An Efficient and Effective Pilot Space-Time Adaptive Algorithm for Mobile Communication Systems

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Abstract. *In this paper we present a new adaptive space-time algorithm for mitigating the effects of CCI and ISI and minimizing the probability of error in mobile communication systems, and evaluate its performance for different mobile velocities. The proposed algorithm is computationally efficient and provides better performance than the conventional RLS algorithm.*

Keywords

Channel estimation, multipath fading channels, adaptive array signal processing, mobile-cellular radio communications.

1. Introduction

Intersymbol interference (ISI) and co-channel interference (CCI) are known to severely limit the capacity of wireless communications systems [1]. Adaptive antenna arrays, particularly the joint space-time adaptive processing techniques, provide effective means in combating both ISI and CCI and improving the system capacity and the communications quality [1-3]. In many systems known pilot symbols are inserted in each data frame and used by an adaptive algorithm to estimate the array response. For example, IS-54/136 TDMA cellular system uses $q = 14$ known pilot symbols in each user slot of $Q = 162$ symbols; in [2], [3] the standard LMS and RLS algorithms were used to estimate the array response based on this known pilot sequence in each frame. This approach may be subject to large estimation errors since the estimation is solely based on this short sequence of pilot symbols. There are several ways to improve the performance of the array. For instance, one might use decision feedback approach where decisions of transmitted symbols are fed back and used instead of pilots, when pilots are not available. One can also use a different algorithm during the data part of the slot, such as the blind constant modulus algorithm (CMA) [4] that does not require the use of pilot signals. The drawback with decision feedback is that decisions need to be correct; otherwise performance degradation is possible due to the *error propagation* problem. Blind algorithms, on the other hand, have slow convergence and may therefore have poor tracking capabilities. In this letter, we present a simple and effective space-time processing algorithm based on the recursive least squares (RLS) criteria, which we refer to as the *Phase recovery RLS (P-RLS) algorithm*. The algorithm has two modes of operation: standard RLS mode when training sequence is available, and a phase-sensing and correction mode when training sequence is not available. This algorithm provides better performance than the stand-alone standard RLS algorithm.

2. Problem Formulation

We consider a wireless cellular communication system with P mobile users transmitting signals, which are received by an antenna array at a base station. We denote the first user, $p = 1$, as the *desired user* and the remnant users as *interferers*. The transmitted equivalent bandpass signal from user p may be written as:

$$\tilde{s}_p(t) = a_p(t)\exp\left(j2\pi f_d t\right)\exp\left(j\theta_p(t)\right) \qquad (1)$$

where $a_p(t)$ and $\theta_p(t)$ are the envelope and the phase of the signal $s_p(t)$, respectively and f_d is the Doppler shift. The received signal $y_k(t)$ at the base station's reference antenna $k = 1$ is modeled as the convolution sum of the transmitted signals from each device and an impulse response $h_{p,m}$ from user p to the antenna, where the impulse response contains M components or taps. The received equivalent bandpass signal $y_1(t)$ can be written as:

$$\tilde{y}_1(t) = \sum_{p=1}^{P}\sum_{m=1}^{M} h_{p,m}\tilde{s}_p(t-\tau_{p,m})\exp\left(-j2\pi f_c \tau_{p,m}\right) + \tilde{n}_1(t) \qquad (2)$$

where f_c and $\tau_{p,m}$ are the carrier frequency and the delays, respectively. Here we have also added a noise term, $\tilde{n}_1(t)$, containing additive white Gaussian noise (AWGN).

The signal received at each antenna element will be the same with the exception of a phase shift that depends on the angle of arrival and the geometry of the antenna array. The difference in phase between the received signal at the antenna in the origin and the antenna positioned at the coordinate (x_k, y_k, z_k) is [5]:

$$\Delta\theta(p,k) = \beta x_k \cos\phi_p \sin\varphi_p + \beta y_k \sin\phi_p \sin\varphi_p + \beta z_k \cos\varphi_p \qquad (3)$$

where $\beta = 2\pi/\lambda$ is the phase propagation factor, λ the wavelength of the signal and (Φ_p, φ_p) the direction of arrival for the signal from user p. If the elevation angle φ_p represent the horizon ($\varphi_p = 90°$) and we consider an array with all elements distributed with equal distance Δx on the x-axis (i.e., $y_m = z_m = 0$), equation (3) can be re-written as

$$\Delta\theta(p,k) = \beta k \Delta x \cos\phi_p . \quad (4)$$

Equation (4) corresponds to the phase shift for a linear antenna array. For a circular antenna array with users and antennas on the plane $z_m = 0$ the phase shift (3) equals to

$$\Delta\theta(p,k) = \beta x_k \cos\phi_p + \beta y_k \sin\phi_p . \quad (5)$$

Taking this phase shift into account, the received signal at antenna k can be represented as

$$\tilde{y}_k(t) = \sum_{p=1}^{P} \Big\{ \exp\big(j\Delta\theta(p,k)\big) \cdot \left(\sum_{m=1}^{M} h_{p,m} \tilde{s}_p(t-\tau_{p,m}) \exp\big(-j2\pi f_c \tau_{p,m}\big) \right) \Big\} + \tilde{n}_k(t) \quad (6)$$

where $k = 1$ represents the antenna in the origin (i.e. the reference antenna).

The signal on each antenna is sampled and filtered by FIR-filters with adaptive coefficients, and the output signals from each FIR-filter are summed together, resulting in the so-called *adaptive beamformer* or *array*. The output from the beamformer at sample instant n is given by $\hat{s}_1(n) = \mathbf{w}^H(n)\mathbf{x}(n)$, where $\mathbf{x}(n)$ is the data vector. The objective of the beamformer is to continuously adjust the beamformer weight vector $\mathbf{w}(n)$ in order to minimize the ISI for the desired user and the CCI from the interfering users.

3. The Phase Recovery RLS Algorithm

There are several algorithms which can be used to adapt the array as explained in the introduction. In this letter, we present a low complexity solution which we refer to as the *Phase recovery RLS (P-RLS) algorithm*. The P-RLS may be divided into two stages of operation, in which the adaptation process is performed in different ways:

Initialization:

Initialize the parameters and variables of the RLS algorithm and the step size α in (10) below.

Stage A:
Training sequence is available (i.e., $n = 1, 2, \ldots, q$)

Use the standard RLS algorithm to update the weight vector and calculate the beamformer output $\hat{s}_1(n)$.

Stage B:
No training sequence available (i.e., $n = q+1, q+2, \ldots, Q$)

1. Calculate the output using old weights:

$$\hat{s}_1(n) = \mathbf{w}^H(n-1)\mathbf{x}(n) . \quad (7)$$

2. Calculate phase errors:

$$\psi_b(n) = \angle\hat{s}_1(n) - \left(\frac{\pi}{4} + b\frac{\pi}{2}\right), b = 0,1,\ldots,3 . \quad (8)$$

3. Find the minimum phase error:

$$\psi(n) = \psi_B(n),\ B = \arg \min_{b\in\{0,1,2,3\}} |\psi_b(n)| . \quad (9)$$

4. Update the *phase offset*:

$$\zeta(n) = \zeta(n-1) - \alpha\psi(n),\ \zeta(q) = 0 . \quad (10)$$

5. Calculate a new weight vector:

$$\mathbf{w}(n) = \mathbf{w}(q)\exp\big(-j\zeta(n)\big) . \quad (11)$$

6. Recalculate the output using new weight vector:

$$\hat{s}_1(n) = \mathbf{w}^H(n)\mathbf{x}(n) . \quad (12)$$

The main idea behind the algorithm is that, assuming that the weights converged in stage A, the channel will change during stage B (due to noise and interference) and the constellation rotates, hence the message points become less precise (phase-shifted). The task of the algorithm in stage B is to focus on the rotation of the constellation by monitoring and adjusting for the variations of the phase, thereby improving the overall performance. This approach is computationally efficient because of its low complexity.

Three parameters have to be specified in the initialization of the P-RLS algorithm:

- The forgetting factor λ. This parameter is used in the RLS to ensure that past data are forgotten in order to able the algorithm to follow the variations of the channel. When $\lambda = 1$ the memory of the RLS algorithm is infinite. The memory is finite for values below 1.
- The inverse correlation matrix $\mathbf{P}$. This matrix is used in the RLS update and should be initialized as $\chi\mathbf{I}$, where χ is the so-called regularization parameter and $\mathbf{I}$ is the identity matrix. In practice, χ should be assigned a small value when the signal-to-noise ratio (SNR) is high and a large value when the SNR is low.
- The step size α. This parameter is a part of the update of the phase offset $\zeta(n)$ in equation (10) and determines how much impact the current phase error $\psi(n)$ has on $\zeta(n)$.

The reader is referred to [6] for more information on the RLS algorithm and its initialization.

4. Simulation Results

We have evaluated the proposed P-RLS algorithm by means of Monte Carlo computer simulations and compared its performance to the conventional standard RLS (S-RLS) algorithm (updating using Stage A only) and the continuous RLS (C-RLS) algorithm (adapts the weights during the whole slots, using all data as pilots).

The simulations were carried out on an IS-136 Cellular system, where QPSK modulation is used and each slot consists of $Q = 162$ symbols, where the first $q = 14$ are pilots. The carrier frequency was set to 850 MHz and the sampling frequency of the simulator was 24300 Hz. An 8 element circular array was used with half wavelength spacing and omnidirectional antenna elements. We consider one desired user at 0°and seven interferers at 55°, 80°, 140°, 182°, 221°, 265° and 323°, respectively. All users are transmitting with a power of 2. The users are fully uncorrelated. The channel impulse response for the desired user is $h_{1,m}=\delta(m)+0.25\delta(m\text{-}1)$. and for the interferers $h_{p,m}=0.16\delta(m)$, $p\neq1$, where $\delta(0)=1$ and $\delta(m)=0$, $m\neq0$. The power for the AWGN was set to 0.05 resulting in a signal-to-noise plus interference ratio (SNIR) at each antenna of approximately 9.5 dB. The forgetting factor λ was fixed to 0.99 in all simulations and the step size α was chosen as 0.1. Eight coefficients were used for each antenna in the equalizer.

Figs. 1 and 2 show the symbol error rate (SER) and mean squared error (MSE) at different mobile velocities for the various algorithms. It is clearly evident from these figures that the P-RLS algorithm provides better performance (lower SER and MSE values) compared to S-RLS and C-RLS, respectively, for mobile velocities between 20 and 75 km/h; for higher velocities the performance is virtually the same.

5. Conclusions

In this paper we presented a computationally efficient and effective space-time processing algorithm based on the recursive least squares (RLS) criteria, which we refer to as the Phase recovery RLS (P-RLS) algorithm. Simulation results have shown that the algorithm provides improved performance for low and medium velocities in comparison to the standard RLS algorithm. Future work comprises investigating the convergence of the algorithms, the performance for varying SNR's, tuning the parameters of the algorithm to see the possibility of getting better performance at high speeds and comparing with the linear antenna array configuration. Also, the symbol error ratio should be simulated and compared. The CPU-time demands of the P-RLS algorithm should be computed and compared to a decision feedback equalizer.

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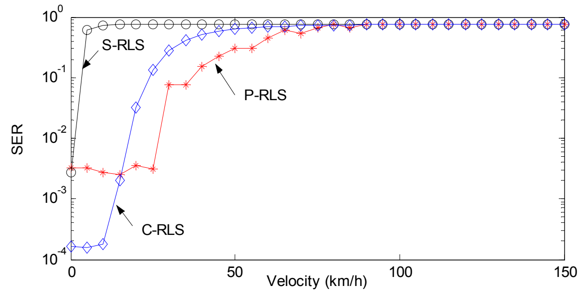

Fig. 1. Symbol error rate versus mobile velocity for the different tested algorithms.

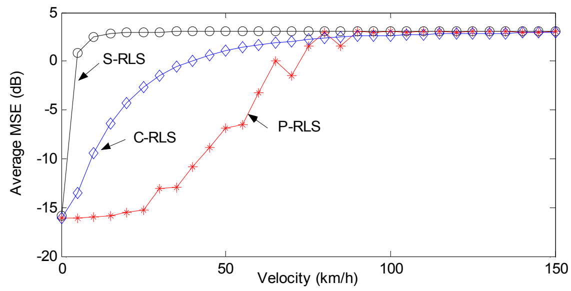

Fig. 2. Mean squared error versus mobile velocity for the different tested algorithms.

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