Introduction to Computer Networks
Fall 2021
Homework 1 (**Due: 11/08/2021**)

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

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?

(1) Asymmetric bandwidth for upstream and downstream.
(2) As general users need higher download speed.

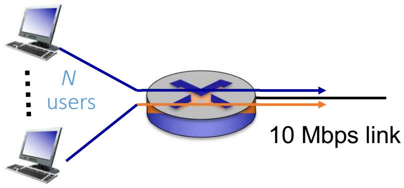
2. (8 points) **Packet Loss:** (1) Please give two different root causes of packet losses. (2) In TCP, what machenisms are designed to resolve the two issues?

(1) Router buffer overflow and destination buffer overflow.
(2) 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.

(1) packet switching is not reserved for a path while circuit switching is.
(2) store the entire packet in the router before sending it forward to the next link.
(3) it is simpler as it requires no allocation and it supports more users at the same time; however, it may lead to packet loss and more delay and it's less suitable for real-time applications.
(4) if there's lots of users are sharing the same link, we should wait in queue to transmit our packets, but if there's few users transmitting at the same time, we may get less delay, thus the performance is not guaranteed.

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.)

$$\frac{10\times10^6}{500\times10^3} = 20 \Rightarrow C_0^N\, 0.9^N + C_1^N\, 0.9^{N-1}\cdot 0.1 + \cdots + C_{20}^N\, 0.9^{N-20}\cdot 0.1^{20} \geq 0.01$$

$$\Rightarrow \sum_{k=0}^{20}\left(C_k^N\cdot 0.9^{N-k}\cdot 0.1^k\right) \geq 0.01$$

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(b) (4 points) If 1,000 users join the system and each is avtive 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.)

$$1-\sum_{k=0}^{20}\left(C_k^{1000}\cdot 0.9^{1000-k}\cdot 0.1^k\right)$$

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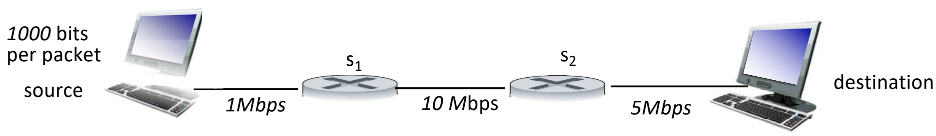
5. (16 points) **Delay:**

(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 predidct? Why?

(1) nodal processing delay : Time required to examine the packet's header and determine where to go.
queueing delay : Wait in the buffer for being transmitted onto the link.
transmission delay : Time required to push all the packet's bits into the link.
propogation delay : Time required to propagate from the beginning of the link to another end point.
(2) nodal processing delay is constant.
(3) queueing delay is hard to predict as we don't know how many user are transmitting the data and how many time we should wait.

(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)



the time s2 start to forward the packet equals to the sum of the first two transmission delay (as there's only one packet , i.e. no queueing delay), that is:
t = 1000/10^6+1000/10^7 = 0.0011(s)

(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)

the time the destination receives all the 10 packets equals to the sum of three transmission delay of the 10 packets and the queueing delay at source:
transmission delay: 1000/10^6+1000/10^7+1000/(5*10^6) =0.0013(s)
queueing delay of the last packet = the transmission time of the first 9 packets in source:
0.001*9 = 0.009
t = 0.0013+0.009 = 0.0103(s)

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

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

   host-to-host means the communication in the network layer,which is an end to end communication, while process-to-process communication is in the tranport layer, which stresses the communication of requesting for a particular service.

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

   use port number to distinguish different processes in the same host
   the unique identity of a process is "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.

   (1) DNS (domain name system)
   (2) applications that requires fast response and doesn't care about few packet loss are suitable for using UDP.

   (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?

   (1) As users request the same data to the same proxy server, it may cause conjestion, while CDN have copies of the same data in different CDN servers so that it will trace your location and give you the data from the nearest one.
   (2) separate the data of the content into small partitions and copy them respectively, then build those CDN servers in every area, so that the time requesting for data still fast and it will not exceed the limited storage size.

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.

persistent HTTP send objects over a single TCP connection while non-persistent HTTP send objects over distinct TCP connection.

persistent HTTP has lower latency as it only requires one round trip time to request for the data, while non-persistent HTTP requires two round trip time as each time it requests for the data, it must send request to the server to open the link.

(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?

(1) 1+(3+1)/2 = 3(RTTs)
(2) 1+(3+1)/4 = 2(RTTs)

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?

(1) bit errors indicates the packet is received but with some bits wrong, packet losses means some packets are not pushed into the buffer due to overflow.

(2) the receiver will send back a feedback packet, if there's bit errors(NAK), the sender will re-transmmit the data packet.

(3) add a 1-bit 0,1,0,1... sequence into the packet header, if the sequence is wrong, then discard the repeated packet.

(4) give up after timeout and then the sender will automatically retransmit the data.

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

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.

2,4

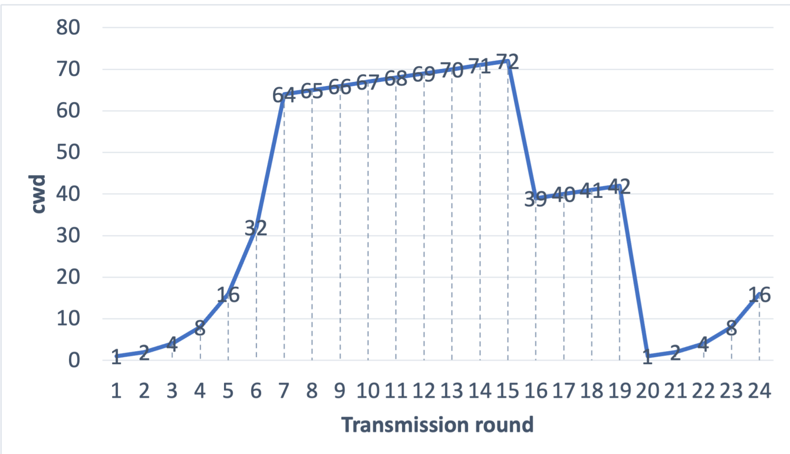
(c) (3 points) Consider a path between a source and a destination with the bandwidth of 10Mb/s. Assume that 10% of data packets are lost. Assume that another 10% of the data packets are received correctly but their ACK cannot be sent back to the source by the timeout. Assume all the retransmissions can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

assume their are x Mb datas . it requires $\frac{x}{10}$ s

and 20% of x requires retransmission , which requires $\frac{0.2x}{10}$ s

$\Rightarrow$ the average throughput = $\dfrac{x}{\frac{x+0.2x}{10}} = \frac{10}{1.2} = \frac{25}{3}$ Mb/s #

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.

slow start: let congestion window grows in a exponential way.
congestion avoidance: prevent from satuarating the available bandwidth.

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

[1,7] U [20,24]

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

[7,15] U [16,19]

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

5th round: 64
18th round: 32