只寫答案而沒有解釋說明,扣一半分數

- 1. (a) Explain how TCP Fast Retransmit works. (6%) (26% total)
 - (b) How TCP does its flow control? (6%)
 - (c) What values are used by TCP and UDP to identify their sockets? (6%)
 - (d) Describe four operations to provide reliable data transfer over channels with errors and loss? (8%)
- 2. (a) List TCP seven characteristics (7%)
 - (b) Consider TCP protocols. Suppose the sequence number space is of size *k*. What is the largest allowable sender window *w*? (5%)
- 3. (a)Which tool allows the host running the tool to query any specified DNS server for a DNS record? (2%)
 - (b) How to run the tool in (a) to execute "Please send me the host names of the authoritative DNS for ncue.edu.tw" operation? (4%)
 - (c) How to run the tool in (a) to execute "Please send me the host names of www.ncue.edu.tw, but we want to the query sent to the DNS server *dns.hinet.net* rather than to the default DNS server" operation? (4%)
 - (d) Which tool can be used to show your current TCP/IP information? (2%)
 - (e) How to empty the DNS cache in your host? (2%) (14% 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 300 and 2</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. UDP and TCP uses 1's complement for their checksums. Suppose you have the following three 8-bit byptes: 00100011, 01001110, 01010100. What is the 1's complement for the sum of these 8-bit bytes? Show all work. (要寫出過程 6%) 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%) (10% total)
- 6. List and compare two pipelined transport protocols with these two figures. (寫出 Window=? 與各標號處的動作 10%)



- 7. (a) Describe how TCP Reno does its congestion control. (8%) (12% total)(b) Answer and justify the following questions. (4%)
 - a. After the 16th transmission round, is segment loss detected by a triple duplicate ACK or by a timeout? (2%)
 - b. During what transmission round is the 70^{th} segment sent? (2%)



- 8. 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) If n=3, α =0.2 and *EsitmatedRTT*¹ = 0, what are <u>coefficients of SampleRTT</u>¹ and <u>SampleRTT</u>² when calculating EstimatedRTT? (4%)

只寫答案而沒有解釋說明,扣一半分數

- 1. (a) Explain how TCP Fast Retransmit works. (6%) (26% total)
 - (b) How TCP does its flow control? (6%)
 - (c) What values are used by TCP and UDP to identify their sockets? (6%)
 - (d) Describe four operations to provide reliable data transfer over channels with errors and loss? (8%)

Ans:

- (a) Explain how TCP Fast Retransmit works. (6%)
 If sender receives <u>3 ACKs for the same data</u>, (2%) it supposes that <u>segment after ACKed data was</u> lost (2%): resend segment before timer expires (2%) (6% total)
- (b) How TCP does its flow control? (6%)
 <u>Rcvr advertises spare room by including value of **RcvWindow** in segments (2%)

 <u>Sender limits unACKed data to **RcvWindow**</u> (2%) for guaranteeing receive buffer doesn't overflow (2%)
 </u>
- (c) UDP socket identified by two-tuple: (dest IP address, dest port number) (2%)

TCP socket identified by 4-tuple: (4%) source IP address source port number dest IP address dest port number

(d) sender adds <u>sequence number to each pkt</u> to detect duplicate pkts (2%) receiver uses <u>checksum</u> to detect bit errors (2%) receiver sends <u>ACK with seq # of last pkt received OK</u> (2%) sender <u>waits "reasonable" amount of time for ACK</u>, retransmits if no ACK received in this time (2%)

- 2. (a) List TCP seven characteristics (7%)
 - (b) Consider TCP protocols. Suppose the sequence number space is of size k. What is the largest allowable sender window w? (5%)

Ans:

- (a) (7%)
 - point-to-point: one sender, one receiver
 - reliable, in-order byte steam:
 - pipelined: TCP congestion and flow control set window size
 - send & receive buffers
 - full duplex data: bi-directional data flow in same connection
 - connection-oriented: handshaking (exchange of control msgs) init's sender, receiver state before data exchange
 - flow controlled: sender will not overwhelm receiver
- (b) The sequence number space must be at least twice as large as the window size, $k \ge 2w$. (5%)
- (a)Which tool allows the host running the tool to query any specified DNS server for a DNS record? (2%)
 - (b) How to run the tool in (a) to execute "Please send me the host names of the authoritative DNS for ncue.edu.tw" operation? (4%)
 - (c) How to run the tool in (a) to execute "Please send me the host names of www.ncue.edu.tw, but we want to the query sent to the DNS server *dns.hinet.net* rather than to the default DNS server" operation? (4%)
 - (d) Which tool can be used to show your current TCP/IP information? (2%)
 - (e) How to empty the DNS cache in your host? (2%)

(14% total)

Ans:

(a) nslookup (2%)

- (b) nslookup <u>-type=NS ncue.edu.tw</u> (4%)
- (c) nslookup <u>www.ncue.edu.tw</u> <u>dns.hinet.net</u> (4%)

(d) ipconfig (2%)

- (e) ipconfig /flushdns (2%)
- 4. Draw the flow of the TCP three way handshake to explain its operations. Suppose the <u>initial sequence</u> <u>numbers of the client and the server are 300 and 2</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%)

Ans: Three way handshake:

<u>Step 1:</u> 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. UDP and TCP uses 1's complement for their checksums. Suppose you have the following three 8-bit byptes: 00100011, 01001110, 01010100. What is the 1's complement for the sum of these 8-bit bytes? Show all work. (要寫出過程 6%) 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%) (10% total)

Ans:

+

+

One's complement = 00111010 (2%)

To detect errors, the receiver <u>adds the four words (the three original words and the checksum</u>). <u>If the sum contains a zero, the receiver knows there has been an error</u>. OR <u>check if computed checksum</u> equals checksum field value. If NO, error is detected. (2%)

<u>All one-bit errors will be detected (2%)</u>, but <u>two-bit errors can be undetected</u> (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).

6. 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%)



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%)
- 7. (a) Describe how TCP Reno does its congestion control. (8%) (12% total)(b) Answer and justify the following questions. (4%)
 - c. After the 16th transmission round, is segment loss detected by a triple duplicate ACK or by a timeout? (2%)
 - d. During what transmission round is the 70^{th} 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 16th transmission round, packet loss is recognized by a <u>triple duplicate ACK event</u>. (2%)
- 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 70 is sent in <u>the 7th</u> transmission round. (說明 1%, 答案 1%, 共 2%)
- 8. 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) If n=3, α =0.2 and *EsitmatedRTT*¹ = 0, what are <u>coefficients of SampleRTT</u>¹ and <u>SampleRTT</u>² when calculating EstimatedRTT? (4%)
- Ans: (a) Exponential weighted moving average => influence of past sample decreases exponentially fast. <u>據</u> <u>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%)</u>

(b) As n=3, α =0.2 and EsitmatedRTT¹ = 0, (4%) EstimatedRTT³ = $\alpha \times SampleRTT^{3-1} + \alpha(1-\alpha) \times SampleRTT^{3-2} + (1-\alpha)^2 \times EsitmatedRTT^{3-2}$ = $\alpha \times SampleRTT^2 + \alpha(1-\alpha) \times SampleRTT^1 + (1-\alpha)^2 \times EsitmatedRTT^1$ = 0.2× SampleRTT² + 0.2×(1-0.2)× SampleRTT¹ = 0.2× SampleRTT² + 0.2×0.8× SampleRTT¹