An Improved PN Code Acquisition Algorithm Using Adaptive Filter

이동훈, 전경훈, 김정창
포항공과대학교 전자컴퓨터공학과
cheun@postech.ac.kr

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Donghoon Lee, Kyungwhoon Cheun, and Jeongchang Kim
Division of Electrical and Computer Engineering, Pohang University of Science and Technology
cheun@postech.ac.kr

Abstract

A hybrid code acquisition system based on a least-mean-square adaptive filter, interpreted as a channel estimator, is proposed for direct-sequence spread-spectrum systems under frequency selective Rayleigh fading channels. Closed form expressions for the detection and false alarm probabilities are derived. Compared to the previous channel equalizer based system, the proposed system achieves smaller mean acquisition times and is more robust to adaptation step-size selection.

I. Introduction

There have been numerous attempts to improve the PN code acquisition performance of direct-sequence spread-spectrum (DSSS) under frequency selective multipath fading channels. In [1], an optimal detection rule based on the maximum-likelihood criterion was derived for parallel acquisition systems and in [2], chip level post-detection integration method was suggested to improve the performance of serial matched filter acquisition systems under multipath channels. On a different note, a novel partially parallel hybrid acquisition system employing a least-mean-square (LMS) adaptive equalizer was proposed in [3]. Although specific PN phase search strategies proposed in [1]-[3] are different, a commonality is that some form of windowing is employed in order to integrate the signal power spread over the multipath components and seek diversity.

In this paper, we propose a hybrid acquisition system based on an LMS adaptive filter interpreted as *a channel estimator* and analyze its performance. In contrast to the equalizer based system [3], the proposed structure takes the local PN code as input to the adaptive filter and the received signal as the reference (training) signal (see Fig. 1). Owing to this role reversal, the mean convergence rate of the tap-weight vectors and the value of the step-size required for convergence become independent of the received signal-to-noise ratio (SNR) and the given channel response. This makes the proposed system more robust to SNR and channel variations compared to the equalizer based algorithm of [3]. Also, the fact that no multiplication operations are required for tap-weight updates and only one received sample per chip is required results in low hardware complexity.

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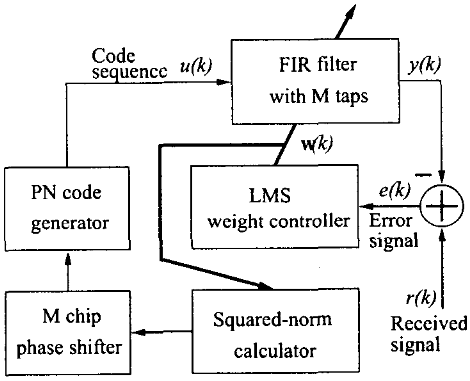

Fig. 1. Proposed code acquisition system.

II. Proposed code acquisition system

The proposed code acquisition system is shown in Fig. 1 is based on an LMS adaptive filter updating the tap weight vector according to the stochastic gradient algorithm [5], attempting to minimize the mean-squared-error (MSE) between the filter output and the reference signal. Note that roles played by the locally generated PN code and the received signal are reversed to those for the equalizer based acquisition system of [3]. The present and past $M-1$ samples of the complex PN code $u(k)$ form the input vector and the filter output is computed as $y(k) = \mathbf{w}(k)^H\mathbf{u}(k)$ where $\mathbf{w}(k) = [w_0(k)\ w_1(k)\ \cdots\ w_{M-1}(k)]^T$ and $\mathbf{u}(k) = [u(k)\ u(k-1)\ \cdots\ u(k-M+1)]^T$ denote the filter tap weight and input vectors at time k, $u(k)$ are assumed to be i.i.d. (independent and identically distributed) random variables with real and imaginary parts taking on the values of $\pm 1/\sqrt{2}$ with equal probability and $\mathbf{a}^T$, $\mathbf{a}^H$ denote the transpose and the Hermitian transpose of the complex vector $\mathbf{a}$, respectively. The reference signal for the adaptive filter consists of the complex baseband signal sampled at the