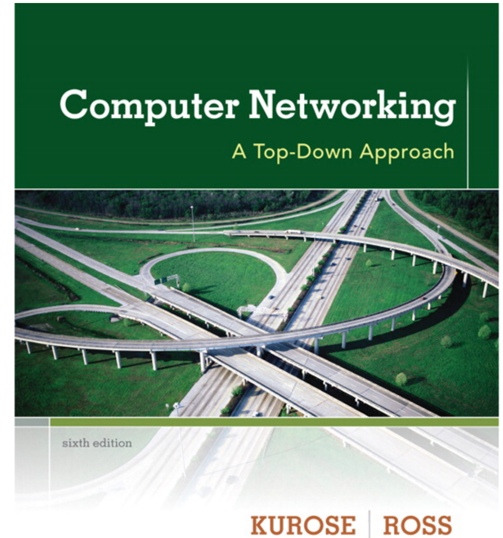
Chapter 3
Transport Layer



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a.a. 2019/2020

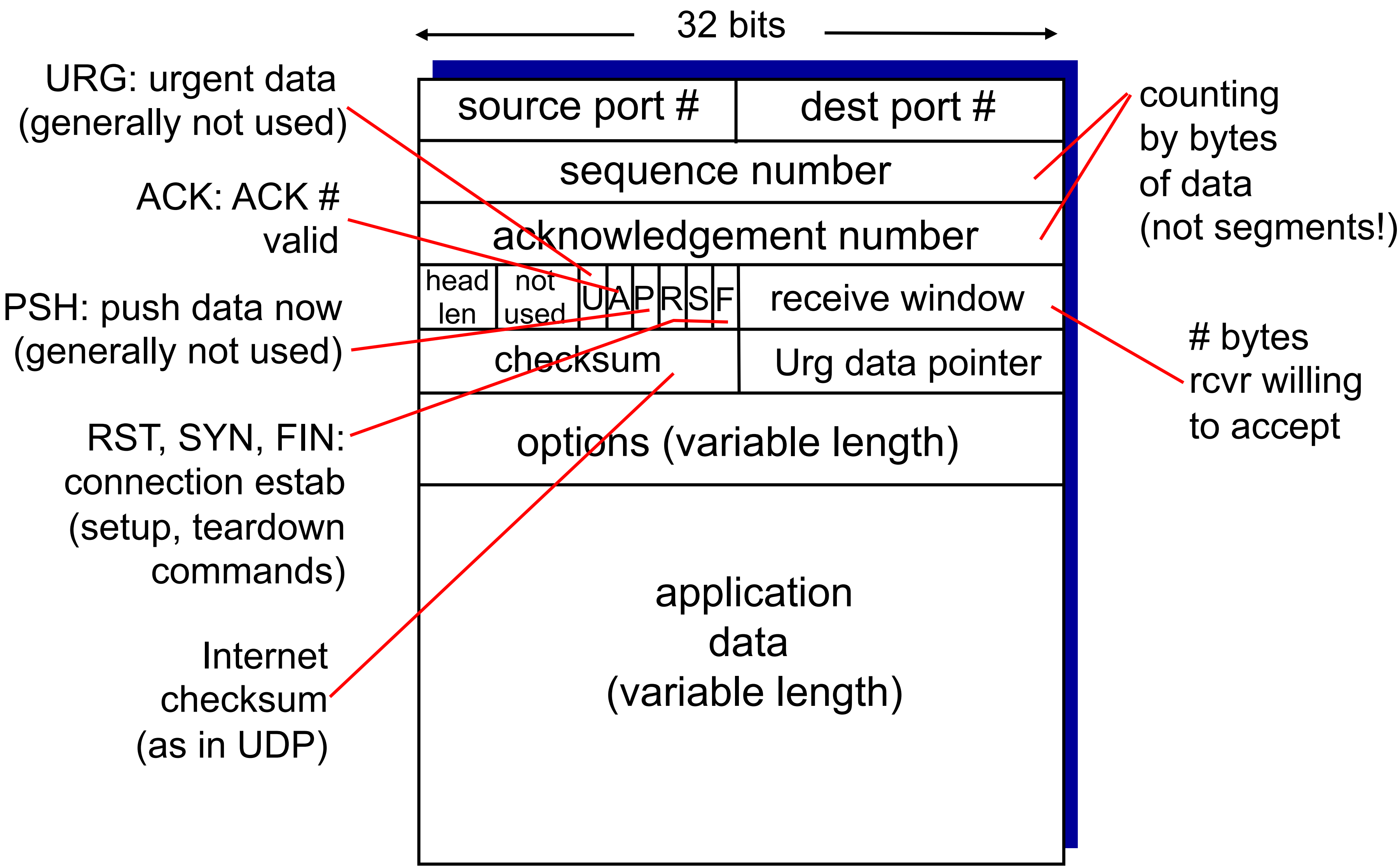
Computer Networking: A Top Down Approach
6th edition
Jim Kurose, Keith Ross
Addison-Wesley
March 2012

We thank for the support material Prof. Kurose-Ross

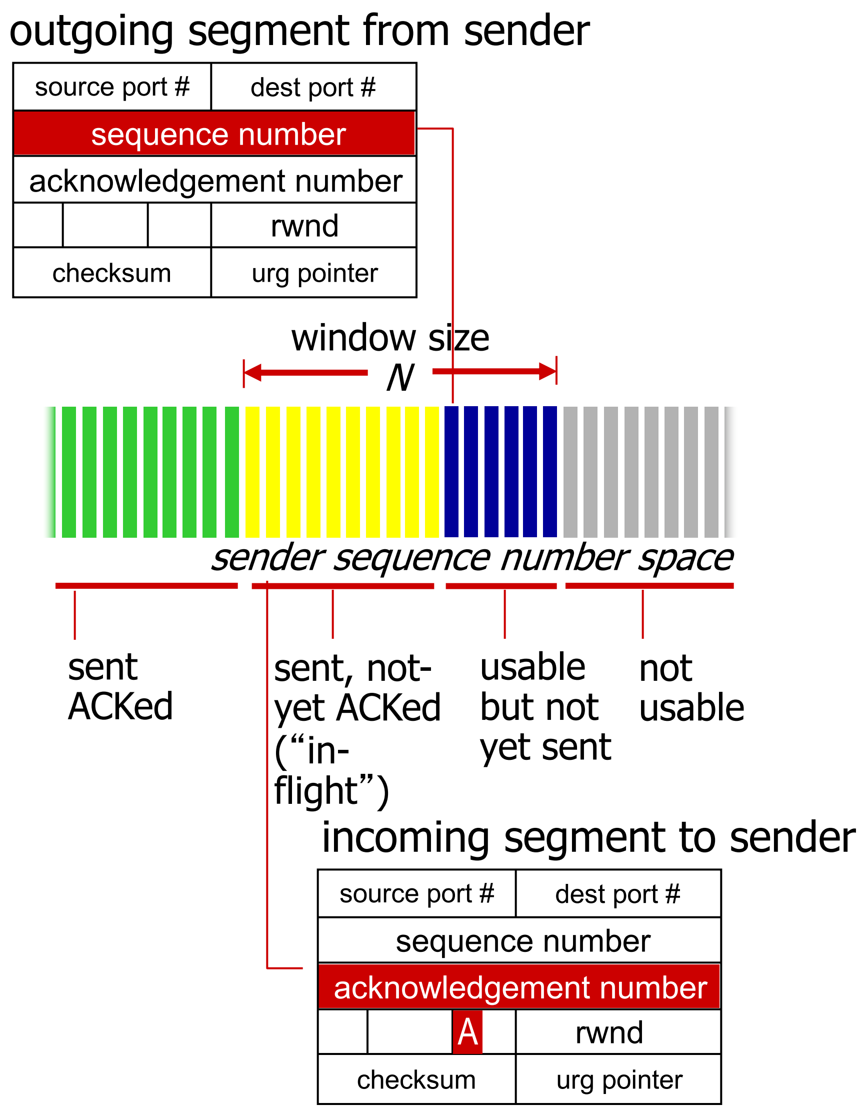
TCP: Overview RFCs: 793,1122,1323, 2018, 2581

- 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



TCP seq. numbers, ACKs



sequence numbers:

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

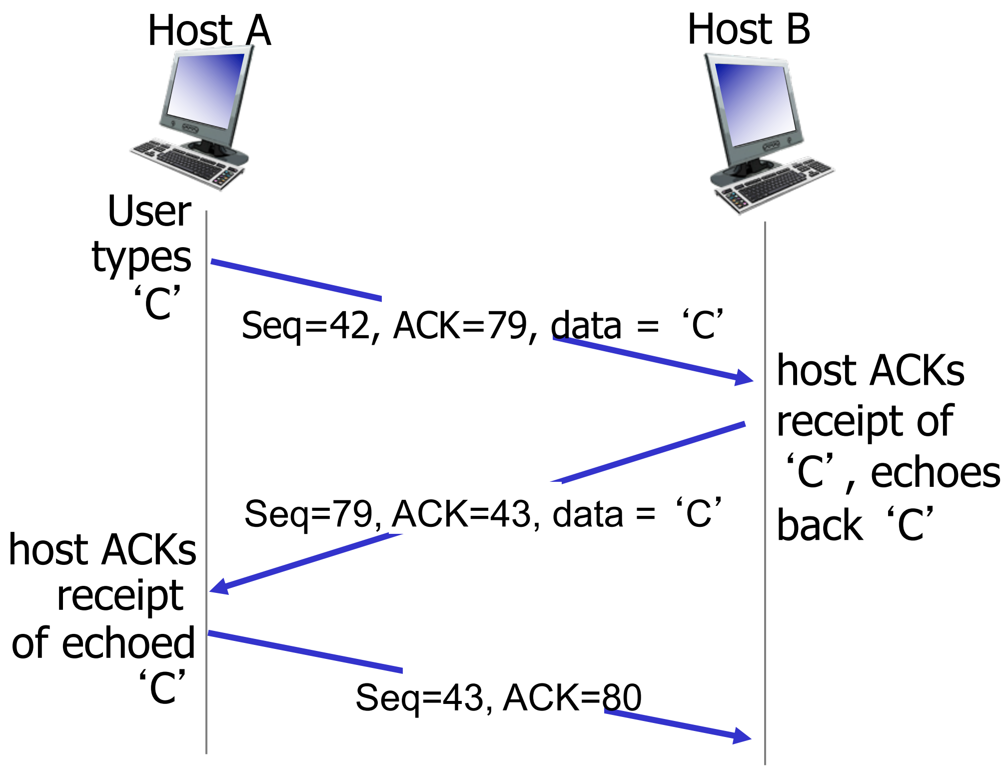
acknowledgements:

- 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

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

3.6 principles of congestion control

3.7 TCP congestion control

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)

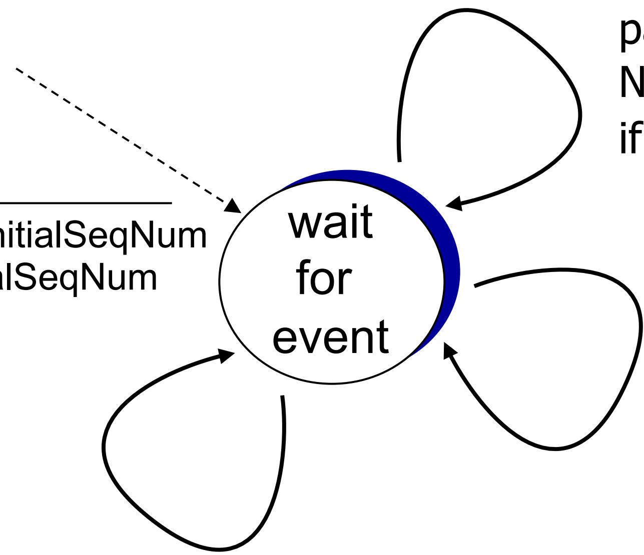
data received from application above

---

create segment, seq. #: NextSeqNum
pass segment to IP (i.e., “send”)
NextSeqNum = NextSeqNum + length(data)
if (timer currently not running)
    start timer

Λ

NextSeqNum = InitialSeqNum
SendBase = InitialSeqNum



timeout

---

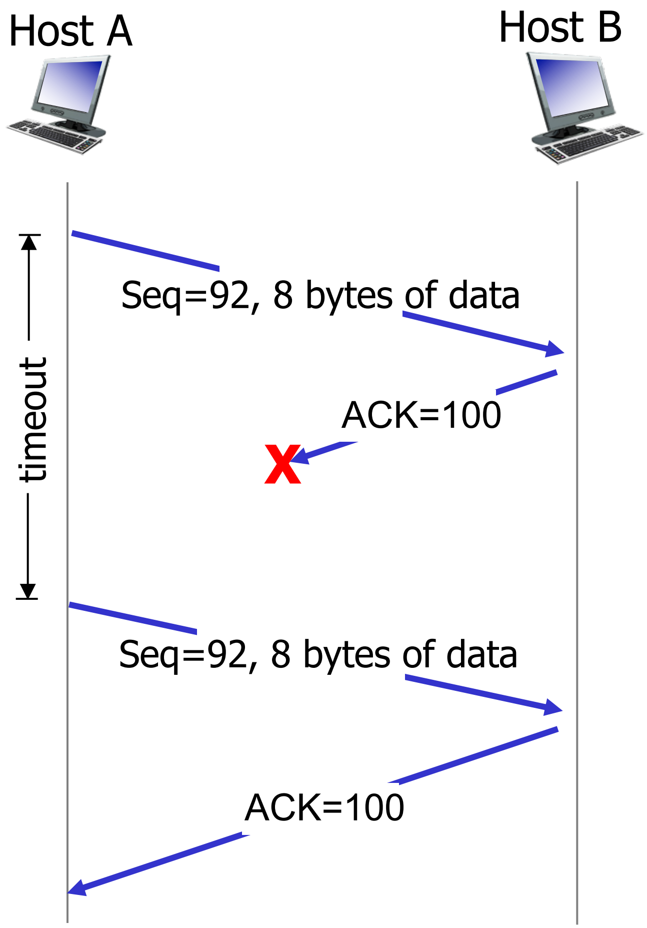
retransmit not-yet-acked segment
with smallest seq. #
start timer

ACK received, with ACK field value y

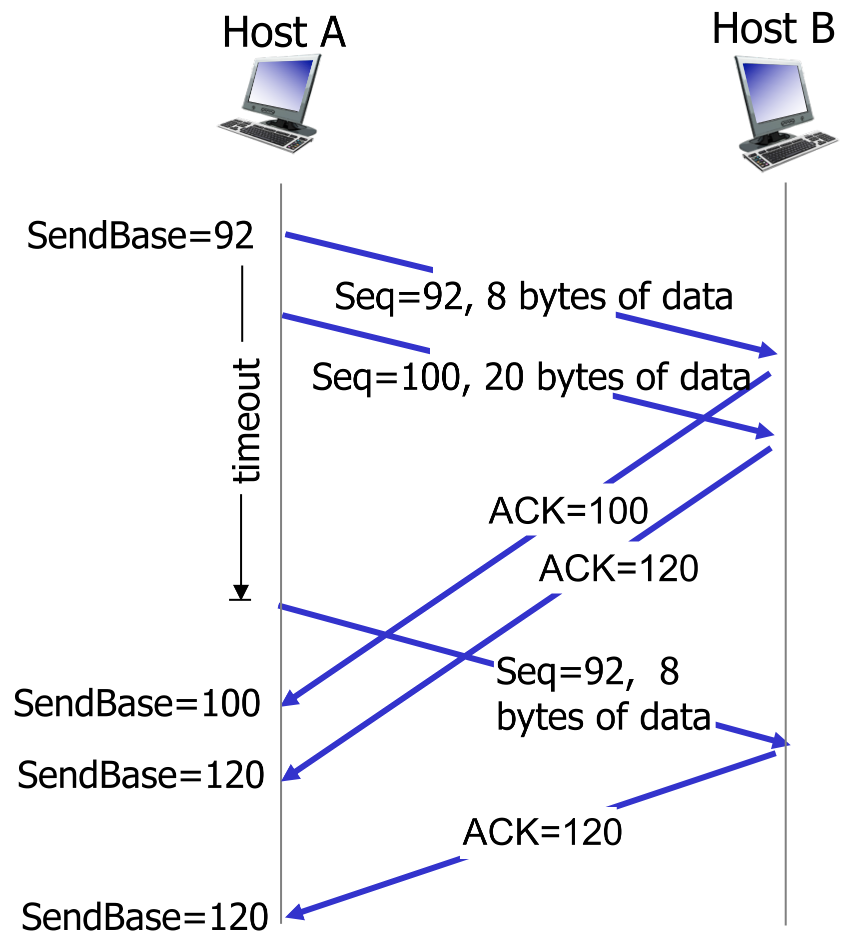
---

if (y > SendBase) {
    SendBase = y
    /* SendBase–1: last cumulatively ACKed byte */
    if (there are currently not-yet-acked segments)
        start timer
      else stop timer
 }

TCP: retransmission scenarios

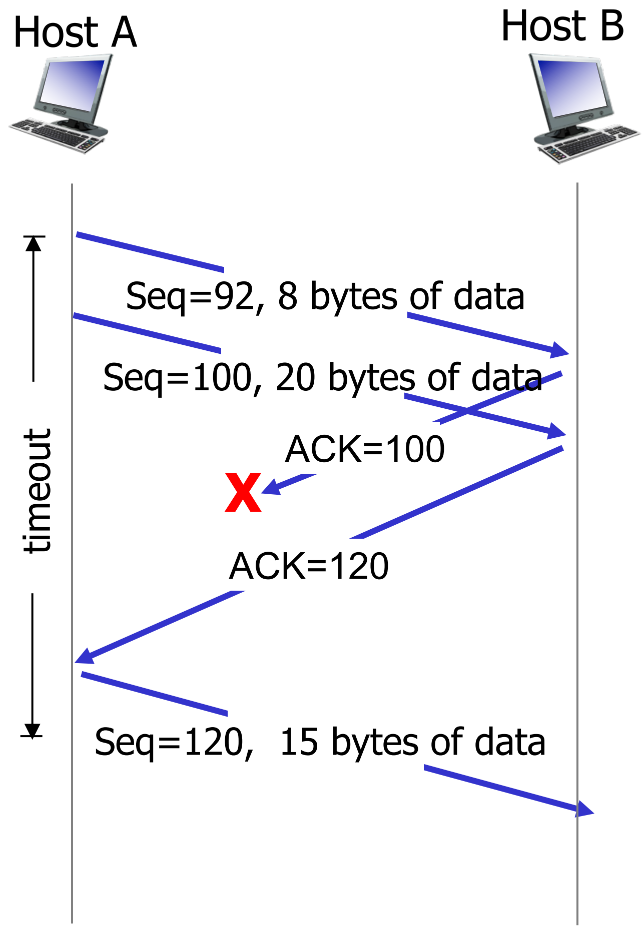


lost ACK scenario



premature timeout

TCP: retransmission scenarios



cumulative ACK

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

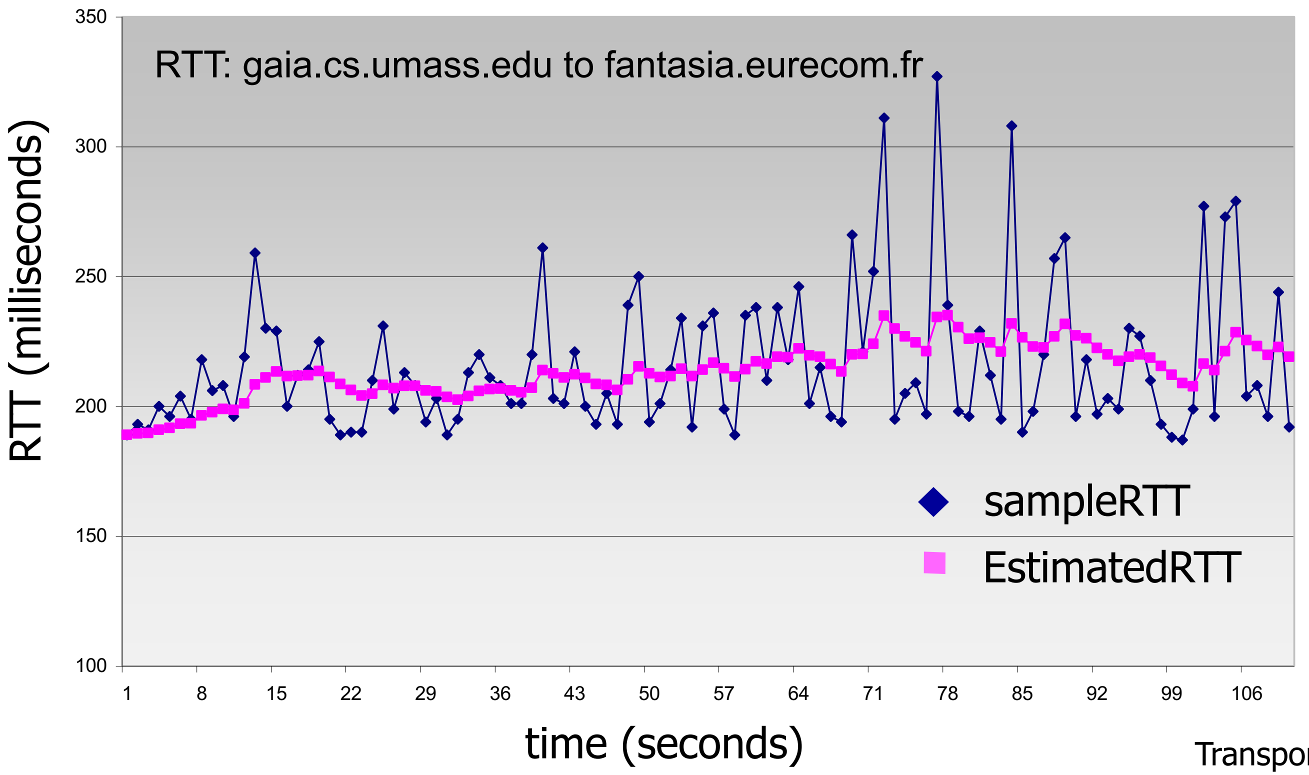
Q: how to estimate RTT?

- **`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- α)*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:

$$\texttt{DevRTT} = (1-\beta)*\texttt{DevRTT} + \beta*|\texttt{SampleRTT}-\texttt{EstimatedRTT}|$$

$$(\text{typically, } \beta = 0.25)$$

$$\texttt{TimeoutInterval} = \underbrace{\texttt{EstimatedRTT}}_{\text{estimated RTT}} + \underbrace{4*\texttt{DevRTT}}_{\text{"safety margin"}}$$

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

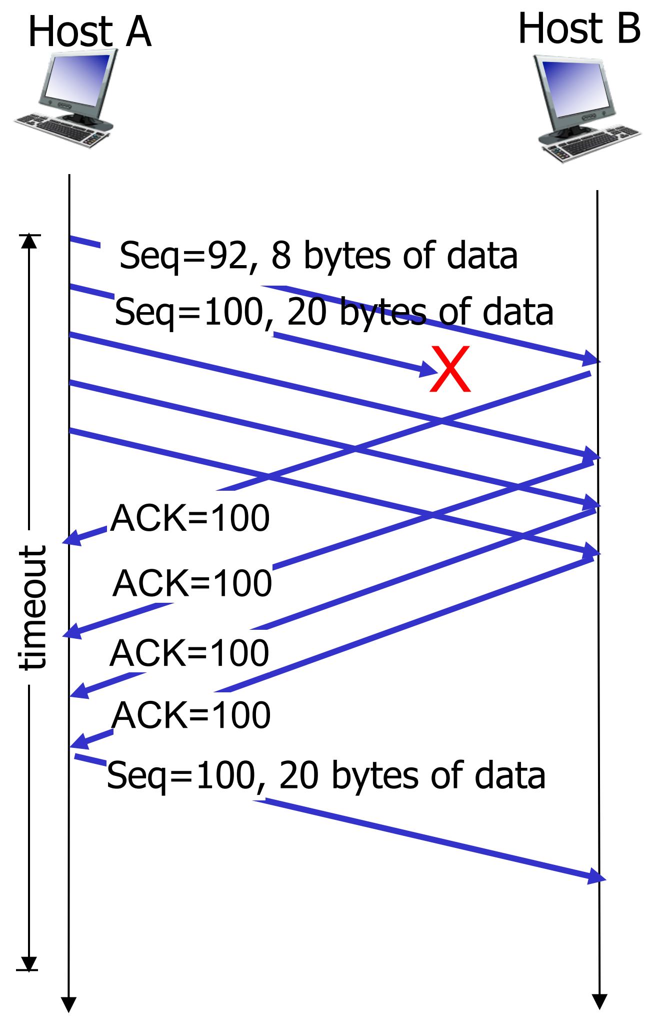
- time-out period often relatively long:
  - long delay before resending lost packet
- detect lost segments via duplicate ACKs.
  - sender often sends many segments back-to-back
  - if segment is lost, there will likely be many duplicate ACKs.

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



fast retransmit after sender receipt of triple duplicate ACK

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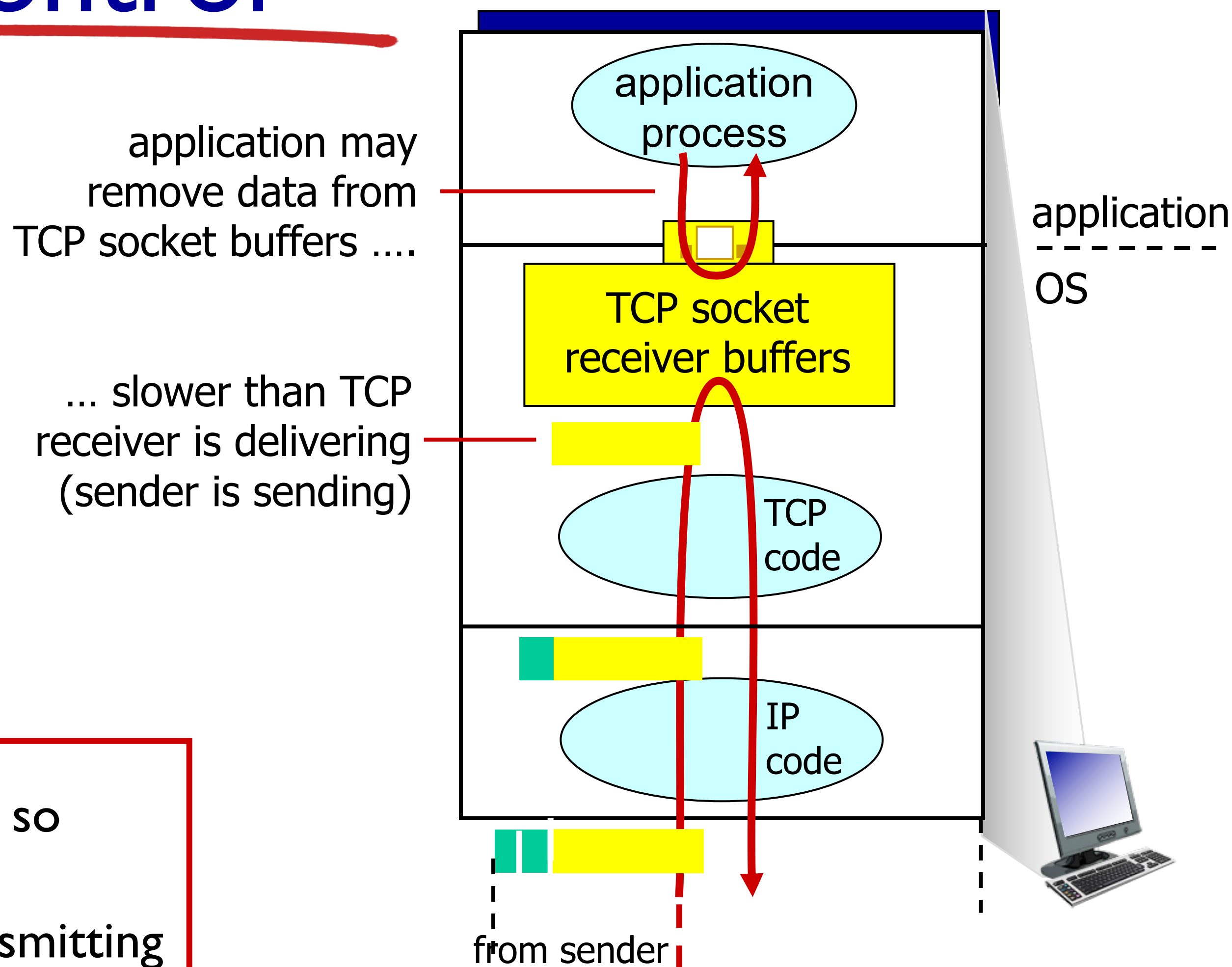
TCP flow control



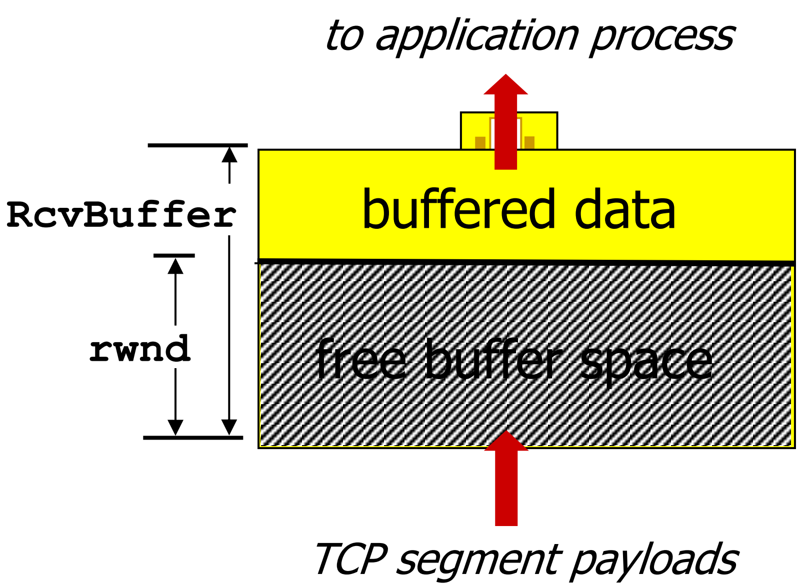
receiver protocol stack

flow control

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

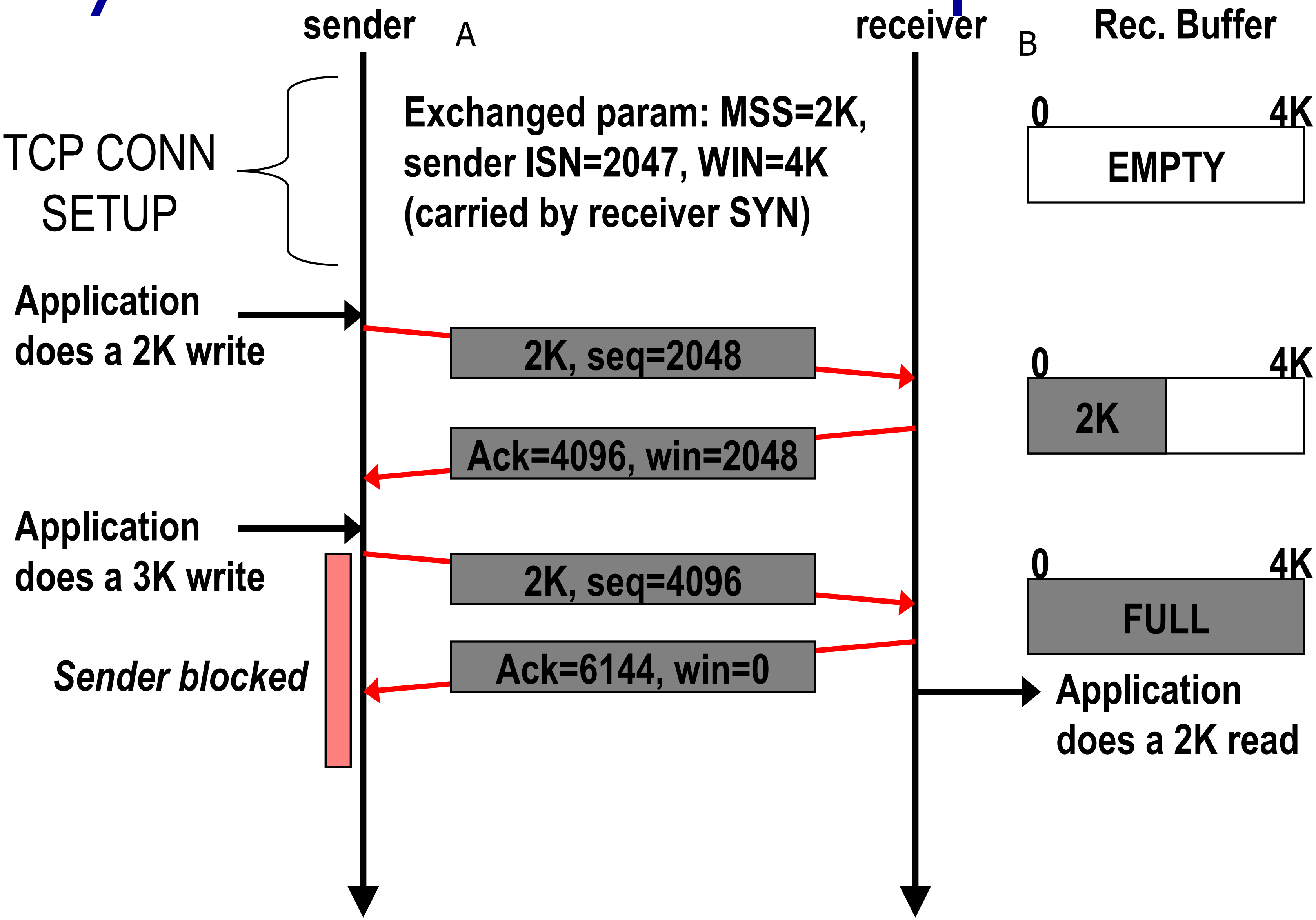
TCP flow control

- 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

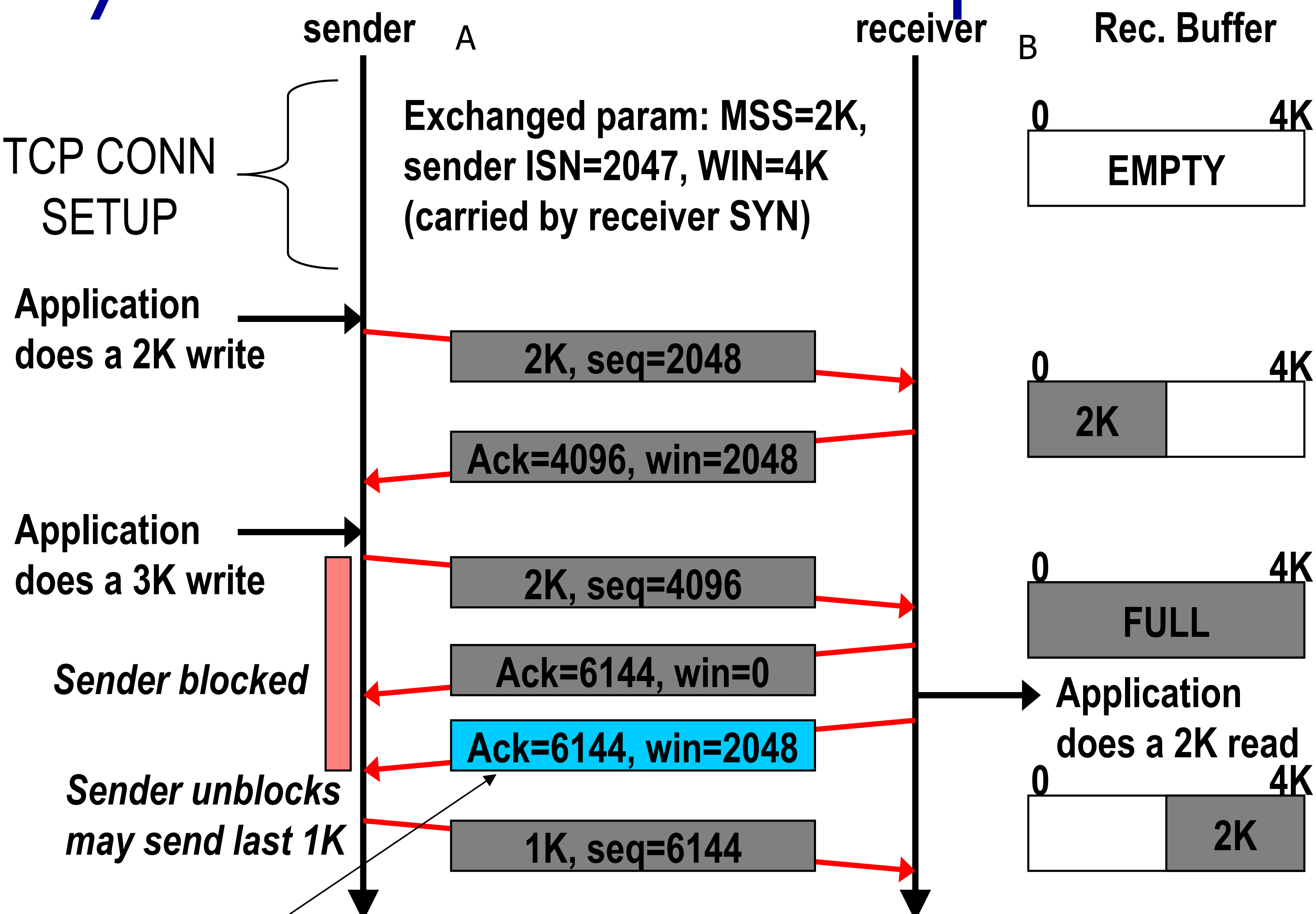


receiver-side buffering

Dynamic window - example



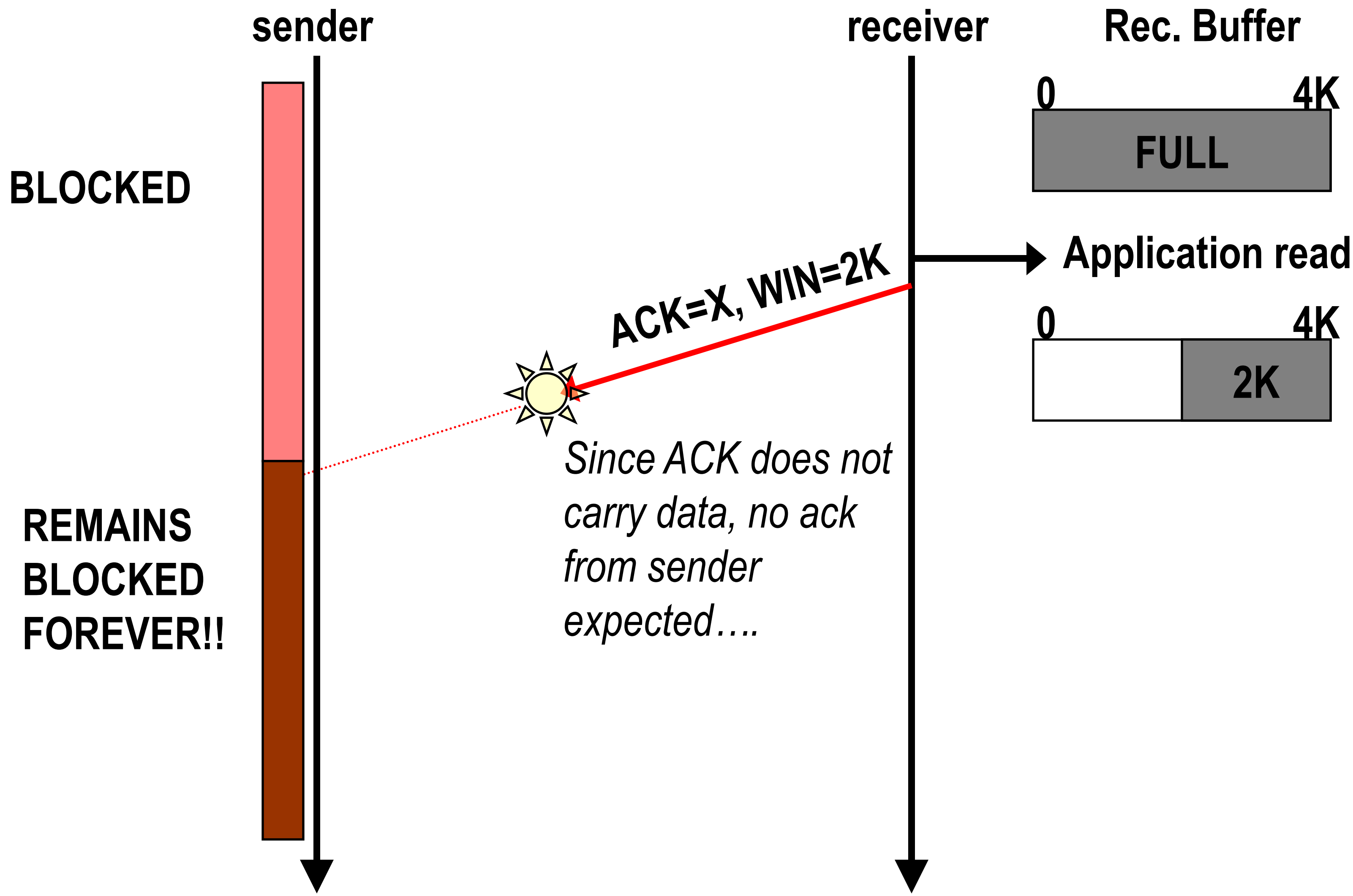
Dynamic window - example



Piggybacked in a packet sent from B to A

Window thus source rate limited by reading speed and buffer size at the receiver

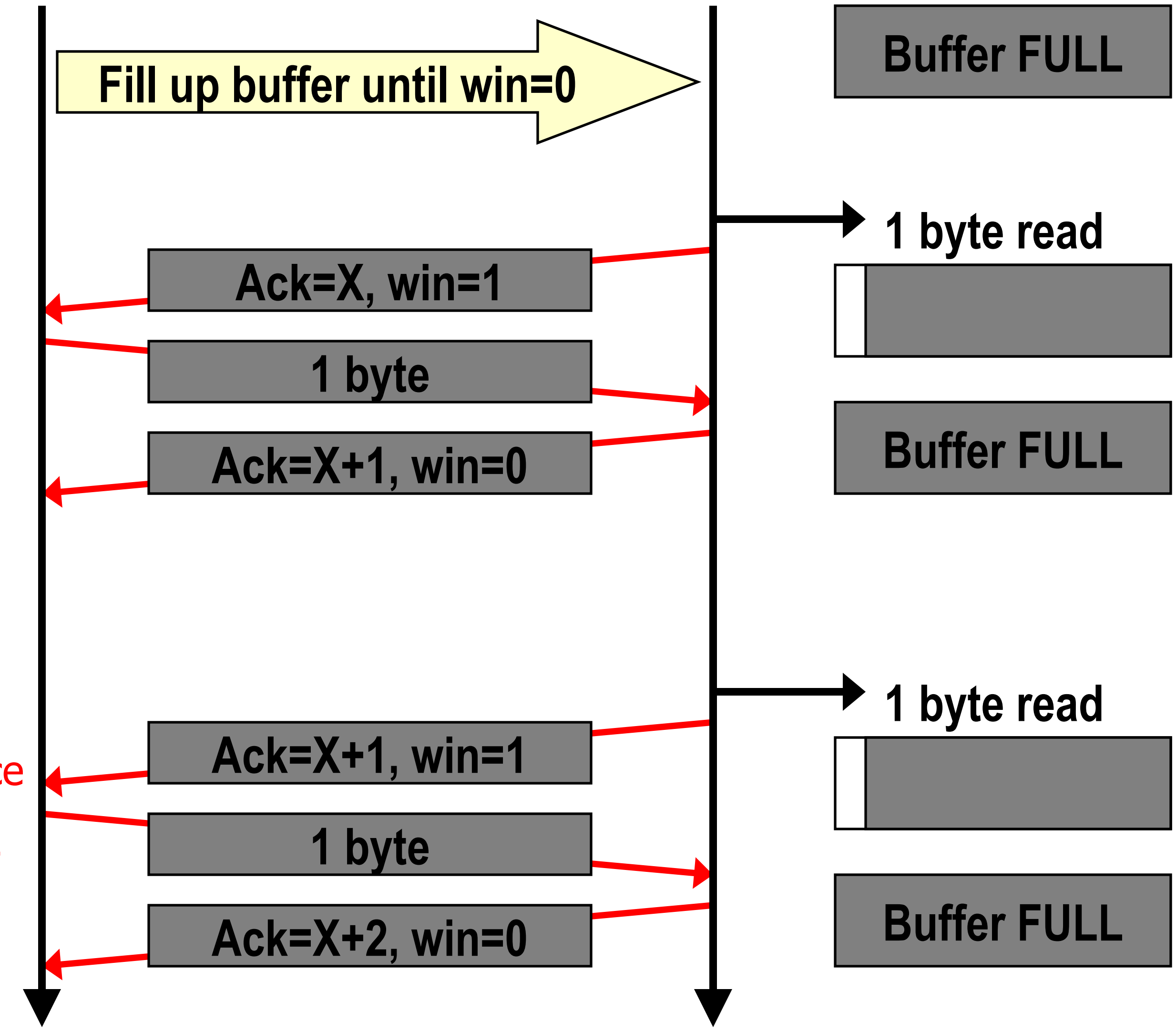
Blocked sender deadlock problem



Solution: Persist timer

- When win=0 (blocked sender), sender starts a "persist" timer
    - Initially 500ms (but depends on implementation)
- When persist timer elapses AND no segment received during this time, sender transmits "probe"
    - Probe = 1byte segment; makes receiver reannounce next byte expected and window size
        - this feature necessary to break deadlock
        - if receiver was still full, rejects byte
        - otherwise acks byte and sends back actual win
- Persist time management (exponential backoff):
    - Doubles every time no response is received
    - Maximum = 60s

The silly window syndrome



Network loaded with tinygrams (40bytes header + 1 payload!!)

Forever!

Anche se il ricevitore e' veloce
A passare i dati al livello
applicativo inviare segmenti
piccoli in un bulk di dati
ha questo effetto

Silly window solution

- Problem discovered by David Clark (MIT), 1982
- easily solved, by preventing receiver to send a window update for 1 byte
- rule: send window update when:
  - receiver buffer can handle a whole MSS

    or
  - half received buffer has emptied (if smaller than MSS)
- sender also may apply rule
  - by waiting for sending data when win low