Soft Combining for Improved Matrix Parity Decoding in Error Prone OFDM Systems

A Project Report

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CHITRA K R

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THESIS CERTIFICATE

This is to certify that the thesis titled **Soft Combining for Improved Matrix Parity Decoding in Error Prone OFDM Systems**, submitted by **Chitra K R**, to the Indian Institute of Technology, Madras, for the award of the degree of **Bachelor and Master of Technology**, is a bona fide record of the research work done by her under our supervision. The contents of this thesis, in full or in parts, have not been submitted to any other Institute or University for the award of any degree or diploma.

Dr. K Giridhar
Research Guide
Professor
Dept. of Electrical Engineering
IIT-Madras, 600 036

Dr. Sheetal Kalyani
Research Guide
Assistant Professor
Dept. of Electrical Engineering
IIT-Madras, 600 036

Place: Chennai

Date: 10th June 2014

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ABSTRACT

KEYWORDS: Error Control Coding; Matrix Parity; Interference; Joint Estimation; OFDM Systems; Chirp Detection

Wireless communication systems are ubiquitous. Over the years, tremendous amount of research has gone in to develop the various paradigms of wireless communication viz. modulation schemes, error control coding, channel estimation, data acquisition and estimation schemes, securitisation of wireless links, receiver architecture etc. However, more often than not, wireless communication links are constrained by various aspects like bandwidth, channel conditions, receiver complexity, operational conditions etc and hence it is imperative to choose the right set of schemes for each paradigm of wireless communication addressing the specific constraints for a system. In this project we have developed a communication framework for a multiuser aeronautical system which is constrained by a severe Doppler offset, fairly high data rate requirements, asynchronous users leading to interference and low latency.

We then shift our focus mainly to the error control coding scheme and effectively using the scheme in not only error control but also in joint estimation of data. Error control coding are techniques that enable reliable delivery of digital data over unreliable communication channels. This makes them an essential element in systems that are highly error prone, just like ours. We explore various soft combining and matrix parity based methods of efficiently combining repeated copies in time and frequency. We also study the effects of modulated interference on these combining schemes and select the best for implementation.

We also venture into jamming signals and look at a specific type of jamming signal used frequently called chirp signals. We explore its properties and look at some effective algorithms used to detect chirp and remove them completely.

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ABBREVIATIONS

OFDM	Orthogonal Frequency Division Multiplexing
QPSK	Quadrature Phase Shift Keying
IBI	Inter Block Interference
ICI	Inter Carrier Interference
LBR	Low Bit Rate
MBR	Medium Bit Rate
HBR	High Bit Rate
FFT	Fast Fourier Transform
GCL	Generalized Chirp Like
MMSE	Minimum Mean Square Error
LMMSE	Linear Minimum Mean Square Error
MRC	Maximal Ratio Combining
ML	Maximum Likelihood
AWGN	Additive White Gaussian Noise

CHAPTER 1

Introduction

Wireless communication framework is dependent on the type of resources available and the requirements of the users. Cellular communication is between users spread all over the world. There are smaller scales of communication systems like WiFi and Bluetooth which are spread over tens of meters and have different channel characteristics and environmental issues compared to other types of communication systems. Each type of system needs its own special framework and design to tackle the issues specific to that system.

Our system is a set of transmitters and receivers in an aeronautical scenario and we need to design a communication framework for the same. This system has the availability of a W Hz bandwidth. Unlike conventional cellular communication systems, the number of users is limited to a maximum of M users. Also, there are no major obstacles in the vicinity, and there is almost always a direct line of sight path from the transmitter to the receiver. All the users are not synchronized, so different users start transmission at different instances of time.

1.1 Motivation

In any communication link, coding and decoding form an integral part of the transmit and receive chain. We have multiple antennas for better reception at the receiver. These antennas give rise to diversity and enable combining of the signals received at various antennas to get a better version of the received signal. Similarly, we also have multiple copies of data by repetition of symbols in time and frequency domain. This can be viewed as data received from slightly different channels, which also allows us to apply data combining methods. The main thrust of this thesis is to explore the diversity in the different copies of data obtained and how to effectively combine data in the presence of interference.

Soft combining and decoding is important when we have multiple copies of data. Since we are not receiving the multiple copies of symbols from different antennas, there is a lot of scope to explore the characteristics of the various copies we receive and extract meaningful data from them. The main aim of my project is to explore how we can effectively get the most useful data from various copies in the presence and absence of interference, using knowledge of the channel and the system framework.

There is a requirement for the system to be designed on an OFDM framework. The next section explains OFDM, and the coding, decoding and combining is part of the constellation mapping and equalization.

1.2 OFDM

OFDM - Orthogonal Frequency Division Multiplexing is a method of encoding digital data on multiple carrier frequencies. A large number of closely spaced orthogonal sub carrier signals are used to carry data on several parallel data streams or channels. Each sub carrier is modulated with a conventional modulation scheme at a low data rate, which in our case will be QPSK modulation.

The primary advantage of OFDM over single-carrier schemes is its ability to cope with severe channel conditions - attenuation of a certain set of frequencies or narrow-band jamming, interference, and frequency selective multi-path fading. Channel equalization is also simplified because OFDM can be viewed as using many slowly modulated narrowband signals rather than one rapidly modulated wideband signal.

Hence we design our required system with OFDM framework and modify those aspects which are specific to the aeronautical characteristic of our model.

1.3 System

In our work, we will specifically be looking at the receiver architecture for Equalization, Soft Combining and Decoding. The conventional OFDM structure needs to be improvised to cater to the specific requirements of the link. The main requirements imposed are:

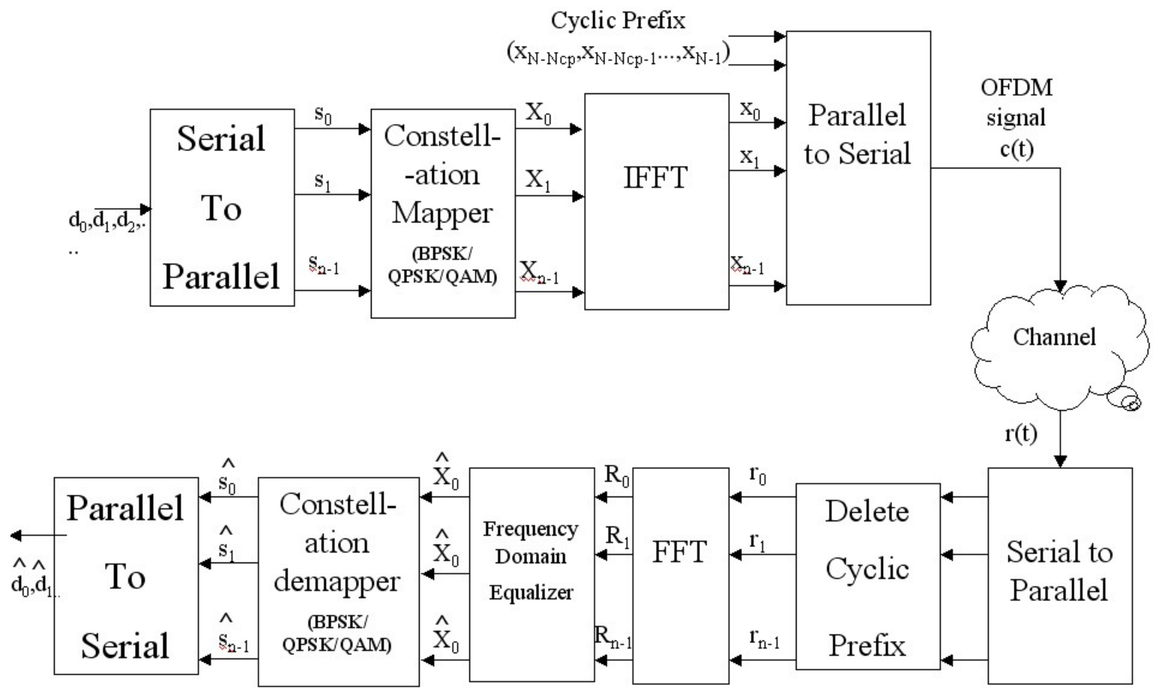

Figure 1.1: OFDM block diagram

- Need to have low latency in decoding and fast-acquisition in high doppler environment.
- The error correction needs to be as simple as possible for lower latency and complexity in processing the data.
- Due to the presence of strong interferers, the decoded data needs to be highly reliable.
- This system is also affected by narrow band frequency jammers, to account for which the data needs to be distributed intelligently.
- The number of users in the system is variable. Correspondingly, variable data rates need to be available based on the number of users currently using the resources.

The various issues affecting the system have also been highlighted above. Another issue is that the transmission from various users is asynchronous. Considering each of these issues, we have designed an OFDM framework for the transmitter and receiver chain. This design modifies the way frequency subcarrier allocation, hopping, equalisation is done. All these design aspects will be explained further in the next chapter on System Model.

1.4 Scope of work

As described in Figure 1.1, the OFDM communication model is made up of a number of blocks. These blocks have been combined into modules and I have worked on the data combining, coding and decoding part of the receiver chain. The various requirements and issues with the system have been explained in the previous section. Every issue is addressed with specific additions in the framework design, each of which is explained in the next chapter.

To test the coding, decoding and combining part of the receiver chain, we have implemented various algorithms on our coding-decoding test simulation in Matlab, for scenarios with and without interference. This simulator is explained in the Appendix.

For testing the entire system from the transmitter to the receiver with synchronization, equalization, channel estimation, coding, decoding and combining, we have worked on the simulator provided by CEWiT - Center for Excellence in Wireless Technology, at the IIT Madras Research Park. This work has just been started and there is a lot of scope to continue simulations of various test cases on this simulator. We have integrated the code on combining various copies, but the test cases can be refined further and the algorithm can be improved.

Chapter 2 explains the System Model in detail, about each aspect of the OFDM framework and how we have had to modify the various aspects of conventional OFDM systems. It also explains the entire receiver architecture, coding and modulation schemes used, frame structure and subcarrier allocations.

Chapter 3 explains a simple Maximal Ratio Combining of the multiple copies of data we receive from repetition in time and frequency domain. This is the most primitive form of combining and can be improved further by incorporating interference aware detection schemes.

Chapter 4 is about Maximum Likelihood Estimation and joint Maximum Likelihood based combining for multiple copies. The ML criterion has been derived for detection in the presence and absence of interference, and for interference aware and interference ignorant detectors.

The final chapter is a conclusion of results obtained and about what further work can be done.

This project is a combined effort of myself and another dual degree student Ankit Behura, and other important aspects of the system design can be found in his thesis, Behura (2014).

CHAPTER 2

System Model

This system is primarily an OFDM Communication link for aeronautical purposes. There are multiple users sharing the same band for communication. Each user intends to transmit to one receiver over a W Hz wideband spectrum. The transmitters are in constant motion at speeds over 800 kmph resulting in high doppler shifts in received signal. There are multiple such transmitter-receiver pairs in the vicinity, each of which are asynchronous, leading to strong interference. Hence there is a need to design the sharing of bandwidth, hopping sequences, frame structure and decoding algorithm for the link. This chapter outlines the important design aspects and explain how they help in combating the challenges presented by this communication model.

2.1 Frequency domain subcarrier allocation

Unlike conventional systems, the entire bandwidth here is not used for transmission. This is a W Hz system, but the entire bandwidth is divided into two halves, which, without loss of generality, is referred to as the upper band and the lower band. All data in the upper band is repeated in the lower band, and hence at this stage, only half the bandwidth is effectively being used.

Further, all the subcarriers in any one band are not used for data transmission. Every J^{th} subcarrier is allotted for preamble data with $J-1$ data sub-carriers between them. In a single frame, when the preamble is being transmitted, the data subcarriers are null and the preamble subcarriers have the respective preamble data on them. Similarly, in that frame when data symbols are being transmitted, the preamble subcarriers are null and the data subcarriers carry data. The frequency domain design can be illustrated as in Figure 2.1.

As can be seen from the figure, there is repetition of data in the frequency domain. This scheme has been proposed mainly for two reasons. Firstly, the presence of frequency jammers in the vicinity can wipe out data in any narrow band set of frequencies.

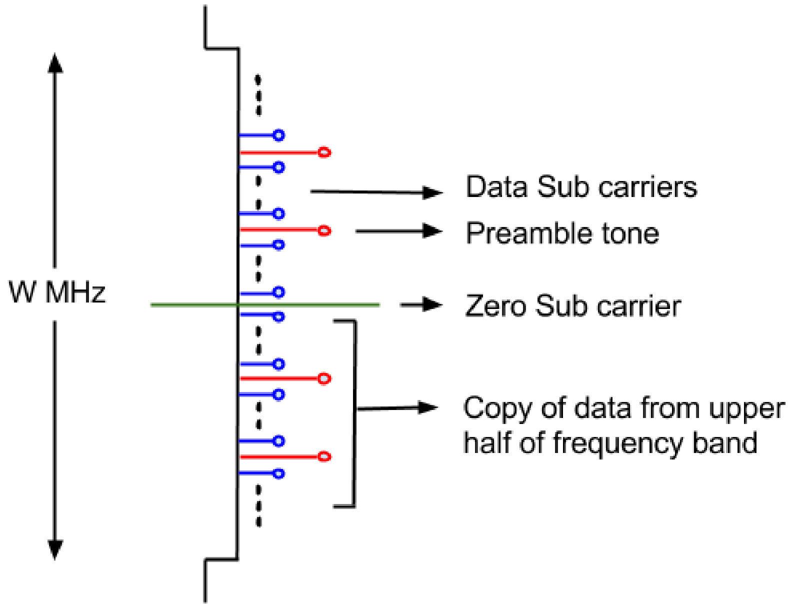

Figure 2.1: Frequency domain subcarrier allocation

To avoid this, repetition of data ensures that there is at least one copy of data. Secondly, this system is prone to strong asynchronous interferers. In the event that one copy of data is contaminated by an interferer, the copy in the other band remains safe.

Figure 2.1 shows data repeated in the same pattern in both the bands. This is just for understanding and the repeated data is not necessarily contiguous. It is actually essential that the data is not contiguous, so that the repeated data is well distributed across the band and whatever contaminates the data in the upper band does not do so in the lower band and vice versa.

2.2 Frequency Hopping

Frequency hopping is a technique of transmitting radio signals by rapidly switching a carrier among many frequency channels, using a pseudo random sequence known to both the transmitter and the receiver. Such a method of transmission offers two main advantages over fixed-frequency transmission:

- Such a system is highly resistant to narrow band interference.
- These signals are difficult to intercept, thus aiding our purpose. A narrowband receiver just sees these signals as an increase in the background noise and it is difficult to intercept the signal without knowing the pseudo random hopping sequence.

Section 2.1 described how repetition of data in frequency domain helps preserve at least one copy of data. This might not always ensure the availability of one good copy, because interference can always affect the other copy. To recover data from the jammer, frequency hopping is also necessary. In conventional OFDM systems, the hopping sequence varies over the entire set of subcarriers as described above. Interference is possible because the users transmit data asynchronously, and even if the design is a hopping sequence for the entire bandwidth, the irregularity in the start times may lead to complete interference. Hence, the proposed design is a framework where users are paired up, each pair of users are allotted some subcarriers and they hop only in those allotted subcarriers. This reduces the risk of complete contamination of data.

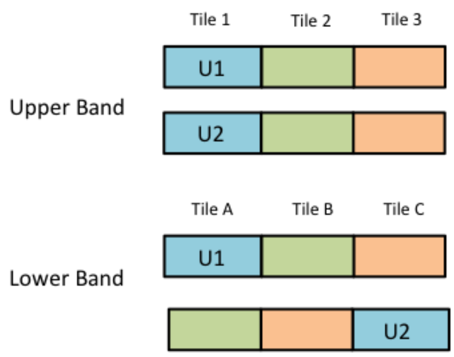

Figure 2.2: Tile hopping pattern

The pairing scheme is explained in Figure 2.2. Each set of $\frac{J-1}{3}$ subcarriers is called a tile. Users are paired up and each pair of users are assigned to three tiles in the upper band and three in the lower band. For example, assume that user 1 and user 2 are paired. Also suppose that these two users are assigned tiles A1, B1 and C1 in the upper band, and tiles A2, B2 and C2 in the lower band. The hopping pattern is so designed that no other users will transmit data from these subcarriers. Also, due to asynchronous transmission, if user 1 and user 2 happen to transmit data in tile A1 in the upper band, then that translates to user 1 transmitting from A2 and user 2 transmitting from B2 in the lower band. In such a scenario, one copy is always interference free.

2.3 Frame Structure

The time domain frame is so designed that it aids in acquisition of symbol boundary in the presence of asynchronous interferers and reduces the effect of Inter Block Interference. Every frame has a preamble in the first symbol, followed by D data symbols, a pilot symbol, followed by D data symbols again. To overcome the issue of asynchronous transmission, the frame is designed such that it contains $2D + 2$ symbols, and each symbol is made of two sub symbols and a cyclic prefix. The two sub symbols are identical and are just repeated in time. Figure 2.3 describes the frame structure in detail.

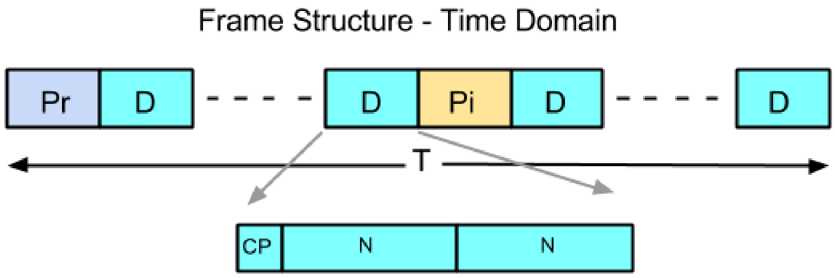

Figure 2.3: Frame Structure

2.4 Data Rates

One of the requirements posed to this system is to have variable data transfer rates. Three data rates have been specified, with corresponding limit on maximum number of users to be supported in the system simultaneously. The three data rates are:

- LBR with M_1 users
- MBR with M_2 users
- HBR with M_3 users

2.5 Control flow

The entire control flow of the receiver has been modularised (refer Figure 2.4) and divided among all students involved in this project.

As in the flow diagram, data is first generated, coded and transmitted. In the receiver architecture, the first step is synchronization, followed by channel estimation, equalization and antenna diversity combining. My work focuses on the next part of the receiver chain, which is soft decoding, interference mitigation and symbol combining.

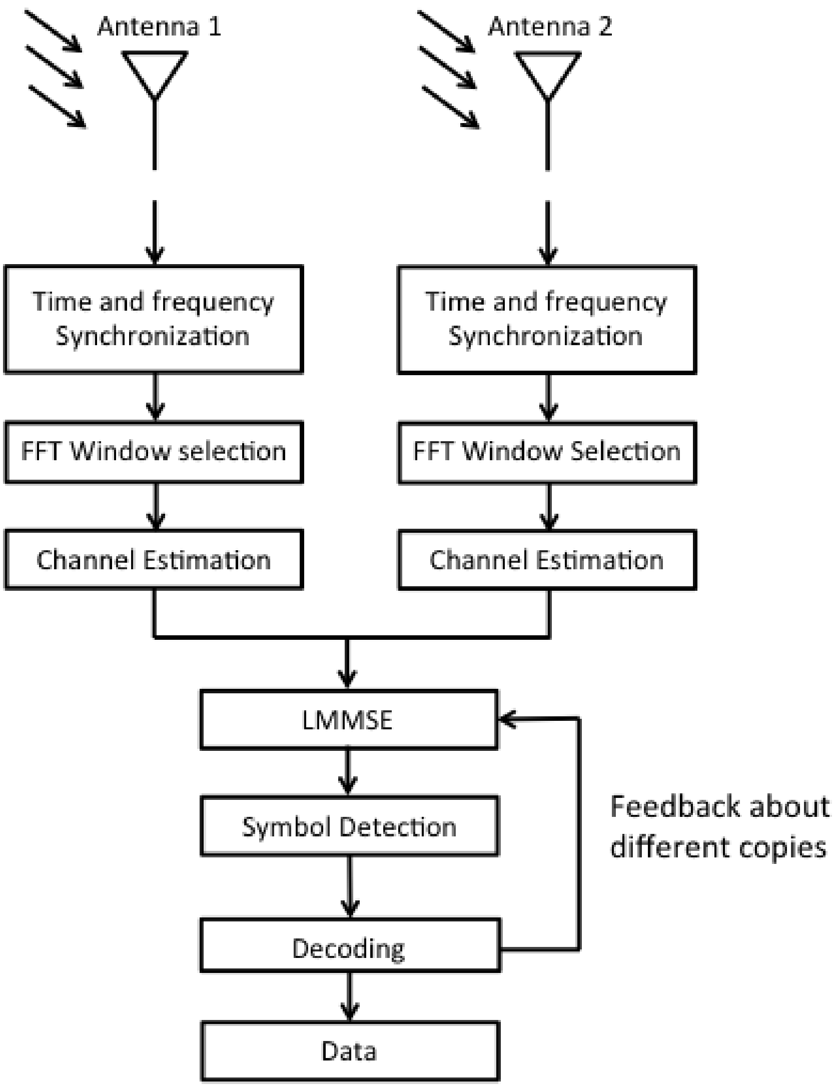

Figure 2.4: Receiver Block Diagram

2.6 Coding and Decoding

Coding is a system of rules to convert information from its basic form to a more efficient form for storage in a medium or transmission through a communication channel. k bits of message can be encoded into n bits of code with $n \geq k$. These extra bits will then help during the decoding process to figure out if there was something wrong in the received bits, and if the coding scheme facilitates it, they could help correct the error too.

Coding is essential to provide error detection and correction capabilities in data received. While there are various methods available for efficient error correction, like LDPC, turbo decoders, etc., this system requires simplicity in implementation on the hardware, and low latency in decoding. Hence, matrix parity coding and decoding is implemented on QPSK modulated data - one set of matrix parity coding on the I phase and one set of matrix parity coding on the Q phase.

Matrix parity coding is very simple and can correct single bit errors. The data is first written down in the form of a 2-dimensional matrix. In our case, there will be 8 bits of data to be encoded at a time. These 8 bits of data can be written as a 4x2 matrix. Figure 2.5 explains this structure. D refers to the data bits intended to be sent. Now the parity bits are assigned. These parity bits are represented by P in the figure. Every row in the matrix is assigned one parity bit, such that the extended row now is always of even parity. That is, the number of 1s is always even. Similarly, every column is assigned one such bit and appended at the end. These parity bits are then considered and an overall parity bit is assigned to them, which will be the last bit in the matrix.

D	D	P
D	D	P
D	D	P
D	D	P
P	P	P

Figure 2.5: Matrix structure for matrix parity

This coding and decoding scheme is simple because while decoding, the rows and columns can be checked if the even parity condition still holds. If there has been a one bit error during transmission, then one row parity and one column parity will fail. This row and column indices will point to the location of the bit which is in error. Flipping that bit will then correct the error.

Thus an 8/15 code rate matrix parity is used - 8 bits of information is to be sent, encoded as 15 bits essentially. Thus there are 15 bits for in-phase transmission and 15 such bits for quadrature phase transmission. The data is then QPSK modulated and two bits per symbol are sent.

While decoding, there are four copies of data - two copies from repetition in time domain and two copies from repetition of data in frequency domain. This improves the performance over a single copy of data, since data can be combined from various copies and the effects of jamming and interference can be mitigated. Data combining is a vast area of research and the following chapters explain more about data combining methods used, and their performance for this system model.

CHAPTER 3

MRC based Soft Combining

Maximal Ratio Combining is a method of diversity combining in which the signals from each channel are added together and the gain of each channel is made proportional to the RMS signal level and inversely proportional to the mean square noise level in that channel. That is, if the received signal is represented as

$$y_i = h_i s + n_i \tag{3.1}$$

then the MRC solution to the estimation problem can be written as

$$\hat{s} = \frac{h_1^* y_1 + h_2^* y_2 + \cdots + h_N^* y_N}{|h_1|^2 + |h_2|^2 + \cdots + |h_N|^2} \tag{3.2}$$

In this system, there are multiple copies of the same data, two copies from time domain repetition and two copies from frequency domain repetition. Every copy contains different channel information and different deforming factors. This can essentially be seen as signals received from different channels because the four copies have slightly varying channel conditions. Thus the diversity here can be exploited and the copies can be combined to obtain a better estimate of data received. While it is possible that two given copies cannot be independently decoded without error, it may happen that on soft combining, previously erroneously received copies can give us enough information to correctly decode.Thus we will now see how MRC based combining affects our system.

First consider the system where there is no interference and only the channel effects distort the receive signal. We will now derive the MRC result for our case.

The received signal can be described as

$$\begin{aligned}
y_i =& h_i x + n_i \\
\text{where, } y =& \text{Received Signal} \\
h =& \text{Channel Coefficient} \\
n =& \text{Noise} \\
i =& \text{Index of data copy}
\end{aligned} \tag{3.3}$$

This equation can me manipulated to obtain x as follows:

$$\begin{aligned}
h_i^* y_i &= h_i^* h_i x + h_i^* n_i \\
&= |h_i|^2 x + h_i^* n_i \\
\text{Summing to } N & \\
\hat{x} &= \frac{\sum_{i=1}^{N} |h_i|^2 x + h_i^* n_i}{\sum_{i=1}^{N} |h_i|^2} \\
&= x + \frac{\sum_{i=1}^{N} h_i^* n_i}{\sum_{i=1}^{N} |h_i|^2} \\
\text{Therefore} & \\
\hat{x} &= \frac{\sum_{i=1}^{N} h_i^* y_i}{\sum_{i=1}^{N} |h_i|^2} \\
\text{where, } N &= \text{Number of copies}
\end{aligned} \tag{3.4}$$

This effectively is Maximal Ratio Combining on multiple copies of data.

Simulations have been carried out for testing the proposed design and algorithm in Matlab. In these simulations, one frame of data has been simulated with repetition only in frequency domain. One assumption is that perfect synchronization has been achieved and that perfect channel estimation has been done. There is no notion of time. Hence there are two copies of data, on which the performance of the proposed design and combining logic has been tested and verified.

Figure 3.1 shows comparison of the performance of MRC with Matrix Parity Decoding as opposed to the performance of just decoding all the copies independently and choosing the copy which has the least number of errors in decoding the matrix parity block. MRC is seen to be better here because both the copies have identical channel

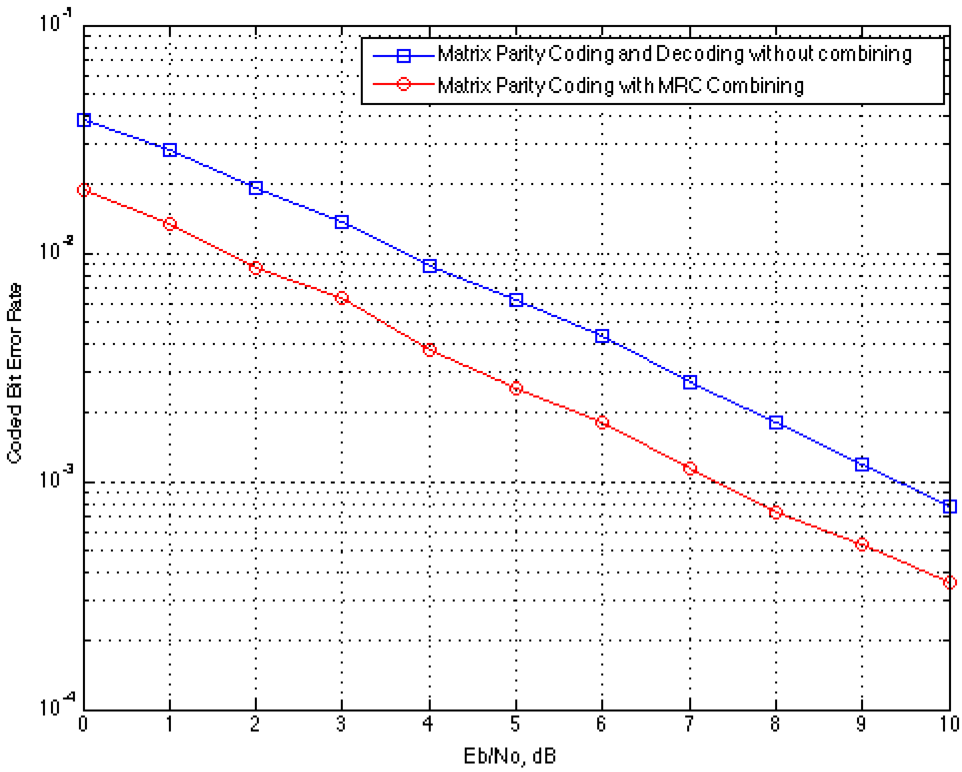

Figure 3.1: Performance of MRC Soft Combining without interference

conditions and more often than not, the two copies will have equal number of errors. In such a scenario, it is seen that the better choice is to combine the two copies.

In the presence of interference, combining all the copies of data might lead to decoding wrong data for the whole frame. One bad copy when combined with decent copies results in bad decoding. This can be overcome by either providing weights to the copies which are combined based on their matrix parity results, or by completely eliminating the interference affected copy. In the current scenario when we do not know which copy is affected by interference, all copies are treated equally and hence the results thus obtained are described by the plot in Figure 3.2, for different strengths of the interferer.

In this method, interference has been ignored. As can be seen from the figure, interference detection and elimination is necessary to know which copy of data is corrupted and derive the best performance from Maximal Ratio Combining.

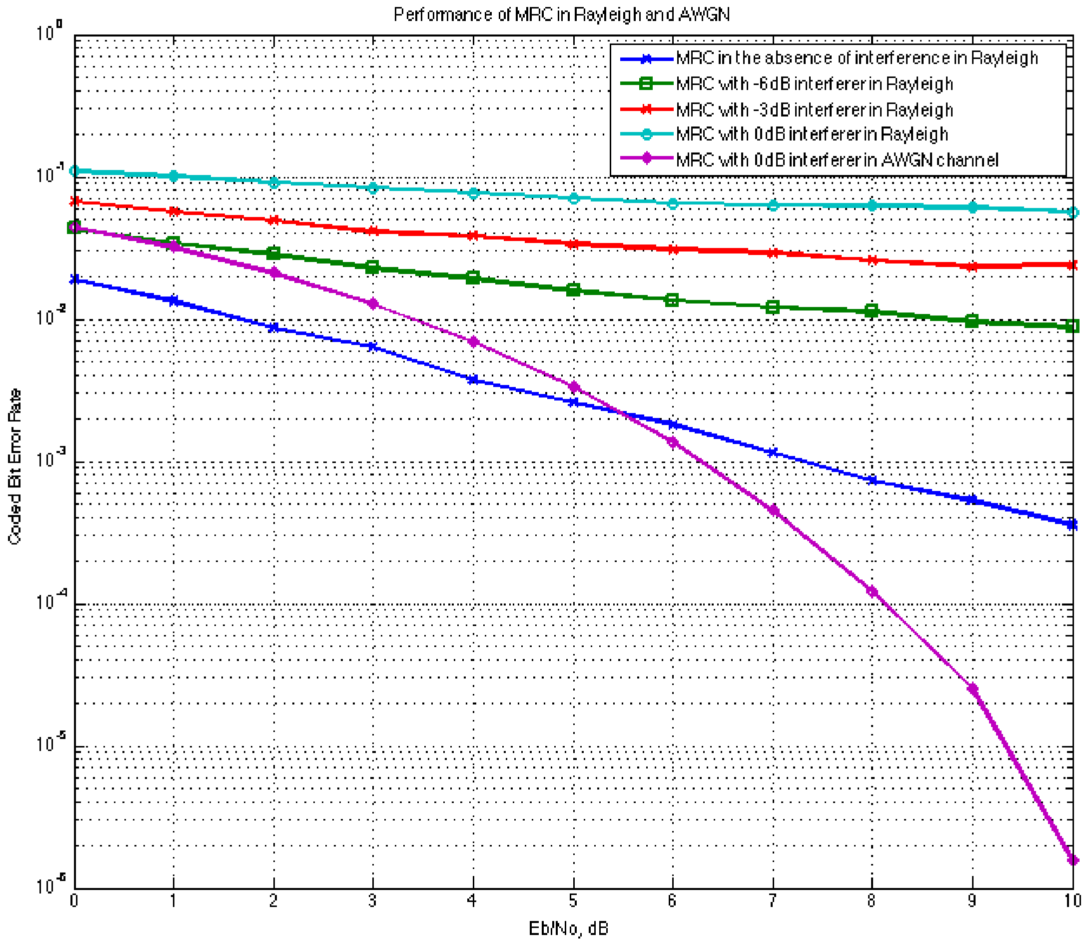

Figure 3.2: Performance of MRC Soft Combining in the presence of interference

CHAPTER 4

ML based Soft Combining

Maximum Likelihood Estimation is a method of estimating the parameters of a statistical model. The method of maximum likelihood selects the set of values of the model parameters that maximises the likelihood function. Intuitively, this maximises the agreement of the selected model with the observed data. Maximum likelihood estimation gives a unified approach to estimation, which is well defined in the case of normal distribution, as we will see in section 5.1. However, in some complicated problems, like in the presence of interference, as we will see in Section 5.2, maximum likelihood estimators are unsuitable, or do not exist. Here the estimator has to be modified to suit the situation.

4.1 ML in the absence of Interference

This section can also be described as the conventional ML detector which functions as the interference ignorant detector. That is, when we have no or very low interference, it can be lumped together with the background noise. In the case of a single copy of data, this detector simply maps the received symbol to the nearest point in the constellation. We will now derive the Maximum Likelihood criterion for our system which has multiple data copies. Consider the scenario where there is no interference in the subcarriers where the user is transmitting. We will derive the likelihood function and the condition which maximises it.

Consider the received signal after equalisation. It can be represented as

$$\begin{aligned} y_i &= x + n_i \\ \text{where, } y &= \text{signal after equalisation} \\ x &= \text{transmitted signal} \\ n &= \text{noise} \\ i &= \text{Index of data copy} \end{aligned} \tag{4.1}$$

Suppose that there are N copies of data available. Our data is QPSK modulated and hence there are four possible constellation points for x. All the four are equally likely. Let X denote the set of possible constellation points which x can take. The Joint ML criterion for the copies can be derived as follows:

$$\begin{aligned}
\arg\max_{x\in X} P(y_1, y_2, \ldots, y_N|x) &= \arg\max_{x\in X} \prod_{i=1}^{N} P(y_i|x) \\
&= \arg\max_{x\in X} \prod_{i=1}^{N} \frac{1}{\sqrt{2\pi\sigma^2}} e^{-\frac{(y_i-x)^2}{2\sigma^2}} \\
&= \arg\max_{x\in X} \frac{1}{(2\pi\sigma^2)^{\frac{N}{2}}} e^{\frac{-1}{2\sigma^2}\sum_{i=1}^{N}(y_i-x)^2} \\
&= \arg\min_{x\in X}\{(y_1-x)^2 + (y_2-x)^2 + \cdots + (y_N-x)^2\}
\end{aligned} \tag{4.2}$$

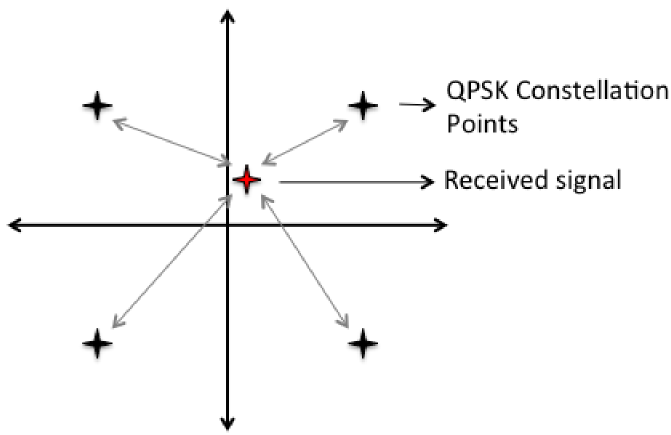

Figure 4.1: ML criterion with one copy

The result signifies that the optimal ML criterion for decision making is the argument (constellation point) which minimises the sum of squares of distances of the received signal from the four possible constellation points(refer Figure 4.1). For example, suppose that the four constellation points are q_1, q_2, q_3 and q_4. First, for q_1, calculate $\sum_{i=1}^{N}(y_i - q_1)^2$ (where N is the number of copies available). Calculate the same metric for q_2, q_3 and q_4 also. Among these four values, the constellation point which has the minimum value is then chosen as $\hat{x}$, the estimated symbol for that received signal. In the absence of interference or in the presence of very low interference, this method of combining is optimal.

The performance of this algorithm was tested on two copies of data in our simulation. In the absence of interference, the results are as expected. A plot of Coded BER vs SNR is shown in Figure 4.2.

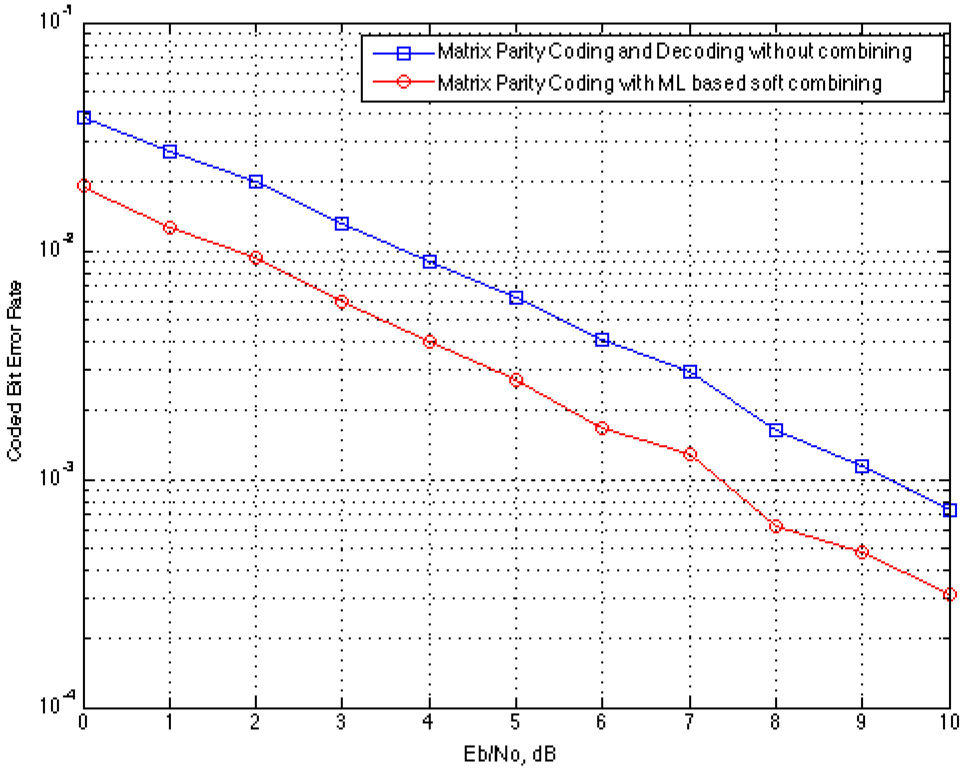

Figure 4.2: Performance of ML Criterion - Plot of Coded BER vs SNR

4.2 ML in the presence of Interference

4.2.1 Interference Ignorant Detection

Interference is commonly viewed as contributing to an error floor and the performance cannot be improved even at higher SNRs. We have seen this in the case of MRC Soft Combining in the previous chapter, where interference forms the error floor, thus inhibiting further improvements.

The result in the previous section assumes that there is very little or no interference. If we use the same expression to estimate the most likely symbol, we are essentially carrying out interference ignorant detection. The performance is obviously much worse than the case with no interference. The performance can be compared in the plot in Figure 4.3.

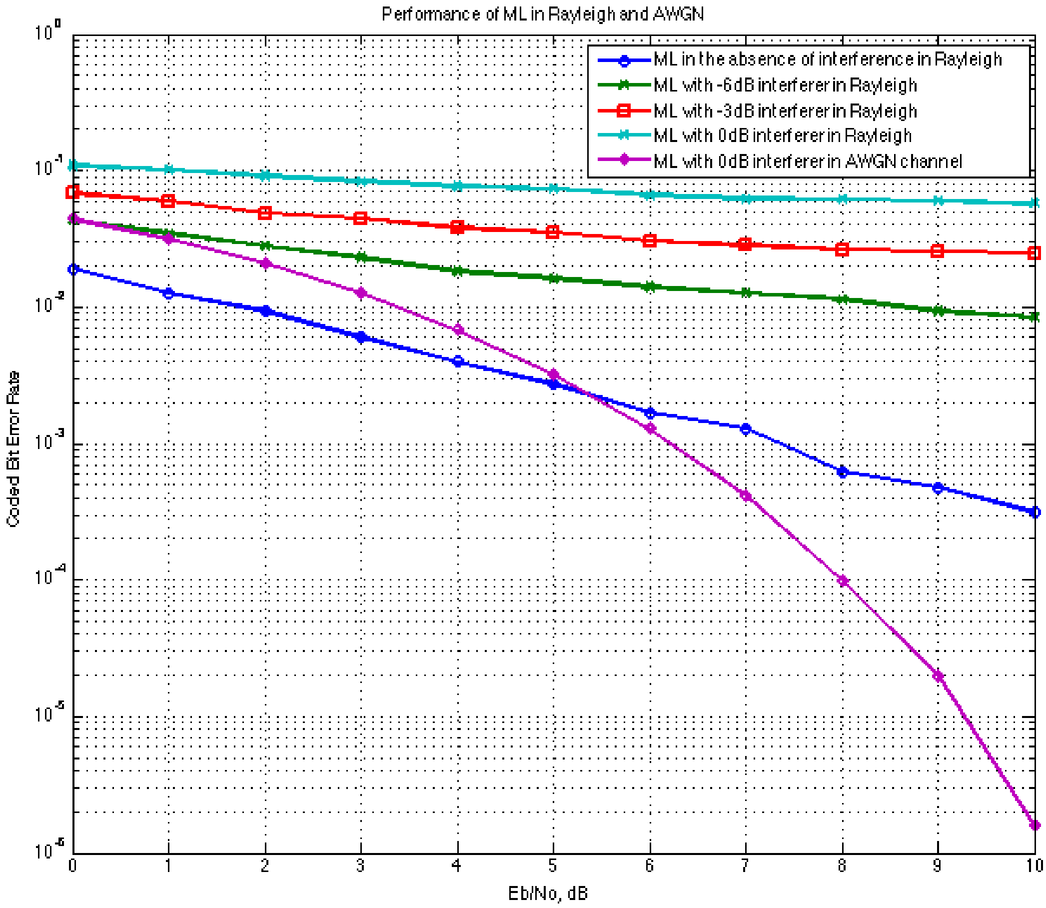

Figure 4.3: Performance of Interference Ignorant Detector - Plot of Coded BER vs SNR

In our simulations, we looked at various combinations which can improve the performance of the interference ignorant detector. We will now describe one such method which gave marginally better results compared to the conventional ML detector.

In the previous section, the detector was:

$$\hat{x} = \arg\min_{x \in X} \{(y_1 - x)^2 + (y_2 - x)^2 + \cdots + (y_N - x)^2\} \tag{4.3}$$

We can modify the condition as:

$$\hat{x} = \arg\min_{x \in X} \left[\min_{i=1,\ldots,N} (y_i - \mathbf{X})^2 \right] \tag{4.4}$$

Here, instead of summing over distance to each constellation point over all copies, we are taking the metric for decision as the minimum of the distances for each constellation point, over all copies. For example, suppose the constellation points of QPSK are q_1, q_2, q_3 and q_4. We measure the distance of the received signal from constellation point q_1, for all the copies of signal available. There will be N such distances for q_1. Now we choose the minimum of these N distances assign this minimum to q_1. Similar procedure is repeated for the other constellation points as well. The constellation point which has the minimum value among these four minimums is chosen as the estimated symbol $\hat{x}$.

The performance of this scheme is slightly better, which can be seen in Figure 4.4.

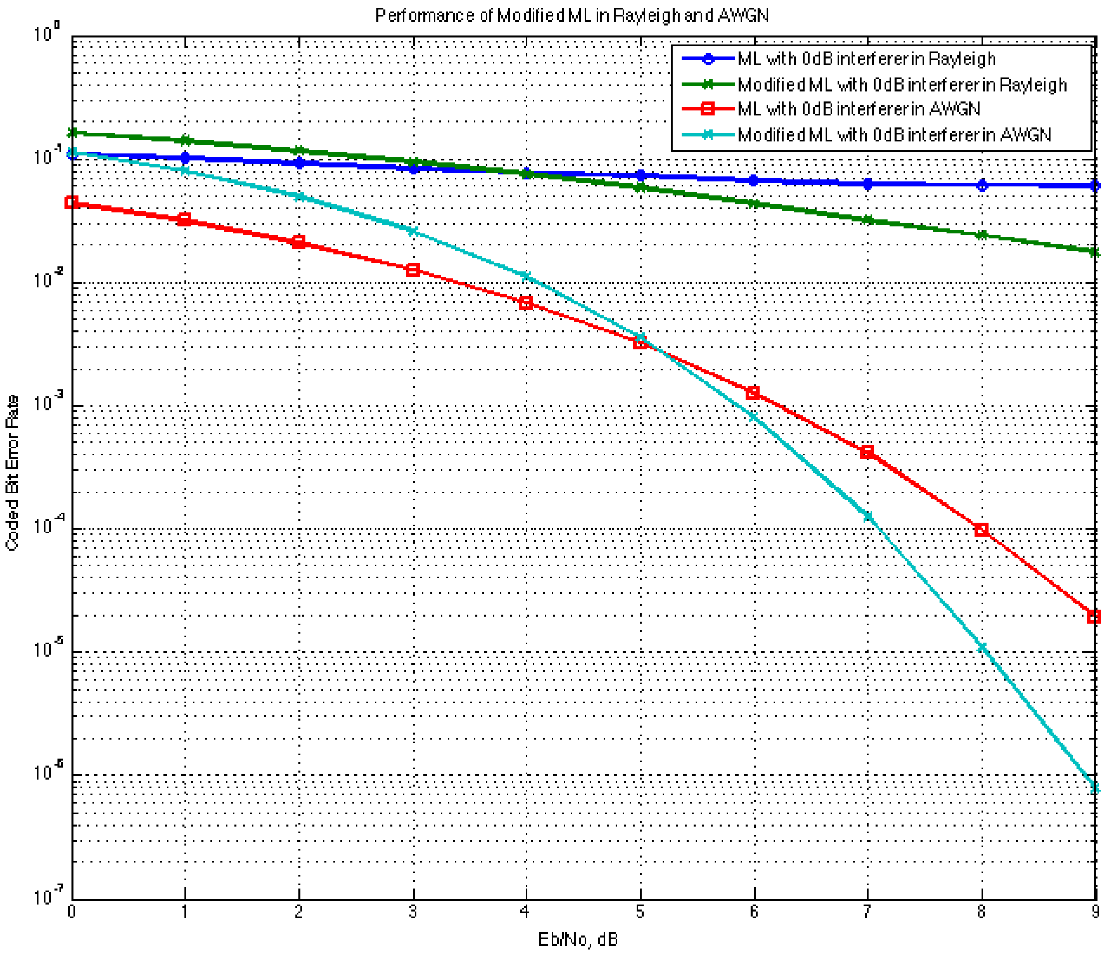

Figure 4.4: Performance of modified Interference Ignorant Detector - Plot of Coded BER vs SNR

4.2.2 Interference Aware Detection

Interference is seen to drastically hit the performance of detectors. However, interference may often be less detrimental than noise of equal power, because contrary to Gaussian noise, the signals from the interferers belong to discreet constellations. We could make use of the fact that the interference is modulated and try to nullify its effect.

The joint ML detector derived in Section 4.1 will not suffice for this case since the interference is assumed to be lumped together with noise. Here, we will treat them as separate entities and derive the results. A reference for this derivation can be found in the work by Lee *et al.* (2011). This method cannot be directly used for our system model, since the inherent assumption for this method is that the channel gain from each transmitter to the receiver is known. It is unknown in our system, and this method needs to be modified to suit our system.

Suppose y is the signal received at Receiver 1. We implement the joint ML estimator on this signal and estimate $\hat{x}_1$, the data sent by Transmitter 1. The signal received at Receiver 1 is given by

$$y = \sum_{u=1}^{M} h_u x_u + z \tag{4.5}$$

where x_u is the signal of Transmitter u, h_u is the gain of the channel from Transmitter u which we assume we have knowledge about, z is the background noise of Receiver 1 and M is the total number of users. By definition, the ML estimate of x_1 is

$$\hat{x}_1 = \arg\max_{x_1} f(y|x_1), \text{where} \tag{4.6}$$

$$\begin{aligned} f(y|x_1) &= \sum_{m_2=0}^{3} \cdots \sum_{m_M=0}^{3} P\{X_2 = x_{2,m_2}, \dots X_M = x_{M,m_M}\}.f(y|x_1,\dots,x_M) \\ &= \frac{1}{4^{M-1}} \sum_{m_2=0}^{3} \cdots \sum_{m_M=0}^{3} f_z(y - h_1 x_1 - \sum_{u=2}^{M} h_u x_{u,m_u}) \end{aligned} \tag{4.7}$$

In the above equation, x_{1,m_1} refers to the m_1^{th} constellation point among the 4 possible points for x_1.

From this we can derive that the ML rule is

$$\hat{x}_1 = \arg\max_{x_1} \sum_{m_2=0}^{3} \cdots \sum_{m_M=0}^{3} exp\left(-\frac{|y - h_1 x_1 - \sum_{u=2}^{M} h_u x_{u,m_u}|^2}{N_0}\right) \tag{4.8}$$

To understand the above expression better, assume that there is only one other interferer apart from the signal we want to receive. So for this case, $M = 2$.

The received signal can be written as

$$y = h_1 x_1 + h_2 x_2 + z \tag{4.9}$$

The ML estimate of x_1 can be written as

$$\hat{x}_1 = \arg\max_{x_1} f(y|x_1) \tag{4.10}$$

The pdf f is

$$\begin{aligned} f(y|x_1) &= \sum_{m=0}^{3} P\{X_2 = x_{2,m}\}.f(y|x_1, x_2) \\ &= \frac{1}{4}\sum_{m=0}^{3} f_z(y - h_1 x_1 - h_2 x_{2,m}) \end{aligned} \tag{4.11}$$

In writing the pdf f, we sum over $f(y|x_1, x_2)$ weighted with the probability of each constellation of x_2. The probability of each of the constellation point is $\frac{1}{4}$. The pdf can be written as

$$\begin{aligned} f(y|x_1, x_2) &= f(h_1 x_1 + h_2 x_2 + z|x_1, x_2) \\ &= f_z(z|x_1, x_2) \\ &= f_z(z) \\ &= f_z(y - h_1 x_1 - h_2 x_2) \end{aligned} \tag{4.12}$$

where x_2 takes on different constellation points. The overall pdf is a sum of the above pdf over all possible constellation points for x_2.

The ML criterion for this case can then be derived as

$$\hat{x}_1 = \arg\max_{x_1} \sum_{m=0}^{3} exp\left(-\frac{|y - h_1 x_1 - h_2 x_{2,m}|^2}{N_0}\right) \tag{4.13}$$

To find the argument which maximizes the above expression, we need to calculate the sum of exponential functions and Euclidian distance from received signal to all combined signal constellation points. To get a log likelihood criterion, which is simpler than the above expression, it can be approximated as follows in high SNR conditions.

$$\sum_{m_2=0}^{3} \cdots \sum_{m_M=0}^{3} e^{-\frac{|y-h_1x_1-\sum_{u=2}^{M} h_u x_{u,m_u}|^2}{N_0}} \approx \max_{x_2,\ldots,x_M} e^{-\frac{|y-h_1x_1-\sum_{u=2}^{M} h_u x_{u,m_u}|^2}{N_0}} \tag{4.14}$$

Using this approximation, the ML detector can be expressed as

$$\hat{x}_1 = \arg\min_{x_1} \left[\min_{x_2,\ldots,x_M} \left| y - h_1x_1 - \sum_{u=2}^{M} h_u x_{u,m_u} \right|^2 \right] \tag{4.15}$$

This for the case of two users will simplify to

$$\hat{x}_1 = \arg\min_{x_1} \left[\min_{x_2} |y - h_1x_1 - h_2x_{2,m}|^2 \right] \tag{4.16}$$

This equation means that for every possible constellation point for x_1, we find the minimum value of $|y{-}h_1x_1{-}h_2x_2|^2$ over all possible constellation points of x_2. We then have four such minimum values, one for each constellation point of x_1. The argument (constellation point of x_1) which corresponds to the minimum among these four values is the estimated symbol $\hat{x}_1$.

This ML detector can aptly be named as joint minimum distance detector because the estimate of x_1 is obtained from the combined signal constellation points of all interferers.

In this detector, we assume that we know the channel gain from all the transmitters to the receiver we are interested in. The performance of this detector could not be tested on our simulation since we do not know the individual channel gains from each transmitter to the receiver. This performance can be tested on the simulator provided by CEWiT, the description of which is given in the Appendix.

CHAPTER 5

Conclusion

In this thesis, I have described the various combining methods and their performance in different scenarios. MRC is seen to be simple and effective when there is no significant contribution from the interferers or when there is absolutely no interference, but in the presence of interference, the results deteriorate quickly, owing to the contribution from the interferer in the combining equation.

Joint Maximum Likelihood estimation is seen to be better in the presence of interference. There is also scope to improve the algorithm further as described in Interference Aware Detection in section 4.2.2. This method has not been tested on the simulator, and can be part of the future work to be done on this project.

Matrix parity decoding on multiple copies independently does not give the required accuracy in data estimation. This could probably be improved by selectively combining certain copies and then using matrix parity decoding to eliminate the worse copies. A variation of this selective combining can be found in Ankit's work - Matrix Parity with selective MRC based combining.

Various other algorithms can be derived by a combination of known combining and decoding procedures, to finally adopt the most efficient algorithm which works for all the possible scenarios. The appendix describes the structure of the simulator used by us, and the simulator provided by CEWiT. All the plots and results described in this thesis are from the simulator designed by us on Matlab.

APPENDIX A

Simulator

A.1 Coding Decoding test Simulator

This simulator is what we used on Matlab to test and verify the algorithms for coding, combining and decoding. For these simulations we assumed that the synchronization and channel estimation are done perfectly, and that we know the channel gain at the receiver. The various steps involved in the flow of the simulation is as follows:

- First generate random QPSK data which needs to be encoded.
- Separate out the I and Q phases and perform matrix parity coding on both individually.
- Combine the I and Q phases of the coded bits and form coded QPSK symbols.
- These symbols are present on the subcarriers in frequency domain, so we now repeat the same data in the other frequency band - for frequency domain repetition of data.
- This completely is the frequency domain structure, which we then take an IFFT and add noise and interference of varying SNRs and SIRs.
- The received signal is then converted to frequency domain by an FFT. We now have the received symbols, on which we implement the algorithms discussed in this thesis.
- Various methods have been tried - Matrix Parity combining, MRC based soft combining, Combination of hard decoding and MRC, ML and Joint ML combining.

A.2 CEWiT Simulator

This simulator is what has been provided by CEWiT. Here all the receiver side architecture has been simulated, with separate files for each process. The main configuration file, simulationConfigFile.m has all the configuration parameters required for the simulation. Each of the receiver blocks have been given separate files for including our algorithms - Synchronization, channel estimation and equalization, coding and decoding. The simulator can be executed by running the main.m file, which then shows the BER at each receiver for the specified number of iterations.

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