Master Course
Computer Networks
IN2097

Prof. Dr.-Ing. Georg Carle
Christian Grothoff, Ph.D.
Lecturer today: Dr. Nils Kammenhuber

Chair for Network Architectures and Services
Institut für Informatik
Technische Universität München
http://www.net.in.tum.de

Chapter 3: Transport Layer

Our goals:

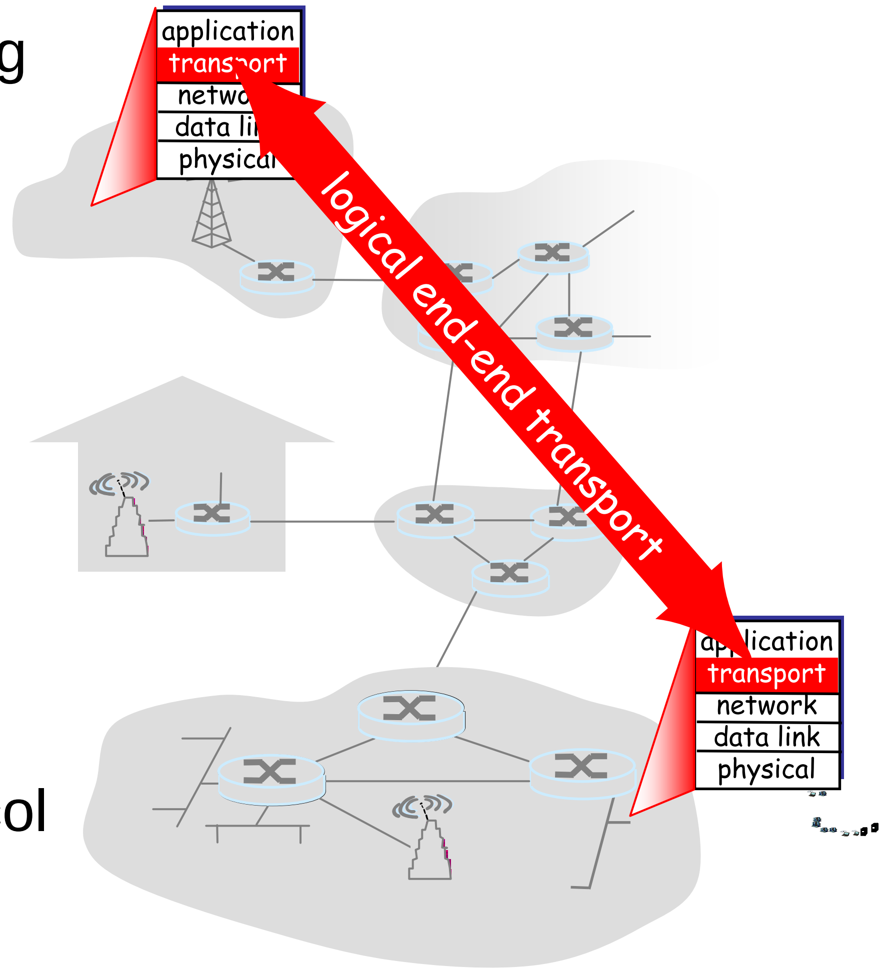
- understand principles behind transport layer services:
  - multiplexing/demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- learn about transport layer protocols in the Internet:
  - UDP: connectionless transport
  - TCP: connection-oriented transport
  - TCP congestion control

Chapter 3 outline

- **3.1 Transport-layer services**
- 3.2 Multiplexing and demultiplexing
- 3.3 Connectionless transport: UDP
- 3.4 largely omitted
- 3.5 Connection-oriented transport: TCP
  - segment structure
  - reliable data transfer
  - flow control
  - connection management
- 3.6 largely omitted
- 3.7 TCP congestion control

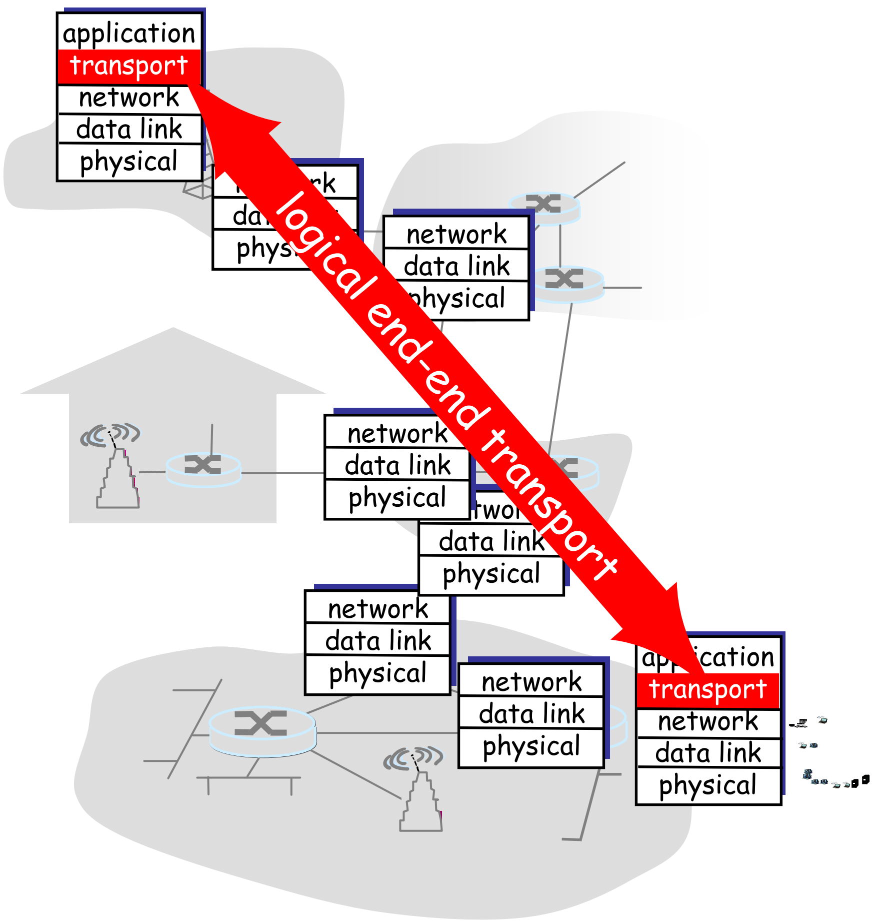
Transport services and protocols

- provide *logical communication* between app processes running on different hosts
- transport protocols run in end systems
  - send side: breaks app messages into segments, passes to network layer
  - rcv side: reassembles segments into messages, passes to app layer
- more than one transport protocol available to apps
  - Internet: TCP and UDP



Internet transport-layer protocols

- reliable, in-order delivery (TCP)
  - congestion control
  - flow control
  - connection setup
- unreliable, unordered delivery: UDP
  - no-frills extension of "best-effort" IP
- services not available:
  - delay guarantees
  - bandwidth guarantees



Chapter 3 outline

- 3.1 Transport-layer services
- **3.2 Multiplexing and demultiplexing**
- 3.3 Connectionless transport: UDP
- -
- 3.5 Connection-oriented transport: TCP
  - segment structure
  - reliable data transfer
  - flow control
  - connection management
- -
- 3.7 TCP congestion control

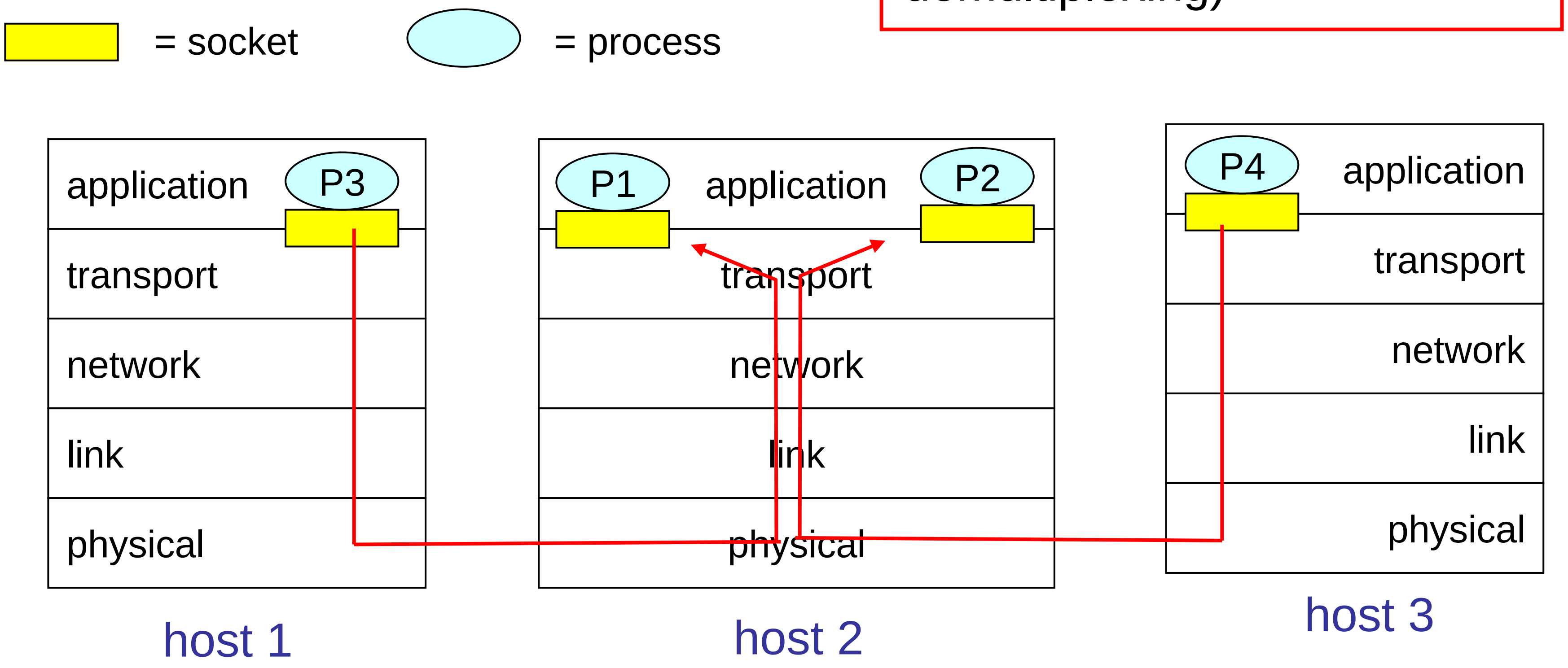
Multiplexing/demultiplexing

Demultiplexing at rcv host:

delivering received segments to correct socket

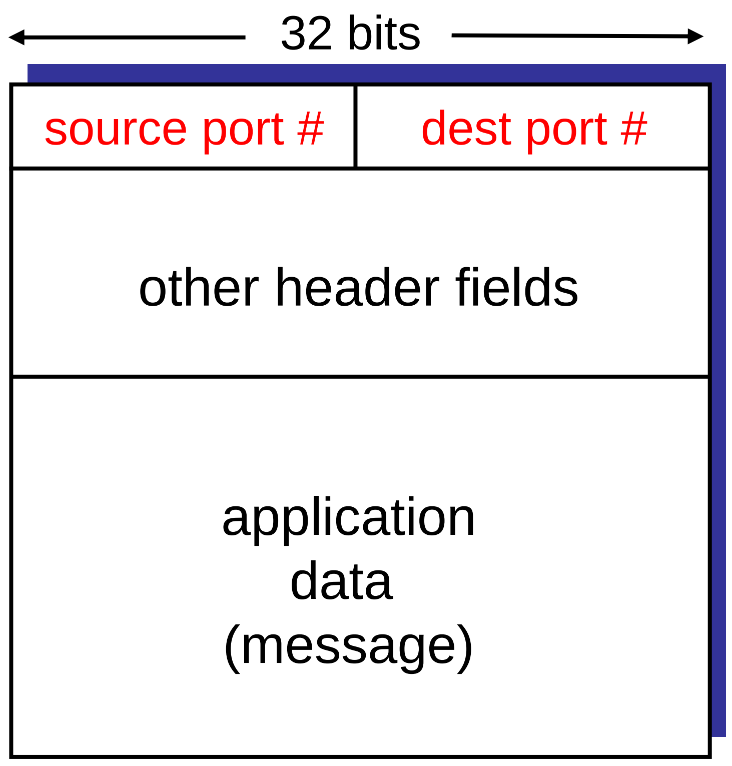
Multiplexing at send host:

gathering data from multiple sockets, enveloping data with header (later used for demultiplexing)



How demultiplexing works

- host receives IP datagrams
  - each datagram has source IP address, destination IP address
  - each datagram carries 1 transport-layer segment
  - each segment has source, destination port number
- host uses IP addresses & port numbers to direct segment to appropriate socket



TCP/UDP segment format

Connectionless demultiplexing

- Create sockets with port numbers:

```
DatagramSocket mySocket1 = new DatagramSocket(12534);
DatagramSocket mySocket2 = new DatagramSocket(12535);
```

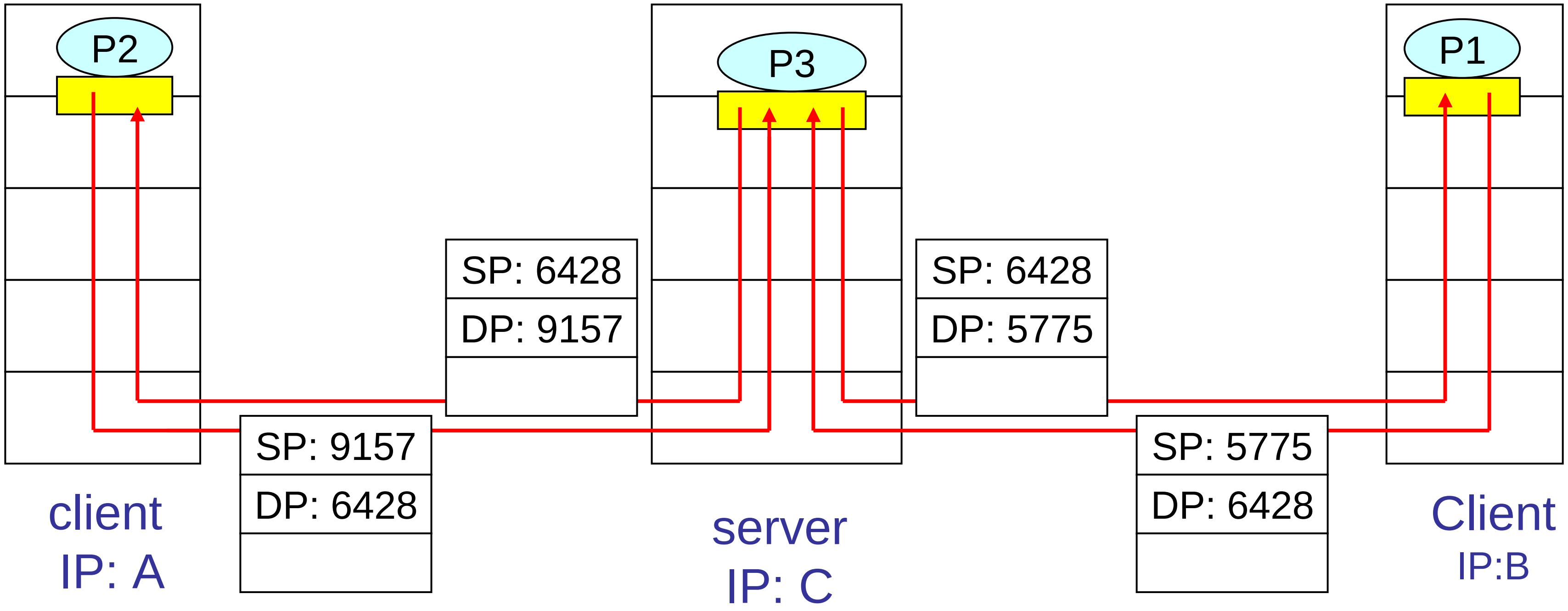
- UDP socket identified by two-tuple:

  (dest IP address, dest port number)

- When host receives UDP segment:
  - checks destination port number in segment
  - directs UDP segment to socket with that port number
- IP datagrams with different source IP addresses and/or source port numbers directed to same socket

Connectionless demux (cont)

```
DatagramSocket serverSocket = new DatagramSocket(6428);
```

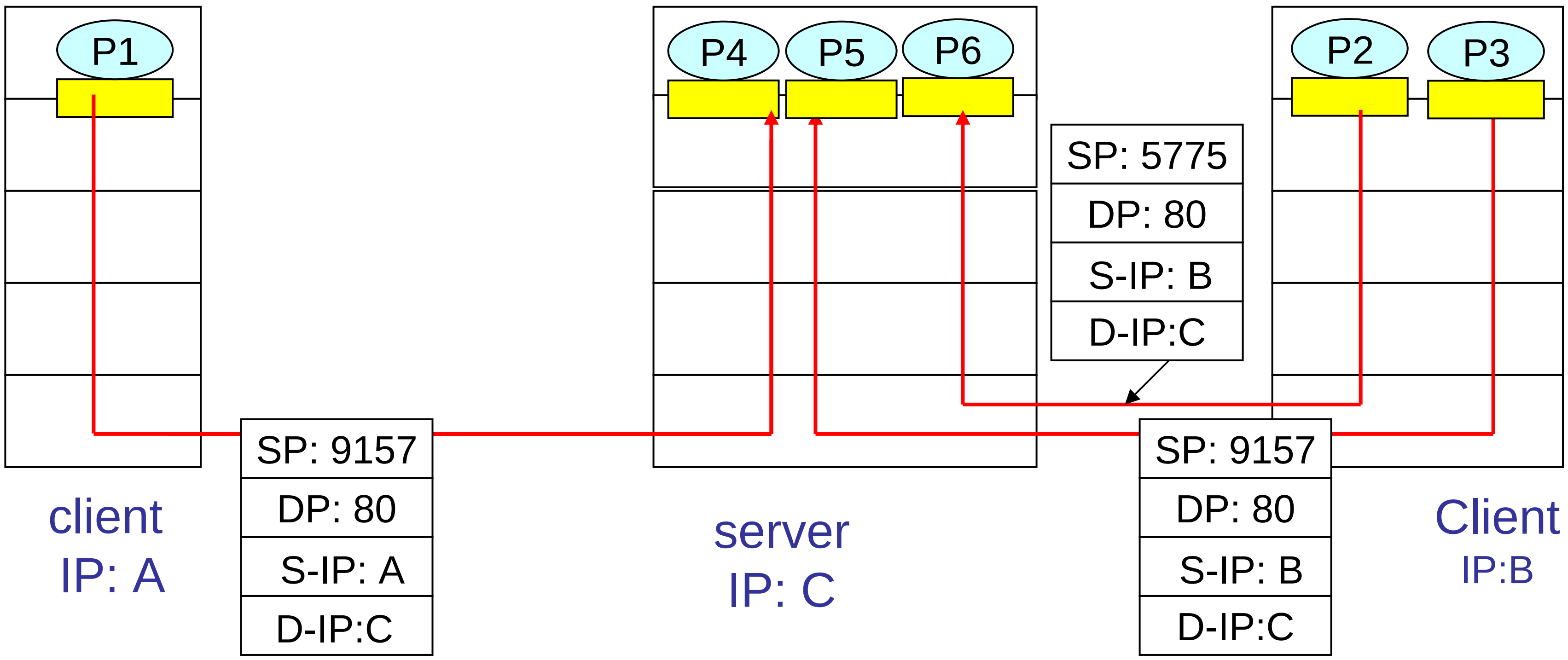


Source Port (SP) provides “return address”

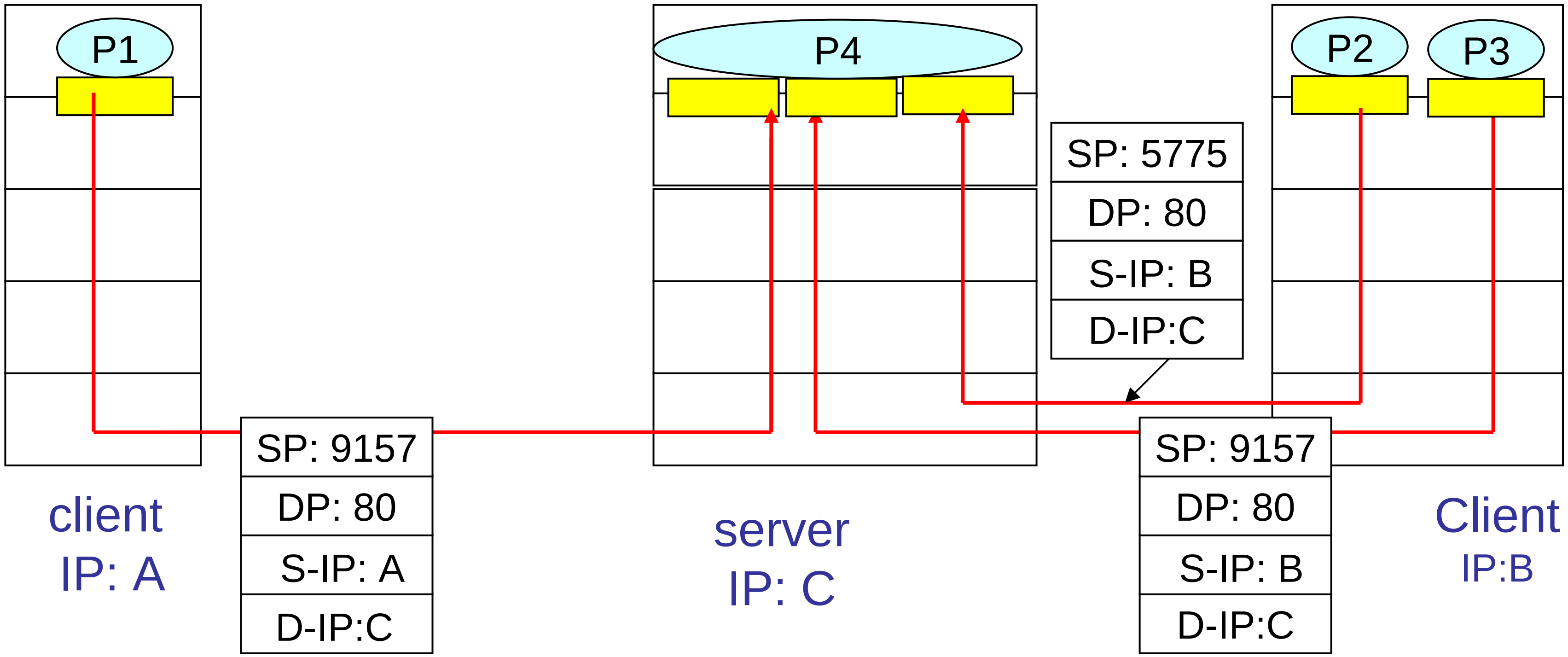
Connection-oriented demux

- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- recv host uses all four values to direct segment to appropriate socket
- Server host may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
- Web servers have different sockets for each connecting client
  - non-persistent HTTP will have different socket for each request

Connection-oriented demux (cont)



Connection-oriented demux: Threaded Web Server



Chapter 3 outline

- 3.1 Transport-layer services
- 3.2 Multiplexing and demultiplexing
- **3.3 Connectionless transport: UDP**
- -
- 3.5 Connection-oriented transport: TCP
  - segment structure
  - reliable data transfer
  - flow control
  - connection management
- -
- 3.7 TCP congestion control

UDP: User Datagram Protocol [RFC 768]

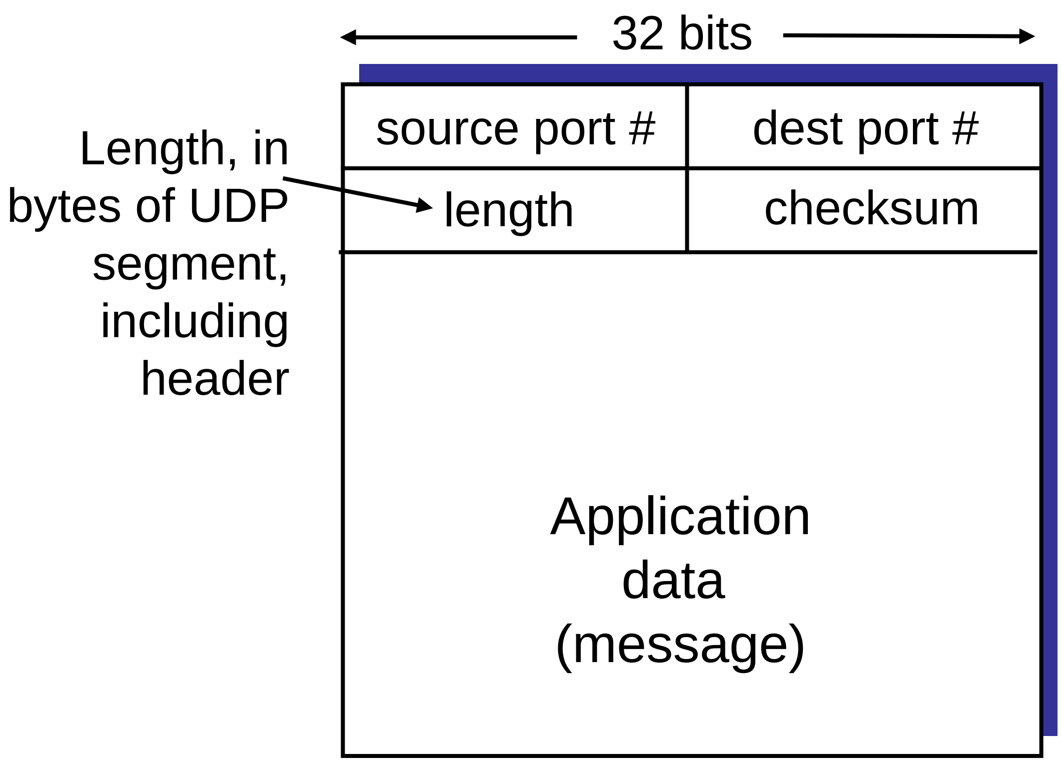
- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out of order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

Why is there a UDP?

- no connection establishment (which can add delay)
- simple: no connection state at sender, receiver
- small segment header
- No congestion control: UDP can blast away as fast as desired

UDP: more

- often used for streaming multimedia apps
  - loss tolerant
  - rate sensitive
- other UDP uses
  - DNS
  - SNMP
- reliable transfer over UDP: add reliability at application layer
  - application-specific error recovery!



UDP segment format

UDP checksum

Goal: detect "errors" (e.g., flipped bits) in transmitted segment

Sender:

- treat segment contents as sequence of 16-bit integers
- checksum: addition (1's complement sum) of segment contents
- sender puts checksum value into UDP checksum field

Receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - NO - error detected
  - YES - no error detected. *But maybe errors nonetheless?* More later ….

Internet Checksum Example

- Note
  - When adding numbers, a carryout from the most significant bit needs to be added to the result
- Example: add two 16-bit integers

		1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0
		1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
sum		1	0	1	1	1	0	1	1	1	0	1	1	1	1	0	0
checksum		0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	1

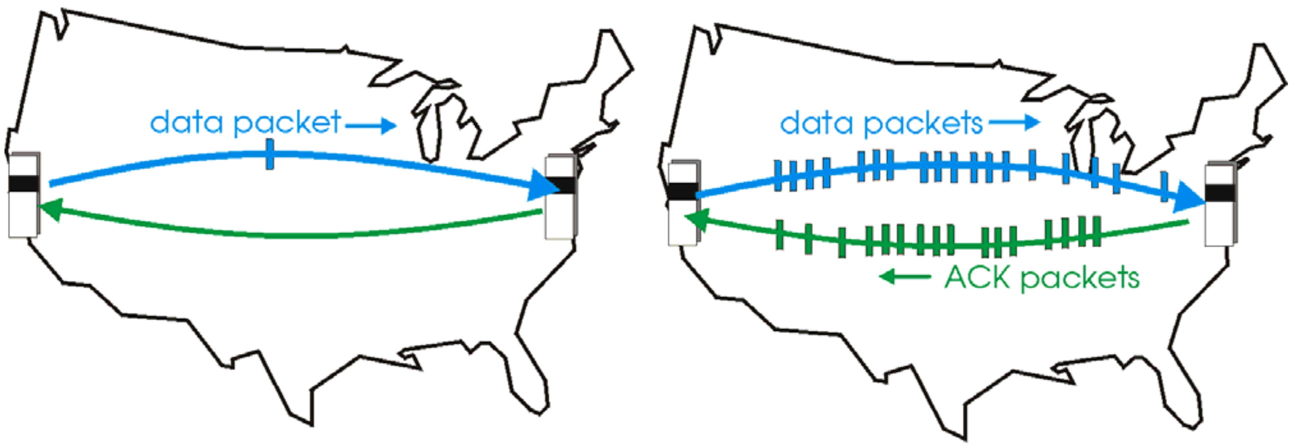
Chapter 3 outline

- 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

Pipelined protocols

Pipelining: sender allows multiple, “in-flight”, yet-to-be-acknowledged pkts

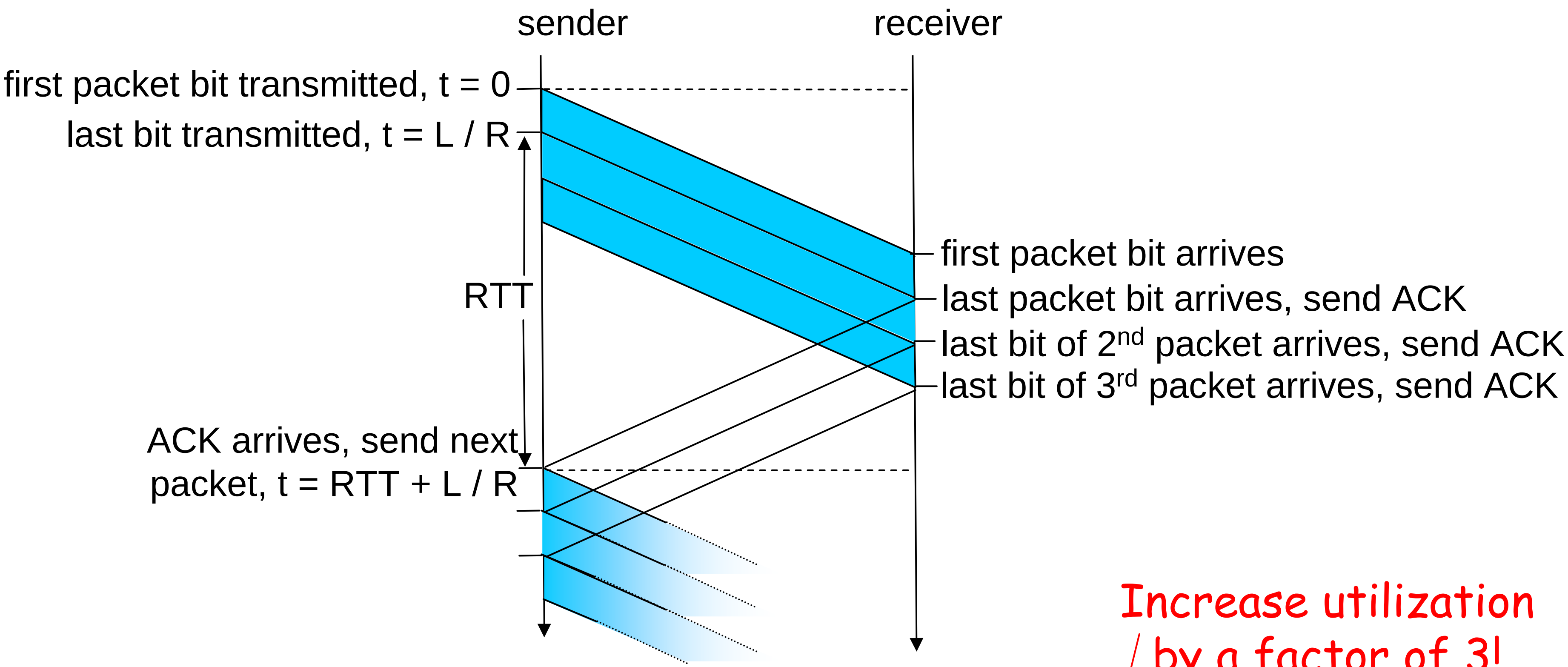
- range of sequence numbers must be increased
- buffering at sender and/or receiver



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

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

Pipelining: increased utilization



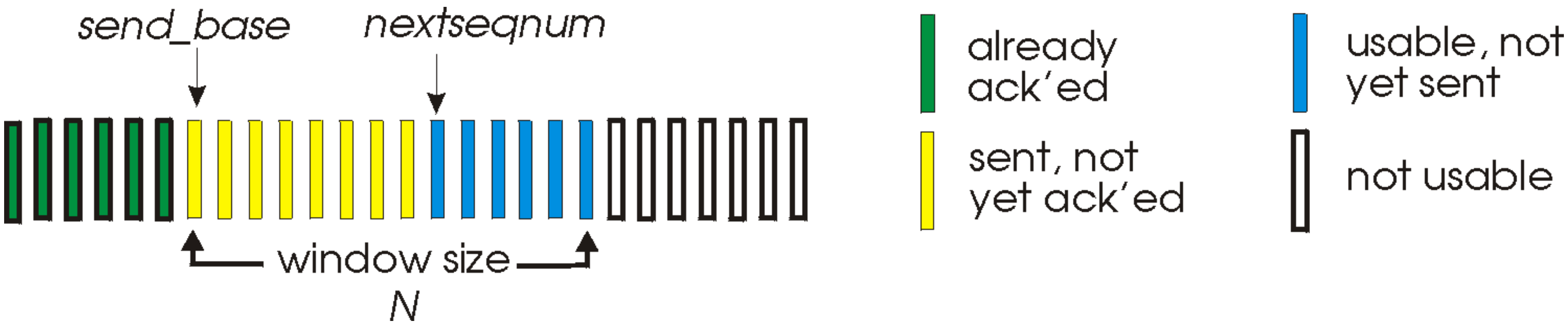
Increase utilization by a factor of 3!

$$U_{sender} = \frac{3 * L / R}{RTT + L / R} = \frac{.024}{30.008} = 0.0008$$

Go-Back-N

Sender:

- k-bit seq # in pkt header
- "window" of up to N, consecutive unack'ed pkts allowed



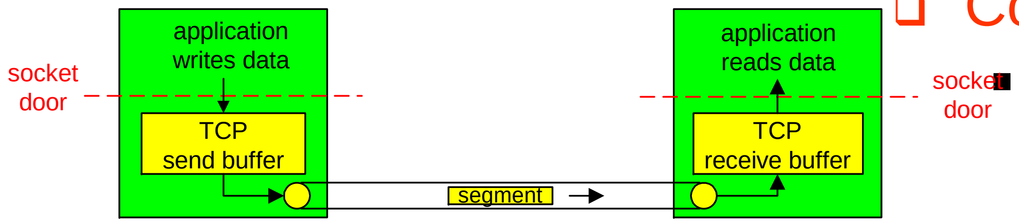
- ACK(n): ACKs all pkts up to, including seq # n - "cumulative ACK"
  - may receive duplicate ACKs (see receiver)
- timer for each in-flight pkt
- *timeout(n):* retransmit pkt n and all higher seq # pkts in window

Chapter 3 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

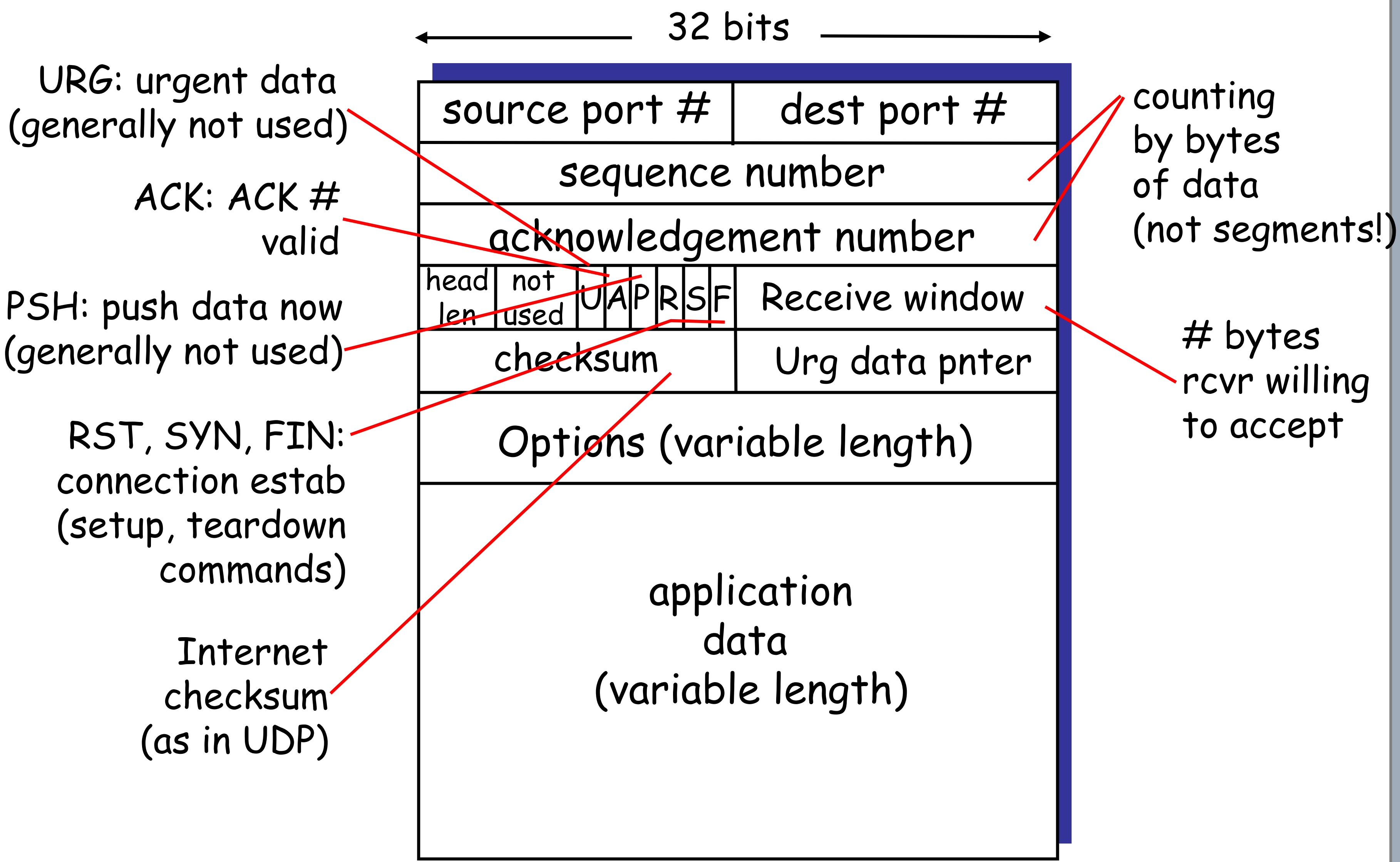
TCP: Overview RFCs: 793, 1122, 1323, 2018, 2581

- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte steam:*
  - no “message boundaries”
- pipelined:
  - TCP congestion and flow control set window size
- *send & receive buffers*



- full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
- connection-oriented:
  - handshaking (exchange of control msgs) init’s sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver
- Congestion controlled:
  - Will not overwhelm network

TCP segment structure



TCP seq. #'s and ACKs

Seq. #'s:

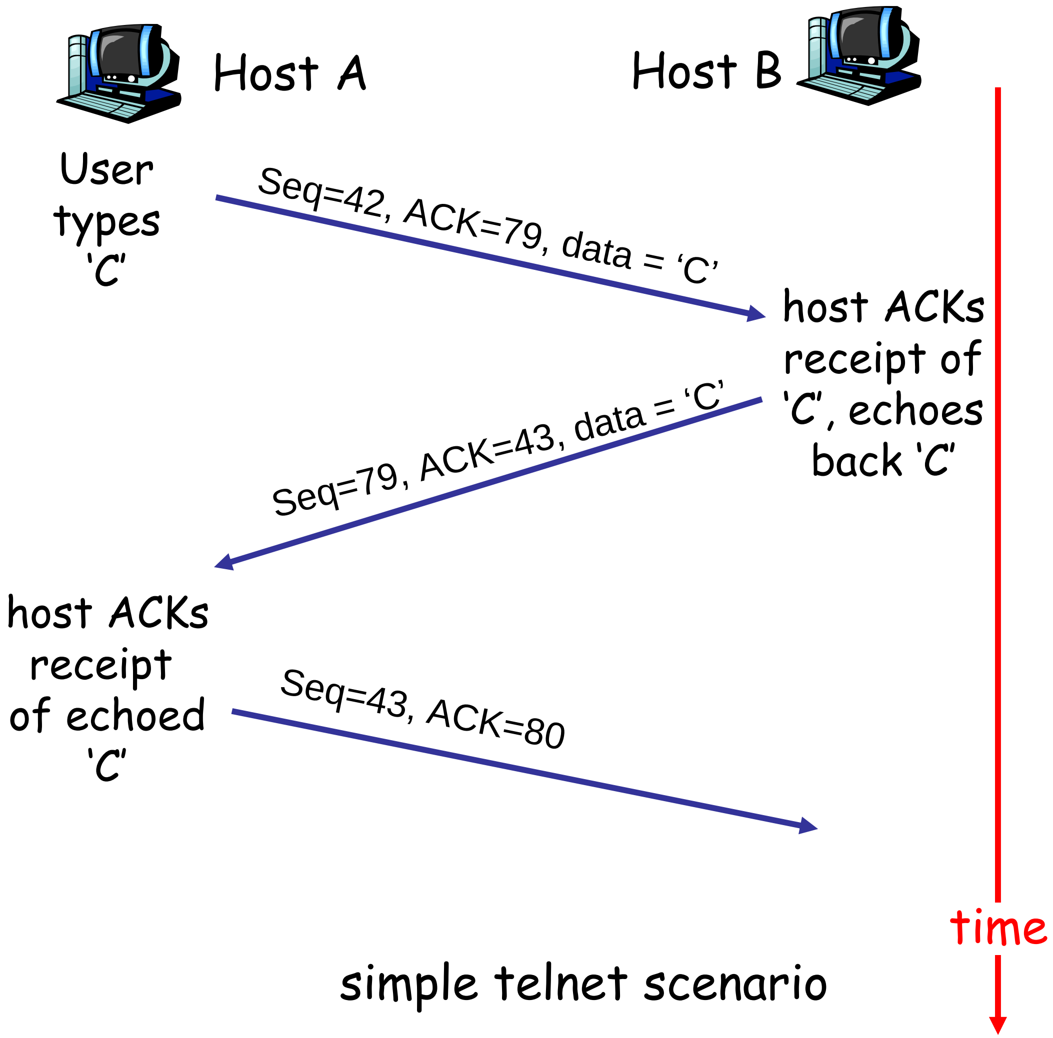
- byte stream "number" of first byte in segment's data

ACKs:

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

Q: how receiver handles out-of-order segments

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



simple telnet scenario

TCP Round Trip Time and 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

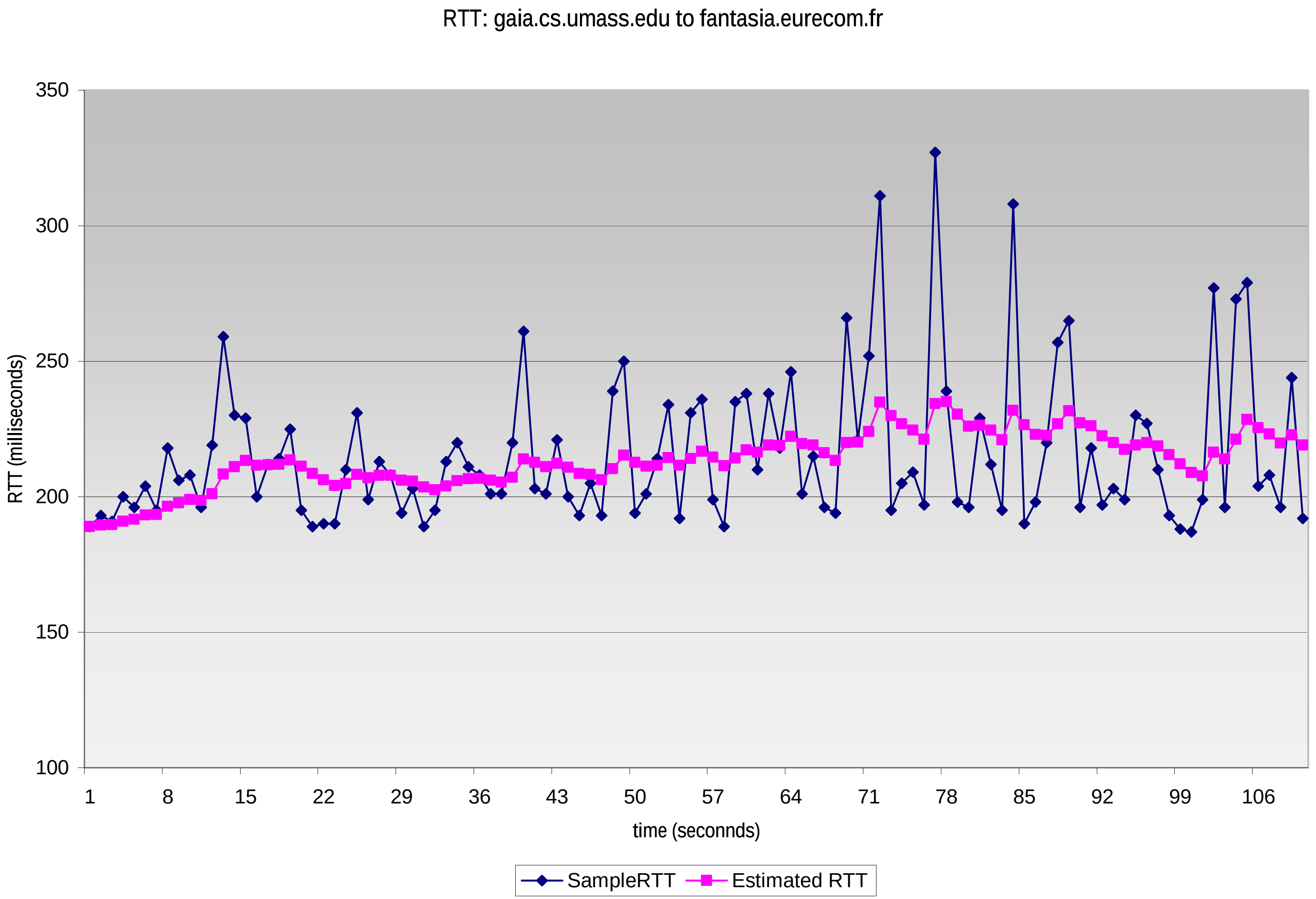
Q: how to estimate RTT?
- `SampleRTT`: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- `SampleRTT` will vary, want estimated RTT "smoother"
  - average several recent measurements, not just current `SampleRTT`

TCP Round Trip Time and Timeout

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

- Exponential weighted moving average
- influence of past sample decreases exponentially fast
- typical value: α = 0.125

Example RTT estimation:



TCP Round Trip Time and Timeout

Setting the timeout

- **`EstimtedRTT`** plus "safety margin"
  - large variation in **`EstimatedRTT ->`** larger safety margin
- first estimate of how much SampleRTT deviates from EstimatedRTT:

$$\texttt{DevRTT} = (1-\beta)*\texttt{DevRTT} + \beta*|\texttt{SampleRTT-EstimatedRTT}|$$

$(\texttt{typically, } \beta = 0.25)$

Then set timeout interval:

`TimeoutInterval = EstimatedRTT + 4*DevRTT`

Chapter 3 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 reliable data transfer

- TCP creates rdt service on top of IP's unreliable service
- Pipelined segments
- Cumulative acks
- TCP uses single retransmission timer

- Retransmissions are triggered by:
  - timeout events
  - duplicate acks
- Initially consider simplified TCP sender:
  - ignore duplicate acks
  - ignore flow control, congestion control

TCP sender events:

data rcvd from app:

- Create segment with seq #
- seq # is byte-stream number of first data byte in segment
- start timer if not already running (think of timer as for oldest unacked segment)
- expiration interval: `TimeOutInterval`

timeout:

- retransmit segment that caused timeout
- restart timer

Ack rcvd:

- If acknowledges previously unacked segments
  - update what is known to be acked
  - start timer if there are outstanding segments

TCP sender (simplified)

```
NextSeqNum = InitialSeqNum
SendBase = InitialSeqNum
 loop (forever) {
   switch(event)

   event: data received from application above
        create TCP segment with sequence number NextSeqNum
        if (timer currently not running)
             start timer
        pass segment to IP
        NextSeqNum = NextSeqNum + length(data)

    event: timer timeout
        retransmit not-yet-acknowledged segment with
               smallest sequence number
        start timer

    event: ACK received, with ACK field value of y
        if (y > SendBase) {
             SendBase = y
            if (there are currently not-yet-acknowledged segments)
                   start timer  }
  }  /* end of loop forever */
```

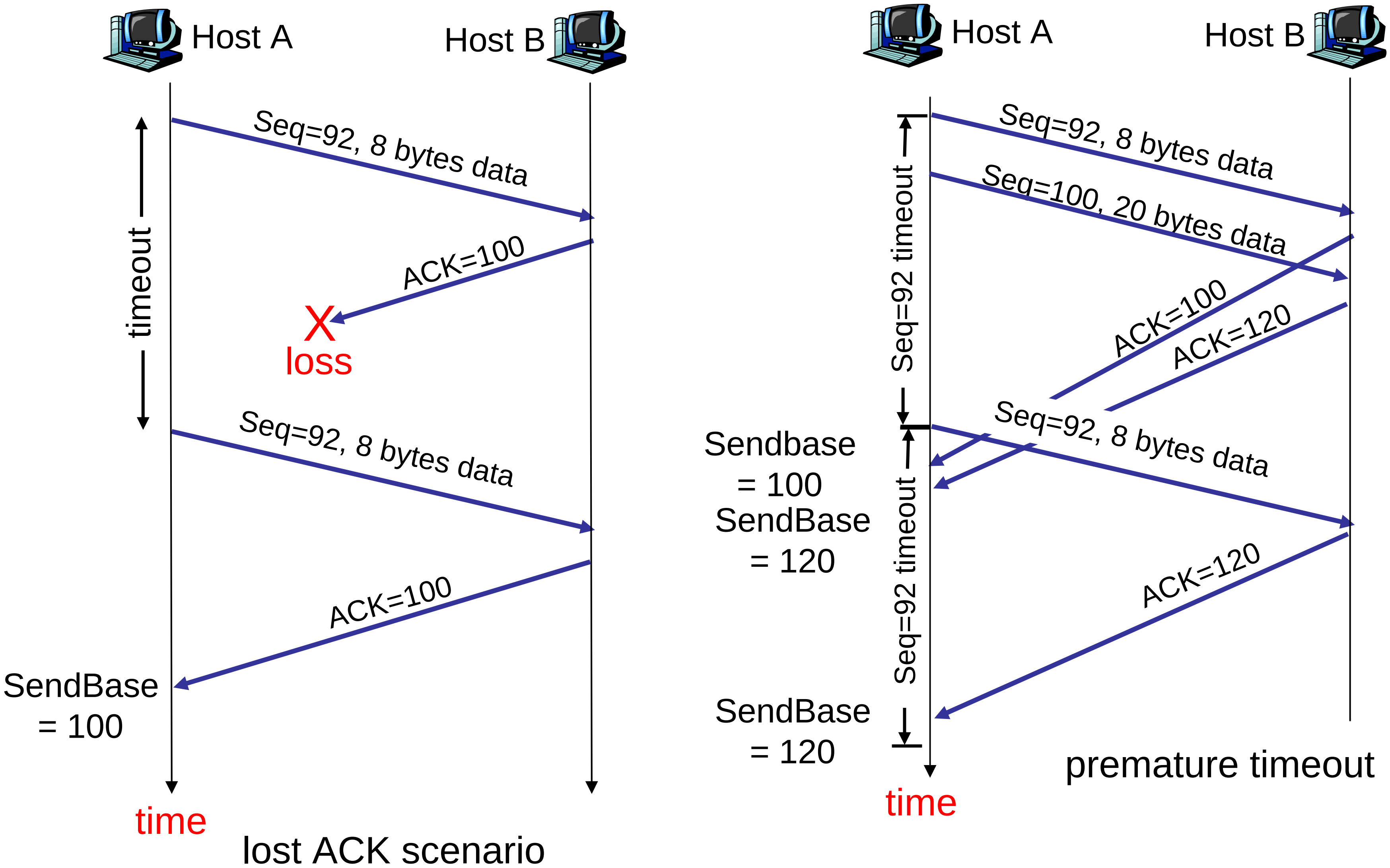
<u>Comment:</u>

- SendBase-1: last cumulatively ack'ed byte

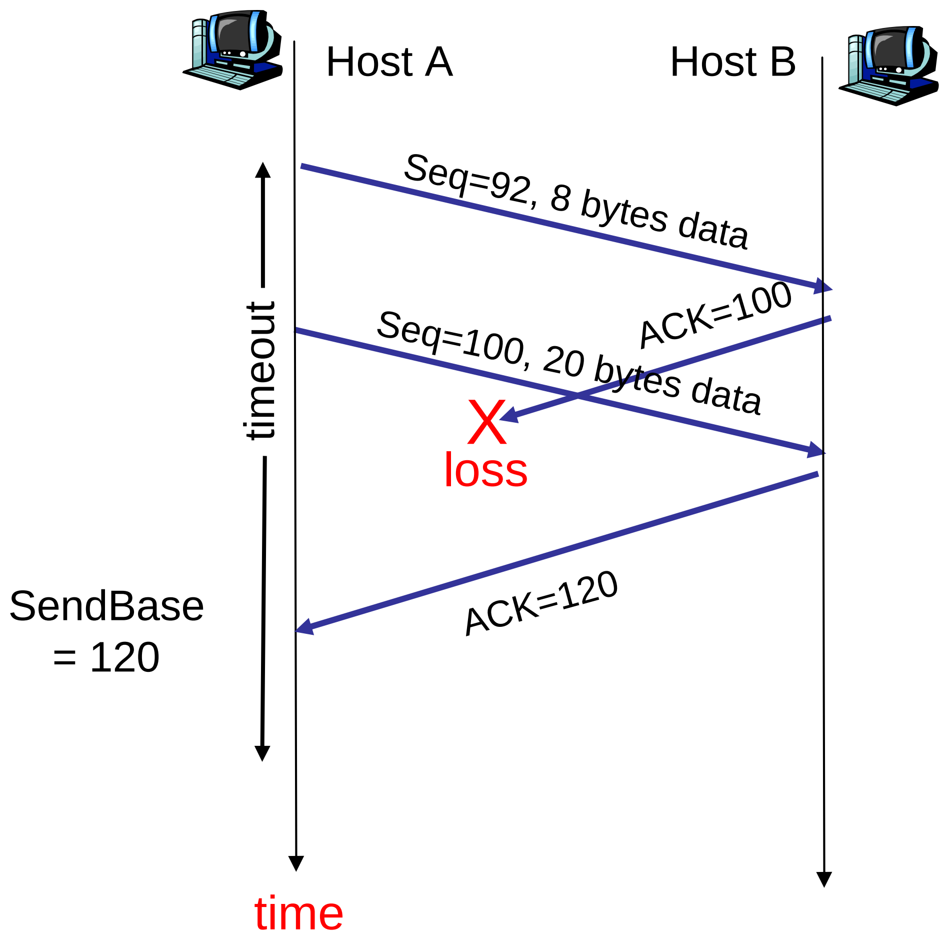
<u>Example:</u>

- SendBase-1 = 71; y= 73, so the rcvr wants 73+ ; y > SendBase, so that new data is acked

TCP: retransmission scenarios



TCP retransmission scenarios (more)



Cumulative ACK scenario

TCP ACK generation [RFC 1122, RFC 2581]

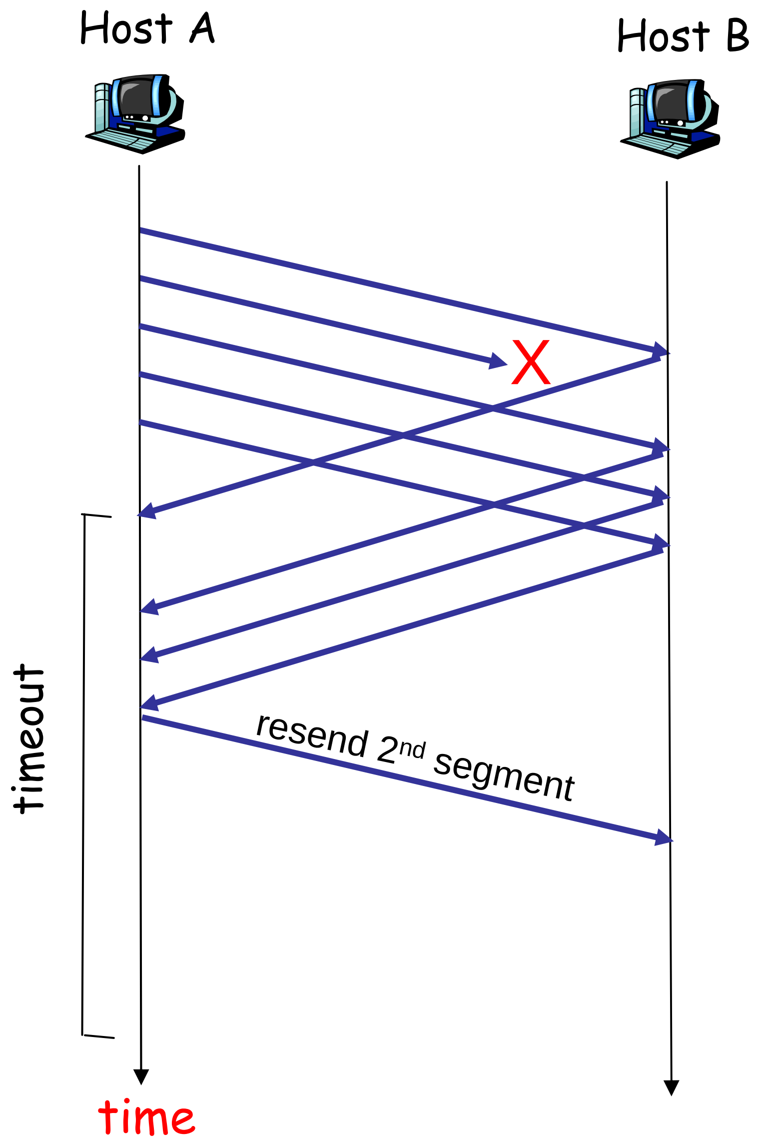
Event at Receiver	TCP Receiver action
Arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed	Delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK
Arrival of in-order segment with expected seq #. One other segment has ACK pending	Immediately 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

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.

- If sender receives 3 ACKs for the same data, it supposes that segment after ACKed data was lost:
  - fast retransmit: resend segment before timer expires

Resending a segment after triple duplicate ACK



Fast retransmit algorithm:

```
event: ACK received, with ACK field value of y
          if (y > SendBase) {
              SendBase = y
              if (there are currently not-yet-acknowledged segments)
                  start timer
            }
          else {
               increment count of dup ACKs received for y
               if (count of dup ACKs received for y = 3) {
                   resend segment with sequence number y
               }
```

a duplicate ACK for already ACKed segment

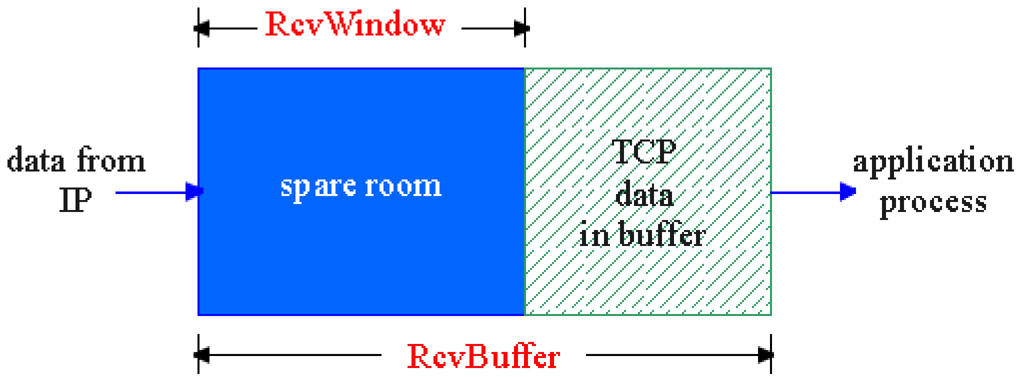
fast retransmit

Chapter 3 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 Flow Control

- receive side of TCP connection has a receive buffer:

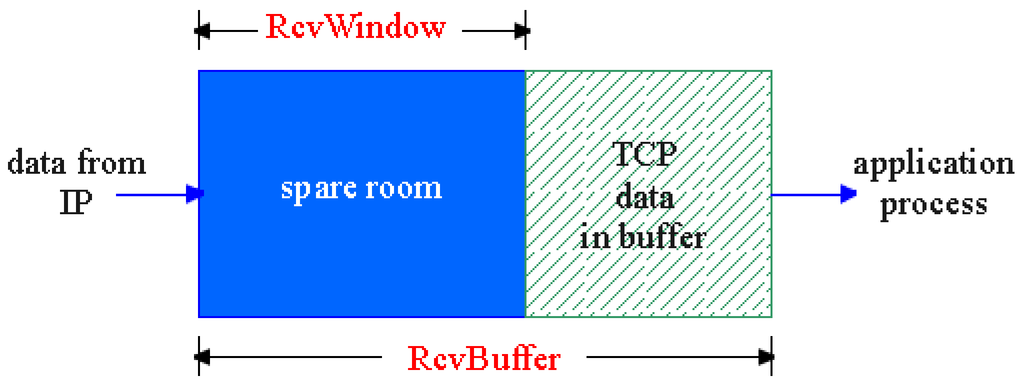


flow control

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

- app process may be slow at reading from buffer
- speed-matching service: matching the send rate to the receiving app's drain rate

TCP Flow control: how it works



(Suppose TCP receiver discards out-of-order segments)

- spare room in buffer

= **`RcvWindow`**

= **`RcvBuffer-[LastByteRcvd - LastByteRead]`**

- Rcvr advertises spare room by including value of **`RcvWindow`** in segments
- Sender limits unACKed data to **`RcvWindow`**
  - guarantees receive buffer doesn't overflow

Chapter 3 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 Connection Management

Recall: TCP sender, receiver establish "connection" before exchanging data segments

- initialize TCP variables:
  - seq. #s
  - buffers, flow control info (e.g. **`RcvWindow`**)
- *client:* connection initiator

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

- *server:* contacted by client

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

Three way handshake:

Step 1: client host sends TCP SYN segment to server

- specifies initial seq #
- no data

Step 2: server host receives SYN, replies with SYNACK segment

- server allocates buffers
- specifies server initial seq. #

Step 3: client receives SYNACK, replies with ACK segment, which may contain data

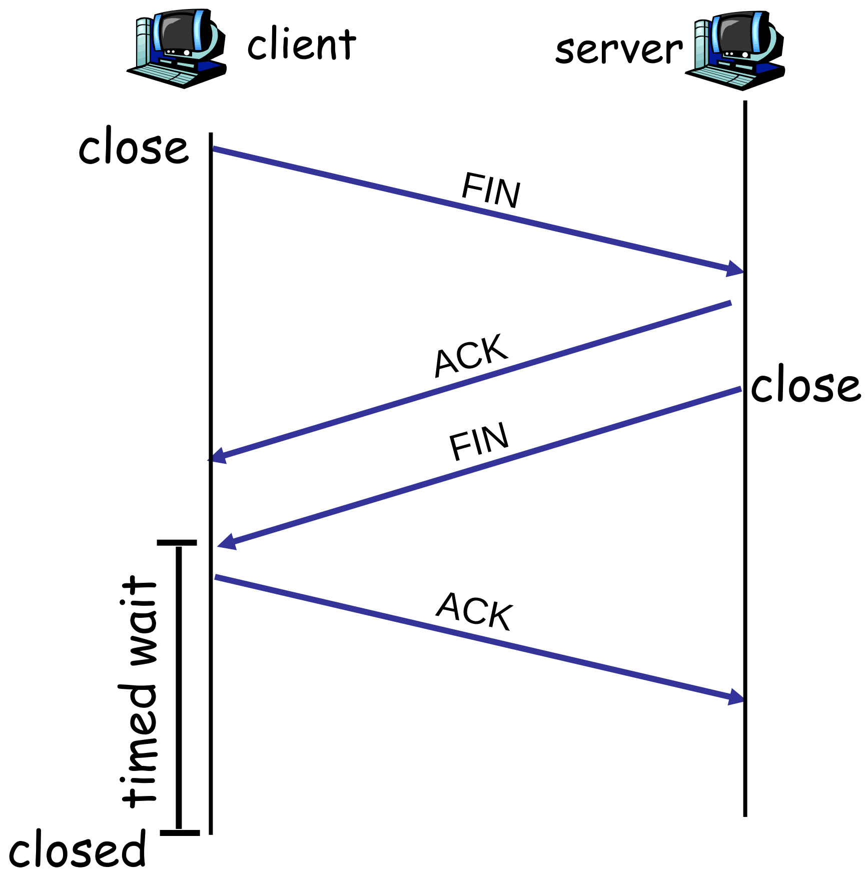
TCP Connection Management (cont.)

Closing a connection:

client closes socket:
`clientSocket.close();`

Step 1: client end system sends TCP FIN control segment to server

Step 2: server receives FIN, replies with ACK. Closes connection, sends FIN.



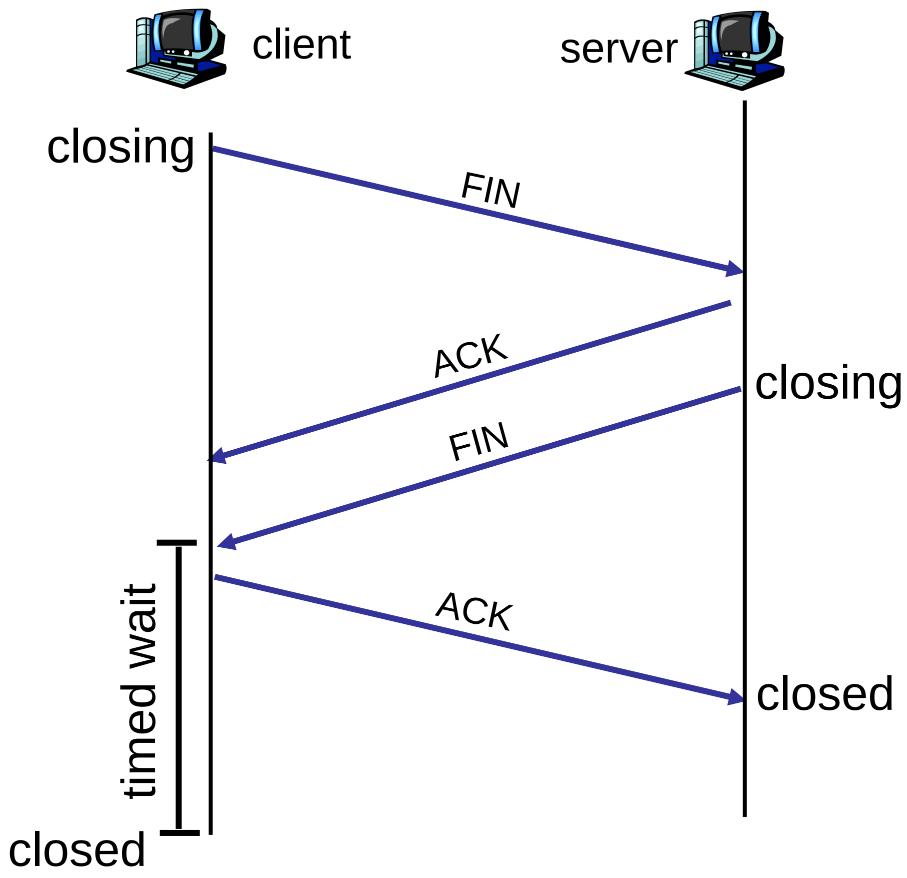
TCP Connection Management (cont.)

Step 3: client receives FIN, replies with ACK.

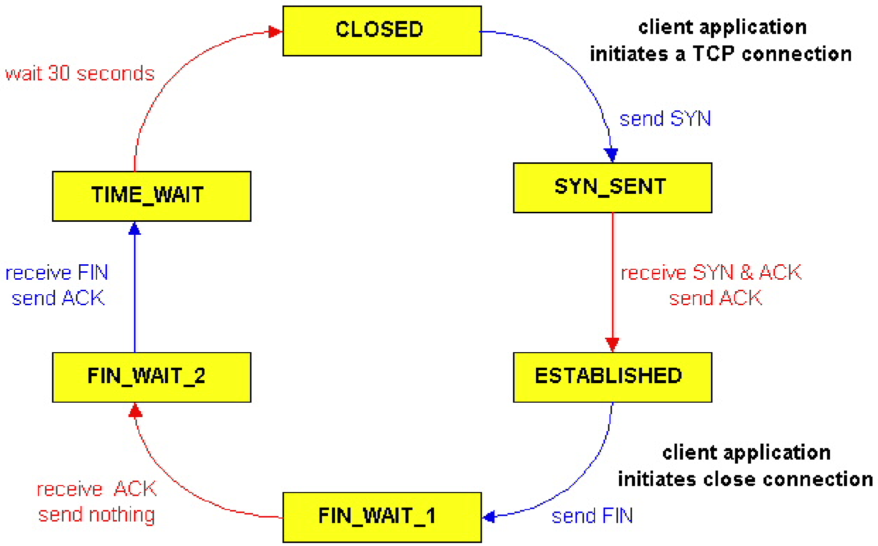
- Enters “timed wait” - will respond with ACK to received FINs

Step 4: server, receives ACK. Connection closed.

Note: with small modification, can handle simultaneous FINs.

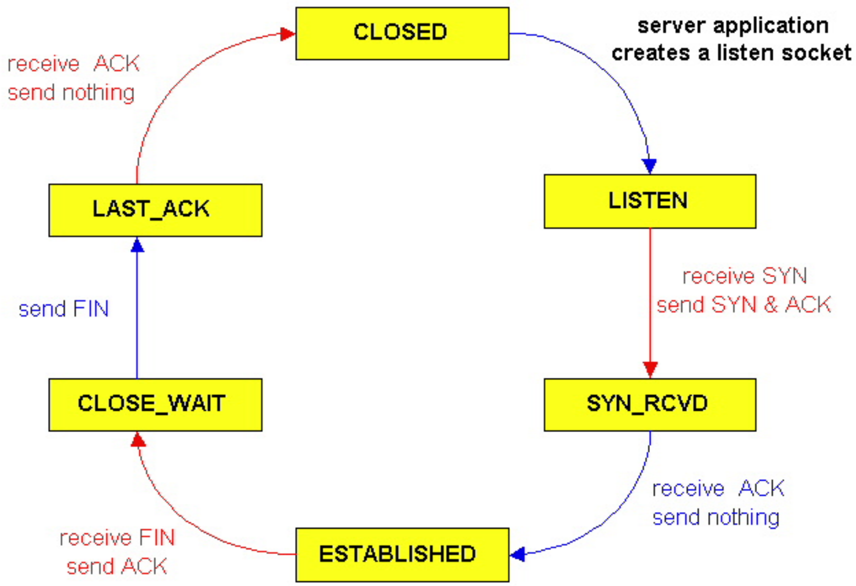


TCP Connection Management (cont)



TCP client lifecycle

TCP server lifecycle



Chapter 3 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

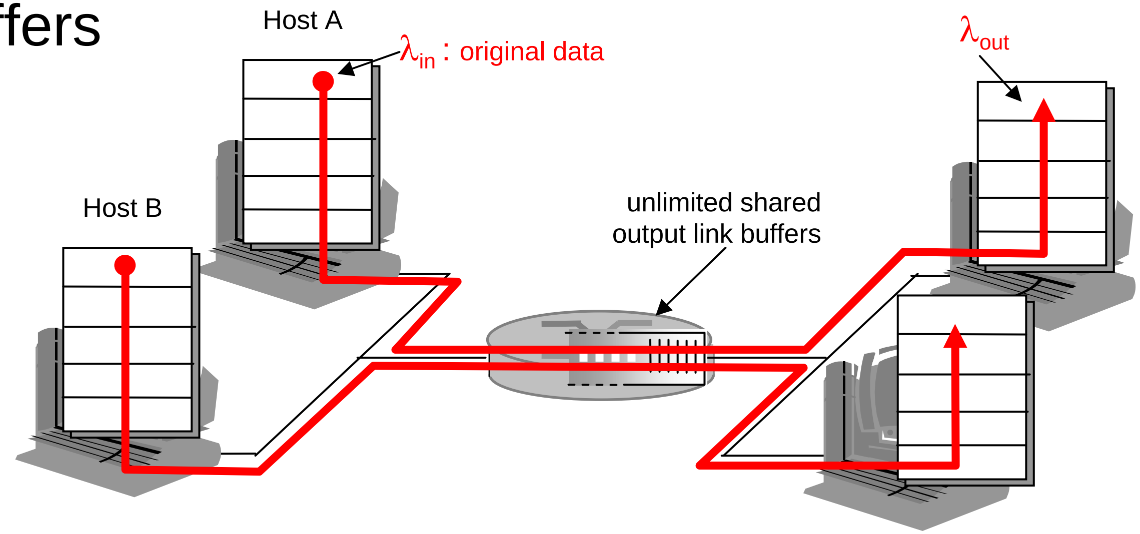
Principles of Congestion Control

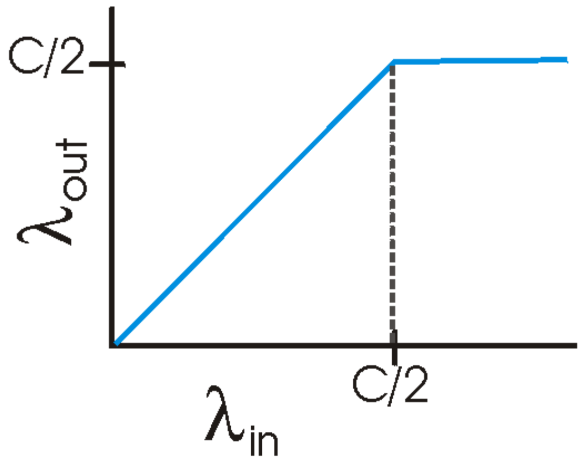
Congestion:

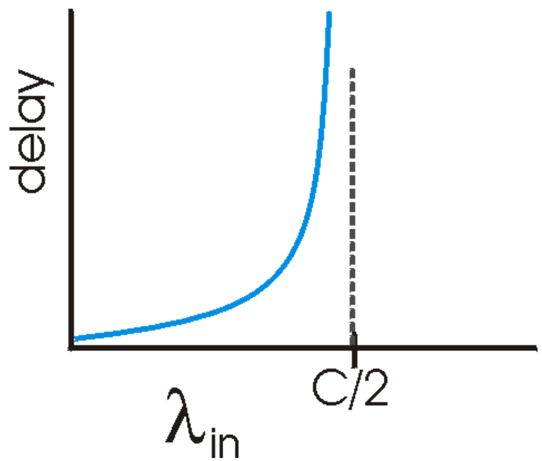
- informally: “too many sources sending too much data too fast for *network* to handle”
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Causes/costs of congestion: scenario 1

- two senders, two receivers
- one router, infinite buffers
- no retransmission



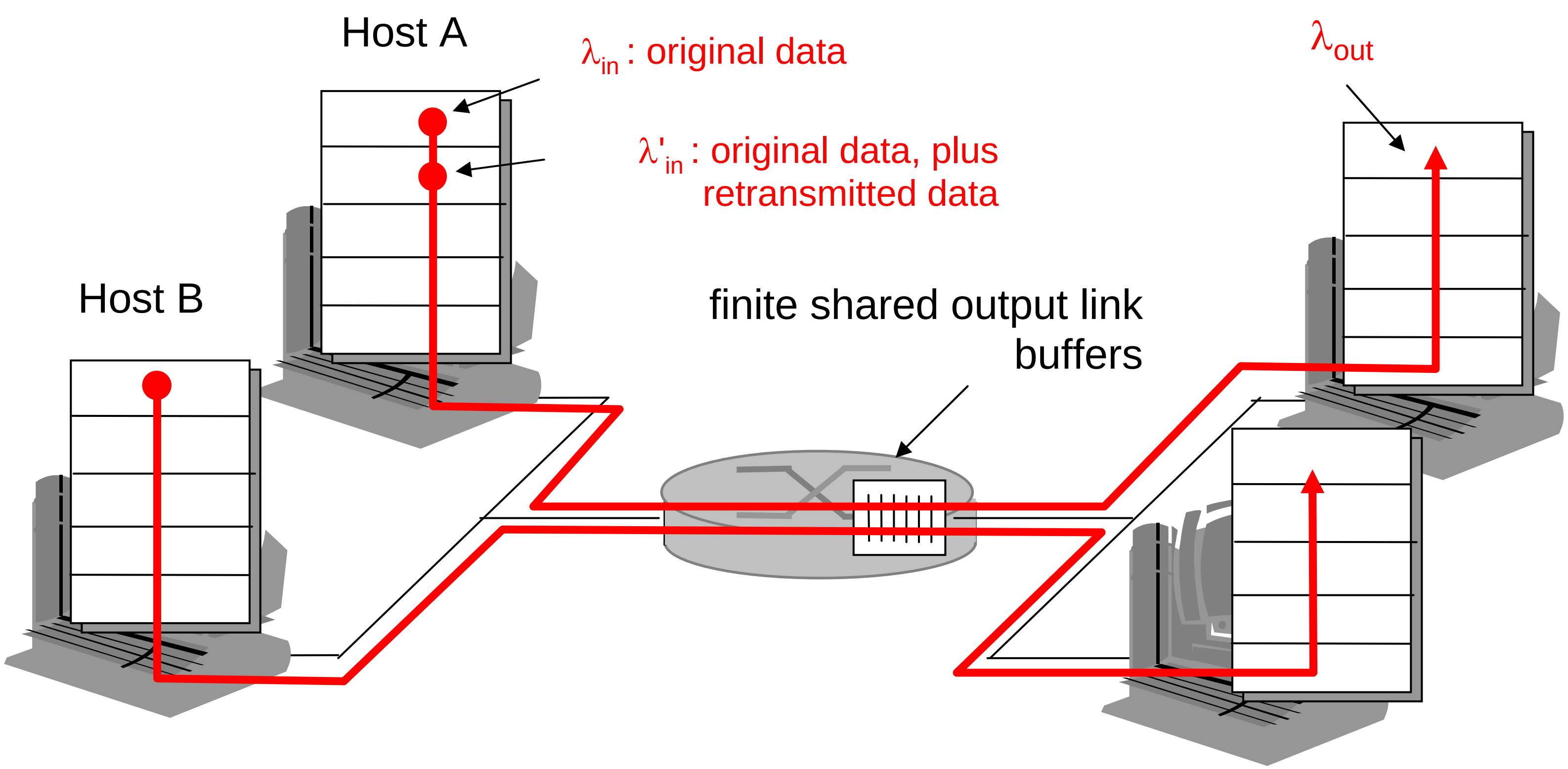




- large delays when congested
- maximum achievable throughput

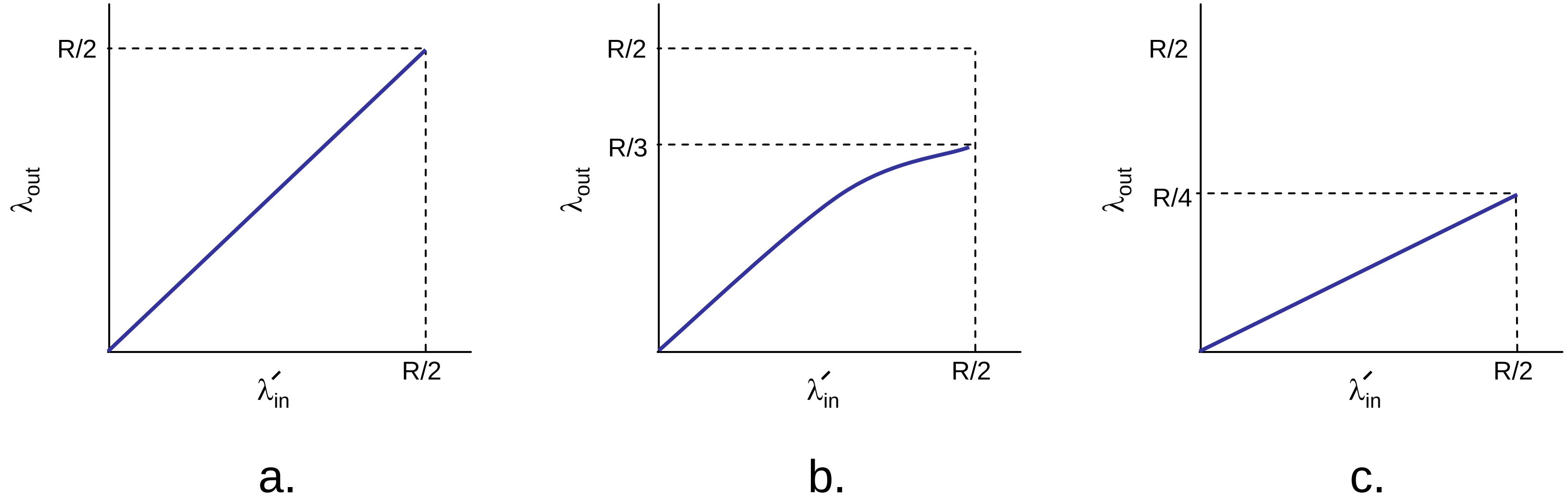
Causes/costs of congestion: scenario 2

- one router, *finite* buffers
- sender retransmission of lost packet



Causes/costs of congestion: scenario 2

- always: $\lambda_{in} = \lambda_{out}$ (goodput)
- "perfect" retransmission only when loss: $\lambda'_{in} > \lambda_{out}$
- retransmission of delayed (not lost) packet makes λ'_{in} larger (than perfect case) for same λ_{out}



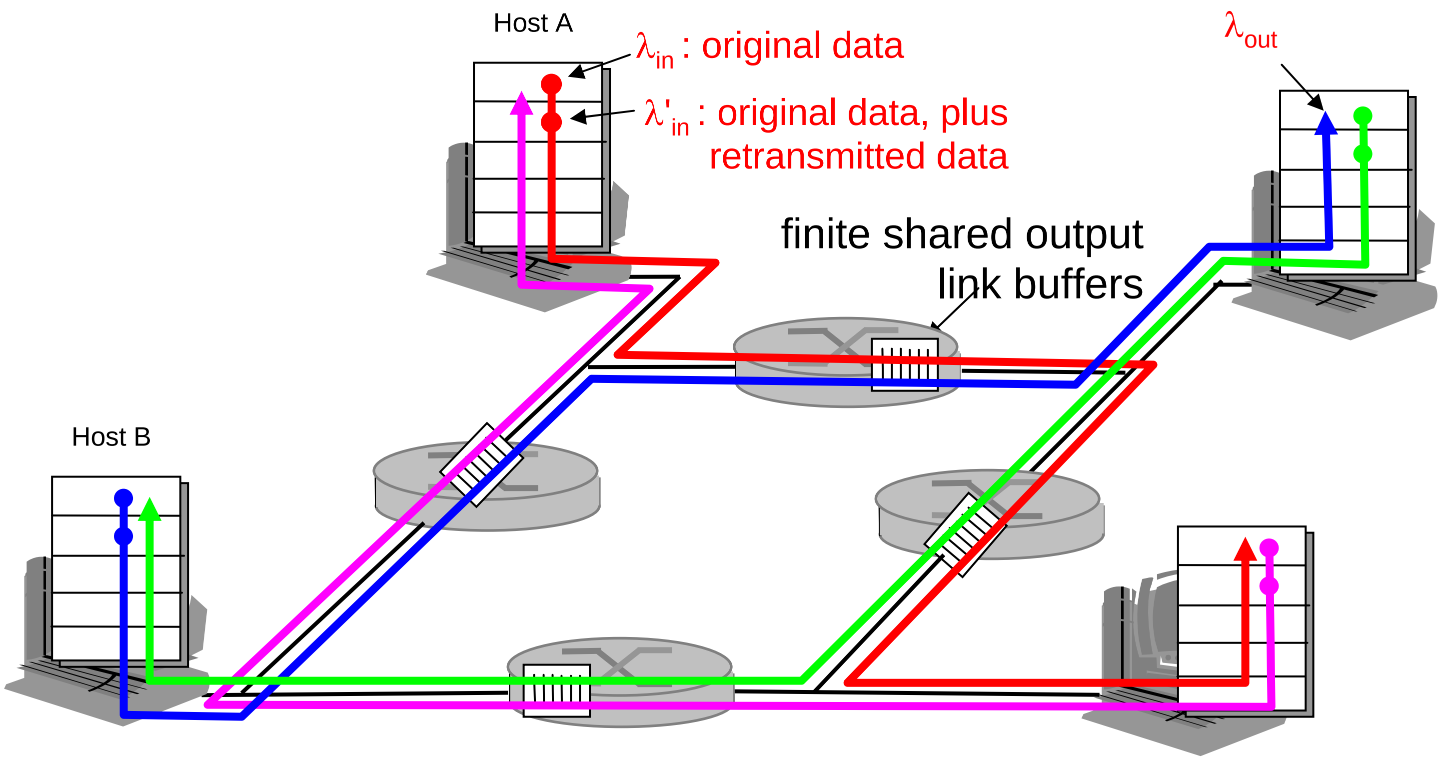
"costs" of congestion:

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

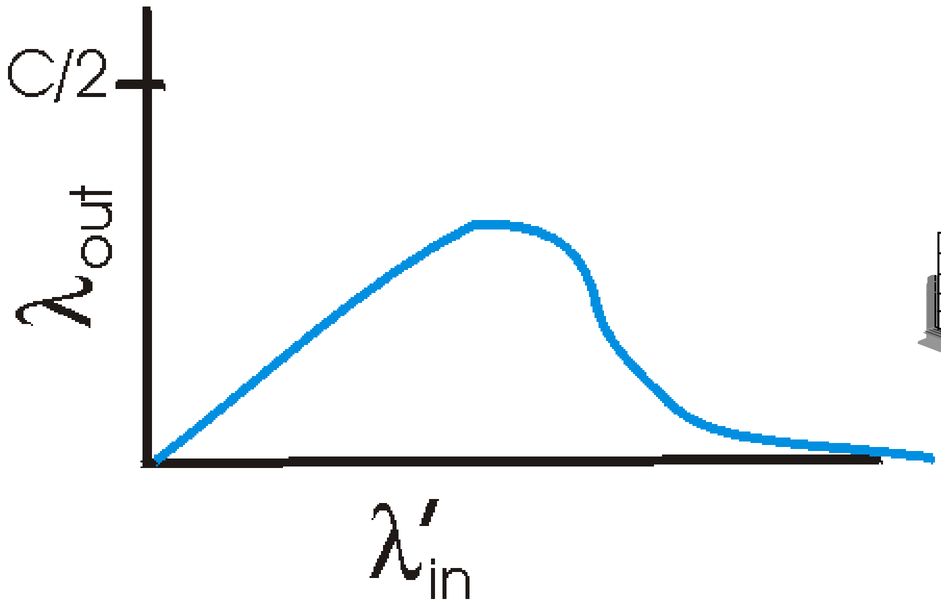
Causes/costs of congestion: scenario 3

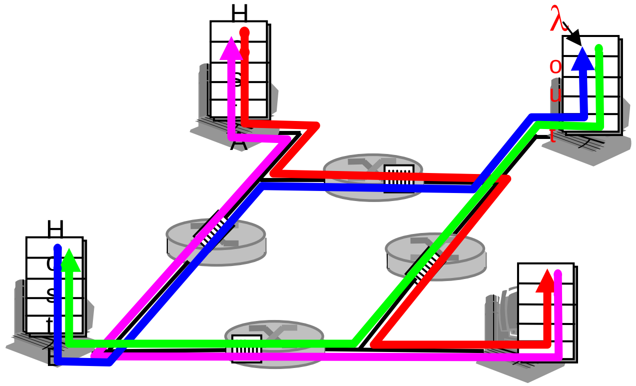
- four senders
- multihop paths
- timeout/retransmit

Q: what happens as λ_{in} and λ'_{in} increase ?



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 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 sender should send at

Case study: ATM ABR congestion control

ABR: available bit rate:

- "elastic service"
- if sender's path "underloaded":
  - sender should use available bandwidth
- if sender's path congested:
  - sender throttled to minimum guaranteed rate

RM (resource management) cells:

- sent by sender, interspersed with data cells
- bits in RM cell set by switches ("*network-assisted*")
  - NI bit: no increase in rate (mild congestion)
  - CI bit: congestion indication
- RM cells returned to sender by receiver, with bits intact

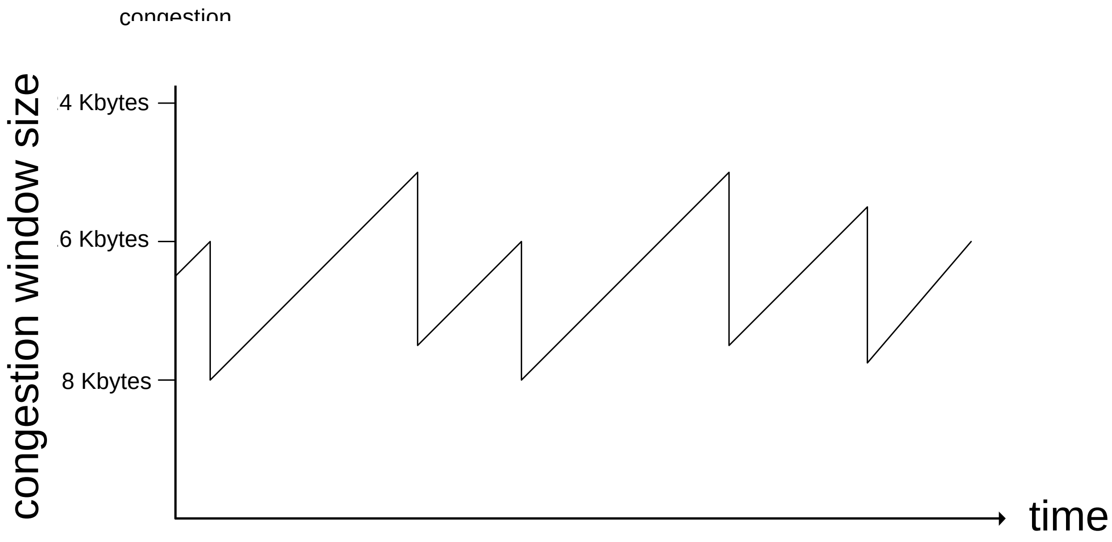
Chapter 3 outline

- 3.1 Transport-layer services
- 3.2 Multiplexing and demultiplexing
- 3.3 Connectionless transport: UDP
- 3.4 Principles of reliable data transfer
- 3.5 Connection-oriented transport: TCP
  - segment structure
  - reliable data transfer
  - flow control
  - connection management
- 3.6 Principles of congestion control
- **3.7 TCP congestion control**

TCP congestion control: additive increase, multiplicative decrease

- *Approach:* increase transmission rate (window size), probing for usable bandwidth, until loss occurs
  - *additive increase:* increase **CongWin** by 1 MSS every RTT until loss detected
  - *multiplicative decrease*: cut **CongWin** in half after loss

Saw tooth behavior: probing for bandwidth



TCP Congestion Control: details

- sender limits transmission:

  `LastByteSent-LastByteAcked` $\leq$ `CongWin`

- Roughly,

$$\boxed{\text{rate} = \frac{\text{CongWin}}{\text{RTT}} \text{ Bytes/sec}}$$

- **`CongWin`** is dynamic, function of perceived network congestion

How does sender perceive congestion?

- loss event = timeout *or* 3 duplicate acks
- TCP sender reduces rate (**`CongWin`**) after loss event

three mechanisms:

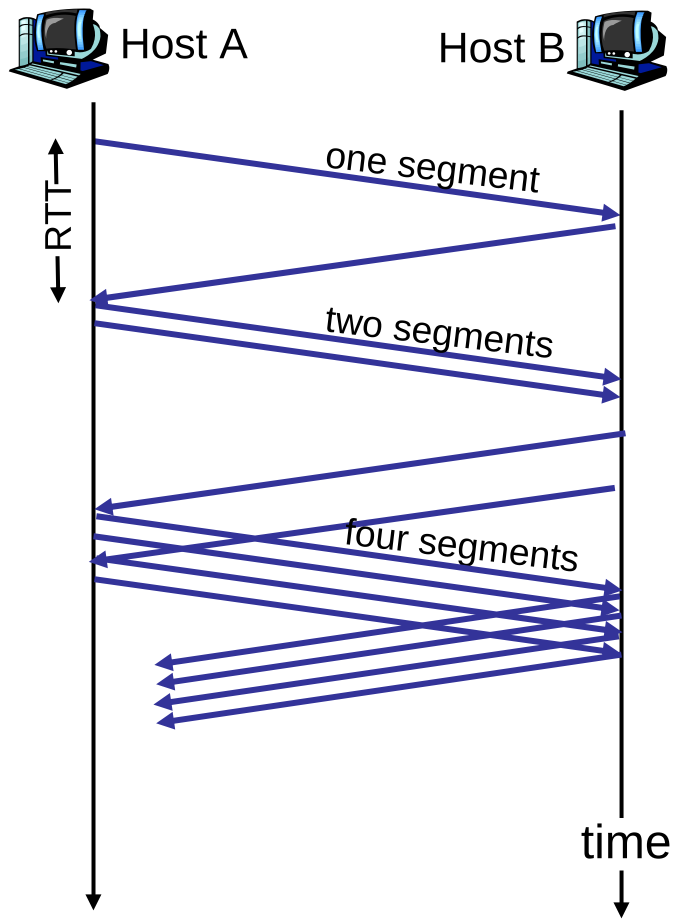
- AIMD
- slow start
- conservative after timeout events

TCP Slow Start

- When connection begins, **`CongWin`** = 1 MSS
  - Example: MSS = 500 bytes & RTT = 200 msec
  - initial rate = 20 kbps
- available bandwidth may be >> MSS/RTT
  - desirable to quickly ramp up to respectable rate
- When connection begins, increase rate exponentially fast until first loss event

TCP Slow Start (more)

- When connection begins, increase rate exponentially until first loss event:
  - double **`CongWin`** every RTT
  - done by incrementing **`CongWin`** for every ACK received
- <u>Summary:</u> initial rate is slow but ramps up exponentially fast



Refinement: inferring loss

- After 3 dup ACKs:
  - `CongWin` is cut in half
  - window then grows linearly
- <u>But</u> after timeout event:
  - `CongWin` instead set to 1 MSS;
  - window then grows exponentially
  - to a threshold, then grows linearly

Philosophy:

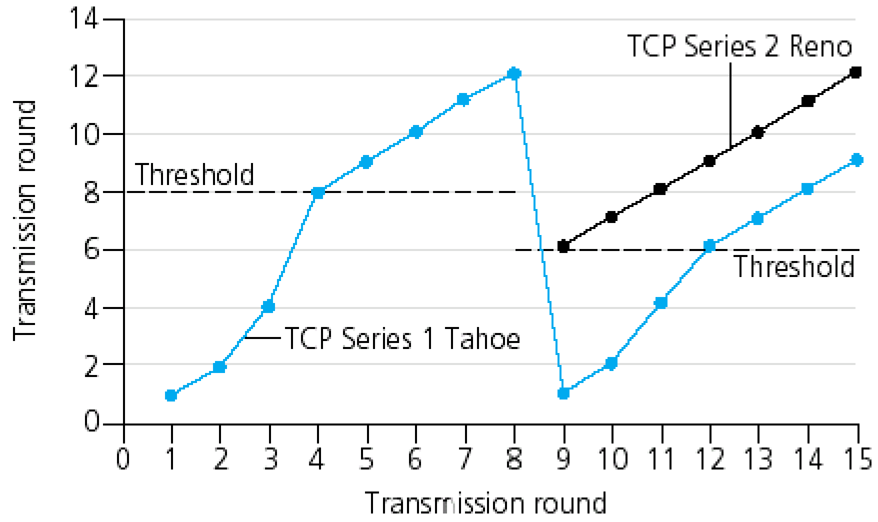
- 3 dup ACKs indicates network capable of delivering some segments
- timeout indicates a "more alarming" congestion scenario

Refinement

- Q: When should the exponential increase switch to linear?
- A: When CongWin gets to 1/2 of its value before timeout.

Implementation:

- Variable Threshold
- At loss event, Threshold is set to 1/2 of CongWin just before loss event



Summary: TCP Congestion Control

- When **`CongWin`** is below **`Threshold`**, sender in slow-start phase, window grows exponentially.
- When **`CongWin`** is above **`Threshold`**, sender is in congestion-avoidance phase, window grows linearly.
- When a triple duplicate ACK occurs, **`Threshold`** set to **`CongWin/2`** and **`CongWin`** set to **`Threshold`**.
- When timeout occurs, **`Threshold`** set to **`CongWin/2`** and **`CongWin`** is set to 1 MSS.

TCP sender congestion control

State	Event	TCP Sender Action	Commentary
Slow Start (SS)	ACK receipt for previously unacked data	CongWin = CongWin + MSS, If (CongWin > Threshold) set state to “Congestion Avoidance”	Resulting in a doubling of CongWin every RTT
Congestion Avoidance (CA)	ACK receipt for previously unacked data	CongWin = CongWin+MSS * (MSS/CongWin)	Additive increase, resulting in increase of CongWin by 1 MSS every RTT
SS or CA	Loss event detected by triple duplicate ACK	Threshold = CongWin/2, CongWin = Threshold, Set state to “Congestion Avoidance”	Fast recovery, implementing multiplicative decrease. CongWin will not drop below 1 MSS.
SS or CA	Timeout	Threshold = CongWin/2, CongWin = 1 MSS, Set state to “Slow Start”	Enter slow start
SS or CA	Duplicate ACK	Increment duplicate ACK count for segment being acked	CongWin and Threshold not changed

TCP summary

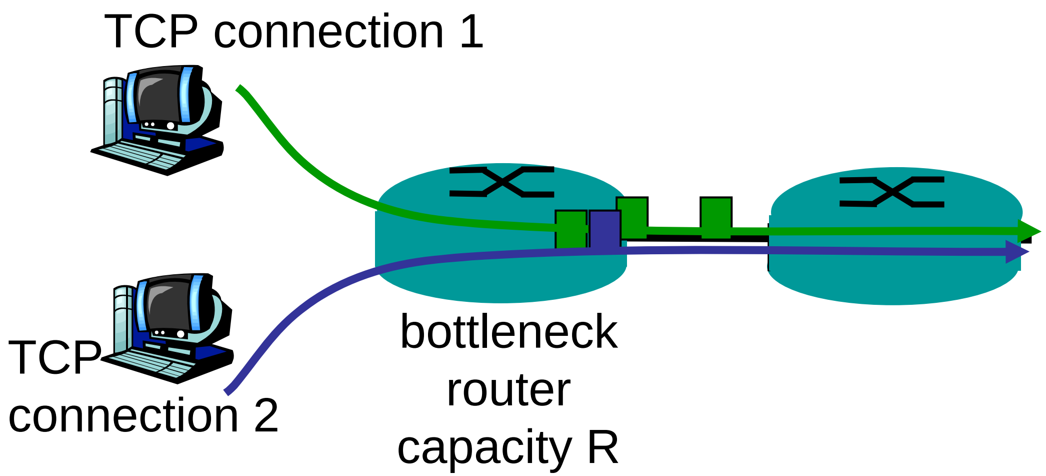
- Connection-oriented: SYN, SYNACK; FIN
- Retransmit lost packets; in-order data: sequence no., ACK no.
- ACKs: either piggybacked, or no-data pure ACK packets if no data travelling in other direction
- Don’t overload receiver: rwin
  - rwin advertised by receiver
- Don’t overload network: cwin
  - cwin affected by receiving ACKs
- Sender buffer = min { rwin, cwin }
- Congestion control:
  - Slow start: exponential growth of cwin
  - Congestion avoidance: linear groth of cwin
  - Timeout; duplicate ACK: shrink cwin
- Continuously adjust RTT estimation

TCP throughput

- What’s the average throughout of TCP as a function of window size and RTT?
  - Ignore slow start
- Let W be the window size when loss occurs.
- When window is W, throughput is W/RTT
- Just after loss, window drops to W/2, throughput to W/2RTT.
- Average throughout: 0.75 W/RTT

TCP Fairness

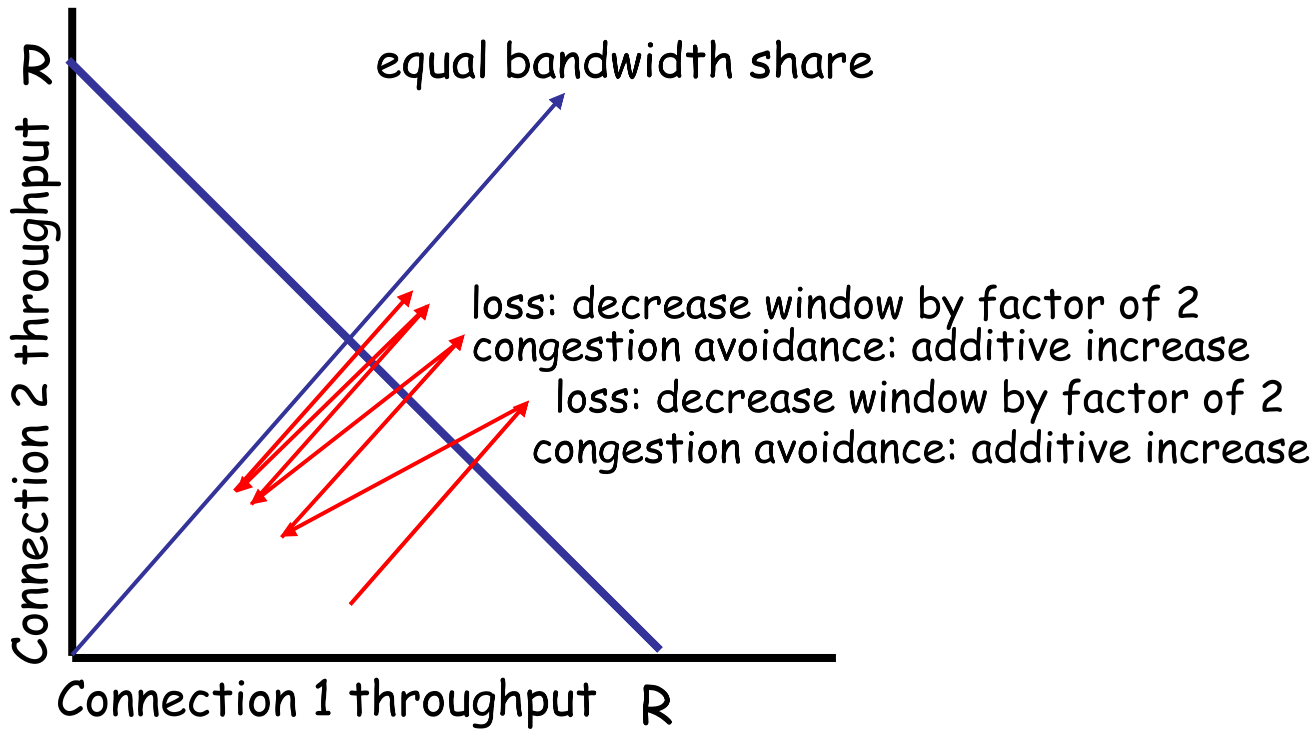
Fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



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:
  - pump audio/video at constant rate, tolerate packet loss
- Research area: TCP friendly

Fairness and parallel TCP connections

- nothing prevents app from opening parallel connections between 2 hosts.
- Web browsers do this
- Example: link of rate R supporting 9 connections;
  - new app asks for 1 TCP, gets rate R/10
  - new app asks for 11 TCPs, gets R/2 !

Chapter 3: Summary

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

Next:

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