Advanced Networks

Transport layer: TCP

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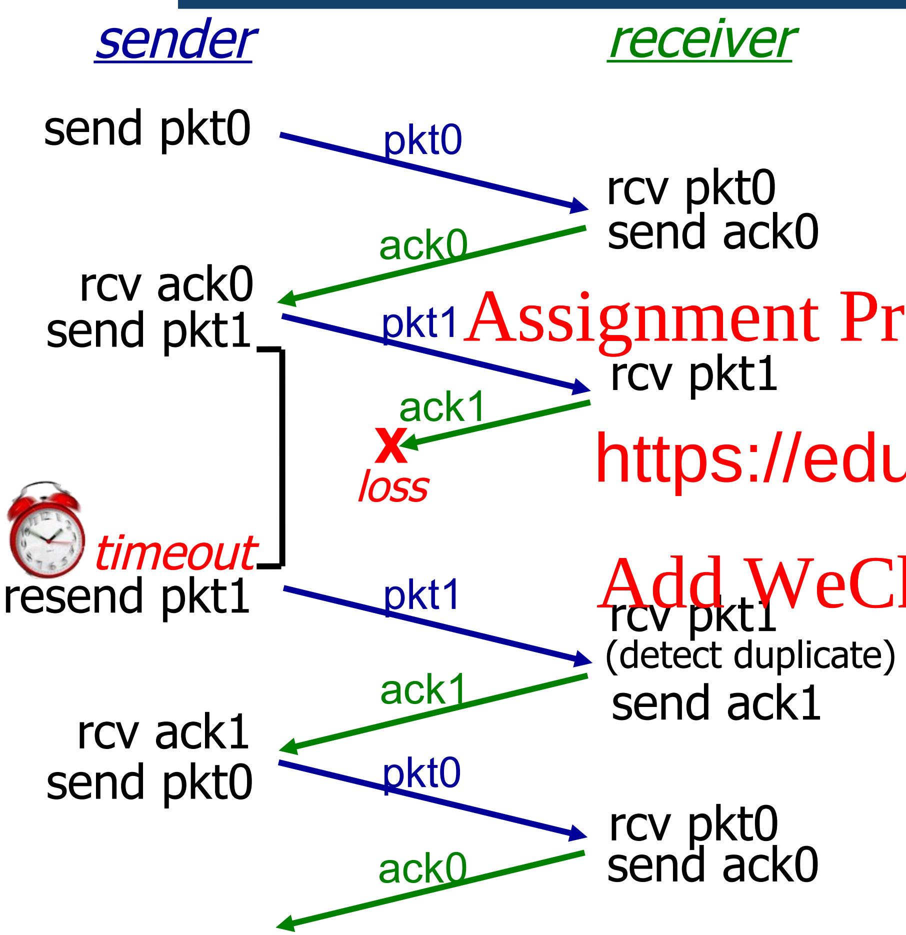
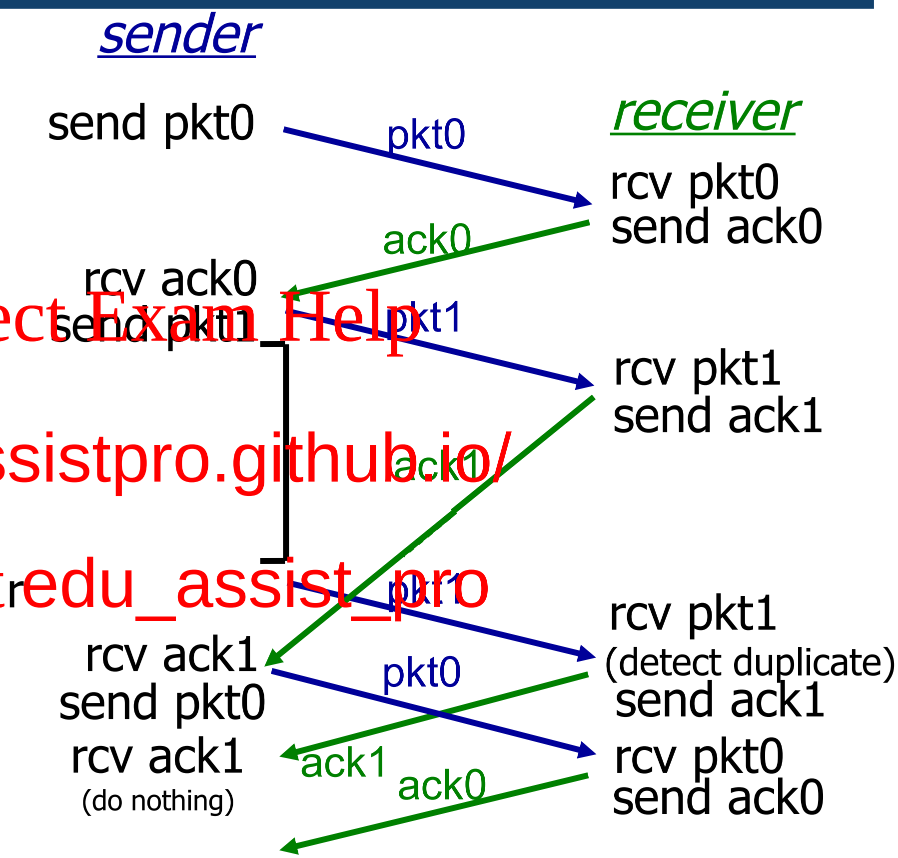
rdt3.0 in action



(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:

$$D_{trans} = \frac{L}{} = \frac{8000\ bits}{} = 8\ microsecs$$

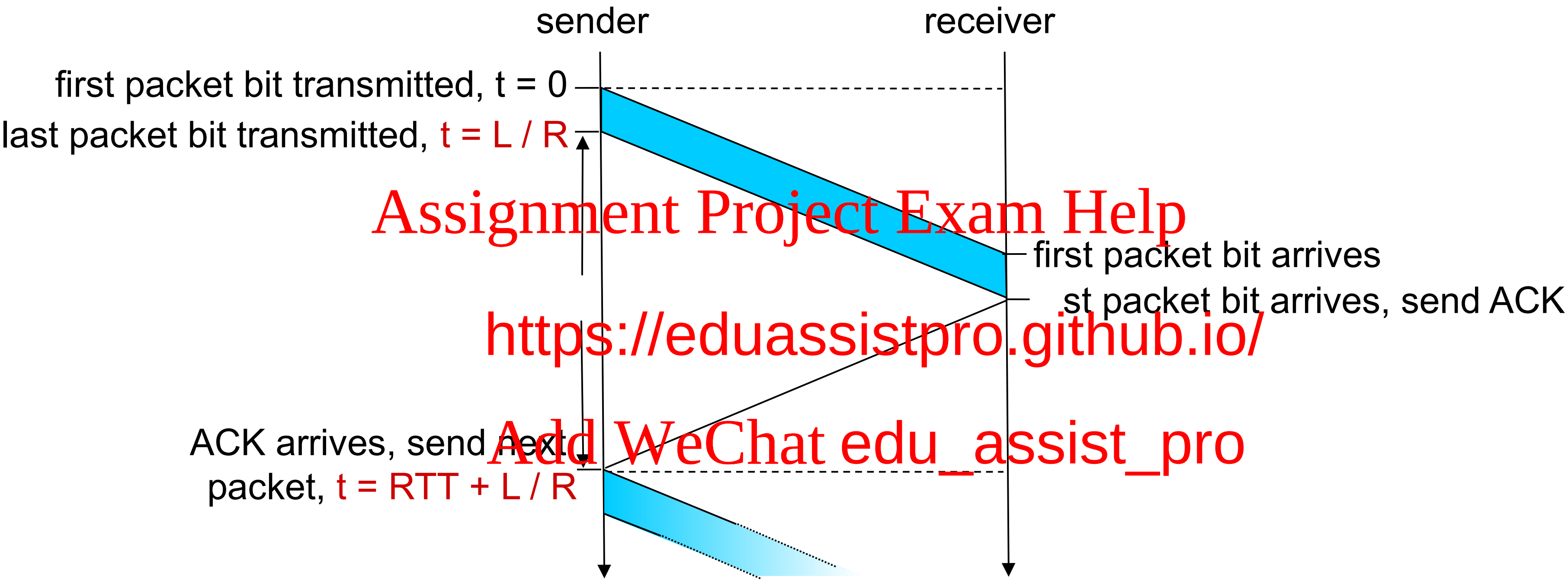
- U $_{sender}$: *utilization* – fraction of r busy sending

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

- 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



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

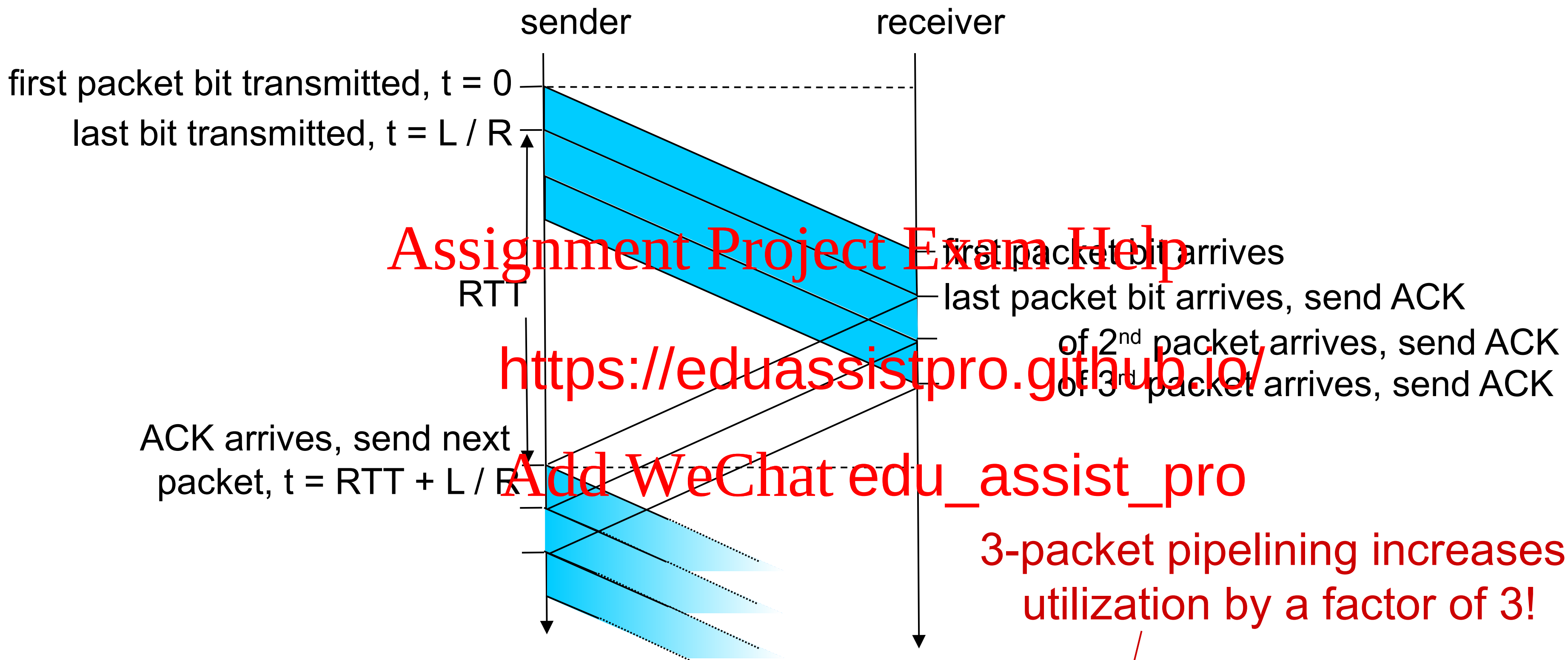
Pipelined protocols

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

- range of sequence numbers must be increased
- buffering at send

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

Pipelining: increased utilization



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

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

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Pipelined protocols: overview

Go-back-N:

- sender can have up to N unacked packets in pipeline
- receiver only se
  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* packet
- ntains timer for ked 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"*
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- timer for oldest in-flight pkt
- *timeout(n):* retransmit packet n and all higher seq # pkts in window

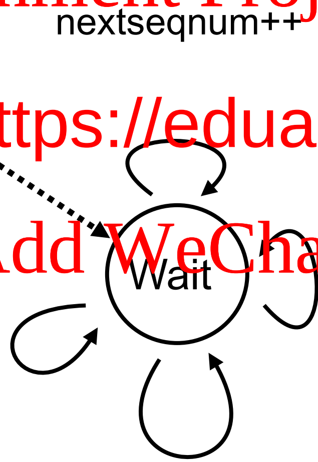
GBN: sender extended FSM

rdt_send(data)

if (nextseqnum < base+N) {
    sndpkt[nextseqnum] = make_pkt(nextseqnum,data,chksum)
    udt_send(sndpkt[nextseqnum])
    if (base == nextseqnum)
        start_timer
    nextseqnum++

Λ

base=1
nextseqnum=1



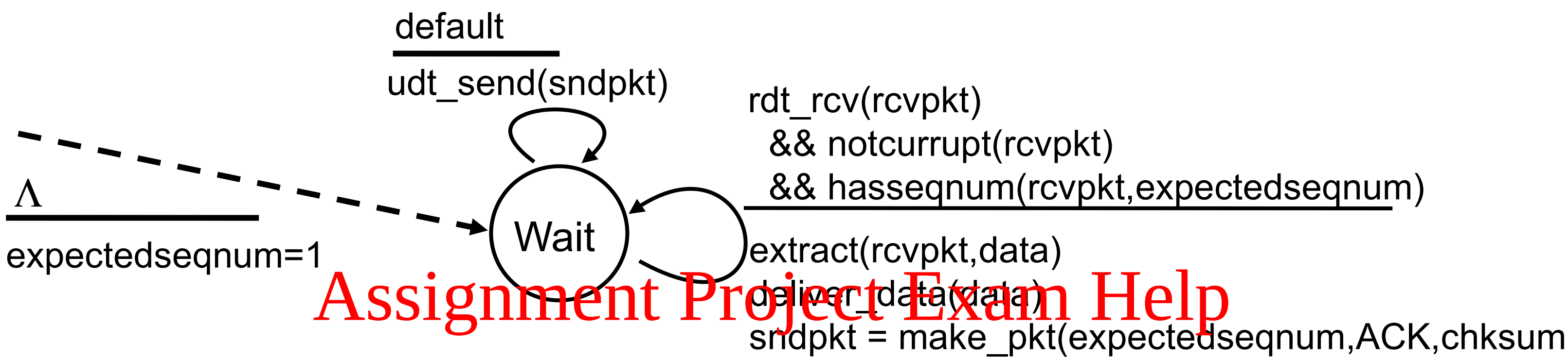
[base])
[base+1])
…
udt_send(sndpkt[nextseqnum-1])

rdt_rcv(rcvpkt)
  && corrupt(rcvpkt)

rdt_rcv(rcvpkt) &&
  notcorrupt(rcvpkt)

base = getacknum(rcvpkt)+1
If (base == nextseqnum)
    stop_timer
else
    start_timer

GBN: receiver extended FSM



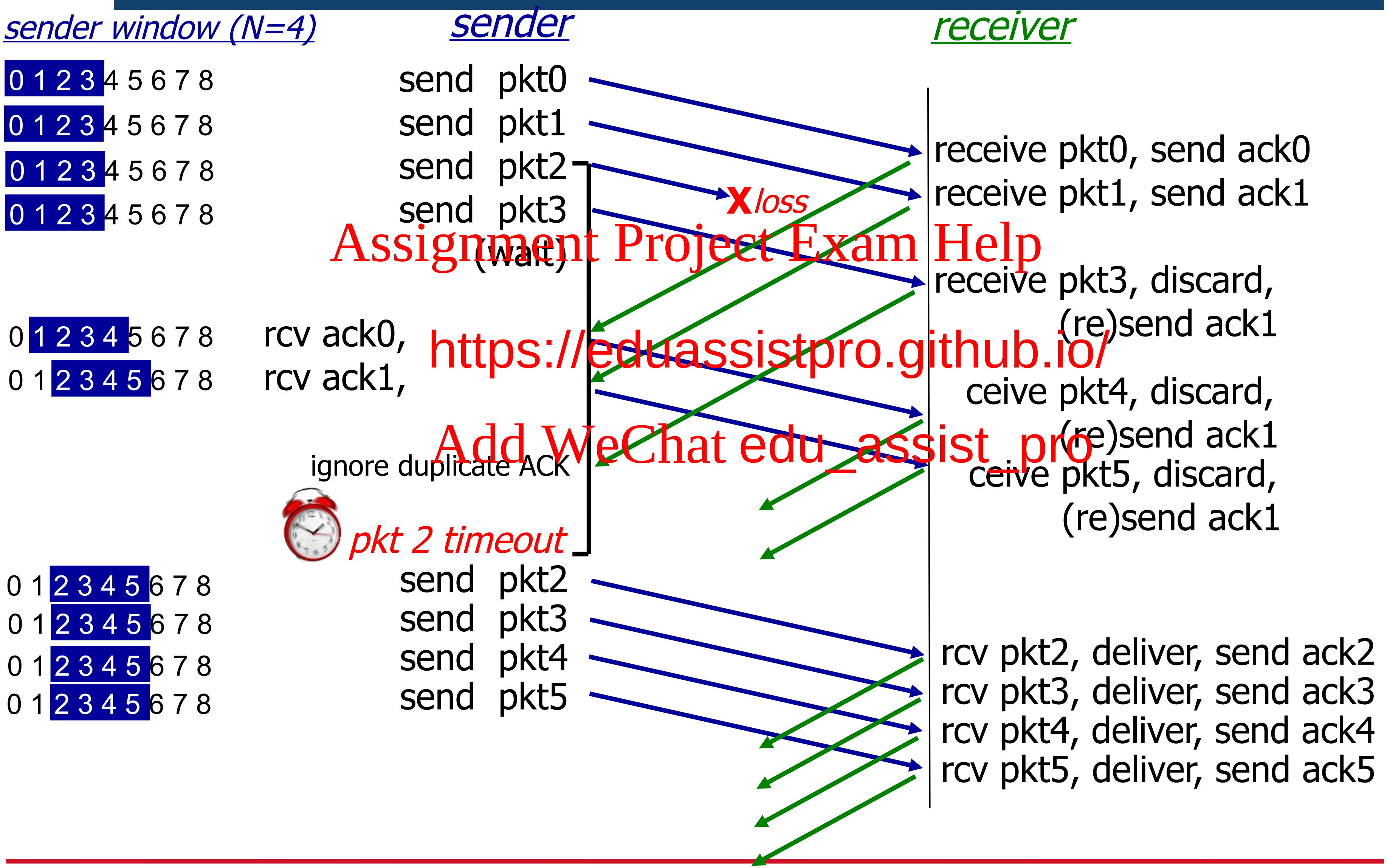
ACK-only: always send ACK [illegible]ly-received pkt with highest *in-order* seq #

- 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



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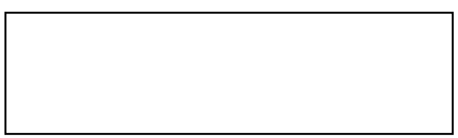
Selective repeat

- receiver *individually* acknowledges all correctly received pkts
  - buffers pkts, as needed, for eventual in-order delivery to upper layer
- sender only re CK not received
  - sender timer for each unACKe
- 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, res

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 #

receiver

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

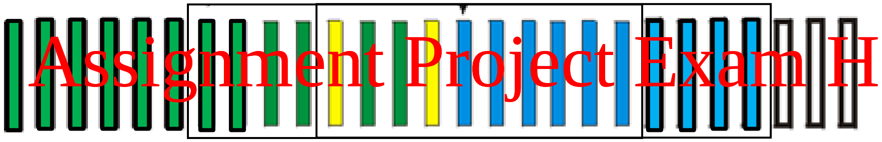
- send ACK(n)
- out-of-order: buffer
- der: deliver (also er buffered, in-order indow to t-yet-received pkt

[rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore



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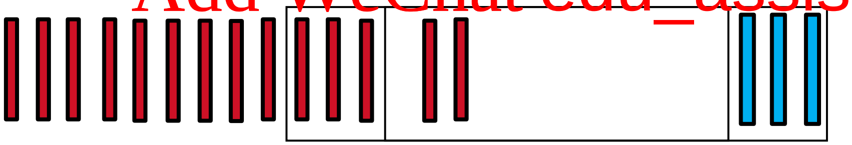
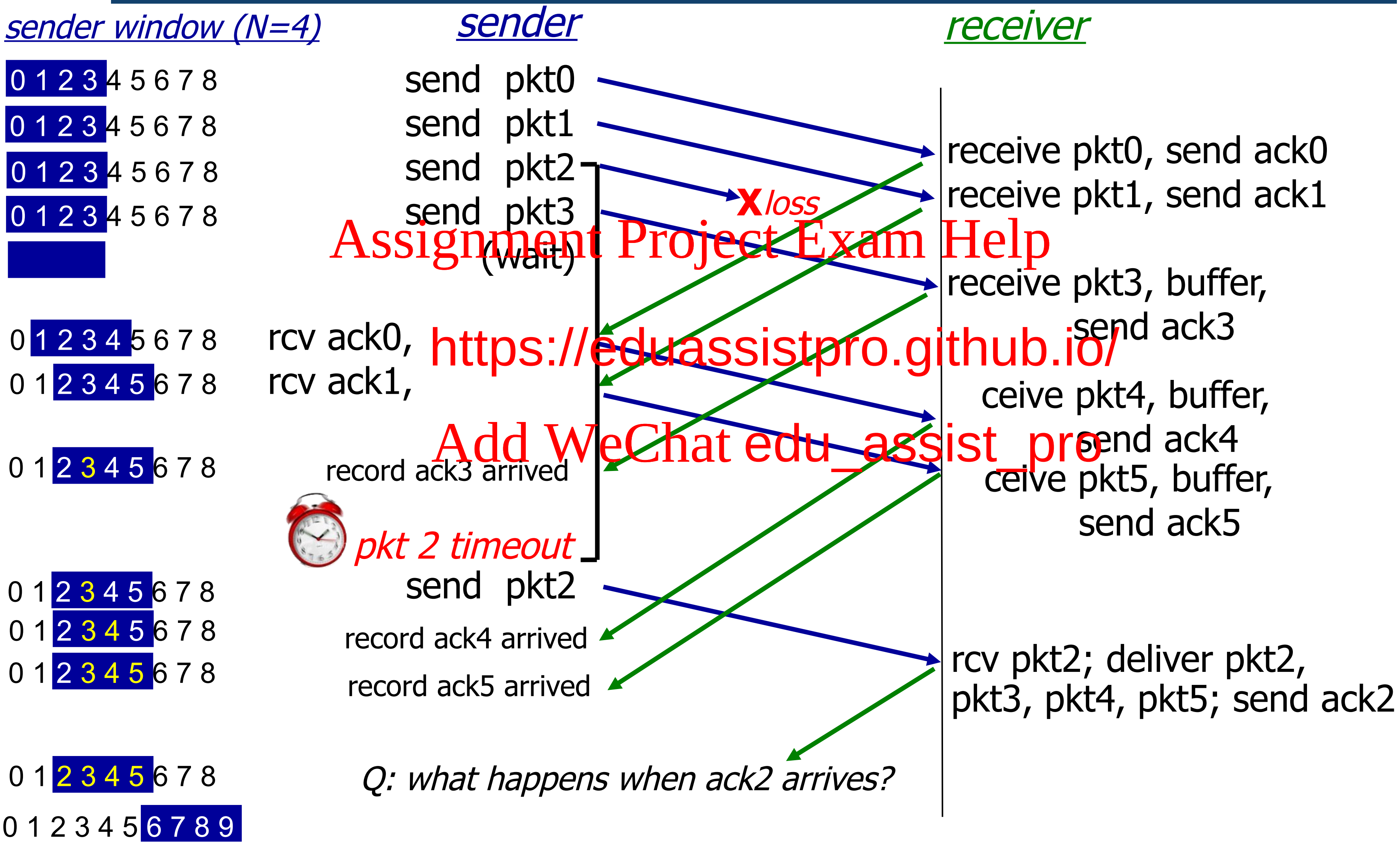


Selective repeat in action



Connection-oriented Transport 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
  - imum segment size
- **n-oriented:**
  - (exchange of control
  - ender, receiver state before data exchange
- **flow controlled:**
  - sender will not overwhelm receiver

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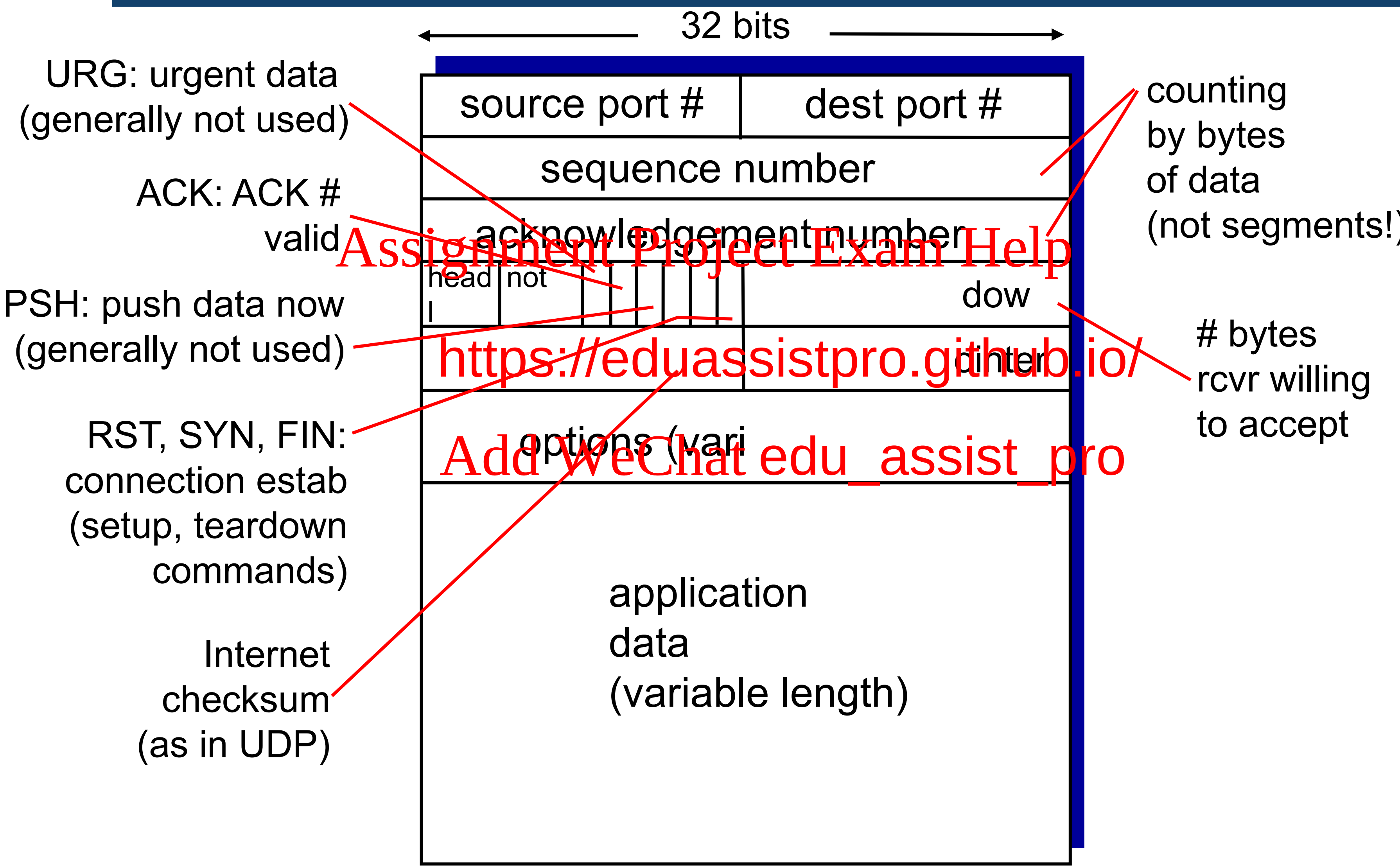
TCP segment structure



TCP seq. numbers, ACKs

sequence numbers:

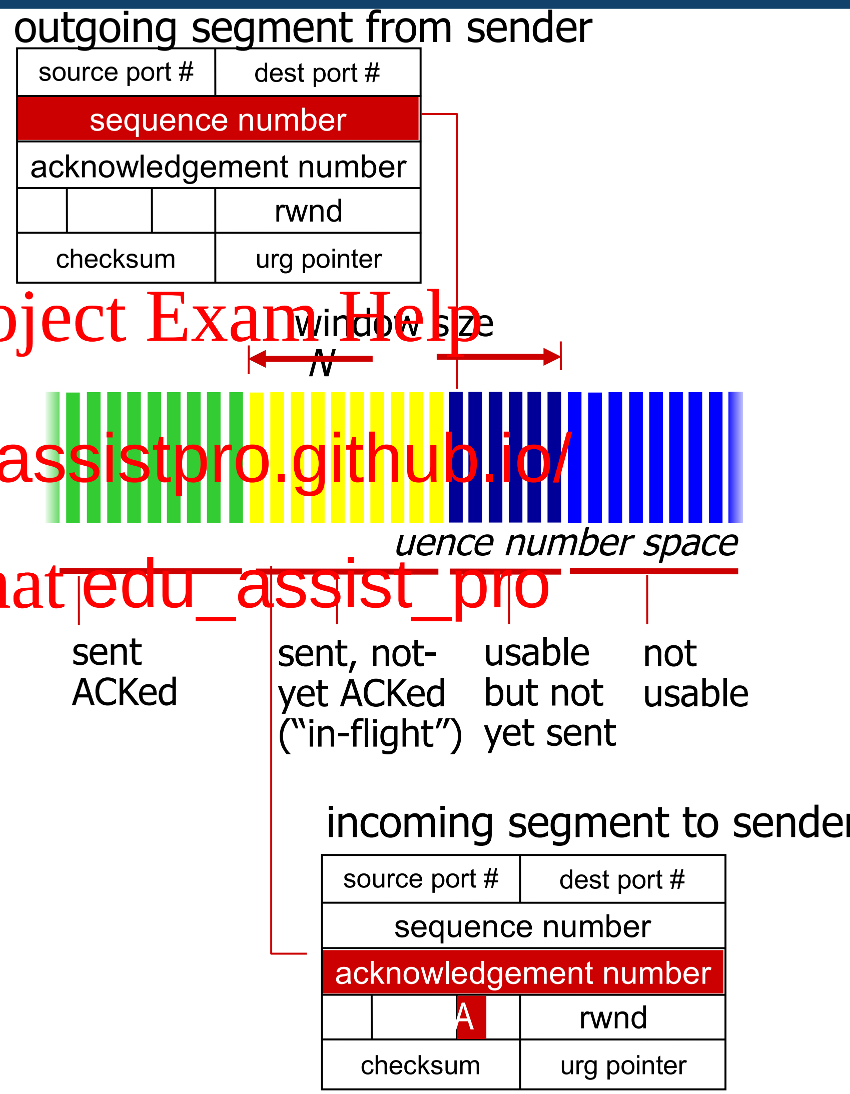
- "number" of first byte in segment's data

acknowledgements:

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

Q: how receiver handles out-of-order segments

- A: TCP spec doesn't say,
- up to implementor
- Most will store, but still use cumulative ACK



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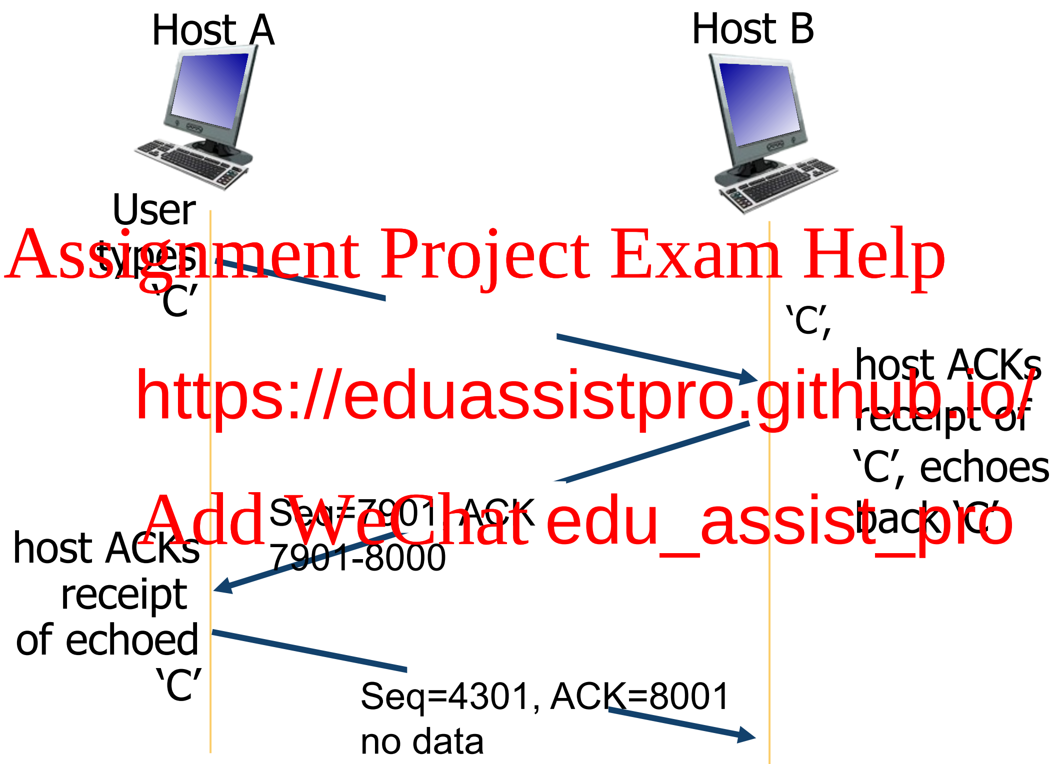
TCP seq. numbers, ACKs



simple telnet scenario

TCP round trip time, timeout

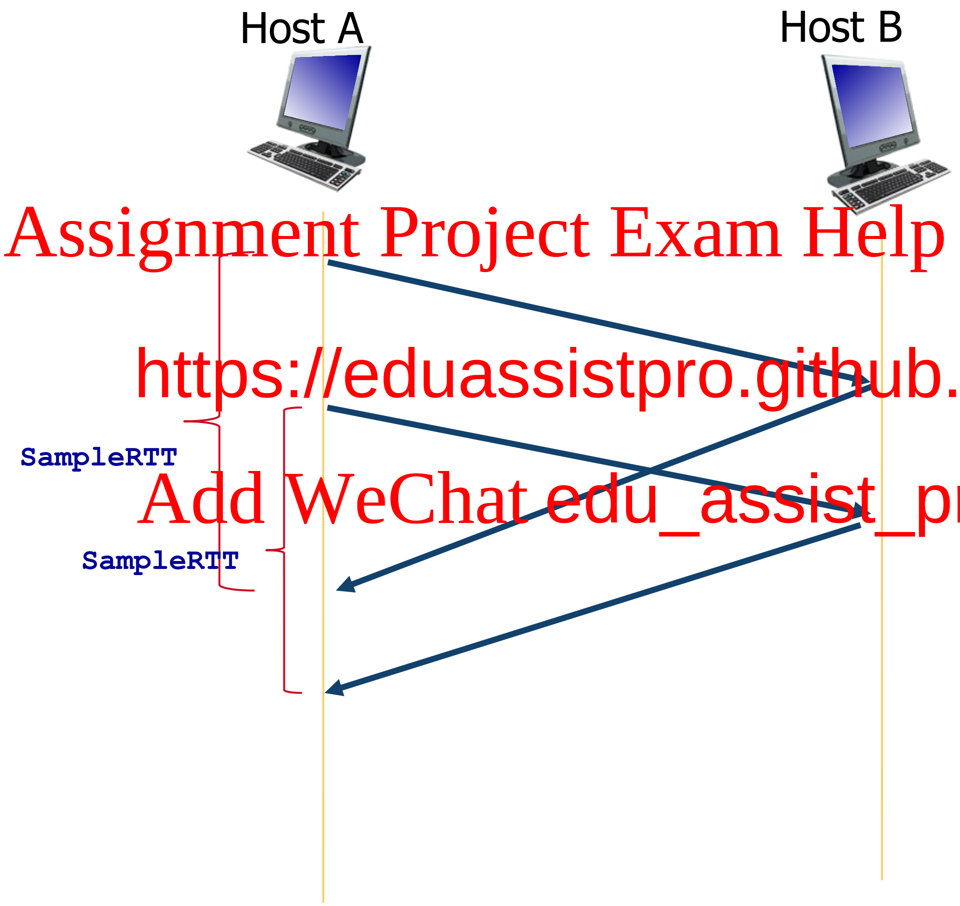
Q: how to set TCP timeout value?

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

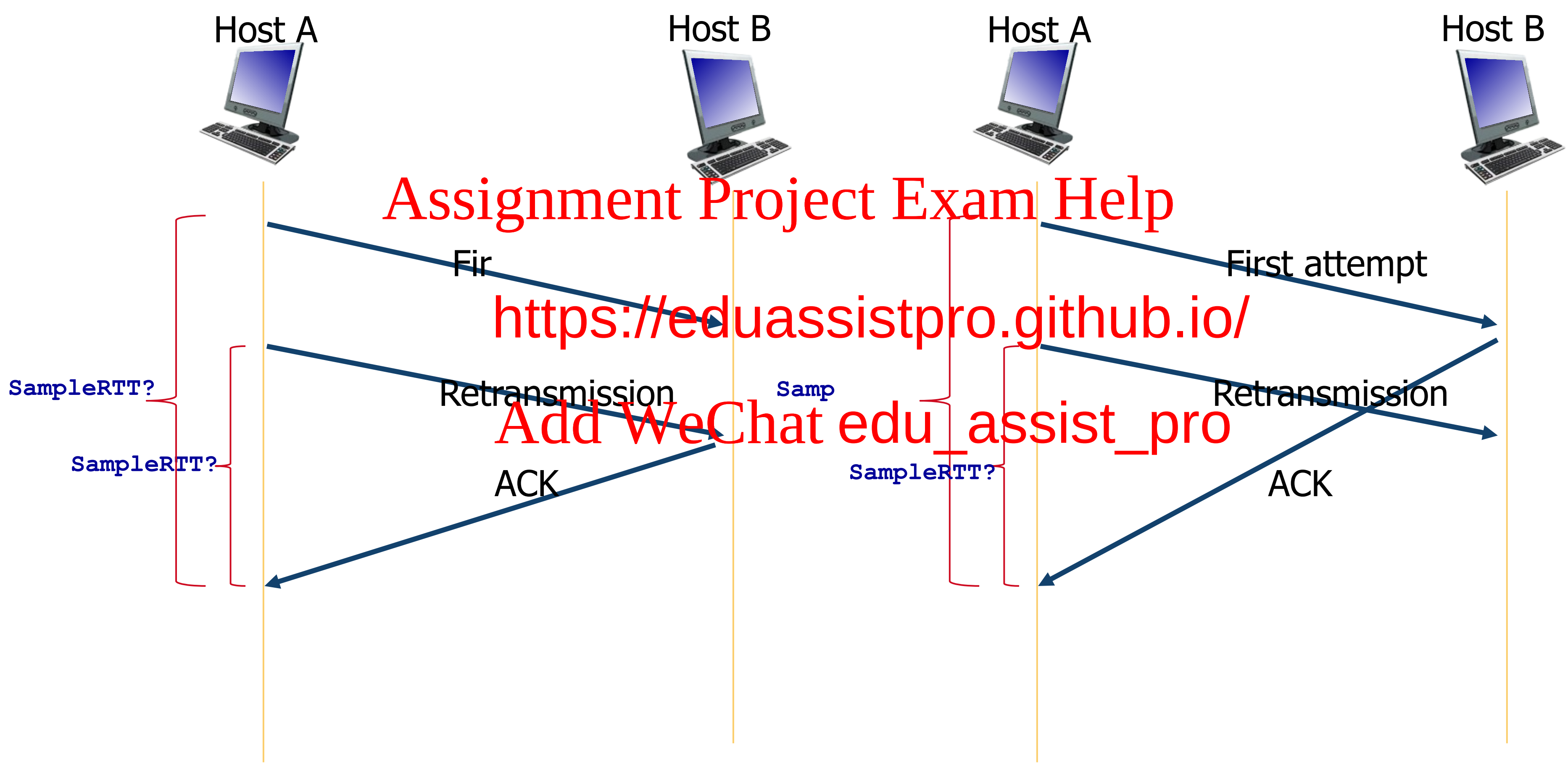
Q: how to estimate RTT?

- **SampleRTT**: measured time from segment transmission ... receipt
- ... etransmissions
- ... T will vary, want estimated RTT "smoother"
  - weighted average of several *recent* measurements, not just current **SampleRTT**

SampleRT



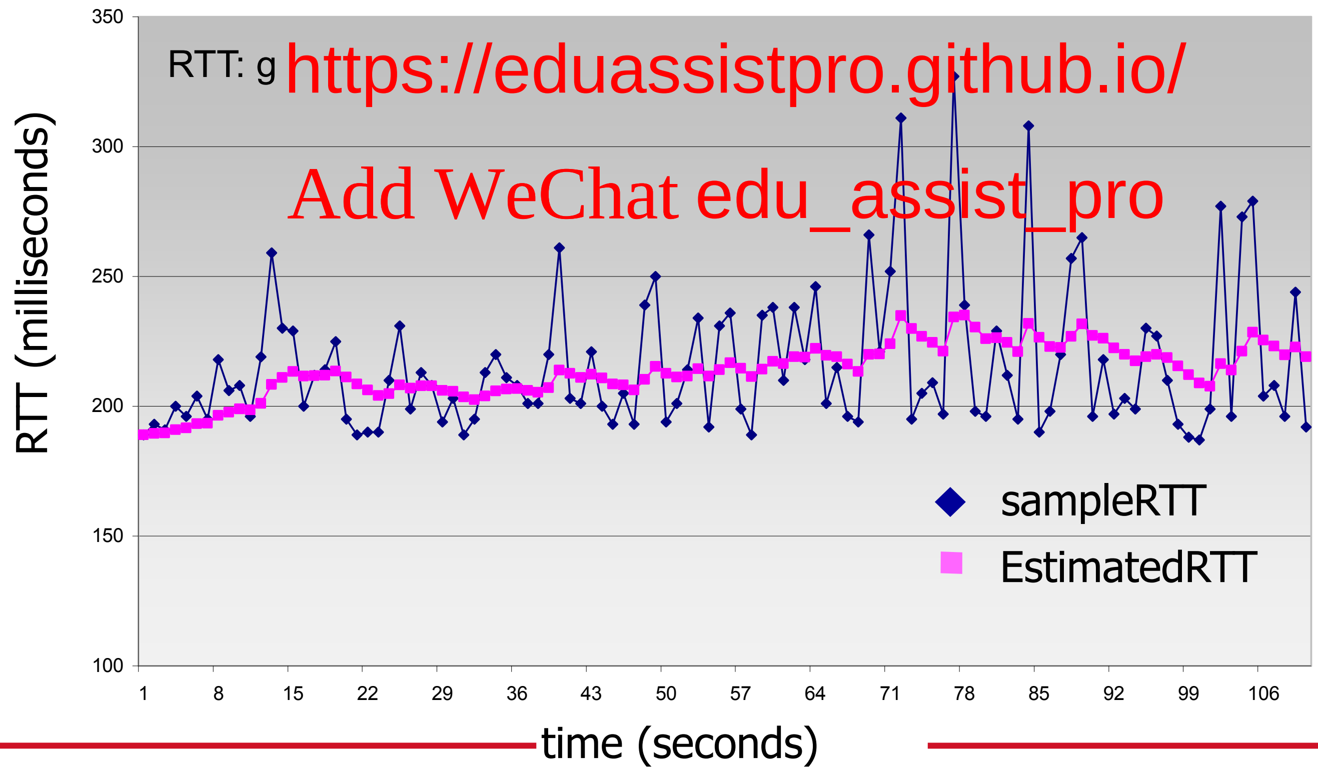
Ignore retransmissions



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 = (
tedRTT|
```



```
(typically,
```

```
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 segmen
  - cumulative acks
  - single retransmission timer
- retransmissions triggered by:
  - timeout events
  - duplicate acks

y consider TCP sen

icate 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 running
  - think of timer as for oldest unacked segment
  - expiration interval: **`TimeOutInterval`**

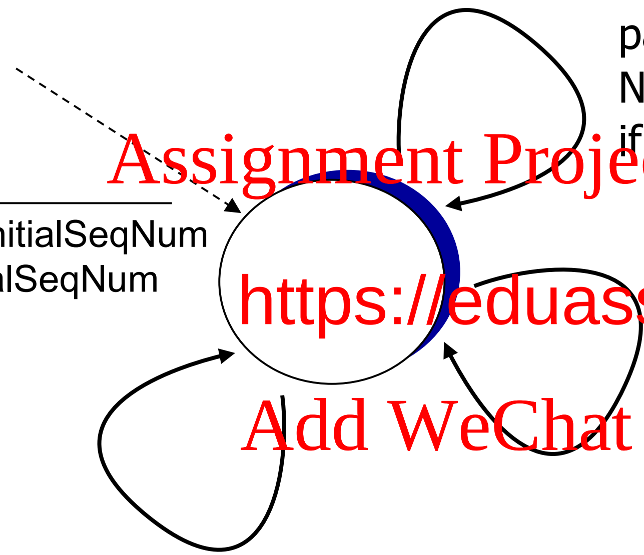
timeout:

- retransmit segment that caused timeout
- restart timer

if edges pr cked 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

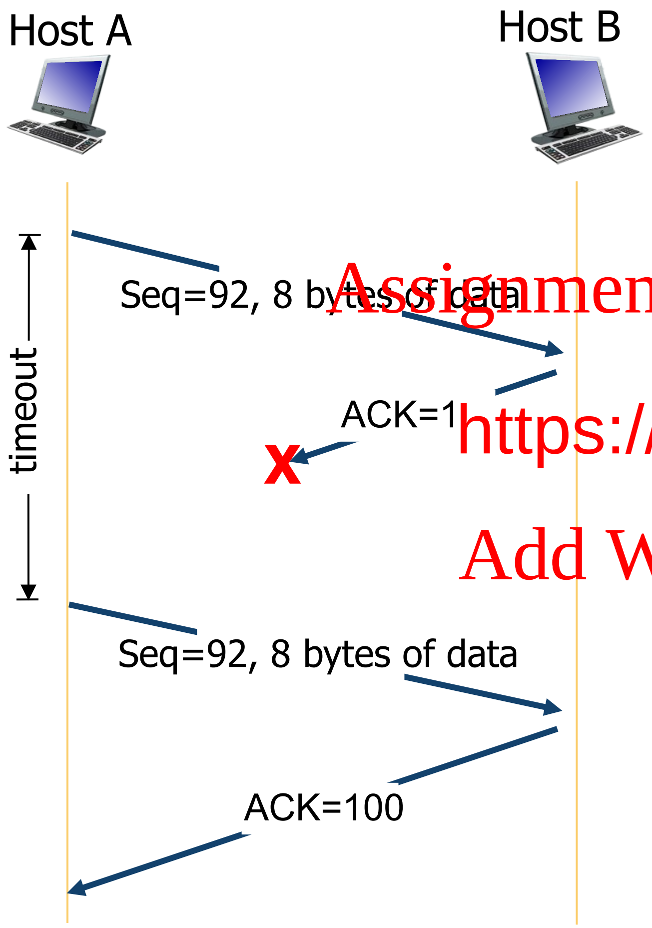
et-acked segment
mallest seq. #

start timer

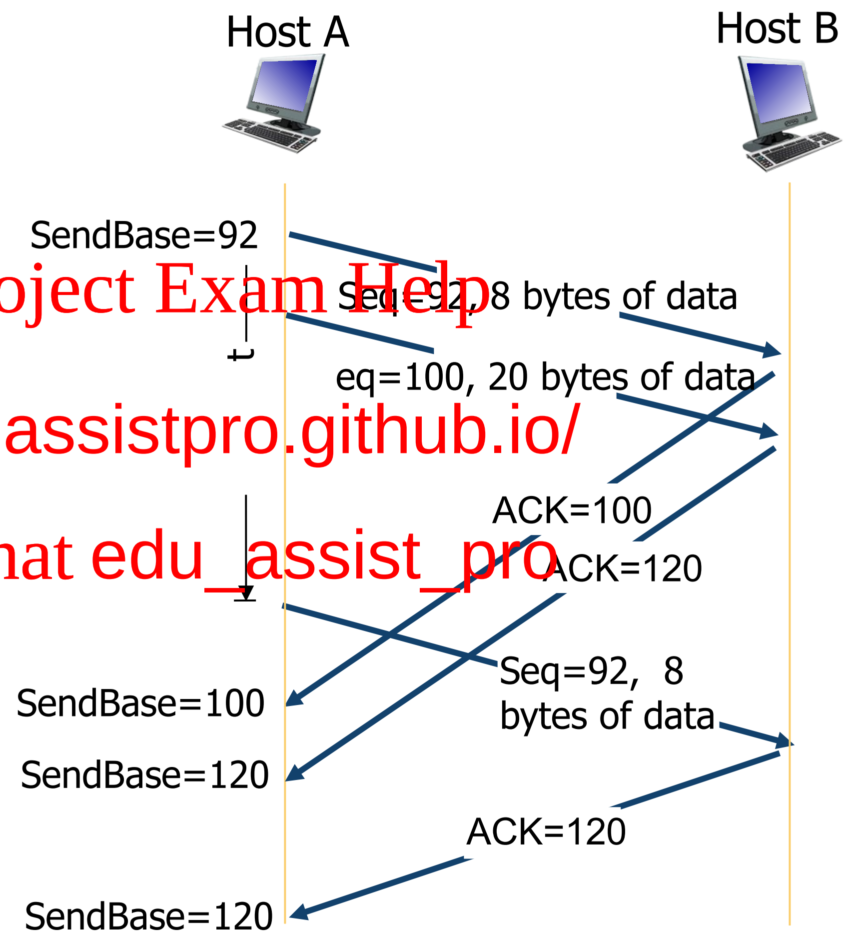
ACK received, with ACK field value y

```
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

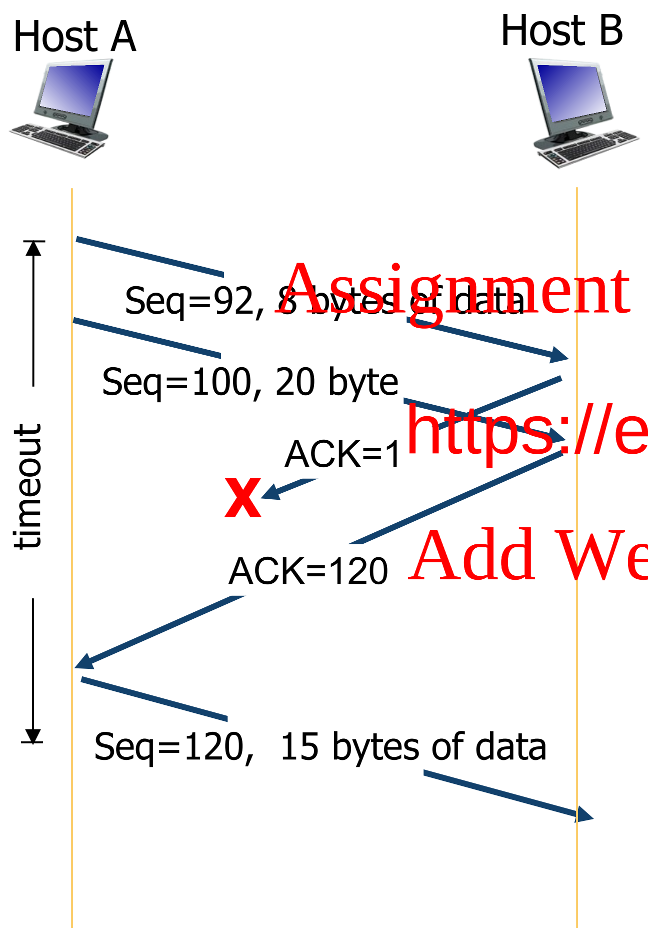


lost ACK scenario



premature timeout

TCP: retransmission scenarios

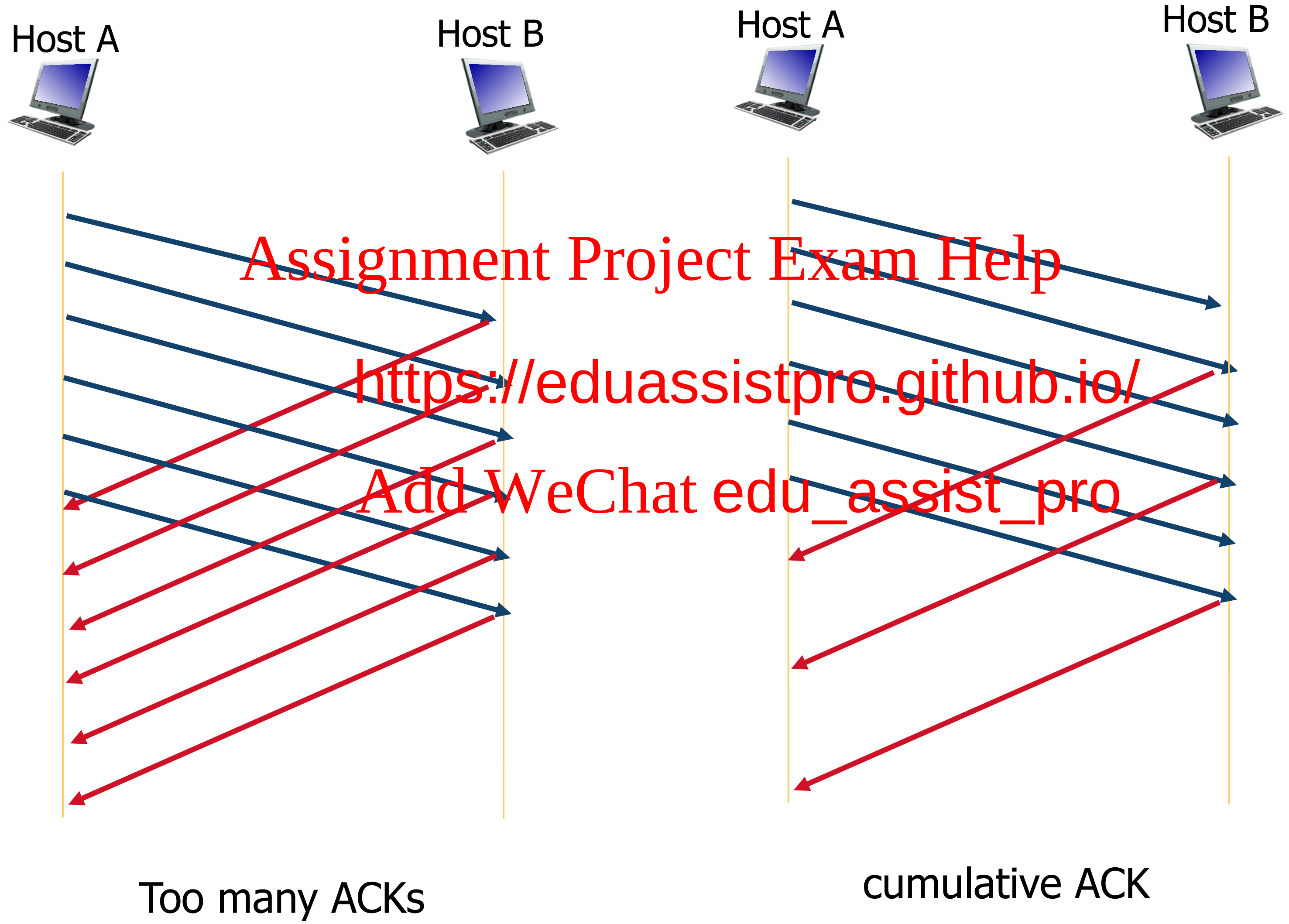


cumulative ACK

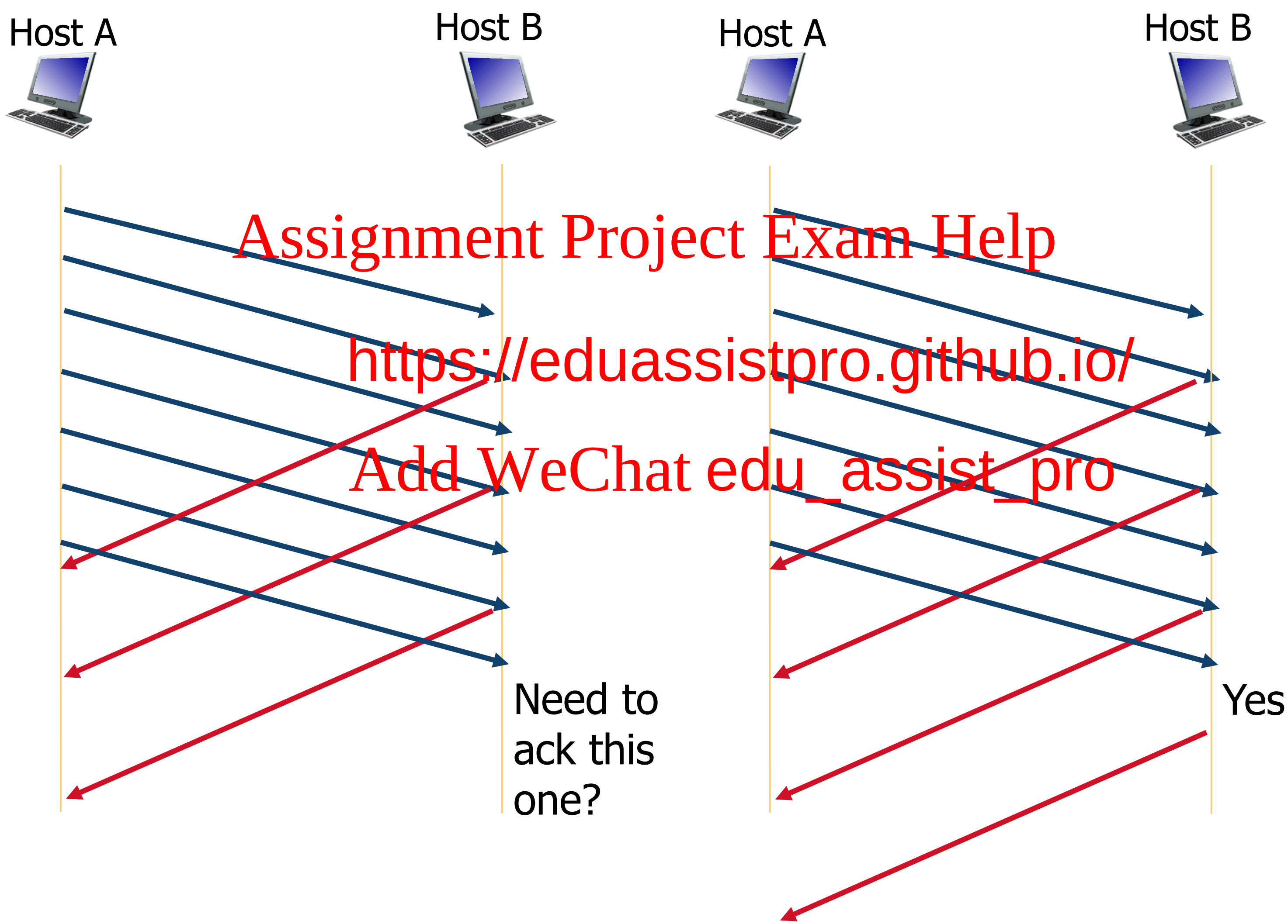
TCP ACK generation [RFC 1122, RFC 2581]

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 se expected seq #. One other segment has ACK pending	end single cumulative A th 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 ACK



TCP ACK



TCP fast retransmit

› time-out period often relatively long:
- long delay before resending lost packet

› detect lost segm[...] duplicate ACKs.
- sender often sends many segments back-to-back
- if segment is lost, there will likely be many duplicate ACKs.

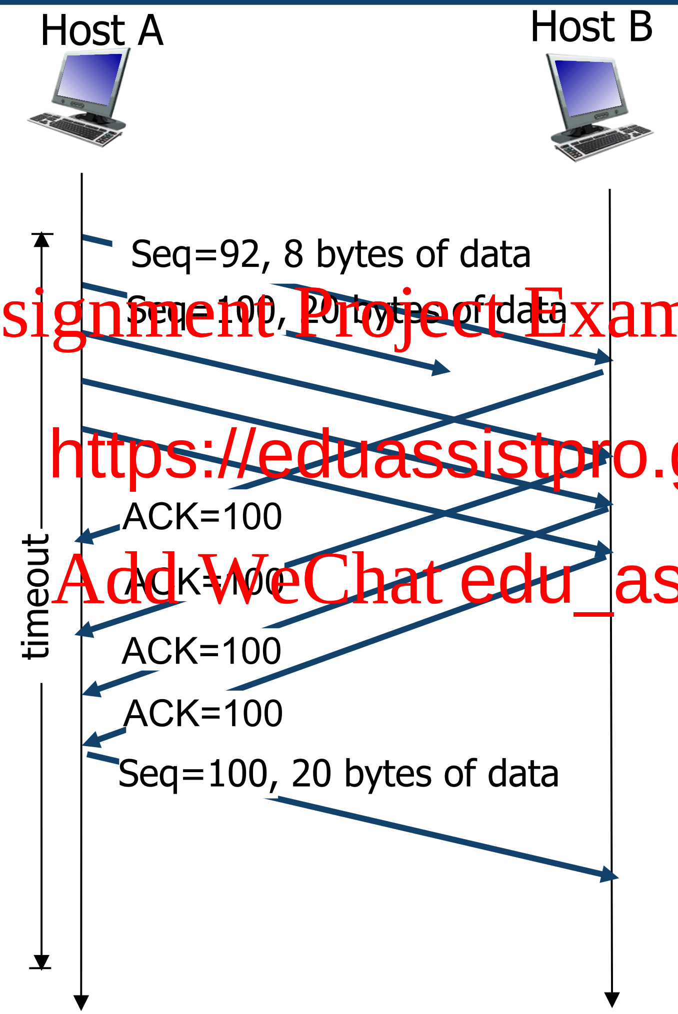
TCP fast retransmit

if sender receives 3 duplicate ACKs for same

icate ACKs”),
acked seg
t seq #

- likely that unacked segment lost, so don’t wait for timeout

TCP fast retransmit

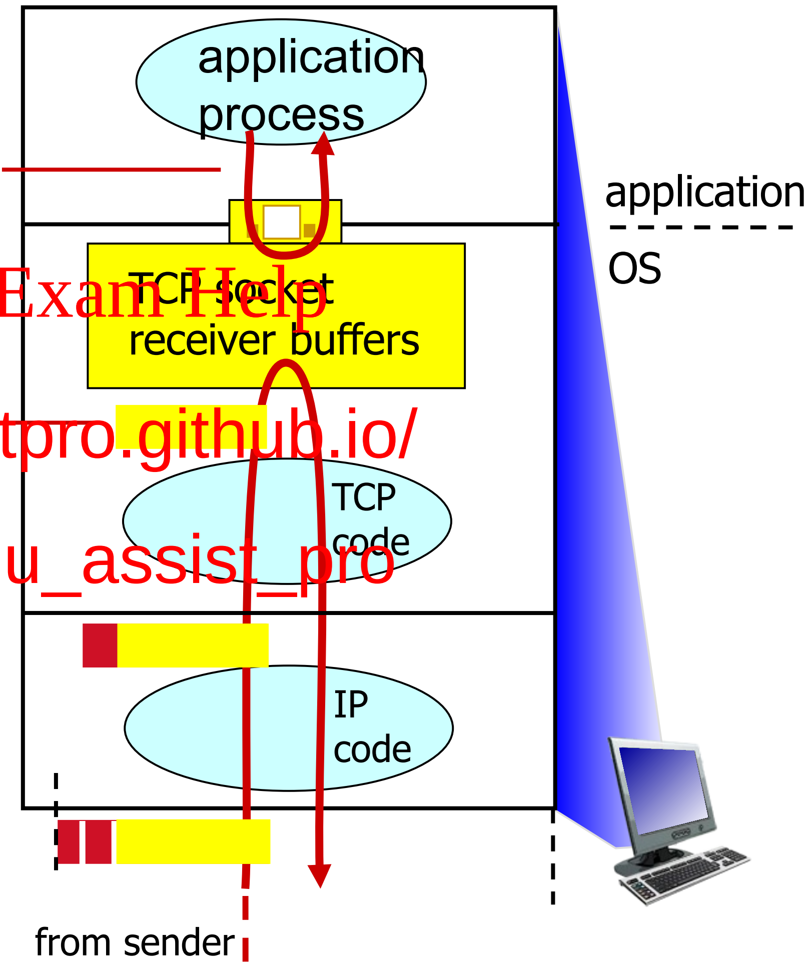


fast retransmit after sender
receipt of triple duplicate ACK

Flow Control in TCP

TCP flow control

application may remove data from TCP socket buffers ….



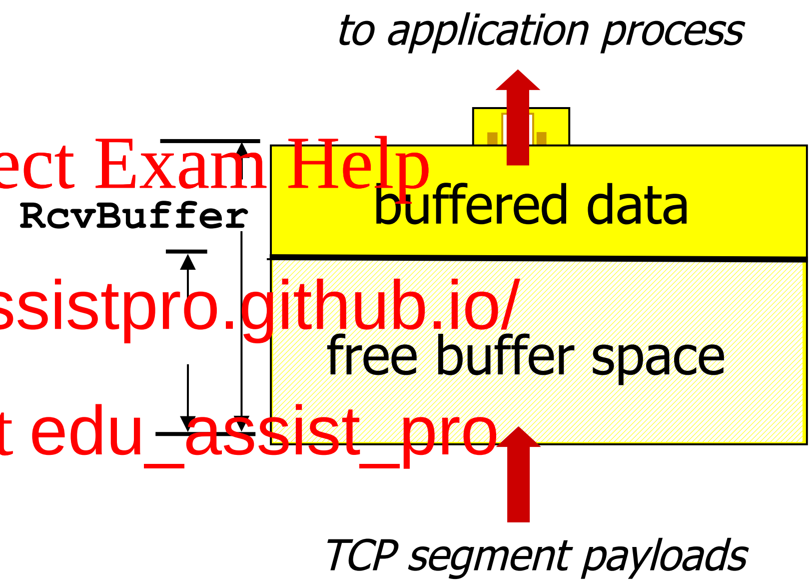
receiver protocol stack

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 se options (typical defau
  - many operating systems autoadjust **`RcvBuffer`**
- sender limits amount of unacked ("in-flight") data to receiver's **`rwnd`** value
- guarantees receive buffer will not overflow



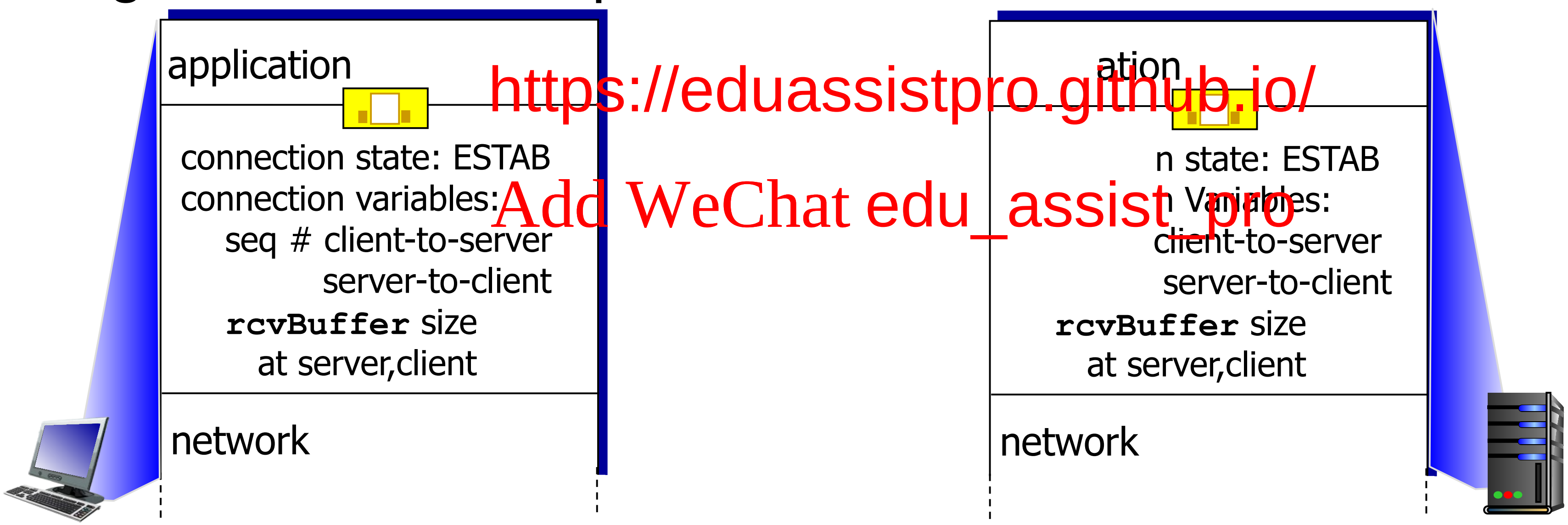
receiver-side buffering

Connection Management in TCP

Connection Management

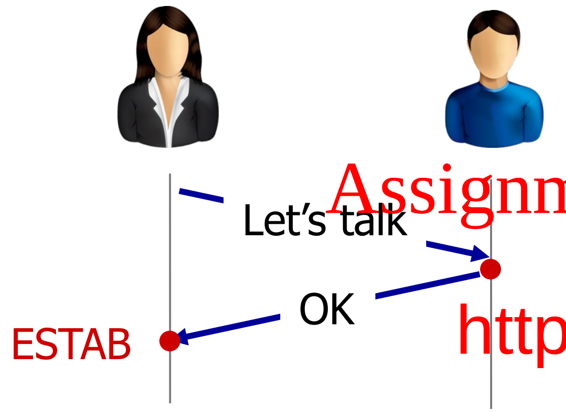
before exchanging data, sender/receiver "handshake":

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



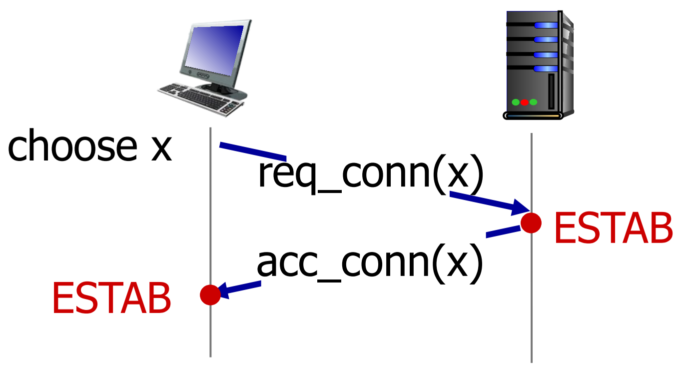
Agreeing to establish a connection

2-way handshake:



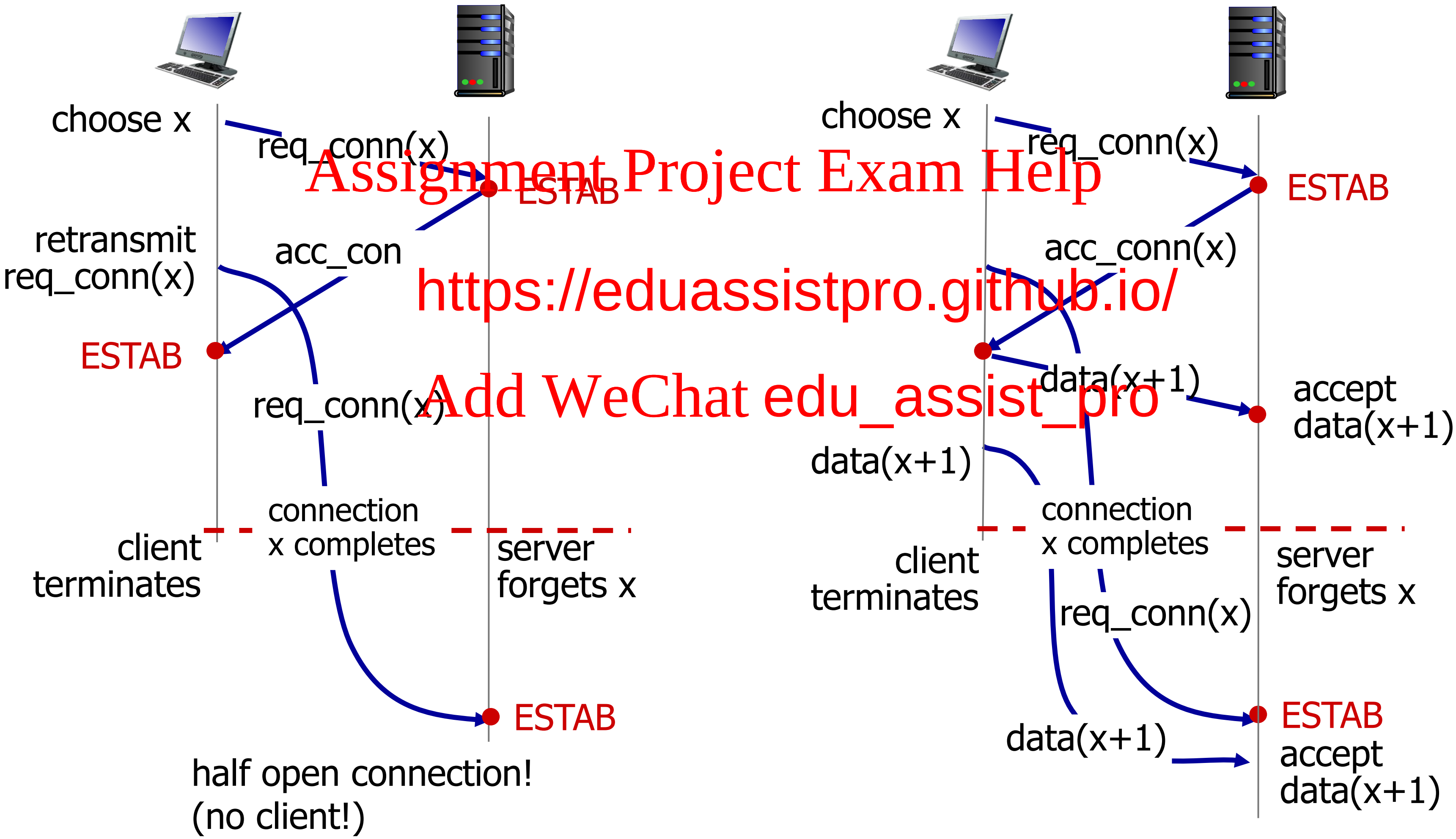
Q: will 2-way handshake always work in network?

- variable delays
- essages (e.g.
- ue to message loss
- ering
- da side

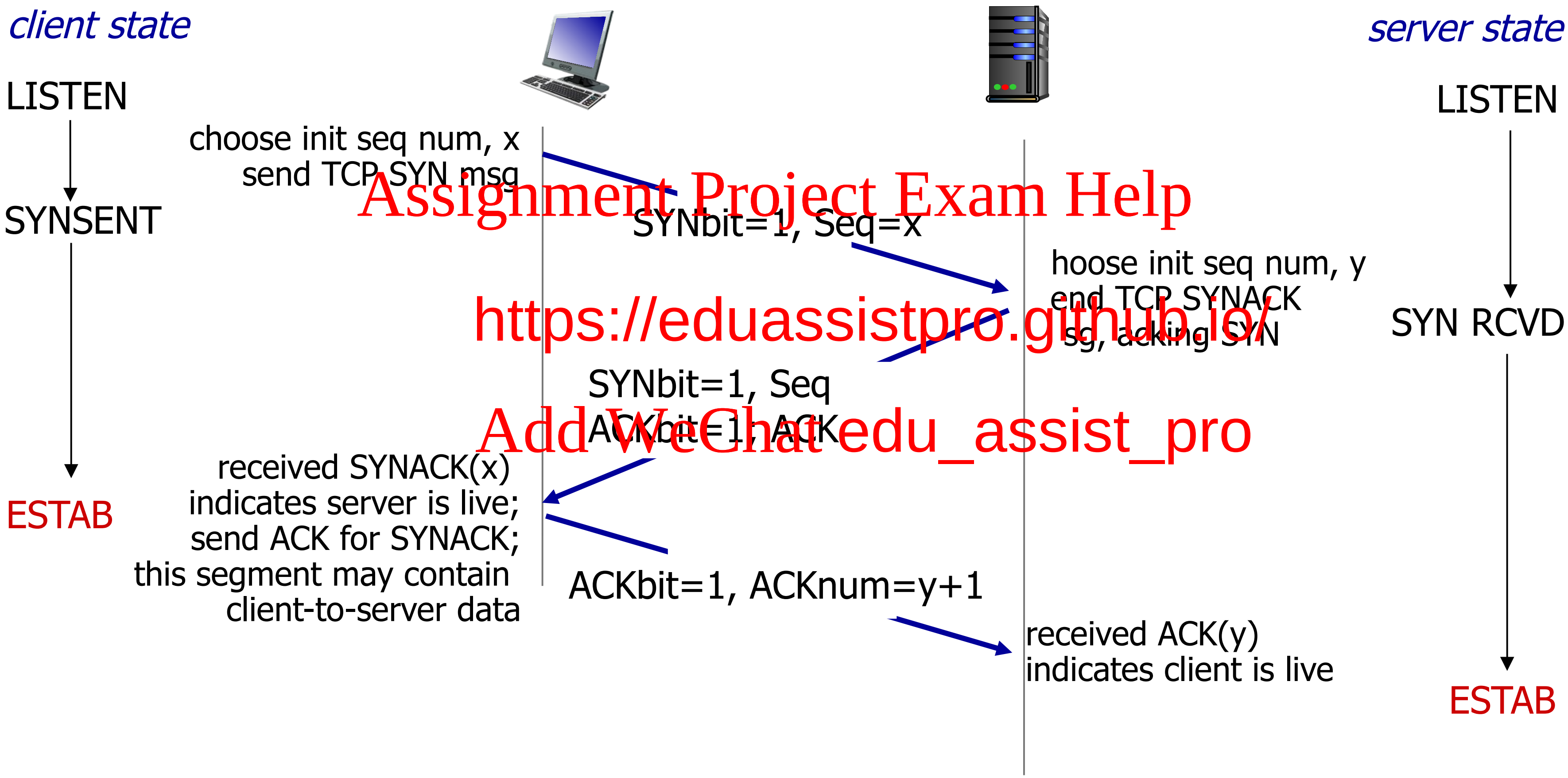


Agreeing to establish a connection

2-way handshake failure scenarios:



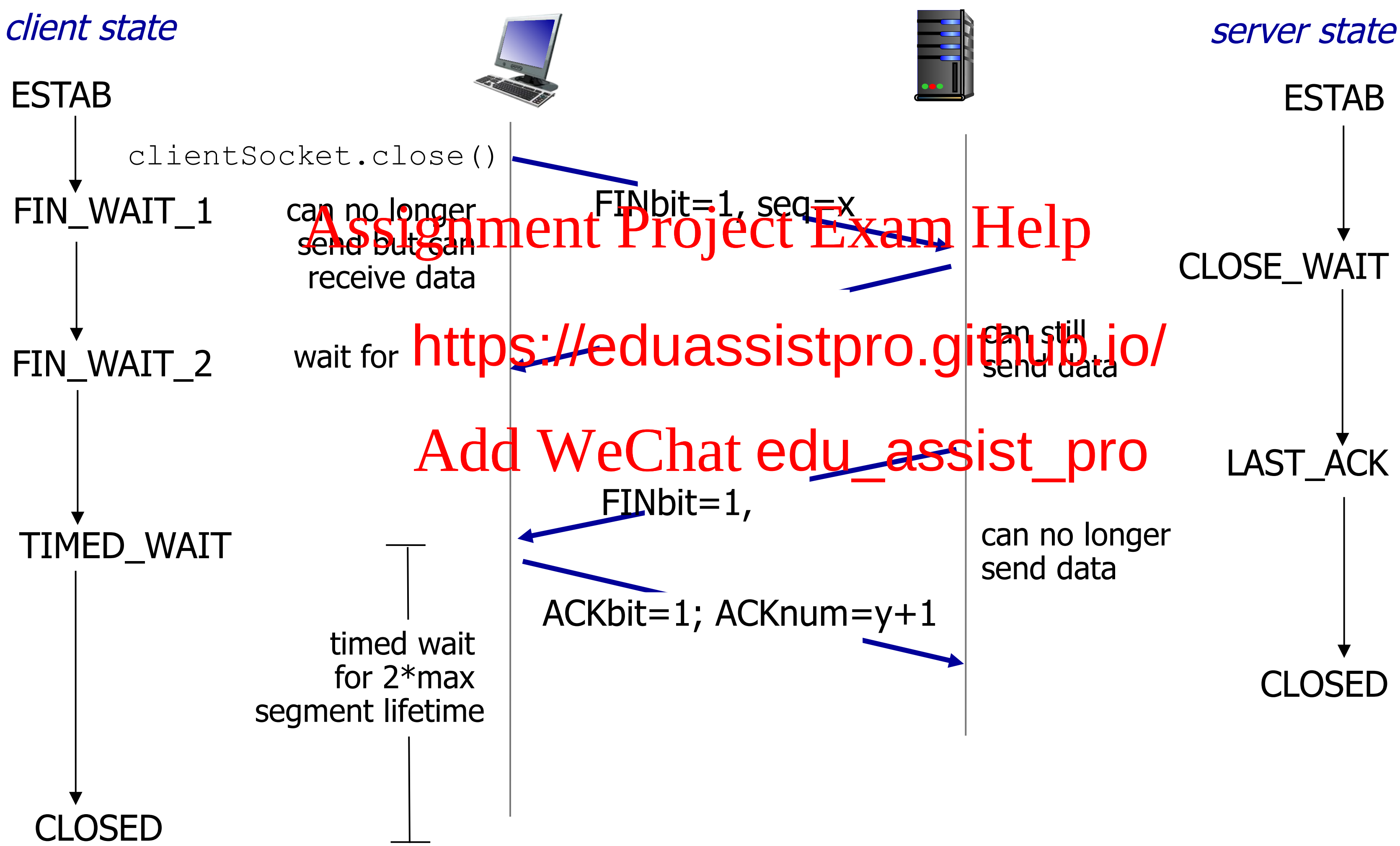
TCP 3-way handshake



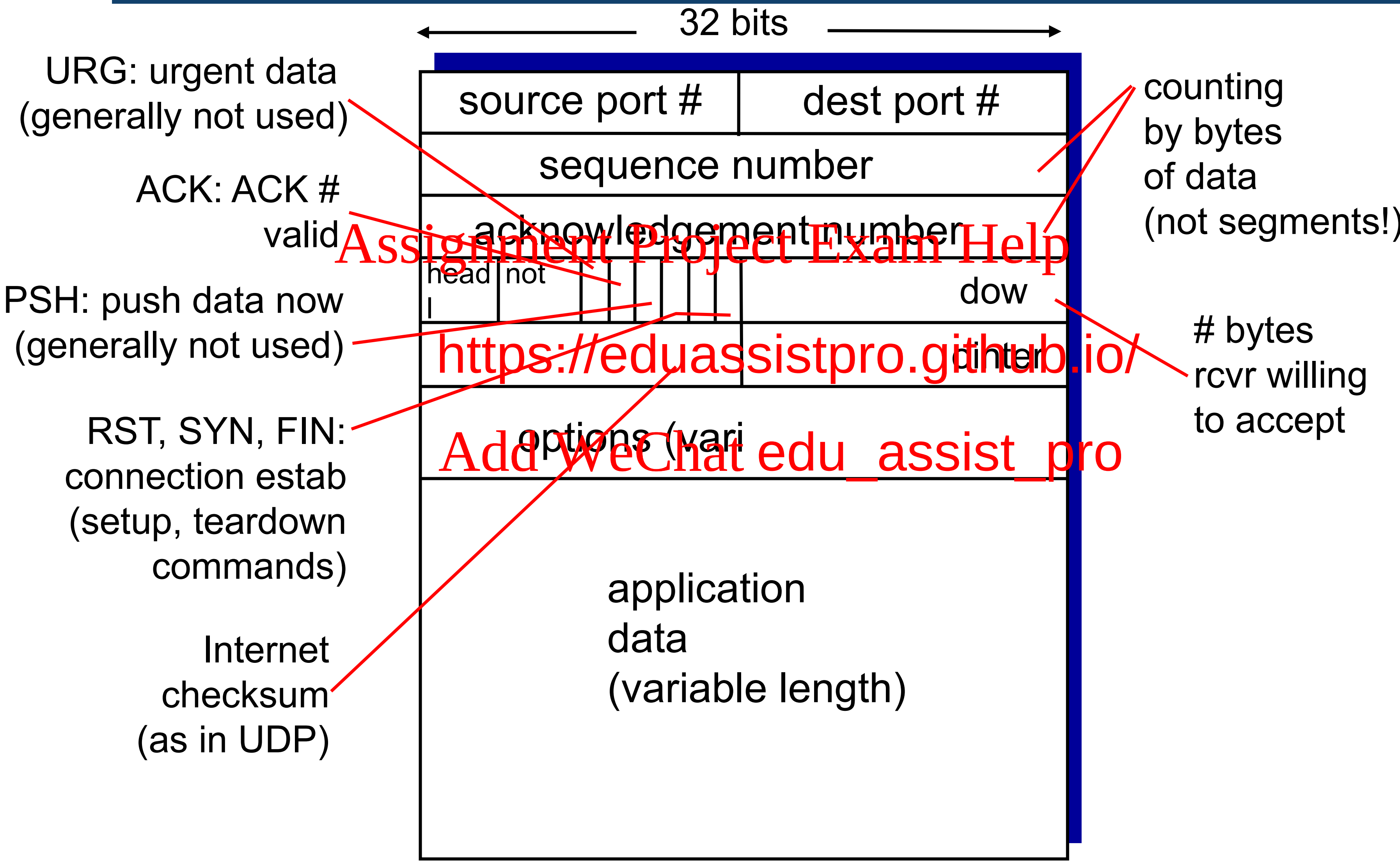
TCP: closing a connection

› 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



Principles of Congestion Control

Principles of congestion control

congestion:

- informally: "too many sources sending too much data too fast fo

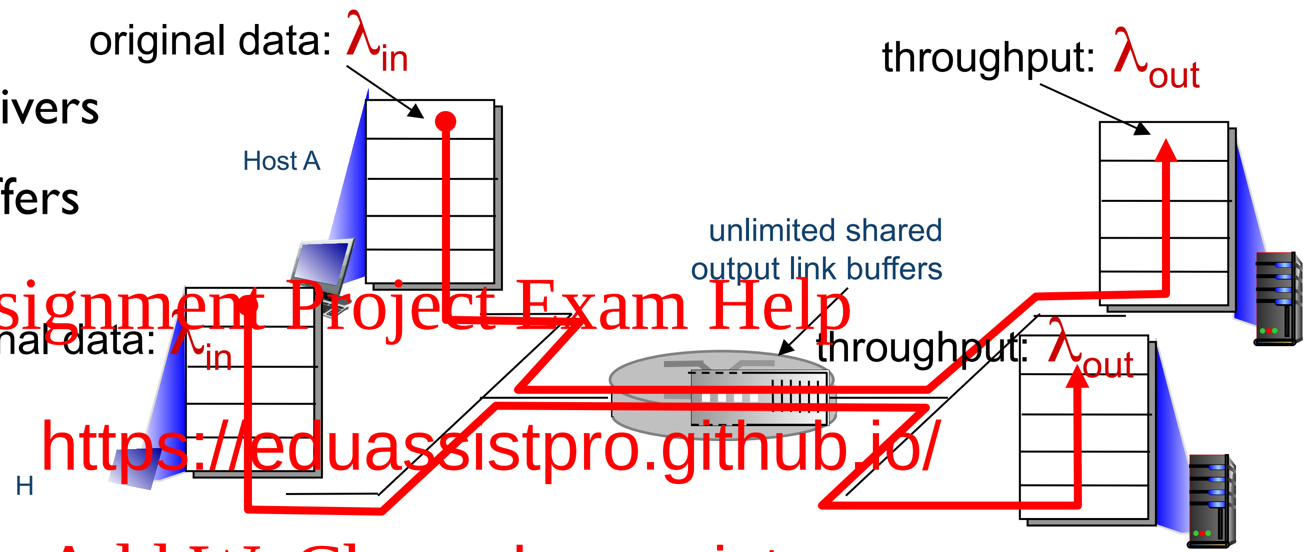
- different from

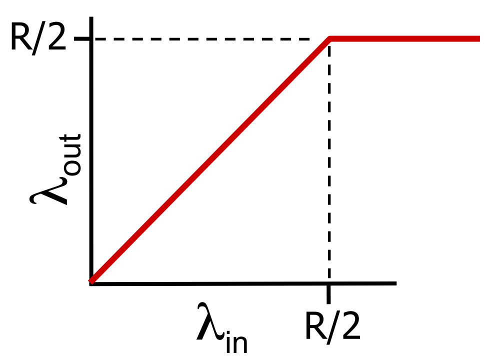
- manifestations:

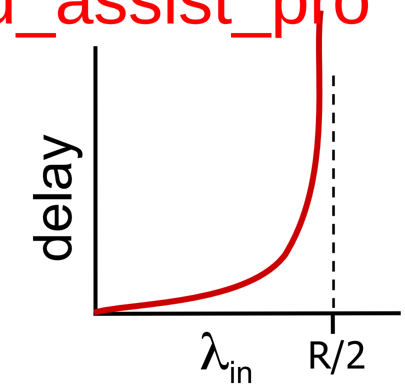
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Causes/costs of congestion: scenario 1

- two senders, two receivers
- one router, infinite buffers
- output link capacity: R
- no retransmission



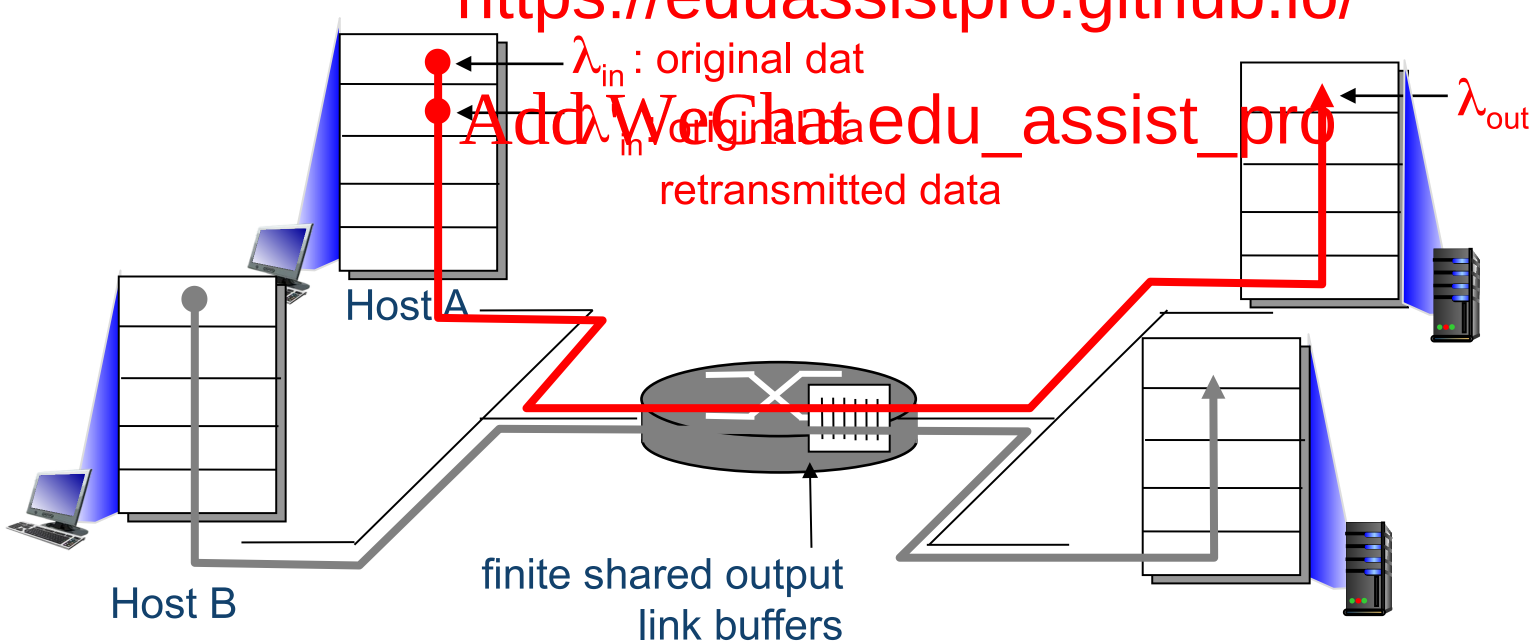




- maximum per-connection throughput: R/2
- large delays as arrival rate, λ_{in}, approaches capacity

Causes/costs of congestion: scenario 2

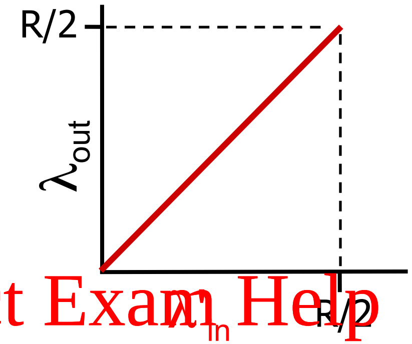
- one router, *finite* buffers
- sender retransmission of timed-out packet
  - application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$ 、
    - Goodput
  - transport-layer input includes *retransmissions* : λ'_{in} λ_{in} ≥

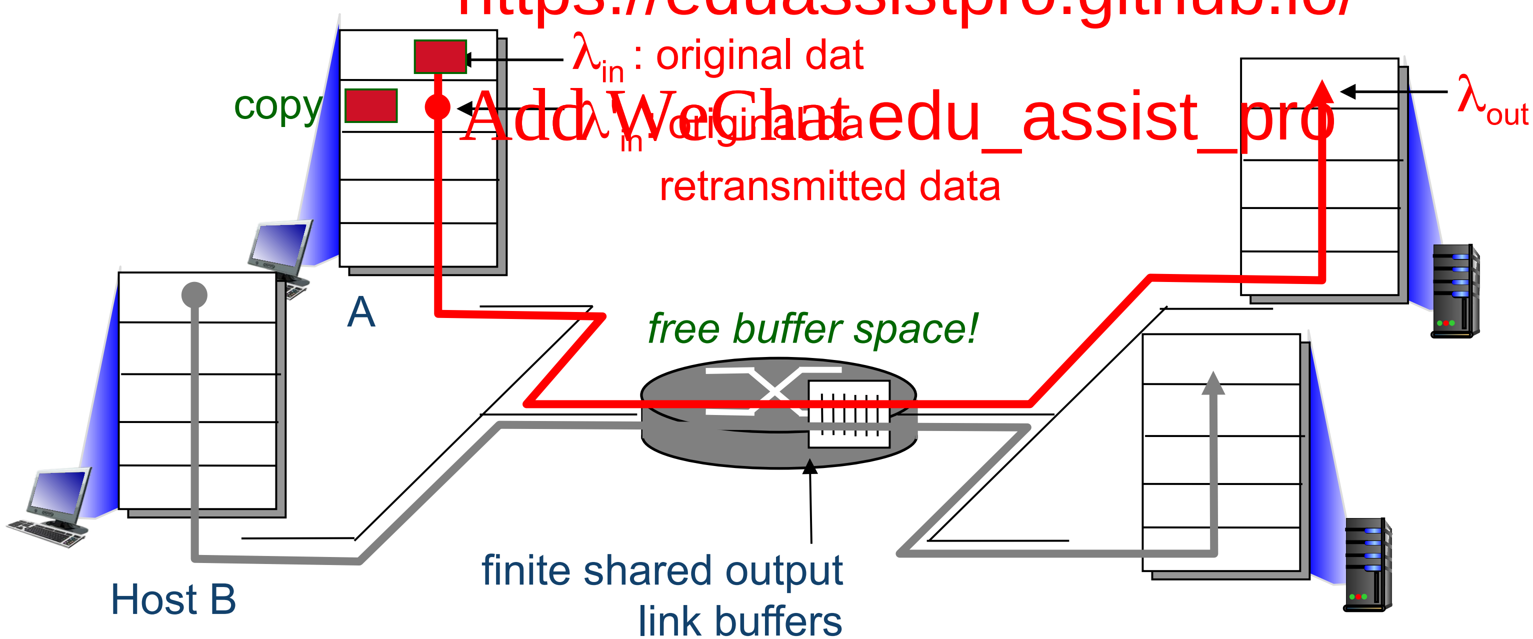


Causes/costs of congestion: scenario 2

idealization: perfect knowledge

- sender sends only when router buffers available

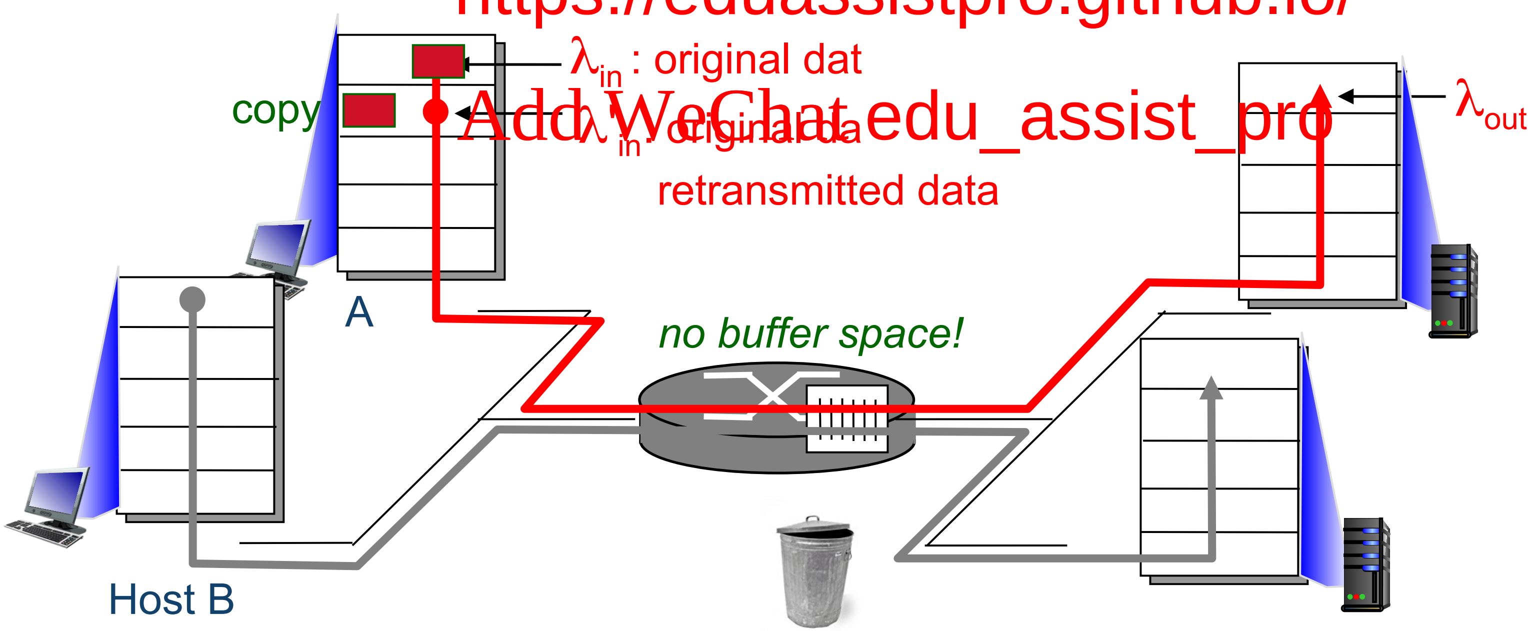




Causes/costs of congestion: scenario 2

Idealization: known loss
packets can be lost, dropped at router due to full buffers

- sender only resends if packet *known* to be lost



Causes/costs of congestion: scenario 2

Idealization: ***known loss***

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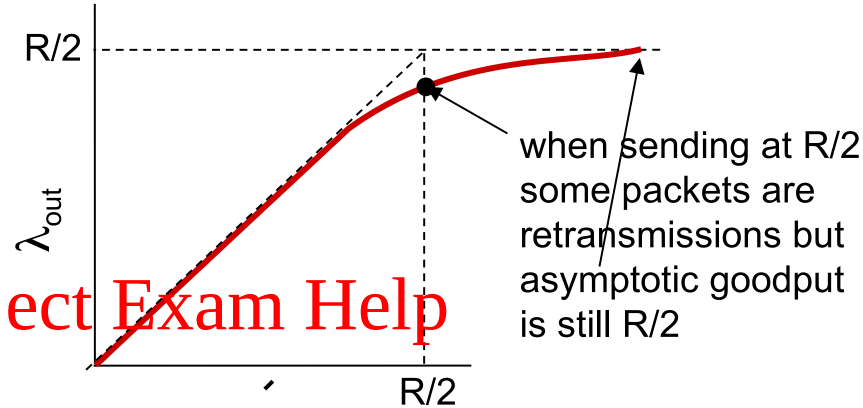
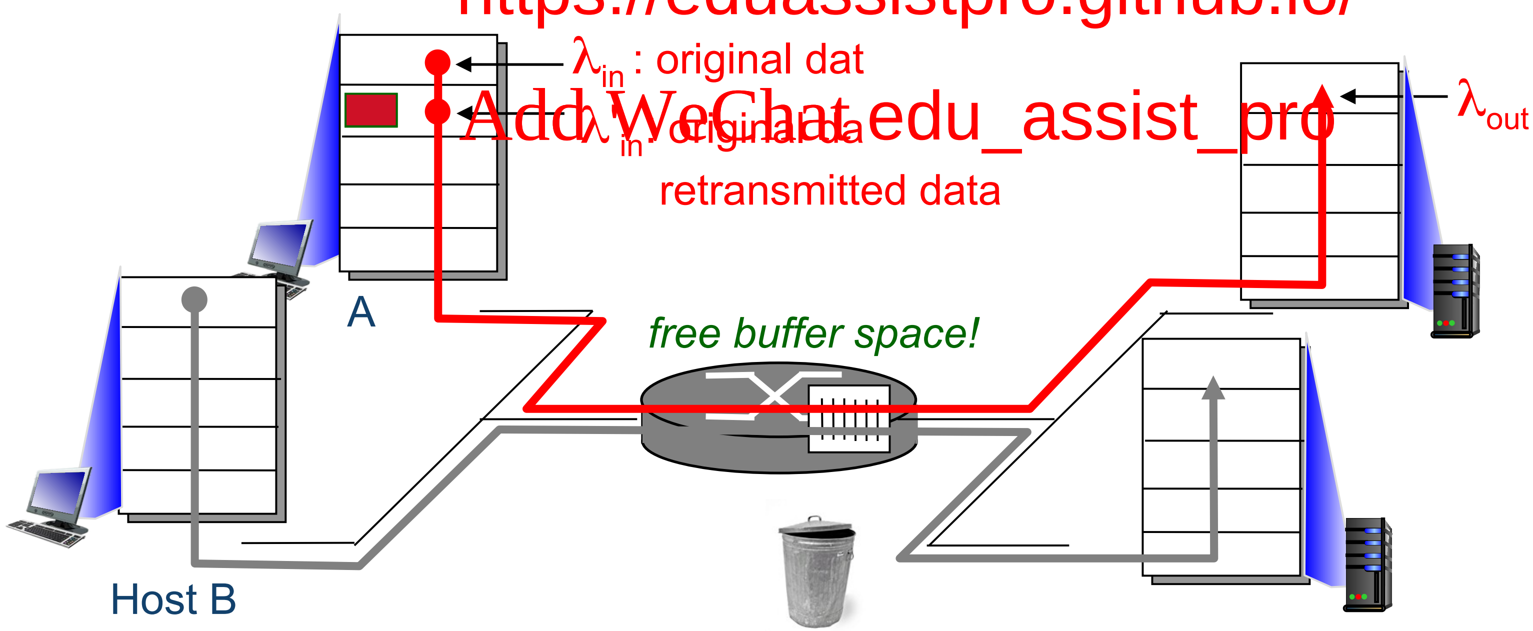
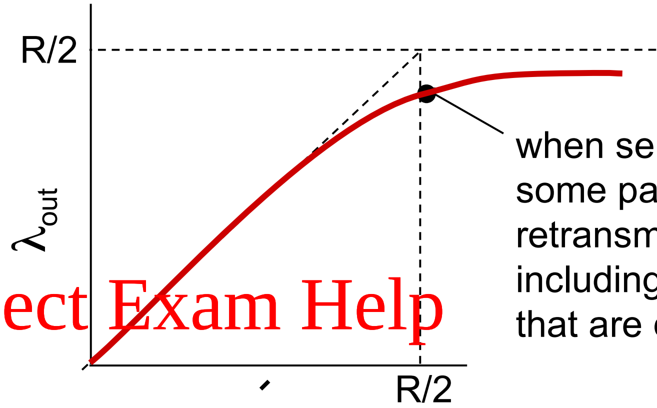


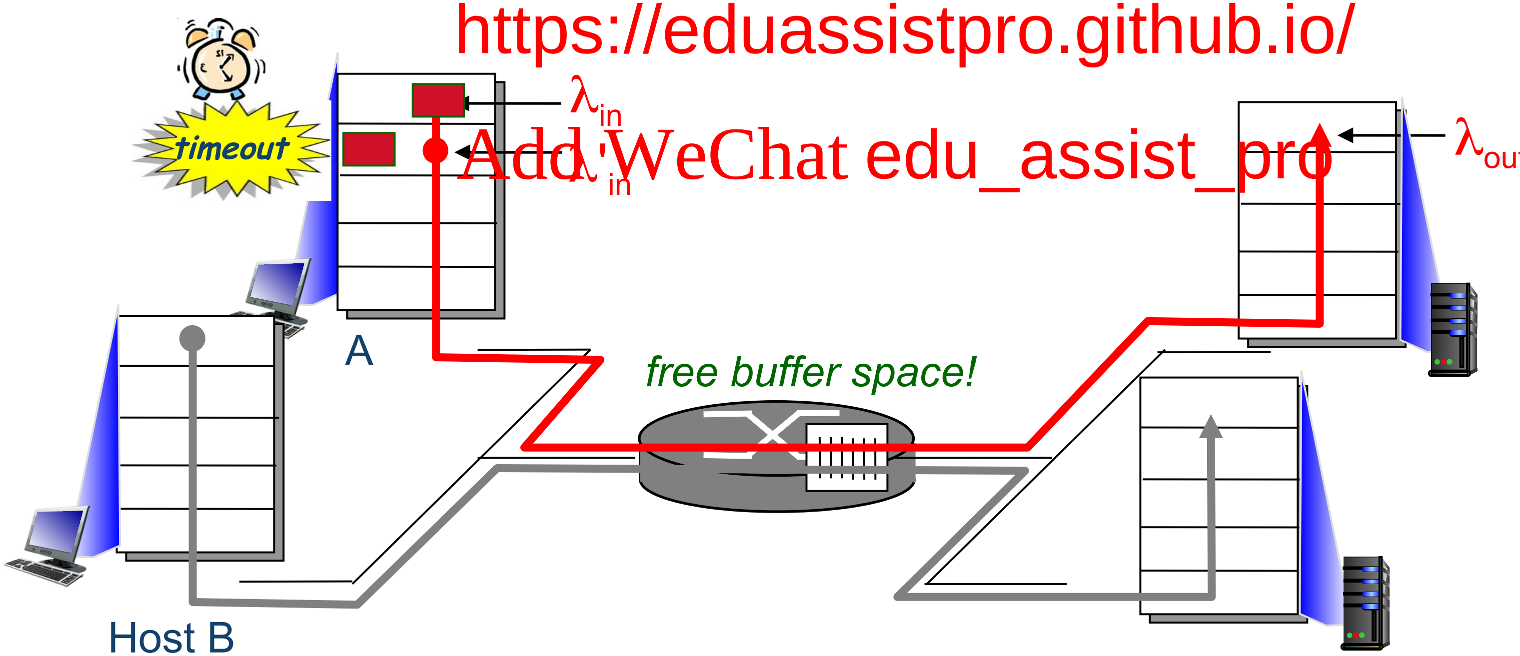


Causes/costs of congestion: scenario 2

Realistic: duplicates

- packets can be lost, dropped at router due to full buffers
- sender times out prematurely, sending *two* copies, both of which are delive

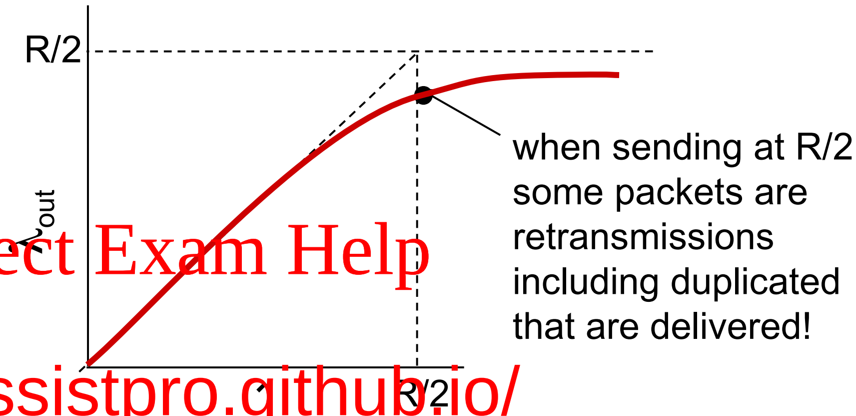




Causes/costs of congestion: scenario 2

Realistic: duplicates

- packets can be lost, dropped at router due to full buffers
- sender times out prematurely, sending *two* copi
which are delive



"costs" of congestion:

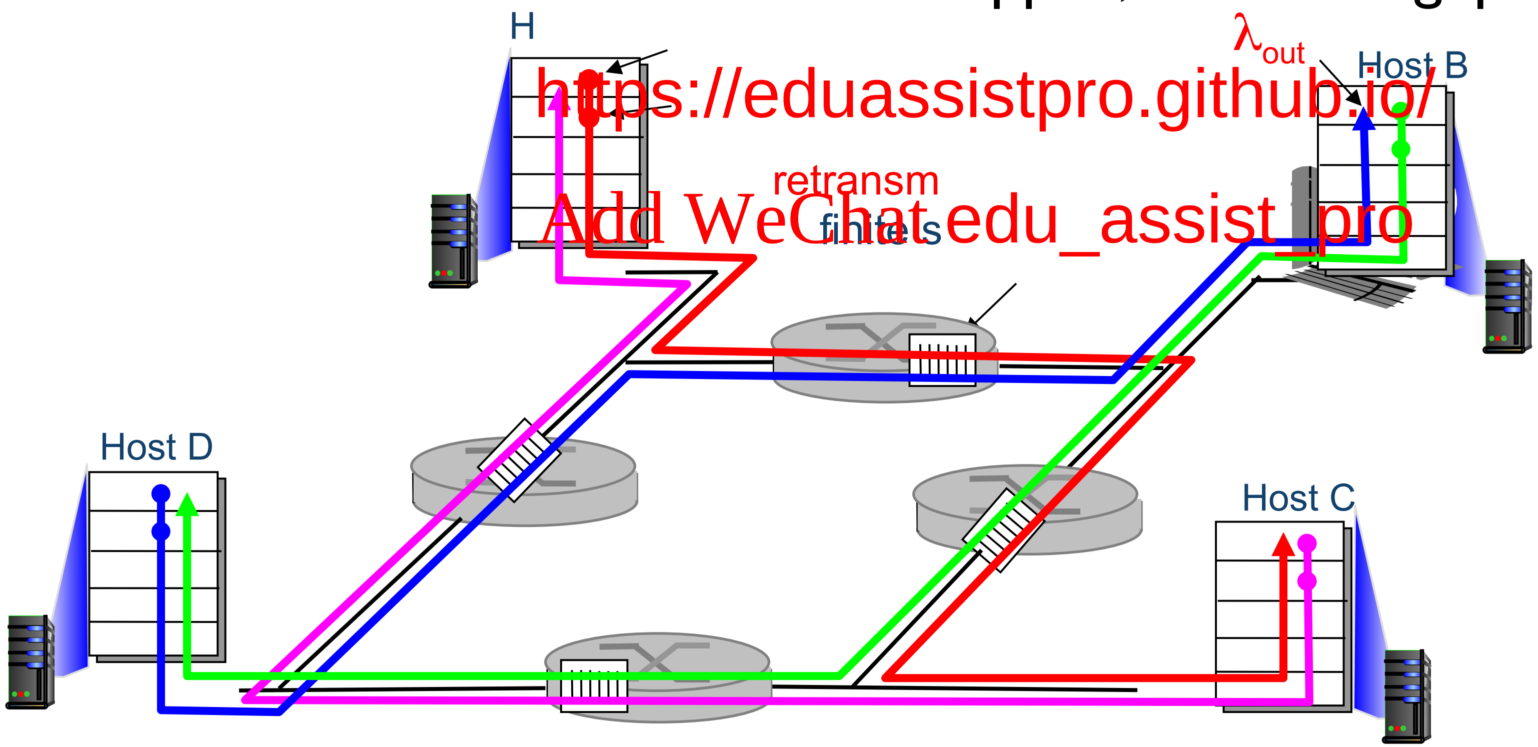
- more work (retrans) for given "goodput"
- unneeded retransmissions: link carries multiple copies of pkt
  - decreasing goodput

Causes/costs of congestion: scenario 3

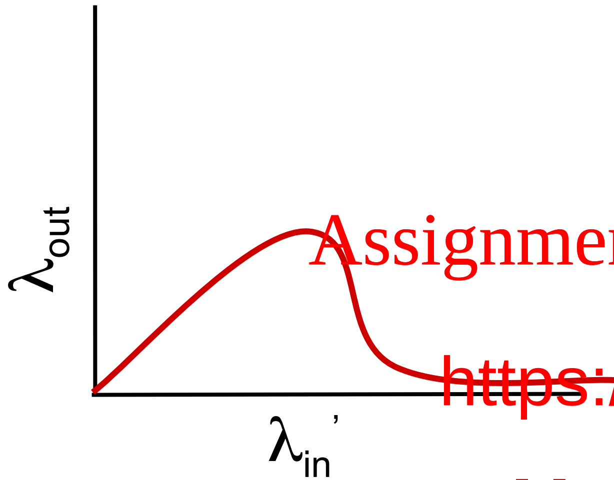
- four senders
- multihop paths
- timeout/retransmit

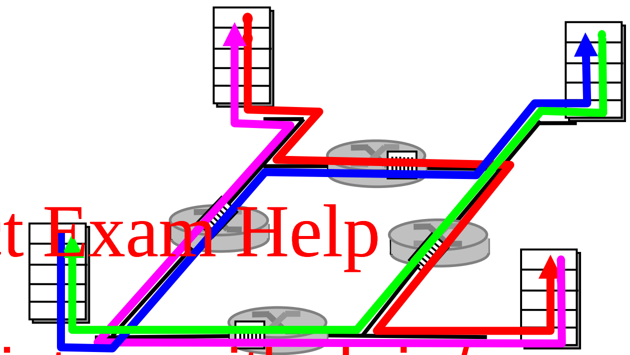
Q: what happens as λ_{in}' increases ?

A: as red λ_{in}' increases, all arriving blue pkts at upper queue are dropped, blue throughput → 0



Causes/costs of congestion: scenario 3





another "cost" of congestion:

- when packet dropped, any "upstream transmission capacity used for that packet was wasted!

Approaches towards congestion control

two broad approaches towards congestion control:

end-end co control:

- no explicit feed from network
- congestion inferred from end-system observed loss, delay
- approach taken by TCP

network-assisted tion control:

- provid tems
  - single bit indicating congestion
  - explicit rate for sender to send at

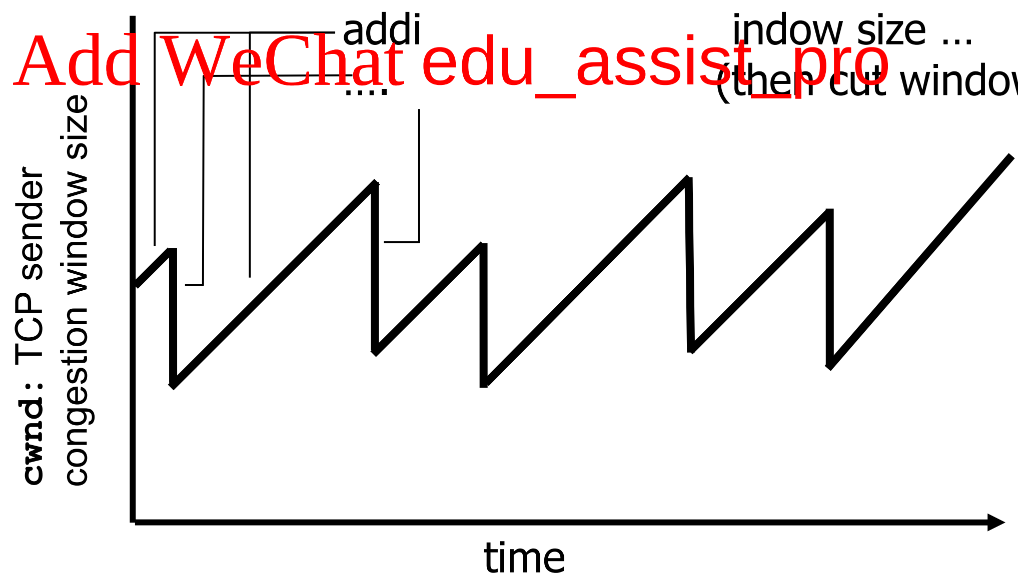
TCP Congestion Control

TCP congestion control

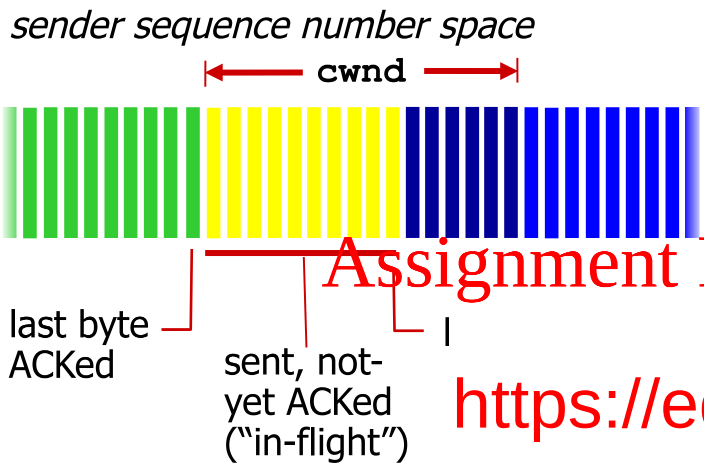
Additive increase multiplicative decrease (AIMD)

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

AIMD saw tooth behavior: probing for bandwidth



TCP Congestion Control: details



TCP sending rate:

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

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

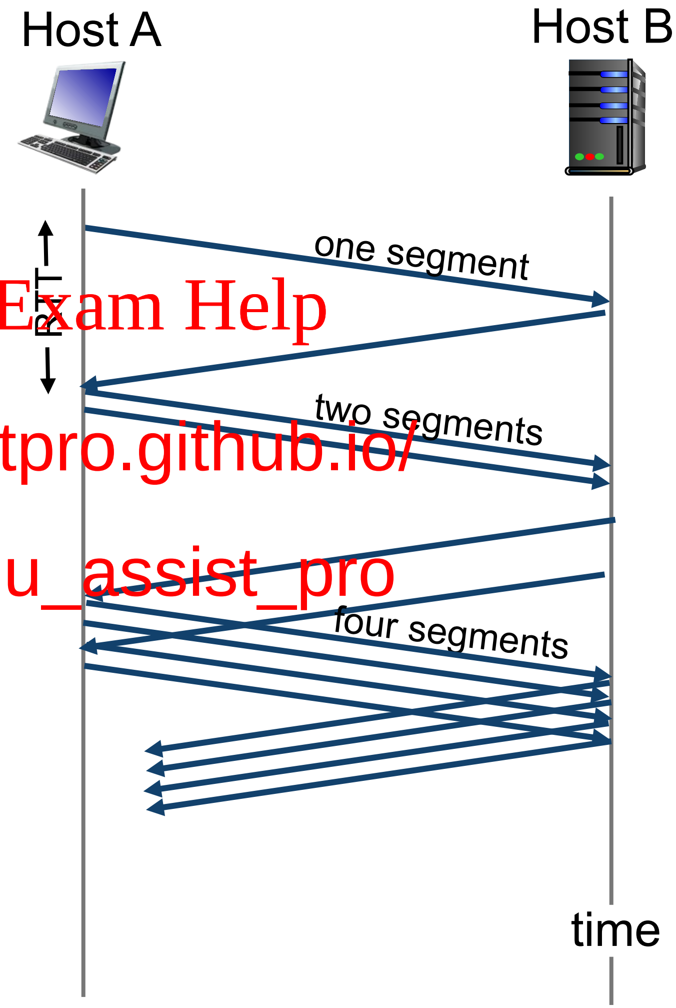
- sender limits transmission:

`LastByteSent- LastByteAcked` $\leq$ `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?

cwnd=12 ssthresh=6

loss!

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

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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

  - TCP Tahoe, same as loss indicated by ti 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

additive increase

additive increase

tiplicative decease

additive increase

slow start

slow start

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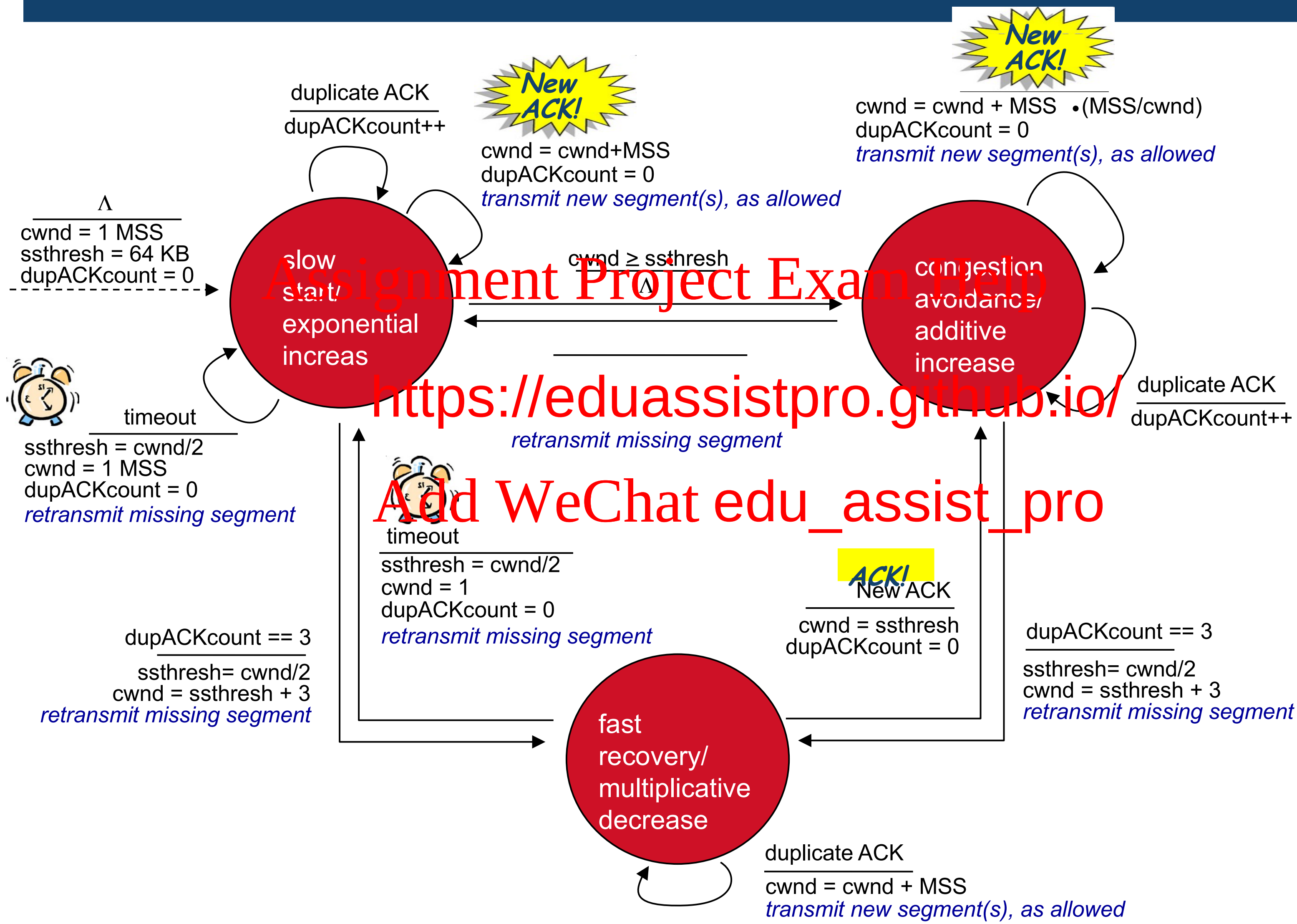
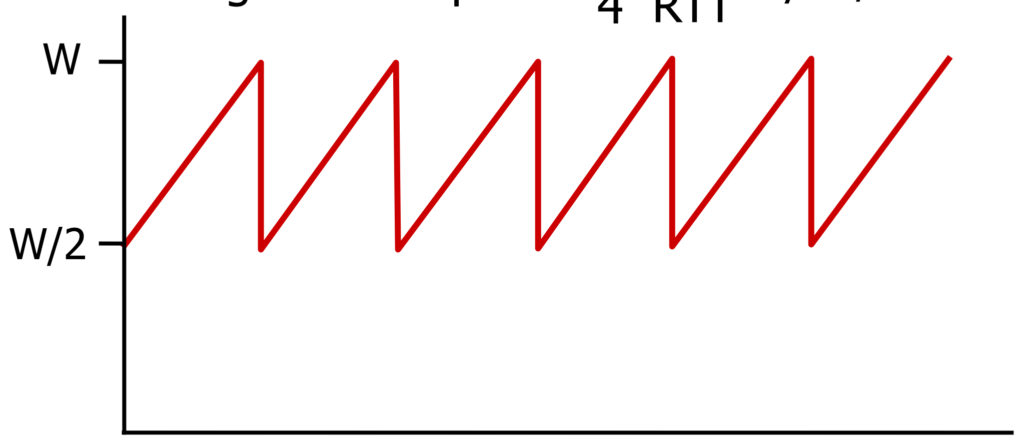
Summary: TCP Reno Congestion Control



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

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



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TCP Futures: TCP over “long, fat pipes”

- example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput
- requires W = 83,333 in-flight segments

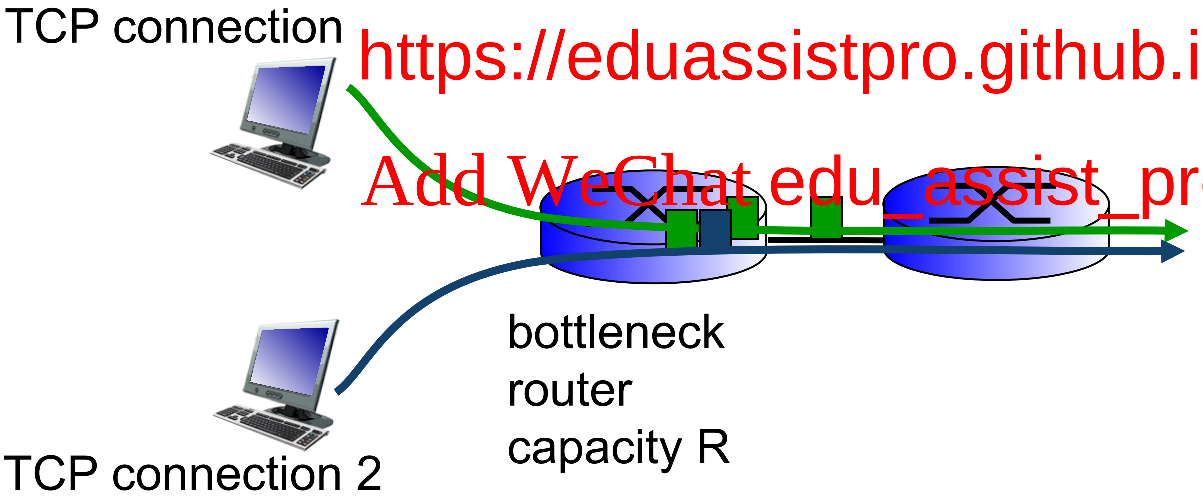
- throughput in ter ... ility, L [Mathis 1997]:

$$\text{TCP throughput} = \frac{1\ldots}{\ldots\sqrt{\ldots}}$$

➜ to achieve 10 Gbps throughput, need a loss rate of L = $2 \cdot 10^{-10}$ *– a very small loss rate!*

- new versions of TCP for high-speed
  - Vegas, Westwood, CUBIC, etc.

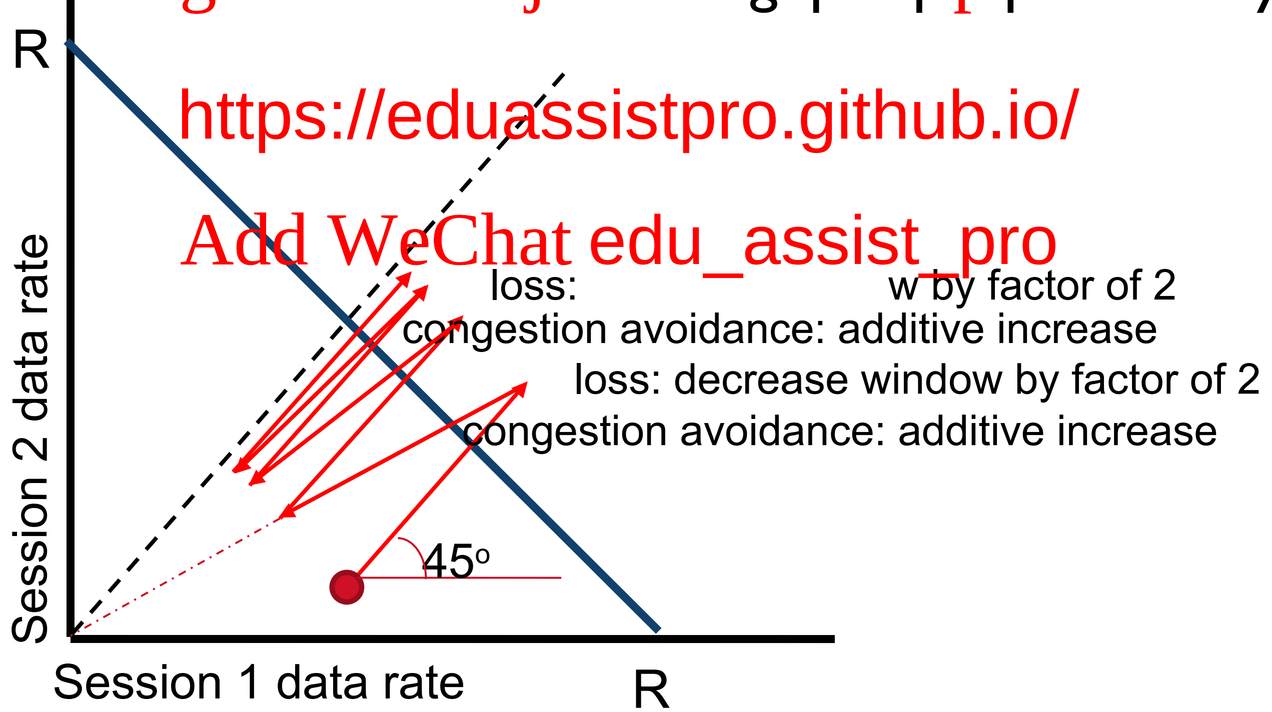
Fairness: K TCP sessions share same bottleneck link of bandwidth R, each has average rate of R/K



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

f rate R

- 1 TCP, gets 0.1R
- 9 TCPs, gets 0.9R

Final word

Window size = min (rwnd, cwnd)

receive window *n window*

flow control *congestion control*

Conclusion

- principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - connection setu
  - flow control
  - congestion control
- instantiation, implementation in th
  - UDP
  - TCP