Experiment No. 1

Title: Simulate to elaborate operations of multiple access techniques for CDMA

Date of Performance: **Roll No:**

Date of Submission: **University Seat No:**

Signature of Staff:

Experiment No. 1

Simulate to elaborate operations of multiple access techniques for CDMA

Title: To implement a basic function of Code Division Multiple Access (CDMA) to test the orthogonally and autocorrelation of a code to be used for CDMA operation. Write an application based on the above concept.

Theory:

What is CDMA?

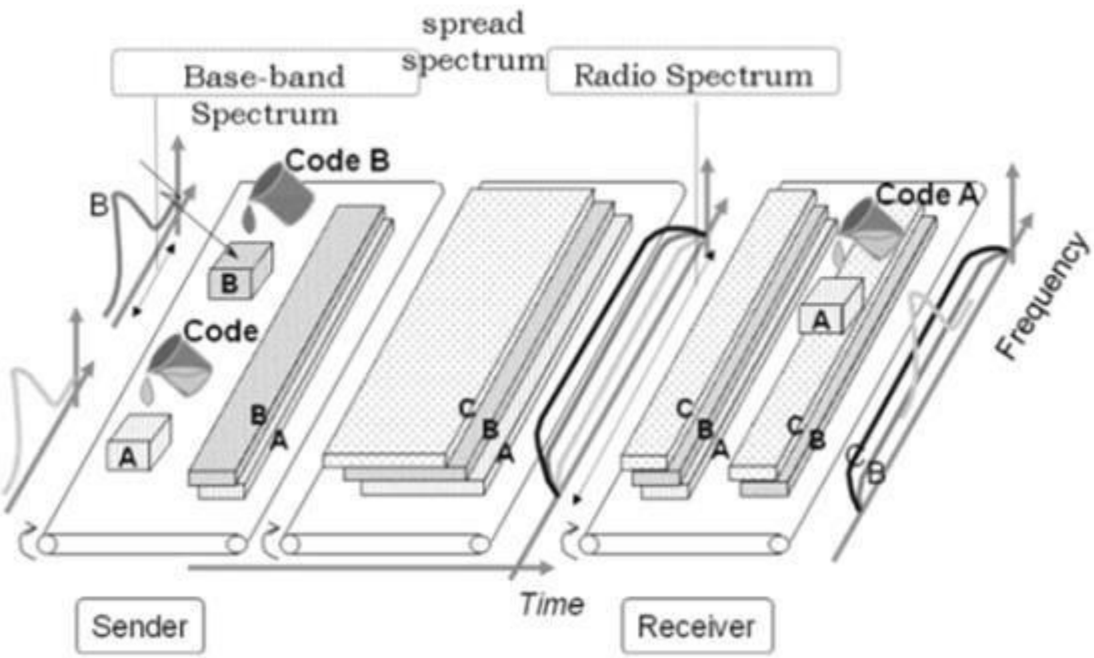
- CDMA stands for Code Division Multiple Access.
- It is a digital cellular standard that utilises spread-Spectrum Technology.
- It spreads the signal over a fully available spectrum or over multiple channels through division.
- It is a more secure and private line.
- It has good voice and data communication capabilities.
- The information is sent simultaneously through several transmitters over a single communication channel.

How Does CDMA Work?

CDMA allows up to 61 concurrent users in a 1.2288 MHz channel by processing each voice packet with two PN codes. There are 64 Walsh codes available to differentiate between calls and theoretical limits. Operational limits and quality issues will reduce the maximum number of calls somewhat lower than this value.

In fact, many different "signals" baseband with different spreading codes can be modulated on the same carrier to allow many different users to be supported. Using different orthogonal codes, interference between the signals is minimal. Conversely, when signals are received from several mobile stations, the base station is capable of isolating each as they have different orthogonal spreading codes.

The following figure shows the technicality of the CDMA system. During the propagation, we mixed the signals of all users, but by that you use the same code as the code that was used at the time of sending the receiving side. You can take out only the signal of each user.



Procedure:

1. The station encodes its data bit as follows. If bit = 1 then +1 If bit = 0 then -1
   no signal(interpreted as 0) if station is idle
2. Each station is allocated a different orthogonal sequence (code) which is N bit long for N stations
3. Each station does a scalar multiplication of its encoded data bit and code sequence.
4. The resulting sequence is then stored on the channel.
5. Since the channel is common, amplitudes add up and hence resultant channel sequence is the sum of sequences from all channels.
6. If station 1 wants to listen to station 2, it multiplies (inner product) the channel sequence with code of station S2.
7. The inner product is then divided by N to get data bit transmitted from station 2.

Example: Assume 4 stations S1, S2, S3, S4. We'll use a 4×4 Walsh Table to assign codes to them.

C1 = [+1 +1 +1 +1]

C2 = [+1 -1 +1 -1]

C3 = [+1 +1 -1 -1]

C4 = [+1 -1 -1 +1]

Let their data bits currently be:

D1 = -1

D2 = -1

D3 = 0 (Silent)

D4 = +1

Resultant channel sequence = C1.D1 + C2.D2 + C3.D3 + C4.D4

= [-1 -1 -1 -1] + [-1 +1 -1 +1] + [0 0 0 0]

+ [+1 -1 -1 +1]

= [-1 -1 -3 +1]

Now suppose station 1 wants to listen to station 2.

Inner Product = [-1 -1 -3 +1] x C2

= -1 + 1 - 3 - 1 = -4

Data bit that was sent = -4/4 = -1.

Conclusion:

Name of Student: **Roll No:**

Title of Experiment: Simulate to elaborate operations of multiple access techniques for CDMA.

To implement a basic function of Code Division Multiple Access (CDMA) to test the orthogonally and autocorrelation of a code to be used for CDMA operation. Write an application based on the above concept.

Code:

```
import numpy as np
c1=[1,1,1,1]
c2=[1,-1,1,-1]
c3=[1,1,-1,-1]
c4=[1,-1,-1,1]
rc=[]

print("Enter the data bits :")

d1=int(input("Enter D1 :"))
d2=int(input("Enter D2 :"))
d3=int(input("Enter D3 :"))
d4=int(input("Enter D4 :"))
r1=np.multiply(c1,d1)
r2=np.multiply(c2,d2)
r3=np.multiply(c3,d3)
r4=np.multiply(c4,d4)
resultant_channel=r1+r2+r3+r4;
print("Resultant Channel",resultant_channel)
Channel=int(input("Enter  the station to listen for C1=1 ,C2=2, C3=3 C4=4 : "))

if Channel==1:
  rc=c1
```

```
elif Channel==2:
  rc=c2
elif Channel==3:
  rc=c3
elif Channel==4:
  rc=c4
inner_product = np.multiply(resultant_channel,rc)

print("Inner Product",inner_product)
res1=sum(inner_product)

data = res1/len(inner_product)
print("Data bit that was sent",data)
```

Output:

```
Enter the data bits :
Enter D1 :23
Enter D2 :5
Enter D3 :456
Enter D4 :56
Resultant Channel [ 540  418 -484 -382]
Enter the station to listen for C1=1 ,C2=2, C3=3 C4=4 : 1
Inner Product [ 540  418 -484 -382]
Data bit that was sent 23.0
```

Experiment No. 2

Title: Study of GSM architecture and signaling techniques

Date of Performance: **Roll No:**

Date of Submission: **University Seat No:**

Signature of Staff:

Experiment No. 2

Study of GSM architecture and signaling techniques

Title: Study of Global System for Mobile Communication its Architecture and signalling techniques.

Theory:

GSM stands for Global System for Mobile Communication. GSM is an open and digital cellular technology used for mobile communication. It uses 4 different frequency bands of 850 MHz, 900 MHz, 1800 MHz and 1900 MHz . It uses the combination of FDMA and TDMA. This article includes all the concepts of GSM architecture and how it works.

GSM is having 4 different sizes of cells are used in GSM :

1. Macro: In this size of cell, Base Station antenna is installed.
2. Micro: In this size of cell, antenna height is less than the average roof level.
3. Pico: Small cells' diameter of few meters.
4. Umbrella: It covers the shadowed (Fill the gaps between cells) regions.

Features of GSM:

1. Supports international roaming
2. Clear voice clarity
3. Ability to support multiple handheld devices.
4. Spectral / frequency efficiency
5. Low powered handheld devices.
6. Ease of accessing network
7. International ISDN compatibility.
8. Low service cost.
9. New features and services.

GSM – Architecture:

A GSM network comprises of many functional units. These functions and interfaces are explained in this chapter. The GSM network can be broadly divided into –

1. The Mobile Station (MS)
2. The Base Station Subsystem (BSS)
3. The Network Switching Subsystem (NSS)
4. The Operation Support Subsystem (OSS)

1. GSM - The Mobile Station

The MS Functions

The MS consists of the physical equipment, such as the radio transceiver, display and digital signal processors, and the SIM card. It provides the air interface to the user in GSM networks. As such, other services are also provided, which include –

- Voice Teleservices
- Data bearer services
- The features' supplementary services

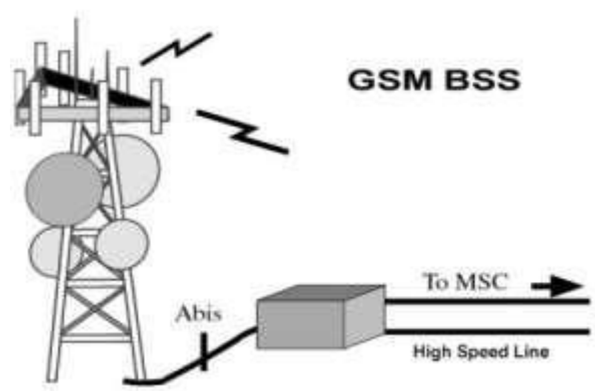
The MS also provides the receptor for SMS messages, enabling the user to toggle between the voice and data use. Moreover, the mobile facilitates access to voice messaging systems. The MS also provides access to the various data services available in a GSM network. These data services include –

- X.25 packet switching through a synchronous or asynchronous dial-up connection to the PAD at speeds typically at 9.6 Kbps.
- General Packet Radio Services (GPRSs) using either an X.25 or IP based data transfer method at the speed up to 115 Kbps.
- High speed, circuit switched data at speeds up to 64 Kbps.

What is SIM?

The SIM provides personal mobility so that the user can have access to all subscribed services irrespective of both the location of the terminal and the use of a specific terminal. You need to insert the SIM card into another GSM cellular phone to receive calls at that phone, make calls from that phone, or receive other subscribed services.

2. GSM - The Base Station Subsystem (BSS)



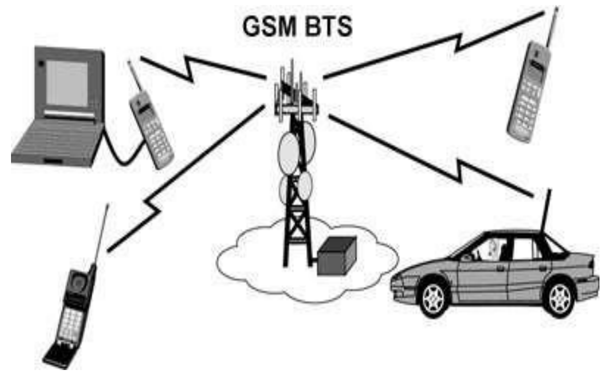
The BSS is composed of two parts –

- The Base Transceiver Station (BTS)

- The Base Station Controller (BSC)

The BTS and the BSC communicate across the specified Abis interface, enabling operations between components that are made by different suppliers. The radio components of a BSS may consist of four to seven or nine cells. A BSS may have one or more base stations. The BSS uses the Abis interface between the BTS and the BSC. A separate high-speed line (T1 or E1) is then connected from the BSS to the Mobile MSC.

The Base Transceiver Station (BTS)



The BTS houses the radio transceivers that define a cell and handles the radio link protocols with the MS. In a large urban area, a large number of BTSs may be deployed.

The BTS corresponds to the transceivers and antennas used in each cell of the network. A BTS is usually placed in the center of a cell. Its transmitting power defines the size of a cell. Each BTS has between 1 and 16 transceivers, depending on the density of users in the cell. Each BTS serves as a single cell. It also includes the following functions –

- Encoding, encrypting, multiplexing, modulating, and feeding the RF signals to the antenna
- Transcoding and rate adaptation
- Time and frequency synchronizing
- Voice through full- or half-rate services
- Decoding, decrypting, and equalizing received signals
- Random access detection
- Timing advances
- Uplink channel measurements

3. **The Base Station Controller (BSC)**

The BSC manages the radio resources for one or more BTSs. It handles radio channel setup, frequency hopping, and handovers. The BSC is the connection between the mobile and the MSC. The BSC also translates the 13 Kbps voice channel used over the radio link to the standard 64 Kbps channel used by the Public Switched Telephone Network (PSDN) or ISDN.

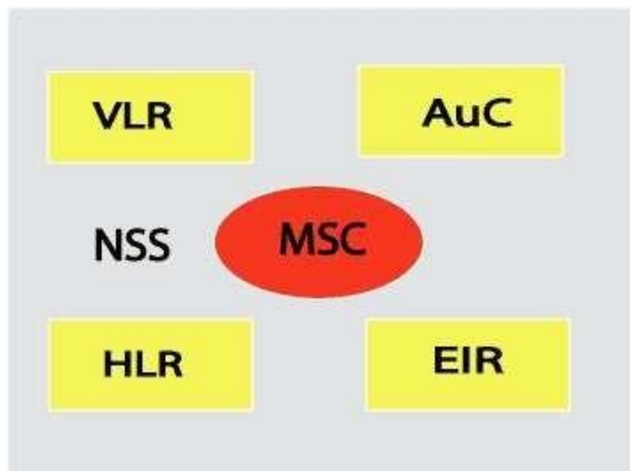
It assigns and releases frequencies and time slots for the MS. The BSC also handles intercell handover. It controls the power transmission of the BSS and MS in its area. The function of the BSC is to allocate the necessary time slots between the BTS and the MSC. It is a switching device that handles the radio resources.

The additional functions include–

- Control of frequency hopping
- Performing traffic concentration to reduce the number of lines from the MSC
- Providing an interface to the Operations and Maintenance Center for the BSS
- Reallocation of frequencies among BTSs
- Time and frequency synchronization
- Power management
- Time-delay measurements of received signals from the MS

4. GSM - The Network Switching Subsystem (NSS)

The Network switching system (NSS), the main part of which is the Mobile Switching Center (MSC), performs the switching of calls between the mobile and other fixed or mobile network users, as well as the management of mobile services such as authentication.



The switching system includes the following functional elements –

Home Location Register (HLR)

The HLR is a database used for storage and management of subscriptions. The HLR is considered the most important database, as it stores permanent data about subscribers, including a subscriber's service profile, location information, and activity status. When an individual buys a subscription in the form of SIM, then all the information about this subscription is registered in the HLR of that operator.

Mobile Services Switching Center (MSC)

The central component of the Network Subsystem is the MSC. The MSC performs the switching of calls between the mobile and other fixed or mobile network users, as well as the management of mobile services such as registration, authentication, location updating, handovers, and call routing to a roaming subscriber. It also performs such functions as toll ticketing, network interfacing, common channel signalling, and others. Every MSC is identified by a unique ID.

Visitor Location Register (VLR)

The VLR is a database that contains temporary information about subscribers that is needed by the MSC in order to service visiting subscribers. The VLR is always integrated with the MSC. When a mobile station roams into a new MSC area, the VLR connected to that MSC will request data about the mobile station from the HLR. Later, if the mobile station makes a call, the VLR will have the information needed for call setup without having to interrogate the HLR each time.

Authentication Center (AUC)

The Authentication Center is a protected database that stores a copy of the secret key stored in each subscriber's SIM card, which is used for authentication and ciphering of the radio channel. The AUC protects network operators from different types of fraud found in today's cellular world.

Equipment Identity Register (EIR)

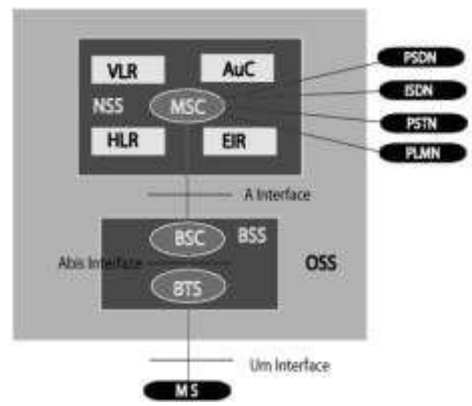
The Equipment Identity Register (EIR) is a database that contains a list of all valid mobile equipment on the network, where its International Mobile Equipment Identity (IMEI) identifies each MS. An IMEI is marked as invalid if it has been reported stolen or is not type approved.

GSM - The Operation Support Subsystem (OSS)

The operations and maintenance center (OMC) is connected to all equipment in the switching system and to the BSC. The implementation of OMC is called the operation and support system (OSS).
Here are some of the OMC functions−

- Administration and commercial operation (subscription, end terminals, charging, and statistics).
- Security Management.
- Network configuration, Operation, and Performance Management.
- Maintenance Tasks.

A simple pictorial view of the GSM architecture is given below –

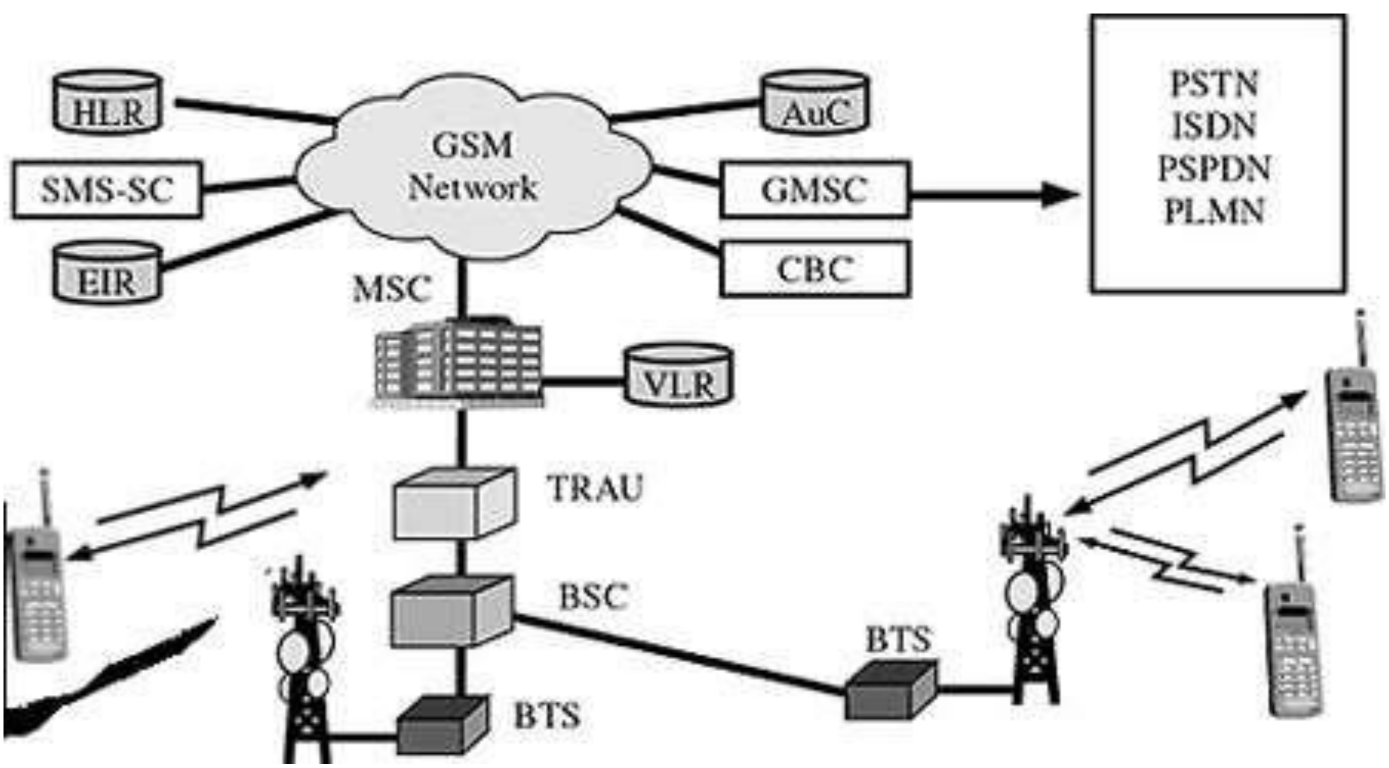


The additional components of the GSM architecture comprise of databases and messaging systems functions –

- Home Location Register (HLR)
- Visitor Location Register (VLR)
- Equipment Identity Register (EIR)
- Authentication Center (AuC)
- SMS Serving Center (SMS SC)
- Gateway MSC (GMSC)
- Chargeback Center (CBC)
- Transcoder and Adaptation Unit (TRAU)

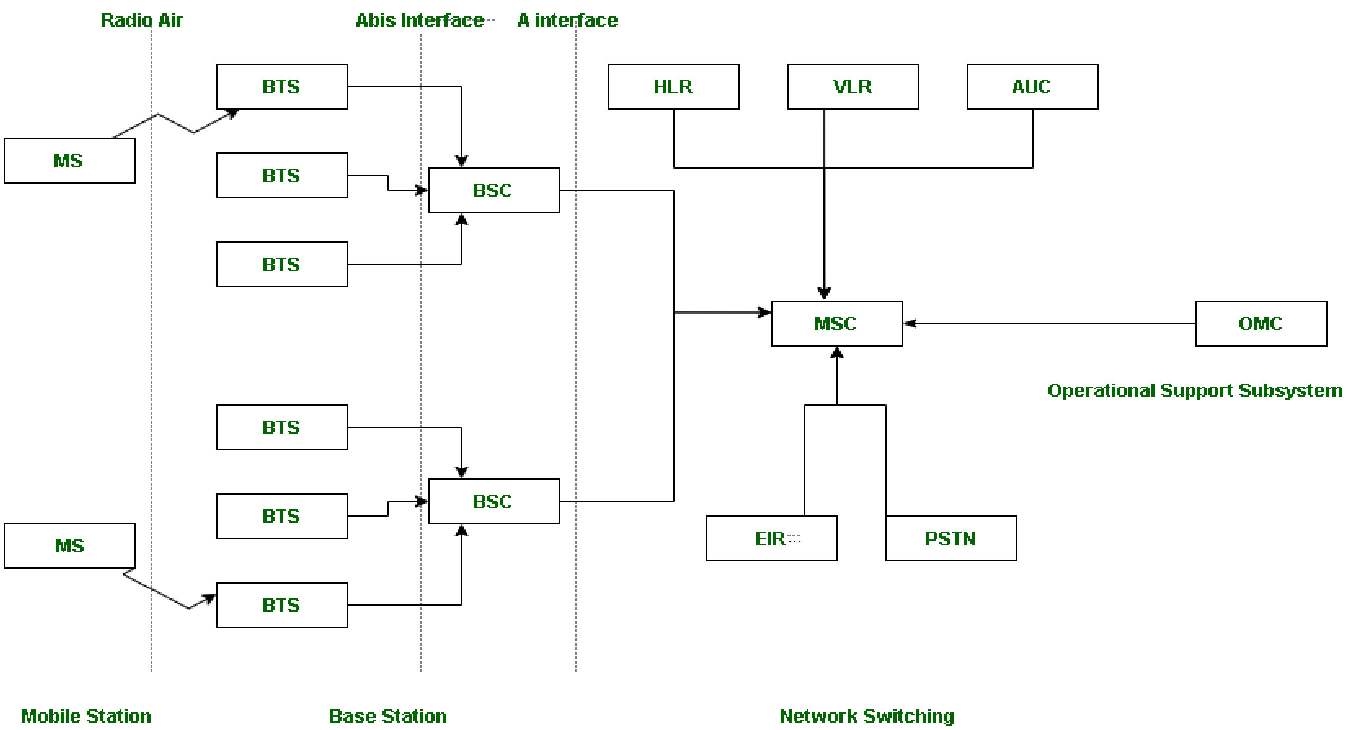
The following diagram shows the GSM network along with the added elements –

The MS and the BSS communicate across the Um interface. It is also known as the *air interface* or the *radio link*. The BSS communicates with the Network Service Switching (NSS) center across the *A* interface.



GSM network areas

Example:- Suppose there are 3 Mobile stations which are connected with the tower and that tower is connected to BTS through TRX, then further connected to BSC and MSC.



Three subsystem BSS, NSS and OSS are connected with each other via some interfaces. Total three interfaces are there:

1. **Air Interface :** Air interface is also known as UM interface. Interface between MS and BTS is called as UM interface because it is mobile analog to the U interface of ISDN.
2. **Abis Interface :** It is a BSS internal interface linking with BTS and BSC.
3. **A interface :** It provides communication between BSS and MSC.

Conclusion:

Experiment No. 3

Title: Study of GPRS services

Date of Performance: **Roll No:**

Date of Submission: **University Seat No:**

Signature of Staff:

Experiment No. 3

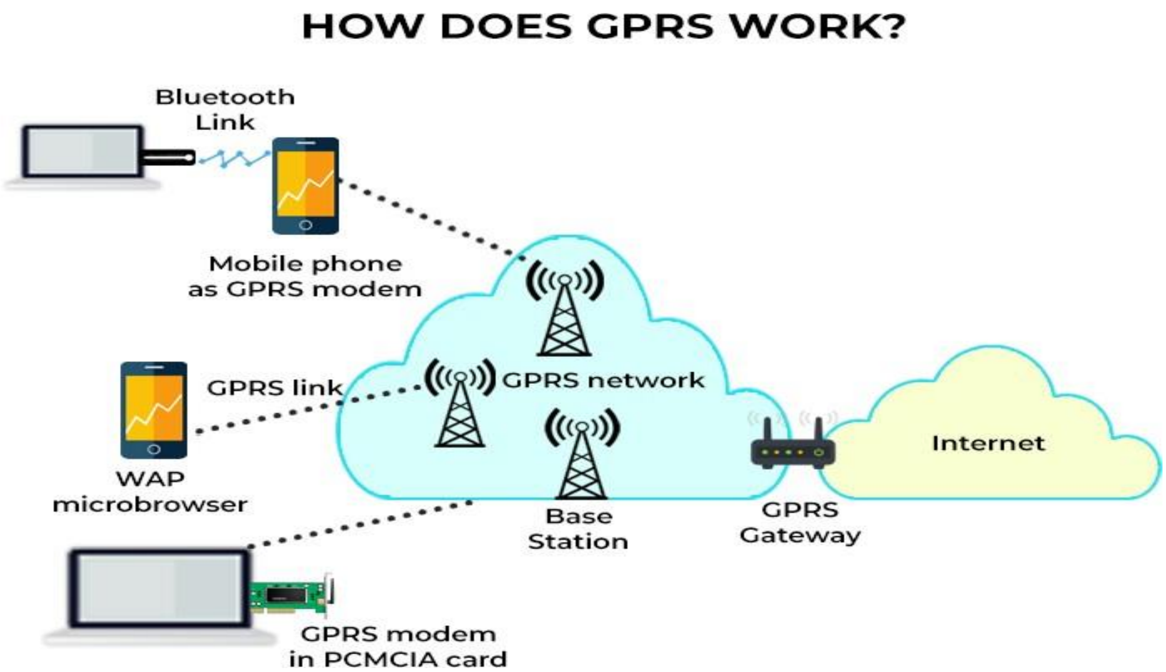
Study of GPRS services

Title: Study of Global System for Mobile Communication its Architecture and signalling techniques.

Theory:

What Is GPRS?

General packet radio service (GPRS) is defined as a mobile communications standard that operates on 2G and 3G cellular networks to enable moderately high-speed data transfers using packet-based technologies.



GPRS adds several new essential features to aid the end-to-end transmission of IP-based data packets. The GSM standards created GPRS, which resulted in a structure with specified attributes, interfaces, and inter-network operations for roaming assistance.

GPRS enhances the architecture of the GSM standard to enable packetized communication services at speeds up to 114 kbit/s in reality. Data transmission uses the network whenever it is essential because of the packet transfer mode. As a result of the GPRS standard, users could be paid based on the volume exchanged instead of the connection length, allowing them to remain connected without incurring further charges. GPRS enables network operators to have a more effective method to distribute data to their customers by removing the requirement to use speech channels for data transmission. Customers get four times the rate for the most typical dial-up connections when using a GPRS connection. The overall average speed of dial-up data services is 9.6 kbps, whereas GPRS speeds range from 40 to 172.2 kbps. Users can also access rich media applications and information using GPRS connectivity. It is not necessary to download any data to view or use it. As a result, GPRS technology is suitable for cellphone internet use. Because of the GPRS network, one can access rich media apps via mobile networks.

Further, because of circuit-switched data speed constraints and the 160-character short message service (SMS) limit, many applications were previously inaccessible in GSM networks. The GPRS system enables mobile services to use apps that have been previously only available on desktop laptops and computers. This unlocks a slew of new technological and creative potential.

GPRS networks can also allow for internet access. It suggests that online services such as web surfing, email, chat, and file transfer protocol (FTP) could be accessible via cell phones. The GPRS connection could be seen as a sub-network of the Internet because it uses similar protocols, and GPRS phones could be regarded as mobile hosts since they are using the same methods. GPRS operates on cell phones, laptops, and other portable devices with a GPRS modem or a GPRS Subscriber Identity Module (SIM) card. A GPRS modem is ideal for sending and receiving messages from a computer.

This packet-based platform was one of the first to offer multimedia messaging and push-to-talk features, similar to a walkie-talkie. However, GPRS devices are rarely capable of transmitting longer voice recordings. GPRS technology, for instance, couldn't allow individuals to send a given text to a transcription system since the audio files became too huge to be transferred at acceptable rates.

The idea of Quality of Service (QoS) is incorporated into GPRS. It refers to the capacity to adapt a service to the requirements of an app. The following QoS conditions apply – priority, reliability, delay, and throughput. The inclusion of new network nodes called GSN (GPRS support nodes) situated on a backbone network is required to integrate GPRS into a GSM architecture:

The SGSN (Serving GPRS Support Node) is a router that controls the locations of nearby stations and offers a packet transit interface to the GGSN portal. It is in charge of GPRS mobile device authentication, network registration, mobility management, and data collection on pricing for using the air interface.

GPRS wireless packet data has three main characteristics:

- An always-online function that eliminates the need for dial-up, enabling programs accessible with just one click.

- The upgraded usual systems mean that carriers do not have to update their equipment; instead, GPRS is put on top of the current equipment.
- An essential part of future 3G networks, meaning the packet data core network for 3G systems EDGE and WCDMA, GPRS is a necessary foundation for the future.

GPRS provides the services listed below:

- Broadcasting and SMS messaging
- Cellular-based push-to-talk
- Presence and instant messaging
- Service for multimedia messaging
- Services such as point-to-point and point-to-multipoint

Benefits Of GPRS:

• Mobility: The capacity to keep up consistent voice and information interchanges while moving.

• Cost Efficient: Communication via GPRS is cheaper than through the regular GSM network.

• Immediacy: Allows customers to obtain connectivity when needed, regardless of location and without a lengthy login session.

• Localization: Enables customers to acquire data applicable to their present area.

• Easy Billing: GPRS packet transmission offers an easier to use billing than that offered by circuit switched administrations. GPRS is an innovation that numerous GPS beacons are using to get up to the minute data with tracking. When the GPS gadget records the information, it would then be able to be transmitted through GPRS to another central location, for example, a PC or through an email. It is the GPRS innovation that takes into consideration ongoing updates to GPS following frameworks. It is this direct GPRS association that gives the client of the GPS system the most reliable information available today.

Conclusion:

Experiment No. 4

Title: BER Performance over Rayleigh Fading wireless channel with BPSK Transmission

Date of Performance: **Roll No:**

Date of Submission: **University Seat No:**

Signature of Staff:

Experiment No. 4

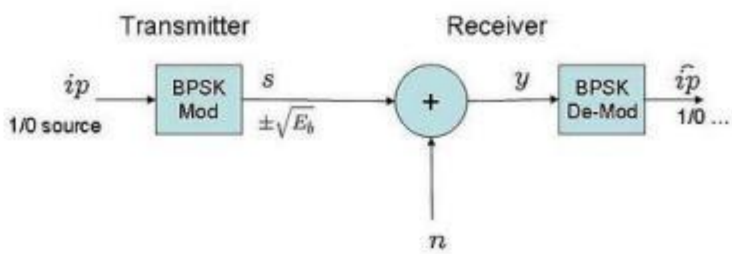
BER Performance over Rayleigh Fading wireless channel with BPSK Transmission

Title: Simulate BER Performance over Rayleigh Fading wireless channel with BPSK Transmission for SNR 0 to 60 db.

Theory:

In this experiment, we will derive the theoretical equation for bit error rate (BER) with Binary Phase Shift Keying (BPSK) modulation scheme in Additive White Gaussian Noise (AWGN) channel. The BER results obtained using Matlab/Octave simulation scripts show good agreement with the derived theoretical results.

With Binary Phase Shift Keying (BPSK), the binary digits 1 and 0 maybe represented by the analog levels $+\sqrt{E_b}$ and $-\sqrt{E_b}$ respectively. The system model is as shown in the Figure below.



Channel Model:

The transmitted waveform gets corrupted by noise , typically referred to as Additive White Gaussian Noise (AWGN).

Additive : As the noise gets 'added' (and not multiplied) to the received signal

White : The spectrum of the noise if flat for all frequencies.

Gaussian : The values of the noise follows the Gaussian probability distribution function,

$$p(x) = \frac{1}{\sqrt{2\pi\sigma^2}} e^{\frac{-(x-\mu)^2}{2\sigma^2}} \text{ with } \mu = 0 \text{ and } \sigma^2 = \frac{N_0}{2}.$$

Computing the probability of error:

Using the derivation provided

The received signal,

$y = s_1 + n$ when bit 1 is transmitted and

$y = s_0 + n$ when bit 0 is transmitted.

The conditional probability distribution function (PDF) of for the two cases are:

$$p(y|s_0) = \frac{1}{\sqrt{\pi N_0}} e^{\frac{-(y+\sqrt{E_b})^2}{N_0}}$$

$$p(y|s_1) = \frac{1}{\sqrt{\pi N_0}} e^{\frac{-(y-\sqrt{E_b})^2}{N_0}}$$

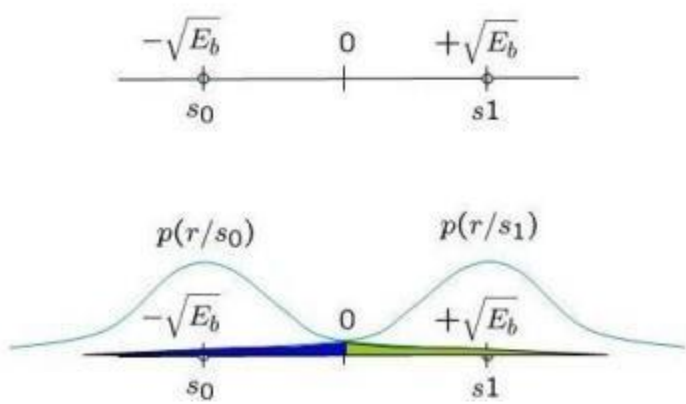


Figure: Conditional probability density function with BPSK modulation

Assuming that S1 and S0 are equally probable i.e. $p(s_1) = p(s_0) = 1/2$, The threshold 0 forms the optimal decision boundary.

- if the received signal is y is greater than 0, then the receiver assumes S1 was transmitted.
- if the received signal is is less than or equal to 0, then the receiver assumes was transmitted.

i.e.

y>0=S1 and

y≤0=S0

Probability of error given S1 was transmitted.

With this threshold, the probability of error given is transmitted is (the area in blue region):

$$p(e|s_1) = \frac{1}{\sqrt{\pi N_0}} \int_{-\infty}^{0} e^{\frac{-(y-\sqrt{E_b})^2}{N_0}} dy = \frac{1}{\sqrt{\pi}} \int_{\sqrt{\frac{E_b}{N_0}}}^{\infty} e^{-z^2} dz = \frac{1}{2} erfc\left(\sqrt{\frac{E_b}{N_0}}\right),$$

where,

$$erfc(x) = \frac{2}{\sqrt{\pi}} \int_{x}^{\infty} e^{-x^2} dx$$

Probability of error given S0 was transmitted

Similarly the probability of error given S0 is transmitted is (the area in green region):

$$p(e|s_0) = \frac{1}{\sqrt{\pi N_0}} \int_{0}^{\infty} e^{\frac{-(y+\sqrt{E_b})^2}{N_0}} dy = \frac{1}{\sqrt{\pi}} \int_{\sqrt{\frac{E_b}{N_0}}}^{\infty} e^{-z^2} dz = \frac{1}{2} erfc\left(\sqrt{\frac{E_b}{N_0}}\right).$$

Total probability of bit error

$$P_b = p(s_1)p(e|s_1) + p(s_0)p(e|s_0).$$

Given that we assumed that S1 and S0 are equally probable i.e. $p(s_1) = p(s_0) = 1/2$, the bit error probability is,

$$P_b = \frac{1}{2} erfc\left(\sqrt{\frac{E_b}{N_0}}\right).$$

Procedure:

Python source code for computing the bit error rate with BPSK modulation from theory and simulation. The code performs the following:

(a) Generation of random BPSK modulated symbols +1′s and -1′s

(b) Passing them through Additive White Gaussian Noise channel

(c) Demodulation of the received symbol based on the location in the constellation

(d) Counting the number of errors

(e) Repeating the same for multiple Eb/No value.

Conclusion:

Name of Student: **Roll No:**

Title of Experiment: Simulate BER Performance over Rayleigh Fading wireless channel with BPSK Transmission for SNR 0 to 60 db.

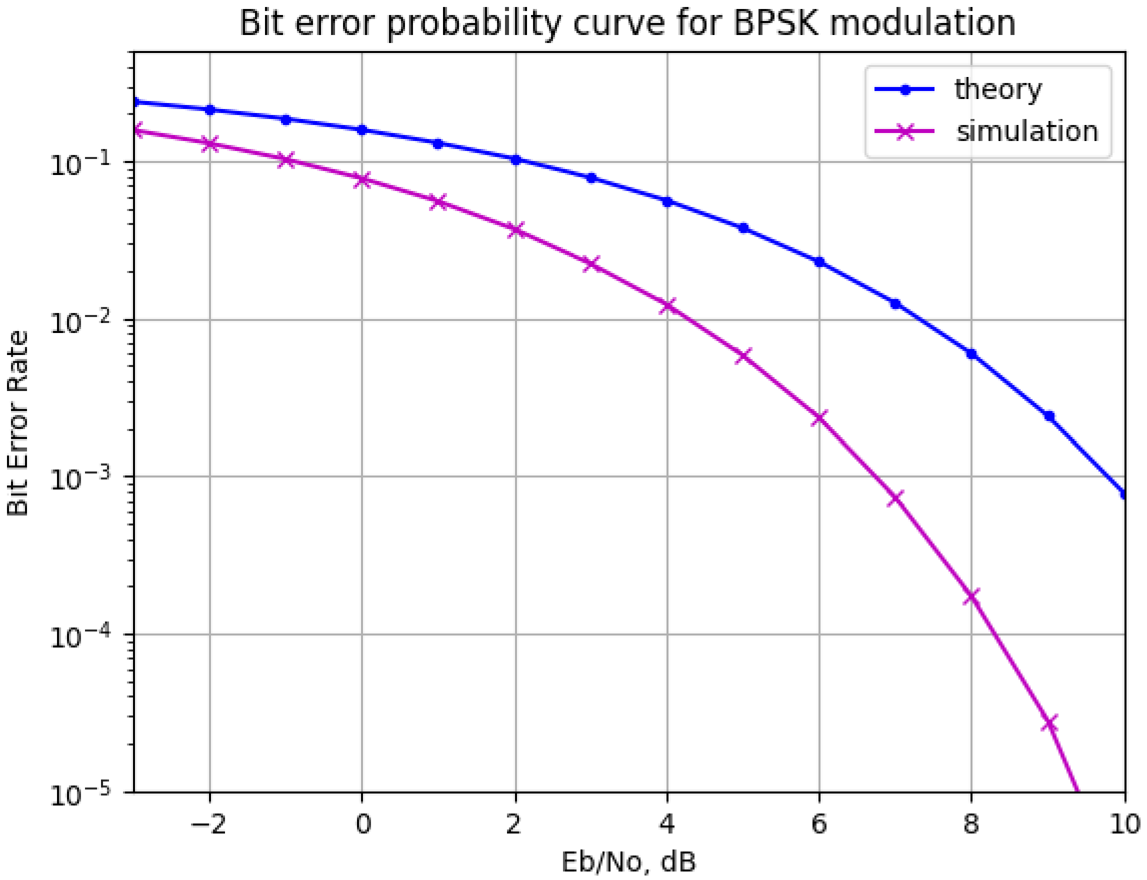
Simulate BER Performance over Rayleigh Fading wireless channel with BPSK Transmission for SNR 0 to 60 db.

Code:

```
import numpy as np
from scipy.special import erfc
import matplotlib.pyplot as plt

np.random.seed(100)
np.random.randn(200)

# Transmitter
N = 10**6
ip = np.random.randint(2, size=N)
s = 2*ip - 1
n = 1/np.sqrt(2)*(np.random.randn(N) + 1j*np.random.randn(N))

Eb_N0_dB = np.arange(-3, 11, 1)

# Loop over various Eb/N0 values
nErr = np.zeros(len(Eb_N0_dB))
for ii in range(len(Eb_N0_dB)):
   # Noise addition
   y = s + 10**(-Eb_N0_dB[ii]/20)*n

   # Receiver - hard decision decoding
   ipHat = np.real(y) > 0

   # Counting the errors
   nErr[ii] = np.sum(ip != ipHat)

# Simulated bit error rate
simBer = nErr/N

# Theoretical bit error rate
```

```
theoryBer = 0.5*erfc(np.sqrt(10**(Eb_N0_dB/10)/2))

# Plot
plt.figure()
plt.semilogy(Eb_N0_dB, theoryBer, 'b.-', label='theory')
plt.semilogy(Eb_N0_dB, simBer, 'mx-', label='simulation')
plt.axis([-3, 10, 10**-5, 0.5])
plt.grid(True)
plt.legend(loc='upper right')
plt.xlabel('Eb/No, dB')
plt.ylabel('Bit Error Rate')
plt.title('Bit error probability curve for BPSK modulation')
plt.show()
```

OUTPUT:



Note: This practical is performed on online python simulator.
Link for simulator:
https://trinket.io/embed/python3/a5bd54189b

Experiment No. 5

Title: Configuring a Cisco Router as a DHCP Server

Date of Performance: **Roll No:**

Date of Submission: **University Seat No:**

Signature of Staff:

Experiment No. 5

Configuring a Cisco Router as a DHCP Server

Title: Configuring a Cisco Router as a DHCP Server

Software Requirements: Open-source Linux operating system, Cisco Packet Tracer.

Theory:

Dynamic Host Configuration Protocol (DHCP) is a standardized client/server network protocol that dynamically assigns IP addresses and other related configuration information to network devices. Every device on a TCP/IP-based network must have a unique unicast IP address to access the network and its resources. Without DHCP, IP addresses for new computers or computers that are moved from one subnet to another must be configured manually.

Configuring the DHCP server:

The DHCP server uses address pools when responding to DHCP client requests. Address pools contains specific IP configuration details that the DHCP server can allocate to a client. You can configure multiple address pools on the device for different networks. To configure an address pool, you must:

1. Create the pool and enter its configuration mode.

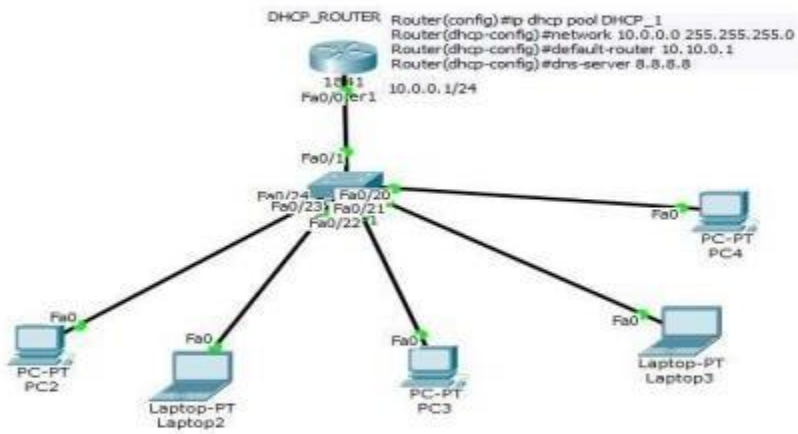
2. Define the network the pool applies to.

3. Define the range of IP addresses that the server can allocate to clients. You can specify multiple address ranges for each pool.

4. Set the lease for the clients. This defines whether the clients receive a dynamic, permanent, or static IP address.

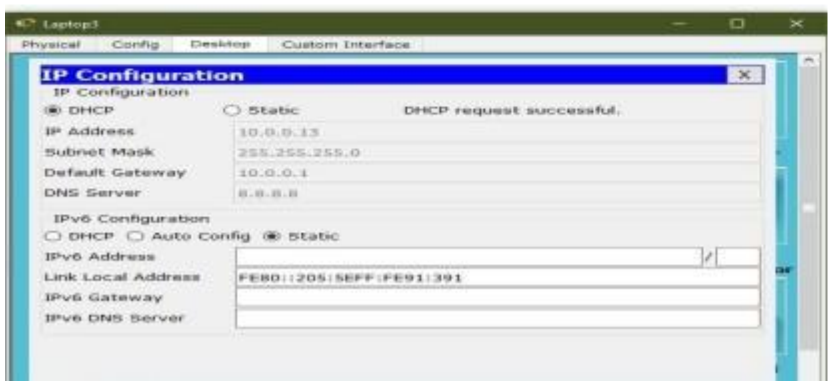
5. Set the options (standard and user-defined) that the clients of a pool require when configuring their IP details.

6. After configuring the address pools, enable the DHCP server by using the command:



On Client Side:

Select IP allocation as a Dynamic allocation.



DORA Process:

The following diagram shows the changing port numbers and the source and destination addresses used during the DHCP transaction. UDP port 68 is reserved for DHCP clients, and UDP port 67 is reserved for DHCP servers.

Step 1

DHCP Discover Sent by the client looking for the IP address. The source IP is 0.0.0.0 because the client doesn't. have an IP address. The destination is 255.255.255.255, which is the broadcast address, as the client doesn't know where the DHCP server is located, so it broadcasts to all devices on the network.

Step 2: DHCP Offer

Sent by the DHCP server offering an IP address to the client. The source address is the DHCP server address. The DHCP server doesn't know the client address yet, so it broadcasts the offer to all devices on the network.

Step 3: DHCP Request

Sent by the client to the DHCP server to say "I will take that IP address, thanks." The client IP address is still 0.0.0.0 and it is again broadcast to all so that any other servers on the network that may have offered an IP address will know to stop communicating with the client for now.

Step 4: DHCP Acknowledgment

Sent by the DHCP server to the client. It confirms the IP address and other details such as subnet mask, default gateway, and lease time with the client. The source address is the DHCP server and the destination is still the broadcast address.

The DHCP process:

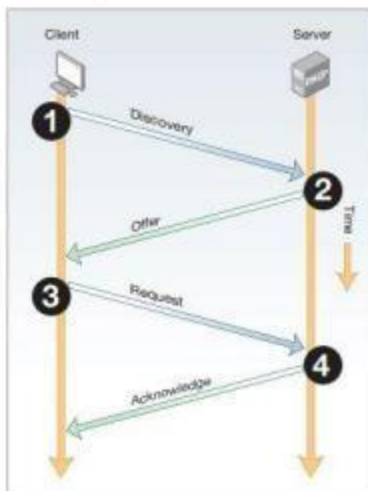
There are four basic steps the DHCP process follows when a client connects to the network:

1. The client broadcasts a DHCP Discover message to say "I need an IP address, are there any DHCP servers out there?"

2. Multiple DHCP servers may respond (via broadcast) with an OFFER for a leased IP address back to the client.

3. The client will choose a DHCP server offer and then broadcast a DHCP REQUEST back to the DHCP server(s) to say "Thanks, I have selected an offer from this DHCP server." All servers will see which offer the client selected.

4. Finally, the selected DHCP server will send (broadcast) an ACKNOWLEDGEMENT back to the client to confirm the IP address, lease time, and other details.



Conclusion:

Experiment No. 6

Title: To understand the handover mechanism

Date of Performance: **Roll No:**

Date of Submission: **University Seat No:**

Signature of Staff:

Experiment No. 6

To understand the handover mechanism

Title To study the effect of handover threshold and margin on SINR and call drop probability and handover probability

Software Requirements:

Operating System: Windows 7

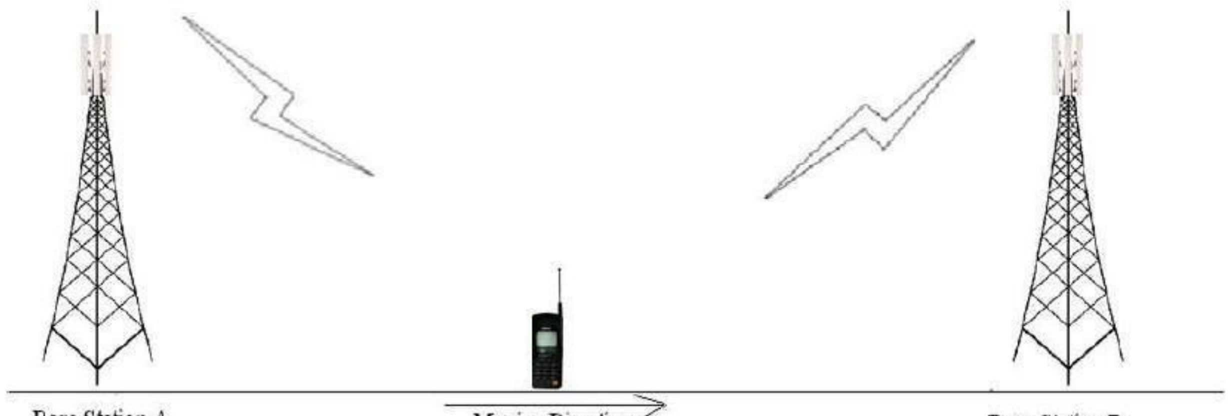
Java Version: 6 only

Mozilla Firefox: version: 47.0.1 Link to download software:
https://drive.google.com/uc?id=0B9mNeu43jUidckFYVTlnenpJRGs&export=download

Theory:

Handoff:

Consider the figure below Initially say the mobile M is quite close to the base station A $P_{r_x}^A > P_{r_x}^B$ and hence receives signal strength from A As the mobile moves away from the base station. A and goes towards B then the signal strength from A keeps falling (pathloss increases).Let there be a minimum sensibility level $P_{r_x}^0$ for the mobile, i.e. if the signal from the B.S.to which the mobile is connected falls below $P_{r_x}^0$ then the call drops. In order to prevent call drop the mobile monitors receive signal strength from the neighbouring 3-6 B.S.. These neighboring 3-6 B.S. also monitor Rx signal strength from the M.S.



The mobile should get connected to B.S. which has the highest signal strength. However, if the M.S. continuously attaches itself to the B.S. with instantaneous height signal strength then the h/o rate may very high in server condition.

Thus, some hysten's condition is used for h. If $P_{r_x}^T$ (T= target B.S.) $> P_{r_x}^h$ higher h/o threshold $P_{r_x}^h$ and $\overline{P_{r_x}^c}$ (c=current B.S.) < minimum h/o threshold the execute h/o to B·ST from B·Sc. Thus, it is threshold impeditive to study in part of the handoff process.

$$\Delta_\gamma = P_{r_x}^h - P_{r_x}^l$$

A successful handoff is one where the call gets from and continuous without call or in other words the h occurs before h/o $P_{r_x}^c$ $P_{r_x}^c < P_{r_x}^0$ becomes If $P_{r_x}^c < P_{r_x}^0$ then call drop event occurs.

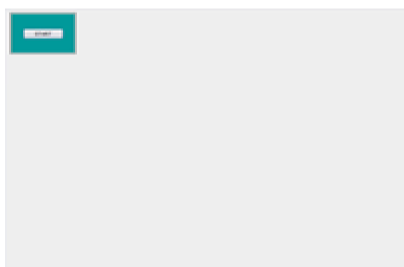
One would like to minimize the no of handoff events as well as minimize call drop probability. The experiment provides opportunity to study the inherent of these three parameters on h/o.

Further the averaging window for calculating $P^T_{r_*}$ and $P^c_{r_*}$ also plays a role in the process. In the experiment small scale fading is not considered and hence the averaging considered only shadowing.

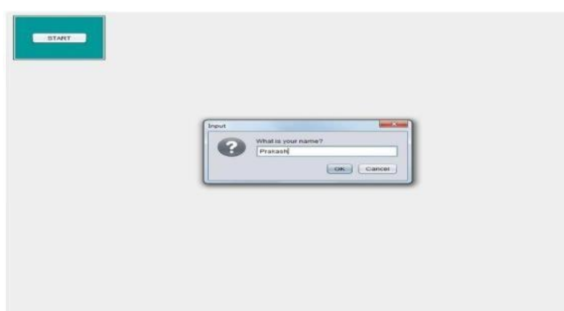
Follow the instructions given below to perform the experiments.

1.1 Starting the Experiments: -

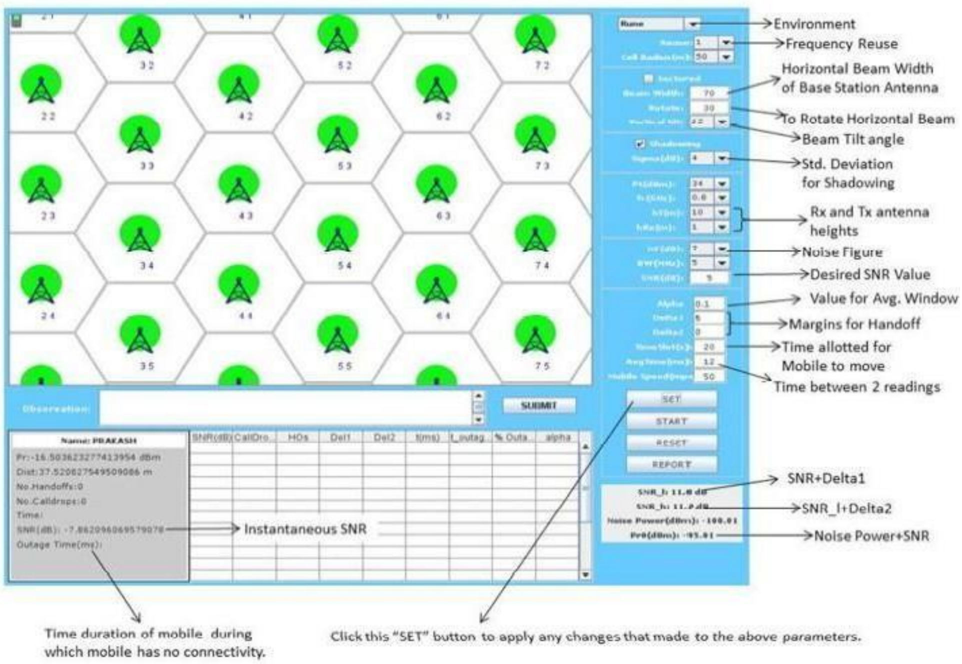
Step1: Click on START button to start experiment.



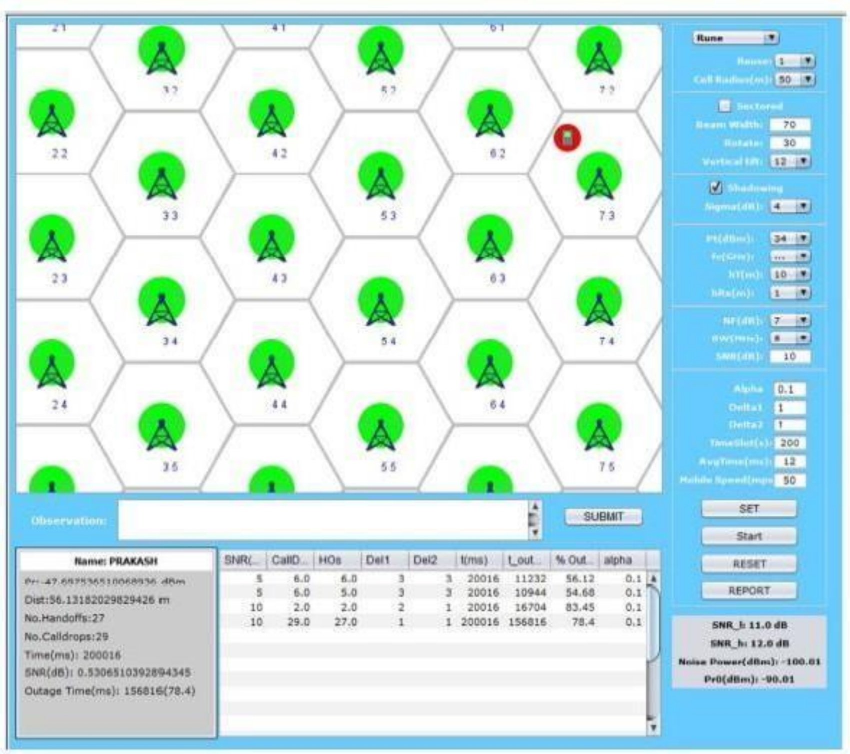
Step2: Enter your name then click OK button.



Step3: Select the parameters (e.g.: Reuse, Environment, Beamwidth, Carrier frequency etc.)



Step4: Click on START button and observe No. of Call Drops and No. of Handoffs.

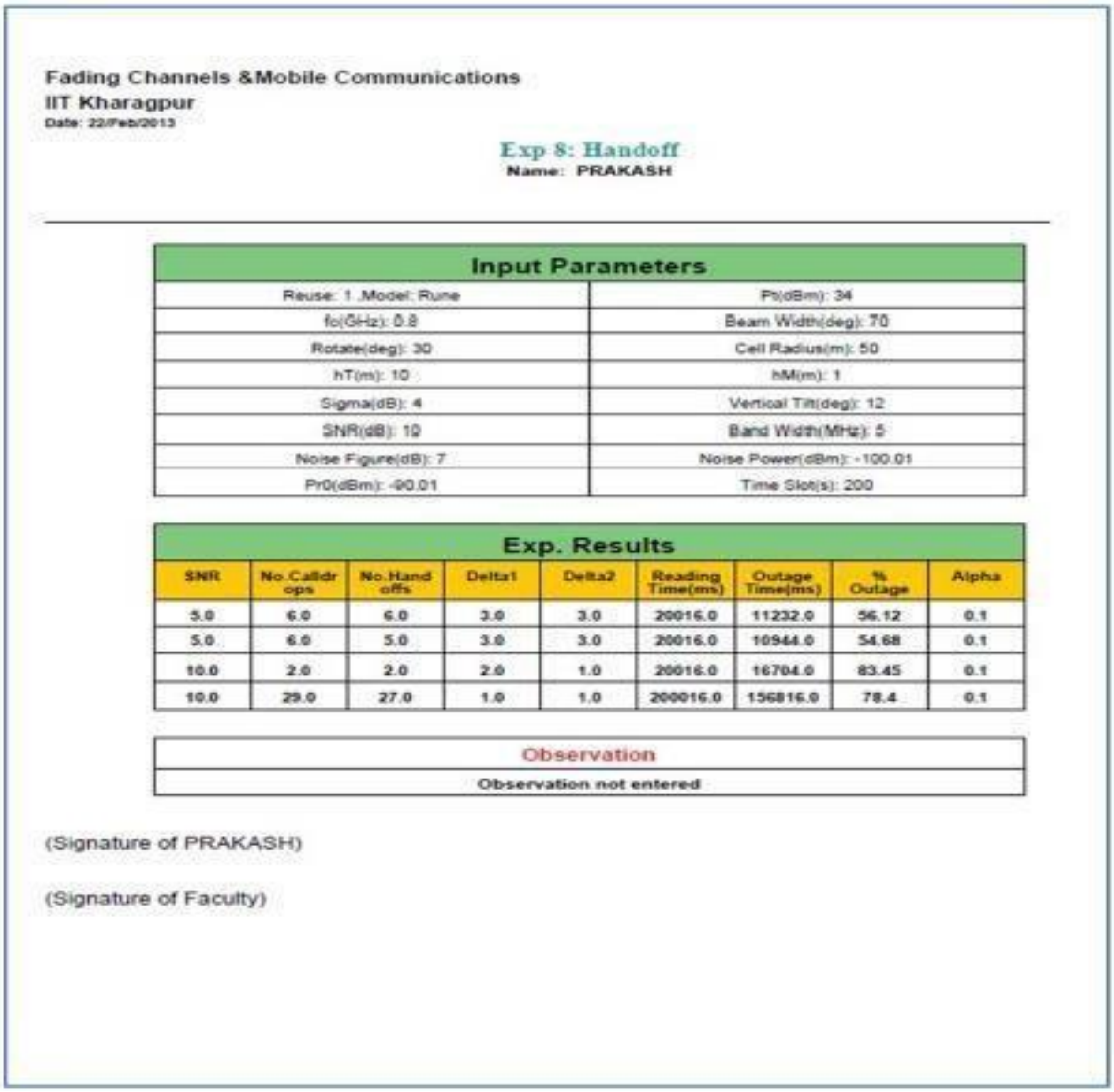


Step5: Enter your observation in the OBSERVATION box and Click on SUBMIT button.

Step6: Finally, click on REPORT to generate PDF report of the experiment.

Step7: After PDF report generation you will get following message.

Step8: PDF report will appear like this.



Fading Channels &Mobile Communications
IIT Kharagpur
Date: 22/Feb/2013

Exp 8: Handoff
Name: PRAKASH

Input Parameters	
Reuse: 1 ,Model: Rune	Pt(dBm): 34
fc(GHz): 0.8	Beam Width(deg): 70
Rotate(deg): 30	Cell Radius(m): 50
hT(m): 10	hM(m): 1
Sigma(dB): 4	Vertical Tilt(deg): 12
SNR(dB): 10	Band Width(MHz): 5
Noise Figure(dB): 7	Noise Power(dBm): -100.01
Pr0(dBm): -90.01	Time Slot(s): 200

Exp. Results

SNR	No.Calldrops	No.Handoffs	Delta1	Delta2	Reading Time(ms)	Outage Time(ms)	% Outage	Alpha
5.0	6.0	6.0	3.0	3.0	20016.0	11232.0	56.12	0.1
5.0	6.0	5.0	3.0	3.0	20016.0	10944.0	54.68	0.1
10.0	2.0	2.0	2.0	1.0	20016.0	16704.0	83.45	0.1
10.0	29.0	27.0	1.0	1.0	200016.0	156816.0	78.4	0.1

Observation
Observation not entered

(Signature of PRAKASH)

(Signature of Faculty)

Step9: To redo experiment click on RESET button.

Observation Table:

Reuse	No of Hand Off	Mobile Speed	Outage	Outage Percentage
1				
3				

Keep reuse ratio 3 and set mobile speed to 50 mps and 100 mps and record the below data. What do we observe after increasing the speed of the mobile station?

Reuse	Mobile Speed	No of Hand off	Outage	Outage Percentage
3	50			
3	100			

Conclusion:

Experiment No. 7

Title: To study the outage probability, LCR & ADF in SISO for Selection Combining and MRC.

Date of Performance: **Roll No:**

Date of Submission: **University Seat No:**

Signature of Staff:

Experiment No. 7

To study the outage probability, LCR & ADF in SISO for Selection Combining and MRC.

Title: To study the outage probability, LCR & ADF in SISO for Selection Combining and MRC.

Theory:

Small scale fading characterizes the fluctuation of signal (strength) over a spatial distance of fraction of wavelength. The fluctuation is also observed in both time and frequency domain at a gain location. The variation of signal (strength) at the receiver is due to random interference between the different copies of the transmitted signal. The interference is sometimes constructive and sometimes destructive. The multiple copies of the transmitted signal are generated due to scattering, reflection, and diffraction due to obstacle present in the path of radio signal between the Tx and Rx movement of the Tx and Rx or the obstacle cause time domain variation of the signal (strength) and the phenomenon is called Doppler effect. Since each path of the radio wave may exhibit difference doppler its cumulative effect results in spread of the carrier/ frequency content of the signal and hence is also known as Doppler spread.

If v is the maximum velocity (m/s) then the maximum Doppler shift is given by ;

`f_m=(upsilon(m//s))/c`

Where,

- C =Velocity light=3**10^8m/s.
- fc =Carrie frequency.

Coherence time is defined as interval in time over which the signal remains correlated. It is defined as ;

`T_c=9/(16Pif_m) (s)`

If symbol duration Ts << Tc it experience slow fading while if Ts>Tc it experience fast fading. The enveloped level crossing rate is defined as the rate at which the signal envelope crosses a specified level R in the positive (or negative) going direction.

It requires the joint pdf (α,.α) of the enveloped level α=|r| and enveloped slope .α=|r|

`L_R=sqrt(2pi(k+1))f_m rho e^(-k-(k+1)rho^2) I_0(2 rho sqrt(k(k+1))) rho=R/sqrt(Omega_p)=R/R_(rms)`

`R_(rms)=sqrt(Omega_p)` is the enveloped level

Rayleigh fading (k=0) and isotropic scattering

`L_R=sqrt(2pi)f_m rho e^(-rho^2)`

Level Crossing Rate For Selection Combining

`L_r=f_m sqrt(pi)M gamma/sqrt(sigma) exp(-gamma^2/(2sigma)) [1-exp(-gamma^2/(2sigma))]^(M-1)`

Where,

- fm is the Maximum doppler frequency.
- σ is the r.m.s value of the received signal voltage.
- γ is the threshold voltage.
- M = No. of channels Average enveloped fade duration

Level Crossing Rate for MRC Combining

`L_r=0.5f_m [(sqrt(2pi)(gamma/sqrt(M^2+M))^(M-1/2))/((M-1)!)]exp(-gamma/sq

The average fade duration

The average fade duration the enveloped remains below a specified level R.

`t=1/N_R P_r[r <=R]`

Average fade duration For Selection Combining

`ADF=(sqrt(rho)**exp(gamma^2/(2sigma)-1))/(sqrt(2pi)f_d M_(gamma))`

For Rayleigh distribution fading

`P_r[r <=R]=int_0^R P_r(dr)=1-exp(-rho^2)`

`bar t=(e^(rho^2-1))/(rho f_m sqrt(2pi))`

In case of flat fading the plot of signal enveloped of transmitting 'r' is given as;

p(r)=rσ2exp(−r^2/(2rho2))(0≤r≤∞) =0(r <0)

Where,

- σ is the r.m.s value of the received voltage signal before detection.
- σ2is the time average power of the received signal before enveloped detection.

Probability of outage is defined as;

P(R)=Pr(r≤R)=∫R_0[p(r)dr=1−exp(−R^2/(2σ^2))]

The mean value rmean of rayleigh distribution is given by;

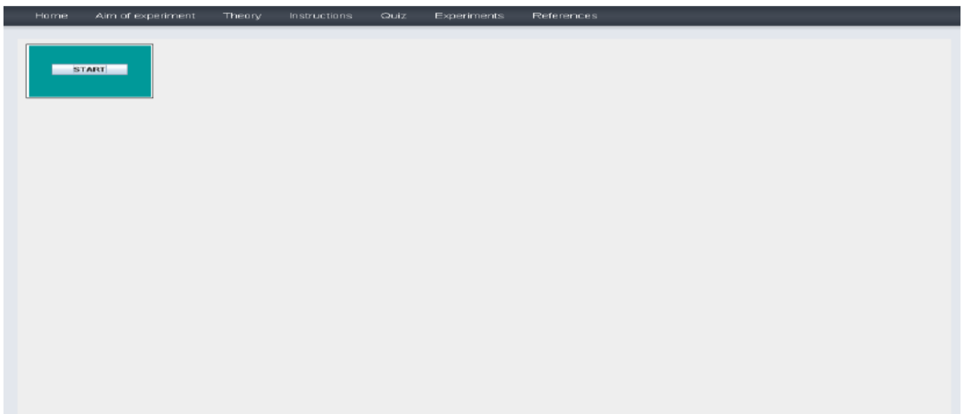
rmean=E[r]=∫∞_0[rp(r)dr]=σ√(π/2)=1.2533σ

$$\sigma^2 r = E[r^2] - E^2[r] = \int_0^\infty [r^2 p(r) dr - (\sigma^2 \pi)/2$$

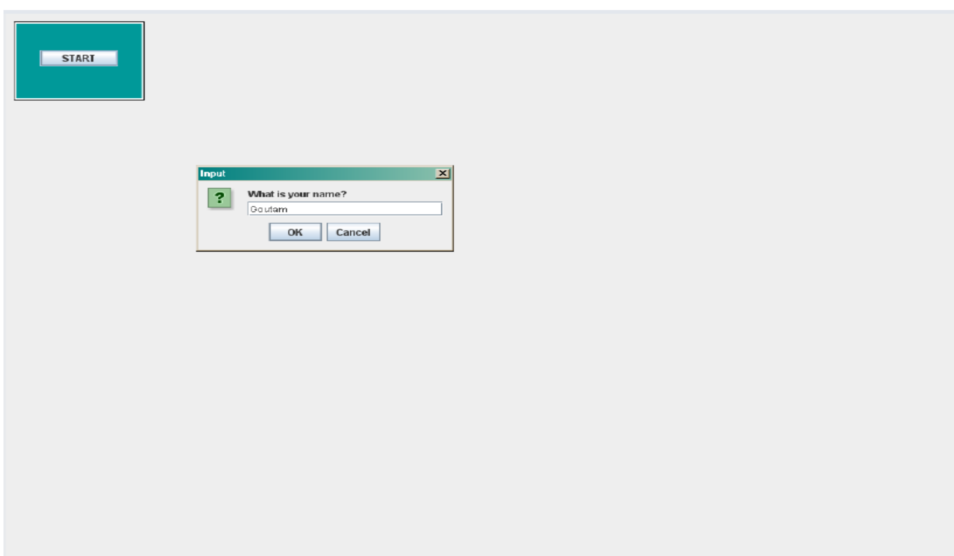
$$= \sigma^2(2 - \pi/2) = 0.4292\sigma^2$$

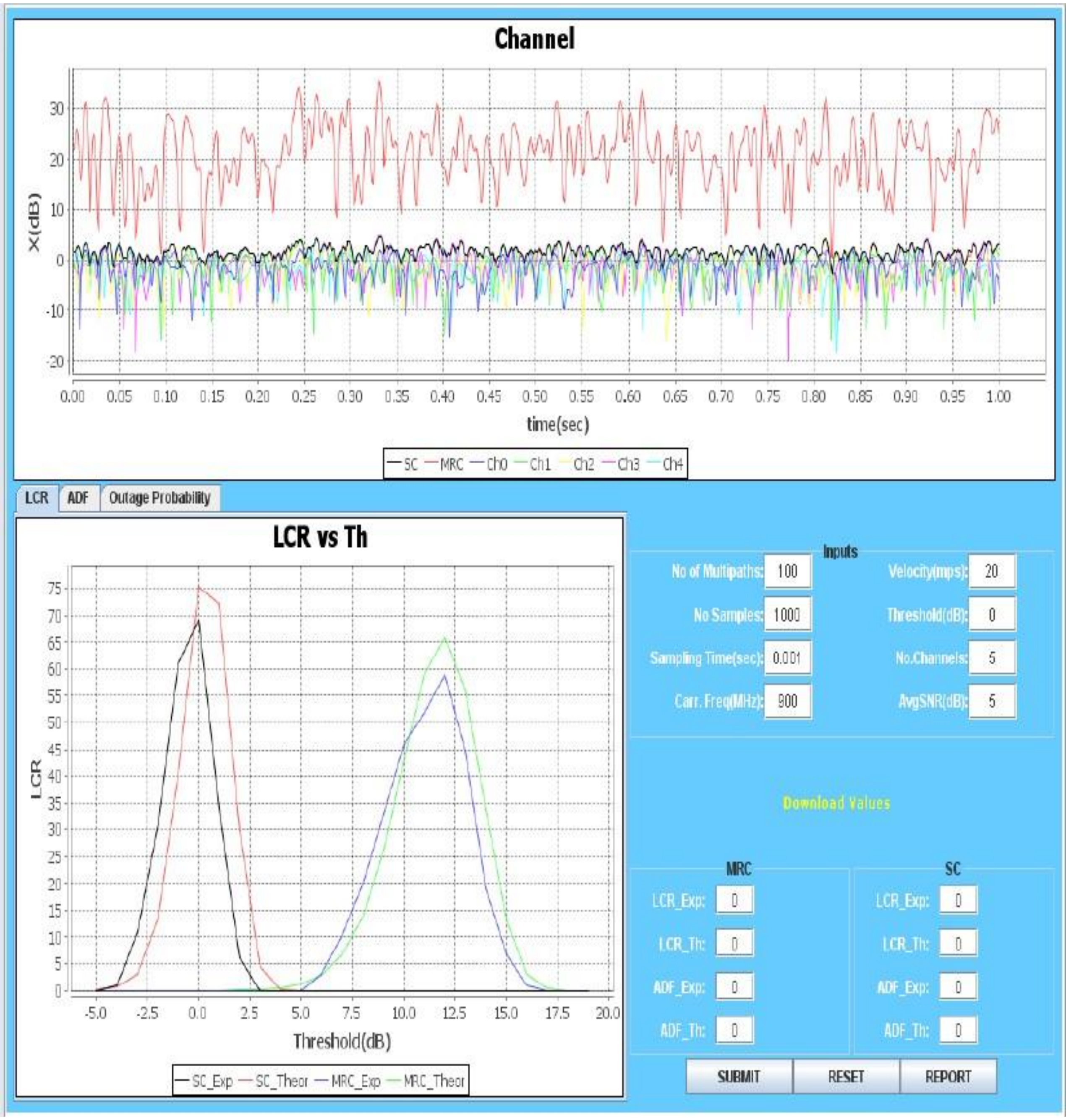
Procedure:

- Step1:- Click on the button START. A page appears with a dialogue box asking for your name.

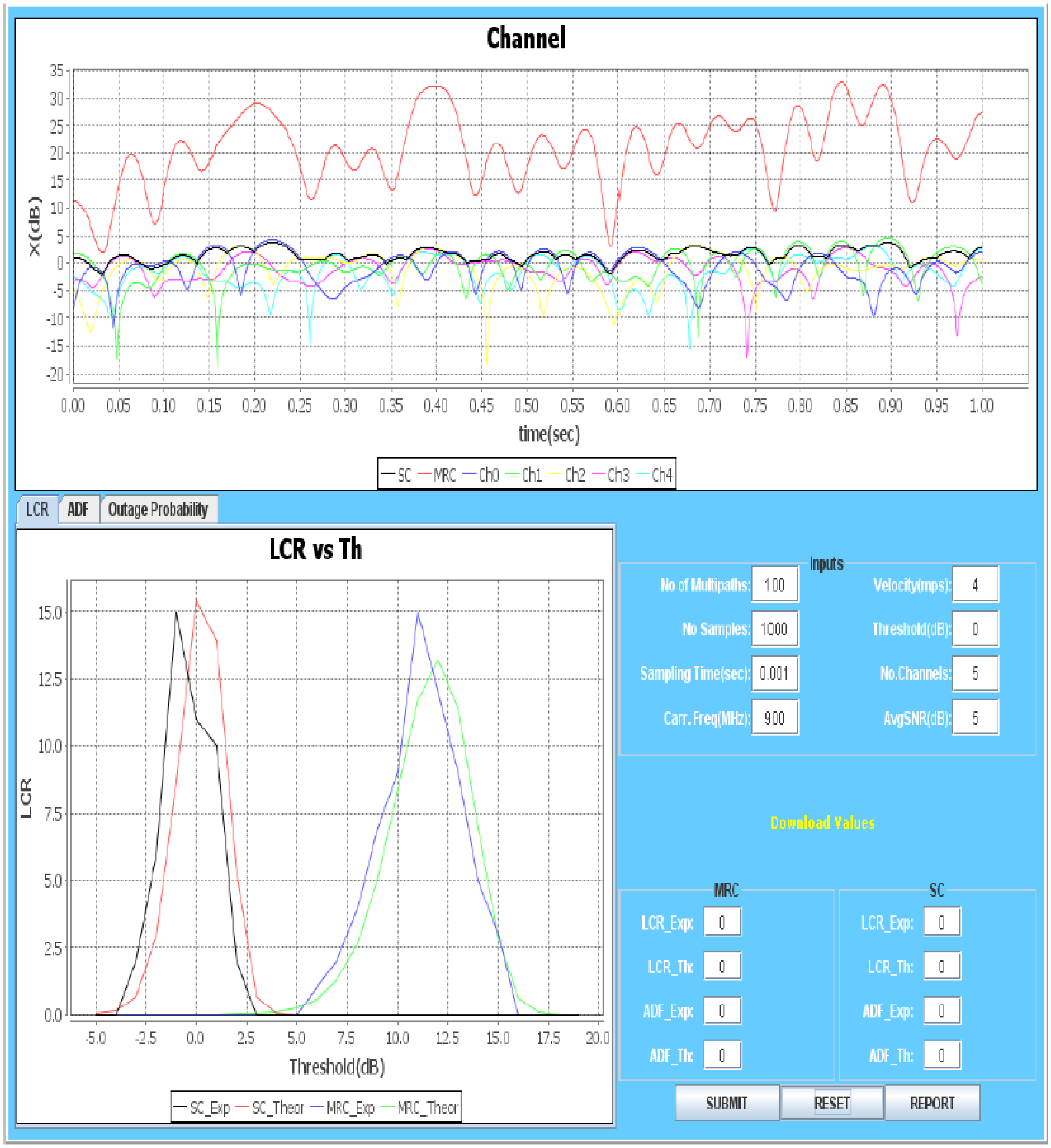


- Step 2:- Enter your name then Click Ok.

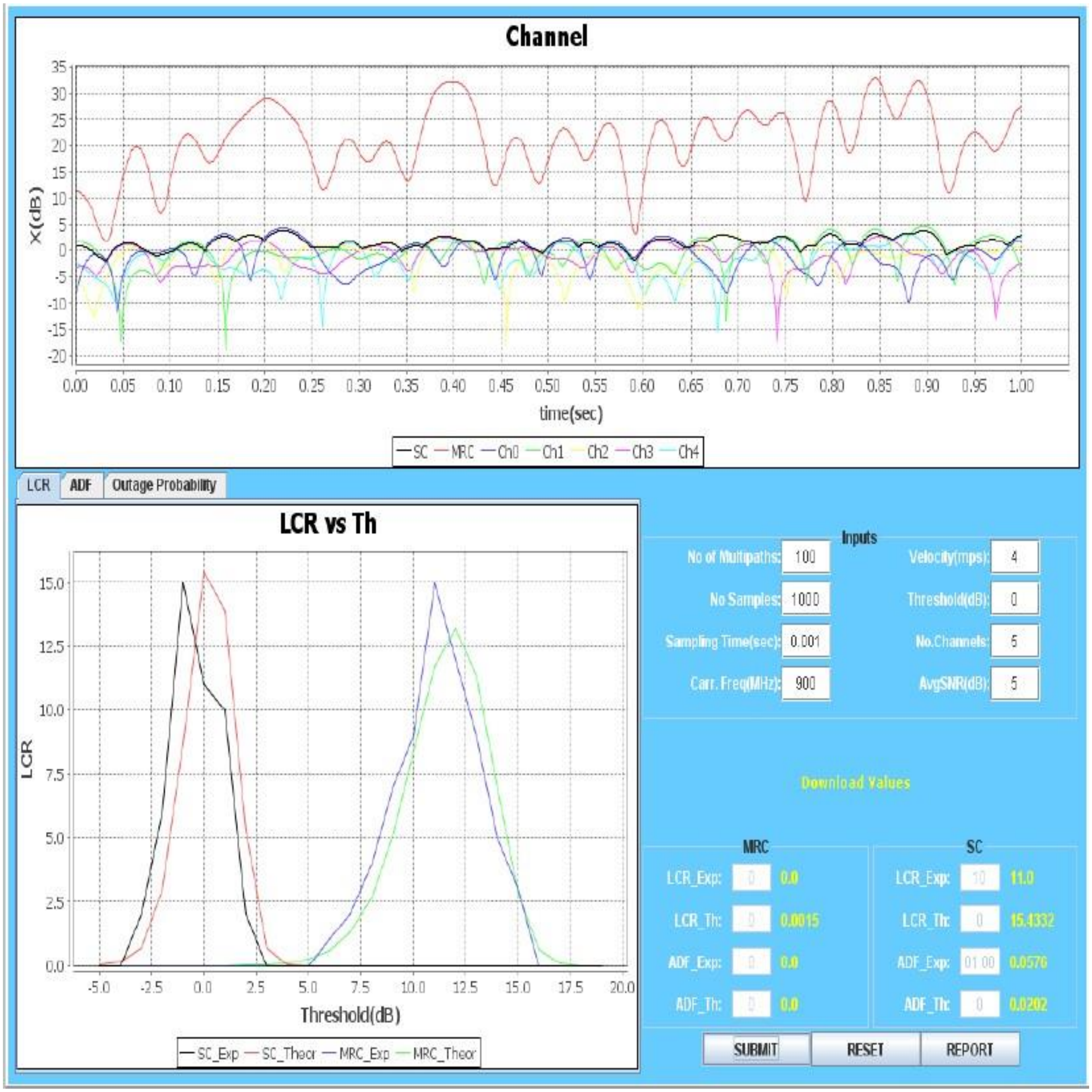




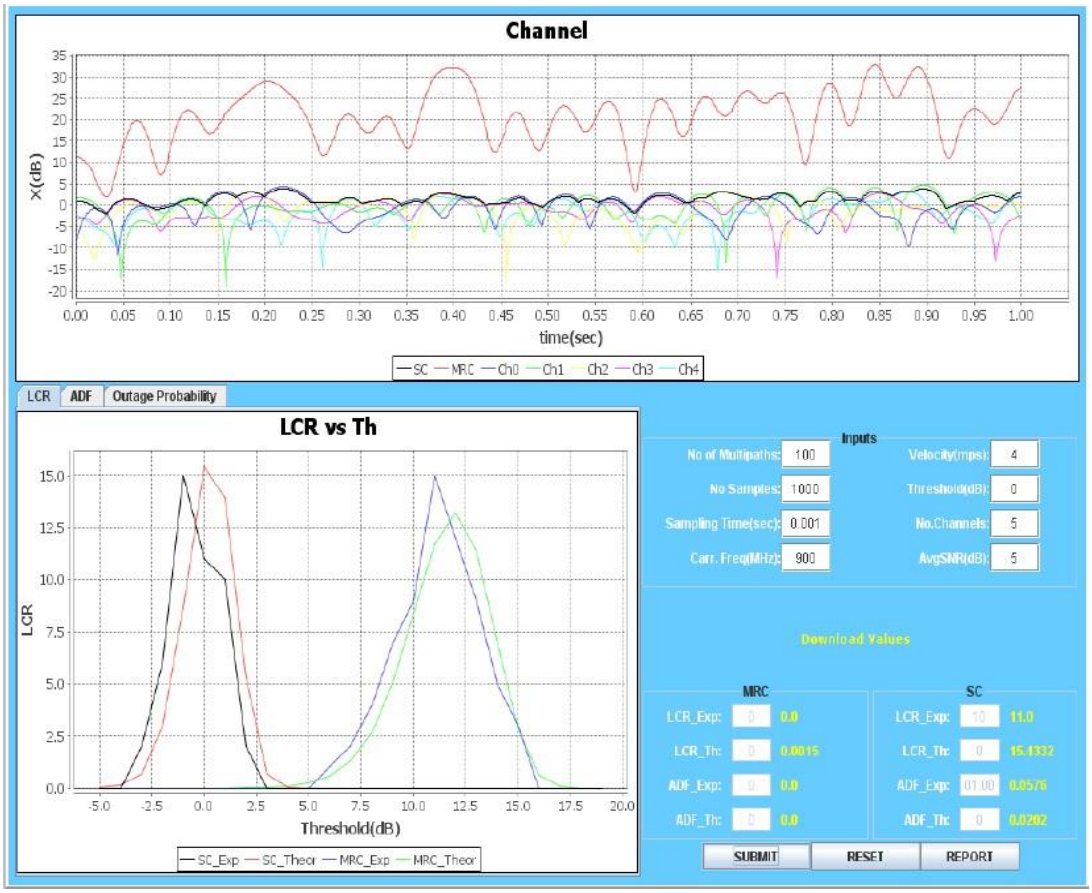
- Step3:- Enter the input parameters value. Then click on "RESET" Button.Observed the waveform.



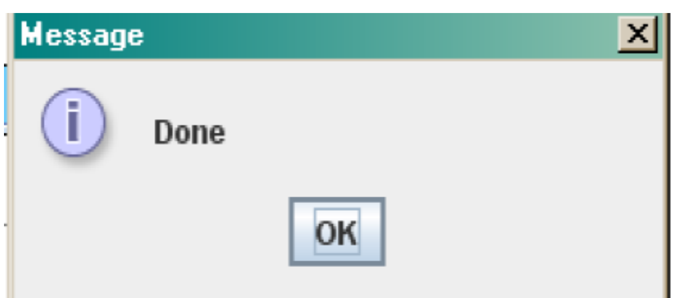
- Step4:- Enter value of LCR Exp and ADF Exp in both MRC and SC from the waveform.Then Click on "SUBMIT" Button.



- Step5:- Click on the "Report" button.



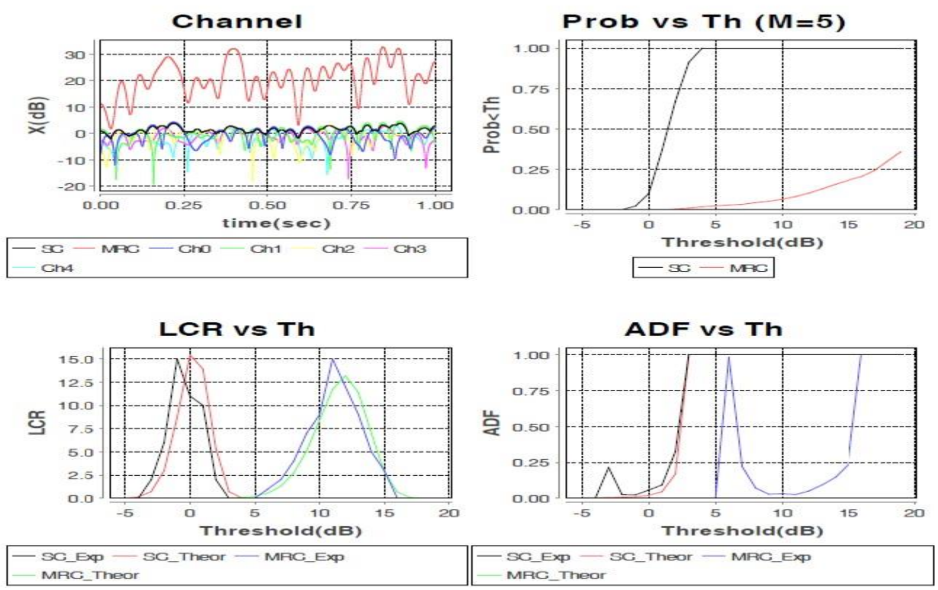
- Step6:- PDF report of the experiment is generated. Save report in Documents.
- Step7:-After generation of the Report you will get following message.



Step8:- Click on the "Ok" and you will get your report.

Fading Channels &Mobile Communications
IIT Kharagpur
Date: Nov/6/2012

Exp9: Flat Fading
Name: GOUTAM



Input Parameters	
No.of Multipaths	100
No.Samples	1000
Sampling Time(sec)	0.001
fc(Hz)	9.0E8
Velocity(mps)	4.0
Threshold(dB)	0.0
No.of Branches	5
Avg SNR(dB)	5.0

Results

Type	LCR(Exp)	LCR(Th)	ADF(Exp)	ADF(Th)
SC(Actual)	11.0	15.4332	0.0576	0.0202
SC(Entered)	10.0	0.0	1.0	0.0
MRC(Actual)	0.0	0.0015	0.0	0.0
MRC(Entered)	0.0	0.0	0.0	0.0

(Signature of GOUTAM)

(Signature of Faculty)

- Step9:- To Redo the experiment click on "RESET" button.

Conclusion:

Experiment No. 8

Title: File sharing by using TCP Protocol

Date of Performance: **Roll No:**

Date of Submission: **University Seat No:**

Signature of Staff:

Experiment No. 8

File sharing by using TCP Protocol

Title: To Perform File Transfer in Client & Server Using TCP/IP.

Software Requirements:

Python, Open-source Linux operating system.

Theory:

What is TCP?

TCP stands for Transmission Control Protocol. It is a communication protocol that is designed for end-to-end data transmission over a network. TCP is basically the " standard" communication protocol for data transmission over the Internet.

It is a highly efficient and reliable communication protocol as it uses a three-way handshake to connect the client and the server. It is a process that requires both the client and the server to exchange synchronization (SYN) and acknowledge (ACK) packets before the data transfer takes place.

Some of the features of the TCP are as follows:

- It provides end-to-end communication.
- It is a connection-oriented protocol.
- It provides error-checking and recovery mechanisms.

What is mean by Socket?

Sockets allow communication between two different processes on the same or different machines. To be more precise, it's a way to talk to other computers using standard Unix file descriptors. In Unix, every I/O action is done by writing or reading a file descriptor. A file descriptor is just an integer associated with an open file and it can be a network connection, a text file, a terminal, or something else.

To a programmer, a socket looks and behaves much like a low-level file descriptor. This is because commands such as read() and write() work with sockets in the same way they do with files and pipes.

Types of Socket

A Unix Socket is used in a client-server application framework. A server is a process that performs some functions on request from a client. Most of the application-level protocols like FTP, SMTP, and POP3 make use of sockets to establish connection between client and server and then for exchanging data. **Socket Types**

There are four types of sockets available to the users. The first two are most commonly used and the last two are rarely used. Processes are presumed to communicate only between sockets of the same type but there is no restriction that prevents communication between sockets of different types.

Stream Sockets

Delivery in a networked environment is guaranteed. If you send through the stream socket three items "A, B, C", they will arrive in the same order − "A, B, C". These sockets use TCP (Transmission Control Protocol) for data transmission. If delivery is impossible, the sender receives an error indicator. Data records do not have any boundaries.

Datagram Sockets Delivery in a networked environment is not guaranteed. They're connectionless because you don't need to have an open connection as in Stream Sockets − you build a packet with the destination information and send it out. They use UDP (User Datagram Protocol).

Raw Sockets

These provide users access to the underlying communication protocols, which support socket abstractions. These sockets are normally datagram oriented, though their exact characteristics are dependent on the interface provided by the protocol. Raw sockets are not intended for the general user; they have been provided mainly for those interested in developing new communication protocols, or for gaining access to some of the more cryptic facilities of an existing protocol.

Sequenced Packet Sockets

They are similar to a stream socket, with the exception that record boundaries are preserved. This interface is provided only as a part of the Network Systems (NS) socket abstraction, and is very important in most serious NS applications. Sequenced-packet sockets allow the user to manipulate the Sequence Packet Protocol (SPP) or Internet Datagram Protocol (IDP) headers on a packet or a group of packets, either by writing a prototype header along with whatever data is to be sent, or by specifying a default header to be used with all outgoing data, and allows the user to receive the headers on incoming packets.

The client-server model

The client-server model is one of the most commonly used communication paradigms in networked systems. Clients normally communicate with one server at a time. From a server's perspective, at any point in time, it is not unusual for a server to be communicating with multiple clients. Client need to know of the existence of and the address of the server, but the server does not need to know the address of (or even the existence of) the client prior to the connection being established. The client and the server on the same local network (usually called LAN, Local Area Network), the client and the server may be in different LANs, with both LANs connected to a Wide Area Network (WAN) by means of routers

Transmission Control Protocol (TCP)

TCP provides a connection oriented service, since it is based on connections between clients and servers. TCP provides reliability. When a TCP client sends data to the server, it requires an acknowledgement in return. If an acknowledgement is not received, TCP automatically retransmit the data and waits for a longer period of time for acknowledgement.

TCP Socket API

The sequence of function calls for the client and a server participating in a TCP connection is presented in following Figure.

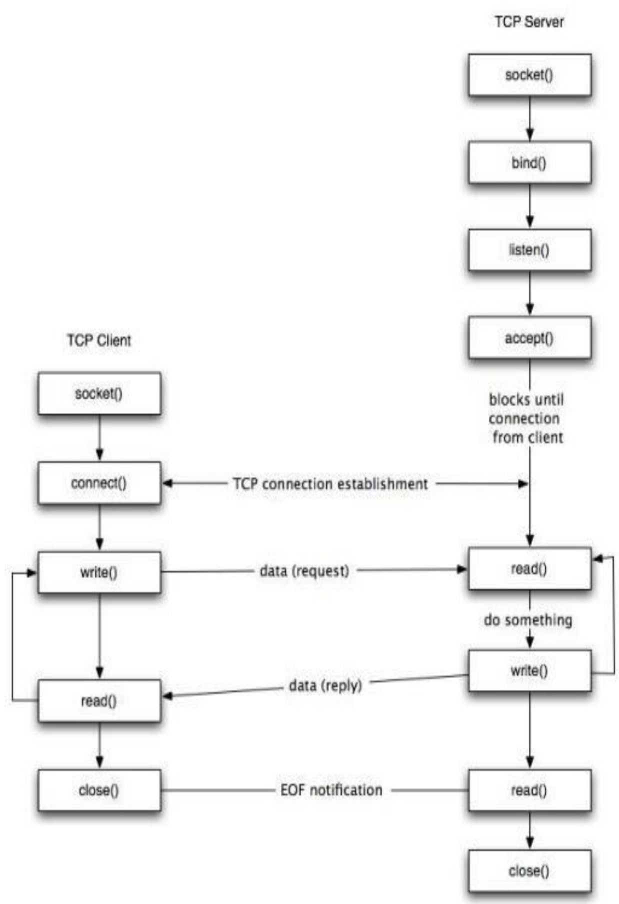


Figure: TCP client-server.

As shown in the figure, the steps for establishing a TCP socket on the client side are the following:

➢ Create a socket using the socket() function;

➢ Connect the socket to the address of the server using the connect() function;

➢ Send and receive data by means of the read() and write() functions.

➢ Close the connection by means of the close() function.

➢ The steps involved in establishing a TCP socket on the server side are as follows:

➢ Create a socket with the socket() function;

➢ Bind the socket to an address using the bind() function;

➢ Listen for connections with the listen() function;

➢ Accept a connection with the accept() function system call. This call typically blocks until a client connects with the server.

➢ Send and receive data by means of send() and receive().

➢ Close the connection by means of the close() function.

Procedure:

Server-side:

1. Import the required libraries: socket and os.
2. Define the server IP address and port number.
3. Create a TCP socket using the socket.socket() method and bind it to the server IP address and port number using the socket.bind() method.
4. Set the socket to listen for incoming connections using the socket.listen() method.
5. Accept incoming connections using the socket.accept() method. This returns a new socket object and the address of the client.
6. Receive the file data from the client using the socket.recv() method in a loop until all data is received.
7. Write the received data to a file using the os library.
8. Close the socket and the file.

Client-side:

1. Import the required libraries: socket and os.
2. Define the server IP address and port number, and the file path.
3. Create a TCP socket using the socket.socket() method and connect to the server using the socket.connect() method.
4. Read the file data from the file system using the open() method and the read() method.
5. Send the file data to the server using the socket.sendall() method.
6. Close the socket.

Conclusion: