DYNAMIC EVOLVING NEURAL FUZZY INFERENCE SYSTEM EQUALIZATION SCHEME IN MODE DIVISION MULTIPLEXING FOR OPTICAL FIBER TRANSMISSION

AWAB NOORI (817063)

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School of Computing, College of Art and Sciences

Universiti Utara Malaysia

Dynamic Evolving Neural Fuzzy Inference System Equalization Scheme in Mode Division Multiplexing for Optical Fiber Transmission

Prepared by

Awab Noori (817063)

Supervisors:

Mr.Suwannit Chareen Chit

Assoc. Prof . Dr . Angela Amphawan

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Abstrak

Prestasi pemultipleks pembahagian mod optik (MDM) dipengaruhi oleh gangguan antara simbol (ISI) oleh kecacatan saluran tidak linear yang timbul daripada gandingan mod peringkat lebih tinggi dan penyebaran modal dalam gentian multimod. Walau bagaimanapun, algoritma penyamaan MDM sedia ada hanya boleh mengurangkan kecacatan saluran linear dalam, tetapi algoritma penyamaan tersebut tidak boleh menangani kecacatan saluran tidak linear secara tepat dalam isyarat tersebut. Oleh itu, terdapat keperluan untuk meneroka cara bagaimana ISI boleh memulihkan isyarat yang dihantar. Kajian ini bertujuan untuk mengawal perluasan saluran tindak balas isyarat MDM dan mengurangkan kecatatan yang tidak diingini dalam saluran MMF dengan membentuk semula isyarat pada penerima. Skim penyamaan sistem pengembangan dinamik inferens neural kabur (DENFIS) telah digunakan untuk mencapai matlamat ini. Kajian ini dijalankan melalui beberapa langkah, bermula dengan membina model MDM dalam Optsim dan mengumpul data. Kemudian, parameter pembentukan semula isyarat telah ditentukan. Selepas itu, penyamaan DENFIS, penyamaan min kuasa dua terkecil (LMS) dan kuasa dua terkecil rekursif (RLS) dilaksanakan dan dinilai. Keputusan kajian menunjukkan bahawa skim penyamaan DENFIS tak linear boleh meningkatkan saluran tindak balas isyarat MDM yang diterima pada ketepatan yang lebih tinggi daripada skim persamaan linear sebelumnya. Penyamaan DENFIS menunjukkan ketepatan pembentukan semula isyarat yang lebih baik dengan purata ralat punca kuasa min (RMSE) 0.0338 dan mengatasi prestasi LMS linear dan skim penyamaan RLS dengan purata nilai RMSE yang tinggi iaitu masing-masing pada 0.101 dan 0.1914. Pengurangan nilai RMSE menunjukkan bahawa skim penyamaan DENFIS dapat mengurangkan ISI dalam saluran linear dengan lebih berkesan. Kesan ini mendahului kadar penghantaran data pada MDM. Selain itu, kejayaan pelaksanaan penyamaan DENFIS secara luar talian pada MDM menggalakkan pelaksanaan penyaman DENFIS dalam sistem optik terbenam secara atas talian pada masa akan datang.

Kata kunci: Pemultipleksan mod pembahagi, DENFIS, gangguan antara simbol, penyamaan, tak linear, gandingan mod

Abstract

The performance of optical mode division multiplexing (MDM) is affected by inter-symbol interference (ISI) from nonlinear channel impairments arising from higher-order mode coupling and modal dispersion in multimode fiber. However, the existing MDM equalization algorithms can only mitigate the linear distortion, but they cannot address nonlinear distortion in the signal accurately. Therefore, there is a need to explore how ISI can be mitigated to recover the transmitted signal. This research aims to control the broadening of the MDM signal and minimize the undesirable distortion among channels in MMF by signal reshaping at the receiver. A dynamic evolving neural fuzzy inference system (DENFIS) equalization scheme has been used to achieve this objective. This research was conducted through a few steps commencing with modelling the MDM system in Optsim and collecting the data. Then, the signal reshaping parameters were determined. After that, DENFIS equalization, least mean square (LMS) and recursive least squares (RLS) equalizations were implemented and evaluated. Results illustrated that nonlinear DENFIS equalization scheme can improve MDM signal at a higher accuracy than previous linear equalization schemes. DENFIS equalization demonstrates better signal reshaping accuracy with an average root mean square error (RMSE) of 0.0338 and outperformed linear LMS and RLS equalization schemes with high average RMSE values of 0.101 and 0.1914 respectively. The reduced RMSE implies that DENFIS equalization scheme mitigates ISI more effectively in a nonlinear channel. This effect can hasten data transmission rates in MDM. Moreover, the successful offline implementation of DENFIS equalization in MDM encourages future online implementation of DENFIS equalization in embedded optical systems.

Keywords: Mode division multiplexing, DENFIS, Inter-symbol interference, Equalization, Nonlinear, Mode coupling

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In the name of ALLAH, Most Gracious, Most Merciful. Praise be to ALLAH, The Lord of the worlds and blessing be upon all his Prophets and upon the last Prophet and messenger, Mohammed and upon his family and his companions.

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List of Abbreviations

MMF	Multimode Fiber
SMF	Single Mode Fiber
MDM	Mode Division Multiplexing
LAN	Local Area Network
MIMO	Multi-Input-Multi-Output
ISI	Inter Symbol Interference
ECOS	Evolving Connectionist Systems
FIR	Finite Impulse Response
IIR	Infinite Impulse Response
MLP	Multilayer Perceptron
SLM	Spatial Light Modulator
ANN	Artificial Neural Network
BER	Bit Error Rate
LMS	Least Mean Squares
RMSE	Root Mean Square Error
MD	Model Dispersion
CMA	Constant Modulus Algorithm
DENFIS	Dynamic Evolving Neural Fuzzy Inference System
RLS	Recursive Least Squares
RMSE	Root Mean Square Error
ECM	Evolving Clustering Method
FIS	Fuzzy Inference System
DSP	Digital Signal Processing

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CHAPTER ONE
INTRODUCTION

1.1 Introduction

Optical fiber has been used in telecommunication systems for many years due to its advantages, such as excellent transmission performance, high capacity, and no electromagnetic interference. Over the years, the demand for optical fiber has increased rapidly. Worldwide data traffic is growing at a rate estimated to exceed 50% annually [1, 2], spurred on by high-definition video streaming, multimedia file sharing, cloud computing, mobile networking, online gaming, and other information technologies over optical fiber backbones.

However, the continued growth of data traffic will congest the optical fiber systems and cause them to reach its capacity limit in the next couple of decades. This has triggered the need for multiplexing techniques to enhance the performance of fiber optic system. Many multiplexing techniques have been explored and have already been implemented in optical systems to increase the capacity. However, some of them have already reached the peak of their performance [3].

The purpose of this research is to develop an equalization scheme for compensating and recover the received signal in mode division multiplexing over multimode fibre. Furthermore, in this research the expression of channel impulse response is to describe received signal.

This chapter presents an overview of this study, with a background and important terminologies used in it. This chapter begins by providing a brief outline of fiber optics, followed by the concept of equalization, the essential focus of this research, key principles on the channel impulse response and neural-fuzzy systems. The research background is presented in Section 1.2, the research motivation in Section 1.3, research problem in Section 1.4, research questions and research objectives presented in Sections 1.5 and 1.6 respectively. Then, the research scope is discussed in Section 1.7. Finally, the significance of the research is outlined in Section 1.8.

1.2 Research Background

Fiber optics can support active traffic growth in our communication and information society. However, dispersion in fiber optic systems is a phenomenon that causes optical pulses to broaden as they propagate through the fiber; thus giving rise to inter-symbol interference (ISI). As data rates become higher and link lengths are stretched to meet the needs of modern fiber optic communications, it is becoming an increasing problem. Consequently, compensating effectively for dispersion is becoming a more and more important issue [4].

This research aims at proposing an equalization scheme, for a system with a specific end goal, to relieve the noise and dispersion represented by ISI and the channel impulse response of the received signal at the receiver side of the mode division multiplexing system. This is performed to increase the data transfer capacity and mitigate inter-symbol interference. Moreover, to create the development for this research, several concepts must be defined.

The optical fiber has two main types of fiber optics, namely single-mode optical fiber and multi-mode optical fiber. Single Mode Fiber (SMF) provides the high capacity to transmit information because of its ability to maintain the transmitted signal over long distances. It is used mostly in long-haul networks. However, SMF will reach its capacity limits [5]. Meanwhile, Multi-Mode Fiber (MMF) is a type of optical fiber made with a bigger core diameter than SMF. Because of its large core diameter, many pulses will propagate inside the core, and consequently face many types of noise and dispersion to the network. It is used mostly in Local Area Networks [4].

Mode division multiplexing in a multimode optical fiber communication system can be an excellent choice to increase capacity in current and future optical fiber communication systems. Mode division multiplexing (MDM) is an appealing technique for expanding the limits of optical fiber links using light modes that propagate inside multimode fiber [4]. Despite its advantages, MDM has some obstacles that downgrade its performance and increase the distortion of transmitted signal. Mode coupling and mode dispersion which lead to inter-symbol interference are the main obstacles in MDM (reviewed in detail in Chapter Two).

Equalization is the method of compensating and reducing the distortion introduced by the transmission medium in communication systems. In other words, equalization refers to any signal processing operation that minimizes the distortion. The goal of an equalizer is to provide error-free communications by ensuring that signals transmitted over a link will recover back at the receiver like the original signals [6].

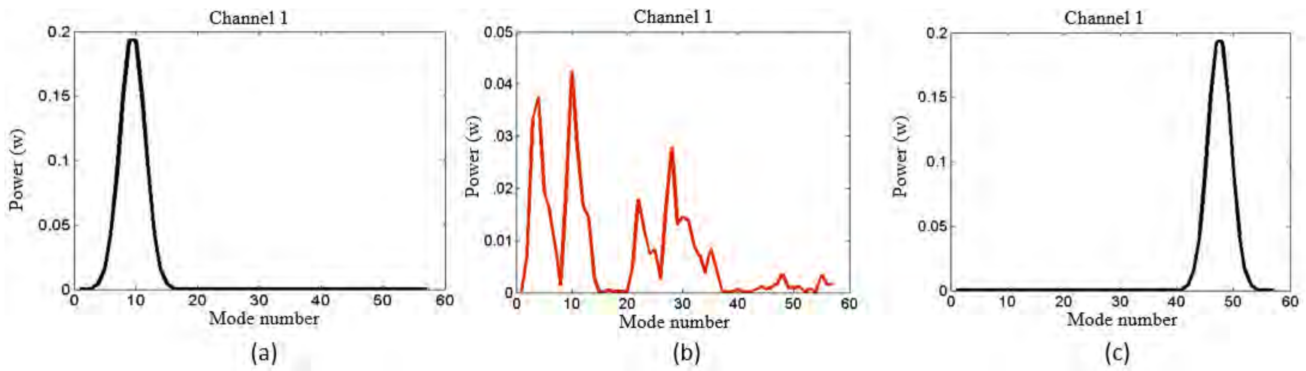


Figure 1.1: Channel impulse response before and after equalization

Figure 1.1 describes the transmitted signal and its status when it is in the transmitter as shown in (a), and the distortion that will occur to it during the propagation inside the link as illustrated in (b). As a result, the signal which reached the receiver will be distorted, and its status will be different compared with the transmitted signal. Equalization techniques provide a solution to recover that distortion on the receiver side from being close to the transmitted signal or the original signal. Hence, figure 1.1 illustrates the received signal before the equalization as shown in (b) and after applying the equalization process as figured in (c).

Moreover, equalization is one of the several important signal processing functions in mode division multiplexing systems [1], because it does not make a real effect on the system's architecture. In other words, equalization does not need to reconstruct the infrastructure of the target systems [7]. Therefore, adaptive equalization, adaptive algorithm, and adaptive filter are, as a rule, broadly considered for the purpose of distortion compensation, because channel equalization is an important subsystem in a communication receiver [8]. In fact, this is the principal reason for this research and it is discussed more in the second chapter.

1.3 Research Motivation

Internet traffic is dramatically growing and will keep increasing in the future [9]. Fiber optic communications continue to evolve due to the capacity demand [4]. However, the channel transmission capacity will be reached [10, 11]. According to Cisco's annual report, broadband speeds will double by 2019 [12]. Figure 1.2 illustrates that there will be 24 billion devices in 2019; up from 14 billion in 2014, and IP traffic will surpass the Zettabyte (1000 Exabyte) threshold in 2016, and the two Zettabyte threshold in 2019 [12].

These facts emphasize the demand for increasing the capacity of optical fiber systems. Therefore, to increase system capacity, new optical transmission models are required [13, 14]. Mode division multiplexing in multimode fiber has turned out to be a viable alternative technology to increase the capacity of optical fiber networks [15] and is expected to fulfill the communication demands that single mode fiber cannot handle due to their capacity limitations [1, 16].

Furthermore, many multiplexing schemes have been designed attempting to compensate for the spread in received signal or channel impulse response and the impact of the distortion. Devices like mode couplers [17], spatial photodetectors [4], spatial light modulation [18], photonic crystal lanterns [19], and several equalization techniques [7] have been proposed. However, ISI still exists due to mode coupling [4, 5, 20]. This research proposes DENFIS equalization scheme to tackle this limitation and mitigate the effect of ISI on the received signal in the mode division multiplexer system over multimode fiber.

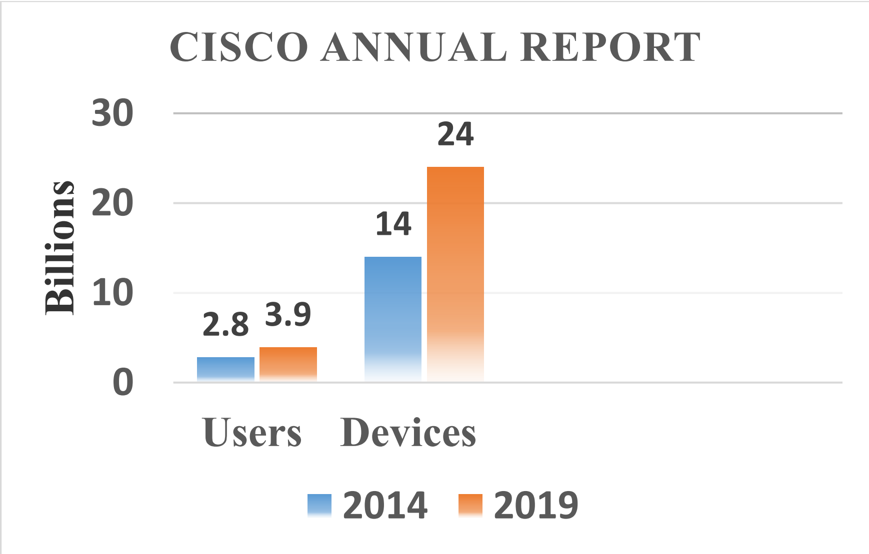


Figure 1.2: Cisco annual report for future traffic demands [12]

1.4 Research Problem

Fiber optics is considered to be the main medium in communication systems that forms the backbone of the Internet [1] because of its advantages such as speed and high capacity. However, fiber optic transmission capacity is close to its maximum performance limit [5], especially with the increasing demand for cloud computing, social media, and big data applications. Therefore, improving the performance of fiber optics is an important issue to transfer a huge amount of data to comply with the global demands for the internet. Multiplexing techniques are used to provide a better performance of fiber optic system. Many of multiplexing techniques have been implemented in optical systems to increase the capacity. Nevertheless, most of them have already reached the peak of their limits of performance [3].

An emerging multiplexing technique called mode division multiplexing is a promising technique to improve the limitation of fiber and increase the capacity of data transmission. However, mode coupling and modal dispersion which cause inter-symbol interference [1] are inevitable physical issues of mode division multiplexing [4, 20] that arises because of the imperfection of manufacturing multimode fiber. This kind of distortion in mode division multiplexing is considered as non-linear distortion [21]. Some modes tend to exchange their powers randomly through spreading inside the fiber causing inter-symbol interference between modes [22, 23]. As a result, the signals will overlap each other. At the receiver side, it will be difficult to decode or recover the transmitted signal correctly. Consequently, the error rate increases and the data rate decreases, and this causes a huge loss of data throughput.

To mitigate the unwanted impact of ISI, equalization will filter the received signal to improve the channel impulse response and mitigate the effect of noise [1]. Furthermore, current equalization techniques in mode division multiplexing are based on Recursive Least Squares (RLS) and Least Mean Squares (LMS) algorithms. However, these algorithms can solve the linear distortion, but cannot deal with nonlinear distortion. Accordingly, this research proposes an equalization scheme based on Dynamic Evolving Neural-Fuzzy Inference System (DENFIS) [24] to mitigate the nonlinear distortion represented by Inter-symbol interference due to mode coupling.

1.5 Research Question

The main research question is:

How to mitigate ISI in the received channel impulse response of a mode division multiplexing system in the presence of mode coupling in order to recover the transmitted channel impulse response?

1.6 Research Objectives

The goal of this research is to develop an equalization scheme using neural fuzzy systems to mitigate the effect of ISI in the channel impulse response of the received signal in mode division multiplexing over multimode fiber. The research objectives are represented as follows:

1. To determine a pulse shaping filter parameters at the receiver of an MDM system based on the number of channels and number of excited modes at the receiver so as to mitigate ISI.

2. To simulate DENFIS equalization scheme to recover the transmitted channel impulse response in MDM system from the distorted channel impulse response based on the designed pulse shaping filter.

3. To evaluate DENFIS equalization scheme between response channel impulse response and the pulse shaping filter.

1.7 Research Scope

This research aims to develop an equalization scheme for mode division multiplexing over the multimode fiber based on neural-fuzzy systems, more specifically based on the proposed DENFIS equalization scheme as shown in Figure 1.3. Furthermore, DENFIS equalization scheme is located on the receiver side of mode division multiplexer system.

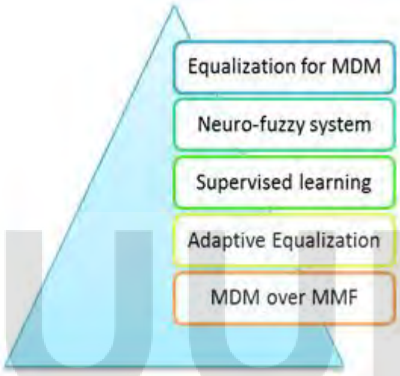


Figure 1.3: Research Scope

DENFIS equalization is performed to improve the overall performance, improve the channel impulse response of the received signal, reduce the error rate in the received signal and minimize the ISI of the received signal at the receiver side of the mode division multiplexing system. Moreover, the power coefficient of the modes collected from the simulation is used in this study as input data that were utilized in equalization scheme. Also, the simulation has been employed because of the lack of equipment to conduct a real experiment for extracting the data. As a result, offline data were employed in this research to carry out DENFIS equalization scheme.

1.8 Significance of Research

This research aims to mitigate one of the major obstacles in the way of achieving a strong mode division multiplexing system over multimode fiber in the fiber optic communication system. Therefore, this can be done by developing an equalization scheme. The significance of solving this problem is to decrease the distortion in the received signal of mode division multiplexing over multimode fiber. Moreover, reducing the mode coupling and ISI will lead to the adapting of mode division multiplexing over multimode fiber for more applications in the future.

CHAPTER TWO
LITERATURE REVIEW

2.1 Introduction

This chapter reviewed studies on the fiber-optic communication system besides some challenges that are considered as obstacles in this system. This chapter also illustrated the equalization techniques as a proposed solution for those challenges. Equalization techniques discussed in detail in this chapter as they represent the focus of this study. Section 2.2 provided a general overview of fiber optic communication system. Section 2.3 illustrated the concept of mode division multiplexing (MDM) so as to comprehend its characteristics. The main challenges in the fiber optic system are presented in Section 2.4. Subsequently, Section 2.5 presented an overview on pulse shaping filters. Then, Sections 2.6 and 2.7 provided a detailed description of equalization techniques, adaptive equalization, adaptive algorithms and adaptive filters. A review of previous equalization schemes for mode division multiplexing system was provided in Section 2.8 to understand what kind of equalizations have been implemented in the mode division multiplexing system. Additionally, previous equalization schemes in the radio systems were presented in Section 2.9. This section was followed by an overview of a neural fuzzy system in Section 2.10. Section 2.11 illustrated the algorithm utilized in this research in detail. Then, Section 2.12 presented the motivation of using the neural fuzzy system. Finally, a summary was provided at the end of this chapter.

2.2 Fiber Optic Communication System

Fiber optics has been used in telecommunication systems for years. Fiber optics is a technology to transmit information from the sender to the receiver by sending pulses of light. Light is used to transform information. It has many advantages such as speed, high capacity, and free of electromagnetic interferences [4].

Multimode fiber is a kind of optical fiber made with a bigger core diameter compared with single-mode fiber. With a casual tolerance for alignment, they were utilized as a part of the expense in data centers and local area networks. Because of this large core diameter, a pulse of light as an input will energize numerous modes to propagate inside the fiber. Typically, these modes touch the output point with various speeds affected by modal dispersion. Modal dispersion in multimode fiber causes ISI [1], which is the restricting factor in this sort of optical networks. Because of this, multimode fiber has been utilized for local area applications [13].

Recently, experts realized that single mode fiber is rapidly approaching its limit for transmission capacity [5]. As a result, they turn to use mode division multiplexing system over multimode fiber as an alternative to exceed the single mode fiber limitation. Increasing fiber capacity can be achieved by increasing spatial dimensionality using multimode fiber [25]. However, multimode fiber has some limitations that degrade its performance.

2.3 Mode Division Multiplexing

Mode division multiplexing is a technique used for expanding the limit of optical fiber links by using light modes that propagate inside a multimode fiber [4]. Hypothetically, light modes are orthogonal to one another. Hence, it is possible to consider each mode as an independent channel [1]. Long distance spread of optical modes in multimode fiber causes mode coupling between the diverse modes, prompting crosstalk and interference [22]. The signal processing system can be utilized to discrete the coupled modes or channels [1]. Figure 2.1 demonstrates the overview of mode division multiplexing over multimode fiber. In mode division multiplexing, each mode is considered as an independent channel transmitting separate signals, as shown in Figure 2.1; each signal is mapped to a specific mode and a multiplexer at the transmitter couples, and all signals are in a single transmission fiber. The opposite operation occurs on the receiver.

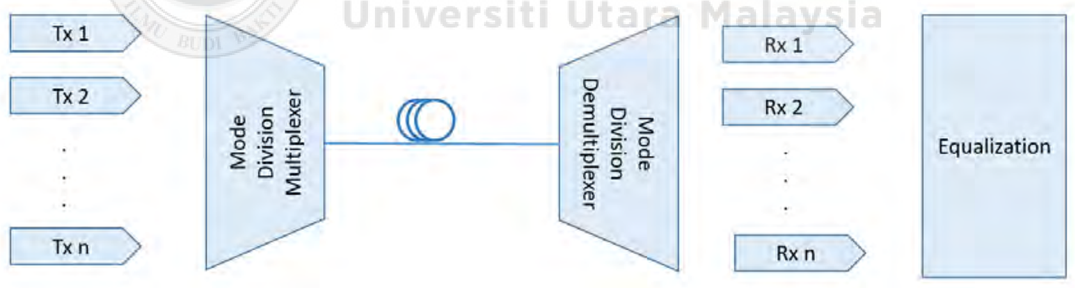


Figure 2.1: Mode division multiplexing basic block diagram [1]

2.4 Obstacles of Mode Division Multiplexing in MMF

The current communication systems transmit high-speed data via the communication channels. During this process, the transmitted signals are corrupted due to some distortions [26], such as mode coupling, modal dispersion and inter-symbol interference (ISI).

2.4.1 Mode Coupling

Mode coupling is an inescapable physical issue that emerges because of multimode fiber industrial malformations [4, 20], such as fiber manufacturing twisting and micro- and macro-bending. Therefore, some modes tend to exchange their power randomly through their spreading inside the fiber. For example, three modes are propagating inside the link as can be seen in Figure 2.2. During the propagation, the three modes will clash with each other. The result of this clash is that the modes split to many modes with less power causing distortions among modes conveyed by these specific modes which decrease the distance capacity of the system and produce a higher error rate [22, 23]. Compensating the channel distortion calls for channel equalization techniques at the receiver side to reconstructing the transmitted signal correctly.

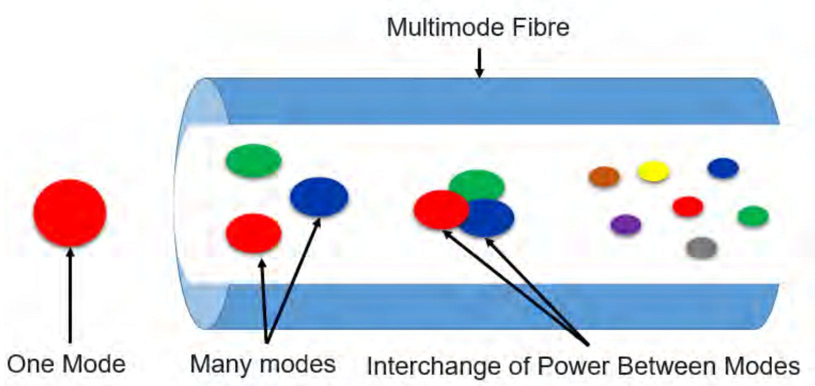


Figure 2.2: Mode coupling concept

2.4.2 Modal Dispersion and Inter-Symbol Interference

The optical power from an optical pulse launched into the multimode fiber is distributed over all of the guided modes. These modes then propagate along the fiber to the receiving end. In practical systems, there are often interchanges of power among the modes (mode coupling). The problem with multimode fiber is that the propagation speed of each one of the guided modes is different. This means that at the receiving end, the received signal or pulse is spread out over time resulting in pulse dispersion. The pulse spread becomes larger if the fiber length increases. This type of dispersion is known as modal dispersion [3], as shown in Figure 2.3. This dispersion is the main limitation of signals transmission on multimode fiber [1].

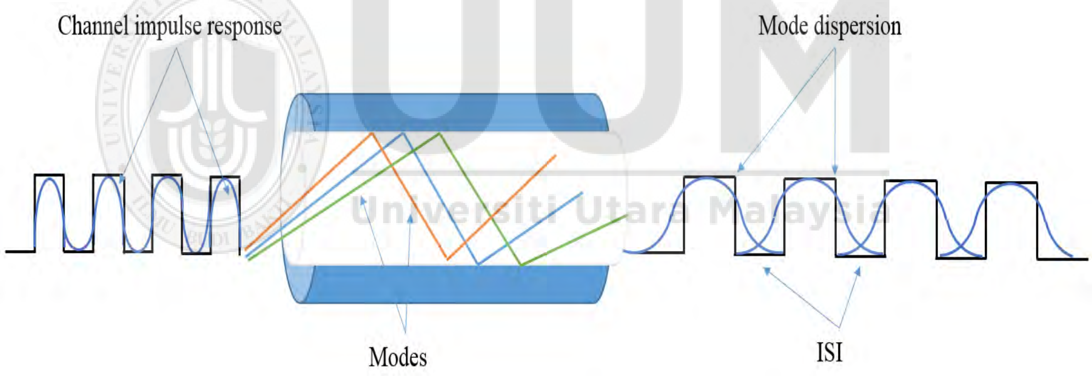


Figure 2.3: Modal dispersion and inter-symbol interference [4]

However, inter-symbol interference (ISI) arises when the signal or pulse for each mode is placed into next time slots [1]. The pulses would have rounded tops instead of flat ones with a restricted bandwidth, as in Figure 2.3 which shows the overlapping between the signals. At this point, ISI occurs due to the distortion effect of the pulse. This effect will cause unwanted contributions from the adjacent pulses that are likely to degrade performance and raise the error rate.

The multiple modes in multimode fiber are considered as a negative effect on the communication link due to the mode coupling, modal dispersion, and ISI (which is explained above). On the other hand, the multiple modes are now seen as a way to increase the capacity of multimode fiber through mode division multiplexing. Mode division multiplexing in multimode fiber is a promising method to increase fiber optic capacity [22].

2.5 Pulse Shaping Filters

The procedure of changing the waveform of transmitted impulses is known as Pulse shaping filtering. Deterring a pulse shaping filter is the first objective of this research. Before reviewing the pulse shaping filter, it is important to understand the basic functioning of the transmission and receiving channel. In any data transmission system, the transmitter sends and transmits the pulses, whereas the receiver detects and receives the pulses. The primary objective at the receiving end is to sample the received signal in a manner such that it enhances the probability of better puls or signal.

The primary purpose of Digital Signal Processing (DSP) is transformation, convolution, and correlation of signals or pulses. Digital filtering is one of the most powerful tools of signal processing. It is eliminating the errors and improving the performance. Pulse shaping filters help different algorithms to keep the signal within allotted bandwidth, maximize the rate of data transmission and reduce transmission errors. The main function of pulse shaping filter is to minimize inter-symbol interference (ISI) to get a bit error rate as low as possible.

These filters are used with the adaptive algorithms. These filters work as a reference and guide the algorithm to get the ideal signal by minimizing the error value [27]. There are many types of pulse shape filters. Sinc filter, Raised Cosine Pulse and Gaussian Pulse filters are the most popular types [28]. As shown in Figure 2.4, the Sinc filter is usually used in the frequency domain as its frequency response is rectangular.

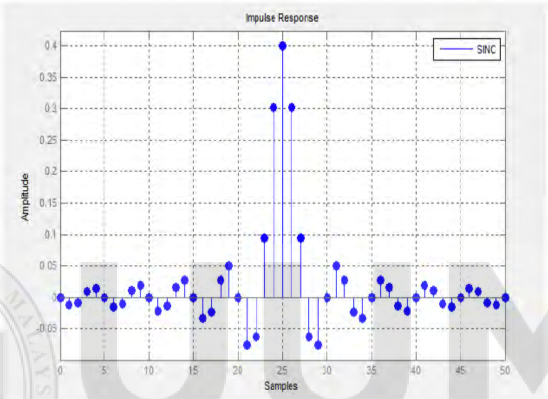


Figure 2.4: Impulse response of Sinc filter [29]

The Raised Cosine filters are requisite to avoid inter-symbol interference and limit the amount of bandwidth necessary for transmission. Raised Cosine is a suitable filter for pulse shaping as its transition band is shaped similar to a cosine curve. Figure 2.5 describes the raised cosine shape. However, this filter works with both the positive and negative values. Raised Cosine filter is used in electronics, such as cellular phones and wireless devices, to increase speed and power consumption [29]. Furthermore, Jafar [30] used Raised Cosine filter to equalize the channel impulse response by using Genetic Algorithm. After that, he compared his outcome with both LMS and RLS algorithms.

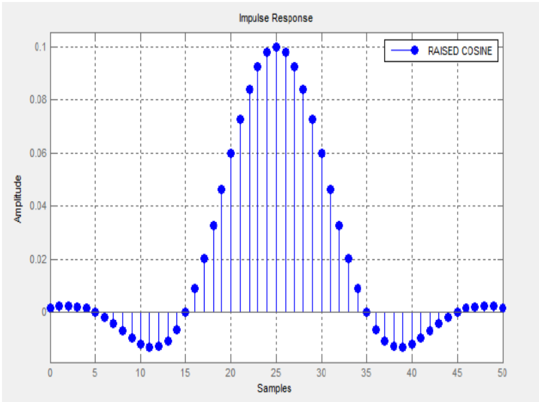


Figure 2.5: Impulse response of Raised Cosine filter [29]

In signal processing, a Gaussian filter has an impulse response as a Gaussian function [28]. Gaussian filters are considered to have the minimum group delay and to provide no overshoot to a step function input at the time of reducing the rise and fall, as shown in Figure 2.6.

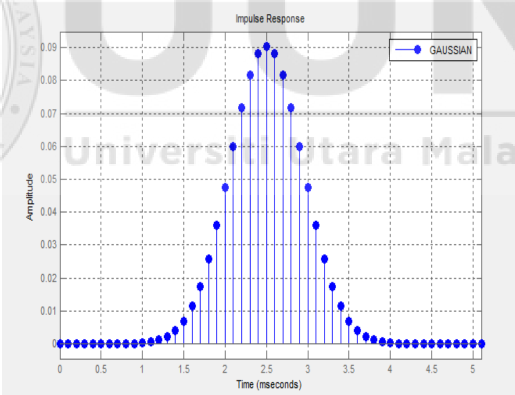


Figure 2.6: Impulse response of Gaussian filter [28]

Moreover, Gaussian filter is used in GSM and image processing. According to a comparison among the three filters conducted by Priyanka Agrawal [29], Gaussian filter gives the best results and better performance than the other two filters [28, 29]. Moreover, many researchers who are working with optic equalization have used

Gaussian pulse filter in their studies [31, 32]. This concept has been used to compensate the dispersion and maximize the delay spread in MDM [31]. Lawan [33] used the Gaussian pulse to study the effects of chromatic dispersion and attenuation in SMF. Moreover, Ladislav [34] used Gaussian pulse filter to reduce chromatic dispersion by operating two simulations in OptSim software.

2.6 Equalization

Equalization is the method of compensating and reducing the distortion introduced by the transmission in fiber communication systems. In other words, equalization refers to any signal processing operations that minimize distortion and ISI. The goal of equalization is to provide error-free communications by ensuring that transmitted signals over the link are recovered at the receiver side as the original signals [6].

Channel characteristics have a significant role in causing distortions, and the response of the channel is time-variant, that is channel characteristics are not known before. The time-variant obligates the equalizers to be designed to adjust themselves to the channel response and to adapt themselves to the variations of time in the reaction of the channel to compensate for the channel changes. Such equalizers are called adaptive equalizers, and they have received a considerable attention because of their superior features [35]. In this research, equalization process must be adaptive equalization due to the reasons mentioned above.

Generally, equalization techniques can be classified based on type, structure and algorithm. As shown in Figure 2.7, classification based on type can be divided into linear equalizers and non-linear equalizers [36]. This classification is based on how the output of an adaptive equalizer used for subsequent control of the equalizer. If an adaptive equalizer has a feedback loop, then it is considered as a non-linear equalizer; whereas if it only has a forward loop and there is no feedback, then it is a linear equalizer. So, the basic difference between linear and non-linear equalization is that in linear equalization, the output signal is not used in a feedback path to adapt the equalizer. On the contrary in nonlinear, the output signal is feedback to change the following outputs of the equalizer.

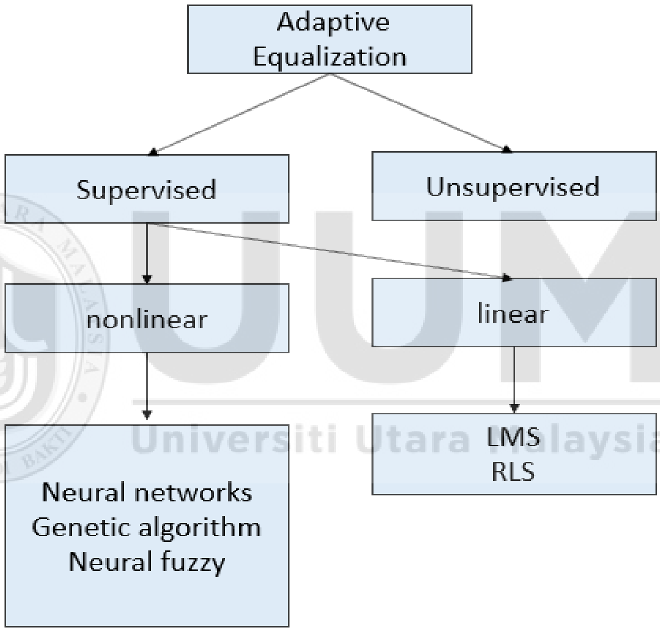


Figure 2.7: Adaptive equalization techniques

Non-linear equalization can be classified based on structure into non-adaptive and adaptive (as mentioned and discussed above). Classification is based on algorithms, such as adaptive algorithms (like LMS and RLS), genetic algorithms and fuzzy algorithms, which are kind of algorithms that behave depending on the available information that may be received and this information may be a history of data or other data resources. In this research, the proposed algorithm dealt with a Gaussian pulse shaping filter and distorted channel impulse response.

2.7 Adaptive Equalization

The signal on the receiver side *y(n)* is different from the original signal *x(n)* because it was affected by the distortion that occurs by channel transfer function *C(z)*. To get the original signal *x(n)*, *y(n)* must be processed by using an equalizer *W(z)* to compensate the distortion, as shown in Figure 2.8. In practice, the channel is unknown and is time varying due to the nature of transmission medium; thus, adaptive equalization is needed to provide an exact compensation over a time-varying channel. The adaptive filter needs the Gaussian pulse shaping filter *d(n)* to compute the error signal *e(n)* for adaptive algorithm. The two operating modes of adaptive equalizers are training mode and tracking mode.

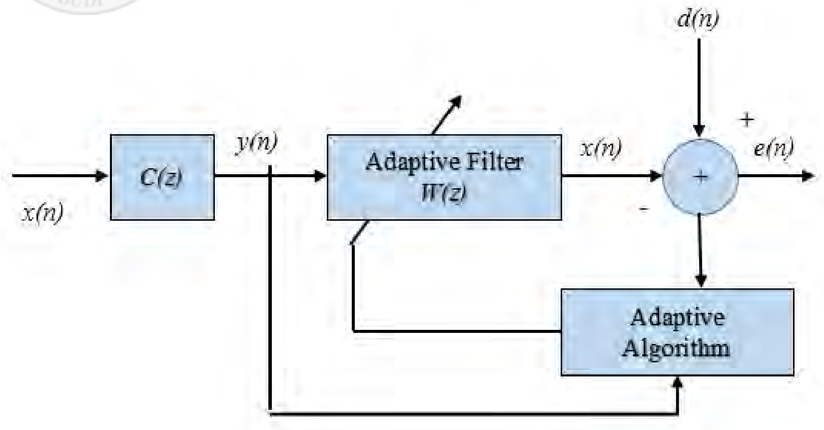


Figure 2.8: Adaptive equalizer [37]

2.7.1 Training and Tracking Modes in Adaptive Equalization

Training mode is one of the processes that are involved in adaptive equalization. Training mode starts when the transmitter sends a known fixed length Gaussian pulse shaping filter so that the receiver's equalizer may average to a proper setting. The Gaussian pulse shaping filter is a fixed known prescribed bit pattern usually put in the standard. In this study, the Gaussian pulse shaping filter was represented as a Gaussian function (Gaussian pulse) shape to show the channel impulse response of the Gaussian pulse shaping filter. After that, normalization process was implemented to normalize the Gaussian pulse shaping filter to be the same pattern or in the same range with the distorted channel impulse response [38].

The distorted channel impulse response was collected from the simulation tool. Immediately after milliseconds following the Gaussian pulse shaping filter, the user data, which is the distorted channel impulse response, were sent. The Gaussian pulse shaping filter is designed to permit the receiver to acquire the proper filter coefficients in the worst possible channel conditions, like maximum delay and maximum ISI. Hence, when the training mode finished, the filter coefficients are a near-optimal signal which is the Gaussian pulse shaping filter. The proposed algorithm at the receiver evaluated the channel and estimated filter coefficients to compensate the errors which occur during the transmission of the channels. Therefore, it has to be a recursive method that finally converges to the exact Gaussian pulse shaping filter or close to the Gaussian pulse shaping filter [37].

When the distorted channel impulse response received, the proposed algorithm tracked the changing channel and continuously changes of the filter characteristics over time.

These changes can be made by comparing each value of Gaussian pulse (Gaussian pulse shaping filter) with the value of distorted pulse (actual signal). If they are equal or almost equal, that means the error is almost zero so these values will stay in the same position. Otherwise, the distorted channel impulse response will pass again through the tracking process, and the comparison will start again. This process will happen with many iterations until the distorted channel impulse response fits the Gaussian pulse shaping filter, and this is the principle of tracking mode.

2.7.2 Adaptive Equalization Filter

There are two primary filters used in the equalization process, if the equalizer has feed forward taps, then it is called Finite Impulse Response (FIR); whereas the equalizer that has both feeds forward and feedback taps, it is referred to as an Infinite Impulse Response (IIR). These IIR filters tend to be unstable when the strongest pulse arrives after the other pulses. So, IIR filters are sometimes unstable and for that reason, IIR is not used much. These filters have been employed in many applications, such as signal processing and control, and the result was significant [39].

In this research, adaptive equalization filters were implemented and integrated within the proposed equalization scheme. In other words, the proposed equalization performed the function of the adaptive equalization filter besides its main function.

2.7.3 Adaptive Equalization Algorithm

In machine learning and optimization, some algorithms are adaptive, which means that the algorithm automatically adjusts its parameters according to the information (e.g. the rate of convergence). The error signal controls the adaptive algorithm; error signal is derived by comparing the output of the equalizer with Pulse Shaping Filter. The adaptive algorithm uses error signal to minimize the cost function, such as Root Mean Square Error (RMSE) and uses the equalizer weights in a way that it minimizes the cost function iteratively. The number of iterations depends on the algorithm that are utilized and the distortion level of the distorted channel impulse response. Furthermore, the two most popular algorithms used in equalization process in mode division multiplexing are Least Mean Squares (LMS) and the Recursive Least Squares (RLS) algorithms. The LMS algorithm is based on minimizing the error between the desired data and the actual data. Though RLS algorithm has the same function of LMS, RLS on the other side is a much faster technique [39].

After reviewing equalization and its techniques, the previous equalization schemes that have been applied to mode division multiplexing are reviewed in the following section to comprehend the earlier works, their outcome and figure out their limitations.

2.8 Previous Equalization in Mode Division Multiplexing

To mitigate the distortion that occurs in the transmitted channel impulse response over the link, equalization technique is used to compensate that distortion in the transmitted channel impulse response. Many equalization schemes for mode division multiplexing (Table 2.1) have been conducted. Some of them used LMS and RLS algorithms and compared between them as in [1, 40]. Simple error ratio measures the performance of both studies to the number of training blocks.

LMS is considered as a linear equalizer and is suitable for solving the linear distortion in the channel. LMS needs training sequence, but this increases the system overhead

[3]. This training sequence is considered as a disadvantage of this algorithm. Though RLS algorithm has the same function of LMS, but its performance was better in terms of convergence time and higher efficiency. Despite that, ISI which is considered to be nonlinear distortion cannot be treated by RLS.

Zhao et al. [41] proposed a new method called recursive least squares constant modulus algorithm (RLSCMA) by combining both RLS and CMA algorithms. The aim of this scheme is to mitigate the channel distortion and ISI of the transmitted channel. RLS converges more rapidly than LMS. The distorted channel impulse responses restored with fast convergence speed. This method was able to tackle the linear distortion that accrues to the signal during the transmission at the receiver more than RLS algorithm. To evaluate the algorithm outcomes, RLSCMA has been compared with stochastic gradient descent CMA method. Moreover, RLSCMA can overcome weak mode coupling modal delay efficiently with fast convergence speed. However, the study could not overcome the nonlinear distortion. Square error convergence and BER have been used to measure performance. Whereas Pan et al. [3] compared the same algorithm RLSCMA with CMA algorithm which has faster speed. In spite of its simple implementation, but CMA algorithm is slow and unstable. The performance was measured by BER.

Xiang et al. [42] have used CMA algorithm. In their study, they compared CMA algorithm with data-aided time domain equalization (DA-TDE). Hence, CMA algorithm achieved a higher efficiency. In this study, BER and OSNR were used to measure the performance of DA-TDA and CAM. A hardware equalization represented by Spatial Light Modulator (SLM) has been conducted in [43], but this kind of equalization is expensive and difficult to update. Also, this study utilized BER and OSNR in measuring the performance.

Table 2.1: Previous work on equalization in mode division multiplexing

Title	Author and year	Equalization Algorithm	Summary	Citation
MIMO Signal Processing For Mode Division Multiplexing with RLSCMA Algorithm	Haiyuan Zhao et al. 2014	recursive least squares constant modulus algorithm (RLSCMA)	RLSCMA algorithm has been applied; simulation results proved that this algorithm can solve model delay and mode coupling	[41]
2x56-Gb/s Mode-Division Multiplexed Transmission Over 2km of OM2 Multimode fiber Without MIMO Equalization	Joel Carpenter et al. 2012	A hardware Spatial Light Modulator (SLM)	Transmit signal without MIMO equalization	[43]
Adaptive Frequency-Domain Equalization in Mode-Division Multiplexing Systems	Sercan Arýk et al. 2014	least mean squares (LMS) and recursive least squares (RLS)	RLS outperforms LMS in most key respects.	[40]
MIMO Signal Processing for Mode-Division Multiplexing	Sercan Ö. Arık et al. 2014	Review two major algorithms LMS and RLS	RLS adaptation would be effective but at the cost of potentially prohibitive complexity	[1]
Performance Comparison Of DA-TDE And CMA For MIMO Equalization Multimode fiber Multiplexing Systems	Xin Xiang et al. 2015	modulus algorithm (CMA) and data-aided time domain equalization (DA-TDE) algorithm(consist of LMS inside)	Researchers have compared the CMA and DA-TDE. They found that CMA algorithm can achieve a higher efficiency than DA-TDE and improve the maximum OSNR	[42]
Fast convergence equalization algorithm for mode division multiplexing system	Xiaolong Pan et al. 2015	Recursive least squares (RLS)-based constant modulus algorithm (CMA) algorithm	RLSCMA algorithm has faster convergent speed than regular CMA	[3]

The main weakness of those existing equalization techniques in mode division multiplexing is that they can deal with linear distortion, but they cannot solve the non-linear distortion, such as mode coupling and inter-symbol interference which exist in mode division multiplexing system [44]. Therefore, this research attempted to solve the non-linear distortion in mode division multiplexing system represented by inter-symbol interference. Moreover, LMS needs training sequence, but this increases the system overhead. CMA needs to choose suitable learning steps because it converges very slowly with small steps and this makes it unable to converge with significant steps in the estimated time. The performance of RLS was better in terms of convergence, higher efficiency. However, RLS, CMA, and LMS are still unable to tackle the problem of nonlinear distortion.

An effective way to overcome this weakness is the use of hybridization techniques. This technique can be made by combining neural network with a fuzzy system to introduce neural-fuzzy technique. Neural-fuzzy techniques have many features that make them a promising approach [6, 45]. For this purpose, some equalization schemes that have been conducted in radio systems based on neural fuzzy systems are reviewed to determine the advantages and disadvantages of each scheme and review their outcome.

2.9 Previous Equalization Techniques in Radio Systems

The transmitted signal through a channel faces kind of distortions that cause ISI. Channel equalization is required to mitigate this ISI. Several equalizer schemes have been implemented to reduce these distortions like adaptive filtering in digital communication system [39].

Linear equalizers are simple in structure and easy to train and have been used in channel equalization to recover the linear distortion [46]. Linear equalizers were able to solve the linear distortion but cannot handle the nonlinear distortion. To overcome the poor performance of a linear equalizer when time-varying is one of channel characteristics, adaptive equalization based on digital filtering was used in multilayer perceptron (MLP) [47] which is better than linear and decision feedback equalizers regarding the probability of error. SNR and BER measured the performance. However, its weakness is that it consumes very long training time besides its sensitivity to the initial network parameters [48, 49].

The radial basis function (RBF) [50] equalizer has received a considerable attention because of its simplicity, requiring less time, besides better efficiency learning compared with MLP [51]. However, the RBF equalizer requires a significant number of centers in the hidden layer which increases the computational complexity. A learning algorithm called Wilcoxon learning algorithm [51] was proposed. The function of this algorithm was to mitigate ISI by using radial basis function equalizer (RBF) [52, 53] and minimize the effect of channel interference by using multilayer perceptron neural network [47]. The equalizer was compared with RBF equalizer and linear equalizer was trained with LMS algorithm. Moreover, Chebyshev artificial neural networks were used to equalize the linear and nonlinear channels [54]. To evaluate the performance of Chebyshev artificial neural network, FLANN and ChNN based equalizer were designed.

The main problem of the existing equalization techniques employed to provide a significant performance is the computational complexity and the time of training [6]. An effective way for developing adaptive equalizers for nonlinear channels is the use of fuzzy technology [55]. This type of adaptive equalizers can process numerical data and linguistic information in the natural form. Human experts determine fuzzy IF-THEN rules using input- output data pairs of the channel. These rules were used for constructing the filter for the nonlinear channel. In these systems, the incorporation of linguistic and numerical information improves the performance.

Sometimes, the construction of proper fuzzy rules for equalizers is difficult. One of the effective technologies for the construction of equalizer's knowledge base is the use of neural networks. Much effort has been devoted to the development and improvement of fuzzy neural network models. The structures of most neural-fuzzy systems mainly implement the TSK-type or Mamdani-type fuzzy reasoning mechanisms. Adaptive neural-fuzzy inference system (ANFIS) implements the TSK-type fuzzy system, where the following parts include linear functions. This fuzzy system [56] can describe the considered problem using a combination of linear functions.

Recently, an efficient method of equalization is by combining fuzzy logic and neural network. This approach provides better performance than traditional methods. Moreover, the type-2 fuzzy set can manage uncertainty better than type-1 fuzzy set. As a result, the combination of type-2 fuzzy system and the neural network has been used. Furthermore, many applications [57, 58] were used, such as time series prediction and modeling. This describes the clustering method for generating the rules and the trend learning algorithm for identifying the parameters. By the use of these algorithms [6], time-varying systems can be determined, equalization of time-varying channel and control of uncertain system.

Nonlinear neural-fuzzy structure for equalization process of channel distortion has been illustrated in [59]. The non-linear function of this equalizer was used to boost the computational power. However, another nonlinear equalizer [60] was proposed to recover both linear and nonlinear distortion by using the artificial neural network and finite impulse response (FIR). In [61], C-means is presented to control the uncertainty of the parameters in the type-2 fuzzy set. In [62], an equalizer based on genetic algorithm is proposed. In [63], a feedback adaptive equalizer based on LMS algorithm is presented for the application of mobile and wireless communication.

Other studies are presented in Table 2.2. These studies focus on the implementation of the neural fuzzy system in equalization process for the radio systems. In [64], the researcher proposed to decrease complexity and training time of parameters of neural-fuzzy network equalizer. In this study, there is no implementation of another previous equalization scheme. The result of this study proved that the application of fuzzy neural technology in channel equalization was efficient and capable of solving the nonlinear distortion. On the other hand, DFNN was proposed in [46], which is a combination of adopted fuzzy rules and neural networks. Its function is similar to Takagi-Sugeno-Kang (TSK) fuzzy system with learning ability of Radial Basis Function (RBF) neural network. This study revealed that the DFNN-based equalizer has better performance than other existing equalizers in terms of minimizing mean square error.

Similarly, [65] proposed an equalizer that is close to a maximum optimum probability (MAP) equalizer, which is considered as the optimal equalizer. The proposed equalizer reduced the computational complexity. This equalizer was evaluated by comparing it with MLP equalizer and BER as performance metrics. Furthermore, a self-constructing fuzzy neural network was proposed in [66]. It is a channel equalizer that can recover the channel distortion. The researchers compared the performance of the adaptive based network fuzzy inference system (ANFIS) and the optimal Bayesian solution. The performance of the proposed equalizer was close to ANFIS and Bayesian optimal solution.

Moreover, [67] provided a comparison between neural-fuzzy System and fuzzy-neural System models to reduce of complexity. The results of this study demonstrated that fuzzy-neural System has lower BER, lower processing speed and better accuracy than neural-fuzzy System. Mean square error (MSE) was used to measure the performance. In [68], the simulation results showed that the proposed scheme is effectively useful in equalization cases with less number of rules and less number of parameters that will be updated. This scheme was evaluated by comparing it with the ANFIS scheme. For the performance calculation, RMSE has been used.

Table 2.2: Previous Neural-fuzzy based equalization work on radio systems

No	Author/ year	Title	Problem	Equalization Method	Results	Citation
1	Rahib H. Abiyev, 2005	Neuro-Fuzzy System for Equalization Channel Distortion	channel distortion	The use of neural-fuzzy technology to decrease training time of parameters and reduce the complexity.	Satisfies the efficiency of application	[64]
2	M. J. Er, et al. 2009	Channel Equalization Using Dynamic Fuzzy Neural Networks	distortion and interference effects	Using (DFNN) which is a combination of fuzzy rules, neural networks and Simulation	DFNN equalizer has better performance than other existing equalizer based on (BER) and efficient method for linear and nonlinear channel equalization	[46]
3	Kumar Sahu, 2002	Nonlinear Channel Equalization using Computationally Efficient Neuro-Fuzzy Channel Equalizer	Channel equalization in digital cellular radio (DCR)	Propose a computationally efficient neuro- fuzzy system based equalizer	The performance of this equalizer is close to an optimal equalizer.	[65]
5	WAN-DE WENG, 2005	The Design of an SCFNN Based Nonlinear Channel Equalizer	Recover the channel distortion	Self-constructing fuzzy neural network(SCFNN) compared with that of the (ANFIS) and Simulations were carried out	The performance of SCFNN can be close to the optimal solution, and hardware	[66]

					requirement is much lower	
6	Kandarpa Kumar el al. 2012	MIMO Channel Modelling: Suitability between Neuro-Fuzzy and Fuzzy-Neural Approaches	Reduced complexity in MIMO	Either Neuro-Fuzzy System or Fuzzy-Neural System forms and compare them	Better precision, lower processing speed and bit error rate (BER)	[67]
7	Rahib H. Abiyev, el al. 2011	A type-2 neuro-fuzzy system based on clustering and gradient techniques applied to system identification and channel equalization	Identification of time-varying systems and equalization	Develop a novel type-2 neuro-fuzzy system for time-varying systems and equalization based on fuzzy clustering and gradient learning algorithm	The performance of the type-2 TSK FNS is much better	[68]

The previous studies show that neural fuzzy has been successfully used in radio systems. It has been proven that neural fuzzy systems can solve the nonlinear distortion in a radio system. This distortion represented by the effect of ISI in the transmitted signal in the receiver, which is the problem of this research. This nonlinearity which represents the gap of this research can be bridged by using neural fuzzy system. Moreover, nonlinear equalizers have the ability to minimize the effect of both linear and non-linear distortions as mentioned in the Table 2.2.

2.10 Neural-Fuzzy Systems

Neural networks and fuzzy systems are considered as alternative methods for signal processing. Both of them have some advantages over the traditional methods, especially with uncertain data. On the other hand, they have some weaknesses when used individually. Hence, combinations of both fuzzy systems and neural networks have been suggested where both models complement each other [69]. In this way, neural-fuzzy systems overcome the individual weaknesses and offer more features.

For two decades, many research efforts have been made to integrate the fuzzy logic systems and neural networks. This combination of fuzzy systems and neural networks seems possible because the two methods have a connection with Intelligent Systems from different dimensions. Fuzzy logic deals with logic or reasoning in high level, whereas neural networks provide algorithms for learning, optimization and classification [6]. As a result, the two methods complement each other [46].

There are several structures available to model neural-fuzzy networks. Fuzzy logic provides the processing for uncertain or approximate data, whereas neural networks provide algorithms for numeric optimization and classification. By integrating fuzzy logic techniques into a neural network, more flexibility can be attained. Neural-fuzzy networks can provide higher processing speed and more robust than traditional neural networks [6]. The integration of neural network and fuzzy logic has different approaches, including: Preprocess/post process Approach, Input-output Approach, and Hybrid System Approach. The hybrid system is a combination of the fuzzy system and neural networks wherein some phases of processing implemented with the fuzzy system and some phases with the neural network [69].

Hybrid systems are those composed by more than one intelligent system (fuzzy system and neural network). Such combinations are expected to be powerful because they combine different intelligent techniques. The main advantages of the hybrid system are that when the classification based on training samples, a neural network can be used, and when the classification based on experts rules, the fuzzy system can be used [69].

The most popular hybrid systems are sequential hybrid systems and incorporated hybrid systems. Incorporated hybrid systems represent the greatest degree of integration. The first system contains the second one or reverse. An example is a neural-fuzzy system, where a fuzzy inference system is implemented using a neural network Structure. In this system, neural networks were used to perform a fuzzy inference system. A fuzzy inference system consists of three components, including: a rule base which contains some of the fuzzy rules; a database which defines the membership functions used in the rules; and, finally, a reasoning mechanism which carries out the inference procedure on the rules and given facts [46].

This study has considered and conducted the development of a hybrid neural-fuzzy system that implements fuzzy inference mechanism in neural network structure including the nonlinear function for channel equalization.

2.11 Dynamic Evolving Neural Fuzzy Inference System

Evolving Connectionist Systems (ECOS) are architectures that facilitate modeling of evolving processes and knowledge discovery [24]. ECOS is a kind of neural network with specific characteristics. They learn in online and offline, with one-pass through the data and in incremental mode. They have evolving structures, use constructive learning and can learn in extended learning mode [45]. ECOS facilitate a different kind of knowledge representation and extraction.

ECOS networks also have several advantages over other constructive algorithms, as follows:

- They are not limited to a particular application domain; so, they can be applied to both classification and function approximation.

- They do not require multiple presentations of the training data set, as is the case with some of the constructive algorithms in existence.
- They learn quickly.
- They can continue learning and not restricted to learning a single training set as some other constructive algorithms.

One of the ECOS models is called Dynamic Evolving Neural Fuzzy Inference System (DENFIS) [24]. It enhances and inherits the features of other models which make DENFIS suitable for online adaptive systems [45]. It uses Takagi-Sugeno type of fuzzy inference engine. In [70], Soltic mentioned that DENFIS is recommended for prediction applications. When unknown data becomes available, the DENFIS will adapt its structure and extract the output. Moreover, the DENFIS creates rules through the learning process.

In DENFIS, clusters of data are created based on a similarity between data samples in the input space. If samples that have a distance to an existing neural node (i.e. cluster center) are less than a certain threshold, then they will be allocated to the same cluster. As for samples that do not fit into existing clusters, they will be added to form new clusters. Cluster center continues adjusting itself according to new data samples, and new clusters will be created incrementally [71]. Furthermore, DENFIS requires Pulse Shaping Filter to perform its function. The implementation of Pulse Shaping Filter can be done by using one of pulse shaping filter methods. During the process, DENFIS will train the distorted channel impulse response to mimic the Pulse Shaping Filter (i.e. curve fitting) [72].

2.12 Evaluation techniques

To assess the fitting and predictive accuracy of the equalization schemes, the results were evaluated using different alternative schemes. In some studies, researchers compared the results of their proposed equalization scheme with the previous equalization schemes, as shown in the literature review in Section 2.9. Therefore, this research evaluated by using two of most algorithms have been used in previous studies LMS and RLS.

2.13 Performance indices

Many previous Nero fuzzy equilization studies have used different performance indices to calculate and conclude their results. In [64], [46], and [65], BER have been used, whereas RMSE besides BER have been used in [66]. In [67], MSE was used to calculate the performance and to describe the results. Furthermore, RMSE was used in [68] to illustrate the outcome of the study.

2.14 Motivation for Using DENFIS Equalization

As mentioned earlier, the main problem with the existing equalization techniques to achieve a significant performance is the time of training and nonlinear distortion [6]. To tackle this weakness, this study utilized hybridization techniques, such as hybridization between fuzzy logic and neural networks, which have been applied to different nonlinear distortion problems [6]. The use of neural-fuzzy technology allows the use of a small number of parameters, fast and easy train equalizer which will lead to making a significant result [45].

There is no principal difference between a fiber optic channel and radio channel regarding ISI as mentioned in [4, 14, 23]. The received signal distorted in a similar manner in both systems [23]. For instance, symbols spread out over neighboring symbols as they propagate through the channel. In more detail, both systems share the multipath transmitted channel which leads to overlapping in the signal and in different time interval. This is called mode dispersion in fiber optic system and fading in radio system, as illustrated in Figure 2.9.

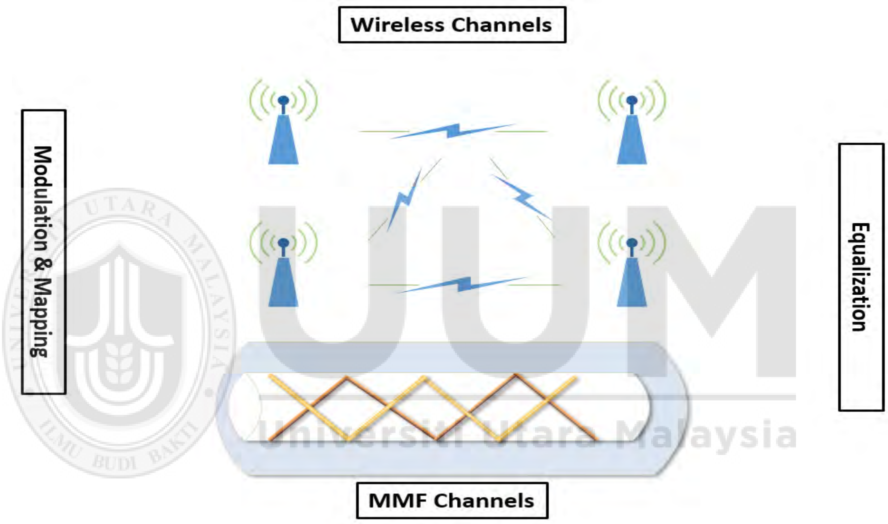


Figure 2.9: The multipath nature of wireless and optical Fiber [73]

As a result, equalization technique used for radio and other systems should be viable for fiber optic links as well. Moreover, using the neural fuzzy system to equalize the channel distortion in optical fiber is possible, regarding the fact that neural fuzzy system has been used successfully in radio systems equalization.

2.15 Summary

This chapter presented an overview of the types of dispersion that the signal faces in optical fiber. It discussed in detail equalization techniques as a solution to solve the problem of this research. Subsequently, the previous studies on equalization that have been conducted for optical fiber were able to mitigate the noise, but not the sever noise represented by ISI which is considered as nonlinear distortion. However, neural fuzzy equalization which has been implemented in radio systems can be applied to solve the nonlinear distortion. This is because of the fact that neural fuzzy equalization was used successfully in radio system to mitigate both linear and nonlinear distortion.

CHAPTER THREE

METHODOLOGY

This chapter aims to illustrate the methodology that was used to achieve the objectives of the research and the research plan. This chapter is divided into sections. Section 3.1 illustrated the research approach to explain the main research phases. This section consists of three subsections that describe in detail the phases that led to achieving the research objectives. Finally, a summary is presented at the end of this chapter.

3.1 Design Research phases

To achieve the goals of this research, many phases were applied. These phases were discussed to fulfill the objectives of this study. The methodology of this research is divided into three different phases, namely Prescriptive Study-I, Prescriptive Study-II and Descriptive Study-II phases, as illustrated in Figure 3.1.

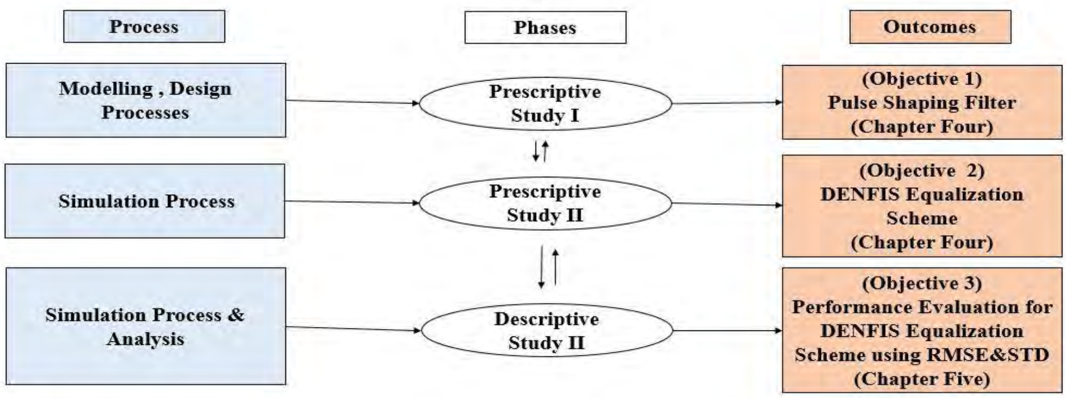


Figure 3.1: Research phases

Figure 3.2 shows the general framework of this research and how the objectives of this work have been done.

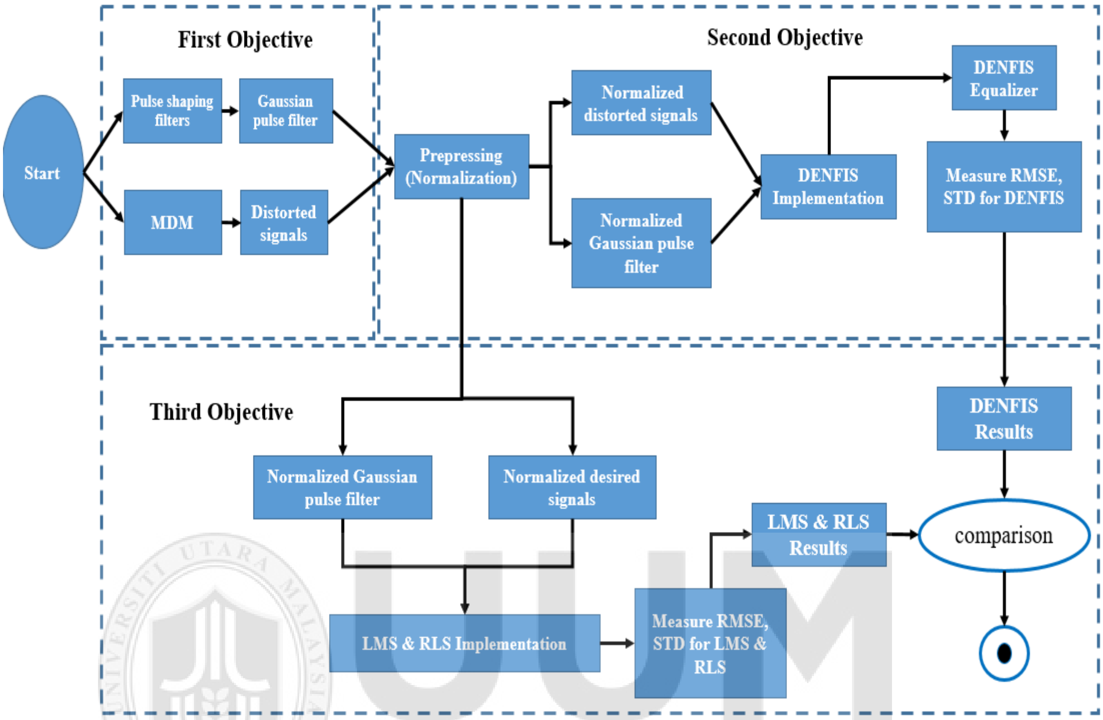


Figure 3.2: DENFIS equalization scheme framework

Before applying the research phases, some steps have been taken. These steps began with a review of literature for pulse shaping filtering which is the first step to achieve the first objective and to provide the understanding of the existing literature regarding the pulse shaping filter. The purpose behind this review was to specify a pulse shaping filter that was used to represent the ideal channel impulse response for the channel to be employed in DENFIS equalization scheme. Pulse shaping filters have been discussed in Section 2.5. Furthermore, the equalization schemes were discussed to realize the problem of mode coupling and ISI in the mode division multiplexing system over multimode fiber. Therefore, gaining an overview of previous equalization schemes in MDM and understanding them is required. Previous equalization schemes of MDM system have been mentioned and discussed in Section 2.8 of this research.

Moreover, the previous equalization schemes in radio system were reviewed to gain more understanding for different equalization schemes. Then, the previous equalization schemes that have been implemented in MDM were presented because there is no principal difference between a fiber optic and radio systems, where the distortion in the received channel impulse response is similar to both systems [1]. The previous equalization schemes in radio systems have been discussed in Section 2.9. The final step in this phase was to make a conclusion from both equalization schemes, MDM equalization scheme and equalization scheme in radio. In fact, this was very useful and helped to determine a suitable equalization scheme that was used in this research.

After reviewing the pulse shaping filters, reviwing previous equalization schemes in MDM and radio systems, the first phase have been taken to achieve the first objective of this research.

3.1.1 Prescriptive Study-I: Design of Pulse Shaping Filter (Objective 1)

Prescriptive Study-I phase is considered as a critical phase, because it has the parameters that have been determined for the pulse shaping filter to be used as a guide to the supervised DENFIS equalization scheme. The different steps of this phase are illustrated below in Figure 3.3.

The first step in this phase was to collect the distorted channel impulse response from MDM. The second step was to choose a proper pulse shaping filter to meet the requirement of the objective. The third step of this phase was to determine the parameters value of the pulse shaping filter, this was performed according to the collected distorted channel impulse responses from MDM (i.e. number of channels, modes number). Finally, the output signal from Gaussian pulse shaping filter was gathered.

For more details, this phase helped to identify the pulse shaping filter that has been used in this research to choose the proper one based on the ideal signal in a communication system. The next step was to collect the data (i.e. distorted channel impulse response) from three distinct spot modes over one wavelength of MDM. Then, the obtained signal was analysed regarding the number of modes, power distribution among channels, and the amount of distortion, as shown in Figure 3.3 Step 1.

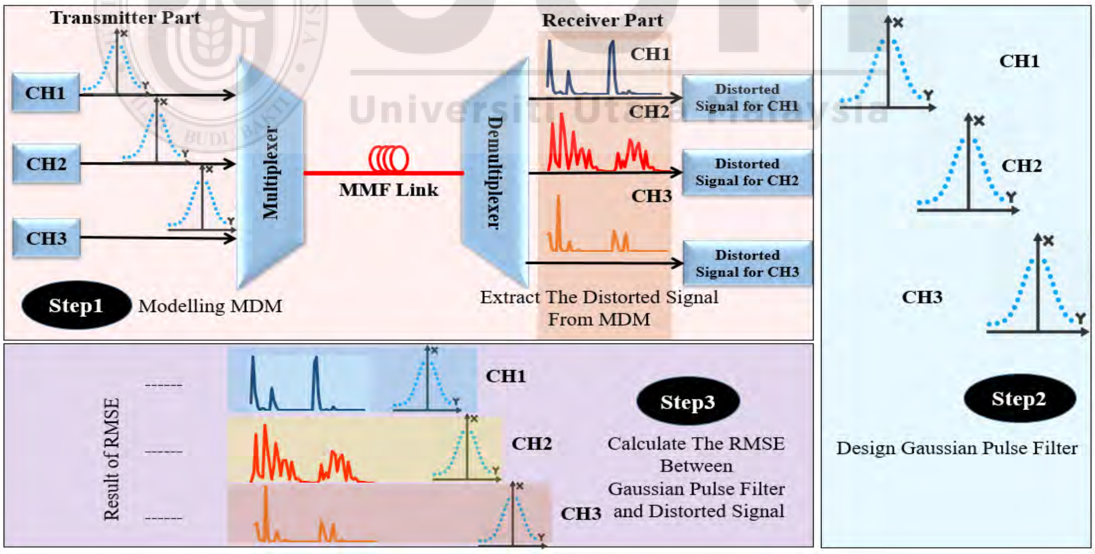


Figure 3.3: MDM model and Gaussian pulse shaping filter design

This information helped the researcher to determine the parameter of the corresponding Gaussian pulse shaping filter for each channel regarding the number of channels, the number of modes and time interval for each channel to avoid the ISI among channels, as illustrated in Figure 3.3 Step 2. Finally, the last step was to calculate the performance using root mean square error (RMSE) to determine the error value between the distorted channel impulse response and Gaussian pulse shaping filter, as shown in Figure 3.3 Step 3. Moreover, modeling the mode division multiplexing has been implemented in OptSim 5.2 simulator [74].

In the proposed MDM model illustrated in Figure 3.5 Step1, data are sent over three channels with specific data rate and a specific length of MMF. The MDM model has three different phases, they are: the input, processing phase, and the output phase. The main goal of modeling MDM and its implementation in the simulation is to collect the received signal from the receiver side which is distorted channel impulse response. After that, this distorted channel impulse response is mapped to Gaussian pulse shaping filter, which was implemented using MATLAB.

The main goal of determining the parameters to a Pulse Shaping Filter is to be employed in DENFIS equalization scheme as a guide to be followed to mitigate the effect of ISI in the received signal of MDM as have been mentioned in research objectives. Many pulse shaping filters have been reviewed in Section 2.5. According to Priyanka Agrawal [29], Gaussian Pulse is the most proper filter to shape the ideal signal comparing with Raised Cosine Pulse and Sinc pulse. The implementation of ideal status for the received channel impulse response can be performed by using the Gaussian pulse shaping filter [29]. Therefore, Gaussian Pulse filter was employed in this research to represent the target pulse shaping due to the fact that Gaussian plus filter works in the time domain. This is in contrary to other filters such as Raised

Cosine and Sinc which have simple frequency response besides working in the frequency domain.

Moreover, Gaussian plus filter does not have negative values like Raised Cosine filter, as shown in Figure 2.6, and the collected signal from MDM will not contain any negative values. In contrast, the other Raised Cosine and Sinc filters have negative values. For that reason, Gaussian filter is suitable for the collected channel impulse response. Moreover, the Gaussian pulse is used because the optical sources have a distribution of power with a wavelength that is approximately Gaussian distribution in the form [33].

To determine the parameters of Gaussian pulse shaping filter, two important steps have been considered. The first step is to identify the number of channels and the number of excited modes that have been collected from MDM. The second step is to separate the channels into different time slots or time periods. Determining the time interval for each channel is a vital step because it avoids the interference that may occur among channels and cause ISI. Figure 3.4 illustrates the steps that were taken to determine the parameters of Gaussian pulse shaping filter.

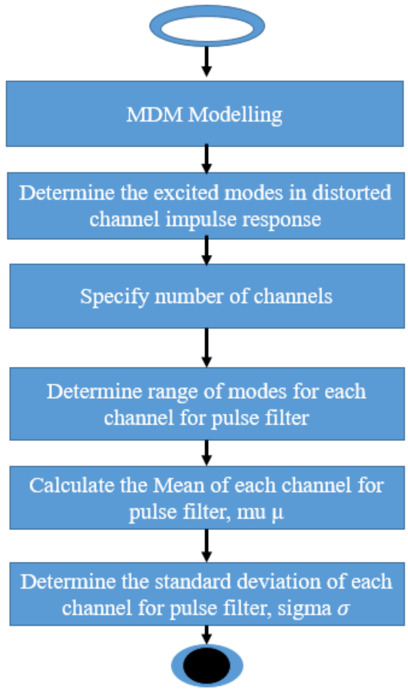


Figure 3.4: Prescriptive Study-I the parameters determination process of pulse shaping filter

In this research, many signals from many channels have been extracted from the MDM model. As a result, the parameters of Gaussian pulse shaping filter that have been applied must be equal to the number of distorted channel impulse response that was collected from the receiver of MDM. After that, each Gaussian pulse shaping filter was mapped to the distorted channel impulse response. The parameters of Gaussian pulse shaping filter g have been determined using MATLAB tool; so, each channel was mapped into its distorted channel impulse response.

After getting the distorted channel impulse responses from the mode division multiplexing and implementing the Gaussian pulse shaping filter (i.e. Objective One), the next step was Prescriptive Study-II phase which represented the implementation of the proposed equalization scheme based on DENFIS.

The outcome of this phase is:

- Chapter Four (Objective 1)
    - Determine the paramaters of Pulse shaping filter to mitigate ISI based on the distorted channel impulse response of an MDM system.

3.1.2 Prescriptive Study-II: DENFIS Equalization Scheme (Objective 2)

Before going through DENFIS implementation process, the distorted channel impulse response that has been extracted from MDM in the previous step of this phase and the Gaussian pulse shaping filter that has been designed are in different range. Therefore, the distorted channel impulse response and the Gaussian pulse shaping filter must be in the same range so that they can be implemented in DENFIS equalization scheme. In this research, linear normalization was used; the min and max of the data values are 0 and 1. Normalization process was applied to both the distorted channel impulse response and the Gaussian pulse shaping filter in MATLAB, as illustrated in Figure 3.5 Step 1.

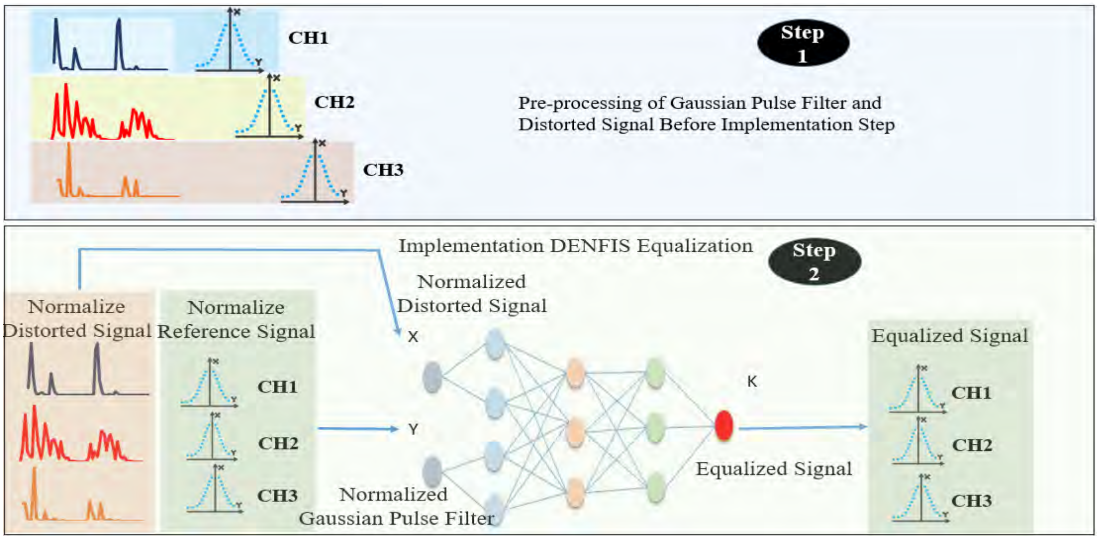


Figure 3.5: Pre-processing and DENFIS implementation

In Prescriptive Study-II, the implementation of DENFIS equalization scheme for the mode division multiplexing system has been conducted, as shown in Figure 3.5 Step 2. The goal of this is to improve the channel impulse response, reduce ISI effects in mode division multiplexer over multimode fiber and solve the nonlinear dispersion. The different steps of this phase are illustrated in Figure 3.6.

Implementation of equalization scheme by using DENFIS was the second objective of this research. DENFIS was proposed by Kasabov [24]. This method combines both neural network and fuzzy logic to introduce a powerful hybrid method by taking advantage of the adaptive learning capability of neural networks and the reasoning capability of fuzzy logic. DENFIS learns the membership functions and rules incrementally from data using evolving clustering method (ECM) algorithm [24].

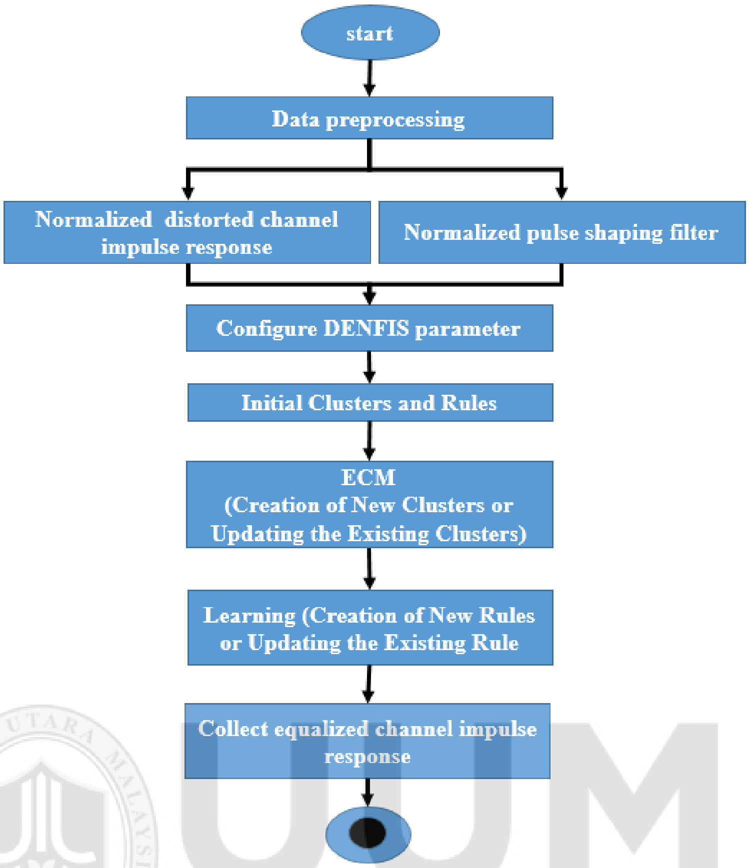


Figure 3.6: Prescriptive Study II main process

ECM is a type of grid partitioning method based on a distance measure. The distance measure used here is the normalized euclidean distance. The fuzzy inference system (FIS) utilized is a first order Takagi-Sugeno-Kang (TSK) FIS. The triangular membership function was used in DENFIS as a standard membership function. Moreover, triangular membership function has been employed in this research as standard membership function.

DENFIS equalization process started at the moment the user inserts the distorted channel impulse response and Gaussian pulse shaping filter to each channel, and then to specify the value for D*thr* parameter. D*thr* is a clustering parameter and the only e parameter to tune. Furthermore, the performance of DENFIS depends mostly on the choice of D*thr* value [72]. After that, the initial data passed to create at least *m* rules and so on until DENFIS equalization finishing its learning process, as shown in Figure 3.6.

Verification and Validation (V&V)

They were used to ensure that the proposed equalization scheme fulfils the requirements to achieve the research objectives. Verification and Validation are independent, and therefore should not affect each other.

On the one hand, verification relates to the authentication of whether the equalization based DENFIS is successful and that it gives results. On the other hand, validation process for the DENFIS equalization scheme occurs after the implementation process to ascertain that the proposed equalization scheme meets the requirements. Furthermore, validation can be obtained by conducting studies in real experiment environment. However, this study used simulation by taking advantage of the fact that some researchers used the same simulation for their studies. In other words, this refers to calculating the errors before and after the equalization process and comparing those in terms of root mean squared error (RMSE). Moreover, it is used to compare the result of DENFIS equalization scheme with the results of both RLS and LMS previous equalization schemes that have been implemented in MDM system over MMF. The equalization scheme will be valid if the channel impulse response has improved and ISI has been reduced in the transmitted signal.

The outcome of Prescriptive Study II is:

- Chapter Four (Objective 2)

    - The equalized channel impulse response using DENFIS scheme.

3.1.3 Descriptive Study-II: Evaluation of DENFIS Equalization Scheme (Objective 3)

The core point of this phase is to evaluate the DENFIS equalization scheme for the mode division multiplexing system over multimode fiber. Achieving this phase meets the third objective of this research. Furthermore, selecting an evaluation technique is a critical phase in all performance evaluation projects. The performance evaluation of DENFIS equalization can be done by using one of the three traditional approaches, namely measurement, analytical modeling, and simulation, as illustrated below.

3.1.3.1 Analytical Modelling

Analytical modeling is a set of equations done by using mathematical symbols to describing the performance of an actual system. A mathematical model can be investigated using computer programming. The results of analytical modeling can be represented by using a graphical representation generated from the output of the program. Analytical modeling has many advantages, such as low cost, less time required, and easy in trade-off evaluation. However, analytical modeling has low accuracy as compared to other techniques in performance evaluation.

3.1.3.2 Measurement

Measurement based performance is another approach of evaluation which can be conducted by implementing a real network or using a test-beds. This method provides very accurate results, but at the cost of equipment needed. Moreover, setting a real experiment for the proposed equalization scheme is expensive. However, this method is not repeatable.

3.1.3.3 Simulation

Simulation allows the researcher to perform experiments on the operations of network models without setting up the real network. On the other hand, simulation is defined as an order of a planning method of the network system, within a theoretical framework, executing and analyzing the executed output, and using network simulation [75].

Many researchers adopted simulation as their main research methodology [1]. Simulation has been suggested to mimic the real network model, test or investigate the operations of a network, without affecting the original network. Selecting the network simulation may contribute to conducting an accurate study in developing networks. Therefore, numerous researchers adopted network simulation as their research method to explore network communication and share their findings in published papers [76].

Based on the above advantages, it is evident that simulation is a suitable approach for the performance evaluation of this research. Simulator software was used to implementing and optimizing DENFIS equalization performance. OptSim simulator and MATLAB were utilized in this work. OptSim simulation tool was used to get the data (distorted channel impulse responses) that were applied in the proposed equalization scheme, whereas MATLAB was used to implement the Gaussian pulse shaping filter and DENFIS equalization scheme.

3.1.3.3.1 OptSim Simulation

OptSim by RSoft has been commercially available since 1998 and is used by leading engineers in both academic and industrial organizations worldwide. The use of OptSim [74] simulation varies in optical connections, for instance in optimizing all types of optical broadband networks, such as virtual optical connection. It has extensive database activity and large components including power, loss, wavelength and other related parameters, such as wavelength spacing, transmission capacity, power level, etc. All the mentioned parameters can be set by the user to optimize the system performance of the OptSim simulation software.

3.1.3.3.2 MATLAB

MATLAB r2013b is a software that has high ability in handling signal processing and artificial intelligence [77]. To carry out the process of equalization, MATLAB r2013b was the main platform to develop the proposed equalization for MDM using DENFIS.

MATLAB is an integrated and extensible technical computing environment that delivers powerful computation, visualization, and application development tools. It is a high-performance language for technical computing. It integrates computation, visualization, and programming environment. Furthermore, MATLAB is a modern programming language environment: it has sophisticated data structures, contains built-in editing and debugging tools, and supports object-oriented programming.

These factors make MATLAB an excellent tool for teaching and research [83]. In addition to that, MATLAB is an effective tool for digital signal processing which is employed to mitigate the noise or fix the imperfection for the communication signal. All of these factors motivate the researchers in engineering fields to use it as a tool to solve their research problems.

3.1.3.4 Evaluation

To assess the performance of DENFIS scheme, comparison was made between the results of DENFIS equalization scheme and those of other existing equalization schemes, namely LMS and RLS, as shown in Figure 3.7. LMS algorithm is a class of adaptive filter used to mimic the desired filter by finding the filter values that relate to producing the least mean squares of the error signal, which is the difference between the pulse shaping filter and the distorted channel impulse response. DENFIS equalization scheme uses RMSE in the evaluation process.

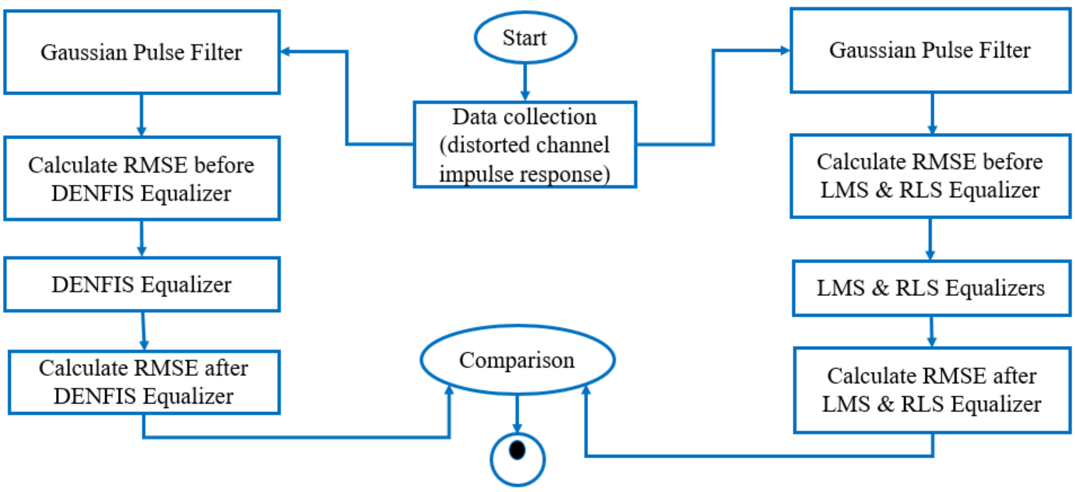


Figure 3.7: DENFIS performance evaluation

RLS adaptive filter is an algorithm which recursively finds the filter coefficients that minimize a weighted linear least squares cost function relating to the input signals. Furthermore, RLS is in contrast to other algorithms, such as LMS, that aim to reduce the mean square error.

To implement LMS and RLS schemes, the distorted channel impulse response and Gaussian pulse shaping filter values for different channels have to be available. Therefore, the distorted and Gaussian pulse shaping filters will be the same signals used in DENFIS equalization scheme. In this research, LMS and RLS have been implemented in MATLAB. LMS has two main parameters the StepSize and LeakageFactor. LeakageFactor must be between 0 and 1. StepSize is used for learning purpose for LMS. In contrast, RLS has one parameter which is *forgetfactor*. *Forgetfactor* is a real number between 0 and 1, and it is used for learning purpose [75].

3.1.3.5 Evaluation technique

The main purpose of conducting this research is to find an equalization scheme to minimize the noise represented by ISI in the mode division multiplexing system to minimize the distortion in the received signal. To determine whether DENFIS equalization meets this demand or not, performance metrics were used.

- Root mean squared error (RMSE) is an estimator that measures the average of the squares of the errors, which is the difference between the distorted channel impulse response and Gaussian pulse shaping filter [72]. Moreover, RMSE has been used to calculate the fitting accuracy of a received signal with respect to a pulse filter [71, 78].

$$RMSE = \sqrt{\frac{1}{N}\sum_{i=1}^{N}(Oi - Pi)2} \quad \ldots..(3.1)$$

Where N is the length of the channel impulse response of the concerned channel, i is the index of the channel impulse response, O is the distorted channel impulse response, and P is the parameters Gaussian pulse filter.

3.2 Summary

This chapter illustrated the methodology utilized to conduct this study. The method consisted of three different phases. These phases started with reviewing pulse shaping filters to achieve the first objective. Then, reviewing previous equalization schemes in MDM and radio systems was in the second phase and apply DENFIS equalization scheme to achieve the second objective of this research. This have been done by simulating the MDM modeling, extracting the distorted channel impulse response, determining the parameters of Gaussian pulse shaping filter, implementing DENFIS equalization, and obtaining results. The third phase is the evaluation of the proposed equalization by comparing DENFIS results with other algorithms that have been used in mode division multiplexing equalization scheme.

CHAPTER FOUR
DETERMINE THE PULSE SHAPING FILTER PARAMETERS AND IMPLEMENTATION OF DENFIS EQUALIZATION SCHEME

4.1 Introduction

This chapter described the process of determining of Gaussian pulse shaping filter parameters and the implementation of the Dynamic Evolving Neural Fuzzy Inference System (DENFIS) equalization scheme used to recover the received channel impulse response to its original form. This has been achieved by dividing the implementation goals to smaller sub-goals. Each sub-goal was reached and integrated with other sub-goals to accomplish the implementation phase. The chapter started with Section 4.2 that described the process of determining the parameters of Gaussian pulse shaping filter. Gaussian pulse shaping filter is an important thing to trigger the process of DENFIS method and without Gaussian pulse shaping filter DENFIS equalization scheme cannot function. Next, Section 4.3 detailed the implementation of DENFIS equalization scheme. Finally, a summary was presented at the end of this chapter

4.2 Determining the Gaussian pulse shaping filter parameters (Objective 1)

To determine pulse shaping filter parameters that were used in the DENFIS equalization scheme, distorted channel impulse response was collected from simulated division multiplexing (MDM). Therefore, this step was achieved by modeling the MDM in OptSim 5.2 simulator [74]. The proposed MDM model is illustrated in Figure 4.1.

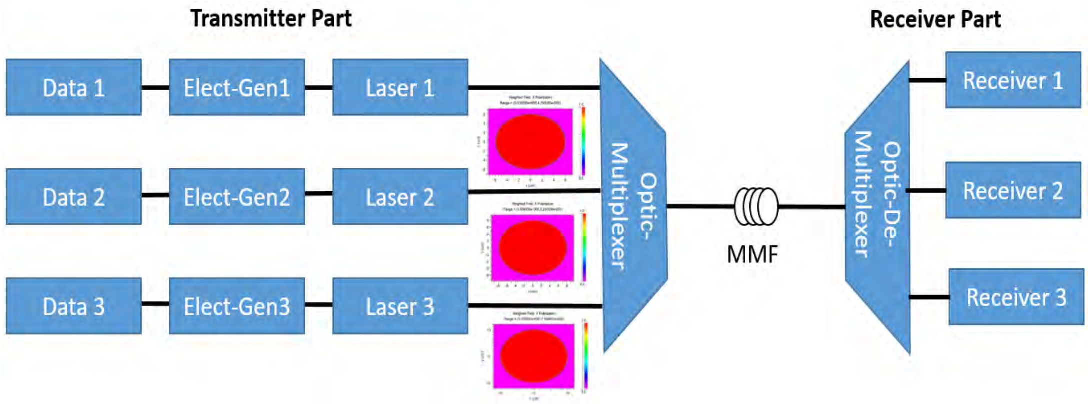


Figure 4.1: Mode division multiplexing model

Data are sent over three channels with a total data rate of 90 G/s over 5 km length of MMF. The MDM model comprises of three distinct phases, namely the input, processing phase, and the output phase. In the entry phase, the signal is generated using a Pseudo-Random Binary Sequence (PRBS). It is used to generate sequences, which are offset from one another, to ensure that each signal is not propagating the same bit value at all time points. After that, the signal generator is attached to each channel through which the transmission bit rate proposed.

Then, all previous components are connected to a Vertical-Cavity Surface-Emitting Laser (VCSEL) array, which is a type of semiconductor laser diode with a laser beam that creates the modes to transfer the signal to the other side. The three channels signals are then combined using a mode division multiplexer. Moreover, in this proposed model, a type of modes called a spot is mode used to transfer the signal to the receiver at wavelength 1550.12 nanometer. Then, the three channels are propagated through the multimode fiber for 5 km. After that, the transmitted signal arrived at the demultiplexer. Finally, the channels signal reached the receiver, as shown in Figure

4.1. For more details, Table 4.1 below illustrates the scenario and the parameters used for MDM model.

Table 4.1: MDM simulation scenario

Data Generator

No	Name	
1	Channel	Three channel (spot mode)
2	Bitrate for each channel	30 Gbps
3	Bitrate for three channels	90 Gbps
4	Pattern type	PRBS
	Pattern length	7

Elec- Generator

No	Name	
1	Drive-Type	On-Off
2	Modulation-Type	NRZ
	Signal type	CURRENT

SpVCSEL

No	Name	
1	Wavelength	1550.12e-9
2	Mode-Type	spot mode
3	Spot 01	x-outer: 6 um
4	Spot 02	x-outer: 8 um
5	Spot 03	x-outer: 10 um

MMF

No	Name	
1	Operational-mode	parabolic
2	Core-radius	25
3	Length	5000
4	Attenuation	0.3

Spatial-coupler

No	Name	
1	X offset	13
2	Y offset	-12

Farthermore, many scenarios have been tested until figuring out the proper scenario, as shown in Table 4.1. The model is executed, and data propagated through the input phase, processing phase and finally, the output phase. Three channels have been implemented Channel 1, Channel 2 and Channel 3 respectively. These channels are combined and sent to their destinations. After modeling the MDM and executing it in the simulation, the distorted channel impulse response has been gathered from the MDM. The next step is to prepare this data to be suitable for equalization process. The next sections show why performing data preparation is important and how will be implemented.

The collected signal or distorted channel impulse response from the MDM model consists of three channels. Each channel has different value range between 0 and 1. This value represents the power coefficient of the distorted channel impulse response. Moreover, the collected signals have a very severe distortion and the power units are scattered overall the input space of each channel. Furthermore, there is overlapping among the channels causing Inter-Symbol Interference (ISI), which is considered as non-linear dispersion or high order distortion. In other words, each channel interferes with a time interval of another channel, as illustrated in Figure 4.2.

However, DENFIS equalization scheme cannot function with the distorted channel impulse response only. To make DENFIS equalization scheme possible, DENFIS equalization scheme requires a Pulse Shaping Filter. This Pulse Shaping Filter was used in DENFIS equalization to reshape the distorted channel impulse response during the equalization process.

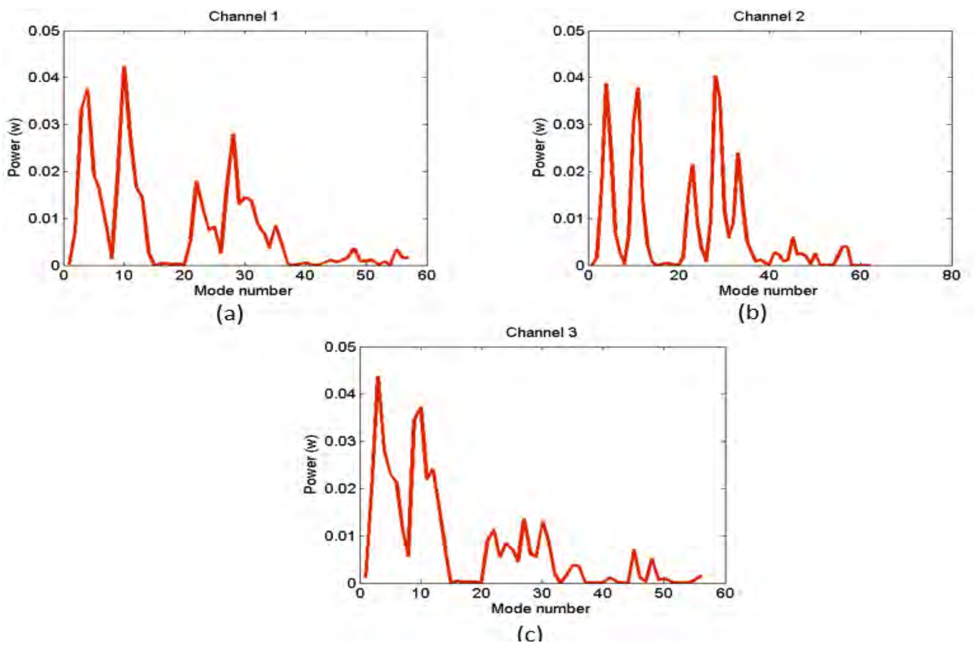


Figure 4.2: The distorted channels: (a) distorted channel impulse response of Channel 1 (b) distorted channel impulse response of Channel 2 (c) distorted channel impulse response of Channel 3

The aim of determining the parameters for Pulse Shaping Filter is to be used as pulse shaping with DENFIS equalization scheme to mitigate distortion represented by Inter-Symbol Interference (ISI) in the received channel impulse response as have been mentioned in research objectives. The design of ideal status for the received channel impulse response can be done by using the Gaussian pulse shaping filter [29]. This pulse shaping filter is one of filters used to represent the ideal signal in communication systems. Moreover, Gaussian plus filter does not have negative values, as can be seen in Figure 2.8, and the collected channel impulse response from MDM will not contain any negative values also. However, the other Raised Cosine and Sinc filters have negative values. For this reason, Gaussian filter is suitable for the collected data from simulated MDM. Furthermore, the Gaussian pulse is used because the optical sources have a distribution of power with a wavelength that is approximately Gaussian distribution in the form [33].

Therefore, Gaussian filter was used to determine Gaussian pulse shaping filter parameters. To determine the parameters, some channels and a number of excited modes collected from MDM must be determined. Moreover, putting the channels into different time slots is very important step to avoid the interference that may occur among channels and causes ISI, as shown in Figure 4.3.

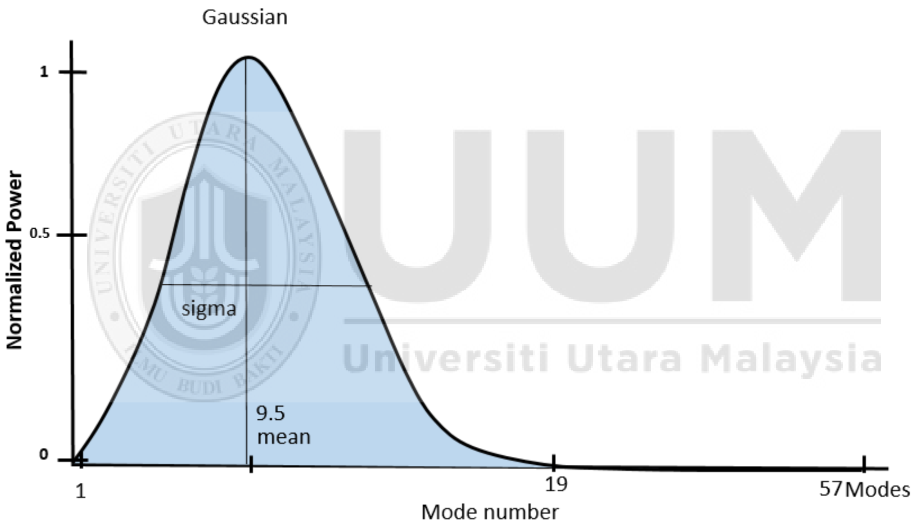


Figure 4.3: Gaussian pulse shaping filter parameters

The Gaussian function is presented in Equation (4.1). Where $\boldsymbol{x}$ is the mode number which is based on the modes number that arrived to the receiver. In this research, they were 57 Modes divided by 3. μ is the mode at which the channel impulse response is maximum to mimic the ideal shape of the channel and also it is median of each channel. It has been calculated by computing the length of each channel and dividing it by 2; so, μ = 9.5 for each channel. Moreover, $\boldsymbol{\sigma}$ is the standard deviation which is the channel impulse width. In the proposed Gaussian pulse shaping filter, the standard deviation is $\boldsymbol{\sigma}$ = 2. The value of standard deviation has been determined based on the collected modes number and the number of channels. Furthermore, in this scenario, the standard deviation value cannot be higher or lower than 2 because choosing higher or lower value will make the Gaussian pulse filter not accurate enough to include all the data points in the input space.

$$\boldsymbol{f(x) = \frac{1}{\sigma\sqrt{2\pi}}\ e^{-(x-\mu)^{2/(}\,2\sigma^{2})}\ \ldots\ldots\ldots..(4.1)}$$

This parameter mentioned above have been implemented in MATLAB to produce three reference channels and map those channels with the three channels that have been collected from the MDM. As a result, each channel is assigned to its distorted channel impulse response. The result is three curves; these curves represent the Gaussian pulse shaping filter for the three channels as one curve for each channel. In other words, the first distorted channel will be mapped to the first Gaussian pulse filter; the second distorted channel will be assigned to the second Gaussian pulse filter, and the third distorted channel will be mapped to the third Gaussian pulse filter channel, as shown in Figure 4.4. Furthermore, each channel has its time interval; therefore, there will be no overlapping among these channels.

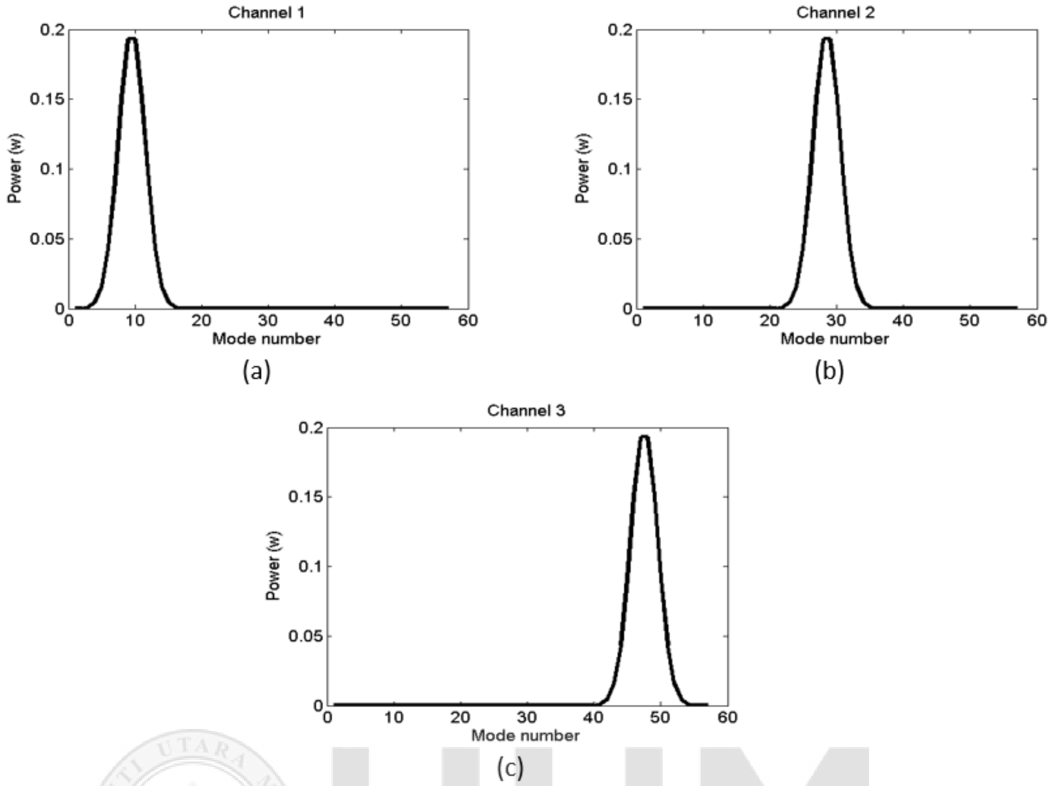


Figure 4.4: The shape of Gaussian pulse filter: (a) Gaussian pulse 1 (b) Gaussian pulse 2 (c) Gaussian pulse 3

Hence, the collected distorted channel impulse response and Gaussian pulse shaping filter in its status are not suitable to be applied to the equalization process because they are in different ranges. For tackleing data range problem, data normalization process was implemented to make this data suitable for equalization in this research. Moreover, data normalization has been applied to both the distorted channel impulse response and Gaussian pulse shaping filter.

Data normalization is a process where distorted channel impulse response, which is brought from MDM model, should be fitted with the Gaussian pulse shaping filter or be in the same range. In this research, min and max normalization was used. The min and max of the data values are 0 and 1, as shown in Equation (4.2), where the actual value is x, the minimum value in the range is x_{min} and x_{max} is the maximum value in the range [79].

$$\boldsymbol{x_{norm} = \frac{x - x_{min}}{x_{\max} - x_{min}}} \quad \dots \dots \dots .. (\mathbf{4.2})$$

Where x represents the value of distorted channel impulse response or Gaussian pulse filter, x_{min} is the minimum value in channel impulse response and x_{max} is the maximum value of channel impulse response. Normalization process is applied to both the distorted channel impulse response (i.e. the distorted channel impulse response from MDM model) and the Gaussian pulse shaping filter. Normalizing the data is very important when comparing different variables with different units as this will scale all variables to the same range making it easier to find any similarity in the behavior. Furthermore, another application is the Neural Networks and Fuzzy Logic where the input data is required to lie within a [0, 1] range as this will enhance the forecasting accuracy.

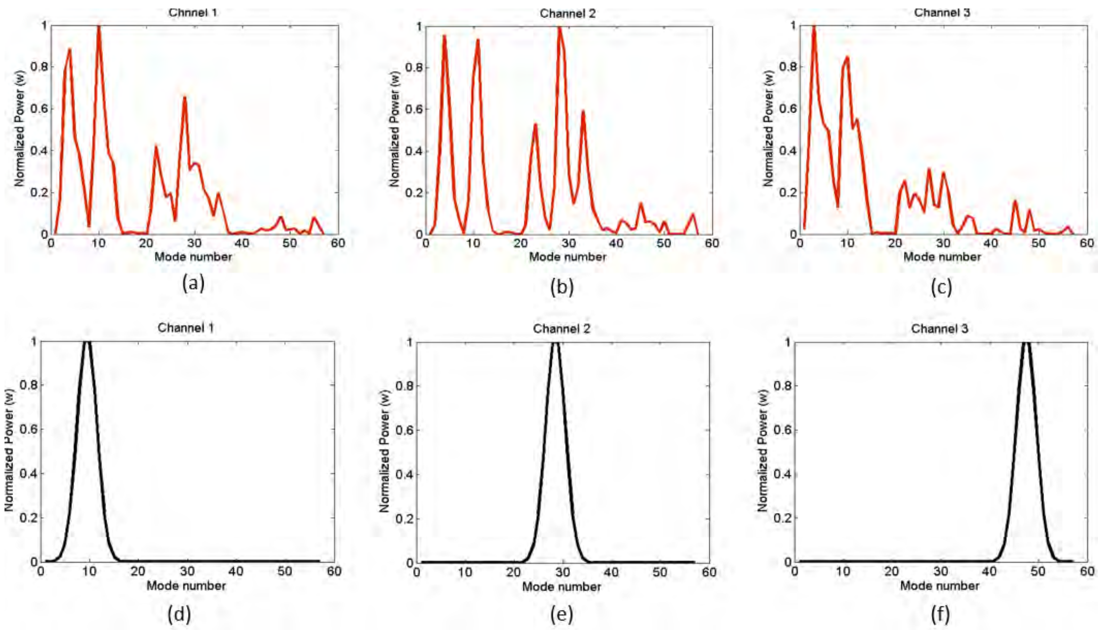


Figure 4.5: Normalized channels: (a,b,c) distorted channel impulse response (d,e,f) Gaussian pulse shaping filters

Figure 4.5 illustrates the normalized distorted channel impulse response and the normalized Gaussian pulse shaping filter after normalization process. Data normalization is an important process which will make sure that no values will be ignored inside the input space. Moreover, data normalization process has been accomplished, and both distorted channel impulse response and Gaussian pulse shaping filter are ready to go through the next step which is the implementation of DENFIS equalization.

The different three Pulse Shaping Filters have been designed and used in DENFIS equalization as a target to equalize the distorted channel impulse responses. To design the Pulse Shaping Filter, distorted channel impulse response represented by three channels has been collected from the simulated MDM, as shown in Figure 4.6. After that, depending on the number of channels and number of modes in each channel, the

Gaussian pulse shaping filter has been designed in MATLAB to get the results of Gaussian pulse shaping filter for the three channels, as illustrated in Figure 4.7. The Gaussian pulse shaping filter has been developed depending on Gaussian function. The benefit of this Gaussian pulse shaping filter is to be used in DENFIS equalization scheme to serving as a guide for DENFIS during the learning process.

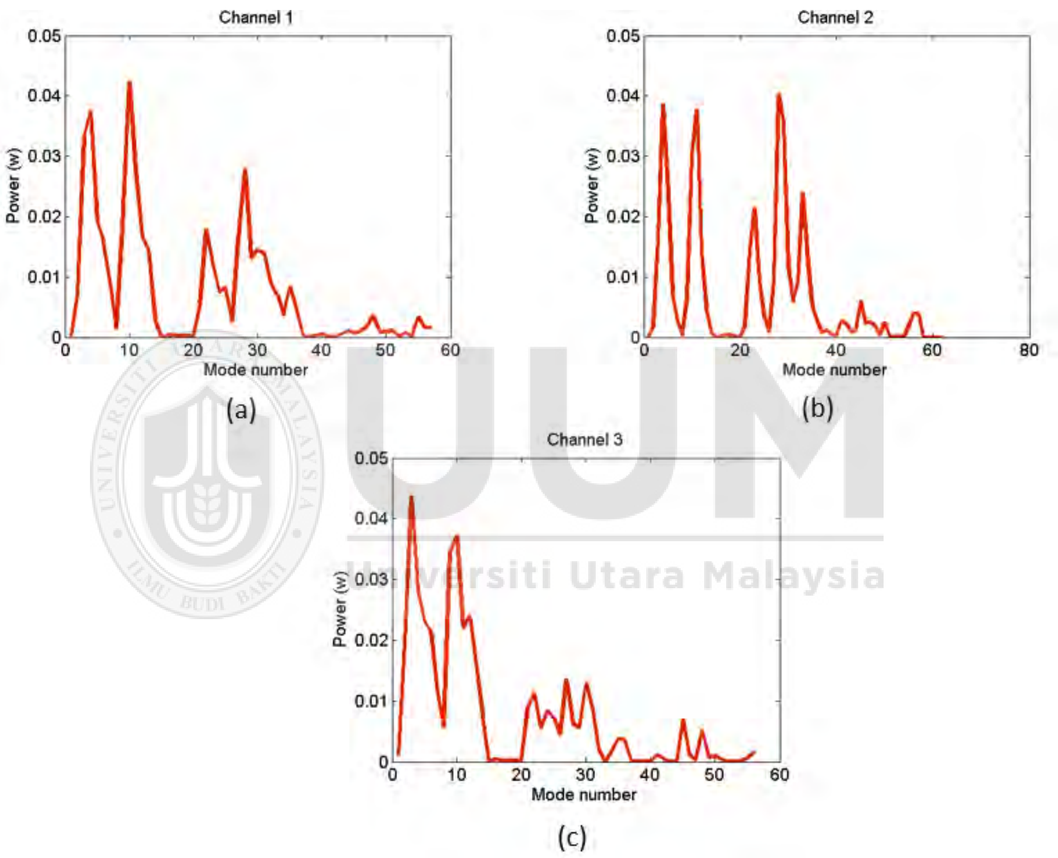


Figure 4.6: Distorted channels: (a) distorted channel impulse response for Channel 1 (b) distorted channel impulse response for Channel 2 (c) distorted channel impulse response for Channel 3

The collected channel impulse response from MDM is waving from a high level of distortion and the modes spread overall the channel without considering the time interval of each channel which leads to the signals interfere with each other and causing ISI. Moreover, the shape of channel impulse response is scattered, and it has many fluctuations. However, the designed Gaussian pulse filter has its time interval which will lead to control the scattering of the data point in the input space to avoid the ISI between the signals.

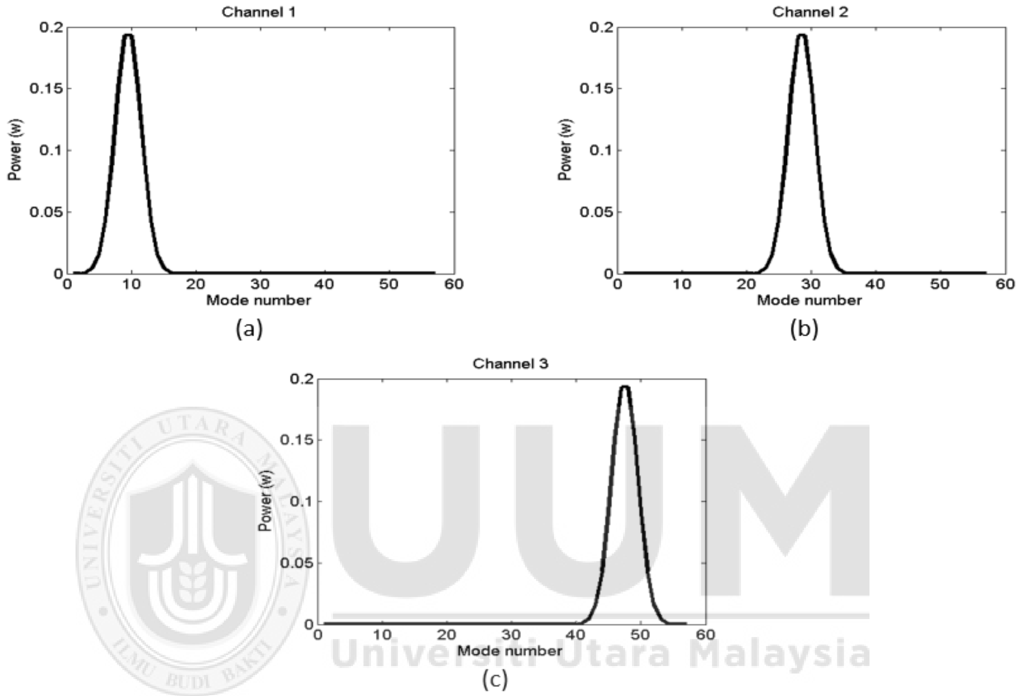


Figure 4.7: The Gaussian pulse shaping filter: (a) Gaussian pulse filter for Channel 1 (b) Gaussian pulse filter for Channel 2 (c) Gaussian pulse filter for Channel 3

Figure 4.7 illustrates the three designed Pulse shape filter to be used in DENFIS equalization for three distorted channel impulse response. In contrast with the distorted channel impulse responses that have been collected from MDM, the designed pulse shaping filter for each channel has its own time interval and it has ideal shape for the communication channel to avoid the overlapping among channels, as can be seen in Figure 4.7.

4.3 Simulation of Equalization Scheme Using DENFIS (Objective 2)

Achieving the first objective of this research (i.e. determining the Gaussian pulse shaping filter parameters and modeling the MDM model, distorted channel impulse response collection) means that all requirements are available to achieve the second objective of this research. In other words, the second objective cannot be accomplished without achieving the first objective of this research. Implementation of equalization scheme using DENFIS is the second purpose of this research. DENFIS was proposed by Kasabov [24]. This method combines both Neural Network and Fuzzy Logic to introduce a powerful hybrid method by taking advantage of the adaptive learning capability of Neural Networks and the reasoning capability of Fuzzy Logic. The DENFIS system learns the Membership Functions and rules incrementally from data using Evolving Clustering Method (ECM) algorithm [24].

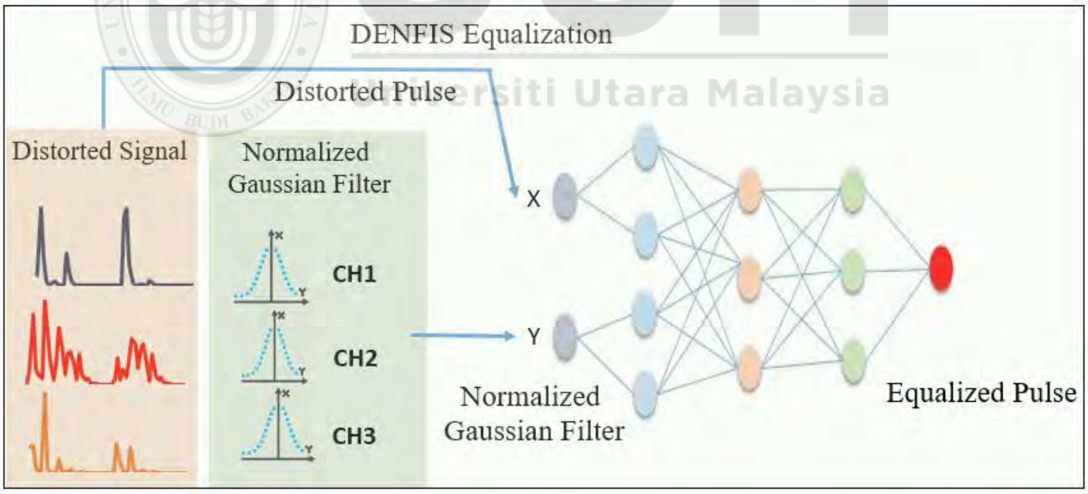


Figure 4.8: DENFIS implementation

ECM is a type of grid partitioning method based on a distance measure. The distance measure used here is the normalized euclidean distance. The Fuzzy Inference System

(FIS) utilized is a first order Takagi-Sugeno-Kang (TSK) FIS. The triangular membership function is used in DENFIS as a standard membership function.

Evolving clustering method (ECM) [79] is used in DENFIS to create and update the number of clusters depending on the Dthr value. The process of dividing a big dataset into groups of similar objects that have similar properties within and different properties outside is called clustering. ECM [79] is a distance based grid partitioning method that can be applied offline by including constrained minimization criteria. The important issue to be ensured when using this approach is that the input data must be normalized as a first step. Otherwise, variables that will be used have a different unit; small or large data might be neglected due to the different input units. A threshold value is defined at the starting of the process which will determine the number of clusters to be created [77]. ECM is illustrated below:

Create the first cluster center C_0 from the first example I_0
for each subsequent vector I_n **do**
Find the minimum distance $D_{\min}$ between I_n and each
cluster centre C_n
if $D_{\min}$ is less than any cluster radius **then**
Add I_n to the nearest cluster
else
Find the cluster *a* with minimum value of $S_{i,j}$, where
$S_{i,j}$ = $D_{i,j}$ +R_i, $D_{i,j}$ is the distance between the
cluster center and vector *j*, and R_i is the radius of
cluster *i*
if $S_{i,a}$> $2D_{thr}$ **then**
Create a new cluster
else
Update *a*
end if
end if
end for

Although ECM appears to be a useful clustering method in itself, its primary function is to support the inference of fuzzy rules from data in DENFIS. This is done in two phases, firstly forming the antecedents and then followed by the consequent functions. The antecedents are formulated by finding which combination of input membership functions (MF) is the most highly activated for the centre of the cluster, that is, the values represented by the cluster centre are fuzzified by the input MF set and the winning, most highly activated, MF are taken as the antecedents for that rule.

The consequent functions are then found using a Least Means Estimation process over the examples within the cluster. Thus, each cluster is used as the basis of a single rule. Clustering and reformulation of the rules are performed whenever a new training example is presented to the network. For any input vector I, the output of the DENFIS is calculated as the combined output of the most strongly activated m rules. There is no adjustment of the MF during training.

DENFIS implementation begins with entering the normalized distorted channel impulse response along with normalized Gaussian pulse shaping filter signal to each channel individually, as shown in Figure 4.6. Then, the value for D*thr* (i.e. Dthr is a threshold value defined at the beginning of the process which will determine the number of clusters that will be created) parameter has been specified to be 0.001 and epochs is set to 2 and the number of nodes to 3 as standard values [78].

The performance of DENFIS depends mostly on the choice of *Dthr* value. As mentioned above, D_{thr} is a threshold value defined at the beginning of the process which will determine the number of clusters that will be created. In this research, D_{thr}

parameter was changed from 0.09 to 0.001 steps to get the most suitable values to achieve the optimal results. D*thr* is a clustering parameter and the only parameter to tune in DENFIS. Therefore, the performance of DENFIS depends mostly on the choice of D*thr* value [72].

This initial number of clusters is produced which leads to establishing some rules that equal to the number of clusters. Moreover, the number of extracted rules is equal to the number of clusters that will be generated by the ECM [79]. Then, the role of ECM is to create a cluster or update existing clusters by reposition the clusters and update the radius of each cluster.

In ECM phase, the first cluster has been created and a zero value assigned to its radius. During ECM process, three scenarios happened. The first one happened when the first data were inserted and cluster was created. Then, when a new data point arrived, the ECM will calculate the distance between the new arrival data point and the centers of all clusters that have been created. This process will find the Min distance using the normalized Euclidean distance between the cluster center and the new data point. Then, if the distance is less than the value of the radius, this new data point will belong to an existing cluster, none of the existing clusters will be updated, nor any new cluster is added, and this is the first scenario.

The second scenario was that if the minimum distance is more than the radius of this cluster, then a new distance is calculated as the distance from each cluster plus the radius of each cluster. If the new minimum distance is more than twice the threshold level, then a new cluster is created with its center as the data point position and its radius is zero.

The third scenario happened when the minimum new distance is less than twice the threshold level, then the radius of this cluster will be updated to the minimum new distance divided by 2 and the cluster center is moved along the line connecting the new data pair and the old cluster center. In this way, the maximum distance from any data pair belonging to a cluster and its cluster center is kept below the threshold defined at the beginning.

Finally, new rules have been created and some of the rules have been updated depending on the number of clusters that have been set up or maintained by ECM. All the channels are trained and tested with the same data set. For more details, DENFIS consists of 5 main layers, which are briefly described below:

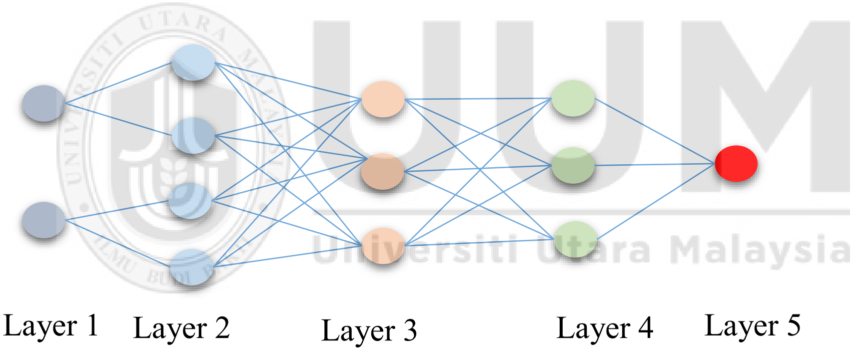


Figure 4.9: DENFIS layers

The distorted channel impulse response and Gaussian pulse filter are input variables. The first layer of the network contains the membership function. The standard triangular membership function has been used to map crisp input values to fuzzy linguistic expressions. In this layer, quantization process has been done for both distorted channel impulse responses and Gaussian pulse filter. Then, in layer two the incoming data points have been multiplied, and the product represents the weight of each rule. After that, normalization occurs in layer three by calculating the ratio between each weight of rule and the sum of all weights. Next, the normalized weight multiplied with linear activation function called Least Means Estimation process that has consequent parameters in layer four. Finally, the overall output has been computed by the summation of all incoming data points in the fifth layer.

Different results from various channels have been discussed and analyzed in terms of the shape of channel impulse response, RMSE and the time interval for each channel before and after the DENFIS equalization. DENFIS equalization scheme simulated for the distorted channel impulse response of the three different channels. The results of three channels are summarized in Table 5.1. The table illustrates the values before DENFIS equalization and after DENFIS implementation. Furthermore, RMSE has been used to calculate the performance.

Table 4.2: The results of DENFIS equalization scheme for the three channels

	Dthr	**N of R**	**Iteration**	**RMSE(before)**	**RMSE(after)**
CH1	0.001	43	30	0.2119	0.0365
CH2	0.0001	52	30	0.2124	0.0294
CH3	0.0001	53	40	0.2376	0.0356

Figure 4.10 (a) presents the distorted channel impulse response of Channel 1 before the implementation of DENFIS. DENFIS equalization used Gaussian plus filter to mitigate the distortion in the Channel 1, as seen in Figure 4.10 (b). The final result of the equalized channel impulse response for Channel 1 is illustrated in Figure 4.10 (c) after applying the DENFIS equalization scheme.

In Figure 4.10, the equalization process in Channel 1 was performed and the nonlinear distortion represented by ISI is dramatically decreased. This is noticed in the shape of channel impulse response of equalized channel impulse response, as illustrated in Figure 4.10 (c).

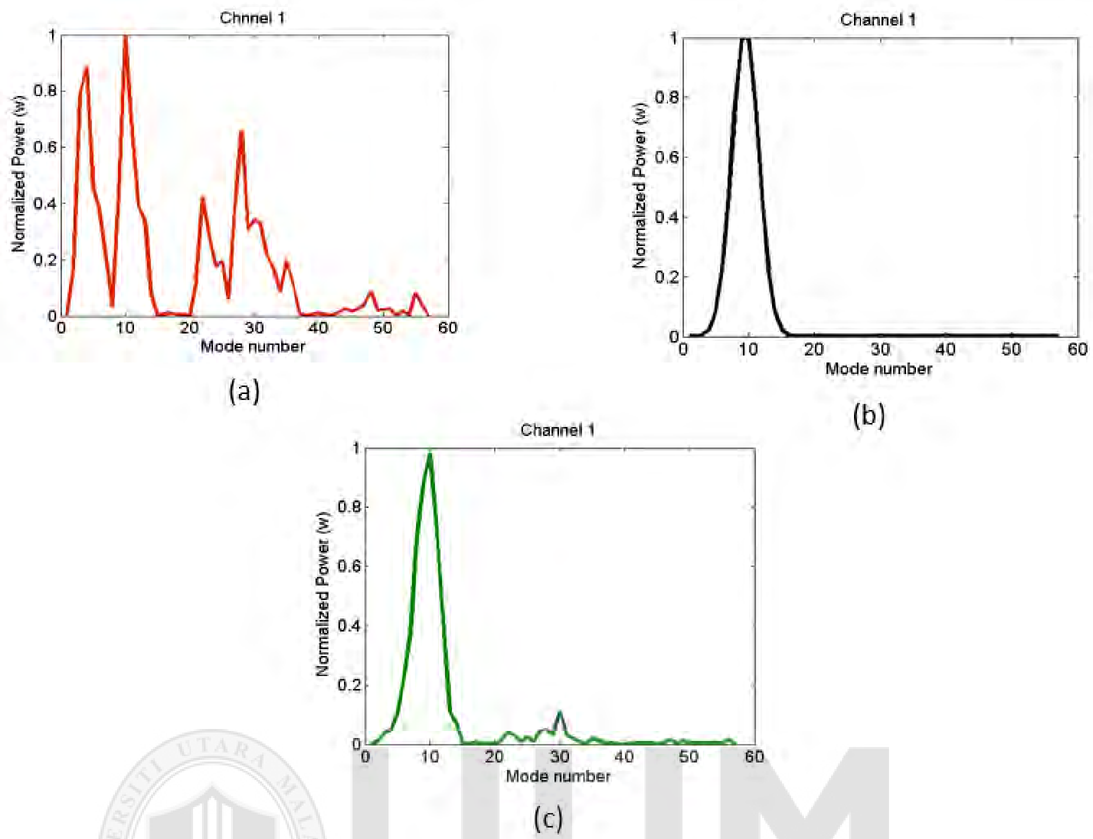


Figure 4.10: Effect of equalization on channel impulse response: (a) distorted channel impulse response for Channel 1 (b) Gaussian Pulse filter for Channel 1 (c) equalized channel impulse response for Channel 1 using DENFIS

Furthermore, the modes are centered in their proper position; so, there is no overlapping with other signals. The results indicated that Channel 1 required less number of rules and fewer iteration numbers because most of Channel 1 values were positioned in the Channel 1 time interval or near of it. The distortion amount in the distorted channel impulse response of Channel 1 before equalization was assembled mostly between 1 and 40 modes period, as shown in Figure 4.10 (a). However, the distortion value in the equalized signal of Channel 1 after equalization was focused in Channel 1 time interval, as shown in Figure 4.10 (c). However, there was still a little distortion in the equalized signal of Channel 1 and cannot be totally eliminated.

Moreover, the result of Channel 1 with the setting of D*thr* (i.e. DENFIS parameter) showed that RMSE before equalization has decreased from (0.2119) to be (0.0365) after equalization. The value of RMSE after equalization was lower before applying DENFIS equalization. This means that the scattering of the data point in the input space has been increased. Furthermore, the value of RMSE before equalization was extracted by calculating the value of distorted channel impulse response with the value of Gaussian pulse filter. Furthermore, the RMSE after equalization has been calculated by measuring the RMSE of equalized signal 1 and the RMSE of the Gaussian pulse filter.

The result of Channel 2 and its improved channel impulse response is illustrated in Figure 4.11 (a) before the equalization, and Figure 4.11 (c) shows the shape of Channel 2 after DENFIS equalization. The equalization process in Channel 2 provided the best RMSE result which indicates a real improvement in the shape of channel impulse response and no overlapping between different signals, as shown in Figure 4.11 (c). Moreover, the form of equalized impulse shape has almost the same impulse shape as the pulse shaping filter, as shown in Figure 4.11 (b). The result presented that Channel 2 used 52 numbers of rules which means that D*thr* has created and updated 52 cluster centers during the equalization process and 30 iteration numbers because most of Channel 2 values are positioned in the Channel 2 modes period.

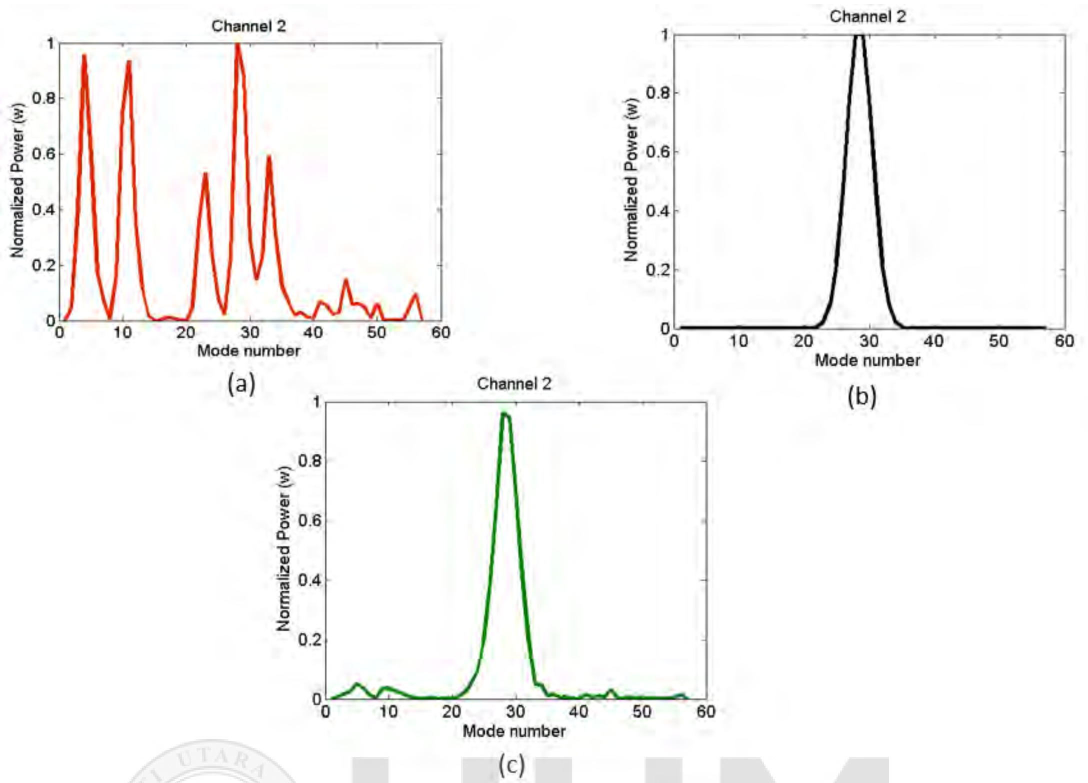


Figure 4.11: Effect of equalization on channel impulse response: (a) distorted channel impulse response for Channel 2 (b) Gaussian Pulse filter 2 (c) equalized channel impulse response for Channel 2 using DENFIS

According to Figure 4.11 (a), the distortion amount in the distorted channel impulse response of Channel 2 before equalization was assembled mostly between 1 and 15 modes period from the left side of Channel 2 and from 36 to 50 in the right side of Channel 2. However, the distortion value in the equalized signal of Channel 2 after equalization was centralized in Channel 2 period, as can be seen in Figure 4.11 (b). Moreover, there was still a little distortion in the equalized signal of Channel 2 and cannot be totally mimic the Gaussian pulse shaping filter.

The result of Channel 2 illustrated that RMSE before equalization has decreased from (0.2124) to be (0.0294) after equalization and STD value reduced, as shown in Table 4.2. This reduction in RMSE value emphasizes that the possibility of ISI will be slight.

The same concept implemented in Channel 1 has been applied to Channel 2. The value of RMSE before equalization is extracted by calculating the value of distorted channel impulse response with the value of Gaussian pulse filter. Furthermore, the RMSE after equalization has been calculated by measuring the RMSE of equalized Channel 2 and the RMSE of the desired Channel 2.

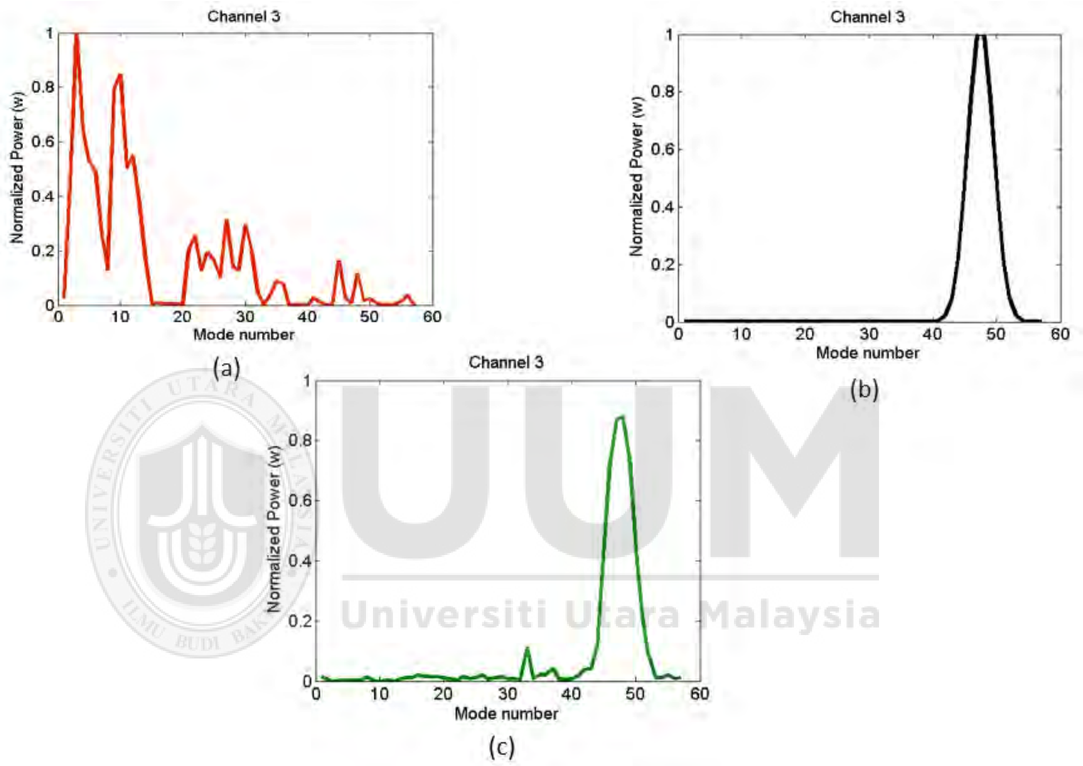


Figure 4.12: Effect of equalization on channel impulse response: (a) distorted channel impulse response for Channel 3 (b) Gaussian Pulse filter for Channel 3 (c) equalized channel impulse response for Channel 3 using DENFIS

The result of Channel 3 before equalization and its response after applying the DENFIS equalization is shown in Figure 4.12 (c). The result in numbers is presented in Table 4.2 with the values of the parameters that have been used for each channel. Table 4.2 shows the value of Channel 3 before and after applying DENFIS equalization scheme.

In Figure 4.12, the equalization process for Channel 3 improved the shape of channel impulse response significantly and the points in the input space integrated under the Channel 3 time interval, as shown in Figure 4.12 (c). This indicates that the effect of ISI in this channel has been mitigated and no more overlapping between channels. The results illustrate that Channel 3 provided a good RMSE result but at the cost of a high number of rules and a large number of iterations. According to Figure 4.12 (a), the distortion amount in the distorted channel impulse response of Channel 3 before equalization was assembled mostly between 1 modes period and 17 modes period from the left side of Channel 3 and from 20 to 35 for the same side. Furthermore, there was still a little distortion in the equalized signal of Channel 3.

The result of Channel 3 illustrates that RMSE before equalization has reduced from (0.2376) to be (0.0356) after equalization. The value of RMSE after equalization was minimized which means that data points scattering is reduced. The same idea implemented in Channel 1 and Channel 2 has been applied to Channel 3. The value of RMSE before equalization is extracted by calculating the distorted channel impulse response value with the value of Gaussian pulse filter. Furthermore, the RMSE after equalization has been calculated by measuring the RMSE of equalized Channel 3 and the RMSE of the desired Channel 3.

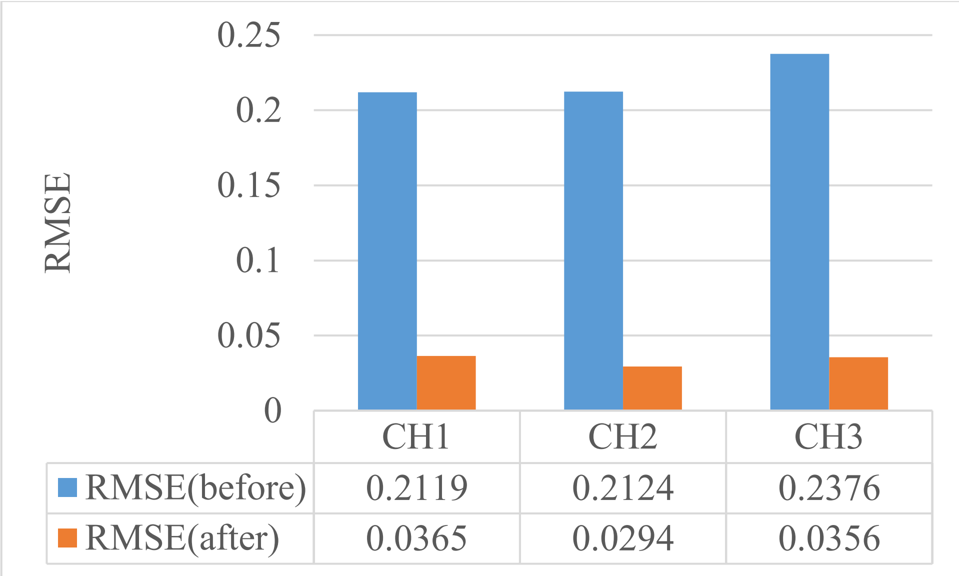


Figure 4.13: RMSE for all channels using DENFIS

According to the results discussion of channels mentioned above, it can be concluded that the value of RMSE before equalization has a significant influence on the results and decreeing the RMSE value, as shown in Figure 4.13. This means that if the scattering or spreading of data points in the input space is high, then the RMSE after equalization will be minimized sharply. The reason of this is that ECM algorithm will be able to produce a greater number of clusters which will lead to increase the number of rules in DENFIS. So, the more spread of data, the lower root main square error. Furthermore, the change of *Dthr* value in DENFIS scheme is necessary because when *Dthr* = 0.9, the ECM algorithm found a kind of difficulty to determine the number of clusters used in DENFIS equalization scheme to determine the number of rules. As a result, the best result was obtained when *Dthr* was 0.001 and 0.0001 for the different three channels.

DENFIS can deal with significant amount of data. Therefore, increasing the number of channels and the rate of data may provide more accurate analysis in the DENFIS equalization scheme. In this research, an offline DENFIS equalization scheme was carried out to mitigate the amount of ISI in the channel impulse response. For better results, a real-time compensation is required.

4.4 Summary

This chapter described the systemic phases of research comprising the channel impulse response collection from a MDM system, determining the parameters of a Gaussian pulse shaping filter and the simulation of DENFIS equalization scheme. Furthermore, this chapter discussed the pulse shaping filter results to achieve first objective of this research, as well as DENFIS equalization scheme results which represent the second objective.

CHAPTER FIVE
EVALUATION OF DENFIS SCHEME

5.1 Introduction

The Gaussian pulse shaping filter parameters used in DENFIS equalization scheme has been determined and implemented in the previous chapter to fulfill the first and second objectives of this research. Besides the results discussion, the third objective of this research has been discussed and has been illustrated in this chapter. This has been done through evaluating DENFIS equalization scheme by comparing its results with other previous equalization schemes, namely Least Mean Squares (LMS) and Recursive Least Squares (RLS). This comparison has been done based on root mean squared error (RMSE). In this chapter, evaluation of DENFIS equalization scheme is mentioned in Section 5.2. The DENFIS versus least mean squares (LMS) Algorithm comparison is presented in Section 5.2.1. Section 5.2.2 discussed the evaluation process of DENFIS equalization scheme versus recursive least squares (RLS) algorithm. Finally, a summary is presented at the end of this chapter.

5.2 Evaluation of DENFIS equalization scheme with respect to existing MDM algorithms (Objective 3)

DENFIS results have been compared with channel impulse response of LMS and channel impulse response of RLS algorithms to observe the effect of DENFIS equalization scheme in terms of RMSE value. This has been done by implementing LMS and RLS in MATLAB and setting up the parameters of each algorithm. Moreover, distorted channel impulse response and Gaussian pulse shaping filter executed in DENFIS equalization scheme have been implemented in LMS and RLS schemes. The main concepts of LMS and RLS have been illustrated below; besides the main parameters used in the equalization process of both LMS and RLS. The result has been collected and discussed for both LMS and RLS.

5.2.1 DENFIS versus Least Mean Squares Algorithm

For the LMS algorithm, the channel impulse response at the equalizer input is formed in Equation (5.1):

$$y(n) = \sum_{j=0}^{n-1} h(j)x(n-1) + v(n) \ldots\ldots (5.1)$$

Where $x(n)$ denotes the data sample at time index n, $v(n)$ is the additive noise and $h(j)$ is the channel impulse response. The weighting coefficients in the LMS algorithm are updated according to the following expression (5.2).

$$w(n+1) = w(n) + \mu e^{H}(n)X(n) \ldots\ldots (5.2)$$

Here, μ is the step size which controls the rate of convergence of the LMS algorithm. LMS has two main parameters the *StepSize* and *LeakageFactor*. LeakageFactor must be between 0 and 1. Where StepSize is used for learning purpose for LMS [77]. This parameter has been adjusted to get the optimal result by applying the values of parameters from previous studies and using trial and error [13]. The distorted channel impulse response and Gaussian pulse shaping filter for different three channels have been imported. The imported channels are the same channels that have been used in DENFIS proposed equalization scheme.

Table 5.1: Results of Least Mean Squares algorithm

	No. Of Weights	Step size	Leakage factor	RMSE (before)	RMSE (after)
Channel 1	15	0.4	1	0.26241951	0.1431572
Channel 2	25	0.4	0.7	0.28983271	0.0997181
Channel 3	50	0.4	1	0.37664307	0.0603506

The result of LMS indicated an improvement in channel impulse response for Channel 1 and decreased RMSE, as illustrated in Table 5.1, and aligned inside its time interval for each channel, as shown in Figure 5.1 (c). However, RMSE in Channel 1 is so far to be like DENFIS equalization scheme, as shown in Figure 5.1 (d). The value of RMSE in DENFIS Channel 1 is (0.0365), whereas RMSE is higher for the same channel in LMS equalization, as shown in Table 5.1. Moreover, LMS can reduce the RMSE value more efficiently than DENFIS equalization.

Furthermore, LMS can reduce the RMSE slightly, and this is because LMS is considered as a linear equalizer. As a result, LMS cannot reduce it sharply to the level that mitigates the nonlinear distortion represented by ISI.

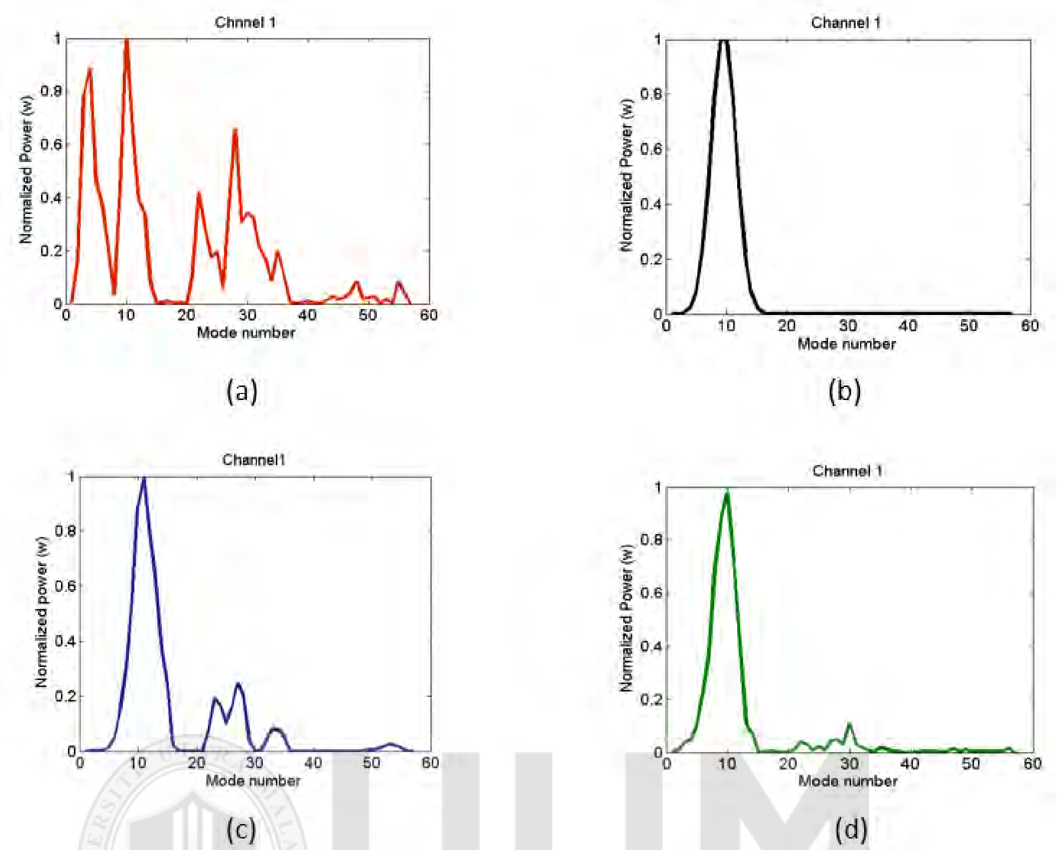


Figure 5.1: : Comparison of DENFIS and LMS equalization schemes for Channel 1: (a) distorted channel impulse response (b) Gaussian Pulse filter (c) equalized channel impulse response using LMS (d) equalized channel impulse response using DENFIS

In Channel 2, the RMSE has been decreased, as shown in Table 5.1, because most of the data points of the Chunnel 2 are positioned in the time interval of Channel 2 which will make the learning process easier, specifically between Mode 20 to 40, as shown in Figure 5.2 (a). However, the shape of channel impulse response in DENFIS Channel 2, shown in Figure 5.2 (d), is more accurate and almost mimicked the Gaussian pulse filter for the same channel in Figure 5.2 (b). Furthermore, the reduction in RMSE value of DENFIS equalization is better compared with the reduction of RMSE in LMS for Channel 2.

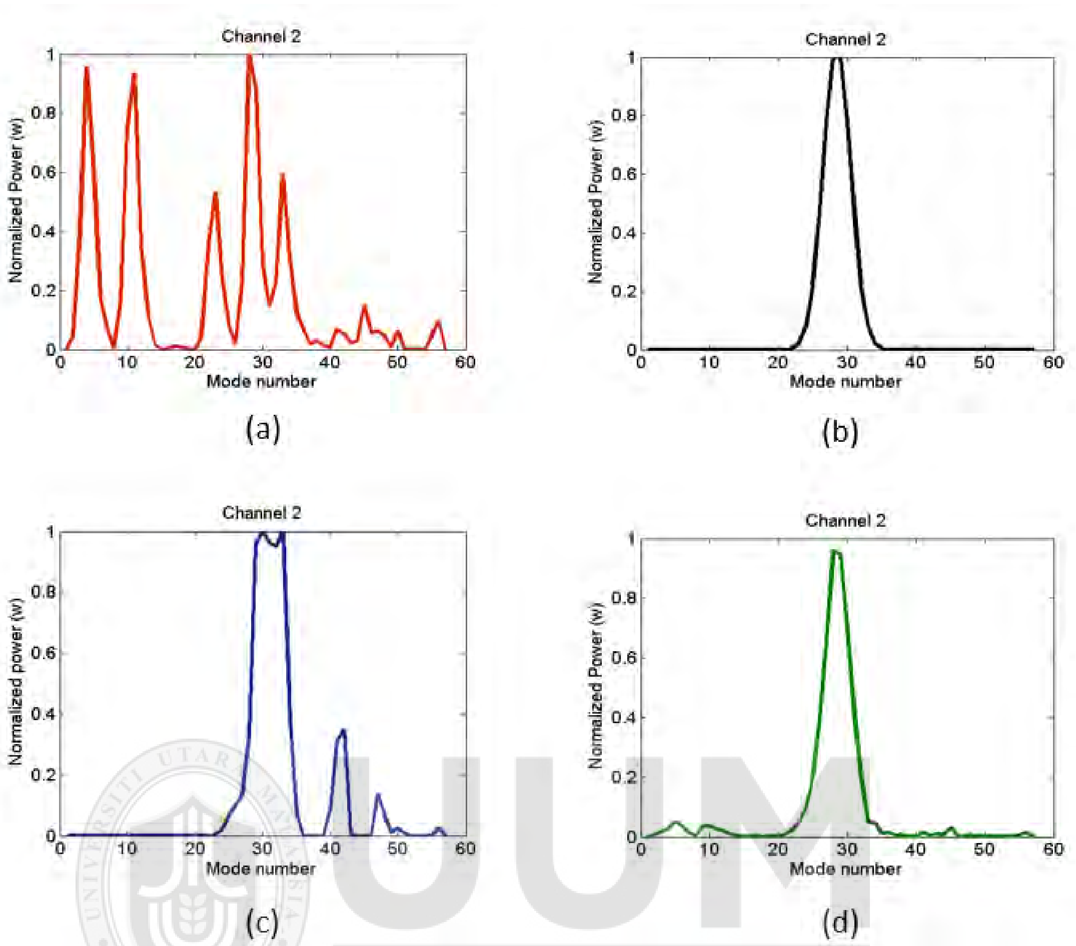


Figure 5.2: : Comparison of DENFIS and LMS equalization schemes for Channel 2: (a) distorted channel impulse response (b) Gaussian Pulse filter (c) equalized channel impulse response using LMS (d) equalized channel impulse response using DENFIS

The result of LMS of Channel 3 presented the best shape of channel impulse response for Channel 3 comparing with other channels of LMS and DENFIS in Channel 3, decreased RMSE, as illustrated in Table 5.1, and data points located inside time interval for each channel, as shown in Figure 5.3 (c). However, the minimization of RMSE value is smaller than the reduction that has been occurred to the Channel 3 in DENFIS equalization, as figured in 5.3 (d). The value of RMSE in DENFIS Channel 1 is (0.0365), whereas RMSE is higher for the same channel in LMS equalization, as shown in Table 5.1. Moreover, LMS can reduce the RMSE value more efficiently than DENFIS equalization.

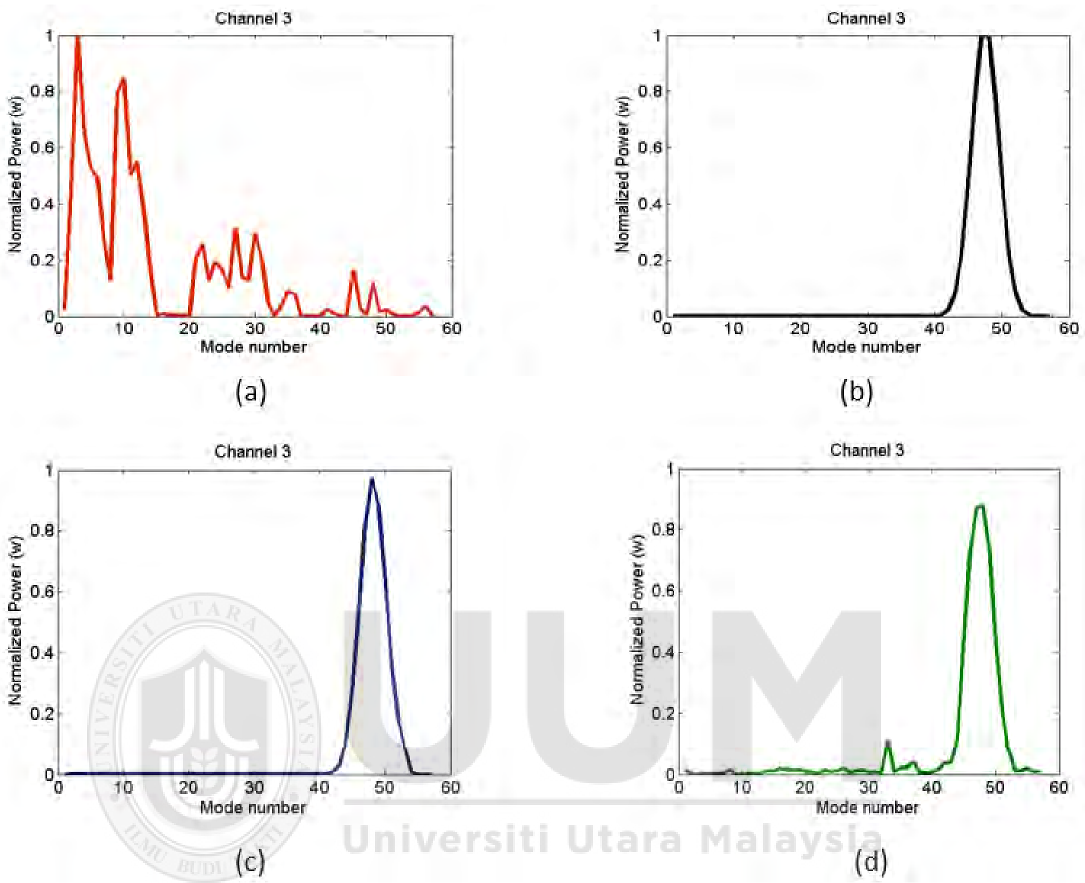


Figure 5.3: : Comparison of DENFIS and LMS equalization schemes for Channel 3: (a) distorted channel impulse response (b) Gaussian Pulse filter (c) equalized channel impulse response using LMS (d) equalized channel impulse response using DENFIS

5.2.2 DENFIS versus Recursive Least Squares Algorithm

RLS is an algorithm which recursively finds the filter coefficients that minimize a weighted linear least squares cost function relating to the input signals. This is in contrast to other algorithms, such as LMS, that aim to reduce the mean square error. The RLS has fast convergence. However, this benefit comes at the cost of high computational complexity and potentially poor tracking performance when the filter to be estimated changes [39].

As for this equalization scheme, the distorted channel impulse response and Gaussian pulse shaping filter for different three channels have been imported. The imported channels are the same channels that have been used in DENFIS proposed equalization scheme.

RLS has one parameter which is *Forgetfactor*. *Forgetfactor* is a real number between 0 and 1, and it is used for learning purpose [13] [77] as in Equation (5.3).

$$y(n) = \frac{1}{N}\sum_{j=0}^{n-1} \lambda^{i} e^{2}(n-1) \ldots\ldots (5.3)$$

Where N represents the filter length, e^2 is the instant value and λ is the forgetting factor. The same concept has been applied here that is the parameter has been changed to get the right result by applying the values of parameters from previous studies and using trial and error [13], as shown in Table 5.2.

Table 5.2: Recursive Least Squares results

	No. Of Weights	F.Factor	Initialization Parameter	RMSE (before)	RMSE (after)
Ch 1	15	0.6	0.612	0.2624195	0.1428285
Ch 2	15	0.9	0.112	0.2898327	0.2382519
Ch 3	9	0.1	0.912	0.3766430	0.1932356

The result of RLS revealed an improvement in channel impulse response for Channels, decreased RMSE and the position of different channels is almost within the time interval for each channel, as shown in Figures 5.4, 5.5 and 5.6. However, the nonlinear distortion represented by ISI still exists and the RMSE is still high after RLS equalization because the same channel remains high. This indicates that nonlinear distortion cannot be solved by using this algorithm.

For more details, the results of Channel 1 with the main parameters setting of Forget Factor and Initialization Parameter, as illustrated in Table 5.2, indicate that RMSE before equalization is minimized from (0.26241) to be (0.14282) after equalization. Furthermore, the shape of channel impulse responses is improved but still has some of the distortions that may increase the chances to face ISI among channels, as shown in Figure 5.4 (c). This reduction in RMSE is not vital enough to mitigate ISI and prevent the different channels from interfering each other.

However, RMSE value in DENFIS equalization for the same channel has more desirable results, and the shape of channel impulse response is more enhanced than the form of channel shape in LMS equalization. Furthermore, the value of RMSE before equalization is extracted by calculating the value of distorted channel impulse response with the value of Gaussian pulse filter and RMSE after equalization has been calculated by measuring the RMSE of equalized signal 1 with the Gaussian pulse filter.

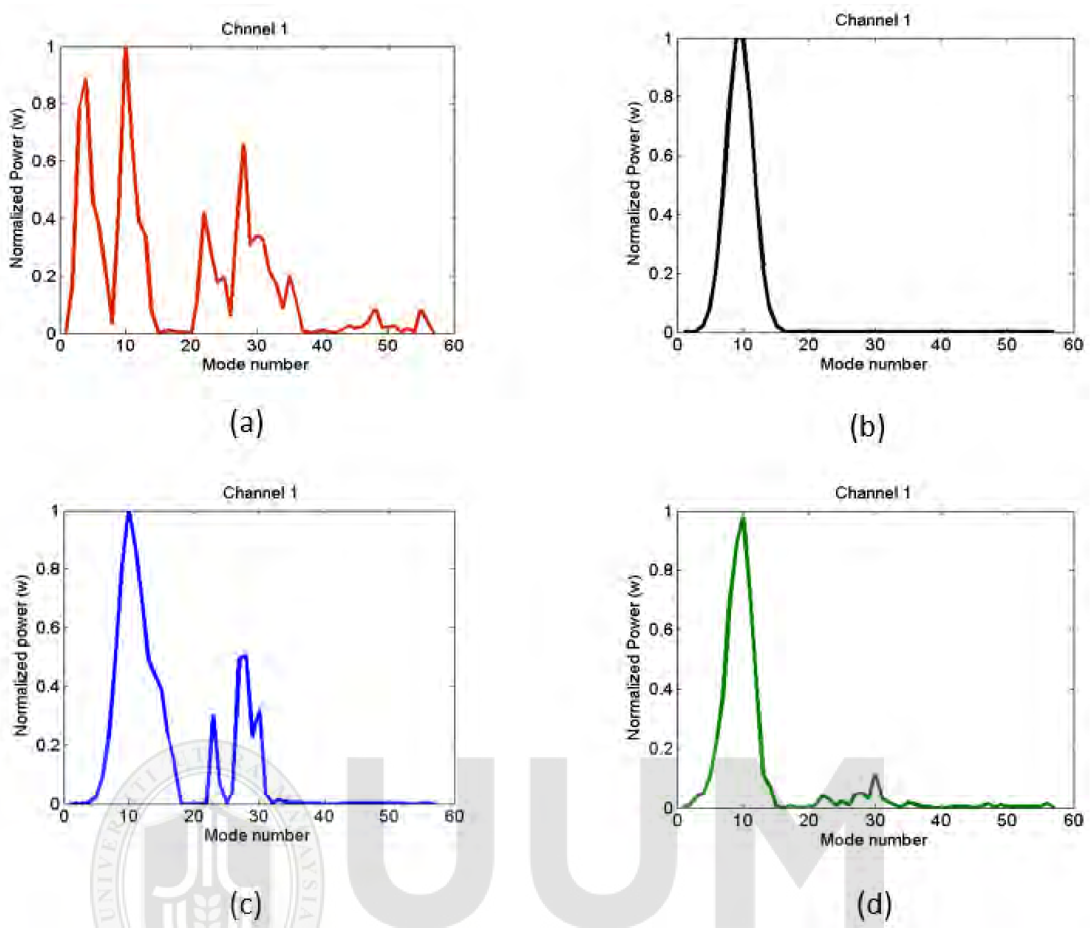


Figure 5.4: Comparison of DENFIS and RLS equalization schemes for Channel 1: (a) distorted channel impulse response (b) Gaussian Pulse filter (c) equalized channel impulse response using RLS (d) equalized channel impulse response using DENFIS

The result of Channel 2 has the best channel impulse response, as shown in Figure 5.5 (c). In contrast, Channel 3 has the worst channel impulse response with the main parameters, as illustrated in Table 5.2 and Figure 5.6 (c). However, DENFIS channel impulse response in Channel 2 has a better shape than RLS algorithm for the same channel, as illustrated in Figure 5.5 (d).

Table 5.2 shows that RMSE before equalization is minimized from (0.28983) and (0.37664) to be (0.23825) and (0.19323) after equalization respectively. The value of

RMSE before equalization is extracted by calculating the value of distorted channel impulse response with the value of Gaussian impulse filter. Then, RMSE after equalization has been calculated by measuring the RMSE of equalized channel impulse response for Channels 2 and 3 with the Gaussian pulse filter.

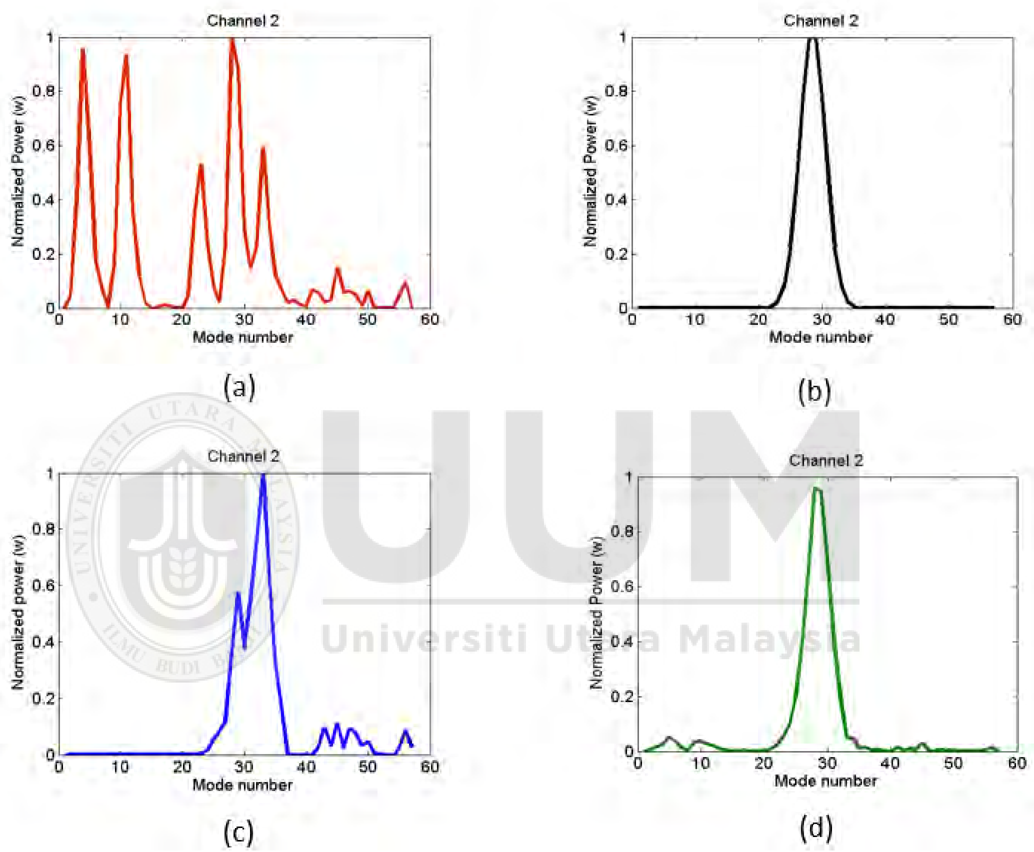


Figure 5.5: Comparison of DENFIS and RLS equalization schemes for Channel 2: (a) distorted channel impulse response (b) Gaussian Pulse filter (c) equalized channel impulse response using RLS (d) equalized channel impulse response using DENFIS

Although RLS algorithm provides better channel impulse response for the different three channels, yet the value of RMSE is not minimized significantly. This is because RLS algorithm works to minimize the weight, whereas DENFIS is working on update the rules or creating new ones. As a result, the appearance of ISI as a nonlinear distortion is resident. Furthermore, the interference between channels is the main obstacle to applying RLS to tackle the nonlinear distortions.

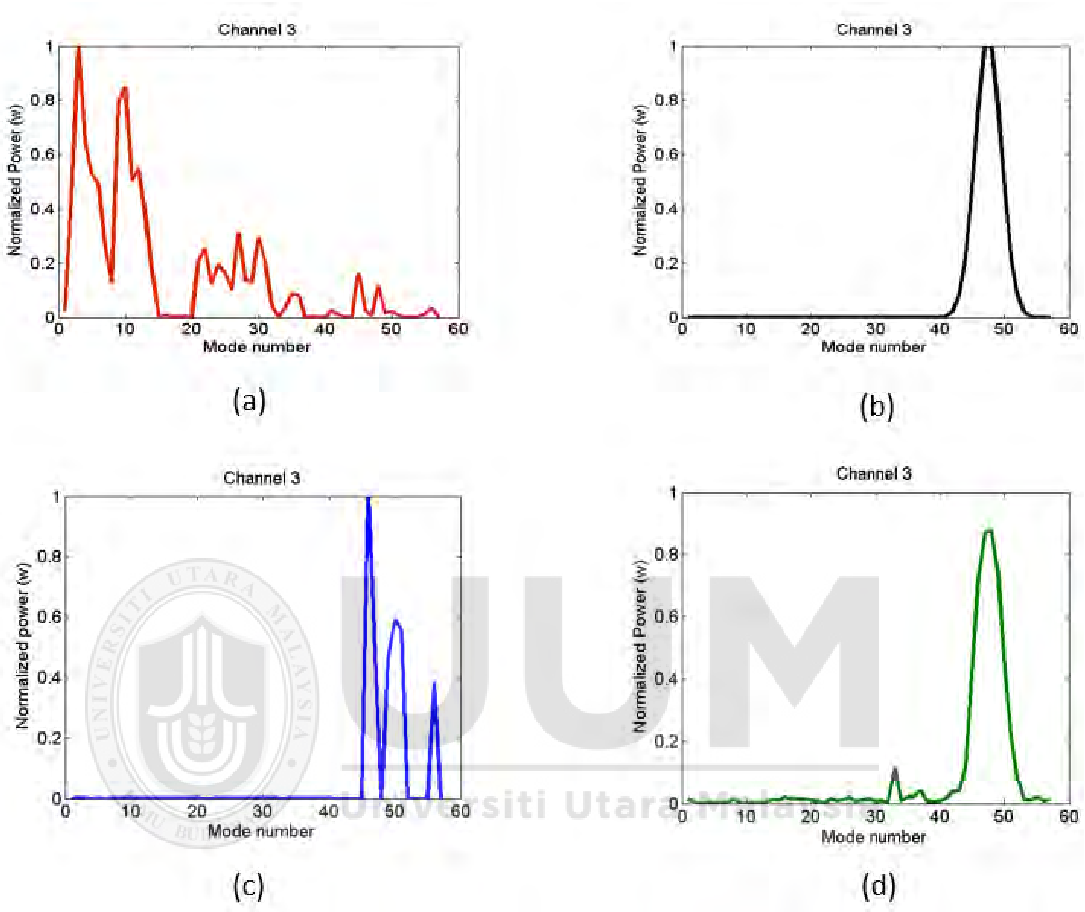
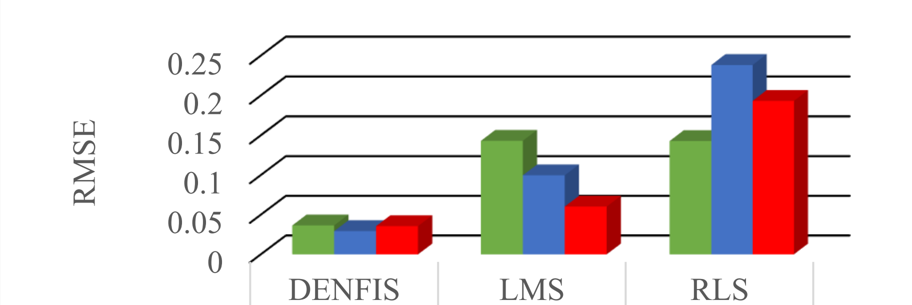


Figure 5.6: Comparison of DENFIS and RLS equalization schemes for Channel 3: (a) distorted channel impulse response (b) Gaussian Pulse filter (c) equalized channel impulse response using RLS (d) equalized channel impulse response using DENFIS

DENFIS result has been compared with LMS and RLS results to observe the effect of DENFIS equalization scheme to the RMSE and to achieve the third objective this research. The results of the comparison showed that DENFIS performs better than LMS and RLS in terms of RMSE rate for Channel 1, Channel 2 and Channel 3, as illustrated in Figure 5.7. In contrast, LMS has less performance than DENFIS but produced better results comparing with RLS, as shown in Figure 5.7. Moreover, DENFIS equalization scheme can reshape the channel impulse response of the received signal with the appearance of severe ISI in the channels to be almost like the transmitted channel impulse response.



	DENFIS	LMS	RLS
Channel 1	0.0365	0.1431	0.1428
Channel 2	0.0294	0.0997	0.2382
Channel 3	0.0356	0.0603	0.1932

Figure 5.7: RMSE comparison of DENFIS equalization scheme with respect to LMS and RLS equalization schemes

5.3 Summary

This chapter illustrated the achievement of the third objective of this research represented by the evaluation process of DENFIS equalization scheme and comparing it with another previous equalization scheme that has been implemented in MDM. In addition, the performance comparison of DENFIS equalization versus LMS and RLS equalization revealed that DENFIS equalization outperforms LMS and RLS equalization schemes in relation to the minimization of error between the distorted channel impulse response and the Gaussian pulse filter.

CHAPTER SIX
CONCLUSION

This chapter presented the conclusion, the limitation and future work of this research. Firstly, Section 6.1 illustrated the conclusion of this research. Then, Section 6.2 stated the limitation of this research. Finally, Section 6.3 discussed the recommendations for future work.

6.1 Conclusion

As have been mentioned in Chapter One, the main objectives of this research is to specify an equalization scheme for mode division multiplexing (MDM) system in multimode fiber (MMF). This study has been conducted to tackle the problem of nonlinear dispersion represented by mode coupling and inter-symbol interference (ISI) in MDM system over MMF. The equalization scheme has been illustrated in Chapter Four.

In this research, parameters for Gaussian pulse shaping filter have been determined and then used in dynamic evolving neural fuzzy inference system (DENFIS) equalization scheme. The shaping filter was used to recover the transmitted channel impulse response in MDM system from the distorted channel impulse response. The MDM model has been modeled to extract the distorted signal that was used to conduct this research. The DENFIS equalization scheme for MDM over MMF has been simulated and applied to the distorted signal that has been extracted from MDM model.

The results indicated that DENFIS as equalization scheme for the transmitted signal in MDM system over MMF is possible. According to the result, the proposed equalization scheme can reduce the amount of noise and errors that affect the transmitted signal which has been represented as root mean squared error (RMSE). Moreover, using DENFIS equalization scheme can reduce channel distortion and ISI, which is categorized as non-linear dispersion.

To evaluate the performance of DENFIS equalization scheme, two popular algorithms, namely least mean squares (LMS) and recursive least squares (RLS) have been suggested and implemented in MATLAB to make a comparison with DENFIS results. The result of comparison indicated that DENFIS equalization scheme has superior performance than other equalization schemes in terms of RMSE and a narrower channel impulse response.

Using such equalization scheme for MDM in MMF maybe will help to transfer signals in higher data rate with longer distance which might lead to adapt MMF for more applications in the future.

6.2 Limitations

The data of MDM system over MMF for DENFIS equalization scheme have been simulated instead of making a real experiment. Moreover, increasing the number of channels inside the MDM simulation and the data rate may provide more accurate analysis regarding the result of DENFIS equalization scheme. As for ISI mitigation, a real-time compensation of signal is required, but, in this research, only offline compensation was carried out.

6.3 Future Work

Implementing DENFIS equalization scheme in real experiment environment instead of the simulation environment will be an interesting topic in the future. The successful implementation of DENFIS equalization in a MDM system over MMF promotes more exploration of DENFIS equalization using larger data sets. Moreover, an online DENFIS equalization scheme is predictable for rapidly fluctuating nonlinear channel impairments.

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