EX.NO 6 Study of Network Simulator

- Ns programming: A Quick start
- Case study I: A simple Wireless network
- Case study II: Create a new agent in Ns
- Ns Status
- Periodical release (ns-2.26, Feb 2003)
- Platform support
- FreeBSD, Linux, Solaris, Windows and Mac

Ns Functionalities

Routing, Transportation, Traffic sources,queuing disciplines, QoS

Wirele
s

Ad hoc routing, mobile IP, sensor-MAC Tracing, visualization and various utilitiesNS(Network Simulators) Most of the commercial simulators are GUI driven, while some network simulators are CLI driven. The network model / configuration describe the state of the network (nodes, routers, switchesand links) and the events (data transmissions, packet error etc.). The important outputs of simulations are the trace files. Trace files log every packet, every event that occurred in the simulation and are used for analysis. Network simulators can also provide other tools to facilitate visual analysis of trends and potential trouble spots.

Most network simulators use discrete event simulation, in which a list of pending "events" is stored, and those events are processed in order, with some events triggering future events such as the event of the arrival of a packet at one node triggering the event of the arrival of that packet at a downstream node. Simulation of networks is a very complex task. For example, if congestion is high, then estimation of the average occupancy is challenging because of high variance. To estimate the likelihood of a buffer overflow in a network, the time required for an accurate answer can be extremely large. Specialized techniques such as "control variants" and "importance sampling" have been developed to speed

simulatio
n.

Examples of network simulators

There are many both free/open-source and proprietary network simulators. Examples of notable

network simulation software are, ordered after how often they are mentioned in research papers:

- ns (open source)
- OPNET (proprietary software)
- NetSim (proprietary software)

Uses of network simulators

Network simulators serve a variety of needs. Compared to the cost and time involved in setting up an entire test bed containing multiple networked computers, routers and data links, network simulators are relatively fast and inexpensive. They allow engineers, researchers to test scenarios that might be particularly difficult or expensive to emulate using real hardware - for instance,simulating a scenario with several nodes or experimenting with a new protocol in the network. Network simulators are particularly useful in allowing researchers to test new networking protocols or changes to existing protocols in a controlled and reproducible environment. A typical network simulator encompasses a wide range of networking technologies and can help the users to build complex networks from basic building blocks such as a variety of nodes and links. With the help of simulators, one can design hierarchical networks using various types of nodes like computers, hubs, bridges, routers, switches, links, mobile units etc. Various types of Wide Area Network (WAN) technologies like TCP, ATM, IP etc. and Local Area Network (LAN) technologies like Ethernet, token rings etc., can all be simulated with a typical simulator and the user can test, analyse various standard results apart from devising some novel protocol or strategy for routing etc. Network simulators are also widely used to simulate battlefield networks in Network-centric warfare There are a wide variety of network simulators, ranging from the very simple to the very complex. Minimally, a network simulator must enable a user to represent a network topology, specifying the nodes on the network, the links between those nodes and the traffic between the nodes. More complicated systems may allow the user to specify everything about the protocols used to handle traffic in a network. Graphical applications allow users to easily visualize the workings of their simulated environment. Text-based applications may provide a less intuitive interface, but may permit more advanced forms of customization.

Packet loss occurs when one or more packets of data travelling across a computer network fail to reach their destination. Packet loss is distinguished as one of the three main error types encountered in digital communications; the other two being bit error and spurious packets caused due to noise. Packets can be lost in a network because they may be dropped when a queue in the network node overflows. The amount of packet loss during the steady state is

another important property of a congestion control scheme. The larger the value of packet loss, the more difficult it is for transport layer protocols to

maintain high bandwidths, the sensitivity to loss of individual packets, as well as to frequency and patterns of loss among longer packet sequences is strongly dependent on the application itself.

Throughput

This is the main performance measure characteristic, and most widely used.In communication networks, such as Ethernet or packetradio, throughput or network throughput is the average rate of successful message delivery over a communication channel. The throughput is usually measured inbitsper second (bit/s or bps), and sometimes in data packets per second or data packets per time slot This measure how soon the receiver is able to get a certain amount of data send by the sender. It is determined as the ratio of the total data received to the end to end delay. Throughput is an important factor which directly impacts the network performance

Delay

Delay is the time elapsed while a packet travels from one point e.g., source premise or network ingress to destination premise or network degrees. The larger the value of delay, the more difficult it is for transport layer protocols to maintain highband widths. We will calculate end to end delay

Queue Length

A queuing system in networks can be described as packets arriving for service, waiting for service if it is not immediate, and if having waited for service, leaving the system after being served. Thus queue length is very important characteristic to determine that how well the active queue management of the congestion control algorithm has been working.

EX.NO 7. Study of TCP/UDP Performance using Simulation Tool

Aim: To Study of TCP/UDP Performance using Simulation Tool

CASE STUDY – 1 TCP

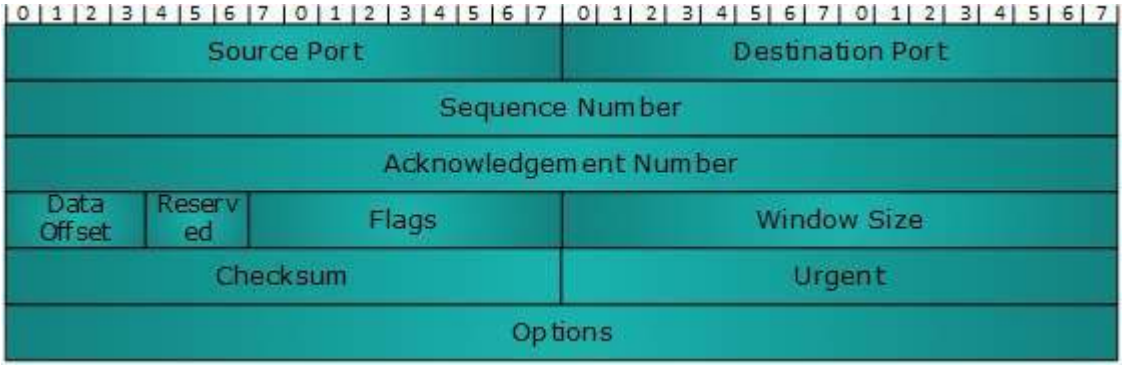
The transmission Control Protocol (TCP) is one of the most important protocols of Internet Protocols suite. It is most widely used protocol for data transmission in communication network such as internet.

Features

- TCP is reliable protocol. That is, the receiver always sends either positive or negative acknowledgement about the data packet to the sender, so that the sender always has bright clue about whether the data packet is reached the destination or it needs to resend it.
- TCP ensures that the data reaches intended destination in the same order it was sent.
- TCP is connection oriented. TCP requires that connection between two remote points be established before sending actual data.
- TCP provides error-checking and recovery mechanism.
- TCP provides end-to-end communication.
- TCP provides flow control and quality of service.
- TCP operates in Client/Server point-to-point mode.
- TCP provides full duplex server, i.e. it can perform roles of both receiver and sender.

Header

The length of TCP header is minimum 20 bytes long and maximum 60 bytes.



- **Source Port (16-bits)** - It identifies source port of the application process on the sending device.
- **Destination Port (16-bits)** - It identifies destination port of the application process on the receiving device.
- **Sequence Number (32-bits)** - Sequence number of data bytes of a segment in a session.
- **Acknowledgement Number (32-bits)** - When ACK flag is set, this number contains the next sequence number of the data byte expected and works as acknowledgement of the previous data received.
- **Data Offset (4-bits)** - This field implies both, the size of TCP header (32-bit words) and the offset of data in current packet in the whole TCP segment.
- **Reserved (3-bits)** - Reserved for future use and all are set zero by default.

- **Flags (1-bit each)**
  - **NS** - Nonce Sum bit is used by Explicit Congestion Notification signaling process.
  - **CWR** - When a host receives packet with ECE bit set, it sets Congestion Windows Reduced to acknowledge that ECE received.
  - **ECE** -It has two meanings:
    - If SYN bit is clear to 0, then ECE means that the IP packet has its CE (congestion experience) bit set.
    - If SYN bit is set to 1, ECE means that the device is ECT capable.
  - **URG** - It indicates that Urgent Pointer field has significant data and should be processed.
  - **ACK** - It indicates that Acknowledgement field has significance. If ACK is cleared to 0, it indicates that packet does not contain any acknowledgement.
  - **PSH** - When set, it is a request to the receiving station to PUSH data (as soon as it comes) to the receiving application without buffering it.
  - **RST** - Reset flag has the following features:
    - It is used to refuse an incoming connection.
    - It is used to reject a segment.
    - It is used to restart a connection.
  - **SYN** - This flag is used to set up a connection between hosts.
  - **FIN** - This flag is used to release a connection and no more data is exchanged thereafter. Because packets with SYN and FIN flags have sequence numbers, they are processed in correct order.
- **Windows Size** - This field is used for flow control between two stations and indicates the amount of buffer (in bytes) the receiver has allocated for a segment, i.e. how much data is the receiver expecting.
- **Checksum** - This field contains the checksum of Header, Data and Pseudo Headers.
- **Urgent Pointer** - It points to the urgent data byte if URG flag is set to 1.
- **Options** - It facilitates additional options which are not covered by the regular header. Option field is always described in 32-bit words. If this field contains data less than 32-bit, padding is used to cover the remaining bits to reach 32-bit boundary.

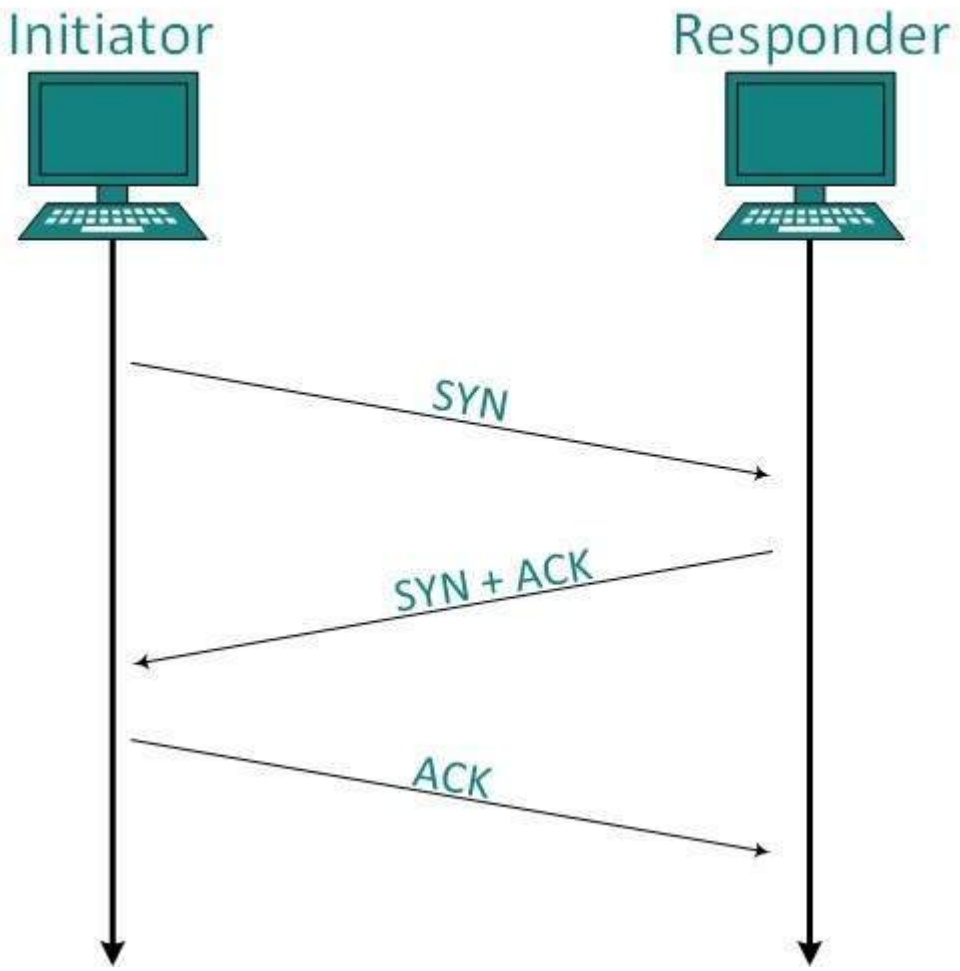
Addressing

TCP communication between two remote hosts is done by means of port numbers (TSAPs). Ports numbers can range from 0 – 65535 which are divided as:

- System Ports (0 – 1023)
- User Ports (1024 – 49151)
- Private/Dynamic Ports (49152 – 65535)

Connection Management

TCP communication works in Server/Client model. The client initiates the connection and the server either accepts or rejects it. Three-way handshaking is used for connection management.



Establishment

Client initiates the connection and sends the segment with a Sequence number. Server acknowledges it back with its own Sequence number and ACK of client's segment which is one more than client's Sequence number. Client after receiving ACK of its segment sends an acknowledgement of Server's response.

Release

Either of server and client can send TCP segment with FIN flag set to 1. When the receiving end responds it back by ACKnowledging FIN, that direction of TCP communication is closed and connection is released.

Bandwidth Management

TCP uses the concept of window size to accommodate the need of Bandwidth management. Window size tells the sender at the remote end, the number of data byte segments the receiver at this end can receive. TCP uses slow start phase by using window size 1 and increases the window size exponentially after each successful communication.

For example, the client uses windows size 2 and sends 2 bytes of data. When the acknowledgement of this segment received the windows size is doubled to 4 and next sent the segment sent will be 4 data bytes long. When the acknowledgement of 4-byte data segment is received, the client sets windows size to 8 and so on.

If an acknowledgement is missed, i.e. data lost in transit network or it received NACK, then the

window size is reduced to half and slow start phase starts again.

Error Control &and Flow Control

TCP uses port numbers to know what application process it needs to handover the data segment. Along with that, it uses sequence numbers to synchronize itself with the remote host. All data segments are sent and received with sequence numbers. The Sender knows which last data segment was received by the Receiver when it gets ACK. The Receiver knows about the last segment sent by the Sender by referring to the sequence number of recently received packet.

If the sequence number of a segment recently received does not match with the sequence number the receiver was expecting, then it is discarded and NACK is sent back. If two segments arrive with the same sequence number, the TCP timestamp value is compared to make a decision.

Multiplexing

The technique to combine two or more data streams in one session is called Multiplexing. When a TCP client initializes a connection with Server, it always refers to a well-defined port number which indicates the application process. The client itself uses a randomly generated port number from private port number pools.

Using TCP Multiplexing, a client can communicate with a number of different application process in a single session. For example, a client requests a web page which in turn contains different types of data (HTTP, SMTP, FTP etc.) the TCP session timeout is increased and the session is kept open for longer time so that the three-way handshake overhead can be avoided.

This enables the client system to receive multiple connection over single virtual connection. These virtual connections are not good for Servers if the timeout is too long.

Congestion Control

When large amount of data is fed to system which is not capable of handling it, congestion occurs. TCP controls congestion by means of Window mechanism. TCP sets a window size telling the other end how much data segment to send. TCP may use three algorithms for congestion control:

- Additive increase, Multiplicative Decrease
- Slow Start
- Timeout React

Timer Management

TCP uses different types of timer to control and management various tasks:

Keep-alive timer:

- This timer is used to check the integrity and validity of a connection.
- When keep-alive time expires, the host sends a probe to check if the connection still exists.

Retransmission timer:

- This timer maintains stateful session of data sent.
- If the acknowledgement of sent data does not receive within the Retransmission time, the data segment is sent again.

Persist timer:

- TCP session can be paused by either host by sending Window Size 0.
- To resume the session a host needs to send Window Size with some larger value.
- If this segment never reaches the other end, both ends may wait for each other for infinite time.
- When the Persist timer expires, the host re-sends its window size to let the other end know.
- Persist Timer helps avoid deadlocks in communication.

Timed-Wait:

- After releasing a connection, either of the hosts waits for a Timed-Wait time to terminate the connection completely.
- This is in order to make sure that the other end has received the acknowledgement of its connection termination request.
- Timed-out can be a maximum of 240 seconds (4 minutes).

Crash Recovery

TCP is very reliable protocol. It provides sequence number to each of byte sent in segment. It provides the feedback mechanism i.e. when a host receives a packet, it is bound to ACK that packet having the next sequence number expected (if it is not the last segment).

When a TCP Server crashes mid-way communication and re-starts its process it sends TPDU broadcast to all its hosts. The hosts can then send the last data segment which was never unacknowledged and carry onwards.

CASE STUDY – 2 UDP

The User Datagram Protocol (UDP) is simplest Transport Layer communication protocol available of the TCP/IP protocol suite. It involves minimum amount of communication mechanism. UDP is said to be an unreliable transport protocol but it uses IP services which provides best effort delivery mechanism.

In UDP, the receiver does not generate an acknowledgement of packet received and in turn, the sender does not wait for any acknowledgement of packet sent. This shortcoming makes this protocol unreliable as well as easier on processing.

Requirement of UDP

A question may arise, why do we need an unreliable protocol to transport the data? We deploy UDP where the acknowledgement packets share significant amount of bandwidth along with the actual data. For example, in case of video streaming, thousands of packets are forwarded towards its users. Acknowledging all the packets is troublesome and may contain huge amount of bandwidth wastage. The best delivery mechanism of underlying IP protocol ensures best efforts to deliver its packets, but even if some packets in video streaming get lost, the impact is not calamitous and can be ignored easily. Loss of few packets in video and voice traffic sometimes goes unnoticed.

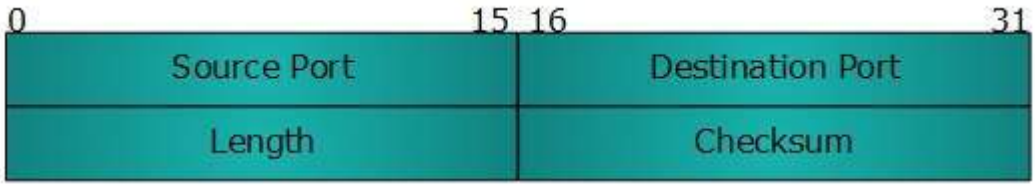
Features

- UDP is used when acknowledgement of data does not hold any significance.

- UDP is good protocol for data flowing in one direction.
- UDP is simple and suitable for query based communications.
- UDP is not connection oriented.
- UDP does not provide congestion control mechanism.
- UDP does not guarantee ordered delivery of data.
- UDP is stateless.
- UDP is suitable protocol for streaming applications such as VoIP, multimedia streaming.

UDP Header

UDP header is as simple as its function.



UDP header contains four main parameters:

- **Source Port** - This 16 bits information is used to identify the source port of the packet.
- **Destination Port** - This 16 bits information, is used identify application level service on destination machine.
- **Length** - Length field specifies the entire length of UDP packet (including header). It is 16-bits field and minimum value is 8-byte, i.e. the size of UDP header itself.
- **Checksum** - This field stores the checksum value generated by the sender before sending. IPv4 has this field as optional so when checksum field does not contain any value it is made 0 and all its bits are set to zero.

UDP application

Here are few applications where UDP is used to transmit data:

- Domain Name Services
- Simple Network Management Protocol
- Trivial File Transfer Protocol
- Routing Information Protocol
- Kerberos