COMP 445 Data Communications & Computer networks Winter 2022

Transport Layer

- ✓ Transport-Layer services
- ✓ Multiplexing and demultiplexing
- **√** UDP
- ✓ Reliable data transfer
- **✓** TCP
- ✓ Principles of congestion control
- ✓ TCP Congestion control

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TCP: Overview RFCs: 793,1122,1323, 2018, 2581

- point-to-point:
 - one sender, one receiver
- reliable, in-order byte steam:
 - no "message boundaries"
- pipelined:
 - TCP congestion and flow control set window size

full duplex data:

- bi-directional data flow in same connection
- MSS: maximum segment size
- connection-oriented:
 - handshaking (exchange of control msgs) inits sender, receiver state before data exchange
- flow controlled:
 - sender will not overwhelm receiver

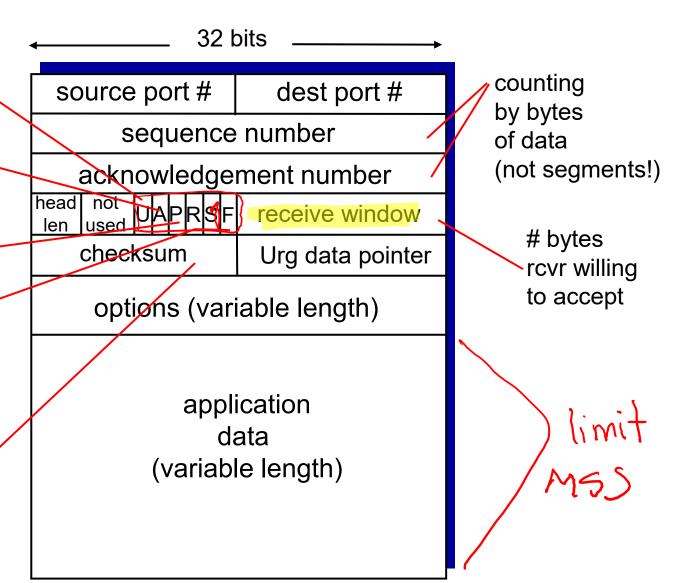
TCP segment structure

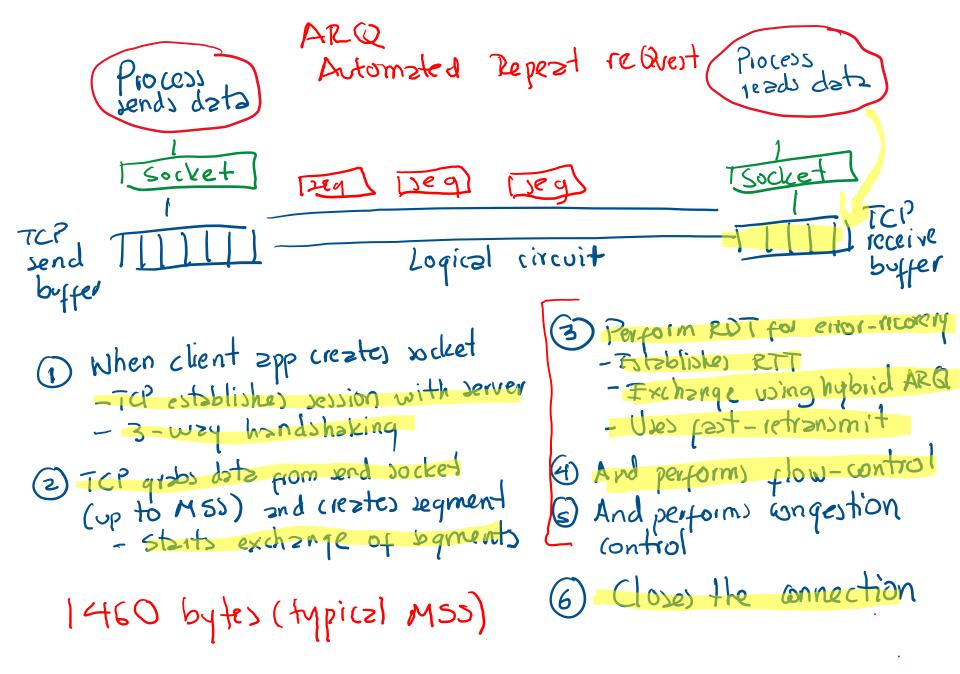
URG: urgent data (generally not used) ACK: ACK # valid

PSH: push data now (generally not used)

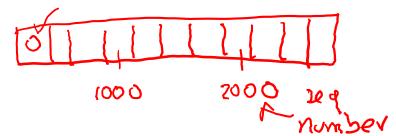
RST, SYN, FIN: connection estab (setup, teardown commands)

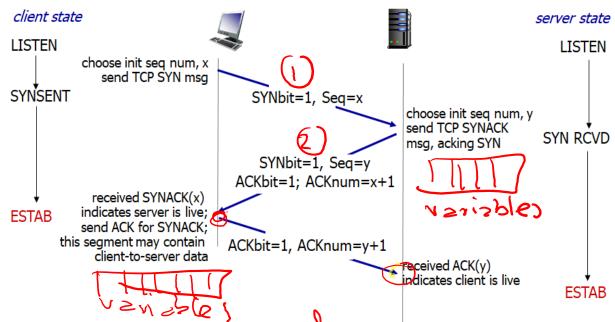
Internet checksum' (as in UDP)





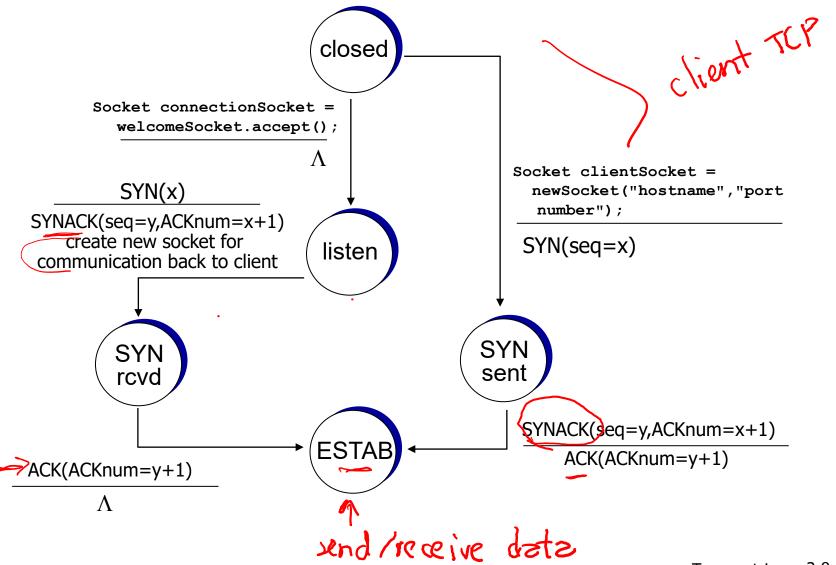
TCP 3-way handshake





- 1) SYN. no dete, client seq number (2) SYNACIC, no dete, ACK client seq num, server seq number
- ACK, dete (optional), ACK server seq number

TCP 3-way handshake: FSM



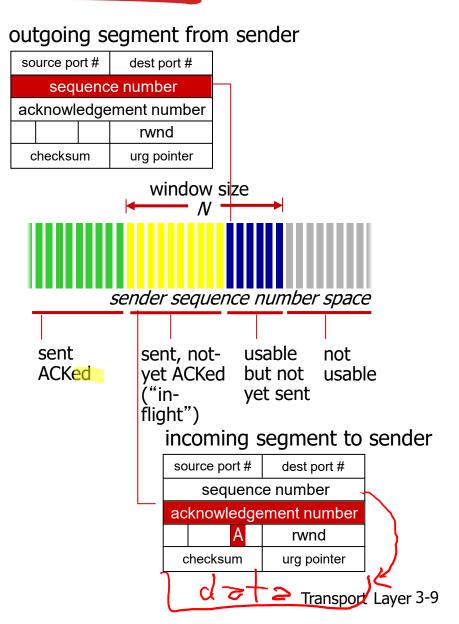
TCP seq. numbers, ACKs

sequence numbers:

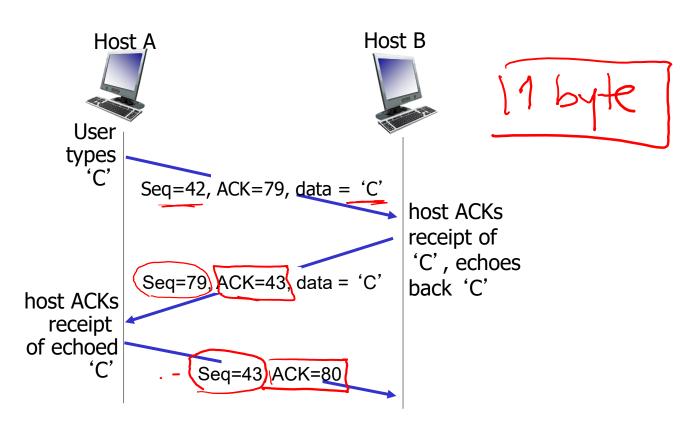
 byte stream "number" of first byte in segment's data

acknowledgements:

- seq # of next byte expected from other side
- cumulative ACK
- Q: how receiver handles out-of-order segments
 - A: TCP spec doesn't say,
 - up to implementor



TCP seq. numbers, ACKs



simple telnet scenario

TCP round trip time, timeout

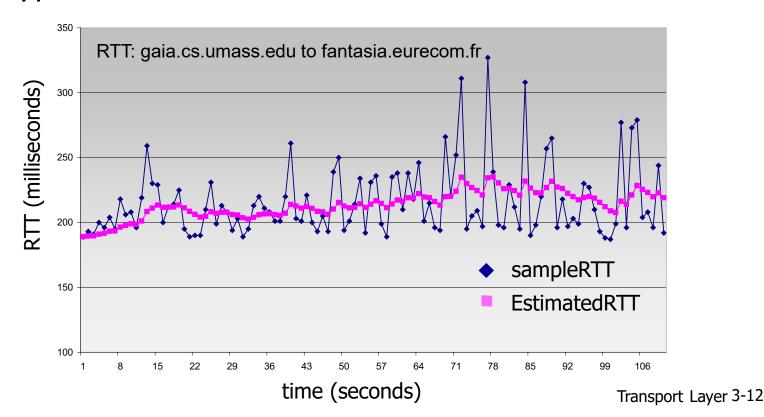
- Q: how to set TCP timeout value?
- longer than RTT
 - but RTT varies
- too short: premature timeout, unnecessary retransmissions
- too long: slow reaction to segment loss

- Q: how to estimate RTT?
- SampleRTT: measured time from segment transmission until ACK receipt
 - ignore retransmissions
- SampleRTT will vary, want estimated RTT "smoother"
 - average several recent measurements, not just current SampleRTT

TCP round trip time, timeout

EstimatedRTT = $(1-\alpha)$ *EstimatedRTT + α *SampleRTT

- exponential weighted moving average
- influence of past sample decreases exponentially fast
- typical value: $\alpha = 0.125$



TCP round trip time, timeout

- timeout interval: EstimatedRTT plus "safety margin"
 - large variation in **EstimatedRTT** -> larger safety margin
- estimate SampleRTT deviation from EstimatedRTT:

```
DevRTT = (1-\beta)*DevRTT + \beta*|SampleRTT-EstimatedRTT| (typically, \beta = 0.25)
```

TimeoutInterval = EstimatedRTT + 4*DevRTT



estimated RTT

safety margin"

^{*} Check out the online interactive exercises for more examples: http://gaia.cs.umass.edu/kurose ross/interactive/

TCP reliable data transfer

- TCP creates rdt service on top of IP's unreliable service
 - pipelined segments
 - cumulative acks
 - single retransmission timer
- retransmissions triggered by:
 - timeout events
 - duplicate acks

let's initially consider simplified TCP sender:

- ignore duplicate acks
- ignore flow control, congestion control

TCP sender events:

data rcvd from app:

- create segment with seq #
- seq # is byte-stream number of first data byte in segment
- start timer if not already running
 - think of timer as for oldest unacked segment
 - expiration interval: TimeOutInterval

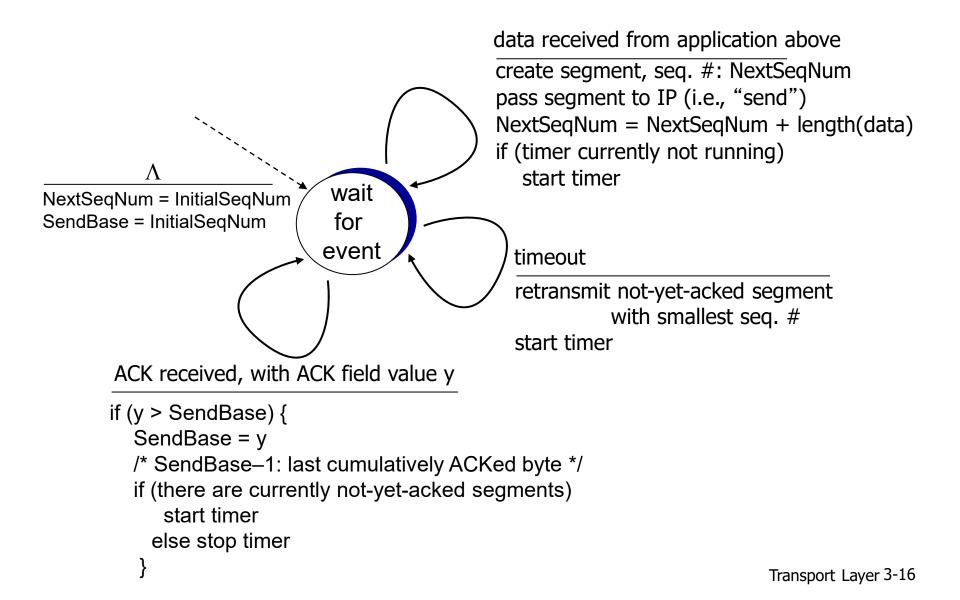
timeout:

- retransmit segment that caused timeout
- restart timer
 ack rcvd:
- if ack acknowledges previously unacked segments
 - update what is known to be ACKed
 - start timer if there are still unacked segments

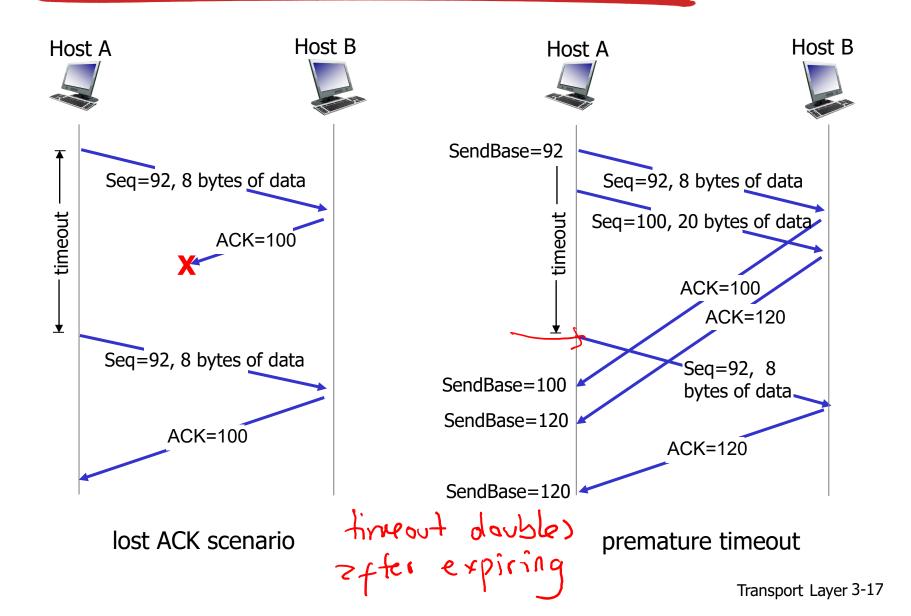
extinuted KTT devRTT

iony post,

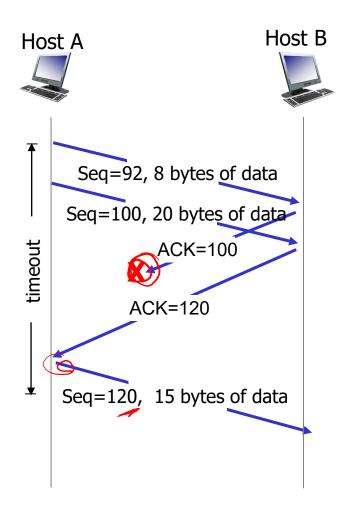
TCP sender (simplified)



TCP: retransmission scenarios



TCP: retransmission scenarios



cumulative ACK

TCP ACK generation [RFC 1122, RFC 2581]

event at receiver	TCP receiver action
arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed	delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK
arrival of in-order segment with expected seq #. One other segment has ACK pending	immediately send single cumulative ACK, ACKing both in-order segments
arrival of out-of-order segment higher-than-expect seq. # . Gap detected	immediately send duplicate ACK, indicating seq. # of next expected byte
arrival of segment that partially or completely fills gap	immediate send ACK, provided that segment starts at lower end of gap

TCP fast retransmit

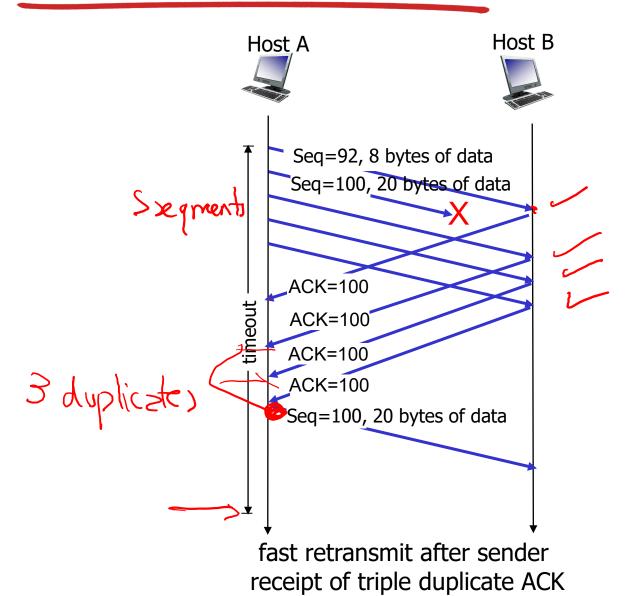
- time-out period often relatively long:
 - long delay before resending lost packet
- detect lost segments via duplicate ACKs.
 - sender often sends many segments backto-back
 - if segment is lost, there will likely be many duplicate ACKs.

TCP fast retransmit

if sender receives 3 ACKs for same data ("triple duplicate ACKs"), resend unacked segment with smallest seq #

 likely that unacked segment lost, so don't wait for timeout

TCP fast retransmit



TCP flow control

application may remove data from TCP socket buffers

... slower than TCP receiver is delivering (sender is sending)

application process application OS TCP socket receiver buffers TCP code ĬΡ code from sender

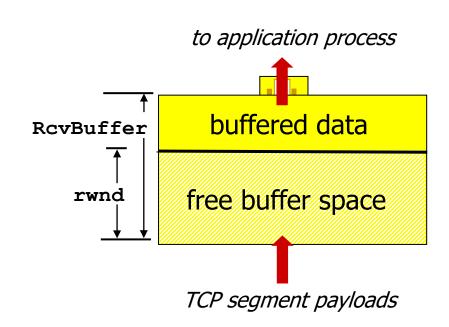
receiver protocol stack

flow control

receiver controls sender, so sender won't overflow receiver's buffer by transmitting too much, too fast

TCP flow control

- receiver "advertises" free buffer space by including rwnd value in TCP header of receiver-to-sender segments
 - RcvBuffer size set via socket options (typical default is 4096 bytes)
 - many operating systems autoadjust RcvBuffer
- sender limits amount of unacked ("in-flight") data to receiver's rwnd value
- guarantees receive buffer will not overflow

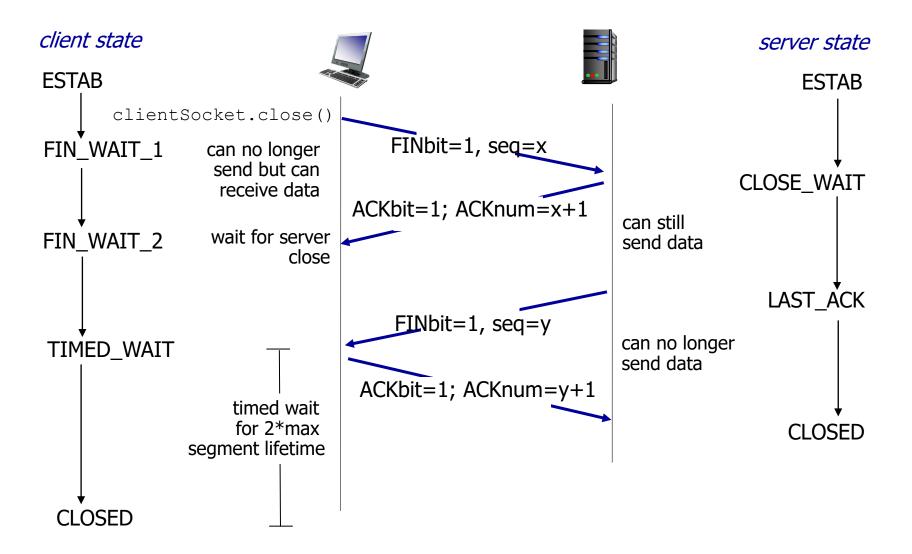


receiver-side buffering

TCP: closing a connection

- client, server each close their side of connection
 - send TCP segment with FIN bit = I
- respond to received FIN with ACK
 - on receiving FIN, ACK can be combined with own FIN
- simultaneous FIN exchanges can be handled

TCP: closing a connection



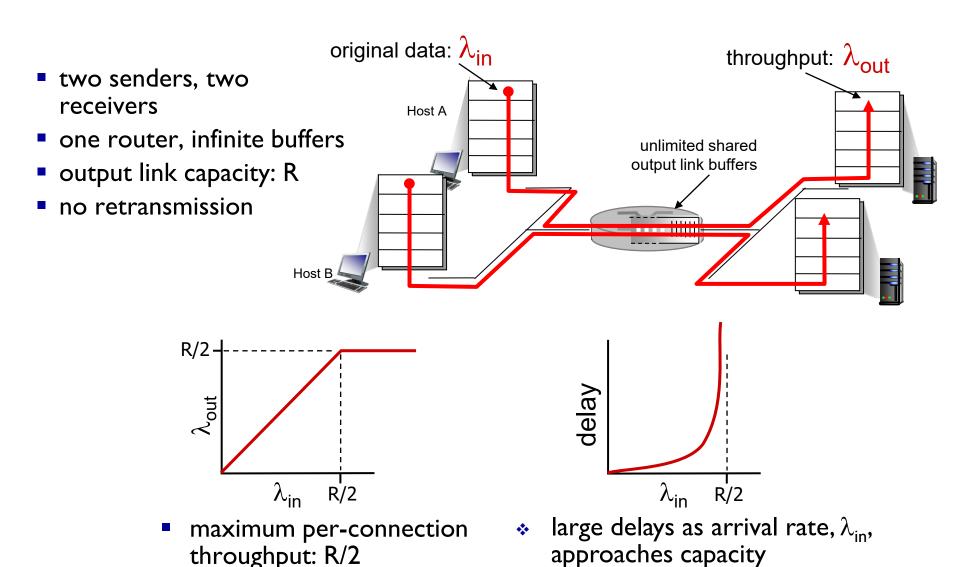
Transport Layer

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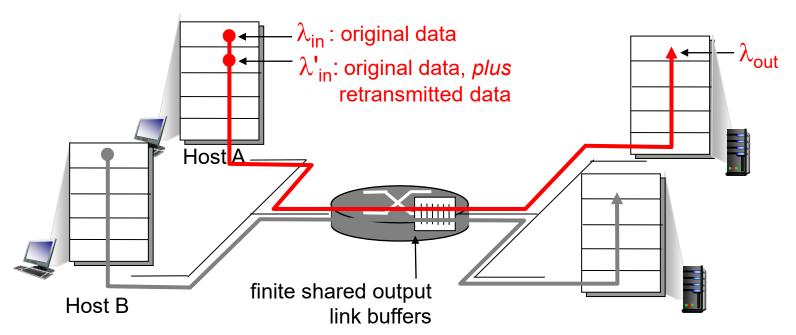
Principles of congestion control

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!

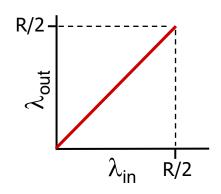


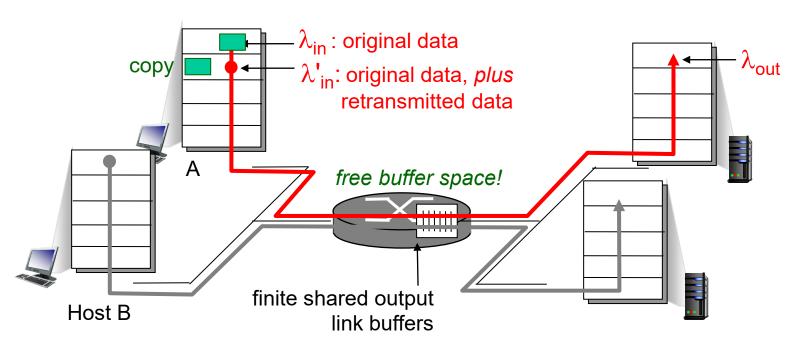
- one router, finite buffers
- sender retransmission of timed-out packet
 - application-layer input = application-layer output: λ_{in} = λ_{out}
 - transport-layer input includes retransmissions : $\lambda_{in} \ge \lambda_{in}$



idealization: perfect knowledge

 sender sends only when router buffers available

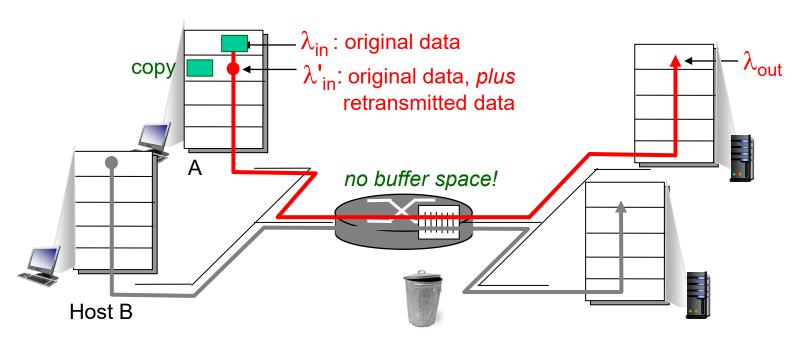




Idealization: known loss

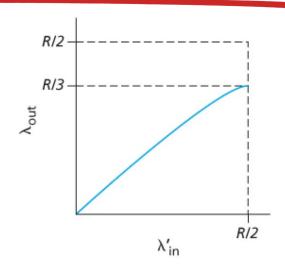
packets can be lost, dropped at router due to full buffers

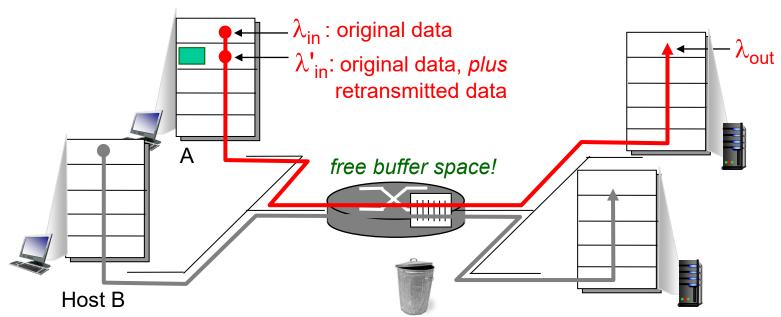
sender only resends if packet known to be lost



Idealization: known loss packets can be lost, dropped at router due to full buffers

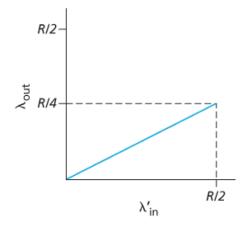
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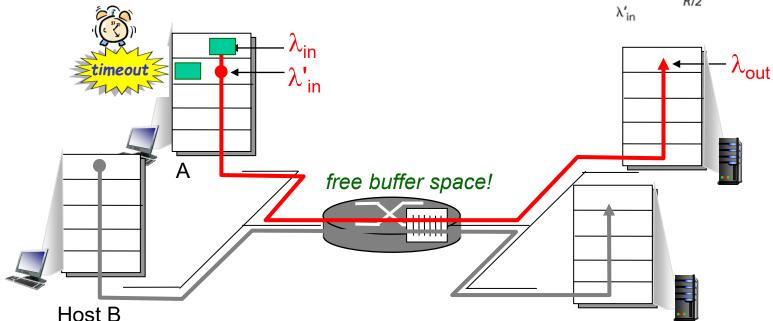




Realistic: duplicates

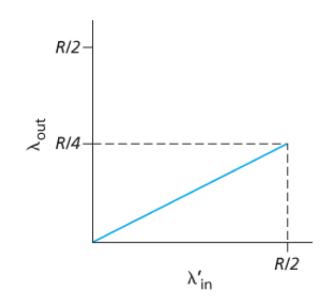
- packets can be lost, dropped at router due to full buffers
- sender times out prematurely, sending two copies, both of which are delivered





Realistic: duplicates

- packets can be lost, dropped at router due to full buffers
- sender times out prematurely, sending two copies, both of which are delivered



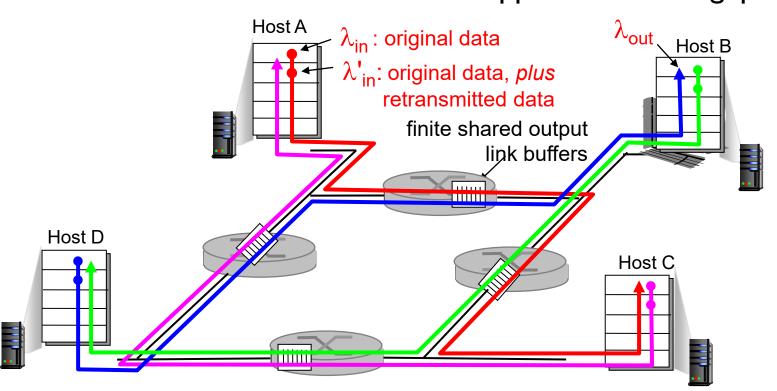
"costs" of congestion:

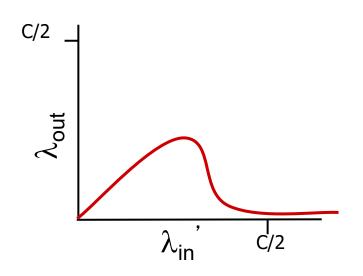
- more work (retrans) for given "goodput"
- unneeded retransmissions: link carries multiple copies of pkt
 - decreasing goodput

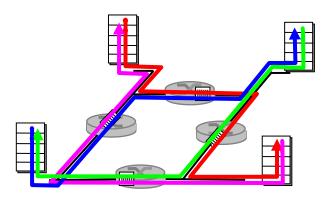
- four senders
- multihop paths
- timeout/retransmit

Q: what happens as λ_{in} and λ_{in} increase?

A: as red λ_{in} increases, all arriving blue pkts at upper queue are dropped, blue throughput $\rightarrow 0$







another "cost" of congestion:

when packet dropped, any "upstream transmission capacity used for that packet was wasted!

Transport Layer

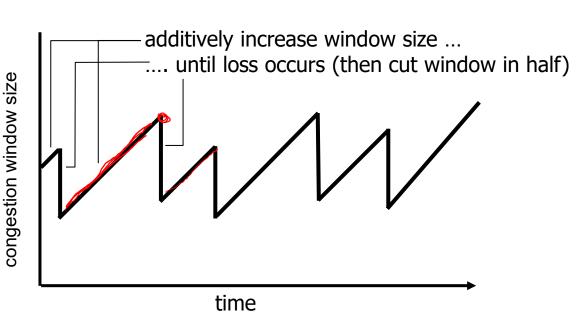
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TCP congestion control: additive increase multiplicative decrease

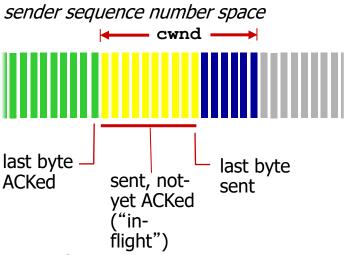
- approach: sender increases transmission rate (window size), probing for usable bandwidth, until loss occurs
 - additive increase: increase cwnd by I MSS every RTT until loss detected
 - multiplicative decrease: cut cwnd in half after loss

AIMD saw tooth behavior: probing for bandwidth

cwnd: TCP sender



TCP Congestion Control: details



sender limits transmission:

 cwnd is dynamic, function of perceived network congestion

TCP sending rate:

 roughly: send cwnd bytes, wait RTT for ACKS, then send more bytes

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

min (round, cound)

v Slow start

v Congestion avoidance

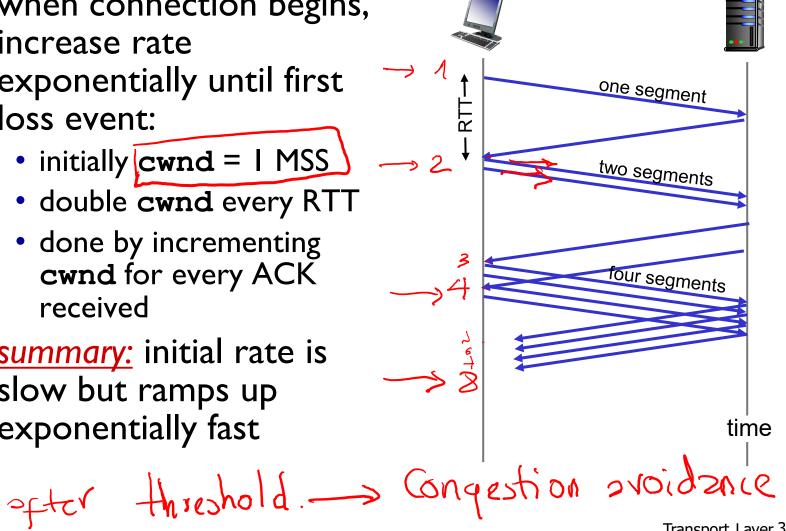
V Tast recovery

Transport Layer 3-39

TCP Slow Start

when connection begins, increase rate exponentially until first loss event:

- initially cwnd = I MSS
- double cwnd every RTT
- done by incrementing cwnd for every ACK received
- summary: initial rate is slow but ramps up exponentially fast



Host A

Transport Layer 3-40

Host B

TCP: detecting, reacting to loss

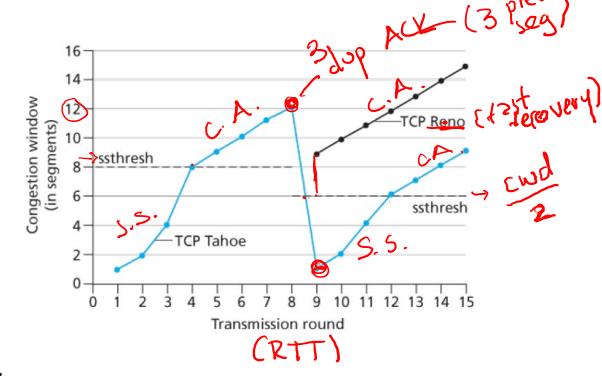
- loss indicated by timeout: MSS (3)
 cwnd set to I MSS;

 - window then grows exponentially (as in slow start) to threshold, then grows linearly -> Conq. Avoidance
- loss indicated by 3 duplicate ACKs: TCP RENO
 - dup ACKs indicate network capable of delivering some segments -> fzst recovery
 - cwnd is cut in half window then grows linearly
- TCP Tahoe always sets cwnd to I (timeout or 3 duplicate acks)

TCP: switching from slow start to CA

Q: when should the exponential increase switch to linear?

A: when **cwnd** gets to 1/2 of its value before timeout.

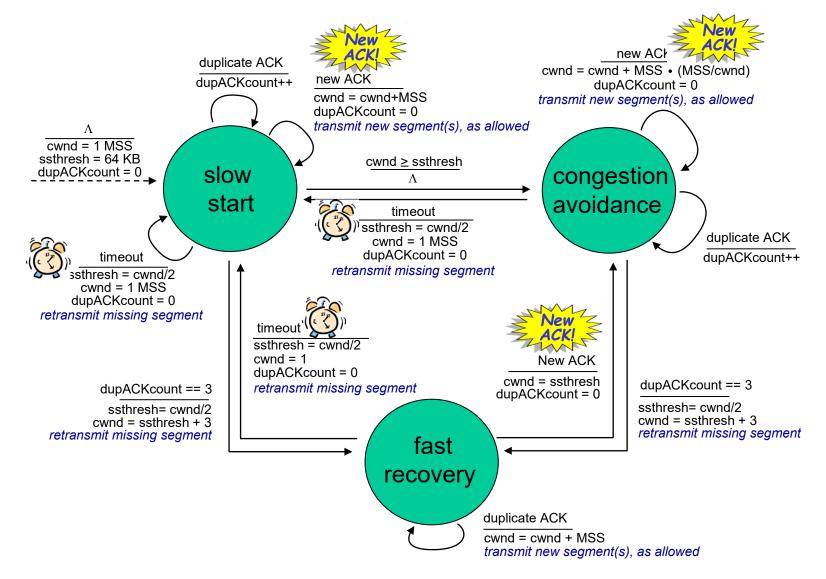


Implementation:

- variable ssthresh
- on loss event, ssthresh is set to 1/2 of cwnd just before loss event

^{*} Check out the online interactive exercises for more examples: http://gaia.cs.umass.edu/kurose_ross/interactive/

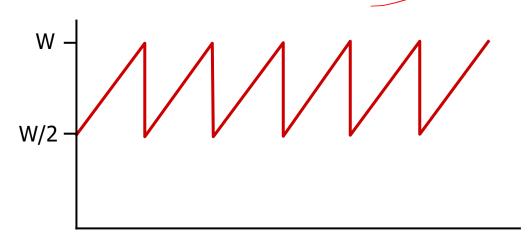
Summary: TCP Congestion Control



TCP throughput

- avg. TCP thruput as function of window size, RTT?
 - · ignore slow start, assume always data to send
- W: window size (measured in bytes) where loss occurs
 - avg. window size (# in-flight bytes) is 3/4 W
 - avg. thruput is 3/4W per RTT

avg TCP thruput =
$$\frac{3}{4} \frac{W}{RTT}$$
 bytes/sec



W+W RTT ZRTT

TCP Futures: TCP over "long, fat pipes"

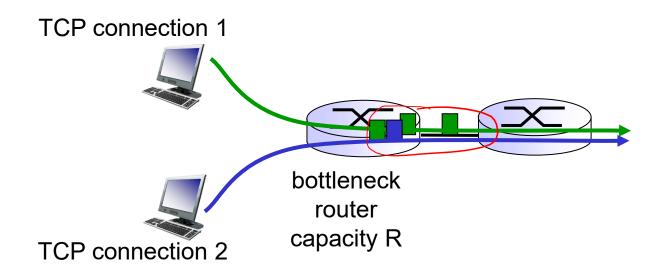
- example: I500 byte segments, I00ms RTT, wantI0 Gbps throughput
- requires W = 83,333 in-flight segments
- throughput in terms of segment loss probability, L [Mathis 1997]:

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

- ⇒ to achieve 10 Gbps throughput, need a loss rate of L = $2 \cdot 10^{-10}$ a very small loss rate!
- new versions of TCP for high-speed

TCP Fairness

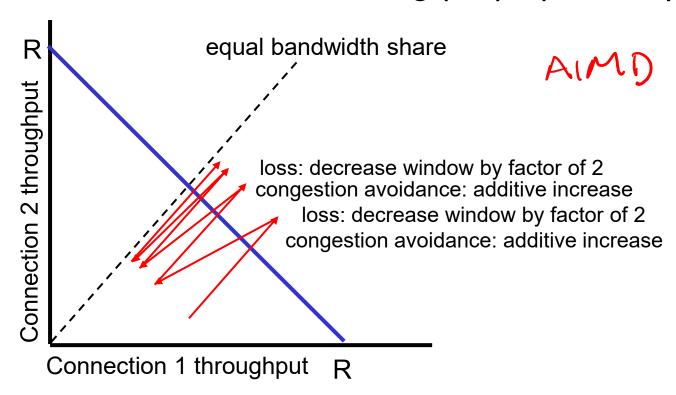
fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



Why is TCP fair?

two competing sessions:

- additive increase gives slope of I, as throughput increases
- multiplicative decrease decreases throughput proportionally



Fairness (more)

Fairness and UDP

- multimedia apps often do not use TCP
 - do not want rate throttled by congestion control
- instead use UDP:
 - send audio/video at constant rate, tolerate packet loss

Fairness, parallel TCP connections

- application can open multiple parallel connections between two hosts
- web browsers do this
- e.g., link of rate R with 9 existing connections:
 - new app asks for I TCP, gets rate R/I0
 - new app asks for 11 TCPs, gets R/2

Chapter 3: 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"
- two network layer chapters:
 - data plane
 - control plane

References

Figures and slides are taken/adapted from:

 Jim Kurose, Keith Ross, "Computer Networking: A Top-Down Approach", 7th ed. Addison-Wesley, 2012. All material copyright 1996-2016 J.F Kurose and K.W. Ross, All Rights Reserved