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Department of Electronics and Telecommunication Engineering

Mobile Computing Lab Manual

B.E. E&TC

WEF Academic Year 2022-23

Experiment No. –

Subject: - Mobile Computing

Name of the Student:___________________________________Roll No.____________

Date: _______________ Marks & Signature: -

Subject Teacher

Dr D Y Patil Prathishthan's

D. Y. PATIL COLLEGE OF ENGINEERING, Akurdi, Pune

Department of Electronics and Telecommunication

Subject: Mobile Computing (Elective - V)

Class: BE(E&TC)

List of Experiments:

Sr. No.	Title of the experiment
1	Simulate to elaborate operation of multiple access techniques for CDMA.
2	Study of GSM architecture and signaling techniques.
3	Study of GPRS services.
4	Simulate BER performance over Rayleigh Fading wireless channel with BPSK transmission for SNR 0 to 60 dB.
5	Configuring a Cisco Router as a DHCP Server.
6	To understand the handover mechanism.
7	To study the outage probability, LCR & ADF in SISO for Selection Combining and MRC (Flat Fading).
8	To Perform File Transfer in Client & Server Using TCP/IP.
9	Case Study on different real time mobile computing services.

Experiment No.–1

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.

AIM:

Basic function of Code Division Multiple Access (CDMA).

OBJECTIVES:

To understand function of CDMA used to test orthogonally and autocorrelation of a code.

SOFTWARE & HARDWARE REQUIREMENTS:

OS.: Unix or windows 7/8/10,

Processor: i3/i5/i7

Software: Python (Jupyter Notebook) or java

THEORY-CONCEPT

CDMA stands for Code Division Multiple Access. It is a digital cellular standard that utilizes spread-Spectrum Technology. It spreads the signal over a fully available spectrum or over multiple channels through division. It is a channelization protocol for Multiple Access, where information can be sent simultaneously through several transmitters over a single communication channel.

It is achieved in below steps: A signal is generated which extends over a wide bandwidth. The code which performs this action is called spreading code. Later, a specific signal can be selected with a given code even in the presence of many other signals. It is mainly used in mobile networks like 2G and 3G. It is a more secure and private line. It has good voice and data communication capabilities.

Procedure or Working

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.

How does CDMA work?

To see how CDMA works, we must understand orthogonal sequences (also known as chips).

Let N be the number of stations establishing multiple access over a common channel.

Then the properties of orthogonal sequences can be stated as follows:

An orthogonal sequence can be thought of as a 1xN matrix.

Eg: [+1 -1 +1 -1] for N = 4.

Scalar multiplication and matrix addition rules follow as usual.

Eg: 3.[+1 -1 +1 -1] = [+3 -3 +3 -3]

Eg: [+1 -1 +1 -1] + [-1 -1 -1 -1] = [0 -2 0 -2]

Inner Product: It is evaluated by multiplying two sequences element by element and then adding all elements of the resulting list.

Inner Product of a sequence with itself is equal to N

[+1 -1 +1 -1].[+1 -1 +1 -1] = 1 + 1 + 1 + 1 = 4

Inner Product of two distinct sequences is zero

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

Ref 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)
```

CONCLUSION:

ORAL QUESTIONS:

1. What is CDMA?
2. Write down difference between FDMA TDMA and CDMA

Experiment No.–2

Title: Study of GSM architecture and signaling techniques.

INTRODUCTION

The development of Global System for Mobile Communication (GSM) started in 1982 when the Conference of European Posts and Telegraphs (CEPT) formed a study group called Groupe Special Mobile (the initial meaning of GSM) whose aim was to study and develop a pan-European public cellular system in the 900 MHz range [6]. Some of the basic criteria for their proposed system were

- Good subjective speech quality
- Low terminal and service cost
- Support for international roaming
- Ability to support handheld terminals
- Support for range of new services and facilities
- Spectral efficiency
- ISDN compatibility

Commercial operation of GSM networks started in mid-1991 in European countries. By the beginning of 1995, there were 60 countries with operational or planned GSM networks in Europe, the Middle East, the Far East, Australia, Africa, and South America, with a total of over 5.4 million subscribers. GSM uses a mixture of both Frequency Division Multiple Access (FDMA) and Time Division Multiple Access (TDMA) [2]. FDMA parts include the division by frequency of the 25 MHz bandwidth into 124 carrier frequencies spaced 200 KHz for GSM900. TDMA further divides each carrier frequencies into 8-time slots such that each carrier frequency is shared by 8 users. In GSM, the basic radio resource is a time slot with duration of 577 μs. 8 Time slots of 577 μs constitute a 4.615 ms TDMA Frame. GSM uses Gaussian Minimum Shift Keying (GMSK) modulation scheme to transmit information over Air Interface [1]. GSM uses number of channels to carry data over air interface; these channels are broadly divided into following two categories:

i. Physical Channels

ii. Logical Channels

This paper is divided into five parts. Starting with an introduction (Section-I), next section covers the need of GSM (Section-II). Moving ahead, GSM architecture is discussed (Section-III). Next section is about the interfaces in between the GSM subsystems (Section-IV). After that GSM channels are discussed (Section-V). This section is about the call origination using GSM channels (Section-VI). Finally, conclusions summarize the last section (Section-VII).

NEED OF GSM

Features of GSM that account for its popularity and wide acceptance are listed below.

i. Improved spectrum efficiency

ii. International roaming

iii. Low-cost mobile sets and base stations (BSs)

High-quality speech

iv. Compatibility with Integrated Services Digital Network (ISDN) and other telephone company services

v. Support for new services.

Architecture

GSM architecture is classified into three subsystems i.e., BSS, NSS, and OSS.

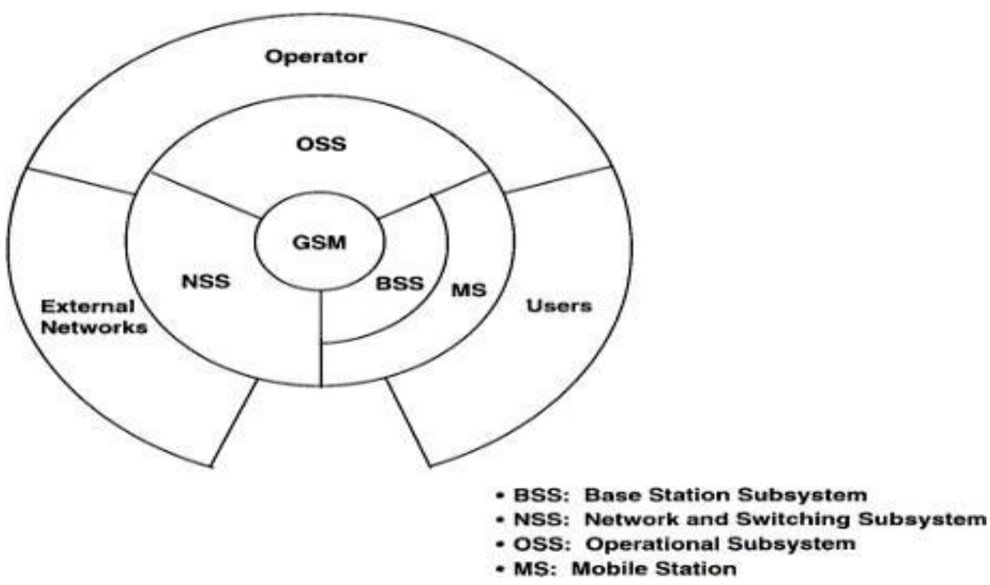


Fig : Basic Architecture

The BSS provides and manages communication paths between the MSs and the NSS. It includes management of the radio interface between MSs and the whole GSM system. The NSS has the province of managing communications and linking MSs to the relevant networks or other MS's. Neither NSS is in direct contact with the MSs, nor is the BSS in direct contact with external networks [10], [5]. In the GSM, interaction between the subsystems can be grouped into two main parts:

i. Operational: External networks to or from NSS to or from BSS to or from MS to or from subscriber

ii. Control: OSS to or from service provider

1. GSM Subsystem Entities

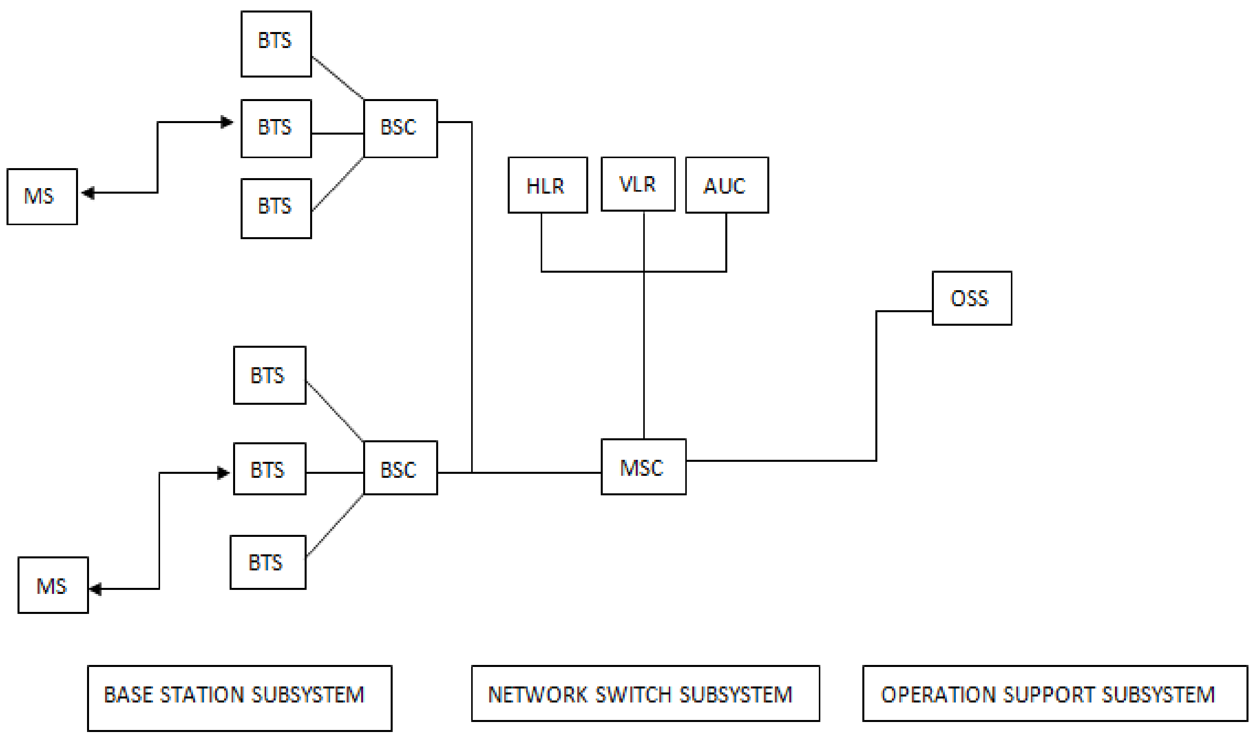


Fig: GSM Architecture

a) Mobile Station (MS):

A Mobile Station is used by a mobile subscriber to communicate with the cellular system. GSM MSs consist of:

i. A mobile terminal
ii. A Subscriber Identity Module (SIM)

b. Base Transceiver System (BTS) –
The BTS is the radio equipment. It handles the radio interface to the mobile station. A group of BTSs is controlled by a BSC.
ii. Base Station Controller (BSC) - The BSC is a high-capacity switch which provides all the control functions and physical links between the MSC and BTS such as handover, cell configuration data, and control of radio frequency (RF) power levels in base transceiver stations. Several BSCs are served by an MSC.

c) Network Switching Subsystem (NSS):
The NSS main role is to manage the communications between GSM and other network users. Another task of it includes the main switching functions of GSM, databases required for the subscribers, and mobility management.
i. Mobile Station Controller (MSC): The MSC is the central unit of NSS. It performs the telephony switching functions of the system. It controls the traffic among all the BSCs.
ii. Home Location Receiver (HLR): The HLR is the functional unit used for management of mobile subscribers. The information stored in it are subscriber’s identity, location, and authentication. It acts as a permanent store for a person’s subscription information until that subscription is canceled.
iii. Visitor Location Receiver (VLR): The VLR is the functional unit which temporarily stores subscription information so that the MSC can service all the subscribers currently visiting that MSC service area. The VLR is regarded as distributed HLR because it holds a copy of the HLR information stored about the subscriber.
iv. Authentication Centre (AUC): The AUC is a database connected to the HLR which provides it with the authentication parameters and ciphering keys used to ensure network security.

d) 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 support system (OSS). The OMC provides a single point for the maintenance personnel to maintain the entire system. One OMC can serve multiple MSCs.

I. GSM INTERFACES

The interfaces between various elements of the GSM network facilitates the information interchange. It also enables the case that network elements from different manufacturers can be used [2].

i. Um interface: This interface is linking the Mobile Equipment (ME) and the Base Station (BTS/BSC). It exchanges information in air.

ii. Abis interface: This interface is linking the BSC and a BTS, and it has not been totally standardized. It is an internal interface as the BTS and the BSC are both part of the BSS. The Abis interface allows control of the radio equipment and radio frequency allocation in the BTS.

iii. A interface: This interface is linking the BSS and the MSC. This interface carries information to enable the channels, timeslots. The messaging required within the network to enable handover etc. to be undertaken is carried over this interface.

iv. B interface: This interface is linking the MSC and the VLR. It uses a protocol known as the MAP/B protocol, the letter “B” indicating that the protocol is used for “B” interface. This is an internal interface as the VLR is a part of the MSC. The interface is used by the MSC when it needs to access data regarding a MS located in its area [8].

v. C interface: This interface is linking the HLR and the MSC. The protocol used for communication is MAP/C, the letter "C" indicating that the protocol is used for the "C" interface. It helps in communication between the MSC and the HLR regarding various aspects, for example, subscriber information.

vi. D interface: This interface is linking the VLR and the HLR. It uses the MAP/D protocol, the letter "D" indicating that the protocol is used for the "D" interface. It is used to exchange the data related to the location of the Mobile Equipment and to the management of the subscriber.

vii. E interface: This interface is linking two MSCs. The E interface exchanges data related to handover between the two MSCs. It uses the MAP/E protocol, the letter "E" indicating that the protocol is used for the "E" interface.

viii. F interface: This interface is linking the MSC and EIR. It uses the MAP/F protocol, the letter "F" indicating that the protocol is used for the "F" interface. The communications along this interface are used to confirm the status of the International Mobile Equipment Identity (IMEI) of the Mobile Equipment (ME) gaining access to the network [9].

ix. H interface: This interface is linking the MSC and the AUC. It uses the MAP/H protocol, the letter "H" indicating that the protocol is used for the "H" interface. It transfers short messages.

In a system, information flows forward, backward and sideways. This information flow is referred to as communication. Communication channels refer to the way this information flows within the system [1][4].

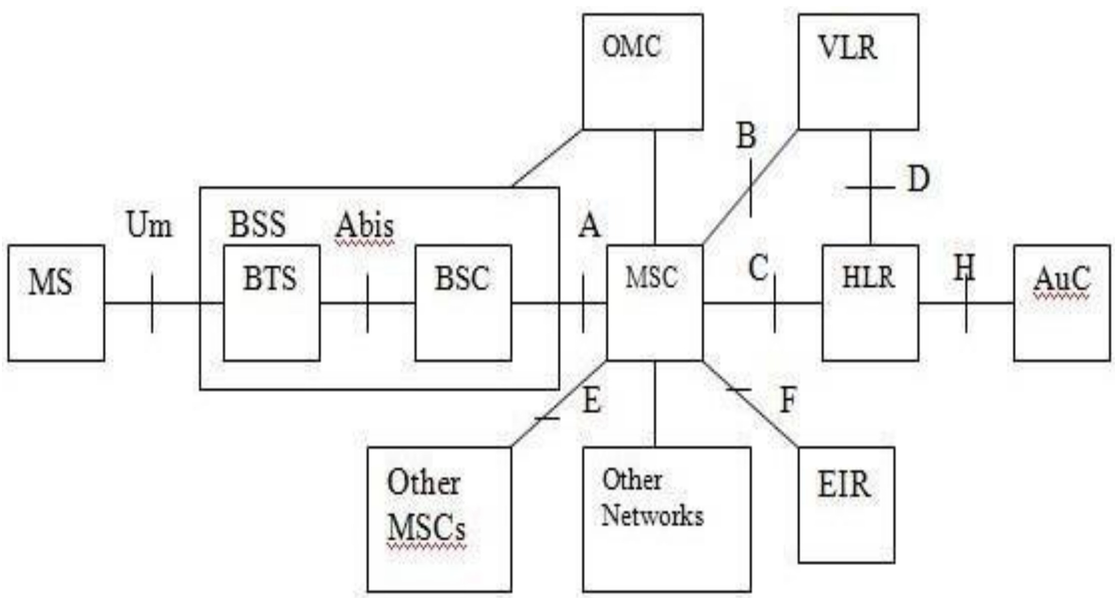


Fig: GSM Interfaces

I. GSM CHANNELS

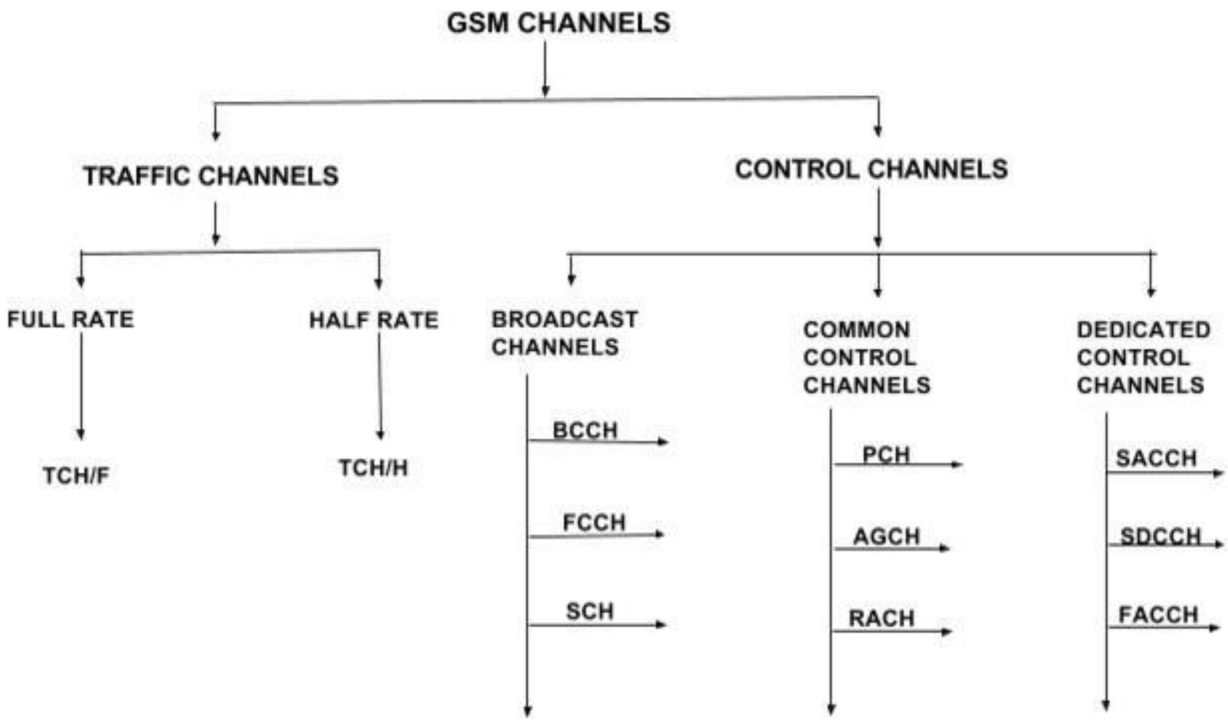


Fig : GSM Channels [1]

1. Traffic Channels (TCH): This channel carries digitally encoded user's speech or data and has identical functions and formats on both the forward and reverse link.

a) Full Rate Traffic Channels (TCH\F): This channel carries information at a rate of 22.8 Kbps.

b) Half Rate Traffic Channels (TCH\H): This channel carries information at a rate of 11.4 Kbps.

2. Control Channels (CCH): This channel carries control information to enable the system to operate correctly. There are three main categories of control channels in GSM which are further divided into several categories: i. Broadcast Channels (BCH): The broadcast channel operates on the forward link of a specific Absolute Radio Frequency Channel Number (ARFCN) within each cell, and transmits data only in the first time slot (TS0) of certain GSM frames.

ii. Common Control Channels (CCCH): This channel is responsible for transferring control information between all mobiles and the BTS. It is necessary for call origination.

a) Broadcast Control Channels (BCCH) – DOWNLINK

This channel broadcasts network and cell specific information required to identify the network and gain access.

b) Frequency Control Channels (FCCH) – DOWNLINK

This channel contains frequency correction bursts which are used by the mobiles for frequency correction. It bears information for frequency synchronization. It allows each subscriber unit to synchronize its internal frequency standard to the exact frequency of the base station.

c) Synchronization Channel (SCH) – DOWNLINK

This channel is used by the MS to learn the Base Station Information Code (BSIC) as well as the TDMA frame number (FN). There are 6 bits of BSIC which have two parts; 3 bits NCC and 3 bits BCC. NCC stands for Network Colour Code and used to identify the BTS for which measurement is made. BCC stands for Base Station Colour Code and used for a better transmission in case of interference [6]. BICS avoids ambiguity or interference.

a) Paging Channel (PCH) – DOWNLINK

This channel is used for alerting to Mobile Subscribers for incoming calls, SMS, and other mobility services. Every MS in a cell periodically listens to this channel.

b) Random Access Channel (RACH) – UPLINK

This channel is used to send a request to the network for a dedicated resource to the MS. If the request is not granted within a specific time period by the network, the MS repeats the request on the RACH.

c) Access Grant Channel (AGCH) - DOWNLINK

This channel is used by a BTS to notify the MS of the assignment of a dedicated control channel.

iii. Dedicated Control Channels (DCCH):

Like traffic channels, they are bidirectional and have the same format and function on both the forward and reverse links.

a) Standalone Dedicated Control Channel (SDCCH) – UPLINK/DOWNLINK

In response of RACH, the network allocates SDCCH over AGCH for further communication between MS and BTS. This channel is used for the Location Update, Voice Call Setup and SMS.

b) Fast Associated Control Channel (FACCH) – UPLINK/DOWNLINK

This channel is used to convey Handover information. There is no time slot and frame allocation dedicated to this channel. This channel can be associated with SDCCH or TCH and works on the principle of stealing. The burst of TCH is replaced by FACCH signaling when required.

c) Slow Associated Control Channel (SACCH) – UPLINK/DOWNLINK

This channel is always associated with TCH or SDDCH used for control and supervision of signals associated with the traffic channels.

II. CALL FLOW USING CHANNELS

The call origination made by the Mobile Station (MS) is a multi-step process on the network side. First of all, broadcast channels are already being transmitted by the BTS continuously. When a number is dialed by the MS, Random Access Channels (RACH) is sent to the BSC for allocation of the channel. In respond to that Access Grant Channel (AGCH) is send by BSC as a confirmation. Standalone Dedicated Control Channel (SDCCH) can be used for both downlinks as well as uplink [7]. Hence, it is used for multiple purposes in this flow. It is sent by the MS to MSC with its location update information.

Through SDCCH MSC request for the International Mobile Subscriber Identity (IMSI) number and then this information is sent back to MSC by MS by SDCCH [2]. Along with IMSI to MSC, SDCCH gives location update to HLR and information about equipment to Equipment Identity Register (EIR) which is a part of Authentication Centre (AUC). Using the same channel security and ciphering of call is done. After all, this process connection is established and ringing begins [6]. Also, the Traffic Channel (TCH) is allocated for the call. Hence, the conversation starts between the two MS's.

Conclusion:

FAQ:

1. What is GSM and architecture?
2. What is GSM Signalling?
3. What is GSM technique?

Experiment No.–3

Title: Study of GPRS services.

GPRS is an expansion Global System for Mobile Communication. It is basically a packet-oriented mobile data standard on the 2G and 3G cellular communication network's global system for mobile communication. GPRS was built up by European Telecommunications Standards Institute (ETSI) because of the prior CDPD, and I-mode packet switched cell advances.

GPRS overrides the wired associations, as this framework has streamlined access to the packet information's network like the web. The packet radio standard is utilized by GPRS to transport client information packets in a structured route between GSM versatile stations and external packet information networks. These packets can be straightforwardly directed to the packet changed systems from the GPRS portable stations.

History Of GPRS

GPRS was one of the main advances that empowered a cell system to interface with Internet Protocol systems, accomplishing across the board reception in the mid-2000s. The capacity to peruse the web from a telephone whenever through "dependably on" data networking, while underestimated in a great part of the world today, was yet an oddity when it was introduced. Indeed, even now, GPRS keeps on being utilized in parts of the world where it has been too expensive even to consider upgrading cell organize framework to move up to newer alternatives.

According to a study on the history of GPRS development Bernhard Walke and his student, Peter Decker, are the inventors of GPRS – the first system providing universal mobile Internet access.

Goals Of GPRS:

1. Consistent IP services

2. Leverage industry investment in IP

3. Open Architecture

4. Service innovation independent of infrastructure

Services Offered:

1. SMS messaging and broadcasting

2. Push-to-talk over cellular.

3. Instant messaging and presence

4. Multimedia messaging service

5. Point-to-Point and Point-to-Multipoint services

Protocols supported:

1. Internet Protocol (IP)

2. Point-To-Point Protocol (PPP)

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:

FAQ:

1. What is GPRS?

2. What are features of GPRS?

Experiment No.–4

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

Problem Statement:

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

Objectives:

- What is a Rayleigh Fading
- Study of BPSK transmission

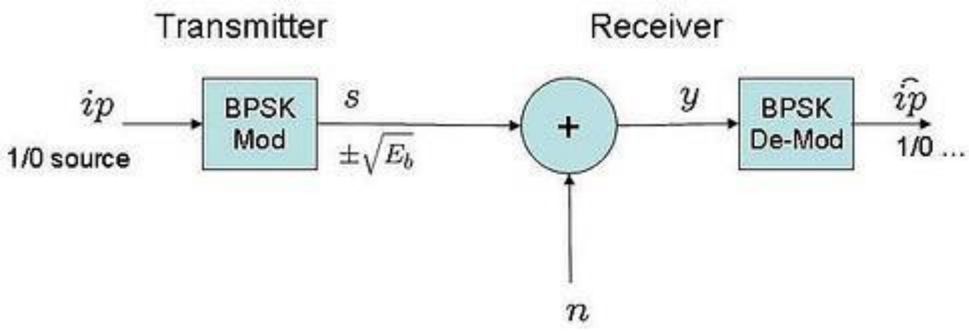
Software Requirements:

- Windows 7/10/11
- Matlab

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 n, 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 n follows the Gaussian probability distribution function, $p(x) = \frac{1}{\sqrt{2\pi\sigma^2}} e^{\frac{-(x-\mu)^2}{2\sigma^2}}$ with $\mu = 0$ 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 y for the two cases are:

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

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

.

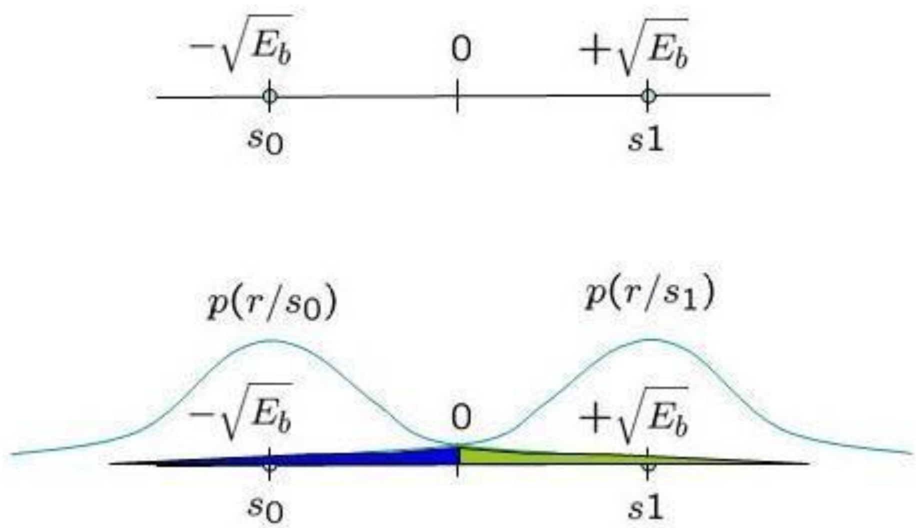


Figure: Conditional probability density function with BPSK modulation

Assuming that s_1 and s_0 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 s_1 was transmitted.
- if the received signal is y is less than or equal to 0, then the receiver assumes s_0 was transmitted.

i.e.

$y > 0 \Rightarrow s_1$ and $y \leq 0 \Rightarrow s_0$.

Probability of error given s_1 was transmitted.

With this threshold, the probability of error given s_1 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 s_0 was transmitted

Similarly the probability of error given s_0 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 s_1 and s_0 are equally probable i.e. $p(s_1) = p(s_0) = 1/2$, the **bit error probability** is,

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

Simulation model

Matlab/Octave 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

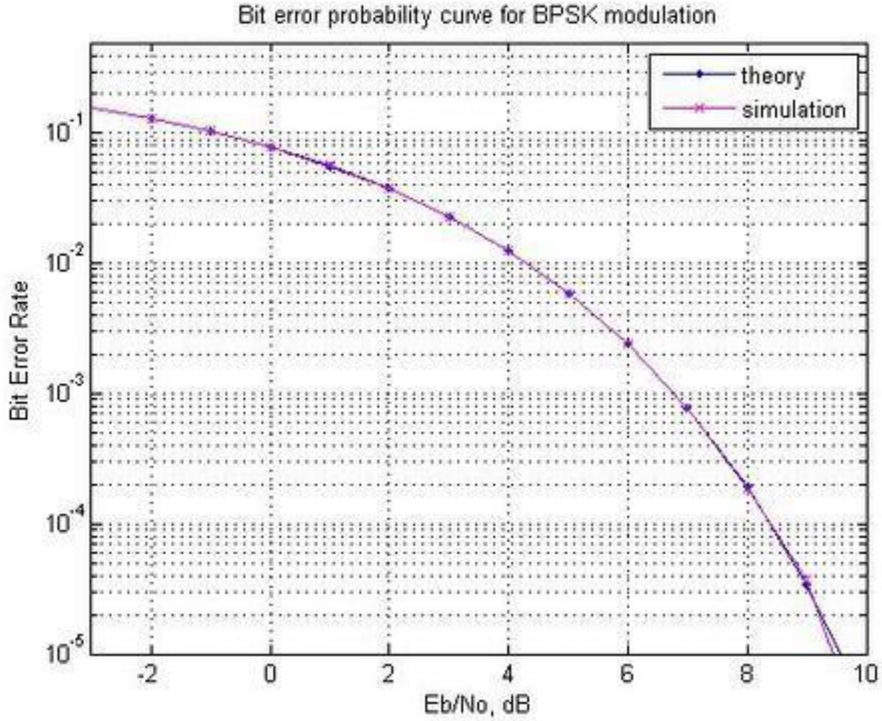
(d) Counting the number of errors

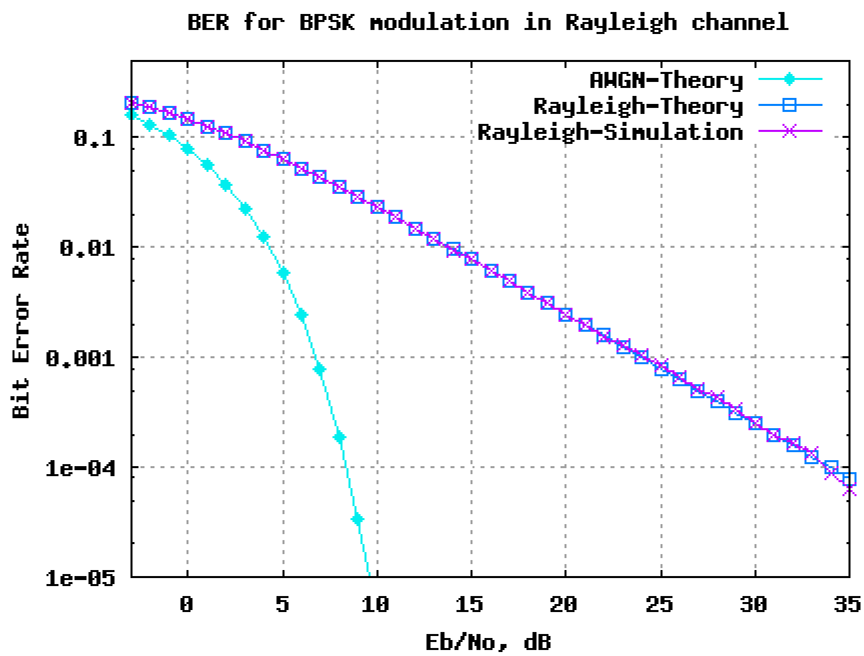
(e) Repeating the same for multiple Eb/No value.

Ref Code:

```
clear

N = 10^6 % number of bits or symbols

 rand('state',100); % initializing the rand() function

randn('state',200); % initializing the randn() function

% Transmitter

ip = rand(1,N)>0.5; % generating 0,1 with equal probability

s = 2*ip-1; % BPSK modulation 0 -> -1; 1 -> 1

n = 1/sqrt(2)*[randn(1,N) + j*randn(1,N)]; % white gaussian noise, 0dB variance

Eb_N0_dB = [-3:10]; % multiple Eb/N0 values

for ii = 1:length(Eb_N0_dB)

% Noise addition

y = s + 10^(-Eb_N0_dB(ii)/20)*n; % additive white gaussian noise

% receiver - hard decision decoding

ipHat = real(y)>0;

% counting the errors

nErr(ii) = size(find([ip- ipHat]),2);

end

simBer = nErr/N; % simulated ber

theoryBer = 0.5*erfc(sqrt(10.^(Eb_N0_dB/10))); % theoretical ber

% plot

close all

figure

semilogy(Eb_N0_dB,theoryBer,'b.-');

 hold on

semilogy(Eb_N0_dB,simBer,'mx-');

axis([-3 10 10^-5 0.5])

grid on

legend('theory', 'simulation'); xlabel('Eb/No, dB'); ylabel('Bit Error Rate');

title('Bit error probability curve for BPSK modulation');
```

Observations





Conclusion:

Experiment No.–5

Title: Configure a Cisco Router as a DHCP Server

Problem Statement:

Configuring a Cisco Router as a DHCP Server.

Objectives:

- What is a DORA PROCESS?
- Benefits of DHCP
- How it works?

Outcome:

Deploy Client-Server architectures and prototypes by the means of correct standards and technology to assign IP address.

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.

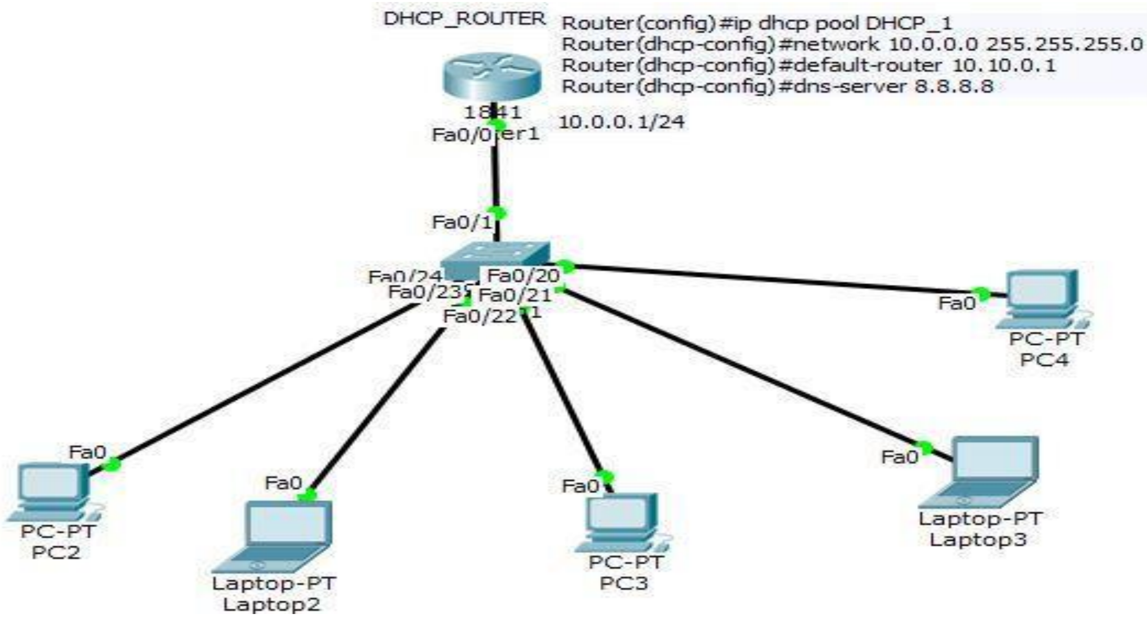
To configure an address pool, you must:

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.

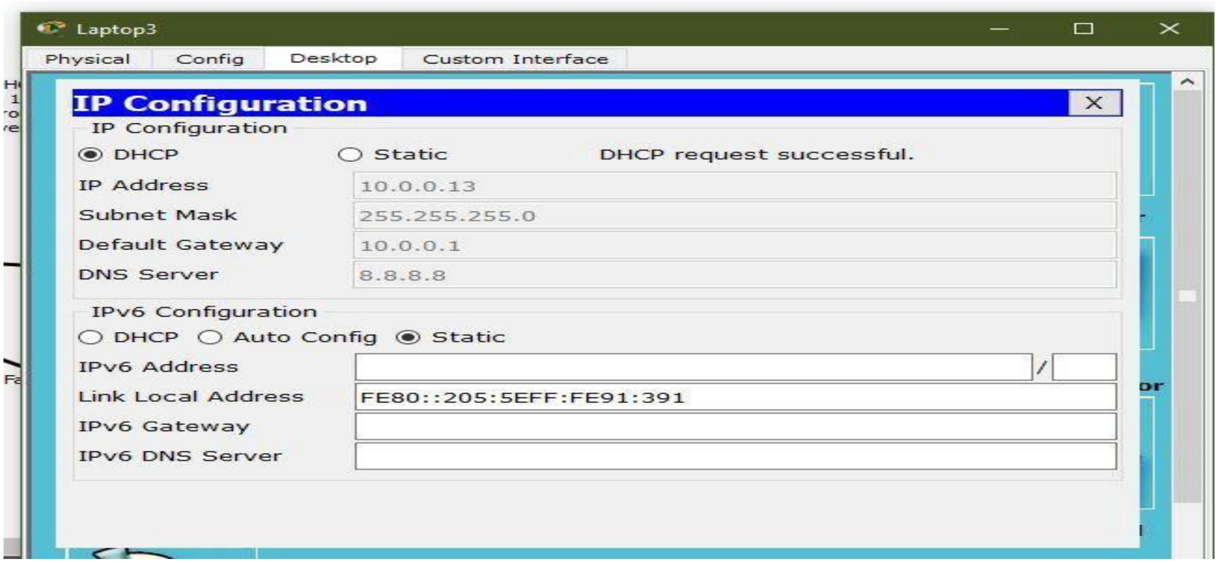
5. Set the options (standard and user-defined) that the clients of a pool require when configuring their IP details.

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.

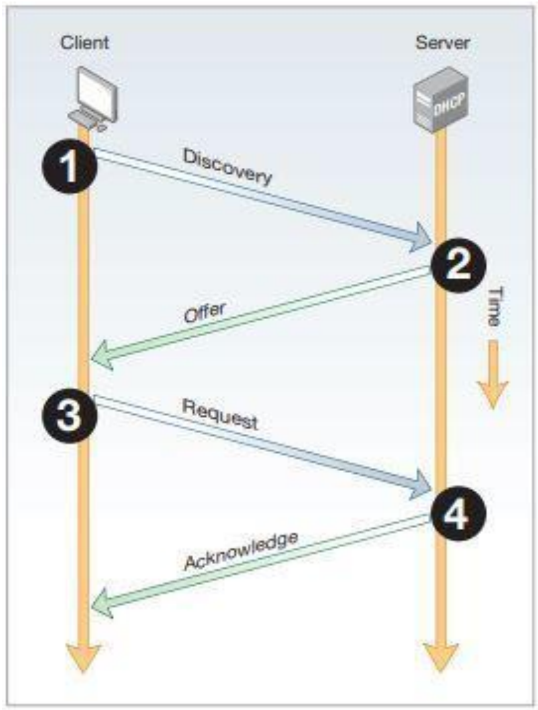
The DHCP process

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:

Frequently asked questions: -

1] Explain DORA Process?

2] What are benefits of DHCP Process?

3] Why DORA works on UDP...?

Experiment No.–6

Aim: To understand the handover mechanism.

Objectives:

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

Prerequisite:

Operating System: Windows 7

Java Version: 6 only

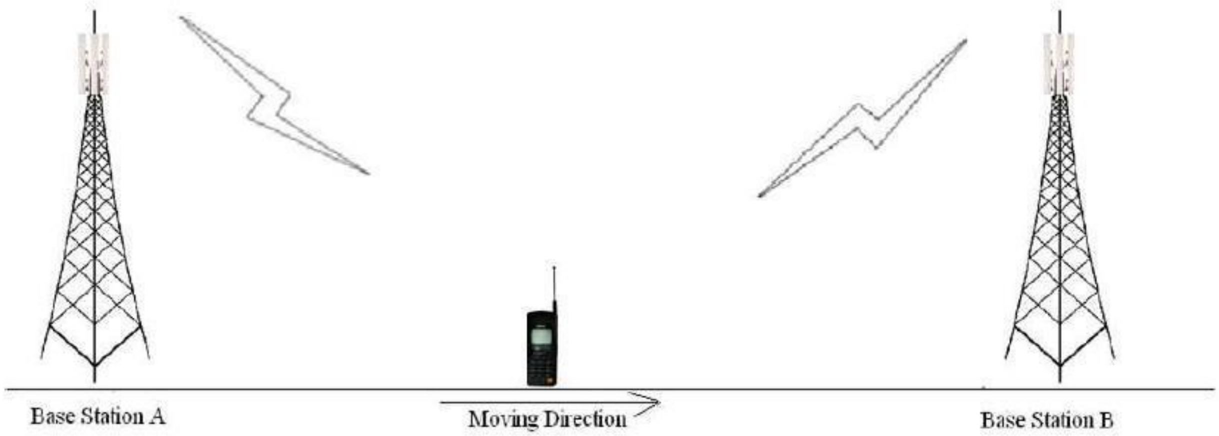
Mozilla Firefox: version: 47.0.1

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 and hence receives signal strength from A PArx>PBrx . 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 P0rx0 for the mobile, i.e. if the signal from the B.S.to which the mobile is connected falls below P0rx0 then the call drops. In order to prevent call drop the mobile monitors receive signal strength from the neighboring 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 PTrx (T= target B.S.) > Phrx$h$ higher h/o threshold and Pcrx (c=current B.S.) < Phrx$h$ 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.

Δγ=Phrx−PlrxΔ=h-

A successful handoff is one where the call gets from and continuous without call or in other words the h occurs before h/o Pcrx becomes <P0rx<0. If Pcrx<P0rx0 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 PTrx and Pcrx also plays a role in the process. In the experiment small scale fading is not considered and hence the averaging considered only shadowing.

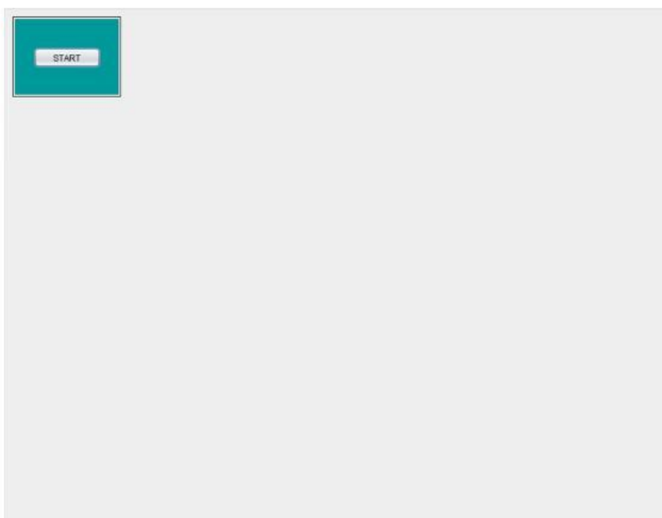
Students conducting the experiment is expected to study the impact of these on h/0. He/She is encouraged to respect the experiment for several sets of values of these parameters these draw conclusion.

Instruction

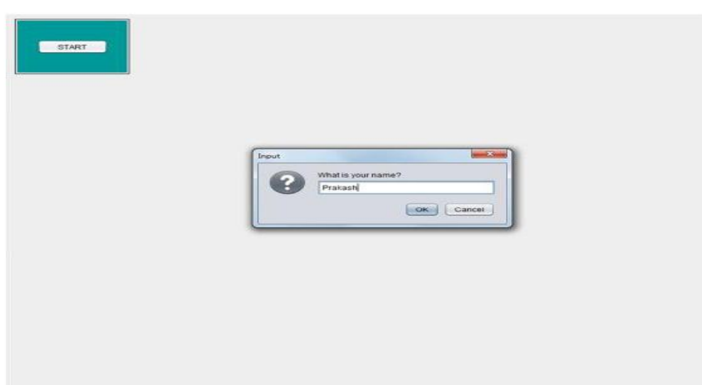
Follow the instructions given below to perform the experiments.

1.1 Starting the Experiments: -

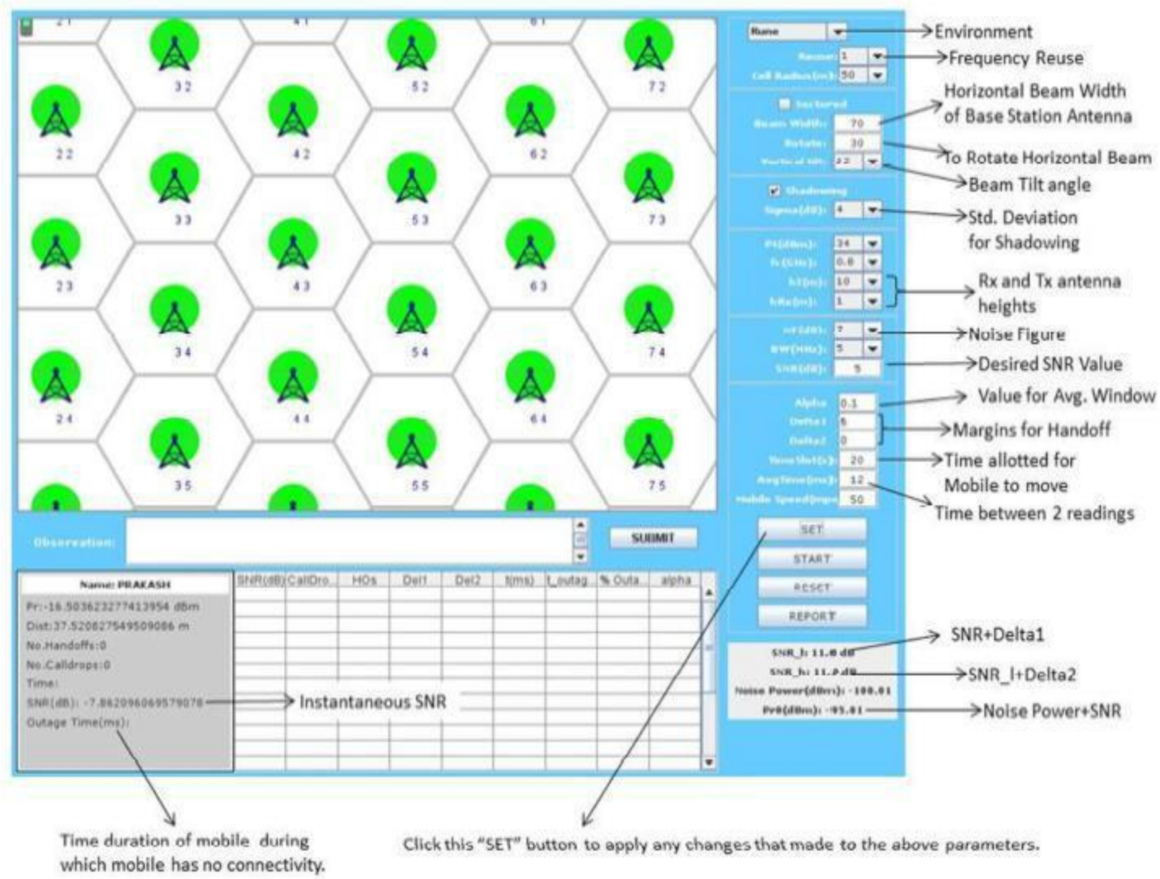
•. Step1: Click on START button to start experiment.



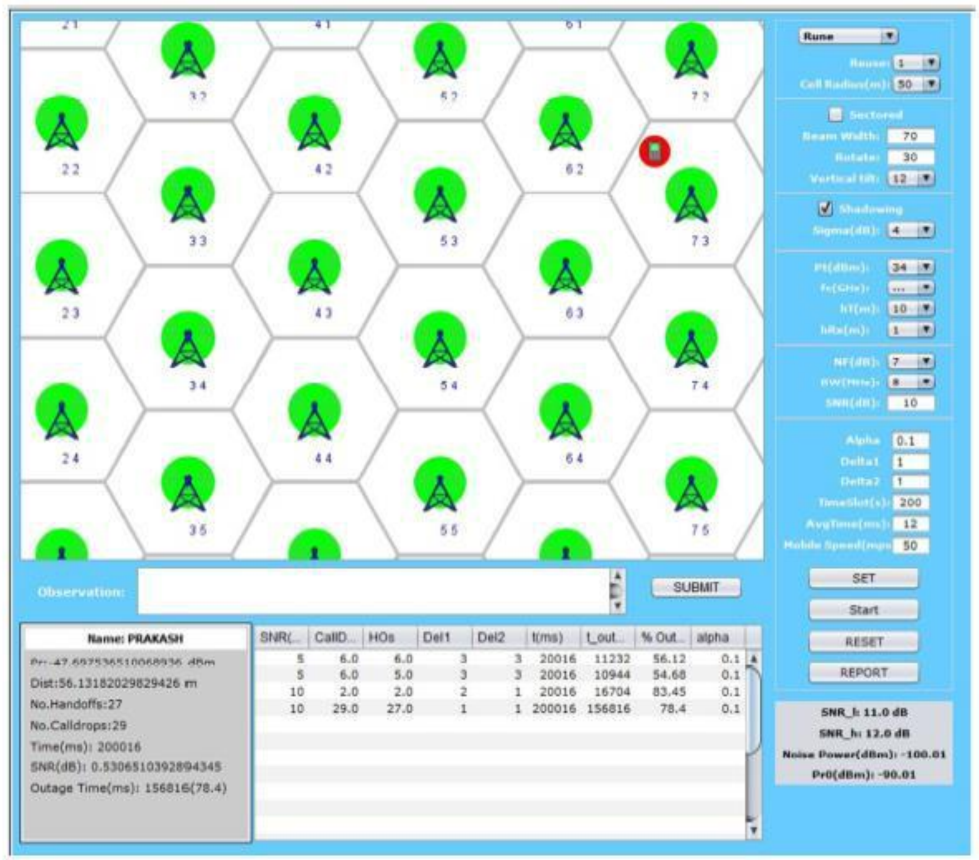
•Step2: Enter your name then click OK button.



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

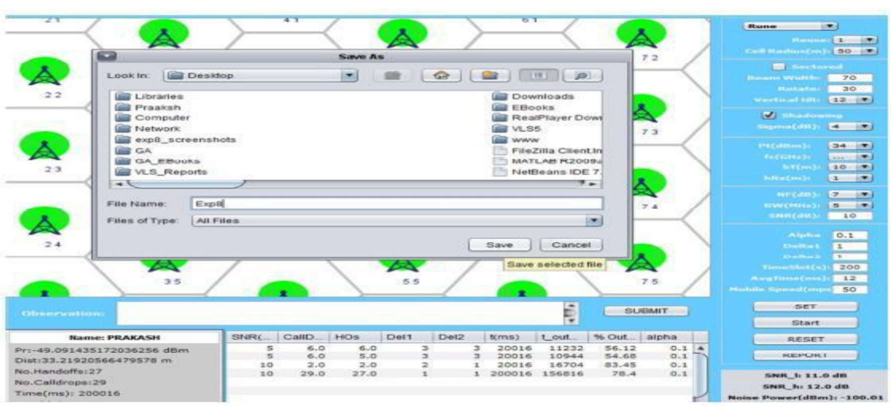


- Step4: Click on START button and observe No. of Call Drops and No. of Handoffs.



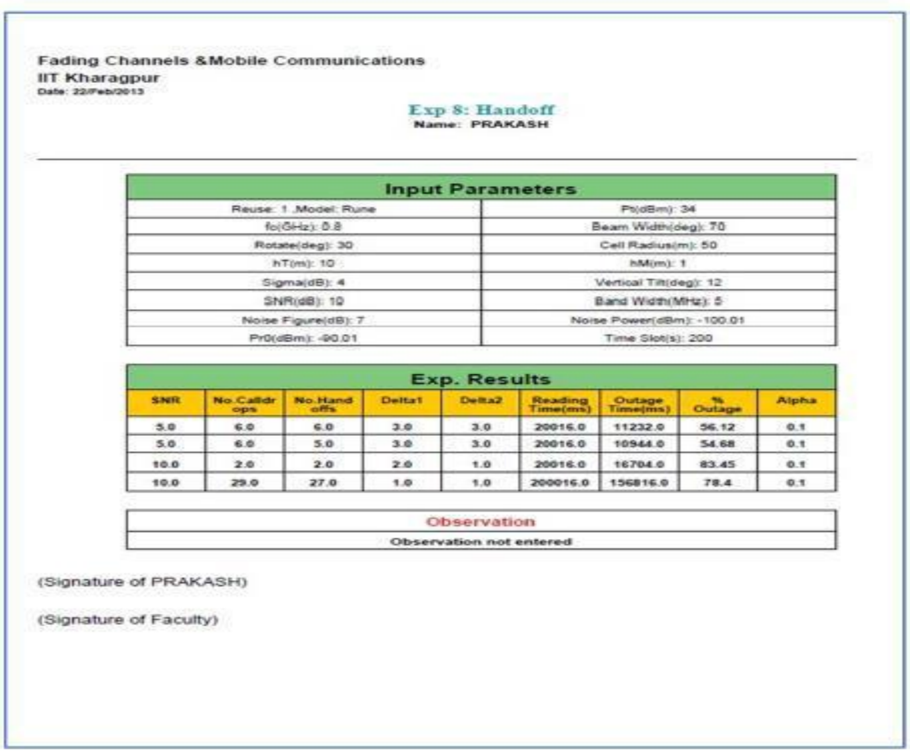
- Step5: Enter your observation in the OBSERVATION box and Click on SUBMIT button.

- 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.Calldr ops	No.Hand offs	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			

FAQ:

1. What is handoff?
2. What is the condition for handoff?
3. Explain Handoff and its types.

Experiment No.–7

Aim: 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 fm=υ(m/s)c=

Where,

•c=velocitylight=3∗108m/s.=h=3∗10*8

•fc=carrierequency.

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

Tc= 9/16Πfm (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|

LR=√2π(k+1)fmρe−k−(k+1)ρ2I0(2ρ√k(k+1))ρ=R√Ωp=RRrms

Rrms=√Ωp is the enveloped level
Rayleigh fading (k=0) and isotropic scattering LR=√2πfmρe−ρ2 Level Crossing Rate For Selection Combining Lr=fm√πMγ√σexp(−γ22σ)[1−exp(−γ22σ)]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

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

t=1NRPr[r≤R]
Average fade duration For Selection Combining

ADF=√ρ∗exp(γ22σ−1)√2πfdMγ

For Rayleigh distribution fading

$$Pr[r \le R]=\int_0^R Pr(dr)=1-\exp(-\rho^2)$$

$$\bar{t}=\frac{e^{\rho^2}-1}{\rho f_m\sqrt{2\pi}}$$

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

$$p(r)=\frac{r}{\sigma^2}\exp\left(-\frac{r^2}{2\rho^2}\right)(0\le r\le\infty)$$

$$=0 (r<0)$$

Where,

• σ is the r.m.s value of the received voltage signal before detection.

• σ^2 is the time average power of the received signal before enveloped detection.

Probability of outage is defined as

$$P(R)=Pr(r\le R)=\int_0^R p(r)dr=1-\exp\left(-\frac{R^2}{2\sigma^2}\right)$$

The mean value rmean of rayleigh distribution is given by

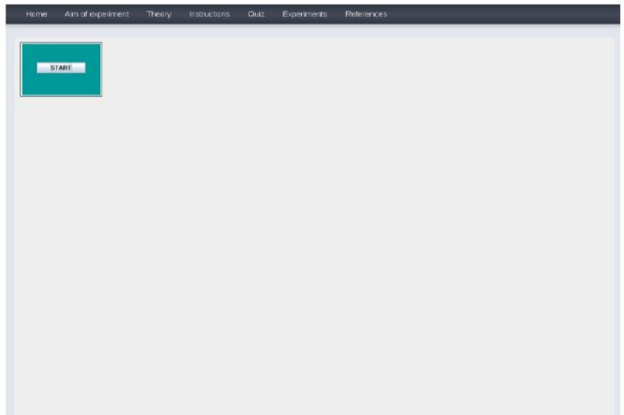
$$r_{mean}=E[r]=\int_0^\infty r p(r)dr=\sigma\sqrt{\frac{\pi}{2}}=1.2533\sigma=1.2533$$

$$\sigma_r^2=E[r^2]-E^2[r]=\int_0^\infty r^2 p(r)dr-\frac{\sigma^2\pi}{2}$$

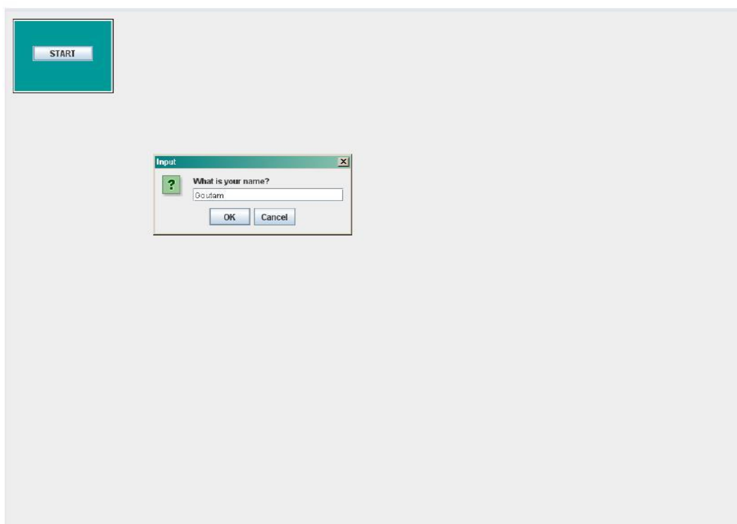
$$=\sigma^2\left(2-\frac{\pi}{2}\right)=0.4292\sigma^2$$

Instructions for Experiment: - Flat Fading

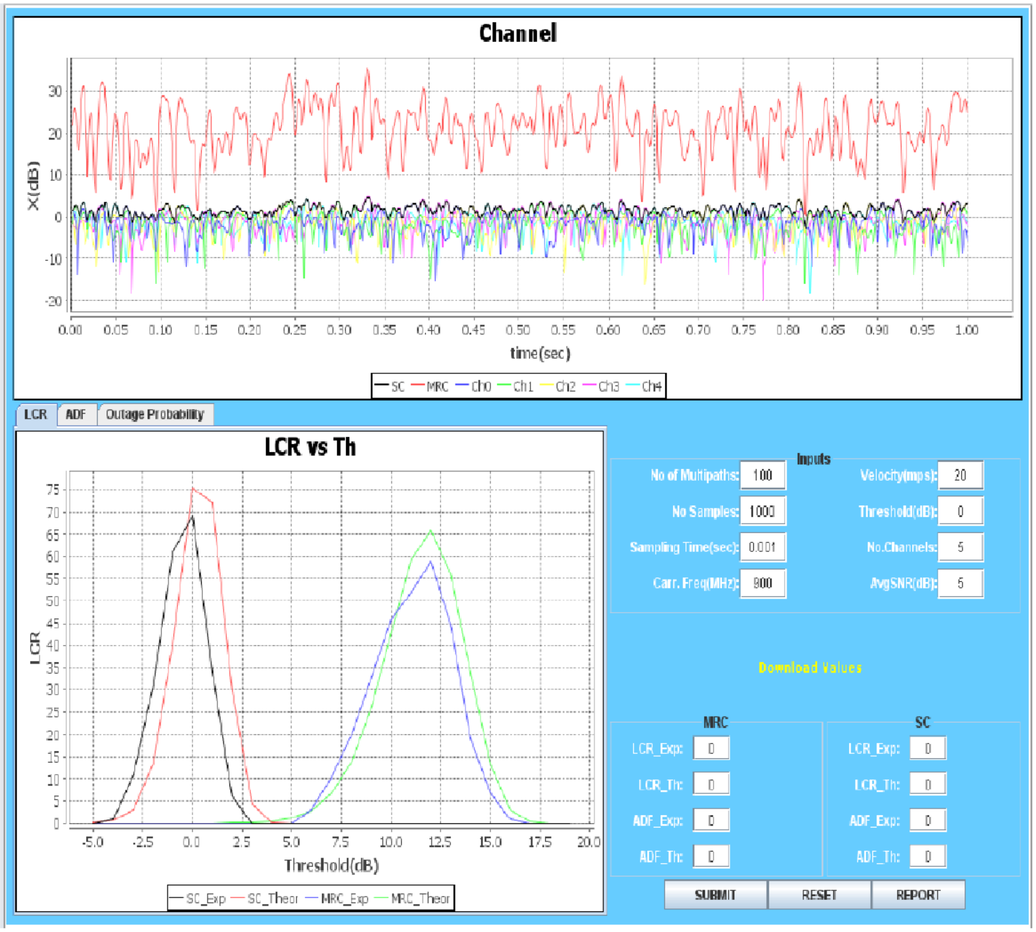
• Step1:- Click on the button START. A page appears with a dialogue box asking for your name.

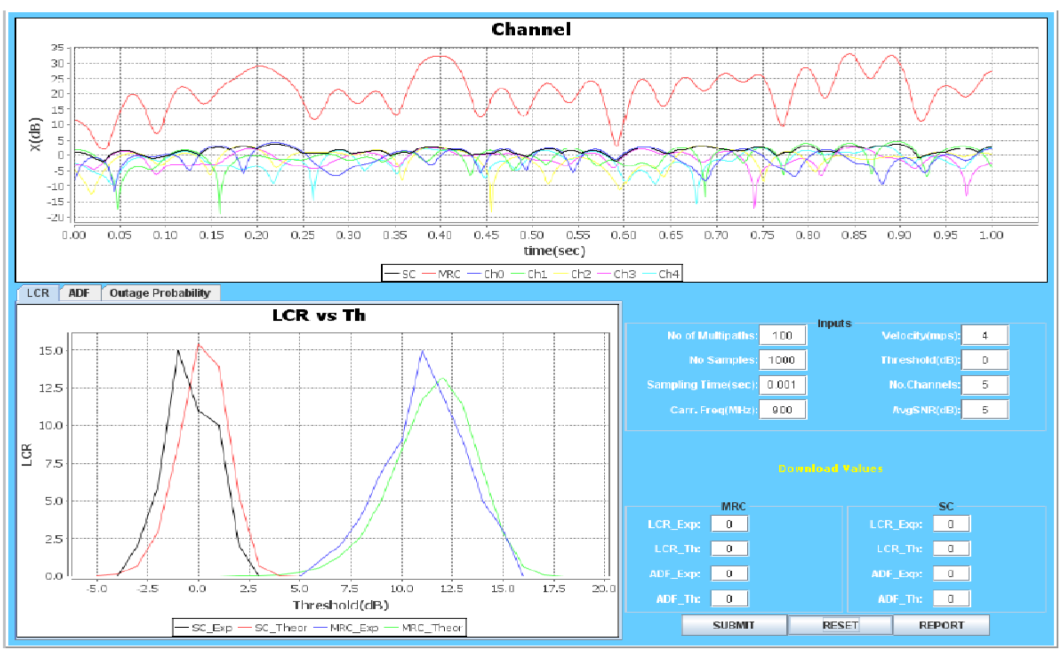


•Step 2:- Enter your name then Click Ok.

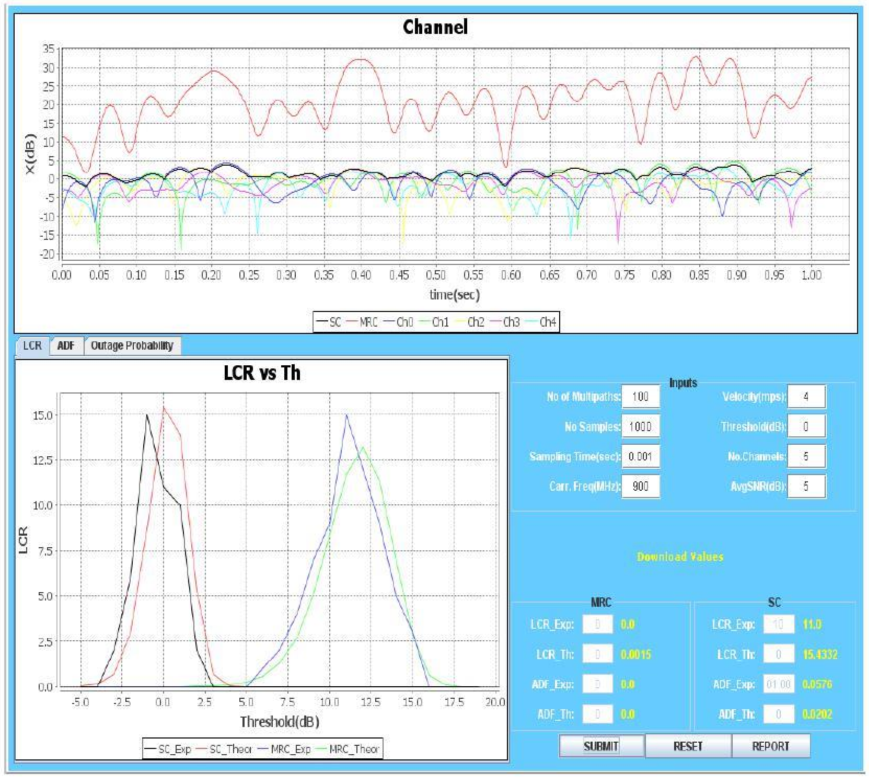


Step3: - Enter the input parameters value. Then click on "RESET" Button. Observed the waveform.

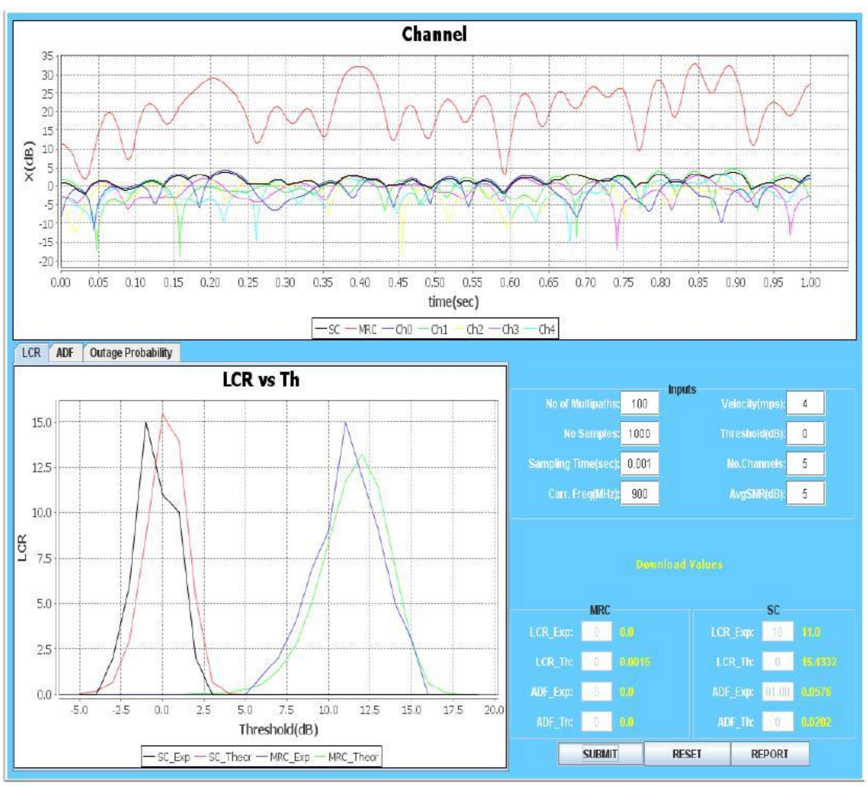




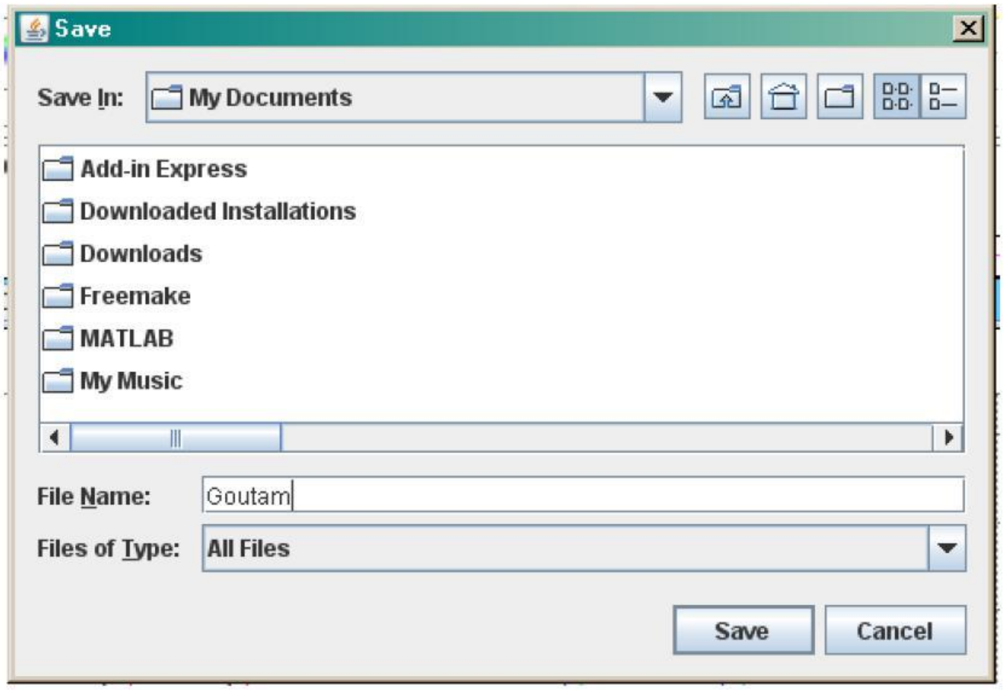
4. Step4: - Enter value of LCR Exp and ADF Exp in both MRC and SC from the waveform. Then Click on "SUBMIT" Button.



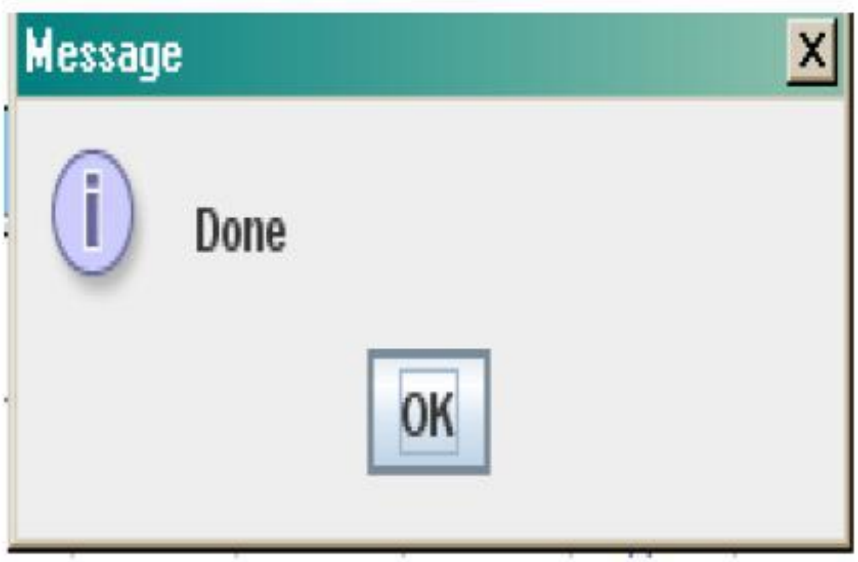
•Step5:- Click on the "Report" button.



•Step6:- PDF report of the experiment is generated.



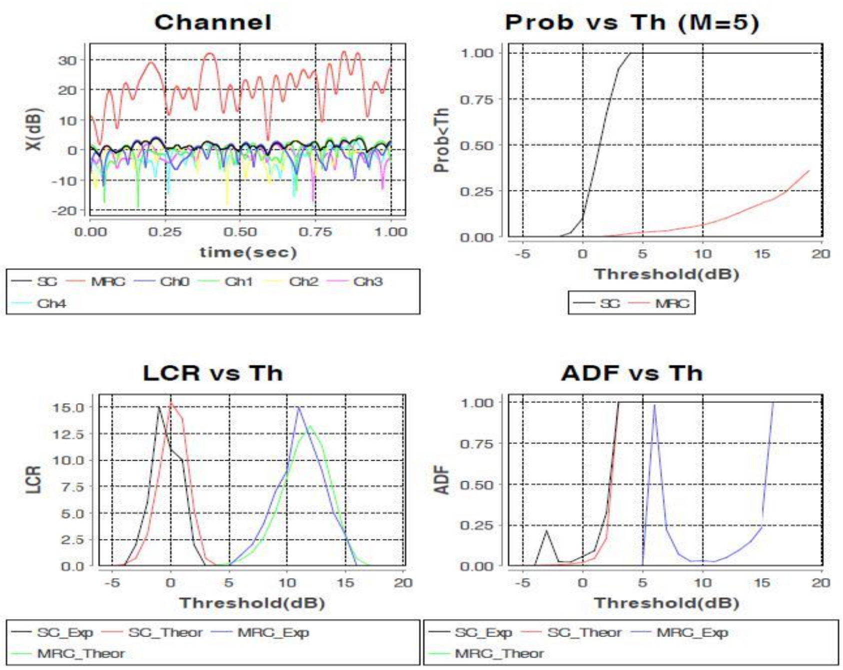
•Step7:-After generation of the Report you will get following message.



•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:

FAQ;

1. What is Flat Fading?

2. What is the difference between flat and selective fading?

3. What are the types of fading?

Experiment No.–8

Title: File sharing by using TCP Protocol

Problem Statement:

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

Objectives:

- What is a socket?
- The client-server model.
- Remote Communication

Outcome:

Develop Client-Server architectures and prototypes by the means of correct standards and technology.

Software Requirements:

Python, Open-source Linux operating system.

THEORY:

The basics

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

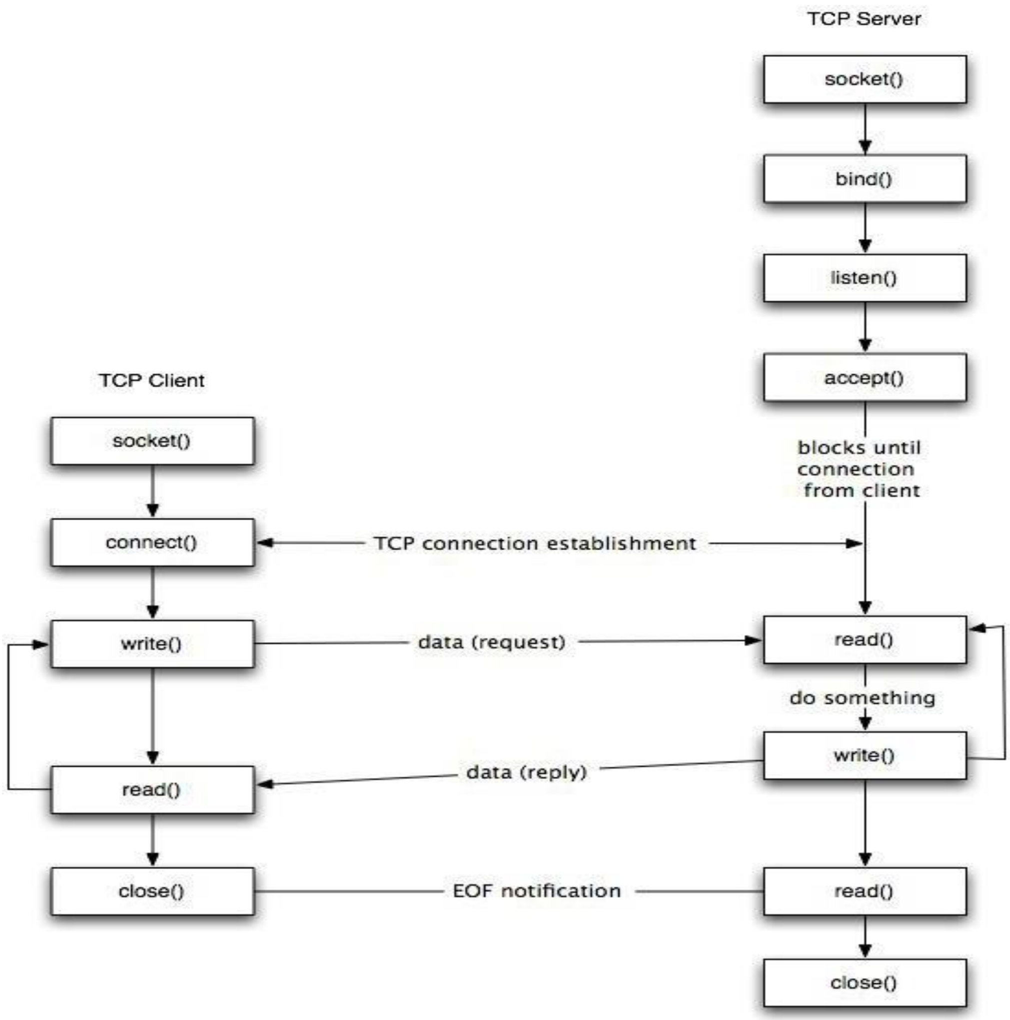


Figure: TCP client-server.

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.

Conclusion:

Experiment No.–9

Title: Case Study on different real time mobile computing services.

<u>Attach different real time mobile computing services as case study report.</u>