CS353 Spring 2016
Homework 2
Due: April 29th at 10am

Name: ______________________________

USC-ID: ______________________________

Grade Table (for staff use only)

Question	Points	Score
1	13	
2	9	
3	20	
4	20	
5	12	
6	16	
Total:	90	

Instructions

- Submit this homework on BlackBoard in pdf format
- To submit, you can print out this document, write your answers on it, scan it and upload it. Or you can also use a pdf markup program to insert your answers into this document in the provided spaces, Or create a separate solution pdf (without the questions), each answer neatly labeled, and upload only solutions.
- If you need more space for work attach sheets at the end of the homework when you submit
- If you have questions, please ask on Piazza
- Some potentially useful information:
  1 Byte = 8 bits, 1 Mbps = $10^6 \frac{\text{bits}}{\text{sec}}$, 1 Gbps = $10^9 \frac{\text{bits}}{\text{sec}}$, 1 millisecond (or msec) = 10^{-3} sec, 1 microsecond (or μsec) = 10^{-6} sec, 1 nanosecond (or nsec) = 10^{-9}. Assume speed of light is 2.0 x 10^8 m/s (speed in optical fibre)
- **READ THIS:** In this homework there are several questions that go beyond what we covered in class. When you start reading the question and it doesnot sound familiar, just work through the problem step-by-step and the answers should come out. Some of the problems involve some basic algebra and probability. They are mainly intuitive, but if you are having trouble with these concepts please attend office hours.

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1. **Short Questions**

(a) (2 points) DHCP allows a computer to acquire a new IP address whenever it moves to a new subnet. Why is this not always enough to address the communications needs of mobile hosts

> **Solution:** A correspondent node has no way of knowing that the IP address of a mobile node has changed, and hence no way to send it a packet. A TCP connection will break if the IP address of one endpoint changes.

(b) (2 points) what is the main downside of requiring traffic destined to a mobile node to be sent first to its home agent?

> **Solution:** The home agent and the mobile node may be very far apart, leading to suboptimal routing.

(c) (3 points) Suppose a client C repeatedly connects via TCP to a given port on a server S, and that each time it is C that initiates the close.How many TCP connections a second can C make here before it ties up all its available ports in TIME WAIT state? Assume client ephemeral ports are in the range of 1024 to 5119, and that TIME WAIT lasts 60 seconds.

> **Solution:** We have 4096 ports; we eventually run out if the connection rate averages more than $4096/60 = 70$ per sec. (The range used here for ephemeral ports, while small, is typical of older TCP implementations.)

(d) (3 points) Explain the fundamental conflict between tolerating burstiness and controlling network congestion

> **Solution:** It is easier to allocate resources for an application that can precisely state its needs, than for an application whose needs vary over some range. Bursts consume resources, and are

hard to plan for.

(e) (3 points) When TCP sends a (SYN, SequenceNum = x) or (FIN, SequenceNum = x) , the consequent ACK has Acknowledgment = x + 1; that is, SYNs and FINs each take up one unit in sequence number space. Is this necessary? If so, give an example of an ambiguity that would arise if the corresponding Acknowledgment were x instead of x + 1; if not, explain why.

Solution: Incrementing the Ack number for a FIN is essential, so that the sender of the FIN can determine that the FIN was received and not just the preceding data. For a SYN, any ACK of subsequent data would increment the acknowledgment number, and any such ACK would implicitly acknowledge the SYN as well (data cannot be ACKed until the connection is established). Thus, the incrementing of the sequence number here is a matter of convention and consistency rather than design necessity.

2. An organization has been assigned the prefix 212.1.1/24 (class C) and wants to form subnets for four departments, with hosts as follows: A 75 hosts
B 35 hosts
C 20 hosts
D 18 hosts
There are 148 hosts in all.

   (a) (4 points) Give a possible arrangement of subnet masks to make this possible.

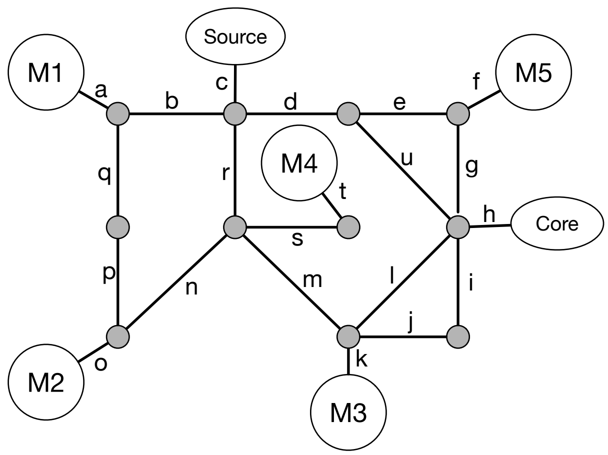
   Solution: Giving each department a single subnet, the nominal subnet sizes are 27, 26, 25, 25 respectively; we obtain these by rounding up to the nearest power of 2. For example, a subnet with 128 addresses is large enough to contain 75 hosts. A possible arrangement of subnet numbers is as follows. Subnet numbers are in binary and represent an initial segment of the bits of the last byte of the IP address; anything to the right of the / represents host bits. The / thus represents the subnet mask. Any individual bit can, by symmetry, be flipped throughout; there are thus several possible bit assignments. * A 0/ one subnet bit, with value 0; seven host bits
   * B 10/
   * C 110/
   * D 111/
   The essential requirement is that any two distinct subnet numbers remain distinct when the longer one is truncated to the length of the shorter.

   (b) (5 points) Suggest what the organization might do if department D grows to 32 hosts.

Solution: We have two choices: either assign multiple subnets to single departments, or abandon subnets and buy a bridge. Here is a solution giving A two subnets, of sizes 64 and 32; every other department gets a single subnet of size the next highest power of 2: * A 01/
001/
* B 10/
* C 000/
* D 11/

3. Multicast

Consider the following network, where a source S is sending a packet to group G, which has members m_1, m_2, m_3, m_4, m_5. Unicast routing is shortest-path, with all links having a cost of one. Multicast routing delivers a single sent packet to multiple destinations; below you will be asked to write the path a packet takes from the source to a particular destination (written as a series of links, such as k-j-i-h).



(a) Assume that the network uses DVMRP for multicast.

i. (2 points) Path traveled by packet from S to m_1
ii. (2 points) Path traveled by packet from S to m_2
iii. (2 points) Path traveled by packet from S to m_3
iv. (2 points) Path traveled by packet from S to m_4
v. (2 points) Path traveled by packet from S to m_5

(b) Assume that the network uses CBT for multicast.

i. (2 points) Path traveled by packet from S to m_1
ii. (2 points) Path traveled by packet from S to m_2
iii. (2 points) Path traveled by packet from S to m_3
iv. (2 points) Path traveled by packet from S to m_4
v. (2 points) Path traveled by packet from S to m_5

Solution: i. (2 points) Path traveled by packet from S to m1 c - b - a
ii. (2 points) Path traveled by packet from S to m2 c - r - n - o
iii. (2 points) Path traveled by packet from S to m3 c - r - m - k
iv. (2 points) Path traveled by packet from S to m4 c - r - s - t
v. (2 points) Path traveled by packet from S to m5 c - d - e - f
(b) Assume that the network uses CBT for multicast.
i. (2 points) Path traveled by packet from S to m1 c - b - a
ii. (2 points) Path traveled by packet from S to m2 c - d - u - l - m - n - o
iii. (2 points) Path traveled by packet from S to m3 c - d - u - l - k
iv. (2 points) Path traveled by packet from S to m4 c - d - u - l - m - s - t
v. (2 points) Path traveled by packet from S to m5 c - d - u - g - f

4. (20 points) Consider a flaky link where the initial transmission of a data packet is dropped if its number is prime (in other words, the initial transmissions of D2, D3, D5, D7, D11, D13 are dropped, but subsequent transmissions are ok). Note that the ACKs are cumulative and numbered according to the next expected packet (hence, A4 indicates the receipt of D1, D2, and D3). Hosts x and y are using a transport protocol with sliding window flow control with a constant window size of 5 packets and selective repeat. Three duplicate ACKs trigger a retransmission (hint: consider how many total ACKs makes for three duplicates?). Assume that the latency of the link is significantly longer than the transmission time of 5 packets and that the retransmit timeout is much longer than the RTT.

   Below, fill in the first 20 packets sent from host x (you donot need to indicate what ACKs are generated, though it may be helpful and we have entered the first few entries below). Mark which packets are retransmits due to timeouts and which are retransmits due to duplicate acknowledgements.

1.	D1	A2
2.	D2(dropped)	-
3.	D3 (dropped)	-
4.	D4	A2
5.	D5 (dropped)	-
6.		
7.		
8.		
9.		
10.		
11.		
12.		
13.		
14.		
15.		
16.		
17.		
18.		
19.		
20.		

Solution:

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1. D1 A2
2. D2 (dropped) -
3. D3 (dropped) -
4. D4 A2
5. D5 (dropped) -
6. D6 A2
7. D2 (timeout) A3
8. D7 (dropped) -
9. D3 (timeout) A5
10. D8 A5
11. D9 A5
12. D5 (timeout) A7
13. D10 A7
14. D11 (dropped) -
15. D7 (timeout) A11
16. D12 A11
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17. D13 (dropped) -
18. D14 A11
19. D15 A11
20. D11 (dupack/retransmit) A13
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5. Assume that TCP implements an extension that allows window sizes much larger than 64 KB. Suppose that you are using this extended TCP over a 1-Gbps link with a latency of 50 ms to transfer a 10-MB file, and the TCP receive window is 1 MB. If TCP sends 1-KB packets (assuming no congestion and no lost packets):

   (a) (4 points) How many RTTs does it take until slow start opens the send window to 1 MB?

   Solution: In slow start, the size of the window doubles every RTT. At the end of the ith RTT, the window size is 2^i KB. It will take 10 RTTs before the send window has reached 2^10 KB = 1 MB.

   (b) (4 points) How many RTTs does it take to send the file?

   Solution: After 10 RTTs, 1023 KB = 1 MB ? 1 KB has been transferred, and the window size is now 1 MB. Since we have not yet reached the maximum capacity of the network, slow start continues to double the window each RTT, so it takes 4 more RTTs to transfer the remaining 9MB (the amounts transferred during each of these last 4 RTTs are 1 MB, 2 MB, 4 MB, 1 MB; these are all well below the maximum capacity of the link in one RTT of 12.5 MB). Therefore, the file is transferred in 14 RTTs.

   (c) (4 points) If the time to send the file is given by the number of required RTTs multiplied by the link latency, what is the effective throughput for the transfer? What percentage of the link bandwidth is utilized?

   Solution: It takes 0.7 seconds (14 RTTs) to send the file. The effective throughput is (10MB / 0.7s) = 14.3MBps = 114.3Mbps. This is only 11.4% of the available link bandwidth.

6. Fair Queueing
   (a) Consider four flows traversing a single link with bandwidth 1Gbps. The four flows pass through a router implementing Fair Queueing before they reach the link. At their sources, the four flows are sending at the following rates:
   - Flow A: 100Mbps
   - Flow B: 200Mbps
   - Flow C: 400Mbps
   - Flow D: 500Mbps

   Assuming there are no other flows in the network, how much bandwidth does each flow get (in Mbps) after passing through the FQ router?

   i. (2 points) Flow A:

   ii. (2 points) Flow B:

   iii. (2 points) Flow C:

   iv. (2 points) Flow D:

   Solution: i. (1 point) Flow A: 100 Mbps ii. (1 point) Flow B: 200 Mpbs iii. (1 point) Flow C: 350 Mbps iv. (1 point) Flow D: 350 Mbps

   (b) Now assume that the router doesn't use fair queueing, but instead drops each flow at a specific rate to ensure that the resulting rates of undropped packets achieve the fair share rates above. What are the dropping rates (expressed as a fraction of dropped packets)? More specifically, for a given flow, what probability of dropping p will restrict that flow's throughput to its fair share?

   i. (2 points) Flow A:

   ii. (2 points) Flow B:

   iii. (2 points) Flow C:

   iv. (2 points) Flow D:

Solution: i. (1 point) Flow A: 0 ii. (1 point) Flow B: 0 iii. (1 point) Flow C: .125 iv. (1 point) Flow D: .3