Final Exam SOLUTION

Computer Networks

Spring 2010

Prof. Cheng-Fu Chou

<u>Question 1: ``Quickies''(30%)</u> Answer each of the following questions *briefly, i.e., in at most a few sentences.*

a) (5%) Where can queueing occur in a router? Briefly explain the conditions that lead to such queueing.

Queueing can occur at both the input ports and the output ports of a router. Queueing occurs at the output port when the arriving rate of packets to the outgoing link exceeds the link capacity. Queue occurs on an input port when the arriving rate of packets exceeds the switch capacity; head-of-the-line blocking can also cause queueing at the input ports.

1. 沒講出在哪邊發生或者沒有討論發生在 input port 和 output port 的不同，只有提到 input rate>output rate，可得 3 分
2. 沒講出在哪邊發生，但有分別針對
   packet arrival rate vs. switch fabric rate
   switch fabric rate vs. output link capacity
   討論，可得 5 分
3. 只有討論 queueing in input port 或 output port 其中一種，可得 3 分.

b) (5%) What is HOL (Head-of-Line) blocking? Does it occur in input ports or output ports?

The switch fabric can only switch the packets at the head of the buffer per cycle. HOL arises in input ports when packets arriving at different input ports are destined for the same output port. If the HOL packet of a certain buffer at the input cannot be switched to an output port because of contention, the <u>rest of the packets in that buffer are blocked by that Head-of-Line packet, even if there is no contention at the destination output ports for those packets</u>.

發生位置寫對: 2 分

敘述 HOL-blocking 正確: 3 分; 只有說 input port buffer 前頭的 packet 擋到後面的 packet，但沒說明是 output port contention 造成後面要送到別的 output port 的 packet 被 block，僅得 2 分.

只有說兩個 packet 到同一 output port 造成 contention，但是沒說明後面的 packet 會被前頭的 block，僅得 1 分.

c) (5%) What is meant by the term “tunneling”?

In tunneling, an IP datagram (e.g., an Ipv6 datagram) is carried as the payload of another IP datagram.

只有點到 IPV6 跟 IPV4 間的轉換，可得 2 分

點到 packet format 轉換，沒說怎麼轉換，可得 3 分

d) (5%) Biefly describe TCP’s slowstart algorithm. What causes the TCP slowstart algorithm to end, and TCP congestion avoidance to begin?

When connection begins, TCP’s congestion window is 1 MSS. For each ACK received, the congestion window size is double. This causes the number of segments sent per RTT to increase exponentially. Slowstart ends when a threshold window size is reached.

點到初始值是 1，得 1 分

點到倍數增加，得 2 分

說明何時結束 slow start，得 2 分;說明是以 3 duplicated ACK 進入 congestion avoidance，可得 1 分.

e) (5%) Two hosts simultaneously send data through a link of capacity 1Mbps. Host A generates data with a rate of 1Mbps and uses TCP. Host B uses UDP and transmits a 100bytes packet every 1ms Which host will obtain higher throughput?

B host. Flow A (1Mbps) + B (0.8Mbps) > link capacity (1Mbps). While a packet loss (congestion) occurs, host A can decrease the congestion window size since the congestion control of TCP. On the other hand, host B keep sending data at rate 0.8Mbps without ACK. Therefore, host B can occupy more capacity of a link.

回答 **host A,** 但能給予合理解釋**,** 可得 **2~3** 分

回答 **host B,** 解釋 **UDP** 不需等 **ACK** 或 **overhead** 較低**,** 也可得 **4** 分

回答 **host B,** 但解釋錯誤**,** 得 **1** 分

f) (5%) What are different inter-AS and intra-AS protocols used in the Internet

Routers are aggregated into autonomous systems (ASs). Within an AS, all routers run the same intra-AS routing protocol. Special gateway routers in the various ASs run the inter-autonomous system routing protocol that determines the routing paths among the ASs. The problem of scale is solved since an intra-AS router need only know about routers within its AS and the gateway router(s) in its AS.

寫出在單一 **AS** 中用 **intra-AS,** 在 **AS** 間用 **inter-AS** 即可得滿分**,** 只寫對一個 **2%**

g) (5%) What is the difference between a group-shared tree and a source-based tree in the context of multicast routing?

In a group-shared tree, all senders send their multicast traffic using the same routing tree. With source-based tree, the multicast datagrams from a given source are routed over s specific routing tree constructed for that source; thus each source may have a different source-based tree and a router may have to keep track of several source-based tress for a given multicast group.

source-based tree: one tree per source, e.g. shortest path trees/reverse path forwarding

group-shared tree: group uses one tree, e.g. minimal spanning trees/center-based trees

寫出在 group-shared tree 中, 所有 source 共享同一個 tree, 而在 source-based tree 中, 須爲每個 source 建立一個獨立的 tree 即可得滿分

只寫出可採用的演算法名稱可是沒講差異/原理 0 分

Source 寫成 node/host 之類的會酌量扣分或不給分

Problem 2: Congestion Control and TCP (40%)

a) (5%) What is the difference between congestion control and flow control?

Flow control is about matching the speed of a sender to the capabilities of the receiver.
Congestion occurs when senders overutilize the resources within the network.
Flow control means preventing the source from sending data that the sink will end up dropping because it runs out of buffer space. 也就是 source 送太快, 導致 receiver 的 buffer 爆掉
Congestion control means preventing (or trying to prevent) the source from sending data that will end up getting dropped by a router because its queue is full. 也就是避免網路總流量過大, 塞爆 router 的 queue

上述兩組答案任寫出一組皆可得滿分, 只對一個看答題完整性給 2%或 3%

b) (5%) It is said that a TCP connection "probes" the network path it uses for available bandwidth. What is meant by that?

TCP keeps increasing its send rate (by increasing its window size) until loss occurs (at which point congestion has set in). TCP then sets its rate lower but again begins increasing its send rate to again determine the point at which congestion sets in. In this sense, TCP is contantly probing the network to see how much bandwidth it can use.

有寫出概念(e.g. SS→CA→(CA 發生時得到當時網路最大值)→SS→CA)就可以拿滿分

c) Suppose that in TCP, the sender window is of size N, the base of the window is at sequence number x, and that the sender has just sent a complete window's worth of segments. Let RTT be the sender-to-receiver-to-sender round trip time, and let MSS be the segment size.

i. (10%) Is it possible that there are ACK segments in the receiver-to-sender channel for segments with sequence numbers lower than x? Justify your answer.

It is possible. Suppose that the window size is N=1. The sender sends packet x-1, which is delayed and so it timeouts and retransmits x-1. There are now two copies of x-1 in the *network. The receiver receives the first copy of x-1 and ACKs. The receiver then receives the 2nd copy of x-1 and ACKs. The sender receives the first ACK and sets it window base to x. At this point, there is still an ACK for x-1 propagating back to the sender.*

只寫可能發生, 給 2 分

只寫可能發生, 而沒有寫出正確原因, 或寫出不清楚的原因可得 3 分

只寫可能發生, 並寫出封包 delay/timeout 重送…, 但沒有詳細正確說明可得 6 分

ii. (5%) Assuming no loss, what is the throughput (in packets/sec) of the sender-to-receiver connection?

Assume that N is measured in segments. The sender can thus send N segments, each of size MSS bytes every RTT secs. The throughput is this ***N·MSS/RTT. or N/RTT***

因為題目沒講清楚, 所以兩者任寫一個都算對

iii. (5%) Suppose TCP is in its congestion avoidance phase. Assuming no loss, what will the window size be after the N segments are ACKed?

N+1

呃..對就是對, 錯就是錯

d) (10%) Consider the use of TCP over a wireless channel, which is much more prone to bit-level errors than a wire or optical fiber. Comment on how well suited TCP's (i) error detection mechanism, and (ii) its window-based congestion control algorithm are for such a bit-error-prone environment.

TCPs checksum method is not very strong (compared to other techniques such as CRC) in detecting bit level errors.

要寫出 checksum 不是很強, 因為無法很好的偵測 bit level error 的原因, 沒寫出正確原因, 則視答題內容給 2~3%, 這部分占 5 分

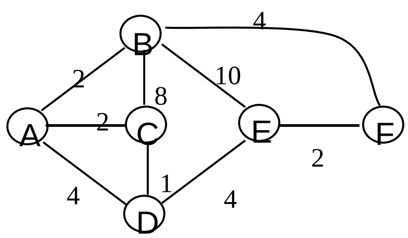
TCPs window based congestion control algorithm interprets the absence of an ACK as an indication of congestion (buffer overflow) in the network. If the receiver is not sending ACKs because of bit level corruption (which is not related to congestion control), then the sender will incorrectly interpret the absence of ACKs as a congestion indication and will unnecessarily decrease its window. Thus the window based congestion control algorithm is not well suited for an environment in which packets can be lost due to bit level errors.

須寫出因為 bit-level error, ACK 封包可能會沒傳到, TCP 可能因此誤認為 congestion 而降低傳輸效能, 沒寫原因則視答題內容給 2~3%, 這部分占 5 分

Question 3: Routing Algorithms(20%)

a) (10%) Consider the network shown below. Show the operation of Dijkstra's (Link State) algorithm for computing the least cost path from **D** to all destinations.

N	D(A),p(A)	D(B),p(B)	D(C),p(C)	D(E),p(E)	D(F),p(F)
D	4,D	infty	1,D	4,D	infty
DC	3,C	9,C		4,D	infty
DCA		5,A		4,D	infty
DCAE		5,A			6,E
DCAEB					6,E
DCAEBF					



過程中的 parent node 佔 2 分
過程中的 distance 佔 4 分
最終結果 distance 佔 3 分
最終結果 parent node 佔 1 分
每寫錯一項, 扣 1 分

b) (10%) Consider a node Z, which has only two neighbors – X and Y. The link cost from Z to X is 2 and the link cost from Z to Y is 3. Suppose X and Y have the distance tables shown below, which they send to Z.

D^X	s_1	s_2	s_3
f	13	2	4
g	5	7	9

D^Y	t_1	t_2	t_3
f	6	5	8
g	2	9	7

Complete the following table in node Z after it receives the distance tables from its neighbors X and Y:

D^Z	X	Y
f	4	8
g	7	5

$D^Z(f,X) = c(Z,X) + mincost(X,f) = 2 + 2 = 4$
$D^Z(g,X) = c(Z,X) + mincost(X,g) = 2 + 5 = 7$
$D^Z(f,Y) = c(Z,Y) + mincost(Y,f) = 3 + 5 = 8$
$D^Z(g,Y) = c(Z,Y) + mincost(Y,g) = 3 + 2 = 5$

Question 4: NAT: Network Address Translation (15%)

a) (5%) Suppose host A is behind a NAT, and host B has a public IP. The TCP connection between host A and B has been established. While host A sends datagram to host B and host B replies datagram to host A, what does the NAT router do?

Refer to Chapter 4 slide, page 54.
須寫出 NAT 記錄 A 的 IP 與 Port, 並在自己的 routing table 中加入對應 WAN IP 和 Port 的項目, 而當 B 回應資料時, NAT 在 WAN 收到後, 依照 routing table 查出此 Port 的對應的內部 IP 與 Port, 再將資料轉送給 A
有寫出 NAT 轉送資料, 但沒寫 routing table 項目的建構和使用過程, 則得 2 分
只有寫 IP 轉換, 但沒寫出 Port 在 Mapping 時的重要性, 則是答題內容給 2~3 分

b) (10%) NATs on P2P applications: Suppose a peer A discovers through query that peer B has a file it wants to download. Also, suppose that A and B are both behind a basic NAT which does not support any other functions. Is there a way to allow A to establish a TCP connection with B directly without application-specific NAT configuration? Discuss why and how to do it (if it is possible).

Impossible.
Consider what happens when A attempts to establish a TCP connection with B. A might send a TCP SYN packet with B's WAN IP as the destination address and some destination port number, say, x. When the B's NAT receives this TCP SYN packet, it doesn't know to which internal host it should direct the packet, since it doesn't have an entry for a connection initiated from the WAN side. Thus the NAT will drop the SYN packet.
寫不行但理由亂寫或不合理, 則給 3 分
寫不行但沒有說明原因, 則給 2 分
寫不行, 雖沒寫出為什麼, 但有寫出可能的解法, 給 5 分
請注意題目(紅色部分)已經寫清楚 A 與 B 間需為**直接連線**, 並且**無法設定 NAT**, NAT 也僅提供**最基本的功能**, 所以是沒辦法穿透的

Question 5: Subnets (10%)

Consider the topology shown below. Denote the three subnets with hosts (starting clockwise at 12:00) as Networks A, B, and C. Denote the subnets without hosts as Networks D, E, and F.

(a) Assign network addresses to each of these six subnets, with the following constraints: All addresses must be allocated from 214.97.254/17; Subnet A should have enough addresses to support 250 interfaces; Subnet B should have enough addresses to support 120 interfaces; and Subnet C should have enough addresses to support 120 interfaces. Of course, subnets D, E and F should each be able to support two interfaces. For each subnet, the assignment should take the form a.b.c.d/x or a.b.c.d/x – e.f.g.h/y.

possible assignments are

Subnet A: 214.97.255/24 (256 addresses)

Subnet B: 214.97.254.0/25 – 214.97.254.0/29 (128 – 8 =120 addresses)

Subnet C: 214.97.254.128/25 (128 addresses)

Subnet D: 214.97.254.0/31 (2 addresses)

Subnet E: 214.97.254.2/31 (2 addresses)

Subnet F: 214.97.254.4/30 (4 addresses)

原則上，只要不會出現 confuse 都會給對. 但如果給 sub-net D, E, F 的 range 卻比給 sub-net A, B, C 還大，扣 3 分

(b) Using your answer of part (a), provide the forwarding tables (using longest prefix matching) for each of the three routers.

Router 1

Longest Prefix Match	*Outgoing Interface*
11010110 01100001 11111111	*Subnet A*
11010110 01100001 11111110 0	*Subnet D*
11010110 01100001 11111110 1	*Subnet F*

Router 2

Longest Prefix Match	*Outgoing Interface*
11010110 01100001 11111111	*Subnet D*
11010110 01100001 11111110 0	*Subnet B*
11010110 01100001 11111110 1	*Subnet E*

Router 3

Longest Prefix Match	*Outgoing Interface*
11010110 01100001 11111111	*Subnet F*
11010110 01100001 11111110 0	*Subnet E*

11010110 01100001 11111110 1 *Subnet C*

沒有寫出如何到其他主要 sub-net A, B, C, 個別扣 0.5 分. EX : R1 需要知道 packet 是送往 sub-net A, B, or C 的話, 要由哪個 interface 出去

沒寫到 sub-net D ,E, F, 不扣分.