National University of Computer and Emerging Sciences
Islamabad Campus

Computer Networks (CS3001)

Final Examination

Total Time (Hrs): 3
Total Marks: 100
Total Questions: 11

Course Instructor(s):
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Section(s): CS All Sections

Date: Dec 28, 2024

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Roll No **Course Section** **Student Signature**

Do not write below this line.

Attempt all the questions.

Section I: Transport Layer [30 Marks]

Q 1: Short Questions **[15 marks]**

a. If the RTT from Lahore to Islamabad is 120ms and all links in the network have a 155 Mbits/second data-rate, how much data can fit in the “pipe” (data in transit)? Express your answer in bytes.(2 marks)

Solution:

To calculate the amount of data that can fit in the "pipe" (also referred to as the bandwidth-delay product), use the formula:

Data in transit=Bandwidth×RTT

Given Values:

Bandwidth = 155 Mbits/sec
$=155\times10^6$ bits/sec
RTT = 120 ms=0.120 seconds

Step 1: Calculate data in transit (in bits):

Data in transit=155×106×0.120=18.6×10^6 bits

Step 2: Convert to bytes:

Data in transit=$18.6\times10^6/8=2.325\times10^6$ bytes

Final Answer:

The amount of data that can fit in the pipe is **2,325,000 bytes**.

b. In a connection, the value of cwnd is 3000 and the value of rwnd is 5000. The host has sent 2000 bytes, which have not been acknowledged. How many more bytes can be sent? (2 marks)

Given:

Congestion window size (cwnd) = 3000 bytes
Receiver window size (rwnd) = 5000 bytes
Unacknowledged bytes = 2000 bytes

Step 1: Calculate the effective window size:

Effective window size = min(cwnd,rwnd)-Unacknowledged bytes

Effective window size = min(3000,5000)−2000 =1000 bytes

Final Answer:

The host can send **1000 more bytes.**

c. Client uses UDP to send data to a server. The data length is 16 bytes. Calculate the efficiency of this transmission at the UDP level (ratio of useful bytes to total bytes). (2 marks)

Given:

- Data length (payload) = 16 bytes
- UDP header size = 8 bytes
- Total size = Data length+UDP header size
- Total size=16+8=24 bytes

Efficiency Formula:

Efficiency=Useful data (payload) / Total size×100

Calculation:

Efficiency=(16/24) ×100=66.67%

Final Answer:

The efficiency of the UDP transmission is **66.67%**.

d. In a TCP connection, the window size fluctuates between 60,000 bytes and 30,000 bytes. If the average RTT is 30 ms, what is the throughput of the connection? (2 marks)

Solution:

Throughput = Average window size / RTT

Average window size = 60,000+30,000/2=45,000 bytes

RTT = 30 ms=0.030 seconds

Calculation:

Throughput=45,000 bytes / 0.030 seconds=1,500,000 bytes/sec

Final Answer:

Throughput=**1.5MB/sec**

e. A client residing on a host with IP address 122.45.12.7 sends a message to the corresponding server residing on a host with IP address 200.112.45.90. If the well-known port is 161 and the ephemeral port is 51000, what are the pair of socket addresses used in this communication? (2 marks)

Answer:

The pair of socket addresses used in this communication is:

(122.45.12.7, 51000) and (200.112.45.90, 161)

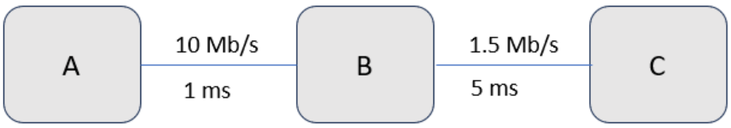
f. What are the four aspects related to the reliable delivery of data? (2 marks)
The four aspects are:
1. Error Detection and Correction
2. Flow Control
3. Acknowledgment and Retransmission
4. Sequencing

g. An HTTP client opens a TCP connection using an initial sequence number (ISN) of 14,534 and the ephemeral port number of 59,100. The server opens the connection with an ISN of 21,732. Show the three TCP segments during the connection establishment if the client defines the rwnd of 4000 and the server defines the rwnd of 5000. Ignore the calculation of the checksum field. (3 marks)

Summary of the Three-Way Handshake:
1. **Client → Server:** SYN, Seq = 14,534, Ack = -, rwnd = 4000
2. **Server → Client:** SYN-ACK, Seq = 21,732, Ack = 14,535, rwnd = 5000
3. **Client → Server:** ACK, Seq = 14,535, Ack = 21,733, rwnd = 4000

Q2: **[15 marks]**

Consider the following scenario that connects the host computers A and C with a store and forward switch B. The propagation delays from A TO B is 1ms from B to C is 5ms and bandwidths between A TO B is 10Mb/s and between B and C is 1.5Mb/s of the respective links are given in the Figure.



a. Assume A wants to send a frame to C. If the frame size is 6000 bits, how long does it take to deliver the entire frame at the switch B? (5 points)
b. How long does it take to deliver the entire frame from host A to host C? (5 marks)

c. Suppose A continuously sends frames to C (one after the other/back to back/without giving a break). However, host B can store at most 10 packets. At what time will B start to drop packets? What will be the packet dropping rate in terms of packets/sec? (5 marks)

Solution:

(a) Time to deliver the entire frame at switch B

To determine how long it takes for the frame to be delivered to switch B, we need to calculate the transmission delay from A to B and add the propagation delay from A to B.

1. **Transmission Delay (A to B):**

Transmission Delay=Frame Size/Bandwidth from A to B
Transmission Delay=6000/107=0.0006 s=0.6 ms

2. **Total Time to Deliver to B:**

Total Time=Transmission Delay+Propagation Delay
Total Time=0.0006+0.001=0.0016 s=**1.6 ms**

(b) Time to deliver the entire frame from host A to host C

To calculate the total time for the frame to travel from A to C:

1. The frame must first reach B (computed in part (a)).
2. Once the frame is completely at B, it will be transmitted to C. The total time from B to C includes:

Transmission Delay (B to C): Transmission Delay=Frame Size/Bandwidth from B to C
Transmission Delay=6000/(1.5×10^6)=0.004 s=4 ms
Propagation Delay (B to C): 0.005 s=5 ms
Total Time (B to C)=0.004+0.005=0.009 s=9 ms

Total Time from A to C:

Total Time (A to C)=Time (A to B)+Time (B to C)
Total Time (A to C)=0.0016+0.009=0.0106 s=**10.6 ms**

C.

1. **Rate of Frame Arrival at B (A to B):**

   The arrival rate is determined by the transmission rate from A to B:

$$\text{Arrival Rate} = \frac{1 \text{ Frame}}{\text{Transmission Delay (A to B)}} = \frac{1}{0.0006} = 1666.67 \text{ frames/sec}$$

2. **Rate of Frame Departure from B (B to C):**

   The departure rate is determined by the transmission rate from B to C:

$$\text{Departure Rate} = \frac{1 \text{ Frame}}{\text{Transmission Delay (B to C)}} = \frac{1}{0.004} = 250 \text{ frames/sec}$$

3. **Frame Accumulation Rate at B:**

$$\text{Accumulation Rate} = \text{Arrival Rate} - \text{Departure Rate}$$

$$\text{Accumulation Rate} = 1666.67 - 250 = 1416.67 \text{ frames/sec}$$

4. **Time for Buffer to Fill:**

   Switch B has a buffer capacity of 10 frames. The time it takes to fill this buffer is:

$$\text{Time to Fill Buffer} = \frac{\text{Buffer Capacity}}{\text{Accumulation Rate}}$$

$$\text{Time to Fill Buffer} = \frac{10}{1416.67} \approx 0.00706 \text{ s} = 7.06 \text{ ms}$$

5. **Packet Dropping Rate:**

   Once the buffer is full, packets are dropped at the accumulation rate:

$$\text{Dropping Rate} = \text{Accumulation Rate} = 1416.67 \text{ packets/sec}$$

Final Answers

Final Answers:

(a): 1.6 ms

(b): 10.6 ms

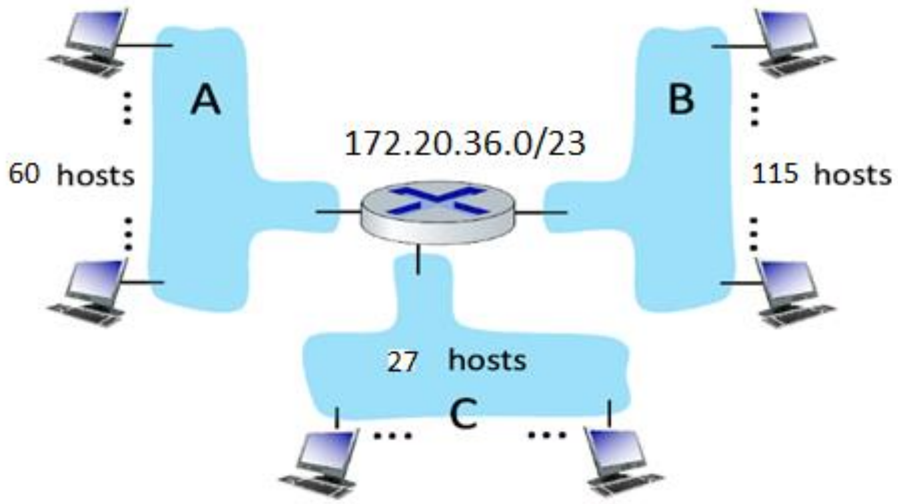
(c):

- **B starts to drop packets at 7.06 ms**
- **Packet dropping rate: 1416.67 packets/sec**

Section II: Network Layer (Data Plane) [20 Marks]

Q 3: Subnetting [**10 Marks**]

Consider the router and the three attached subnets below (A, B, and C). The number of hosts is also shown below. The subnets share the 23 high-order bits of the address space: 172.20.36.0/23. Assign subnet addresses to each of the subnets (A, B, and C) so that the amount of address space assigned is minimal, and at the same time leaving the largest possible contiguous address space available for assignment if a new subnet were to be added. Then answer the questions below.



a. Is the address space public or private? (0.5 mark)
b. How many hosts can there be in this address space? (0.5 mark)
c. What are the subnet addresses of subnet A, B and C? (CIDR notation) (3 marks)
d. What are the broadcast addresses of subnet A, B and C? (3 marks)
e. What are the starting and ending addresses of subnet A, B and C? (3 marks)

Solution:

a) The address space **172.20.36.0/23** belongs to the private IP range **172.16.0.0 – 172.31.255.255**.

b) A /23 network provides $2^{32-23}=2^9=512$ total IP addresses. Subtracting 2 for the network and broadcast addresses, we have 510 usable IPs.

c) We need to allocate IP addresses to each subnet while minimizing waste.
Subnet A: Requires 60 hosts. The nearest power of 2 is 2^6=64. Therefore, Subnet A requires a **/26** network.
Subnet B: Requires 115 hosts. The nearest power of 2 is 2^7=128. Therefore, Subnet B requires a **/25** network.
Subnet C: Requires 27 hosts. The nearest power of 2 is 2^5=32. Therefore, Subnet C requires a **/27** network.
Subnet A: 172.20.36.0/26 (64 addresses)
Subnet B: 172.20.36.64/25 (128 addresses)
Subnet C: 172.20.36.192/27 (32 addresses)

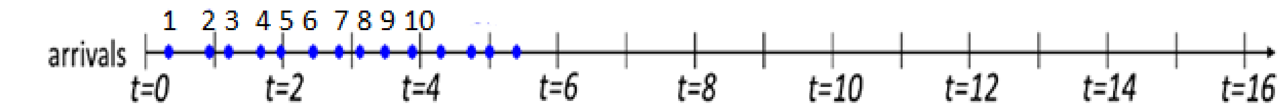
d) **Subnet A:** Broadcast address is 172.20.36.63
Subnet B: Broadcast address is 172.20.36.191

Subnet C: Broadcast address is 172.20.36.223

e) **Subnet A:** Starting address: 172.20.36.1, Ending address: 172.20.36.62.
Subnet B: Starting address: 172.20.36.65, Ending address: 172.20.36.190.
Subnet C: Starting address: 172.20.36.193, Ending address: 172.20.36.222.

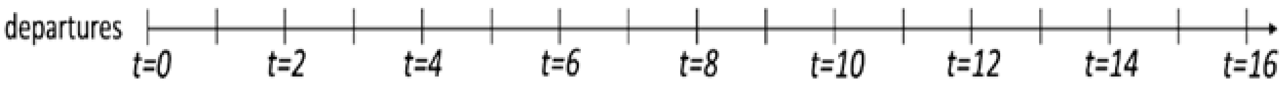
Q 4: Packet Scheduling **[5 Marks]**

Consider the arrival of 10 packets to an output link at a router in the interval of time [0, 5], as indicated by the figure below. We'll consider time to be "slotted", with a slot beginning at t = 0, 1, 2, 3, etc. Packets can arrive at any time during a slot, and multiple packets can arrive during a slot. At the beginning of each time slot, the packet scheduler will choose one packet, among those queued (if any), for transmission according to the packet scheduling discipline (that you will select below). Each packet requires exactly one slot time to transmit, and so a packet selected for transmission at time t, will complete its transmission at t+1, at which time another packet will be selected for transmission, among those queued.



Packets (#: time): 1: 0.34, 2: 0.93, 3: 1.21, 4: 1.67, 5: 1.96, 6: 2.42, 7: 2.79, 8: 3.09, 9: 3.45, 10: 3.84

Packets (#: class): 1: 3 , 2: 3 , 3: 1 , 4: 1 , 5: 1 , 6: 2 , 7: 2 , 8: 1 , 9: 1 , 10: 1



Consider a Weighted Fair Queuing (WFQ) packet scheduling mechanism with three classes of traffic (1, 2, 3), with lower class numbers having higher priority. The scheduling weights are 0.5, 0.3, and 0.2 for classes 1, 2, and 3, respectively.
For the given times from t=1 to t=10:
Show the steps to determine which packet is selected for each time interval based on the WFQ mechanism and specify the packet number on the departure line for each time interval or write 'n/a' if no packet is sent.

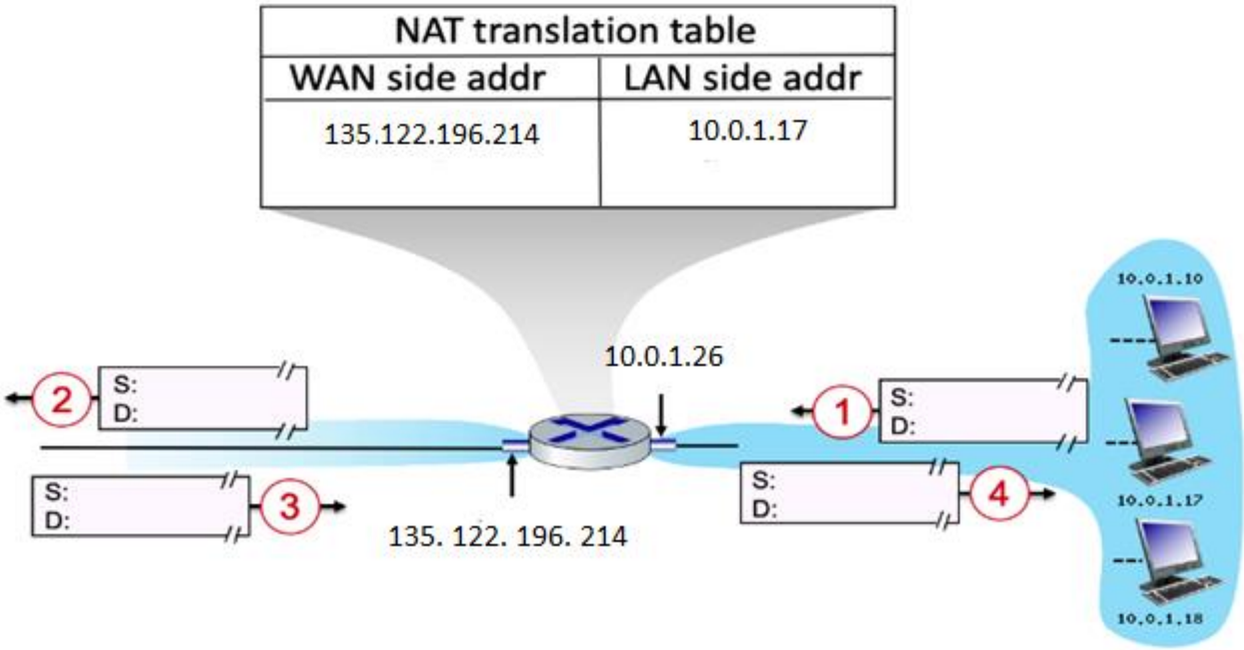
Solution:

Time (t)	Arrived Packets	Packets in Queue	Packet Selected (Class)	Departure Time (t)
t = 0	1	1	1 (Class 1)	t = 1
t = 1	2	2, 3	3 (Class 1)	t = 2
t = 2	3, 4, 5	2, 4, 5	4 (Class 1)	t = 3
t = 3	6, 7	2, 5, 6, 7	5 (Class 1)	t = 4
t = 4	8, 9, 10	2, 6, 7, 8, 9, 10	8 (Class 1)	t = 5
t = 5	—	2, 6, 7, 9, 10	9 (Class 1)	t = 6
t = 6	—	2, 6, 7, 10	10 (Class 1)	t = 7
t = 7	—	2, 6, 7	6 (Class 2)	t = 8
t = 8	—	6, 7, 2	7 (Class 2)	t = 9
t = 9	—	7, 2	2 (Class 3)	t = 10

1 mark is for showing working steps.

Q 5: Network Address Translation (NAT) **[5 Marks]**

Consider the scenario below in which three hosts, with private IP addresses 10.0.1.10, 10.0.1.17, 10.0.1.18 are in a local network behind a NAT'd router that sits between these three hosts and the larger Internet. IP datagrams being sent from, or destined to, these three hosts must pass through this NAT router. The router's interface on the LAN side has IP address 10.0.1.26, while the router's address on the Internet side has IP address 135.122.196.214. Suppose that the host with IP address 10.0.1.17 sends an IP datagram destined to host 128.119.161.188. The source port is 3453, and the destination port is 80.



a. Consider the datagram at step 1, after it has been sent by the host but before it has reached the router. What is the source and destination IP address for this datagram?

b. Now consider the datagram at step 2, after it has been transmitted by the router. What is the source and destination IP address for this datagram?
c. At step 2, Will the source port have changed?
d. Now consider the datagram at step 3, just before it is received by the router. What is the source and destination IP address for this datagram?
e. At step 4, after it has been transmitted by the router but before it has been received by the host. What is the source and destination IP address for this datagram?

Solution:

a. The source address will be the local host's IP, which is 10.0.1.17and the destination address will be the remote machine's IP, which is 128.119.161.188
b. The source address will be the router's public IP, which is 135.122.196.214 and destination address will be remote machine's IP 128.119.161.188
c. Yes, the NAT will change the source port.
d. The source address will be the remote machine's IP, which is 128.119.161.188 and the destination address will be the router's public IP, which is 135.122.196.214
e. The source address will be the remote machine's IP, which is 128.119.161.188 and the destination address will be the local host's IP, which is 10.0.1.17.

Section III: Network Layer (Control Plane) [30 Marks]

Q 6: Short Questions [13 Marks]

a. Given a network with multiple gateways, analyze how hot potato routing might impact the load on an intra-AS network compared to the inter-AS costs. (2 marks)

   Hot potato routing reduces the load on intra-AS resources at the cost of potentially higher inter-AS costs and suboptimal routing paths. This trade-off is especially significant in networks with multiple gateways where balancing intra-AS load and inter-AS cost is critical.

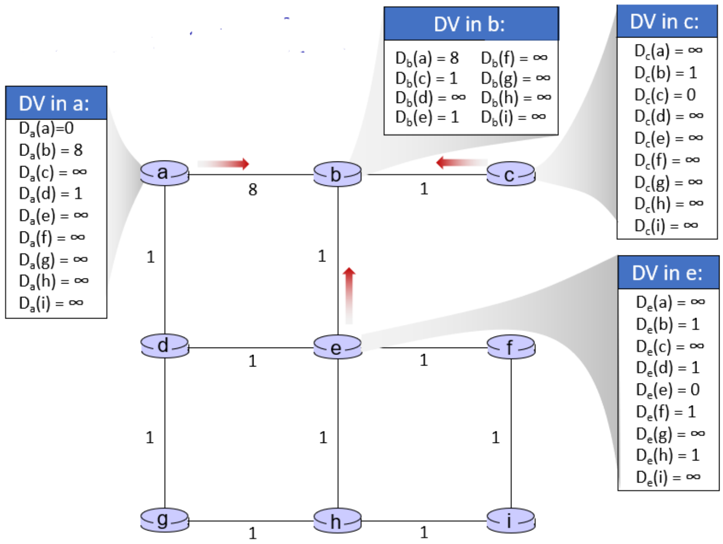
b. Consider a scenario where an AS wants to avoid routing traffic through a specific neighboring AS due to security concerns. How can the AS enforce this policy using BGP attributes? (1 mark)

   The AS can enforce this policy by configuring **BGP import policies** to reject route advertisements containing the specific neighboring AS in the **AS-PATH** attribute, effectively avoiding routes through that AS.

c. If an ICMP message with type 3 and code 1 is received at the source, what does it signify, and how should the source respond? (1 mark)

   An ICMP message with **type 3, code 1** indicates that the destination host is unreachable.

d. Compute updated forwarding table of **b** by looking at its neighbors. Use Bellman ford equation to do so. (3 marks)



$$D_b(a) = \min\{c_{b,a}+D_a(a),\ c_{b,c}+D_c(a),\ c_{b,e}+D_e(a)\} = \min\{8,\infty,\infty\} = 8$$

$$D_b(c) = \min\{c_{b,a}+D_a(c),\ c_{b,c}+D_c(c),\ c_{b,e}+D_e(c)\} = \min\{\infty,1,\infty\} = 1$$

$$D_b(d) = \min\{c_{b,a}+D_a(d),\ c_{b,c}+D_c(d),\ c_{b,e}+D_e(d)\} = \min\{9,2,\infty\} = 2$$

$$D_b(e) = \min\{c_{b,a}+D_a(e),\ c_{b,c}+D_c(e),\ c_{b,e}+D_e(e)\} = \min\{\infty,\infty,1\} = 1$$

$$D_b(f) = \min\{c_{b,a}+D_a(f),\ c_{b,c}+D_c(f),\ c_{b,e}+D_e(f)\} = \min\{\infty,\infty,2\} = 2$$

$$D_b(g) = \min\{c_{b,a}+D_a(g),\ c_{b,c}+D_c(g),\ c_{b,e}+D_e(g)\} = \min\{\infty, \infty, \infty\} = \infty$$

$$D_b(h) = \min\{c_{b,a}+D_a(h),\ c_{b,c}+D_c(h),\ c_{b,e}+D_e(h)\} = \min\{\infty, \infty, 2\} = 2$$

$$D_b(i) = \min\{c_{b,a}+D_a(i),\ c_{b,c}+D_c(i),\ c_{b,e}+D_e(i)\} = \min\{\infty, \infty, \infty\} = \infty$$

e. In a Distance Vector routing scenario, what happens when a router detects an increase in a link cost? Illustrate the "count-to-infinity" problem with an example. (1 mark)

When a link cost increases, routers update distances and propagate them. In the "count-to-infinity" problem, routers repeatedly exchange increasing path costs, causing delays in convergence. Example: A → B (1), B → C (1). If B→C breaks, A and B loop updates indefinitely.

f. How do OpenFlow controller-to-switch and switch-to-controller messages facilitate SDN operation and network programmability? (1 mark)

OpenFlow **controller-to-switch** messages configure switches by managing flow tables, while **switch-to-controller** messages report events or forward packets for decision-making. Together, they enable dynamic SDN control and programmability.

g. Describe how a logically centralized control plane in SDN provides enhanced flexibility and avoids misconfigurations compared to traditional per-router control. (2 marks)

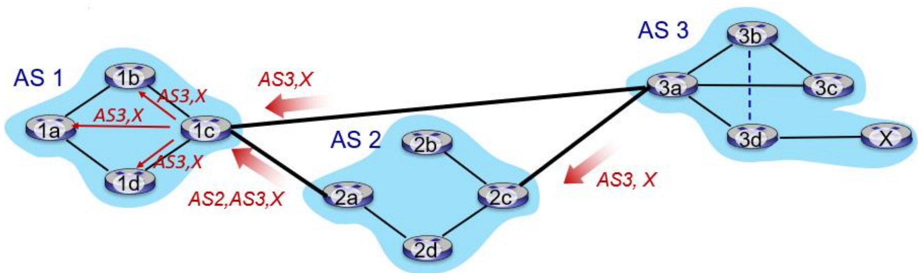
A logically centralized SDN control plane provides a global network view, enabling optimized routing and traffic engineering. It avoids misconfigurations by centralizing control logic, eliminating inconsistencies from distributed configurations in traditional per-router control.

h. Explain the role of AS-PATH and NEXT-HOP attributes in BGP route advertisement and selection. (2 marks)
The **AS-PATH** attribute lists ASs traversed by a route, helping prevent loops and influencing route selection based on path length. The **NEXT-HOP** attribute identifies the next router for forwarding traffic, ensuring accurate routing within and between ASs.

Q 7: [7 Marks]

Consider a network with three autonomous systems (AS1, AS2, AS3). Each AS uses OSPF for intra-domain routing and BGP for inter-domain routing.

a. Draw a topology where AS1 is connected to both AS2 and AS3. Assume AS1 prefers routing through AS2 unless AS2's path becomes unavailable. (3 marks)



b. Describe the updates and route advertisements exchanged using eBGP and iBGP when a link in AS2 fails. (2 marks)

Updates and Route Advertisements in BGP:

1. **Before Failure**:
   - **eBGP**: AS2 advertises routes to AS1; AS3 advertises its routes to AS1. AS1 selects the path via AS2 based on preferences.
   - **iBGP**: Within AS1, R1 shares the preferred path (via AS2) with other AS1 routers.
2. **After Failure in AS2**:
   - **eBGP**: AS2 stops advertising its routes. AS1 detects the failure and switches to the route advertised by AS3.
   - **iBGP**: R1 updates all other AS1 routers to use the backup path via AS3.

c. Analyze how SDN could simplify or optimize this process using its centralized control plane. (2 marks)

1. **Global View**:

   SDN's centralized control plane provides a global view of network state, detecting AS2's link failure instantly.

2. **Dynamic Rerouting**:

   SDN dynamically updates routing tables in AS1 to use AS3 without relying on BGP's convergence time.

3. **Policy Optimization**:

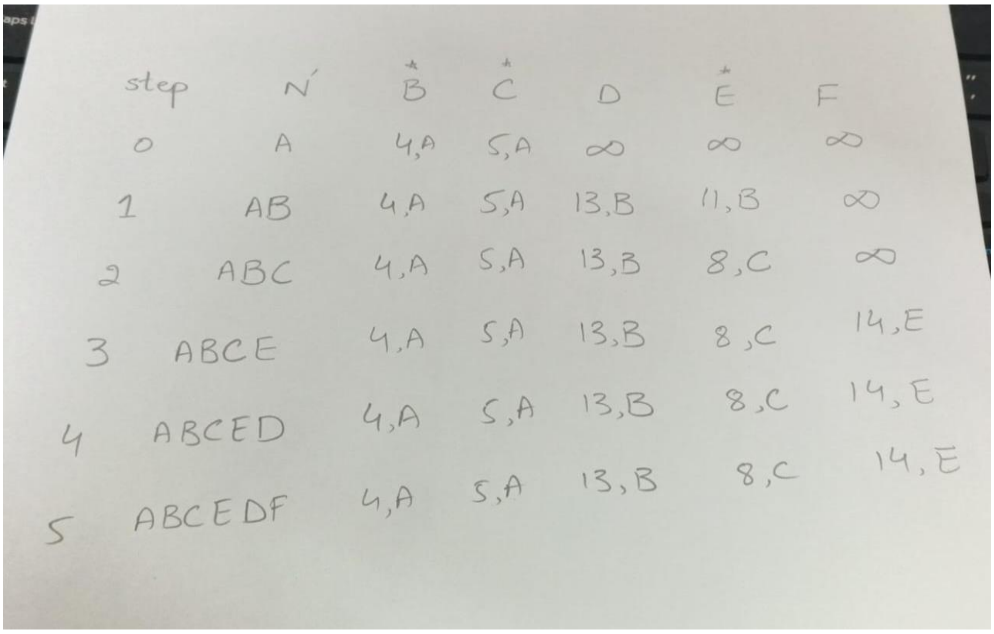
   SDN can enforce routing preferences directly using traffic engineering policies, eliminating dependency on distributed BGP updates.

Q 8: **[10 Marks]**

Consider the following network. With the indicated link costs, use Dijkstra's shortest-path algorithm to compute the shortest path from A to all network nodes.

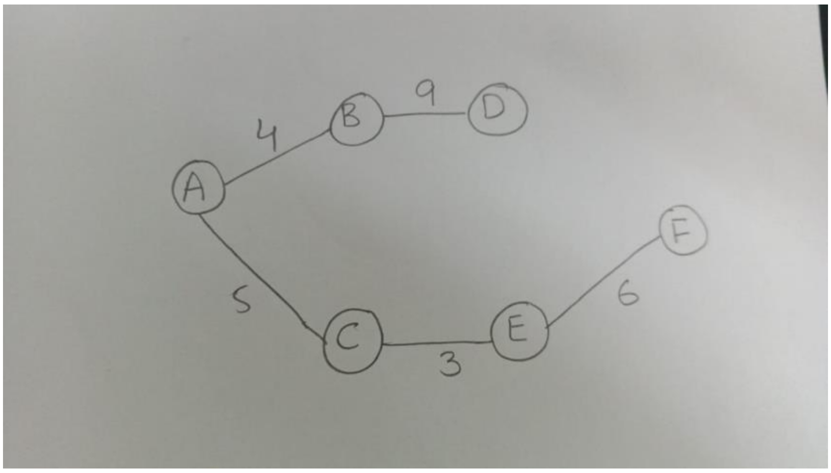
Consider the following network. With the indicated link costs, use Dijkstra's shortest-path algorithm to compute the shortest path from A to all network nodes.

a. Show how the algorithm works, show all necessary steps in the form of table. [marks 5]

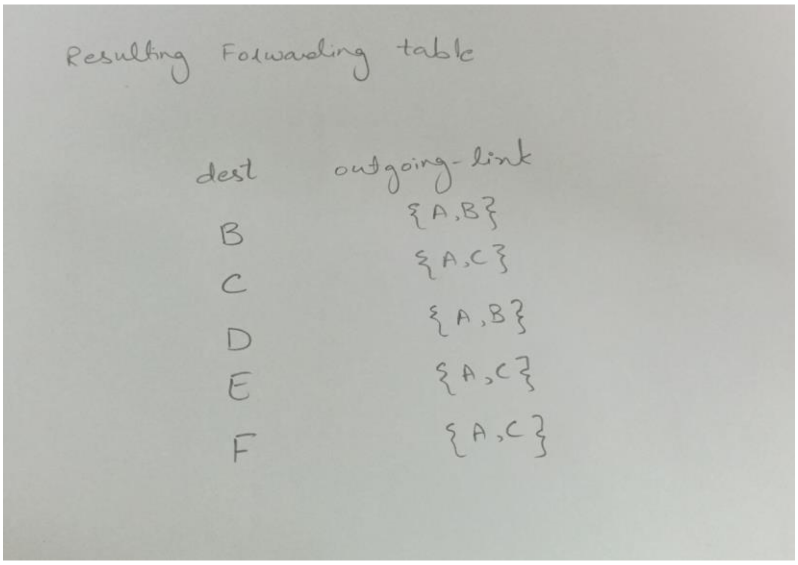


step	N'	B	C	D	E	F
0	A	4,A	5,A	∞	∞	∞
1	AB	4,A	5,A	13,B	11,B	∞
2	ABC	4,A	5,A	13,B	8,C	∞
3	ABCE	4,A	5,A	13,B	8,C	14,E
4	ABCED	4,A	5,A	13,B	8,C	14,E
5	ABCEDF	4,A	5,A	13,B	8,C	14,E

b. Draw the resulting shortest-path tree from A to all other nodes. [marks 3]



c. Provide the resulting forwarding table in A. [marks 2]

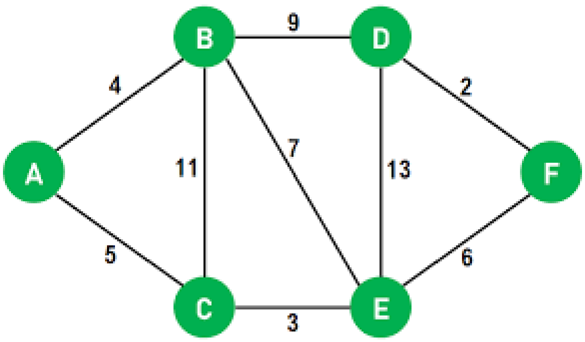


Resulting Forwarding table

dest	outgoing-link
B	{A,B}
C	{A,C}
D	{A,B}
E	{A,C}
F	{A,C}

NOTE

Please note that you derive (b) and (c) from (a). If (a) is not correct then (b) and (c) will be automatically considered incorrect.



Section IV: Data Link Layer [20 Marks]

Q 9: **[10 Marks]**

A. . How does slotted ALOHA mitigate the drawbacks of pure ALOHA while still being considered a random access protocol? [2 Marks]

Answer: Both are random access for shared channels, but pure ALOHA allows transmissions at any point, leading to **high collision probability and low efficiency (max 18.4%).** Slotted **ALOHA synchronizes transmissions to fixed time slots**, reducing collisions to a maximum of 36.8% by limiting potential overlap. **This improves channel utilization** despite still being random in nature.

B. Explain how carrier sensing and collision detection in CSMA/CD enhances efficiency. [2 Marks]

Answer: Carrier sensing (CS) **allows a node to listen before transmitting; if the channel is busy, it waits.** Collision Detection (CD) continuously monitors during transmission to detect overlapping signals**. If a collision occurs, both nodes stop, use backoff algorithms, and retransmit later. CS prevents unnecessary transmissions on occupied channels, while CD stops collisions early instead of wasting entire packets as in pure CSMA.**

C. Why is an ARP query broadcast while its response is unicast? [2 Marks]

Answer: ARP maps IP addresses to MAC addresses. **A query seeks the MAC for a specific IP, so it's broadcast to reach all nodes on the subnet potentially holding that address.** The response, **knowing the recipient's MAC from the query,** is unicast to that specific node.

D. Describe how the token in a Token Ring network ensures fair access to the shared channel while preventing collisions. [2 Marks]

Answer: In Token Ring, a single "token" circulates around the ring. **A node can transmit only when it possesses this token.** This serialized access ensures that **nodes take turns transmitting, effectively preventing simultaneous transmissions (collisions).** When a node transmits, it regenerates a new token after sending its frame, passing it along to the next node in line for its turn.

E. Explain how "backoff" in the context of CSMA/CD helps mitigate network collisions. [2 Marks]

Answer: Backoff is an exponential algorithm used by Ethernet stations after detecting a collision. **It involves waiting for a random amount of time (increasing exponentially with each successive collision) before attempting to retransmit data. This randomized delay spreads out transmissions, reducing the likelihood of another simultaneous transmission and causing further collisions.**

Q 10: **[5 Marks]**

Suppose that a packet's payload consists of **10 eight-bit values** (e.g., representing ten ASCII-encoded characters) shown below. (Here, we have arranged the ten eight-bit values as five sixteen-bit values):

Figure 1:

01110101	00010100
00111110	11011000
00101000	00001110
01011000	00001010
11010110	00110100

Figure 2:

Both the payload and parity bits are shown. One of these bits is flipped.

00010101	10001010	0
00000000	11101101	0
00100100	00000000	0
11101110	00011101	0
01000110	01011011	0
00011001	00100001	0

Figure 3:
Both the payload and parity bits are shown; Either one or two of the bits have been flipped.

11011110	11111110	1
01101110	11001000	1
10110000	00101100	0
10001001	10000000	0
01011111	10000010	0
11010010	00011000	0

Using the above figures, answer the following questions:

a. For figure 1, compute the two-dimensional parity bits for the 16 columns. Combine the bits into one string. [2 Marks].
   1110110111111100
b. For figure 1, compute the two-dimensional parity bits for the 5 rows (starting from the top). Combine the bits into one string. [2 Marks]
   11110
c. For figure 1, compute the parity bit for the parity bit row from question 1. Assume that the result should be even. [2 Marks]
   0
d. For figure 2, indicate the row and column with the flipped bit (format as: x,y), assuming the top-left bit is 0,0. [2 Marks]
   0,5
e. For figure 3, is it possible to detect and correct the bit flips? Yes or No. [2 Marks]
   Yes

Q 11: **[5 Marks]**

Suppose that the 4-bit generator (G) is 1001 and that r = 3. What are the CRC bits (R) associated with the data payload D when D is:

a. 10011111 (2.5 marks)
   110

b. 10011101 (2.5 marks)
   100