COMPUTER NETWORK SAMPLE SET

- **What are the three criteria necessary for an effective and efficient network?**

The three criteria are performance, reliability, and security

- **For n devices in a network, what is the number of cable links required for a mesh, ring, bus, and star topology?**

The number of cables for each type of network is:

a. Mesh: n (n – 1) / 2

b. Star: n

c. Ring: n – 1

d. Bus: one backbone and n drop lines

- **A color image uses 16 bits to represent a pixel. What is the maximum number of different colors that can be represented?**

With 16 bits, we can represent up to 2^{16} different colors.

- **A client's browser sends an HTTP request to a website. The website responds with a handshake and sets up a TCP connection. The connection setup takes 3.3 ms, including the RTT. The browser then sends the request for the website's index file. The index file references 23 additional images, which are to be requested/ downloaded by the client's browser. Assuming all other conditions are equal, how much longer would non-persistent HTTP take than persistent HTTP?**

Given that the connection setup takes 3.3 ms (including the RTT), and the index file references 23 additional images, we can calculate the time difference as follows:

Non-Persistent HTTP

For each of the 23 images, a new TCP connection needs to be established. So, the total time taken would be:

23 images * 3.3 ms = 75.9 ms

Persistent HTTP

In a persistent HTTP connection, the same TCP connection is reused for all the 23 images. So, the total time taken would be:

1 * 3.3 ms = 3.3 ms

Therefore, the time difference between non-persistent HTTP and persistent HTTP would be: 75.9 ms (Non-Persistent) - 3.3 ms (Persistent) = 72.6 ms

So, non-persistent HTTP would take 72.6 ms longer than persistent HTTP under the given conditions.

- **Define the analog hierarchy used by telephone companies and list different levels of the hierarchy.**

To maximize the efficiency of their infrastructure, telephone companies have traditionally multiplexed analog signals from lower-bandwidth lines onto higher-bandwidth lines. The analog hierarchy uses voice channels (4 KHz), groups (48 KHz), supergroups (240 KHz), master groups (2.4 MHz), and jumbo groups (15.12 MHz)

- **Assume that a voice channel occupies a bandwidth of 4 kHz. We need to multiplex 10 voice channels with guard bands of 500 Hz using FDM. Calculate the required bandwidth**

To multiplex 10 voice channels, we need nine guard bands. The required bandwidth is then B = (4 KHz) × 10 + (500 Hz) × 9 = 44.5 KHz

- **Difference between centralized P2P network and de centralized P2P network**

➢ However, it represents a single point of failure which reduces the reliability of the system. In completely decentralized P2P systems, a central authority for storing data and handling all the queries is not available.

➢ Interconnected peers are able to participate in transactions by interacting with each other and make local autonomous decisions to achieve their objectives. Peers are responsible for storing, sharing information and handling the queries.

➢ Peers act as clients and request services from other peers as well as servers and provide services to other peers. These systems provide improved robustness and enhanced scalability compared to centralized systems.

➢ The fundamental difference between the two approaches is that one prioritizes robustness, while the other prioritizes efficiency.

➢ A decentralized index approach tends to be more robust (no single point of failure), but it is usually tricky to make it as efficient as a centralized approach. In terms of scalability, decentralized approaches have a bigger potential, but it is not trivial to ensure that a given decentralized system actually scales well from both a theoretical and a practical point of view.

- **Differentiate between FQDN and PQDN with example ?**

FQDN - A fully qualified domain name is made up of a host name and a domain name. For example, www.example.com is an FQDN.

PQDN - A partially qualified domain name is a domain name that's missing some of the information needed to be an FQDN. For example, example.com is a PQDN for the site www.example.com

- **Calculate the required bandwidth, if in a communication channel the signal power is 100 W and noise power is 10 W and the information transmission rate is 10kbps.**

Data rate = bandwidth * log (1 + (Signal power / Noise power))

10 = bandwidth * log (1 + (100 / 10)) { log (1 + (100 / 10)) = 3.4594 approx. equal to 4 is taken here}

10 = bandwidth * 4 Bandwidth = 2.5 KHz

- **List the advantages and disadvantages of star topology**

Advantages of Star Topology:

1) As compared to Bus topology it gives far much better performance, signals don't necessarily get transmitted to all the workstations.

2) Easy to connect new nodes or devices. In star topology new nodes can be added easily without affecting rest of the network. Similarly components can also be removed easily.

3) Centralized management. It helps in monitoring the network.

4) Failure of one node or link doesn't affect the rest of network. At the same time it's easy to detect the failure and troubleshoot it.

Disadvantages of Star Topology:

1) Too much dependency on central device has its own drawbacks. If it fails whole network goes down.

2) The use of hub, a router or a switch as central device increases the overall cost of the network.

3) Performance and as well number of nodes which can be added in such topology is depended on capacity of central device.

- **During TCP transmission, how many sequence numbers are consumed by a SYN, FIN and SYN+ACK ?**

In Transmission Control Protocol (TCP),

- SYN packet consumes one sequence number
- ACK packet consumes one sequence number
- SYN + ACK packet consumes two sequence number

- **Consider the Go back N protocol with a sender's window size of 'n'. Suppose that at time 't', the next inorder packet the receiver is expecting has a sequence number of 'K'. Assume that the medium does not reorder messages. What are the possible sets of sequence numbers inside the sender's window at time 't'. Assume the sender has already received the ACKs.**

In Go back N protocol, the receiver window size is 1.

It is given that receiver expects the packet having sequence number 'K'.

It means it has processed all the packets ranging from 0 to K-1.

It is given that sender has received the acknowledgement for all these packets.

So, outstanding packets in sender's window waiting for the acknowledgement starts from K.

Sender window size = n.

Therefore, last packet in sender's window will have sequence number K+n-1.

- **If the packet size is 1 KB (binary unit) and propagation time is 15 msec, the channel capacity is 10^9 b/sec, find the transmission time and utilization of sender in stop and wait protocol**

0.0003413 %

- **In a TCP connection, the initial sequence number at the client site is 4000. The client opens the connection, sends three segments, the second of which carries 2000 bytes of data, and closes the connection. What is the value of the sequence number in each of the following segments sent by the client?**

a. The SYN segment

b. The data segment

c. The FIN segment

The SYN segment 4000

The data segment 4001 - 6001

The FIN segment 6002

- **Discuss the working of email using SMTP and POP ?**

SMTP is a push protocol; it pushes the message from the client to the server. In other words, the direction of the bulk data (messages) is from the client to the server. On the other hand, retrieving messages from mail boxes needs a pull protocol; the client must pull messages from the server. The direction of the bulk data is from the server to the client. The third stage uses a message access agent (MAA) such as POP3 or IMAP4.

- **Explain recursive resolution and iterative resolution ?**

In recursive resolution the client queries just one server. In iterative resolution the client queries more than one server

- **Calculate the total time required to transfer a 3 MB file in the following cases, assuming a RTT of 60 ms, a packet size of 2 KB and an initial 200 ms of "handshaking" before data is sent. The bandwidth is 10 Mbps, and data packets can be sent continuously.**

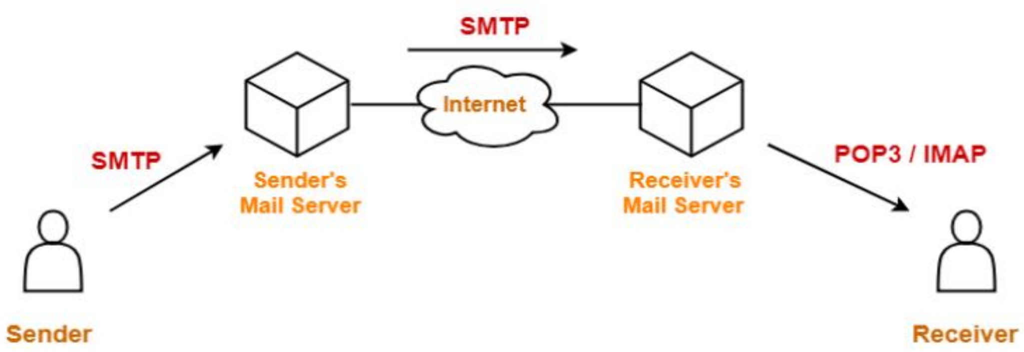
Total time = initial handshaking + network delay

Initial handshaking = 2*RTT = 2*60 ms = 160 ms.

Delay = propagation delay + transmission delay (assuming processing & queuing delays are not significant).

- **Working of Simple Mail Transfer Protocol**

- SMTP server is always on a listening mode.
- lient initiates a T P connection with the SMTP server.
- SMTP server listens for a connection and initiates a connection on that port.
- The connection is established.
- lient informs the SMTP server that it would like to send a mail.
- Assuming the server is OK, client sends the mail to its mail server.
- lient's mail server use DNS to get the IP Address of receiver's mail server.
- Then, SMTP transfers the mail from sender's mail server to the receiver's mail server.



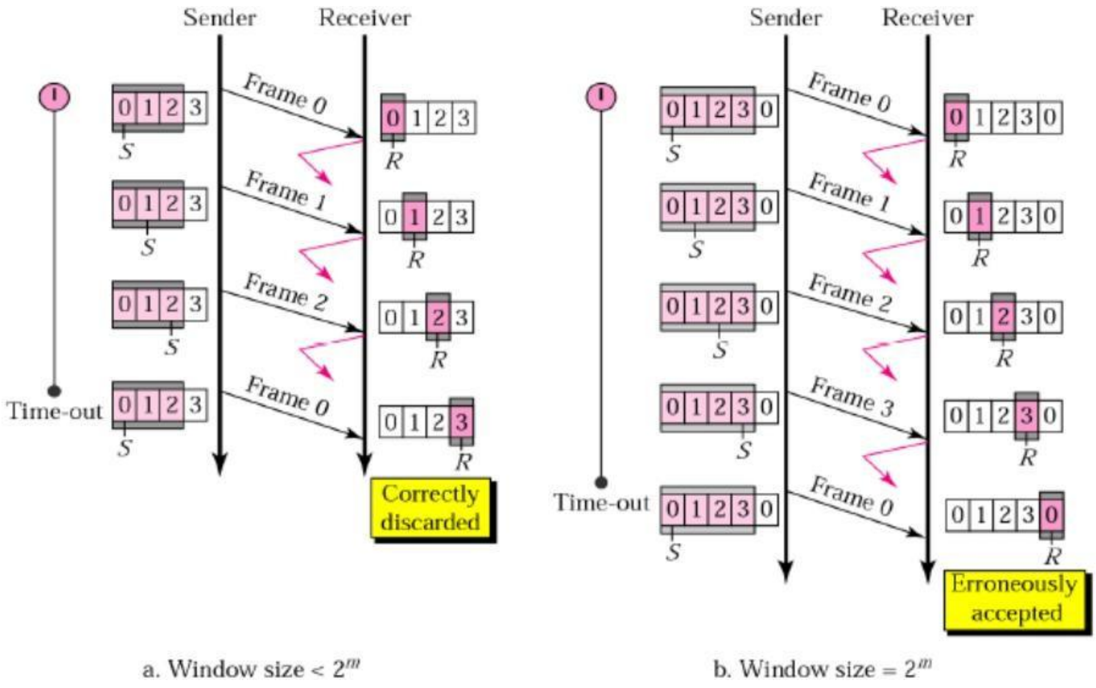
While sending the mail, SMTP is used two times-

1. Between the sender and the sender's mail server
2. Between the sender's mail server and the receiver's mail server

- **In our reliable data transfer protocol (rdt), why did we need to introduce sequence numbers?**

Sequence numbers are required for a receiver to find out whether an arriving packet contains new data or is a retransmission.

- **Explain why the size of the sender window must be less than 2^m for Go Back-N ARQ.**



- **The distance from earth to a distant planet is approximately 9×10^{10} m. What is the channel utilization if a stop-and-wait protocol is used for frame transmission on a 64 Mbps point-to-point link? Assume that the frame size is 32 KB and the speed of light is 3×10^{8} m/s.**

Distance = 9×10^{10} m

Datarate = 64 Mbps

Size = 32 KByte = 256 Kbits

Propagation Speed = 3×10^{8} m/s

Transmission Delay = Packet Size / Datarate = 256 Kb/ 64 Mbps = 0.004 s

Propagation Delay = Distance / Propagation Speed = $9\times10^{10} / 3\times10^{8} = 300$ s

Assuming no processing delay on receiver and ack size is negligible, only one packet is sent in RTT ie 600 s

Utilization = 0.004 / (0.004 + 600) = 6×10^{-6}

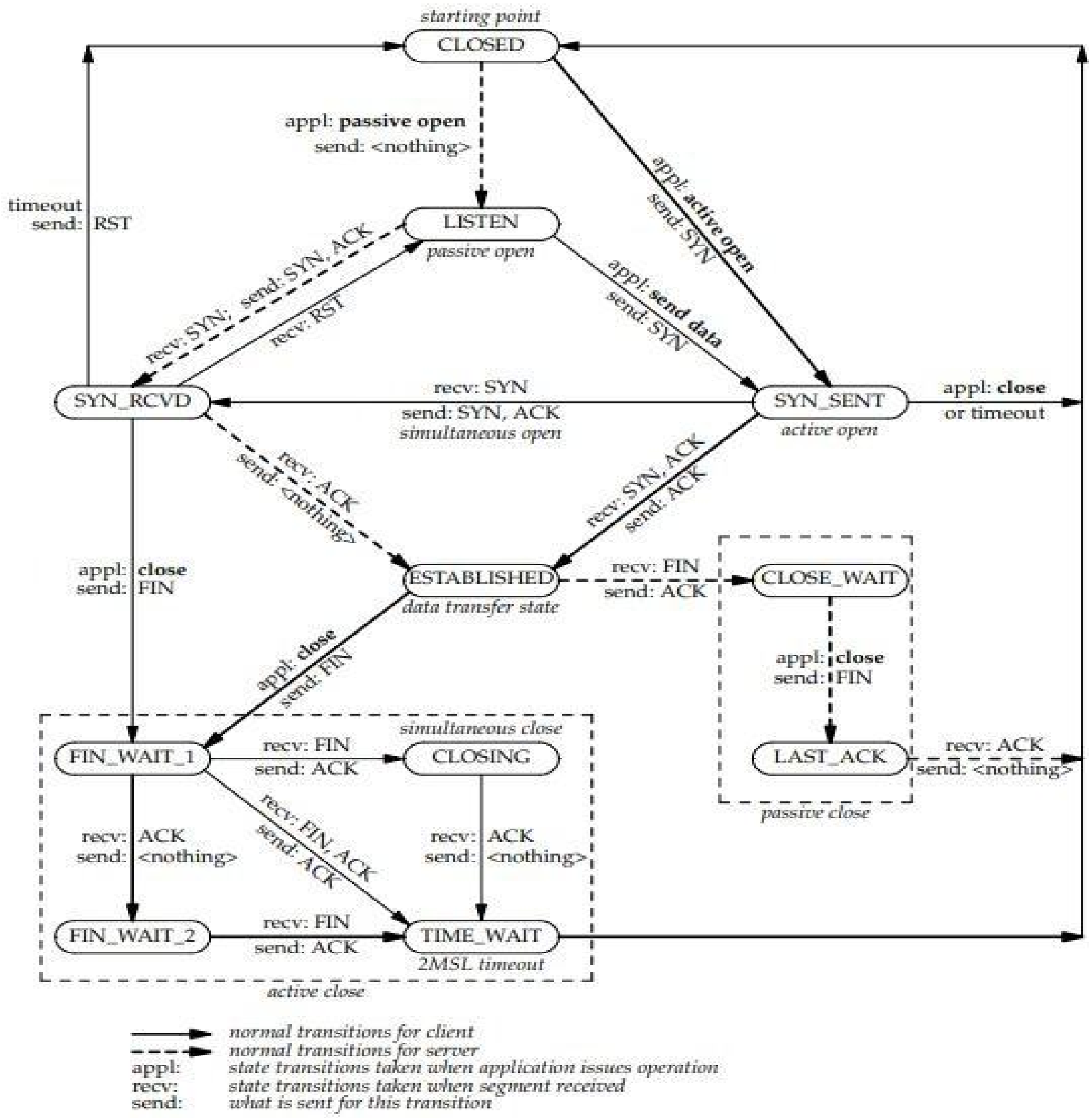
- **If the bandwidth of the line is 1.5 Mbps, RTT is 45 msec & packet size is 1 KB, find the link utilization in stop and wait**

10.8%

- **Discuss the TCP State Transition Diagram**

A connection progresses through a series of states during its lifetime. The states are: LISTEN, SYN-SENT, SYNRECEIVED, ESTABLISHED, FIN-WAIT-1, FIN-WAIT-2, CLOSE-WAIT, CLOSING, LAST-ACK, TIME-WAIT, and the fictional state CLOSED. CLOSED is fictional because it represents the state when there is no TCB, and therefore, no connection. Briefly the meanings of the states are:

- LISTEN represents waiting for a connection request from any remote TCP and port.
- SYN-SENT represents waiting for a matching connection request after having sent a connection request.
- SYN-RECEIVED represents waiting for a confirming connection request acknowledgment after having both received and sent a connection request.
- ESTABLISHED represents an open connection, data received can be delivered to the user. The normal state for the data transfer phase of the connection.
- FIN-WAIT-1 represents waiting for a connection termination request from the remote TCP, or an acknowledgment of the connection termination request previously sent.
- FIN-WAIT-2 represents waiting for a connection termination request from the remote TCP.
- CLOSE-WAIT represents waiting for a connection termination request from the local user.
- CLOSING represents waiting for a connection termination request acknowledgment from the remote TCP.
- LAST-ACK represents waiting for an acknowledgment of the connection termination request previously sent to the remote TCP (which includes an acknowledgment of its connection termination request).
- TIME-WAIT represents waiting for enough time to pass to be sure the remote TCP received the acknowledgment of its connection termination request. CLOSED represents no connection state at all.



- **Non-persistent vs Persistent HTTP connection**

Non-persistent HTTP connection	Persistent HTTP connection
Non-persistent HTTP connection is one that is used for serving exactly one request and sending one response.	Persistent HTTP connection is one that can be used for serving multiple requests.
HTTP server closes the T P connection automatically after sending a HTTP response.	HTTP server closes the T P connection only when it is not used for a certain configurable amount of time.
A new separate T P connection is used for each object.	A single T P connection is used for sending multiple objects one after the other.

Suppose a TCP connection is transferring a file of 5000 bytes. The first byte is numbered 10,00l. What are the sequence numbers for each segment if data are sent in five segments, each carrying 1000 bytes?

The following shows the sequence number for each segment:

Segment 1	Sequence Number: 10,001 (range: 10,001 to 11,(00)
Segment 2	Sequence Number: 11,001 (range: 11,001 to 12,000)
Segment 3	Sequence Number: 12,001 (range: 12,001 to 13,000)
Segment 4	Sequence Number: 13,001 (range: 13,001 to 14,000)
Segment 5	Sequence Number: 14,001 (range: 14,001 to 15,000)

Assume that, in a Stop-and-Wait ARQ system, the bandwidth of the line is 1 Mbps, and 1 bit takes 20 ms to make a round trip. What is the bandwidth-delay product? Ifthe system data frames are 1000 bits in length, what is the utilization percentage of the link?

The bandwidth-delay product is

$$(1 \times 10^{6}) \times (20 \times 10^{-3}) = 20{,}000 \text{ bits}$$

The system can send 20,000 bits during the time it takes for the data to go from the sender to the receiver and then back again. However, the system sends only 1000 bits. We can say that the link utilization is only 1000/20,000, or 5 percent.

What is the utilization percentage of the link, if we have a protocol that can send up to 15 frames before stopping and worrying about the acknowledgments

The bandwidth-delay product is still 20,000 bits. The system can send up to 15 frames or 15,000 bits during a round trip. This means the utilization is 15,000/20,000, or 75 percent.

Using 5-bit sequence numbers, what is the maximum size of the send and receive windows for each of the following protocols?
a. Stop-and-Wait ARQ
b. Go-Back-N ARQ
c. Selective-Repeat ARQ

Stop-And-Wait ARQ	send window = **1**	receive window = **1**
Go-Back-N ARQ	send window $= 2^5 - 1 = \mathbf{31}$	receive window = **1**
Selective-Repeat ARQ	send window $= 2^4 = \mathbf{16}$	receive window = **16**

A sender sends a series of packets to the same destination using 5-bit sequence numbers. If the sequence number starts with 0, what is the sequence number after sending 100 packets?

A five-bit sequence number can create sequence numbers from 0 to 31. The sequence number in the Nth packet is (N mod 32). This means that the 101th packet has the sequence number (101 mod 32) or 5.

What can you say about the TCP segment in which the value of the control field is one of the following?
a. 000000
b. 000001
c. 010001

a. None of the control bits are set. The segment is part of a data transmission without piggybacked acknowledgment.
b. The ***FIN*** bit is set. This is a FIN segment request to terminate the connection.
c. The ***ACK*** and the ***FIN*** bits are set. This is a ***FIN+ACK*** in response to a received ***FIN*** segment.

In TCP, if the value of HLEN is 0111, how many bytes of option are included in the segment?

0111 in decimal is 7. The total length of the header is 7×4 or 28. The base header is 20 bytes. The segment has 8 bytes of options.

TCP is sending data at 1 Mbyte/s. If the sequence number starts with 7000, how long does it take before the sequence number goes back to zero?

The largest number in the sequence number field is $2^{32} - 1$. If we start at 7000, it takes $[(2^{32} - 1) - 7000] / 1{,}000{,}000 =$ **4295** seconds.

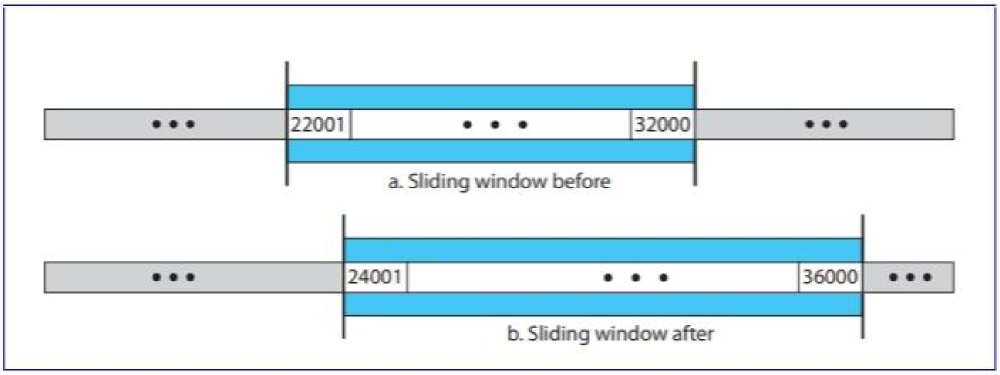
To make the initial sequence number a random number, most systems start the counter at 1 during bootstrap and increment the counter by 64,000 every 0.5 s. How long does it take for the counter to wrap around?

Every second the counter is incremented by $64{,}000 \times 2 =$ **128,000**. The sequence number field is 32 bits long and can hold only $2^{32}-1$. So it takes $(2^{32}-1)/(128{,}000)$ seconds or **33,554** seconds.

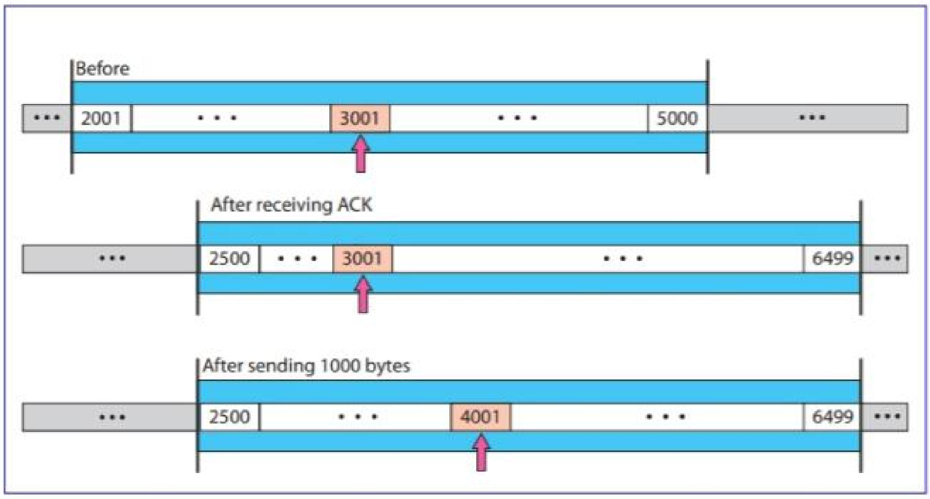
A client uses TCP to send data to a server. The data are 16 bytes. Calculate the efficiency of this transmission at the TCP level

16 bytes of data / (16 bytes of data + 20 bytes of TCP header) $\approx$ 0.44 or 44 percent

A TCP connection is using a window size of 10,000 bytes, and the previous acknowledgment number was 22,001. It receives a segment with acknowledgment number 24,001 and window size advertisement of 12,000. Draw a diagram to show the situation of the window before and after.



A window holds bytes 2001 to 5000. The next byte to be sent is 3001. Draw a figure to show the situation of the window after the following two events.
a. An ACK segment with the acknowledgment number 2500 and window size advertisement 4000 is received.
b. A segment carrying 1000 bytes is sent.



The following is a dump of a UDP header in hexadecimal form:

`06 32 00 0D 00 1C E2 17`

What is the

(a) Source port number
(b) Destination port number
(c) Total length of the UDP
(d) Length of tha data
(e) Considering that an IP frame can have a maximum total length of 65 535 bytes, what is the maximum length of the data in a UDP frame?

The UDP header has four parts, each of two bytes. That means we get the following interpretation of the header.

(a) Source port number $= 0632_{16} = 1586$

(b) Destination port number $= 000D_{16} = 13$

(c) Total length $= 001C_{16} = 28$ bytes

(d) Since the header is 8 bytes the data length is $28 - 8 = 20$ bytes.

(e) The IP header is minimum 20 bytes, which gives the maximum payload 65515 bytes. To fit a UDP frame in this with header of 8 bytes we get data $65515 - 8 = 65507$ bytes.

A client uses UDP to send data to a server. The data are 16 bytes. Calculate the efficiency of this transmission at the UDP level (ratio of useful bytes to total bytes).

Data are 16 bytes, length of UDP header is 8 bytes, so the ratio is $\frac{16}{16+8} = \frac{2}{3}$.

A client uses TCP to send data to a server. The data are 16 bytes. Calculate the efficiency of this transmission at the TCP level (ratio of useful bytes to total bytes, assume no options).

Data are 16 bytes, length of TCP header (no options) is 20 bytes, so the ratio is $\frac{16}{20+16} = \frac{4}{9}$.