# UNIVERSITI PUTRA MALAYSIA

# ADAPTIVE MODULATION AND CODING USING SIGNAL TO NOISE RATIO SWITCHING THRESHOLD

FAEZAH JASMAN



# ADAPTIVE MODULATION AND CODING USING SIGNAL TO NOISE RATIO SWITCHING THRESHOLD

**BY**

**FAEZAH JASMAN**

**Thesis Submitted to the School of Graduate Studies, University Putra Malaysia, in Fulfilment of the Requirement for the Degree of Master of Science**

**June 2009**

# DEDICATION

This thesis is dedicated to

ALL I LOVE

Specially

MY BELOVED PARENTS

Abstract of thesis presented to the Senate of Universiti Putra Malaysia in fulfilment of the requirement for the degree of Master of Science

**ADAPTIVE MODULATION AND CODING USING SIGNAL TO NOISE RATIO SWITCHING THRESHOLD**

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**June 2009**

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**Faculty : Engineering**

Orthogonal frequency division multiplexing (OFDM) is one of the key enabling technologies for fourth generation (4G) wireless system. It offers high data rate transmission with high spectral efficiency, immunity to multipath fading and simple implementation using fast Fourier transform (FFT). However inefficient utilization of the channel will result when OFDM system is designed for worst-case channel conditions. Thus adaptive transmission scheme that can be adjusted to channel conditions is one of the techniques to improve the performance of OFDM systems.

In this thesis, three types of subband adaptive transmission scheme namely adaptive modulation (AM), adaptive coding (AC) and adaptive modulation and coding (AMC) based on SNR switching threshold are investigated. The performances of these systems are evaluated using an efficient adaptation algorithm. The efficient adaptation algorithm is based on the average value of the SNR of the subcarriers in the subband.

First the performance of adaptive modulation using quadrature amplitude modulation (QAM) and phase shift keying (PSK) system are evaluated. The results obtained showed that a significant improvements in terms of bit error rate (BER), spectral efficiency and throughput can be achieved. To further enhance the system, convolutional coding is employed. However convolutional coding causes the maximum throughput to be limited. To solve this problem adaptive coding schemes which provides another area of flexibility is investigated. Finally the combination of adaptive modulation and adaptive coding is examined.

Simulations results have shown that the performance of adaptive transmission schemes are superior compared to fixed (nonadaptive) transmission schemes. A high throughput performance can be achieved without sacrificing the BER. The performance comparisons of the proposed system with the conservative system showed that the proposed scheme is able to meet the BER target of $10^{-3}$ with a slightly better throughput performance around 0.3 Bps/Hz.

Abstrak tesis yang dikemukakan kepada Senat Universiti Putra Malaysia
sebagai memenuhi keperluan untuk ijazah Master of Sains

**SKEMA MODULASI DAN PENGEKODAN ADAPTIF MENGGUNAKAN HAD MINIMUM NISBAH ISYARAT TERHADAP HINGAR**

Oleh

**FAEZAH JASMAN**

**Jun 2009**

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Pemultipleksan pembahagi frekuensi ortogonal adalah satu daripada teknologi utama untuk sistem tanpa wayar generasi ke empat (4G). Ia menawarkan penghantaran data yang tinggi dengan kecekapan yang tinggi, kekebalan terhadap pemudaran berbilang laluan dan pelaksanaan mudah menggunakan penjelmaan Fourier pantas (FFT). Namun penggunaan laluan yang kurang efisien akan berlaku apabila sistem tersebut digubah untuk keperluan laluan minimum. Oleh itu teknologi penghantaran adaptif yang berupaya untuk diubahsuai terhadap keadaan mutu isyarat merupakan salah satu cara untuk meningkatkan kualiti sistem OFDM.

Di dalam tesis ini tiga jenis komunikasi adaptif akan dibincangkan iaitu modulasi adaptif, pengekodan adaptif dan juga modulasi dan pengekodan adaptif menggunakan algoritma penyesuaian yang dicadangkan. Algoritma penyesuaian yang dicadangkan adalah berasaskan nilai purata nisbah isyarat terhadap hingar (SNR) yang dikira daripada subpembawa di dalam subjalur.

Pertama sekali modulasi adaptif dibincangkan dengan menggunakan sistem permodulatan amplitude kuadratur (QAM) dan juga kekunci anjak fasa (PSK). Hasil kajian menunjukkan bahawa pencapaian yang baik dari segi kadar ralat bit (BER) dan kadar penerimaan data diperolehi. Selepas itu teknik pengekodan digunakan untuk menambahbaikkan pencapaian sistem ini. Walau bagaimanapun, teknik pengekodan telah menyebabkan kadar penerimaan data maksimum dihadkan. Seterusnya teknik pengekodan adaptif dikaji sebagai salah satu cara untuk mengatasi masalah ini. Akhirnya teknik menggabungkan modulasi adaptif dan pengekodan adaptif digabungkan untuk memaksimumkan pencapaian.

Hasil daripada simulasi yang dijalankan, menunjukkan bahawa pencapaian skema transmisi adaptif adalah lebih baik berbanding skema transmisi tetap. Kadar penerimaan data yang tinggi dapat dicapai tanpa perlu mengorbankan nilai kadar ralat data (BER). Perbandingan antara skema yang dicadangkan dengan skema yang lazim telah menunjukkan bahawa skema yang dicadangkan mampu menepati nilai BER yang ditetapkan di samping nilai kadar data  yang lebih baik sebanyak 0.3 Bps/Hz.

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This thesis submitted to the Senate of Universiti Putra Malaysia and has been accepted as fulfillment of the requirement for the degree of Master of Science.
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## DECLARATION

I hereby declare that the thesis is based on my original work except for quotations and citations which have been duly acknowledged. I also declare that it has not been previously or concurrently submitted for any other degree at UPM or other institutions.

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**FAEZAH JASMAN**

Date: 11 August 2008

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| | |
|---|---|
| **3G** | Third Generation |
| **4G** | Fourth Generation |
| **AC** | Adaptive Coding |
| **ADC** | Analog to Digital Converter |
| **ADSL** | Assymetric Digital Subscriber Line |
| **AM** | Adaptive Modulation |
| **AMC** | Adaptive Modulation and Coding |
| **AMPS** | Advance Mobile Phone Service |
| **AWGN** | Additive White Gaussian Noise |
| **BER** | Bit Error Rate |
| **BPSK** | Binary Phase Shift Keying |
| **CDMA** | Code Division Multiple Access |
| **CDMA 2000** | Code Divison Multiple Access 2000 |
| **DAB** | Digital Audio Broadcast |
| **dB** | Decibel |
| **DS/SS** | Direct Sequence Spread Spectrum |
| **DS-CDMA** | Direct Sequence Code Division Multiple Access |
| **DSL** | Digital Subscriber Line |
| **DVB-T** | Terrestial Digital Video Broadcast |
| **EDGE** | Enhanced Date Rates for GSM Evolution. |
| **FDCHTF** | Frequency Domain Channel Transfer Function. |
| **FDD** | Frequency Division Duplexing |
| **FDM** | Frequency Division Multiplexing |
| **FDMA** | Frequency Division Multiple Access |
| **FEC** | Forward Error Control |
| **FFT** | Fast Fourier Transform |
| **FH/SS** | Frequency Hoping Spread Spectrum |
| **GPRS** | General Packet Radio Service |
| **GSM** | Global System for Mobile communication |
| **GSTN** | General Switched Telephone Network |
| **HDSL** | High Bit Rate Digital Subscriber Line |
| **HSDPA** | High Speed Data Packet Access |
| **HYPERLAN2** | High Performance Local Area Network |
| **IFFT** | Inverse Fast Fourier Transform |

| | |
|---|---|
| **IMMT** | International Mobile Telecommunications |
| **IMT2000** | International Mobile Telecommunications 2000 |
| **IS-95** | Interim Standard 95 |
| **ISI** | Inter Symbol Interference |
| **LAN** | Local Area Network |
| **MAC** | Media Access Control |
| **MCS** | Modulation and Coding Scheme |
| **MIMO** | Multiple Input Multiple Output |
| **MMAC** | Mobile Multimedia Access Communication |
| **OFDM** | Orthogonal Frequency Division Multiplexing |
| **PDC** | Personal Digital Communication System |
| **PSK** | Phase Shift keying |
| **QAM** | Quadrature Amplitude Modulation. |
| **SNR** | Signal to Noise Ratio |
| **TDD** | Time Division Duplexing |
| **TDMA** | Time Division Multiple Access |
| **TH/SS** | Time Hoping Spread Spectrum |
| **WCDMA** | Wideband- Code Division Multiple Access |
| **MPSK** | M-ary Phase shift keying |
| **MQAM** | M-ary Quadrature Amplitude Modulation |

# CHAPTER 1

# INTRODUCTION

## 1.1 Background

Wireless communication has become one of the most rapidly growing industries in the world. This tremendous growth in demand for a global system that support interactive multimedia service has gives birth to the 4$^{th}$ generation (4G) mobile system. Technically, 4G stands for one integrated, IP-based environment for all telecommunication requirements including voice, video, broadcasting media and Internet that utilizes both fixed and wireless networks [Govil 07].

Several technologies that are essential in the success of the 4G systems include Orthogonal Frequency Division Multiplexing (OFDM), multiple input/multiple output (MIMO), universal mobile telecommunication system (UMTS) and Time Division Synchronous Code Division Multiple Access (TDSCDMA). All of these methods are typified by high rates of data transmission and packet-switched transmission protocols [Govil 08].

OFDM is a special form of multi-carrier transmission where all the subcarriers are orthogonal to each other. OFDM technologies are able to deliver high data rate by splitting the data into a number of lower rate streams that are transmitted

simultaneously. The main advantage of OFDM is its ability to deal with multipath fading and narrowband interference without using complicated channel equalization.

Another approach to increase the data rate and spectral efficiency is adaptive transmission techniques or also known as link adaptation. The essence of adaptive transmission techniques is to dynamically adjust the transmission parameters such as modulation level, code rate, symbol rate, weight of transmission antenna, transmitted power or any combination of these parameters according to the channel condition. Few examples of adaptive transmission techniques are adaptive modulation, and adaptive coding and adaptive power control [Goldsmith 05].

By taking advantage of the time varying nature of the wireless fading channel, all these adaptive techniques are trying to use both power and spectrum more efficiently to realize a higher bit rate transmission without sacrificing the Bit Error Rate (BER) performance. For example, adaptive modulations method can provide higher bit rates relative to fixed (nonadaptive) transmission systems by transmitting at a high rate under favorable channel conditions. When channel conditions are poor, energy efficient schemes such as BPSK or QPSK are used. As channel quality improves, 16 QAM or 64 QAM is used.

To further enhance the communication system, the design of the adaptive transmission schemes is integrated with OFDM [Cao 06]. Hence in this thesis, an investigation of adaptive transmission based on OFDM system will be conducted and the results will be presented at the end of the work.

## 1.2 Problem Statement and Motivation

The fundamental limitation of wireless systems is constituted by their time and frequency domain channel fading which cause fixed (nonadaptive) mode transceivers to suffer from burst of transmission errors. Normally the design of a fixed transmission system is based on worst case channel condition. Since Rayleigh fading can cause a signal power loss of up to 30 dB, designing for the worst-case channel conditions can result in very inefficient utilization of channel [Goldsmith 05]. Furthermore in fixed (nonadaptive) transmission schemes the bandwidth is wasted for good channel conditions since the transmitter and receiver are not optimized for current channel conditions [Hanzo 02].

Specifically in OFDM, the BER of different subcarriers transmitted in time dispersive channels depends on the frequency domain channel transfer function. The bit errors are normally concentrated in a few severely faded subcarriers. In the rest of the subcarriers , there are normally no bit errors. If the high BER subcarriers can be identified and applied with more powerful forward error correction codes, the overall bit error rate of the whole OFDM frame will be improved. Since the frequency domain fading deteriorates the SNR of certain subcarriers, but improves others above the average SNR value, the potential loss of throughput due to the exclusion of faded subcarriers can be mitigated by employing higher order modulation modes on the subcarriers exhibiting high SNR values.

The current OFDM systems such as HIPERLAN/2, IEEE 802,11a and IEEE 802.16e does not have the capability to adapt to the variation of the fading properties of the individual subcarriers. Only one code or modulation level is employed depending on the average Carrier to Interference ratio (C/I) [Askary 06]. Hence the subcarriers with low C/I do not have have sufficient error correcting capabilities resulting in multiple errors at the receiving end. Since the same number of bits is used for every subcarrier, the signal spectrum is also not optimized.

Motivated by the above mentioned performance limitations of fixed OFDM systems, an extensive research has been conducted on adaptive transmission for OFDM systems. Specifically three types of adaptive transmission schemes namely adaptive modulation, adaptive coding and adaptive modulation and coding are discussed. In adaptive modulation, the spectral efficiency will increase with increasing channel SNR while in fixed modulation the spectral efficiency is constant. Thus the average spectral efficiency of the adaptive scheme is improved while at the same time the BER is better suited to the requirement of the application. Thus the adaptive link becomes much more efficient for data transmission [Svensson 07].