Computer Networks Final (104/1)

只寫答案而沒有解釋說明，扣一半分數

1. 針對 63.107.172.1 這個 IP address，(以十進位表示，要寫完整過程) (17%)
   a. 這一個 IP 屬於那個 Class 的網路？以二進位說明(1%) 其所屬的 IP 網路表示法為何？(2%) 可用 IP 範圍？(2%) 共有幾個 IP 可用？(1%) mask 的值為何？(1%)
   b. 將此 IP 網路分成 7 subnets，subnet mask 的值為何？(2%) 請列出第 7 個 subnet 的網路表示法 (2%) 可用 IP 範圍？(2%) 共有幾個 IP 可用？(1%)
   c. 手動設定電腦的網路時，至少要設定哪三個項目的資訊，才可以上網？(3%)

2. (a) Draw a figure to show four components of a router (8%) (b) Draw three types of switching fabrics with their names. (1% each) (c) What is Head-of-the-Line (HOL) blocking? (3%) (14% total)

3. (a) Consider the two 16-bit words (shown in binary) below. Recall that to compute the Internet checksum of a set of 16-bit words, we compute the one's complement sum of the two words. That is, we add the two numbers together, making sure that any carry into the 17th bit of this initial sum is added back into the 1's place of the resulting sum); we then take the one's complement of the result. Compute the Internet checksum value for these two 16-bit words: (8%)

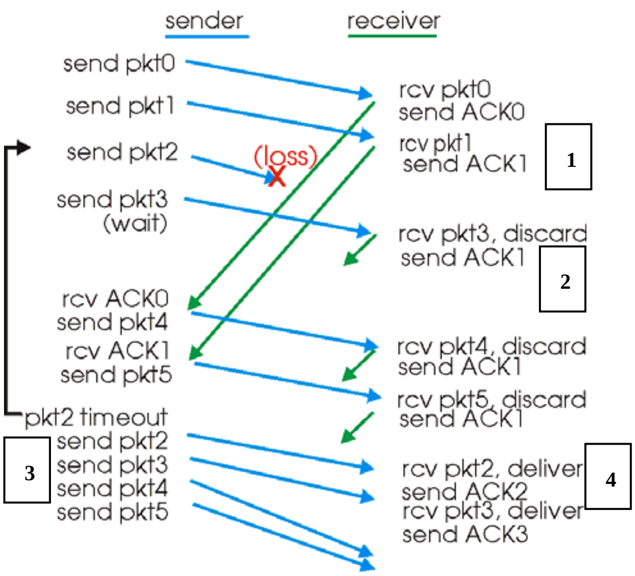
   10100000 10010010 this binary number is 41106 decimal (base 10)

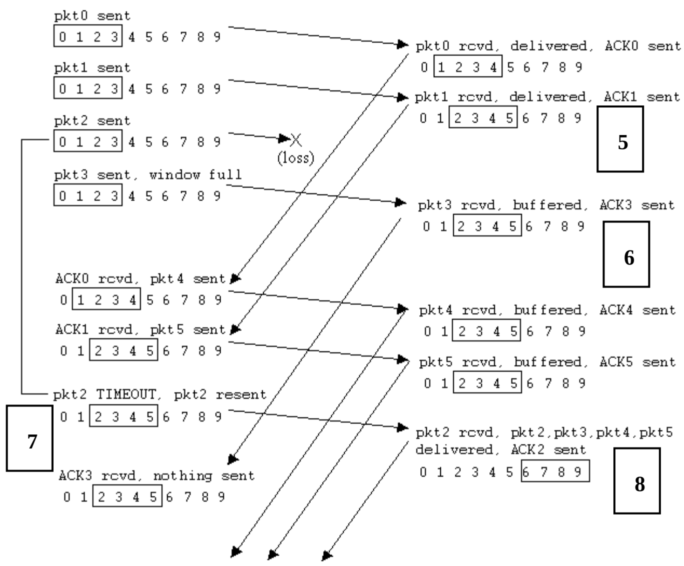
   01111110 10011111 this binary number is 32415 decimal (base 10)

   (b) With the 1's complement scheme, how does the receiver detect errors? Is it possible that 1-bit error will go undetected? (2%) How about a 2-bit error? (2%) (12% total)

4. Draw the flow of the TCP three way handshake to explain its operations. Suppose the <u>initial sequence numbers of the client and the server are 1 and 200</u>, respectively. 必須在圖上分別清楚標示出 TCP 必要的 flag, sequence number, and ACK number. (10%)

5. List and compare two pipelined transport protocols with these two figures. (寫出 Window=? 與各標號處的動作 10%)





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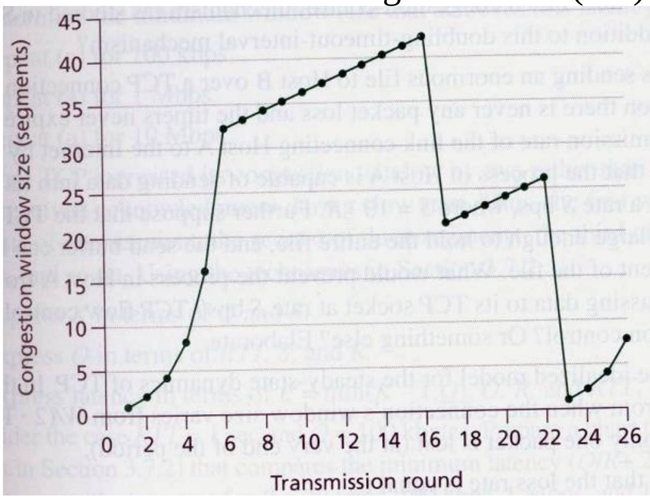
6. (a) Describe how TCP Reno does its congestion control. (8%) (12% total)
   (b) Answer and justify the following questions.
   After the 22th transmission round, is segment loss detected by a triple duplicate ACK or by a timeout? (2%)
   During what transmission round is the 50th segment sent? (2%)



7. Consider the TCP procedure for estimating RTT
   ($EstimatedRTT^{n} = \alpha \times SampleRTT^{n-1} + (1-\alpha) \times EstimatedRTT^{n-1}$).
   (a) Why TCP uses this function? (2%)
   (b) Let $SampleRTT^{n}$ be the most recent sample RTT, let $SampleRTT^{n-1}$ be the next most recent sample RTT, and so on. Express $EstimatedRTT^{n}$ in terms of n SampleRTTs if $EsitmatedRTT^{1} = 0$. (要有兩次疊代過程(各 2%)後寫出通式與以 summation 總和符號表示(各 2%) (10% total)

8. (a) Explain how TCP Fast Retransmit works. (6%) (15% total)
   (b) How TCP does its flow control? (6%)
   (c) Suppose the TCP sequence number space is of size *k*. What is the largest allowable sender window *w*? (3%)

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   b. 將此 IP 網路分成 7 subnets，subnet mask 的值為何？(2%) 請列出第 7 個 subnet 的網路表示法(2%) 可用 IP 範圍？(2%) 共有幾個 IP 可用？(1%)
   c.手動設定電腦的網路時，至少要設定哪三個項目的資訊，才可以上網？(3%)

Ans:

a.

63.107.172.1 的二進位表示法為 00111111. XXXXXXXX.XXXXXXXX.XXXXXXXX，由前 1 個 bits 0 可判斷為 Class A 的 IP。(1%)
此 IP 所屬於的 Class A 的網路表示法為 63.0.0.0 (2%)
所有 host ID 部分的 24 個 bit 的 X 不可以全為 0 或 1，
因此第一個可用 Host ID 為 00111111. 00000000.00000000. 00000001 = 63.0.0.1 (1%)
最後一個可用 Host ID 為 00111111.11111111.11111111.11111110 = 63.255.255.254 (1%)
->共有 $2^{24}-2$ 個可用 Host ID (1%)
Mask: 255.0.0.0 (1%)

b.

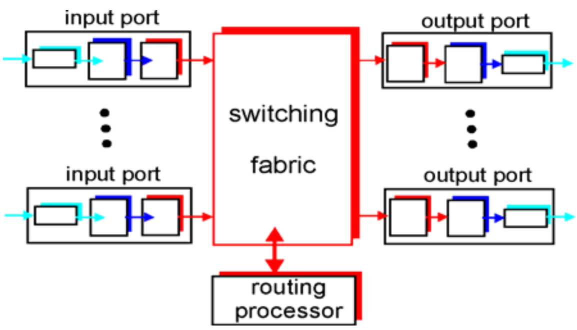
將此 Class A 網路分成 7 個 subnet，加上全為 0 與全為 1 的兩個不能用的 subnet ID，最少需要 7+2=9 <= 2^4, subnet mask 的值 => 需要 Host ID 的前 4 個 bits 當作 subnet ID。所以新的 subnet mask 是由原本 Class A 的 default subnet mask 255.0.0.0 來改，改成 255.11110000.00000000.00000000=> 255. 240.0.0 (2%)

subnet 的 ID 要從此 Class A Network ID 00111111.XXXXXXXX.XXXXXXXX.XXXXXXXX 來改，需要 Host ID 的前 4 個 bits 當作 subnet ID，不可全為 0 或 1。因此第 7 個 subnet ID 為 00111111. 01110000.00000000.00000000 => 63.112.0.0(2%)
因此第一個可用 Host ID 為 00111111.01110000.00000000. 00000001 = 63.112.0.1 (1%)
最後一個可用 Host ID 為 00111111.01111111.11111111. 11111110 = 63.127.255.254 (1%)
->共有 $2^{20}-2$ 個可用 Host ID (1%)

c. IP address, subnet mask, default gateway (3%)

2. (a) Draw a figure to show four components of a router (8%) (b) Draw three types of switching fabrics with their names. (1% each) (c) What is Head-of-the-Line (HOL) blocking? (3%) (14% total)

Ans:
(a) (2% each, 8% total)



(b) (3%)
switching via memory; (1%)
switching via a bus; (1%)
switching via an interconnection network (1%)

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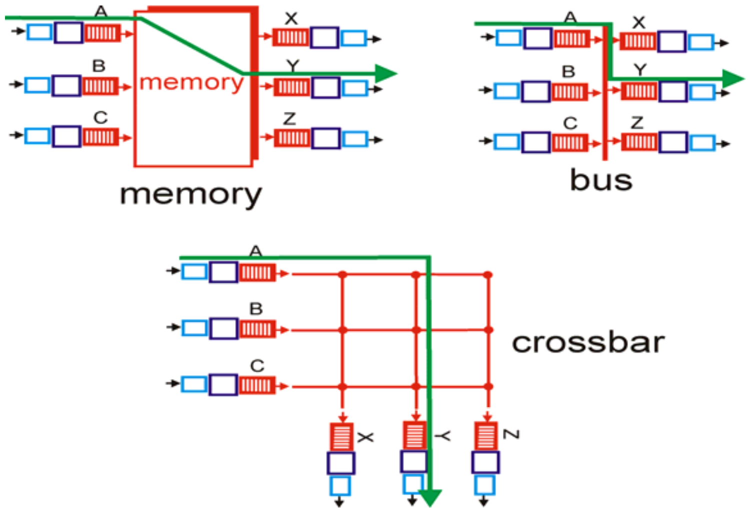
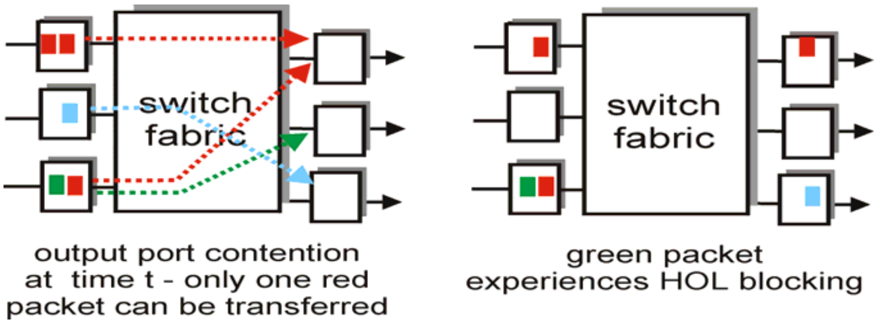


(c) queued datagram at front of queue prevents others in queue from moving forward (3%)



3. (a) Consider the two 16-bit words (shown in binary) below. Recall that to compute the Internet checksum of a set of 16-bit words, we compute the one's complement sum of the two words. That is, we add the two numbers together, making sure that any carry into the 17th bit of this initial sum is added back into the 1's place of the resulting sum); we then take the one's complement of the result. Compute the Internet checksum value for these two 16-bit words: (8%)

10100000 10010010 this binary number is 41106 decimal (base 10)

01111110 10011111 this binary number is 32415 decimal (base 10)

(c)With the 1's complement scheme, how does the receiver detect errors? Is it possible that 1-bit error will go undetected? (2%) How about a 2-bit error? (2%) (12% total)

Ans:

(a) When we add these first two numbers together, we get:

```
  10100000 10010010
  01111110 10011111
  -------- --------
1 00011111 00110001    note the carry into the 17th bit   (4%)
```

Since there is a carry, we need to add a 1 in the ones place (rightmost bit) of the rightmost 16-bit quantity above, giving: 00011111 00110010 (2%) and then we need to take the ones complement of this value, to get the Internet checksum: 11100000 11001101 (2%)

(b)

All one-bit errors will be detected (2%), but two-bit errors can be undetected (2%) (e.g., if the last digit of the first word is converted to a 0 and the last digit of the second word is converted to a 1).

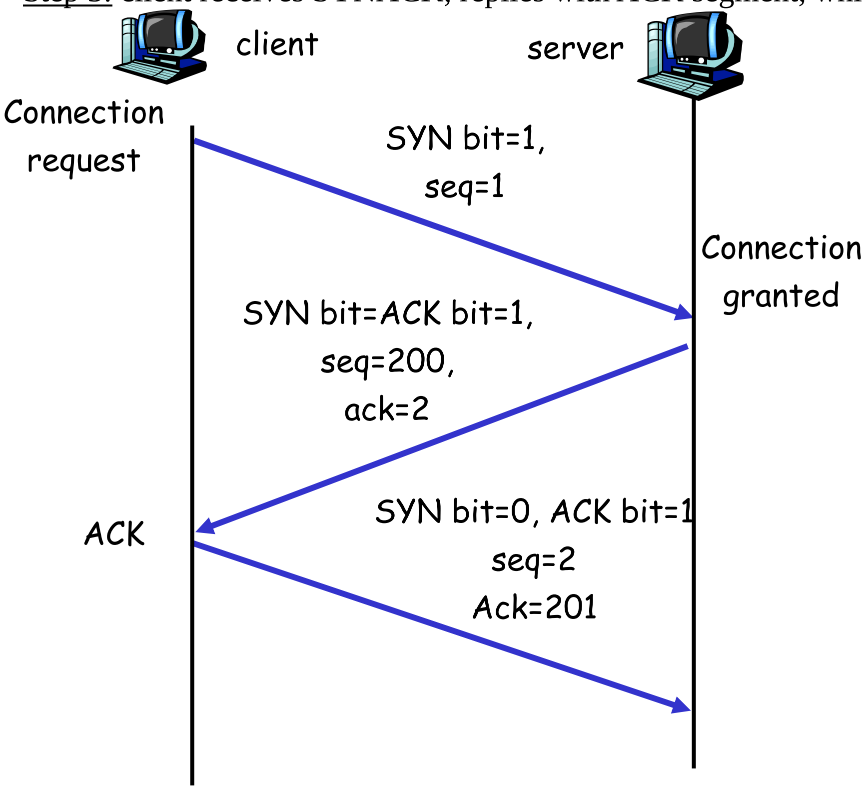
4. Draw the flow of the TCP three way handshake to explain its operations. Suppose the initial sequence numbers of the client and the server are 1 and 200, respectively. 必須在圖上分別清楚標示出 TCP 必要的 flag, sequence number, and ACK number. (10%)

Ans: Three way handshake:

Step 1: client host sends TCP SYN segment to server (搭配圖要正確 2%)

Step 2: server host receives SYN, replies with SYNACK segment (4%)

Step 3: client receives SYNACK, replies with ACK segment, which may contain data (4%)



上圖每個符號含內容 1 分，標示不全者，視狀況扣分，共 10 分

5. List and compare two pipelined transport protocols with these two figures. (寫出 Window=? 與各標號處的動作 10%)

Ans:

Go-back-N (5%)

- "window" of up to N, consecutive unack'ed pkts allowed (window = 4) (1%)

(1) ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq # (1%)

(2) out-of-order pkt:

- discard (don't buffer) -> no receiver buffering! (1%)
- Re-ACK pkt with highest in-order seq # (1%)

(3) timeout(n): retransmit pkt n and all higher seq # pkts in window (1%)

(4) deliver in-order segments to upper layer. (1%)

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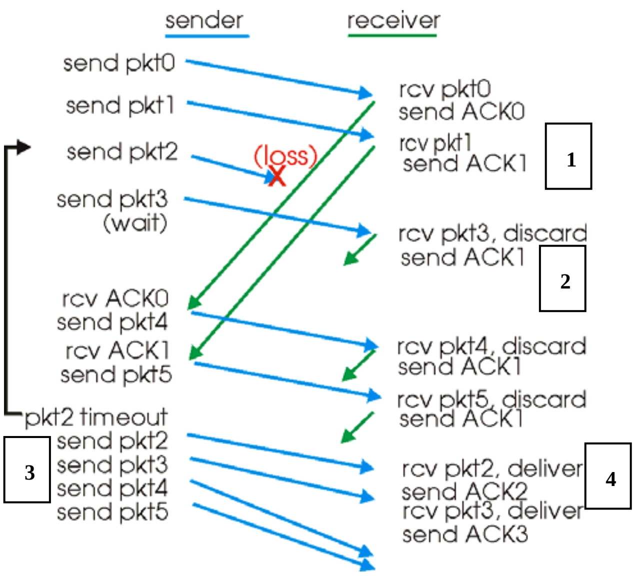
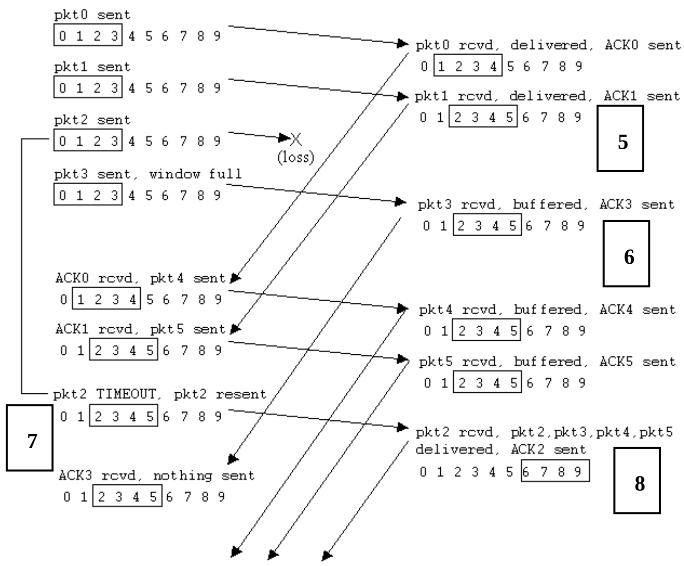
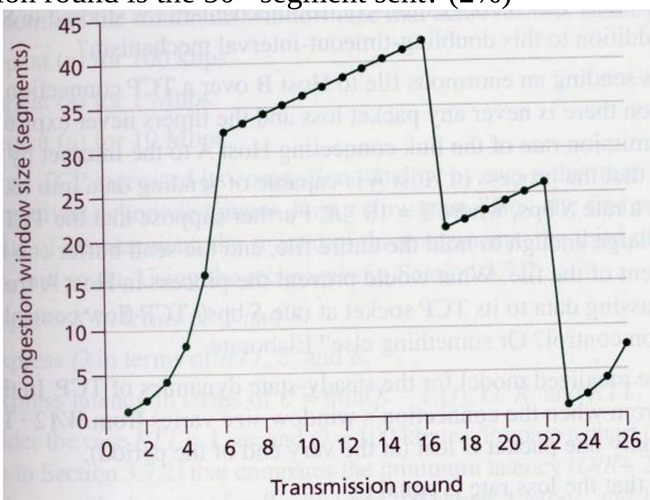




Selective Repeat (4%)
(5) receiver *individually* acknowledges all correctly received pkts (1%)
(6) buffers out-of order pkts (1%)
(7) sender only resends pkts for which ACK not received when timeout (1%)
(8) deliver total in-order pkts to upper layer (1%)

6. (a) Describe how TCP Reno does its congestion control. (8%) (12% total)
(b) Answer and justify the following questions.
After the 22th transmission round, is segment loss detected by a triple duplicate ACK or by a timeout? (2%)
During what transmission round is the 50th segment sent? (2%)



Ans: (8%)
When **CongWin** is below **Threshold** (1%), sender in slow-start phase, window grows exponentially (1%).
When **CongWin** is above **Threshold** (1%), sender is in congestion-avoidance phase, window grows linearly (1%).
When a triple duplicate ACK occurs (1%), **Threshold** set to **CongWin/2** and **CongWin** set to **Threshold** (1%).
When timeout occurs (1%), **Threshold** set to **CongWin/2** and **CongWin** is set to 1 MSS (1%).
(b) (8%)
a. After the 22th transmission round, packet loss is recognized by a timeout. (2%)

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b. During the 1st transmission round, packet 1 is sent; packet 2-3 are sent in the 2nd transmission round; packets 4-7 are sent in the 3rd transmission round; packets 8-15 are sent in the 4th transmission round; packets 16-31 are sent in the 5th transmission round; packets 32-63 are sent in the 6th transmission round; packets 64 – 96 are sent in the 7th transmission round. Thus packet 50 is sent in the 6th transmission round. (說明 1%, 答案 1%，共 2%)

7. Consider the TCP procedure for estimating RTT ($EstimatedRTT^{n} = \alpha \times SampleRTT^{n-1} + (1-\alpha)\times EstimatedRTT^{n-1}$).

(a) Why TCP uses this function? (2%)

(b) Let *SampleRTT^{n}* be the most recent sample RTT, let *SampleRTT^{n-1}* be the next most recent sample RTT, and so on. Express *EstimatedRTT^{n}* in terms of n SampleRTTs if $EsitmatedRTT^{1} = 0$. (要有兩次疊代過程(各 2%)後寫出通式與以 summation 總和符號表示(各 2%) (10% total)

Ans: (a) Exponential weighted moving average => influence of past sample decreases exponentially fast. 據測量出來的 SampleRTT，估計下一次的 EstimatedRTT，用來設定下一次的 Timeout 時間 (2%)

(b)

$$EstimatedRTT^{n} = \alpha \times SampleRTT^{n-1} + (1-\alpha)\times EstimatedRTT^{n-1}$$

$$= \alpha \times SampleRTT^{n-1} + (1-\alpha)\times[\alpha \times SampleRTT^{n-2} + (1-\alpha)\times EsitmatedRTT^{n-2}]$$

$$= \alpha \times SampleRTT^{n-1} + \alpha(1-\alpha)\times SampleRTT^{n-2} + (1-\alpha)^{2}\times EsitmatedRTT^{n-2}$$

$$= \alpha \times SampleRTT^{n-1} + \alpha(1-\alpha)\times SampleRTT^{n-2} + (1-\alpha)^{2}\times$$
$$[\alpha \times SampleRTT^{n-3} + (1-\alpha)\times EsitmatedRTT^{n-3}]$$

$$= \alpha \times SampleRTT^{n-1} + \alpha(1-\alpha)\times SampleRTT^{n-2} + \alpha(1-\alpha)^{2}\times$$
$$SampleRTT^{n-3} + (1-\alpha)^{4}\times EsitmatedRTT^{n-3}$$

$$= ......$$

$$= \alpha \times SampleRTT^{n-1} + \alpha(1-\alpha)\times SampleRTT^{n-2} +$$
$$\alpha(1-\alpha)^{2}\times SampleRTT^{n-3} + ... + \alpha(1-\alpha)^{n-2}\times SampleRTT^{n-(n-1)}$$
$$+(1-\alpha)^{n-1}\times EsitmatedRTT^{n-(n-1)}$$

$$== \alpha\sum_{j=1}^{n-1}(1-\alpha)^{j-1}SampleRTT^{n-j} + (1-\alpha)^{n-1}EsitmatedRTT^{1}$$

$$= \alpha\sum_{j=1}^{n-1}(1-\alpha)^{j-1}SampleRTT^{n-j} (\because EsitmatedRTT^{1} = 0)$$

8. (a) Explain how TCP Fast Retransmit works. (6%) (15% total)

(b) How TCP does its flow control? (6%)

(c) Suppose the TCP sequence number space is of size *k*. What is the largest allowable sender window *w*? (3%)

Ans:

(a) Explain how TCP Fast Retransmit works. (6%)

If sender receives 3 ACKs for the same data, (2%) it supposes that segment after ACKed data was lost (2%): resend segment before timer expires (2%) (6% total)

(b) How TCP does its flow control? (6%)

Rcvr advertises spare room by including value of **RcvWindow** in segments (2%)

Sender limits unACKed data to **RcvWindow** (2%) for guaranteeing receive buffer doesn't overflow (2%)

(c) The sequence number space must be at least twice as large as the window size, $k \geq 2w$. (3%)