Selection Based Successive Interference Cancellation for Multicode Multicarrier CDMA Transceiver

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Abstract- A Selection based Interference algorithm for efficient decoding of MC-MC-DS-CDMA Transceiver with Multipath Rayleigh fading is presented in this paper.The Interference Cancellation algorithm is capable of increasing signal strength and reducing probability of error using two stages of decoding namely , Selection phase and Threshold decoding phase .The numerical results have shown that effective decoding with respect to frequency increases SNR levels.

Key-Words: - MC –MC DS CDMA (Multicode Multicarrier Direct Sequence CDMA),SIC(Succesive Interference Cancellation) SNR(Signal to Noise Ratio)

1. Introduction

The conventional Multicarrier- Code Division Multiple Access Direct Sequence CDMA system is affected by Inter Symbol Interference, Multi-user Interference, Multiple Access Interference when the number of users increases and system performance degrades as there is increased complexity in the circuit. The proposed Transceiver belongs to the class of overlapping-block transmission for a fixed set of transmitting or receiving filters, the design problem of Maximizing SIR is a great importance. The algorithms are developed such that the signals are preserved from Multipath fading, ISI and other interferences that arise due to channel noise. A Multiple Successive Interference Cancellation (MUSIC) Transceiver provides more flexibility in evaluation of system performance and maintains the signal at high SNR , based on the algorithms developed. For Multipath fading channels, the transceiver performs very well and at the same time the system complexity reduces as compared to conventional CDMA systems.

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. 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 MC-MC-CDMA system clearly performs both single-code multicarrier CDMA (MC-CDMA) and single-carrier multi-code CDMA by allowing the system to have a Filterbank Transceiver .This indicates that the Transceivers based on filterbanks for MC-MC-CDMA system should be more seriously considered for the next generation cellular systems.

In the receiver, when the received power of the desired and the interfered signal is nearly the same, the probability of detection error becomes high, because there are several combinations of signal candidates that result in similar replicas with the minimum Euclidean distance. In such a case, it is difficult or the CCI canceller to distinguish the desired signal from the interference signal. Therefore, even if a convolution code is employed, performance is not improved much. On the other hand, when the received power of the desired and the interference signals is different, the CCI canceller can easily distinguish the signals. Thus, the probability of detection error become slow.In multicarrier communication systems, each subcarrier is subject to different fading, and thus has different received signal power. However, the effects of fading on adjacent subcarriers can be regarded as the same in general. Therefore, when interference signals have nearly the same power level with the desired signal, some of consecutive subcarrier has nearly the same power level too. This leads to burst detection errors in a frequency domain, and the advantage of using convolutional codes is reduced. As a result, the BER performance is degraded.

A number of receiver techniques have been proposed to mitigate this effect, the two main categories being linear receivers and non-linear interference cancellation receivers. With a linear receiver, a linear transform is applied to the received signal to restore the orthogonality between the users, although the effect of additive noise also needs to be taken into account. With the non-linear interference cancellation receivers, tentative estimates of user data are used to reconstruct interfering users signals and these are subtracted to leave an interference free signal. The interfered signal approach the performance of the maximum likelihood receiver. This optimum receiver compares the received signal with every possible combined transmitted signal, and requires an exponentially complex algorithm, which prohibits its use if the number of CDMA users is larger than about 10.

In the despreading (decorrelation) and demodulation process the recived signal is converted into a narrow band positive baseband Signal-to-Interference (S/I) component. The baseband S/I have to be sufficiently large to relatively lead to a relatively low Pe(Probability of Error). The baseband self-interference, S/I are frequently designed to several decibels higher than the baseband Signal-to-Thermal noise (S/N).

Filter bank techniques provide better transmitting and receiving filters in terms of performance. In several applications we need to either decompose a signal into a number of components or to assemble a number of signals into one signal. These two operations are called Analysis (one signal to many signals) and Synthesis (many signals to one signal), respectively and they are performed by filters. In a typical spectral decomposition problem, the frequency spectrum is subdivided into number of bands, with the intent of determining the signal component within each band. The goal can be either to be actually estimate a time-varying frequency spectrum or to add flexibility in encoding a signal for compression and transmission, based on the fact that all frequencies contribute equally to analysis and synthesis networks the total energy of the signal.

A bank of Bandpass filters with transfer functions H_0 (Z)……….H_{M-1}(Z), tuned at different frequencies provides for necessary decomposition called ANALYSIS FILTER bank. Each output can be decimated by a factor L, [1] downsampling is represented as D and upsampling as U, generally with restriction$1<=L<=M$.The case L=M is maximally decimated and it is easy to see in this case, that the data rate at both sides of the analysis filter bank is same. When L<M the analysis network increases the data rate, when L>M, that is when the filter bank lowers the data rate clearly there is a loss of information and is discarded. The transmitting and receiving filters have excellent frequency separation property inherited from good filter bank design. For frequency selective channels there is intraband and crossband interference in Filter bank transceivers [1]. Modulated filter bank transceivers achieve ISI free transmission over AWGN channel is considered. The rest of the paper is organized as follows. In section 2, Transceiver conditions are derived and are formulated using matrix representation. The algorithms for mitigating interference and error control coding is studied in section 3.A MUSIC receiver is constructed to enhance receiver performance is studied in section 4. The proposed algorithms provide desired performance .Simulation results are provided in section 5 to demonstrate the usefulness of proposed transceiver.

2. DFT Modulated Filter Bank Transceiver

The Transceiver proposed in this scheme is capable of achieving desirable performance with number of sub carriers increased which is the principle behind Multi Carrier CDMA system. The system involves Multi Carrier through which the desired signal is extracted from number of multi-carriers transmitted. The algorithms involved increase system performance to accommodate large number of users at higher data rate. By increasing the throughput, the probability of occurrence of error at the receiver can decreased. The system performance is measured by considering the parameters such as Bandwidth, SIR, SNR and average throughput. By appropriate use of modulation scheme along with the above considered steps, desired performance can be achieved by increasing the number of users as well as the average throughput.

The system is designed in such a way that it uses M number of sub bands and N is the upsampling and downsampling ratio. The conditions are derived such that ISI free condition [1] is possible when the ratio is greater than or equal to one and the difference between them yields number of redundant samples added to combat Intra-band and cross bend ISI.

Considering only FIR filters with

$F_i(Z)=f_i\,(0)+f_i\,(1)z^{-1}+\ldots f_i\,(n_f)^{z-nf}$ (1)

$H_i\,(Z)=h_i\,(0)+hi(1)z\text{-}1+\ldots hi\,(n_f)^{z-nf}$ (2)

Where (n_f+1) and (n_h+1) are respectively the lengths of transmitting and receiving filters. The values of n_f and n_h can be larger than N.F_i (z) and $F_h(z)$ are transmitting and receiving filters. The proposed system incorporates itself the ability to offer higher degree of synchronization by selecting appropriate values of M and N.The selection involves in identifying the optimized value that offers required synchronization.

The selection parameters involve

M-Number of sub bands
N-Downsampling and Upsampling ratio
N> = M-ISI free solution is possible
N-M-Number of redundant samples added to combat intraband and crossband isi
N=M-Maximally decimated

From the above parameter set desired carrier is selected from multiple carriers transmitted.

2.1. Multi Code Multi Carrier Direct Sequence CDMA (MC-MC-DS- CDMA) :

Multi-code Multi-Carrier CDMA is a combination of two techniques. First, an OFDM system is used to provide a number of orthogonal carriers, free from ISI [2]. Second , an individual code chip, to provide a spread spectrum system, which modulates each carrier. The main advantage of doing this is that when the multiple-access interference becomes a problem, the resulting linear detectors are much simpler to implement, as only a single tap equalizer is required for each channel. Rake reception[3] can also be employed to exploit the channel diversity by channel matching in the frequency domain allowing optimal reception for a single use capacity and data rate of wireless communication systems are limited by time varying characteristics of dispersive fading channel. In order to support maximum number of users at high signal strength we propose two algorithms searching algorithm and a refining algorithm.

2.2.Characterization of multipath fading channel

There are several large obstacles between a basestation (BS) and a mobile station (MS), and also many local scatterers (e.g., neighboring buildings)[3] in the vicinity of the MS. Reflection of the signal by large obstacles creates propagation paths with different time delays; each path is a cluster of irresolvable multipaths created by reflection or diffraction, by local scatterers, of the transmitted signal reaching the surroundings of an MS. They interfere and the received signal power changes rapidly in a random manner with a period of about half-carrier wavelength when the MS moves. Such a multipath channel can be viewed as a time varying linear filter of impulse response *h* (t, *t*) observed at time *t*.

2.2.1 System Model for MC-MC-DS-CDMA

Consider k user's, where the k^{th} user's serial symbol data stream where each symbol has duration T_d is first serial to parallel (S/P) converted into M sub channels, and the new symbol duration in each sub channel is T_s=M $T_{d.}$

Let M S/P converted signals at the time i for user k be given by

$S_i^{(k)}=[s_0^{(k)}(i),s_1^{(k)}(i),\ldots\ldots\ldots\ldots.\ s^k_{M-1}(i)]^T$ (3)

$S_q^{(k}(i)=[\ S_q^{(k}(i),\ S_{q+Q}^{(k}(i),\ldots\ldots\ S_{q+(D-1)Q}(i)]^T$ (4)

Where M=DQ and Q and D are number of symbol streams and the number of subcarriers in each symbol stream respectively signals spread in frequency domain are then respread in time domain by using k^{th} user's specific spreading code or signature sequence $c_n^{(k)}$ where n=0,1…..N-1 where N is the processing or spreading gain for user k. Thus respread signals are assigned to each subcarrier via an interleaver, and the multicarrier modulation is implemented by using the Inverse Discrete Fourier Transform (IDFT). Finally, the modulated signals are parallel transmitted over M subcarriers.

In MC-CDMA, after recovery of the subcarriers, the signals at the output of the FFT have to be 'unspread', by applying the inverse code matrix. However some weighing is needed to optimize performance and to mitigate the effects of the channel. At this point we restrict ourselves to the class of (linear) receivers, which makes decisions based on linear combinations of all subcarrier signals. We explicitly introduce the FFT, the inverse code matrix C^{-1} and a generic weigh matrix *W*. This allows us to address a simple implementation for the receiver, where the weighing reduces to a simplified adaptive diagonal matrix, while the FFT and C^{-1} are non adaptive, and can be implemented efficiently using standard butterfly topologies. Inversion is needed to find the MMSE of the signal in the presence of noise, MAI and ICI.

3. A Enhanced MLSD Method:

A Fast MLSD method involves selection algorithm and refining algorithm

3.1. Selection Algorithm:

After frequency spreading, M low rate data symbols $x_0^{(k)}$, $x_1^{(k), \cdots\cdots}$ $x_{M-1}^{(k)}$ are superimposed with Code Division Multiplexing (CDM). The sequence generated by CDM is a Multiamplitude sequence, which consists of M superimposed chips .The Multiamplitude sequence, is called a CDM block, and each superimposed chip is called a CDM chip[3]. Now let M received signal

$\gamma_0^{(k)}, \gamma_1^{(k)}, \ldots\ldots.., \gamma_{M-1}^{(k)}$ be subdivided into Q symbol streams. Like $s_q^{(k)}$and $x_q^{(k)}$, we define

$r_q^{(k)}=[r_q^{(k)}, r_{q+Q}^{(k)} \ldots\ldots.. r_{q+(D-1)}^{(k)}]^T$ where $q=0,1 \ldots\ldots..Q-1$

$$n_q^{(k)}=[n_q^{(k)}, n_{q+Q}^{(k)} \ldots\ldots.. n_{q+(D-1)}^{(k)}]^T \quad (5)$$

Consider the qth CDM block of user k .For convenience, we denote

$r=[r_{0,} r_1,\ldots r_{D-1}]^T=[r_q^{(k)}, r_{q+Q}^{(k)} \ldots r_{q+(D-)}^{(k)}]^T$

$n=[n_{0,} n_1,n_{D-1}]^T=[n_q^{(k)}, n_{q+Q}^{(k)} \ldots n_{q+(D-)}^{(k)}]^T$

$s=[s_{0,} s_1,\ldots s_{D-1}]^T =[s_q^{(k)} s_{q+Q}^{(k)} s_{q+(D-1)}^{(k)}]^T$

and let $h=[h_0, h_1 \ldots..h_{D-1}]^T$ be the equivalent gain vector corresponding to CDM block.

$$r=E.diag(h)(1/sqrt(D)\ W_s)+ n \quad (6)$$

where E is the energy of the transmitted signal and diag (h)=diag ($r_{0,}$ $r_1,\ldots r_{D-1}$). In order to determine the transmitted signal vector $s=[s_{0,} s_1, s_{D-1}]^T$ [3]from the received signal vector $r=[r_{0,} r_1,\ldots\ldots. r_{D-1}]^T$ in which all of G possible symbol vectors $s_{1\ldots\ldots\ldots}s_G$ is searched to find an optimal symbol vector S_{opt} as

$$S_{opt=}arg_{g=1.G}^{min}[E.diag(h)(WS_g/sqrt(D))-r]^2. \quad (7)$$

The best vector from the number of transmitted signal vectors has been chosen and the method is called as SELECTION process.

The fast Maximum Likelihood Decoding algorithm employed at the receiver is capable of decoding the appropriate vector, given from the number of vectors transmitted through multiple carriers, which is considered to be as the searching step and the refining step involves in identifying the error bit position, and calculating the probability of occurrence of error for a particular transmitted signal. It selects only the probability of successful transmission by which desired system performance can be achieved.The desired signal is chosen in such a way that it satisfies the calculated BER which is set as threshold.

Multi rate transmission for single carrier CDMA systems in AWGN channel, the code assignment is limited by number of orthogonal codes for the short spreading factor, and multipath can be problematic for the higher data rates since the spreading factor is short. The proposed Multicode multicarrier CDMA system does not require variable spreading factors. It uses the same code book to support various data rates for different users.

The number of simultaneously higher data rate users in a multi carrier CDMA system will be less than the number of equal data rate users in a traditional CDMA system. A variation of the Multi code scheme, which supports variable data rates by varying the set of code sequences assigned to each of the users, has been proposed. The users communicate their data by choosing one sequence from their code set to transmit over common channel.

4. MUSIC Algorithm

After retrieving the best signal from searching process our aim is to find the relationship between the transmitted and the received signal which in turn detects error as well as interference added to the signal while transmitting. The motto is to decide the number of sub bands, proper upsampling and down sampling ratio which tend to optimize the sampling frequency. If the received signal satisfies the constraints of the MUSIC algorithm SNR is calculated, if not refining process tend to retransmit the signal which is progressing in particular frequency until the desired output is reflected at the decoder section.

The main advantage of doing this is that when the multiple-access interference becomes a problem, the resulting linear detectors are much simpler to implement, as only a single tap equalizer is required for each channel. Rake reception can also be employed to exploit the channel diversity by channel matching in the frequency domain allowing optimal reception for a single user.

In the uplink another advantage of MC-CDMA can be exploited. If the signals can be synchronized to arrive within a small fraction of the symbol time (e.g. indoor, or very small cell environment) then this asynchronism can be overcome by cyclically-extending the signal further allowing synchronous reception of all signals, with no ISI from other users.

In MC-CDMA, after recovery of the sub-carriers, the signals at the output of the FFT have to be 'un-spread', by applying the inverse code matrix. However some weighing is needed to optimize performance and to mitigate the effects of the channel. At this point we restrict ourselves to the class of (linear) receivers, which make decisions based on linear combinations of all sub-carrier signals. We explicitly introduce the FFT, the inverse code matrix C^{-1} and a generic weigh matrix W. This allows us to address a simple implementation for the receiver, where the weighing reduces to a simplified adaptive diagonal matrix, while the FFT and C^{-1} are non-adaptive, and can be implemented efficiently using standard butterfly topologies.

A joint optimization can be derived from the following MMSE model. It reduces the joint effects of noise, MAI and ICI. The Minimum Mean-Square Error Estimate of the user data is equal to the conditional expectation $\mathrm{E}B|Y$. We can rewrite this as $\mathrm{E}B|Y = \mathrm{E}C^{-1}A|Y = C^{-1}\mathrm{E}A|Y$

Constitutes for over all instances of the modulation, but keeping the channel fixed. It shows that without loss of performance one can estimate the modulation of each sub-carrier as A, and then perform an inverse of the code matrix, using $B = C^{-1}A$ for the user data. Let A be a linear combination of Y, namely $A = WY$.

The optimum choice of matrix W follows from the orthogonality principle that the estimation error is uncorrelated with the received data, viz., E $(A - A)\ YH = 0_N$ with 0_N an all-zero matrix of size N by N. Thus we arrive at $W = \mathrm{E}\ [AY^H]\ \mathrm{R}_{YY}^{-1}$, for the optimum estimation matrix. Here $Y = HA + N$, where channel matrix H has the components $H_{nm} = \mathrm{b}_{nm}$.

In such case,

$\mathrm{E}\ (AY^{\mathrm{H}}) = \mathrm{E}\ (A\ (HA)^{\mathrm{H}}) + \mathrm{E}\ (AN^{\mathrm{H}}) = \mathrm{E}AA^{\mathrm{H}}H^{\mathrm{H}} = C\ \mathrm{E}BB^{\mathrm{H}}C^{\mathrm{H}}H^{\mathrm{H}} = H^{\mathrm{H}}$. (8)

Also, R_{YY}, the covariance matrix of Y, becomes

$\mathrm{R}_{YY}=\mathrm{E}YY^{\mathrm{H}}=H\mathrm{E}(AA^{\mathrm{H}})\ H^{\mathrm{H}} +\mathrm{E}NN^{\mathrm{H}} =HH^{\mathrm{H}} +N_0T_sI_N$ (9)

Inversion is needed to find the MMSE of the signal in the presence of noise, MAI and ICI.

For simplicity, we initially review the special case of a channel without Doppler spread, thus with $H = T_s$ diag ($\mathrm{b}_{0,0}, \dots \mathrm{b}_{N-1,\,N-1}$), as it was proposed in [21]. Then E (AY^{H}) reduces to that is, each sub-carrier is weighed by a factor, which only depends on the signal strength in that sub-carrier and the noise, we interpret this as an automatic gain control and a phase corrector. In the more general case of time-varying channels with ICI implementation of this MMSE solution is quite involved because W does not reduce to a diagonal matrix. This implies that the optimum filter requires a (channel-adaptive) matrix inversion. Mostly, such studies assume a limited number of (dominant) propagation paths (small I_w), so in this respect these differ from our Rayleigh model. In practice, it may not always be feasible or economic to estimate all b* accurately, invert the covariance matrix in real-time, while adapting fast enough for the time-variations of the channel.

Applications of interest are in instantaneous spectral decomposition and in digital communications. In a typical spectral decomposition problem we subdivide the frequency spectrum into a number of bands, with the intent of determining the signal component within each band. The goal can be either to actually estimate a time-varying frequency spectrum or to add flexibility in encoding a signal for compression and transmission, based on the fact that all frequencies contribute equally to the total energy of the signal.

A Bank of band pass filters with the transfer functions $H_0(Z)$……..$H_{M-1}(Z)$, tuned at different frequencies, provides for the necessary decomposition .That is what is called the Analysis filter bank. Each output can be decimated by a factor L , as shown in the figure , generally with the restriction1< =L< =M.The case L=M is called MAXIMALLY DECIMATED and it is easy to see in this case ,the data rate at both sides of the analysis filter bank is same. When L<M , the analysis network increases the data rate ,when L>M that is, when the filter bank lowers the data rate-clearly there is a loss of information and we disregard this case.

Considering only FIR Filters with

$Fi(z)=fi(0)+fi(1)z^{-}+\dots\dots\dots fi(n_f)z^{-nf}$ (10)

$Hi(z)=hi(0)hi(1)z^{-1}+\dots\dots\dots hi(n_f)z^{-nf}$ (11)

where (n_f+1) and (n_h+1) are, respectively, the length of transmitting and receiving filters. The values of n_f and n_h can be larger than N. Fi(z),Hi(z)-Transmitting and receiving filters. Digital filter is called a FIR Filter if its impulse response has a finite number of non-zero entries such as h[n] for n=0,1…….N. Generally we assume implicitly h[0] is not equal to zero and h[N] is not equal to zero.

The filter has utmost N+1 non zero entries and is said to have length N+1 nonzero entries and is said to have length N+1Let x[n] and y[n] be the input and output of the filter. Then they can be described by the convolution

Y[n]=h[0]x[n]+h[1]x[n-1]+………..+h[N]x[n-N] (11)

Filter is said to have order N.A Nth order FIR Filter clearly has length N+1.

Applying Z-Transform and applying zero initial conditions we obtain the transfer function of the filter as

$$Y(Z)/X(Z)=H(Z) = h[0]+h[1]z^{-1}+\ldots\ldots h[N]z^{-N}$$

$$Y(Z)=h(0)Z^{N}+h[1]Z^{N-1}+\ldots h[N]/z^{N} \quad (12)$$

It has N poles and N zeros. All its poles are located at z=0.Thus every FIR filter is stable .This also follows from the fact that its impulse response is always absolutely summable.

FIR Filters that have linear phase will be designed to approximate a desired frequency response of the form

$$Hd(e^{jw})=D(W)e^{-jMw} \quad (13)$$

Where M-Positive constant

D(W)-Desired magnitude response

At the transmitter or receiver we only need to implement a prototype filter of ordernf(or corresponding order nh) and an M by M DFT Matrix that can be implemented efficiently using FFT. In many applications ,it is often desired to have transmitting or receiving filters with good frequency responses .For many wireless communication systems the preference is to have transmitting filters with better frequency responses so that the transmitter outputs will have smaller out-of-band energy. In this case $F_0(z)$ is designed to be a good low pass filter and Fi(z) will be a good band pass filter.

Depending on the application the design problem is either

1. Given a good lowpass transmitting prototype filter F0(z),design the receiving prototype filter H0(z) to achieve ISI Free property or SIR Maximization.
2. Given a good low pass receiving prototype filter H0(z),design the transmitting prototype filterF0(z) to achieve ISI Free property or SIR Maximization.

Consider the transmitting channel to be slowly varying and that it van be modeled as an FIR LTI channel c(z) and an additive noise v(n).Let L be the maximum possible order of the channel c(z).Then c(z) can be expressed as

$$C(z)= \sum_{n=0}^{L} C(n)z^{-n} \quad (14)$$

the order of L can be larger than number of redundant samples (N-M)

In multicarrier communication systems, each subcarrieris subject to different fading. If adjacent subcarriers have independent fading in a frequency domain,consecutive detection error does not occur. However, the effects of fading on adjacent subcarriers can be regarded as the same in general. Therefore, the desired and the interference signals within some of consecutive subcarriers have the same power. This causes consecutive detection errors

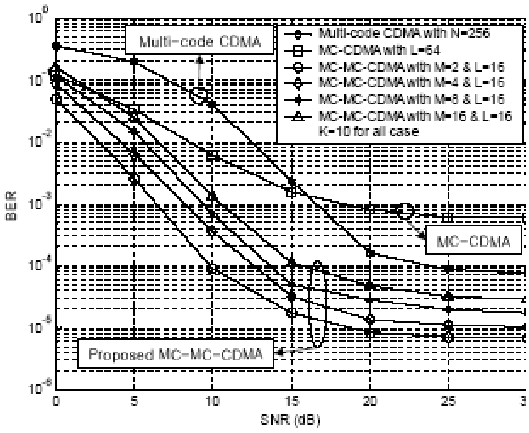

Figure 1 Comparison of MC-CDMA-and MC-MC-DS-CDMA in terms of SNR and BER

The above figure investigates the relation between each subcarrier separate by certain frequency which amounts in canceling ISI[5] which tend to degrade the signal strength the choice of best subcarrier is chosen such that the available bandwidth in the channel as well as the signal with maximum strength in terms of resisting itself form interference

5. Simulation Results

The result shown below proceeds with declaring unified samples that tends to undergo searching and refining algorithm yielding a better SNR and maximum achievable BER at the desirable data rate. The conventional method yields BER of 10-1 which does not provide required resistance for the transmitted signal which in turn adds up with noise generated from channel as well as interference from other users.

By proper frequency selection BER is optimized to retain original signal strength at the receiver which is considered to be best signal at receiver. The BER performance versus the number of users for both systems with an SNR of 30dB is shown in result. At the same BER, data rate per user, and consumed bandwidth, the MC-MC-CDMA system can support more users than the

MC-CDMA system. For example, at the BER of 10^{-3}, the number of users supported by the MC-MC-CDMA system is about 30, while it is about 7 for the MC-CDMA system. These are both uncoded systems with a total spreading gain of 64 before detection versus SNR with the various K and M. In this system, the mean of all interference power is assumed to be equal. The received SINR of the MC-MC-CDMA system varies according to the variation of K and SNR, but not M. Since the length of the code sequence N is fixed over all different value of M, the received SNR is not changed according to M. It means that the proposed MC-MC-CDMA system can support higher data rate without increasing the interference unlike the multi-code CDMA system. However, because higher M causes the proportionally reduced minimum distance of the code sequence, the BER after despreading increases with M. Here BER up to 10^{-6} is achieved by proper usage of proposed algorithms at decoder where by increasing SNR.

The result shown below proceeds with declaring unified samples that tend to undergo searching and refining algorithm yielding a better SNR and maximum achievable BER at the desirable data rate

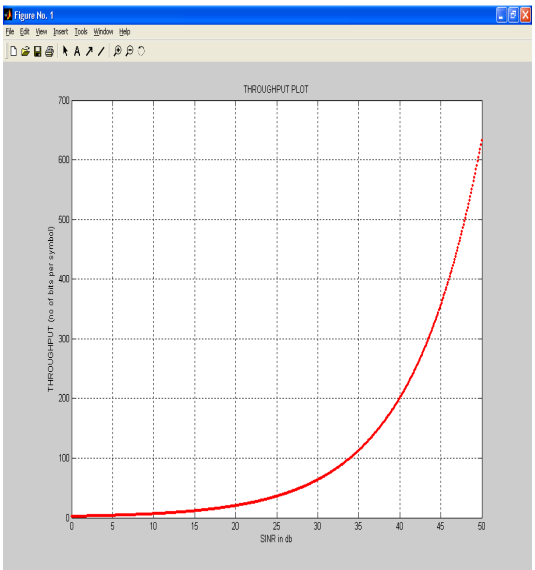

Figure 2 Response for SINR Vs Throughput

The above result show that for transmitting n bits, considering n different combinations of input for a Multi Code Multicarrier Direct Sequence CDMA System yields Better Throughput

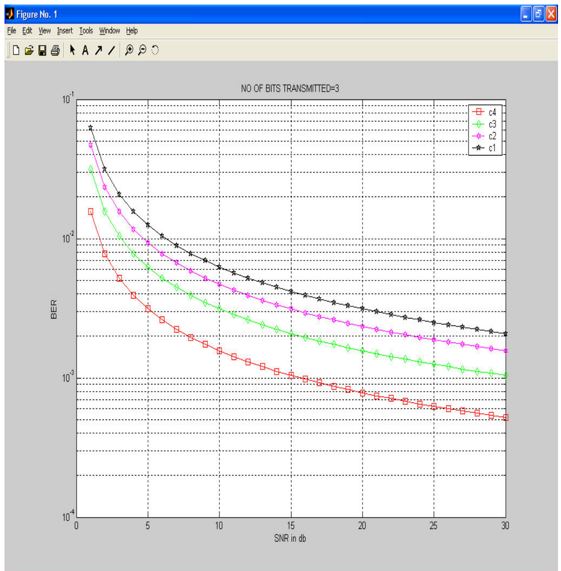

Figure 3 BER Plot for 3 Transmitted Bits

The above result show that for transmitting 3 bits, considering 8 different combinations of input for a Multi Code Multicarrier Direct Sequence CDMA System yields SNR OF 30 db and BER of 10^{-3}.

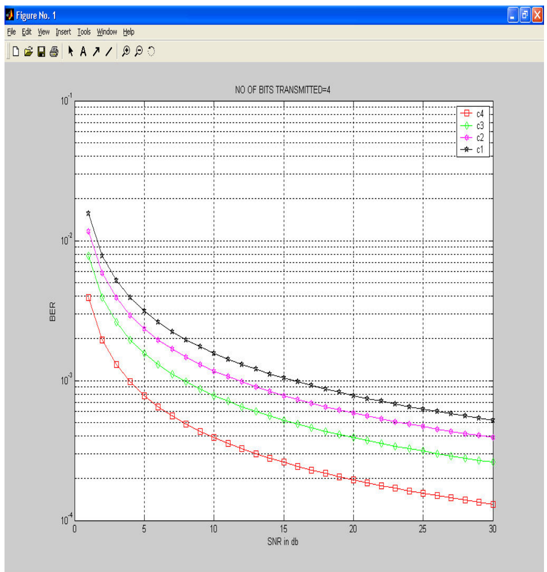

Figure 4 BER Plot for 4 Transmitted Bits

The above result show that for transmitting 4 bits, considering 16 different combinations of input for a Multi Code Multicarrier Direct Sequence CDMA System yields SNR OF 30 db and BER of 10^{-4}.

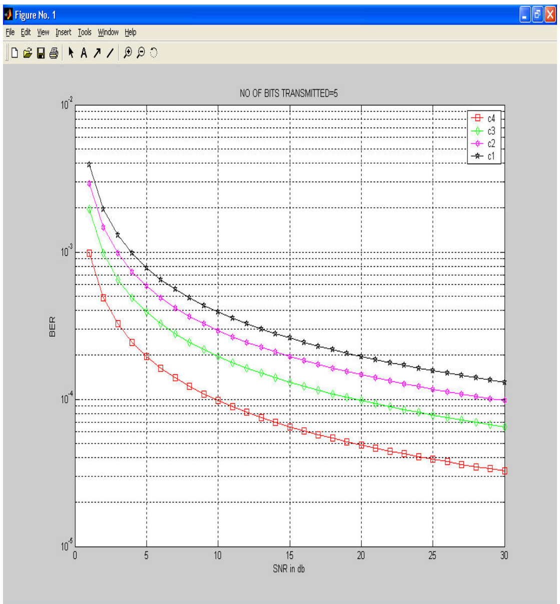

Figure 5 BER Plot for 4 Transmitted Bits

The above result show that for transmitting 5 bits, considering 32 different combinations of input for a Multi Code Multicarrier Direct Sequence CDMA System yields SNR OF 30 db and BER of 10^{-5}.

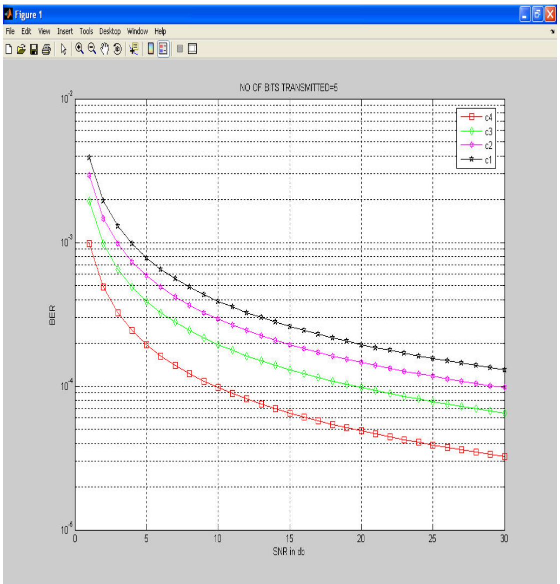

Figure 6 BER Plot for 5 Transmitted Bits

The above result show that for transmitting 6 bits, considering 64 different combinations of input for a Multi Code Multicarrier Direct Sequence CDMA System yields SNR OF 30 db and BER of 10^{-5}.

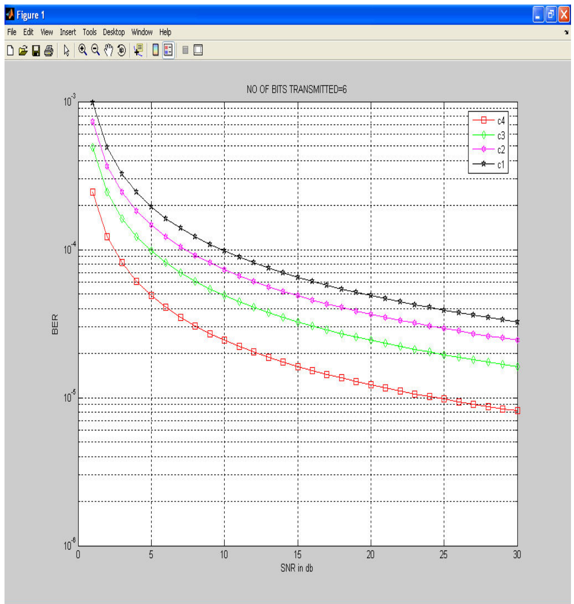

Figure 7 BER Plot for 6 Transmitted Bits

The above result show that for transmitting 6 bits, considering 64 different combinations of input for a Multi Code Multicarrier Direct Sequence CDMA System yields SNR OF 30 db and BER of 10^{-5}.

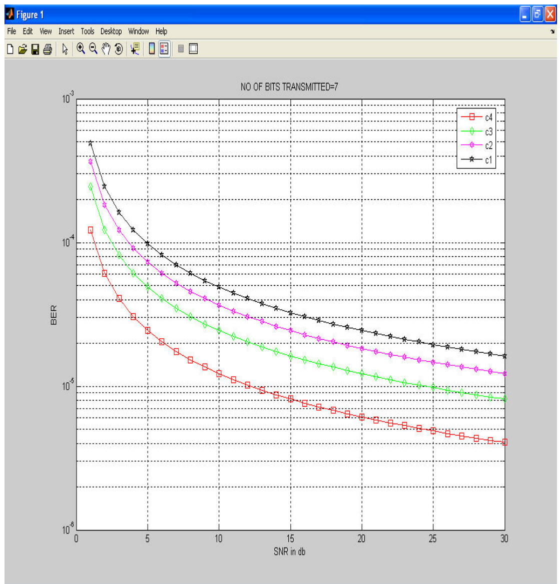

Figure 8 BER Plot for 7 Transmitted Bits

The above result show that for transmitting 7 bits, considering 128 different combinations of input for a Multi Code Multicarrier Direct Sequence CDMA System yields SNR OF 30 db and BER of 10^{-5}.

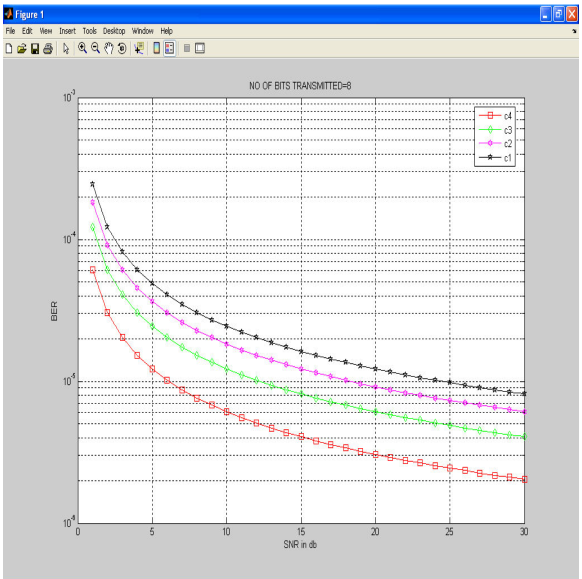

Figure 9 BER Plot for 8 Transmitted Bits

The above result show that for transmitting 8 bits, considering 256 different combinations of input for a Multi Code Multicarrier Direct Sequence CDMA System yields SNR OF 30 db and BER of 10^{-5}.

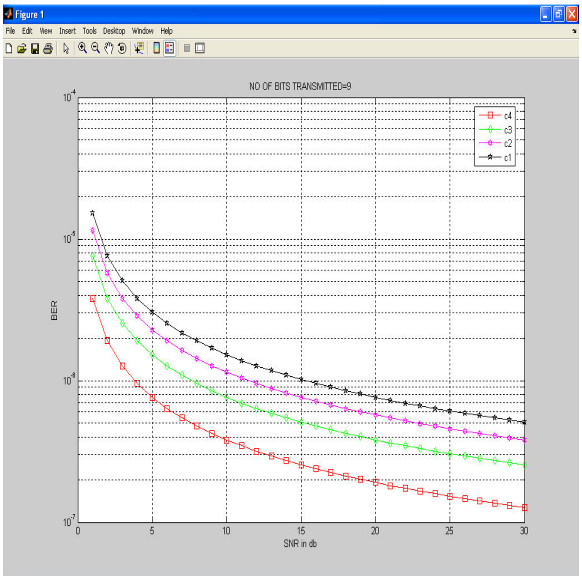

Figure 10 BER Plot for 9 Transmitted Bits

The above result show that for transmitting 9 bits, considering 512 different combinations of input for a Multi Code Multicarrier Direct Sequence CDMA System yields SNR OF 30 db and BER of 10^{-6}.

6. Conclusion

In this paper, multi-code multicarrier CDMA is shown to be a promising method for supporting variable data rates for a large number of user's in future cellular systems. By using the multi-code concept, the MC-MC-CDMA system achieves two-dimensional spreading gain as well as frequency diversity. In addition, various data rates can easily be supported by changing the size of the code sequence set. With the same total bandwidth, both analytical and simulation

In this paper, we considered multi code multi carrier DS CDMA transceiver and proposed algorithms to obtain better signal strength and mitigate interferences which tend to degrade the signal strength at the receiver section. The two algorithms namely the selection algorithm and MSIC algorithm regain the desired signal strength and decode properly for a system with very high speed data rate. The system designed using filter banks is capable of achieving greater synchronization at minimum received power and maximize signal to interference ration by which Bit error rate is reduced to the desired level. Using the Multicarrier modulation technique and achieving orthogonality for multiple codes generated is capable of mitigating all forms of interferences by selecting the proper carrier at the receiver yielding the desired output with minimum bit error rate and at the same time maximizing Signal to Noise ratio. The interference power is reduced by calculating the probabilities of best received vector with minimum bit error probability. The target of achieving higher signal strength and greater synchronization with minimum bit error rate is achieved by considering the algorithms proposed. Multi-code multicarrier CDMA was shown to be a promising method for supporting variable data rates for a large number of users in future cellular systems. By using the multi-code concept, the MC-MC-CDMA system achieves two-dimensional spreading gain as well as frequency diversity. In addition, various data rates can easily be supported by changing the size of the code sequence set. With the same total bandwidth, both analytical and simulation.In this proposed work, we considered multi code multi carrier DS CDMA transceiver and proposed algorithms to obtain better signal strength and mitigate interferences which tend to degrade the signal strength at the receiver section. the two algorithm namely the searching algorithm and refining algorithm along with MUSIC algorithm regain the desired signal strength and decode properly for a system with very high speed data rate. For number of bits to be transmitted decided by the user the system is capable of generating bit error probabilities at minimum error rate

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