# CDA 4506
# Design and Implementation of Data Communication Networks

Lecture Set 4

Dr. R. Lent

# Chapter 3: Transport Layer

Our goals:

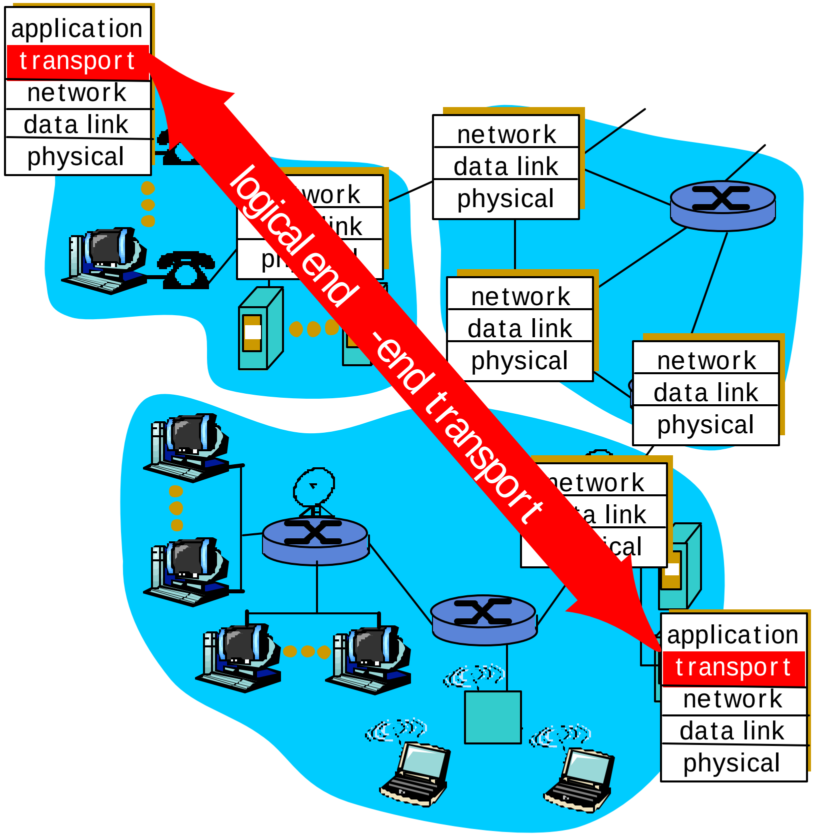
- understand principles behind transport layer services:
  - multiplexing/demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- learn about transport layer protocols in the Internet:
  - UDP: connectionless transport
  - TCP: connection-oriented transport
  - TCP congestion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

# Chapter 3 outline

- 3.1 Transport-layer services
- 3.2 Multiplexing and demultiplexing
- 3.3 Connectionless transport: UDP
- 3.4 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

# 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



# Transport vs. network layer

- *network layer:* logical communication between hosts
- *transport layer:* logical communication between processes
  - relies on, enhances, network layer services

Household analogy:

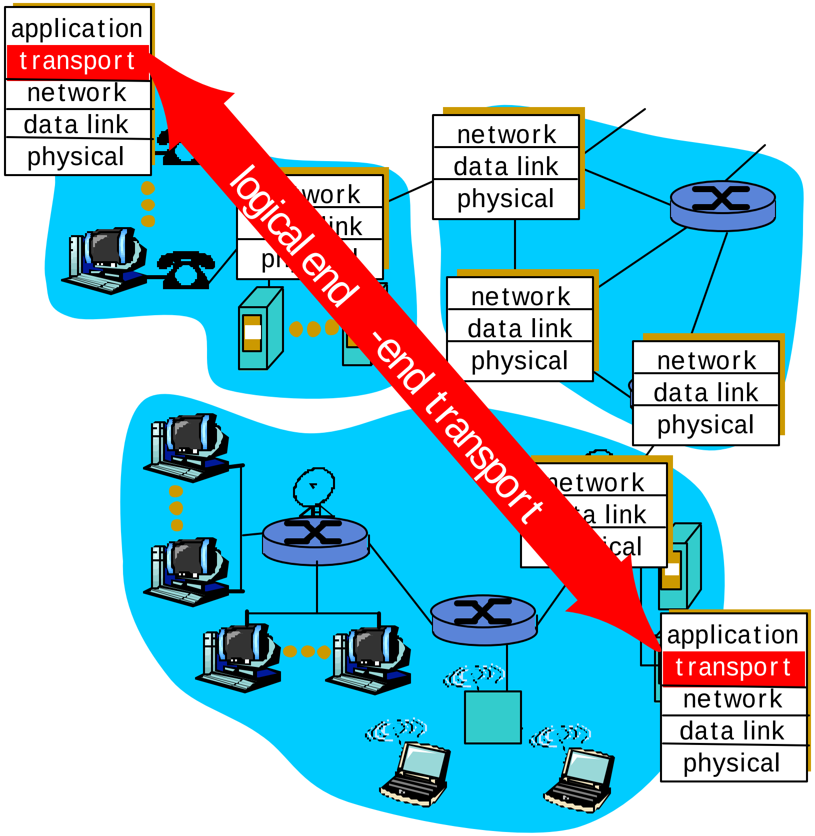
*12 kids sending letters to 12 kids*

- processes = kids
- app messages = letters in envelopes
- hosts = houses
- transport protocol = Ann and Bill
- network-layer protocol = postal service

# Internet transport-layer protocols

1. reliable, in-order delivery (TCP)
    1. congestion control
    2. flow control
    3. connection setup
2. unreliable, unordered delivery: UDP
    1. 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
- 3.4 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

# Multiplexing/demultiplexing

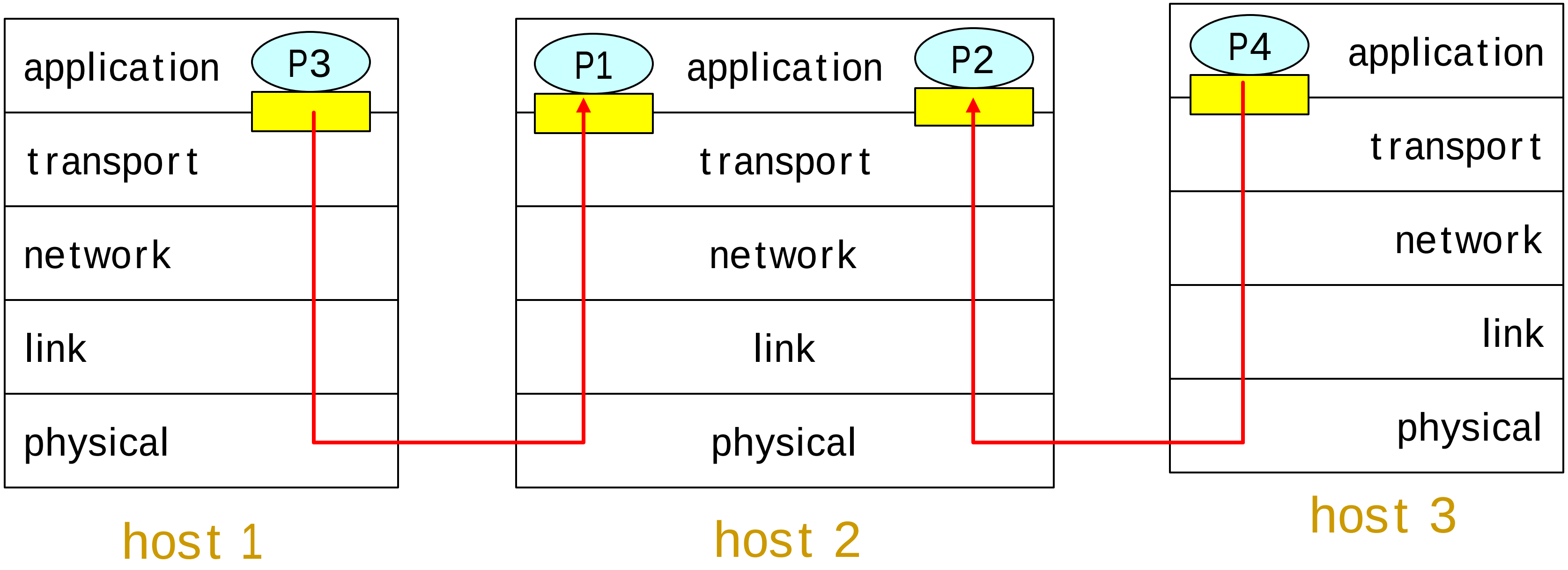
Demultiplexing at rcv host:

delivering received segments to correct socket

Multiplexing at send host:

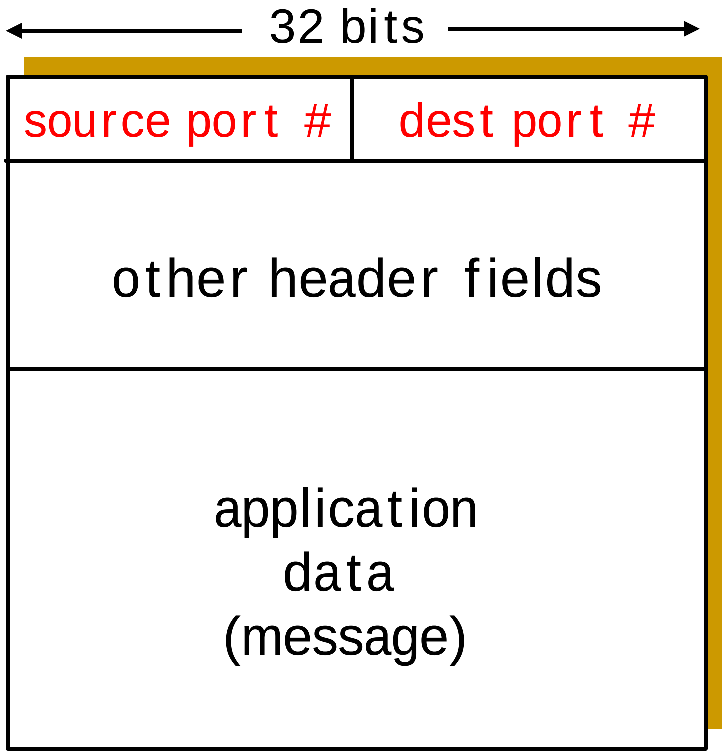
gathering data from multiple sockets, enveloping data with header (later used for demultiplexing)

🟨 = socket ◯ = process



# 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 (recall: well-known port numbers for specific applications)
- host uses IP addresses & port numbers to direct segment to appropriate socket



32 bits

| source port # | dest port # |
|---|---|
| other header fields | |
| application data (message) | |

TCP/UDP segment format

# Connectionless demultiplexing

- Create sockets with port numbers:

```
DatagramSocket mySocket1 = new
   DatagramSocket(99111);
DatagramSocket mySocket2 = new
   DatagramSocket(99222);
```

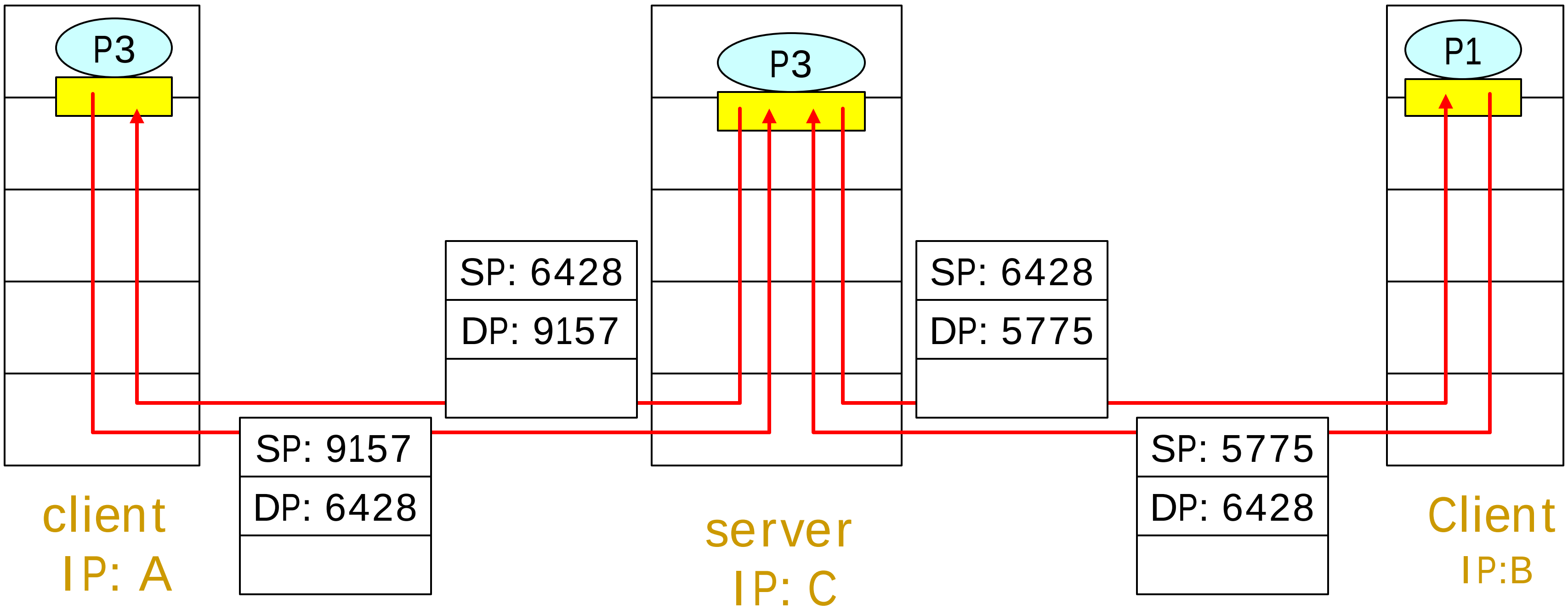
- UDP socket identified by two-tuple:

(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);
```



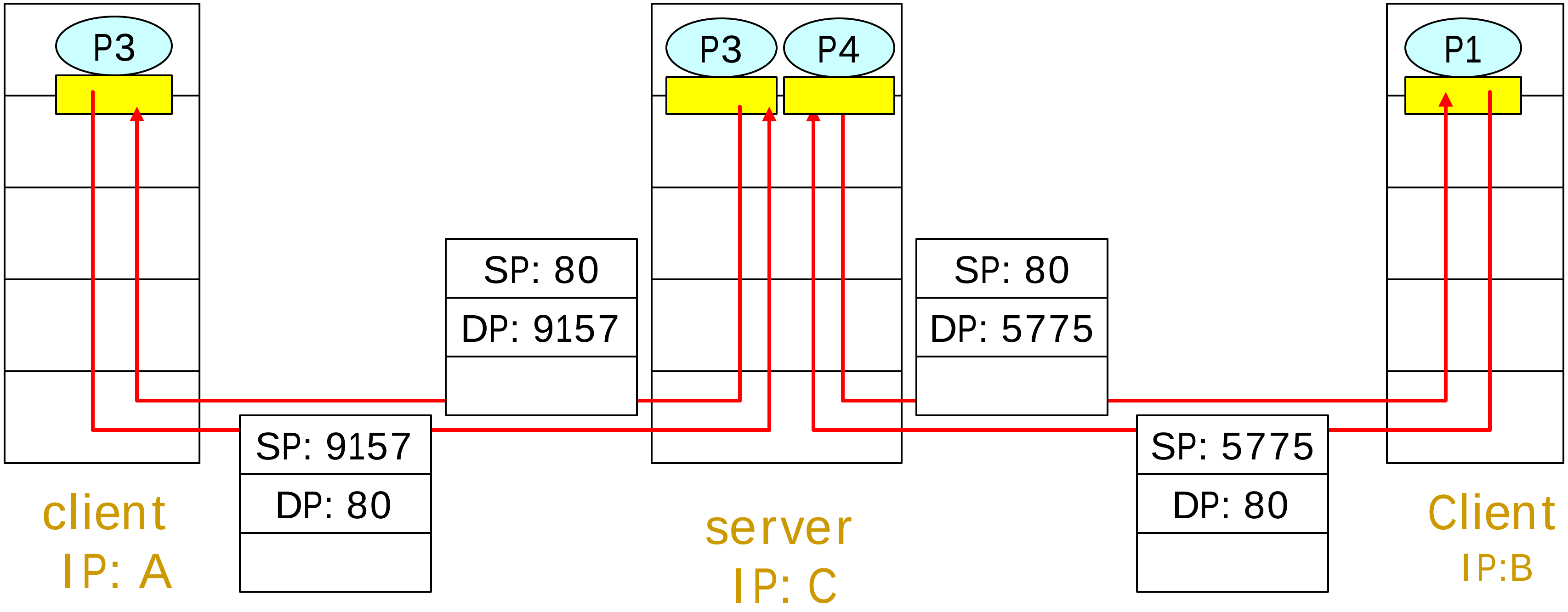
SP provides “return address”

# 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)



# 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

# UDP: User Datagram Protocol [RFC 768]

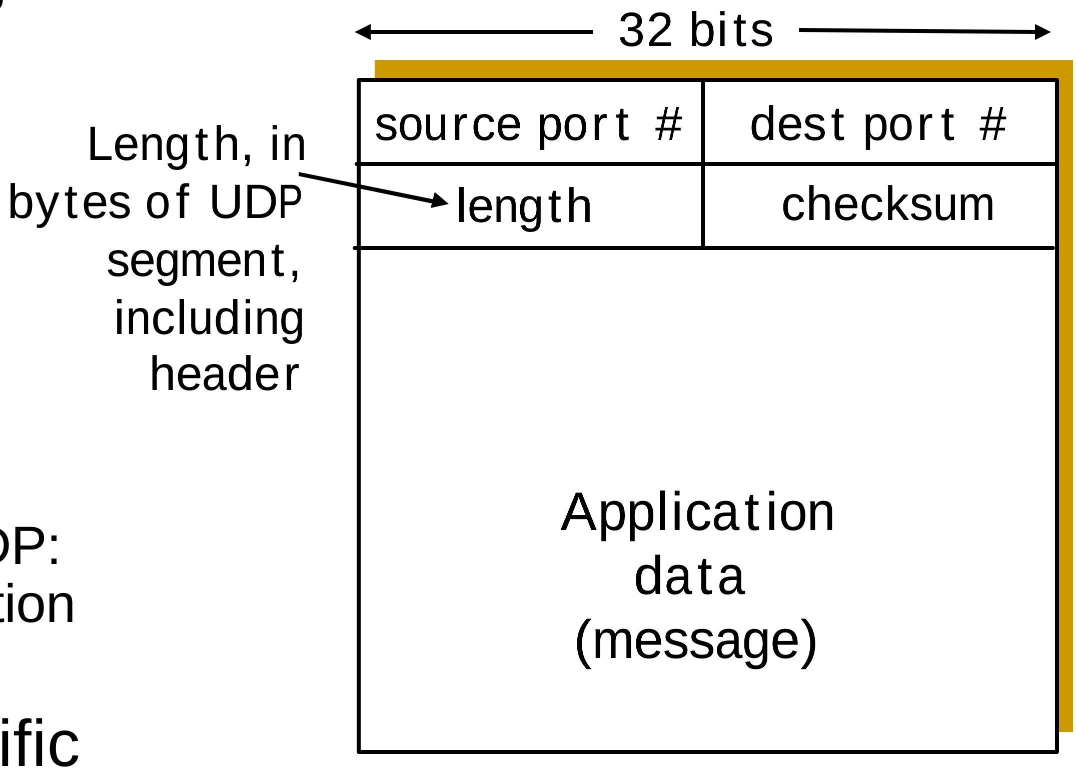
- “no frills,” “bare bones” Internet transport protocol
- “best effort” service, UDP segments may be:
  - lost
  - delivered out of order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

## 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!



32 bits

source port # | dest port #

length | checksum

Application data (message)

Length, in bytes of UDP segment, including header

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:

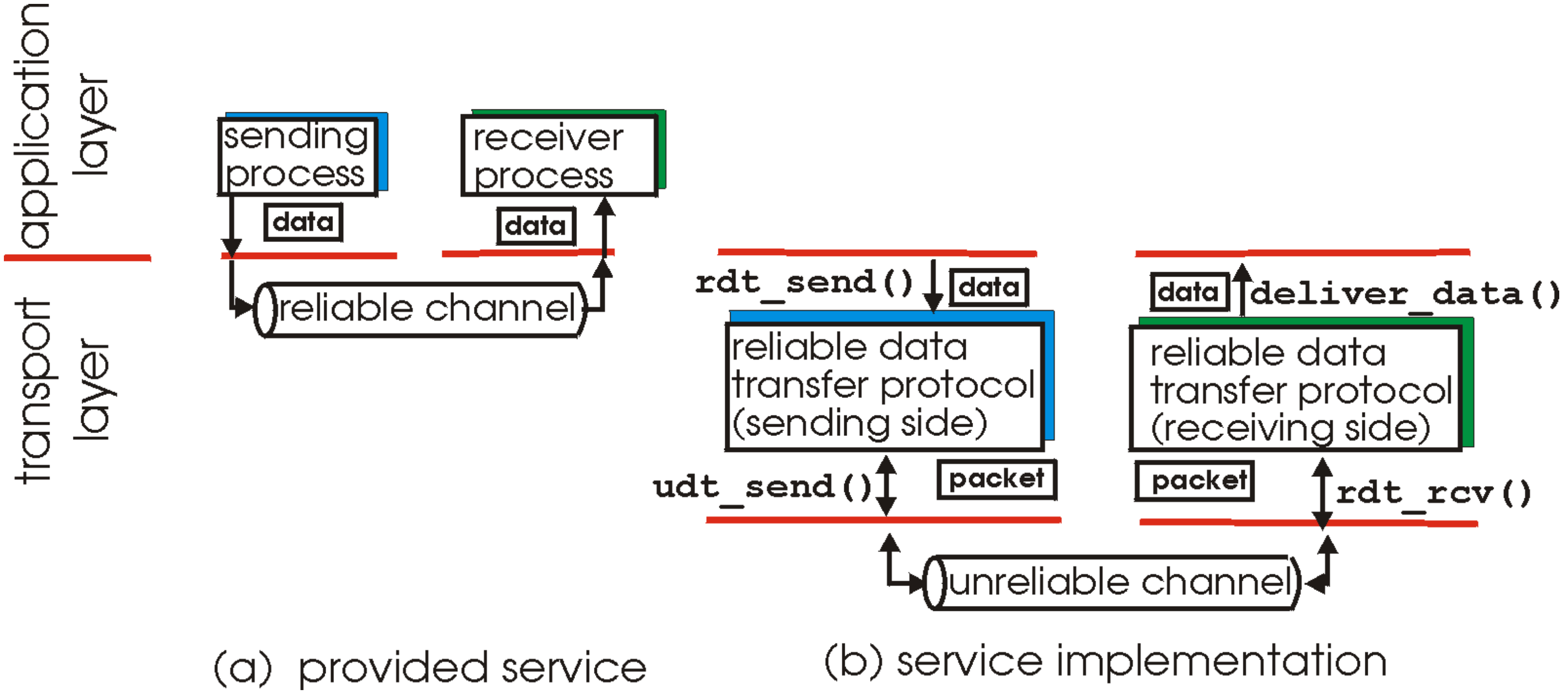
- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - NO - error detected
  - YES - no error detected. *But may be errors nonetheless?* More later ….

# 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 Reliable data transfer

- important in app., transport, link layers
- top-10 list of important networking topics!

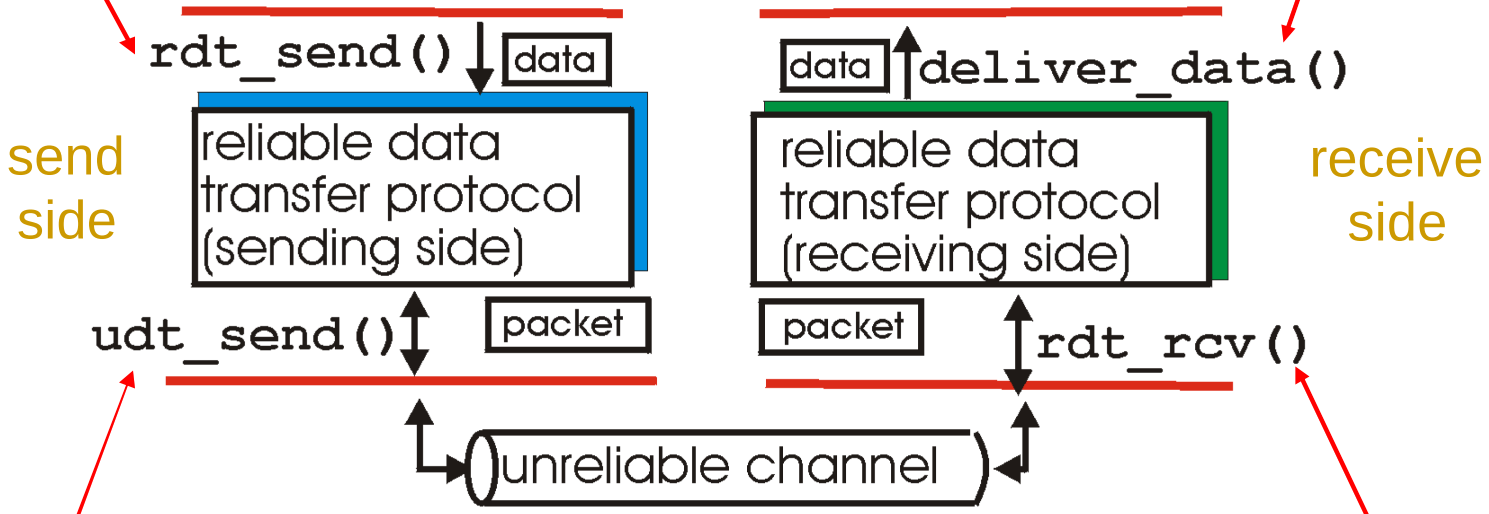


- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

# Reliable data transfer: getting started

**rdt_send():** called from above, (e.g., by app.). Passed data to deliver to receiver upper layer

**deliver_data():** called by **rdt** to deliver data to upper



**udt_send():** called by rdt, to transfer packet over unreliable channel to receiver

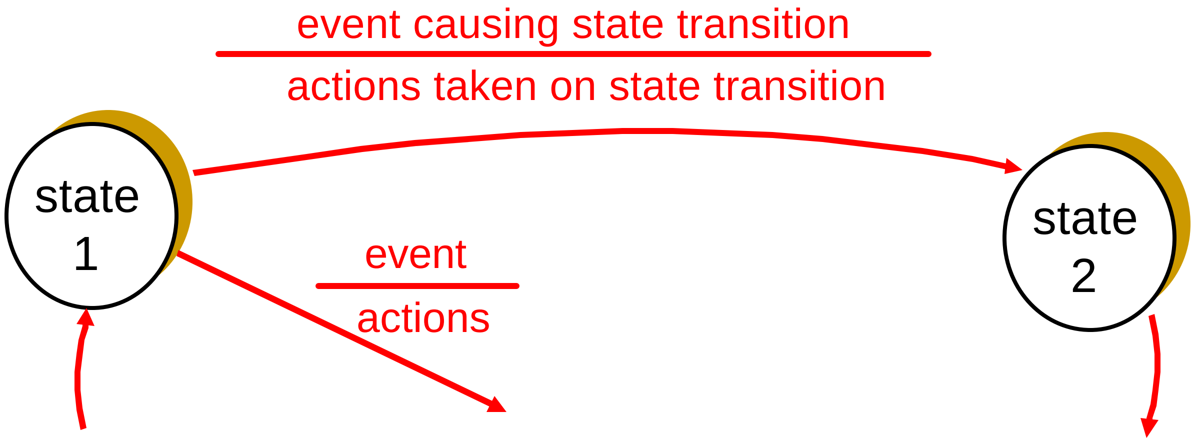
**rdt_rcv():** called when packet arrives on rcv-side of channel

# Reliable data transfer: getting started

We'll:

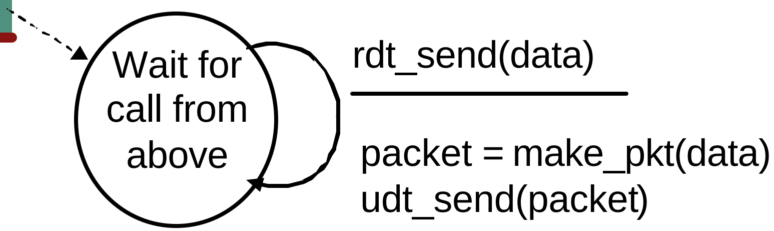
- incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- consider only unidirectional data transfer
  - but control info will flow on both directions!
- use finite state machines (FSM) to specify sender, receiver

state: when in this "state" next state uniquely determined by next event


event causing state transition
actions taken on state transition
state 1
event
actions
state 2

# Rdt1.0: reliable transfer over a reliable channel

- underlying channel perfectly reliable
  - no bit errors
  - no loss of packets
- separate FSMs for sender, receiver:
  - sender sends data into underlying channel
  - receiver read data from underlying channel

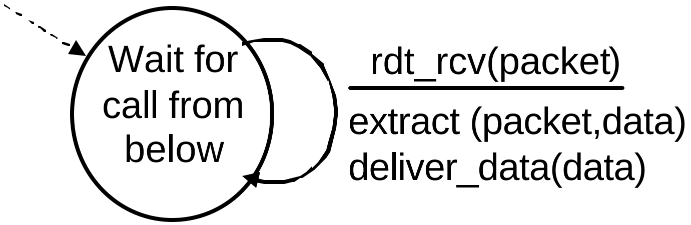


Wait for call from above

rdt_send(data)

packet = make_pkt(data)
udt_send(packet)

sender



Wait for call from below

rdt_rcv(packet)

extract (packet,data)
deliver_data(data)

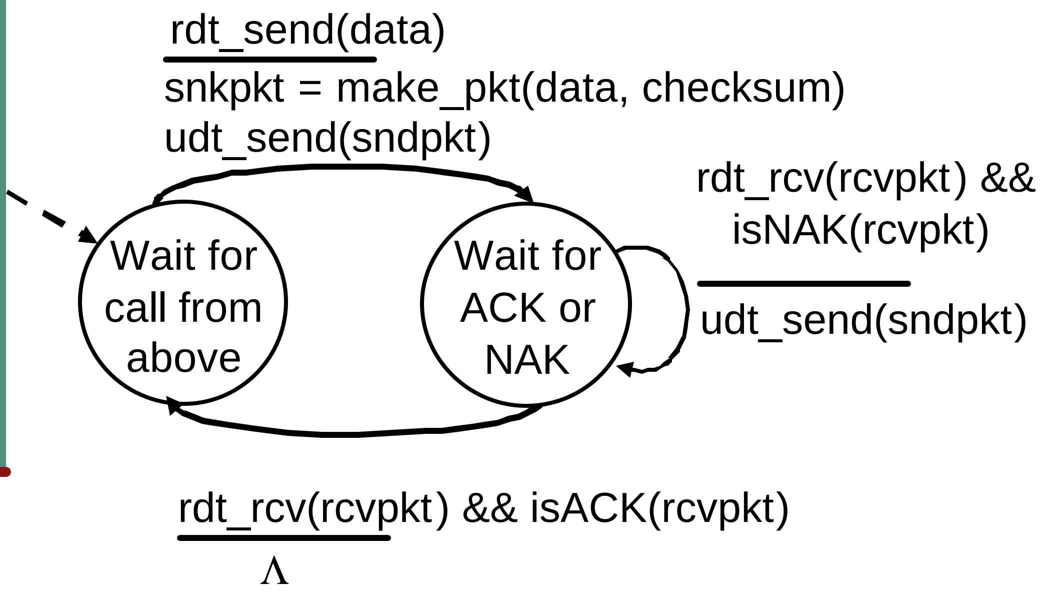
receiver

# Rdt2.0: channel with bit errors

- Underlying channel may flip bits in packet
  - recall: UDP checksum detects bit errors
- *The* question: how to recover from errors:
  - *acknowledgements (ACKs):* receiver explicitly tells sender that pkt was received OK
  - *negative acknowledgements (NAKs):* receiver explicitly tells sender that pkt had errors
  - sender retransmits pkt on receipt of NAK
- New mechanisms in **`rdt2.0`** (beyond **`rdt1.0`**):
  - error detection
  - receiver feedback: control msgs (ACK,NAK) rcvr->sender

# Rdt2.0: FSM specification

sender



rdt_send(data)

snkpkt = make_pkt(data, checksum)
udt_send(sndpkt)

Wait for call from above

Wait for ACK or NAK

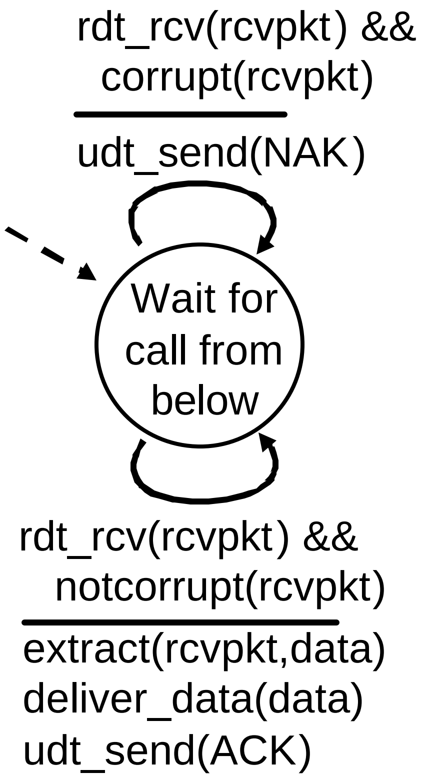
rdt_rcv(rcvpkt) &&
isNAK(rcvpkt)

udt_send(sndpkt)

rdt_rcv(rcvpkt) && isACK(rcvpkt)

Λ

receiver



rdt_rcv(rcvpkt) &&
corrupt(rcvpkt)

udt_send(NAK)

Wait for call from below

rdt_rcv(rcvpkt) &&
notcorrupt(rcvpkt)

extract(rcvpkt,data)
deliver_data(data)
udt_send(ACK)

# rdt2.0 has a fatal flaw!

What happens if ACK/NAK corrupted?

- sender doesn't know what happened at receiver!
- can't just retransmit: possible duplicate

What to do?

- sender ACKs/NAKs receiver's ACK/NAK? What if sender ACK/NAK lost?
- retransmit, but this might cause retransmission of a correctly received pkt!

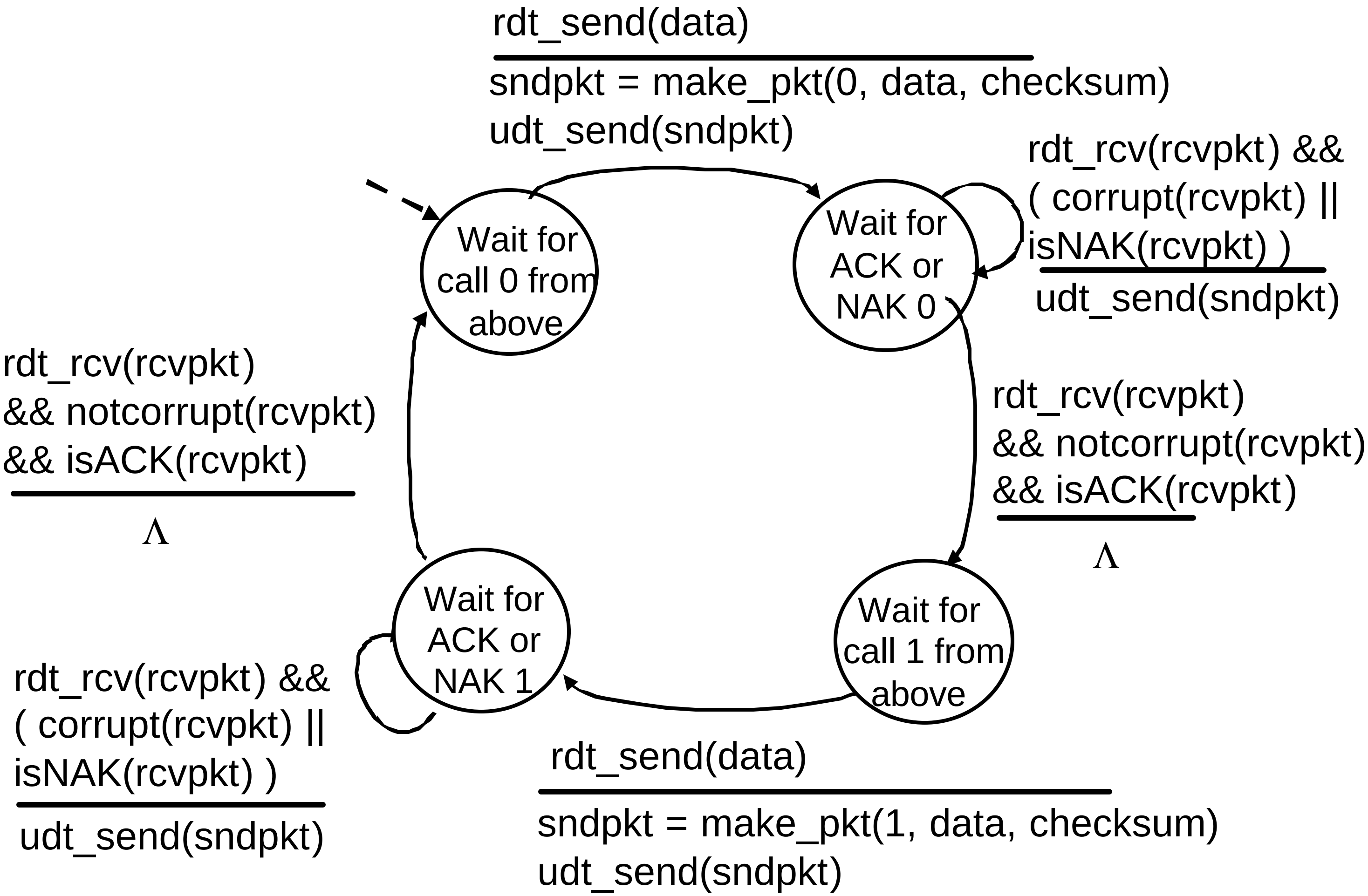
Handling duplicates:

- sender adds *sequence number* to each pkt
- sender retransmits current pkt if ACK/NAK garbled
- receiver discards (doesn't deliver up) duplicate pkt

stop and wait

Sender sends one packet, then waits for receiver response

# rdt2.1: sender, handles garbled ACK/NAKs



Wait for call 0 from above

Wait for ACK or NAK 0

Wait for call 1 from above

Wait for ACK or NAK 1

rdt_send(data)
sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) &&
( corrupt(rcvpkt) ||
isNAK(rcvpkt) )
udt_send(sndpkt)

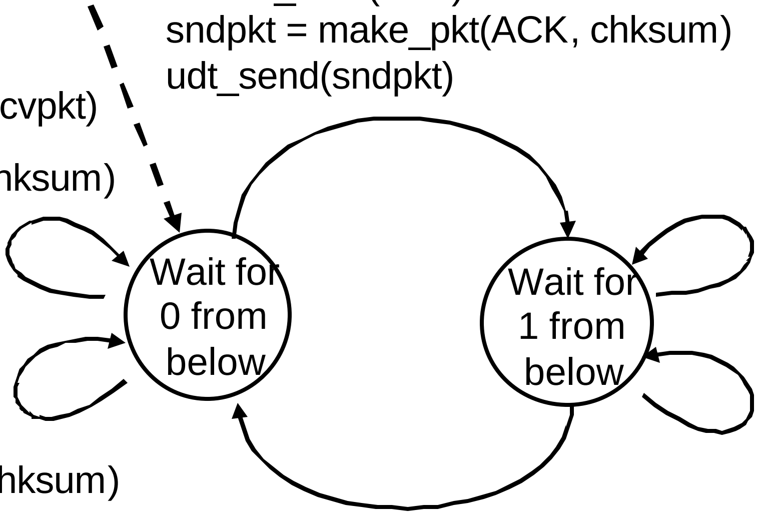
rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt)
Λ

rdt_send(data)
sndpkt = make_pkt(1, data, checksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) &&
( corrupt(rcvpkt) ||
isNAK(rcvpkt) )
udt_send(sndpkt)

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt)
Λ

# rdt2.1: receiver, handles garbled ACK/NAKs



rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
&& has_seq0(rcvpkt)

extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) && (corrupt(rcvpkt)

sndpkt = make_pkt(NAK, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) &&
not corrupt(rcvpkt) &&
has_seq1(rcvpkt)

sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) && (corrupt(rcvpkt)

sndpkt = make_pkt(NAK, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) &&
not corrupt(rcvpkt) &&
has_seq0(rcvpkt)

sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
&& has_seq1(rcvpkt)

extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)

# rdt2.1: discussion

Sender:

- seq # added to pkt
- two seq. #'s (0,1) will suffice. Why?
- must check if received ACK/NAK corrupted
- twice as many states
  - state must "remember" whether "current" pkt has 0 or 1 seq. #

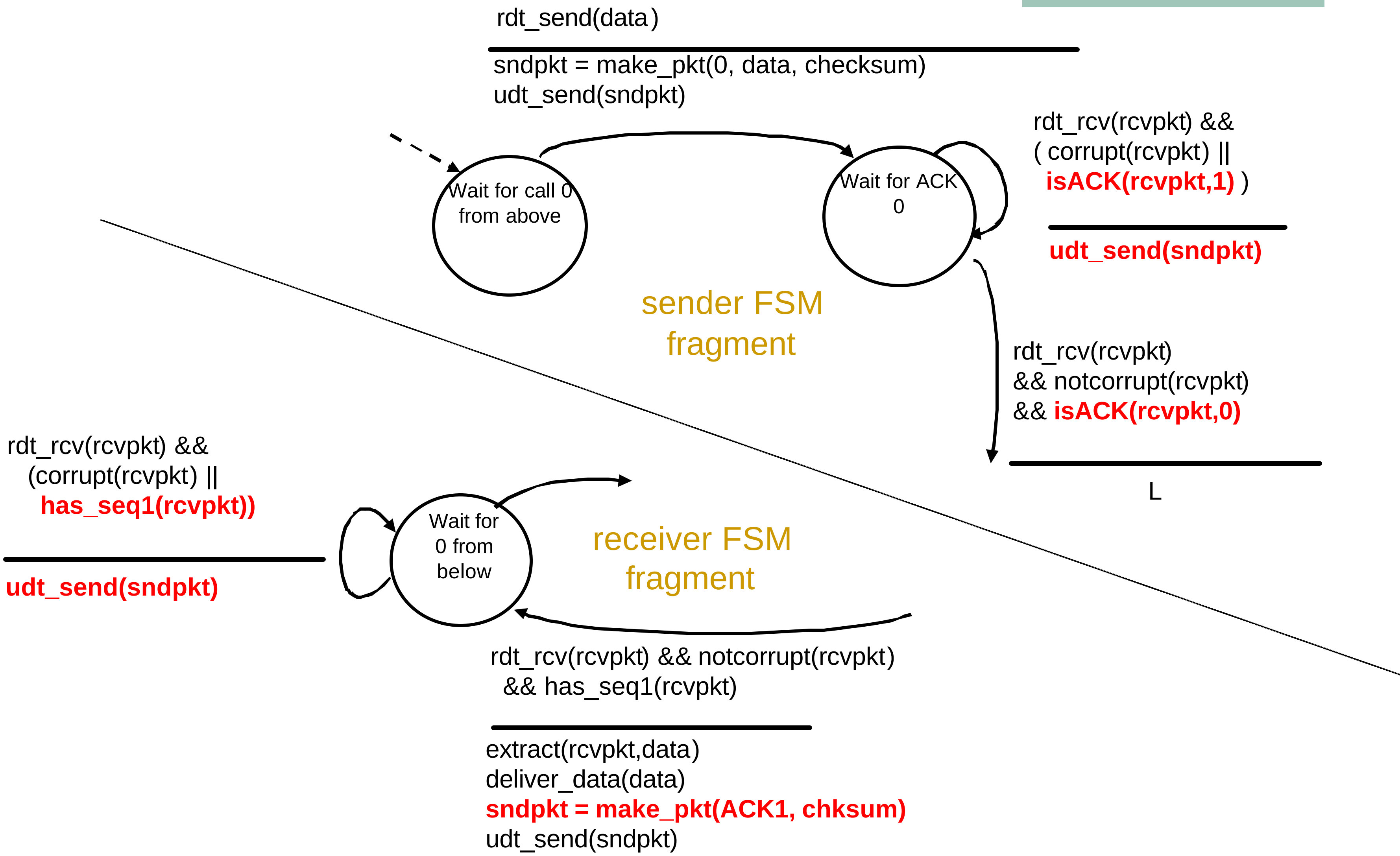
Receiver:

- must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected pkt seq #
- note: receiver can *not* know if its last ACK/NAK received OK at sender

# rdt2.2: a NAK-free protocol

- same functionality as rdt2.1, using NAKs only
- instead of NAK, receiver sends ACK for last pkt received OK
    - receiver must *explicitly* include seq # of pkt being ACKed
- duplicate ACK at sender results in same action as NAK: *retransmit current pkt*

# rdt2.2: sender, receiver fragments



rdt_send(data)

sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)

Wait for call 0 from above

Wait for ACK 0

rdt_rcv(rcvpkt) &&
( corrupt(rcvpkt) ||
**isACK(rcvpkt,1)** )

**udt_send(sndpkt)**

sender FSM fragment

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& **isACK(rcvpkt,0)**

L

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
**has_seq1(rcvpkt))**

**udt_send(sndpkt)**

Wait for 0 from below

receiver FSM fragment

rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
&& has_seq1(rcvpkt)

extract(rcvpkt,data)
deliver_data(data)
**sndpkt = make_pkt(ACK1, chksum)**
udt_send(sndpkt)

# rdt3.0: channels with errors *and* loss

New assumption: underlying channel can also lose packets (data or ACKs)

- checksum, seq. #, ACKs, retransmissions will be of help, but not enough

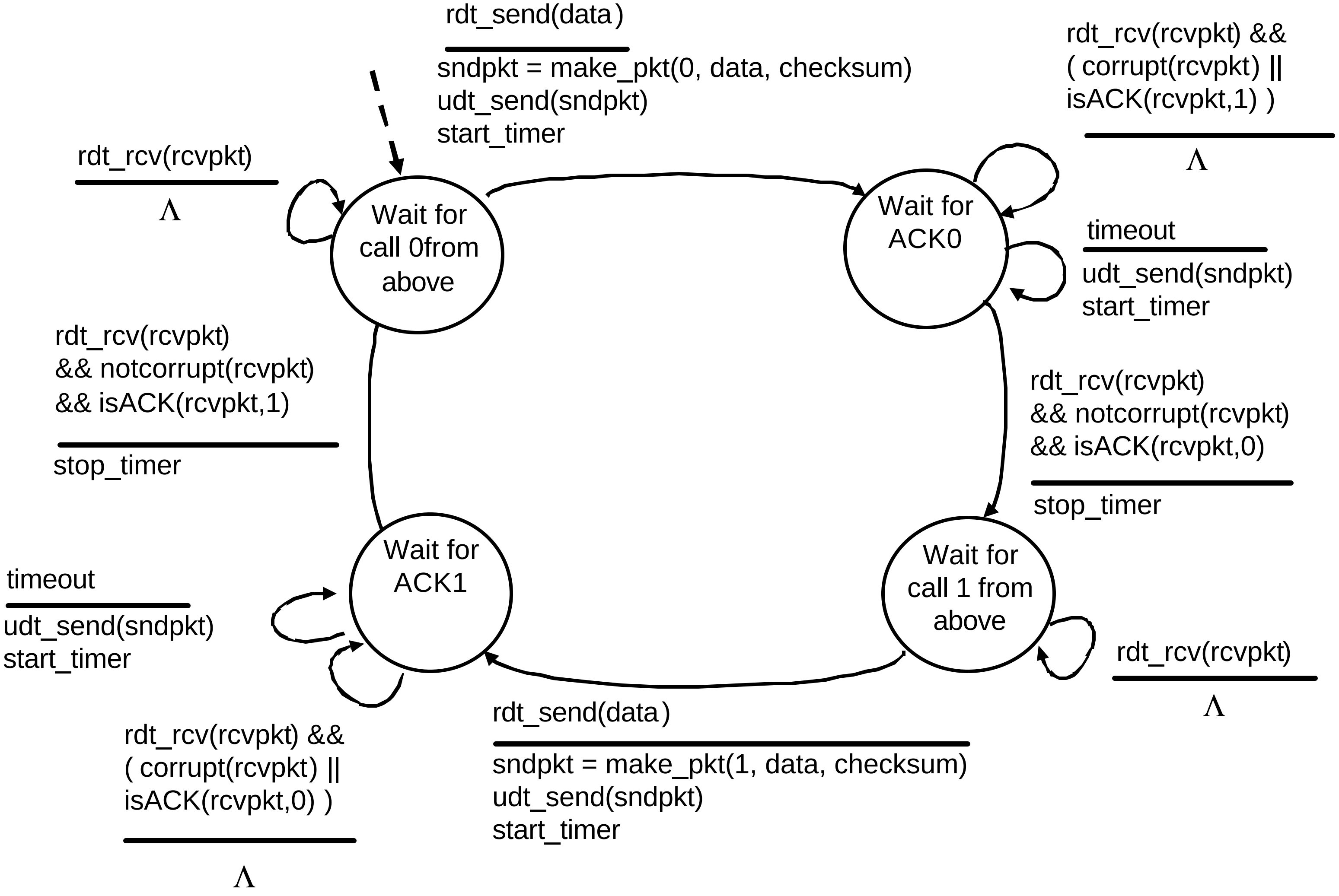
Q: how to deal with loss?

- sender waits until certain data or ACK lost, then retransmits
- yuck: drawbacks?

Approach: sender waits "reasonable" amount of time for ACK

- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but use of seq. #'s already handles this
  - receiver must specify seq # of pkt being ACKed
- requires countdown timer

# rdt3.0 sender



rdt_send(data)

sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)
start_timer

rdt_rcv(rcvpkt)

Λ

Wait for call 0from above

Wait for ACK0

rdt_rcv(rcvpkt) &&
( corrupt(rcvpkt) ||
isACK(rcvpkt,1) )

Λ

timeout

udt_send(sndpkt)
start_timer

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt,1)

stop_timer

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt,0)

stop_timer

Wait for ACK1

Wait for call 1 from above

timeout

udt_send(sndpkt)
start_timer

rdt_rcv(rcvpkt)

Λ

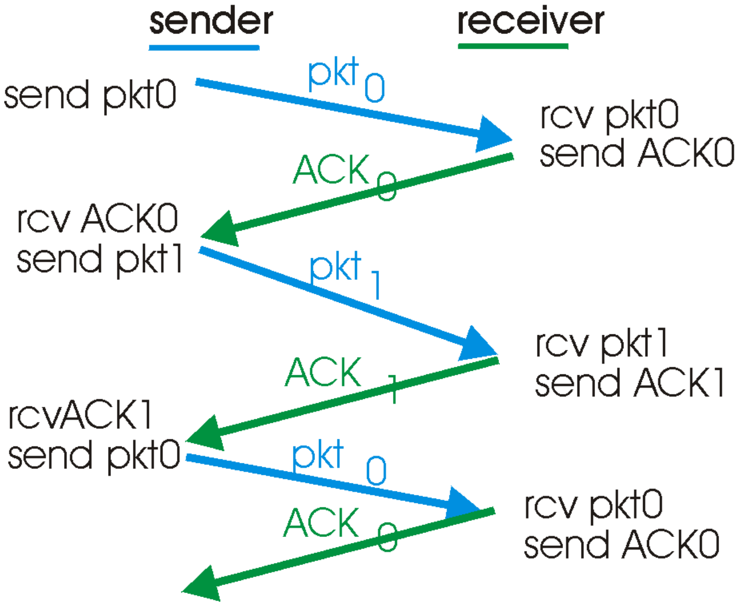
rdt_rcv(rcvpkt) &&
( corrupt(rcvpkt) ||
isACK(rcvpkt,0) )

Λ

rdt_send(data)

sndpkt = make_pkt(1, data, checksum)
udt_send(sndpkt)
start_timer

# rdt3.0 in action



sender

receiver

send pkt0

pkt 0

rcv pkt0
send ACK0

ACK 0

rcv ACK0
send pkt1

pkt 1

rcv pkt1
send ACK1

ACK 1

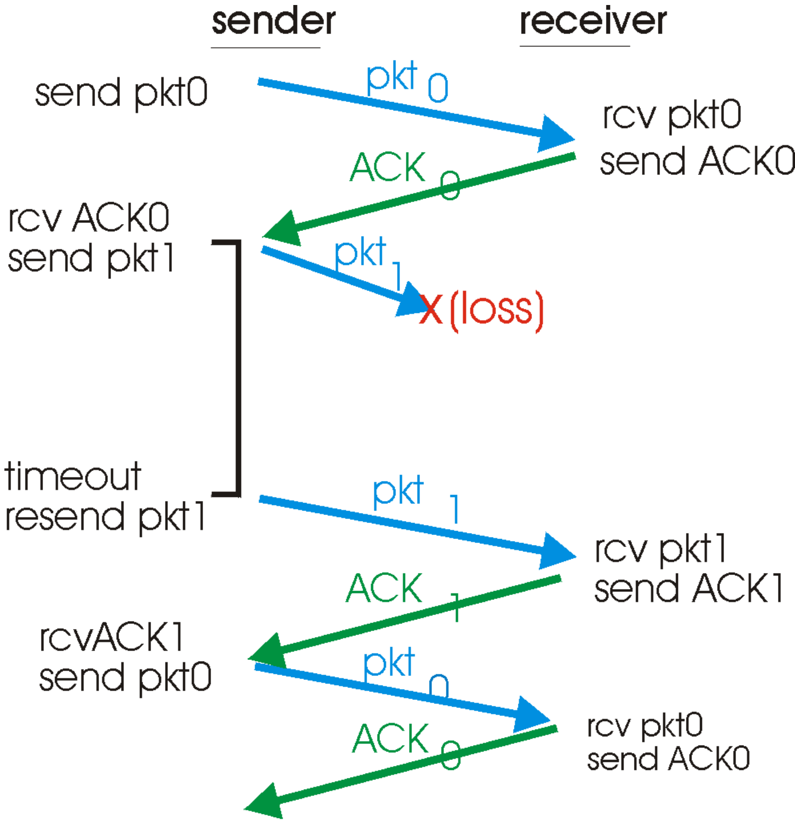
rcvACK1
send pkt0

pkt 0

rcv pkt0
send ACK0

ACK 0

(a) operation with no loss



sender

receiver

send pkt0

pkt 0

rcv pkt0
send ACK0

ACK 0

rcv ACK0
send pkt1

pkt 1

X (loss)

timeout
resend pkt1

pkt 1

rcv pkt1
send ACK1

ACK 1

rcvACK1
send pkt0

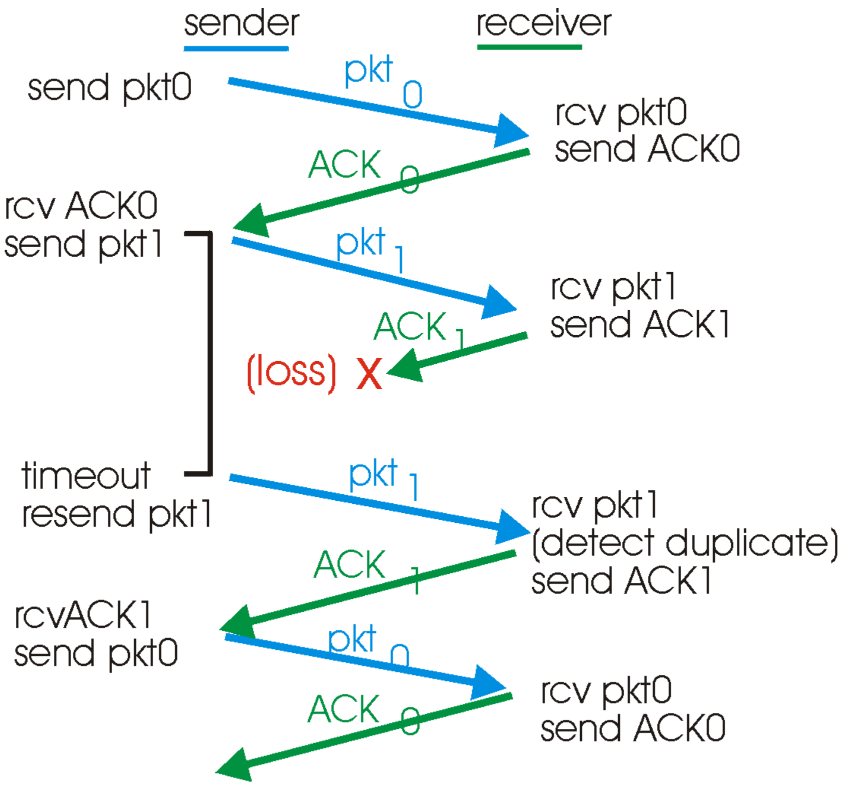
pkt 0

rcv pkt0
send ACK0

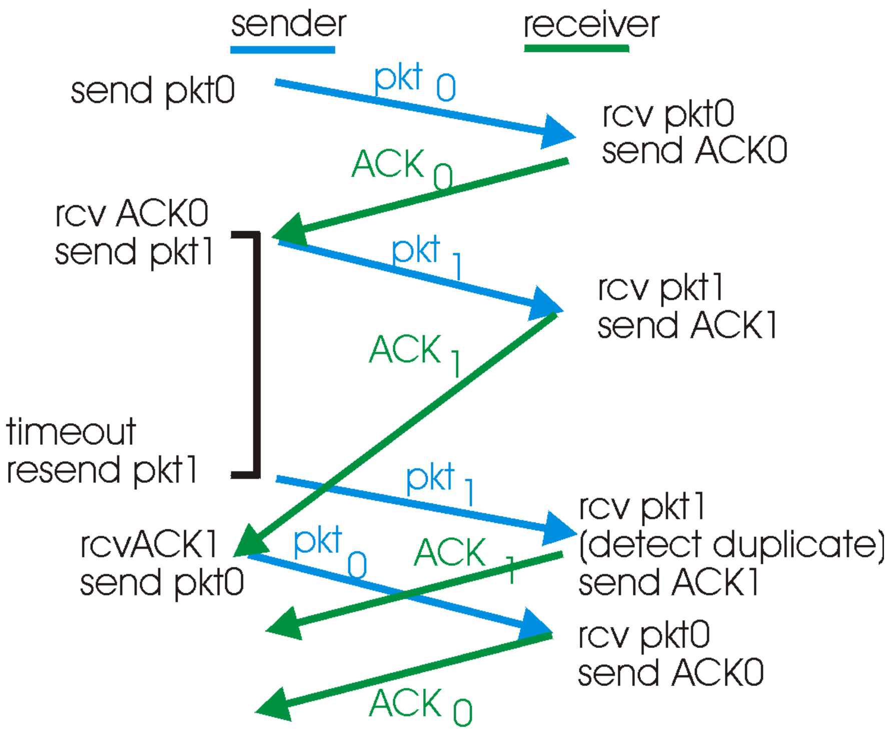
ACK 0

(b) lost packet

# rdt3.0 in action



(c) lost ACK



(d) premature timeout

# Performance of rdt3.0

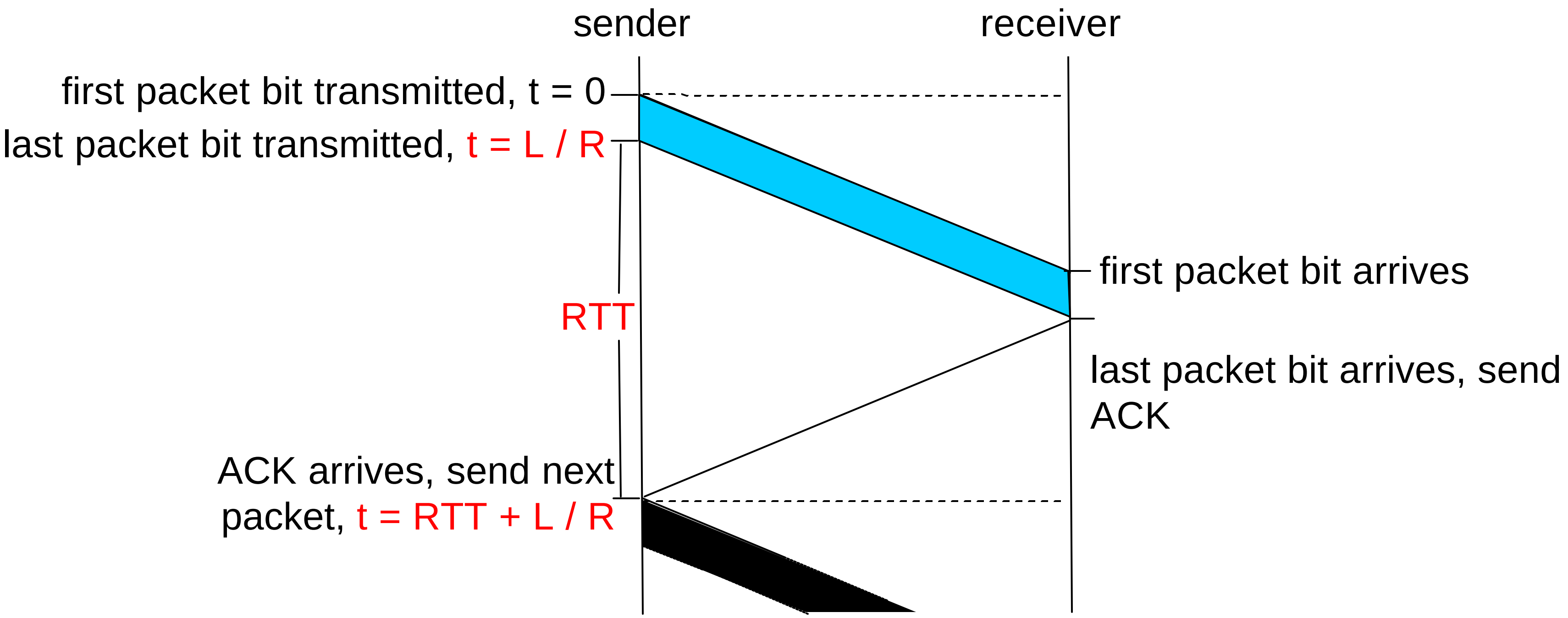
- rdt3.0 works, but performance is poor
- example: 1 Gbps link, 15 ms e-e prop. delay, 1KB packet:

$$T_{\text{transmit}} = \frac{\text{L (packet length in bits)}}{\text{R (transmission rate, bps)}} = \frac{\text{8kb/pkt}}{\text{10**9 b/sec}} = \text{8 microsec}$$

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

- $U_{\text{sender}}$: utilization – fraction of time sender busy sending
- 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

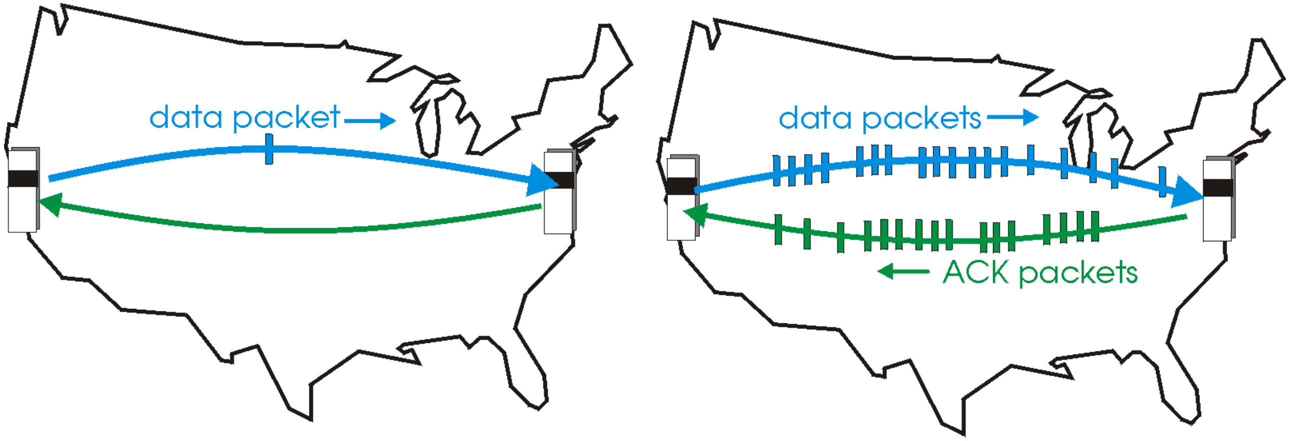

sender
receiver
first packet bit transmitted, t = 0
last packet bit transmitted, t = L / R
RTT
first packet bit arrives
last packet bit arrives, send ACK
ACK arrives, send next packet, t = RTT + L / R

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

# Pipelined protocols

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

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

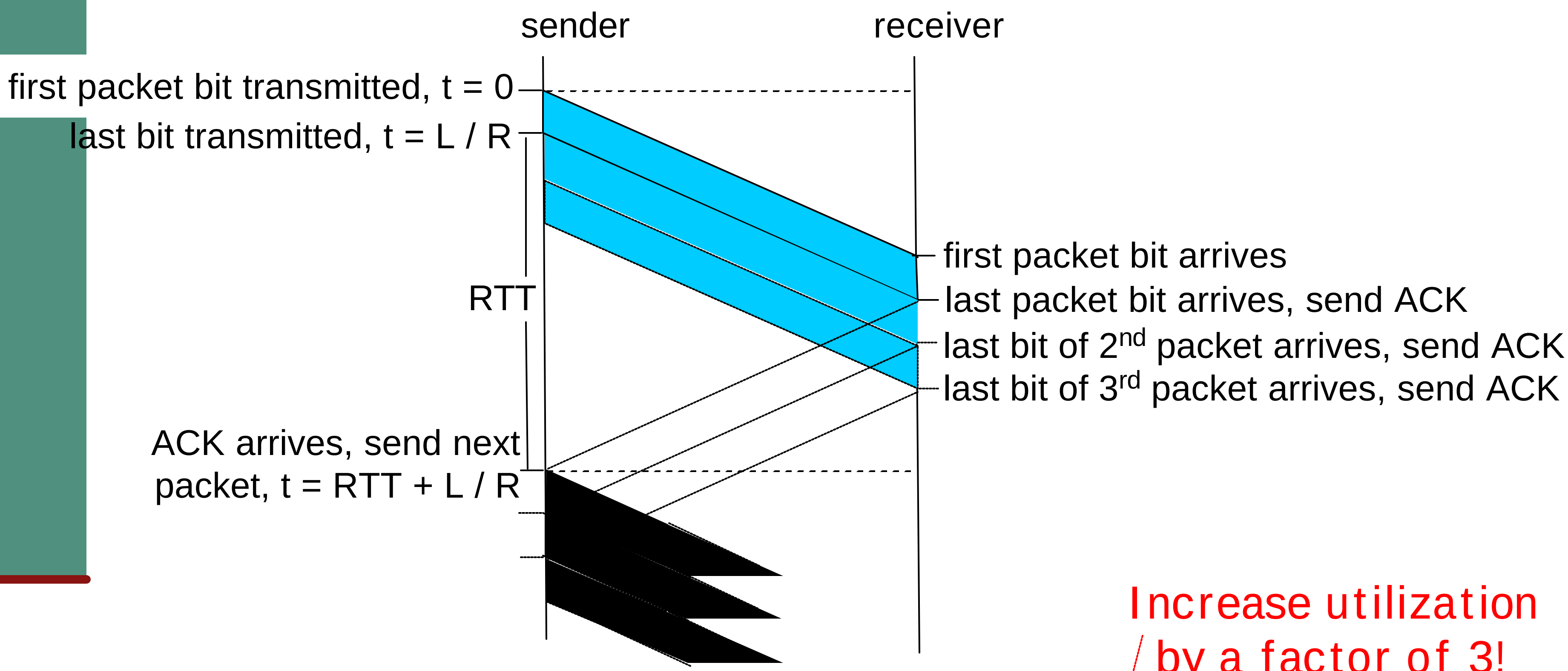


(a) a stop-and-wait protocol in operation

(b) a pipelined protocol in operation

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

# Pipelining: increased utilization



sender

receiver

first packet bit transmitted, t = 0

last bit transmitted, t = L / R

RTT

first packet bit arrives

last packet bit arrives, send ACK

last bit of 2nd packet arrives, send ACK

last bit of 3rd packet arrives, send ACK

ACK arrives, send next
packet, t = RTT + L / R

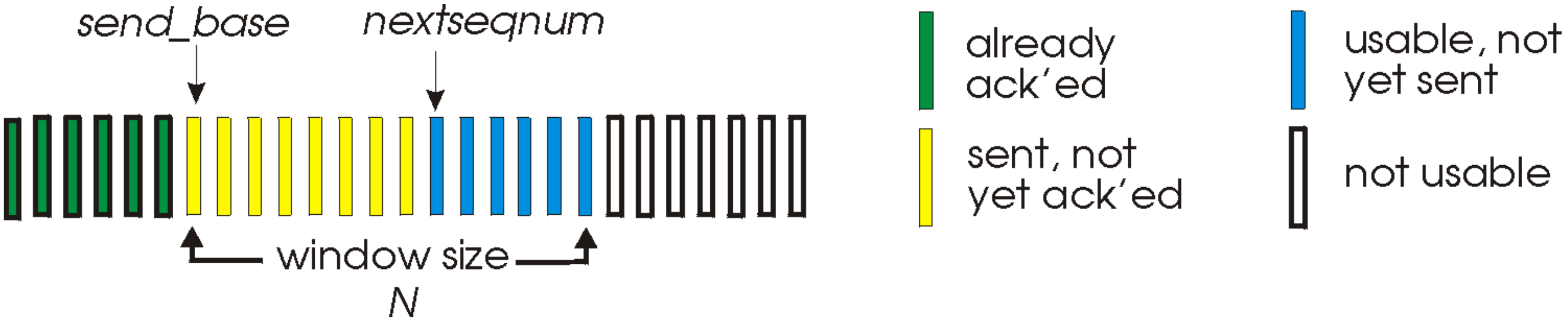
$$U_{\text{sender}} = \frac{3 * L / R}{RTT + L / R} = \frac{.024}{30.008} = 0.0008$$

Increase utilization
by a factor of 3!

# Go-Back-N

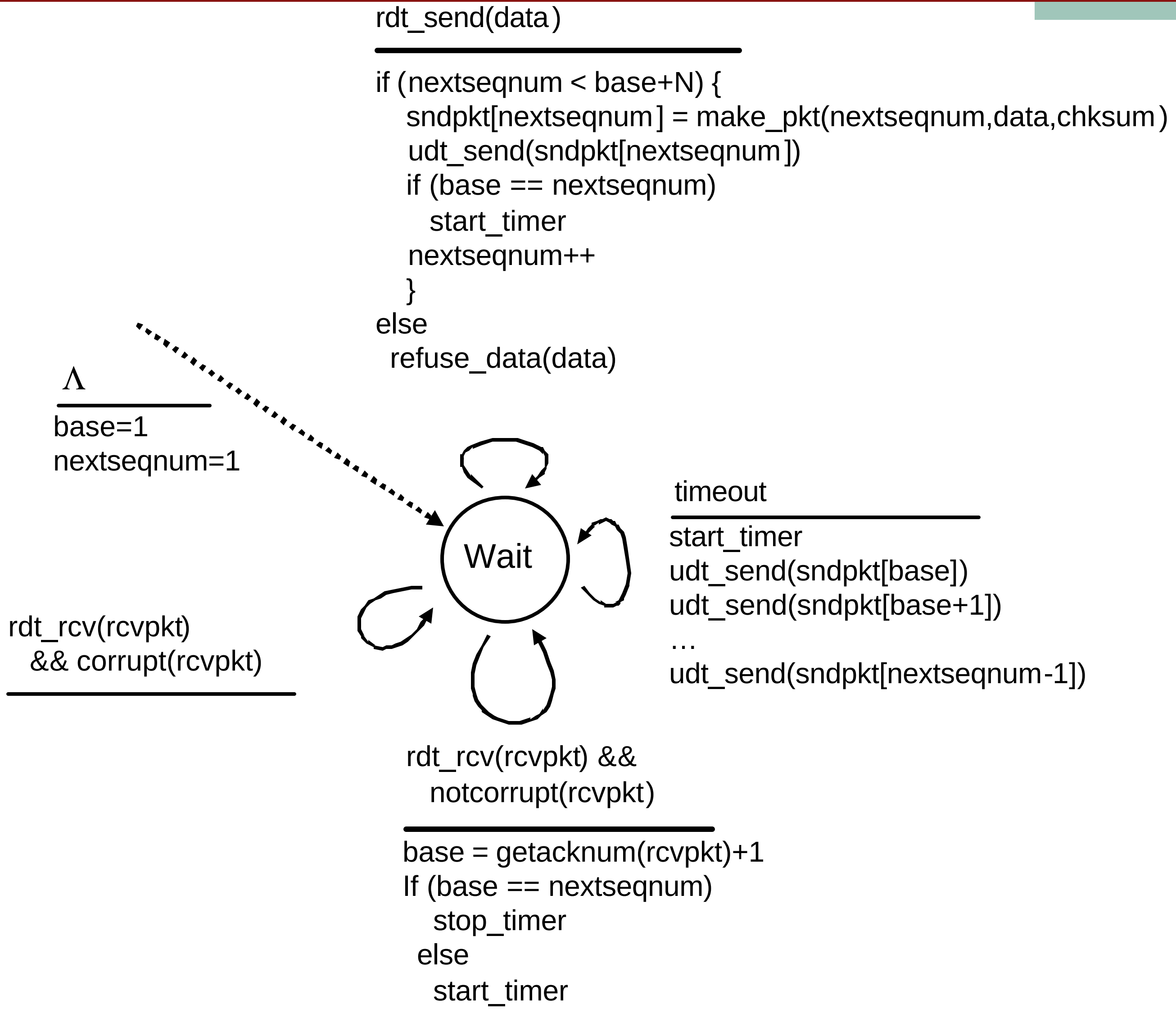
Sender:

- 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 deceive duplicate ACKs (see receiver)
- timer for in-flight pkts
- *timeout(n):* retransmit pkt 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++
    }
else
  refuse_data(data)
```

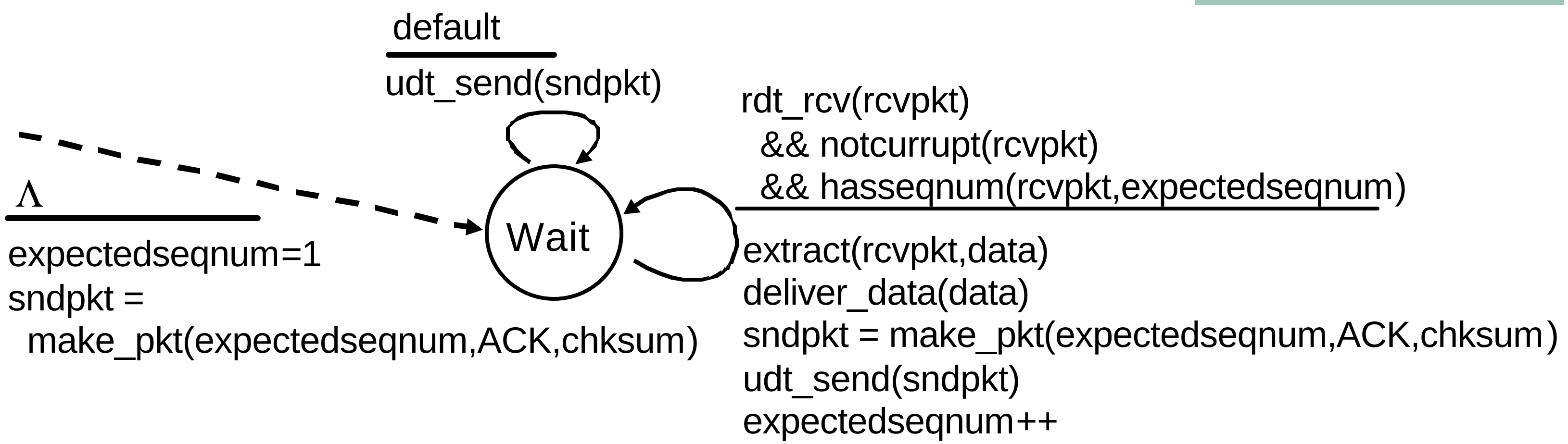
```
Λ
________________
base=1
nextseqnum=1
```

```
timeout
________________________________
start_timer
udt_send(sndpkt[base])
udt_send(sndpkt[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



default

udt_send(sndpkt)

Λ

expectedseqnum=1
sndpkt =
make_pkt(expectedseqnum,ACK,chksum)

Wait

rdt_rcv(rcvpkt)
&& notcurrupt(rcvpkt)
&& hasseqnum(rcvpkt,expectedseqnum)

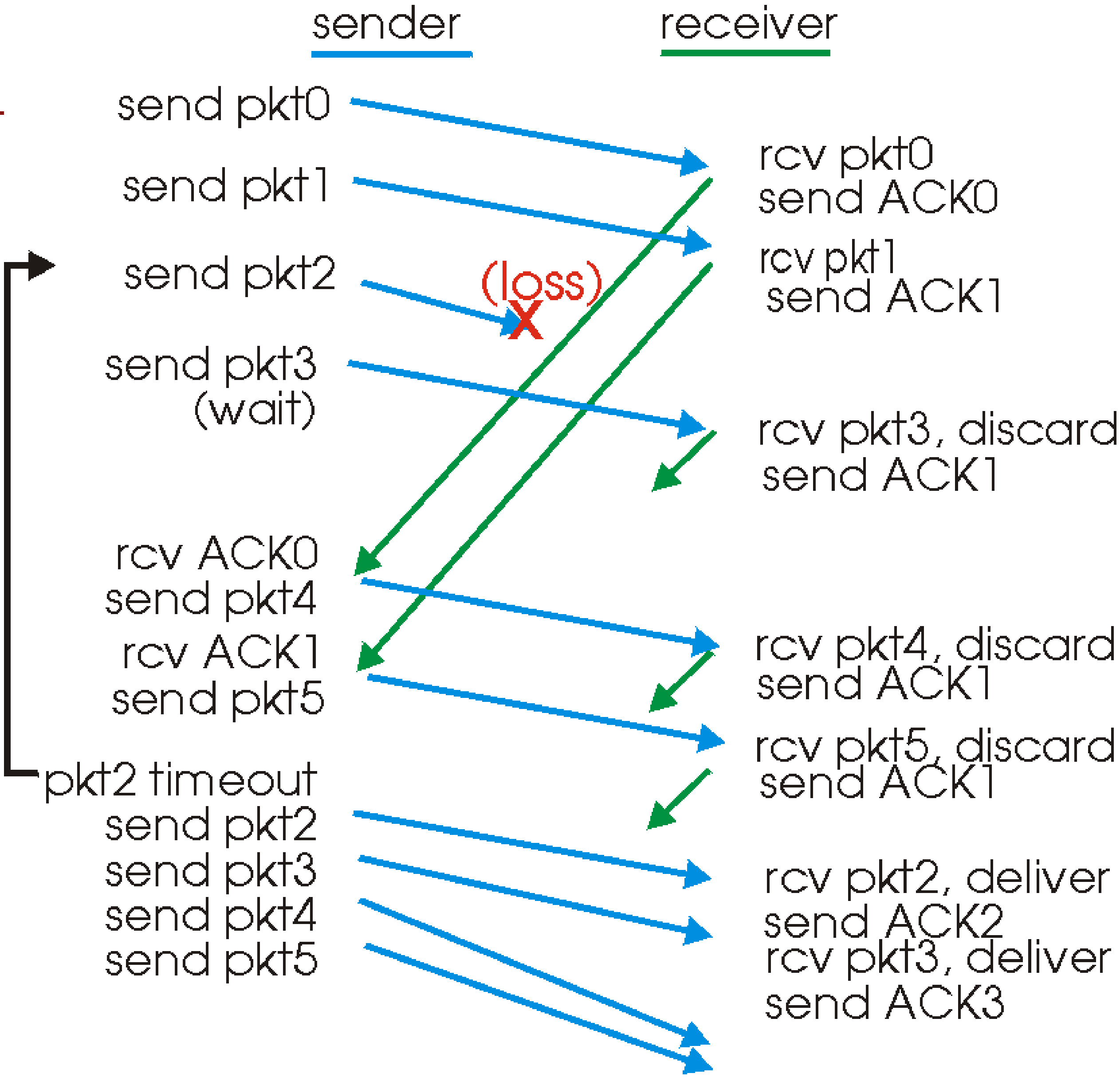
extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(expectedseqnum,ACK,chksum)
udt_send(sndpkt)
expectedseqnum++

ACK-only: always send ACK for correctly-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



sender

receiver

send pkt0

rcv pkt0
send ACK0

send pkt1

rcv pkt1
send ACK1

send pkt2

(loss)
X

send pkt3
(wait)

rcv pkt3, discard
send ACK1

rcv ACK0
send pkt4

rcv pkt4, discard
send ACK1

rcv ACK1
send pkt5

rcv pkt5, discard
send ACK1

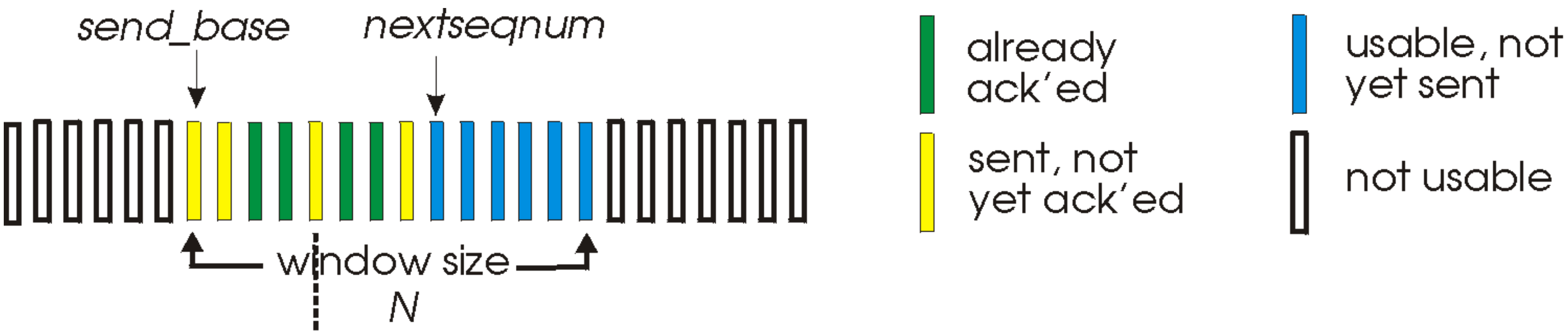
pkt2 timeout
send pkt2
send pkt3
send pkt4
send pkt5

rcv pkt2, deliver
send ACK2
rcv pkt3, deliver
send ACK3

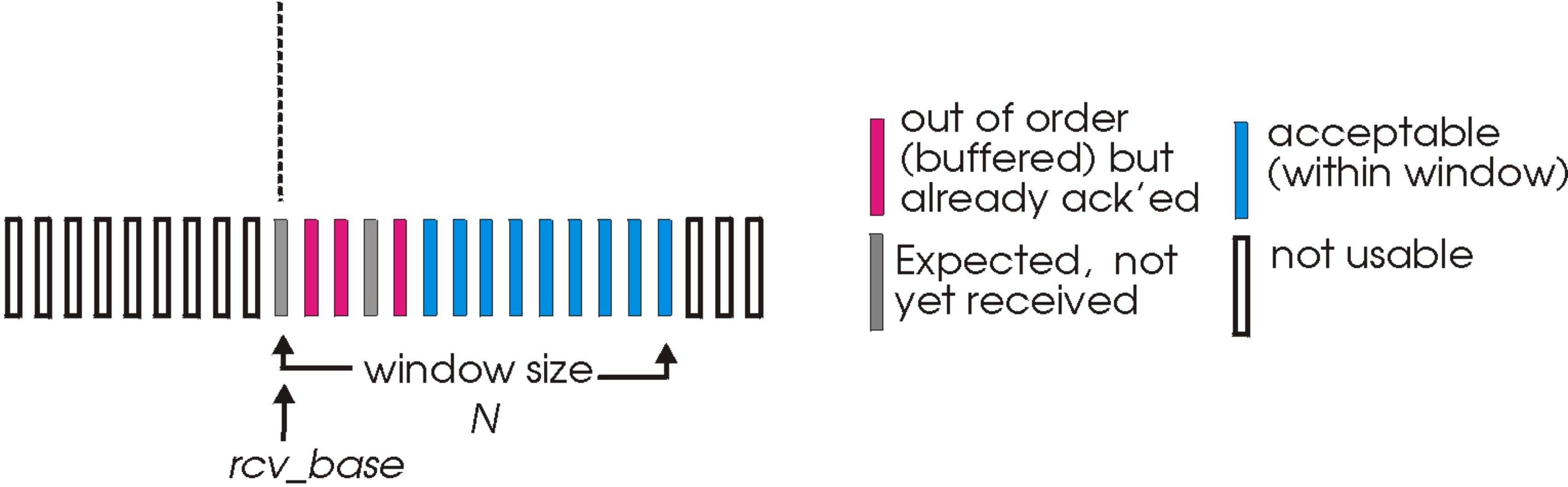
# Selective Repeat

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

# Selective repeat: sender, receiver windows



(a) sender view of sequence numbers


out of order (buffered) but already ack'ed
acceptable (within window)
Expected, not yet received
not usable
window size
N
rcv_base

(b) receiver view of sequence numbers

# Selective repeat

## sender

data from above :

- if next available seq # in window, send pkt

timeout(n):

- resend pkt n, restart timer

ACK(n) in [sendbase,sendbase+N]:

- mark pkt n as received
- if n smallest unACKed pkt, advance window base to next unACKed seq #

## receiver

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

- send ACK(n)
- out-of-order: buffer
- in-order: deliver (also deliver buffered, in-order pkts), advance window to next not-yet-received pkt

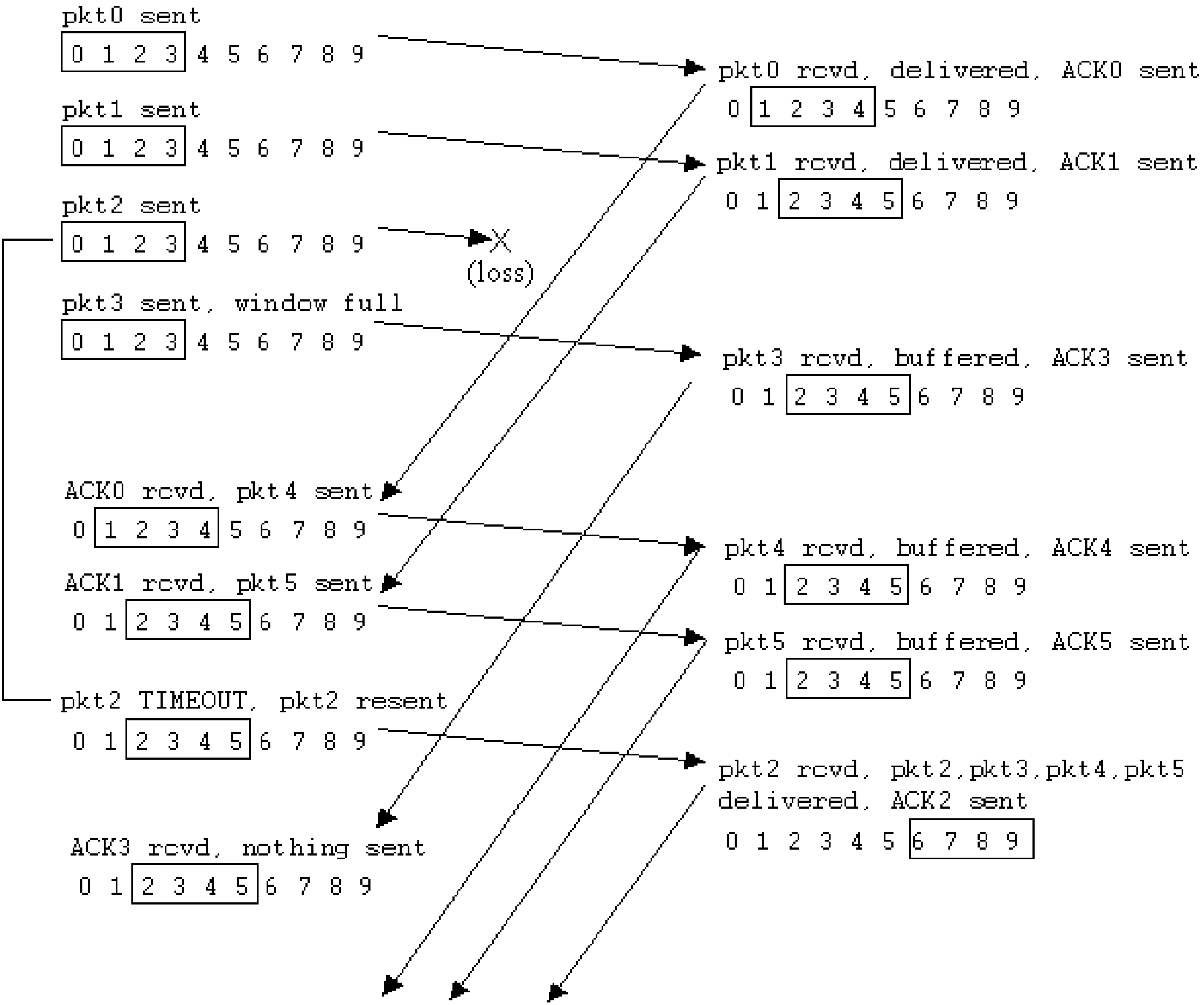
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore

# Selective repeat in action



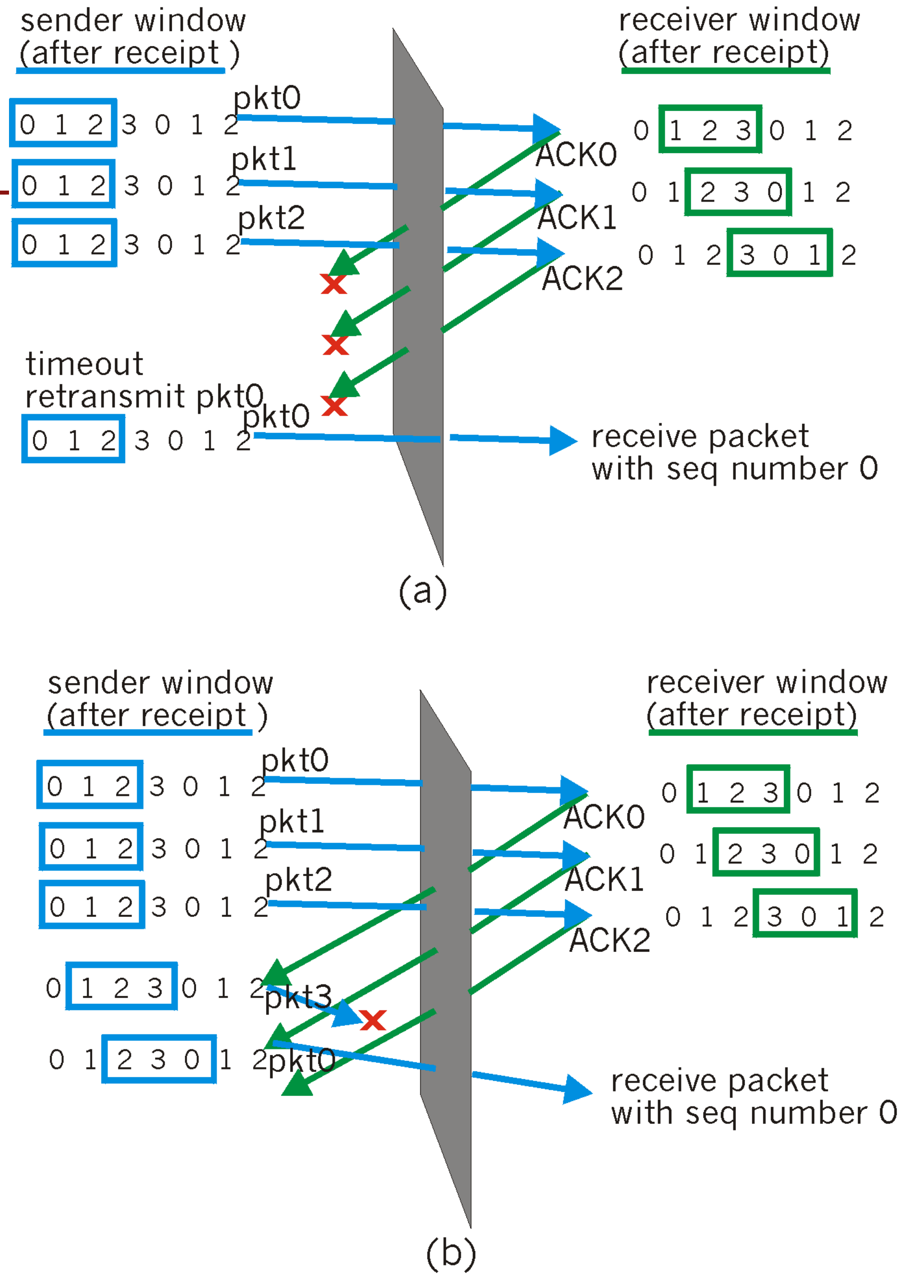
# Selective repeat: dilemma

Example:

- seq #'s: 0, 1, 2, 3
- window size=3

- receiver sees no difference in two scenarios!
- incorrectly passes duplicate data as new in (a)

Q: what relationship between seq # size and window size?



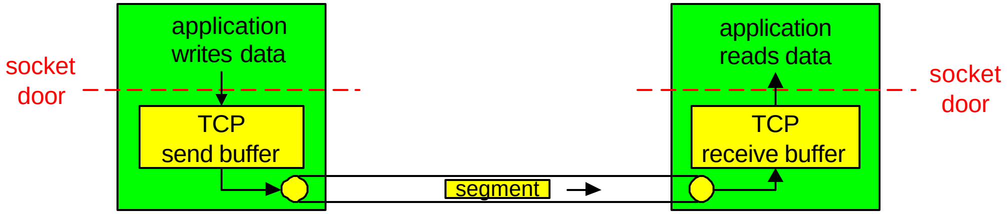
# Chapter 3 outline

- 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

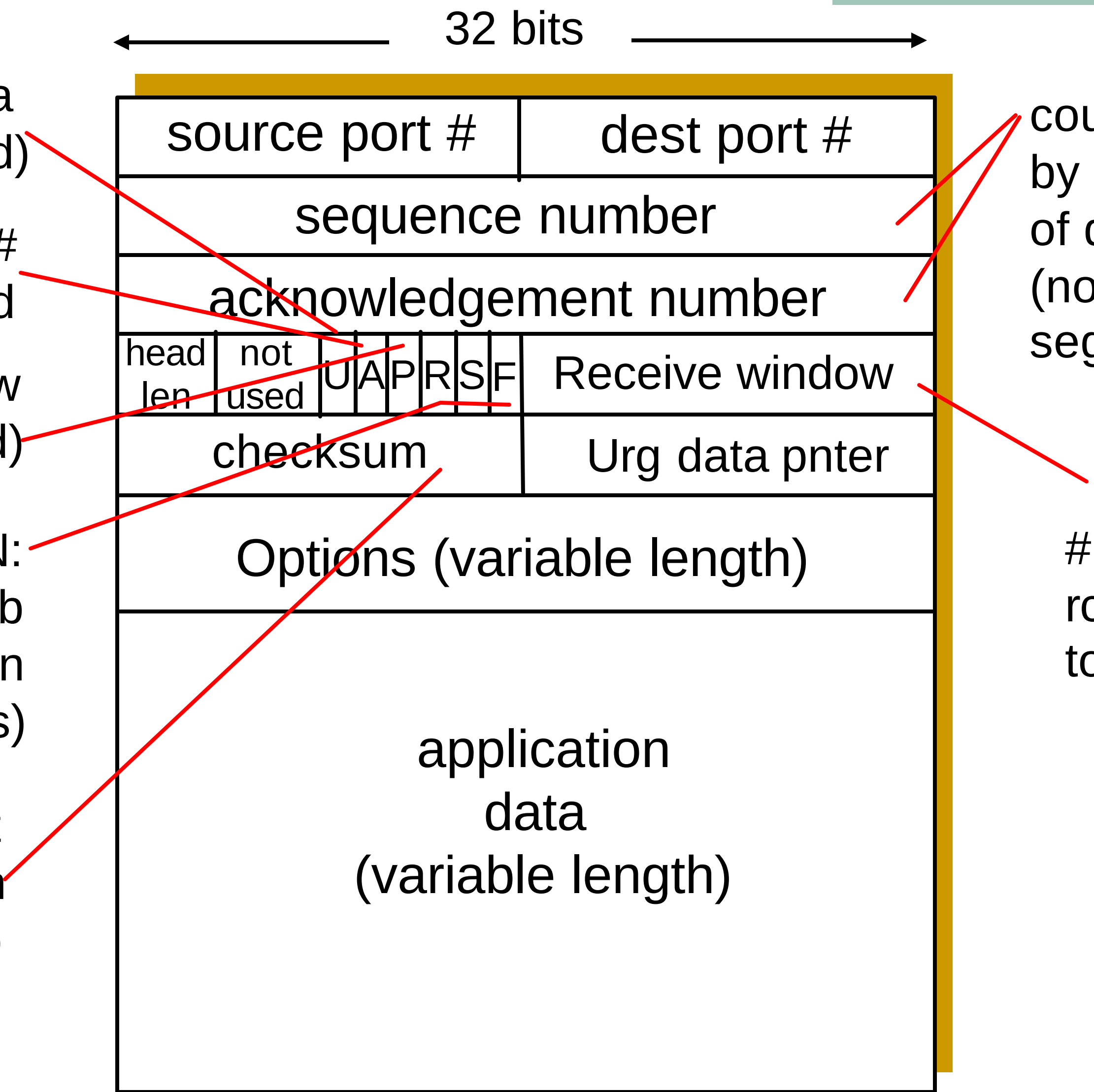
RFCs: 793, 1122, 1323, 2018, 2581

- 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

# TCP segment structure



32 bits

source port # | dest port #

sequence number

acknowledgement number

head len | not used | U | A | P | R | S | F | Receive window

checksum | Urg data pnter

Options (variable length)

application data (variable length)

URG: urgent data (generally not used)

ACK: ACK # valid

PSH: push data now (generally not used)

RST, SYN, FIN: connection estab (setup, teardown commands)

Internet checksum (as in UDP)

counting by bytes of data (not segments!)

# bytes rcvr willing to accept

# TCP seq. #’s and ACKs

Seq. #’s:

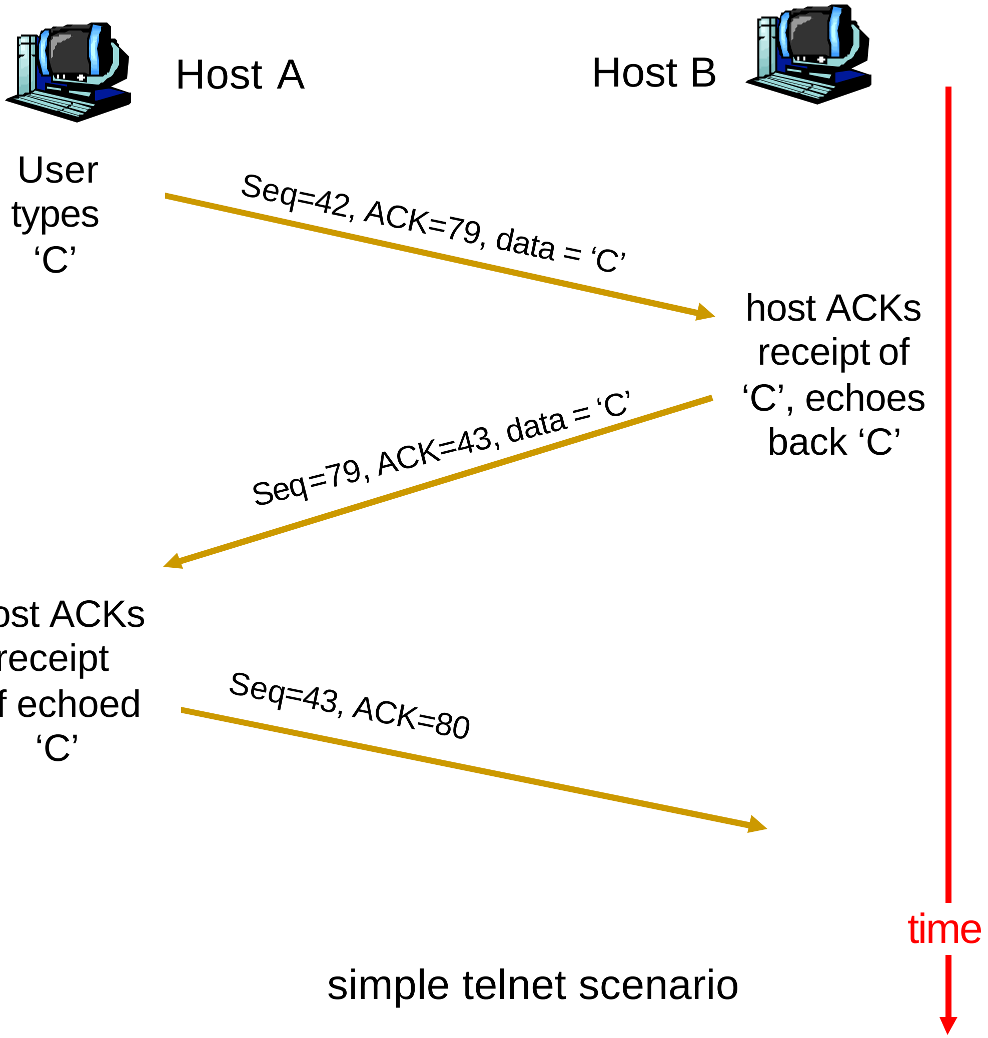
- byte stream “number” of first byte in segment’s data

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



Host A

Host B

User types ‘C’

Seq=42, ACK=79, data = ‘C’

host ACKs receipt of ‘C’, echoes back ‘C’

Seq=79, ACK=43, data = ‘C’

host ACKs receipt of echoed ‘C’

Seq=43, ACK=80

time

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?

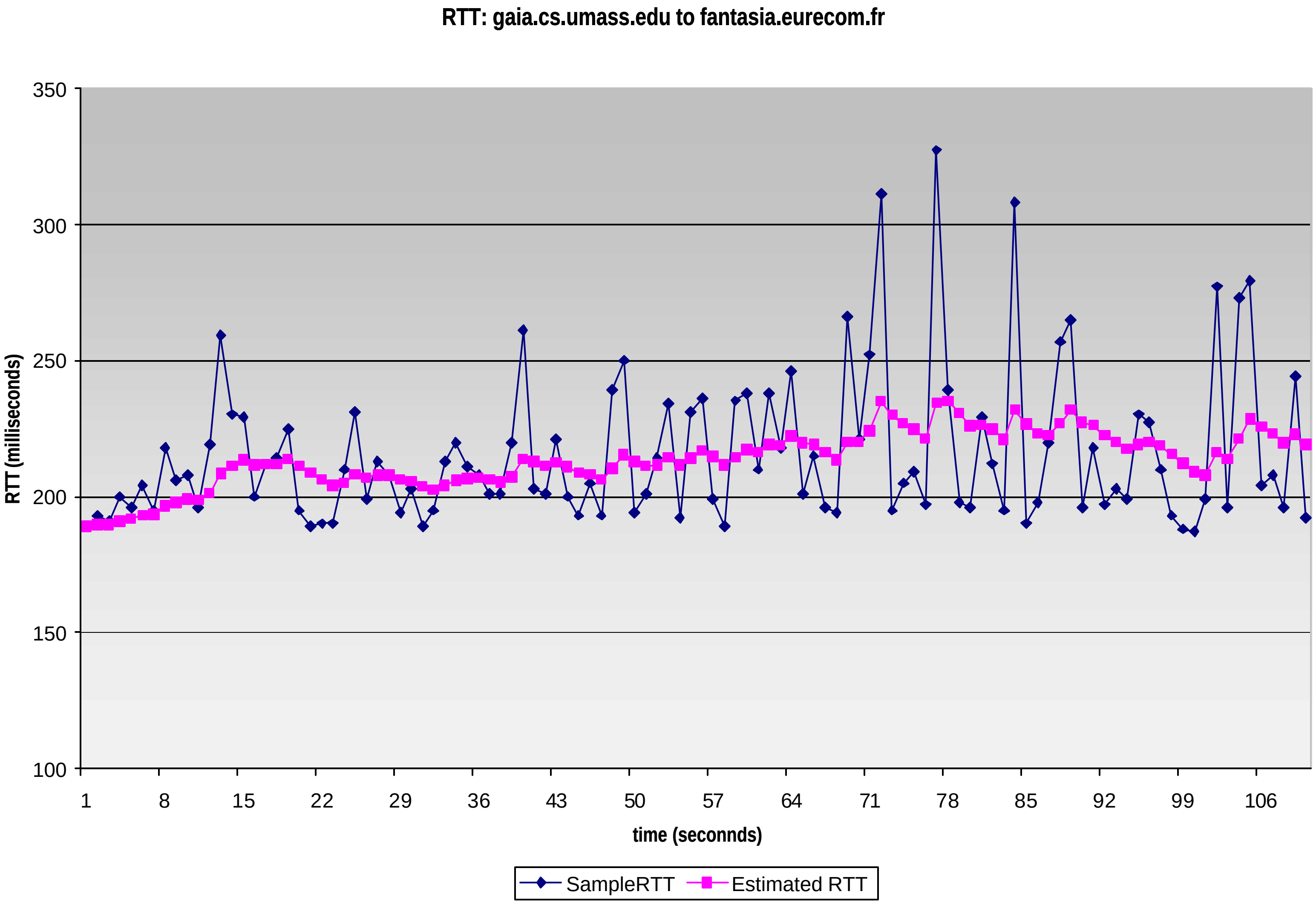
- **`SampleRTT`**: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- **`SampleRTT`** will vary, want estimated RTT “smoother”
  - average several recent measurements, not just current **`SampleRTT`**

# TCP Round Trip Time and Timeout

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

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

# Example RTT estimation:



# TCP Round Trip Time and Timeout

## Setting the timeout

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

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

**(**typically**, b = 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 overview

- 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 (simplified)

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:

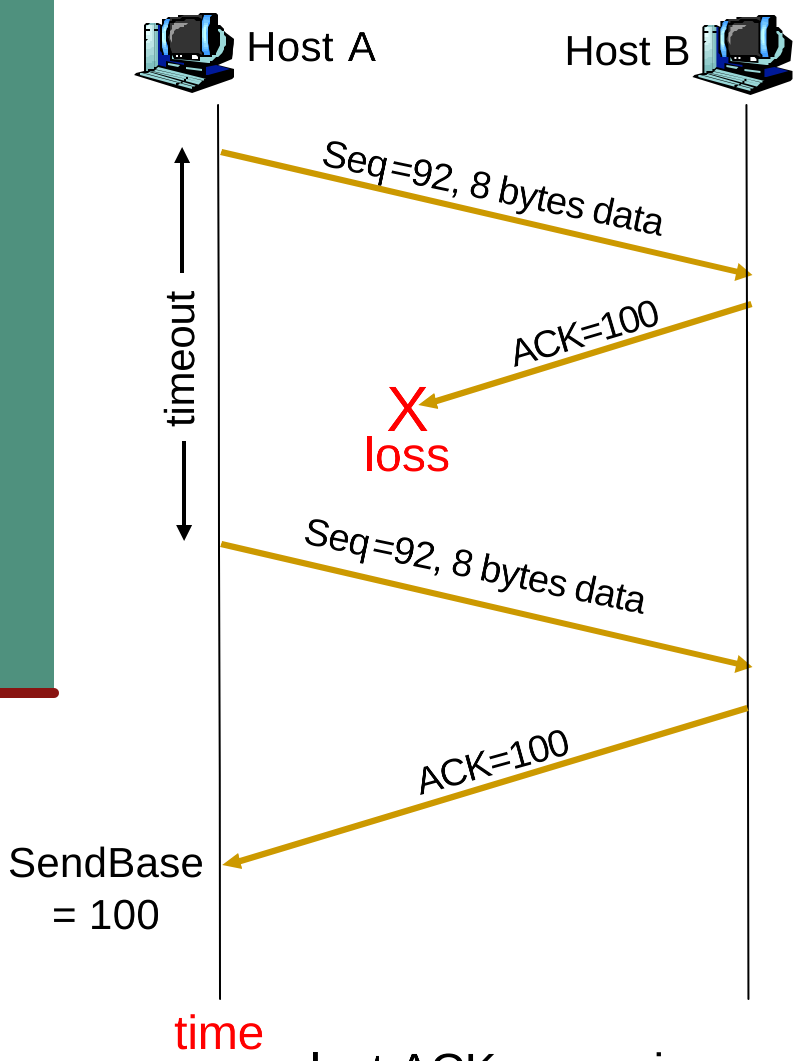
- If acknowledges previously unacked segments
  - update what is known to be acked
  - start timer if there are outstanding segments

# TCP sender

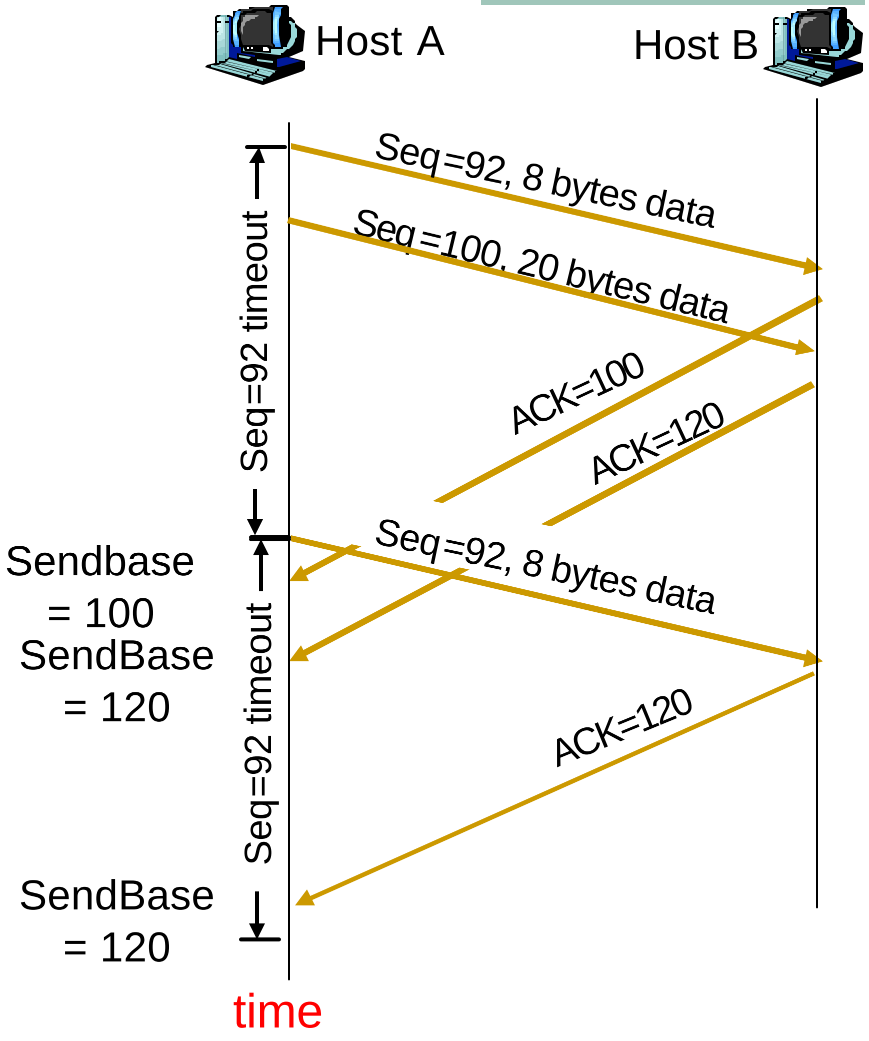
(simplified)

```
NextSeqNum = InitialSeqNum
SendBase = InitialSeqNum

loop (forever) {
 switch(event)

    event: data received from application above

           create TCP segment with seq# 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 */
```

# TCP: retransmission scenarios

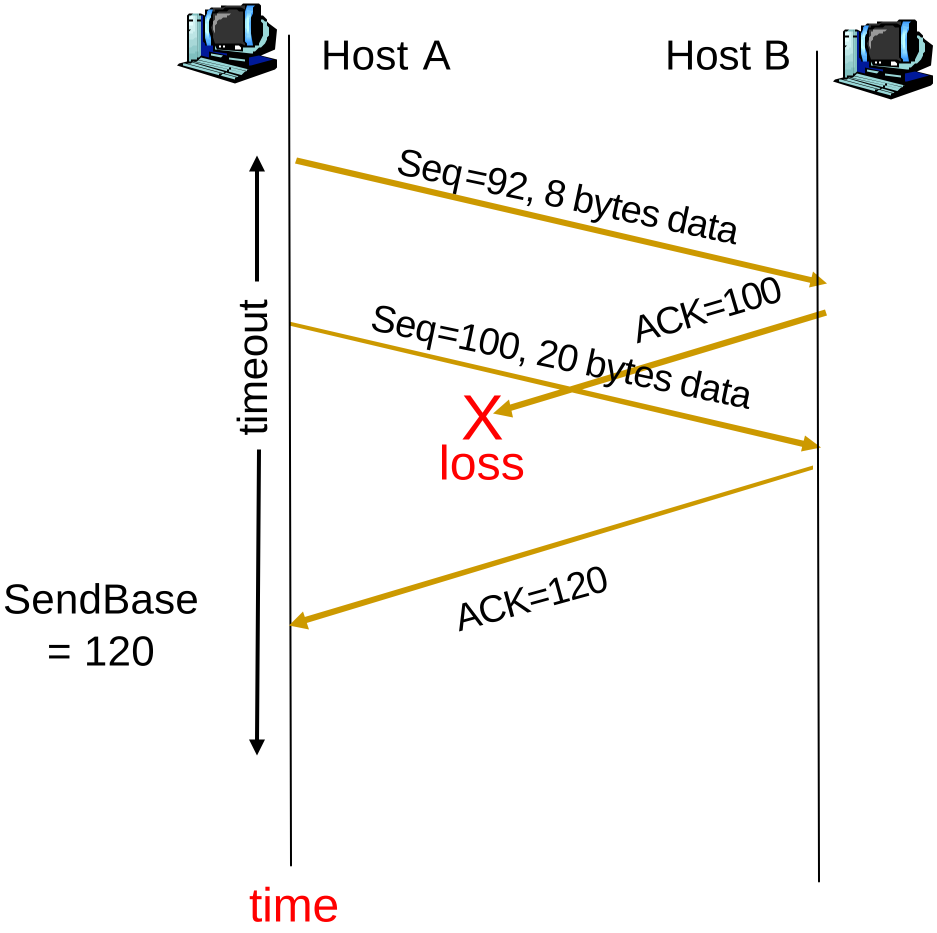


lost ACK scenario



premature timeout

# TCP retransmission scenarios (more)



Cumulative ACK scenario

# Modifications

1. Doubling the timeout interval
   - After a timeout event:
     - TCP retransmits the not yet ack’d segment with the smallest sequence number
     - TCP sets the next timeout interval to twice the previous value (rather than deriving it from **`EstimatedRTT`** and **`DevRTT`**)
   - Provides a limited form of congestion control
2. Fast retransmit

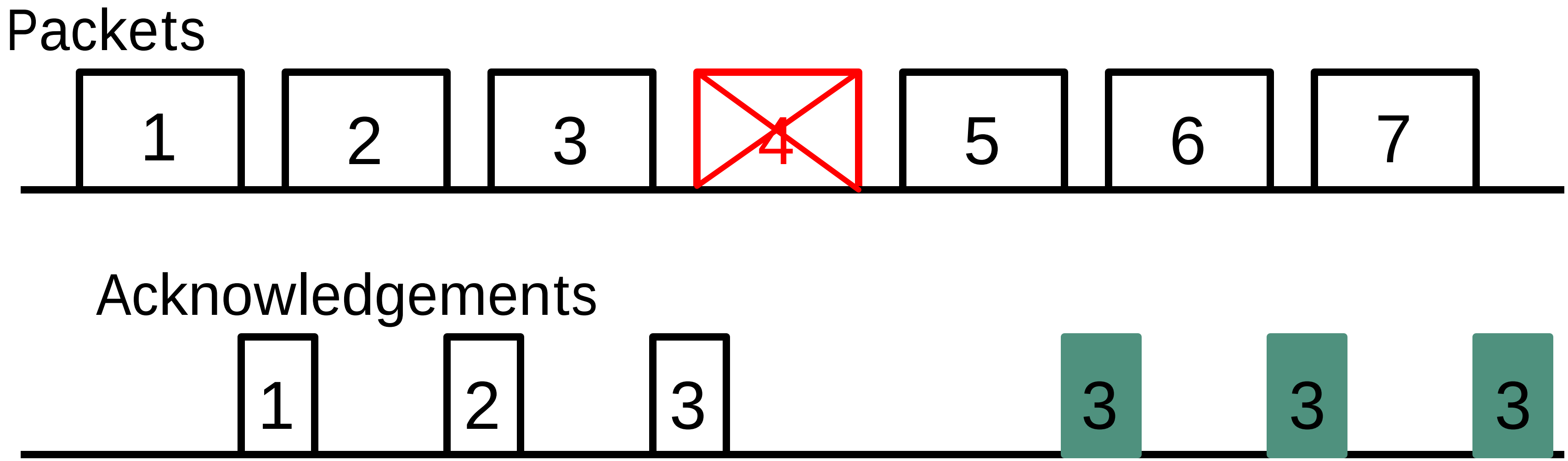
# Fast  Retransmit

- 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

# Packet Loss

Packet loss detected by

- Retransmission timeouts
- Duplicate ACKs (at least 3)



Packets

1 2 3 4 5 6 7

Acknowledgements

1 2 3 3 3 3

# 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

# Flow Control

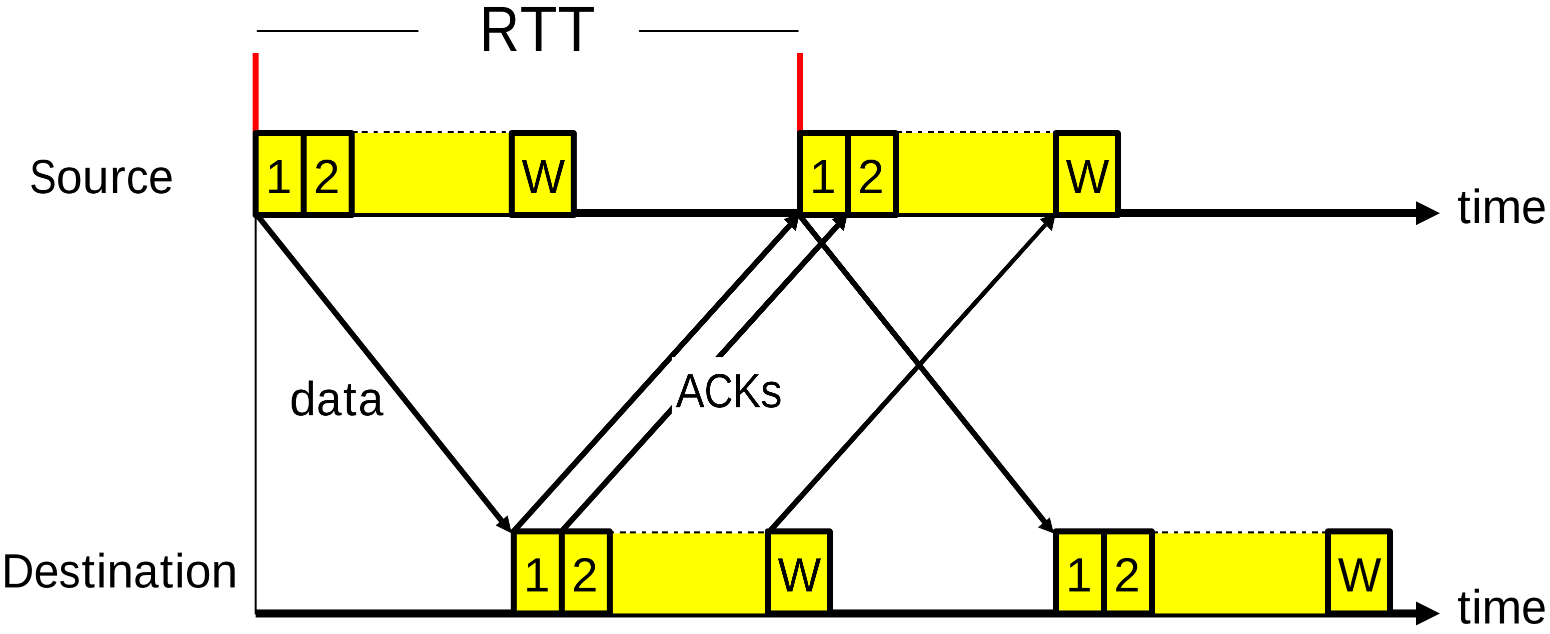
- speed-matching service: matching the send rate to the receiving app’s drain rate

**flow control**

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

- How can it be implemented?

# Window Flow Control



At most W packets per RTT

packet size <= MSS bytes

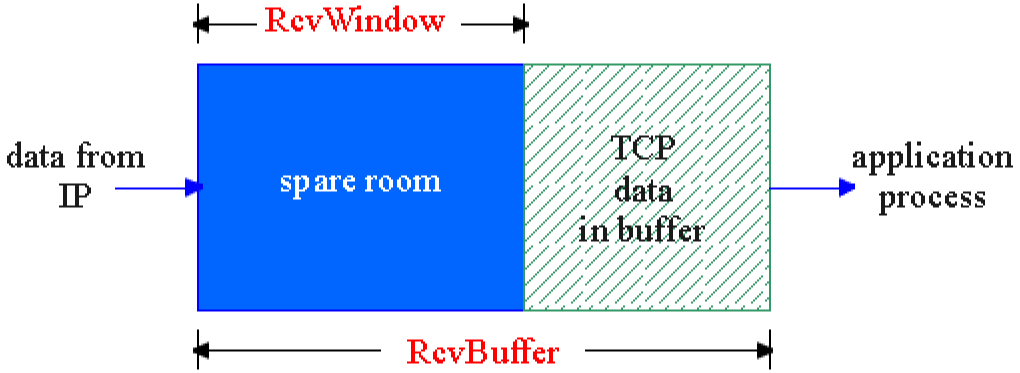
# Window Flow control

- Limit the number of packets in the network to window W
- Source rate = $\dfrac{W \times MSS}{RTT}$ bps
- Notes:
  - If W too small then rate « capacity (low utilization)
    If W too big then rate > capacity => congestion
  - Solution: Adapt W to network (and conditions)

W x MSS = BW x RTT

# TCP Flow Control

- receive side of TCP connection has a receive buffer:



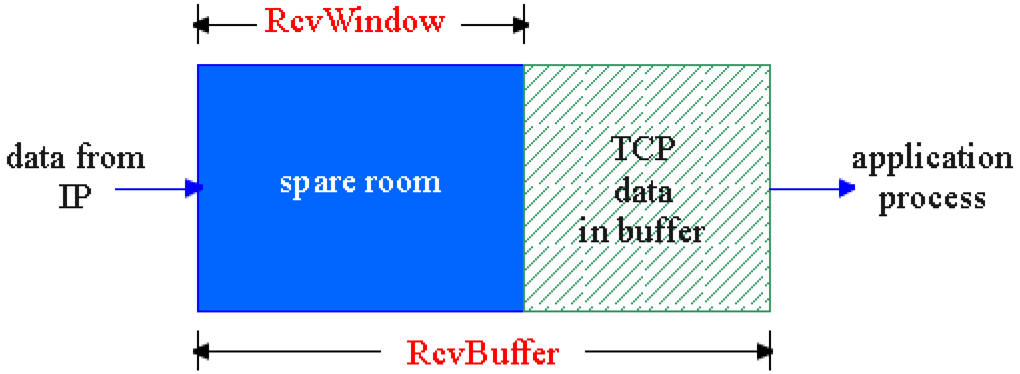
- app process may be slow at reading from buffer

flow control

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

- 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.
- During this phase, TCP initializes variables:
  - Sequence numbers
  - Buffers: flow control information (**`RcvWindow`**)
- TCP also provides mechanisms to close connections

# TCP Three-way Handshake

Step 1: client host sends TCP SYN segment to server

- specifies initial seq #
- no data

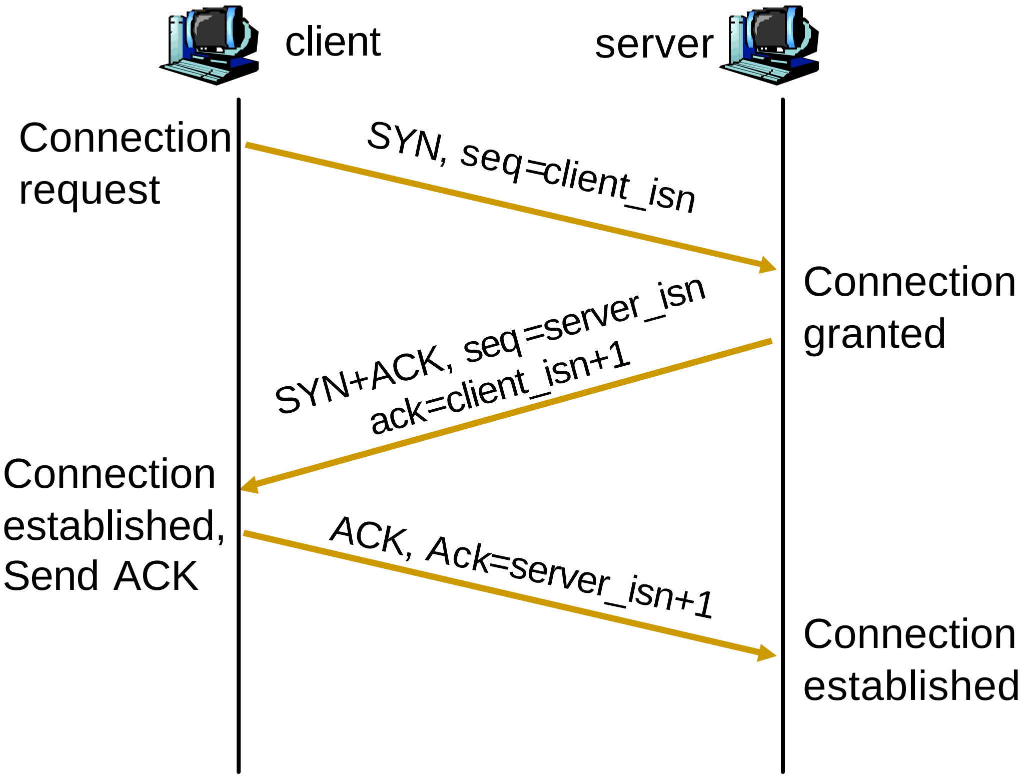
Step 2: server host receives SYN, replies with SYNACK segment

- server allocates buffers
- specifies server initial sequence number
- No data

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

*server:*
```
Socket connectionSocket =
welcomeSocket.accept();
```

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



client

server

Connection request

SYN, seq=client_isn

Connection granted

SYN+ACK, seq=server_isn
ack=client_isn+1

Connection established, Send ACK

ACK, Ack=server_isn+1

Connection established

# Closing a TCP Connection

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

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

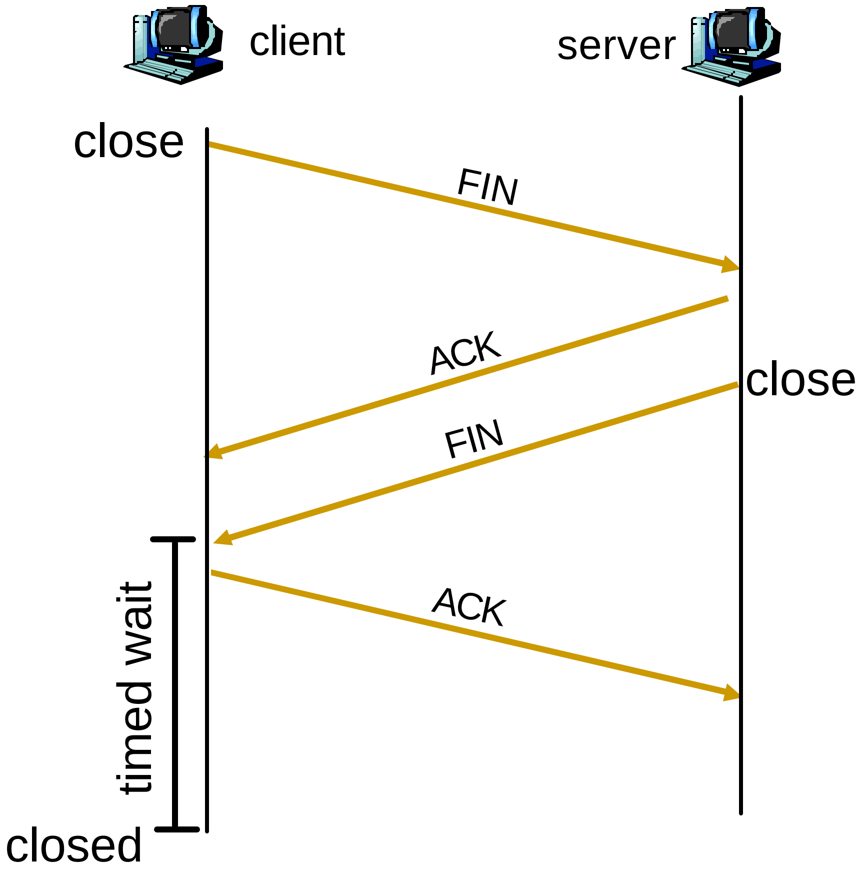
Step 3: client receives FIN, replies with ACK.

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

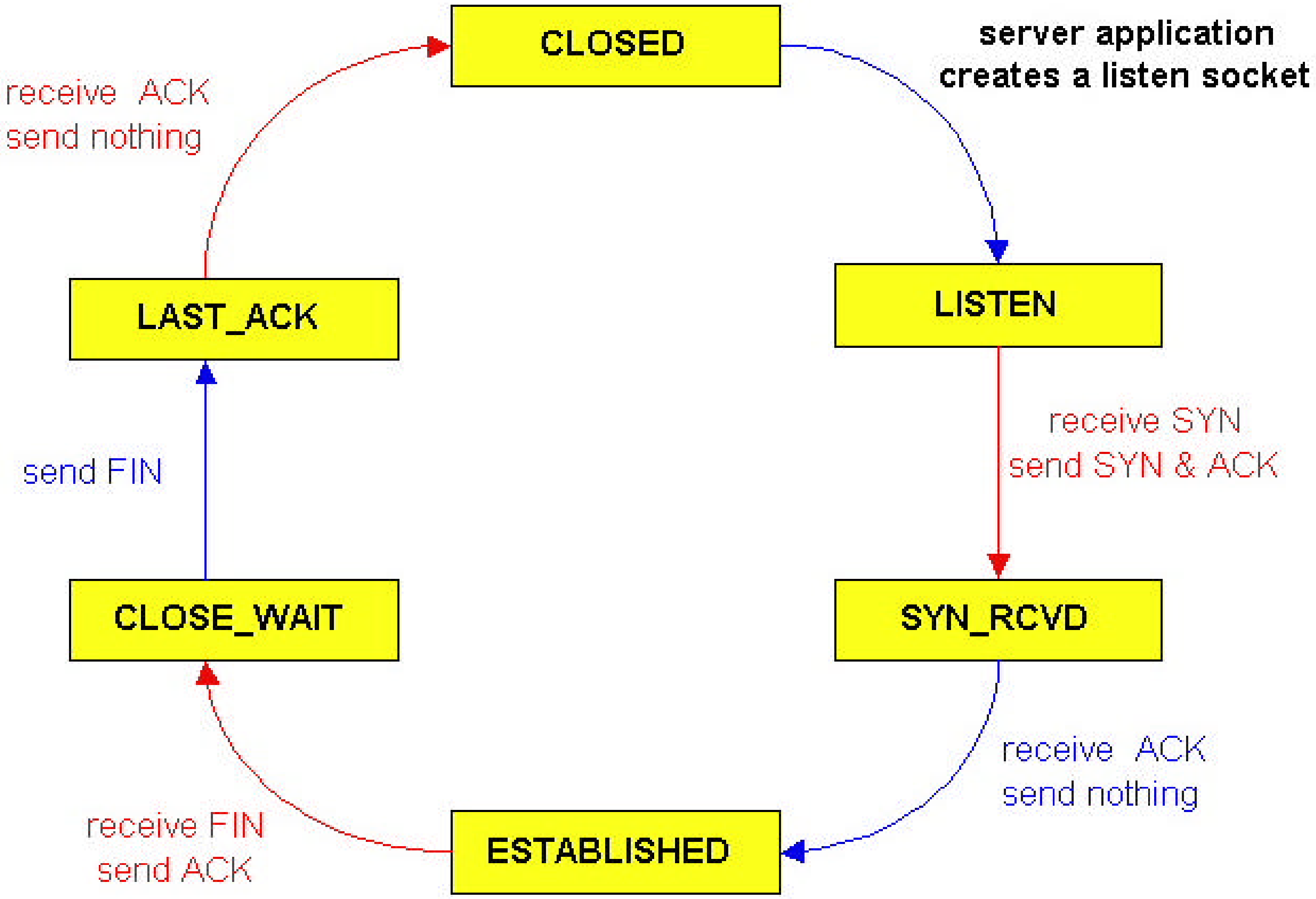
Step 4: server, receives ACK. Connection closed.

client closes socket:

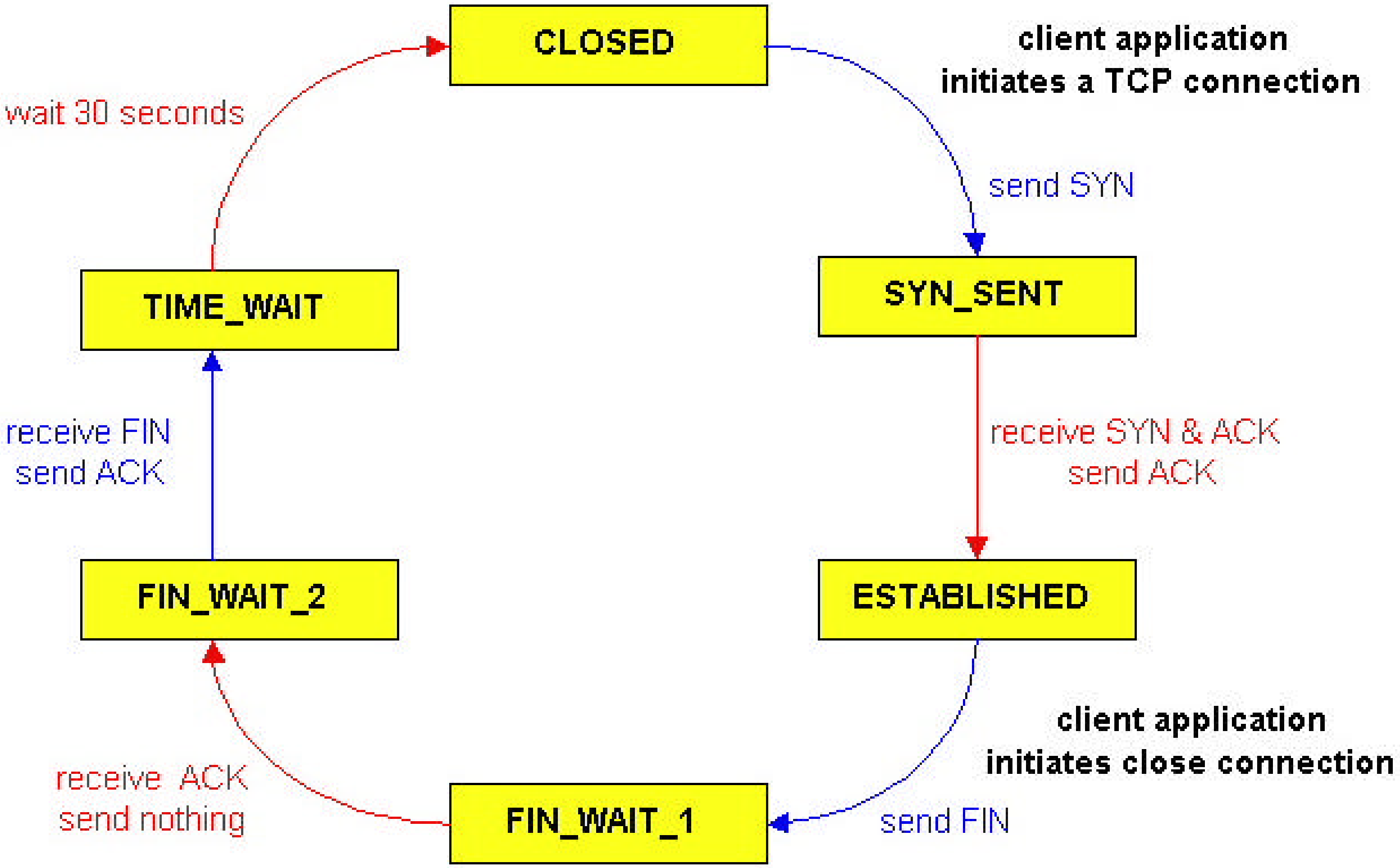
```
clientSocket.close();
```



# Typical TCP Server Lifecycle



# Typical TCP Client Lifecycle



# Chapter 3 outline

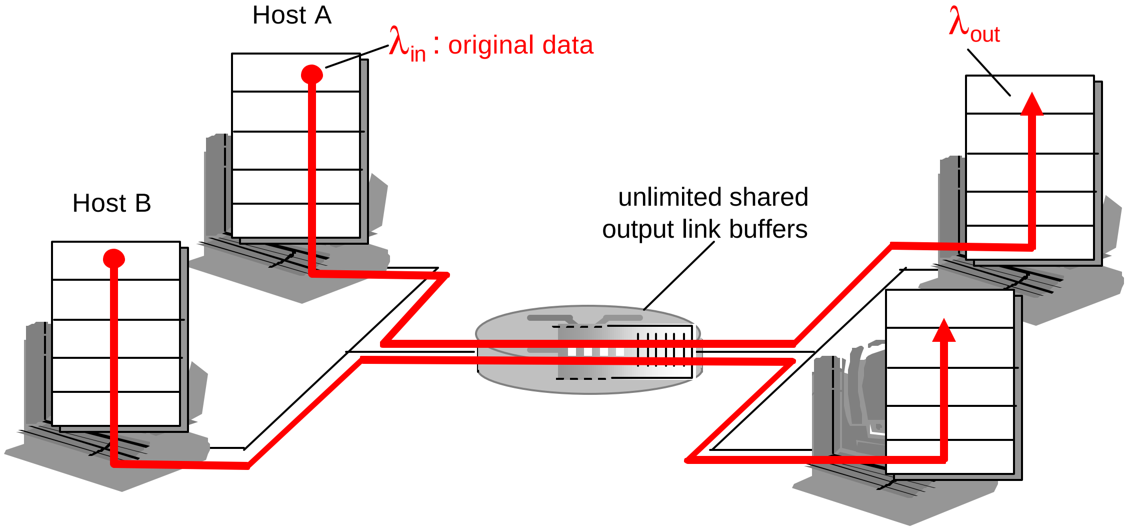
- 3.1 Transport-layer services
- 3.2 Multiplexing and demultiplexing
- 3.3 Connectionless transport: UDP
- 3.4 Principles of reliable data transfer
- 3.5 Connection-oriented transport: TCP
  - segment structure
  - reliable data transfer
  - flow control
  - connection management
- 3.6 Principles of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

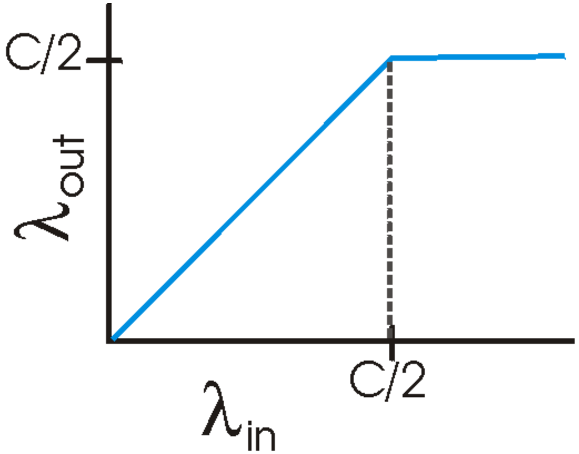
# Network Congestion

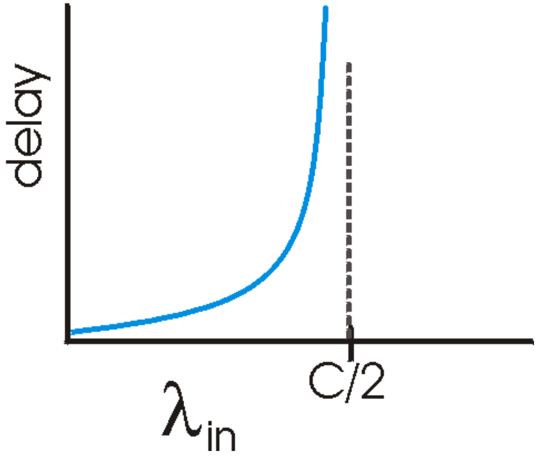
- Informally: “too many sources sending too much data too fast for *network* to handle”
- Different from flow control! (why?)
- Effects of congestion:
  - Packet loss (buffer overflow at routers)
  - Retransmissions
  - Reduced throughput
  - long delays (queueing in router buffers)
  - Network collapse:
    - Unnecessarily retransmitted packets
    - Undelivered or unusable packets

# Causes of congestion: scenario 1

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


Host A
$\lambda_{in}$ : original data
Host B
unlimited shared output link buffers
$\lambda_{out}$

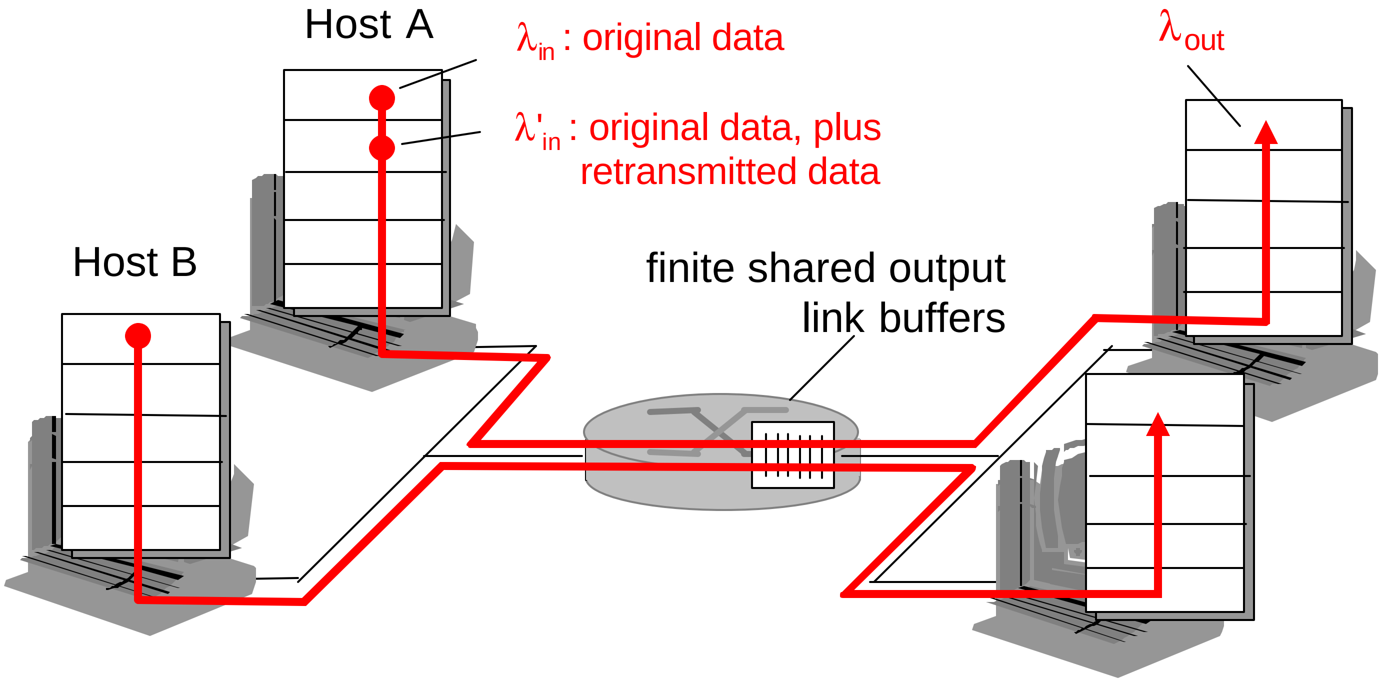




- large delays when congested
- maximum achievable throughput

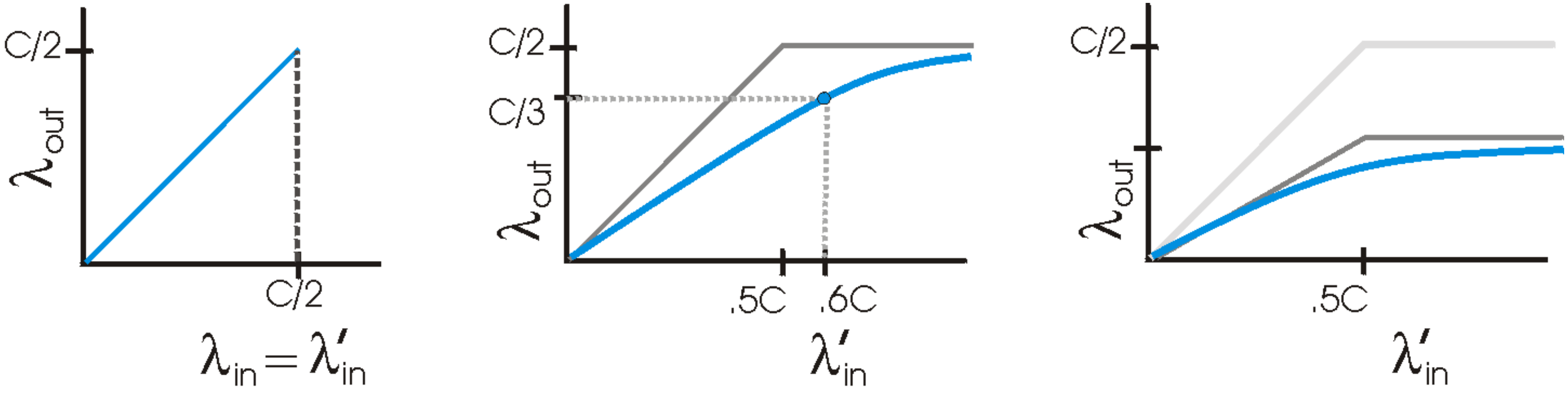
# Causes of congestion: scenario 2

- one router, *finite* buffers (packet loss!)
- sender retransmission of lost packets


Host A
$\lambda_{in}$ : original data
$\lambda'_{in}$ : original data, plus retransmitted data
$\lambda_{out}$
Host B
finite shared output link buffers

# Causes of congestion: scenario 2 (cont’d)

- $\lambda_{in} = \lambda_{out}$
- Retransmission when packet loss: $\lambda'_{in} > \lambda_{out}$
- Retransmission of delayed (not lost) packet makes $\lambda'_{in}$ even larger

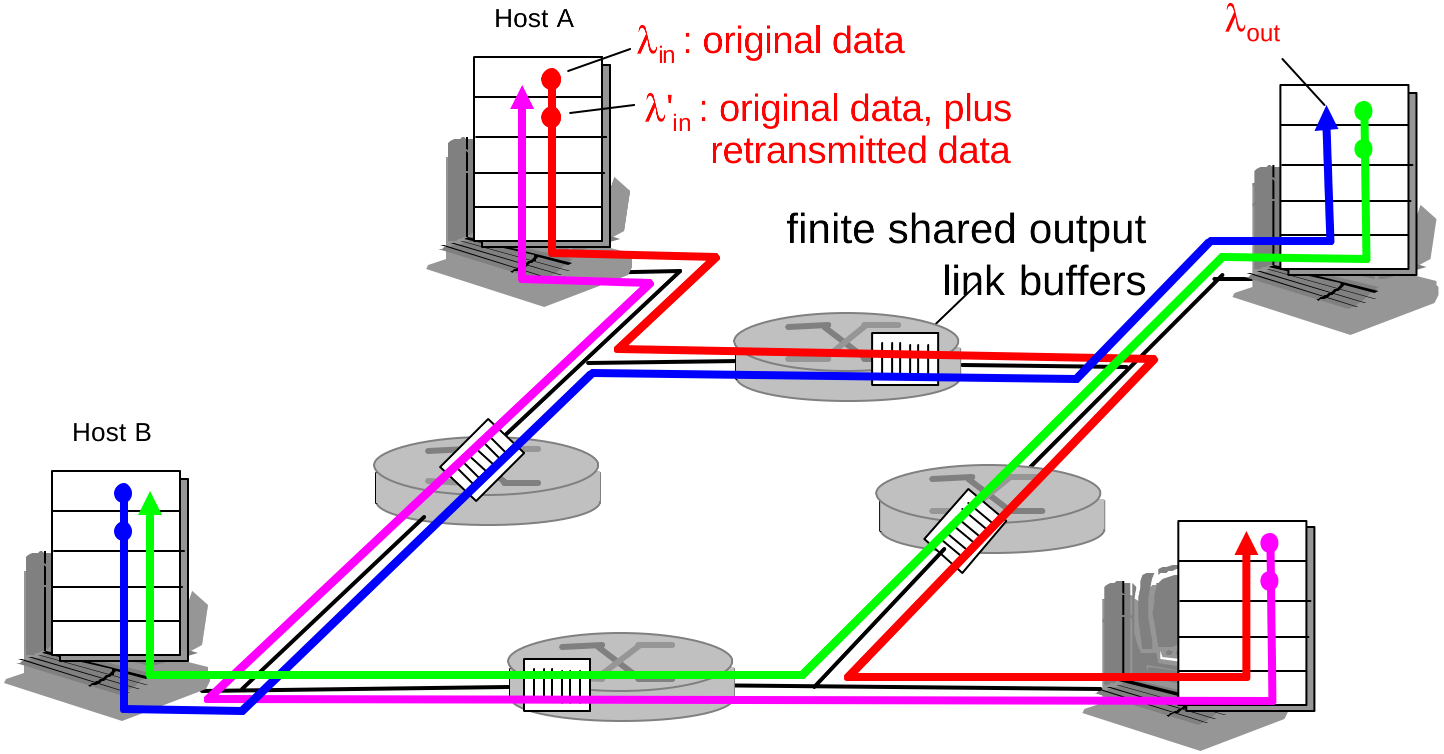


(a) (b) (c)

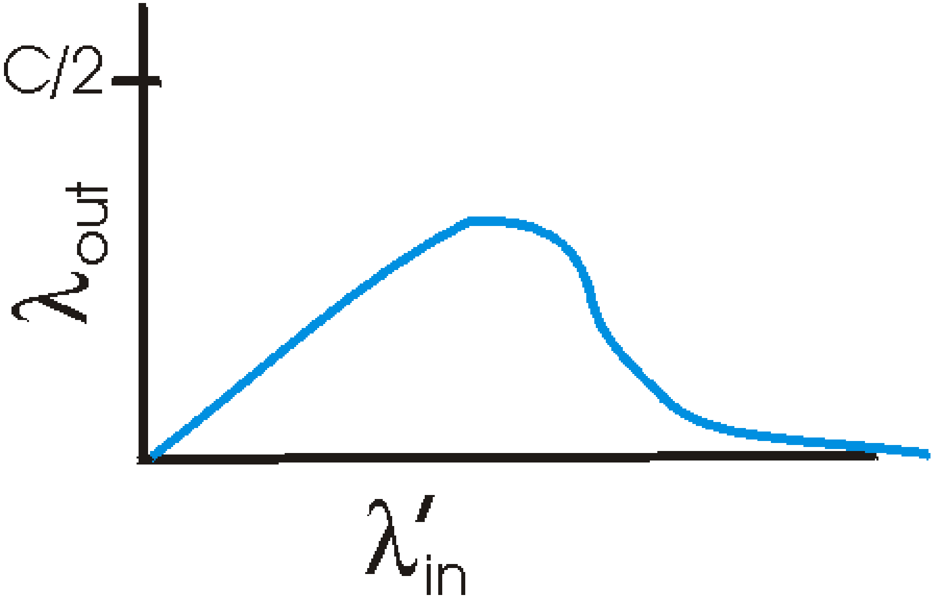
# Causes of congestion: scenario 3

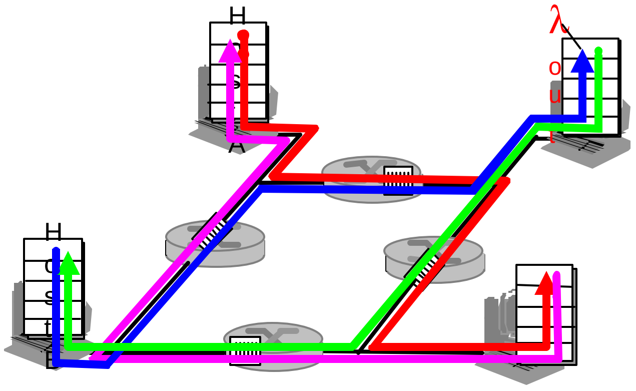
- four senders
- multihop paths
- timeout/retransmission

Q: what happens as $\lambda_{in}$ and $\lambda'_{in}$ increase ?


Host A
$\lambda_{in}$ : original data
$\lambda'_{in}$ : original data, plus retransmitted data
$\lambda_{out}$
finite shared output link buffers
Host B

# Causes/cost of congestion: scenario 3





when a packet is dropped, any “upstream transmission capacity used for that packet is 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:

- To sender (Choke packet), or
- To receiver and from there to sender. E.g. single bit indicating congestion (SNA, DECbit, TCP/IP ECN, ATM)

# TCP Congestion Control

- end-end control (no network assistance)
- sender limits transmission:

  **`LastByteSent-LastByteAcked`**
  **`£ CongWin`**

- Roughly,

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

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

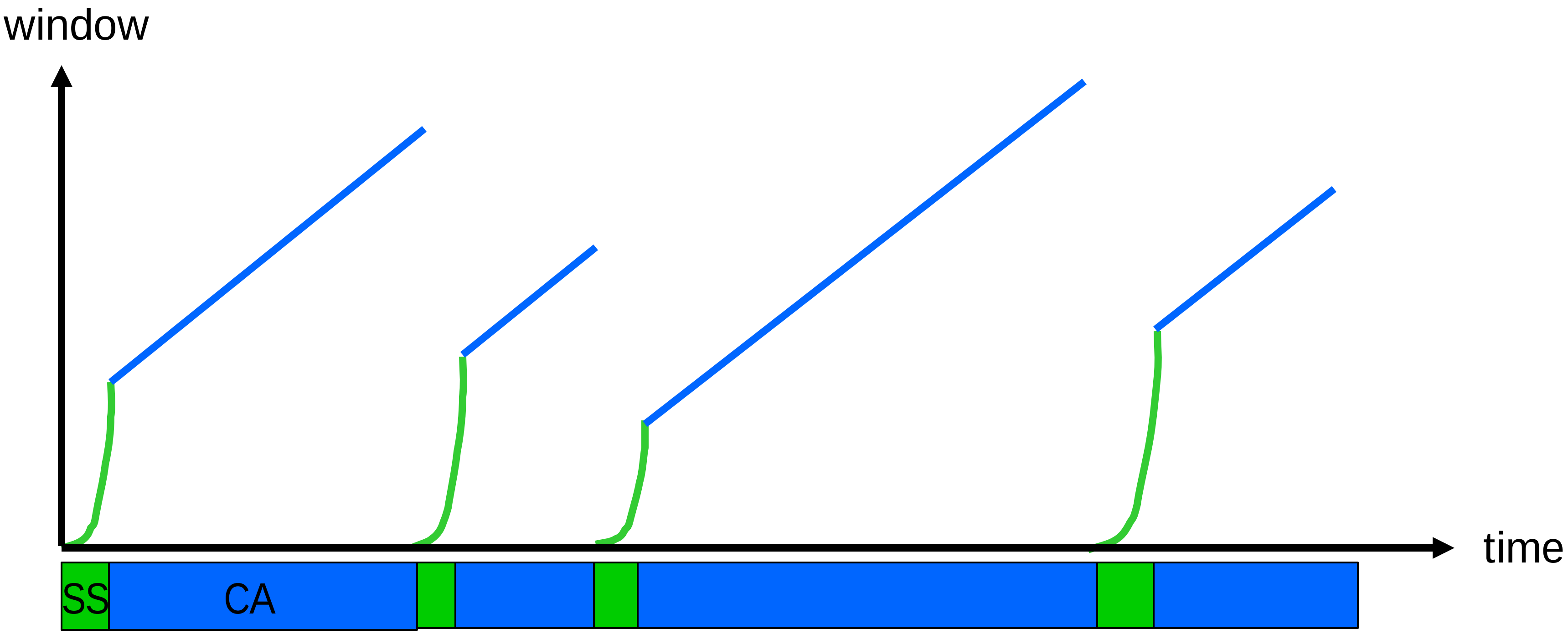
How does sender perceive congestion?

- loss event = timeout *or* 3 duplicate acks

Two mechanisms:

- Slow start
- Congestion Avoidance:
  - AIMD
  - Reaction to timeout events

# TCP Tahoe (Jacobson 1988)

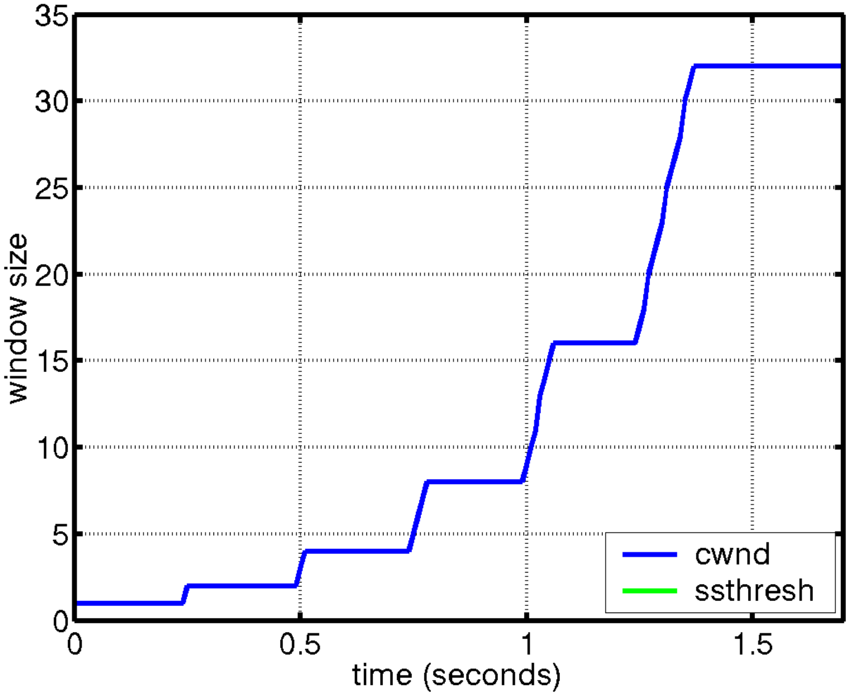


SS: Slow Start

CA: Congestion Avoidance

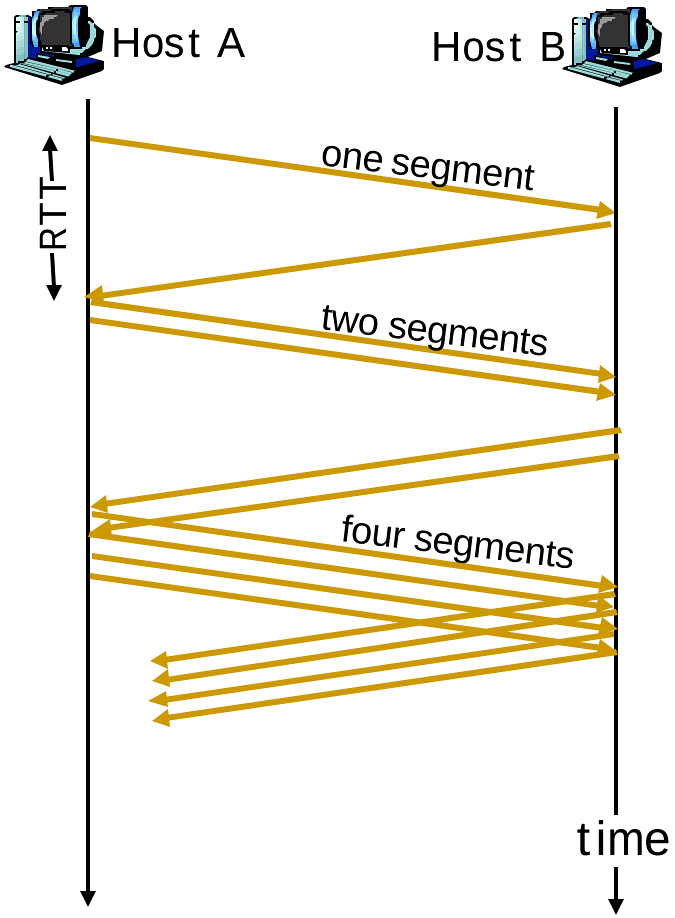
# TCP Slow Start

- 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)

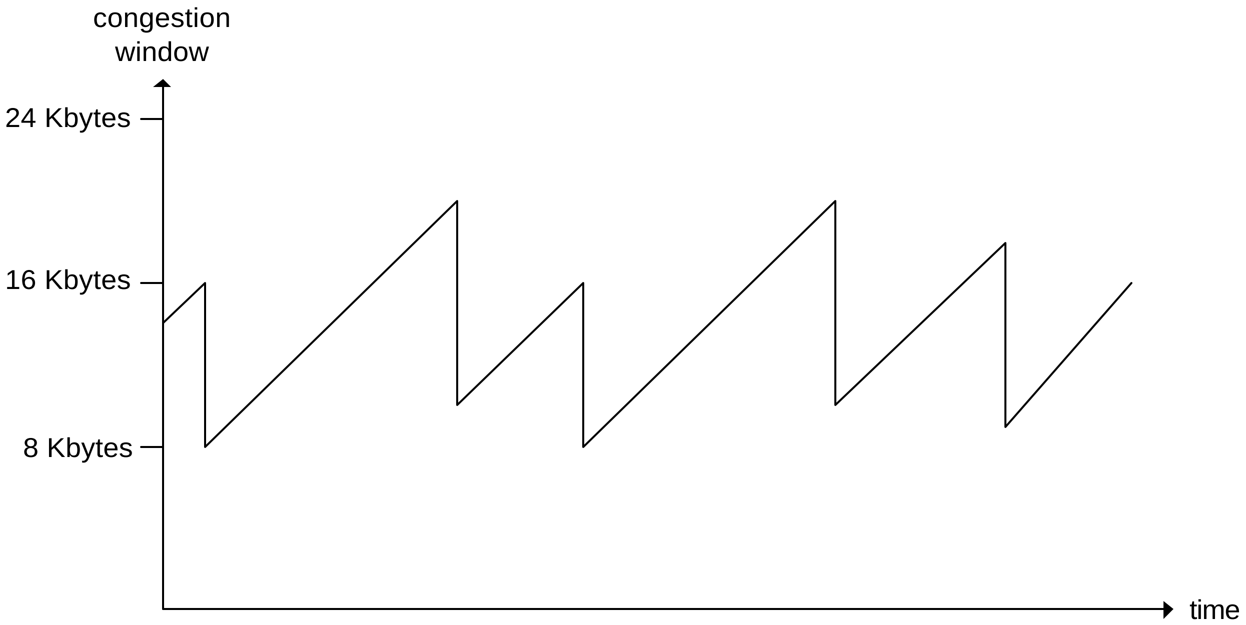
- When connection begins, increase rate exponentially until first loss event:
    - double **`CongWin`** every RTT
    - done by incrementing **`CongWin`** for every ACK received
- Summary: initial rate is slow but ramps up exponentially fast



# TCP AIMD

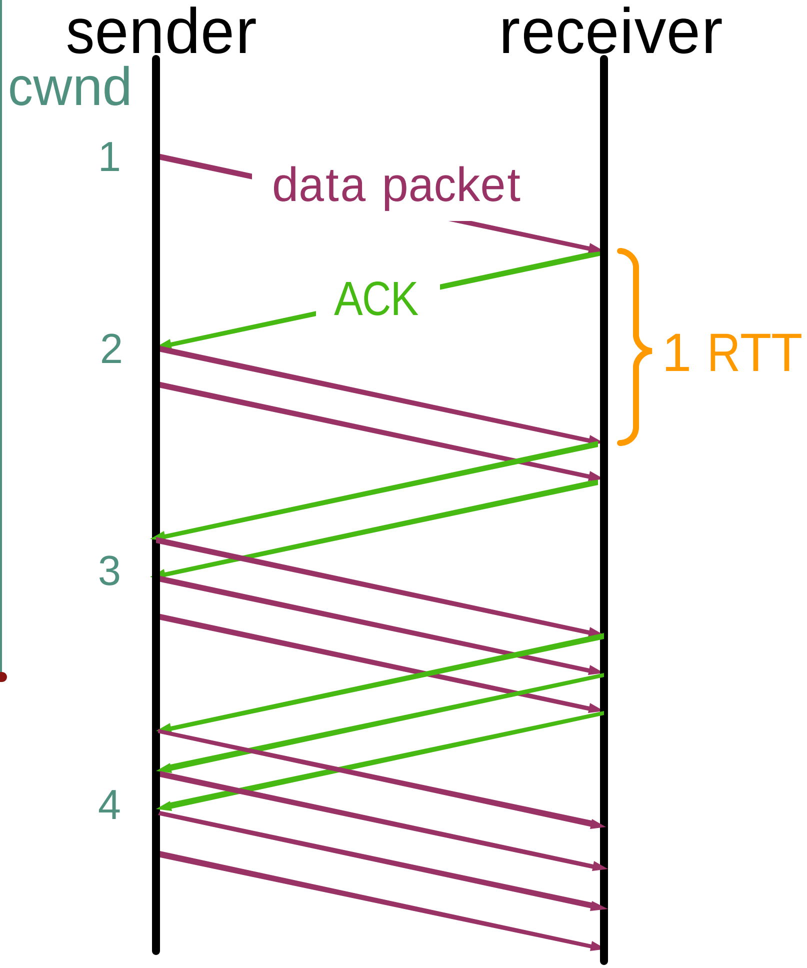
additive increase: increase **CongWin** by 1 MSS every RTT in the absence of loss events: *probing*

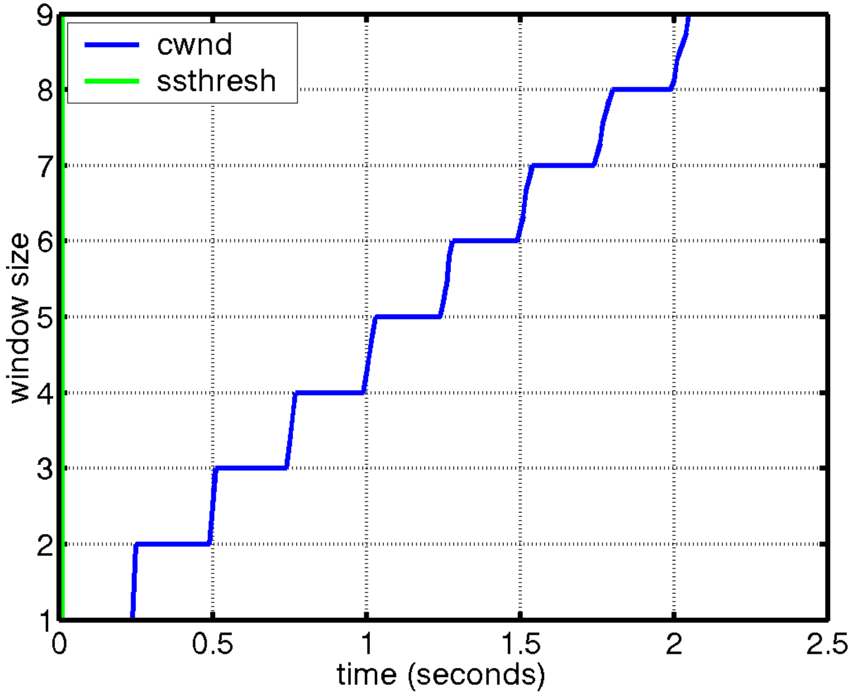
multiplicative decrease: cut **CongWin** in half after loss event



Long-lived TCP connection

# Congestion Avoidance





cwnd ← cwnd + 1 (for each cwnd ACKS)

# Reaction to Timeout Events

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

Philosophy:

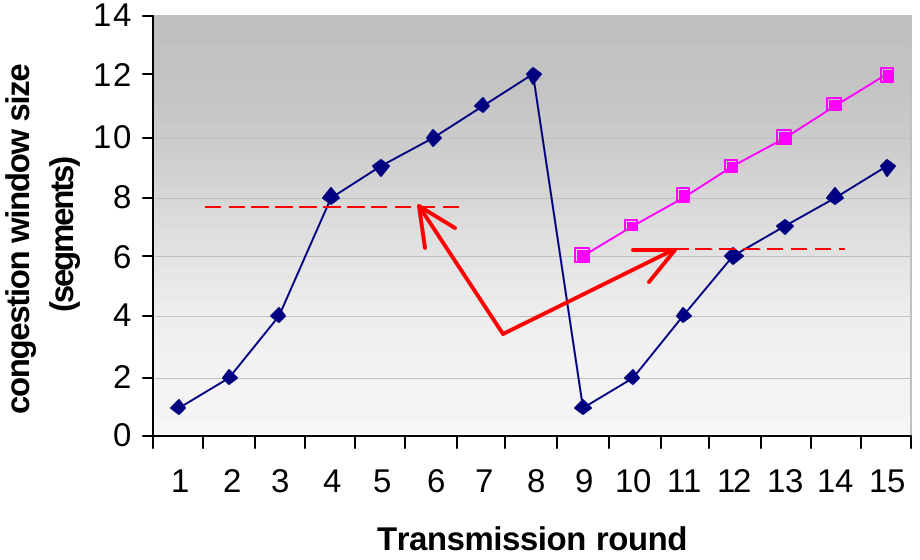
• 3 dup ACKs indicates network capable of delivering some segments

•timeout is “more alarming”

# Refinement

Q: When should the exponential increase switch to linear?

A: When **`CongWin`** gets to 1/2 of its value before timeout.



## Implementation:

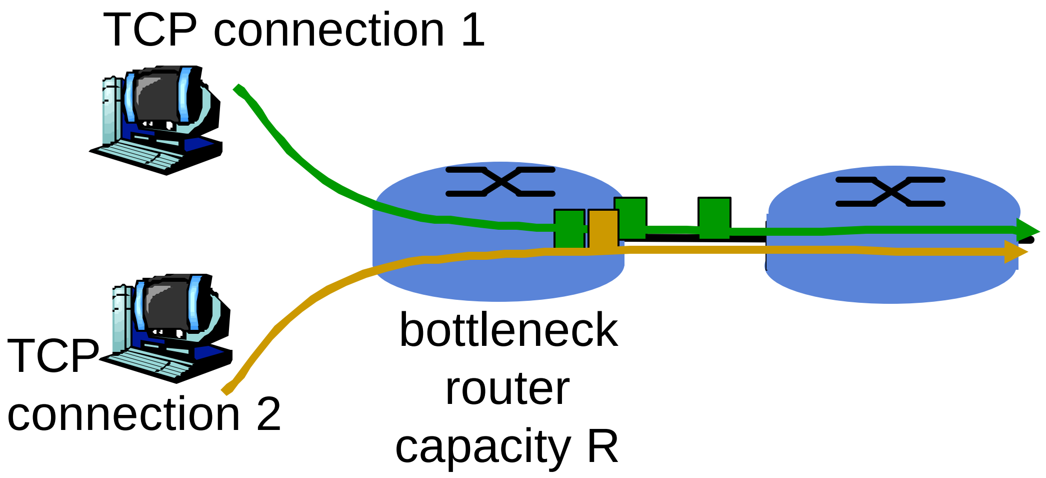
- Variable Threshold
- At loss event, Threshold is set to 1/2 of CongWin just before loss event

# 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 Fairness

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



TCP connection 1

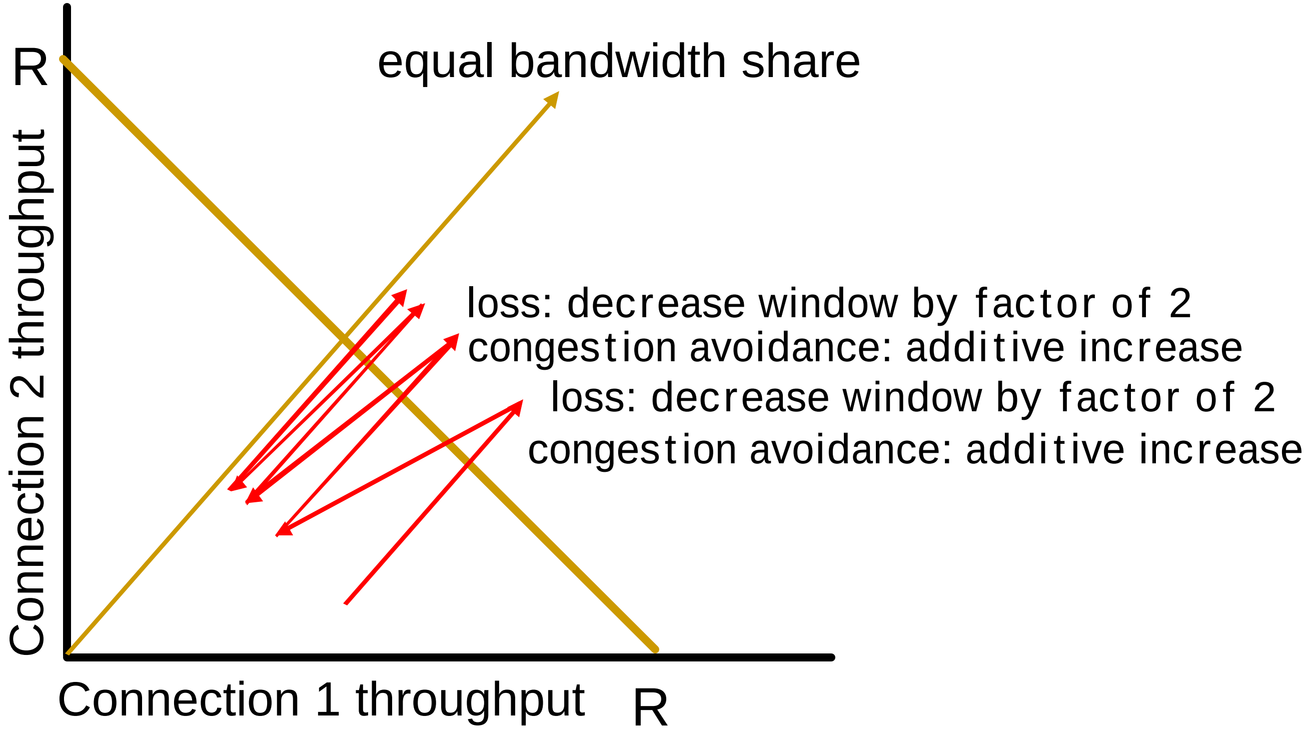
TCP connection 2

bottleneck router capacity R

# Why is TCP fair?

Two competing sessions:

- Additive increase gives slope of 1, as throughout increases
- multiplicative decrease decreases throughput proportionally


R
equal bandwidth share
Connection 2 throughput
loss: decrease window by factor of 2
congestion avoidance: additive increase
loss: decrease window by factor of 2
congestion avoidance: additive increase
Connection 1 throughput
R

# 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

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