DPU

**Dr. D. Y. Patil Institute of Technology
Pimpri Pune-411018**

Department of Artificial Intelligence and Data Science

Laboratory Manual

**Savitribai Phule Pune University
Third Year of Artificial Intelligence and Data Science (2019 Course)**

**Subject Code: 317524
Computer Networks Laboratory**

**Prepared by
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Academic Year
2024-2025

DPU

Dr. D. Y. Patil Institute of Technology
Pimpri Pune-411018

Department of Artificial Intelligence and Data Science

Vision & Mission of the Institute

Vision of the Institute
- Empowerment through knowledge

Mission of the Institute

- Developing human potential to serve the Nation
- Dedicated efforts for quality education.
- Yearning to promote research and development.
- Persistent endeavor to imbibe moral and professional ethics.
- Inculcating the concept of emotional intelligence.
- Emphasizing extension work to reach out to the society.
- Treading the path to meet the future challenges.

Dr. D. Y. Patil Institute of Technology
Pimpri Pune-411018

Department of Artificial Intelligence and Data Science

Vision & Mission of the Department

Vision of the Department

- To produce globally competent engineers in the field of Artificial Intelligence and Data Science with human values

Mission of the Department

- To develop students with a sound understanding in the area of Artificial Intelligence, Machine Learning and Data Science.
- To enable students to become innovators, researchers, entrepreneurs and leaders globally.
- Equip the department with new advancement in high performance equipments and software to carrying out research in emerging technologies in AI and DS.
- To meet the pressing demands of the nation in the areas of Artificial Intelligence and Data Science.

Computer Networks Laboratory
317524

Teaching Scheme	Credit Scheme	Examination Scheme and Marks
Practical: 02 Hours/Week	01	Term Work: 25 Marks Practical: 25 Marks

Course Objectives:
1. To learn computer network hardware and software components
2. To learn computer network topologies and types of network
3. To develop an understanding of various protocols, modern technologies and applications
4. To learn modern tools for network traffic analysis
5. To learn network programming

Course Outcomes:
On completion of the course, learner will be able to–
CO1: Analyze the requirements of network types, topology and transmission media
CO2: Demonstrate error control, flow control techniques and protocols and analyze them
CO3: Demonstrate the subnet formation with IP allocation mechanism and apply various routing
algorithms
CO4: Develop Client-Server architectures and prototypes
CO5: Implement web applications and services using application layer protocols

Practical No.	Assignment to be covered
4 Assignments from Group A	
1	Demonstrate the different types of topologies and types of transmission media by using a packet tracer tool.
2	Setup a WAN which contains wired as well as wireless LAN by using a packet tracer tool. Demonstrate transfer of a packet from LAN 1 (wired LAN) to LAN2 (Wireless LAN).
3	Use packet Tracer tool for configuration of 3 router networks using one of the following protocols RIP/OSPF/BGP.
4	Write a program to demonstrate Sub-netting and find subnet masks.
6 Assignments from Group B	
5	Socket Programming using C/C++/Java. a. TCP Client, TCP Server b. UDP Client, UDP Server
6	Write a program using TCP socket for wired network for following a. Say Hello to Each other b. File transfer
7	Write a program using UDP Sockets to enable file transfer (Script, Text, Audio and Video one file each) between two machines.
8	10. Capture packets using Wireshark and accomplish the following and save the output in file: a. Capture all TCP traffic to/from Facebook, during the time when you log in to your Facebook account b. Capture all HTTP traffic to/from Facebook (other website), when you log in to your Facebook account c. Write a DISPLAY filter expression to count all TCP packets (captured under item #1) that have the flags SYN, PSH, and RST set. Show the fraction of packets that had each flag set. d. Count how many TCP packets you received from / sent to Facebook (other website), and how many of each were also HTTP packets.
9	Study and Analyze the performance of HTTP, HTTPS and FTP protocol using Packet tracer tool.
10	To study the SSL protocol by capturing the packets using Wireshark tool while visiting any SSL secured website (banking, e-commerce etc.).
2 Assignments from Group C	
11	Installing and configuring DHCP server and assign IP addresses to client machines using DHCP server.
12	Write a program for DNS lookup. Given an IP address input, it should return URL and vice versa.

GROUP A

Assignment No.: 1

PROBLEM STATEMENT: Demonstrate the different types of topologies and types of transmission media by using a packet tracer tool.

OBJECTIVES: 1. To learn different types of topologies and transmission media

2. To learn packet tracer tool

OUTCOME: Students will be able to,

1. get familiar with packet tracer tool
2. implement different topologies using packet tracer tool
3. understand different transmission media

THEORY-CONCEPT:

Network topology is the geometric representation of relationship of all the links connecting the devices or nodes. Network topology represent in two ways one is physical topology that define the way in which a network is physically laid out and other one is logical topology that defines how data actually flow through the network.

Cisco Packet Tracer (CPT) is multi-tasking network simulation software to perform and analyze various network activities such as implementation of different topologies, select optimum path based on various routing algorithms, create DNS and DHCP server, sub netting, analyze various network configuration and troubleshooting commands. In order to start communication between end user devices and to design a network, we need to select appropriate networking devices like routers, switches, hubs and make physical Connection by connection cables to serial and fast Ethernet ports from the component list of packet tracer. Networking devices are costly so it is better to perform first on packet tracer to understand the concept and behavior of networking.

DESIGNING OF TOPOLOGY

1. Bus Topology In local area network, it is a single network cable runs in the building or campus and all nodes are connected along with this communication line with two endpoints called the bus or backbone. In other words, it is a multipoint data communication circuit that is easily control data flow between the computers because this configuration allows all stations to

receive every transmission over the network. For bus topology we build network using three generic pc which are serially connected with three switches using copper straight through cable and switches are interconnected using copper cross over cable shown in fig1.
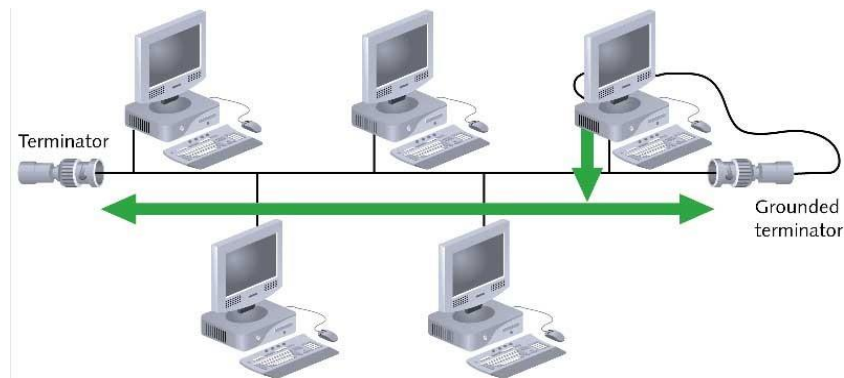

Fig 1. Bus topology

2. Star Topology, all the cables run from the computers to a central location where they are all connected by a device called a hub. It is a concentrated network, where the end points are directly reachable from a central location when network is expanded. Ethernet 10 base T is a popular network based on the star topology. For star topology we build network using five generic pc which are centrally connected to single switch 2950-24 using copper straight through cable shown in fig2.
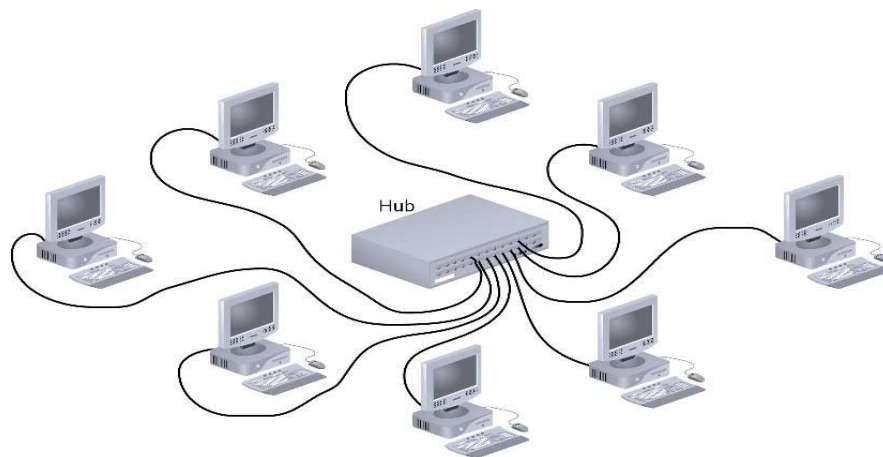

Fig 2. Star Topology

3. Mesh Topology, every device has a dedicated point to point link to every other device. The term dedicated stand for link carries traffic only between two devices it connects. It is a well-

connected topology; As shown in fig3., in this every node has a connection to every other node in the network. The cable requirements are high and it can include multiple topologies. Failure in one of the computers does not cause the network to break down, as they have alternative paths to other computers star topology, all the cables run from the computers to a central location.
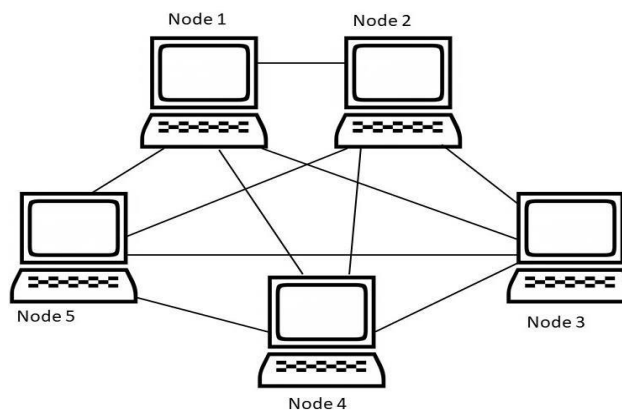

Fig 3. Mesh Topology

TRANSMISSION MEDIA

o   Transmission media is a communication channel that carries the information from the sender to the receiver. Data is transmitted through the electromagnetic signals. The main functionality of the transmission media is to carry the information in the form of bits through LAN (Local Area Network). It is a physical path between transmitter and receiver in data communication. In a copper-based network, the bits in the form of electrical signals. In a fiber-based network, the bits in the form of light pulses. In OSI(Open System Interconnection) phase, transmission media supports the Layer 1. Therefore, it is considered to be as a Layer 1 component. The electrical signals can be sent through the copper wire, fiber optics, atmosphere, water, and vacuum. The characteristics and quality of data transmission are determined by the characteristics of medium and signal. Transmission media is of two types are wired media and wireless media. In wired media, medium characteristics are more important whereas, in wireless media, signal characteristics are more important. Different transmission media have different properties such as bandwidth, delay, cost and ease of installation and maintenance. The transmission media is available in the lowest layer of the OSI reference model, i.e., Physical layer. Transmission Media is broadly classified into the following types:
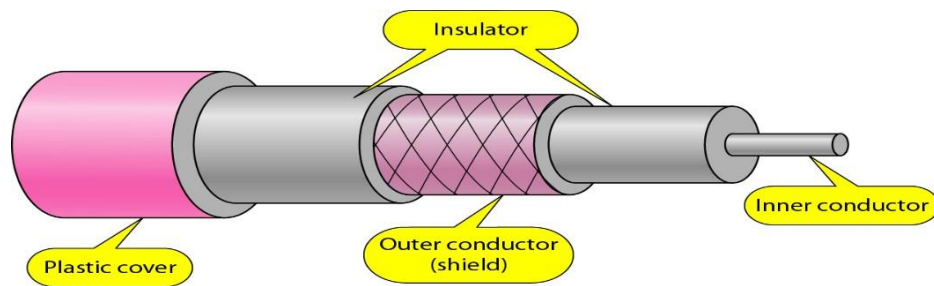
1. Guided
2. Unguided

1. Guided Transmission media: Transmission capacity depends on the distance and on whether the medium is point-to-point or multipoint. There are three types of guided transmission media.
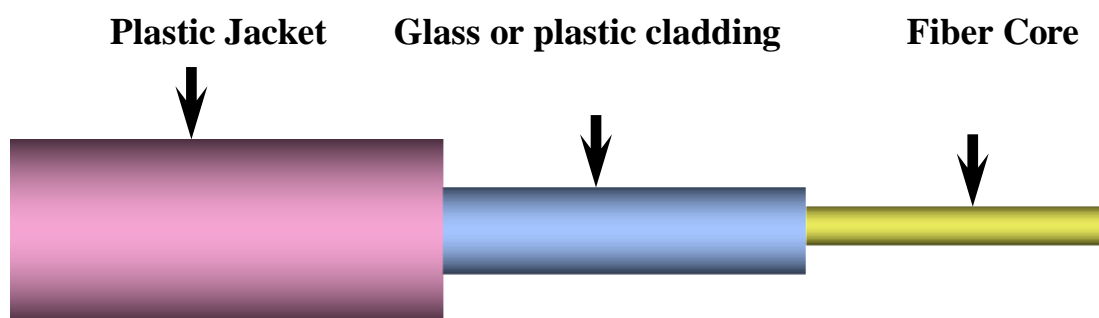
    a) Twisted pair wires
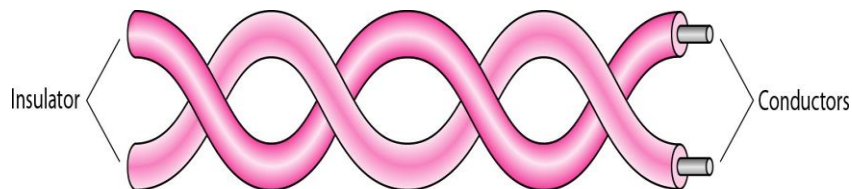
    b) Coaxial cables

    c) Optical fiber

2. Unguided transmission Media: It is also referred to as Wireless or Unbounded transmission media. No physical medium is required for the transmission of electromagnetic signals.

Features:

- The signal is broadcasted through air
- Less Secure
- Used for larger distances

There are 3 types of Signals transmitted through unguided media:

Radio waves – These are easy to generate and can penetrate through buildings. The sending and receiving antennas need not be aligned. Frequency Range:3KHz – 1GHz. AM and FM radios and cordless phones use Radio waves for transmission.

Microwaves – It is a line-of-sight transmission i.e., the sending and receiving antennas need to be properly aligned with each other. The distance covered by the signal is directly proportional to the height of the antenna. Frequency Range:1GHz – 300GHz. These are majorly used for mobile phone communication and television distribution.

Infrared – Infrared waves are used for very short distance communication. They cannot penetrate through obstacles. This prevents interference between systems. Frequency Range:300GHz – 400THz. It is used in TV remotes, wireless mouse, keyboard, printer, etc.

CONCLUSION: We have learned how to implement different topologies using different transmission media in packet tracer tool.

FREQUENTLY ASKED QUESTIONS:

1. What is hub & switch?

2. What is star topology?

3. Explain types of transmission media.

Assignment No: 2

PROBLEM STATEMENT: Setup a WAN which contains wired as well as wireless LAN by using a packet tracer tool. Demonstrate transfer of a packet from LAN 1 (wired LAN) to LAN2 (Wireless LAN).

OBJECTIVES: 1. To configure wired LAN & wireless LAN.

2. To configure access point.

OUTCOME: Students will be able to,

4. Configure wired LAN & wireless LAN.

5. Configure access point.

THEORY-CONCEPT:

Wired LAN (Local Area Network):

A wired LAN refers to a network of devices (computers, printers, servers, etc.) within a limited geographical area, such as a home, office, or building, that are connected using physical cables. Ethernet cables, typically Cat 5e or Cat 6, are commonly used to establish wired connections. Wired LANs offer high-speed, reliable, and secure data transmission, making them suitable for situations where consistent and fast network connectivity is required. Wired LANs often use devices like switches and routers to manage the network traffic.

Wireless LAN (Local Area Network):

A wireless LAN, commonly known as Wi-Fi, is a network that connects devices without the need for physical cables. Instead, it uses radio waves to transmit data between devices and access points. Wireless LANs provide mobility and flexibility since devices can connect to the network from various locations within the coverage area. Wi-Fi is prevalent in homes, offices, cafes, airports, and other public places. While wireless networks offer convenience, they can be susceptible to interference and have potential security concerns if not properly secured.

WAN (Wide Area Network):

A WAN is a network that covers a larger geographical area, often spanning multiple cities, countries, or even continents. It connects multiple LANs and other network devices over long distances. The internet itself is a massive example of a WAN. WANs use various technologies, including leased lines, fiber optics, satellite links, and more, to establish long-distance connections. WANs are typically slower than LANs due to the extended distances involved, and they can be more complex to manage. They play a crucial role in connecting remote offices, data centers, and providing access to cloud services.

CONCLUSION: We have learned how to setup WAN which contains wired as well as wireless LAN by using a packet tracer tool.

FREQUENTLY ASKED QUESTIONS:

1. What is LAN & WAN?

2. What is Access Point?

3. Explain Wireless LAN.

Assignment No: 3

PROBLEM STATEMENT: Use packet Tracer tool for configuration of 3 router networks using one of the following protocols RIP/OSPF/BGP.

OBJECTIVES: 1. To configure router.

2. To understand routing protocols.

OUTCOME: Students will be able to,

4. Configure router networks using Packet Tracer.

5. Understand routing protocols.

THEORY-CONCEPT:

Routing Information Protocol (RIP):

The Routing Information Protocol (RIP) is one of the oldest distance-vector routing protocols which employs the hop count as a routing metric. RIP prevents routing loops by implementing a limit on the number of hops allowed in a path from source to destination. The largest number of hops allowed for RIP is 15, which limits the size of networks that RIP can support.

Step 1: First, open the Cisco packet tracer desktop and select the devices given below:

S.NO	Device	Model Name	Qty.
1.	PC	PC	6
2.	Switch	PT-Switch	3
3.	Router	PT-router	3

IP Addressing Table:

S.NO	Device	IPv4 Address	Subnet mask	Default Gateway
1.	PC0	192.168.10.2	255.255.255.0	192.168.10.1
2.	PC1	192.168.10.3	255.255.255.0	192.168.10.1
3.	PC2	192.168.20.2	255.255.255.0	192.168.20.1
4.	PC3	192.168.20.3	255.255.255.0	192.168.20.1
5.	PC4	192.168.30.2	255.255.255.0	192.168.30.1
6.	PC5	192.168.30.3	255.255.255.0	192.168.30.1

- Then, create a network topology as shown below the image.
- Use an Automatic connecting cable to connect the devices with others.
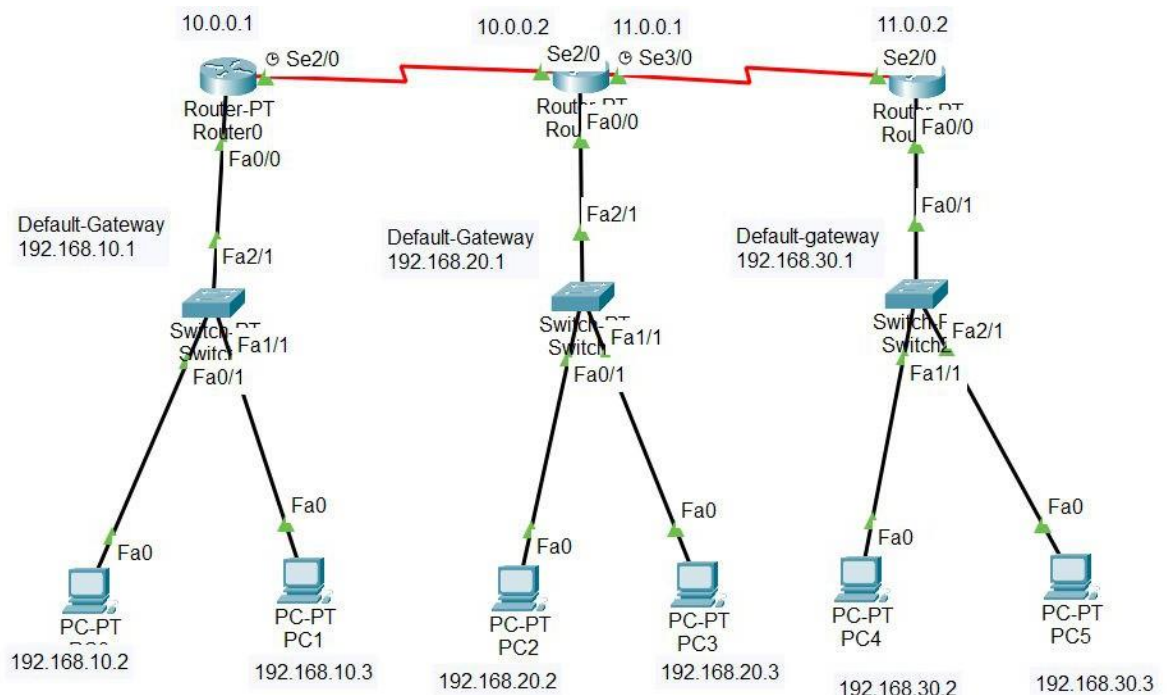
Step 2: Configure the PCs (hosts) with IPv4 address and Subnet Mask according to the IP addressing table given above.

- To assign an IP address in PC0, click on PC0.
- Then, go to desktop and then IP configuration and there you will IPv4 configuration.
- Fill IPv4 address and subnet mask.
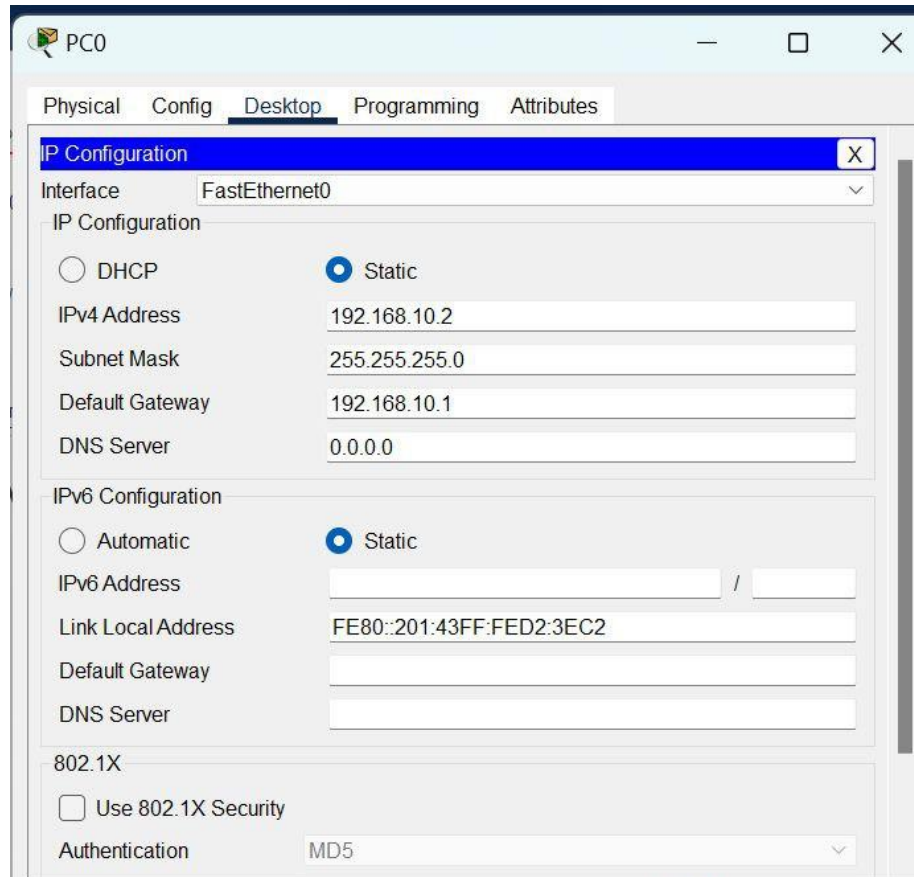
- Assigning an IP address using the ipconfig command, or we can also assign an IP address with the help of a command.
- Go to the command terminal of the PC.
- Then, type ipconfig <IPv4 address><subnet mask><default gateway>(if needed)
- Example: ipconfig 192.168.10.2 255.255.255.0 192.168.10.1
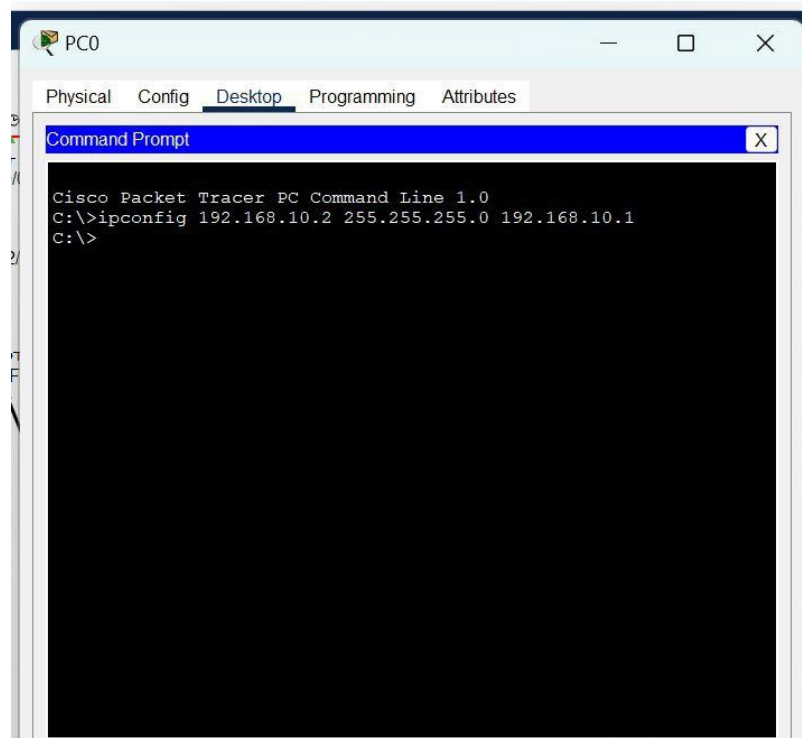
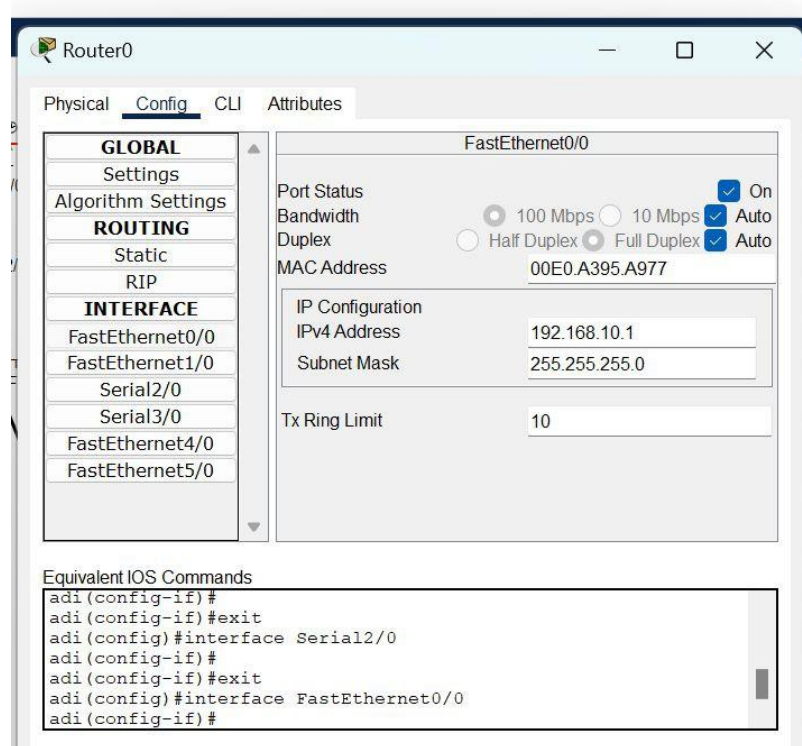- Repeat the same procedure with other PCs to configure them thoroughly.

Step 3: Configure router with IP address and Subnet mask.

IP Addressing Table Router:

S.NO	Device	Interface	IPv4 Address	Subnet mask
1.	router0	FastEthernet0/0	192.168.10.1	255.255.255.0
		Serial2/0	10.0.0.1	255.0.0.0
2.	router1	FastEthernet0/0	192.168.20.1	255.255.255.0
		Serial2/0	10.0.0.2	255.0.0.0
		Serial3/0	11.0.0.1	255.0.0.0
3.	router2	FastEthernet0/0	192.168.30.1	255.255.255.0
		Serial2/0	11.0.0.2	255.0.0.0

- To assign an IP address in router0, click on router0.
- Then, go to config and then Interfaces.

- Make sure to turn on the ports.
- Then, configure the IP address in FastEthernet and serial ports according to IP addressing Table.
- Fill IPv4 address and subnet mask.

- Repeat the same procedure with other routers to configure them thoroughly.

Step 4: After configuring all of the devices we need to assign the routes to the routers.

To assign RIP routes to the particular router:

- First, click on router0 then Go to CLI.
- Then type the commands and IP information given below.

CLI command : network <network id>

- RIP Routes for Router0 are given below:

Router(config)#network 192.168.10.0
Router(config)#network 10.0.0.0

- RIP Routes for Router1 are given below:

Router(config)#network 192.168.20.0
Router(config)#network 10.0.0.0
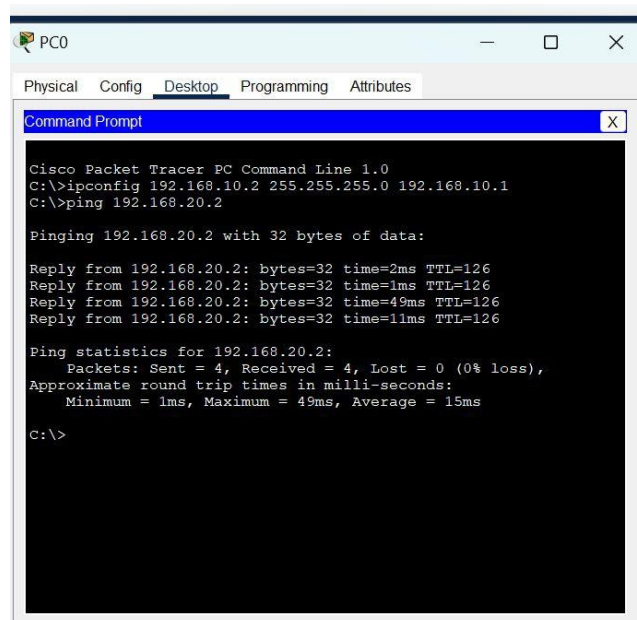Router(config)#network 11.0.0.0

- RIP Routes for Router2 are given below:

Router(config)#network 192.168.30.0
Router(config)#network 11.0.0.0

Step 5: Verifying the network by pinging the IP address of any PC.

- We will use the ping command to do so.
- First, click on PC0 then Go to the command prompt.
- Then type ping <IP address of targeted node>.
- As we can see in the below image we are getting replies which means the connection is working properly.

Example: ping 192.168.20.2

CONCLUSION: We have learned to configure 3 router networks using RIP in Cisco Packet Tracer tool.

FREQUENTLY ASKED QUESTIONS:

1. What is router?

2. What is IP address & default gateway

3. Explain RIP?

Assignment No: 4

PROBLEM STATEMENT: Write a program to demonstrate Sub-netting and find subnet masks.

OBJECTIVES: 1. To learn network address classes

4. To understand the concept of subnetting

OUTCOME: Students will be able to

1. memorize network address classes

2. implement subnetting and subnet masks

THEORY-CONCEPT:

SUBNETTING

Subnetting is a process of dividing large network into the smaller networks based on layer 3 IP address. Every computer on network has an IP address that represent its location on network. Two version of IP addresses are available IPv4 and IPv6. In this article we will perform subnetting on IPv4.

IPv4

IP addresses are displayed in dotted decimal notation, and appear as four numbers separated by dots. Each number of an IP address is made from eight individual bits known as octet. Each octet can create number value from 0 to 255. An IP address would be 32 bits long in binary divided into the two components, network component and host component. Network component is used to identify the network that the packet is intended for, and host component is used to identify the individual host on network. IP addresses are broken into the two components:

Network component: - Defines network segment of device.

Host component: - Defines the specific device on a particular network segment

IP Classes in decimal notation

1. Class A addresses range from 1-126
2. Class B addresses range from 128-191
3. Class C addresses range from 192-223
4. Class D addresses range from 224-239
5. Class E addresses range from 240-254
● 0 [Zero] is reserved and represents all IP addresses.
● 127 is a reserved address and is used for testing, like a loop back on an interface.
● 255 is a reserved address and is used for broadcasting purposes

SUBNET MASK

Subnet mask is a 32 bits long address used to distinguish between network address and host address in IP address. Subnet mask is always used with IP address. Subnet mask has only one purpose, to identify which part of an IP address is network address and which part is hostaddress.

For example, how will we figure out network partition and host partition from IP address 192.168.1.10. Here we need subnet mask to get details about network address and host address.

● In decimal notation subnet mask value 1 to 255 represent network address and value 0 [Zero] represent host address.
● In binary notation subnet mask **ON** bit [1] represent network address while **OFF** bit [0]represent host address.

In decimal notation

    IP address   192.168.1.10

    Subnet mask   255.255.255.0

Network address is 192.168.1 and host address is 10.

In binary notation

    IP address   11000000.10101000.00000001.00001010

    Subnet mask   11111111.11111111.11111111.00000000

Network address is 11000000.10101000.00000001 and host address is 00001010

The following table shows IP class associated with default subnet

Table 1

IP Class	Default Subnet	Network bits	Host bits	Total hosts	Valid hosts
A	255.0.0.0	First 8 bits	Last 24 bits	16, 777, 216	16, 777, 214
B	255.255.0.0	First 16 bits	Last 16 bits	65,536	65,534
C	255.255.255.0	First 24 bits	Last 8 bits	256	254

Network ID

First address of subnet is called network ID. This address is used to identify one segment or broadcast domain from all the other segments in the network.

Block Size

Block size is the size of subnet including network address, hosts addresses and broadcast address.

Broadcast ID

There are two types of broadcast, direct broadcast and full broadcast.

Direct broadcast or local broadcast is the last address of subnet and can be heard by all hostsin subnet.

Full broadcast is the last address of IP classes and can be heard by all IP hosts in network. Full broadcast address is 255.255.255.255

The main difference between direct broadcast and full broadcast is that routers will not propagate local broadcasts between segments, but they will propagate directed broadcasts.

Host Addresses

All address between the network address and the directed broadcast address is called host address for the subnet. You can assign host addresses to any IP devices such as PCs, servers, routers, and switches.

Single class C IP range can fulfill this requirement, still you have to purchase 2 class C IP range, one for each network. Single class C range provides 256 total addresses and we need only 30 addresses, this will waste 226 addresses. These unused addresses would make additional route advertisements slowing down the network.

With subnetting you only need to purchase single range of class C. You can configure router to take first 26 bits instead of default 24 bits as network bits. In this case we would extend default boundary of subnet mask and borrow 2 host bits to create networks. By taking two bits from the host range and counting them as network bits, we can create two new subnets, and assign hosts them. As long as the two new network bits match in the address, they belong to the same network. You can change either of the two bits, and you would be in a new subnet.

Default subnet mask

Class	Subnet Mask	Format
A	255.0.0.0	Network.Host.Host.Host
B	255.255.0.0	Network.Network.Host.Host
C	255.255.255.0	Network.Network.Network.Host

ADVANTAGE OF SUBNETTING

- Subnetting breaks large network in smaller networks and smaller networks are easier to manage.
- Subnetting reduces network traffic by removing collision and broadcast traffic, that overall improve performance.
- Subnetting allows you to apply network security polices at the interconnection between subnets.
- Subnetting allows you to save money by reducing requirement for IP range.

CONCLUSION: We have successfully implemented the subnetting and subnet mask program.

FREQUENTLY ASKED QUESTIONS:

1. What is subnetting and subnet mask?

2. What is network & host address?

3. Explain IP Classes.

Group B

Assignment No: 5

PROBLEM STATEMENT: Socket Programming using C/C++/Java.

           a. TCP Client, TCP Server

           b. UDP Client, UDP Server

OBJECTIVE: 1. To learn socket programming.

        2. To classify client server communication

        3. To learn TCP and UDP.

OUTCOME: Students will be able to

1. apply Socket programming for client server communication.

2. understand client server communication using TCP and UDP

THEORY-CONCEPT

USER DATAGRAM PROTOCOL (UDP)

UDP is a simple transport-layer protocol. The application writes a message to a UDP socket, which is then encapsulated in a UDP datagram, which is further encapsulated in an IP datagram, which is sent to the destination. There is no guarantee that a UDP will reach the destination, that the order of the datagrams will be preserved across the network or that datagrams arrive only once.

The problem of UDP is its lack of reliability: if a datagram reaches its final destination but the checksum detects an error, or if the datagram is dropped in the network, it is not automatically retransmitted. Each UDP datagram is characterized by a length. The length of a datagram is passed to the receiving application along with the data. No connection is established between the client and the server and, for this reason, we say that UDP provides a connection-less service

UDP SOCKET API

There are some fundamental differences between TCP and UDP sockets. UDP is a connection-less, unreliable, datagram protocol (TCP is instead connection-oriented, reliable and stream based). There are some instances when it makes to use UDP instead of TCP. Some popular applications built around UDP are DNS, NFS, SNMP and for example, some Skype services and streaming media.

Figure 1 shows the interaction between a UDP client and server. First of all, the client does not establish a connection with the server. Instead, the client just sends a datagram to the server using the sendto function which requires the address of the destination as a parameter. Similarly, the server does not accept a connection from a client. Instead, the server just calls the recvfrom function, which waits until data arrives from some client. recvfrom returns the IP address of the client, along with the datagram, so the server can send a response to the client.

As shown in the Figure 1, the steps of establishing a UDP socket communication on the client side are as follows:

- Create a socket using the socket() function;
- Send and receive data by means of the recvfrom() and sendto() functions.

The steps of establishing a UDP socket communication on the server side are as follows:

- Create a socket with the socket() function;
- Bind the socket to an address using the bind() function;
- Send and receive data by means of recvfrom() and sendto().

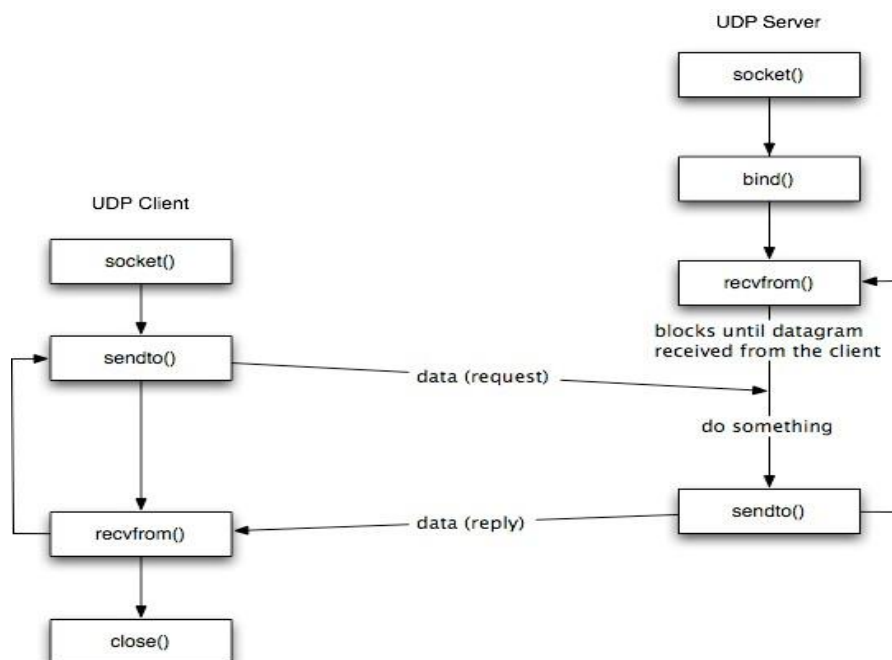**Fig 1. Client Server communication using UDP**

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 send 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. We have mentioned that UDP datagrams are characterized by a length. TCP is instead a byte-stream protocol, without any boundaries at all.

TCP SOCKET API

The sequence of function calls for the client and a server participating in a TCP connection is presented in Figure 2
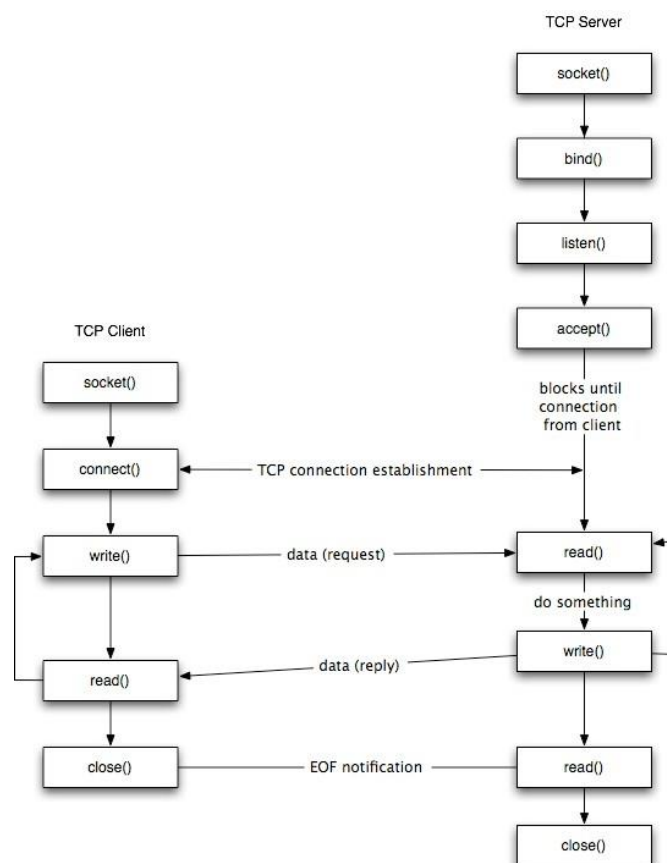

Fig 2 Client Server communication using TCP

CONCLUSION: In this experiment we implemented client server communication using TCP and UDP

FREQUENTLY ASKED QUESTIONS:

1. What is client-server communication?

2. What is socket programming?

3. Explain UDP.

Assignment No:6

PROBLEM STATEMENT: Write a program using TCP socket for wired network for following

a. Say Hello to Each other

b. File transfer

OBJECTIVE: 1. To learn TCP socket for wired network

OUTCOME: Students will able to

1. implement socket programming using TCP.

THEORY-CONCEPT:

SOCKET PROGRAMMING

The Berkeley socket interface, an API, allows communications between hosts or between processes on one computer, using the concept of a socket. It can work with many different I/O devices and drivers, although support for these depends on the operating system implementation. This interface implementation is implicit for TCP/IP, and it is therefore one of the fundamental technologies underlying the Internet. It was first developed at the University of California, Berkeley for use on Unix systems. All modern operating systems now have some implementation of the Berkeley socket interface, as it has become the standard interface for connecting to the Internet. Programmers can make the socket interfaces accessible at three different levels, most powerfully and fundamentally at the RAW socket level. Very few applications need the degree of control over outgoing communications that this provides, so RAW sockets support was intended to be available only on computers used for developing Internet related technologies.

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.

Socket Types

There are four types 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.

Stages for server

1. Socket creation:

     int sockfd = socket(domain, type, protocol)

   sockfd: socket descriptor, an integer (like a file-handle)

   domain: integer, communication domain e.g., AF_INET (IPv4 protocol) , AF_INET6 (IPv6 protocol)

   type: communication type

SOCK_STREAM: TCP(reliable, connection oriented)

SOCK_DGRAM: UDP(unreliable, connectionless)

protocol: Protocol value for Internet Protocol(IP), which is 0. This is the same number which appears on protocol field in the IP header of a packet.(man protocols for more details)

2. Bind:

   int bind(int sockfd, const struct sockaddr *addr, socklen_t addrlen);

   After creation of the socket, bind function binds the socket to the address and port number specified in addr(custom data structure). In the example code, we bind the server to the localhost, hence we use INADDR_ANY to specify the IP address

3. Listen:

   int listen(int sockfd, int backlog);

   It puts the server socket in a passive mode, where it waits for the client to approach the server to make a connection. The backlog, defines the maximum length to which the queue of pending connections for sockfd may grow. If a connection request arrives when the queue is full, the client may receive an error with an indication of ECONNREFUSED.

4. Accept:

   int new_socket= accept(int sockfd, struct sockaddr *addr, socklen_t *addrlen);

   It extracts the first connection request on the queue of pending connections for the listening socket, sockfd, creates a new connected socket, and returns a new file descriptor referring to that socket. At this point, connection is established between client and server, and they are ready to transfer data.

   Stages for Client

   1. Socket connection: Exactly same as that of server's socket creation

   2. Connect:

   int connect(int sockfd, const struct sockaddr *addr, socklen_t addrlen);

   The connect() system call connects the socket referred to by the file descriptor sockfd to the address specified by addr. Server's address and port is specified in addr.

CONCLUSION:

We have successfully implemented the TCP socket programming

FREQUENTLY ASKED QUESTIONS:

1. Explain different type of sockets?

2. What is bind and listen in socket programming?

3. Explain TCP.

Assignment No:7

PROBLEM STATEMENT: Write a program using UDP Sockets to enable file transfer (Script, Text, Audio and Video one file each) between two machines.

OBJECTIVE: 1. To learn how to transfer file using socket programming.

OUTCOME: Students will able to

1. implement socket programming using UDP.

THEORY-CONCEPT:

A transport layer protocol usually has several responsibilities. One is to create a process-to-process communication; UDP uses port numbers to accomplish this. Another responsibility is to provide control mechanisms at the transport level. UDP does this task at a very minimal level. There is no flow control mechanism and there is no acknowledgment for received packets. UDP, however, does provide error control to some extent. If UDP detects an error in the received packet, it silently drops it. UDP is a connectionless, unreliable transport protocol. It does not add anything to the services of IP except for providing process-to-process communication instead of host-to-host communication. UDP packets, called user datagrams, have a fixed-size header of 8 bytes.

❑ Source port number. This is the port number used by the process running on the source host. It is 16 bits long, which means that the port number can range from 0 to 65,535. If the source host is the client (a client sending a request), the port number, in most cases, is an ephemeral port number requested by the process and chosen by the UDP software running on the source host. If the source host is the server (a server sending a response), the port number, in most cases, is a well-known port number

❑ Destination port number. This is the port number used by the process running on the destination host. It is also 16 bits long. If the destination host is the server (a client sending a request), the port number, in most cases, is a well-known port number. If the destination host is the client (a server sending a response), the port number, in most cases, is an ephemeral port number. In this case, the server copies the ephemeral port number it has received in the request packet.

❑ Length. This is a 16-bit field that defines the total length of the user datagram, header plus data. The 16 bits can define a total length of 0 to 65,535 bytes. How- ever, the total length needs to be much

less because a UDP user datagram is stored in an IP datagram with the total length of 65,535 bytes. The length field in a UDP user datagram is actually not necessary. A user datagram is encapsulated in an IP datagram. There is a field in the IP datagram that defines the total length. There is another field in the IP datagram that defines the length of the header. So if we subtract the value of the second field from the first, we can deduce the length of the UDP datagram that is encapsulated in an IP datagram. UDP length = IP length − IP header's length However, the designers of the UDP protocol felt that it was more efficient for the destination UDP to calculate the length of the data from the information provided in the UDP user datagram rather than ask the IP software to supply this information. We should remember that when the IP software delivers the UDP user datagram to the UDP layer, it has already dropped the IP header.

❏ Checksum. This field is used to detect errors over the entire user datagram (header plus data). The checksum is discussed in the next section.

UDP SERVICES

UDP provides process-to-process communication using sockets, a combination of IP addresses and port numbers. Several port numbers used by UDP are shown in Table 1.

Table 1: Port numbers used by UDP

Port	Protocol	Description
7	Echo	Echoes a received datagram back to the sender
9	Discard	Discards any datagram that is received
11	Users	Active users
13	Daytime	Returns the date and the time
17	Quote	Returns a quote of the day
19	Chargen	Returns a string of characters
53	Domain	Domain Name Service (DNS)
67	Bootps	Server port to download bootstrap information
68	Bootpc	Client port to download bootstrap information
69	TFTP	Trivial File Transfer Protocol
111	RPC	Remote Procedure Call
123	NTP	Network Time Protocol
161	SNMP	Simple Network Management Protocol
162	SNMP	Simple Network Management Protocol (trap)

Connectionless Services: As mentioned previously, UDP provides a connectionless service. This

means that each user datagram sent by UDP is an independent datagram. There is no relationship between the different user datagrams even if they are coming from the same source process and going to the same destination program. The user datagrams are not numbered. Also, there is no connection establishment and no connection termination as is the case for TCP. This means that each user datagram can travel on a different path.

Flow Control: UDP is a very simple protocol. There is no flow control, and hence no window mechanism. The receiver may overflow with incoming messages. The lack of flow control means that the process using UDP should provide for this service, if needed. Error Control: There is no error control mechanism in UDP except for the checksum. This means that the sender does not know if a message has been lost or duplicated. When the receiver detects an error through the checksum, the user datagram is silently dis- carded. The lack of error control means that the process using UDP should provide for this service if needed.

Checksum: UDP checksum calculation is different from the one for IP. Here the checksum includes three sections: a pseudo header, the UDP header, and the data coming from the application layer. The pseudo header is the part of the header of the IP packet in which the user datagram is to be encapsulated with some fields filled with 0s If the checksum does not include the pseudoheader, a user datagram may arrive safe and sound. However, if the IP header is corrupted, it may be delivered to the wrong host. The protocol field is added to ensure that the packet belongs to UDP, and not to TCP.

Encapsulation: When a process has a message to send through UDP, it passes the message to UDP along with a pair of socket addresses and the length of data. UDP receives the data and adds the UDP header. UDP then passes the user datagram to IP with the socket addresses. IP adds its own header, using the value 17 in the protocol field, indicating that the data has come from the UDP protocol. The IP datagram is then passed to the data link layer. The data link layer receives the IP datagram, adds its own header (and possibly a trailer), and passes it to the physical layer. The physical layer encodes the bits into electrical or optical signals and sends it to the remote machine.

De-encapsulation: When the message arrives at the destination host, the physical layer decodes the signals into bits and passes it to the data link layer. The data link layer uses the header (and the trailer) to check the data. If there is no error, the header and trailer are dropped and the datagram is passed to IP. The IP software does its own checking. If there is no error, the header is dropped and the user datagram is passed to UDP with the sender and receiver IP addresses. UDP uses the

checksum to check the entire user datagram. If there is no error, the header is dropped and the application data along with the sender socket address is passed to the process. The sender socket address is passed to the process in case it needs to respond to the message received. When a process has a message to send through UDP, it passes the message to UDP along with a pair of socket addresses and the length of data. UDP receives the data and adds the UDP header. UDP then passes the user datagram to IP with the socket addresses. IP adds its own header, using the value 17 in the protocol field, indicating that the data has come from the UDP protocol. The IP datagram is then passed to the data link layer. The data link layer receives the IP datagram, adds its own header (and possibly a trailer), and passes it to the physical layer. The physical layer encodes the bits into electrical or optical signals and sends it to the remote machine. Multiplexing: At the sender site, there may be several processes that need to send user datagrams However, there is only one UDP. This is a many-to-one relationship and requires multiplexing. UDP accepts messages from different processes, differentiated by their assigned port numbers. After adding the header, UDP passes the user datagram to IP. De-multiplexing: At the receiver site, there is only one UDP. However, we may have many processes that can receive user datagrams. This is a one-to-many relationship and requires demultiplexing. UDP receives user datagrams from IP. After error checking and dropping of the header, UDP delivers each message to the appropriate process based on the port numbers.

UDP APPLICATIONS

The following shows some typical applications that can benefit more from the services of UDP than from those of TCP.

❑ UDP is suitable for a process that requires simple request-response communication with little concern for flow and error control. It is not usually used for a process such as FTP that needs to send bulk data.

❑ UDP is suitable for a process with internal flow and error-control mechanisms. For example, the Trivial File Transfer Protocol (TFTP) process includes flow and error control. It can easily use UDP.

ALGORITHM

Server Program

1. Open the Server Socket: ServerSocket server = new ServerSocket(PORT);

2. Wait for the Client Request: Socket client = server.accept();

3. Create I/O streams for communicating to the clientDataInputStream is = new DataInputStream(client.getInputStream());

 DataOutputStreamos = new DataOutputStream(client.getOutputStream());

4. Perform communication with client Receive from client: String line = is.readLine(); Send to client: os.writeBytes("Hello\n")

5. Close socket: client.close();

Client Program

1. Create a Socket Object: Socket client = new Socket(server, port_id);

2. Create I/O streams for communicating with the server. is = new DataInputStream(client.getInputStream());

os = new DataOutputStream(client.getOutputStream());

 3. Perform I/O or communication with the server: Receive data from the server: String line = is.readLine(); Send data to the server: os.writeBytes("Hello\n");

4. Close the socket when done:

5. client.close()

CONCLUSION

Thus, we have successfully implemented UDP socket for file transfer between two machines.

FREQUENTLY ASKED QUESTIONS:

1. What are the applications of UDP?

2. Explain port numbers used in UDP.

3. Explain UDP Services.

Assignment No:8

PROBLEM STATEMENT: Capture TCP and http packets using Wireshark.

OBJECTIVE: 1. Learn TCP, HTTP protocol.

PROBLEM STATEMENT: Capture packets using Wireshark and accomplish the following and save the output in file:

 a. Capture all TCP traffic to/from Facebook, during the time when you log in to your Facebook account

 b. Capture all HTTP traffic to/from Facebook (other website), when you log in to your Facebook account

 c. Write a DISPLAY filter expression to count all TCP packets (captured under item #1) that have the flags SYN, PSH, and RST set. Show the fraction of packets that had each flag set.

 d. Count how many TCP packets you received from / sent to Facebook (other website), and how many of each were also HTTP packets.

OUTCOME: Students will able to

1. Capture tcp and http packets using Wireshark tool.

THEORY-CONCEPT:

HTTP:

The Hypertext Transfer Protocol (HTTP) is an application-level protocol for distributed, collaborative, The Hypertext Transfer Protocol (HTTP) is an application-level protocol for distributed, collaborative, hypermedia information systems. This is the foundation for data communication for the World Wide Web (i.e. internet) since 1990. HTTP is a generic and stateless protocol which can be used for other purposes as well using extensions of its request methods, error codes, and headers.

Basically, HTTP is a TCP/IP based communication protocol, that is used to deliver data (HTML files, image files, query results, etc.) on the World Wide Web. The default port is TCP 80, but other ports can be used as well. It provides a standardized way for computers to communicate with each other. HTTP specification specifies how clients' request data will be constructed and sent to the server, and

how the servers respond to these requests.

HTTP Transaction

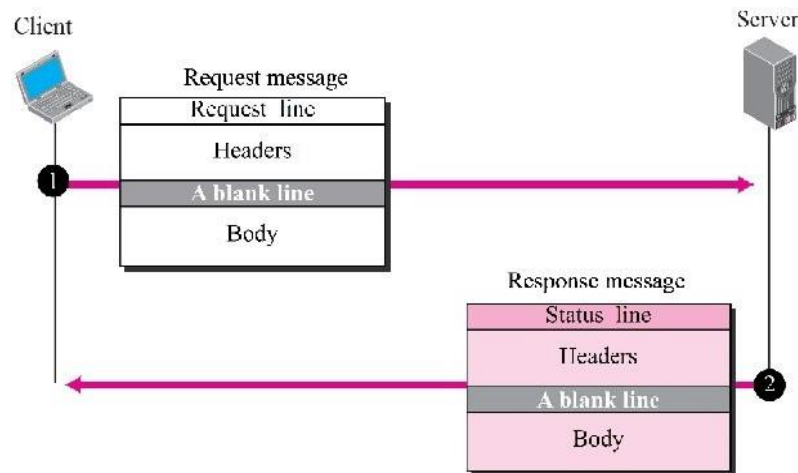Following fig 1 illustrates the HTTP transaction between the client and server.

Fig 1. HTTP Transaction

Request Message

The format of the request is shown in below. A request message consists of a request line, a header, and sometimes a body.
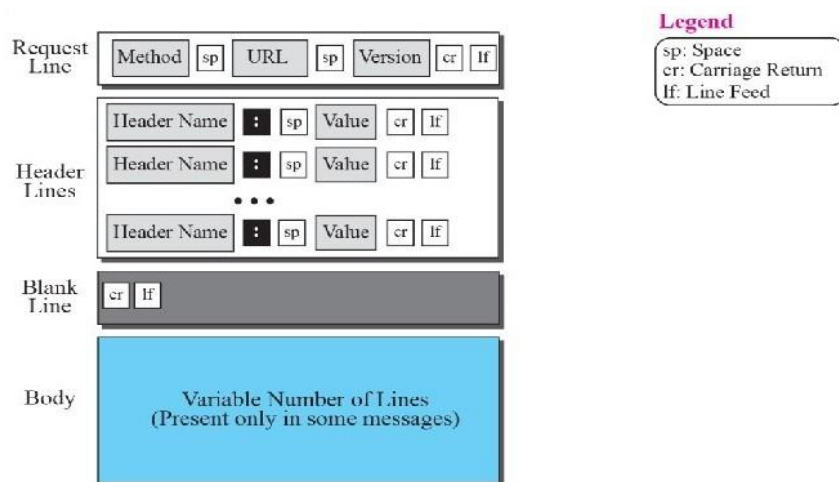

Fig 2. Format of request message

Response Message

The format of the response message is shown below. A response message consists of a status line, header lines, a blank line and sometimes a body.
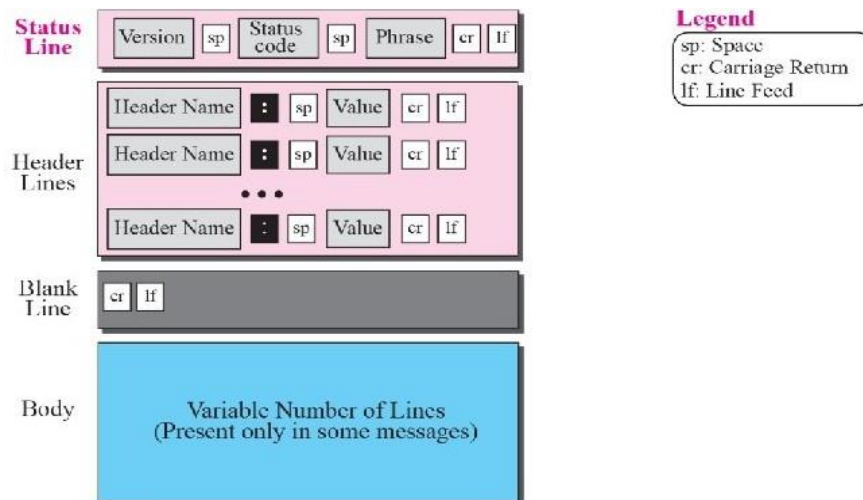

Fig 3. Format of response message

TCP

TCP lies between the application layer and the network layer, and serves as the intermediary between the application programs and the network operations. TCP, unlike UDP, is a connection-oriented protocol. when a process at site A wants to send to and receive data from another process at site B, the following three phases occur:

1. The two TCPs establish a virtual connection between them.

2. Data are exchanged in both directions.

3. The connection is terminated.

TCP FEATURES

1. Numbering System

Although the TCP software keeps track of the segments being transmitted or received, there is no field for a segment number value in the segment header. Instead, there are two fields called the sequence number and the acknowledgment number. These two fields refer to a byte number and not a segment number.

2. Flow Control

TCP, unlike UDP, provides flow control. The sending TCP controls how much data can be accepted from the sending process; the receiving TCP controls how much data can to be sent by the sending

TCP . This is done to prevent the receiver from being overwhelmed with data. The numbering system allows TCP to use a byte oriented flow control.

3. Error Control

To provide reliable service, TCP implements an error control mechanism. Although error control considers a segment as the unit of data for error detection (loss or corrupted segments), error control is byte-oriented.

4. Congestion Control

TCP, unlike UDP, takes into account congestion in the network. The amount of data sent by a sender is not only controlled by the receiver (flow control), but is also determined by the level of congestion, if any, in the network.

PACKET FORMAT

A packet in TCP is called a segment. The format of a segment is shown in Figure . The segment consists of a header of 20 to 60 bytes, followed by data from the application program. The header is 20 bytes if there are no options and up to 60 bytes if it contains options.


a. Segment

b. Header
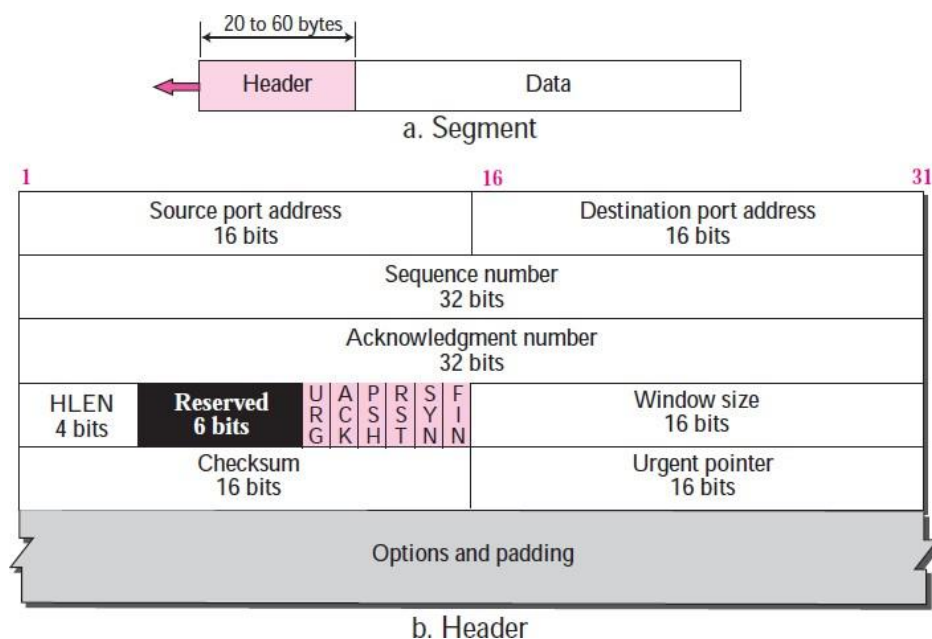
TCP CONNECTION

TCP transmits data in full-duplex mode. When two TCPs in two machines are connected,

they are able to send segments to each other simultaneously. This implies that each party must initialize communication and get approval from the other party before any data are transferred.

Connection establishment using three-way handshaking
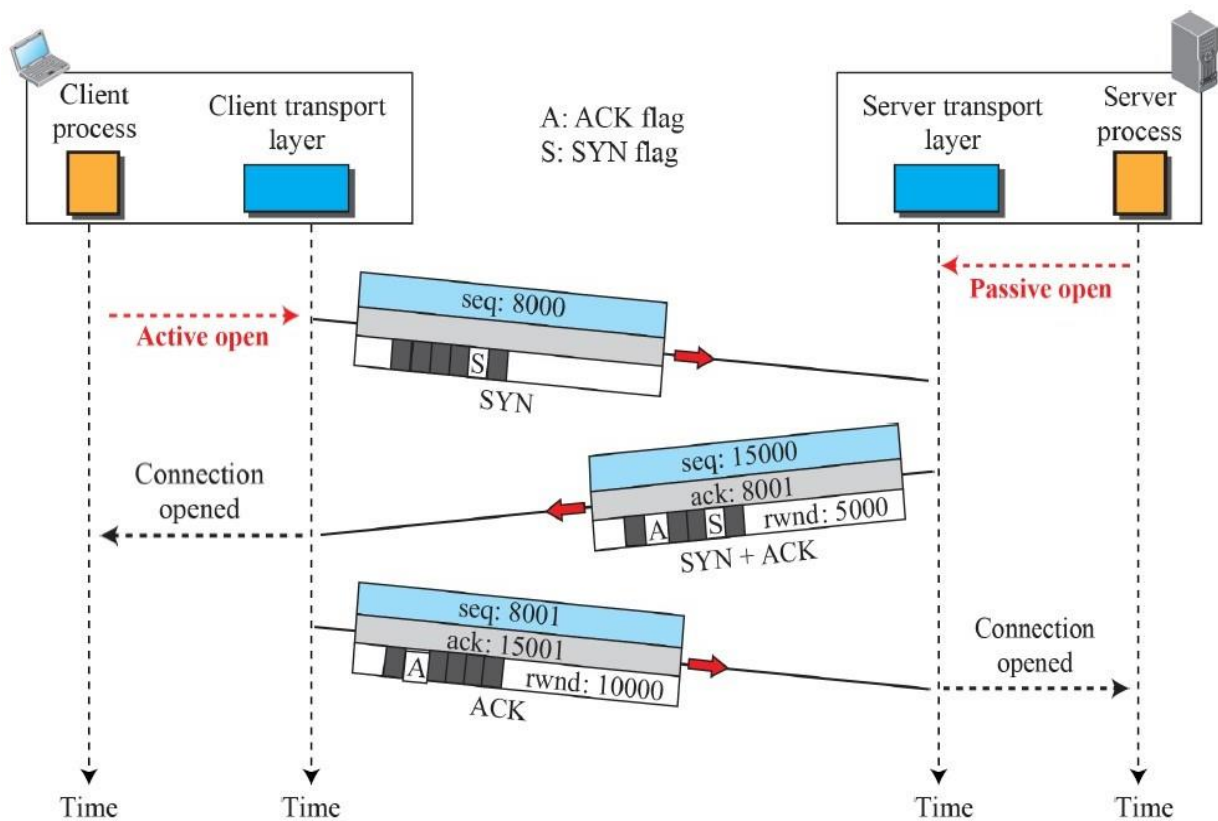
CONCLUSION: We successfully captured tcp and http packets using Wireshark tool.

FREQUENTLY ASKED QUESTIONS:

1. What is Wireshark tool?
2. What is HTTP request & response?
3. Explain TCP packet format.

Assignment No:9

PROBLEM STATEMENT: Study and Analyze the performance of HTTP, HTTPS and FTP protocol using Packet tracer tool.

OBJECTIVE: 1. Learn HTTP, HTTPS and FTP

OUTCOME: Students will able to

1. analyze the performance of HTTP, HTTPS and FTP using packet tracer tool.

THEORY-CONCEPT:

HTTP:

The Hypertext Transfer Protocol (HTTP) is an application-level protocol for distributed, collaborative, hypermedia information systems. This is the foundation for data communication for the World Wide Web (i.e. internet) since 1990. HTTP is a generic and stateless protocol which can be used for other purposes as well using extensions of its request methods, error codes, and headers.

Basically, HTTP is a TCP/IP based communication protocol, that is used to deliver data (HTML files, image files, query results, etc.) on the World Wide Web. The default port is TCP 80, but other ports can be used as well. It provides a standardized way for computers to communicate with each other. HTTP specification specifies how clients' request data will be constructed and sent to the server, and how the servers respond to these requests.

There are three basic features that make HTTP a simple but powerful protocol:

1. **HTTP is connectionless:** The HTTP client, i.e., a browser initiates an HTTP request and after a request is made, the client waits for the response. The server processes the request and sends a response back after which client disconnect the connection. So client and server knows about each other during current request and response only. Further requests are made on new connection like client and server are new to each other.

2. **HTTP is media independent:** It means, any type of data can be sent by HTTP as long as both the client and the server know how to handle the data content. It is required for the client as well as the server to specify the content type using appropriate MIME-type.

3. **HTTP is stateless:** As mentioned above, HTTP is connectionless and it is a direct result of HTTP being a stateless protocol. The server and client are aware of each other only during a current request. Afterwards, both of them forget about each other. Due to this nature of the protocol, neither the client nor the browser can retain information between different requests across the web pages.

The following diagram shows a very basic architecture of a web application and depicts where HTTP sits:
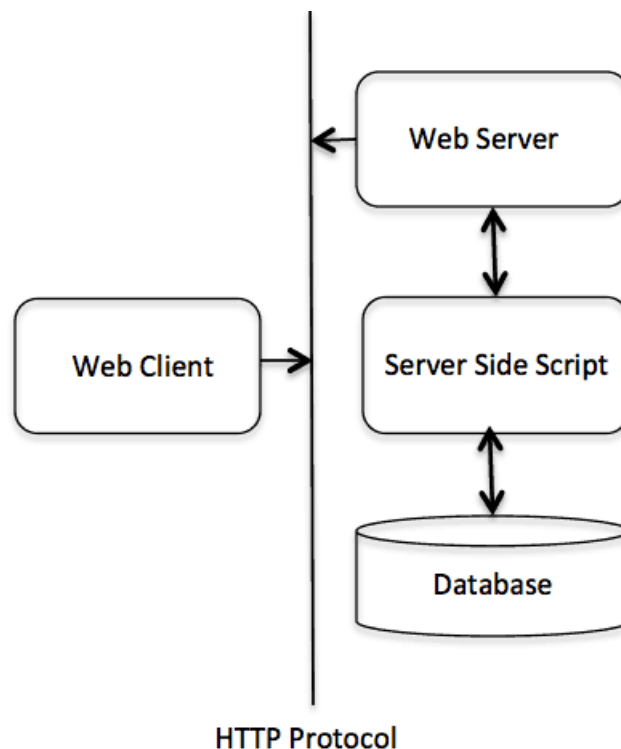

Fig 1. Basic architecture of a web application

The HTTP protocol is a request/response protocol based on the client/server based architecture where web browsers, robots and search engines, etc. act like HTTP clients, and the Web server acts as a server.

Client

The HTTP client sends a request to the server in the form of a request method, URI, and protocol version, followed by a MIME-like message containing request modifiers, client information, and possible body content over a TCP/IP connection.

Server

The HTTP server responds with a status line, including the message's protocol version and a success or error code, followed by a MIME-like message containing server information, entity meta information, and possible entity-body content.

HTTPS:

HTTPS is an abbreviation of Hypertext Transfer Protocol Secure. It is a secure extension or version of HTTP. This protocol is mainly used for providing security to the data sent between a website and the web browser. It is widely used on the internet and used for secure communications. This protocol uses the 443 port number for communicating the data. This protocol is also called HTTP over SSL because the HTTPS communication protocols are encrypted using the SSL (Secure Socket Layer).

By default, it is supported by various web browsers. Those websites which need login credentials should use the HTTPS protocol for sending the data.

Advantages of HTTPS

- The main advantage of HTTPS is that it provides high security to users.
- Data and information are protected. So, it ensures data protection.
- SSL technology in HTTPS protects the data from third-party or hackers. And this technology builds trust for the users who are using it.
- It helps users by performing banking transactions.

Disadvantages of HTTPS

- The big disadvantage of HTTPS is that users need to purchase the SSL certificate.
- The speed of accessing the website is slow because there are various complexities in communication.
- Users need to update all their internal links.

The **File Transfer Protocol (FTP)** is a standard network protocol used for the transfer of computer files between a client and server on a computer network.

FTP employs a **client-server** architecture whereby the client machine has an **FTP client** installed and establishes a connection to an **FTP server** running on a remote machine. After the connection has been established and the user is successfully authenticated, the data transfer phase can begin.

Let's now do FTP configuration in Packet Tracer:

1.Build the network topology.
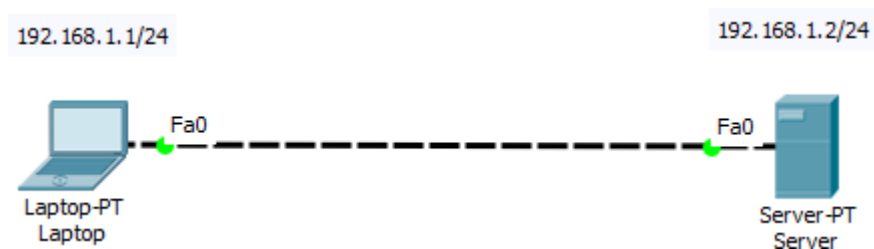
2. Configure static IP addresses on the Laptop and the server.

Laptop: **IP address:** 192.168.1.1 Subnet **Mask:** 255.255.255.0

Server: **IP address:** 192.168.1.2 Subnet **Mask:** 255.255.255.0

3. Now try using an **FTP client** built in the Laptop to send files to an **FTP server** configured in the Server.

From the Laptop's command prompt, FTP the server using the server IP address by typing:

ftp 192.168.1.2

Provide the **username**(cisco) and **password**(cisco) [which are the defaults] for ftp login.

You are now in the FTP prompt .

PC0 has an **FTP client** which can be used to read, write, delete and rename files present in the FTP server.

The **FTP server** can be used to read and write configuration files as well as IOS images. Additionally, the FTP server also supports file operations such rename, delete and listing directory.

With that in mind, we can do something extra. So let's do this:

Create a file in the Laptop then **upload** it to the server using **FTP**.

To do this, open the **Text Editor** in the Laptop, create a file and give it your name of choice.

Type any text in the editor then **save** your file. e.g. myFile.txt.

Now upload the file from the Laptop to the server using FTP. (An FTP connection has to be started first. But this is what we've done in step 3)

So to do an FTP upload, we'll type:

put MyFile.txt

```
ftp>
ftp>put MyFile.txt

Writing file MyFile.txt to 192.168.1.2:
File transfer in progress...

[Transfer complete - 47 bytes]

47 bytes copied in 0.023 secs (2043 bytes/sec)
ftp>
```

Once file upload is successful, go to the Server **FTP directory** to verify if the file sent has been received . To do this, go to **Server-> Services->FTP**. Here look for MyFile.txt sent from the laptop.
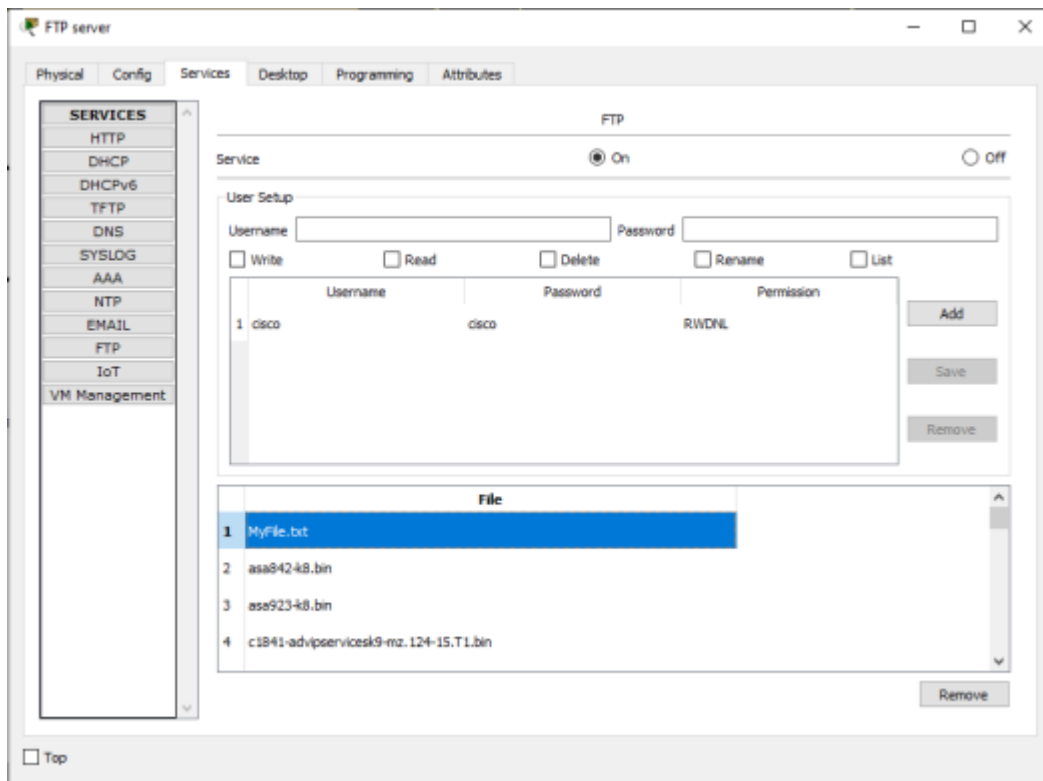
CONCLUSION:

We have successfully analyzed the performance of HTTP, HTTPS and FTP.

FREQUENTLY ASKED QUESTIONS:

1. What is FTP?
2. Differentiate between HTTP and HTTPS.
3. What are the advantages and disadvantages of HTTPS?

Assignment No:10

PROBLEM STATEMENT: To study the SSL protocol by capturing the packets using Wireshark tool while visiting any SSL secured website (banking, e-commerce etc.).

OBJECTIVE: 1. To learn SSL protocol.

        2. To get familiar with Wireshark tool.

OUTCOME: Students will able to

1. understand SSL protocol.

2. Implement SSL protocol using Wireshark tool.

THEORY-CONCEPT:

SSL, or Secure Sockets Layer, is an encryption-based Internet security protocol. It was first developed by Netscape in 1995 for the purpose of ensuring privacy, authentication, and data integrity in Internet communications. SSL is the predecessor to the modern TLS encryption used today.

How does SSL/TLS work? In order to provide a high degree of privacy, SSL encrypts data that is transmitted across the web. This means that anyone who tries to intercept this data will only see a garbled mix of characters that is nearly impossible to decrypt. SSL initiates an authentication process called a handshake between two communicating devices to ensure that both devices are really who they claim to be. SSL also digitally signs data in order to provide data integrity, verifying that the data is not tampered with before reaching its intended recipient. There have been several iterations of SSL, each more secure than the last. In 1999 SSL was updated to become TLS.

Why is SSL/TLS important?

Originally, data on the Web was transmitted in plaintext that anyone could read if they intercepted the message. For example, if a consumer visited a shopping website, placed an order, and entered their credit card number on the website, that credit card number would travel across the Internet unconcealed. SSL was created to correct this problem and protect user privacy. By encrypting any data that goes between a user and a web server, SSL ensures that anyone who intercepts the data can only see a scrambled mess of characters. The consumer's credit card number is now safe, only visible to the shopping website where they entered it.

SSL also stops certain kinds of cyber-attacks: It authenticates web servers, which is important because attackers will often try to set up fake websites to trick users and steal data. It also prevents attackers from tampering with data in transit, like a tamper-proof seal on a medicine container.

Are SSL and TLS the same thing?

SSL is the direct predecessor of another protocol called TLS (Transport Layer Security). In 1999 the Internet Engineering Task Force (IETF) proposed an update to SSL. Since this update was being developed by the IETF and Netscape was no longer involved, the name was changed to TLS. The differences between the final version of SSL (3.0) and the first version of TLS are not drastic; the name change was applied to signify the change in ownership. Since they are so closely related, the two terms are often used interchangeably and confused. Some people still use SSL to refer to TLS, others use the term "SSL/TLS encryption" because SSL still has so much name recognition.

CONCLUSION:

We have successfully implemented SSL protocol using Wireshark tool.

FREQUENTLY ASKED QUESTIONS:

1. What is SSL protocol?
2. Why is SSL/TLS important?

GROUP C

Assignment No:11

PROBLEM STATEMENT: Installing and configuring DHCP server and assign IP addresses to client machines using DHCP server**.**

OBJECTIVE: To learn DHCP server installation and configuration.

OUTCOME: Students will able to

1. install and configure DHCP server.

THEORY-CONCEPT:

A DHCP Server is a network server that automatically provides and assigns IP addresses, default gateways and other network parameters to client devices. It relies on the standard protocol known as Dynamic Host Configuration Protocol or DHCP to respond to broadcast queries by clients.

A DHCP server automatically sends the required network parameters for clients to properly communicate on the network. Without it, the network administrator has to manually set up every client that joins the network, which can be cumbersome, especially in large networks. DHCP servers usually assign each client with a unique dynamic IP address, which changes when the client's lease for that IP address has expired.

How to Install the DHCP Service

Before you configure the DHCP service, you must install it on the server. DHCP is not installed by default during a typical installation of Windows Standard Server 2003 or Windows Enterprise Server 2003. You can install DHCP during the initial installation of Windows Server 2003, or after the initial installation is completed.

How to Install the DHCP Service on an Existing Server

1. Click Start, point to Control Panel, and then click **Add or Remove Programs**.
2. In the **Add or Remove Programs** dialog box, click Add/Remove Windows Components.
3. In the Windows Components Wizard, click Networking Services in the Components list, and then click Details.

4. In the Networking Services dialog box, click to select the **Dynamic Host Configuration Protocol (DHCP)** check box, and then click OK.

5. In the Windows Components Wizard, click Next to start Setup. Insert the Windows Server 2003 CD-ROM into the computer's CD-ROM or DVD-ROM drive if it is prompted to do so. Setup copies the DHCP server and tool files to your computer.

6. When Setup is completed, click Finish.

How to Configure the DHCP Service

After you have installed the DHCP service and started it, you must create a scope. The scope is a range of valid IP addresses available for lease to the DHCP client computers on the network. Microsoft recommends that, each DHCP server in your environment has at least one scope that does not overlap with any other DHCP server scope in your environment. In Windows Server 2003, DHCP servers in an Active Directory-based domain must be authorized to prevent *rogue* DHCP servers from coming online. Any Windows Server 2003 DHCP Server that determines itself to be unauthorized will not manage clients

How to Create a New Scope

1. Click Start, point to Programs, point to Administrative Tools, and then click DHCP.

2. In the console tree, right-click the DHCP server on which you want to create the new DHCP scope, and then click New Scope.

3. In the New Scope Wizard, click Next, and then type a name and description for the scope. The name can be anyone that you want, but it should be descriptive enough so that you can identify the purpose of the scope on your network (for example, you can use a name such as "Administration Building Client Addresses"). Click Next.

4. Type the range of addresses that can be leased as part of this scope. For example, use a range of IP addresses from a starting IP address of 192.168.100.1 to an ending address of 192.168.100.100. Because these addresses are given to clients, they must all be valid addresses for your network and not currently in use. If you want to use a different subnet mask, type the new subnet mask. Click Next.

5. Type any IP addresses that you want to exclude from the range that you entered. These addresses include any one in the range described in step 4 that may have already been statically assigned to various computers in your organization. Typically, domain

controllers, Web servers, DHCP servers, Domain Name System (DNS) servers, and other servers, have statically assigned IP addresses. Click Next.

6. Type the number of days, hours, and minutes before an IP address lease from this scope expires. It determines how long a client can hold a leased address without renewing it. Click Next, and then click **Yes, I want to configure these options now** to extend the wizard to include settings for the most common DHCP options. Click Next.

7. Type the IP address for the default gateway that should be used by clients that obtain an IP address from this scope. Click Add to add the default gateway address in the list, and then click Next.

8. If you are using DNS servers on your network, type your organization's domain name in the **Parent domain** box. Type the name of your DNS server, and then click Resolve to make sure that your DHCP server can contact the DNS server and determine its address. Click Add to include that server in the list of DNS servers that are assigned to the DHCP clients. Click Next, and then follow the same steps. If you are using a Windows Internet Naming Service (WINS) server, by adding its name and IP address, click Next.

9. Click **Yes, I want to activate this scope now** to activate the scope and allow clients to obtain leases from it, and then click Next.

10. Click Finish.

11. In the console tree, click the server name, and then click Authorize on the Action menu.

CONCLUSION:

We have successfully installed and configured DHCP server.

FREQUENTLY ASKED QUESTIONS:

1. What is DHCP?
2. How does DHCP work?
3. What is the difference between DHCP and static IP addressing?

Assignment No: 12

PROBLEM STATEMENT: Write a program for DNS lookup. Given an IP address input, it should return URL and vice versa.

OBJECTIVE: 1. To learn DNS concept.

                2. To get the host name and IP address.

                3. To map the host name with IP address and Vice-versa

OUTCOME: Students will able to

           1.   summarize DNS lookup.

THEORY-CONCEPT:

NEED FOR DNS

To identify an entity, TCP/IP protocols use the IP address, which uniquely identifies the connection of a host to the Internet. However, people prefer to use names instead of numeric addresses. Therefore, we need a system that can map a name to an address or an address to a name. When the Internet was small, mapping was done using a host file. The host file had only two columns: name and address. Every host could store the host file on its disk and update it periodically from a master host file. When a program or a user wanted to map a name to an address, the host consulted the host file and found the mapping. Today, however, it is impossible to have one single host file to relate every address with a name and vice versa. The host file would be too large to store in every host. In addition, it would be impossible to update all the host files every time there is a change. One solution would be to store the entire host file in a single computer and allow access to this centralized information to every computer that needs mapping. But we know that this would create a huge amount of traffic on the Internet. Another solution, the one used today, is to divide this huge amount of information into smaller parts and store each part on a different computer. In this method, the host that needs mapping can contact the closest computer holding the needed information. This method is used by the Domain Name System (DNS).
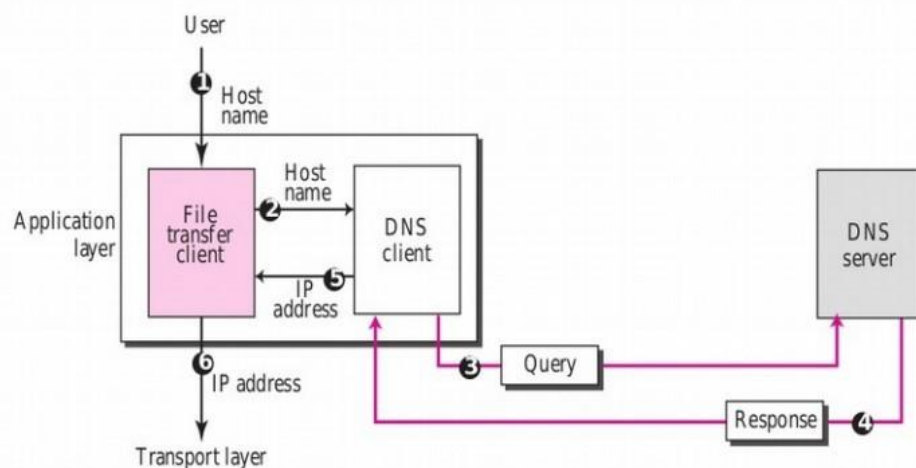
Figure 1: How TCP/IP uses a DNS client and a DNS server to map a name to an address; the reverse mapping is similar.

In Figure 1, a user wants to use a file transfer client to access the corresponding file transfer server running on a remote host. The user knows only the file transfer server name, such as forouzan.com. However, the TCP/IP suite needs the IP address of the file transfer server to make the connection. The following six steps map the host name to an IP address.

1. The user passes the host name to the file transfer client.

2. The file transfer client passes the host name to the DNS client.

3. We know that each computer, after being booted, knows the address of one DNS server. The DNS client sends a message to a DNS server with a query that gives the file transfer server name using the known IP address of the DNS server.

4. The DNS server responds with the IP address of the desired file transfer server.

5. The DNS client passes the IP address to the file transfer server.

6. The file transfer client now uses the received IP address to access the file transfer server. **NAME SPACE**

To be unambiguous, the names assigned to machines must be carefully selected from a name space with complete control over the binding between the names and IP addresses. In other words, the names must be unique because the addresses are unique. A name space that maps each address to a unique name can be organized in two ways: flat or hierarchical.

FLAT NAME SPACE

In a flat name space, a name is assigned to an address. A name in this space is a sequence of characters without structure. The names may or may not have a common section; if they do, it has no meaning. The main disadvantage of a flat name space is that it cannot be used in a large system such as the Internet because it must be centrally controlled to avoid ambiguity and duplication.

HIERARCHICAL NAME SPACE

In a hierarchical name space, each name is made of several parts. The first part can define the nature of the organization, the second part can define the name of an organization, the third part can define departments in the organization, and so on. In this case, the authority to assign and control the name spaces can be decentralized. A central authority can assign the part of the name that defines the nature of the organization and the name of the organization.

DOMAIN NAME SPACE

To have a hierarchical name space, a domain name space was designed. In this design the names are defined in an inverted-tree structure with the root at the top. The tree can have only 128 levels: level 0 (root) to level 127 (see Figure 2)
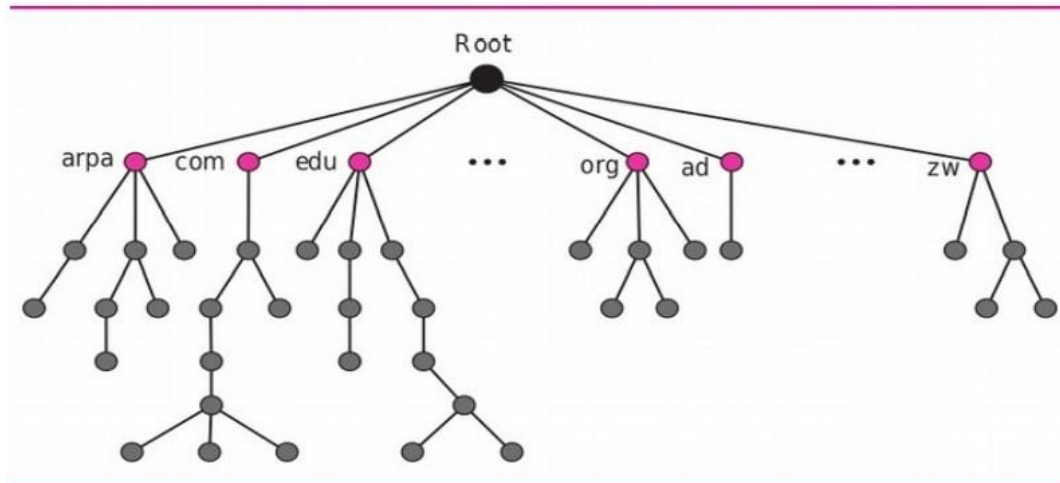

Figure 2 : Domain Name Space

LABEL

Each node in the tree has a label, which is a string with a maximum of 63 characters. The root label is a null string (empty string). DNS requires that children of a node (nodes that branch from the same node) have different labels, which guarantees the uniqueness of the domain names. **DOMAIN NAME**

Each node in the tree has a domain name. A full domain name is a sequence of labels separated by dots (.). The domain names are always read from the node up to the root. The last label is the label of the root (null). This means that a full domain name always ends in a null label, which means the last character is a dot because the null string is nothing.
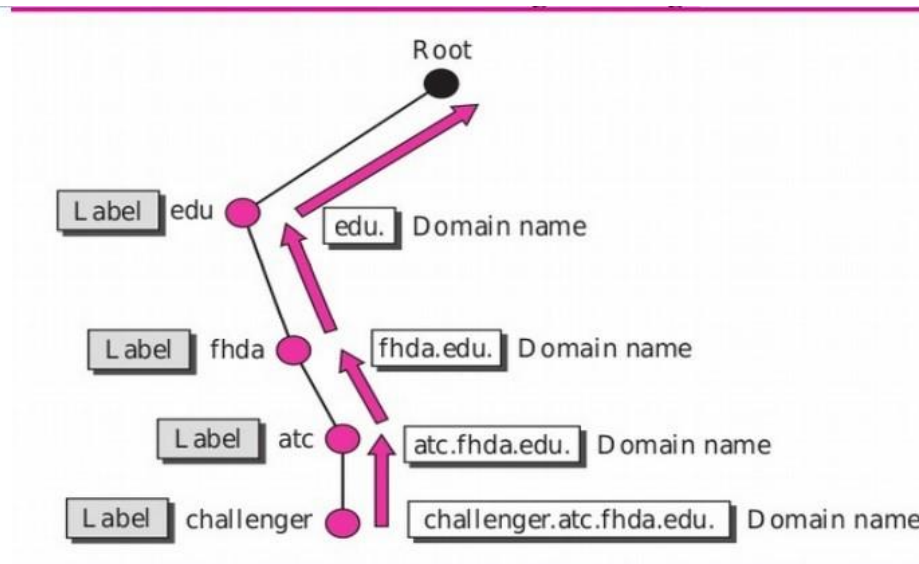

Figure 3 shows some domain names.

DOMAIN

A domain is a subtree of the domain name space. The name of the domain is the name of the node at the top of the subtree. Figure 4 shows some domains. Note that a domain may itself be divided into domains (or subdomains as they are sometimes called).
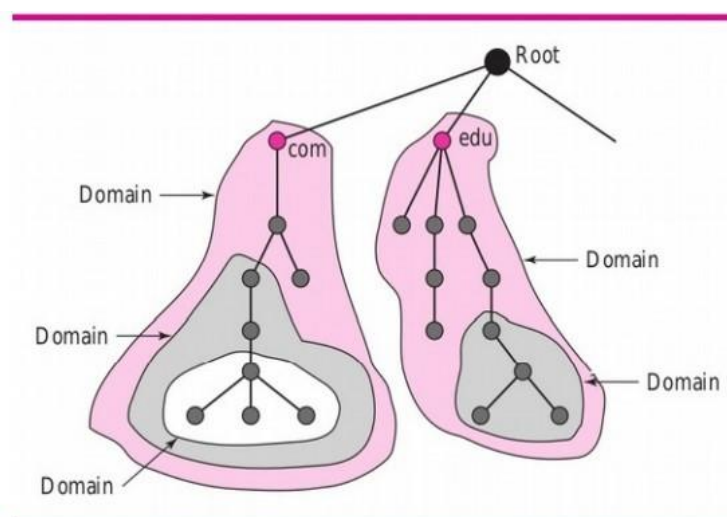

Figure 4 Domain

RESOLUTION

Mapping a name to an address or an address to a name is called name-address resolution. Resolver: DNS is designed as a client-server application. A host that needs to map an address to a name or a name to an address calls a DNS client called a resolver. The resolver accesses the closest DNS server with a mapping request. If the server has the information, it satisfies the resolver; otherwise, it either refers the resolver to other servers or asks other servers to provide the information. After the resolver receives the mapping, it interprets the response to see if it is a real resolution or an error, and finally delivers the result to the process that requested it. Mapping Names to Addresses: Most of the time, the resolver gives a domain name to the server and asks for the corresponding address. In this case, the server checks the generic domains or the country domains to find the mapping. If the domain name is from the generic domains section, the resolver receives a domain name such as "chal.atc.fhda.edu.". The query is sent by the resolver to the local DNS server for resolution: If the local server cannot resolve the query, it either refers the resolver to other servers or asks other servers directly. If the domain name is from the country domains section, the resolver receives a domain name such as "ch.fhda.cu.ca.us.". The procedure is the same. Mapping Addresses to Names A client can send an IP address to a server to be mapped to a domain name. As mentioned before, this is called a PTR query. To answer queries of this kind, DNS uses the inverse domain. However, in the request, the IP address is reversed and two labels, in-addr and arpa, are appended to create a domain acceptable by the inverse domain section. For example, if the resolver receives the IP address 132.34.45.121, the resolver first inverts the address and then adds the two labels before sending. The domain name sent is "121.45.34.132.in-addr.arpa.", which is received by the local DNS and resolved. Recursive Resolution: The client (resolver) can ask for a recursive answer from a name server. This means that the resolver expects the server to supply the final answer. If the server is the authority for the domain name, it checks its database and responds. If the server is not the authority, it sends the request to another server (the parent usually) and waits for the response. If the parent is the authority, it responds; otherwise, it sends the query to yet another server. When the query is finally resolved, the response travels back until it finally reaches the requesting client
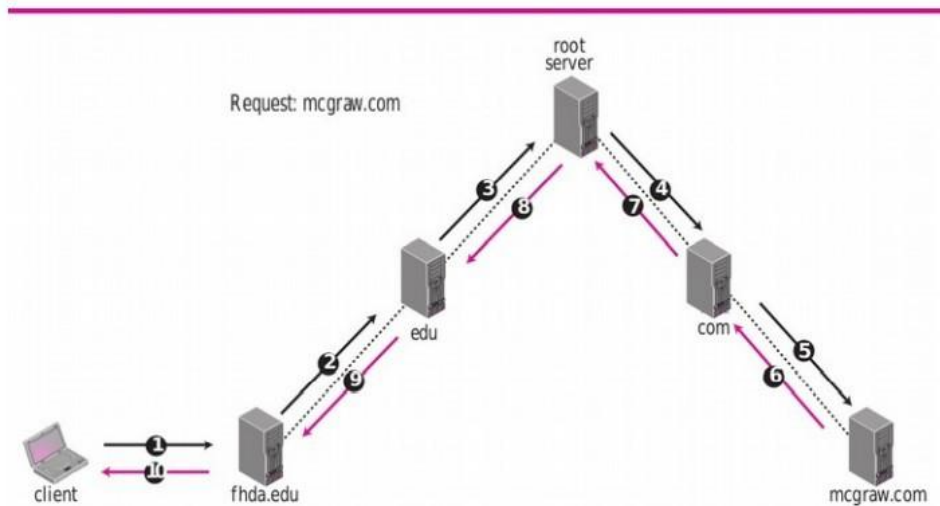
Figure 5: Recursive resolution.

CONCLUSION:

We have successfully performed DNS lookup program.

FREQUENTLY ASKED QUESTIONS:

1. What is DNS and what is its primary purpose?
2. How does the DNS resolution process work?
3. What is a DNS server?