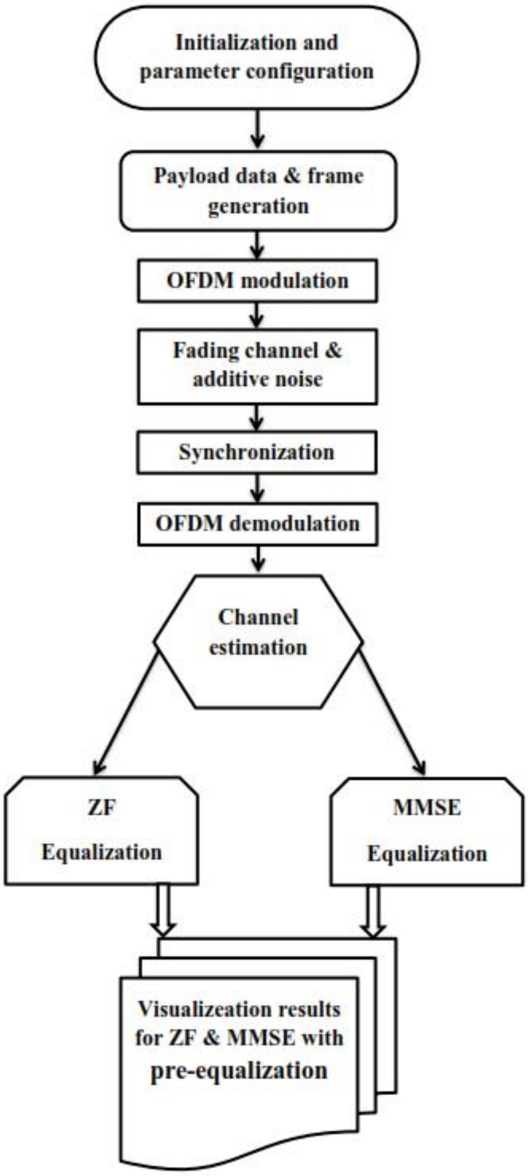

Figure 3.1: The Implementation of Equalization In LTE Downlink

3.2 Simulation Description

The simulation model of transmit and receive chains and the propagation channel model used in this study are shown in the following block diagram:

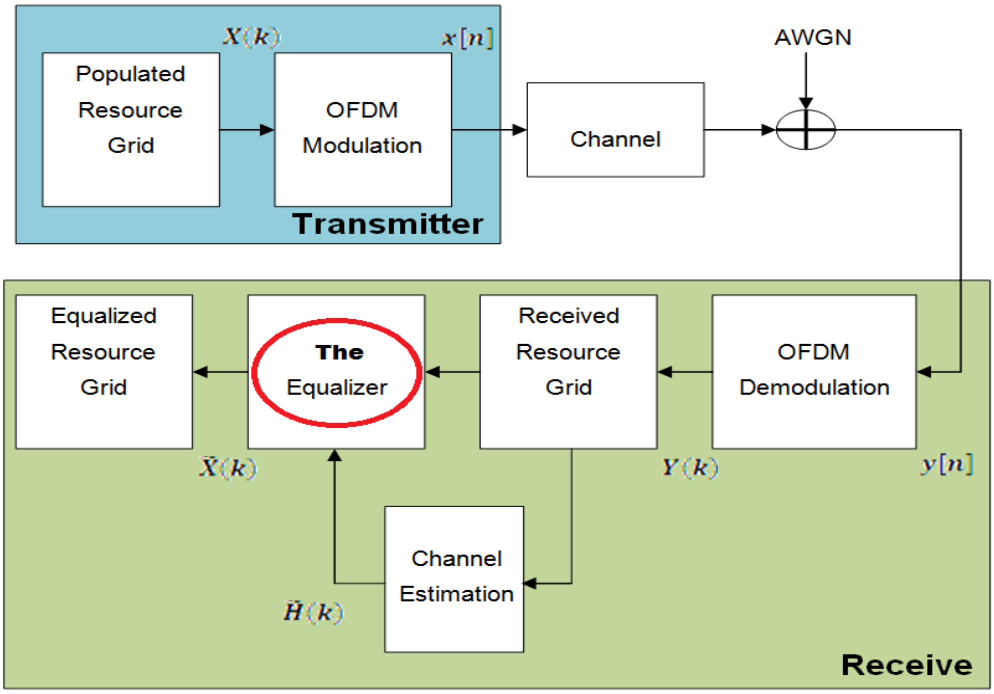

Figure 3.2: Block Diagram of the Simulation Model

From figure 3.2 the populated resource grid represents a number of subframes containing data. This grid is then OFDM modulated and passed through the model of the propagation channel.

Channel noise in the form of additive white Gaussian noise (AWGN) is added before the signal enters the receiver. Once inside the receiver the signal is OFDM demodulated and a received resource grid can be constructed.

The received resource grid contains the transmitted resource elements which have been affected by the complex channel gains and the channel noise. Using the known pilot symbols to estimate the channel, it is possible to equalize the effects of the channel and reduce the noise on the received resource grid.

The Table 3.1 below represents simulation parameter that configured in this study which includes cell configuration, SNR configuration, channel configuration, and channel estimator configuration for the model.

Table 3.1 Simulation Parameters

Parameter	**Value**
Bandwidth	1.4 MHz
Number of resource blocks	15
Sampling rate	3840000
NFFT	256
Duplex Mode	FDD
Cyclic Prefix length	Normal
Cell ID	10
Number of receive antenna	1,2
Number of transmit antenna	1,2
Desired SNR	22 dB = (12.58 Linear SNR)
Delay spread profile	EVA
Doppler frequency	120 Hz
Channel model type	Rayleigh fading
Frequency x Time averaging window in Res	9 x 9
Interpolation Type	Cubic
Modulation	QPSK

Generally, there are two cases studied, the first using a single transmitter and receiver to maintain a single resource grid handles the OFDM frame transmission and stablishing channel estimation and equalization with the two equalizers to obtain their effects.

The second case used two transmitters and receiver to maintain two frames transmission with channel specific configuration and MIMO correlation. In the two cases the error vector magnitude is calculated and discussed.

3.3 Simulation Scenario

The simulation scenario describes the generation of the first resource grid, the same process is used in the second one.

The resource grid represents the payload data carrier, the payload data is the randomly generated data to be used in the simulation, and the populated data represents the payload data in the resource grid.

The simulation script generates a frame worth of data on one antenna port. As no transport channel is created in this study the data is random bits, QPSK modulated and mapped to every symbol in a subframe.

A cell specific reference signal and primary and secondary synchronization signals are created and mapped to the subframe. 10 subframes are individually generated to create a frame.

The frame is OFDM modulated, passed through an Extended Vehicular a Model (EVA5) fading channel, additive white Gaussian noise added and demodulated.

Finally, the evaluation for each of MMSE and ZF equalizations using channel and noise estimation is applied and then the received and equalized resource grids are visualized in order to evaluate the performance of each equalizer and the advantageous one.

The operating SNR is configured in decibels by the value 22 dB which is also converted into a 12.58 Linear SNR, a fading channel with an Extended Vehicular (A) model (EVA) delay profile and 120Hz Doppler frequency is used.

Doppler frequency is required only if there is a Delay Profile set. The cell identification number is used to assign pilot symbols positions within the subframe.

The delay profiles are selected to be representative of low, medium and high delay spread environments. The resulting model parameters are defined in Table B.2.1-1 from [17] and the tapped delay line models are defined in Tables B.2.1-2, B.2.1-3 [Appendix.A.2].

The normalize delay profile power is meant by the Rayleigh fading process such that the expected value of the path gains' total power is 1.

Table 3.2: Simulation Parameters

Number of resource blocks (NRB)	Windowing samples for normal and extended cyclic prefix
6	4
15	6
25	4
50	6
75	8
100	8

The number of windowing samples is a compromise between the effective duration of cyclic prefix, and therefore the channel delay spread tolerance, and the spectral characteristics of the transmitted signal, not considering any additional FIR filtering. For a larger amount of windowing, the effective duration of the cyclic prefix is reduced but the transmitted signal spectrum has smaller out-of-band emissions.

3.5.1 Minimum Mean Square Error (MMSE)

In MIMO wireless communication, an equalizer is employed, which is a network of elements that makes an attempt to recover a signal that has suffers with an ISI or the additive noise and proves the BER characteristics and maintains a good SNR.

A Minimum Mean Square Error (MMSE) estimator is a method in which it minimizes the mean square error (MSE), which is a common measure of estimator quality. Minimum mean-square error equalizer, which does not usually eliminate ISI completely but instead, minimizes the total power of the noise and ISI components in the output. The MMSE estimator is then defined as the estimator achieving minimal MSE.

Generally, it very difficult to determine a closed form for the MMSE estimator; in these cases, one possibility is to seek the technique minimizing the MSE within a particular class, such as the class of linear estimators. The linear MMSE filter can be calculated as

$$W = (H^H H + \sigma^2 I_M)^{-1} H^H = G^{-1} H^H, \tag{3.4}$$

Where H is the channel matrix, σ^2 is the noise variance, $(.)^H$ is the complex conjugate transpose and I_M is a M × M identity matrix. The received signal vector is then filtered to obtain the equalized signals as:

$$\check{\chi}_{MMSE} = W_y \tag{3.5}$$

Hence, the LMMSE detector minimizes the mean square error between the output ($\check{\chi}_{MMSE}$) and the transmitted signal(χ).

The LMMSE filter can also be calculated using an extended channel matrix ($\underline{H} \in C^{(M+N)\times N}$) , as:

$$W = (\underline{H}^H \underline{H})^{-1} \underline{H}^H, \tag{3.6}$$

Where the extended channel matrix and received vector ($\underline{y} \in C^{(M+N)\times N}$), can be expressed as:

$$\underline{H} = [H \ \sigma I_N] \text{, } and\ \underline{y} = [y \ 0_{N\times 1}] \tag{3.7}$$

Where $0_{N\times 1}$ denotes a vector of zeros. The equalized signals can then be obtained from:

$$\check{\chi}_{MMSE} = \underline{H}^{+} \underline{y} \tag{3.8}$$

Where $(.)^+$ denotes the Moore-Penrose pseudo-inverse of the matrix [19]. This bias in the linear MMSE filter can be removed by W=BG-1HH [20] where B is a diagonal matrix with (i,i^{th}) element:

$$B_{i,i} = \frac{p_i + 1}{p_i} \tag{3.9}$$

Then the SINR on stream (i) can be calculated as:

$$p_i = \frac{1}{\sigma^2 G_{i,i}^{-1}} - 1 \tag{3.10}$$

3.5.2 Zero Forcing (ZF)

The Least Square (LS) or Zero forcing (ZF) decoder is the simplest method used for decoding. The complexity is also the least. It is described in the following equation:

$$G_k = \frac{1}{H_k} \tag{3.11}$$

)

Where H_k is assumed to be the known channel. With the ZF equalizer, the interference is forced to zero, or assumed to be zero, which in turn amplifies the noise and can be seen from the following mathematical expression for OFDM if the first received symbol is expressed as $\hat{a}$ and the second is expressed as $\hat{b}$:

$$\hat{a} = xa + yb + n, \text{ and } \hat{b} = xb + ya + n \tag{3.12}$$

The first equation shows that for the received symbol a, the interference of received symbol b is also available. So if one is to use the zero forcing of the equation above, then the noise n is amplified.

In case of MIMO detection, the ZF equalizer is given by:

$$W_{ZF} = (H^H H)^{-1} H^H = H^{\therefore} \quad (3.13)$$

)

Where, (H) is the channel matrix, and $(\cdot)^{\therefore}$ is the pseudo inverse of the matrix [19]. The ZF receiver suppresses the interference between the MIMO streams, but it enhances the noise and the performance is far from optimal.

3.6 The Error Vector Magnitude Calculation

The EVM of an LTE signal is defined as the magnitude of the difference between a complex transmitted data symbol and an ideal reference symbol, normalized to the ideal reference signal, and averaged over a 1 ms subframe. Unlike cellular standards such like WCDMA, however, the EVM of an LTE signal is measured in intervals of 180 kHz referred to as Resource Blocks. The number of Resource Blocks varies depending on the bandwidth of the LTE carrier.

For example, a 1.4 MHz LTE signal contains 6 Resource Blocks, whereas a 20 MHz LTE signal contains 100 Resource Blocks. Figure (3.7) shows the basic EVM measurement check points block diagram.

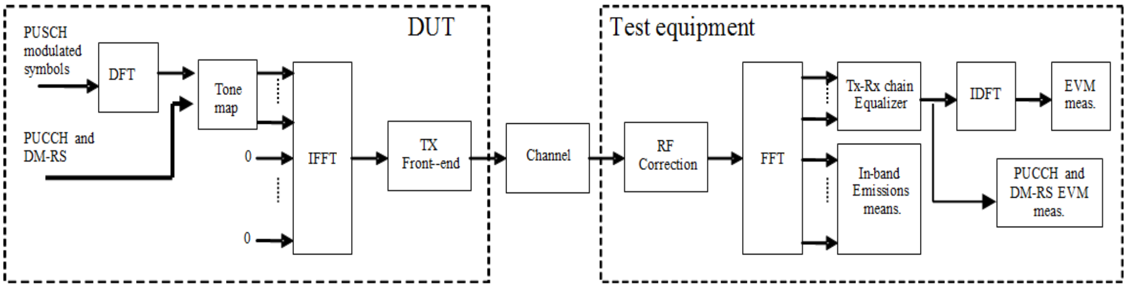

Figure 3.8: EVM Measurement Points

The EVM is the difference between the ideal waveform and the measured waveform for the allocated RB(s), the total EVM of an LTE signal is calculated as per 3GPP TS 36.104, Annex E.7 [1]:

$$EVM = \sqrt{\frac{\sum_{v \in T_m} \left| z'(v) - i(v) \right|^2}{|T_m| \cdot P_0}} \tag{3.14}$$

Where T_m is a set of $|T_m|$ modulation symbols with the considered modulation scheme being active within the measurement period, $z'(v)$ are the samples of the signal evaluated for the EVM, $i(v)$ is the ideal signal reconstructed by the measurement equipment, P_0 is the average power of the ideal signal and for normalized modulation symbols P_0 is equal to 1.

CHAPTER FOUR

RESULTS AND DISCUSSION

CHAPTER FOUR

RESULTS AND DISCUSSION

The script of system model simulation is generated with all required functions from the LTE toolbox which is generally depending on the "mw LTE library" which is the C language library includes the Standard-compliant models for the releases 8, 9, and 10 of 3GPP LTE and LTE-Advanced systems.

The system evaluation model consists of two cases, the first includes one resource grid sent from and to one single antenna, in the second two resource grids generated and each one sent from and to different antenna, so in the second we used two antennas for each receiver and transmitter.

4.2 Single Input Single Output

As all parameters are set as in the chapter 3 for system model, in the first case the generated payload data and the resource grid received after fading channel and additive noise processes is shown below in 3D visualization in figure 4.1.

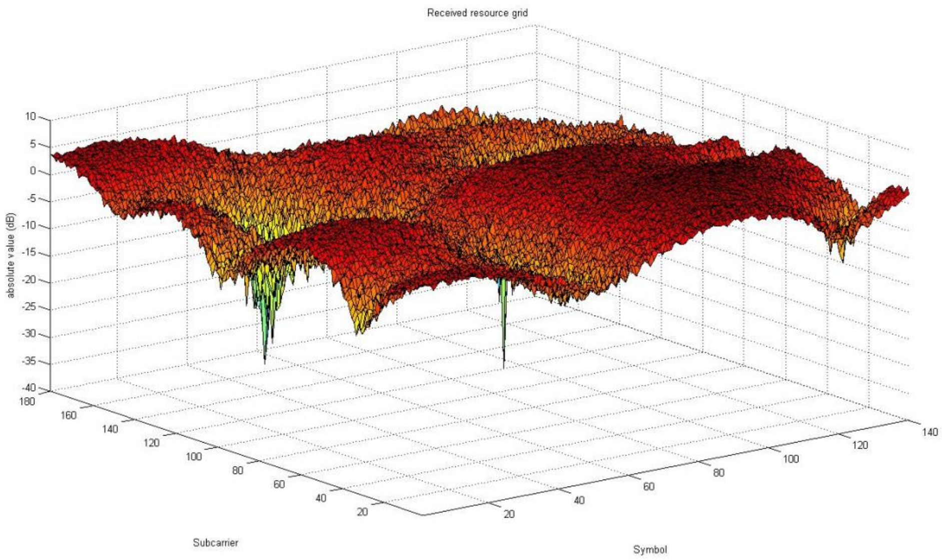

Figure 4.1 Received Resource Grid (exp1)

The resource grid generated in the transmitter was compact and flat, the figure (4.1) shows the effect of multipath fading channel and noise, the RMC percentage of EVM for the received resource grid is 20.682% to the original transmitted resource grid.

After channel estimation and equalization there is two results; one of the Zero Forcing (ZF) equalizer, which is in the figure (4.2) and the equalization effect is clear to be seen but with some distortions, the RMC percentage value of the ZF equalized signal becomes 3.941% to the original transmitted resource grid, which introduce good recovered resource grid.

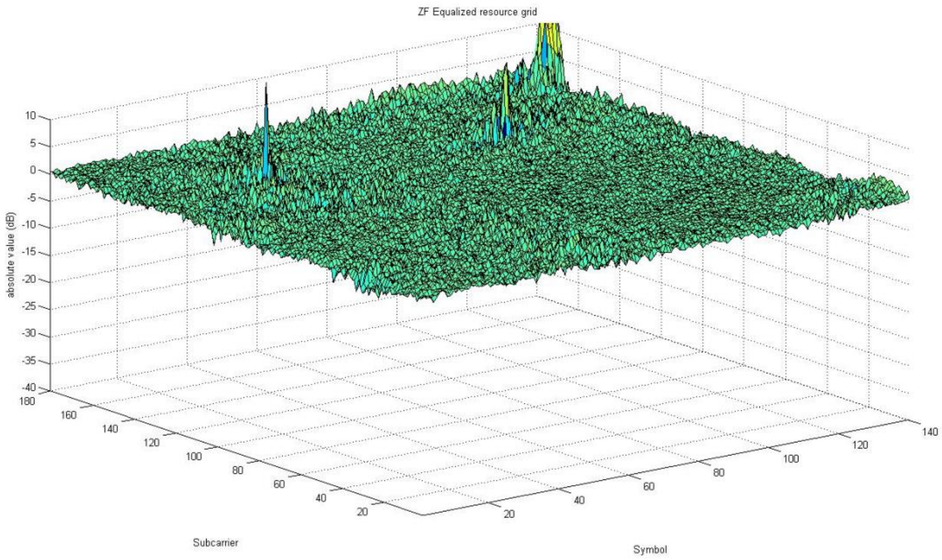

Figure 4.2: The ZF Equalized Resource Grid (exp1)

Figure 4.3 shows the Minimum Mean Square Error (MMSE) equalizer effect upon the received resource grid, which introduce better equalization as visualized and by the RMS percentage of the EVM calculation, the percentage RMS EVM of the MMSE equalized signal is 1.423% represent better performance than the ZF equalizer.

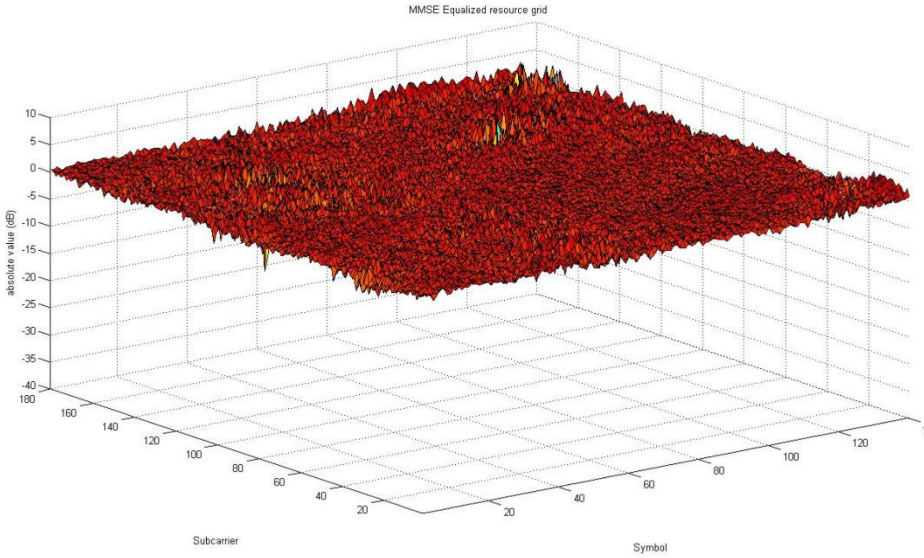

Figure 4.3: The MMSE Equalized Resource Grid (exp1)

Table 4.1 RMS EVM Percentage Summary

	Received Grid	ZF equalized	MMSE equalized
RMS EVM %	20.682%	3.941%	1.423%

In this point we realize that the traditional MMSE equalizer is more efficient than the ZF equalizer in case of Single Input Single Output (SISO).

4.3 Multiple Input Multiple Output

The second part of this study is experimenting the effect of the equalizers were two antennas are used. This part of experiment has two cases which are with and without MIMO correlation.

In the first case there is no correlation between antennas, in the second case the MIMO correlation was used is the 'High' correlation level which is applicable to tests defined in [8].

4.3.1 Without MIMO Correlation

There are two received resource grid from each antenna ports the first one shown in figure (4.4) and the second in (4.7), the equalization is done in a parallel form each resource grid is equalized alone to attempt recovering the original resource grid and to allow the calculation the difference between the equalized grid and the original grid by the EVM.

As seen in the figure (4.4) the first received grid is distorted by the effect of the multipath propagation and noise and it has 27.180% RMS EVM value which is worse than the received grid in the first case.

While the second received grid in this case in figure (4.7) has 35.002% RMS EVM which is worse than the first received grid.

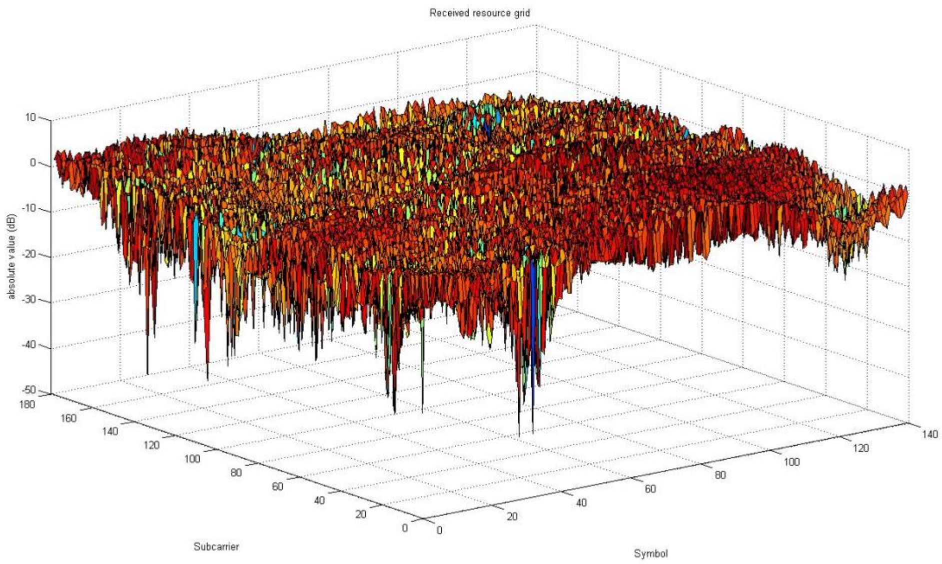

Figure 4.4: Resource Grid Received in the First Port (27.180%)

The ZF equalization of the first received grid is shown below in figure (4.5) and it has 7.525% RMS EVM, which is much better than its received grid by approximately 20%.

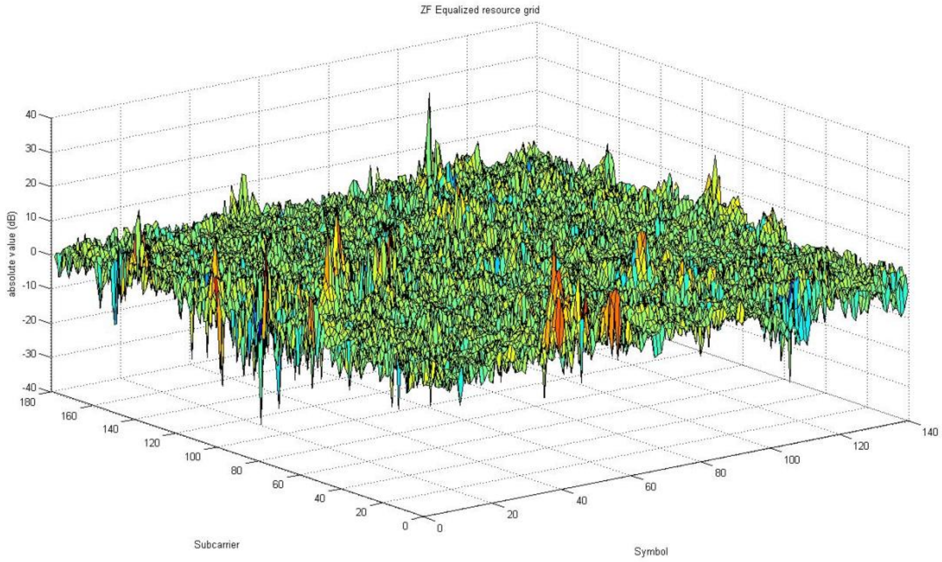

Figure 4.5: ZF Equalization of the Resource Grid Received in the First Port (7.525%)

The MMSE equalization of the first received grid is shown in figure (4.6) and it has 16.252% RMS EVM, which is just better than its received grid by approximately 10%.

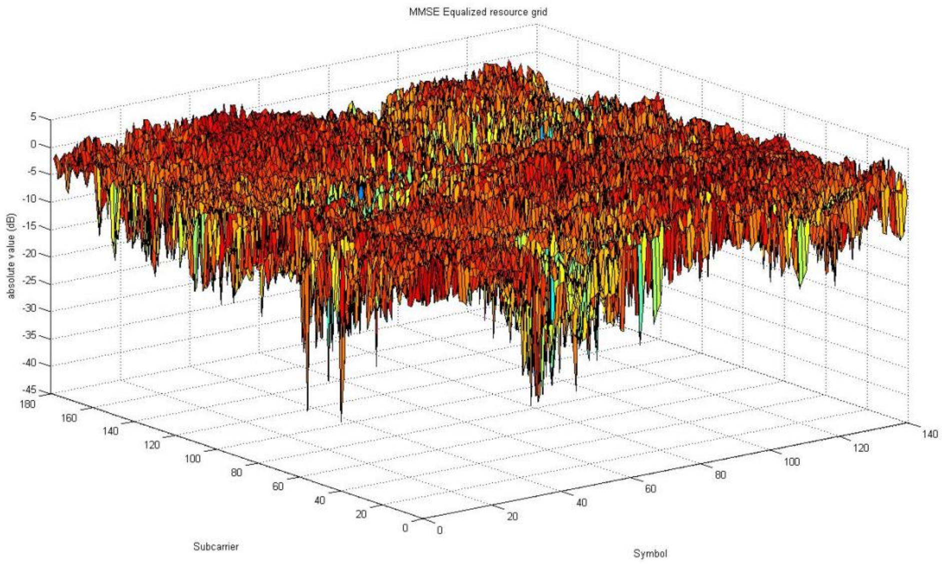

Figure 4.6 MMSE Equalization of the Resource Grid Received in the First Port (16.252%)

The figure (4.7) shows the second received grid and as expected it’s also distorted by the effect of the multipath propagation and noise and it has 35.002% RMS EVM value which is worse than the received grid in the first port.

The first port received grid in this case was in figure (4.4) were it has 27.180% RMS EVM.

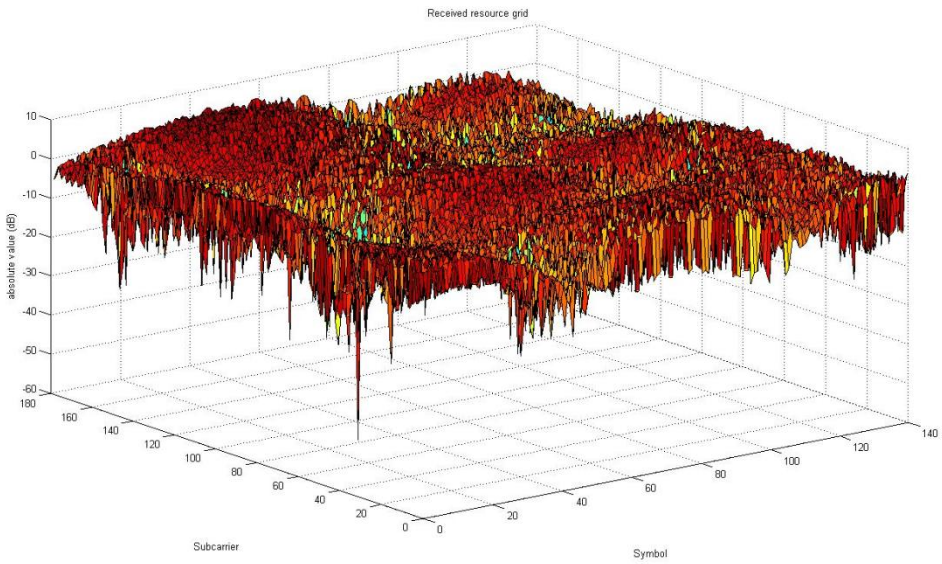

Figure 4.7: Resource Grid Received in the Second Port (35.002%)

The ZF equalization of the second received grid is shown in figure (4.8) and it has 27.498% RMS EVM, which is much worse than the ZF equalization in the first one.

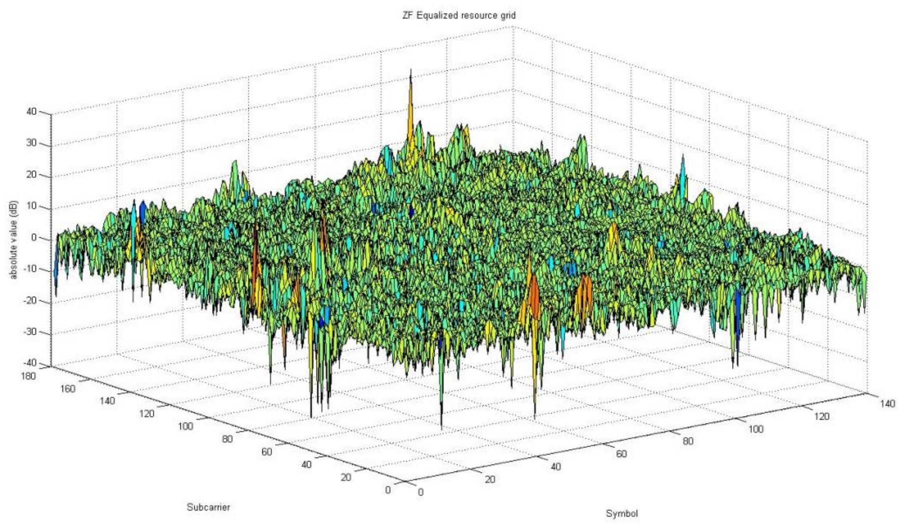

Figure 4.8: ZF Equalization of the Resource Grid Received in the Second Port (27.498%)

The MMSE equalization of the second received grid is shown in figure (4.9) and it has 48.143% RMS EVM, which is not acceptable were the original grid RMS EVM was 35.002%, so in this case the MMSE equalization becomes inefficient.

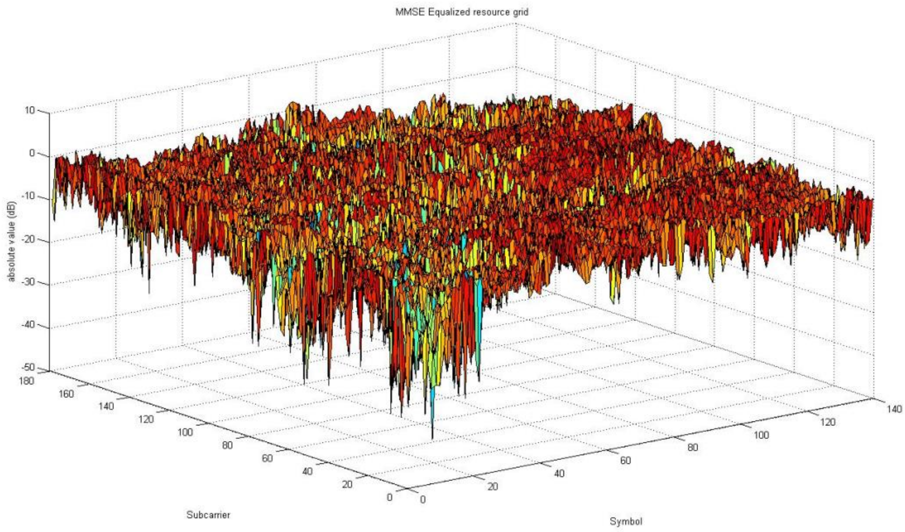

Figure 4.9: MMSE Equalization of the Resource Grid Received in the Second Port (48.143%)

4.3.1 With MIMO Correlation

This is the second part of this case, introduces the usage of MIMO correlation in the channel, in order to investigate the effect we follow the same methodology used before in the first case, so the figure (4.10) shows the first received grid and as expected it's also distorted by the effect of the multipath propagation and noise and it has 24.934% RMS EVM value which is better than the first received grid in the first case.

The first received grid in this first case was in figure (4.4) were it had 27.180% RMS EVM.

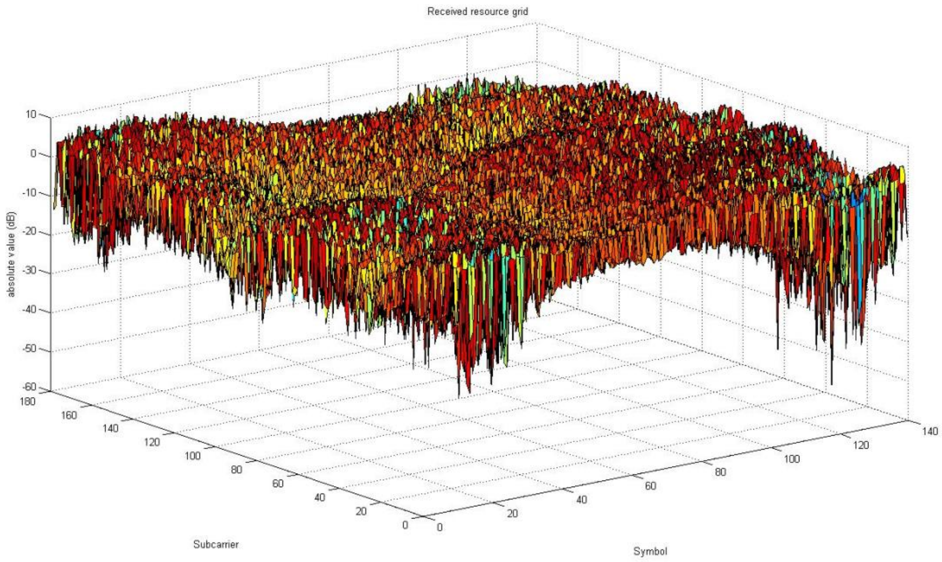

Figure 4.10: Resource Grid Received in the First Port (exp2:case2) (24.934%)

The ZF equalization of the first received grid above is shown in figure (4.11) and it has 13.671% RMS EVM, its good when compared with it's the received grid, but it's worse than the ZF equalized without MIMO cancellation, so in this point we realize that the channel MIMO cancellation doesn't affect the performance of the equalizer, it's just affect the received grid because that the RMS EVM of the same grid without MIMO correlation was 27.180% and with MIMO correlation becomes 24.934%.

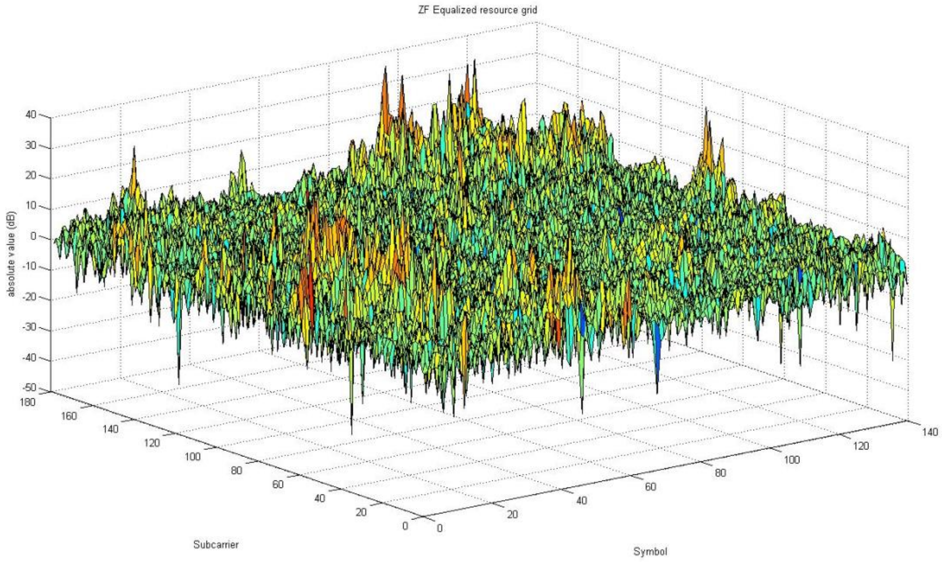

Figure 4.11: ZF Equalized Resource Grid Received in the First Port (exp2:case2) (13.671%)

The MMSE equalization of the first received grid in this case is shown below in figure (4.12) and it has 29.429% RMS EVM, which is also worse than its received grid.

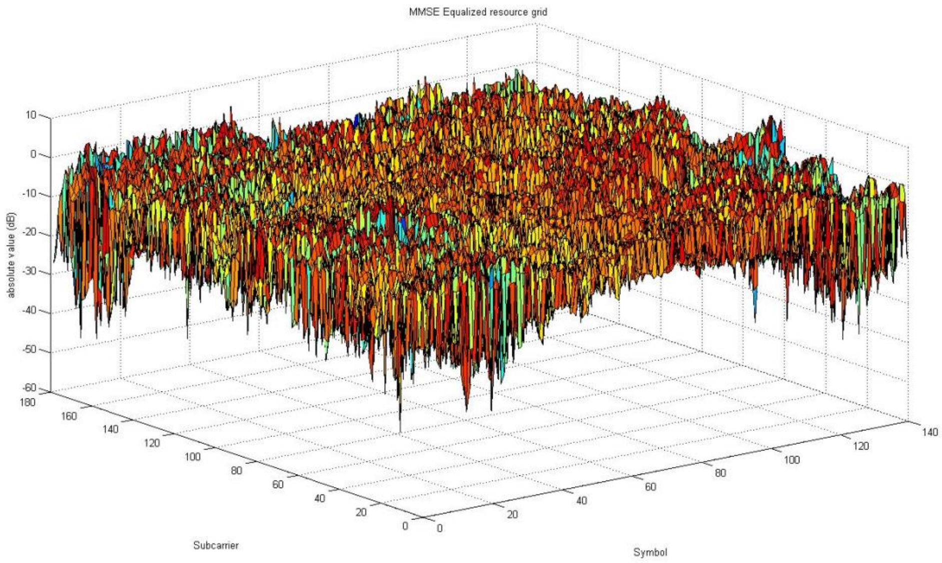

Figure 4.12: MMSE Equalized Resource Grid Received in the First Port (exp2:case2) (29.429%)

The figure (4.13) shows the second received grid of this case and it has 30.534% RMS EVM value which is better than its equivalent received grid in the first case. The second received grid in the first case was in figure (4.7) were it has 35.002% RMS EVM. This ensures the point of the channel MIMO correlation mentioned before.

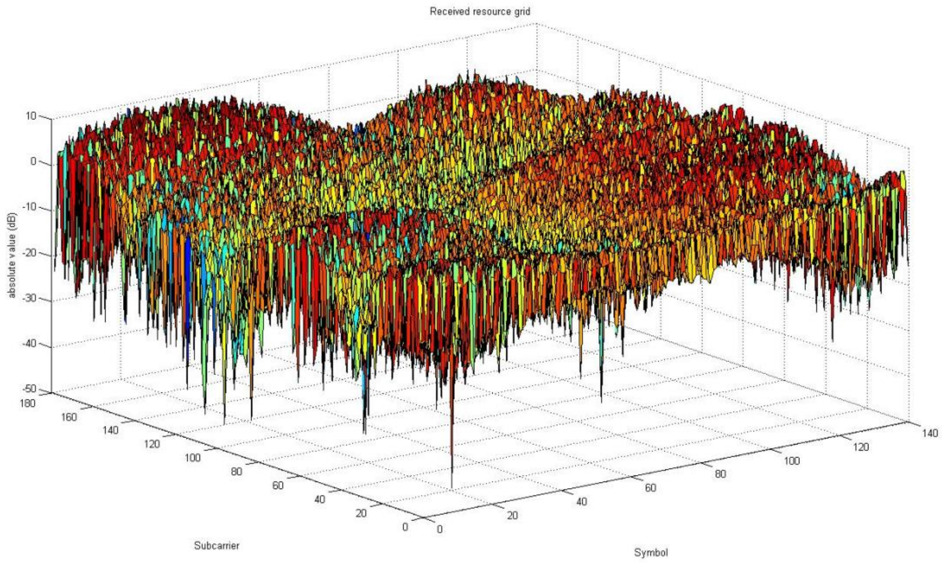

Figure 4.13: Resource Grid Received in the Second Port (exp2: case2) (30.534%)

The ZF equalization of the second received grid above is shown in figure (4.14) and it has 17.684% RMS EVM, and it’s good by comparing with the original received grid.

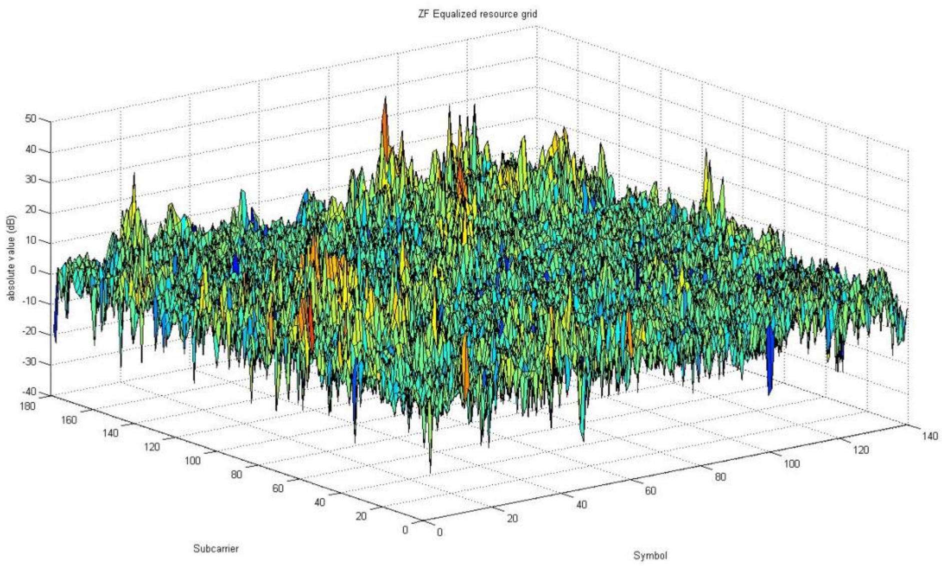

Figure 4.14: ZF Equalized Resource Grid Received in the Second Port (exp2:case2) (17.684%)

The MMSE equalization of the second received grid in this case is shown below in figure (4.15) and it has 33.965% RMS EVM, which is ensures that the traditional MMSE equalization is inefficient in case of MIMO.

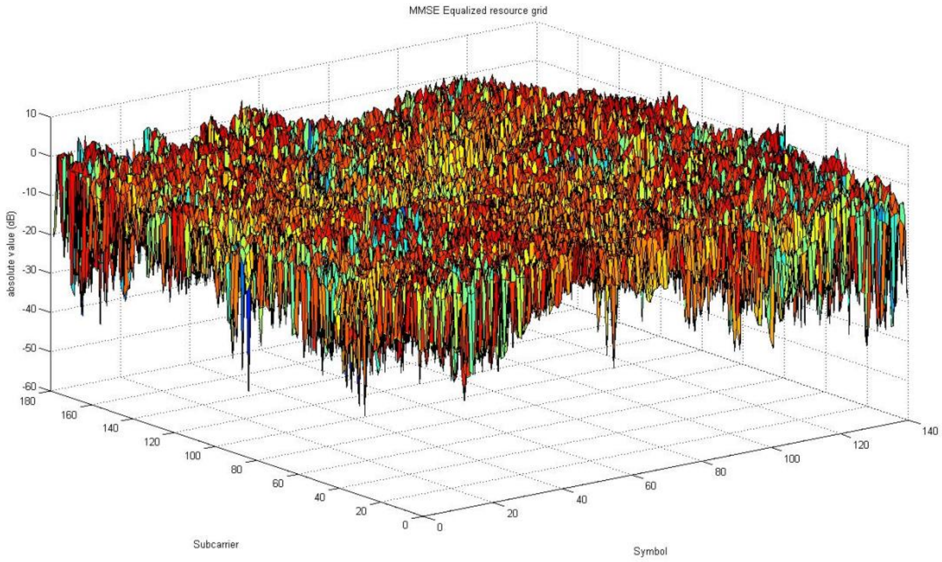

Figure 4.15: MMSE Equalized Resource Grid Received in the Second Port (exp2 case2) (33.965%)

Table 4.2: Percentage RMS of EVM value in MIMO

	With MIMO Correlation			Without MIMO Correlation		
	Received	ZF	MMSE	Received	ZF	MMSE
First Grid	24.934%	13.671%	29.429%	27.180%	7.525%	16.252%
Second Grid	30.534%	17.684%	33.965%	35.002%	27.498%	48.143%

Table (4.2) summarizes the equalization of MIMO with and without channel MIMO correlation, realization of this table is that the effect of the channel MIMO correlation in the received grids it's obvious, and the traditional MMSE equalizer becomes inefficient in case of MIMO, while the ZF equalizer still effectible and shows good results.

BER Results:

As development of this work a new physical layer scenarios is developed to deploy physical layer characteristics of the network and gather statistics for Bit Error Rate BER and the results are taken for different cases, here figure (4-16) below shows the BER for ZF equalizer in different cases such as SISO, MIMO, and MIMO with correlations, and the realization of this result is expectable from previous results as SISO case has the highest BER records while MIMO has medium BER values and MIMO with correlation has the lowest BER records.

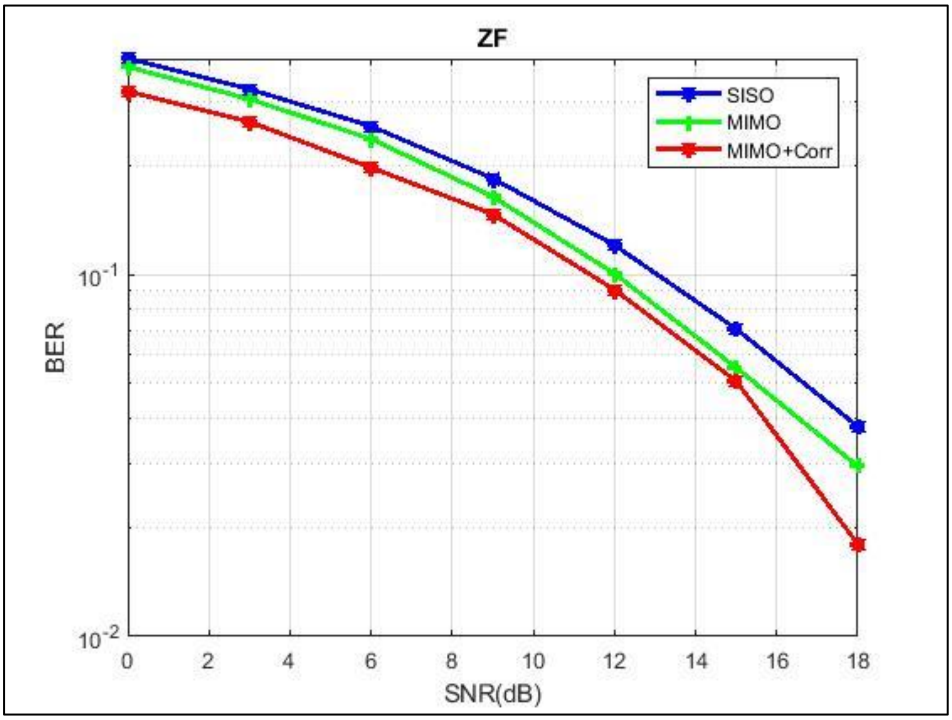

Figure 4.16: comparison of ZF with SISO, MIMO, and MIMO with correlation

In case of MMSE also same results is achieved that higher BER is figured in SISO case and MIMO with correlation has the lowest BER value as shown in figure (4-17) but here BER generally is less than ZF in previous result.

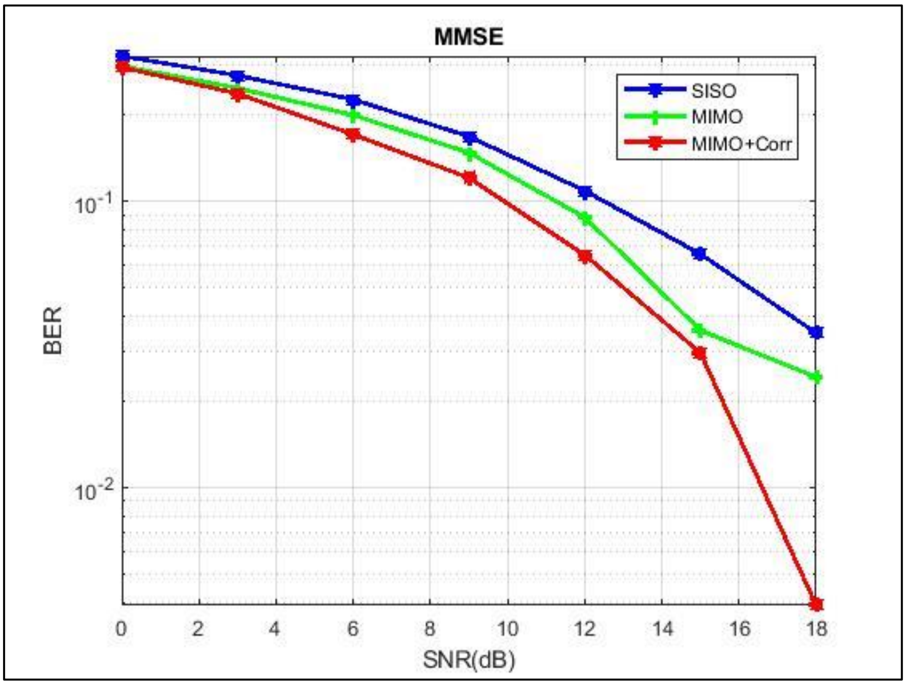

Figure 4.17: comparison of MMSE with SISO, MIMO, and MIMO with correlation

As further analysis a comparison between ZF and MMSE is also performed in term of BER as shown in figure (4-18) below, which shows ZF vs MMSE in case of SISO and the result shows that MMSE has outperformed the ZF equalizer by achieving lower BER values.

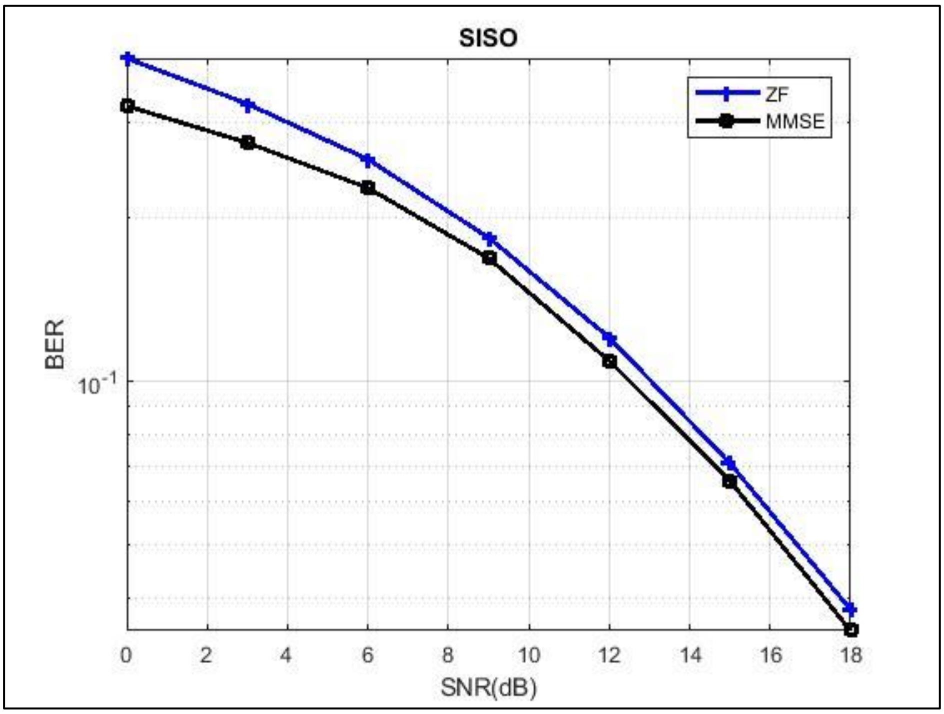

Figure 4.18: comparison of MMSE vs ZF in case of SISO

The final result is shown in figure (4-19) below shows both equalizers BER in case of MIMO with and without correlation, and it clear to say that MMSE with correlation has the best lowest BER values and ZF without correlation achieved the worst highest records, while ZF with correlation has better performance than MMSE without correlation as in MIMO case.

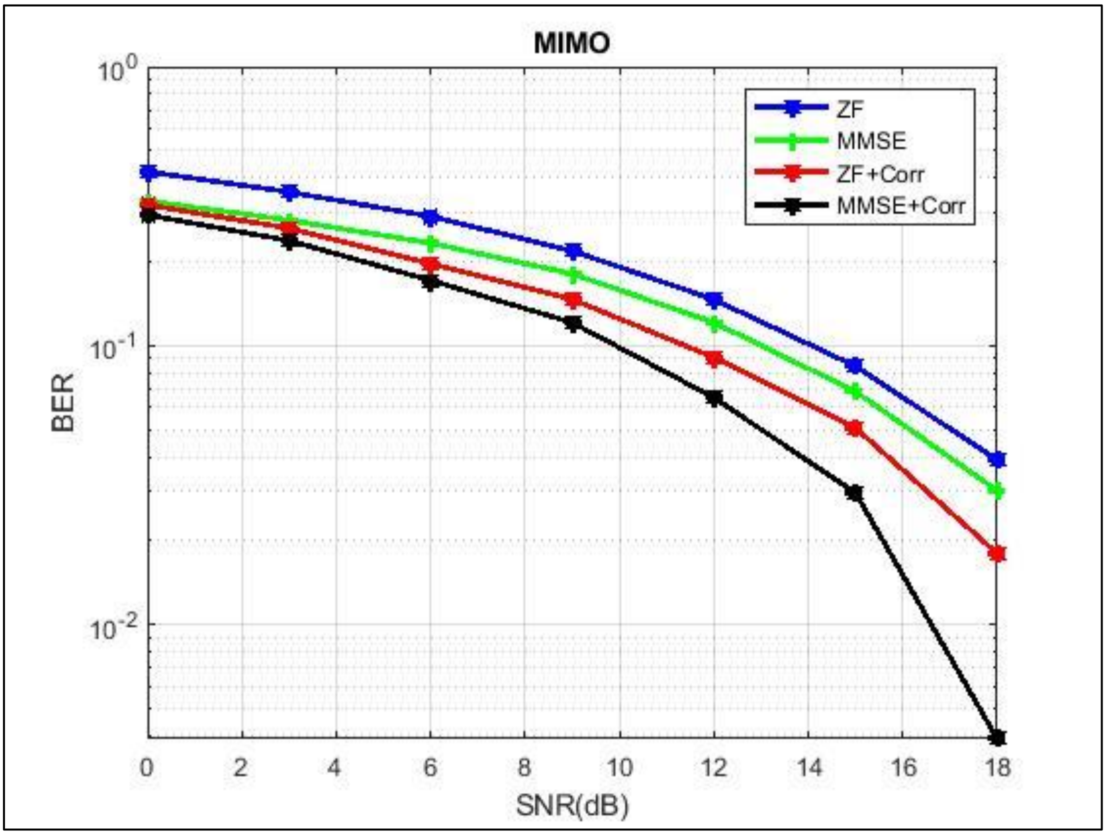

Figure 4.18: comparison of MMSE vs ZF in case of MIMO with & without correlation.

CHAPTER FIVE

CONCLUSION AND RECOMMENDATIONS

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CONCLUSION AND RECOMMENDATIONS

5.1 Conclusion

While performing this study many aspects are realized and some problems are occurs, by the end of the study we figure out that OFDM is sensitive to carrier frequency offset, which resulting in Inter Carrier Interference, also OFDM exhibits a high resilience against the ISI. Incorporating channel coding techniques into OFDM systems, which results in Coded OFDM, allows us to maintain robustness against frequency-selective fading channels, where busty errors are encountered at specific subcarriers in the FD.

In theoretical manner, in classic single carrier systems, the distortion of a signal introduced by multi-path propagation in which one symbol interferes with subsequent symbols this distortion known as Inter Symbol Interference (ISI). Complex equalizers have to be employed at the receiver for the sake of mitigating this ISI.

There are two familiar equalizers are verified by this study, the Zero forcing (ZF) and the Minimum Mean Square Error (MMSE), the ZF is meant by bushing the error value to zero, the MMSE minimize the mean square error value.

From the simulation we realize that the MMSE has better results than the ZF equalizer in case of single antennas configuration SISO, but the ZF equalization has the advantage in case of multiple antennas MIMO, the channel with MIMO correlation maintains better received grid than without MIMO correlation.

5.2 Recommendations

As a development of this work we suggest that the use of ZF in implementation of MIMO receiver because its simple design and effective results introduce, and for SISO the usage of the MMSE is preferred which introduces better results in linearity processing.

And also we suggest that the configuration of MIMO correlation as specified for channels to be used, for better signal recovering.

- **Recommendation for Future work**

As a development of this work, an as the rapid development of LTE standard the MMSE is redefined for MIMO but with more complexity in addition to more specific channels configuration, so we suggest to introduce an advanced study of this work with the dedicated MMSE channel estimation and equalization for MIMO with transmit diversity transmission mode, and also as improvement of channel MIMO correlation we suggest to introduce the custom configuration for channel MIMO correlation matrices.

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APPENDIX (A)

A.1 Static propagation condition

For 1 port transmission the channel matrix is defined in the frequency domain by

$$\mathbf{H} = \begin{pmatrix} 1 \\ 1 \end{pmatrix}.$$

For 2 port transmission the channel matrix is defined in the frequency domain by

$$\mathbf{H} = \begin{pmatrix} 1 & j \\ 1 & -j \end{pmatrix}.$$

For 4 port transmission the channel matrix is defined in the frequency domain by

$$\mathbf{H} = \begin{bmatrix} 1 & 1 & j & j \\ 1 & 1 & -j & -j \end{bmatrix}$$

For 8 port transmission the channel matrix is defined in the frequency domain by

$$\mathbf{H} = \begin{bmatrix} 1 & 1 & 1 & 1 & j & j & j & j \\ 1 & 1 & 1 & 1 & -j & -j & -j & -j \end{bmatrix}$$

A .2: Multi-path fading propagation conditions

Table A.2.1-1 Delay profiles for E-UTRA channel models

Model	Number of channel taps	Delay spread (r.m.s.)	Maximum excess tap delay (span)
Extended Pedestrian A (EPA)	7	45 ns	410 ns
Extended Vehicular A model (EVA)	9	357 ns	2510 ns
Extended Typical Urban model (ETU)	9	991 ns	5000 ns

Table A.2-2 Extended Vehicular A model (EVA)

Excess tap delay [ns]	Relative power [dB]

0	0.0
30	-1.5
150	-1.4
310	-3.6
370	-0.6
710	-9.1
1090	-7.0
1730	-12.0
2510	-16.9

APPENDIX (B)

B.1: Timing offset

As a result of using a cyclic prefix, there is a range of $\Delta\tilde{t}$, which, at least in the case of perfect Tx signal quality, would give close to minimum error vector magnitude. As a first order approximation, that range should be equal to the length of the cyclic prefix. Any time domain windowing or FIR pulse shaping applied by the transmitter reduces the $\Delta\tilde{t}$ range within which the error vector is close to its minimum.

B.2: Window length

The window length W affects the measured EVM, and is expressed as a function of the configured cyclic prefix length. In the case where equalization is present, as with frequency domain EVM computation, the effect of FIR is reduced. This is because the equalization can correct most of the linear distortion introduced by the FIR. However, the time domain windowing effect can't be removed.

B.3: Window length for normal CP

The table below specifies the EVM window length at channel bandwidths 1.4, 3, 5, 10, 15, 20 MHz, for normal CP. The nominal window length for 3 MHz is rounded down one sample to allow the window to be centred on the symbol.

Table B.3-1 EVM window length for normal CP

Channel Bandwidth MHz	Cyclic prefix length[1] N_{cp} for symbol 0	Cyclic prefix length[1] N_{cp} for symbols 1 to 6	Nominal FFT size	Cyclic prefix for symbols 1 to 6 in FFT samples	EVM window length W in FFT sampl	Ratio of W to CP for symbols 1 to 6 [2]

					es	
1.4	160	144	128	9	5	55.6
3			256	18	12	66.7
5			512	36	32	88.9
10			1024	72	66	91.7
15			1536	108	102	94.4
20			2048	144	136	94.4
Note 1: The unit is number of samples, sampling rate of 30.72MHz is assumed. Note 2: These percentages are informative and apply to symbols 1 through 6. Symbol 0 has a longer CP and therefore a lower percentage.						

B.4: Window length for Extended CP

The table below specifies the EVM window length at channel bandwidths 1.4, 3, 5, 10, 15, 20 MHz, for extended CP. The nominal window lengths for 3 MHz and 15 MHz are rounded down one sample to allow the window to be centered on the symbol.

Table B.4-1 EVM window length for extended CP

Channel Bandwidth MHz	Cyclic prefix length[1] N_{cp}	Nominal FFT size	Cyclic prefix in FFT samples	EVM window length W in FFT samples	Ratio of W to CP [2]
1.4	512	128	32	28	87.5
3		256	64	58	90.6
5		512	128	124	96.9

10		1024	256	250	97.4
15		1536	384	374	97.4
20		2048	512	504	98.4
Note 1: The unit is number of samples, sampling rate of 30.72MHz is assumed. Note 2: These percentages are informative					

APPENDIX (C)

C.1: Receiver antenna capability

The performance requirements are based on UE(s) that utilize one or more antenna receivers. For all test cases, the SNR is defined as:

$$SNR = \frac{\sum_{j=1}^{N_{RX}} \hat{E}_s^{(j)}}{\sum_{j=1}^{N_{RX}} N_{oc}^{(j)}}$$

Where N_{RX} denotes the number of receiver antenna connectors and the superscript receiver antenna connector j. The above SNR definition assumes that the REs are not precoded. The SNR definition does not account for any gain which can be associated to the precoding operation. The relative power of physical channels transmitted is defined in Table C.3.2-1. The SNR requirement applies for the UE categories and CA capabilities given for each test. For enhanced performance requirements type A, the SINR is defined as

$$SINR = \frac{\sum_{j=1}^{N_{RX}} \hat{E}_s^{(j)}}{\sum_{j=1}^{N_{RX}} N_{oc}^{(j)'}}$$

where N_{RX} denotes the number of reciver antenna connectors and the superscript receiver antenna connector j. The above SINR definition assumes that the REs are not precoded. The SINR definition does not account for any gain which can be associated to the precoding operation. The relative power of physical channels transmitted is defined in Table C.3.2-1. The SINR requirement applies for the UE categories given for each test.

For the performance requirements specified in this clause, it is assumed that N_{RX}=2 unless otherwise stated.