行政院國家科學委員會補助專題研究計畫成果報告

利用週期性調變通訊頻道之盲判別

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計畫編號: NSC-91-2219-E009-049

執行期限: 91年8月1日 至 92年7月31日

主持人: 林清安

計畫參與人員: 吳卓諭

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Blind Identification with Periodic Modulation

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中文摘要

本計畫研究通訊頻道盲判別的問題, 提出一個利用週期性調變的方式來做有限脈衝頻道盲判別的方法。所提出的方法利用接收信號的相關矩陣與頻道參數乘積之間的線性關係, 來提高頻道參數估測的準確性。除了判別的方法之外, 本研究提出簡單的頻道可判別的條件及週期性頻變序列的最佳設計。模擬結果顯示, 本研究所提出的方法優於文獻上子空間的判別方法。

關鍵字: 無線通訊、盲判別、週期性調變

Abstract

We propose a method for blind identification of FIR channels with periodic modulation. The time-domain formulation in terms of block signals is simple compared with existing frequency-domain formulations. It is shown that the linear equations relating the products of channel coefficients and the autocorrelation matrix of the received signal can be further arranged into decoupled groups. The arrangement reduces computations and improves accuracy of the solution; it also leads to very simple identifiability conditions and a very natural formulation of the optimal modulating sequence selection problem. The proposed optimal selection minimizes the effects of channel noise and error in autocorrelation matrix estimation; it results in a consistent channel estimate when the channel noise is white. Simulation results show the method yields good performance: it compares favorably with existing subspace modulation- induced-cyclostationarity method and it is robust with respect to channel order overestimation.

Keywords: wireless communication, blind identification, periodic modulation.

1 Introduction

To achieve high-speed reliable communication, channel identification and equalization is necessary to reduce intersymbol interference (ISI) in many communication environments. Channel identification and equalization can be achieved either by sending training sequences, or by designing the equalizer based on *a priori* knowledge of the channel. *A priori* knowledge is often not available in a radio (wireless) communication environment and sending training sequences reduced data transmission rate. Blind channel identification and equalization, which does not assume *a priori* channel knowledge or send training data, have traditionally relied on high-order statistics (HOS) of the stationary received data, but this usually requires relatively long data which is needed to accurately estimate the HOS [9]. This motivates approaches using only induced second-order cyclostationary statistics.

Blind identification and equalization of finite impulse response channel (FIR) channels which exploits cyclostationarity of second-order statistics of the received data is first proposed in [1]. Various schemes have since been proposed, *e.g.*, [2, 3]. Cyclostationarity can be induced at the receiver or at the transmitter. While receiver induced cyclostationarity is always through oversampling or multiple sensing, many different schemes have been proposed to induce cyclostationarity at the transmitter. They include periodic modulation [4, 5], repetition coding [6] and a combination of repetition and modulation[7] and filter bank precoding [8].

We study the problem of blind channel identification with periodic modulation of source symbols. We formulate the problem in time-domain and in terms of block signals. The method exploits the linear relation between the products of channel coefficients and the autocorrelation matrix of the received signal and computes the products first by solving a set of linear equations. The channel coefficients are then obtained (to within a scalar ambiguity) by computing the dominant eigenvector of an associated Hermitian matrix. We show that the set of linear equations relating the products of coefficients and the autocorrelation matrix can be further arranged into decoupled groups. The arrangement reduces computations and improves accuracy of the solution; it also leads to very simple identifiability conditions, which depend on the modulating sequence alone, and a very natural formulation of the optimal modulating sequence selection problem. The proposed optimal selection minimizes the effects of channel noise and error in autocorrelation matrix estimation.

The report is organized as follows. Section 2 is the problem statement and preliminary. Section 3 establishes the identifiability conditions, proposes an identification algorithm and discusses numerical aspects associated with it. In Section 4, the problem of selecting the modulating sequence is formulated and solved. In Section 5, simulation examples are given to illustrate the performance of the proposed method. Section 6 is conclusions.

2 Problem statement and preliminary

2.1 Problem statement

We consider the baseband transmission system. The source symbol sequence $s(n)$ is modulated by a (real) periodic sequence $p(n)$ with period N to obtain the modulated sequence $w(n) = p(n)s(n)$ which is sent through the channel. The channel is modelled as an FIR filter, whose input-output relation is

$$z(n) = \sum_{l=0}^{L} h(l)w(n-l) \tag{2.1}$$

where $h(n)$ is the impulse response of the channel and L is the channel order.

The received signal sequence $x(n)$ id the sum of the filtered signal $z(n)$ and an additive noise, *i.e.*,

$$x(n) = z(n) + v(n) \tag{2.2}$$

We propose a method for identifying $h(n)$ using second order statistics of $x(n)$ and a method for optimal design of the modulation sequence $p(n)$. The following assumptions are made throughout.

(A1) The source $s(n)$ is zero mean, white, and with unit variance.

(A2) The noise $v(n)$ is stationary with zero mean and is uncorrelated with $s(n)$.

(A3) An upper bound $\hat{L}$ on channel order L is known and the period $N > \hat{L} + 1$.

2.2 Preliminary

Define the block received signal $\bar{x}(n) := \begin{bmatrix} x(nN) & \cdots & x(nN+N-1) \end{bmatrix}^T$ and let the block signals $\bar{w}$, $\bar{s}$, $\bar{z}$, and $\bar{v}$ be similarly defined. In terms of block signals the channel relation (2.1) and (2.2) can be written as

$$\bar{x}(n) = H_0 G \bar{s}(n) + H_1 G \bar{s}(n-1) + \bar{v}(n) \tag{2.3}$$

where $H_0 \in \mathbf{C}^{N\times N}$ is a lower triangular Toeplitz matrix with $\begin{bmatrix} h(0) & \cdots & h(L) & 0 & \cdots & 0 \end{bmatrix}^T$ as its first column, $H_1 \in \mathbf{C}^{N\times N}$ is an upper triangular

Toeplitz matrix with $\begin{bmatrix} 0 & \cdots & 0 & h(L) & \cdots & h(1) \end{bmatrix}$ as its first row, and $G \in \mathbf{R}^{N\times N}$ is an diagonal matrix whose jth diagonal entry is $p(j-1)$. Equation (2.3) is a time-invariant description of the channel in terms of block signals.

3 Channel identification

3.1 Identification equation: noise free case

We consider first the noise free case, *i.e.*, $x(n) = z(n)$. We assume for the moment that the channel order is known. The autocorrelation matrix of $\bar{x}(n)$ can be computed from (2.3) as

$$R_{\bar{x}}(0) = E\bar{x}(n)\bar{x}(n)^* = H_0 G^2 H_0^* + H_1 G^2 H_1^* \quad (3.1)$$

The equation is quadratic in the channel coefficients $h(0)$, $\cdots$, $h(L)$. If we consider the products $h(k)h(l)$ as unknowns, then (3.1) is a system of $N(N+1)/2$ linear equations (we need only to consider the upper triangular part.) It can be further divided into $L+1$ decoupled groups of equations with smaller dimensions, by exploiting the Toeplitz structure of H_0 and H_1. We describe precisely the equations below.

We define $\Gamma_j[Q] \in \mathbf{C}^{N-j}$ as the vector consists of the jth upper diagonal entries of the matrix $Q \in \mathbf{C}^{N\times N}$, and define the vector $f_j \in \mathbf{C}^{L-j+1}$ as

$$f_j = \begin{bmatrix} h(0)h(j)^* & h(1)h(j+1) & \cdots & h(L-j)h(L) \end{bmatrix}$$

The $L+1$ decoupled groups of equations can then be expressed as

$$\Gamma_j[R_{\bar{x}}(0)] = M_j f_j, \quad 0 \le j \le L \quad (3.2)$$

where the the klth entry of $M_j \in \mathbf{R}^{(N-j)\times(L-j+1)}$ is

$$(M_j)_{kl} = \begin{cases} p(0)^2, & \text{if } k = l; \\ p(k-l)^2, & \text{if } k > l; \\ p(N-l+k)^2, & \text{if } k < l. \end{cases}$$

We note that the matrix M_0 is a complex $N \times (L+1)$ circulant matrix with $\begin{bmatrix} p(0)^2 & \cdots & p(N-1)^2 \end{bmatrix}^T$ as its first column. The matrix M_j can be obtained from M_0 by deleting its last j rows and last j columns. Solving these $L+1$ sets of equations we would get the products $h(k)h(l)^*$, $k = 0, \cdots, L$, $l \ge k$.

3.2 Identifiability condition

Since we consider noise free case and choose $N > L+1$, so every set of equations in (3.2) is overdetermined and consistent. If each matrix M_j is full column rank then the products of channel coefficients can be solved uniquely as

$$f_j = (M_j^T M_j)^{-1} M_j^T \Gamma_j[R_{\bar{x}}(0)] \quad (3.3)$$

Let Q be the Hermitian matrix whose ijth element is $h(i)h(j)^*$. The channel coefficients $h(0), \cdots, h(L)$ can be determined to within a scalar ambiguity by computing the eigenvector of Q associated with its largest eigenvalue. We thus have the following sufficient condition for identifiability.

Identifiability condition: The channel is identifiable if each M_j in (3.2) is full column rank.

Since M_j depends only on the modulating sequence $p(n)$, by choosing $p(n)$ properly we can always make M_j full rank. We thus conclude that every channel is identifiable.

3.3 Identification algorithm

Based on the discussions so far, we propose the following algorithm for the computation of channel coefficients. Assume that the modulating sequence has been chosen so that each M_j is full rank.

Identification algorithm

step1: Compute estimate of correlation matrix $R_{\bar{x}}(0)$ via time average

$$\hat{R}_{\bar{x}}(0) = \frac{1}{K}\sum_{i=1}^{K} \bar{x}(i)\bar{x}(i)^*$$

where K is the number of data block.

step2: Compute the product coefficients by (3.3).

step3: Form the matrix Q defined previously and compute its unit-norm eigenvector associated with the largest eigenvalue.

4 Optimal modulation sequence

We consider the general case, that is, the channel noise is present, and discuss the problem of selecting the modulating sequence $p(n)$. We first propose an optimality criterion to select $p(n)$ to reduce the effect of noise. We will find a class of solutions which are optimal for noise attenuation. Among this class of solutions, we then choose the "best" $p(n)$, with which the channel coefficients can be most reliably computed.

4.1 Optimality criterion

Assume that the additive channel noise is white. Then

$$R_{\bar{x}}(0) = H_0 G^2 H_0^* + H_1 G^2 H_1^* + \sigma^2 I_N \quad (4.1)$$

where σ^2 is the noise variance. From (4.1), noise has contribution to only the diagonal entries of $R_{\bar{x}}(0)$. Thus the $L+1$ groups of equations in (3.2) remains the unchanged except that the $j=0$ group becomes

$$\Gamma_0[R_{\bar{x}}(0)] = M_0 f_0 + \sigma^2 b \quad (4.2)$$

where $b = \begin{bmatrix} 1 & \cdots & 1 \end{bmatrix}^T \in \mathbf{R}^N$. Since σ^2 is not known, f_0 can not be determined from (4.2). Instead the least squares solution $\hat{f}_0$ can be computed as

$$\begin{aligned} \hat{f}_0 &= (M_0^T M_0)^{-1} M_0^T \Gamma_j [R_{\bar{x}}(0)] \\ &= f_0 + \sigma^2 (M_0^T M_0)^{-1} M_0^T b \end{aligned}$$

To eliminate the effect of noise, we should choose the modulating sequence so that the null space of M_0^T contains b, or equivalently, we should make $\mathcal{R}(M_0)$ orthogonal to b. But this is impossible since all entries of b and columns of M_0 are positive. However, this suggests that we should choose $p(n)$ so that the angle between $\mathcal{R}(M_0)$ and b is as close to $\pi/2$ as possible. We note that each column of M_0 makes the same angle with b since M_0 is circulant. Let q be the first column of M_0, *i.e.*, $q = \begin{bmatrix} p(0)^2 & \cdots & p(N-1)^2 \end{bmatrix}^T$ and define

$$\gamma = \frac{q^T b}{\|q\|_2 \|b\|_2}.$$

We formulate the problem of selecting optimal $p(n)$ as the following optimization problem:

Minimize γ by choosing $p(0), p(1), \cdots, p(N-1)$

Subject to

$$\frac{1}{N} \sum_{n=0}^{N-1} p(n)^2 = 1 \;\text{ and }\; p(n)^2 \geq \delta > 0 \;\text{ for all }\; n$$

4.2 Optimal solution

It turns out that for a given positive $\delta < 1$, there are N optimal sequences:

$$|p(m)| = \sqrt{N(1-\delta)+\delta}, \;\; |p(n)| = \sqrt{\delta} \;\text{ for }\; n \neq m$$

where m is any integer between 0 and $N-1$. The optimal modulating sequences assume only two values with a single peak and the rest assuming the lower bound. In theory, where the peak occurs does not matter since each optimal choice gives the same γ and thus the same noise effect. In practice, it does matter since different choices define different M_0 and thus all the submatrices M_j. In particular, the condition numbers of the matrices $M_j^T M_j$ will be different. Making the condition number small is crucial in obtaining reliable least squares solutions in (3.3). Thus the proposed optimal choice of modulating sequence is one of the N optimal sequences that results in the smallest condition number of $M_j^T M_j$.

5 Simulation results

To illustrate the performance of the proposed channel identification method, we consider the five-tap channel used in [4]:
$h(0) = 0.459 + 0.265j,\; h(1) = -0.2078 - 0.12j,\; h(2) = -0.467 - 0.27j,\; h(3) = 0.095 + 0.055j,\; h(4) = -0.031 - 0.018j$.

The input source symbols are drawn from an i.i.d QPSK constellation. The additive channel noise is white with normal distribution. The channel identification performance is measured by the normalized root-mean-squares error (NRMSE) defined as

$$\text{NRMSE} := \frac{1}{\|h\|_2} \sqrt{\frac{1}{I} \sum_{i=1}^{I} \|\hat{h}^{(i)} - h\|_2^2}$$

where I is the number of Monte Carlo runs and $\hat{h}^{(i)}$is the estimate of the channel impulse response vector in the ith trial. For computation purpose, the scalar ambiguity is removed by a least squares fitting. The signal-to-noise ratio (SNR) is defined as

$$\text{SNR} := \sqrt{\frac{\frac{1}{N}\sum_{n=0}^{N-1} E|z(n)|^2}{E|v(n)|^2}}$$

For all simulations, $I = 100$.

Figure 5.1 shows the dependence of NRMSE on the number of samples used in the computation for different choice of peak value index m in the optimal modulating sequences. The choices $m = 0$ and $m = 1$ give the best condition number and thus yield smallest NRMSE. In this simulation, $N = 6, \delta = 0.5878$, and SNR $= 10$ dB.

Figure 5.2 and Figure 5.3 compare the performance of the proposed method with those of the one cycle subspace method [5] and the structured subspace method [4]. In this simulation, $N = 6$, $\delta = 0.5878$, and $m = 0$. In Figure 5.2, the SNR is fixed at 10 dB and in Figure 5.3 the number of samples is fixed at 1000. The results show that the proposed method gives better performance than the two subspace methods do.

Figure 5.4 shows the effect of channel order over-estimation. For each channel order upper bound $\hat{L}$, $4 \leq \hat{L} \leq 12$, the length of modulation sequence $N = \hat{L} + 2$. The SNR is fixed at 10 dB and the number of samples is 1000. The result shows that as the channel upper bound increased from 4 to 12, the NRMSE increases only 5 dB.

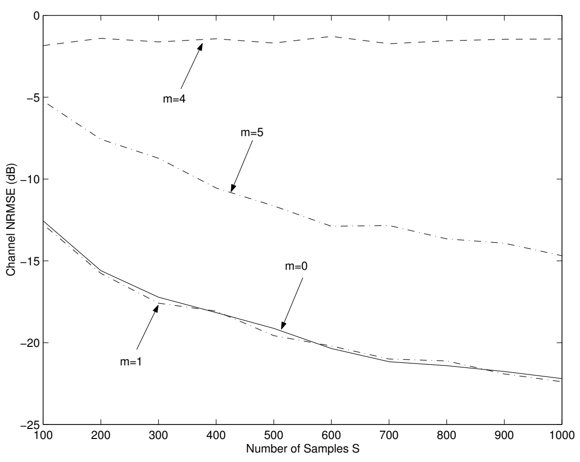

Figure 5.1: NRMSE for optimal $p(n)$ with different peak value index m

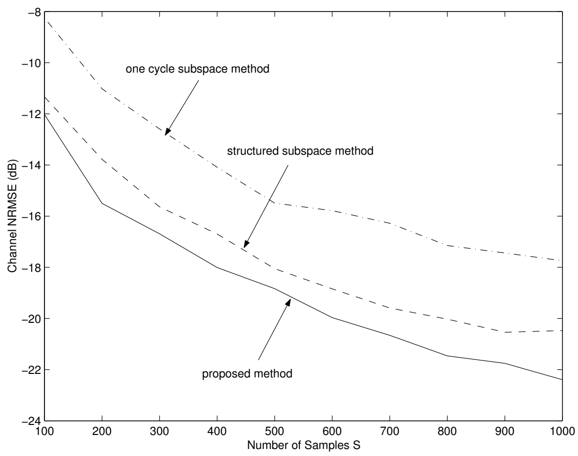

Figure 5.2: Comparison with subspace methods

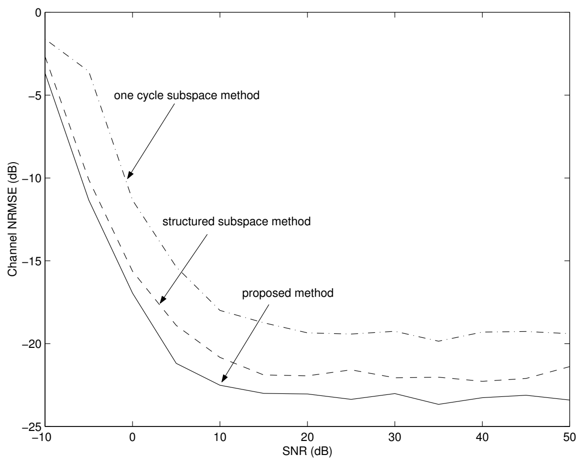

Figure 5.3: Comparison with subspace methods

6 Conclusions

We propose a method for blind identification of FIR channels with periodic modulation of source symbols. The time-domain formulation in terms of block signals is simple compared with existing frequency-domain approaches. The method exploits the linear relation between the products of channel coefficients and the autocorrelation matrix of the received signal as well as the decoupled structure of the resulting linear system of equations. The identifiability conditions so derived are particular simple: they depend on the modulating sequence alone. Indeed, with the proposed method, any FIR channel is identifiable with an appropriate choice of the periodic modulating sequence provided that the modulation period $N \geq$

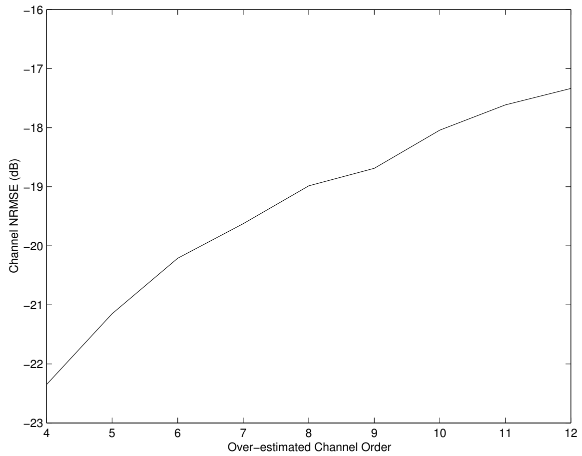

Figure 5.4: Robustness to channel order overestimation

$L+2$, where L is the channel order. In fact, *almost* all periodic modulating sequences yield the channel identifiable. The optimal modulating sequence selection problem formulated as one of minimizing the effects of channel noise and error in estimating the autocorrelation matrix is straightforward and easy to solve. The proposed optimal solution also results in a consistent channel estimate when the channel noise is white. Simulation results show that the method yields good performance: it compares favorably with existing subspace modulation-induced-cyclostationarity methods and it is robust with respect to channel order overestimation.

References

[1] L. Tong, G. Xu and T. Kailath, " Blind identification and equalization based on second-order statistics: A time-domain approach," *IEEE Trans. Inform. Theory*, Vol. 40, No. 3, pp. 340-349, 1994.

[2] E. Moulines, P. Duhamel, J. Cardoso and S. Mayargue, "Subspace methods for blind identification of multichannel FIR filters," *IEEE Trans. Signal Processing*, Vol. 43, No. 2, pp. 516-525, 1995.

[3] K. Abed-Meraim, E. Mouline, and P. Loubaton, "Prediction error method for second-order blind identification," *IEEE Trans. Signal Processing*, Vol. 43, No. 3, pp. 694-705, 1997.

[4] A. Chevreuil, E. Serpedin, P. Loubaton and G. B. Giannakis, " Blind channel identification and equalization using periodic modulation precoders: performance analysis," *IEEE Trans. Signal Processing*, Vol. 48, No. 6, pp. 1570-1586, 2000.

[5] E. Serpedin and G. B. Giannakis, " Blind channel identification and equalization with modulation-induced cyclostationarity," *IEEE Trans. Signal Processing*, Vol. 46, No. 7, pp. 1930-1944, 1998.

[6] M. Tsatsanis and G. B. Giannakis, " Transmitter induced cyclostationarity for blind channel equalization," *IEEE Trans. Signal Processing*, Vol. 45, No. 7, pp. 1785-1794, 1997.

[7] P. Ciblat, A. Chevreuil and P. Loubaton, " α-repetition/modulation and blind second order identification," *IEEE Trans. Signal Processing*, Vol. 48, No. 11, pp. 3153-3161, 2000.

[8] G. B. Giannakis, " Filteranks for blind channel identification and equalization," *IEEE Signal Processing Letters*, Vol. 4, No. 6, pp. 184-187, 1997.

[9] B. Proat and B. Friedlander, " Blind equalization of communication channel using high-order statistics," *IEEE Trans. Signal Processing*, Vol. 39, No. 2, pp. 522-526, 1991.