

CS3640

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# Transport Layer (3): Congestion Control

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*Computer Science*

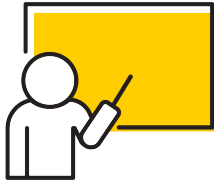
*The University of Iowa*

# Midterm Format and Question

Category	Example questions and topics	Weight
Networking <b>Principles</b>	<i>End-to-end argument; Routing and forwarding; Protocol layering</i>	25%
Networking <b>Protocols</b>	<i>TCP vs. UDP; HTTP headers and extensions; Designing CDNs</i>	25%
Networking <b>Problems</b>	<i>Understanding delays; Mitigating congestion; Security challenges</i>	25%
Network <b>Programs</b>	<i>Explain how traceroute works; Socket programming; Video Streaming</i>	25%

*There will be an optional **bonus question** carrying 10% extra points (expect it to be challenging)*

# Preparations and logistics



Revisit the **lectures and slides**:

<https://shastri.info/teaching/cs3640>



Read the **textbook**:

[Kurose-Ross chapters 1-3](#)



Midterm **schedule**:

[3/10 Thursday at 6:30PM in 3655 SC](#)

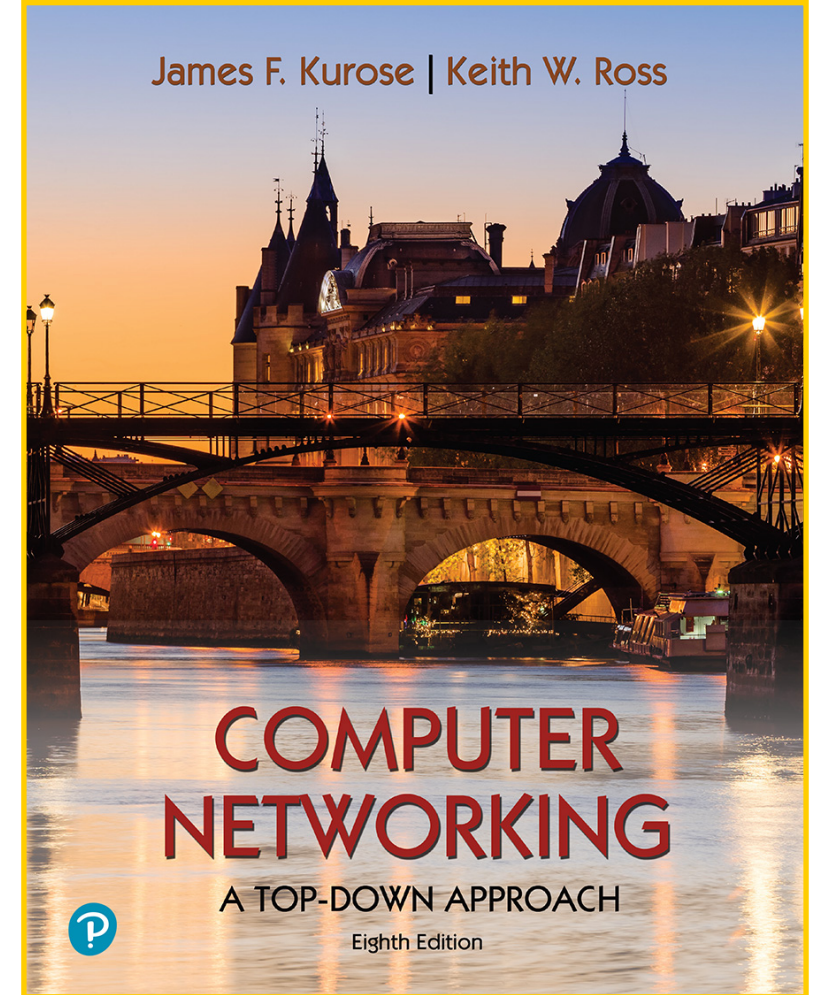
It is a 1-hour pen-and-paper exam (closed book, closed notes, closed electronics)

# Lecture goals

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*continued discussion of reliable data transfer, followed by congestion control*

- *Pipelined RDT protocols*
- *Congestion control*

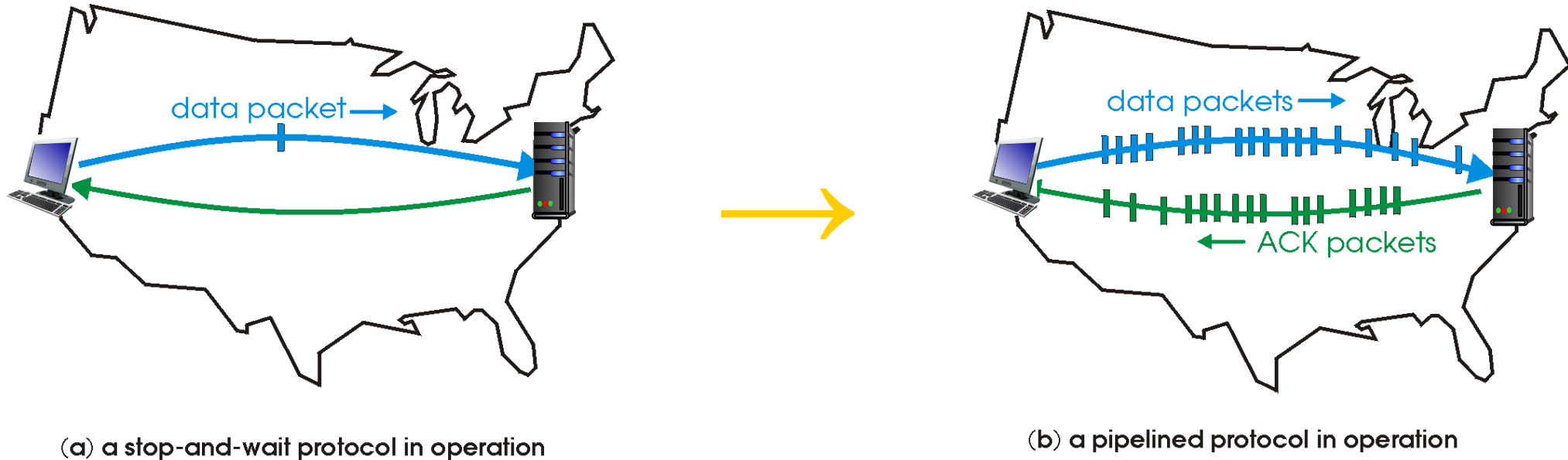


Chapter 3.4, 3.6

# Reliable Data Transfer: techniques and mechanisms

<b>Checksum</b>	detect bit errors in packet
<b>ACK</b>	report reception of a packet correctly
<b>NAK</b>	report error(s) in a received packet
<b>Sequence numbers</b>	detect any missing packets
<b>Timers</b>	detect and recover from packet loss
<b>Pipelining</b>	increase channel utilization

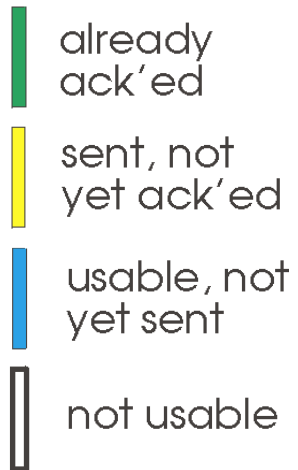
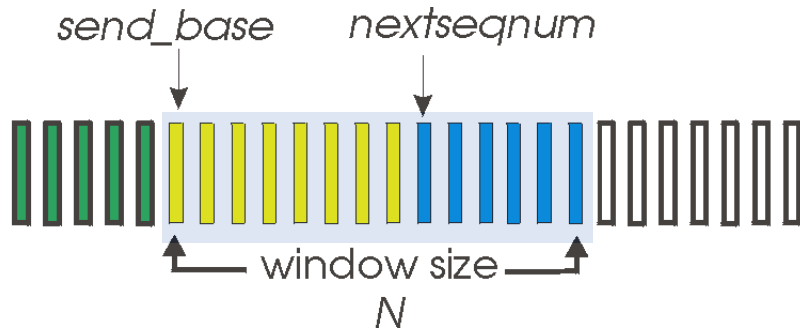
# Increased Utilization w/ Pipelining



## Pipelined transfer of packets, where several could be in-transit and unacknowledged

- change-1: increase the range of sequence# in RDT protocols
- change-2: add buffer capability to both sender and receiver sides
- two variants: *Go-Back-N* and *Selective Repeat*

# Go-Back-N (GBN) Protocol

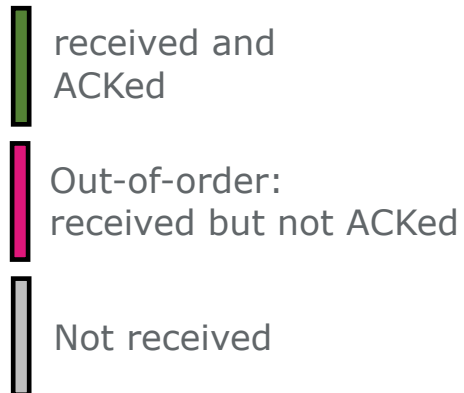
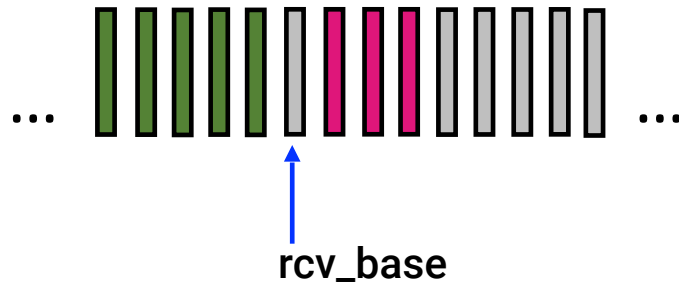


## Sender

- **Packet window:** sender defines a window of up to  $N$  consecutive transmitted but not yet ACKed packets
- **Cumulative ACK:** if the sender receives  $\text{ACK}(k)$ , then it considers all packet up to  $k$  to be ACKed i.e., it moves the window forward to  $k+1$
- **Timer:** sender always maintains a timer for the oldest not yet ACKed packet
- **Retransmissions:** Upon timer interrupt, sender retransmits the first not yet ACKed packet followed by all higher seq# packets in window



# Go-Back-N (GBN) Protocol



## Receiver

- **Out-of-order packets:** are typically discarded
- Discarding results in lower channel utilization, whereas buffering introduces extra state management
- **Acknowledging:** always send ACK for any correctly-received packet; however, the ACK will always carry the seq# of the highest in-order packet
- This ACK scheme may generate duplicate ACKs



# Go-Back-N in action

sender window (N=4)

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

sender

send pkt0

send pkt1

send pkt2

send pkt3

(wait)

rcv ack0, send pkt4

rcv ack1, send pkt5

ignore duplicate ACK



pkt 2 timeout

send pkt2

send pkt3

send pkt4

send pkt5

receiver

receive pkt0, send ack0

receive pkt1, send ack1

receive pkt3, discard, (re)send ack1

receive pkt4, discard, (re)send ack1

receive pkt5, discard, (re)send ack1

rcv pkt2, deliver, send ack2

rcv pkt3, deliver, send ack3

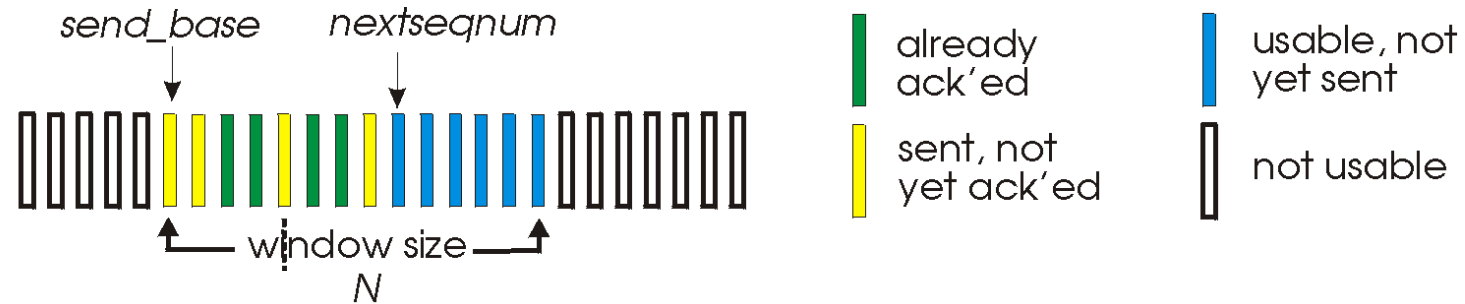
rcv pkt4, deliver, send ack4

rcv pkt5, deliver, send ack5

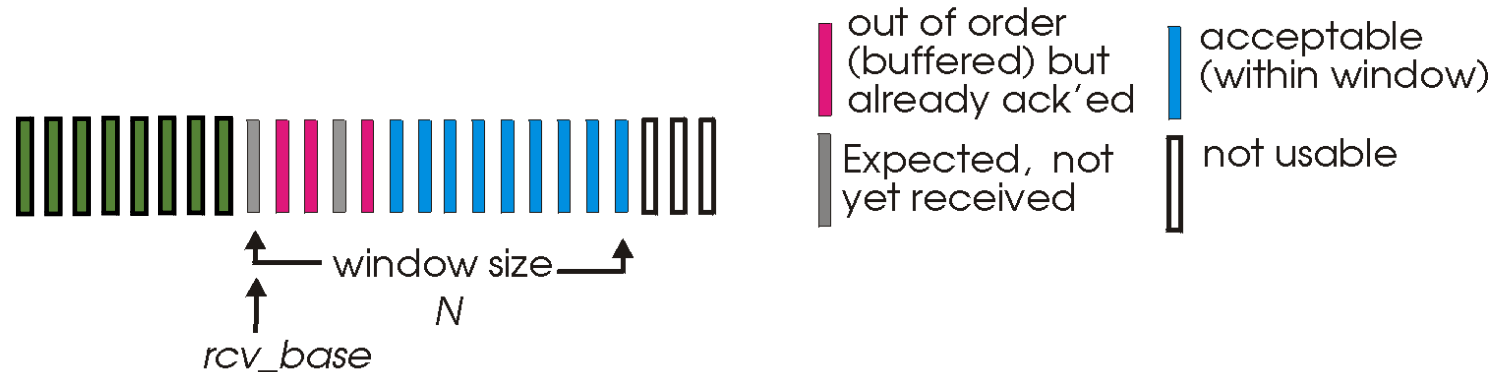
# Selective Repeat (SR) Protocol

	Go-Back-N	Selective Repeat
ACKs	<b>Cumulative</b> i.e., ACK(k) will ACK all packets up to and including #k	<b>Individual</b> i.e., ACK(k) just ACKs packet #k
Out of order packets	Receiver discards all out of order packets	Buffers out-of-order packet for later delivery
Buffer size	Sender buffer = N; receiver buffer = 1	Sender buffer = N; Receiver buffer = N
Sender timer	Set for only the oldest unacknowledged packet	Set for every transmitted packet

# Selective Repeat: sender and receiver windows



(a) sender view of sequence numbers



(b) receiver view of sequence numbers

# Selective Repeat: sender and receiver

## sender

*data from above:*

- if next available seq # in window, then send packet

*timeout(n):*

- resend packet n, restart timer

*ACK(n) in [sendbase, sendbase+N]:*

- mark packet n as received
- remove the timer for packet n
- if n smallest unACKed packet, advance window base to next unACKed seq #

## receiver

*packet n in [rcvbase, rcvbase+N]*

- send ACK(n)
- out-of-order: buffer
- in-order: deliver (also deliver buffered, in-order packets), advance window to next not-yet-received packet

*packet n in [rcvbase-N, rcvbase-1]*

- ACK(n)

*otherwise:*

- ignore

# Selective Repeat in action

sender window (N=4)

0 1 2 3 4 5 6 7 8  
0 1 2 3 4 5 6 7 8  
0 1 2 3 4 5 6 7 8  
0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8  
0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8  
0 1 2 3 4 5 6 7 8  
0 1 2 3 4 5 6 7 8  
0 1 2 3 4 5 6 7 8

sender

send pkt0  
send pkt1  
send pkt2  
send pkt3  
(wait)

rcv ack0, send pkt4  
rcv ack1, send pkt5

record ack3 arrived



pkt 2 timeout  
send pkt2  
(but not 3,4,5)

receiver

receive pkt0, send ack0  
receive pkt1, send ack1  
receive pkt3, buffer, send ack3  
receive pkt4, buffer, send ack4  
receive pkt5, buffer, send ack5

rcv pkt2; deliver pkt2,  
pkt3, pkt4, pkt5; send ack2

Q: what happens when ack2 arrives?

# Congestion Control

# Distinguishing **Flow Control** from **Congestion Control**



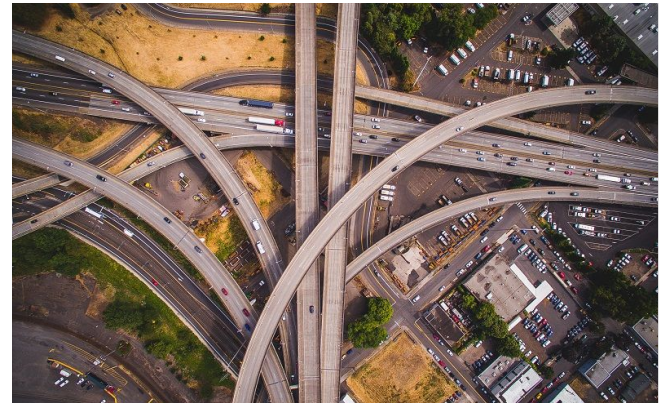
## flow control

*one sender too fast  
for one receiver*



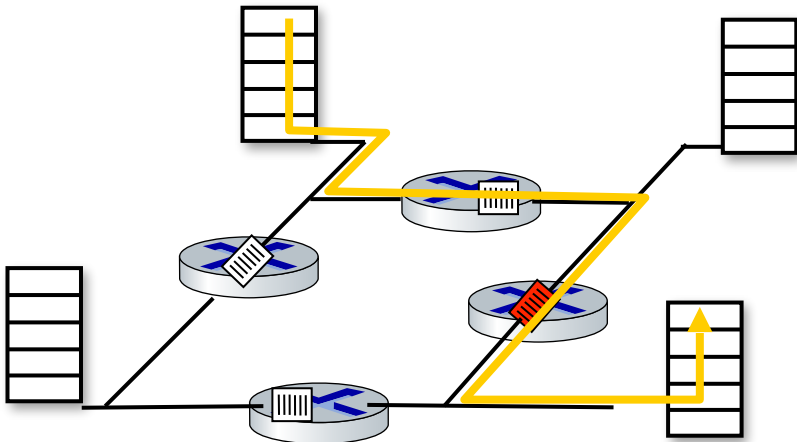
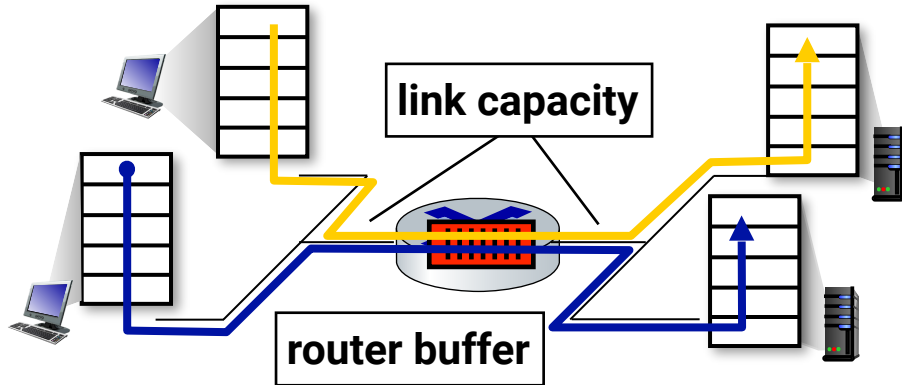
## congestion control

*many senders, too fast for  
the network to handle*





# Causes and Implications of Congestion



## 1. When the router's link capacity gets saturated

- packets experience queuing delays
- senders and receivers experience lower throughput

## 2. When the router's buffer overflows

- some packets get dropped, causing sender to retransmit
- some packets get delayed, causing sender to retransmit duplicate packets
- both cases lead to lower throughput and higher latency

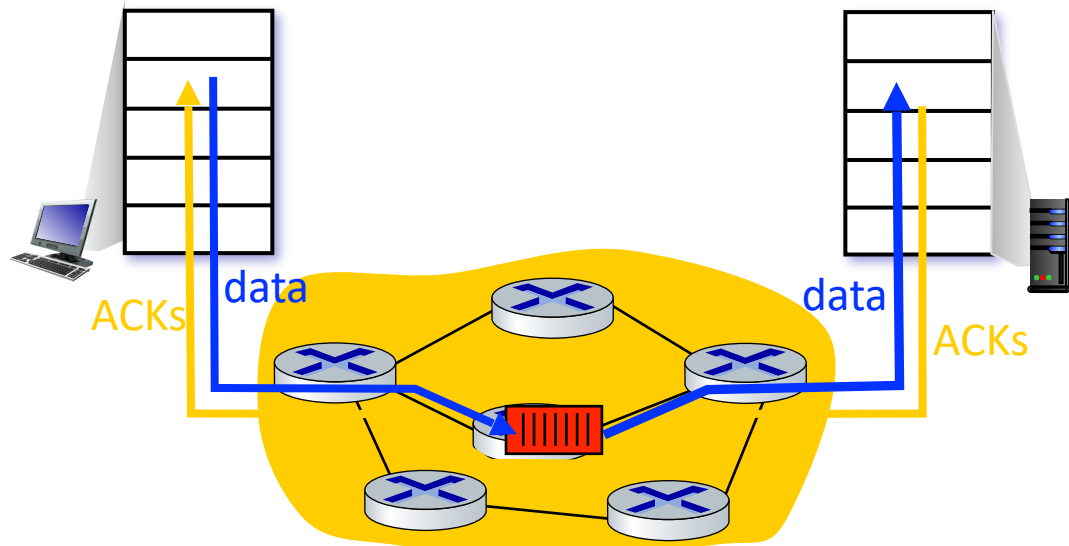
## 3. When an upstream router drops a packet

- transmission capacity used up by the packet so far is wasted
- this reduces the utilization of the overall network

# Approaches towards congestion control

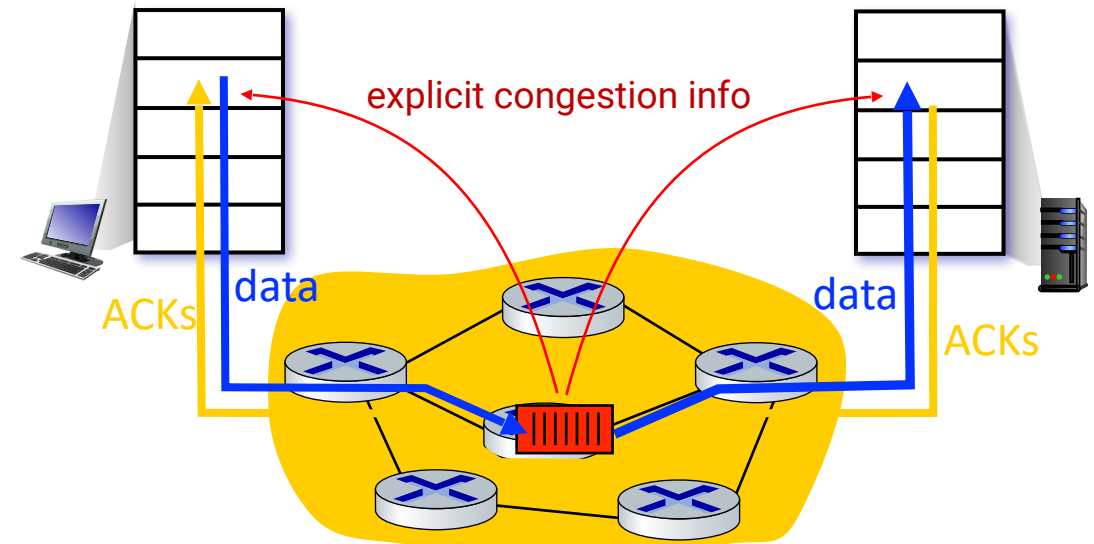
## End-to-end (reactive)

- no explicit feedback from network layer
- congestion is **inferred** from observed loss and delay (via timeouts, duplicate ACKs, or RTT measurements)
- this is the approach taken by the original TCP



## Network-assisted (proactive)

- a congested router provides **direct** feedback to all hosts with flows passing through it
- router acts proactively to indicate its congestion level or to explicitly set sending rate
- RFC3168 (in 2001) added explicit congestion notification (ECN) to TCP and IP

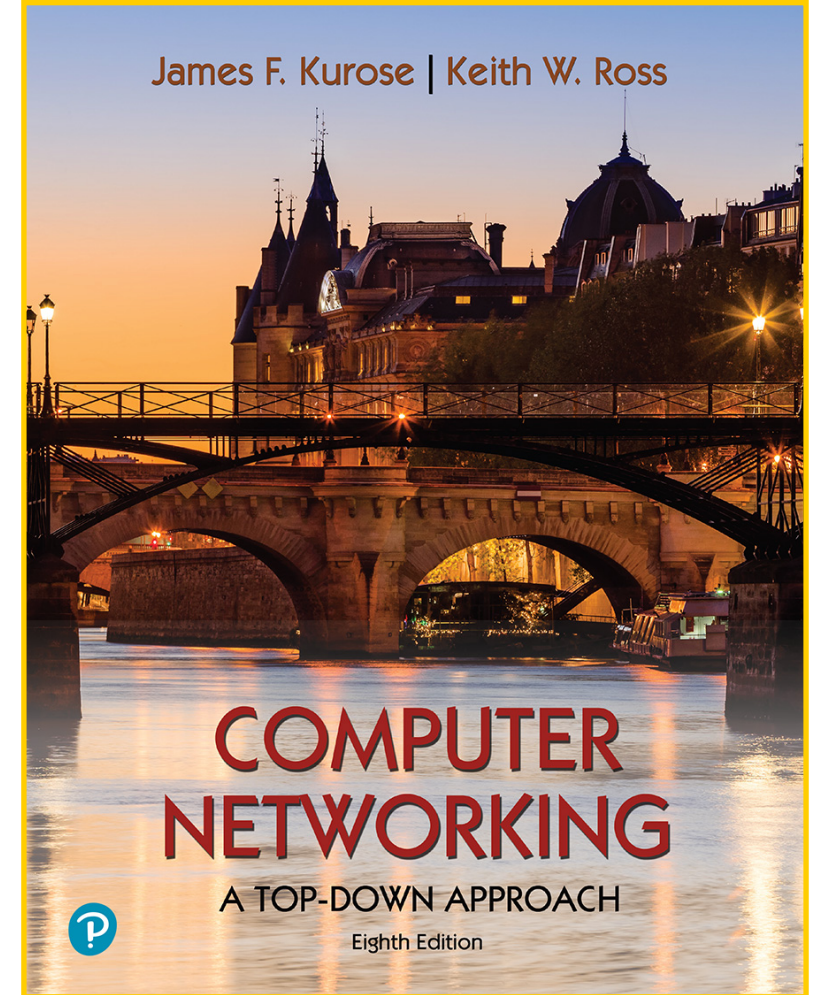


# Next lecture

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*from principles to practice: design and operation of TCP*

- *Protocol structure*
- *Connection management*
- *Reliable data transfer*
- *Flow and congestion control*



Chapters 3.5, 3.7

# **Spot Quiz (ICON)**