Multiuser Blind Channel Estimation and Spatial Channel Pre-Equalization*

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Abstract

In this paper, we study two of the fundamental operations of a TDD smart antenna system, namely, the uplink *channel and sequence identification* and downlink *selective transmission*. For uplink, our focus is on the development of a *blind* estimation algorithm which is capable of resolving a multiuser system without the use of training sequence or any input statistics. For downlink, we propose a spatial channel pre-equalization scheme which simultaneously eliminates the intersymbol interference (ISI) and the co-channel interference (CCI) for all users using FIR filters. Both algorithms were validated by RF experiments using the smart antenna testbed developed in the University of Texas at Austin.

1 Introduction

Recently, Smart Antenna Systems (SAS) [1, 2, 3] have been proposed to substantially increase the capacity and coverage of wireless communication systems by exploiting the spatial diversity. Two of the fundamental operations of a smart antenna system are the uplink sources separation and the downlink selective transmission. Since the uplink signals are usually distorted by both ISI and CCI, one needs the knowledge of the spatial channel responses between each user and the base-station antenna array. Conventional approaches relying on the same training sequence as in current wireless standards, *e.g.*, GSM and IS-45, are not so useful in a multiuser scenario. Blind estimation algorithm needs to be applied to separate different co-channel signals. However, most algorithms to date consider only one source [4, 5, 6, 7, 8, 9] with an exception of [10]. The main objective of this paper is to introduce a recursive blind estimation algorithm to resolve a multiuser system without relying on the input statistics information.

In Time-Division-Duplex (TDD) systems, the uplink and downlink channels have the same characteristics. It is suggested in [3] that one can transmit signals with the same reception antenna pattern since the channels are reversible. Such a scheme takes advantage of the spatial and temporal diversities among the users and can effectively reduce the interference at the user receivers. This idea, although conceptually simple, is difficult to implement especially in the presence both short- and long-delay multipath. In this case, the pre-compensation of FIR channels requires linear equalization with an infinite length tapped delay line [3].

The second goal of this paper is to propose a practical channel pre-equalization technique for wireless systems with frequency-selective fading. The novel contribution is exploitation of structure information of an antenna array system which allows *perfect* pre-equalization (removes ISI) and cancellation (removes CCI) of other co-channel FIR channels by FIR filters. In particular, we design a set of FIR filters to preprocess the signals such that outputs from user receivers are interference free.

2 Uplink Blind Identification

We consider the case where the composite channels of a wireless system can be perfectly modeled as FIR filters. We also assume that the information bearing symbols are drawn from a finite set of alphabets, *e.g.*, BPSK or QPSK. Under the above assumptions, it is shown in [11] that by means of spatial oversampling (antenna antenna) or/and temporal oversampling, the system output is related to its input and the channel responses by $\mathbf{y}(n) = \sum_{l=0}^{L} \mathbf{h}(l)s(n-l)$. In the case of d ($d > 1$) co-channel users, it becomes $\mathbf{y}(n) = \sum_{i=1}^{d} \mathbf{h}_i \otimes s_i(n)$. The problem under consideration is to estimate $\{\mathbf{h}_i\}$ and $\{s_i(n)\}$ from a finite number of system outputs $\mathbf{y}(1), \cdots, \mathbf{y}(N)$ without *any* knowledge of the inputs.

We begin by reviewing a multiuser blind estimation algorithm [10] for the simplest case of a multiuser system where all the channels have the same order. Next, we extend our study to a general system and focus on the deconvolution of channels with nonidentical orders. Finally, we establish a complete implementation procedure.

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2.1 A Simple Case

Given a finite number of data samples, the channel output vectors can be arranged in a matrix form,

$$\mathbf{Y}(K) = \begin{bmatrix} \mathbf{y}(1) & \mathbf{y}(2) & \cdots & \mathbf{y}(N-K+1) \\ \mathbf{y}(2) & \mathbf{y}(3) & \cdots & \mathbf{y}(N-K+2) \\ \vdots & \vdots & \cdots & \vdots \\ \mathbf{y}(K) & \mathbf{y}(K+1) & \cdots & \mathbf{y}(N) \end{bmatrix} = \underbrace{[\mathbf{H}_1(K) \ \cdots \ \mathbf{H}_d(K)]}_{\mathbf{H}(K)} \underbrace{\begin{bmatrix} \mathbf{S}_1(K+L) \\ \vdots \\ \mathbf{S}_d(K+L) \end{bmatrix}}_{\mathbf{S}(K+L)}, \quad (1)$$

where K is defined as the *smoothing factor*, and

$$\mathbf{S}_i(K+L) = \begin{bmatrix} s_i(1-L) & \cdots & s_i(N-L-K+1) \\ s_i(2-L) & \cdots & s_i(N-L-K+2) \\ \vdots & \cdots & \vdots \\ s_i(K) & \cdots & s_i(N) \end{bmatrix}.$$

Our approach to resolving this multiuser system involves two steps [10],

Step 1. Channel Deconvolution: Assuming $\mathbf{H}(K)$ is of full column rank, we can see that the row span of $\mathbf{Y}(K)$ is then the same as that of $\mathbf{S}(K+L)$, *i.e.*, a least-squares approach for accomplishing the deconvolution was provided in [11]. The method is summarized as fellows,

1. From $\mathbf{Y}(K)$, compute the null space $\mathbf{V}_o(K)$.
2. Construct the following matrix

$$\mathbf{V}(K+L) = \underbrace{\begin{bmatrix} \mathbf{V}_o(K) & \cdots & \cdots & \mathbf{0} \\ \mathbf{0} & \mathbf{V}_o(K) & \cdots & \vdots \\ \vdots & \cdots & \ddots & \mathbf{0} \\ \mathbf{0} & \cdots & \cdots & \mathbf{V}_o(K) \end{bmatrix}}_{K+L \ blocks}, \quad (2)$$

where $\mathbf{0}$ is a column vector.

3. The *synchronized and deconvolved* input matrix $\mathbf{S}(1)$ is related to the null space of $\mathbf{V}(K+L)$ by $\mathbf{V}^{\perp}(K+L) = \mathbf{W}\mathbf{S}(1)$, where $\mathbf{W}$ is a $d \times d$ full rank matrix.

The inputs are readily determined from $\mathbf{V}^{\perp}(K+L)$ if there is only one user. When $d > 1$, the blind estimation problem reduces to the identification of $\mathbf{S}(1)$ and $\mathbf{W}$ from $\mathbf{V}^{\perp}(K+L)$.

Step 2. Symbol Identification: To identify $\mathbf{S}(1)$ from $\mathbf{V}^{\perp}(K+L)$, notice that for most digital communication signals, the information bearing symbols are from a finite set of alphabets. It was proved in [12, 13] that blind symbol estimation can be achieved given sufficient data samples. We adopt a so-called Iterative Least-Squares with Projection (ILSP) algorithm to accomplish this, see [13] for details.

Initialization:
Choose the smoothing factor K
Determine the highest order of the channels L
Construct $\mathbf{Y}(K)$
Computer the null space $\mathbf{V}_o(K)$ from $\mathbf{Y}(K)$
For $l = L, \cdots, 1$, if $d_l \neq 0$, do:
Construct $\mathbf{V}(K+l)$ as in (2)
Calculate its null subspace $\mathbf{V}^{\perp}(K+l)$
Apply $\mathbf{V}^{\perp}(K+l)$ and $\mathbf{S}^{l+1}(2), \cdots, \mathbf{S}^{L}(L-l+1)$ to the Partial ILSP algorithm, identify $\mathbf{S}^{l}(1)$
Output:
The symbols from each user
The channels between the users and the base-station

Table 1: Recursive Blind Identification

2.2 General Cases

Rearrange the inputs and channels into L groups such that each group has d_l users with the same channel order, $l+1$. Denote $\mathbf{H}^l(K)$ and $\mathbf{S}^l(K+l)$ as the channel and input matrix for a subsystem. Consequently

$$\mathbf{Y}(K) = \underbrace{[\mathbf{H}^1(K) \ \cdots \ \mathbf{H}^L(K)]}_{\mathbf{H}(K)} \underbrace{\begin{bmatrix} \mathbf{S}^1(K+1) \\ \vdots \\ \mathbf{S}^L(K+L) \end{bmatrix}}_{\mathbf{S}(K)}.$$

A *recursive* identification procedure is established in Table 1. The key techniques are explained in the following.

Channel Deconvolution By the Hankel structure of $\mathbf{S}^l(K+l)$ and the fact that $\mathbf{V}(1) = \mathbf{V}_o(K) \perp \{\mathbf{S}^L(K+L), \cdots, \mathbf{S}^1(K+1)\}$, we may properly rearrange the corresponding blocks and obtain the following:

$$\begin{array}{llllll} \mathbf{V}(K+1) & \perp & \mathbf{S}^L(L+1), & \mathbf{S}^{L-1}(L), & \cdots & \mathbf{S}^1(1); \\ \vdots & \vdots & \vdots & \vdots & & \\ \mathbf{V}(K+L-1) & \perp & \mathbf{S}^L(2), & \mathbf{S}^{L-1}(1); & & \\ \mathbf{V}(K+L) & \perp & \mathbf{S}^L(1) & & & \end{array} \quad (3)$$

Clearly, $\{\mathbf{V}(K+l)\}$ have *inherent* null vectors $\mathbf{N}(l) \stackrel{\text{def}}{=} [\mathbf{S}^l(1)^T \ \mathbf{S}^{l+1}(2)^T \ \cdots \ \mathbf{S}^L(L-l+1)^T]^T$. It is proved in [14] that in general, $\{\mathbf{N}(l)\}$ form the null space of $\mathbf{V}(K+l)$. Therefore, the span of the input vectors can be obtained by calculating the null space of $\{\mathbf{V}(K+l)\}$.

Partial ILSP Algorithm The identification of $\mathbf{S}(L)$ can be accomplished directly using ILSP. For subsystems with lower orders, part of the symbols in $\mathbf{N}(l)$ which correspond to the subsystems with higher order channels already identified, we propose modified ILSP with uses the *a priori* knowledge for fast convergence.

Let $\mathbf{P}$ be the *unknown* symbol matrix and let $\mathbf{Q}$ be the *known* symbol matrix. Denote $\mathbf{O} = \mathbf{W}[\mathbf{P}^T \ \mathbf{Q}^T]^T$ where $\mathbf{W}$ is a full rank square matrix, a Partial ILSP method which identifies $\mathbf{P}$ given $\mathbf{O}$ and $\mathbf{Q}$ can be easily extended from the original ILSP algorithm.

1. Given $[\mathbf{P}_k]_{ij}$ with each element an alphabet,
2. $k := k+1$,
   - $\mathbf{P}_{k+1} = \mathbf{P}_k [\mathbf{P}_k^H \ \mathbf{Q}^H](\mathbf{O}[\mathbf{P}_k^H \ \mathbf{Q}^H])^{-1}\mathbf{O}$,
   - Project $[\mathbf{P}_{k+1}]_{ij}$ to the closest alphabets,
3. Continue until $(\mathbf{P}_{k+1} - \mathbf{P}_k) = 0$.

Recursive Blind Identification We have explained all the key operations for an Recursive Blind Identification (RBI) scheme for a general system. The estimation procedures in Table 1 can now be well understood:

- The first step is to choose the initial parameters such as K, L and calculate the null space of $\mathbf{Y}(K)$, $\mathbf{V}_o(K)$.
- The next part of RBI is a loop indexed by l which gradually reduces the channel orders and determine all the subsystems. In each loop, the rank of $\mathbf{V}(K+l)$ is first studied to determine whether or not $d_l = 0$. Then $\mathbf{V}^\perp(K+l)$ and the already identified $\mathbf{S}^{l+1}(1), \cdots, \mathbf{S}^L(1)$ are passed to the Partial ILSP algorithm to obtain $\mathbf{S}^l(1)$.
- When the loop is completed, the channels can be identified by least-squares fitting.

3 Downlink Selective Transmission

Since the channels characterize the spatial and the temporal diversities between the users and the base-station antenna, one should try to perform selective transmission using antenna pattern which inverts the uplink channels. In this section, we propose a practical scheme which we term as *spatial channel pre-equalization* to perfectly equalize (removes ISI) and cancel (removes CCI) FIR channels using FIR filters. In particular, we design a set of FIR filters to pre-process the signals such that outputs from user receivers are free of interference [15, 8].

Derivation

Consider a single user case, if the base-station transmits $\mathbf{y}(n)$, the user receives $s(n) = \mathbf{h}(n) \otimes \mathbf{y}(n) \stackrel{\text{def}}{=} \sum_{l=0}^{L} \mathbf{y}^H(n-l)\mathbf{h}(l)$. The goal of pre-equalization is to find a *vector* FIR filter $\mathbf{g}(n) = [g^1(n) \cdots g^M(n)]^T$, such that $\mathbf{y}(n) = \mathbf{g}(n) \otimes s^*(n)$ and $\mathbf{h}(n) \otimes \mathbf{y}(n) = s(n)$. Restate the problem in a matrix form,

$$\underbrace{\begin{bmatrix} \mathbf{h}(L)^T & & 0 \\ \mathbf{h}(L-1)^T & \ddots & \vdots \\ \vdots & \ddots & \mathbf{h}(L)^T \\ \mathbf{h}(0)^T & \ddots & \mathbf{h}(L-1)^T \\ & \ddots & \vdots \\ 0 & & \mathbf{h}(0)^T \end{bmatrix}}_{\mathbf{H}(K+1)} \underbrace{\begin{bmatrix} \mathbf{g}^*(0) \\ \mathbf{g}^*(1) \\ \vdots \\ \mathbf{g}^*(K) \end{bmatrix}}_{\mathbf{g}} = \underbrace{\begin{bmatrix} 0 \\ \vdots \\ 0 \\ 1 \\ 0 \\ \vdots \\ 0 \end{bmatrix}}_{\mathbf{1}(m)}. \tag{4}$$

Notice that due to the oversampling, each element in the left side of (4) is now a vector. The dimension of $\mathbf{H}(K+1)$ is $(L+K+1) \times M(K+1)$. As long as the number of channels $M \geq 2$, there exists such a K such that $\mathbf{H}(K+1)$ has more columns than rows. It is readily seen that $\mathbf{g}$ can be found as $\mathbf{H}^\dagger(K+1)\mathbf{1}(m)$, where $\dagger$ denotes the pseudo-inverse.

The Multiuser Case For a system with d ($d > 1$) users, let $\mathbf{h}_i(n)$ be the channel characteristics between the i^{th} user and the base-station, the task of pre-equalization can be expressed as

$$\underbrace{\begin{bmatrix} \mathbf{H}_1(K+1) \\ \vdots \\ \mathbf{H}_d(K+1) \end{bmatrix}}_{\mathbf{H}(K+1)} \underbrace{[\mathbf{g}_1 \cdots \mathbf{g}_d]}_{\mathbf{G}} = \underbrace{\begin{bmatrix} \mathbf{1}(m) & & 0 \\ 0 & \ddots & 0 \\ 0 & \cdots & \mathbf{1}(m) \end{bmatrix}}_{\mathbf{T}}, \tag{5}$$

where $\mathbf{H}_i(K+1)$ and $\mathbf{g}_i$ with the subscripts as user indices have the structure defined in (4).

The dimension of the multiuser $\mathbf{H}(K+1)$ matrix is now $d(L+K+1) \times M(K+1)$. As long as $M > d$, there exist a K such that (5) becomes underdetermined and has a perfect solution.

4 Experimental Results

We validated all our algorithms by RF field experiments using the smart antenna testbed developed at the University of Texas at Austin. Our facilities include a fully adaptive 8-element uniform linear antenna array, several remote transceivers, and central control units. The carrier frequency is 900MHz and the message signal is 50K symbols/s BPSK sequence.

4.1 Uplink Blind Identification

We used multiple transmitters to generate two uplink signals, each with long-delay multipath. 50 sample vectors were collected at the baud rate. We applied the proposed RBI method to identify the inputs and channels. The channels were then used to separate and equalized the original data from the antenna outputs. The results are illustrated in Figure 1.

4.2 Downlink Selective Transmission

We calculated the pre-equalization filters according to the identified uplink channels. Each message signal was convolved with its corresponding filter and the superposition of the resulting signals were transmitted from the base-station antenna array. The signals received by each user were plotted in Figure 2. Comparing to the constellations in uplink, the degradation may attributed to the mismatch of the antenna array, or an actual change of the environment.

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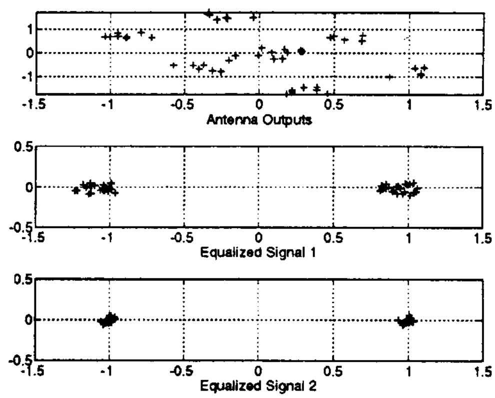

Figure 1: Uplink Source Separation

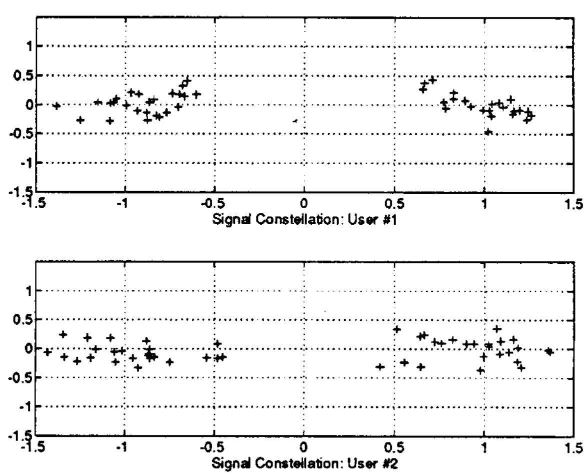

Figure 2: Downlink Selective Transmission