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Software-based Implementation of Digital Communication System including Equalization, Synchronization, and Symbol Detection.

by

Leul Wuletaw

Supervisor

Dr.-Ing. Dereje Hailemariam

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Abstract

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The role of digital communication system in our day-to-day life is drastically increasing. State of the art technologies are being deployed now and then to cope up with the increasing data rate and high quality of service demands of users. Fulfilling these demands, in turn, require having digital communication systems that are more reliable and pervasive. For digital communication to be feasible and applicable in the real world, care must be taken of processes occurring between the transmitter and receiver. Hence, this thesis tries to examine major blocks present in digital communication system and develops an easy-to-use GUI (Graphical User Interface) for designing such systems. The communication system consists of source encoder and modulator from the transmitter side and synchronization (both time and frequency), equalization and detection in the receiver side. Moreover, the channel considered is frequency selective with Additive White Gaussian Noise. In the channel equalization, where existing algorithms are reviewed and implemented to compensate for signal distortions caused by the channel. Then synchronization techniques are used to accurately estimate carrier and symbol timing offsets of the received signal. Finally, symbol detection mechanisms are utilized in order to make an optimal decision of incoming symbols and output a decoded binary 1s and 0s for further processing.

Acknowledgements

My greatest gratitude from the depth of my heart goes to God for endowing me with courage, strength as well as health through-out my journey. I would like to express my deep gratitude to Dr.-Ing. Dereje Hailemariam for his patient guidance, enthusiastic suggestions and useful critiques for the last three semesters of project work. I would also like to thank Tewodros Amsalu for his advice and technical assistance during the implementation stage of the thesis. Finally, I wish to thank my family and friends for their support and encouragement throughout my study.

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Abbreviations

AWGN	**A**dditive **W**hite **G**aussian **N**oise
BER	**B**it **E**rror **R**ate
BPF	**B**and **P**ass **F**ilter
BPSK	**B**inary **P**hase **S**hift **K**eying
CMA	**C**onstant **M**odulus **A**lgorithm
DFE	**D**ecision **F**eedback **E**qualizer
FSA	**F**ractionally **S**paced **A**lgorithm
GUI	**G**raphical **U**ser **I**nterface
ISI	**I**nter **S**ymbol **I**nterference
JTC	**J**oint **T**echnical **C**ommittee
LS	**L**east **S**quares
MAP	**M**aximum **A**posteriori **P**robability
ML	**M**aximum **L**ikleihood
MLSE	**M**aximum **L**ikleihood **S**equence **E**stimation
MMA	**M**ulti **M**odulus **A**lgorithm
MMSE	**M**inimum **M**ean **S**quare **E**rror
PAM	**P**ulse **A**mplitude **M**odulation
PCM	**P**ulse-**C**ode **M**odulation
PD	**P**hase-**D**etector
PLL	**P**hase **L**ocked **L**oop
QAM	**Q**uadrature **A**mplitude **M**odulation
QPSK	**Q**uadrature **P**hase **S**hift **K**eying
SGD	**S**tochastic **G**radient **D**escent
SNR	**S**ignal power-to-**N**oise **R**atio
SRCA	**S**ignal **R**educed **C**onstellation **A**lgorithm

SRRC	**S**quare **R**oot **R**aised **C**osine Filter
VCO	**V**oltage **C**ontrolled **O**scillator
ZF	**Z**ero **F**orcing

Chapter 1

INTRODUCTION

1.1 Introduction

A communication system delivers information between a source and destination that are some distance apart. In the same token, a digital communication system is used to convey information digitally.

A digital communication system has achieved wider applications than its analog counterpart. Now a day, it is used pervasively from mobile handsets in our hand to high bandwidth satellite communication, mainly due to reasons such as robustness to noise and interference, reliability and cost of implementation. Moreover, it also supports features such as encryption making it more secure than analog systems. Figure 1.1 shows a functional block diagram representation of digital communication.

Initial blocks present at the transmitter side are encoder and modulator where source analog signals are sampled and converted in to digital data and transmitted using a specific modulation scheme. The channel has a statistical behavior that introduces distortion to signal transmitted towards the receiver. It presents noise, fading and delay through which the transmitted signal passes. Finally, equalization, synchronization, demodulation and decoding are performed to reverse the transmission effects and retrieve the transmitted information at the receiver.

Despite the tremendous impact and wide applications they have, digital communication systems are still on the infant stage, especially in developing countries like Ethiopia, as these countries play insignificant role in system development. This is due to lack of a broad understanding of the design of digital systems and analysis of various factors arising in and out of the functional entities. Some of these factors are where and when the digital communication system is to be deployed, type of source to be transmitted, available signal-to-noise power ratio (SNR), type of channel, so and so forth. For this reasons, a communication system design and analysis tools are deemed necessary for testing and building user specific systems as per own requirement.

The main subject of this thesis lies on three entities of a digital communication system, i.e. carrier and symbol timing recovery, equalization, and symbol detection.

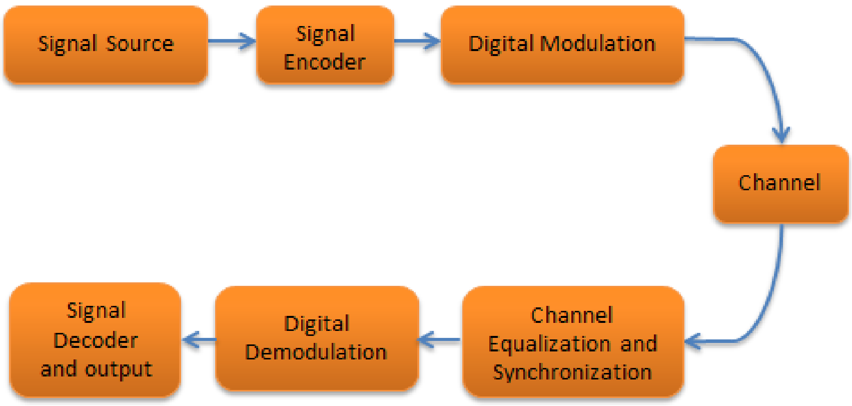

FIGURE 1.1: Functional block diagram of typical digital communication system.

1.2 Literature Review

1.2.1 Historical Background

The practical application of digital systems started immediately before the period of World War II, through the landmark invention of pulse-code modulation (PCM) by Alex Reeves [4]. He used this technique for digital encoding of speech signals and was further developed during the war to enable the encryption of speech signals. But the theoretical foundation for signal transmission was laid prior to the war, a classic paper being published in 1928 by Harry Nyquist. He developed mechanism for the successful reception of transmitted signal over dispersive channel [5].

The turning point in digital communication system was the invention of transistor, which stimulated the application of electronics to switching and digital communications, such as [5]:

- A stored program control system was placed at Bell Laboratories in March 1958.
- The first digital switch based commercial telephone service began in Morris, Illinois, in June 1960.
- Again Bell Laboratories founded the first T-1 carrier system transmission in 1962.

In 1943, D. O. North devised the famous matched filter used for optimizing the signal power to noise ratio. The same result was independently reached by J. H. Van Vleck and D. Middleton, who coined the term matched filter [5].

A major breakthrough providing a theoretical foundation of digital communications were laid by Claude Shannon in a classic paper entitled A Mathematical Theory of Communication in 1948, later amended to The Mathematical Theory of Communications. He disproved that increasing information rate doesnt necessarily result increase in the probability of error, provided that the channel capacity is not exceeded [5].

Following Shannons paper [5]:

- Error correcting codes were developed by M. J. E Golay and Richard W. Hamming in 1950.

- Turbo codes were developed by C. Berrou, A. Glavieux, and P. Thitimajshima 1993.

Communication between computers and terminals becomes possible at low speed (300 to 1200 bits/sec) in the early 1950s. Furthermore, Robert Lucky and G. Ungerboek pioneered adaptive equalization and efficient modulation techniques to increase this data rate. In tandem with error-control methods by Viterbi, this gave birth to high speed digital communication. This was further extended to the provision of the internet, satellite communication, and optical communication [4].

1.2.2 System Model

Most digital communication systems encode and modulate the source data prior to transmission. The physical channels through which this modulated signals are transmitted are usually bandlimited and causes dispersion of the signal in time domain. As a result gives rise to a phenomenon known as inter-symbol interference (ISI). ISI causes the channel to be frequency selective in frequency domain. The severity of ISI depends on the operating environment (channel bandwidth) and the application at hand (coherence bandwidth), i.e. narrowband or broadband. At the receiver channel equalization is needed to mitigate the effect of the ISI induced by a dispersive channel [1][6].

Propagation delay in the channel can produce phase and frequency shifts. Moreover, clocks and oscillators at the transmitter and receiver are free running, which in addition to the delay causes loss of timing and synchronization [1]. Hence, synchronization is one major task of the receiver and must take account for the phase, frequency and time shifts as well as synchronizing the clocks and oscillators at the transmitter and receiver.

The core purpose of the demodulator is to perform decisions, i.e. symbol detection of the transmitted symbols.

1.2.2.1 Channel

Two phenomena are observed in wireless communication. Primarily, due to the presence of scatters in the environment, there is a delayed and scaled arrival of signals which leads to ISI. Consequently, ISI causes the channel to be selective in frequency. Secondly,

mobility of transmitter, receiver, or objects in the environment results in Doppler spread which in turn causes the channel to be time-selective [6]. Both ISI and Doppler spread severely degrade the performance of the wireless communication.
In addition to the fading, receiver thermal noise is the commonest impairment in most communication systems to the transmission medium itself. Hence, good channel models are critical to the design of efficient communication systems.

1.2.2.2 Demodulator

Consider that the transmitted signal waveforms are generated from N orthonormal basis functions. The demodulator is used at the baseband to convert incoming corrupted signal waveform in to an N-dimensional vector. It can be performed using N correlators or N-linear matched filters. The correlator uses basis functions to compute the received signals projection. On the other hand, matched filters use filters whose impulse response is matched to the basis functions [1].
In an alternative approach, the tasks of the demodulator may include synchronization, channel equalization, and detector that are explained in the subsequent sections.

1.2.2.3 Carrier and Symbol Synchronization

In digital communication received signal at the demodulator must be periodically sampled once every symbol interval in order to recover the transmitted information. Nonetheless, the unknown propagation delay between the transmitter and receiver distort the timing and also results in phase offset. Additionally, further carrier offsets may result from drifting in time between the oscillators at the transmitter and receiver. Hence, the symbol timing and carrier offset must be estimated and corrected for proper sampling and coherent reception of the received signal.
Two methods for carrier phase estimation are [1]:

- Training based estimation: Use of pilot signal that allows the receiver to extract the carrier frequency and delay of the received signal. Pilot signal is unmodulated carrier component that is tracked by a Phase Locked Loop (PLL) which is designed to be narrowband.

- Derive the carrier phase estimate directly from the modulated signal. This is a blind estimation technique that requires no training sequence. This is a widely used practice and the total transmitter power is used to transmit the information bearing signal only.

1.2.2.4 Channel Equalization

Equalizer is an optimal inverse filter that compensates for the ISI effects on the received signal.

The filter is used to collect back the dispersed symbol energy as a result of the channel. Viewed in the frequency domain, the equalizer enhances the frequency components with small amplitudes and attenuates those with large amplitudes. The main target of equalization is to satisfy the zero-ISI condition, for which the combined effect of the pulse shaping filters, channel filter and equalizer filter will be a flat composite-received frequency response and linear phase [1]. Matching the equalizer filter to the cascade of the pulse shaping filters and channel filter guarantees optimality.

The non-stationarity and time varying characteristics of the wireless channel poses a problem in incorporating filters to remove the channel effect. This is when adaptive equalizer comes in to the picture for which no knowledge of the channel is sought.

Equalizers can be realized in a number of ways; some of the common ones are Maximum likelihood sequence estimation (MLSE), Linear transverse filter equalizer, Decision Feedback Equalizer (DFE), and Adaptive equalizers [1]. All except adaptive equalizers are based on a priori information about channel impulse response.

Adaptive equalizer is based on the adaptation of the filter coefficients as it processes the data to attain some specified performance value. Adaptive equalizer can either be linear or decision-feedback equalizers type. Adaptive equalizer is based on training period or blind equalization. In the training sequence based equalization, a training sequence (i.e. data sequence in the training period), which is known to the equalizer is transmitted. Based on the least squares criterion (LS) criterion, the equalizer adaptively adjusts the filter coefficients to minimize the mean square error. On the other side, blind equalization methods come to rescue the bandwidth wasted on transmitting the training signals, which makes them viable solution for achieving high spectral efficiency. This method solely depends on the knowledge of signal structure and statistics to perform equalization. Some of the algorithms are Constant Modulus Algorithm (CMA), Multi Modulo Algorithm (MMA), Fractionally Spaced Algorithm (FSA), and Signed Reduced Constellation Algorithm (SRCA).

1.2.2.5 Symbol Detection Techniques

Once the effect of the time-varying wireless channel is known and possibly compensated by using mitigation techniques, further symbol detection schemes are required to discern between signal, on one hand, and random noise and interference, on the other hand.

In the simpler case that the signal is corrupted by noise only, the detector decides in each symbol interval based on the observation from demodulator (correlator or the matched filter type). An optimal detector is implemented based on either the maximum a posteriori probability (MAP) criterion or the maximum likelihood (ML) criterion [1].

A range of symbol detection techniques exist and can broadly be classified as linear and non-linear detection schemes. Linear symbol detection is the simplest form of symbol detection that treats all transmitted signals as interferences except for the desired stream from the target user. Typical examples are based on zero forcing (ZF) or Minimum Mean-Square Error (MMSE) criterion. On the other hand, non-linear detection schemes include ML detection which checks through all vector constellations for the most probable transmitted signal vector and estimates according to the Maximum Likelihood principle. It is considered the optimal detector since it minimizes the probability of error.

1.3 Problem Statement

Increasing demand of digital communication systems locally necessitates the use of efficient schemes and techniques subject to the requirement of the designer (vendor). As a result, various schemes may be put forward as a candidate. But selecting the one that suits best for a given application requires thorough investigation. And this all requires conducting a broad survey of the available schemes.

There is lack of a customized digital communication system designed in a developing or third world countries. Almost all of the products in the market are costly, making their adoption very difficult.

Despite availability of independent implementations and analysis of specific algorithms and blocks (entities) of digital communication system conducted by experts, there is lack of an end-to-end performance analysis made which would serve as a platform for comparing various schemes and techniques.

1.4 Objectives

1.4.1 General Objective

The general objective of this research thesis is to design and implement channel equalization, synchronization and symbol detection techniques for digitally modulated signals.

1.4.2 Specific Objectives

More specifically the objectives are broken down as:

Channel Equalizer

- Review linear, nonlinear and blind channel equalization techniques and algorithms;
- Investigate the performance of the reviewed algorithms and select appropriate equalization technique and see its implementation in software.

Time and carrier synchronization

- Review available time and carrier synchronization techniques (i.e., blind and non-blind) and algorithms;
- Select appropriate estimation techniques and implementation in software.

Demodulation

- Identify modulation scheme and order;
- Possible modulation types are: Phase modulation (BPSK, QPSK), Quadrature amplitude modulation.

Symbol detection techniques

- Review and select a symbol detection technique;
- Implement the selected detection technique for various modulation type and order, e.g., PAM, 4QAM;

- The symbol detector accepts symbols from subsequent units and generates binary 0s and 1s.

Final implementations

- Implement relevant techniques to retrieve information from a recorded practical signal and image. Decision on which blocks to include depends on the prior information about the recorded signal and implementation considerations.

1.5 Methodology

The methods employed to achieve the objectives of this thesis are:

1.5.1 Literature review

Includes reading books, journals, articles, simulation tools and other resources related to synchronization, channel equalization, symbol detection, and practical implementation.

1.5.2 System modeling

Involves modeling of the system under study including the information source, modulator at the transmitter, transmission channel with AWGN and equalization, synchronization, demodulation and symbol detection at the receiver. Figure 1.2 presents below the modeling of a digital communication system.

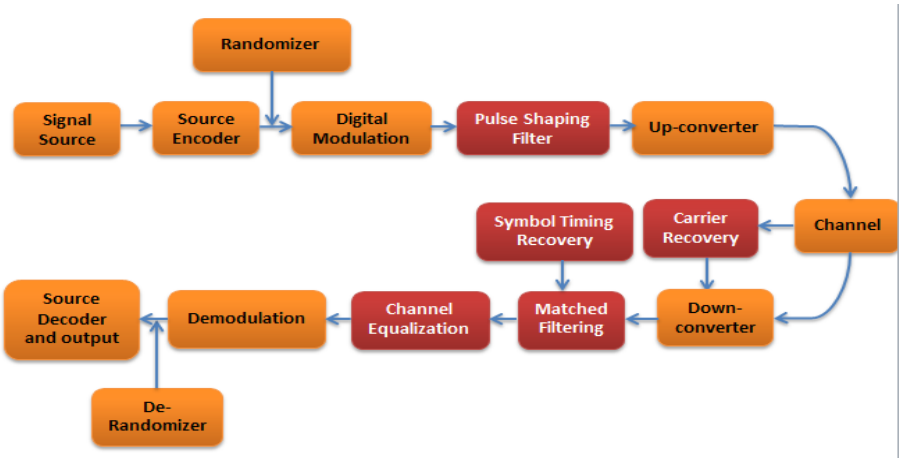

FIGURE 1.2: Functional block diagram of digital communication system.

1.5.3 Synchronization

1.5.3.1 Carrier Recovery

Of all the various techniques available to track the incoming carrier, Phase Locked Loop (PLL), Costas loop, and Squaring loop were analyzed.

- PLL: - is used to follow an unmodulated carrier. As shown in Figure 1.3, a conventional PLL consists of three elements, namely, phase detector (PD), loop filter, and voltage controlled oscillator (VCO). It uses a feedback loop to keep track of the phase of the incoming signal. The loop filter transfer function is given by [1]:

$$G(s) = k * \frac{(1+\tau_2 s)}{(1+\tau_1 s)} \tag{1.1}$$

Where k is loop filter gain, $\tau_1$and τ_2 are design parameters. The closed loop transfer function of the PLL is [1]:

$$H(s) = k * \frac{kG(s)/s}{1+kG(s)/s} \tag{1.2}$$

The denominator of *H(s)* is expressed in standard form as [1]:

$$D(s) = \omega_n^2 + 2\xi\omega_n + \xi^2 \tag{1.3}$$

Where ξ is called the loop damping factor and ω_n is the natural frequency of the loop, and are given by [1]:

$$\omega_n = \sqrt{k/\tau_1} \tag{1.4}$$

$$\xi = \omega_n(\tau_2 + 1/k) \tag{1.5}$$

Where $\xi < 1$ results in underdamped response,
$\xi = 1$ results in critically damped response, and
$\xi > 1$ produces in overdamped response.
The one-sided noise equivalent bandwidth of the loop is [1]:

$$B_{eq} = \frac{1+\tau_2\omega_n^2}{8\xi/\omega_n} \tag{1.6}$$

Equation (1.6) describes the tradeoff involved between the speed of response and noise in the phase estimate during the selection of the bandwidth of the PLL.

- Costas loop: - is sufficiently capable of tracking modulated suppressed carrier which was the main drawback of conventional PLL. Figure 1.4 presents the Costas loop which uses the same elements of PLL, but in a different way.

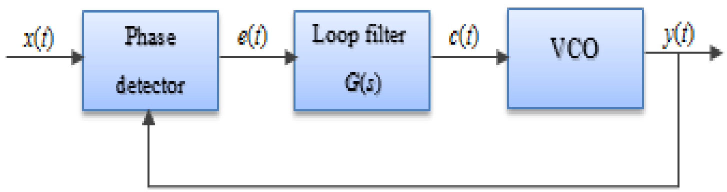

FIGURE 1.3: Phase Locked Loop [1]

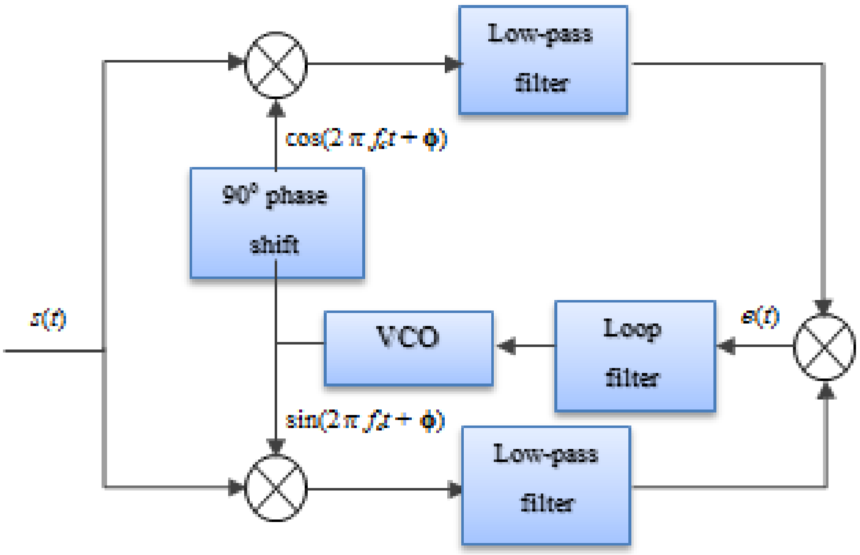

FIGURE 1.4: Block diagram of Costas Loop [1]

As can be seen from Figure 1.4, each quadrature outputs from the VCO are fed in the phase detectors. The incoming received signal is applied to both phase detectors and passed through low-pass filter. A third PD is used to multiply the outputs from PDs and generates an output which is scaled and filtered through a loop filter to control the VCO. Costas loop has a phase ambiguity of 1800, hence it must be used with differential encoding at the transmitter and differential decoding at the receiver.

- Squaring loop: - it squares the received signal for tracking. As shown in Figure 1.5, it employs a bandpass filter to pass the double frequency component of the squared signal [1].

  Additionally, Squaring loop contains a frequency divider, which is used to output a signal at the same frequency as the initial received signal. The same to Costas loop, a Squaring loop requires the use of differential encoding and differential decoding to avoid phase ambiguity.

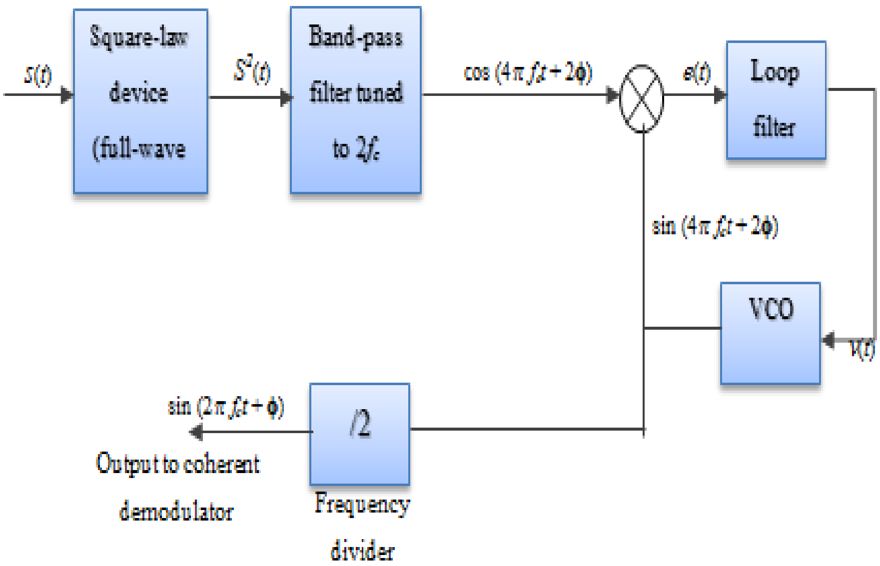

FIGURE 1.5: Squaring Loop [1]

1.5.3.2 Symbol Timing Recovery

Early-late gate and gradient based algorithms are designed and implemented to perform the timing synchronization. They belong to the class of non-decision directed (blind) recovery, i.e. they do not rely on any knowledge of the modulation symbols [1].

- Early-late gate: - one of the most common method which takes advantage of the symmetrical property of the received symbols about the peak sample of the symbol. From Figure 1.6, one can understand that the center of the symbols is an optimum timing phase for maximum signal power to noise ratio (SNR) [2].

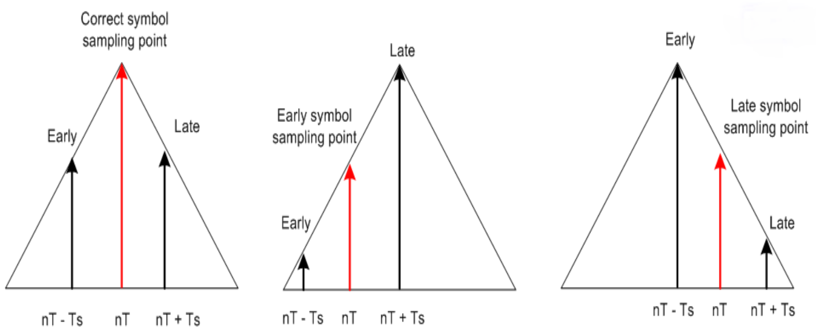

FIGURE 1.6: Early-late timing error computation [2]

This timing recovery algorithm generates its error by using samples that are early and late compared to the ideal sampling point. The generation of the error requires at least three samples per symbol. The samples indicate the direction towards to the center of the symbol. The error signal is computed using the following equation:

$$e = K * (y_n^2 - y_{n-1}^2) \tag{1.7}$$

Where K is the step size parameter and the timing phase deviation between y_n and y_{n-1} is δ.

- Gradient based: - is an optimized version of the early-late gate algorithm used for synchronization in digital receivers. From Figure 1.7, we can observe that the algorithm uses two consecutive samples with an interval of half period. Thus, requiring a maximum of two samples per symbol.

  The error signal for the Gradient based algorithm is computed using the following equation [2].

  $$e = K * (y_n * y_{n-1}) \tag{1.8}$$

  Where K is the step size parameter and the timing phase deviation between y_n and y_{n-1} is T/2 seconds.

  Gradient based algorithm has the advantage of being insensitive to carrier offsets and robustness during symbol transitions.

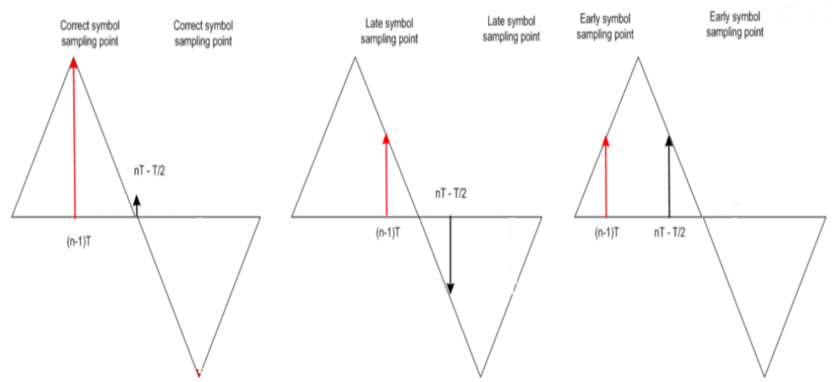

FIGURE 1.7: Gradient based timing error computation

1.5.4 Equalization

From a group of equalization techniques reviewed in the literature, blind adaptive equalization technique was of a best interest. It is blind in the sense that it is capable to equalize a signal solely from the transmitted data. This ensures that no extra bandwidth is necessary for its implementation. Furthermore, ample of blind equalization algorithms exist out of which CMA and MMA has been selected and implemented in this thesis, mainly for their computational simplicity and MMAs interesting ability to simultaneously track phase offsets (carrier recovery) and perform equalization.

- Constant Modulus Algorithm: - it takes advantage of the constant modularity of a set of random signals. It corrects any deviation of the received signal constant module caused by distortion in the channel. The cost function for CMA is given by [7]:

$$J_2(k) = E[(|y(k)|^2 - R_2)^2] \tag{1.9}$$

Where *y(k)* is the equalizers instantaneous output and R_2 is called constant modulus and the constant modulus is defined mathematically as [7]

$$R_2(k) = \frac{E[|a(k)|^4]}{E[|a(k)|^2]} \tag{1.10}$$

Where *a(k)* is the instantaneous transmitted symbol.

The N-tap weighted equalizers output is given by [7]

$$y(k) = W^T * X \tag{1.11}$$

Where X the input sequence for the equalizer and W the equalizer weight which can be found using [7]:

$$X(k) = [x(k), x(k-1), ., x(k-N+1)]^T \tag{1.12}$$

$$W(k) = [w_0(k), w_1(k-1), ., w_{N-1}(k-N+1)]^T \tag{1.13}$$

The equalizer weight is updated using [7]:

$$W(k+1) = W(k) - \mu \bigtriangledown J_2(W(k)) \tag{1.14}$$

Using the SGD algorithm Equation (14) is simplified to [7]:

$$W(k+1) = W(k) - \mu e(k) X^*(k)) \tag{1.15}$$

Where *e(k)* is the instantaneous error and given by:

$$e(k) = y(k)(|y(k)|^2 - R_2) \tag{1.16}$$

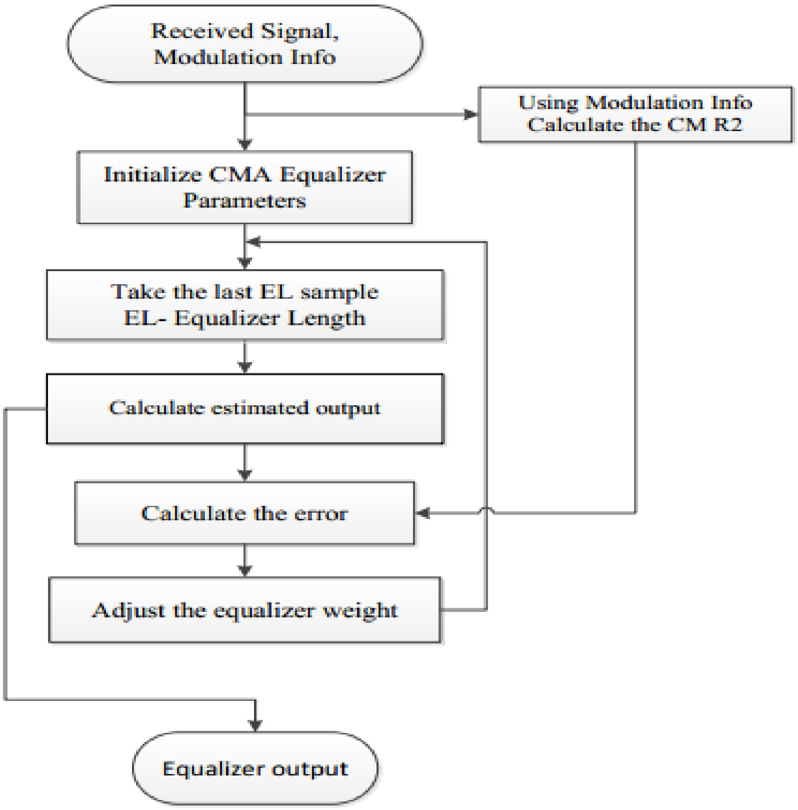

FIGURE 1.8: CMA Implementation flow chart [3].

- Multi Modulus Algorithm: -like CMA uses the constant modularity of signals to minimize dispersions but considers the real and imaginary parts of y(n) separately in its cost function. This enables it to carry the channel phase distortion [8].
  The MMA adjusts the dispersion of $y_R(k)$ and $y_I(k)$ separately and the cost function is given by [9]:

$$J_{MMA}(w) = J_R(w) + J_I(w) \tag{1.17}$$

Where $J_R(w)$ and $J_I(w)$ are the cost functions for the real and imaginary part and given by [9]:

$$J_R(w) = E[|y_R(w)|^2 - R_{2,R}^2] \tag{1.18}$$

$$J_I(w) = E[|y_I(w)|^2 - R_{2,I}^2] \tag{1.19}$$

Where $R_{2,R}$ and $R_{2,I}$ are the constant modules for the real and imaginary parts respectively and can be calculated using [9]:

$$R_{2,R}(k) = \frac{E[|a_R(k)|^4]}{E[|a_R(k)|^2]}, R_{2,I}(k) = \frac{E[|a_I(k)|^4]}{E[|a_I(k)|^2]} \tag{1.20}$$

The equalizer weight is updated using [9]:

$$W(k+1) = W(k) - \mu \bigtriangledown J_{MMA}(W(k)) \tag{1.21}$$

Using SGD algorithm Equation (21) is simplified to

$$W(k+1) = W(k) - \mu e(k) X^*(k)) \tag{1.22}$$

Where *e(k)* is calculated from the real and imaginary error as [9]:

$$e(k) = e_R(k) + e_I(k) \tag{1.23}$$

$$e_R(k) = y_R(k)(|y_R(k)|^2 - R_{2,R}), and \tag{1.24}$$

$$e_I(k) = y_I(k)(|y_I(k)|^2 - R_{2,I}) \tag{1.25}$$

Then the output of the equalizer can be found using:

$$y(k) = W^T * X \tag{1.26}$$

Where X is the input sequence for the equalizer and W is the equalizer weight which can be found using [9]:

$$X(k) = [x(k), x(k-1), ., x(k-N+1)]^T \tag{1.27}$$

$$W(k) = [w_0(k), w_1(k-1), ., w_{N-1}(k-N+1)]^T \tag{1.28}$$

1.5.5 Channel Models

Three channel models are used in the simulation to evaluate the performance of equalization techniques. Figure 1.10 clearly depicts the channels for raised cosine pulse.

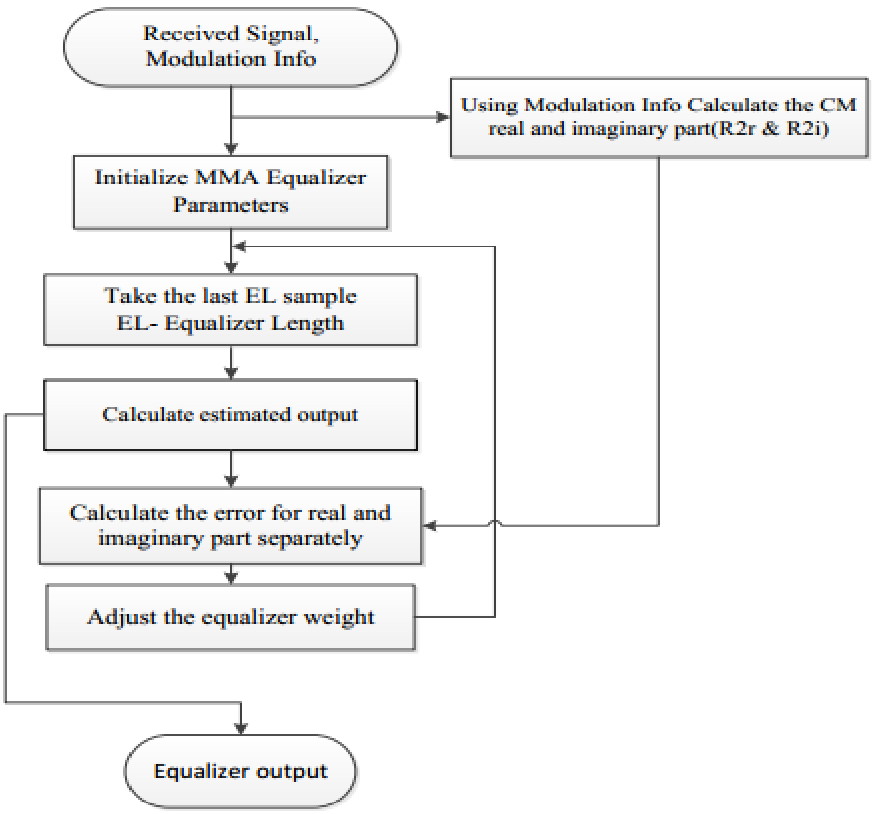

FIGURE 1.9: MMA Implementation flow chart [3].

1.5.6 Symbol Detection

Of all the possible detection mechanisms, a non-linear detection scheme known as ML detection is used. It has increased performance when used in conjunction with a raised cosine pulse shaping filter. A raised cosine filter enhances received SNR when used as a matched filter.

Figure 1.11 presents the plot of a square root raised cosine filter. Here, the filter has a roll-off factor of a =1.

Matched in the sense that the shape of the filter is similar with the transmitted signal pulse. The matched filter is used:

- As a low pass filter to filter off unwanted signal from received data.
- For correlating the received signal with the transmit pulse shape.

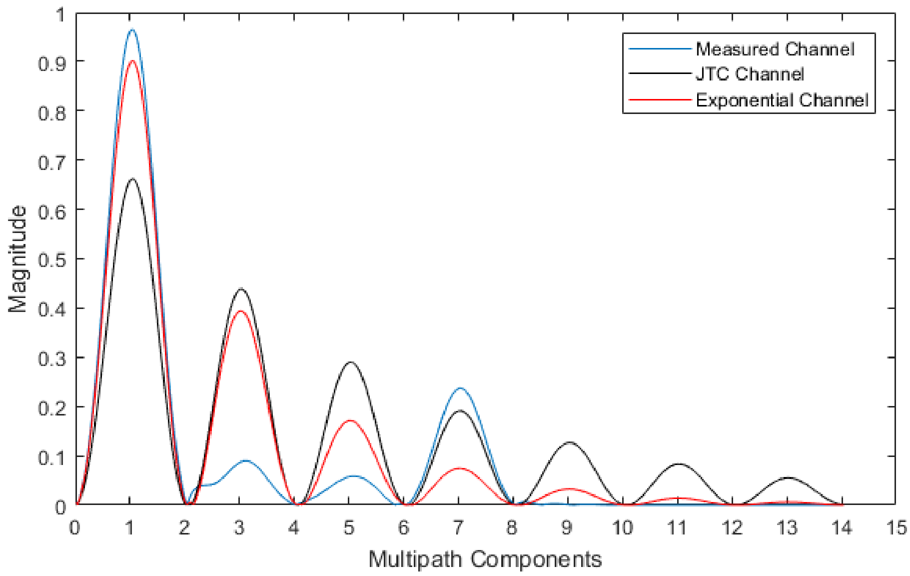

FIGURE 1.10: Channel impulse response for raised cosine.

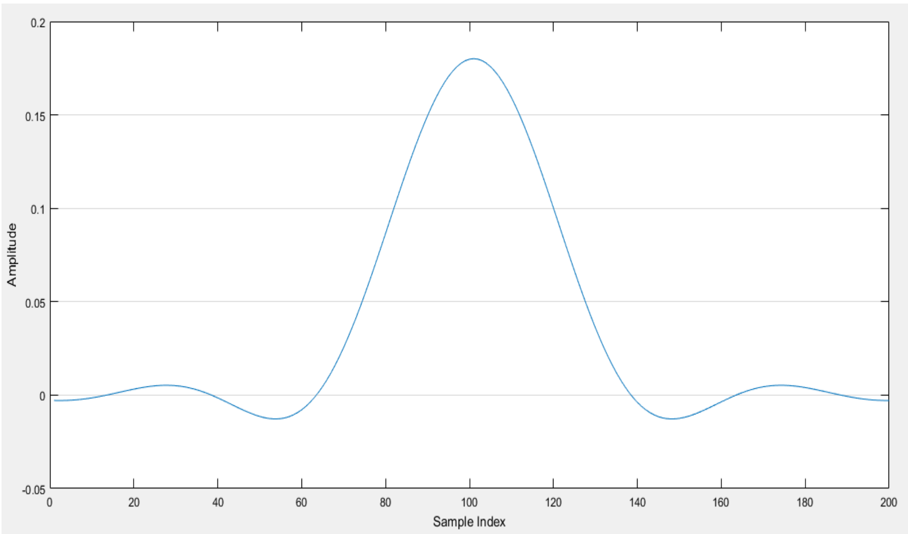

FIGURE 1.11: Raised cosine pulse.

1.5.7 Simulation and Practical Implementation

The following specific tasks are performed:

- Simulating the modeled communication system using MAT LAB and then incorporating implementations of equalization, synchronization, demodulation and detection methods into the simulated system.
- Implement all these techniques to retrieve information from a recorded practical signal and image.

- Develop a customized GUI for designing and simulating a digital communication system.

1.5.8 Performance comparison

- Evaluating the performance of implemented systems and comparing them.
- Candidate performance metrics are: bit error rate, probability of correct estimation and complexity analysis.

Chapter 2

RESULT AND EVALUATION

This chapter discusses the results obtained by the MATLAB-based simulation of synchronization and equalization algorithms. The simulations were done for BPSK and 4-QAM constellation with unit energy symbols. The pulse shape used is square root raised cosine filter (SRRC) pulse with 100 excess bandwidth. SNR used for performing simulation ranges between 10 and 20dB. A 32-bit, 4GB RAM host computer is used for testing runtime (complexity) of algorithms. The equalization algorithms are simulated in typical channels representing the transmission environment. The simulation for synchronization is conducted both in the presence and absence of an AWGN channel.

2.1 Equalization

The simulation results are organized in sections for each equalization technique, which itself contains subsections for each channel models. Then the results are presented using three channel models.

2.1.1 Constant Modulus Algorithm

The CMA described in Section 1.5.4 was simulated using MATLAB according to the flow chart in Figure 1.8.

2.1.1.1 Measured Channel Impulse Response

Simulation is conducted using a randomly generated data and exposed to a measured channel impulse response shown in Figure 1.10. The received signal and the equalized signal for different values of SNR are illustrated in Figure 2.1. The concentration of the equalized signals to the desired value increases as SNR increases.

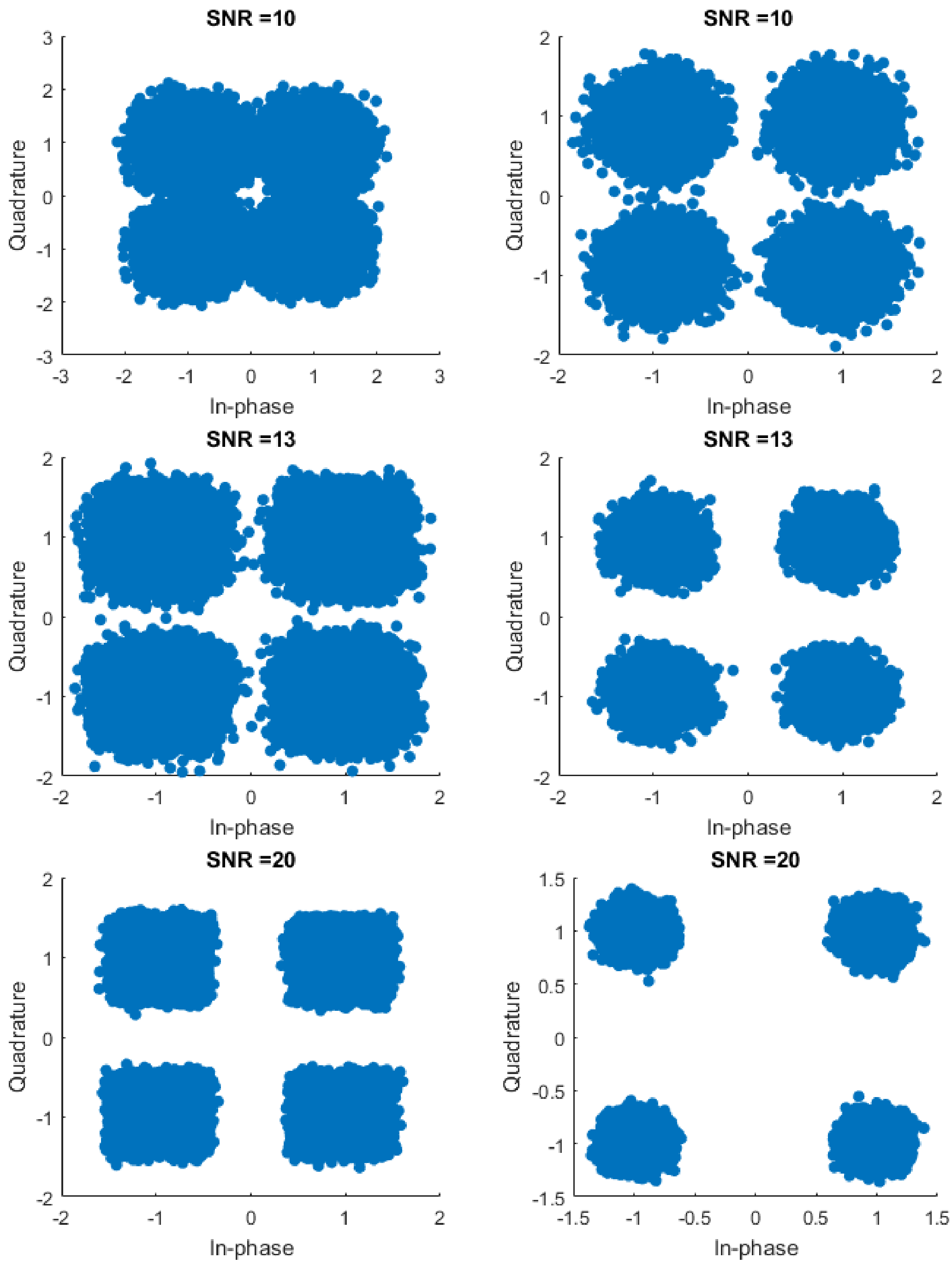

FIGURE 2.1: CMA Equalizer for measured channel response.

2.1.1.2 Joint Technical Committee Model

Randomly generated data source is transmitted across the JTC channel impulse response shown in Figure 1.10. From Figure 2.2, one can observe that the constellation of equalized signals become distinct enough at 13dB. We can also observe that the concentration to the desired value increases as SNR increases.

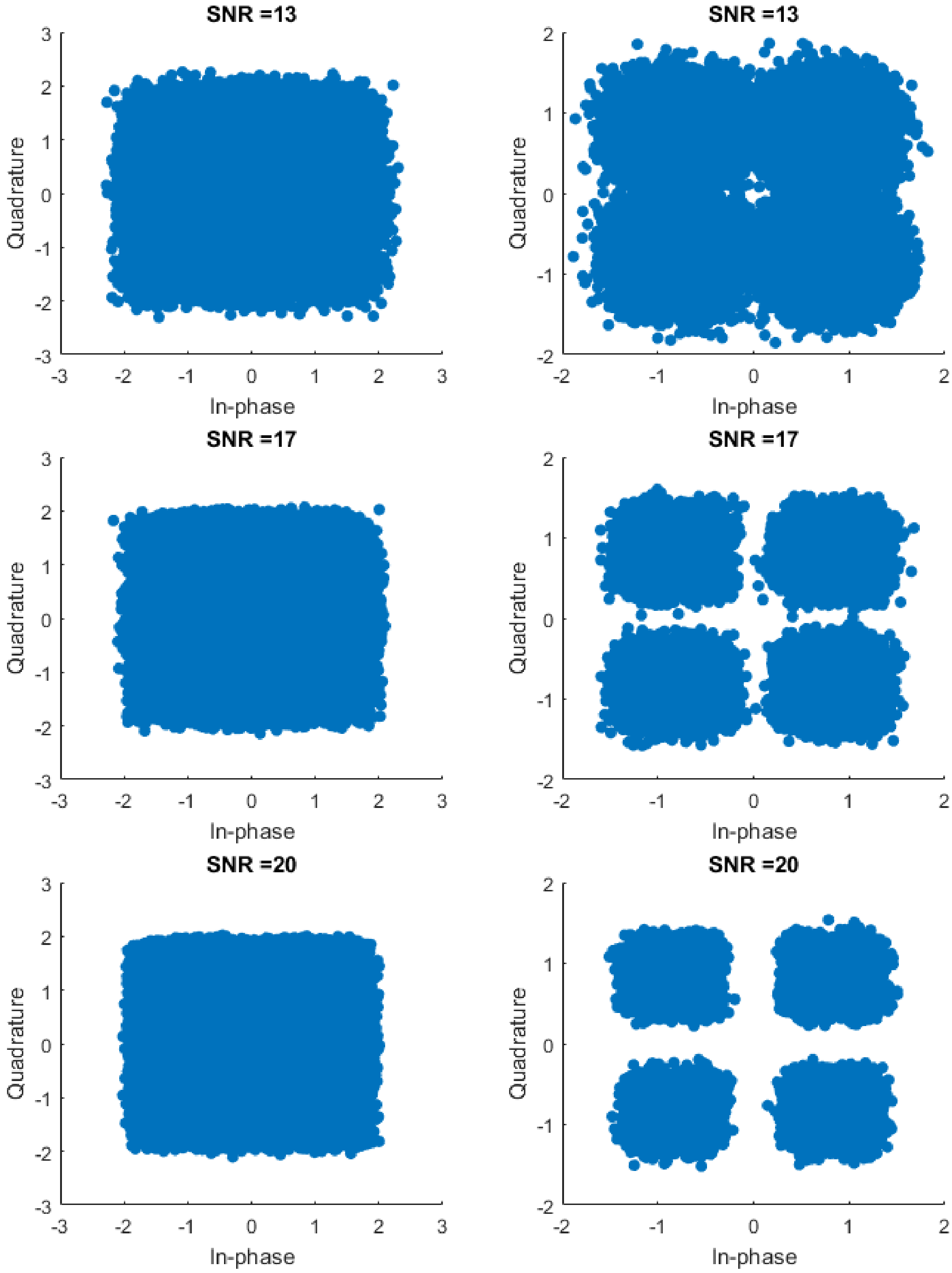

FIGURE 2.2: CMA Equalizer for JTC channel.

2.1.1.3 Phase Shift Recovery

Figure 2.3 shows that the CMA equalizer do not have capability to recover phase shift. Exponential channel with phase shift shown in Figure 1.10, is used to introduce phase shift to the transmitted signal.

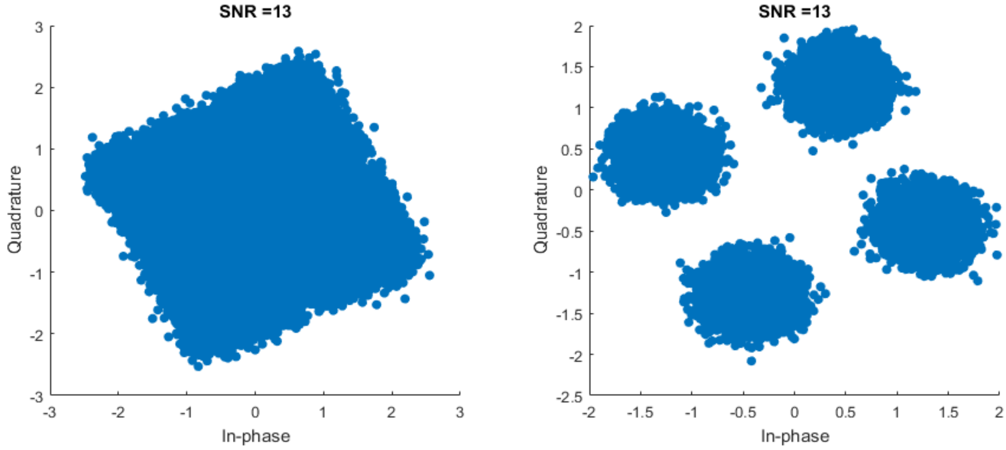

FIGURE 2.3: CMA Equalizer for a complex channel at 13 dB SNR.

2.1.1.4 BER Performance

The BER simulation result when exposed to the three channels is shown in Figure 1.10, the result indicate that the JTC channel poses more bit error rate. Best BER performance is observed for transmission through measured channel.

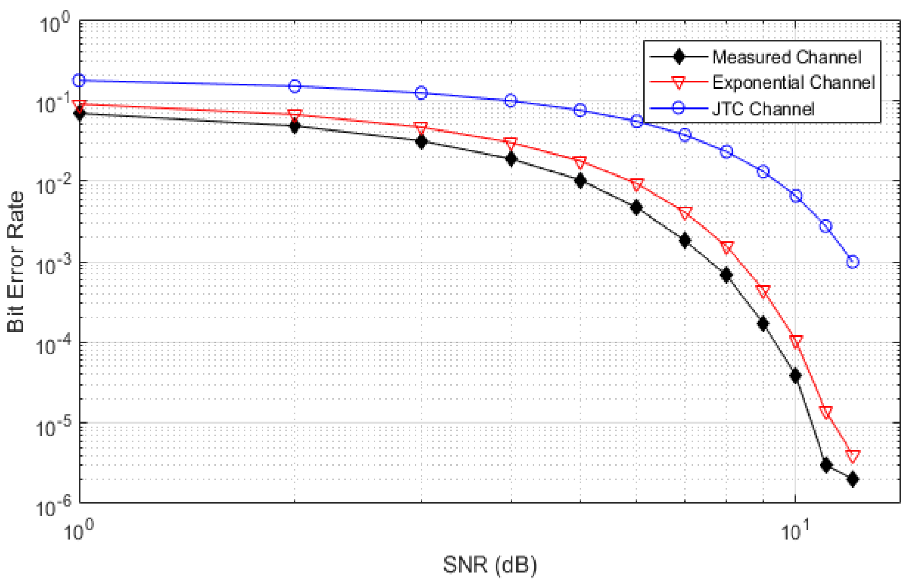

FIGURE 2.4: BER vs SNR for CMA.

2.1.1.5 Complexity

The CMA algorithm uses 1 sample per symbol period; it only considers the magnitude component of parameters in process of finding the equalizer weights, which minimizes the processing time.

2.1.2 Multi Modulus Algorithm

Based on the flow chart in Figure 1.9, the MMA was simulated in MATLAB. A randomly generated data is exposed to the three channels.

2.1.2.1 Measured Channel Impulse Response

Simulation is conducted using a randomly generated data and measured channel impulse response shown in Figure 1.10. The received signal and the equalized signal for different values of SNR are illustrated in Figure 2.5. The concentration of the equalized signals to the desired value increases as SNR increases. At 10dB, the equalized signals are distinct and concentrated in the constellation.

2.1.2.2 Joint Technical Model

Randomly generated data source is transmitted across the JTC channel impulse response shown in Figure 1.10. Similar to CMA, one can observe that the constellation of equalized signals for MMA become distinct enough at 13dB. One can also observe that the concentration to the desired value increases as SNR increases.

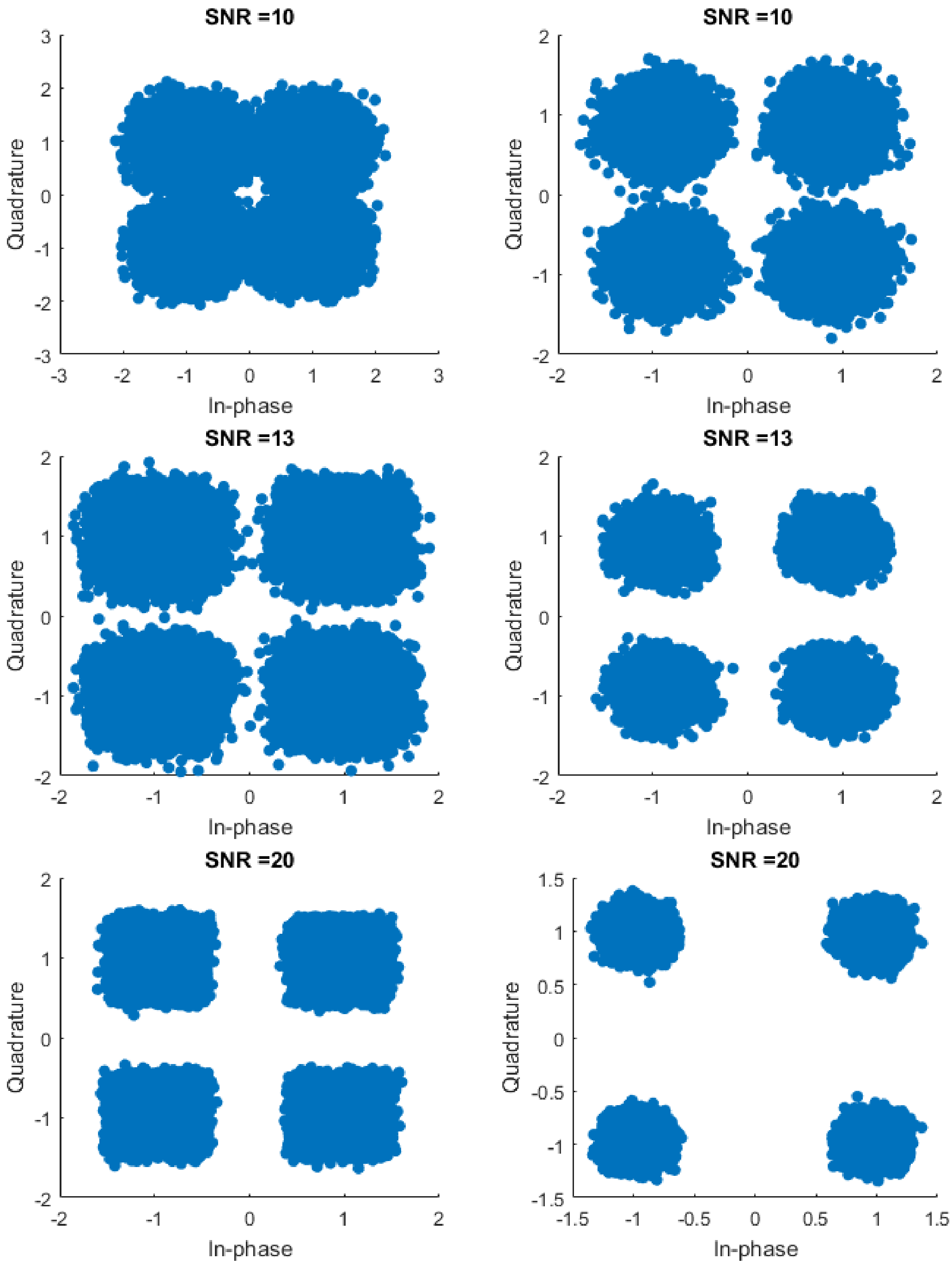

FIGURE 2.5: MMA Equalizer for measured channel response.

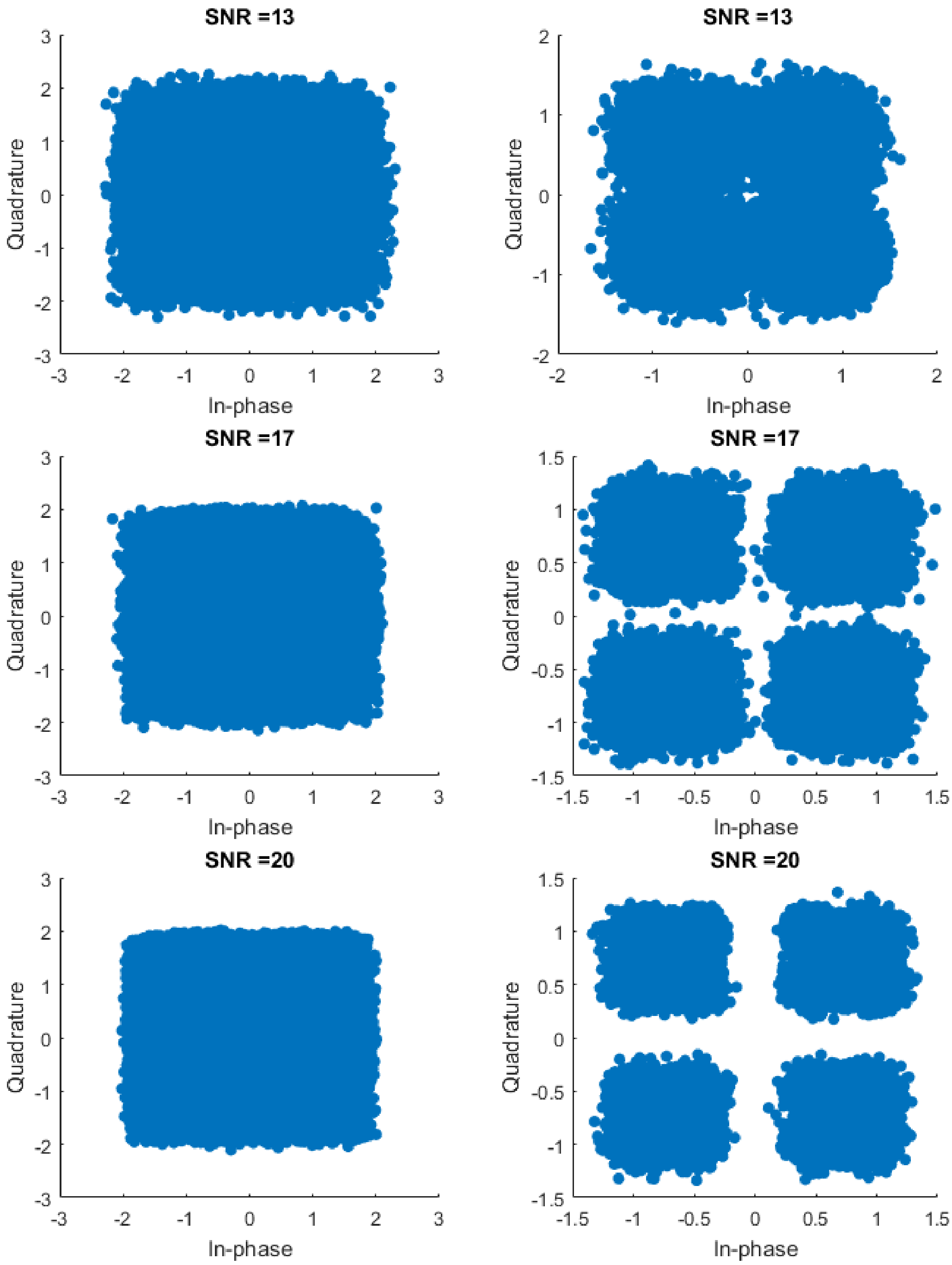

FIGURE 2.6: MMA Equalizer for JTC channel.

2.1.2.3 Phase Shift Recovery

Unlike the CMA equalization, the MMA equalization has the capability to recover phase shift introduced by the exponential channel. The phase shift introduced by the channel and the equalized signal constellations are illustrated in Figure 2.7.

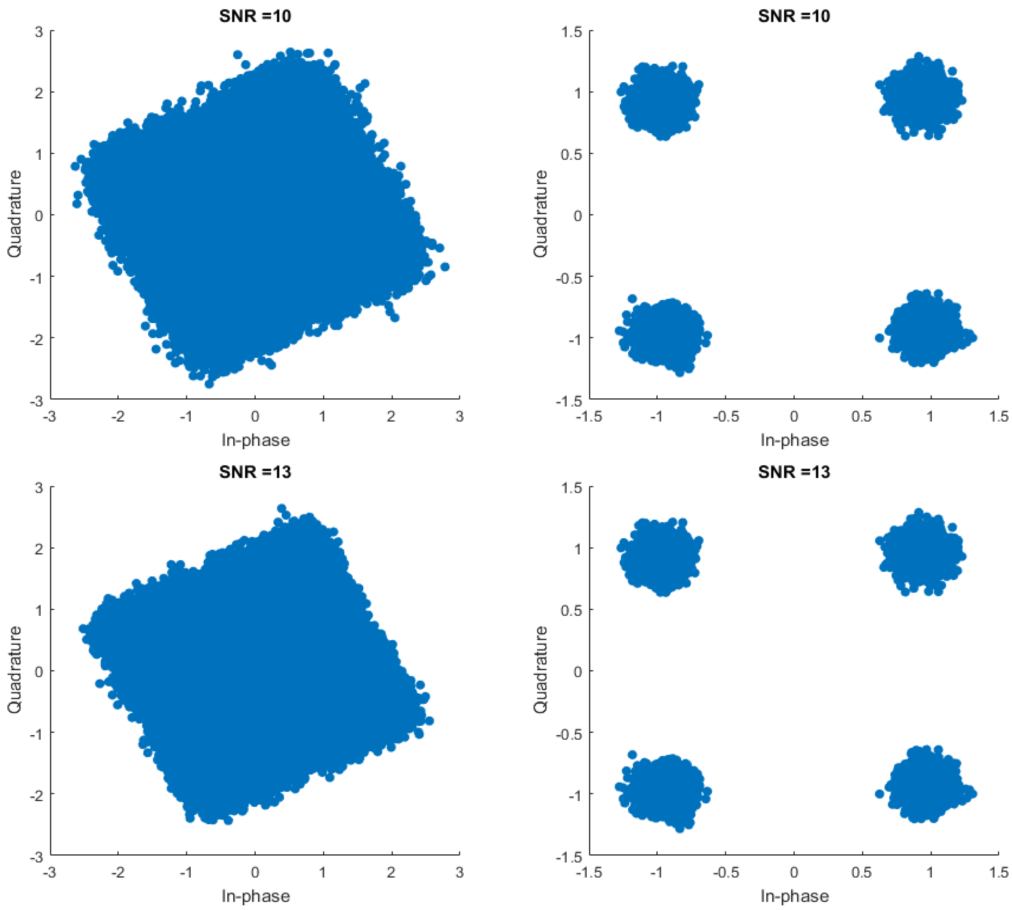

FIGURE 2.7: MMA Equalizer for a complex channel.

2.1.2.4 BER Performance

The BER simulation result when exposed to the three channels is shown in Figure 1.10, the result indicate that the JTC channel poses more bit error rate. Best BER performance is observed for transmission through measured channel. The BER performance of MMA is almost the same to BER performance of CMA shown in Figure 2.4.

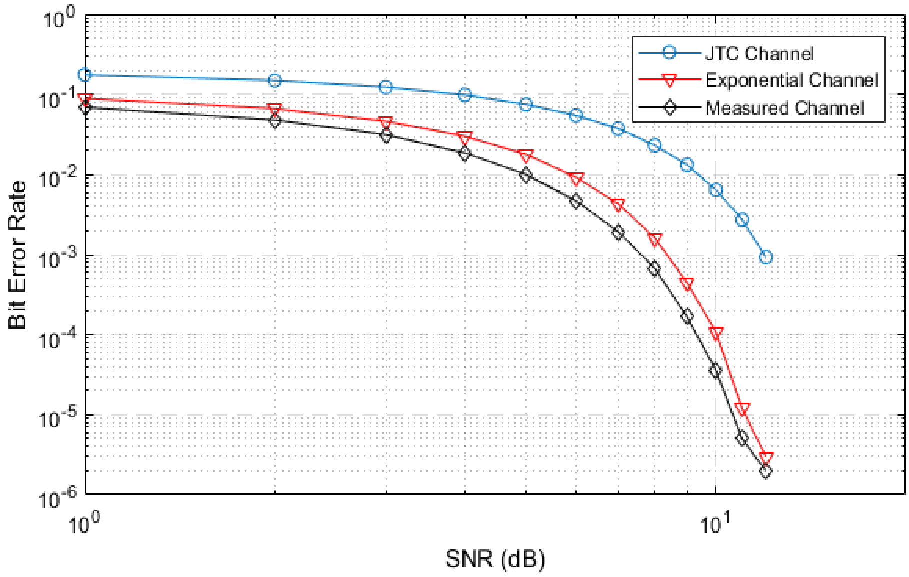

FIGURE 2.8: BER vs SNR for MMA.

2.1.2.5 Complexity

Like CMA algorithm, MMA requires one sample per symbol period. However, it uses additional parameter to consider the phase component during the process of finding the equalizer weight which in turn adds complexity.

2.2 Carrier Recovery Techniques

The simulation results are organized in sections for each equalization technique, which itself contains subsections for each channel models. Then the results are presented using three channel models.

2.2.1 PLL

The PD, loop filter and VCO blocks shown in Figure 1.3, were used for the complete system simulations. An unmodulated carrier signal is used for the simulation.

As shown in Figure 2.9, the phase tracking capability of PLL has been simulated for different values of k. While the fc remains constant at 0.05*. The result implies that as k increases, the curve reaches rapidly to the desired value but undergoes oscillation before settling. The PLL achieved convergence before the 150th sample index. It must be noted that a PLL is not capable of tracking modulated (suppressed) carrier input signal.

Figure 2.10 shows the simulation results of PLL for different values of fc. And the k is fixed to 0.15. The result shows that as fc increases, the curve undergoes an oscillation about the desired value. This clearly shows the effect of increased noise in to the system due to higher fc. From Figure 2.10, one can observe that the PLL converges before the 30th sample index.

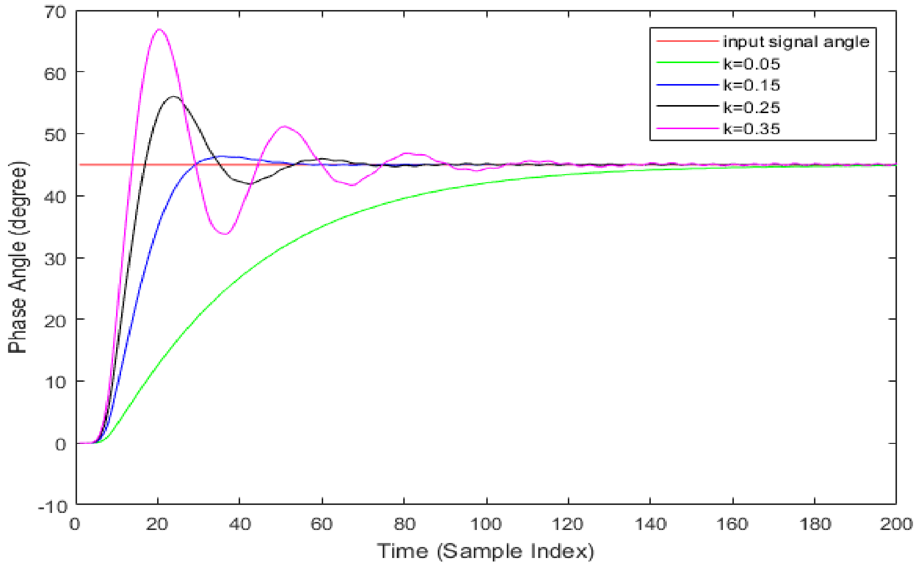

FIGURE 2.9: PLL carrier tracking for different loop coefficients.

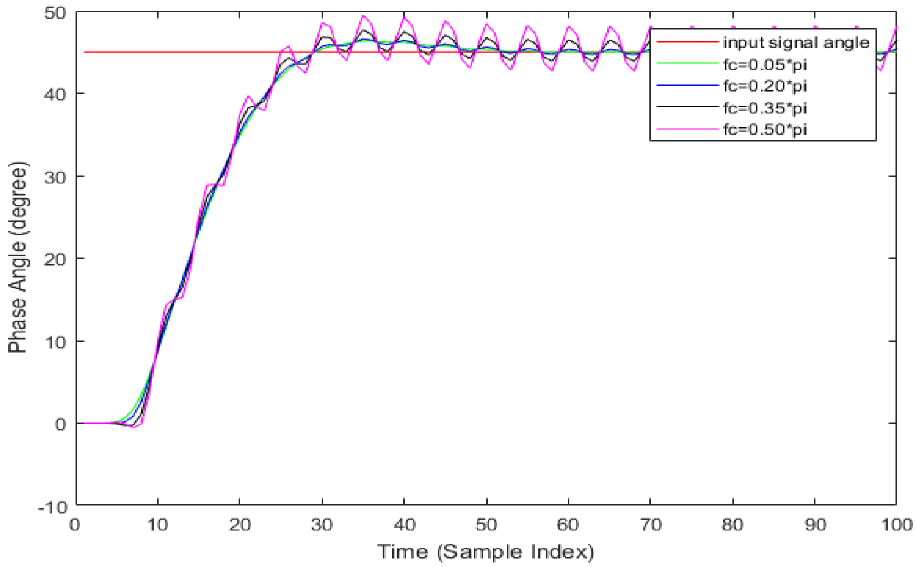

FIGURE 2.10: PLL carrier tracking for different LPF cutoff frequency.

2.2.2 Costas Loop

The two PDs, loop filter and VCO blocks shown in Figure 1.4, were used for the complete system simulations. The binary phase shift keying (BPSK) modulation is used for the simulation of the system.

Figure 2.11 shows the resulting phase tracking capability of Costas loop, which have been simulated for different values of k. And the fc used for this case remains constant at 0.05*. What this result implies is that as k increases, the curve reaches rapidly to the proximity of the desired value but undergoes oscillation. The Costas loop achieved convergence before the 700th sample index. And it must be noted that a Costas loop is capable of tracking modulated (suppressed) carrier input signal.

Figure 2.12 shows the simulation results of Costas loop for different values of fc. And the k is fixed to 0.15. The result implies that as fc increases, the curve tends to be more inaccurate and undergoes an oscillation about the proximity of the desired value. This clearly depicts the effect of increased noise in to the system as a result of higher fc. From Figure 2.12, one can observe that the Costas loop converges before the 200th sample index.

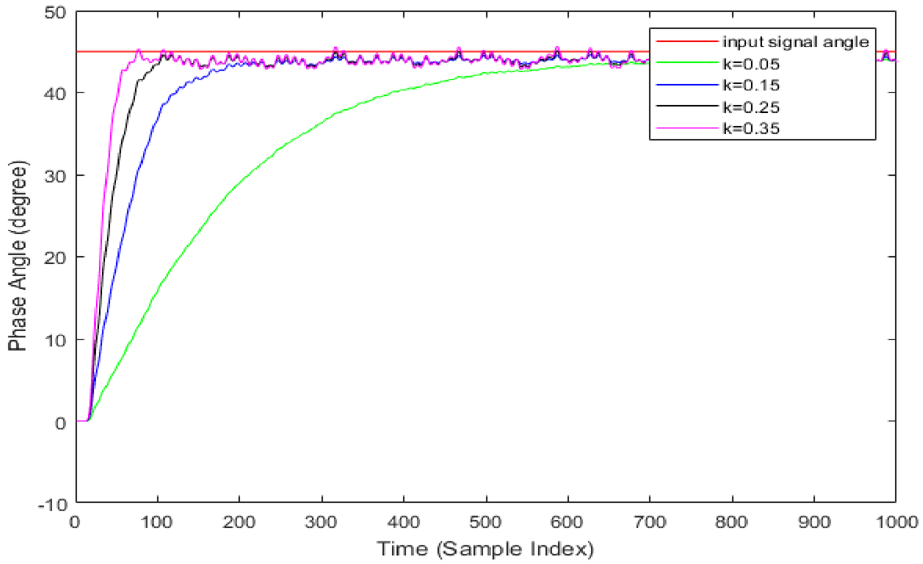

FIGURE 2.11: Costas loop carrier tracking for different loop coefficients.

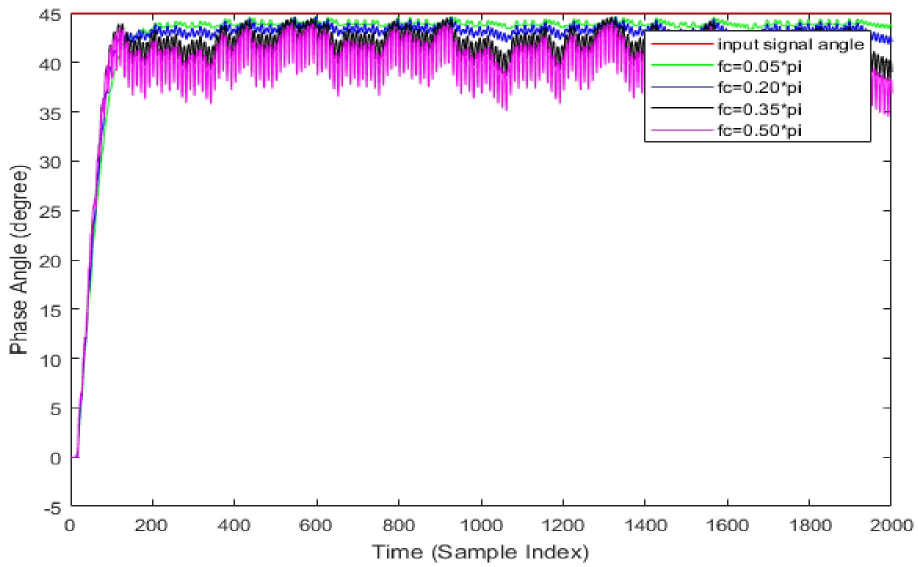

Figure 2.12: Costas loop carrier tracking for different LPF cutoff frequency.

2.2.3 Squaring Loop

The Squaring Law Device, bandpass filter (BPF), loop filter and VCO blocks shown in Figure 1.5, were used for the complete system simulations. The BPSK modulation is used for the simulation of the system.

Figure 2.13 depicts the resulting phase tracking capability of Squaring loop, which have been simulated for different values of k. While the fc remains constant at 0.05*. The result tells that as k increases, the curve reaches rapidly to the proximity of the desired value but more inaccurately than Costas loop. The squaring loop achieved convergence before the 8000th sample index. It must be noted that a squaring loop is capable of tracking modulated (suppressed) carrier input signal.

The simulation results of Squaring loop is also taken for different values of fc and it is shown in Figure 2.14. And the k is fixed to 0.5. What the result implies is that as fc increases, the curve tends to be more inaccurate and undergoes an oscillation. This clearly depicts the effect of increased noise in to the system as a result of higher fc. From Figure 2.14, one can observe that the squaring loop converges before the 200th sample index.

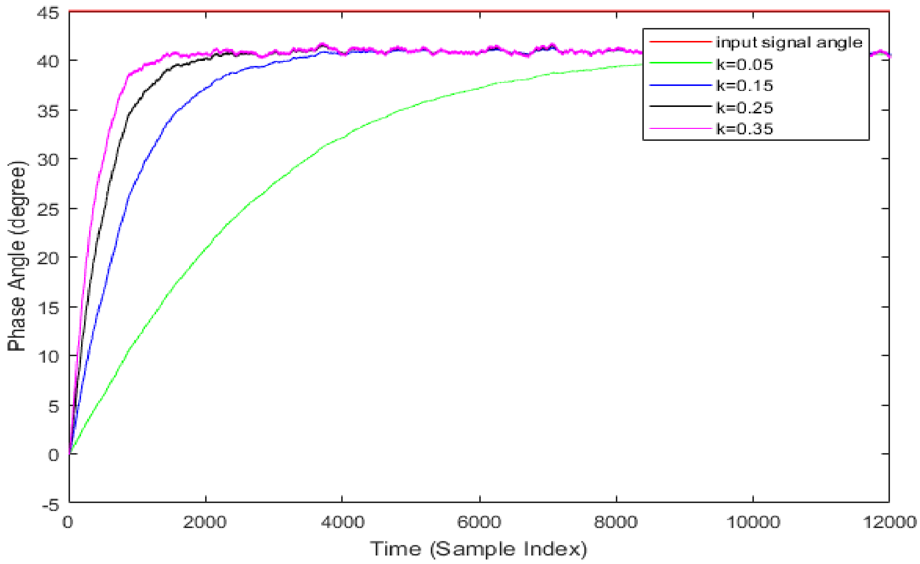

FIGURE 2.13: Squaring loop carrier tracking for different loop coefficients.

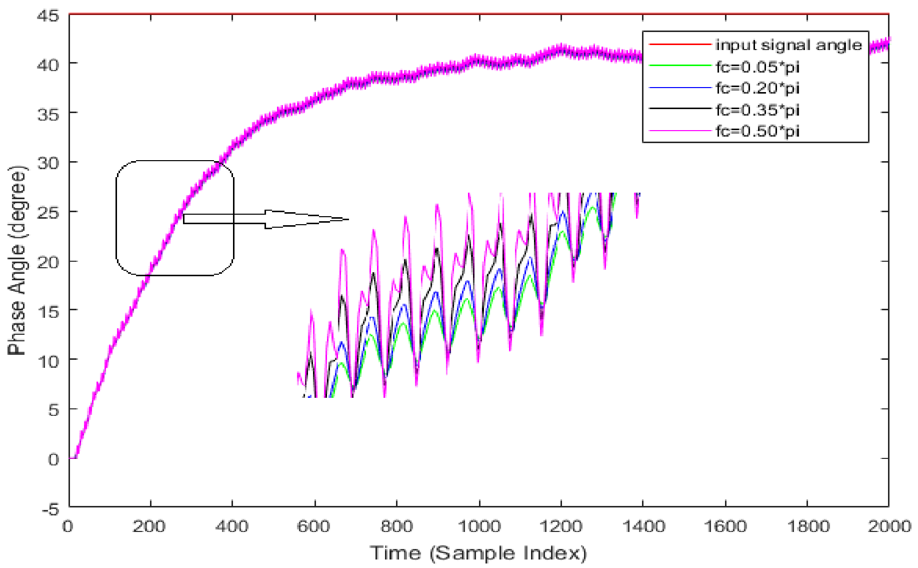

FIGURE 2.14: Squaring loop carrier tracking for different LPF cutoff frequency.

2.3 Symbol Timing Recovery

2.3.1 Early-Late Gate Algorithm

The error signal described in Section 1.5.3.2 was simulated using MATLAB, and the following result was obtained. The error signal is the output of the timing detector, which gives the information about the convergence of the system towards a solution. The parameters that change in Equation(1.7) are K and . Figures 2.15 and 2.16 show the error signal curves for different values of K both in the presence and absence of noise respectively with 0.25T timing phase deviation. In noise less case, one can observe that the error signal amplitude is convergent for average values of K. For step size between 1.25 and 30, the error signal converges. At 10dB the same pattern is observed. We can also observe that being initially convergent in the absence of noise, lower values of K has improved BER and stable performance in the presence of noise. This is illustrated in Figure 2.16.

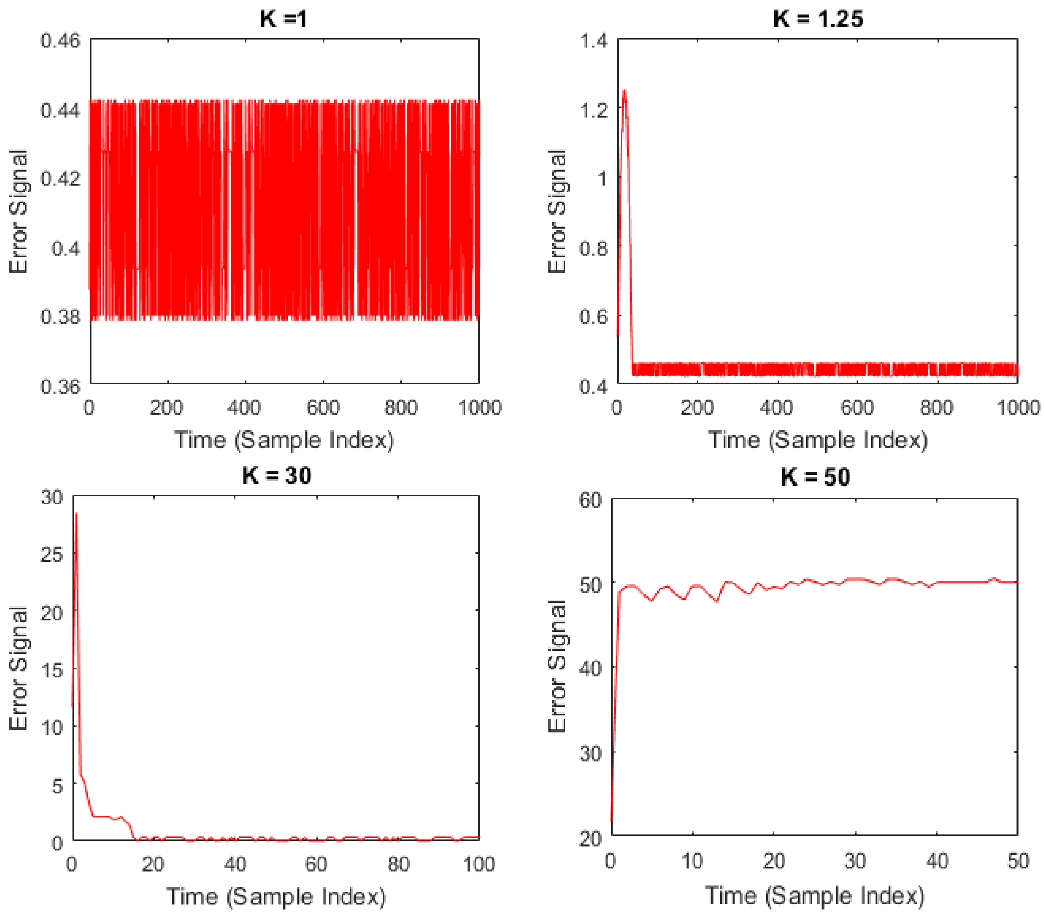

FIGURE 2.15: Error signal of early late gate for different step size.

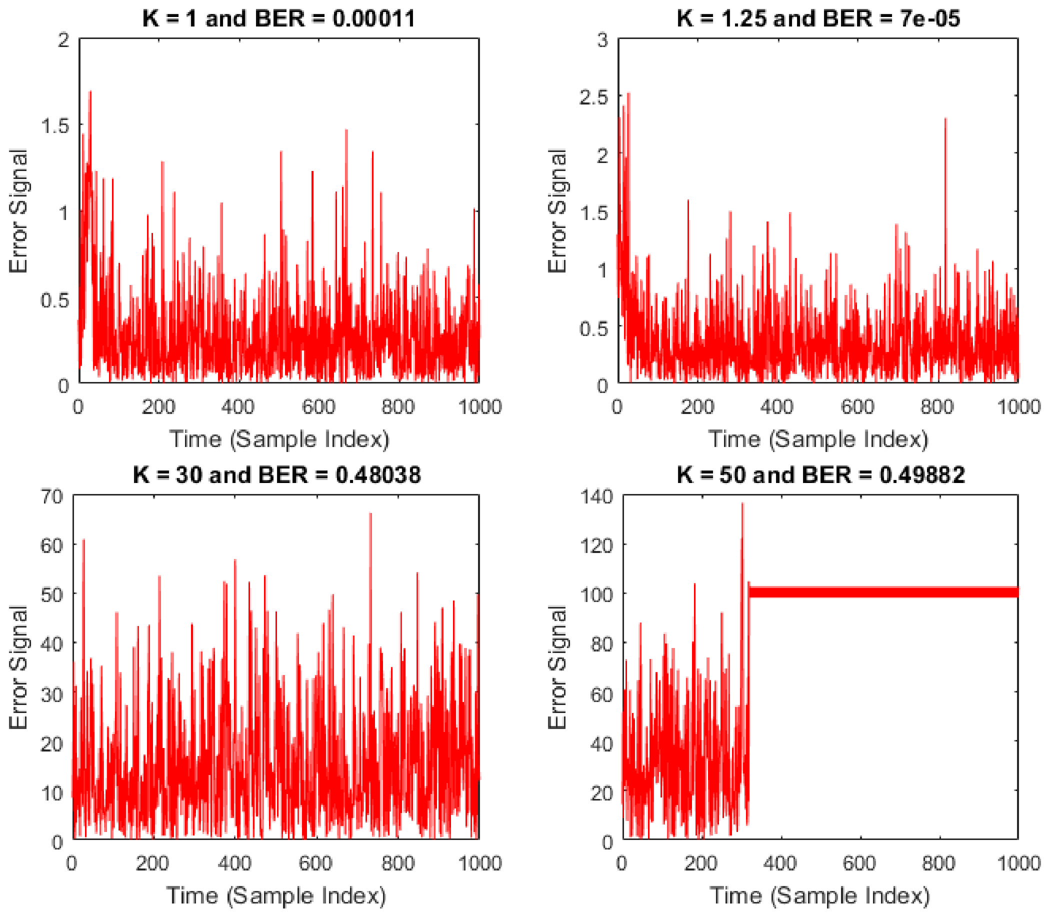

FIGURE 2.16: Error signal of early late gate for different step size in AWGN.

Figures 2.17 and 2.18 show the error signal curves for different values both in the presence and absence of noise respectively. The step size used is 1.25. In noise less case, one can observe that the error signal amplitude is convergent for between 0.25T and 0.35T. The same pattern resulted at 10dB.

- Complexity

  The early-late gate algorithm uses 3 samples per symbol period; it only considers two of the samples for computing error signal and the center sample for peforming decision, which inturn adds complexity. Early-late gate takes on average 1.3 ms to process 10,000 randomly generated data.

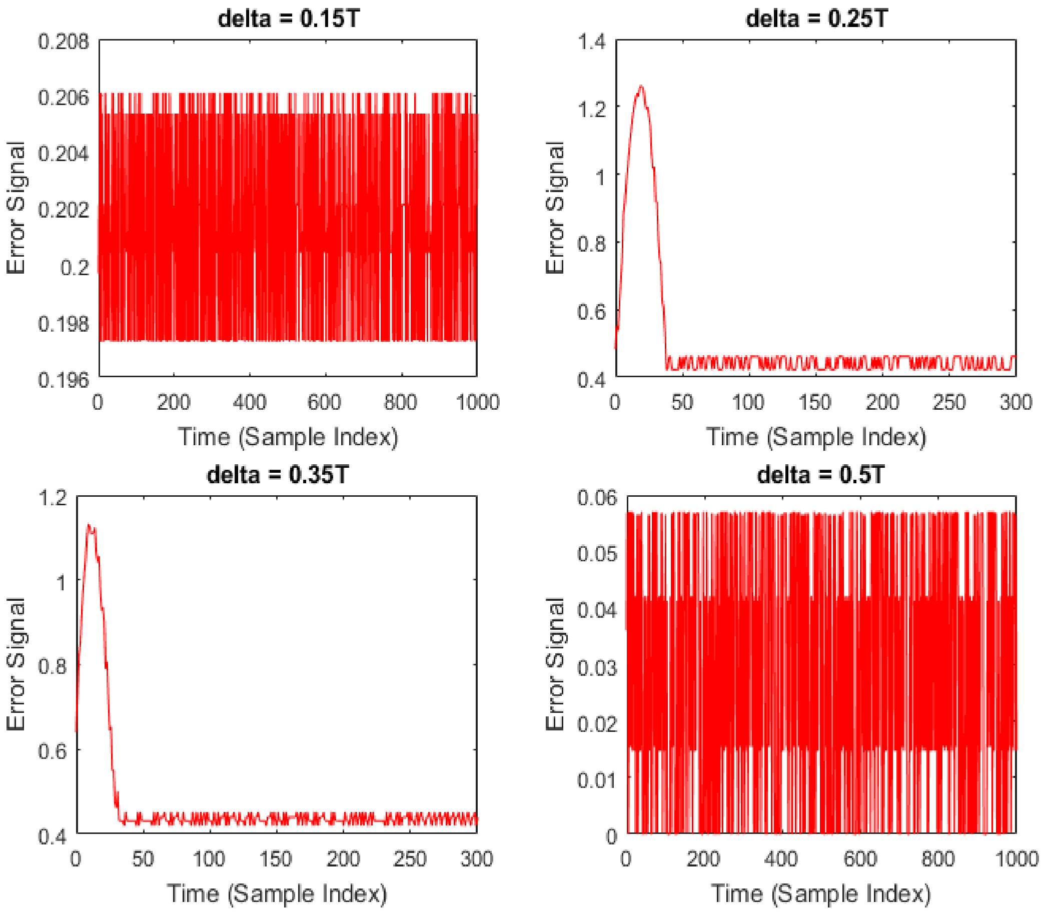

FIGURE 2.17: Error signal of early late gate for different timing phase deviation.

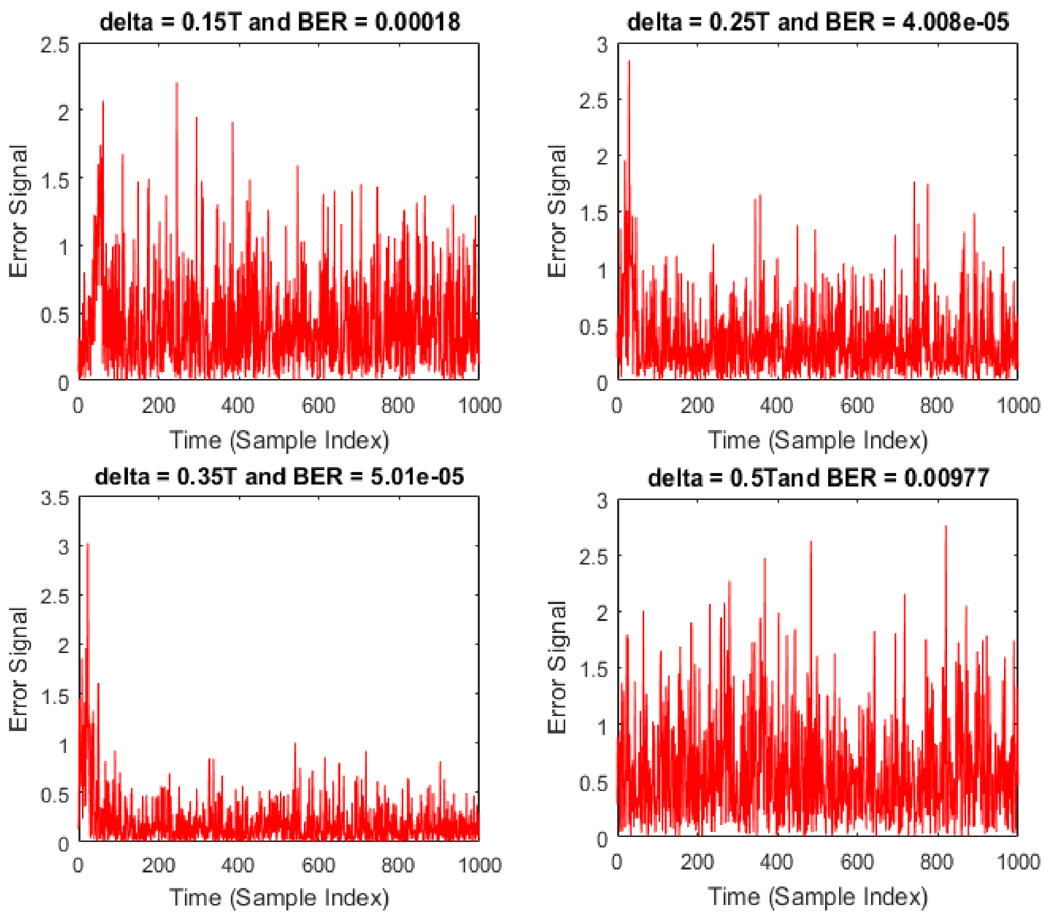

FIGURE 2.18: Error signal of early late gate for different timing phase deviation in AWGN.

2.3.2 Gradient Based Algorithm

The error signal described in Section 1.5.3.2 was implemented using MATLAB, and the following result was obtained. The parameter that varies in Equation(1.8) is K. Figures 2.19 and 2.20 show the error signal curves for different values of K both in the presence and absence of noise respectively. For a noiseless system, one can observe that the error signal amplitude converges for step size between 3 and 4. At 10dB lower values of K start converging. Being initially non-converging in the absence of noise, lower values of K has improved BER and stable performance in the presence of noise. This is illustrated in Figure 2.20.

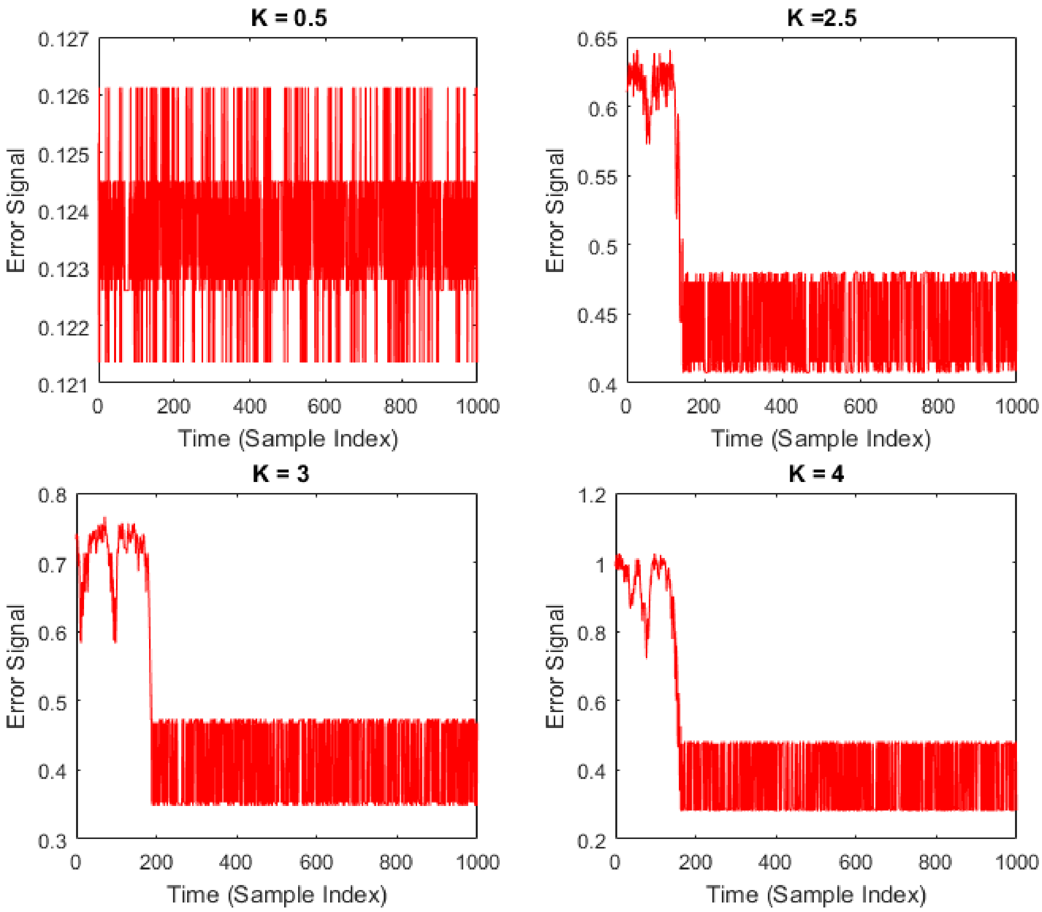

FIGURE 2.19: Error signal of gradient based for different step size.

- Complexity

  The gradient based algorithm uses 2 samples per symbol period for computing error signal; and considers the latest sample for performing decision, which minimizes the

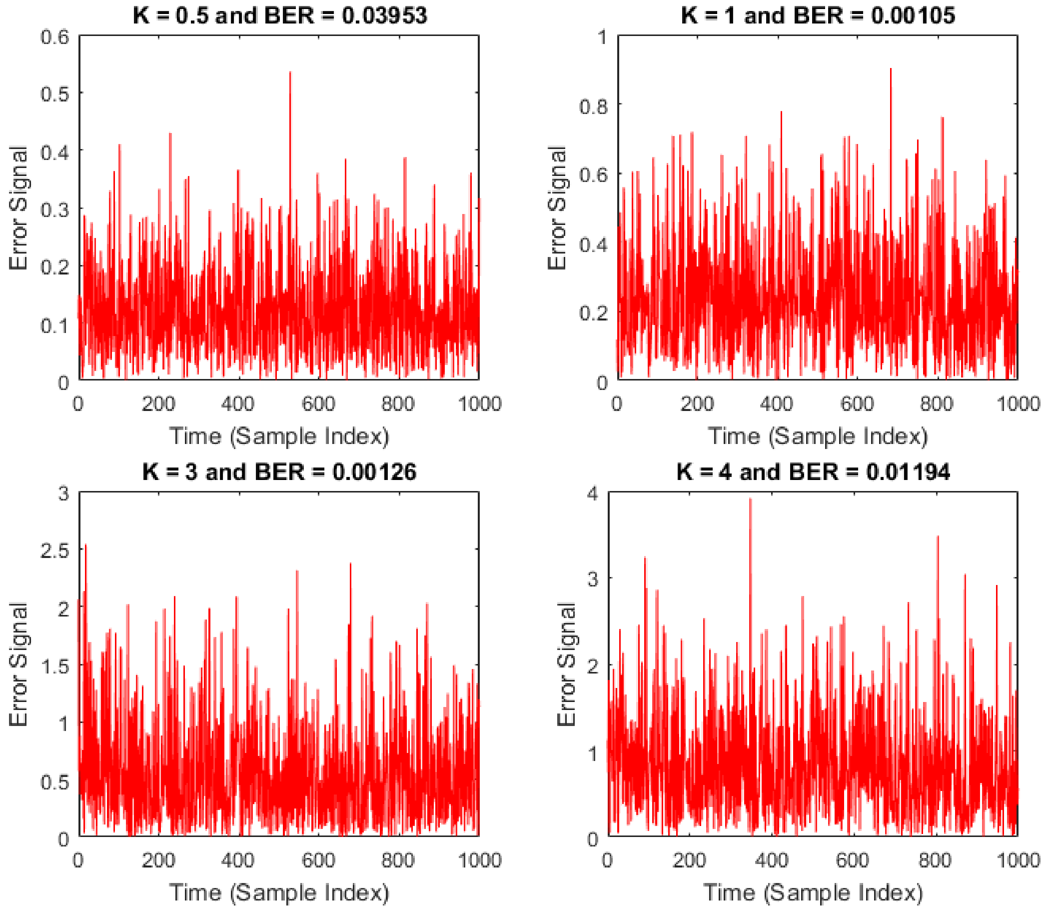

FIGURE 2.20: Error signal of gradient based for different step size in AWGN.

required processing time as compared to early-late gate. Gradient based algorithm takes on average 1.2 ms to process 10,000 randomly generated data.

2.4 Graphical User Interface (GUI)

A customised GUI has been developed using MAT LAB app designer (see Appendix A for description of App designer) for simulating an end-to-end digital communication system. The GUI is used for the simulation of audio and image files trasnmission through the digital communication entities described in the thesis.

The GUI further enables any user to design and test parameters belonging to specific blocks in the system. It abstracted the tiresome code development phase during the design of a digital communication system.

The application makes it easy for users and designers to view not only numerical results but also their graphical and visual depictions.

As shown in Figure 2.21, the main page gives the general components of the GUI with its objective.

FIGURE 2.21: Main tab of the Application

Figure 2.22 shows the design and transmission tab which provides:

- End-to-end simulation using a randomly generated data.
- Users a browsing and transmitting platform for audio and image files selected from the host computer.
- Easy configuration of blocks present in the transmitter side, channel, and receiver side.
- A schematic description of signal variation across each block.

Figure 2.23 shows the performance analysis tab where performance of specific algorithms in equalization, carrier recovery and symbol detection are analyzed based on randomly generated data.

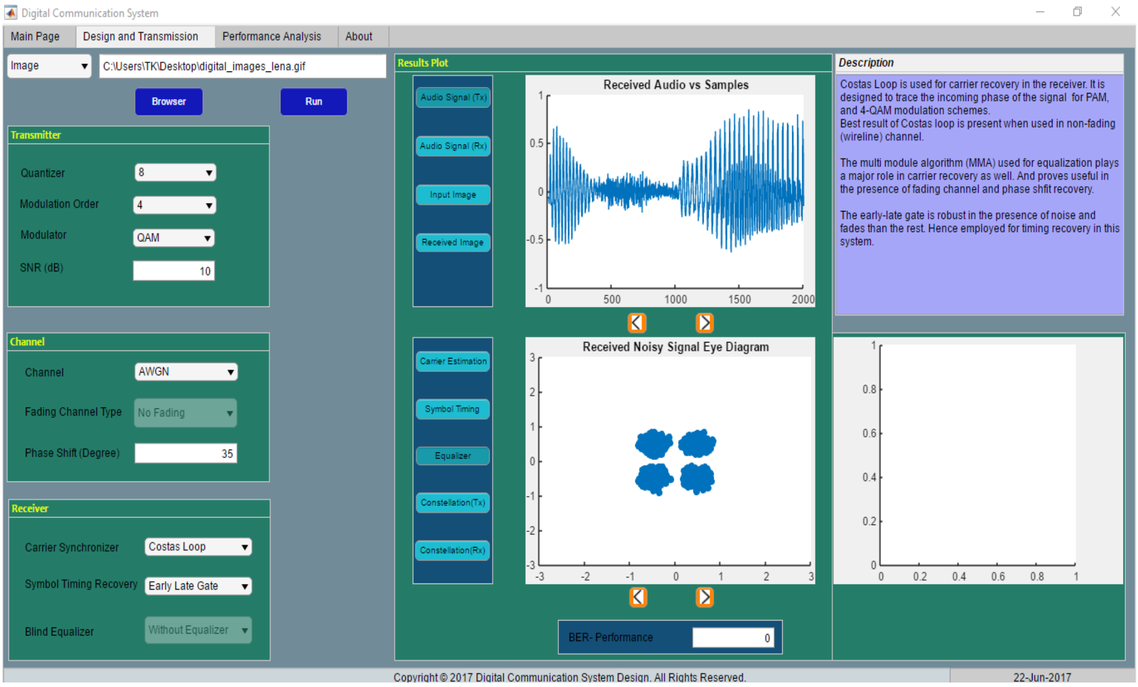

FIGURE 2.22: Design and Transmission Tab

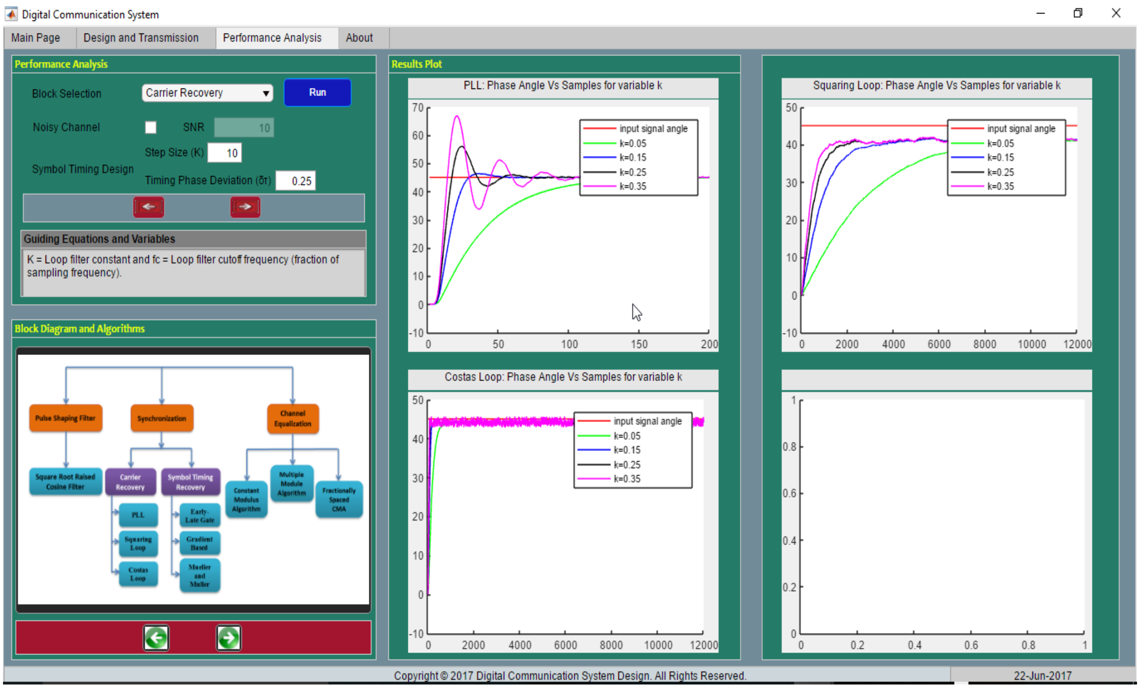

FIGURE 2.23: Performance Analysis Tab.

Chapter 3

CONCLUSIONS AND FUTURE WORKS

3.1 Conclusions

In this thesis, equalization and synchronization techniques in conjunction with symbol detection mechanism have been discussed. The analysis holds good for any constellation though the results presented are for BPSK and 4-QAM constellation. The equalization techniques are discussed when the signals are exposed to three channel models. CMA and MMA are nearly identical for all performance measuring parameters except the phase recovery. MMA has the ability to recover transmitted signal without the need for phase recovery loop. The carrier recovery techniques are compared in terms of their tracking capability by varying the loop filter constants and low pass filter cutoff frequency. Unlike PLL, Costas loop can track a modulated signal and exercises better accuracy than squaring loop. Finally, symbol timing recovery techniques are analyzed using their error equations and complexity. The early-late gate achieves good BER performance than the gradient based algorithm, but at the expense of increased complexity. Finally, the above functional blocks of a digital communication system have been incorporated to a customized GUI. The GUI simulates the transmission of audio and image files through the end to end digital communication system and enables performance analysis of the above specific functional blocks.

3.2 Future Works

The following points are recommended as a future work:

- Upgrading the GUI to include additional blocks of a digital communication, such as source coder and channel coder in order to make the application all-inclusive.
- Adopting more techniques for each blocks in the digital communication, such as higher modulation schemes and training based equalization techniques.
- Extending and integrating the thesis by introducing and incorporating related works from others.
- A schematic description of signal variation across each block.

Appendix A

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.

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.

A.1 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).

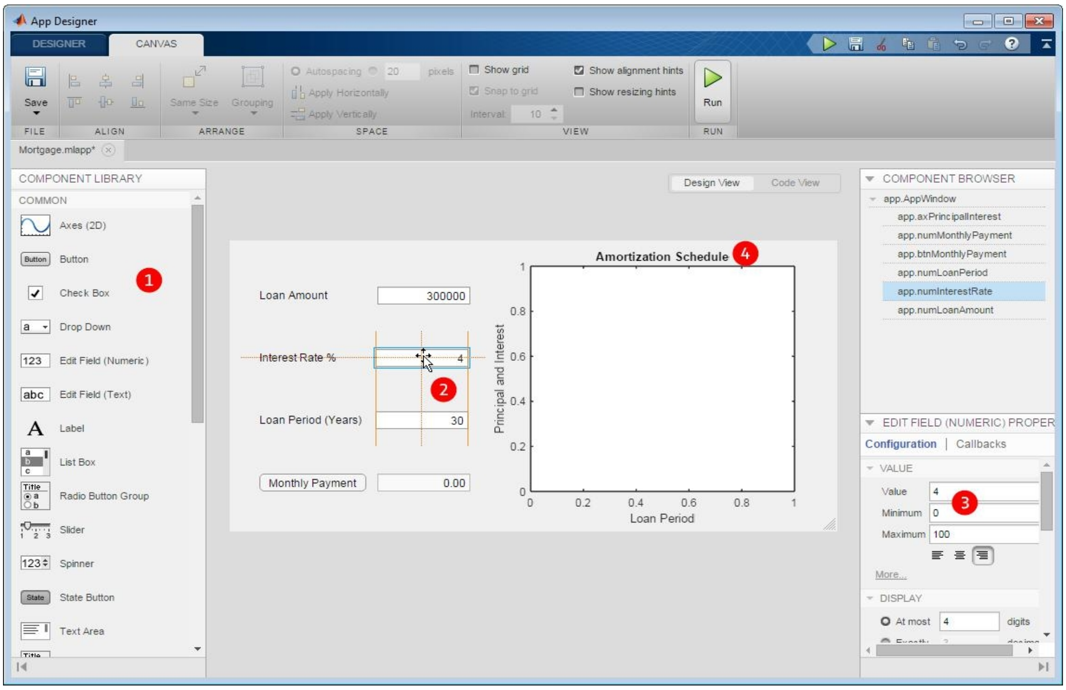

FIGURE A.1: App designer interactive design environment.

A.2 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).

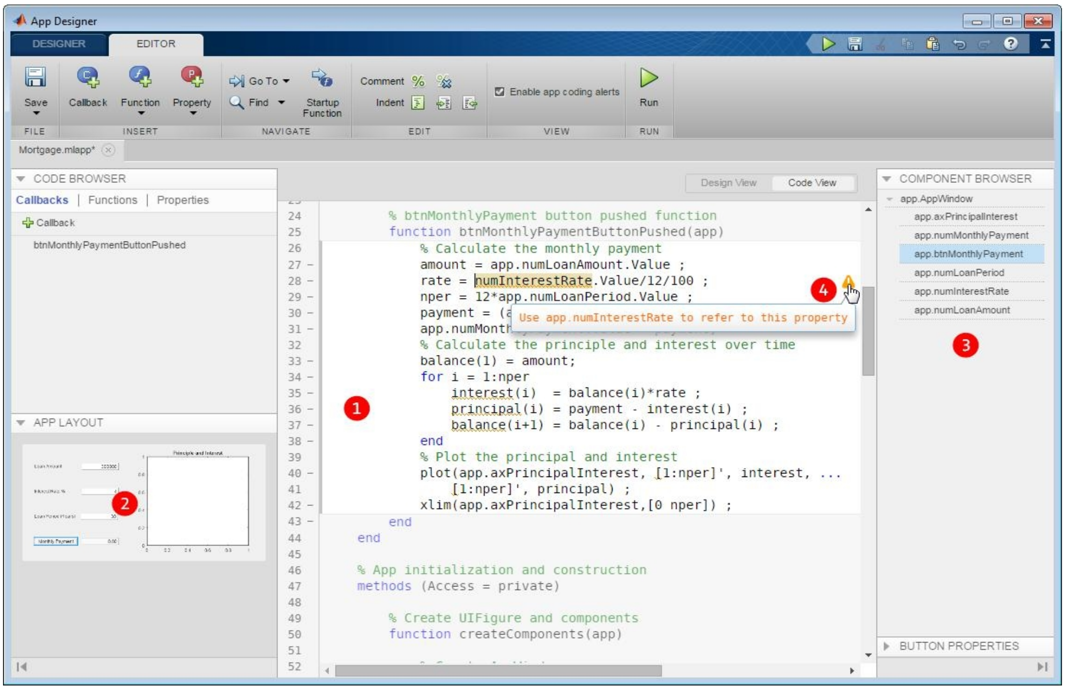

FIGURE A.2: App designer built-in editor integration.

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