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Performance analysis and comparison of blind to non-blind least-squares equalization with respect to effective channel overmodeling

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Abstract

The object of this work is the study of a direct blind equalization algorithm which appeared recently in the literature. It is a least-squares (LS) equalization method in the blind context, assuming a linear FIR communication channel and a linear equalizer. If channel order is known, blind LS equalizers can be constructed that entirely suppress intersymbol interference in noiseless signal transmission. In practice, though, channels may be comprised of a few "big" consecutive taps, which we call "significant part", surrounded by a lot of smaller leading and/or trailing "tail" terms. In such an environment, channel order is harder to define while the value used by the algorithm is critical to its performance. We carry out both theoretical analysis, making use of perturbation theory arguments, and simulations for the cases where channel order determination procedure has yielded an estimate greater than ("effective overmodeling") or equal to the order of the significant part. Our purpose is to compare the performance of blind LS algorithm with that of its non-blind counterpart. We conclude that (a) when channel does not possess leading tail terms, blind LS is robust to effective overmodeling, meaning that it behaves very much like non-blind LS, and (b) when leading tail terms are present, blind LS will generally not work satisfactorily in the effective overmodeling scenario. In either case, when the order of the significant part is identified correctly and the actual significant parts of subchannels are sufficiently diverse, the algorithm behaves well. © 2002 Elsevier Science B.V. All rights reserved.

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1. Introduction

Intersymbol interference (ISI) is one of the main factors obstructing reliable digital communications. It indicates the spreading in time of the transmitted symbols by the propagation medium and may be destructive at high enough symbol rates. In order to remove the corrupting effects of ISI, a special device is employed at the receiver called an *equalizer*.

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Conventional equalizer design techniques rely on the periodic transmission of data already known to the receiver, referred to as *training sequences*. A priori knowledge of such data allows for either the direct computation of the equalizer or the computation of the channel coefficients as a preliminary step before equalizer determination [9].

Training sequences clearly result in a waste of some of the channel's capacity. In order to allocate the maximum possible transmitting capacity to the users, *blind equalization* algorithms have received extensive attention. They do *not* make use of training sequences but rely solely on the output of the communication channel to achieve the desired equalization task.

The more traditional of these techniques sample the output of the channel at the baud rate. Therefore, they inevitably use higher-than second-order statistics (HOS) of the sampled symbols since only in this way is it possible to retrieve channel phase information. Nonetheless, this characteristic is responsible for two important disadvantages, namely: need for large sample sizes and potential capture in undesirable local minima.

In the pioneering work of [13] it is proved that channel phase information is present in channel output second-order statistics (SOS) if the input is observed through more than one sufficiently diverse channels. This amounts to oversampling the channel output and/or using several antennas at the receiver. These implementations are equivalent at a higher level of abstraction since they can all be modeled as a number of separate "virtual" channels driven by the same input. This setting has been named SIMO in the literature after "single-input multiple-output". Hence, the equalizers follow a "multiple-input single-output" (MISO) setting, in that they exploit each of the multiple virtual channel outputs to yield the equalized output for the actual channel.

SOS techniques alleviate the problems of HOS techniques and are therefore advantageous. Several algorithms have been developed with the SOS-SIMO setting in mind that either directly estimate the equalizer [2,3,8,11] or estimate the channel [1,7,14] at an initial step. What will ultimately determine the usefulness of these techniques is their robustness to real-world conditions which, more often than not, stray from theoretical assumptions. It is well known that most blind channel identification methods are very sensitive to channel overmodeling. Direct blind equalization algorithms were developed in the hope of overcoming this kind of sensitivity. It remains to be studied, however, if this is really the case.

One representative of the class of blind SOS algorithms that directly compute the equalizer is described in [10]. It is the blind analog of non-blind LS equalization. Briefly put, if the order of the channel is M and its output is oversampled by a factor of p, then an equalizer of order $L_{\mathrm{eq}} \geqslant M/(p-1)-1$ can be found that will *entirely* suppress the ISI introduced in the noiseless transmission of a white input sequence. In a typical implementation, a channel order determination procedure is employed to furnish an estimate that is subsequently fed into the algorithm.

Of particular interest is the case where a rather long channel of order M is incorrectly detected to be of order $L+1$ where $L+1<M$. Long channels appear in the context of microwave radio links [5,6] and they are usually comprised of a few big consecutive taps (with, probably, some small intermediate taps), called "significant part" throughout the paper and whose order we symbolize by L^*+1, while the rest of them are rather small leading and trailing terms and are referred to as "tails".

In this work, we attempt to examine the robustness properties of [10] in the 1-input/2-output channel context when the channel has a total order of M and the equalizer order is $L<M-1$, that is, shorter than required for perfect input reconstruction. Furthermore, we assume that all participating statistical quantities are known with infinite precision and the system is noiseless. Our aim is to unveil potential sensitivity of the algorithm to the channel-order mismatch. Statistical inaccuracies and additive channel noise are naturally expected to deteriorate system performance. It is interesting to remark at this point that the effect of long, small tail terms is equivalent with the presence of coloured noise in the system.

The rest of our paper is organized as follows: In Section 2, we present the channel model used and we review the algorithm developed in [10]. Section 3 is devoted to our contribution, i.e., the performance analysis of blind LS. In Sections 3.1 and 3.2, we decompose the "equalization" of the Mth-order channel