13.1 INTRODUCTION

There are a number of applications where the reference signal is either not available or consists of a training signal that in communication systems implies in reduction of useful data transmission. In those cases, we should utilize some alternative objective functions applied to the available data as well as some knowledge related to the nature (properties) of the signals involved.

In this chapter, some adaptive-filtering algorithms are presented which do not utilize reference signal that are collectively known as blind adaptive-filtering algorithms. The algorithms are also called training-less or unsupervised algorithms since their learning do not include any reference or training signal. This chapter makes no attempt to cover this subject in breadth and in depth, but the interested reader can consult some books [1]-[4] for further details.

There are two main types of blind signal processing procedures widely discussed in the literature, namely blind source separation and blind deconvolution. In the former case several signal sources are mixed by an unknown environment and the objective of the blind signal processor is to separate these signal sources [2]. On the other hand, the blind deconvolution aims at removing the effect of a linear time-invariant system on a signal source where the only assumptions are the observation of the signal before the deconvolution process and the probability density of the input signal source.

Blind deconvolution is obviously closely related to blind equalization, and the distinction lies on the fact that in the equalization case it is usually assumed that the input signal belongs to a prescribed finite set (constellation) and the channel is a continuous-time channel. These features of the equalization setup are assets that can be exploited by allowing nonlinear channel equalization solutions, whereas blind deconvolution employs linear solutions because its input signal cannot be considered to belong to a finite set constellation. However, it is fact that several solutions for both problems are closely related and here we emphasize the blind equalization case.
In blind equalization the channel model is either identified explicitly or implicitly. The algorithms utilizing as objective function the minimization of the MSE or generating a zero-forcing (ZF) solution\(^1\) in general do not estimate the channel model explicitly. On the other hand, nonlinear solutions for channel equalization such as maximum likelihood sequence detector (MLSD) [8] and the DFE require explicit estimation of the channel model.

As a rule, the blind signal processing algorithms utilize second and higher order statistics indirectly or explicitly. The high-order statistics are directly employed in algorithms based on cumulants, see [9] for details, and they usually have slow convergence and high complexity. There is yet another class of algorithms based on models originated from information theory [3].

This chapter deals with blind algorithms utilizing high-order statistics implicitly for the single-input single-output (SISO) equalization case, e.g. constant modulus algorithm (CMA), and algorithms employing second-order statistics for the single-input multi-output (SIMO) equalization case. Unfortunately the SISO blind solutions have some drawbacks related to the multiple minima solutions, slow convergence, and difficulties in equalizing channels with nonminimum phase\(^2\). In the SIMO case we are usually dealing with oversampled received signal, that is, the received signal is sampled at rate multiple of the symbol rate (at least twice). Another SIMO situation is whenever we use multiple receive antennas that can be proved to be equivalent to oversampling. Such sampling higher than baud rate results in received signals which are cyclostationary allowing the extraction of phase information of the channel. In the case of baud rate sampling and WSS inputs, the received signal is also WSS and only minimum-phase channels can be identified from second-order statistics since the channel phase information is lost. Under certain assumptions the SIMO configuration allows the identification of the channel model as well as blind channel equalization utilizing only second-order statistics. In particular, this chapter presents the Godard, CM, and Sato algorithms for the SISO case. We also discuss some properties related to the error surface of the CMA. Then we derive the blind CM affine projection algorithm which is then applied to the SISO and SIMO setups.

### 13.2 CONSTANT-MODULUS ALGORITHM

In this section we present a family of blind adaptive-filtering algorithms that minimizes the distance between the modulus of the equalizer output and some prescribed constant values, without utilizing a reference signal. These constant values are related to the modulus of constellation symbols, denoted by \(C\), of typical modulations utilized in many digital communication systems. The earlier blind equalization proposals addressed the case of Pulse Amplitude Modulation (PAM) for the case the channel model is considered a linear time-invariant Single-Input Single-Output (SISO) system [5]-[6], operating at symbol rate. This approach was latter generalized in [7] by modifying the objective function to consider higher order statistics of the adaptive-filter output signal that accommodates the case of Quadrature Amplitude Modulation (QAM).

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1In the ZF solution the equalized signal is forced to be equal to the transmitted signal, a solution not recommended whenever the environment noise is not negligible, due to noise enhancement. The ZF equalizer aims at estimating a channel inverse in order to eliminate intersymbol interference.

2Channels whose discrete-time models have poles and zeros outside the unit circle.