COMP 3331/9331: Computer Networks and Applications

Week 6

Congestion Control (Transport Layer)

Reading Guide: Chapter 3, Sections: 3.6, 3.7

Transport Layer: Outline

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

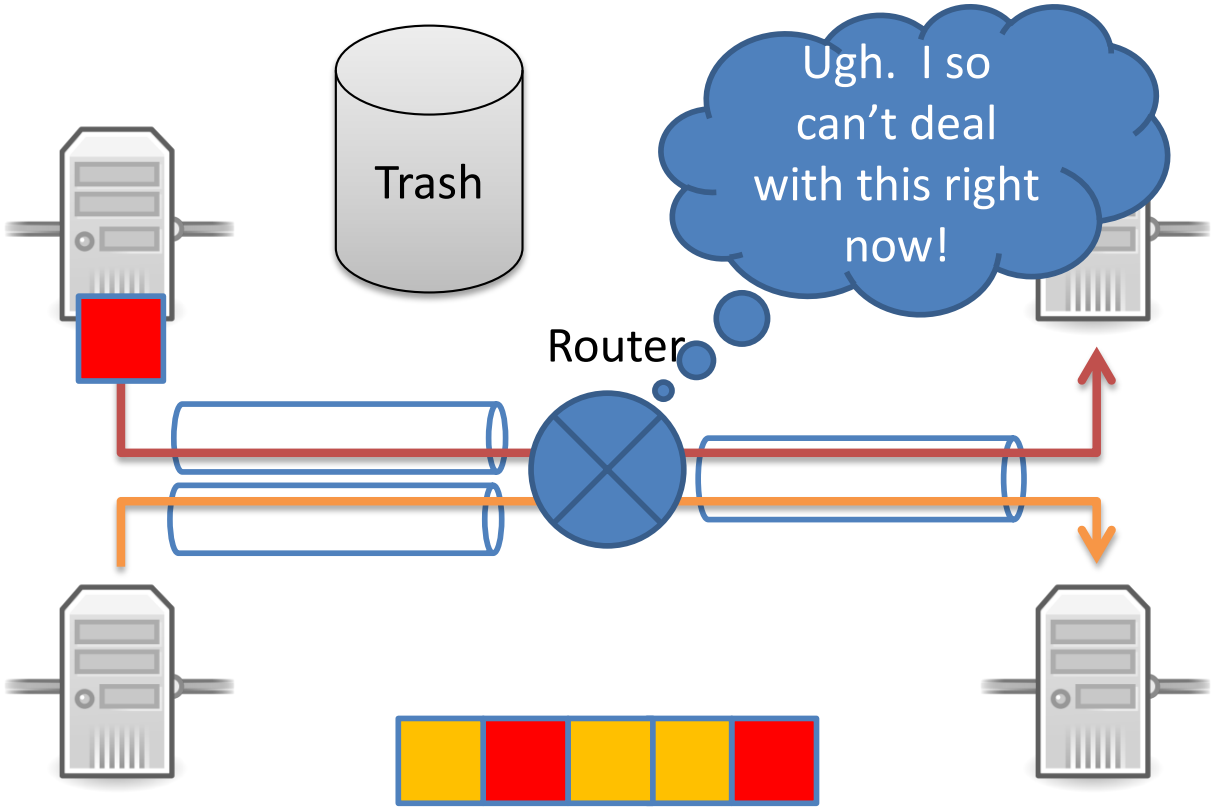
3.7 TCP congestion control

Principles of congestion control

congestion:

- informally: "too many sources sending too much data too fast for *network* to handle"
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Congestion



Incoming rate is faster than outgoing link can support.

Quiz: What's the worst that can happen?

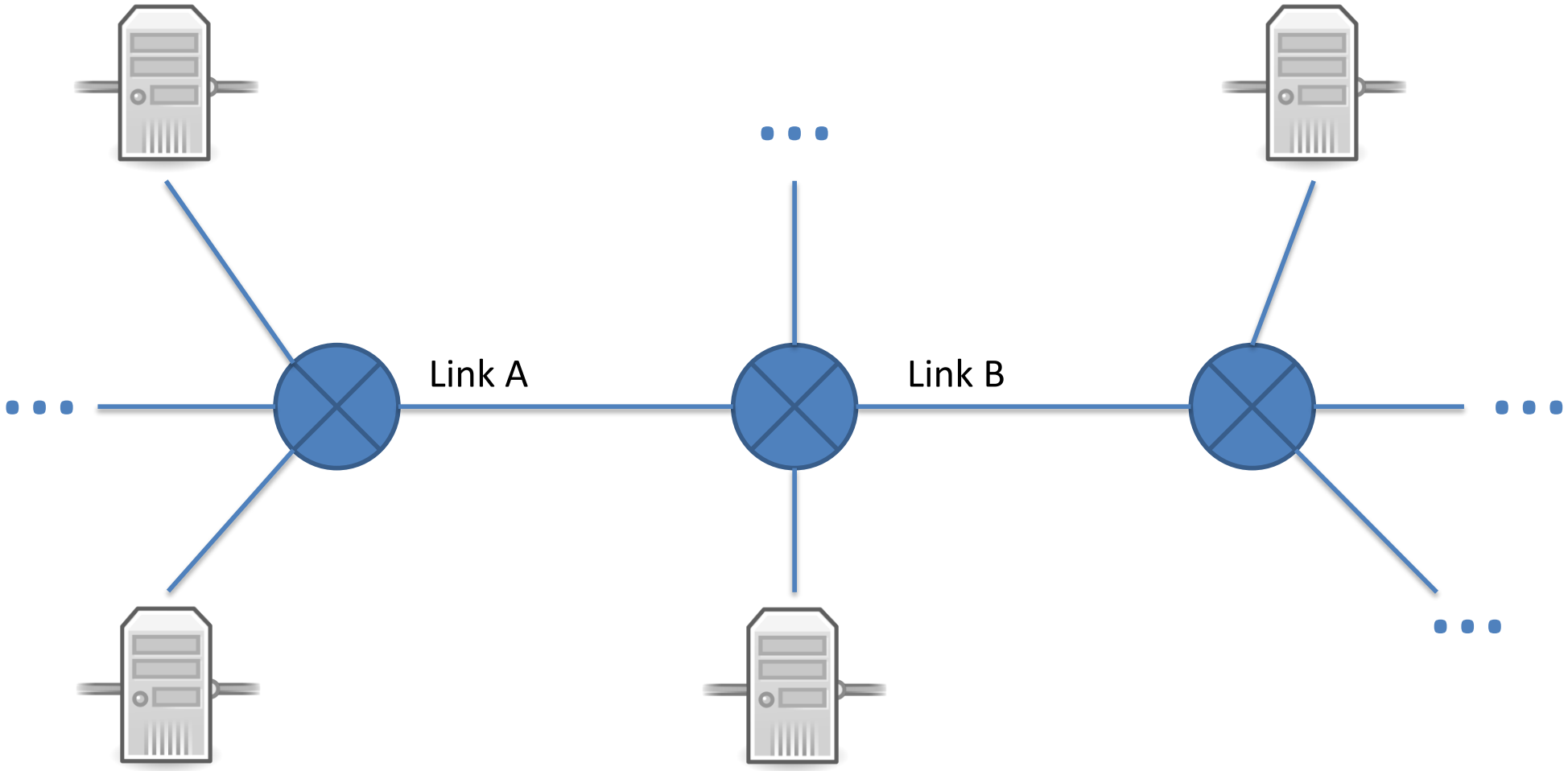
A: This is no problem. Senders just keep transmitting, and it'll all work out.

B: There will be retransmissions, but the network will still perform without much trouble.

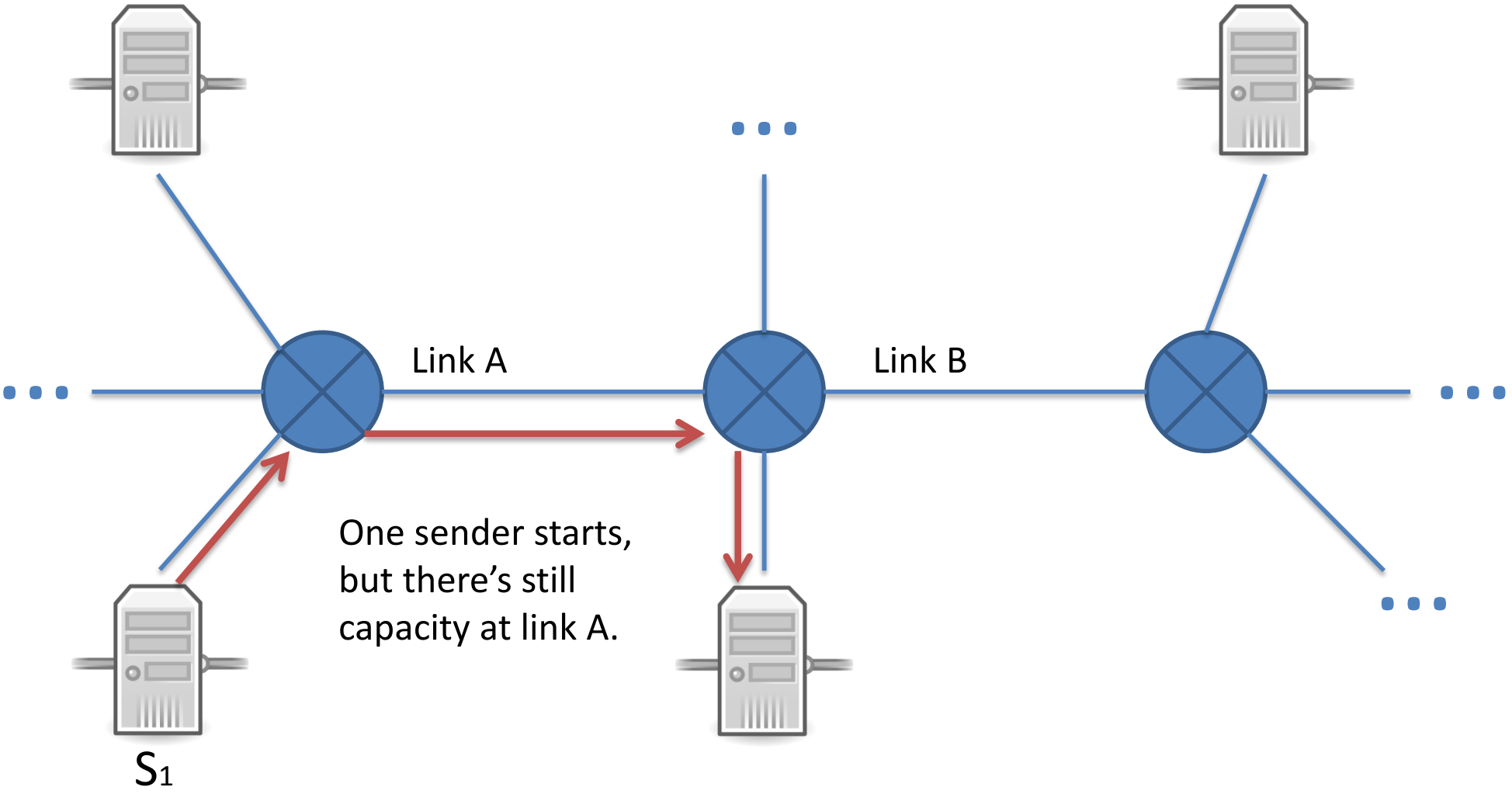
C: Retransmissions will become very frequent, causing a serious loss of efficiency

D: The network will become completely unusable

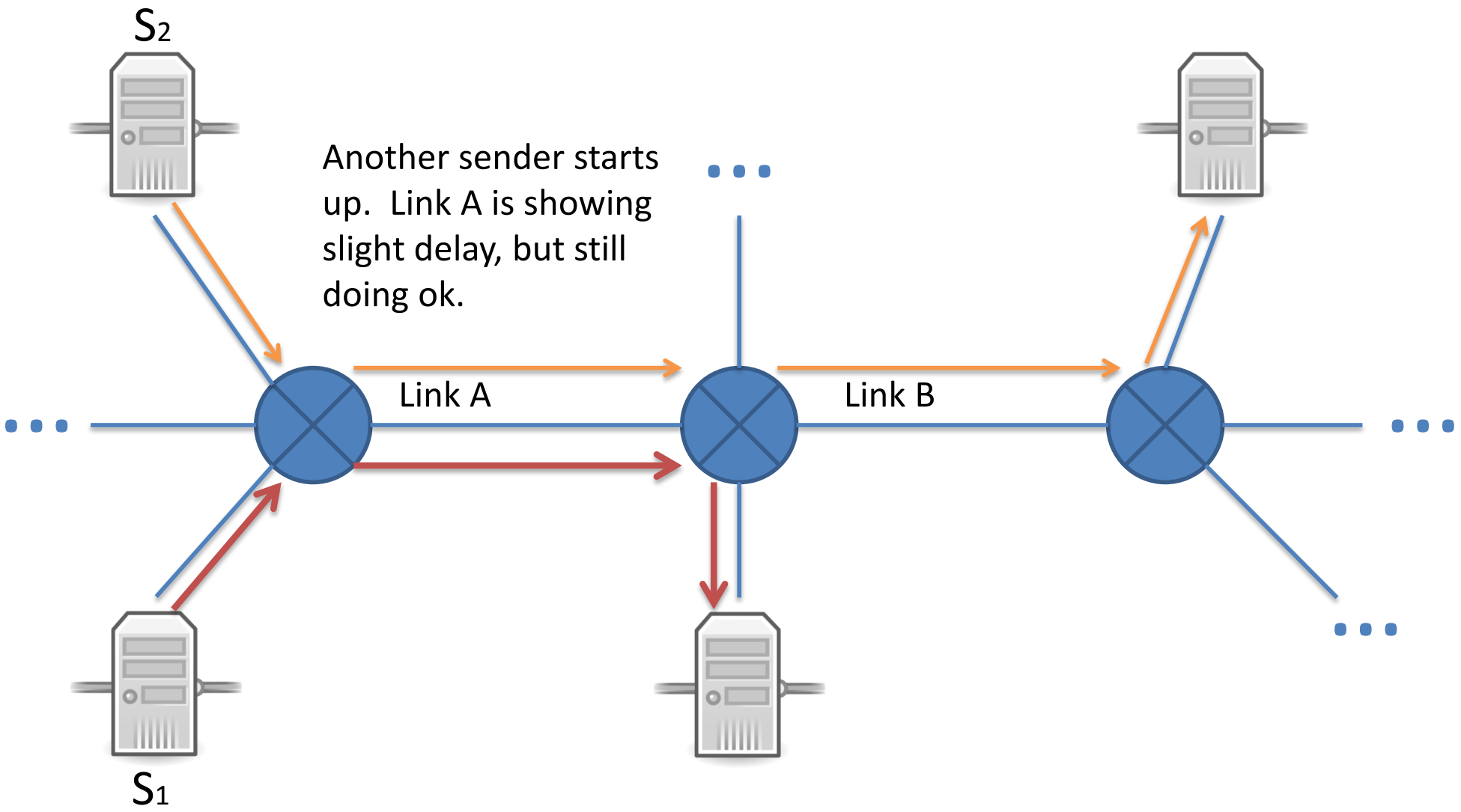
Congestion Collapse



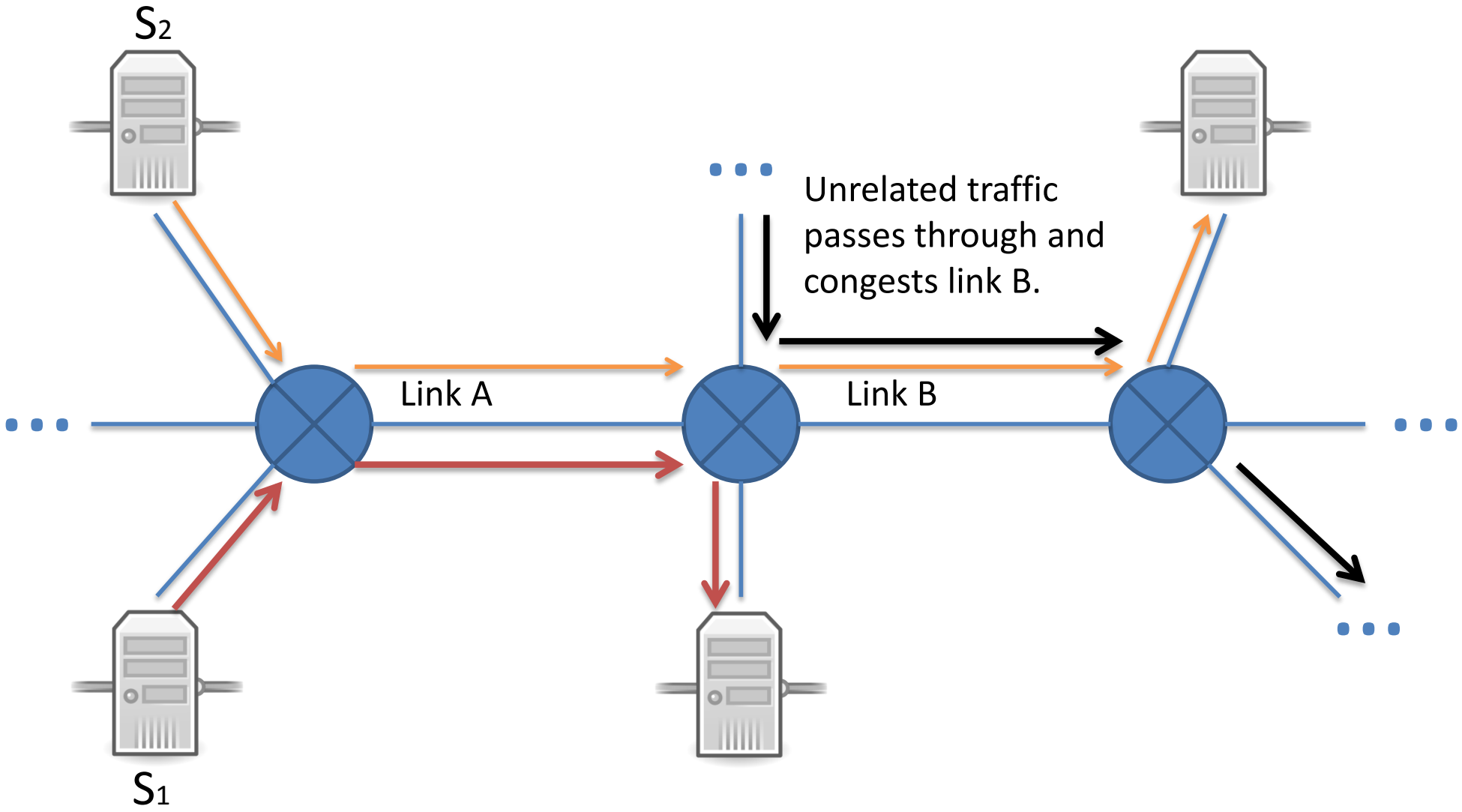
Congestion Collapse



Congestion Collapse

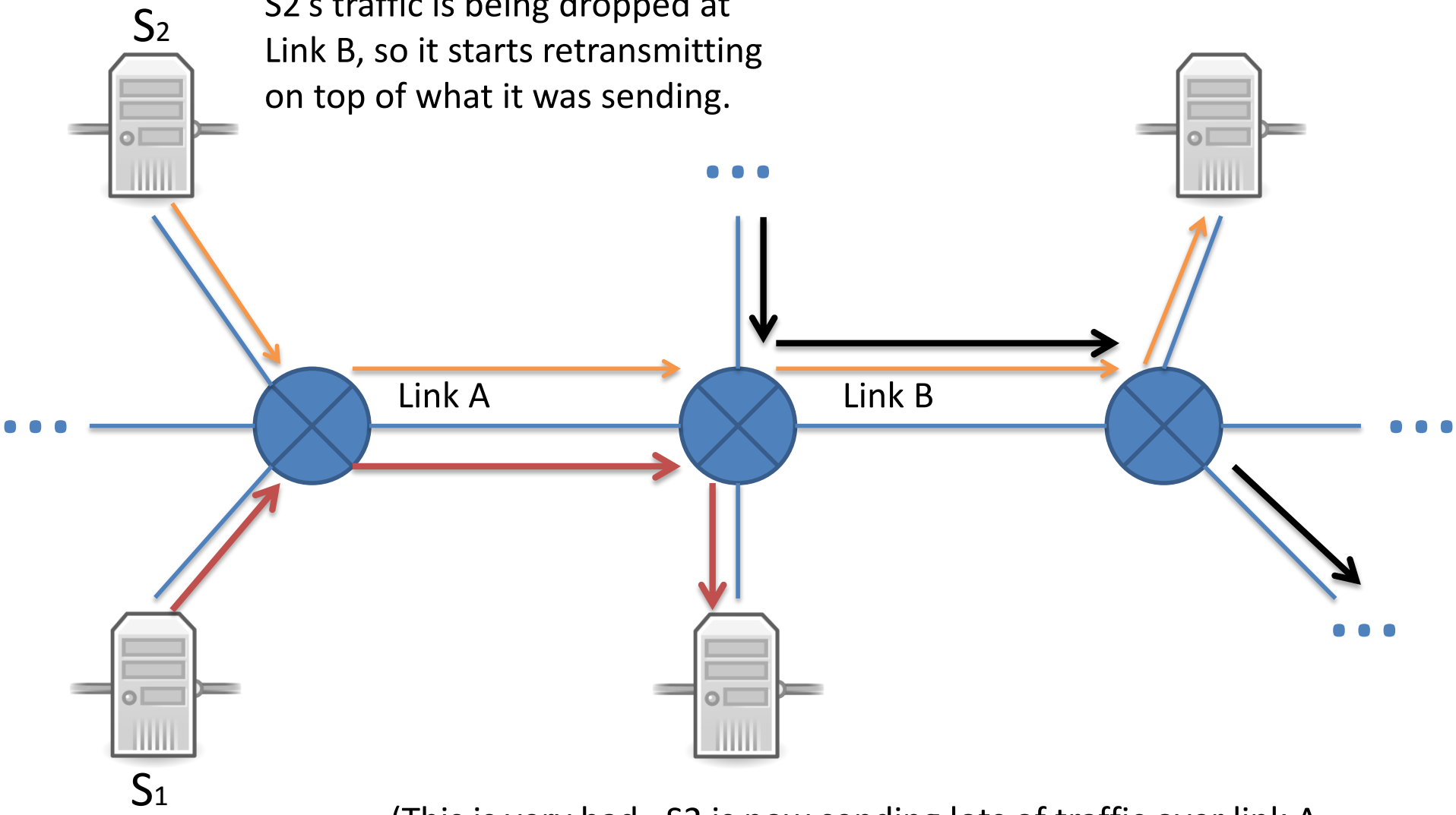


Congestion Collapse



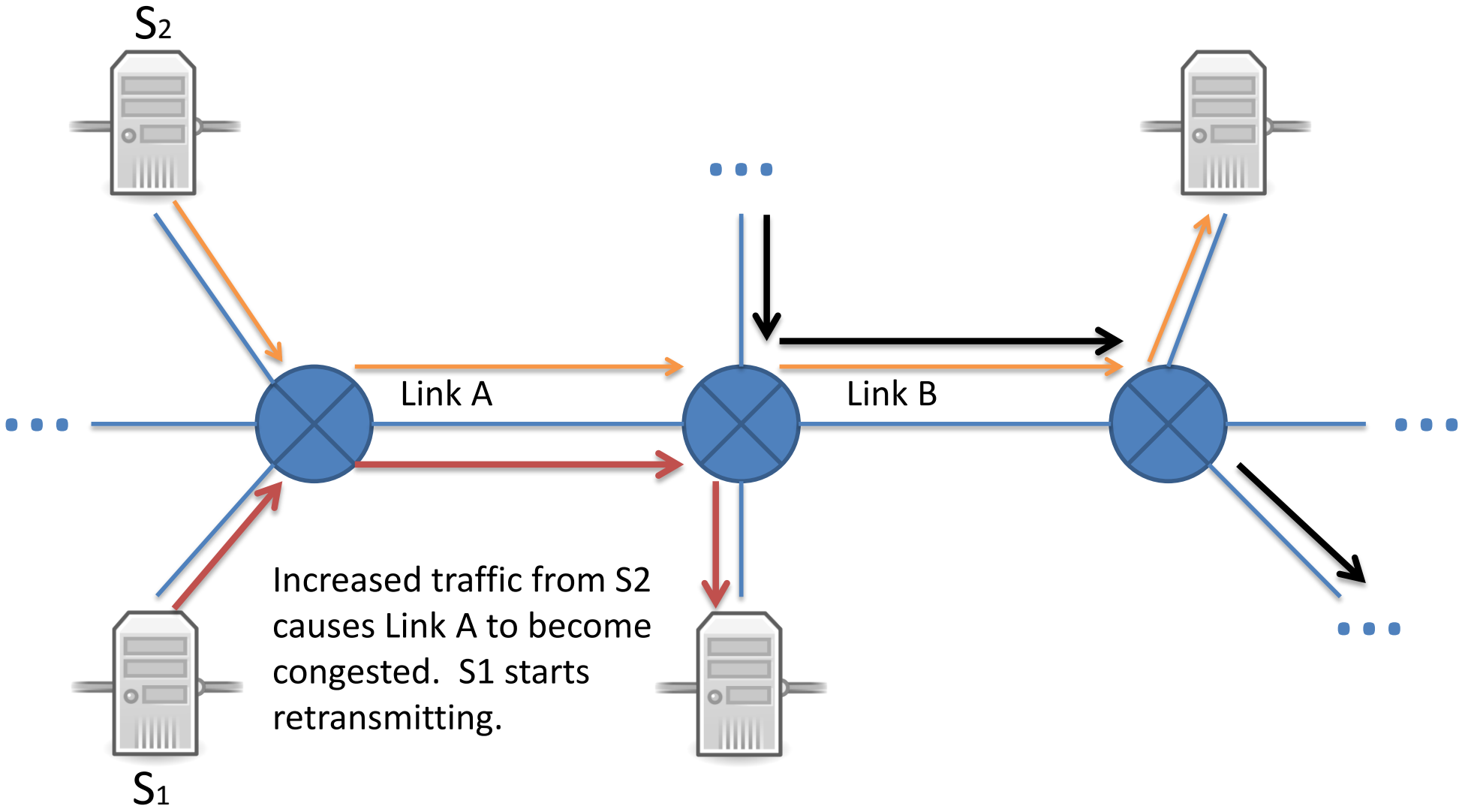
Congestion Collapse

S2's traffic is being dropped at Link B, so it starts retransmitting on top of what it was sending.

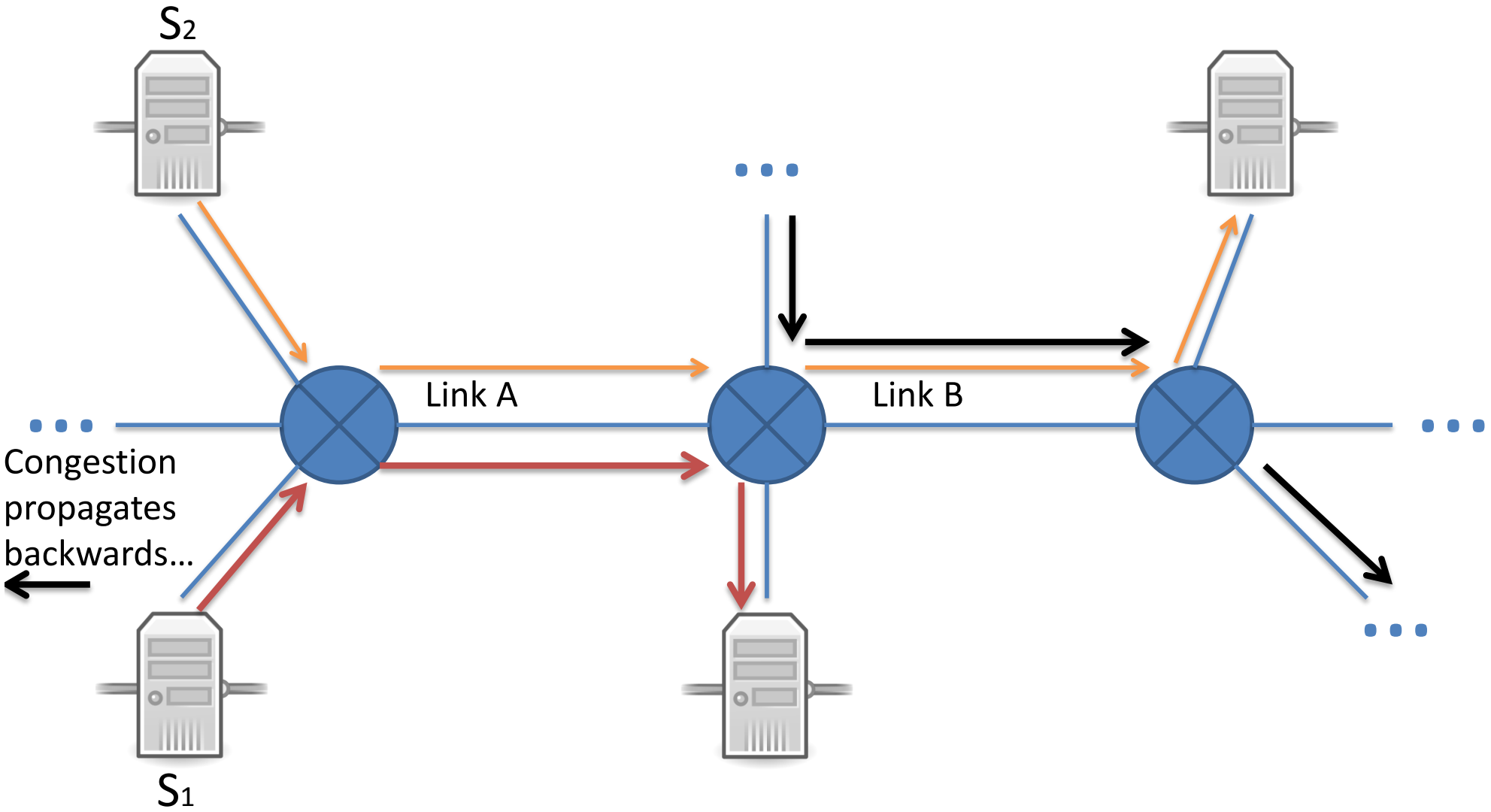


(This is very bad. S2 is now sending lots of traffic over link A that has no hope of crossing link B.)

Congestion Collapse



Congestion Collapse



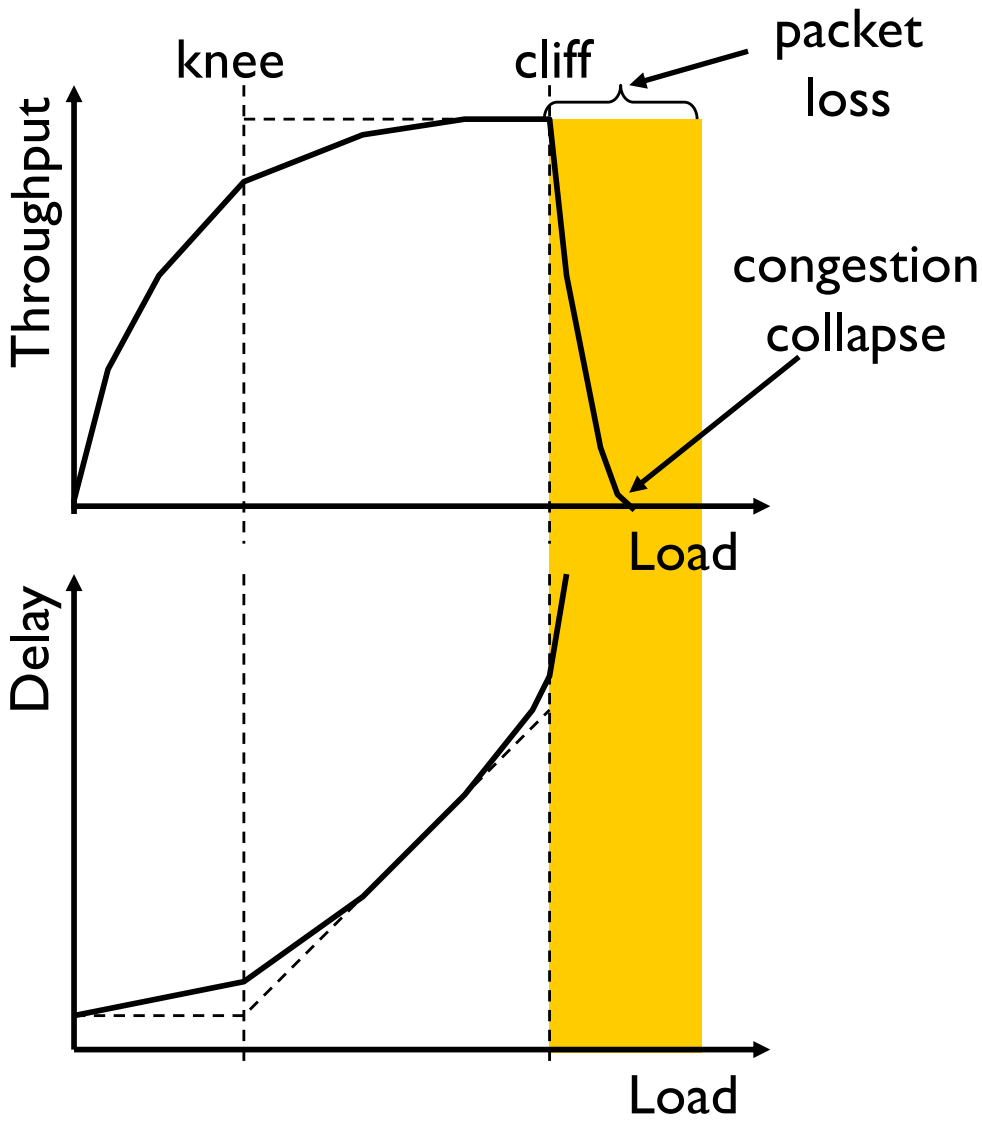
Without congestion control

congestion:

- Increases delivery latency
  - Variable delays
  - If delays > RTO, sender retransmits
- Increases loss rate
  - Dropped packets also retransmitted
- Increases retransmissions, many unnecessary
  - Wastes capacity of traffic that is never delivered
  - Increase in load results in decrease in useful work done
- Increases congestion, cycle continues …

Cost of Congestion

- Knee – point after which
  - Throughput increases slowly
  - Delay increases fast
- Cliff – point after which
  - Throughput starts to drop to zero (congestion collapse)
  - Delay approaches infinity



Congestion Collapse

This happened to the Internet (then NSFnet) in 1986

- Rate dropped from a *blazing* 32 Kbps to 40bps
- This happened on and off for *two years*
- In 1988, Van Jacobson published "Congestion Avoidance and Control"
- The fix: senders voluntarily limit sending rate

Approaches towards congestion control

two broad approaches towards congestion control:

end-end congestion control:

- no explicit feedback from network
- congestion inferred from end-system observed loss, delay
- approach taken by TCP

network-assisted congestion control:

- routers provide feedback to end systems
  - single bit indicating congestion (SNA, DECbit, TCP/IP ECN, ATM)
  - explicit rate for sender to send at

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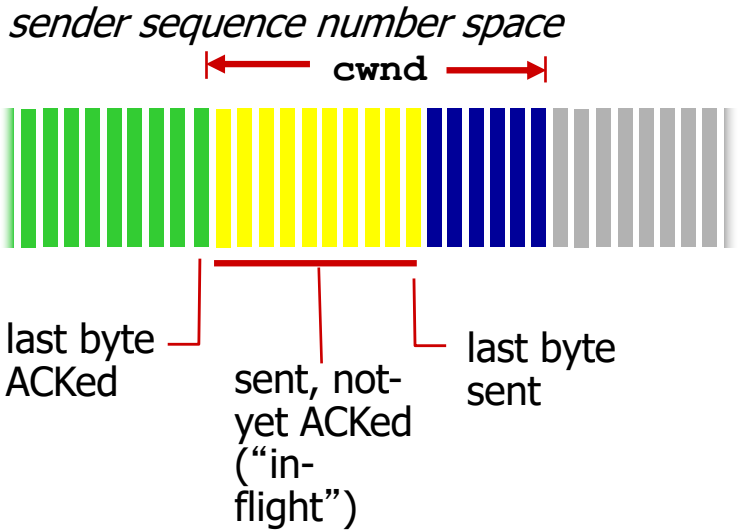
3.7 TCP congestion control

TCP's Approach in a Nutshell

- TCP connection has window
  - Controls number of packets in flight

- *TCP sending rate:*
  - *roughly:* send cwnd bytes, wait RTT for ACKS, then send more bytes

$$\text{rate} \approx \frac{\text{cwnd}}{\text{RTT}} \text{ bytes/sec}$$



- Vary window size to control sending rate

All These Windows…

- Congestion Window: CWND
  - How many bytes can be sent without overflowing routers
  - Computed by the sender using congestion control algorithm
- Flow control window: Advertised / Receive Window (RWND)
  - How many bytes can be sent without overflowing receiver's buffers
  - Determined by the receiver and reported to the sender
- Sender-side window = minimum{CWND, RWND}
  - Assume for this lecture that RWND >> CWND

CWND

- This lecture will talk about CWND in units of MSS
  - (Recall MSS: Maximum Segment Size, the amount of payload data in a TCP packet)
  - This is only for pedagogical purposes

- Keep in mind that real implementations maintain CWND in bytes

Two Basic Questions

- How does the sender detect congestion?

- How does the sender adjust its sending rate?

Quiz: What is a "congestion event"

A: A segment loss (but how can the sender be sure of this?)

B: Increased delays

C: Receiving duplicate acknowledgement (s)

D: A retransmission timeout firing

E: Some subset of A, B, C & D (what is the subset?)

Quiz: How should we set CWND?

A: We should keep raising it until a "congestion event" then back off slightly until we notice no more events.

B: We should raise it until a "congestion event", then go back to 1 and start raising it again

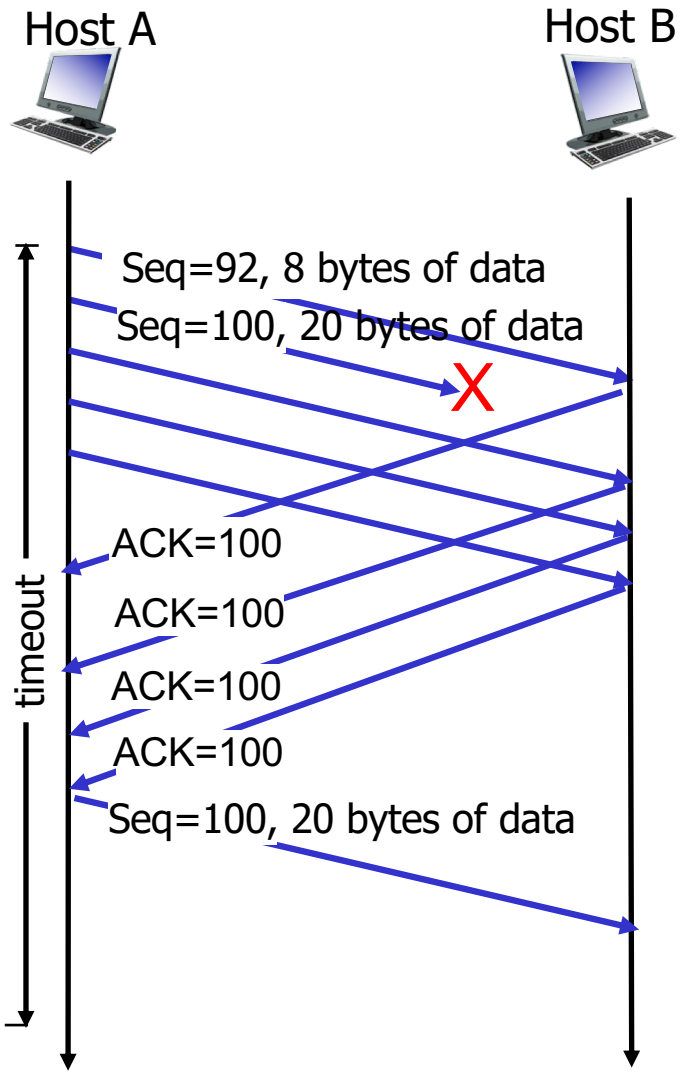
C: We should raise it until a "congestion event", then go back to median value and start raising it again.

D: We should sent as fast as possible at all times.

Not All Losses the Same

- Duplicate ACKs: isolated loss
  - dup ACKs indicate network capable of delivering some segments
- Timeout: much more serious
  - Not enough dup ACKs
  - Must have suffered several losses
- Will adjust rate differently for each case

RECAP: TCP fast retransmit



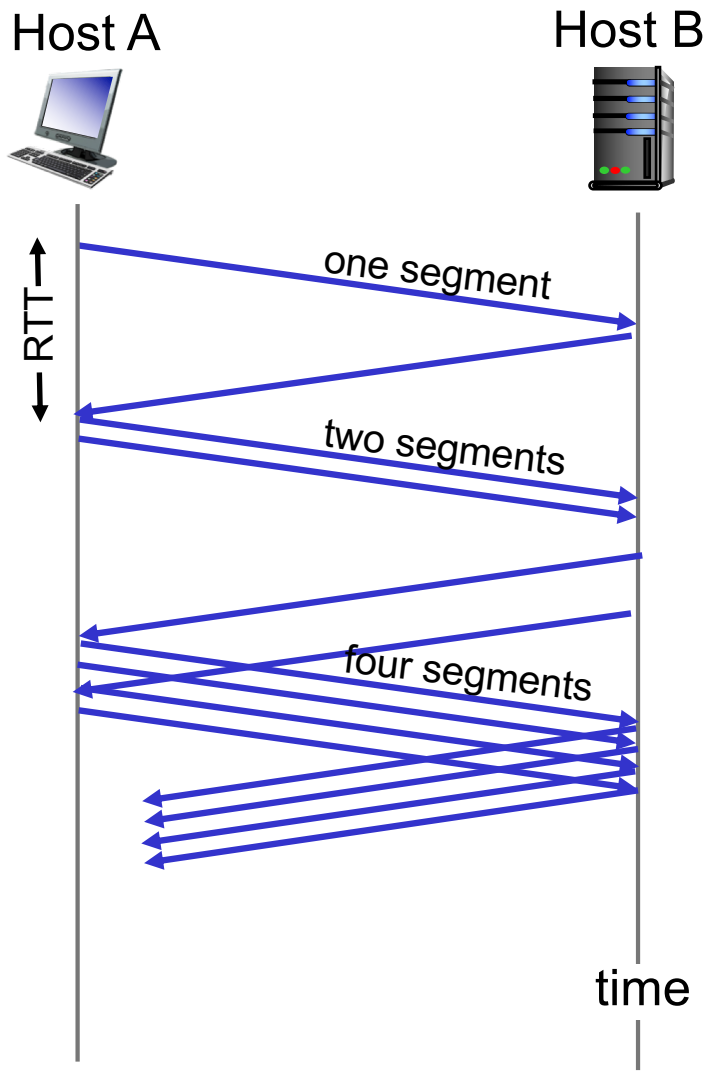
fast retransmit after sender receipt of triple duplicate ACK

Rate Adjustment

- Basic structure:
  - Upon receipt of ACK (of new data): increase rate
  - Upon detection of loss: decrease rate
- How we increase/decrease the rate depends on the phase of congestion control we're in:
  - Discovering available bottleneck bandwidth *vs.*
  - Adjusting to bandwidth variations

TCP Slow Start (Bandwidth discovery)

- when connection begins, increase rate exponentially until first loss event:
  - initially **`cwnd`** = 1 MSS
  - double **`cwnd`** every RTT (full ACKs)
  - Simpler implementation achieved by incrementing **`cwnd`** for every ACK received
    - cwnd += 1 for each ACK
- *summary:* initial rate is slow but ramps up exponentially fast



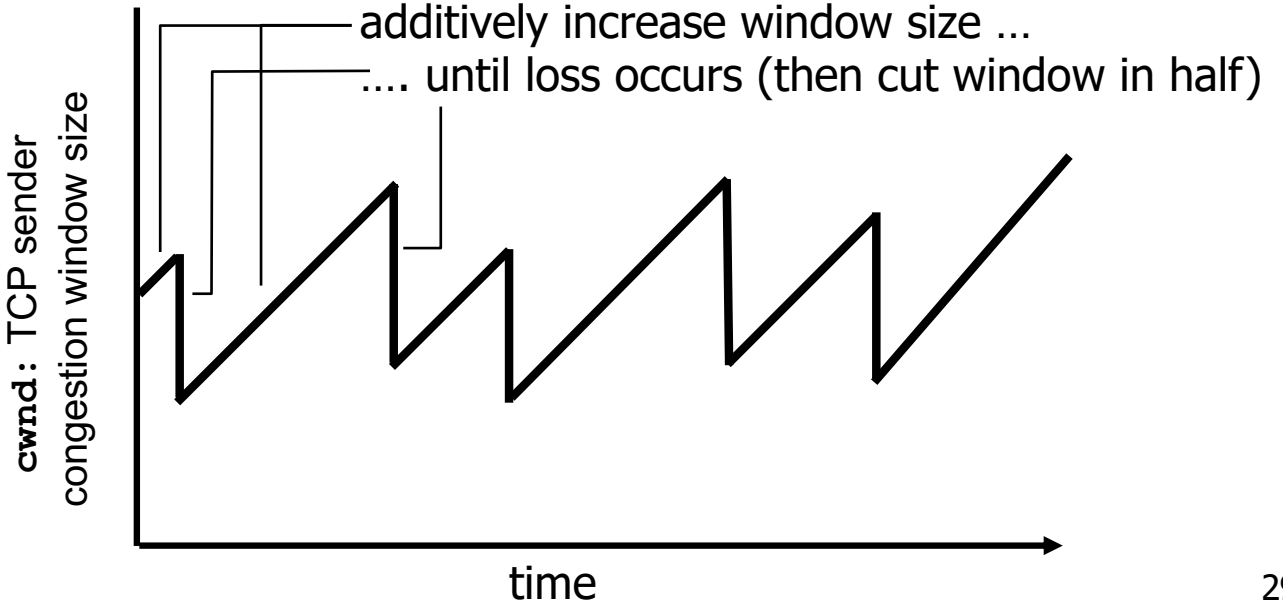
Adjusting to Varying Bandwidth

- Slow start gave an estimate of available bandwidth
- Now, want to track variations in this available bandwidth, oscillating around its current value
  - Repeated probing (rate increase) and backoff (rate decrease)
  - Known as Congestion Avoidance (CA)
- TCP uses: "Additive Increase Multiplicative Decrease" (AIMD)
  - We'll see why shortly…

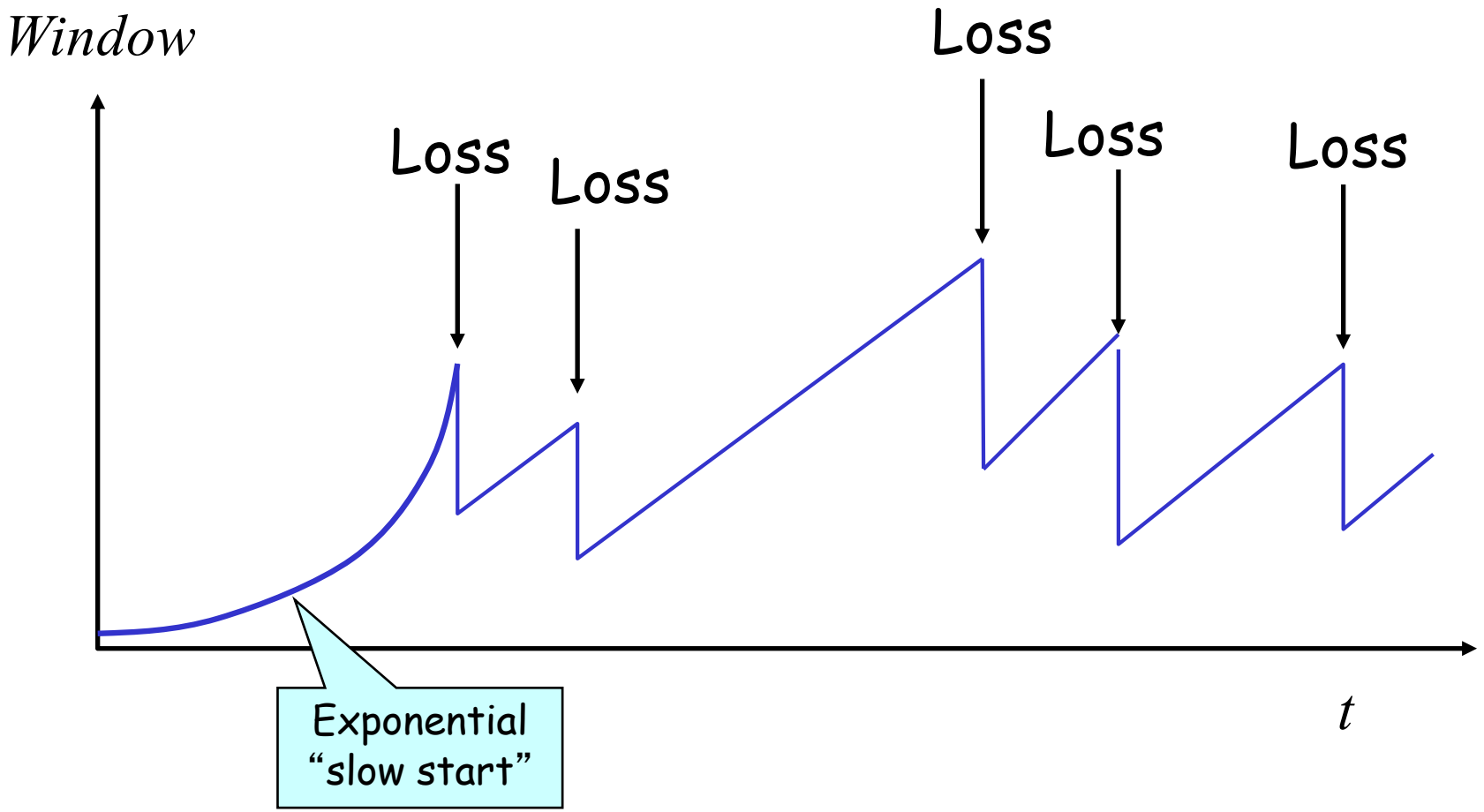
AIMD

- *approach:* sender increases transmission rate (window size), probing for usable bandwidth, until another congestion event occurs
  - *additive increase:* increase **`cwnd`** by 1 MSS every RTT until loss detected
    - For each successful RTT (all ACKS), **`cwnd = cwnd +1`**
    - Simple implementation: for each ACK, **`cwnd = cwnd + 1/cwnd`**
  - *multiplicative decrease:* cut **`cwnd`** in half after loss

AIMD saw tooth behavior: probing for bandwidth



Leads to the TCP "Sawtooth"



Slow-Start vs. AIMD

- When does a sender stop Slow-Start and start Additive Increase?

- Introduce a "slow start threshold" (ssthresh)
  - Initialized to a large value

- Convert to AI when cwnd = ssthresh, sender switches from slow-start to AIMD-style increase
  - On timeout, ssthresh = CWND/2

Implementation

- State at sender
  - CWND (initialized to a small constant)
  - ssthresh (initialized to a large constant)
  - [Also dupACKcount and timer, as before]

- Events
  - ACK (new data)
  - dupACK (duplicate ACK for old data)
  - Timeout

Event: ACK (new data)

- If CWND < ssthresh
  - CWND += 1

- *2*MSS packets per ACK*
- *Hence after one RTT (All ACKs with no drops):*
  CWND = 2xCWND

Event: ACK (new data)

- If CWND < ssthresh
  - CWND += 1

Slow start phase

- Else
  - CWND = CWND + 1/CWND

"Congestion Avoidance" phase (additive increase)

- *Hence after one RTT (All ACKs with no drops):*
  CWND = CWND + 1

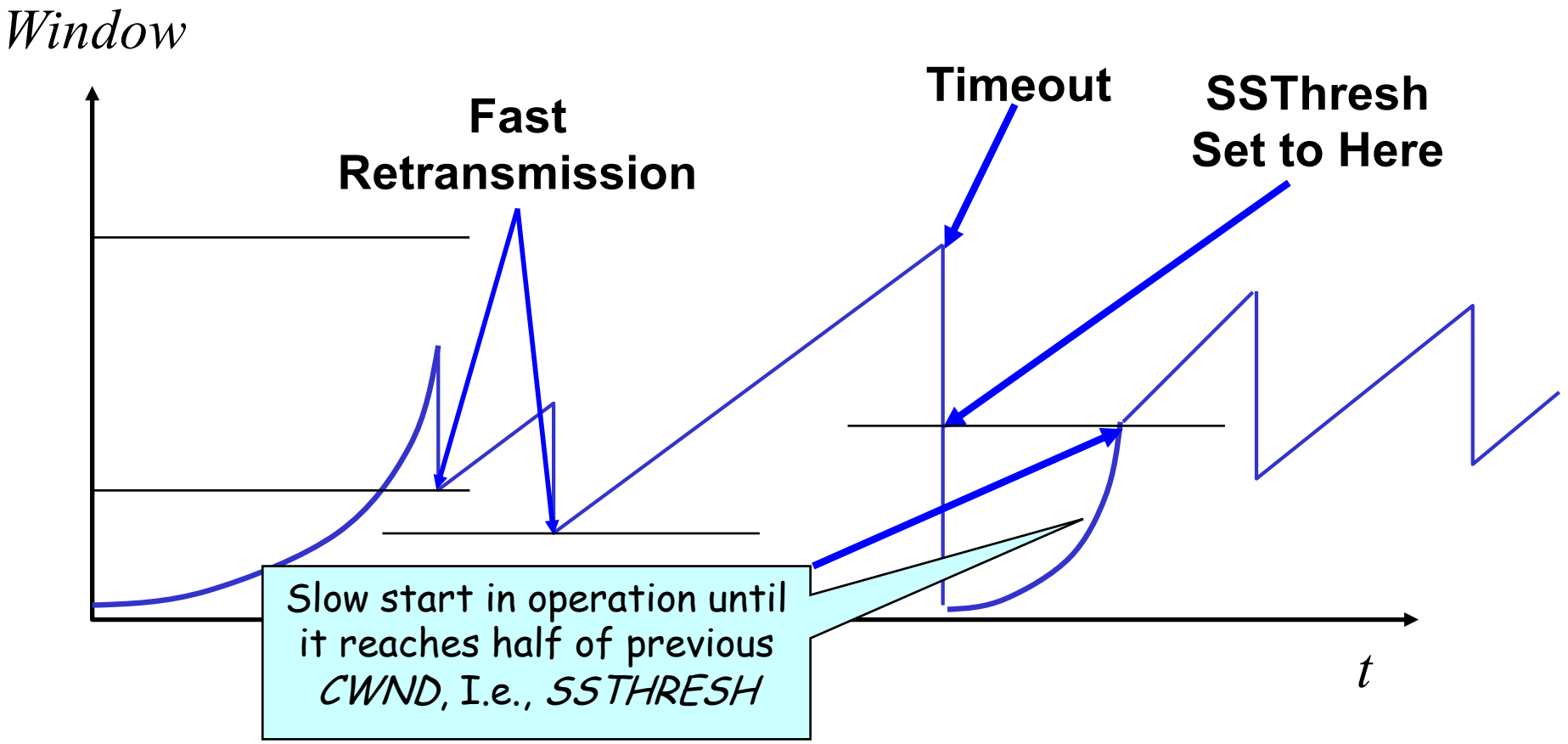
Event: dupACK

- dupACKcount ++
- If dupACKcount = 3 /* fast retransmit */
  - ssthresh = CWND/2
  - CWND = CWND/2

Event: TimeOut

- On Timeout
  - ssthresh ← CWND/2
  - CWND ← 1

Example



Slow-start restart: Go back to CWND = 1 MSS, but take advantage of knowing the previous value of CWND

One Final Phase: Fast Recovery

- The problem: Fast retransmit slow in recovering from an isolated loss

Example (window in units of MSS, not bytes)

- Consider a TCP connection with:
  - CWND=10 packets (of size MSS, which is 100 bytes)
  - Last ACK was for byte # 101
    - i.e., receiver expecting next packet to have seq. no. 101

- 10 packets [101, 201, 301,…, 1001] are in flight
  - Packet 101 is dropped
  - What ACKs do they generate?
  - And how does the sender respond?

Timeline

- ACK 101 (due to 201) cwnd=10 dupACK#1 (no xmit)
- ACK 101 (due to 301) cwnd=10 dupACK#2 (no xmit)
- ACK 101 (due to 401) cwnd=10 dupACK#3 (no xmit)
- RETRANSMIT 101 ssthresh=5 cwnd= 5
- ACK 101 (due to 501) cwnd=5 + 1/5 (no xmit)
- ACK 101 (due to 601) cwnd=5 + 2/5 (no xmit)
- ACK 101 (due to 701) cwnd=5 + 3/5 (no xmit)
- ACK 101 (due to 801) cwnd=5 + 4/5 (no xmit)
- ACK 101 (due to 901) cwnd=5 + 5/5 (no xmit)
- ACK 101 (due to 1001) cwnd=6 + 1/5 (no xmit)
- ACK 1101 (due to 101) ← only now can we transmit new packets
- Plus no packets in flight so ACK "clocking" (to increase CWND) stalls for another RTT

Solution: Fast Recovery

Idea: Grant the sender temporary "credit" for each dupACK so as to keep packets in flight

- If dupACKcount = 3
  - ssthresh = cwnd/2
  - cwnd = ssthresh + 3
- While in fast recovery
  - cwnd = cwnd + 1 for each additional duplicate ACK
- Exit fast recovery after receiving new ACK
  - set cwnd = ssthresh

Example

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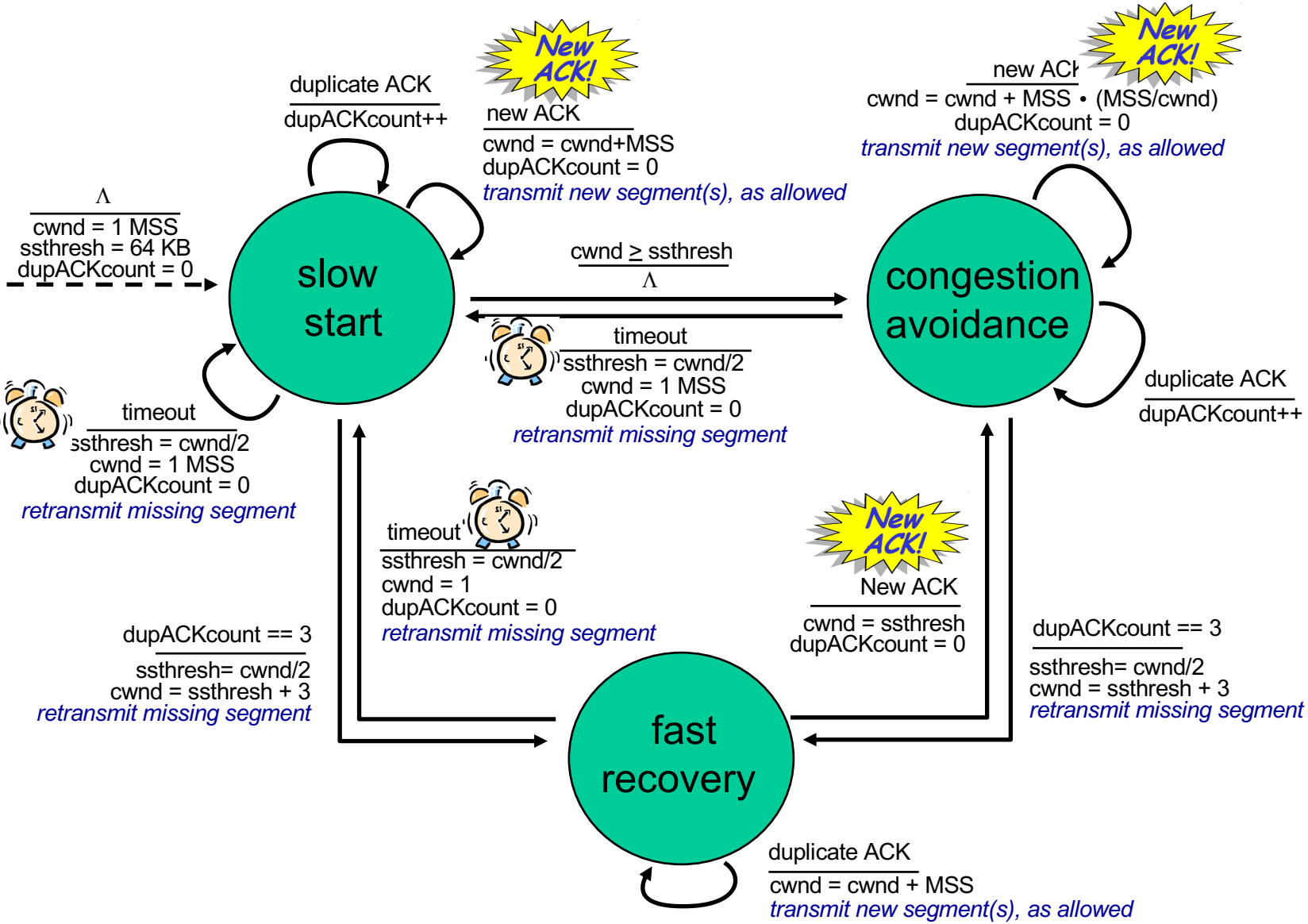
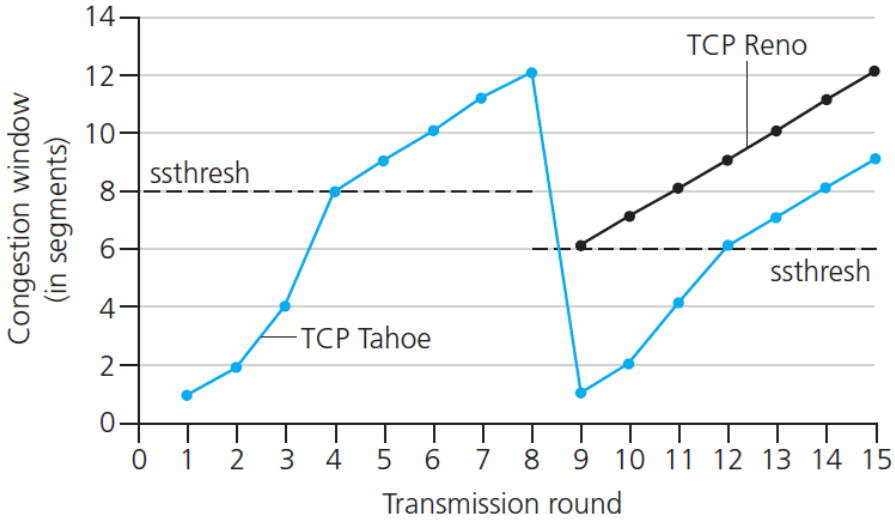
Timeline

- ACK 101 (due to 201) cwnd=10 dup#1
- ACK 101 (due to 301) cwnd=10 dup#2
- ACK 101 (due to 401) cwnd=10 dup#3
- REXMIT 101 ssthresh=5 cwnd= 8 (5+3)
- ACK 101 (due to 501) cwnd= 9 (no xmit)
- ACK 101 (due to 601) cwnd=10 (no xmit)
- ACK 101 (due to 701) cwnd=11 (xmit 1101)
- ACK 101 (due to 801) cwnd=12 (xmit 1201)
- ACK 101 (due to 901) cwnd=13 (xmit 1301)
- ACK 101 (due to 1001) cwnd=14 (xmit 1401)
- ACK 1101 (due to 101) cwnd = 5 (xmit 1501) ← exiting fast recovery
- Packets 1101-1401 already in flight
- ACK 1201 (due to 1101) cwnd = 5 + 1/5 ← back in congestion avoidance

Summary: TCP Congestion Control



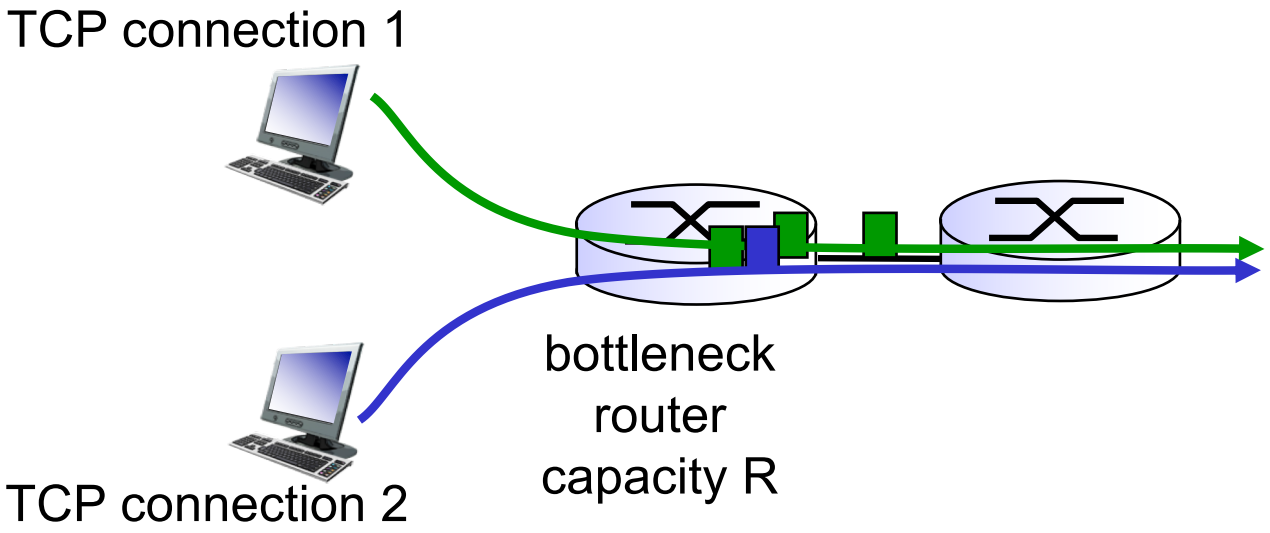
TCP Flavours



- TCP-Tahoe
  - cwnd =1 on triple dup ACK & timeout
- TCP-Reno
  - cwnd =1 on timeout
  - cwnd = cwnd/2 on triple dup ACK
- TCP-newReno
  - TCP-Reno + improved fast recovery
- TCP-SACK (NOT COVERED IN THE COURSE)
  - incorporates selective acknowledgements

TCP Fairness

fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



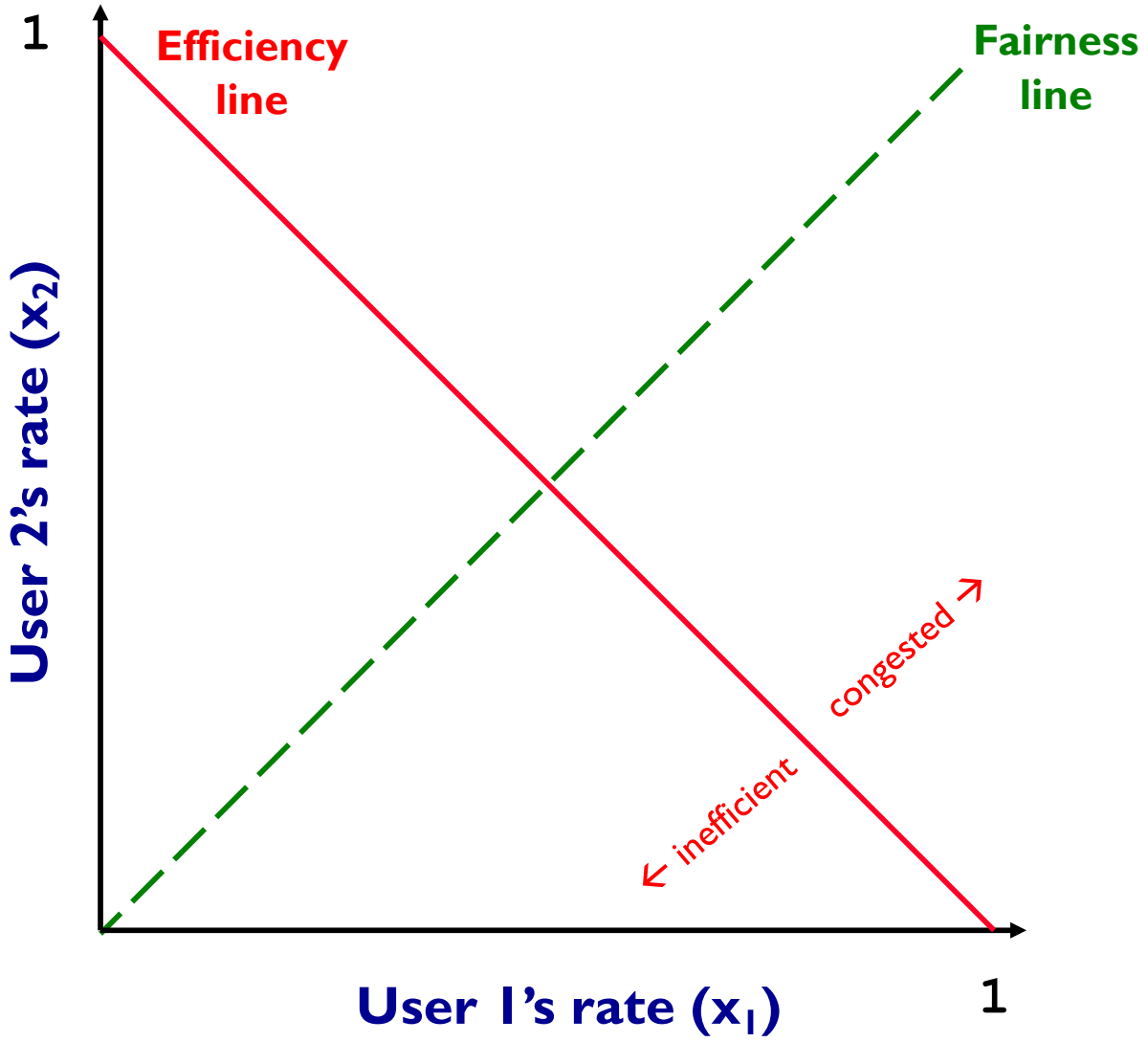
Why AIMD?

- Some rate adjustment options: Every RTT, we can
  - Multiplicative increase or decrease: CWND→ a*CWND
  - Additive increase or decrease: CWND→ CWND + b

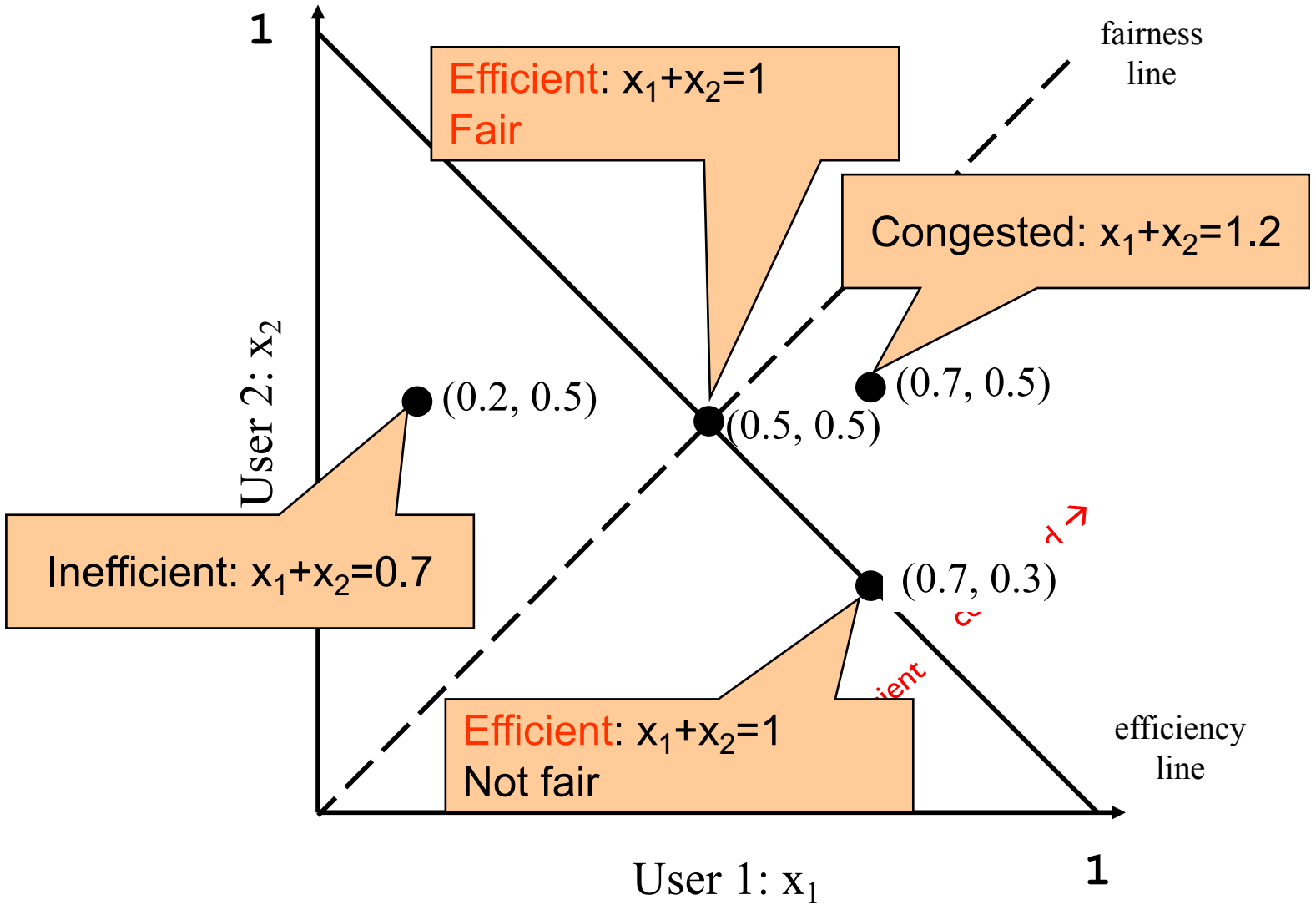
- Four alternatives:
  - AIAD: gentle increase, gentle decrease
  - AIMD: gentle increase, drastic decrease
  - MIAD: drastic increase, gentle decrease
  - MIMD: drastic increase and decrease

Simple Model of Congestion Control

- Two users
  - rates x_1 and x_2
- Congestion when $x_1+x_2 > 1$
- Unused capacity when $x_1+x_2 < 1$
- Fair when $x_1 = x_2$

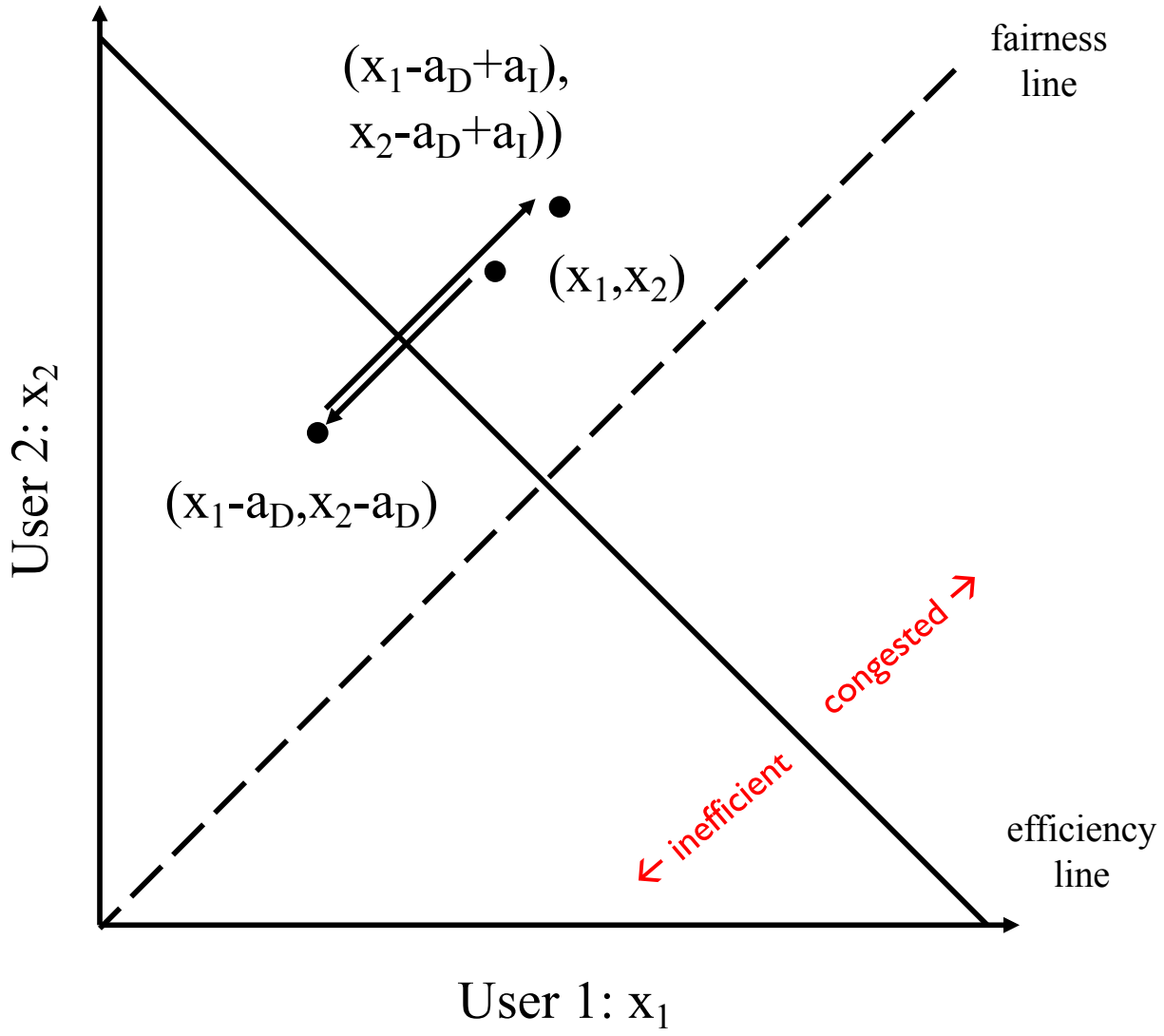


Example

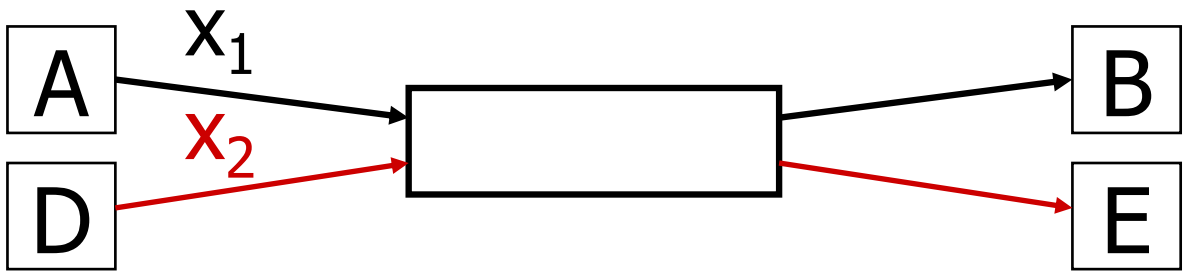

1
fairness line
Efficient: $x_1+x_2=1$
Fair
Congested: $x_1+x_2=1.2$
User 2: x_2
(0.2, 0.5)
(0.5, 0.5)
(0.7, 0.5)
Inefficient: $x_1+x_2=0.7$
(0.7, 0.3)
Efficient: $x_1+x_2=1$
Not fair
efficiency line
User 1: x_1
1

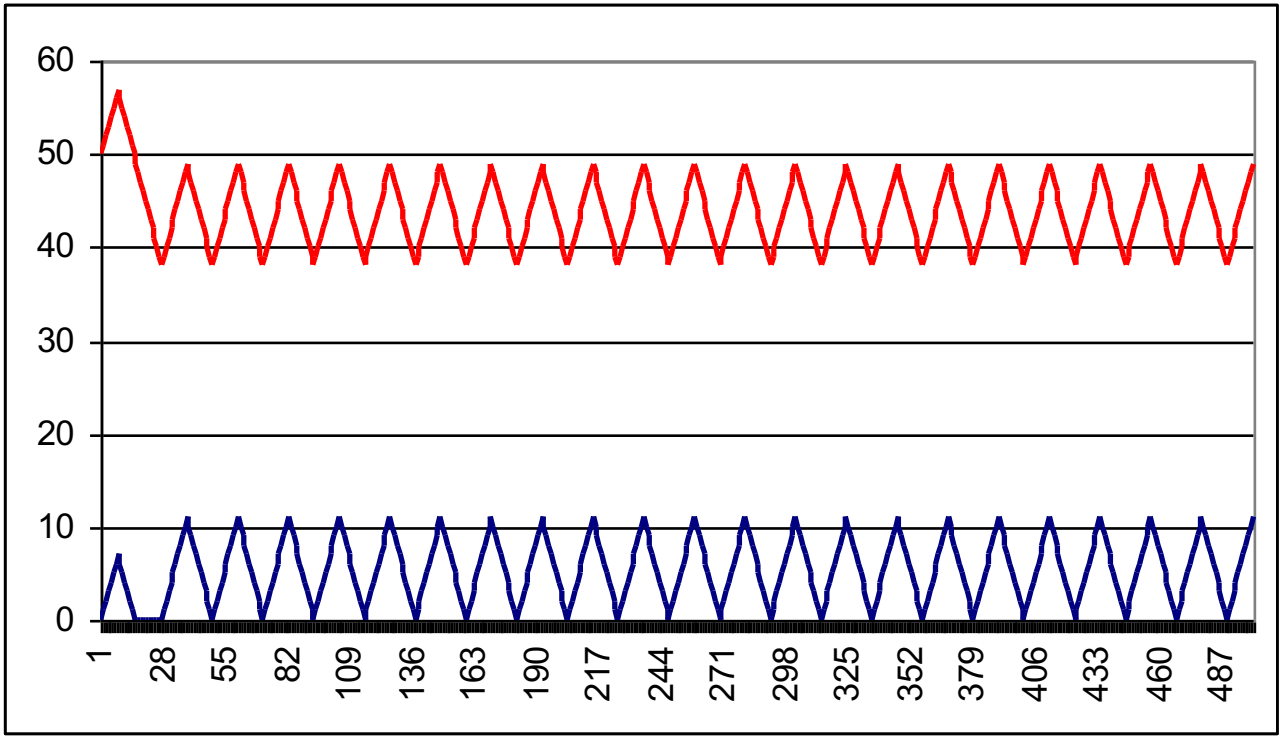
AIAD

- Increase: $x + a_I$
- Decrease: $x - a_D$
- Does not converge to fairness



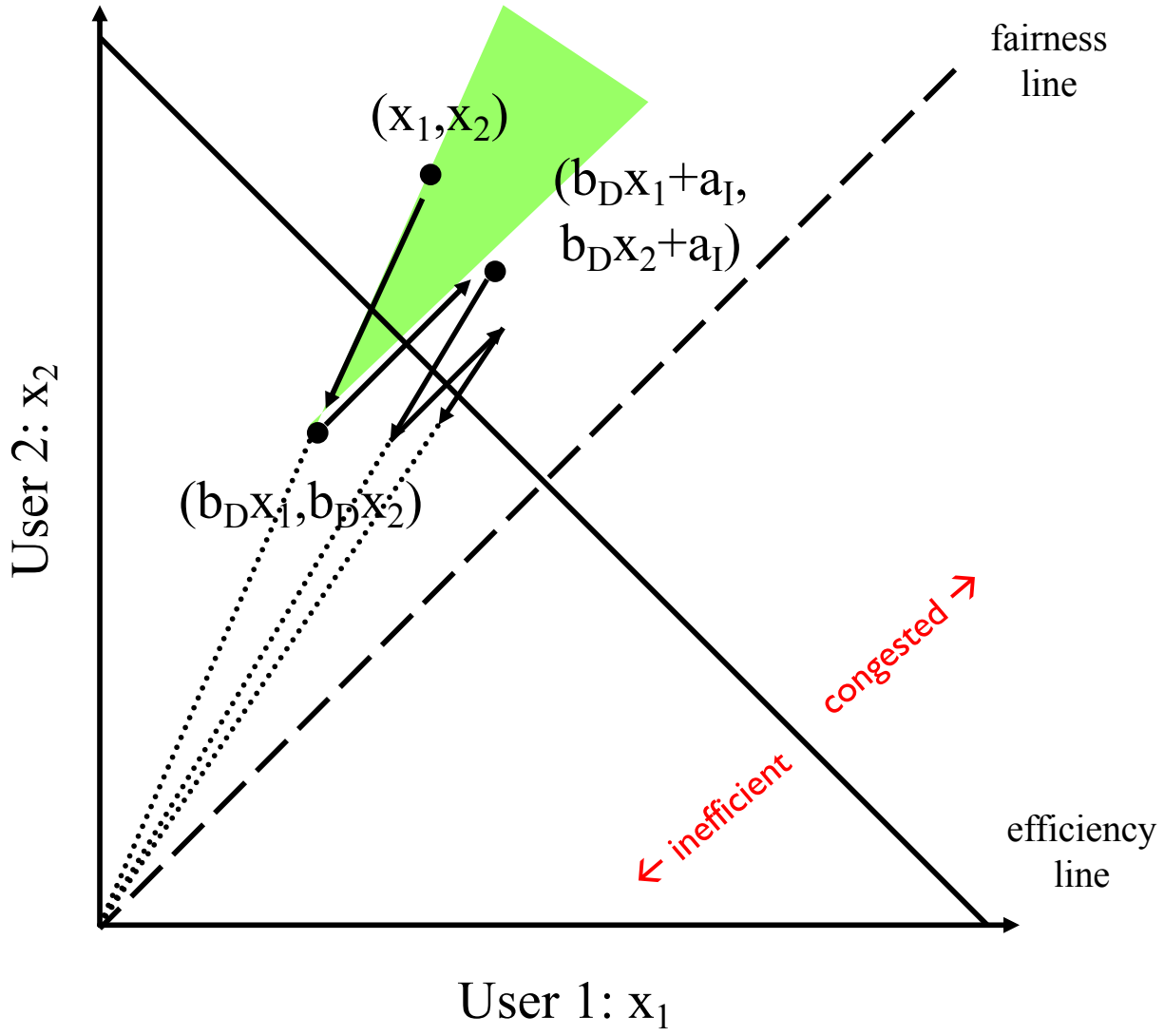
AIAD Sharing Dynamics



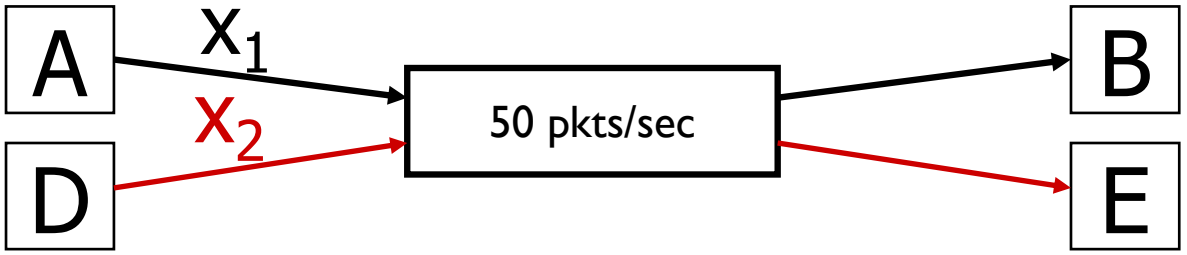


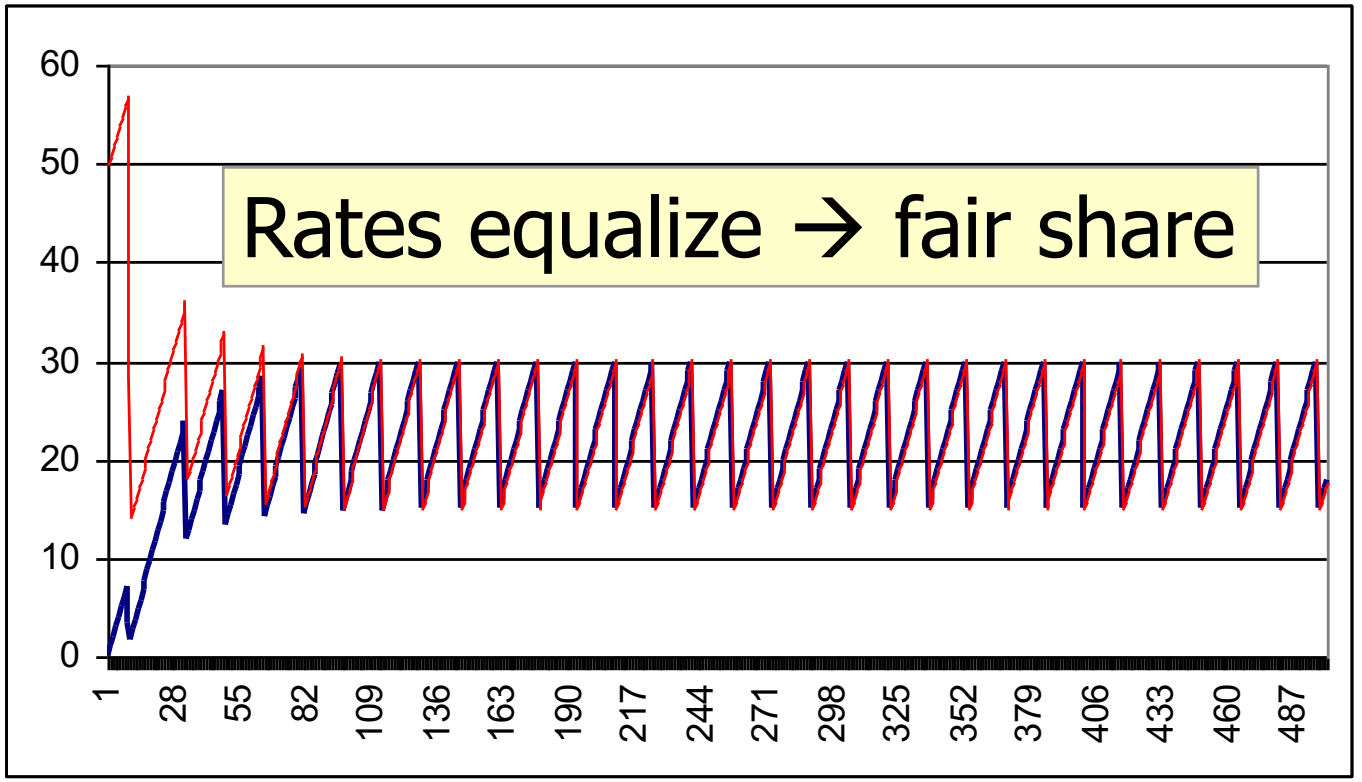
AIMD

- Increase: $x+a_I$
- Decrease: $x*b_D$
- Converges to fairness

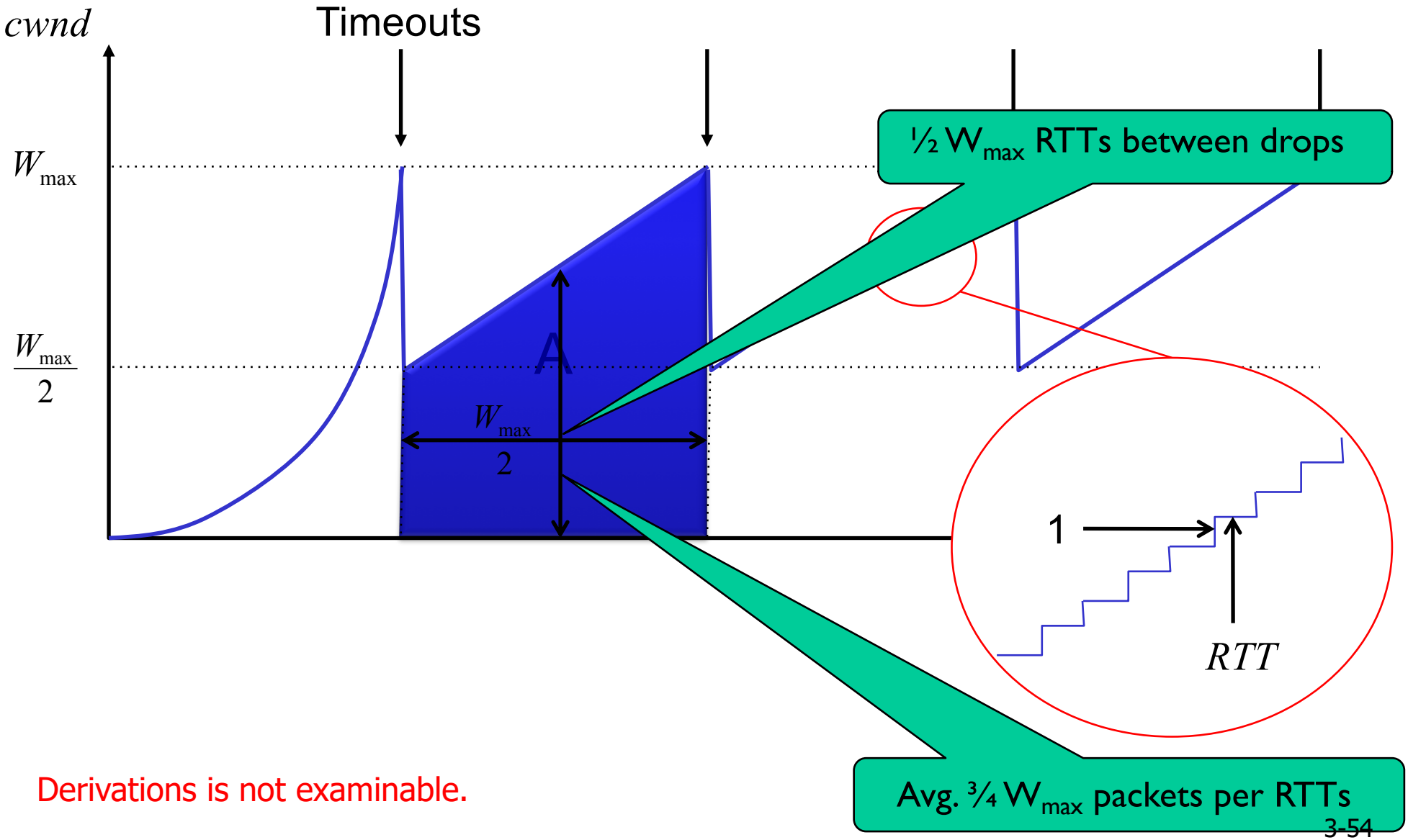


AIMD Sharing Dynamics



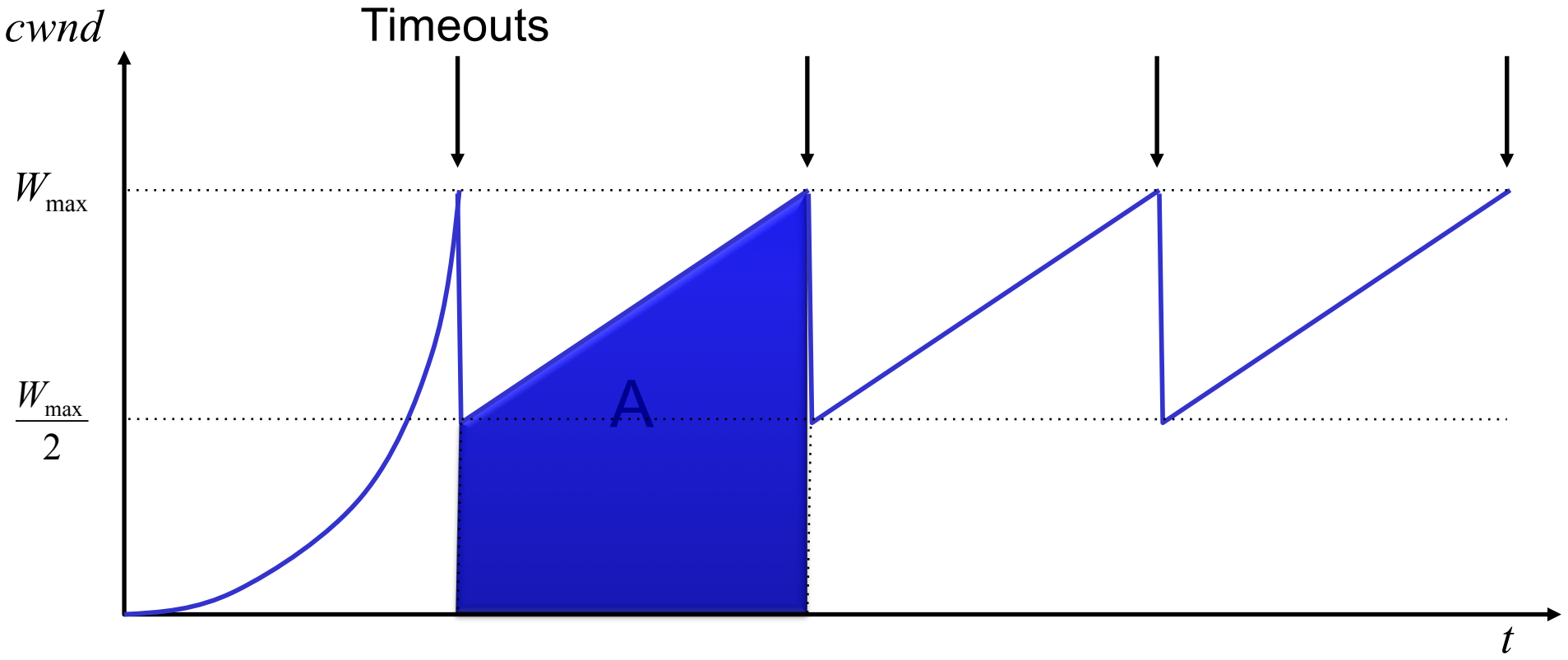


A Simple Model for TCP Throughput



Derivations is not examinable.

A Simple Model for TCP Throughput



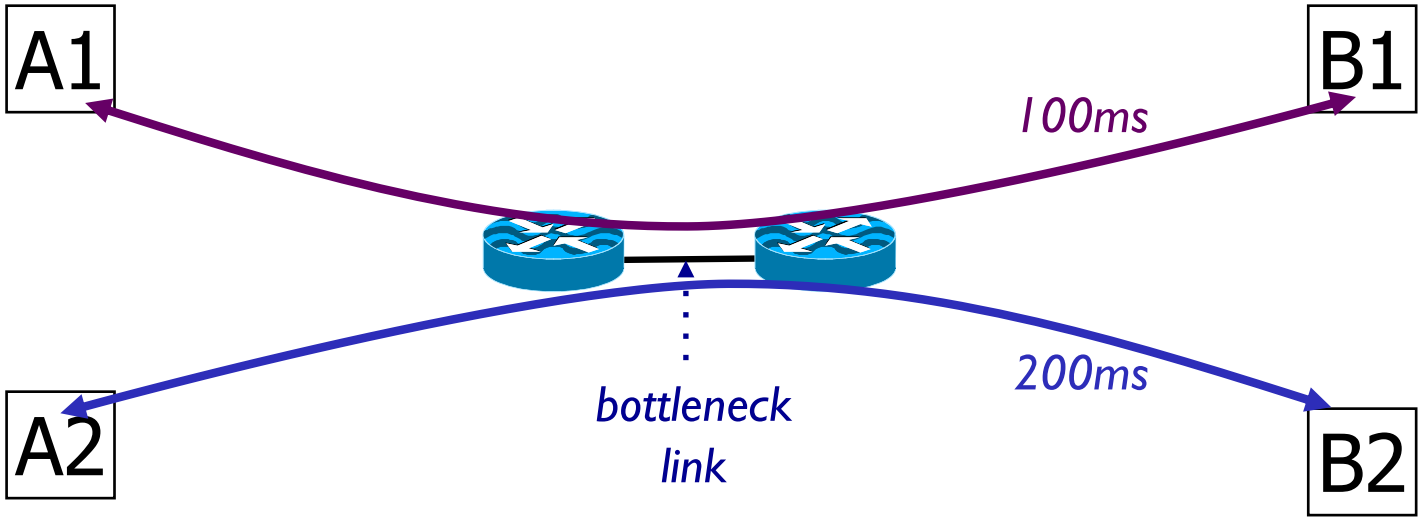
Packet drop rate, $p = 1/A$, where $A = \frac{3}{8} W_{max}^2$

Throughput, $B = \frac{A}{\left(\frac{W_{max}}{2}\right) RTT} = \sqrt{\frac{3}{2}} \frac{1}{RTT\sqrt{p}}$

Implications (1): Different RTTs

$$\text{Throughput} = \sqrt{\frac{3}{2}}\frac{1}{RTT\sqrt{p}}$$

- Flows get throughput inversely proportional to RTT
- TCP unfair in the face of heterogeneous RTTs!



Implications (2): Loss not due to congestion?

- TCP will confuse corruption with congestion
- Flow will cut its rate
  - Throughput ~ 1/sqrt(p) where p is loss prob.
  - Applies even for non-congestion losses!

Implications: (3) How do short flows fare?

- 50% of flows have < 1500B to send; 80% < 100KB
- Implication (1): short flows never leave slow start!
  - short flows never attain their fair share
- Implication (2): too few packets to trigger dupACKs
  - Isolated loss may lead to timeouts
  - At typical timeout values of ~500ms, might severely impact latency

Implications: (4) TCP fills up queues → long delays

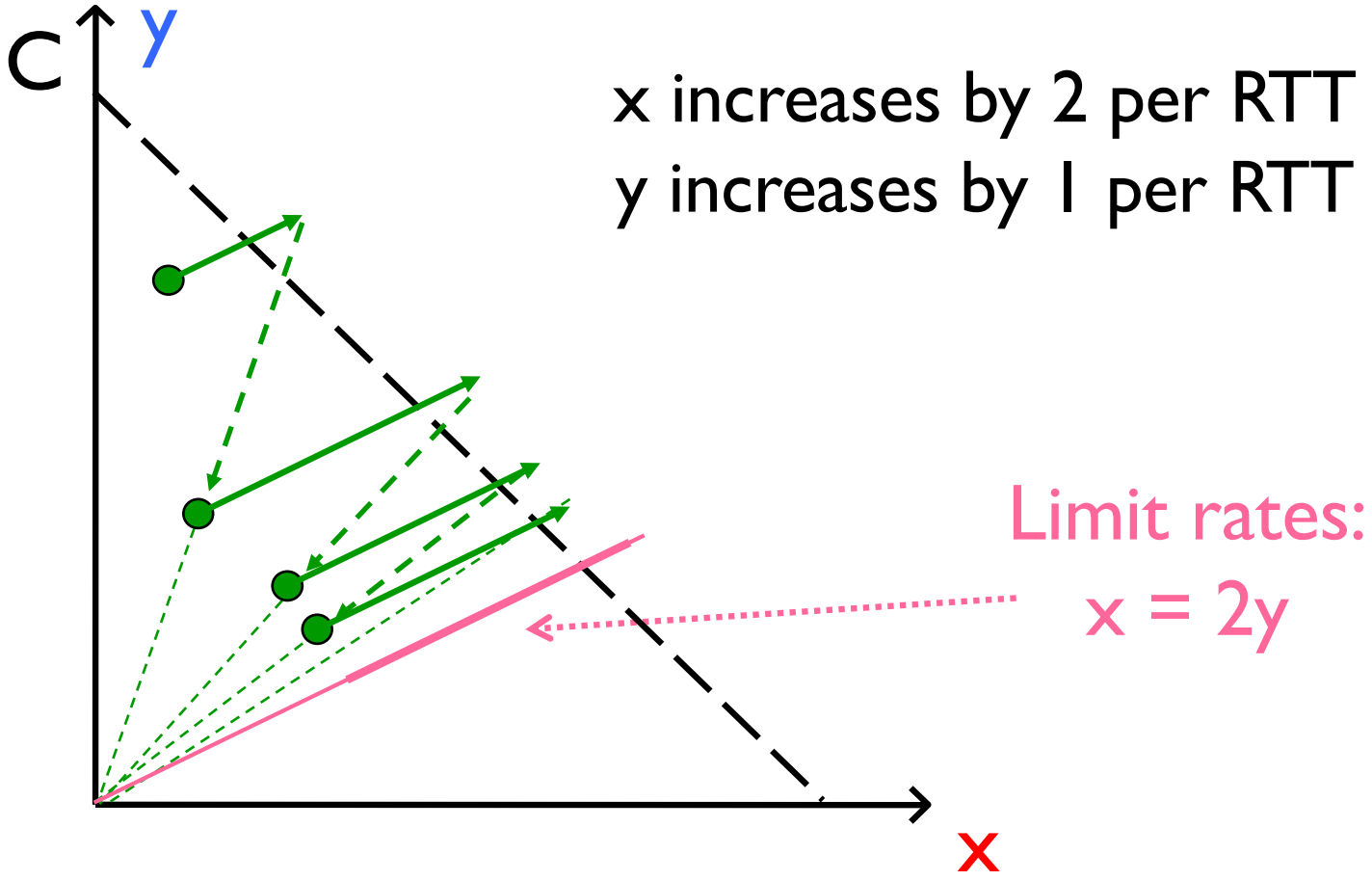
- A flow deliberately overshoots capacity, until it experiences a drop

- Means that delays are large for *everyone*
  - Consider a flow transferring a 10GB file sharing a bottleneck link with 10 flows transferring 100B

Implications: (5) Cheating

- Three easy ways to cheat
  - Increasing CWND faster than +1 MSS per RTT

Increasing CWND Faster



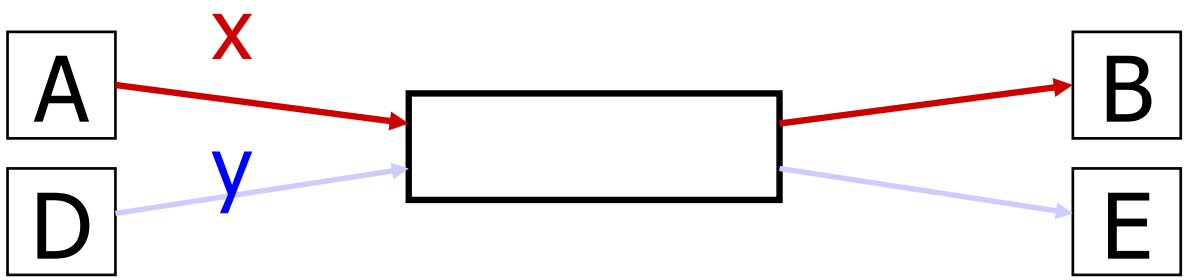
x increases by 2 per RTT
y increases by 1 per RTT

Limit rates:
x = 2y

Implications: (5) Cheating

- Three easy ways to cheat
  - Increasing CWND faster than +1 MSS per RTT
  - Opening many connections

Open Many Connections



Assume

- A starts 10 connections to B
- D starts 1 connection to E
- Each connection gets about the same throughput

Then A gets 10 times more throughput than D

Implications: (5) Cheating

- Three easy ways to cheat
  - Increasing CWND faster than +1 MSS per RTT
  - Opening many connections
  - Using large initial CWND

Transport Layer: Summary

- principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- instantiation, implementation in the Internet
  - UDP
  - TCP

next:

- leaving the network "edge" (application, transport layers)
- into the network "core"