Internet Protocols EBU5403 The Transport Layer Part 2

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	Week 1	Week 2	Week 3	Week 4
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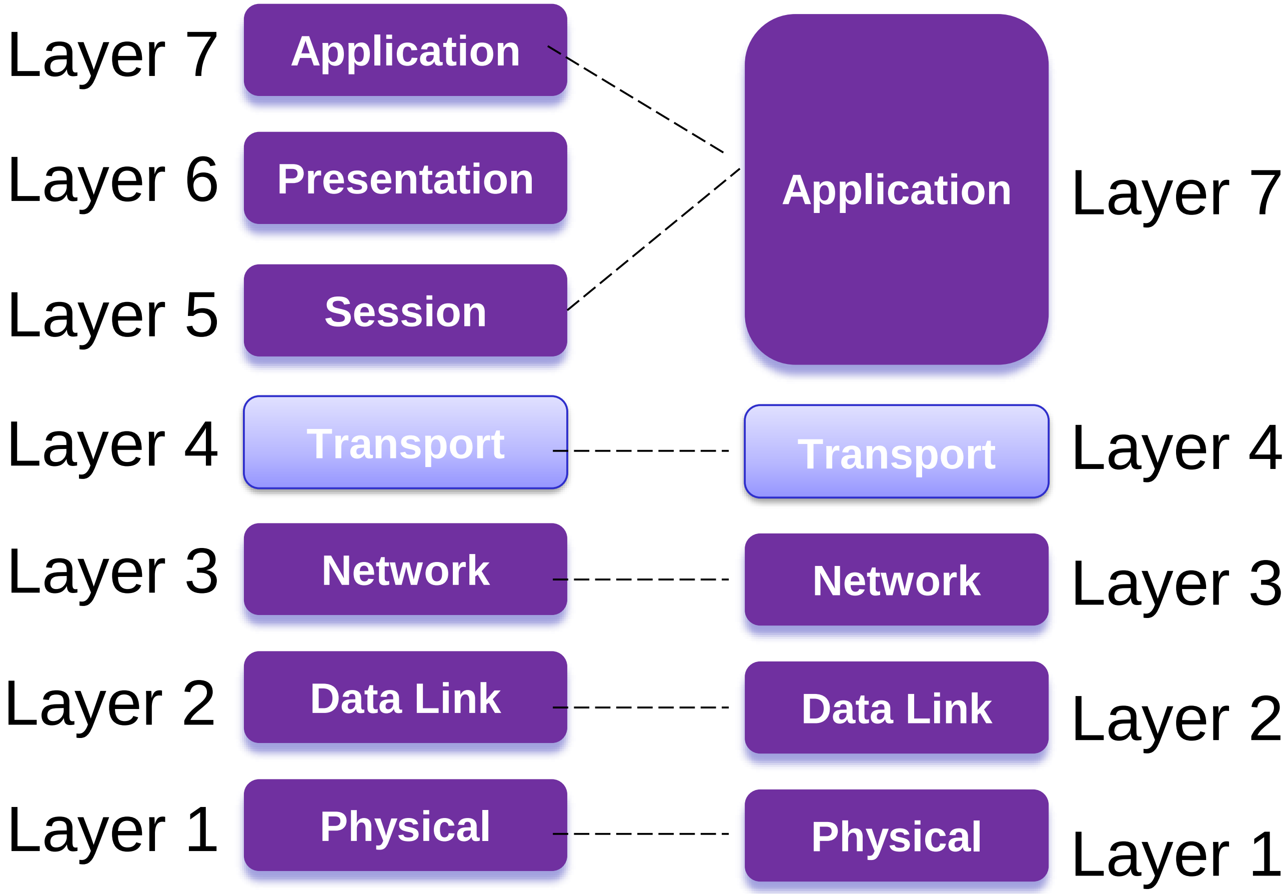
Structure of course

- Week 1
  - Introduction to IP Networks
  - The Transport layer (part 1)
- Week 2
  - The Transport layer (part II)
  - The Network layer (part I)
  - Class test
- Week 3
  - The Network layer (part II)
  - The Data link layer (part I)
  - Router lab tutorial (assessed lab work after this week)
- Week 4
  - The Data link layer (part II)
  - Network management and security
  - Class test

Transport Layer



Transport layer 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

Recap of rdt 1.0 2.0, 2.1, 2.2, 3.0

- rdt 1.0 assumed no losses or errors
- rdt 2.0 introduced ACK (ACKnowledgment) and NACK to say a packet had been received or an error occurred.
- rdt 2.1 introduced the concept of the SEQuence number to deal with errors in ACKs.
- rdt 2.2 introduced the repeated ACK – ACK with repeated sequence number means same as NACK.
- rdt 3.0 introduced timeout in case packet lost completely not just corrupted.

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

Performance of rdt3.0

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

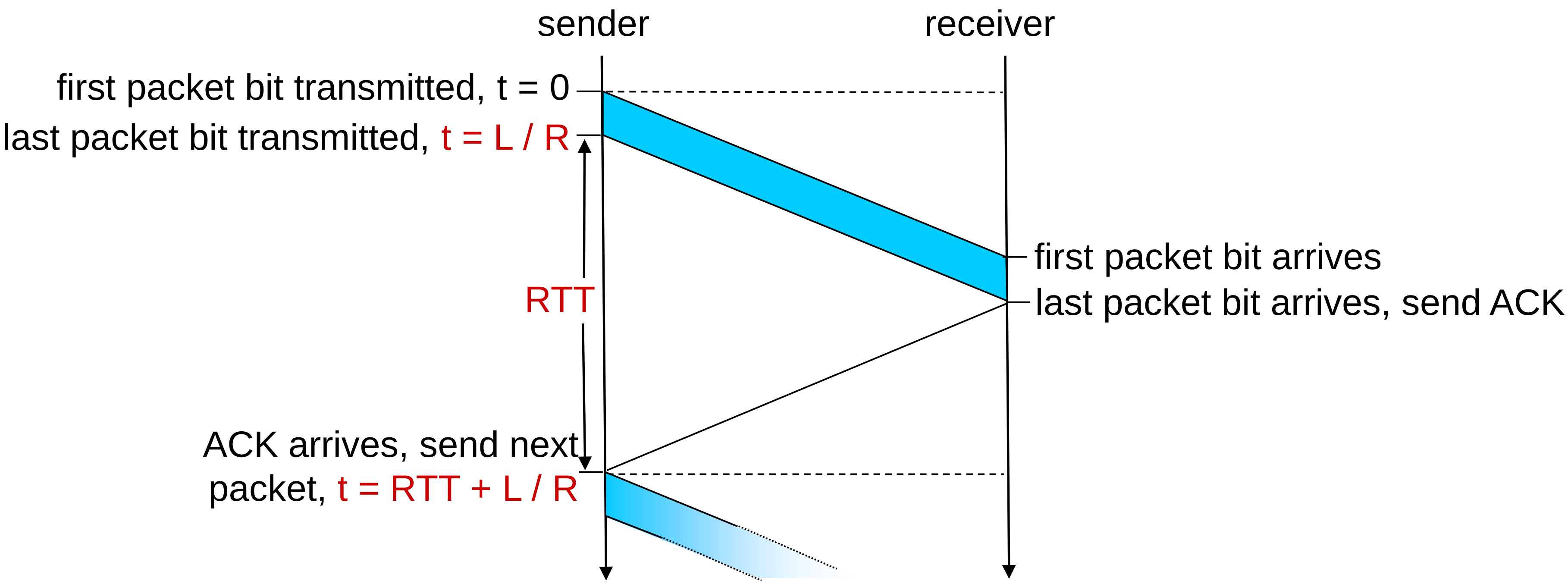
$$D_{trans} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 8 \text{ microsecs}$$

  - 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 throughput over 1 Gbps link
- Badly designed protocol has limted how we use the resource.

rdt3.0: stop-and-wait operation

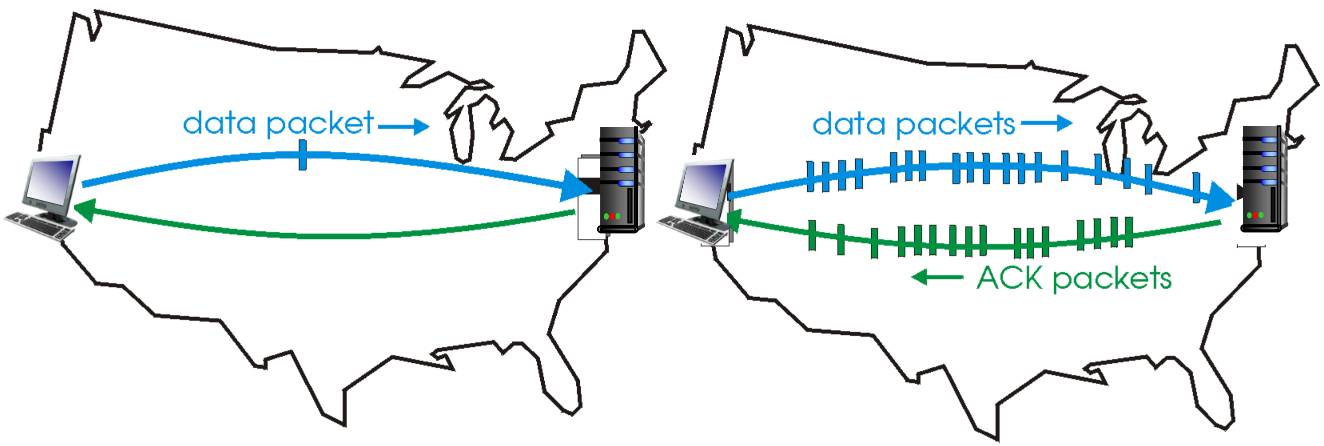


$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols

pipelining: sender allows multiple, "in-flight", yet-to-be-acknowledged packets (packets with no ACK as yet)

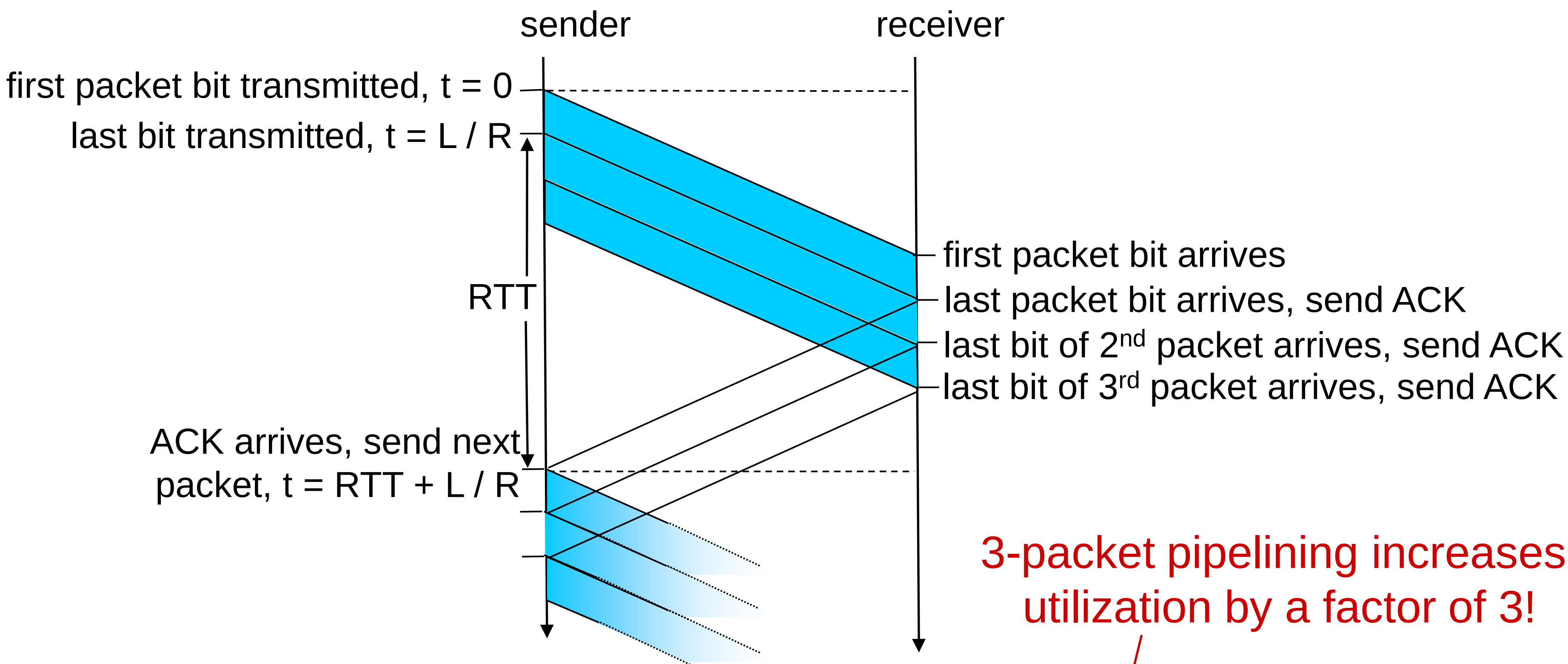
- range of sequence numbers must be increased
- buffering at sender and/or receiver



(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!

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

Pipelined protocols: overview

Go-back-N:

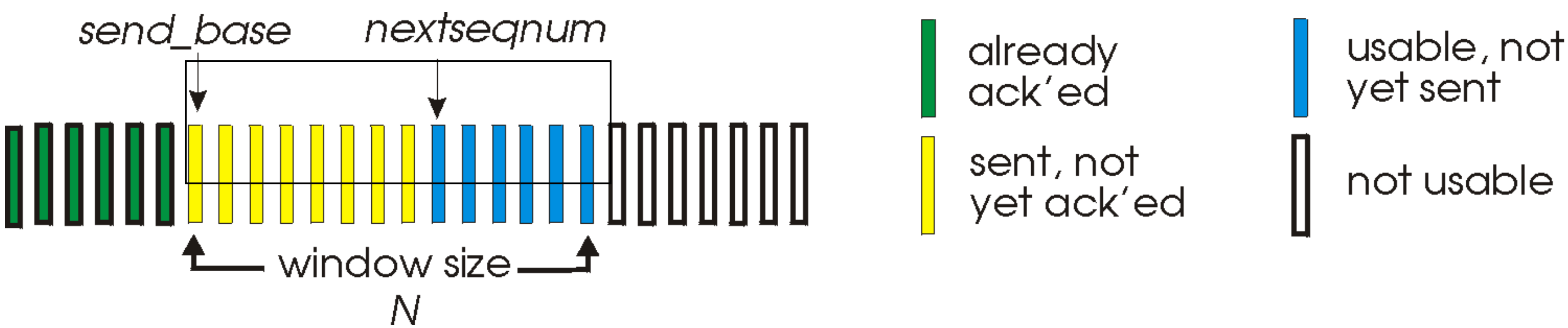
- sender can have up to N unacked packets in pipeline
- receiver only sends *cumulative ACK*
  - doesn't ACK packet if there's a gap
- sender has timer for oldest unacked packet
  - when timer expires, retransmit *all* unacked packets

Selective Repeat:

- sender can have up to N packets which it has not seen ACK for in "pipeline"
- Receiver sends *individual ACK* for each packet
- sender maintains timer for each packet with no ACK
  - when timer expires, retransmit only that unacked packet

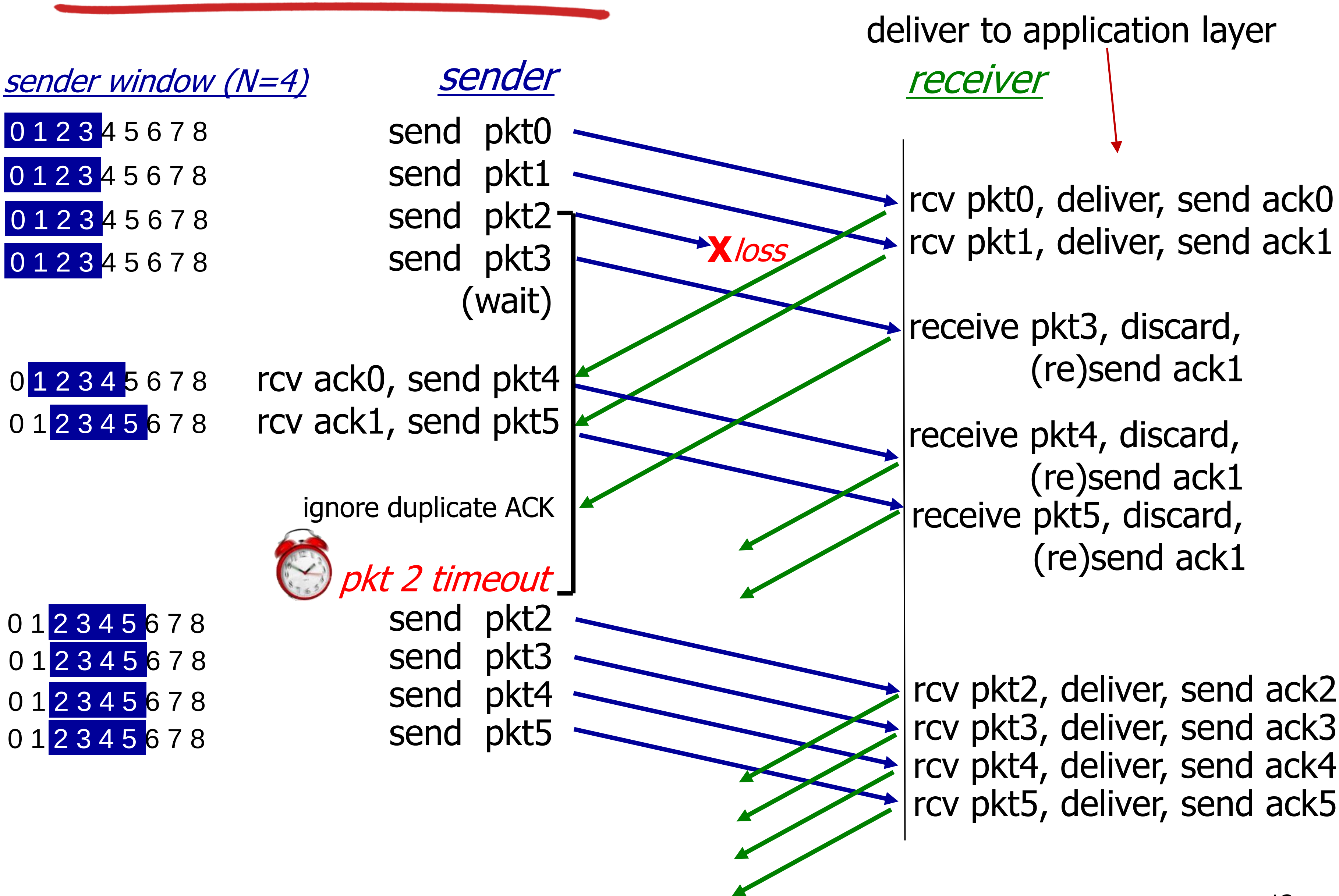
Go-Back-N: sender

- Now we need more sequence numbers k-bit sequence number in packet header
- "window" of up to N, consecutive packets with no ACK



- ACK(n): ACKs all packets up to, including sequence # 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

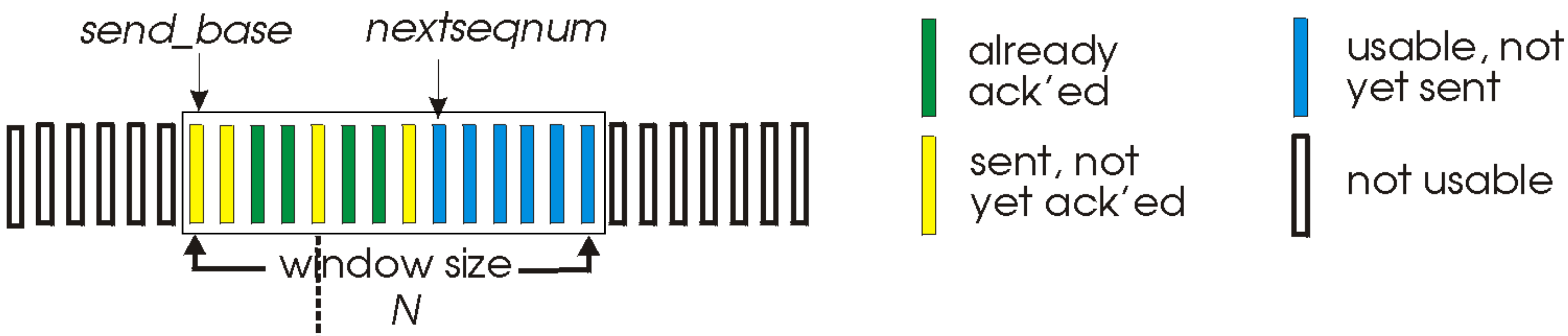
GBN in action



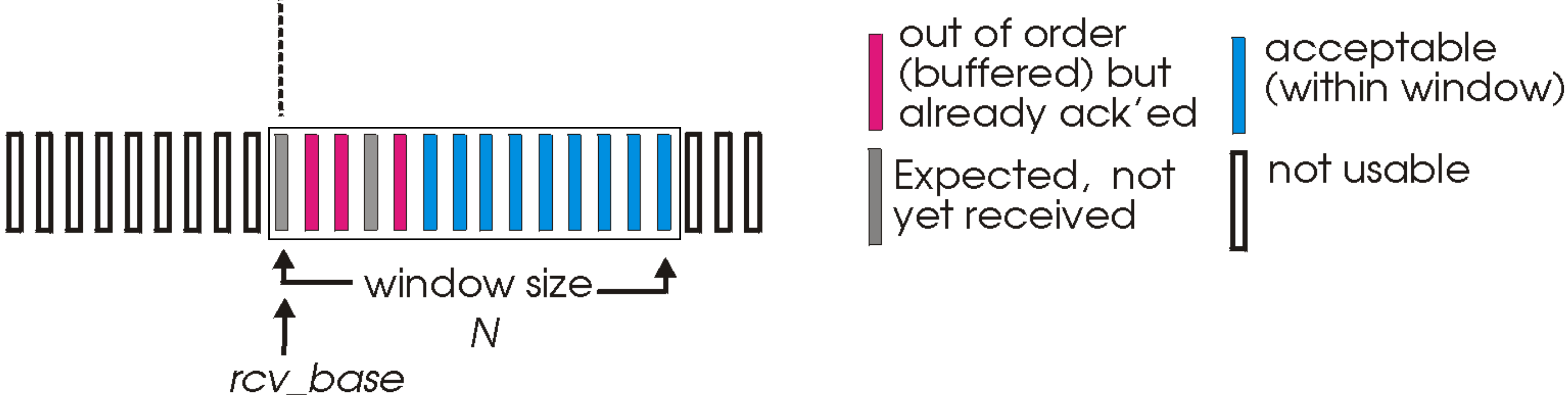
Selective repeat

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

Selective repeat: sender, receiver windows



(a) sender view of sequence numbers



(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 #

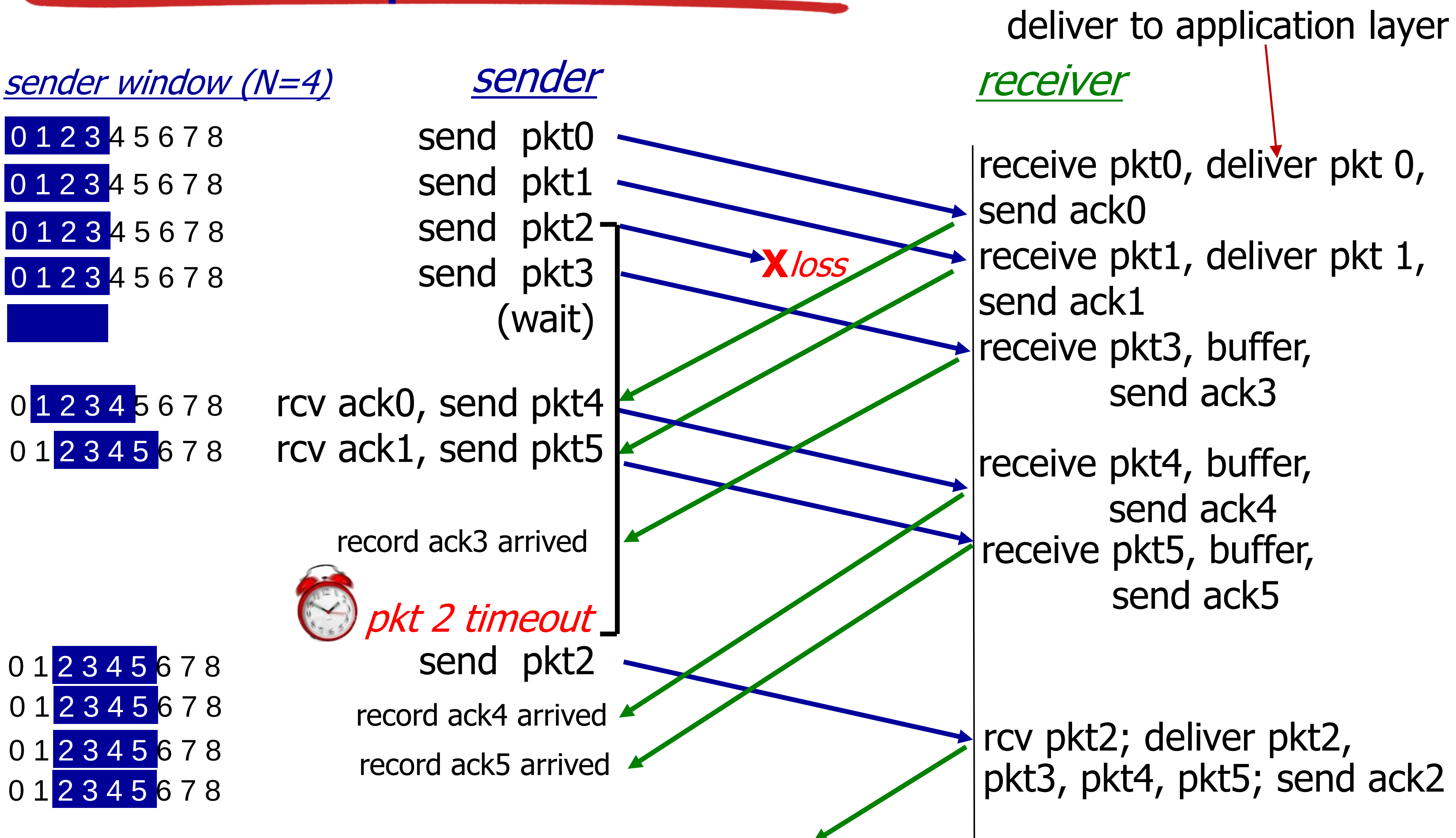
receiver

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-1]
- ACK(n)

otherwise:
- ignore

Selective repeat in action



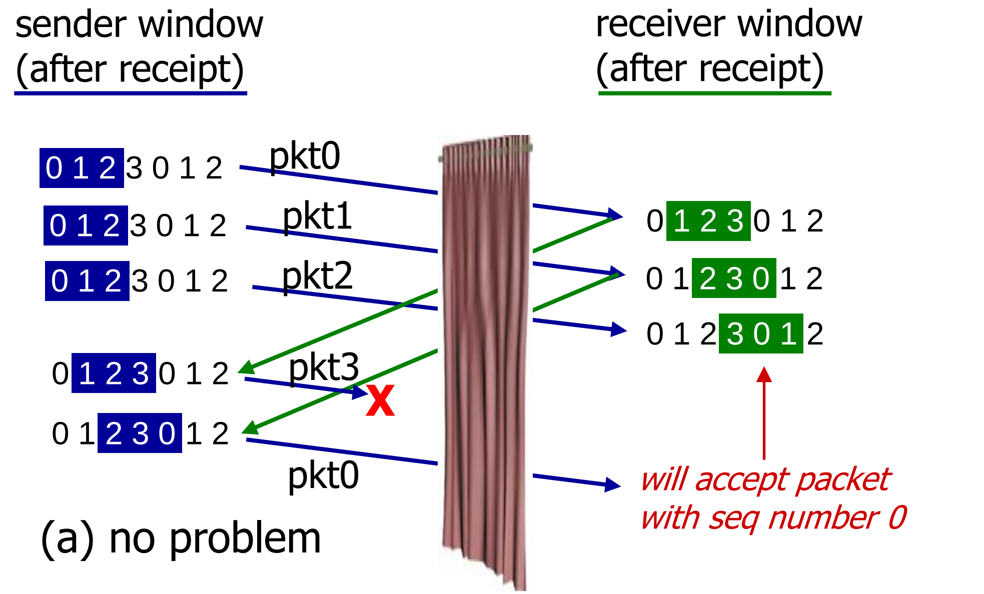
Q: what happens when ack2 arrives?

Selective repeat: dilemma

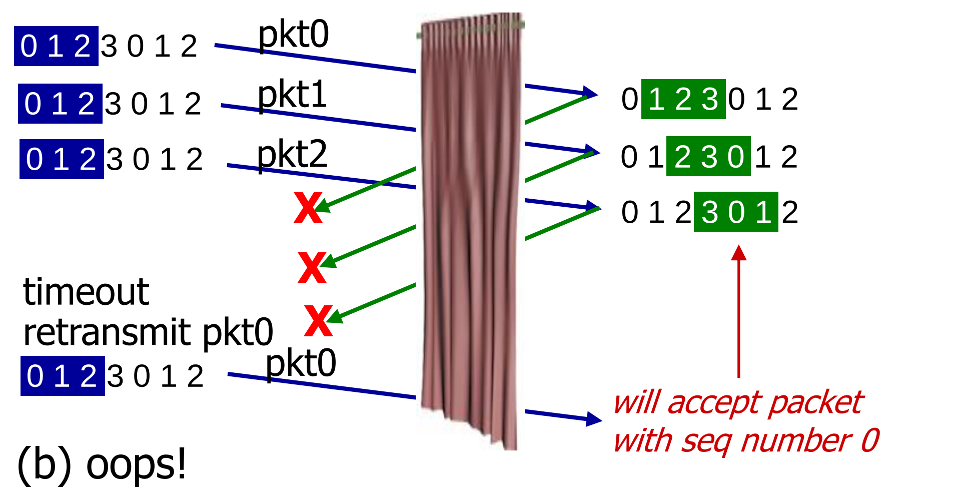
example:

- 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)?



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



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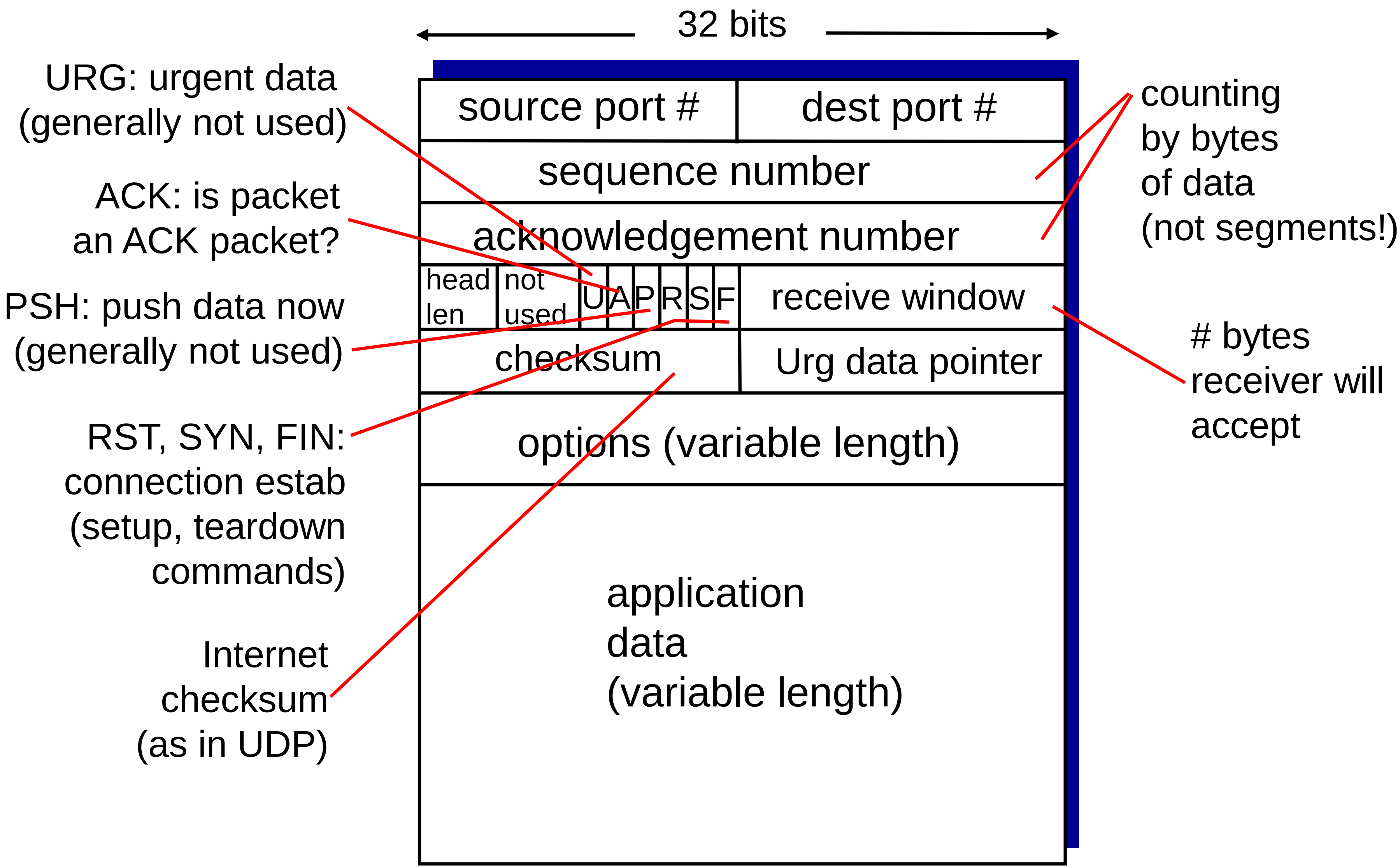
TCP: Overview RFCs: 793,1122,1323, 2018, 2581

- 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 header


32 bits
source port #
dest port #
sequence number
acknowledgement number
head len
not used
U A P R S F
receive window
checksum
Urg data pointer
options (variable length)
application data (variable length)
URG: urgent data (generally not used)
ACK: is packet an ACK packet?
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 receiver will accept

TCP seq. numbers, ACKs

sequence numbers:

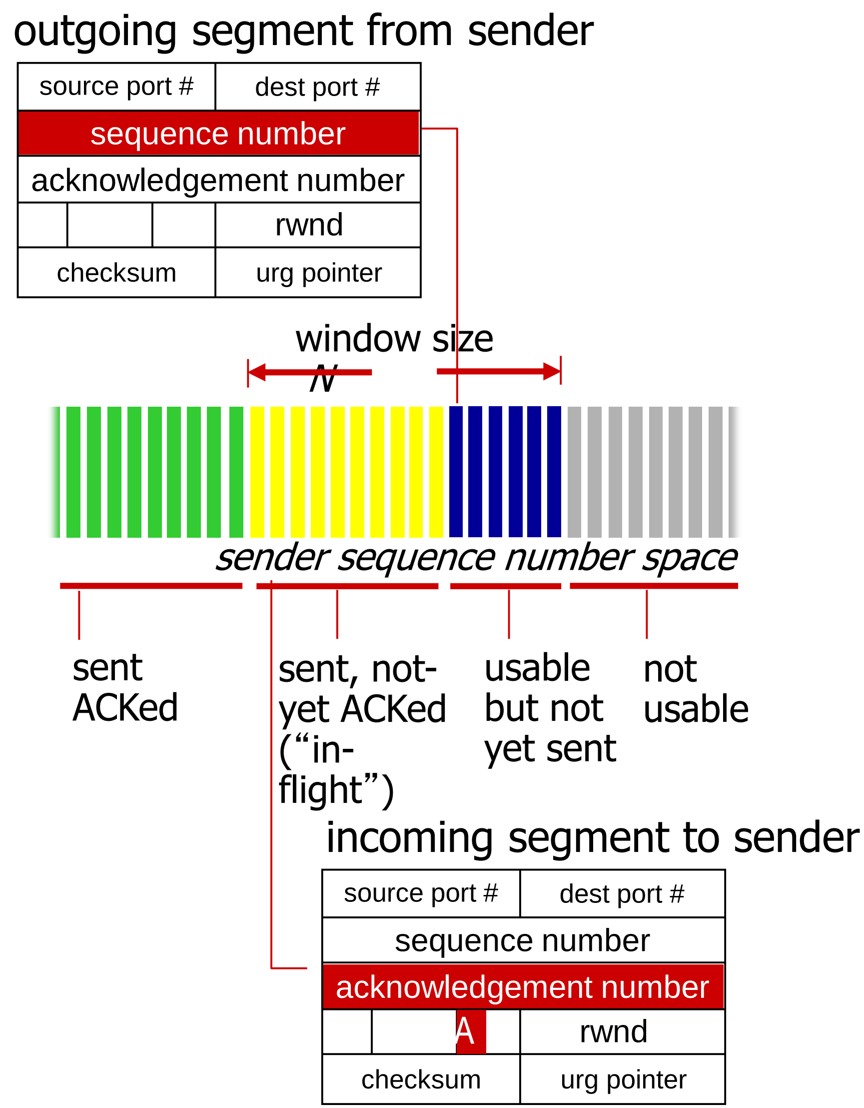
- byte stream "number" of first byte in segment's data

acknowledgements:

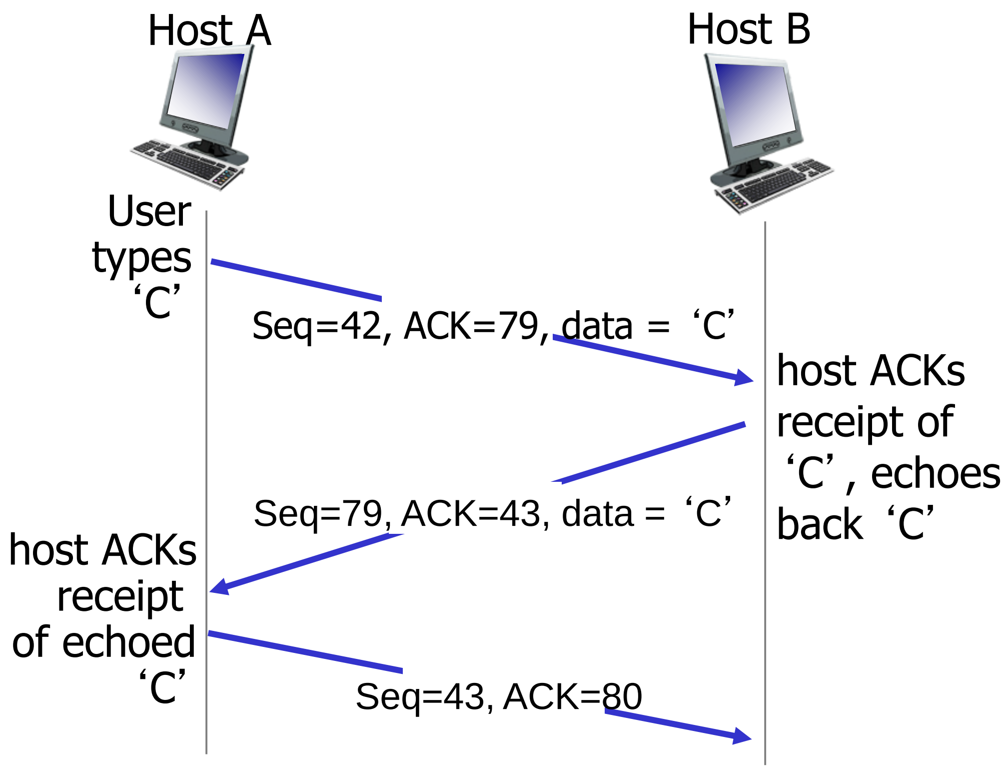
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

- A: TCP spec does not say, - up to those writing TCP code as long as data is delivered in order to application layer



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:* early timeout, unnecessary retransmissions
- *too long:* slow reaction to segment loss

Q: how to estimate RTT?

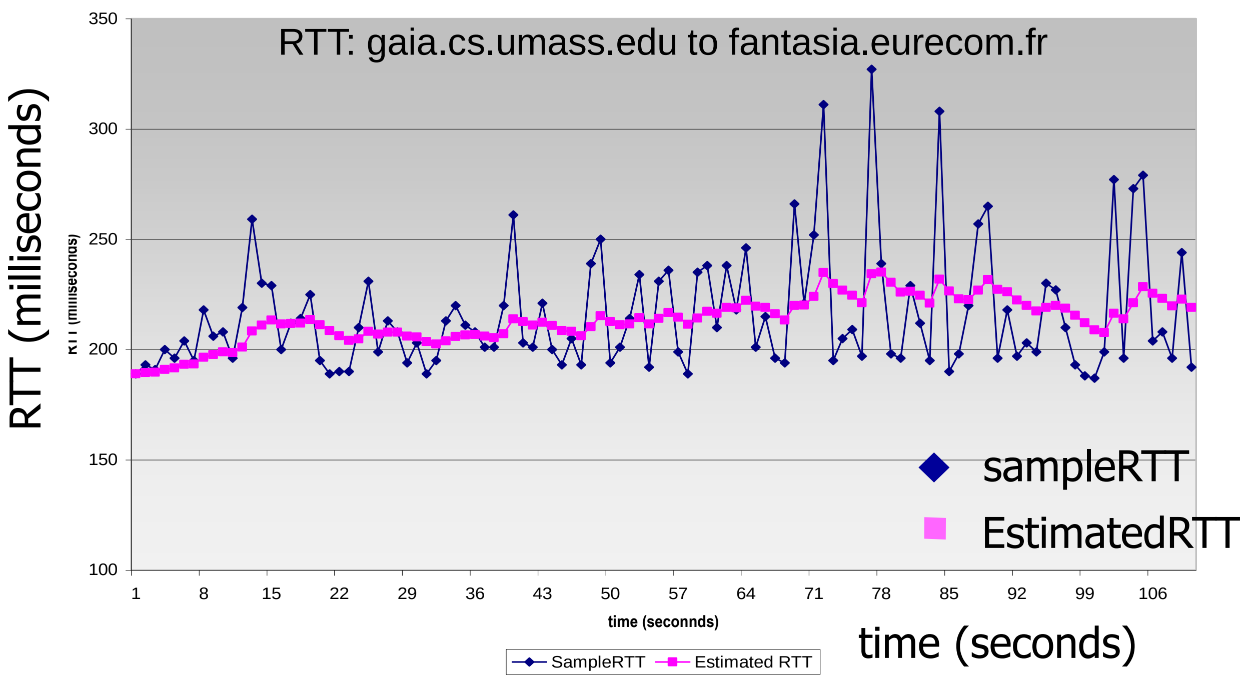
- **`SampleRTT`**: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- **`SampleRTT`** will vary, want estimated RTT "smoother"
  - average several *recent* measurements, not just current **`SampleRTT`**

TCP round trip time, timeout

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

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

EstimatedRTT_new = 0.875 · EstimatedRTT + 0.125 · SampleRTT



TCP round trip time, timeout

- Jacobsen/Karel's Algorithm
- 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)
```

Where |x| means the "absolute" value – x made positive.

```
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 received from application:

- 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`**

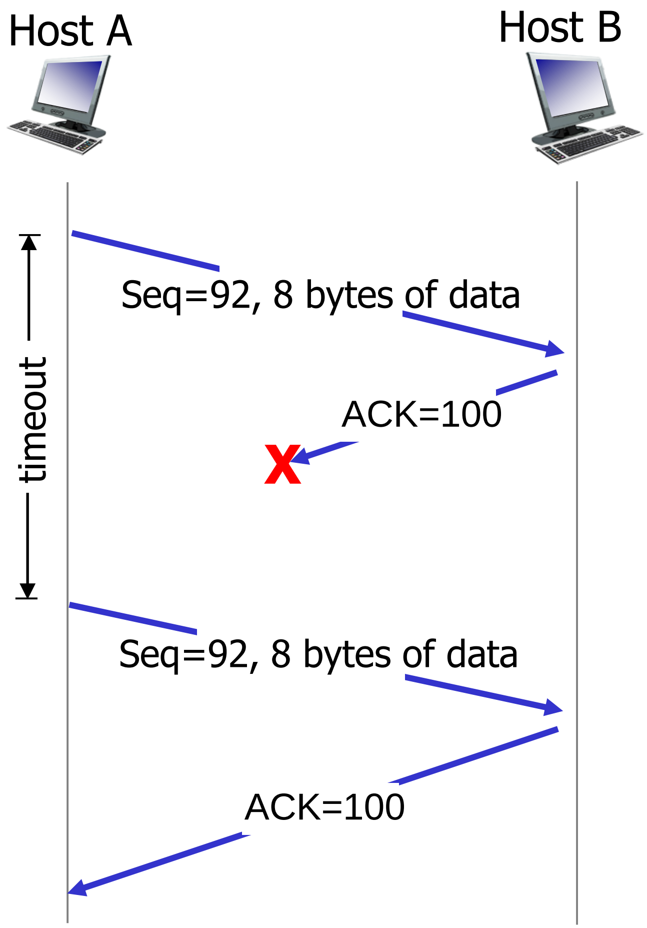
timeout:

- retransmit segment that caused timeout
- restart timer

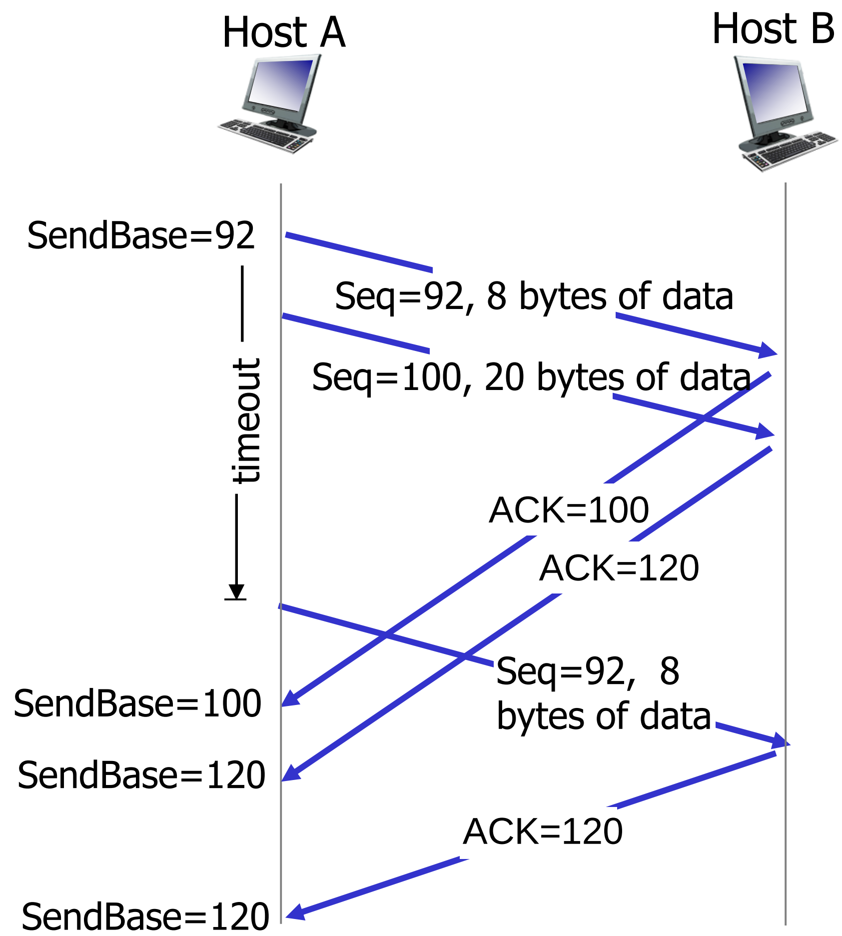
ack rcvd:

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

TCP: retransmission scenarios

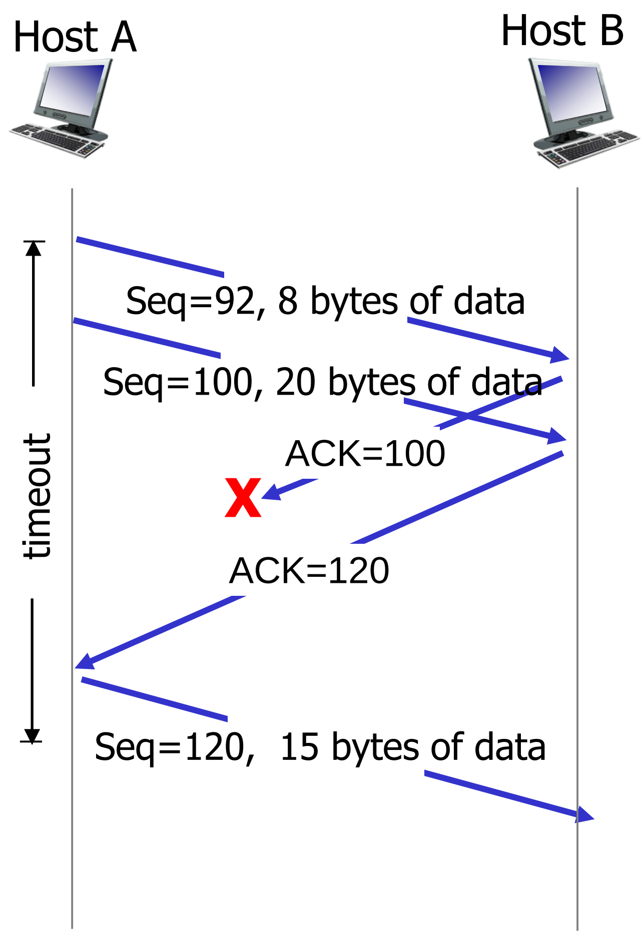


lost ACK scenario



premature timeout

TCP: retransmission scenarios



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.

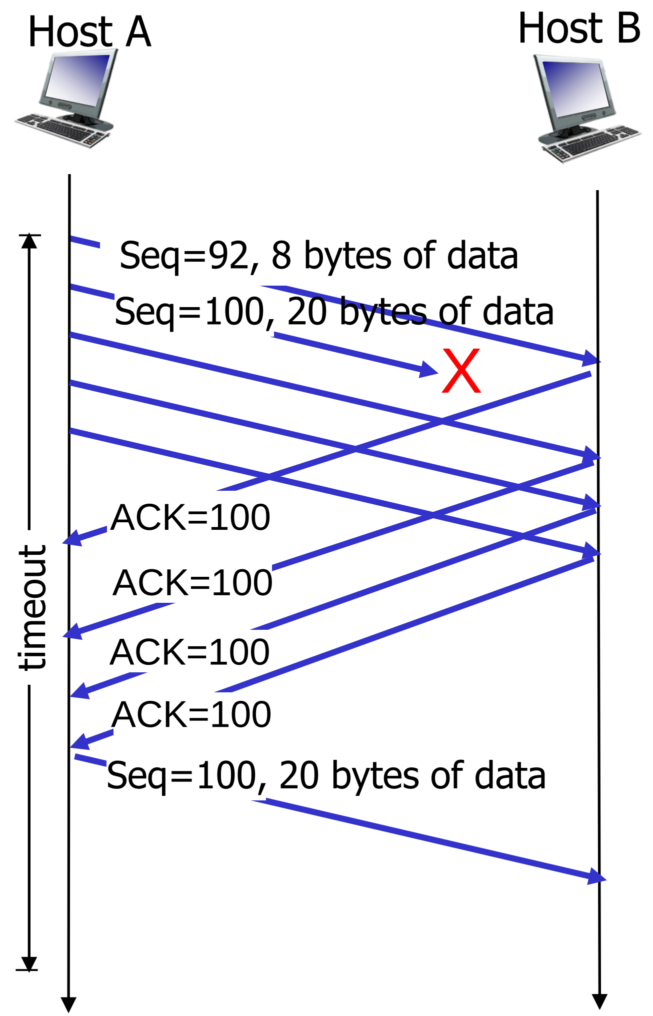
TCP fast retransmit

if sender receives 3 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

What have we learned?

- We learned about "pipelining" – sending more than one packet "in flight" at once. This makes transfer much more efficient.
- "Go back N" and "Selective repeat" – two ways to send several packets.
- We have seen how TCP uses SEQ and ACK to send the correct packets and retransmit missing ones.
- Triple duplicate ACK used to hint packet is “really lost” not just out of order – fast restransmit
- Jacobsen/Karel's Algorithm used to get "smooth" but “safe”estimate of RTT (and variance) and hence set timeout.

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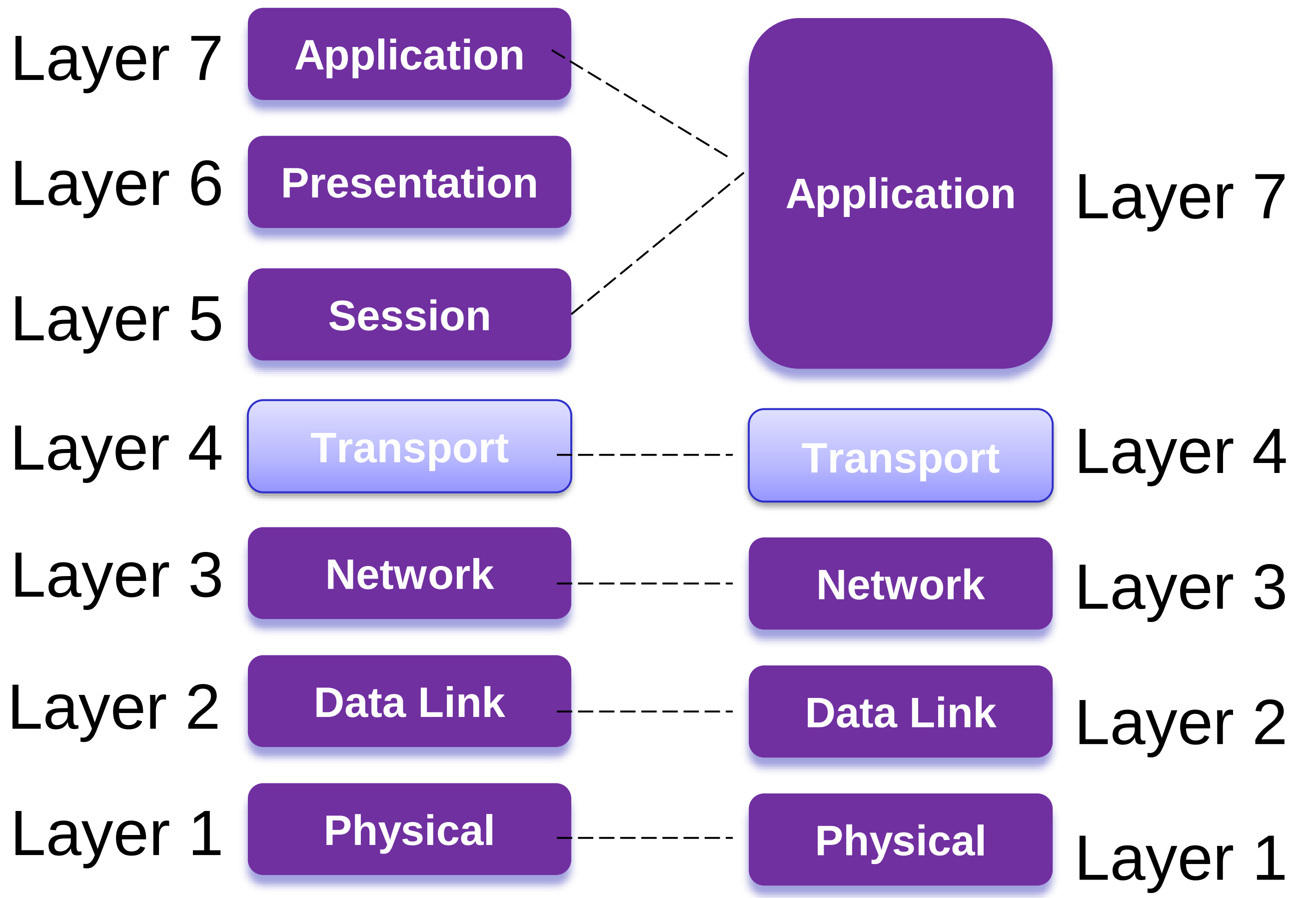
Transport Layer



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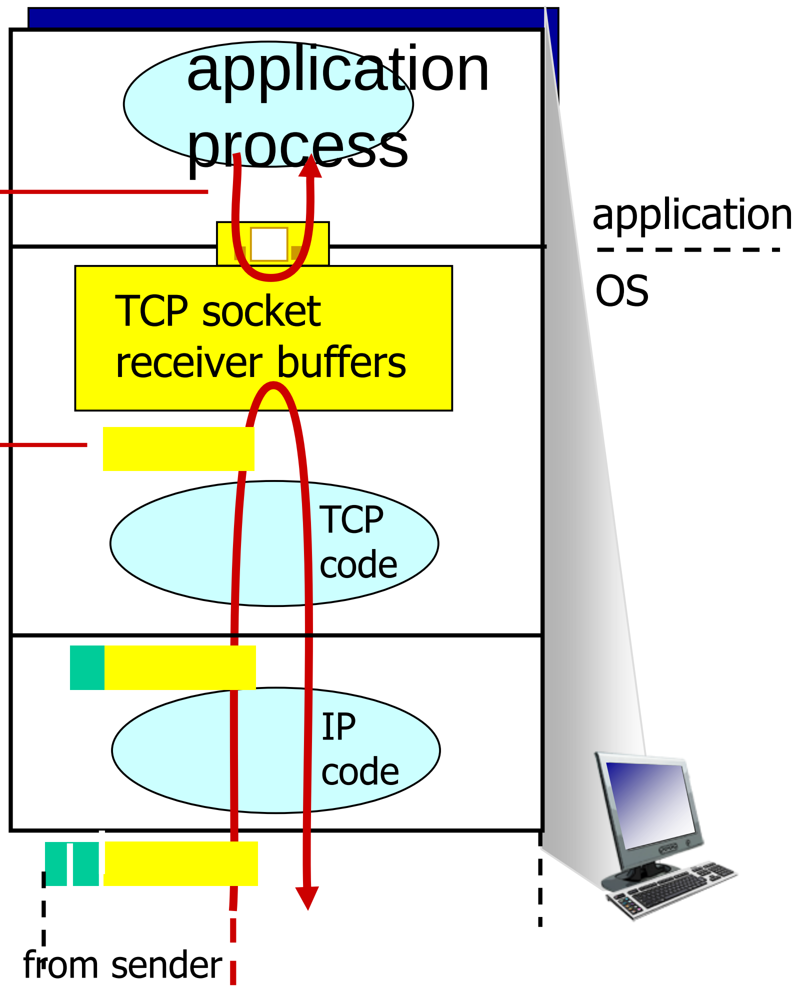
TCP flow control

application may remove data from TCP socket buffers ….

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



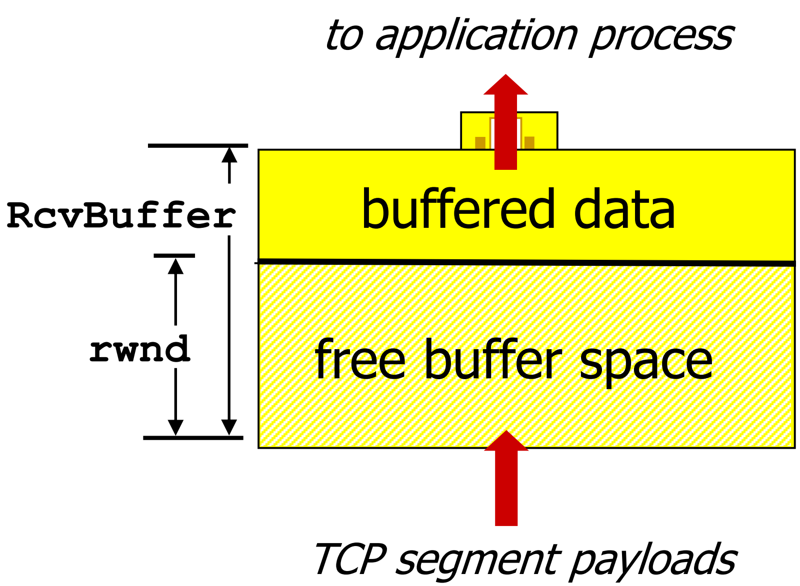
receiver protocol stack

flow control

receiver controls sender, so sender will not overflow receiver 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

Nagle's algorithm

- A problem can occur when an application generates data very slowly.
- Consider, ssh or telnet that generate data only when a user types.
- TCP will send the data as it arrives at the send buffer if there is space left in the send buffer.
- This means (for ssh/telnet) one packet sent every time user hits key.
- Overhead of this is huge (TCP header + IP header + frame header to send one byte).
- Cure is known as "Nagle's algorithm".

Nagle's Algorithm

1. The sending TCP sends the first piece of data it receives – no matter no small or large
2. Sending TCP accumulates data in the buffer and waits until one of the following before sending the segment:
   - The receiving TCP sends an acknowledgement
   - Data has accumulated to fill a maximum size segment
3. Repeat step 2

- Note: Sometimes Nagle's algorithm should be switched off – e.g. when fast interaction is vital and you want small packet sizes to be sent.

Nagle's Algoritm

```
When the application produces data to send
   if both the available data and the window ≥ MSS
       send a full segment
   else
       if there is unACKed data in flight
             buffer the new data until an ACK arrives
       else
             send all the new data now
```

Note: MSS is usually set to the size of the largest segment TCP can send without causing the local IP to fragment. That is, MSS is set to the maximum transmission unit (MTU) of the directly connected network, minus the size of the TCP and IP headers

Silly Window Syndrome

- Silly Window Syndrome occurs when the TCP system is forced to send very small packets. Named because window size is "silly".
- This can happen in two separate ways:

1. Sender produces data very slowly.
    - Same problem as Nagle's algorithm.
2. Receiver processes data very slowly.
    - Single byte or small number removed from full receive buffer.
    - Sender is informed of opportunity to send small number of bytes and immediately sends filling buffer.
    - Process repeats.

Silly Window Syndrome

- Sender side: If there is data to send but the window is open less than MSS (maximum segment size), how long should TCP wait for the window to be filled before sending?
  - Send as dictated by Nagle's algorithm if it is switched on.
  - If no ACK is expected or Nagle is switched off, send data.
- Receiver side: Avoid advertising very small window sizes by keeping a record and waiting until window size is sufficient for sender to send reasonable amount of data.

MSS and MTU

- MSS (mentioned in Nagle algorithm) is a parameter specifying the largest amount of data in a single IP datagram that should be sent by a remote host.
- MTU is a parameter specifying the largest amount of data that a communication protocol or system can pass onwards. For example, standards (e.g. Ethernet) can fix the size of an MTU, or systems (such as point-to-point serial links) may set MTU at connect time.
- MSS size is set according to MTU:

  MSS = MTU – IP header size – TCP header size.

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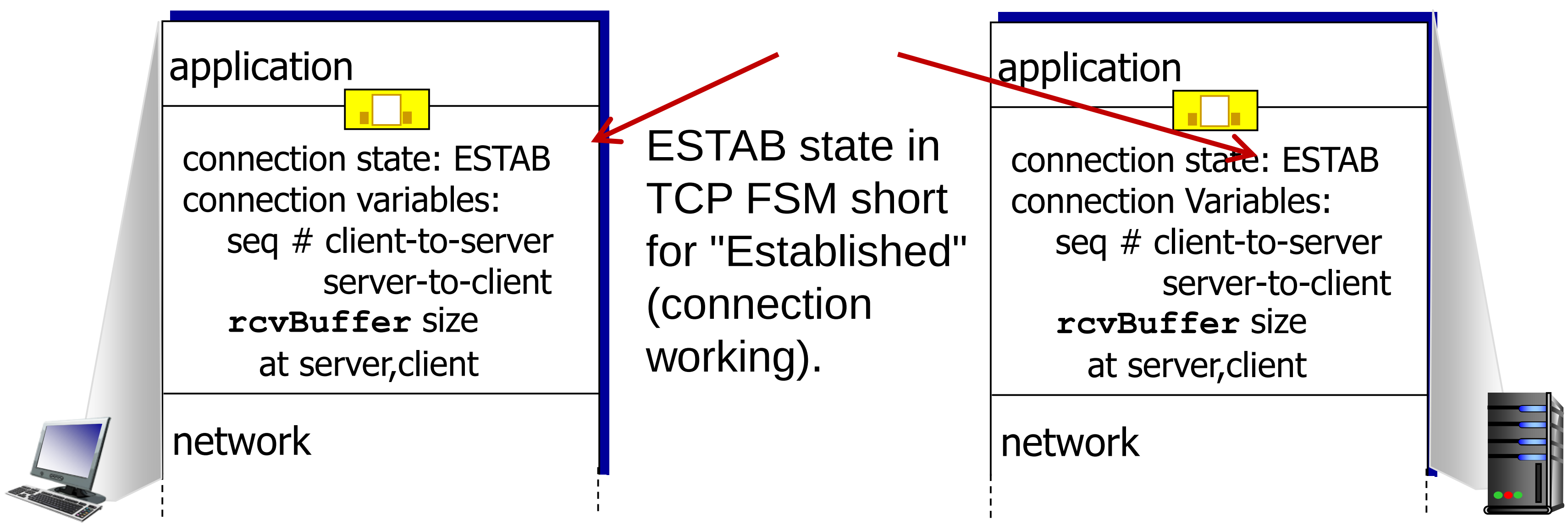
Connection Management

before exchanging data, sender/receiver "handshake":

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

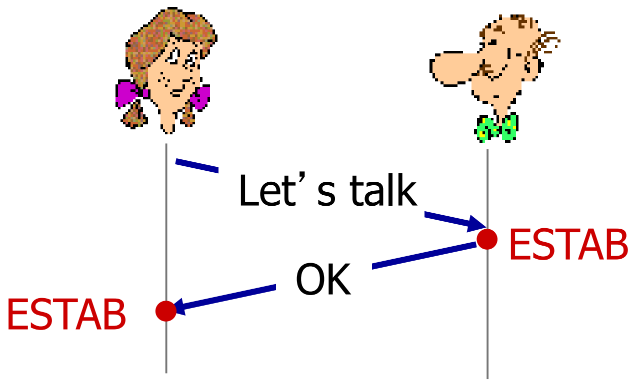


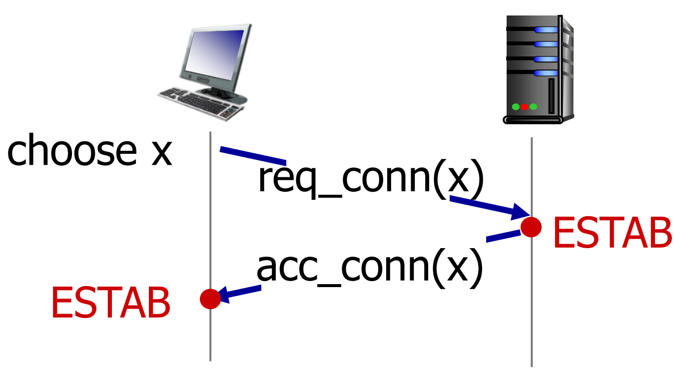

```
Socket clientSocket =
  newSocket("hostname","port
   number");
```

```
Socket connectionSocket =
  welcomeSocket.accept();
```

Agreeing to establish a connection

2-way handshake:



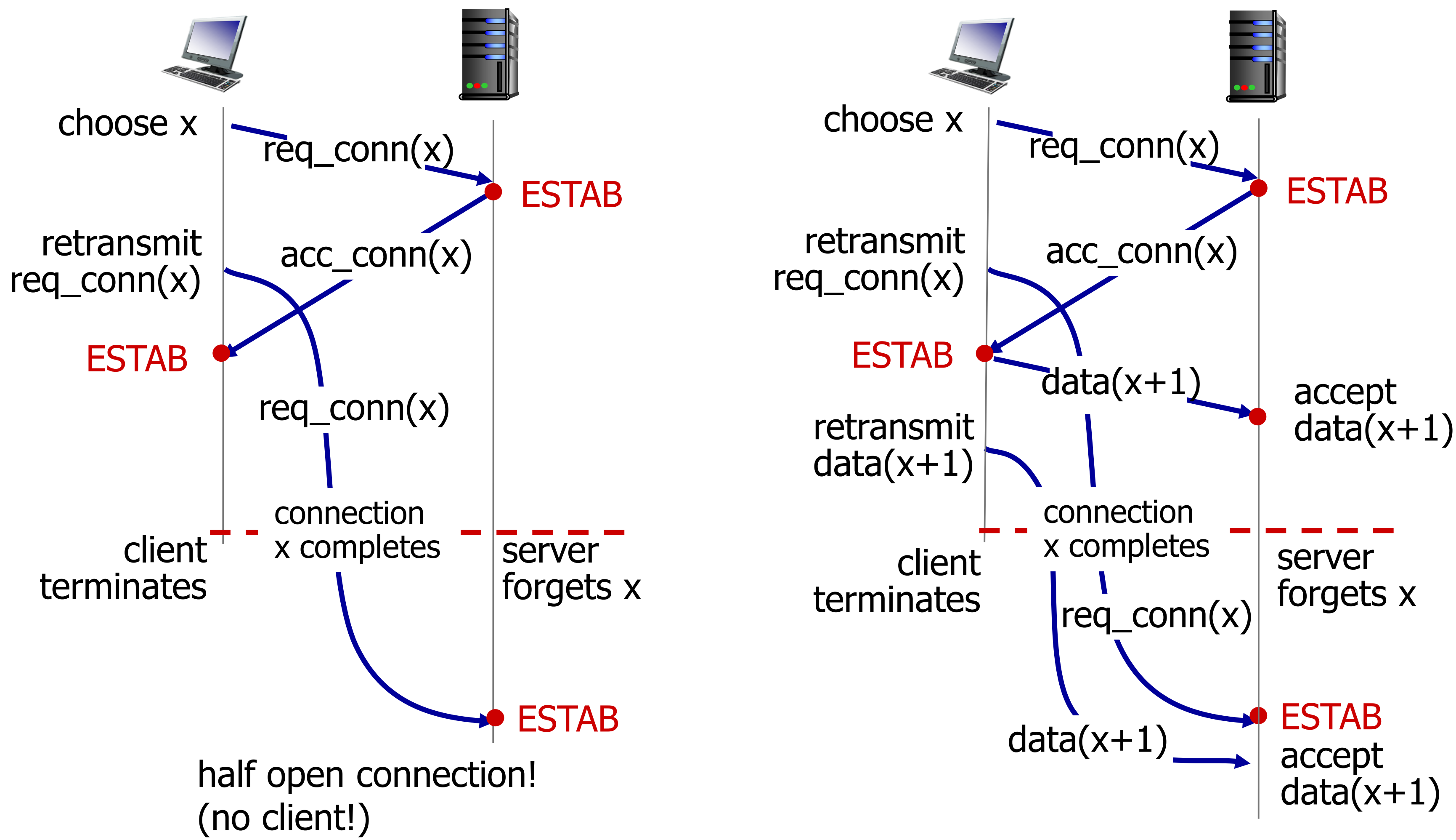


Q: will 2-way handshake always work in network?

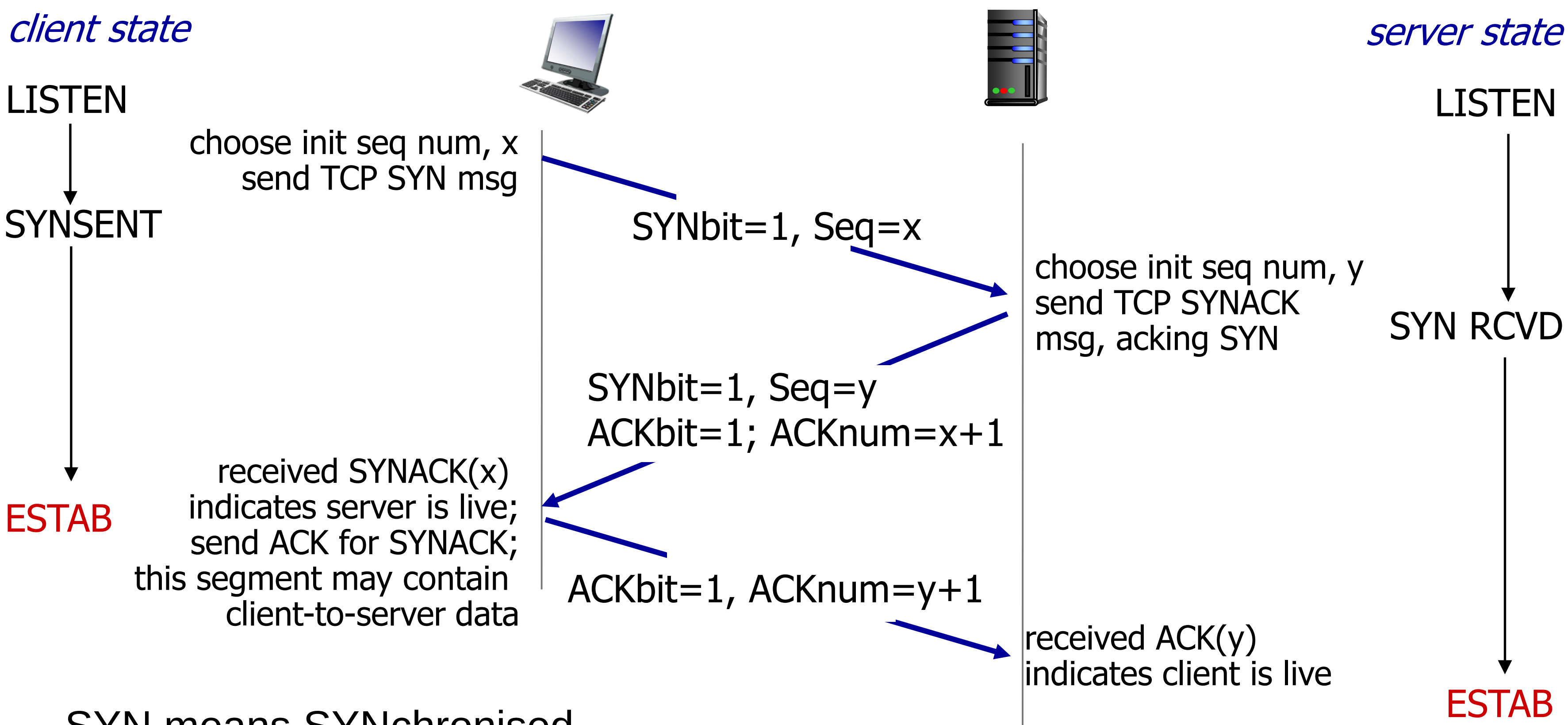
- variable delays
- retransmitted messages (e.g. req_conn(x)) due to message loss
- message reordering
- cannot "see" other side

Agreeing to establish a connection

2-way handshake failure scenarios:

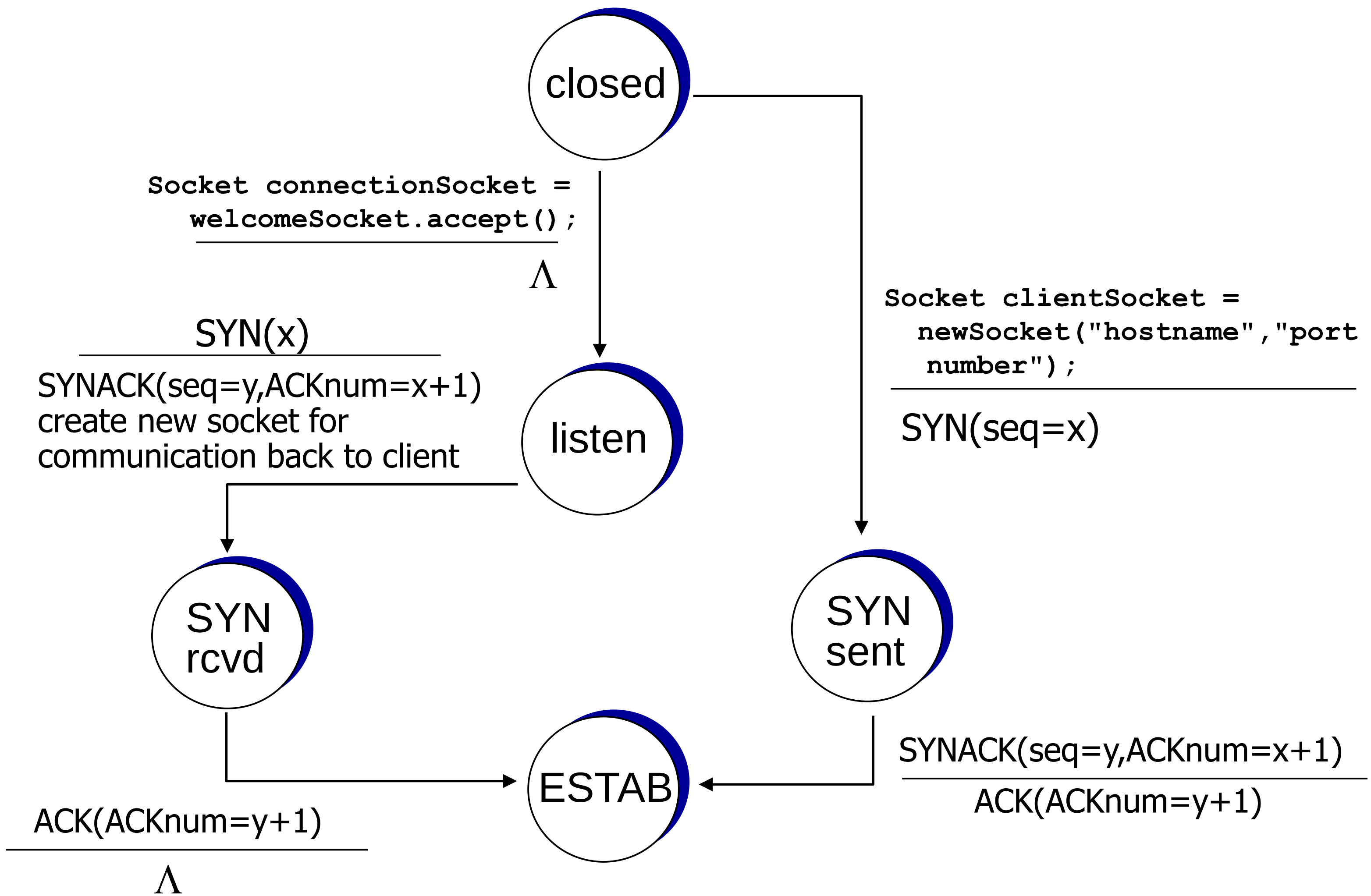


TCP 3-way handshake



SYN means SYNchronised
SYNbit = bit in TCP header saying this is SYN packet
ACKbit = bit in TCP header saying this is ACK packet

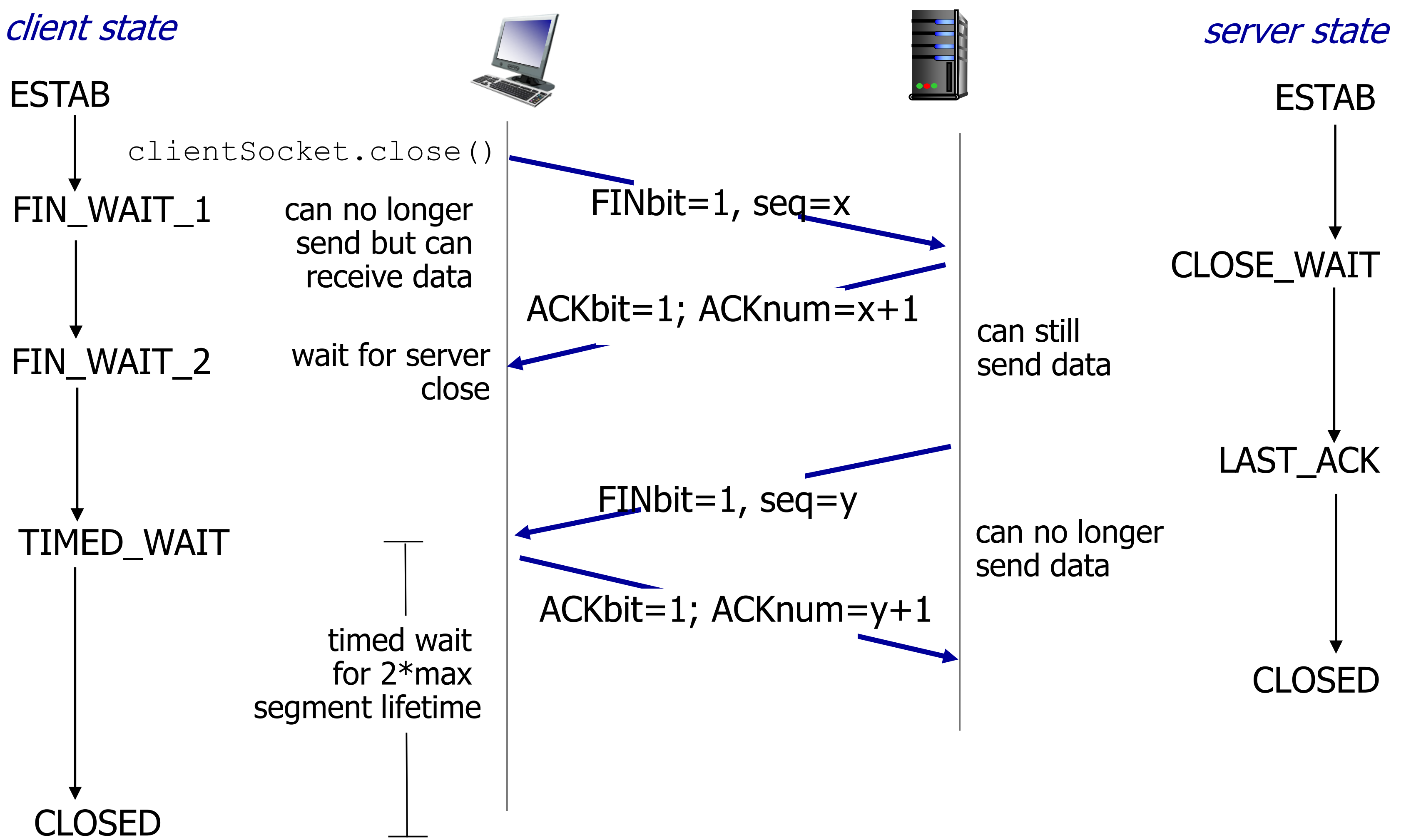
TCP 3-way handshake: Finite State Machine



TCP: closing a connection

- client, server each close their side of connection
  - send TCP segment with FIN bit = 1
- 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



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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!
- how does this look?
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Causes and costs of congestion

- Too much traffic enters router – buffer fills up, this increases delay (and hence reduces throughput).
- Much too much traffic enters router – buffer overfills and causes loss. Packet needs to be retransmitted.
- If packet is lost after several "hops" then many resources are wasted. (e.g. Packet travels from A to B to C to D then lost at D – it has taken up space at A, B and C unnecessarily).
- Useful concept: goodput – this is the rate at which data reaches the application layer. Different from throughput because of:
  - loss
  - retransmission
  - corrupted packets

Transport layer: 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

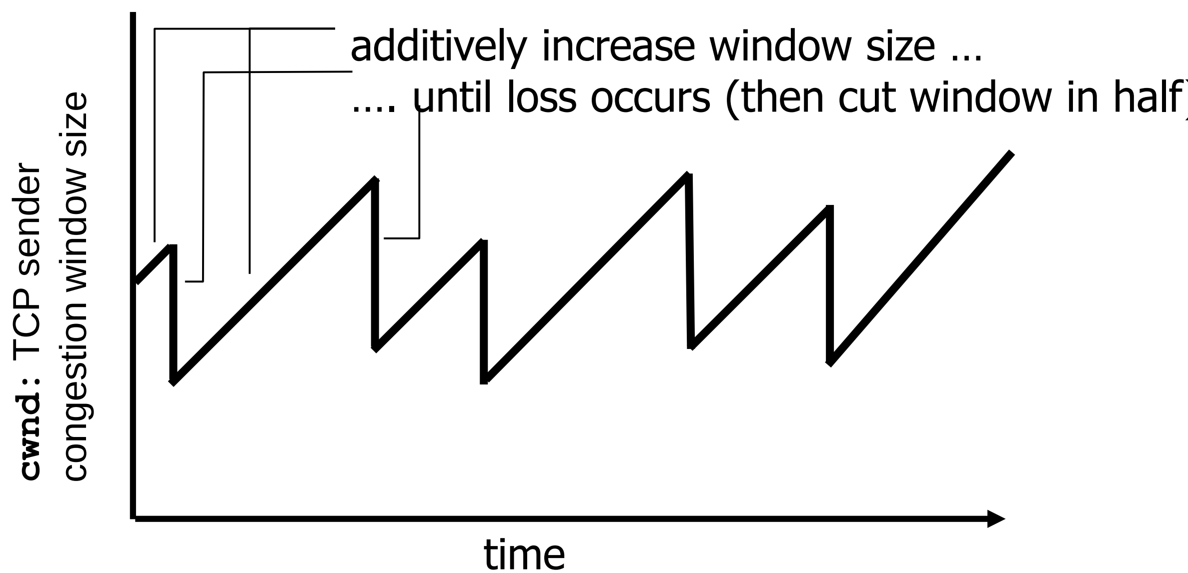
3.6 principles of congestion control

3.7 TCP congestion control

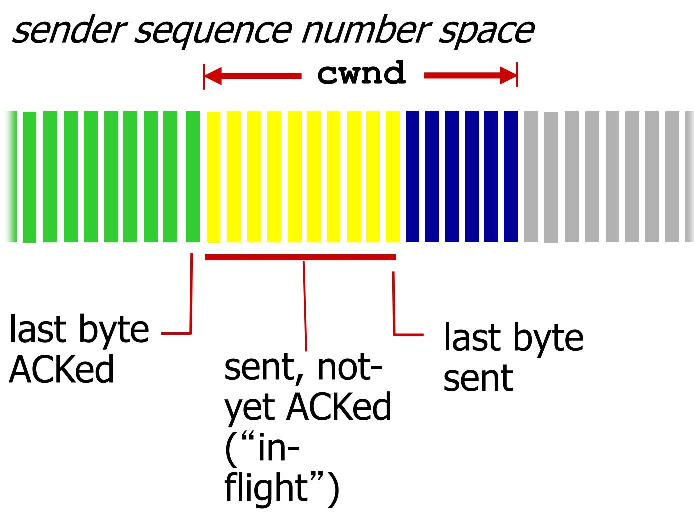
TCP congestion control: additive increase multiplicative decrease

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

AIMD saw tooth behavior: probing for bandwidth



TCP Congestion Control: details



- sender limits transmission:

$$\text{LastByteSent} - \text{LastByteAcked} \leq \text{cwnd}$$

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

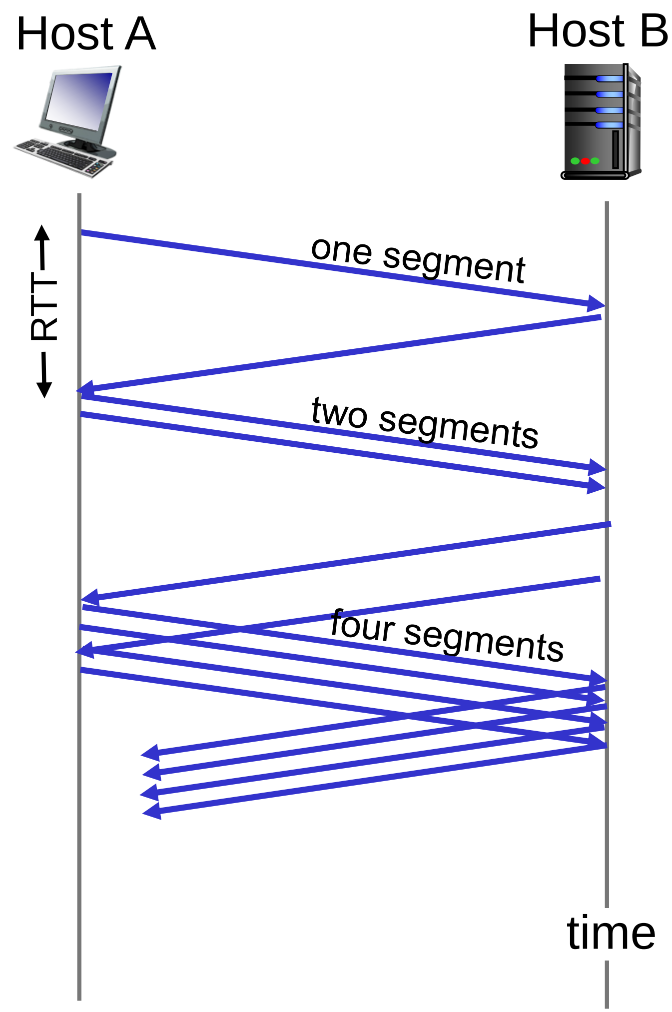
TCP sending rate:

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

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

TCP Slow Start

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



TCP flavours

- There are lots of implementations of TCP.
- The protocol specifies certain things but leaves others free.
- For example TCP protocols can choose how they want to react to duplicate ACKs or what their initial window sizes are.
- TCP protocols are sometimes named after places with casinos (gambling): Reno, Tahoe, New Reno

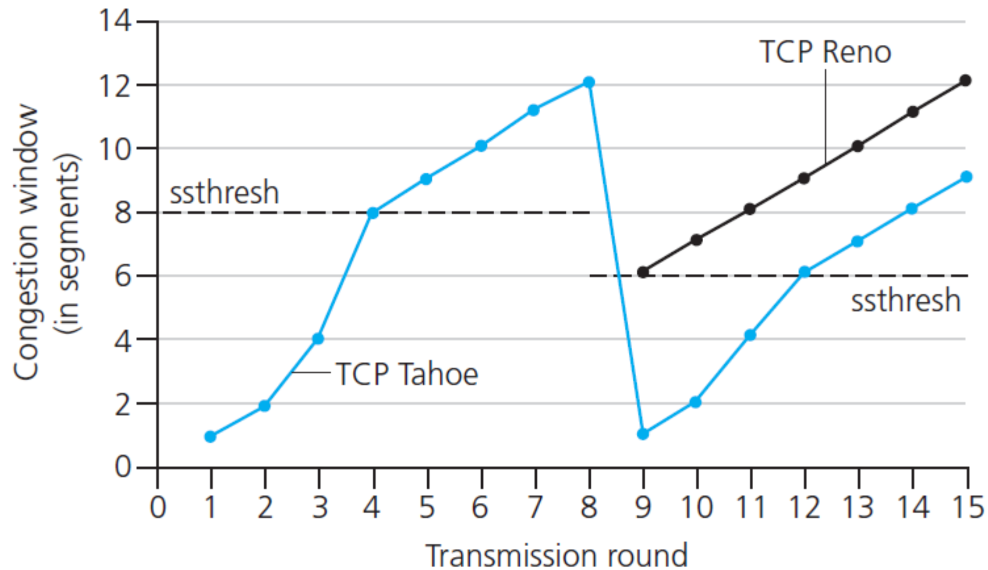
TCP: detecting, reacting to loss

- loss indicated by timeout:
  - **`cwnd`** set to 1 MSS;
  - window then grows exponentially (as in slow start) to threshold, then grows linearly
- loss indicated by 3 duplicate ACKs: TCP RENO
  - dup ACKs indicate network capable of delivering some segments
  - **`cwnd`** is cut in half window then grows linearly
- TCP Tahoe always sets **`cwnd`** to 1 (timeout or 3 duplicate acks)

TCP: switching from slow start to Congestion Avoidance

Q: when should the exponential increase switch to linear?

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



Implementation:

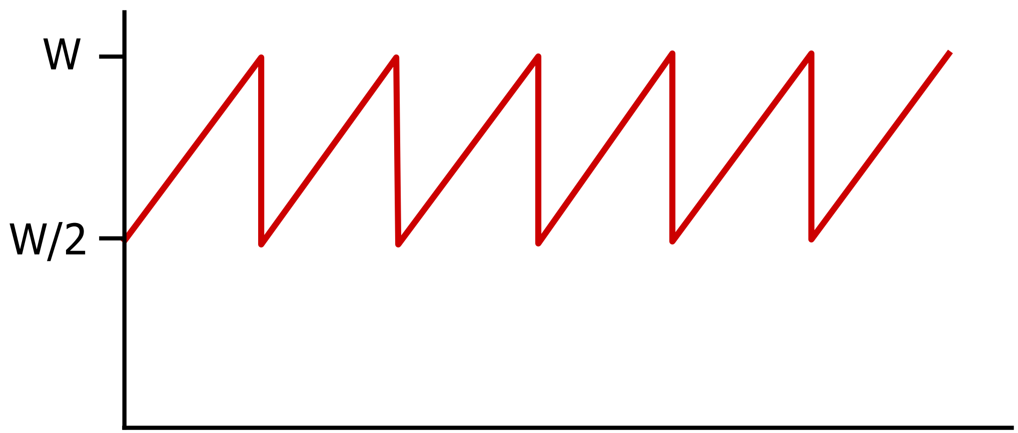
- variable **`ssthresh`**
- on loss event, **`ssthresh`** is set to 1/2 of **`cwnd`** just before loss event

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

TCP throughput

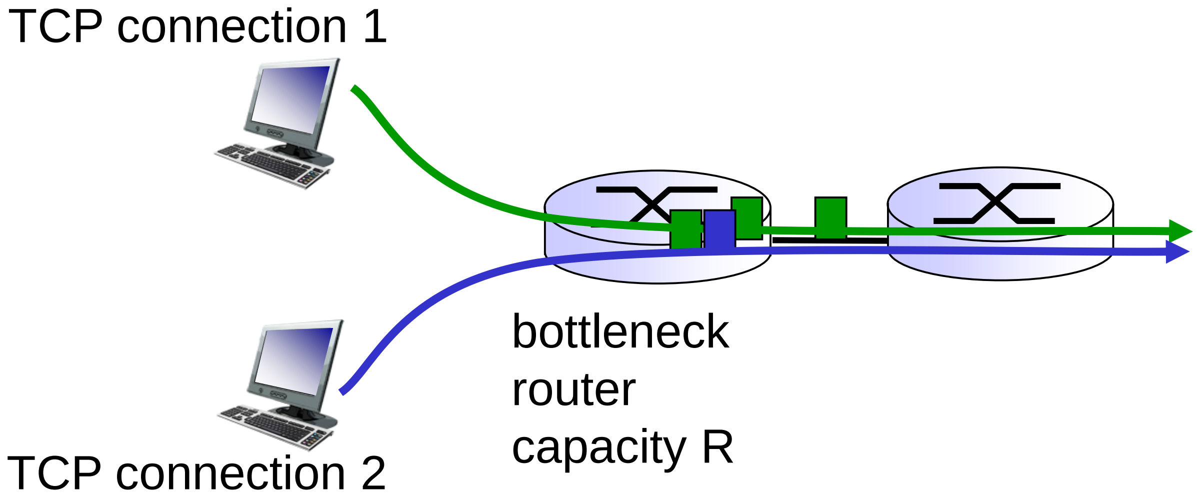
- avg. TCP thruput as function of window size, RTT?
  - ignore slow start, assume always data to send
- W: window size (measured in bytes) where loss occurs
  - avg. window size (# in-flight bytes) is ¾ W
  - avg. thruput is 3/4W per RTT

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



TCP Fairness

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



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

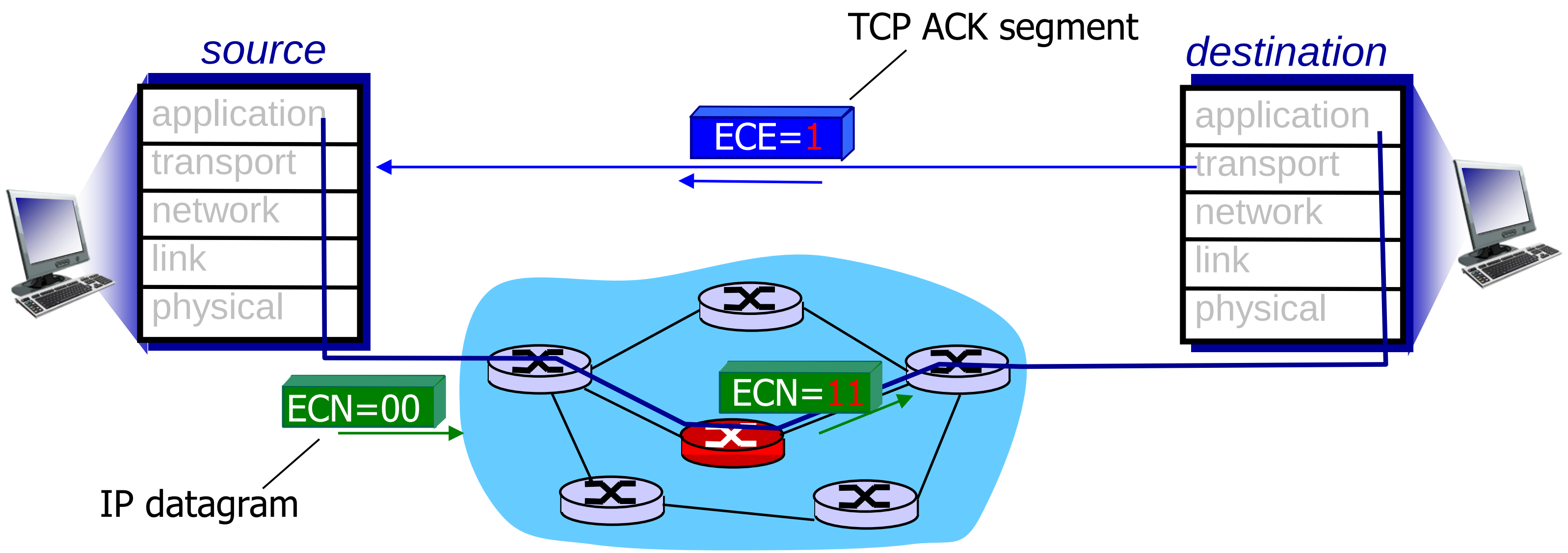
Fairness, parallel TCP connections

- 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 1 TCP, gets rate R/10
  - new app asks for 11 TCPs, gets R/2

Explicit Congestion Notification (ECN)

network-assisted congestion control:

- two bits in IP header (ToS field) marked *by network router* to indicate congestion
- congestion indication carried to receiving host
- receiver (seeing congestion indication in IP datagram)) sets ECE bit on receiver-to-sender ACK segment to notify sender of congestion



What have we learned?

- Completed our description of TCP
- Learned TCP mechanisms:
  - Set up of connection SYN – SYNACK – ACK
  - Closing a connection FIN – FINACK
  - Windows – used to control data "in flight" in the network.
  - Flow control – controlled by receiver limits window for connection. Stops receiver getting too much traffic.
  - Congestion control – reacts to network itself being overloaded. Stops network getting too much traffic.

Transport Layer: summary

- principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- Two protocols in the Internet
  - UDP
  - TCP

next:

- leaving the network "edge" (application, transport layers)
- into the network "core"
- Network layer chapters:
  - data plane
  - control plane