Question 1:

(a) VoIP uses the Internet, which is a packet-switching network. In packet switching, there is no resource reservation and the available network bandwidth is shared between multiple communication sessions. Therefore, VOIP services can't guarantee the same quality of service throughout the duration of the call.

(b) POTS is a circuit-switched network. In circuit switching, a dedicated circuit is set up between the two communicating end points for the duration of the call and adequate resources are reserved. These resources are dedicated to the call and are not shared with any other call. Therefore, POTS can guarantee the same quality of service throughout the duration of the call.

Question 2:

(a) - processing delay: look at destination address and determine output link.
- queueing delay: time packet are queued at routers waiting to be serviced.
- transmission delay: packet length (bits) /link bandwidth (bps).
- propagation delay: time it takes for data to propagate between 2 nodes.

(b) BDW = link bandwidth * latency or round trip time. The unit used to express the BDP product can be bits, bytes, packets, etc.

(c) It measures the amount of data that can be in transit inside the network at any given time. Therefore, it is a way to measure network capacity.

(d) It is the rate (e.g., bits/sec) at which data is transferred between sender and receiver over a period of time.

(e) It is the link on end-end path that constrains the end-end throughput. In other words, it is the link on the end-to-end path that has the least capacity.

(f) average throughput = min{channel bandwidth (bits per second on the end-end path)
= min{10Mbps, 100Mbps} = 10Mbps

(e) Since each user user gets a fair share of the networks capacity, the average throughput is divided by all 20 of them. Therefore average throughput = 10Mbps/20

Average throughput= min{channel bandwidth (bits per second on the end-end path)/ number of users = min {10Mbps, 100Mbps} /20 = 0.5 Mbps = 500 Kbps

Question 3:

Let's say a user wants to retrieve example.com

(1) on the user's host: If the page for example.com has already been cached on the user's host, the browser satisfies the user's request without involving the cache at the user's institutional network or the one at the content provider's network. Therefore, it reduces the time

the user gets the object back. It also decreases the load on the institutional network and on the Internet.

(2) on the user's institutional network: if example.com is cached on the institutional network, the user's request will be serviced by the institutional cache. This will again decrease the time to get the object to the user; while it will not decrease the load on the institutional network, it will decrease Internet traffic load.

(3) on the content provider's network: the cache located at the user's institutional network will act as a client to satisfy the original http request and send the http request to the cache located at the content provider's network. The response in this scenario will likely take longer comparing to scenarios 1 and 2. It may improve load on the content provider's network.

Question 4

(a) propagation delay: length of physical link/ propagation speed in medium

$= 24{,}000km / 24 * 10^7 m/s$.

(b) U_{sender} = (transmission time)/(transmission time + RTT)

transmission time = packet length (bits) /link bandwidth (bps)
RTT = propagation delay * 2

(c) Usender*10

(d) Two sequence numbers for scenario (b). For scenario (c), we need 11 sequence numbers (i.e., window size +1) if we assume in order delivery.

Question 5 :
RTT_{CP}: The round trip time between the client and the proxy server
RTT_{SP}: The round trip time between the proxy server and the origin server

Assumption 1: TCP handshake NOT included in the retrieval time

a) Non-persistent HTTP:

- With 100% miss and 5 objects: response time = $5RTT_{CP} + 5RTT_{PS}$
- With 100% hit and 5 objects: response time = $5RTT_{CP}$

The average time to retrieve the main page plus 4 embedded objects:

$0.40(5*(RTT_{CP} + RTT_{SP}))+ 0.60(5*RTT_{CP})$

b) Persistent HTTP:

- With 100% miss and 5 objects: response time = $5RTT_{CP} + 5RTT_{PS}$
- With 100% hit and 5 objects: response time = $5RTT_{CP}$

The average time to retrieve the main page plus 4 embedded objects:

$0.40(5*(RTT_{CP} + RTT_{SP}))+ 0.60(5*RTT_{CP})$

Assumption 2: TCP handshake included in the retrieval time

a) Non-persistent HTTP:
- With 100% miss and 5 objects: response time = $5RTT_{CP}$ (TCP connection) + $5RTT_{CP}$ (HTTP requests)+ $5RTT_{PS}$ (TCP connection) + $5RTT_{PS}$ (HTTP requests)
- With 100% hit and 5 objects: response time = $5RTT_{CP}$ (TCP connection) + $5RTT_{CP}$ (HTTP requests)

The average time to retrieve the main page plus 4 embedded objects:

$0.40(5*(2RTT_{CP} + 2RTT_{PS}))+ 0.60(10RTT_{CP})$

b) Persistent HTTP:
- With 100% miss and 5 objects: response time = $1RTT_{CP}$ (TCP connection) + $5RTT_{CP}$ (HTTP requests)+ $1RTT_{PS}$(TCP connection) + $5RTT_{PS}$(HTTP requests)
- With 100% hit and 5 objects: response time = $1RTT_{CP}$ (TCP connection) + $5RTT_{CP}$ (HTTP requests)

The average time to retrieve the main page plus 4 embedded objects:

$0.40(6RTT_{CP} + 6RTT_{PS}) + 0.60(6*RTT_{CP})$

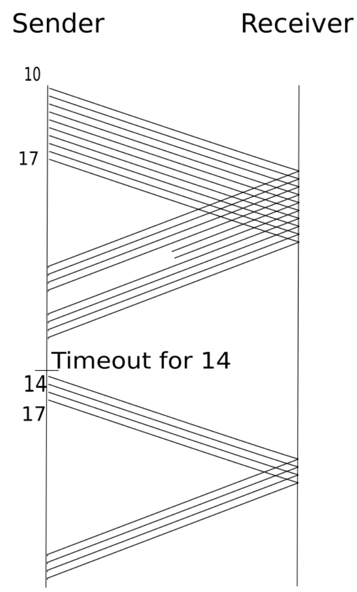
Question 6:

a) Sequence of steps:
- Host1 requests tucano.br's NS to resolve www.pbs.org.
- Tucano.br's NS requests contacts the DNS root NS
- The DNS root NS responds with address for DNS NS for .org TLD
- Tucano.br's NS requests www.pbs.org to .org DNS NS
- .org DNS NS responds with address for the DNS NS for pbs.org
- Tucano.br's NS requests www.pbs.org's IP address from the pbs.org DNS NS
- pbs.org DNS NS responds with www.pbs.org's IP address
- Tucano.br's NS forwards www.pbs.org address to host1

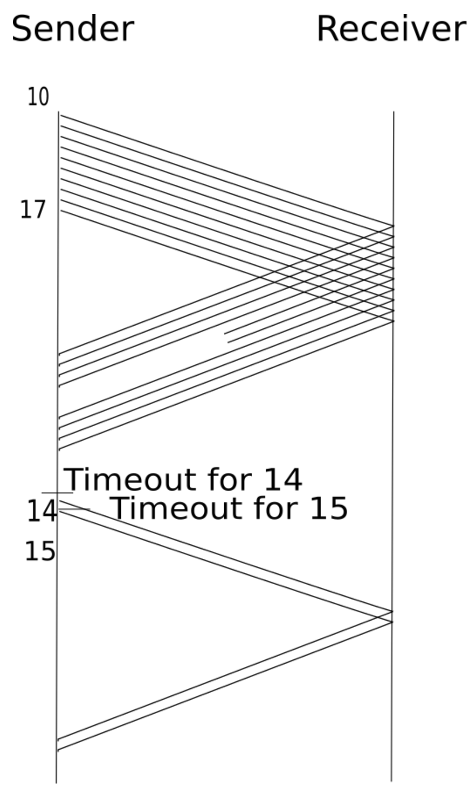
b) Sequence of steps:
- Host1 requests tucano.br's NS to resolve www.pbs.org.
- Tucano.br's NS requests www.fordfoundation.org to .org DNS NS server
- .org DNS NS server responds with address for the DNS NS for fordfoundation.org
- Tucano.br's NS requests www.fordfoundation.org's IP address from the fordfoudation.org DNS NS
- fordfoundation.org DNS NS responds with www.fordfoundation.org's IP address
- Tucano.br's NS forwards www.fordfoundation.org address to host1

Question 7:

a) Yes, because the sender can send packets while it waits for the acknowledgements of other packets.
b) For Go-Back-N we need 9 sequence numbers, while for Selective-Repeat we would need 16 sequence numbers. In either case 4 bits would be enough, as those allow us to represent 16 different values.
c) Yes, because it can have up to 8 unacknowledged segments at the same time, and it only has 5 at the moment. It can then sends segments 5, 6, and 7.
d) Segments 14, 15, 16, and 17 will be retransmitted, because Go-Back-N retransmits all segments starting from the first loss.



f) Only segments 14 and 15, because in Selective-Repeat the receive can keep packets that arrive out of order and only require retransmission of the ones that didn't arrive.



g) Cumulative ACKs can be be used for the Go-Back-N, because the protocol only needs to maintain information about the last packet that was received before losses. For the Selective Repeat Cumulative ACKs by themselves aren’t enough to provide information to avoid retransmitting packets.

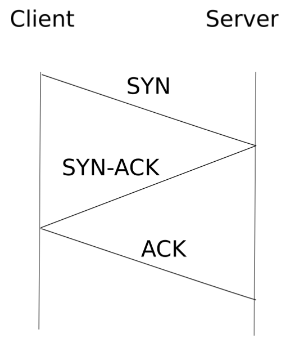
h) Selective-Repeat receivers must keep track of packets received out of order on the receiver, while the sender must keep track of timers for each packet sent.
The Go-Back-N receiver, in contrast, discards packets that arrive out of order and only keeps a timer for the oldest un’acked packet, since all packets will be retransmitted if that packet was lost.

Question 8:

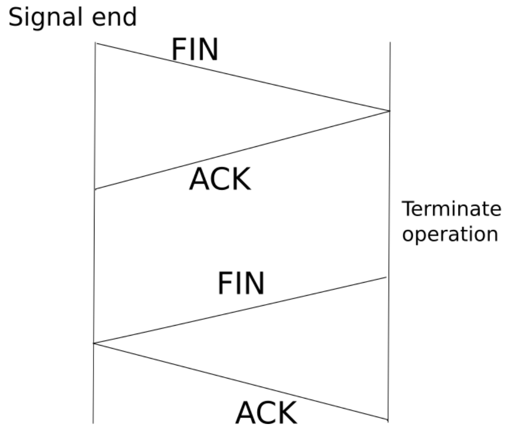
a) No, because UDP is a connectionless transport protocol.
b) UDP uses the checksum to detect transmission errors on packets as they are received. When the sender prepares the UDP segment, it calculates a checksum based on the data to be transmitted and adds it to the UDP segment header. When the segment is received, the receiver calculates the checksum of the received data and compares it to the checksum received. If they differ, then the packet is discarded.

Question 9:

a) TCP uses the three-way handshake. It consists of a message sent by the connection starter (SYN), a confirmation sent by the other part (SYN-ACK), and an acknowledgement sent by the starter for the confirmation (ACK).



b) At least 180ms.
c) Because it is important to make sure that all packets were received before ceasing to receive packets. It is also important to reclaim resources.

d) 

Question 10:

a) Sequence numbers: allow packets to be ordered when the arrive, despite out-of-order arrivals and losses.

Retransmission of packets, to ensure arrival of packets that might have been lost.

Checksum, to ensure that transmitted packets are error-free.

Feedback to check whether data has been received at the receiver.

Retransmission timer to trigger a potential packet loss.

b) TCP uses a smooth RTT because RTT can vary greatly throughout a connection, so the smoothing is used to avoid adapting to short-lived transient behavior.
c) It should be closer to 0. Since we know that the conditions do not change very often, we want favor the general trend that we have observed, instead of the very last measurement, which could not be representative.