CSCE 463/612 Networks and Distributed Processing Fall 2020

Transport Layer V

Dmitri Loguinov
Texas A&M University

October 15, 2020

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 control

3.7 TCP congestion control

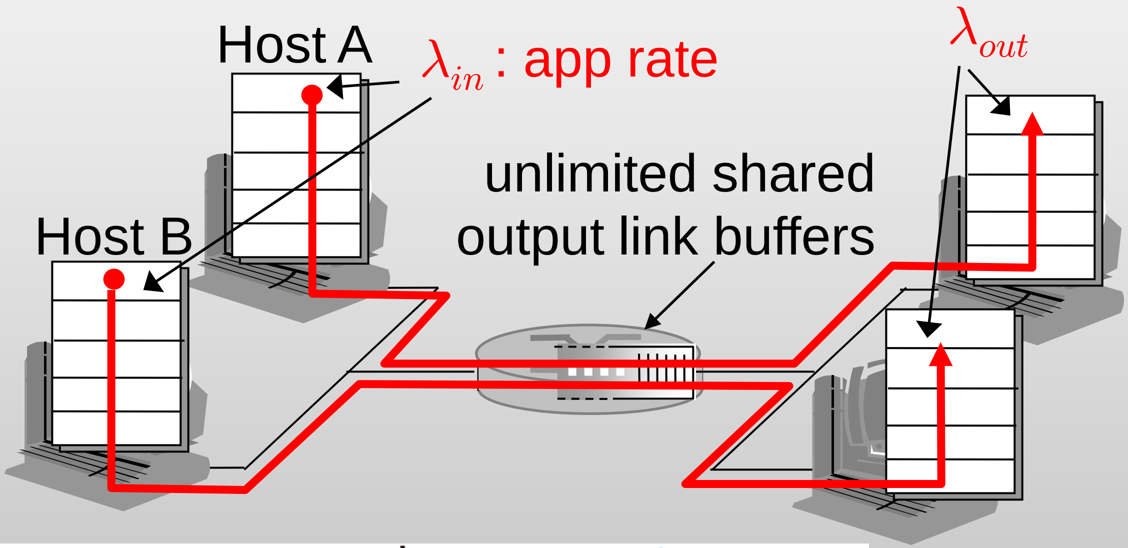
Principles of Congestion Control

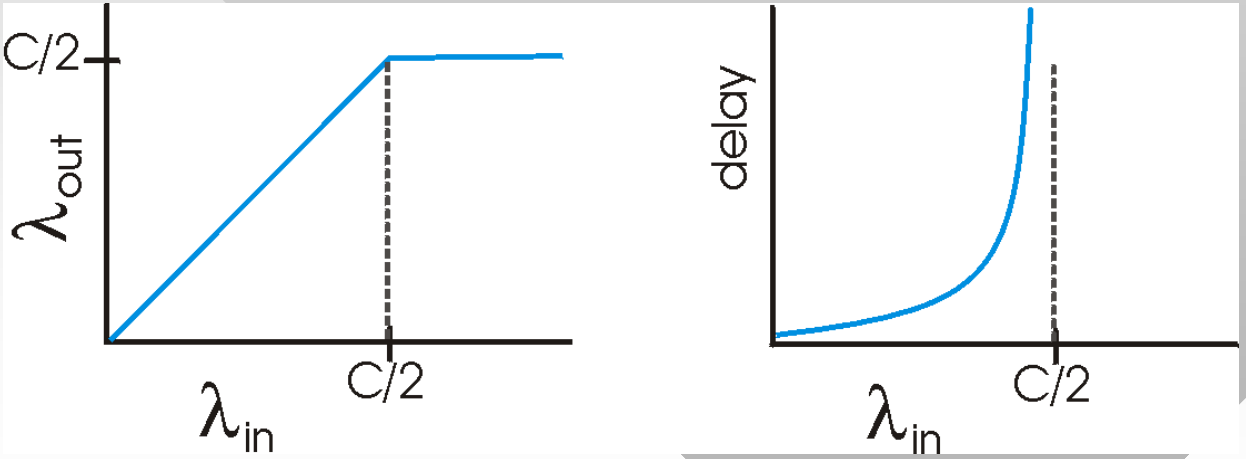
Congestion:

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

Causes/Costs of Congestion: Scenario 1

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

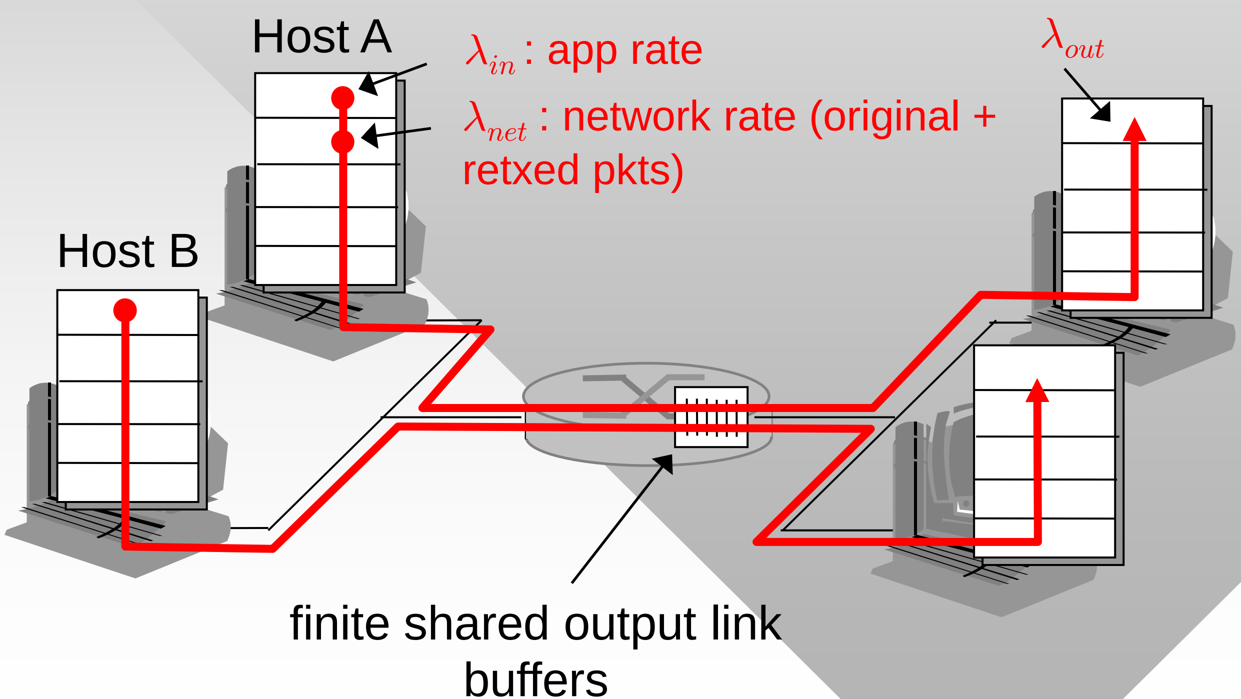




Cost 1: queuing delays in congested routers

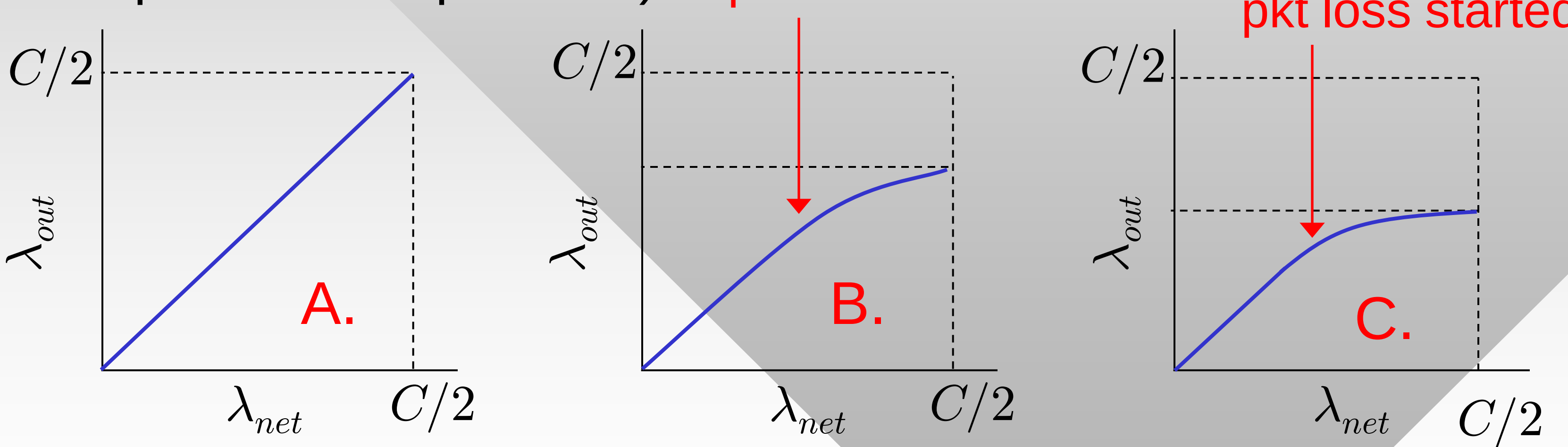
Causes/Costs of Congestion: Scenario 2

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



Causes/Costs of Congestion: Scenario 2

- We call $\lambda_{in} = \lambda_{out}$ *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)
  - <u>Case C</u>: same as B, but timer is not perfect (duplicate packets are possible)

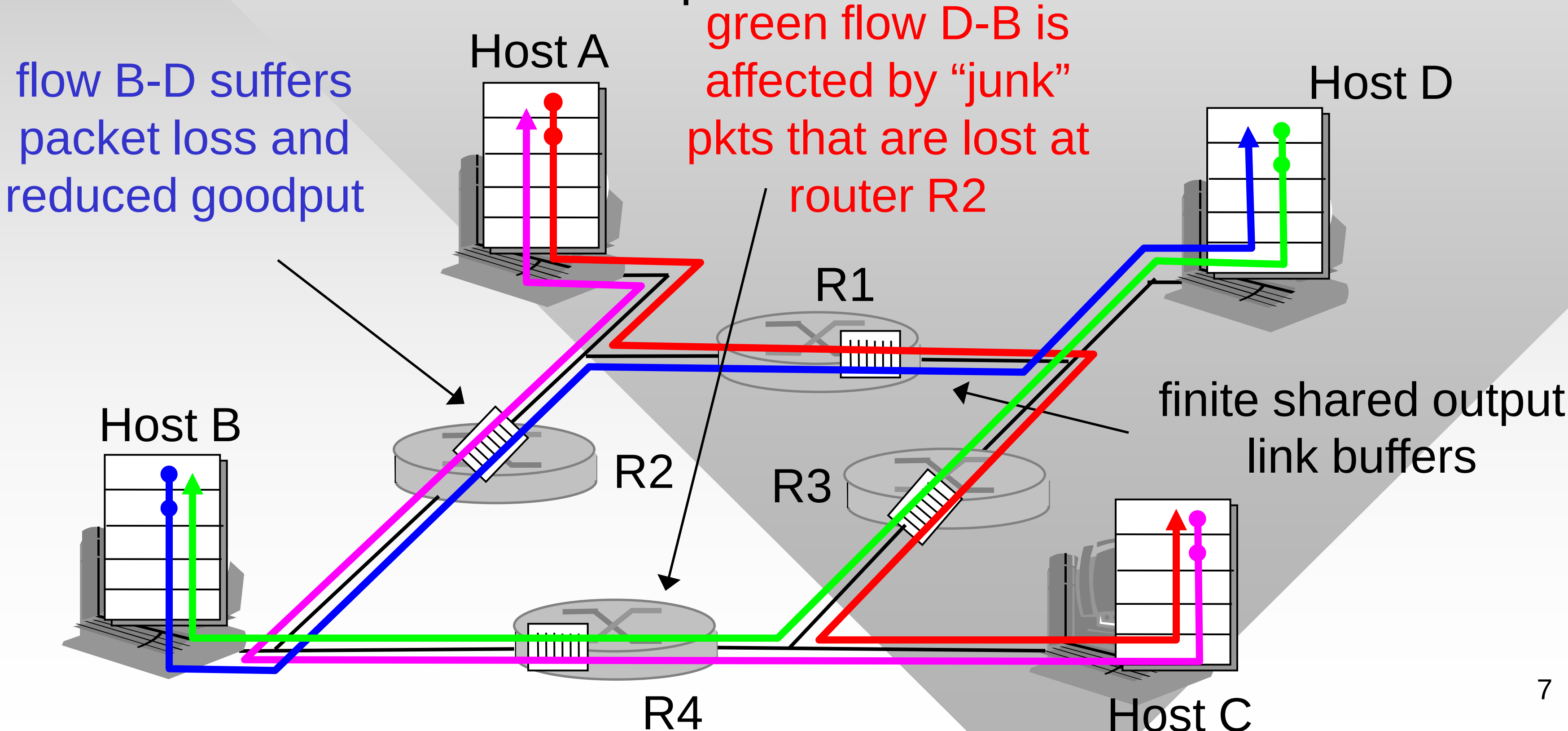


<u>Cost 2</u>: retransmission of lost packets and premature timeouts increase network load, reduce *flow's own* goodput

Causes/Costs of Congestion: Scenario 3

- Multihop case
  - Timeout/retransmit
  - R2 = 50 Mbps, R1 = R3 = R4 = 100 Mbps
  - Flow C-A: sends 90 Mbps

Cost 3: congestion causes goodput reduction for *other* flows



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)

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)

ATM = Asynchronous Transfer Mode

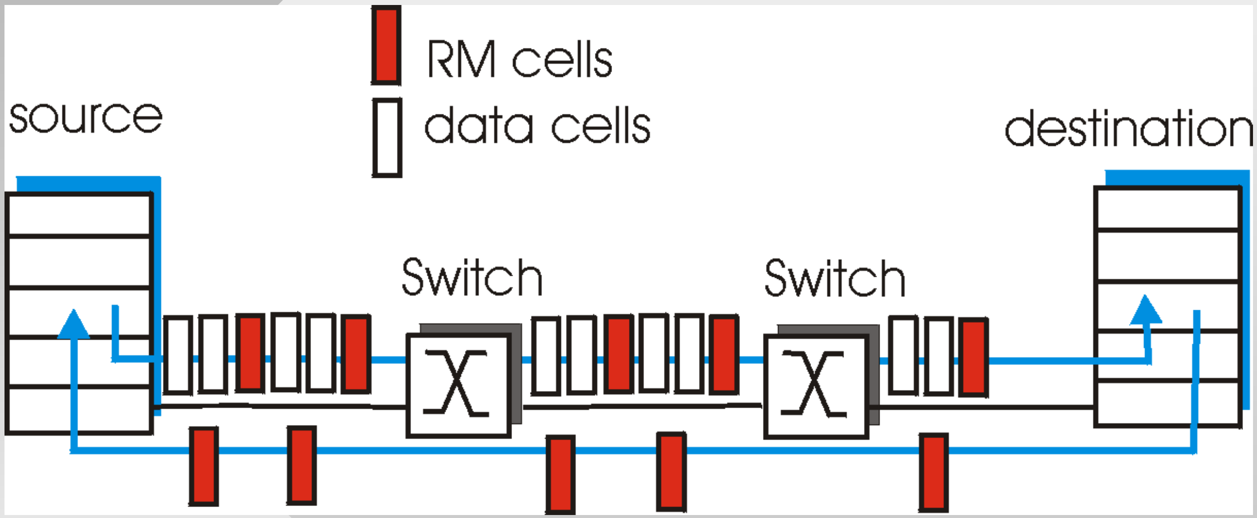
Case Study: ATM ABR Congestion Control

- 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?

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 control
3.7 TCP congestion control

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

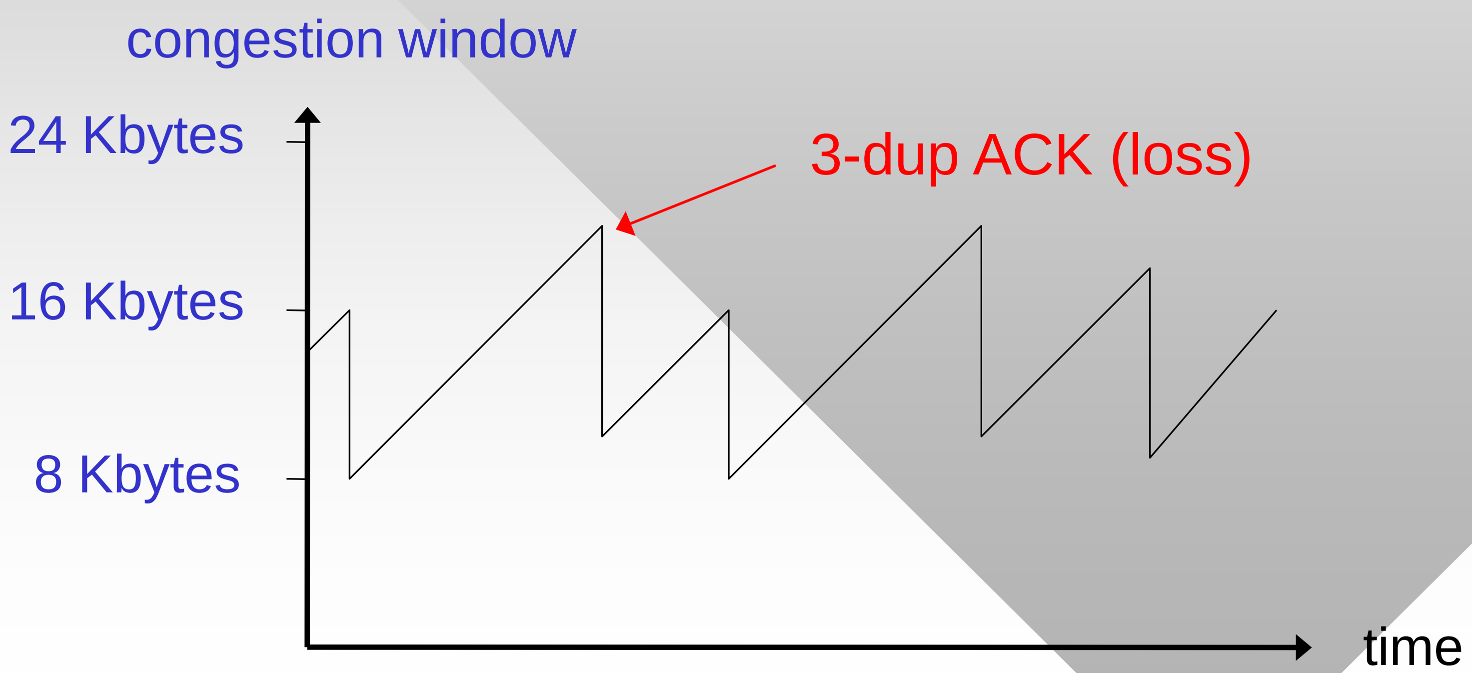
- How does sender perceive congestion?
  - Loss event = timeout or 3 duplicate acks
- TCP sender reduces rate (`CongWin`) after loss event
- Three mechanisms:
  - AIMD (congestion avoidance)
  - Slow start
  - Conservative after timeout events

TCP AIMD (Additive Increase, Multiplicative Decrease)

Additive increase: increase `CongWin` by 1 MSS every RTT in the absence of loss events: *probing*

Multiplicative decrease: cut `CongWin` in half after fast retransmit (3-dup ACKs)

Peaks are different: # of flows or RTT changes



TCP Equations

- To better understand TCP, we next examine its AIMD equations (congestion avoidance)
- Assume that W is the window size in pkts and $B =$ `CongWin` is the same in bytes ($B = MSS * W$)
- General form (loss detected through 3-dup ACK):

$$W = \begin{cases} W + \frac{1}{W} & \text{per ACK} \\ W/2 & \text{per loss} \end{cases}$$

- 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

TCP Equations

$$W = \begin{cases} W + \frac{1}{W} & \text{per ACK} \\ W/2 & \text{per loss} \end{cases}$$

- 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:

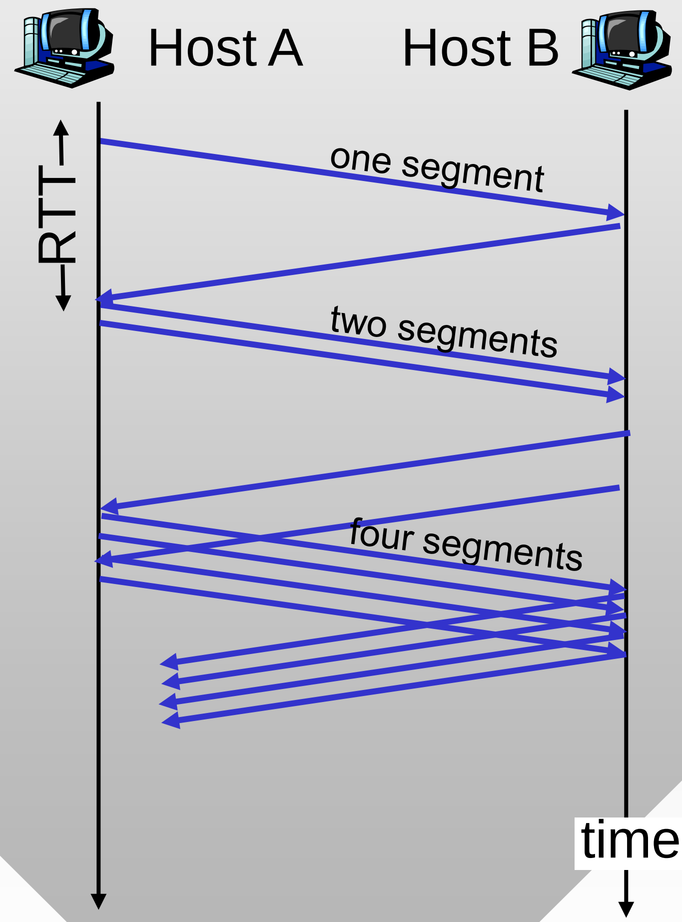
$$r = \frac{B}{RTT + L/R} \approx \frac{B}{RTT} = \frac{MSS * W}{RTT}$$

TCP 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)

- Slow start
  - Double `CongWin` every RTT
- Done by incrementing `CongWin` for every ACK received:
  - $W = W+1$ per ACK (or $B = B + MSS$)
- Summary: initial rate is slow but ramps up exponentially fast

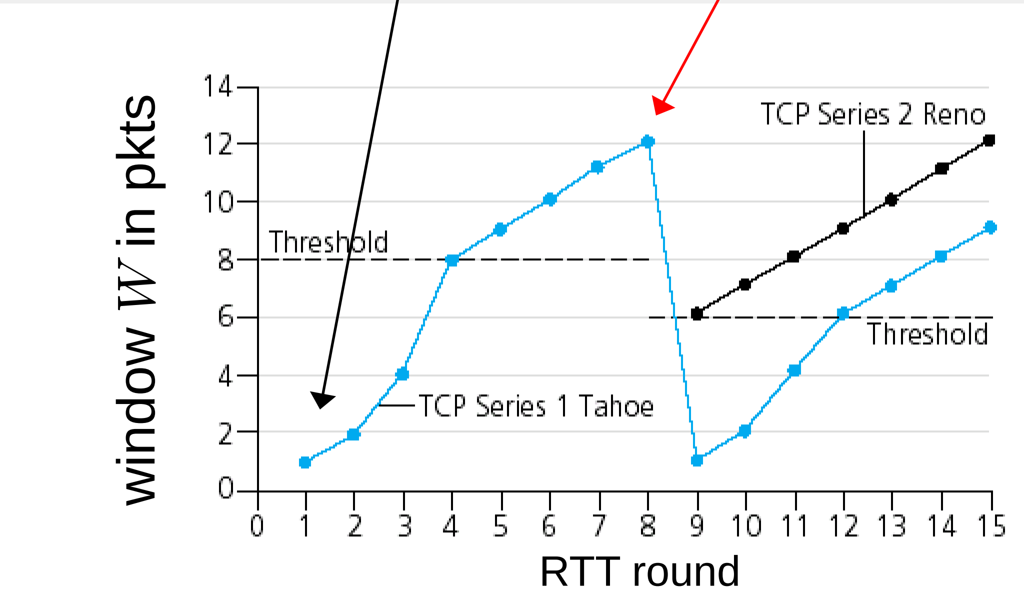


Refinement

- TCP Tahoe responds only to timeouts:
  - `Threshold = CongWin/2`
  - `CongWin` is set to 1 MSS
  - Slow start until `threshold` is reached; then move to AIMD congestion avoidance
- TCP Reno loss:
  - Timeout: same as Tahoe
  - 3 dup ACKs: `CongWin` is cut in half (original idea was called fast recovery, now part of AIMD)

loss detected via triple dup ACK

previous timeout



Fast Recovery Philosophy:

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

Timeout before 3-dup ACK is more alarming

Refinement (More)

- Initial slow start ends when either
  - Loss occurs
  - Initial threshold is reached
- Initial threshold is usually set to the receiver's advertised window

Implementation:

- Variable `ssthresh` is the "slow start threshold"
- At loss events, `ssthresh` is set to `CongWin/2`

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 Congestion Control

- Summary of TCP Reno: