Adaptively Modulating Multiple User on OFDM Using Multiple Concatenated Codes

by

Beenish Niaz

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i Orthogonal frequency division multiplexing
ii Mobile Communication System
iii Wireless communication system

Certificate

It is certified that the research work contained in this thesis by Beenish Niaz [137-FET/MSEE/F07], has been carried out under my supervision. In my opinion it is fully adequate, in scope and quality, as a thesis for the degree of Master of Science in Telecommunication Engineering awarded by International Islamic University, Islamabad.

Supervisor
Dr. Ijaz Mansoor Qureshi
Dean, FET, IIU Islamabad

External Examiner
Dr. Abdul Jalil
Associate Professor, PIEAS Islamabad

Internal Examiner
Dr. Aqdas Naveed Malik
Assistant Professor, IIU Islamabad

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Beenish Niaz
137-FET/MSEE/F07

بسم الله الرحمن الرحيم

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Dedicated to my Mother.

Abstract

Recently, orthogonal frequency-division multiplexing (OFDM) has received a considerable amount of interests for high-rate wireless communications. Because OFDM increases the symbol duration and transmitting data in parallel, it has become one of the most effective modulation techniques for combating multipath delay spread over mobile wireless channels. In this thesis, the problem of Adaptive Coded Modulation through an OFDM system over parallel flat fading channels is considered.

In this thesis, an adaptive coding scheme for OFDM based systems using *Multiple Concatenated Codes(MCC)* is presented. The structure of these codes is simple and relies on the concatenation of two or more codes of shorter length. These codes can be designed to have large diversity which makes them attractive for use in fading channels. The criterion of study is the optimization of the throughput of the system.

Here, the problem of Adaptive Code and Adaptive Modulation scheme selection over parallel flat fading channel is considered. Through the adaptive scheme, a modulation selection based on the quality of the Sub-channels is proposed, and a code assigning method for the different sub-channels, that adapts to the channel conditions. The proposed scheme protects the information better than the ones in the conventional systems, since the latter do not take advantage of the individual sub-channel condition.

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CHAPTER 1

INTRODUCTION

1.1 Background

Wireless communication technology, a fundamental part of modern information infrastructure, is evolving at a frantic pace. In order to achieve high-speed transmission of data on a wireless channel, a reliable and spectrally efficient transmission scheme is needed which is the main thirst of the whole world. Reliable data communication is one of the major problems in modern wireless channels. Radio channels should be able to tolerate the effects of signal fading channels and Inter Symbol Interference (ISI). *ISI* occurs due to multipath propagation of the signal and interference from other communication systems and the background noise. Signals tend to propagate along different paths due to reflection, scattering, and diffraction from obstructing objects. The received signal will then be a sum of randomly delayed signal components, which will add either constructively or destructively, causing rapid fluctuations in the received signal level. This is called *multipath fading*.

Mobile data communication is even more challenging. It requires continuous measurement and updating the Channel State Information (CSI) and adaptation of the coding/modulation techniques according to the new environmental conditions. It also involves proper power distribution techniques [1].

Channel coding is one of the main tools that increase the transmission reliability at higher data rates. It has become an essential part in communication systems. The aim of channel coding is to protect the information against disturbances during the transmission. Thereby, redundancy is added for error detection and correction. It protects data against transmission errors to ensure adequate transmission quality (bit or frame error rate). It is power efficient i.e. compared to the case where the data is not coded, the same error-rates are achieved with much less transmit power at the expense of a bandwidth expansion. Error control coding deals with techniques for detecting and correcting errors in a signal Error control coding is especially useful in wireless communications systems.

With the rapid growth of digital wireless communication in recent years, the need for high-speed mobile data transmission has increased. New modulation techniques are being implemented to keep up with the desire more communication capacity. Processing power has increased to a point where Orthogonal Frequency Division Multiplexing (OFDM) has become feasible and economical. Although, OFDM was proposed in the 1960's it was not widely employed until the 1990's, largely because of significant circuit design issues, such as spurious frequency components and linearity of amplifiers. Today, OFDM is a major contender for 4G (fourth generation) wireless applications with significant potential performance enhancements over existing wireless technology. Since many wireless communication systems being developed use OFDM, it is a worthwhile research topic. Some examples of current applications using OFDM include DSL, DAB (Digital Audio Broadcasting), HDTV broadcasting, IEEE 802.11 (wireless networking standard). Orthogonal Frequency Division Multiplexing (OFDM) is a powerful technique employed in communications systems suffering from frequency selectivity.

Recently, orthogonal frequency-division multiplexing (OFDM) has received a considerable amount of interests for high-rate wireless communications. In order to solve the bandwidth efficiency problem, orthogonal frequency division multiplexing was proposed, which employs orthogonal tones to modulate the signals. The tones are spaced at frequency intervals equal to the symbol rate and are capable of separation at the receiver. OFDM increases the symbol duration while transmitting data in parallel. Therefore, it's become one of the most effective modulation techniques for combating multipath delay spread over mobile wireless channels. OFDM increases capacity by splitting a data-bearing radio signal into multiple sets and then modulating each onto a different sub carrier. These sub carriers are spaced orthogonally so that they can be packed closely together without interference and transmitted simultaneously. In OFDM, the spectra of the sub carriers overlap, and their spacing is chosen so that each sub carrier is orthogonal to all other sub carriers. The common method of obtaining orthogonality of sub carriers is to choose their frequency spacing equal to the inverse of the sub-carrier symbol duration.

Channel coding for OFDM-based systems provides high-rate, high reliable and secure data it is especially suitable for high-speed communication due to its resistance to ISI. As communication systems increase their information transfer speed, the time for each transmission necessarily becomes shorter. Since the delay time caused by multipath remains constant, ISI becomes a limitation in high-data-rate communication [2]. OFDM avoids this problem by sending many low speed transmissions simultaneously.

The advantage of OFDM is that each sub channel is relatively narrowband and is assumed to have flat fading. However, it is entirely possible that a given sub channel has a low gain, resulting in a large Bit Error Rate (BER). Thus, it is desirable to take advantage of sub channels having relatively good performance; this is the motivation for adaptive modulation. In the context of time-varying channels, there is a non-correlation time associated with each frequency-selective channel instance. Thus, a new adaptation is implemented each time the channel becomes non-correlated.

Adaptive modulation is an important technique that yields increased data rates over non-adaptive encoded schemes. An inherent assumption in channel adaptation is some form of channel knowledge at both the transmitter and the receiver. Given this knowledge, both the transmitter and receiver can have an agreed upon modulation scheme for increased performance.

A design principle focusing more on spectral efficiency is *rate-adaptive transmission.* Where the basic concept is to exploit and track the time varying characteristics of the wireless channel to transmit with as high information rate as possible when the channel quality is good, and to lower the information rate (and trade it for link reliability) when the channel quality is reduced. With such a transmission scheme, a feedback channel is required, on which the receiver reports channel state information (CSI) to the transmitter. Based on the reported CSI, the transmitter can make a decision on which rate to employ for the next transmission period. In particular, the transmitter may choose to select symbols from the biggest constellation meeting a predefined bit-error-rate (BER) requirement, to ensure that the spectral efficiency is maximized for an acceptable (target) BER. A promising method is to vary the constellation size and the channel-coding scheme (error control) according to the channel conditions, in which case a rate-adaptive transmission scheme is called *Adaptive Coded Modulation* (ACM).

The severity of wireless communications is much greater than that for AWGN channels. For AWGN channels, good codes can be designed by making the distance between the code words as large as possible, or, alternatively, to make the number of code words at small Euclidean distance as small as possible. However, for fading channels, the severity of channels is mainly tackled by the diversity in communication. Diversity in communication means sending a copy of the message, or part of a copy, on different paths or channels in order to increase the reliability. It can be achieved by diversity in carrier frequency, space diversity, by coding or a combination of all methods. In this work the main concentration is on diversity by coding [3] [4].

1.2 Related work

Concatenated codes are used to obtain long and powerful codes by using simple constituent codes. Product codes are a kind of concatenated codes and are presented by Elias [5]. They can be represented as a set of matrices such that each row in these matrices is a codeword in one constituent code and each column is a codeword in another constituent code. One important property of product codes is burst error correction.

Product codes are serially concatenated codes [6]. The concept of product codes is very simple and powerful at the same time where very long block codes can be constructed by using two or more shorter constituent codes. Consider two block codes A' and B' with parameters $[n, k_A, d_A]$ and $[m, k_B, d_B]$, respectively [1]. The rates of the codes A' and B' are denoted by R_A and R_B, respectively, and are equal to:

$$\boldsymbol{R_A = \frac{k_A}{n}} \qquad \textbf{1.1}$$

$$\boldsymbol{R_B = \frac{k_B}{m}} \qquad \textbf{1.2}$$

The product code C, is obtained from the codes A' and B' in the following manner:

1. Place $k_A \times k_B$ information bits in an array of k_B rows and k_A columns.
2. Coding the k_B rows using the code A'. Note that the result will be an array of k_B rows and n columns.
3. Coding the n columns using the code B'.

The construction of the product code C with constituent codes A' and B' is illustrated in Figure 1.1.

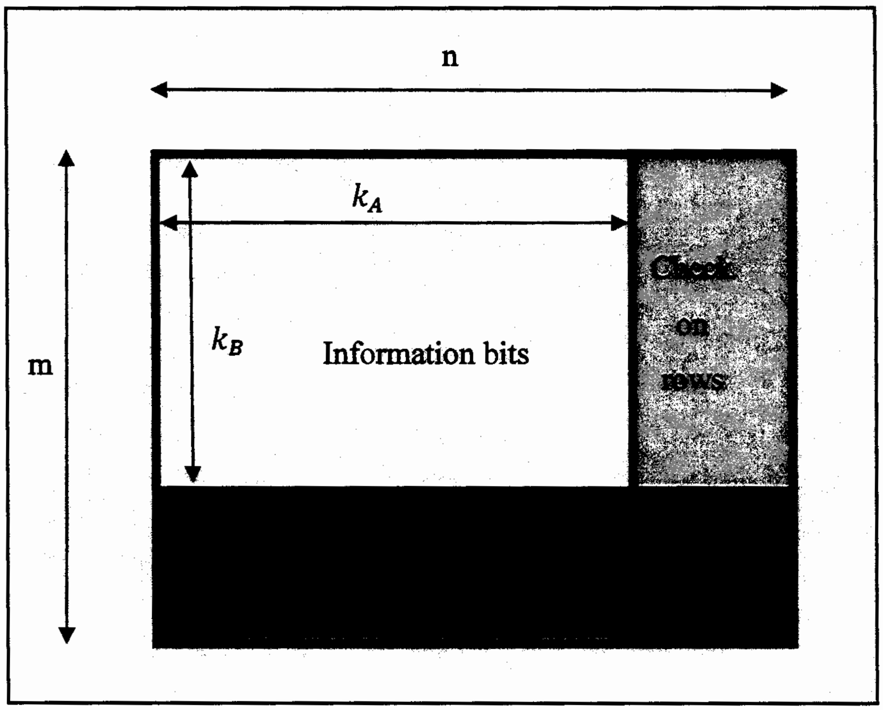

Figure 1.1: Structure of product codes.

The parameters of the resulting product code will be $[mn, k_A k_B, d_A d_B]$ and its rate will be equal to $R_A * R_B$. Therefore, we can construct long block codes starting by combining two short codes.

Product codes have much smaller minimum distance. Therefore, error-correcting potential of product codes is quite large. They have burst error correction capability. The covering radius of product codes is, usually, much greater than half the minimum distance of the code [7]. Forney was the first to propose concatenated codes [8]. The aim was to find a code that approaches the channel capacity with practical decoding complexity that increase polynomialy with the length of the code [9]. The proposed code was a concatenation of a relatively short inner code with an outer long Reed-Solomon code [6]

Generalized concatenated codes are further generalization of concatenated codes. They can be viewed as binary matrices with the rows and columns belonging to many different block codes. They were introduced by Zinoviev [10].

Concatenated codes are efficient in wireless communication channels because they have comparatively high minimum distances and they have proper structure for burst error correction without the need for extra interleaving. Interleaving is used to transform burst errors into random errors, which then can be corrected by forward error control codes. Concatenated codes have higher reliability with slight increase in complexity. They use different codes for encoding the rows and columns instead of restricting oneself to only one code. This leads to different data rates.

Adaptive modulation has also recently gained momentum in wireless systems because it provides a more efficient use of channel. Adaptive modulation for OFDM systems was investigated in [3][(11]

The question of adaptive modulation and coding according to the instantaneous channel conditions was previously treated in [12]. Adaptive modulation for OFDM systems are investigated according to Average Channel Conditions by [4]. The idea of application in OFDM system is presented in [13]. The method used for designing the codes bear great similarities to the ideas presented by Omar-Al-Askary in [1] with some modifications that take into consideration the channel conditions. The following publication is closely related to the proposed project.

Omar Al-Askary presented adaptive coding scheme for OFDM based systems using generalized concatenated codes is presented. The performance of HIPERLAN/2; after encoding with the proposed coding scheme is studied and compared to the standard convolution coding scheme currently used. The criterion of study is the optimization of the throughput. Through the adaptive scheme, a modulation selection based on the average quality of the channel, and a code assigning method for the different sub-channels, that adapts to the channel conditions is proposed. Simulations show that lower packet error rate can be accomplished for even higher throughput.

1.3 Contribution and Outline of the thesis

1.3.1 **Signal-to-Noise Ratio**

The Signal-to-Noise Ratio (SNR) plays an important role in deciding the modulation scheme for each channel. The theoretical values of error probability are calculated using the SNR for each sub-channel. Then these theoretical values are compared with the error probabilities calculated for non-coded data . Since the sub-channels are Rayleigh channels therefore their respective SNR are also called the Carrier to Interference ratio (C/I).

1.3.2 **Channel State information Feedback**

Once the Carrier-to-Interference ratio and the SNR has been calculated at the receiver along with the suitable modulation scheme; then all this information is required to be sent back to the transmitter. For all these calculations, one needs to know the Channel State Information (CSI). In this thesis, CSI has been assumed to be in perfect knowledge. Since the criterion is decided at the receiver therefore there is no bind demodulation required.

1.3.3 **Coherence Time Factor**

Whenever adaptive modulation is used, one factor that is required for reliable communication is environment. The environment effects the choice of a modulation scheme at a great extends. If the surrounding conditions are changing rapidly with time then the method of modulation needs to be assessed with same rate. Otherwise, the use of adaptive modulation would be just wastage of time & resources. The environment factor can be calculated by keeping track on the coherence time factor, which varies according to the type of environment.

1.3.4 **Multiple Concatenated Codes**

In this thesis, the problem of Adaptive Coded Modulation through an OFDM based system over parallel frequency-selective fading channels using *Multiple Concatenated Codes (MCC)* is considered. The structure of these codes is simple and relies on the concatenation of two or more codes of shorter length. These codes can be designed to have large diversity, which makes them attractive for use in fading channels. MCC are based on binary cyclic codes. However, non-binary cyclic or non-cyclic binary codes can be used as building blocks. The decoding of the proposed codes is performed by GMD decoding. These codes can be used in current wireless systems where approaching channel capacity is, usually, of less importance than the processing delay, energy consumption or integrated circuit chip size [1].

The method used for designing the codes bear great similarities to the ideas presented by Omar-Al-Askary in [1] with some modifications that take into consideration the channel conditions.

1.3.5 Detailed Contributions

The main contribution of the thesis can be summarized as follows:

A coding scheme, Multiple Concatenated Codes, with good error correction capabilities especially in fading channels. Choosing optimum coding dimensions according to the individual channel conditions and proper selection of codes for row blocks.

To show, through detailed examples, the potential of these codes for use in field in communications other than the, conventional, one path communication channels. The channels used are OFDM channels.

Designing an effective adaptive coded modulation algorithm for each sub-channel such, that throughput is maximized, while keeping the complexity of the system to a minimum.

Through the adaptive scheme, a modulation selection based on the quality of the Sub-channels is proposed, and a code assigning method for the different sub-channels, that adapts to the channel conditions. The proposed scheme protects the information better than the ones in the conventional systems, since the latter do not take advantage of the individual sub-channel conditions

The criterion of study is the optimization of the throughput of the system.

CHAPTER 2
OFDM AND COFDM

Orthogonal Frequency Division multiplexing (OFDM), also sometimes called discrete multi-tone modulation (DMT), is a particular form of Multi-carrier transmission and is suited for frequency selective channels and high data rates. This technique transforms a frequency-selective wide-band channel into a group of non-selective narrowband channels, which makes it robust against large delay spreads by preserving orthogonality in the frequency domain.
OFDM splits the stream that is to be transmitted into several parallel bit streams. The available frequency spectrum is divided in to several sub-channels, and each low rate bit stream is transmitted over one sub-channel by modulating a sub-carrier using a standard modulation scheme, like QAM, PSK, etc. The sub-carrier frequencies are chosen so that the modulated data streams are orthogonal to each other.

Channel equalization is simplified by using many slowly modulated narrowband signals instead of one rapidly modulated wideband signal. The primary advantage of OFDM is the ability to cope up with channel conditions for example, multipath and narrow band interference without complex equalization filters.

Recently, OFDM is used as an emerging technology for high data rates. In particular, three wireless local Area Networks systems have been normalized simultaneously in three regions of world: the USA, EUROPE, and JAPAN, respectively, IEEE 802.11a, ETSI BRAN

HIPERLAN/2 and ARIB MMAC. They all occupy 5 GHz band, with several channels spaced by 20 MHz offering data rates of up to 54Mbits/s, adequate for the current and future applications.

2.1 Introduction to Multiple Access Schemes:

Multiple access schemes are a very important means of allowing a large number of mobile users to access a finite amount of spectrum simultaneously. By doing the capacity can be increased. In simple words, they ensure the efficient use of the spectrum. They are also preferred over Radio spectrums since they are very expensive. These schemes can be grouped into two categories.

1. **Narrow band:** Narrow band systems have bandwidths that are way smaller then the carrier frequency. In other words, it may be related to the bandwidth of a single channel to the expected coherence bandwidth of the channel [14].
2. **Wide band**: Wideband systems are the one in which the transmission bandwidth of each channel will be higher than the coherence bandwidth. Therefore, the effects of the multipath are combated very effectively. In wide band systems, the frequency fades are not a problem since it affects only a small portion at a given time.

2.2 Types of Multiple Access Schemes:

There are many ways to multiple accesses for accommodating multiple users. These can be categorized as follows:

1. Frequency Division Multiple access
2. Time Division Multiple access
3. Spread Spectrum Multiple Access
4. Space Division Multiple Access

2.2.1 **Frequency Division Multiple Access:**

Frequency Division Multiple Access (FDMA) has N separate frequency channels that are attained by dividing the available bandwidth with N. Each user is assigned one channel at a

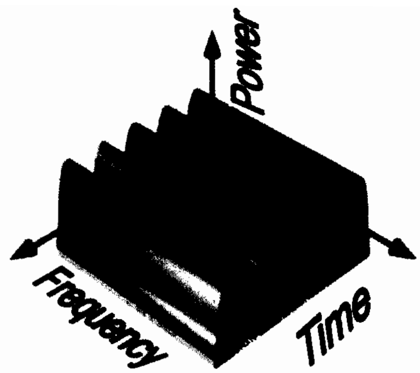

Figure 2.1: FDMA scheme where each user is assigned a different frequency.

single time. Channel allocation is on demand bases. Once a channel is assigned to a user then no other user will be able to share it until the first user does not complete the task. This can be demonstrated by the Figure 2.1. Some portion is reserved for guard band that is required to ensure that there is no overlapping of the channels. FDMA can result in loss of resources since the allocated channel cannot be used by any other user even if the allotted user is idle. FDMA is indented for narrow band systems. It does not require equalization since the inter symbol interference is very low.

2.2.2 Time Division Multiple Access

Time Division Multiple Access (TDMA) is a scheme in which each user is assigned a unique time slot. Thus, each user will be using the whole bandwidth in a sequential manner depending on their time slot number. The slots are arranged in a group called frame and this frame is repeated continuously. Here too guard intervals are required but between time slots. It can be said that TDMA systems can transmit data in a *buffer and bust* method [14]. The following Figure 2.2 represents a TDMA system. In TDMA, the transmission rates are usually, high due to which the equalization is required. However, that is it here a great deal of overhead is required in order to synchronize the data. Without this, the TDMA would be a disaster.

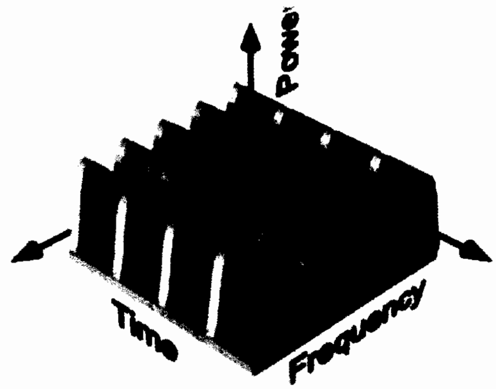

Figure 2.2: TDMA Scheme where each channel represents a time slot.

2.2.3 **Spread Spectrum multiple Access:**

Spread Spectrum Multiple Access (SSMA) is a scheme in which the bandwidth of the data that is being transmitted is much larger than the channel bandwidth. This scheme is used with wide band systems. In simple word, the signal is widened by adding noise. This noise is called the pseudo random noise. Therefore, by multiplying the signal with this nose it becomes a noise signal with characteristics of a wide noise. At the receiver, the data is extracted by multiplying the wide noise signal with the same sequence of noise. It is very good to immune the signal for multipath fading. There are two types of SSMA schemes:

2.2.3.1 Frequency Hopped Multiple Access:

Here each user is pseudo randomly assigned a frequency from a pool of allowable or defined frequencies. This frequency will function as a carrier and transmit data through a wide band channel. Thus, each user will be assigned a different frequency at a time. No two or more user is allowed to access the same carrier frequency at the same time. In this fashion all, the user will be using the entire band at once. Thus, it can be called a multi carrier technique.

2.2.3.2 Code Division Multiple Access:

Here a spreading signal is used to spread the signal to wide band. Each user is assigned a different spreading code. The key is to keep all the spreading codes orthogonal to each other. If for some reason, the orthogonality is lost then the recovery of the signal is impossible.

2.2.4 **Space Division Multiple Access**

Apart from these, there are many other schemes. These are usually the upgrade of the previous or a hybrid of ant two. One of which is very important to us is called Orthogonal Frequency Division Multiplexing.

2.3 Orthogonal Frequency Division Multiplexing

Orthogonal Frequency Division Multiplexing (OFDM) is very similar to the well-known and used technique of Frequency Division Multiplexing (FDM). OFDM uses the principles of FDM to allow multiple messages to be sent over a single radio channel. It is however in a much more controlled manner, allowing an improved spectral efficiency [15]. OFDM is a special form of Multi Carrier scheme with densely spaced orthogonal subcarriers [16]. The basic idea is to transmit N symbols in parallel over N different subcarriers while enlarging the symbol duration N times [17]. This enlargement of symbol duration reduces the bandwidth utilization by the same factor N. Major advantages of using OFDM is the fact that it does not need an equalization even when the channel is highly dispersive. There will be no co-channel interference since each user will be assigned a different orthogonal carrier and they will all be using the (same) entire channel. Not only that it is one of the best ways to fight the fading cased by the multipath but also has that it effectively fights the Inter Symbol Interference. In addition to all this, it has a very high spectral efficiency and operates on a single frequency channel as mentioned earlier. The problem of time synchronization is not an issue with OFDM.

2.3.1 Characteristics of OFDM:

In order to understand OFDM one needs to explore the characteristics of OFDM. By doing so it will enlighten us with the aspects that would result in better performance through its usage. The major factors that govern the OFDM are as follows:

2.3.1.1 Orthogonality of OFDM:

The Orthogonality of the OFDM implies that all the sub-carries are orthogonal to each other. In other words, it means that the dot product of any two subcarriers frequencies is zero. These orthogonal subcarrier will minimize the cross talk between the sub channels. Thus, the guards that were required in the FDMA in order to separate the carriers are no more needed. We know that a good transmission rate is depicted by the Nyquist theorem. The orthogonally will result in rates that will ensure higher rates thus higher spectral efficiency. For the implementation of OFDM FFT algorithms are used. The use of FFT allows us to incorporate efficient modulators and demodulators. OFDM generally has a nearly 'white' spectrum, giving it benign electromagnetic interference properties with respect to other co-channel users [18].

These carriers are chosen in a very precise manner such that the spacing between them is the inverse of the symbol period. This will result in the orthogonality of the carriers. At the receiver end only those packets will be extracted that will have their respective code of orthogonality.

Due to these factors, it requires a less complex transmitter and receiver. In addition, almost all the bandwidth is utilized by using orthogonal multiple carries. Since efficient modulation can be incorporated due to the use of FFT the adaptive modulation becomes quite an attraction with OFDM.

With all these benefits of Orthogonality here are some sensitive factors that need to be handled carefully. One of them being the Inter Carrier Interference (ICI). It results when the carriers are no more orthogonal. Orthogonality is usually lost when the frequencies at the receiver and the transmitter are not synchronized in sub carrier frequencies. The result will be a crosstalk between the carriers and the information will be lost. Synchronization is not always the only cause for loss of Orthogonality. Other factors that contribute to it are Doppler Shift and Multipath. Doppler shift will be there whenever there is a movement by the receiver, transmitter or both. If this Doppler is combined with the effects of multipath, the results are further worsened. The cost of removing or lessening these effects is the increase in the complexity of transmitter and receiver. Thus, we can categorize the loss of orthogonality for two reasons one being due to offset in frequency i.e. δ and the other one in time τ.

2.3.1.2 Inter Symbol Interference (ISI) in OFDM:

In OFDM, each carrier supports a lower data rate. When combined all together the rate becomes quite high. By doing so the ISI can be fought very effectively, that results from the multipath effects. In order to transmit low data rate on the individual sub carriers the modulation schemes with low data rate are used. Since the rate of each carrier is lowered, it implies that the duration of the information symbol will be increased. For this reason, a guard interval will be required which will be positioned between the symbols. Thus, there would not be any overlapping therefore no interference. This leads to no Inter Symbol Interference (ISI). This guard interval will not only keep the ISI minimum but also serve in the problem of synchronization of the carriers. Therefore, it will not only separate the symbols but also contain information that will be used for synchronization. In other words, it will combat the ISI and ICI at the same time. Due to which capacity of the OFDM is reduced. This is the reason why not all of the available bandwidth is used. The loss in bandwidth depends upon the length of the guard interval [19].

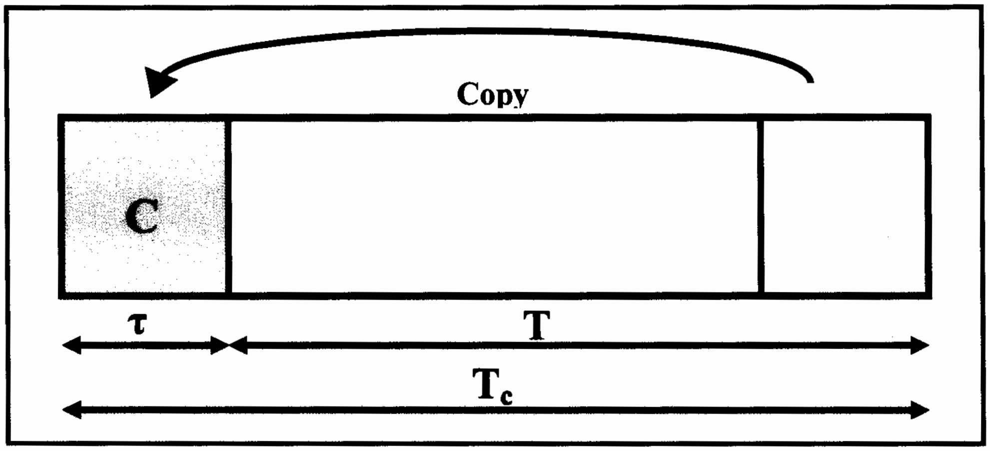

Figure 2.3: Representation of Cyclic Prefix in OFDM (21).

During this guard interval, a cyclic prefix is transmitted. This cyclic prefix consists of the end part of the OFDM symbol, which is copied on to the guard interval followed by the OFDM symbol itself. Thus, it fights the multipath effects. A typical representation can be expressed as in Figure 2.3. Where CP implies the cyclic prefix. The basic function of CP is to accommodate the transient behavior of the previous symbol. This process of accommodating the transient behavior can be explained by the following example. Suppose that we have a symbol of 8 periods of a frequency f1, which is being sent on the channel using the carrier without the use of cyclic prefix. In this case, the received symbol will experience a transition period that will overlap the next and the previous symbol. This will result in the loss of orthogonality. This can be explained by the Figure 2.4.

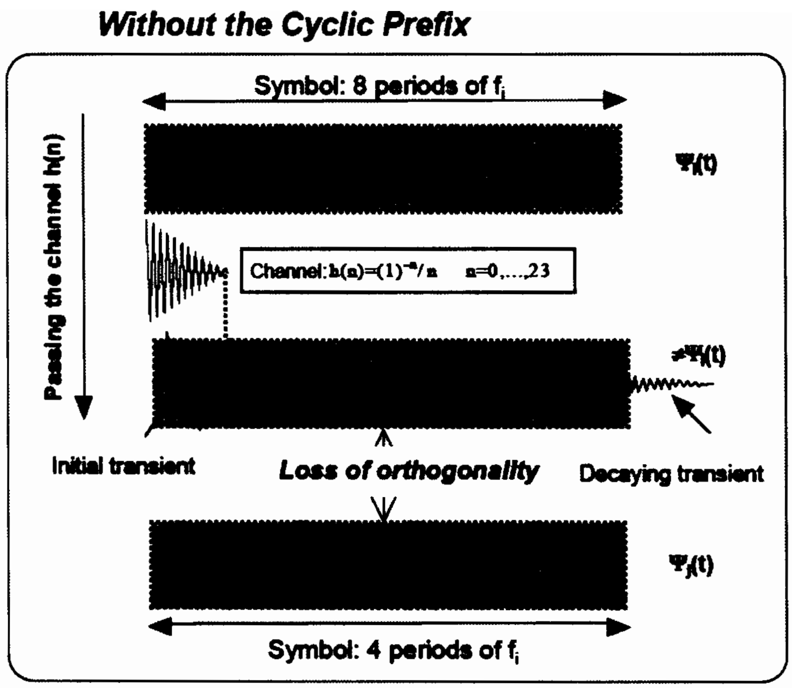

Figure 2.4: Performance of OFDM without CP (21).

Now if the guard interval is included then the transient behavior will be eliminated and t he orthogonality will be left intact. This can be explained by the following Figure 2.5.

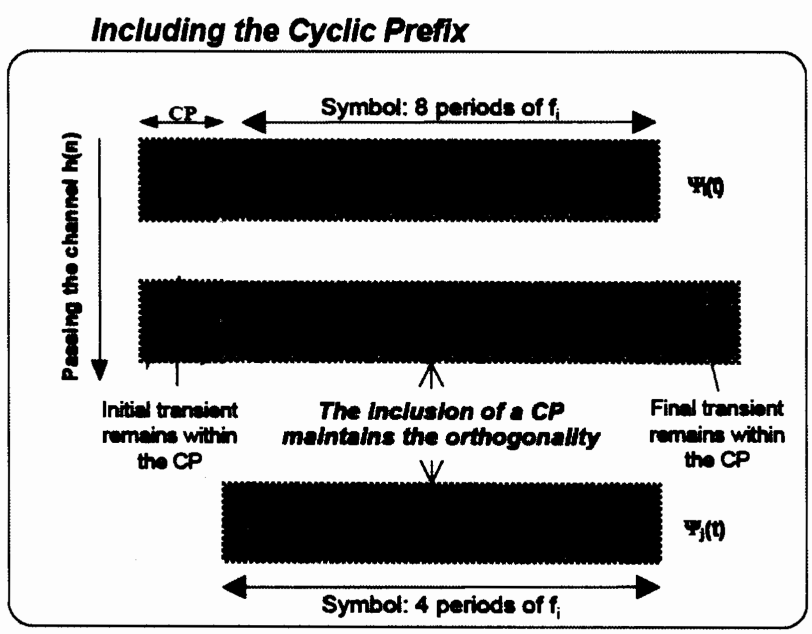

Figure 2.5: Performance of OFDM with CP (21).

Thus, a cyclic prefix will play an important role in the time synchronization of symbols since long salience will result in loss of orthogonality and it ensure the frequency synchronization as well.

2.3.1.3 *Forward Error Correction in OFDM:*

Forward error correction in OFDM implies that a code is used along with the subcarrier. In the past and still now, convolution codes are used. This is done in order to fight the burst errors. Many recent systems use codes hat perform close to the Shannon limit. Usually the Reed-Solomon codes or the BCH codes are used.

Apart from that interleaving in either frequency or time can be used so that the error correcting codes can be used. Frequency interleaving is good to fight the fading in signal. It fights faded part of the bandwidth. Time interleaving fights the effects of fading due to high Doppler shifts.

2.3.1.4 *Adaptive Transmission on OFDM*

By adaptive transmission, it implies that the rate at which the data is sent on the channel in changed according to the changes in the channel on which the data is being sent. The adaptive transmission can be accommodated by the use of modulation schemes or the coding. The rates of the modulation and coding schemes are chosen respective to the channel conditions. The conditions of the channel are a important factor that needs to be known in order to accommodate the techniques that correspond to the rates that would result in the best-optimized outputs. This is done by a feedback process that will sent data about the channel conditions to the transmitter from the receiver end. For all this to be effective power, allocation of the OFDM sub channels is to be maintained so that no sub carrier is bad enough to cause very level of interference or attenuation.

2.3.1.5 *Multiple Access in OFDM:*

In its traditional sense, OFDM is a modulation technique. It is utilized for transferring one-bit stream over one communication channel using one sequence of OFDM symbols [18]. However, it can be exploited to incorporate multiple users. For this to happen all the sub carriers are assigned a different independent user. This method is in the uplink of the IEEE 802.16 Wireless MAN standard, commonly referred to as WiMAX [18].

In this thesis, this concept of multiple access in OFDM is exploited to accommodate multiple user with data rates that varies in coding and in modulation. Thus the thesis would be adaptive multiple access on the OFDM.

2.3.1.6 Diversity in OFDM:

In OFDM, diversity may be added in much number of ways. Some of them involve the use of antenna arrays, MIMO and single frequency networks. All these method are way to add spatial diversity.

2.3.1.7 PAPR in OFDM:

PAPR stands for peak-to-average power ratio. One of the major reasons of limitation of OFDM usage is the fact that it requires high PAPR. This will result in combination of sub-carriers that will lead to distortion. In order to handle the PAPR signal needs to be a linear and the transmitter and receiver need to have high-resolution converter. If the linearity condition is not met the there will be rise in the noise floor, ISI will also surface and the inter-modulation distortion will be experienced.

2.3.2 **System Model:**

The OFDM system can be idealized over an AWGN channel, which is time invariant. It can be analyzed over the transmitter and receiver. Fortunately, the apparently very complex processes of modulating (and demodulating) thousands of carriers simultaneously are equivalent to Discrete Fourier Transform operations, for which efficient Fast Fourier Transform (FFT) algorithms exist. Thus, integrated circuit implementations of OFDM demodulators are feasible for affordable mass-produced receivers [20].

2.3.2.1 Transmitter:

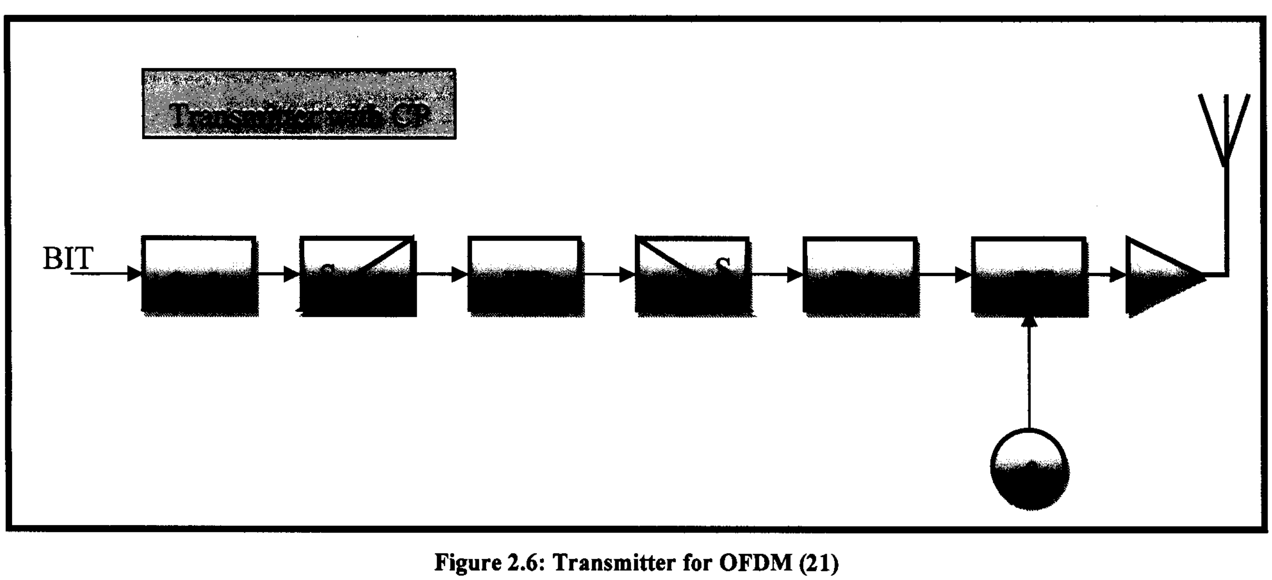

Figure 2.6: Transmitter for OFDM (21)

2.3.2.2 *Receiver:*

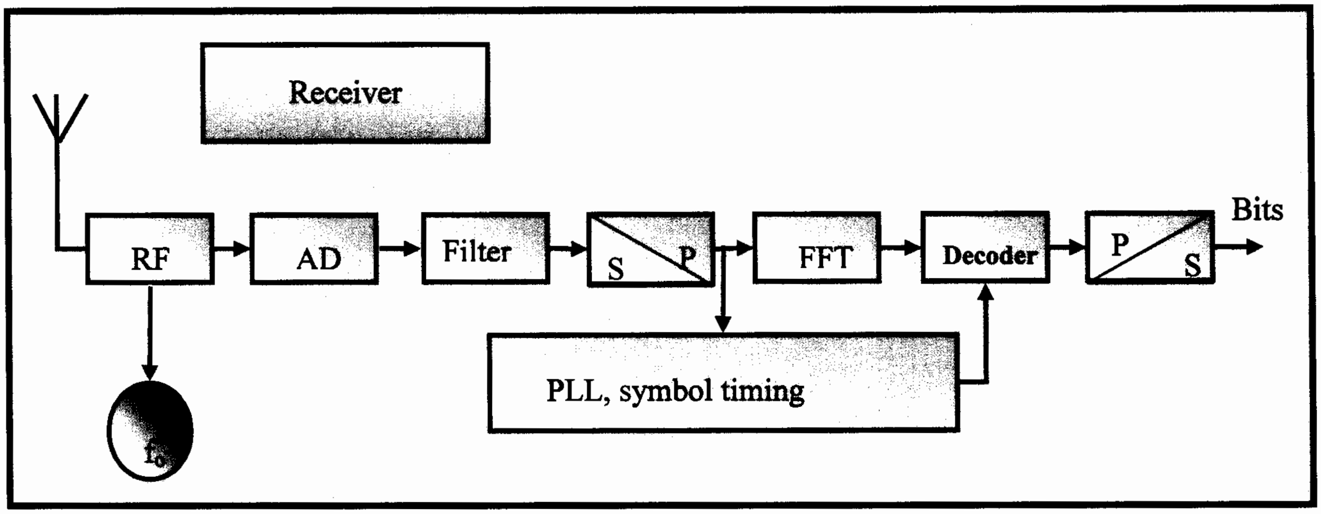

Figure 2.7: Receiver for OFDM (21)

2.4 Coded OFDM

Coded OFDM (COFDM) is the combination of OFDM with the coding of the data in such a way that the results are improved. The coding is integrated by means of channel state information. Therefore, whenever the forward error coding is combined with the OFDM we have COFDM. More information about COFDM can be found in [22] [23]. COFDM has been specified for digital broadcasting systems for both audio -- Digital Audio Broadcasting (DAB) [24] and (terrestrial) television -- Digital Video Broadcasting (DVB-T) [20] [25] [19] [23]. COFDM is very effective with the multipath fading even better then the simple OFDM. It becomes even more effective if the special diversity is incorporated. For example if COFDM is used with the special diversity of Single-Frequency Networks (SFNs) in which all transmitters radiate the same signal on the same frequency. A receiver may thus receive signals from several transmitters, normally with different delays and thus forming a kind of 'unnatural' additional multipath. Provided the range of delays of the multipath (natural or 'unnatural') does not exceed the designed tolerance of the system (slightly greater than the guard interval) all the received-signal components contribute usefully [20]. COFDM is slightly more complicated than the un-coded OFDM. Nevertheless, in order to modulate data on the channel using the multiple carriers we need to know more about the channel if full advantage is to be extracted from the usage of the COFDM. This more information is said to be the *a priori information is usually known as channel-state information* (CSI) [20]. If this information is known then the decoding will be more efficient. Moreover, the inclusion of CSI will greatly improve the performance of the COFDM even in a highly frequency selective fading environment in addition to the interference.

2.5 COFDM in this Thesis

In this thesis, COFDM is used. First, the channel is subdivided into N number of sub-channels. Depending upon the channel state information of the sub-channels the carrier to interference ratio and the AWGN noise is calculated. Based on this information the modulation and the encoding are preformed. Thus, the COFDM in this thesis will exploit the channel state information in order to achieve the best results that would ensure the highest throughput. Channel code is a widely used term which mostly refers to the forward error correction code and bit interleaving in communication and storage where the communication media or storage media is viewed as a channel. The channel code is used to protect data sent over it for storage or retrieval even in the presence of noise. There are many ways to encode the OFDM. They can be classified from simple linear encoders to the complex encoder involving convolutional encoders or their combination. This process is further explained in the chapter that focuses on the proposed predicament.

CHAPTER 3
ADAPTIVE MODULATION

The type of digital modulation used plays a very important role in any type of communication. In today's communication requirement the modulation can play a crucial role. It can very well define the criterion of performance of any type of wireless communication. In this section, the different types of modulation schemes are defined along with their characteristics. At the end of this chapter the schemes that have been used along with their reason of use is stated. Techniques described include Quadrature Phase Shift Keying (QPSK) and Quadrature Amplitude Modulation (QAM) and how these techniques can be used to increase the capacity and speed of a wireless network. These modulation techniques are the basis of communications for systems like cable modems, DSL modems, CDMA, 3G, Wi-Fi* (IEEE 802.11) and WiMAX (IEEE 802.16) [26].

3.1 Introduction

Wireless communication consists of Radio waves that propagate through space and are responsible for delivering any type of data that has been embedded into them. These Radio waves are actually electromagnetic waves that have the property of travelling at the speed of light. There is a range of frequencies that define these electromagnetic waves in the Electromagnetic Spectrum as the Radio ways. Any of these frequencies can be used in order to modulate data. Within this range, each frequency has its own characteristics.

The process by which a frequency from the Radio Wave Spectrum is used as a carrier wave on to which some data is embedded for a destination is called modulation. In other words, the modulation can be stated as a process of modifying the baseband signal so that the transmission over a medium can be facilitated. The medium can be anything. In wireless communication, it is called channel. Thus, prior to [27] transmission the signal is modified by some parameter of some high-frequency carrier signal. The carrier signal is a high frequency sinusoid whose parameters are amplitude and frequency. These parameters are varied in order to get the desired modification. Modulation can be of two types according to the type of the signal:

Analog Modulation: If the signal is an analog signal then the modulation is called Analog modulation. There are two ways to do analog modulation. One is the Amplitude Modulation (AM) and the other being the Frequency Modulation (FM). In AM, a low frequency signal is multiplied by a high frequency sinusoid. It is a technique used to broadcast AM radio. The AM signal is a product of the form

$$x(t) = v(t)\cos(2\pi f_c t) \quad 3.1$$

Here it is assumed that the frequency of the cosine term *i.e.* f_c is much greater that any frequency contained in the spectrum of $v(t)$. The Cosine wave $\cos(2\pi f_c t)$ is called the Carrier signal and its frequency is the carrier frequency [28]. The above equation can be further elaborated by the following Figure 3.1.

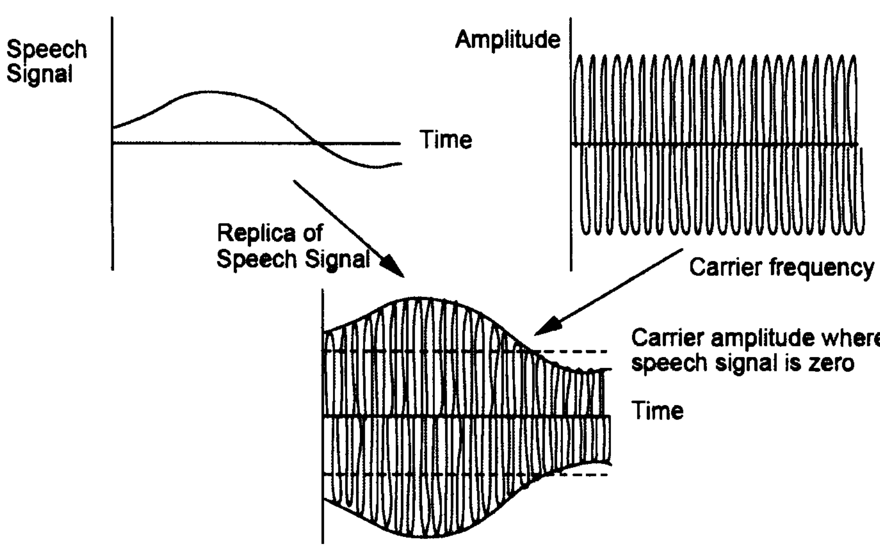

Figure 3.1: Visualization of Amplitude Modulation (29).

Here the speech signal is the $v(t)$ and the carrier frequency the f_c.

The other method of analog modulation is the Frequency Modulation (FM). In FM the signal to be modulated is multiplied by carrier that is function whose angle is time varying. This can be stated in the mathematical form as follows:

$$x(t) = \mathcal{R}e\{Ae^{j(\omega_o t+\phi)}\} = A\cos(\omega_o t + \phi) \qquad 3.2$$

This equation can be represented in following

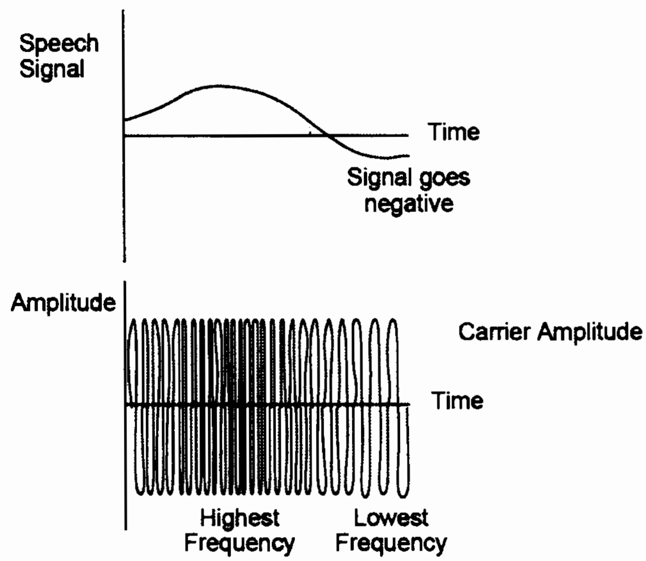

Figure 3.2: Visualization of Frequency Modulation (29).

Digital Modulation: If the signal is a digital signal then it would be digital modulation. The three most basic methods r this type of modulation are Amplitude Shift Keying (ASK), Frequency Shift Keying (FSK) and Phase Shift Keying (PSK). The concept of digital modulation is of our interest and would be further explained in the following context.

3.2 Digital Modulation

Whenever we are talking about digital communication, only specific values of the modulating signal can exist [30]. Therefore, when a digital signal is modulated by any of these methods then their respective characteristics will be incorporated in the signal, which is to be transmitted. These characteristics are listed in the following Table 3.1 [30]. There are many advantages of digital modulation in the communications of any kind. One of the major advantages of digital modulation is the fact that it provides more security since there are more ways of encoding data. Apart from that, they can be sent independently and in a group. Then there are some disadvantages to it as well. For example in a given bandwidth it is less efficient then the AM. The basic principle of using the digital modulation is to combine the Shift keying with the modulating characteristics of a carrier signal i.e. the amplitude, frequency and the phase. They are explained as below in Table 3.1 as a case of binary:

Table 3.1: The key characteristics of AM, FM and PM compared.

Characteristic	AM	FM	PM
[redacted]			
Spectrum Complexity	Simple	Complex	Complex
[redacted]			
Ease of Modulation/Demodulation	Simple	Difficult	Difficult
[redacted]			

1. **Amplitude Shift Keying:** Amplitude shift keying (ASK) involves increasing the amplitude (power) of the wave in step with the digital signal and is used in AM radio [26]. In simple words the transmission is either turned on or off according to the bit that would be either 1 or 0. This method is very sensitive since it uses the amplitude to modulate which is the key factor effected by the noise.
2. **Frequency Shift Keying:** In Frequency Shift Keying (FSK), the frequency is varied with the digital information. It means that in binary case we have two frequencies representing the 1's and the 0's. This method is very resilient to the noise since the modulating factor here is frequency, which is not directly affected by the noise.
3. **Phase Shift Keying:** In Phase Shift Keying (PSK), each bit is associated with a different phase value. In other words, the data is modulated by changing the phase of the transmitting signal with respect to the value of the bit. For example in binary data, 1's may be represented by the 0-degree phase and the 0's by the 180-degree phase.

Once the signal is transmitted then demodulation is preformed at the receiver in order to retrieve the signal. Apart from these methods, there are other methods that are a hybrid of these three. Nevertheless, before discussing them let us explore the criteria for selecting any of these methods that is detection methods, there efficiency and the Shannon limit etc.

Selection Criteria:

Modulation schemes should be chosen in such a manner that the spectral efficiency, power efficiency, bit error rate, channel interference, cost and complexity of implementation be optimized. Therefore, whenever a modulation scheme is to be chosen there should be major focus on one of the above criteria as a requirement and a tradeoff between the rest of the factors in order to achieve the set target. Usually the Bit Error Rate (BER) is the one that sets the standard of the communication. It needs to stay below a particular threshold. Usually for voice, it can be set at the order of 10^{-2} or less. The factor that would result in the increase of BER is the fading experienced by the signal due to the fading, AWGN noise and the other type of interferences.

Spectral efficiency also plays an important role in the choice of modulation scheme. The idea is to keep the spectral efficiency of the system as high as possible without compromising the limitations of the system bandwidth. It can be considered as a trade-off between the pulse width and the data rate. It can be expressed in the form of a equation as follows

$$\boldsymbol{S_\circ = B + 2\Delta f} \qquad \textbf{3.3}$$

Where B is the bandwidth occupied by the signal and the Δf is the maximum carrier frequency. Bandwidth can be defined as the ratio of the data rate of the channel and the spectral efficiency i.e.

$$\boldsymbol{B = \frac{R_d}{n}} \qquad \textbf{3.4}$$

If these two equations are combined, then we will have the following representation

$$\boldsymbol{S_\circ = \frac{R_d}{n} + 2\Delta f} \qquad \textbf{3.5}$$

in order to get higher spectral efficiency is to minimize the spectral occupancy. To do so one will need to decrease the rate of the encoder; the tradeoff to this will be the loss of data or signal with lower power. Another way is to improve the spectral efficiency of the modulation at the cost of higher complexity or to improve the oscillators at both ends [31]. Along with this, the power efficiency is also very important. Here the idea is to keep the fidelity of the signal at power levels that are very low. This can be done by increasing the power of the signal but at the cost of the amplifiers, which may lead to higher BER.

So what we need to do is to find a scheme that would not only provide with a good BER but should have a high power and spectral efficiency, robust to multipath effects, lower co-channel and co-carrier interference, lower transit behavior outside of the band. With all this in mind, the cost and the complexity of the implementation should be acceptable.

3.3 Adaptive Modulation

In wireless systems, the mobility of the users or subscribers causes the channel to vary rapidly with time. This variation is due to the Doppler Effect. Apart from this, the signal is known to shifts due different or varying speed of each user, which will result in different phase rotation [32]. This multipath fading will cause the signal to have different SNR at different times. In these circumstances, using a fixed modulation technique will limit the system from performing to its best. The system would have to be built for such a standard, which would take care of the worst-case scenario of the channel to offer an acceptable bit error rate [33]. Such systems would be spectrally inefficient. Good communication would be the one, which is spectrally efficient and robust altogether. Thus for a good communication over the fading channels the solution would be to use an adaptive scheme that would allow us to chose a method of modulation that would suit best the then conditions of the channel. In simple words, it is the exploitation of the channel characteristics so that the transmission power of the signal, the constellation size and the method of modulations are varied according to the requirements [34]. So; the key over here is to being able to estimate the Channel State Information. Once the CSI is retrieved then the adaptation would not be that difficult. This is done typically by making a channel estimate at the receiver and transmitting this estimate back to the transmitter [33]. Thus, by adaptively changing the modulation method of the channel by using the CSI the result is higher throughput. This process of increasing the throughput by incorporating the adaptive modulation has to be kept in check for a particular threshold of probability of error. This is done by allocating a higher order of modulation method to a channel with a low fading or higher gain; will improve our instantaneous SNR and a lower order of modulation method to a channel, which is very bad with higher fades thus a lower effective SNR. In simpler words, it is a tradeoff between the spectral efficiency and the bit error rate. Thus, an optimization is reached the fading dynamics of the channel. A general estimate of the channel conditions needed for different modulation techniques provided in the following Figure 3.3 [26]

3.3.2 Modulation Selection

Depending upon the CSI the best possible modulation scheme is selected. The following discussion will focus upon the fact that the CSI has been calculated; and the proper modulation order and type need to be predicted. Along with that, all those methods that have been used in this thesis are explained along with their pros and cons. Apart from that the algorithm followed is also stated.

3.3.3 Detection Parameters

At the receiver end, one need to inform which modulation scheme was used for transmitting data. If receiver is not informed; then the demodulator will attempt to do blind demodulation. This will result in the loss of time. On the other hand, the information about the modulation scheme could be sent within the transmission package. This issue discussed in detail under the heading of system model.

3.4 Modulation Schemes

The modulation schemes that are used in this thesis are discussed in this section. Each sub-channel is a Rayleigh in nature. So the processing of sending data on any one of the can be represented by the multiplication of the data and the channel state information and then adding the AWGN noise. This process can be represented by the following equation

$$\boldsymbol{r(t) = c(t) \cdot s(t) + n(t)} \qquad \textbf{3.6}$$

Where $r(t)$ is the received signal, $c(t)$ id the Channel State Information, $s(t)$ is the data signal and $n(t)$is the AWGN noise. This Channel State Information and the AWGN noise is used to determine the modulation order. For this, the water-filling algorithm is used to calculate the Carrier-to-Interference ratio. This Carrier-to Interference ration will used to calculate the theoretical probability of error. There are four types of modulations used in this thesis. These methods are

1. QPSK
2. 8QAM
3. 16QAM
4. 64QAM

Using the above equation their behavior can be depicted in the following Figure 3.4.

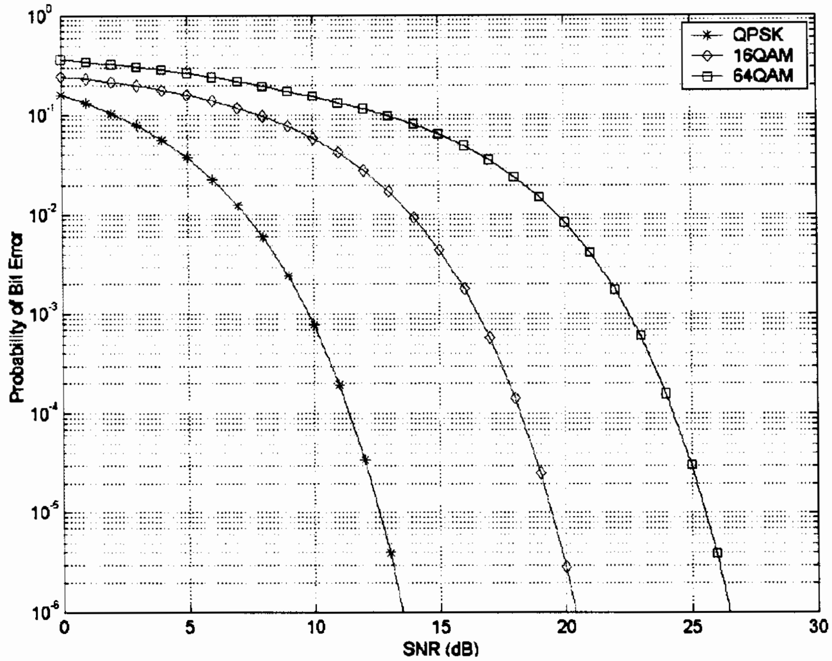

Figure 3.4: Bit Error Probability of Channel.

From this graph we need to chose a point that will serve as threshold point for the maximum allowable Bit Error Rate. From the above figure, we can see that QPSK is most robust of them all. Therefore, by choosing a maximum probability of error for this method will provide us with a threshold for a minimum operate able SNR. In our case let the maximum allowable error be 10^{-3}. This performance would be possible only for the SNR greater than 10dB at least. From the graph it can be seen that there is no modulation method that would be able to perform better than the QPSK for a given probability of error. Therefore, we will assign this method whenever the channel is very bad and there is no possible way for the increase of throughput without compromising the system performance with respect to error probability.

3.4.1 Algorithm for Adaptive Modulation

The advantage of using a wideband channel is that, it provides a higher range of diversity in frequency. By diversity in frequency it is implied that the independent path with their individual fading characteristics. Thus if the frequency diversity is increased it implies that the number of individual fading paths has been increased. This increased diversity will be exploited by the use of adaptive modulation to further improve the performance of the system. This performance can be the measure of either a better rate of probability or higher throughput. In the current scenario, we need to increase the throughput of the system by maintaining an acceptable error rate. For adaptive modulation to be fulfilled two basic conditions needs to be fulfilled. These conditions will ensure that there is an increase in the data rate:

1. Channel needs to be divided into sub-channels that are orthogonal to each other. This implies the use of OFDM channel.
2. Each of these OFDM sub-channel needs to be allotted a modulation scheme that would follow an acceptable probability of error. This selection is based upon the channel state information.

The above two conditions gives on overview of how the algorithm for the adaptive modulation needs to be shaped. Therefore, to adaptively assign each sub-channel a separate modulation scheme a series of steps needs to be followed. These steps are the key to shape the adaptive algorithm. However, in order to follow these steps one need to have some information about the system altogether. This information is very crucial for a system that needs to operate adaptively to the changing channel characteristics due to the surrounding environment. This crucial information set consists of the estimation of Carrier-to-Interference Ratio, the Channel State information and the Coherence Time Factor.

3.4.1.1 Carrier-to-Interference Ratio

The Carrier-to-Interference ratio (C/I) plays a very important role in the determination of the right modulation scheme. Depending on the value of the C/I the right modulation scheme is decided at the receiver. This process involves the calculation of theoretical values of error probability using the C/I for each sub-channel. These values are then compared with the values of those that were obtained with the non-coded data. Along with the values of the C/I one other parameter is also required to predict the modulation scheme. This parameter is also used in the determination of the C/I, these measurements are done at the receive end.

3.4.1.2 Channel State information Feedback

Once the Carrier-to-Interference ratio and the SNR has been calculated at the receiver along with the required modulation scheme; then all this information needs to be sent at the transmitter. For all these calculations, one needs to know the channel state information. In this thesis, it has been assumed that its perfect knowledge is available. Since the criterion is decided at the receiver therefore, there is no bind demodulation required.

3.4.1.3 Coherence Time Factor

Whenever adaptive modulation is used, the one factor that is required to keep the communication at its best is the environment factor. The environment effects the choices of modulation to a great extend. If the environment or the surrounding conditions are changing rapidly with time then the method of modulation needs to be assessed with the same rate. If this is not done, then the use of adaptive modulation will be of no good use but just waste of time and resources. This factor can be calculated by keeping track on the coherence time factor. This factor varies differently according to the type of environment. For indoor the coherence time has been measured to be 18ms if the carrier frequency is of about 5.25 GHz and the Doppler effect of 5km/hr [14].

3.4.1.4 Algorithm

Once all these grounds are covered, we can look into the algorithm for the adaptive allocation of the modulation scheme. The algorithm has the following steps

Step 1 : The channel is divided into N number of OFDM sub-channels. Each of these sub-channels will have its own Channel State Information and SNR. With this CSI the respective Carrier-to-Interference ratio is calculated and stored. These variables are represented in the following Table 3.2:

Table 3.2: Variable representation for each Sub-channel

Variable Types	Sub-Channel 1	Sub-Channel2	Sub-Channel N
[redacted]			
Carrier-to-Interference Ratio	C/I_1	C/I_2	C/I_N
[redacted]			

Step 2 : Calculate the probability of error for each channel by using the Carrier-to-Interference ratio for each method i.e. QPSK, 8QAM, 16QAM and 64QAM. This is done by the following procedure:

1) $Initialize\ data\ for\ calculating\ the\ practical\ probability\ of\ error\ as\ Test_{data}$
2) $for\ i = 1 \rightarrow N; repeat\ the\ following$:
3) $Modulate\ the\ test\ data\ of\ each\ subchannel\ i;\ using\ all\ the\ methods\ and\ store\ in\ variables$:
   i) mod_{QPSK_i}
   ii) mod_{8QAM_i}

iii) mod_{16QAM_i}

iv) mod_{64QAM_i}

4) *Filter the modulated data through the respective subchannel i, add noise and transmit:*

   i) $data_{tx_{QPSK_i}}$

   ii) $data_{tx_{8QAM_i}}$

   iii) $data_{tx_{16QAM_i}}$

   iv) $data_{tx_{64QAM_i}}$

5) *The transmitted data is then demodulated for each subchannel i;*

   i) $demod_{QPSK_i}$

   ii) $demod_{8QAM_i}$

   iii) $demod_{16QAM_i}$

   iv) $demod_{64QAM_i}$

6) *Calculate the probability of error by comparing the recieved data with the* $data_{test}$ *for each i;*

   i) $P_{b_{QPSK_i}}$

   ii) $P_{b_{8QAM_i}}$

   iii) $P_{b_{16QAM_i}}$

   iv) $P_{b_{64QAM_i}}$

   These values of probability of error for each channel and modulation method calculated from the above steps are represented as follows in Table 3.3:

Table 3.3: Probability of error for each sub-channel and method.

Method	Sub-Channel 1	Sub-Channel 2	...	Sub-Channel N
QPSK	[illegible]	[illegible]	[illegible]	[illegible]
8QAM	$P_{b_{8QAM_1}}$	$P_{b_{8QAM_2}}$	$\cdots$	$P_{b_{8QAM_N}}$
16QAM	[illegible]	[illegible]	[illegible]	[illegible]
64QAM	$P_{b_{64QAM_1}}$	$P_{b_{64QAM_2}}$	$\cdots$	$P_{b_{64QAM_N}}$

Step 3 : Calculate the theoretical error probability for each method

1) $for\ i = 1 \rightarrow N, repeat\ the\ following\ steps$
2) $for\ each\ subchannel\ i, calculate\ the\ theoretical\ probability\ of\ error\ for\ each\ modulation\ scheme$

   i) $P_{b_{QPSK_i}} = Q\left(\sqrt{\left(\frac{C}{I}\right)_i}\right)$

   ii) $P_{b_{8QAM_i}} = 2 \times Q\left(\sqrt{\left(2 \times \left(\frac{C}{I}\right)_i \times \left(\sin\left(\frac{\pi}{8}\right)\right)\right)}\right)$

   iii) $P_{b_{16QAM_i}} = \left(\frac{3}{4}\right)\left(Q\left(\sqrt{2 \times \left(\frac{C}{I}\right)_i}\right)\right)$

   iv) $P_{b_{64QAM_i}} = \left(\frac{7}{12}\right)\left(Q\left(\sqrt{2 \times \left(\frac{C}{I}\right)_i}\right)\right)$

3) *end.*

The above steps will result in a matrix of probability of error calculated by using the theoretical formulae, which is in the following Table 3.4:

Table 3.4: Theoretical Probability of error for each modulation method.

Modulation method	Sub-Channel 1	Sub-Channel 2	...	Sub-Channel N
[redacted]				
8QAM	$th\_P_{b_{8QAM_1}}$	$th\_P_{b_{8QAM_2}}$	...	$th\_P_{b_{8QAM_N}}$
[redacted]				
64QAM	$th\_P_{b_{64QAM_1}}$	$th\_P_{b_{64QAM_2}}$	...	$th\_P_{b_{64QAM_N}}$

Step 4: Compare the theoretical probability of error to the practical probability of error for each sub-channel and decide the modulation method with the highest throughput while maintaining the probability of error to an acceptable threshold. For this purpose the following steps are followed:

1) $for\ i = 1 \rightarrow N, repeat\ the\ following\ steps$
2) $if\ th\_P_{b_{64QAM_i}} \leq P_{b_{64QAM_i}};$

   $then\ method_i = 64QAM$
3) $else\ if\ th\_P_{b_{16QAM_i}} \leq P_{b_{16QAM_i}};$

   $then\ method_i = 16QAM$
4) $if\ th\_P_{b_{8QAM_i}} \leq P_{b_{8QAM_i}};$

   $then\ method_i = 8QAM$
5) $else\ method_i = QPSK$
6) $end.$

The above process will be repeated for every 't' less than the coherence time. In the above process if the probability of error is too high to be less the any of the corresponding theoretical values then the one with the lowest probability of error is selected. In our case, this condition is fulfilled by the QPSK method. Once this is done, we get the following Table 3.5; that will show us the methods that are selected for each of the sub-channel:

Table 3.5: Modulation method list for each Sub-channel.

Channel	Sub-channel 1	Sub-channel 2	...	Sub-channel N
[illegible]	[illegible]	[illegible]	[illegible]	[illegible]

The receiver sends this information along with the Channel State Information and SNR to the transmitter. The transmitter then will modulate the data of the users using this scheme. The receiver will then receive this information and demodulate it.

In this thesis, we have used four methods of modulation for the adaptive allocation of each sub-channel. The reference thesis considers the average probability of error in order to compute the modulation scheme. Each of the sub-channels will be using the same modulation scheme. For this purpose the individual C/I ratio of each channel was computed and then averaged out. This averaged out C/I is then used the compute the modulation scheme. The average probability is determined by the following equation:

$$\frac{C}{I_{avg}} = \frac{\left(\frac{C}{I_1}+\frac{C}{I_2}+\cdots+\frac{C}{I_N}\right)}{N} \qquad 3.7$$

This average $\frac{C}{I_{avg}}$ is then used to compute the theoretical probability of error for each method. The same $\frac{C}{I_{avg}}$ is then used to compute the practical probability of error. These are then compared to select the best possible method. This method is indeed simpler and less complex to implement. The drawback is that the sub-channel is not used to its maximum potential. Since the average is taken, a sub-channel with very bad C/I might end up with a modulation scheme that it cannot support. This will deeply affect the probability of error, which will end up rising. In the same manner a sub-channel with a good C/I might be allotted a modulation scheme that would have a lower throughput. The result is waste of resources. Moreover, the fact that OFDM with flat fading is used implies that the system would give terrible performance if the flat fading sub-channels becomes worse i.e. Doppler effects become stronger etc. All these facts can be fought by just incorporating the adaptive modulation. In this thesis, we have used four types of modulations namely QPSK, 8QAM, 16QAM and 64QAM. In the reference thesis, four modulations are used as well. The methods used in the reference are BPSK, QPSK, 16QAM and 64QAM. The difference here lies in the way the OFDM is encoded. Since the reference thesis uses the Generalized Concatenated Codes there is no problem in adjusting the BPSK. But in this thesis the Multi-Level Concatenated Codes (MCC) are used. There is no problem in using the QPAK, 16QAM and 64QAM with the MCC structure. However, the BPSK cannot be accommodated. This thing is explained in the Coded OFDM chapter.

Therefore, in general we have used the Adaptive Modulation in this thesis in order to accommodate the maximum number of users in each channel while maintaining the acceptable level of probability of error and maximizing the throughput. Thus, resources are not wasted and the COFDM is more efficiently used. This is because the adaptive modulation not only increases the throughput but also helps to encode the OFDM to better suit the channel conditions i.e. the Channel State Information, Carrier-to-Interference ratio and the SNR of each sub-channel.

CHAPTER 4

ADAPTIVELY MODULATING MULTIPLE USER ON OFDM USING MULTIPLE CONCATENATED CODES

The method of adaptive modulation has a great significance. The benefits of the adaptive modulation can further be enhanced by adaptively coding in terms of spectral efficiency and probability of bit error along with accommodating multi rate users. The study will include slow Raleigh fading channels. Findings will hold the advantages of using the predictive measures depending on the Channel State Information (CSI) in order to adjust the modulation scheme, rate and the number of users accordingly.

4.1 Proposed Predicament

To verge on a methodology that would not only attempt to adaptively modulate on an OFDM system but also effectively encode data from multiple users that support different rates.

In the proposed problem the concept of Multilevel Concatenated Codes (MCC) is exploited for the facilitation of multiple users that operate at multiple rates that are independent of each other. Since the new technology focus on high speed wireless data transmission; the employment of orthogonal frequency division multiplexing (OFDM) in extension to the MCC would not only increase the throughput but also at a very agreeable probability of error. The use of MCC is very effective since a large amount of data can be transmitted and at the same time, very long codes can be ensured. These codes are a combination of generalized concatenated codes with multilevel coding. The structure of these codes is simple and relies on the concatenation of two or more codes of shorter length. These codes can be designed to have large diversity, which makes them attractive for use in fading channels.

I am trying to investigate ways to take advantage of Rayleigh fading by means of adaptive modulation and coding along with the accommodating multiple multi-rate users. Through this method, the modulation and/or coding used by the transmitter is to change in response to the changing conditions of each sub-channel. In essence, it is optimization of the transmission scheme according to the CSI of each sub-channel for a required fidelity. For example, when the sub-channel has a lower SNR, the reduction in the signal constellation size will result in improved reliability. Conversely, when the sub-channel has a higher SNR, an increase in the signal constellation size will result higher achievable data rates. The same argument can be made for forward error correction coding. In periods of deep fade, we can lower the code rate and make the transmission more resilient to errors. In case the fade is way too deep the sub-channel is closed.

The adaptive accommodation of multiple multi-rate users is best possible through the use of MCCs. The reason is that more than one block code will be available horizontally and only one vertically. The horizontal codes will accommodate the multiple users. The block codes used will represent the rate of the respective user

4.2 System Model

Here is a complete description and definition of the system that is being investigated in this thesis. It will be the platform for comparing our algorithm with reference paper in both performance and complexity. In the thesis, we will only consider linear binary block codes on Rayleigh fading channel. The algorithms and the analytical results can easily be extended to non-binary codes, linear or non-linear and other channel distributions.

Link adaptation is required whenever best or the maximum utilization of the channel is required. There are two ways to achieve; either by variable rate and/or variable coding is used. For a given CSI a power adopting channel estimates the received signal and sends back information to transmitter so that best possible power and code rate can be selected. There are two types of adopting channels:

Channel with silence

Ignoring receiver

With adopting channels, the transmitter will stop sending when the fading is high and thus save power by some restriction on maximum power. Resulting effect of the channels with and without the adaptation at the decoder will be same but the average SNR will be lower. In addition, the transmitting power will be same for both cases; when there is no fading or the channel is good. The constellation capacities of these adopting channels will be estimated in terms of decoders. There are two types of decoders one being Optimal decoder and the other one Scaled output decoder. In our case, the receiver with optimal decoder is considered. In order to understand the working the following assumptions are required:

Behavior of the channel is decided on the CSI estimation, which is assumed perfect by the receiver and the transmitter.

- No signal point should be at the origin or close to it
- Complex noise with $var = ½$ per dimension
- Average energy is equal to SNR but not in units
- Fading coefficient will be complex Gaussian

Channel model with the above assumptions can be represented as Figure 4.1.

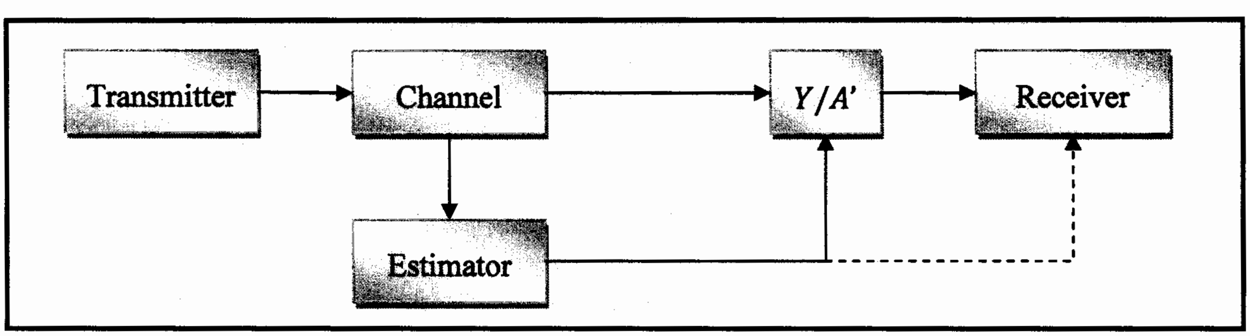

Figure 4.1: Channel Model for receiver with optimal decoder.

Here the two types of adopting channels will be discussed with reference to the constellation capabilities of a optimal decoder. Due to the absence of the transmission for some period of time a new parameter is characterized as A_{α}. Here it is assumed that the channel will shut down when the magnitude of a sample from A' is less than α.

Channel with silence:

Transmission is stopped when fading is deep. The Advantages here are that power is saved. In addition, no more interference is added to the channel. The model can be represented as in Figure 4.2.

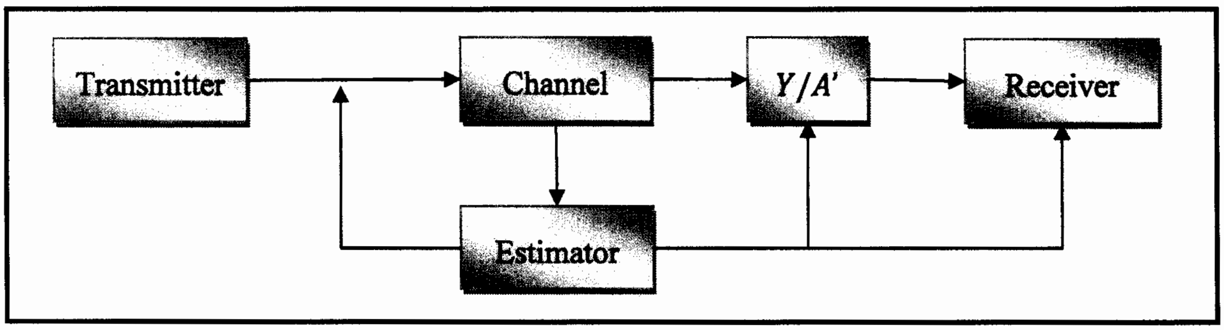

Figure 4.2: Optimal receiver on channel with silence.

The only difference in the calculation of the constellation capacity for an adopting and non-adopting channel is that of the difference in PDF of A' and $A\alpha$. Analytical solution is very hard therefore, the numerical solution is opted. Not much gain is observed with adaptation v/s non-adopting. Another advantage is that it is possible to attain capacity very close to channel capacity in a channel with silence by using long codes. Here for high values of alpha it is better to be silent.

Channel with Ignoring Receiver:

Here the receiver ignores the part of the signal that is unreliable. The unreliable symbols of the signal are called Erasure symbols. Due to ignoring, there will be less mutual information therefore no gain in the capacity. Its model is presented as below:

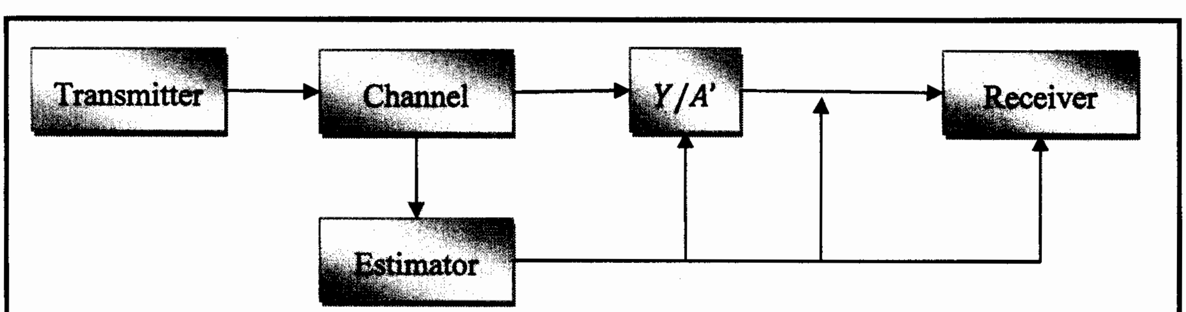

Figure 4.3:Channel model for a optimal ignoring receiver.

In this case, not much difference in observed in the capacities. It implies that some part of the message can be ignored without much loss of information. While the system used in our thesis is as follows Figure 4.4: Overview of our System.Figure 4.4:

4.2.1 Rayleigh Fading Channel

Rayleigh fading is a statistical model for the effect of a propagation environment on a radio signal, such as that used by wireless devices. It assumes that the magnitude of a signal that has passed through such a transmission medium will vary randomly, or fade, according to a Rayleigh distribution i.e.; the radial component of the sum of two uncorrelated Gaussian random variables. This fading is most applicable when there is no dominant propagation along a line of sight between the transmitter and receiver.

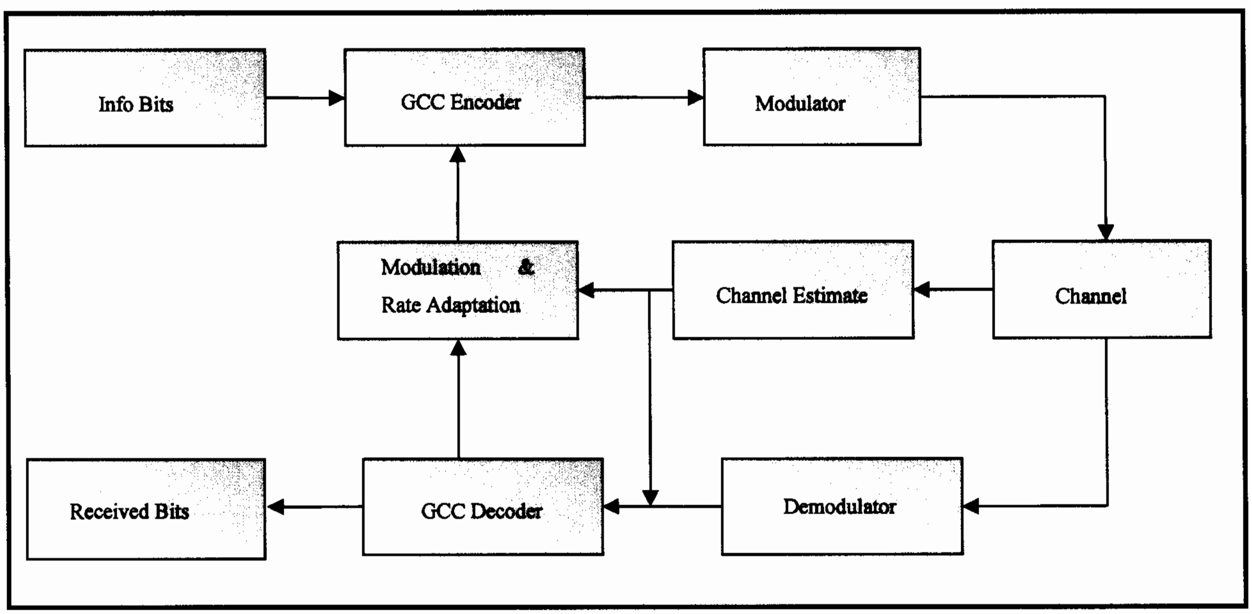

Figure 4.4: Overview of our System.

This is a reasonable model when there is many objects in the environment that scatter the radio signal before it arrives at the receiver. The central limit theorem holds that, if there is sufficiently much scatter, the channel impulse response will be well modeled as a Gaussian process irrespective of the distribution of the individual components. If there is no dominant component to the scatter, then such a process will have zero mean and phase evenly distributed between 0 and 2π radians. The envelope of the channel response will therefore be Rayleigh distributed. A Rayleigh fading channel itself can be modeled by generating the real and imaginary parts of a complex number according to independent normal Gaussian variables.

Rayleigh fading is a small-scale effect. There will be bulk properties of the environment such as path loss and shadowing upon which the fading is superimposed. How rapidly the channel fades will be affected by how fast the receiver and/or transmitter are moving. Motion causes Doppler shift in the received signal components. The Doppler power spectral density of a fading channel describes how much spectral broadening it causes. This shows how a pure frequency e.g. a pure sinusoid which is an impulse in the frequency domain, is spread out across frequency when it passes through the channel. It is the Fourier transform of the time-autocorrelation function.

4.2.2 Channel model

The model used for analysis is a statistical, time uncorrelated, fading model with Gaussian noise. This model is a good approximation of an interleaved flat fading channel with only limited delay spread and where the received signal is the sum of many signal components [38].

Here, we assume a frequency non-selective fading channel. Let x be the sent signal and let y be the received signal. This channel is a slowly varying, where channel coefficients are constant over at least one symbol interval. The receiving signal sample during the i^{th} symbol interval can be written as:

$$y_i = a_i \cdot x_i + z_i \qquad 4.1$$

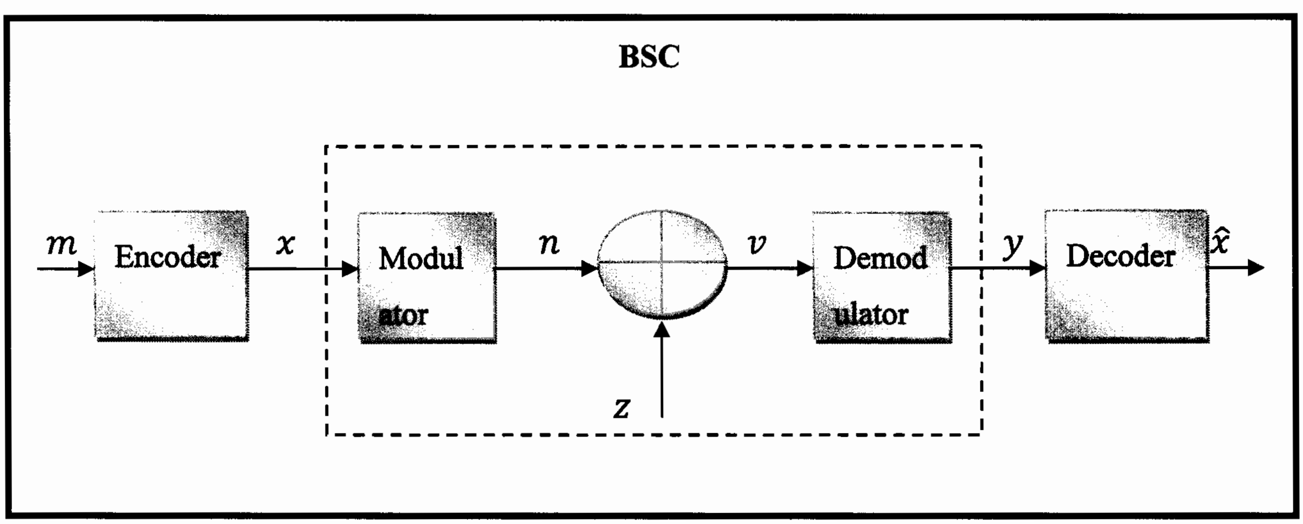

Figure 4.5:Binary symmetric channel

Where x_i, a_i and z_i are respectively the transmitted symbol, the fading coefficient and the noise in the channel. This equation can be represented in the graphical form in the Figure 4.5. We assume that the sequence of multiplicative coefficients, $\{a_i\}$ is independent of both the sequence of transmitted symbols $\{x_i\}$ and the noise sequence $\{z_i\}$. We also assume that $\{a_i\}$ is ergodic and therefore, we can investigate the stochastic channel output given as:

$$\mathbf{Y} = \mathbf{AX} + \mathbf{Z} \qquad \mathbf{4.2}$$

Where, both A and Z ~ $N_c(0, 1/2)$, i.e., complex Normal distribution [1].

When the channel state information is available at the receiver, we model it a

$$\mathbf{A}' = \mathbf{A} + \mathbf{W} \qquad \mathbf{4.3}$$

Where $W \cong N\_c\ (0, \sigma_w{}^2)$ represents the error in the channel estimate.

It is quite rare for wireless channels to be time invariant. Therefore, a time variant fading model should be used for simulation purposes. The Doppler frequency shift [38], denoted by f_D and is equal to:

$$\boldsymbol{f_D} = \left(\frac{\boldsymbol{v}}{\boldsymbol{c}}\right) * \boldsymbol{f_c} \qquad \mathbf{4.4}$$

Where, v is the velocity of the mobile, f_c the carrier frequency and c is the propagation speed. The Doppler shift, or Doppler spread, marks the spread in signal frequency at the receiver due to mobility. A very simple and efficient model is Jake's model (39) (40). Jake's method is a way to simulate the fading process based on the sum of independent oscillators.

4.2.3 **Channel Estimation**

In order to select the best possible parameters we need to know the Channel State Information. This CSI will determine the criteria that would allow us to select the right parameters for transmitting the data. This CSI has to be reliable in order to produce good results.

In wireless communication systems, the channel estimation is performed by transmitting a training sequence [41] [42], i.e. a sequence of symbols that are known to the receiver in advance. This sequence of symbols is used to estimate the parameters of the channel.

For example, a Pseudo Random (pn) sequence can be used. Let the number of symbols in the training sequence be equal to N. Assuming that the fading is very slow then, without loss of generality we can assume that the training sequence begins at time i. We can rewrite 4.2 as:

$$\boldsymbol{y_i = a \cdot x_i + z_i, \qquad i \in 1, \ldots, N} \qquad \mathbf{4.5}$$

I.e. we assume that the fading coefficient is constant during the transmission of the current packet. Since we know the values of the transmitted symbols $x_1, x_2, \ldots, x_Q$ then, we can calculate an estimate of the fading coefficient denoted by â as follows:

$$\hat{a} = \frac{1}{Q}\sum_{i=1}^{Q}\left(a + \frac{z_i}{x_i}\right)$$
$$= a + \frac{1}{Q}\sum_{i=1}^{Q}\left(\frac{z_i}{x_i}\right) \qquad 4.6$$

Since we assumed that a is constant during the transmission period and that the noise at i is complex normal, then we deduced that the channel estimate $\hat{a}$ is a sample, a realization, from a random variable A', with complex normal distribution that has:

$$E(\acute{A}) = a \qquad 4.7$$

$$V(\acute{A}) = \frac{V(Z)}{QE_s} = \frac{1}{QSNR} \qquad 4.8$$

Where E and V are, respectively, the expectation and variance of a random variable and SNR is the signal to noise ratio at the receiver. We deduce from these equations that in the case of very slow fading, we can assume the error in estimation to be complex normal and that its variance is inversely proportional with the length of the training sequence and the signal to noise ratio. The fading is, usually, not constant during the transmission, which is usually the case in practical systems. In this case, the estimate will also experience errors due to the variations of the channel. As the channel is modeled as Rayleigh, this extra error will also be complex Gaussian and we should expect an estimate having a variance larger than that given by the above equation.

4.3 Power Allocation of Sub-Channel

In this thesis, OFDM is used. OFDM is a multiple access technique that has become quite promising for the next generation of wireless communication [14][43]. Since the OFDM divides the available bandwidth into N number of sub-channel, each can be used for different user. For this to happen, each sub-channel needs to have some resource allocation. This process involves the assignment of the total power of the channel to all the sub-channels. There are two ways to perform this; one being the fixed allocation of resources and the other the dynamic allocation of resources. The example of fixed allocation of resources includes Time Division Multiple Access (TDMA) and Frequency Division Multiple Access (FDMA). Since a wireless channel is time varying therefore it would be better to use dynamic allocation of resources. However, there is a catch here. This dynamic assignment needs to be handled carefully since it not as easy as it sounds. The idea here is to allot resources in such a manner that the output either decreases the probability of error or maximizes the throughput. This process is called the optimization of resource allocation. There are two ways to do that:

1. Margin Adaptive (MA) allocation
2. Rate Adaptive (RA) allocation

In Margin Adaptive optimization the idea is to achieve, the minimum transmit power while maintaining a constant probability of error or the throughput [16]. The area of our interest is the Rate Adaptive allocation. In RA allocation the idea is to maximize the throughput while maintaining an acceptable probability of error and keeping the transmit power constant for all users [17]. Many researchers have looked into the optimization in rate adaptive allocation. In [17] it was suggested that channel's capacity can be maximized when each user is assigned a sub-channel that would have the best available gains. Thus, it will lead to less degrading of the signal. While doing so the power of the channel is redistributed by using the Water-Filling algorithm. All is good up till here except that this Water-Filling might lead to the creation of sub-channel with power distribution that would result in negative Carrier-to-Interference ratio. If this happens, the channel will shut down and end up in communication loss. In order to avoid this there should be a track to power distribution. This process of power allocation in order to maximize the throughput while maintain the probability of error elaborated in the section of algorithm.

4.4 Adaptive Modulation

In this thesis, the choice of modulation scheme plays a very important role. This is so because it would be a balancing act between the throughput and the probability of error. This means that the order of modulation needs to be selected in such a manner that the maximum allowable probability of error is never exceeded. However, the catch is the throughput must be maximized. When this is done, there would be a compromise with the probability of error. This is were the balancing act comes in.

By this far, we have established that the modulation scheme shall be playing a major role since the right choice would be crucial for the required outcome. This can be explained by an example. Suppose we have two modulation schemes say BPSK and 16QAM. If these two are compared, we will see that BPSK would provide with better probability of error while 16QAM will provide higher throughput.

In order to perform the adaptive modulation a set of modulation schemes is required. This set will provide the adaptive algorithm to choose the right scheme in order to maximize the throughput on each of the sub-channels. The modulation schemes, which are considered in this thesis for this very purpose, are listed below:

1. QPSK
2. 8QAM
3. 16QAM
4. 64QAM

The choice of these methods depends on two reasons

1. Their ability to provide higher throughput with acceptable bit error rate
2. Their contribution to the construction of MCC's

Each sub-channel will be assigned one of these modulation schemes. This process depends a lot on the Channel State Information (CSI) of each sub-channel. The Water-Filling algorithm which computes the Carrier-to-Interference $\left({}^{C}/_{I}\right)$ ratio will use the CSI. Thus the $\left({}^{C}/_{I}\right)$ of each sub-channel will provide the criteria for the selection for the method that would provide us with the best throughput. The carrier to interference ratio will provide us with an upper bound on the probability of error for each method. For this, the following formulae are used:

$$\mathbf{P}_{bth} = \mathbf{Q}\left(\sqrt{\frac{\mathbf{C}}{\mathbf{I}}}\right) \quad ,\boldsymbol{for\ QPSK} \qquad \mathbf{4.9}$$

$$\mathbf{P}_{bth} = \mathbf{2} \times \mathbf{Q}\left(\sqrt{\left(\mathbf{2} \times \left(\frac{\mathbf{C}}{\mathbf{I}}\right)_{\mathbf{i}} \times \left(\mathbf{sin}\left(\frac{\boldsymbol{\pi}}{\mathbf{8}}\right)\right)\right)}\right) \quad ,\boldsymbol{for\ 8QAM} \qquad \mathbf{4.10}$$

$$\mathbf{P}_{bth} = \left(\frac{\mathbf{3}}{\mathbf{4}}\right)\left(\mathbf{Q}\left(\sqrt{\mathbf{2} \times \frac{\mathbf{C}}{\mathbf{I}}}\right)\right) \quad ,\boldsymbol{for\ 16QAM} \qquad \mathbf{4.11}$$

$$\mathbf{P}_{bth} = \left(\frac{\mathbf{7}}{\mathbf{12}}\right)\left(\mathbf{Q}\left(\sqrt{\mathbf{2} \times \frac{\mathbf{C}}{\mathbf{I}}}\right)\right) \quad ,\boldsymbol{for\ 64QAM} \qquad \mathbf{4.12}$$

These formulae will give us the theoretical bound on the probability of error. This bound is computed for each sub-channel for each of these methods. Once this is computed then a comparison is preformed in order to determine which modulation scheme to be used. For this purpose the practical probability of error is calculate for every sub-channel using each of the above mentioned modulation schemes. After this is done, a comparison is performed. In this comparison, the method that would yield the maximum throughput is selected. In the worst case scenario QPSK would be selected so as the probability of error never exceeds the allowable level. So in order to select the right modulation scheme the following criteria has to meet

$$P_b \leq P_{b_{th}} \qquad \textbf{4.13}$$

where the practical probability of error is P_b. The process of selection the right modulation scheme is explained in the sixth chapter. Following is a figure that will give an overview of how the system will perform i.e. how each sub-channel will be allotted a modulation scheme that will in turn assign multiple users by the use of MCC.

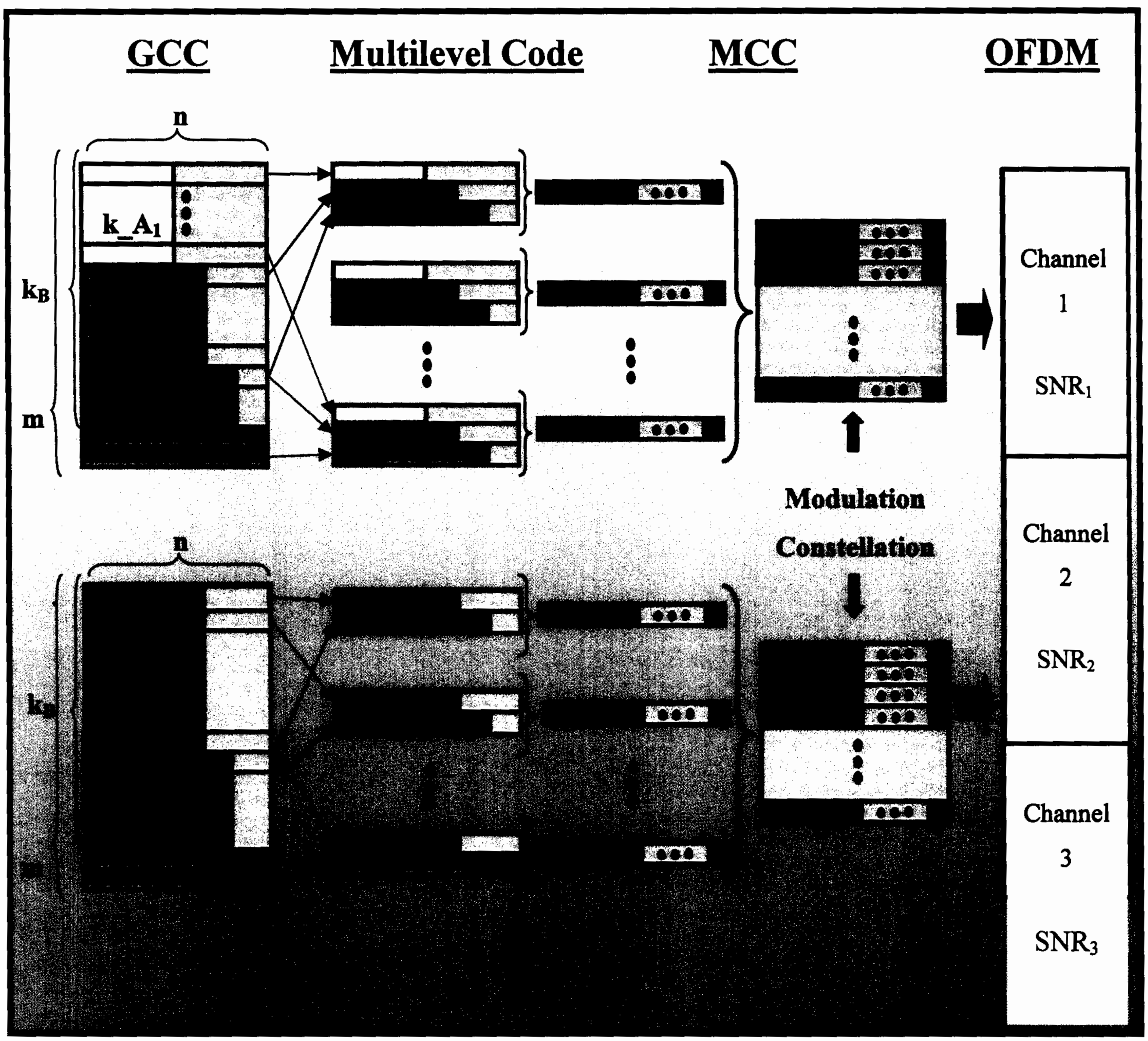

Figure 4.6: Adaptively selecting modulation and MCC scheme for OFDM.

There is more to the choice of modulation then the maximization of throughput. It will contribute to the selection of number of users that would be accommodated on the respective sub-channel. Suppose if 16QAM is the selected modulation scheme for any sun-channel then that sub-channel will accommodate four users. Each of these users will have a different rate. This information will be exploited by using Forward Error Control. This is done by using different encoders for each user to create the GCC and in turn construct the MCC. This process is explained in the following section. One crucial point here is that the modulation schemes that are used in this thesis shows a relationship with the number of encoders and thus the BCH encoders. For this to work the minimum number of users that can be accommodated on a channel is two. The reason for this is that the construction of MCC requires a least two encoders. Otherwise, there would be no multilevel in the GCC. Thus, there would be no MCC. Therefore the order of any of the modulations schemes used is greater the two.

4.5 Forward Error Control

Channel coding is used in communication systems to eliminate or greatly reduce the errors that are introduced by the channel, so that a reliable communication can be possible. All codes are, basically, a pre-selected subset of sequences from the total space of signal sequences.

The coding scheme used in this thesis is an extension of Generalized Concatenated Codes (GCC). This scheme can be viewed as a two dimensional coding scheme. Since OFDM modulation varies both in time and frequency, we can come up with a scheme that takes advantage of the different fading properties of each sub-channel. GCC consists of multiple horizontal encoders and one vertical encoder. This GCC is then in turn used to create the required encoded format. The encoding scheme used in this thesis is called the Multilevel Concatenated Codes (MCC). In order to understand these codes along with their structure and working the following

Figure 4.7 will play an important role.

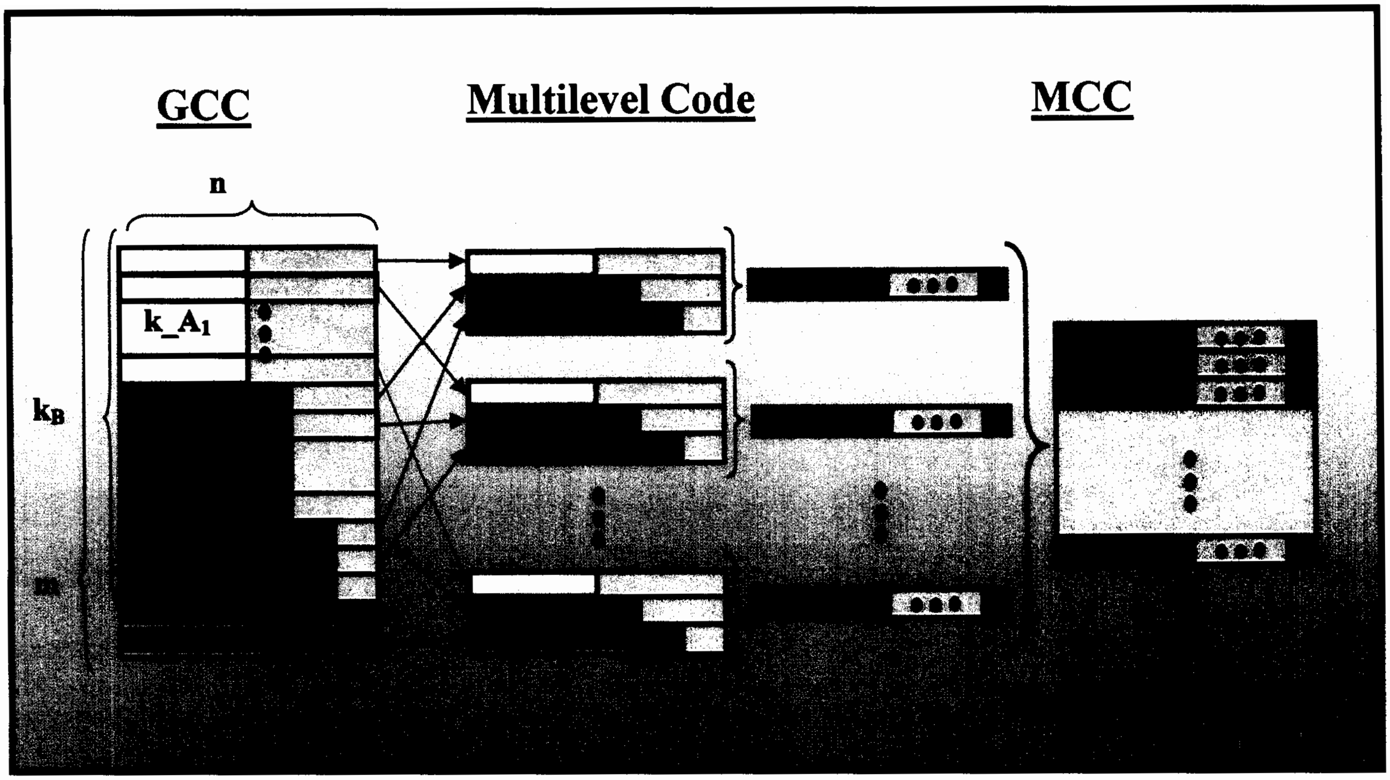

Figure 4.7: Multilevel Concatenated codes

4.5.1 **Multilevel Concatenated Codes**

The information required in order to build the Multilevel Concatenated Code is the modulation scheme. In our OFDM system there will be N number of sub-channels. Each of these sub-channels will have its own modulation scheme. The algorithm for the adaptive selection of the modulation schemes adaptively computes this modulation scheme. Once it has been done, we have the following information

1. Method of Modulation
2. Order of Modulation

This order of modulation plays the key role in selecting the no of horizontal encoders. Thus, the order of modulation and the number of horizontal encoders have a one to one correspondence. Let us say that the order of the selected modulation scheme is O_i where i represent the i^{th}sub-channel. The equation that determined the order of modulation is expressed as

$$O_i = 2^{L_i} \qquad \textbf{4.14}$$

The i corresponds to the i^{th} sub-channel and the L_i will represent the number of users that the i^{th} sub-channel. For example if 16QAM is used for one of the sub-channel, then the order is 16 and the level is four by using the above equation. This implies that this sub-channel will be encoded using four horizontal encoders. Thus, four users with different rates will be incorporated on this channel. This is shown in the figure that four encoders are used. Let A represents a horizontal encoder and B represents a vertical encoder. In our case, there will be $L_i = 4$ encoders. If the set of encoder for i^{th} sub-channel is represented by the following equation

$$A_i = \{A_1, A_2, A_3, \dots, A_{L_i}\} \qquad \textbf{4.15}$$

Then in our case it can be represented as

$$A_i = \{A_1, A_2, A_3, A_4\} \qquad \textbf{4.16}$$

Each of this encoder uses Bose-Chaudhuri-Hochquenghem (BCH) Binary Codes. In BCH codes each row code is subset of each other. The column encoding procedure follows, utilizing one binary BCH code. BCH codes are used because they are considered to be good codes from the perspective of rate and minimum distance [6]. Moreover, they conform to the way GCC codes are constructed. That is, the row codes should be subsets of each other so that we are getting different data rates to use best channel conditions. It is also possible to use other codes or non-binary codes. The encoded length, the information length and the minimum distance in the form of $\left[n, k_{A_{L_i}}, d_{A_{L_i}}\right]$ can express these encoders. The encoder could be represented as $[m, k_B, d_B]$. In this thesis after a large number of iteration the set of encoders that was selected is $k_A = \{30, 36, 39, 45, 51, 57\}$ and for the vertical encoder $k_A = 57$ and $n = m = 63$. using these parameters GCC is constructed. It's shown in the

Figure 4.7 the GCC also has a symmetrical distribution of rows per user. This means that each user will be allotted the same number of rows. This is required for the construction of MCC. Below is a Figure 4.8, which represents a GCC.

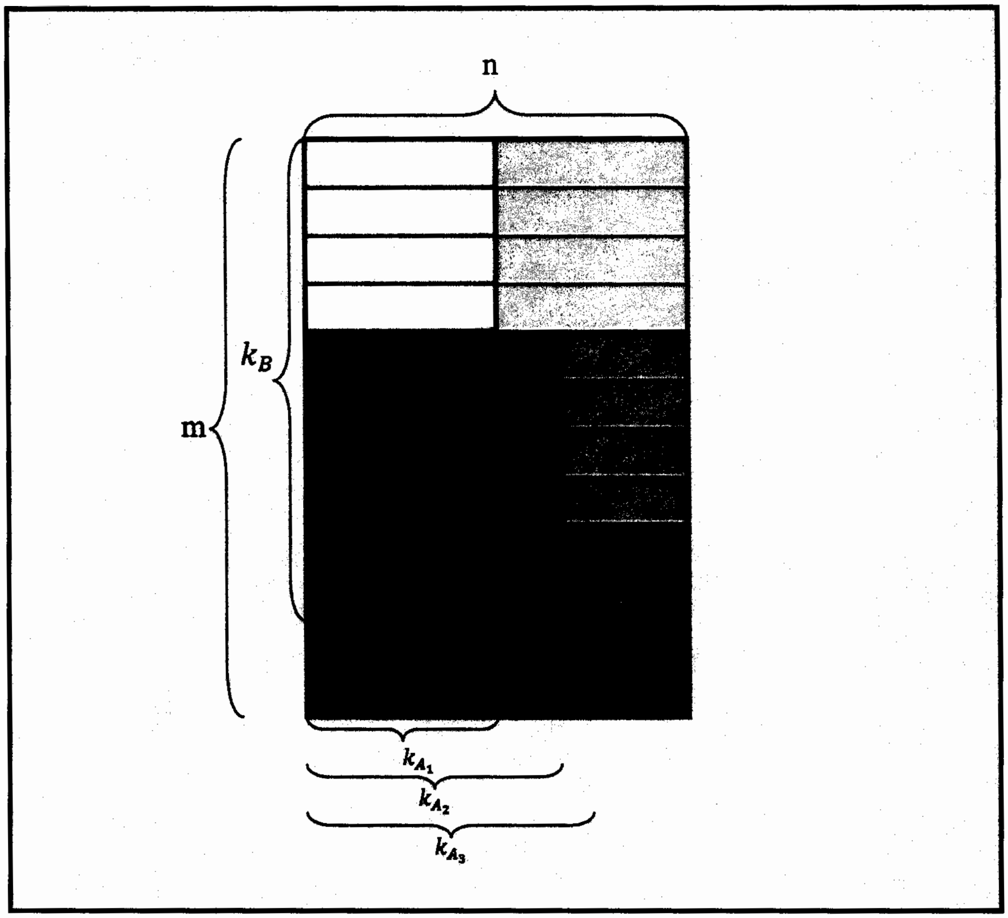

Figure 4.8: GCC.

Row codes, the modulation scheme, and the desired upper bound for bit error rate P_b, are selected according to the sub-channel condition after row decoding. We can find an upper bound on the bit error rate P_b after demodulation, according to the modulation scheme used and the sub-channels' condition. The sub-channels' condition is measured by C/I, which is the power of the desired signal divided by the power of the interference signals.

It shows that each user is allotted the same number of rows while each encoder might have a slightly different distribution. The number of rows of information in the GCC's last encoder/ user will be less than the rest. This is the case when the GCC is yet to be encoded vertically. Once it is done, it is required for each user to be assigned the same number of rows. If it is still not possible then embedment is used. Once this is done the GCC is the outcome. Once this is done, we construct the multilevel code by concatenating one row from each encoder. Then modulating it into a Euclidean code gives us the final product of MCC. This process is shown in the

Figure 4.7. Once all the rows form each group are encoded we get MCC. The number of rows in MCC corresponds to the number of rows that each user is assigned. The data rate of MCC is same as the one that respective sub-channel can support. Which implies that data rate of MCC is the same as the one provided by the modulation scheme that was selected for the respective sub-channel by the adaptive algorithm. The rate of multilevel code that consists of all the horizontal block codes will be defined as:

$$R_A = \frac{\sum_{i=1}^{q} A_i}{n} \qquad 4.17$$

The rate of the vertical block code is:

$$R_B = \frac{k_B}{m} \qquad 4.18$$

The rate of the MCC is:

$$R_c = R_B \frac{\sum_{i=1}^{q} R_{A_i}}{q} - \left(1 - R_B\right) \sum_{i=1}^{q-1} \left(R_{A_q} - R_{A_i} \right) \qquad 4.19$$

For higher code rates it can be approximated as follows:

$$R_c \lesssim R_B \frac{\sum_{i=1}^{q} R_{A_i}}{q} \qquad 4.20$$

This process is repeated for each sub-channel. Thus, the total rate of the channel will be the summation R_{c_i} of each sub-channel i. Therefore, each sub-channel is assigned a different modulation scheme. The scheme selected for each sub-channel will in turn determine the number of horizontal encoders. This will also represent the number of user for its respective sub-channel. This process will be repeated for each sub-channel. This implies that each sub-channel will have its own MCC. Therefore, the number of sub-channels corresponds to the number of MCC. Each MCC will contain different/ same number of user.

4.5.2 Code Selection of the Rows

The main idea is to maintain the GCC codeword error rate P_{GCC} lower than a certain limit P_{cw}. This is done by introducing appropriate redundancy in order to provide a scheme with lower error probability than the desired one. For this purpose each sub-channel is assigned a particular set of codes. Keeping in mind that the decoder will first decode the columns and then the rows, we need to restrict the column error probability lower than a certain probability P_c, after it has been decoded:

$$P_{cw} = 1 - (1 - P_c)^n$$

$$P_c = 1 - (1 - P_{cw})^{1/n} \qquad 4.21$$

The column code can correct up to t_B errors, where $t_B = \lfloor (d_B - 1)/2 \rfloor$ depending on the number of sub-channels used for redundancy. The error probability of the bits in one column before its decoding shall be lower than p, where:

$$P_c \leq \sum_{i=t_{B+1}}^{m} \binom{m}{i} (1-p)^{m-i} p^i \qquad 4.22$$

We can now select our row codes depending on the sub-channel condition, the modulation scheme, and the desired upper bound for bit error rate p after row decoding. We can find an upper bound on the bit error rate P_b after demodulation according to the modulation scheme used and the sub-channels' condition [44][45].

Suppose the rows of each one sub-channel are encoded by codes with t_j error correction capability, where $t_j = \lfloor (d_{A_j} - 1)/2 \rfloor$ and j coresponds to q which is the level of the modulation. After row decoding, the resulting bit error rate can be bounded [46] by:

$$\sum_{i=t_j+1}^{n} \left(\frac{t_j+i}{n}\right) \binom{n}{i} (1 - P_B)^{n-i} {P_B}^i \qquad 4.23$$

Here n represents the modulation order q. Subsequently, we can find the lowest t_j, for each sub-channel, to keep the above probability lower than p. The value t_j directly indicates the code that should be used. So by using the above formula a set of appropriate codes can be computed for each sub-channel. This method will ensure that the overall GCC codeword error rate will always be lower than P_{cw}.

Since the encoding depends on the modulation scheme that was allotted to each sub-channel the rate of the MCC will a little different. The following equations will be used:

Row code rate of per sub-channel i: $R_{A_i} = \alpha \sqrt{\frac{N}{qC}} C_i \qquad 4.24$

Column code rate of per sub-channel i: $$R_{B_i} = \alpha \sqrt{\frac{C}{qN}} \quad 4.25$$

Code rate per channel i: $$R_i = R_{A_i} R_{B_i} \quad 4.26$$

Total Rate: $$R_t = \sum_{i=1}^{N} R_i \quad 4.27$$

Here N implies the total number of sub-channels, the q implies the modulation order of the respective i^{th} sub-channel, Ci is the Shannon capacity of the respective i^{th} sub-channel while C corresponds to the Shannon capacity of the channel itself. The value of α provides us with a tradeoff between the throughput and probability of error. As we are interested in the maximization of throughput we will take α to be equal to 1. If the probability is to be kept at the best level the α has to be near 0. There is a process that can be followed to compute the optimum value. This method is presented in this paper. These are used to plot the throughput of the system as well.

4.6 Algorithm

The following algorithm will demonstrate how the modulation scheme is selected for each of the sub-channels along with the respective rates. The process of power allocation is also covered within these steps. The proposed concept is summarized in the following algorithm.

Step 1: Power Distribution among Sub-Channels:

1. Compute the SNR of each sub-channel as Carrier-to-interference ratio C/I
2. Compute the initial power distribution for all the sub-channels
3. Check if any of the sub-channel is allotted negative power. If not then go to the next step otherwise go through the following procedure:
    a. Those indexes that have a positive value of power will be marked.
    b. These marked indexes will refer to those values of C/I that correspond to positive values of sub-channel power. Thus power will be redistributed

among theses marked C/I sub-channels while ignoring those with negative power.

c. This negative power test is repeatedly preformed until no negative power is assigned to any channel.

d. If there is no more negative sub-channel power then go to the step (following) 4.

4. The computed values of C/I and initial values of allocated power per sub-channel are set as final values for further computation.

This process can be explained through the following figure:

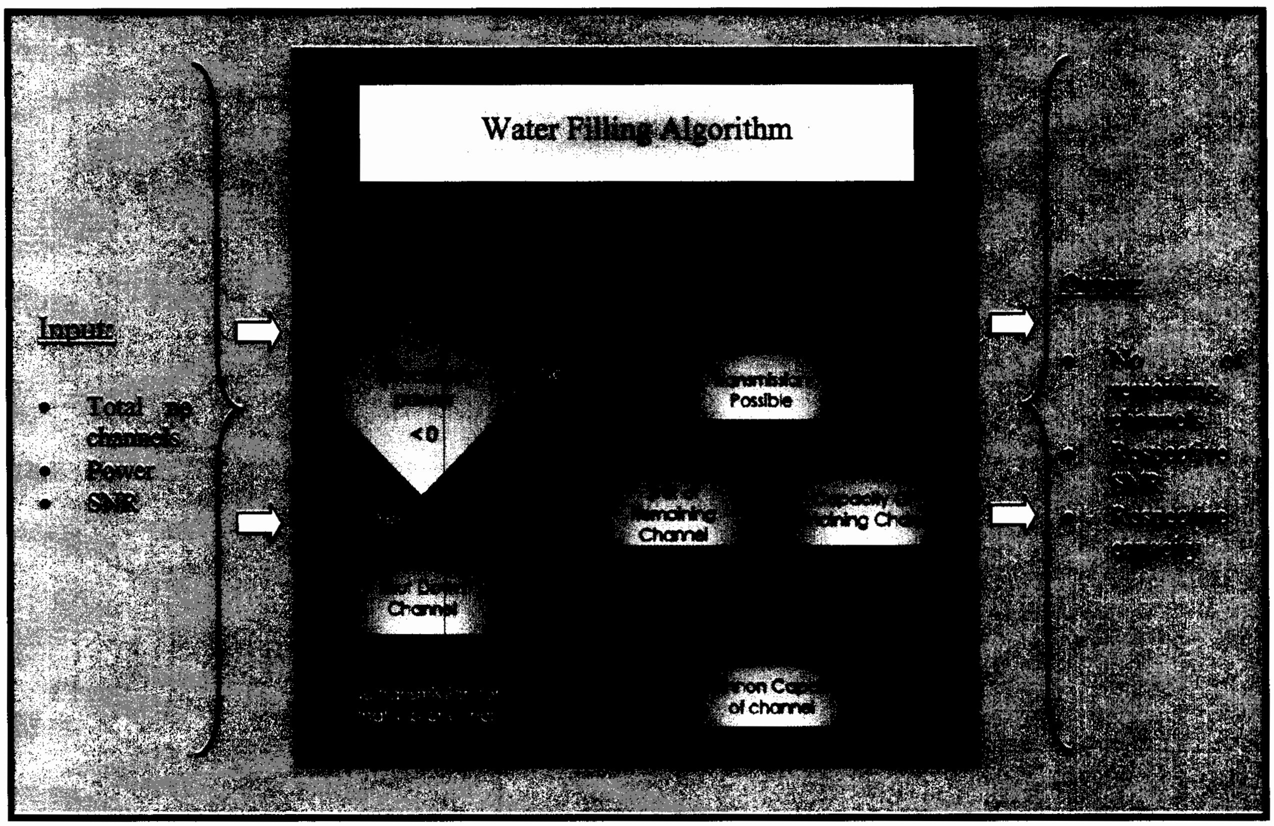

Figure 4.9: Flowchart of Water Filling Algorithm.

Step 2: Compute P_b for Each Sub-Channel:

1. Using the computed C/I of each sub-channel the empirical values of the probability of error is computed.
2. Along with this the practical probability of error is also computed for each sub-channel.

The above steps are preformed for each of the four modulation schemes. This process is explained in detail in the section 3.4.1.4 of chapter 3.

Step 3: Coding and Modulation Scheme Assignment Per Sub-Channel:

1. Compare the empirical and practical values of error probability for each sub-channel
2. Selecting the best scheme for maximization of throughput
3. Based on the modulation scheme select respective coding scheme

The following tables determine the modulation schemes used and the respective coding schemes:

Table 4.1: Used modulation schemes.

Scheme	Method	$M=2^q$	Order=q
[redacted]			
8QAM	QAM	8	3
[redacted]			
64QAM	QAM	64	6

Table 4.2: Proposed coding schemes.

Scheme	Method	$M=2^q$	Order=q
[redacted]			
8QAM	QAM	8	3
[redacted]			
64QAM	QAM	64	6

Step 4: Load the Information Bits and Modulate

1. Each sub-channel will be allotted a MCC
2. Each MCC will have a unique assignment of modulation and coding scheme depending on C/I
3. Data is modulated on the channel

CHAPTER 5

SIMULATION AND RESULTS

When analyzing a communication system, it is usually considered that the Channel State Information (CSI) is perfectly known at the receiver only or both at the transmitter and the receiver. This is actually not far from the truth for the current systems. However, since the current demand is to achieve higher bit rates at the same signal to noise ratios or, alternatively, to achieve the same data rates at a lower signal to noise ratios, then, the assumption of perfect CSI at the transmitter and receiver becomes untrue [47].

Current systems utilize convolution coding, which does not have the capability to adapt to the variation of fading properties of the individual sub-channel. Therefore, always there is a need to use the codes which have capability to adapt to the variation of fading properties of the individual sub-channel. Sub-channels with a low C/I do not have sufficient error correcting capabilities resulting in multiple errors in the received signals. Alternatively, codes with much lower rates should be used in order to keep the error rate less than the required value [1].

According to individual sub-channel, conditions, appropriate codes can be implemented. By this, more error correcting and detecting capability can be achieved and sub-channels with high C/I use less redundancy and achieve high data rates. Linear Generalized concatenated codes can also be used which load some of the sub-channels with redundancy for a second dimension [40].

5.1 Comparison

In this thesis the work of Omer-al-Askary was followed. Almost all the parts of his thesis; deserve further investigation. However, the most important ones are as follows:

- It is possible to randomly interleave each row of the concatenated code before modulation.
- Non-binary codes can be used to encode several rows at the same time.
- In the case of OFDM channels, it is sometimes required that different sub- channels use different modulation schemes as well as different code rates depending on the quality of the channel. In this paper, all the sub-channels use the same modulation scheme but with different code rates. A further improvement on the structure of the code is that the rows of a codeword block can be combined in different ways such that the transmitted symbols for one codeword block belong to different modulation schemes.

It's the last one that has been a focus of our research. The main idea is to adaptively implement the OFDM to accommodate multiple users with different rates.

5.1.1 System

In order to understand any algorithm it is important to understand the system on which its implemented as well. The system proposed by Omer-al-Askary can be viewed as an enhancement to the H/2 standard that was introduced by the European Telecommunications Standards Institute (ETSI). The modulation scheme implemented in the H/2 physical layer is OFDM.

The modulation schemes implemented in H/2 are: Binary PSK (BPSK), Quadrature PSK (QPSK), 16QAM and 64QAM. In order to compare designed OFDM system with the H/2 standard, same modulation schemes are used. The OFDM system including the proposed coding scheme has been depicted in Figure 5.1:

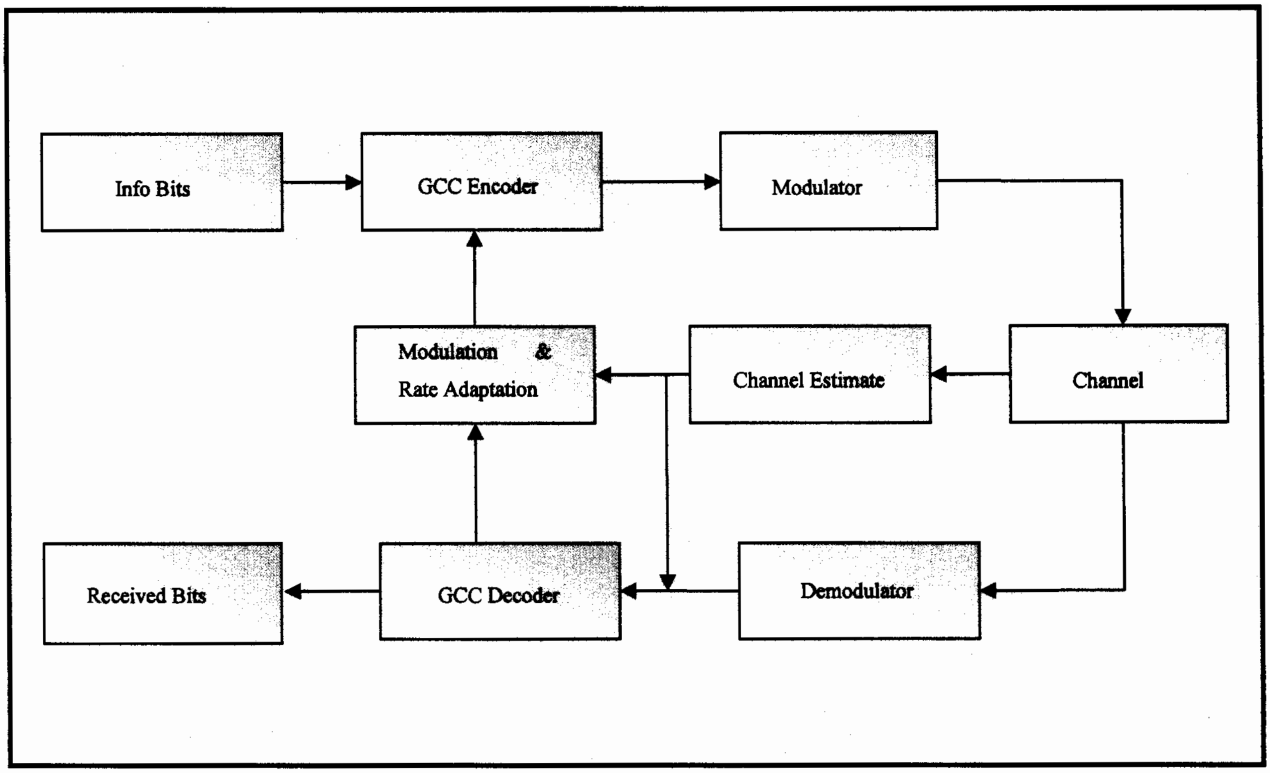

Figure 5.1: Overview of System

5.1.2 Modulation and Coding

Two aspects were considered in the structuring of the system: *choosing a modulation type* and *deciding on the codes* used for the individual sub-channels. These two aspects are interrelated since choosing a different modulation scheme affects the choice of the optimal code for each sub-channel. The criterion is to maximize the throughput of the system.

5.1.2.1 Modulation Scheme

According to the mean C/I, a modulation threshold is assigned, that depicts the modulation scheme to be used. These thresholds are found after excessive simulations on the adaptive GCC system for each of the modulation schemes. This technique conforms to the H/2 standard [48] and can be easily transformed into an integrated circuit in terms of implementation.

5.1.2.2 Coding Scheme

The coding scheme implemented in the system is a simple type of GCC. Initially the scheme decides upon the structure of the GCC. Each row will be loaded by a different number of information bits depending on the code used. Once the information bits are loaded, the row encoding procedure is initiated, using different binary Bose-Chaudhuri-Hochquenghem (BCH) codes. The column encoding procedure follows, utilizing one binary BCH code.

The GCC scheme is a two dimensional coding scheme. Since OFDM modulation varies both in time and frequency, this scheme takes advantage of the different fading properties of each sub-channel. In this paper, each sub-channel corresponds to a row, whereas all the bits transmitted at the same time form a column or a group of consecutive columns. Each codeword in the GCC can be considered as a $m \times n$ binary matrix. In Omer-al-Askary's paper, m is the number of the sub-channels in the OFDM system and n is the length of the binary BCH codes used for the row encoding. The bits in each row are grouped in groups of q-bits, where q is the number of bits required for a specific modulation scheme. This way, q columns in the codeword matrix will constitute an OFDM symbol and will be sent at the same time instant. Note, however, that sometimes, the length of the codes used for the rows, n, does not divide by q and thus, the last OFDM symbol will be loaded by bits from two different codeword matrices. Let $L_A = \{A_1, A_2, \dots A_L\}$ be a list of BCH codes with parameters $[n_i, k_{A_i}, d_{A_i}]$ and $i \in \{1, 2, \dots, L\}$. Each row in the codeword matrix will be a member of one of the codes $A_i \in L_A$. In addition, each column in the codeword matrix will be a member of the $[m, k_B, d_B]$ code B. The adaptive method should choose for each row a code from L_A that has a rate and minimum distance appropriate for the quality of the sub-channel.
After extensive simulations, 4 different codes for the row coding were used by Omer-al-Askary. The optimum codes were proved to be: (63,63,1), (63,57,3), (63,51,5) and (63,36,11). The proposed improvements required more bandwidth in the reverse link due to reporting purposes.

The principle of construction of Generalized Concatenated Codes is based upon serial outer encoders and inner encoder. This could be generalized for any number of outer codes and inner codes. The codes can be used in combination with some q-level bandwidth efficient modulation, e.g., $2^q PSK$ or $2^q QAM$. From each subcode q bits (q is the number of bits required for a particular modulation scheme allotted to a sub-channel through CSI) are taken at each time, modulated and transmitted on each one of the equivalent channels which may be OEDM or MIMO etc. In this thesis the criteria of code selection depends on the channel conditions that is the CSI. Therefore, the structure of GCC is exploited to construct the required MCC. Therefore, this CSI determines the modulation scheme, which in turn determines the coding scheme. Thus, depending on the modulation scheme, a set of usable codes is picked. This set of usable codes is called the coding scheme and it depends on the order of modulation, which is the power of 2. Therefore, each scheme will have a different order of modulation. Following Table 5.1 gives an overview of how the selection is made:

Table 5.1: Code assignment scheme

[illegible]	2	3	4	6
[illegible]	{30 36 }	{30 36 39 }	{30 36 39 45 }	{30 36 39 45 51 57}
[illegible]	63	63	63	63

5.1.3 Adaptive Algorithm

The proposed concept of Omer-al-Askary can be summarized in the following algorithm.

Step 1: Distribute power equally among the sub-channels, compute the average C/I of the sub-channels and choose the modulation scheme

Step 2: Compute P_b for each sub-channel

Step 3: Assign a certain code $A_i \in A_L$ for each sub-channel, according to the code assigning thresholds and its corresponding P_b.

Step 4: If for any sub-channel no coding scheme can be assigned, shut down the sub-channel with the lowest C/I condition, redistribute its power equally among the remaining sub-channels, and go back to Step 2.

Step 5: Load the information bits and modulate.

The suggested decoding method for the MCC is GMD decoding with hard decision decoding of the message, because of the low complexity required for this algorithm. GMD decoding is performed by decoding the rows using a BCH decoder, followed by decoding of the columns using a similar BCH decoder.

The referenced algorithm was designed to be comparable with H/2 standards. In the specifications of H/2 there are 48 sub-channels used accordingly. The column codes are one of the BCH codes with parameters (48, k, d). In the referenced algorithm the codes were derived from the shortening of the (63, k, d) codes. The encoded length of the codes is chosen to be n=63. The shortening method makes sure that the resultant codes have k that is shorter than 63. The use of the same mother code results in a much simpler hardware implementation. The channel properties remain constant for the duration of the whole GCC codeword. This implies that the duration of the largest codeword used is less than the coherence time of the channel. In the simulations, we assume perfect knowledge of the sub-channel's conditions.

An overview comparison in the algorithm of the proposed algorithm and that of the Omer-al-Askary is given below:

Table 5.2: Comparison

Proposed	**Omer-al-Askary**
[redacted]	[redacted]
Each sub-channel has its own modulation scheme	Sub-channels use same modulation scheme
[redacted]	[redacted]
Number of codes in MCC equal modulation order	Number of rows in GCC equal number sub-channels
[redacted]	[redacted]
Higher redundancy	Lower redundancy

The complexity in decoding, was measured by the number of operations per decoding procedure. It was comparable to current commercial OFDM systems. However, in our case the coding scheme was also dependent on the modulation scheme not on the channel only. While his criteria of channel also depended on the code's ability to enhance the channel ours was to provide the same average rate of codes used to that of the channels conditions. These conditions were monitored by the adaptive selection of the modulation scheme.

5.2 Results of Proposed Predicament

The main idea while erecting the simulation to support the acclamations the main idea was to keep the whole process as simple as possible. At the same time, it was also essential to imitate the reality as much as possible. This purpose involved the some assumptions. It was understood that there was a perfect knowledge of sub-channels' conditions. It was a channel model [49] for office environment [50]. Following will a series of comparisons in terms of error probability and throughput. These are displayed in categories of system with and without encoding and in suggestion to the referenced paper used.

5.2.1 **System without Encoding**

Following are the results of the proposed algorithm without encoding. They were attained by calculating the probability of error for each sub-channel and comparing it with the practical error probability.

5.2.1.1 Error Probability

Probability of error for non-coded system of six sub-channels is given in the following Figure 5.2. It can be seen that the error probability is quite acceptable. Not only this, it even got better with the increasing signal to noise ratio. The outcome of error probability greatly depends on the condition of each sub-channel. Since the channel is flat fading, the Channel State Information would not be changing as often. Due to which the decaying of the error probability will be somewhat smooth. In addition, the displayed figure is a mean of all the sub-channels. In our case there are six sub-channels and the sampling time is 1/Mbps. Further computation showed that by increasing the no of bits per second there was degradation in the error probability. Therefore, to further decrease the probability there are two options. One is to decrease the no of bits per second and the other option is to introduce the redundancy through encoding.

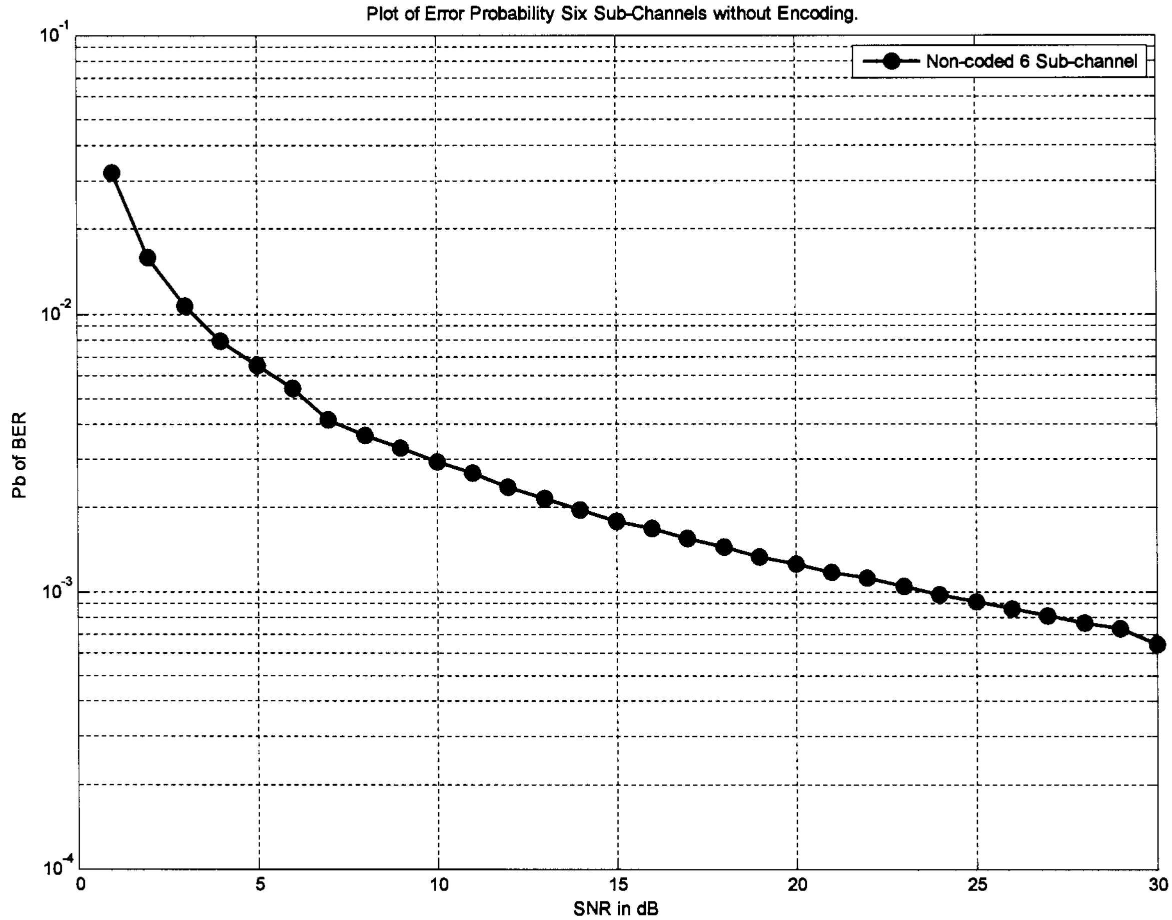

Figure 5.2: Error probability of non-coded system.

5.2.1.2 Throughput

It's seen in the following Figure 5.3 that the throughput is maximized by three times to the rate at which the bits are sent in one second. In addition, it can be noted that the maximum is attained at an SNR of about 8dB. This remains constant for the rest of the SNR range. Throughput is inversely proportional to the error probability. So if the throughput is maximized the probability of error would increase. Now since our aim is to maintain a reasonable error probability while maximizing the throughput there are two ways to do so. One is to increase the number of bits per second while the other is to use encoding. The following figure is attained for a sampling time of 1/Mbps. It can be increased but at the expense of error probability.

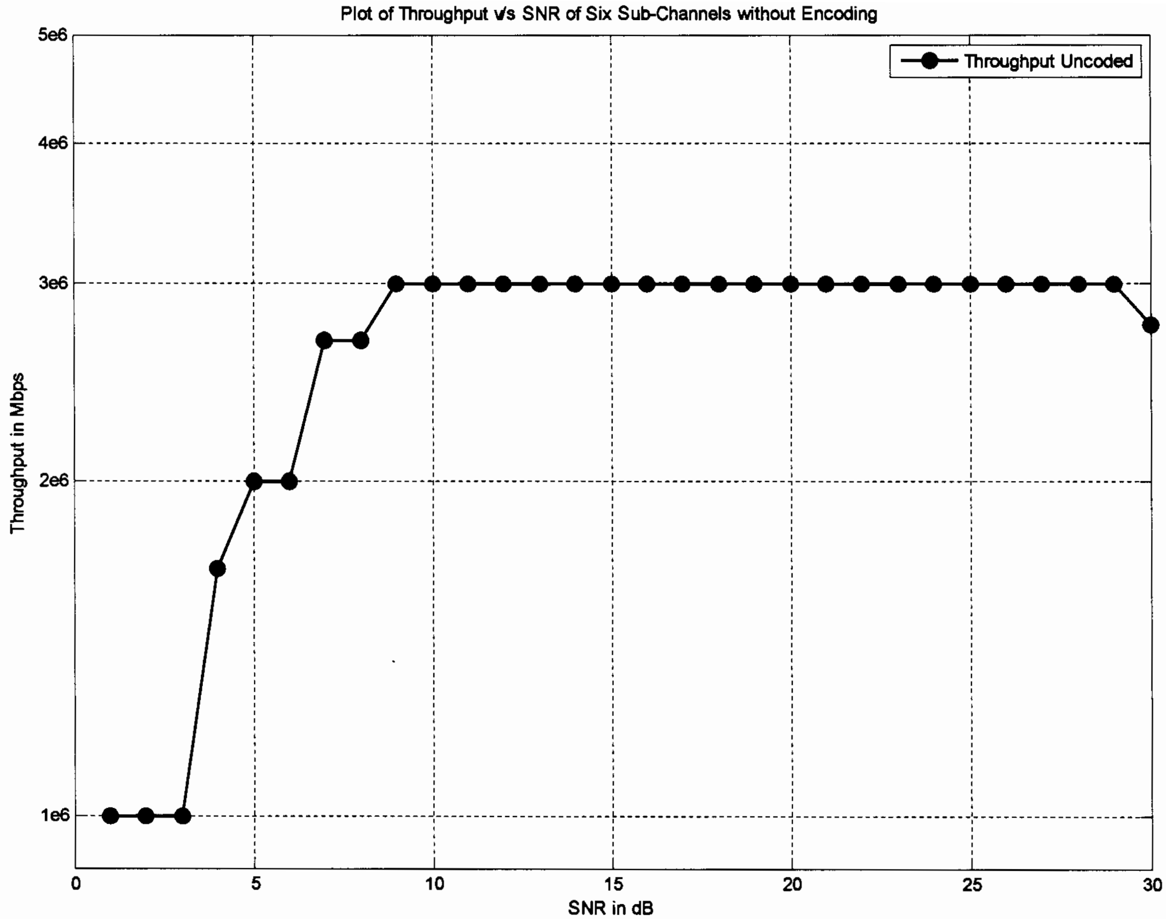

Figure 5.3: Throughput v/s SNR non-coded system

5.2.2 System with Encoding

The encoding in the proposed algorithm follows the structure of a Multilevel Concatenated Codes. These play an important role, not only in decreasing the probability of error but also in busting the throughput. These were covered in the fourth chapter. Following are the results of the encoded system.

5.2.2.1 Error Probability

The following Figure 5.4 illustrates the error probability of an encoded system. It is observed that error probability does decay but not in a smooth curve. There are some abrupt changes, which signify the fact that there is switching between the modulation orders but since it is a flat fading environment the overall pattern is a decaying pattern, as the SNR increases. If in cases it was a rapidly changing environment the curve might be more fluctuating.

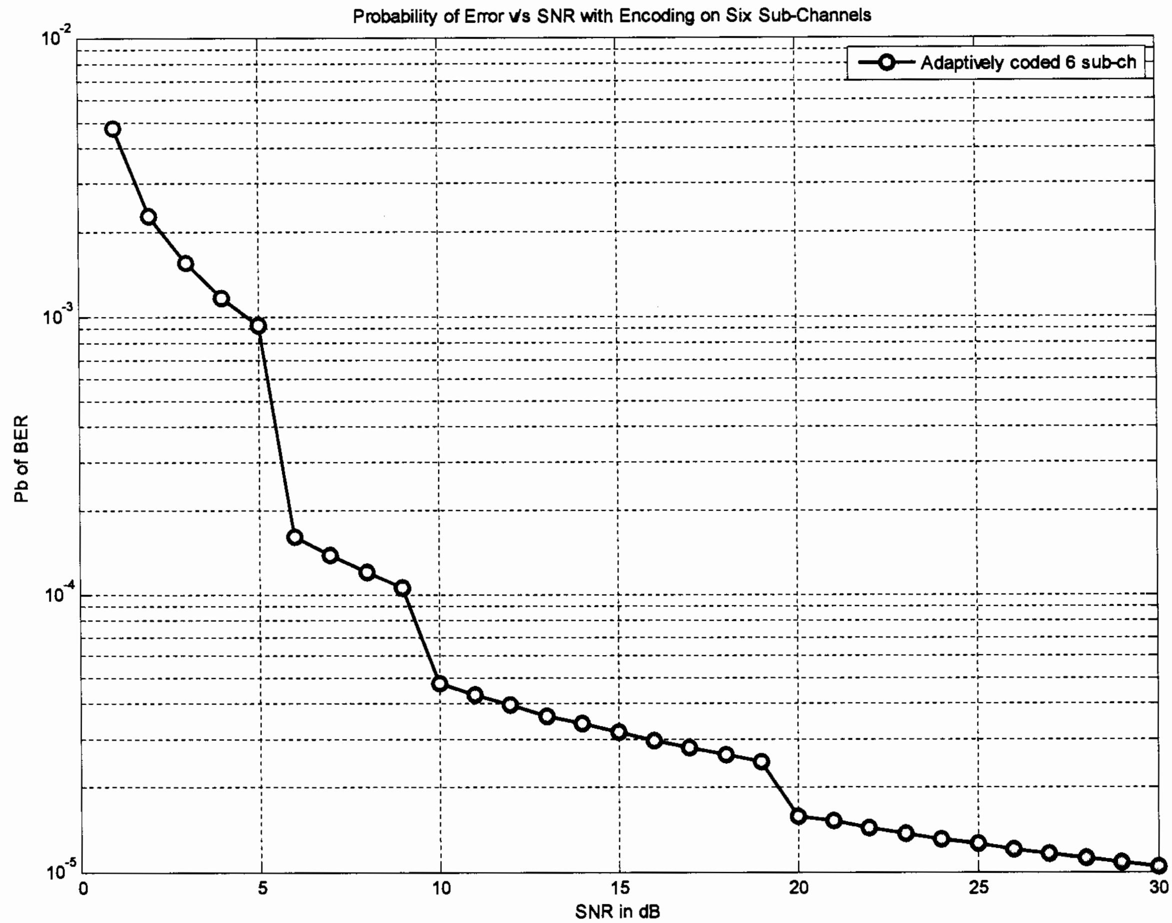

Figure 5.4: Error probability of Encoded system

5.2.2.2 Throughput

The throughput of the encoded system depends on two factors one the modulation scheme and other the Shannon capacity of the sub-channel and the channel itself. As the capacity of the channel increases as the SNR increases, the capacity of the sub-channels also increases respectively. This increase in capacity combined with the modulation order determines the overall throughput of the system. The factors that affect the throughput of the system are the available bandwidth, the signal power and the bit rate per second. Slight change in any of the above mentioned variables and the results will show a significant change.

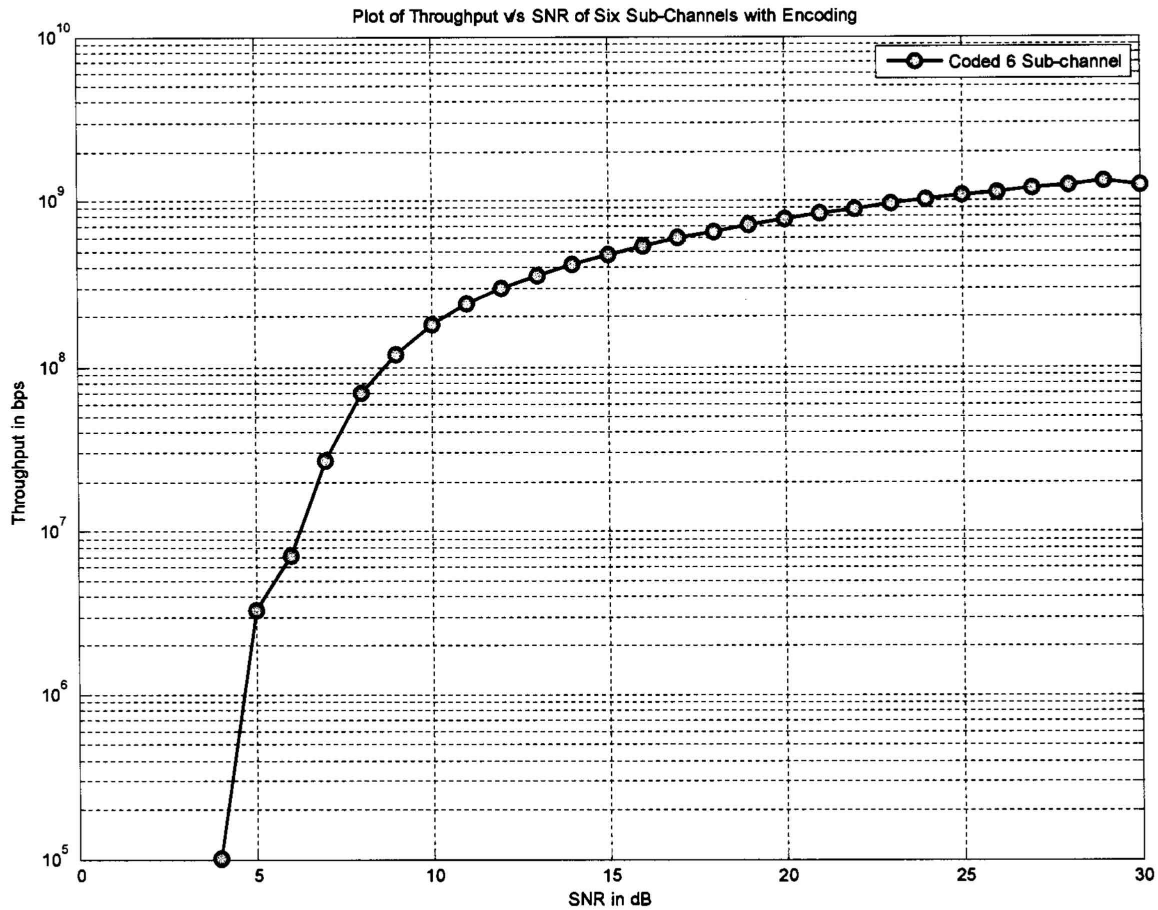

Figure 5.5: Throughput of encoded system.

5.2.3 Comparisons between Referenced and Proposed Predicament

The figures below illustrates that the referenced adaption of H/2 displays a finer performance in terms of throughput, while sustaining a low PER. It was also possible to attain a certain throughput while using less power which would lead to less interference and, hence, to an increase in the system's capacity. Furthermore, the enhancement can outperform H/2 in applications where the probability of error requirements is tight - constrained to less than 1%. The complexity of the computation was measured by the number of operations per decoding, for decoding proposed codes is usually much less or at worst comparable to that for decoding the convolution codes used in current commercial OFDM systems.

Nevertheless, these results when compared with that of the proposed predicament the findings are quite pleasant. The throughput is maximized while attaining the minimum probability of error. Therefore, as in general the proposition was able to obtain an optimally maximized throughput.

5.2.3.1 Error Probability

In the following Figure 5.6 there is a comparison between the error probability of the referenced and the proposition along with the Hyperlan system's error probability. It can be seen that the error probability is of the proposed system is quite good. Not only that it is more reliable but also converging. Since the systems discussed are all flat fading therefore, it will be interesting to compare the results with a fast fading system. If the proposition was implemented with a multipath fading channel then the results in light to our current findings would be similar to performance of the referenced algorithm in terms of error probability. To further improve the performance, one way would be to decrease the number of bits sent per second. Since the throughput of our system has profoundly increased to the magnitude of up to 1000Mbps, the decreased sampling time would result in lower error probability. However, the throughput of the system would still be quite high.

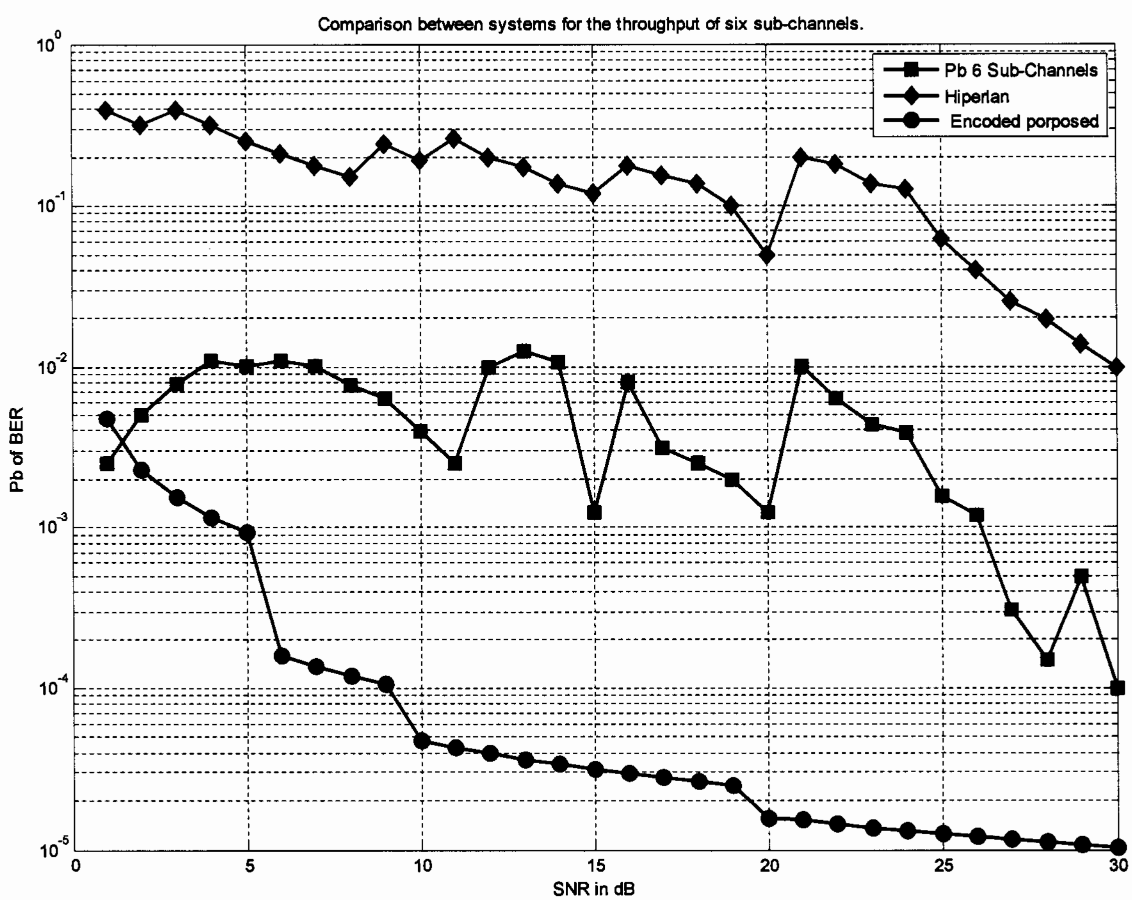

Figure 5.6: Comparison of error probability of the referenced and proposed algorithm.

5.2.3.2 *Throughput*

The main idea of the proposition was to maximize the throughput. This goal was achieved and is displayed in the following Figure 5.7. As it can be seen, the throughput has been maximized to the magnitude 100Mbps. This has been achieved for a system that has a bandwidth of about 1MHzs with a sampling time of 1/1Mbps. The formulae of throughput for our system depends upon of these will affect the Shannon capacity of the channel and the sub-channel. Not only this the power distribution among the sub-channels will also be affected. This in turn affects the throughput. The drive to increase the throughput further would result in degradation of error the signal power, bandwidth and sampling time. Changing anyone probability. If for a time being we neglect the fact of degradation in error probability, throughput of profound magnitudes can be achieved. The factors that would help in increasing the error probability is increasing the sampling time. Power and bandwidth are quite hardwired variables and would costly to change them to suet are intentions. The figure below shows us that initially the performance of the referenced algorithm are better but after SNR = 4dB the performance of increase in signal power or bandwidth or both and our proposition outperforms and provides us with very high throughput.

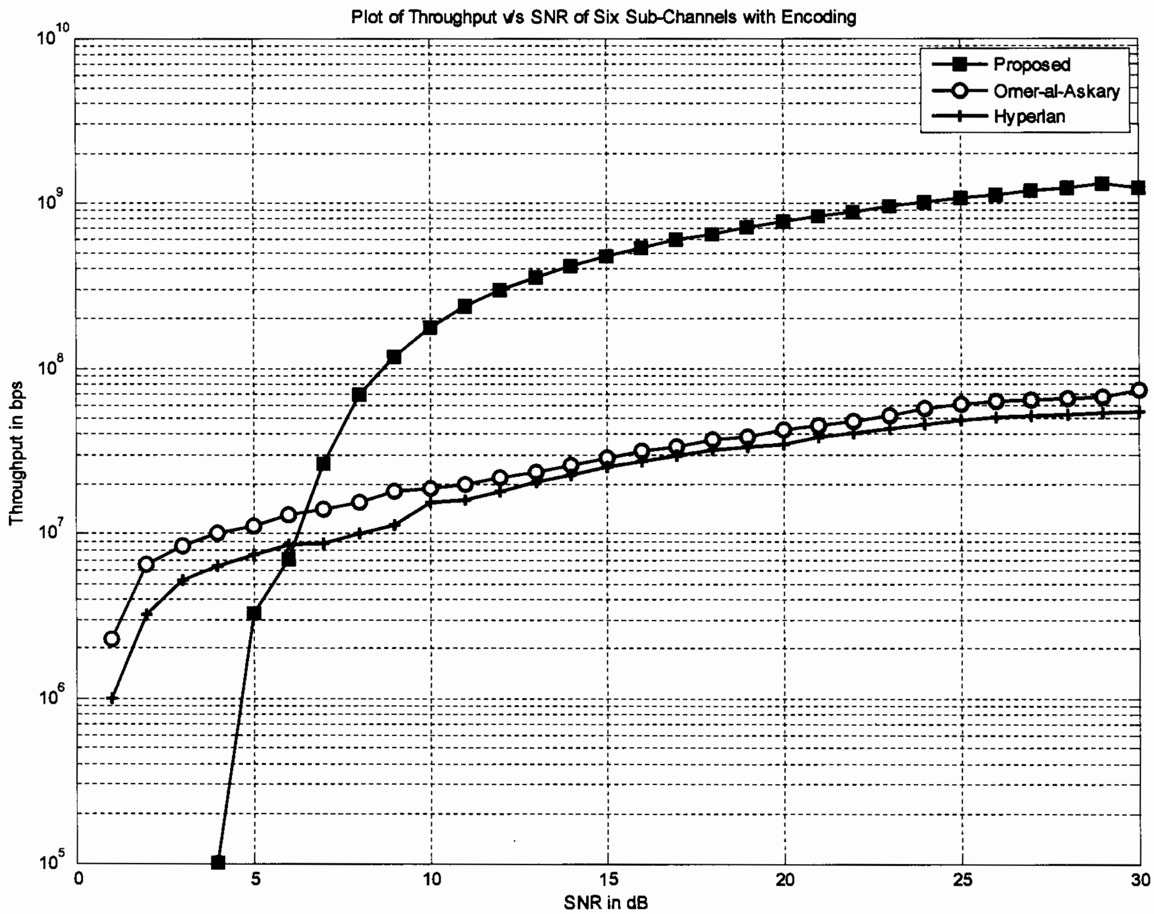

Figure 5.7: Throughput comparison.

5.2.4 Comparisons between systems with different channels

The following diagrams will give a comparison between the outcomes of the system with different number of channels. Originally, the system in this thesis was compared to that of the Omer-al Askary's for six sub-channels. In this section, the comparison has been extended to system's performance for 2, 4, 12 and 24 sub-channels in addition to that of 6 sub-channels. These comparisons were carried out in order to understand and evaluate the system under different conditions.

5.2.4.1 Error Probability of uncoded system

The following Figure 5.8 gives a comparison in the performance of the proposed system in five different scenarios. The comparison is in the form of probability of bit error rate without coding. It is seen that the probability of error decreases as the number of channels decreases. In the following case, the probability of error is the least for the system with two sub-channels. In addition, as the number of sub-channels increases the probability of error increases. However, the increase in bit error rate is not very significant as the number of sub-channels increases. This implies that the performance of the system with more sub-channels has a comparable performance to that of less number of sub-channels. Therefore, the performance of the adaptively modulated system is more or less the same in terms of bit error rate.

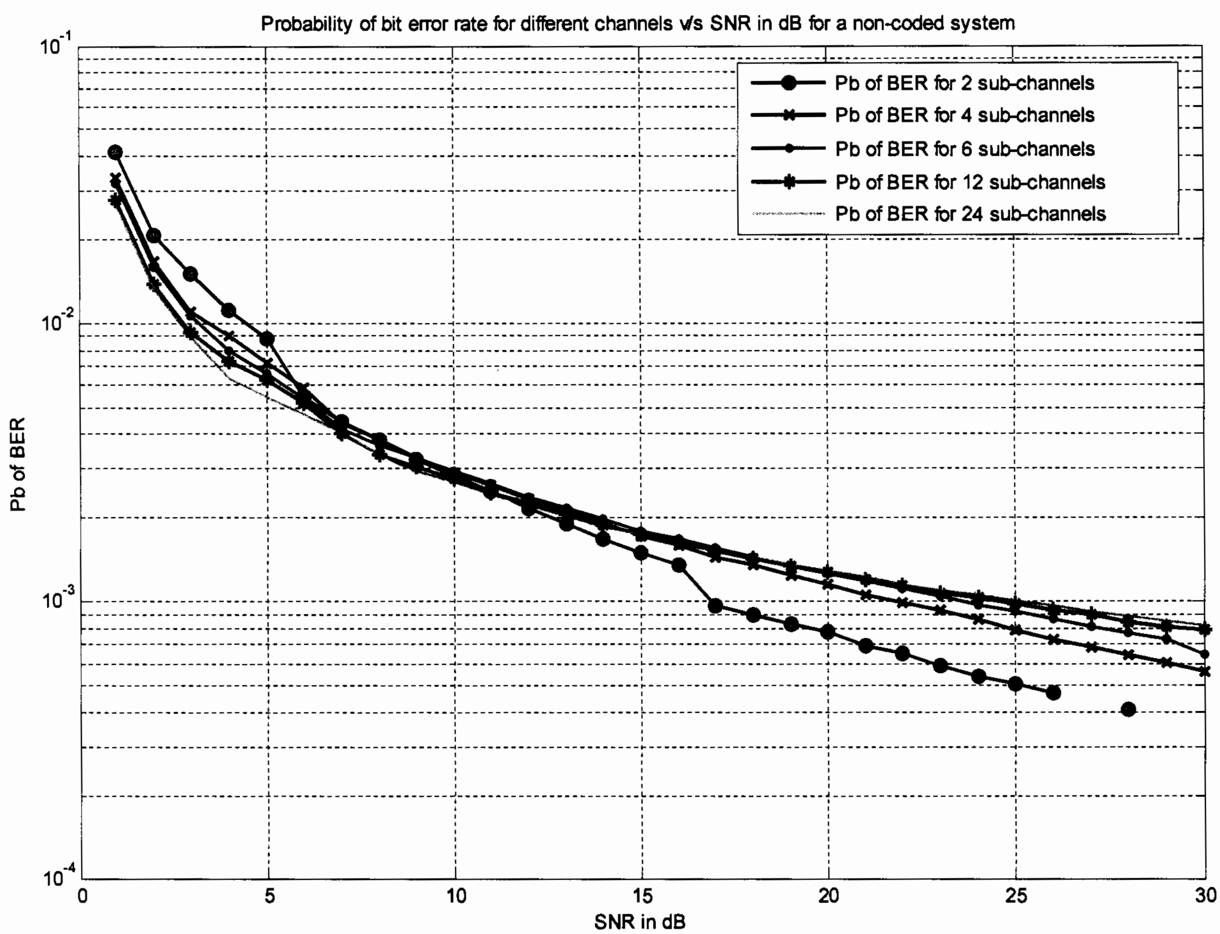

Figure 5.8: Probability of error for different channel numbers without coding.

5.2.4.2 Error Probability of coded system

In the above Figure 5.9, the comparison between different channel numbers in a uncoded system was presented. The following Figure 5.9, a comparison between the same sub-channel number is presented but for a coded system. This is done because the main idea is to improve the bit error rate. The comparison is between 2, 4, 6, 12 and 24 sub-channels. The criteria itself of this thesis is to achieve the highest attainable throughput while maintaining the probability of error to an acceptable level. It is seen in the figure that probability of error decreases as the SNR increases. Also the probability of error for 2 sub-channels has been improved in comparison to the probability of error for a 2 sub-channels in a uncoded system. Furthermore, the probability of bit error rate increases as the number of sub-channels increases. But it is seen that the probability or error is still acceptable.

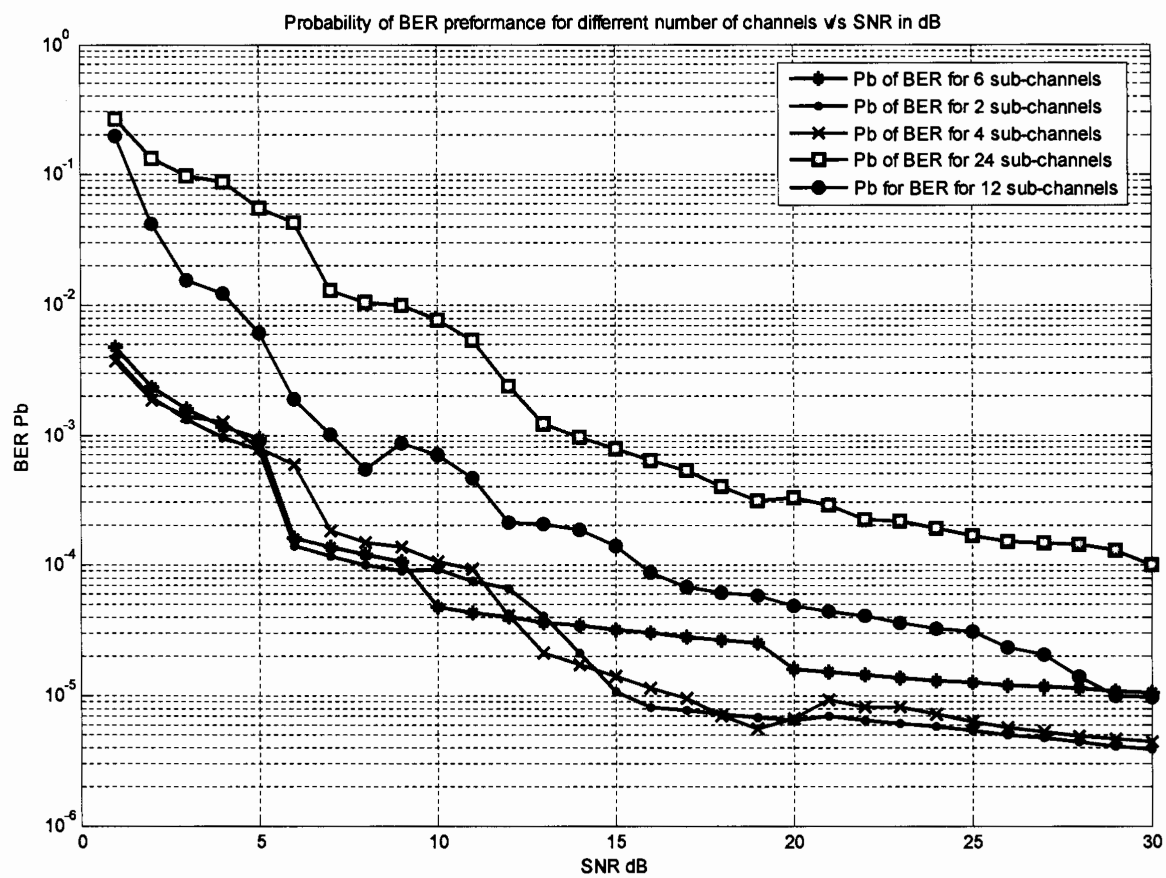

Figure 5.9: Probability of error for different channel numbers with encoding.

5.2.4.3 Throughput of uncoded system

In this thesis, the main idea is to increase the throughput. In Figure 5.10, the comparison of different number of sub-channels is presented. In which it is seen that the throughput increases as the number of sub-channels increase. The throughput of 2 sub-channels is about 10Mbps less than the throughput of 24 sub-channels for the same conditions. This behavior is as expected. The cost of increase in the throughput is the increase in probability of error.

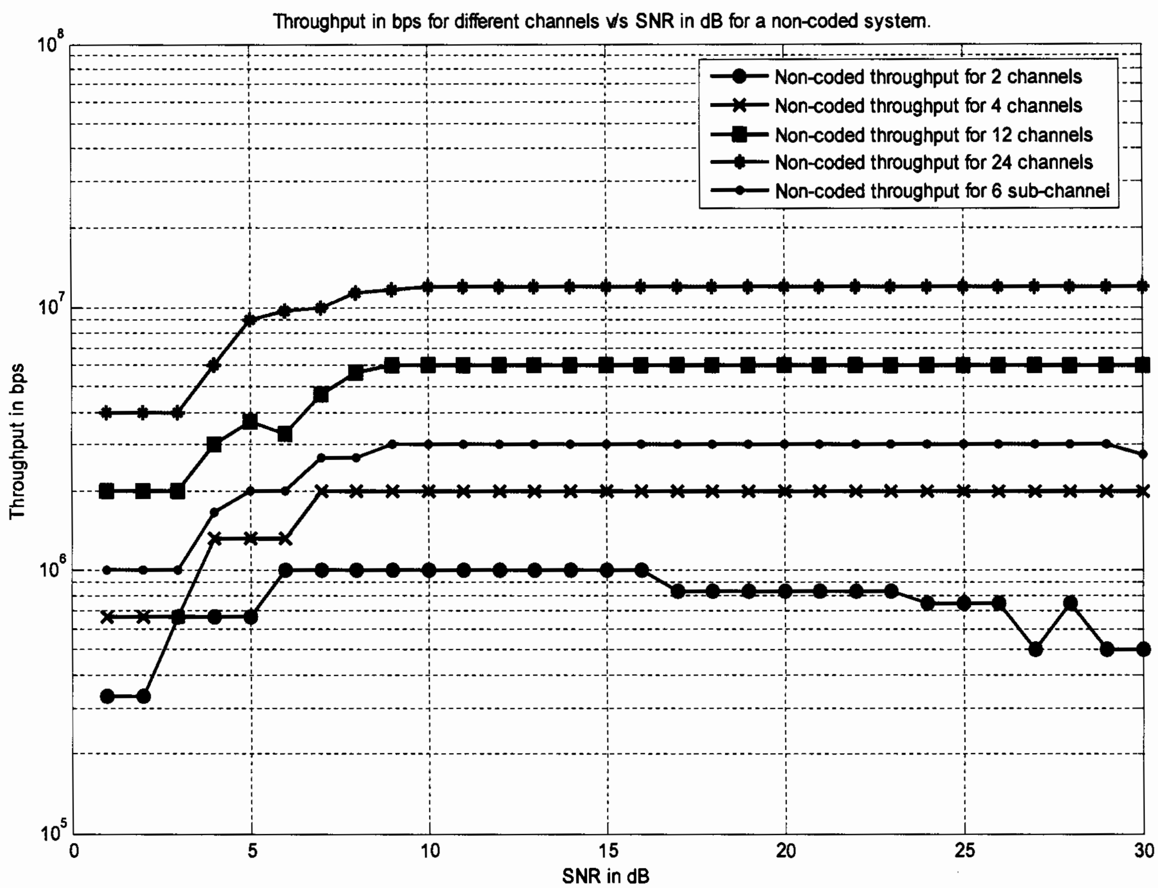

Figure 5.10: Throughput of uncoded system for different number of channels.

5.2.4.4 Throughput of coded system

In this section, the results of the proposed predicament are displayed in the context of different number of sub-channels. In the

Figure 5.11, the throughput for different number of sub-channels is displayed. It is clear that as the number of sub-channels increases the throughput increases. This behavior is similar to that of the uncoded system. The main difference is in the magnitude of the throughput attained. The results show that for 2 sub-channels the throughput is maximized for 10^8 at 30 dB. The increase is log-exponential starting from 20bps at 1dB. For 4 to 6 sub-channels the throughput is zero for the initial values of SNR. The performance of 12 sub-channels is

3×10^9bps as maximum attainable throughput. The throughput although is nil for the first 6 dB's. On the other hand, the 24 sub-channel has attained throughput greater then 10^{10} bps.

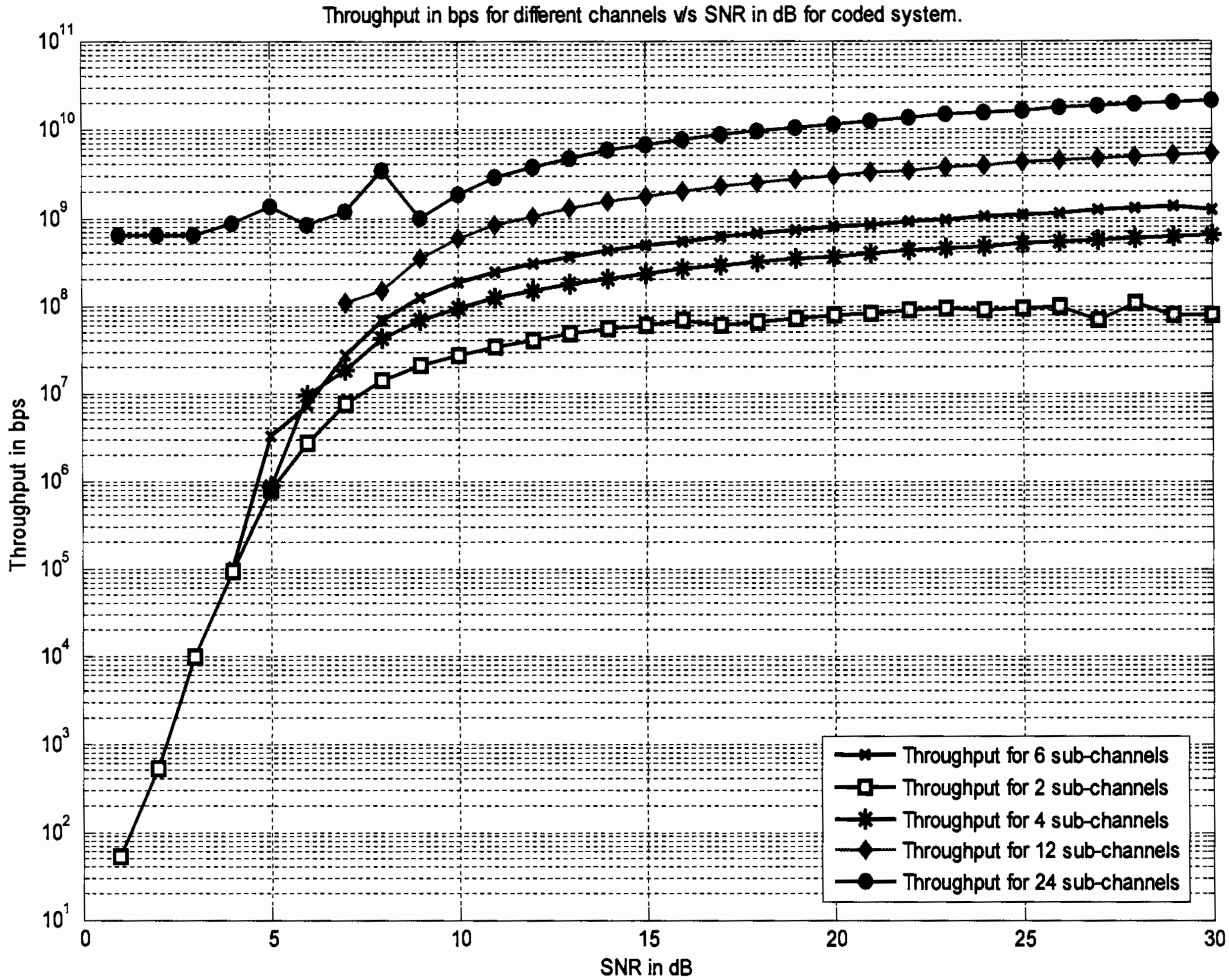

Figure 5.11: Throughput for different number of channels for coded system.

The comparison of our system to that of Omer-al-Askary and Hyperlan are given in the Figure 5.12. The results show that the performances of the Hyperlan and reference paper are out done even for 4 sub-channels. While the Hyperlan and the reference paper results are true for 6 sub-channels. The only difference is that for all SNR their throughput is present and steady.

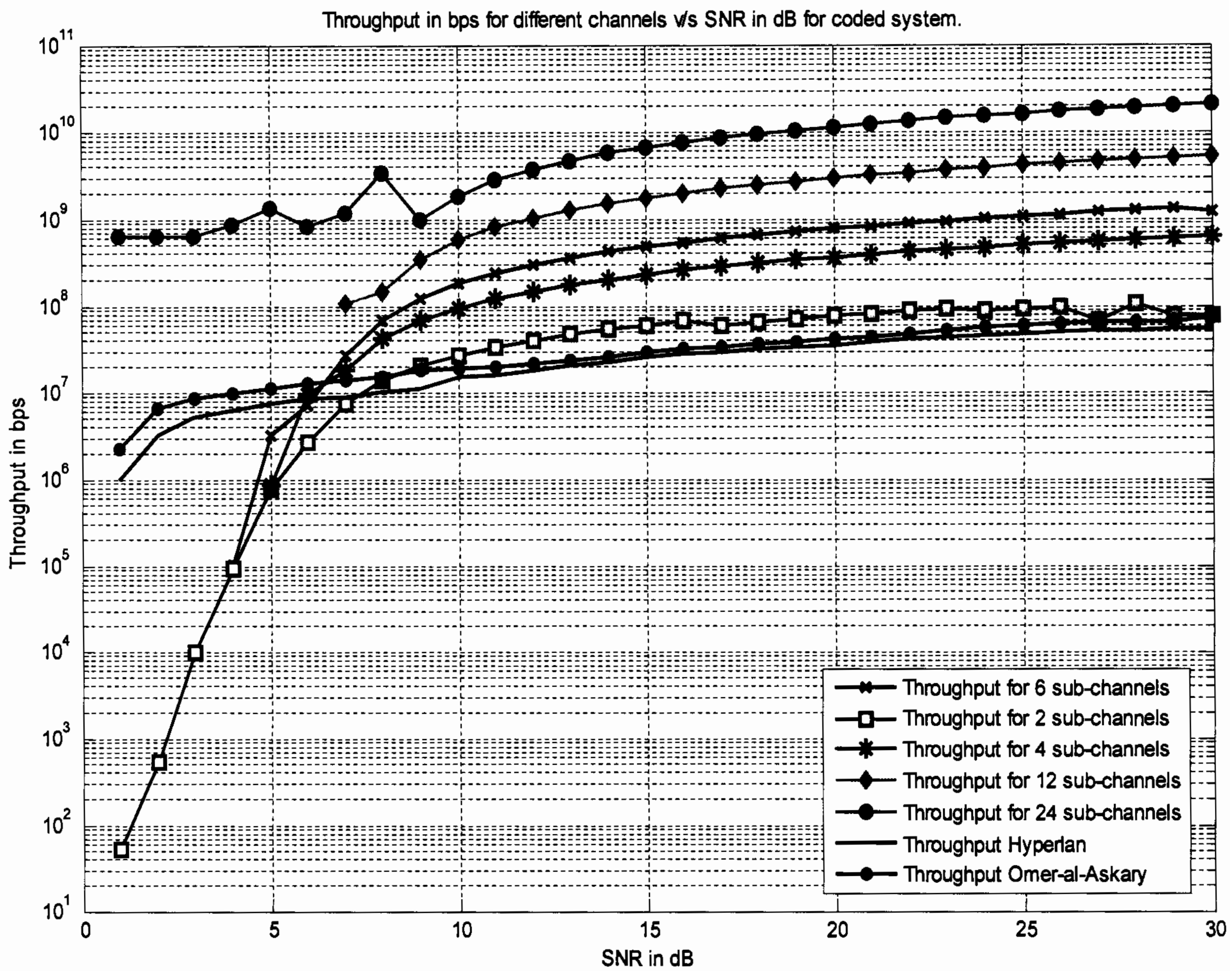

Figure 5.12: throughput comparison of proposed system to that of Omer and Hyperlan.

5.2.5 Comparisons between Systems with and without Encoding

This is a comparison of the adaptive modulation in terms of encoded and non-coded system. Performance measurement parameters are error probability and throughput of a encoded and non-coded system.

5.2.5.1 Error Probability

It is quite obvious that the error probability of our system with encoding outran the non-coded system. Although it is not a smooth curve but the error rate is lower. The unsmooth decay of error is due to the adaptive modulation criterion. A smoother curve is only possible by totally ignoring the error rate. However, this would result in disastrous communication. A communication in which there is no reliability is not good communication. Such algorithms are useless. Therefore, it would be impossible to ignore the error rate.

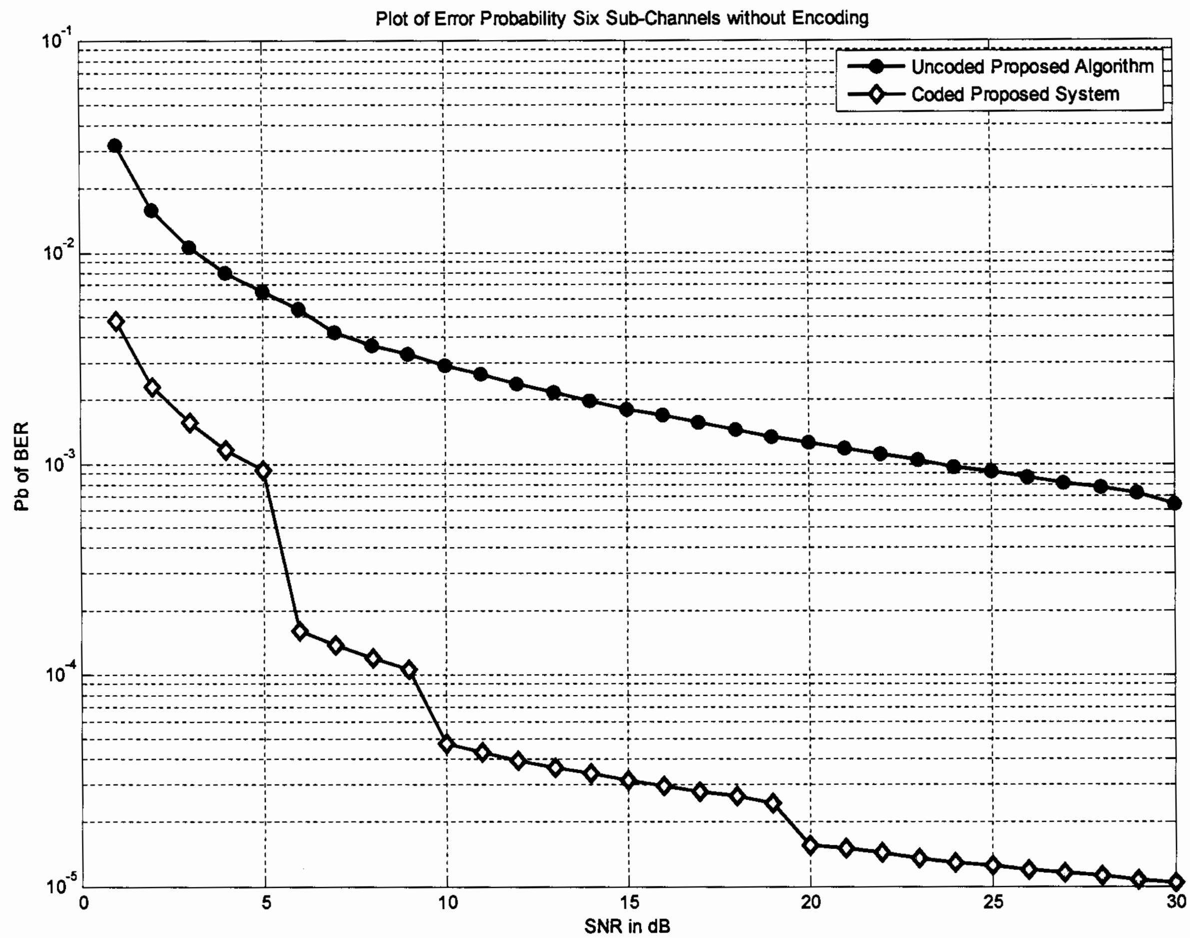

Figure 5.13: Comparison of error probability of encoded and non-coded system.

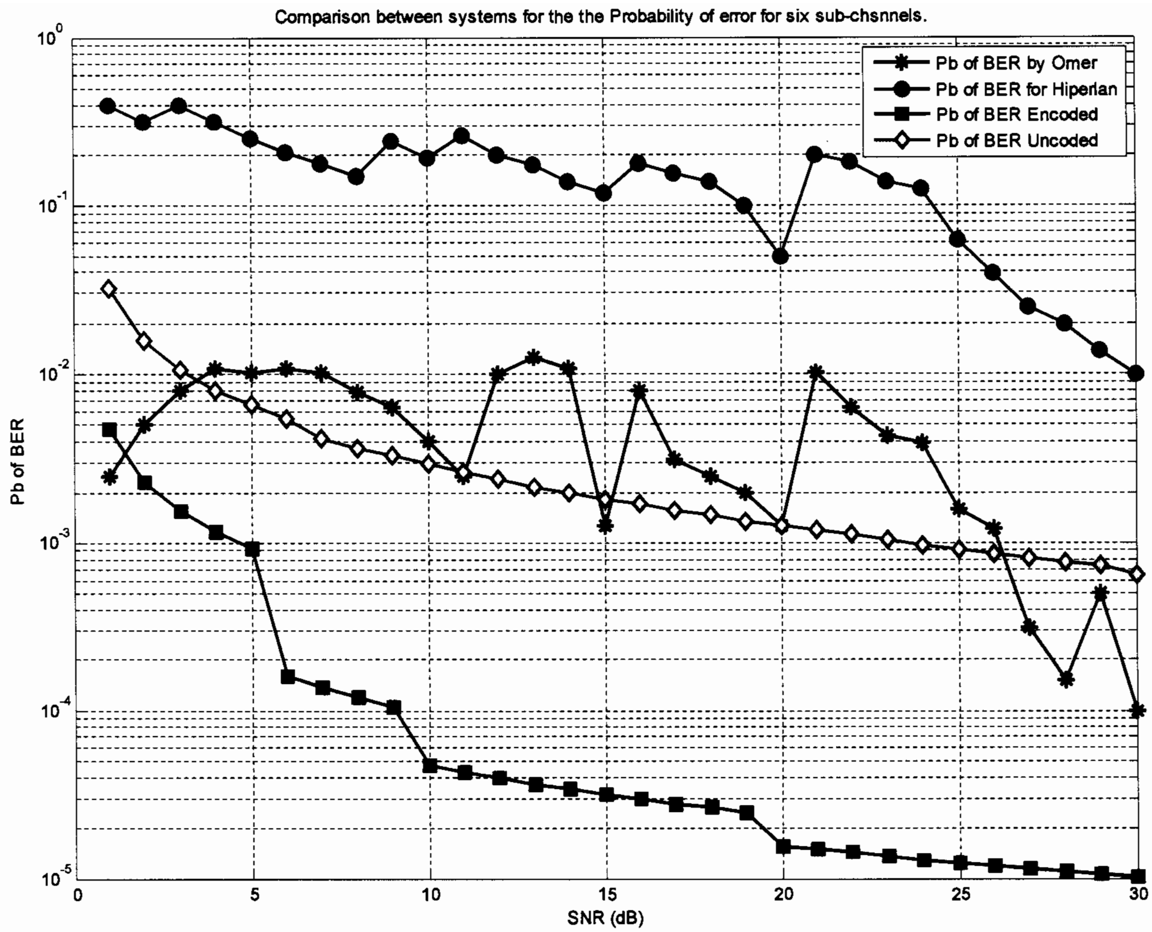

Figure 5.14: Comparison of error probability for all the systems.

5.2.5.2 Throughput

There is great increase in the throughput of an encoded system to non-coded system. This result is shown in Figure 5.15.

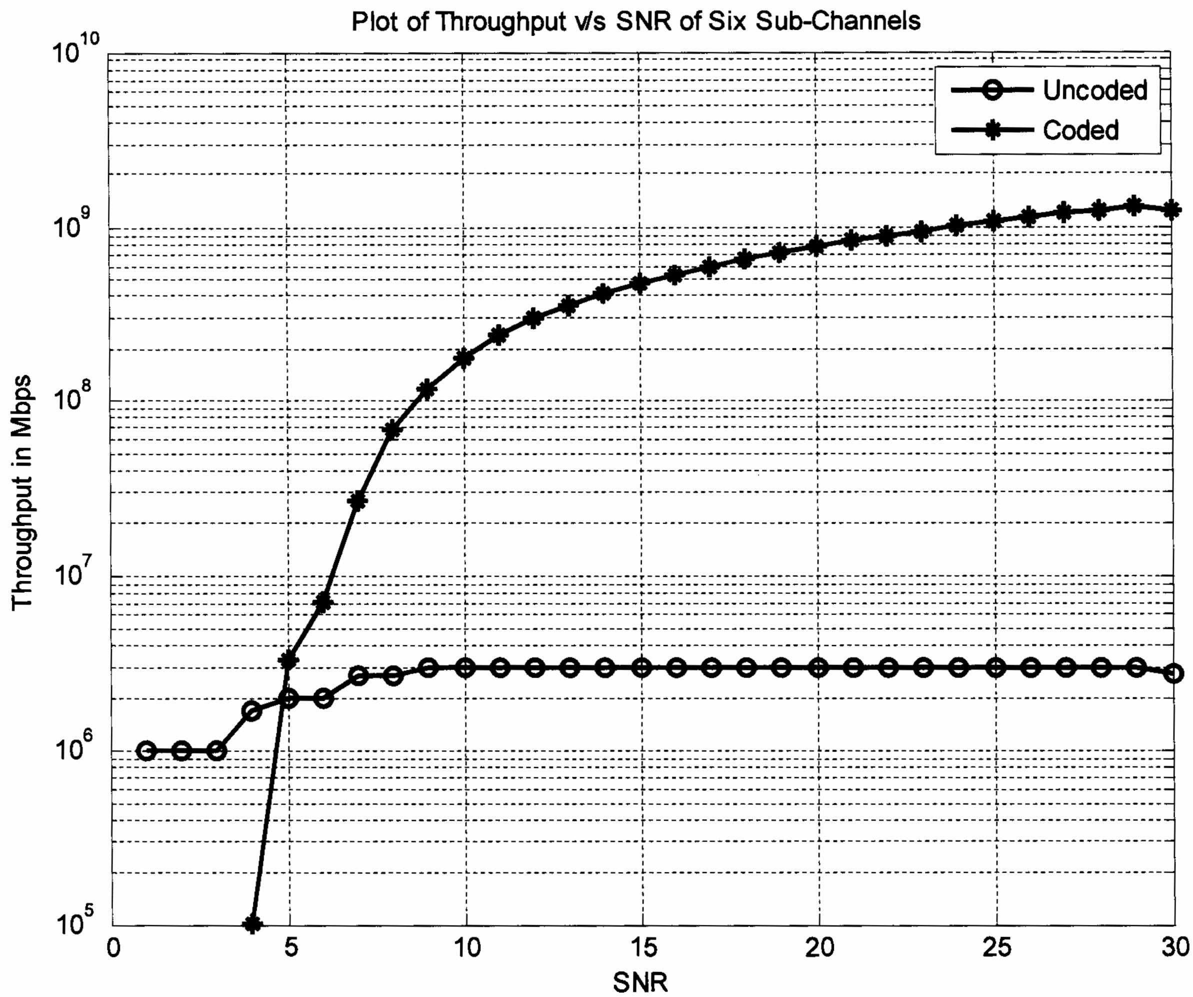

Figure 5.15: Comparison of throughput of the two systems.

5.2.6 **Complexity**

The proposed system is implemented for 6 sub-channels. This is done as the referenced paper and the Hyperlan are both implemented for 6 sub-channels. From the above figures, it is seen that the proposed system has out done the algorithm by Omer-al-Askary. First the system performance of the uncoded system was compared. The performance of the coded system out ran the uncoded system. The BER was also with in an acceptable range. The main requirement was to attain high throughput. As it can be seen the results were quit satisfying. The catch was the increase in the execution time. The reason for this was the fact that more data per sub-channel is being sent now. Which implies that the there is more decoding operations required. Thus, more additions and multiplications are to be executed per sub-channel. So in general the higher throughput is attained at the cost of complexity in decoding which results in increase in execution time. The Table 5.3, is formulated for an overview of the execution time for our proposal and that of Omer's paper. In this table, the execution time of different number of sub-channels is also depicted. The table shows the proposed system does have a longer execution time due higher complexity.

Table 5.3: Listing of Execution Time.

System	Number of Channels	Execution Time in seconds (s)	Execution time in Hours (hr)
[redacted]	[redacted]	[redacted]	[redacted]
	4 sub-channels	7334.508	3.04 hr
	[redacted]	[redacted]	[redacted]
	12 sub-channels	27397.495	7.61 hr
	[redacted]	[redacted]	[redacted]
Omer-al-Askary	6 sub-channels	13727.94	3.641 hr

CHAPTER 6
CONCLUSION AND FUTURE WORK

6.1 Conclusion

It has been shown in the section 5.2 of proposed predicament that the results of the modification are better in both aspects that is the throughput and the probability of error as well. Thus the modification better utilizes the channel condition. We have been able to achieve these results because in state of deep fading the transmission would shutdown. Since a main focus was to improve our throughput; it was observed that whenever the channel was in good condition the modulation scheme with the highest rate would be selected while maintaining the acceptable probability of error. This in turn will adaptively select a coding scheme for each sub-channel. Here, the complexity, measured by the number of operations per decoding, is much less or at worst comparable to that for decoding the convolutional codes used in current commercial OFDM systems [51].

Since each sub-channel has its own MCC therefore more redundancy is added. Therefore, data is better protected as compared to the conventional systems. In addition, since each sub-channel accommodates different horizontal codes with multiple rates, therefore multiple users can be accommodated on each sub-channel. The rate of the code and the user will have a one to one correspondence. The idea here is that the combined rate of all the codes horizontally into the rate of the vertical code will best fit the rate supported by that sub-channel's

adaptively selected modulation scheme. Thus more number of users are accommodated then the number of available sub-channel. The minimum number users that can be accommodated at a given instance is 2*N, where N the number of sub-channels that are available at that time. If the rate of the user and that of the horizontal codes do not correspond then some interleaving can be used to accommodate different users of lower rate into the codes of higher rates. By doing this the resources will not be wasted since no sub-channel would sit idle.
Here the arrangement of the sub-channels in a particular order is not required. Also there is no need to have a particular code arrangement.

Here adaptive coding and adaptive selection of modulation scheme according to individual channel conditions results in better performance of MCC. MCC is build up on linear binary codes and these codes are able to provide good performance in channels with burst errors without the need of an interleaver/de-interleaver. These codes also proved to perform well in fading channels.

Proposed algorithm results in great performance gains as compared to conventional systems but at the cost of system complexities.
Here transmission is closed when the channel is in deep fade. Therefore, it is possible to obtain some gain in total rate by switching off the transmission when the quality of the channel degrades below a certain level. This is a simple form of adaptation similar to that presented by Goldsmith et al in [52]. However, it is maintained that the net gain in rate is very small especially for high signal to noise ratios, which is the same conclusion in [52].
GMD decoding of generalized concatenated codes requires very low complexity. This decoding method decodes all errors of weight less than half the minimum distance. GMD decoding performs very well if the minimum distance of the concatenated code is very large, i.e., close to the maximum possible value. However, if the minimum distance is not very large, most of the error correction potential lies in correcting error patterns slightly greater than half the minimum distance [1]. Proposed algorithm result in minimum system complexity and results in great performance gains as compared to conventional systems

6.2 Future Work

This is an open field to work for anyone further. Here we list some future enhancements that we believe are the most important ones:

- The process of power allocation to each sub-channel should be investigated further in order to improve the performance in terms of both the throughput and the probability of error.
- The proposed predicament is implemented for higher throughput. One scenario may be to improve the performance in terms of error probability. Since the results showed good error probability for higher throughput in our case, it implies that for an acceptable throughput very good error probability could be achieved.
- In this thesis, the adaptively selected modulation scheme was the criteria of code selection. One enhancement could be, to either let the code and modulation but be selected depending on the sub-channel condition. Moreover, both should have a say in the keeping and discarding of sub-channels with bad conditions.
- Implement the proposed predicament for a multipath fast fading Rayleigh channel.
- Improving the error probability of the system by implementing the iterative decoding.
- One further enhancement can be the implementation of MIMO along with OFDM.
- Here MCC can further interleave before modulation. For large codes, the interleaving minimizes the probability that a symbol error will result in several errors in the same column. This, in turn, minimizes the probability of an uncorrectable error in one column [1].
- We are just using Binary BCH codes. Non-binary codes can be used to encode MCC. Binary codes can be used to encode several rows at the same time. For example, a single code can be used to encode all or some of the rows of the first level, a second code for all or some of the rows in the second level, etc. This results with fewer granularities in the structure of the code and possibly lower performance. However, a single decoding pass decodes many rows at the same time instead of decoding each row separately, which decreases the decoding complexity [1].
- A further improvement on the structure of the code is that the rows of a codeword block can be combined in different ways such that the transmitted symbols for one codeword block belong to different modulation schemes [1].
- Different decoding schemes can be used.
- The proposed criteria can be implemented in 4G technologies like Wi-Max.

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