Performance Improvement of Synchronous Multiuser Communication Systems by Using Process Sharing Techniques

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Abstract: - This paper presents new techniques that improve the performance of Code Division Multiple Access (CDMA) system for transmission of digital data over time varying channels such as high frequency mobile channels. The main ideas presented here are to share the equalization process between the transmitter and the receiver of the system with a certain ratio that maximize the signal to noise ratio (SNR). The channel characteristics should be known at the transmitter and receiver, it is the requirement for all systems that employ coding at the transmitter.

Key-Words:- Signal processing, communications, sharing, coding, CDMA.

1 Introduction

Recently, mobile communication services are penetrating into our society at an explosive growth rate. All the current cellular communication systems have adopted digital technology. The demand for a variety of wideband services such as high-speed internet access and video/high-quality image transmission will continuously increase. CDMA have been designed to support wideband services at high data rates with the same quality as fixed networks [1].

Wireless communication systems are playing a major role in providing portable access to future information services. The demand for new services to the internet and advanced image and video applications presents key technical challenges: multimedia information access requires high-bandwidth and low-latency network connections to many users, mobility requires adaptation to time varying channel conditions; and portability imposes severe constraints on receiver size and power consumption [1, 2].

Block Transmission System has recently been proposed for transmission of digital data over time varying and time dispersive channels such as mobile radio channels [3, 4, 5, 6]. In this system, each data block is detected as a unit in contrast to the recursive symbol-by-symbol detection approach usually employed.

In order to reduce the size of the receiver, Ghani, *et al.* [7] have proposed a system that moves the equalization process from the receiver to the transmitter which leads to no processing in the receiver, except of testing against threshold level. This pre-coding system [7] is shown in Figure 1. The signal at the input to the transmitter is a sequence of *k*-level element values $\{s_i\}$, where k = 2, 4, 8,… and the elements $\{s_i\}$ are considered to be statistically independent and equally likely to have any of the possible values. The buffer-store at the input to the transmitter holds *m* successive element values $\{s_i\}$.

In the pre-coder, the *m* $\{s_i\}$ are converted into the corresponding *m* coded signal-elements. The pre-coder performs a linear transformation on the *m* $\{s_i\}$ to generate the corresponding sequence of impulses that is fed to the baseband channel $y(t)$ which is assumed to be either time invariant or varies slowly with time [8].

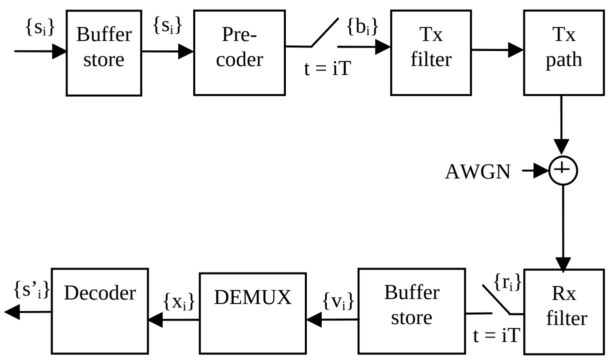

Fig.1: Pre-coding system model [7]

Additive White Gaussian Noise (AWGN), with zero mean and variance σ^2 [8], is assumed to be added to the data signal at the output of the transmission path, so, the waveform $w(t)$ is added to the transmitted signal through the channel **C**.
The sampled impulse-response of the baseband channel in Figure 1 is given by $(g+1)$ component vector [3, 4, 10]:

$$y_i = y(iT) = y_o \quad y_1 \quad \dots \quad y_g \qquad (1)$$

where $y_o \neq 0$, and $y_i = 0$ for i < 0 and i > g.
The received waveform $r(t)$ at the output of the baseband channel is sampled at the time instants $\{iT\}$, for all integers $\{i\}$.
The $\{r_i\}$ are fed to the buffer-store that contains two separate stores. While one of these holds a set of the received $\{r_i\}$ for a detection process, the other receives the next set of $\{r_i\}$ in preparation for the next detection process.
A group of *m* multiplexed signal-elements are detected simultaneously in a single detection process, from the set of $\{r_i\}$ that depends only on these elements. The receiver uses the knowledge of $\{y_i\}$ in the detection of the *m* element values $\{s_i\}$ from the received samples $\{r_i\}$. A period of *nT* seconds is available for the detection process, where $n = m + g$ [7].
The decoder and demultiplexer in Figure 1 together retrieve from the appropriate set of received $\{r_i\}$ the *m* estimates $\{x_i\}$ of the *m* element-values $\{s_i\}$ in a received group of elements. Each x_i is an unbiased estimate of the corresponding s_i such that $x_i = s_i + u_i$, where u_i is a zero mean Gaussian random variable [8]. The detector detects each s_i by testing the corresponding x_i against appropriate thresholds. The detected value of s_i is designated as $s_i^{'}$ [7].
This paper presents new methods that depend on the idea of sharing the equalization process between the transmitter and the receiver in order to improve the system performance and is organized as follows: in Section 2, we proposed the system models, both systems will be discussed in details in this section. The sharing processes and a review for other block code systems are presented in Section 3. In Section 4, numerical results are presented and the systems performances are compared with those where all the processing are carried out at the receiver or at the transmitter [3, 7]. Finally, Section 5 presents the conclusions and recommendations of our study.
Notation: All bold faces variables in this paper denote vectors and matrices.

2 Systems Models

2.1 First system

Figure 2 shows the basic model of the sharing system considered. The signal at the input to the transmitter is a sequence of *k*-level element values $\{s_i\}$, where k = 2, 4, 8,… and the $\{s_i\}$ being statistically independent and equally likely to have any of the possible values. The buffer-store at the input to the transmitter holds *m* successive element values $\{s_i\}$ to form the $1 \times m$ data vector **S**. The transmitter's processor, $\mathbf{F_1}$, is an $m \times n$ matrix, so, in this processor, **S** is converted into the corresponding *n* elements vector **B** which is the convolution between **S** and $\mathbf{F_1}$ that is fed to the baseband channel.

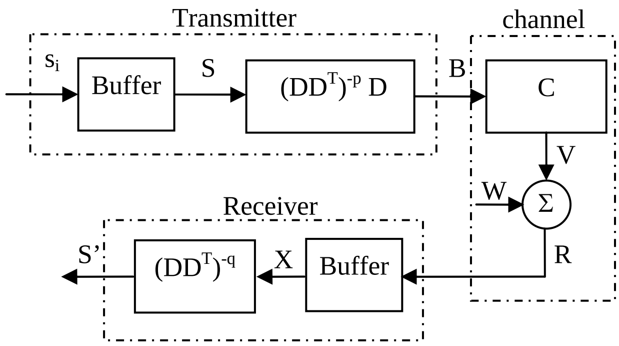

Fig. 2: First system model

The value of *n* is chosen to be the Algebraic sum of the length of the input data vector *m* and the channel's length *g* [3, 10].
The linear baseband channel has an impulse response y(t) and includes all transmitter and receiver filters used for pulse shaping and linear modulation and demodulation [8]. White Gaussian noise is introduced at the output of the transmission path. The noise has zero mean and a two sided power spectral density of σ^2, giving the zero mean Gaussian waveform w(t) at the output of the receiver filter. The sampled impulse-response of the baseband channel in figure 2 is given by the $(g+1)$ component vector $y_i = y_o \quad y_1 \quad \dots \quad y_g$ where $y_o \neq 0$, and $y_i = 0$ for i < 0 and i > g [8].
Due to transmission in blocks of *n* elements, the baseband channel can be represented in matrix form. From now on, the channel will be represented by the $n \times (n+g)$ matrix **C** and its i[th] row is [3]:

$$\mathbf{C_i} = \overbrace{0 \quad \dots \quad 0}^{i-1} \quad \overbrace{y_o \quad y_1 \quad \dots \quad y_g}^{g+1} \quad \overbrace{0 \quad \dots \quad 0}^{n-i} \qquad (2)$$

The output of the channel will be $1\times(n+g)$ vector **V,** which is the convolution between **B** and **C**, i.e.

$$\mathbf{V} = \mathbf{BC} \qquad (3)$$

where $\mathbf{V} = \lfloor v_1 \quad v_2 \quad \cdots \quad v_{n+g} \rfloor$ is the $1\times(n+g)$ received signal

The received vector **R** will be the vector **V** with the $1\times(n+g)$ AWGN vector **W** added on. i.e.

$$\mathbf{R} = \mathbf{V} + \mathbf{W} \qquad (4)$$

The receiver buffer store chooses the central *m* component of the vector **R** to form the vector **X**, which will be fed to the receiver's processor matrix $\mathbf{F_2}$ [7].

In the sharing process studied here, the transmitter's processor operates as a pre-coding scheme on the transmitted signal, and the receiver's processor completes the detection process on the received vector to obtain the detected value of **S**. In each case, it has an exact prior knowledge of **Y**, derived from the knowledge of the sampled impulse response of the channel. In case of time-varying channel, the rate of change in **Y** is assumed to be negligible over the duration of a received group of *m* signal elements, and sufficiently slow to enable **Y** to be correctly estimated from the received data signal [4, 6, 9].

2.2 Second system

The basic model of this system is shown in Figure 3. The signal at the input to the transmitter is the same as the previous system. The transmitter's processor, $\mathbf{F_1}$, here is an $m\times m$ matrix, so, now **S** is converted into the corresponding *m* elements vector **B** instead of *n* elements in the previous system.

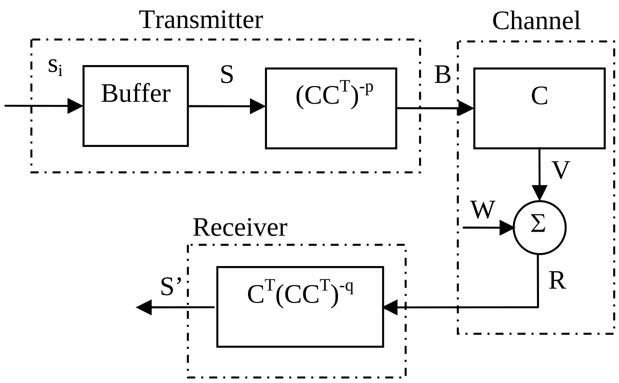

Fig.3: Second system model

The data will be transmitted in a channel with the same characteristics as mentioned before, but the matrix size will differ due to the transmission of a block with *m* elements now, the baseband channel will be represented by the $m\times n$ matrix **C** and its i[th] row is [3]:

$$\mathbf{C_i} = \overbrace{0 \ \ldots \ 0}^{i-1} \ \overbrace{y_o \quad y_1 \ \ldots \ y_g}^{g+1} \ \overbrace{0 \ \ldots \ 0}^{m-i} \qquad (5)$$

The output of the channel will be the $1\times n$ vector **V,** which is the convolution between **B** and **C**, i.e.

$$\mathbf{V} = \mathbf{BC} \qquad (6)$$

where $\mathbf{V} = [v_1 \quad v_2 \quad \cdots \quad v_n]$ is the $1\times n$ received signal

The received vector **R** will be the vector **V** with the $1\times n$ AWGN vector **W** added on. i.e [8].

$$\mathbf{R} = \mathbf{V} + \mathbf{W} \qquad (7)$$

This vector **R** will be fed to the $n\times m$ receiver's processor matrix $\mathbf{F_2}$, so, the data at the output of the receiver will be the same as the $1\times m$ transmitted vector.

3 Design and Analysis of the Sharing Process

The block code system in [4] is an equalizer at the receiver with the equation $y^T(yy^T)^{-1}$ which ensures that the total equation of the system from the input to the output in the absence of noise is:

$$\mathbf{R} = \mathbf{SYY^T(YY^T)^{-1}} = \mathbf{S} \qquad (8)$$

while the precoding system in [7] is an equalizer at the transmitter with the equation $(DD^T)^{-1}D$, where D is the $m\times n$ matrix of rank *m* whose related to the channel matrix **C** and its i[th] row is [7]:

$$\mathbf{D_i} = \overbrace{0 \ \ldots \ 0}^{i-1} \ \overbrace{y_g \quad y_{g-1} \ \ldots \ y_o}^{g+1} \ \overbrace{0 \ \ldots \ 0}^{m-i} \qquad (9)$$

so, the total equation of the system in the absence of noise:

$$\mathbf{R} = \mathbf{S}(\mathbf{DD}^T)^{-1}\mathbf{DD}^T = \mathbf{S} \qquad (10)$$

In the first system proposed in this paper, we split the process given in [7] between the transmitter and the receiver, so that the transmitter's share of the process is the $m\times n$ matrix:

$$\mathbf{F}_1 = (\mathbf{DD}^T)^{-p}\mathbf{D} \qquad (11)$$

and the receiver's share is the $m\times m$ matrix:

$$\mathbf{F}_2 = (\mathbf{DD}^T)^{-q} \qquad (12)$$

where:

$$0 \le p \le 1 \qquad (13)$$

$$q = 1 - p \qquad (14)$$

So, the total equation from the input to the output is:

$$\mathbf{R} = \mathbf{SF}_1\mathbf{CF}_2 \qquad (15)$$

$$= \mathbf{S}(\mathbf{DD}^T)^{-p}\mathbf{DD}^T(\mathbf{DD}^T)^{-q} = \mathbf{S} \qquad (16)$$

Here, only the central *m* components of the vector **V**, i.e., $v_{g+1} \quad v_{g+2} \quad \ldots \quad v_{g+m}$, will be taken into consideration as they give information about the transmitted data, so, we can assume that $\mathbf{C} = \mathbf{D}^T$ [7].

In absence of AWGN, it is clear from the equation above that there is no need for any further processing after the receiver's share of the equalization process, but when noise is present [8],

$$\mathbf{R} = (\mathbf{S}\mathbf{F}_1\mathbf{C} + \mathbf{W})\mathbf{F}_2 \quad (17)$$

$$= \mathbf{S}\mathbf{F}_1\mathbf{C}\mathbf{F}_2 + \tilde{\mathbf{W}} = \mathbf{S} + \tilde{\mathbf{W}} \quad (18)$$

Thus the detector can now detect the values of the signal elements by comparing the corresponding $\{r_i\}$ with the appropriate thresholds.

In the second system, the transmitter's share of the process is the $m \times m$ matrix:

$$\mathbf{F}_1 = (\mathbf{C}\mathbf{C}^T)^{-p} \quad (19)$$

and the receiver's share of the process is the $n \times m$ matrix:

$$\mathbf{F}_2 = \mathbf{C}^T(\mathbf{C}\mathbf{C}^T)^{-q} \quad (20)$$

where:

$$0 \le p \le 1 \quad (21)$$

and:

$$q = 1 - p \quad (22)$$

So, the total equation of the system from the input to the output is:

$$\mathbf{R} = \mathbf{S}\mathbf{F}_1\mathbf{C}\mathbf{F}_2 \quad (23)$$

$$= \mathbf{S}(\mathbf{C}\mathbf{C}^T)^{-p}\mathbf{C}\mathbf{C}^T(\mathbf{C}\mathbf{C}^T)^{-q} = \mathbf{S} \quad (24)$$

and when noise is present,

$$\mathbf{R} = (\mathbf{S}\mathbf{F}_1\mathbf{C} + \mathbf{W})\mathbf{F}_2 \quad (25)$$

$$= \mathbf{S}\mathbf{F}_1\mathbf{C}\mathbf{F}_2 + \tilde{\mathbf{W}} = \mathbf{S} + \tilde{\mathbf{W}} \quad (26)$$

Also, the detector will use a comparator to detect the signal elements.

4 Performance Evaluation

As discussed above, the channel's impulse response has no effect on the total performance of the system, so that, the only effective element is the AWGN. In order to study the performance of the system, we must find the tolerance to noise from the transmitter's and the receiver's shares.

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 $\{\mathbf{D}_i\}$ (or $\{C_i\}$) have unit length. Since there are *m* *k*-level signal elements in a group, the vector **S** has k^m possible values each corresponds 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 input data vector **S**, then in order to make the transmitted signal energy per bit equal to unity, the transmitted signal must be divided by [8]:

$$\ell = \sqrt{\frac{e}{mk^m}} \quad (27)$$

The *m* sampled values of the received signal from where the corresponding s_i are detected, are the components of:

$$\mathbf{R}' = \frac{\mathbf{B}\mathbf{D}^T}{\ell} + \mathbf{W} \quad (28)$$

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

$$\mathbf{R} = \ell\mathbf{R}' = \mathbf{B}\mathbf{D}^T + \ell\mathbf{W} = \mathbf{B}\mathbf{D}^T + \mathbf{U} \quad (29)$$

where $\mathbf{U}$ is an *m* component row vector whose components are sample independent Gaussian random variables with zero mean and variance $\eta_T^2 = \ell^2\sigma^2$. Thus, the tolerance to noise of the transmitter's share is determined by η_T^2.

In the receiver, the total variance of the matrix $\mathbf{F_2}$ can be calculated by:

$$\eta_R^2 = \frac{1}{m}\sum_{j=1}^{m}\sum_{i=1}^{m}(f_2)_{ij}^2 \quad (30)$$

So, the total tolerance to noise from both the transmitter's and the receiver's shares is:

$$\eta = \sqrt{\ell^2\eta_R^2} = \ell\eta_R \quad (31)$$

In case of no distortion, the signal to noise ratio $(SNR)_{ND}$ is given by:

$$(SNR)_{ND} = \frac{E_b}{\sigma^2} \quad (32)$$

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

$$(SNR)_C = \frac{E_b}{\eta^2\sigma^2} \quad (33)$$

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

$$(SNR)_{relative} = \frac{(SNR)_C}{(SNR)_{ND}} = \frac{1}{\eta^2} \quad (34)$$

or in dB:

$$(SNR)_{relative} = 10\log_{10}\left(\frac{1}{\eta^2}\right) \quad (35)$$

Tables 1 and 2 show the numerical results of the previous equation for a given channel $Y = [1 \quad 2 \quad 1]$ after being normalized, for both systems. Figures 4 and 5 show the effect of the sharing factor *p* of the signal to noise ratio relative to no distortion channel.

P	ℓ	η_R^2	$\ell^2\eta_R^2$	$\ell\eta_R$	SNR$_{relative}$
0.00	1.00	1798.7	1798.7	42.41	-32.55
0.10	0.96	699.40	643.38	25.37	-28.09
0.20	0.93	273.63	237.22	15.40	-23.75
0.30	0.92	108.14	91.67	9.57	-19.62
0.40	0.94	43.47	38.21	6.18	-15.82
0.50	1.00	18.00	18.00	4.24	-12.55
0.60	1.14	7.85	10.25	3.20	-10.11
0.70	1.42	3.73	7.56	2.75	-8.79
0.75	1.64	2.70	7.27	2.70	-8.62
0.80	1.93	2.03	7.56	2.75	-8.79
0.90	2.80	1.31	10.25	3.20	-10.11
1.00	4.24	1.00	18.00	4.24	-12.55

Table 1: Numerical results of system 1

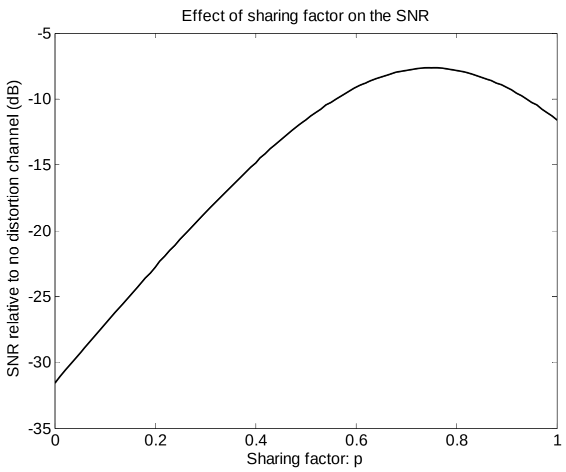

Fig. 4: Effect of factor p on the SNR for system 1

P	ℓ	η_R^2	$\ell^2\eta_R^2$	$\ell\eta_R$	SNR$_{relative}$
0.00	1.00	18.00	18.00	4.24	-12.55
0.10	1.14	7.85	10.25	3.20	-10.11
0.20	1.42	3.73	7.56	2.75	-8.79
0.25	1.64	2.70	7.27	2.70	-8.62
0.30	1.93	2.03	7.56	2.75	-8.79
0.40	2.80	1.31	10.25	3.20	-10.11
0.50	4.24	1.00	18.00	4.24	-12.55
0.60	6.59	0.88	38.21	6.18	-15.82
0.70	10.40	0.85	91.67	9.57	-19.62
0.80	16.54	0.87	237.22	15.40	-23.75
0.90	26.45	0.92	643.38	25.37	-28.09
1.00	42.41	1.00	1798.70	42.41	-32.55

Table 2: Numerical results of system 2

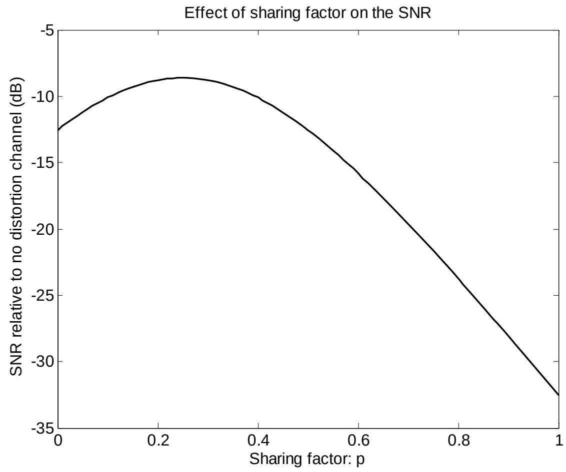

Fig. 5: Effect of factor p on the SNR for system 2

It is clear that system 1 has the best performance at $p = 0.75$, while it is at $p = 0.25$ for system 2. Both systems give the same improvement with about 4dB gain better than the pre-coding system $(p = 1)$.

The bit error rate for the systems described in this paper is shown in Figure 6. The two systems have the same performance, so that only one of them appears in the figure. The sharing systems improved the performance approximately 4 dB which is a good improvement in badly scattered channels. The signal elements are binary antipodal having possible values as +1 or -1. There are 8 elements in a group and these are equally likely to have any of the two values. The sampled impulse response of the channel is $\{y_i\} = [1 \quad 2 \quad 1]$. It has a second order null in the frequency domain and introduces severe signal (amplitude) distortion [9]. For comparison, the bit error rate of the block linear equalizer and the pre-coding system are also given.

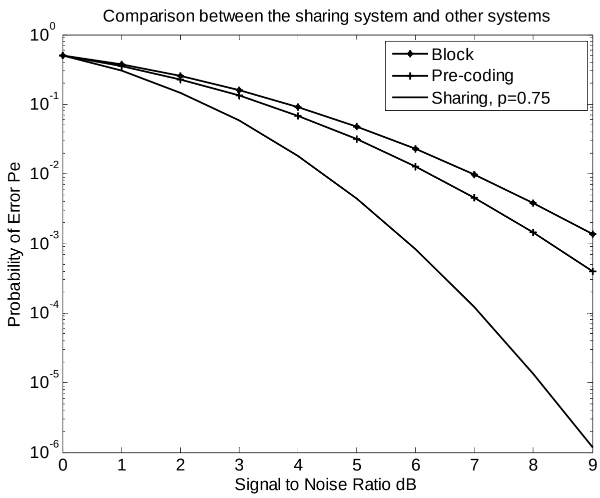

Fig. 6: Probability of error for the sharing system

5 Conclusion

In this paper, we have developed sharing strategies between the transmitter and the receiver for the downlink of a synchronous multiuser communication system in fading multipath environment. The sharing is such that 75% of the equalization is done at the transmitter, while 25% of the process is done at the receiver for the first system. The second system has 25% in the transmitter and 75% in the receiver. This results in a 4 dB enhancement in comparison with the precoding system, where all the equalization process is done at the transmitter and leaves the receiver quite simple. In applications where the transmitted signal faces a badly scattering channel, this 4dB can make a difference in the total performance of the system, so that one can accept a little processing at the receiver in order to gain 4dB enhancement. It is assumed that the transmitter has prior knowledge of the multipath channels. There are a number of techniques that are available for channel estimation and available in the published literature.

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