Real-Number DFT Codes on a Fading Channel

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Abstract —
The utilization of real-number DFT codes for a multiplicative channel is introduced in this paper. By the proposed encoding procedure, some redundancies can be added into the transmitted data. With these redundancies, syndromes for the parameters of a fading channel can be obtained from the received data. The decoding algorithm for real-number DFT codes can be used to calculate the fading parameters with these syndromes.

I. Introduction

In 1981, Marshall first defined error control codes for real or complex data and suggested that real-number codes could have applications similar to those of Reed-Solomon codes. Wolf, with a different view, took real-number codes as a new technique for solving signal processing problems such as impulsive noise cancellation in information transmission.

A common feature of previous studies is that the channel error model is assumed to be additive. In this paper, the real-number decoding method for multiplicative channel error model (which corresponds to the situation of transmitting over a fading channel in practice) will be investigated.

II. Encoding and Decoding Scheme for a Fading Channel

Usually the effect of a fading channel is modeled by a slowly varying component multiplying the transmitted signal, that is

$$r_i = y_i \cdot e_i + n_i \tag{1}$$

where y_i is the transmitted signal, e_i the multiplicative parameters of a fading channel, n_i the background noise, and r_i the received signal. In a block coding scheme, we can also assume that the index i is in the range of $0, 1, 2, \ldots, N-1$.

A multiplication can be transformed into an addition by taking logarithm. However, since the signals under consideration are assumed to be complex, complex logarithm function are required. It can be easily derived from eqn. (1) that

$$\log_c r_i = \log_c y_i + \log_c e_i + \hat{n}_i \tag{2}$$

where $\hat{n}_i = \log_c(1 + \frac{n_i}{y_i \cdot e_i})$. It should noted that when $n_i \ll y_i \cdot e_i$, $\hat{n}_i$ will approach 0. Since e_i is slowly varying, both e_i and $\log_c e_i$ can be viewed as a lowpass signal. Therefore, it is reasonable to assume that $\log_c e_i$ can be obtained from the sum of some unknown low frequency components E_k, that is

$$\log_c e_i = \sum_{l=1}^{\nu} E_{k_l} \cdot e^{2\pi \frac{ik_l}{N}} \tag{3}$$

where k_l is the location for a nonzero frequency components, and E_{k_l} is the magnitude of that component. Now suppose that y_i is encoded as

$$y_i = \begin{cases} 1 & i = 0, 1, \ldots, N-K-1 \\ x_i & i = N-K, N-K+1, \ldots, N-1 \end{cases} \tag{4}$$

The first $N-K$ equations in eqn. (2) become the desired syndromes

$$S_i = \log_c r_i = \log_c e_i + \hat{n}_i \tag{5}$$

These noisy syndromes can readily be input to some decoding algorithms for the DFT codes. to compute E_k, provided that the number of nonzero terms of E_k in eqn. (5). After E_k is computed, an estimation of y_i can then be derived. In this way, one can estimate out the channel parameters e_i and, at the same time, the transmitted data x_i.