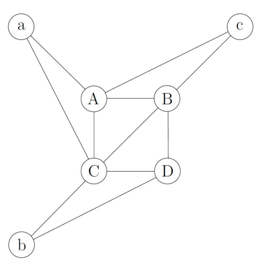
Feedback — Final (Q10 - Q20)

You submitted this exam on **Sat 11 May 2013 4:12 AM PDT -0700**. You got a score of **10.00** out of **10.00**.

This exam covers Chapters 10 - 20. Good luck!

Question 1

Consider the topology in the figure below. Nodes a,b, and c are end hosts and nodes A, B, C, and D are routers. Each link is 15 Mbps. There are 2 concurrent sessions: node a sends to node c, and node b sends to node c. Each session can split traffic on multiple paths. A link's bandwidth is shared equally between the two sessions if they happen to use the same link.



(a) How much bandwidth are a and b able to concurrently send to c?

(b) How much bandwidth are a and b able to concurrently send to c, if router A fails?

(c) How much bandwidth are a and b able to concurrently send to c, if routers A and D fail?

(d) How much bandwidth are a and b able to concurrently send to c, if routers A and C fail?

Your Answer		Score	Explanation
(a) 30 Mbps (b) 15 Mbps (c) 15 Mbps (d) 15 Mbps	✓	1.00	
(a) 30 Mbps (b) 30 Mbps (c) 30 Mbps			

(d) 30 Mbps

(a) 30 Mbps (b) 30 Mbps (c) 15 Mbps (d) 15 Mbps		
(a) 30 Mbps (b) 30 Mbps (c) 30 Mbps (d) 15 Mbps		
(a) 15 Mbps (b) 15 Mbps (c) 15 Mbps (d) 15 Mbps		
Total	1.00 / 1.00	

Question Explanation

Think about overall link capacities for bandwidth more than sharing bandwidth.

Question 2

Consider a single user within an ISP, where the user has a utility function $U(x) = U_\alpha(x) = \frac{x^{1-\alpha}}{1-\alpha}$ $(i.e.,$ alpha-fair utility). The price charged has a flat-rate (g) and a usage-based component (h), for a total $g + hx$.

(a) Given g and h from the ISP, what is the user's demand x^* (as a function of h and α)?
(b) Assume the ISP has monopoly price-setting power, and will push the user's net utility to zero when determining the flat-rate and usage-based prices. In this pricing scheme, what is the expression for g (as a function of h and α)?

Your Answer		Score	Explanation
(a) $x^* = ((1-\alpha)h)^{-\frac{1}{1-\alpha}}$ (b) $g = h^{\frac{\alpha-1}{\alpha}}$			
(a) $x^* = h^{-\frac{1}{\alpha}}$ (b) $g = \frac{\alpha}{1-\alpha} h^{\frac{\alpha-1}{\alpha}}$	✓	1.00	
(a) $x^* = h^{\frac{1}{\alpha}}$ (b) $g = \frac{\alpha}{\alpha-1} h^{\frac{1-\alpha}{\alpha}}$			

(a) $x^* = h^{-\frac{1}{\alpha}}$
(b) $g = h^{\alpha-1}$

Total	1.00 / 1.00

Question Explanation

(a) To determine the demand, we solve the net-utility maximization problem:

$$\frac{d}{dx}\left(U(x) - (g + hx)\right) = 0$$

$$x^{-\alpha} - h = 0$$

$$x^* = h^{-\frac{1}{\alpha}}.$$

(b) In a monopoly ISP, the relationship between g and h is determined by setting the net utility to zero:

$$U(x) - (g + hx) = 0.$$

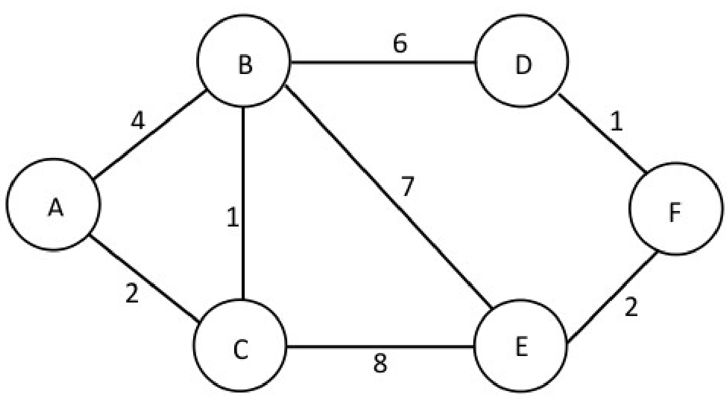
Using the result from (a), we have:

$$\frac{1}{1-\alpha} h^{\frac{\alpha-1}{\alpha}} - g - h^{\frac{\alpha-1}{\alpha}} = 0$$

$$g = \frac{\alpha}{1-\alpha} h^{\frac{\alpha-1}{\alpha}}.$$

Question 3

Consider the following router graph:



Run RIP, assuming that each router's table consists only of itself at t = 0. Assume that in the case of two path costs being the same, the node will choose to forward to the one with the lower ID.

(a) What neighbor does A forward to in order to get to F at t = 3? What is the resulting path to F?

(b) What neighbor does A forward to in order to get to F at t = 4? What is the resulting path to F?

(c) Suppose that link DF fails for t > 4. D immediately detects the failure. Will RIP re-converge? If so, what will A's final path to F be?

Your Answer		Score	Explanation
(a) No path to F is yet discovered. (b) Forwards to B; ABDF. (c) Will not re-converge.			
(a) Forwards to C; ACBDF. (b) Forwards to C; ACBDF. (c) Will re-converge; ACBEF.			
(a) Forwards to B; ABDF. (b) Forwards to C; ACBDF. (c) Will re-converge; ACBEF.	✓	1.00	
(a) Forwards to B; ABDF. (b) Forwards to C; ACBDF. (c) Will not re-converge.			
Total		1.00 / 1.00	

Question Explanation

(a, b) We will show the evolution of A's routing table, beginning at t = 0. The entries are in the format "NodeID   DestinationID   Cost   Next Node".

t = 0:

A	A	0	A

t = 1:

A	A	0	A
A	B	4	B
A	C	2	C

t = 2:

A	A	0	A
A	B	3	C
A	C	2	C
A	D	10	B
A	E	10	C

t = 3:

A	A	0	A
A	B	3	C
A	C	2	C
A	D	9	C
A	E	10	C
A	F	11	B

t = 4:

A	A	0	A
A	B	3	C
A	C	2	C

A	D	9	C
A	E	10	C
A	F	10	C

At t = 3, A will forward to B to get to F, and at t = 4, he will forward to C.

(c) RIP will re-converge because there still exists a path between each node. All routes that are currently using DF (A -> F, B -> F, C -> F, D -> F) will see their costs keep increasing, until an alternative is preferable. To see this, consider node D. When DF fails, he sees B's message "B F 7" and will re-compute the routing entry to F as "D F 13 B". Then, B will decide to route through E to get to F instead, since 9 > 13 + 6. So B advertises "B F 9 E", and D recomputes this as well. A's path becomes ACBEF.

Question 4

Recall that the congestion window length w changes with time as follows during TCP Reno's congestion avoidance phase. (1) If an ACK is received, then increase w by $\frac{1}{w}$; (2) If congestion is detected, then decrease w by $\frac{w}{2}$. Suppose the probability of failed transmission is p, and the probability of a successful transmission is $1-p$. The transmission rate is $x = \frac{w}{RTT}$ packets per second.

Find the transmission rate x_r at equilibrium.

Your Answer		Score	Explanation
◉ $x_r = \frac{1}{RTT}\sqrt{\frac{2(1-p)}{p}}$	✓	1.00	
○ $x_r = \frac{1}{RTT}\sqrt{\frac{p}{2(1-p)}}$			
○ $x_r = \frac{1}{RTT}\left(\frac{2(1-p)}{p}\right)^2$			
○ $x_r = \frac{1}{RTT}\left(\frac{p}{2(1-p)}\right)^2$			
○ $x_r = RTT\,\frac{2(1-p)}{p}$			
Total		1.00 / 1.00	

Question Explanation

Recall that equilibrium implies no change (on average) in w.

Question 5

Consider a P2P network with one dedicated server and 10 peers. The upload bandwidth of the server U_s is

1 MB/s. All the peers are uniform with an upload bandwidth U_i of 0.4 MB/s and a download speed D_i of 5 MB/s.

(a) If a 10 MB file has to be distributed amongst all the peers, what is the shortest possible time in which this can happen?
(b) The admin of this network decides to improve performance by adding another peer. If he wants to cut the time calculated in part (a) by half, what should the minimum upload bandwidth of the new peer be ?

Your Answer		Score	Explanation
(a) 10 seconds (b) 0.8 MB/s			
(a) 20 seconds (b) 6 MB/s	✓	1.00	
(a) 15 seconds (b) 2 MB/s			
(a) 5 seconds (b) 10 MB/s			
Total		1.00 / 1.00	

Question Explanation

(a) The shortest possible time is given by:

$$T = \max\{\frac{F}{u_s}, \frac{F}{d_{min}}, \frac{NF}{u_s + \sum_{i=1}^{N} u_i}\} = \max\{\frac{10}{1}, \frac{10}{5}, \frac{10\times 10}{1+0.4\times 10}\} = \max\{10, 2, 20\}\text{sec} = 20\text{sec}.$$

(b) If we cut the time in half, we are still limited by the third term in the above equation, because $10 \geq 10 > 2$. Hence, when we add a user with an upload bandwidth x, we want:

$T = \frac{11\times 10}{1+0.4\times 10+x} = \frac{110}{5+x} = 10$. Solving this gives $x = 6\text{MB/s}$.

Question 6

For a particular 5-stage Clos network defined by $(2, 2, 8)$, we wish to construct this by only using 2x2 switches to save money by buying less complex switches.

How many 2x2 switches will we need if we wish to build a folded Clos network using only 2x2 switches?

Your Answer		Score	Explanation
32	✓	1.00	

- 16
- 64
- 8
- 128

Total	1.00 / 1.00

Question Explanation

Expand the $(2, 2, 8)$ network into 2x2 switches, and count the 2x2 switches in the first ceil($\frac{x}{2}$) stages, where x is the number of stages in the expanded Clos Network.

Question 7

Consider a playback buffer with three packets 1, 2, and 3. Packet 1 is transmitted at t = 0, Packet 2 at t = 1, and Packet 3 at t = 2 sec, i.e. a unit step function. Each packet arrives at the receiver at a probabilistic time given by a Pareto distribution, which means the arrival times are Pareto random variables. The probability density function of the arrival time of packet i is then:

$$p_i(t) = \frac{\alpha t_i}{t^{\alpha+1}}, t \geq t_i; 0, t < t_i;$$

where t_i and α are parameters for the packet i.

Let $\alpha = 5$ for each packet i, and $t_i = 5, 6, 8$ for $i = 1, 2, 3$, respectively. What is the optimal playback time of the first packet, p^*, that minimizes latency but ensures that all packets have been received with at least 95% probability? You may assume that the arrival times are independent of one another.

Your Answer		Score	Explanation
$p^* = 12.6$ sec	✓	1.00	
$p^* = 9.1$ sec			
$p^* = 6.0$ sec			
$p^* = 10.9$ sec			
Total		1.00 / 1.00	

Question Explanation

First, we find the CDF of each Pareto distribution, which will tell us the probability that the arrival times A_i

are at least t:

$P(A_i > t) = 1 - (\frac{t_i}{t})^{\alpha}$.

We use this to find the times which guarantee that each packet has been received with at least 95% probability. For packet 1, $P(A_1 > t) = 0.95 = 1 - (\frac{5}{t})^5$, giving us $t = 9.1$ sec. For packet 2, $P(A_2 > t) = 0.95 = 1 - (\frac{6}{t})^5$, giving us $t = 10.9$ sec. And for packet 3, $P(A_3 > t) = 0.95 = 1 - (\frac{8}{t})^5$, giving us $t = 14.6$ sec. Out of 9.1, $10.9 - 1$, and $14.6 - 2$, $14.6 - 2 = 12.6$ sec is the highest, therefore requiring $p^* = 12.6$ sec to ensure smooth playback with 95% probability.

Question 8

There is a simpler random access protocol than CSMA that is just as famous. It is called Aloha, as it was invented in Hawaii in the early 1970s, and further led to the development of packet radio technologies. The operation of (the slotted time version of) Aloha is easy to describe. During each timeslot, each of a given set of N users chooses to transmit a packet with probability p. We assume that if two or more users transmit at the same timeslot, all packets are lost. This is the only feedback available at each transmitter. Each lost packet is retransmitted with probability p too. We assume this process continues until a packet is eventually transmitted successfully.

Suppose three active nodes A, B and C are competing for access to a channel using slotted Aloha. Assume each node has an infinite number of packets to send. The first slot is numbered slot 1, the second slot is numbered slot 2, an so on.

(a) What is the probability that node A succeeds for the first time in slot A?

(b) What is the probability that some node (A, B or C) succeeds in slot 2?

(c) What is the probability that the first success occurs in slot 4?

Your Answer		Score	Explanation
(a) $p(1-p)^2$ (b) $3p(1-p)^2$ (c) $(1-3p(1-p^2))^3 * 3p(1-p)^2$	✓	1.00	
(a) p (b) $p + 2(1-p)$ (c) $(1-3p(1-p))^2$			
(a) p (b) $(p(1-p)^2)^3$ (c) $3p(1-p)^2$			

(a) $p(1-p)^3$
(b) $3p(1-p)^3$
(c) $(1-3p(1-p^3))^4 * 3p(1-p)^3$

Total 1.00 / 1.00

Question Explanation

(a) We need A to transmit, and the other two not to transmit. This probability is simply $p \times (1-p) \times (1-p)$.

(b) Here, we need one station to transmit, and the other two not to transmit. But it could be any node, whereas in (a), we were looking for one in particular. So this is $P(\text{A Successful}) + P(\text{B Successful}) + P(\text{C Successful})$, which is just $3 \times P(\text{A Successful})$.

(c) This is $P(\text{Failure in first three slots}) \times P(\text{Success in the fourth slot})$. Here, we need failure in the first three slots. The probability of a failure in one slot is $1 - P(\text{Success}) = 1 - 3p(1-p^2)$. For three in succession, we need this to occur three times: $P(\text{Failure in first three slots}) = (1-3p(1-p^2))^3$. And the success in the fourth slot is the same as before. Hence, we have $(1-3p(1-p^2))^3 * 3p(1-p)^2$.

Question 9

As mentioned in Chapter 14, TCP starts with a small congestion window, which is initially set to 1 MSS (maximum segment size), and goes through a slow start phase. During this time, the congestion window increases multiplicatively, i.e. 2 MSS, 4 MSS, 8 MSS, and so on for every round trip time, until the slow start threshold is reached and the congestion avoidance phase begins. Suppose you open a web browser and begin downloading a webpage of 100 MSS.

(a) Assuming there is no packet lost, how many RTTs are required in order to download the webpage? Remember to add one RTT of handshake to begin the connection.

(b) If RTT = 70 ms, what is the average throughput in kbps? Assume 1 MSS = 536 Bytes.

Your Answer	Score	Explanation
(a) 7 MSS (b) 875 kbps		
(a) 7 MSS (b) 683 kbps		
(a) 8 MSS (b) 428 kbps		

(a) 8 MSS (b) 748 kbps	✓	1.00
Total		1.00 / 1.00

Question Explanation

(a) We need to keep adding the MSSs transmitted for successive RTTs until we hit 100 MSS:

$$1 + 2 + 4 + 8 + 16 + 32 + (64 - 27) = 100.$$

Hence, we need 7 RTTs for actual downloading, and 1 extra to set up the TCP connection, for a total of 8 RTTs.

(b) The amount of kb transferred is given by:

$$100\text{MSS} \times 536\,\frac{\text{B}}{\text{MSS}} \times 8\,\frac{\text{b}}{\text{B}} \times \tfrac{1}{1024}\,\frac{\text{kb}}{\text{b}} = 418.75\text{kb}.$$

The time it takes in sec is given by:

$$8\text{RTTs} \times 70\,\frac{\text{ms}}{\text{RTT}} \times \tfrac{1}{1000}\,\frac{\text{s}}{\text{ms}} = 0.56\text{s}.$$

Hence, the rate is $\frac{418.75}{0.56} = 748\text{kbps}.$

Question 10

The Ultimatum Game is a game where two players interact to decide how to divide a sum of money between them. The first player proposes how to divide the sum, and the second player can either accept or reject the proposal. If the offer is accepted, then the division is implemented. If the offer is rejected, the no one receives anything. The game is only played once.

Alice and Bob are to divide a 1-foot-long sandwich between the two of them. Alice knows that Bob will not accept any offer less than x foot (where x is some fraction between 0 and 1), but only has some estimate for Bob's probability density function of x:

$p(x) = 4x$ if x is less than 0.5

$p(x) = 4(1 - x)$ if x is greater than or equal to 0.5

How will Alice split the sandwich, and what is her expected share?

Your Answer		Score	Explanation
Alice offers Bob 0.59 ft, and expects to receive 0.27 ft.	✓	1.00	

- Alice offers Bob 0.5 ft, and expects to receive 0.5 ft.
- Alice offers Bob 0.42 ft, and expects to receive 0.48 ft.
- Alice offers Bob 0.21 ft, and expects to receive 0.79 ft.

Total	1.00 / 1.00

Question Explanation

Maximize the expected return for Alice.