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ADDIS ABABA INSTITUTE OF TECHNOLOGY

SCHOOL OF ELECTRICAL AND COMPUTER ENGINEERING

Performance Analysis of Blind Equalization Algorithms for Broadcast Channel

BY

Nebiyu Nane

<u>Advisor</u>

Dr. -Ing. Dereje Hailemariam

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APPROVAL BY BOARD OF EXAMINERS

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Chair Person	Signature
Dr. -Ing. Dereje Hailemariam	____________________
Advisor	Signature
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Internal Examiner	Signature
____________________	____________________
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Declaration

I, the undersigned, declare that this thesis work is my original work, has not been presented for a degree in this or any other universities, and all sources of materials used for the thesis work have been fully acknowledged.

Nebiyu Nane ____________________

Name Signature

Place: **Addis Ababa, Ethiopia**

November 9, 2017

Date of Submission

This thesis has been submitted for examination with my approval as a university advisor.

Dr. -Ing. Dereje Hailemariam ____________________

Advisor Signature

Abstract

Wireless channels exhibit multipath and time dispersion that lead to attenuation and inter-symbol interference (ISI). In order to compensate the effect of ISI caused by channels, equalization techniques are introduced on receiver ends. The increasing demand for high-speed data transmission over band limited channels makes the applications of training-based equalization methods more challenging due to constant transmission of the pilot signals/training sequences which consumes a lot of bandwidth and power. Blind equalizations, a category of equalization techniques, requires no training sequences to track and compensation the effect of ISI channel. These equalizations are applicable in many practical areas such as audio broadcasting in which training-based equalizations are not mainly applied and use only the received signals to equalize the channel.

This thesis focuses on the performance analysis of three blind equalization techniques; namely, Variable Step Size Constant Modulus Algorithm (VSS-CMA), Decision Feedback Equalizer Constant Modulus Algorithm (DFE-CMA) and joint VSSCMA-Decision Directed Least Mean Square (DDLMS) algorithm for application in audio broadcasting. To ease visualization of results and see the effect of changing various parameters a Graphical User Interface (GUI) is developed. Comparison of the results of the above three blind equalization algorithms were compared with two additional blind equalization algorithms: Constant Modulus Algorithm (CMA) and Multi Modulus Algorithm (MMA).

The simulation results are evaluated using Mean Square Error (MSE), symbols constellation plots, convergence rate, computation time and complexity as performance metrics. Rectangular 16-Quadrature Amplitude Modulation (QAM) scheme, ISI channel with Additive White Gaussian Noise and three other channel models for GUI simulation are considered.

From the simulation results VSSCMA-DDLMS algorithm has better/faster convergence rate and low level of steady-state error or MSE values compared to VSS-CMA and DFE-CMA algorithms.

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List of Acronyms

ADC	Analog to Digital Converter
AM	Amplitude Modulation
AWGN	Additive White Gaussian Noise
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
CM	Constant Modulus
CMA	Constant Modulus Algorithm
Codec	Coder-decoder
DAB	Digital Audio Broadcasting
DAC	Digital to Analog Converter
DDB	Decision Directed
DDLMS	Decision Directed Least Mean Square
DFE	Decision Feedback Equalizer
DFE-CMA	Decision Feedback Equalizer Constant Modulus Algorithm
DQPSK	Differential Quadrature Phase Shift Keying
DVB-T	Digital Video Broadcasting (-Terrestrial)
FBF	Feedback Filter
FEC	Forward Error Correction
FFF	Feed Forward Filter
FFT	Fast Fourier Transform
FIR	Finite Impulse Response
FM	Frequency Modulation
GMMA	Generalized Multi-Modulus Algorithm
GUI	Graphical User Interface
HF	High Frequency

HiFi	High Fidelity
HTML	Hypertext Mark-up Language
IBOC	In-Band On-Channel
IF	Intermediate Frequency
ISDB	Integrated Service Digital Broadcasting
ISDB (-T)	Integrated Service Digital Broadcasting (-Terrestrial)
ISI	Inter-symbol Interference
JTC	Joint Technical Committee
LMS	Least Mean Square
LOS	Line of Sight
MMA	Multi Modulus Algorithm
MPEG	Moving Pictures Experts Group
MSE	Mean Square Error
NHK	Japan Broadcasting Corporation
OFDM	Orthogonal Frequency Division Multiplexing
PC	Personal Computer
PDF	Probability Density Function
PM	Phase Modulation
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RCA	Reduced Constellation Algorithm
RDI	Receiver Data Interface
RF	Radio Frequency
RLS	Recursive Least Square
SCA	Square Contour Algorithm
SER	Symbol Error Rate

SFN	Single Frequency Network
SNR	Signal to Noise Ratio
TV	Television
VSS-CMA	Variable Step Size Constant Modulus Algorithm

CHAPTER 1

INTRODUCTION

1.1 Introduction

Nowadays, digital based communication systems are used for the transmission of voice, data and video. One of the most important advantages of the digital transmission systems for voice, data and video communications is their higher reliability in noise environment in comparison with that of their analog counterparts [1, 2]. In digital communication, the information being transferred is represented in digital form, most commonly as binary digits, or bits. This is in contrast to analog information, which takes on a continuum of values. Most communication systems used for transferring information today are either digital, or are being converted from analog to digital [3].

The increasing demand for digital communications needs high-speed data transmission over band-limited channels. Unfortunately, most often the digital transmission of information is accompanied with a phenomenon known as inter-symbol-interference (ISI) [2]. ISI arises when the data is transmitted through band-limited channel. Such channel causes spreading of transmitted data/pulse that result in overlapping of adjacent pulse, which in turn lead to interference in the transmitted signals [1]. ISI is recognized as the major obstacle for high speed data transmission systems [1]. Thus, it is very crucial to compensate the effect of ISI in modern digital communications by developing efficient equalization algorithms which provide good filtering performance and fast convergence.

Adaptive equalization technique is one of the effective methods used to reduce the effect of ISI. An adaptive equalizer is an equalization filter that automatically adapts to random and time-varying properties of the wireless broadcast channel [2]. Adaptive equalization can be categorized as training-based, semi-blind and blind equalizations.

In training-based equalization, a known training sequence is transmitted over the channel prior to the transmission of any useful data. The receiver uses the relationship between the known training sequence and the sequence it actually receives to construct an approximation of the inverse transfer function for the channel. In blind equalizations compensation and tracking of the channel can be achieved without using a training sequence. And the purpose of semi-blind equalizations is to combine both training sequence and blind information and exploit the positive aspects of both techniques [24].

As majority of communication systems often struggle with limited bandwidth constraint, it is desirable for the receiver to obtain optimum blind channel equalization without consuming much channel bandwidth [4]. Blind equalization has a wide range of applications, for example in digital communications it is used for removal of ISI due to non-ideal channel and multi-path propagation; in speech recognition it used for removal of the effects of the microphones and the communication channels; in correction of distorted images; analysis of seismic data; de-reverberation of acoustic gramophone recordings etc [5].

There are different types of blind equalization algorithms applied in various applications; such as Constant Modulus Algorithm (CMA), Multi Modulus Algorithm (MMA), Fractional Spaced Constant Modulus Algorithm (FS CMA),Variable Step Size Constant Modulus Algorithm (VSS-CMA),Decision Feedback Equalizer Constant Modulus Algorithm (DFE-CMA) and joint VSSCMA-Decision Directed Least Mean Square (DDLMS).

1.2 Statement of the Problem

To mitigate the effect of ISI in audio broadcasting system, blind equalization algorithms are used as one of the available techniques. Before applying equalization techniques from the receiver end of the system; understanding the nature wireless broadcast channel and identifying equalization algorithms with good performance are the major challenges. The wireless broadcast channel is random and time varying in nature and the performances of blind equalization techniques vary depending on the application areas they are being applied. However, the problem in a real implementation is not only limited on identifying algorithms with good performance and understanding the characteristics of broadcast channels. It requires investigation of the end-to-end blocks in digital communication system using various simulation tools. Hence, the thesis tries to examine all end-to-end blocks in digital communication system and integrate Graphical User Interface (GUI) application with selected blind equalization algorithms to perform the test simulations. GUI application is used in the thesis as an end-to-end performance analysis tool.

1.3 Objective

1.3.1 General Objective

The general objective of the thesis is to perform analysis of VSS-CMA, DFE-CMA and VSSCMA-DDLMS; then recommending the best performed algorithm for real application areas specifically in audio broadcasting systems.

1.3.2 Specific Objectives

The specific objectives of this thesis can be summarized as follows:

- To understand appropriate models for broadcast channel.
- To analyze VSS-CMA, DFE-CMA and joint VSSCMA-DDLMS dual-mode algorithms for ISI channel with AWGN through MATLAB simulations: considering MSE, convergence rate, computation time, complexity and symbols constellation plots as performance metrics.
- To apply simulation procedures to test the performance of equalization techniques;
- To integrate GUI blocks with selected blind algorithms and channel models
- To apply GUI with app designer application for test simulations.

1.4 Literature Review

To mitigate the effect of ISI channel, lots of researches are conducted with focus on blind channel equalization algorithms. In this thesis some of the related prior works in blind channel equalization algorithms are reviewed as follows:

Ali Ozen *et al.* proposed a VSS-CMA blind algorithm to resolve contradiction between steady state error and convergence rate of the fixed step-size conventional CMA. The authors verified the performance of VSS-CMA with simulations in frequency selective Rayleigh fading channels. Symbol constellation plots showing transmitted, received and equalized outputs are not included in simulations as performance parameter [6].

Ying Xiao analyzed the performance of decision feedback blind equalization based on Recursive Least Square (RLS) algorithm for forward filter updates and CMA for feedback filter updates by considering the underwater acoustic channel. Quadrature Phase Shift Keying (QPSK) modulation is used for simulation. Applying two algorithms; namely, RLS for forward filter and CMA for feedback filter will increase the complexity of the algorithm structure even if RLS algorithm has faster convergence rate and lowest convergence steady-state error [7].

Kaiyu Qin *et al.* presented dual-mode blind equalization algorithm for high-order QAM. The proposed algorithm consists of Generalized Multi-Modulus Algorithm (GMMA) for initial convergence during blind startup, and DDLMS algorithm which is used to enhance the performance. From the results, both GMMA and proposed dual-mode equalization algorithms can recover the arbitrary phase error due to the channel, and the constellation of proposed dual-mode equalization algorithms is more concentrated /equalized outputs are extracted clearly/ than the GMMA's [8].

Sun Yunshan *et al*. introduced variable step-size in the CMA. Theoretical analysis and simulation result indicated the VSS-CMA has more superior restraining performance (has quicker convergence rate) than the conventional CMA blind equalization. The performance metric used by the authors' is only MSE [9].

Nigel McGinty discussed the issue of transitioning from a CMA blind equalizer to a DFE based DD (Decision Directed) blind equalization method. A cascaded structure consisting of a transversal filter, updated using the CMA algorithm, and a DFE updated using a DDLMS algorithm was considered. The performance of the algorithm has been illustrated using QPSK signal as an input [10].

Ram Nishanth*et al.* compared the performance of stop-and-go decision directed algorithm, CMA and Wei Rao's modified CMA blind equalization algorithms for QAM constellation across linear band-limited channel. The paper concludes that stop-and-go algorithm has a better performance than others when compared for MSE [11].

PratimaManhas and M.K Soni dealt with the channel equalization techniques: Least Mean Square (LMS), RLS and CMA used for Orthogonal Frequency Division

Multiplexing (OFDM) system. They performed comparative analysis of equalization techniques in terms of Bit Error Rate (BER). Based on simulation results, they concluded that, CMA equalizer is used to enhance the performance of OFDM system due to its minimum value of BER as compared with LMS and RLS equalizer [12].

Wei Rao*et al*. proposed DFE operate with CMA and DD algorithm to overcome ISI faster in the communication system without the aid of training sequences. For illustrating the performance of the proposed DFE, they carried out a numerical simulation using 4-QAM and 16-QAM data symbols transmitted through underwater acoustic channel and compared proposed DFE with DFE-CMA based on the received symbols constellation plots. Consequently the proposed DFE has the faster convergence rate than the DFE-CMA which can improve the performance for the system. In this paper, the comparative study has been performed based on only DFE extensions [13].

G.R. Mishra *et al.* presented the performance analysis of RLS and CMA algorithm in presence of noisy audio signal. The analysis shows that in CMA algorithm the convergence rate is low as compare to RLS algorithm whereas CMA algorithm requires low computing power and relatively better performance. Dual-mode blind equalization algorithms are not considered in the paper [14].

Tewodros Amsalu investigated the performance of three blind adaptive equalization techniques; namely, CMA, MMA and FS CMA for application in audio broadcasting. Performance metrics used for simulations are Symbol Error Rate (SER), rate of convergence, stability, complexity and audibility. From the simulation results, FS CMA has much better SER performance than the CMA and MMA [15].

1.5 Methodology

Firstly, review of books, journals and articles about adaptive equalization techniques has been conducted giving high attention to blind adaptive equalization algorithms. Based on the findings, VSS-CMA, DFE-CMA and joint VSS-CMA-DDLMS blind algorithms are identified and selected for performance analysis. Secondly; simulation parameters are selected, channel models are identified and flow charts are developed for each of the respective algorithms. Finally, simulations are done using MATLAB R2014a software and GUI with app designer application to compare performances of selected algorithms.

1.6 Contributions

The contributions of the thesis can be listed as follows:

- Creating better understanding of the three selected blind equalization algorithms including programmable simulation flowcharts;
- Performance analysis of blind equalization techniques for audio broadcasting system by considering different broadcast channel models;
- Applying a customized GUI with app designer application as performance analysis tool to identify algorithms with better performance through test simulations.

1.7 Organization of the Thesis

This paper is organized into five separate chapters. Chapter 1 provides a brief introduction to the topic and objectives of the work. Chapter 2 discusses the mathematical explanations and working principles of adaptive blind equalization algorithms. Chapter 3 is about digital radio broadcasting, standards, channel models and characteristics. Chapter 4 tells about results and discussions. Finally, conclusions and recommendations of this thesis are given in Chapter 5.

CHAPTER 2

Channel Equalization Techniques

2.1 Introduction

Equalization is a process in which symbols sent by the transmitter can be recovered correctly from the received signals that suffer from additive noise and channel distortion, known as the ISI. ISI can severely corrupt the transmitted signal and make it difficult for the receiver to directly recover the send symbol [16]. In order to minimize/remove the effect of ISI, equalization techniques are widely used.

2.2 Adaptive Equalizer

The wireless broadcast channel is random and time varying in nature, thus a variety of adaptive equalizers can be used to truck the time varying characteristics of the channel. An adaptive equalizer is an equalization filter that automatically adapts to time-varying properties of the channel. It is a filter that self-adjusts its transfer function according to an optimizing adaptive algorithm [2,17].

Adaptive equalizations used in modern communication technology are classified as training-based (data aided), semi-blind and blind equalizations. The three classifications are discussed in the next sections giving high attention to blind adaptive equalization.

2.2.1 Training-based Equalization

In training-based equalization technique the transmitter sends a fixed-length sequence which is either a known pseudo-random binary sequence or a predetermined pattern of bits. The equalizer uses this training sequence to establish the short-term channel characteristics and to optimize its settings. Following the training sequence the user data is sent. The adaptive algorithm in the equalizer tracks the changing channel and the equalizer continually updates its settings to follow changes in the channel characteristics [18].

The training-based equalization method has a quick conversion rate, better efficiency and has simple application. This method is considered best for environment where fast fading is required with high Doppler spread and little coherence time. The downside to these equalizers; however, is that they constantly need pilot signals that consumes a lot of bandwidth and power [19, 20].

Two of the most commonly used algorithms in adaptive training equalization practice are LMS and RLS algorithms.

LMS Algorithm:

LMS is the most common form of adaptive equalization. The LMS uses stochastic gradient descent for updating the equalizer weights during its operation, which means that the gradient of the error performance surface with respect to the free parameter vector changes randomly from one iteration to the next[19,21].

The LMS filter is built around a transversal (i.e. tap delay line) structure, which is responsible for performing the filtering process. And the weighting control mechanism is performing the adaptive control process on the tap weight of the transversal filter as shown in Figure 2.1 below [22].

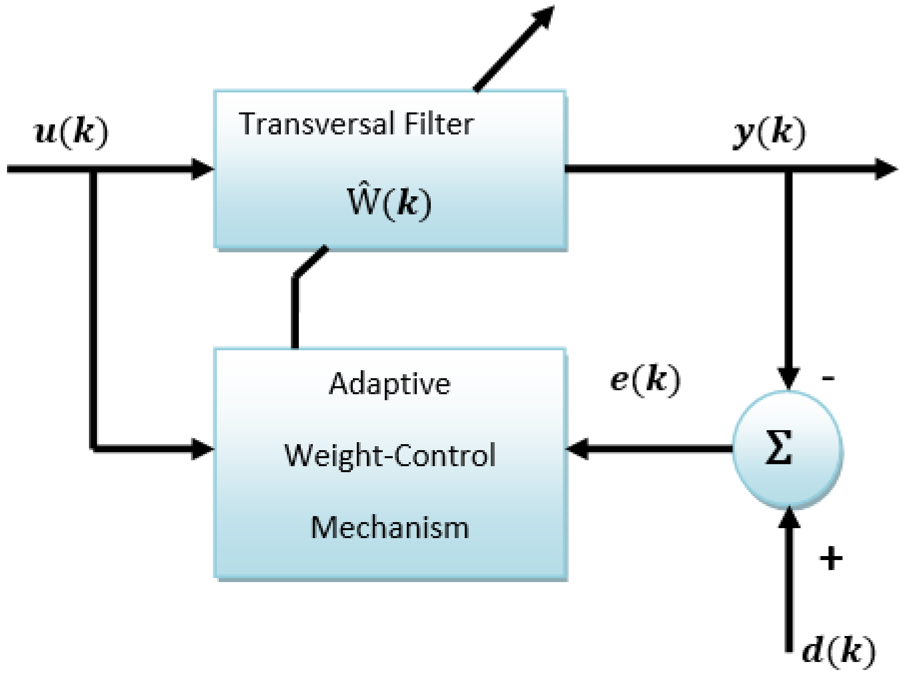


Figure 2.1 Basic block diagram of adaptive transversal filter employing LMS / RLS algorithms [22].

In Figure 2.1, $\boldsymbol{u(k)}$ is input signal, $\boldsymbol{e(k)}$ is error signal, $\boldsymbol{d(k)}$ is desired signal, $\boldsymbol{y(k)}$ is Transversal Finite Impulse Response (FIR) filter output and $\hat{W}(\boldsymbol{k})$ is weight vector of Transversal FIR filter.

The LMS algorithm updates the linear filter coefficients such that the MSE cost function is minimized [23].

LMS perform the following operation to update coefficients of the adaptive filter.

The error estimation $\boldsymbol{e(k)}$ is

$$\boldsymbol{e(k) = d(k) - y(k)} \quad \textbf{(2.1)}$$

$$\boldsymbol{y(k) = w(k) * u(k)} \quad \textbf{(2.2)}$$

Coefficient updating equation is

$$\boldsymbol{w}(\boldsymbol{k}+\mathbf{1}) = \boldsymbol{w}(\boldsymbol{k}) + \boldsymbol{\mu e}(\boldsymbol{k})\boldsymbol{u}(\boldsymbol{k}) \qquad \textbf{(2.3)}$$

where $\boldsymbol{\mu}$ is the step size of the adaptive filter that controls the convergence characteristics of the LMS algorithm, $\boldsymbol{w}(\boldsymbol{k})$ is weight vector and $\boldsymbol{u}(\boldsymbol{k})$ is the input signal vector [23].

RLS Algorithm:

RLS algorithm is another type of adaptive algorithm which has a higher computational complexity than its counterpart LMS [19].

As shown in Figure 2.1, RLS algorithm has the same structure with LMS algorithm, except that it provides a tracking rate sufficient for fast fading channel; moreover, RLS algorithm is known to have the stability issues due to the covariance update formula $\boldsymbol{p}(\boldsymbol{k})$, which is used for automatics adjustment in accordance with the estimation error as follows [2]:

$$\boldsymbol{p}(\mathbf{0}) = \boldsymbol{\delta}^{-\mathbf{1}}\boldsymbol{I} \qquad \textbf{(2.4)}$$

Where $\boldsymbol{p}$ is inverse correlation matrix and $\boldsymbol{\delta}$ is regularization parameter, positive constant for high Signal to Noise Ratio (SNR) and negative constant for low SNR.

For each instant of time $\boldsymbol{k}$**=1, 2, 3……**

$$\boldsymbol{\pi}(\boldsymbol{k}) = \boldsymbol{p}(\boldsymbol{k}-\mathbf{1})\boldsymbol{u}(\boldsymbol{k}) \qquad \textbf{(2.5)}$$

$$\boldsymbol{z}(\boldsymbol{k}) = \frac{\boldsymbol{\pi}(\boldsymbol{k})}{[\boldsymbol{\lambda}+\boldsymbol{u}^{\boldsymbol{H}}(\boldsymbol{k})\boldsymbol{\pi}(\boldsymbol{k})]} \qquad \textbf{(2.6)}$$

Time varying gain vector

$$\boldsymbol{\xi}(\boldsymbol{k}) = \boldsymbol{d}(\boldsymbol{k}) - \hat{\mathrm{W}}^{\boldsymbol{H}}(\boldsymbol{k}-\mathbf{1})\boldsymbol{u}(\boldsymbol{k}) \qquad \textbf{(2.7)}$$

Then the priori estimation error

$$\boldsymbol{w(k) = w(k-1) + z(k)\xi(k)} \tag{2.8}$$

The performance of most RLS-type algorithms in terms of convergence rate, tracking, misadjustment, and stability depends on the forgetting factor $\boldsymbol{\lambda}$. It is known that when the forgetting factor is very close to one, the algorithm achieves low misadjustment and good stability, but its tracking capabilities are reduced. A smaller value of $\boldsymbol{\lambda}$ improves the tracking but increases the misadjustment, and it could affect the stability of the algorithm [24].

2.2.2 Semi-blind Adaptive Equalization

In a semi-blind equalization both blind information of received symbols and known pilot symbols will be used to design the equalizers. Semi-blind techniques can avoid some the possible pitfalls of blind equalization techniques because they incorporate the information of known symbols. Furthermore, exploiting the blind information in addition to the known pilot symbols allows the estimation of longer channel impulse responses than possible with a certain training sequence length. This is a feature that of interest for the application of mobile communications in mountainous areas. Semi-blind equalization techniques are particularly promising when training-based and blind equalization techniques fail separately; the combination of both can be successful in such cases [25, 26].

The aim of this thesis is applying adaptive blind channel equalization with the selected blind algorithms.

2.2.3 Blind Adaptive Equalization

In the blind equalization no training sequence is used. However, blind equalization is capable to compensate amplitude and delay distortion of a communication channel.

The term blind indicate that the equalization is performed blindly on the data without a reference signal. The blind equalization depends on the statistical information of the received signal to adjust the parameter adaptively without the assistance of the training sequence [14]. Blind equalizer overcomes the disadvantage of the traditional data aided equalizer. It can be applied in high data rate, band limited digital communication systems in which the transmission of a training sequence is impractical or very costly [1, 27].

There have been many blind adaptive equalization techniques; such as CMA, Square Contour Algorithm (SCA),Reduced Constellation Algorithm (RCA), MMA, FS CMA, VSS-CMA, DFE-CMA and VSSCMA-DDLMS [13,16,19,29,63,64]. The three blind equalization techniques included in this thesis are: VSS-CMA, DFE-CMA and VSS-CMA-DDLMS and will be discussed in the following subsections.

2.2.3.1 Variable Step Size Constant Modulus Algorithm

Many communication signals, such as Frequency Modulation (FM), Phase Modulation (PM),Binary Phase Shift Keying (BPSK), and QAM have the Constant Modulus (CM) property, which means the amplitude of the signal is constant after these modulation schemes are applied, so the blind equalization algorithm are widely used for these techniques [16].

The CMA is one widely used blind equalization algorithm for QAM signals. CMA has the disadvantages of the slow convergence rate and large steady state MSE and phase-blind nature. Several new algorithms have been introduced to overcome the disadvantages of CMA [28]. In this thesis, VSS-CMA is proposed as one of the new algorithms (extensions of CMA).

The traditional CMA algorithm uses the fixed step-size. The less the step-size is, the smaller the stable state remainder error (shown in equation 2.17) is after algorithm restraining, and the smaller BER is. But the convergence rate will become also slower. In order to speed up the convergence rate, we must increase the step-size, but this way can cause the big remainder error and the BER, even causes radiation. Therefore, in the CMA algorithm, the convergence rate and the restraining precision are contradictory when the step-size was adjusted. The best method to solve this contradictory is utilizing the variable step-size. Namely in the algorithm iteration initial period, the lager step-size was adopted to enhance the convergence rate; after algorithm restraining the step-size was reduced to increase the restraining precision [29].

The new algorithm (i.e. VSS-CMA) utilized a nonlinear function of remainder error to control the step-size. The step-size would change in the algorithm iterative process [29].

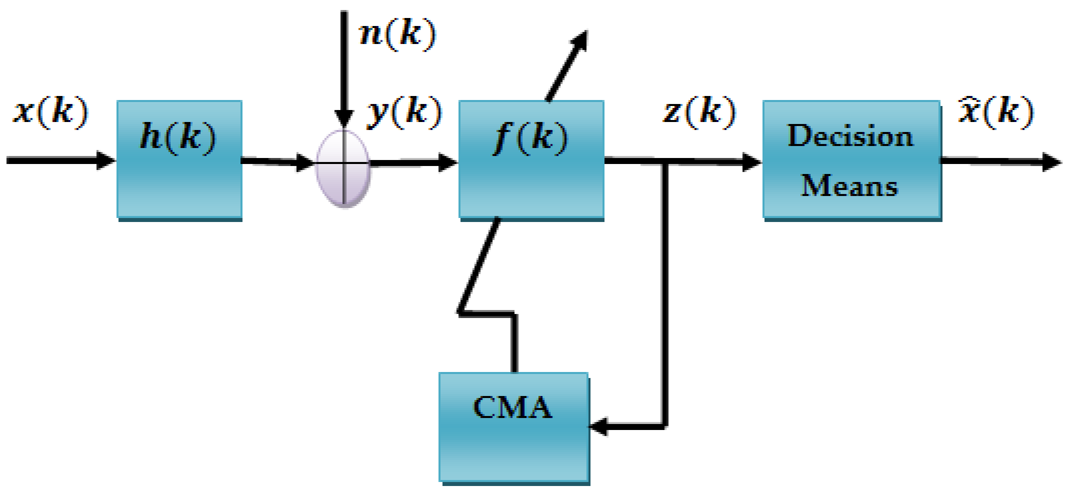


Figure 2.2 Structure of CMA [30].

In Figure 2.2, $\boldsymbol{x}(\boldsymbol{k})$ is the sending sequence, $\boldsymbol{h}(\boldsymbol{k})$ is the channel impulse response, $\boldsymbol{n}(\boldsymbol{k})$ is the noise sequence, $\boldsymbol{y}(\boldsymbol{k})$ is the receiving sequence and it is also the input signal of blind equalizer, $\boldsymbol{z}(\boldsymbol{k})$ is the output restoration signal of blind equalizer, $\widehat{\boldsymbol{x}}(\boldsymbol{k})$ is the decision output signal.

The structure of transversal filter is shown in Figure 2.3.

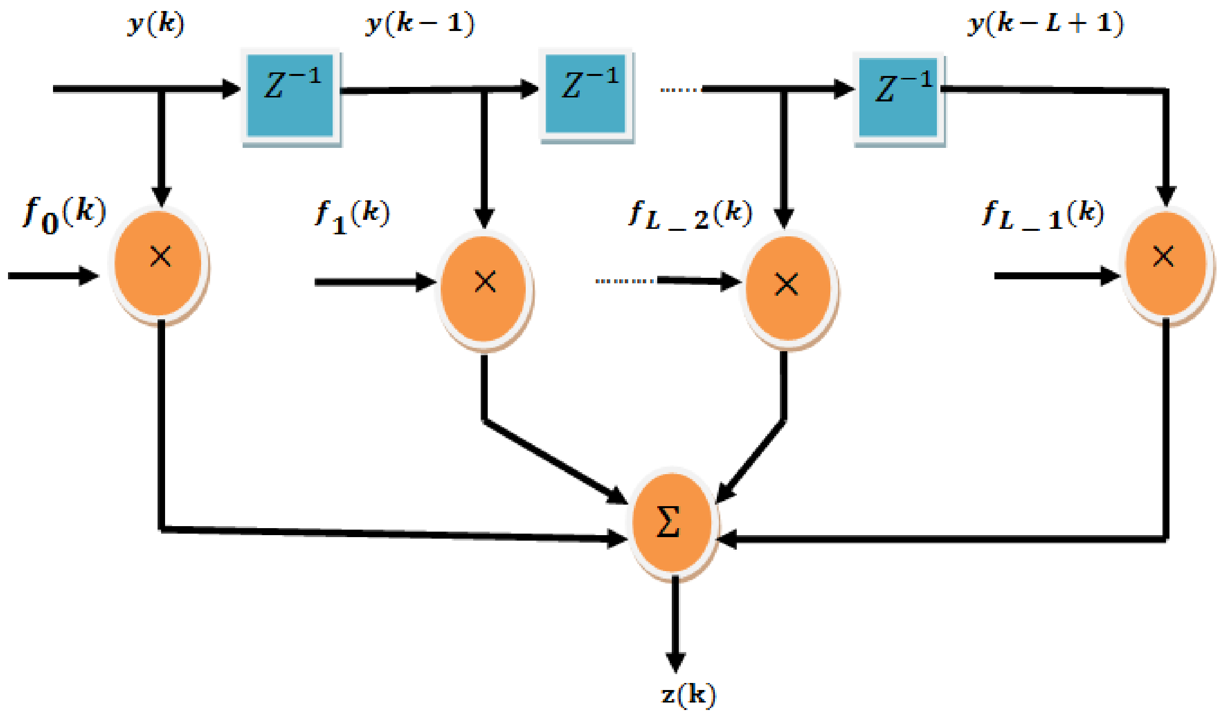


Figure 2.3 Transversal filter structure [29].

In Figure 2.3, $\boldsymbol{y(k)}$ is the input sequence of the filter, $\boldsymbol{L}$ is the length of transversal filter and $\boldsymbol{f(k)}$ is the tap coefficient of the equalizer. According to the signal processing theory, operator "*"represents convolution, and

$$\boldsymbol{y(k) = x(k) * h(k) + n(k)}$$

$$= \boldsymbol{\sum_i h(k) x(k-i) + n(k)} \quad \textbf{(2.9)}$$

The equalizer output is

$$\boldsymbol{z(k) = f(k) * y(k)} \quad \textbf{(2.10)}$$

Let transversal filter input sequences vector $\boldsymbol{y(k)}$ in Figure 2.3 is

$$\boldsymbol{y(k) = [y(k), y(k-1), \ldots, y(k-L+1)]^T} \quad \textbf{(2.11)}$$

$$\boldsymbol{\mathrm{f}(k) = [f_0(k), f_1(k-1), \ldots, f_{L-1}(k-L+1)]^T} \quad \textbf{(2.12)}$$

The cost function of CMA is

$$J(k) = E[(|z(k)|^2) - R^2] \quad \textbf{(2.13)}$$

The error function of CMA is

$$e(k) = z(k)(|z(k)|^2 - R) \quad \textbf{(2.14)}$$

R is the constant depending only on the transmitted data statistics.

It is defined as:

$$R = \frac{E[|x(k)|^4]}{E[|x(k)|^2]} \quad \textbf{(2.15)}$$

In the CMA, the iteration formula of the tap coefficient is

$$\mathbf{f}(k+1) = \mathbf{f}(k) - \mu e(k)\mathbf{y}^*(k) \quad \textbf{(2.16)}$$

where μ is the step size factor which is usually a quite small positive constant [29].

But the fixed step-size of the conventional CMA makes the convergence rate and the convergence precise become a contradiction, which makes the development and the application of CMA limited [31].

By utilizing variable step-size to the CMA can solve this contradictory that is we increase the step size to accelerate the speed of convergence in the initial and reduce the step size to improve the precision of convergence when it is close to the convergence state.

The residual error$d(k)$ is [29].

$$d(k) = \hat{x}(k) - z(k) = W^T(k)y(k) + \zeta(k) \quad \textbf{(2.17)}$$

where $W(k)$ is called weight error vector, $\zeta(k)$ is the Gaussian noise which is independent and identically distributed and has zero mean value.

In the process of implementation of the VSS-CMA, the value of weight error vector $W(k)$ is tending to zero and the value of the residual error also decreases. After the algorithm converged, $d(k)$ tends to be a very small number, it is proper to use $\boldsymbol{d(k)}$ to control the value of step size. However, because of the existence of $\boldsymbol{\zeta(k)}$,$\boldsymbol{d(k)}$ is sensitive to the interference signal. In some unstable channel, $\boldsymbol{d(k)}$ can suddenly be very large if there is a strong interference. In this situation we cannot directly use $\boldsymbol{d(k)}$ to control the step-size, otherwise, the step-size will be too large and the convergence of the algorithm cannot be assured. So we use a nonlinear function and set $\boldsymbol{d(k)}$ as a parameter of the function to make sure the step-size will be influenced by the residue error but will be varied in a reasonable scope [29].

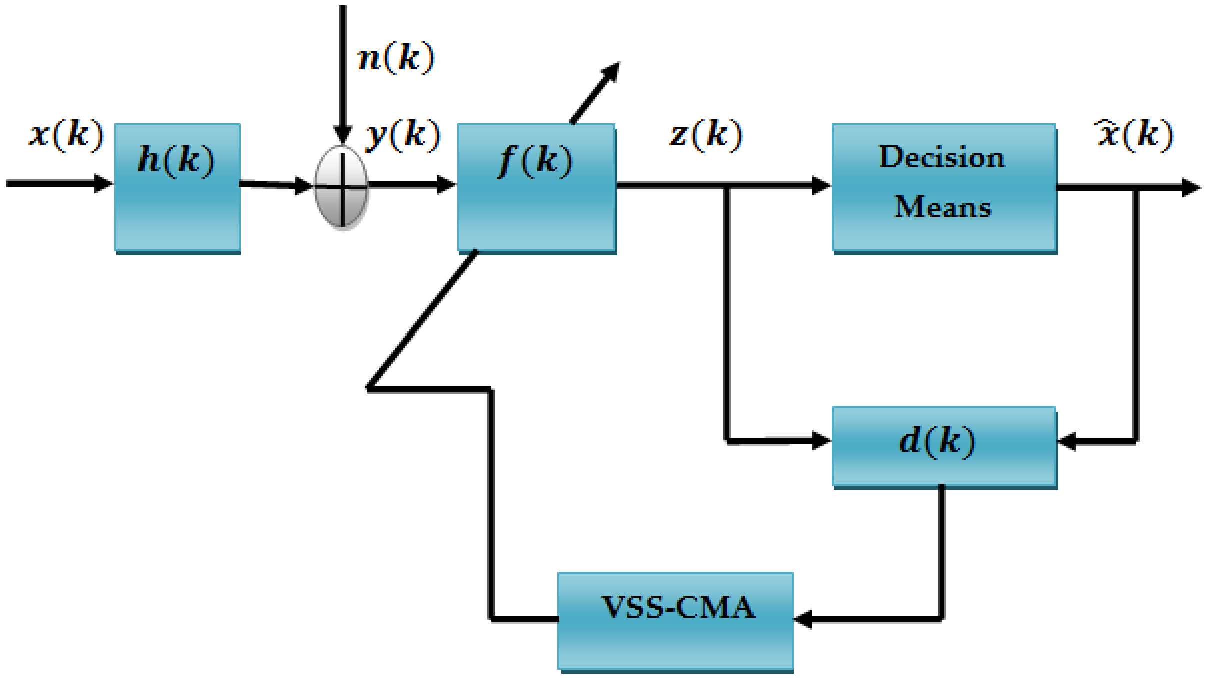
The step function controlled by the residual error is [29]

$$\boldsymbol{\mu(k)} = \frac{\boldsymbol{\delta|d(k)|^{\alpha}}}{\boldsymbol{(1+|d(k)|^{\alpha})(1+\exp(-|d(k)|^{\beta}))}} \quad \textbf{(2.18)}$$

The iteration formula of the tap coefficient is

$$\mathbf{f}(\boldsymbol{k+1}) = \mathbf{f}(\boldsymbol{k}) - \boldsymbol{\mu(k)e(k)}\mathbf{y}^*(\boldsymbol{k}) \quad \textbf{(2.19)}$$

In (2.19), $\boldsymbol{\beta}$, $\boldsymbol{\delta}$, $\boldsymbol{\alpha}$, are all constants. $\boldsymbol{\delta}$ controls the whole range of values of $\boldsymbol{\mu(k)}$ so that it can adjust the convergence rate of the algorithm. $\boldsymbol{\alpha}$ and $\boldsymbol{\beta}$ are used to change the shape of the function, different values of$\boldsymbol{\alpha}$ and $\boldsymbol{\beta}$ are taken, different curvatures of the function $\boldsymbol{\mu(k)}$ can be achieved. So by taking the appropriate values, we can make that if $\boldsymbol{d(k)}$ is small then $\boldsymbol{\mu(k)}$ is small, if $\boldsymbol{d(k)}$ is increasing then $\boldsymbol{\mu(k)}$ is also increasing at a reasonable speed and amplitude. This heuristic is very useful in a variety of control and signal processing applications [32].



`Figure 2.4 Basic Structure of VSS-CMA [29].

The convergence of the VSS-CMA is as follows: when

$0 \leq \frac{|d(k)|^{\alpha}}{(1+|d(k)|^{\alpha})(1+\exp(-|d(k)|^{\beta}))} < 1$, the value scope of $\mu(k)$ satisfies: $0 \leq \mu(k) < \delta$.

In order to guarantee the algorithm restrained, the step size must ensure that

$$0 \leq \mu(k) < \frac{2}{3 * tr(\mathbf{R}yy)} \tag{2.20}$$

where $\mathbf{R}yy$ is the autocorrelation matrix of the receiving sequence$y(k)$, $\mathbf{tr}(\mathbf{R}yy)$ is the trace of $\mathbf{R}yy$[29].

2.2.3.2 Decision Feedback Equalizer Constant Modulus Algorithm

Among blind equalization techniques, the linear equalizer should not compensate the channel with deep spectral zeros though its easy structure. DFE can adapt to different types of channels because the feedback filter of the DFE has the characteristics of non-linear [33]. A DFE incorporates a Feed-Forward Filter (FFF) operating on the received signal to suppress pre-cursor ISI, and a Feed Back Filter (FBF) operating on previously detected symbols to suppress post-cursor ISI. DFE uses a nonlinear decision device at the output, and the output represents a noise-free replica of the transmitted symbol assuming that the probability of decision error is small [34].

The DFE usually adopts the CMA, because CMA is one kind of blind equalization which operating without any explicit training sequence, unlike the conventional adaptive equalization, can save the bandwidth effectively, and CMA has strong robustness and can be easily implemented [13].

The structure of DFE-CMA is shown in Figure below.

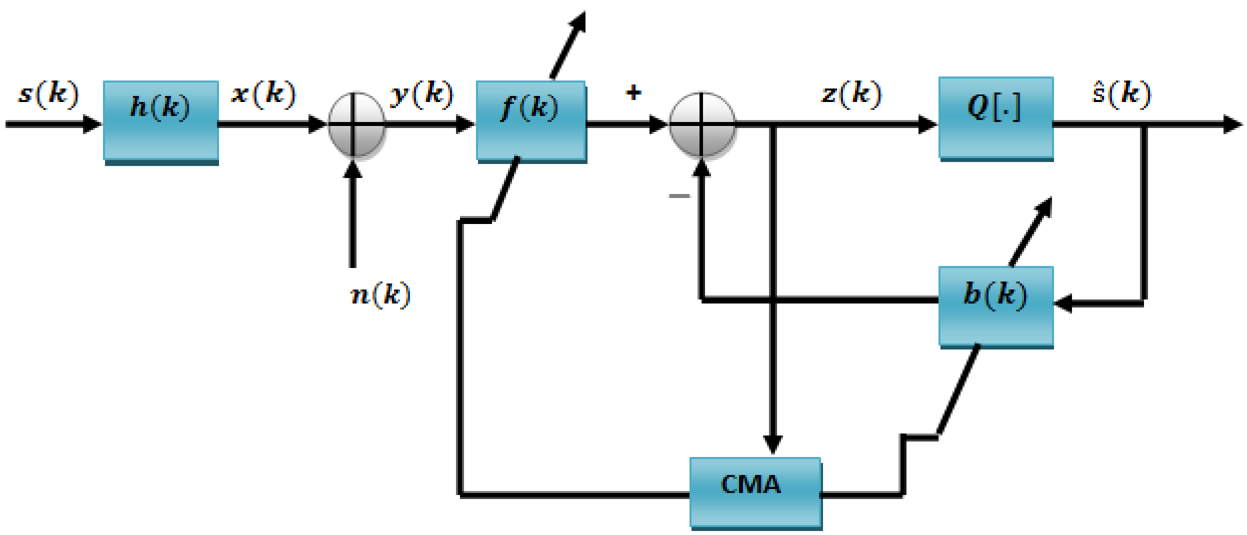


Figure 2.5 Structure of the DFE-CMA [33].

In Figure 2.5, $\boldsymbol{x(k)}$ is the channel input sequence, $\boldsymbol{h(k)}$ is the impulse response of the channel, $\boldsymbol{n(k)}$ is additive white gaussian noise, $\boldsymbol{Q[.]}$is the decision device, $\hat{s}(\boldsymbol{k})$ is the output of decision device, $\boldsymbol{f(k)}$ is the weight vector of FFF and $\boldsymbol{b(k)}$ is the weight vector of FBF.

The received signal [33]

$$y(k) = h(k) * s(k) + n(k) \quad (2.21)$$

Here "*" denotes convolution, $s(k)$ is the transmitted symbol, $h(k)$ is the channel impulse response and $n(k)$ is the additive white Gaussian noise.

The FFF's taps weight vector with the length of N_f is defined as

$$f(k) = [f(k), f(k-1), L, f(k-N_f+1)]^T \quad (2.22)$$

The taps weight vector of the FBF with the length of N_b is defined as

$$b(k) = [b(k), b(k-1), L, b(k-N_b+1)]^T \quad (2.23)$$

The input recurrent sequence of FBF is defined as

$$\hat{A}(k) = [\hat{s}(k), \hat{s}(k-1), L, \hat{s}(k-N_b+1)]^T \quad (2.24)$$

Then, the equalizer weight vector of DFE adjusts by CMA is [13]

$$y(k) = x(k) + n(k) = h^T(k)s(k) + n(k) \quad (2.25)$$

$$z(k) = f^T(k)y(k) - b^T(k)\hat{A}(k) \quad (2.26)$$

The error signal $e(k)$ used to update the weight vectors of both FFF and FBFs is defined by

$$e(k) = z(k)[|z^2(k)| - R^2] \quad (2.27)$$

The cost function of CMA is defined by

$$J(k) = E[(|z(k)|^2 - R^2)^2] \quad (2.28)$$

where $E[.]$ indicates statistical expectation. R^2 is the constant depending only on the input data symbol $s(k)$ and it is defined as [13]

$$R^2 = E[|s(k)|^4/E[|s(k)|^2] \quad (2.29)$$

According to the stochastic gradient descent algorithm, decision feedback blind equalization updates weights based on CMA is given by [7]

$$f(k) = f(k-1) + \mu y^*(k)e(k) \quad (2.30)$$

$$b(k) = b(k-1) - \mu \hat{A}^*(k)e(k) \quad (2.31)$$

where μ is the step size and the symbol "*" denotes conjugate operation.

2.2.3.3 Joint VSSCMA-DDLMS Dual-Mode Algorithm

Compared with the DDLMS algorithm which is commonly used in the adaptive equalizer, the performance of the VSS-CMA is unsatisfactory. We can combine VSS-CMA with DDLMS algorithm to form a new dual-mode blind equalization algorithm which has a better equalization performance and the convergence rate is fast while the steady state error is smaller. The basic principle of DDLMS algorithm is the same as the LMS algorithm [29].

The structure of blind equalization system based on the joint VSSCMA-DDLMS algorithm is shown in Figure 2.6.

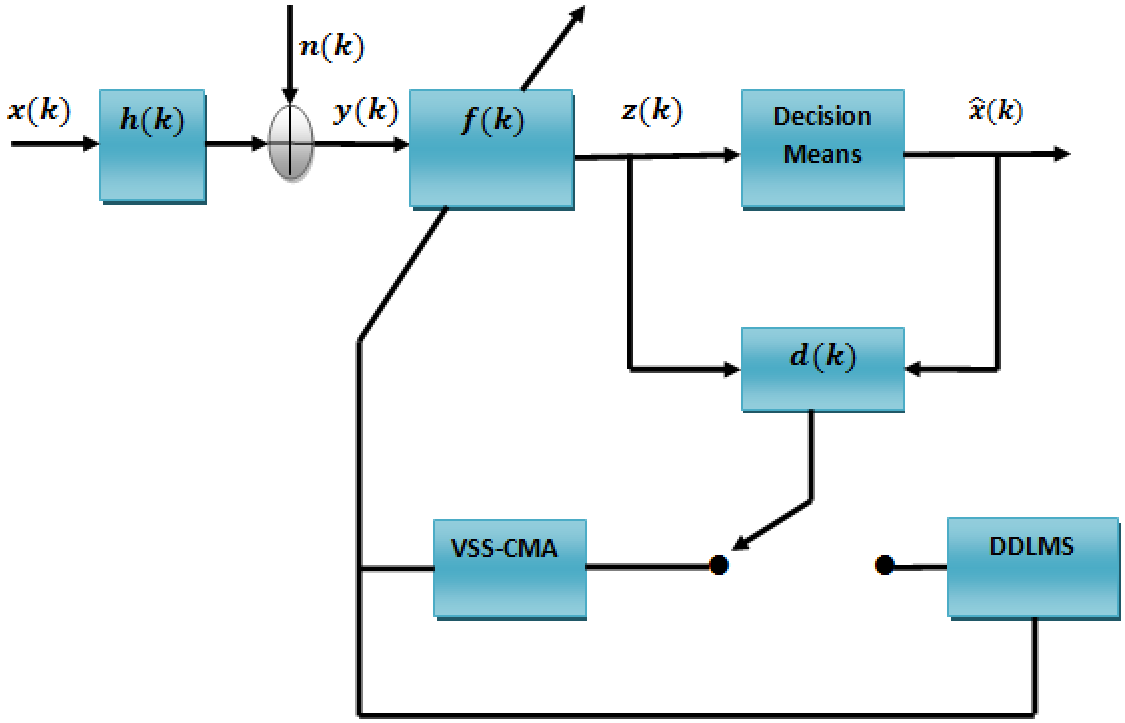


Figure 2.6 Joint VSSCMA-DDLMS dual-mode algorithm structure [29].

In Figure 2.6, $\boldsymbol{x(k)}$ is the sending sequence, $\boldsymbol{h(k)}$ is the channel impulse response, $\boldsymbol{n(k)}$ is the noise sequence, $\boldsymbol{y(k)}$ is the receiving sequence , $\boldsymbol{z(k)}$ is the output of blind equalizer, $\boldsymbol{\hat{x}(k)}$ is the decision output signal and $\boldsymbol{d(k)}$ is the residual error.

The LMS algorithm uses the square of the error between the equalizer output and the ideal response instead of the mean square error as the cost function. The algorithm uses the steepest descent method that is along the opposite direction of the gradient vector of the cost function to adjust the equalizer tap coefficient vector [29].

The iteration formula of the tap coefficient is

$$\boldsymbol{f(k+1) = f(k) - \mu_{LMS} e(k) y^*(k)} \quad \textbf{(2.32)}$$

The error of the DDLMS algorithm is defined as the difference between the received signal of the decision means and the output signal of the decision means.

$$\boldsymbol{e(k)_{DDLMS} = z(k) - \hat{x}(k)} \tag{2.33}$$

The iteration formula of the tap coefficient is the same as that of the LMS algorithm.

$$\boldsymbol{f(k+1) = f(k) - \mu_{DDLMS} e(k) y^*(k)} \tag{2.34}$$

The cost function is

$$\boldsymbol{J(k) = E[e(k)^2]} \tag{2.35}$$

The initial idea of combining VSS-CMA with DDLMS algorithm is using the switching dual-mode. In the beginning of the equalization, the error between the received signal and the hoped signal is large, VSS-CMA the convergence rate and the property of which are good can be used at this stage.

When the equalizer working for a while, the eye diagram of the channel is opening, it is time to switch DDLMS mode to obtain faster convergence rate and smaller steady state error. When the equalized points fall on the interval $\boldsymbol{Cmin}$ which means $|\boldsymbol{d(k)}| = |\boldsymbol{\hat{x}(k) - z(k)}| \leq \boldsymbol{Cmin}$,the error is small, DDLMS algorithm can be used to control the tap coefficients iterative formula.

If the points fall out of the interval$\boldsymbol{Cmin}$, VSS-CMA can be chosen. The large value of $\boldsymbol{Cmin}$ may lead to larger steady state error or the algorithm does not converge. However, if the value of $\boldsymbol{Cmin}$ is too small, the convergence process will have a big time delay, lead to the loss of the superiority of the joint algorithm [35]. In order to overcome the contradictions of such a choice, two option radiuses can be set, namely$\boldsymbol{Cmin}$ and when $|\boldsymbol{d(k)}| \geq \boldsymbol{Cmax}$,VSS-CMA algorithm can be chosen. When $|\boldsymbol{d(k)}| \leq \boldsymbol{Cmin}$, we use DDLMS.

So the iteration formula of the tap coefficient is [29]

$$\begin{cases} \boldsymbol{f(k+1) = f(k) - \mu(k)_{VSSCMA} e(k)_{VSSCMA} y^{*}(k) for |d(k)| \geq Cmax} \\ \\ \boldsymbol{f(k+1) = f(k) - \mu(k)_{DDLMS} e(k)_{DDLMS} y^{*}(k) for |d(k)| \leq Cmin} \end{cases} \quad \textbf{(2.36)}$$

where, $\boldsymbol{e(k)_{VSSCMA}}$ and $\boldsymbol{e(k)_{DDLMS}}$ are given in equation (2.14) and (2.33) respectively.

CHAPTER 3

Channel Models In Digital Broadcasting System

3.1 Introduction

Radio broadcasting is one of the most widespread electronic mass media comprising of hundreds of programme providers, thousands of High Frequency (HF) transmitters and billions of radio receivers worldwide. Since the broadcasting began in the early 1920s, the market was widely covered by the Amplitude Modulation (AM) services. Then came the FM and now we live in a world of digital communication systems and services. Digital telecommunication has advantages over analog systems such as storage capacity, reliability, and quality of service, miniaturization and many more. The new digital radio system, Digital Audio Broadcasting (DAB) has the capability to replace the existing AM and FM audio broadcast services in many parts of the world in near future [36].

3.2 Digital Radio Standards

The digital radio standards have extensive advantages over analog systems. Among some of these include: audio quality, service reliability, programming, coverage flexibility receiver, spectrum, and power efficiency, as well as new business opportunities for the broadcaster. Due to inadequate spectrum resources and increasing demands for high-quality multimedia services, traditional analog audio broadcasting is being migrated into digital radio around the world [37].

Recently various digital radio broadcasting techniques and standards have been proposed, and many countries have been considering converting their analog radio broadcasting into digital broadcasting. However, since a small number of countries have achieved the digital conversion of their analog radio broadcasting (some European countries and the United States), most countries still face converting their analog radio broadcasting service into digital service [37].

The most common specifications for DAB are proposed in the Europe, Japan, and United States. In the next subsections some of these standards are briefly covered.

3.2.1 Digital Audio Broadcasting

The DAB standard has been developed within an European project called Eureka 147 [37]. The Eureka 147 DAB system is very reliable, multi-programme, digital radio broadcasting system, intended mainly for robust reception by mobile, portable and fixed receivers, using simple antennas [38].

DAB is a whole digital radio broadcasting system that not only is intended for audio services as usual, but also for low data rate, graphs, Hypertext Mark-up Language (HTML) pages, ancillary data, just to name a few. It is available mainly in a terrestrial propagation system but it is also thought to be supported by satellite coverage and cable networks.

The DAB standard comes up from the necessity to exploit the benefits of the digital technology in terrestrial broadcasting in order to improve the characteristics and the services provided by the analogue radio networks. The radio is one of the most widespread communication media, but in the analogue versions, both AM and FM, do not provide high quality for mobile receivers [39]. Analogue networks are able, of course, to provide radio services for the most of the mobile and portable users, but they do not offer any kind of protection against the multipath interference which is impossible to be avoided when we are talking about mobile communication systems or even the AM signal often suffers fading and interference problems [39].

The Eureka 147 DAB system is the most significant advancement in radio broadcasting technology since the introduction of FM and AM. It offers both listeners and broadcasters a unique combination of benefits and opportunities. These include the following [38]:

- Both music and data services (such as text information, news, graphics, still or moving pictures) can be received using the same receiver.
- The DAB system has universal and well standardized system layout and a wide range of receiving equipment including fixed (stationary), mobile and portable radio receivers.
- Efficient use of the available radio frequency spectrum.
- Provides flexible bit rates between 8 and 384 Kbit/s.
- DAB services can be transmitted in a flexible multiplex configuration.
- DAB transmitter networks can be designed as Single Frequency Networks (SFNs).
- Low transmitting power.
- High robustness against multipath reception.
- Larger coverage area than current AM and FM systems.

Today, the various types of DAB receivers can be categorized as follows [40]:-

- Car radios, either audio only or audio and data receivers.
- Personal Computer (PC)-card receivers. Some of these solutions are completely hardware based, whereas other implementations decode the digital part in software on the PC.
- Home receivers including High Fidelity (HiFi) tuners and kitchen radios.
- Portable receivers.
- Monitor receivers for network monitoring.

The DAB system comprises a number of elements that are related and which are needed to implement a complete digital radio transmission chain. These include analog audio signal sampling, digital compression, data encoding, digital signal multiplexing, time stamping of the ensemble signal, digital distribution via telecommunication circuits, channel encoding and modulation, transmission, and finally reception [41]. Figure 3.1 presents the block schematic of a typical DAB receiver.

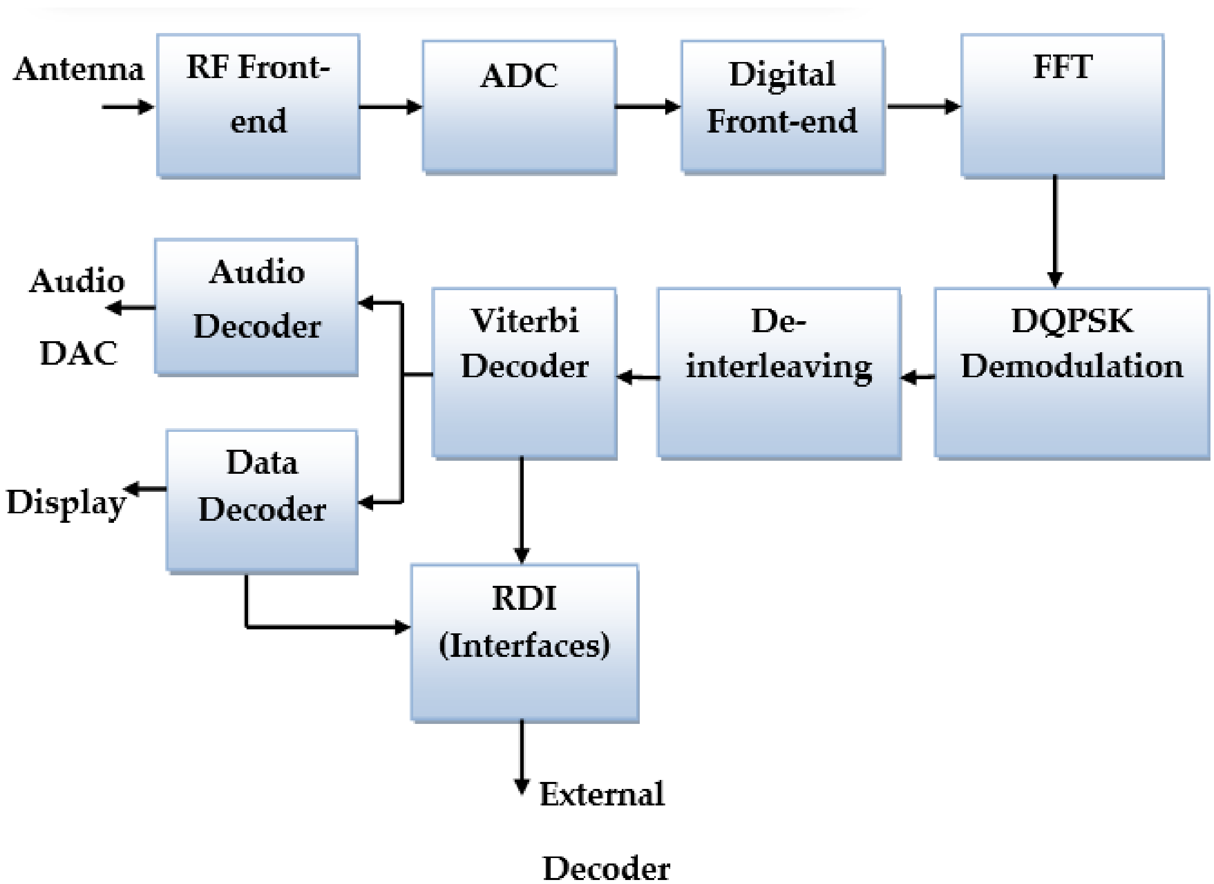


Figure 3.1 Receiver block schematic [40].

Radio Frequency (RF)-Front End

The signal received from the antenna is processed in the RF front-end, filtered and mixed to an intermediate frequency or directly to the complex baseband.

Digital Baseband Processing

Digital baseband processing is the generic term for all signal processing steps starting directly after digitization of the Intermediate Frequency (IF) signal using an Analog to Digital Converter (ADC) until the source coded data become available after Viterbi decoding. In the case of DAB, baseband processing includes the following processing steps [40]:

- Generation of the complex baseband signal
- Orthogonal Frequency Division Multiplexing (OFDM) demodulation, possibly combined with compensation for the frequency drift of the baseband signal
- Demodulation of the Differential Quadrature Phase Shift Keying (DQPSK) modulated carriers
- Time and frequency de-interleaving
- Channel decoding using the Viterbi algorithm
- Synchronization of time, frequency and phase

Audio Decoder

The audio coding scheme used in DAB is Moving Pictures Experts Group (MPEG)-1 and MPEG-2 layer II. For DAB use; these standards have been extended to provide further information for the detection of transmission errors in those parts of the bit-stream with the highest error sensitivity. This is useful for error concealment. Furthermore; the system provides a mechanism to reduce the dynamic range of the decoded audio signal at the receiver which is useful, especially in noisy environments like vehicles.

Interfaces

In principle, DAB receivers can be equipped with two types of interfaces: those that carry data to or from DAB receivers and those that carry control information.

Data Interfaces: The Receiver Data Interface (RDI) is a data interface which was specifically developed for DAB. It is suitable for connecting an external data decoder to a DAB receiver.

Control Interfaces: It is used to exchange control information, like the frequency, that the front-end should tune in [40].

3.2.2 In-band On Channel Digital Audio Broadcasting

In the United States a variant of DAB is being developed and has been licensed in some areas for broadcast trials. It is known as "In Band On Channel DAB" (IBOC DAB) because it shares the same frequency bands as FM signals. The IBOC DAB signal is broadcast from the same transmitter as is used for the analog FM service. This makes it attractive to broadcasters who can re-use existing equipment for analog and digital radio. The digital audio information is transmitted in each sideband and only one sideband needs be received to decode the signal. This signal redundancy not only protects against corruption of one sideband but also increases the performance of the system if both sidebands are decoded together [41].

It differs from the earlier described European DAB system in the areas of error correction, interleaving, and audio coding. These changes were made to allow compatibility of the IBOC DAB system with the existing analog FM and AM services [41].

IBOC systems are comprised of four building blocks similar to the Eureka 147/DAB's major subsystems [42].

Audio Source Coding: An audio Coder-decoder (Codec) is a source encoding device that filters out those parts of an analog signal that are irrelevant to the human ear. When decoded, the signal will not be identical to the original but will be perceived to be the same.

Channel Coding: The output stream from the audio codec is encoded using Forward Error Correction (FEC) and interleaving in the transmission system. This greatly improves the reliability of the transmitted information by carefully adding redundant information used to correct errors occurring in the transmission path. Advanced FEC coding techniques have been specifically designed for AM systems based on detailed interference studies to exploit the non-uniform nature of interference in the AM bands.

Modulation/Demodulation Techniques: A modem is a device that modulates a signal or demodulates it. AM systems use QAM scheme in conjunction with OFDM. FM systems also uses OFDM modulation but with the carriers modulated with QPSK modulation scheme.

Blending: Blending is a technique employed in IBOC DAB systems to seamlessly switch between digital-to-analog signals. Essentially, blending allows transition from the instantly acquired analog signal to the digital signal (after the receiver has acquired, decoded, and processed the signal). Once the digital signal is acquired, the receiver will transition to it in a seamless fashion. Should the digital signal become corrupted the receiver will seamlessly switch to the analog signal [42].

3.2.3 Integrated Services Digital Broadcasting

In Japan, the NHK (Japan Broadcasting Corporation) Science and Technical Research Laboratories have proposed a concept called Integrated Services Digital Broadcasting (ISDB). From this approach a system was created which can be configured for terrestrial and satellite broadcasts of radio, TV and multimedia services. For the terrestrial system (ISDB-T) the most important aims are rugged reception with portable and mobile receivers, use of SFNs to achieve frequency efficient coverage and to have a flexible scheme in order to be future-proof [43]. From the point of view of the relation to DAB, ISDB-T basically uses the same principles and hence provides similar features as do DAB and Digital Video Broadcasting Terrestrial (DVB-T), respectively. By using the MPEG-2 standard for coding, the audio and video quality will be comparable. The advantage of ISDB-T is that the radio and TV systems are based on the same basic building blocks (although the coding and modulation will be different), which allows for reuse of some circuits in receivers, whereas DAB and DVB-T differ in many details. Another advantage of ISDB-T is the availability of a narrow-band mode (i.e. a single segment) which can deliver one or two radio programmes to small regions or communities, although potentially with some reception problems due to flat fading. With this segmentation ISDB-T can also flexibly use the frequencies not occupied by analogue services in the introductory simulcast phase. Such possibilities are not present in DAB [43].

3.3 Radio Broadcasting Channel Models

In wireless communication systems the channel introduces random variations with time and/or frequency in both the amplitude and phase of the transmitted signal. These phenomena are collectively known as fading and dispersion. Fading originates due to various causes. The most common is the presence of multiple propagation paths, that is, the existence of a number of paths along which an electromagnetic signal propagates from a transmitting antenna to the receiving one [44].

The impacts of multipath include constructive and destructive interference and phase shifting of the signal. In digital radio communications, multipath propagation can affect the quality of communications [45].This multipath effect is illustrated in Figure

3.2.

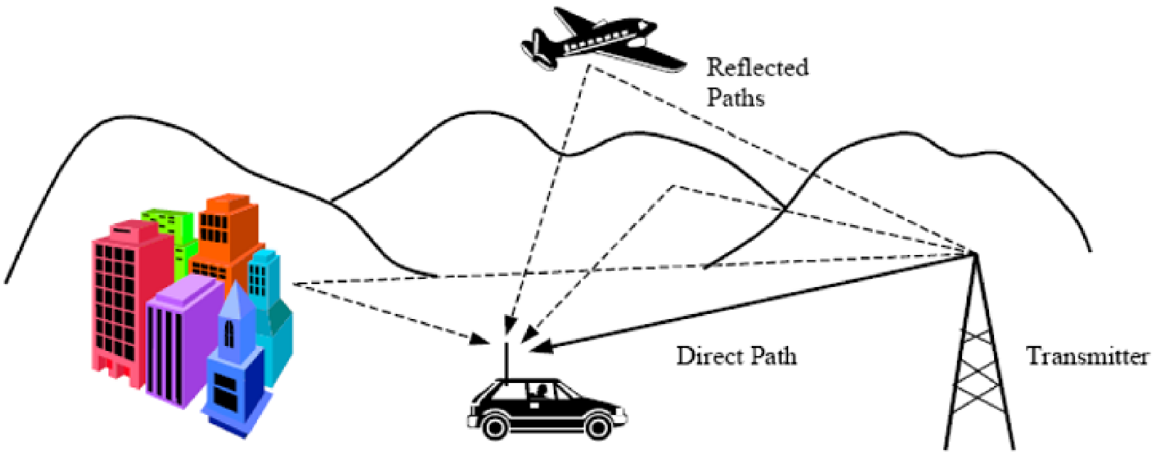


Figure 3.2 Effect of multipath on mobile receiver [46].

For mobile radio applications, the channel is time-varying because motion between the transmitter and the receiver results in propagation path changes.

Channel models discussed in this thesis are: AWGN channel, mobile radio channel (Rayleigh, Rician), ISI channel, measured channel impulse response, exponential decaying, and Joint Technical Committee (JTC) channel models.

3.3.1 AWGN Channel

AWGN channel is a universal channel model for analyzing modulation schemes. In this model, the channel does nothing but add a white Gaussian noise to the signal passing through it. This implies that the channel's amplitude frequency response is flat (thus with unlimited or infinite bandwidth) and phase frequency response is linear for all frequencies so that modulated signals pass through it without any amplitude loss and phase distortion of frequency components. Fading does not exist. The only distortion is introduced by the AWGN [47].

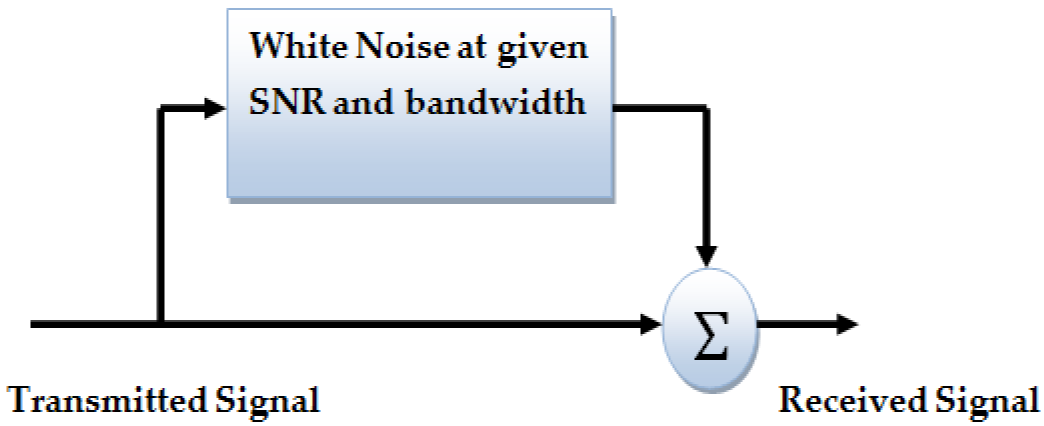


Figure 3.3 Block diagram of AWGN channel model [48].

Strictly speaking, the AWGN channel does not exist since no channel can have an infinite bandwidth. However, when the signal bandwidth is smaller than the channel bandwidth, many practical channels are approximately an AWGN channel. For example, the LOS radio channels, including fixed terrestrial microwave links and fixed satellite links, are approximately AWGN channels when the weather is good. Wideband coaxial cables are also approximately AWGN channels since there is no other interference except the Gaussian noise [47].

Additive Gaussian noise is ever present regardless of whether other channel impairments such as limited bandwidth, fading, multipath and other interferences exist or not. Thus the AWGN channel is the best channel that one can get [47].

3.3.2 The Mobile Radio Channel

The mobile radio channel is characterized by two types of fading effects: large-scale fading and small scale fading as shown in Figure 3.4.Large-scale fading is the slow variation of the mean (distant-dependent) signal power over time. This depends on the presence of obstacles in the signal path and on the position of the mobile unit. The large-scale fading is assumed to be a slow process and is commonly modeled as having log normal statistics. Small-scale fading is also called Rayleigh or Rician fading because if a large number of reflective paths is encountered the received signal envelope is described by a Rayleigh or a Rician Probability Density Function (PDF) [49].

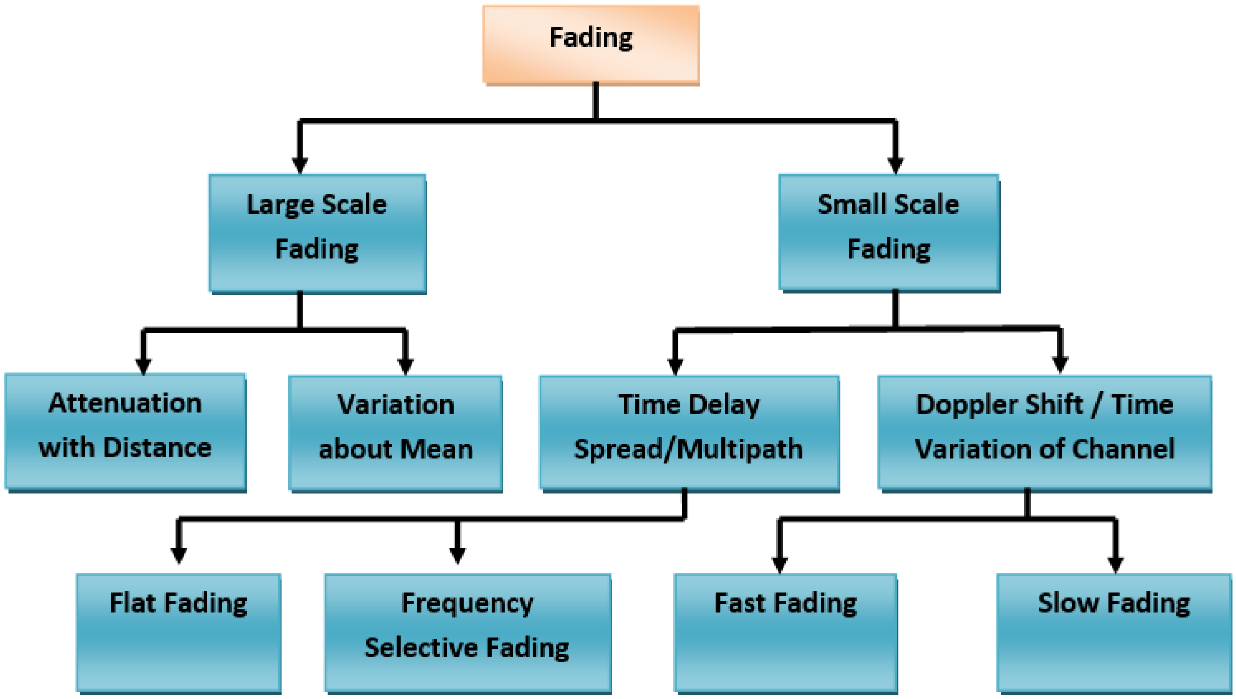


Figure 3.4 Fading manifestations [50].

3.3.2.1 Rayleigh Channel Model

Rayleigh fading is a statistical model for the effect of a propagation environment on a radio signal, such as that used by wireless devices. Rayleigh fading models assume that the magnitude of a signal that has passed through a communication channel will vary randomly, or fade, according to a Rayleigh distribution-the radial component of the sum of two uncorrelated Gaussian random variables [51].

Rayleigh fading is viewed as a reasonable model for tropospheric and ionospheric signal propagation as well as the effect of heavily built-up urban environments on radio signals. Rayleigh fading is most applicable when there is no dominant propagation along a line of sight between the transmitter and receiver. If there is a dominant line of sight, Rician fading may be more applicable [51].

The envelope of the received signal is statistically described by a Rayleigh PDF, given by [49]

$$f_R(r) = \begin{cases} \frac{r}{\sigma^2}\exp\left[-\frac{r^2}{2\sigma^2}\right] & for \;\; r \geq 0\,, \\ 0 & for \;\; r < 0 \end{cases} \qquad \textbf{(3.1)}$$

where r is the envelope amplitude of the received signal and σ is the rms value of the received voltage signal before envelope detection.

For mobile-radio channels, the Rayleigh distribution is widely used to describe the statistical time- varying nature of the received envelope of a flat-fading signal, or of an individual multipath component [52].

3.3.2.2 Rician Channel Model

It is a stochastic model for radio propagation anomaly caused by partial cancellation of a radio signal by itself-the signal arrives at the receiver by several different paths (hence exhibiting multipath interference), and at least one of the paths is changing. Rician fading occurs when one of the paths, typically a LOS signal, is much stronger than the others. In Rician fading, the amplitude gain is characterized by a Rician distribution [51].

Rayleigh fading is the specialized model for stochastic fading when there is no LOS signal, and is sometimes considered as a special case of the more generalized concept of Rician fading. In Rayleigh fading, the amplitude gain is characterized by a Rayleigh distribution [51].

This LOS signal adds a deterministic component to the multipath signal and has a PDF given by [49]

$$\begin{cases} f_R(r) = \frac{r}{\sigma^2} \exp\left[-\frac{(r^2+A^2)}{2\sigma^2} I_0\left(\frac{Ar}{\sigma^2}\right)\right] for \quad r \geq 0\,, \\ \qquad\qquad\qquad\qquad 0 \qquad\qquad for \quad r < 0 \end{cases} \tag{3.2}$$

where A is the peak amplitude of the dominant signal and I_0 is the modified Bessel function of the first kind and zero-order.

3.3.3 ISI Channel Model

Suppose that $u[n]$ is the source signal of interest and is distorted by ISI channel $h[n]$ as shown in Figure 3.6:

The received data $y[n]$is given as [53]:

$$y[n] = x[n] + w[n] \tag{3.3}$$

where

$$\mathrm{x[n]} = h[n] * u[n] = \sum_{k=-\infty}^{\infty} \mathrm{h[k]u[n-k]} \tag{3.4}$$

$\mathrm{x[n]}$is the noise-free signal and $\mathrm{w[n]}$ is the additive noise accounting for measurement noise as well as physical effects not explained by $x[n]$.

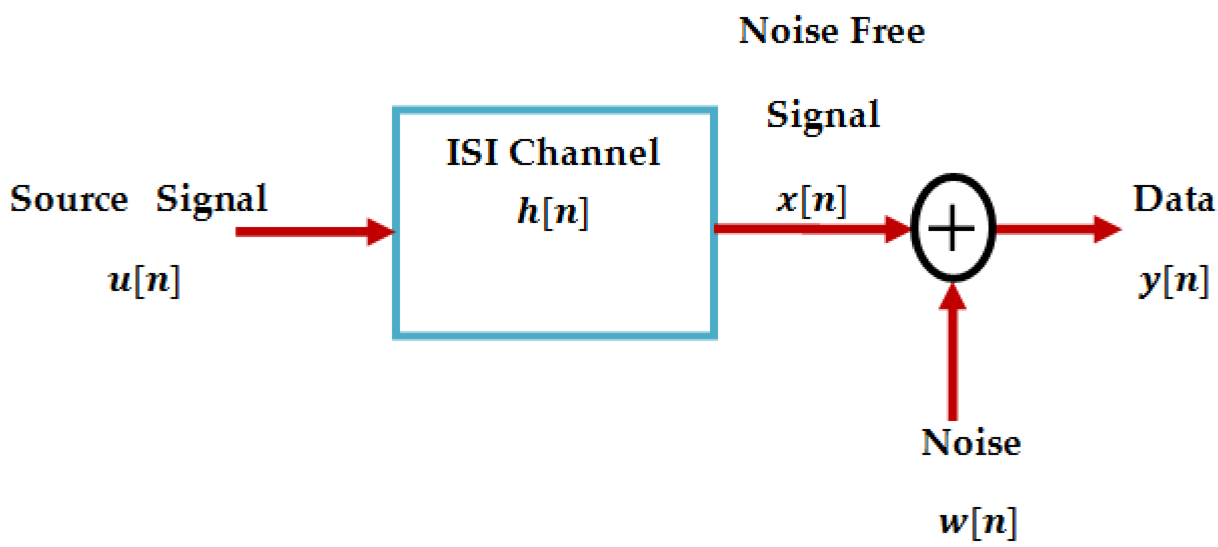


Figure 3.5 Block diagram of ISI channel model [53, 54].

From **(3.3)** and **(3.4)**, it follows that

$$y[n] = h[0]u[n] + \underbrace{\sum_{k=-\infty,k\neq 0}^{n} \mathrm{h[k]u[n-k]}}_{\textbf{ISI Terms}} + \mathrm{w[n]} \qquad \textbf{(3.5)}$$

From **(3.5)**, one can see that the desired sample (or symbol) $\boldsymbol{u[n]}$, scaled by $\boldsymbol{h[0]}$, in the first term is not only corrupted by the noise $\mathbf{w[n]}$ but also interfered with by other samples (or symbols) in the second term. This latter effect is called ISI [53].

Due to the ISI terms, the desired signal$\boldsymbol{u[n]}$cannot be clearly observed from the received signal. Therefore, it is necessary to eliminate the ISI terms to extract the desired signal [54].

This problem arises in a variety of engineering and science areas such as seismic exploration, digital communications, speech signal processing, ultrasonic nondestructive evaluation, underwater acoustics, radio astronomy, and so on [53].

The presence of ISI in the system introduces error in decision device at the receiver output. Therefore, in the design of transmitting and receiving filters, the objective is to minimize the effects of ISI and; thereby, deliver the digital data to its destination with the smallest error rate possible [55].In this thesis ISI channel model is used for simulation of three selected blind algorithms and its channel response is depicted in Figure 4.5 (d).

For GUI simulations the following three channel models [15] are considered in addition to ISI channel model.

- Measured Channel Impulse Response: the response of the channel used for simulation is shown in Figure 4.5 (b).
- Exponential Decaying Channel: the exponential channel model is the ideal channel model for software simulation and performance comparisons due it is easy to generate and it is a reasonably accurate model of the real world. Its impulse response is depicted in Figure 4.5 (c).
- Joint Technical Committee Model:-multipath propagation model recommended by the JTC for simulation of radio propagation is used as one of the channel for the simulation and impulse response of JTC is shown in Figure 4.5 (d).

CHAPTER 4

RESULTS AND DISCUSSION

This Chapter presents the results of simulations using MATLAB 2014a to examine the performances of blind adaptive equalization algorithms described in Chapter two. Comparative study of five adaptive blind channel equalization algorithms; namely, CMA, MMA, VSS-CMA, DFE-CMA and VSSCMA-DDLMS is also performed using GUI. GUI application is based on MATLAB software. Additional information can be found in appendices.

Before presenting the simulation results, it is helpful to look at discussion on performance parameters:-MSE, symbols constellation plots, convergence rate, computation time and complexity in the following sections.

4.1 Performance Parameters

The parameters that are used in this thesis to check the performance of an adaptive algorithm are listed as follows.

4.1.1 Mean Square Error

Mathematically, MSE is defined as [29]:

$$\mathbf{MSE} = \mathbf{10lg}[\sum_{k=0}^{N-1} |\mathbf{z}(\mathbf{k}) - \hat{x}(k)|^2/\mathbf{N}] \qquad \mathbf{(4.1)}$$

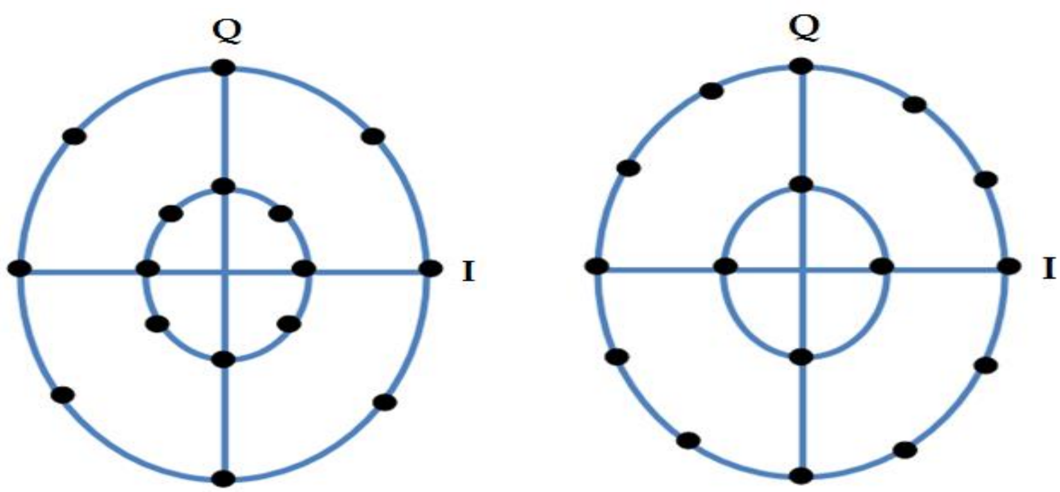
where, $\mathbf{z}(\mathbf{k})$ is output of blind equalizer,$\hat{x}(k)$ is decision output signal and $\mathbf{N}$ is number of iterations discussed on chapter two.

The MSE curve (graphical plot) shows how the error function behaves with the increase in the number of iterations.

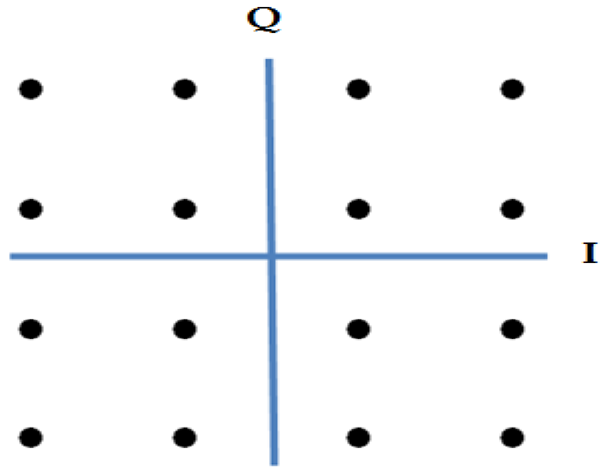
4.1.2 Symbols Constellation Plots

This type of performance measures is usually useful to see if the algorithm used is working or not. It is a constellation diagram of the observed signals and equalizers outputs. It is also called scatter or eye diagram [56].

A geometric representation called constellation is a very clear way of describing a QAM signal set. And a QAM signal is represented by a point (or vector, or phasor) with coordinates [57].The two axes sometimes are simply labeled as In phase (I)-axis and Quadrature (Q)-axis, and sometimes are even left unlabeled. Three types of QAM constellations are shown in Figure 4.1



a) Type I QAM constellation b) Type II QAM constellation



c) Type III QAM constellation

Figure 4.1 Three types of QAM constellations [57].

Type III system offered a very small improvement in performance over the type II system, but its implementation would be considerably better than that of type I and II. Due to this, the type III constellation has been the most widely used system [57].

For M-ary square QAM signals, (I_i, Q_i) are a pair of independent integers which determine the location of the signal point in the constellation. The minimum values of (I_i, Q_i) are $(\pm 1, \pm 1)$. The pair (I_i, Q_i) is an element of the $L \times L$ matrix [57]:

$$[\boldsymbol{I_i}, \boldsymbol{Q_i}] = \begin{bmatrix} (-\boldsymbol{L}+\mathbf{1}, \boldsymbol{L}-\mathbf{1}) & (-\boldsymbol{L}+\mathbf{3}, \boldsymbol{L}-\mathbf{1}) & \ldots & (\boldsymbol{L}-\mathbf{1}, \boldsymbol{L}-\mathbf{1}) \\ (-\boldsymbol{L}+\mathbf{1}, \boldsymbol{L}-\mathbf{3}) & (-\boldsymbol{L}+\mathbf{3}, \boldsymbol{L}-\mathbf{3}) & \ldots & (\boldsymbol{L}-\mathbf{1}, \boldsymbol{L}-\mathbf{3}) \\ \cdot & \cdot & & \cdot \\ \cdot & \cdot & & \cdot \\ \cdot & \cdot & & \cdot \\ (-\boldsymbol{L}+\mathbf{1}, -\boldsymbol{L}+\mathbf{1}) & (-\boldsymbol{L}+\mathbf{3}, -\boldsymbol{L}+\mathbf{1}) & \ldots & (\boldsymbol{L}-\mathbf{1}, -\boldsymbol{L}+\mathbf{1}) \end{bmatrix} \quad \textbf{(4.2)}$$

Where $\boldsymbol{L} = \sqrt{\boldsymbol{M}}, \boldsymbol{M} = \mathbf{4}^{\boldsymbol{n}}, \boldsymbol{n} = \mathbf{1}, \mathbf{2}, \mathbf{3}, \ldots$

For the 16-QAM used for simulation, L=4, the matrix is

$$[I_i, Q_i] = \begin{bmatrix} (-3,3) & (-1,3) & (1,3) & (3,3) \\ (-3,1) & (-1,1) & (1,1) & (3,1) \\ (-3,-1) & (-1,-1) & (1,-1) & (3,-1) \\ (-3,-3) & (-1,-3) & (1,-3) & (3,-3) \end{bmatrix} \quad \textbf{(4.3)}$$

4.1.3 Convergence Rate

The convergence rate is defined as the number of iterations required for the algorithm to converge to its steady state MSE [58].A fast rate of convergence allows the algorithm to adapt rapidly to a stationary environment of unknown statistics. Furthermore, it enables the algorithm to track statistical variations when operating in a non-stationary environment [59].

4.1.4 Computation Time

This is an important parameter from a practical view point. The computation time depends on the number of operations required for one complete iteration of the algorithm along with the amount of memory needed to store the mathematical calculations [58]. Table 4.3 shows computation time of the three algorithms.

4.1.5 Complexity

The complexity of the algorithm is an important indicator to measure the merits of the algorithm, and also an important indicator to determine whether the algorithm is conducive to hardware implementation [29].The complexity of the algorithms can be measured in two approaches. One is by considering number of addition, multiplication and exponentiation operations of algorithms iteration formula or based on the computation time of the algorithms. In this thesis the computation time is used to measure the complexity of algorithms.

4.2 Simulation Parameters

The parameters used for the simulation are identified, defined and summarized in the Table 4.1.

Table 4.1 Simulation Parameters

Parameter	Value/Technique
Audio Signal type	.mp3 audio file
Sampling frequency	48kHz
Quantization	8-bits
SNR Range	5,10,15 and 20 dB
Smoothing Length	11
Step Size for VSS-CMA	0.0001
Step Sizes for VSSCMA-DDLMS	0.0001,0.002
Step Size for DFE-CMA	0.0001
Modulation type	16-QAM
Rolloff factor	1
Filter span in symbols	4

4.3 Simulation Block Diagram

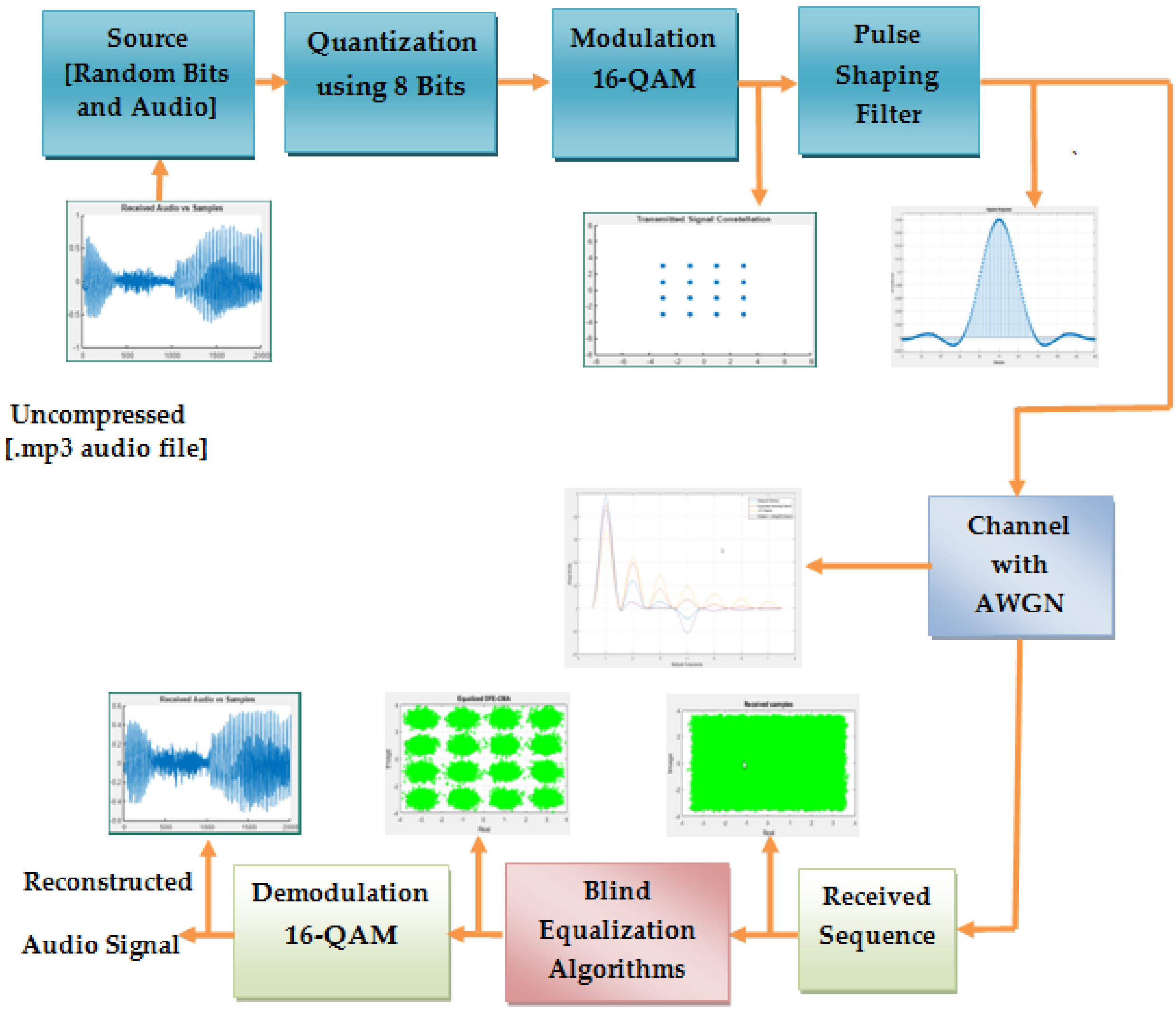


Figure 4.2 Simulation block diagram.

Major blocks of simulation diagram are explained in the following section.

- **Source:** random bits and audio file (.mp3) are used for simulation in the thesis.
- **Quantization:** is the process of converting continuous analog audio signal to a digital signal with discrete numerical values and applied to convert each sample of audio signal source to 8 bit as shown in Figure 4.2.
- **Modulation:** 16-QAM modulation whose constellation shown in Figure 4.1 (c) is used for the simulation.
- **Pulse Shaping Filter:** The raised cosine transmit filter block up samples and filters the input signal using raised cosine FIR filter. Raised cosine FIR filter with roll off factor (beta= 1) and filter span in symbols of 4 is considered in the thesis.

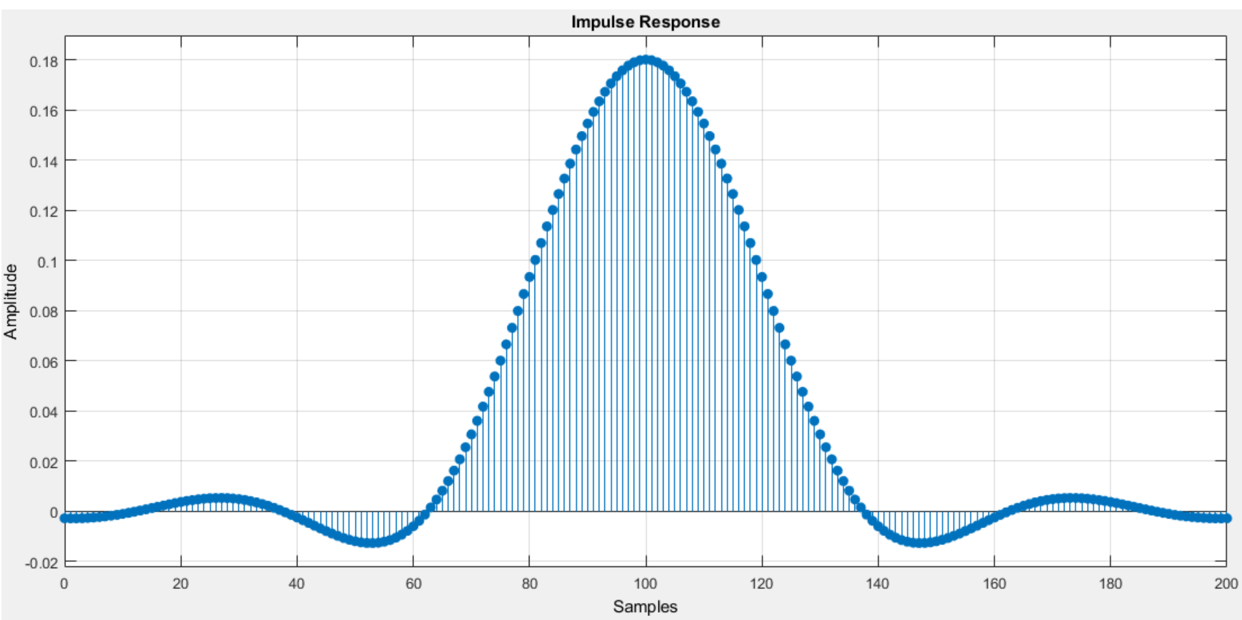


Figure 4.3: Raised cosine pulse for rolloff factor of (beta=1).

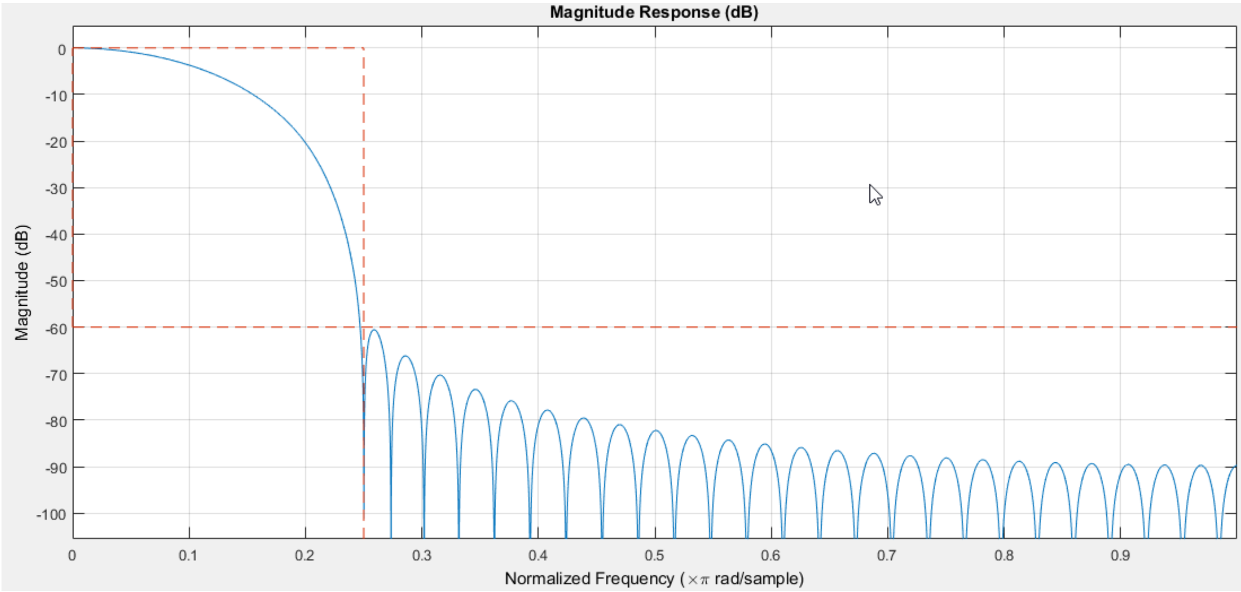
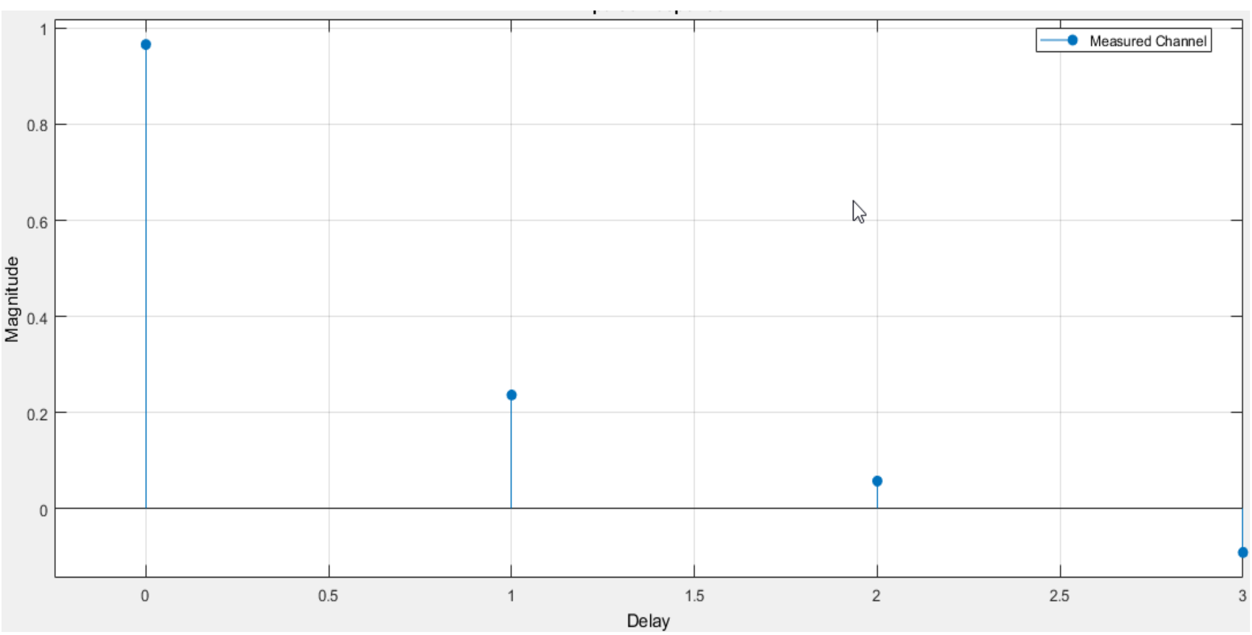
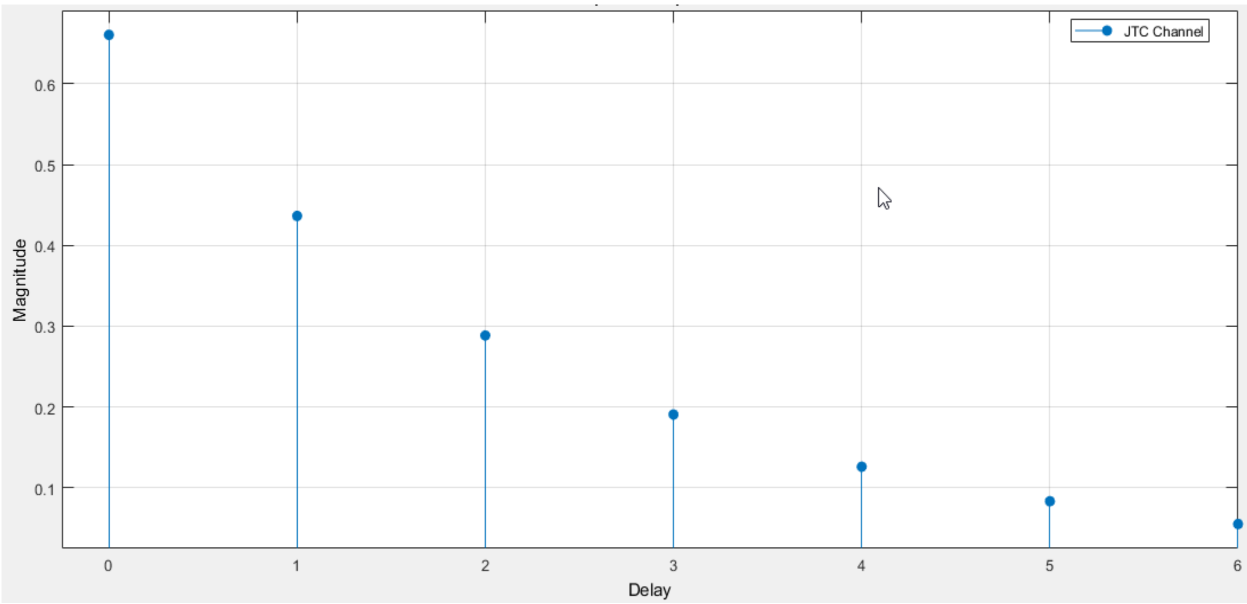


Figure 4.4: Raised cosine pulse frequency response for rolloff factor (beta=1).

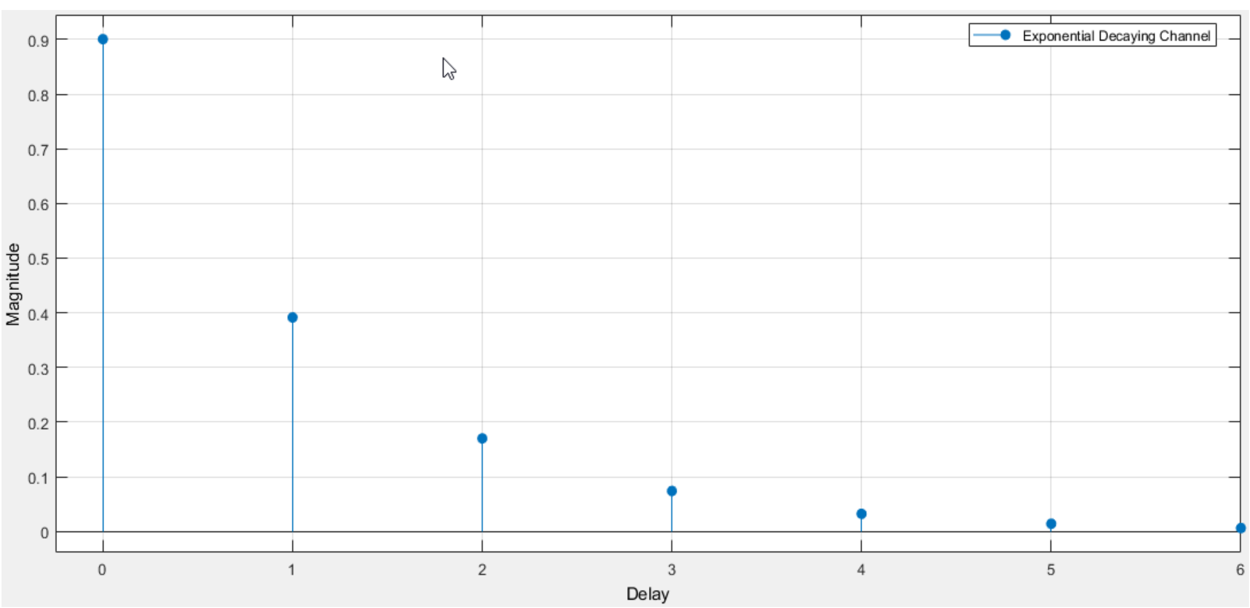
- **Channel: -** ISI channel model with AWGN is used for simulation of random bits as input source. Other channel models: - exponential decaying, Joint Technical Committee (JTC), and measured channels [15] are also used for GUI test simulation. Each channel impulse responses are shown in the Figure 4.5 (a)-(d).



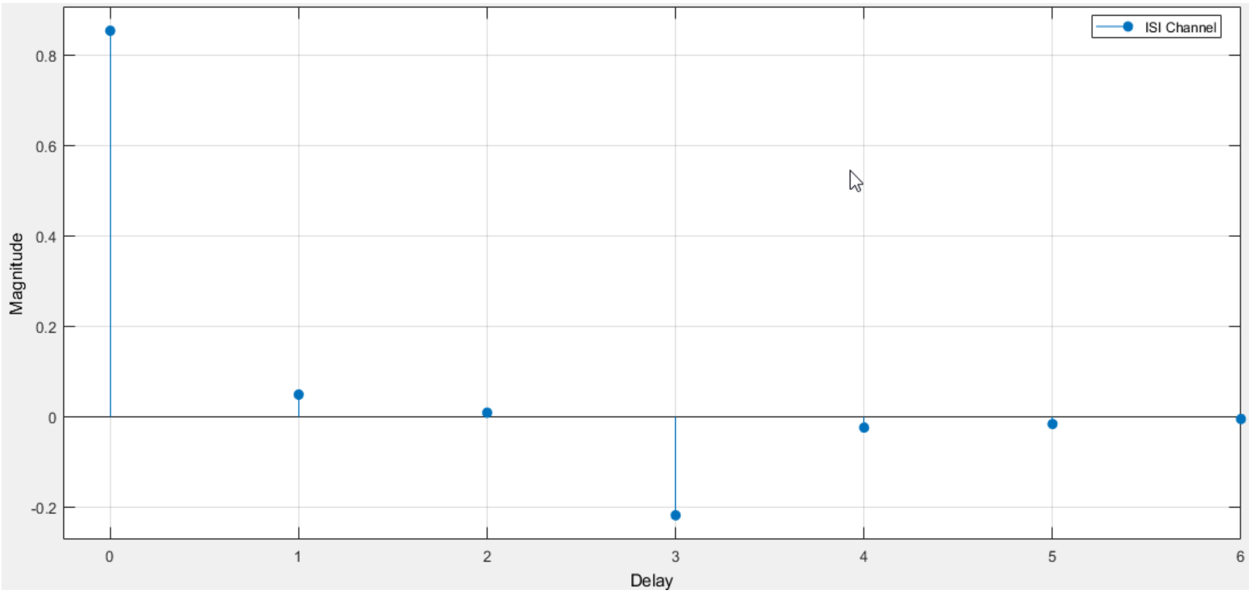
(a) Measured Channel Impulse Response.



(b) JTC Channel Impulse Response.



(c) Exponential Decaying Channel Impulse Response.



(d) ISI Channel Impulse Response

Figure 4.5 (a)-(d) Channels impulse responses [15, 60].

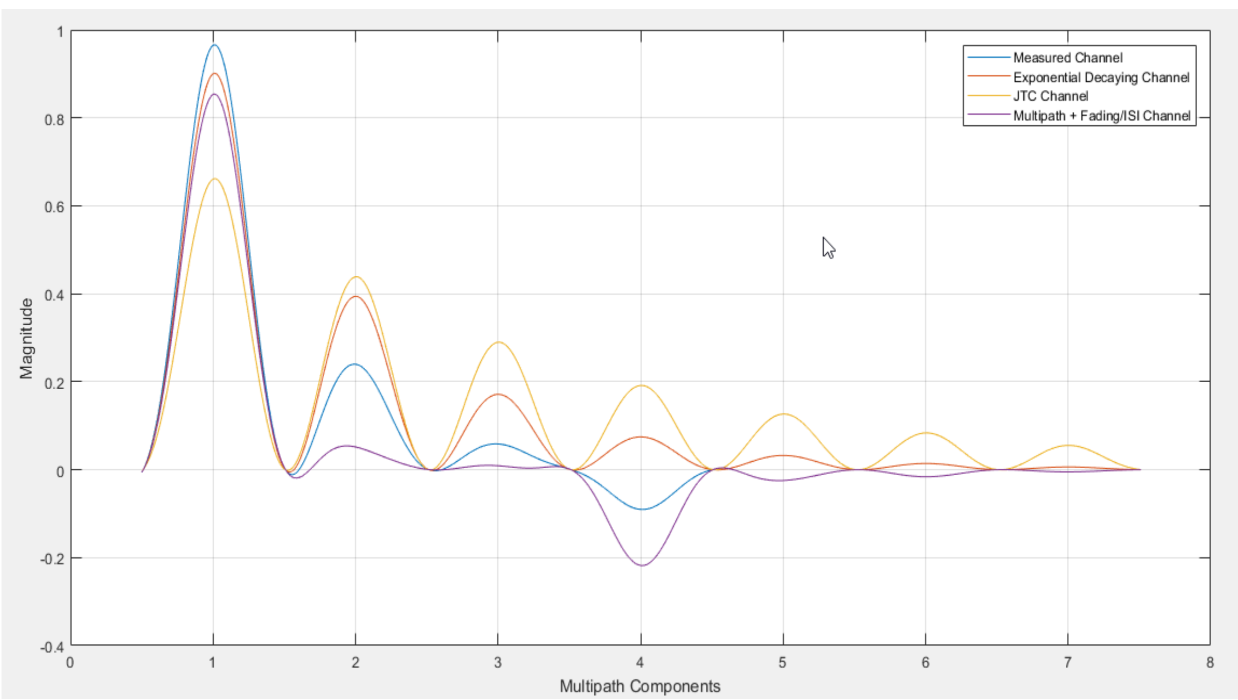


Figure 4.6 Channels impulse response for raised cosine [15, 60].

- **Adaptive Blind Equalization: -**VSS-CMA, DFE-CMA and VSSCMA-DDLMS are used for simulation. CMA and MMA are used for GUI simulation comparisons.

4.4 Simulation Results

Simulation results are discussed in two sections. In this section, simulation of three selected adaptive blind equalization techniques is done using randomly generated bits as an input source and ISI channel as transmission medium with AWGN. In section 4.5, both random bits and audio signal (.mp3 sample file) are used as an input data for GUI simulations. GUI application is applied as an end-to-end performance analysis tool and simulation results of five blind adaptive equalization algorithms including CMA and MMA are demonstrated.

In Figure 4.7 shown below, symbol constellation plots represents simulation of the original transmitted symbols of random bits. And Figure 4.8 shows received symbols (the distorted signal due to ISI channel and AWGN present in the medium) without applying equalization techniques. To extract the equalized outputs from the distorted received symbols, three adaptive blind equalization algorithms are applied and their equalized simulation outputs are discussed in the following sections.

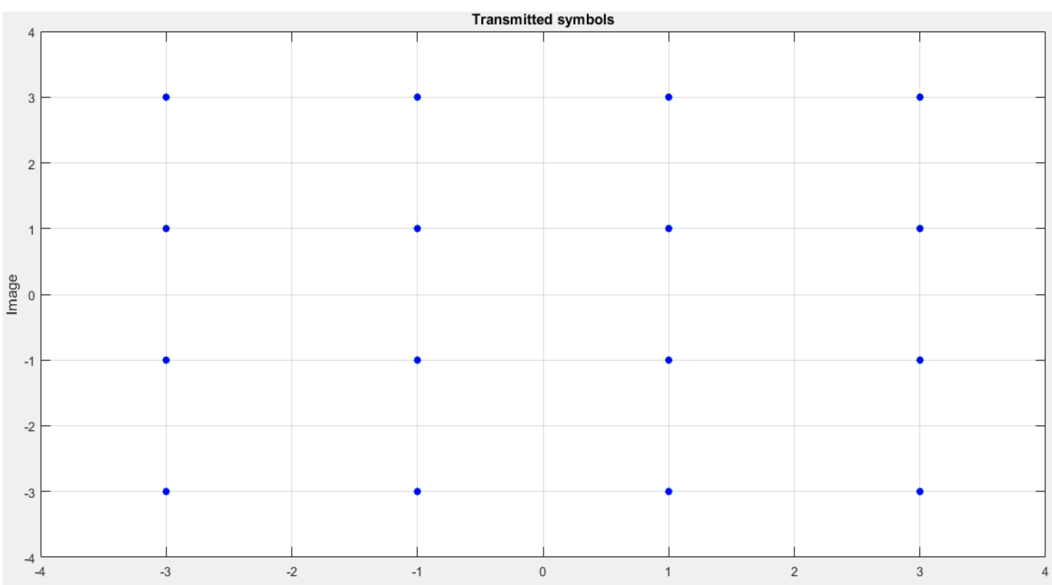
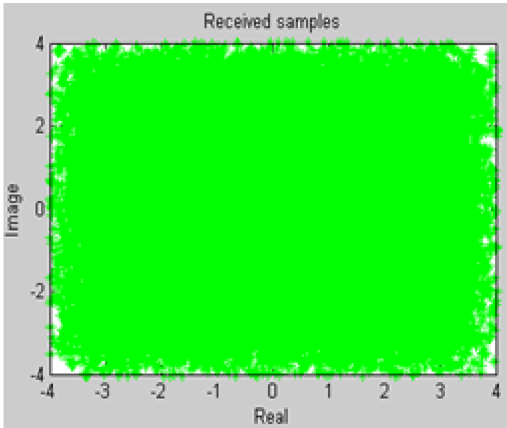
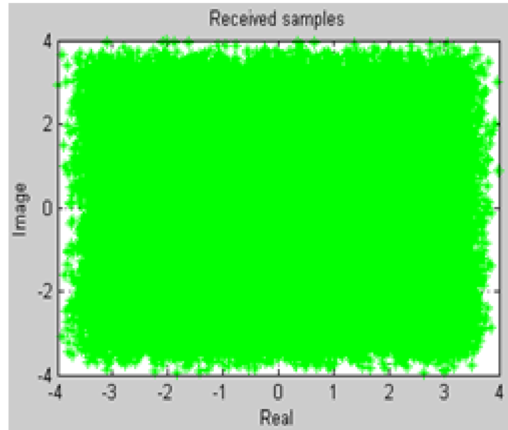


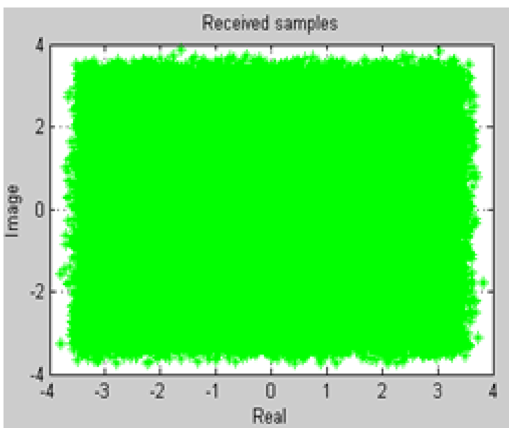
Figure 4.7 Representation of 16 QAM transmitted signal.



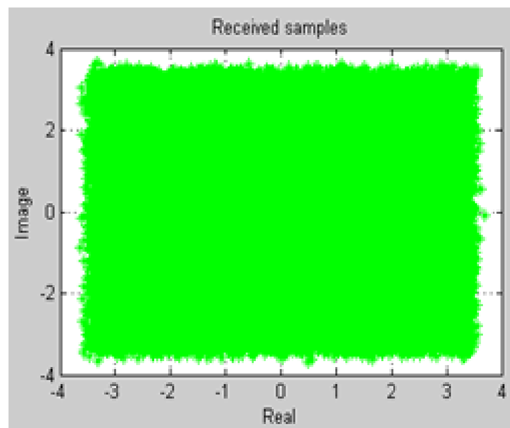
(a) Received Symbols, 5dB SNR



b) Received Symbols, 10dB SNR



c) Received Symbols, 15dB SNR



d) Received Symbols, 20dB SNR

Figure 4.8 Received symbol representations for different SNR values.

Simulation flow charts of the three adaptive blind equalization algorithms are shown in Figures 4.9, 4.11 and 4.13. Mathematical expressions used in flow charts are discussed in Chapter two. Following the flow charts, simulation codes are developed for each of the respective algorithms using MATLAB 2014a. The developed simulation flowcharts of VSS-CMA, DFE-CMA and VSSCMA-DDLMS are shown in Figures 4.9, 4.11 and 4.13 respectively. Outputs of symbol constellation plots are depicted in Figures 4.10, 4.12 and 4.14.

The simulation results are presented in the next sections based on the parameters listed in Table 4.1 and performance metrics discussed in Section 4.1.

VSS-CMA Simulation Results

The simulation is done using input values from Table 4.1 with 10,000 of randomly generated bits as an input source and ISI channel impulse response shown in Figure 4.5 (d).

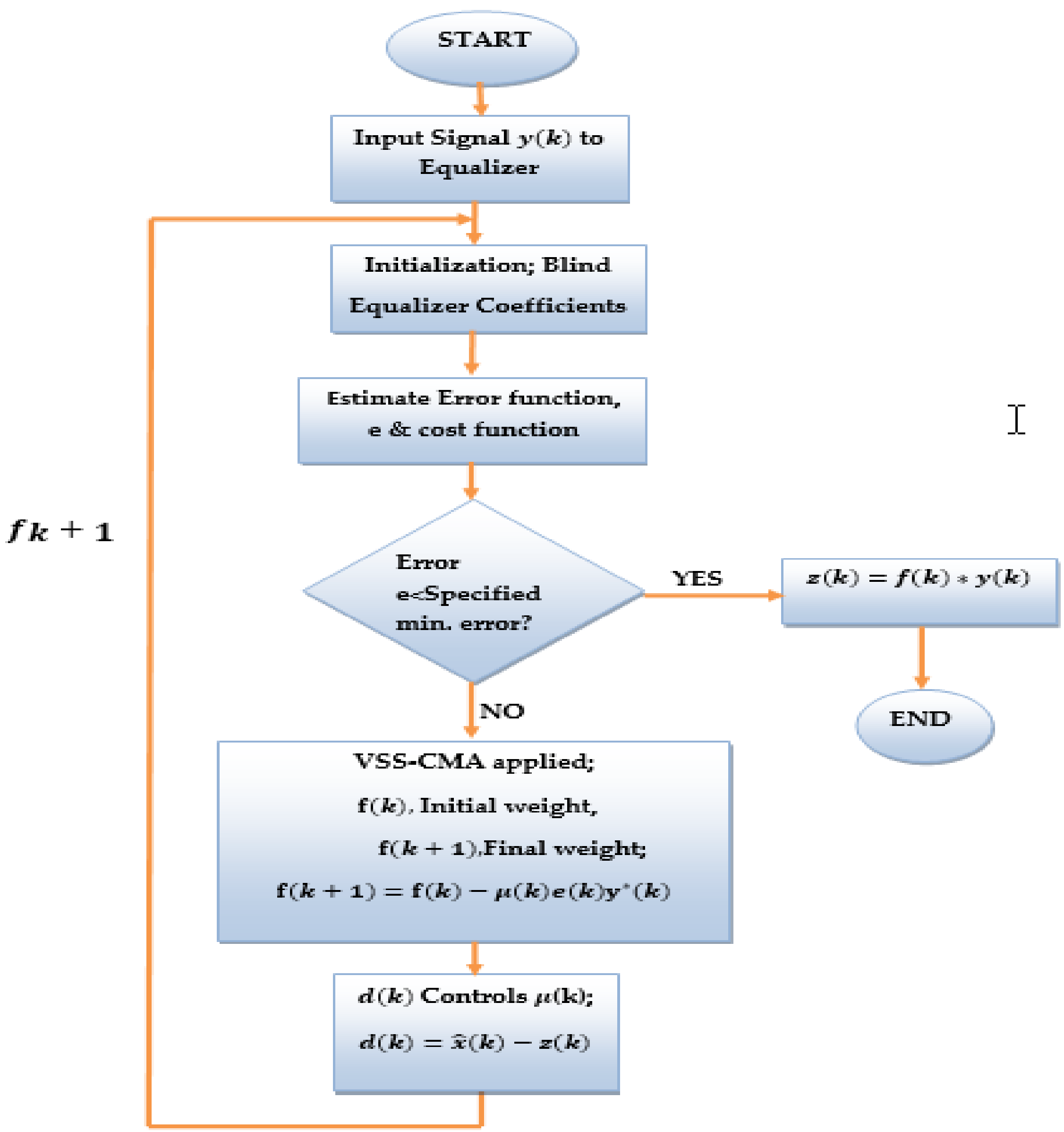
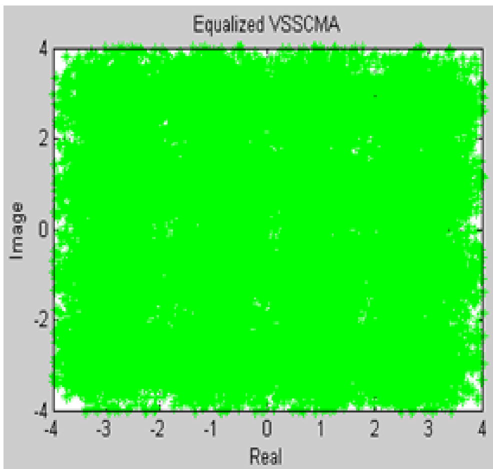
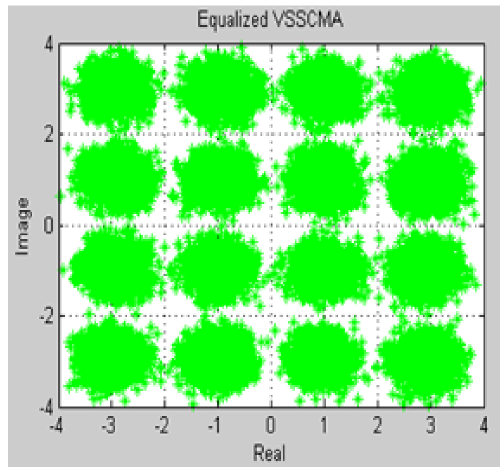


Figure 4.9 Simulation flowchart of VSS-CMA.

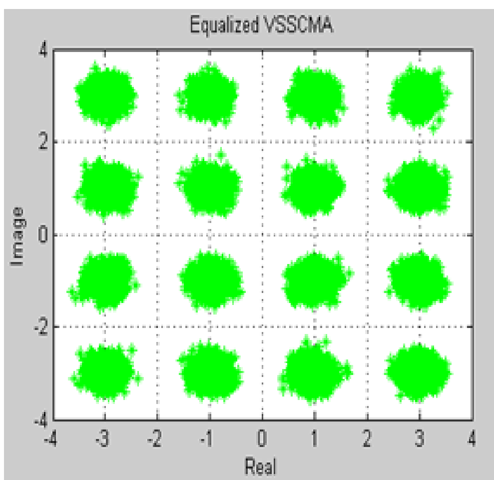
For different values of SNR, the equalized symbol simulation plots are investigated as shown in Figure 4.10.From the results, when SNR value increases the concentration of the equalized symbols to the desired value increases and separation in the symbol constellation plots are not shown clearly for SNR value of 5dB.



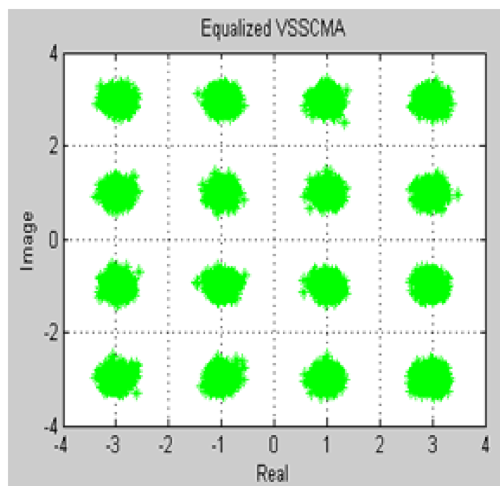
a) Equalized VSS-CMA, SNR 5dB



b) Equalized VSS-CMA, SNR 10dB



c) Equalized VSS-CMA, SNR 15dB



d) Equalized VSS-CMA, SNR 20dB

Figure 4.10 Equalized symbol plots by applying VSS-CMA for different SNR Values.

Measured Values of MSE for equalized symbol constellation plots in Figure 4.10aredepicted in Table 4.2 below.

Algorithm	SNR Values (dB)	MSE Values of Equalized Outputs
VSS-CMA	5	6.854×10^{-3}
	10	3.16×10^{-3}
	15	4.296×10^{-3}
	20	2.3×10^{-4}

Table 4.2 VSS-CMA Equalized symbol constellation plots MSE values

From measured MSE values of VSS-CMA symbols constellation plots in Figure 4.10, the algorithm starts to extract the transmitted signal for MSE value of 3.16×10^{-3} at 10dB SNR using step size of 0.0001.As SNR value increases, the value of MSE decreases and concentration of equalized symbols increases.

DFE-CMA Simulation Results

The simulation is done using same input values and channel type with VSS-CMA.

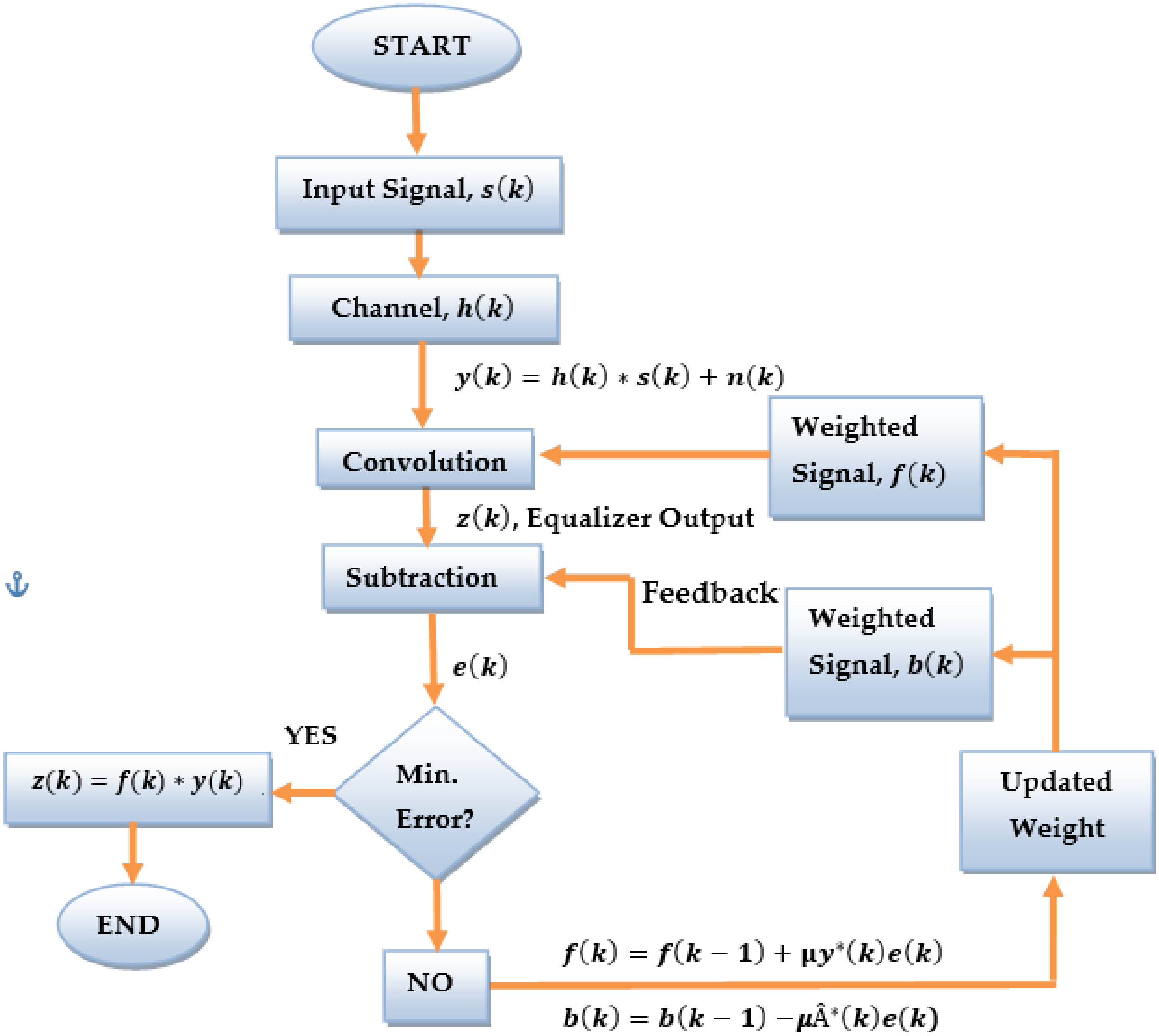
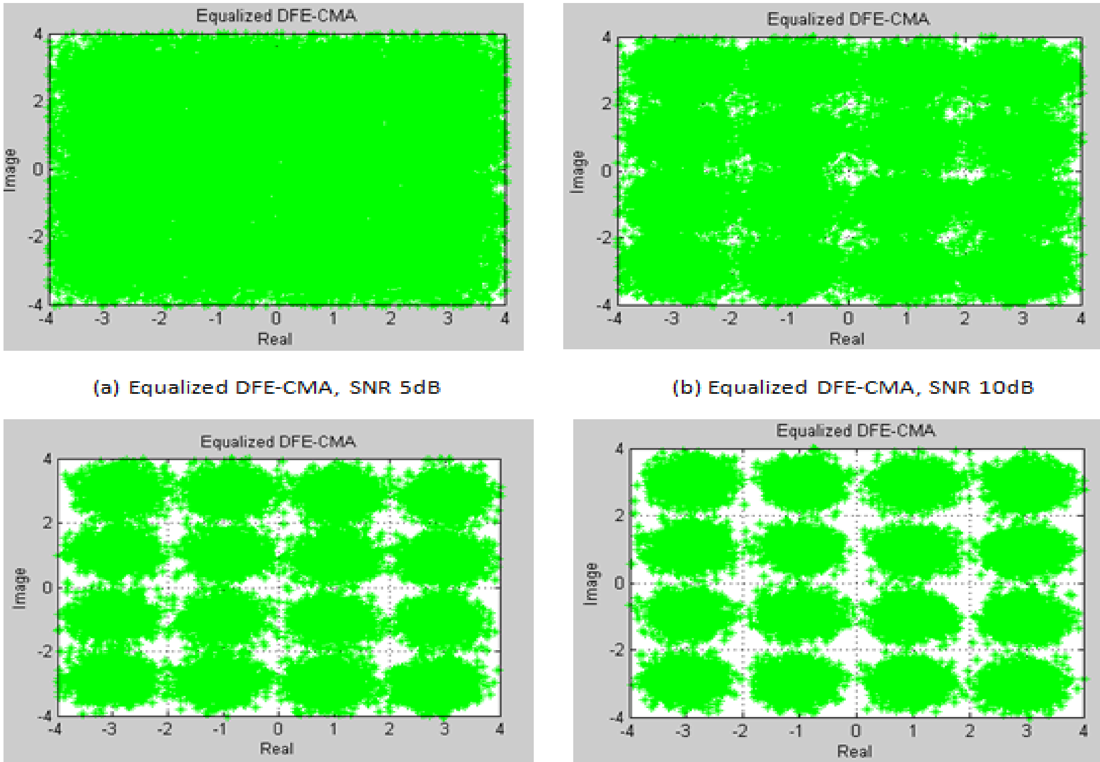


Figure 4.11 DFE-CMA simulation flowchart.

Similarly, from the simulation results in Figure 4.12 and 4.14, concentration of the equalized symbols to the desired value increases as the value of SNR increases after applying DFE-CMA and VSSCMA-DDLMS respectively.



(a) Equalized DFE-CMA, SNR 5dB

(b) Equalized DFE-CMA, SNR 10dB

(c) Equalized DFE-CMA, SNR 15dB

(d) Equalized DFE-CMA, SNR 20dB

Figure 4.12 Equalized symbol plots by applying DFE-CMA for different SNR values.

MSE values for symbol constellation plots of Figure 4.12 is depicted in Table 4.3 below.

Algorithm	**SNR Values (dB)**	**MSE Values of Equalized Outputs**
DFE-CMA	5	3.951×10^{-2}
	10	4.72×10^{-3}
	15	5.542×10^{-3}
	20	3.94×10^{-4}

Table 4.3 DFE-CMA Equalized symbols constellation plots MSE values

From measured values of MSE for DFE-CMA symbols constellation plots in Figure 4.12, the algorithm able to recover the transmitted signal for MSE value of 5.542×10^{-3}at 15dB SNR using 0.0001 step size value. When the value of MSE decreases, the concentration of equalized symbols increases and SNR values also increases.

VSSCMA-DDLMS Simulation Results

Simulation is done using the same input values that are used with VSS-CMA and DFE-CMA algorithms.

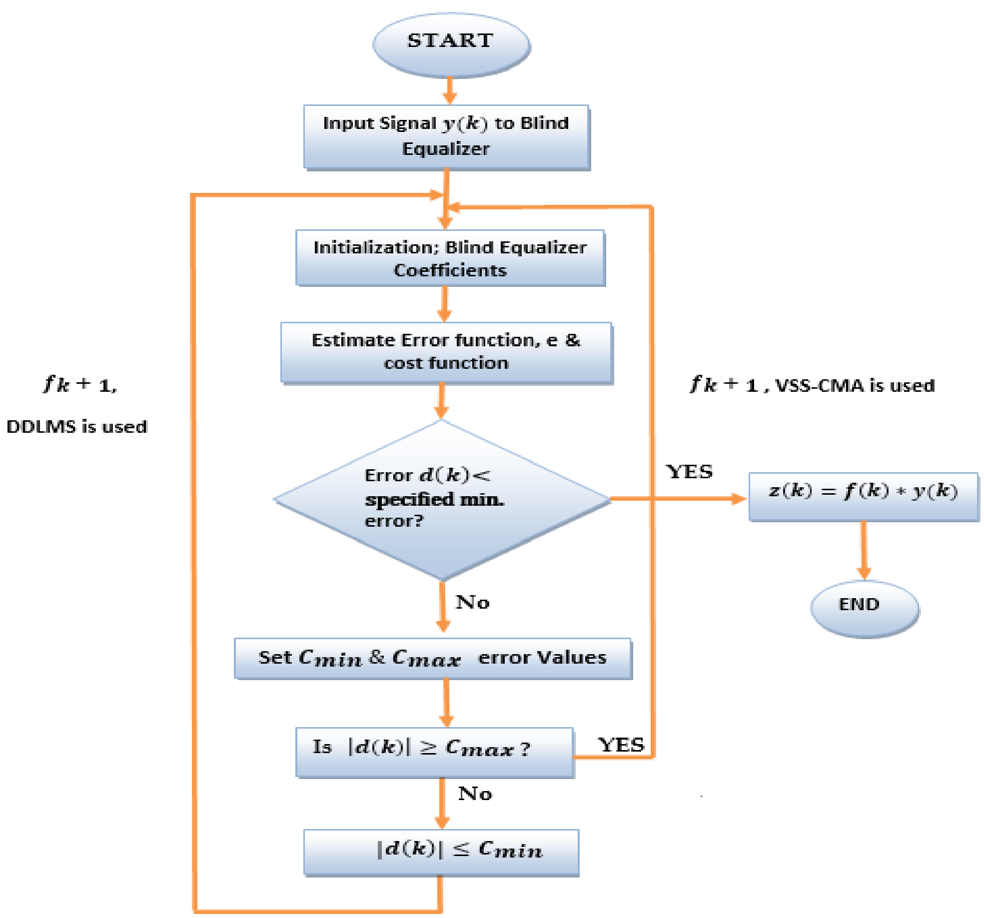
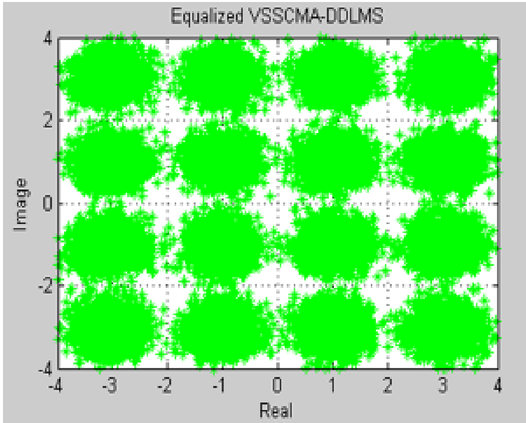
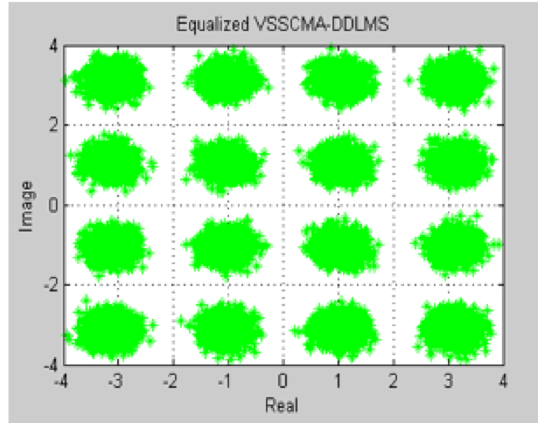


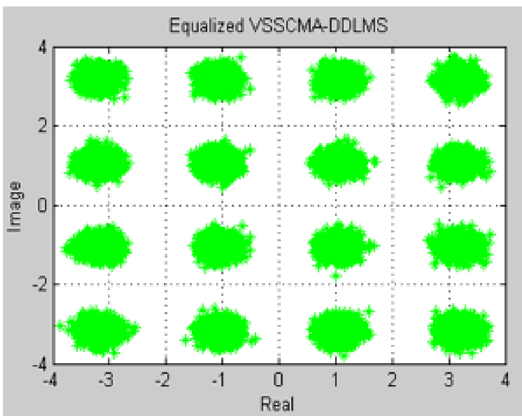
Figure 4.13 VSSCMA-DDLMS simulation flowchart.



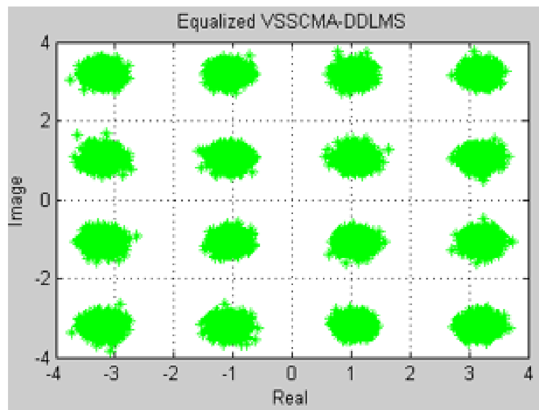
(a) Equalized VSSCMA-DDLMS, 5dB SNR



(b) Equalized VSSCMA-DDLMS, 15dB SNR



(c) Equalized VSSCMA-DDLMS, 20dB SNR



(d) Equalized VSSCMA-DDLMS, 25dB SNR

Figure 4.14 Equalized symbol plots using VSS-CMA-DDLMS for different SNR values

Algorithm	**SNR Values (dB)**	**MSE Values of Equalized Outputs**
VSSCMA-DDLMS	5	4.62×10^{-3}
	10	8.704×10^{-4}
	15	1.616×10^{-4}
	20	7.872×10^{-5}

Table 4.4 VSSCMA-DDLMS Equalized Outputs MSE Values

The result in Figures 4.10, 4.12 and 4.14 shows that; the symbol constellation plot of the DFE-CMA is the least concentrated, VSS-CMA is more concentrated than DFE-CMA and less concentrated constellation points compared to VSS-CMA-DDLMS.

From measured MSE values of symbols constellation plots in Tables 4.2, 4.3 and 4.4, VSSCMA-DDLMS has minimum MSE value of 7.872×10^{-5} at 20dB SNR. Using step sizes of 0.0001 for VSSCMA and 0.0002 for DDLMS, the algorithm extracts the original transmitted signal at 5dB SNR for 4.62×10^{-3} MSE value and the concentration of equalized symbols clearly visible as the SNR value increases.

Among the three algorithms VSSCMA-DDLMS algorithm has better ability for equalizing the original transmitted signal for SNR values from 5dB to 20dB.The three algorithms are also compared at 20dB SNR and randomly generated bits set to 10,000, 20,000 and 30,000 respectively.

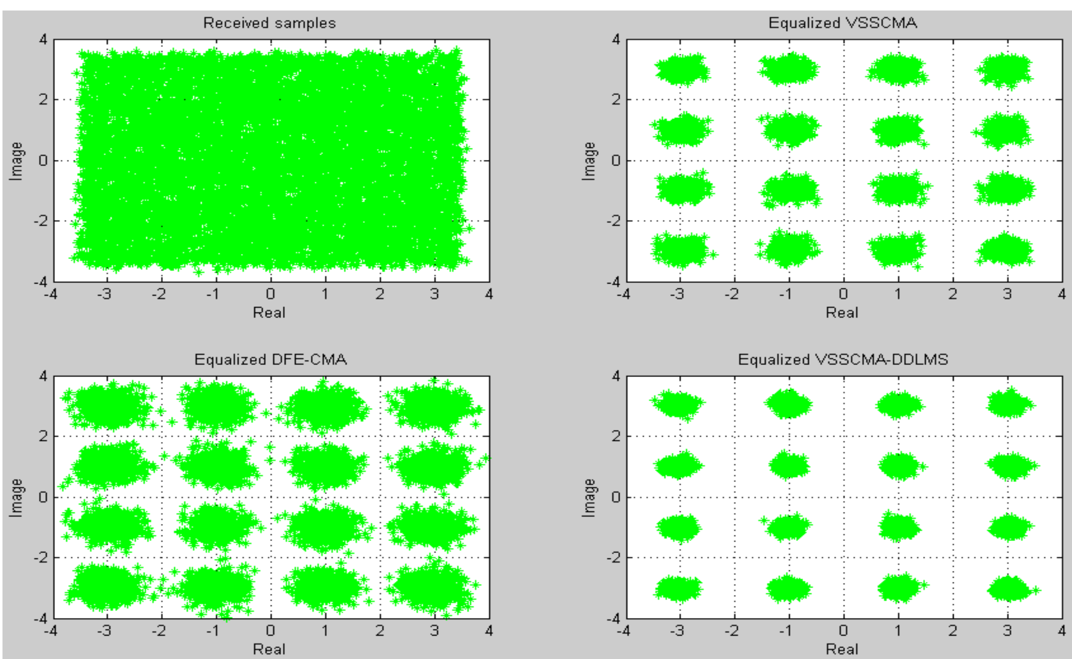


Figure 4.15 Comparison of three algorithms for number of symbols set to 10,000 and SNR=20dB

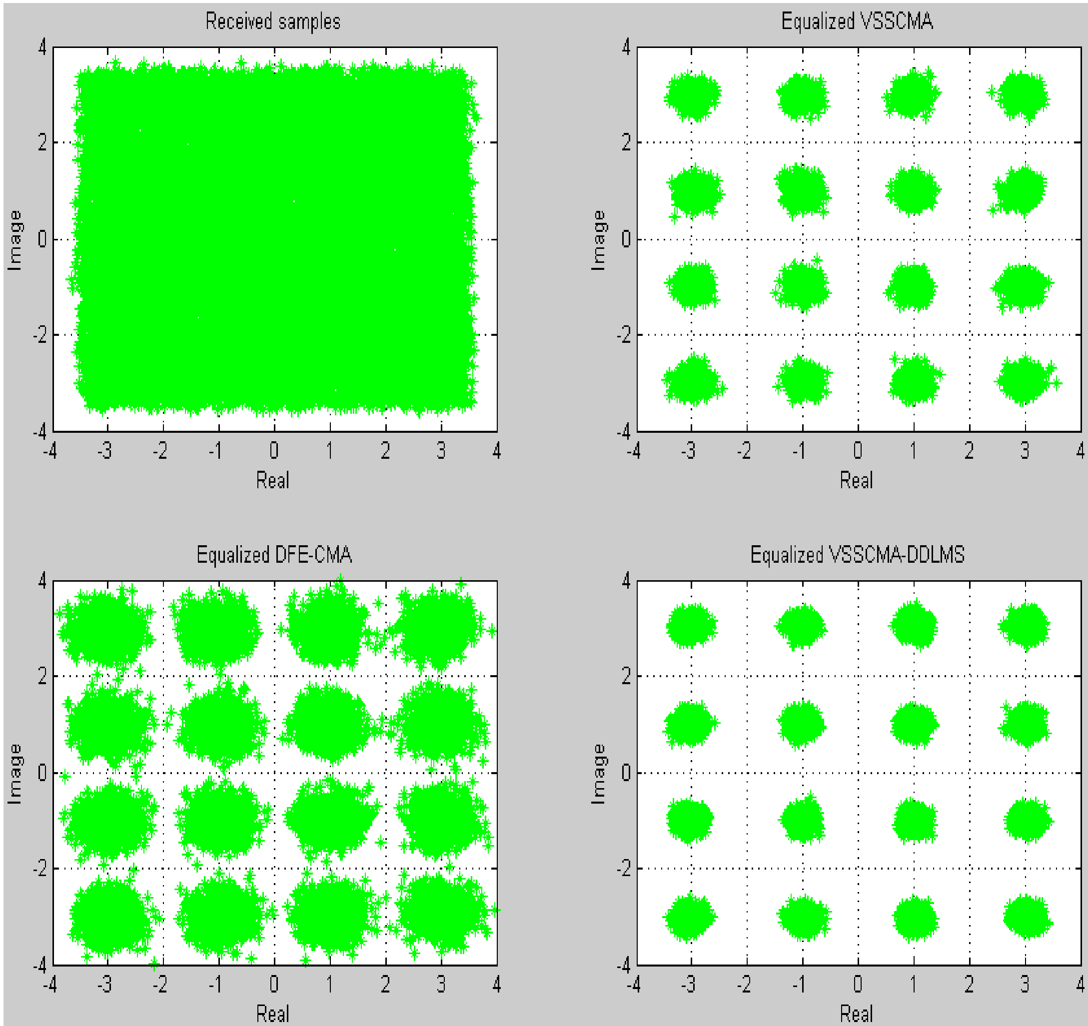


Figure 4.16 Comparison of three algorithms for number of symbols set to 20,000 and SNR=20dB

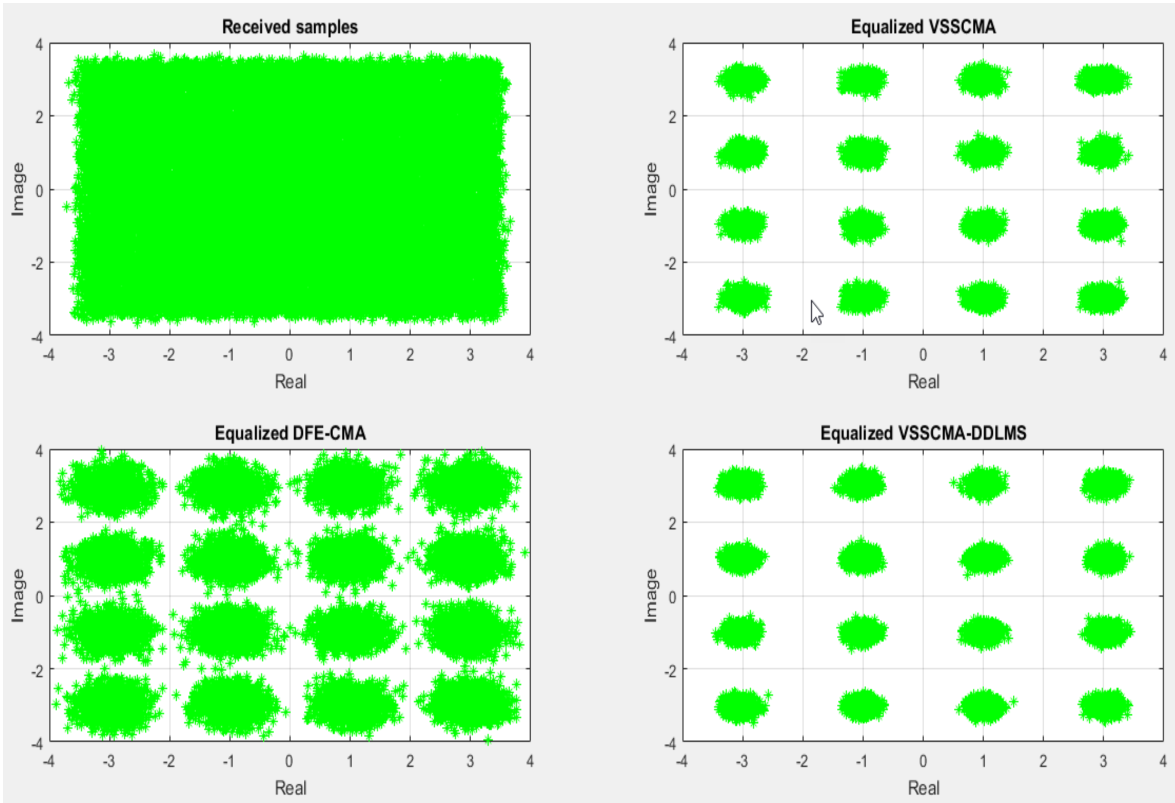


Figure 4.17 Comparison of three algorithms for number of symbols set to 30,000 and SNR=20dB

From the results in Figures 4.15, 4.16 and 4.17; for number of symbols set to 10,000 and 20,000 values VSSCMA-DDLMS is with more concentrated constellation points than both DFE-CMA and VSS-CMA. When the number of symbols increases beyond 30000, the concentration of symbol constellation plots of the VSSCMA-DDLMS and VSS-CMA are nearly similar. In all the above cases, DFE-CMA is the least concentrated compared to the two algorithms.

In general, among the three algorithms, VSSCMA-DDLMS algorithm has better ability for equalizing the original signal.

Considering same parameters used with symbol constellation plot comparisons above, next section leads to simulation results of algorithms using MSE, convergence rate, complexity and computation time as performance metrics.

- **MSE and Convergence Rate**

Figures 4.18, 4.19 and 4.20 shows comparisons of convergence curves for three blind equalization algorithms for different iterations. Algorithms are compared using MSE and convergence rate as performance parameters. As the number of iterations increases beyond 6000, convergence rate of VSS-CMA decreases as illustrated in Figure 4.18.

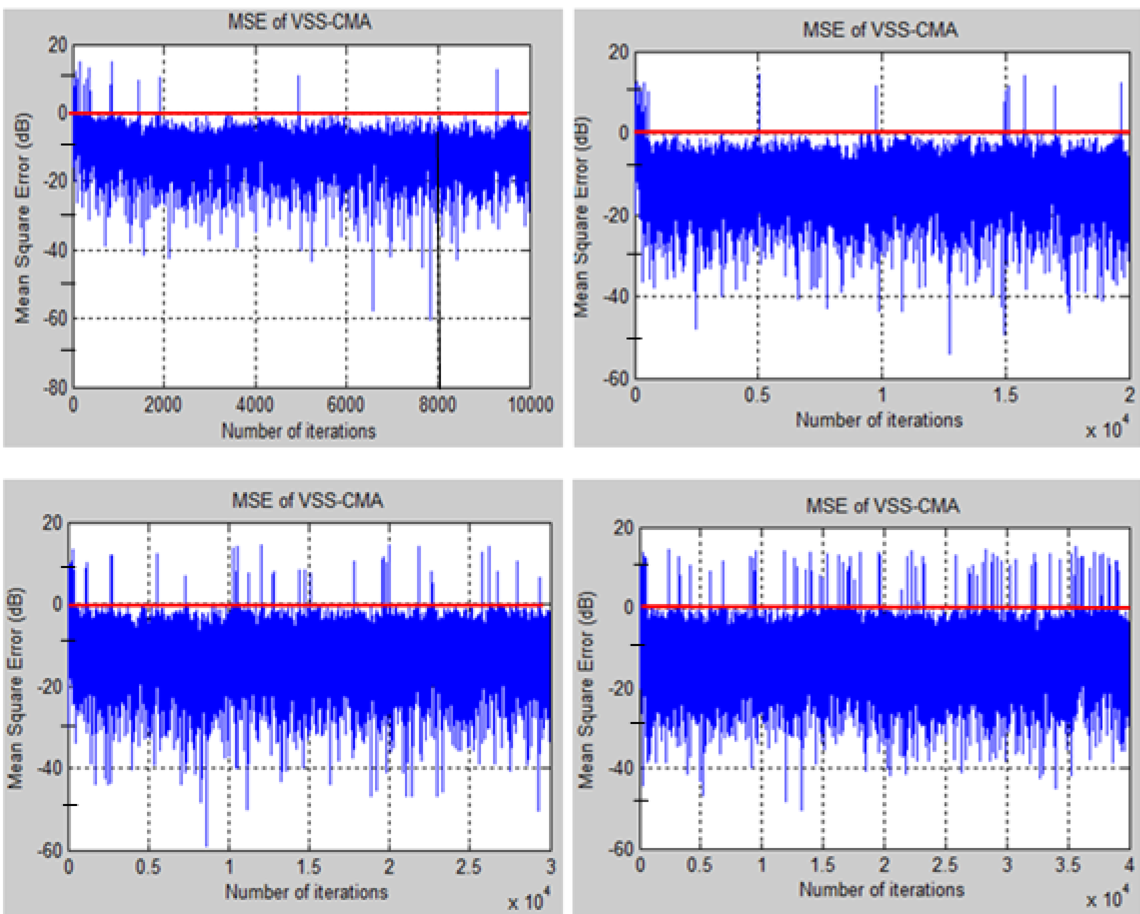


Figure 4.18 MSE versus iteration plots of VSS-CMA.

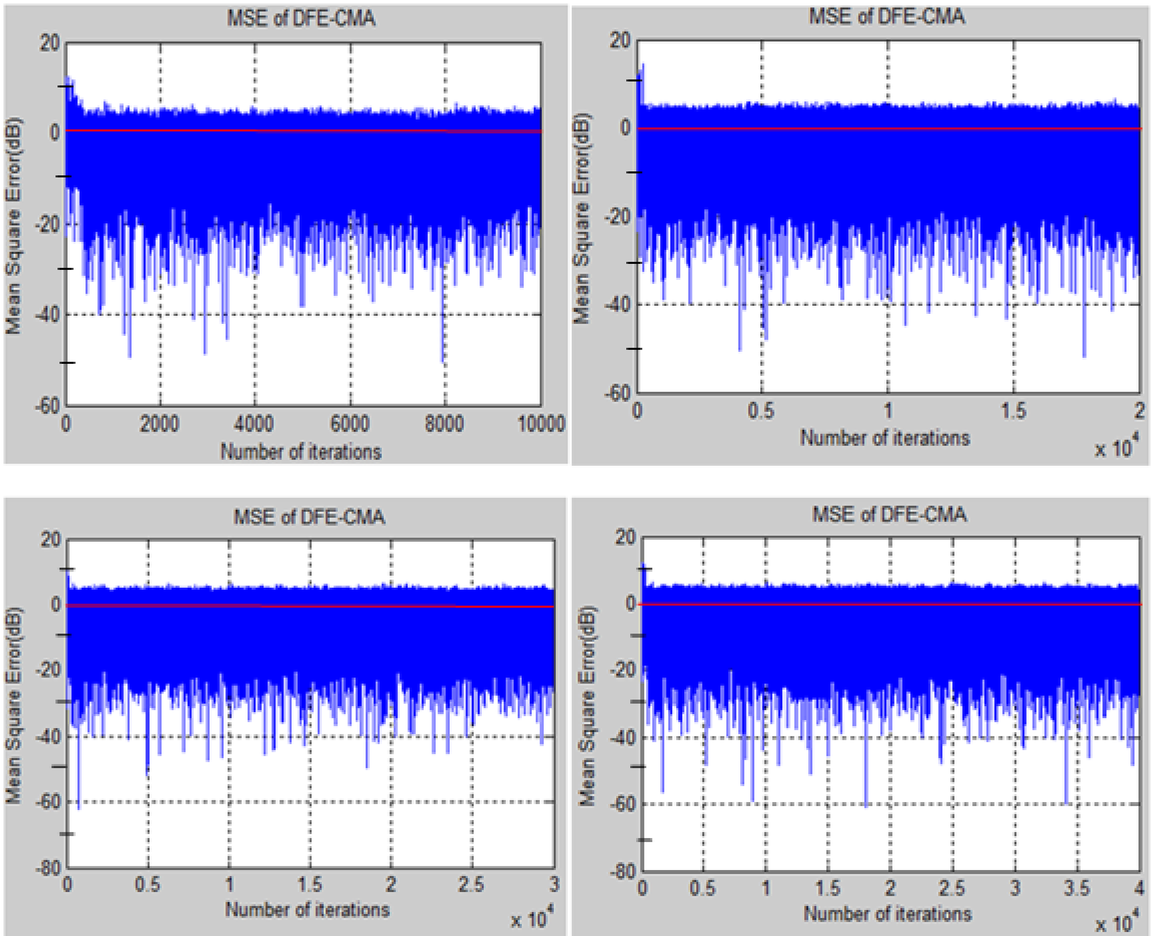


Figure 4.19 MSE versus iteration plots of DFE-CMA.

Figure 4.19 shows that the performance of DFE-CMA is declined as iterations increase beyond 10,000 and from results in Figure 4.20 VSSCMA-DDLMS performs better than the two algorithms having less iteration times of 4000.

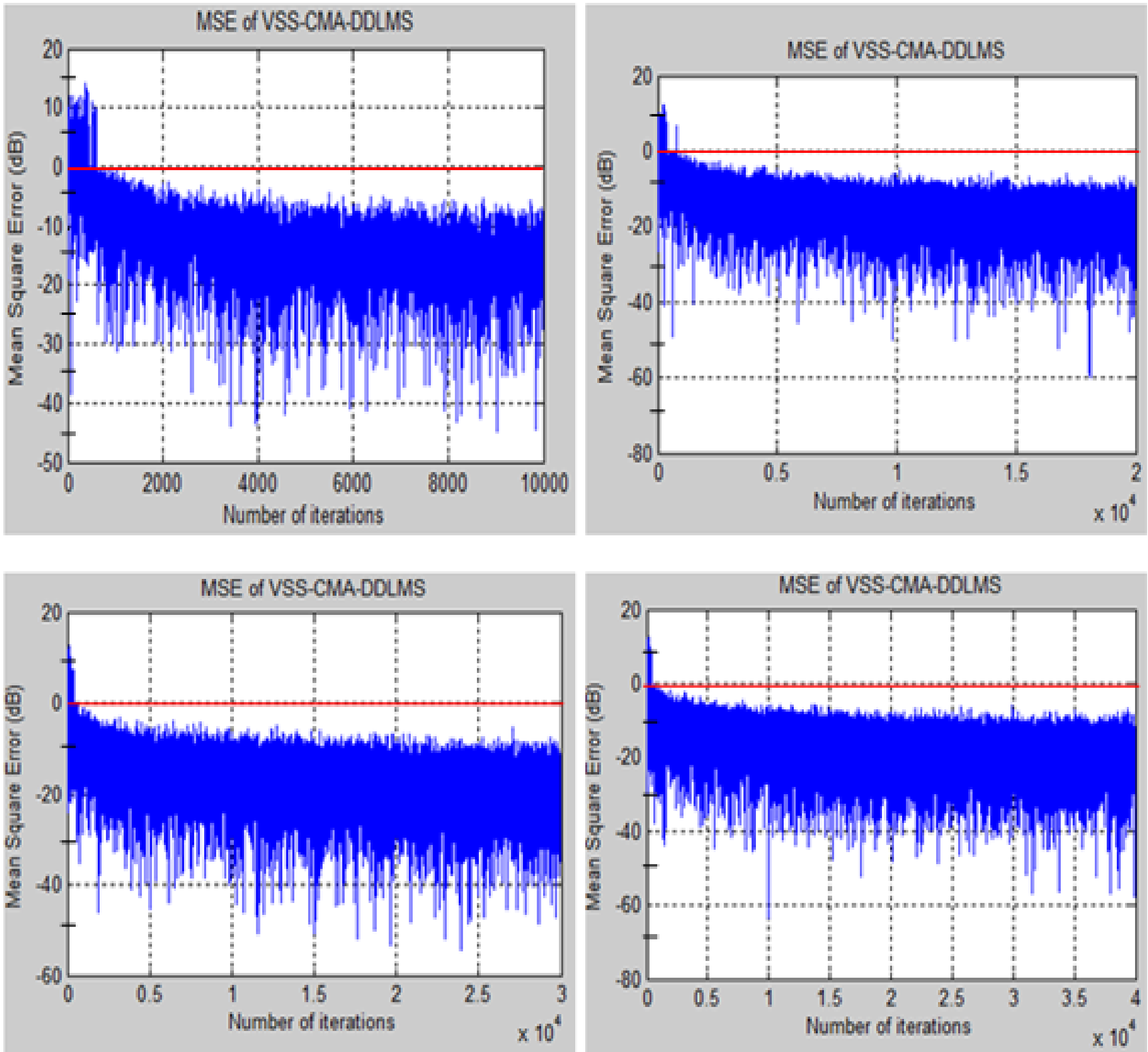


Figure 4.20 MSE versus iterations plots of VSS-CMA-DDLMS.

From the simulation results in Figures 4.18, 4.19 and 4.20; the values of average MSE are given in Table 4.2:

Algorithm	MSE in dB
VSS-CMA	-5
DFE-CMA	5
VSS-CMA-DDLMS	-10

Table 4.5 MSE values in dB.

From Figures 4.18, 4.19 and 4.20; MSEs of the three algorithms decrease with the number of iterations increasing. Taking zero line as a reference for MSE versus iteration plots in Figures 4.18-4.20; the average value of MSE for VSS-CMA is ~-5dB when the algorithm is iterated 10000,20000,30000 and 40000 times respectively. For the same values of iterations MSE of DFE-CMA can reach to ~+5dB and the MSE of VSSCMA-DDLMS algorithm is reduced to ~-10dB. From the above results, VSSCMA-DDLMS algorithm has both a better convergence rate and low level of steady-state error or MSE value than VSS-CMA and DFE-CMA.

- **Computation Time and Complexity**

Computation time of the algorithms also checked by setting number of symbols to 5,000 and SNR to 20dB. The result shows that computation time of VSSCMA-DDLMS is greater than DFE-CMA and VSS-CMA. To check the complexity of an algorithm, computation time can be used as one of the available indicator. From the result obtained, complexity of VSS-CMA is less compared to VSSCMA-DDLMS and DFE-CMA.

Algorithm	Computation Time (seconds)
VSS-CMA	2.4
DFE-CMA	4.2
VSS-CMA-DDLMS	4.85

Table 4.6 Computation time in seconds.

4.5 GUI Simulations

In this section, integration of the block diagram shown in Figure 4.2 with GUI application set up of MATLAB 2017a app designer is performed. This application is used in the thesis as a test tool to simulate the performance analysis of five blind equalization algorithms mentioned in Section 4.4.

GUI application main, design-transmission and performance analysis pages are shown in the following Figure 4.21.

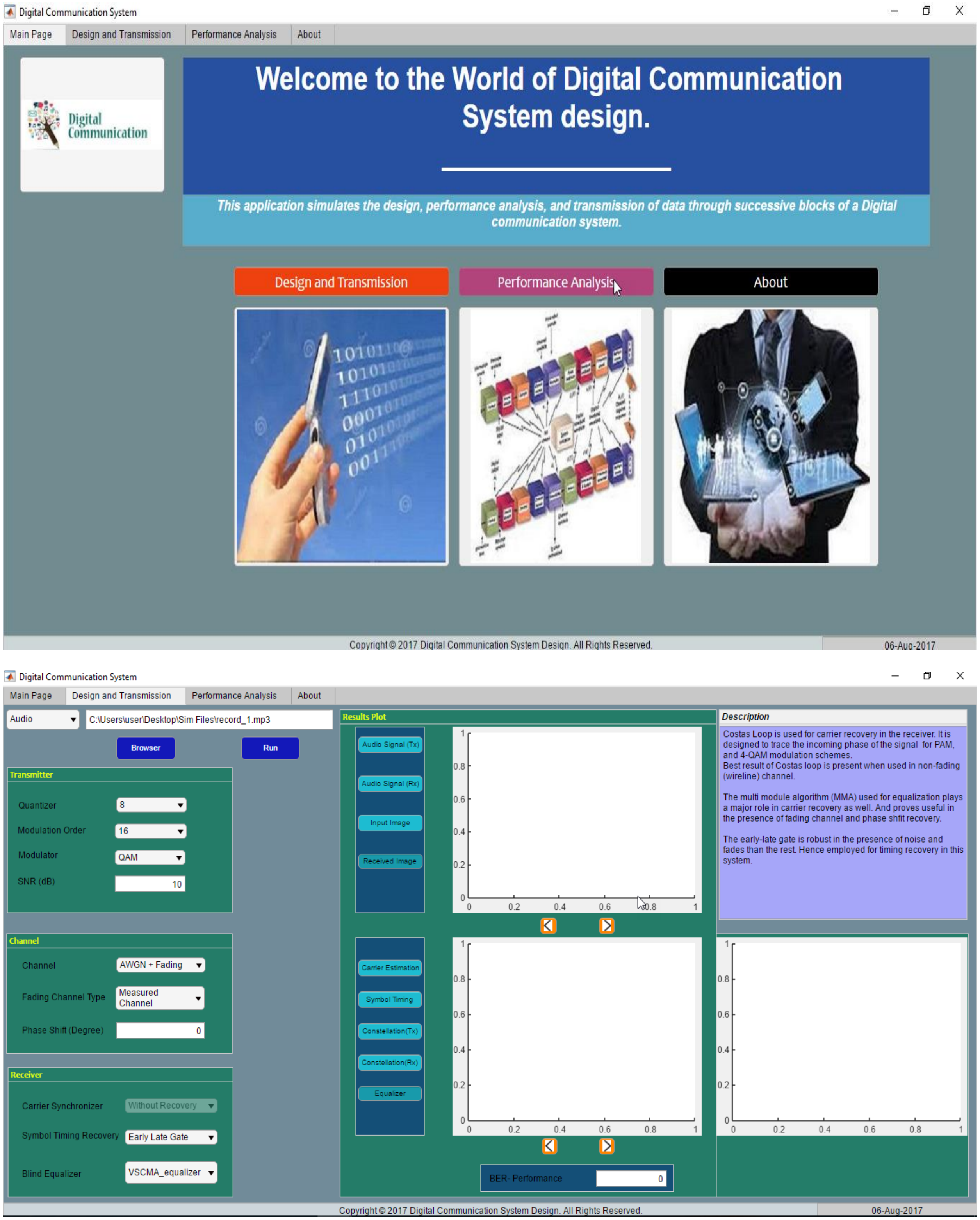


Figure 4.21 GUI main page.

Selected simulation parameters on GUI design-transmission page are: - random bits, quantization value in bits, audio file (.mp3 sample file), SNR values and channel types. And all needed parameters from Table 4.1 are incorporated in GUI blocks using app designer application. Five adaptive blind equalization algorithms:-three selected algorithms from Section 4.4, CMA and MMA from [15] are used for test simulation. As a test simulation, symbol constellation plot is used as performance metric to make comparative study of algorithms and simulation results are shown in the next sections.

Firstly, in Section 4.5.1 below, random bits are selected as an input data on GUI design-transmission page along with needed input parameters and simulation results obtained are shown in Figures 4.22 (for measured channel), 4.23 (for JTC channel), 4.24 (for exponential decaying channel),and 4.25 (for ISI channel) respectively. Simulation results are obtained independently using symbol constellation plots for five algorithms and combined for comparisons. Secondly, in Section 4.5.2, audio signal is used as an input data on GUI design-transmission page and needed parameters are also selected for simulation. Simulation result shows; transmitted audio signal, received noisy signal constellation, equalized signal constellation and reconstructed audio signal in Figures 4.27 (for measured channel) and 4.28 (for JTC channel).

4.5.1 Random Bits as Input Data

Selected simulation parameters: 30,000 random generated bits, 20dB SNR, four channel types, and five blind adaptive equalization algorithms one by one for particular simulation.

A) Measured Channel Outputs

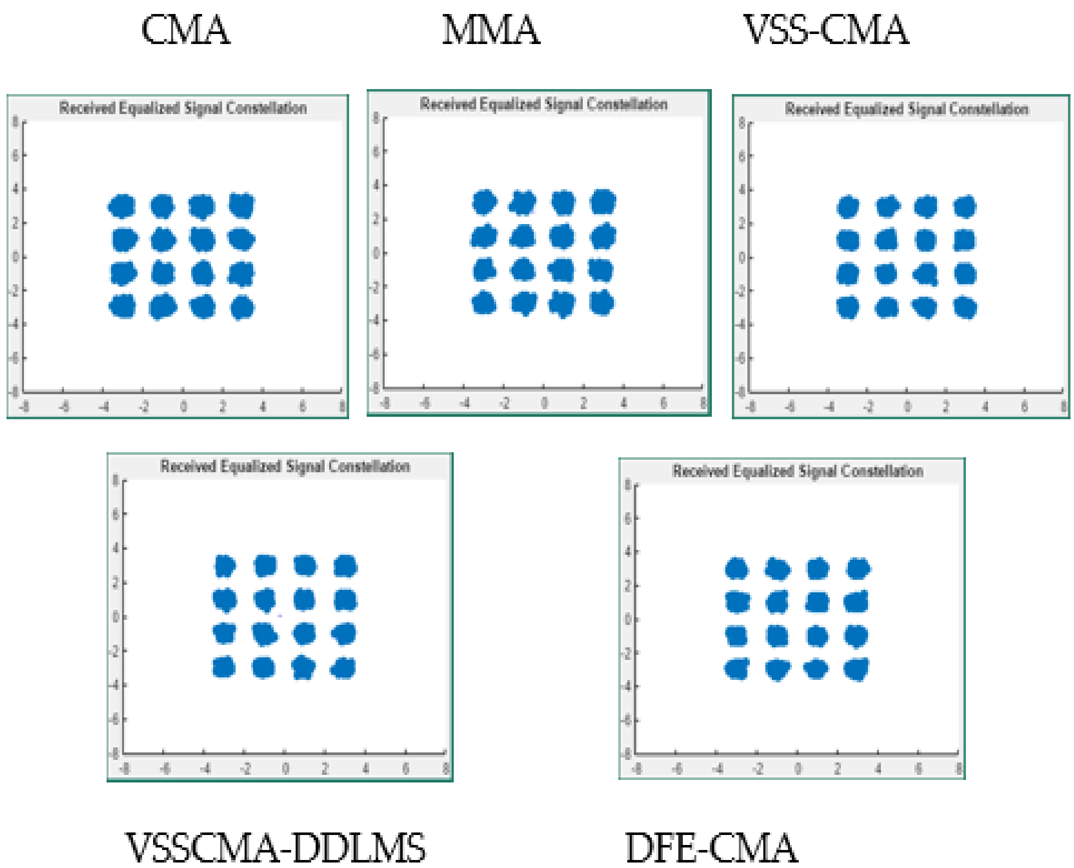


Figure 4.22 Symbol constellation plots in measured channel

From simulation results in Figure 4.22, using measured channel as transmission medium with AWGN, concentration of equalized symbol constellation plots of five algorithms are nearly the same.

B) JTC Channel Outputs

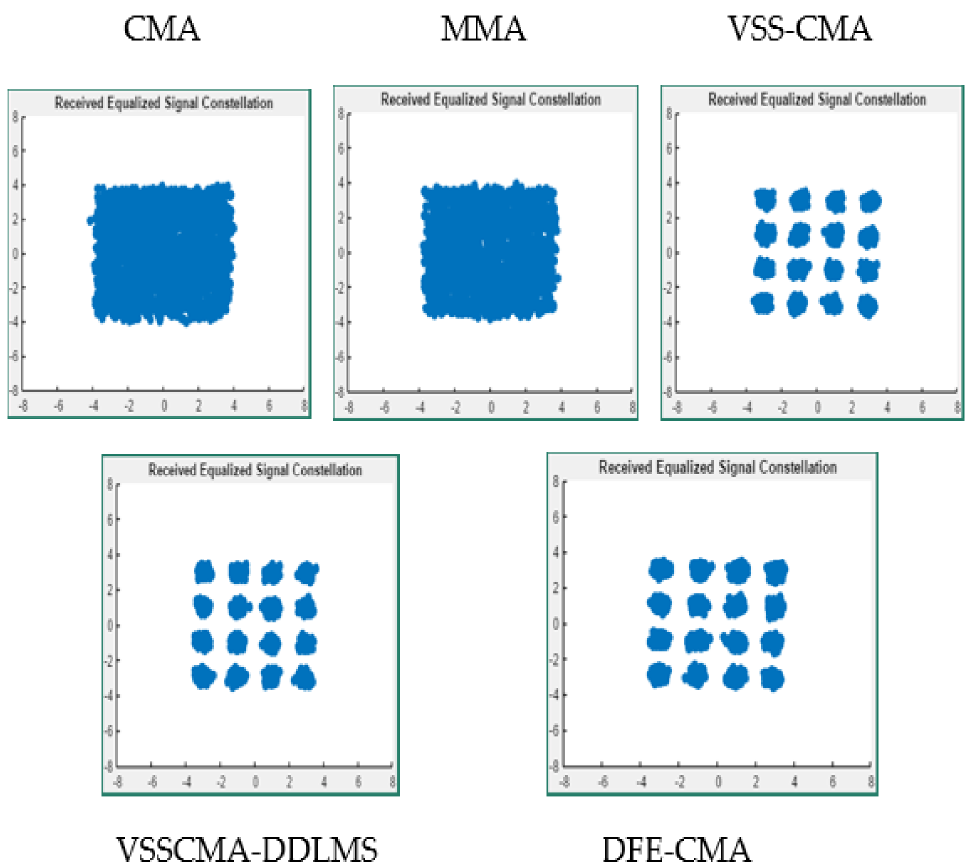


Figure 4.23 Symbol constellation plots using JTC channel

Using JTC channel as a transmission medium, three algorithms: - VSS-CMA, DFE-CMA and VSSCMA-DDLMS shows better equalized symbol constellation plots. But CMA and MMA unable to equalize the transmitted random bits as depicted in Figure 4.23 above.

C) Exponential Decaying Channel Outputs

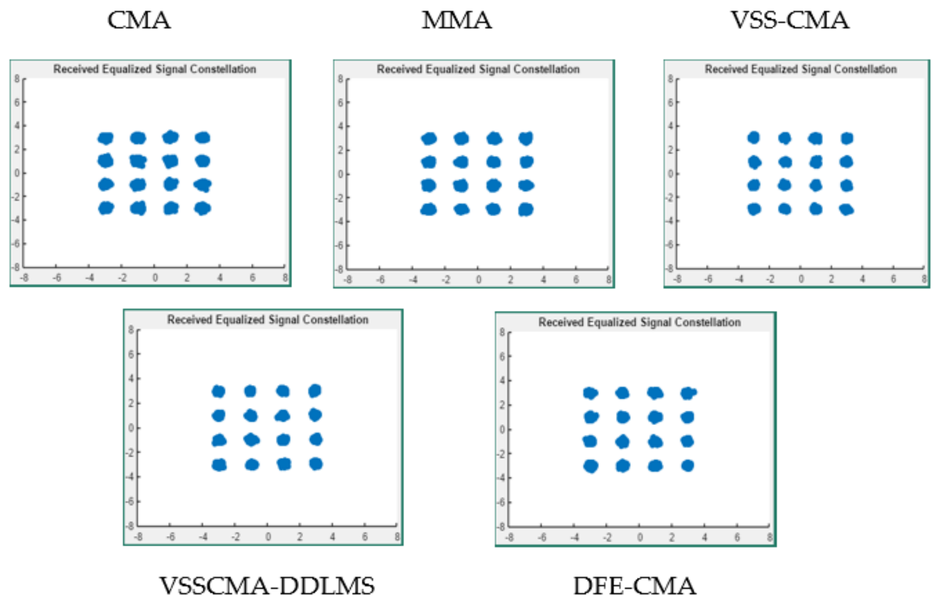


Figure 4.24 Symbol constellation plots in exponential decaying channel

D) ISI Channel Outputs

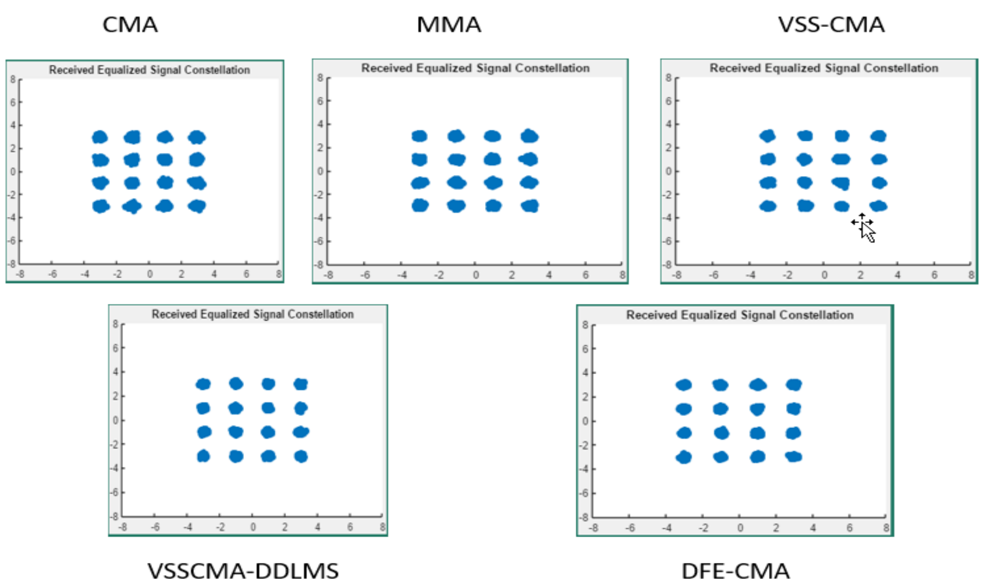


Figure 4.25 Constellation plots in ISI channel

The above results show that, all algorithms extract original transmitted symbols in measured, exponential decaying and ISI channels. But in JTC channel, VSS-CMA, DFE-CMA and VSSCMA-DDLMS algorithms extract the equalized outputs. Equalized outputs are not extracted in CMA and MMA.

4.5.2 Audio as Input Data

Selected simulation parameters: 20dB SNR, audio (.mp3 sample file), raised cosine rolloff factor (beta=1), quantization (8 bits), sampling frequency (48 kHz), filter span in symbols 4 and two channel types are used to avoid redundancy in outputs (JTC and exponential decaying channels). And five blind adaptive equalization algorithms are used for simulation. Symbol constellation plots, transmitted and received audio signal plots are depicted in Figures below.

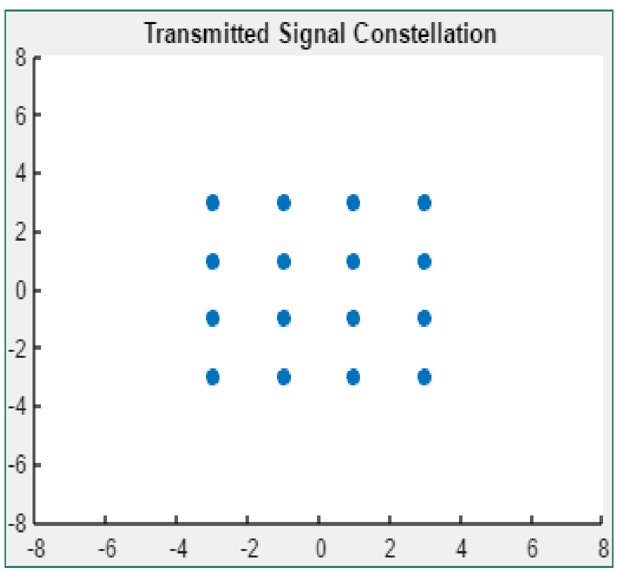
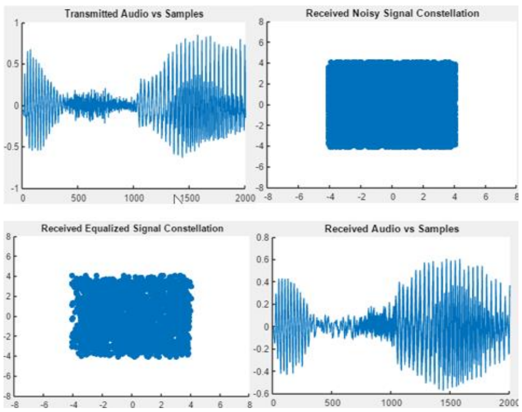


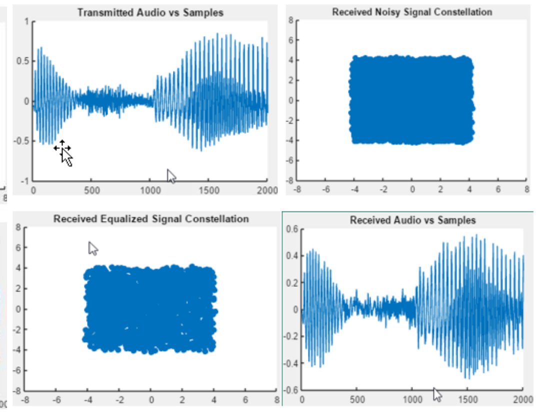
Figure 4.26 Transmitted symbol plots.

A) Measured Channel Outputs

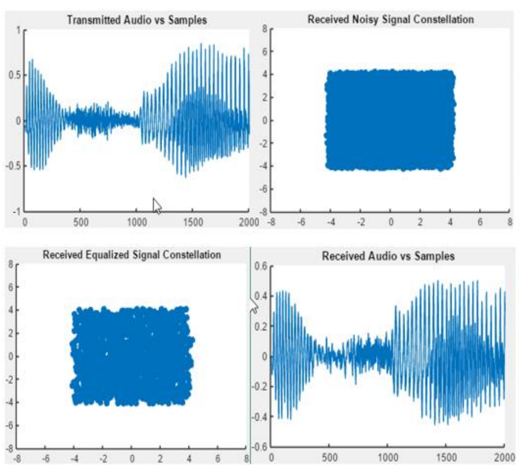
CMA



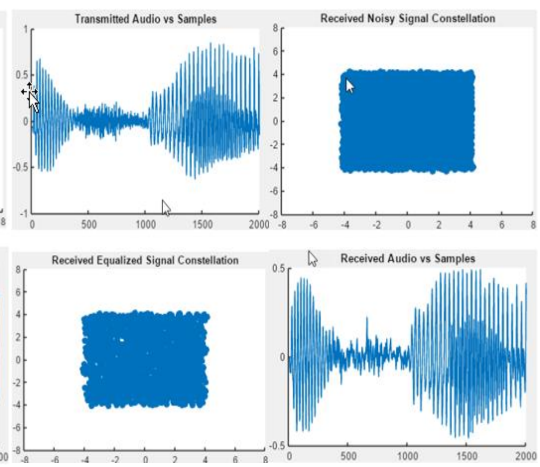
VSS-CMA



VSSCMA-DDLMS



DFE-CMA



MMA

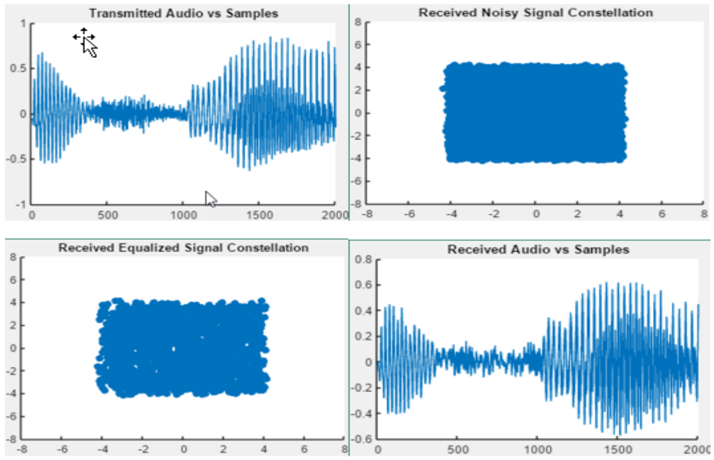
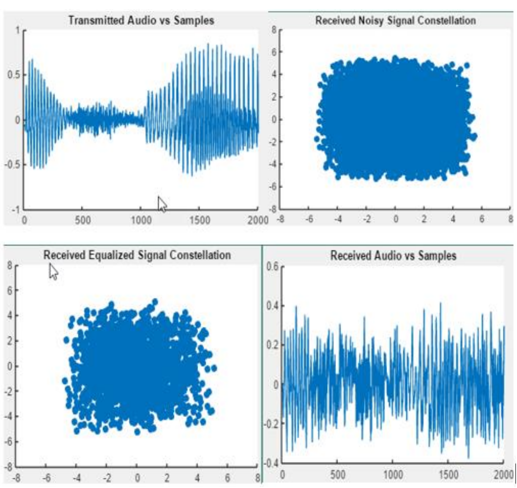
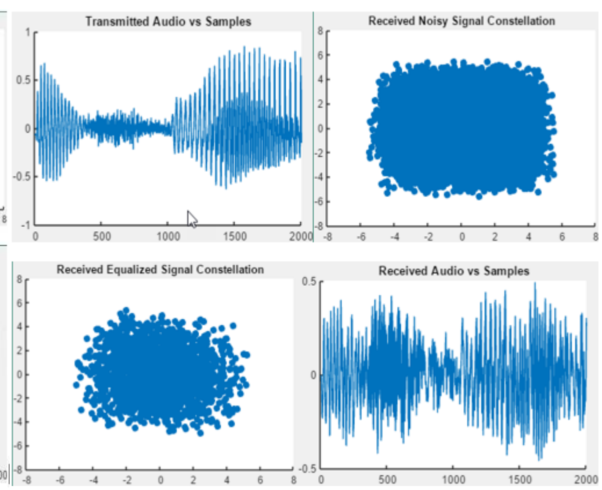


Figure 4.27 (a)-(e) Simulation outputs in measured channel

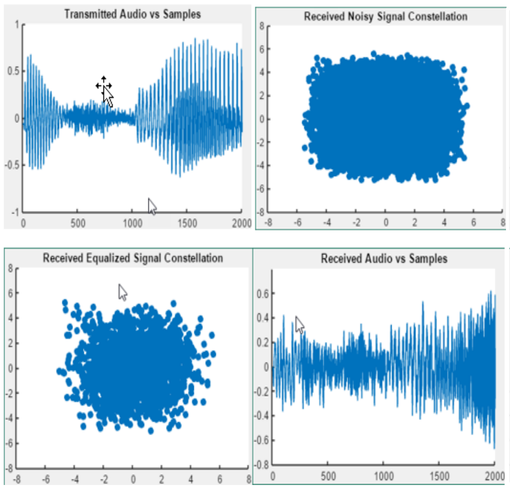
B) JTC Channel Outputs

CMA **VSS-CMA**

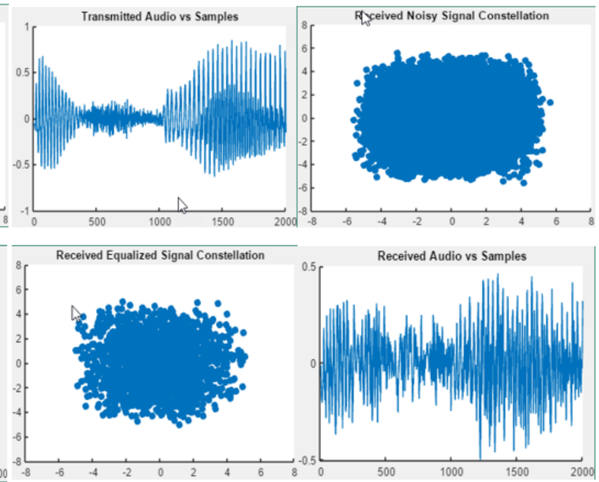


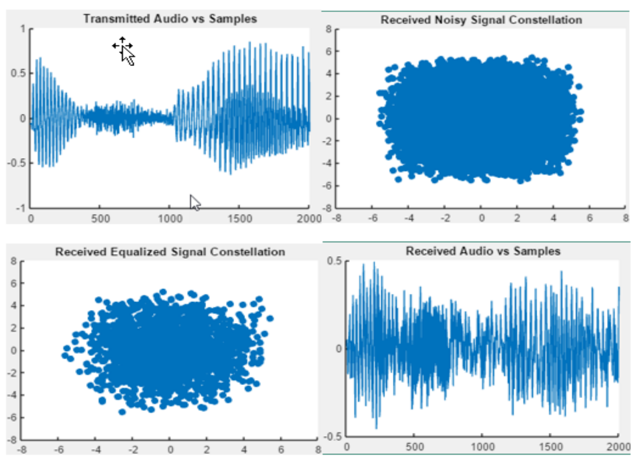


VSSCMA-DD LMS



DFE-CMA





MMA

Figure 4.28 Simulation outputs in JTC channel

In Figure 4.27 simulation outputs indicates all algorithms will have capability of extracting equalized constellation symbol plots and works for audio signals as an input source. And in Figure 4.28 the outputs of JTC channel shows, equalized constellation plot and re-constructed transmitted audio signal are not clearly extracted.

CHAPTER 5

CONCLUSION AND FUTURE WORKS

5.1 Conclusion

In this thesis performance analysis of the three adaptive blind equalization algorithms is done by setting simulation parameters. Using symbols constellation plots as performance metric; original transmitted symbols, received symbols before applying blind algorithms and equalized outputs after applying the three selected blind algorithms have been checked for SNR values from 5 to 20dB in ISI channel with AWGN. MSE versus iteration plots, convergence rate and computation time are also checked for the selected algorithms through simulations. As simulation result indicates VSSCMA-DDLMS has better ability for equalizing the original transmitted signal compared to VSS-CMA and DFE-CMA for using random data bits as input source. It also has both a better convergence rate and low level of MSE value than VSS-CMA and DFE-CMA.

In the second place, using GUI application as an end to end digital communication system and as a test tool for simulation, comparative performance analysis of five blind equalization algorithms is done. For random bits used as an input source through four channel models, the results indicates all of the algorithms extract original transmitted symbols in measured, exponential decaying and ISI channels. But in JTC channel; VSS-CMA, DFE-CMA and VSSCMA-DDLMS algorithms extract the equalized outputs. Equalized outputs are not extracted in CMA and MMA.When audio signal is used as an input data, algorithms perform better in measured channel than JTC channel model.

From simulation results obtained, the general performance of VSSCMA-DDLMS is good compared with DFE-CMA and VSS-CMA. Thus, VSSCMA-DDLMS can be applied practically in audio broadcasting system as one of the available blind mitigation technique with better overall performance.

5.2 Future Works

The results presented in simulation indicates algorithms perform in different way based on the parameters selected for analysis, input data type and channel models used. In this thesis VSSCMA-DDLMS algorithm has its own positive characteristic compared with DFE-CMA and VSS-CMA. Performing comparative study of VSSCMA-DDLMS with semi-blind channel algorithms considering various simulation parameters, application areas and channel models can be one of an open area for research. Deeply studying adaptive blind techniques in context of OFDM to enhance its performance will also be another research area.

Making GUI as an all-inclusive end-to-end performance analysis simulation tool by upgrading app designer section to incorporate additional simulation parameters, semi-blind, training based equalization techniques, higher order modulation schemes, and different channel models can also be another angle of extending the research.

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APPENDICES

Appendix A: MATLAB GUI

GUIs (also known as graphical user interfaces or UIs) provide point-and-click control of software applications, eliminating the need to learn a language or type commands in order to run the application [61].

MATLAB® apps are self-contained MATLAB programs with GUI front ends that automate a task or calculation. The GUI typically contains controls such as menus, toolbars, buttons, and sliders. Many MATLAB products, such as Curve Fitting Toolbox™, Signal Processing Toolbox™, and Control System Toolbox™ include apps with custom user interfaces. We can also create our own custom apps, including their corresponding UIs, for others to use [61].

For simulating an end-to-end digital communication system, customised GUI has been developed using MATLAB app designer. This customised GUI is used for the simulation of random data bits and audio files trasnmitted through the digital communication blocks illustrated in the thesis.Furthermore, it enables any user to design and test parameters belonging to specific blocks in the system.

Appendix B: MATLAB App Designer

App Designer is an environment for building MATLAB® apps. It simplifies the process of laying out the visual components of a user interface. It includes a full set of standard user interface components, as well as a set of gauges, knobs, switches, and lamps to create control panels and human-machine interfaces. Most 2-D plots are also supported. Use app designer for apps that do not require graphics beyond 2-D plots and images. App Designer generates code that is structured to facilitate app development and data sharing across the app [62].

App Designer integrates the two primary tasks of app building – laying out the visual components and programming app behavior. You can quickly move between visual design in the canvas and code development in an integrated version of the MATLAB Editor. The embedded editor allows you to add new properties, callbacks, and other functions with a single click.

App Designer generates object-oriented code. This format makes it easy to share data between parts of the app. The compact structure of the code makes it easier to understand and maintain. Apps are stored as a single file containing both layout and code. You can share apps using this single file, or you can package them with supporting code and data and install them in the App Gallery [62].

App Designer Features

❖ **Interactive Design Environment**

- Drag and drop visual components from the Component Library to the design canvas (1).
- Use alignment hints to get a precise layout of user interface components (2).
- Specify common component properties through specialized property sheets (3).

- Set axes properties in the design environment (4).

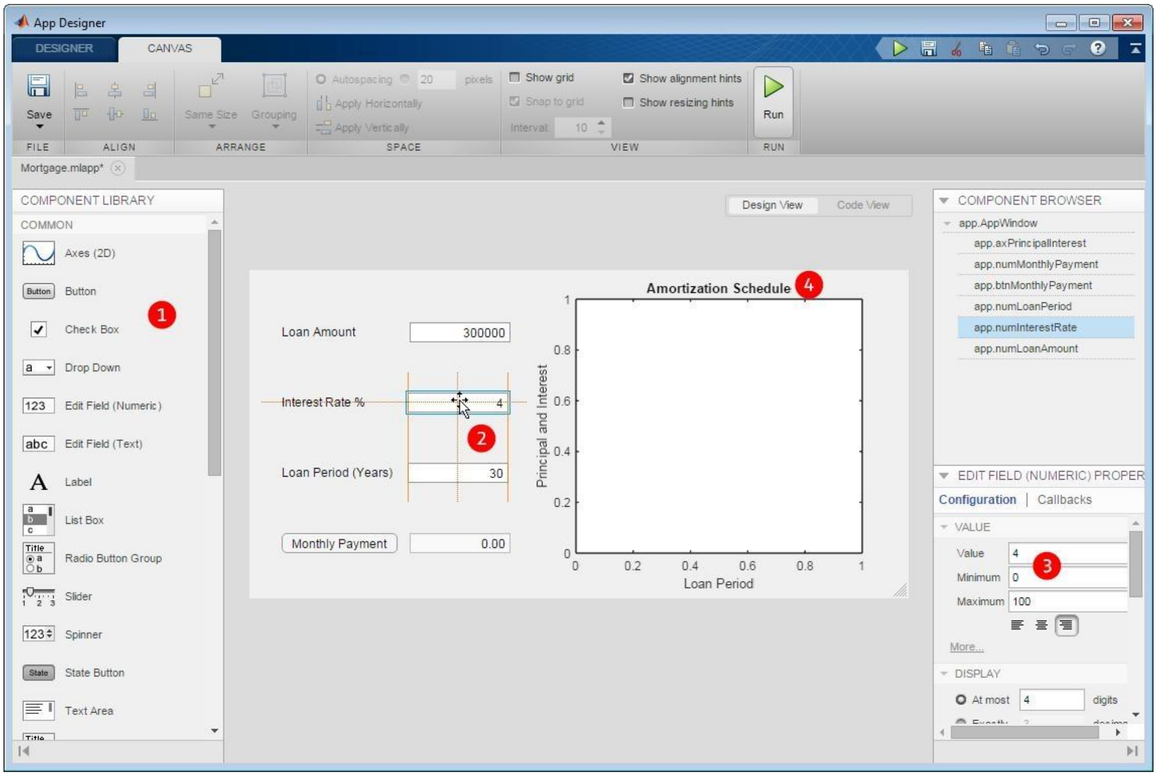

Figure B-1: App designer interactive design environment [62].

❖ **Built-In Editor Integration**

- Edit app code within App Designer using an integrated version of the MATLAB Editor (1).
- Use the App Layout pane to identify the names of the components in the code (2).
- Use the Component Browser to add callbacks or navigate to existing callbacks (3).
- Use programming alerts to avoid common coding errors (4).

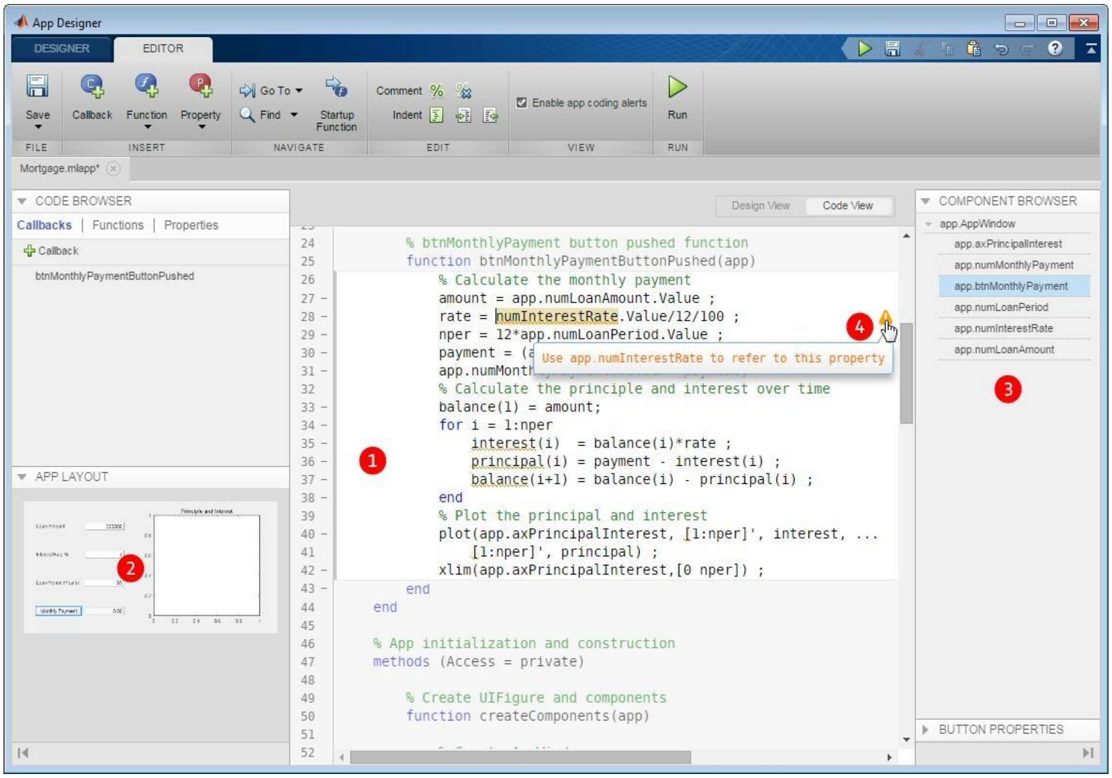

Figure B-2: App designer built-in editor integration [62].

❖ Code Format for Apps

- Implement the behavior of your app as an object-oriented program (1).
- Access the user interface components as properties of the app (2).
- Create custom properties for data that is shared between different parts of the app (3).
- Define callback functions as methods to control the behavior of the app (4).

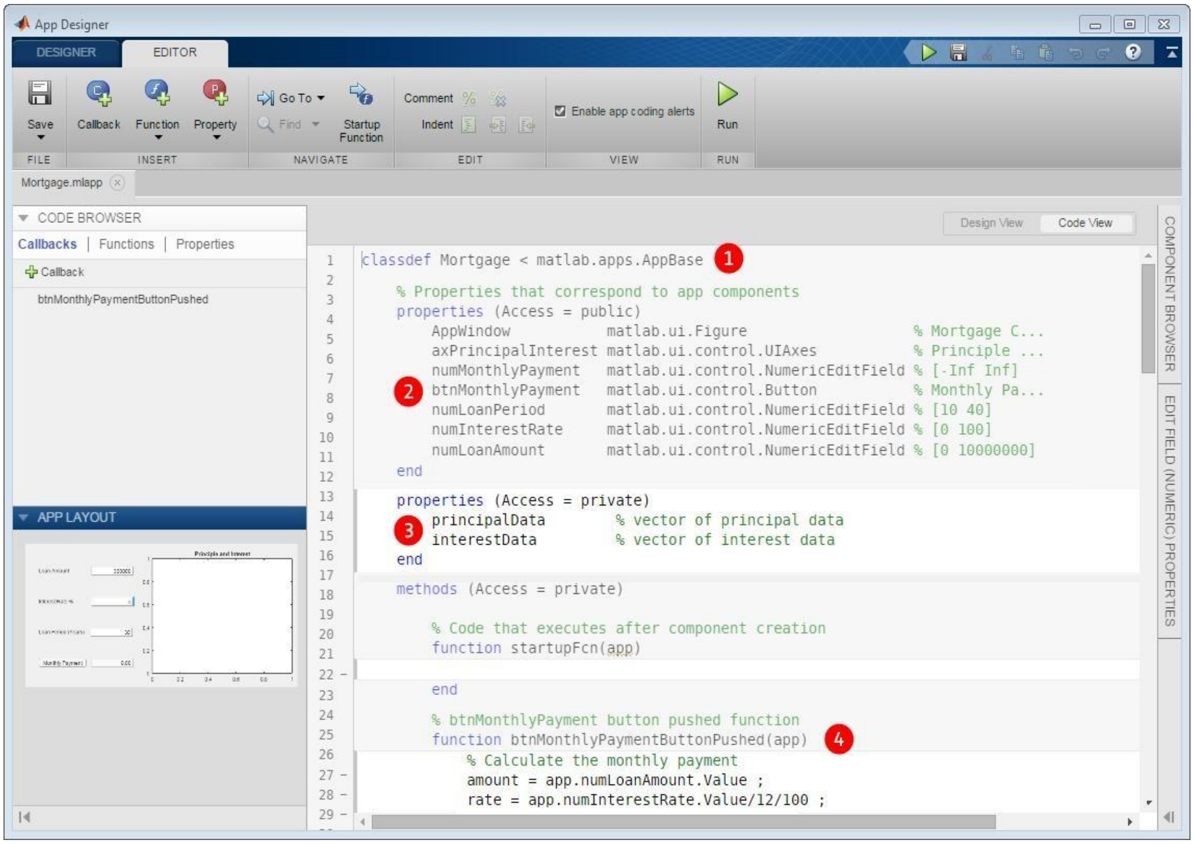

Figure B-3: App designer code format for Apps [62].