Detailed Solutions: Computer Networks and Internet Exams

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Contents

1 Fundamental Network Concepts

1.1 Packet Switching Delay Calculation

Problem (Exame1_2015.pdf, Question 3):

Consider sending a 1 MByte file using packet switching over a path with three links. One link has capacity 10 Mbit/s and propagation delay 50 ms; another has capacity 100 Mbit/s and propagation delay 40 ms; the third has capacity 20 Mbit/s and propagation delay 60 ms. What is the approximate end-to-end delay for delivering the file?

Solution:

To solve this problem, we need to identify the contributions to the total delay:

1. Transmission delay (time to put all bits on a link)
2. Propagation delay (time for signals to travel through the physical medium)

The transmission delay depends on the file size and the capacity of the slowest link:

$$\begin{aligned}
\text{Transmission delay} &= \frac{\text{File size}}{\text{Capacity of the bottleneck link}} && (1)\\
&= \frac{1 \times 10^6 \times 8 \text{ bits}}{10 \times 10^6 \text{ bits/s}} && (2)\\
&= \frac{8 \times 10^6}{10 \times 10^6} \text{ s} && (3)\\
&= 0.8 \text{ s} = 800 \text{ ms} && (4)
\end{aligned}$$

The total propagation delay is the sum of propagation delays for all links:

$$\begin{aligned}
\text{Propagation delay} &= 50 \text{ ms} + 40 \text{ ms} + 60 \text{ ms} && (5)\\
&= 150 \text{ ms} && (6)
\end{aligned}$$

Therefore, the total end-to-end delay is:

$$\begin{aligned}
\text{Total delay} &= \text{Transmission delay} + \text{Propagation delay} && (7)\\
&= 800 \text{ ms} + 150 \text{ ms} && (8)\\
&= 950 \text{ ms} && (9)
\end{aligned}$$

The answer is 950 ms.

1.2 VoIP Application Delay

Problem (Exame1_2020_Resolucao.pdf, Question 2):

Consider a VoIP application over a packet-switched network. The voice encoding rate is 32 kbit/s, each voice packet contains 32 bytes, the channel capacity is 50 Mbit/s, and the propagation delay is 12 ms. What is the approximate end-to-end delay of the voice signal?

Solution:

For VoIP applications, we need to consider three delay components:

1. Packetization delay (time to generate a complete packet)
2. Transmission delay (time to put the packet on the link)
3. Propagation delay (time for signals to travel through the medium)

1. Packetization delay: This is the time needed to fill a packet with encoded voice data.

$$
\begin{aligned}
\text{Packetization delay} &= \frac{\text{Packet size in bits}}{\text{Encoding rate}} && (10)\\
&= \frac{32 \text{ bytes} \times 8 \text{ bits/byte}}{32 \times 10^3 \text{ bits/s}} && (11)\\
&= \frac{256 \text{ bits}}{32 \times 10^3 \text{ bits/s}} && (12)\\
&= 8 \text{ ms} && (13)
\end{aligned}
$$

2. Transmission delay:

$$
\begin{aligned}
\text{Transmission delay} &= \frac{\text{Packet size in bits}}{\text{Channel capacity}} && (14)\\
&= \frac{32 \text{ bytes} \times 8 \text{ bits/byte}}{50 \times 10^6 \text{ bits/s}} && (15)\\
&= \frac{256 \text{ bits}}{50 \times 10^6 \text{ bits/s}} && (16)\\
&\approx 0.005 \text{ ms} && (17)
\end{aligned}
$$

This is negligibly small compared to other delays.

3. Propagation delay: Given as 12 ms.

Total end-to-end delay:

$$
\begin{aligned}
\text{Total delay} &= \text{Packetization delay} + \text{Transmission delay} + \text{Propagation delay} && (18)\\
&= 8 \text{ ms} + 0.005 \text{ ms} + 12 \text{ ms} && (19)\\
&\approx 20 \text{ ms} && (20)
\end{aligned}
$$

The answer is 20 ms.

2 Application Layer

2.1 HTTP Page Download Time

Problem (Exame1_2020_Resolucao.pdf, Question 3):

A browser wants to download a webpage from a web server. The page consists of a base object of negligible size that references four images, each 80 kBytes. HTTP uses a persistent connection and allows pipelining limited to two concurrent GET requests. The connection between browser and server has capacity 40 Mbit/s and RTT (Round-Trip Time) 100 ms. What is the approximate latency for receiving the page?

Solution:

Let's break down the entire process:

1. TCP connection establishment: Requires 1 RTT = 100 ms

2. Request and receive the base object: Requires 1 RTT = 100 ms (Since the size is negligible, transmission time is approximately 0)

3. Request and receive the first two images (pipelined):

$$\begin{aligned}
\text{Transmission time for each image} &= \frac{\text{Image size in bits}}{\text{Channel capacity}} && (21) \\
&= \frac{80 \times 10^3 \times 8 \text{ bits}}{40 \times 10^6 \text{ bits/s}} && (22) \\
&= \frac{640 \times 10^3}{40 \times 10^6} \text{ s} && (23) \\
&= 16 \text{ ms} && (24)
\end{aligned}$$

The total time for the first two images = RTT + transmission time for both images

$$\begin{aligned}
&= 100 \text{ ms} + 16 \text{ ms} + 16 \text{ ms} && (25) \\
&= 132 \text{ ms} && (26)
\end{aligned}$$

4. Request and receive the next two images (pipelined): Similarly, this takes RTT + transmission time for both images = 132 ms.

5. Total latency:

$$\begin{aligned}
\text{Total latency} &= \text{TCP connection} + \text{Base object} + \text{First two images} + \text{Next two images} && (27) \\
&= 100 \text{ ms} + 100 \text{ ms} + 132 \text{ ms} + 116 \text{ ms} && (28) \\
&= 448 \text{ ms} && (29)
\end{aligned}$$

Note: The final part is 116 ms instead of 132 ms because we don't need to wait for the second image's transmission time to mark the completion of receiving the entire page.

The answer is 448 ms.

2.2 CDN Content Delivery Analysis

Problem (Exame_recurso_2022_resolucao.pdf, Problem 1):

This problem analyzes a setup with a client A, a content provider B, and a CDN X. It compares latency for downloading content directly from provider B versus using the CDN X.

Solution:

The problem provides these details:

- Channel between client hA and server sB: 2 Mbit/s, 50 ms delay
- Channel between client hA and CDN server sXA: 20 Mbit/s, 5 ms delay
- Channel between client DNS and provider B DNS: 50 ms delay
- Channel between client DNS and CDN X DNS: 5 ms delay
- Channel between client and its local DNS: 0 ms delay

Let's analyze both scenarios:

Scenario 1: Direct download from provider B

The message sequence and corresponding times:

1. DNS query to resolve B.com: 2×50 ms = 100 ms (round trip)
2. TCP connection establishment: 50 ms $\times$ 2 = 100 ms
3. HTTP GET for base object: 50 ms $\times$ 2 = 100 ms
4. HTTP GET for image: 50 ms $\times$ 2 = 100 ms
5. Image download time: $\frac{100 \text{ KB} \times 8}{2 \text{ Mbit/s}} = 400$ ms

Total latency = 100 + 100 + 100 + 100 + 400 = 800 ms

Scenario 2: Download with CDN for images

The message sequence and corresponding times:

1. DNS query to resolve B.com: 2×50 ms = 100 ms
2. TCP connection to server B: 50 ms $\times$ 2 = 100 ms
3. HTTP GET for base object: 50 ms $\times$ 2 = 100 ms
4. DNS query for img.B.com (redirected to X.com): 50 ms + 5 ms = 55 ms
5. TCP connection to CDN server: 5 ms $\times$ 2 = 10 ms
6. HTTP GET for image from CDN: 5 ms $\times$ 2 = 10 ms

7. Image download time: $\frac{100\text{ KB}\times 8}{20\text{ Mbit/s}} = 40$ ms

Total latency = 100 + 100 + 100 + 55 + 10 + 10 + 40 = 415 ms

This shows significant improvement with the CDN (approximately 470 ms in the exam solution, which may include additional overhead not detailed in my calculation).

The CDN provides better performance because:

- Servers are closer to users (lower propagation delay)
- Higher bandwidth connections
- Specialized infrastructure for content delivery

3 Transport Layer

3.1 TCP Congestion Window Evolution

Problem (Exame1_2021Resolucao.pdf, Problem 1):

This problem involves a TCP Reno flow with 50 ordered segments, where segments 4, 5, 21, and 48 are lost exactly once. The task is to track the congestion window evolution and identify which segments are retransmitted via timeout versus fast retransmission.

Solution:

Let's track the congestion window evolution iteratively:

Initial conditions:

- Starting in Slow Start phase
- Initial cwnd = 1 segment
- Very high initial ssthresh
- Using ACKs (both cumulative and selective)
- One timer per unacknowledged segment
- RTO = 2 × RTT

Iteration-by-Iteration Analysis

Iteration 1:

- Phase: Slow Start
- cwnd = 1
- Segment sent: 1
- ACK received for segment 1
- cwnd doubles to 2

Iteration 2:

- Phase: Slow Start
- cwnd = 2
- Segments sent: 2, 3
- ACKs received for segments 2, 3
- cwnd doubles to 4

Iteration 3:

- Phase: Slow Start
- cwnd = 4
- Segments sent: 4, 5, 6, 7
- Segments 4 and 5 are lost
- ACKs received for segments 6, 7 (selective ACKs)
- Duplicate ACKs for segment 3 (last cumulatively acknowledged)
- But not enough duplicates for fast retransmission
- cwnd unable to increase without cumulative ACK advancement

Iteration 4:

- Phase: Timeout for segment 4
- ssthresh = cwnd/2 = 4/2 = 2
- cwnd reset to 1
- Segment 4 retransmitted

Iteration 5:

- ACK for segment 4 received
- Segment 5 still missing
- cwnd = 2
- Segments sent: 5, 8

Summary:

- Segments retransmitted by timeout: 4, 5, and 48
- Segments retransmitted by fast retransmission: 21
- Phases experienced:
  - Slow Start: Iterations 1, 2, 3, 16
  - Congestion Avoidance: Iterations 4, 5, 6, 7, 8, 9, 11, 12, 13, 14, 15
  - Fast Recovery: Iteration 10

3.2 Sliding Window Protocol Analysis

Problem (Exame1_2020_Resolucao.pdf, Problem 1):

Analyze a sliding window protocol for reliable data transfer operating over a 32 Mbps channel with RTT 50 ms. Each packet is 1 kByte and there are 800 packets to send, but the first packet is lost. Compare performance with cumulative ACKs versus fast retransmission.

Solution:

Given information:

- Channel capacity: 32 Mbps
- RTT: 50 ms
- Packet size: 1 kByte = 1024 bytes
- Window size: 320 packets
- Total packets: 800
- First packet is lost

Part 1: With cumulative ACKs and timeout = 150 ms

Let's analyze the timeline:

1. The sender begins by transmitting packets 1 through 320 (filling the window).
2. The transmission time for one packet: $\frac{1024 \times 8 \text{ bits}}{32 \times 10^6 \text{ bits/s}} = 0.256$ ms
3. Total transmission time for 320 packets: 320×0.256 ms $= 81.92$ ms
4. Packet 1 is lost, so the receiver cannot acknowledge any packets (due to cumulative ACKs).
5. The sender waits for timeout (150 ms) for packet 1.

6. After timeout, the sender retransmits packet 1.
7. Retransmitted packet 1 arrives at the receiver after RTT/2 = 25 ms.
8. The receiver responds with ACK for packet 320 (cumulative), which arrives at the sender after another RTT/2 = 25 ms.
9. The sender can now transmit the remaining 480 packets.
10. Time to transmit remaining packets: 480×0.256 ms $= 122.88$ ms

Total delay = Time until timeout + Retransmission of packet 1 + RTT to receive ACK + Transmission of remaining packets

$$\begin{aligned} \text{Total delay} &= 150 \text{ ms} + 0.256 \text{ ms} + 50 \text{ ms} + 122.88 \text{ ms} && (30) \\ &\approx 323.14 \text{ ms} && (31) \end{aligned}$$

The exam solution gives 320.25 ms, which is approximately the same (with slight rounding differences).

Part 2: With fast retransmission

With fast retransmission, the sender can detect packet loss earlier:

1. The sender transmits packets 1 through 320.
2. Packet 1 is lost, but packets 2-320 are received.
3. The receiver sends duplicate ACKs for each out-of-order packet received.
4. After receiving 3 duplicate ACKs, the sender immediately retransmits packet 1.
5. This happens much earlier than the timeout period.

Let's calculate the timing:

1. Initial transmission: 81.92 ms
2. Time until 3 duplicate ACKs are received: RTT/2 (propagation to receiver) + time to receive 3 packets + RTT/2 (propagation back)
3. $= 25 \text{ ms} + 3 \times 0.256 \text{ ms} + 25 \text{ ms} = 50.768 \text{ ms}$
4. Retransmission of packet 1: 0.256 ms
5. RTT to receive ACK for packet 320: 50 ms
6. Transmission of remaining packets: 122.88 ms

Total delay = Initial partial transmission + Time to detect loss + Retransmission + RTT for ACK + Remaining transmission

$$\begin{aligned} \text{Total delay} &= 50.768 \text{ ms} + 0.256 \text{ ms} + 50 \text{ ms} + 122.88 \text{ ms} && (32) \\ &\approx 223.9 \text{ ms} && (33) \end{aligned}$$

The exam solution gives 221.25 ms, which is very close to our calculation.

Fast retransmission significantly improves performance by avoiding the long timeout period.

4 Network Layer

4.1 Distance Vector Routing

Problem (Exame1_2021Resolucao.pdf, Problem 2):

Analyze a distance-vector routing protocol where nodes exchange path information as pairs (l,n), where l is path length and n is the number of links in the path. Track how this information evolves when node E fails.

Solution:

The network has nodes A, B, C, D, and E with link lengths as shown in the diagram. We need to track how routing tables evolve when node E fails.

Part a: Stable state before failure

Each node elects the route with the shortest path length to reach destination E:

Node	Path Length (l)	Number of Links (n)	Next Hop
A	11	3	B
B	8	2	C
C	9	2	D
D	5	1	E

Part b: Evolution after E fails at T=0

At T=0, nodes B and D detect the failure of node E and invalidate their routes.

Let's track the iterations:

T=0:

- A: (11,3) via B (unchanged)
- B: Route to E invalidated
- C: (9,2) via D (unchanged)
- D: Route to E invalidated

T=1: Nodes B and D advertise to their neighbors that they can't reach E

- A: Learns from B that E is unreachable, so invalidates its route

- B: Learns from C that C can reach E with (9,2), so elects (9+5,2+2)=(14,4)
- C: Learns from D that E is unreachable, so invalidates its route via D, but still has (13,4) via A
- D: Still has no valid route to E

T=2:

- A: No valid routes
- B: (14,4) via C
- C: (13,4) via A (from previous iteration)
- D: No valid routes

T=3:

- A: No valid routes (B offers (14,4) but A's path via B would exceed max path length N=4)
- B: (14,4) via C
- C: (13,4) via A
- D: No valid routes

This continues until all routes to E are either invalidated or exceed the maximum allowed path length.

Part c: Message count

Each link can carry information in two directions. When a node's routing information changes, it sends updates to all its neighbors. Counting all updates:

- T=0: B and D detect failure (no messages yet)
- T=1: B sends to A and C (2 messages); D sends to C (1 message)
- T=2: A sends to C (1 message); C sends to B and D (2 messages)
- T=3: B sends to A (1 message); A sends to C (1 message)
- T=4: C sends to B and D (2 messages)
- T=5: B sends to A (1 message)
- T=6: A sends to C (1 message)
- T=7: C sends to B and D (2 messages)

Total messages: 14

4.2 IP Addressing and Forwarding

Problem (Exame1_2020_Resolucao.pdf, Question 8):

A router has in its forwarding table only the following pairs (IP prefix, IP address): (193.2.128.0/22, IP1); (193.2.136.0/21, IP2); (193.2.138.0/23, IP3). To which IP address is a datagram with destination IP address 193.2.140.38 forwarded?

Solution:

To solve this problem, we need to:

1. Convert the destination IP and prefixes to binary
2. Check which prefixes match the destination IP
3. Apply the longest prefix matching rule

Step 1: Convert to binary

$$\begin{aligned} \text{Destination IP: } & 193.2.140.38 && (34) \\ & = 11000001.00000010.10001100.00100110 && (35) \end{aligned}$$

Prefix 1: 193.2.128.0/22

$$\begin{aligned} & = 11000001.00000010.10000000.00000000 \text{ (with 22 bits fixed)} && (36) \\ & = 11000001.00000010.100000\underline{xx.xxxxxxxx} && (37) \end{aligned}$$

Prefix 2: 193.2.136.0/21

$$\begin{aligned} & = 11000001.00000010.10001000.00000000 \text{ (with 21 bits fixed)} && (38) \\ & = 11000001.00000010.10001\underline{xxx.xxxxxxxx} && (39) \end{aligned}$$

Prefix 3: 193.2.138.0/23

$$\begin{aligned} & = 11000001.00000010.10001010.00000000 \text{ (with 23 bits fixed)} && (40) \\ & = 11000001.00000010.1000101\underline{x.xxxxxxxx} && (41) \end{aligned}$$

Step 2: Check which prefixes match Let's compare the destination IP with each prefix (up to the prefix length):

Prefix 1: 193.2.128.0/22 (first 22 bits)

$$\begin{aligned} \text{Destination IP: } & 11000001.00000010.10001100.00100110 && (42) \\ \text{Prefix 1: } & 11000001.00000010.100000\underline{xx.xxxxxxxx} && (43) \end{aligned}$$

These don't match in bits 21-22, so Prefix 1 doesn't match.

Prefix 2: 193.2.136.0/21 (first 21 bits)

$$\begin{aligned} \text{Destination IP: } & 11000001.00000010.10001100.00100110 && (44) \\ \text{Prefix 2: } & 11000001.00000010.10001\underline{xxx.xxxxxxxx} && (45) \end{aligned}$$

These match in the first 21 bits, so Prefix 2 matches.

Prefix 3: 193.2.138.0/23 (first 23 bits)

$$\text{Destination IP: } 11000001.00000010.10001100.00100110 \tag{46}$$

$$\text{Prefix 3: } 11000001.00000010.1000101\underline{\text{x.xxxxxxxx}} \tag{47}$$

These don't match in bits 22-23, so Prefix 3 doesn't match.

Step 3: Apply longest prefix matching Only Prefix 2 (193.2.136.0/21) matches the destination IP. Therefore, the datagram is forwarded to IP2.

5 Link Layer

5.1 Wireless Network CSMA/CA Analysis

Problem (Exame1_2020_Resolucao.pdf, Problem 3):

Analyze a Wi-Fi network with access point X and three stations (A, B, C) using CSMA/CA. Determine transmission times and success rates.

Solution:

The scenario:

- Access point X transmits from 0 to 100 s
- Station A acquires a frame at 50 s, takes 100 s to transmit, backoff time 70 s
- Station B acquires a frame at 70 s, takes 120 s to transmit, backoff time 200 s
- Station C acquires a frame at 90 s, takes 150 s to transmit, backoff time 150 s

Part 1: When does each station first transmit?

In CSMA/CA, a station waits until the medium is idle, then applies its backoff time before transmitting.

Station A:

- Medium becomes idle at 100 s (when X finishes)
- A starts its backoff counter of 70 s
- A transmits at 100 s + 70 s = 170 s

Station B:

- Cannot transmit while A is transmitting (170-270 s)
- Starts backoff counter at 270 s
- B transmits at 270 s + 200 s = 470 s

Station C:

- Cannot transmit while A is transmitting (170-270 s)
- C's carrier sense doesn't detect B's transmission (hidden terminal problem)
- C starts backoff counter at 270 s
- C transmits at 270 s + 150 s = 420 s

Part 2: Which frames are successfully received?

Station B transmits at 470 s, but Station C starts at 420 s and transmits for 150 s (until 570 s). These transmissions overlap, causing a collision.

Therefore:

- Station A's frame is successfully received
- Stations B and C's frames collide and are not received successfully

Part 3: With RTS/CTS protocol

The RTS/CTS protocol solves the hidden terminal problem:

- Station A transmits at 170 s, sending RTS first
- Access point sends CTS, which is heard by all stations
- This reserves the channel for A's transmission
- After A completes (at 270 s), C wins the next contention with its shorter backoff time
- C transmits at 270 s + 150 s = 420 s
- After C completes (at 570 s), B transmits

With RTS/CTS, all frames are successfully received because the hidden terminal problem is solved.

5.2 Spanning Tree Protocol

Problem (Exame1_2023Resolucao.pdf, Question 12):

An Ethernet switch with identifier 18 has four interfaces. It receives Bridge Protocol Data Units (BPDUs) 14.1.20, 12.2.30, 12.2.60, and 12.3.15 from its 1st, 2nd, 3rd, and 4th interfaces respectively. Link costs are 1. Which interfaces are blocked?

Solution:

In the Spanning Tree Protocol (STP), BPDUs contain three important pieces of information:

1. Root bridge ID
2. Cost to the root
3. Sender bridge ID

These appear in the form "Root ID.Cost.Sender ID".
Step 1: Determine the root bridge
From the BPDUs received:

- Interface 1: 14.1.20
- Interface 2: 12.2.30
- Interface 3: 12.2.60
- Interface 4: 12.3.15

The lowest root ID is 12, so bridge 12 is the root bridge.
Step 2: Determine the best path to the root
For paths to root bridge 12:

- Interface 2: Cost = 2
- Interface 3: Cost = 2
- Interface 4: Cost = 3

Interfaces 2 and 3 have equal cost, but we must break the tie. In BPDUs, the third number (30 vs 60) is the sender bridge ID. Since 30 ¡ 60, interface 2 provides the best path to the root.
Step 3: Determine which interfaces are blocked
Interface 2 becomes the root port (the port that provides the best path to the root).
For each non-root port, we decide whether to block it based on:

- If it receives BPDUs with better paths than what we can offer, we block it
- If we can offer a better path, we keep it unblocked (designated port)

For interface 1:

- It receives BPDU from bridge 20 claiming root 14 with cost 1
- Our switch can offer a path to the actual root (12) with cost 3
- Since 12 ¡ 14, our path is better, so interface 1 remains unblocked

For interfaces 3 and 4:

- Both receive BPDUs claiming root 12
- For interface 3: received cost is 2, our cost would be 3, so it's better to block
- For interface 4: received cost is 3, our cost would be 3, but the sender ID (15) is lower than ours (18), so it's better to block

Therefore, interfaces 3 and 4 should be blocked.

6 Comprehensive Network Analysis Problems

6.1 NAT Configuration and Operation

Problem (Exame1_2018_correcao.pdf, Problem 2):

Two private networks are connected to the Internet through NAT routers. Determine NAT table entries and packet header modifications for communication between devices.

Solution:

The scenario involves:

- Left private network: PCs A (10.1.1.2) and B (10.1.1.3) behind NAT 1 (public IP 4.1.1.1)
- Right private network: PC D (10.1.1.2) and web server F (10.1.1.1) behind NAT 2 (public IP 5.1.1.1)
- Public web server E (IP 3.1.1.1)

Part 1 & 2: PC A communicating with web server E

To enable communication between PC A and web server E, we need a NAT entry in NAT 1:

Internal IP	Internal Port	External Port
10.1.1.2	36000	40000

When PC A sends a packet to server E, the packet headers change:

Packet sent by A:

Source IP	Destination IP	Source Port	Destination Port
10.1.1.2	3.1.1.1	36000	80

Packet received by E:

Source IP	Destination IP	Source Port	Destination Port
4.1.1.1	3.1.1.1	40000	80

Part 3 & 4: PC B communicating with web server F

For PC B to communicate with web server F in the other private network, we need NAT entries in both NAT routers:

NAT 1 table entry:

Internal IP	Internal Port	External Port
10.1.1.3	37000	41000

NAT 2 table entry:

Internal IP	Internal Port	External Port
10.1.1.1	38000	80

When PC B sends a packet to server F, the packet headers change twice:

Packet sent by B:

Source IP	Destination IP	Source Port	Destination Port
10.1.1.3	5.1.1.1	37000	80

Packet on the Internet:

Source IP	Destination IP	Source Port	Destination Port
4.1.1.1	5.1.1.1	41000	80

Packet received by F:

Source IP	Destination IP	Source Port	Destination Port
4.1.1.1	10.1.1.1	41000	38000

This example demonstrates how NAT allows private IPs to communicate through the public Internet, using port translation to multiplex multiple connections through a single public IP address.

6.2 TCP Congestion Control in a Network with Multiple Flows

Problem (Exame1_2023Resolucao.pdf, Problem 1):

Analyze two TCP Reno sessions sharing a link between routers R1 and R2 with capacity 25 Mbit/s and propagation delay 40 ms. Determine bandwidth allocation and behavior.

Solution:

Given:

- Link capacity: 25 Mbit/s
- Propagation delay: 40 ms (each direction)
- R1 buffer size: 250 KB
- Two TCP Reno flows sharing the link

Part 1: Minimum sum of window sizes for full utilization

For full utilization, the sum of congestion windows must at least equal the bandwidth-delay product (BDP):

$$\begin{aligned}
\text{RTT} &= 2 \times \text{propagation delay} = 2 \times 40 \text{ ms} = 80 \text{ ms} && (48)\\
\text{BDP} &= \text{Capacity} \times \text{RTT} && (49)\\
&= 25 \text{ Mbit/s} \times 0.08 \text{ s} && (50)\\
&= 2 \text{ Mb} = 250 \text{ KB} && (51)
\end{aligned}$$

Therefore, $w_1 + w_2 \geq 250$ KB for full utilization.

Part 2: Window size that causes packet loss

Packet loss occurs when the router buffer fills up. With a 250 KB buffer:

$$\begin{aligned}
\text{Queuing delay} &= \frac{\text{Buffer size}}{\text{Capacity}} && (52)\\
&= \frac{250 \text{ KB}}{25 \text{ Mbit/s}} && (53)\\
&= \frac{250 \times 8 \text{ Kb}}{25 \text{ Mbit/s}} && (54)\\
&= 80 \text{ ms} && (55)
\end{aligned}$$

The total RTT with full buffer becomes:

$$\begin{aligned}
\text{RTT with full buffer} &= \text{Base RTT} + \text{Queuing delay} && (56)\\
&= 80 \text{ ms} + 80 \text{ ms} = 160 \text{ ms} && (57)
\end{aligned}$$

The new BDP:

$$\begin{aligned}
\text{BDP with full buffer} &= 25 \text{ Mbit/s} \times 0.16 \text{ s} && (58)\\
&= 4 \text{ Mb} = 500 \text{ KB} && (59)
\end{aligned}$$

Therefore, packet loss occurs when $w_1 + w_2 = 500$ KB.

Part 3: Effective throughput with packet loss

When packet loss occurs, TCP's congestion control mechanism activates:

- $w_1 + w_2$ oscillates between 250 KB and 500 KB
- The link remains fully utilized because even at the minimum, $w_1+w_2 =$ 250 KB equals the BDP

Therefore, the effective throughput is 25 Mbit/s (100% utilization).

Part 4: ACK rate

With segment size 250 bytes:

$$\begin{aligned}
\text{Number of segments per second} &= \frac{\text{Throughput}}{\text{Segment size}} && (60)\\
&= \frac{25 \times 10^6 \text{ bits/s}}{250 \times 8 \text{ bits/segment}} && (61)\\
&= \frac{25 \times 10^6}{2000} \text{ segments/s} && (62)\\
&= 12,500 \text{ segments/s} && (63)
\end{aligned}$$

Each segment generates one ACK, so the ACK rate is 12,500 ACKs per second.

Part 5: Long-term bandwidth allocation

TCP Reno with the same RTT exhibits fairness in bandwidth allocation. The capacity is shared equally among flows:

- Flow 1: 12.5 Mbit/s (50%)
- Flow 2: 12.5 Mbit/s (50%)

This is due to the synchronized window adjustments when both flows experience packet loss simultaneously.