Internet Protocols EBU5403
The Transport Layer Part 2

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	Part 1	Part 2	Part 3	Part 4
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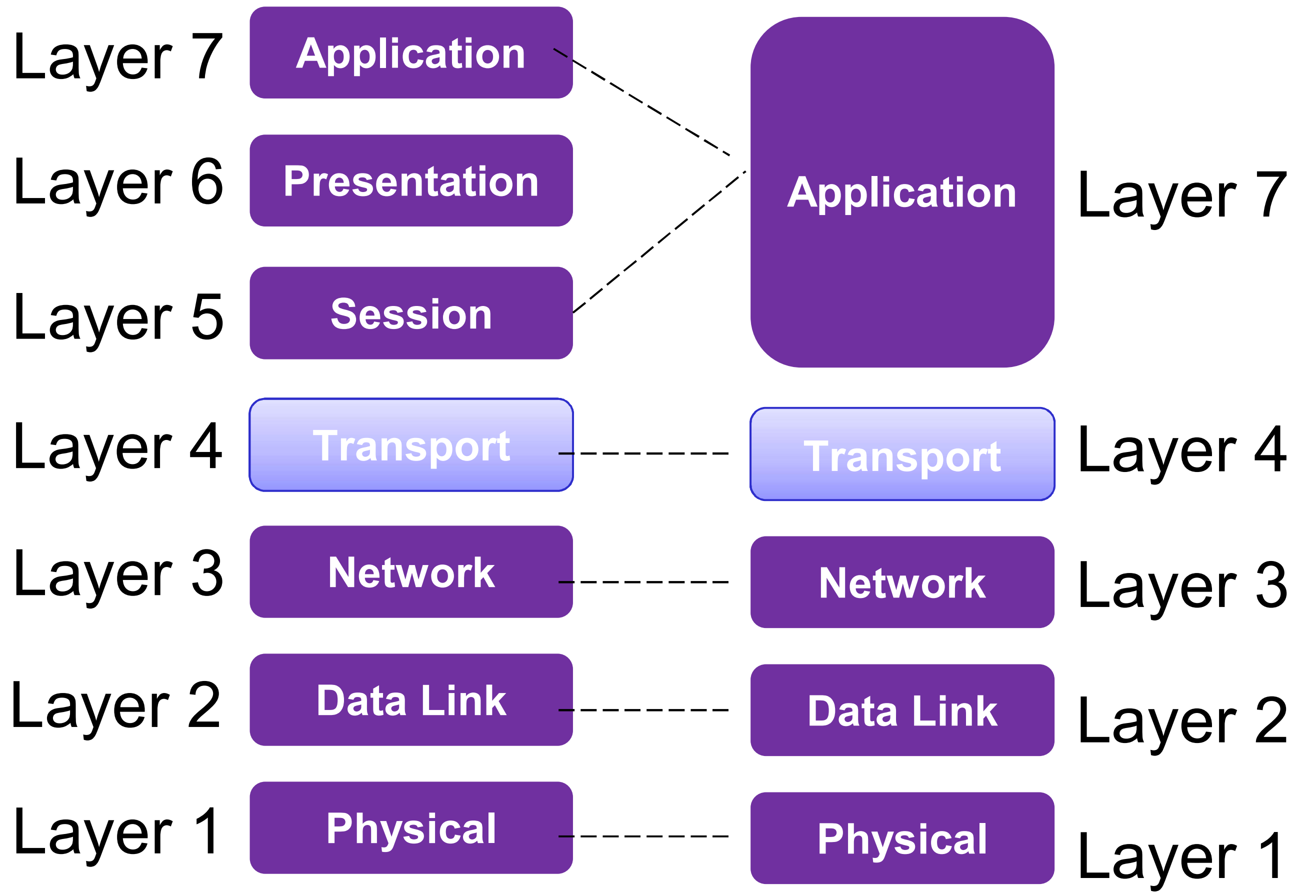
Structure of course

- Part A
  - Introduction to IP Networks
  - The Transport layer (part 1)
- Part B
  - The Transport layer (part II)
  - The Network layer (part I)
  - Class test
- Part C
  - The Network layer (part II)
  - The Data link layer (part I)
  - Router lab tutorial (assessed lab work after this week)
- Part D
  - 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.

Test your understanding

- Which version of rdt?
  - Assumes no losses or corruption
  - Works when packets can be corrupted but not lost.
  - Works with corruption and packets being lost or delayed.
  - Pause the video to write down your answer.

Test your understanding

- Which version of rdt?
  - Assumes no losses or corruption
    - rdt 1.0 – assumes a reliable channel
  - Works when packets can be corrupted but not lost.
    - rdt 2.2 – introduces ACK and sequence number to deal with corruption
  - Works with corruption and packets being lost or delayed.
    - rdt 3.0 – introduces timeout to cope with loss
  - If you got it wrong review rdt.

Performance of rdt3.0

- rdt3.0 is correct, but performance is very bad
- e.g.: 1 Gbps link, 15 ms delay, 8000 bit packet
- Define round-trip-time as time to propagate there and back -- (2 x end-to-end delay)

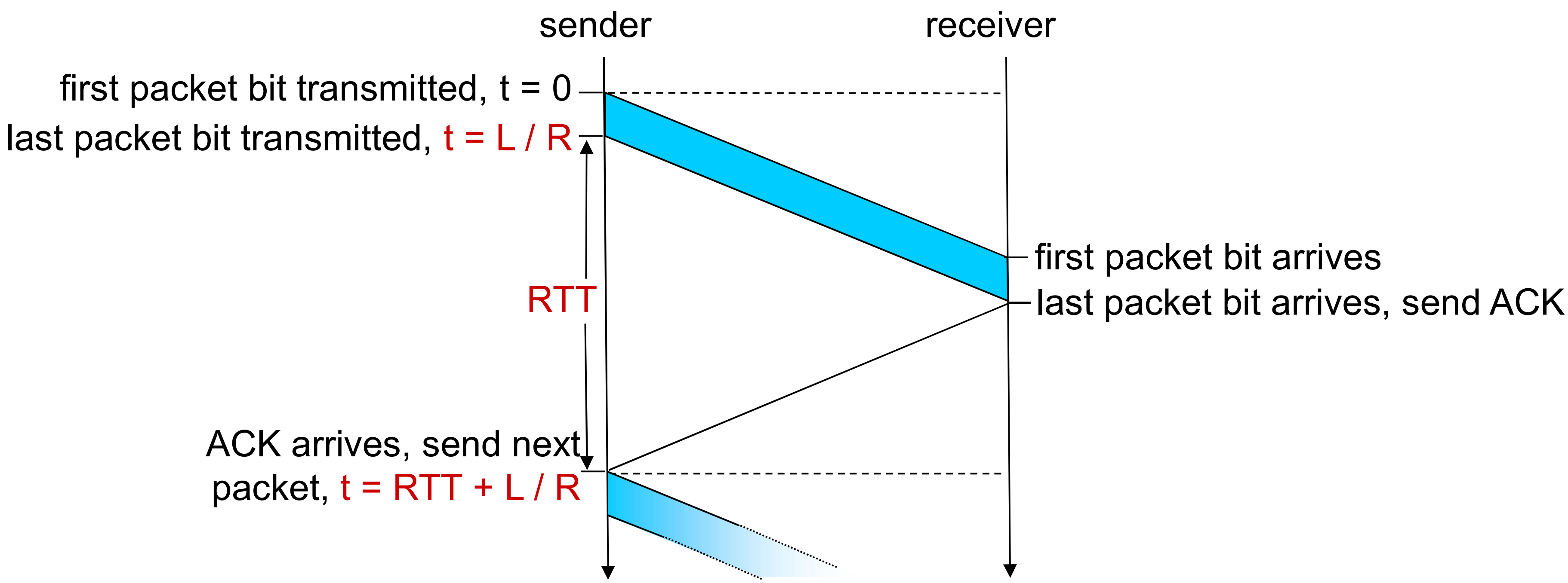
$$D_{trans} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 8 \text{ microsecs}$$

  - 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 limts how we use resource.

rdt3.0: stop-and-wait operation

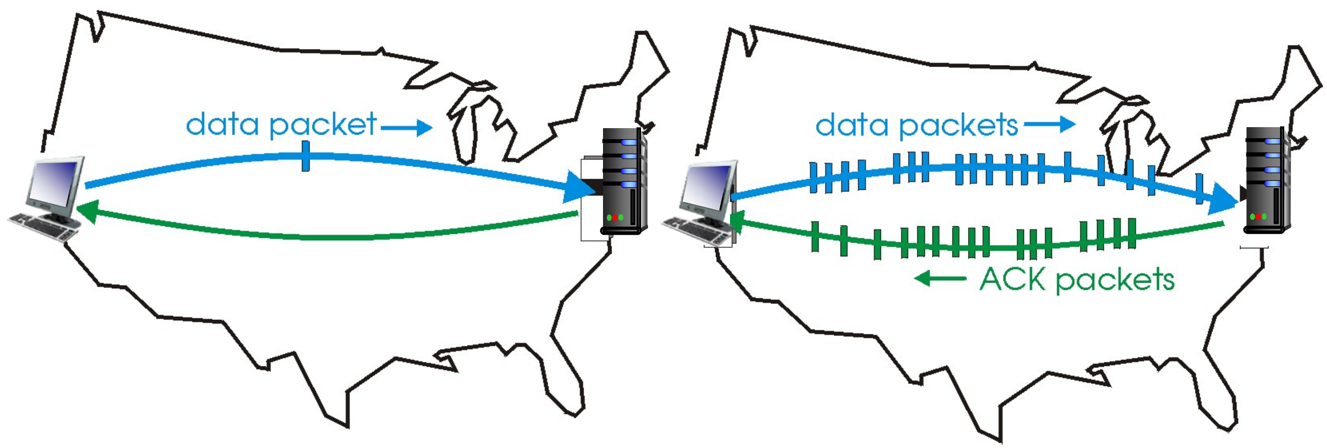


$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols

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

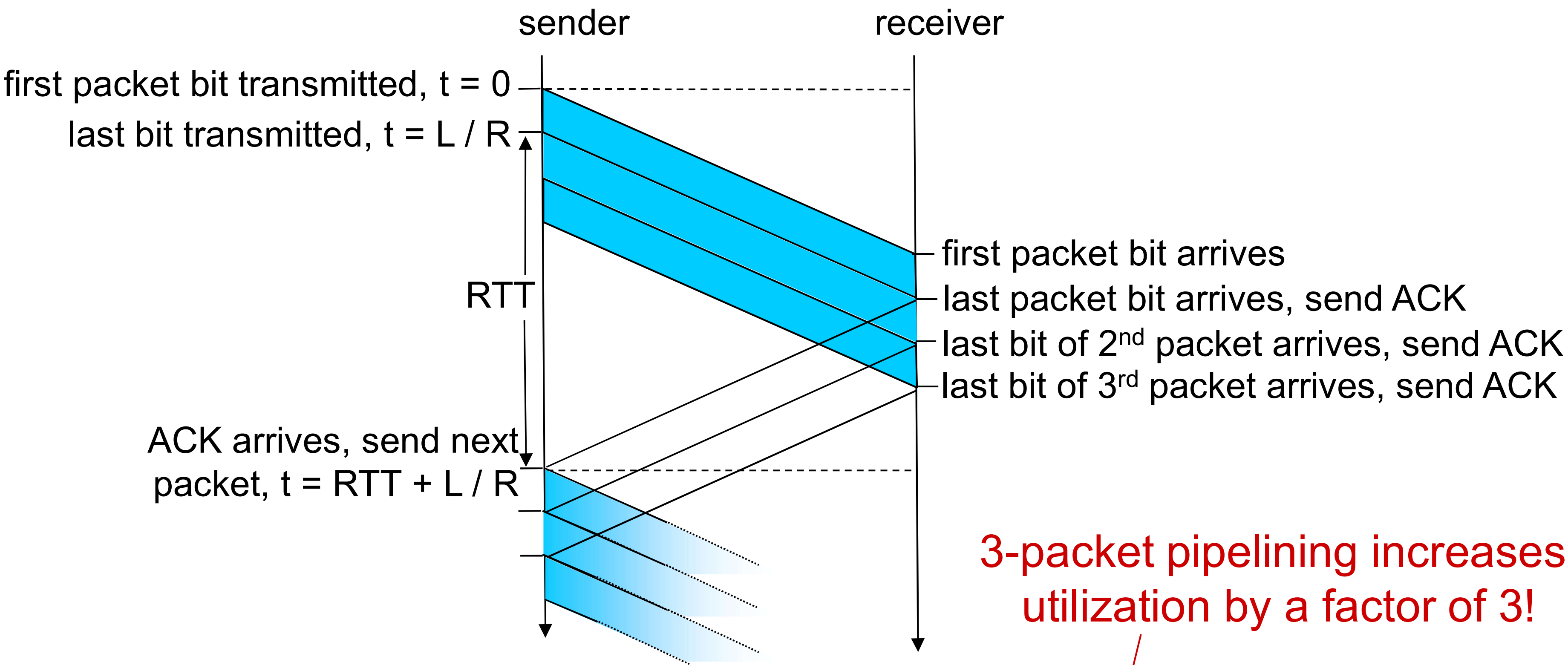
- range of sequence numbers must be increased
- buffering at sender and/or receiver



(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!

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

Test your understanding

- 1250 byte packet. 100 Mb/s link.
- End to end delay = 5ms

- What is the utilisation if we use stop and wait.

- What is the utilisation if we pileline 25 packets.

Test your understanding

- 1250 byte packet. 100 Mb/s link.
- End to end delay = 5ms

  L= 1250x8 bits = 10,000 bits

  Transmission delay = L/R = 10,000/100,000,000 = 0.0001 s = 0.1ms
- What is the utilisation if we use stop and wait.
- U = (L/R)/ (RTT+L/R) = 0.1/(10 + 0.1) ~ 0.01*
- What is the utilisation if we use stop and wait.
- U= 25* (L/R)/ (RTT+L/R) ~0.25
  If you did't get it review slides 8-11

Pipelined protocols: overview

Go-back-N:

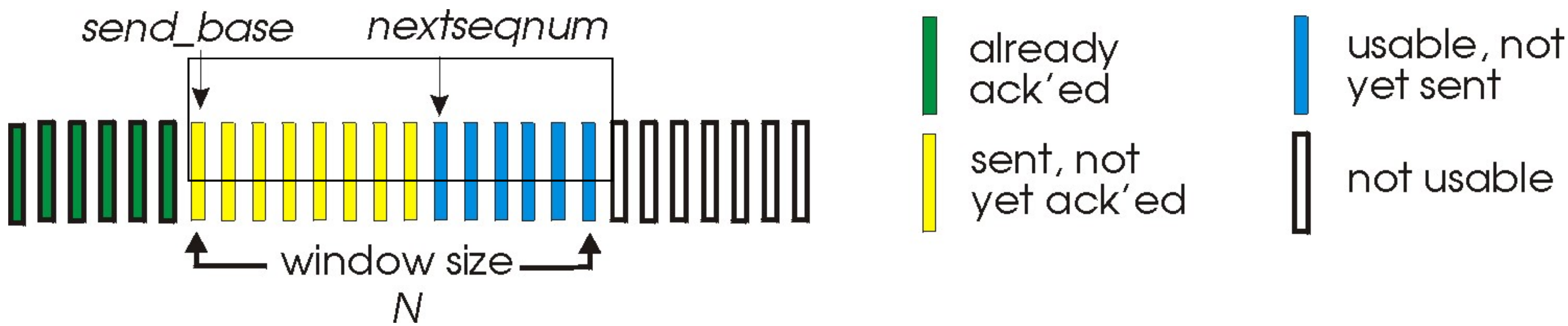
- sender can have up to N unacked packets in pipeline
- receiver only sends *cumulative ACK*
  - doesn't ACK packet if there's a gap
- sender has timer for oldest unacked packet
  - when timer expires, retransmit *all* unacked packets

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

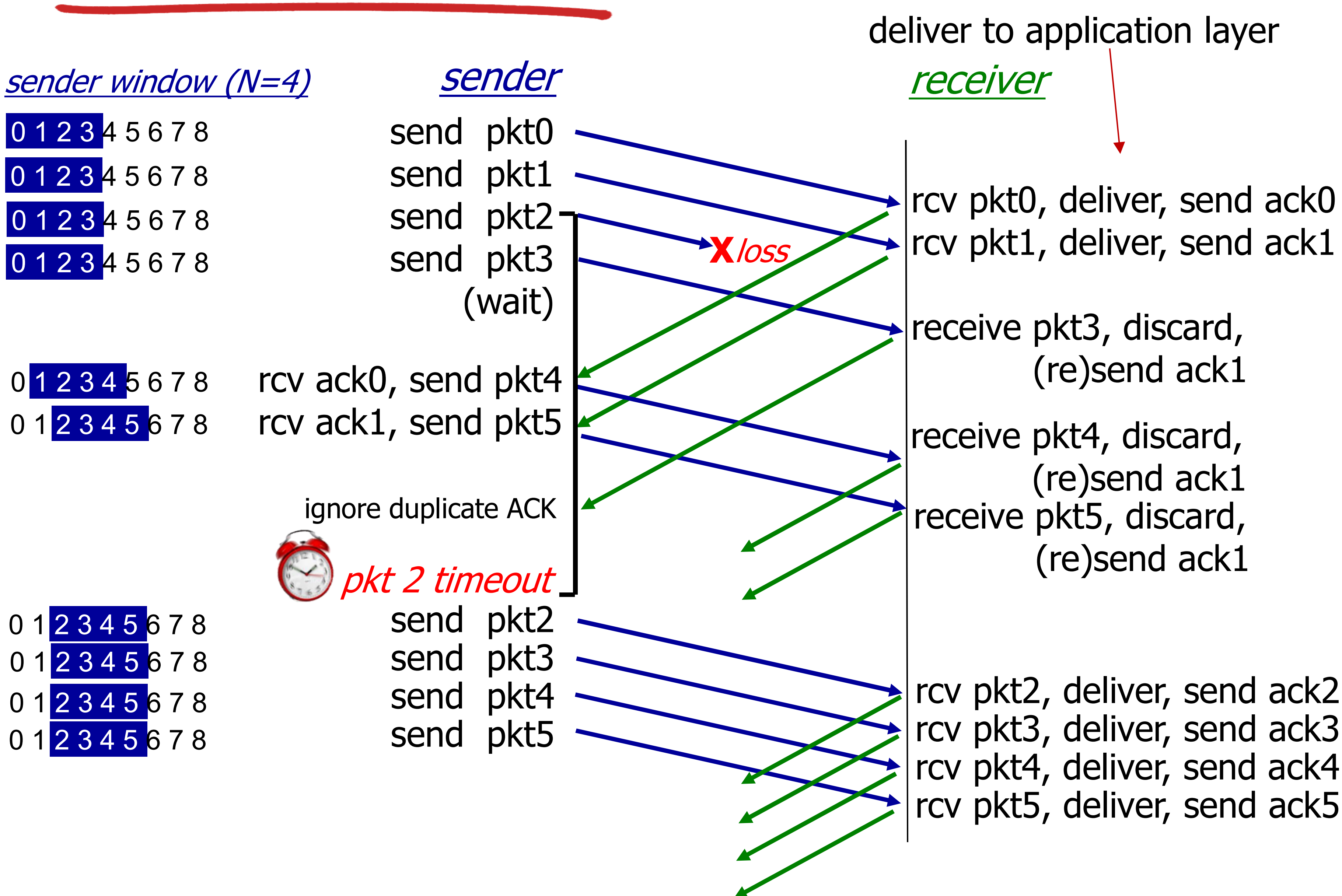
Go-Back-N: sender

- 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

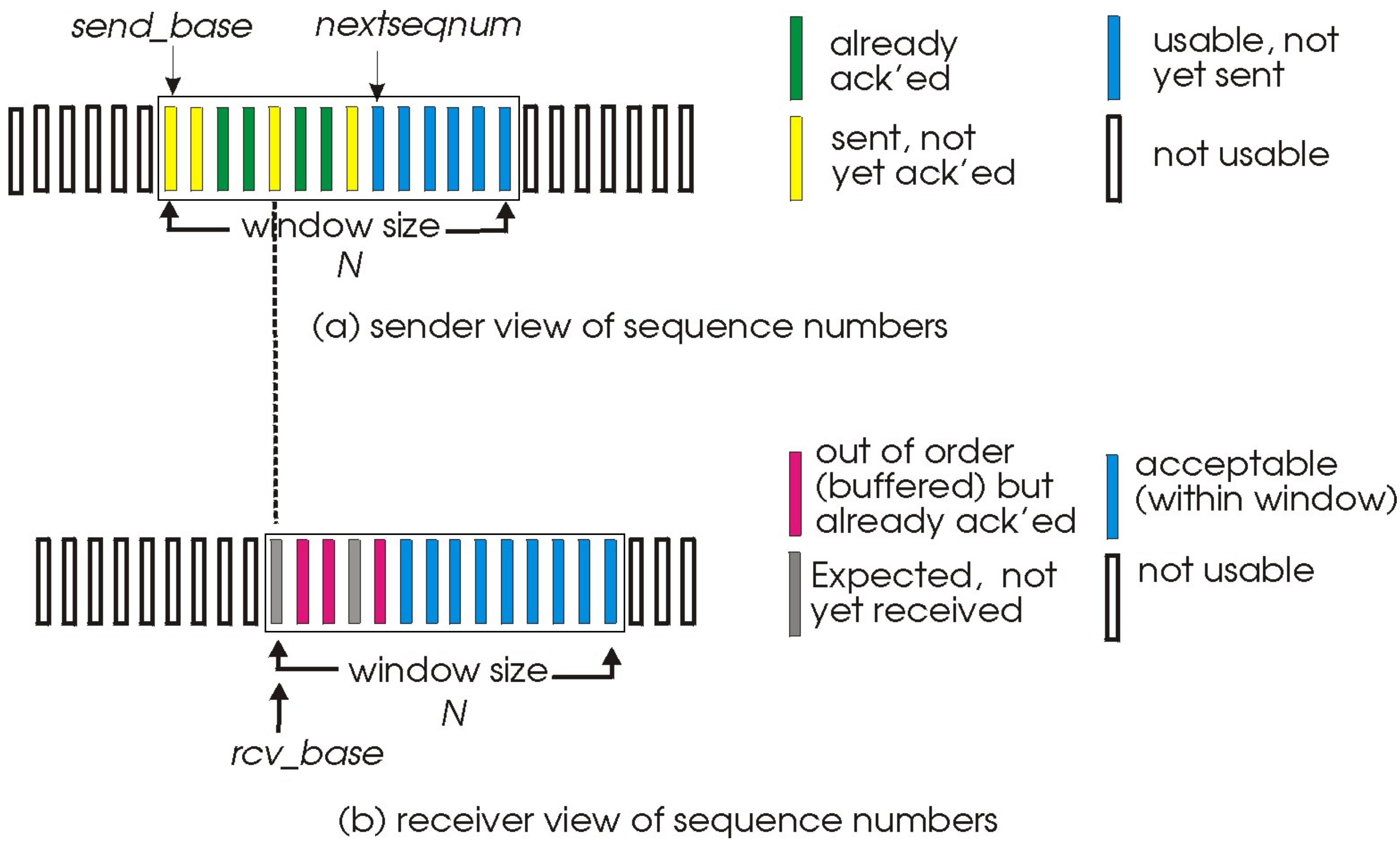
GBN in action



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



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 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

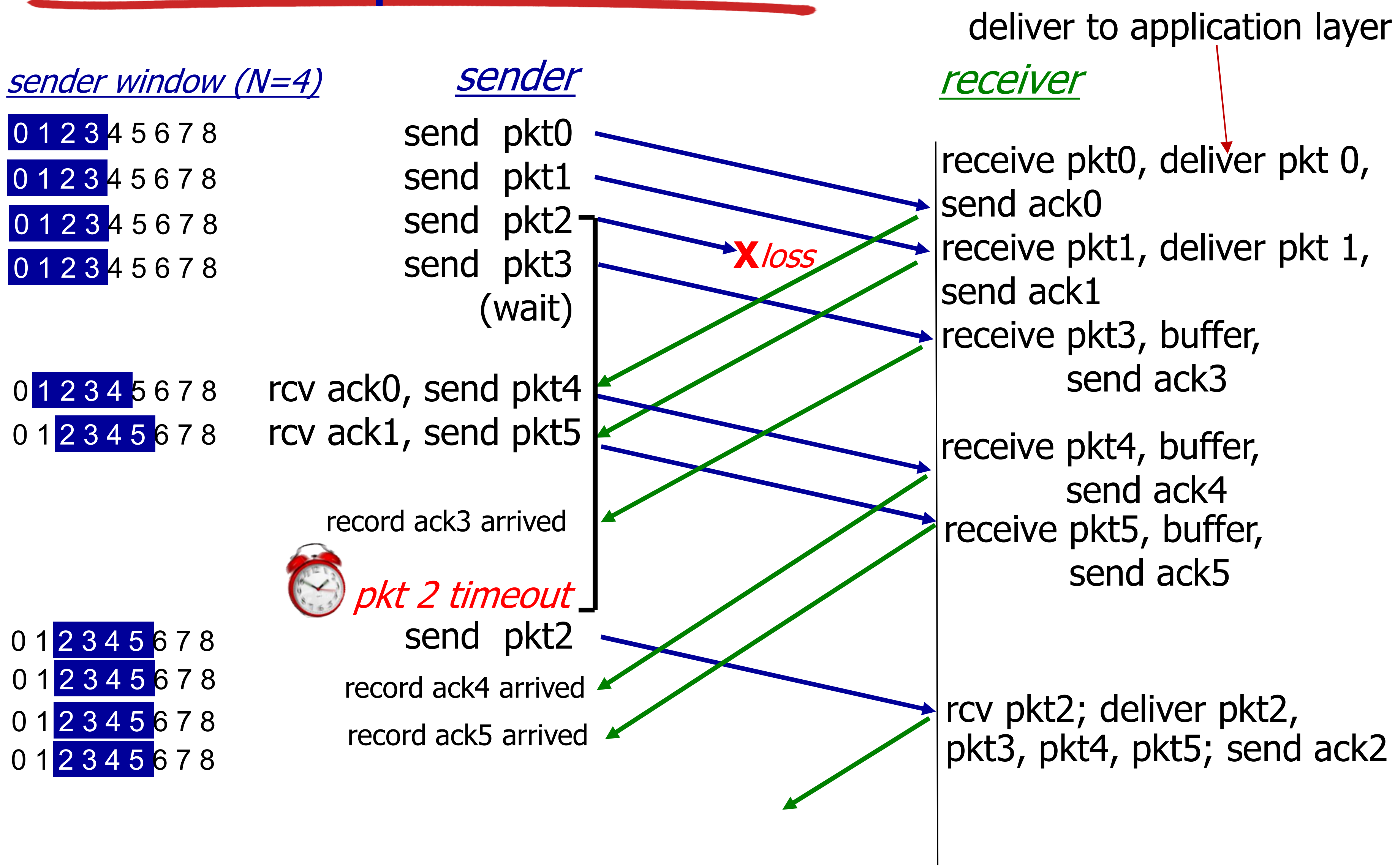
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

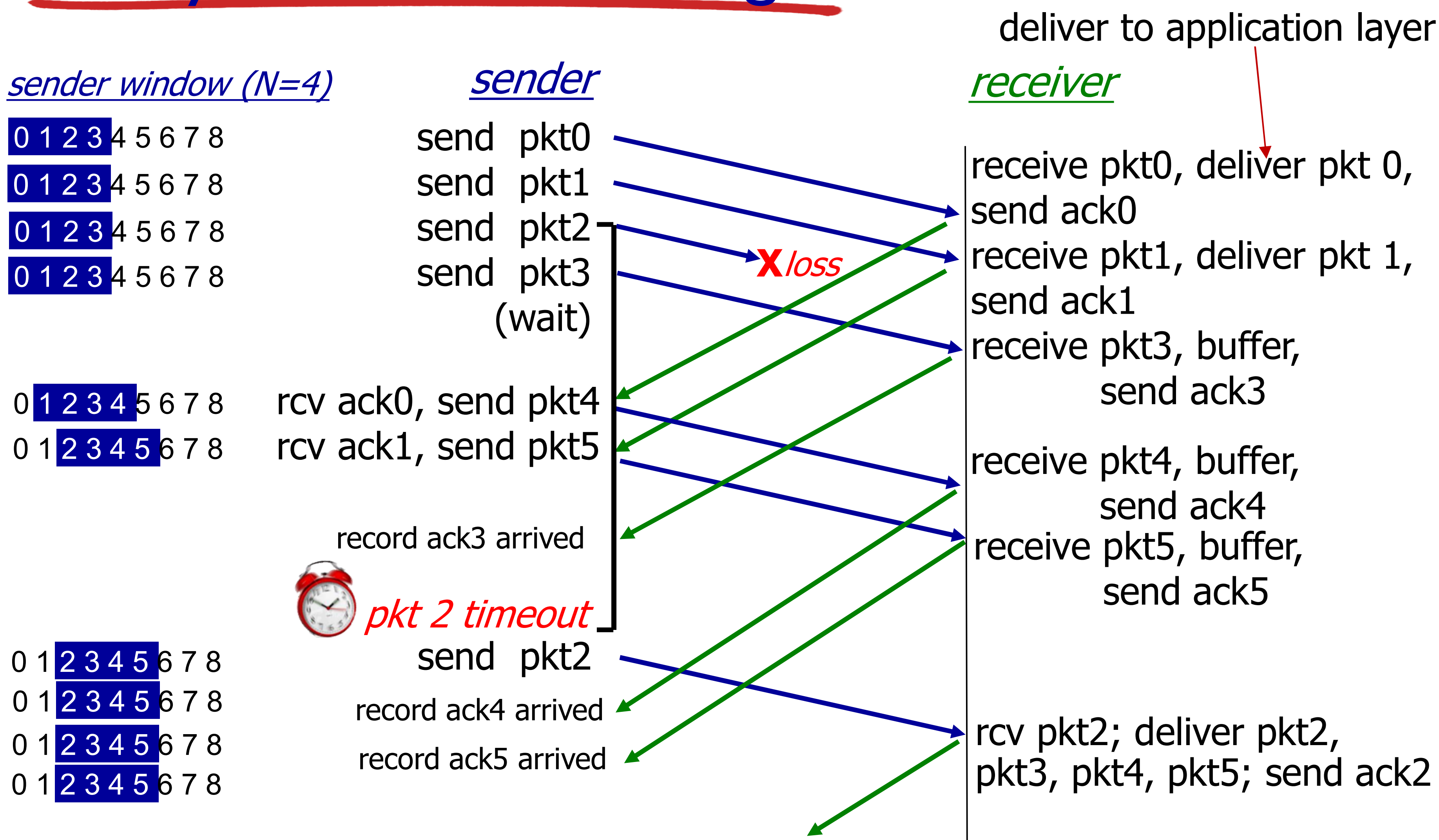
otherwise:

- ignore

Selective repeat in action

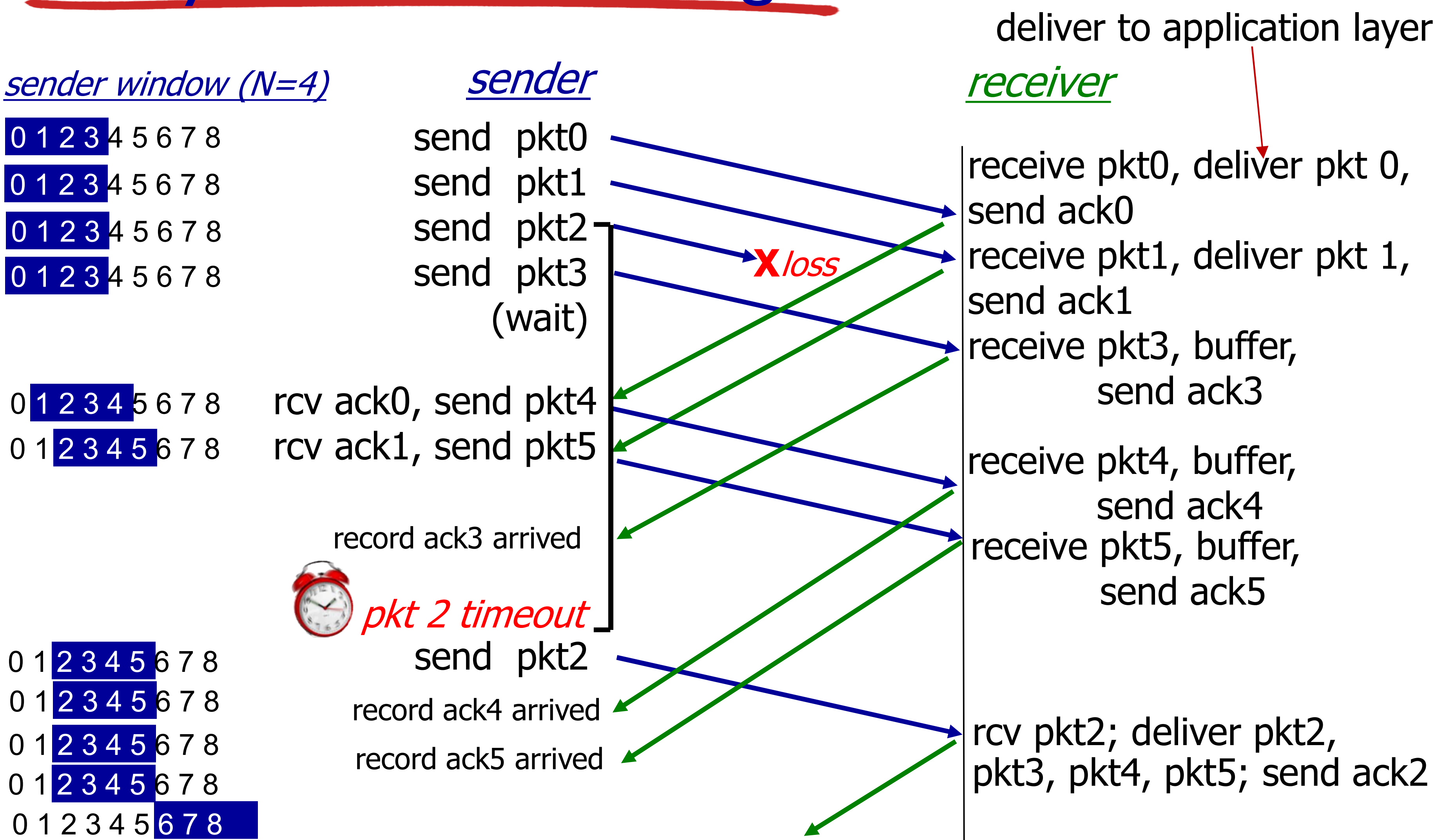


Test your understanding



What happens to the window when ACK 2 arrives?
Pause the video and write down your answer

Test your understanding



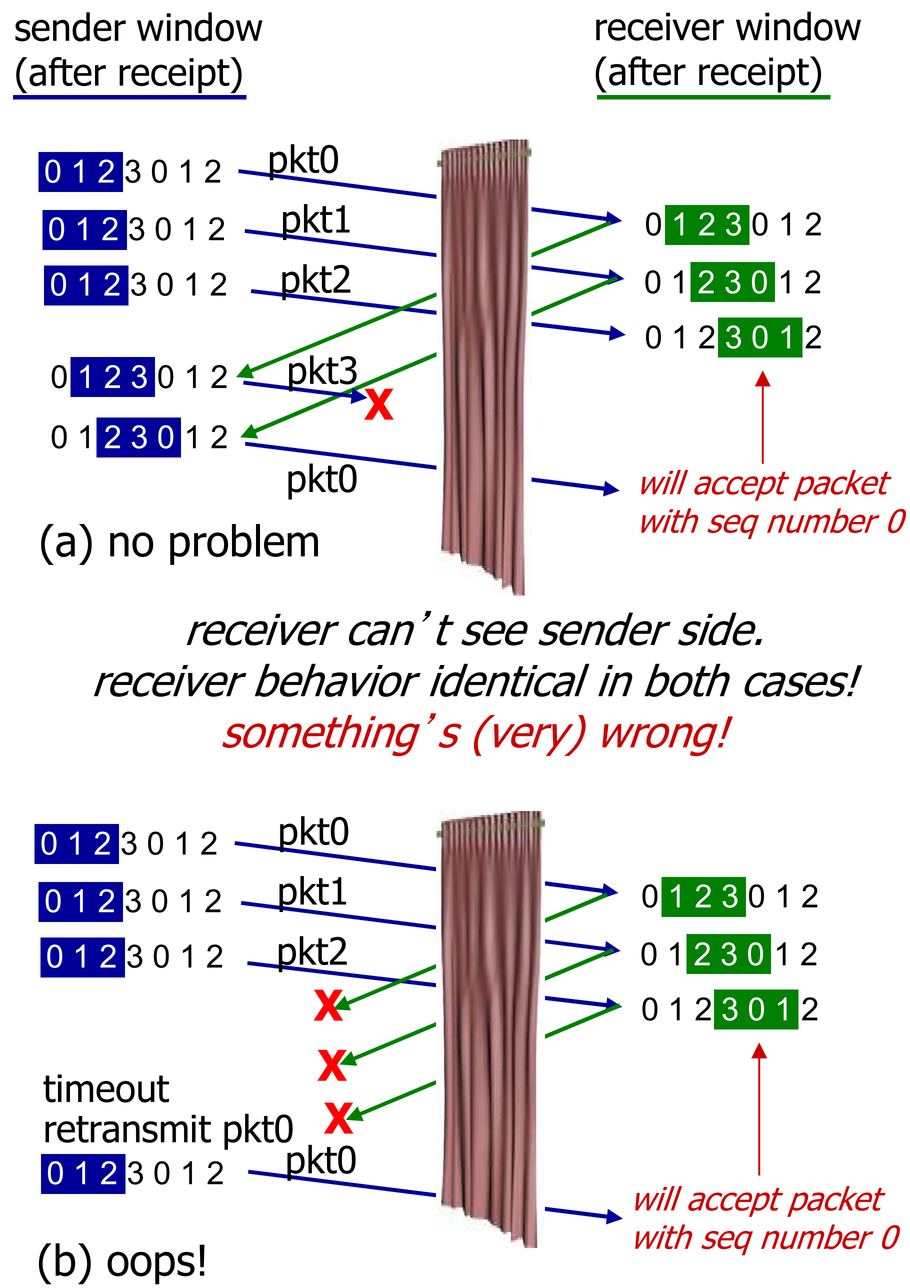
The window advances to 6 7 8 since ACKs 1-5 arrived
If you got it wrong review selective repeat

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)?



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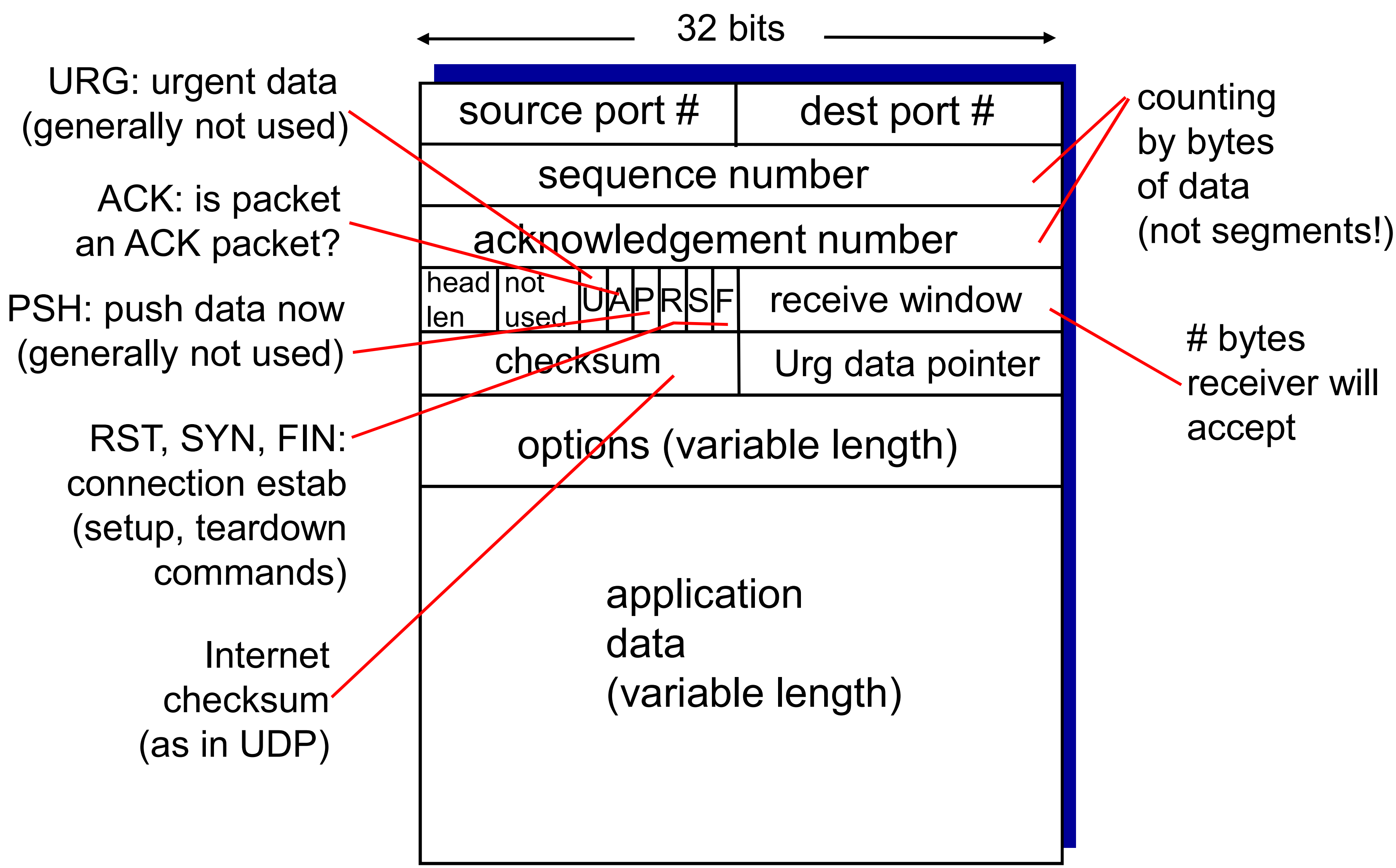
TCP: Overview RFCs: 793,1122,1323, 2018, 2581

- 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

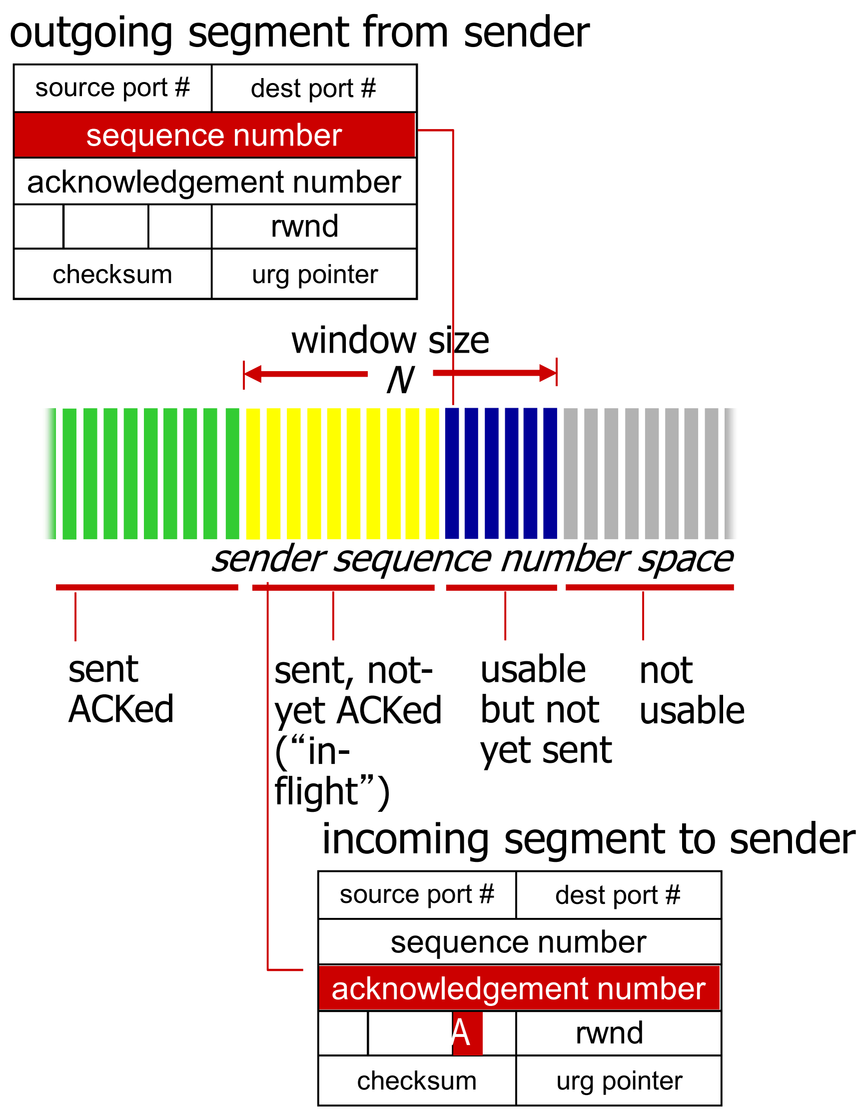
acknowledgements:

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

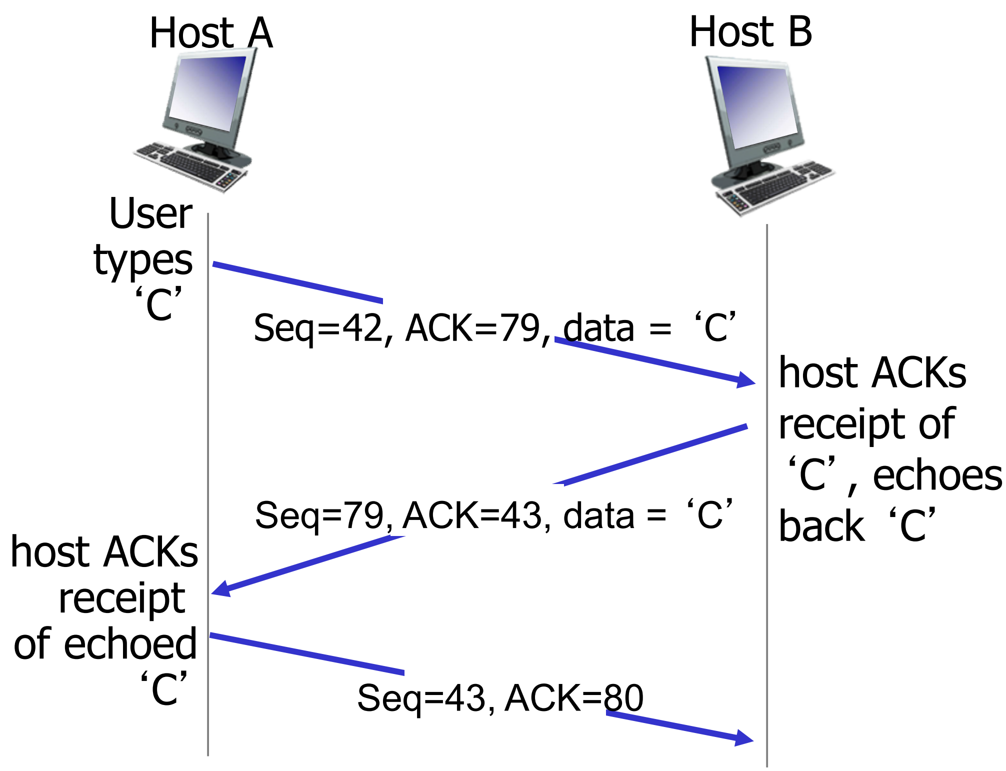
  (cumulative ACK)

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?

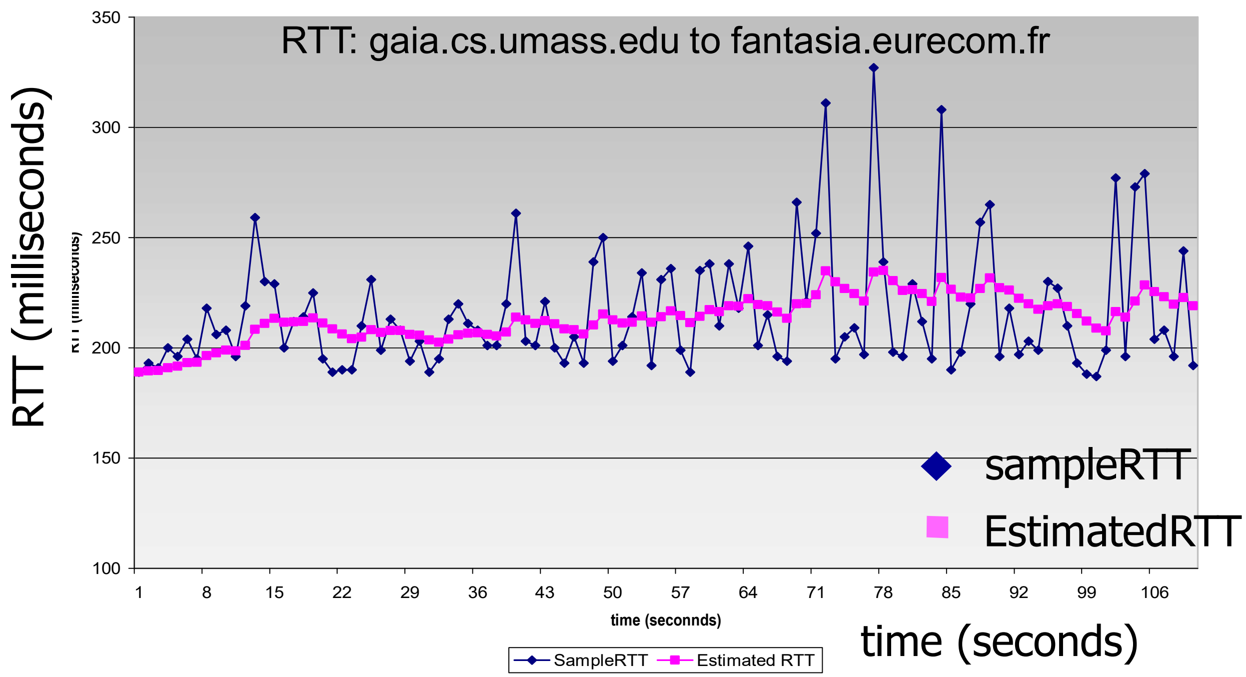
- **`SampleRTT`**: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- **`SampleRTT`** will vary, want estimated RTT "smoother"
  - average several *recent* measurements, not just current **`SampleRTT`**

TCP round trip time, timeout

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

- exponential weighted moving average
- influence of past sample decreases exponentially fast
- typical value: α = 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
```

$\uparrow$ estimated RTT $\uparrow$ "safety margin"

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

Test your understanding

- What is the problem if a TCP timeout is set too short?

- What is the problem if a TCP timeout is set too long?

- Estimated RTT is 12ms α = 0.125, Sample RTT is 8ms. Calculate a new Estimated RTT

Pause the video and write down your answer

Test your understanding

- What is the problem if a TCP timeout is set too short?
  - Packets are retransmitted needlessly (they were delayed not lost) wasting bandwidth
- What is the problem if a TCP timeout is set too long?
  - Packets take a long time to retransmit meaning your connection is slow.
- Estimated RTT is 12ms $\alpha = 0.125$, Sample RTT is 8ms. Calculate a new Estimated RTT
  - Estimated RTT= $(1- \alpha)\ 12 + \alpha\ 8$ ms = $(0.875).12 + 0.125\ .8$ ms = 10.5+1ms = 11.5 ms

If you did not get it right revise this section

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TCP sender events (simplified)

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`

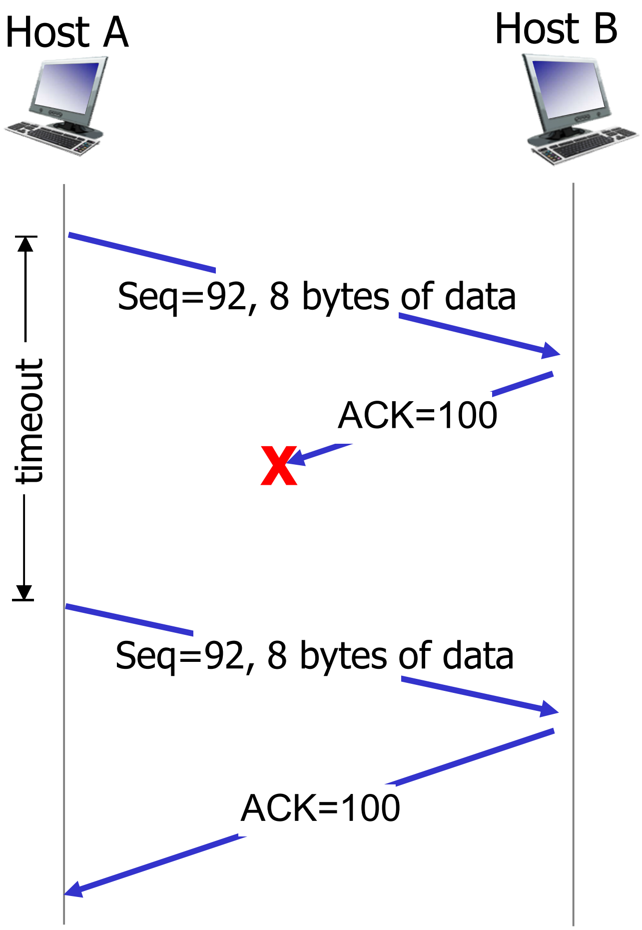
timeout:

- retransmit segment that caused timeout
- restart timer

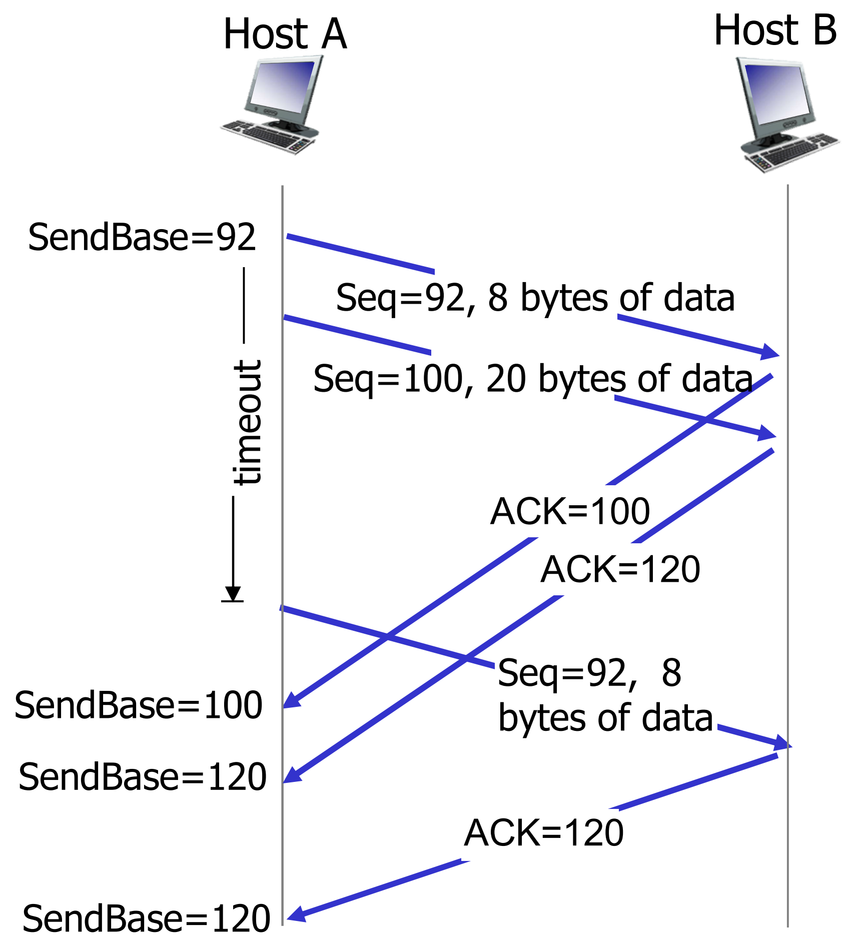
ack recieved:

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

TCP: retransmission scenarios

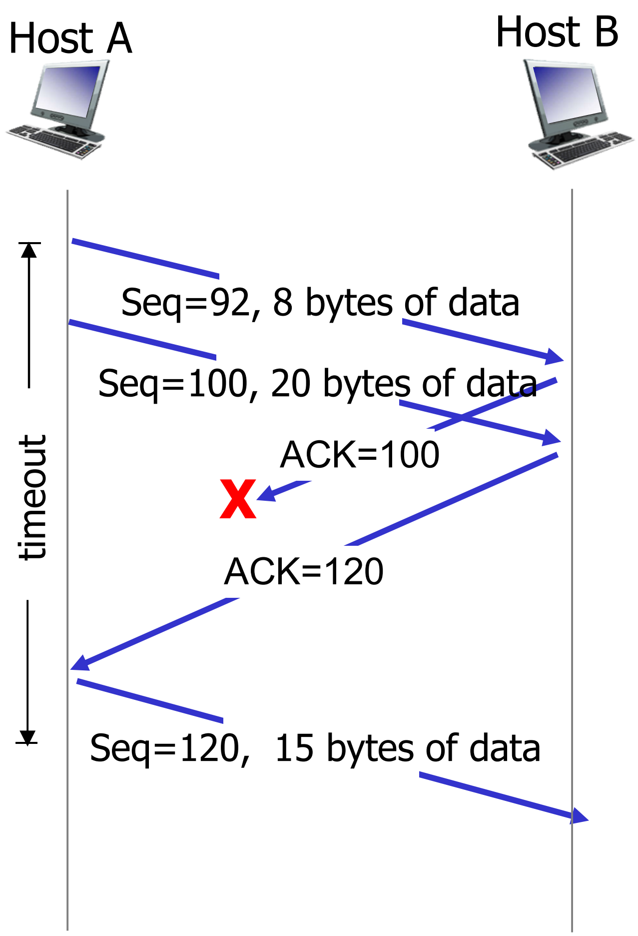


lost ACK scenario



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.

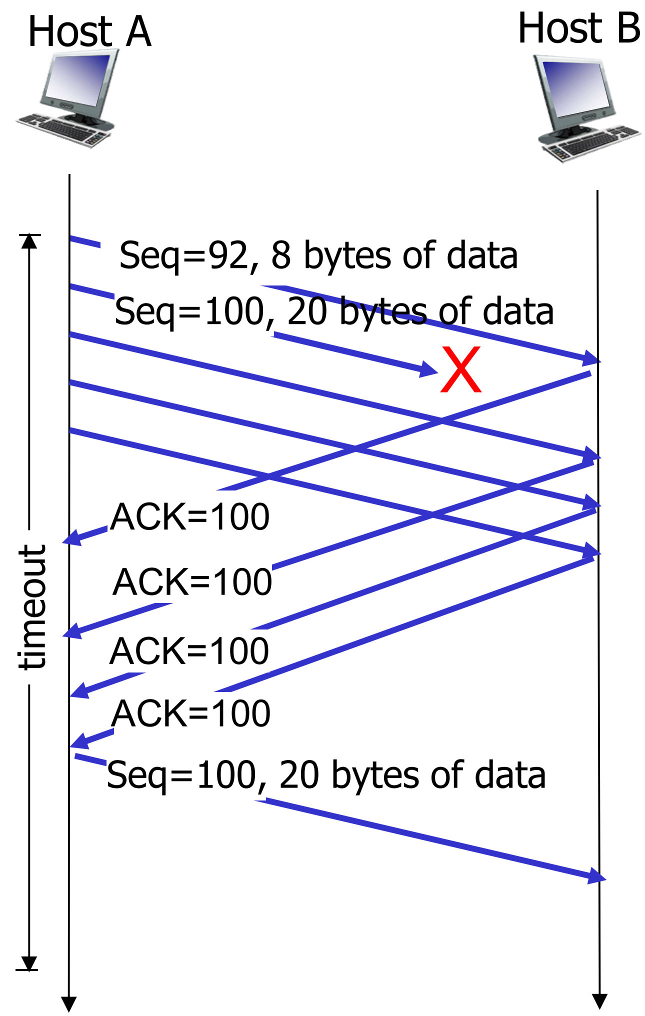
TCP fast retransmit

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

Test your understanding

- Simple TCP sender has a window size of 5. It has 10 packets to send. It sends packets 1,2,3,4,5.
- What should it do if it receives the following ACKS?
  - (What do you think happened?)
- Receives ACK for 1, 2, 3, 4, 5
- Receives ACK for 1, 3.
- Receives ACK for 1 four times.

Pause the video and write down your answer.

Test your understanding

- Simple TCP sender has a window size of 5. It has 10 packets to send. It sends packets 1,2,3,4,5.
- Receives ACK for 1, 2, 3, 4, 5
  - Send packets 6,7,8,9,10 (everything got there fine).
- Receives ACK for 1, 3.
  - Send packet 6, 7, 8. Wait for timeout for packet 4. (ACK 3 tells us packets 2 and 3 must have arrived – if packet 2 had not arrived ACK 1 would be generated.)
- Receives ACK for 1 four times.
  - Send packet 6. Send packet 2 (Triple duplicate). Packet 2 is lost, corrupted or delayed (hence receiver sends duplicate when it receives packets 3, 4, 5)

If you did not get the right answer review the last section.

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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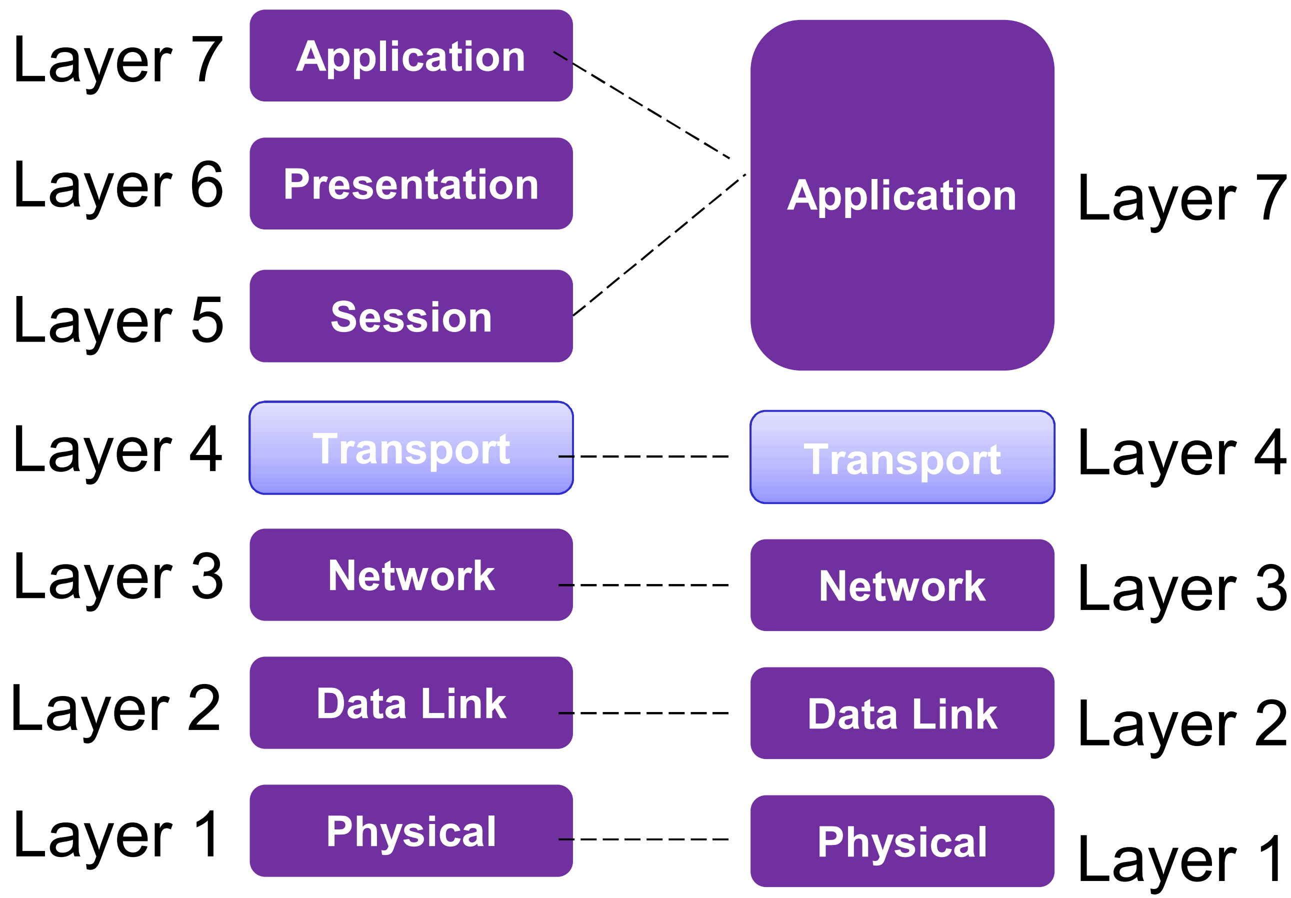
Transport Layer



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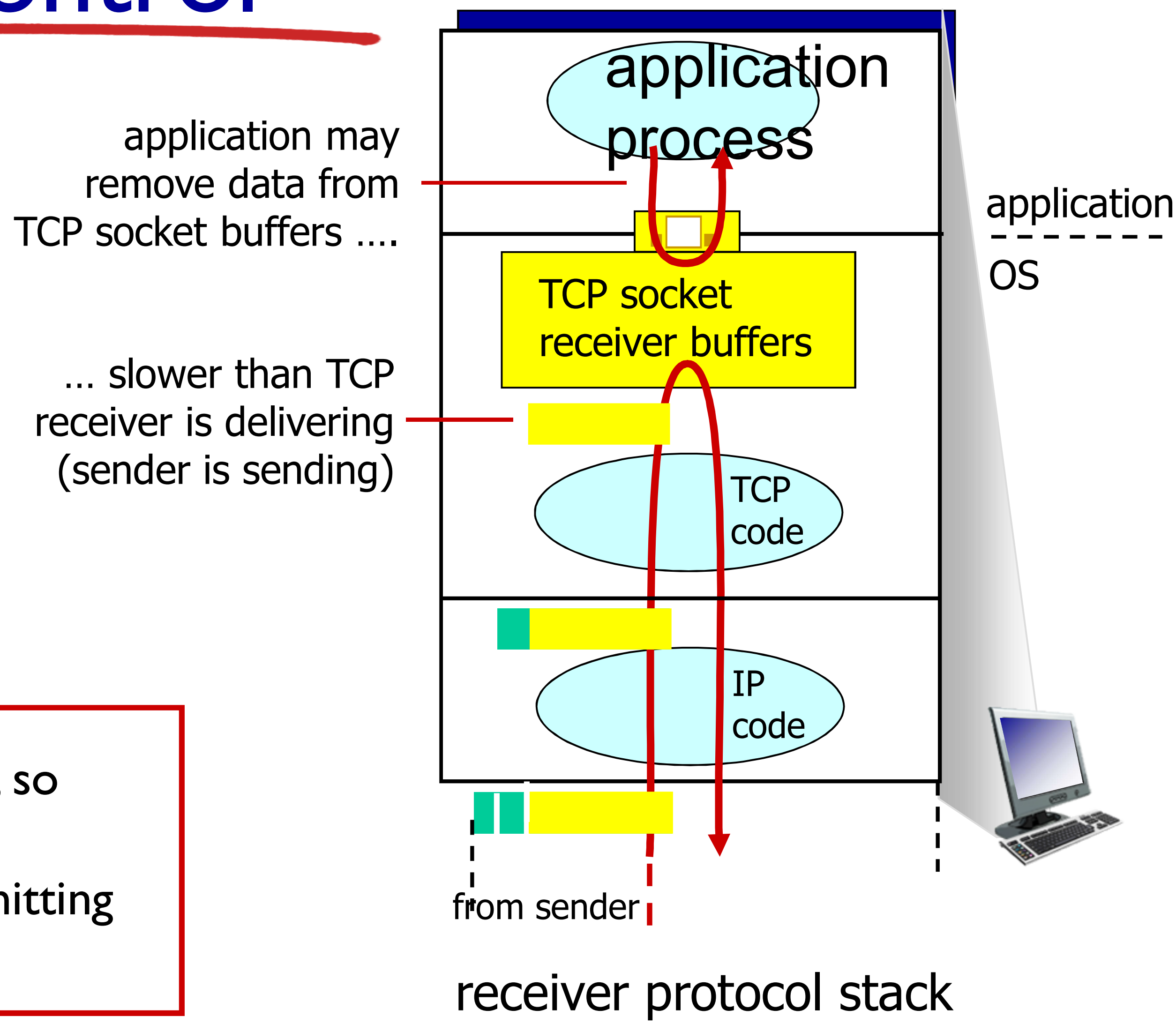
TCP flow control

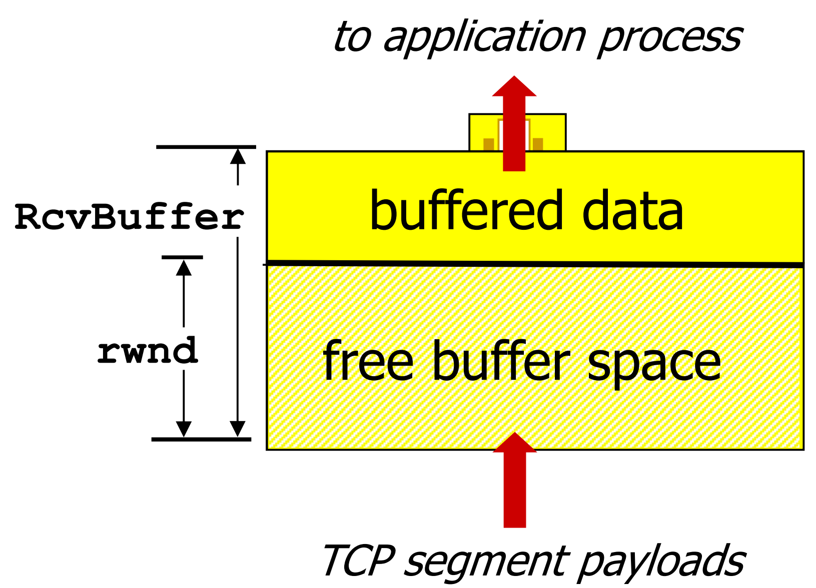


flow control

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



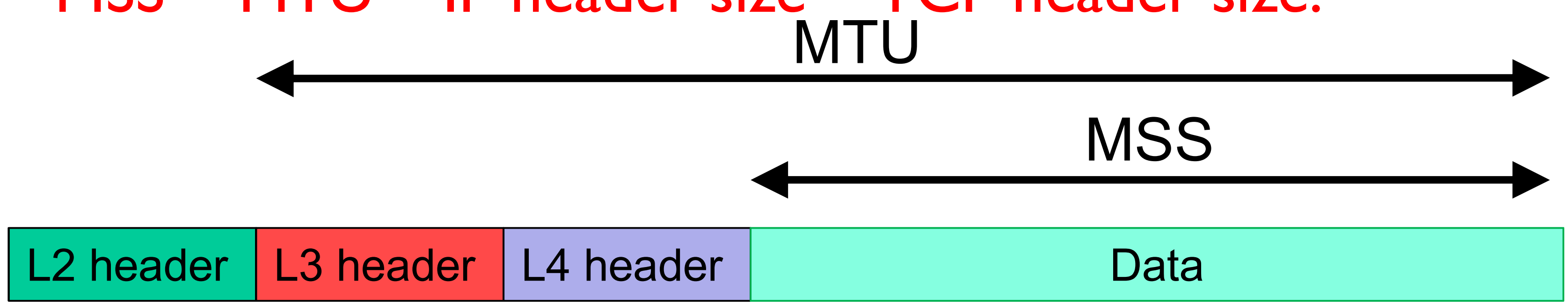
receiver-side buffering

`rwnd` = receive window

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.



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.

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.
   - Cure – receiver does not advertise windows that would cause sender to send small amounts of data.

Check your understanding

- What is the reason for flow control?
- Who sends rwnd?
- What problem might be caused by a very very small rwnd? (How might this happen?)
- Pause the video write down your answers.

Check your understanding

- What is the reason for flow control?

  Stop the sender sending too much for the receiver

- Who sends rwnd?

  The receiver sends it – it is the “receiver window”. It says how much buffer space the receiver has.

- What problem might be caused by a very very small rwnd? (How might this happen?)

  Sender might send small (inefficient) packets. This might happen if receiver is very slow – processing one or two bytes at a time.

If you are puzzled review the previous section.

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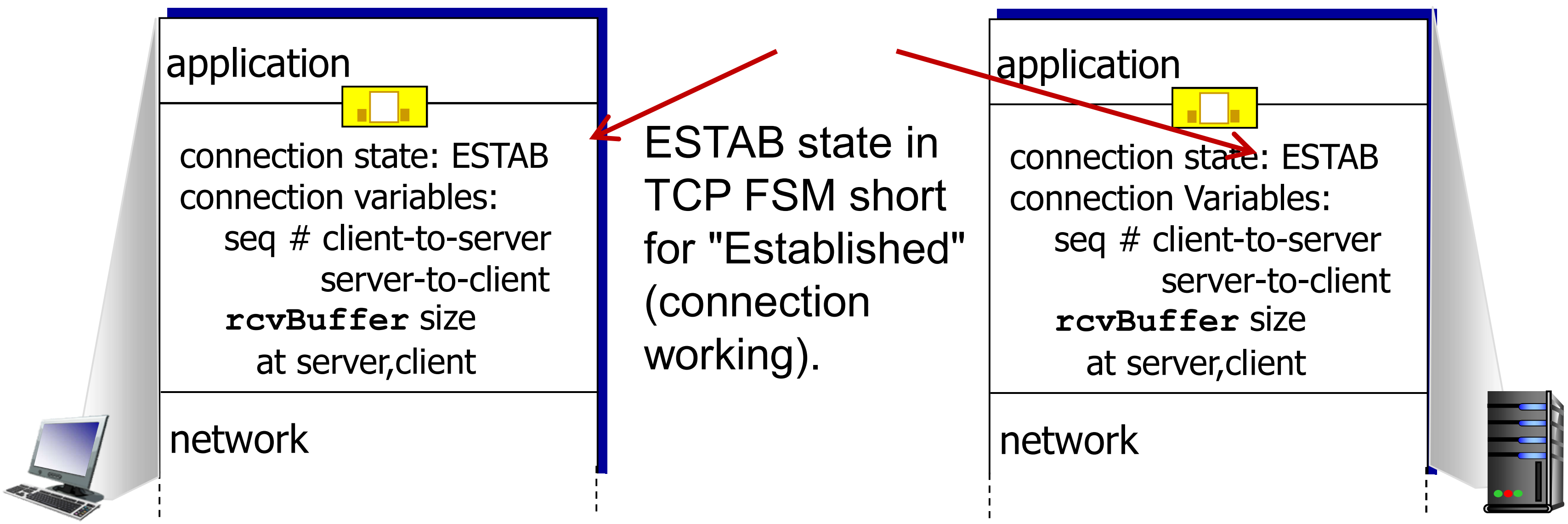
Connection Management

before exchanging data, sender/receiver "handshake":

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

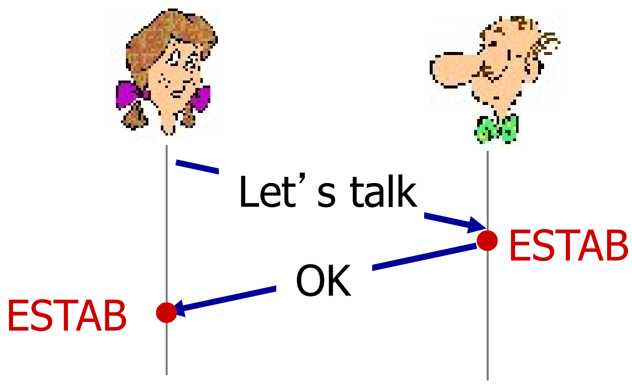


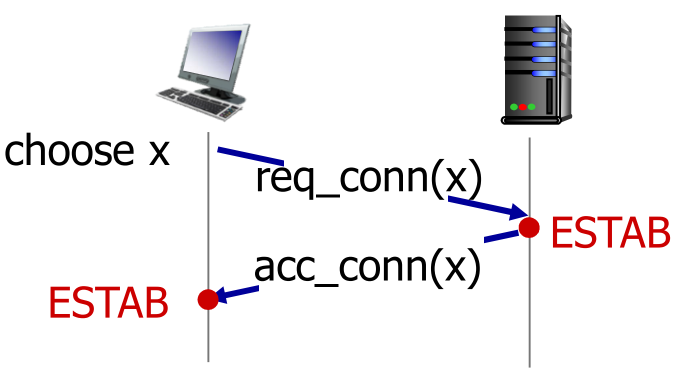

```
Socket clientSocket =
  newSocket("hostname","port
   number");
```

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

Agreeing to establish a connection

2-way handshake:



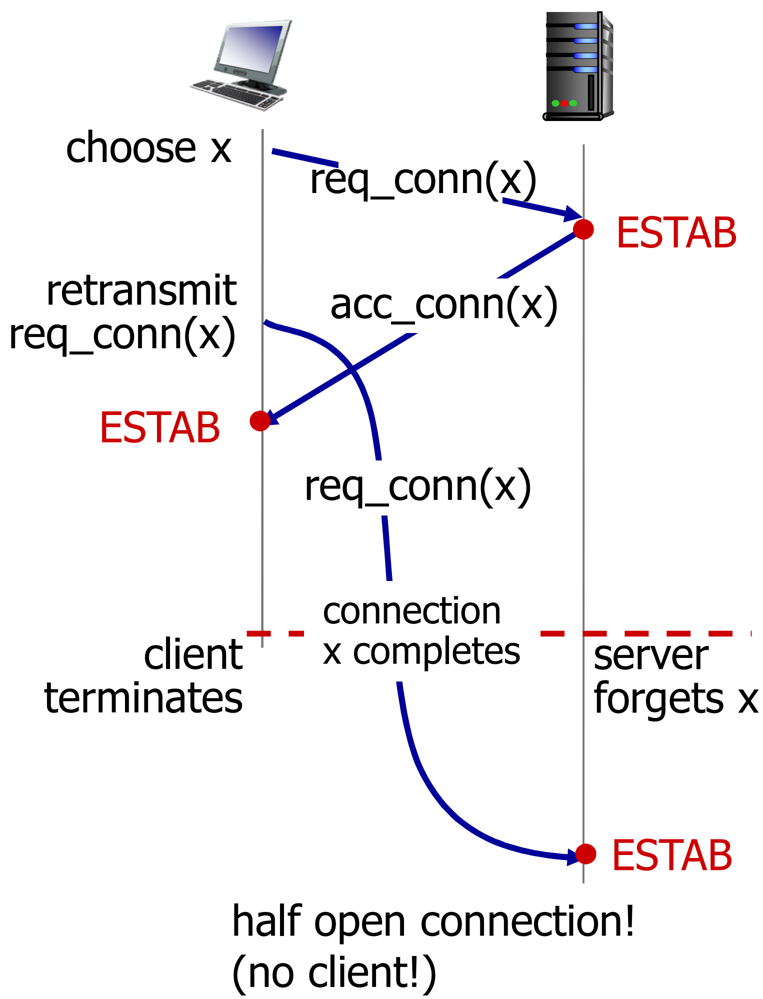


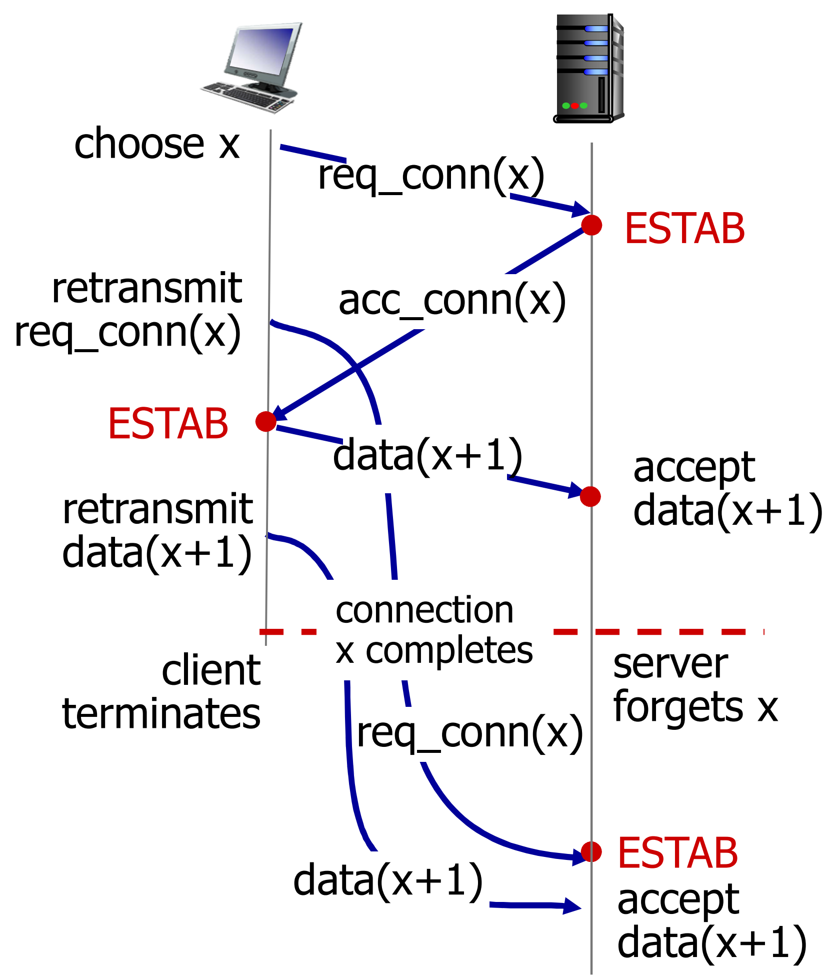
Q: will 2-way handshake always work in network?

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

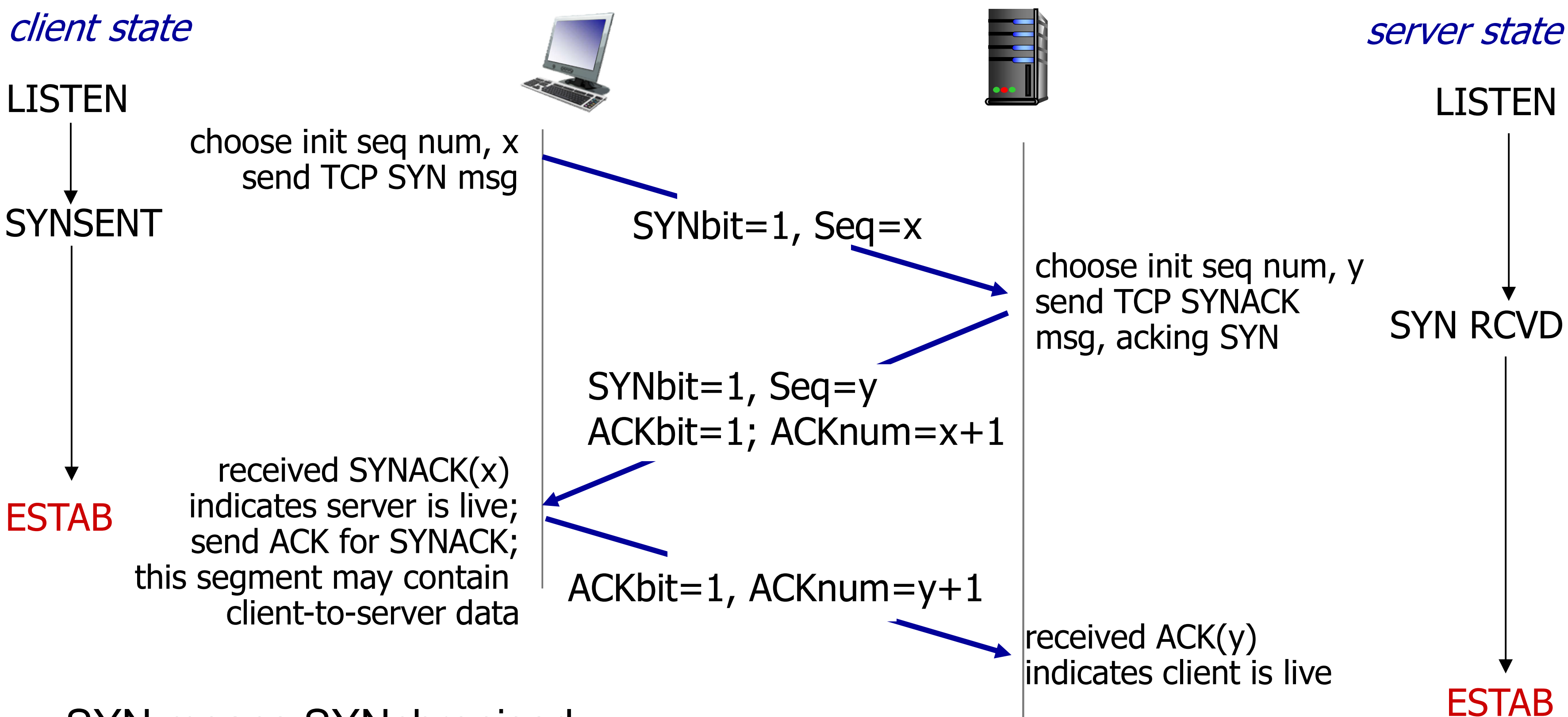
Agreeing to establish a connection

2-way handshake failure scenarios:



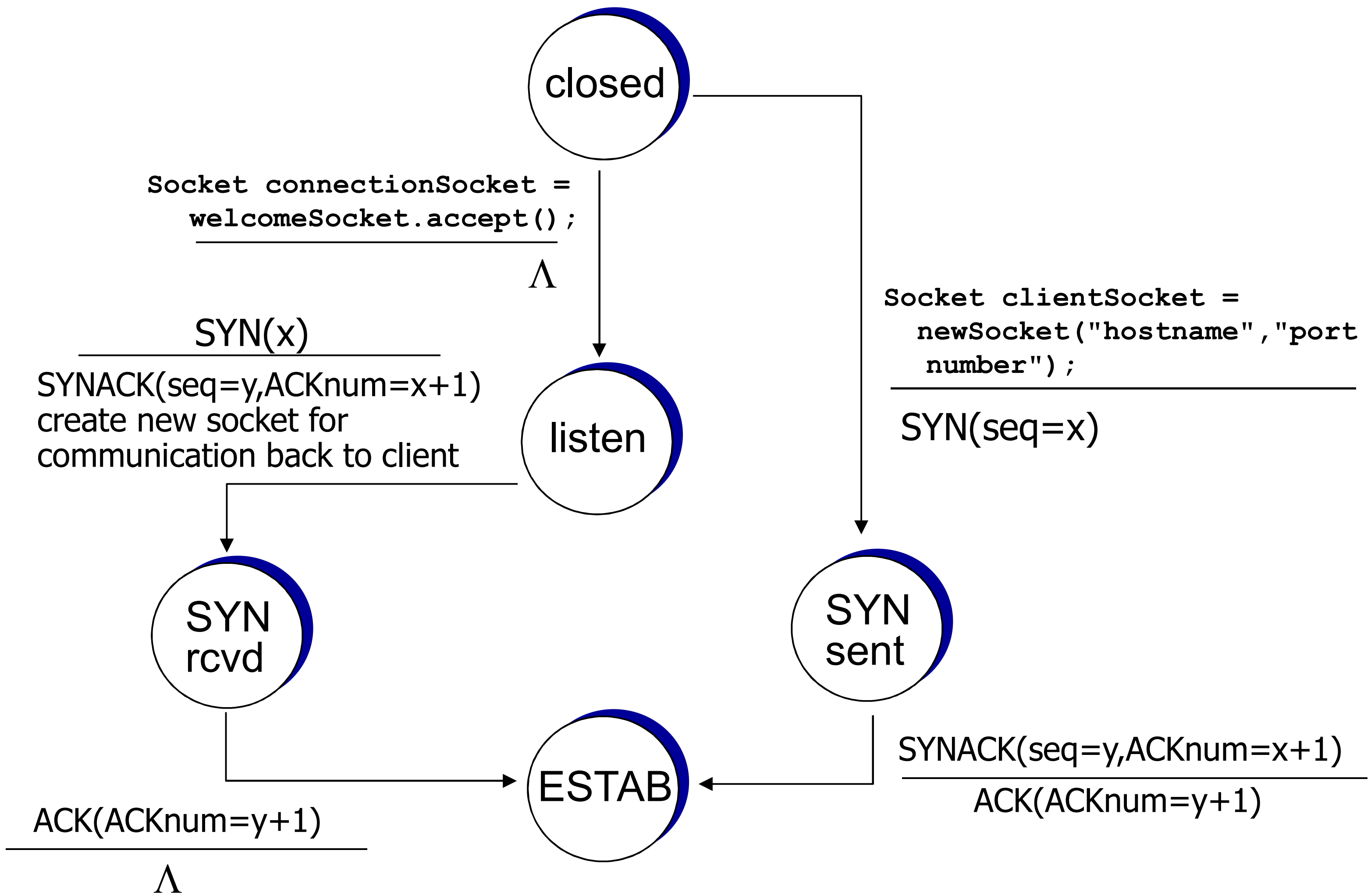


TCP 3-way handshake



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

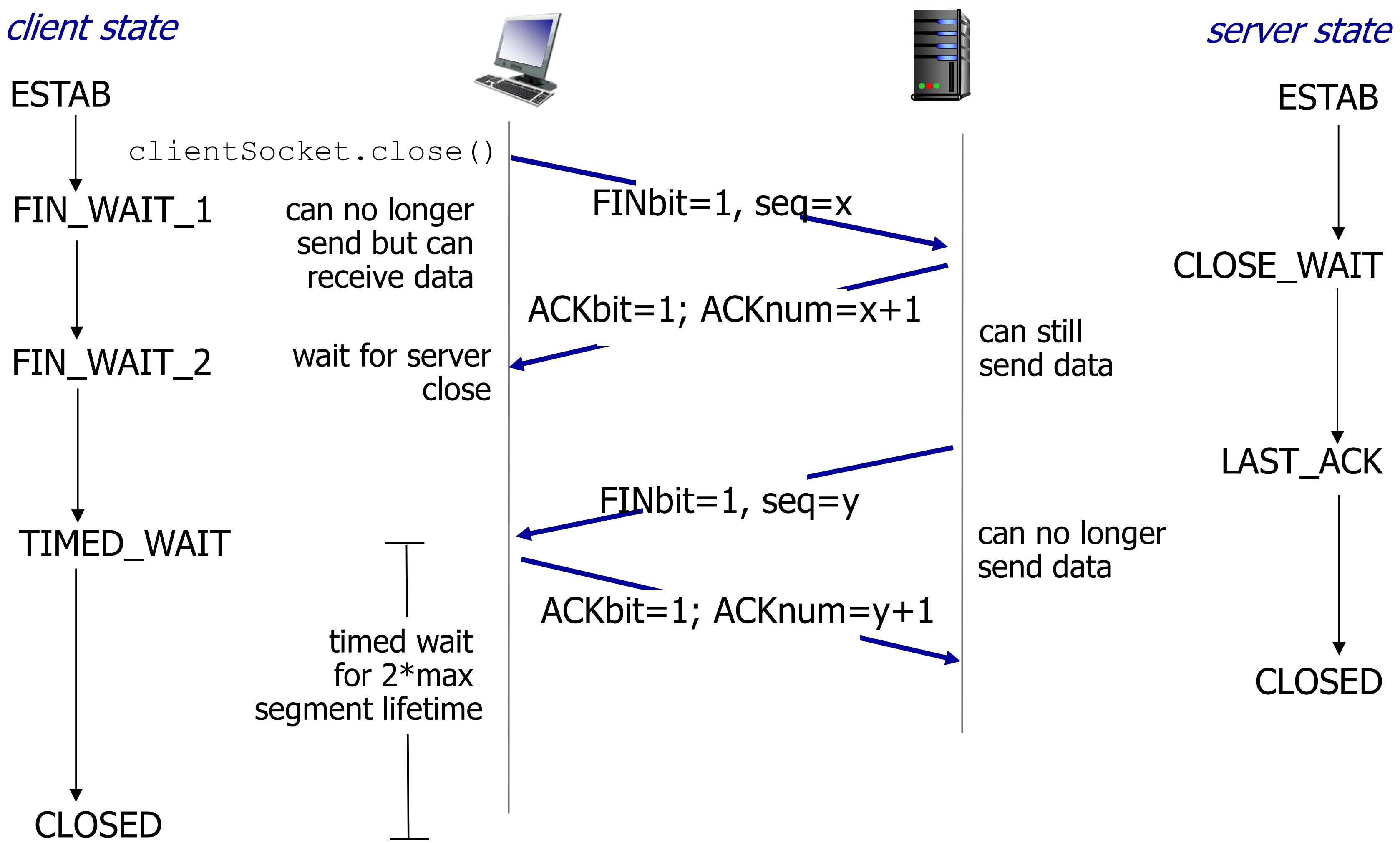
TCP 3-way handshake: Finite State Machine



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



Test your understanding

- What is the meaning of the following TCP flag combinations:

  1) SYN – ACK
  2) FIN
  3) FIN – ACK

- What are the flag combinations for the three packets in the three way handshake?
- Pause the video write down your answer

Test your understanding

1) SYN – ACK -- Response to SYN, will accept a new connection

2) FIN – I want to close the connection

3) FIN – ACK – I agree to close the connection

- What are the flag combinations for the three packets in the three way handshake?
- SYN SYN—ACK ACK
- If you had problems review the previous section

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

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

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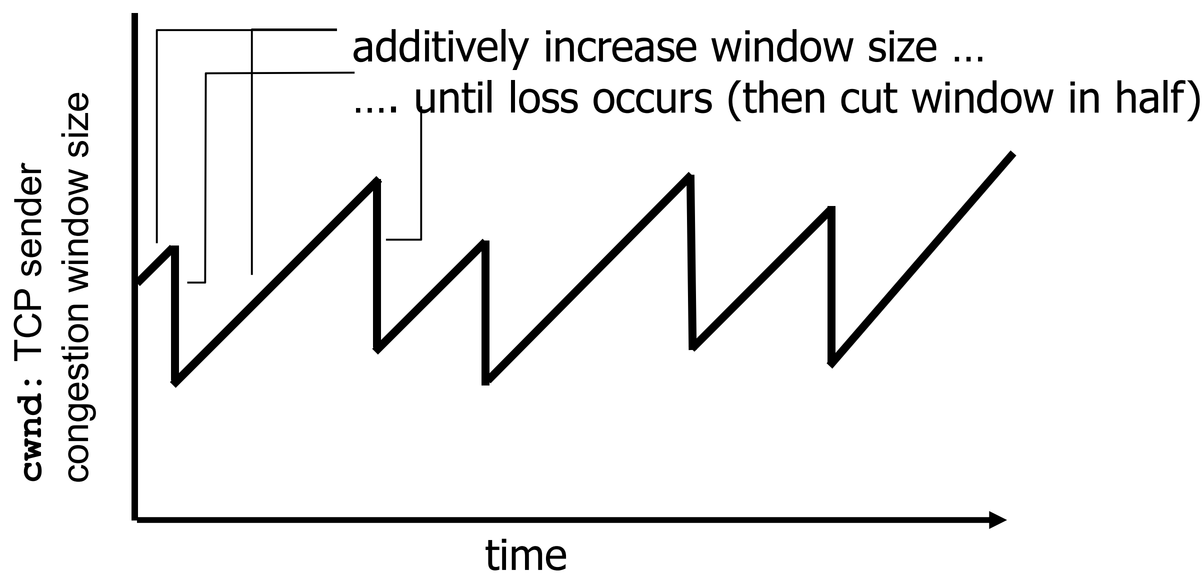
3.7 TCP congestion control

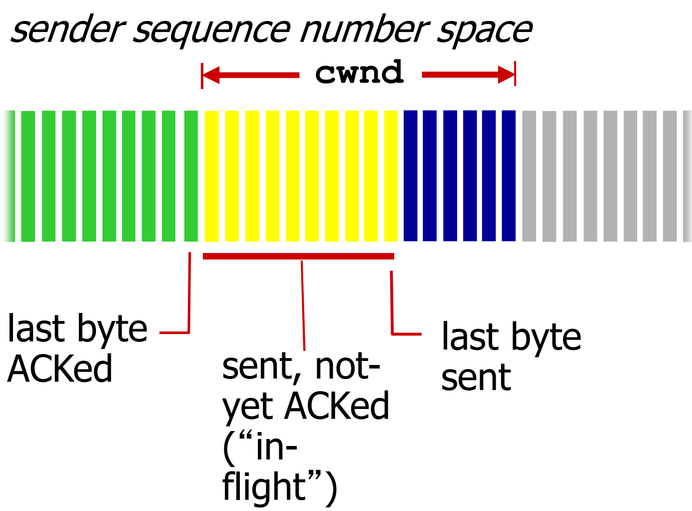
TCP congestion control: additive increase multiplicative decrease

- *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:

```
LastByteSent-
  LastByteAcked  ≤  cwnd
```

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

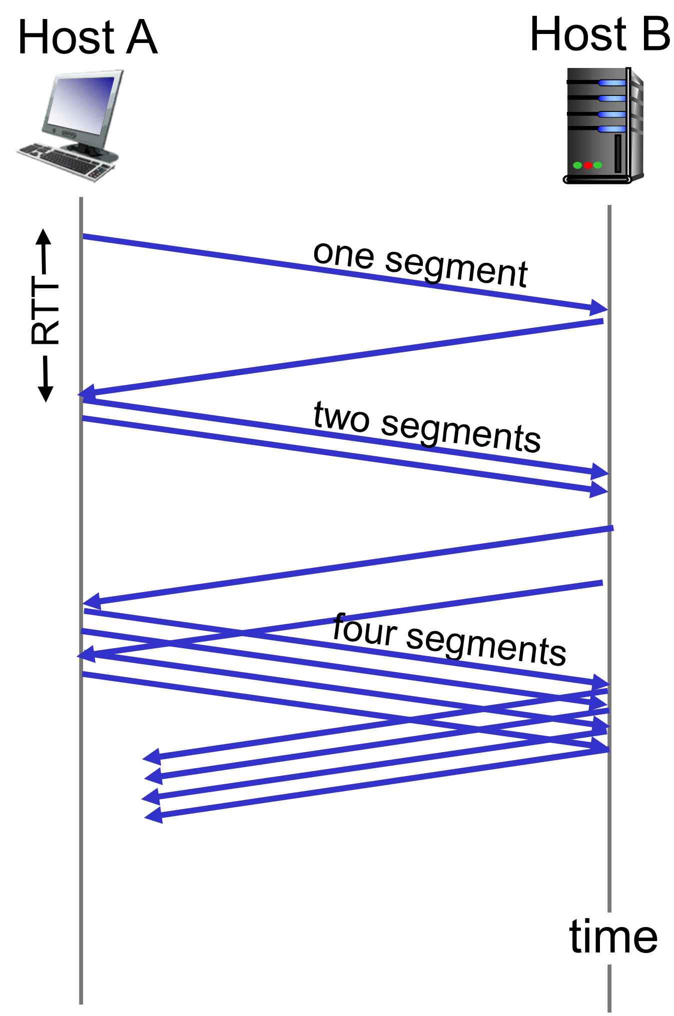
TCP sending rate:

- *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

Test your understanding

- What are the purposes of windows rwnd and cwnd?
- In additive increase multiplicative decrease what triggers TCP to reduce cwnd?
- How does a receiver indicate is buffer is getting full and the sender must slow down?
- Pause the video write down your answer.

Test your understanding

- What are the purposes of windows rwnd and cwnd?

  rwnd = flow control

  (not too fast for receiver)

  cwnd = congestion control

  (not too fast for network)

- In additive increase multiplicative decrease what triggers TCP to reduce cwnd?

  loss (or timeout) of a packet

- How does a receiver indicate is buffer is getting full and the sender must slow down?

  Sends smaller rwnd

- If you had problem review previous section.

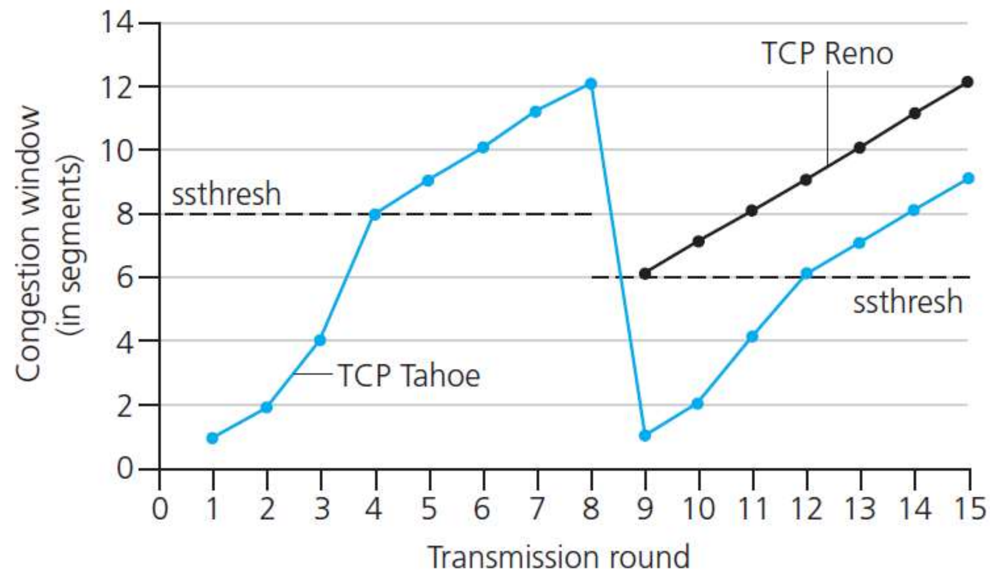
TCP: detecting, reacting to loss

- 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:

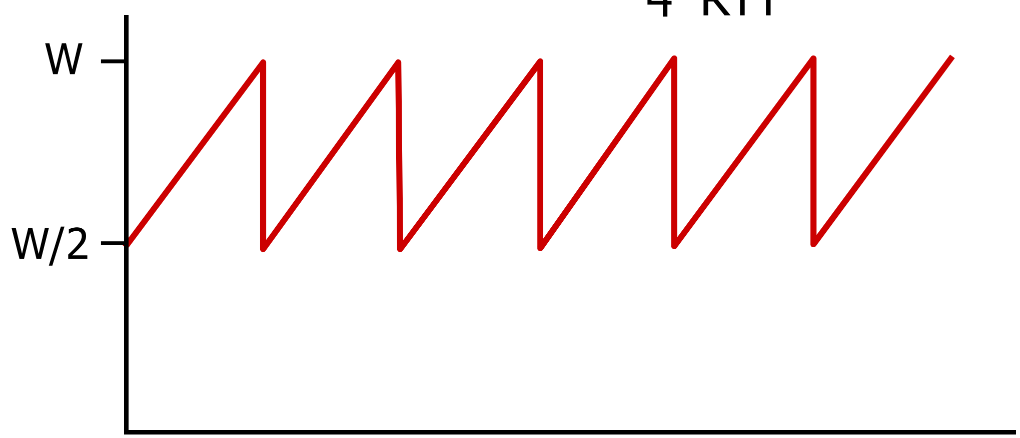
- variable **ssthresh**
- on loss event, **ssthresh** is set to 1/2 of **cwnd** just before loss event

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

TCP throughput

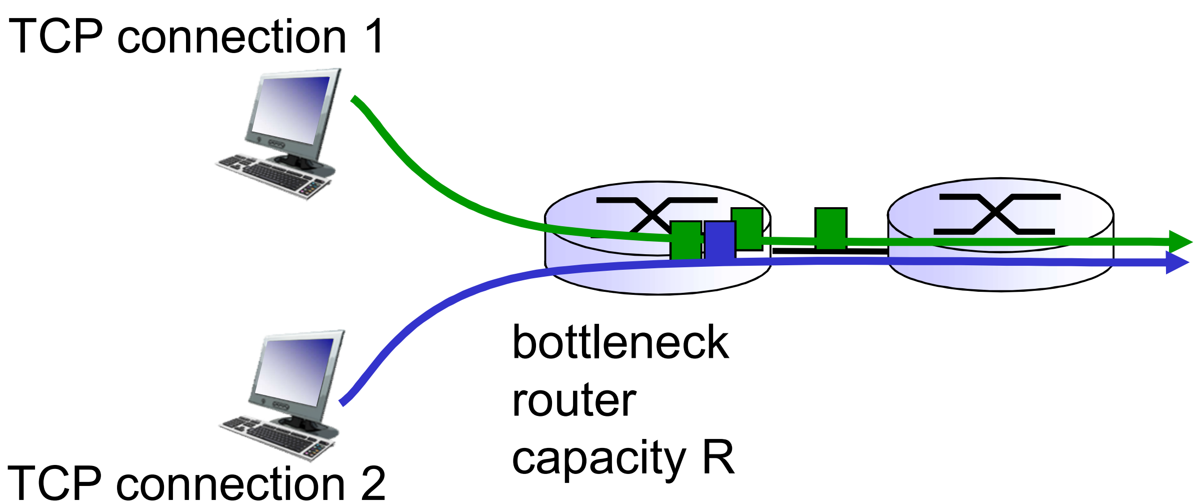
- avg. TCP throughput as function of window size, RTT?
  - ignore slow start, assume always data to send
- W: window size (measured in bytes) where loss occurs
  - avg. window size (# in-flight bytes) is ¾ W
  - avg. throughput is 3/4W per RTT

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



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 just over R/2

Test your understanding

- Here are some “clever” TCP ideas – what is wrong with them?
- The receiver will only send a very small rwnd that way it can never get too much traffic.
- The sender will only reduce window size very slowly in case of congestion (AIAD).
- We can be very quick if we send another packet if we have only one duplicate ACK (not triple)
- Pause the video write down your answers.

Test your understanding

- The receiver will only send a very small rwnd that way it can never get too much traffic.

  This will be inefficient. The sender can only have a small window (like rdt 3.0)

- The sender will only reduce window size very slowly in case of congestion.

  We may lead to lots of lost packets.

  We may no longer respect fairness.

- We can be very quick if we send another packet if we have only one duplicate ACK (not triple)

  We will sends lots of unnecessary packets

- If you are confused review the previous section.

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:
  - data plane
  - control plane