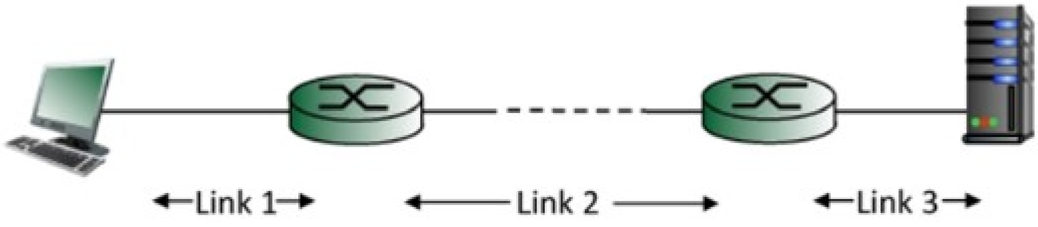
# CS3201 mid-term exam solution

Computer Networks (City University of Hong Kong)

1. Consider the connection between the following two hosts.




Suppose that we upgrade the bandwidth of Link 3 but do not change any other parameters of the connection (i.e., the routers and Links 1 and 2 remain unchanged). Mark the delay(s) that will be affected by this upgrade: Queueing delay and transmission delay

2. What is DNS used for?
   To resolve the IP address of hostnames and to provide multiple names to the same machine

3. Give an application example that it may prefer to use UDP instead of TCP and why? Moreover, Give an application example that it may perfer to use TCP instead of UDP and why?
   UDP: Multi-media data such as skype communication (UDP), since losing data or lower resolution is acceptable.
   TCP: Bank account transactions, ftp files (TCP), reliable, and in order data transfer, no loss is allowed.

4. Web caches: can significantly reduce the response time during web browsing; Web caches may reduce the traffic intensity on a network's access link; The local web proxy is a client, and it is also a server.

5. (a) Difference between SMTP and a mail-access protocol such as IMAP: SMTP is used to push mails from the client to the mail server and communicate between email servers, while IMAP is used to fetch mails from a user mailbox on the mail server.

   (b) If a person sends an email to someone else, SMTP would be used.

   (c) If a person wants to read emails from his/her mailbox, IMAP should be adopted.

6. Consider file distributions in the context client-server and of P2P architectures. There is a server hosting a file, and we would like to distribute the file to a group of N peers. Mark all true statements:

   The minimum distribution time of a client-server architecture scales linearly with the number of clients that want to obtain the file; Once the number of clients N becomes large, a P2P architecture is likely to outperform the client-server architecture.

7. Consider the following two arrival patterns of packets at a router:
   - (A) 3N packets arrive periodically. That is, the first N packets arrive simultaneously in the 1st timestep, and no other packets arrive during the N-1 following timesteps, then after the N timesteps, the next N packets arrive simultaneously, and after 2N timesteps, the last N packets arrive simultaneously again.

- (B) There is a sequence of 3N packets with one packet arriving every second timestep, where forwarding (or relaying) each packet at the router takes one timestep.

Mark all true statements:

The traffic intensity is exactly 1 for (A); (A) will incur a larger average queuing delay compared to (B).

8. In the context of HTTP protocol, mark all false statements:
   The HTTP protocol requires both the client and the server to keep track of several variables such as rcv_base. Persistent HTTP can achieve a speedup by bypassing the transport layer protocol; When pipelining is used, a server will can pack multiple objects into the same HTTP response message.

9. Suppose that we use 8-bit sums instead of 16-bit sums to compute the UDP checksum. Consider the following scenario of a segment transfer between a sender and a receiver:

| Transmitted by the sender | Received by the receiver |
|---|---|
| 11011011 (byte 1)<br>11011101 (byte 2)<br>10111001 (checksum) | 11011101 (byte 1)<br>11011011 (byte 2)<br>10110101 (checksum) |

Can the receiver detect an error in the segment transmission?

No, as the receiver cannot detect the error because 2 bits flipped in the same column, and the sum will be the same as the one without the 2 bits flipping.

10. Difference between the Go-back-N protocol and Repeat-Selective protocol:
    GBN: there is only one timer for each sliding window, the ack is always the last successful one without out of the order. All received packets need to be retransmitted if some packet prior to these packets is missed, wasting lots of bandwidth and resulting in longer delays.
    SR: there is a timer for each packet in the sliding window, acknowledge each successful received packet. Only failed packets are retransmitted, but all out of order packets received in the receiver must be buffered, the protocol needs more timer in the sender side and more storage resource in the receiver side.

11. Suppose that 3 bits are used to represent sequence numbers in the Go-Back-N protocol.
    (a) What is the maximum sender window size such that the protocol works correctly?
    Since maximum sender windows size = 8, which means not larger than 8.
    (b) List all sequence numbers in the sender window.
    #seq 0,1,2,3,4,5,6,7

12. Assume that there are three routers R1, R2 and R3 between Alice's host and Bob's host. The bandwidth of the first link, i.e., from Alice's host to router R1, is B1 bits/sec. The second link connecting router R1 to router R2 has a bandwidth B2 bits/sec, the third link connecting router R2 to router R3 has a bandwidth of B3 bits/sec Bob, and the last link connecting router R3 to Bob has a bandwidth of B4 bits/sec.

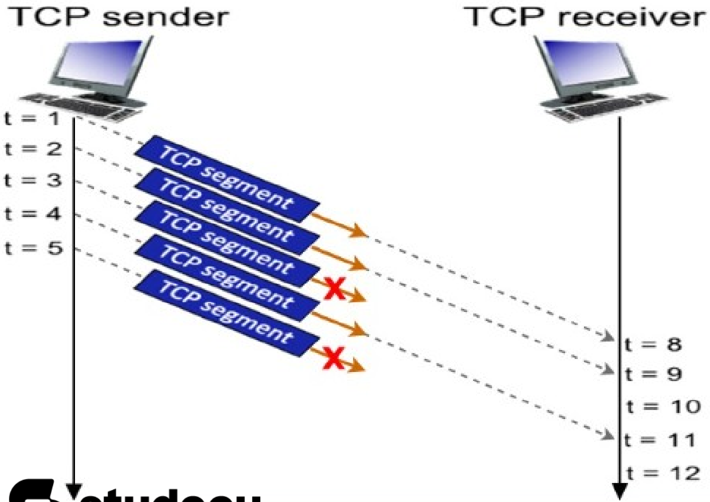
(a) What is the end-to-end delay for a packet of L bits that is sent from Alice's host to Bob's host? assuming that queueing, nodal, and propagation delays are ignored.
end-to-end throughput = L/B1 + L/B2 + L/B3+L/B4.

(b) Give a formula to calculate the throughput of the connection between the two hosts.
throughput is min{B1, B2, B3, B4}.

13. Suppose that hosts A and B communicate with each other using the TCP protocol. What is the size of the TCP send window for host A if the value of rwnd (receiver window) advertised by host B is 1500 bytes, the value of cwnd (congestion window) is 2500 bytes, and assuming the link between A and B has a bandwidth of 4500 bytes/sec?
As the TCP sender window is determined by the receiver window through its ACK advertises rwnd size, in this case, it is 1500, thus, the send window size is 1500, otherwise, the packets will be lost due to exceeding the buffer size in the receiver side B.

14. Suppose that there are six packets arrived at a receiver, which contain the following address information, respectively
Source: 221.177.0.7, port: 8158;
Destination: 183.168.20.5, port: 1000
Source: 221.177.0.12, port: 9600;
Destination: 183.168.20.5, port: 1000
Source: 221.177.0.7, port: 9600;
Destination: 183.168.20.5, port: 1000
Source: 221.177.0.12 port: 9600;
Destination: 183.168.20.5, port: 1300
Source: 220.177.0.12 port: 9600;
Destination: 183.168.20.5, port: 1200
Source: 221.177.0.12, port: 9600;
Destination: 183.168.20.5, port: 1000
Which of the six packets will be delivered to the same socket?
(a) Assume UDP is used. → 1,2,3,6
(b) Assume TCP is used. → 2,6

15. Consider the figure below in which a TCP sender and receiver communicate over a connection in which the sender-to-receiver segments may be lost. The TCP sender sends an initial window of 5 segments. Suppose the initial value of the sender-to-receiver sequence number is 20, and the number of bytes contained in the first 5 segments are 125, 19, 231, 1200 and 53, respectively. The delay between the sender and receiver is 7 time

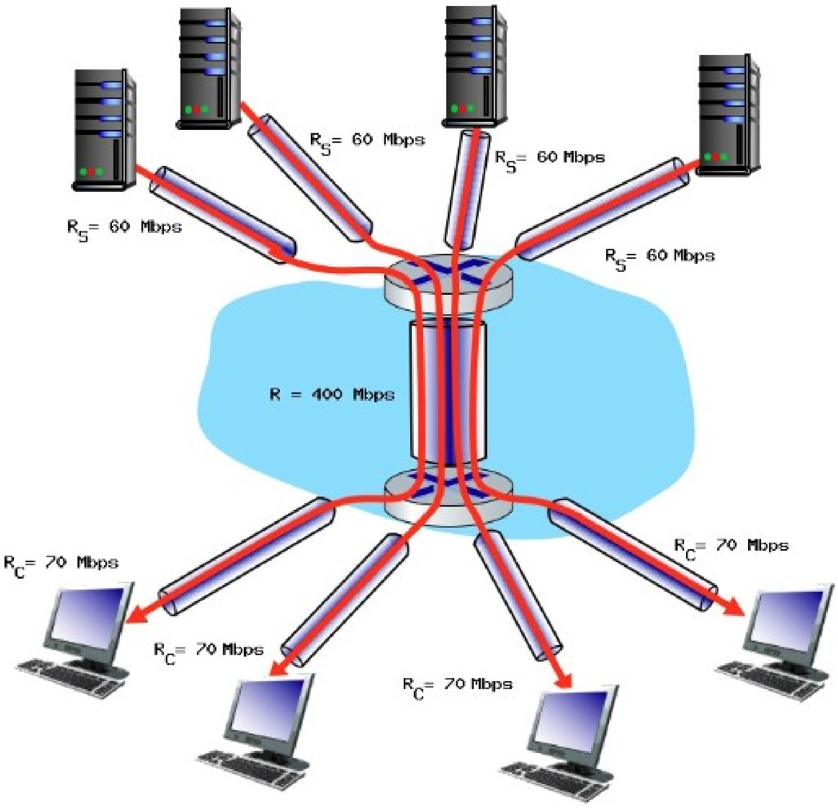


units, and so the first segment arrives at the receiver at t=8. As shown in the figure below, 2 of the 5 segment(s) are lost between the sender and the receiver, i.e., segments 3 and 5 are lost.

(a) Give the ACK number the receiver sends in response to each of the segments.
   ACK01: 145; ACK02: 164; ACK04: 164;

(b) Give the sequence number associated with each of the 5 segments sent by the sender.
   S1: 20; s2: 20+125 = 145; s3: 145+19 = 164; s4: 164+231 = 395; s5: 395+1200 = 1595;

16. Consider the scenario shown below, there are four different servers connected to four different clients over four three-hop paths, where the four pairs share a common middle hop with a transmission capacity of R = 400 Mbps. Each of the four links from the servers to the shared link has a transmission capacity of RS = 60 Mbps, while each of the four links from the shared middle link to each client has a transmission capacity of RC = 70 Mbps.




- Which link is the bottleneck link in any routing path between a server and a client?
  each of the four routing paths from the shared middle link has a transmission capacity of RC = 40 Mbps is the bottleneck link.
- What is the maximum throughput (in Mbps) for each of four client-to-server pairs, assuming that the transmission rate of the middle link is fairly shared (divides its transmission rate equally) among 4 routing paths?
  The maximum throughput is min{85, 280/4, 40}=40Mbps