

# **Tutorial - 6**

## **Additional Solved Example for Mid-term Exam**

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# Solved Example - 1

(1) (4 pts) A CIDR lookup Table is given below.

IP address	output port
144.45.24.12/22	port 1
144.45.14.10/23	port 2
144.45.28.0/23	port 3
144.45.12.0/24	port 2
0/0	port 1

An coming IP packet with the destination address:

**144.45.31.22**

What is the output port for this packet? Explain.

# How to think?

## Points to note :

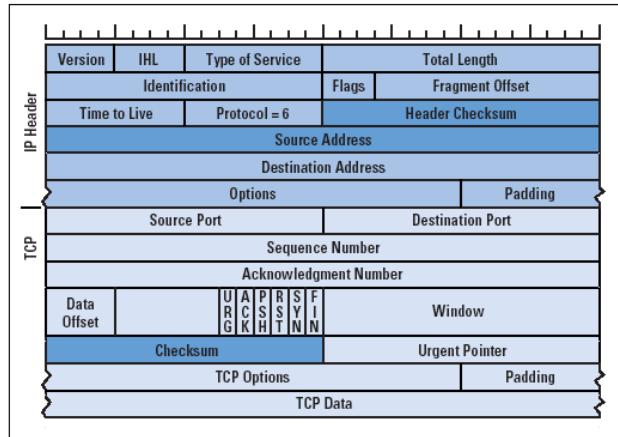
1. IP Address in such a format “144.45.24.12/22” denotes a range of IP addresses
2. All the IP addresses in the example are 144.45.\*.\*
3. Write out all the IP addresses & masks. No need to convert the 1st two numbers to binary (because they are common in all the IPs here).
4. 0/0 stands for default

# Self Practice - 1

(5) (4 pts) Modern operating systems allow us to set up filters to block incoming traffic. A filter specifies i) which bits that need to be checked and ii) their corresponding bit values. If matched, the packet will be dropped. Suppose we like to block all incoming TCP connections from the network 194.64.35.0/22. What are the bits in the TCP/IP header we need to check? and what are their values that correspond to a blocking condition?

TCP/IP Packet for your reference

Figure 1: TCP/IP Header  
Fields Altered by NATs  
(Outgoing Packet)



Another point to note is, for incoming IP packets, ACK=0, SYN=1.

Also check the “Type of Service”, this should be 6 if the packet has TCP data.

# Self Practice - 2

(3 pts) A filter type firewall blocks some applications based on certain bits in the protocol. Suppose a company uses a proxy server (such as the one shown in Problem 4) to block all HTTP connections originating from the company. What fields in the IP/TCP header and which bits of these fields will be examined by the proxy server? Assume that all HTTP servers use TCP port 80.

# Solved Example - 2

**(3)(6 pts)** A data link protocol uses a sliding window for error recovery. Assume the total number of IDs is 8 (i.e. 0, 1, .., 7). The size of the sending window is 5 and the size of the receiving window is 3. At the beginning, the sending window is 0 1 2 3 4 and the receiving window is 0 1 2. A while later, the sender has sent out 4 packets 0 1 2 and 3, but the receiver only receives 0 and 1.

- (a)** All ACKs sent back by the receiver are lost. What is the current sending window and receiving window?
- (b)** Suppose later ACK 1 arrives at the sender. What is the sending and receiving window after that?

# How to think?

Things you should know to solve this question

1. Know what is sliding window protocol (ref to tutorial - 4, see the video)
2. Draw a diagram & visualize what is happening

# Solved Example - 3

(2 pts) Assume that the unit of the TCP window is byte. The RTT between a sender and a receiver of a TCP channel is about 100 ms. What is the maximum bandwidth achievable for this TCP channel?

Points to note:

Unit of TCP window is byte => It is safe to take 1packet as 1byte

# How to think?

- RTT is 100ms. This means it takes 100 ms to send a packet and receive its acknowledgment.
- This also means, if we send 1 packet at a time it takes 100ms to send transfer it from sender to receiver.
- Need to calculate rate, ie. number of packets per sec. Thus 10 packets/sec.
- Now if we send  $N$  packets together we can have  $10N$  packets/sec. Seems like there is no upper bound on how much data can be sent. This is ofcourse not true.
  - maximum window size in a IP packet is  $2^{16}-1$  (65535)
  - Also the rate depends upon congestion in the network.

# Solved Example - 4

(20 pts.) Hosts A and B are separated by 20 kilometers and all the links connected these two hosts are 1Gbps. A 0.1M bits message is to be sent from A to B. Suppose the propagation speed is  $2 \times 10^8$  m/s. The processing delay of each router is 5  $\mu$ s and the links are uncongested.



- a. Find the propagation delay from A to B. (2pts)
- b. Suppose the whole message is sent as a big packet and the switches use store-and-forward packet switching, what is the total time to move the message from A to B? (3pts)
- c. Now suppose the message is segmented into 50 packets  
How long will it take to move the first packet from A to B? (2pts)

# Things to know & how to think

## Understand the terms

**Propagation time** - Time to move packet over the full distance

**Transmission time** - Time to load the packet onto the network link

**Nodal processing** - Time taken by routers to process

**Queuing delay** - Extra time taken due to congestion.

Generally depends on traffic intensity on the router (ratio of arrival rate and departure rates).

Eg. Object size : 80k bits, 10 requests / sec on a 2Mbps link

TI	0.1	0.2	0.3	0.4	0.5	0.6	0.7	0.8	0.9	1
$d_{queue}$	0.01μs	0.02μs	10μs	100μs	2ms	0.1s	2s	30s	2min	8min

**Total time** is sum of these 4

(a) Find propagation time

$$20 \text{ Km} / (2*10^8 \text{ m/s})$$

$$= (2*10^4 / 2*10^8) \text{ sec}$$

$$= 10^{-4} \text{ sec}$$

$$= 100 \text{ micro-sec}$$

(b) Whole packet is sent in 1 go. What is total time?

**propagation time** = 100 micro-sec

**transmission time** =  $3 * (0.1 \text{ Mb} / 1 \text{ Gbps})$   
= 300 micro-sec

3 is because, the packet is put onto the link  
3 times, 1st by the PC then by the 2 routers  
in the way

**Queue time** = 0 sec (since, links are un-congested)

**Nodal processing** =  $2 * 5 \text{ micro-sec}$  (since there are 2 routers in the way)

Total = 100 micro-sec + 300 micro-sec + 10 micro-sec = 410 micro-sec

(c ) Packet is divided into 50 parts. How long does it take for 1st part to arrive at destination?

Now, the packet size is  $0.1\text{Mbits} / 50 = 2000 \text{ bits}$

propagation time =  $20 \text{ Km} / 2*10^8 = 100 \text{ micro-sec}$

Transmission time =  $3 * (2000 / 1 \text{ Gbps}) = 6 \text{ micro-sec}$

queuing time = 0sec

nodal processing = 10 micro-sec

total =  $100 + 6 + 10 = 116 \text{ micro-sec}$

side question : How long does it take for 2nd part to arrive?

Note : The 2nd packet will be launched as soon as the 1st packet is on the link, ie. 2 micro-sec later.

Note : 2nd packet has to wait at the routers, since when 2nd arrives, 1st is being processed by the router.

Visualize that R1 starts to process 2nd packet only after it is done with 1st packet. Thus, the initial lag of 2 micro-sec is now 5 micro-sec (nodal processing)

Thus at  $t= 118$  (116+5) micro-sec the 2nd packet arrive.

Another side question: When does the 50th packet arrive?

2nd packet arrives at  $t = 116 + 5$

3rd packet should arrive 5 micro-sec after 2nd packet

3rd packet arrives at  $t = (116 + 5) + 5$

4th packet  $t = (116 + 5 + 5) + 5$

in general

nth packet  $t = 116 + (n-1)*5$

Thus, 50th packet arrives at  $t=116 + 49*5 = 361$

# Multiple Choice Questions

# Q1

In the transmitting process of packet X from host A to host B, which of the following delays are proportional to the size of packet X?

- a. Processing delay
- b. Queuing delay
- c. Transmission delay
- d. Propagation delay
- e. None of the above

Processing time : Is generally fixed and depends on the quality of router

Queue time : depends on the amount of traffic in the network

Transmission time : size of packet / link speed

Propagation time : distance / speed

Answer : C

# Q2

Your computer will not be able to connect to other machines on the Internet if the domain name server that your machine attempts to use is down.

- a. True
- b. False

DNS is used only to translate domain name to IP address

Even though your institutional DNS server is down, you can still connect to a remote PC if you know its IP address

Answer : B

# Q3

My friend Sam tells me that his network program uses UDP but can still achieve reliable data transfer.

His statement is

- A. True
- B. False

Sam can himself design a mechanism in his program to handle corrupt or lost packets.

Answer : True