

**ANALYSIS OF PERFORMANCE FOR VARIOUS BANDWIDTH EFFICIENT
QAM SCHEMES**

P.KHIDMAT MAKLUMAT AKADEMIK
UNIMAS



1000126922

JELIHA BINTI BUJANG

This project is submitted in partial fulfilment of
the requirements for the degree of Bachelor of Engineering with Honours
(Electronic & Telecommunication Engineering)

Faculty of Engineering
UNIVERSITI MALAYSIA SARAWAK
2004

To my beloved father and mother, En Bujang Taha and Puan Sapiah Wahid;

My greatest motivation!.....

ACKNOWLEDGEMENT

First of all I would like to give my greatest appreciation to my supervisor, En Hushairi Zen for his continuous help, guidance, ideas and suggestion during the development of this final year project.

This appreciation also goes to the lecturers and the staffs of Faculty of Engineering UNIMAS.

To my family who always gives incredible support and wonderful advice.

Finally, to my friends; Halimah, Rosmelianny, Jam'aah, Norhidayah, Dahlia and Hasyimah for their understanding and tremendous support. Thank You!

ABSTRAK

Pada masa sekarang, keperluan untuk komunikasi digital pada kadar yang tinggi memberikan minat yang meluas terhadap teknik modulasi digital. Salah satu daripada teknik tersebut ialah *quadrature amplitude modulation* atau ringkasnya QAM. QAM adalah skim modulasi di mana dua pembawa *sinusoidal*, salah satu daripadanya keluar 90 darjah daripada fasa dengan merujuk kepada yang satu lagi digunakan untuk menghantar data pada saluran fizikal yang disediakan. QAM menggabungkan modulasi amplitud dengan modulasi anjakan kunci fasa bertujuan untuk memperbaiki kecekapan jalur pada kos persembahan *error probability*. Untuk projek akhir tahun ini, analisa telah dibuat untuk mengkaji persembahan pelbagai skim QAM. Skim ini terdiri daripada peringkat 16 *square* QAM, peringkat 16 *star* QAM dan *trellis code modulation*. Kemudian, perbandingan telah dibuat untuk menganalisa skim manakah yang terbaik dari segi persembahan BER dan kecekapan jalur. Selain itu, program simulasi telah dibuat menggunakan MATLAB dengan tujuan untuk membuat perbandingan bagi persembahan BER skim-skim tersebut.

ABSTRACT

Today, the necessity of digital communication at high rates has provided a great interest on digital modulation techniques. One of these techniques is quadrature amplitude modulation (QAM). QAM is a modulation scheme where two sinusoidal carriers, one exactly 90 degrees out of phase with respect to the other, are used to transmit data over a given physical channel. It combines amplitude modulation and phase shift keying modulation together to improve bandwidth efficiency at the expense of error probability performance. For this final year project, an analysis is made to study the performance of various QAM schemes. These schemes are 16-Square QAM, 16-Star Differential QAM and Trellis Code Modulation. Then, comparison is made to analyst which schemes that is superior in terms of its BER and bandwidth efficiency. Simulation program that compare the performance of BER for those schemes is developed using MATLAB.

Table of Content

	Page
Project title	
Dedication	
Acknowledgement	
Abstrak	i
Abstract	ii
Table of contents	iii
List of Figures	vi
List of Abbreviations	vii
Chapter 1	
Introduction	
1.1 An Overview	1
1.2 Objectives	3
1.3 Outline	4
Chapter 2	
Literature Review	
2.1 Elements of Digital Communication	5
2.2 Communication channels and their characteristics	7
2.2.1 Wireline channels	8
2.2.2 Fiber optic channels	9
2.2.3 Wireless electromagnetic channels	10
2.2.4 Underwater acoustic channels	10
2.2.5 Channel fading	10
2.3 Basic digital modulation techniques	11
2.3.1 Frequency shift keying	11
2.3.2 Phase Shift Keying	12
2.3.3 <i>M</i> -ary encoding	13
2.3.4 Quadrature Amplitude Modulation	15
2.3.4.1 The Principles of QAM	15
2.4 <i>M</i> -ary signal constellations	18
2.5 Square QAM	19
2.5.1 Basic concepts	20

2.6	Star QAM	20
2.7	Trellis Code Modulation	21
2.7.1	Key elements of TCM	22
2.7.2	Philosophy of TCM	23
2.7.3	Set Partitioning	23
Chapter 3		
	Methodology	26
3.1	Introduction	26
3.2	Bandwidth efficiency	26
3.2.1	Shannon Channel Coding Theorem	27
3.3	Probability of error	27
3.4	Q -Function	30
3.5	Probability of error for M -ary QAM	31
3.6	Probability of error for square QAM	36
3.7	Star QAM	40
3.7.1	Theoretical calculation of BER	40
3.8	Bit error probabilities from symbol error probabilities	41
3.9	Error probability estimation and free distance calculation for TCM	42
3.10	MATLAB program	43
3.10.1	Basic Plotting	44
3.10.2	Creating a Plot	44
3.10.3	Multiple Data Sets in One Graph	44
3.10.4	Legend	45
3.10.5	Logspace	45
3.10.6	Semilogx, Semilogy	45
3.10.7	xlabel, ylabel and zlabel	46
3.10.8	Grid	46
Chapter 4		
	Results and Discussion	47
4.1	Introduction	47
4.2	Probability of error versus E_b/N_0	47
4.2.1	16-square QAM	47
4.2.2	16 star QAM	48
4.2.3	BER performance for bandwidth QAM schemes (square QAM, star QAM and TCM)	50
4.2.4	BER performances for M -ary QAM	52
4.3	Bandwidth efficiency	54
Chapter 5		
	Recommendations and Conclusion	56

5.1	Conclusion	56
5.2	Recommendation	57
References		59
Bibliography		60
APPENDIX A		61
APPENDIX B		62
APPENDIX C		63
APPENDIX D		64
APPENDIX E		65
APPENDIX F		67

List of Figures

Figure		Page
2.1	Basic elements of a digital communication system	7
2.2	Frequency range for guided wire channel	9
2.3	(a) truth table; (b) phasor diagram; (c) constellation diagram	16
2.4	The constellation of star signal points	21
2.5	Trellis code modulation	22
2.6	Ungerboeck partitioning of 16-QAM signals	24
2.7	Uncoded 4-PSK and its one-state trellis diagram	25
3.1	The constellation of signal points	34
3.2	Type I, II, and III decision regions for 16-QAM	35
3.3	The constellation of signal points for 16-level square QAM	36
3.4	The constellation of signal points for 4-level PAM	37
4.1	The BER performance of 16 level square QAM	48
4.2	BER performances for 16-level QAM	50
4.3	BER performance for three bandwidth QAM schemes	51
4.4	BER performance for M-ary QAM	52
4.5	Symbol error probability versus E_b/N_0 for M-ary QAM	53
4.6	Bandwidth efficiency for M-ary QAM	55

List of Abbreviations

B

BPSK – Binary phase shift keying

BER – Bit Error Rate

C

CPFSK – Continuous-phase frequency shift keying

CW – Continuous Wave

F

FSK – Frequency Shift Keying

F_m – Mark Frequency

F_s – Space Frequency

f_b – Bit rate

FEC – Forward Error Correction

L

LED – Light Emitting Diode

M

MSK – Minimum shift-keying

N

NLF – Non-linear Filtering

O

OQPSK – Offset QPSK

P

PAM – **P**ulse **a**mplitude **m**odulation

PSK – Phase Shift Keying

PRK – Phase reversal keying

PLL – Phase Lock Loop

PCN – Personnel Communication Network

Q

QAM – Quadrature Amplitude Modulation

QPSK – Quaternary phase shift keying

S

SNR – Signal-to-Noise Ratio

T

TCM – Trellis Code Modulation

CHAPTER 1

INTRODUCTION

1.1 Overview

The purpose of communication system is to transfer information from one place to another. However, the criteria that is more concerned here is the transmission of information by electrical means via digital communication techniques. In the digital communications, the information is in digital form where it is represented in a series of discrete message.

In most communication systems, the main purpose is to use the bandwidth and transmitted power resources efficiently. Bandwidth efficiency is defined as the ratio of data rate to signal bandwidth whereas power efficiency is characterized by the probability of making a reception error as a function of signal-to-noise ratio [4]. Often, a system may be required to provide higher bandwidth efficiency but at the same time maintain the low error of transmission. Therefore, the type of modulation method is important in order to cater this requirement.

Mobile radio systems require bandwidth efficient digital modulation schemes because the available radio spectrum is limited [4]. Thus, in future, the available spectrum for the personal communications network (PCN) will be at a premium since the number of subscribers increased [6]. Therefore, a binary modulation technique is no longer appropriate and solutions to overcome these problems are introduced. One of the solutions that being considered is 16-level quadrature amplitude modulation (QAM). There are two

rules for mapping a 4 bit symbol onto two dimensional spaces. There are square mapping and star mapping.

Digital communication that transmits 16-level QAM signals over Rayleigh fading channels experience problems of having unacceptable high bit error rate (BER). This occurs because Phase Lock Loop (PLL) at the receiver is unable to track absolute phase during fades. It locks onto different quadrant than it required. In order to solve the problems, differential encoding is used because it reduced false locking. However, the standard square QAM constellation suffers false lock at positions 26° and 53° .

Star QAM is a constellation that does not have false locking position. By using star constellation, the nuisance that related to the standard square constellation is diminished. Star constellation is a twin 8-level PSK constellation and it allows differential encoding and decoding that mitigate the effect of Rayleigh fading. However, star QAM systems also encounter problems in Rayleigh fading. Error occurs when noise and changes of phasor amplitude or phase caused by fading are combined and force the incoming signal level above a decision boundary.

Another scheme that being considered is trellis code modulation. Trellis code modulation (TCM) was introduced by Ungerboeck by combining the coding and modulation in order to gain noise immunity without increasing the signal bandwidth [5]. One of the innovative aspects about TCM is the convolutional encoding and modulation is not being treated as separated entities but as a unique operation. Therefore, instead of demodulating the received signal first and then decoded, the signal is processed by combining the demodulation and decoding in a single step. Thus, the free Hamming distance of the convolutional code is not set as the parameter that governs the transmission system. Instead, it is using the free Euclidean distance between the transmitted signal

sequences. Therefore, the optimization of TCM design is based on the Euclidean distance and the selection of the code and the signal constellation will not be performed in separate steps.

Then, instead of using hard decision, the detection process will involve soft decisions. This means, the received signal will be processed before the decision of which transmitted symbols they are correspond to be made. TCM is commonly used in high-speed voice band modems. Without coding, high-speed modems achieved data rate up to 9600 bps with $M=16$ QAM signal constellation. The added coding gain provided by trellis coded modulation has made it possible to increase the speed of the transmission by at factor of 2 [5].

1.2 Objectives

Basically, the aim of this thesis is to study performance of the various bandwidth QAM schemes. The objectives of this project are as follows:

1. To be able to identify the digital modulation techniques such as frequency shift keying (FSK), phase shift keying (PSK) and quadrature amplitude modulation (QAM) and have the ability to distinguish those techniques. In addition, the different level of QAM is also studied and comparison is made between those levels.
2. To recognize and generalize the MATLAB software that being used for simulation purpose.
3. Develop programs that plot performances for various schemes based on the information from the research.

4. Make a comparison for various schemes in terms of their BER performance and bandwidth efficiency.

1.3 Outline

The thesis report is arranged as following:

Chapter 1 gives an overview of various bandwidth QAM schemes. It explains some of the basic characteristics of those schemes. Apart from that, this chapter also includes the objectives as well as the outline of the thesis.

Chapter 2 gives a literature review of basic digital communication such as AM/FM, PSK and QAM. It also covers more detail on various bandwidth efficient QAM schemes as well as different level of QAM. Three types of schemes that are being considered here are 16-level square QAM, 16-level star QAM and trellis code modulation. Some of their basic concept such as bandwidth efficiency, basic philosophy, probability of error and set partitioning are all covered in this chapter.

Chapter 3 explains the methodology that being used and explains the program in details. It also explains two parameters that are used for comparison purposes. There are BER performance and bandwidth.

Chapter 4 shows the equation for finding the probability of error for those schemes and the simulation results. Discussion is also included in this chapter.

Finally, conclusions and some recommendation about the future works are given as a summary in chapter 5.

CHAPTER 2

LITERATURE REVIEW

2.1 Elements of a Digital Communication System

Figure 2-1 shows the basic elements of a digital communication system. The source output is either an analog signal (audio or video signal) or a digital signal (binary coded number). In both cases, each message produced by the source is converted into a sequence of binary digit. The process of efficiently converting the output of either an analog or digital source into a sequence of binary digits is called source encoding or data compression. The sequence of binary digits (information sequence) from the source encoder is passed to the channel encoder. Channel encoder initiate redundancy in the binary information sequence at the receiver to overcome the effects of noise and interference in the transmission of the signal through the channel. Thus, the added redundancy increased both the reliability of the received data and signal. Redundancy added helps the receiver to decode the desired information sequence.

The binary sequence at the output of the channel encoder is passed to the digital modulator, which will be the interface to the communication channel. Since most of the communication channels in practice are capable of transmitting electrical signals (waveforms), the main purpose of the digital modulator is to map the binary information sequence into signal waveforms. For example, the coded information sequence is to be transmitted one bit at a time at some uniform rate R bits per second (bits/s). The digital modulator map the binary digit 0 into a waveform $s_0(t)$ and the binary digit 1 into a

waveform $s_1(t)$. Each bit from the channel encoder is transmitted separately. This is called binary modulation. Alternatively, the modulator transmit b coded information bits at a time by using $M = 2^b$ distinct waveforms $s_i(t)$, $i = 0, 1, \dots, M - 1$, one waveform for each of the 2^b possible b -bit sequences. This is called M -ary modulation ($M > 2$). [1]

The communication channel is the physical medium that is used to send the signal from the transmitter to the receiver [1]. In wireless transmission, the channel would be the atmosphere (free space). On the other hand, telephone channels typically employ a variety of physical media. This includes wire lines, optical fiber cables and wireless (microwave radio). Regardless of the physical medium that being used for the transmission, the transmitted signal will corrupted in a random manner by a variety of possible mechanisms such as additive thermal noise generated by electronic devices, man-made noise and atmospheric noise.

At the receiving end of a digital communication system, the digital demodulator processes the channel-corrupted transmitted waveform and reduces the waveforms to a sequence of numbers that represent estimates of the transmitted data symbols (binary or M -ary). This sequence of numbers is passed to the channel decoder. This will reconstruct the original information sequence from knowledge of the code used by the channel encoder and the redundancy contained at the received data.

The performance of the demodulator and decoder is measured by the frequency of errors occur in the decoded sequence. More precisely, the average probability of a bit-error at the output of decoder is a measure of the performance of demodulator-decoder combination. Generally, the probability of error is a function of the code characteristics, the types of waveforms used to transmit the information over the channel, the transmitter

power, and the characteristics of the channel and the method of demodulation and decoding.

Finally, when an analog output is desired, the source decoder accepts the output sequence from the channel decoder and by using the knowledge of the source encoding method; it will reconstruct the original signal from the source. Because of channel decoding errors and possible distortion introduced by the source encoder and decoder, the signal at the output of the source decoder is an approximation to the original source output. The difference between the original and reconstructed signal is a measure of the distortion introduced by the digital communication system.

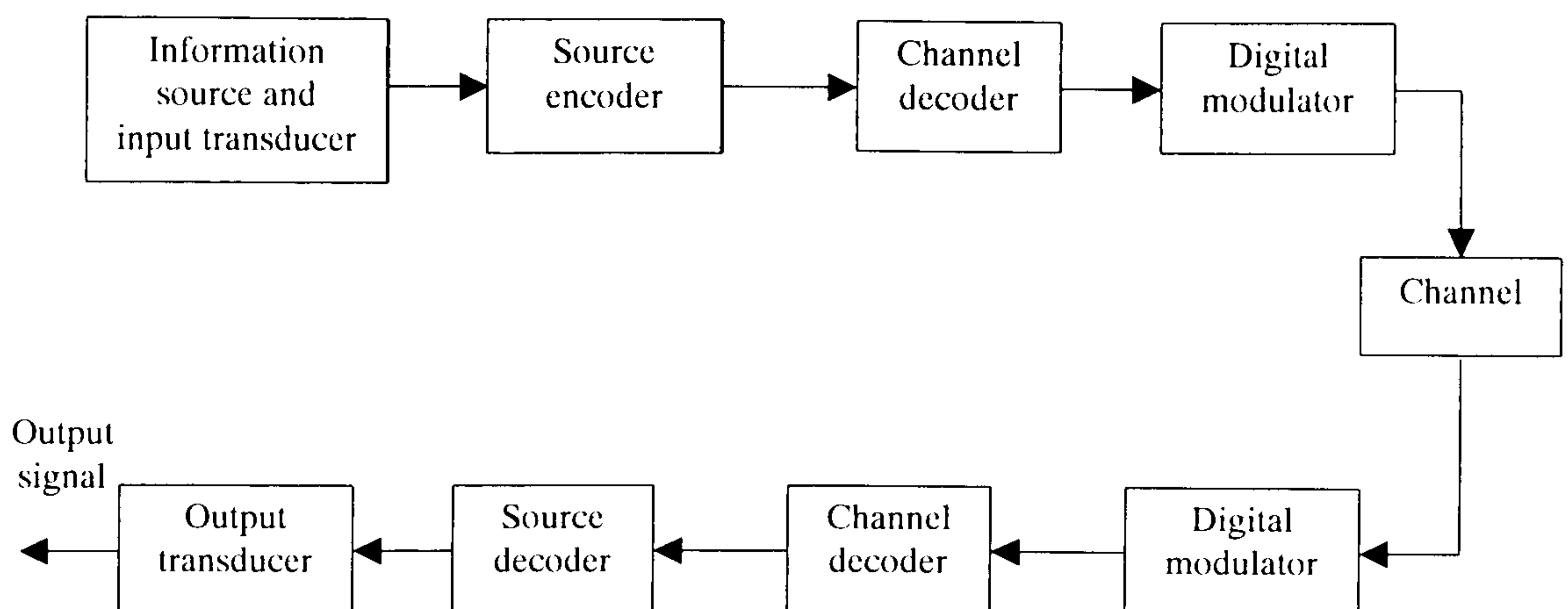


Figure 2-1 Basic elements of a digital communication system

(Adapted from: Proakis, John G. (2001). *Digital Communications*. McGraw-Hill)

2.2 Communication channels and their characteristics

As described earlier, the communication channel provides the connection between the transmitter and the receiver. The physical channel would be a pair of wires that carry the electrical signals, or an optical fiber that carries information on a modulated light beam,

or an underwater ocean channel in which the information is transmitted acoustically, or free space over which the information-bearing signal is radiated by use of an antenna.

One regular problem in signal transmission through any channel is additive noise. This noise is generated within the components. This noise is also known as thermal noise. Other sources of noise and interference can occur externally to the system. For example, the interference from other users of the channel. The effect of noise and interference can be minimized by proper design of the transmitted signal and its demodulator at the receiver. Other types of signal degradations that occur in the transmission over the channel are signal attenuation, amplitude and phase distortion and multipath distortion.

Increasing the power in the transmitted signal can minimize the effect of noise. However, equipment and other practical restriction limit the power level in the transmitted signal. Another basic limitation is the available channel bandwidth. A bandwidth restriction is usually due to the physical limitations of the medium and the electronic components used to implement the transmitter and the receiver. These two limitations restrict the amount of data that can be transmitted reliably over any communication channel.

2.2.1 Wireline channels

A wire line is use for voice signal transmission as well as data and video transmission. Twisted pair wire lines and coaxial cable guided electromagnetic channels that provide modest bandwidth. Telephone wire has a bandwidth of several kilo-Hertz (kHz) while coaxial cable has usable bandwidth of several mega-Hertz (MHz) [1]. Figure 2-2 shows the frequency range of guided electromagnetic channels that provide waveguides and optical fibers. Signals transmitted through such channels are distorted in

both amplitude and phase and further corrupted by additive noise. Twisted-pair wireline channels are also prone to crosstalk interference from physically adjacent channels.

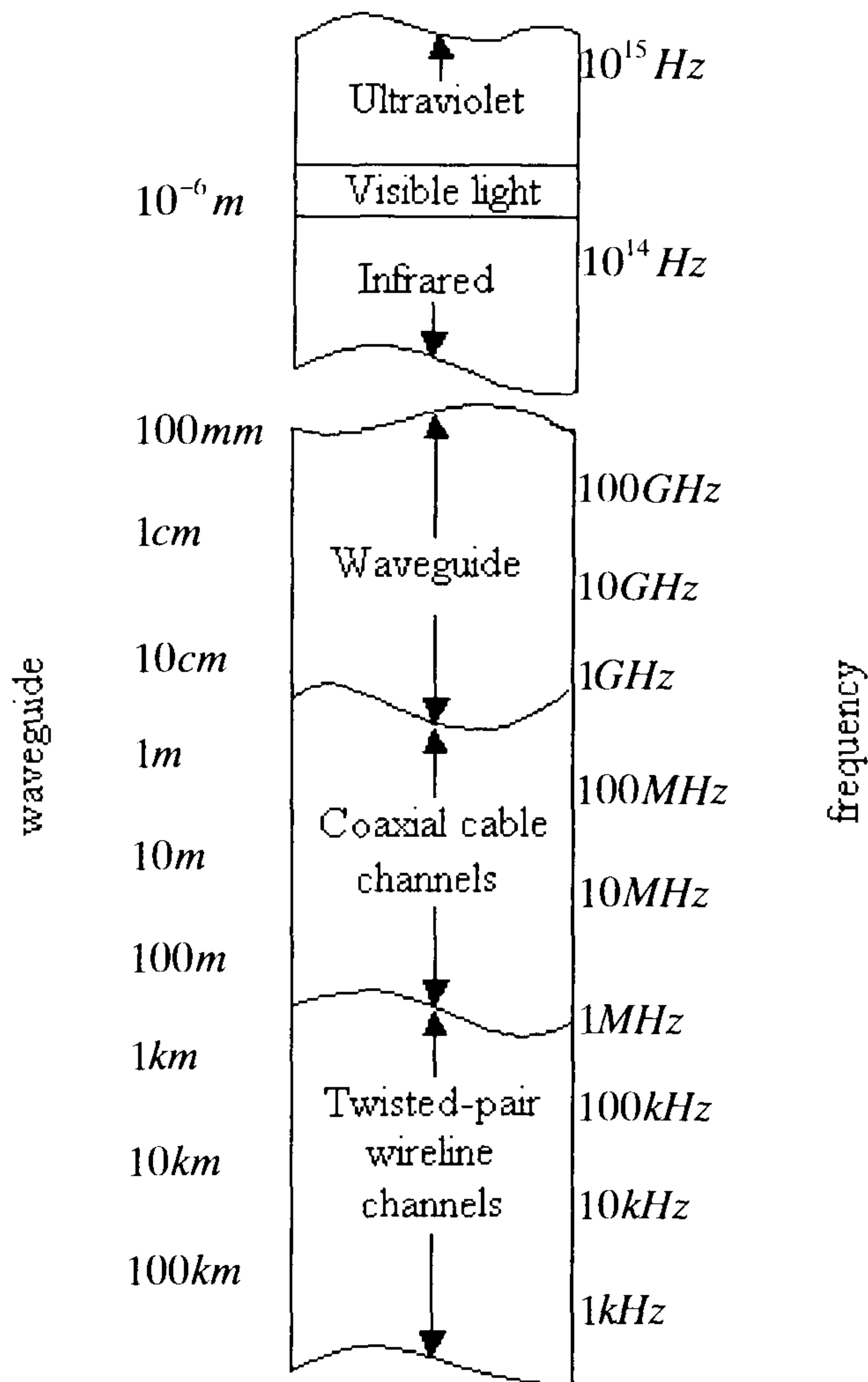


Figure 2-2 Frequency range for guided wire channel

(Adapted from: Proakis, John G. (2001). *Digital Communications*. McGraw-Hill)

2.2.2 Fiber optic channels

Optical fibers offer a channel bandwidth that is several orders of magnitude larger than coaxial cable channels. The transmitter or modulator in a fiber optic is a light source,

either light-emitting diode (LED) or a laser [1]. Information is transmitted by varying (modulating) the intensity of light source with the message signal.

2.2.3 Wireless electromagnetic channels

In wireless communication systems, electromagnetic energy is coupled to the propagation medium by an antenna, which serves as the radiator. The physical size and the configuration of the antenna depend primarily on the frequency of operation.

2.2.4 Underwater acoustic channels

An underwater acoustic channel is characterized as a multipath channel due to signal reflections from the surface and the bottom of the sea. Because of wave motion, the signal multipath components undergo time-varying propagation delays that result in signal fading. [1]

2.2.5 Channel fading

Fading means rapid fluctuations of amplitude of a radio signal over short period of time or distance. It happened in multipath waves. Fading can be categorized into two categories. First category is called large-scale fading. This happened due to the motion over large area (hills or forest). Large scale fading can be described in terms of a mean path loss with the addition of variation around the mean. Whereby the second category called small-scale fading, also known as Rayleigh fading, happened due to small changes in positions. The degrading effects of a Rayleigh fading channel are due to the superposition of multiple, time varying propagation paths. The impairments manifest themselves as deep fades in both the frequency and time domains [6].

Rayleigh fading occurs on time-varying multipath channels. This occurs when the medium is time varying, such as under-sea acoustic transmission, radio transmission through the upper atmosphere and indoor radio transmission where moving people cast shadows or it can occur when the transmitter and receiver are in motion as in mobile radio [8]. Because the distances along the multiple propagation paths are changing, the receiver observes multiple Doppler shifted versions of the transmitted signal

2.3 Basic digital modulation techniques

The basic techniques for transmission of digital data through communication channels are introduced in this chapter. Basically, there are three digital modulation techniques that are commonly used in digital communication systems. There are frequency shift keying (FSK), phase shift keying (PSK) and quadrature amplitude modulation (QAM).

2.3.1 Frequency shift keying

Frequency shift keying is a simple, low-performance form of digital modulation. Binary FSK is a form of constant-amplitude angle modulation similar to conventional frequency modulation except that the modulating signal is a binary pulse stream that differs between two discrete voltage levels rather than a continuously changing analog waveform [3]. The general expression for a binary FSK signal is

$$v(t) = V_c \cos\left[\left(\omega_c + \frac{V_m(t)\Delta\omega}{2}\right)t\right] \quad (2-1)$$

where

V_c = binary FSK waveform

$v(t)$ = peak unmodulated carrier amplitude

ω_c = radian carrier frequency

$V_m(t)$ = binary digital modulating signal

$\Delta\omega$ = change in radian output frequency

Minimum shift-keying FSK

Minimum shift-keying FSK (MSK) is a form of continuous-phase frequency shift keying (CPFSK) [4]. Basically, MSK is binary FSK except that the mark and space frequencies are synchronized with the input binary bit rate. This means that there is a precise time relationship between the two but it does not mean they are equal. With MSK, the mark and space frequencies are selected in such a way that they are separated from the center frequency by an exact odd multiple of one-half of the bit rate [f_m and $f_s = n (f_b/2)$, where f_m = mark frequency, f_s = space frequency and n = any odd number]. This is to make sure the smooth phase transition in the analog output signal when it changes from a mark to a space frequency, or vice versa, there is a sudden phase discontinuity in the analog output signal. When this happens, the demodulator has difficulty following the frequency shift. As a result, an error may happen.

2.3.2 Phase Shift Keying

Phase shift keying is another form of angle modulated, constant amplitude digital modulation [3]. PSK is similar to conventional phase modulation except that with PSK the input signal is a binary digital signal and limited number of output phases is possible.

Binary phase shift keying

In binary phase shift keying (BPSK), two output phases are possible for a single carrier frequency (“binary” meaning “2”). One output phase represents logic 1 and the other logic 0. As the input digital signal changes state, the phase of the output carrier shifts between two angles that are 180° out of phase. Other names for BPSK are phase reversal keying (PRK) and biphase modulation. BPSK is a form of suppressed carrier, square wave modulation of a continuous wave (CW) signal. [4]

2.3.3 *M*-ary encoding

M-ary is a term derived from the word “binary”. *M* is a digit representing the number of conditions possible. So far, the two digital modulation techniques that being discussed (binary FSK and BPSK) are binary systems. They only have two possible output conditions. One represents logic 1 and the other logic 0 [2]. Thus, they are *M*-ary systems where $M = 2$. In digital modulation, it is better to encode at a level higher than binary. For example, a PSK system with four possible output phases is an *M*-ary system where $M = 4$. If there were eight possible output phases, $M = 8$, and so on. Mathematically,

$$N = \log_2 M$$

where

N = number of bits

M = number of output conditions possible with N bits

Quaternary phase shift keying

Quaternary phase shift keying (QPSK) or quadrature PSK is another form of angle modulated, constant digital. QPSK is an *M*-ary encoding technique where $M = 4$.