選擇題(20%）DCDAB BACBC

1. 何謂 UDP 中盡力而為的投遞服務（best-effort delivery service）?
   (A)保證區段依序送達目的地，保證區段完整性。
   (B)保證區段依序送達目的地，但不保證區段完整性。
   (C)不保證區段依序送達目的地，但保證區段完整性。
   (D)不保證區段依序送達目的地，也不保證區段完整性。
2. TCP 用以避免任何一筆 TCP 連線用大量的流量塞滿通訊主機之間連結的方式稱為？(A)碰撞控制（collision control） (B)糾纏控制(tangle control) (C)壅塞控制(congestion control) (D)障礙控制(obstacle control)。
3. 有關 TCP 與 UDP 的異同下列何者敘述錯誤？ (A) TCP 是雙向傳輸 UDP 是單向 (B) TCP 可靠性比 UDP 高 (C) TCP 傳送比 UDP 慢 (D) TCP 與 UDP 均提供一個連線導向(Connection Oriented)的傳輸
4. 在 rdt（Reliable Data transfer）模型中，為了避免封包遺失與資料錯誤的情形，除了使用 ACK 機制外，另外在傳送端還多了何項機制？
   (A)增加倒數計時器
   (B)傳送端同時傳送多個封包
   (C)增加暫存區
   (D)持續等待封包送達，不送達不會有下個動作。
5. 下列有關 GBN(Go-Back N)與 SR(Selective Repeat)的敘述，下列何者「錯誤」？
   (A) GBN 單一封包的錯誤，將導致其後的所有在 window 內的封包均需重傳(重新傳送大量封包)。
   (B) GBN 接收端會暫存順序不正確的封包。
   (C) SR 傳送端只重傳沒有收到 ACK 的封包
   (D) SR 傳送端，每一個未確認的封包需要一個計時器
6. 有關用戶端跟伺服器端關閉 TCP connection 的順序何者是正確的？
   (甲)伺服器收到ACK訊息，連線關閉
   (乙)用戶端發送TCP FIN到伺服端
   (丙)伺服器端接收到FIN、以ACK回應，關閉連線，傳送FIN
   (丁)用戶端收到FIN，回應ACK訊息，進入等待計時
   (A)甲乙丙丁(B)乙丙丁甲(C)甲丙乙丁(D)乙甲丙丁
7. AIMD (Additive-Increase, Multiplicative-Decrease) congestion control:中，當網路塞車發生時，cwnd 會如何改變？（A）變為一半（B）變為 2 倍（C）變為 0（D）變為 1MSS
8. TCP Slow Start 中，若每個 RTT 訊息傳送沒有 loss 的話，傳送速率會如何改變？（A）維持原速率（B）每回 RTT 中增加 1MSS（C）每回 RTT 中速率都倍增（D）只在第一回 RTT 中速率倍增，之後維持原速率
9. TCP congestion control 中，當每回 RTT 訊息沒有遺失，但 cwnd 超過 ssthresh 時，對於 cwnd 會採取什麼動作？（A）維持現狀（B）cwnd 都加 1（C）cwnd 都倍增（D）cwnd 降為一半
10. TCP congestion control 中，何時會將網路速率下降？（A）網路速率永遠不會下降（B）當訊息沒有遺失，但 cwnd 超過 ssthresh 時（C）收到 3 次重複 ACK 時（D）以上皆非

問答題（80%）

1. 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: (要寫出計算過程，8%)

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

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

Ans: When we add these first two numbers together, we get:

```
    10100000 10010010
    01111110 10011111      每個 bit 執行二進位加法 (4%)
   --------------------------
  1 00011111 00110001      將進位第 17 bit 加到第一個 bit  (2%)
+                   1
   --------------------------
    00011111 00110010
    11100000 11001101      取 1 補數 (2%)
```

2. (a) Which tool can be used to show your all TCP/IP information? (2%)
(b) How to see these cached records of the DNS cache in your host? (2%)
(c) How to run the tool to query specified DNS server to execute "Please send me the host names of the authoritative DNS for google.com" operation? (2%)
(d) How to run the tool to query DNS server 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.google.com" operation? (2%)
(e) Explain iterated query and recursive query (4%) (12%)

Ans:
(a) ipconfig /all (2%)
(b) ipconfig /displaydns (2%)
(c) nslookup –type=NS google.com (2%)
(d) nslookup *www.ncue.edu.tw* dns.google.com (2%)
(e) iterated query: (2%)
- contacted server replies with name of server to contact

recursive query: (2%)
- contacted server forwards the DNS query to next server and waits for the reply

3. 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 *EstimatedRTT*^{1}*=0*. (要有兩次疊代過程，直到 *SampleRTT*$^{n-2}$，每次各 2%)後寫出通式(以 summation 總和符號表示)(4%) (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)

$$
\begin{aligned}
&\text{EstimatedRTT}^n = \alpha \times \text{SampleRTT}^{n-1} + (1-\alpha) \times \text{EstimatedRTT}^{n-1} \\
&= \alpha \times \text{SampleRTT}^{n-1} + (1-\alpha) \times [\alpha \times \text{SampleRTT}^{n-2} + (1-\alpha) \times \text{EstimatedRTT}^{n-2}] \\
&= \alpha \times \text{SampleRTT}^{n-1} + \alpha(1-\alpha) \times \text{SampleRTT}^{n-2} + (1-\alpha)^2 \times \text{EstimatedRTT}^{n-2} \\
&= \alpha \times \text{SampleRTT}^{n-1} + \alpha(1-\alpha) \times \text{SampleRTT}^{n-2} + (1-\alpha)^2 \times [\alpha \times \text{SampleRTT}^{n-3} + (1-\alpha) \\
&\qquad \times \text{EstimatedRTT}^{n-3}] \\
&= \alpha \times \text{SampleRTT}^{n-1} + \alpha(1-\alpha) \times \text{SampleRTT}^{n-2} + \alpha(1-\alpha)^2 \times \text{SampleRTT}^{n-3} + (1-\alpha)^4 \\
&\qquad \times \text{EstimatedRTT}^{n-3}] \\
&= \ldots\ldots\ldots. \\
&= \alpha \times \text{SampleRTT}^{n-1} + \alpha(1-\alpha) \times \text{SampleRTT}^{n-2} + \alpha(1-\alpha)^2 \times \text{SampleRTT}^{n-3} \\
&\qquad + \cdots + \alpha(1-\alpha)^{n-2} \times \text{SampleRTT}^{n-(n-1)} + (1-\alpha)^{n-1} \times \text{EstimatedRTT}^{n-(n-1)}
\end{aligned}
$$

$$= \alpha \sum_{j=1}^{n-1} (1-\alpha)^{j-1}\,\text{SampleRTT}^{n-j} \; + (1-\alpha)^{n-1}\text{EstimatedRTT}^{1}$$

$$= \alpha \textstyle\sum_{j=1}^{n-1}(1-\alpha)^{j-1}\,\text{SampleRTT}^{n-j} \;\; (\because \text{EstimatedRTT}^{1} = 0)$$ （8%）

4. 針對 193.107.172.1 這個 IP address，(以十進位表示，要寫完整過程） (16%)
   a. 這一個 IP 屬於那個 Class 的網路？以二進位說明(2%) 其所屬的 IP 網路表示法為何？(2%) 可用 IP 範圍？(4%) 共有幾個 IP 可用？(2%) mask 的值為何？(2%)
   b. 手動設定電腦的網路時，除了 default gateway 的 IP 外，至少要設定哪兩個項目的資訊，才可以上網？(4%)

Ans:

a.

193.107.172.1 的二進位表示法為 11000001.XXXXXXXX.XXXXXXXX.XXXXXXXX，由前 3 個 bits 110 可判斷為 Class C 的 IP。(2%)

此 IP 所屬於的 Class C 的網路表示法為 193.107.172.0 (2%)

所有 Host IP 部分的 8 個 bit 的 X 不可以全為 0 或 1，

因此第一個可用 Host IP 為 193.107.172. 00000001 = 193.107.172.1 (2%)

最後一個可用 Host IP 為 193.107.172.11111110 = 193.107.172.254 (2%)

->共有 2^8-2=254 個可用 Host IP (2%)

Mask: 255.255. 255.0 (2%)

b. IP address, subnet mask, (4%)

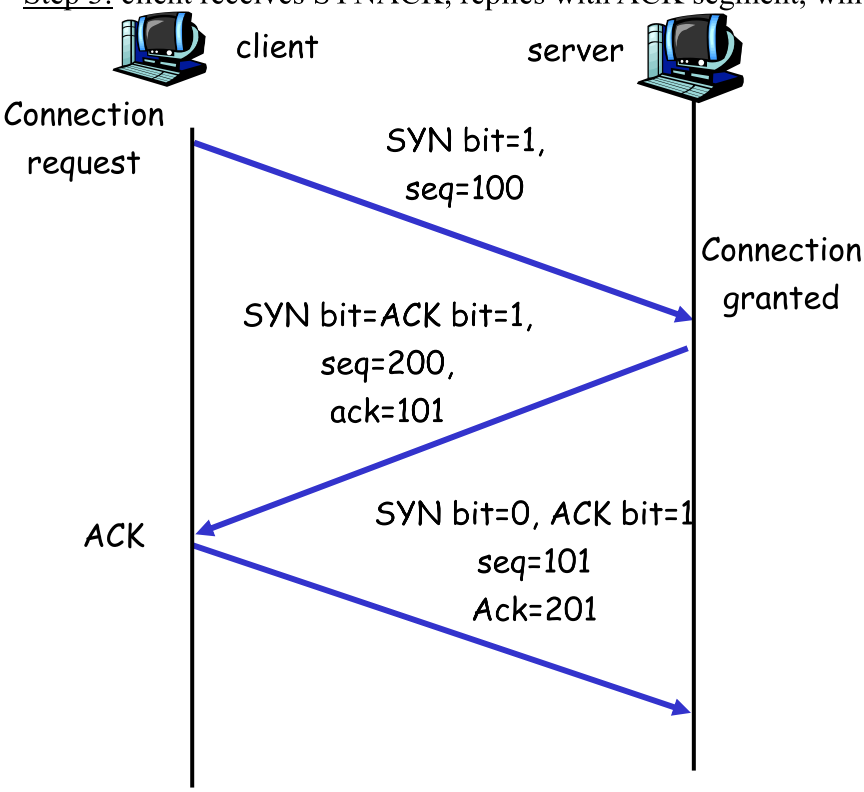
5. Draw and write the flow of the TCP three way handshake to explain its operations. Suppose the initial sequence numbers of the client and the server are 100 and 200, respectively. 必須寫出三步驟的過程，在圖上分別清楚標示出 TCP 必要的 flag, sequence number, and ACK number. (10%)

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

6. List and compare two pipelined transport protocols with these two figures. (左右圖分別是哪一個 pipelined transport protocols？(2% each)(寫出 Window=? 與各標號處的動作 10%) (14%)

Ans:

左圖是 Go-back-N (2%) (6%)

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

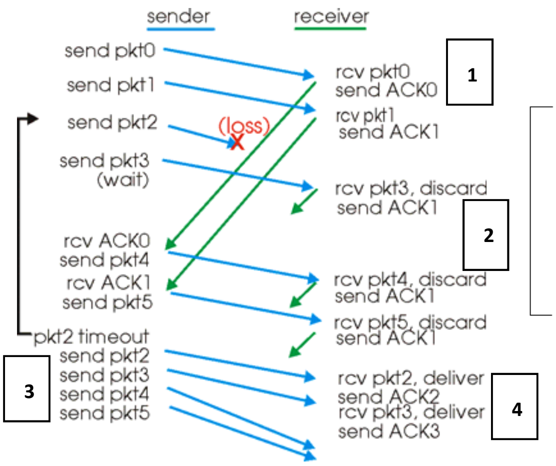
(1) ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq # (1%)

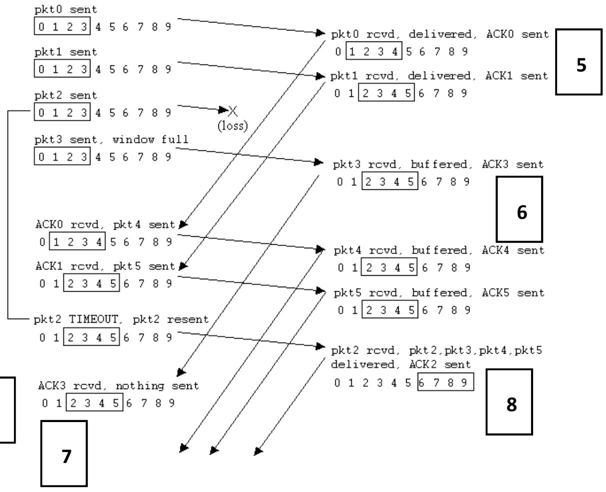
(2) out-of-order pkt:

- 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%)





右圖是 Selective Repeat (2%) (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) Explain how TCP Fast Retransmit works. (3%) (10% total)

(b) How TCP does its flow control? (3%)

(c) What values are used by TCP to identify their sockets? (4%)

Ans:

(a) Explain how TCP Fast Retransmit works. (3%)

If sender receives <u>3 ACKs for the same data</u>, (1%) it supposes that <u>segment after ACKed data was lost</u> (1%): <u>resend segment before timer expires</u> (1%) (3% total)

(b) How TCP does its flow control? (3%)

<u>Rcvr advertises spare room by including value of **RcvWindow** in segments</u> (1%)

<u>Sender limits unACKed data to **RcvWindow**</u> (1%) for <u>guaranteeing receive buffer doesn't overflow</u> (1%)

(c) TCP socket identified by 4-tuple: (4%)

source IP address

source port number

dest IP address

dest port number