COMP 3331/9331: Computer Networks and Applications

Week 5

Transport Layer (Continued)

Reading Guide: Chapter 3, Sections: 3.5 – 3.8

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

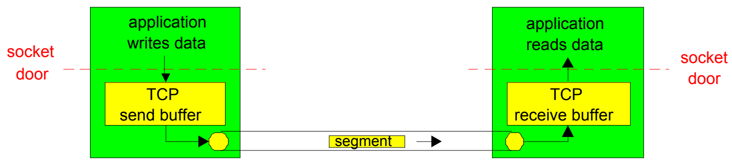
3.5 connection-oriented transport: TCP
- segment structure
- reliable data transfer
- flow control
- connection management

3.6 principles of congestion control

3.7 TCP congestion control

TCP: overview RFCs: 793,1122, 2018, 5681, 7323

- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte stream:*
  - no “message boundaries"
- full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size



- cumulative ACKs
- pipelining:
  - TCP congestion and flow control set window size
- connection-oriented:
  - handshaking (exchange of control messages) initializes sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

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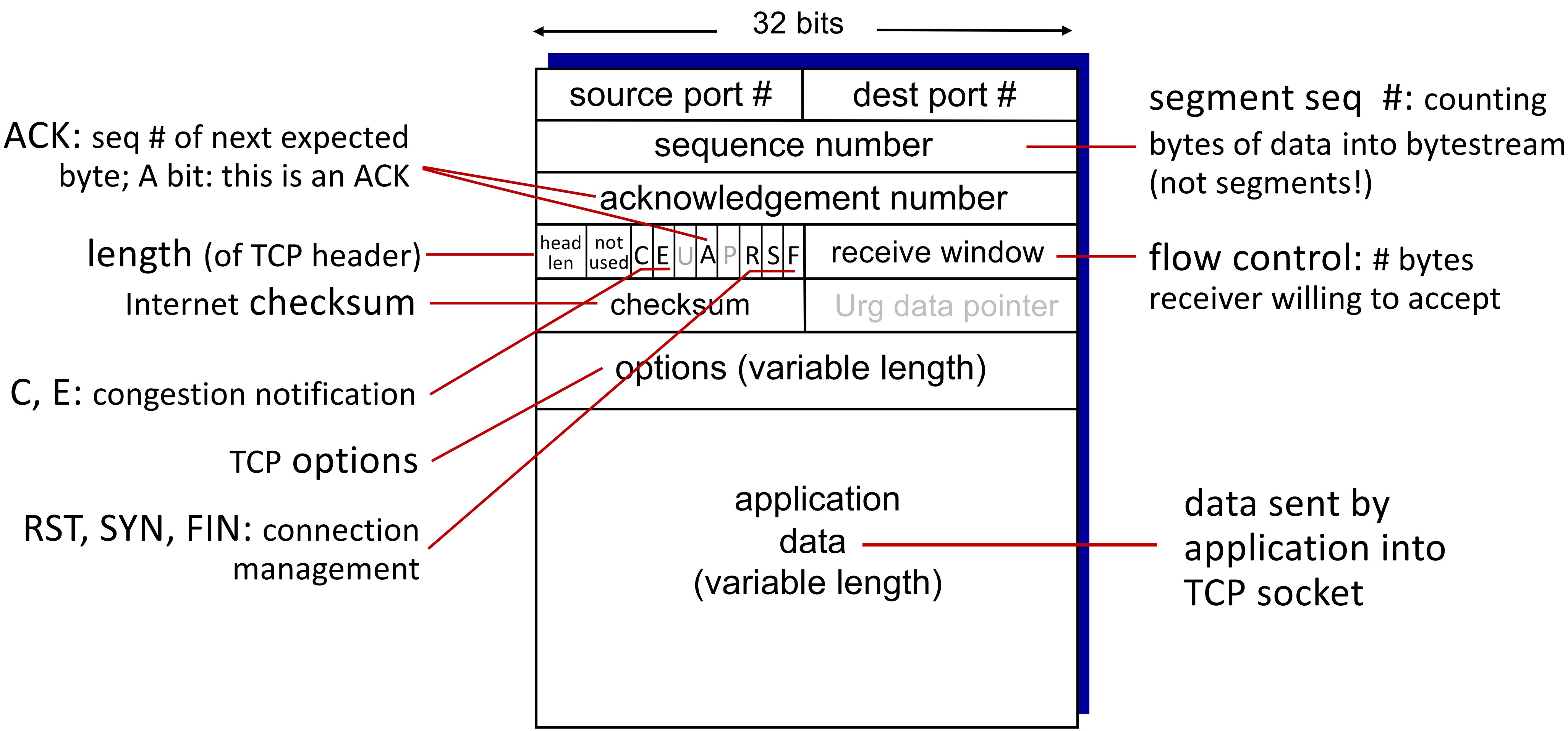
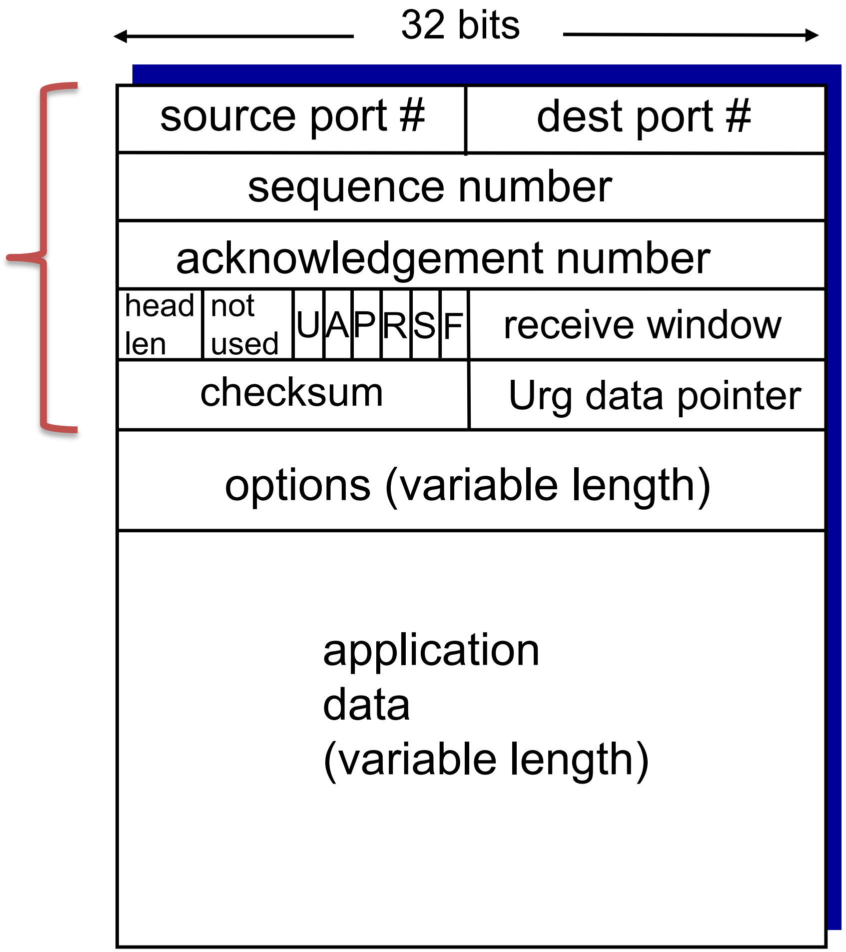
TCP segment structure



TCP segment structure

20 Bytes

(UDP was 8)



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Recall: Components of a solution for reliable transport

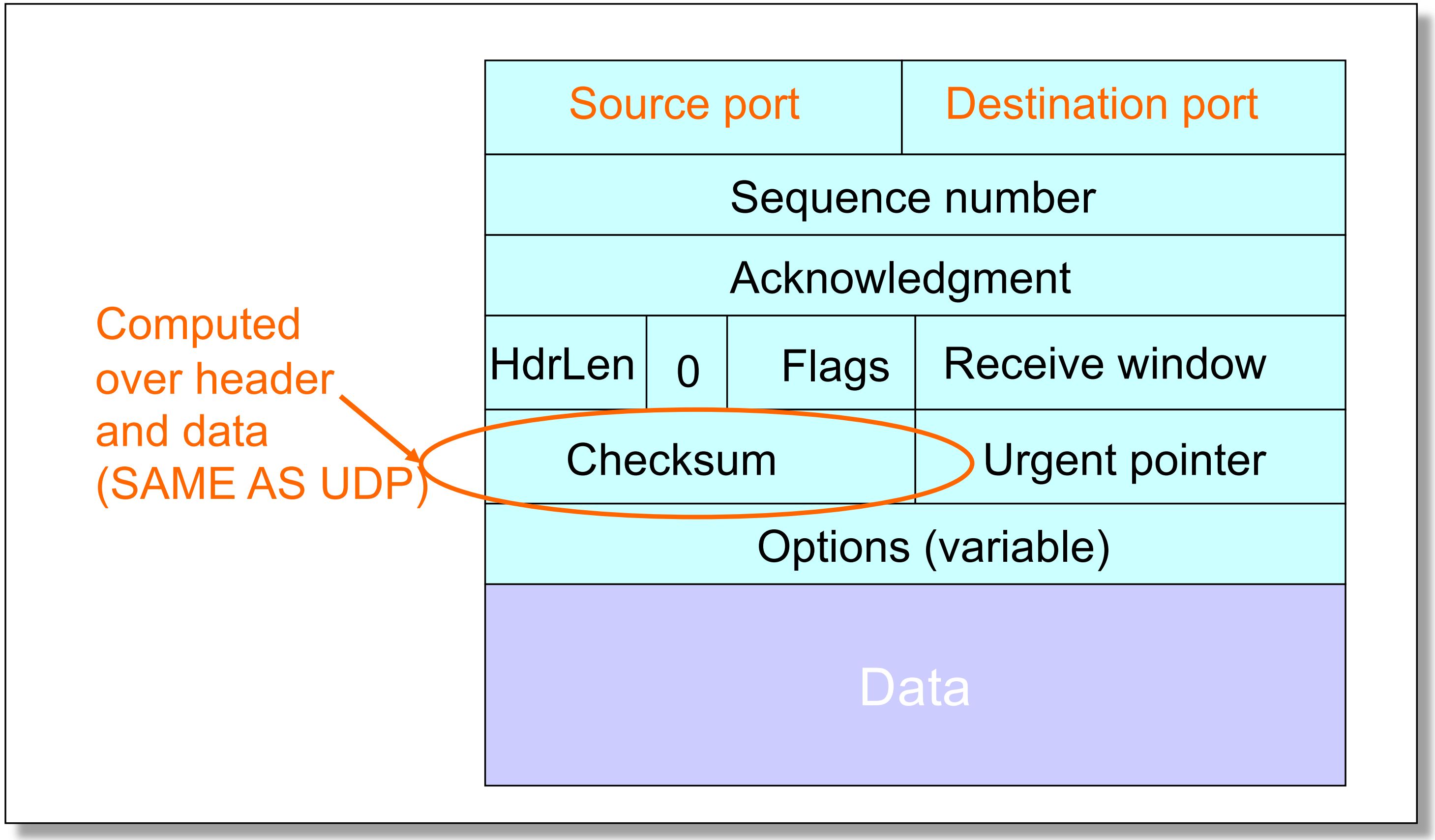
- Checksums (for error detection)
- Timers (for loss detection)
- Acknowledgments
  - Cumulative
  - Selective
- Sequence numbers (duplicates, windows)
- Sliding Windows (for efficiency)
  - Go-Back-N (GBN)
  - Selective Repeat (SR)

What does TCP do?

Many of our previous ideas, but some key differences

- Checksum

TCP Header


Source port
Destination port
Sequence number
Acknowledgment
HdrLen
0
Flags
Receive window
Computed over header and data (SAME AS UDP)
Checksum
Urgent pointer
Options (variable)
Data

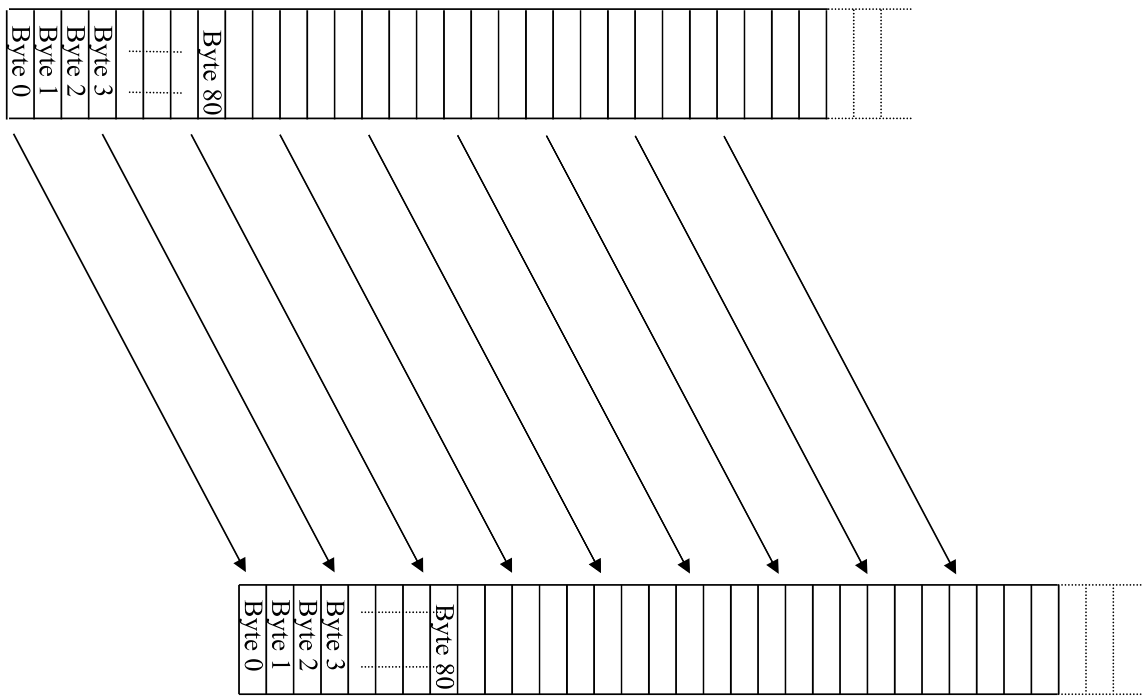
What does TCP do?

Many of our previous ideas, but some key differences

- Checksum
- **Sequence numbers are byte offsets**

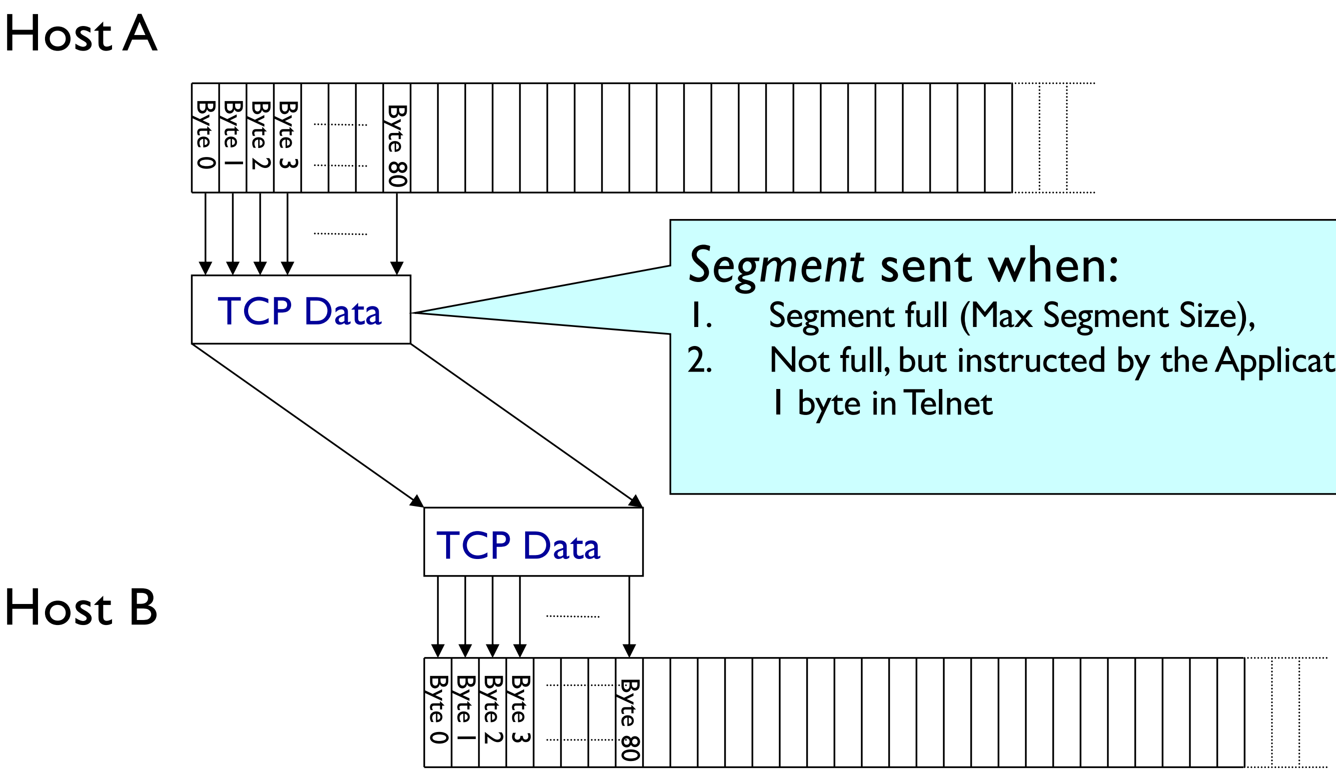
TCP “Stream of Bytes” Service ..

Application @ Host A



Application @ Host B

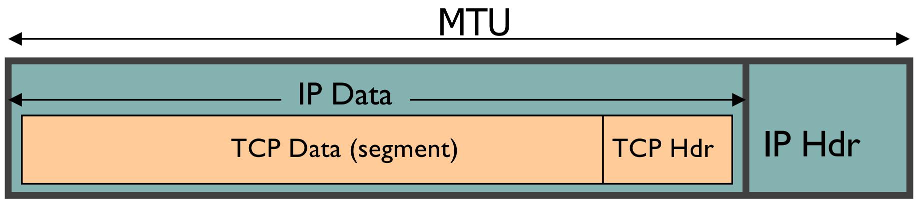
.. Provided Using TCP "Segments"



Segment sent when:

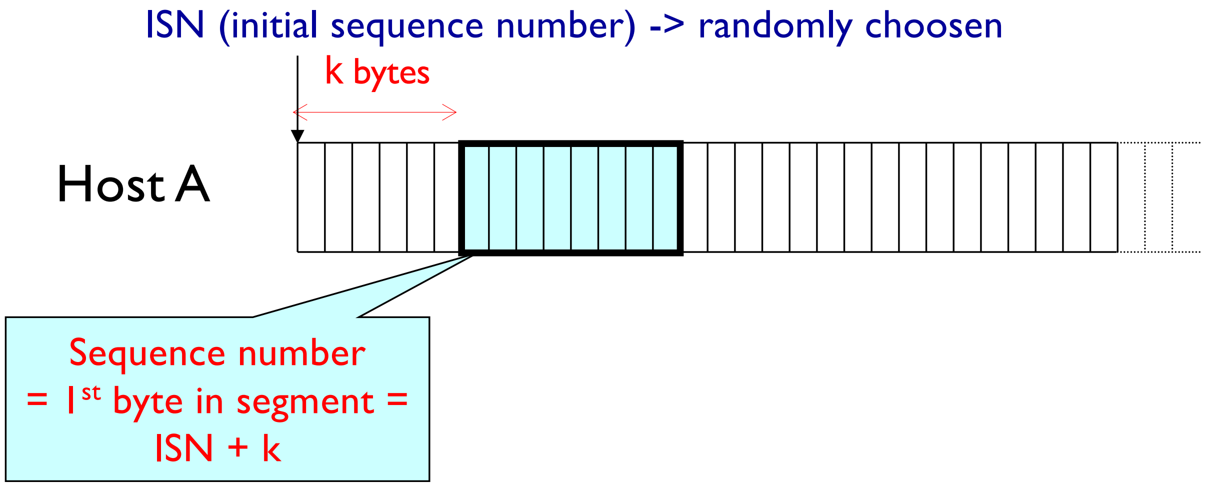
1. Segment full (Max Segment Size),
2. Not full, but instructed by the Application e.g., 1 byte in Telnet

TCP Maximum Segment Size



- IP packet
  - No bigger than Maximum Transmission Unit (MTU) of link layer
  - E.g., up to 1500 bytes with Ethernet
- TCP packet
  - IP packet with a TCP header and data inside
  - TCP header $\geq$ 20 bytes long
- TCP **segment**
  - No more than Maximum Segment Size (MSS) bytes
  - E.g., up to 1460 consecutive bytes from the stream
  - MSS = MTU – 20 (minimum IP header) – 20 (minimum TCP header)

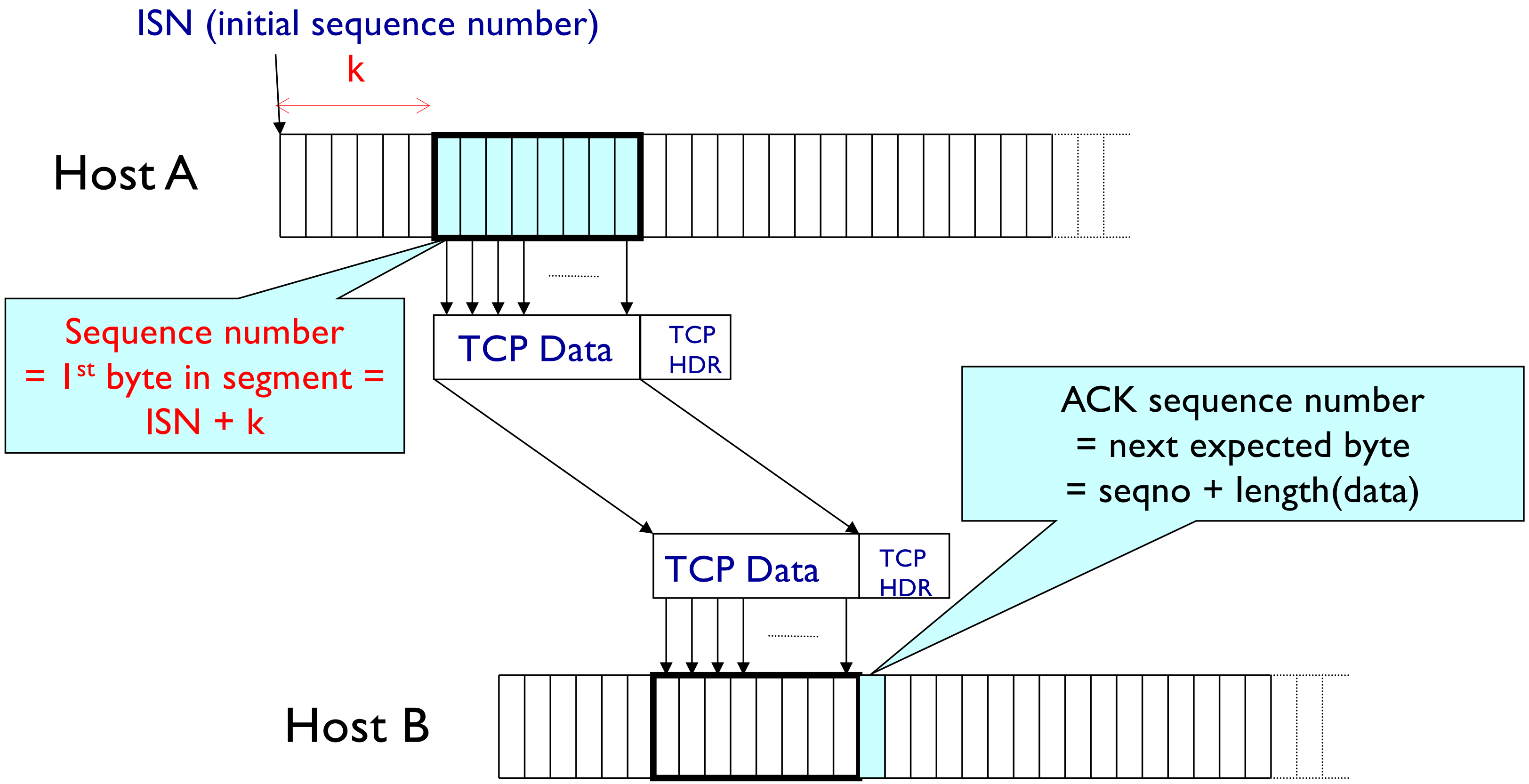
Sequence Numbers



Sequence numbers:

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

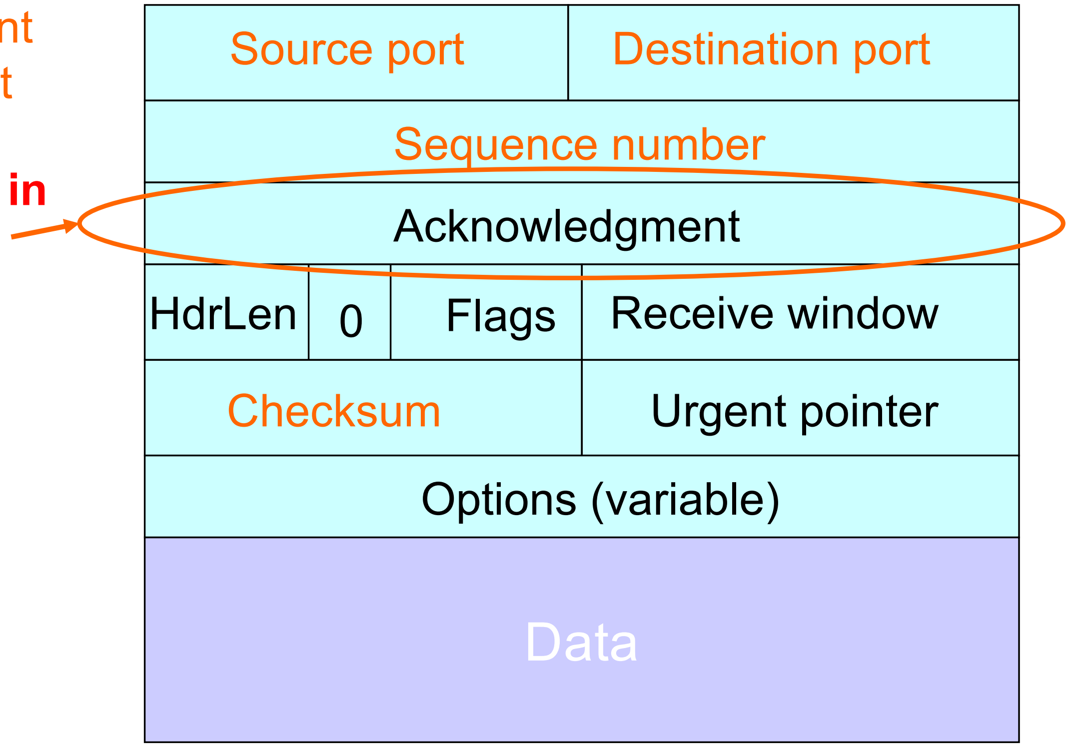
Sequence & Ack Numbers



TCP Header

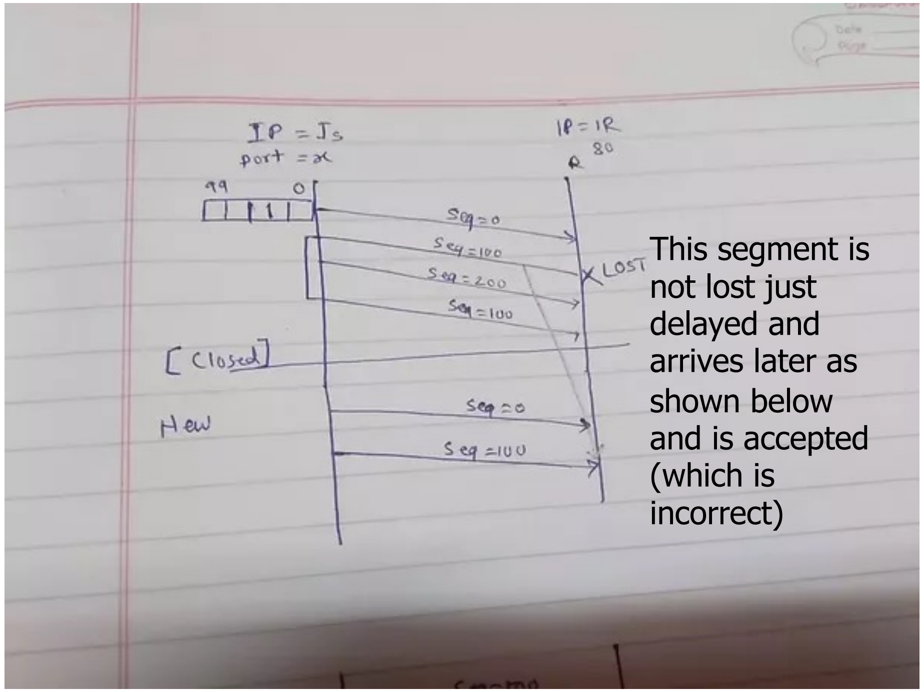
Acknowledgment gives seqno just beyond highest seqno received **in order**

("What Byte is Next")

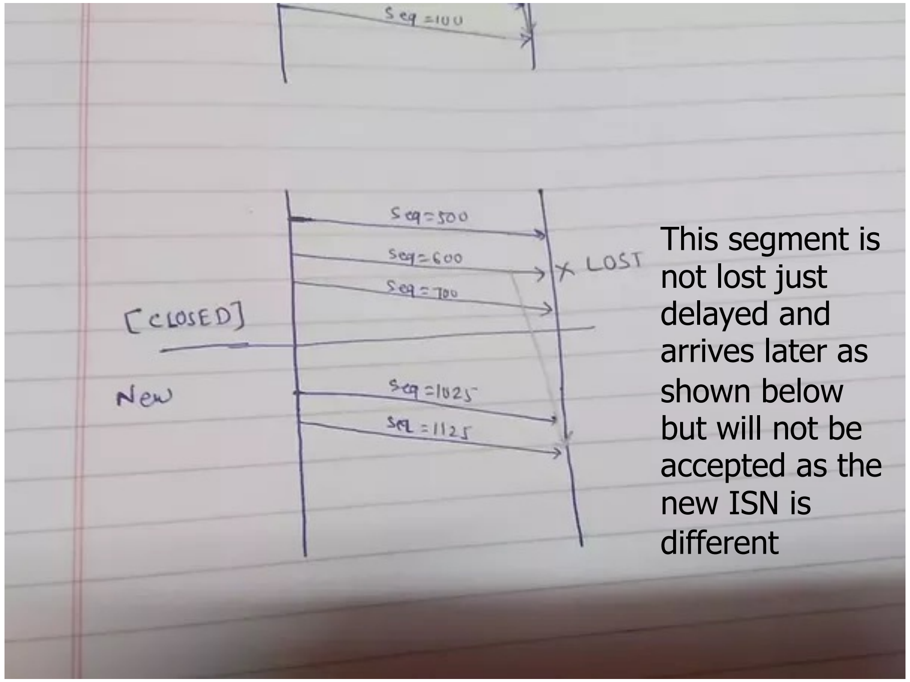


Why choose random ISN?

- Avoids ambiguity with back-to-back connections between same end-points



(a) When ISN=0



(b) When ISN is random

- Potential security issue if the ISN is known

What does TCP do?

Most of our previous tricks, but a few differences

- Checksum
- Sequence numbers are byte offsets
- Receiver sends cumulative acknowledgements (like GBN)

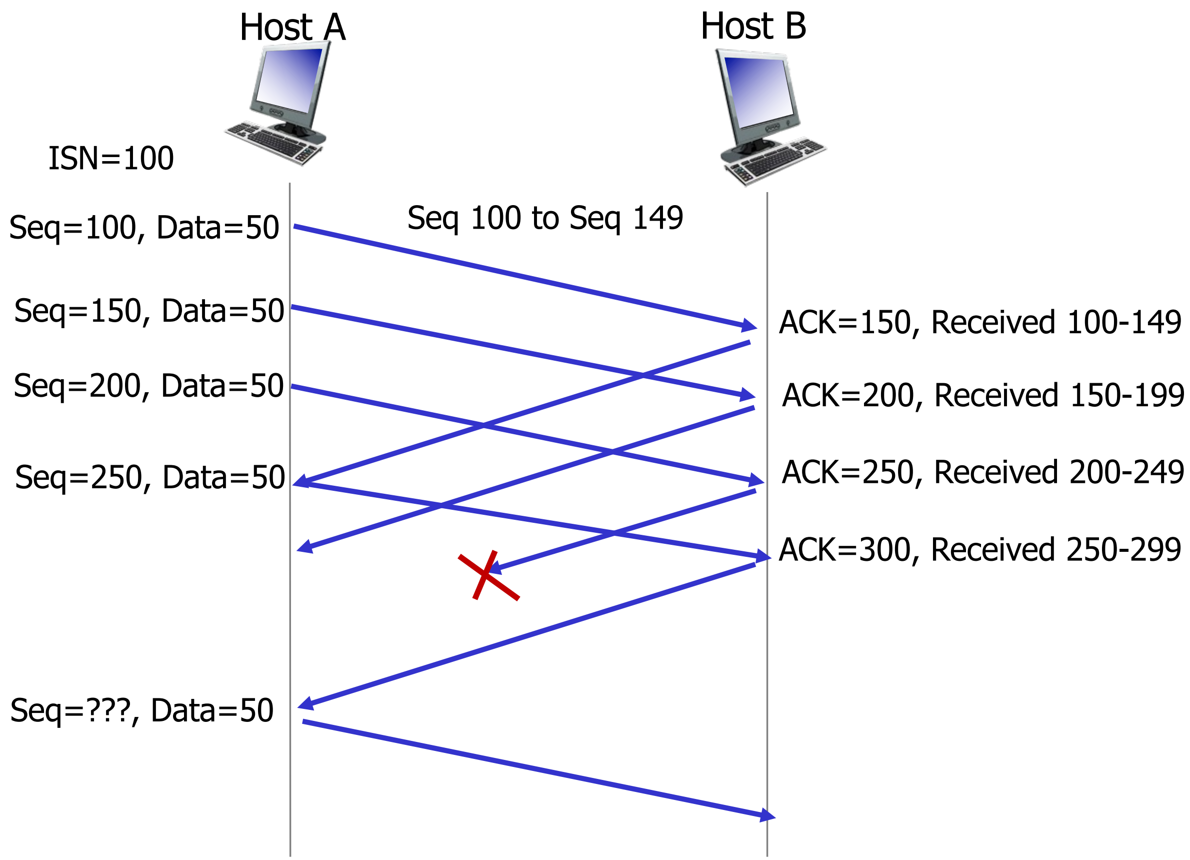
ACKing and Sequence Numbers

- Sender sends packet
  - Data starts with sequence number X
  - Packet contains B bytes [X, X+1, X+2, ….X+B-1]

- Upon receipt of packet, receiver sends an ACK
  - If all data prior to X already received:
    - ACK acknowledges X+B (because that is next expected byte)

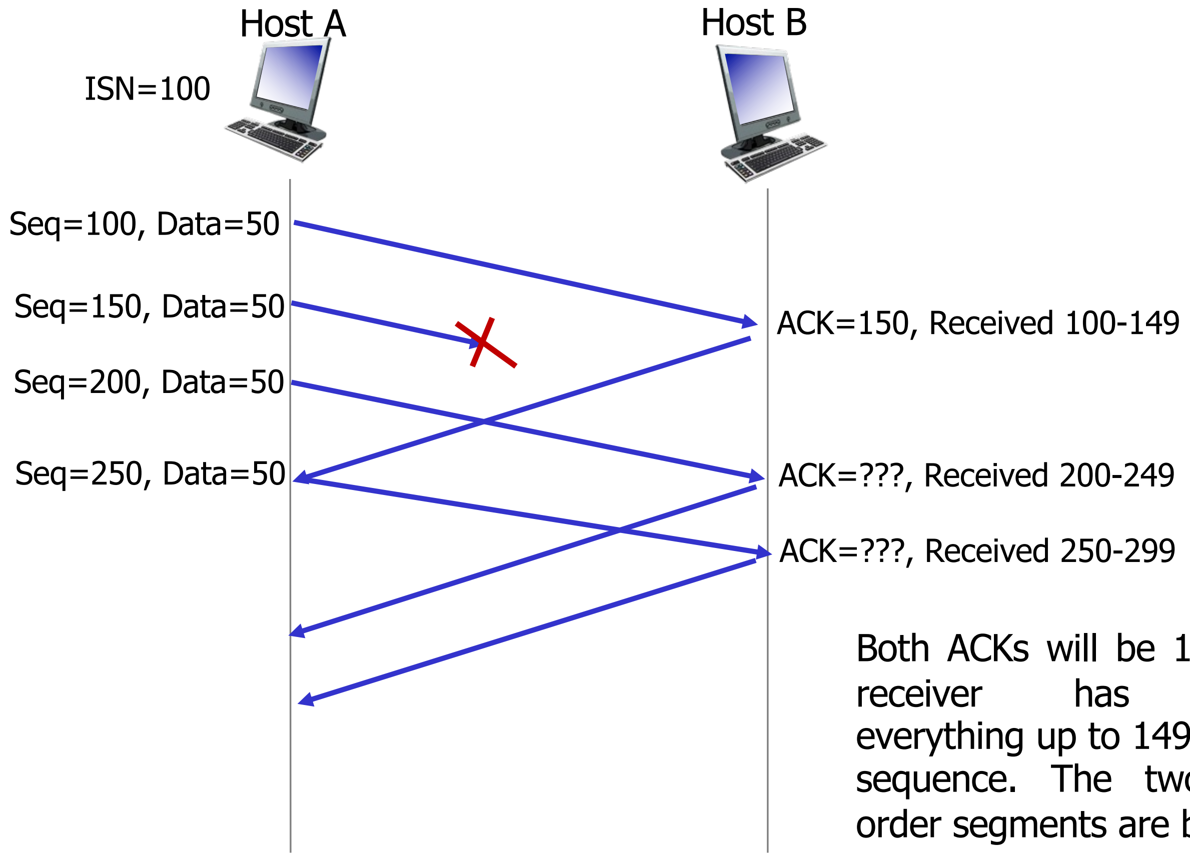
  - If highest in-order byte received is Y such that (Y+1) < X
    - ACK acknowledges Y+1
    - Even if this has been ACKed before

An Example



Seq = 300 (new segment)
Since TCP uses cumulative ACKs, the receipt of ACK 300 before a timeout (for seg with sequence number 200) implies the receiver has received all 4 segments sent above

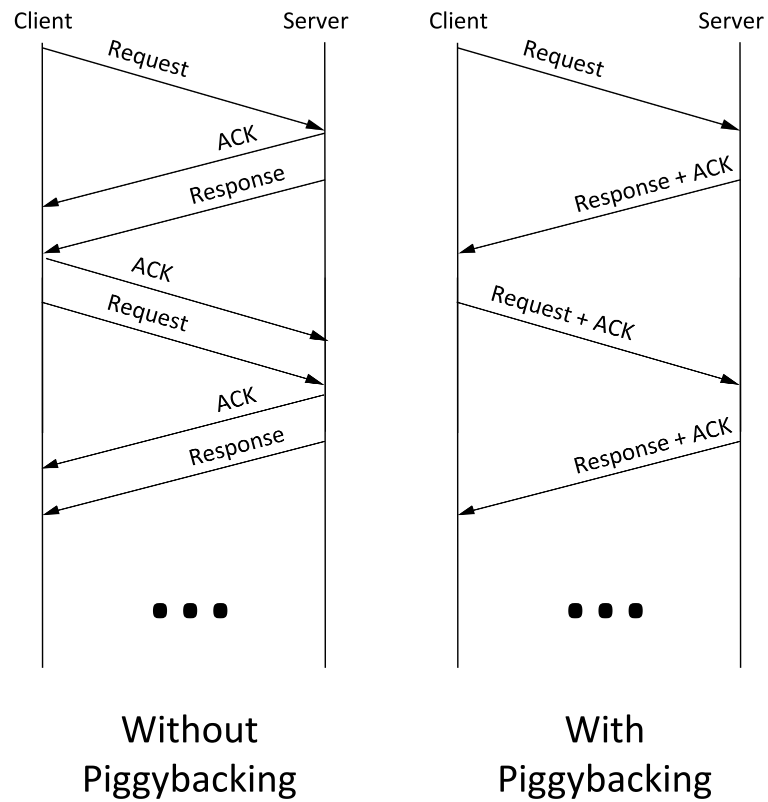
Another Example



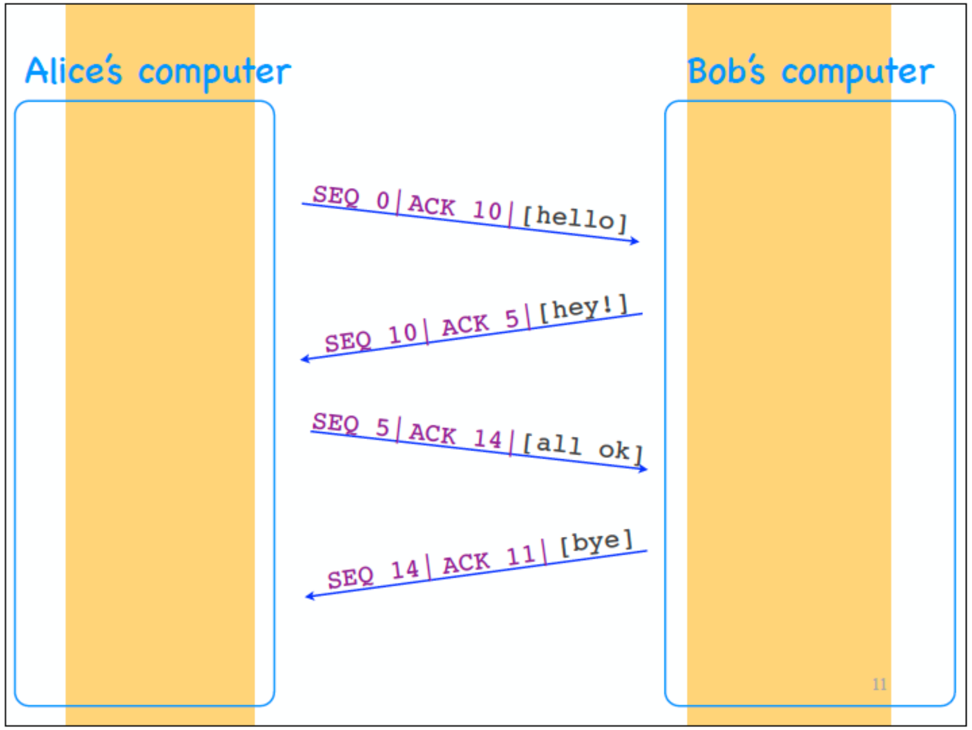
Both ACKs will be 150 as the receiver has received everything up to 149 in correct sequence. The two out of order segments are buffered.

Piggybacking

- So far, we've assumed distinct "sender" and "receiver" roles
- Usually both sides of a connection (i.e. the application processes) send some data

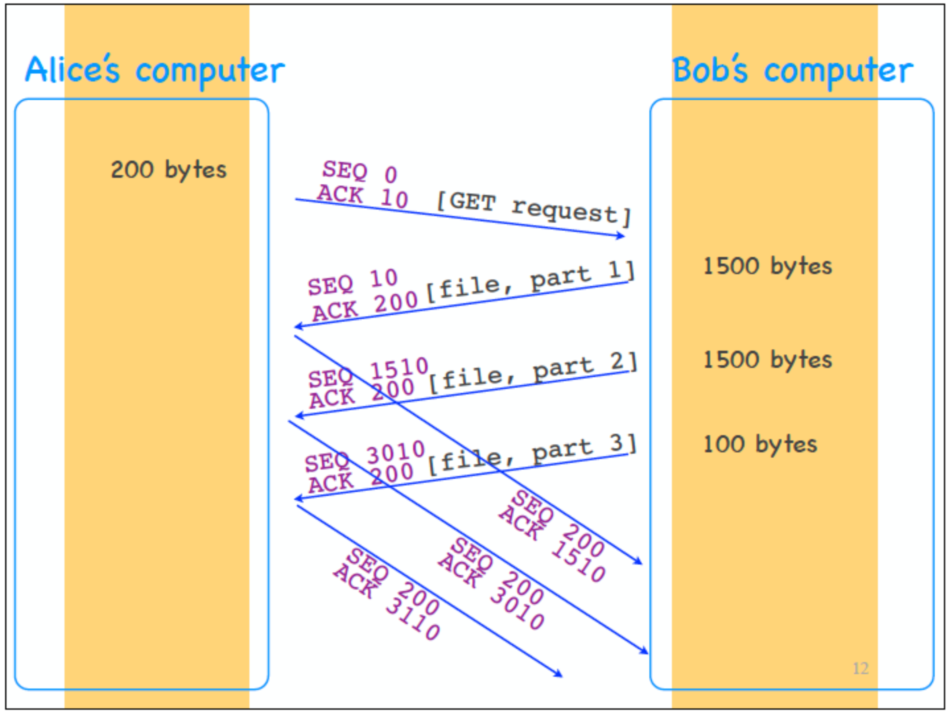


Example



Note: Connection establishment not shown. Alice's end point selects the initial sequence number as 0 while Bob's end point selects the initial sequence number as 10

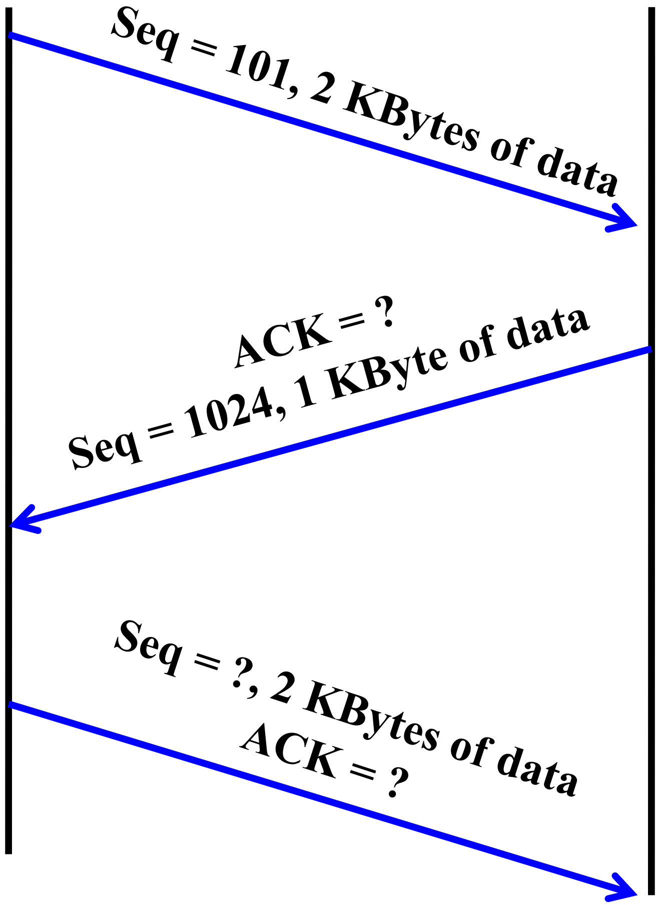
Another Example



Note: Connection establishment not shown. Alice's end point selects the initial sequence number as 0 while Bob's end point selects the initial sequence number as 10

HTTP response split into 3 segments (MSS = 1500 bytes)

Quiz



ACK =101 + 2048 = 2149

Seq = 2149

ACK = 1024 + 1024 = 2048

What does TCP do?

Most of our previous tricks, but a few differences

- Checksum
- Sequence numbers are byte offsets
- Receiver sends cumulative acknowledgements (like GBN)
- Receivers can buffer out-of-sequence packets (like SR)

Loss with cumulative ACKs

- Sender sends packets with 100 bytes and sequence numbers:
  - 100, 200, 300, 400, 500, 600, 700, 800, 900, …

- Assume the fifth packet (seq. no. 500) is lost, but no others

- 6th packet onwards are buffered

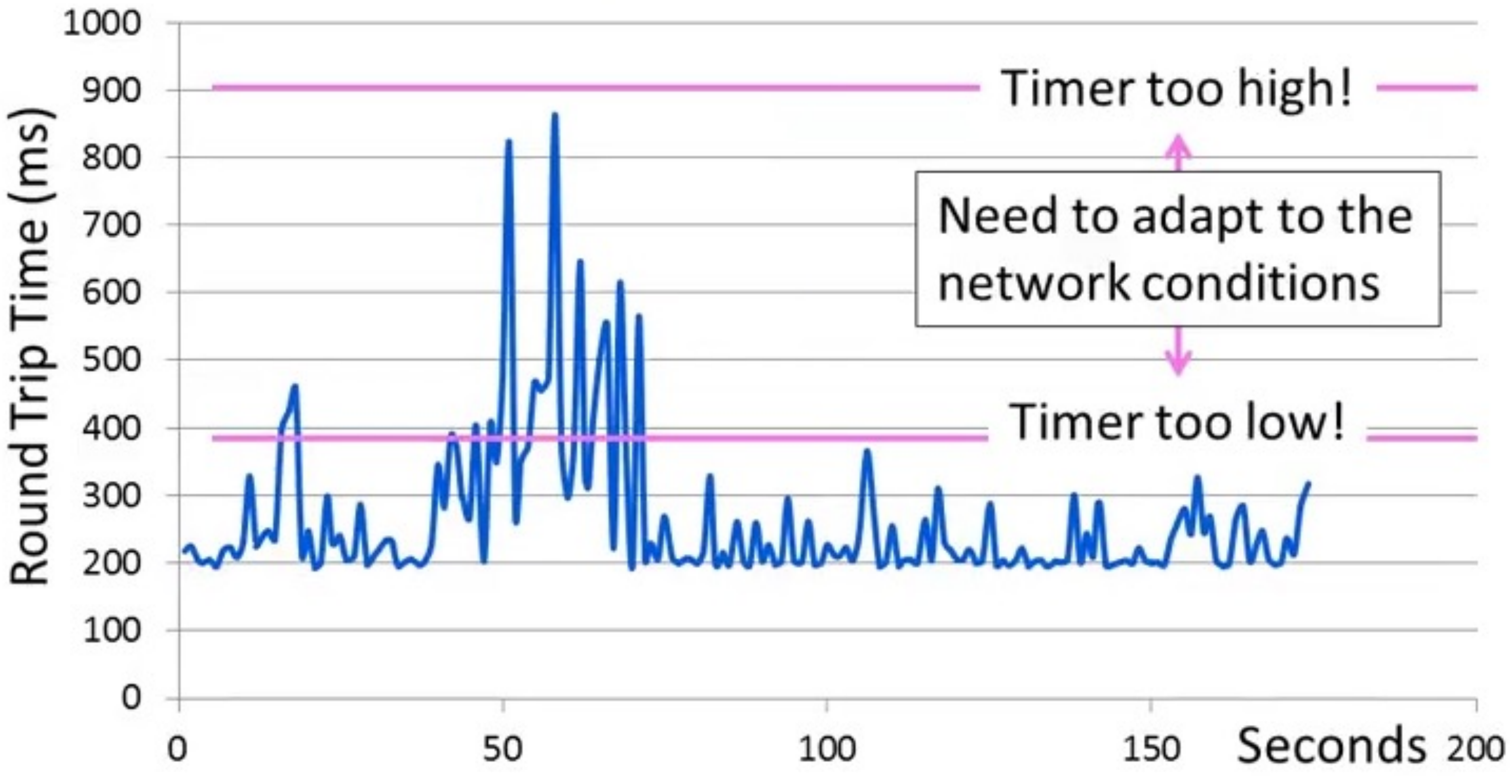
- Stream of ACKs will be:
  - 200, 300, 400, 500, 500, 500, 500,…

What does TCP do?

Most of our previous tricks, but a few differences

- Checksum
- Sequence numbers are byte offsets
- Receiver sends cumulative acknowledgements (like GBN)
- Receivers do not drop out-of-sequence packets (like SR)
- Sender maintains a single retransmission timer (like GBN) and retransmits on timeout (*how much?*)

TCP round trip time, timeout



TCP round trip time, timeout

Q: how to set TCP timeout value?

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

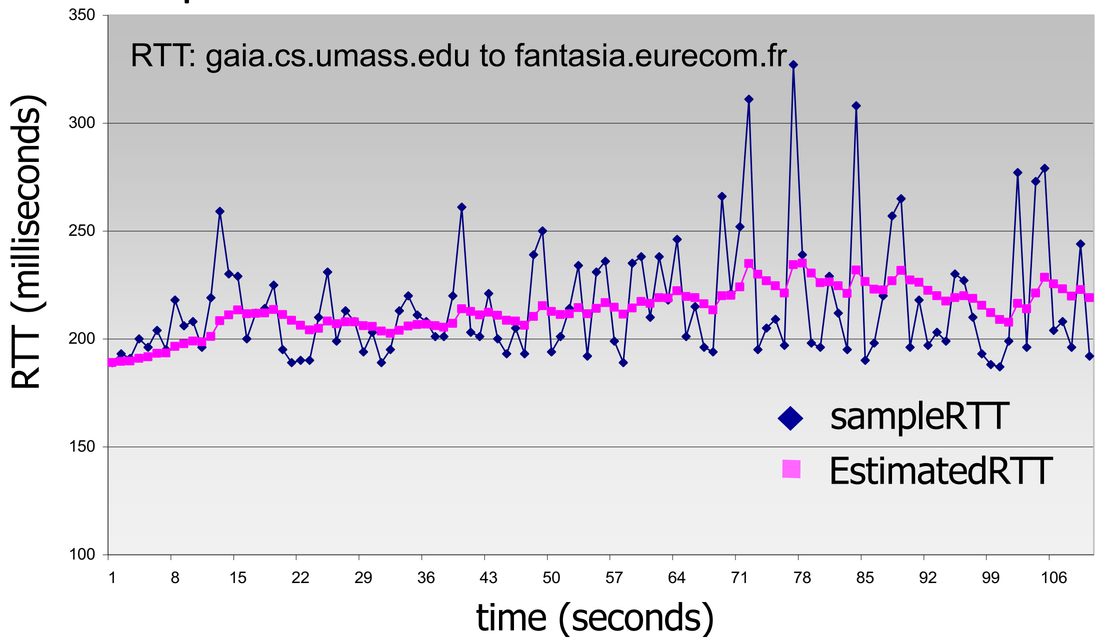
Q: how to estimate RTT?

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

TCP round trip time, timeout

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

- exponential weighted moving average (EWMA)
- influence of past sample decreases exponentially fast
- typical value: α = 0.125
- For the first sample, EstimatedRTT = SampleRTT



TCP round trip time, timeout

- timeout interval: **`EstimatedRTT`** plus "safety margin"
  - large variation in **`EstimatedRTT`:** want a larger safety margin

```
TimeoutInterval = EstimatedRTT + 4*DevRTT
```

estimated RTT "safety margin"

- **`DevRTT`**: EWMA of **`SampleRTT`** deviation from **`EstimatedRTT`**:

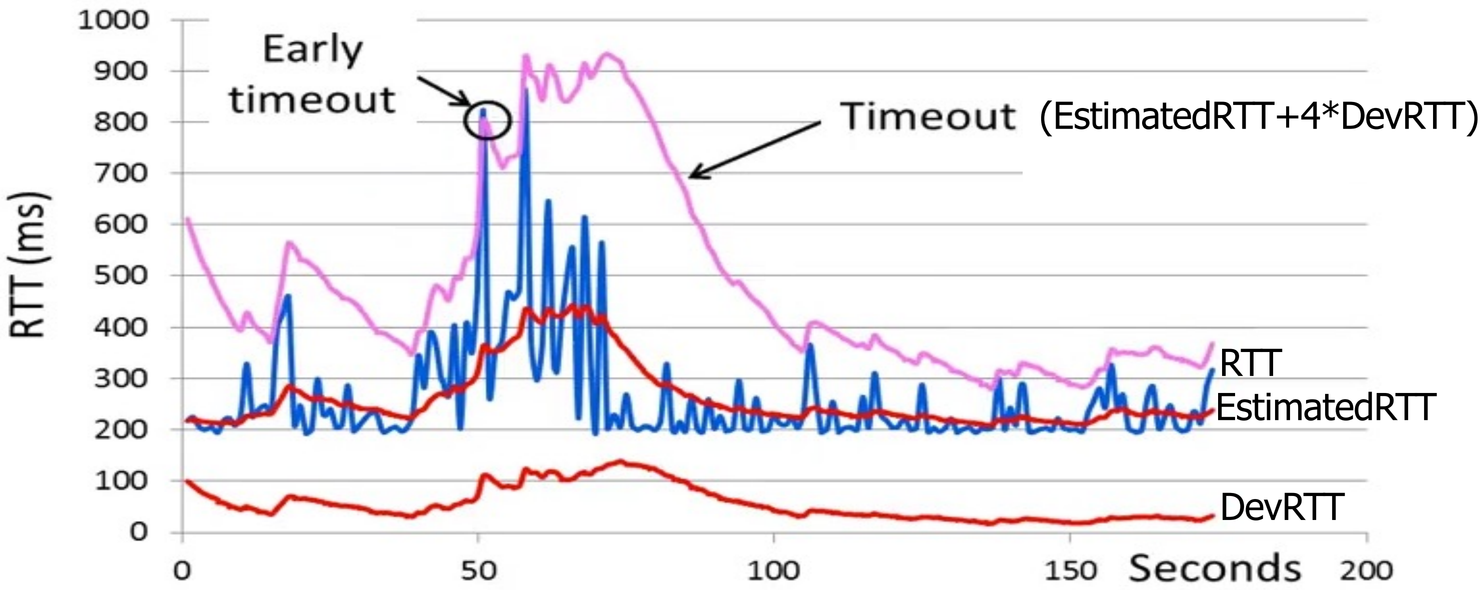
```
DevRTT = (1-β)*DevRTT + β*|SampleRTT-EstimatedRTT|
```

(typically, β = 0.25)

For the first sample, DevRTT = SampleRTT/2

Practice Problem:
http://wps.pearsoned.com/ecs_kurose_compnetw_6/216/55463/14198700.cw/index.html

TCP round trip time, timeout




```
TimeoutInterval = EstimatedRTT + 4*DevRTT
```

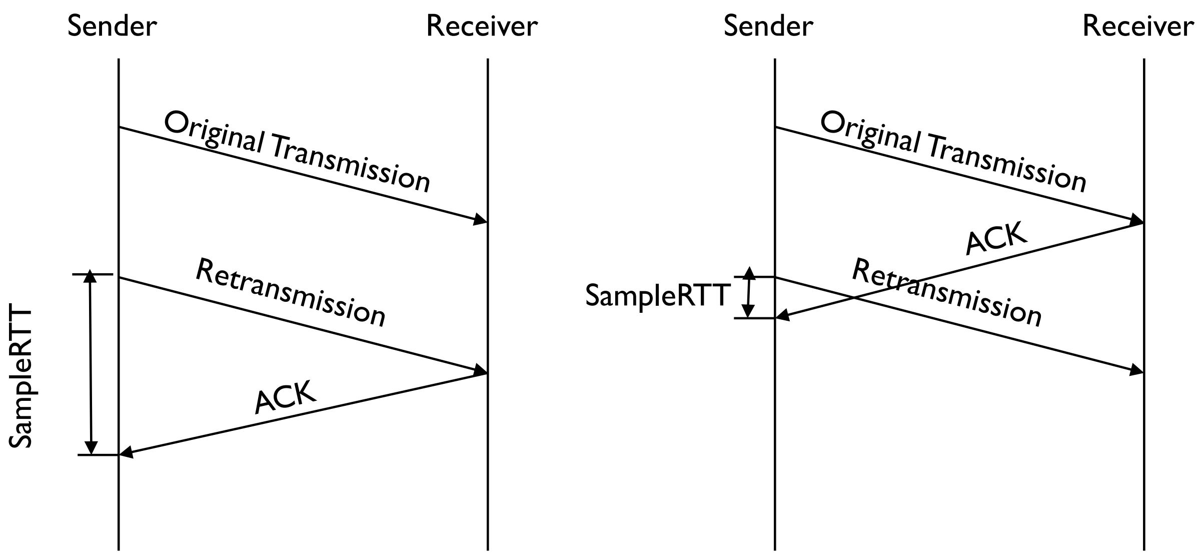
estimated RTT

"safety margin"

Figure: Credits Prof David Wetherall UoW

Why exclude retransmissions in RTT computation?

- How do we differentiate between the real ACK, and ACK of the retransmitted packet?
- Sender cannot differentiate between the two scenarios shown below



TCP Sender (simplified)

PUTTING IT TOGETHER

event: 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`**

event: timeout

- retransmit segment that caused timeout
- restart timer (double of previous value -> limited congestion control)

event: ACK received

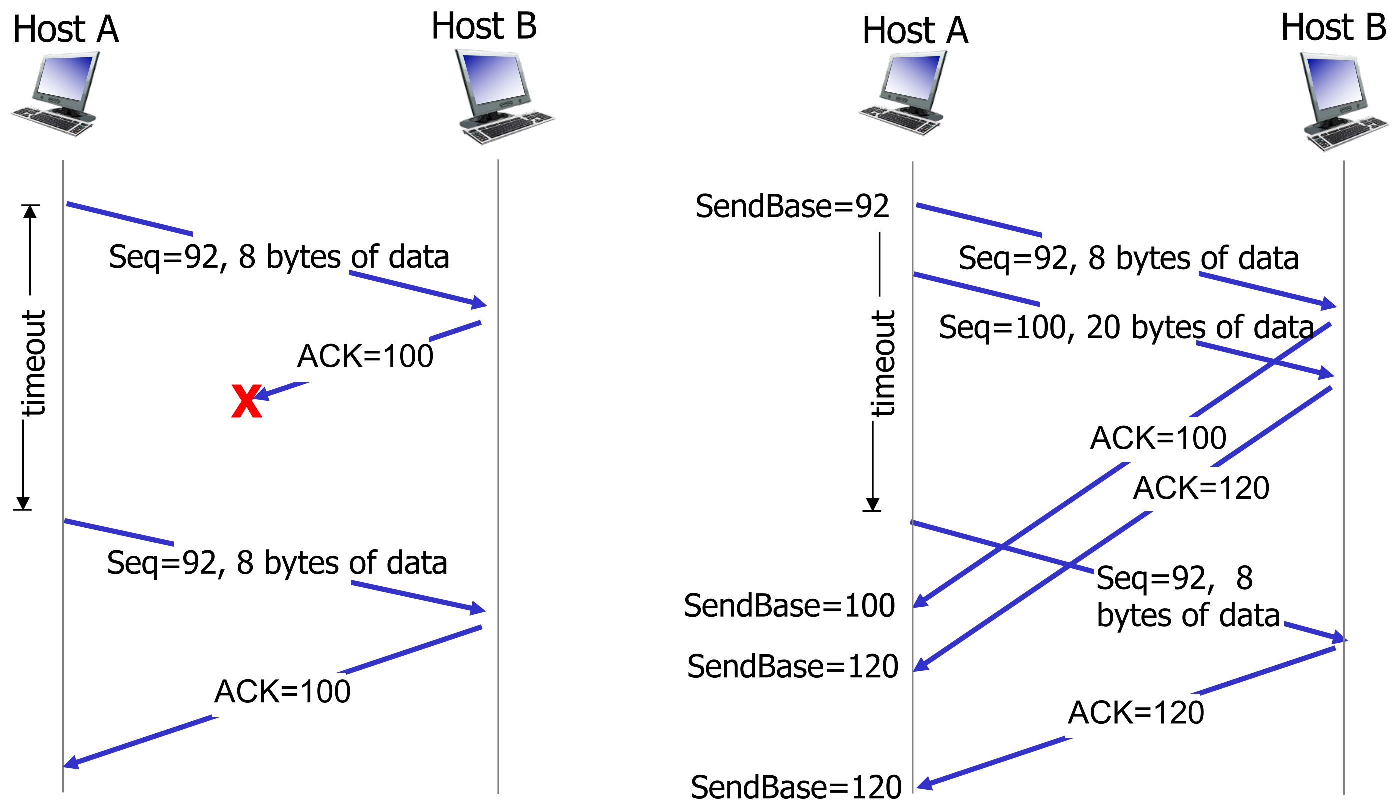
- if ACK acknowledges previously unACKed segments
  - update what is known to be ACKed
  - start timer if there are still unACKed segments

TCP ACK generation [RFC 1122, RFC 2581]

Note: You may neglect delayed ACKs in the exam unless explicitly told to consider it

event at receiver	*TCP receiver action*
arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed	delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK
arrival of in-order segment with expected seq #. One other segment has ACK pending	immediately send single cumulative ACK, ACKing both in-order segments
arrival of out-of-order segment higher-than-expect seq. # . Gap detected	immediately send *duplicate ACK,* indicating seq. # of next expected byte
arrival of segment that partially or completely fills gap	immediate send ACK, provided that segment starts at lower end of gap

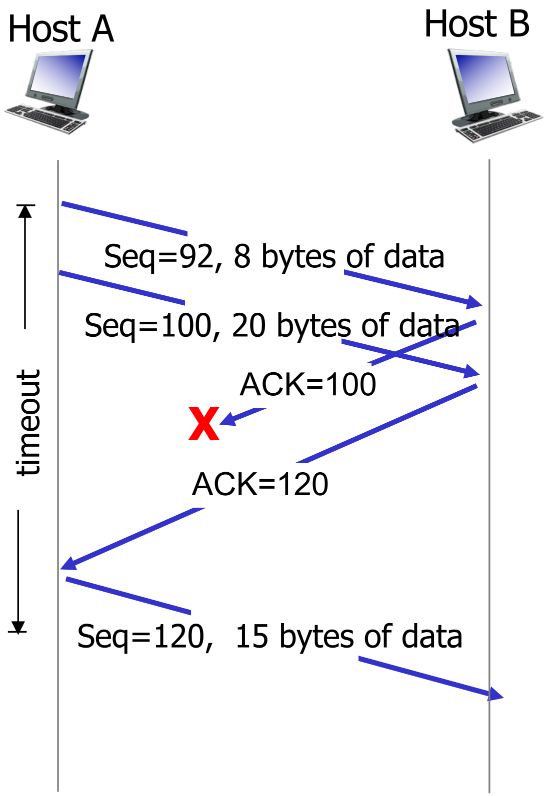
TCP: retransmission scenarios



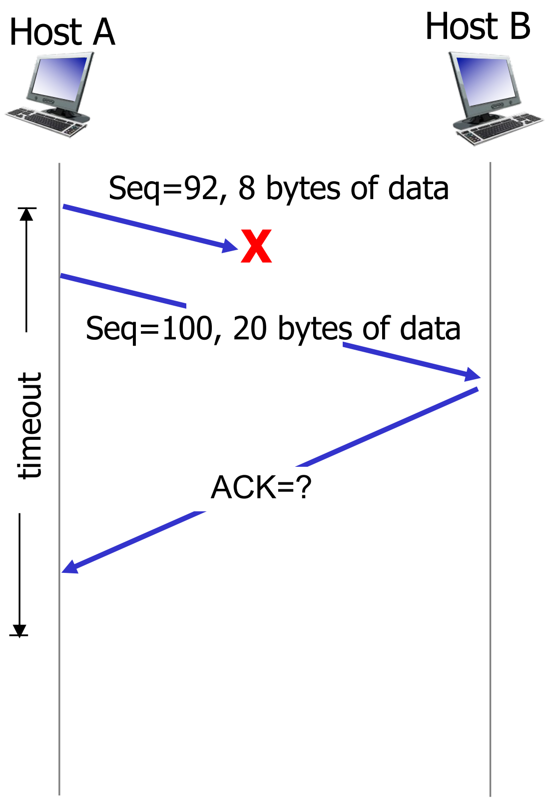
lost ACK scenario

premature timeout

TCP: retransmission scenarios



cumulative ACK



cumulative ACK

What does TCP do?

Most of our previous tricks, but a few differences

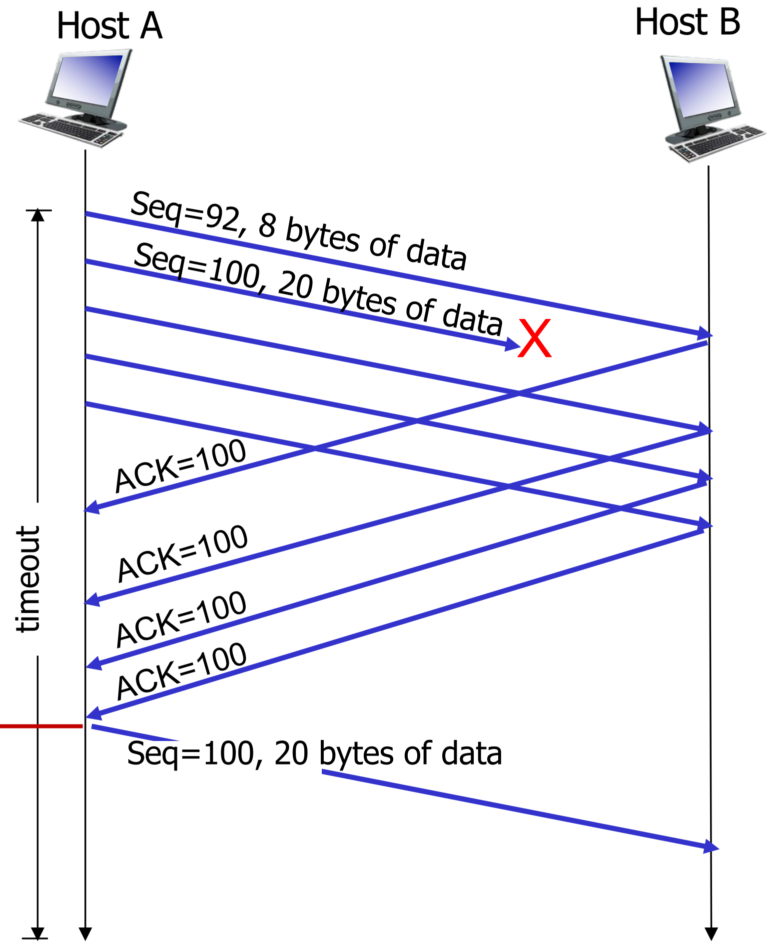
- Checksum
- Sequence numbers are byte offsets
- Receiver sends cumulative acknowledgements (like GBN)
- Receivers may not drop out-of-sequence packets (like SR)
- Sender maintains a single retransmission timer (like GBN) and retransmits on timeout
- Introduces fast retransmit: optimisation that uses duplicate ACKs to trigger early retransmission

TCP fast retransmit

TCP fast retransmit

if sender receives 3 additional ACKs for same data ("triple duplicate ACKs"), resend unACKed segment with smallest seq #

- likely that unACKed segment lost, so don't wait for timeout



Receipt of three duplicate ACKs indicates 3 segments received after a missing segment – lost segment is likely. So retransmit!

Quiz: TCP Sequence Numbers?

A TCP Sender is about to send a segment of size 100 bytes with sequence number 1234 and ack number 436. What is the highest sequence number up to (and including) which this sender has received from the receiver?

A. 1233
B. 436
C. 435
D. 1334
E. 536

Answer : C
Cumulative ACK

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Quiz: TCP Sequence Numbers?

A TCP Sender is about to send a segment of size 100 bytes with sequence number 1234 and ack number 436. Is it possible that the receiver has received byte number 1335?

A. Yes

B. No

Answer: A. It is possible that this packet being transmitted may be a retransmission and the next packet (in sequence) may have been already received

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Quiz: TCP Sequence Numbers?

The following statement is true about the TCP sliding window protocol for implementing reliable data transfer

A. It exclusively uses the ideas of Go-Back-N
B. It exclusively uses the ideas of Selective Repeat
C. It uses a combination of ideas of Go-Back-N and Selective-Repeat
D. It uses none of the ideas of Go-Back-N and Selective-Repeat

Answer: C

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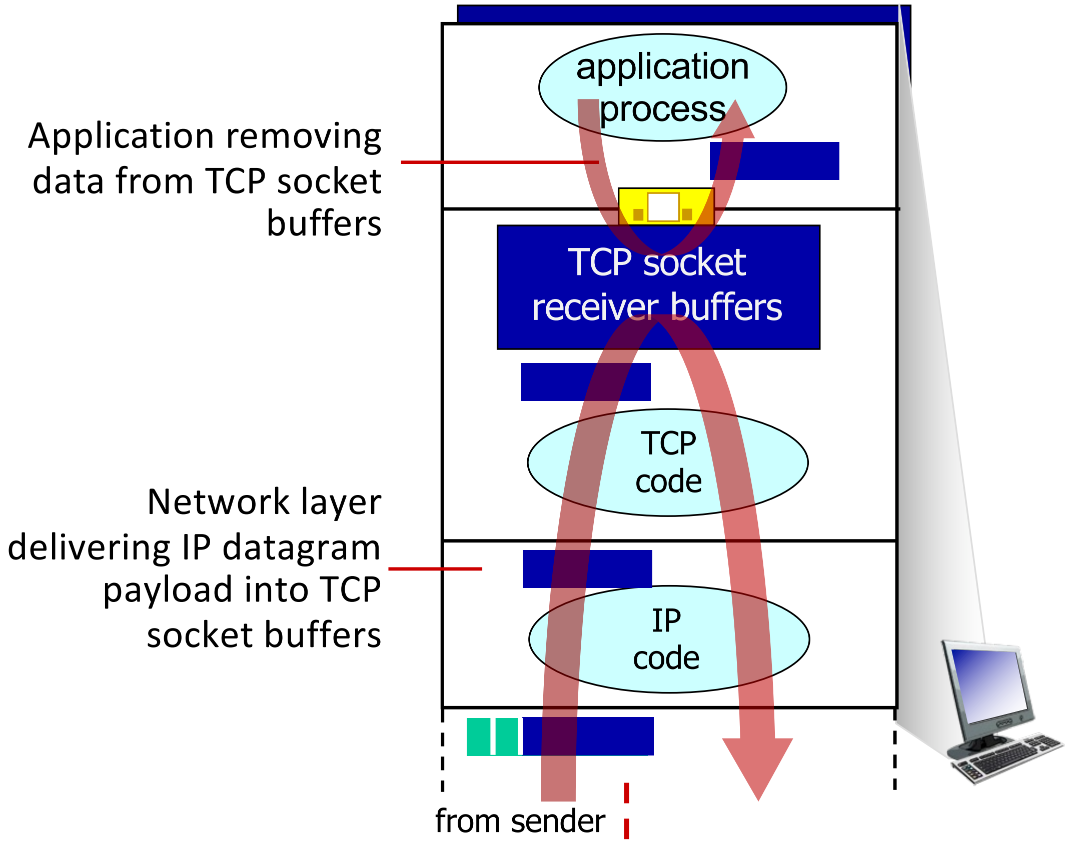
TCP flow control

Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?



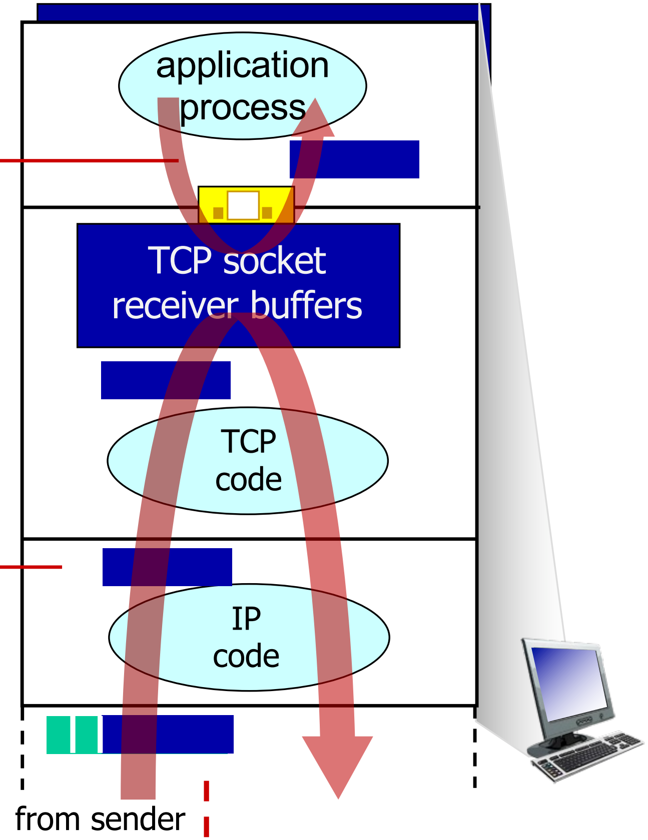
receiver protocol stack

TCP flow control

Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?

Application removing data from TCP socket buffers

Network layer delivering IP datagram payload into TCP socket buffers



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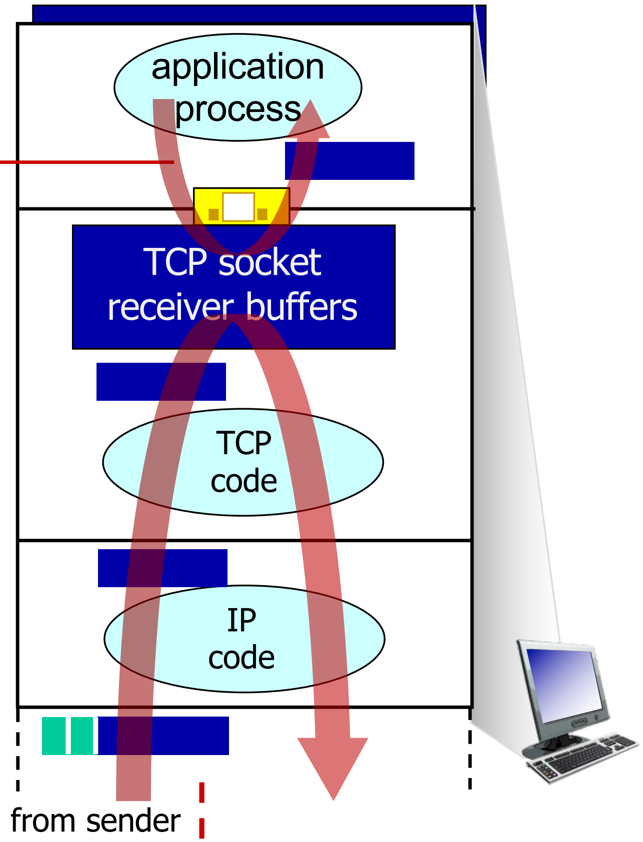
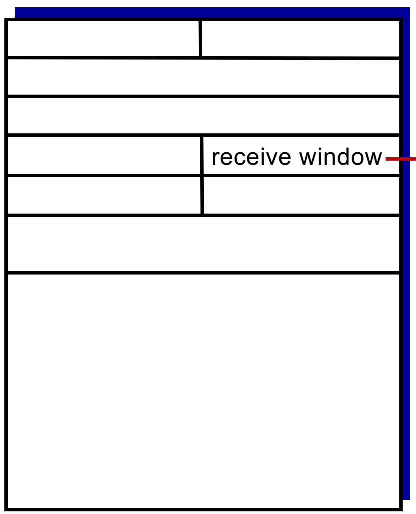


receiver protocol stack



flow control: # bytes receiver willing to accept

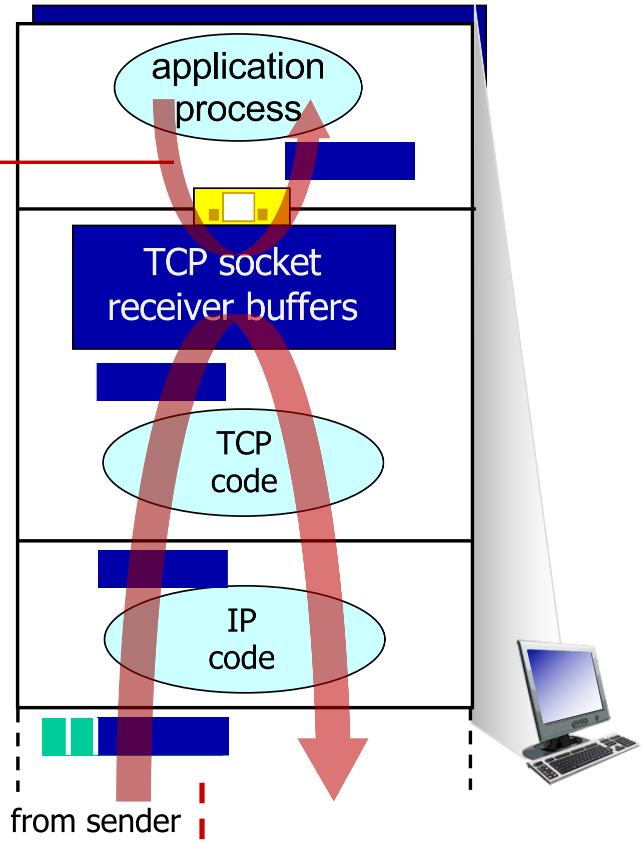
TCP flow control

Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?

flow control

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

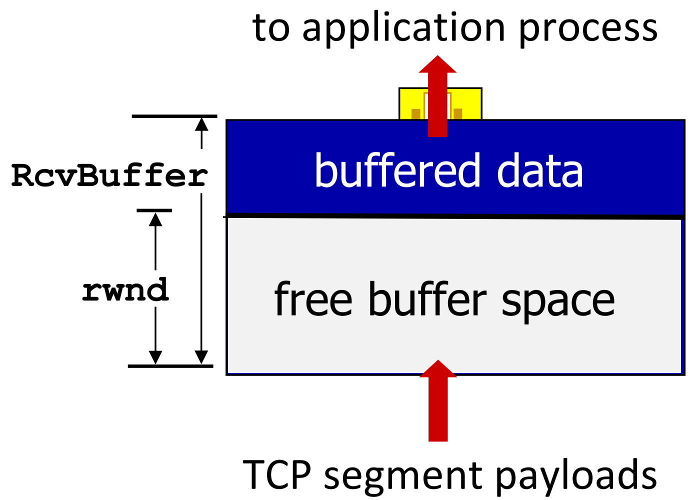
Application removing data from TCP socket buffers



receiver protocol stack

TCP flow control

- TCP receiver "advertises" free buffer space in **`rwnd`** field in TCP header
  - **`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 received **`rwnd`**
- guarantees receive buffer will not overflow



TCP receiver-side buffering

TCP flow control

- TCP receiver "advertises" free buffer space in **rwnd** field in TCP header
  - **RcvBuffer** size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust **RcvBuffer**
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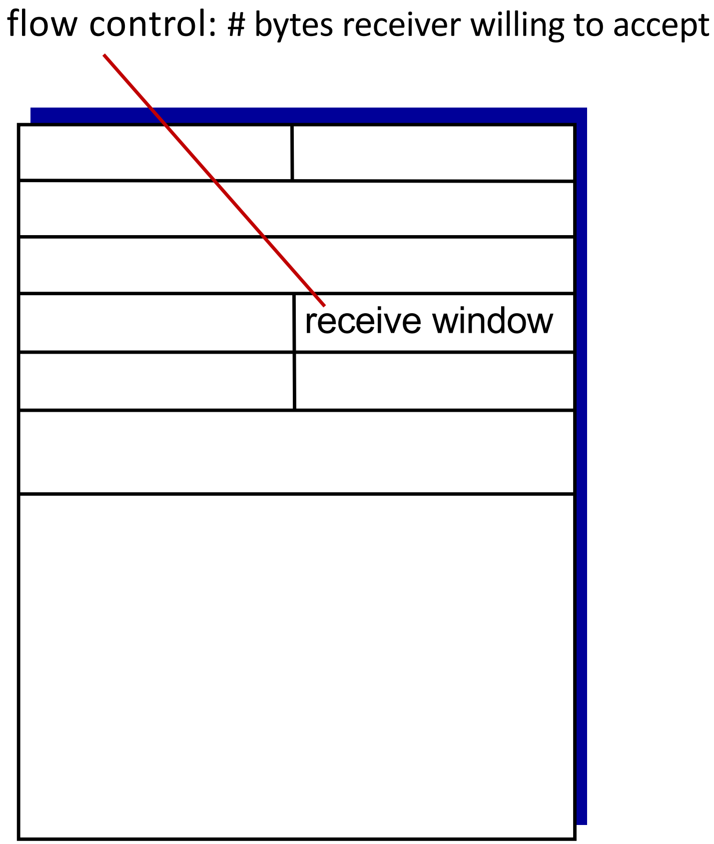

TCP segment format

TCP flow control

- What if `rwnd = 0`?
  - Sender would stop sending data
  - Eventually the receive buffer would have space when the application process reads some bytes
  - But how does the receiver advertise the new `rwnd` to the sender?
- Sender keeps sending TCP segments with one data byte to the receiver
- These segments are dropped but acknowledged by the receiver with a zero-window size
- Eventually when the buffer empties, non-zero window is advertised

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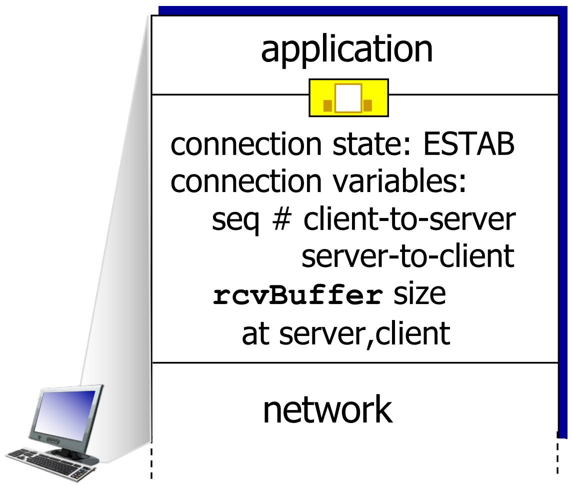
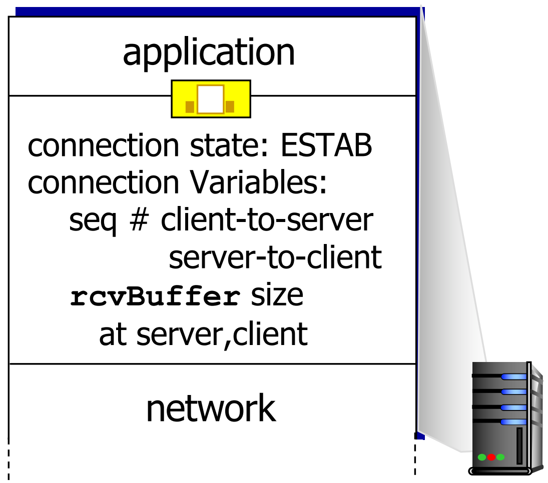
TCP connection management

before exchanging data, sender/receiver “handshake”:

- agree to establish connection (each knowing the other willing to establish connection)
- agree on connection parameters (e.g., starting seq #s)



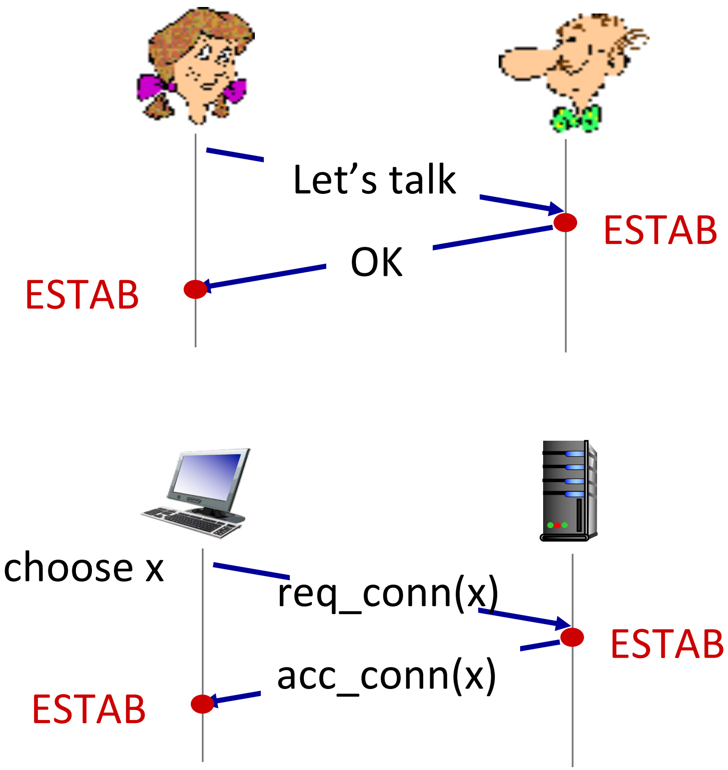



```
Socket clientSocket =
  newSocket("hostname","port number");
```

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

Agreeing to establish a connection

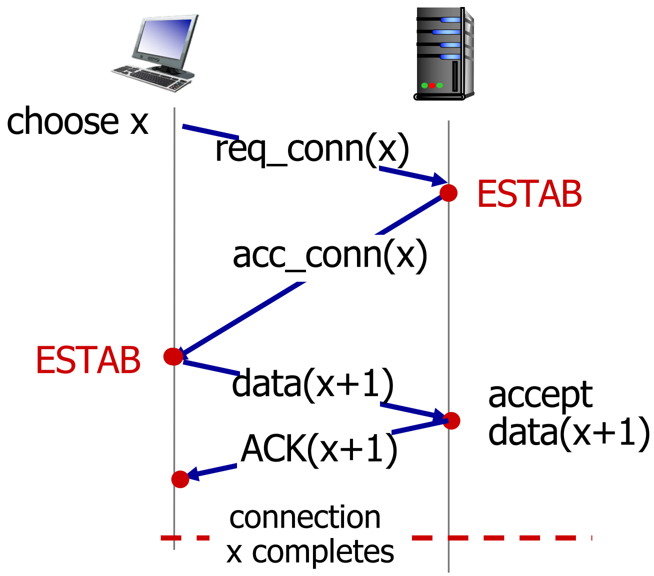
2-way handshake:



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

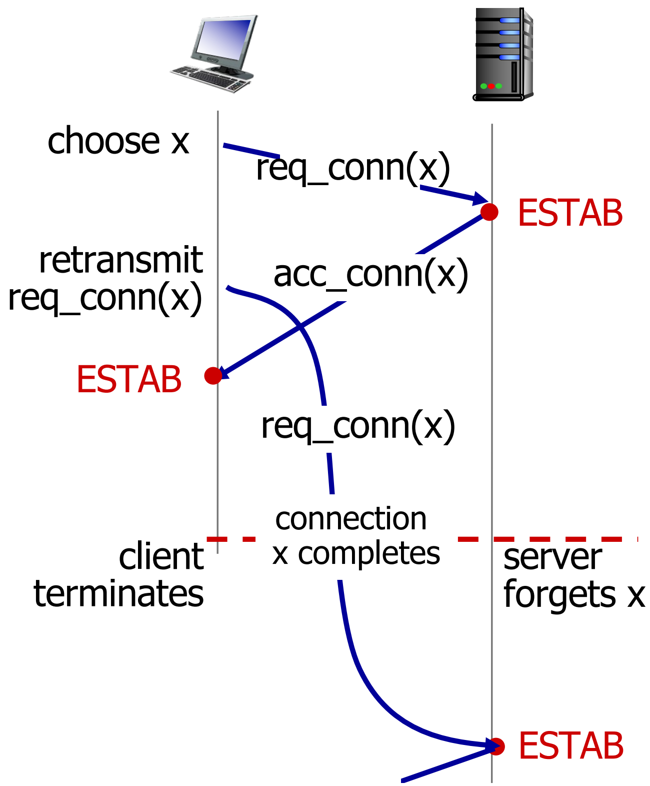
- variable delays
- retransmitted messages (e.g. req_conn(x)) due to message loss
- message reordering
- can't "see" other side

2-way handshake scenarios



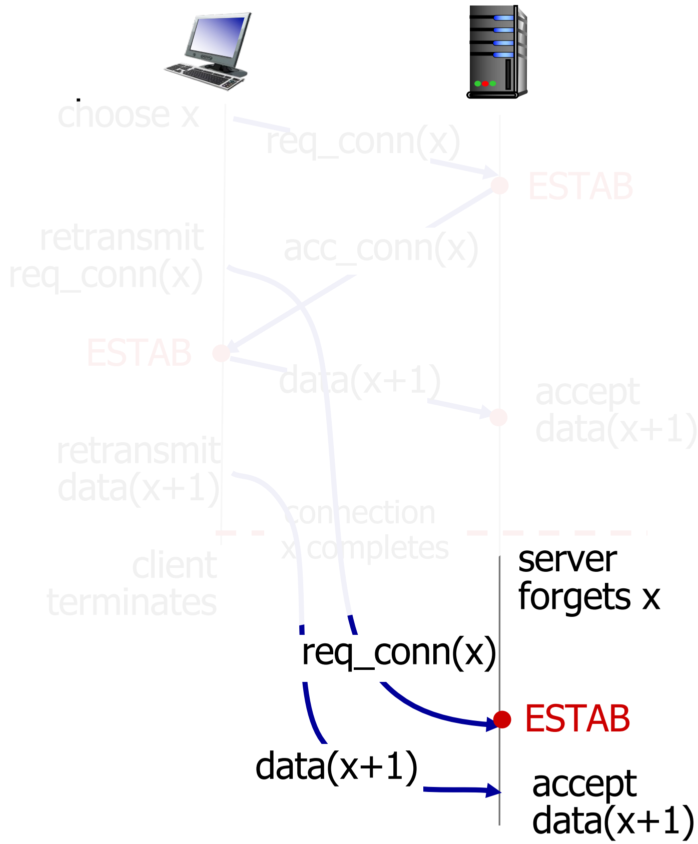
No problem!

2-way handshake scenarios



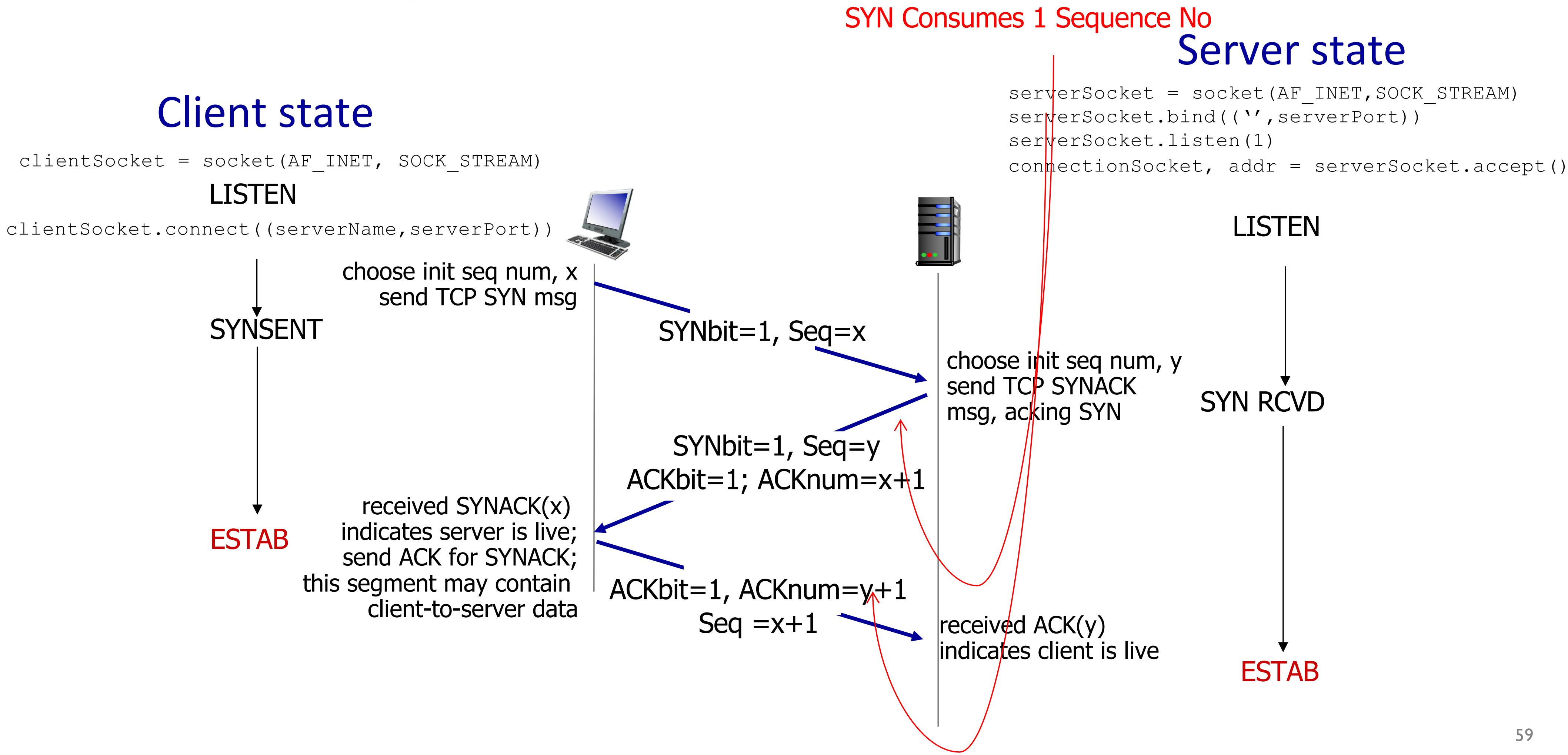
Problem: half open connection! (no client)

2-way handshake scenarios



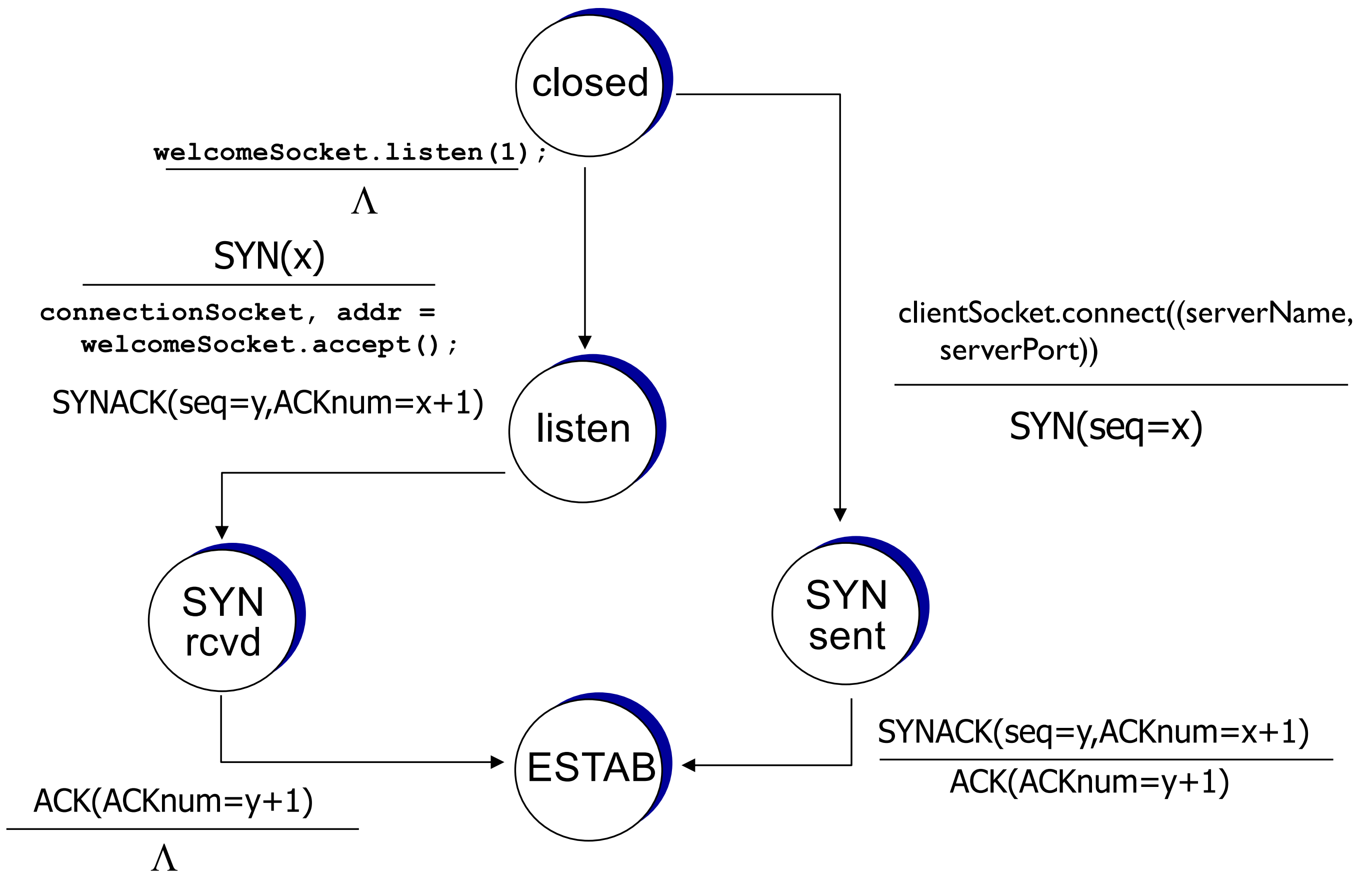
Problem: dup data accepted!

TCP 3-way handshake



A human 3-way handshake protocol

TCP 3-way handshake: Partial state machine



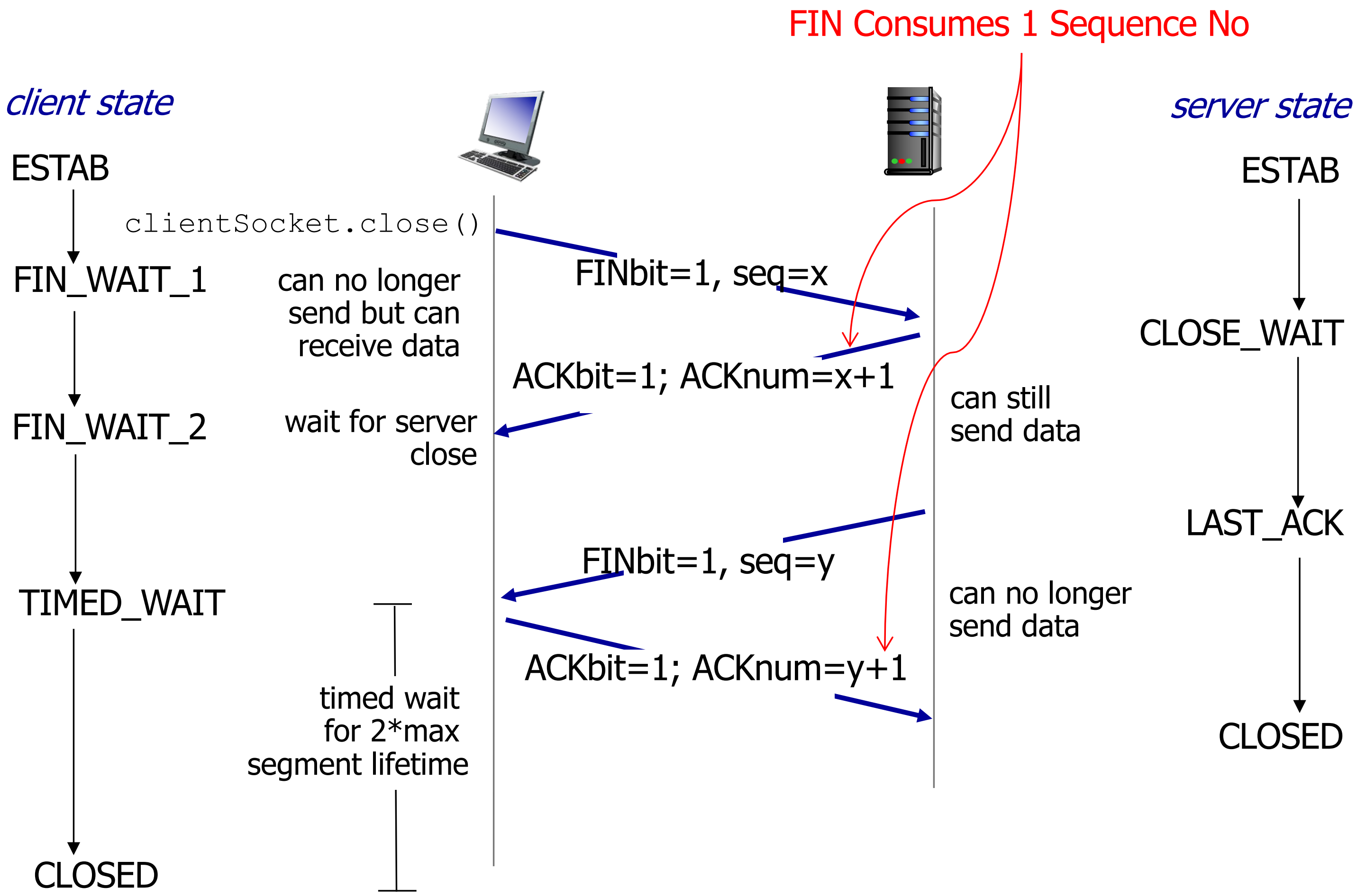
What if the SYN Packet Gets Lost?

- Suppose the SYN packet gets lost
  - Packet is lost inside the network, or:
  - Server discards the packet (e.g., it's too busy)
- Eventually, no SYN-ACK arrives
  - Sender sets a timer and waits for the SYN-ACK
  - … and retransmits the SYN if needed
- How should the TCP sender set the timer?
  - Sender has no idea how far away the receiver is
  - Hard to guess a reasonable length of time to wait
  - **SHOULD** (RFCs 1122,2988) use default of 3 second, RFC 6298 use default of 1 second

TCP: closing a connection

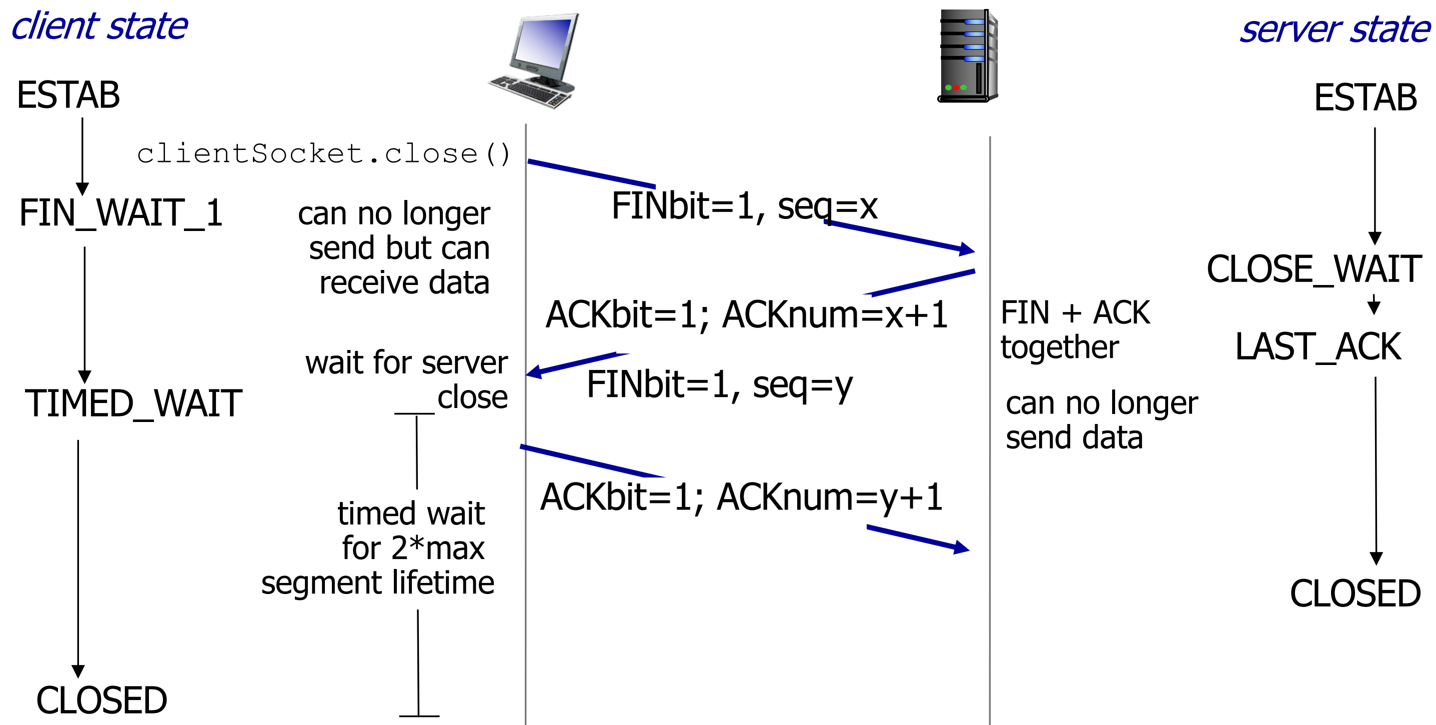
- client, server each close their side of connection
  - send TCP segment with FIN bit = 1
- respond to received FIN with ACK
  - on receiving FIN, ACK can be combined with own FIN
- simultaneous FIN exchanges can be handled

Normal Termination, One at a Time

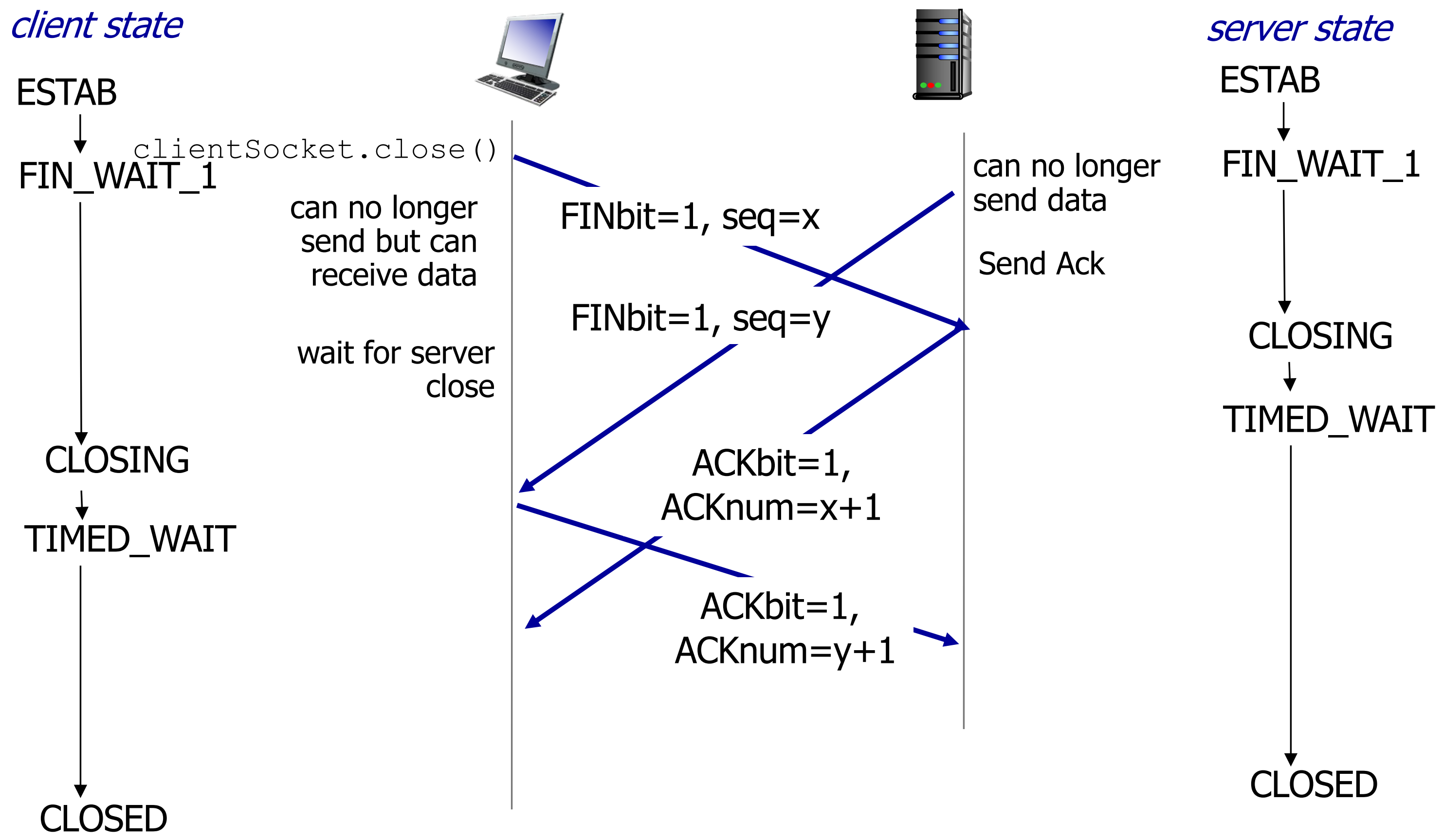


TIMED_WAIT: Can retransmit ACK if last ACK is lost

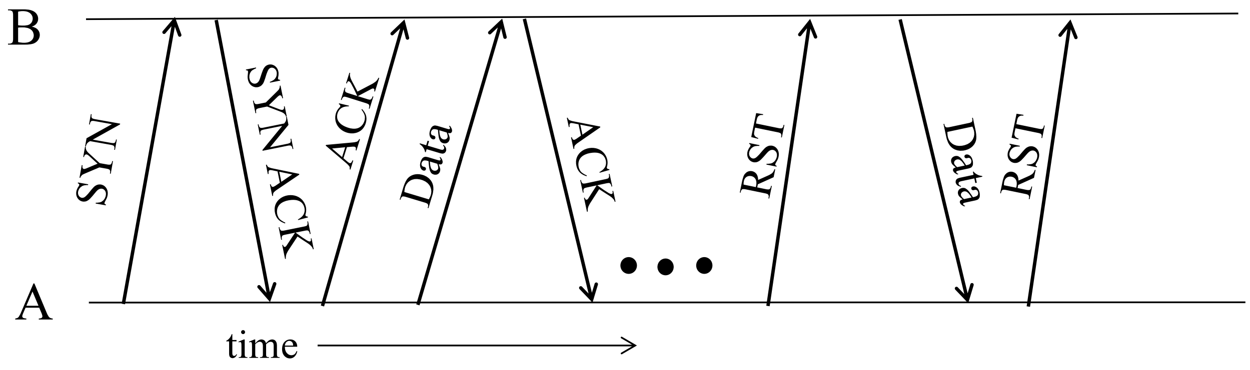
Normal Termination, Both Together



Simultaneous Closure

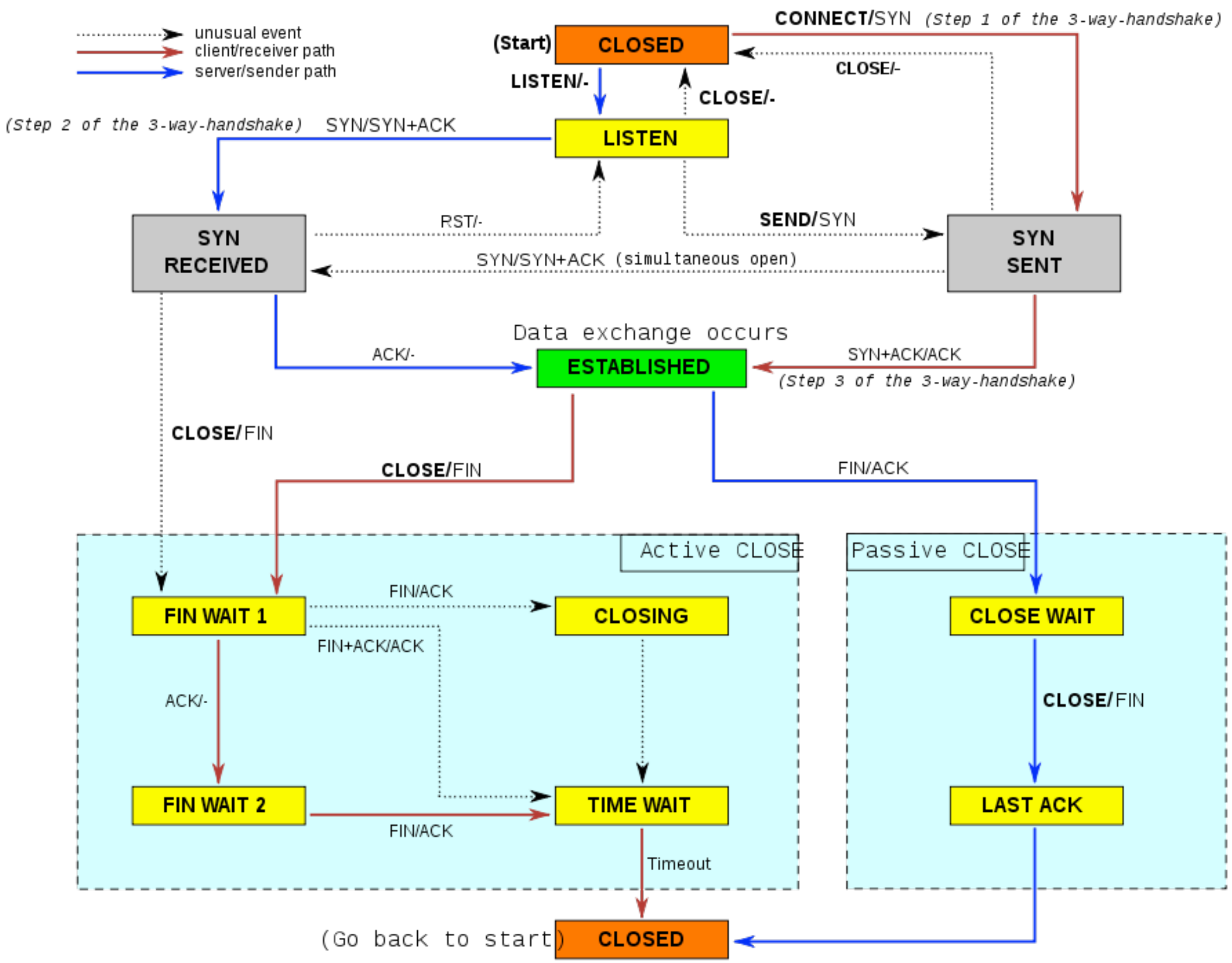


Abrupt Termination



- A sends a RESET (**RST**) to B
  - E.g., because application process on A crashed
- That's it
  - B does not ack the **RST**
  - Thus, **RST** is not delivered reliably
  - And: any data in flight is lost
  - But: if B sends anything more, will elicit another **RST**

TCP Finite State Machine



TCP SYN Attack (SYN flooding)

- Miscreant creates a fake SYN packet
  - Destination is IP address of victim host (usually some server)
  - Source is some spoofed IP address
- Victim host on receiving creates a TCP connection state i.e allocates buffers, creates variables, etc and sends SYN ACK to the spoofed address (half-open connection)
- ACK never comes back
- After a timeout connection state is freed
- However for this duration the connection state is unnecessarily created
- Further miscreant sends large number of fake SYNs
  - Can easily overwhelm the victim
- Solutions:
  - Increase size of connection queue
  - Decrease timeout wait for the 3-way handshake
  - Firewalls: list of known bad source IP addresses
  - TCP SYN Cookies (explained on next slide)

TCP SYN Cookie

- On receipt of SYN, server does not create connection state
- It creates an initial sequence number (*init_seq*) that is a hash of source & dest IP address and port number of SYN packet (secret key used for hash)
  - Replies back with SYN ACK containing *init_seq*
  - Server does not need to store this sequence number
- If original SYN is genuine, an ACK will come back
  - Same hash function run on the same header fields to get the initial sequence number (*init_seq*)
  - Checks if the ACK is equal to (*init_seq*+1)
  - Only create connection state if above is true
- If fake SYN, no harm done since no state was created

http://etherealmind.com/tcp-syn-cookies-ddos-defence/

Quiz: TCP Connection Management?

Assume that one end of a TCP connection selects an initial sequence number 120. The first TCP segment containing data sent by this end point will have a sequence number of ____

A. 120

B. 121

C. 122

D. 128

E. 0

ANSWER: B (because SYN uses 1 seq no.)

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Quiz: TCP Connection Management?

Assume that one end point of the TCP connection sends a FIN segment. If it never receives an ACK, what should it do?

A. Assume that the connection is closed and do nothing

B. Retransmit the FIN

C. Transmit an ACK

ANSWER: B

D. Start crying

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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

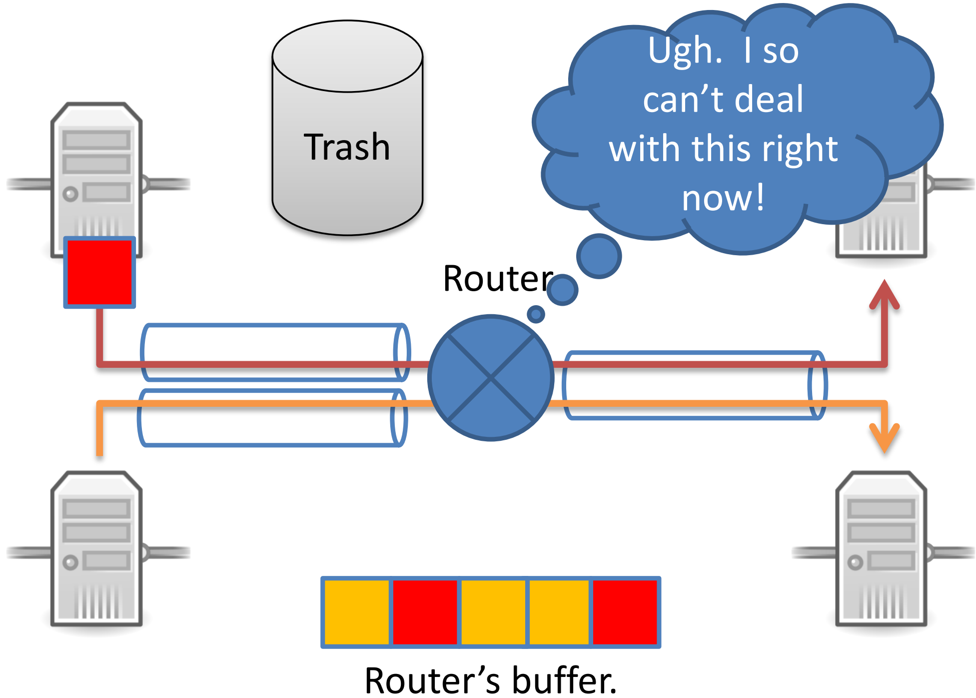
3.7 TCP congestion control

Principles of congestion control

congestion:

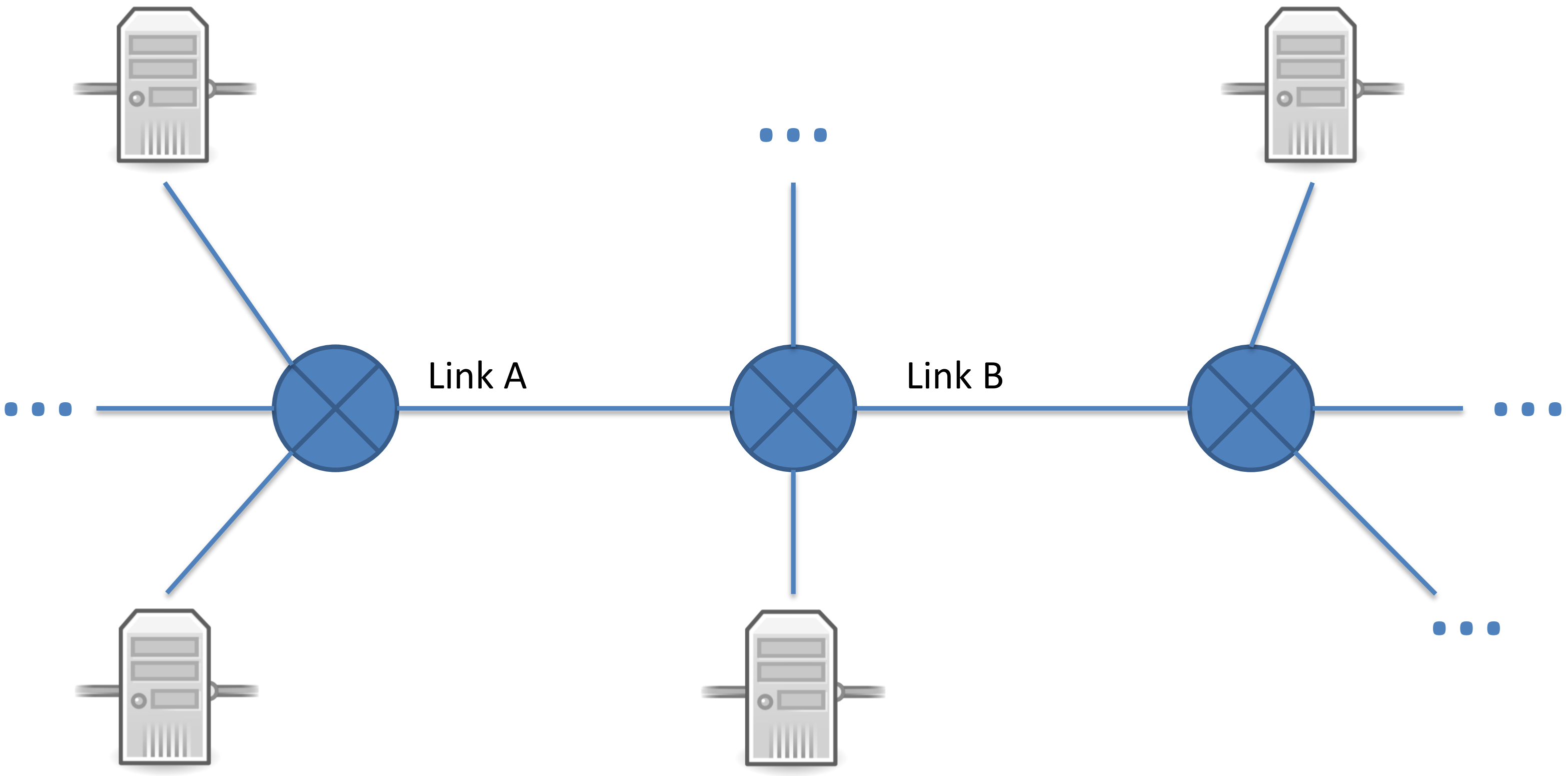
- informally: "too many sources sending too much data too fast for *network* to handle"
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Congestion

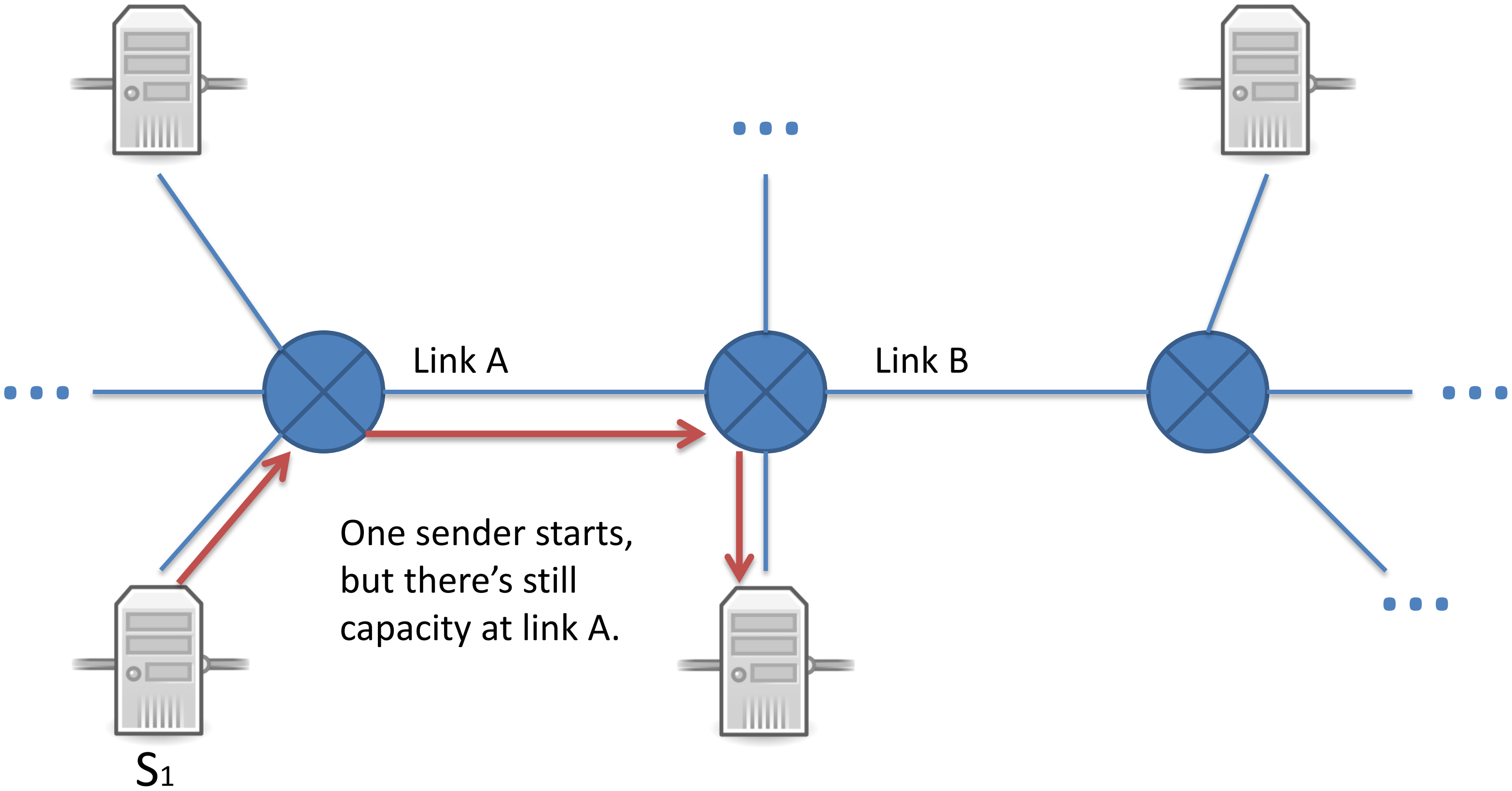


Incoming rate is faster than outgoing link can support.

Congestion Collapse

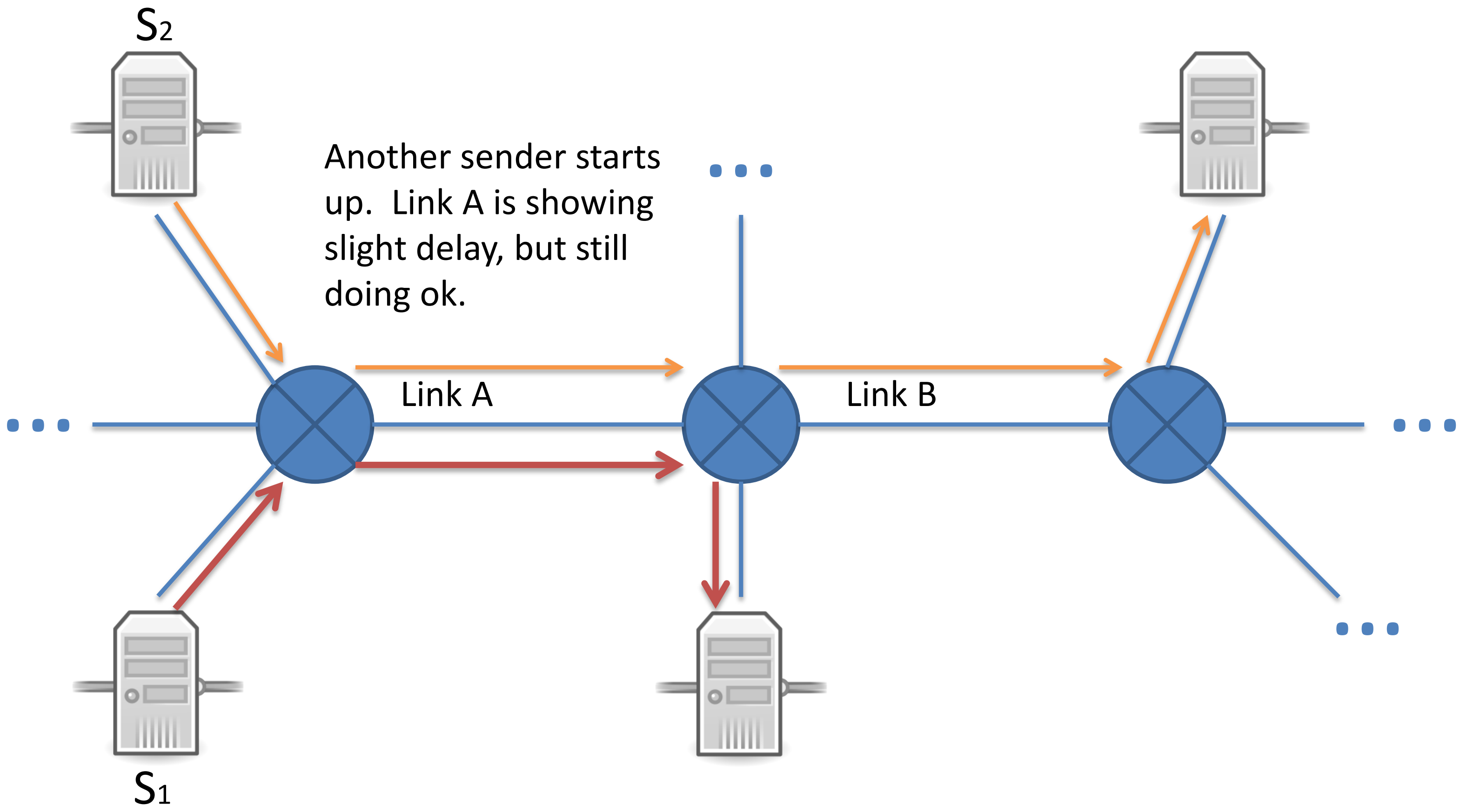


Congestion Collapse

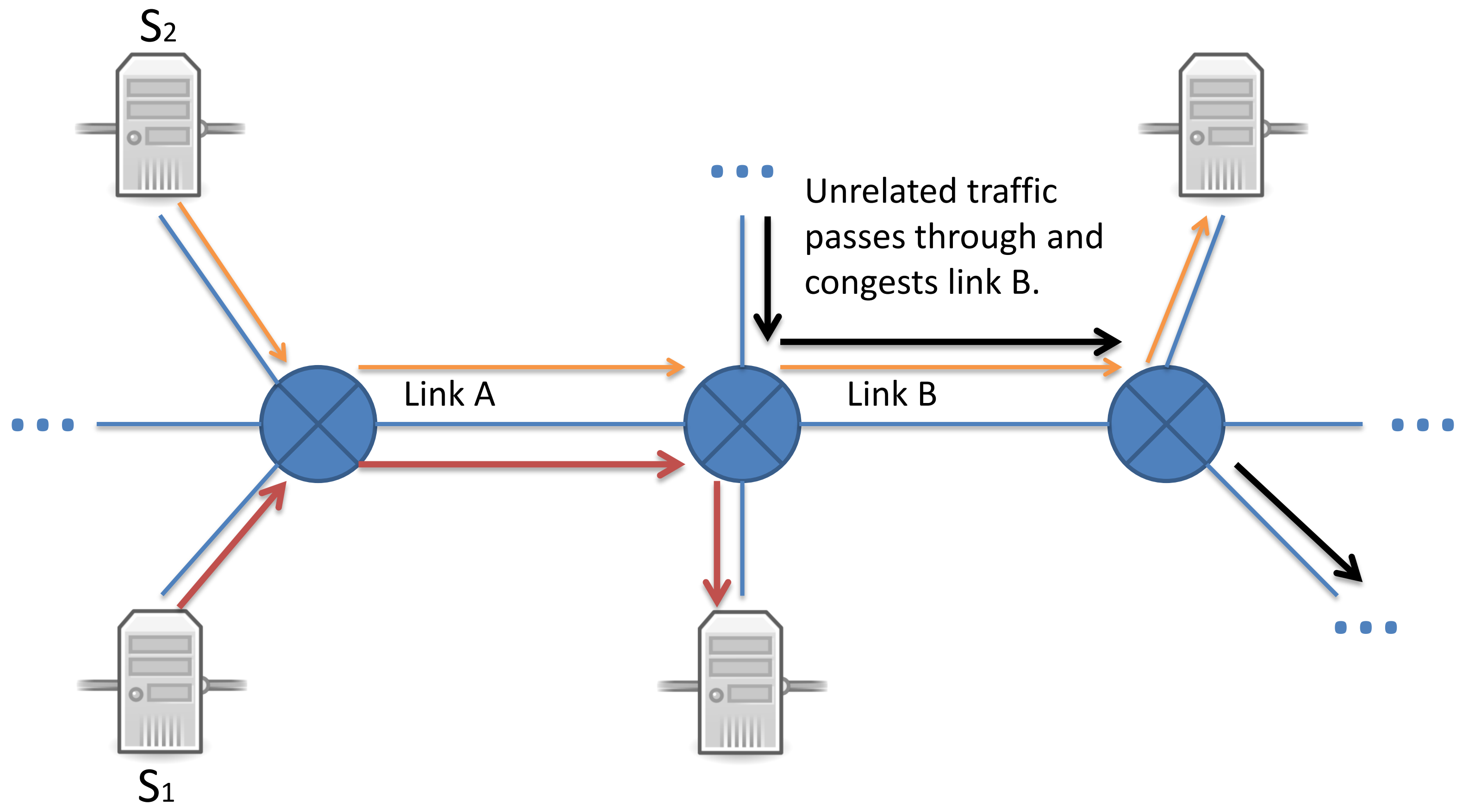


One sender starts, but there's still capacity at link A.

Congestion Collapse

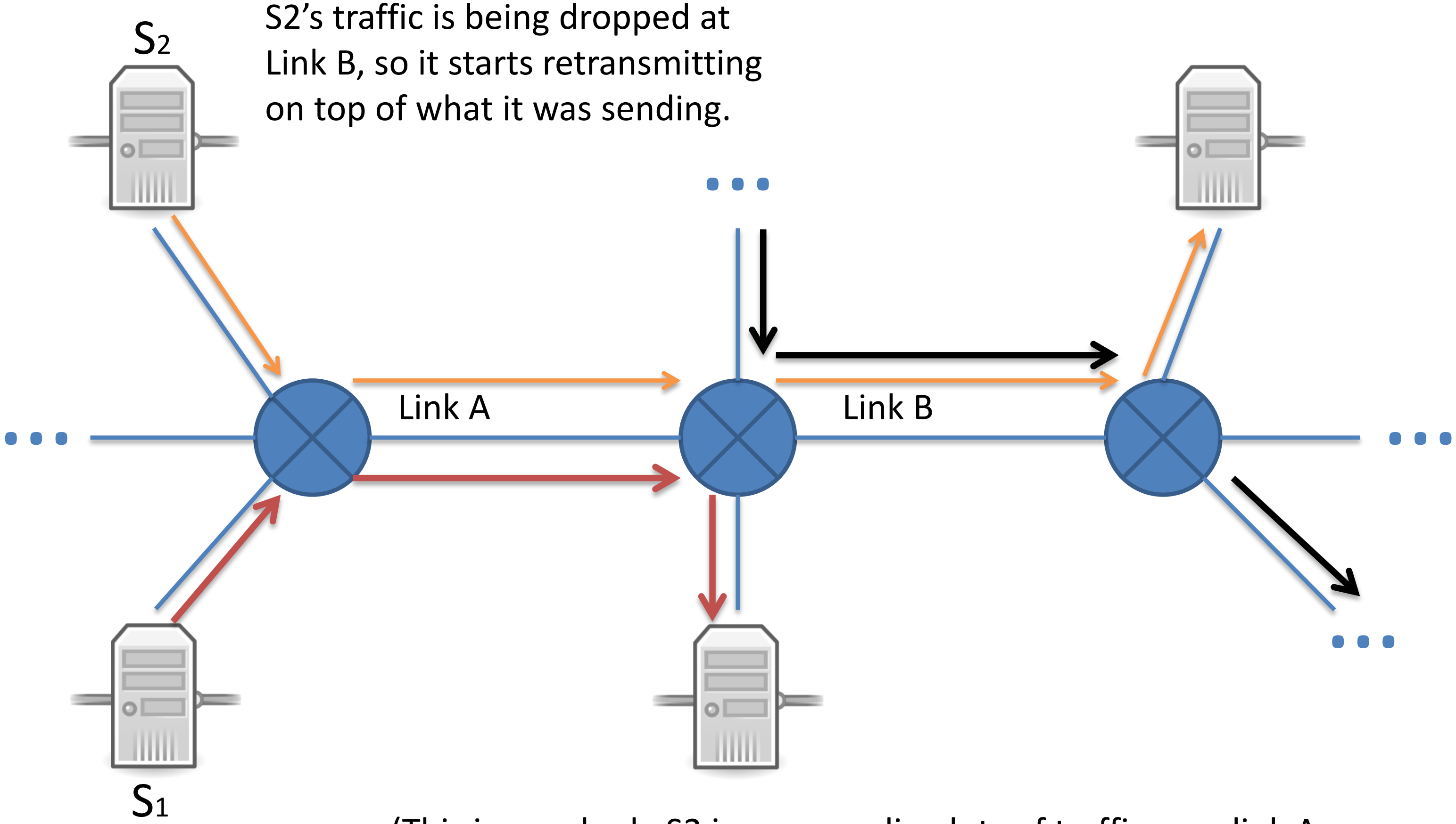


Congestion Collapse



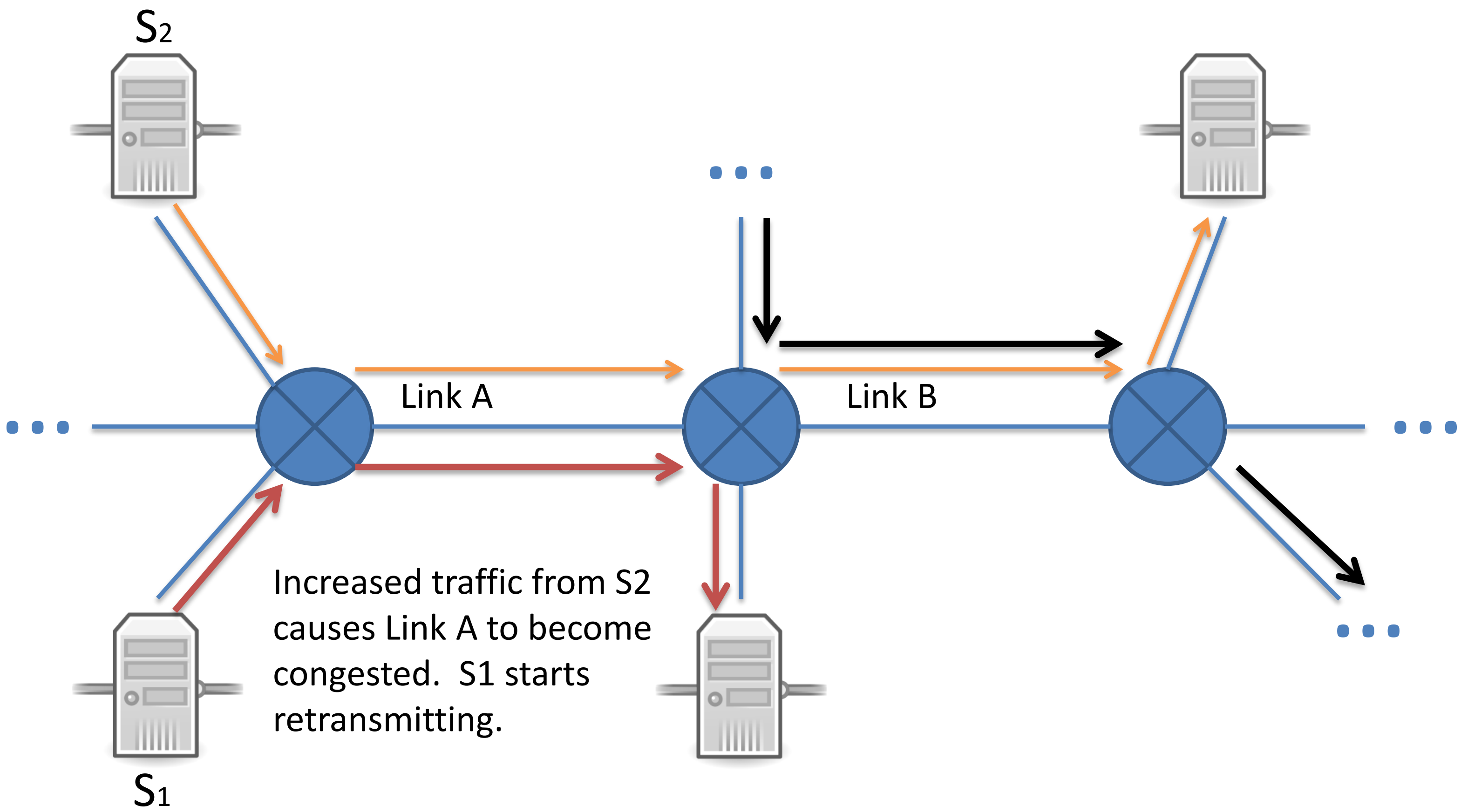
Congestion Collapse

S2's traffic is being dropped at Link B, so it starts retransmitting on top of what it was sending.

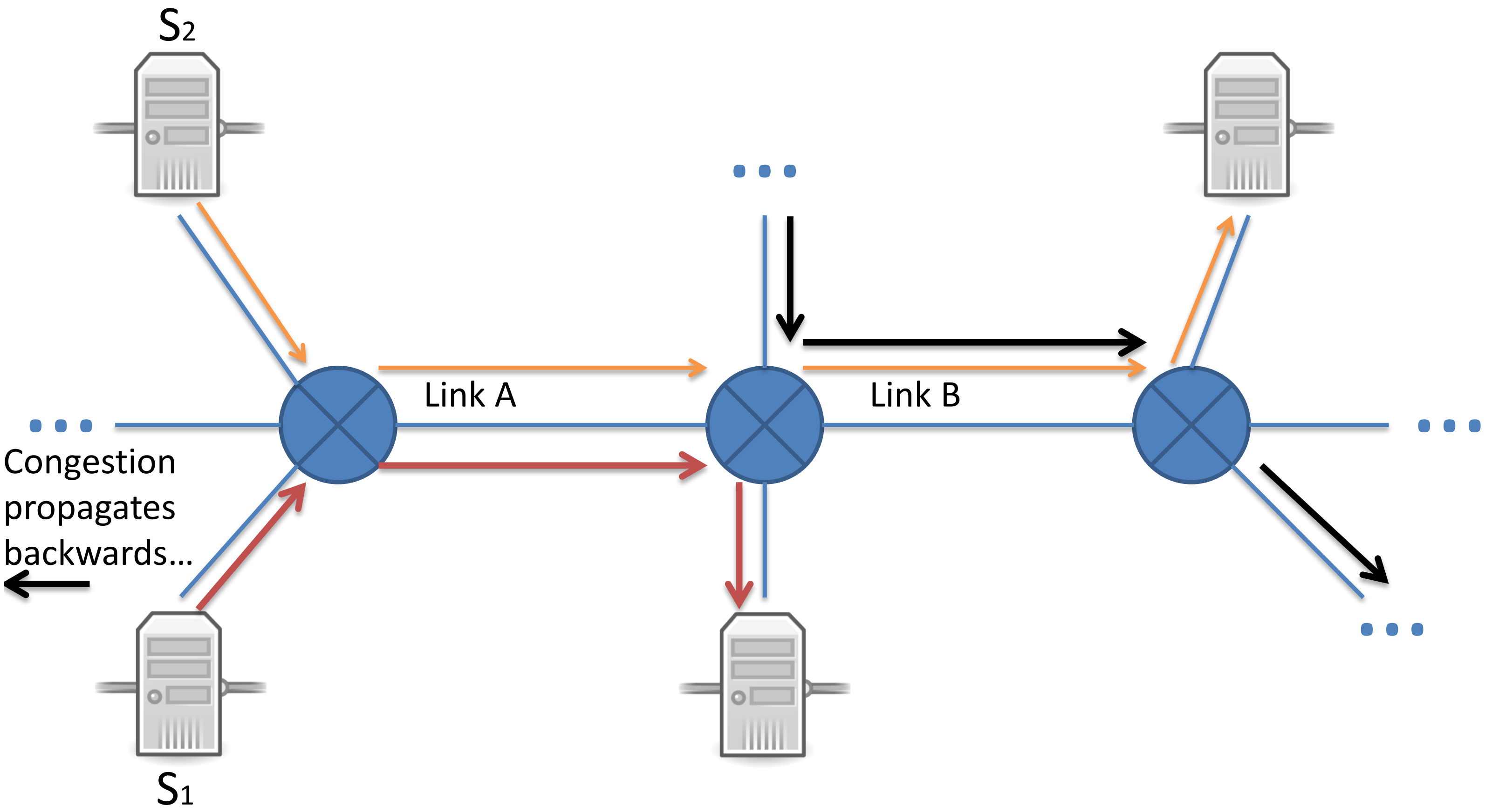


(This is very bad. S2 is now sending lots of traffic over link A that has no hope of crossing link B.)

Congestion Collapse



Congestion Collapse



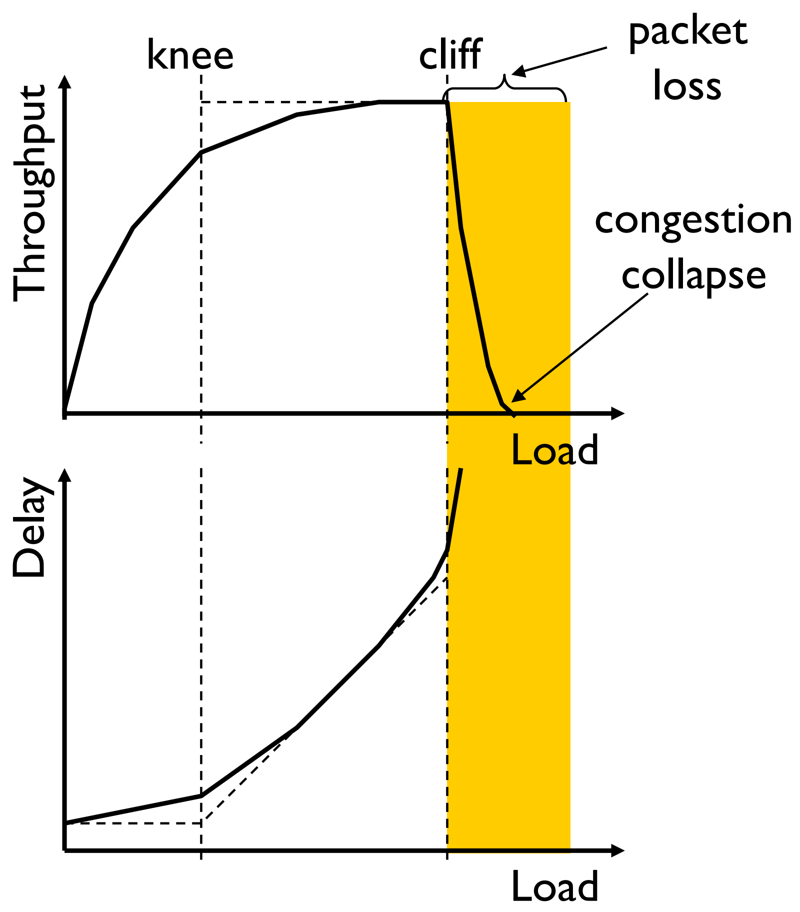
Without congestion control

congestion:

- Increases delays
  - If delays > RTO, sender retransmits
- Increases loss rate
  - Dropped packets also retransmitted
- Increases retransmissions, many unnecessary
  - Wastes capacity of traffic that is never delivered
  - Increase in load results in decrease in useful work done
- Increases congestion, cycle continues …

Cost of Congestion

- Knee – point after which
  - Throughput increases slowly
  - Delay increases fast
- Cliff – point after which
  - Throughput starts to drop to zero (congestion collapse)
  - Delay approaches infinity



Congestion Collapse

This happened to the Internet (then NSFnet) in 1986

- Rate dropped from a *blazing* 32 Kbps to 40bps
- This happened on and off for *two years*
- In 1988, Van Jacobson published “Congestion Avoidance and Control”
- The fix: senders voluntarily limit sending rate

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 (SNA, DECbit, TCP/IP ECN, ATM)
  - explicit rate for sender to send at

Transport Layer: Outline

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- connection management

3.6 principles of congestion control

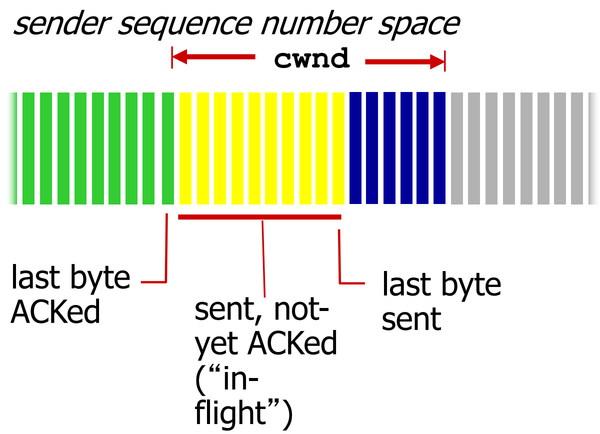
3.7 TCP congestion control

TCP's Approach in a Nutshell

- TCP connection maintains a window
  - Controls number of packets in flight

- *TCP sending rate:*
  - *roughly:* send cwnd bytes, wait RTT for ACKs, then send more bytes

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



- Vary window size to control sending rate

All These Windows…

- Congestion Window: CWND
  - How many bytes can be sent without overflowing routers
  - Computed by the sender using congestion control algorithm

- Flow control window: Advertised / Receive Window (RWND)
  - How many bytes can be sent without overflowing receiver's buffers
  - Determined by the receiver and reported to the sender

- Sender-side window = minimum{CWND, RWND}
  - Assume for this discussion that RWND >> CWND

CWND

- This lecture will talk about CWND in units of MSS
  - (Recall MSS: Maximum Segment Size, the amount of payload data in a TCP packet)
  - This is only for pedagogical purposes

- Keep in mind that real implementations maintain CWND in bytes

Two Basic Questions

- How does the sender detect congestion?

- How does the sender adjust its sending rate?

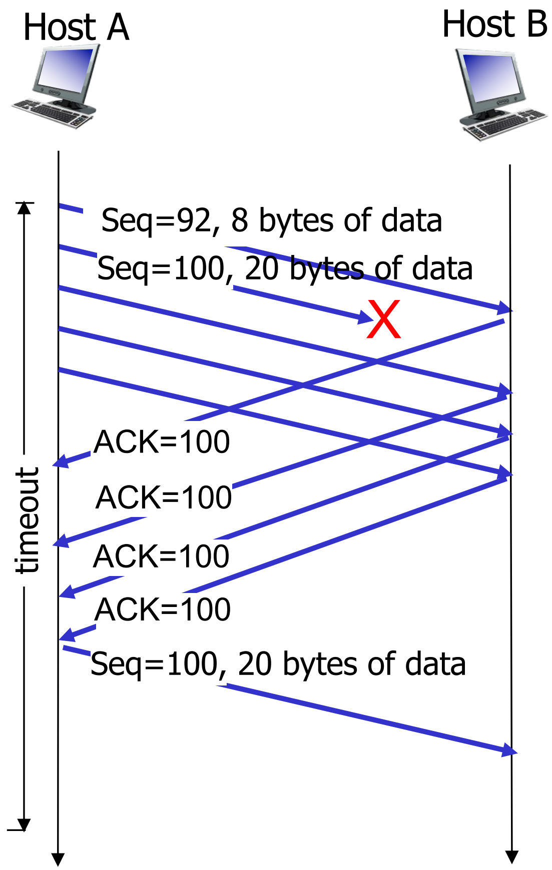
Detection Congestion: Infer Loss

- Duplicate ACKs: isolated loss
  - dup ACKs indicate network capable of delivering some segments

- Timeout: much more serious
  - Not enough dup ACKs
  - Must have suffered several losses

- Will adjust rate differently for each case

RECAP: TCP fast retransmit (dup acks)



fast retransmit after sender receipt of triple duplicate ACK

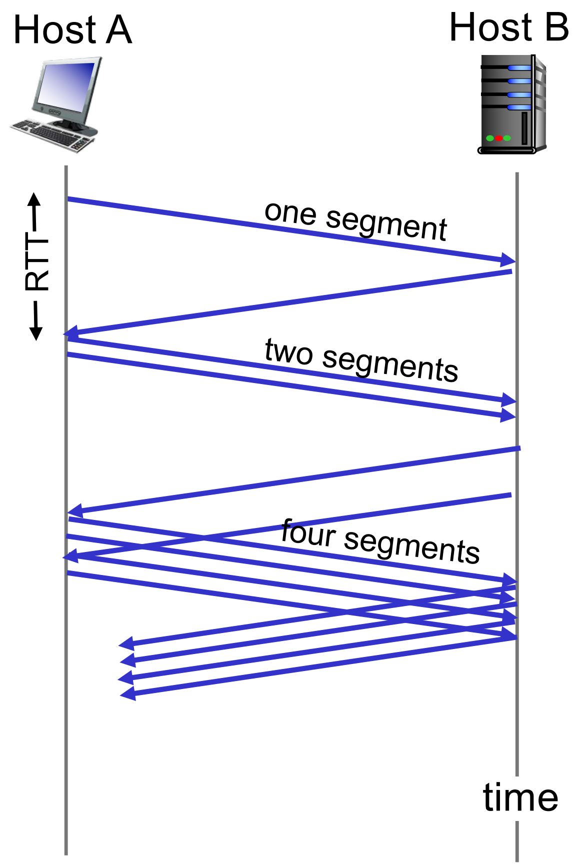
Rate Adjustment

- Basic structure:
  - Upon receipt of ACK (of new data): increase rate
  - Upon detection of loss: decrease rate

- How we increase/decrease the rate depends on the phase of congestion control we're in:
  - Discovering available bottleneck bandwidth *vs.*
  - Adjusting to bandwidth variations

TCP Slow Start (Bandwidth discovery)

- when connection begins, increase rate exponentially until first loss event:
  - initially **cwnd** = 1 MSS
  - double **cwnd** every RTT (all ACKs)
  - Simpler implementation achieved by incrementing **cwnd** for every ACK received
    - cwnd += 1 for each ACK
- *summary:* initial rate is slow but ramps up exponentially fast



Adjusting to Varying Bandwidth

- Slow start gave an estimate of available bandwidth

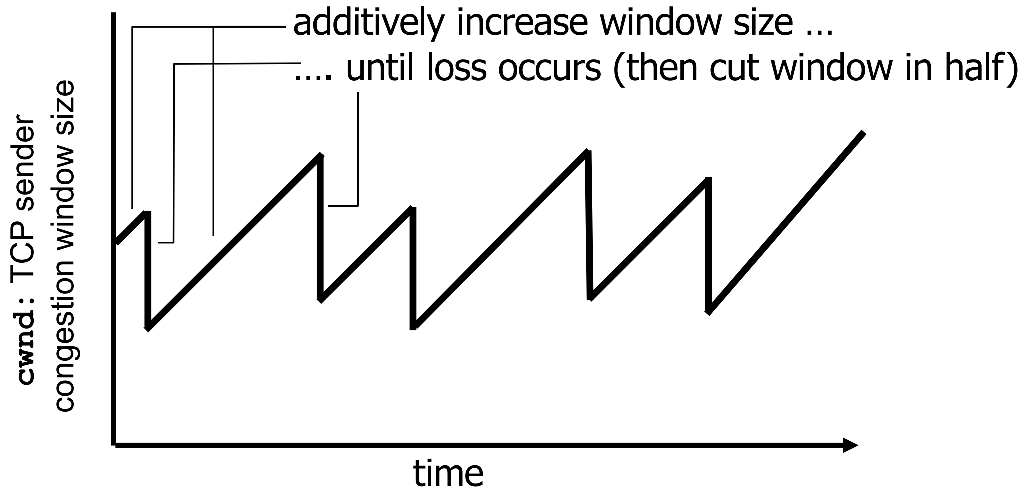
- Now, want to track variations in this available bandwidth, oscillating around its current value
  - Repeated probing (rate increase) and back-off (rate decrease)
  - Known as Congestion Avoidance (CA)

- TCP uses: "Additive Increase Multiplicative Decrease" (AIMD)

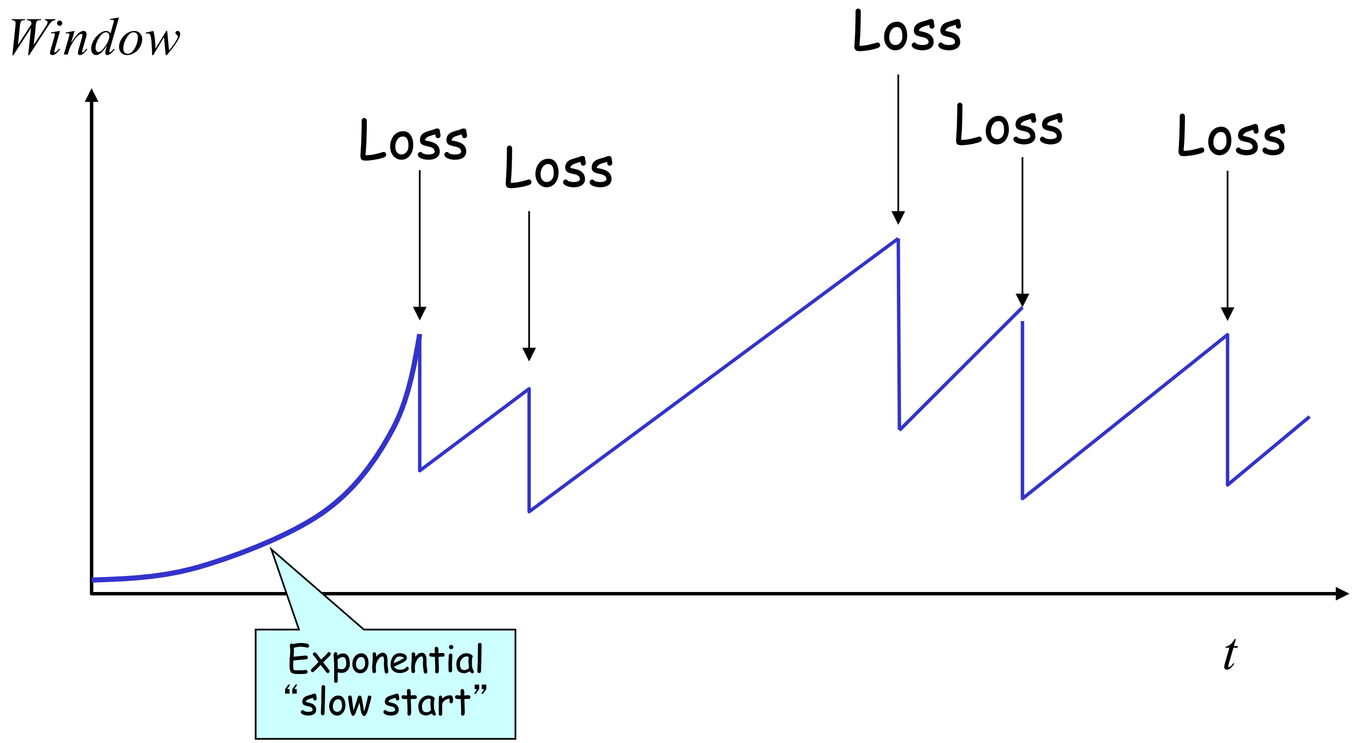
AIMD

- *approach:* sender increases transmission rate (window size), probing for usable bandwidth, until another congestion event occurs
  - *additive increase:* increase **`cwnd`** by 1 MSS every RTT until loss detected
    - For each successful RTT (all ACKs), **`cwnd = cwnd +1`** (in multiples of MSS)
    - Simple implementation: for each ACK, **`cwnd = cwnd + 1/cwnd`** (since there are **`cwnd/MSS`** packets in a window)
  - *multiplicative decrease:* cut **`cwnd`** in half after loss

AIMD saw tooth behavior: probing for bandwidth



Leads to the TCP "Sawtooth"



Slow-Start vs. AIMD

- When does a sender stop Slow-Start and start Congestion Avoidance?

- Introduce a "slow start threshold" (ssthresh)
  - Initialized to a large value

- Convert to CA when cwnd = ssthresh, sender switches from slow-start to AIMD-style increase
  - On loss, ssthresh = CWND/2

Implementation

- State at sender
  - CWND (initialized to a small constant)
    - the slides use multiple of MSS
  - ssthresh (initialized to a large constant)
  - [Also dupACKcount and timer, as before]

- Events
  - ACK (new data)
  - dupACK (duplicate ACK for old data)
  - Timeout

Event: ACK (new data)

- If CWND < ssthresh
  - CWND += 1

- *Hence after one RTT (All ACKs with no drops):*
  CWND = 2xCWND

Event: ACK (new data)

- If CWND < ssthresh
  - CWND += 1

Slow start phase

- Else
  - CWND = CWND + 1/CWND

"Congestion Avoidance" phase (additive increase)

- *Hence after one RTT (All ACKs with no drops):*
  CWND = CWND + 1

Event: dupACK

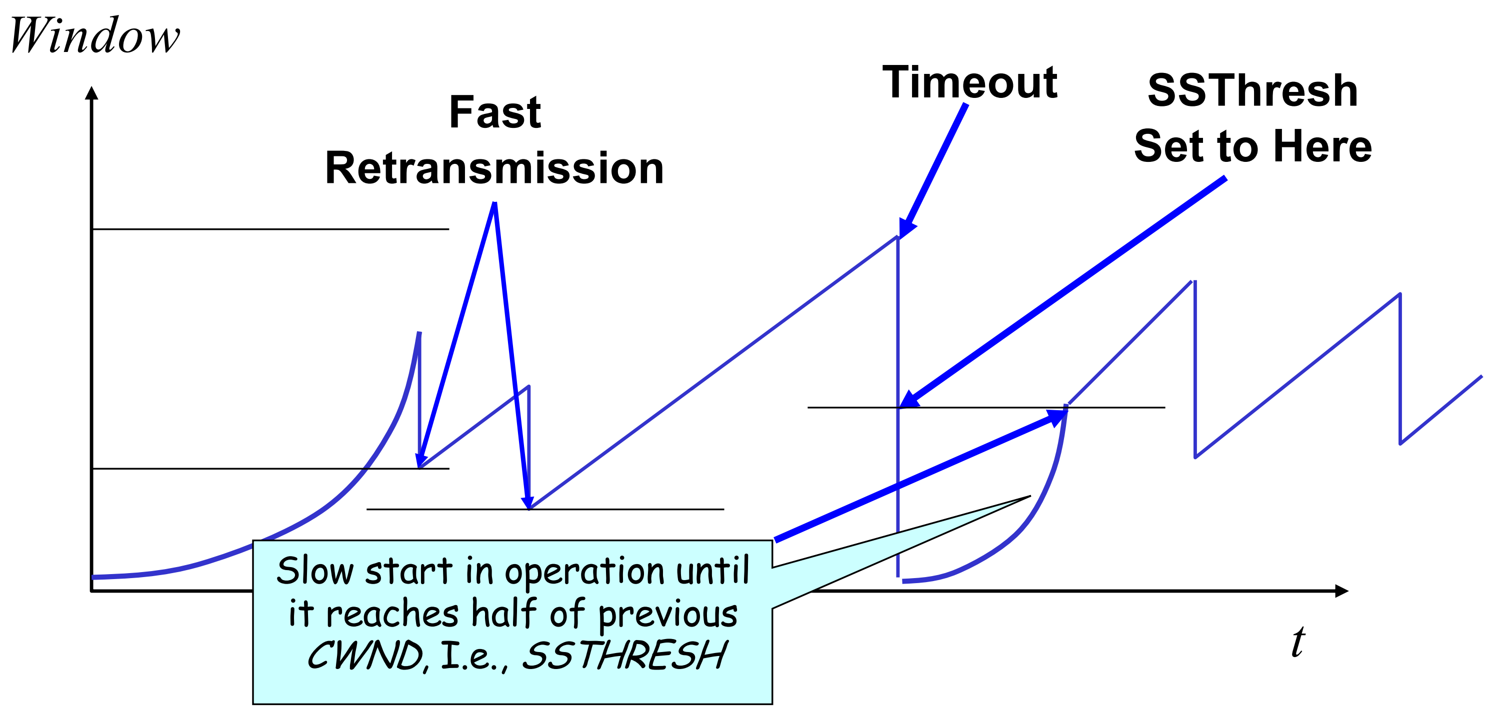
- dupACKcount ++

- If dupACKcount = 3 /* fast retransmit */
  - ssthresh = CWND/2
  - CWND = CWND/2

Event: TimeOut

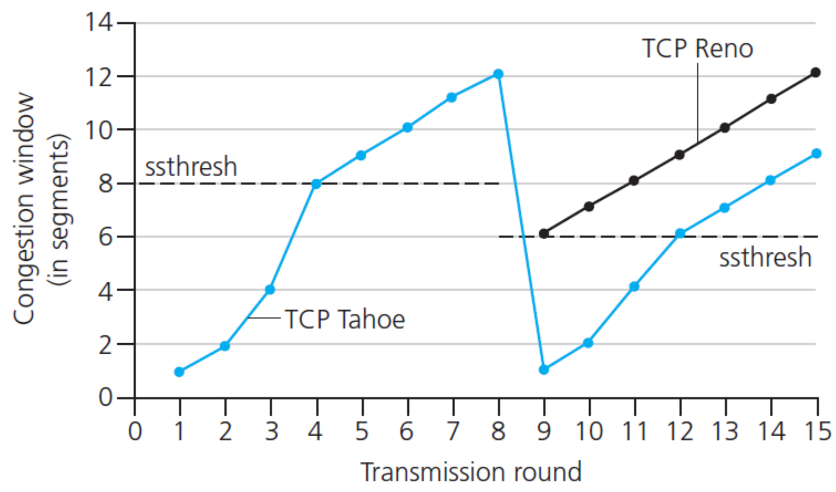
- On Timeout
  - ssthresh $\leftarrow$ CWND/2
  - CWND $\leftarrow$ 1

Example



Slow-start restart: Go back to CWND = 1 MSS, but take advantage of knowing the previous value of CWND

TCP Flavours

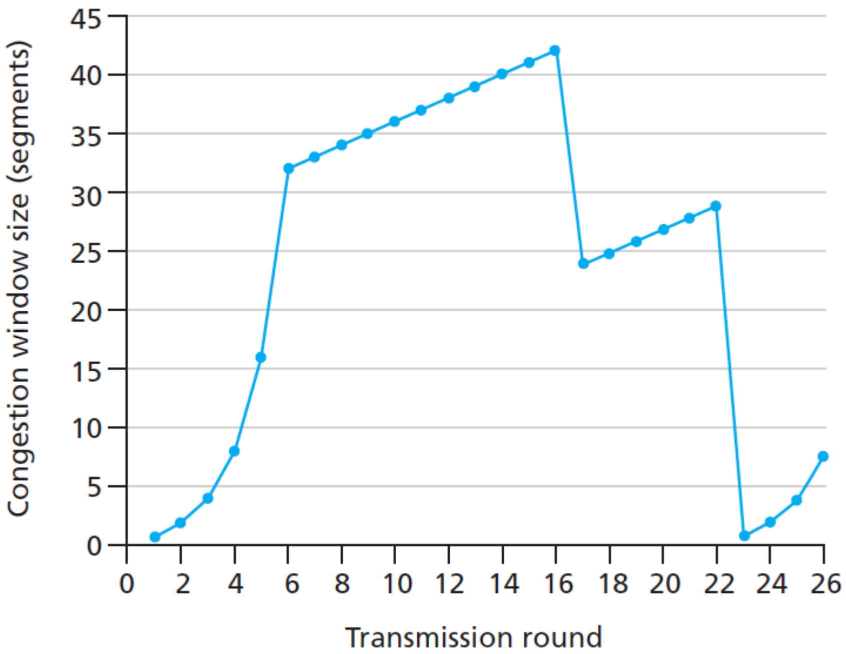


- TCP-Tahoe
  - cwnd =1 on triple dup ACK & timeout
- TCP-Reno
  - cwnd =1 on timeout
  - cwnd = cwnd/2 on triple dup ACK
- TCP-newReno
  - TCP-Reno + improved fast recovery (SKIPPED AND NOT ON EXAM)
- TCP-SACK (NOT COVERED IN THE COURSE)
  - incorporates selective acknowledgements

Quiz: TCP Congestion Control?

In the figure how many congestion avoidance intervals can you identify?

A. 0
B. 1
C. 2
D. 3
E. 4



Answer: C
6 to 16 ,17 to 22

Note: the transition at round 17 is not entirely acurrate, the window should reduce to 21 (currently 24)

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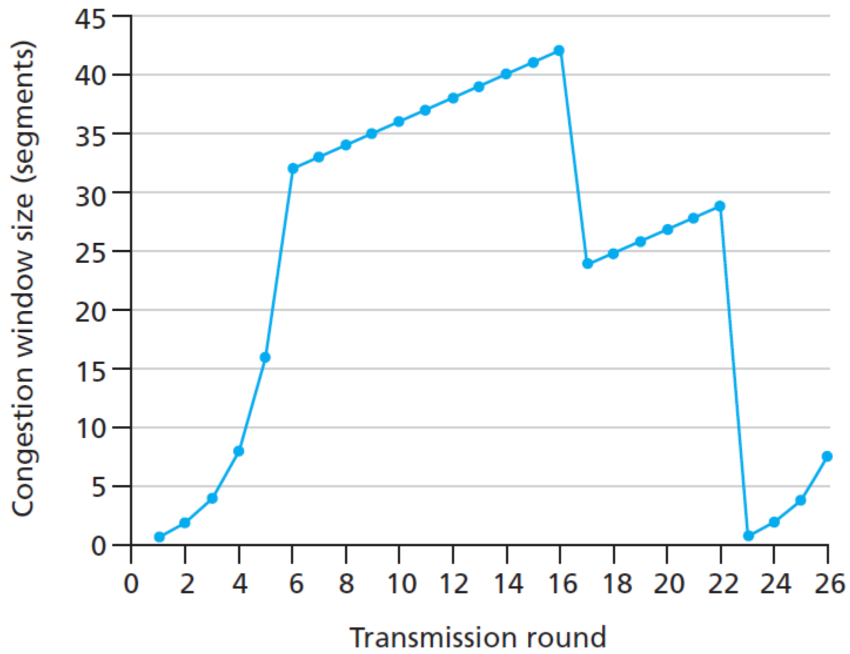
Quiz: TCP Congestion Control?

In the figure how many slow start intervals can you identify?

A. 0
B. 1
C. 2
D. 3
E. 4



Answer: C
Round 1 – 6, and Round 23-26

Note: the transition at round 17 is not entirely acurrate, the window should reduce to 21 (currently 24)

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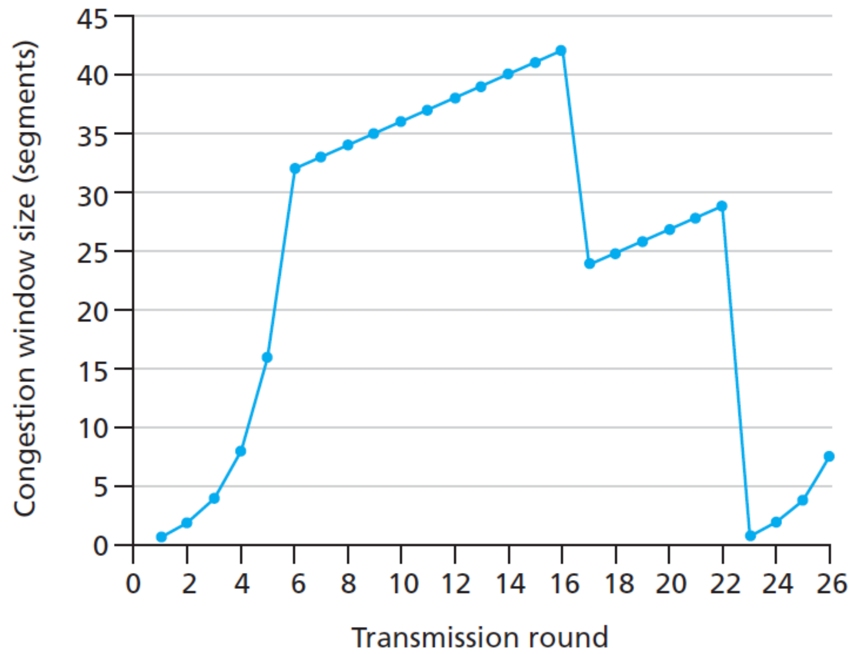
Quiz: TCP Congestion Control?

In the figure after the 16th transmission round, segment loss is detected by ________ ?

A. Triple Dup Ack

B. Timeout



Answer: A as the window is cut to half the previous value

Note: the transition at round 17 is not entirely acurrate, the window should reduce to 21 (currently 24)

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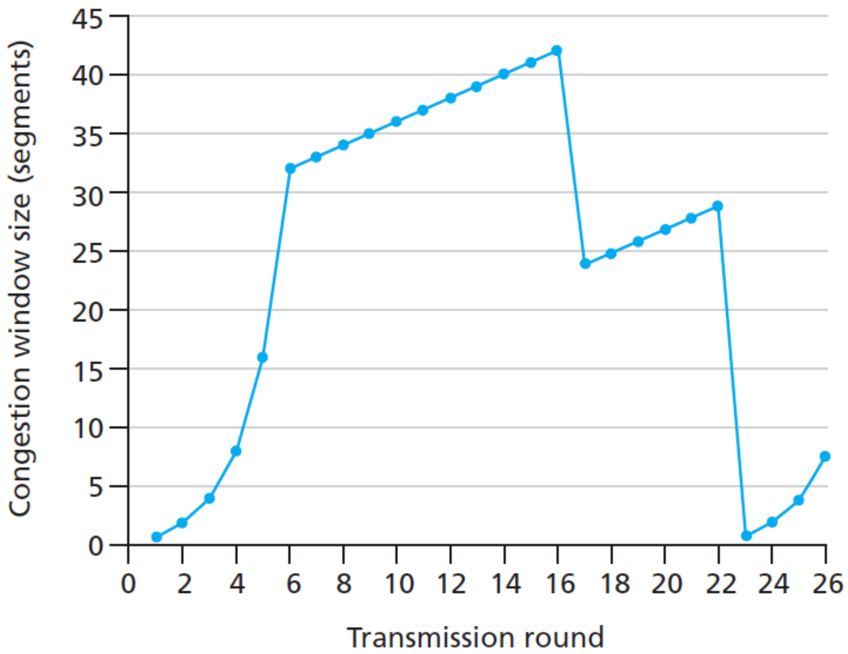
Quiz: TCP Congestion Control?

In the figure what is the initial value of sstresh (steady state threshold)?

A. 0
B. 28
C. 32
D. 42
E. 64



Answer: C (In Round 6, there is a transition from slow start to Congestion avoidance when the window is equal to 32 (sstresh)

Note: the transition at round 17 is not entirely acurrate, the window should reduce to 21 (currently 24)

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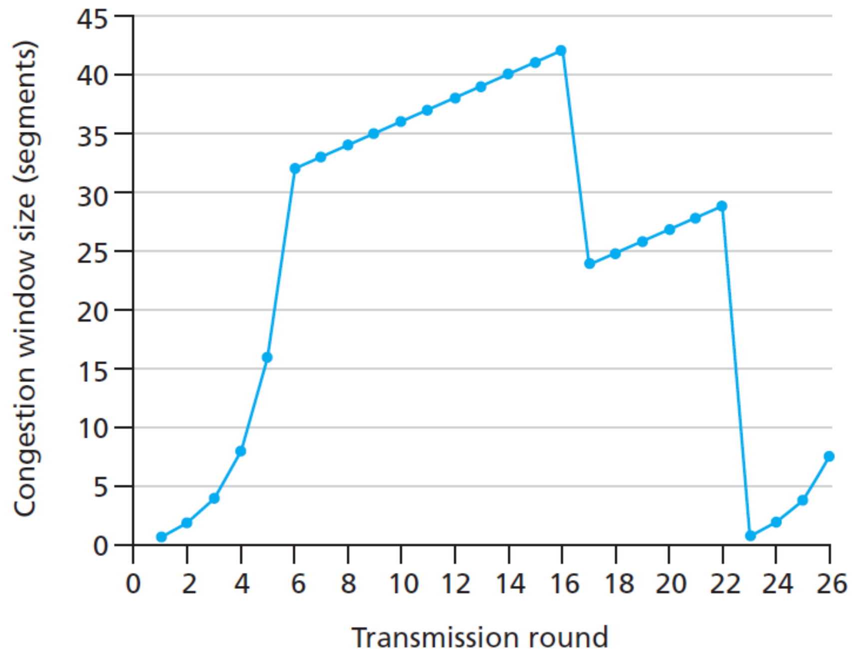
Quiz: TCP Congestion Control?

In the figure what is the value of sstresh (steady state threshold) at the 18^{th} round?

A. 1
B. 32
C. 42
D. 21
E. 20



Answer: D
(sstresh is set to 21 when a triple dup ack event is encountered in the 16^{th} round)

Note: the transition at round 17 is not entirely acurrate, the window should reduce to 21 (currently 24)

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NOT ON EXAM

Evolving transport-layer functionality

- TCP, UDP: principal transport protocols for 40 years
- different "flavors" of TCP developed, for specific scenarios:

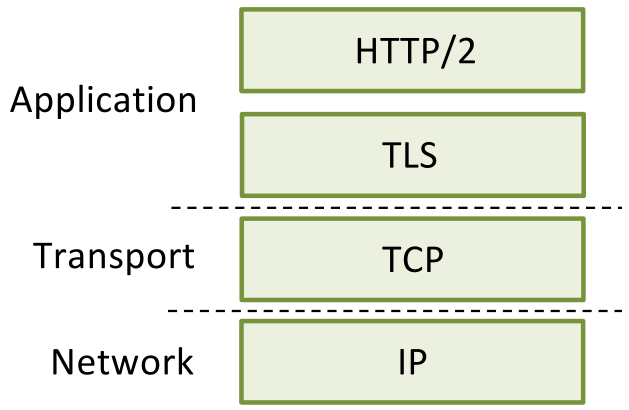
Scenario	Challenges
Long, fat pipes (large data transfers)	Many packets "in flight"; loss shuts down pipeline
Wireless networks	Loss due to noisy wireless links, mobility; TCP treat this as congestion loss
Long-delay links	Extremely long RTTs
Data center networks	Latency sensitive
Background traffic flows	Low priority, "background" TCP flows

- moving transport–layer functions to application layer, on top of UDP
  - HTTP/3: QUIC

NOT ON EXAM

QUIC: Quick UDP Internet Connections

- application-layer protocol, on top of UDP
  - increase performance of HTTP
  - deployed on many Google servers, apps (Chrome, mobile YouTube app)



HTTP/2 over TCP

NOT ON EXAM

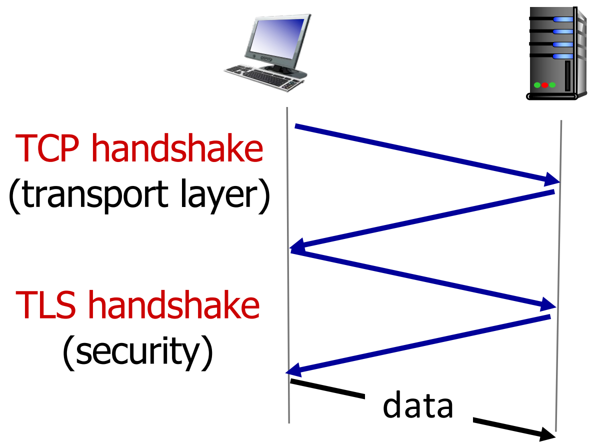
QUIC: Quick UDP Internet Connections

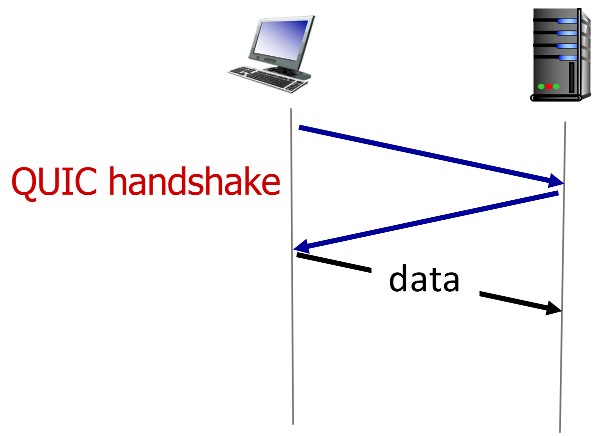
adopts approaches we've studied in this module for connection establishment, error control, congestion control

- **error and congestion control:** "Readers familiar with TCP's loss detection and congestion control will find algorithms here that parallel well-known TCP ones." [from QUIC specification]
- **connection establishment:** reliability, congestion control, authentication, encryption, state established in one RTT

- multiple application-level "streams" multiplexed over single QUIC connection
  - separate reliable data transfer, security
  - common congestion control

QUIC: Connection establishment





TCP (reliability, congestion control state) + TLS (authentication, crypto state)

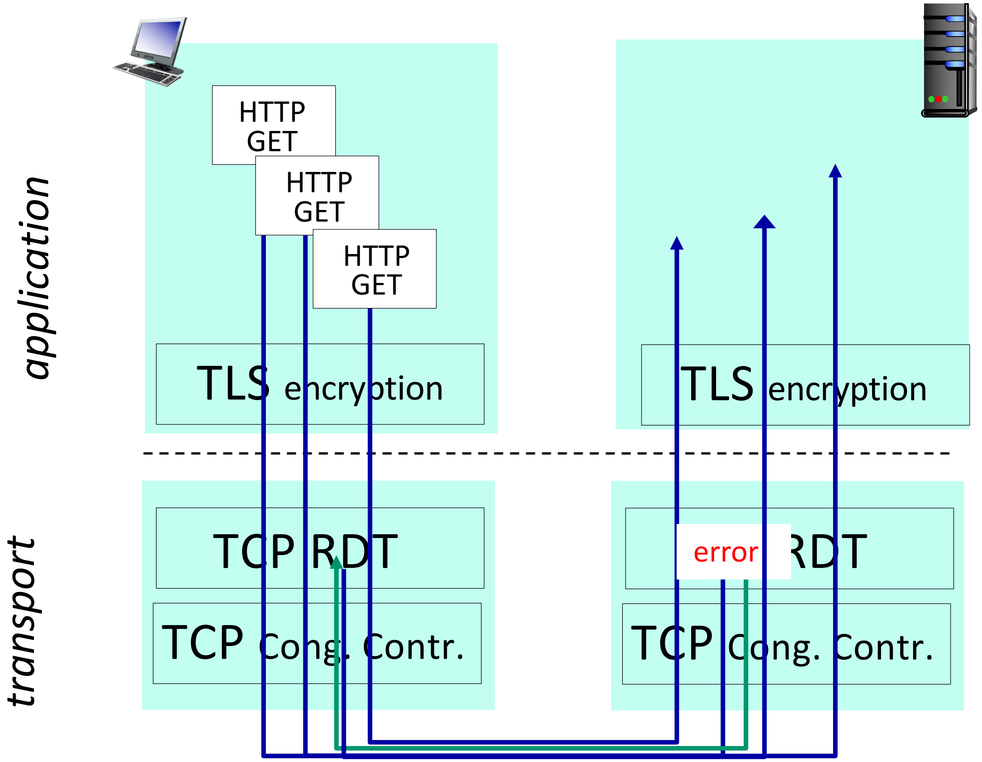
- 2 serial handshakes

QUIC: reliability, congestion control, authentication, crypto state

- 1 handshake

NOT ON EXAM

QUIC: streams: parallelism, no HOL blocking



(a) HTTP 1.1

Transport Layer: Summary

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

<u>next:</u>

- leaving the network "edge" (application, transport layers)
- into the network "core"