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978-0-521-86486-2 - Adaptive and Iterative Signal Processing in Communications
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Excerpt
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1 Introduction

Signal processing and communications are closely related; indeed, various signal processing techniques are adopted for communications at both transmitters and receivers. In particular, the role of signal processing is very important in receiver design. For example, signal processing techniques are applied to carrier and clock synchronization, channel equalization, channel estimation, interference rejection, etc. It is our aim in this book to introduce adaptive and iterative signal processing techniques for receiver design in interference-limited environments.

1.1 Communications in interference-limited environments

Generally, the performance of communication systems is limited by the interference, of which there are various sources. For example, multipaths of a radio channel cause intersymbol interference (ISI), as shown in Fig. 1.1. The received signal becomes a sum of delayed transmitted signals with different attenuation factors. Although the background noise is negligible, the ISI can degrade the performance because the received signal is distorted by the ISI.

In a multiuser system, such as the one shown in Fig. 1.2, the other users' signals become interfering signals. Thus, it is important to alleviate interfering signals to achieve a satisfactory performance.

Throughout this book, we introduce a few communication systems under interference-limited environments. For each communication system, adaptive and/or iterative signal processing methods are discussed to overcome the interference.

We will briefly review some approaches for the interference mitigation in each interference-limited channel below.

1.1.1 *ISI channels*

A nonideal dispersive communication channel can introduce the ISI, and the receiver has to mitigate the ISI to achieve a satisfactory performance. Since a dispersive channel is seen as a linear filter, a linear filtering approach can be used to equalize a dispersive channel. The resulting approach is called the linear equalization. A block diagram for a communication system with a linear equalizer is shown in Fig. 1.3, in which $H(z)$ and $G(z)$ denote the $\mathcal{Z}$-transforms of an ISI channel and a linear equalizer, respectively. The transmitted signal,

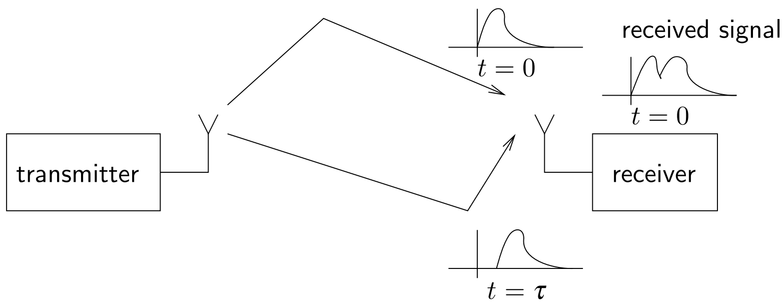

Figure 1.1. Radio channel with multipaths. In this illustration, the propagation delay of the upper path is zero, while the propagation delay of the lower path is $\tau > 0$. The received signal is a superposition of the multipath signals.

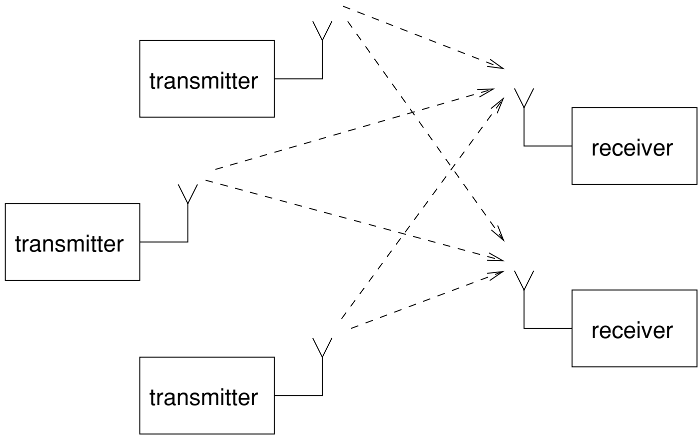

Figure 1.2. Multiuser system.

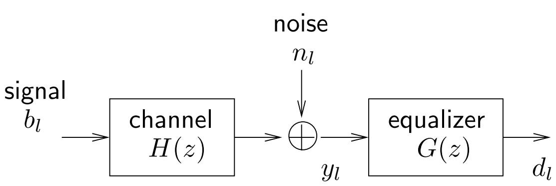

Figure 1.3. Dispersive channel and linear equalizer.

noise, channel output, and equalizer output at discrete time l are denoted by b_l, n_l, y_l, and d_l, respectively. The role of the linear equalizer is to equalize the frequency response of the channel. The zero-forcing (ZF) linear equalizer has the inverse frequency response of the channel so that the overall frequency response becomes flat and there is no ISI. That is, if $G(z) = 1/H(z)$ and the noise is negligible, we can expect that $d_l = b_l$. However, this intuitive approach often fails. If the frequency response of an ISI channel is small at some frequencies, the inverse of the frequency response, $1/H(z)$, can be large at those frequencies. The noise components corresponding to those frequencies can be enhanced, resulting in a low signal-to-noise ratio (SNR) after the equalization. To avoid this problem, other approaches can be considered.

The minimum mean square error (MMSE) filtering is well established in statistical signal processing. A linear filter is designed to minimize the difference between a desired signal and the output of the filter so that the output of the filter can be used as an estimate of the desired signal. The desired signal is a random signal and the input signal to the filter is generally correlated with the desired signal. Generally, the mean square error (MSE) is chosen to optimize the filter since the MMSE criterion is mathematically tractable.

The MMSE equalization is based on the MMSE filtering. In the MMSE equalizer, the error consists of the ISI as well as the noise. Since the MSE is to be minimized, the noise is not enhanced. The MMSE equalizer generally outperforms the ZF equalizer.

Extension of the linear equalization is possible by incorporating the cancelation of (detected) past symbols. This results in the decision feedback equalization. The main difference between this and the linear equalizer is that the decision feedback equalizer (DFE) is nonlinear. However, under certain assumptions, the DFE can be considered as a linear equalizer. The DFE consists of a feedback filter and a feedforward filter. The role of the feedforward filter is to combine dispersed signals, while the role of the feedback filter is to suppress ISI components. Even though this approach looks simple, it is very insightful, and the idea is applicable to other systems suffering from interfering signals.

A different approach, based on the sequence detection, is available that allows effective detection of signals over an ISI channel. The data sequence that maximizes the likelihood function is chosen using the maximum likelihood (ML) criterion. This approach is called ML sequence detection (MLSD) and the solution sequence is called the ML sequence. An exhaustive search, in which the likelihoods for all the possible data sequences are obtained, is carried out to find the ML sequence. Clearly, this approach becomes prohibitive because the number of candidate sequences grows exponentially with the length of the data sequences. For example, there are 2^L candidate data sequences for a binary sequence of length L.

Fortunately, there are computationally efficient algorithms for MLSD. The Viterbi algorithm (VA) is an example. The VA can find the ML sequence, and its complexity grows linearly with the length of data sequence.

Both sequence detection and channel equalization require channel estimation; this is crucial because the channel estimation error can degrade the performance. The channel estimation and data detection are closely coupled. Usually, joint channel estimation and data detection approaches can outperform decoupled approaches (in which channel estimation and data detection are carried out separately).

1.1.2 CDMA channels

For multiuser communications, there are various multiple access schemes. A multiple access scheme allows multiple users to share a common communication channel, for example the same frequency band. In frequency division multiple access (FDMA), a frequency band is divided into multiple subbands so that each user can use one of the subbands without the interference from the other users in the same system. In time division multiple access (TDMA), multiple users share the same frequency band. However, they use different time

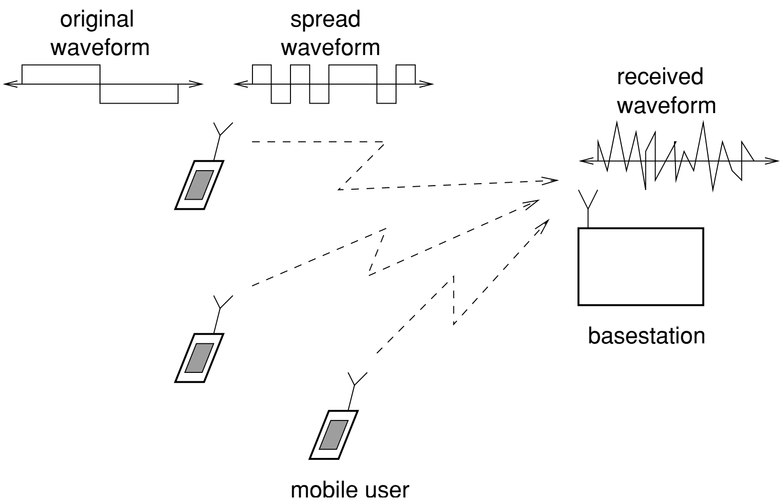

Figure 1.4. Uplink channels with spread signals.

slots to avoid multiple access interference (MAI). Code division multiple access (CDMA) is another multiple access scheme. In CDMA, users can transmit signals simultaneously over the same frequency band during the same time interval. However, to distinguish each user from the others, users' signals are spread by different spreading codes. Figure 1.4 shows the uplink channels[†] of CDMA systems, where the mobile users transmit spread signals simultaneously. CDMA is chosen for some cellular systems as it has a number of advantages over the others.

CDMA becomes interference-limited if spreading codes are not orthogonal. Since there are multiple users in a CDMA system, the signals from the other users become the interfering signals or the MAI. As the MAI limits the performance, it is desirable to mitigate the MAI to achieve a good performance. Linear multiuser detection and interference cancelation are introduced to deal effectively with the MAI. The main idea behind linear multiuser detection is similar to that behind linear equalization; that is, a linear transform is used to recover users' signals. There is also a strong relationship between interference cancelation and decision feedback equalization. If signals are detected, they are canceled to allow better detection of the as yet undetected signals.

Both ZF and MMSE criteria can be employed to design linear multiuser detectors. In addition to the linear multiuser detector and the interference canceler, the ML multiuser detector is also available. The ML multiuser detector is analogous to the MLSD. Owing to the ML criterion, all the users' signals are simultaneously detected. However, a straightforward implementation requires an exhaustive search.

1.1.3 *MIMO channels*

To increase the channel capacity of wireless communications, multiple antennas were introduced. Since there are multiple transmit and multiple receive antennas, the channel is called

[†] The uplink channel is the name given to a channel from a user to a basestation.

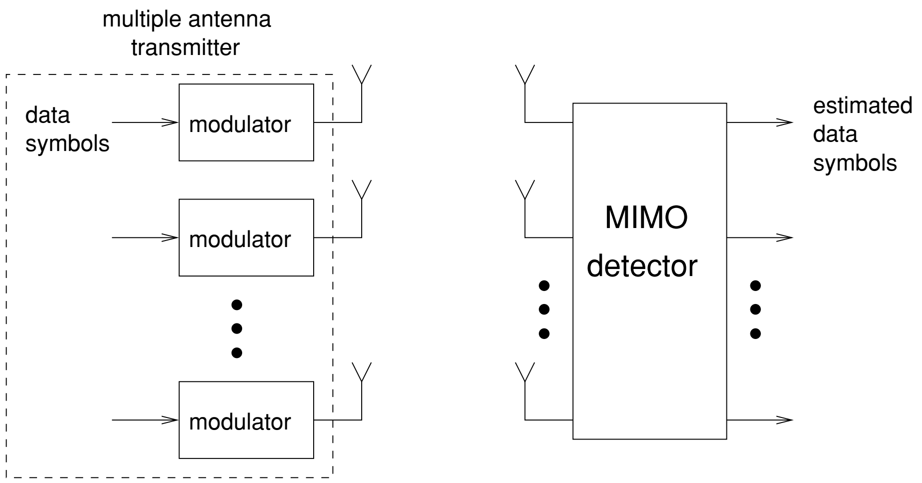

Figure 1.5. Block diagram for MIMO systems.

the multiple input multiple output (MIMO) channel. It is shown that the MIMO channel capacity increases linearly with the minimum of the numbers of transmit and receive antennas. This is an important finding, because MIMO systems can be used to provide high data rate services over wireless channels.

Various transmission schemes are studied, including a simple scheme in which each antenna transmits different data sequences. If a receiver is able to detect all the different data sequences, the data rate increases linearly with the number of transmit antennas. A receiver equipped with multiple receive antennas, as shown in Fig. 1.5, can detect multiple signals simultaneously and the resulting detection is called the MIMO detection. An optimal MIMO detector is the ML detector. Finding the ML solution vector that maximizes the likelihood function may require an exhaustive search; since the number of the candidate vectors grows exponentially with the number of transmit antennas, the ML detector becomes impractical. Thus, suboptimal, but less complex, detection algorithms are required. A combination of linear filtering and cancelation can lead to computationally efficient MIMO detection algorithms.

1.2 Issues in receiver design

There are a number of factors to consider in receiver design, and we focus on two particular issues in this book. The first is the relationship between data detection and channel estimation, and the second is the computational complexity.

1.2.1 *Data detection and channel estimation*

In practice, the receiver performs two major tasks: channel estimation and data detection. Even though channel estimation and data detection are tightly coupled, they are often dealt with separately. In the conventional approach, the channel estimation is carried out first to obtain a channel estimate. Then, the data detection is performed with the estimated channel. In fact, the channel estimation and data detection can be carried out jointly to achieve better performance.

There are various methods employed in data detection and channel estimation. In general, the following three groups of methods can be considered.

(i) Group 1: joint channel estimation and data detection methods;
(ii) Group 2: data detection oriented methods;
(iii) Group 3: channel estimation oriented methods.

For an ISI channel, a method in Group 1 attempts to find the estimates of the channel impulse response and data sequence simultaneously under a certain performance criterion, for example the maximum likelihood (ML) criterion.

Group 2 methods focus on the data detection, and the channel estimation is often implicitly carried out within the data detection process. For example, if each candidate data sequence has its own channel estimate, an exhaustive search can find an optimal data sequence (and its corresponding channel estimate) under a certain performance criterion. To avoid an exhaustive search, less complex suboptimal approaches could be investigated. In particular for an ISI channel, the trellis structure is exploited to reduce the complexity of the search. Some Group 2 methods apply to joint channel estimation and data detection (i.e., some Group 2 methods can be seen as Group 1 methods).

Group 3 methods are mainly investigated using statistical signal processing techniques, because the channel estimation problem can be seen as a parameter estimation problem. The main idea of Group 3 methods is based on the conventional approach for receiver design. As mentioned earlier, the conventional receiver has two separate steps: channel estimation and then data detection. If the channel is ideally estimated, the ensuing data detection will not experience any performance degradation. In one Group 3 method, the data detection can be carried out implicitly as part of the channel estimation.

It is difficult to compare these three groups in terms of performance and complexity. Each group has advantages and disadvantages over the others. However, throughout this book we mainly focus on Group 2 and 3 methods as they are suitable for adaptive and iterative receivers.

1.2.2 *Complexity*

The complexity, especially the computational complexity, is of extreme importance in receiver design. Fundamentally, it is possible to design an ideal (or optimal) receiver if an exhaustive search is affordable. However, in general, an exhaustive search becomes impractical because the complexity, which grows exponentially with the length of data sequences and/or number of users, is too high. A high complexity can result in undesirable results, such as long processing delays and high power consumption.

Adaptive and iterative signal processing techniques play a key role in deriving computationally efficient algorithms for receivers. The resulting receivers are adaptive or iterative.

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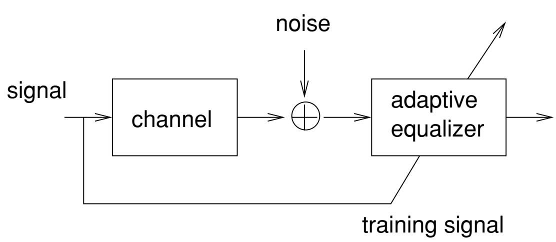

Figure 1.6. Adaptive equalization.

1.3 Adaptive signal processing

Adaptive signal processing has a wide range of applications, including its use in communication systems. The main idea of adaptive signal processing is learning.

For a channel equalizer, optimal equalization (coefficient) vectors can be found in terms of the channel impulse response (CIR) of an ISI channel. In practice, since the CIR is unknown, it has to be estimated. Then, with the estimated CIR as the true CIR, the corresponding optimal equalization vectors can be found. However, this approach has drawbacks. If the CIR varies, a new CIR should be estimated and the equalization vectors should be updated accordingly. In addition, due to the estimation error, the optimality would not be guaranteed.

Adaptive algorithms can find the equalization vectors without explicit knowledge of the channel. In general, an adaptive equalizer can find an equalization vector using a training sequence, as shown in Fig. 1.6. If adaptive algorithms are capable of tracking a time-varying CIR, the equalization vector can be continuously updated according to the channel variation.

Adaptive algorithms are also applicable to the channel estimation for an ISI channel. Adaptive channel estimation can be carried out in conjunction with the sequence detection. This results in an adaptive joint channel estimation and data detection.

For CDMA systems, adaptive algorithms are applied to the multiuser detection as the spreading codes of the other users may not be known. Adaptive linear multiuser detectors learn the statistical properties of interfering signals to suppress interfering signals.

In CDMA systems, statistical properties of the MAI can vary when there are incoming or outgoing users. Thus, the statistical properties of interfering signals are time-variant. In this case, the tracking ability of adaptive algorithms becomes essential in order to adjust the coefficients of a multiuser detector. Adaptive receivers are also important in tracking the variation of fading channels.

1.4 Iterative signal processing

There are various forms of iterative signal processing for communication systems. Iterative signal processing is an attractive idea as the iterations are expected to produce better performance. Generally, iterative signal processing techniques are off-line algorithms, while

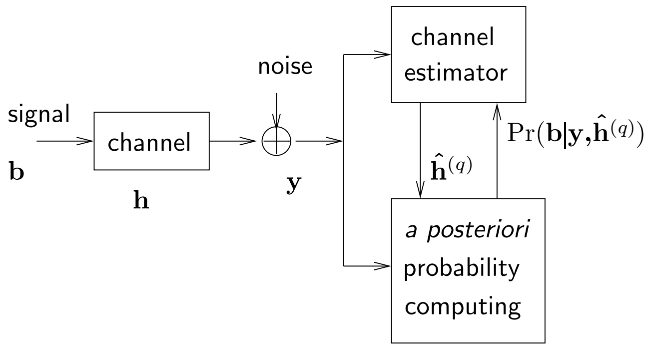

Figure 1.7. Iterative channel estimation approach.

adaptive signal processing techniques are on-line algorithms. In this book, we focus on two different methods for iterative signal processing: one is based on the expectation-maximization (EM) algorithm, and other is based on the turbo principle.

The EM algorithm is an iterative method used to solve various maximum likelihood (ML) estimation problems. If the likelihood function is highly nonlinear, a closed-form solution may not be available. In this case, numerical approaches can be used to find the ML solution. The main advantage of the EM algorithm over other numerical approaches is that it is numerically stable with each EM iteration increasing the likelihood. The EM algorithm has been widely used in a number of signal processing areas. The channel estimation is a typical example in which the EM algorithm is applicable.

Different approaches could be used to apply the EM algorithm to the channel estimation. Figure 1.7 shows a block diagram of an iterative channel estimation approach based on the EM algorithm approach, where $\mathbf{b}$, $\mathbf{h}$, and $\mathbf{y}$ denote the transmitted signal vector, the channel vector, and the received signal vector, respectively. The channel vector $\mathbf{h}$ may be estimated from $\mathbf{y}$. In general, channel estimation becomes difficult if $\mathbf{b}$ is unknown; in this case it is known as blind channel estimation. The iterative channel estimation provides a sequence of channel estimates through iterations. Since $\mathbf{b}$ is not available, we attempt to find the *a posteriori* probability of $\mathbf{b}$ from the received signal vector, $\mathbf{y}$, and, say, the qth estimate of $\mathbf{h}$, $\hat{\mathbf{h}}^{(q)}$, for the next estimation to obtain $\hat{\mathbf{h}}^{(q+1)}$, which is the $(q+1)$th estimate of $\mathbf{h}$. Through iterations, if the *a posteriori* probability of $\mathbf{b}$ becomes more reliable, a better estimate of $\mathbf{h}$ is expected. According to the principle of the EM algorithm, the channel estimate converges to the ML estimate if an initial estimate is sufficiently close to the ML estimate.

The iterative channel estimation approach is generic. It is applicable to ISI, CDMA, and MIMO channels. In addition, the channel estimator in EM-based channel estimation can also be modified as long as it can provide good performance. The main requirement of the channel estimator in Fig. 1.7 is that it should exploit the *a posteriori* probability of $\mathbf{b}$. There are other channel estimators which meet this requirement.

The major problem with the iterative channel estimation approach in Fig. 1.7 is that the complexity of finding the *a posteriori* probability of $\mathbf{b}$ can be too great. A straightforward implementation of the calculation of the *a posteriori* probability of $\mathbf{b}$ has the complexity

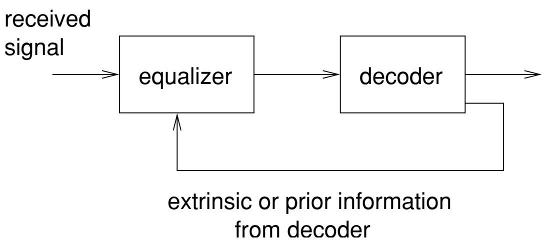

Figure 1.8. Turbo equalizer within the iterative receiver.

growing exponentially with the length of **b**. There are computationally efficient approaches that can find the *a posteriori* probability of **b** depending on the structure of the channels. For example, the trellis structure of an ISI channel can be exploited to find the *a posteriori* probability of **b** with a lower complexity.

The turbo principle was originally introduced to decode turbo codes, and it led to the proposal of an iterative decoding algorithm that provided an (almost) ideal performance. The turbo principle is also applicable to channel equalization for coded signals. The main idea behind the turbo equalizer is to exchange extrinsic information between the channel equalizer and channel decoder. Since more reliable prior information from a channel decoder is available, the ISI can be effectively suppressed in the turbo equalizer through iterations. An equalizer within an iterative receiver, as shown in Fig. 1.8, should take advantage of the prior information from the channel decoder.

The extrinsic information is regarded as prior information and is used to detect or decode data symbols together with the received signal. Therefore, the maximum *a posteriori* probability (MAP) principle is suitable for both equalizer and decoder. However, there can be other approaches to incorporating prior information. In particular, for channel equalization, a cancelation based approach can be employed. Using prior information of data symbols, a cancelation method can effectively mitigate the ISI with a low complexity. Thus, the MAP equalizer can be replaced by a cancelation based equalizer such as a DFE at the expense of minor performance degradation.

In CDMA systems, the turbo principle can be applied in the design of iterative receivers. For coded signals, an iterative receiver consists of a multiuser detector and a group of channel decoders. Using decoded signals, a better interference cancelation is expected if an interference cancelation based multiuser detector is used. As the output of the multiuser detector becomes more reliable, the resulting decoding would provide more reliable outputs. Through iterations, an (almost) ideal performance can be achieved.

The iterative receiver has several drawbacks. Generally, it has a long processing delay as several iterations are required. In addition, since iterative receivers are based on off-line processing, a number of memory elements are required to store a block of received signal samples as well as soft information of decoded bit sequences. Even though the iterative receiver has these drawbacks, it will be widely employed as it can provide almost ideal performance. In addition, given that computing power will increase and the cost of memory elements will become cheaper, there is no doubt that the iterative receiver will become popular in the future.

1.5 Outline of the book

The book is divided into three parts as follows.

- Part I: Chapters 2, 3, and 4 belong to Part I, in which the ISI channel is considered. To alleviate the ISI, various approaches are discussed with adaptive signal processing techniques.
  Chapter 2 mainly concentrates on channel equalizers. The sequence detection is an alternative approach to deal with the ISI problem. The adaptive channel estimation is introduced in conjunction with sequence detection in Chapter 3. In Chapter 4, we discuss sequence detection and channel estimation for fading channels, a brief introduction of which is also presented in Chapter 4.
- Part II: Iterative signal processing techniques and iterative receivers that include a channel decoder are introduced in Part II, which consists of Chapters 5, 6, and 7.
  We discuss the EM algorithm and introduce an iterative channel estimation method based on the EM algorithm in Chapter 5. An optimal symbol detection method is studied in conjunction with the iterative channel estimation. In Chapter 6, an iterative receiver is introduced for signal detection/decoding over ISI channels based on the turbo principle. We briefly introduce information theory and channel coding to provide a background for understanding of the iterative receiver. Two different approaches to the design of the iterative receiver for random channels are discussed in Chapter 7.
- Part III: Applications of adaptive and iterative signal processing techniques to other interference-limited communication systems are studied in Part III (Chapters 8, 9, 10, and 11).
  In Chapter 8, CDMA systems are introduced along with multiuser detection. Adaptive linear multiuser detectors are also discussed. The iterative receiver for CDMA systems is the major topic of Chapter 9.
  MIMO channels with multiple antennas are studied in Chapter 10 using diversity techniques. Various MIMO detection algorithms that are suitable for the iterative receiver are also presented.
  Orthogonal frequency division multiplexing (OFDM) is introduced in Chapter 11. OFDM is not interference-limited in an ideal situation. However, OFDM is introduced in this book as it is becoming important for wireless communications and an iterative receiver is included to improve the performance. The code diversity in OFDM systems and the iterative receiver based on the EM algorithm are discussed.

1.6 Limitations of the book

In this book, we do not attempt to introduce a comprehensive guide for communication systems. Rather, we aim at introducing adaptive and iterative signal processing with