<u>CSCE 463/612</u> <u>Networks and Distributed Processing</u> <u>Fall 2024</u>

Transport Layer V

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October 17, 2024

Chapter 3: Roadmap

- 3.1 Transport-layer services
- 3.2 Multiplexing and demultiplexing
- 3.3 Connectionless transport: UDP
- 3.4 Principles of reliable data transfer
- 3.5 Connection-oriented transport: TCP
 - Segment structure
 - Reliable data transfer
 - Flow control
 - Connection management

3.6 Principles of congestion control3.7 TCP congestion control

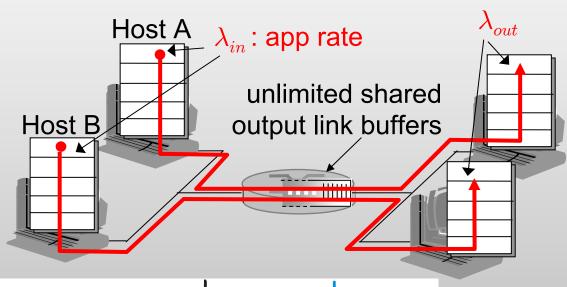
Principles of Congestion Control

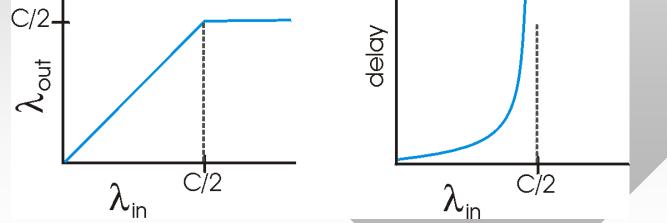
Congestion:

- Informally: "too many sources sending data too fast for the *network* to handle"
- Different from flow control!
- Manifestations:
 - Lost packets (buffer overflows)
 - Delays (queueing in routers)
- Important networking problem



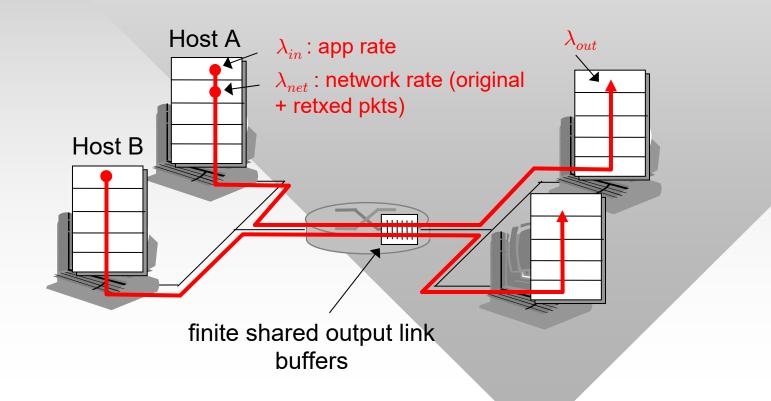
- Two senders, two receivers
- One router of capacity C, infinite buffers, no loss
- No retransmission



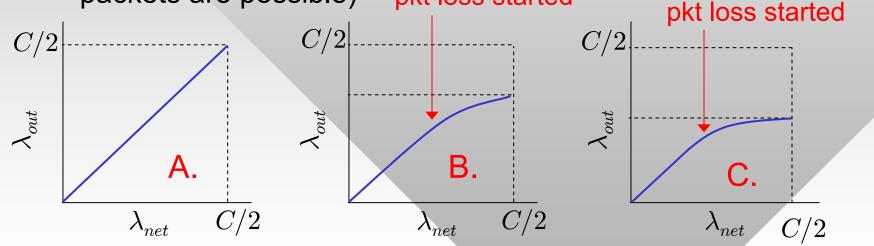


Cost 1: queuing delays in congested routers

- One router, *finite* buffers (pkt loss is possible now)
- Sender retransmission of lost packet
- During congestion $2\lambda_{net} = 2(\lambda_{in} + \lambda_{retx}) = C$



- We call λ_{out} goodput and λ_{net} throughput
 - <u>Case A</u>: pkts never lost while $\lambda_{net} < C/2$ (not realistic)
 - <u>Case B</u>: pkts are lost when λ_{net} is "sufficiently large," but timeouts are perfectly accurate (not realistic either)
 - Case C: same as B, but timer is not perfect (duplicate packets are possible) pkt loss started

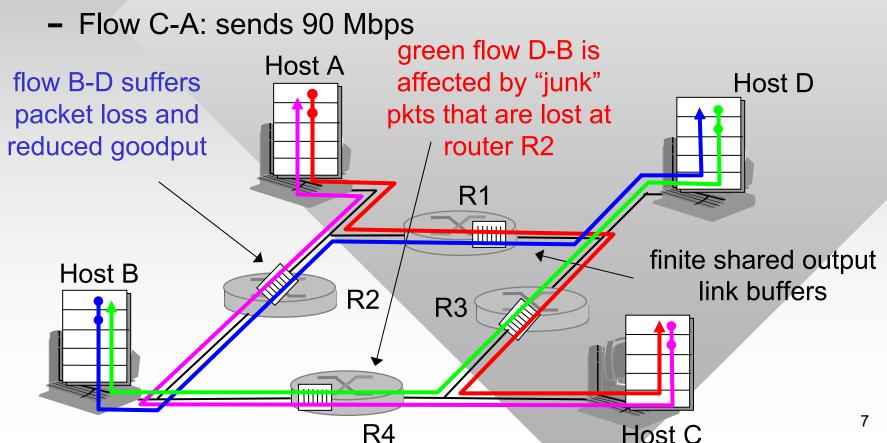


Cost 2: retransmission of lost packets and premature timeouts increase network load, reduce flow's own goodput

- Multihop case
 - Timeout/retransmit

<u>Cost 3</u>: congestion causes goodput reduction for *other* flows

- R2 = 50 Mbps, R1 = R3 = R4 = 100 Mbps



Approaches Towards Congestion Control

Two broad approaches towards congestion control:

End-to-end:

- No explicit feedback from network
- Congestion *inferred* by end-systems from observed loss/delay
 - Approach taken by TCP (relies on loss)

ATM = Asynchronous Transfer Mode

Network-assisted:

- Routers provide feedback to end systems
 - Single bit indicating congestion (DECbit, TCP/IP ECN)
 - Two bits (ATM)
 - Explicit rate senders should send at (ATM)

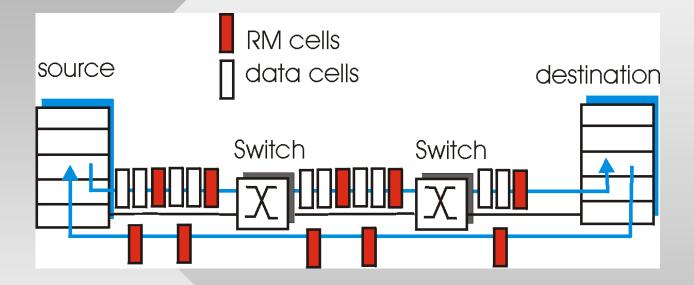
Case Study: ATM ABR Congestion Control

- For network-assisted protocols, the logic can be binary:
 - Path underloaded, increase rate
 - Path congested, reduce rate
- It can also be ternary
 - Increase, decrease, hold steady
 - ATM ABR (Available Bit Rate) profile

RM (resource management) packets (cells):

- Sent by sender, interspersed with data cells
- Bits in RM cell set by switches/routers
 - NI bit: no increase in rate (impending congestion)
 - CI bit: reduce rate (congestion in progress)
- RM cells returned to sender by receiver, with bits intact

Case Study: ATM ABR Congestion Control



- Additional approach is to use a two-byte ER (explicit rate) field in RM cell
 - Congested switch may lower ER value
 - Senders obtain the maximum supported rate on their path
- Issues with network-assisted congestion control?

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TCP Congestion Control

- TCP congestion control has a variety of algorithms developed over the years
 - TCP Tahoe (1988), TCP Reno (1990), TCP SACK (1992)
 - TCP Vegas (1994), TCP New Reno (1996)
 - High-Speed TCP (2002), Scalable TCP (2002)
 - FAST TCP (2004), TCP Illinois (2006)
- Many others: H-TCP, CUBIC TCP, L-TCP, TCP Westwood, TCP Veno (Vegas + Reno), TCP Africa
- Linux: BIC TCP (2004), CUBIC TCP (2008)
- Vista and later: Compound TCP (2005)
 - Server 2019 switched to CUBIC
- Google: BBR (2016)

TCP Congestion Control

- End-to-end control (no network assistance)
- Sender limits transmission:
 LastByteSent LastByteAcked ≤ CongWin
- CongWin is a function of perceived network congestion
- The effective window is the minimum of CongWin, flow-control window carried in the ACKs, and sender's own buffer space

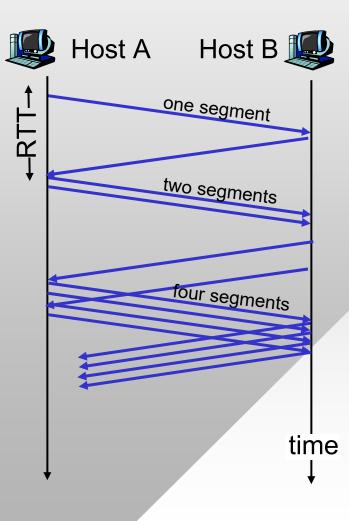
- How does sender
 perceive congestion?
 - Loss event = timeout
 or 3 duplicate acks
- TCP sender reduces rate (CongWin) after loss event
- <u>Three mechanisms:</u>
 - Slow start
 - Conservative after timeouts
 - AIMD (congestion avoidance)

TCP Slow Start

- When connection begins, CongWin = 1 MSS
 - Example: MSS = 500 bytes and RTT = 200 msec
 - Q: initial rate?
 - A: 20 Kbits/s
- Available bandwidth may be much larger than MSS/RTT
 - Desirable to quickly ramp up to a "respectable" rate
- Solution: Slow Start (SS)
 - When a connection begins, it increases rate exponentially fast until first loss or receiver window is reached
 - Term "slow" is used to distinguish this algorithm from earlier TCPs which directly jumped to some huge rate

TCP Slow Start (More)

- Let W be congestion window in pkts and B = CongWin be the same in bytes (B = MSS * W)
- Slow start
 - Double CongWin every RTT
- Done by incrementing CongWin for every ACK received:
 - W = W+1 per ACK (or B = B + MSS)
- <u>Summary:</u> initial rate is slow but ramps up exponentially fast



Congestion Avoidance

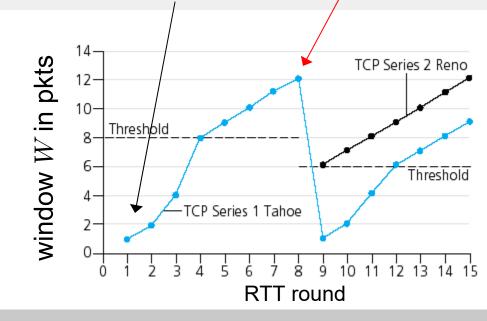
loss detected via triple dup ACK

previous timeout

- TCP Tahoe loss (timeout or triple dup ACK):
 - Threshold = CongWin/2
 - CongWin is set to 1 MSS
 - Slow start until threshold is reached; then move to linear probing

TCP Reno loss:

- Timeout: same as Tahoe
- 3 dup ACKs: CongWin is cut in half (method called fast recovery)



Fast Recovery Philosophy:

Three dup ACKs indicate that network is capable of delivering subsequent segments

Timeout before 3-dup ACK is more alarming

TCP Reno AIMD (Additive Increase, Multiplicative Decrease)

Additive increase: increase CongWin by 1 MSS every RTT in the absence of loss events: probing <u>Multiplicative decrease</u>: cut CongWin in half after fast retransmit (3-dup ACKs)

Peaks are different: # of flows or RTT changes

congestion window 24 Kbytes 16 Kbytes 8 Kbytes time

TCP Reno Equations

- To better understand TCP, we next examine its AIMD equations (congestion avoidance)
- General form (loss detected through 3-dup ACK):

$$W = egin{cases} W + rac{1}{W} & ext{per ACK} \ W/2 & ext{per loss} \end{cases}$$

- Reasoning
 - For each window of size W, we get exactly W acknowledgments in one RTT (assuming no loss!)
 - This increases window size by roughly 1 packet per RTT
- In general, many other protocols also perform actions on packet arrival rather than timers



$$W = egin{cases} W + rac{1}{W} & ext{per ACK} \ W/2 & ext{per loss} \end{cases}$$

• What is the equation in terms of B = MSS * W?

$$B = \begin{cases} B + \frac{MSS^2}{B} & \text{per ACK} \\ B/2 & \text{per loss} \end{cases}$$

- Equivalently, TCP increases B by MSS per RTT
- What is the rate of TCP given that its window size is *B* (or *W*)?
- Since TCP sends a full window of pkts per RTT, its ideal rate can be written as:

$$r = \frac{B}{RTT + L/R} \approx \frac{B}{RTT} = \frac{MSS * W}{RTT}$$

TCP Reno Sender Congestion Control

Event	State	TCP Sender Action	Commentary
ACK receipt for previously unacked data	Slow Start (SS)	CongWin += MSS, If (CongWin >= ssthresh) { Set state to "Congestion Avoidance" }	Results in a doubling of CongWin every RTT
ACK receipt for previously unacked data	Congestion Avoidance (CA)	CongWin += MSS ² / CongWin	Additive increase, resulting in increase of CongWin by 1 MSS every RTT
Loss event detected by triple duplicate ACK	SS or CA	ssthresh = max(CongWin/2, MSS) CongWin = ssthresh Set state to "Congestion Avoidance"	Fast recovery, implementing multiplicative decrease
Timeout	SS or CA	ssthresh = max(CongWin/2, MSS) CongWin = MSS Set state to "Slow Start"	Enter slow start
Duplicate ACK	SS or CA	Increment duplicate ACK count for segment being acked	CongWin and Threshold not changed

TCP Reno Congestion Control

• Summary:

