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Comparative Analysis of ZF and MMSE Receiver for Multicode MC-CDMA Downlink Channels

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Abstract: A new wireless communication system denoted as Multi-Code Multi-Carrier CDMA (MC-MC CDMA), which is the combination of Multi-Code CDMA and Multi-Carrier CDMA, is analyzed in this paper. This system can satisfy multi-rate services using multi-code schemes and multi-carrier services used for high rate transmission. The equalization techniques used in this work are Zero Forcing (ZF) and Minimum mean square error (MMSE) and their performances are compared on the basis of bit error rate (BER). The BER is analyzed with respect to Signal to Noise ratio (SNR). QPSK modulation scheme is implemented for the analysis. The comparative analysis is done through software simulation.

Keywords: **Multi code, multi carrier, CDMA, Rayleigh channel, zero forcing, MMSE, QPSK, Signal to Noise Ratio (SNR), Symbol error ratio (SER).**

I. INTRODUCTION

Multi-Code CDMA and Multi-Carrier CDMA have attracted a lot of attention from researchers due to their perceived high rate transmission capability. In Multi-Code CDMA, researchers have investigated the systems performance in different fading channel [1] and suggested many schemes to improve the performance [2]. In Multi-Code CDMA, the input data streams are first split into several sub-streams in parallel and then orthogonal codes are multiplied for each sub-stream.

In Multi-Carrier CDMA [3], the input data streams are first split into several sub-streams in parallel, like in Multi-Code CDMA and then modulate several subcarriers with each sub-stream before transmitting the signals. Similarly with Multi-Code CDMA, Multi-Carrier CDMA is analyzed with different fading channels, and researchers have suggested schemes to improve the system performance [4], and [5].

Spread spectrum techniques were originally proposed to allow secure communication, by spreading the signal over a wide bandwidth, allowing the signal power spectral density to be reduced. This is achieved by transmitting a higher frequency pseudo-noise sequence in place of a single modulated symbol[6-9].

This signal looks almost like noise and because it is wideband narrowband interference has little effect. The signal can be detected by correlating with the pseudo-noise sequence. In multi-path channels, the multi-path diversity can be exploited by using a channel matched (rake) receiver, giving the optimal performance (for a single user).

By using a number of different pseudo noise sequences, multiple users can transmit simultaneously using the same bandwidth. In all practical cases, at the receiver, all the users' pseudo-noise sequence will not be orthogonal and therefore the capacity of the system will be limited by this multiple access interference.

The results show that the proposed MCMC- CDMA system clearly performs both single code multicarrier CDMA (MC-CDMA) and single-carrier multi-code CDMA by allowing the system to have a Filter bank Transceiver .This indicates that the Transceivers based on filter banks for MC-MC-CDMA system should be more seriously considered for the next generation cellular systems [10-12].

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Also the adaptive equalization techniques have compensated the time dispersion in the channel and thereby increasing the efficiency of data transmission. Many researchers have inclined towards the various processing techniques over the last few years [6,7]. But due to the simplicity, linear equalization techniques have attracted a lot, as they are not optimal in a maximum likelihood sense.

ZF and MMSE are the two key equalization techniques due to their superior features over other equalization techniques. Even though they are not optimal, the MMSE receiver satisfies an alternative criterion, i.e., it minimizes the mean squared error (MSE) and ZF is known to eliminate the interference completely [8-10]. The performance of ZF and MMSE has been extensively analyzed for micro diversity systems so far where there is a communication between co-located diversity antenna at the base station and the distributed users [10-12]. But not much research has been done on macro diversity system where both transmit and receive antennas are widely separated.

The lack of research over macro diversity systems is due to the complexity of its channel matrix. In micro diversity, Wishart form is used in the classical models and Kronecker correlation matrix. However in macro diversity case, Wishart assumption is not followed, which makes its analytical work extremely difficult. Therefore only few results are available in macro diversity case [13, 14].

In this paper, ZF and MMSE equalization techniques are implemented over MCMC-CDMA and their performances are compared. The rest of the paper is laid out as follows. Section II describes the preliminary background of MCMC-CDMA systems. Section III provides preliminary results of ZF and MMSE which will be used throughout the paper. The simulation study is presented in section IV and section V concludes the paper.

II. HISTORICAL BACKGROUND & ADVANCED TECHNIQUE

a) Multicode MC-CDMA Model

Recently a new proposal for a system based on a combination of CDMA and OFDM has gained increasing attention in the research community. This system is called the Multi-Carrier CDMA (MC-CDMA) system and it combines the advantages offered by both OFDM and CDMA. This section describes the basic architecture and the advantages of this system. A novel multi-code system has been proposed in to support variable data rates [2]. In this scheme, each user has a set of M codes called the sequence set. The system is an M-ary modulation where a code sequence represents a sequence of $\log_2 M$ bits. The size of the sequence set depends on the required data rate. In the normal case, the set size is 2, i.e., there are two sequences in the set, one to represent a '0' and one to represent a '1'. When the data rate is to be made L times the standard data rate, the sequence set is made of size 2^L and each sequence of L bits is mapped to one of the 2^L code sequences.

The system model is depicted in Fig.1. It consists of transmitter, receiver and fading channel. The transmitter part converts input data stream $d_k(t)$ of k^{th} user, into J parallel sub streams $d_{kj}(t)$ which is coded by an orthogonal signal $a_j(t)$. The resulting signal $b_k(t)$, called super stream, is serial-to-parallel (S/P) converted again and spread in frequency domain by $c_k(t)$. Each of these branches then modulates one of the H orthogonal subcarriers and the results are summed.

At the receiver part, the received signal $r(t)$ is first demodulated by locally generated carrier, despread by the PN sequence and then P/S converted; his output is then despread again by each orthogonal code for multicode component in order to recover sub stream before correlation. Finally, the sub streams are recovered from the correlated data.

III. FUNDAMENTAL OF EQUALIZATION

Equalization compensates for inter symbol interference (ISI) created by multi path within time dispersive channels. If the modulation bandwidth exceeds the coherence bandwidth of the radio channel, ISI occurs and modulation pulses are spread in time into adjacent symbols. Equalizer must be adaptive since the channel generally unknown and time varying [2]. In a broad sense, the term equalization can be used to describe any signal processing operation that minimizes ISI [1]. An equalizer is usually implemented at base band or at IF in a receiver. Since the baseband complex envelope expression can be used to represent baseband waveform, the channel response, demodulation signal, and adaptive equalizer algorithm are usually simulated and implemented at baseband [1].

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A) Adaptive algorithm

The so-called adaptive filter, is the use of the result of the filter parameters a moment ago, automatically adjust the filter parameters of the present moment, to adapt to the unknown signal and noise, or over time changing statistical properties, in order to achieve optimal filtering [1]. Adaptive filter has "self-regulation" and "tracking" capacities. Adaptive filter can be divided into linear and nonlinear adaptive filter. Non-linear adaptive filter has more signal processing capabilities. However, due to the non-linear adaptive filter more complicated calculations, the actual use is still the linear adaptive filter [2]. As shown in Figure 1.

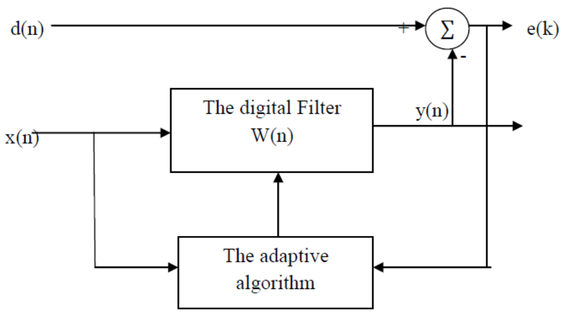

Fig 1. Adaptive Filter Scheme

Figure 1 represents the general adaptive filtering display: digital filter carries on filtering on the input signal $x(n)$, produces output signal $y(n)$. Adaptive algorithm adjusts the filter coefficient included in the vector $w(n)$, in order to let the error signal $e(n)$ to be the smallest. Error signal is the difference of useful signal $d(n)$ and the filter output $y(n)$. Therefore, adaptive filter automatically carry on a design based on the characteristic of the input signal $x(n)$ and the useful signal $d(n)$ [4]. Using this method, adaptive filter can be adapted to the environment set by these signals. When the environment changes, filter through a new set of factors, adjusts for new features[3] .The most important properties of adaptive filter is that it can work effective in unknown environment, and to track the input signal of time-varying characteristics [1].

Adaptive filter has a wide application range used in communications, control and many other systems. Filter out an increase noise usually means that the contaminated signal through the filter aimed to curb noise and signal relatively unchanged. Fixed filter designers assume that the signal characteristics of the statistical computing environment are fully known which is based on the prior knowledge of the signal and noise. However, in most cases it is very difficult to meet the conditions. Adaptive filter operates through the observation of the existing signal to understand statistical properties and adjust parameters automatically to change their performance. Therefore its design does not require of the prior knowledge of signal and noise characteristics. Adaptive filter is used for the cancellation of the noise component which is overlapped with unrelated signal in the same frequency range [1].

B) Zero Forcing Algorithm

Zero Forcing Equalizer is a linear equalization algorithm used in communication systems which inverts the frequency response of the channel. This algorithm was proposed by Robert Lucky. The Zero-Forcing Equalizer applies the inverse of the channel to the received signal, to restore the signal before the channel. The name Zero forcing corresponds to bringing down the Inter Symbol Interference (ISI) to zero in a noise free case. This will be useful when ISI is more predominant when comparing to the noise. By assuming $N_T = N_R$ and H is a full rank square matrix, we can calculate the covariance matrix [35] of the effected noise as:

$$Y = HX + N$$

Where,

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$$Y = \text{Received Symbol Matrix}$$
$$H = \text{Channel matrix}$$
$$X = \text{Transmitted symbol Matrix}$$
$$N = \text{Noise matrix}$$

To solve for X , we need to find a matrix W which satisfies $WH = I$. The zero forcing detectors for meeting this constraint is given by

$$W = \left(H^H H\right)^{-1} H^H$$

where

$$W = \text{Equalization Matrix}$$
$$H = \text{Channel Matrix}$$

This matrix is known as the Pseudo inverse for the general $m \times n$ matrix. Zero forcing equalizer tries to null out the interfering terms when performing the equalization. While doing so, there can be amplification of noise. Hence Zero forcing equalizer is not the best possible equalizer. However, it is simple and reasonably easy to implement.

C) Minimum Mean Square Error

A Minimum Mean Square Error (MMSE) estimator describes the approach which minimizes the mean square error (MSE), which is a common measure of estimator quality. The main feature of MMSE equalizer, is that it does not usually eliminate ISI completely but , minimizes the total power of the noise and ISI components in the output.

For a system defined by

$$Y = HX + N$$

The Minimum mean square error approach tries to find a coefficient W which minimizes the criterion

$$E\left\{\left[W_{y-x}\right]\left[W_{y-x}\right]^H\right\}$$

To solve for x, we need to find a matrix W which satisfies $WH = I$. The Minimum Mean Square Error (MMSE) detector for meeting this constraint is given by,

$$W = \left[H^H H + NI\right]^{-1} H^H$$

This matrix is known as the Pseudo inverse for the general $m \times n$ matrix.

IV. SIMULATION RESULTS

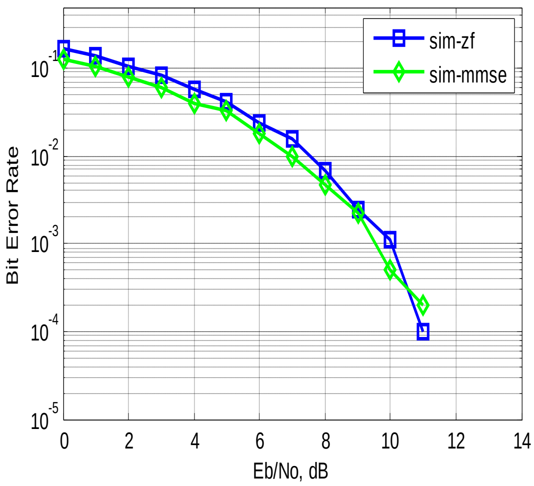

Fig 2. BER performance of MMSE and ZF for MC-MC-CDMA

Simulation is performed to verify the effectiveness of equalization technique over Multi code MCCDMA receivers. The effect of zero forcing equalizer is compared with the effect of MMSE for the respective system. The SER performance is evaluated for ZF and MMSE case with respect to Signal to Noise Ratio (SNR). The performance is reflected from the figures1and 2. The SER is always seems to be on a bit higher side for MMSE case as compared to that for ZF case.

V. CONCLUSION

With the design of the receiving adaptive filters, we can minimize the effects of Inter-symbol Interference, Multipath Interference and Additive Interference and thereby deliver the digital data to its destination with the smallest error rate possible. Hence we can improve the efficiency of a communication system. This script shows the BER performance of adaptive equalizers based on ZF and MMSE in a Rayleigh channel. The simulation results are achieved using in MATLAB environment. From simulation graph shown in results, we can see that as the Signal to Noise ratio increases the Bit Error Rate of ZF and MMSE are decreases. We clearly analyze that the Bit Error Rate (BER) of ZF is more compared to MMSE algorithm for same Signal to Noise Ratio (SNR). Hence we can conclude that the performance of MMSE algorithm is better than ZF algorithm.

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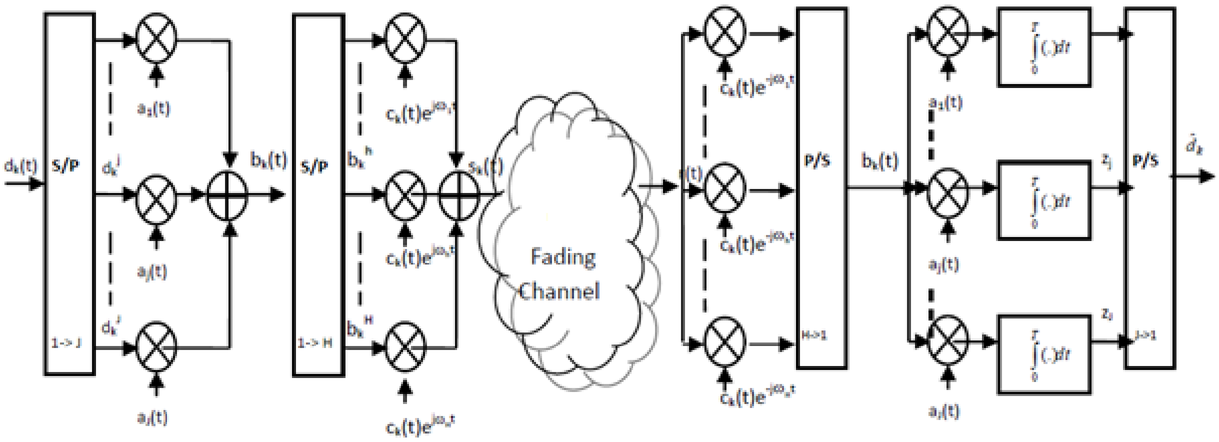

Fig 3.Transmitter/Receiver system model for MC-MC-CDMA