[12.4]
A multicast router would forward datagrams to a given multicast address, say 224.0.64.32 within its attached physical network. What IP address and link-layer address should be specified in a packet?

IP:224.0.64.32
MAC:01:00:5E:00:40:20

[12]
If we want to broadcast a data to all hosts attached on a common physical network, we can use link-layer (ff:ff:ff:ff:ff:ff) or IP-layer (224.0.0.1) broadcast address. What are the differences between both?

ff:ff:ff:ff:ff:ff情況主機會由網卡接收此broadcast，
224.0.0.1情況要網卡支援multicast才會接收

3. [12]
Why do we need to have the multicast MAC address in data-link layer?

雖然使用broadcast MAC address也能達到相同效果，卻會增加ethernet的流量。

20. [13]
Please compare source-based tree and group-based tree. Then given an example protocol for each multicast approach.

Source-based tree: 對每個multicast group建構一shortest path tree，例如：MOSPF。
Group-based tree: 只有一個tree由一center core為root，負責管理及routing，例如：CBT。

1. [13]
In IGMP, is it necessary for multicast router to know how many hosts join in a multicast group? Please justify your answer.

否,router只需知道group裡面還有沒有host。Router發出IGMP Request後，只要有一部主機回覆IGMP即代表此group存在，因此group主機數目不是router所在意的。

16. [13]
For an IGMP report message, what are the differences if the field of the destination IP address is the multicast router, group address, or 224.0.0.1? Are they all correct?

(1)
multicast router時，群組其它成員仍舊做出回應，
group address則可以取消群組其他成員的回應動作，

224.0.0.1為子網路所有系統，最後效果同傳給group address
(2)
yes

16. [14]
(1) What is the point query in DNS?
(2) If the resolver receives the IP address 140.117.176.22, what domain name will be sent to the local DNS and then resolved?

(1) 用來反查IP所對應的網域名稱
(2) 22.176.117.140.in-addr.arpa

18. [16]
Please describe the BOOTP protocol operation.

當client開機使用BOOTP在data-link layer傳送一個destination IP為255.255.255.255，source IP為0.0.0.0的封包去詢問自己的IP，BOOTP server利用ARP cache以unicast傳送給client 或或者broadcast廣播給client。

19. [15]
What is the sorcerer’s apprentice bug?

在TFTP中，由於延遲的ACK導致接下來傳送的封包都傳送兩次，而且ACK也會出現兩次。
both side time out and retransmit

1.
If a host A receive two SYN packets from the same port from remote host B, the second may be either a retransmission of the original or else, if B has crashed and rebooted, an entirely new connection request.
(a) Describe the difference as seen by host A between both cases.
(b) Given an algorithmic description of what the TCP layer needs to do upon receiving a SYN packet. Consider the duplicate/new cases above, and the possibility that nothing is listening to the destination port.

(a)
Host A根據接收到SYN packet上面的sequence number，如果sequence number跟第一個SYN packet相同，則代表這是一個retransmission；如果sequence number不同代表是一個新的連線。
(b)
如果host A收到相同sequence number的SYN packet代表retransmission，host A傳回

sequence number+1的ACK packet。
如果host A收到不同sequence number的SYN packet代表新的連線，host A傳回sequence number+1的ACK packet。
如果host A收到目前沒有聆聽的destination port，TCP discard這個packet。

2.
You are designing a reliable byte-stream protocol that uses a sliding window (like TCP). This protocol will run over a 10-Gbps network. The RTT of the network is 140ms, and the maximum segment lifetime is 60 seconds. How many bits would you include in the AdvertisedWindow and SequenceNum fields of your protocol header?

AdvertisedWindow:
10G * 0.14s = 1.4Gbits = 175Mbytes
$2^x \geqq 175 * 2^{20}$ → $x \geqq 27.xxx$ → Advertised Window = 28bits

SequenceNum:
10G * 60s = 600Gbits = 75Gbytes
$2^x \geqq 75 * 2^{30}$ → $x \geqq 37.xxx$ → Advertised Window = 37bits

11. [17]
(1) Unlike UDP, TCP is a connection-oriented service. Two applications using TCP must establish a TCP connection with each other before they can exchange data. For a TCP connection, are all TCP segments delivered over the same path? Explain that.

(2)Consider FTP as an example of TCP applications. Is it enough to setup only one TCP connection between client and server? Justify your answer.

(1)
No, the delivery path is handled by routers in IP layer.
(2)
No, need two TCP connections, one for data connection and another for control connection.

22. [17]
Why is TCP not suitable for real-time multimedia traffic?

Because of TCP’s features, there cannot allow retransmission of packets for real-time multimedia.

6. [18]
In TCP state transition diagram, there is a TIME_WAIT state (also called 2MSL wait

state). Why does TCP need it?
27.
Please explain the 2MSL Wait State in TCP state transition diagram. What is the purpose for it?

進行active close的機器等待另一方(passive close)送出FIN segment，當active close機器收到FIN segment後會回傳ACK並等待2MSL的時間，因為passive close機器會再沒有收到ACK時又送出另一個FIN segment，而新的FIN segment會在2MSL時間逾時前到達active close機器，這樣active close機器才能重送ACK，以確保passive close機器到ACK。

13. [18]
Assume that during 3 way handshake connection set up in TCP/IP a memory error occurs at one of the intermediate routers. This error is particularly vicious in that it modifies the source address in the IP header. Can the protocol recover from this error? Explain exactly how.

可以，改變了來源端的IP address，並不會影響整個運作，當來源端沒有收到期待的ACK值則會重送，如果其它電腦收到非期待的ACK值(假設其它電腦之間也存在相同的socket pair)，則電腦會丟棄這個封包並回覆期待的ACK值。

14. [18]
Except persist timer, show another three timers maintained by TCP?

persist timer
Retransmission timer
Keekalive timer
2MSL timer

12. [18.18]
When the TCP module receives incoming TCP segments from IP module, are they demultiplexed to the applications just based on the destination TCP port number? Explain that. (You must answer YES or NO first)

[Exercise 18.18]
No. Incoming data segments are demultiplexed using the source IP address, source port number, destination IP address, and destination port number. For incoming connection requests we saw in Section 18.11 that a TCP server can normally prevent connections from being accepted based on the destination IP address.

5. [18]
Is it possible for the application of TELNET to have simultaneous open in TCP? Explain your reasons.

不可能，TELNET屬於client-server的架構，server是被動接受服務，無法進行active open。

4. [19]
Please describe the characteristic of TCP interactive data flow. What algorithm are been using to improve the performance of TCP for the service of interactive data flow? And how do they do?

interactive data flow : 小封包 ex. telnet
[CH19.3]
Delay ACK: 使用delay ACK讓發送ACK的時間能夠延後，如果剛好有資料與ACK的傳輸方向相同，就將資料與ACK一同送出去(the ACK piggyback with the data)。
[CH19.4]
Nagle Algorithm: outstanding packet只能存在一個，也就是說，當送出一個封包之後必須等到

7. [19]
What is the self-clocking behavior of TCP?

利用ACK的回傳速率來決定sender端的發送速率，稱為self-clocking，即ACK多快sender就送多快。送出segment後，等收到ACK才送下一個segment。

4. [20.8]
TCP provides the urgent mode. What is it? What does the content of the urgent pointer mean in the TCP header?

(1)
Urgent mode用來讓sender送出一段資料給receiver，而這段資料不需要按照順序，馬上能被receiver給處理。
(2)
Urgent pointer用來指出segment的資料中，urgent data的結束位置和普通資料的起始位置。

13. [20]
Which parameters of TCP can control the amount of data flow injected into the network? They are imposed by either the sender or the receiver. Please indicate these.

Sliding window is imposed by receiver
congestion window(cwnd) is imposed by sender.

10. [21.6]
Congestion avoidance is a way to deal with lost packets. What can indicate the packet loss?

Receiving of ACK packet is timeout or the receipt of duplicate ACKs.

8. [21.11]
What is the repacketization in TCP protocol?

當TCP封包逾時並重送的情況下，TCP可以傳送比之前更大的segment來增加效率而不用重傳相同大小的segment。

14. [21]
Receipt of the duplicate ACKs tells us a packet may be loss. The TCP does not perform slow start, but still increase the data flow (i.e., increase cwnd). Please describe the reasons.

lost segment, or reordering of segments, 超過3個duplicate ACKs可能是lost segment,
不等 timeout就直接進行retransmission,(fast retransmit)
fast recovery, congestion avoidance

11. [21]
Describe the purpose of the variable ssthresh (slow start threshold) in TCP. When should ssthresh be updated?
15.
Congestion avoidance and slow start are independent algorithms with different objectives. In practice they are implemented together. Please describe when TCP performs congestion avoidance and when it does slow start.

The variable ssthreash is used for congestion avoidance.
When cwnd is below ssthresh, performs slow start.
When cwnd is above ssthresh, performs congestion avoidance.
When congestion occurs, ssthresh is updated to a half of min(cwnd, win).

1. [21 22]
TCP is reliable for data packets. How about for ACK packet? Please explain your answer.

ACK封包的傳送可能遺失在網路上，
TCP利用retransmission timer來重發逾時的ACK
或利用persist timer來要求對方傳送ACK packet。

9. [22]
What is the silly window syndrome (SWS)?

Sender和receiver緩慢的處理資料，Sender和receiver之間以小封包來傳送資料，而不是以一

個完整的segment size封包大小來傳送，稱為silly window syndrome。

17. [22]
A sender on a TCP connection that receives a 0 advertised window periodically probes the receiver to discover when the window becomes nonzero. Why would the receiver need an extra timer if it were responsible for reporting that its advertised window had become nonzero (i.e., if the sender did not probe)?

當receiver的window size不為0時，receiver會傳送ACK來通知sender，如果ACK遺失則sender依然無法傳送資料且receiver可能也不會在重傳ACK，造成雙方都在等待對方的資料。

21.
Consider the operation of NAT (Network Address Translation). Does it violate the TCP reliability? Please explain your answer. If your answer is positive, can you fix this problem? (you must answer “yes” or “no” before explaining. Otherwise you cannot get any points in the problem).

26. [22 23]
Why does TCP need persist timer and keepalive timer? Please explain for each.

Persist timer: 當sender因為receiver的window size為0而等待receiver更新它的window size，receiver則送出更新window size的ACK然後等待sender端傳送資料，如果更新window size的ACK在網路上遺失，則會造成雙方互相等待對方資料的情況(deadlock)，所以sender需要persist timer去詢問receiver的window size。

Keepalive timer: 當server和client之間沒有資料傳輸，讓server可以發送小封包知道client是否還存活的計時器，避免server浪費資源紀錄不存在的client。

6. [24]
Consider a network link connecting host A and B. Let RTT be the average round trip time. If TCP is running over the network, please calculate the bandwidth limit between A and B. You must consider TCP window scale option, in which one-byte shift count is set to be the maximal value 14.
3.
Please show the upper bound of TCP bandwidth (bytes/sec) if no window scale factor is considered (i.e., only 16-bit window size).

2^16 = 1Gbytes
capacity = bandwidth * RTT

2^16 = bandwidth * RTT
bandwidth = 2^16 / RTT
bandwidth取決於RTT

15. [24]
In the long fat network (LFN), the 16-bit window size may not be enough to fill the pipe. Additionally, there may be the problem of wrapped sequence numbers. How does TCP solve these problems?

使用window scale option解決window size的大小；
使用timestamp解決重複的sequence number。

6. [24]
Can timestamp solve the retransmission ambiguity problem? Explain that.

可以，因為timestamp隨著時間改變，並且不會與之前的timestamp重複，所以sequence number與timestamp可以構成唯一的識別碼，以解決the retransmission ambiguity problem。

28. [24.6]
Please describe TCP PAWS (Protection Against Wrapped Sequence Number).

在高速網路的情況下，為了避免因重傳造成舊的segment使用目前的sequence number，使接收端誤以為舊的segment是下一個要接收的區段；PAWS使用timestamp option，讓一個segment的識別號碼為sequence number加上timestamp。

7. [24]
Packet loss in a LFN (Long Fat Network, i.e., a network with large bandwidth-delay products, i.e., large capacity) can reduce throughput drastically. Could you explain the reason?

要是lose more than one packet, 啟動slow start,
因window可達最大,啟動slow start 也下降的最大

10. [24]
(1) In TCP congestion avoidance algorithm, we want to increase congestion window size (cwnd) by at most one segment (i.e., maximum segment size, MSS) each round-trip time (RTT). To estimate a RTT, TCP uses the time to send and receive acknowledgements for the data in one window. Of course, TCP does not wait for an entire window of data to be sent and acknowledged before increasing cwnd. Instead, it adds a small increment to cwnd each time an acknowledgement arrives. The small increment is chosen to make the increase average approximately on MSS over an entire window. Please show the small increment to be MSS^2 /cwnd.

(2) Continue part(1). Explain why computing this increment each time an ACK arrives may not result in the correct increment. Given a more precise definition for this increment. (Hint: A given ACK can acknowledge more or less than one MSS's worth of data)

(1)
1 RTT → 1 segment
1 RTT → N ACK
1 ACK → 1/N segment (N = cwnd/MSS)
1 ACK → MSS/cwnd * segment = MSS/cwnd * MSS = MSS^2/cwnd
(2)
因為不一定每次都是MSS, 所以實際上會比較小

29.
Please describe the multiple streams and multihoming services of SCTP.

Multiple streams: if one of the streams is blocked, the other streams can still deliver their data.
Multihoming services: SCTP allows sending and receiving host defining multiple IP addresses in each end.

30.
TCP uses BYTE as the data unit. How about SCTP?

SCTP uses data chunk as the data unit and uses a transmission sequence number to number the data chunks.