

CS3640

Transport Layer (3): Congestion Control

Prof. Supreeth Shastri

Computer Science

The University of Iowa

Lecture goals

continued discussion of reliable data transfer, followed by congestion control

- *Pipelined RDT protocols*
- *Congestion control*

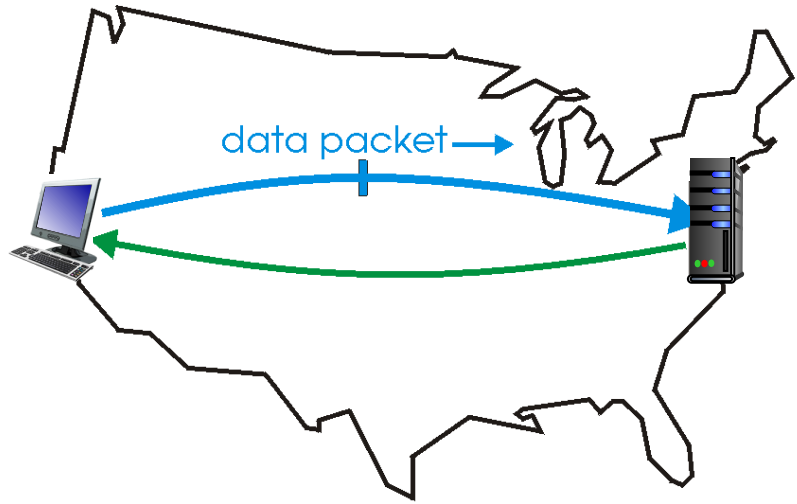


Chapter 3.4, 3.6

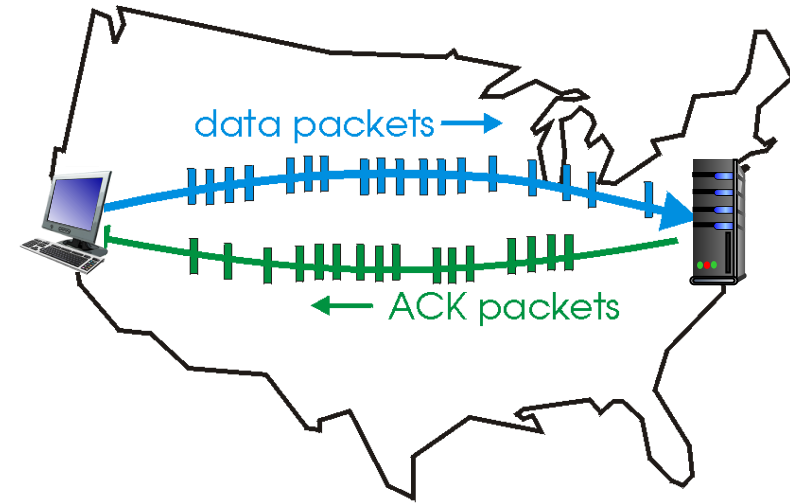
Reliable Data Transfer: techniques and mechanisms

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

Increased Utilization w/ Pipelining



(a) a stop-and-wait protocol in operation

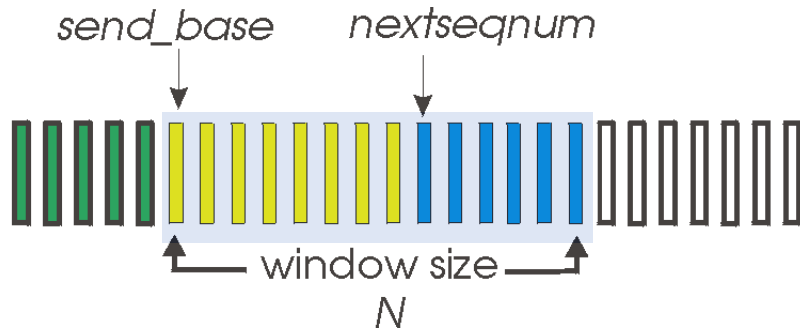


(b) a pipelined protocol in operation

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*

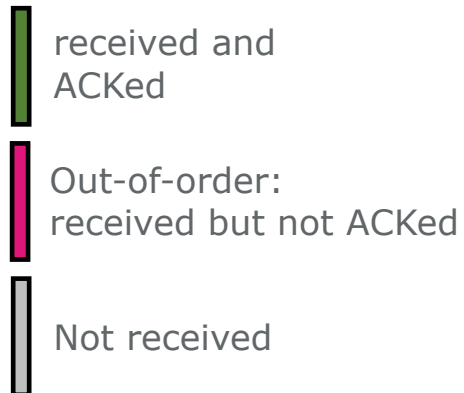
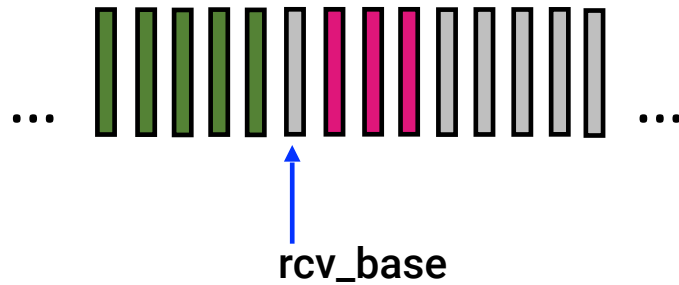
Go-Back-N (GBN) Protocol



Sender

- **Packet window:** sender defines a window of up to N consecutive transmitted but not yet ACKed packets
- **Cumulative ACK:** if the sender receives $ACK(k)$, 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

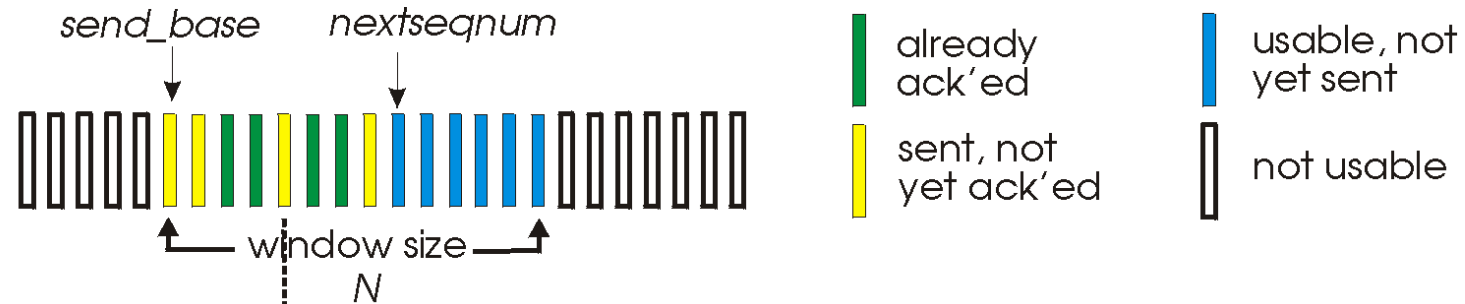
rcv pkt4, deliver, send ack4

rcv pkt5, deliver, send ack5

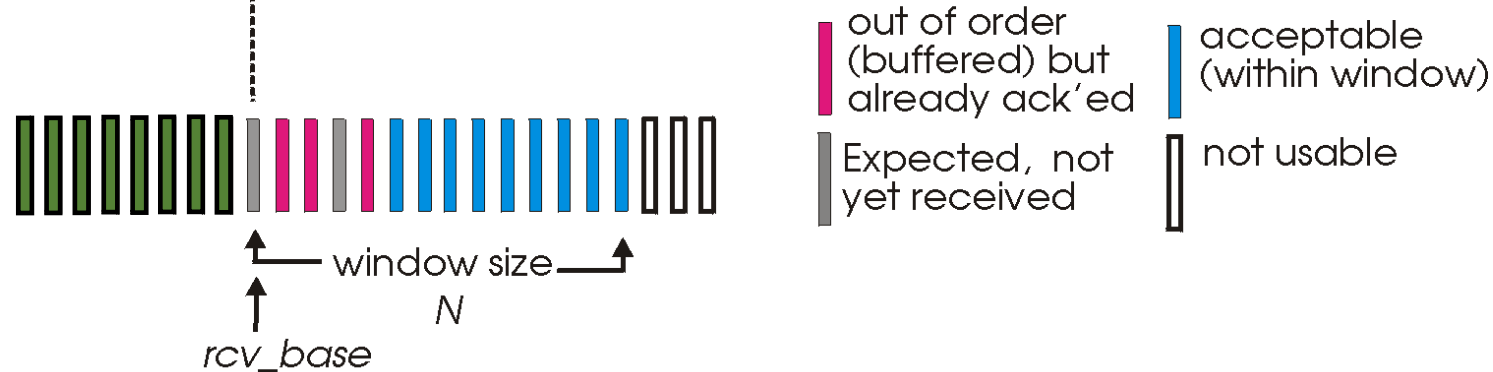
Selective Repeat (SR) Protocol

	Go-Back-N	Selective Repeat
ACKs	Cumulative i.e., ACK(k) will ACK all packets up to and including #k	Individual i.e., ACK(k) just ACKs packet #k
Sender timer	Set for only the oldest unacknowledged packet	Set for every transmitted packet
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

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
- if n smallest unACKed packet, advance window base to next unACKed seq #

receiver

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

- 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?

Selective Repeat: window size vs. sequence#

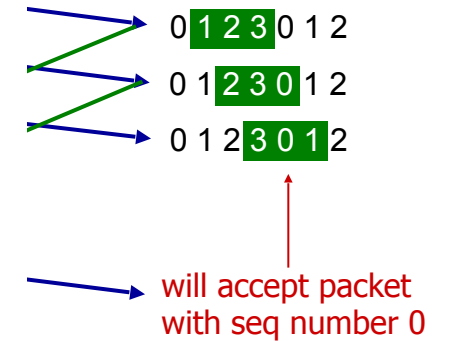
- Consider a window size of 3 and a sequence# set of {0, 1, 2, 3} for the SR protocol
- Remember that sender and receiver cannot see each other's windows nor the state
- Case-1 and Case-2 look identical from receiver's viewpoint \Rightarrow SR protocol is no longer reliable
- Solution:** sequence number set $> 2 * \text{window size}$

sender window
(after receipt)

0 1 2 3 0 1 2
0 1 2 3 0 1 2
0 1 2 3 0 1 2

0 1 2 3 0 1 2
0 1 2 3 0 1 2

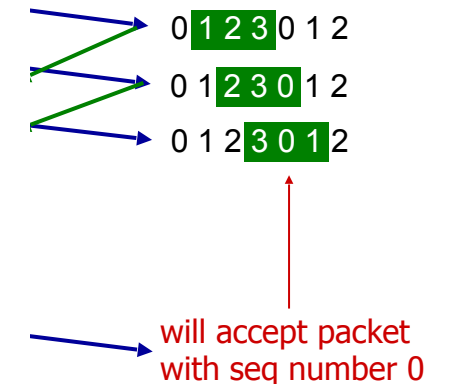
receiver window
(after receipt)



Case-1: no ACKs are dropped

0 1 2 3 0 1 2
0 1 2 3 0 1 2
0 1 2 3 0 1 2

timeout
retransmit |
0 1 2 3 0 1 2



Case-2: all ACKs are lost

Congestion Control

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



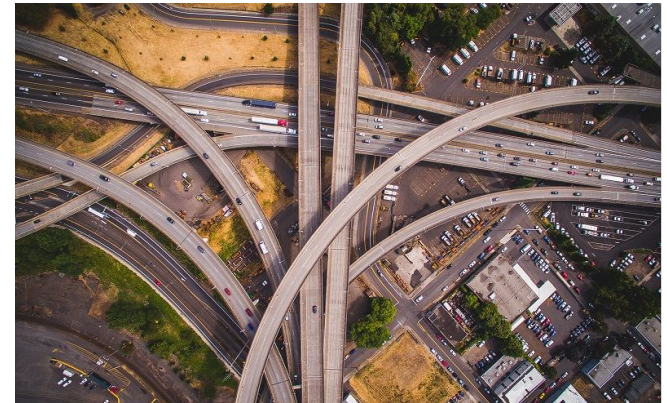
flow control

*one sender too fast
for one receiver*

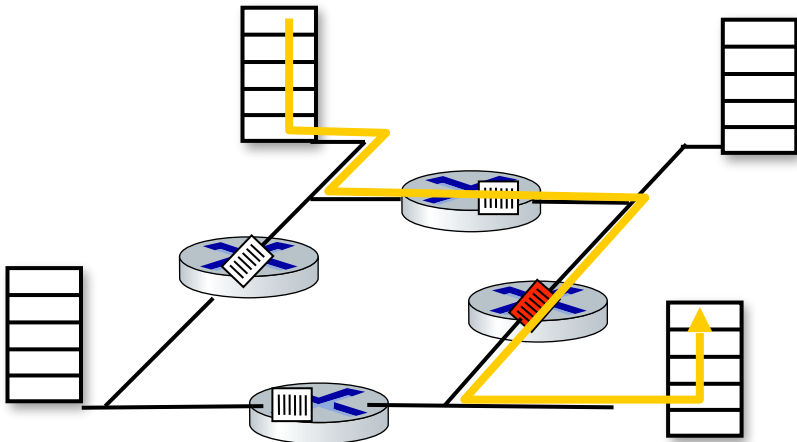
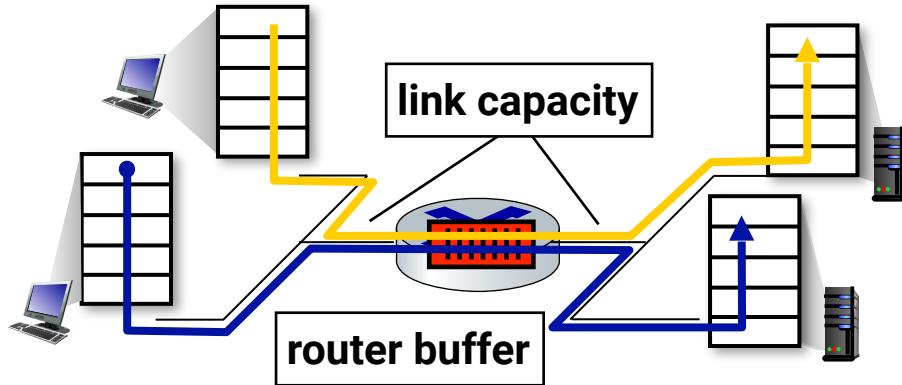


congestion control

*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

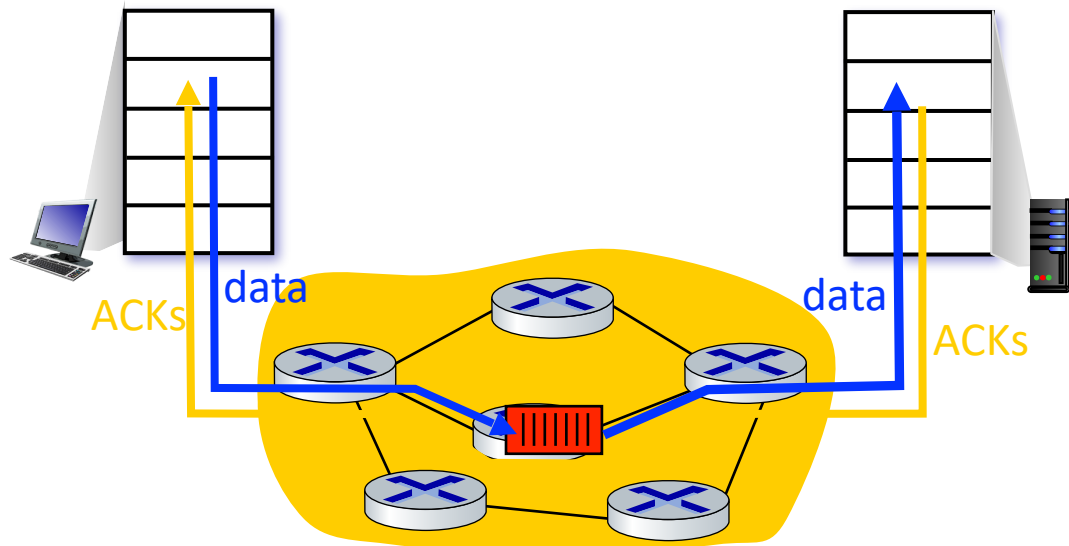
3. When an upstream router drops a packet

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

Approaches towards congestion control

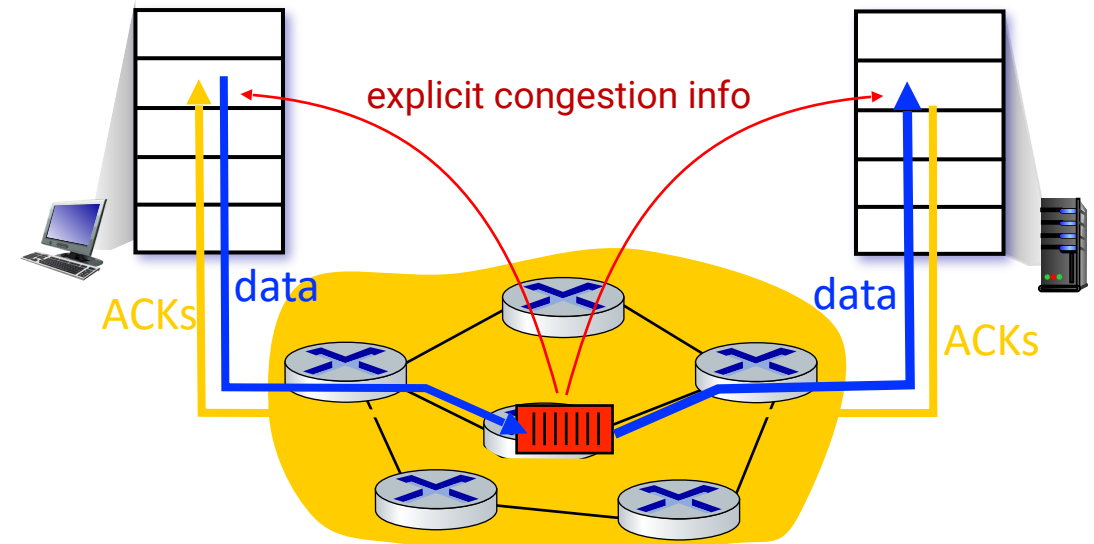
End-to-end (reactive)

- 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)

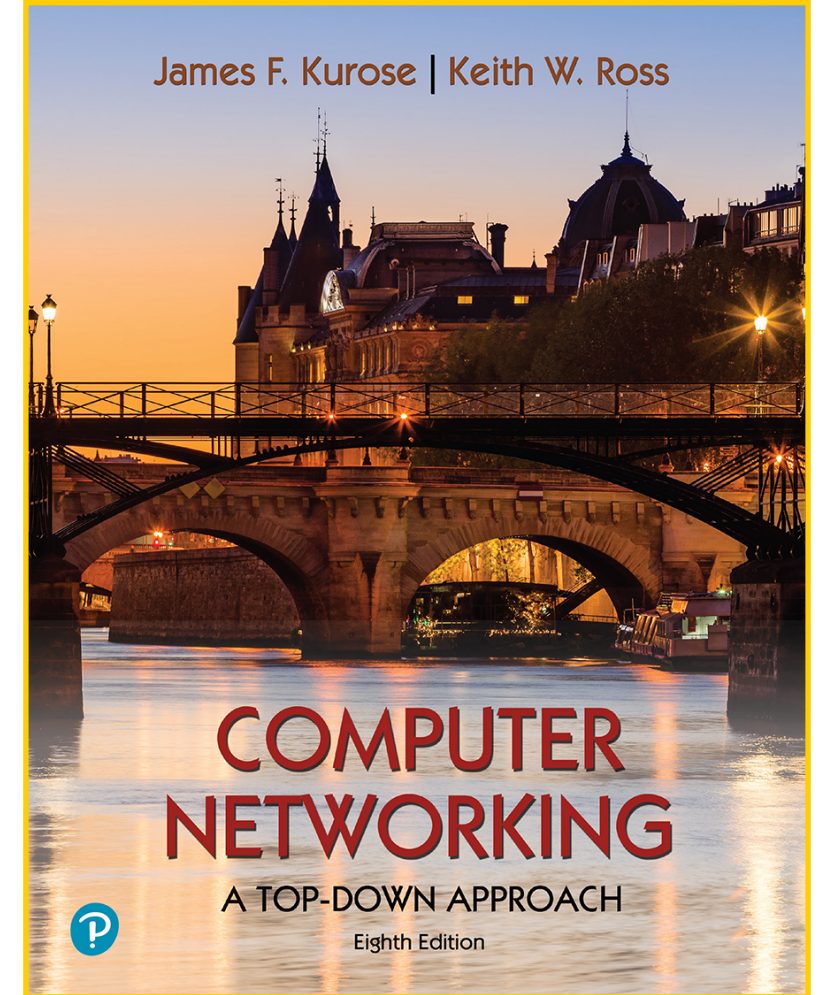
- a congested router provides **direct** feedback to all hosts with flows passing through it
- router acts proactively to indicate its congestion level or to explicitly set sending rate
- RFC3168 (in 2001) added explicit congestion notification (ECN) to TCP and IP



Next lectures

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)