COMP9121: Design of Networks and Distributed Systems

Week 8: Transport Layer 2

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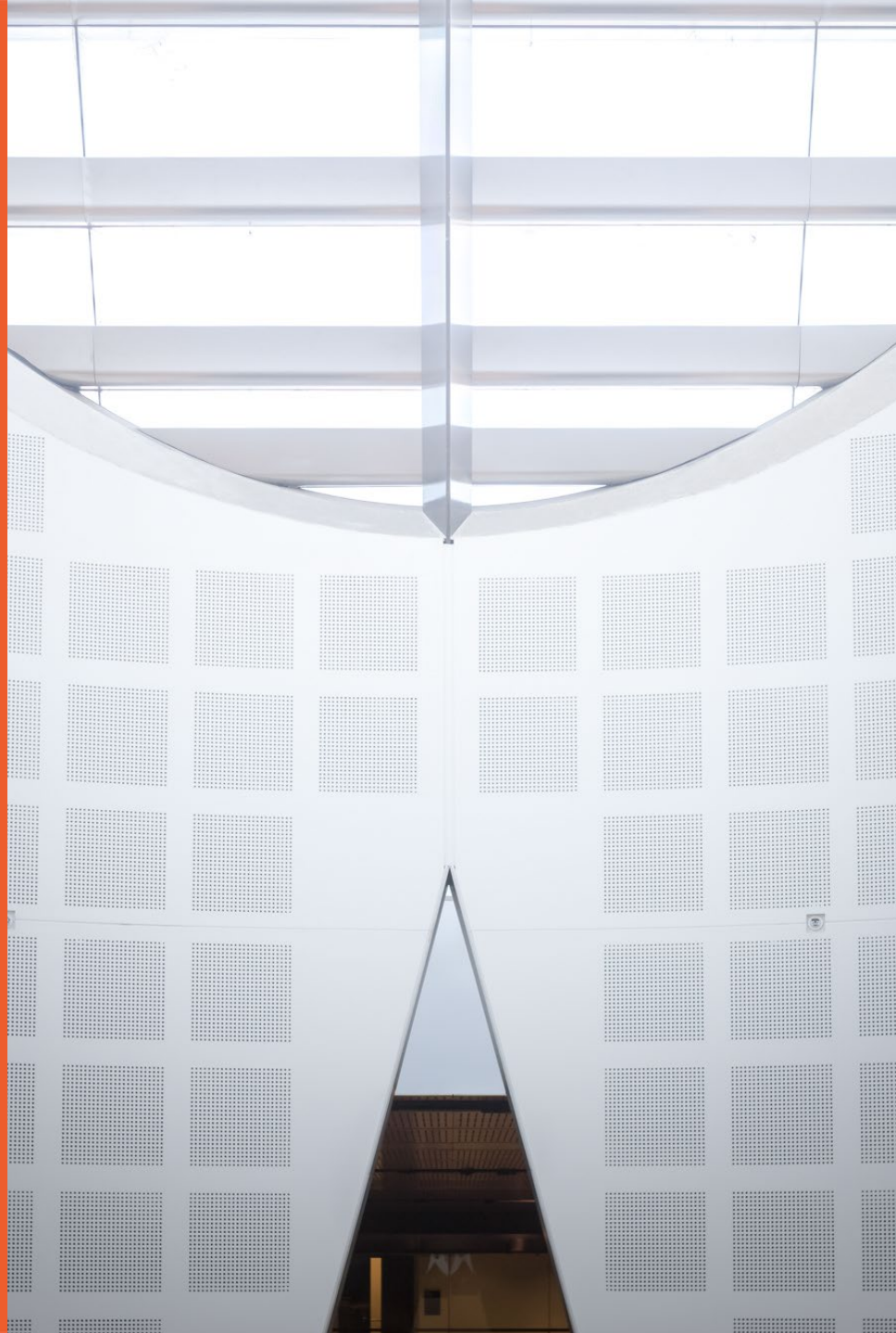


rdt3.0: channels with errors and loss

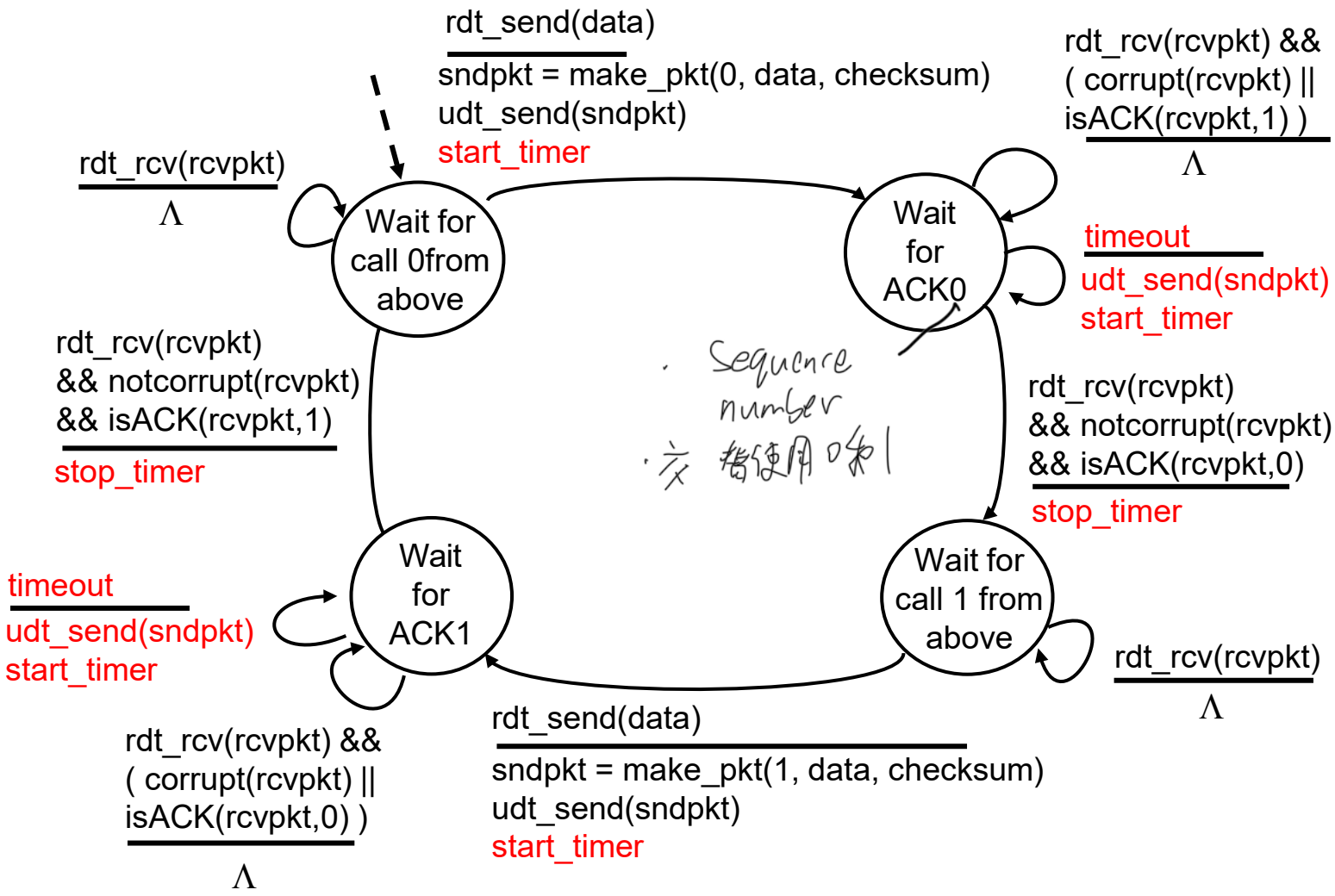
new assumption: underlying channel can also lose packets (data, ACKs)

- checksum, seq. #, ACKs, retransmissions will be of help … but not enough

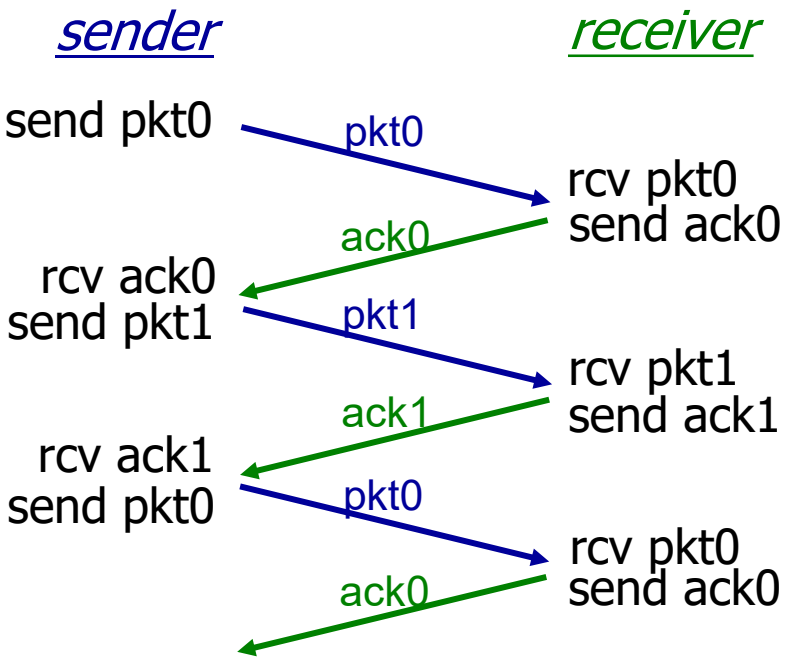
approach: sender waits "reasonable" amount of time for ACK

- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but seq. #'s already handles this
  - receiver must specify seq # of pkt being ACKed
- requires countdown timer

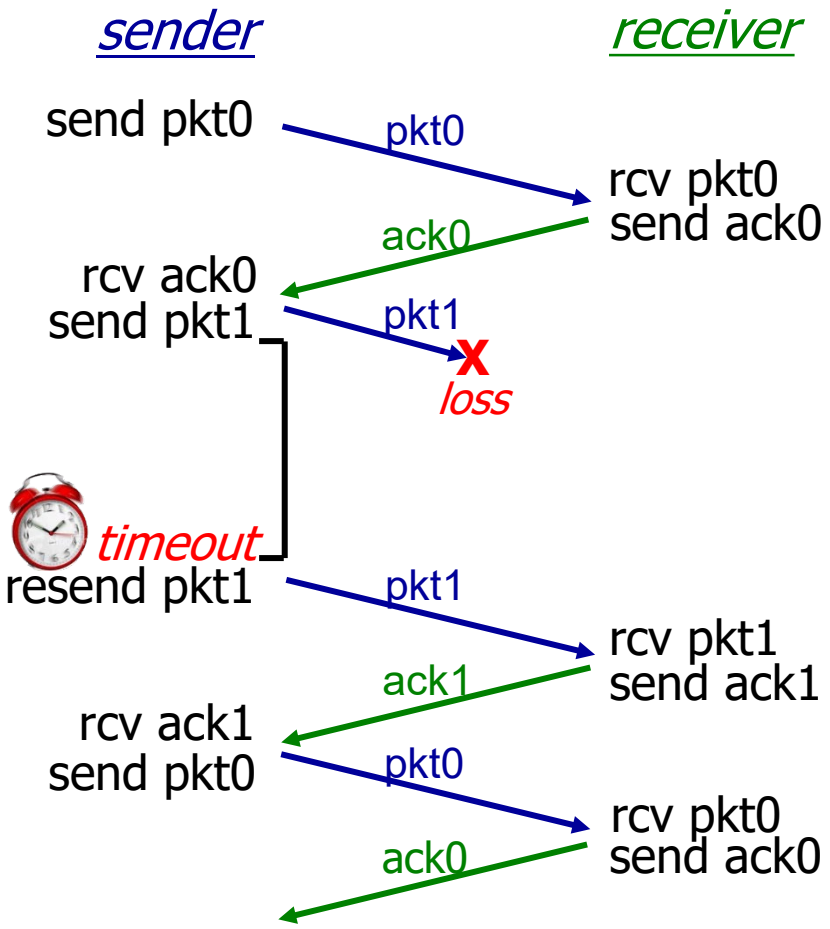
rdt3.0 sender



rdt3.0 in action

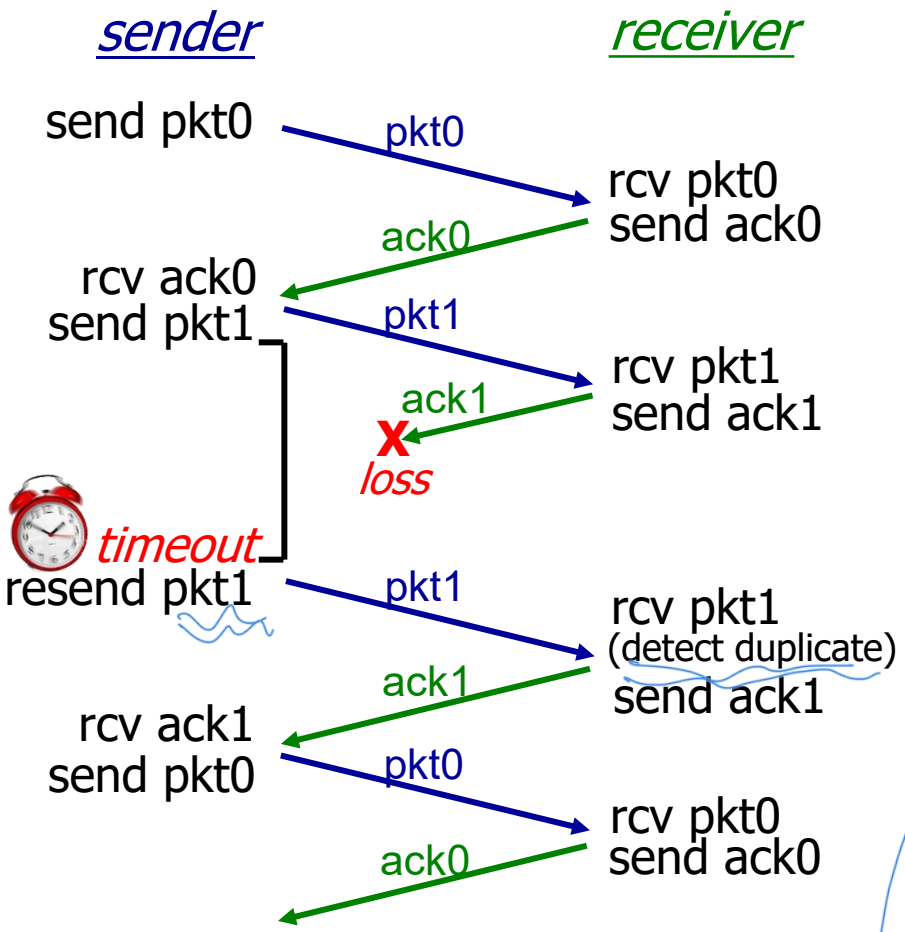


(a) no loss

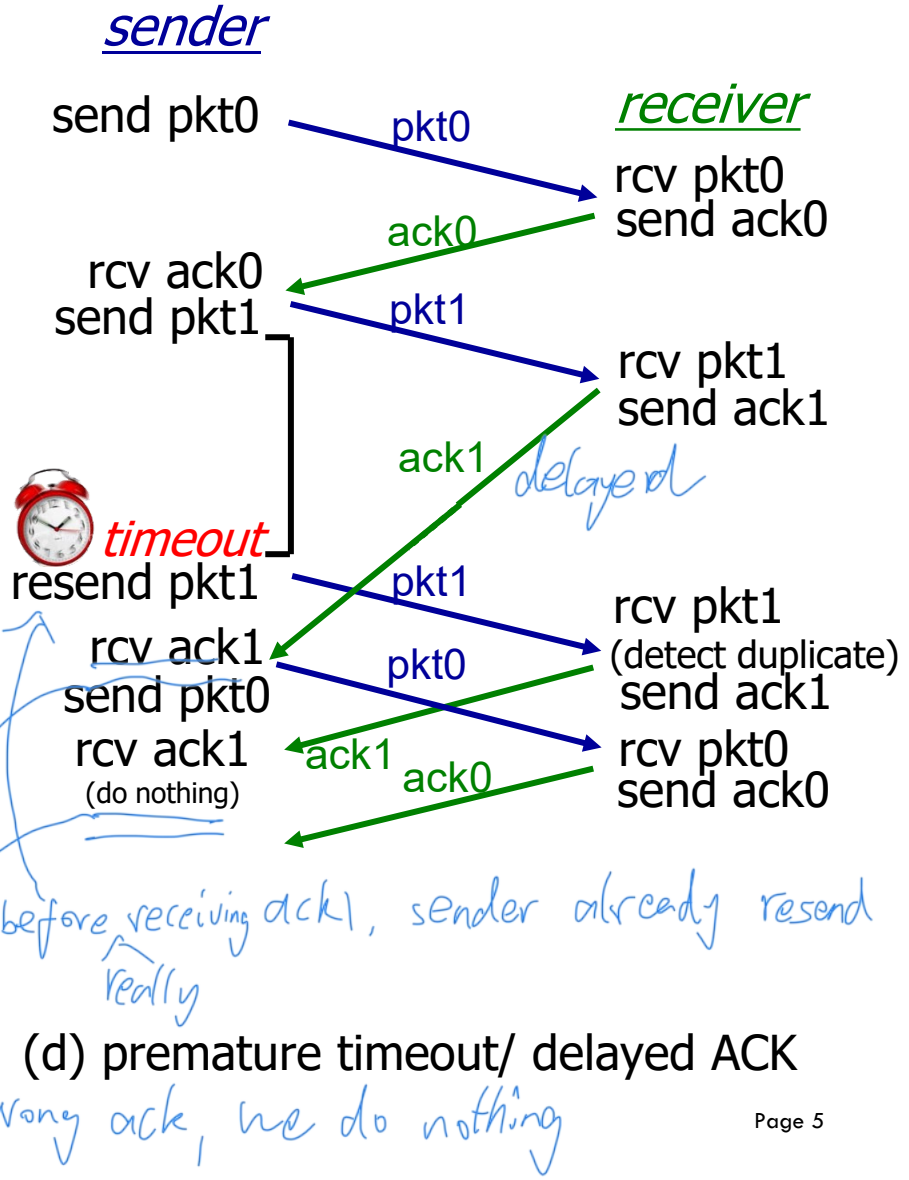


(b) packet loss

rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

Performance of rdt3.0

- rdt3.0 is correct, but performance stinks
- e.g.: 1 Gbps link, 15 ms prop. delay, 8000 bit packet:

$$D_{trans} = \frac{L}{R} = \frac{8000\ bits}{10^9\ bits/sec} = 8\ microsecs$$

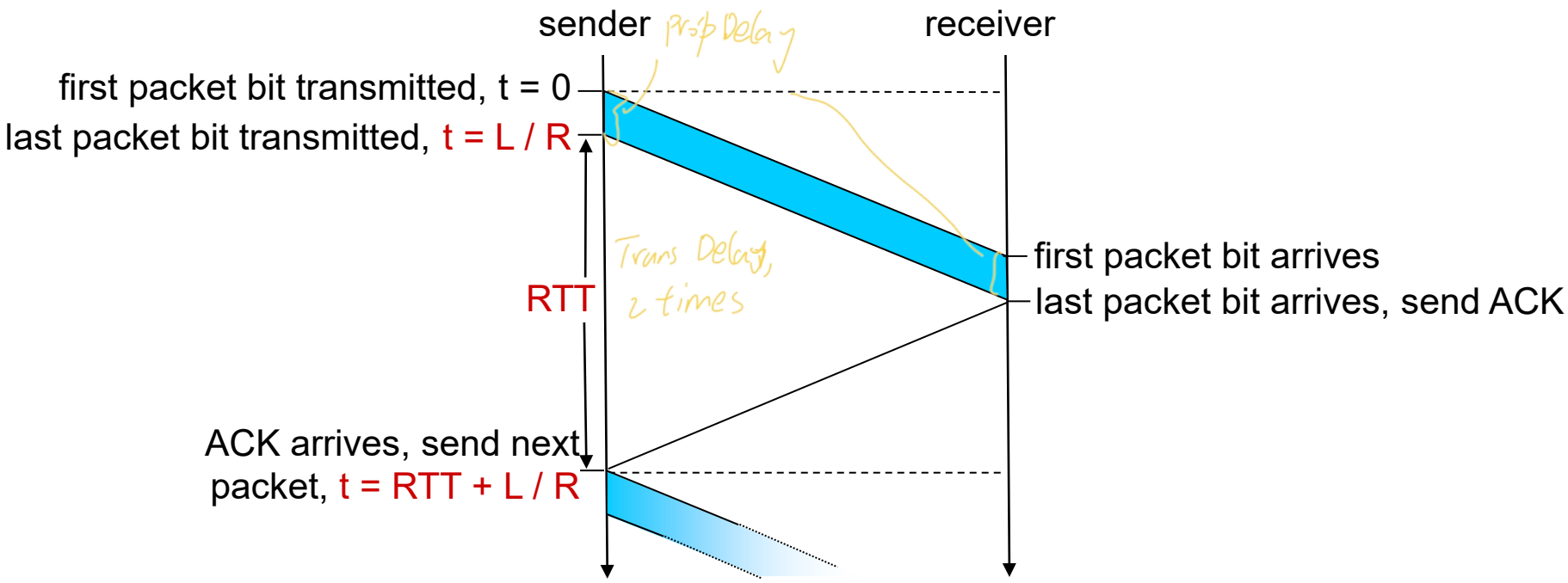
- U_{sender}: *utilization* – fraction of time sender busy sending

$$U_{sender} = \frac{L/R}{RTT + L/R} = \frac{.008}{30.008} = 0.00027$$

- if RTT=30 msec, 1KB pkt every 30 msec: 33kB/sec thruput over 1 Gbps link

❖ network protocol limits use of physical resources!

rdt3.0: stop-and-wait operation

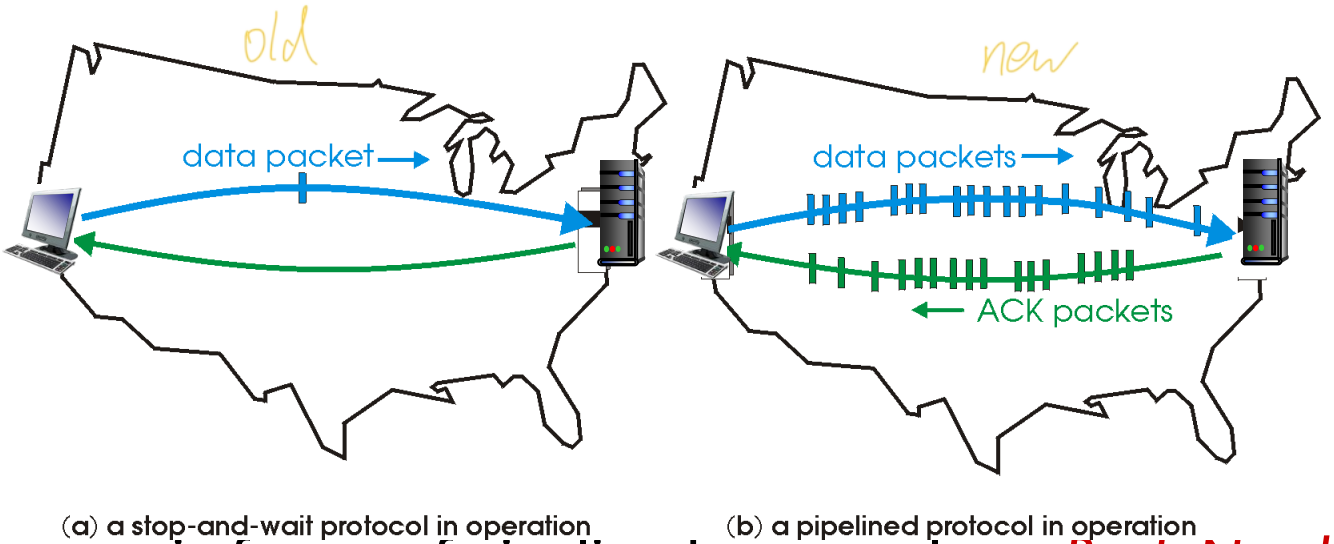


$$U_{sender} = \frac{L/R}{RTT + L/R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols

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

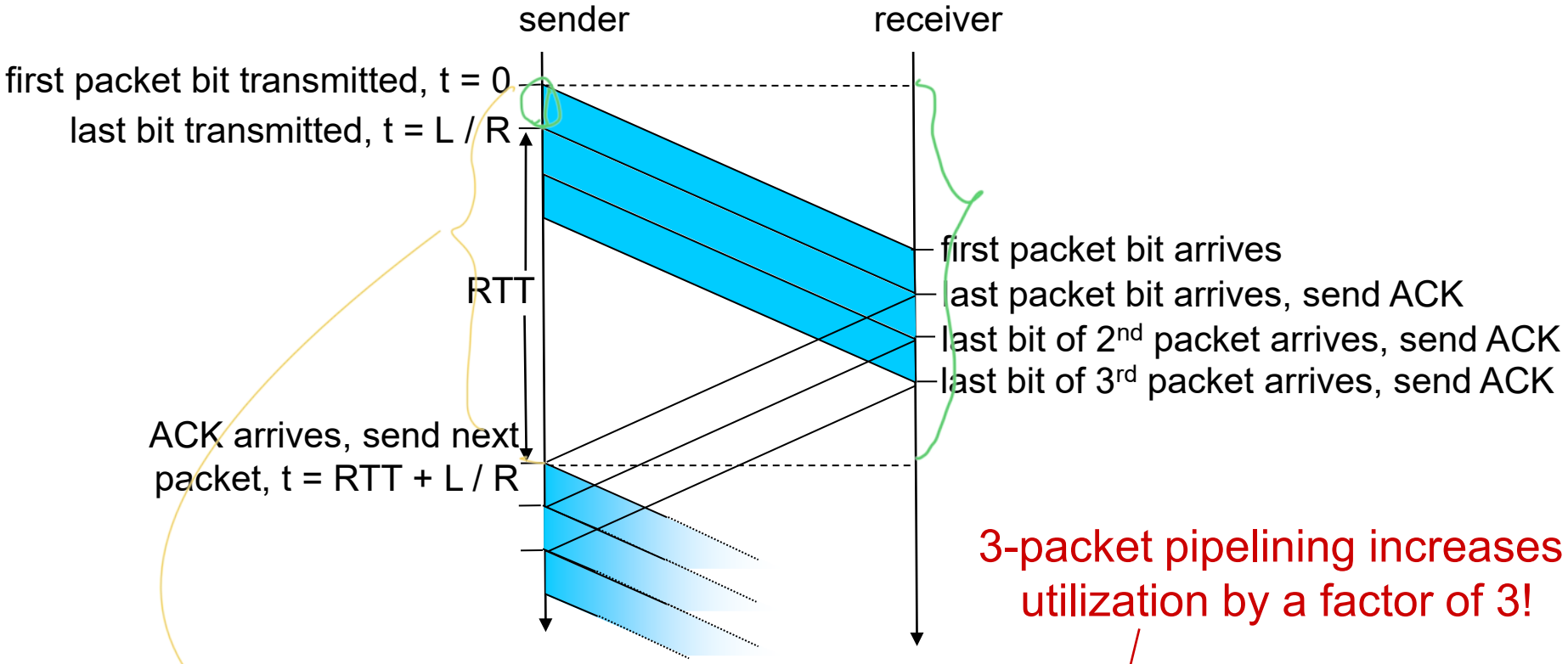
- range of sequence numbers must be increased
- buffering at sender and/or receiver



(a) a stop-and-wait protocol in operation (b) a pipelined protocol in operation

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

Pipelining: increased utilization



3-packet pipelining increases utilization by a factor of 3!

$$U_{sender} = \frac{3L/R}{RTT + L/R} = \frac{.0024}{30.008} = 0.00081$$

为什么乘3

3个pipe比存先高效3倍

蓝色部分 sender在工作，白色的时候没有

Pipelined protocols: overview

Go-back-N:

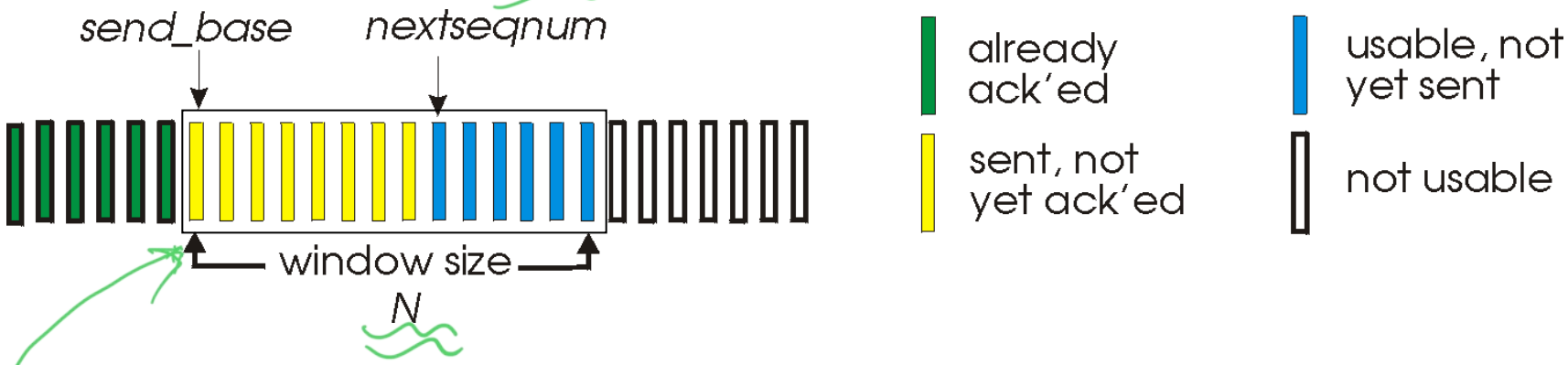
- sender can have up to N unacked packets in pipeline
- receiver only sends *cumulative ack*
  - does not ack packet if there is a gap
- sender has timer for oldest unacked packet
  - when timer expires, retransmit *all* unacked packets

Selective Repeat:

- sender can have up to N unacked packets in pipeline
- receiver 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

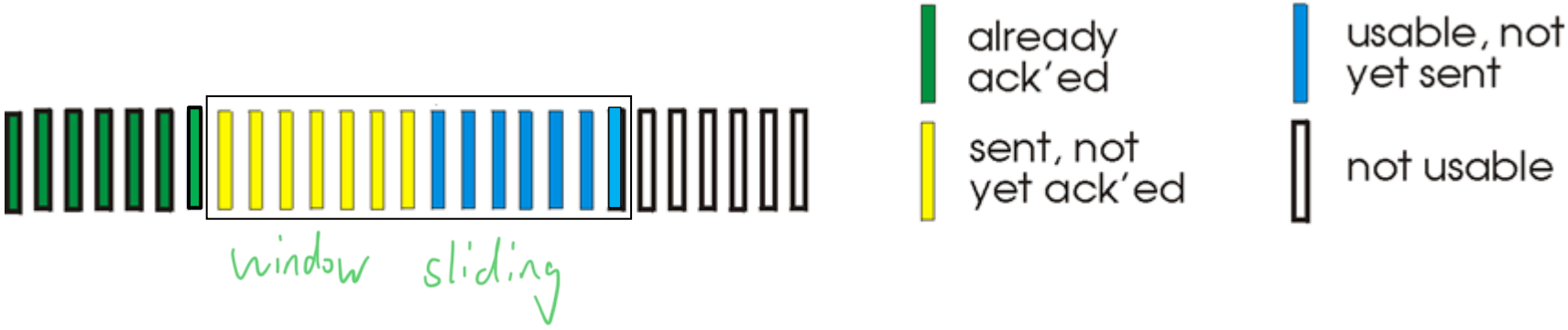
– "window" of up to N, consecutive unacked pkts allowed



- 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

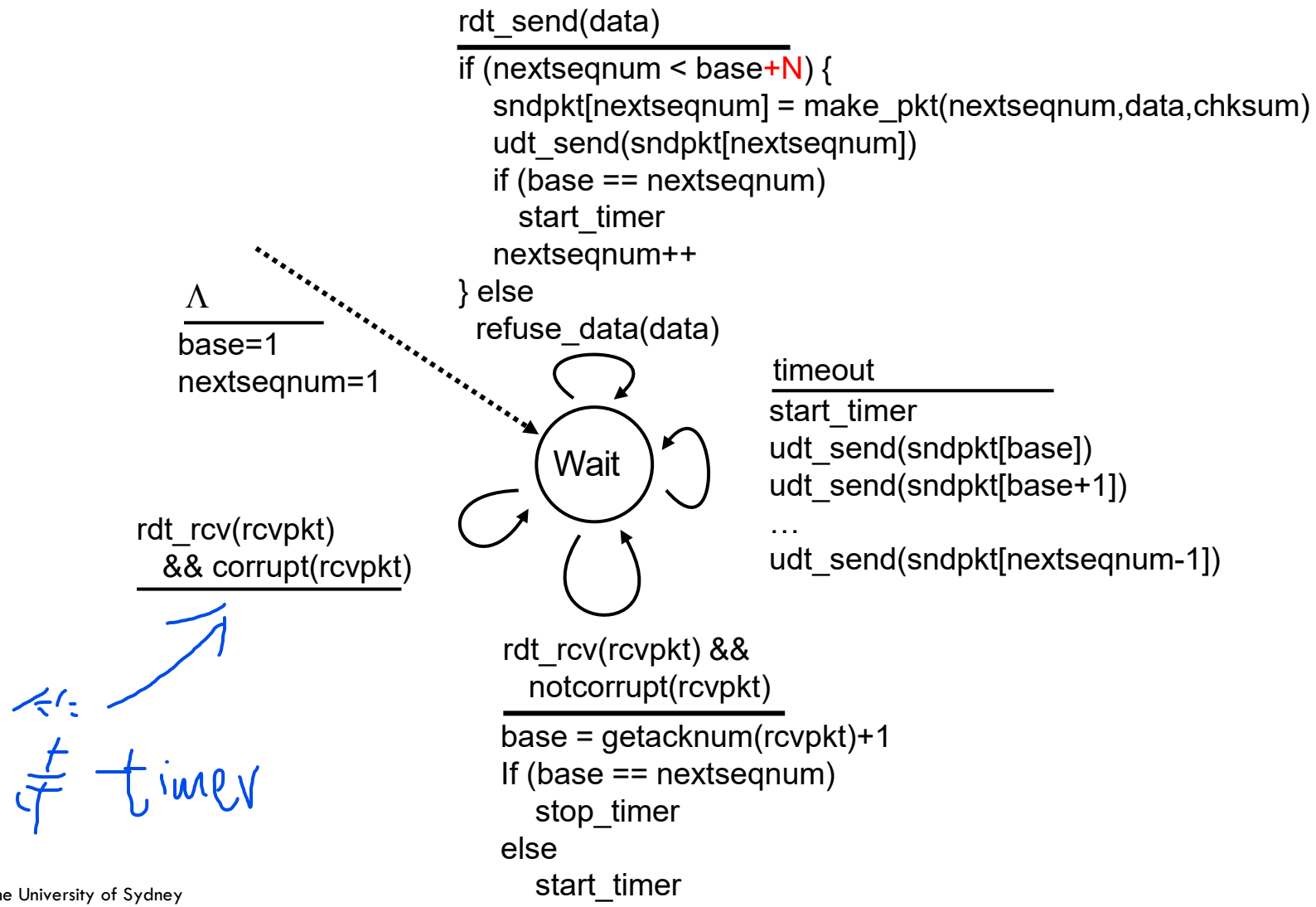
Go-Back-N: sender

– "window" of up to N, consecutive unacked pkts allowed

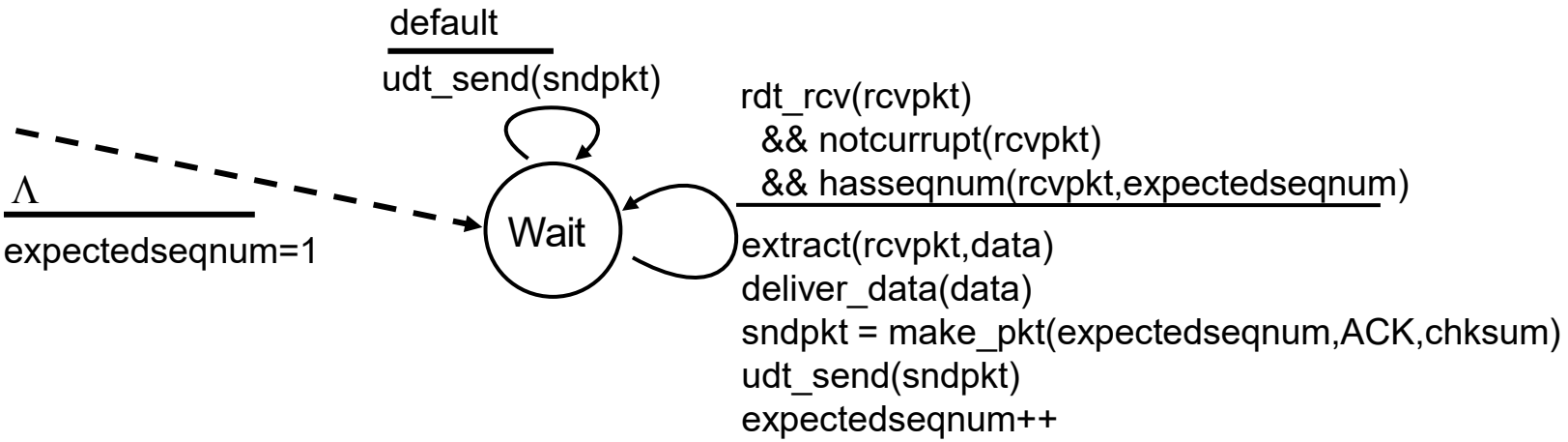


- ACK(n): ACKs all pkts up to, including seq # n - *"cumulative ACK"*
  - may receive duplicate ACKs (see receiver)
- timer for oldest in-flight pkt
- *timeout(n):* retransmit packet n and all higher seq # pkts in window

GBN: sender extended FSM



GBN: receiver extended FSM



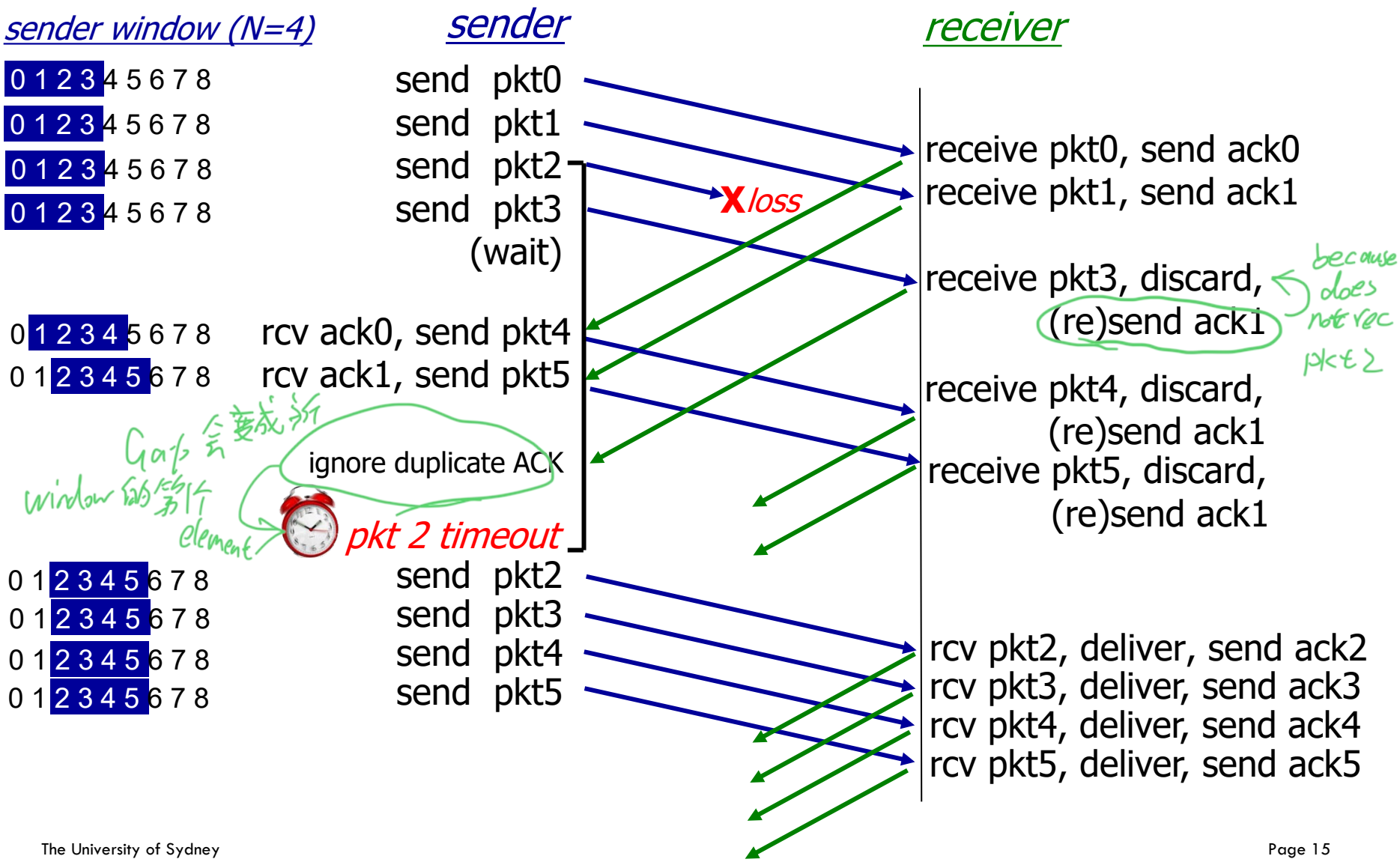
ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- 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



Method 2

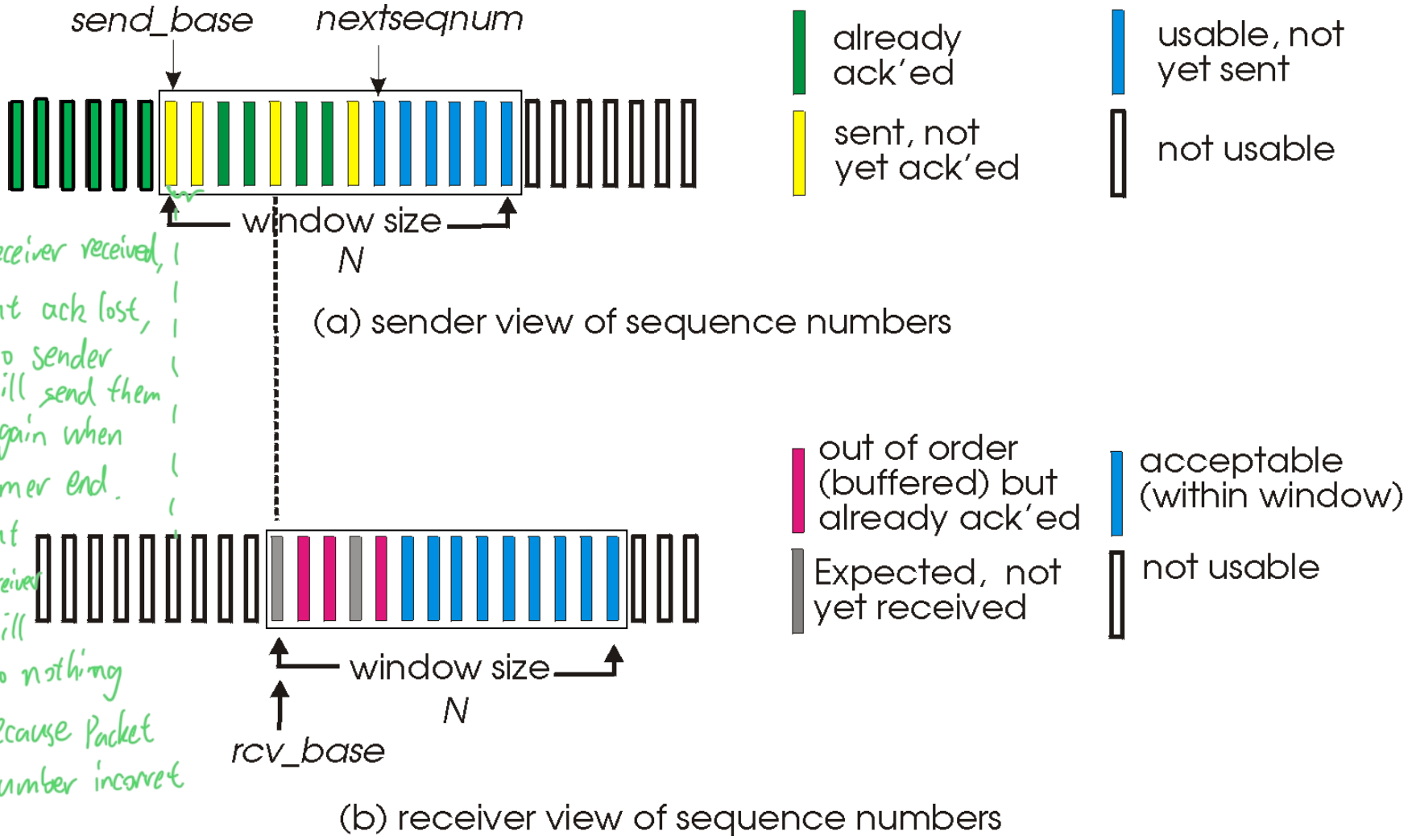
Selective repeat

- receiver *individually* acknowledges all correctly received pkts
  - buffers pkts, as needed, for eventual in-order delivery to upper layer

先存储正确pck，要求sender重传错误pck

- sender only resends pkts for which ACK not received
  - sender timer for each unACKed pkt
- sender window
- receiver window

Selective repeat: sender, receiver windows



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-1]:

– mark pkt n as received

– if n is smallest unACKed pkt, advance window base to next unACKed seq #

window 第一个收到为 acked 会导致 window sliding

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

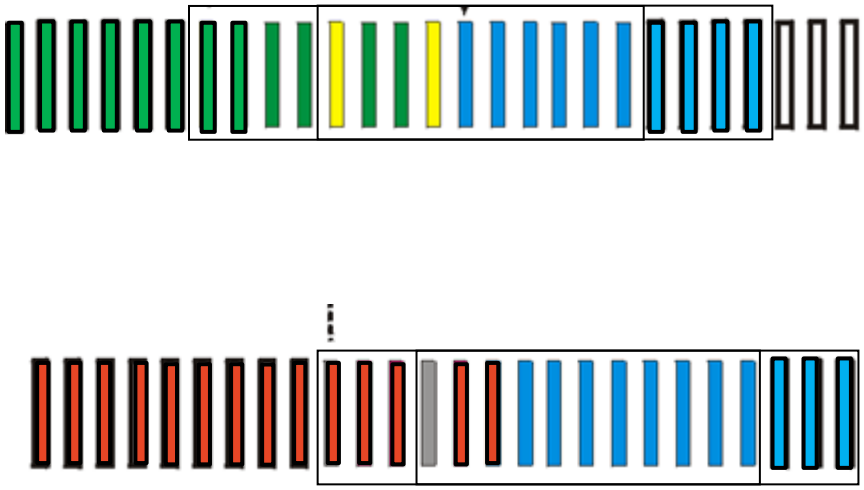
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

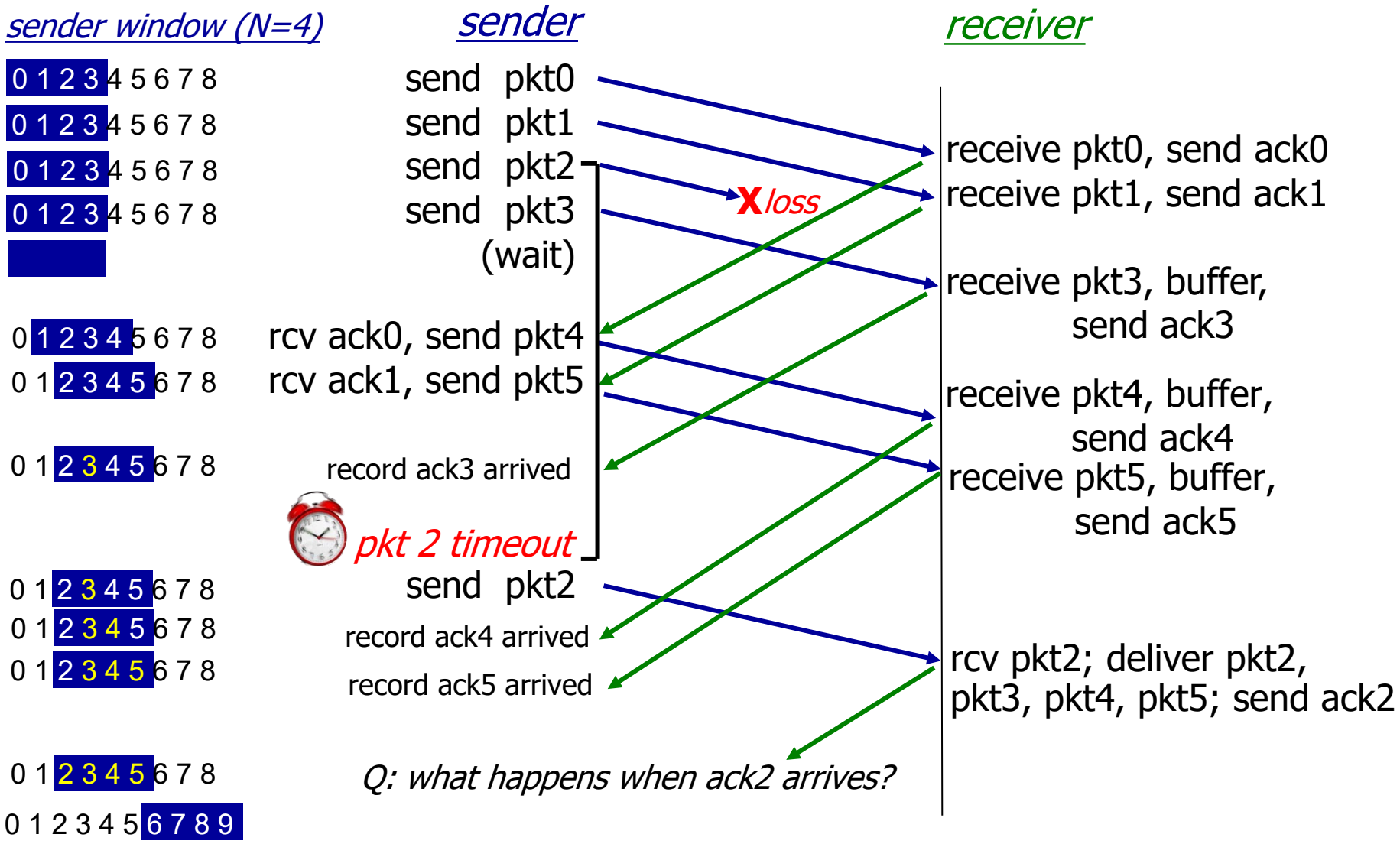
otherwise:

- ignore

Example



Selective repeat in action

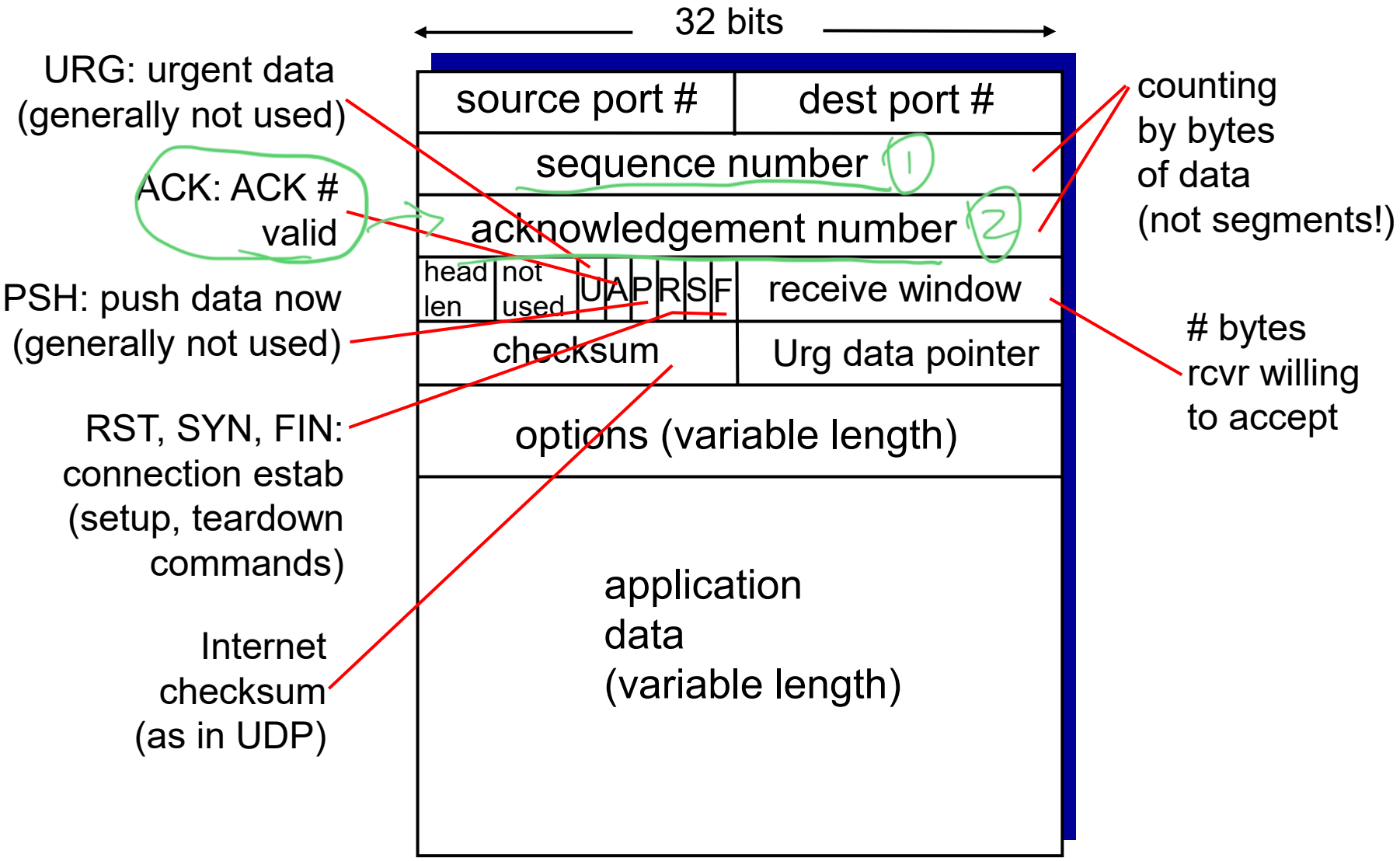


TCP

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

- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte stream*
- 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

1

sequence numbers:

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

2

acknowledgements:

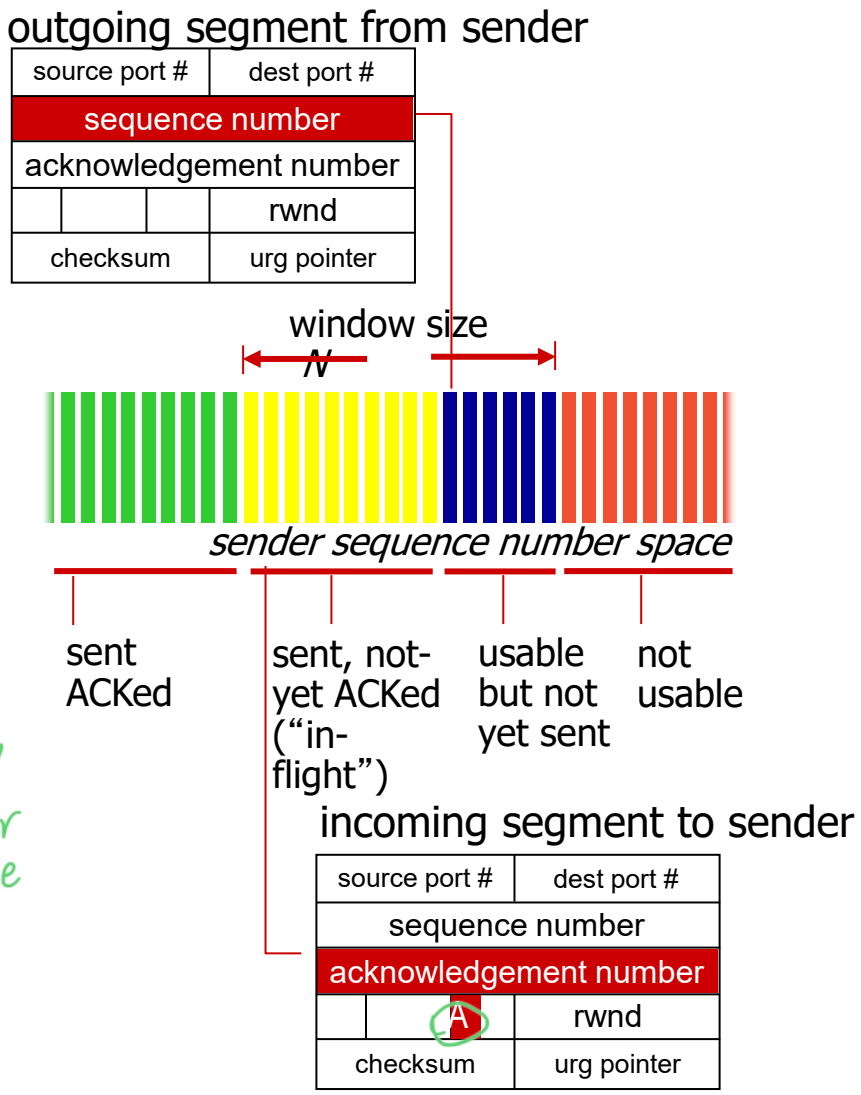
- seq # of **next byte** expected from other side
- cumulative ACK

both side need use these

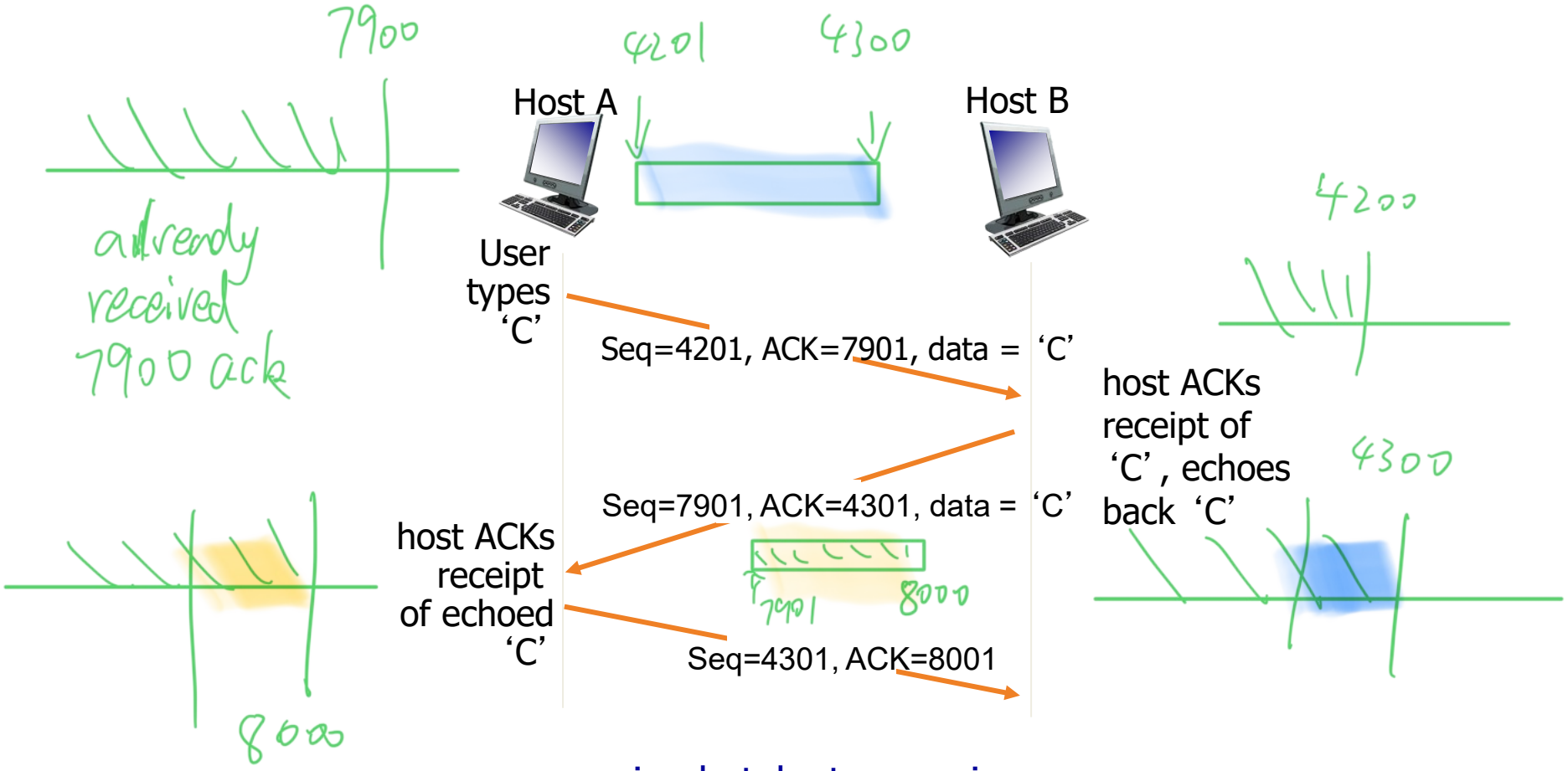
Q: how receiver handles out-of-order segments

- A: TCP spec doesn't say,
- up to implementor

so we say other side



TCP seq. numbers, ACKs



simple telnet scenario

TCP round trip time, timeout

Round Trip Time

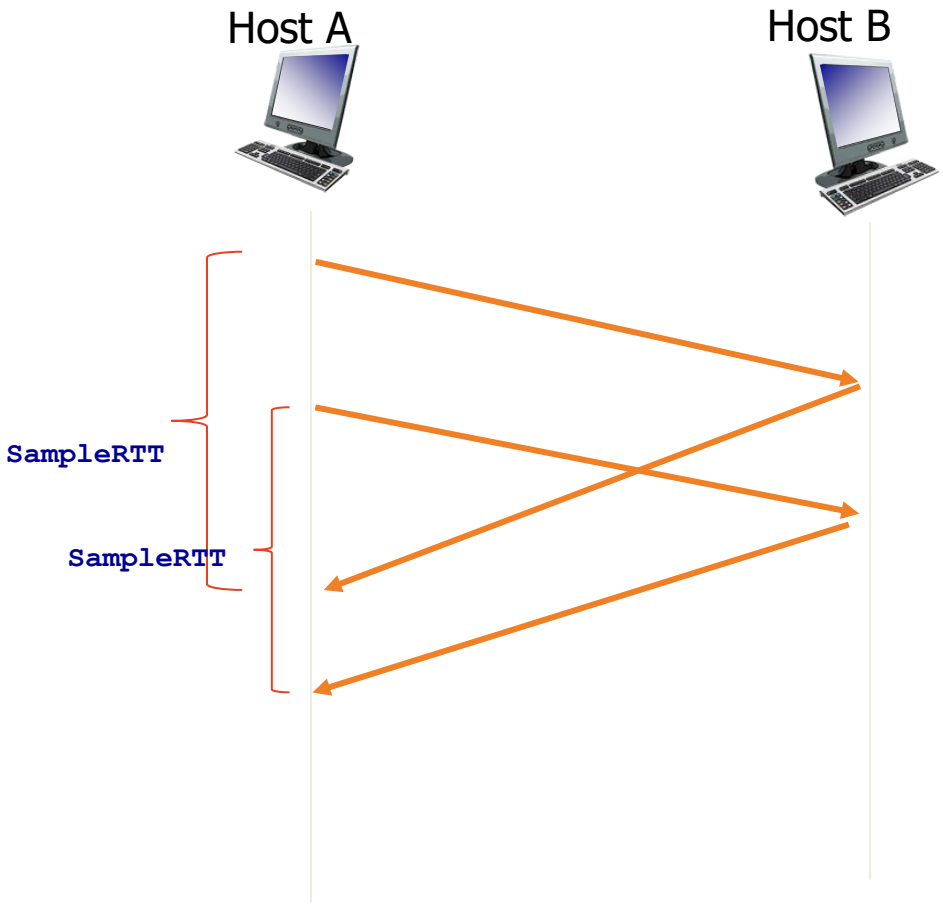
Q: how to set TCP timeout value?

- longer than RTT
  - but RTT varies
- *too short*: premature timeout, unnecessary retransmissions
- *too long*: slow reaction to segment loss

Q: how to estimate RTT?

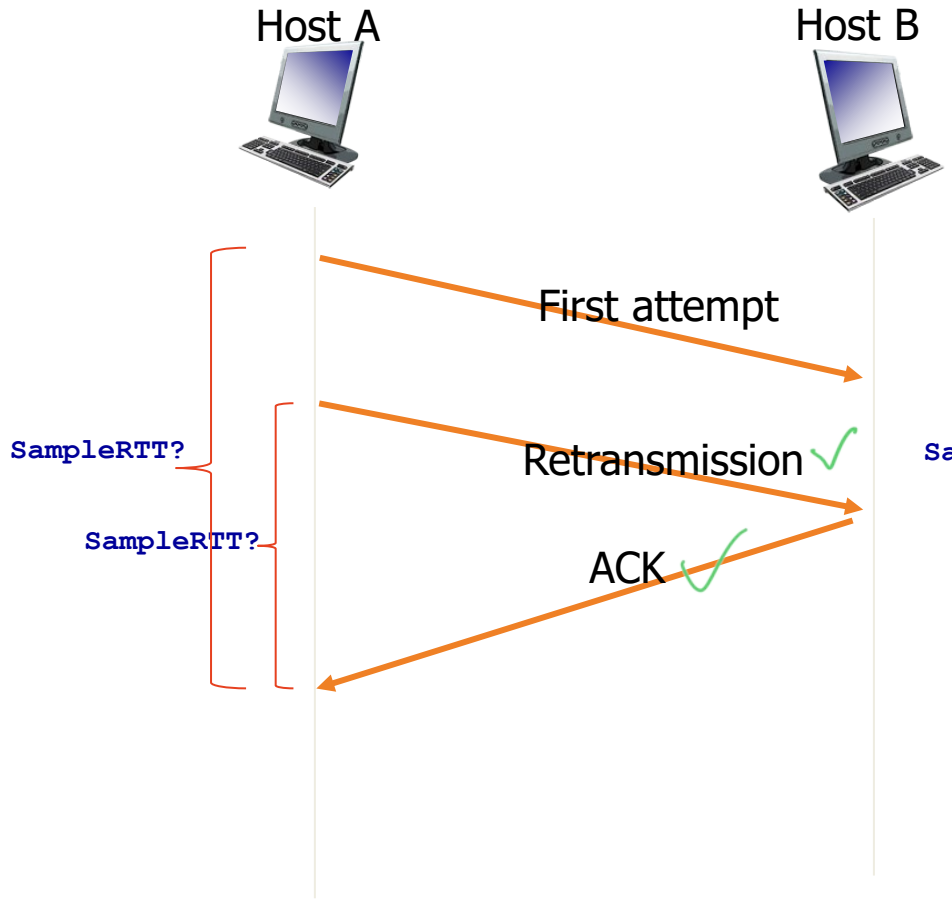
- **`SampleRTT`**: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- **`SampleRTT`** will vary, want estimated RTT "smoother"
  - weighted average of several *recent* measurements, not just current **`SampleRTT`**

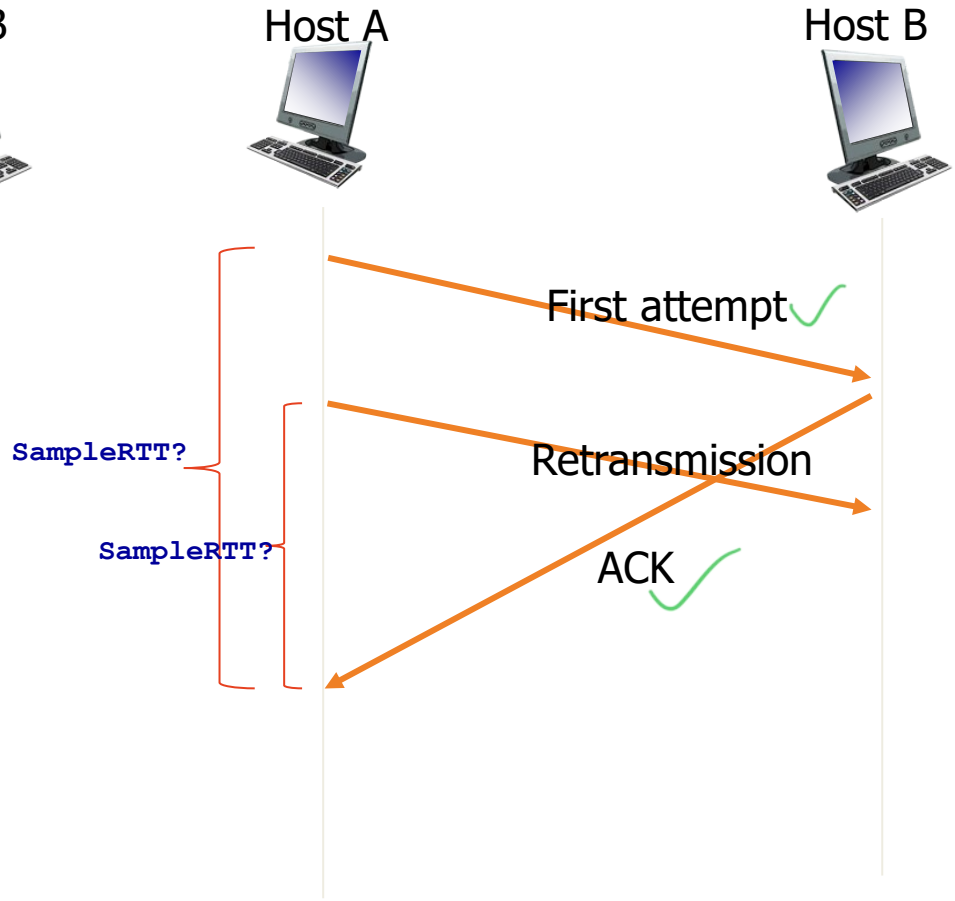
SampleRT



Ignore retransmissions

ambiguity

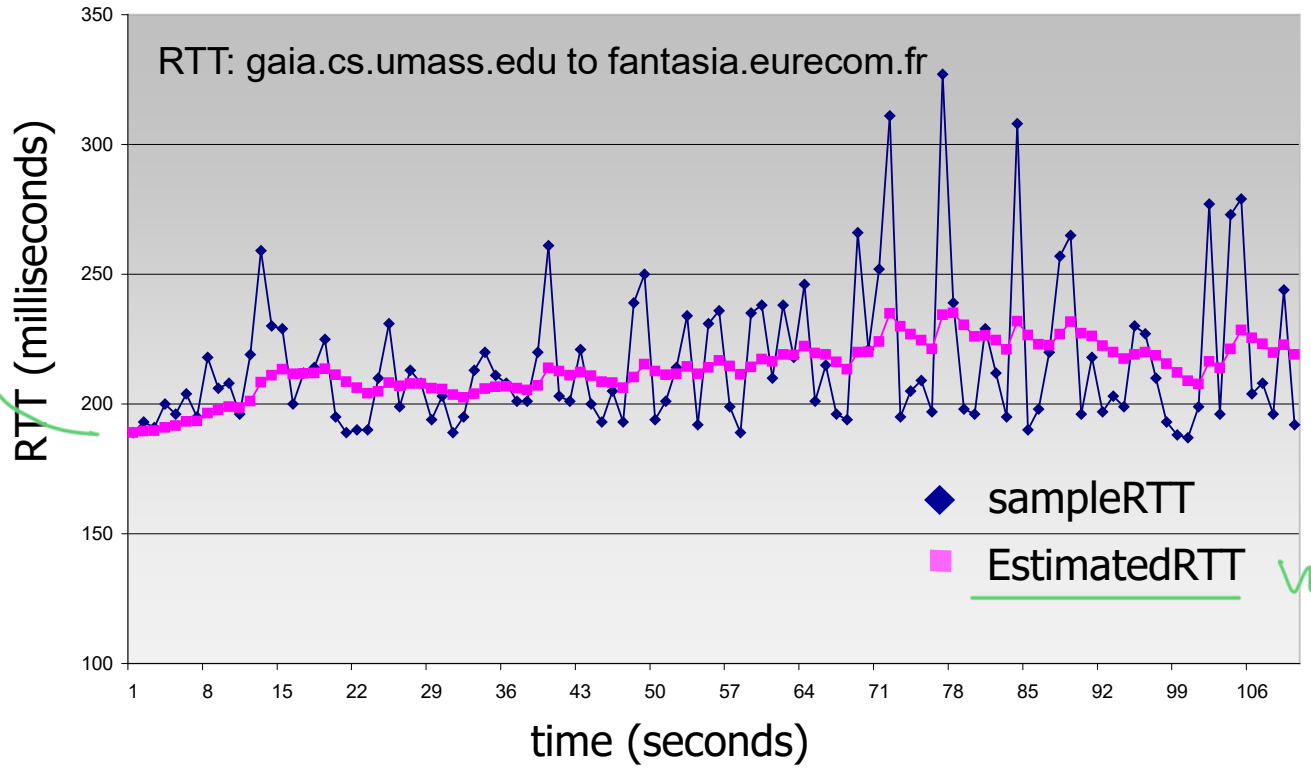




TCP round trip time, timeout

EstimatedRTT = (1- α)*EstimatedRTT + α*SampleRTT

- exponential weighted moving average
- influence of past sample decreases exponentially fast
- typical value: α = 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-β)*DevRTT +
β*|SampleRTT-EstimatedRTT|

(typically, β = 0.25)

TimeoutInterval = EstimatedRTT + 4*DevRTT

estimated RTT

"safety margin"

Reliable Data Transfer in TCP

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`

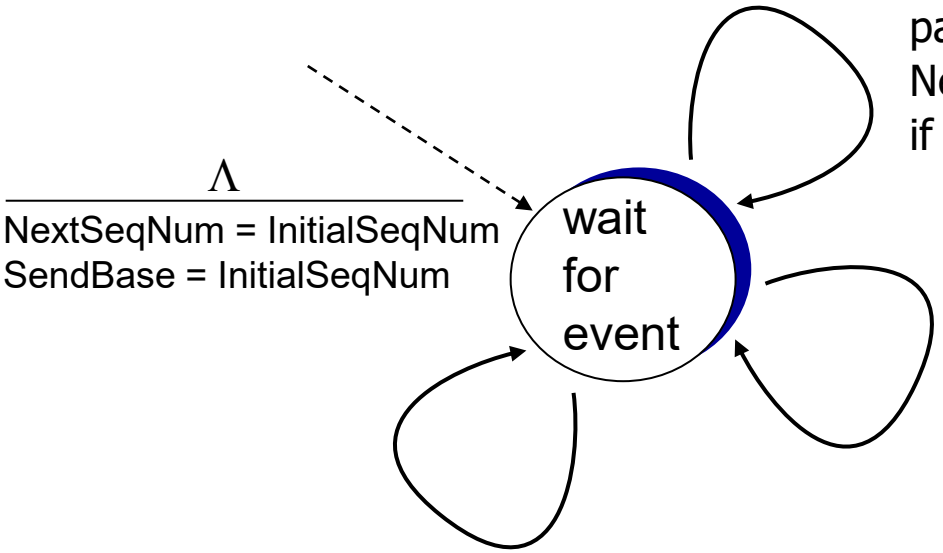
timeout:

- 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

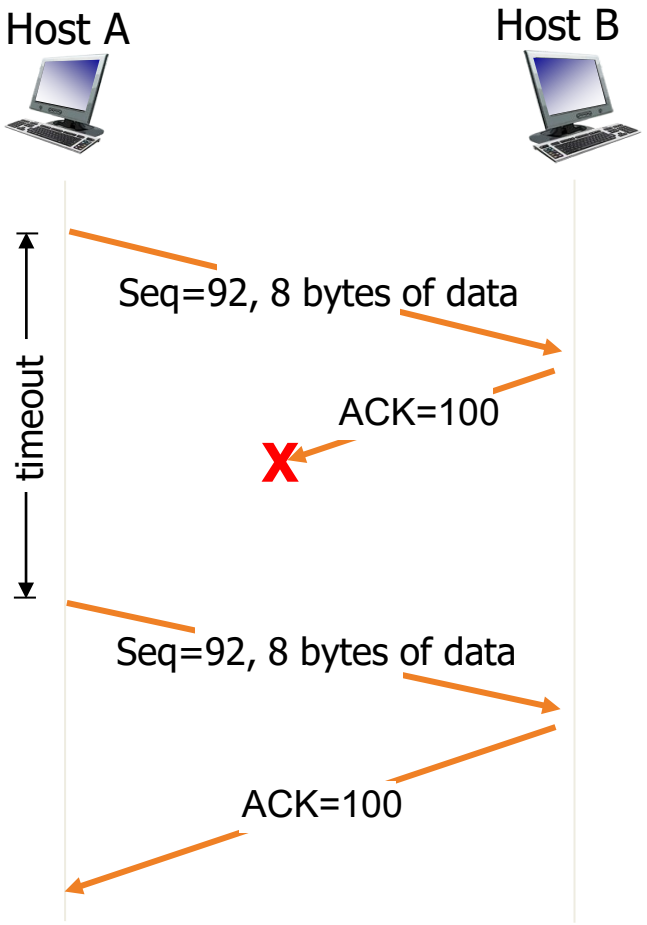
timeout

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

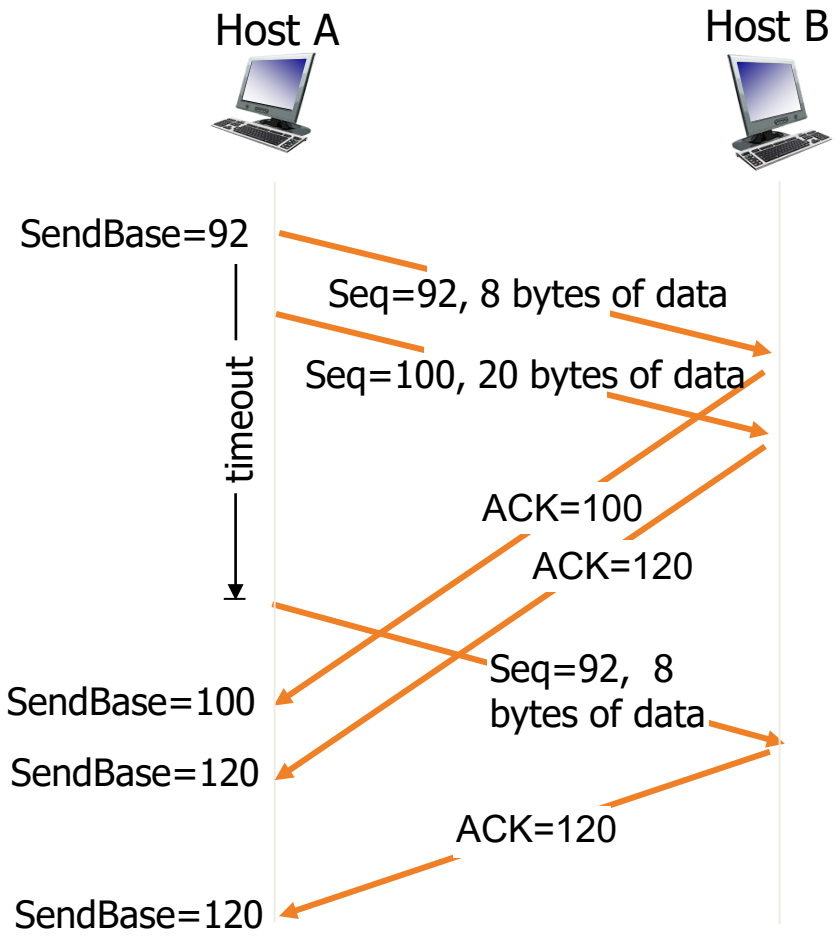
ACK received, with ACK field value y

```
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

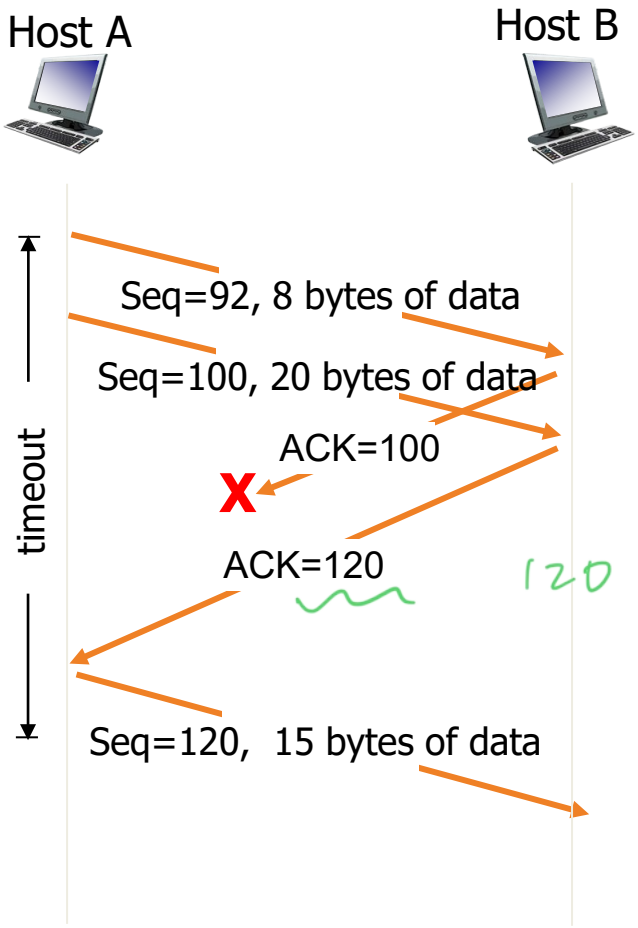


lost ACK scenario



premature timeout

TCP: retransmission scenarios



120 overwrite 100

because ack 120 means ack0 — ack 120 are all correct

cumulative ACK

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.

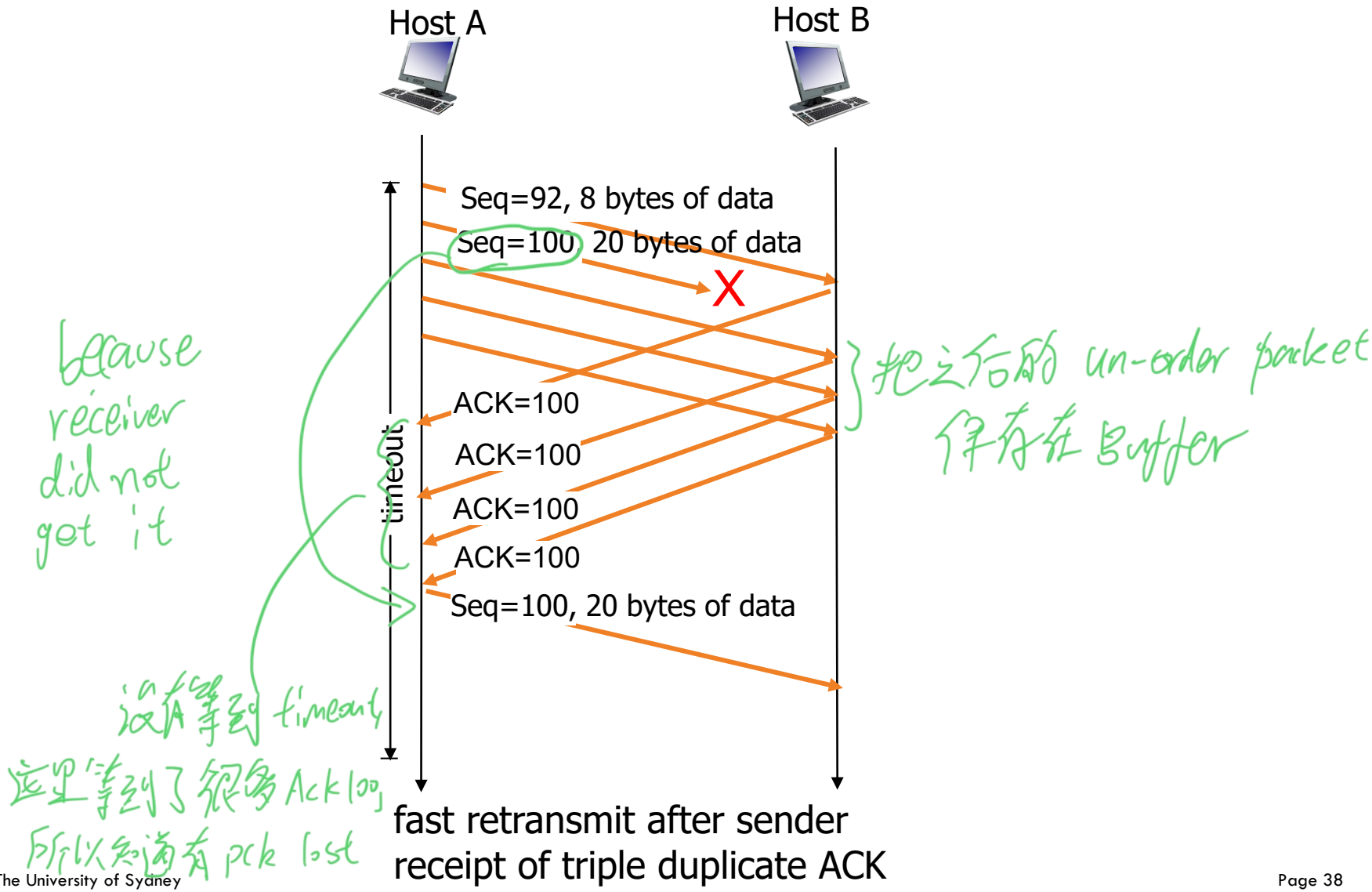
TCP fast retransmit

if sender receives 3 duplicate 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

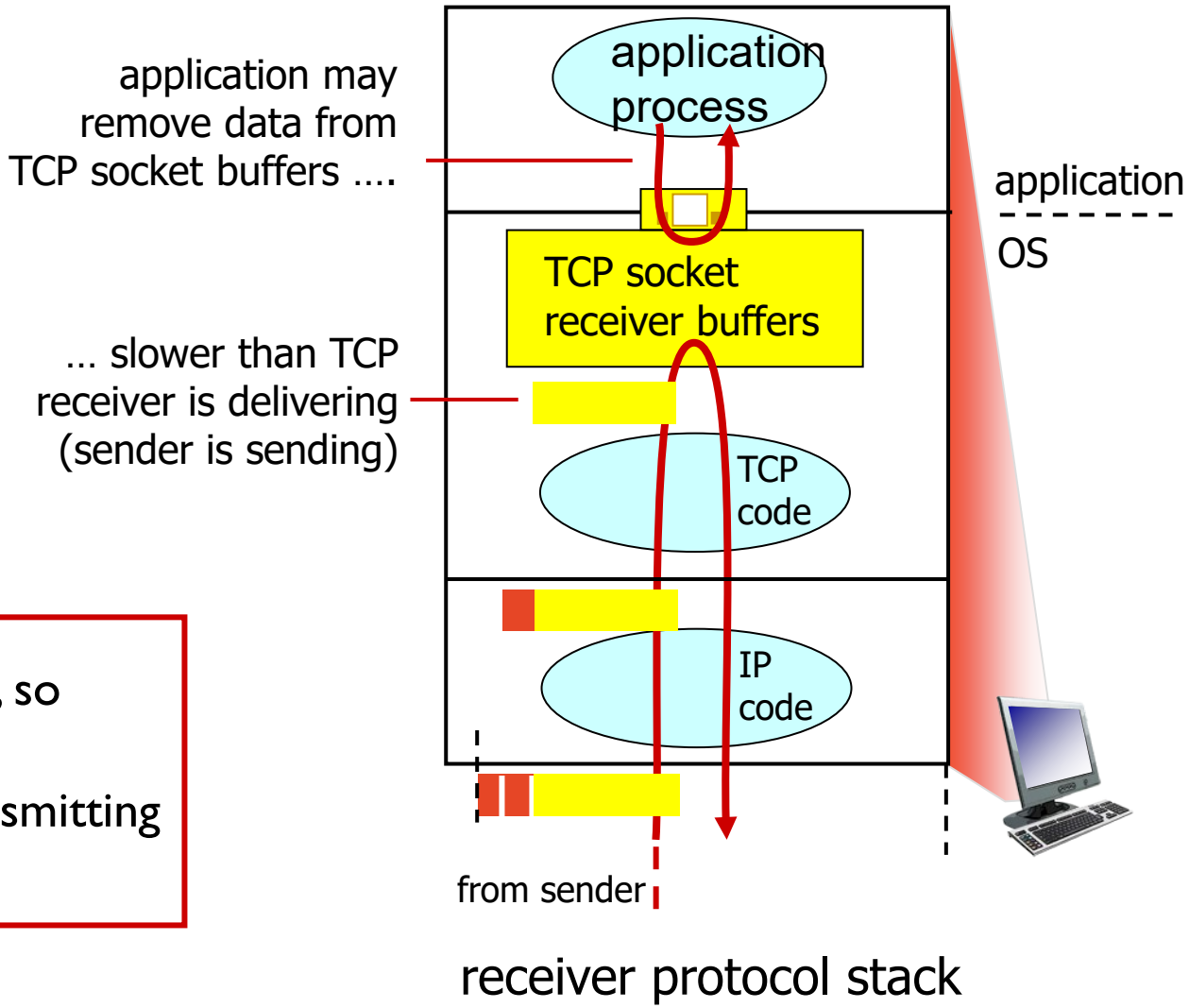


fast retransmit after sender receipt of triple duplicate ACK

Flow Control in TCP

TCP flow control

Receiver buffer is limited

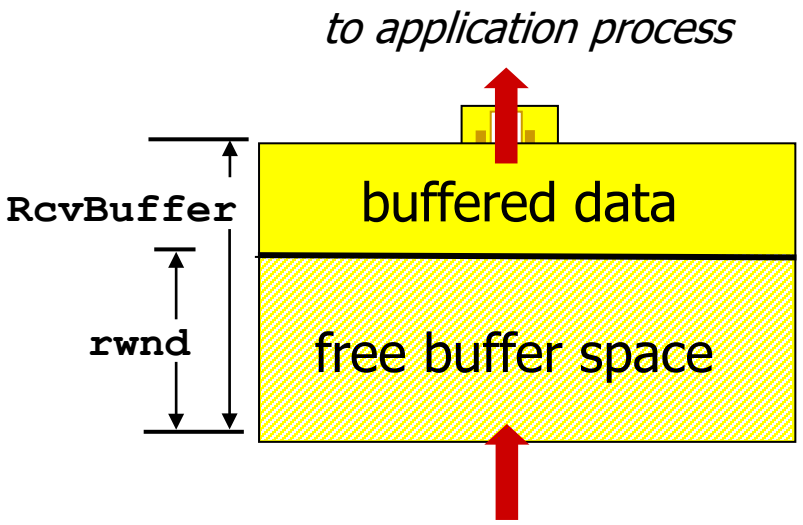


flow control

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



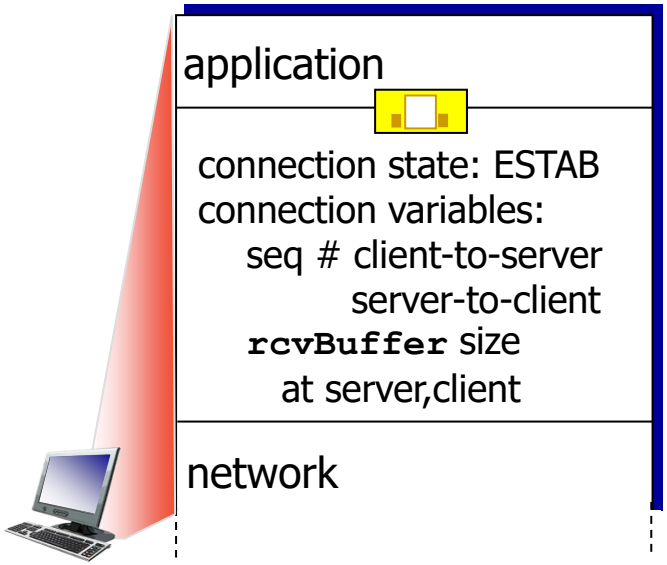
receiver-side buffering

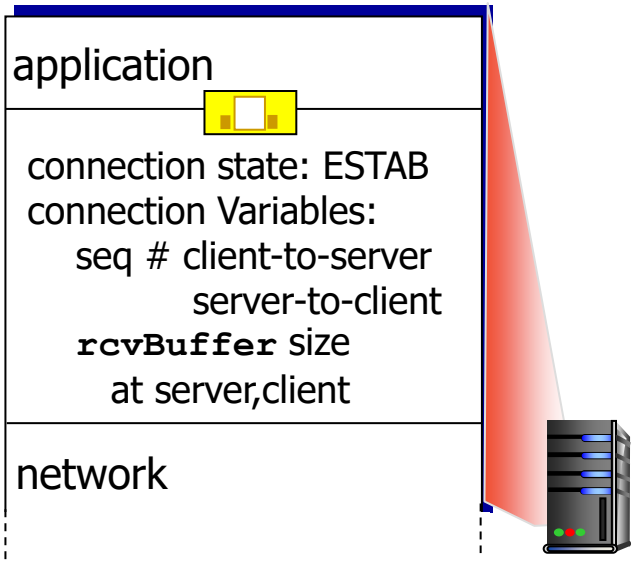
Connection Management in TCP

Connection Management

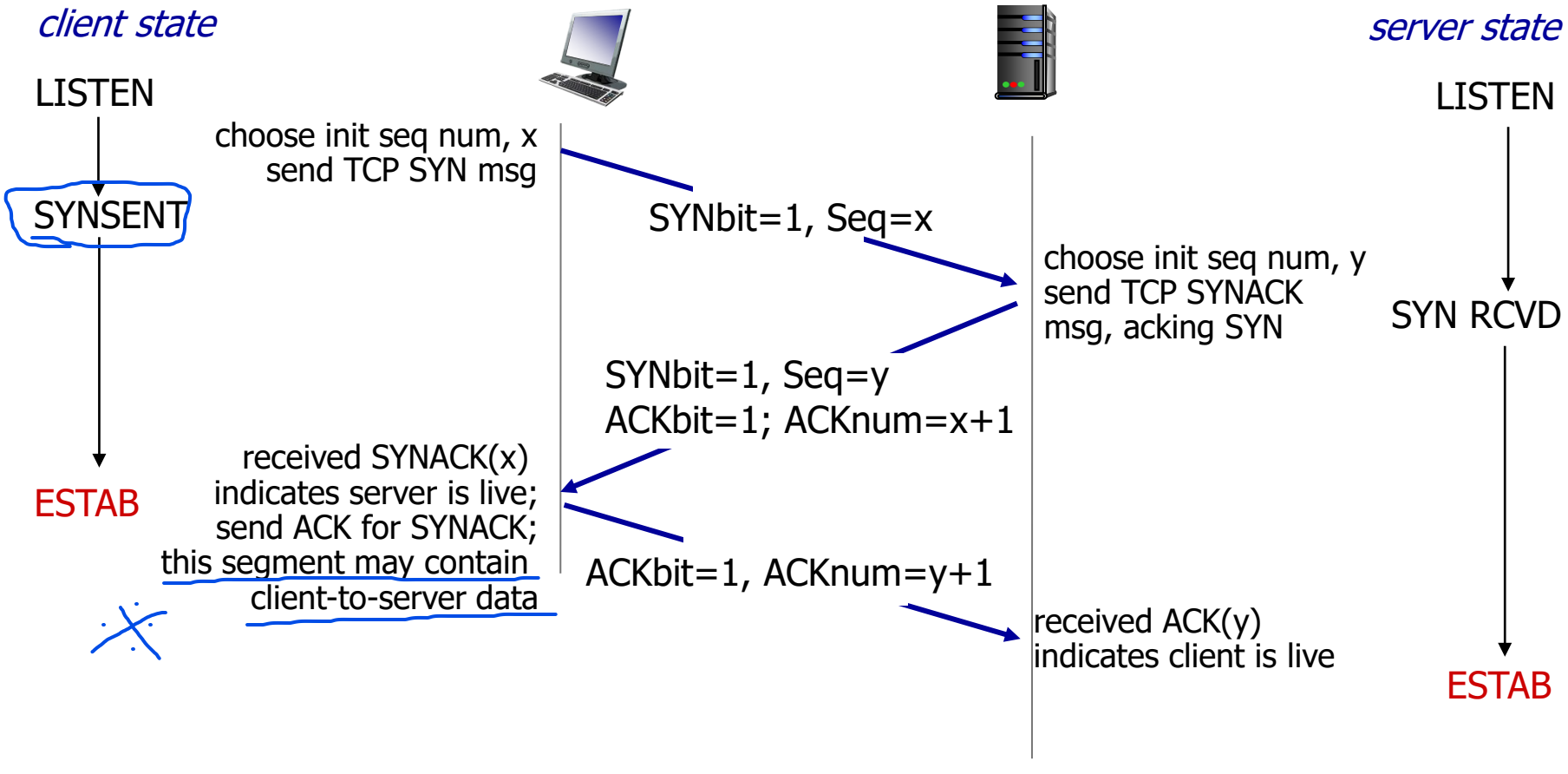
before exchanging data, sender/receiver "handshake":

- agree to establish connection (each knowing the other willing to establish connection)
- agree on connection parameters





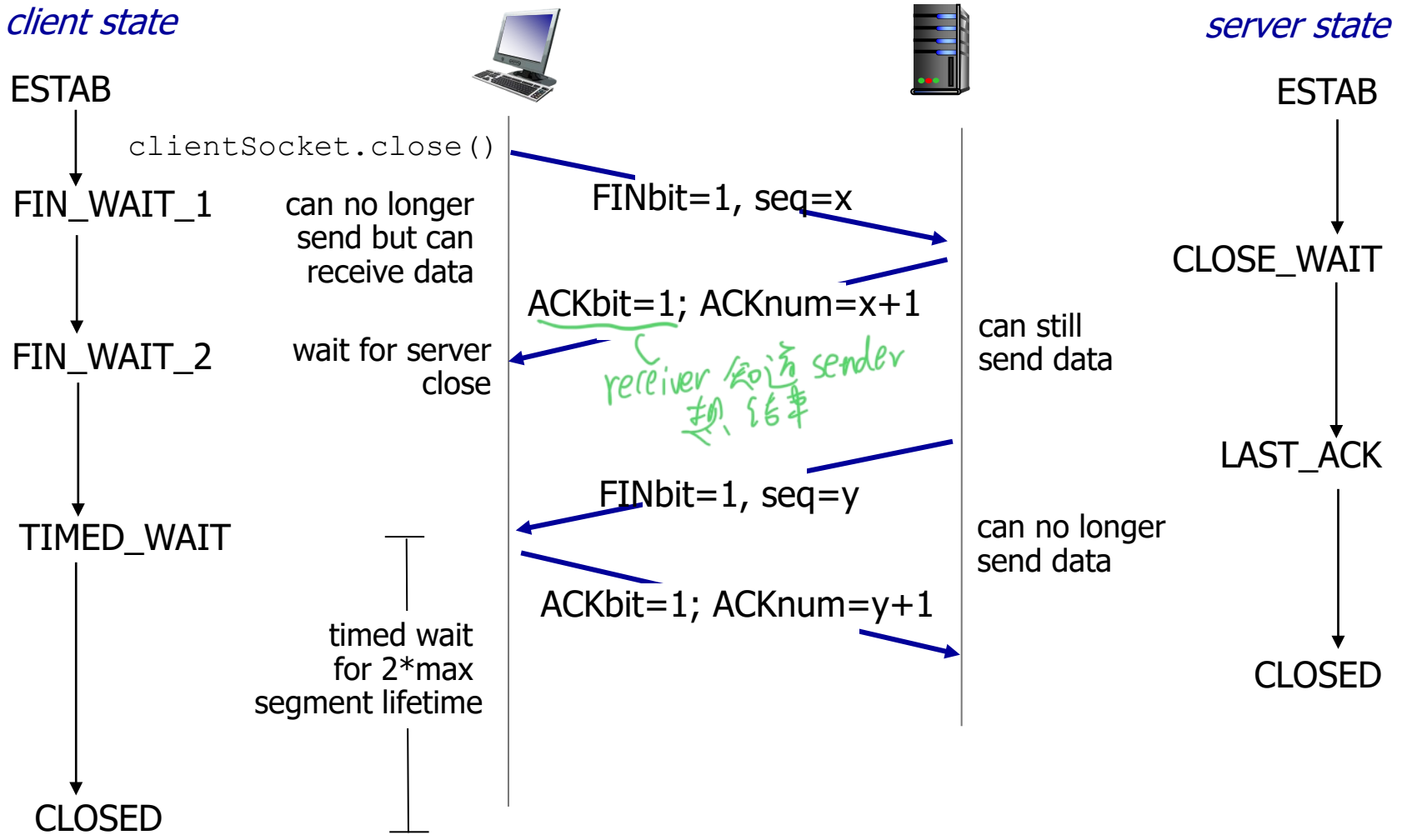
TCP 3-way handshake



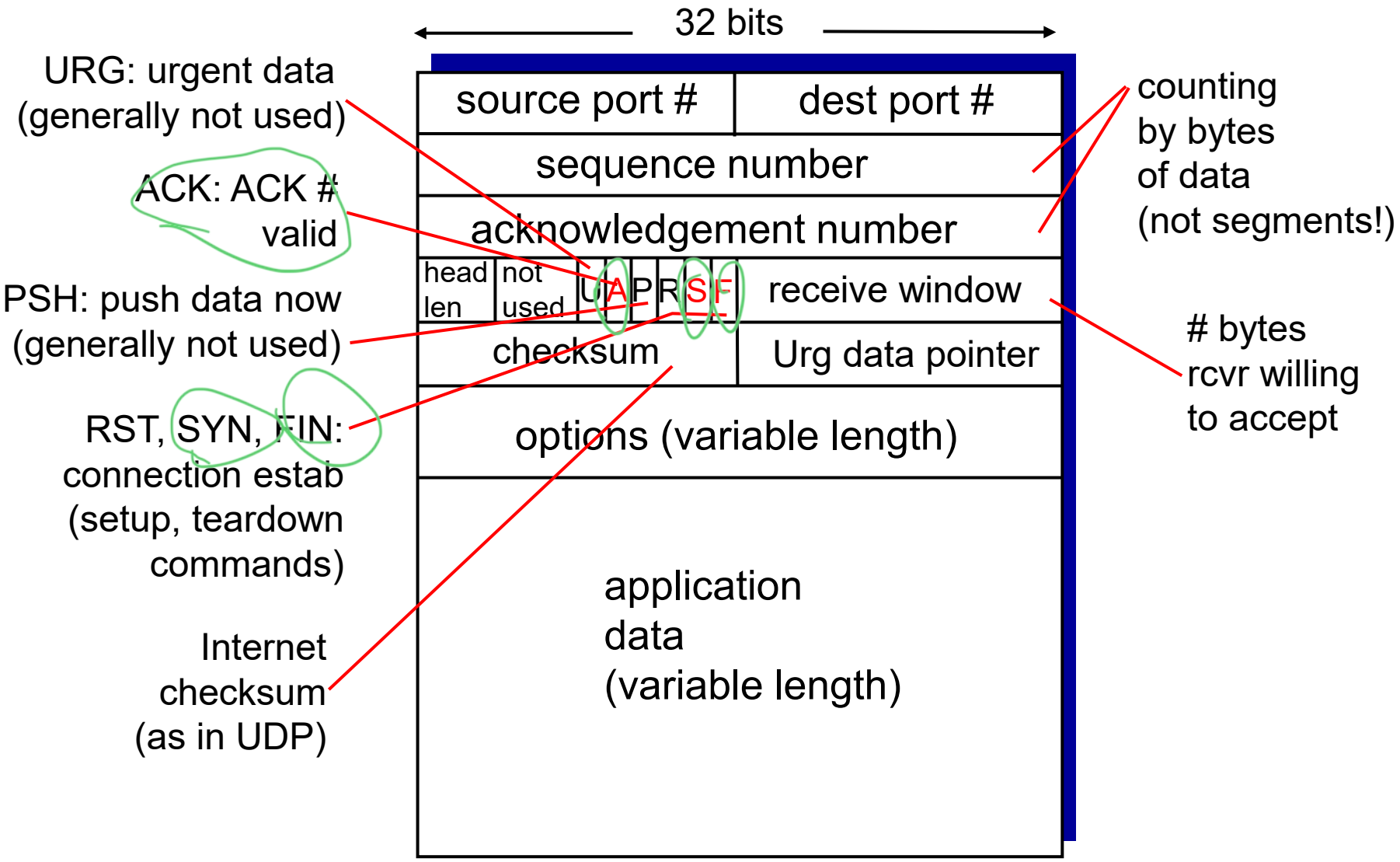
TCP: closing a connection

- client, server each closes their side of connection
  - send TCP segment with FIN bit = 1
- respond to received FIN with ACK

TCP: closing a connection



TCP segment structure



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: 2 cases
  1. lost packets (buffer overflow at routers)
  2. long delays (queueing in router buffers)
- a top-10 problem!

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:

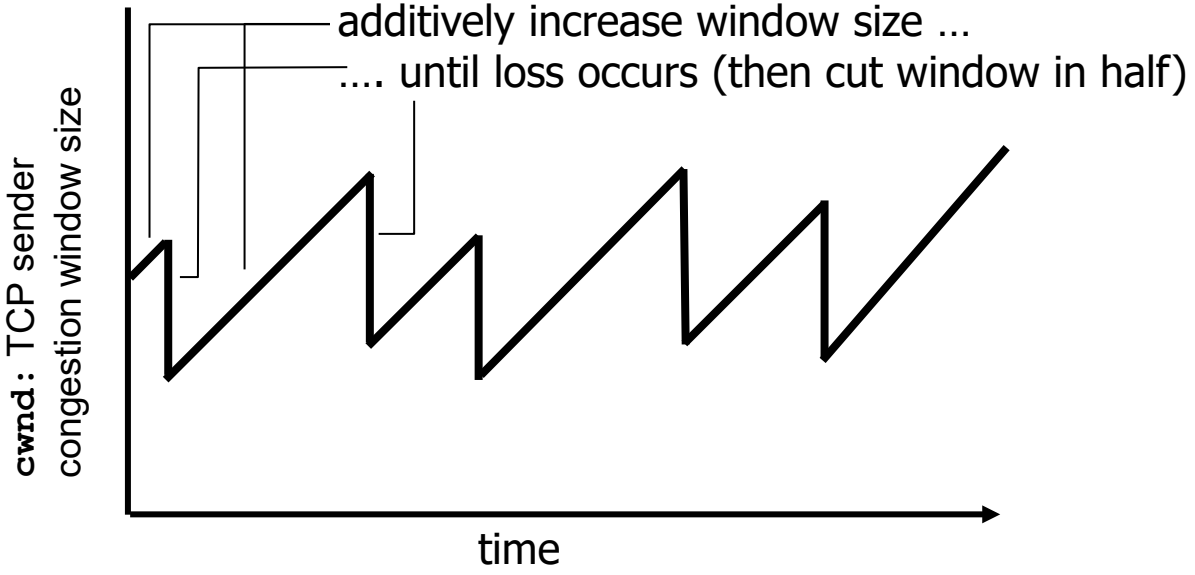
- routers provide feedback to end systems
  - single bit indicating congestion
  - explicit rate for sender to send at

TCP congestion control

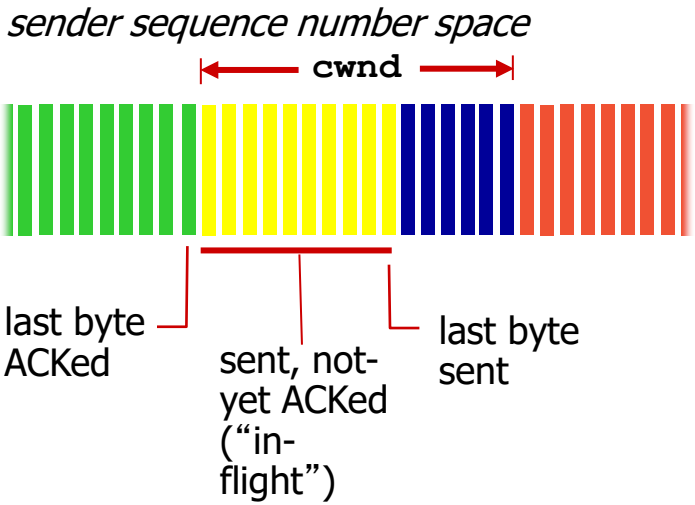
Additive increase multiplicative decrease (AIMD)

- *approach:* sender increases transmission rate (window size), probing for usable bandwidth, until loss occurs
  - *additive increase:* increase **`cwnd`** by 1 MSS (maximum segment size) every RTT until loss detected
  - *multiplicative decrease:* cut **`cwnd`** in half after loss

AIMD saw tooth behavior: probing for bandwidth



TCP Congestion Control: details



TCP sending rate:

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

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

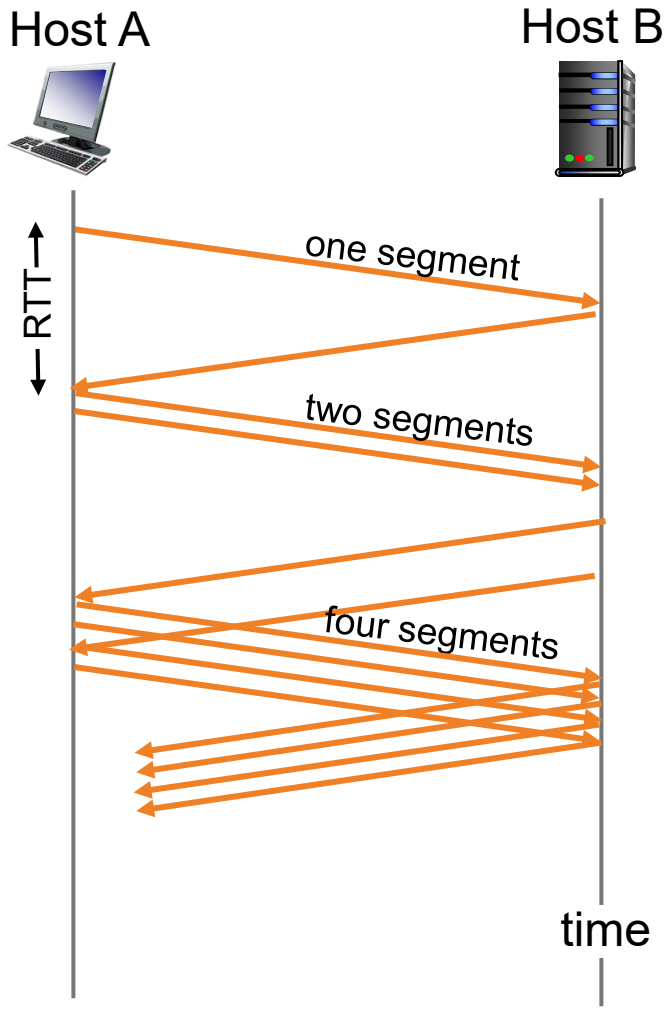
› sender limits transmission:

LastByteSent- LastByteAcked ≤ cwnd

› **cwnd** is dynamic, function of perceived network congestion

TCP Slow Start

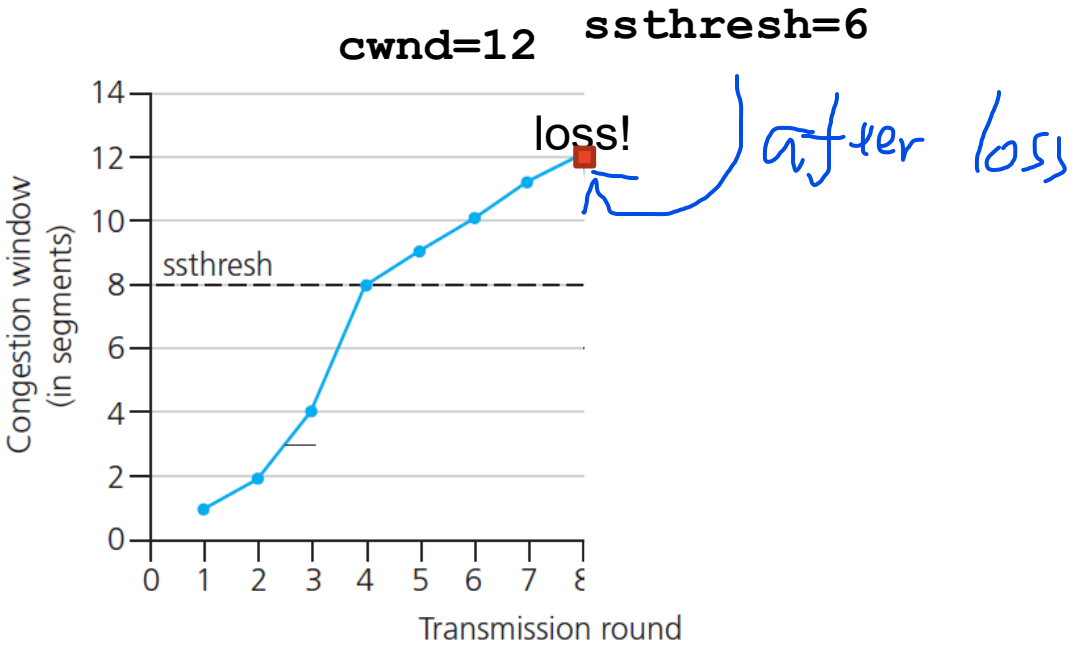
- when connection begins, increase rate exponentially:
  - initially **`cwnd`** = 1 MSS
  - double **`cwnd`** every RTT
  - done by incrementing **`cwnd`** for every ACK received
- *summary:* initial rate is slow but ramps up exponentially fast
- when should the exponential increase switch to linear (additive increase)?



TCP: switching from slow start to CA

Q: when should the exponential increase switch to linear?

A: cwnd reaches **`ssthresh`**



Implementation:

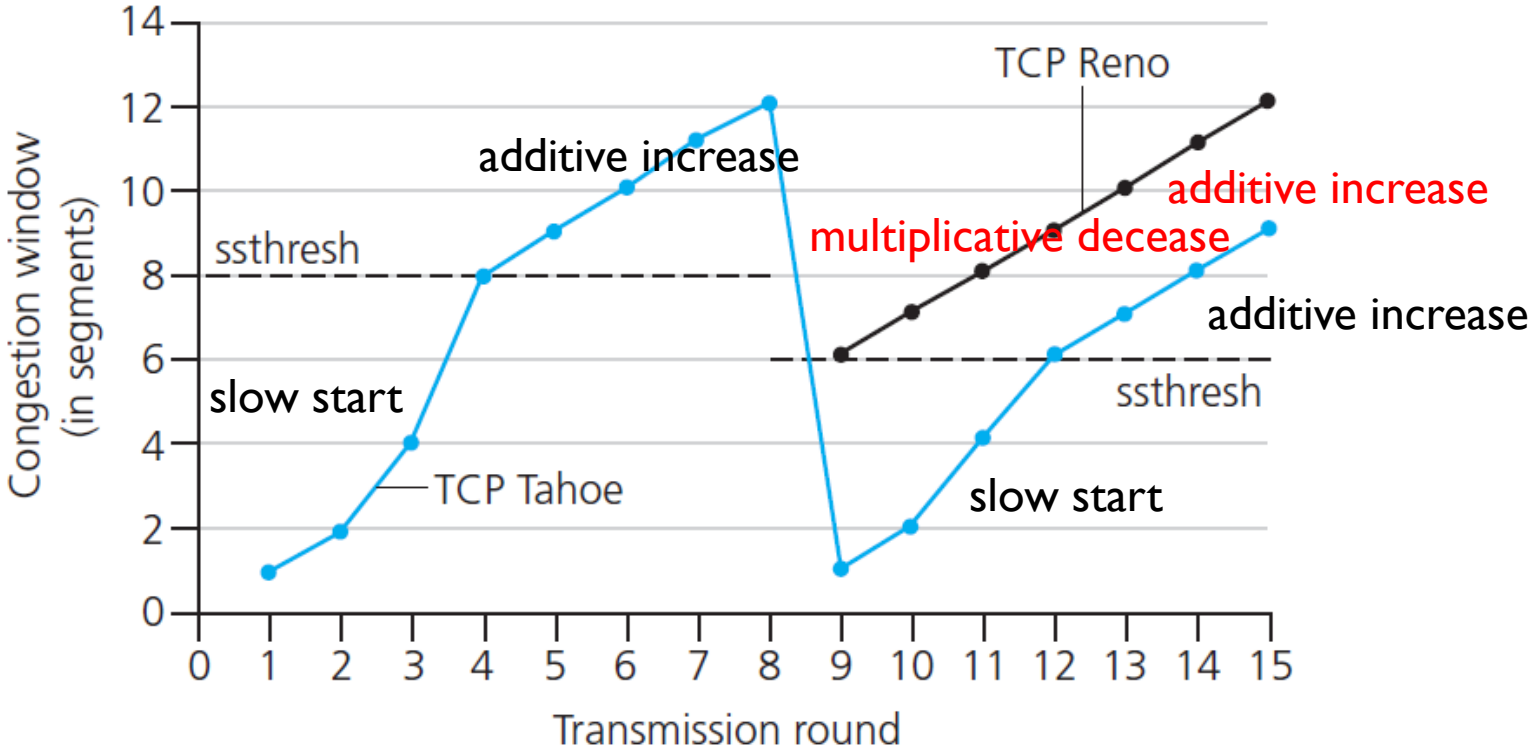
- At beginning **`ssthresh`**, specified in different versions of TCP
- (In this example **`ssthresh=8 segment`**)

› on loss event, **`ssthresh`** is set to 1/2 of **`cwnd`** just before loss event

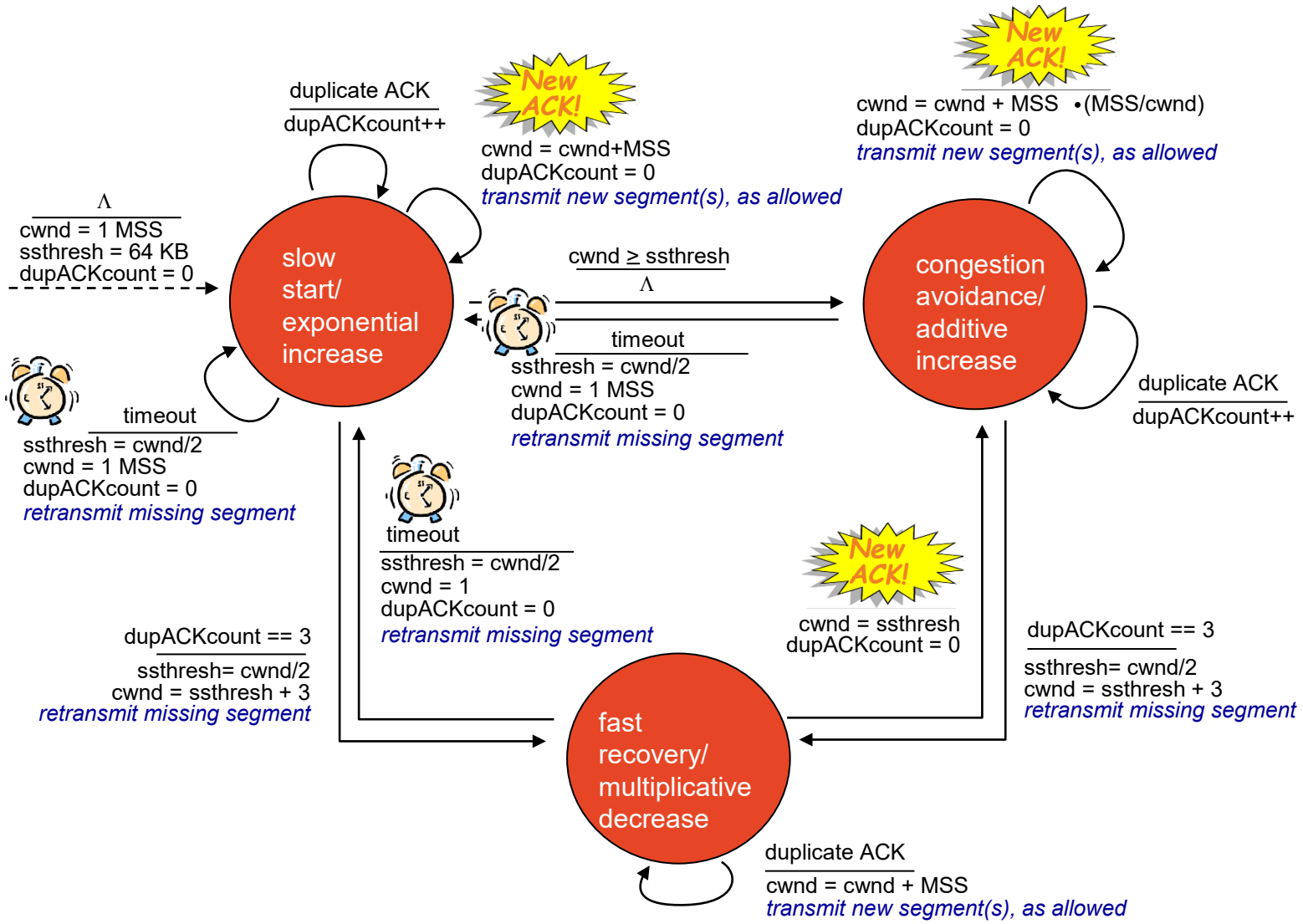
TCP: detecting, reacting to loss

- loss indicated by timeout:
  - **cwnd** set to 1 MSS;
  - window then grows exponentially (as in slow start) to **ssthresh**, then grows linearly
- loss indicated by 3 duplicate ACKs:
  - TCP Tahoe, same as loss indicated by timeout, always sets **cwnd** to 1 (timeout or 3 duplicate acks)
  - TCP RENO
    - **cwnd** is cut in half window then grows linearly (additive increase)
    - fast recovery

TCP: switching from slow start to CA

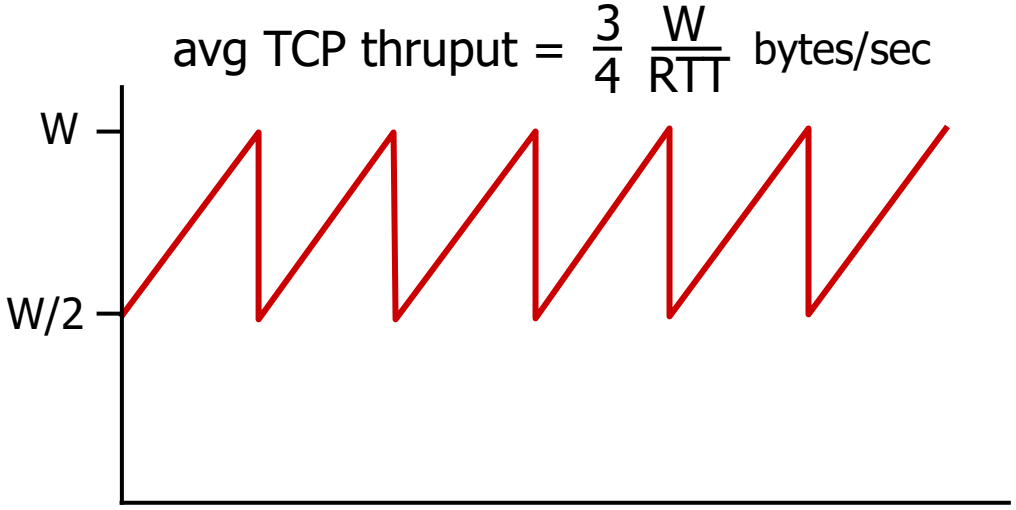


Summary: TCP Reno Congestion Control



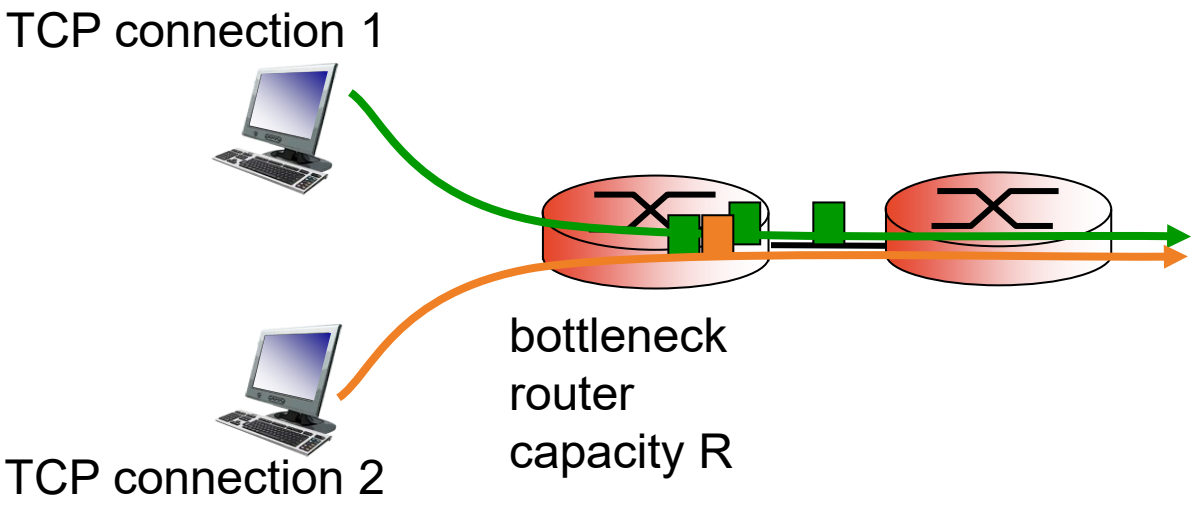
TCP Reno throughput

- 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 ¾ W
  - avg. thruput is 3/4W per RTT



TCP Fairness

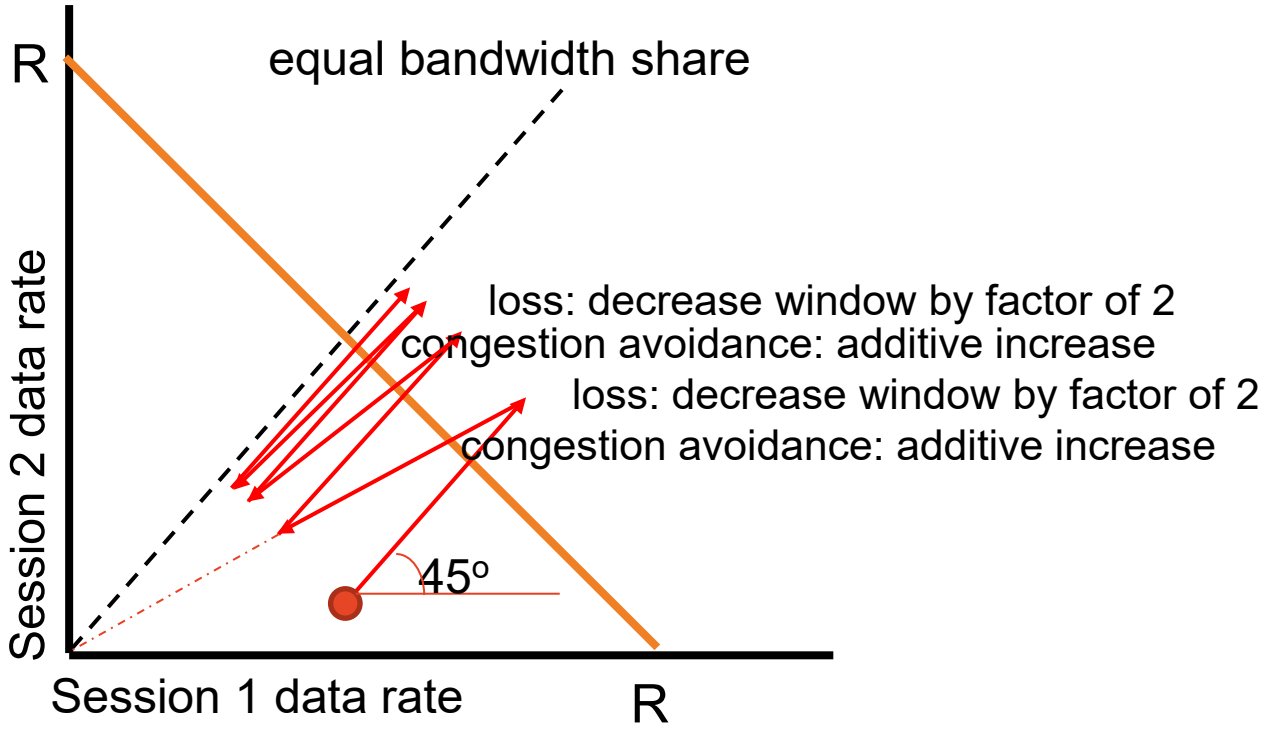
Fairness: K TCP sessions share same bottleneck link of bandwidth R, each has average rate of R/K



Why is TCP fair?

two competing sessions:

- additive increase gives slope of 1, as throughout 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
- e.g., link of rate R
  - App 1 asks for 1 TCP, gets 0.1R
  - App 2 asks for 9 TCPs, gets 0.9R

Final word

Window size = min (rwnd, cwnd)

receive window

flow control

congestion window

congestion control

Conclusion

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