TRANSMITTER PRECODING AND CODE-SHARING TECHNIQUES USING BLOCK TRANSMISSION SYSTEM

by

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**Thesis submitted in fulfillment of the requirements
for the degree of
Doctor of Philosophy**
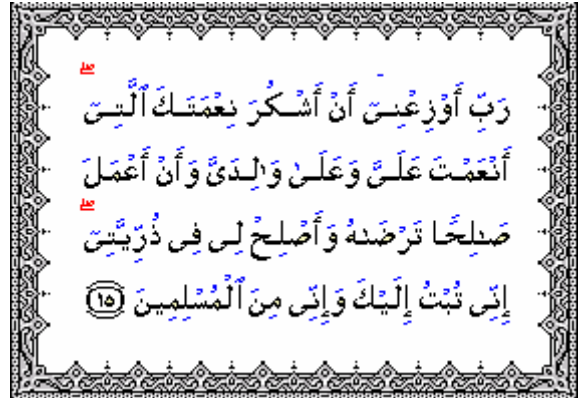
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DEDICATION

To

my parents…

my wife…

my children: Lara, Mumen & Sajed…

for their love, patience and sacrifice during my study

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"O my Lord! grant me that I may be grateful for Thy favour which Thou hast bestowed upon me, and upon both my parents, and that I may work righteousness such as Thou mayest approve; and be gracious to me in my issue. Truly have I turned to Thee and truly do I bow (to Thee) in Islam."

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رَبِّ أَوْزِعْنِى أَنْ أَشْكُرَ نِعْمَتَكَ ٱلَّتِىٓ أَنْعَمْتَ عَلَىَّ وَعَلَىٰ وَٰلِدَىَّ وَأَنْ أَعْمَلَ صَٰلِحًا تَرْضَىٰهُ وَأَصْلِحْ لِى فِى ذُرِّيَّتِىٓ إِنِّى تُبْتُ إِلَيْكَ وَإِنِّى مِنَ ٱلْمُسْلِمِينَ ﴿١٥﴾

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TABLE OF CONTENTS

CHAPTER 6 – CONCLUSION AND FUTURE WORK

LIST OF TABLES

LIST OF FIGURES

LIST OF ABBREVIATIONS

AWGN	Additive White Gaussian Noise
BDFE	Block Decision Feedback Equalizer
BER	Bit Error Rate
BLE	Block Linear Equalizer
CDMA	Code Division Multiple Access
CH	Channel
CSIT	Channel Side Information at the Transmitter
dB	Decibel
DFE	Decision feedback Equalizer
FDMA	Frequency Division Multiple Access
FPGA	Field Programmable Gate Array
GB	Guard Band
IBI	Inter Block Interference
ISI	Inter Symbol Interference
LMS	Least Mean Square
LoS	Line of Site
LTE	Linear Transversal Equalizer
MAI	Multiple Access Interference
MMSE	Minimum Mean Square Error
MSE	Mean Square Error
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
PDF	Power Spectral Density

PEP	Pair-wise Error Probability
PSD	Power Spectral Density
Rxed	Received signal
SBCE	Semi Blind Channel Estimation
SNR	Signal to Noise Ratio
SNR_C	Signal to Noise Ratio in real channel
SNR_{ND}	Signal to Noise Ratio in case of no distortion channel
$SNR_{relative}$	Signal to Noise Ratio relative to no distortion channel
ST	Space-Time
TDMA	Time Division Multiple Access
Txed	Transmitted signal
WiMAX	Worldwide Interoperability for Microwave Access
ZF	Zero Forcing

LIST OF SYMBOLS

σ^2	Variance of the AWGN
ℓ	Normalization factor at the output of the transmitter
η	Variance of the receiver coder.
η_T	Total noise variance at the end of the receiver
ξ_b	Signal energy per bit
B	The vector at the output of the transmitter
C, **D**, **Y**	Channel matrices with different sizes
$\mathbf{C}_i$, $\mathbf{D}_i$, $\mathbf{Y}_i$	One row of the channel matrices **C**, **D**, **Y**
e	The total energy of the vector **B**
ξ_b	Signal energy per block
F	The Precoder matrix
$\mathbf{F}_1$	The transmitter share of the coding process
$\mathbf{F}_2$	The receiver share of the coding process
g	Length of the channel
K	Number of users
k	Number of signal levels
m	Length of the transmitted block
n	The length of the vector at the input of the channel
N_o	AWGN power
p	Sharing factor at the transmitter
q	Sharing factor at the receiver
R	Received vector after the buffering stage

r_i	Element of the received vector after the buffering stage
S	Transmitted data vector
S'	The estimated data vector
s'_i	Element of the estimated data vector
s_i	Element of the transmitted data vector **S**
U	Processed noise vector
u_i	Element of the processed noise vector
V	Received vector at the output of the channel
v_i	Element of the received vector at the output of the channel
W	AWGN vector
$w(t)$	AWGN signal
$y(t)$	Base band channel

TEKNIK PRAPENGEKODAN PENGHANTAR DAN PERKONGSIAN KOD MENGGUNAKAN SISTEM PENGHANTARAN BLOK

ABSTRAK

Dewasa ini, perkhidmatan komunikasi bergerak telah meledak dengan begitu pantas sekali dalam masyarakat kita. Kesemua sistem komunikasi selular yang ada kini menggunakan teknologi digital. Permintaan terhadap perkhidmatan pelbagai jalur lebar seperti capaian Internet berkelajuan tinggi dan penghantaran imej berkualiti tinggi / video, semakin meningkat. Sistem komunikasi bergerak generasi ketiga direka khusus untuk membantu perkhidmatan jalur lebar, dengan kualiti yang sama seperti rangkaian tetap. Sistem komunikasi wayarles dijangka dapat memainkan peranan utama dalam menyediakan capaian mudah alih untuk kegunaan perkhidmatan maklumat masa depan. Permintaan terhadap perkhidmatan baru untuk membantu aplikasi Internet dan video termaju, menjadi cabaran utama dalam bidang teknikal, iaitu capaian multimedia memerlukan hubungan rangkaian lebar jalur - tinggi dan kependaman-rendah kepada ramai pengguna, mobiliti memerlukan adaptasi terhadap keadaan masa saluran yang berubah-ubah dan kemudahalihan, yang boleh menyebabkan kekangan yang teruk terhadap saiz penerima dan penggunaan kuasa.

Batasan fizikal terhadap saluran wayarles mewujudkan cabaran yang besar terhadap komunikasi yang boleh diharapkan. Batasan lebar jalur, kehilangan perambatan, hinggar, gangguan, dan pemudaran pelbagai laluan menyebabkan saluran wayarles seperti suatu saluran paip yang sempit, yang tidak mampu mengendalikan aliran data yang pantas. Justeru, para penyelidik bertegas mereka bentuk suatu sistem yang sesuai beroperasi dalam keadaan yang sedemikian, dalam usaha memperoleh kualiti

perkhidmatan yang tinggi. Begitu juga dengan mobiliti sistem komunikasi yang memerlukan penyelidikan lanjut untuk mengurangkan kekompleksan dan penggunaan kuasa pada penerima. Tesis ini mengemukakan suatu teknik baru bagi memperbaiki prestasi sistem komunkasi yang digunakan untuk penghantaran data digital melalui masa saluran yang berubah-ubah, contohnya saluran bergerak frekuensi tinggi. Semua teknik yang dijelaskan dalam tesis ini dikelaskan sebagai sistem penghantaran blok, iaitu sejenis pensetaraan linear untuk meminimumkan kesan gangguan antara simbol. Data ini dihantarkan dalam bentuk blok daripada simbol data. Blok data yang berjaya dihantar, diasingkan sama ada melalui blok isyarat latihan atau blok tanpa isyarat. Penghantaran isyarat data dalam blok mampu mengurangkan kekompleksan penerima dan juga membantu mengurangkan kesan ralat perambatan. Panjang yang sesuai bagi setiap blok data bergantung pada parameter saluran yang dijangkakan dan kekompleksan yang dibenarkan.

Sistem pertama yang dijelaskan dalam tesis ini ialah sistem prapengekodan. Sistem ini bertujuan mengurangkan kekompleksan penerima melalui pensetaraan yang biasanya dilakukan daripada penerima kepada penghantar. Tiada proses lain diperlukan pada penerima kecuali suatu ujian ambang. Dengan kata lain, teknik prapengekodan ini menghasilkan penguatan 2 dB dalam prestasi sistem, dibandingkan dengan BLE, dengan semua pensetaraan dilakukan pada penerima.

Dalam sistem kedua, strategi perkongsian diaplikasikan dalam sistem prapengekodan untuk meningkatkan prestasi sistem. Melalui pensetaraan yang diagihkan, didapati 75% daripada proses dilakukan pada penghantar, sementara 25% dilakukan dengan lengkap pada penerima. Hal ini menyebabkan sistem lebih tegar terhadap hinggar yang bertambah, dan menghasilkan penguatan 2 dB lebih daripada prakod, dan penguatan 4 dB yang lebih daripada BLE. Disebabkan terdapatnya jalur-adang yang

berlebihan dalam blok yang dihantar, maka ia dikenali sebagai sistem kongsi dengan jalur-adang.

Sistem ketiga merupakan sistem kongsi tanpa jalur-adang. Ia juga bergantung pada idea agihan proses pensetaraan di antara penghantar dan penerima, tetapi diaplikasikan pada BLE. Nisbah kongsi adalah 25% pada penghantar, dan 75% pada penerima. Disebabkan tidak terdapatnya jalur-adang tambahan dalam sistem ini, maka lebar-jalur yang diperlukan adalah kurang, dibandingkan dengan prapengekodan dan sistem kongsi yang lain. Prestasi kadar ralat bit daripada sistem ini adalah sama seperti pada sistem prapengekodan.

TRANSMITTER PRECODING AND CODE-SHARING TECHNIQUES USING BLOCK TRANSMISSION SYSTEM

ABSTRACT

Recently, mobile communications services are penetrating into our society at an explosive growth rate. All of the current cellular communication systems have adopted digital technology. Nowadays, the demands for various wideband services such as high-speed Internet access and video/high-quality image transmission, is increasing. The third-generation mobile communication system has been designed to support wideband services with the same quality as the fixed networks. The wireless communications systems are expected to play more roles in providing portable access to future information services. The demand for new services to support Internet and advanced video applications presents key technical challenges, i.e., multimedia access requires high-bandwidth and low-latency network connections to many users, mobility requires adaptation to time varying channel conditions and portability imposes severe constraints on receiver size and power consumption.

Physical limitations on wireless channels impose huge challenges to reliable communication. Bandwidth limitations, propagation loss, noise and interference make the wireless channel a narrow pipe that does not readily accommodate rapid flow of data. Thus, researches aim to design systems that are suitable to operate in such channels, in order to have high performance quality of service. Also, The mobility of the communication systems require further investigations to reduce the complexity and the power consumption of the receiver.

This thesis presents new techniques that improve the performance of the communication system used for transmission of digital data over time varying channels such as high frequency mobile channels. All the techniques described

through this thesis are classified as block transmission systems, which are type of linear equalization to minimize the effect of intersymbol interference (ISI). The data are transmitted in the form of blocks of data symbols. Successive blocks of data are separated either by blocks of training signals or blocks of no signal. Transmission of data signals in blocks considerably reduces the complexity of the receiver and also helps in reducing error propagation effects. The appropriate length of each data block depends on the channel parameters expected and the allowed complexity.

The first system described in this thesis is the precoding system. This system aims to reduce the complexity of the receiver by moving the equalization which usually done in the receiver to the transmitter. No other processing is needed at the receiver except a threshold testing. On the other hand, this precoding technique results in a 2 dB enhancement in the system performance in comparison with the normal Block Linear Equalizer (BLE) where all the equalization is done at the receiver.

In the second system, a sharing strategy was applied on the precoding system in order to increase the performance of the system. The equalization is divided so that 75% of the process is done at the transmitter, while 25% is completed at the receiver. This makes the system more immune toward additive noise, and results in 2 dB enhancement more than the precoder, and 4 dB more than BLE. Because of the extra guard band in the transmitted block, it is named sharing system with guard band.

The third system is the sharing system without guardband. It is also depends on the idea of dividing the equalization process between the transmitter and the receiver, but it is applied on the BLE. The sharing ratio is 25% at the transmitter, and 75% at the receiver. There is no extra guard bands in this system, so, it needs less bandwidth in comparison with the precoding and the other sharing system. The bit error rate performance of this system is the same as the precoding system.

CHAPTER ONE

INTRODUCTION

1.1 Motivation

In the wireless communication systems, maintaining low complex mobile units is

one of the most important factors that should be taken into considerations [1, 2]. In

the literature, many researchers have investigated techniques that move signal

processing, such as equalization, from the mobile unit to the base station in order to

reduce the mobile complexity [3, 4]. In such techniques, a transmitter-based

interference cancellation is done at the base station and just simple linear processing,

e.g., threshold decision, at the mobile unit. Many researches tried to simplify the

receiver unit, for example, the authors of [4] proposed a precoding technique which

simplifies the receiver. Their system depends on the knowledge of the channel

elements, i.e., the delayed version of the spreading waveform, and the complex

channel fading gain for each user in each path. It depends too on the knowledge of

the channel matrix by using a sophisticated adaptive channel estimation method

given by [5]. Also, the original information can be retrieved at the mobile unit using

a matched filter. The authors in [6, 7] suggested techniques at the transmitter for

synchronous Code Division Multiple Access (CDMA) over Additive White Gaussian

Noise (AWGN) channel. In their design, they used a RAKE receiver.   The

disadvantage of RAKE reception is the sensitivity to channel mismatch, its

performance is generally inferior to MMSE or decorrelator based multiuser interference rejection [4].

In the band limited Inter Symbol Interference (ISI) transmission system, when the delay spread is less than the symbol interval, ISI can be neglected because the delayed pulse will not affect the pervious or the next pulse from the other paths. When the delay spread is large, intersymbol interference can be removed by inserting guard intervals between symbols to ensure that the delayed version of the pulse will not affect the other pulses from other paths [8].

As mentioned earlier, receiver simplicity is a very important issue in the design of the communication system, but, on the other hand, the demand for higher performance systems with perfect service, pushes toward designing systems with very low bit error rate. Users will not accept simple applications if the QoS is not high. So, the system bit error rate level is very important for the designer too. Sometimes, designers may accept some complexity in the receiver in order to get better QoS.

Another important issue is the bandwidth limitation. Frequency spectrum is limited, and the designer has to take the bandwidth needed for the system in addition to the QoS and simplicity.

So, in the design procedure, one must find the optimum model that takes all the mentioned above conditions into considerations.

1.2 Objectives

- To implement a precoding technique that moves the equalization process from the receiver to the transmitter in order to reduce the complexity of the receiver.

- To implement a code-sharing technique that divides the equalization process between the transmitter and the receiver in order to improve the performance.

- To apply the code-sharing technique on the block linear equalizer in order to increase the bandwidth efficiency by using less guard band in the transmitted vector.

1.3 Research methodology

In this research, a block diagram is proposed from each system that clarifies all the stages of the transmitter, channel and the receiver. After that, mathematical representations of the proposed systems are analyzed. These mathematical models will be represented by equations that depend on the active variables that may affect the systems, which can be analyzed using computer programs (MATLAB$^{®}$). The performance of the systems, will be tested, specially the signal to noise ratio, the bit error rate and the immunity against inter-symbol interference.

Depending on the results of the mathematical representations, simulation models will be done on MATLAB$^{®}$ (command line and Simulink$^{®}$), and the results will be compared with those obtained from the mathematical representations.

The parameters effect for each system will be studied to determine the limitation and the boundaries for the design.

The data obtained from the stages for each system will be compared to ensure that the systems have been tested by two methods. Also, the performance of the three systems will be compared to see which one is the best from each point-of-view (bit error rate and bandwidth), with a comparison with another system from the literature.

1.4 Original contributions

In this work, some coding strategies at the transmitter and the receiver were developed for the downlink of the communication system in band-limited ISI channels.

First Contribution:

In the first system, the coding is such that no processing of the received signal is needed except testing these against appropriate threshold levels. This will make the receiver quite simple at the expense of the common transmitter. In situations where a single transmitter is feeding many receivers, it will be an advantage to complicate the base station, and simplify the receiver units . The performance of the system with precoding is expected to be competitive to those systems where all the processing is carried out at the receiver.

Second Contribution:

In the second system, sharing strategy was developed between the transmitter and the receiver for the communication system in band-limited ISI channels. The

equalization process will be divided between the transmitter and the receiver in optimum values. It is expected that this method will give some enhancement in comparison with the precoding system, where all the equalization process is done at the transmitter and leaves the receiver quite simple. In applications where the transmitted signal faces a badly scattering channel, the performance of the communication system is seriously affected. So, in order to increase the performance, extra processing in the receiver is accepted.

Third Contribution:

The third system is also a sharing strategy, but instead of dividing the equalization process of the precoder (as done in the second contribution), the sharing strategy this time will be applied on the Block Linear Equalizer (BLE). It is expected to give the same performance of the precoding system, but with less bandwidth to be consumed during transmission.

For all the mentioned above contribution, the parameters effects are tested for all the effective factor of the system, to find the boundary limits, and the suitable operating conditions for each system.

In all systems, the transmitter is assumed to have prior knowledge of the channel impulse response. This will not be discussed in details, and it will not be a part of this thesis. There are many techniques available for channel estimation and these are available in the published literature.

1.5 Thesis outline

The rest of this thesis is organized as follows:

In Chapter Two, literature review will be introduced to spot the light on the systems used in this thesis, like channel estimation, channel equalization, block transmission systems with a brief description about their types.

In Chapter Three, precoding systems will be discussed in details with practical examples taken from recent systems from the literature

The contribution of this thesis starts in Chapter Four, which introduces the mathematical representation of the three proposed models. The systems models will be explained in details, with studies for the performance, and analysis of the coders in each system.

Chapter Five includes further studies on the proposed systems, where the three systems were studied using simulation program. The parameters effect on each system is presented too, with a comparison between the three systems.

In Chapter Six, conclusions and limitations of this study were included, with some ideas for future work in this field.

CHAPTER TWO

CHANNEL ESTIMATION AND EQUALIZATION

2.1 Introduction

The function of channel estimation is to make an estimate of the changes in amplitude and phase shift caused by the wireless channel, and provide these information to the equalizer which reverse the effect of the wireless channel before the demodulation process.

Wireless channels, in general, are not easy to be estimated because they are frequency selective, time selective, and in some applications, space selective too. Channel characteristics may be simplified, as in Jakes' model [9], to a multipath model with a certain angle for each path, with a complex amplitude and a propagation. Then, some statistical assumptions may be used to develop these parameters [10].

It is possible to use a general channel model, that gives a large class of observed channels only for modeling purposes. But for channel estimation, a complex model with many variables may be determined to be less accurate when the parameters have to be estimated with a finite set of observations [10].

2.2 Channel estimation

Methods of channel estimation may be classified depending on the used symbols in the estimation process, i.e., pilot and data symbols [11]. If only the pilot symbols are used, it is called training pulse based detection. If only the data symbols are used in the estimation process, the estimator can't depend on previously known symbols, so, this channel estimation is considered as blind estimation. There is another method where both the pilot and the data symbols are used in the estimation, and this is called semiblind estimation. All the three methods will be discussed in this section with some practical examples.

2.2.1 Training based estimation

Channels are estimated in some practical systems by using periodic pulses of known training symbols [12]. It is the easiest and most convenient method of estimation. This training-based method uses already known pulses to estimate the characteristics of the mobile channel [13]. Those pulses usually are period of silence, i.e. zeros, or may be a sequence of ones depending on the design of the system, and to be sent after each block of transmitted data as shown in Figure 2.1. For example, in Orthogonal Frequency Division Multiplexing (OFDM) systems, they are called cyclic prefix and are one of the important factors in designing the system. Also, in block transmission systems (like the systems proposed in this thesis), a group of zero pulses are sent after each block of coded data to synchronize the system, and is used later as a guardband.

	data	training	data	
• • •				• • •
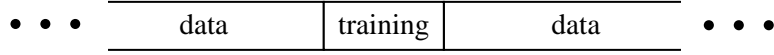
Figure 2.1: Block of transmitted data with training pulses.

The training pulses will convolute with the unknown channel impulse response caused by the surrounding, resulting in a distorted signal at the receiver end. Because of the receiver's knowledge of original transmitted pulses, it can easily estimate the effect of the channel. In a transceiver wireless system (where each terminal of the system acts as a transmitter and a receiver), if the channel characteristics are needed by one of the wireless system terminals, it may use the training pulse transmitted from the other terminal (back channel) to get knowledge about the channel impulse response.

The disadvantage of this method is that the performance of the channel estimator is limited by the power of the pilot layer. Even if the data interference cancellation is perfect, the modified received data is still noisy [14].
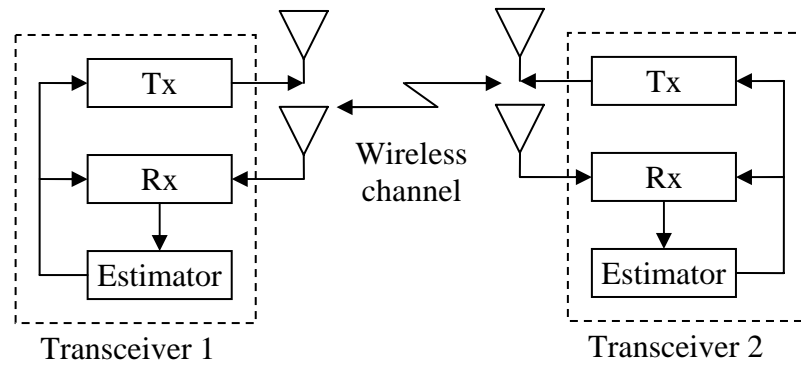
Figure 2.2: Wireless transceiver with channel estimator.

A practical example of this estimation technique is the method applied in the Altera® Worldwide Interoperability for Microwave Access (WiMAX) system [15].

Referring to the manual of this system, the data and the pilot subcarriers are transmitted through the channel, and the pilot subcarrier is known. So, the receiver can guess the changes due the channel. The subchannel is made up of six groups with four adjacent subcarriers in each group. These groups are modulated with a mix of data and pilots over three Orthogonal Frequency Division Multiple Access (OFDMA) symbols. The subcarriers for a particular subchannel are the same over the three OFDMA symbols, and in this way it is possible to estimate the frequency response of the channel. In the first and third OFDMA symbol, the outer carriers of each tile are the pilot subcarriers, and so it is possible to estimate the channel response at these frequencies by comparing the pilot subcarriers with the known reference one. The frequency response of the two inner subcarriers may be estimated by linear interpolation in the frequency domain [15].

2.2.2 Semiblind channel estimation

In systems that depend on previous knowledge of the channel characteristics, an important issue for achieving good performance is to get reliable channel information. Due to the complexity of the channel in environments with crowded surroundings, i.e., cities, applying training based method makes it necessary to use large amount of training symbols in order to estimate the frequently updated channel coefficients, which will reduce the system bandwidth efficiency.

Therefore, some effective channel estimators have been introduced to predict channel information using the data transmitted symbols, and just a few training symbols [16-

18]. This method is called semiblind channel estimation (SBCE). It uses not only the training pluses, but also the data symbols to perform channel estimation [19]. Due to its amazing performance, SBCE has became more popular. Many researchers have used it in their systems and presented many simulation models as in [20-23], and mathematical studies as in [11, 19, 24-26].

Since the data symbols are unknown at the receiver, data detection becomes important. The procedure of SBCE can be done as follows: [19]

4.1 Initial training-based channel estimation;

4.2 Given channel knowledge, detect data symbols;

4.3 Given data knowledge, perform channel estimation by taking the whole block as virtual training;

4.4 Repeat the last two steps until having a stopping order.

Due to it flexibility in estimating the channel impulse response, this method is widely used in many application in the communication systems, specially if there is a guard band or cyclic prefix in the transmitted vector, such as OFDM. Authors in [16] applied the semi blind estimation method to a Space Time OFDM with block transmission. Authors is [18] extended the work to be applied on MIMO OFDM, with interesting results.

Another application that semi blind estimators are seen clearly is the DS-CDMA systems. Authors in [17] studied the performance of SB estimators on DS-CDMA with long codes.

2.2.3 Blind channel estimation

Blind sequence estimation is defined as a technique to estimate data sequence from the received signal without knowing the transmission channel, or knowing the impulse response of the channel without knowing the transmitted bit [27].

Because of its major effect in the wireless communication system, there are many research activities in the literature that introduce blind estimation techniques [4, 5, 28-34].

Also, blind channel estimation has been used recently in may wireless applications, such as MIMO-OFDM with orthogonal codes as in [27, 35]. In these two researches, authors used blind estimation because their designs did not use cyclic prefix to reduce the consumed bandwidth. Blind estimation seems to be the perfect solution for systems without cyclic prefix.

Authors in [36] tested the blind estimation method on fast time varying channels. This kind of channel is considered to be the worst case that may face the transmission signal [36] because of the rapid changing in the channel impule response, which makes them very difficult to be estimated. The results in their research were very promising.

Also, authors in [37] proposed a novel algorithm to blind estimate the channel impulse response in DS-CDMA system, assuming that the noise correlation is unknown. This algorithm was able to deal with arbitrary symbol constellations, without using auxiliary antennas in the transmitter and receiver [37].

One of the practical blind estimation techniques is presented in [4]. The system presented in that paper is a precoding technique, which is similar to the precoding system proposed in this thesis. In that system, the transmitted signal is precoded at

the transmitter, i.e. the base station. It has been assumed that the channel coefficients are perfectly known at the transmitter side, which is essential for designing the precoder. The transmitter must update its knowledge about the channel for each block due to the mobility of the receiver.

The uplink signal transmitted from the mobile unit to the base station during the ith symbol interval can be written as [5]:

$$r[i] = \sum_{k=1}^{K} b_k[i] \sum_{l=1}^{L} s_{l,k} f_{l,k} + n[i] \qquad (2.1)$$

where:

- L is the number of resolvable paths

- $b_k[i]$ is the ith symbol for the kth user

- $s_{l,k}$ is the delayed version of the spreading waveform (with zero-adding when a guard interval is inserted)

- $f_{l,k}$ is the complex channel fading gain corresponding to the lth path of the kth user

- $n[i] \sim \mathrm{N}\left(0, \sigma^2 I_N\right)$ is a complex white Gaussian noise vector.

It is easier to deal with Equation 2.1 in vector form, so, the effect of the channel can be viewed in vectors by using the assumption [4]:

$$S_k \overset{\Delta}{=} \begin{bmatrix} s_{1,k} & s_{2,k} & \cdots & s_{L,k} \end{bmatrix} \qquad (2.2)$$

and

$$\boldsymbol{f}_k \overset{\Delta}{=} \begin{bmatrix} f_{1,k} & f_{2,k} & \cdots & f_{L,k} \end{bmatrix} \qquad (2.3)$$

Then Equation 2.1 can be written as

$$r[i] = \sum_{k=1}^{K} b_k[i] S_k \boldsymbol{f}_k + n[i] \qquad (2.4)$$

It is clear that S_k and f_k give information about the channel, so, they can be replaced by the symbol h_k, which is the channel for each user.

Again, the other summation (the users) can be removed by writing the elements in vector forms assuming that:

$$H = \begin{bmatrix} h_1 & h_2 & \cdots & h_K \end{bmatrix} \tag{2.5}$$

and

$$b[i] = \begin{bmatrix} b_1[i] & b_2[i] & \cdots & b_K[i] \end{bmatrix} \tag{2.6}$$

so, Equation 2.4 can be written again as [4]:

$$r[i] = Hb[i] + n[i] \tag{2.7}$$

A block diagram of the uplink system appears in Figure 2.3.
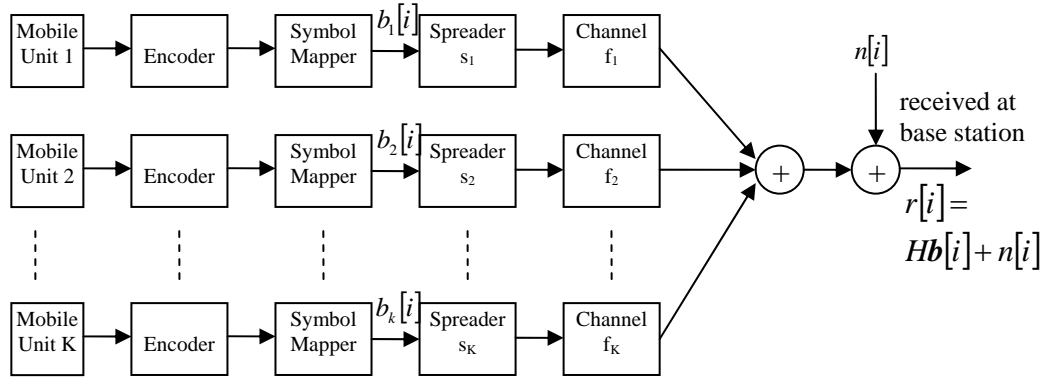
Figure 2.3: The uplink of a *K*-user CDMA system [4].

Let the autocorrelation matrix of the received signal $r[i]$ be [5]:

$$C_r = E\{r[i]r[i]^H\} \tag{2.8}$$

$$= E\{(Hb[i] + n[i])(Hb[i] + n[i])^H\} \tag{2.9}$$

$$= E\{Hb[i]H^H b^H[i]\} + E\{H^H b^H[i]n[i]\} + E\{Hb[i]n^H[i]\} + E\{n[i]n^H[i]\} \tag{2.10}$$

14

$$= E\{HH^H \boldsymbol{b}[i]\boldsymbol{b}^H[i]\} + E\{H^H \boldsymbol{b}^H[i]\}E\{n[i]\} + E\{H\boldsymbol{b}[i]\}E\{n^H[i]\} + E\{n[i]n^H[i]\} \quad (2.11)$$

$(.)^H$ denotes the Hermitian transpose of the matrix.

Assuming that each user's information symbols are independently identically distributed (i.i.d.), and the symbol streams of different users are independent, therefore [5]:

$$E\{\boldsymbol{b}[i]\boldsymbol{b}^H[i]\} = I \quad (2.12)$$

and knowing that the noise AWGN with zero mean, i.e. $E\{n[i]\} = 0$, Equation 2.11 reduced to [5]:

$$C_r = E\{HH^H\} + E\{n[i]n^H[i]\} \quad (2.13)$$

$$= HH^H + \sigma^2 I_N \quad (2.14)$$

Writing Equation 2.14 again in the eigen decomposition form [5]:

$$C_r = U_s \Lambda_s U_s^H + \sigma^2 U_n U_n^H \quad (2.15)$$

where:

- $U_s = [u_1 \quad u_2 \quad \cdots \quad u_K]$ contains the orthonormal eigenvectors of C_r

- $\Lambda_s = diag(\lambda_1 \quad \lambda_2 \quad \cdots \quad \lambda_K)$ contains the K largest eigenvalues of C_r

- $U_n = [u_{K+1} \quad u_{K+2} \quad \cdots \quad u_N]$ contains the orthonormal eigenvectors that correspond to the smallest eigenvalue, σ^2.

Now, for simplicity, one of the users will be studied, i.e. user 1. Authors in [5, 28, 31, 38] proved that:

$$U_n^H h_1 = U_n^H S_1 \boldsymbol{f}_1 = 0 \quad (2.16)$$

so, $\boldsymbol{f}_1$ can be estimated at the base station as [4]:

$$\hat{\boldsymbol{f}}_1 = \arg \min_{\|\mathbf{f}\|=1} \|U_n^H S_1 \boldsymbol{f}\|^2 \quad (2.17)$$

$$= \arg \min_{\|\boldsymbol{f}\|=1} \underbrace{\left(S_1^H U_n U_n^H S_1 \right)}_{Q} \boldsymbol{f} \qquad (2.18)$$

$$= \text{minimum eigenvector of } Q \qquad (2.19)$$

Equation 2.19 can be implemented blindly in a batch or sequential adaptive manner because it specifies $\boldsymbol{f}_1$ up to a scale and phase ambiguity [4].

In batch mode, noise subspace parameters in Equation 2.18 can be replaced with parameters obtained from the eigen decomposition of the sample autocorrelation matrix of the received signal [4]. Also, in sequential adaptive mode, the channel estimated coefficients are updated frequently for each time period, using a tracking algorithm and sequential Kalman Filtering technique described in [4]. The channel estimate should be unique so that H have rank K which is possible if $K \leq N$.

2.3 Channel equalization

In order to track the channel characteristics during the transmission process, adaptive equalization algorithms are applied to adjust the equalizer coefficients to minimize the Mean Square Error (MSE) [39]. Least Mean-Square (LMS) is one of those algorithms. In the receiver, the equalizer, which will reverse the effect of the channel distortion, is used as a correction process before the normal detection process in order to reduce the ISI, and therefore, enhance the performance of the system [40].

2.3.1 Linear equalizers
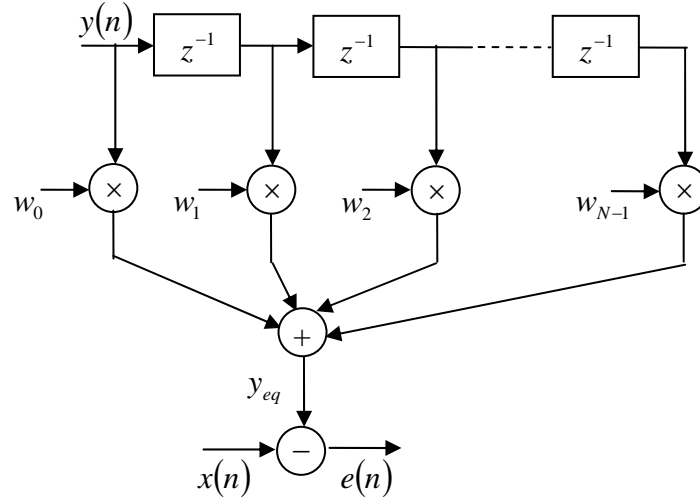
Figure 2.4: Linear Transversal Equalizer (LTE)

Linear Equalizers (LE)

The linear transversal equalizer (LTE) is the most widely used and simplest equalizer among channel equalizers. The block diagram of this equalizer is shown in Figure 2.4. At the output of the equalizer, there are N taps (LTE is also named tapped-delay-line equalizer) at the nth time slot, and is given by [41]:

$$y_{eq}(n) = \sum_{k=0}^{N-1} w(k) y(n-k) \tag{2.20}$$

where $w(k)$ are the equalizer coefficients, and $y(n-k)$ is the input to the equalizer after being delayed k times.

The error is the difference between transmitted signal and the output of the equalizer, that is:

$$e(n) = x(n) - y_{eq}(n) \tag{2.21}$$

This error signal is used to update the tap coefficients of the equalizer periodically in order to track the time varying of the channel response. The distorted data is input to the delay line of the equalizer. The delayed data is then convolved with the equalizer coefficients to get the estimate of the transmitted symbol. The detector quantizes the estimate to the nearest symbol in the signal space. Depending upon the mode of operation, the error signal is generated by subtracting the estimated symbol from either the locally generated true symbol for training mode or from the detector's output for the decision directed mode. This error signal with suitable weighting is then used to update the coefficients of the equalizer.

Linear equalization is effective in practical applications with linear distortions [8]. If there are nulls in the channel amplitude characteristics, the equalizer amplitude characteristics becomes very large near those nulls, it will increase the additive noise. In this case linear transversal equalizer is incapable of performing well.

There are many adaptive algorithms applied in the LTE as Least Mean Square (LMS), Discrete Cosine Transform (DCT)-LMS, and Recursive Least Square (RLS) as presented in [42].

2.3.2 Nonlinear equalizers

Decision Feedback Equalizer (DFE) is a nonlinear equalizer that is widely used in situations where ISI is large. It has been shown that DFE performs significantly better than a linear equalizer of equivalent complexity [43].

The DFE is used to estimate the error and cancel the Intersymbol Interference (ISI) from the symbols that have already been detected. The block diagram of DFE is shown in Figure 2.5. It consists of a feedback and a feedforward linear transversal

filters [44]. The output of both filters is the equalized signal. The decision made on this signal is fed back via the feedback filter in order to cancel ISI caused by previously detected symbols [8, 44].
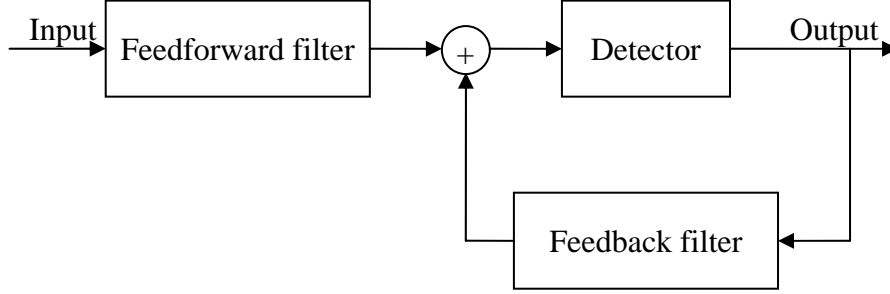
Figure 2.5: Decision feedback equalizer

The basic idea of the DFE is that if the values of the symbols already detected are assumed to be correct, then the ISI contributed by these symbols can be canceled by subtracting past symbol values with appropriate weighting from the equalizer output [8]. The feedback filter reshapes the received signal so that the ISI at the output of the feedforward filter is causal, and the current pulse height is as high as possible with respect to the residual trailing ISI, which is then subtracted in the feedback part without noise enhancement.

When an incorrect fed back decision is taken, its effect will be seen in the following few equalizing steps, which will cause multiple errors following the first one. This is called the propagation error, and it is considered as a major disadvantage to the system [44].

DFE based algorithms are widely discussed in the literature review. Authors in [45] used an algorithm that uses DFE in MIMO-OFDM systems. This algorithm results in accurate equalization and simple detection.

2.3.3 Zero-forcing equalizer

The zero-forcing (ZF) equalizer is used to eliminate the intersymbol interference and the Multiple Access Interference (MAI) [46]. It multiplies the distorted received signal by the inverse of the channel to retrieve the original transmitted signal. Suppose that the system model is a linear filter model with impulse response $\{f_n\}$, followed by an equalizer having an impulse response $\{c_n\}$. This can be represented in one variable q_n which is the convolution between the discrete linear filter and the equalizer, and given by [8]:

$$q_n = \sum_{j=-\infty}^{\infty} c_j f_{n-j} \tag{2.22}$$

It is more convenient to assume that the equalizer have limited number of taps, so the output at the kth sampling interval is:

$$\hat{I}_k = q_o I_k + \sum_{n \neq k} I_n q_{k-n} + \sum_j c_j \eta_{k-j} \tag{2.23}$$

The first term is the transmitted data scaled by q_o (the desired term), while the second term is the ISI effect. The third term is the modulated noise.

In order to eliminate the effect of ISI, the taps should be designed to have a weight [8]:

$$q_n = \sum_{j=-\infty}^{\infty} c_j f_{n-j} = \begin{cases} 1 & n = 0 \\ 0 & n \neq 0 \end{cases} \tag{2.24}$$

In z transform,

$$Q(z) = C(z)F(z) = 1 \tag{2.25}$$

or,

$$C(z) = \frac{1}{F(z)} \tag{2.26}$$

Which is the inverse filter of the linear filter model as mentioned above.

In reality, zero-forcing equalization does not work in most applications, for the following reasons:

- Even though the channel impulse response has finite length, the impulse response of the equalizer needs to be very long (theoretically infinitely long).

- The channel may have zeros in its frequency response that cannot be inverted.

2.4 Block transmission systems

In block transmission systems, the data symbols are grouped in the form of blocks of certain length separated by blocks of known symbols [47]. The known receiver for this kind of systems is the Non-linear Data Directed Estimator (NDDE) introduced in [48, 49], and will be explained in this section.

Block transmission systems are based on the assumption that the channel should be constant within the block, which means that the block duration must be sufficiently short in comparison with the channel profile [47].

Block equalizers are types of the block transmission systems as they adopt the transmission in blocks. They are detectors that represent an extension of conventional equalization techniques discussed in the previous subsections [47]. The Block Linear Equalizer (BLE) and Block Decision Feedback Equalizer (BDFE) have been designed and used in many applications such as HF Modems [47, 50, 51].

These equalization techniques are also applied on block transmission system as in [52], where the estimated channel impulse response is used to compute the coefficients of a Minimum-Mean-Squared-Error Decision Feedback Equalizer (MMSE-DFE).

In order to compare the block equalizers with conventional equalization techniques, Kaleh [47], has derived a block linear equalizer and a block decision-feedback equalizer in both zero forcing (ZF) and minimum-mean-squared-error (MMSE). These equalizers use transformations that depend on channel coefficients without taking noise level into account. Also, the performance of each equalizer is evaluated in the same reference with a comparison in performance with the conventional equalizers.

In this section, after discussing the NDDE, a brief discussion for the BDFE will be introduced. Then a detailed discussion for the BLE will be given because BLE is the base stone for all the proposed systems in this thesis.

2.4.1 Non-linear data directed estimator

Non-linear Direct Data Estimation (NDDE) algorithm, which is a Levinson implementation of the recursive least squares algorithm, is given in [49]. Assuming that M is the data block length and L is the largest spread, the channel correlation elements T_j can be written as:

$$T_j = \sum_{P=0}^{L-j} \hat{g}_{P+j} \hat{g}_P, \quad j = 0, \ldots, M \tag{2.27}$$

where $\hat{g}_P$ are the channel gain coefficients, which are known or may be estimated in the channel estimation stage. Then, the cross correlation coefficients q_k between the $\hat{g}_P$ and the received symbols $Z(k)$ are found, which are expected to be the received data $Y(k)$, corrected by the known intersymbol interference caused by the adjacent training block.

$$q_{M-j+1} = \sum_{P=0}^{L-j} \hat{g}_p^* Z(P+j), \quad j = 0,\ldots,M \tag{2.28}$$

The transmitted data block $\hat{B} = \left[\hat{b}_{M-1},\ldots,\hat{b}_0\right]$ may be retrieved by recursively solving for the least squares solution of [49]:

$$T_M . \hat{B} = Q \tag{2.29}$$

where T_M is the Toeplitz matrix of the channel correlation coefficients and Q is the vector of the cross correlation coefficients. In the following equation, $\hat{b}(m,k)$ and $\hat{\phi}(m,k)$ are two-dimensional variables, where $m = 2,\ldots,M$ denotes the recursion number and k indicates the location of the elements in the vector notation. When the recursion ends, the estimated symbols $\hat{B} = \left[\hat{b}_{M-1},\ldots,\hat{b}_0\right]$ while ϕ is a local variable in the recursion process [49]:

$$\phi(m-2,m-2) = \frac{T_{M-1} - [T_1,\ldots,T_{m-2}] \cdot \begin{bmatrix} \phi(m-3,m-3) \\ \vdots \\ \phi(m-3,0) \end{bmatrix}}{T_0 - [T_1,\ldots,T_{m-2}] \cdot \begin{bmatrix} \phi^*(m-3,0) \\ \vdots \\ \phi^*(m-3,m-3) \end{bmatrix}} \tag{2.30}$$

$$\phi(m-2,k) = \phi(m-3,k) - \phi(m-2,m-2).\phi(m-3,m-3-k), \quad k = 0,\ldots,m-3 \tag{2.31}$$

$$\hat{b}(m-1,m-1) = \frac{q_m - [T_1,\ldots,T_{m-1}] \cdot \begin{bmatrix} \phi(m-2,m-2) \\ \vdots \\ \phi(m-2,0) \end{bmatrix}}{T_0 - [T_1,\ldots,T_{m-1}] \cdot \begin{bmatrix} \phi^*(m-2,0) \\ \vdots \\ \phi^*(m-2,m-2) \end{bmatrix}} \tag{2.32}$$

$$\hat{b}(m-2,k) = \hat{b}(m-2,k) - \hat{b}(m-1,m-1).\phi(m-2,m-2-k), \quad k = 0,\ldots,m-2 \tag{2.33}$$

Since the channel response vector contains nulls, the channel correlation matrix T_M will contain nulls too, that saves computational time, and makes the process faster.

The NDDE algorithm performs iterative detection of data by applying repeatedly the DDE algorithm with dimension M, $M-2$, $M-4$, ..., 2. The dimension is decreased by subtracting the two most reliable extreme symbols that are detected in each iteration [49].

2.4.2 Block Decision Feedback Equalizer

Block Decision Feedback Equalizer (BDFE) is an extended equalization technique to the block transmission system. Whereas conventional equalizer use transversal filter, this one involve the use of matrix transformation. This transformation can be implemented exactly, while the conventional one is approximated by simple filter implementation. Also, the channel impulse response may be easily computed using either Levinson or Schur algorithm [53].

The performance of this system (ZF-BDFE or MMSE-BDFE) is better than the conventional DFE and NDDE, and its complexity is lower too [47, 54, 55].
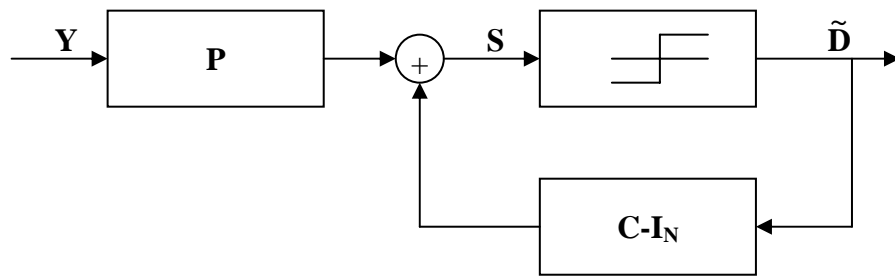
Figure 2.6: Block diagram of BDFE [54].

Both the MMSE-BDFE's and the ZF-BDFE's are composed of a threshold detector, a feedforward transformation, and a feedback transformation, as shown in Figure 2.6.