ADAPTIVE INTERFERENCE CANCELLATION FOR FAST FADING CHANNEL

A Thesis

Submitted to the College of Engineering

of Al - Nahrain University in a Partial Fulfillment

of the Requirements for the Degree of

Master of Science

in

Electronic & Communications Engineering

by

Mohammed Kadhum Hmood
(B.Sc. 2009)

Jamad Alakhar 1437
March 2016

Committee Certificate

We the Examining Committee , after reading this thesis " **Adaptive Interference Cancellation For Fast Fading Channel**" and examining the student "**Mohammed Kadhum Hmood**" in its content, find it is adequate as a thesis for the degree of **Master of Science in Electronic and Communication Engineering.**

Assist. Prof. Dr. Aied K. Mohammed

Chairman

/ /2016

Dr. Bayan Mehdi Sabbar

Member

14 /4/ 2016

Dr. Mohammed Hussein Ali

Member

14 /4/ 2016

M.J.A
14-4-2016

Assist. Prof. Dr. Manal Jamil Al-Kindi

Supervisor and member

14 /4/ 2016

Approved by the College of Engineering /Al-Nahrain University

Assist. Prof. Dr. Jassim Abbood Abbas Aldabbagh

Dean, College of Engineering/Al-Nahrain University

/ /2016

Certification

I certify that the thesis **"Adaptive Noise Cancellation for Fast Fading Channel"** was prepared by Mohammed Kadhum Hmood under my supervision at Al - Nahrain University/ Collage of Engineering in a partial fulfillment of requirements for the degree of **Master of Science in Electronic & Communications Engineering**.

Supervisor

Assist. Prof. Dr. Manal Jamil Alkindi

In view of the available recommendation, I forward this thesis for debate by the Examining Committee.

Assist. Prof. Dr. Manal Jamil Alkindi

(Head, Department of Electronic and Communication Engineering)

ABSTRACT

One of the big problems in wireless communication channels is the multipath fading. Almost all environments suffer for this multipath which is caused when the receiver receive more than one signal path because of diffraction, reflection, shadowing and scattering. Each path has different properties which cause more complexity in the receiver end to recover the transmitted signal especially in the case of fast fading channel when we have fast varying time domain. In a typical orthogonal frequency division multiplexing (OFDM) broadband wireless communication system, a guard interval and cyclic prefix are inserted to decrease the effecting of inter-symbol interference and the inter-carrier interference. The superimposed training sequence is one of the technical methods used to increase the transmission efficiency and decreases the effect of inter-symbol interference (ISI). The superimposed training sequence is the random data added the transmitted frame and it is stored in the receiver for channel estimation and equalization process.

In this thesis, the superimposed training sequence STS will be imposed on OFDM system. The superimposed training system (STS) is done by two methods: the first one is an algebraic sum of training sequence with the OFDM transmitted frame and the second one uses matrix concatenate in the transmitter to add the training sequence with OFDM frame and using the adaptive equalizer in the receiver to improve the received signal .

Two types of equalizers are used in our simulation which are; the LMS and the RLS equalizers and the results of two equalizers were compared between them. The STS with RLS equalizer gives minimum bit error rate (BER) and better system stability and performance compared to LMS equalizer. The modulation schemas

which are in the communication system are; (BPSK, QPSK and 16 QAM). The OFDM system parameters are chosen according to IEEE 802.11a standards.

List of Contents

List of Abbreviations

AWGN	Additive White Gaussian Noise
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
CDMA	Code Division Multiple Access
DFE	Decision Feedback Equalizers
DVB	Digital Video Broadcasting
FFT	Fast Fourier Transform
GI	Guard Interval
GSM	Global System for Mobile Communication
ICI	Inter Carrier Interference
IFFT	Inverse Fast Fourier Transform
ISI	Inter-Symbol Interference
LAN	Local Area Network
LMS	Least Mean Square
MMSE	Minimum Mean Square Error
OFDM	Orthogonal Frequency Division Multiplexing
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RLS	Recursive Least Mean Square
SER	Symbol Error Rate
STS	Superimposed training sequence
TS	Training Sequence
WLAN	Wireless Local Area Network
ZF	Zero Forcing

List of Symbols

x_σ	zero-mean Gaussian random variable
σ	standard deviation
d_0	reference distance
$\overline{L}p\ (d)$	average path loss
r	envelope amplitude of the received signal
$\bar{s}_k$	input data
c_k	training sequence
F^H	complex conjugate transpose
$w(k)$	frequency domain Window function
u	input vector in LMS algorithm
e	error signal
H_k	the channel matrix
T_ρ	the channel coherence time
N_{STM}	the total number of transmitted symbols
T_{sym}	the OFDM symbol time
R	the overall coding rate
N_{data}	the number of used data subcarriers
C_m	normalized factor
rms	the delay spread
f_d	the maximum Doppler shift
τ	time delay
P	power gain
f_s	symbol Rate
λ	Forget Factor

List of Figures, Charts and Flow Charts

List of Tables

CHAPTER ONE

Introduction

1.1 General background

The transmission channel in communication systems is defined as the medium that the transfers information to be sent from transmitter end to the receiver end.

The perfect situation can be obtained when the signal is received without any distortion or losing the transmitted signal. All the interested people and researchers work to reach this situation, but until now this target is considered as difficult to achieve because the difficulties that appear in the transmission medium.

The signal propagation can be divided into two types, direct path (LOS signal) and multipath propagation. The direct path occurs when there is straight path between the transmitter antenna and receiver antenna, the case is defined in communication system by (line of site) so no interference happens in the transmitted frame when passing through channel [1].

The multipath propagation occurs when the transmission medium has obstacles which affect the transmitted frames like buildings, trees, cars and long distance.

In order to design any communication system, the choice of the transmission channel takes in consideration the problems can be caused by these obstacles like attenuation, absorption, Faraday rotation, spherical spreading scintillation, polarization dependence, motion and fading, delay spread, reflection losses, dispersion ,angular spread, Doppler spread, and interference.

In this thesis the fading will be the main object to be define analysis of all factors that cause the fading how it can reduce the effect of signal fading due to channel.

Due to multipath problems, the received signal is fluctuated when it is transferred through the multipath medium so the receiver antenna will receive many copies from the transmitted signal but with different power gain and time delay, these several copies added constructively or destructively depending on the nature of received signal.

Fading can be divided into two parts: fast fading and slow fading, slow fading happens when the signal transfers in channel by slow fluctuation or variation of the signal. This slow fluctuation is caused by scattering or reflection from large object like mountains, trees and buildings [1].

Fast fading happens when the signal transfers on channel by rapid fluctuation over short distance. Fast fading is caused by scattering from moving objects like cares or complex geographical terrain. The Rayleigh fading is an example of this type of fading when there is no straight path (line of sight) between transmitter and receiver. Additionally, fading can be defined as flat or frequency selective fading. In order to reduce the effect of fading on the transmitted signal, many solutions can be considered to reduce this effect, one of these solutions uses the equalization algorithms to recover the received signal and achieve better BER.

Equalization process is widely used in wireless communication systems in order to lower the effects of the channel distortion. Many researchers have proposed equalizers for different modulations techniques. In this research the BER performance of different modulations schemas with the LMS and RLS equalizers is compared in terms of the frequency selective Rayleigh fading channel.

The aim of equalizers is to decrease the effect of ISI caused by fast fading and noise from the multipath wireless networks and mobile communications channels. The equalizers may be broadly classified as linear and non-linear equalizer and used to ease the demodulation process at the wireless receivers. In wireless multipath communication Rayleigh fading is a common phenomenon which exists due to reflection of waves, diffraction, and scattering of the transmitted waves from the large physical structures such as buildings or mountains as shown in Figure 1-1. With the increasing number of wireless users the higher size modulation methods are frequently adopted thus you need to compare the performance of modulation techniques over fading environment. In the current work performance of various linear equalizers is evaluated in terms of the frequency selective fading channels.

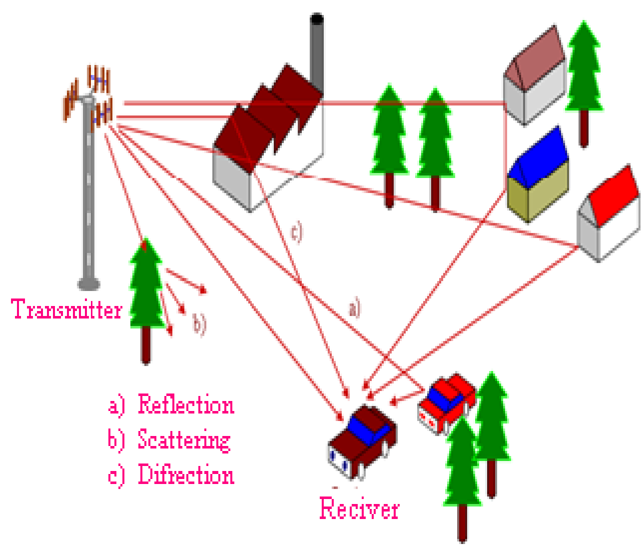


Figure 1-1 Reasons for fading in multipath wireless channels [2]

Several types of equalizer have been used in wireless communication system according to the nature of transmitted data, the characteristics of channel, adaptive noise and the algorithm that is used in receiver side to improve the performance of received signal.

1.2 Main equalization techniques.

Zero Forcing Equalizer is a linear equalization algorithm used in communication systems, which upset the frequency response of the received signal. The expression of Zero Forcing comes from dropping the ISI to zero in a noise environment. This dropping will give high advantage when ISI is highest compared to noise. [3]

A minimum mean square error (MMSE). The main feature of MMSE equalizer is that it does not usually eliminate ISI completely but 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. In many cases, it is not possible to determine a closed form of 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.

Blind equalization is essentially blind de-convolution applied to digital communications. The main advantage of blind equalizer is estimation of the equalizer filter, more than that of the channel impulse response which will be estimated by blind equalizer itself. After estimation process is completed then the result is compared with the received signal to produce estimation of the transmitted information signal [4].

The LMS algorithm method has a low complexity and it is globally convergent if the training values are given correctly. If the training sequence or the stored symbols are not correct, it does not converge. RLS algorithm has better convergence characteristics than the LMS algorithm. But, it has higher computational complexity than LMS algorithm. The general RLS algorithm's complexity grows with (N^2) where N is the number of equalizer coefficients.

There are also RLS algorithms that have computational complexities which grow linearly with the increasing number of equalizer coefficients [2].

Decision Feedback Equalizer (DFE): There are two major categories of a channel equalizer; linear and non-linear. Non-linear equalization is needed when the channel distortion is too severe for the linear equalizer to mitigate the channel impairments. An example of a linear equalizer is a zero-forcing equalizer (ZFE) and as the name implies, it forces ISI to become zero for every symbol decision. A zero-forcing equalizer enhances noise and results in performance degradation. On the other hand, a Minimizes Mean Square Error-Linear Equalizer (MMSE-LE), minimizes the error between the received symbol and the transmitted symbol without enhancing the noise. Although MMSE-LE has better performance than ZFE, its performance is not enough for channels with severe ISI. An obvious choice for channels with severe ISI is a non-linear equalizer. [5]

From the above summary of the equalizer role in wireless communication systems and the widely used to migrate the receiving signal for the adaptive noise and the inter symbol interference ISI and distortion caused by the fading channel and multipath, The main advantages of equalizing the received signal before demodulation techniques are[1]:

i) Reduction in Inter symbol Interference (ISI).

ii) Reduction in Inter Carrier Interference (ICI).

iii) Reduction in noise.

Over More and complementing the using of equalizer on the receiver side and to decrease the inter symbol interference between the transmitted frames many researchers employed some techniques to improve the equalizer work and make the convergence and the performance faster and more stably.

1.3 Main Methods of Channel Estimation.

1- Pilot sequence was employed for equalization purposes by adding PN sequence to the OFDM frame according to some standards to arrange the OFDM frame to insert the pilots sequence between the OFDM slots and demultiplxing in the receiver is used to move this pilot for OFDM frame. The main advantage of pilots sequence is to support the channel estimation work in receiver to estimate the channel coefficients.

Some researchers used the superimposed pilot sequence to improve the channel estimation process by adding a lot of pilots to the OFDM frame to decrease ISI [5].

2- Conventional training sequence is considered as one technique for channel estimation and equalization by adding random data to the OFDM frame in the transmitter and using equalizer like LS, LMS and RLS to remove this training from the OFDM frame, the main benefit of training sequence is to support the channel estimation and estimate the channel coefficients by storing the same training sequence in the receiver [6].

3- Cyclic prefix is used also to prevent the inter symbol interference by adding empty data to the end of OFDM frame to extend it and decrease the interference between the OFDM subcarriers when there is dispersive channel.

4- Superimposed training sequence technique employ training sequence adding algebra or after the OFDM frame slot on the transmitter side and storing the same training sequence on the receiver side. In the receiver the coefficients of equalizer are changed by using equalization process. Thus the output of the equalizer nearly matches the STS. However, the modulation of this STS with the input information adds an overhead and thus has effect on the throughput of the OFDM system [9].

In this research, the superimposed training sequence is the main topic which will study and analyzes the role it plays in communication system to improve the equalizers work by using two types of equalizers LMS and RLS equalizers with different M-PSK and M-QAM modulations over the Rayleigh fading channels presented. In order to compare the performance several of equalizer parameters are varied and Bit error rate (BER) is evaluated and compared [10].

1.4 Literature Survey of Channel Estimation.

December 1999, Hoeher P and Tufvesson F [6], for purpose of channel estimation in receiver, performed a superimposed pilot training sequence technique based on the Viterbi algorithm .

June 2000 F,Mazzenga [7], discovered a method for channel estimation by estimating the efficiency of bandwidth compared with the increasing the power of transmitted signal and adding a pilot sequence to the input frame.

November 2001, C. K. Ho, B. Farhang-Boroujeny and F. Chin [8], added pilot to the OFDM frame by employing channel estimation by semi-blind equalizer for OFDM in dispersive channels.

January 2005 Mounir Ghogho, Des McLernon, Enrique Alameda-Hernandez, and Ananthram Swami [9], showed that best estimation can be got when the training sequence is arithmetically added to the OFDM transmitted data instead of putting in an empty part of time slot.

2006 Yang, Qinghai and Kwak, Kyung Sup [10]. in "Time-Varying Multipath Channel Estimation with Superimposed Training in CP-OFDM Systems", have analyzed superimposed training sequence as CP when compacted with OFDM frame and transmitted through time varying multipath channel.

June 2006 N.Chen and G. T. Zhou [11], describe OFDM transmission frame with superimposed training. They examined the superimposed training sequence with PAR of OFDM signal also derived the function of cumulative and complementary distribution and determined the upper and lower bounds for this function.

February 2007 Qinghai Yang et al [12]: in "Superimposed training for estimating of doubly selective OFDM channels", in superimposed training-based approaches, found a major problem to be handled is that of interference from the transmitted information sequence unknown at the receiver.

November 2007 Yang, Q. and Kwak, KS [13]: in "Optimal superimposed training for estimation of OFDM channels", introduce an optimal method for using superimposed pilots in channel estimation for OFDM system using Wiener filter. This algorithm is a device to find set of the optimal rectangular of time-frequency samples for the complex channel estimation.

October 2011 Abdelhakim Khlifi and Ridha Bouallegue [31]:in "Performance Analysis of LS and LMMSE Channel Estimation Techniques for LTE Downlink Systems", The main purpose of this paper is to study the performance of two linear channel estimators for LTE Downlink systems, the Least Square Error (LSE) and the Linear Minimum Mean Square Error (LMMSE).

March 2013 Ali Asadi and Behzad Mozaffari Tazehkand [32]: in "A New Method to Channel Estimation in OFDM Systems Based on Wavelet Transform" introduce a new threshold based method using wavelet decomposition will be proposed which is based on an initial LS estimation technique.

Jul-Aug 2013 Gunjan Verma, Prof. Jaspal Bagga [33]: in "Performance Analysis of Equalizer Techniques for Modulated Signals" In this work, the performance of two equalizers Least Mean Square (LMS) and Recursive Least Square (RLS) is observed by calculating the BER effect of Rician channels over low Doppler shift and found the best result when used Recursive Least Square (RLS).

1.5 Aim of the Work

The aim of this thesis is to investigate the effect of using superimposed training sequence technique (STS) instead of the conventional training sequence in OFDM systems to meet the fast fading channel requirements. A simulation of OFDM system is to be implemented for the following investigation:

1- The role that is STS plays to decrease the effect of multipath fading channel for fast fading.
2- The significance of STS to equalize the received signal and reduce the effect of inter symbol interference (ISI).
3- Comparison between different modulation techniques.
4- Study the performance of two types of equalizers; Least Mean Square (LMS) and Recursive Least Square (RLS).
5- Calculate the Bit error rate of two equalizers with three of modulations techniques.

1.6 Thesis Outline

This thesis consists of chapters including this introductory chapter.

Chapter One includes a general introduction to wireless communication system and the main problems in wireless channel communication and literature review.

Chapter Two introduces the fading and discusses the main reasons which cause the inter symbol interference and classifies the type of fading channel.

Chapter Three: explains briefly the equalization process and the advantage of superimposed training sequence which is used in equalization process to reduce the effect of ISI.

Chapter Four: introduces the OFDM communication system design with the methods for adding the STS and the channel properties.

In Chapter Five: gives simulation results for OFDM system with training sequence with and without adaptive equalization.

Chapter Six: includes the conclusions from the work and gives suggestions for future work.

CHAPTER TWO

Fading

2.1 Introductions

This chapter includes introductions to the fading and discusses the main reasons which caused the inter symbol interference and classifies the types of fading channel.

In most of the wireless communication environment, a signal exits a transmitter split into wide directions and these rayed signals take many paths and arrive to the receiver in different amplitude and different times, so that, the signal reaching to the receiver is the composite of all the components. When the two signals get combined the summation signal can be an attenuated signal or augmented signal depending on nature of the two signals are combined constructively or destructively.

In wireless communication environments, the signals copies became collective at the receiver and some of them constructively combine and some of them destructively combine[14].The final result of summation of all the received signals become very complicated and the received signal becomes totally different from the original signal transmitted from the input source. The quality of the received signal at the receiver gets weaker than the transmitted signal. This type of process of signal destruction by the multiple fading of a signal is called 'Fading'.

For more understanding of the fading, the below figures show a couple of different aspects of fading which can be measured using various equipment’s. First, compare between faded signals and non-faded signal using a power spectrum scope. The two results show the high fluctuation in the faded spectrum.

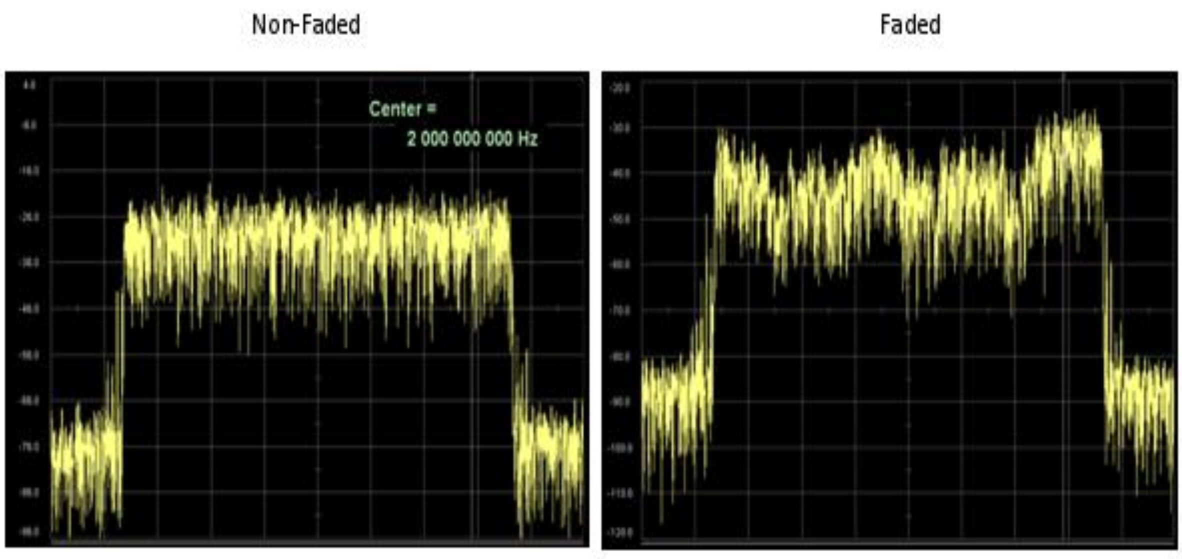


Figure 2-1 Faded and non-faded signals on spectrum analyzer [14]

Figure 2-2 shows the signal constellation for the faded and non-faded signal and it is very clear that the non-faded constellation figure has symmetric poles and without any impurities.

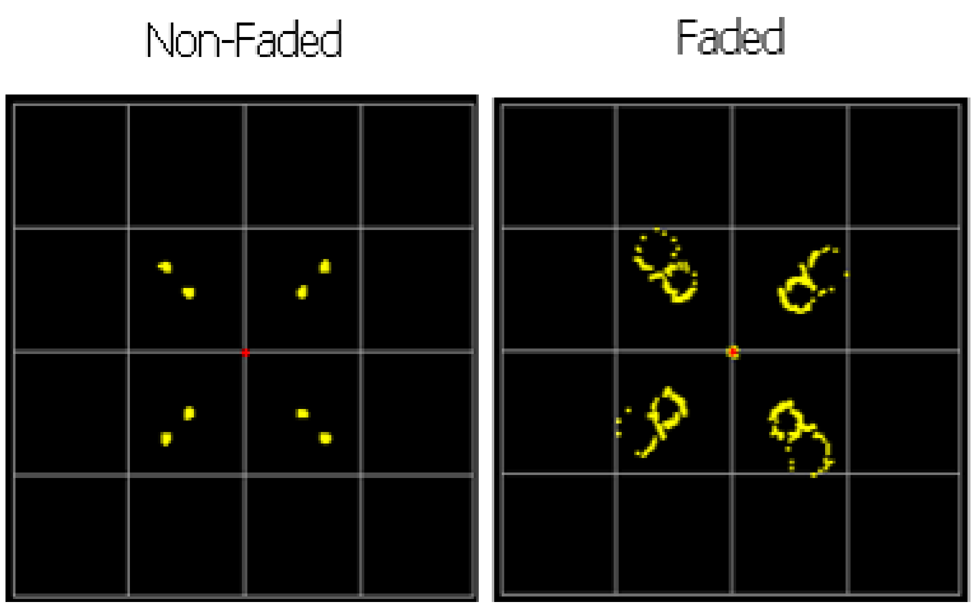


Figure 2-2 Samples of constellation for faded and non-faded signal. [14]

2.2 Major Components of Fading

Fading is generated from many components or factors the list below gives the major factors for fading:[14]

- Variation of the received signal power
- Fluctuations in signal phase
- Received power variation created by multi path
- Fluctuations of angle of arrival of received signal
- Path loss
- Frequency Shift(Doppler shift)
- Reflection and diffractions from various objects

The fading analysis requires brief explanation of each of these factors, Figure 2-3 represents the summation of three paths at the receiver end.

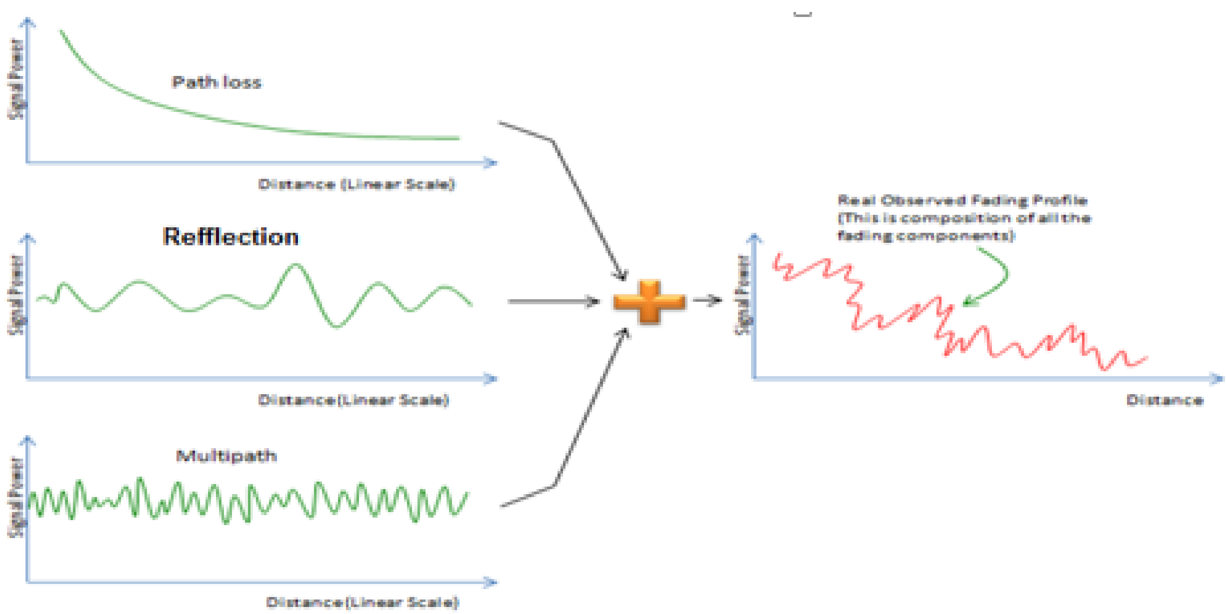


Figure 2-3 The summation of three paths at the receiver end. [14]

Figure 2-3 shows the three paths behavior on transmission channel, the horizontal axis represent the distance.

In the first section is the path loss decreases gradually but does not have any fluctuation. In the second section, notice a low fluctuation of the signal over the distance. In the third section, notice a high fluctuations of signal over the distance. The fourth section (on the right side) is the summation of these three paths.

2.2.1 Fast Fading and Slow Fading

Fast fading is used to describe channels when the coherence time of the channel less than the transmission symbol time. Fast fading describes a condition where the time duration in which the channel behaves in a correlated manner is short compared to the time duration of a symbol. Therefore, it can be expected that the fading characteristic of the channel will change several times while a symbol is propagating, leading to distortion of the baseband pulse shape. Analogous to the distortion described as ISI, the distortion take place because the received signal's components are not all highly correlated throughout time. Hence, fast fading can cause the baseband pulse to be distorted, resulting in a loss of SNR that often obtains an irreducible error rate. Such distorted pulses cause synchronization problems [15].

Slow Fading is used to characterize channels when Coherence time of the channel greater than the transmission symbol time. Slow fading describes a condition where the time duration in which the channel behaves in a correlated manner is long compared to the time duration of a symbol. So that can expect that the channel status remains unchanged approximately during the time of transmitted a symbol. The symbols will not suffer from the distortion in the pulse, the loss in SNR is the primary degradation in a slow-fading channel [15].

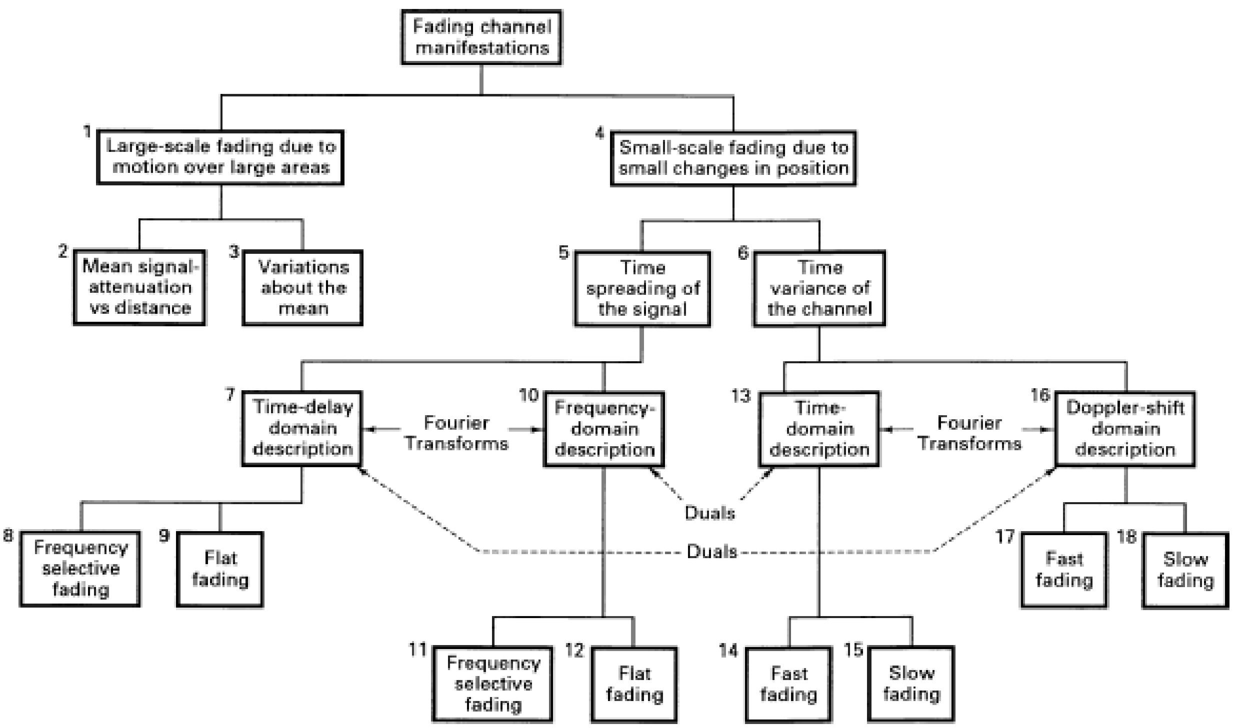


Figure 2-4 fading channel manifestations [29]

Figure 2-4 shows the structure of fading. The fading channel can be divided into two main parts, large scale fading and small scale fading. The main reason for large scale is the movement over the wide areas and this move causes attenuation and fluctuations in the mean signal compared with distance.

The small scale part is caused by the low changes in location and this has effect on the time spreading and time variance of the channel.

The phenomenon of flat and frequency selective fading come from the time spreading of the transmitted signal as indicated in 11,12 blocks when working in the frequency domain and from time delay domain as indicated in blocks 8,9 in Figure 2-4. [29]

The fast and slow fading's emerge from the Doppler shift factor and time variance factor when working in time domain as indicted in blocks 14, 15, 16 and 17.

There are three basic phenomenon that affect signal propagation in a wireless communication system:

- Reflection appears when a propagating wave hits a smooth surface with very large volume compared to the RF signal wavelength (λ).

. • Diffraction which appears when the radio path between the transmitter and receiver is blocked by a heavy form with large volume compared to λ, causing minor waves to be composited behind the heavy form.

- Scattering which appears when a radio wave hits either a large harsh surface or any surface whose volume is on the order of λ or less, causing the energy to be spread out (scattered) or ray in different directions. an example is an urban environment, perfect signal obstructions that produce scattering include foliage, lamp posts, and street signs. [29]

2.2.2 LARGE-SCALE FADING: PATH-LOSS MEAN AND STANDARD DEVIATION

For the radio wireless application, Okumura performed some measurements of the path-loss for the heights of antenna and coverage. Hata [15] transformed Okumura's data into parametric formulas. For the mobile radio application, the mean path loss, $\overline{L}p\ (d)$ is a function of distance, d, between the transmitter and receiver it is proportional to an n_{th} power of d relative to a reference distance $d_{\circ}$. as shown below.

$$\overline{L}p\,(d)_{\,(\mathrm{dB})} = L_s(d_\circ)_{\,(\mathrm{dB})} + 10\,n\,\log_{10}(d\,/d_\circ) \quad (2.1)$$

where $d_\circ$ is the reference distance, the value of $d_\circ$[is taken to be 1 km for large cells, 100 m for microcells, and 1 m for indoor channels] [15]. $\overline{L}p\ (d)$ is the average path loss for a given value of d. Linear regression for a minimum mean-squared estimate (MMSE) fit of $\overline{L}p\ (d)$ versus d on a log-log scale (for distances greater than $d_\circ$) generates a slope equal to 10n dB/decade with straight line. (n) value depends on the antenna height ,frequency, and propagation medium. In free space, n = 2, as seen in equation 2. 1[15], when have a strong wave has guided phenomenon (like urban streets), n can be lower than 2. The path loss $L_s(d_\circ)$ to the reference point at a distance $d_\circ$[from the transmitter is typically found through field measurements or calculated using the free-space path loss given by equation 2.2].[30]

$$L_s[d] = \left(\frac{4\pi d}{\lambda}\right)^2 \qquad (2.2)$$

Thus, path loss $L_s(d)$ can be expressed in terms of $L_\rho(d)$ plus a random variable Xσ, as follows [3]:

$$L_\rho(d)\ (\mathrm{dB}) = L_s(d_\circ)\ (\mathrm{dB}) + 10nlog_{10}\ (d\ /d_\circ) + x_\sigma\ (\mathrm{dB}) \qquad (2.3)$$

Where x_σ denotes a zero-mean Gaussian random variable (in decibels) with standard deviation σ. x_σ is site- and is distance-dependent. The choice of a value for x_σ is often based on measurements; it is not unusual for it to take on values as high as 6–10 dB or greater. Thus, the parameters needed to statistically describe path loss due to large-scale fading for an arbitrary location with a specific transmitter receiver separation are: [29]

- The reference distance $d_\circ$
- The path-loss exponent n
- The standard deviation σ of x_σ .

2.2.3 SMALL-SCALE FADING: STATISTICS AND MECHANISMS

When the received signal is made up of multiple reflective rays plus a significant line-of-sight (non fading) component, the envelope amplitude due to small-scale fading has a Rician, and is referred to as Rician fading [15]. The non faded component is called the specula component. As the amplitude of the specula component approaches zero, the Rician approaches a Rayleigh, expressed as

$$\int \frac{r}{\sigma^2} \exp\left[-\frac{r}{2\sigma^2}\right] \qquad \text{for } r \geq 0 \qquad (2.4)$$

where r is the envelope amplitude of the received signal and 2 is σ^2 which is the predilection mean power of the multipath signal. The Rayleigh fading component is sometimes called the random or scatter or diffuse component. The Rayleigh results from having no specular component of the signal; thus, for a single link it represents the pdf associated with the worst case of fading per mean received signal power. For the remainder of this research, it will be assumed that loss of signal-to-noise ratio (SNR) due to fading follows the Rayleigh model described. It will also be assumed that the propagating signal is in the ultra-high-frequency (UHF) band, encompassing present-day cellular and personal communications services (PCS) frequency allocations — nominally 1 GHz and 2 GHz, respectively

[15]. As indicated in Figure2-4, blocks 4, 5, and 6, small-scale is fading which manifests itself in two mechanisms:

• Time-spreading of the underlying digital pulses within the signal

• A time-variant behavior of the channel due to motion (e.g., a receive antenna on a moving platform).

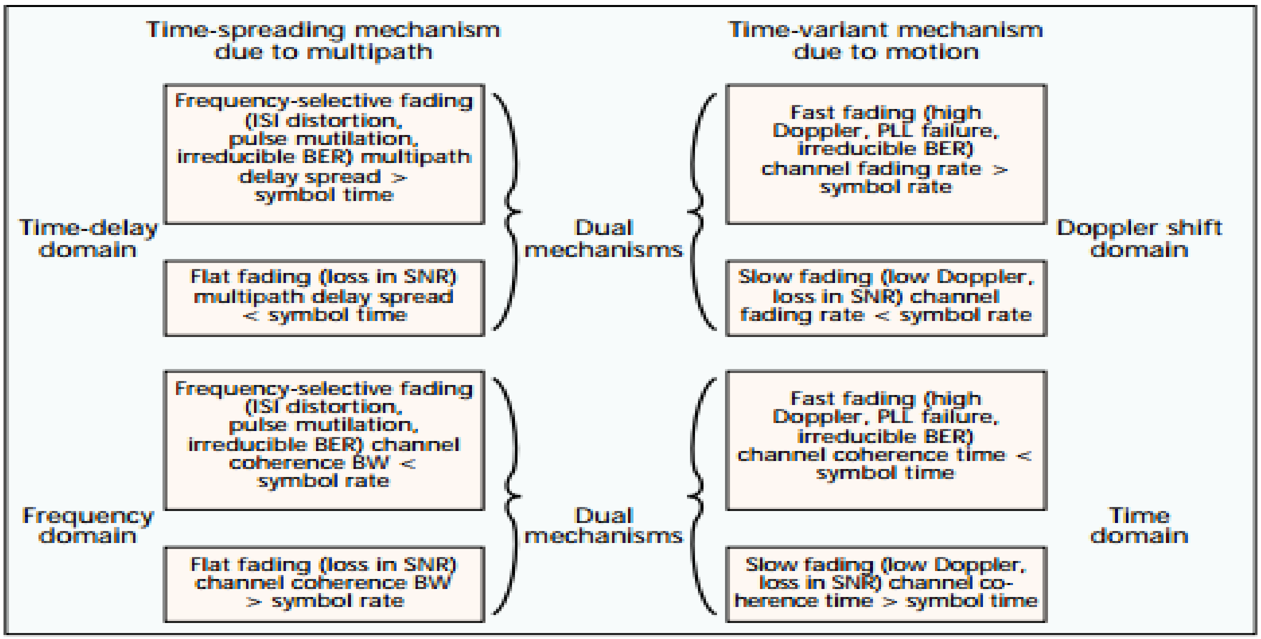


Fig 2-5.Small-scale fading: mechanisms, degradation categories, and effects. [15]

The model treats signal variations arriving with different delays as uncorrelated. It can be shown such a channel is effectively WSS in both the time and frequency domains.

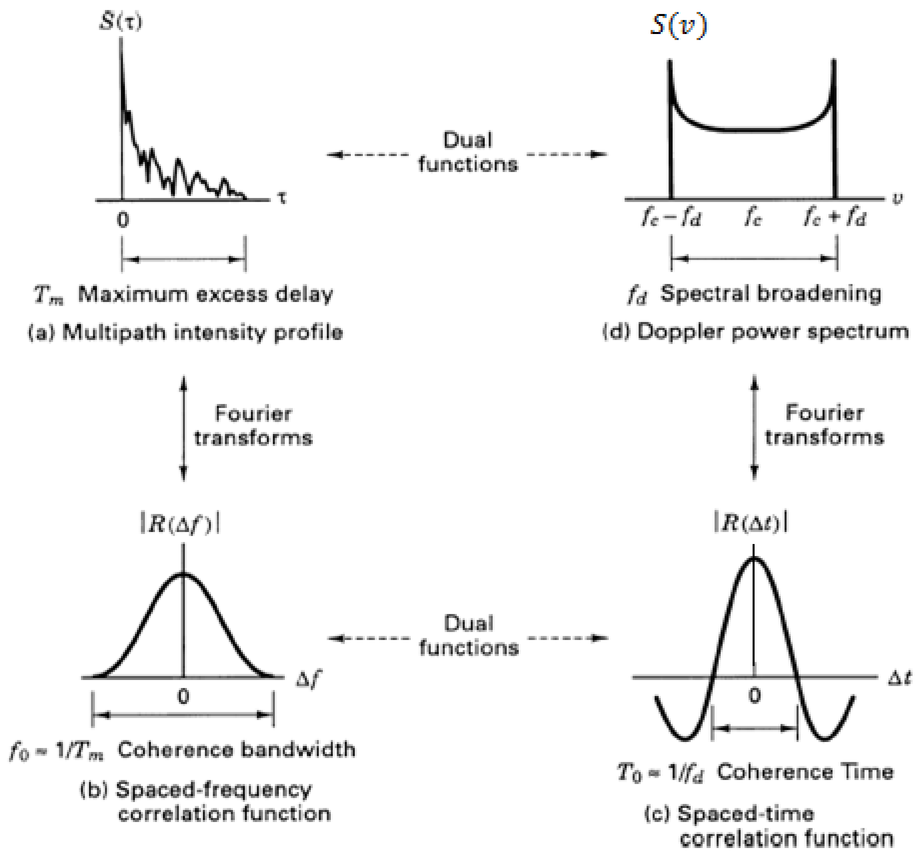


Figure 2-6 Relationships among the channel correlation functions and power density functions. [29]

For the mobile channel, Figure 2-6 contains four functions that make up this model [15]. Starting with Figure. 2-6a and proceeding counterclockwise toward Figure 2-6d. In Figure 2-6a a multipath-intensity profile, S(τ) versus time delay τ, is plotted. Knowledge of S(τ) helps answer the question "For a transmitted impulse, how does the average received power vary as a function of time delay τ. The term "time delay" is used to refer to the excess delay. It represents the signal's propagation delay that exceeds the delay of the first signal arrival at the receiver. For a typical wireless radio channel, the received signal usually consists of several discrete multipath components, sometimes referred to as fingers. For some channels, such as the troposphere scatter channel, received signals are often seen as a continuum of multipath components [15].

For a single transmitted impulse, the time, T_m , between the first and last received component represents the maximum excess delay, during which the multipath signal power drop to some threshold level below that of the strongest component.[30]

The frequency-selective fading happen when $Tm > Ts$. This condition appears whenever the received components of a symbol expand after the symbol's time duration. Such multipath dispersion of the signal yields the same kind of inter symbol interference distortion caused by an electronic filter. [29]

The flat fading happen when $Tm < Ts$. In this situation, all the received components of a symbol reach within the symbol time duration; so that, the components are not destroyed. Here, there is no channel-induced ISI distortion, since the signal time spreading does not result in significant overlap among neighboring received symbols. There are still performance fluctuations since the irresolvable phasor components can be added up destructively to yield a substantial reduction in SNR [29].

2.2.4 Channels with Multipath

The definition properties of the wireless channel are for reinforcement of a model for the fluctuations of the channel strength over time and frequency.

A MS is connected with antenna of BTS receiver and there are a lot of obstacles or buildings between the antenna of receiver and the MS. the receiver antenna would receive the multiple copies of the same signal from MS because the same signal can reach the antenna of receiver in multiple paths as shown in Figure 2-7. Let's take example that has four different paths as shown in Figure 2-7. Two characteristics are in this example, one is time delay and the other one is attenuation. [14]

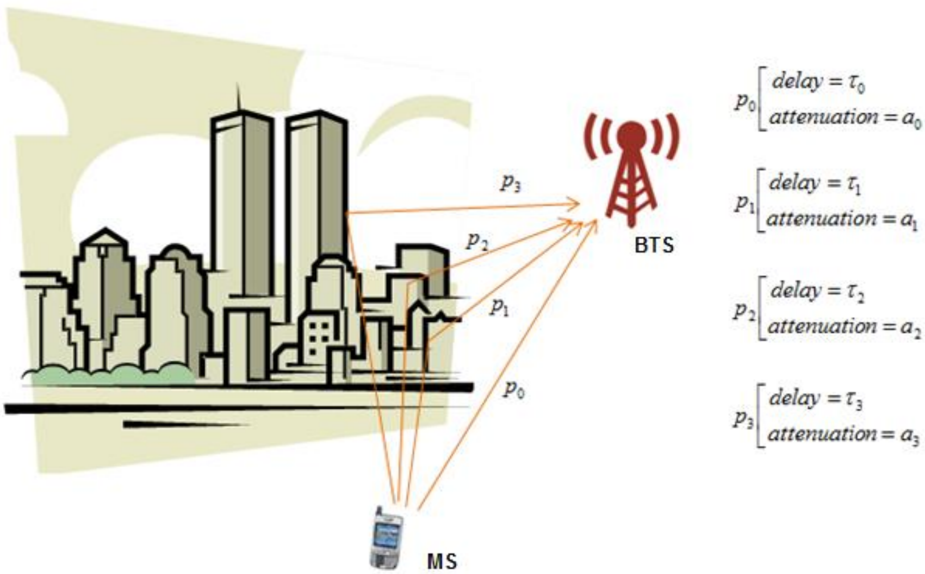


Figure 2-7.Time delay and attenuation for four multipath channels.[14]

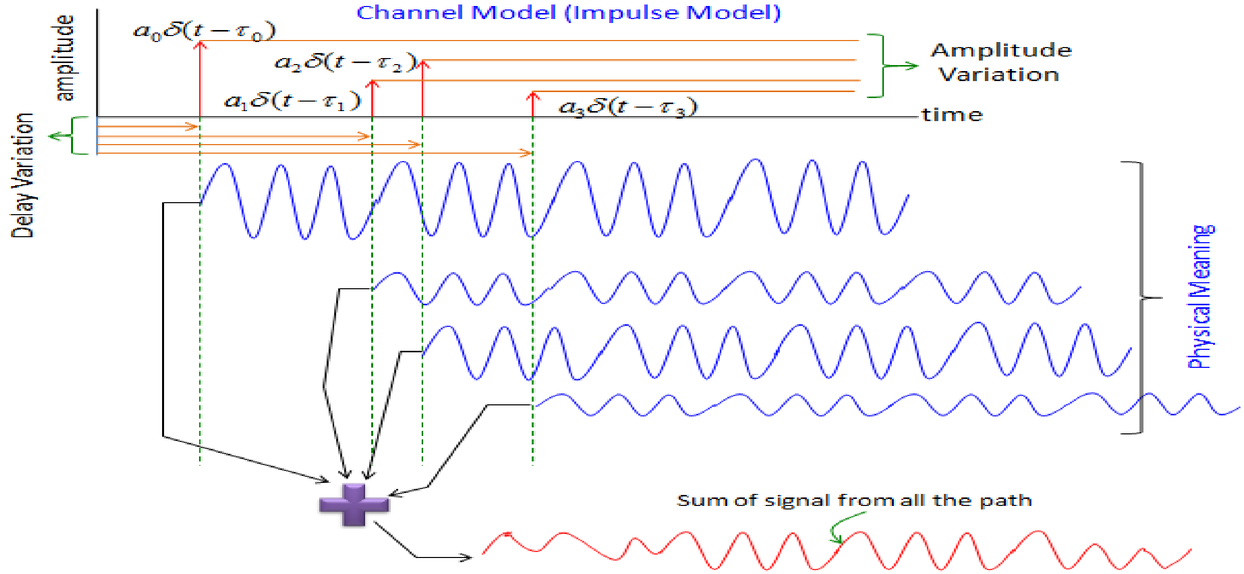


Figure 2-8.Summation for four multipath channels.[14]

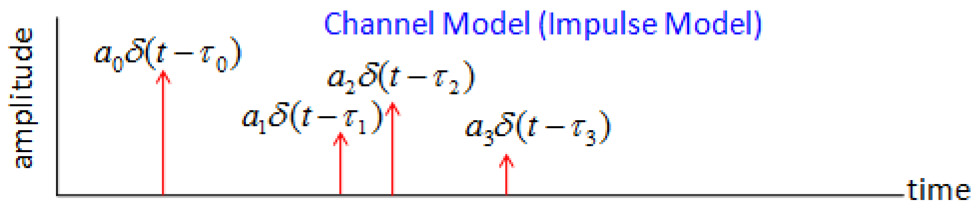


Figure 2-9.Impulse Response for four multipath channels.[14]

$$h(t) = a_\circ\delta(t - t_\circ) + a_1\delta(t - t_1) + a_2\delta(t - t_2) + a_3\delta(t - t_3)$$
$$= \sum_{i=0}^{3} a_i\delta(t - t_i) \quad (2.6)$$

Where$h(t)$ $is\ the\ implse\ responce\ of\ the\ channl.$

Figures 2-8 and 2-9 show the summation of four multipath channel and impulse response for the these four multipath channels respectively at the receiver end, each path has a different amplitude and phase and the red path represent the summation of the four paths, when here is a larger gap in time between the paths then the equalization process becomes very hard to recover the transmitted signal.

Figure 2-10 shows that the amplitude and phase get different from the transmitted signal, but overall shape of the signal is the same as that of the transmitted signal. It means that no distortion happen in by this multipath.

In this case you can suppose that a signal has six different multipath as shown in part (1) (the upper part). The time delay is represented in horizontal axis and the amplitude is represented by vertical axis. Parts (2) ~ (7) show the signal propagates through each path till arrives to Rx antenna. Also it can noted the amplitude and phase are different in each path. Part (8) indicates the summation of all parts from (2) ~ (6), part (8) is the final signal the receiver will have treatment with.

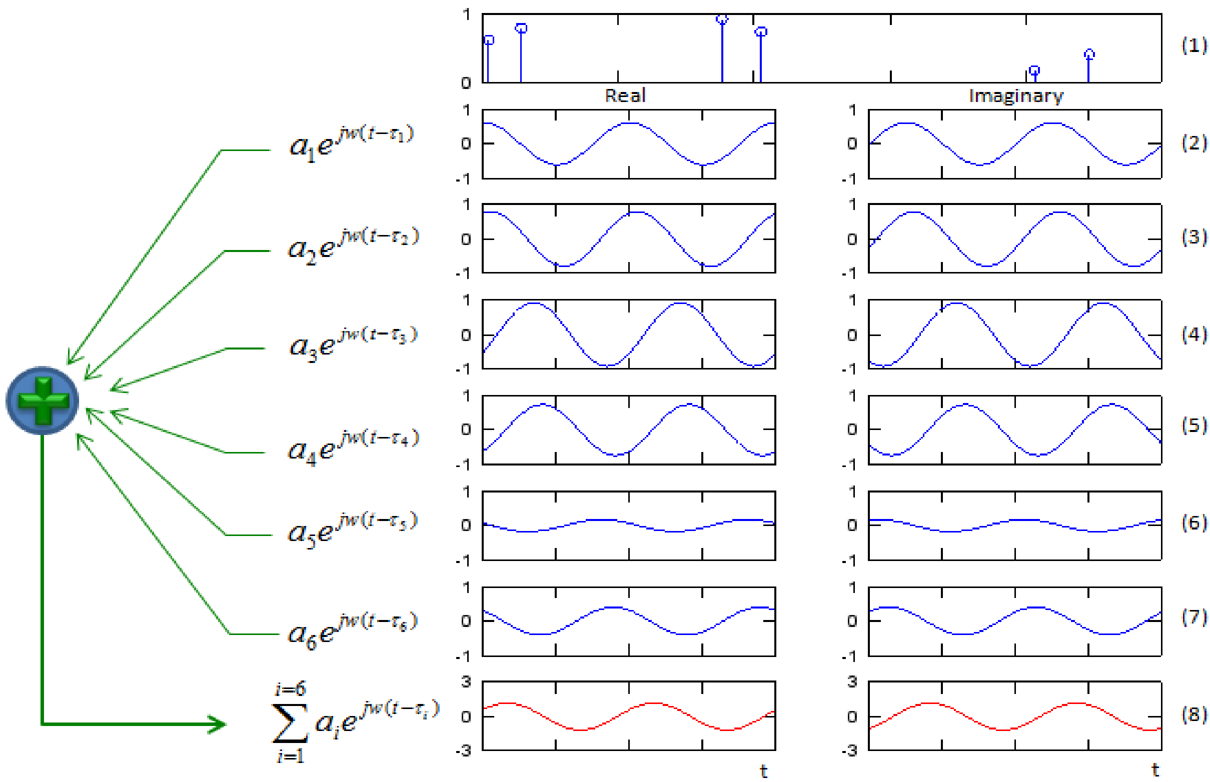


Figure 2-10.The amplitude and phase get different from the transmitted signal.[14]

2.2.5 Doppler Spread

Doppler shift is the famous phenomenon in the wireless communication fields, it is related to the motion of the transmitted signal and the receiver antenna. Doppler shift causes the Doppler spread in the transmitted signal when passing through the channel as shown in the below figure 2-11.

As shown, how much a frequency (f_c) gets speeded by Doppler spread is indicated byf_m. Figure 2-11 show another formula for representing f_m, it will be noticed f_m is determined according to the object velocity (relative velocity between transmitter and receiver) and speed of light and (f_c) is the carrier frequency. Which is constant. Now the only variable is the moving speed (v). According to this equation, f_m and$(v)$are in proportional relation, meaning if (v) gets higher, f_m becomes larger and if (v)gets smaller, f_m becomes smaller [14].
As shown in the below graph the width depends on f_m value.

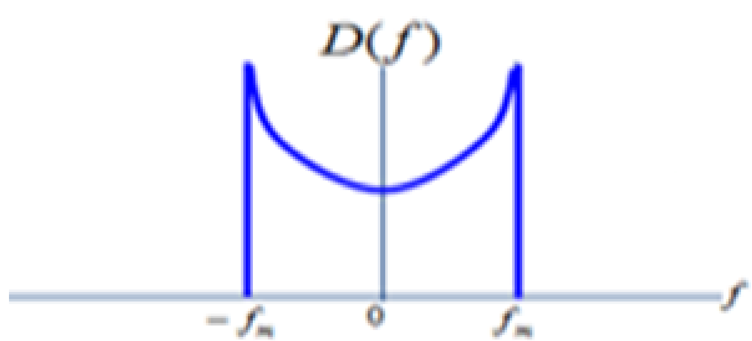


Figure 2-11 Doppler spread [14]

$$D(f) = \frac{1}{\sqrt{1-\left(\frac{f}{fm}\right)^2}} \tag{2.7}$$

$$f_D = \frac{v * f_c}{c} \tag{2.8}$$

v *is moving speed.*

$v.\cos\theta$ *is the relative velocity.*

f_D *is the Doppler spread.*

f_c *is the carrier frequency.*

c *is the speed of light* in free space.

Doppler spread which affect the signal can be divided by two types, the wide Doppler spread and the narrow Doppler spread, below the simulation results for these two cases are shown in Figure 2-12 [14]. The lower part is for wide Doppler spread the upper part is for narrow Doppler spread. Figure 2-12 shows a lot of frequency energy fluctuations vs. time in wide Doppler spectrum compared with narrow band.

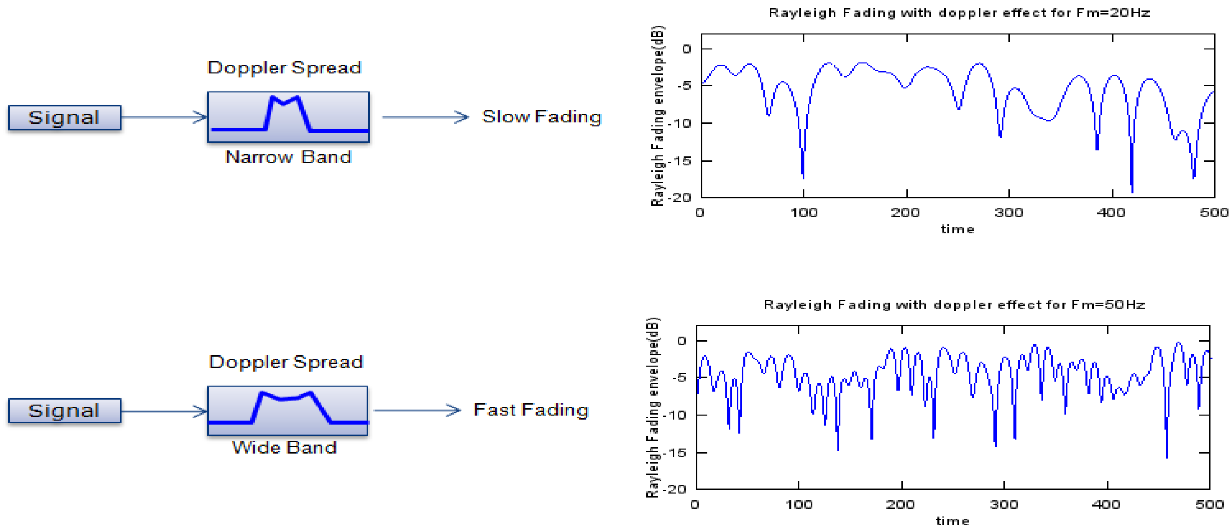


Figure 2-12 Narrow and wide Doppler spread [14]

CHAPTER THREE

Equalization

3.1 Introductions

As is explained briefly in chapter two fast fading channel has effect on transmitted data and the distortion generated causes inter symbol interference (ISI) and the additive noise (AWGN) which already added to the transmitted signal during passed through multipath channels.

This chapter will explain the solution that will decrease the effect of multipath channel and ISI for fast fading channel and steps for equalization in OFDM system to make equalization process before the FFT processing [2].

Equalization is the operation to clear ISI and the effect of noise from the transmission channel. It is put in the receiver of the communication system. The equalizer transfer function is reciprocal of the transfer function of the channel Equalization involves many process of reducing the MSE between input response and output.

Equalization methods are used by communication engineers to mitigate the effects of the inter symbol interference. This thesis studies both inter symbol interference and several equalization methods which amount to different structures for the receiver box. The methods presented in this thesis are not optimal for detection, but rather are widely used sub-optimal cost-effective methods that reduce the ISI. These equalization methods try to convert a band limited channel with ISI into one that shows less memory, hopefully synthesizing a new AWGN-like channel at the receiver output. The designer can then analyze the resulting memory less, and the equalized channel using the methods.

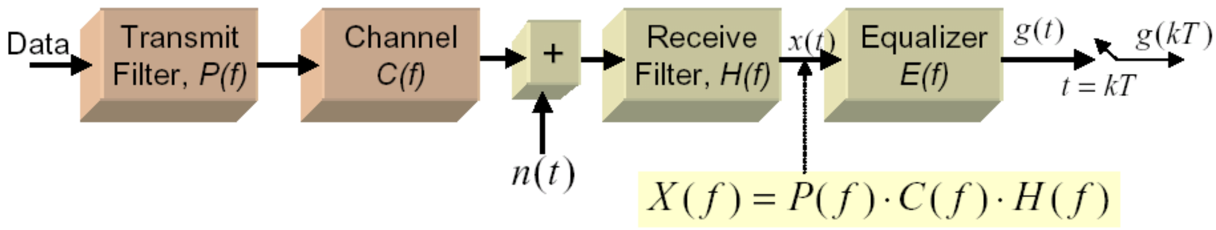


Figure 3-1 the band-limited channel with receiver [16]

3.2 TYPES OF EQUALIZERS:

The equalizer types can be classified depending on the adaptation characteristics and the algorithms which used the equalizers and they are grouped as shown below Figure 3-2:

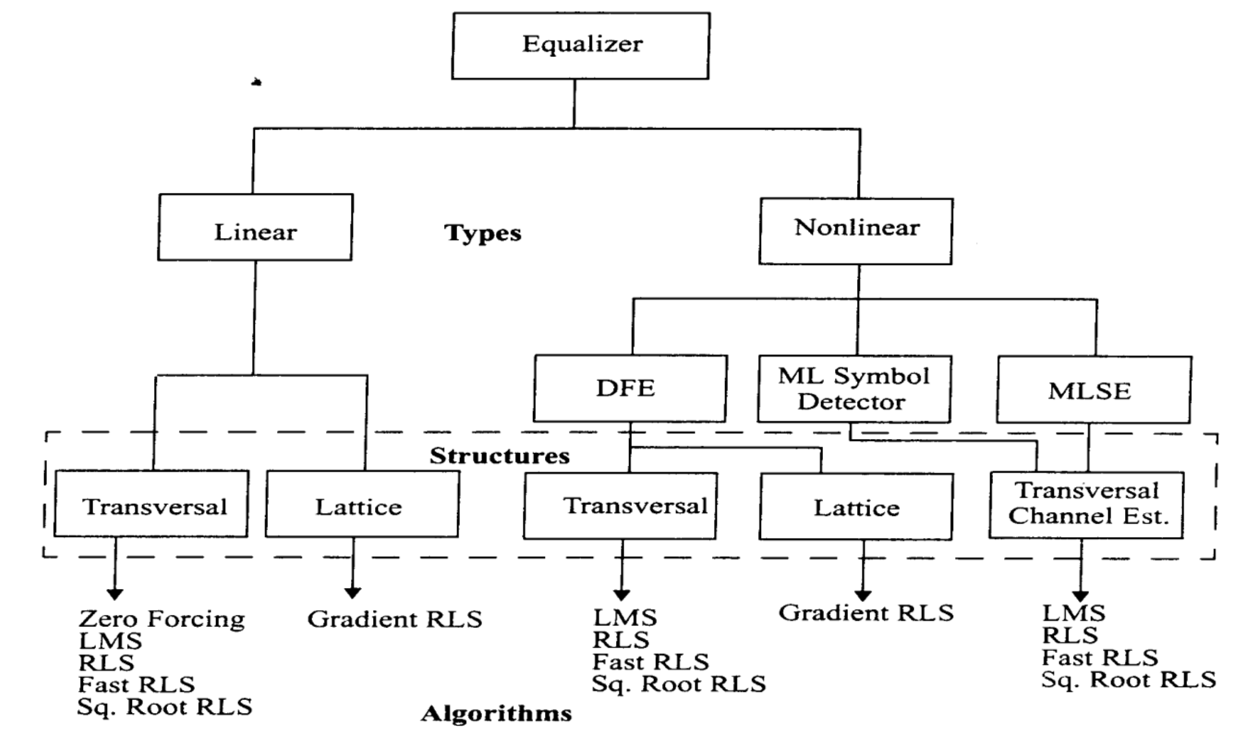


Figure 3-2 structure of equalization types [16]

3.2.1 Zero Forcing (ZF)

The ZF-equalizer is designed using the peak-distortion criterion. It ideally eliminates all ISI. The names zero forcing corresponds to bringing down inter-symbol interference to zero in environment having noise. Zero-forcing equalizers cancel the additive noise and may significantly amplify noise for channels with spectral nulls.

The limitations of ZF Equalizer are:

1. When the impulse response of channel has finite length, the equalizer impulse response needs to be infinitely long. [18]
2. When some weak signals is already received on the received side, to indemnity magnitude of ZF equalizer grows so large. As a result, noise will added after the received channel will take support by large factor and control overall SNR.

Furthermore the channel may have zeros in its frequency responses that cannot be inversed at all.

The ZF Equalizer belongs to the class of preset linear equalizers and it uses the Peak Distortion Criterion to evaluate the equalizer tap weights. Consider the communication system block diagram (with an equalizer) given in Figure 3-3.

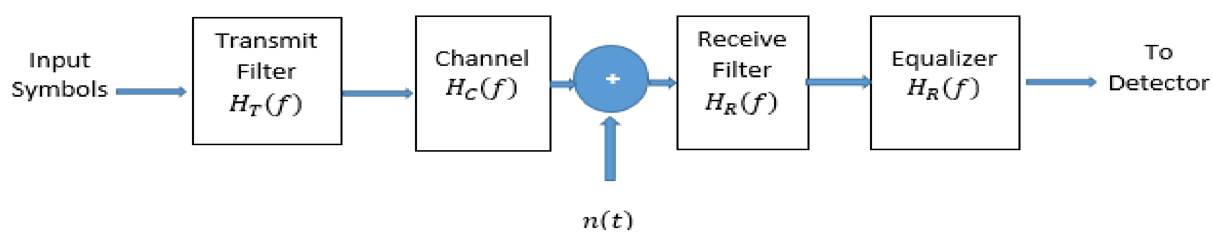


Figure 3-3 Block diagram of Zero forcing equalizer [18]

$$H_R(f) = \begin{cases} \sqrt{H_c(f)}\,.\,e^{-j2\pi f t_{\circ}}\,, & |f| \leq W \\ 0 & |f| \geq W \end{cases} \qquad (3.1)$$

$$H_R(f) = \; H^*{}_T(f)$$

$$H_R(f).H \quad _T(f) = \left|H \quad _T(f)\right|^2$$

$$H_E|(f)| = \frac{1}{H_C(f)} = \frac{1}{|H_C(f)|}e^{-j\theta_c(f)} \quad , \qquad |f| \leq W \qquad (3.2)$$

Where $H_E|(f)|$ is the channel distortion compensate.

$H_R(F)$ is channel frequency response.

$H^*{}_r(F)$ is inverse of the channel frequency response.

$H_{Eq}|(f)|$ is the equalized frequency response.

$H_c(f)$ is the channel impulse response

To restore the transmitted signal, a zero-forcing equalizer is used that applies the channel frequency response inverse. The combination of channel and zero forcing equalizer output gives a flat frequency response with linear phase to obtain the transmitted signal [16].

The equalizer has two input signals, the training symbols which are also used at the transmitter side, and the received symbols. As indicated in Figure 3.4by selecting column block, the received signal is divided into the training symbols and data symbols. The channel estimation is done by dividing the received training symbols by the true training symbols. This estimation is used for the zero-forcing equalizer. The subsystem of the equalizer gains is shown in Figure 3.5.

The estimated channel is the input of the equalizer gain block and the output is the inverse of the channel estimation. According to the zero-forcing equalizer technique with combination of received data and zero forcing equalizer output (inverse of channel estimation), the estimated signal before the channel can be achieved.[17]

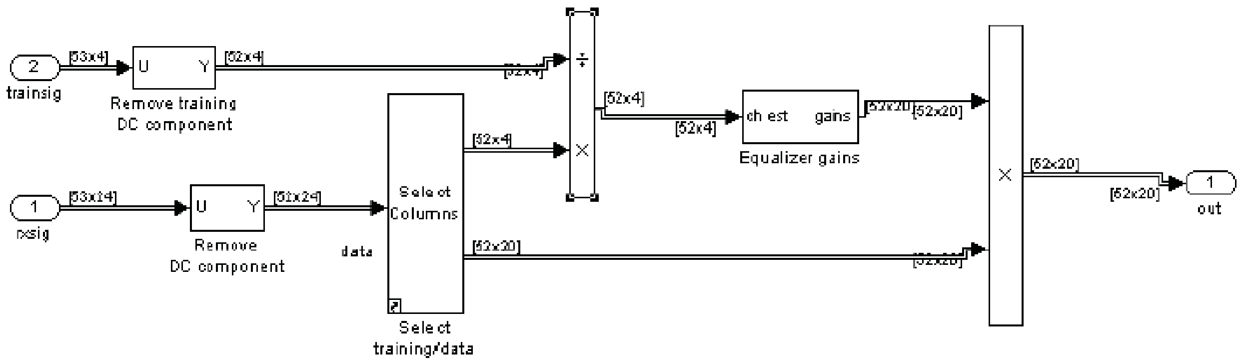


Figure 3.4 Subsystem of equalizer [17]

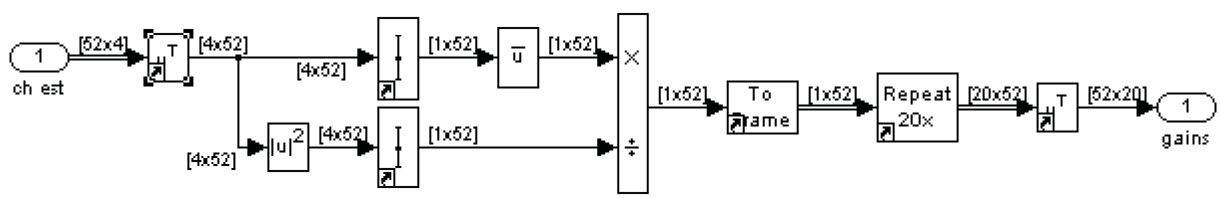


Figure 3.5 Subsystem of equalizer gains [17]

3.2.2 Adaptive Equalizer

An adaptive equalizer is an equalizer that can adapt automatically to fading properties in wireless communication channel. In this research the equalizer is implemented in a training mode. It can be implemented by performing the tap weight adjustments continually by replacing the known training sequence with a sequence of data symbols it can estimated from the output of equalizer and treated as known data.

3.2.2.A Least Mean Square (LMS)

Architecture of the adaptive equalizer is shown in Figure 3-6 below. The coefficients of the filters are called weights of the system and are updated according to the type of equalizer algorithm. The equalizer output is calculated as;

$$y\,(n) = \sum_{k=0}^{M-1} \mathrm{u}\,(\mathrm{n}-\mathrm{k}\,)\mathrm{w}_{\mathrm{k}}^{*}(\mathrm{n}) \qquad (3.3)$$

The error signal is calculated as;

$$e\,(n) = d\,(n) - \mathrm{y}\,(n) \qquad (3.4)$$

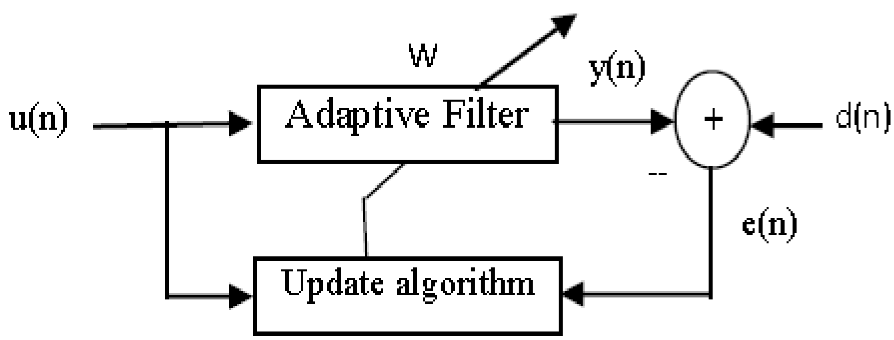


Figure 3-6 Adaptive Equalizer [9]

Where, w: weight vector or coefficients
u: input vector
e: error signal

LMS equalizer is built around a transversal (i.e. tapped delay line) structure. The LMS filter or equalizer employs the small step size u which provides a simple and accurate representation of the filter transient behavior. Let W be the initial weight then [17]:

$$G = \mu * u(n-k) \qquad (3.5)$$

Where μ is the step size of LMS algorithm.
Updated weights are calculated as;

$$w_k\ (n+1) = L_{\text{factor}} * w_k\ (n) + G * e^*(n) \qquad (3.6)$$

Where L_{factor} is the leakage factor which is 1 for standard LMS algorithm [19].

3.2.2. B Recursive Least Square (RLS)

The Recursive least squares (RLS) based adaptive filter algorithm recursively finds the filter coefficients and minimize the weighted linear least squares cost function associated with the input signals. RLS algorithm converges faster than other methods but is computationally complex. The structure of the RLS algorithm is similar to that of LMS as shows in Figure 3.6. The RLS algorithm uses covariance matrix updating formula, which is used for automatics adjustment corresponding to the estimation error. The initial inverse correlation matrix Delta is given as; [19]

$$p(0) = \delta^{-1} I \qquad (3.7)$$

Where, p is the inverse correlation matrix and δ^{-1} is regularization parameter instance n=1, 2,3…. which is positive constant for high SNR and negative for the low SNR value. For any time instance n=1, 2, 3….

$$\pi(n) = p(n-1) * u(n) \qquad (3.8)$$

$u(n-k)$ is the input factor.

$$u(n) = \frac{\pi(n)}{\lambda + u^H(n) * \pi(n)} \qquad (3.9)$$

λ = Forget Factor.

Then priori estimation error is given by

$$e(n) = d(n) - W^{H}(n-1)x(n) \tag{3.10}$$

The updated weights are calculated as;

$$w_k(\mathrm{n}) = w_k\ (\mathrm{n}-1) + u(n) * e^{*}\ (n) \tag{3.11}$$

The converges of LMS is slow compared with RLS and the algorithm updated is slower in RLS than the LMS equalizer.

3.3 Superimposed Training Sequences

We specify the issue of fast fading channel estimation by using the superimposed training sequence, the superimposed training sequence includes algebraic sum of known and unknown sequence , the known sequence is usually identified by research like pilots sequence which is used in communication system to reliable wireless transmission channel .[20]

The researches performed a method which decreases the BER during channel estimation. This method comprises the training sequence in the transmitter and receiver in channel estimation steps to bring the superimposed received data together with the input data, the cross-correlation role to take the received data with an estimate time delay define by the rms delay spread of the channel, the superimposed training sequence stored in a register on the receiver and it is the same training e sequence which is added to the OFDM frame in the transmitter.

At the receiver end, the cross-correlated data proceeds over a length of custom of samples that can be made long to use and compare the coherence time of the channel and operates along with the stored values in the register of the inverse of auto-correlation of STS so as to get a better channel estimate.

In this case and for better performance the OFDM system needs bandwidth efficient channel estimation, training symbols known and stored in the receiver end which are algebraic addition to the data, this method by adding that avoids using additional time slots for training sequence, at the receiver the noise and unknown data are used for channel estimation.

This method is featuring on the pilots sequence by bandwidth efficient so the superimposed training sequence is used in modern communication system like 4G and LTE to control the bandwidth and avoid inter symbol interference. [20]

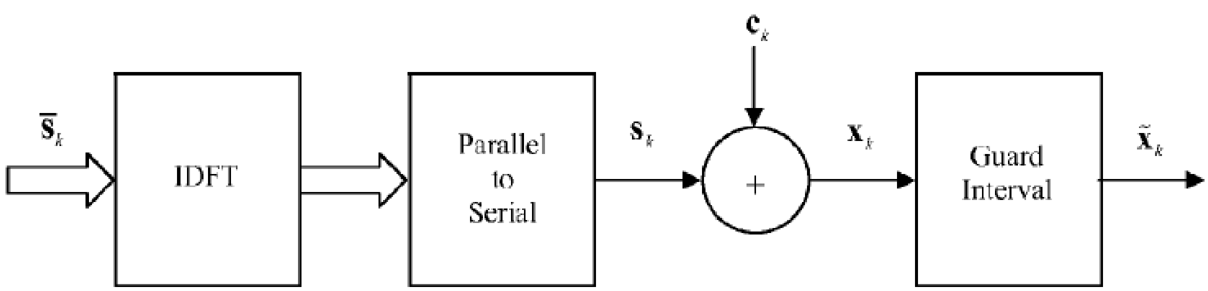


Figure 3-7 the block of the transmitter of a STS based OFDM system. [20]

Figure 3.7 will explain the method and location for adding the training sequence to the OFDM system. Binary data is collected into symbols depending on the level of modulation schema used. $\bar{s}_k$ Vectors of such symbols are formed after those are passed to the IDFT. Here *k* represents the OFDM symbol index. Some of values entries are zero to meet the requirements of the spectral mask of the OFDM system. The output of the IDFT is $s_k = F^H\bar{s}_k$, where F is the normalized N×N DFT matrix with and F^H is the complex conjugate transpose.

$$F(m,n)=\frac{1}{\sqrt{N}\,e^{-\frac{2j\pi mn}{N}}} \qquad (3.12)$$

Here (m,n) is the mth row and nth column of the matrix. A multiplexing is added to the end of transmitter to convert from parallel to serial after exiting from IDFT.

A training sequence c_k is then algebraic is added to this IDFT output `with a specific low training to data power ratio to get,

$$x_k = s_k + c_k \qquad (3.13)$$

The x_k is the summation of the input data and training sequence are pass through OFDM system this data will be modulated and occupy all sub bands of OFDM system by components of matrix as written in equation (3.14) to protect them from inter symbol interference.

$$c_k(m) = e^{j\frac{2km}{N}\left(\frac{m}{2}+1\right)} \qquad (3.14)$$

where m th is the row of the vector. This sequence will be summed with guard bands which are specified according to standard of OFDM system. In this case the training sequence is given by

$$c_k(m) = IDFT_N\left\{DFT_N\left[e^{j\frac{2km}{N}\left(\frac{m}{2}+1\right)}\right] \times W_{(k)}\right\} \qquad (3.15)$$

Where W (k) is the frequency domain window function used to meet the requirements of the spectral mask of the OFDM system. Here w_k can be any spectral mask including a rectangular function defined for $0 \le k \le N-1$ The DFT_N [x (n)] and $IDFT_N$ [X(k)] are defined as,[20]

$$DFT_N[x(k)] = X(k) = \sum_{n=0}^{N-1} x(n)e^{-j\left(\frac{2\pi nk}{N}\right)} \qquad k=0,1,\ldots N-1 \qquad (3.16)$$

$$IDFT_N[x(k)] = x(n) = \frac{1}{N}\sum_{k=0}^{N-1} x(k)e^{j\left(\frac{2\pi nk}{N}\right)}, n=0,1,\ldots N-1 \qquad (3.17)$$

In the guard band part, the cyclic prefix is add between OFDM symbol after IDFT, the benefit from this addition is it decreases the inter block interference (IBI) between IDFT output symbol and to make the equalization easy in the frequency domain at the end of receiver.

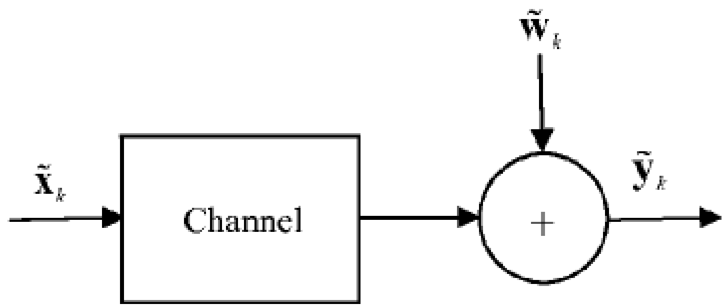


Figure 3-8 A block diagram of the equivalent baseband channel encountered in a typical superimposed training based OFDM system.

In all the OFDM symbols are passed on the transmission channel after exiting from the guard band, the situation will work for transmission channel type is the fast varying time channel that is caused by multipath of three paths with different power gains and time Daley and consider the OFDM symbol passed through adaptive white Gaussian noise AWGN after being delivered for Rayleigh fading channel.

This Rayleigh fading channel and the AWGN to add to the adaptive noise and made distortion on the OFDM symbol which already passed through it and the signal received by receiver antenna at the front of receiver includes the overall channel properties as shown in Figure3. 9.

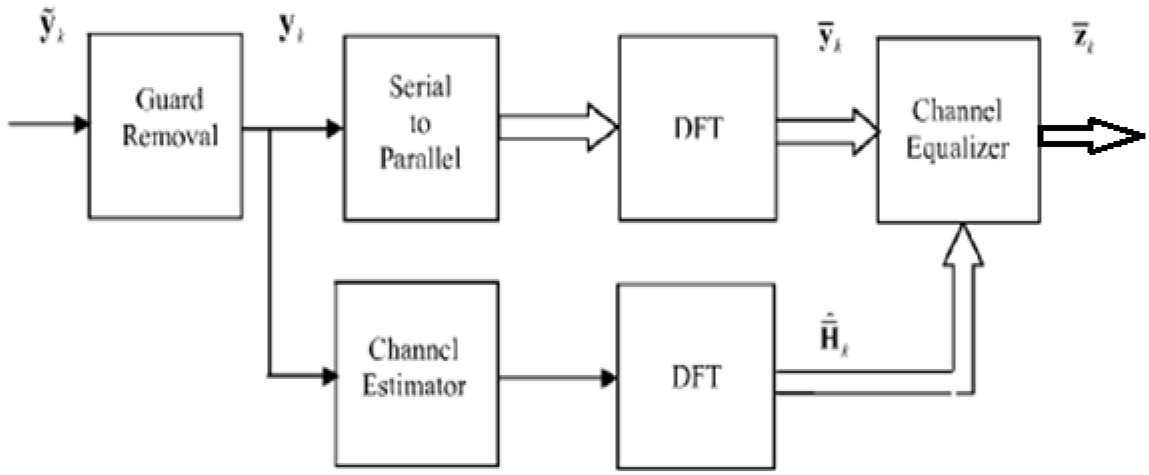


Figure 3-9 A block diagram of the equivalent baseband receiver of the superimposed training based OFDM system.

Figure 3.9 explains the receiver of the OFDM system with superimposed training sequence. The received vectors which get it after guard band removal from the OFDM symbols can be written as in following equation [20]

$$y_k = H_{k(}s_k + c_{k)} + w_k \qquad (3.18)$$

Where w_k is the AWGN with the k th vector and H_k is the channel matrix with dimensions of N×N.

Depending on the fading nature of the transmission channel you can suppose that the channel properties are closely the same for many OFDM symbols. Hence $H_k = H, K = 0,1, \ldots .(T_\rho - 1)$ where T_ρ depends on the channel coherence time. Equation 3 according to above fading nature can be written as following equation [20],

$$y_k = h_k(s_k + c_{k)} + w_k \qquad (3.19)$$

where s_k and c_k are Toeplitz matrices of the data and superimposed training respectively having dimensions of $N \times L$ and $h_k = h, K = 0,1,\ldots.T_\rho \quad (T_\rho - 1)$ is the $L \times 1$ length channel vector.[17]

3.4 The Conventional Training Sequence

The addition process for conventional training sequences is widely used in communication systems because the key role it plays to recover the received signal, according to this significant role, the global system for mobile communication (GSM) set the standardization to add the training sequence like other parameters such as frequency hopping, TDM and time slot, below is the standardized TDM. [21]

Table (1) Training sequence of standardized TDM [21]

Bit Number (BN)	Length of field	Contents of field
0 - 2	3	tail bits
3 - 60	58	encrypted bits (e0 .. e57)
61 - 86	26	training sequence bits
87 - 144	58	encrypted bits (e58 .. e115)
145 - 147	3	tail bits
(148 - 156	8,25	guard period (bits)

And for physical channel they put standard for normal burst and for guard band

Table (2) Training sequence standards in normal burst [21]

Bit Number (BN)	Length of field	Contents of field
0 - 2	3	tail bits
3 - 60	58	encrypted bits (e0 .. e57)
61 - 86	26	training sequence bits
87 - 144	58	encrypted bits (e58 .. e115)
145 - 147	3	tail bits
(148 - 156	8,25	guard period (bits)

The GSM slots are the small specific time period that is found in all mobiles. Different types of data and information are required to be used in mobiles as trainmaster or receiver so that they have defined format. There are shorted transmission frames, the slots are normally used for transmitting 148 bits of data information. This data can be used for carrying voice data, internet, video call, and control and synchronization data.

This structure of GSM is not simultaneous and has time offset and this time offset is very useful to make the mobile transmitter or receiver at the same time and this will reduce the cost by decreasing the using of expansive filters.

3.4.1 GSM Burst

The GSM transmission can implement several of functions. Some GSM bursts are used for transceiver of data and others are occupies for control information. So the GSM bursts can be divided into different types as shown below;.

- Normal burst   uplink and downlink
- Synchronization burst downlink
- Frequency correction burst downlink
- Random Access (Shortened Burst)   uplink

3.4.2 GSM normal burst

This GSM burst is used for the standard communications between the base station and the mobile, and typically transfers the digitized voice call.

The template of the normal GSM burst is accurately defined and follows a Formula canonical as shown in figures 3-11. [22].

- 3 tail bits: These tail bits at the start of the GSM burst give time for the transmitter to ramp up its power
- 57 data bits: This block of data is used to carry information, and most often contains the digitized voice data although on occasions it may be replaced with signaling information in the form of the Fast Associated Control Channel (FACCH). The type of data is indicated by the flag that follows the data field
- 1 bit flag: This bit within the GSM burst indicates the type of data in the previous field.
- 26 bits training sequence: This training sequence is used as a timing reference and for equalization. There are a total of eight different bit sequences that may be used, each 26 bits long. The same sequence is used in each GSM slot, but nearby base stations using the same radio frequency channels will use different ones, and this enables the mobile to differentiate between the various cells using the same frequency.is used to adopt the parameters of the receiver to the current path propagation and to select the strong signal in case of multipath propagation.

- 8.25 bits guard time, at the end of the GSM burst there is a guard period. This is introduced to prevent transmitted bursts from different mobiles overlapping, as a result of their differing distances from the base station.

3.4.3 GSM synchronization burst

Figure 3-10 GSM Normal Burst

The purpose of this form of GSM burst, this is to provide synchronization between the MS and BTS in time [22].

- 3 tail bits: Again, these tail bits at the start of the GSM burst give time for the transmitter to ramp up its power
- 39 bits of information:
- 64 bits of a Long Training Sequence:
- 39 bits Information:
- 3 tail bits Again these are to enable the transmitter power to ramp down.
- 8.25 bits guard time to act as a guard interval.

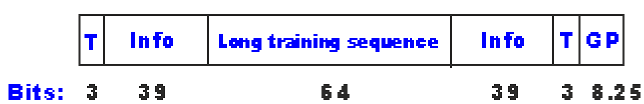


Figure 3-11 GSM synchronization burst

3.4.4 GSM frequency correction burst

With the information in the burst all set to zeros, the burst essentially consists of a constant frequency carrier with no phase alteration to allow the MS to correct the local oscillator to avoid the interference with neighboring channel.

- 3 tail bits: Again, these tail bits at the start of the GSM burst give time for the transmitter to ramp up its power.
- 142 bits all set to zero:
- 3 tail bits Again these are to enable the transmitter power to ramp down.
- 8.25 bits guard time to act as a guard interval.

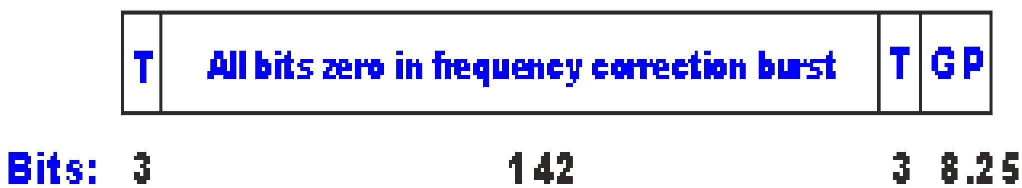


Figure 3-12 GSM frequency correction burst [22]

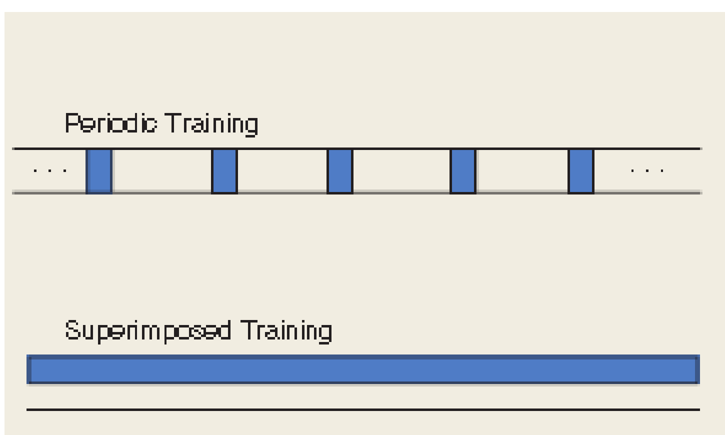


Figure 3-13 Differences between the periodic training and superimposed training Sequence

3.5 Methods for Adding Training Sequences

Super imposed training sequence can be used with OFDM system to prevent the inter symbol interference which appears in fading channels as a result of multipath fading , this sequence is a random sequence which does not have any information which added to the frame of data after passed through modulation mapped.

We have two methods to add the training sequence, the first one is the direct sum with the input data (algebraic summation) after added process is completed, the compacted signal enters the OFDM modulation and is transferred into channel, at receiving end the equalizer is used to remove the effect of training sequence and used as the desired input to equalized the received signal, method is defined as Superimposed training sequence [20].

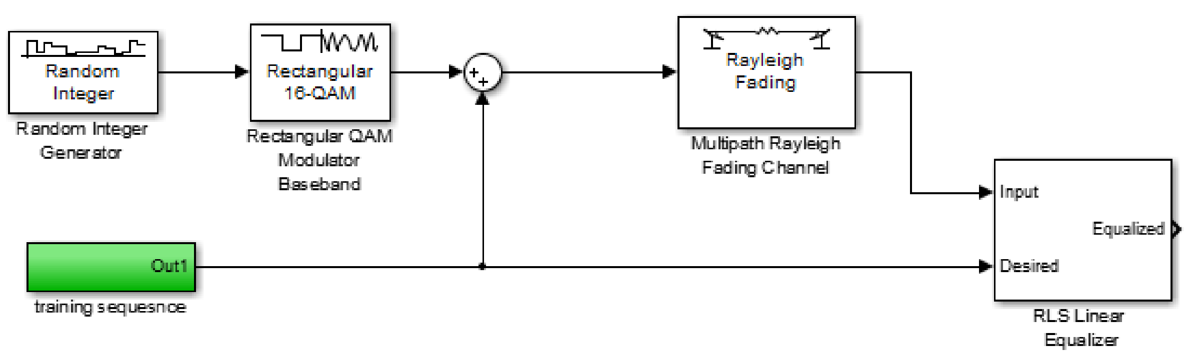


Figure 3-14 Algebraic summation between input data and superimposed training sequence

The second method is defined as conventional training sequence, it uses Matrix Concatenate to add the training sequence, the benefit of this method is to isolate the data frame form the affecting area during the transmutation time and make the insulator between the data frames.

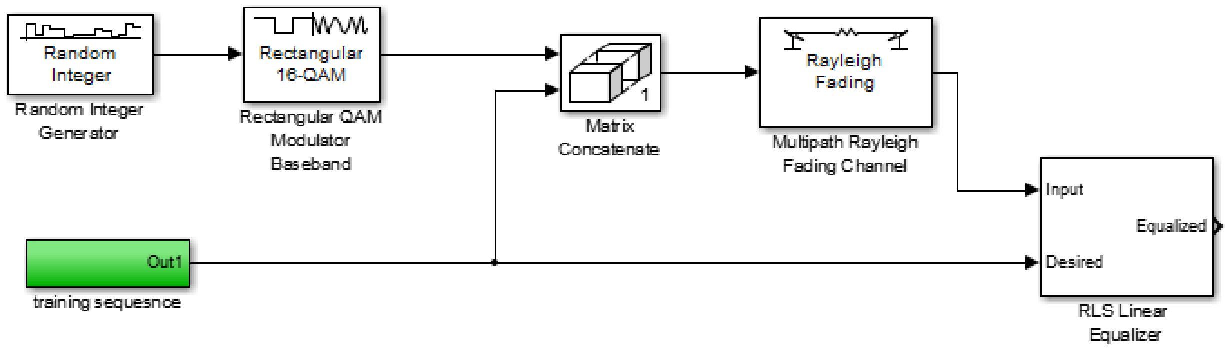


Figure 3-15 Matrix concatenate to collect the input data and ST

In this case, so far the conventional training sequence is similar to the cyclic prefix work but the training sequence has space more than the prefix, in the receiving end the demax tools are used to divided the received signal and terminate the training sequence frame to make the demodulation process more smooth.

CHAPTER FOUR

OFDM System

4.1 Introductions

this chapter introduces the main concepts of the OFDM communication system and discusses the addition parts that improve the work of OFDM system, The structure of the OFDM communication system is explained briefly; emphasis is being placed on how to implement it using Fast Fourier Transform (FFT) technique. The direct-conversion receiver architecture (which will be used later in this work) is also studied as related to OFDM system. The problem of wireless channel is the multipath fading which cause the fast varying in time and frequency domain.

4.2 Concepts of OFDM Communication System

OFDM is a special case of multicarrier transmission, where a single data stream is transmitted over a number of lower rate subcarriers. The basic principle of OFDM is to split a high data rate stream into a number of lower rate streams that are transmitted simultaneously over a number of subcarriers. One of the main advantages of using OFDM is to increase the immunity against frequency selective fading [19].[The word "orthogonal" indicates that there is an accurate mathematical relationship between the frequencies] which are carried to the subcarriers in the OFDM system, which ensures that the integral of the product of any two adjacent subcarriers is equal to zero. In a normal Frequency Division Multiplexing (FDM) system, the subcarriers are spaced apart in such a way that the signals can be received and demodulated. In these receivers, guard bands are introduced between the different subcarriers in the frequency domain which results in a lowering of spectrum efficiency. However, it is possible to arrange the subcarriers in an OFDM system so the sidebands of the independent subcarriers overlap, as shown in Figure 4.1. The signals in OFDM receiver are still received without adjacent carrier interference. To do this, the subcarriers must be mathematically orthogonal. To maintain orthogonal between subcarriers, it is necessary to ensure that the symbol time contains one or more multiple cycles of each sinusoidal carrier waveform [20]. In the case of OFDM, the sinusoids of subcarriers will satisfy this requirement since each is a multiple of a fundamental frequency. Orthogonality is critical since it prevents ICI. ICI occurs when the integral of the carrier products are no longer zero over the integration period, so signal components from one subcarrier cause interference to neighboring subcarriers.

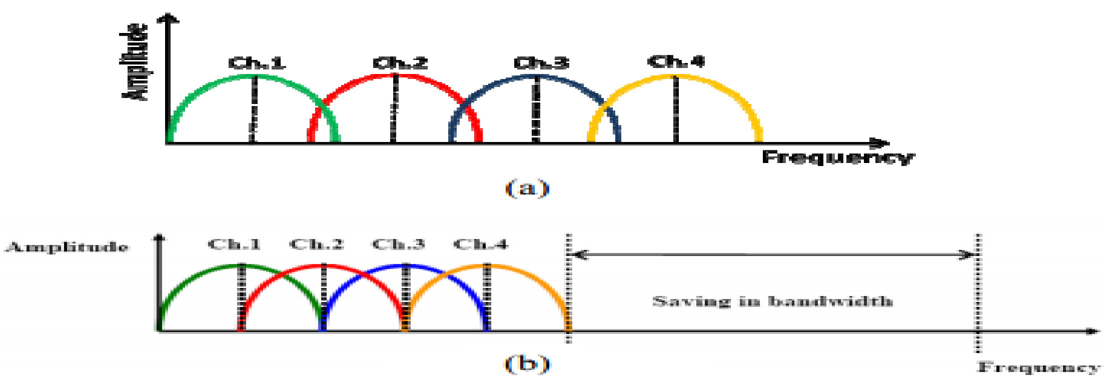


Figure 4-1 Concept of the OFDM signal orthogonal [19]

(a)Non-orthogonal subcarriers (b) orthogonal subcarriers.

Mathematically, the OFDM signal is expressed as a sum of the prototype pulses shifted in the time and frequency directions and multiplied by the data symbols. In continuous-time notation, the k_{th} OFDM symbol is written as [17]

$$s_{RF,k}[t - kT] = \left\{ R_{\circ} \left\{ w(t - kT) \sum_{t=\frac{1}{2}}^{\frac{L-1}{h}} d_{lk} e^{j2\pi\left[f_c + \frac{1}{T_{fft}}\right](t-kT)} \right\} \right\} \quad (4.1)$$

$$kT_{\circ}\text{-}T_{wtn} - T_{guara}$$

where

$s_{RF,k}$ RF received signal

k : index on transmitted symbol

T : OFDM symbol duration.

T_{fft} : FFT time: effective part of the OFDM symbol

T_{guard} : duration of the cyclic prefix

T_{win} : window interval

f_c : center frequency of the occupied frequency spectrum

F-$\frac{1}{T_{fft}}$: frequency spacing between adjacent SCs

L : number of OFDM subcarriers

d_{lk} : complex symbol modulated on the i_{th}SC of the k_{th} OFDM symbol

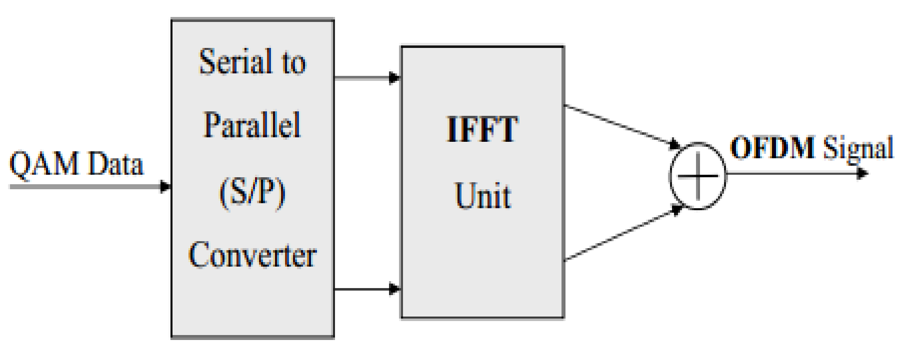


Figure 4-2 OFDM modulator [19]

The real and imaginary parts correspond to the in phase and quadrature parts of the OFDM signal respectively. They have to be multiplied by a cosine and sine of the desired frequency to produce the final OFDM signal. Equation (4.1) represents the general form of complex baseband OFDM signal. Figure (4-2) shows the block diagram for the OFDM modulator. The complex baseband OFDM signal, defined in equation (4.1), is the inverse Fourier transform of L QAM input symbols. The discrete time case is the Inverse Discrete Fourier Transform (IDFT). In practice, this transform can be implemented very efficiently by the Inverse FFT (IFFT). The IFFT drastically reduces the amount of calculations by exploiting the regularity of the operations.

4.3 OFDM Transceiver

Figure 4-3 shows the general OFDM transceiver system. In OFDM system design, the series and parallel converters are considered to realize the concept of parallel data transmission. In a conventional serial data system, the symbols are transmitted sequentially, where the frequency spectrum of each data symbol allow occupying the entire available bandwidth. When the data rate is sufficiently high, several adjacent symbols may be completely distorted over frequency selective fading or multipath delay spread channel. In the parallel data transmission system, the spectrum of an individual data element normally occupies only a small part of available bandwidth. Therefore, because of dividing an entire channel bandwidth into many narrow sub bands, the frequency response over each individual sub channel is relatively flat. A parallel data transmission system offers possibilities for alleviating this problem encountered with serial systems. The signal mapped converts the bits on each sub channel to their corresponding constellation; for example, one of the signal mapped types used in this thesis is the 16QAM modulation.

In OFDM modulator, it is necessary to predetermine the number of sub symbols to be available simultaneously at the inputs of the IFFT unit. For this reason the sequentially received data are temporarily stored, until the required number of sub symbols for parallel transmission has accumulated, and then they read out in parallel. After the IFFT unit, Parallel to Serial (P/S) converter converts the signal from the parallel form to the serial form. After that a Guard Interval (GI) will be added to the data sequence [25]. At the receiver side, the inverse operations are performed in the reverse order to produce the received bit stream.

4.3.1 Source

as described in the standard [17], the information bits must be randomized before the transmission. The randomization operation is used to reduce the potential of transmission of non-modulated subcarriers. The process of randomization is implemented on each burst of data on the downlink and uplink, and on each allocation of a data block (sub channels on the frequency domain and OFDM symbols on the time domain). In our case, instead of performing a randomization process, a binary source that produces random sequences of bits is used. The number of bits that are generated is specified to be frame-based and is calculated from the packet size required in each situation. The packet size depends on the number of transmitted OFDM symbols and the overall coding rate of the system, as well as the modulation alphabet. Equation (4.2) calculates the number of transmitted OFDM symbols in one frame. It depends on the total number of transmitted symbols,[$\mathrm{N_{sym}}$], which also includes the symbols used for the preamble, specified by $\mathrm{N_{train}}$:[17 and 25]

$$N_{ofdm} = N_{Tsym} - N_{train} \tag{4.2}$$

Furthermore, the total number of transmitted symbols is defined as

$$N_{STM} = \frac{T_{frame}}{T_{sym}} \tag{4.3}$$

In the formula, $T_{sym}$is the OFDM symbol time, and T_{frame} denotes the frame duration. The expression that defines T_{sym} as well as the possible values specified for the frame duration can be found in trainmaster part. Once the number of OFDM symbols is known, the number of bits to be sent by the source is calculated:

$$\text{S packet} = N_{ofdm} R N_{data} M_a. \tag{4.4}$$

Here, R represents the overall coding rate, N_{data} is the number of used data subcarriers, and $M_a$defines the modulation alphabet, which is specified by the number of transmitted bits per symbol.

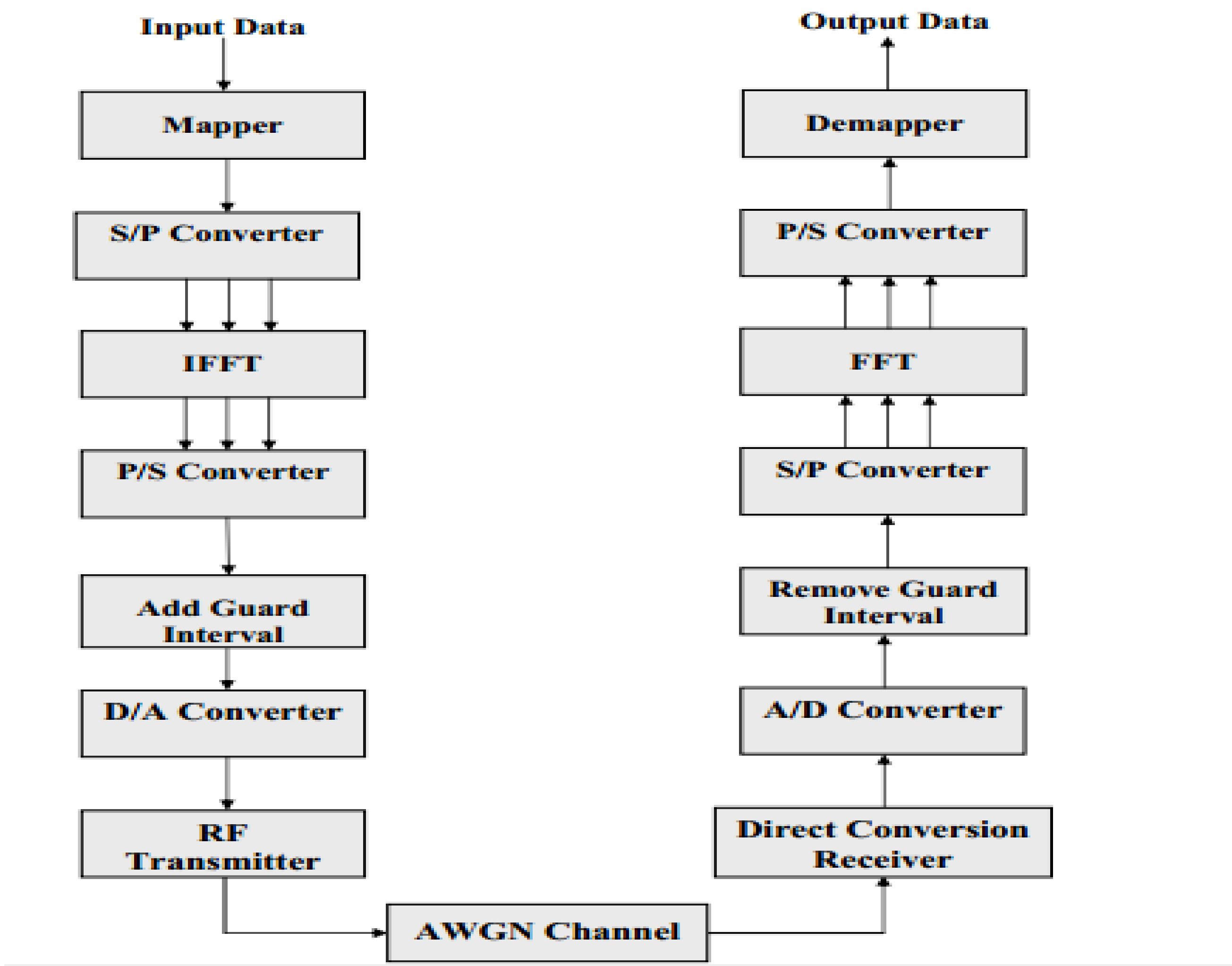


Figure 4-3 OFDM transceiver.[23]

4.3.2 Convolution Encoder

The data bits are encoded by a binary convolution encoder after the RS encoding process. The polynomial generated to derive its two input code bits, denoted X and Y, is specified in the following expressions:[25]

$$G_1 = 171_{OCT} \quad \text{for X} \tag{4.5}$$

$$G_2 = 133_{OCT} \quad \text{for Y} \tag{4.6}$$

A convolution encoder accepts messages of length k∘ bits and generates codeword's of n∘ bits. Generally, t is made up of a shift register of L segments, where L denotes the constraint length. The binary convolution encoder that implements the described code is shown in Figure 4.4 A connection line from the shift register feeding into the adder means a "one" in the octal representation of the polynomials; n_d no connection is represented by a "zero".

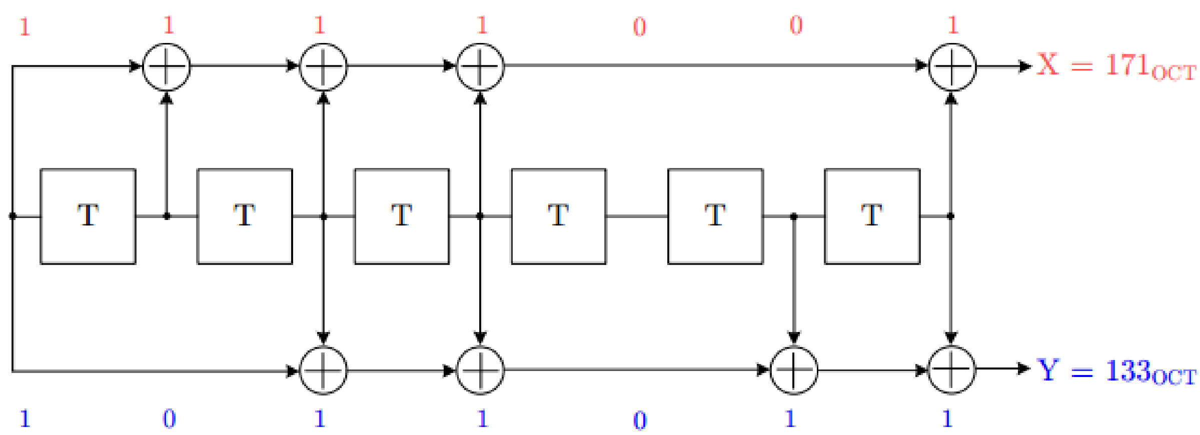


Figure 4-4: Convolution encoder of binary rate 1/2. [17]

4.3.3 Puncturing Process

The process of puncturing is used to create the variable coding rates needed to provide various error protection levels to the users of the system. Puncturing is the process of methodology which removes bits from the output stream of a low-rate encoder in order to decrease the volume of data to be transmitted, thus forming a high-rate code for the OFDM system. The bits are removed according to a perforation matrix.

4.3.4 Modulation mapper

Once the signal has been coded, it enters the modulation block. All wireless communication systems use a modulation scheme to map coded bits to a form that can be effectively transmitted over the communication channel. Thus, the bits are distributed to a subcarrier amplitude and phase, the (IQ) vector is complex in phase and in quadrature phase which represents the distributed bits.

The IQ figure for a modulation scheme explains the transmitted vector for data word group. Gray coding is a method for this specific figure as below so that modified points in the constellation only differ in one single bit. The Gray coding supports reducing the totally BER as it reduces the possibility of multiple bit errors that appear from a single symbol error. 2-PAM, 4-QAM, 16-QAM, and 64-QAM modulations are supported by the system. The support of the last one, the 64-QAM modulation, is optional for license-exempt bands. The constellation maps for 2-PAM, 4-QAM, and 16-QAM modulations are shown in Figure 4-5.

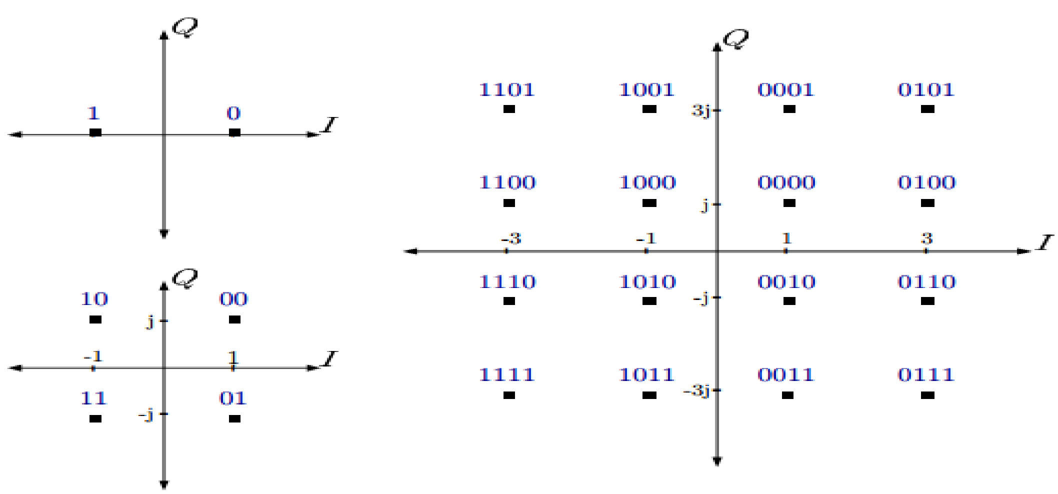


Figure 4-5: 2-PAM, 4-QAM and 16-QAM constellation maps. [24]

All points in the constellation are multiplied by right factor C_m to normalize and achieve equal average power symbol. Table 1 shows the values of C_m factor which achieves equal average symbol power, the constellations described above are normalized by multiplying all of their points by an appropriate factor Cm. Values of this factor C m are given in Table 1 represents the coordinate points in the constellation map of the symbol alphabet and is defined in Table 1.

Table 4-1: Modulation alphabet for the constellation map. [25]

Modulation scheme	Normalization constant for unit average power
2-PAM	$C_m = 1$
4-QAM	$C_m = 1/\sqrt{2}$
16-QAM	$C_m = 1/\sqrt{10}$
64-QAM	$C_m = 1/\sqrt{42}$

Modulation scheme	Symbol alphabet
2-PAM	$A_s = (1, -1)$
4-QAM	$A_s = (1 + j, 1 - j, -1 + j, -1 - j)$
16-QAM	$A = (j, 3j, -j, -3j)$ $A_s = (A + 1, A + 3, A - 1, A - 3)$
64-QAM	$A = (j, 3j, 5j, 7j - j, -3j, -5j, -7j)$ $A_s = (A + 1, A + 3, A + 5, A + 7, A - 1, A - 3, A - 5, A - 7)$

Furthermore, an adaptive modulation and coding mechanism is supported in the downlink with the purpose of allowing the number of transmitted bits per symbol to be varied depending on the channel conditions. A more detailed explanation of adaptive modulation as well as a description of how it is implemented in the simulator is given in Chapter 5.

4.3.5 Pilot Symbols

Pilot sequence can be used to implement frequency offset on the receiver end. Additionally, as recent results showed [23], the pilots sequence is used especially for fast fading channel. This kind of mapping is given by the operations $1 - 2\mathrm{w_k}$, where $\mathrm{w_k}$ is the sequence produced by the PRBS generator, and $\mathrm{w_k}$ denotes the binary inversion. The indices represent the subcarrier numbers where the pilots are going to be inserted:

$$p_{-88} = p_{-38} = p_{63} = p_{88} = 1 - 2w_k \tag{4.7}$$

$$p_{-63} = p_{-13} = p_{13} = p_{38} = 1 - 2\overline{w}_k \tag{4.8}$$

The initialization sequences for the PRBS generator vary depending on the direction of transmission, i.e. the downlink or the uplink. A sequence of all "ones" is used in the downlink while a sequence of alternated "ones" and "zeros", being the first bit equal to "one", is used in the uplink.

4.3.6 Guard Interval

One of the significant properties of OFDM transmissions is the solidity against multipath delay spread. In order to eliminate the Inter Symbol Interference (ISI) almost completely and to preserve the orthogonality among subcarriers, a guard interval which is known as "Cyclic Prefix" is introduced for each OFDM symbol [17, 25].

Figure 4-6 shows the cyclic prefix location, it is described as a mirror, for a part of the end of the symbol waveform is put at the start of the symbol as the guard interval. This effectively extends the length of the symbol, while confirming the orthogonality of the symbol waveform. This cyclic extended symbol is used to meet the requirements to perform the FFT that can take any location over the frame length of the symbol.

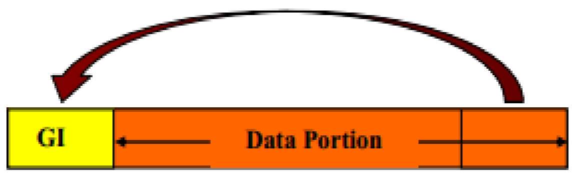


Figure 4-6 Symbol extension with guard interval.[25]

Another type of the guard intervals is zero padding [25]. In zero padding, a number of zero samples are added to the front of each symbol to combat ICI. The number of these zero samples depends on the channel delay of each symbol. Other types of guard intervals are possible. One type is to have half of the period in a cyclic prefix symbol, as in cyclic prefix type, and the other half is zero padded.

4.3.7 Assembler

OFDM system specifies for the 64-point FFT three types of subcarriers; data, pilot and null, as shown in Figure 4-7. 52 of the total 80 subcarriers are used for data and pilot subcarriers, of which pilots are permanently spaced throughout the OFDM spectrum. The remaining 16 active carriers take up the data subcarriers. The rest of the potential carriers are nulled and set aside for guard bands and removal of the center frequency subcarrier.

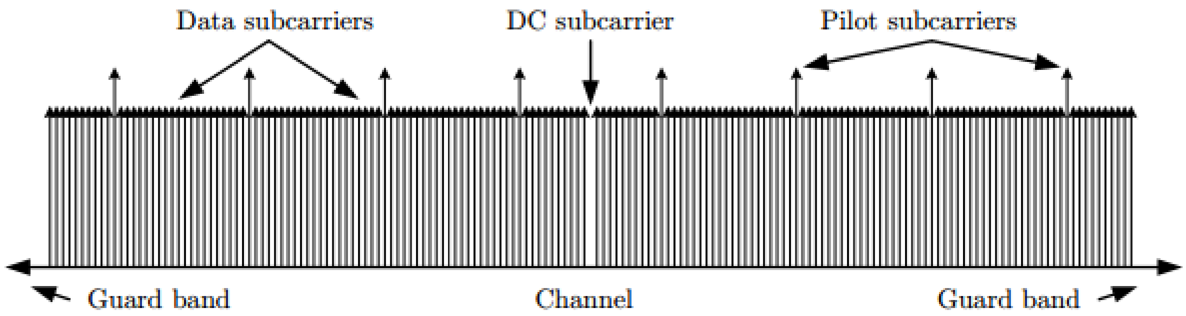


Figure 4-7: OFDM frequency description.[25]

In order to construct an OFDM symbol, a process to rearrange these carriers is needed. for this purpose, the assembler block is inserted in the simulator. It performs this operation in two steps by first inserting the pilot tones and the zero DC subcarrier between data with a process of vertical concatenation, and then appending the training symbols at the beginning of each burst in a horizontal way, as shown in Figure 4.7. It is shown that while the first step performs a concatenation in the frequency domain, the second step does it in the time domain.

4.4 Transmission Channel:

- **Importance of multipath delays**

Data is already transmitted in the time domain as a sequence of symbols. Each symbol has a set duration, for example at 1 Mbaud, each symbol lasts 1 μS.

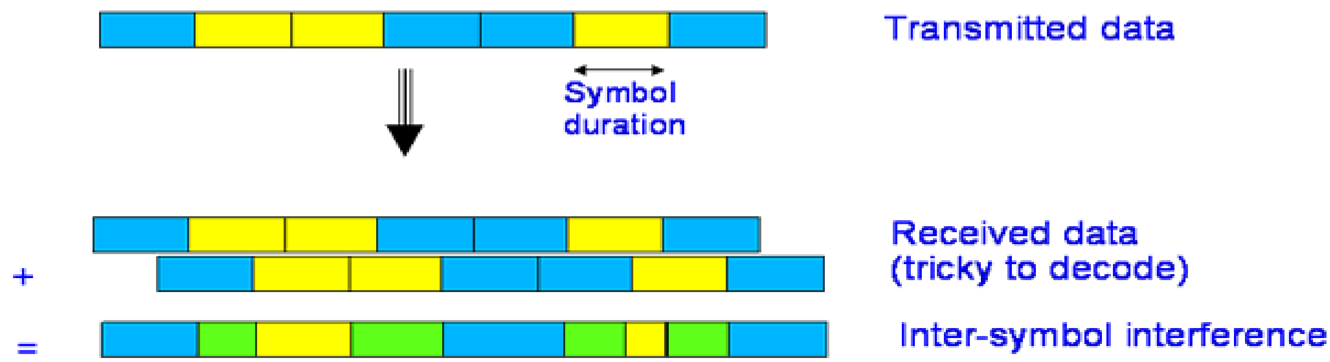


Figure 4-8: Data frame structure [21]

Delay spread causes the Bit error rate which decreases with increased SNR. Also the increase in power will increase the error rate.

The standard of error in digital system in general is about 10^{-12} or more ,and this system can achieve the error rate by using the digital coding techniques and using other techniques to prevent the inter symbol interference ISI and channel distortion in case of having fast varying time channel [23].

- **The tapped delay line model**

In general when work is on multipath channel, the sequence of delayed amplitude will represent the model of multipath.

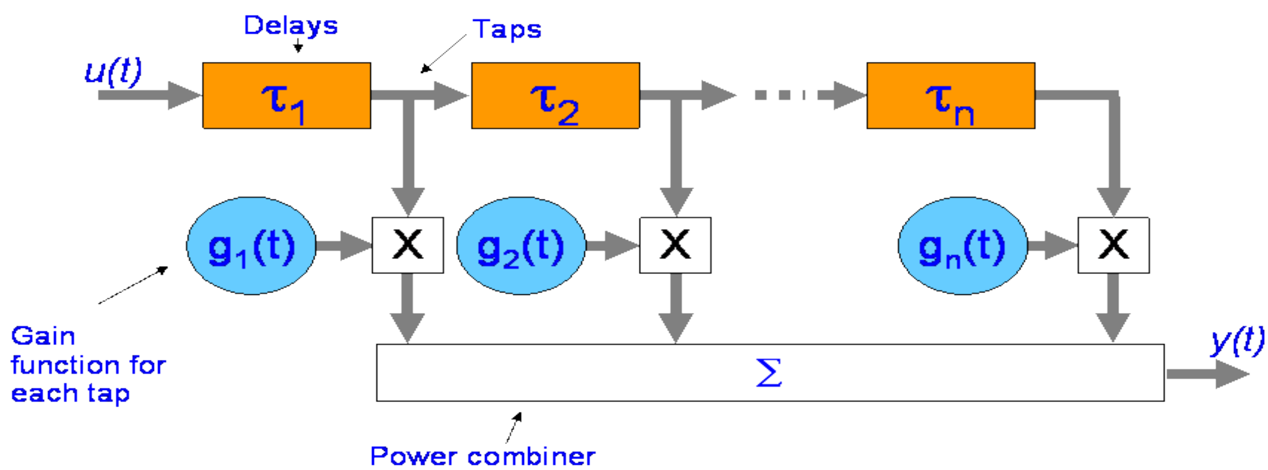


Figure 4-9: Taps delay diagram [27]

All the taps are uncorrelated. The delays $\tau_1 \& \tau_2 \&, \& \tau_3 \ldots \& \tau_n$ are not constant, but function in correspondence to the delay spread.[27]

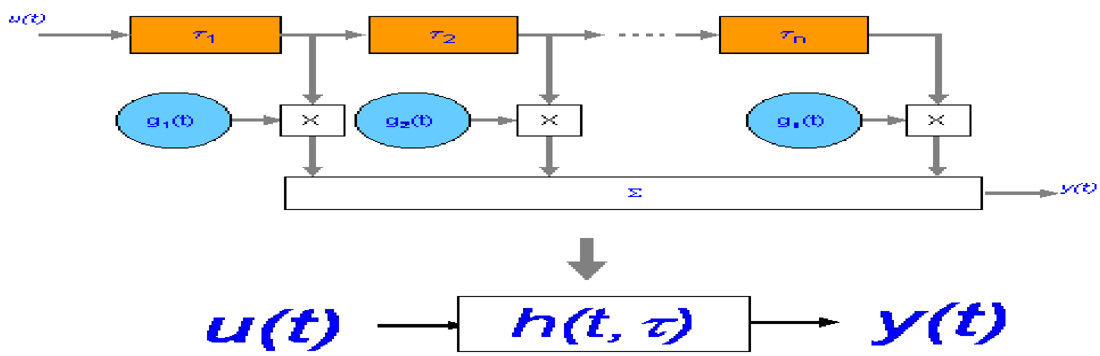


Figure 4-10: Impulse response diagram [27]

The impulse response $h(t, \tau)$ is defined as the delay spread function

$$y(t) = u(t) * h(t, \tau) = \int_{-\infty}^{\infty} h(t, \tau) u(t - \tau). d\tau \quad (4.9)$$

- **Defining delay spread**

Figure 4.11 shows a set of definitions of power gain and the relationship with time delay.

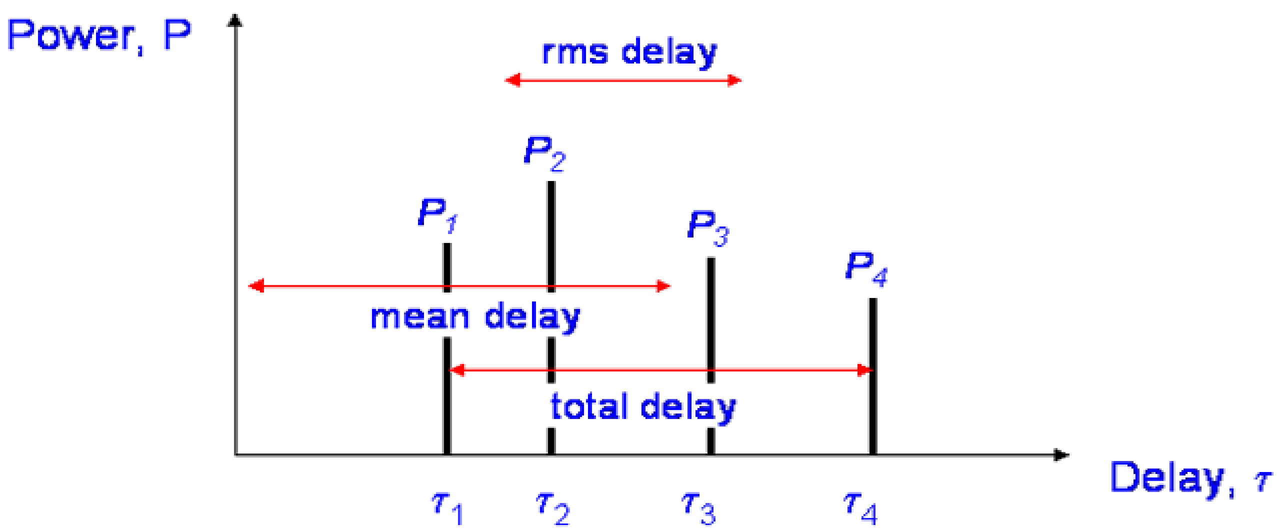


Figure 4-11: Power VS time delay [25]

The received signal, $p(t)$ is made up of N taps $p_1(t)$to $p_N(t)$

Excess delay is the delay of tap relative to the first tap.

Total delay is the delay difference between first and last tap.

Total power is the sum of all the tap powers.

$$P_T = \sum_{i=1}^{N} P_i \qquad (4.10)$$

Mean delay, is the average delay weighted by power

$$\tau_\circ = \frac{1}{P_T} \sum_{i=1}^{N} P_i \qquad (4.11)$$

The rms delay spread

$$\tau_{rms} = \sqrt{\frac{1}{P_T}\sum_{i=1}^{N} P_i \tau_i^2 - \tau_\circ^2} \qquad (4.12)$$

In many environments the rms delay spread is already found. Almost all modern demodulators can de-spread, cancel and make control of the multipath components in the input signal. RAKE receivers are an example of this techniqus16].

- **Delay spread Example**

The table below gives some typical values at 1-2 GHz for GSM mobile systems. ISI occurs when the symbol duration is similar to delay spread. For reference, the Urban and Hilly environment has long time delay so all systems very careful when the communication system is designed in this area.

Table (4.2) rms delay spread for varies environments

Environment	rms **delay spread** (uS)
Indoor	0.01-0.05
Mobile satellite	0.04-0.05
Open area	< 0.2
Suburban microcell	< 1
Urban microcell	1-3
Hilly area microcell	3-10

- **Small scale fading due to multipath.**

A. Spreading in Time: Different paths have different lengths;

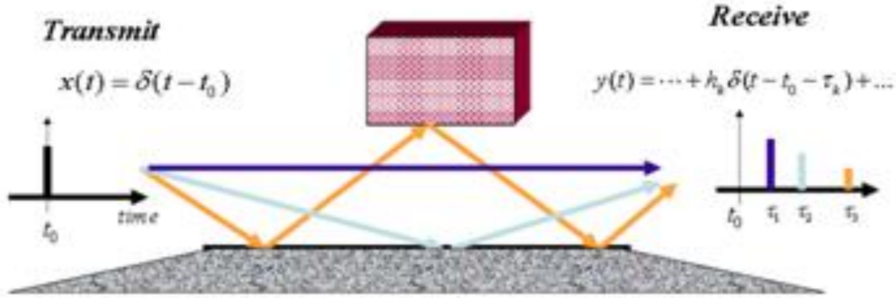


Figure 4-12 the multipath components [27]

Example of 100m path difference we have a time delay

$$\tau = \frac{100}{c} = \frac{10^2}{3 \times 10^8} = \frac{1}{3}\ \mu\, sec$$

B. spreading in frequency: motion causes frequency shift (Doppler)

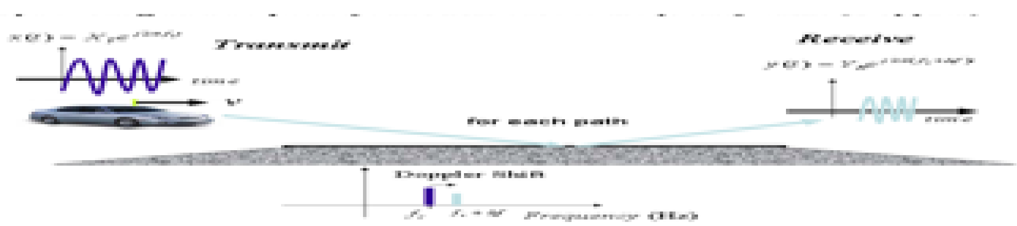


Figure 4-13 The frequency spreading [27]

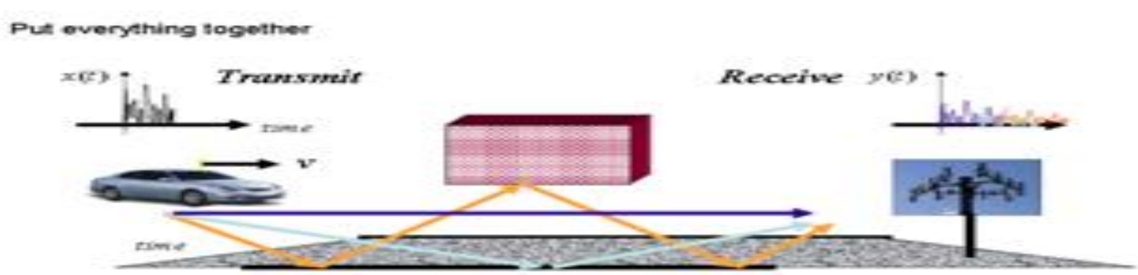


Figure 4-14 The frequency spreading on multipath [27]

$$y(t) = Re\left\{\sum a_k x(t-\tau_k) e^{j2\pi(F_k+\Delta F_k)(\tau-\tau_k)}\right\} \quad (4.13)$$

$where\ a_k$ is the attenuation.

τ_k is the shift in time

ΔF_k is the shift in frequency

- This causes small scale time variation

$$y(t) = Re\left\{\left(\sum a_k\, e^{j2\pi\Delta F_k t}\, e^{-j2\pi(F_c+\Delta F_k)\tau_k} x(t-\tau_k)\right) e^{j2\pi F_c t}\right\}$$

$$= Re\left\{r(t)e^{j2\pi F_c t}\right\} = r_I(t)\cos(2\pi F_c t) - r_Q(t)\sin(2\pi F_c t)$$

Where $r_I(t)\cos(2\pi F_c t)$ Represent the in phase components

$r_Q(t)\sin(2\pi F_c t)$ Represent the in quadrature components

$$r(t) = r_I(t) + jr_Q(t) \cong \sum_k a_k\, e^{j2\pi\Delta F_k t}\, e^{-j2\pi(F_c+\Delta F_k)(\tau_k+\varepsilon_k)} x(t-\tau_k)$$

Assume $x(t) \cong x(t-\varepsilon_k)$

Leading to $r(t) = C_l(t)x(t-\tau_l)$

$$c_l(t) = \sum_k a_k\, e^{j2\pi\Delta F_k t}\ e^{-j2\pi(F_c+\Delta F_k)(\tau_k+\varepsilon_k)} \quad (4.14)$$ random, time varying

Statistical model for the time varying coefficients

$$c_l(t) = \sum_{k=1}^{M} a_k e^{j2\pi\frac{v}{\lambda}\cos\theta_k t}\ \ e^{-j2\pi\left(F_c+\frac{v}{\lambda}\cos\theta_k\right)(\tau_l+\varepsilon_k)}$$

By the CLT $c_l(t)$ is Gaussian with

$E\{c_l(t)\} = 0$ Since θ_k random uniformly distributed $[0,2\pi]$

$$E\{c_l(t)c_l^*(t+\Delta t)\} = \sum_{k=1}^{M}\left\{|a_k|^2\right\} e^{-j2\pi\frac{v}{\lambda}\cos\theta_k \Delta t}$$

Assume $E\{|a_k|^2\} = \frac{P_l}{M}$

$$E\{c_l(t)c_l^*(t+\Delta t)\} = \frac{P_l}{M}\sum_{k=1}^{M} e^{-j2\pi\frac{v}{\lambda}\cos\theta_k \Delta t}$$

$$= P_l E\left\{e^{-j2\pi\frac{v}{\lambda}\cos\theta_k \Delta t}\right\} \quad (4.15)$$

Each coefficient $c_l(t)$ is complex, Gaussian, WSS with autocorrelation

$$E\{c_l(t)c_l^*(t+\Delta t)\} = P_l J_\circ(2\pi F_D \Delta t)$$

$$S_l(F) = \begin{cases} \frac{2P_l}{\pi F_D}\frac{1}{\sqrt{1-(F/F_D)^2}} & |F| < F_D \end{cases} \quad (4.16)$$

where F_D is the maximum Doppler shift.

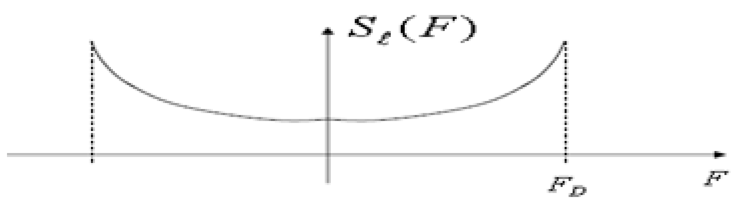


Figure 4-15 Modulation of multipath channel components [27]

Parameters for a Multipath Channel (No Line of Sight):

Time delays:	$\left[\tau_1 \quad \tau_2 \quad \cdots \quad \tau_L\right]$	**sec**
Power Attenuations:	$\left[P_1 \quad P_2 \quad \cdots \quad P_L\right]$	**dB**
Doppler Shift:	F_D	**Hz**

The following Matlab program explains the difference between the flat fading channel and frequency selective channel and compares between them in terms of signal constellation and frequency spectrum:

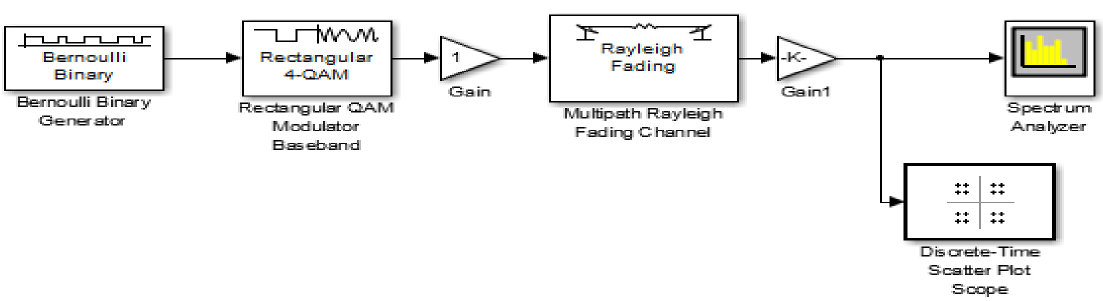


Figure 4-16: flat fading channel simulink

Case one: Flat fading

Channel properties:

Time delays T= [0 $10e^{-6}15e^{-6}$] sec

Power P= [0,-3,-8] dB

Symbol Rate FS= 10 kHz

Doppler Fd = 0.1 Hz

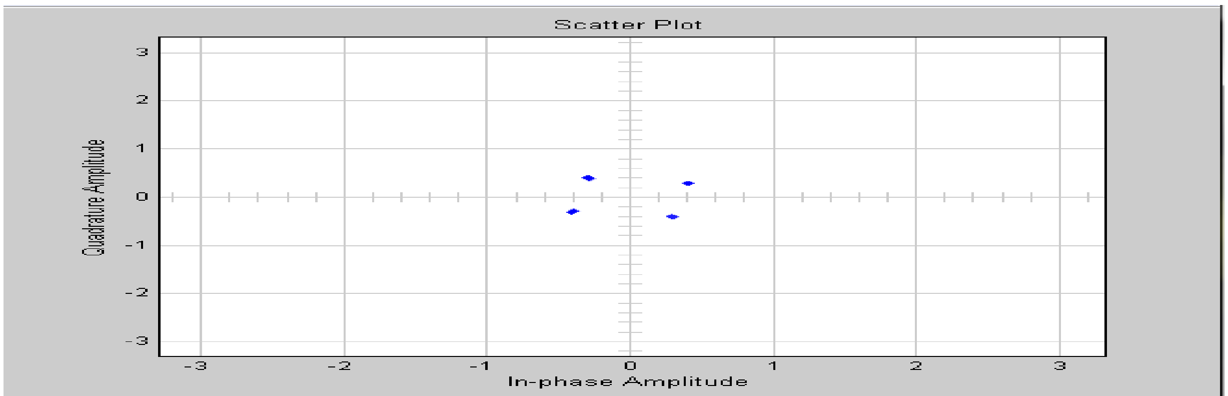


Figure 4-17: Very low inter symbol interference

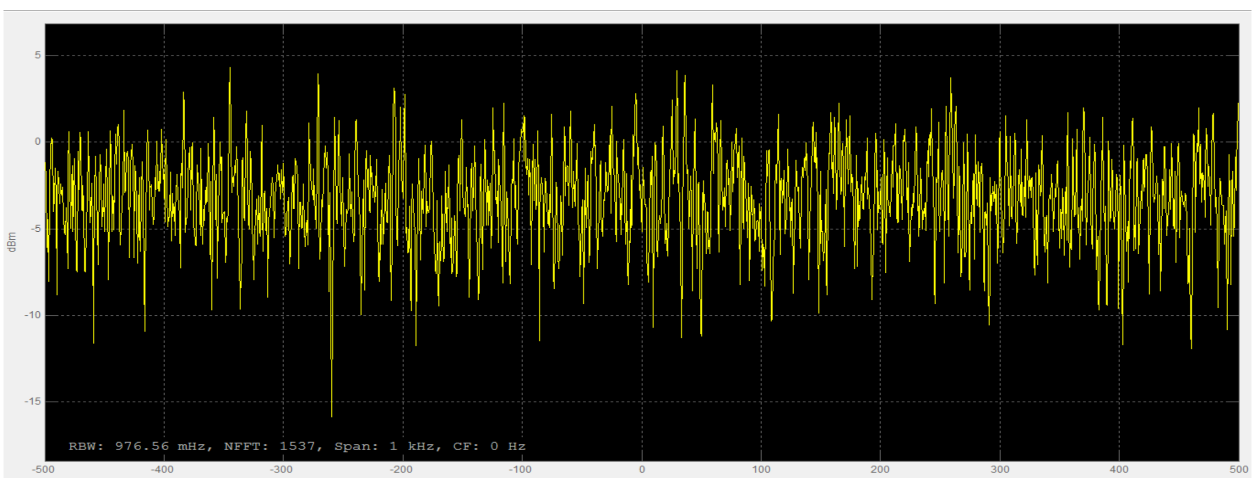


Figure 4-18 Channel spectrum: fairly uniform

Case two: Frequency selective fading

Channel properties:

Time delays T= [0 $10e^{-6}15e^{-6}$] sec

Power P= [0,-3,-8] dB

Symbol Rate FS= 1 MHz

Doppler Fd = 0.1 Hz

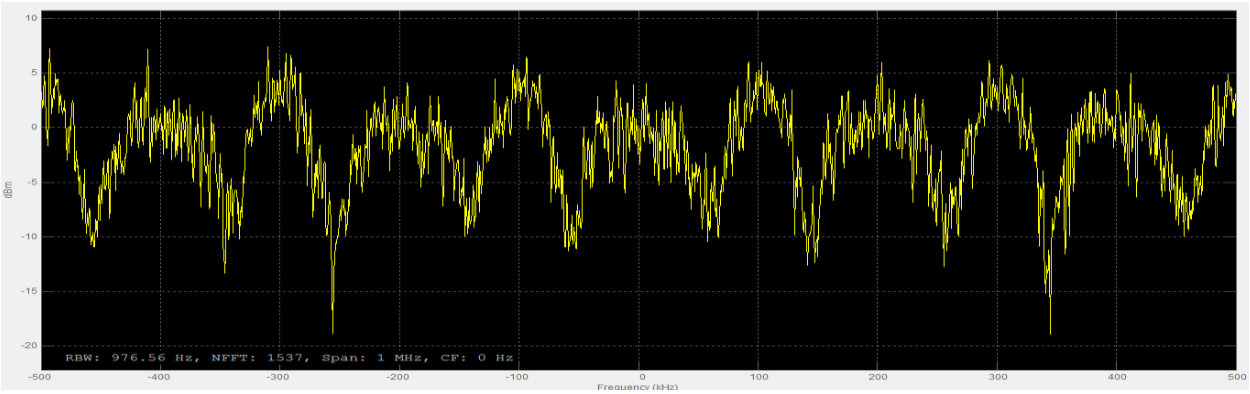


Figure 4-19: Spectrum with deep variations

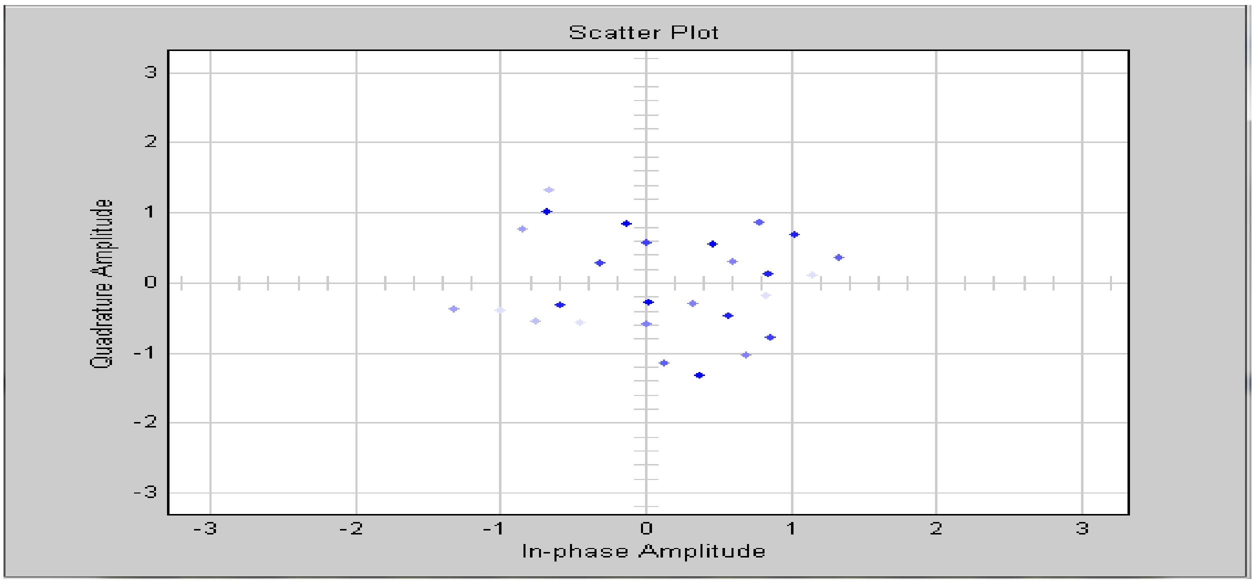


Figure 4-20: High ISI for frequency selective channel

- **Configure channel objects**

In Rayleigh fading channel , there are three topics which specify the behavior of the channel and specify the type of this channel, so any researcher will set the values he needs by more care to show the channel properties . The topics are:[26]

- **Path Delays**

By standardization, the first time delay is typically set to zero. The first delay represents the first path reached to the receiver. [26]

For indoor area, path delays after the first are between 1 ns and 100 ns.

For outdoor area, path delays after the first are typically between 100 ns and 10 μs .such as the areas surrounded by mountains.

The ability of a signal to resolve paths depends on its bandwidth. If the difference between the largest and smallest path delays is less than 1% of the symbol period, then the signal experiences the channel as if it had only one discrete path.[26]

- **Average Path Gains**

The average path gains in the transmission channel parameters indicates the average power gain of each fading path., the computer standard model uses average path gains between -20 dB and 0 dB to represent the different path gains..

The dB values in a vector of average path gains often slope linearly as a function of delay, and delay depends on the propagation environment.[27]

- **Maximum Doppler Shifts**

In GSM systems standard, Doppler shifts is a term of the speed of the mobile. If the mobile moves at speed v (m/s), then the Doppler shift is calculated as shown in Equation 2.6.

When the car moves in the open area it might represent a maximum Doppler of about 80 Hz, when the transmitter moves by it will represent about 4 Hz. A 0 value of maximum Doppler shift represent a constant channel that comes from a Rayleigh .[29]

- **4.4.2 Work with Delays and Channel Properties**

In the next page one Matlab code will write be implanted to explain the channel parameters after modify the channel properties to explain the effect of multipath when set the values for fast fading channel.

For example, the code below shown a random signal passing through a three-path Rayleigh channel:

```
% Three-Path Rayleigh channel

h = Rayleigh chan (1/100000, 130, [0 1.5e^-6 3.2e^-6], [0, -5, -8]);

tx = randi([0 1],500,1);        % Random bit stream

hmod = comm.DBPSKModulator;     % Create DBPSKModulator

dpskSig = step(hmod,tx);        % DPSK signal

h.StoreHistory = true;          % Allow states to be stored

y = filter(h, dpskSig);         % Run signal through channel
```

- **Impulse Response (IR).**

Figure 4-21 shows the magnitude of two impulse responses: the multipath response (three peak bandwidth) and the band limited channel response.

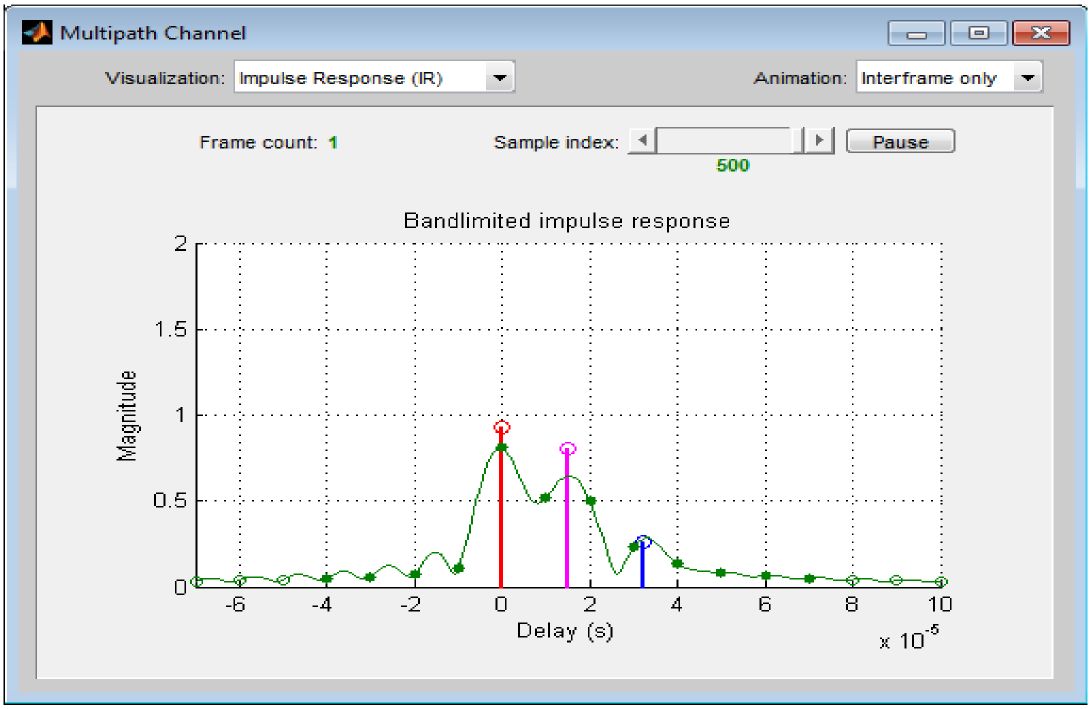


Figure 4-21: Channel impulse response

Each multipath component is represented by stems of the smallest delay value represented by red color and the biggest is represented by blue and the middle value is represented by rose color and the band limited channel response is represented by the green color curve.

The convolving result of multipath response is shown in Figure 4.17 above with a sinc pulse of period, (T), which is equal to the input signal sample period.

The output of the channel filter is the convolution of the input signal (sampled at rate 1/T) with this discrete-time channel filter response .The solid green circles represent the channel filter response sampled at rate (1/T). For computational speed, the response taken capture for a period of time.[26]

The hollow green color circles represent sample values not captured in the channel filter response that is used for processing the input signal.

- **Frequency Response (FR).**

Figure 4-22 shows the magnitude of the frequency response of the multipath channel over the signal bandwidth.

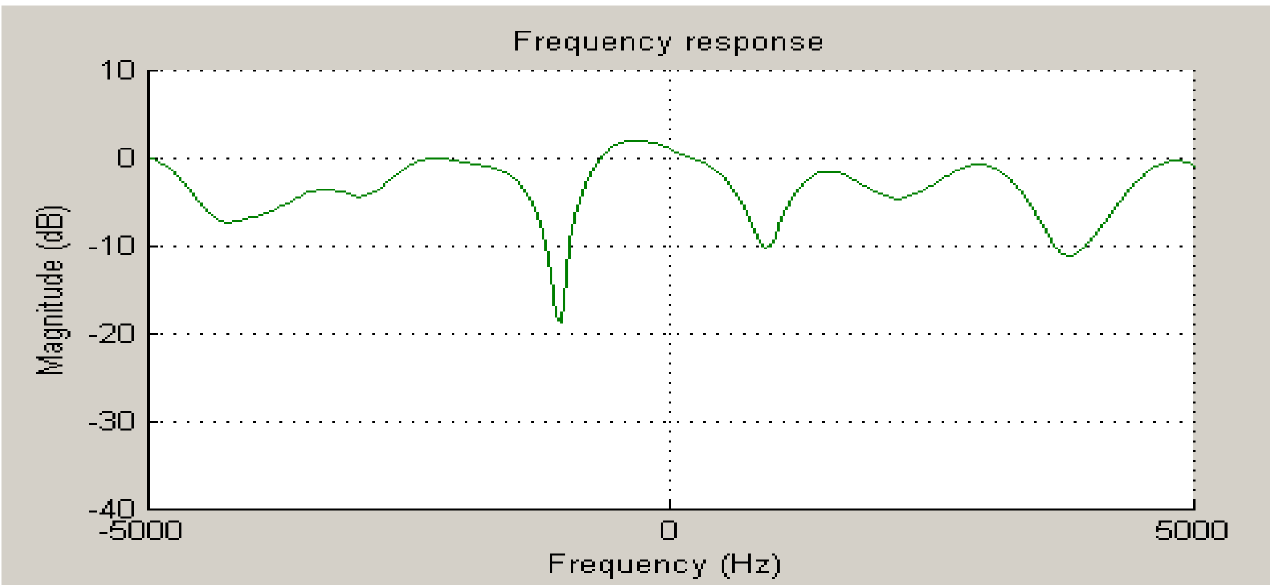


Figure 4-22: Frequency response

Many fluctuations appear in the above plot and these fluctuations decrease with change in the channel parameter to slow fading or change in the environments to open areas.

- **IR waterfall**

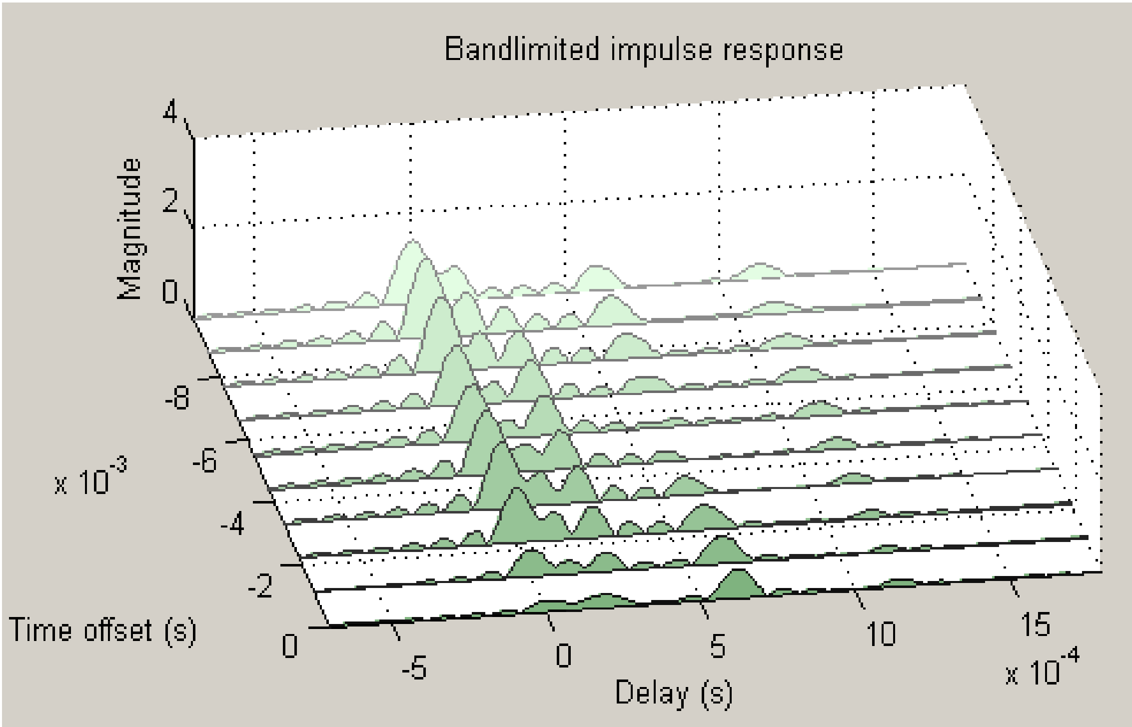


Figure 4-23: IR waterfall response

Figure 4-23 shows the development of the magnitude impulse response over period of time, the darkest part represents the current response and 10 captures represent the band limited channel impulse response within the last frame.

- **Multipath Gain.**

Figure 4-24 shows the total Bandwidth gains for the three multipath channel

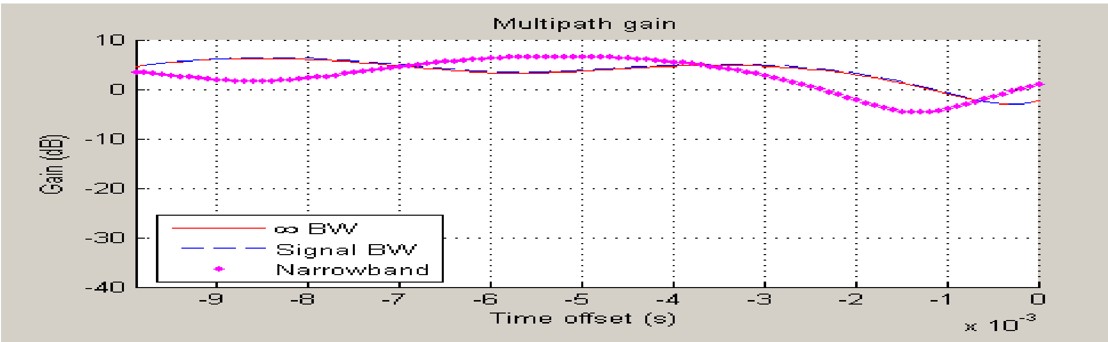


Figure 4-24: Multipath gain

A total gain is the sum of three component magnitudes, as explained in the discretion below:

Narrowband fading envelope is represented by magenta color, current signal bandwidth by (dashed blue line): This is the summation of the magnitudes of the channel filter samples (the solid green). This curve represents the maximum signal energy that can be captured in the receiver. Its value like as BER, is derived from its Infinite bandwidth represented by solid red line, this is the total magnitudes of the multipath fading component gains.

In general, the relationship between the multipath gain and the bandwidth is reciprocal, the fading gain decrease when bandwidth is increased,. If the signal bandwidth curve strongly follows the narrowband curve, you might describe the signal as narrowband. The wideband fading can be obtained when the signal bandwidth curve directly follows the infinite bandwidth curve.[26]

- **Doppler Spectrum**

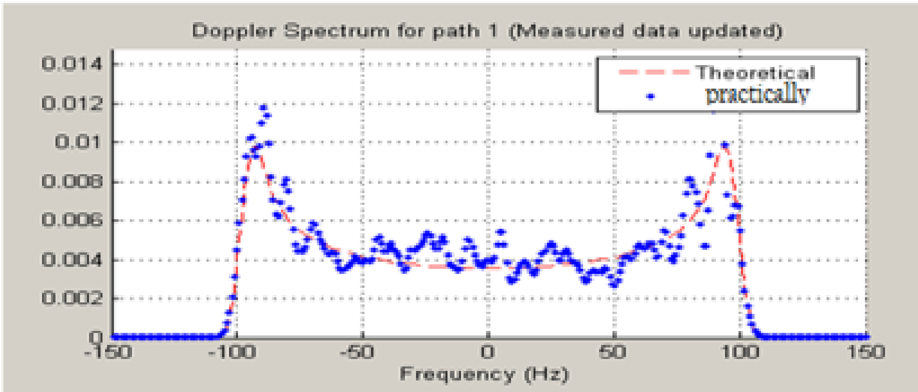


Figure 4-25: Doppler spectrum

Figure 4-25 shows the two Doppler spectrum , the first one is explained by dashed red which represents the theoretical values of Doppler response used in fading channel , the second one is explained by blue dots and it represents the measurement value of Doppler spectrum , the advantage of these measurements appear only when enough path gains generated.

Jakes spectrum was used in the plot of theoretical spectrum and the total power is 1 after normalizing the Doppler spectrum and the capture is taken after the program code is paused to shows the figure.

The best results can be achieved when very near approximation value is used for Doppler spectrum to the theoretical values, in the case the channel model has generated fading gains to allow to represent the channel properties with good representation especially multipath fading channel, such as BER which can give accurate measurements when the specific values an used for maximum Doppler shift and processed enough samples to obtain good results.

4.5 Receiver

As shown in Figure 4-25, the typical receiver implements the opposite operation of the transmitter, a channel estimation is necessary to discover the unknown channel coefficients. This section explains the different steps implemented by the receiver to recover the transmitted bits.

The first step is the cyclic prefix in which removed and received signal is converted to the frequency domain. After that the received signal passes through dissembler to remove the guard band, Zero DC and pilots sequence and prepare the signal for the demodulation, the training sequence is used in the channel estimator with equalizer to calculate the channel coefficients. The channel coefficients can be used in the damper to implement equalization of the data, and so, indemnity the frequency selective fading of the multipath fading channel.

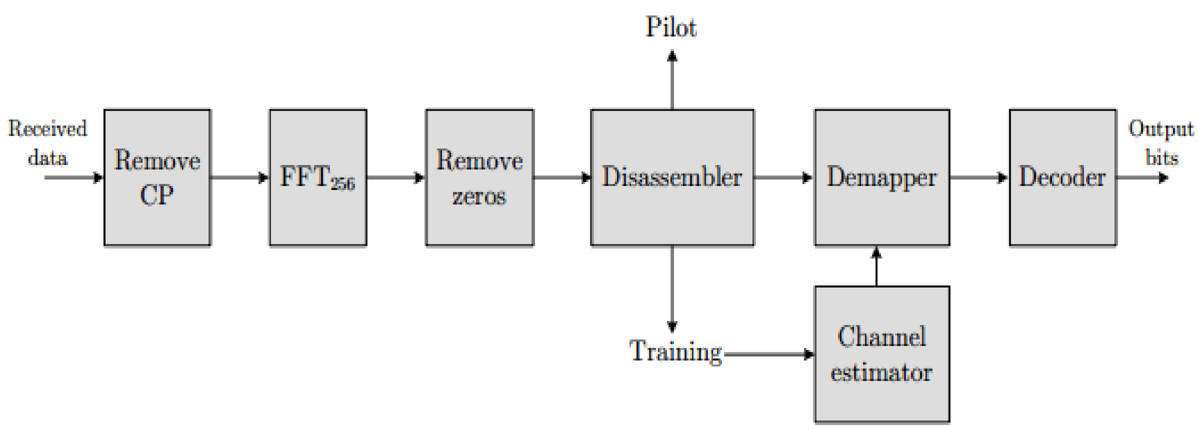


Figure 4-27: Receiver of the OFDM system.[25]

4.5.1 Fast Fourier Transform Algorithm

The IFFT algorithm represents a rapid way for modulating a group of subcarriers in parallel. Either the FFT or the IFFT is a linear pair of processes, therefore the FFT is necessary to convert the signal again to the frequency domain1.

4.5.2 Removing the Guard Bands

When removing the subcarriers that corresponds to the guard bands, the frequency structure has to be taken into account. Although zero padding acting as guard band is appended at the end of the subcarrier structure at the transmitter, a rearrangement of this subcarriers is performed when doing the IFFT operations which removed from the center of the OFDM symbol.

4.5.3 Dissembler

When the received signal comes out from the OFDM demodulation, the data is separated by Dissembler into two parts: desired data, training sequence, the superimposed training sequence can be used into two parts: desired data and pilots sequence when pilots sequence are used.

4.5.4 De-maping

At the receiver, the de-maping provides the interface between the transmission channel and the functions that compute and deliver estimates of the transmitted data bits to the user. Furthermore, the de-maping operates on the waveform that is received in each separate transmission symbol interval and produces a number or a set of numbers that represent an estimate of a transmitted binary or M-ary symbol.

Thus, the de-mapped methods are used for decision metrics with the aim of making a decision about which bit, "zero" or "one", is transmitted. This decision metric can be as simple as hard decision, or more sophisticated, being then a soft decision. Hard de-mapped methods output a hard decision as a function of the input, and this form of output is application-dependent. [27]

4.5.5 Bit Error Rate

The error rate calculation block calculates the bit error rate, by comparing the received data with transmitted data.

It has three inputs, Tx and Rx ports that are used to accept transmitted and received signals and the third port is used to indicate the related frame for computation.

CHAPTER FIVE

Simulation and Results

5.1 Introduction

This chapter gives the simulation results and evaluation of tests on the effect of superimposed training sequence in OFDM systems. The adaptive algorithm of a Recursive last sequence estimation, least mean square and training sequence in OFDM system, introduced in chapter three and four is implemented here using MATLAB environment. Because the parameter estimation depending on structures of the received signals (such as, conventional training sequence), this technique is applicable to a wide range of present and future OFDM based communications standards. The accuracy of the OFDM parameter estimation and consequently the performance of the compensation an arbitrarily scalable, allowing for a flexible tradeoff between performance, computational effort and measurement time. The performance of the simulated systems is evaluated for three modulation techniques BPSK, QPSK and 16QAM. The simulation results are performed using MATLAB (R14). The results are documented for four scenarios; the first is the OFDM with and without equalization, the second is the conventional training sequence with adaptive equalizer, and the third is superimposed training sequence with adaptive equalizer.

By using a simulation program in MATLAB 2014b (R14), the effect of the superimposed training sequence and conventional training sequence parameters will be discussed by employing the signal constellation diagrams, Bit Error Rate (BER) characteristics and the power spectrum for the transmitted and received signals.

Table 5.1 Parameter values used in the simulation of IEEE 802.11a

Parameter	**Specifications**
FFT Size	64
Number of Subcarriers	52
Pilot Ratio	12
Training Sequence	26 and 52
Guard Length	25% (16 samples)
Guard Type	Cyclic Extension
Wireless Channel	Rayleigh Fading Channel + AWGN
Signal Constellation	BPSK,QPSK and 16-QAM
Equalizer Types	Adaptive Equalizer (LMS) Adaptive Equalizer (RLS)

Table (5.1) shows the parameter values which a used in OFDM system according to IEEE 802.11a standards. The model supports all of data rates. The model implements adaptive modulation and coding over dispersive multipath fading channels. Note that the model uses an artificially high channel fading rate to make the data rate change further rapidly and thus make the imagining more animated and useful.

The model includes components that model the important features of the WLAN 802.11a standard, the right side row of blocks comprises the transmitter components though the left side comprises the receiver components. The communication system in this model performs these tasks: Generation of random data at a bit rate that varies during the simulation. The varying data rate is completed by enabling a source block periodically for a period that depends on the favorite data rate. Coding and modulation using one of numerous systems specified in the standard BPSK, QPSK, and 16-QAM modulation. OFDM (orthogonal frequency division multiplexing) carries out transmission using 52 data subcarriers, training sequence, 64-point FFTs, and a 16-sample and cyclic prefixes, dispersive multipath fading channels and receiver equalization and Viterbi decoding.

5.2 OFDM System Without Equalization

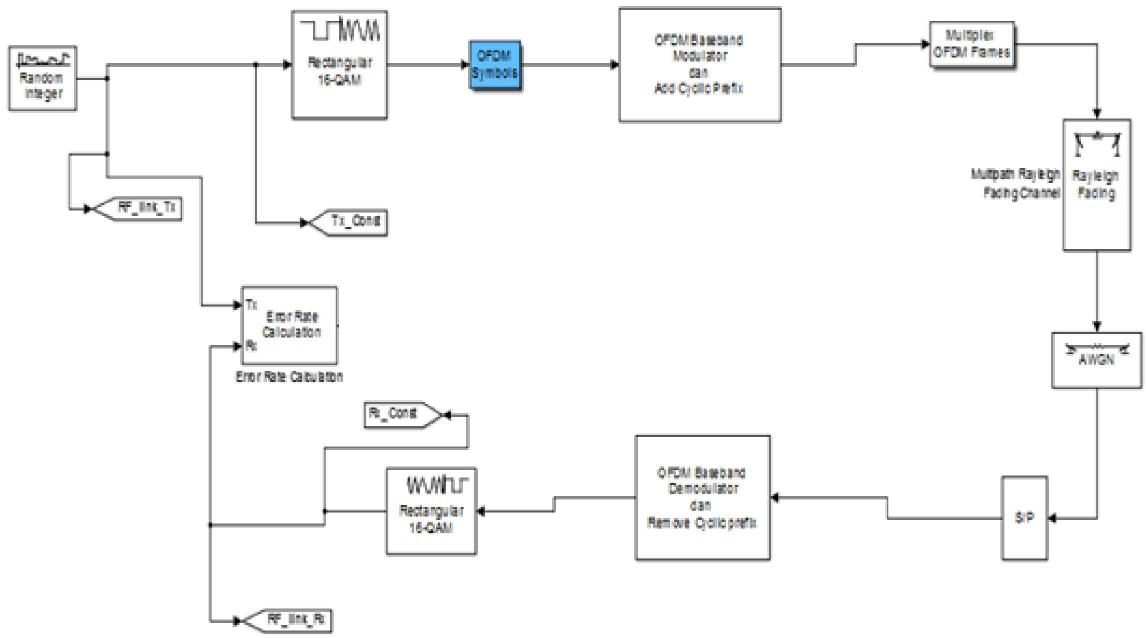


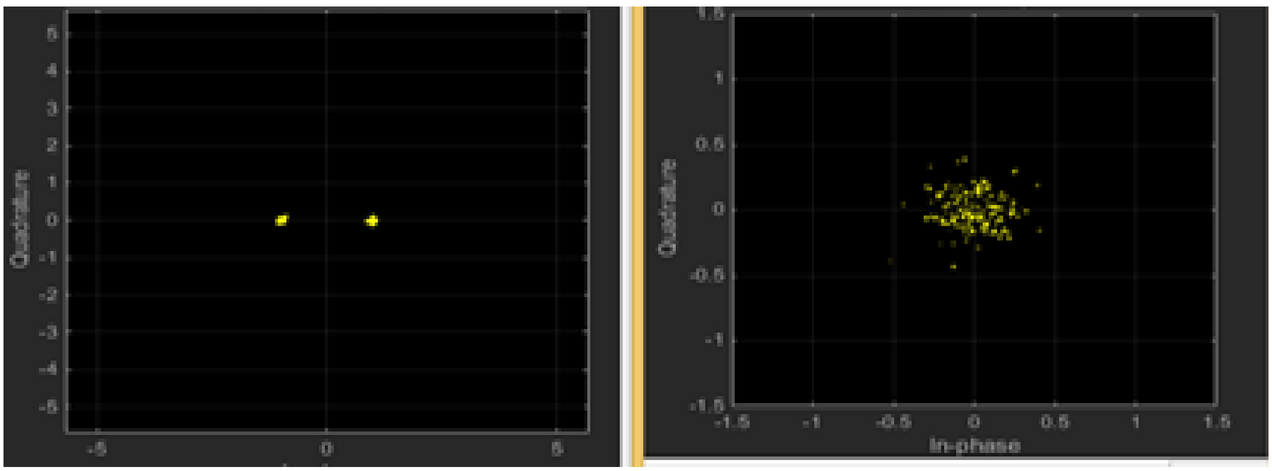
Figure 5-1 The OFDM system without equalization

The first simulation represents the OFDM system without equalization, this system includes the transmitter block in the top row and the received blocks in the down row and the transmission channel in the middle of it.

The communication system in this model performs these tasks: Generation of random integer data and modulation using one of numerous systems specified in the standard. BPSK, QPSK, and 16-QAM modulation .OFDM (orthogonal frequency division multiplexing) transmission using 52 data subcarriers, training sequence, 64-point FFTs, and a 16-sample and cyclic prefixes and. Dispersive multipath fading channels. At the receiver, this system includes the OFDM demodulation block and demodulation blocks to demodulate the received signal and Error rate block to accommodate the difference between input/output.

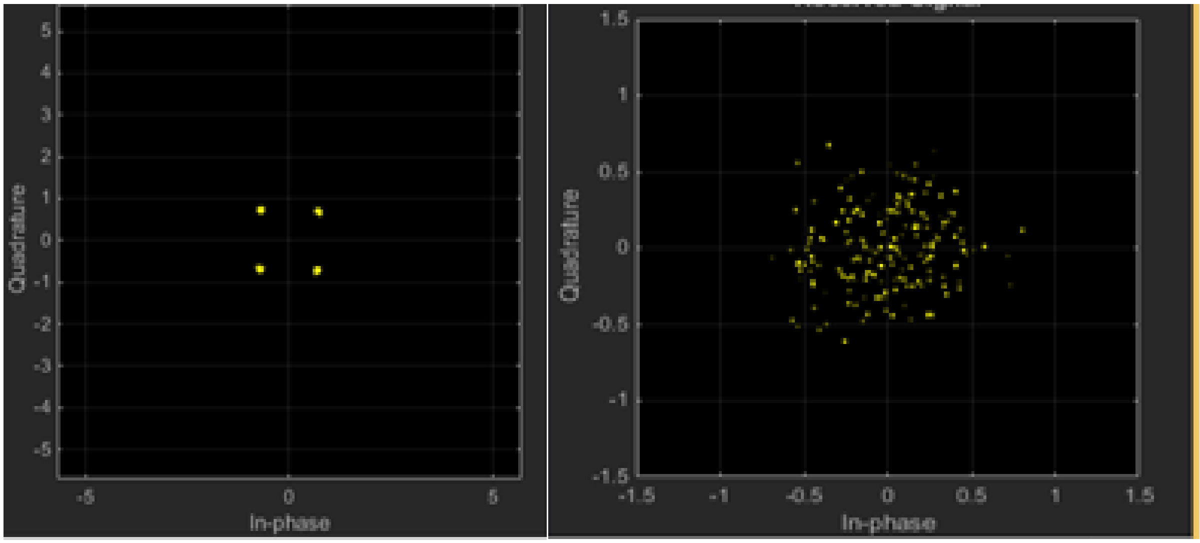
This model will be considered as the reference model in this research, the results obtained here will be compressed with results after adding the new blocks to improve the model performance.

Three modulation techniques are implemented in these models (BPSK, QPSK and 16QAM) and the results show the signal constellation and power spectrum for the transmitted and received signals without using equalization algorithms



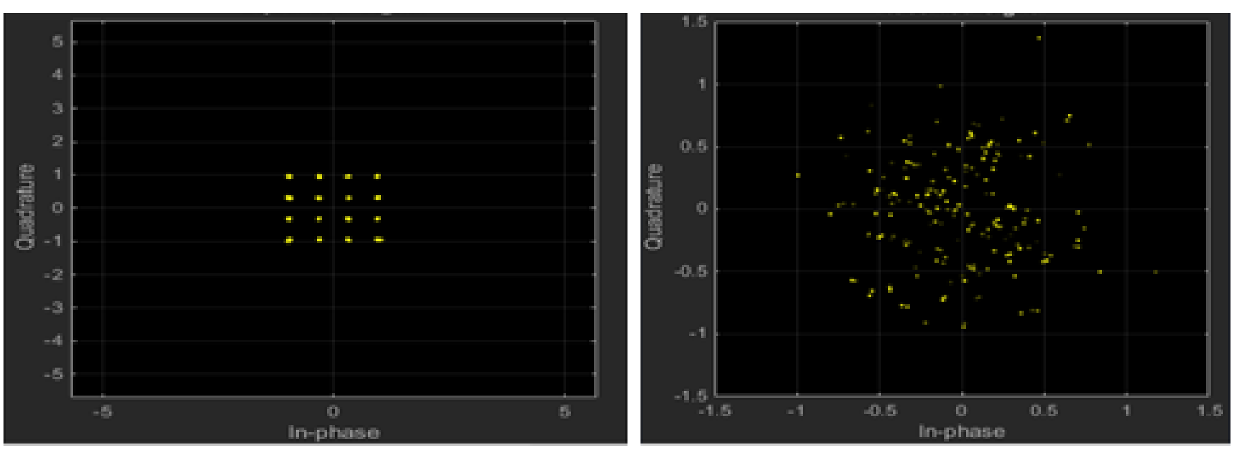
(a) (b)

Figure 5-2 OFDM system without equalization of the signal constellation for BPSK modulation technique. (a) transmitted (b) received



(a) (b)

Figure 5-3 OFDM system without equalization of the signal constellation for QPSK modulation technique. (a) transmitted (b) received

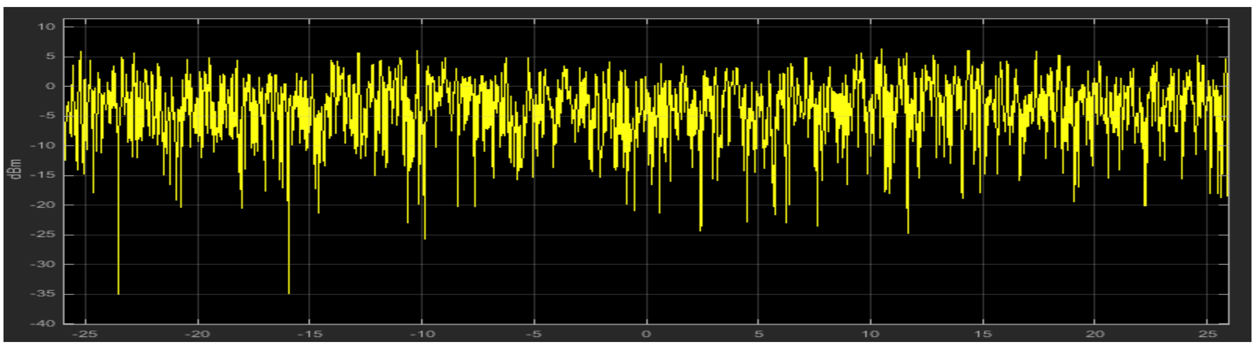


(a) (b)

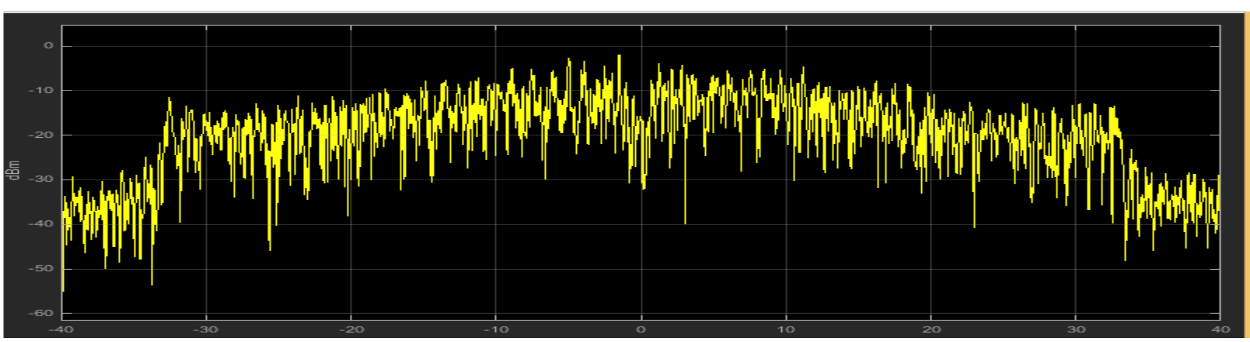
Figure 5-4 OFDM system without equalization of the signal constellation for 16 QAM modulation technique. (a) transmitted (b) received

As showing in above figures of signal constellation, when implement the OFDM system without using equalizer in the receiver end and without any techniques for reducing the inter symbol interference ISI that is caused by multipath fading channel, the received constellation is very distorted in the three cases of modulation and it is clear that cannot be recovered the received signal and the BER is around 0.6.

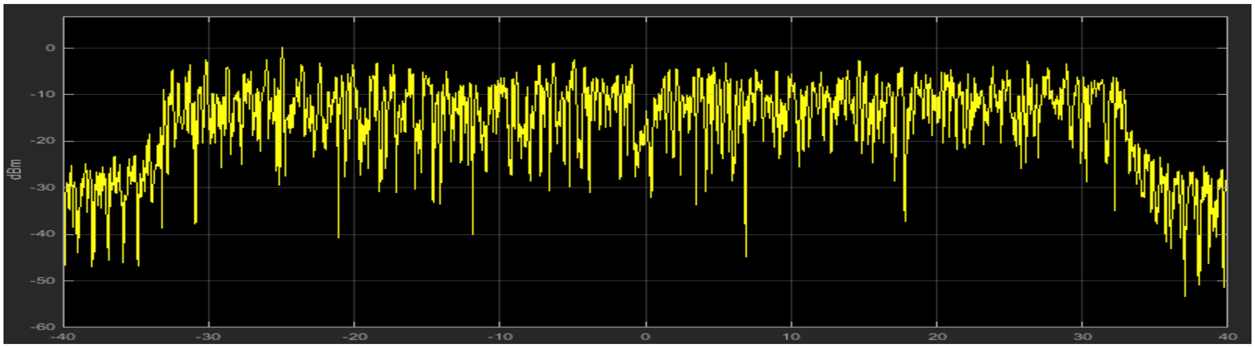
The below figures show the power spectrum of the transmitted and received signal for three modulation techniques. Also it is found the received power spectrum has high fluctuation.



(a)

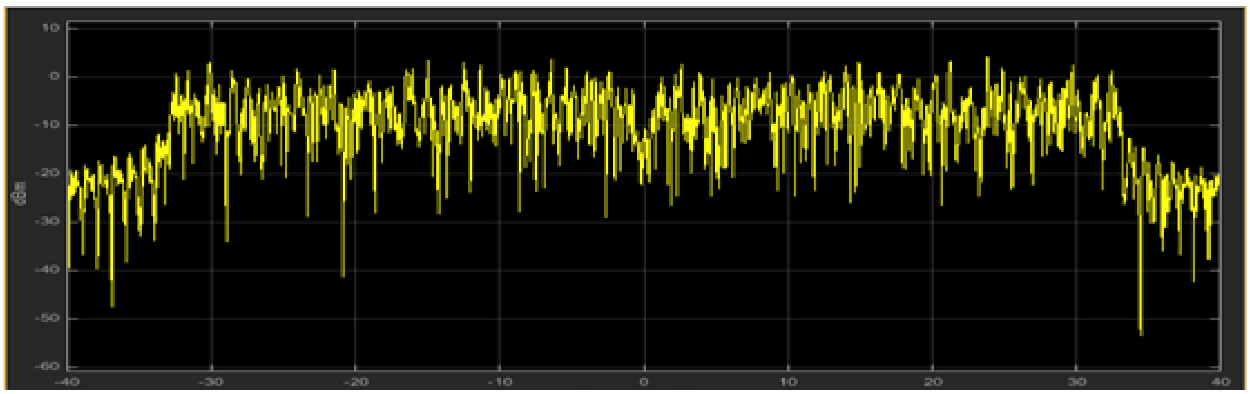


(b)



(c)

c



(d)

Figure 5-5 OFDM system without equalization signal constellation for three modulation techniques. (a) Transmitted (b) received BPSK (c) received QPSK (d) received 16QAM

5.3 Adaptive Equalization with Conventional Training Sequence.

a. Conventional training sequence with OFDM system by using LMS equalizer.

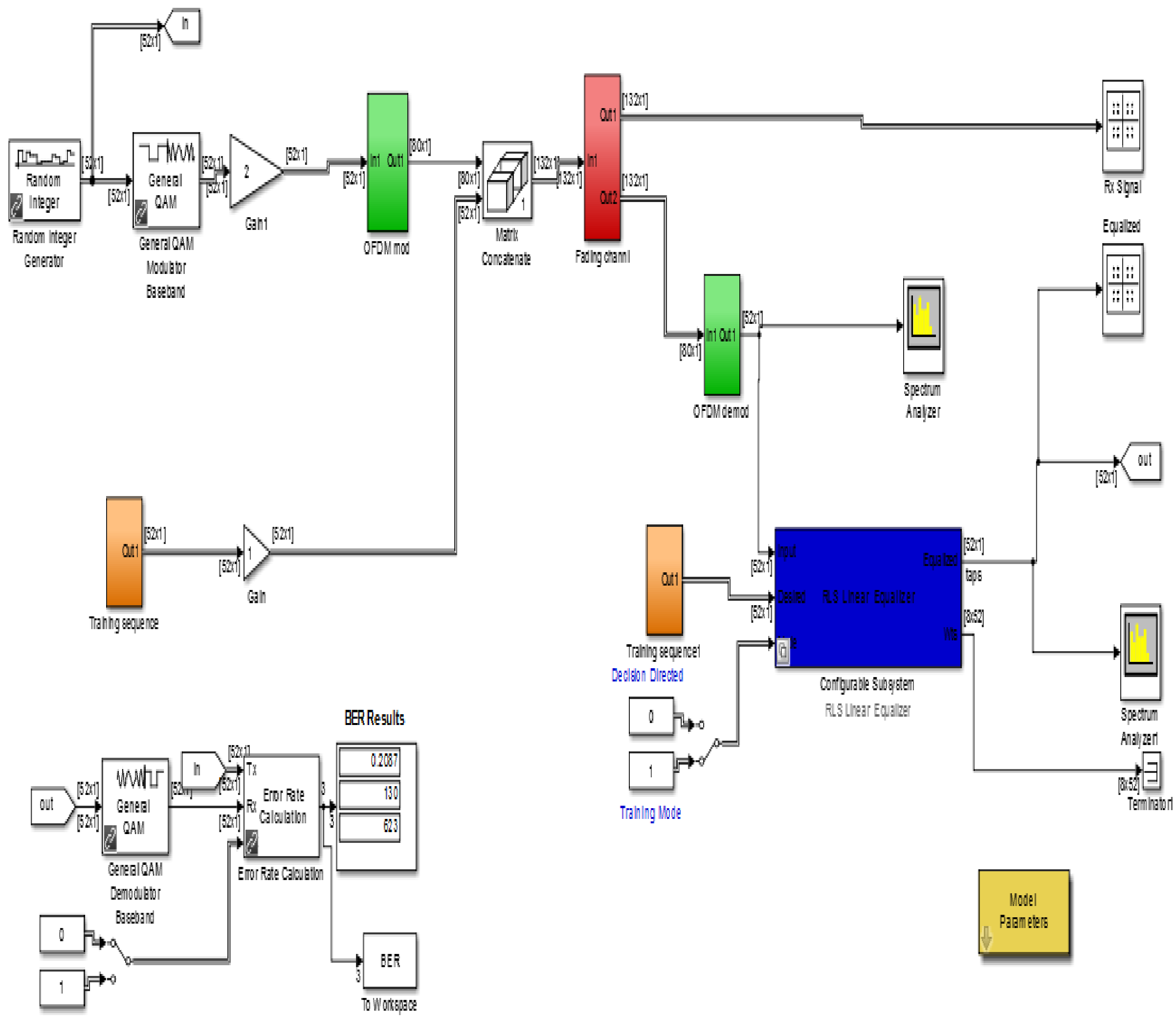


Figure 5-6 The OFDM system with LMS equalizer and conventional training sequence

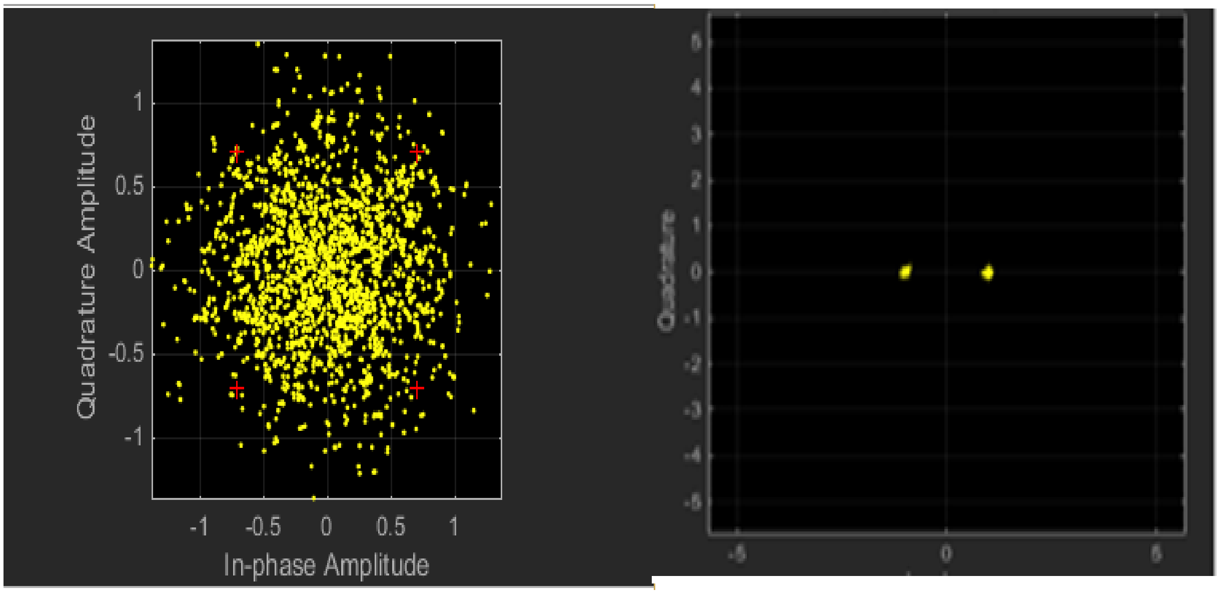
Figure5-6 is illustrates the OFDM system diagram when using adaptive equalization (LMS) in the receiver end and adding the same training sequence in transmitter and receiver.

The training sequence is stored in register on receiver end and the proposed for this store to makes good synchronisation between the transmitter and receiver when the integer data generation starts generating the input data, it is considered the additional work for training sequence.

The LMS equalizer is in set in receiver end and it uses the training sequence to estimate the coefficients of received signal, it has three inputs, the first input is the main input and the received signal enters this equalizer from this pin, the second input is the desired pin and this input is used to receive the training sequence and the third one is the mode pin and this pin is specific to the equalizer and works as training mode or decision directed mode. LMS equalize has two output ports, the first one is the equalized port and the demodulation blocks receives the data from this port and the second is the weight error port.

By using matrix concatenate tool the training sequence is added to the OFDM frame in the transmitter end before passing the channel the power of this training sequence is set to minimum value to decrease the effect on the bandwidth.

The figures below show the signal constellation for the transmitted and received signal for three modulation techniques.



(a) (b)

Figure 5-7 OFDM system with LMS equalization and conventional TS signal constellation for BPSK modulation technique. (a) Before equalizer (b) after equalizer

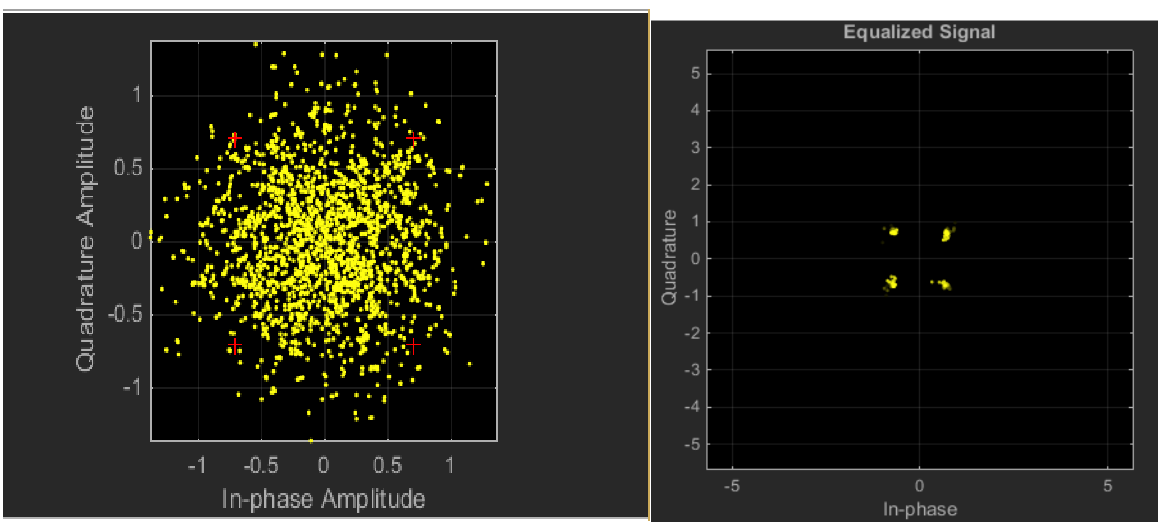


Figure 5-8 OFDM system with LMS equalization of conventional TS signal constellation for QPSK modulation technique. (a) Before equalizer (b) after equalizer

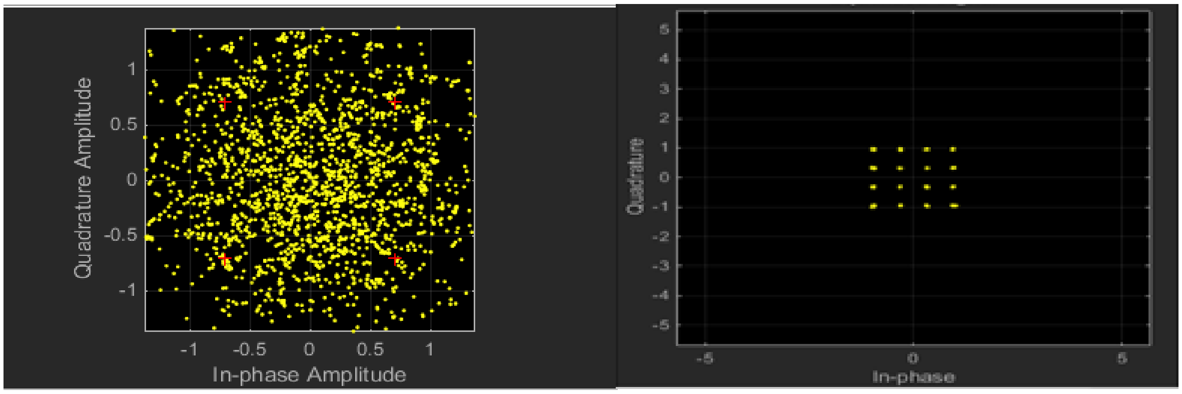
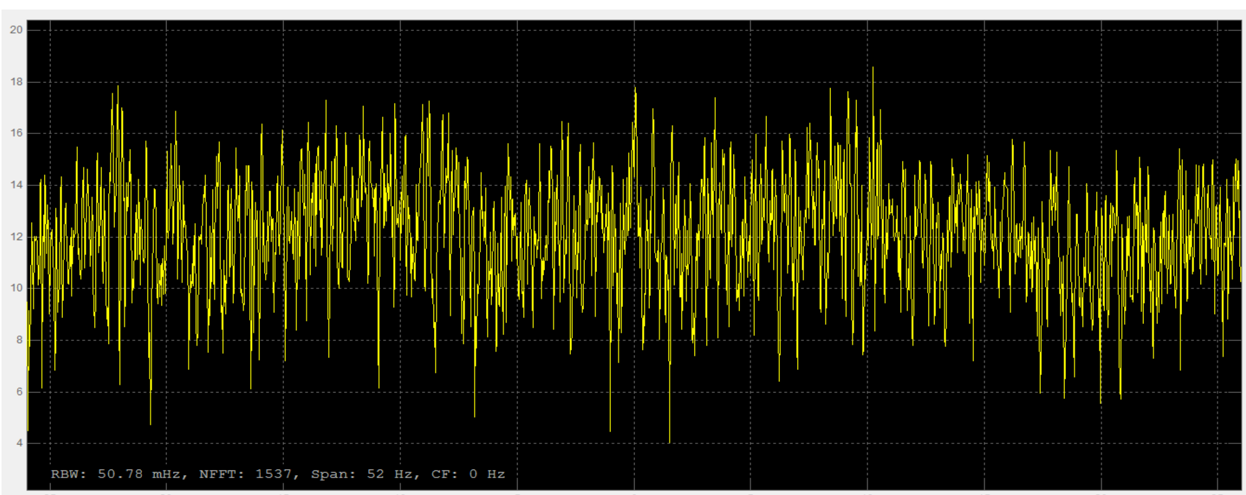


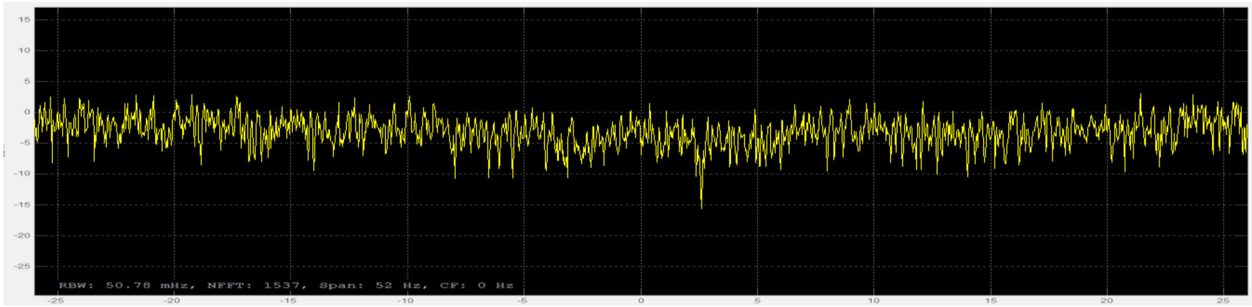
Figure 5-9 OFDM system with LMS equalization of conventional TS signal constellation for 16 QAM modulation techniques. (a) Before equalizer (b) after equalizer

Figure 5-7 shows the difference in signal constellation between the received signals before equalization and after equalized, the equalized constellation shows clearly the role that equalizer plays in improving the received signal by two clear amplitudes in phase amplitude, moreover Figures 5-8 and 5-9 show the signal constellation of QPSK and 16 QAM respectively

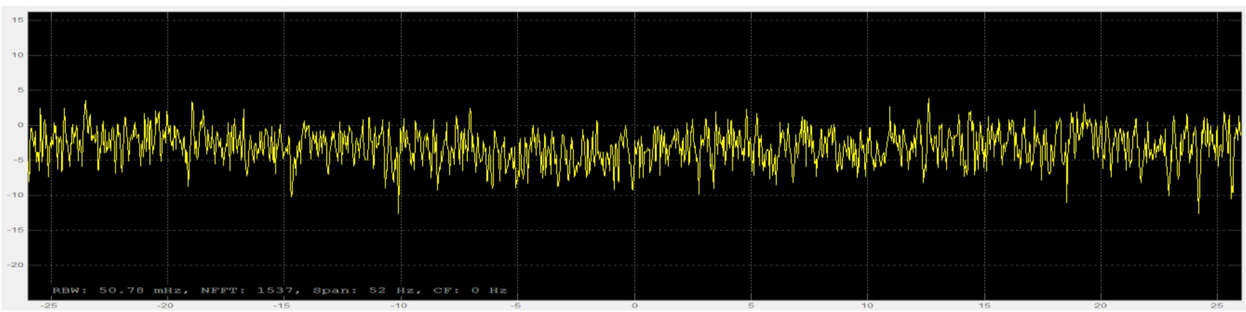
The figures below show the power spectrum of the transmitted and received signals for three modulation techniques.



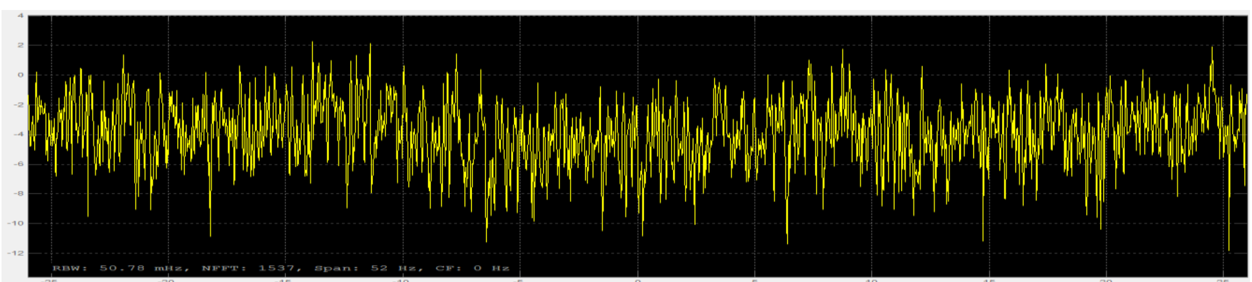
(a)



(b)



(c)



(d)

Figure 5-10 OFDM system with LMS equalizer and conventional TS power spectrum of three modulation techniques. (a) Before equalizer (b) equalized BPSK (c) equalized QPSK (d) equalized 16QAM

Figure 5-10 explains the power spectrum of OFDM system with LMS equalizer by using conventional training sequence when applied to three types of modulation techniques, Figure 5-10(a) for BPSK shows the high fluctuation on frequency axis and the amplitude (dB axis) continuously moving from -60dB to -10dB, compared with Figure 5-10(B) the fluctuation is reduced and the moving on dB axis also reduces and becomes around -20 to zero. Figure 5-10(c) & (d) represents the power spectrum of QPSK and 16 QAM after passing from LMS equalizer.

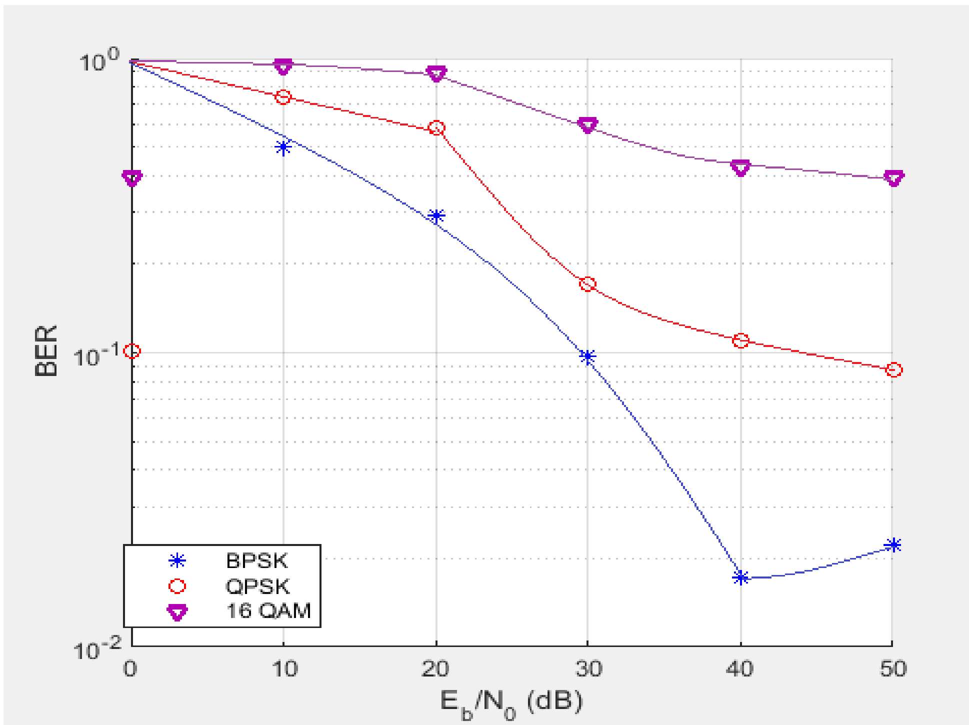


Figure 5-11 BER for OFDM system with LMS equalizer by using conventional training sequence.

The figure above explains the BER values of three types of modulation techniques, as shows Figure 5-11 the best BER values can be achieved in OFDM system with LMS equalizer when using BPSK modulation schema and it's about 10^{-2} when applied to range on $E_b / N_\circ$ starting from 0 to 50 dB and this value can decrease more when the number of multipath on the channel decrease.

b. Conventional training sequence with OFDM system by using RLS equalizer.

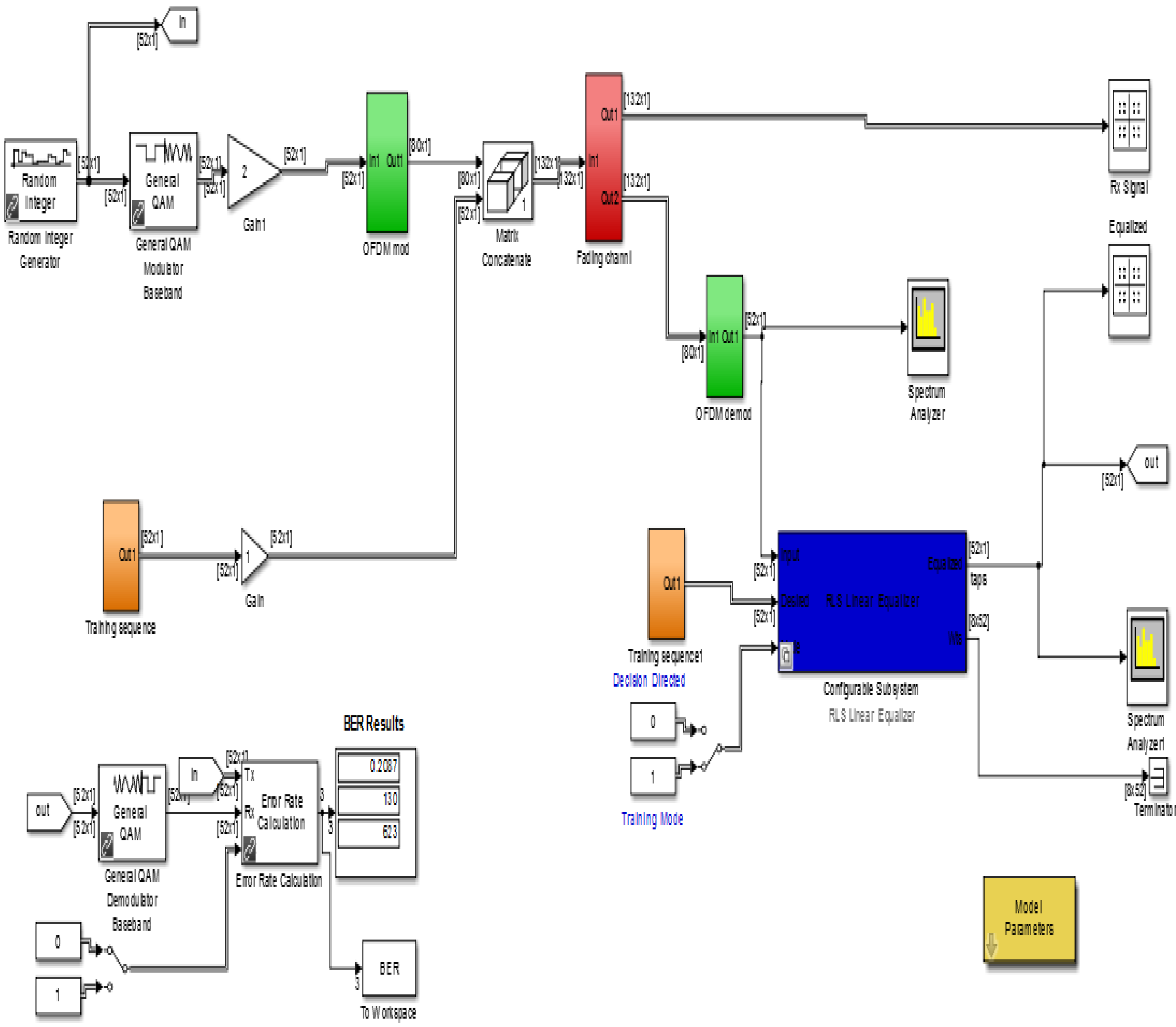
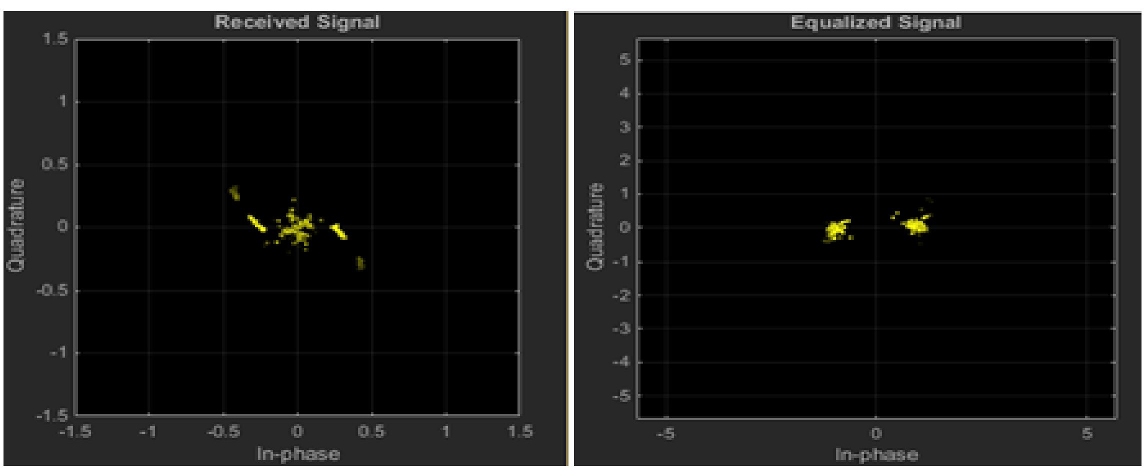


Figure 5-12 The OFDM system with RLS equalizer and conventional training sequence

Figure5-12 illustrates the OFDM system diagram when adaptive equalization (RLS) is used in the receiver end and the same training sequence is added to transmitter and receiver.

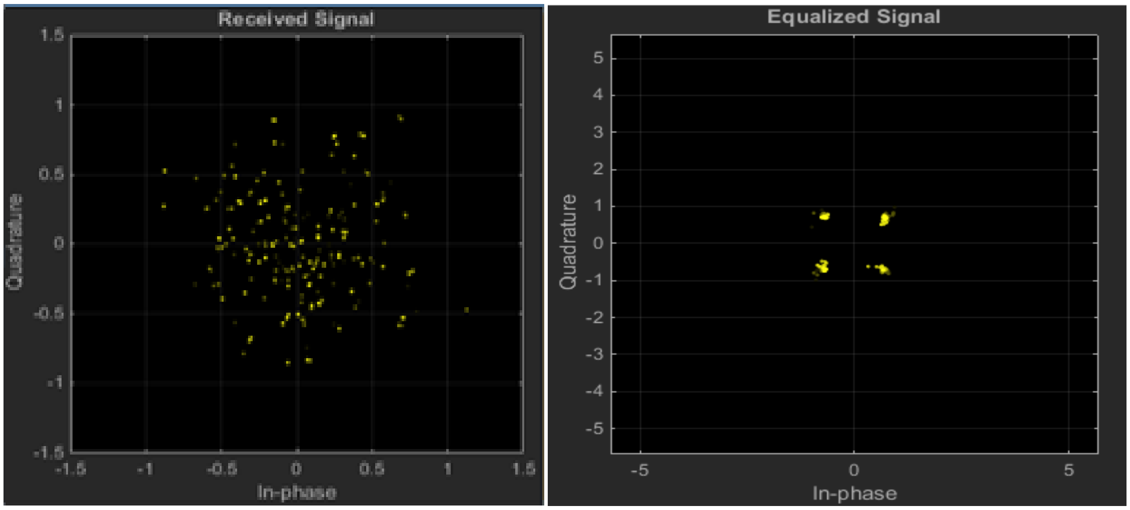
The RLS equalizer is set in receiver end and it uses the training sequence to estimate the coefficients of received signal, it has three input, the first input is the main input and the received signal enters this equalizer from this pin, the second input is the desired pin and it uses this input to receive the training sequence and the third one is the mode pin and this pin is specific to the equalizer works as training mode or decision directed mode. RLS equalizer has two output ports , the first one is the equalized port and the demodulation blocks received the data from this port and the second is the weight error port.

The figures below show the signal constellation for the transmitted and received signal for three modulation techniques.



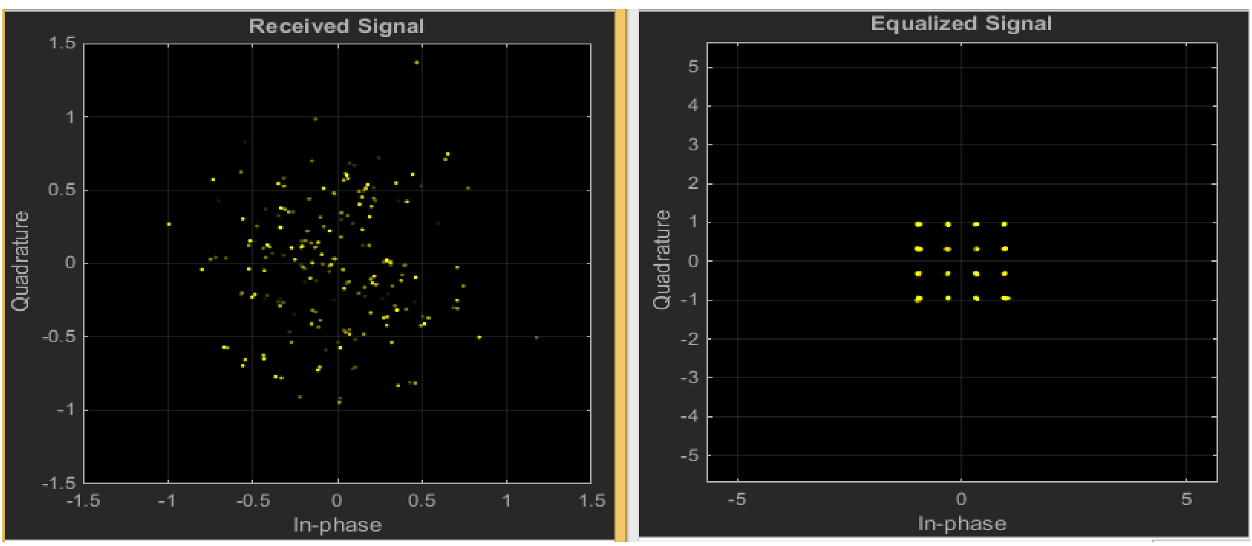
(a) (b)

Figure 5-13 shows OFDM system with RLS equalization and conventional TS signal constellation for BPSK modulation technique. (a)Before equalizer (b) after equalizer



(a) (b)

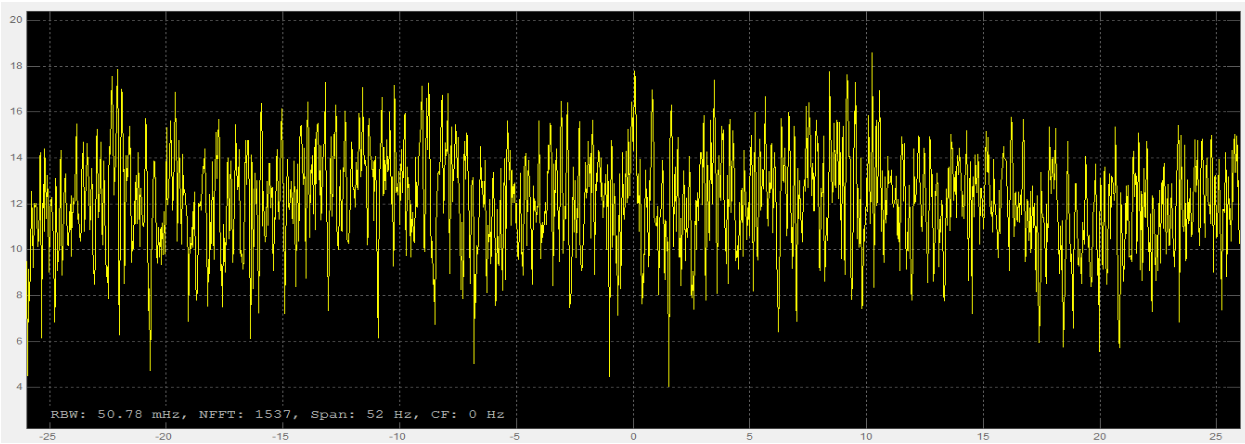
Figure 5-14 OFDM system with RLS equalization and conventional TS signal constellation for QPSK modulation technique. (a)Before equalizer (b) after equalizer



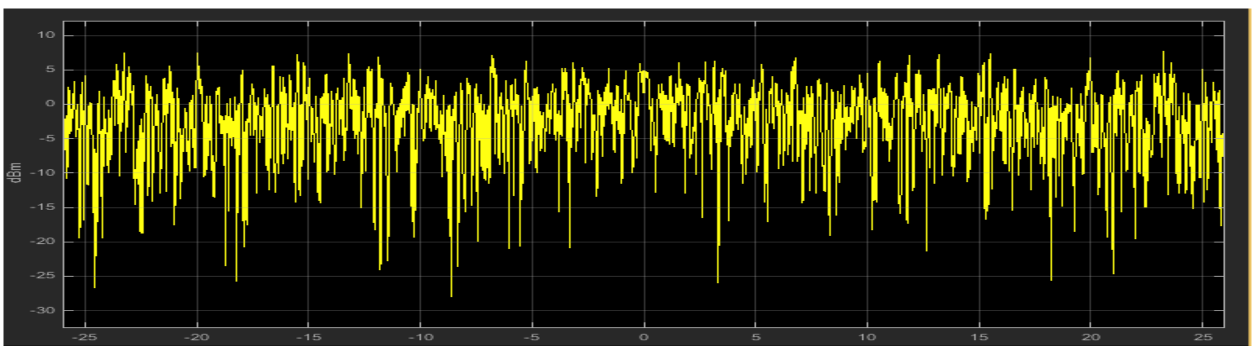
(a) (b)

Figure 5-15 OFDM system with RLS equalization and conventional TS signal constellation for 16 QAM modulation techniques. (a)Before equalizer (b) after equalizer

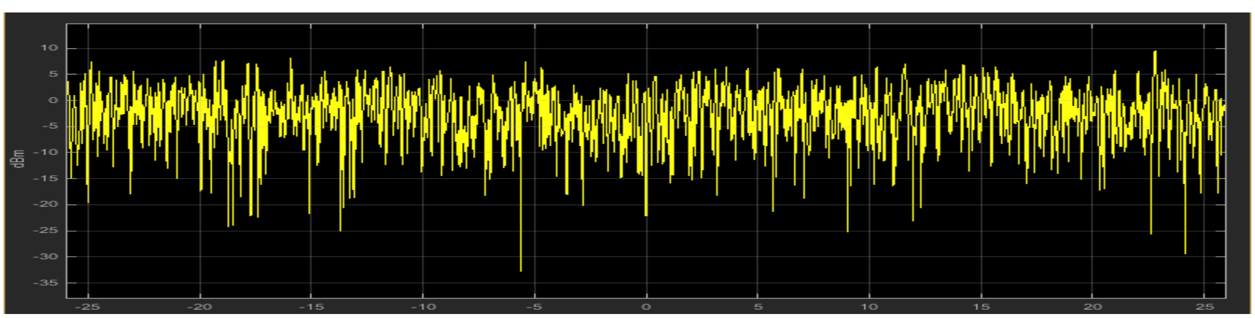
Figure 5-13 shows the difference in signal constellation between the received signals before equalizer and after equalization, the equalized constellation shows more clearly the role of that equalizer plays in improving in the received signal by two clear amplitudes in phase amplitude, moreover Figure 5-14 and 5-15 show the signal constellation of QPSK and 16 QAM respectively



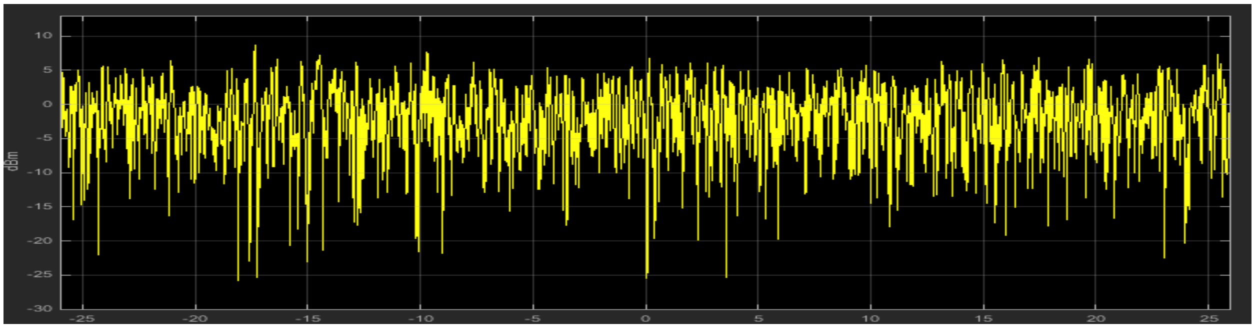
(a)



(b)



(c)



(d)

Figure 5-16 OFDM system with RLS equalizer and conventional TS power spectrum for three modulation techniques. (a) Before equalizer (b) equalized BPSK (c) equalized QPSK (d) equalized 16QAM

The figures above show the power spectrum for the transmitted and received signals of three modulation techniques, Figure 5-16 (a) explains the power spectrum for the received signal before the RLS equalizer passes, the work of the RLS equalizer is to equalize and improve the received signal as shown in the Figure 5-16 (b) and(c) and (d) the fluctuation decreases and the movements on dB axis are less that of LMS equalizer as showed before in Figure 5-10. The spectrum shows approximately satiable on zero values.

Form of signal constellation and power spectrum were presented before, the best results are found when the Binary Phase Shift Keying (BPSK) is used after that the quadture phase shift keying (QPSK).

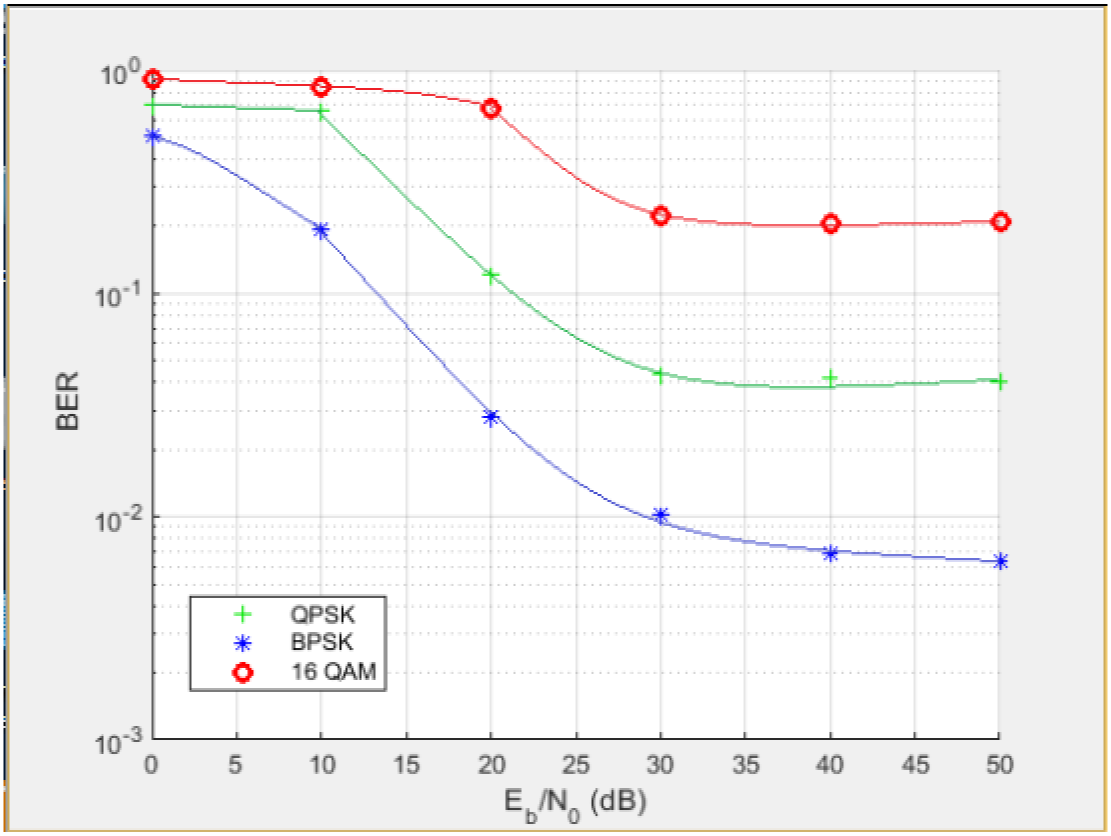


Figure 5-17 BER for OFDM system with RLS equalizer by using conventional training sequence.

Figure 5-17 explains the BER values of three types of modulation techniques, as shows in the above figure, the best BER values can be achieved in OFDM system with LMS equalizer when using BPSK modulation schema and it's about 10^{-3} when applied in the range $E_b / N_\circ$ starting from 0 to 50 dB.

5.4 <u>Adaptive Equalization with Superimposed Training Sequence.</u>

a. Superimposed training sequence with OFDM system by using RLS equalizer.

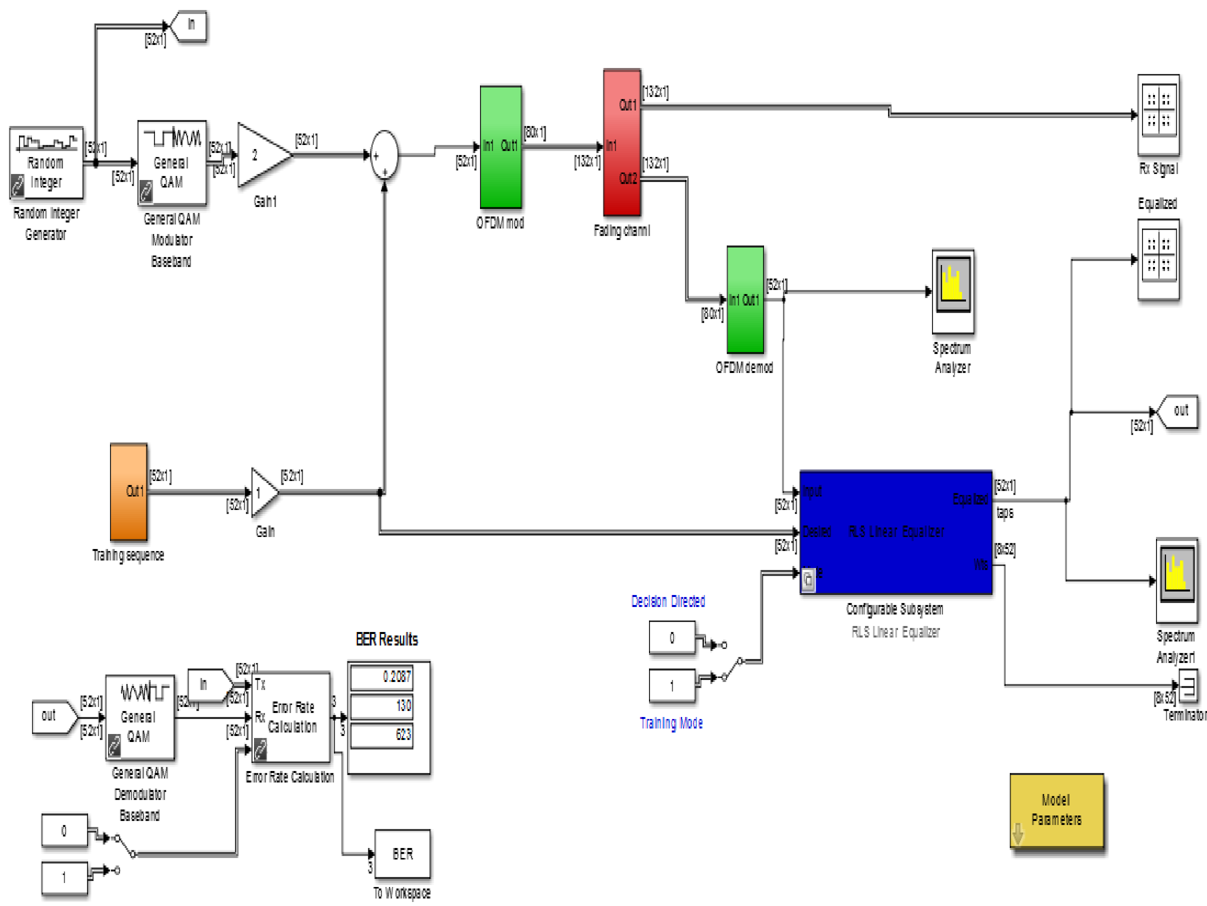


Figure 5-18 The OFDM system with RLS equalizer and superimposed training sequence

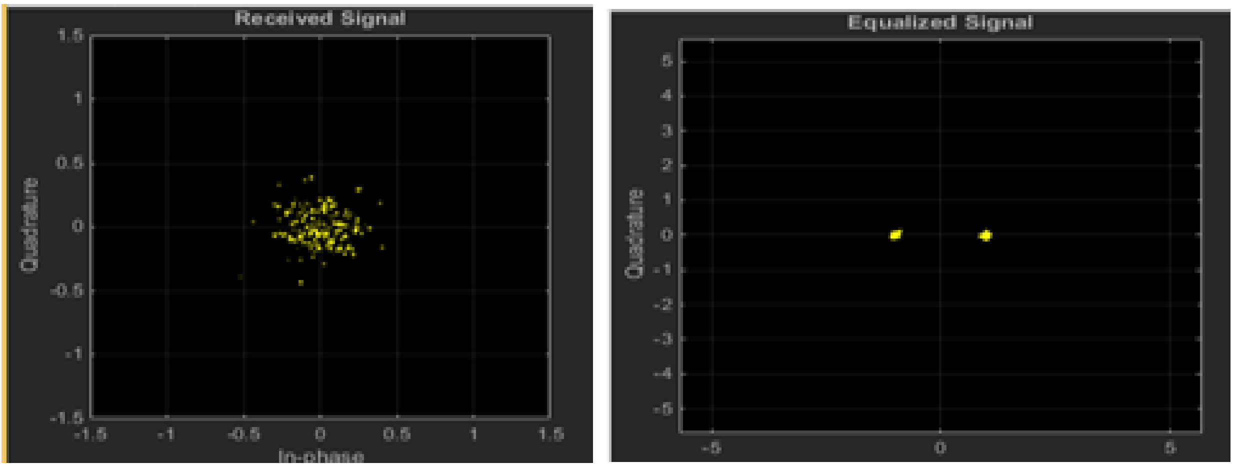
Superimposed training sequence can be used with OFDM system to prevent the inter symbol interference ISI which appears in fading channels as a result of multipath fading , this sequence is a random sequence and does not have any information which is added to the frame of data after passing through modulation mapping.

The addition of training sequence is the direct sum with the input data (algebraic summation) after adding process is completed, the compacted signal enters the OFDM modulation and is transferred into channel, at receiving end the equalizer used to remove the effect of fast fading channel of training sequence which is used as the desired input to equalize the received signal, is the method defined as Superimposed training sequence.

The training sequence is stored in register on receiver end and the proposal for this store is make good synchronisation between the transmitter and receiver when the integer data generation starts generating the input data, it is considered additional work for training sequence.

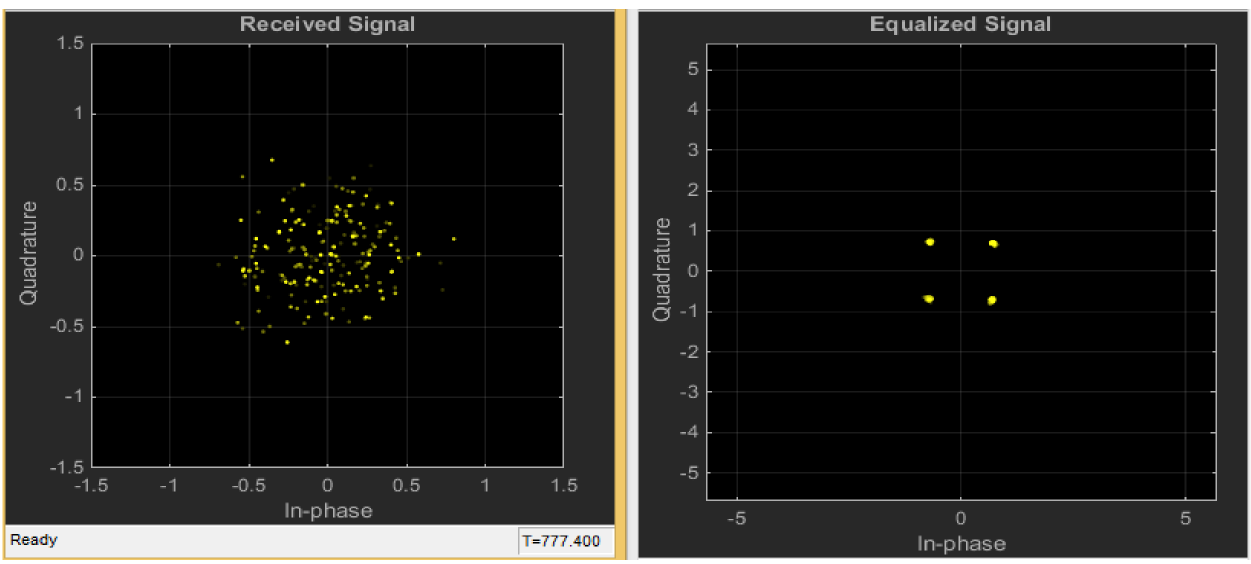
The LMS equalizer is set in receiver end and it uses the training sequence to estimate the coefficients of received signal, it has three input, the first input is the main input and the received signal enter this equalizer from this pin, the second input is the desired pin and it uses this input to receive the training sequence and the third one is the mode pin and this pin is specific to the equalizer works as training mode or decision directed mode. LMS equalize has two output ports, the first one is the equalized port and the demodulation blocks receive the data from this port and the second is the weight error port.

The below figures show the signal constellation for the transmitted and received signal of three modulation techniques.



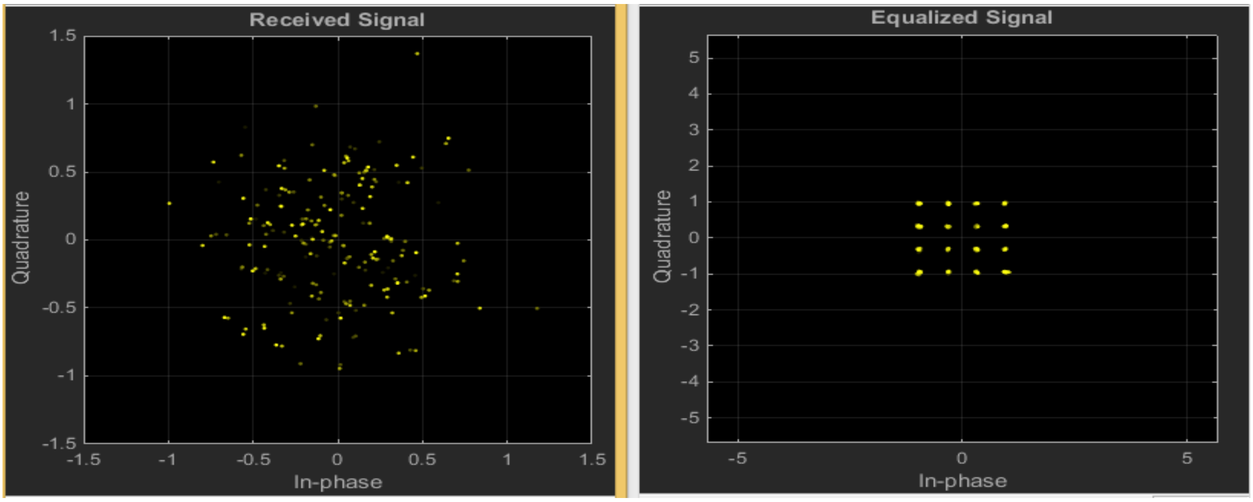
(a) (b)

Figure 5-19 OFDM system with RLS equalizer and STS signal constellation for BPSK modulation technique. (a)Before equalizer (b) after equalizer



(a) (b)

Figure 5-20 OFDM system with RLS equalizer and STS signal constellation for QPSK modulation technique. (a)Before equalizer (b) after equalizer

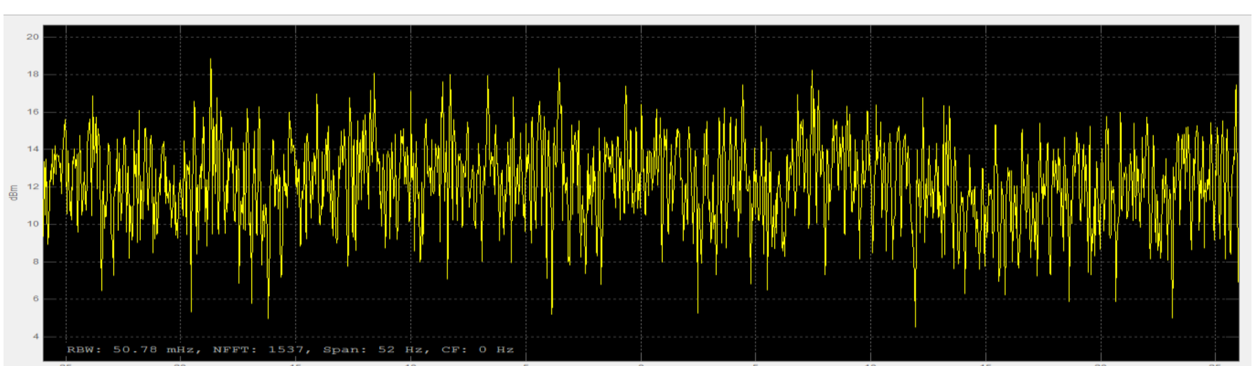


(a) (b)

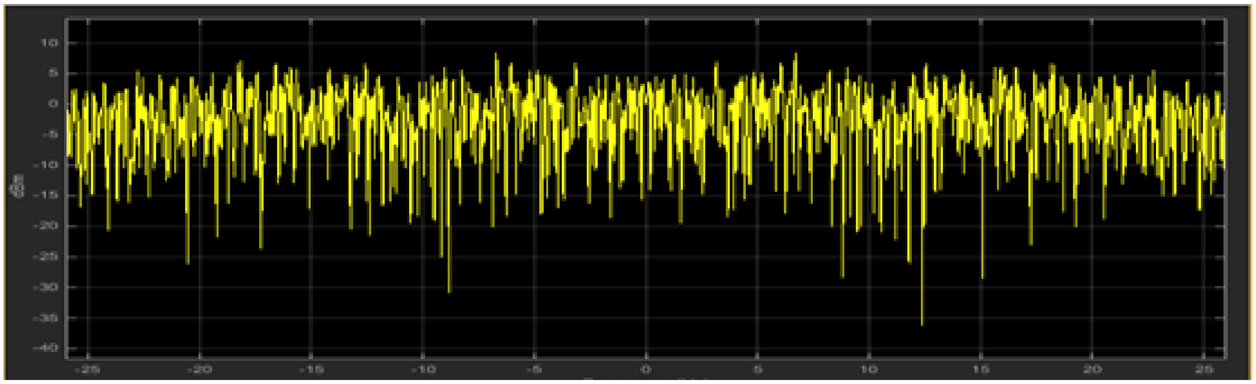
Figure 5-21 OFDM system with RLS equalizer and STS signal constellation for 16 QAM modulation techniques. (a)Before equalizer (b) after equalizer

Figure 5-19 shows the difference in signal constellation between the received signals before equalizer and after equalization , the equalized constellation shows more clearly the role that the equalizer plays in improving the received signal by two clear amplitudes in phase amplitude, moreover Figures 5-20 and 5-21 show the signal constellation of QPSK and 16 QAM respectively.

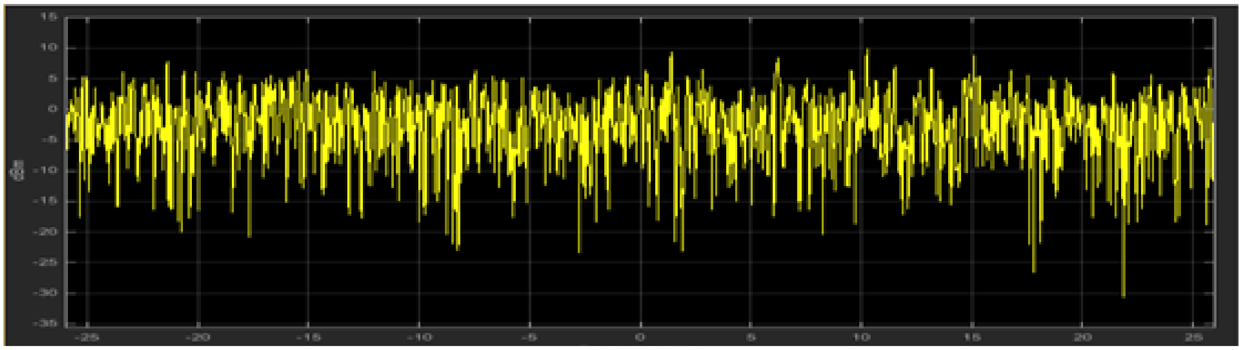
The below figures show the power spectrum for the transmitted and received signal of three modulation techniques



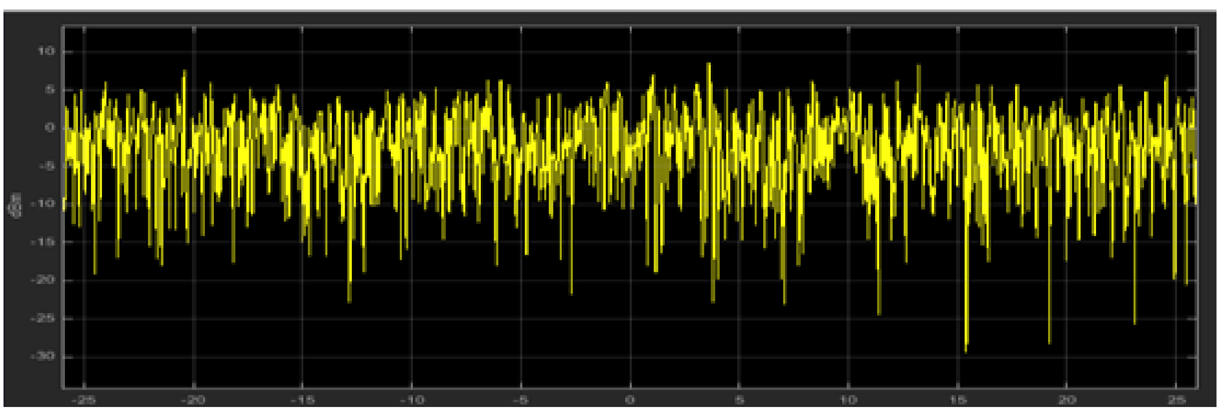
(a)



(b)



(c)



(d)

Figure 5-22 OFDM system with RLS equalizer and STS power spectrum for three modulation techniques(a) Before equalizer (b) equalized BPSK (c) equalized QPSK (d) equalized 16QAM

The above figures show the power spectrum for the transmitted and received signals when using superimposed training sequence for three modulation techniques. Figure 5-22 (a) explains the power spectrum for the received signal before passing the RLS equalizer, the work of the RLS equalizer is to equalize and improve the received signal and show the Figure 5-16 (b) and (c) and (d) show the fluctuation decreases and the movements on dB axis is about -10 to 5 as showed before in Figure 5-22. The spectrum shows approximately satiable zero values.

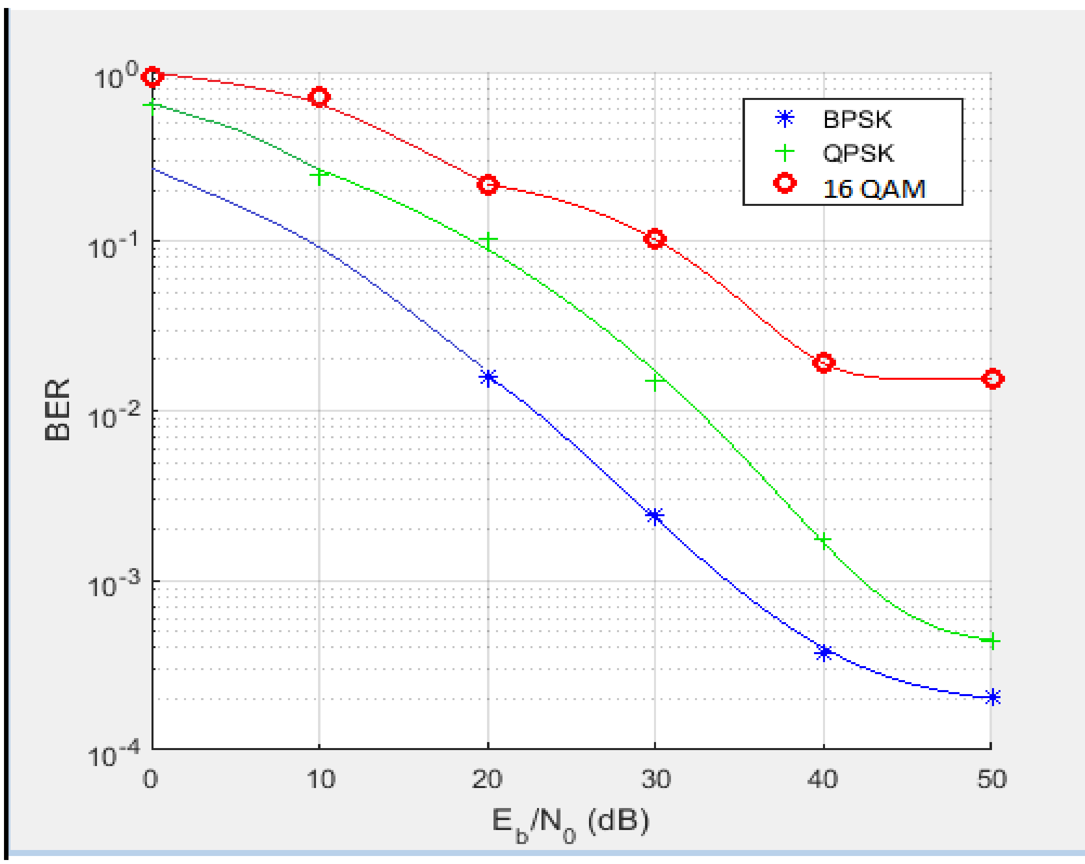


Figure 5-23 BER for OFDM system with RLS equalizer by using superimposed training sequence.

Figure 5-23, shows the Bit error rate (BER) of BPSK,QPSK and 16-QAM modulation techniques compared for RLS adaptive block equalizers. The simulation is performed on MATLAB software. The various input parameters used for the simulation are given in Table 1. In this research the channel is modeled with normalized Channel Impulse Response (CIR). The above figure compares the BER performance of the RLS equalizers respectively for BPSK,QPSK and QAM modulation.

It is found that using superimposed training sequence improves the performance of equalizers significantly. It is found that RLS equalizers can achieve minimum BER of order of10^{-4}. Channel Normalization produces the smooth BER curves and also reduces the error probability.

The BER performance of the different M-PSK with M = 4, 16, is, compared for RLS equalizer under the frequency selective fading channels It can be observed that up to around 16 dB the proposed system with normalized CIR performs approximately similar for all PSK sizes.

The method that is used in this research is Monte Carlo in Bit Error rate analysis tool the MATLAB 2014b (R14) by choosing the range on E_b / $N_\circ$ and choosing the Simulink program to implement BER calculation for it. This method supports the dots plots to plot the BER results after that the line between BER values can be plotted, so the BER figures have some disconnect in the lines which hasn't smooth lines.

Typically, a number of error values of at least 100 produce an accurate error rate. The number of bits value prevents the simulation from running too long, especially at large values of E_b / $N_\circ$. However, if the number of bits value is very small that the simulation collects many errors, the error rate might not be accurate, also confidence intervals can be used to size the accuracy of the error rates that simulation produces; the larger the confidence interval, the less accurate the calculated Bit error rate.

b. Superimposed training sequence with OFDM system by using LMS equalizer.

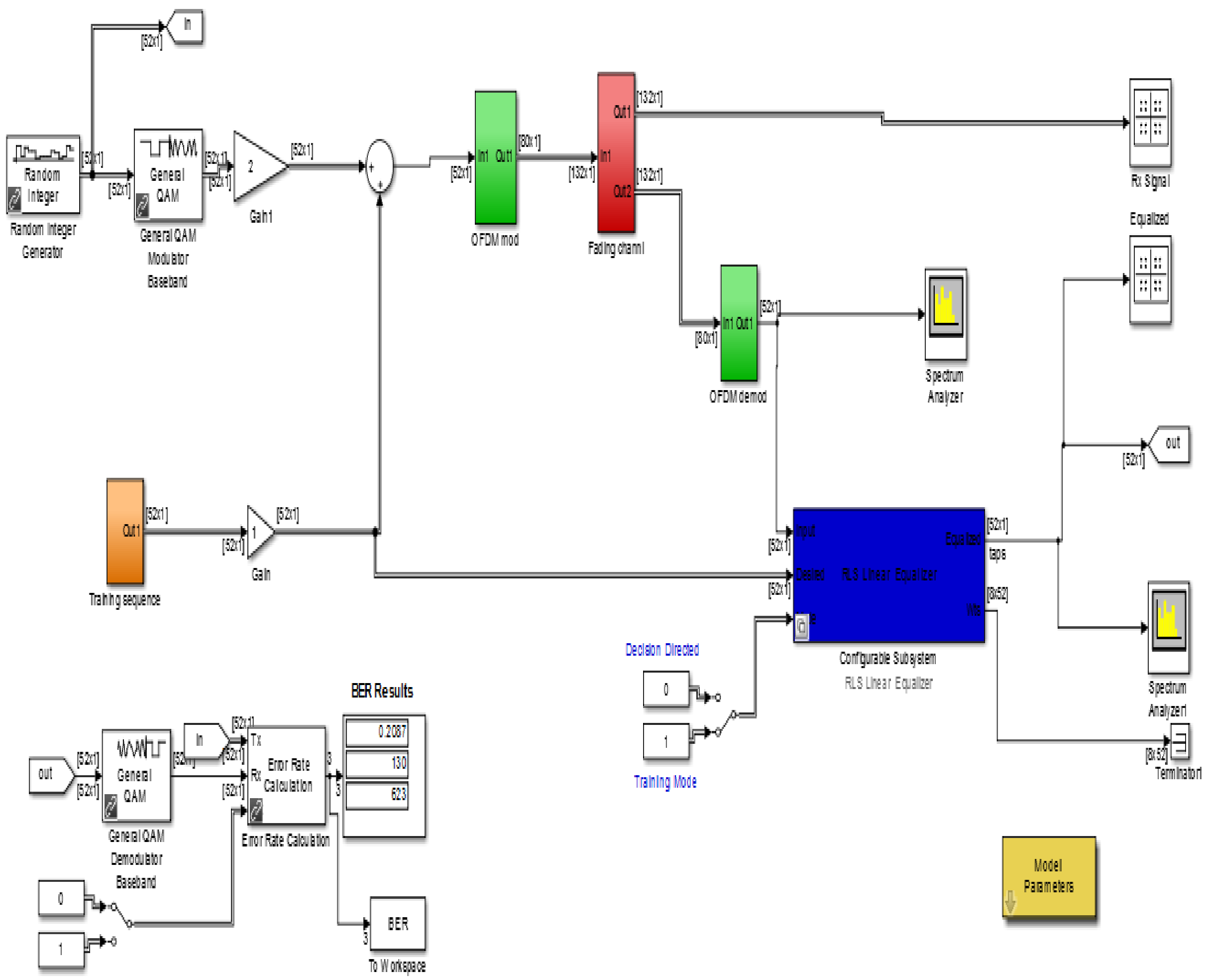
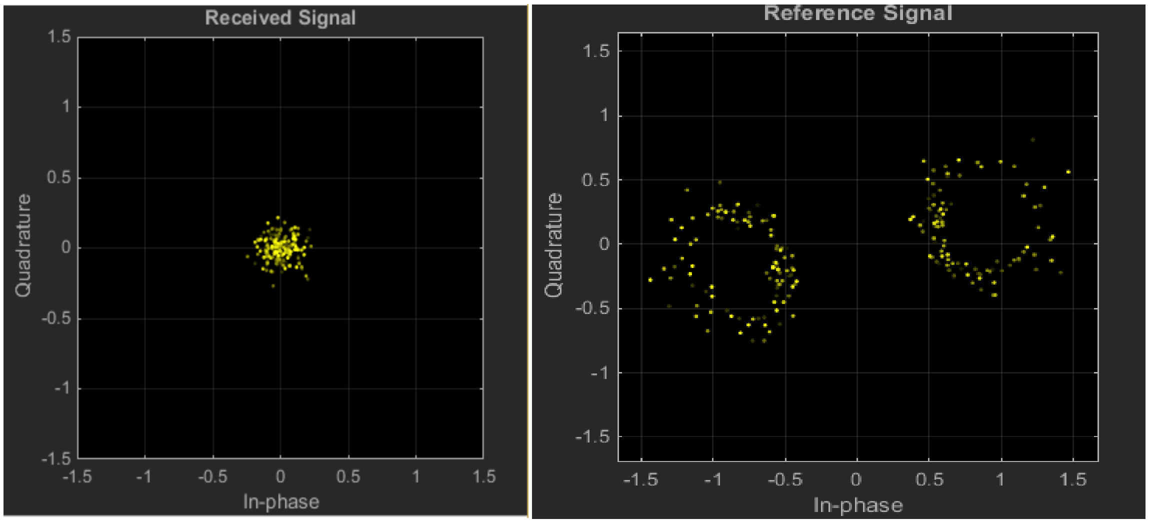
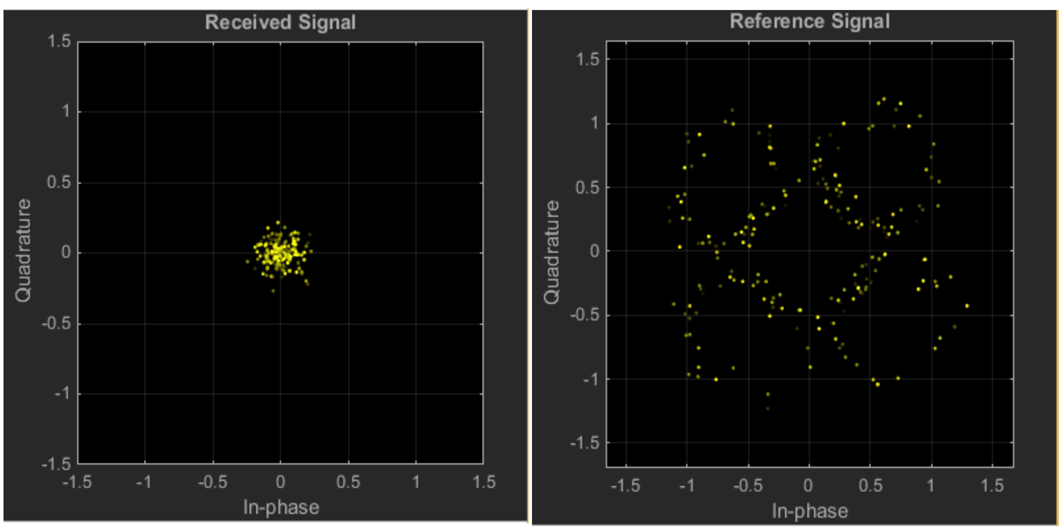


Figure 5-24 The OFDM system with LMS equalizer and superimposed training sequence



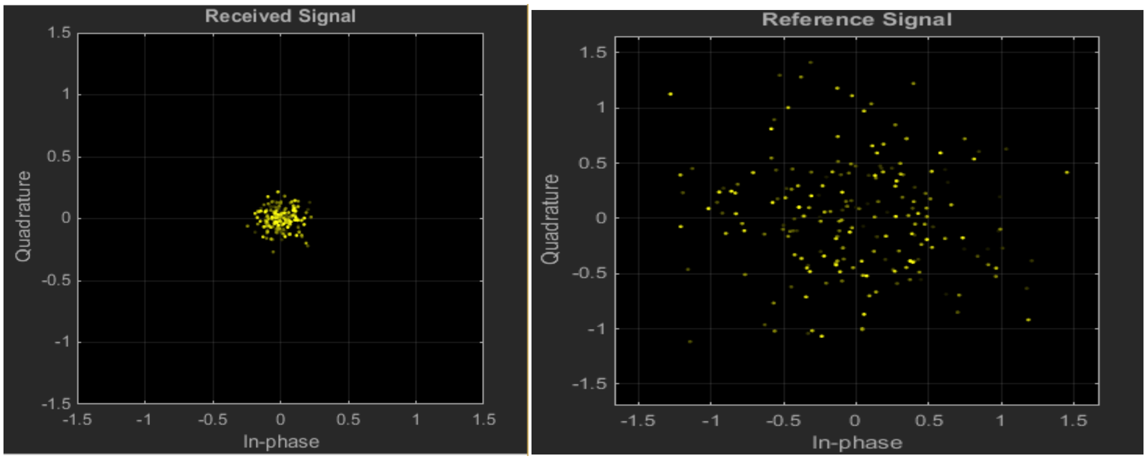
(a) (b)

Figure 5-25 OFDM system with LMS equalizer and STS signal constellation for BPSK modulation technique. (a)Before equalizer (b) after equalizer



(a) (b)

Figure 5-26 OFDM system with LMS equalizer and STS signal constellation for QPSK modulation technique. (a)Before equalizer (b) after equalizer

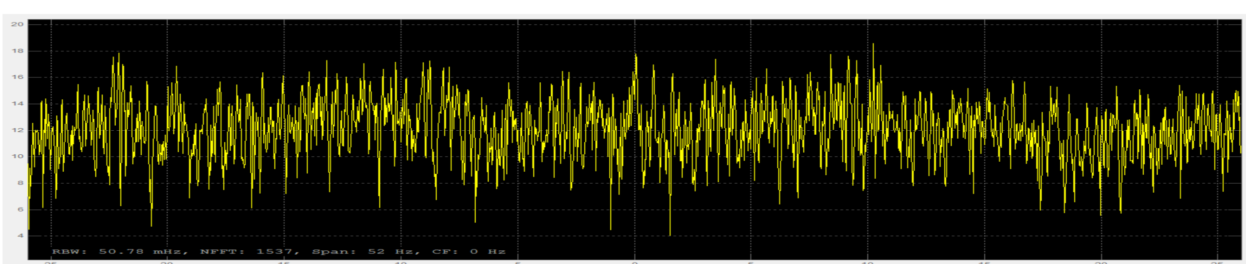


(a) (b)

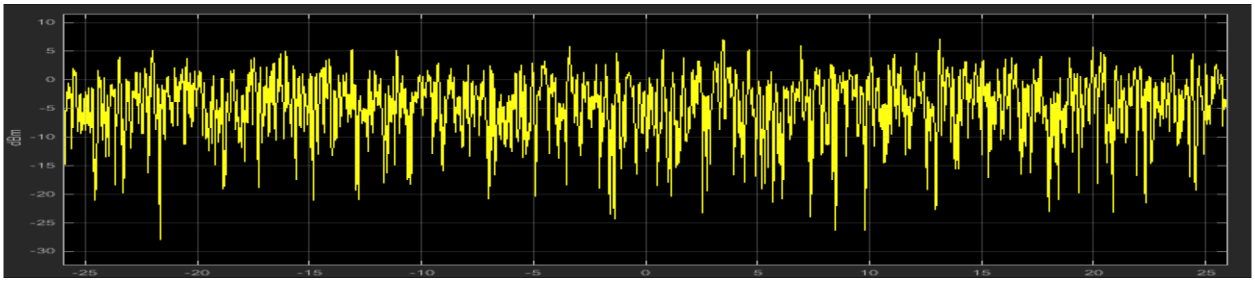
Figure 5-27 OFDM system with LMS equalizer and STS signal constellation for 16QAM modulation technique. (a)Before equalizer (b) after equalizer

Figure 5-25 shows the difference in signal constellation between the received signals before equalizer and after equalization , the equalized constellation shows clearly the role that equalizer plays to improving the received signal by two clear poles amplitude in phase amplitude, moreover the Figure 5-26 and 5-27 show the signal constellation of QPSK and 16 QAM respectively.

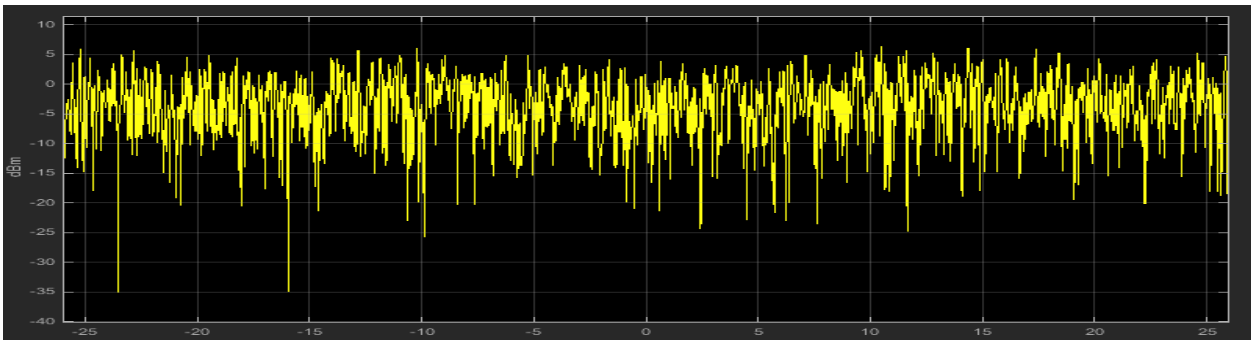
As comparison with RLS equalizer , the results achieved in RLS are better than of LMS when employing Superimposed training sequence for the constellation resolution and the distance between poles , for the power spectrum figures , the RLS results is better than that of LMS in terms of the fluctuations and the movement along the dB axis and the stability on zero axis.



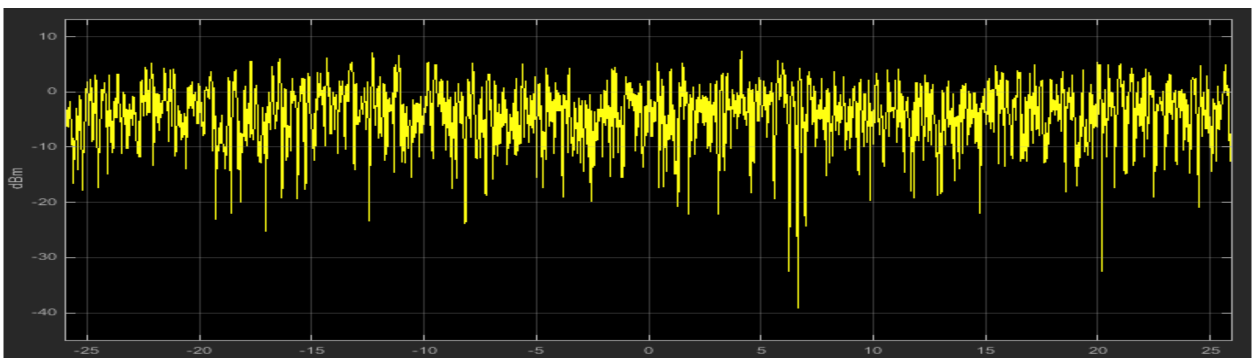
(a)



(b)



(c)



(d)

Figure 5-28 OFDM system with RLS equalizer and STS power spectrum for three modulation techniques. (a) Before equalizer (b) equalized BPSK (c) equalized QPSK (d) equalized 16QAM

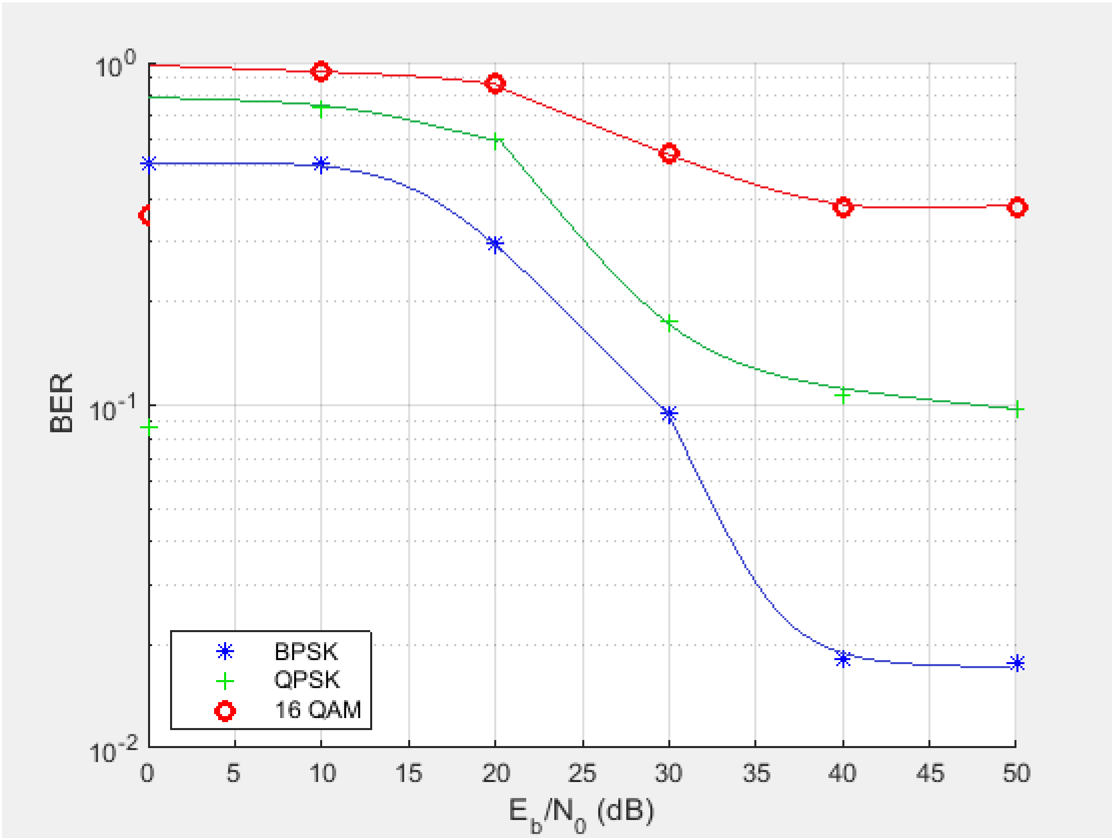


Figure 5-29 BER for OFDM system with LMS equalizer by using superimposed training sequence.

In Figure 5-29, the Bit error rate (BER) of the BPSK, QPSK and 16-QAM modulation techniques are compared for LMS adaptive block equalizers. The simulation is performed on MATLAB software. The same training is appended in the transmitter and receiver. The various input parameters used for the simulation are given in the Table 1. In this research the channel is modeled with normalized Channel Impulse Response (CIR). The above figure compares the BER performance of the RLS equalizers respectively for BPSK, QPSK and QAM modulations. The best BER results achieved are between 10^{-2} and, 10^{-3}.

It is found that using superimposed training sequence improves the performance of equalizers significantly. It is found that RLS equalizers can achieve minimum BER of order of 10^{-3}. Channel Normalization produces the smooth BER curves and also reduces the error probability.
The BER performance of the different M-PSK with M = 4, 16, is, compared for RLS equalizer under the frequency selective fading channels respectively It can be observed that up to around 16 dB the behavior of proposed system with normalized CIR is approximately similar for all PSK sizes.

From the above results you can compared your BER results when STS is used with RLS equalizer with the another public work for using the superimposed pilots sequence with RLS equalizer when using the BPSK modulation techniques [6]. At 30 $E_b / N_\circ$ values, the BER value achieved in two figures is 10^{-3} and approximately all the results are similar with anther values of $E_b / N_\circ$. on the other head , the multipath channel used in this research is more complex and dispersive it include three multipath with highest power gain and time delay than the channel used in [6].

Table 5: the BER and power spectrum range values

Techniques	Factor	**BPSK**	**QPSK**	**16QAM**
Without equalizer	BER	0.6	0.6	0.7
	dB range	-50 to -10	-50 to -10	-50 to -10
Conventional training sequence with LMS	BER	10^{-2}	10^{-1}	0.5
	dB range	-5 to 5	-5 to 5	-10 to 0
Conventional training sequence with RLS	BER	10^{-2}	$10^{-1.5}$	0.3
	dB range	-15 to 5	-15 to 5	-20 to 5
Superimposed training sequence with LMS	BER	10^{-2}	10^{-1}	0.4
	dB range	-15 to 0	-20 to 5	-20 to 5
Superimposed training sequence with RLS	BER	10^{-4}	10^{-3}	10^{-2}
	dB range	-15 to 0	-15 to 5	-10 to 5

CHAPTER SIX

Conclusions and Suggestions for Future Work

6.1 Conclusions

In this work, the objective is to present the superimposed training sequence instead of the conventional training sequence which is applied to OFDM system to resolve the fast fading channel difficulties and to provide better BER performance and reduce the effect of inter symbol interference (ISI). This job is accomplished in three types of modulation techniques: BPSK, QPSK and 16 QAM and with two adaptive equalizers: least mean square (LMS) and recursive least square (RLS) on the multipath channel which consist of Rayleigh fading channel and AWGN.

From the simulation results we can concluded that the superimposed training sequence is the best method can used in channel estimation process compared with conventional training sequence when we used complex multipath channel (Rayleigh fading channel) with RLS adaptive equalizer especially when used BPSK modulation techniques which can achieved 10^{-4} BER value and show the stability of the frequency spectrum along the Zero value in dB axis with small fluctuations with values between -5 to Zero appears when using superimposed training sequence with RLS equalizer.

From the above safely reasons the high advantage can be achieved from the proposed superimposed training sequence on OFDM system and the good results for obtained when use RLS equalizer is used in the receiver end to serve the channel estimation.

6.2 Suggestions for Future Work

1- Applying the same proposed modulation schemes to MC-CDMA systems.

2- Using Forward Error Correction (FEC) codes, such as Reed-Solomon codes, Low Density Parity Check (LDPC) codes, trills code or Turbo codes. These codes give the receiver the ability to detect and correct some errors which occur at the transmission and; therefore, decrease the BER levels.

3- Applying the Bi-directional LMS algorithms instead of the adaptive equalization algorithms to equalize the received signal and analyze the weight updated and LMS equalizer convergence.

4- Implementing the superimposed training sequence on MIMO OFDM systems.

5- Changing the type of the training sequence from random integer to anther data like, PN sequence or chirp sequence and making analysis of the system performance and BER values.

6- Using modulator bank block including different modulation techniques to increase the data rate at which you want to transmit through the channel.

REFERENCES

[1] **John G. Proakis**: "Digital Communications" *Fifth Edition, McGraw-Hill*, 2008.

[2] **Abhishek Kumar, Anoop Tiwari, Ravi Shankar Mishra**,: " Linear Block Equalizers in Rayleigh Fading Channel with Normalized Channel Impulse Response" *International Journal of Computer Applications (0975 – 8887) Volume 93 – No 6,* May 2014 .

[3] **Gunjan Verma, Jaspal Bagga,:** "Performance Analysis of Equalizer Techniques for Modulated Signals" *International Journal of Engineering Researc and Applications (IJERA),* 2013.

[4] Available at :https://en.wikipedia.org/wiki/Blind_equalization.

[5] **Ahsan Aziz**, "Decision Feedback Equalizer for StarCore ®-Based DSPs"**,***Freescal Semiconductor, Application Note*,2007.

[6] **Peter Hoeher,Fredik Tuhvesson**, " channel estimation with superimposed pilot sequence", *Proceedings of the IEEE International Conference on Advanced Communication Technology*, 1999.

[7] **FRANCO MAZZENCA**, "A Hidden Pilot Assisted Demodulation Technique for Channel Estimation in A4-QAM Transmissions", *IEEE International Conference on Advanced Communication Technology Manuscript received on December IS*, 1999.

[8] **Ho, C. K.; Farhang-Boroujeny, B.; and Chin, F**. "Added Pilot Semi Blind Channel Estimation Scheme for OFDM in Fading Channels", *Proceedings of the IEEE-GLOBECOM, pp. 3075-3079,* Nov. 2001.

[9] **Mounir Ghogho, Des McLernon, Enrique Alameda-Hernandez, and Ananthram Swami,** "Channel Estimation and Symbol Detection for Block Transmission Using Data-Dependent Superimposed Training",*IEEE SIGNAL PROCESSING LETTERS, VOL. 12, NO. 3, MARCH* 2005.

[10] **Qinghai Yang and Kyung Sup Kwak**, " Time-Varying Multipath Channel Estimation with Superimposed Training in CP-OFDM Systems" *ETRI Journal, Volume 28, Number 6, December* 2006.

[11] **Chen, Ning and Zhou, G. Tong**. "Superimposed Training for OFDM: A Peak-

to-Average Power Ratio Analysis", *IEEE Transactions on Signal Processing, vol. 54, No. 6, pp. 1-10*. Jun. 2006.

[12] **Qinghai Yang et al**: "Superimposed training for estimating of doubly selective OFDM channels", *Proceedings of the IEEE International Conference on Advanced Communication Technology*, Feb. 1, 2007.

[13] **Angiras R. Varma, Chandra R. N. Athaudage, Lachlan L. H. Andrew, Jonathan H. Manton,** "OPTIMAL SUPERIMPOSED PILOT SELECTION FOR OFDM CHANNEL ESTIMATION", *Proceedings of the IEEE International Conference on Advanced Communication Technology*,2009.

[14] Available at: http://www.sharetechnote.com/html/Handbook_LTE_Fading.html

[15] **Bernard Sklar**, "Rayleigh Fading Channels in Mobile Digital Communication Systems" *IEEE Communications Magazine • July* 1997.

[16] **Deepa.T,** " Linear & non- linear equalizers", *International Journal of Engineering Research and Applications (IJERA)*, 2009.

[17] **Samaneh Shooshtary,** " Development of a MATLAB Simulation Environment for Vehicle-to-Vehicle and Infrastructure Communication Based on IEEE 802.11p", *Master's Thesis in Telecommunications*,2008.

[18] **Prachi Sharma, Piush Gupta and Pradeep Kumar Singh,** "Performance Comparison of ZF, LMS and RLS Algorithms for Linear Adaptive Equalizer" *Advance in Electronic and Electric Engineering*,2014.

[19] **Abhishek Kumar, Anoop Tiwari, Ravi Shankar Mishra**,: " Linear Block Equalizers in Rayleigh Fading Channel with Normalized Chann el Impulse Response" *International Journal of Computer Applications (0975 – 8887) Volume ,93 – No 6*, May 2014 .

[20] **Kumar et al,"** Optimal Training Sequence And Channel Estimation Method And System For Superimposed Training Based OFDM Systems", *U.S. PATENT DOCUMENTS*,2013.

[21] "Digital cellular telecommunications system (Phase 2+);Multiplexing and multiple access on the radio path(GSM 05.02)" *ETSI TC-SM*,.1996.

[22] **Ian Poole, radio-electronic,** Available at

http://www.radio-electronics.com/info/cellulartelecomms/gsm_technical/gsm-radio-air-interface-slot-burst.php.

[23] **L.D. Kabulepa,** "OFDM Basics for Wireless Communications" *Institute of Microelectronic Systems, Darmstadt University of Technology*,2000.

[24] **Deepak Sharma, Praveen Srivastava**, "OFDM Simulator Using MATLAB", *International Journal of Emerging Technology and Advanced Engineering,2013.*

[25] **Dr.techn. Markus Rupp, Dipl.-Ing. Christian Mehlführer,** "Implementation of a WiMAX simulator in Simulink" , *Diploma's Thesis in Telecommunications* Vienna, February 2007.

[26] Available at: http://www.mathworks.com/help/lte/ref/lteofdmmodulate.html

[27] Available at : http://www.mathworks.com/help/comm/ug/multiple-input-multiple-output-mimo.html?searchHighlight=wimax

[28] **Roberto Cristi**, "Wireless Communications with Matlab and Simulink: IEEE802.16 (WiMax) Physical Layer", *Dept of Electrical and Computer Engineering Naval Postgraduate School Monterey, CA 93943 August* 2009.

[29] **Bernard Sklar,** "The Characterization of Fading Channels", *ResearchGate, Communications Engineering Services*,2015.

[30] Available at: http://www.mathworks.com/help/comm/ug/fading-channels.html?searchHighlight=fading

[31] **Abdelhakim Khlifi and Ridha Bouallegue** , "Performance Analysis of LS and LMMSE Channel Estimation Techniques for LTE Downlink Systems" *International Journal of Wireless & Mobile Networks (IJWMN) Vol. 3, No. 5, October*,2011.

[32] **Ali Asadi and Behzad Mozaffari Tazehkand** , "A New Method to Channel Estimation in OFDM Systems Based on Wavelet Transform" *International Journal of Wireless & Mobile Networks (IJWMN) Vol. 4, No.7, October*,2013.

[33] **Gunjan Verma, Prof. Jaspal Bagga, "**Performance Analysis of Equalizer Techniques for Modulated Signals" *International Journal of Engineering Research and Applications (IJERA), Vol. 3, Issue 4, Jul-Aug,* 2013.

استخدامها في الطرف المستلم لتحسين الموجة المستلمة واستخدام نفس المتتالية الترويضية كمدخل المطلوب للمعدل المتكيف لتعويض الموجه الداخلة المراد تعديلها.

نوعين من المعدلات استخدمت في نظام المحاكاة وتم مقارنة النتائج في كلتا الحالتين ووجدنا اقل نسبه خطأ (BER) عند استخدام المتتالية الترويضية مع RLS وتم قياس استقرارية النظام وادائه. في هذا البحث تم استخدام ثلاث تقنيات لتضمين (BPSK,QPSK and 16QAM) وعناصر نظام OFDM تم اختيارها وفقا لمقاييس IEEE 802.11a.

الخلاصه

واحده من اكبر المشكلات في انظمه الاتصالات اللاسلكيه هي خفوت الموجات المرسله و اغلب الاوساط الناقله تعاني من ظاهرة الخفوت والتي تحدث عندما يتسلم الهوائي المستلم عدة اشارات بسبب ظاهره الانكسار, الانعكاس , الضلال و التبديد, كل مسار يمتلك زمن مختلف و طاقه مكتسبة مختلفه عن المسار الاخر والتي تسبب التعقيد فى الاجهزة المستلمة لاسترجاع الموجة المرسلة خصوصا في القنوات الناقلة التي يكون فيها تغيير زمن الخفوت سريع .في النظام المثالي للأتصالات اللاسلكية ذات النطاق العريض (منظومة تقسييم الترددات المتعامدة (OFDM)) يتم ادخال فترة حماية من التداخل بين النواقل والتداخل بين الموجات الناقلة باستخدام التحديد الدوري او متتالية الترويض وهذه الطريقه تقلل من كفاءة الارسال وتزيد من عرض القناة الناقلة وتقليل الكفاءة يؤدي الى تاخير زمن الوصول وتقليل في عدد المستفدين حسب مقايسس IEEE 802.11a LAN (WLAN).

لزيادة كفاءة النقل , المتتالية الترويضية المركبة تضاف الى هيكل الموجة المرسلة لتقليل من تاثير التداخل ويتم اضافتها وتخزينها في نفس الوقت في الطرف المستلم للأستفاده منها في عملية التزامن في انظمة الاتصالات. في العادة, المعدل يستخدم في انظمة (OFDM) لتعديل التأثيرات التي توثر على حافر الاستجابة لقناة الناقلة مع المتتالية الترويضية. توجد هناك انواع متعددة المعدلات التي طورت لتوافق تطبيقات OFDM مثل معدل الاجبار للصفر (ZF) والمعدلات المتكيفية كمعدل التقليل الربعي (LMS) ومعدل التقليل الربعي الفوري (RLS).

في هذا البحث , المتتالية الترويضية تم تنفيذها على نظام OFDM بطريقتين: الاولى عن طريق الاضافة الجبرية المتتالية الترويضية مع هيكل الموجة المرسلة والطريقة الثانية باستخدام سلسلة المصفوفة لدمج المتتالية الترويضية مع الموجة المرسلة .المعدلات تم

الالغاء المتكيف لتداخل في قنوات الخفوت السريع

رسالة

مقدمة الى كليه الهندسة في جامعة النهرين وهي جزء من متطلبات نيل

درجة ماجستير علوم في الهندسة الالكترونيه والاتصالات

من قبل

محمد كاظم حمود

(B.S.c 2009)

جماد الاخره **1437**

اذار **2016**