"I admit that I had read this report and in my opinion this report is sufficient in the manner of scope and quality, to be awarded with a Bachelor Degree of Electronic and Computer Engineering"

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Date : 5 MAY 2006

DESIGN OF MULTIPLE DIFFERENTIAL RECEIVER FOR WIRELESS PERSONAL COMMUNICATION

This Report Is Submitted In Partial Fulfillment Of Requirements For The Bachelor Degree of Electronic Engineering (Computer)

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March 2006

"I admit that this report is done by my own effort except for the summary and statement which I had already mention the source from"

Signature : ....

Name : HASRUL' NISHAM BIN ROSLY

Date : 5 MAY 2006 . ...

Specially dedicated to:

My parents, my love and my friends for all of caring, love and support.

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I have had the opportunity to work with a great group of knowledgeable people that made my project a pleasurable and exciting experience. Their help and support throughout my Degree Final Project is greatly appreciated. Firstly, I would like to give a highest gratitude to my supervisor, Mr. Muhammad Syahrir bin Johal, for his help throughout the project especially during the documentation phase and always helping me work through difficulties and also not disregard thanks to my beloved parents for their non stop encouragement and moral support.

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ABSTRACT

This project deals with the design of multiple differential receiver for wireless personal communications. The receiver is intended to be able to perform signal processing on a corrupted signal using the *differential detection* scheme. The requirements for designing the entire receiver system involve software and hardware development. The type of encoding for the transmitted signal is in the form of Quadrature Phase Shift Keying (QPSK), which has been differentially shifted by 45 degree, leading it to be called pi/4-shift DQPSK. For the signal propagation through the channel, noises are introduced to cause distortion. This corrupted signal will be accepted by receiver in pi/4-shift DQPSK. Then, the received signal will be demodulated in the form of Pi/4-shift DQPSK in order to get the original signal form of QPSK. The corrupted signal from the transmitter will be filtered by filter. Software development is used to design the differential detection algorithm before designing additional peripheral circuit. Programming for the software is achieved in MATLAB. The type of differential detection scheme proposed is a multiple of a conventional differential detection. Simulation on the performance capabilities of both type of differential detection schemes were made. Results reveal the significant improvement of the multiple differential detection schemes over the convention one.

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CHAPTER 1

INTRODUCTION

1.1 Application Background

Through ages, man always tried to find a way for long distance communication. Man used pigeons, and men riding horses to deliver mail. These two methods were such inefficient. The first was limited to certain directions that the pigeons were trained to go to, and the latter took longer time for further distances. Since the early days of electronics, as advances in technology were taking place, the boundaries of both local and global communication began eroding, resulting in a world that is smaller and hence more easily accessible for the sharing of knowledge and information. The pioneering work by Bell and Marconi formed the cornerstone of the information age that exists today and paved the way for the future of telecommunications. These method were used till 19th century, when Alexander Graham Bell succeeded in performing the first wire telephone call (transmitting and electrical signal) in 1875. This invention revolutionized world communication.

Today, local communication was done over wires, as this presented a cost-effective way of ensuring a reliable transfer of information. For long-distance communications, transmission of information over radio waves was needed. Although this was convenient from a hardware standpoint, radio-waves transmission raised doubts over the corruption of the information and was often dependent on high-power transmitters to overcome weather conditions, large buildings, and interference from other sources of electromagnetics.

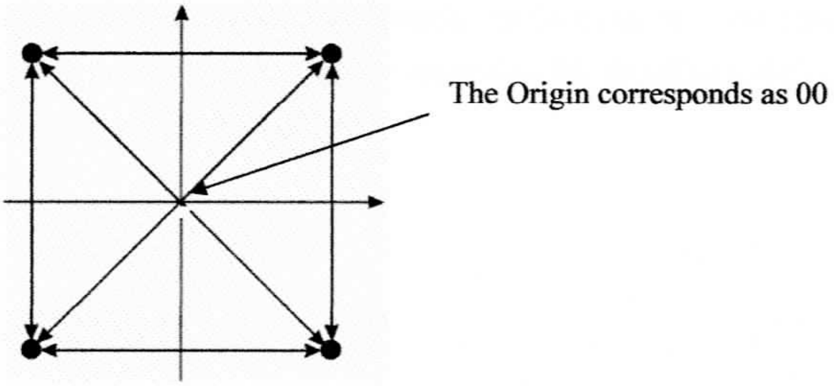
The various modulation techniques offered different solutions in terms of cost-effectiveness and quality of received signals but until recently were still largely analog. Frequency modulation and phase modulation presented a certain immunity to noise, whereas amplitude modulation was simpler to demodulate. However, more recently with the advent of low-cost microcontrollers and the introduction of domestic mobile telephones and satellite communications, digital modulation has gained in popularity. With digital modulation techniques come all the advantages that traditional microprocessor circuits have over their analog counterparts. Any shortfalls in the communications link can be eradicated using software. Information can now be encrypted, error correction can ensure more confidence in received data, and the use of DSP can reduce the limited bandwidth allocated to each service.

As with traditional analog systems, digital modulation can use amplitude, frequency, or phase modulation with different advantages. As frequency and phase modulation techniques offer more immunity to noise, they are the preferred scheme for the majority of services in use today and will be discussed.

1.2 Problem Statement

The use of wireless communication is increasing rapidly. While the available frequency band is very limited, new multiple access schemes in wireless communication are urgently needed. So, pi/4 Shift-DQPSK are one of the solution.

In QPSK modulation, phase ambiguity occurred. Data is read from one constellation to another in 90° or 180° as pictured in the QPSK constellation diagram. For instance, data read from a 01 to 10 will pass through the origin. At the receiver, the origin might be mistaken to correspond as 00. This lead to false information at the receiver.



QPSK constellation diagram

When receiver data in communication channel, noise will occur. This problem can make the data corrupted.

1.3 Project Objective

The objective of this project is to design the software and hardware for the multiple differential receiver. While doing this project, the author also wanted to overcome the problem mentioned by proposing pi/4 shift DQPSK modulation type. In QPSK modulation, phase ambiguity occurred. Data is read from one constellation to another in 90° or 180° as pictured in the QPSK constellation diagram. For instance, data read from a 01 to 10 will passes through the origin. At the receiver, the origin might be mistaken this data to correspond as 00.This lead to false information at the receiver. The author also want to analyze performance between QPSK and pi/4 Shift DQPSK. However, its performance is only moderate in a noisy environment. Therefore, the enhancement of the multiple differential receiver schemes into a multiple is presented in this project to test the capability in improving this decoding technique.

CHAPTER 2

LITERATURE REVIEW

2.1 Digital modulation

There are three major classes of digital modulation techniques used for transmission of digitally represented data:

- Amplitude-shift keying (ASK)
- Frequency-shift keying (FSK)
- Phase-shift keying (PSK)

All of those modulations convey data by changing some aspect of a base signal, the carrier wave, (usually a sinusoid) in response to a data signal. In the case of PSK, the phase is changed to represent the data signal. There are two fundamental ways of utilizing the phase of a signal in this way:

- By viewing the phase itself as conveying the information, in which case the demodulator must have a reference signal to compare the received signal's phase against; or

- by viewing the change in the phase as conveying information — differential schemes, some of which do not need a reference carrier (to a certain extent).

A convenient way to represent PSK schemes is on a constellation diagram. This shows the points in the Argand plane where, in this context, the real and imaginary axes are termed the in-phase and quadrature axes respectively due to their 90° separation. Such a representation on perpendicular axes lends itself to straightforward implementation. The amplitude of each point along the in-phase axis is used to modulate a cosine (or sine) wave and the amplitude along the quadrature axis to modulate a sine (or cosine) wave.

In PSK, the constellation points chosen are usually positioned with uniform angular spacing around a circle. This gives maximum phase-separation between adjacent points and thus the best immunity to corruption. They are positioned on a circle so that they can all be transmitted with the same energy. In this way, the moduli of the complex numbers they represent will be the same and thus so will the amplitudes needed for the cosine and sine waves. Two common examples are binary phase-shift keying (BPSK) which uses two phases, and quadrature phase-shift keying (QPSK) which uses four phases, although any number of phases may be used. Since the data to be conveyed are usually binary, the PSK scheme is usually designed with the number of constellation points being a power of 2.

2.2 Amplitude-shift keying (ASK)

Amplitude-shift keying (ASK) is a form of modulation which represents digital data as variations in the amplitude of a carrier wave. Here is a diagram showing the ideal model for a transmission system using an ASK modulation:

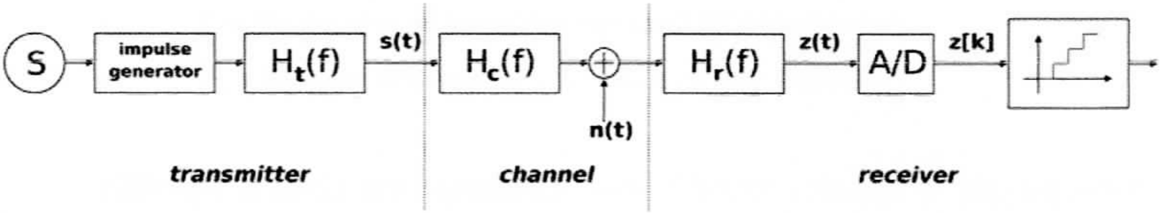


Figure 2.2 Transmission system using an ASK modulation

The simplest and most common form of ASK operates as a switch, using the presence of a carrier wave to indicate a binary one and its absence to indicate a binary zero. This type of modulation is called on-off keying, and is used at radio frequencies to transmit Morse code (referred to as continuous wave operation).

More sophisticated encoding schemes have been developed which represent data in groups using additional amplitude levels. For instance, a four-level encoding scheme can represent two bits with each shift in amplitude; an eight-level scheme can represent three bits; and so on. These forms of amplitude-shift keying require a high signal-to-noise ratio for their recovery, as by their nature much of the signal is transmitted at reduced power.

It can be divided into three blocks. The first one represents the transmitter, the second one is a linear model of the effects of the channel, the third one shows the structure of the receiver. The following notation is used:

- $h_t(t)$ is the carrier signal for the transmission
- $h_c(t)$ is the impulse response of the channel
- $n(t)$ is the noise introduced by the channel
- $h_r(t)$ is the filter at the receiver
- L is the number of levels that are used for transmission
- T_s is the time between the generation of two symbols

Different symbols are represented with different voltages. If the maximum allowed value for the voltage is A, then all the possible values are in the range $[-A,A]$ and they are given by:

$$v_i = \frac{2A}{L-1} i - A; \quad i = 0, 1, \ldots, L-1$$

the difference between one voltage and the other is:

$$\Delta = \frac{2A}{L-1}$$

Considering the figure, the symbols $v[n]$ are generated randomly by the source S, then the *impulse generator* creates impulses with an area of $v[n]$. These impulses are sent to the filter h_t to be sent through the channel. In other words, each symbol with a different carrier wave is sent with the relative amplitude.

Out of the transmitter, the signal *s(t)* can be expressed in the form:

$$s(t) = \sum_{n=-\infty}^{\infty} v[n] \cdot h_t(t - nT_s)$$

In the receiver, after the filtering through $h_r\ (t)$ the signal is:

$$z(t) = n_r(t) + \sum_{n=-\infty}^{\infty} v[n] \cdot g(t - nT_s)$$

where we use the notation:

$n_r(t) = n(t) * h_r(t)$

$g(t) = h_t(t) * h_c(t) * h_r(t)$

where * indicates the convolution between two signals. After the A/D conversion the signal *z[k]* can be expressed in the form:

$$z[k] = n_r[k] + v[k]g[0] + \sum_{n \neq k} v[n]g[k-n]$$

In this relationship, the second term represents the symbol to be extracted. The others are unwanted: the first one is the effect of noise, the second one is due to the intersymbol interference.

If the filters are chosen so that *g(t)* will satisfy the Nyquist ISI criterion, then there will be no intersymbol interference and the value of the sum will be zero, so:

$z[k] = n_r[k] + v[k]g[0]$

the transmission will be affected only by noise.

2.2.1 Probability of error

The probability density function to make an error after a certain symbol has been sent can be modelled by a Gaussian function; the mean value will be the relative sent value, and its variance will be given by:

$$\sigma_N = \int_{-\infty}^{+\infty} \Phi_N(f) \cdot |H_r(f)|^2 df$$

where $\Phi_N(f)$ is the spectral density of the noise within the band and $H_r\ (f)$ is the continuous Fourier transform of the impulse response of the filter $h_r\ (f)$.

The possibility to make an error is given by:

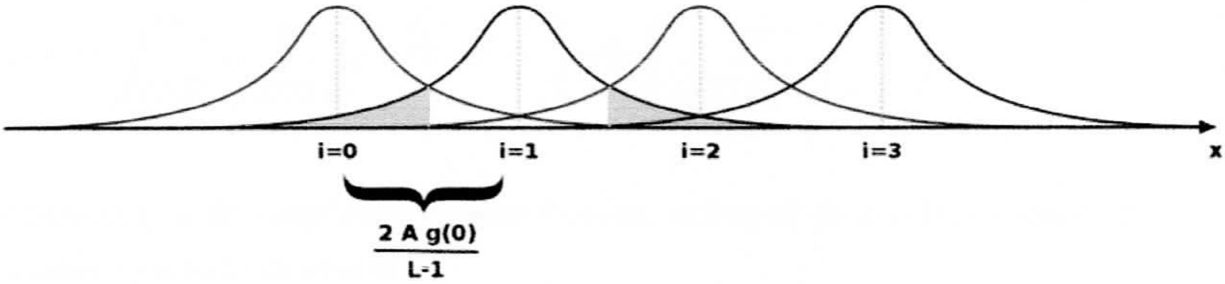
$$P_e = P_{e/H_0} \cdot P_{H_0} + P_{e/H_1} \cdot P_{H_1} + \ldots + P_{e/H_{L-1}} \cdot P_{H_{L-1}}$$

where $P_{e/H_0}$is the conditional probability of making an error after a symbol v_i has been sent and $P_{H_0}$is the probability of sending a symbol v_0.

If the probability of sending any symbol is the same, then:

$$P_{H_i} = \frac{1}{L}$$

If we represent all the probability density functions on the same plot against the possible value of the voltage to be transmitted, we get a picture like this (the particular case of L=4 is shown):



The possibility of making an error after a single symbol has been sent is the area of the Gaussian function falling under the other ones. It is shown in cyan just for one of them. If we call P^+ the area under one side of the Gaussian, the sum of all the areas will be: $2LP^+ - 2P^+$. The total probability of making an error can be expressed in the form:

$$P_e = 2\left(1 - \frac{1}{L}\right)P^+$$

We now have to calculate the value of P^+. In order to do that, we can move the origin of the reference wherever we want: the area below the function will not change. We are in a situation like the one shown in the following picture: