COMP 431
Internet Services & Protocols

The Transport Layer
Congestion control in TCP

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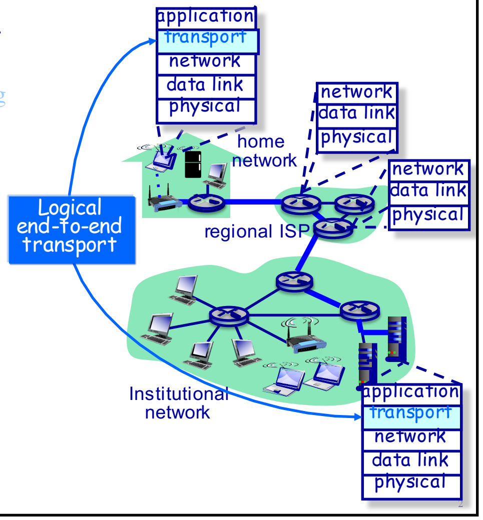
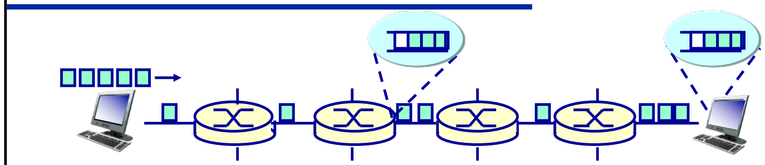
Transport Layer Protocols & Services
Outline

- Fundamental transport layer services
  - » Multiplexing/Demultiplexing
  - » Error detection
  - » Reliable data delivery
  - » Pipelining
  - » Flow control
  - » Congestion control
- Internet transport protocols
  - » UDP
  - » TCP



Congestion Control

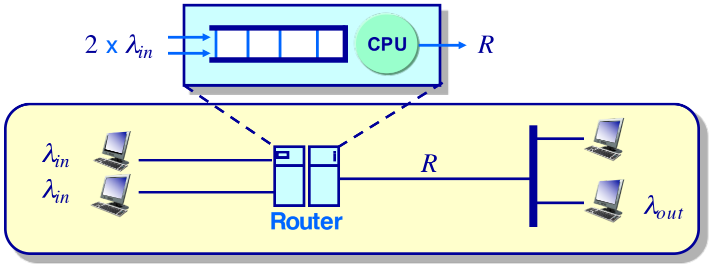
Congestion control *v.* Flow control



- In *flow control* the sender adjusts its transmission rate so as not to overwhelm the receiver
  - » One source is sending data too fast for a receiver to handle
- In *congestion control* the sender(s) adjust their trans-mission rate so as not to overwhelm routers in the network
  - » Many sources independently work to avoid sending too much data too fast for the network to handle
- Symptoms of congestion:
  - » Lost packets (buffer overflow at routers)
  - » Long delays (queuing in router buffers)

The Causes and Effects of Congestion

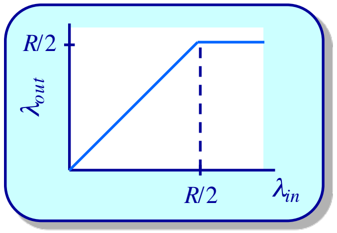
Scenario 1: Two equal-rate senders share a single link



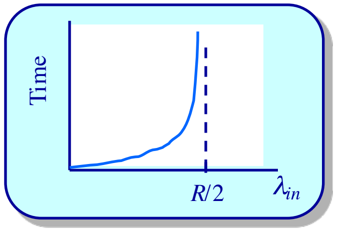
- Two sources send as fast as possible to two receivers across a shared link with capacity *R*
  - » Data is delivered to the application at the receiver at rate λ_{out}
- Packets queue at the router
  - » Assume the router has infinite storage capacity (Thus no packets are lost and there are no retransmissions)

The Causes and Effects of Congestion

Scenario 1: Two equal-rate senders share a single link



Throughput

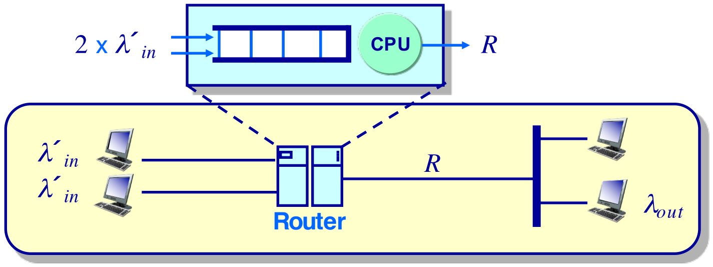


Delay

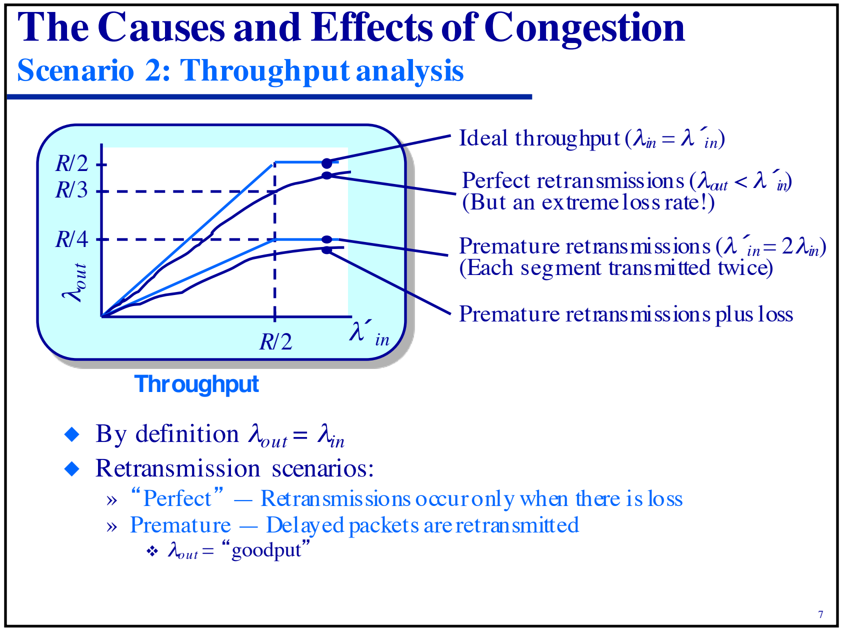
- The maximum achievable per connection throughput is constrained by $^1/_2$ the capacity of the shared link
- Exponentially large delays are experienced when the router becomes congested
  - » The queue grows without bound

The Causes and Effects of Congestion

Scenario 2: Finite capacity router queue



- Assume packets can now be lost
  - » Sender retransmits upon detection of loss
- Define *offered load* as the original transmissions plus retransmissions
  - » $\lambda'_{in} = \lambda_{in} + \lambda_{retransmit}$



The Causes and Effects of Congestion

Scenario 2: Throughput analysis

Ideal throughput ($\lambda_{in} = \lambda'_{in}$)

Perfect retransmissions ($\lambda_{out} < \lambda'_{in}$)
(But an extreme loss rate!)

Premature retransmissions ($\lambda'_{in} = 2\lambda_{in}$)
(Each segment transmitted twice)

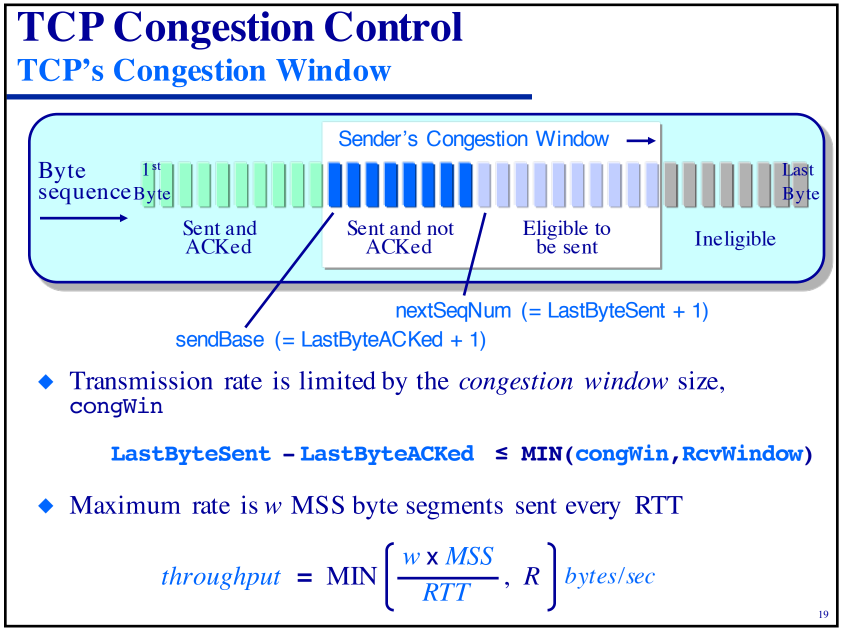
Premature retransmissions plus loss

Throughput

- By definition $\lambda_{out} = \lambda_{in}$
- Retransmission scenarios:
  - » "Perfect" — Retransmissions occur only when there is loss
  - » Premature — Delayed packets are retransmitted
    - ❖ λ_{out} = "goodput"

lambda out cannot be greater than lambda in

lambda out is not less that lambda in because we are only



TCP Congestion Control

TCP's Congestion Window

nextSeqNum (= LastByteSent + 1)

sendBase (= LastByteACKed + 1)

- Transmission rate is limited by the *congestion window* size, `congWin`

`LastByteSent - LastByteACKed ≤ MIN(congWin,RcvWindow)`

- Maximum rate is *w* MSS byte segments sent every RTT

$$throughput = \text{MIN}\left\{ \frac{w \times MSS}{RTT}, R \right\} \; bytes/sec$$

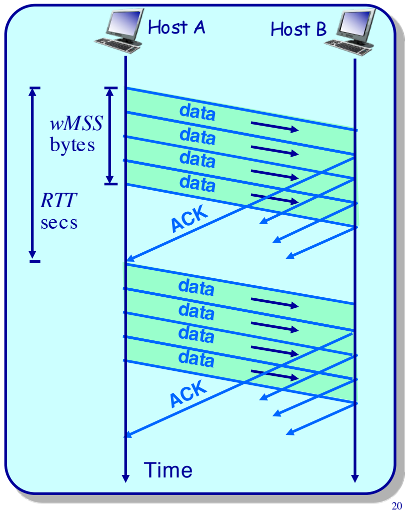
TCP Congestion Control

Congestion window and transmission rate

- If $w \times MSS/R < RTT$, then the maximum rate at which a TCP connection can transmit data is

$$\frac{w \times MSS}{RTT} \ \textit{bytes/sec}$$

- w is the minimum of the number of segments in the receiver's window or the congestion window

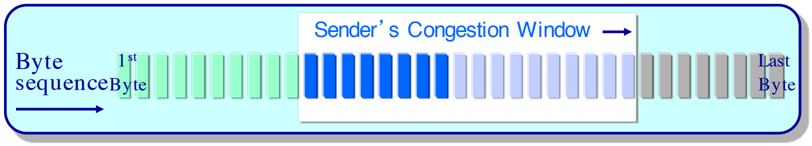


How does R impact RTT: throughput in this case is not influenced by R

to improve increase w or decrease RTT,

TCP Congestion Control

Congestion window control

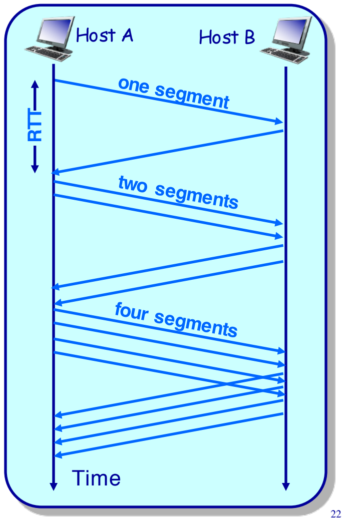


- TCP connections probe for available bandwidth
  - » Increase the congestion window until loss occurs
  - » When loss is detected decrease window, then begin probing (increasing) again
- The congestion window grows in two phases:
  - » *Slow start* — Ramp up transmission rate until loss occurs
  - » *Congestion avoidance* — Keep connection close to sustainable bandwidth
- A window size "threshold" distinguishes between slow start and congestion avoidance phases

TCP Congestion Control

Slowstart

```
congWin = 1 MSS
for (each original ACK received)
          congWin++
  until (loss event OR congWin > threshold)
```



- Exponential increase in window size each RTT until:
  - » Loss occurs
  - » *congWin = threshold*

  (Not so slow!)

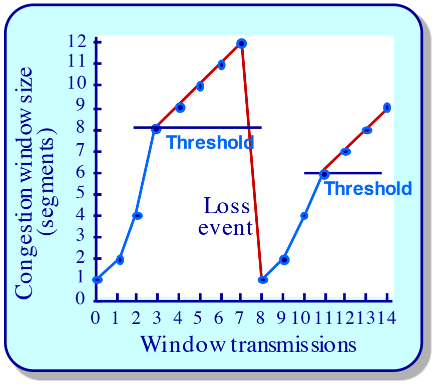
- Note: TCP implementations detect loss using:
  - » Timeout or three duplicate ACKs

TCP Congestion Control

Congestion avoidance

- Increase congestion window by 1 segment each RTT, decrease by a factor of 2 when packet loss is detected
  - » "Additive Increase, Multiplicative Decrease" (AIMD)

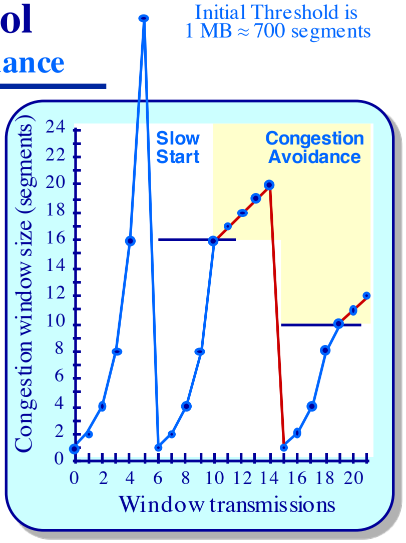
```
/* slowstart is over;
  congWin > threshold
*/
until (loss event) {
  whenever congWin segments
     ACKed:
       congWin++
  }
/* loss event timeout */
threshold = congWin/2
congWin = 1 MSS
perform slowstart
```



TCP Congestion Control

Slow-start *v.* Congestion avoidance

- The threshold is an estimate of a “safe” level of throughput that is sustainable in the network
  - » The threshold specifies a throughput that was sustainable in the recent past
- Slow-start quickly increases throughput to this threshold
- Congestion avoidance slows probes for additional available bandwidth beyond the threshold



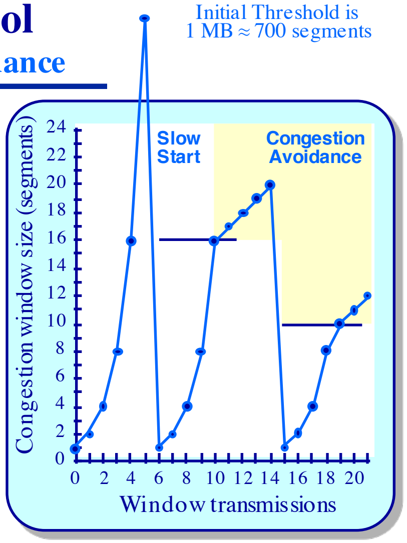
Initial Threshold is
1 MB ≈ 700 segments

Assume $RTT > \frac{w \times MSS}{R}$

TCP Congestion Control

Slow-start *v.* Congestion avoidance

- Loss (at any time) reduces the “safe” throughput estimate to 1/2 of the current throughput
  - » This is the throughput that resulted in loss
- Slow-start begins anew whenever there is loss
- Throughput at initial threshold = 1 MB/*RTT*
  - » At 1st threshold: 16*MSS*/*RTT*
  - » At 2nd threshold: 10*MSS*/*RTT*



Initial Threshold is
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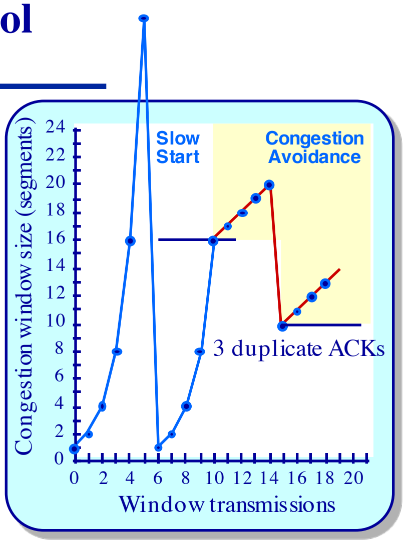
Assume $RTT > \frac{w \times MSS}{R}$

TCP Congestion Control

Major TCP variants

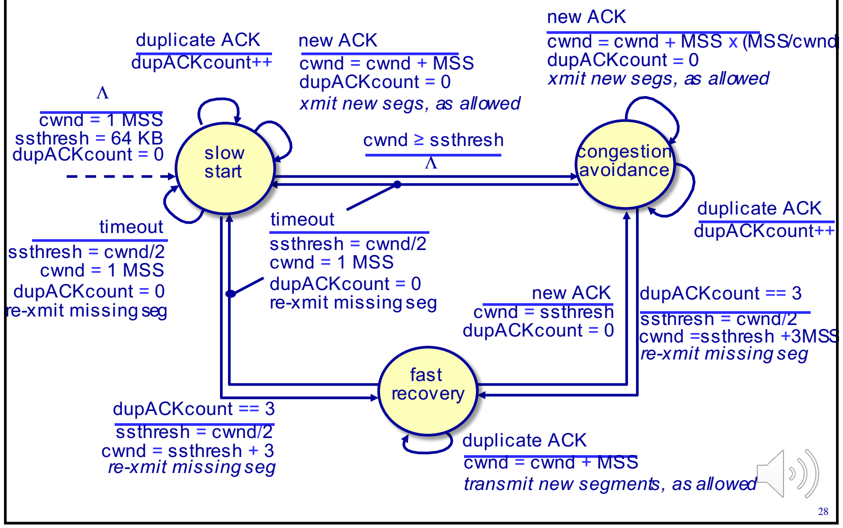
- TCP Tahoe:
  - » Loss signaled by timeout
  - » *threshold* = *congWin*/2
  - » *congWin* = 1 MSS
- TCP Reno:
  - » "Fast retransmit" — Receipt of 3 duplicate ACKs also signals a packet loss
  - » "Fast recovery" — Skips slowstart and continues in congestion avoidance new slowstart threshold
- Others: TCP NewReno, SACK, …



Assume *RTT* > $\frac{w \times MSS}{R}$

TCP Congestion Control

Summary (TCP Reno)



slow start

congestion avoidance

fast recovery

Λ / cwnd = 1 MSS, ssthresh = 64 KB, dupACKcount = 0

duplicate ACK / dupACKcount++

new ACK / cwnd = cwnd + MSS, dupACKcount = 0, *xmit new segs, as allowed*

cwnd ≥ ssthresh / Λ

new ACK / cwnd = cwnd + MSS x (MSS/cwnd), dupACKcount = 0, *xmit new segs, as allowed*

duplicate ACK / dupACKcount++

timeout / ssthresh = cwnd/2, cwnd = 1 MSS, dupACKcount = 0, re-xmit missing seg

timeout / ssthresh = cwnd/2, cwnd = 1 MSS, dupACKcount = 0, re-xmit missing seg

new ACK / cwnd = ssthresh, dupACKcount = 0

dupACKcount == 3 / ssthresh = cwnd/2, cwnd =ssthresh +3MSS, *re-xmit missing seg*

dupACKcount == 3 / ssthresh = cwnd/2, cwnd = ssthresh + 3, *re-xmit missing seg*

duplicate ACK / cwnd = cwnd + MSS, *transmit new segments, as allowed*

Advanced Topics: TCP over “long, fat, pipes”

“High speed TCP”

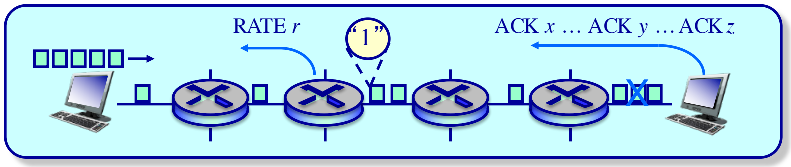
- Can an end system transmit at 10 Gbps over TCP?
  - Assume 1,500 byte segments and a 100 ms RTT
- 10 Gbps would requires W = 83,333 segments (with no loss)
- Throughput in terms of segment loss probability, *L* is

$$\text{TCP throughput} = \frac{1.22 \cdot \text{MSS}}{\text{RTT}\sqrt{L}}$$

  - Thus, to achieve 10 Gbps throughput, we need a loss rate of $L = 2 \cdot 10^{-10}$ (a crazy small loss rate!)
- For these reasons, new versions of TCP for “high-speed” networks exist
  - Beyond a certain window size, the window grows faster each RTT and decreases less on a loss

Approaches to Congestion Control

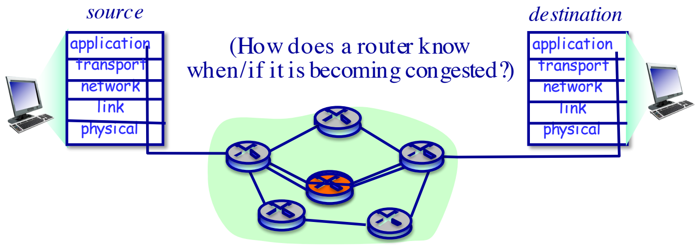
End-to-end *v.* Hop-by-hop



- End-to-end congestion control
  - End-systems receive no feedback from network
  - Congestion inferred by observing loss and/or delay
- Hop-by-hop congestion control
  - Routers provide feedback to end systems
    - Network determines an explicit rate that a sender should transmit at
    - Network signals congestion by setting a bit in a packet’s header (SNA, DECbit, TCP/IP ECN, ATM)

Router-based Congestion Control

Explicit congestion notification (ECN)

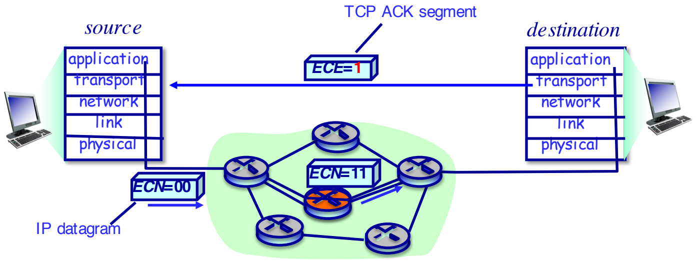


(How does a router know when/if it is becoming congested?)

- A router can detect it is becoming congested before packet loss occurs
- ECN allows a router to "mark" a packet to provide an indication of congestion to an end-system
- This enables end-systems to slow down *before* loss occurs

Router-based Congestion Control

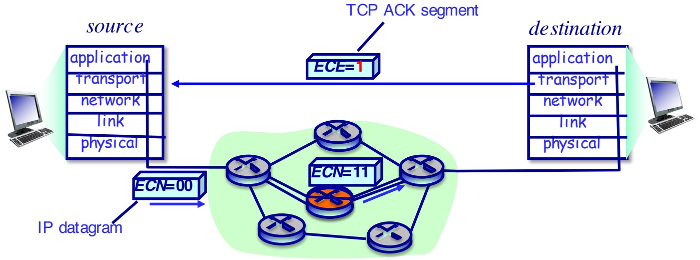
Explicit congestion notification (ECN)



- Two bits in IP header (ToS field) can be set by a router to indicate "early" congestion
- This indication is carried to receiving host
- The receiver detects the congestion indication in an arriving datagram and sets the ECN-Echo (ECE) bit on the next ACK to notify the sender of congestion

Router-based Congestion Control

Explicit congestion notification (ECN)



- When the source receives the ECE indication it reduces its congestion window as for a packet drop
  - » The source acknowledges the congestion indication by sending a segment with the *congestion window reduced* (CWR) bit set
- The receiver keeps transmitting ACKs with the ECE bit set until it receives a segment with the CWR bit set.