# PERFORMANCE ANALYSIS OF A **MULTI-TONE DIRECT SEQUENCE CODE DIVISION MULTIPLE ACCESS** (MT-DS-CDMA) SYSTEM OVER A **NAKAGAMI-*m* FADING CHANNEL**

**Thesis Report**

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## Submitted by


Umme Salma(08110067)

Tanzina Rahman Tanni(08110075)

Nishat Jahan Mourin(08110076)

Humayra Islam (08110097)


Thesis under Supervision of **Dr. Satya Prasad Majumder**

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We also like to thank our parents and family for making it possible for us to study and for their constant help and support. We have worked hard and gave our best. Hopefully, our work will appreciate by our supervisor and respected faculties.

# *Abstract*

**Performance analysis has been carried out for a multi-tone DS-CDMA wireless communication system over a faded channel considering the channel limitations. The analysis is based on MT-DS-CDMA system and a rake receiver with maximal ratio combining (MRC) diversity combining technique. The expressions for the output signal and (Multiple Access Interference) MAI has been derived for the above system model. The expression for the SNR and the unconditional BER at the output of the receiver has also been derived. The performance results has been evaluated numerically in terms of SNR, BER and the power penalty due to channel effect. The optimum system design parameters have been found for the numerical results. Also simulation works has been done for different parameters and comparison has been done with one of the most effective CDMA scheme among all the systems.**

# Chapter 1

# Introduction

Recent development co-relates generally cellular and other types of wireless communications systems and more particularly to code division multiple access (CDMA) systems which use multiple tone modulation to achieve a wide band signal with minimal intra-cell and intra-sector interference. Code division multiple accesses is a multiplexing technique where a number of users simultaneously and asynchronously access a channel by modulating and spread their information-bearing signals with pre-assigned signature sequences. CDMA has different sub divisions. One of them is MT-DS-CDMA. In this paper, we have analysis the numerical results for outdoor and noisy environment MT-DS-CDMA transmission. We are using Nakagami-m fading channel, for the of data transmission. This type of fading channel is used normally for Air to Space station communication or Air to Ground station communication. We consider the generalized frequency-selective Nakagami- fading channels because this model is very general and often best fits the land-mobile and indoor-mobile multipath propagations.

For MT-DS-CDMA transmission, we are considering three types of coding schemes, known as PN-sequence Coding, m-sequence coding and Gold coding. For transmitting the data we are using OFDM modulator because OFDM systems are excellent systems for the application to wireless broadband channels. In receiver side, we are using OFDM demodulator and then Rake combiner, to decode the demodulated signals. We derived results for transmitted and received signal. From the expression of transmitted and received signal, we have calculated the power signals. From the received signal part, we have separated the desired signal, multiple access interference (MAI) and noise. From these expressions, we have calculated Signal to Noise and Interference Ratio (SNIR). Then we have found the expression for instantaneous BER (bit error rate). We also have found the average bit error rate of the system without rake combiner. Then we have run the simulations for derived equation for BER vs. user, vs. Eb/No and also user vs. cross correlation. Then we have compared these results with some of the results of MC-DS-

CDMA which we found via internet. Up to this, our thesis work was done. Next work will be carried out in basis of solutions for the drawbacks we have found in thesis phase.

# Chapter 2

## Code division multiple access (CDMA)

**Code division multiple access** (**CDMA**) is a channel access method used by various radio communication technologies. One of the basic concepts in data communication is the idea of allowing several transmitters to send information simultaneously over a single communication channel. This allows several users to share a band of frequencies. This concept is called Multiple Access. CDMA employs spread-spectrum technology and a special coding scheme to allow multiple users to be multiplexed over the same physical channel. CDMA is a form of spread-spectrum signaling, since the modulated coded signal has a much higher data bandwidth than the data being communicated. We are considering Asynchronous CDMA. Asynchronous CDMA offers a key advantage in the flexible allocation of resources, (such as allocation of a PN codes) to active users. In synchronous CDMA, there are fixed number of codes, time slots and frequency slots in terms of number of simultaneous user. This is the reason choosing Asynchronous CDMA over Synchronous CDMA.

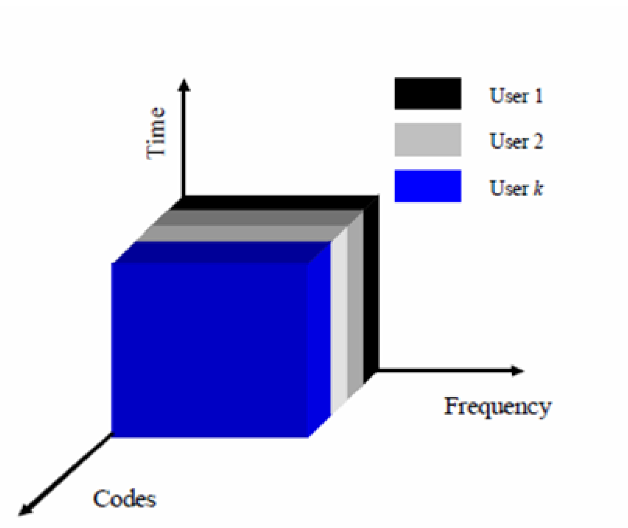


**Figure 1.1:** Code Division Multiple Access

CDMA has different multiple access schemes. These are known as –

- Direct Sequence Code Division Multiple Access (DS-CDMA)
- Multi carrier Code Division Multiple Access (MC-CDMA)
- Multi tone Code Division Multiple Access (MT-CDMA)
- Multi Code Direct Sequence Code Division Multiple Access (MC-DS- CDMA)
- Multi Tone Direct Sequence Code Division Multiple Access (MT-DS- CDMA).

## 2.1. Direct Sequence Code Division Multiple Access (DS-CDMA):

Direct sequence code division multiple access (DS-CDMA) is an attractive allocation technique that allows users to be simultaneously active over the total available bandwidth. In direct sequence transmission, the user data signal is being multiplied by a code sequence. Mostly binary sequences are used. The duration of an element in the code is called the chip time.

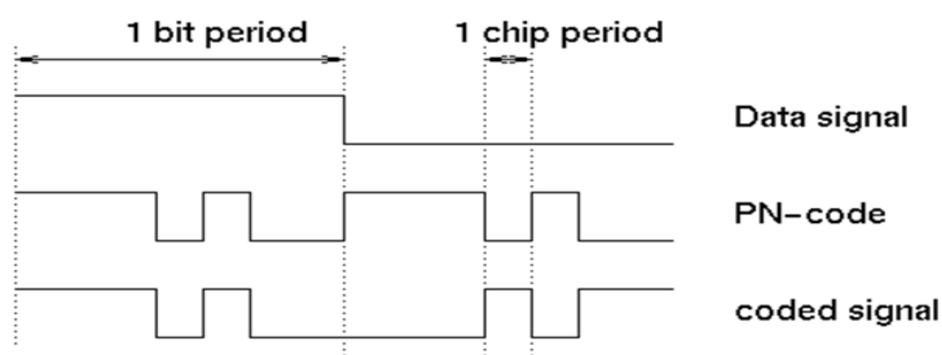


**Figure 1.2.1:** DS-CDMA

In case of direct sequence, one single data stream is being multiplied by a frequency, which is called modulation and then modulated signal is sent through medium. In receiver side, the received signal is being de-modulated and then after filtering using low pass filter, main transmitted data can be achieved.

## 2.1.1. DS-CDMA Transmitter:

In Direct Sequence CDMA system transmission, before modulation, data stream is being encoded with a constant code and then modulation technique is being processed. Here, the user data signal is multiplied by a pseudo random code sequence.

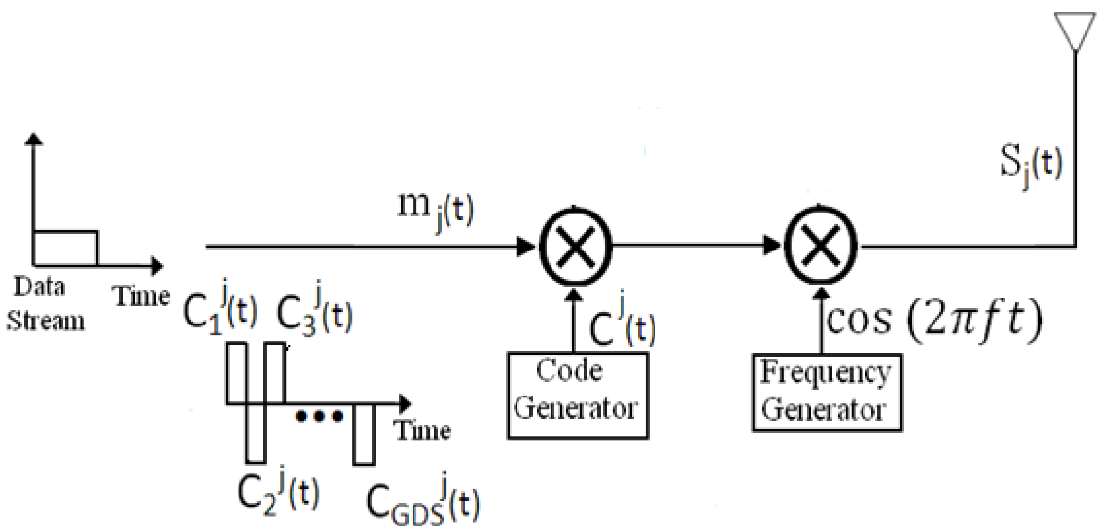


**Figure1.2.2:** DS-CDMA transmitter

In figure 1.2.2 the DS-CDMA transmitter is designed by the capacity of *j* number of user. The DS-CDMA transmitter combines the original data stream using a given spreading code in the time domain. Here t is time, $m_j(t)$ is the data stream of $j^{th}$ user, $c^j(t)$ is the pseudo random code, $m_j(t)c^j(t)$ is $j^{th}$ user data stream after combines the code width. The transmitter generates unique code for each user over one bit period by the code generator. The frequency generator generates one carrier frequency $cos(2\pi ft)$ for each user of Phase Shift Keying modulation technique. After combining and modulation of digital data, it is transmitted by the CDMA antenna over the wireless media like air.

## 2.1.2. DS-CDMA Receiver:

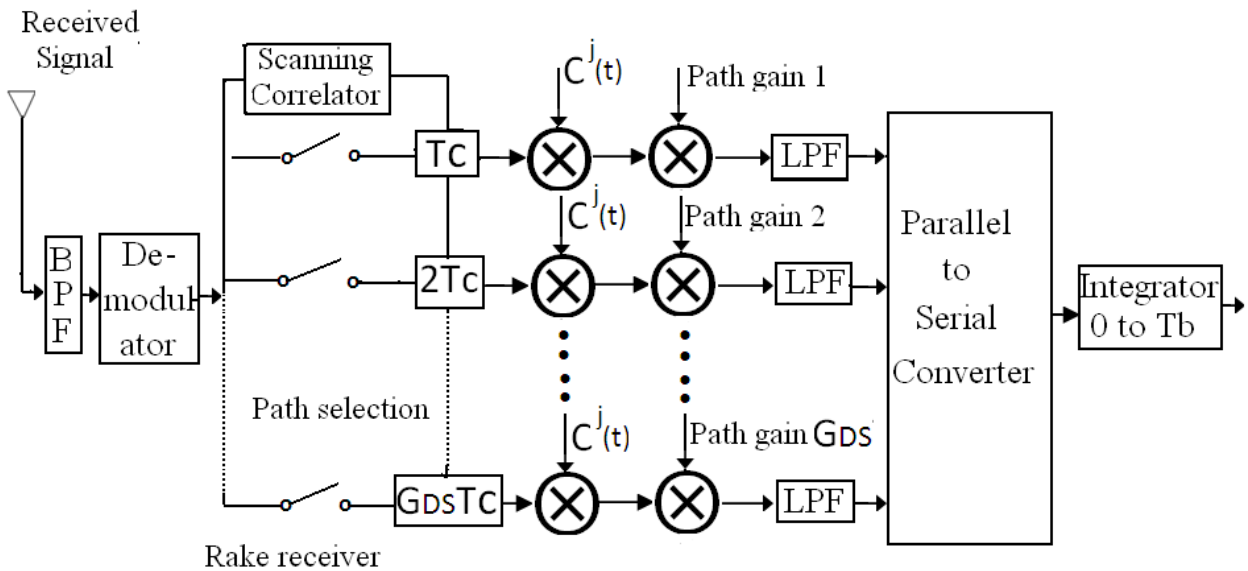


**Figure1.2.3:** DS-CDMA receiver

In figure 1.2.3 the DS-CDMA receiver is designed by the capacity of *j* number of user. Then it will face the MAI for user 1 to user (j-1). At First CDMA antenna received the transmitted signal and then it passes through a band pass filter (BPF). BPF allows a certain band of frequency of transmitted signal and remove any unwanted signal. Demodulator demodulates the signal with $cos(2\pi ft)$ carrier, which is same as the carrier frequency provided by the transmitter. Then, there is a special kind of receiver named rake receiver. It takes signals in time domain and scans signal with respect to $T_c$, $2T_c$….$GDST_c$. Then the received signal is again multiplied by the same code $C^j(t)$. After this the code has been removed, so we get the original transmitted user data. The low pass filter (LPF) restricts the high frequency portion of data signals. Then the parallel to serial (P/S) converter converts parallel data signal to serial data signal. At last, the integrator provides the real $j^{th}$ user digital data. The multi-access interference (MAI) term can be reduced by the integrator.

But the main drawback of this scheme is the transmitted signal is scattered while passing through the medium. The spreading code is used on time domain and all subcarrier remain relationship of orthogonal.

## 2.2. Multi carrier Code Division Multiple Access (MC-DS-CDMA)

MC-DS-CDMA scheme is a combination of code with different chip distribution and same frequency component with a particular separation with reference channel. It improves the performance such that the whole frequency is divided into several small parts and transferred via air medium. In the receiver side, the received signal is being demodulated and then decoded using Rake receiver. After de-multiplexing the main transmitted signal can be achieved.

### 2.2.1. MC-DS-CDMA Transmitter:

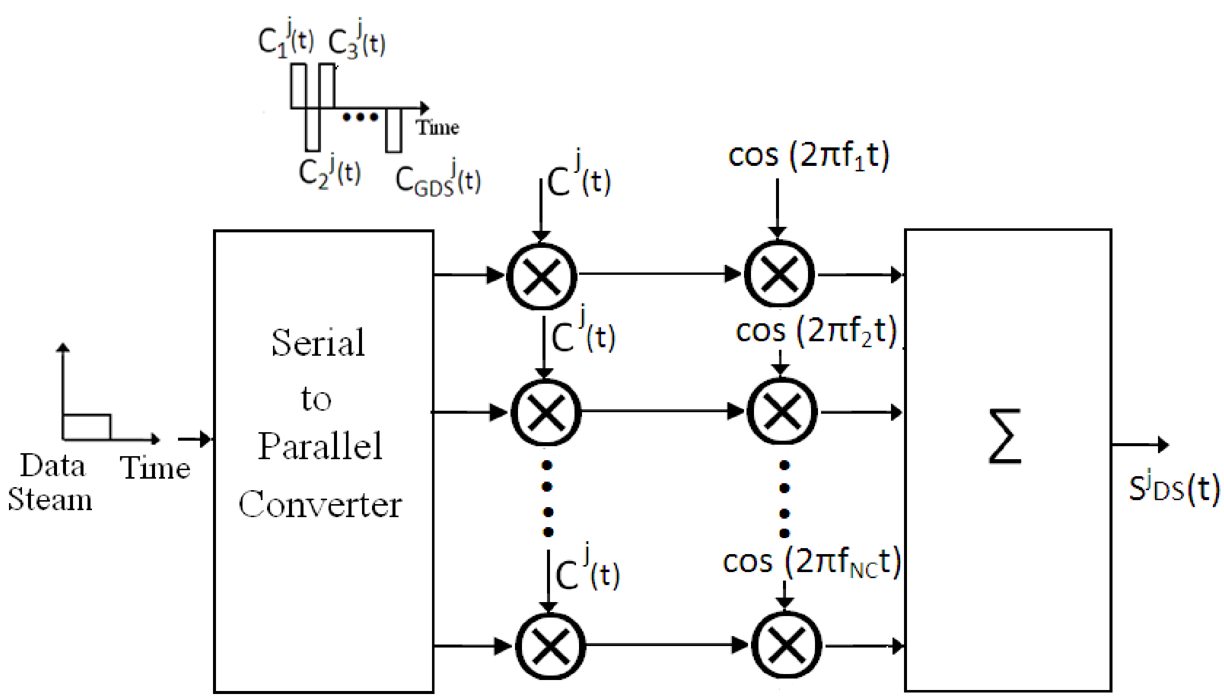


**Figure 1.3.1:** MC-DS-CDMA transmitter

Figure 1.3.1 represents the transmitter of $j^{th}$ user in MC-DS-CDMA scheme, assuming $j^{th}$ user is transmitting. From the MC-DS-CDMA transmitter spreads the original S/P data stream using a given spreading code. Code Generator constructs unique codes for each user as well. Then QPSK modulation is done by the frequency generator. Then a combiner combines the signals and a CDMA antenna transmits the signals over the wireless media.

## 2.2.2. MC-DS-CDMA Receiver:

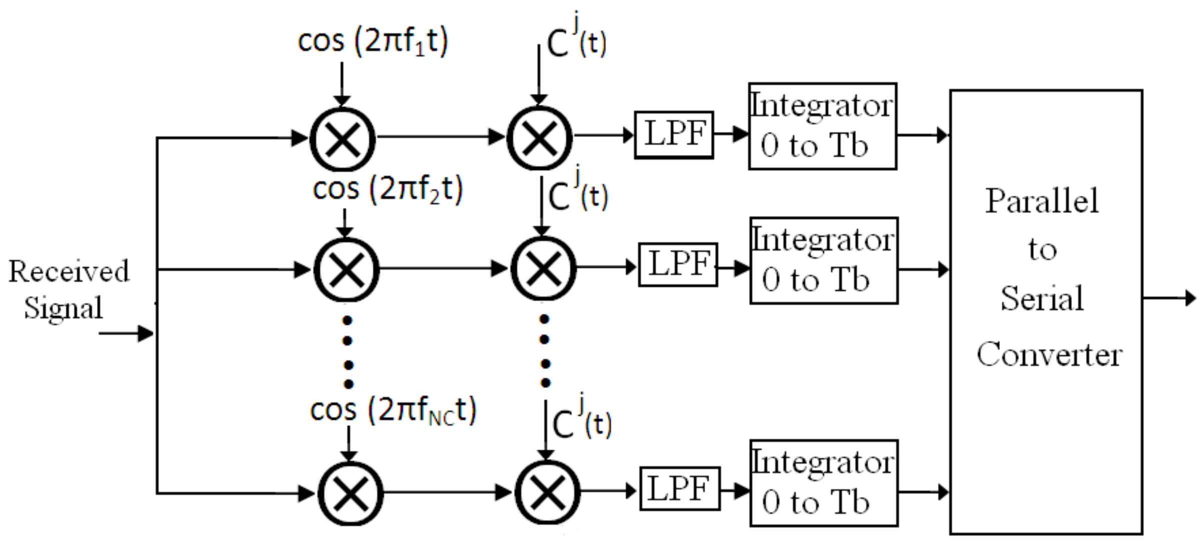


**Figure 1.3.2:** MC-DS-CDMA receiver

Figure 1.3.2 shows the receiver of MC-DS-CDMA scheme, assuming $j^{th}$ user is receiving. MC-DS-CDMA receiver also receives the transmitted signal as a summation of j number of users. It first demodulates the received signal and then dispread the signals with the specific codes given by the receiver code generator. Here, also the code for $j^{th}$ user has to be same both in transmitter and receiver. After that low pass filter restricts the high frequencies portion of the signal and then it modified with the integrator 0 to $T_b$. Finally, the P/S converter presents the actual digital data signal.

Here the spreading code is used on time domain and all subcarrier remain relationship of orthogonal. It is used in uplink and is compatible with IS-95 DS-CDMA at present. Mainly Multi carrier DS-CDMA (MC-DS-CDMA) is a modulation technique that combines OFDM and DS-CDMA.

## 2.3. Multi Tone Direct Sequence (MT-DS-CDMA):

Multi Tone Direct Sequence CDMA (MT-DS-CDMA) is a multiple access technique where a data is transmitted at different frequencies. And at the receiving end, following the frequency selective technique, output of the better frequency is taken.

# Chapter 3

## MT-DS-CDMA Transmitter:

In case of MT-DS-CDMA we are using Nakagami-m fading channel as the intermediate media for data transmission between transmitter and receiver.

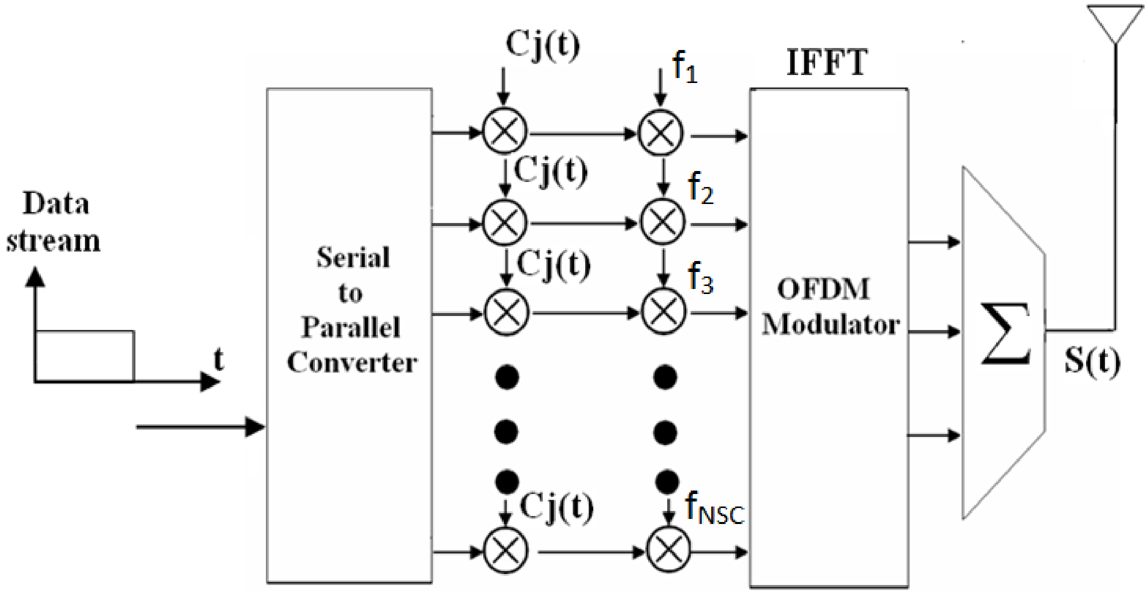


**Figure 1.4:** MT-DS-CDMA transmitter

Figure 1.4 represents the transmitter of $j^{th}$ user in MT-DS-CDMA scheme, assuming $j^{th}$ user is transmitting. In the transmitter side, main data stream is being coded first using pn-sequence code, m-sequence code and gold code. After coding, the encoded data is converted to parallel data streams using serial to parallel converter. Then each data stream is modulated by different frequencies. These frequencies consist of phase difference. Then these parallel streams are entered into the OFDM modulator. Then a combiner combines the signals and a antenna transmits the signals S (t) over the wireless media. The medium of transmission is Nakagami-*m* fading channel.

## 3.1. PN-sequence code:

A Pseudo-random Noise (PN) sequence is a sequence of binary numbers, e.g. ±1, which appears to be random; but is in fact perfectly deterministic. A feature of PN codes is that if the same versions of the PN code are time shifted, then they become almost orthogonal, and can be used as virtually orthogonal codes within a CDMA system.

PN sequences are used for 2 types of spread spectrum techniques:

1) Direct Signal Spread Spectrum (DS-SS)

2) Frequency Hop spread Spectrum (FH-SS)

If Phase Shift Keying (PSK) is used for modulating the PN sequence, then it results in DS-SS and if Frequency Shift Keying (FSK) is used then it results in FH-SS.

### 3.2. M-sequence code:

Shift-register sequences having the maximum possible period for an *r*-stage shift register are called maximal length sequences *or* m-sequences. This code can be generated by doing Ex-OR with two PN sequences code**.**

### 3.3. Gold code:

Gold code can be generated by doing EX-OR with two M-sequence code. Gold codes have bounded small cross-correlations within a set, which is useful when multiple devices are broadcasting in the same range. A set of Gold code sequences consists of $2^n - 1$ sequences each one with a period of $2^n - 1$.

# Chapter 4

## Orthogonal Frequency Division Multiplexing (OFDM):

Orthogonal Frequency Division Multiplexing (OFDM) has been always used in broadcast system. OFDM is a modulation scheme that has gained an immense popularity in the design of wireless communication system. The main advantages of OFDM over a communication system are given below:

- It has high bandwidth efficiency
- It solves the problem of Inter Symbol Interference (ISI)
- It is scalable to high data rates
- It is a flexible modulation scheme
- It is very good at minimizing the effects of time-dispersion

- It does not require channel equalization,
- It does not need phase lock of the local oscillators

## 4.1. OFDM Transmitter:

Suppose that the number of sub carrier is N. In Figure 1.5.1 at the transmitter, the serial data sequence undergoes serial-to-parallel (S/P) conversion to be stacked into one OFDM symbol. After S/P conversion, N-points inverse fast-Fourier transform (IFFT) follows to produce the N dimensional signal in the time domain. The IFFT is at the heart of the OFDM modulation.

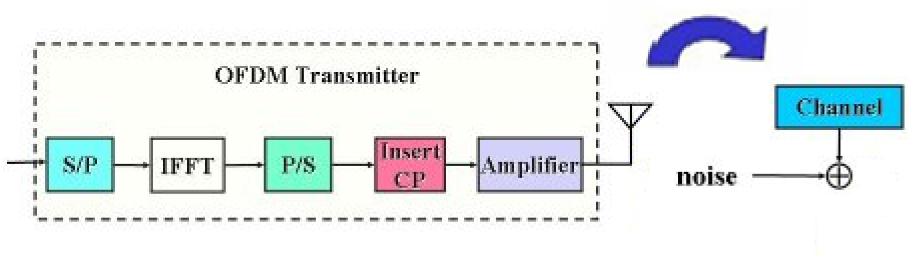


**Figure 1.5.1:** OFDM transmitter

Next, the N dimensional signal is parallel-to-serial (P/S) converted. Note that the time-domain of an OFDM symbol is N times larger than that of a single-carrier system. To moderate interference between closest OFDM symbols caused by the channel time spread and to preserve the orthogonality between the sub carriers, a cyclic prefix (CP) or guard interval of length greater than or equal to the channel length is appended to introduce redundancy.

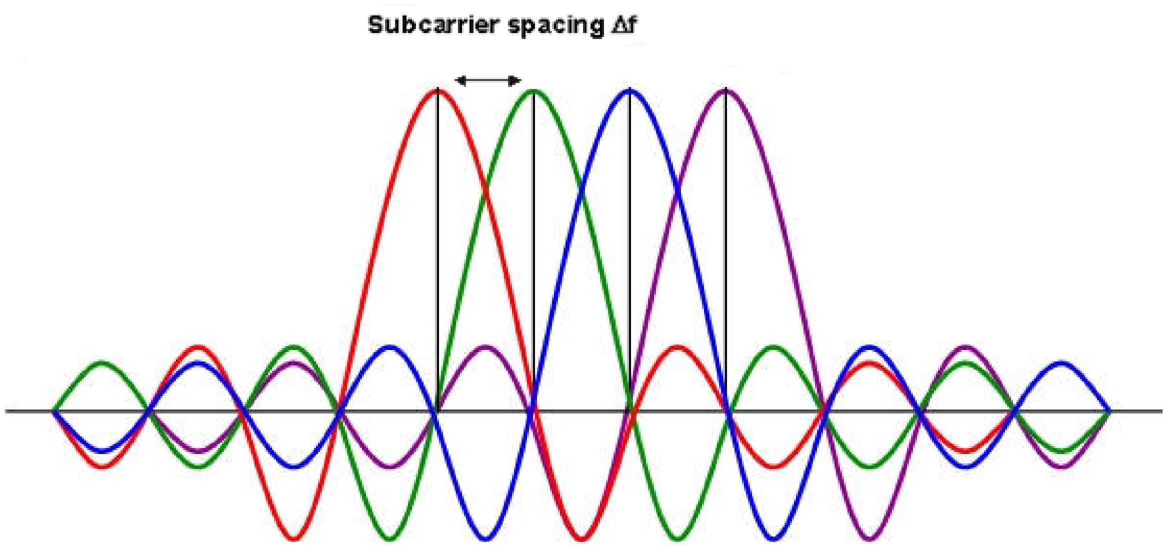


**Figure 1.5.2:** Spectrum of different frequencies

As a result, inter symbol interference (ISI) are completely eliminated. The resultant symbol is then applied to a power amplifier. The OFDM symbol is now ready and it is sent over the channel.

## 4.2. OFDM Receiver:

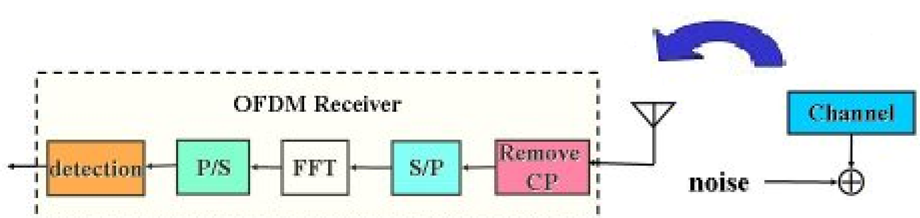

**Figure 1.5.3**: OFDM receiver

In figure 1.5.3 at the receiver, the CP or guard interval is removed from the symbol, and S/P conversion occurs. Then, fast Fourier transform (FFT) is applied to the symbol before P/S conversion take place. The received symbol is then equalized and detected to try to retrieve the original transmitted symbol. This completes our explanation of the modulation, transmission and reception process in OFDM system.

# Chapter 5

## Nakagami-*m* fading channel:

When the frequency of transmitted signal is very high, higher than normal high frequency modulation, then the Nakagami-m fading channel is used. When we modulate the signal, it is normally scattered and it becomes difficult to reconstruct. If Nakagami-*m* fading channel is used, this channel re-collects the scattered signal and then passes to the receiver antenna.


**Figure 1.6:** Nakagami-*m* fading channel

# Chapter 6

## MT-DS-CDMA RECEIVER:

In the receiver side, the re-collected frequencies from the fading channel are received by antenna and demodulated multiplying with respected modulation frequencies. Then each demodulated frequencies are passed through Rake Combiner.

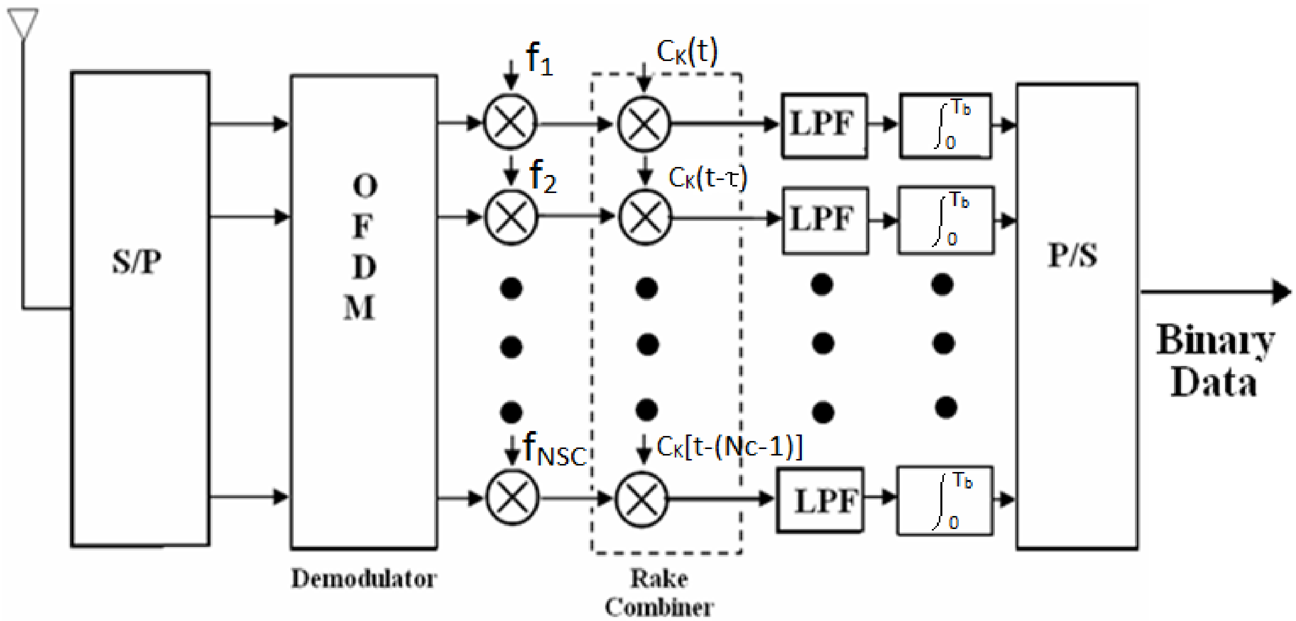


**Figure 1.7:** MT-DS-CDMA receiver

Then the signal should be passed through low pass filter (LPF) to reduce the higher frequency components. By integrating the signal with the integrator 0to $T_b$ the total separated signals can be found. After that to gain the total signal together there is a parallel to serial converter added.

## 6.1. Rake Combiner:

A rake combiner for a CDMA rake receiver, the combiner comprising a receiver for receiving a plurality of multipath components of a signal; a memory; a controller arranged to store a first multipath component in the memory; a summer for summing the first multipath component with a second multipath component to provide a combined signal; wherein the controller is arranged to store the combined signal in the memory. That is, Rake combiner works as multipath coding when the signal is arriving directly, by refection or after distorting, this combiner multiplies the respected code by directly and by delaying the code, as needed for de-coding.

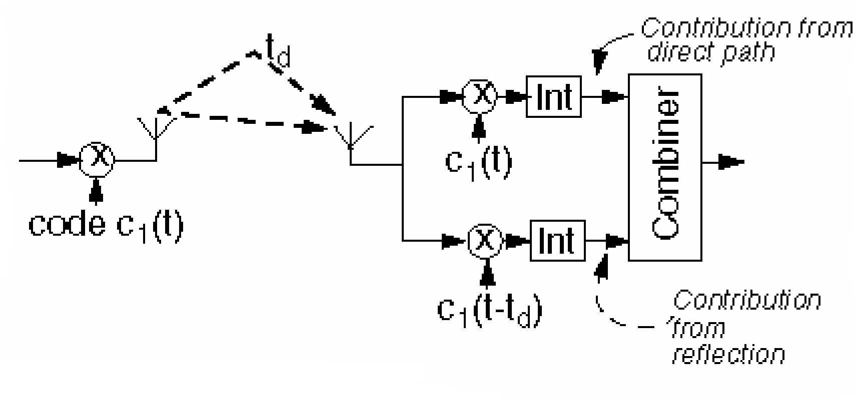


**Figure 1.8:** Rake receiver

After the decoding process occurs, all the signals are then converted to serial data stream using parallel to serial converter. Then we can get the original signal. Of course there will be interferences and noise with the desired signal. Our vision is to design the optimum system in noisy environment.

We have derived the expression for transmitted signal and received signal. In the received signal, there are MAI and noise signal at the same time with desired signal.

# Chapter 7

# Derived Equations:

## 7.1. Symbol expressions:

$d_n$ = Data to be transmitted

i = No. of sub-carrier = 1 to $N_{sc}$

j = No. of user = 1to k

m(t) = Message signal

$S_k(t)$ = Transmitted Signal

$C_k(t)$ = Code added with the signal

r(t) = Received Signal

$$\sum_{x=0}^{Nsc-1} C_k(t - x\tau) = \text{Delayed code in receiver end}$$

$$\eta(t) = \text{Additive White Gaussian Noise (AWGN)}$$

$\theta$ = Phase distortion

$\alpha_D$ = Amplitude distortion

μ = Cross correlation

$\emptyset_K = Phase\ Angle$

## 7.2. Transmitted signal:

### 7.2.1. Data stream:

$$m(t) = \sum_{-\infty}^{\infty} d_n$$

### 7.2.2. Signal of Subcarrier:

General expression for subcarriers:

$$S_C = \sum_{i=1}^{N_{SC}} \sqrt{2P}\, d_n \cos(\omega_i t) \qquad ;\ \omega_i = [\omega_1 + (i-1)\Delta\omega]$$

$$\Delta\omega = 2\pi\Delta f = 2\pi R_s$$

### 7.2.3. Ultimate transmitted signal:

Expression of transmitted signal of k-th user:

$$S_j\,(t) = \sum_{i=1}^{N_{SC}} S_{i,j}\,(t)$$

$S_i\,(t)$ = i-th sub-carrier modulated signal

$$S_j(t) = \sum_{i=1}^{N_{SC}} \sqrt{2P} d_n^k \times C_K(t) \times cos\omega_i\,(t)$$

### 7.3. Received signal:

$$r(t) = \sum_{j=1}^{k} r_j\,(t) = \sum_{j=1}^{k}\sum_{i=1}^{N_{SC}} r_{i,j}\,(t) + \eta\,(t)$$

$$= \sum_{j=1}^{k}[\sqrt{2P}\sum_{i=1}^{N_{SC}} d_{n,i}^{j}\;\left[\sum_{x=0}^{N_{SC}} C_K(t - x\tau)\right]\cos\omega_i\,(t)] + \eta\,(t)$$

$$= \sum_{j=1}^{k}[\alpha_D\sqrt{2P}\sum_{i=1}^{N_{SC}} d_{n,i}^{j}\;\left[\sum_{x=0}^{N_{SC}} C_K(t - x\tau)\right]\cos[\,\omega_i\,(t) + \theta]] + \;\eta(t)$$

#### 7.3.1. With frequency offset due to Doppler effect:

$$r(t) = \sum_{j=1}^{k}\Big[\alpha_D\sqrt{2P}\sum_{i=1}^{N_{SC}} d_{n,i}^{j}\;\left[\sum_{x=0}^{N_{SC}} C_K(t - x\tau)\right]\cos[\omega_1 t + (i-1)\Delta\omega t + \theta + \emptyset_K\; + 2\pi\delta f(t)]\Big] + \eta(t)$$

#### 7.3.2. Coherently de-modulated signal by the carrier and output y(t):

$$y(t) = \sum_{j=1}^{k}[\alpha\sqrt{2P}\sum_{i=1}^{N_{SC}} d_{n,i}^{j}\;\left[\sum_{x=0}^{N_{SC}} C_K(t - x\tau)\right]\cos\{\omega_1 t + (i-1)\Delta\omega t + \theta + \emptyset_K\; + 2\pi\delta f(t)\} \times\;\cos\{\omega_1 t\; + (i-1)\Delta\omega t\}] + \;\eta\,(t) \times \sum_{i=1}^{N_{SC}} \cos\{\omega_1 t + (i-1)\Delta\omega t\}$$

### 7.3.3. Decoded signal:

$$y(t) = \sum_{j=1}^{k} \frac{\alpha_D \sqrt{2P}}{2} [ \sum_{i=1}^{N_{SC}} d_{n,i}^{j} \left[ \sum_{x=0}^{N_{SC}} C_K(t - x\tau) \times C_K(t - x\tau) \right] [\cos\{2\omega_1 t + 2(i-1)\Delta\omega t + \emptyset_k + \theta + 2\pi\delta f(t)\} + \cos\{\theta + \emptyset_k + 2\pi\delta f(t)\}] ] + \eta(t) \times C_K(t - x\tau) \times \sum_{i=1}^{N_{SC}} \cos\{\omega_1 t + (i-1)\Delta\omega t\}$$

### 7.3.4. Separating K-th user term and assuming 'μ' as the cross-correlation among the codes of K-th user and all others:

$$y(t) = \frac{\alpha_D \sqrt{2P}}{2} \sum_{i=1}^{N_{SC}} d_{n,i}^{k} (1) \times [\cos\{2\omega_1 t + 2(i-1)\Delta\omega t + \theta + \emptyset_k + 2\pi\delta f(t)\} + \cos\{\theta + \emptyset_k + 2\pi\delta f(t)\}] + \sum_{j=1}^{k-1} \frac{\alpha_D \sqrt{2P}}{2} \sum_{i=1}^{N_{SC}} d_{n,i}^{j}\, \mu [\cos\{2\omega_1 t + 2(i-1)\Delta\omega t + \theta + \emptyset_k + 2\pi\delta f(t)\} + \cos\{\theta + \emptyset_k + 2\pi\delta f(t)\}] + \eta(t) \times C_K(t - x\tau) \times \sum_{i=1}^{N_{SC}} \cos\{\omega_1 t + (i-1)\Delta\omega t\}$$

If interference is maximum, then $\mu = 1$ and

If interference is zero, then $\mu = 0$.

### 7.3.5. Passing through LPF, high frequency components will be eliminated:

$$y(t) = \frac{\alpha_D \sqrt{2P}}{2} \sum_{i=1}^{N_{SC}} d_{n,i}^{k} [\cos(\theta + \emptyset_K + 2\pi\delta f(t))] + \sum_{j=1}^{k-1} [\frac{\alpha_D \sqrt{2P}}{2} \sum_{i=1}^{N_{SC}} d_{n,i}^{j} (\mu) \times \cos(\theta + \emptyset_K + 2\pi\delta f(t)) ] + \eta(t) \times C_K(t - x\tau) \times \sum_{i=1}^{N_{SC}} \cos\{\omega_1 t + (i-1)\Delta\omega t\}$$

## 7.3.6. Integrating over 0 to $T_b$ :

$$y(t) = \frac{\alpha_D\sqrt{2P}}{2T_b}\int_0^{T_b}[\sum_{i=1}^{N_{SC}} d_{n,i}^k \times \cos\{\theta + \emptyset_K + 2\pi\delta f(t)\}]\,dt + \sum_{j=1}^{k-1}[\frac{\alpha_D\sqrt{2P}}{2T_b}\int_o^{T_b}\sum_{i=1}^{N_{SC}} d_{n,i}^j\,(\mu) \times \cos\{\theta + \emptyset_K + 2\pi\delta f(t)\}dt] + \frac{1}{T_b}\int_0^{T_b}\eta\,(t)\times C_K(t-x\tau) \times \sum_{i=1}^{N_{SC}}\cos\{\omega_1 t + (i-1)\Delta\omega t\}$$

## 7.3.7. Assuming offset = 0:

$$y(t) = \frac{\alpha_D\sqrt{2P}}{2}\sum_{i=1}^{N_{SC}} d_{n,i}^k\cos(\theta + \emptyset_K)\,[\frac{1}{T_b}\int_0^{T_b} dt] + \sum_{j=1}^{k-1}\mu \times \frac{\alpha_D\sqrt{2P}}{2}\sum_{i=1}^{N_{SC}} d_{n,i}^j\cos(\theta + \emptyset_K)[\frac{1}{T_b}\int_0^{T_b} dt] + \frac{1}{T_b}\int_0^{T_b}\eta\,(t)\times C_K(t-x\tau) \times \sum_{i=1}^{N_{SC}}\cos\{\omega_1 t + (i-1)\Delta\omega t\}dt$$

## 7.3.8. Result of integration:

$$y(t) = \frac{\alpha_D\sqrt{2P}}{2}\sum_{i=1}^{N_{SC}} d_{n,i}^k\cos(\theta + \emptyset_K) + \sum_{j=1}^{k-1}\mu \times \frac{\alpha_D\sqrt{2P}}{2}\sum_{i=1}^{N_{SC}} d_{n,i}^j(\cos(\theta + \emptyset_K)) + \frac{1}{T_b}\int_0^{T_b}\eta\,(t)\times C_K(t-x\tau) \times \sum_{i=1}^{N_{SC}}\cos\{\omega_1 t + (i-1)\Delta\omega t\}dt$$

$y(t) = y_j(t) + y_{MAI}(t) + n_0(t)$

$y_j(t) = Desired\ signal$

$y_{MAI}(t) = MAI\ signal$

$n_0$ (t) = noise signal

## 7.4.1. Desired signal:

$$y_j(t) = \alpha_D \sqrt{\frac{P}{2}} \sum_{i=1}^{N_{SC}} d_{n,i}^{k} \cos\left(\theta + \emptyset_K\right)$$

$$Since\ d_n^k = \pm 1\ and \sum_{i=1}^{N_{SC}} d_{n,i}^{k} = N_{sc}$$

Therefore,

$$y_j(t) = \alpha_D N_{sc} \sqrt{\frac{P}{2}} \cos\theta$$

## 7.4.2. Desired power signal:

$$P_k = \frac{1}{2} \alpha_D^2 \ Nsc^2 \ \frac{P}{2}$$

$$= \frac{1}{4} \alpha_D^2 \ Nsc^2 P$$

## 7.5.1. MAI:

$$Y_{MAI}(t) = \sum_{j=1}^{k-1} \mu \alpha_D \sqrt{\frac{P}{2}} \sum_{i=1}^{Nsc} d_{n,i}^{i} \cos(\theta + \emptyset_K)$$

## 7.5.2. MAI Power:

$$P_{MAI} = \sum_{j=0}^{k-1} [\mu^2 \alpha_D^2 \ \frac{P}{2} \ Nsc^2]$$

$$= \frac{1}{4} \mu^2 \alpha_D^2 \ P(k-1) Nsc^2$$

## 7.6.1 Noise:

$$n_0(t) = \frac{1}{T_b} \int_0^{T_b} \omega(t) C_k(t - x\tau) \sum_{i=1}^{Nsc} \cos[\omega_1 t + (i-1)\Delta\omega t]\, dt$$

Now it can be evaluated. Noise power is:

$\sigma_n^2 = \frac{N_0}{4T_b}$ ;

$N_0 = KTR_b$ ; Where K= Boltzmann constant

T= Temperature

$R_b$ = Bit rate

## 7.6.2. SINR:

Signal to Interference & Noise Ratio (SINR):

$$SINR = \frac{P_k}{P_{MAI} + \sigma_n^2}$$

$$= \frac{\frac{1}{4}\ \alpha_D^2\ Nsc^2\ P}{\frac{1}{4}\mu^2\ \alpha_D^2\ \ P(k-1)\ Nsc^2 + \frac{N_0}{4T_b}}$$

$$= \frac{\alpha_D^2\ \ Nsc^2 P\ T_b}{\mu^2\ \ \alpha_D^2\ \ Nsc^2\ P(k-1)\ T_b + N_0}$$

$$= \frac{\alpha_D^2\ \ Nsc^2 E_b}{\mu^2\ \ \alpha_D^2\ \ Nsc^2 E_b\,(k-1) + N_0}$$

$$= \frac{\alpha_D^2\ \ Nsc^2}{\mu^2\ \ \alpha_D^2\ \ Nsc^2\,(k-1) + (\frac{E_b}{N_0})^{-1}}$$

$$SINR\ = \frac{\alpha_D^2\ \ Nsc^2}{\mu^2\ \ \alpha_D^2\ \ Nsc^2(k-1) + (\frac{E_b}{N_0})^{-1}}$$

## 7.6.3. BER:

SINR is a function of α. So for an instantaneous value of α, the instantaneous Bit error rate (BER) is

$$P_b(\alpha) = f\ (SINR)$$

$$= 0.5 * erfc\ (SINR)$$

$$= 0.5 * erfc\ [\frac{\alpha_D^2\ Nsc^2}{\mu^2\ \alpha_D^2\ Nsc^2\ (k-1) + (\frac{E_b}{N_0})^{-1}}]$$

Where,

$\alpha_D$ = amplitude distortion

*Nsc* = number of subcarrieres

$\mu$ = cross-correlation co-efficient

*k* = number of user

*(Eb/No)* = 6db, 10db, 12db …

Now the average bit error rate of the system without rake can be obtained as follows:

$$P_b = \int_0^{\infty} P_b(\alpha)\ f(\alpha) d\alpha$$

Where,

$P_b(\alpha)$ = Instantaneous BER

$f(\alpha)$ = pdf of amplitude distortion coefficient for Nakagami–*m* fading,

or, $f_\alpha(x) = \frac{x}{\sigma_\alpha^2} \exp(-\frac{x^2+\alpha_0^2}{2\sigma_\alpha^2})\ I_0\ (\frac{\alpha_0 x}{\sigma_\alpha^2})\quad ; x \geq 2$

Where $\alpha_0$ is $\alpha_0\ (t)$ at any t.$\alpha_0^2$, is the power of the LOS component and is the non-centrality parameter, $I_0$ (.) is the zero-order modified Bessel function of the first kind and is given by

$$I_0(x) = \frac{1}{2\pi} \int_0^{2\pi} \exp(x\cos\theta)\ d\theta$$

We have considered x=3 for Nakagami-m fading. Then we have multiplied this $I_0$ with function of x.sss

$$\int P_b(\alpha_i) * f(\alpha_i) = \sum_{i=1}^{N=256} P_b(\alpha_i) * f(\alpha_i)$$

# Chapter 8

# Result and discussion

We have taken the value of α from 1.0 to 10. The value of Nsc was taken as power of two, starting from 8. Also the value of user we took was started from 1 up to 80. We took the Eb/No db value, and those have started from 6dB up to 54 dB. µ value was taken as 0.05 up to 0.5 to get better curve. Because the smaller we take the value of µ, more our curves will be better. For our curves, we have considered the standard video and data measurement which is BER of $10^{-6}$ dB. As we know, normal voice measurement can tolerate up to $10^{-3}$ BER. For data we can have $10^{-9}$ BER and for video (in some cases data also) we can have $10^{-6}$ BER.

So we have considered here the standard measurement for video and data.

Our first curve is about BER vs. Eb/No. We have seen the performance which should have different Eb/No dB value.

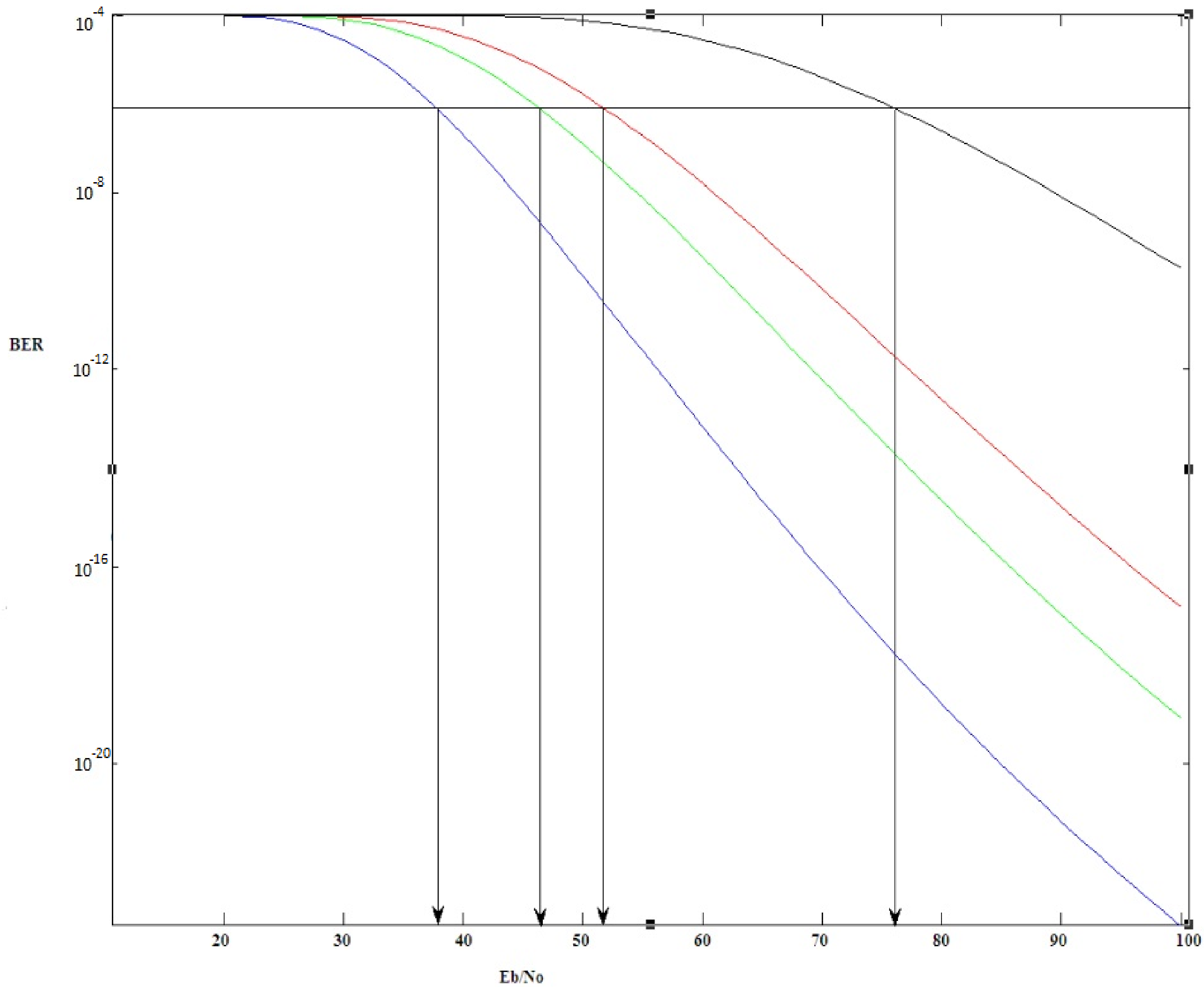


**Figure 1**: BER vs. Eb/No

From this figure above, we can see that for different values of μ we are getting the values of BER.

This figure shows us as we decrease the values of μ, Eb/No requirement is decreasing. From the curve we can see that for lowest value of μ, required Eb/No is around 37 dB. For next higher value of μ, Eb/No requirement is around 52 dB. This way, for highest value of μ, required Eb/No is around 76 dB. This way, with increasing value of μ, Eb/No is increasing.

Our next graph compares the BER with the number of users.

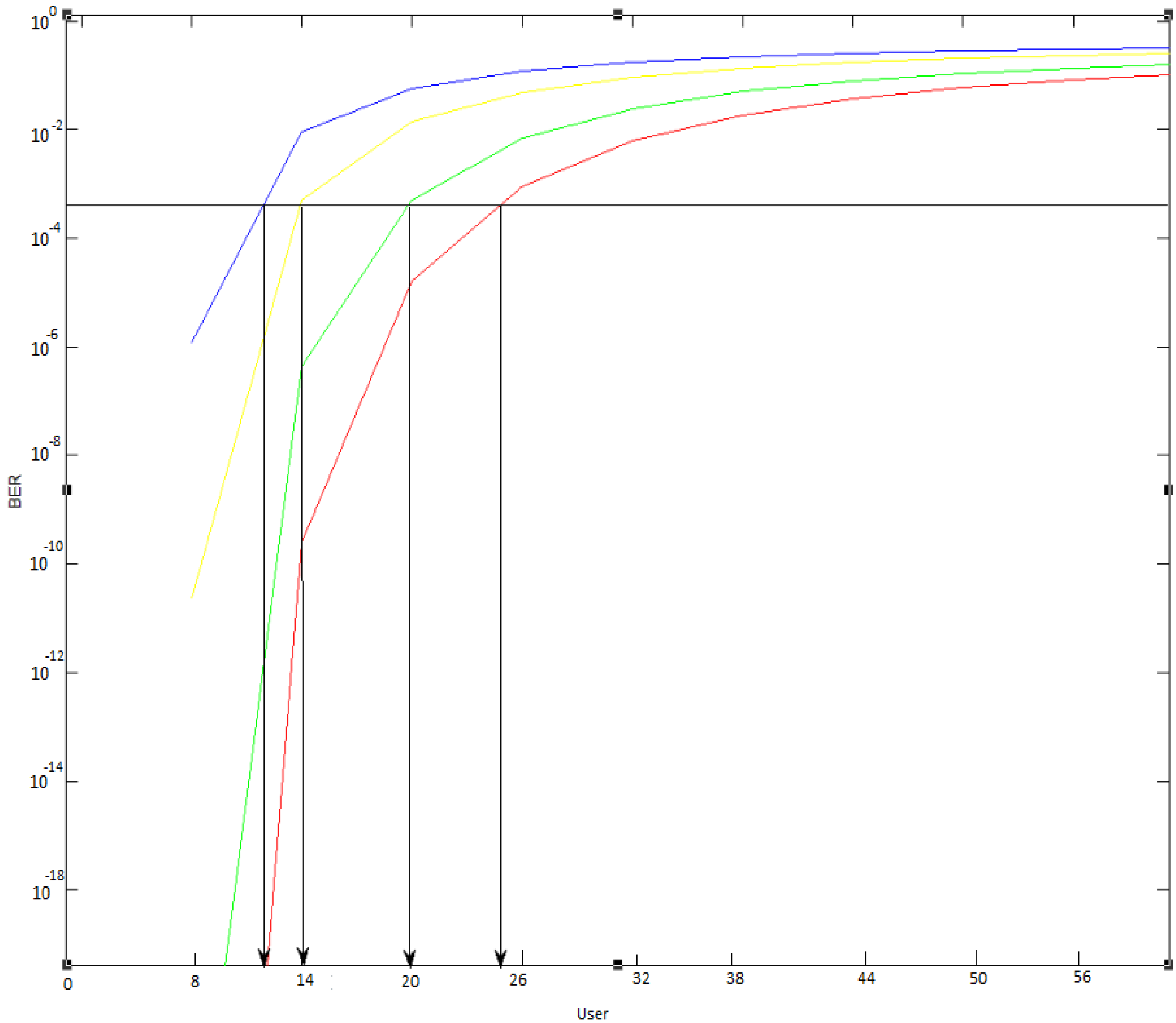


**Figure 2**: BER vs. User for Eb/No= 6 dB.

From this figure, we can see that, for our fixed Eb/No (which is 6 dB), number of user is increasing for a particular μ. As we can see, when the μ value is lowest, user is highest. And reverse order is applicable for higher μ value, where the user number will be lowest. Like we can see from our graph, when μ is taken as 0.05, it supports maximum number of users. That is 24.Then again, when μ value is highest that is 0.085; the number of user supported is being degraded to 12.

From this relation we can come to an end that when μ value increases, due to increase of interference as well, our total number of user decreases.

Our next graph explains clearly the relation of BER with user for another required value of Eb/No. Here we have taken Eb/No value 10dB, where as previous value was 6 dB. So the performance is also degraded due to the change in Eb/No value.

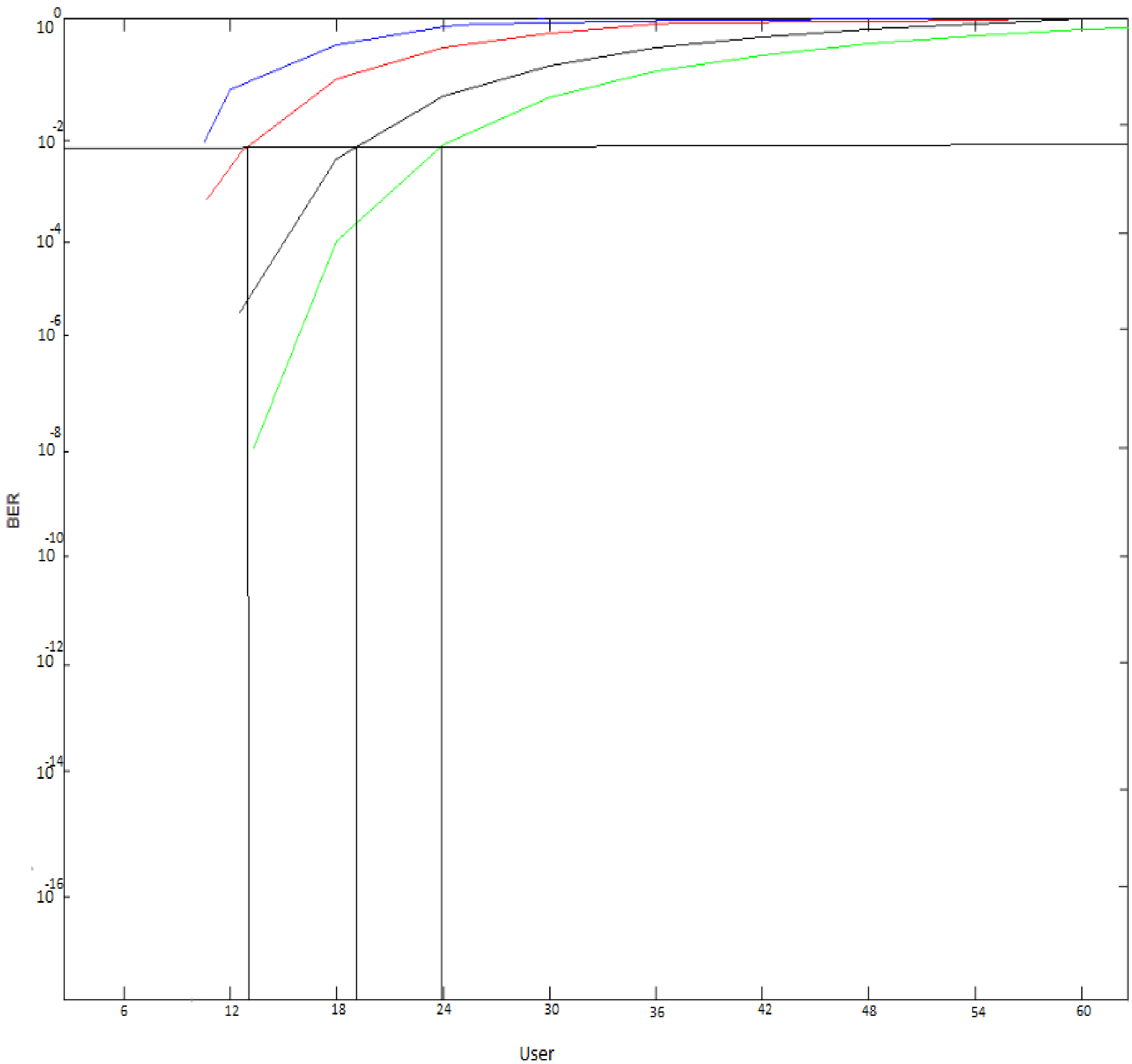


**Figure 3**: BER vs. User for Eb/No= 10 dB.

From the figure above, we can mention that, using Eb/No 10 dB, for μ value 0.05, supported user number is 24. And in the highest case, where the value of μ is 0.085, maximum supported user number is approximately 13. So we can say that this degradation is done due to the change in Eb/No value.

Our next graph shows the relation with user and μ. Here we used different subcarrier channels. As mentioned earlier, we have taken the values of channel as power of two, which have started from 8 up to 32.

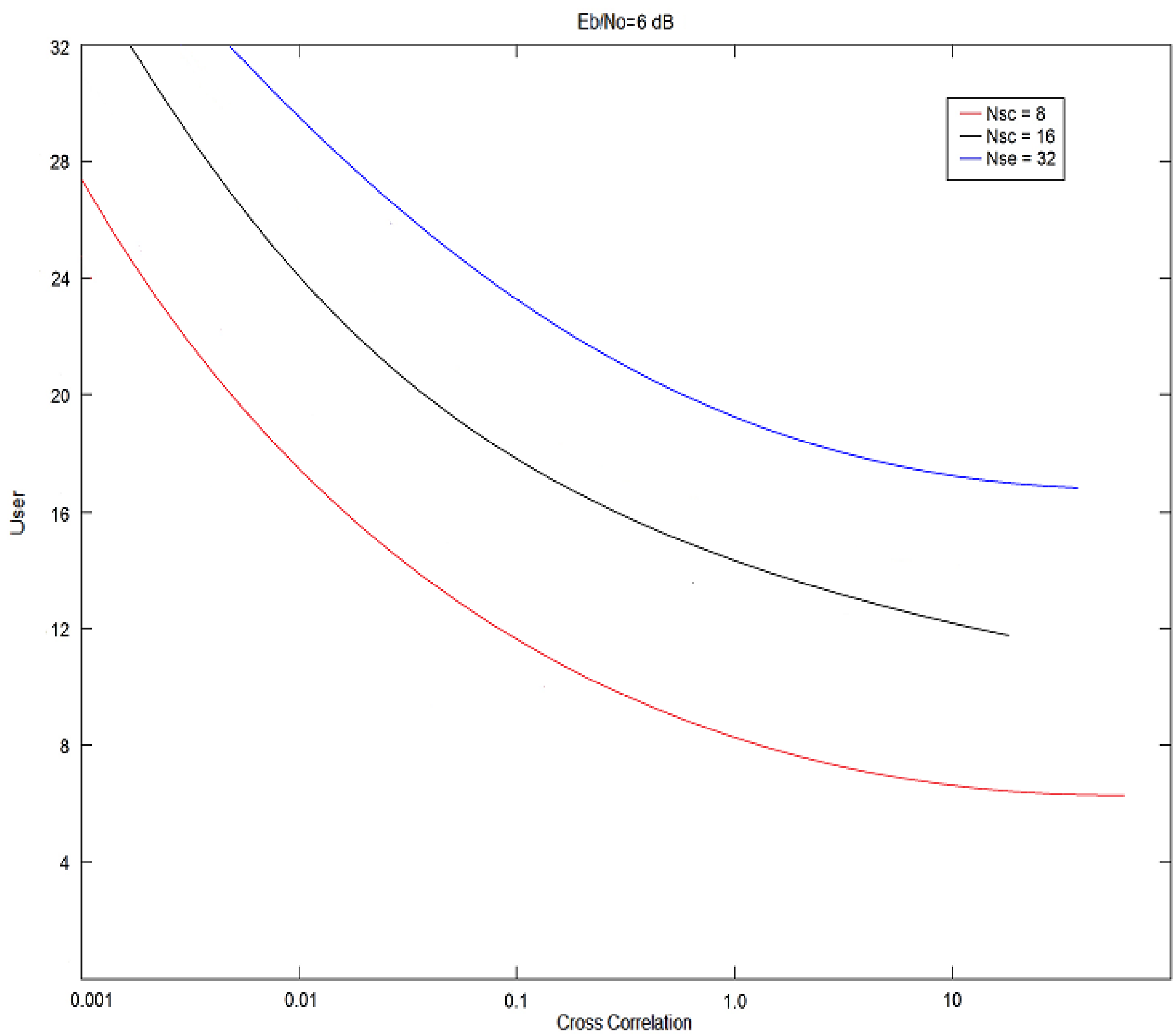


**Figure 4**: User vs. μ with Eb/No=6 dB.

From the above figure, we can see that, for the value of μ =10, for subcarrier channel 8; user supported by the system is approximately 7. For the same μ value, for another subcarrier, Nsc=16, supported user number increases to 14 (approximately). And for subcarrier Nsc= 32, supported user number is almost 18. Using same procedure, we can find the number of users for μ value 0.001. We can see that the performance is being better. That is, for lowest subcarrier, supported number of user is almost 17. For subcarrier 16, number of user increases to 23 almost. And finally for carrier 32, number of user is highest for this particular value of μ that is almost 30. It proves that, due to decrease of μ value, supported user number increases for different subcarrier channels.

In next figure we have shown the effect of adding the probability density function of amplitude distortion coefficient for Nakagami-*m* fading.

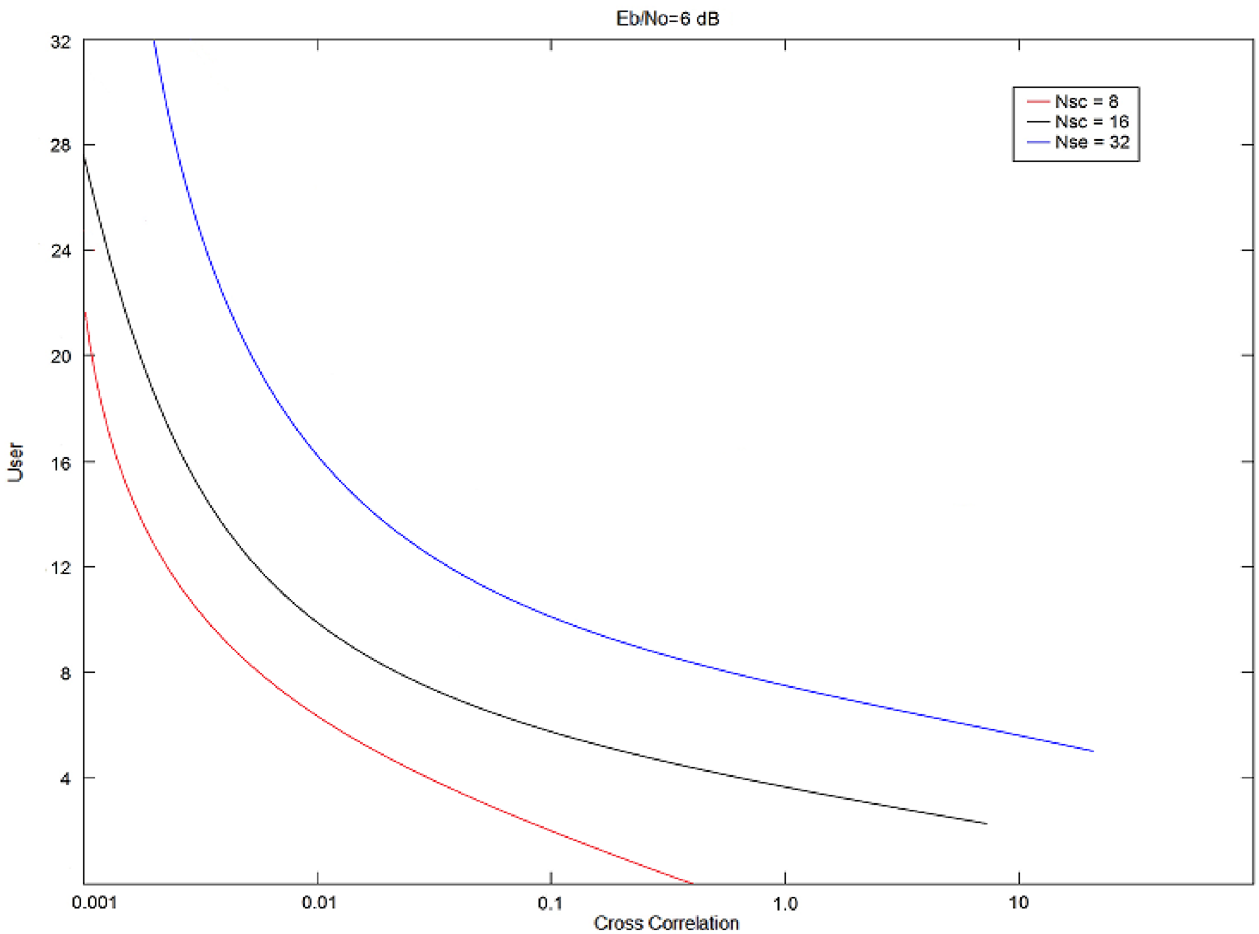


**Figure 5**: User vs. μ after adding Pdf of amplitude distortion co-efficient fading.

We can clearly see from the figure that, after adding the pdf, the curves supports fewer users than before. As we can tell, before adding pdf, at μ value 10, our user was almost 7. Whereas now, after adding pdf, supported user number is less than 5. In case of μ value 0.01, where previously, supported user for subcarrier channels 8, 16, 32 was 17, 23, 30.but now we can see that user number has become 7, 10 and 17. This has surely made the performance less efficient.

Our next figure will be to explain the effect, where Eb/No value is changed to 10dB. Where as previously we used the value Eb/No=6dB.

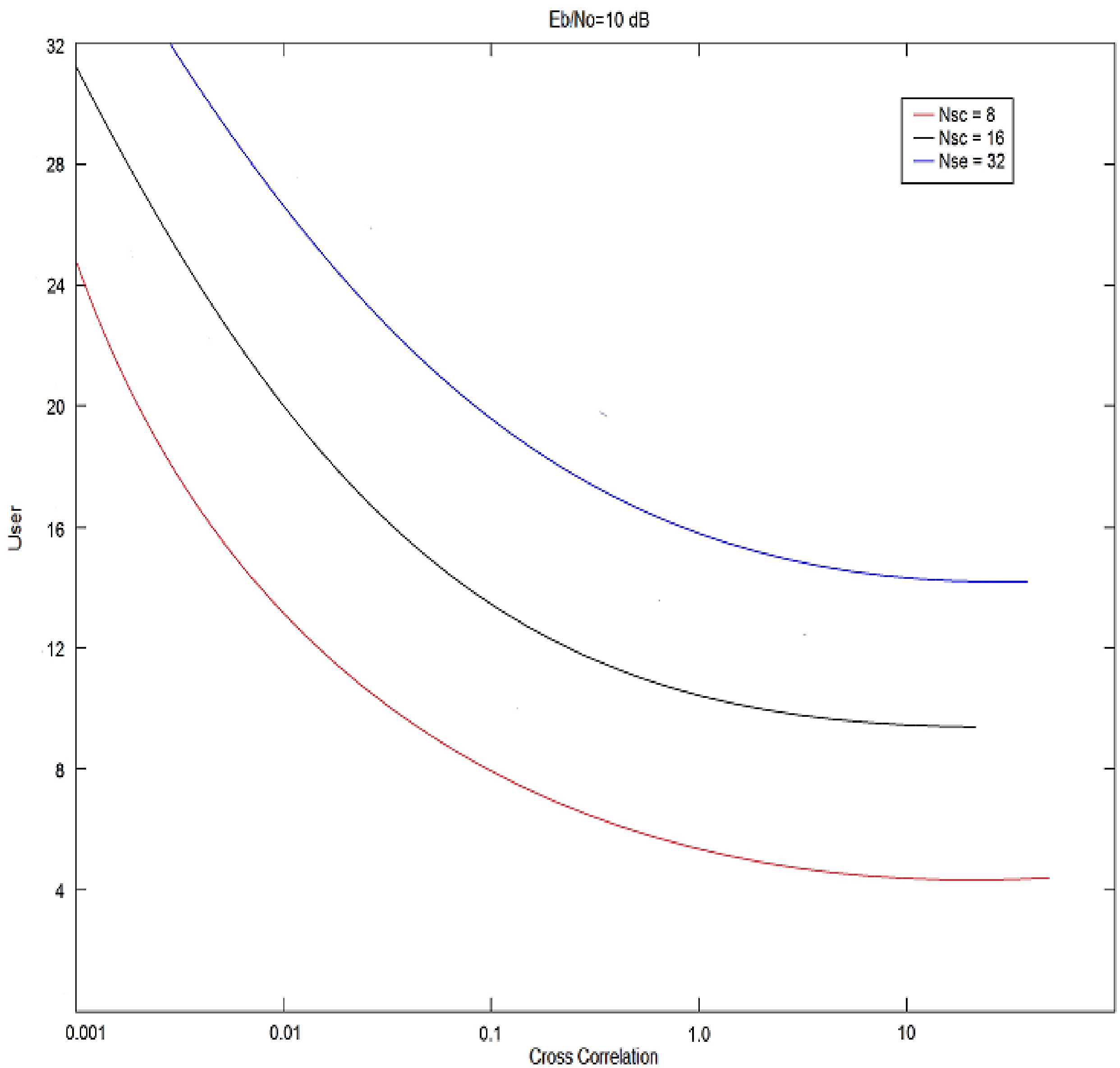


**Figure 6**: User vs. μ for 10 dB without fading affect.

From the above figure, we can see that, when we have increased the value of required value of Eb/No to 10 dB, performance has more degraded. We can take an example of that. When we are taking the value 0.01 of µ value, 13, 20 and 26 are the supported number of users. Where as in case of 6 dB, we had users 17, 23 and 30 for the 0.01 µvalue. So we can clearly say that the performance is being going downwards with the increase of required Eb/No value.

Our next figure we found is, when we added the probability density function of amplitude distortion coefficient for Nakagami-m fading. Here we also can see the difference of the curve with that of without using pdf. The user number for a particular given channel is being decreased. Then again, due to increase of required Eb/No value, the user number is being affected badly.

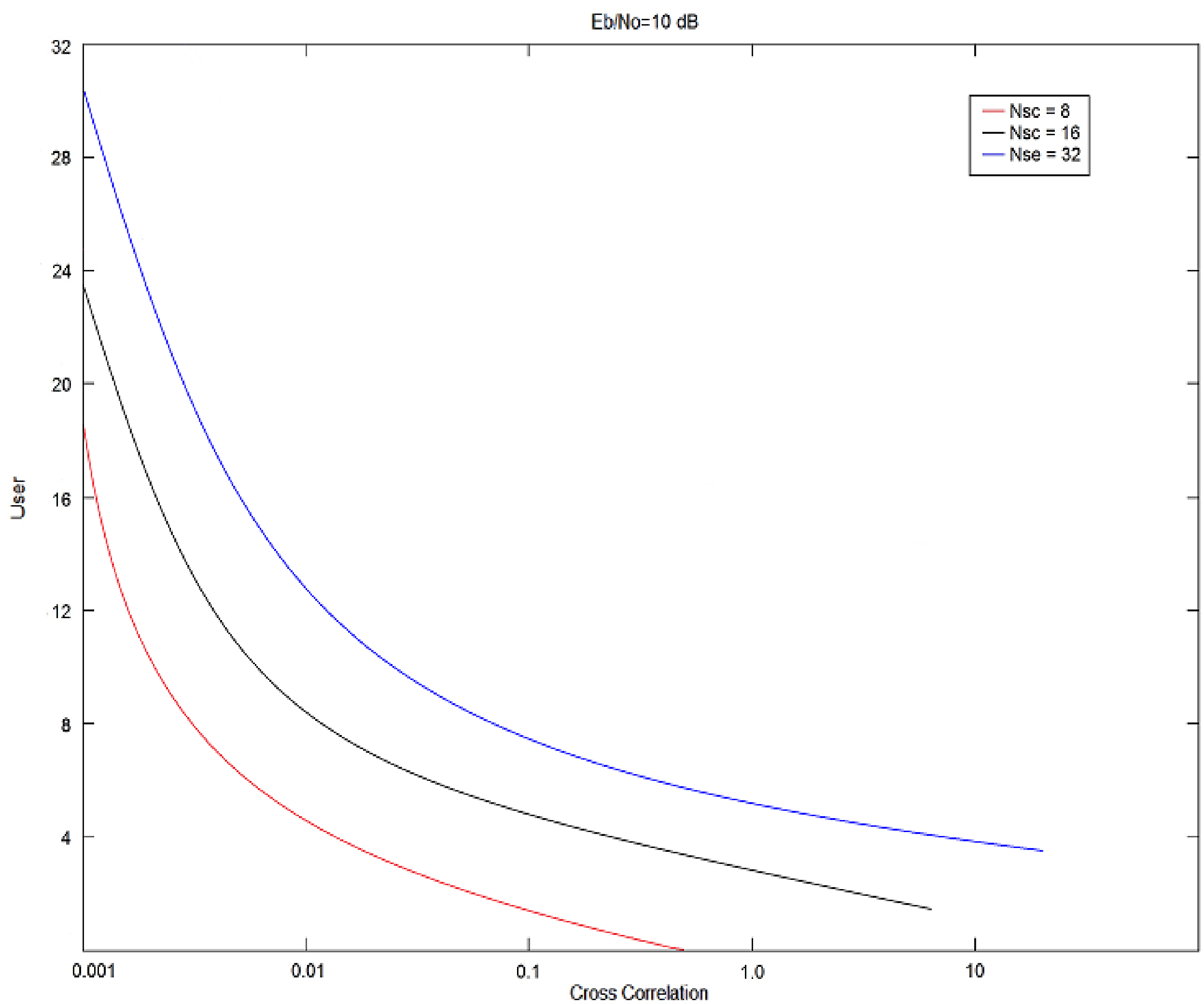


**Figure 7**: User vs. µ for Eb/No=10dB after adding fading.

From the above figure, we can take an example as well. That is, when we are considering the channels Nsc = 8, 16, 32, and Eb/No is fixed and the value is 10dB, for different $\mu$ value, our estimated user number is different.

Like when we are considering the value of $\mu$ say 0.1, for different subcarriers, supported user numbers are 2, 6 and 8 respectively.

Then again, when we decrease the value of $\mu$, say 0.001, then performance gets better. We can see from these curves. When subcarriers are 8, 16 and 32, our supported number users are 5, 9 and 14 respectively. This also satisfies our calculated equation.

# Chapter 9

## Advantages and Comparison of the results found in MT-DS-CDMA with that of MC-DS-CDMA:

Because we have found the results for our thesis work, known as MT-DS-CDMA, for analyzing the performance we can make some comparisons among MC-DS-CDMA and MT-DS-CDMA to understand the performances clearly.

First, we can consider in case of the number of users. That is, when we are considering Eb/No = 6 dB.

From these figures below, we can come to an end, that when the Eb/No ratio is fixed at a certain value, the number of supported users are more in case of MT-DS-CDMA.

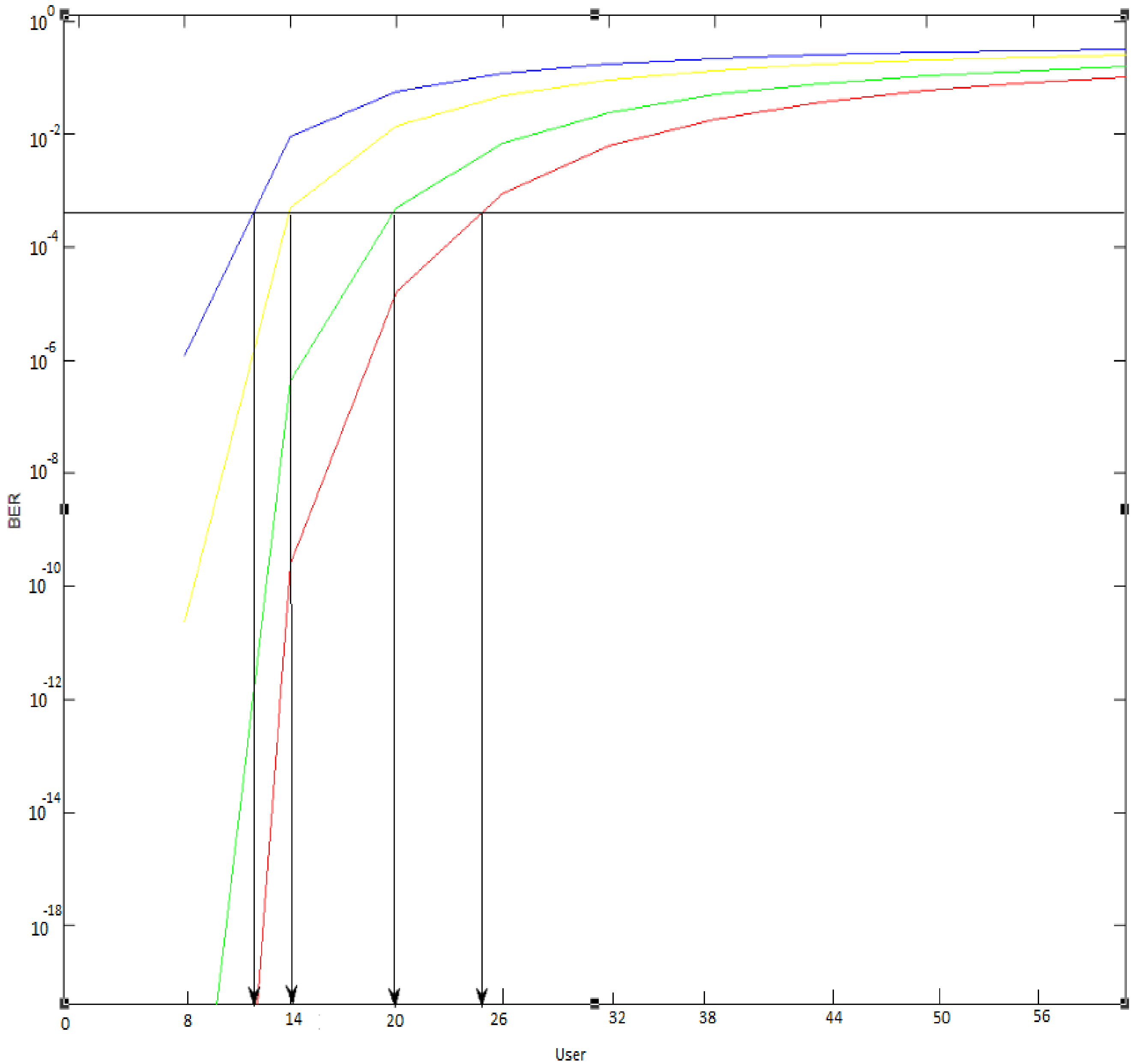


**Figure 8**: For fixed Eb/No, supported number of users for MT-DS-CDMA.

In the figure, for lowest μ value, supported number of user is approximately 25, where as for MC-DS-CDMA is approximately 12. Then again, for maximum μ value, supported user number degrades, but still in case of MT-DS-CDMA, user supported is higher than MC-DS-CDMA.

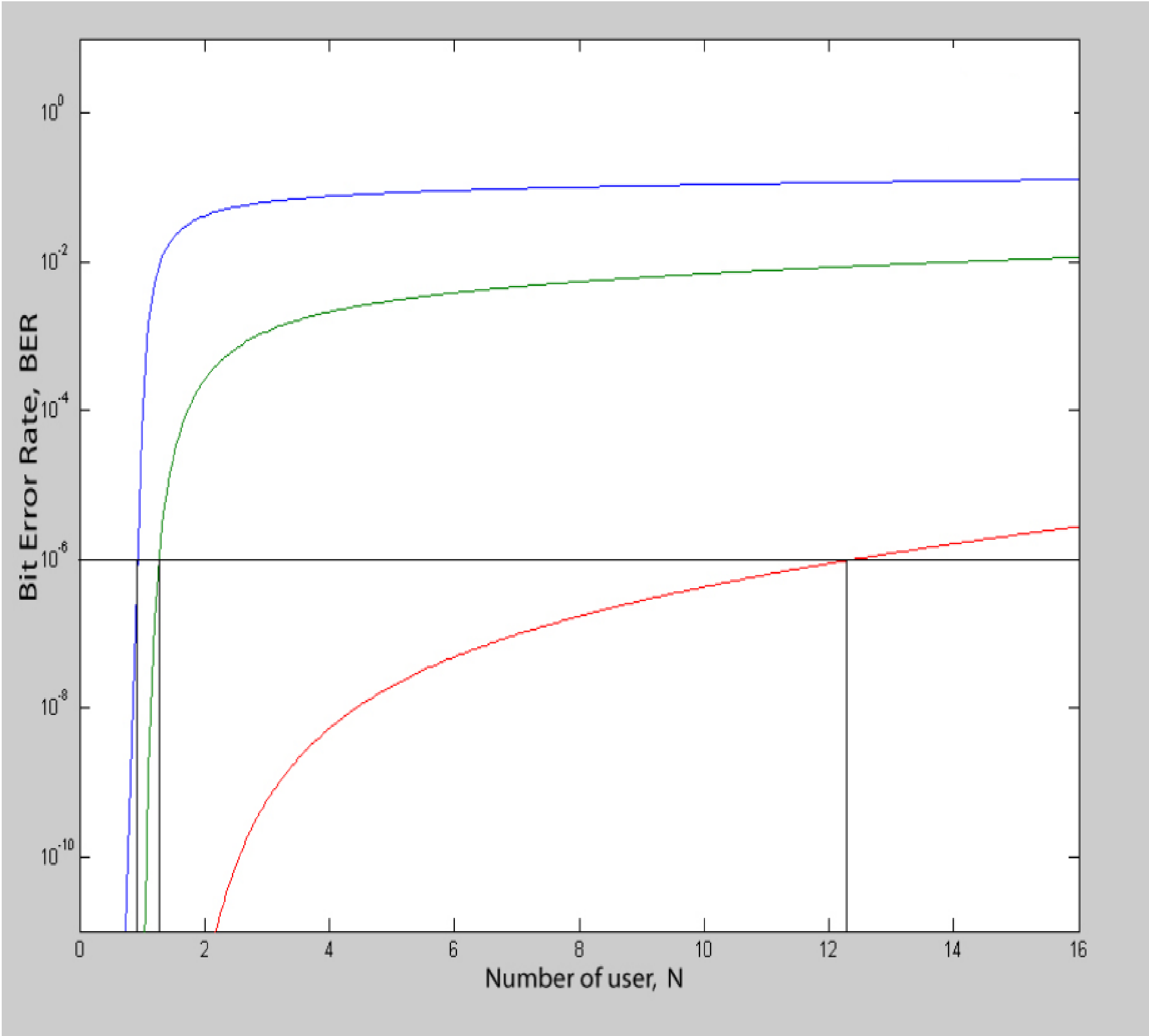


**Figure 9**: For fixed Eb/No, supported number of user for MC-DS-CDMA

That is, when in case of multi-tone, user number is almost 12, for multi-carrier, this value is 1. Which is quite lower than MT-DS-CDMA.

But we also found that, for different Eb/No dB value, sometimes the performance is being better in case of MT-DS-CDMA.

Another factor of comparison is security.

When data is being sent, in case of multi-carrier code division multiple accesses, the whole data is being separated into small parts. Then they are multiplied with different codes and then the coded data's are being multiplied by same frequencies.

On the other hand, in case of multi-tone direct-sequence code division multiple access, the whole data is being multiplied by one code, and then they are being multiplied by different frequencies.

So what happens is that, while transmitting, in both the processes, all the data's are being converted to serial to parallel data using serial to parallel converter.

These transmitted data are passed through very noisy and clumsy channels. So obviously they are being affected by interferences. Due to interferences, data packets can be lost. So in case of multi-carrier code division multiple accesses, any lost data packet can cause harm to the main signal. So we are not getting intact data.

From these figures we can also clearly observe these transmission and reception.

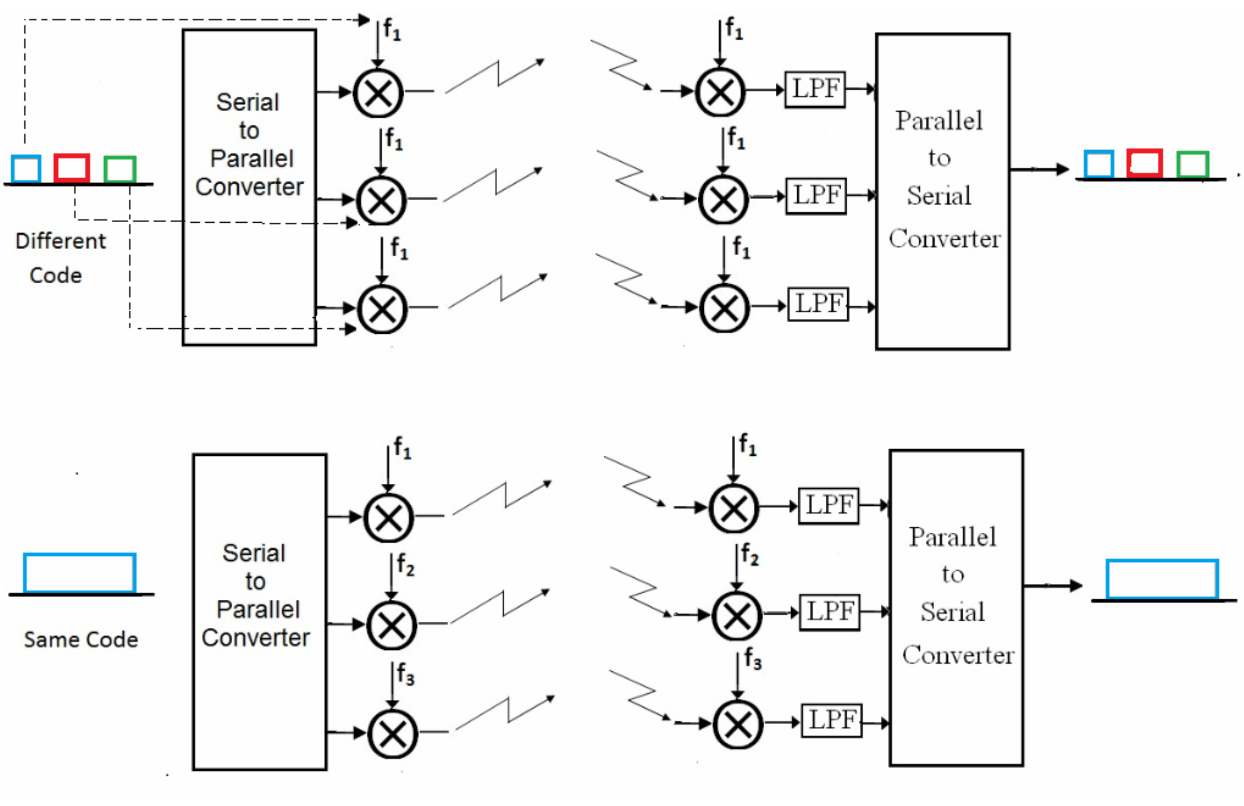


**Figure 1.9.1**: comparison between MT-DS-CDMA and MC-DS-CDMA

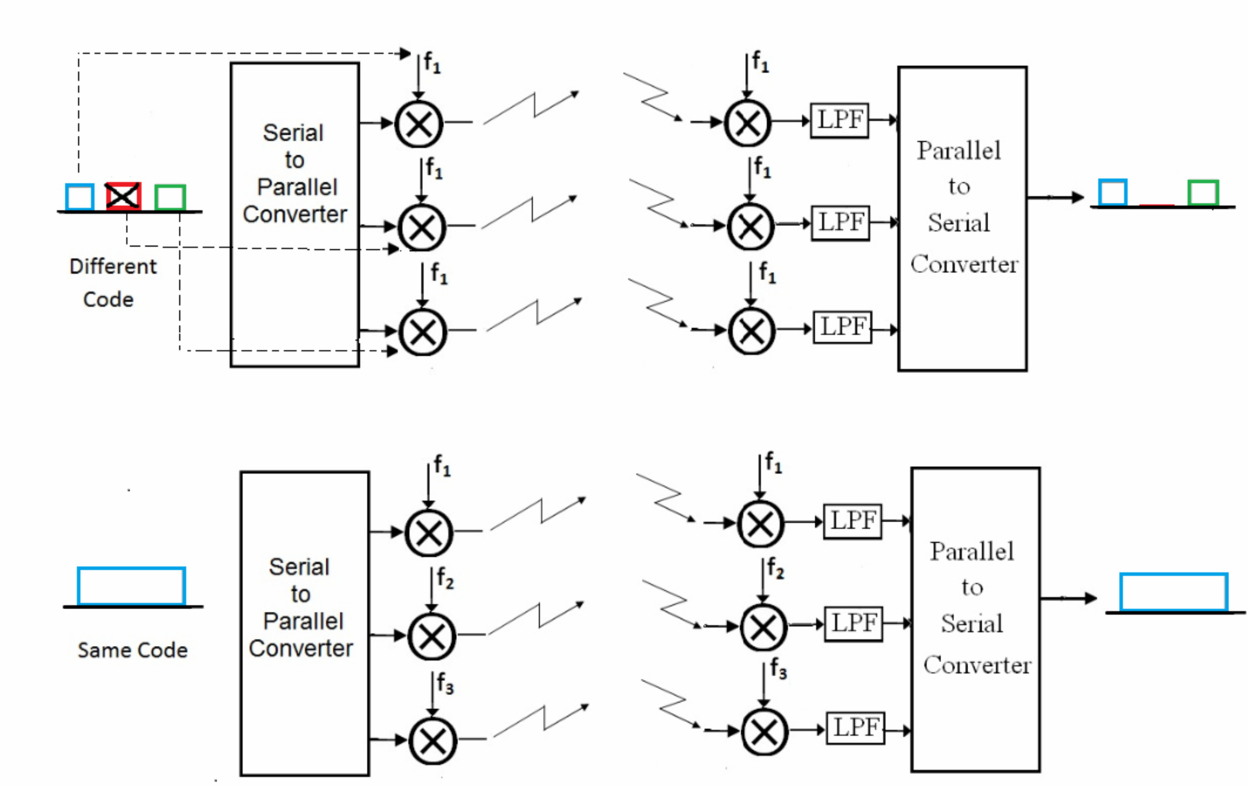


**Figure 1.9.2**: better performance for MT-DS-CDMA

But in case of multi-tone direct-sequence code division multiple accesses; our whole data is being coded with a single code. Then they are sent over noisy and clumsy channels. So if one of the packets is lost, we are getting the whole data packet through another frequency channel. This is due to frequency selective receiver which selects the data from the channels which sends higher and clears frequencies. In any radio transmission, the channel spectral response is not flat. Reflections off near-by objects (e.g. ground, buildings, trees, etc) can lead to multipath signals of similar signal power as the direct signal. This can result in deep nulls in the received signal power due to destructive interference. For narrow bandwidth transmissions if the null in the frequency response occurs at the transmission frequency then the entire signal can be lost. This can be partly overcome in two ways. By transmitting a wide bandwidth signal or spread spectrum as CDMA, any dips in the spectrum only result in a small loss of signal power, rather than a complete loss. Another method is discussed below.

This is how multi-tone direct-sequence code division multiple accesses can ensure the transmission of data safely. This criterion makes multi-tone direct-sequence code division multiple accesses most secure access technique.

Another positive side of multi-tone direct-sequence code division multiple accesses is that, it divides the whole bandwidth into small pieces. It is like compartments of a long train.

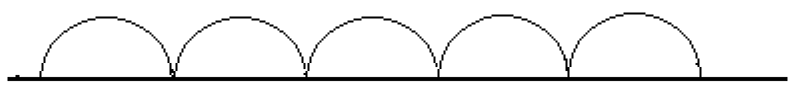

**Figure 1.9.3**: separation of bandwidth.

So, it is easier to handle and less affected by interference and noise. This is another way to overcome the transmitted frequencies, which is to split the transmission up into many small bandwidth carriers, as is done in a COFDM/OFDM transmission. The original signal is spread over a wide bandwidth thus; any nulls in the spectrum are unlikely to occur at all of the carrier frequencies. This will result in only some of the carriers being lost, rather than the entire signal. The information in the lost carriers can be recovered provided enough forward error corrections is sent.

# Chapter 10

# Drawbacks

There are some drawbacks of multi-tone direct-sequence code division multiple accesses. Major one is that due to full data coding, the whole packet transmission takes time. So at the receiving side, delay spread is increasing. Total process is becoming slow.

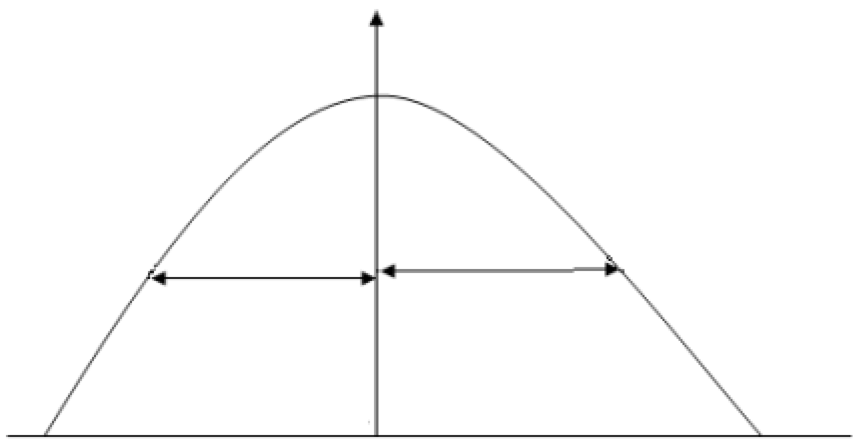

**Figure 2.1**: Delay spread

Another negative side is, for higher Eb/No dB values, user number is being decreasing. Where as in case of multi-carrier code division multiple accesses for a particular Eb/No, user number is higher than MT-DS-CDMA.

Another drawback in this multi-tone code division multiple accesses is that, the system is not cost effective. As it needs more frequencies to send different modulated data, as because we are dividing the whole bandwidth in to small pieces, so cost for hardware is increasing.

# Chapter 11

## Solutions

As every system has some drawbacks, our system does not differ from that. So our main goal was to find the solutions for these drawbacks.

First, we have found that to solve the delay spread problem, we can use orthogonal frequency division multiple access. This system ensures that our received data does not overlap upon one another. So ISI (inter symbol interference) problem can be avoided.

In case of cost, we have to consider that, because where we need higher security, in that place we can't use any system which is not complicated. For users own sake, more components of higher security should be used.

# Chapter 12

## Conclusion and future work:

As far as we have done our thesis work, we can summarize the whole work below:

- We have planned the whole thesis to be done synchronously
- We have found the block diagrams for transmitter and receiver for multi-tone direct-sequence code division multiple accesses.
- We have derived the equations for transmitting and receiving signal
- We have found the equation first without any kind of fading
- Then we have derived those equations by inputting fading effects
- Then we have seen the simulated version of these equations and found that these simulations satisfy out derived equations.
- Then we have taken the values from MC-DS-CDMA simulation and compared with our simulations.
- Then we found both the positive and negative sides of our thesis topic.
- Then we have explains some of the solutions for the drawbacks of our system
- In future some more works has to be done using the drawback solutions we have found in our thesis.
- And there we also have to do OFDM in receiving side.

# Chapter 13

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