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Receiver simplification in synchronous communication systems: simulation study

By

Mohd. Fadzil Ain* Farid Ghani** Mutamed Khatib *
Syed Idris Syed Hassan*

Abstract:

This paper presents a simulation study for code-division multiplex system over time varying channels. The scheme investigated here uses a precoding technique in the transmitter depending on the channel information, so that no need for any signal processing in the receiver rather than testing against a certain threshold. This coding technique minimizes the effect of intersymbol interference (ISI). The data itself is transmitted in the form of blocks of data symbols separated by blocks of no signal. The appropriate length of each data block depends on the channel parameters expected and the allowed complexity. The impulse response is required to be known at the transmitter which is the requirement for all systems that employ coding at the transmitter.

Keywords:

Precoding, signal processing, CDMA and channel estimation

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* School of Electrical & Electronic Eng., University of Science Malaysia (USM)
** University Malaysia Perlis (UNIMAP)

1. Introduction:

Recently, as a result of the importance of maintaining low cost and complexity in wireless systems, researchers have started developing techniques that move necessary signal processing from the mobile unit to the base station, so that the interference cancellation is done at the transmitter and just simple linear processing is done at the mobile unit [1, 2]. This technique is called precoding or pre-equalization.
Reynolds, et al. have proposed a precoding technique that simplifies the receiver [3]. They used a sophisticated channel estimation method to get information about the channel elements, i.e., the delayed version of the spreading waveform, and the complex channel fading gain for each user in each path. The original information can be retrieved at the mobile unit using a matched filter. Vojčić, et al. and Esmailzadeh, et al. suggested precoding techniques for synchronous Code Division Multiple Access CDMA over Additive White Gaussian Noise AWGN channel [4, 5]. In their design, they used a RAKE receiver. The disadvantage for RAKE reception is that it is sensitive to channel mismatch and its performance is generally inferior to MMSE or decorrelator based multi-user interference rejection [3].
We have proposed an effective technique to reduce the complexity of the receiver by using an interesting precoding technique, but that approach was depending on mathematical representation only, and we didn't focus on the effect of the system parameters on the performance of the system [6]. The system described there is a K-user multi-path CDMA system with time disruptive channels, intersymbol interference (ISI), which is introduced when the delay spread is large, is removed by inserting guard intervals between symbols to insure that the delayed version of the pulse will not affect the other pulses from other paths. This will affect the total performance of the system by increasing the bit error rate. When the multi-path delay spread is less than the symbol interval, ISI can be neglected because the delayed pulse will not affect the pervious or the next pulse from the other paths [7]. The precoding technique can achieve both portable unit simplicity and ISI reduction.
An important assumption for this precoding technique is that the transmitter must have information about all channels between it and active receivers. This information can be obtained from receivers via feedback channels [8]. Another important requirement is that the multi-path channel is slow, i.e., that it remains constant over the block of pre-coded bits. Though, the length of the precoding block can be adjusted to match the channel dynamics.
In this paper, we have built a simulation model of [6] to prove the mathematical model, and to study the effect of each variable. Also, we have managed to study the effect of each variable, such as the block length, the guardband length, the channel characteristics and the Signal to Noise Ratio SNR. This paper is organized as follows: in Section 2, we present the system model. The simulation results are compared with those obtained

mathematically in [6] and presented in Section 3. Finally, Section 4 presents the conclusions of our study.
The practical applications of transmitter precoding can be found in wireless local loop, wireless LAN's and indoor communications in general, as well as any other wireless scenario where the precoding block size can be made sufficiently small so that the channel appears slow [4].

2. System Model:

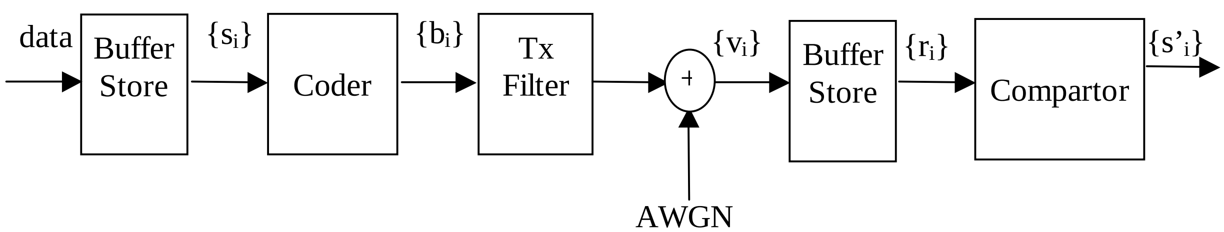

Figure (1): *The downlink of the precoding system*

The system considered in this paper is shown in figure (1). It has been studied mathematically in [6], and here, we have built a simulation model to prove the mathematical model, and to study the effect of each variable.
The sampled impulse-response of the baseband channel is given by the $(g+1)$ component row vector:

$$y_i = y(iT) = y_o \quad y_1 \quad \cdots \quad y_g \tag{1}$$

where: $g+1$ is the length of the channel
$y_o \neq 0$, $y_i = 0$ for i < 0 and i > g.

The signal at the input to the transmitter is a sequence of statistically independent and equally likely data. The buffer-store at the transmitter holds *m* successive elements $\mathrm{S} = [s_1 \quad s_2 \quad \cdots \quad s_m]$. The precoder accepts the input vector S and codes it to form the $1\times n$ signal vector B, which is the convolution between the input vector S and the $m\times n$ coder matrix:

$$\mathrm{F} = \left(\mathrm{D}\mathrm{D}^T\right)^{-1}\mathrm{D} \tag{2}$$

where D is the $m\times n$ matrix of rank *m* whose ith row is:

$$\mathrm{D}_i = \Big[\overbrace{0 \;\cdots\; 0}^{i-1}\;\; \overbrace{y_g \;\; y_{g-1} \;\cdots\; y_o}^{g+1}\;\; \overbrace{0 \;\cdots\; 0}^{m-i}\Big] \qquad (3)$$

After coding, the block will be fed to the baseband channel $y(t)$ which is assumed be either time invariant or varies slowly with time.

White Gaussian noise, with zero mean and variance σ^2, is assumed to be added to the data signal at the output of the transmission path, giving the Gaussian waveform $w(t)$ added to the data signal. At the receiver, the received signal, corresponding to a single group of *m* signal elements, will normally be a sequence of $n+g$ non-zero sample values preceded and followed by zero sample values. The sequence of these $n+g$ sample values in the absence of noise is:

$$v_i = \sum_{j=1}^{n} b_i y_{i-j} \qquad i = 1, 2, \ldots, n+g \qquad (4)$$

where: b_i is the coded data and y_i is the channel elements.

It can be easily shown that the first *g* components of V are dependent in part on the preceding received group of m signal-elements, and the last *g* components of V are dependent in part on the following received group of m elements [6].

Thus there is ISI from adjacent received groups of elements in both the first and the last *g* components of V. However, the central *m* components of V depend only on the corresponding transmitted group of *m* elements, and can therefore be used for the detection of these elements with no ISI from adjacent groups. This will be done in the buffer store at the receiver side. The receiver can now detect the values of the signal elements by comparing the corresponding received samples with the appropriate thresholds.

Assume that the possible values of s_i are equally likely and that the mean square value of S is equal to the number of bits per element. Suppose that the *m* vectors $\{\mathrm{D}_i\}$ have unit length. Since there are *m* *k*-level signal elements in a group, the vector S has k^m possible values each corresponding to a different combination of the *m* *k*-level signal elements. So, the vector B whose components are the values of the corresponding impulses fed to the baseband channel, has k^m possible values. If *e* is the total energy of all the k^m values of the vector B, then in order to make the transmitted signal energy per bit equal to unity, the transmitted signal must be divided by:

$$q = \left(e / mk^m\right)^{1/2} \qquad (5)$$

The m sample values of the received signal from which the corresponding $\{s_i\}$ are detected, are the components of the vector:

$$R' = \frac{BD^T}{q} + W \tag{6}$$

Then, the m sample values which are the components of the vector R', must first be multiplied by the factor q to give the m-component vector:

$$R = qR' = BD^T + qW = S + U \tag{7}$$

where U is an m component row vector whose components are sample independent Gaussian random variables with zero mean and variance $\eta^2 = q^2\sigma^2$. Thus, the tolerance to noise of the system is determined by the value of η^2.

When there is no signal distortion, $(DD^T)^{-1}$ is an identity matrix. Under these conditions, q = 1, so that $\eta^2 = \sigma^2$, and the signal to noise ratio $(SNR)_{ND}$ is given by:

$$(SNR)_{ND} = \frac{E_b}{\sigma^2} \tag{8}$$

where E_b is the energy per bit.

While the signal to noise ratio in the real channel (with noise) is:

$$(SNR)_C = \frac{E_b}{\eta^2\sigma^2} \tag{9}$$

In order to understand the behavior of the system, we calculated the signal to noise ratio relative to no distortion channel:

$$(SNR)_{relative} = \frac{(SNR)_C}{(SNR)_{ND}} = \frac{1}{\eta^2} \tag{10}$$

or in dB:

$$(SNR)_{relative} = 10\log_{10}\left(\frac{1}{\eta^2}\right) \tag{11}$$

2. Simulation Results:

We have built a Matlab program to simulate the whole process. The input of the program is a random data $\{+1 \quad -1\}$ (one million bit is used as input data), and the data will be processed in blocks of different length against different channels.
Also, Matlab built-in generator is used to add AWGN to the transmitted signal with different values of SNR. All these result have been compared with other result of the same system using mathematical model presented in [6].
Figure (2) shows a comparison between the mathematical results obtained in [6], and the output of the Matlab simulation program for $m = 8$ and $Y = [0.5 \quad 1 \quad 0.5]$. The results were similar, so, now we have proved that the model presented earlier is correct. Also, now, we can study the effect of each effective variable of the system, such as the block length, the guardband length, the channel characteristics and the SNR.

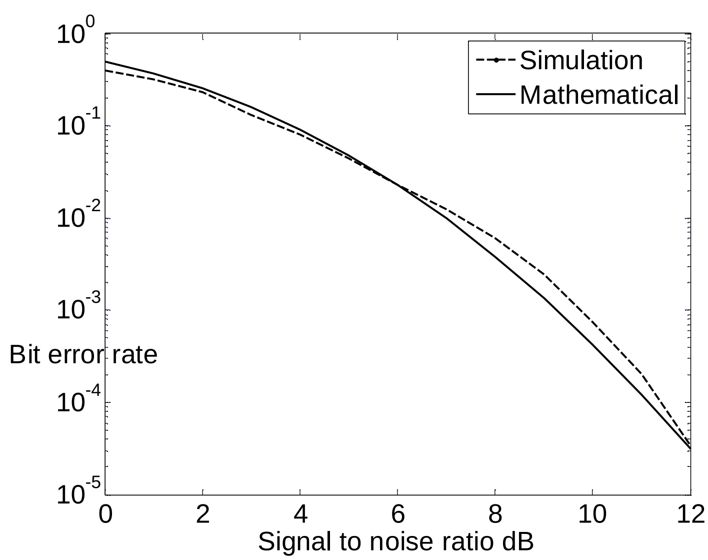

Figure (2): *Comparison between simulation and mathematical results.*

The proposed system depends on transmitting the data in blocks. The source of these data may be serial, i.e. from the same source, or even parallel from different sources. So, the length on the block itself is expected to have a great effect on the performance.
Figure (3) shows the probability of error of the system for different values of SNR using three different length of the block, i.e. $m = 4$, $m = 8$ and $m = 16$, the channel here is assumed to have impulse response $Y = [0.5 \quad 1 \quad 0.5]$.
It is clear from the figure that increasing the block length will reduce the performance of the system and the probability of error becomes worse.
This result is expected because increasing the block length will increase the variance of

the precoder matrix which maximizes the noise variance at the output of the system.

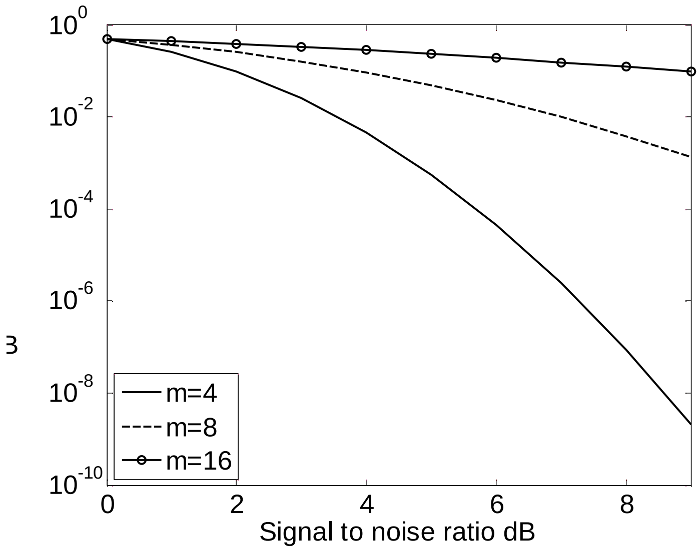

Figure (3): *Effect of block length on the system performance.*

Also, increasing the block length will increase the intersymbol interference inside the block itself (ISI between the blocks is removed by using guardband). Theoretically, the best results will be for $m=1$, which means transmitting each bit separately, and this is not accepted because in this case, each bit will use g bits as a guardband, and this is a great loss in the bandwidth. So, one must find an optimum solution for the block length. Comparing the results with other systems, we can say that $m=8$ is a good choice.

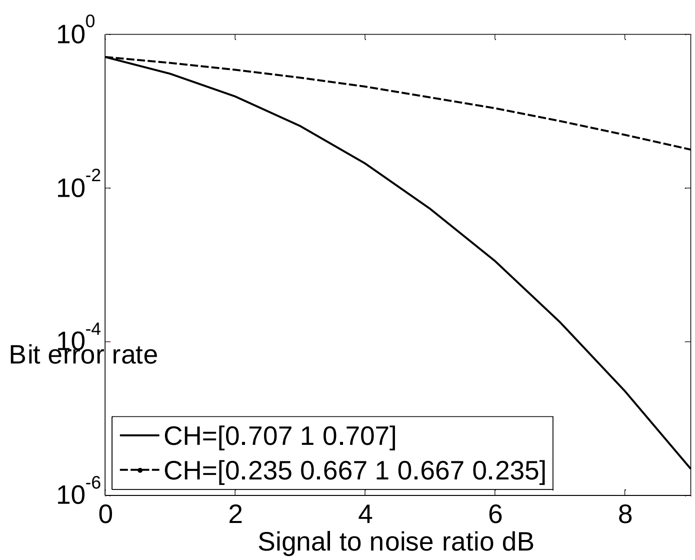

Figure (4): *Effect of channel length.*

In figure (4), we studied the effect of the channel length on the performance of the system. Here, we used two different channels with different lengths $g=2$ and $g=4$, but with the same norm values, as shown in table (1).
Although increasing the channel length will give the system more guardband bits to reduce ISI between blocks, but it will increase the variance η_F^2 of the $m\times n$ precoder matrix F too, affecting an increase in the noise variance η_T^2 at the receiver.

$$\eta_T^2 = \eta_F^2 \sigma^2 \tag{12}$$

where: η_F^2 is the variance of the precoder
σ^2 is the variance of the AWGN.

Table (1): *norm values of the channel vectors*

Channel vector	Channel after normalization	Norm
$[0.235 \quad 0.667 \quad 1 \quad 0.667 \quad 0.235]$	$[0.166 \quad 0.472 \quad 0.707 \quad 0.472 \quad 0.166]$	1.4143
$[0.707 \quad 1 \quad 0.707]$	$[0.5 \quad 0.707 \quad 0.5]$	1.4141
$[0.5 \quad 1 \quad -0.5]$	$[0.408 \quad 0.816 \quad -0.408]$	1.2247
$[0.5 \quad 1 \quad 0.5]$	$[0.408 \quad 0.816 \quad 0.408]$	1.2247
$[1 \quad 2 \quad 1]$	$[0.408 \quad 0.816 \quad 0.408]$	2.4495

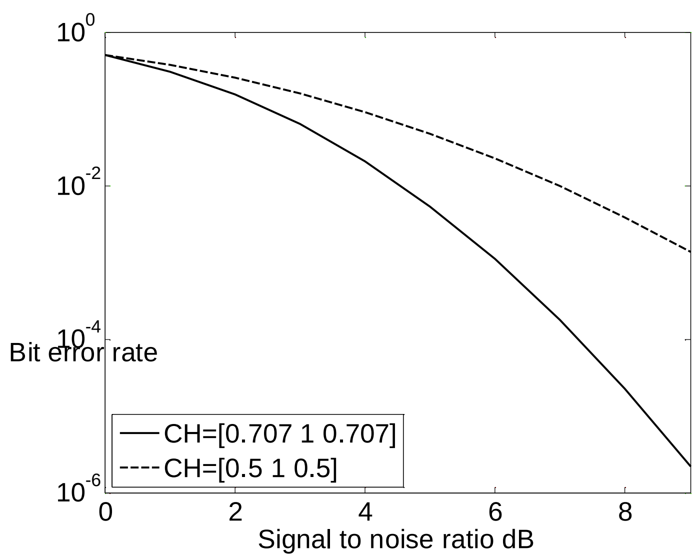

Figure (5): *Effect of channel variance.*

Then we tested the effect of the channel norm value on the performance of the system as shown in figure (5). Here, we used two channels that differ in variance, but similar in length. It is clear that the channels with higher variance (norm) have better performance than those with lower variance. Note that the variance of the channel has no direct effect on the system, it affects the variance of the precoder, and that affects the total performance of the system as shown in equation (13)

$$R = S\left(DD^T\right)^{-1} DD^T = S \tag{13}$$

Now, let us make a look on the effect of the channel symmetry. In figure (6), we used typical channels, but we reversed the sign of one of them at one side, i.e. $[0.5 \quad 1 \quad 0.5]$ and $[0.5 \quad 1 \quad -0.5]$.

The effect was great, asymmetric channels gave much better performance than symmetric one. It is not strange because the symmetric channel increases the coder variance four times more than the asymmetric one.

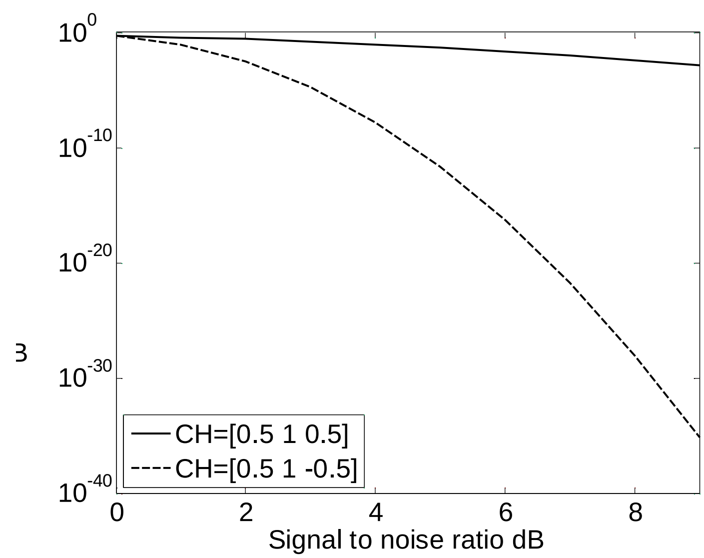

Figure (6): *Effect of channel symmetry.*

The amplitude of the channel will have no effect because if we use typical channels, but different in amplitude, that will lead us to two channels with the same normalized vector, which means the same performance.
To clarify this point, let us make a look on the equation of the system, the channel itself has no effect of the output signal; it affects only the precoder matrix.
Figure (7) is an example.

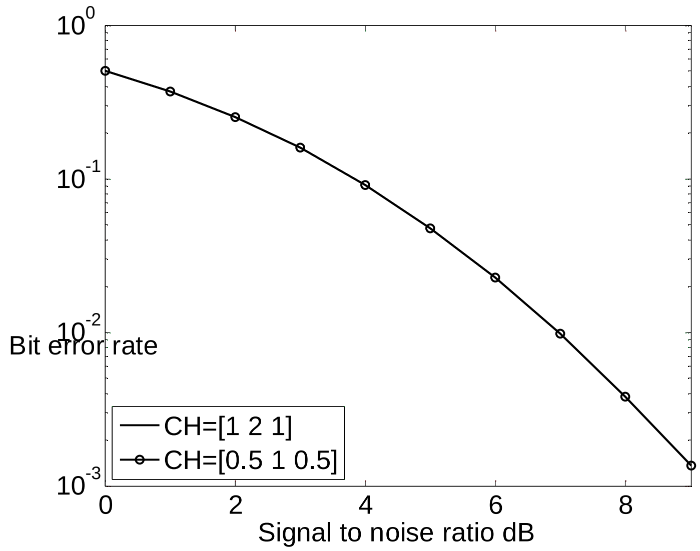

Figure (7): *Effect of channel amplitude.*

4. Conclusions:

In this paper, we have made a simulation study for a promising coding strategy at the transmitter for the downlink of a synchronous multi-user communication system in fading multi-path environment. The coding is such that no processing of the received signal is needed except testing these against appropriate threshold levels. This makes the receiver quite simple at the expense of the common transmitter and hence can be used as an advantage in situations where a single transmitter is feeding many receivers. The performance of the system is tested for different system parameters to see what exactly affect the system. It is assumed that the transmitter has prior knowledge of the multi-path channels. There are a number of techniques that are available for channel estimation and these are available in the published literature.

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References:

[1] T. Rappaport, *Wireless Communication*, 2nd Edition, Prentice-Hall, 1996.

[2] M. Khatib, *Efficient Modulation with OFDM for Next-Generation Mobile Communications Systems*. M.Sc. dissertation, Jordan University of Science and Technology, 2003

[3] D. Reynolds, A. Host-Madsen and X. Wang, *Adaptive Transmitter Precoding for Time Division Duplex CDMA in Fading Multipath Channels: Strategy and Analysis*, EURASIP Journal of Applied Signal Processing, Vol.2002, No.12, P.1365-1376, 2002.

[4] B. Vojčić and W. Jang, *Transmitter precoding in synchronous multiuser communications*, IEEE Transactions on Communications, Vol.46, No.10, P.1346-1355, 1998.

[5] R. Esmailzadeh, E. Sourour, and M. Nakagawa, *Prerake diversity combining in time-division duplex CDMA mobile communications*, IEEE Transactions on Vehicular Technology, Vol.48, No.3, P.795-801, 1999.

[6] F. Ghani, M. Ain and M. Khatib, *Transmitter Precoding in Synchronous Multiuser Communications*, The 8th Seminar on Intelligent Technology and its Applications, Surabaya, Indonesia, P. 1-5, 9-10 May 2007.

[7] J. Proakis, *Digital communications*, 3rd Edition, McGraw Hill, 1995.

[8] D. Borah, *Estimation of Frequency-Selective CDMA Channels with Large Possible Delay and Doppler Spreads*, IEEE Transactions on Vehicular Technology, Vol.55, No.4, P.1126-1136, 2006.

Nomenclatures:

m … Block length before pre-coding
g … Guardband length
n … Block length after pre-coding
S … Transmitted signal
F … Coding matrix
C … Channel matrix
R … Received signal