Lecture 5 – Chapter 3

CIS 5617, Fall 2019
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Based on Slides created by JFK/KWR

7th edition

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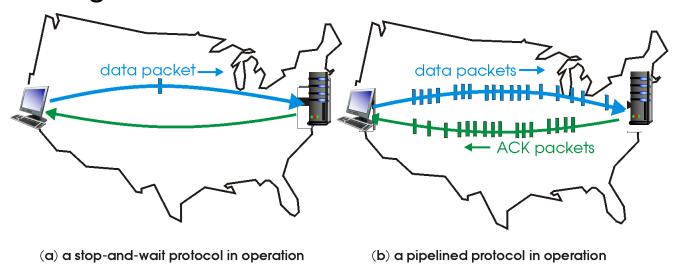
Pearson/Addison Wesley

April 2016

Pipelined protocols

pipelining: sender allows multiple, "in-flight", yetto-be-acknowledged pkts

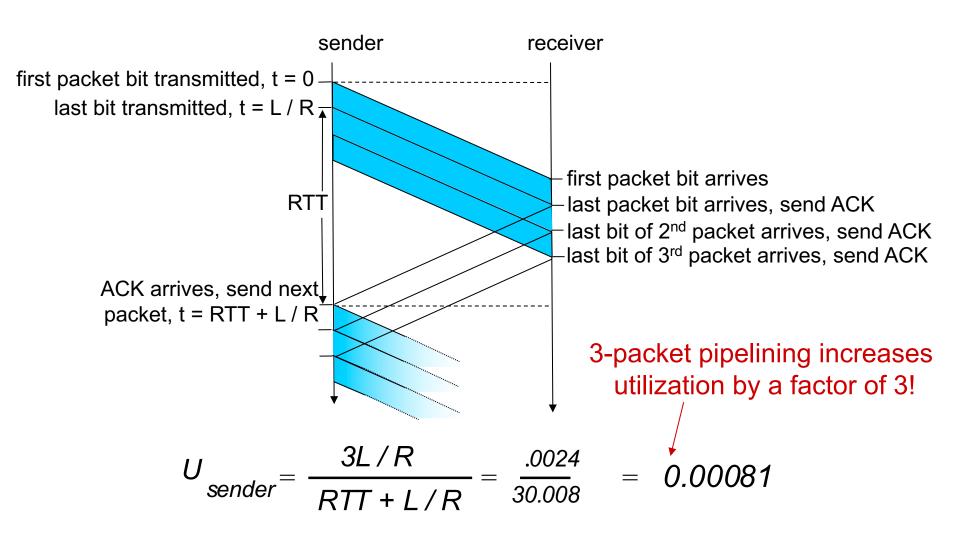
- range of sequence numbers must be increased
- buffering at sender and/or receiver



two generic forms of pipelined protocols: go-Back-N,

selective repeat

Pipelining: increased utilization



Pipelined protocols: overview

Go-back-N:

- sender can have up to N unacked packets in pipeline
- receiver only sends cumulative ack
 - doesn't ack packet if there's a gap
- sender has timer for oldest unacked packet
 - when timer expires, retransmit all unacked packets

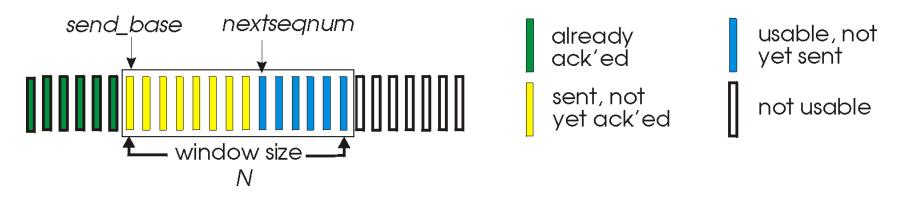
Selective Repeat:

- sender can have up to N unack' ed packets in pipeline
- rcvr sends individual ack for each packet

- sender maintains timer for each unacked packet
 - when timer expires, retransmit only that unacked packet

Go-Back-N: sender

- k-bit seq # in pkt header
- "window" of up to N, consecutive unack'ed pkts allowed

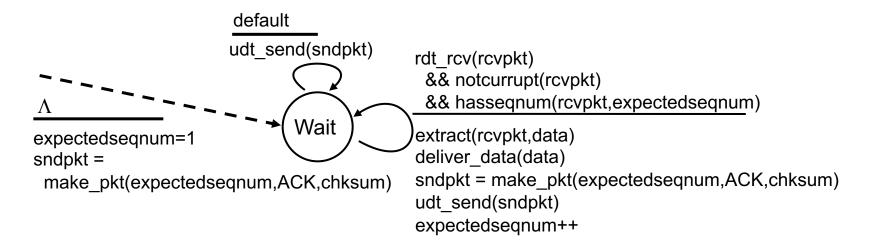


- ACK(n):ACKs all pkts up to, including seq # n "cumulative ACK"
 - may receive duplicate ACKs (see receiver)
- timer for oldest in-flight pkt
- timeout(n): retransmit packet n and all higher seq # pkts in window

GBN: sender extended FSM

```
rdt send(data)
                       if (nextseqnum < base+N) {
                          sndpkt[nextseqnum] = make pkt(nextseqnum,data,chksum)
                          udt send(sndpkt[nextseqnum])
                          if (base == nextseqnum)
                           start timer
                          nextseqnum++
                       else
                        refuse data(data)
  base=1
  nextseqnum=1
                                           timeout
                                           start timer
                             Wait
                                           udt send(sndpkt[base])
                                          udt send(sndpkt[base+1])
rdt rcv(rcvpkt)
 && corrupt(rcvpkt)
                                          udt send(sndpkt[nextsegnum-1])
                         rdt rcv(rcvpkt) &&
                           notcorrupt(rcvpkt)
                         base = getacknum(rcvpkt)+1
                         If (base == nextsegnum)
                           stop timer
                          else
                            start timer
```

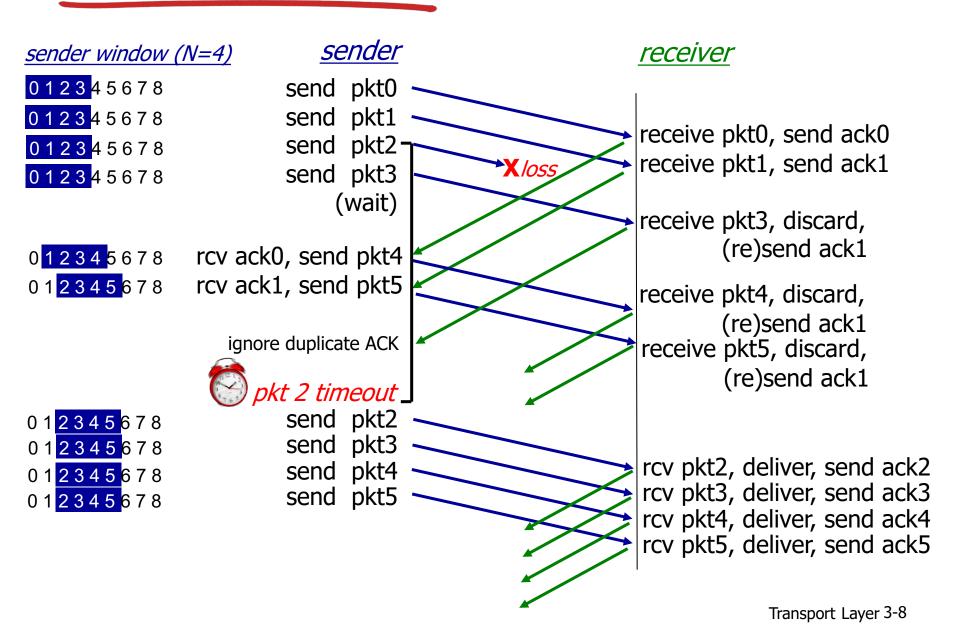
GBN: receiver extended FSM



ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq

- may generate duplicate ACKs
- need only remember expectedseqnum
- out-of-order pkt:
 - discard (don't buffer): no receiver buffering!
 - re-ACK pkt with highest in-order seq #

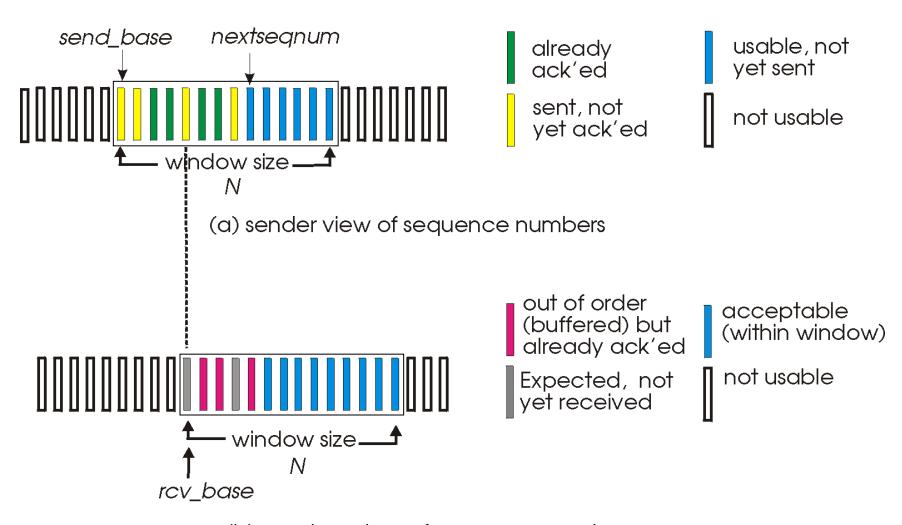
GBN in action



Selective repeat

- receiver individually acknowledges all correctly received pkts
 - buffers pkts, as needed, for eventual in-order delivery to upper layer
- sender only resends pkts for which ACK not received
 - sender timer for each unACKed pkt
- sender window
 - N consecutive seq #'s
 - limits seq #s of sent, unACKed pkts

Selective repeat: sender, receiver windows



(b) receiver view of sequence numbers

Selective repeat

sender

data from above:

if next available seq # in window, send pkt

timeout(n):

resend pkt n, restart timer

ACK(n) in [sendbase,sendbase+N]:

- mark pkt n as received
- if n smallest unACKed pkt, advance window base to next unACKed seq #

receiver

pkt n in [rcvbase, rcvbase+N-1]

- send ACK(n)
- out-of-order: buffer
- in-order: deliver (also deliver buffered, in-order pkts), advance window to next not-yet-received pkt

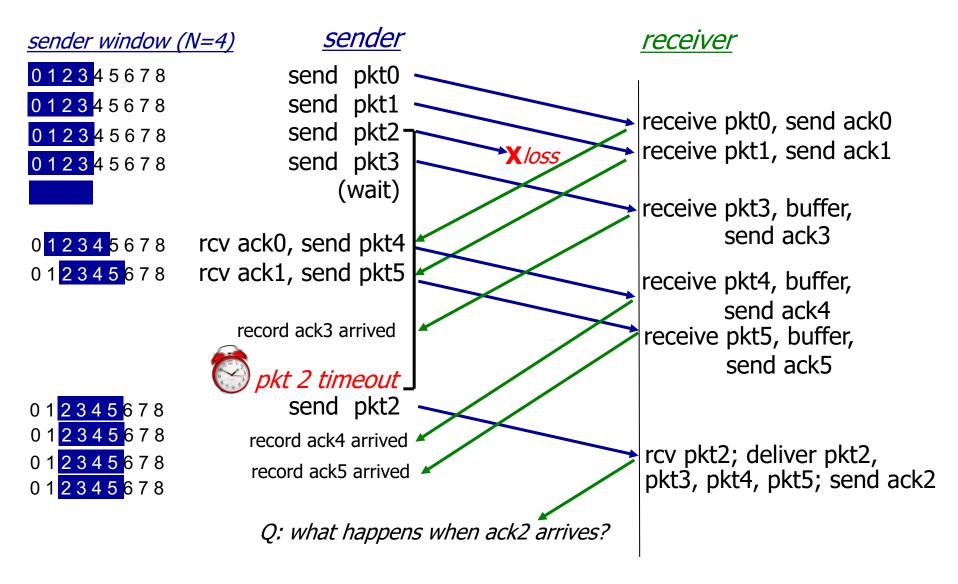
pkt n in [rcvbase-N,rcvbase-I]

ACK(n)

otherwise:

ignore

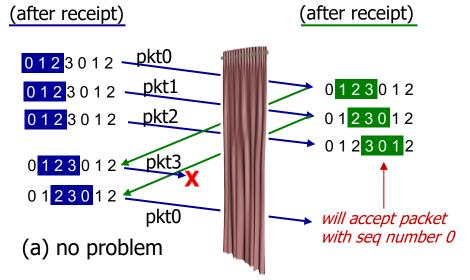
Selective repeat in action



Selective repeat: dilemma

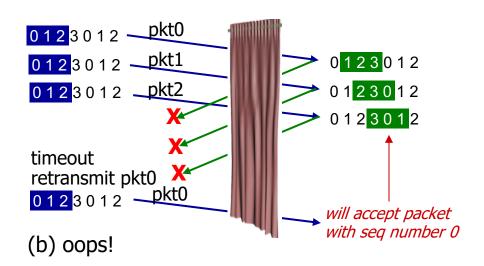
example:

- seq #' s: 0, 1, 2, 3
- window size=3
- receiver sees no difference in two scenarios!
- duplicate data accepted as new in (b)
- Q: what relationship between seq # size and window size to avoid problem in (b)?



sender window

receiver can't see sender side.
receiver behavior identical in both cases!
something's (very) wrong!



receiver window

Chapter 3 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

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

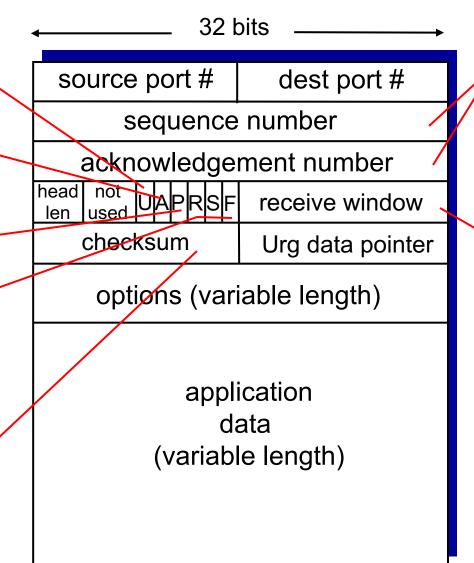
URG: urgent data (generally not used)

ACK: ACK # valid

PSH: push data now (generally not used)

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

Internet checksum (as in UDP)



counting
by bytes
of data
(not segments!)

bytes rcvr willing to accept

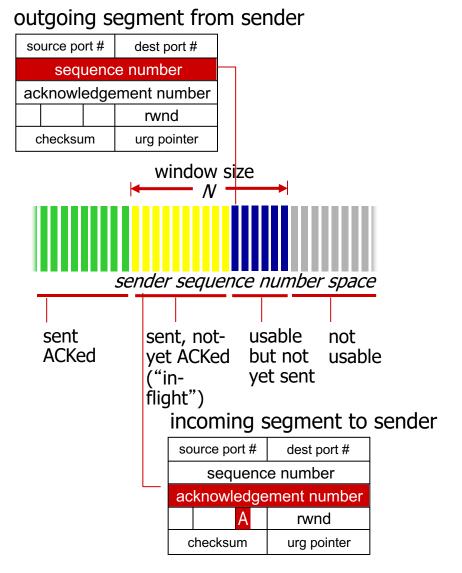
TCP seq. numbers, ACKs

sequence numbers:

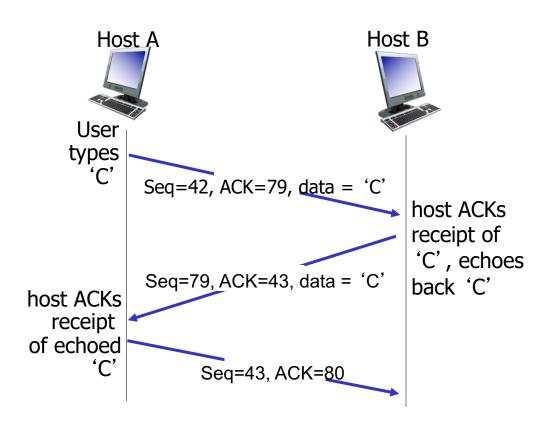
 byte stream "number" of first byte in segment's data

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

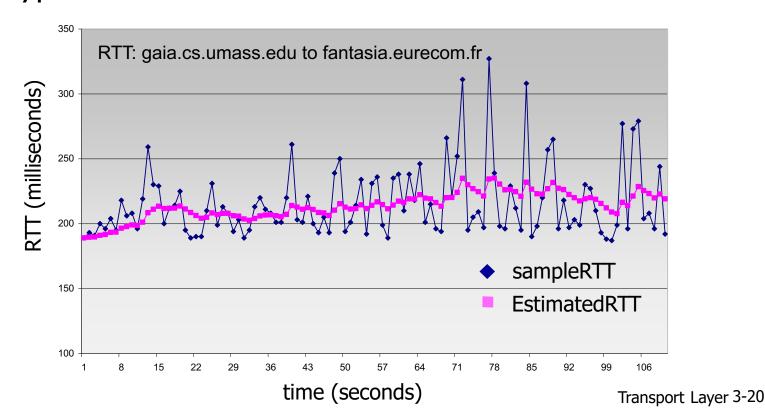
Q: how to estimate RTT?

- 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/

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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

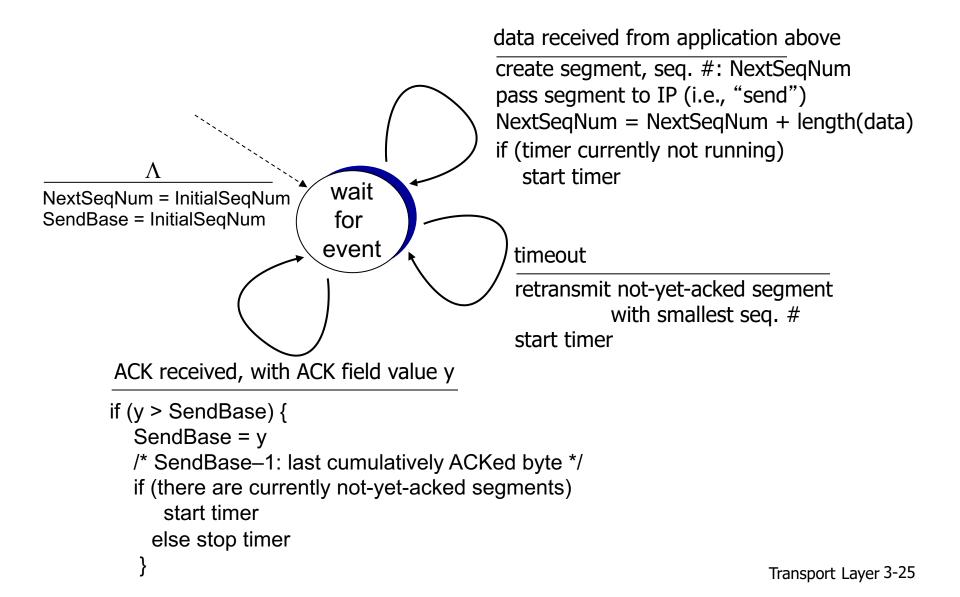
timeout:

- retransmit segment that caused timeout
- restart timer

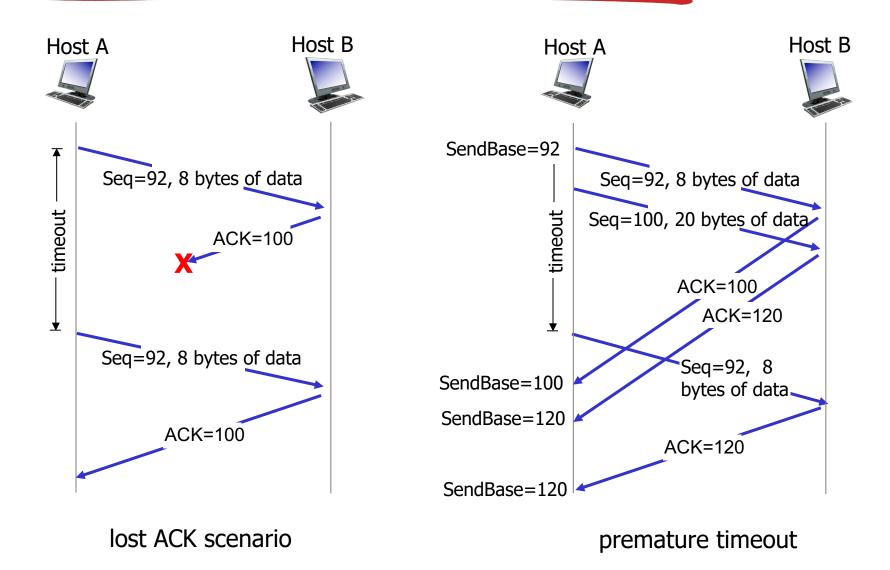
ack rcvd:

- if ack acknowledges previously unacked segments
 - update what is known to be ACKed
 - start timer if there are still unacked segments

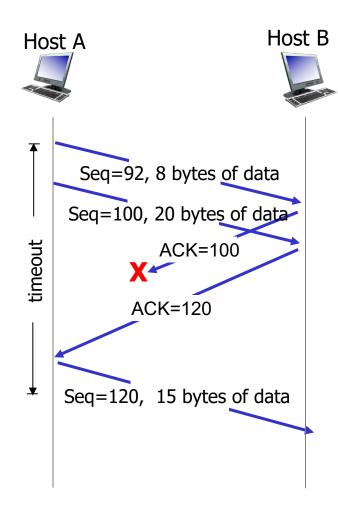
TCP sender (simplified)



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

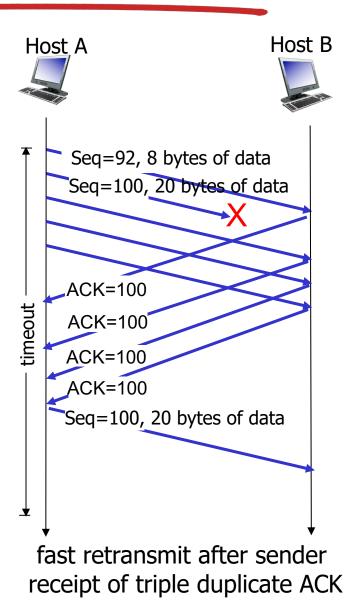
- time-out period often relatively long:
 - long delay before resending lost packet
- detect lost segments via duplicate ACKs.
 - sender often sends many segments backto-back
 - if segment is lost, there will likely be many duplicate ACKs.

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



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