1. Suppose a network uses 8-bit addresses. A router is configured with the following forwarding table:

Prefix	Interface
111	0
1101	0
1100	1
101	Return ICMP destination unreachable
Otherwise	2

a. For each of the three interfaces, give the range of matching addresses and the number of addresses. Assume that 00000000 and 11111111 are also valid addresses. [4]

Interface 0: 111----- : 2^5 addresses = 32 } 224–255, 48 addr (1)

208–255

1101---- : 2^4 " = 16 } 208–223

Interface 1: 1100---- : 2^4 " = 16 addr 192–207 (1)

Interface 2: 100----- : 2^5 " = 32 } 128–159, 160 addr (2)

0------- : 2^7 " = 128 } 0–127

128+64+32 = 224–255 (1)

192

128+64+16 208–223 (2)

(1) for correct range identification

(1) for correct answer

255
32
223

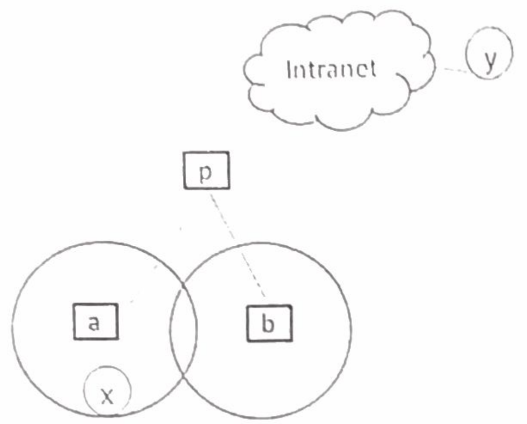
b. Prefix 101 indicates unallocated address space under control of this ISP. A customer organization requests for 4 addresses. Give an example prefix obtained from the unallocated block that can be added to the forwarding table. [1]

Requirement of 4 addresses ⇒ first 6 bits fixed.

A possible prefix is $101\underbrace{000}_{\text{can be anything}}$ (1)

Q2.

2. Here, 'a' and 'b' are wireless access points that work only up to the link layer (ie. they are like switches and do not have IP addresses), 'p' is a switch, and 'y' is a server from which host 'x' is downloading data over FTP.



a. Will the IP address of 'x' change if it moves from 'a' to 'b'? Explain. [1]

No, IP address will not change. Would have been allocated by a DHCP server running in the network which would not differentiate based on AP. 1

b. Why are there chances for the FTP transfer to get affected? [1]

The switch 'p' would have self-learned that 'x' is an link connecting to 'a'. Hence, when 'x' moves, the mapping at the switch may continue pointing to the old access point. 1

c. Describe any one technique that can be used to ensure a smooth handoff? [2]

A make before break technique can be used, where 'x' maintains a connection to both 'a' and 'b' when in range with both of them. It will continue to pick up data sent by the switch to 'a' but send data through 'b' so that 'p' updates the table. If it does not pick up data from 'a', then TCP ~~may~~ will infer a loss for that data.

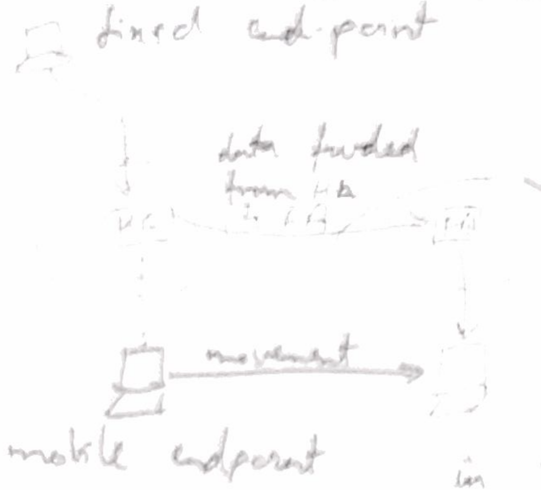
1/2 each for main point

b. If the APs 'a' and 'b' operate like routers at the IP layer and they also run their own DHCP servers which gives IP addresses to clients attached to them, what will happen to the FTP transfer and why? [1]

FTP transfer will break because the IP address for 'x' will change. Hence the four tuple identication for flows will break. 1

½ marks each

c. How can mobile IP be used here? Explain the working using the Home Agent/Foreign Agent terminology in the mobile IP protocol [2]



The home network will continue to receive data sent to the mobile endpoint at its old IP addr, & will encapsulate it with the care-of address for the foreign agent in its new network. Therefore the fixed end-point will not witness any change in the IP for the mobile endpoint.

1. If the time taken by 'x' to move from 'a' to 'b' is large, then despite having mobile IP, the FTP TCP connection may still break because of TCP timeouts. In FTP therefore, a restart option is available, so the transfer can always resume without having to retransmit data that has already been received. Yet, there can be a significant performance drawback in TCP restarts. Explain why. Hint: Think about what happens during a TCP connection initiation [1]

When a new TCP connection is started, it will go through the same slow-start phase to ramp up, because any congestion control parameters discovered earlier will not be carried forward. 1

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g. Suggest a simple feature-addition in TCP to avoid this performance degradation. Assume that congestion is not a problem in the wireless portion of the network. [1]

TCP options can be used to link a new TCP session with an older TCP session, so that congestion control parameters discovered earlier can be reused. 1

3. The figure below shows a set of customers connected to different ASes. BGP is used as the AS level routing protocol.

a. Customers A and B advertise their prefixes to AS-1. After address aggregation, what prefix would AS-1 advertise to AS-2 and AS-3? [1]

A: 202.10.$\underbrace{00100000}_{\text{in binary}}$...

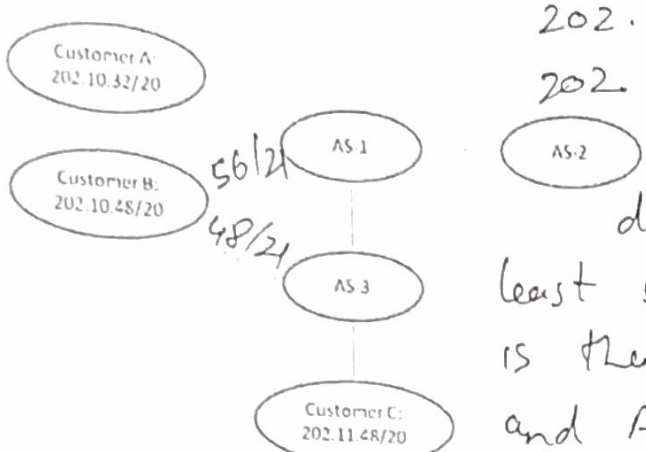
B: 202.10.$\underbrace{00110000}_{\text{first 4 bits are fixed in each (20-16)}}$...

} Aggregation will require first 3 bits to be fixed : 202.10.001-----

⇒ 202.10.32/19

1

b. Customer B now starts to draw its service from AS-3 as well, in a multihomed manner. It advertises 202.10.56/21 to AS-1 and 202.10.48/21 to AS-3 to distribute the load. What prefixes would AS-3 now advertise to AS-1? What prefixes would AS-1 advertise to AS-2? [3]



AS-3 to AS-1 adv. will
(1) → try to aggregate if possible:
202.11.48/20 and
202.10.48/2[illegible]. However
since the prefixes
differ not at the
least significant end, aggr
is therefore not feasible,
and AS-3 will advertise
both prefixes as such.

AS-1 to AS-2 advertisement will try to aggregate all prefixes: 202.10.32/20, 202.10.56/21, 202.10.48/21, 202.11.48/20

(1)

202.10.48.20

(1)

202.10.32/19

Hence, AS-1 will adv. 202.11.48/20 and 202.10.32/19 [1]

c. In continuation of part (b), what path would a packet follow, originating at 202.11.48.1 and destined to 202.10.57.1?

202.11.48.1 is located in customer C's network. 202.10.57.1 is located in customer B's network and has a match with prefix 202.10.56/21 advertized ~~[illegible]~~ by B to AS-1. Hence, the packet will go from C → AS-3 → AS-1 → B

(1)

4. Show that in a steady state TCP connection working in the congestion avoidance phase, the throughput ~ $\frac{1.22 \times MSS}{RTT \times \text{sqrt}(L)}$ where RTT is the round trip time, MSS is the maximum segment size, and L is the loss rate

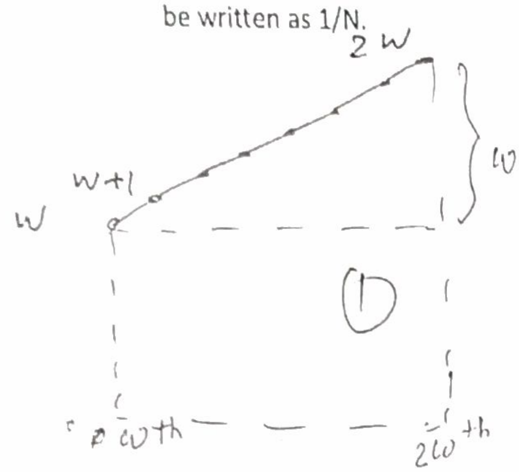
Note that in the congestion avoidance phase, all losses are assumed to be detected through fast retransmits and not timeouts, hence the congestion window rises additively and falls to half its value in a saw-tooth pattern.

$\frac{1}{2} \times w(w) + w \times w$

$= \frac{3}{2} w^2 = \frac{1}{L}$ $\quad w = \sqrt{\frac{2}{3L}}$

Q4.

Hint: If N packets are sent between two consecutive packet loss events, assume that the events happen due to the loss of only one packet in each event, hence the loss rate can be written as 1/N. [5]



Window goes from W to 2W between two consecutive losses. Number of pkts sent = W + W+1 + W+2 + ... 2W

$= W^2 + \frac{W(W+1)}{2} = \frac{3W^2}{2} = N$

2 { Time taken = W . RTT ——— 1
{ Total data sent = $\frac{3W^2}{2}$. MSS — 2

1 Throughput = $\frac{3W^2}{2} \cdot \frac{MSS}{W \cdot RTT}$ $\qquad \frac{②}{①} = \frac{\text{data sent}}{\text{time taken}}$

$= 3/2 \frac{W \cdot MSS}{RTT}$

$= 3/2 \cdot \sqrt{\frac{2}{3} N} \cdot \frac{MSS}{RTT}$ $\qquad \left(W = \sqrt{\frac{2}{3} N} \right)$

$= \frac{MSS}{RTT} \times 1.22 \sqrt{N}$ $\qquad \left(\frac{3}{2} \times \sqrt{\frac{2}{3}} = \sqrt{\frac{3}{2}} = 1.22 \right)$

$= \frac{1.22 \times MSS}{RTT \sqrt{L}}$ $\qquad \left(L = \frac{1}{N} \right)$

1
(correct form)

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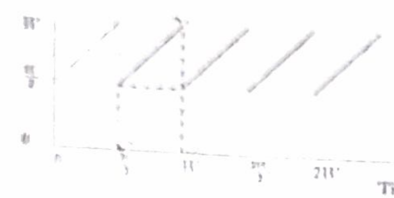
Q4 5) **Throughput for a TCP connection in congestion avoidance phase :**

Throughput for a TCP connection is the rate of data delivery given by -

$$Throughput = \frac{Data\ Sent}{Time\ Taken}$$

The window size increases linearly with each RTT in a TCP connection. Assuming a loss rate of 'L', one can find the number of packets sent on average during two consecutive losses by finding the area under the Window Size vs RTT graph.



— (1)

$$Packets\ sent = \frac{1}{L} = \left(\frac{W}{2}\right)\left(\frac{W}{2}\right) + \left(\frac{1}{2}\right)\left(\frac{W}{2}\right)\left(\frac{W}{2}\right) = \frac{3}{8}W^2$$

$$\therefore Solving,\ W = \sqrt{\frac{8}{3L}}$$

(2)

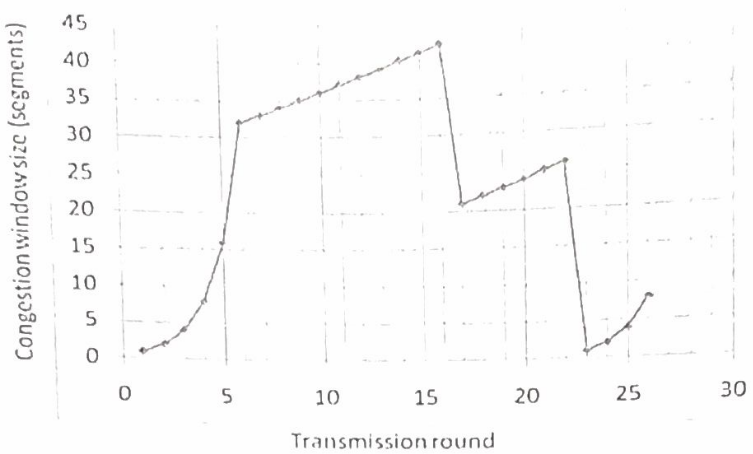
Hence, we can also claim the following.

$$Data\ sent\ between\ losses = MSS \times \frac{3}{8}W^2 \qquad Time\ Taken = RTT \times \left(\frac{W}{2}\right)$$ — (1)

$$\therefore Throughput = \frac{MSS \times \frac{3}{8}W^2}{RTT \times \frac{W}{2}} = \frac{MSS \times \frac{3}{4} \times \sqrt{\frac{8}{3L}}}{RTT} = \frac{\sqrt{\frac{3}{2}} \times MSS}{RTT \times \sqrt{L}}$$

$$\Rightarrow Throughput = \frac{1.22 \times MSS}{RTT \times \sqrt{L}}$$ — (1)

5. Consider the following plot of window size as a function of time for TCP Reno, and answer the following questions.



a. Identify the intervals of time when TCP slow-start is operating. [2]

Rounds 1 – 6 and 23 – 26
1 1

b. Intervals when TCP congestion avoidance is operating. [1]

Rounds 6 – 23 1

c. During the 16th transmission round, is segment loss detected by a triple dup-Ack or by a timeout? [1]

Since the congestion window is halved, it shows that the loss was detected through a triple dup-ack 1

d. How is segment loss detected during the 22nd transmission round? [1]

Here, since the congestion window reduces to 1, the loss was detected through a timeout 1

e. What is the value of slow-start threshold (*ssthresh*) at the 18th transmission round? Explain your answer. [2]

Upon a loss, the ssthresh is set to half the value of the congestion window at the loss event. Here, the ssthresh will be 42/2 = 21.
1 1

f. During what transmission round is the 70th segment sent? Explain your answer. [2]

70th segment will be sent in which round?

$\underbrace{1+2+4+8+16+32}_{\text{64 segments sent in 5 rounds}} + \underbrace{33}_{\text{93 segments sent by round 6.}} \ldots$

Hence 70th segment will be sent in 6th round

6. Short-answer questions

a. If an IP fragment of a large UDP or TCP segment gets lost, the IP fragment is retransmitted. True/false? [1]

False. IP fragments are not re-transmitted.

b. IP multicast uses a special set of IP addresses to identify multicast groups. True/false? [1]

True. It assigns addresses from an IP. address space designated for multicast addr.

c. Name 2 ways that are currently in popular use to tackle IPv4 address scarcity (other than migration to IPv6). [2]

Tunnelling

NAT – Nw. addr translation, for re-using pvt. IP addresses across networks.

Virtual hosting – to have multiple websites running on the same server.

d. The TCP AIMD algorithm has an interesting property of allocating bandwidth to flows in proportion of their demand. True/false? [1]

False. Bw allocation by AIMD happens inversely to RTT.

e. What type of a DNS record is used to identify the mail server for a domain? [1]

MX record.

f. RIP is a link state protocol. True/false? [1]

False. Distance vector protocol.

g. Exponential random backoff algorithms in wired and wireless link layers help estimate the amount of load in the network. True/false? [1]

True. Nodes eventually end up backing off to the extent of load in the

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g. Exponential random backoff algorithms in wired and wireless link layers help estimate the amount of load in the network. True/false? [1]

True. Nodes eventually end up backing off to the extent of load in the nw. which causes collisions.

h. What would you need to do to run a server behind a NAT? [1]

Port forwarding through a static mapping, since on the fly mappings will not exist in this case.

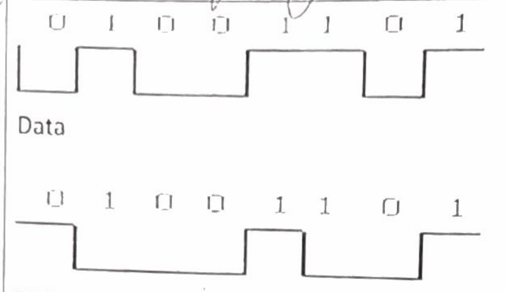
i. Poison reverse is used to solve what kind of problems in distance vector protocols? [1]

Count-to-infinity problems, caused when nodes do not reveal to their neighbours that they are fwding through the same neighbours themselves or looping or infinite looping.

7. Given below is a table for 4B/5B encoding, and an example to help you recall NRZI.

4-Bit Data Symbol	5-Bit Code
0000	11110
0001	01001
0010	10100
0011	10101
0100	01010
0101	01011
0110	01110
0111	01111
1000	10010
1001	10011
1010	10110
1011	10111
1100	11010
1101	11011
1110	11100
1111	11101

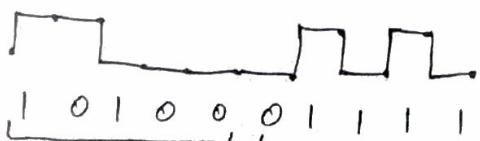
Table 2.4 4B/5B encoding.



a. Encode the following sequence of bits using 4B/5B with NRZI.
00100111 [2]

0010 → 10100 : First 5 bits after encoding
0111 → 01111 : Next 5 bits after encoding } ①

NRZI is change upon a 1. Hence:



Note: They could also show an inverted signal if they started inversely

$\boxed{\tfrac{1}{2} \text{ each}}$

b. Why is 4B/5B used with NRZI? [2]

4B/5B encoding guarantees that in any seq. of bits there will not be more than 3 consecutive zeroes. NRZI already ensures that each one will result in an inversion, which will help re-sync the clock (~~[illegible]~~ for clock recovery). Therefore 4B/5B with NRZI will ensure clock recovery after no more than 3 bits.

c. If the bits can be written on to the medium at a rate of 10 Mbps (1 Mbps = 10^6 bits per sec), what is the physical length of one bit on the medium? Assume c = 2 x 10^8 m/s. [1]

Time for 1 bit $= \frac{1}{10 \times 10^6}$ sec.

Distance (length) of a bit $= 2 \times 10^8 \times \frac{1}{10 \times 10^6}$

$= 20$ m

d. How many bits are on the medium at any time if the propagation delay of the link is 100 ms. [1]

Prop. delay $= 100 \times 10^{-3}$ sec.

Prop. distance $= 2 \times 10^8 \times 100 \times 10^{-3}$

$= 2 \times 10^7$ m

Number of bits $= \frac{2 \times 10^7}{20} = 10^6$ bits at any time on the medium.