National University of Computer and Emerging Sciences
Islamabad Campus

Computer Networks (CS3001)

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

Sessional-II Exam

Total Time (Hrs): 1
Total Marks: 45
Total Questions: 6

Date: Nov 5, 2024

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

Do not write below this line.

Attempt all the questions.

Section-I: Application Layer

Q1: **[5 Marks]**

Consider a file of size F = 1 * 10^9 bits and upload rates u_s = 2 MB/s, u_i = 1.5 MB/s for i = 1 to N and download rates d_{min} = 0.8 MB/s. Calculate the distribution time D_{P2P} (Peer-to-Peer File Distribution) if N is 1,000. (assume 1 byte = 8 bits, and 1000000 bytes = 1 Megabyte).

Conversion **[1 Mark]**:

- File size *F* = 1×10^9 bits,
- Upload rate of the server u_s = 2 MB/s = 2×10^6 bytes/s = 2×8×10^6 bits/s = 1.6×10^7 bits/s,
- Upload rate of each peer u_i = 1.5 MB/s = 1.5×8×10^6 bits/s = 1.2×10^7 bits/s,
- Download rate d_{min} = 0.8 MB/s = 0.8×8×10^6 bits/s = 6.4×10^6 bits/s,
- Number of peers *N* = 1,000.

Using the formula for D_{P2P}:

D_{P2P} = max{ F/u_s, F/d_{min}, $NF/(u_s + \sum u_i)$ }

1. Calculate F/u_s **[0.5 Mark]**:

   1×10^9 bits / 1.6×10^7 bits/s = 62.5 seconds
2. Calculate F/d_{min} **[0.5 Mark]**:

   1×10^9 bits / 6.4×10^6 bits/s = 156.25 seconds
3. Calculate $NF/(u_s + \sum u_i)$ **[2 Mark]**:

   u_s = 1.6×10^7 bits/s
   $\sum u_i$ = 1,000×1.2×10^7 bits/s = 1.2×10^10 bits/s
   $u_s + \sum u_i$ = 1.6×10^7 + 1.2×10^10 = 16,000,000 + 1,200,000,000 = 1.2016 x 10^10 bits/s
   NF = 1,000 × 1×10^9 = 1×10^12 bits

$NF/(u_s + \sum u_i) = 1\times10^{12}$ bits / 1.2016×10^{10} bits/s = 83.22 seconds

Therefore, D_{P2P} is the maximum of these three values:
D_{P2P} = max { 62.5, 156.25, 83.22 } **[1 Mark]**
So D_{P2P} is approximately 156.25 seconds.

Q2: **[5 Marks]**

Q2: Provide short answers to the following questions [5Marks]:

1. What is the primary reason CDN nodes have copies of videos at multiple geographically distributed sites?

Answer: To reduce latency and network congestion for users by serving content closer to them.

2. How might the client-side buffer size impact video streaming performance?

Answer: A larger buffer can compensate for network congestion and variable delays but increases startup latency.

3. What is the main advantage of temporal coding in video streaming?

Answer: It reduces the bit rate needed for transmitting subsequent frames by sending only differences.

4. Why is DASH (Dynamic Adaptive Streaming over HTTP) beneficial for video streaming?

Answer: It allows clients to select the best quality stream based on their current bandwidth.

5. What is the role of a manifest file in DASH?

Answer: It provides URLs for different video chunks, enabling dynamic stream selection.

Section-II: Transport Layer

Q3: **[10 Marks]**

a. In the second segment sent from Host A to B, what are the sequence number, source port number, and destination port number? [2 Marks]

- Sequence number: 207
- Source port number: 302
- Destination port number: 80

b. If the first segment arrives before the second segment, in the acknowledgement of the first arriving segment, what is the acknowledgment number, the source port number, and the destination port number? [2 Marks]

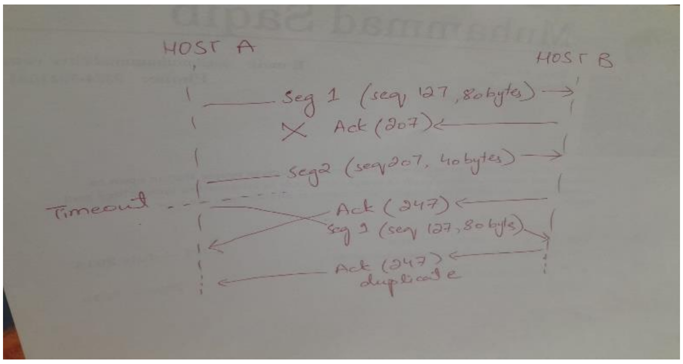
- Acknowledgment number: 207
- Source port number: 80

- Destination port number: 302

c. If the second segment arrives before the first segment, in the acknowledgement of the first arriving segment, what is the acknowledgment number? [2 Marks]

- Acknowledgment number: 127

d. Suppose the two segments sent by A arrive in order at B. The first acknowledgment is lost and the second acknowledgment arrives after the first timeout interval. Draw a timing diagram, showing these segments and all other segments and acknowledgments sent. (Assume there is no additional packet loss.) For each segment in your figure, provide the sequence number and the number of bytes of data; for each acknowledgement that you add, provide the acknowledgment number. [4 Marks]



Q4: **[5 Marks]**

Redraw and complete the following sender and receiver's FSMs for Reliable Data Transfer rdt 2.2**.**

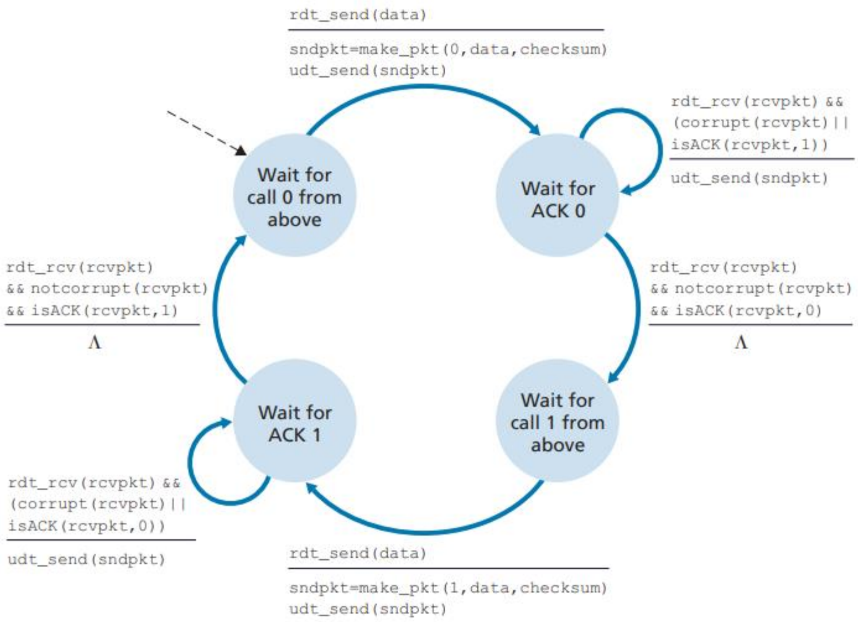


Figure 3.13 ♦ rdt2.2 sender

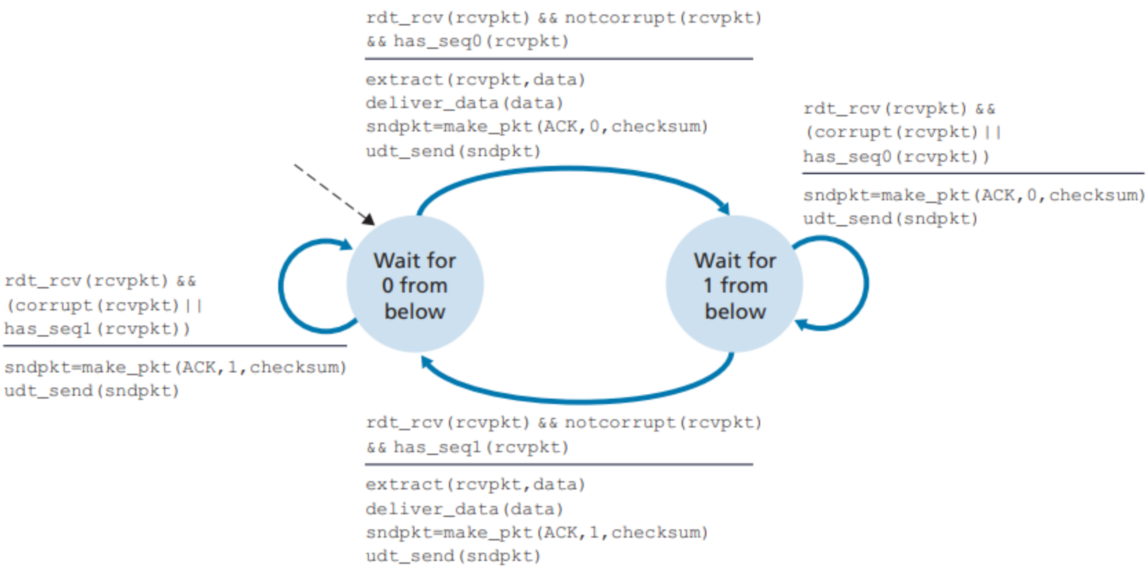


Figure 3.14 ♦ rdt2.2 receiver

Q5: **[10 Marks]**

Consider the following plot of TCP window size as a function of time for two TCP connections A and B. In this problem, we will suppose that both TCP senders are sending large files. We also assume that the packet loss events are independent in connection A and B.

a) Considering the above values of congestion window (cwnd) for these connections, can we identify the type of TCP connections (Reno or Tahoe) that have been used by connection A and B? Justify your answers. [2 Marks]

- For **Connection A**, after packet loss, the cwnd drops significantly but does not go back to 1 segment immediately, indicating that it might be using **TCP Reno** with Fast Recovery.
- For **Connection B**, however, we observe a larger drop in cwnd (possibly back to 1), suggesting that **TCP Tahoe** may be in use for Connection B, as it doesn't seem to use Fast Recovery.

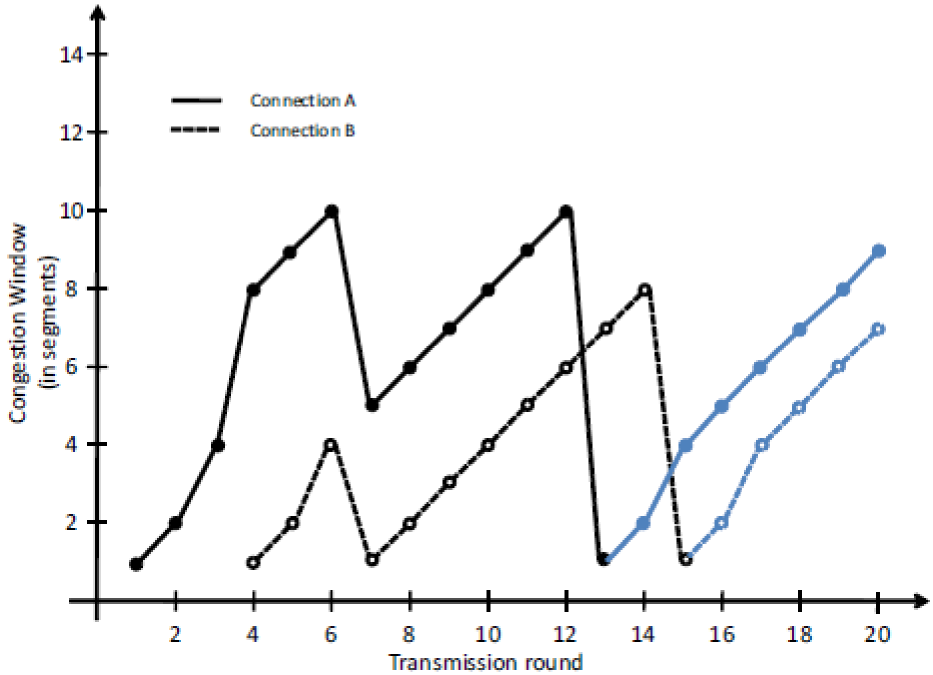
b) How the value of Threshold parameter varies between the 1st and the 14th transmission rounds for each connection? [2 Marks]

Connection A: The value of Threshold is 8 between the first and the sixth transmission round. It is 5 between the sixth and the fourteenth transmission round.
Connection B: With the above plot we cannot identify the exact value of Threshold for connection B between the first and the sixth transmission round. It could have any value larger than 4. From the sixth to the fourteenth transmission round, it is 2 and at the fourteenth transmission round it is 4.

c) At the 12th transmission round for connection A, is segment loss detected by a triple duplicate ACK or by timeout? Justify your answer. [1 Mark]

It is detected by timeout, because cwnd has dropped to 1 at the 13th transmission round.

d) Draw (***using the figure above***) the cwnd values of both connections up to the 20th transmission round, considering that there is no further timeout or duplicate ACK situation for any of the connections.[2 Marks]



e) Consider your solution in part d; assume that the segment size is 1460 bytes and that a total of 87600 bytes have been successfully transmitted over connection A before the 13th transmission round. At which transmission round the cumulative amount of the successful transmitted data is equal to 163520 bytes? Again we assume that there is neither timeout nor duplicate ACK after the 13th transmission round.[3 Marks]

87600 is equal to 87600/1460 = 60 segments. We would like to know at which transmission round the 163520/1460 = 112 segments will be transmitted.
Assume x is the number of segments transmitted in that particular round (the one we are looking for) then, the sum of segments transmitted after round 13 would be:
1 + 2 + 4 + 5 + 6 + 7 + 8 + ……… + x = 112 - 60 = 52
x(x + 1)/2 - 3 = 52
x = 10.
Now round 21 is the transmission round in which 10 segments will be transmitted and cumulative transmitted data will become 163520 bytes.

Section-III: Network Layer

Q6: **[10 Marks]**

The Capital development authority (CDA) Islamabad is planning a "Smart City" network that will connect various city services, including traffic lights, surveillance cameras, public Wi-Fi, and environmental sensors. The network needs to be flexible, scalable, secure, and easily manageable, as it will support thousands of internet devices across the city. In the context of the smart city network, Provide short answer to the following questions [**10 marks**]

a) Which approach offers a more cost-effective solution for scaling the network over time traditional routing or SDN? Why? (1 mark)

SDN is more cost-effective for scaling over time because it allows centralized control and easy network reconfiguration, reducing the need for additional hardware as the network grows.

b) In an area with limited power and computational resources, which approach (traditional routing or SDN) would be more practical to deploy? Why? (2 marks)

Traditional routing would be more practical in areas with limited power and computational resources, as it has distributed control, reducing the need for high-power centralized processing and allowing simple, local decision-making within each router.

c) The city's network routers must prioritize emergency signals over regular data, ensuring critical packets are processed quickly. In traditional routing, Which parts of the router handle decisions about traffic priority versus the actual forwarding of packets, and are these functions typically software-based or hardware-based? (2 marks)

In traditional routing, the **control plane** (software-based) sets traffic priority policies, while the **data plane** (hardware-based) forwards packets based on these policies.

d) Data from various sensors, cameras, and city services flows into the router, it needs to be directed to different parts of the network. Which router component is responsible for determining where each incoming packet should go? (1 mark)

The Input port/**forwarding table** within the router is responsible for determining the direction for each incoming packet based on destination information

e) In Islamabad's Blue Area, where real-time data management is crucial for directing traffic signals and ensuring swift routes to key locations such as PIMS Hospital, which router switching fabric (e.g., crossbar, shared memory, or bus) would best handle this high-priority data flow efficiently and why? (2 marks)

Crossbar switching fabric is best in Islamabad's Blue Area for high-priority, real-time data, as it enables simultaneous data paths without congestion, ensuring speed and reliability for critical traffic signals and emergency routes.

f) In residential sectors of Islamabad, such as G-10, where numerous devices (like public Wi-Fi hotspots, CCTV cameras, and air quality sensors) transmit data simultaneously, what challenges might arise with using a bus-based switching fabric? (1 mark)

In residential areas like G-10, a **bus-based switching fabric** may face challenges with congestion and latency when multiple devices transmit simultaneously, limiting throughput and slowing data processing

g) In the context of the smart city network, what role does the input port play when incoming packets from traffic lights and surveillance systems are delayed due to congestion at the output port? (1 mark)

The **input port** queues incoming packets temporarily when the output port is congested, holding data from traffic lights and surveillance systems until resources are available for processing.

The End ☺