Introduction to Computer Networks **Name:** ______________________

Fall 2021

Homework 1 (Due: 11/08/2021) **ID:** ______________________

This homework contains 9 questions. The deadline is on Nov. 08 (Tue) at 23:59. Please submit your answers to new E3.

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1. (4 points) **Access Network:** ADSL stands for asymmetric DSL. (1) Please explain what does *asymmetric* mean. (2) Why does an ISP prefer to provide ADSL, instead of symmetric DSL?

Solution:
ADSL: Asymmetric Digital Subscriber Line.

Users usually need a higher downlink bandwidth than the uplink bandwidth.

2. (8 points) **Packet Loss:** (1) Please give two different root causes of packet losses. (2) In TCP, what mechanisms are designed to resolve the two issues?

Solution:
Router buffer overflow and Destination buffer overflow.

Congestion control and flow control.

3. (10 points) **Packet switching:** (1) Explain what is the difference of packet switching and circuit switching. (2) Explain what does *store and forward* mean. (3) Give two advantages and disadvantages of packet switching. (4) Explain why packet switching cannot provide performance guarantee.

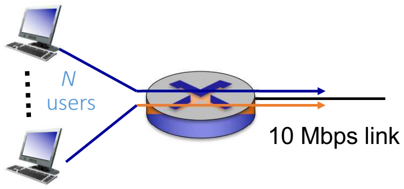
Solution:
(1) Packet switch allows users to compete for resources, while circuit switching reserves a portion of resources for every user.

(2) A switch/router should received the entire packet and then forward the packet to the next hop.

(3) pros: high utilization, fair allocation, simple, more efficient support more users; cons: no performance guarantee, loss and delay, congestion, not suitable for real-time applications

(4) The available bandwidth is determined by how many users compete for the capacity.

4. (8 points) **Bandwidth sharing:** Consider the following scenario, where the outgoing link of the switch is 10 Mb/s and shared by users with packet switching. Assume that each user becomes active for only 10% of time and generates traffic of 500 Kb/s when it is active.



(a) (4 points) If we want to make sure that each user can get a satisfactory service (i.e., rate no lower than 500 Kb/s) with a probability larger than 0.01, at most how many users (denoted by N) can join the system simultaneously? (**Note:** You only need to show your derivation (equation). No need to solve the final result.)

> **Solution:**
> The maximal number of users if $100/0.5 = 200$.
> Find the largest possible x such that
>
> $$\sum_{n=0}^{200} C_n^x (0.1)^n (0.9)^{x-n} \geq 0.01$$
>
> (Just need to write the equation)

(b) (4 points) If 1,000 users join the system and each is active for only 10% of time, then what is the probability that users CANNOT get a satisfactory service? (**Note:** You only need to show your derivation (equation). No need to solve the final result.)

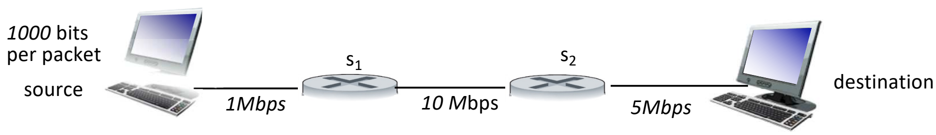
> **Solution:**
>
> $$1 - \sum_{n=0}^{200} C^1 000_n (0.1)^n (0.9)^{1000-n}$$
>
> (Just need to write the equation)

5. (16 points) **Delay:**

   (a) (8 points) (1) List four different delay sources of the end-to-end delay and give a *formal definition* for each delay source. (2) Which types of delay sources are constant? (3) Which type of delay sources is typically hard to predict? Why?

   Solution:
   (Only for reference)
   (1) Nodal processing delay (process time in a router), queueing delay (the latency from a packet is enqued to it is dequeued), transmission delay (the time required to push every bit of a packet to the outgoing link) and propagation delay (the time a packet traverses through a link)
   (2) Constant: transmission and propagation delay Variable: nodal and queueing
   Note: Nodal processing delay depends on the task complexity. If a router implements a fixed task for every packet without any other background process, then nodal processing delay can be constant.
   (3) queueing delay, since it depends on the dynamic number of competing users.

   (b) (4 points) Consider the following scenario, where switches s_1 and s_2 only serve a single flow. Assume there is no propagation delay and nodal processing delay. If a packet of 1,000 bits is sent by the source at time $t = 0$, when does s_2 start to forward the packet? (Show your derivation and answer)



   Solution:
   $\frac{1000}{1} + \frac{1000}{10} = (1000 + 100)(\mu s) = 1.1(ms)$

   (c) (4 points) Consider the same network. Assume that each switch has an infinitely large buffer. If the source now sends 10 back-to-back packets of length 1,000 bits time 0, what is the time that the destination receives all the 10 packets. (Show your derivation)

   Solution:
   Since the bottleneck banwidth (the end-to-end) throughput is 1MBps, it takes $1000 * 10/1 = 10000$ (μs) to send all the bits to the first link.
   Since the second and third links are not bottleneck, the packets do not have to wait in $s1$ and $s2$ but require additional $1000/10+1000/5 = 300$ (μs) to reach the destination.
   The total latency is $100 + 0.3 = 100.3$ (ms).

6. (16 points) **Application layer:**

   (a) (2 points) Explain what is the difference between *host-to-host* and *process-to-process* communications.

Solution:
host-to-host communication means that a host sends packets to another host.

Process-to-process communication means that a process of a host sends segments to a process of the same or another host. A host may include multiple processes.

(b) (4 points) How to distinguish different processes in the same host? What is the unique identity of a process?

Solution:

Use different port numbers. ID = IP address + port number.

(c) (4 points) UDP is a connectionless and unreliable protocol. (1) Give an example application that uses UDP as the transport layer protocol. (2) Explain what kind of applications are suitable for using UDP.

Solution:
DNS

An application that does not require reliable transmissions but need a very short response time.

(d) (6 points) (1) Explain why proxy servers are *less popular* when we have CDN. (2) Assume each CDN server has a limited storage size. If you are the CDN operator, what is the strategy you would adopt to assign a subset of content objects to a CDN server. Why?

Solution:
(1) CDN has duplicated content objects in distributed servers close to the end users.

(2) cach the content objects based on the popularity of the objects based on the preference of local users. By doing this, a user can download the objects of interest from a nearby server.

7. (10 points) **HTTP:**

   (a) (4 points) Explain what is the difference between persistent HTTP and non-persistent HTTP. Give the pros and cons of two mechanisms.

   > **Solution:**
   > persistent: a client only creates a single TCP connection to retrieve all the HTTP objects
   >
   > non-persistent: a client should create an individual TCP connection for downloading every object.
   >
   > non-persistent takes a longer handshaking latency since it should create more TCP connections.

   (b) (6 points) (1) Consider an HTTP client that wants to retrieve a Web page including 3 images. Assume the Web server adopts *persistent HTTP*, which needs 1 RTT to build every TCP connection and 1 RTT to request for a content object. Suppose the server supports at most 2 parallel TCP connections. How many RTTs are required to download the entire Web page? (2) Now, suppose at most 4 parallel TCP connections are allowed. How many RTTs are required?

   > **Solution:**
   > 1 (build connection) + 1 (request for html files) + 1 (request for 3 images) = 3 RTT
   >
   > the same 3 RTT

8. (12 points) **Reliable data transfer:**

   (a) (5 points) (1) Explain what is the difference between bit errors and packet losses. (2) In rdt, what mechanism is used to resolve bit errors of a data packet? (3) In rdt, what mechanism is used to resolve bit errors of a feedback packet? (4) In rdt, what mechanism is used to resolve packet losses?

   > **Solution:**
   > errors: receive the pacekt but the packet is incorrect; loss: the packet is completely lost
   >
   > NACK and retransmission
   >
   > retransmission
   >
   > timeout and retransmission

   (b) (4 points) In the following cases, which cases could trigger unnecessary retransmissions? (multiple choice question)

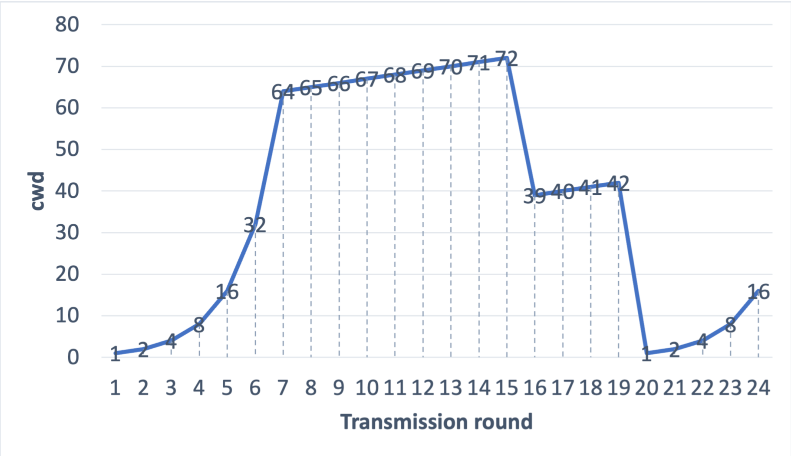
   1. The sender receives a correct NACK from the receiver.
   2. The sender misses the ACK from the receiver.
   3. The sender receives the ACK from the receiver after the timeout.
   4. The sender receives a corrupted ACK from the receiver.

Solution:
2, 3, 4

(c) (3 points) Consider a path between a source and a destination with the bandwidth of 10Mb/s. Assume that 10% of data packets are lost. Assume that another 10% of the data packets are received correctly but their ACK cannot be sent back to the source by the timeout. Assume all the retransmissions can be successful. Then, what is the final throughput?

Solution:
To send x raw data pacekt, we should actually send $1.2x$ packets, including retransmissions. The throughput will be $10 * 1/1.2 = 8.33Mbps$.

9. (16 points) **TCP congestion control:** Consider the following figure. Assuming TCP Reno is the protocol.



(a) (4 points) Explain what is the *design goal* of slow start and congestion avoidance, respectively.

Solution:
slow start: quickly reach the avaiable bandwidth
congestion avoidance: conservatively test the actual upper bound and

(b) (4 points) Identify the intervals of time when TCP slow start is operating.

Solution:
1-7 transmission rounds
20-24 transmission rounds

(c) (4 points) Identify the intervals of time when TCP congestion avoidance is operating.

Solution:
7-19 (or 8-19)

(d) (4 points) What is the initial value of `ssthresh` at the 5th and 18th transmission round, respectively?

Solution:
64 at 5th.

$72/2 = 36$ at 18th