CS132/EECS148 - Instructor: Karim El Defrawy
Midterm – Spring 2013
Time: 1 hour
May 2nd, 2013

Total Points: 25
Attempt all problems.

Problem #1: (5 points, ½ point each)
Choose only one answer. You will not receive any points if you choose multiple answers.

1. In the OSI networking stack, routing is performed by the
   a) Session Layer  b) Network Layer  c) Transport Layer  d) Data-Link Layer  
   e) None of these choices

2. Four bits are used for sequence numbering in a sliding window protocol used in a computer network. What is the maximum window size?
   a) 4  b) 8  c) 15  d) 16  e) None of these choices

3. Which of the following TCP/IP protocols is used for remote terminal connection services?
   a) TELNET  b) FTP  c) RARP  d) UDP  e) None of these choices

4. Which protocol is used for sending email on the Internet?
   a) SMTP  b) SMPP  c) SNMP  d) FTP  e) None of these choices

5. Which one of the following uses the greatest number of layers in the TCP/IP stack?
   a) Switch  b) Repeater  c) Router  d) End Host  e) None of these choices

6. What is the default port number for HTTP?
   a) 21  b) 80  c) 25  d) 8080  e) None of these choices

7. A basic telephone network is an example of
   a) Packet Switching  b) Cell Switching  c) Circuit Switching  d) Message Switching  
   e) None of these choices

8. Which one of the following is used to communicate between different networks?
   a) ADSL  b) HDSL  c) Gateway/Router  d) Modem  e) None of these choices
9. Which of the following OSI layers corresponds to the layer where TCP operates in the TCP/IP stack?
   a) Network Layer  b) Data-Link Layer  c) Session Layer  e) Transport Layer  
e) None of these Choices

10. TCP uses the following mechanism for increasing the congestion window size (cwnd):
    a) additive increase additive decrease  b) additive increase multiplicative decrease  
c) multiplicative increase additive decrease  d) multiplicative increase multiplicative decrease  
e) None of these choices

**Problem #2: (6 points, 1 point each)**
Determine whether each of the following statements is True (T) or False (F). No explanation is necessary; partial credit will not be awarded.

1. All nodes connected to the Internet must implement UDP.  
   ( F )

2. Media access control is a function of the data-link layer.  
   ( T )

3. Forward Error Correction (FEC) can be more efficient than Automatic Repeat Request (ARQ) in a broadcast environment with many receivers.  
   ( T )

4. End hosts implement transport layer protocols but routers do not.  
   ( T )

5. In a packet switched network, all packets belonging to the same transport layer session must follow the same route.  
   ( F )

6. Peer-to-Peer (P2P) systems are scalable because available resources increase with a quadratic factor with each new member that joins the P2P system.  
   ( F )

**Problem #3: (6 points)**
Answer the following questions regarding the Application Layer.

(1 points) 1. Briefly describe what HTTP is and sketch its operation using a simple figure (i.e., the typical messages exchanged during operation of HTTP).

The HyperText Transfer Protocol (HTTP) is a stateless application layer protocol. HTTP is used to transfer web content between a browser application (client) and an HTTP server. All web content is identified by a URL. HTTP is a request response protocol that uses TCP for assured delivery. HTTP uses ASCII encoded headers. The HTTP GET command retrieves HTML files and other objects. The GET header includes the URL of the object and other optional fields such as capability, language, and so on. The response includes a response header with a response code (code 200 is OK and 404 is page not found). Other commands include POST and HEAD.
2. We say that FTP has out-of-band control. What do we mean by that?

We mean that commands and data flow across different TCP connections.

3. What is the minimum value for the timeout of a reliable transmission protocol? Why is there a minimum value?

One RTT; anything shorter would timeout before the ACK had a chance to arrive.

4. What is the difference between congestion control and flow control?

Congestion control prevents overrunning buffers in the network, while flow control prevents overflowing the end hosts (or receivers).

5. Consider 5 users (call them A, B, C, D, and E) that have connections on a single 10 Mb/s link. Connections last for several minutes or perhaps even hours (i.e., they are bounded in time). Assume that user A requests 2 Mb/s for its connection, B requests 1 Mb/s, C requests 3 Mb/s, D requests 4 Mb/s, and E request 10 Mb/s. Note that the total requested bandwidth (20 Mb/s) exceeds the link bandwidth (10 Mb/s). Describe at least three ways of fairly allocating the 10 Mb/s to the five users. Carefully discuss/describe what is “fair”. Carefully discuss the trade-offs between your different definitions of “fair”. Multiple definitions of fair are possible.

One definition of fair can be to allocate resource proportional to need. So,
A gets (2/20)(10) Mb/s = 1.0 Mb/s
B gets (1/20)(10) = 0.5
C gets (3/20)(10) = 1.5
D gets (4/20)(10) = 2.0
E get (10/20)(10) = 5.0

This definition of fairness rewards over-requesting (cheating) to get a larger share against other users. We note that in this example, no user is 100% happy (i.e., has the full allocation that it requested).
Max-min allocation is another definition of fairness. So, all get 2 Mb/s in a first round with A satisfied, B satisfied and with 1 Mb/s to give back, C, D, and E all with bandwidth still needed. The 1 Mb/s is then allocated between C, D, and E (each get 1/3 Mb/s) such that in the end we have:
- A gets 2 Mb/s
- B gets 1 Mb/s
- C gets 2.33 Mb/s
- D gets 2.33 Mb/s
- E gets 2.33 Mb/s

This definition of fairness does not reward cheating (asking for more than you need) and does also allow some users to be 100% happy. This definition of fairness, however, does not fully consider need (i.e., the user requesting the most does not necessarily get the most).

Finally, a random allocation of the entire link to a user can be long term fair, but inefficient (and not short term fair either). So, first give A all its wants. When A is done, give B all it wants, and so on. This method also does not reward cheating since a connection simply uses all it needs and requesting an excess has no benefit.

(8 points) Problem #4:

a) (5 points) Consider a TCP Reno (i.e., one that implements fast retransmit and recovery) flow that has exactly 50 segments to send. Assume that during the transmission, exactly four packets are lost: the 4th, 5th, 22nd, and 48th; no other losses occur. Using the graph below, plot the evolution of the congestion window as each segment is sent. You may measure cwnd and time in whatever units you find convenient. (Do not inflate the window due to duplicate ACKs.)
b) **(3 points)** Label your plot above indicating the regions where slowstart, timeout, congestion avoidance, and fast retransmit occur.

Timeouts, shown in red, occur after the loss of segments 4, 5, 22 and 48. Slowstart periods, depicted in blue above, are until the first loss, and one RTT after each timeout. Everything else is congestion avoidance.

Note 1: I picked 2 RTTs for timeout value, you could use anything 1 RTT or greater.

Note 2: According to the RFC, TCP Reno should drop CWND to 1MSS after a timeout. In this problem to make it simpler, CWND dropped by half in both timeout and 3 duplicate ACKs (fast retransmit).