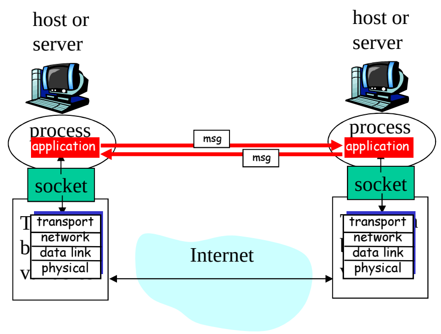
Chapter 3
Transport Layer

Transport Layer

- Central piece of the layered network architecture
  - Provides communications services to applications
  - …relying on the service of the network layer



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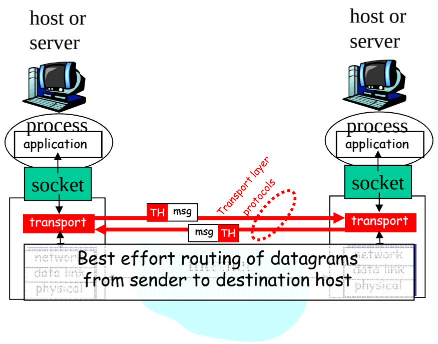
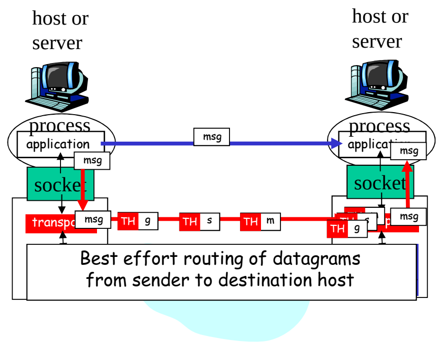


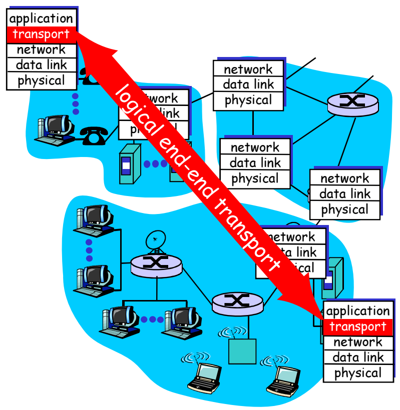
Transport Layer

- Transport entities 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
- Extends host to host datagram delivery to app to app message transfer



Internet transport-layer protocols

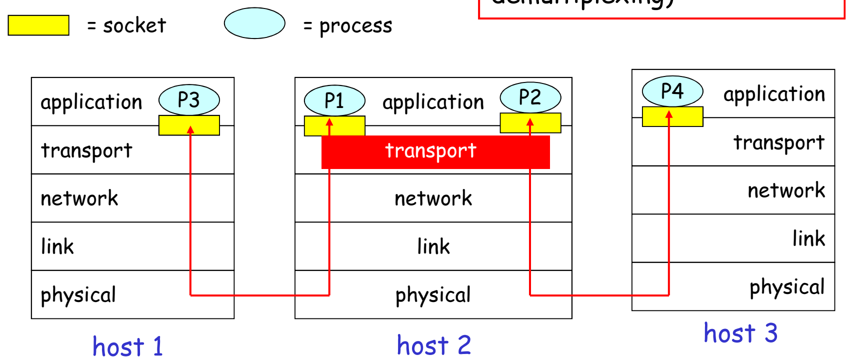
- reliable, in-order delivery (TCP)
  - congestion control
  - flow control
  - connection setup
- unreliable, unordered delivery: UDP
  - no-frills extension of "best-effort" IP
- services not available:
  - delay guarantees
  - bandwidth guarantees



Multiplexing/demultiplexing

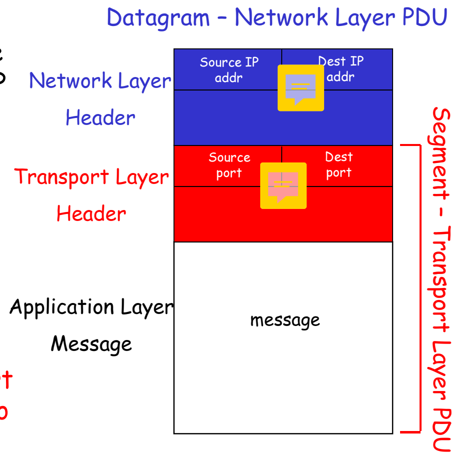
Demultiplexing at rcv host:
delivering received segments to correct socket

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 (in the network layer header)
  - each datagram carries 1 transport-layer segment
  - each segment has source, destination port number (in the transport layer header)
- host uses IP addresses & port numbers to direct segment to appropriate socket



Connectionless demultiplexing

- Create sockets with port numbers:

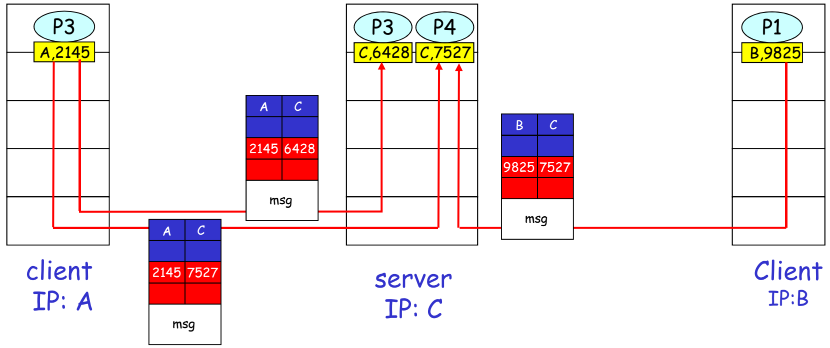
```
DatagramSocket mySocket1 = new
  DatagramSocket(9911);
DatagramSocket mySocket2 = new
  DatagramSocket(9922);
```

- UDP socket identified by two-tuple:

(IP address, 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 # but same destination port # are directed to same socket

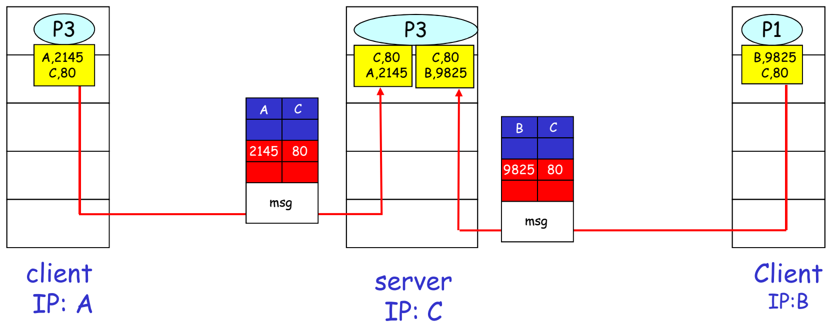
Connectionless demux (cont)



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)



UDP: User Datagram Protocol [RFC 768]

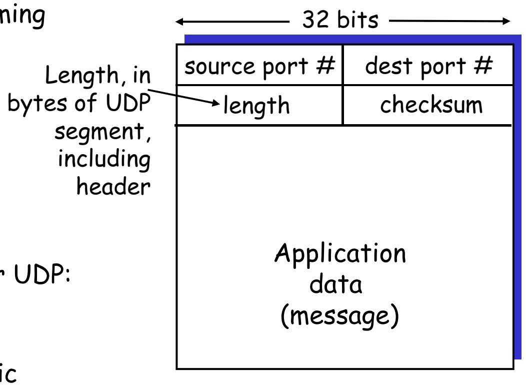
- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out of order
- to bare IP service, UDP adds
  - Mux/demux
  - checksum
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently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

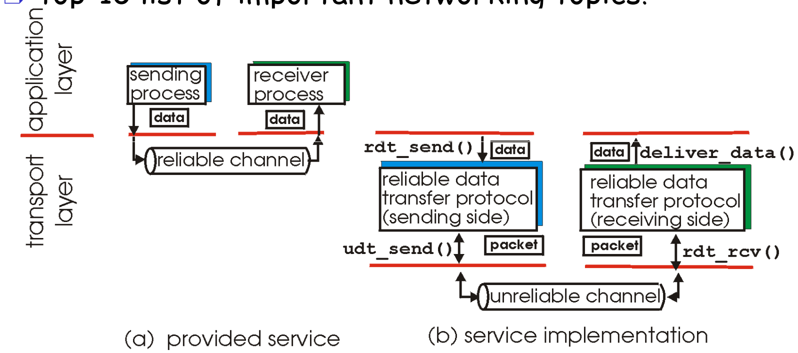
example: add two 16-bit integers

		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

Note: when adding numbers, a carryout from the most significant bit needs to be added to the result

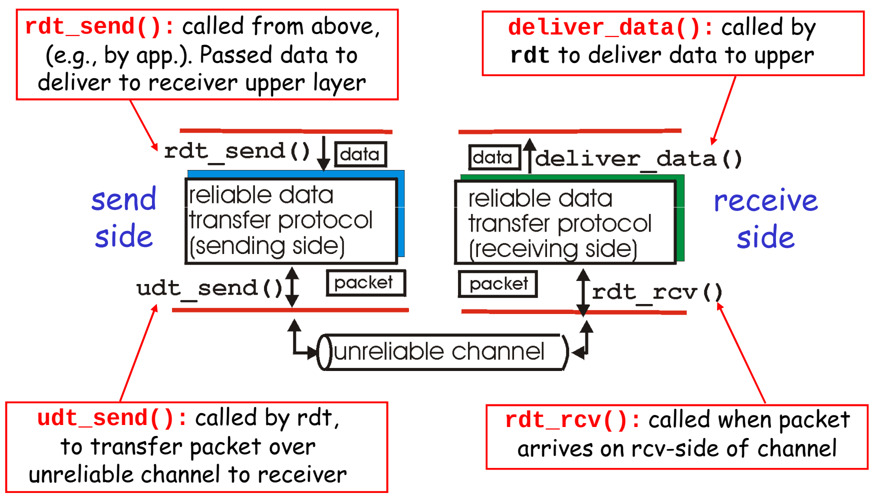
Principles of Reliable data transfer

- 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)

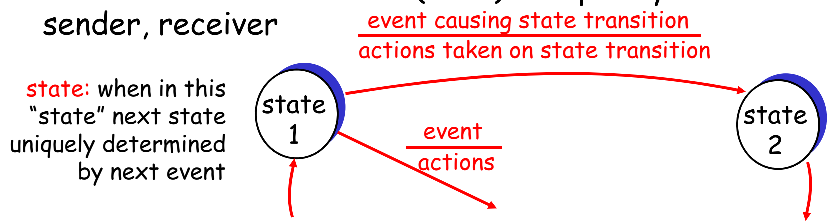
Reliable data transfer: getting started



Reliable data transfer: getting started

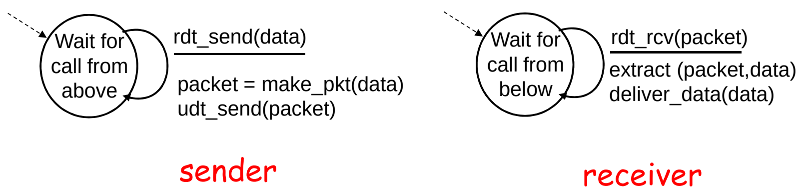
We'll:

- 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



Rdt1.0: reliable transfer over a reliable channel

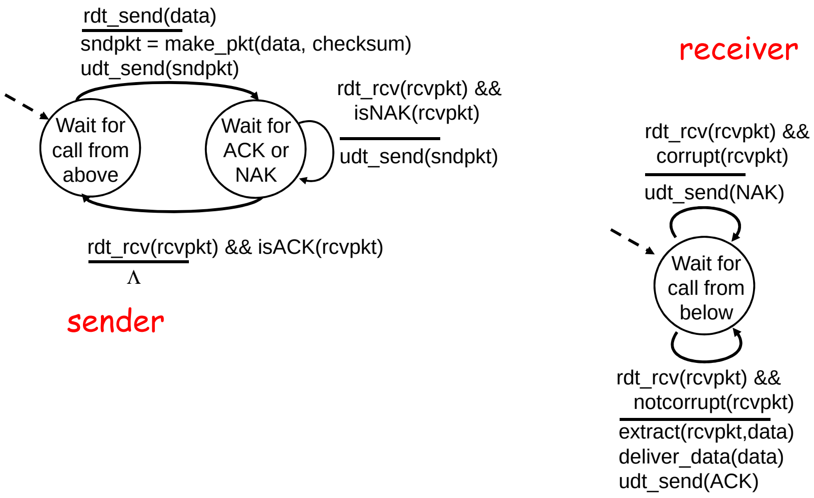
- underlying channel perfectly reliable
  - no bit errors
  - no loss of packets
- separate FSMs for sender, receiver:
  - sender sends data into underlying channel
  - receiver read data from underlying channel



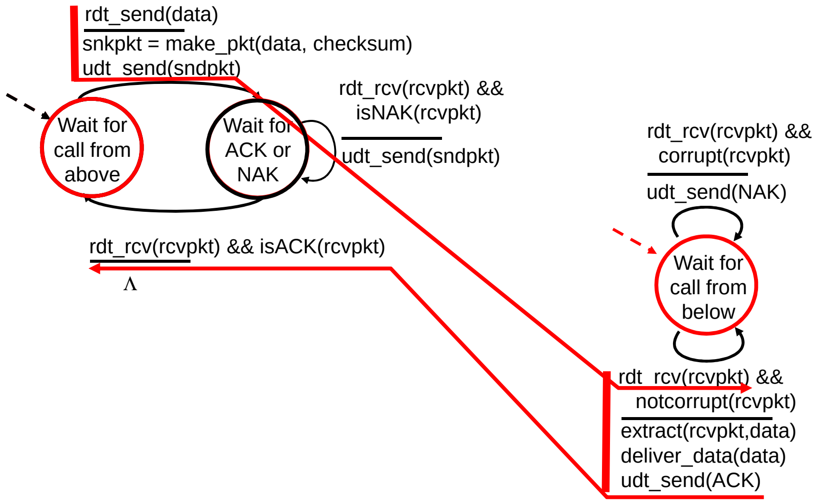
Rdt2.0: channel with bit errors

- underlying channel may flip bits in packet
  - recall: UDP checksum to detect bit errors
- *the* question: how to recover from errors:
  - *acknowledgements (ACKs):* receiver explicitly tells sender that pkt 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

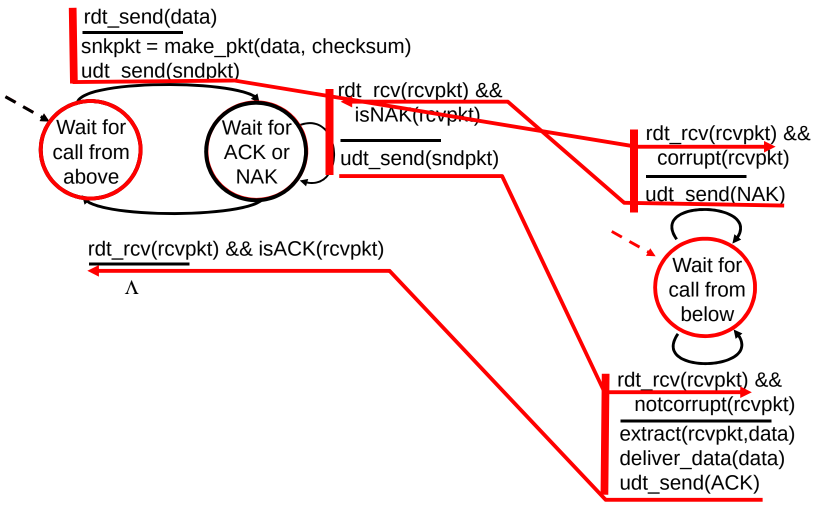
rdt2.0: FSM specification



rdt2.0: operation with no errors



rdt2.0: error scenario



rdt2.0 has a fatal flaw!

What happens if ACK/NAK corrupted?

- sender doesn't know what happened at receiver!

What to do?

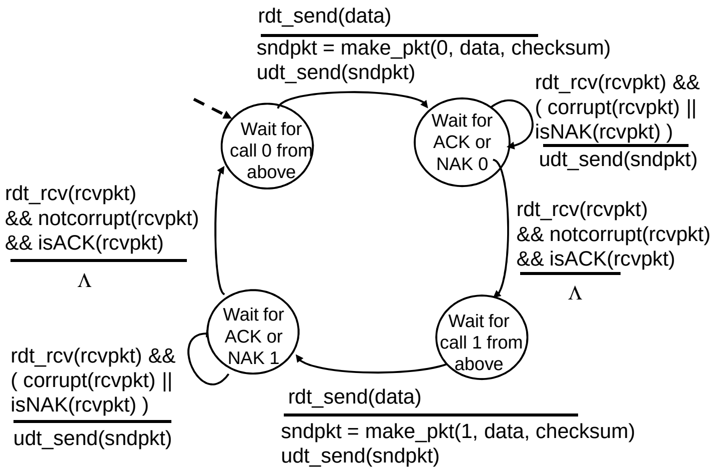
- sender ACKs/NAKs receiver's ACK/NAK? What if sender ACK/NAK lost?
- retransmit, but this might cause retransmission of correctly received pkt!

Handling duplicates:

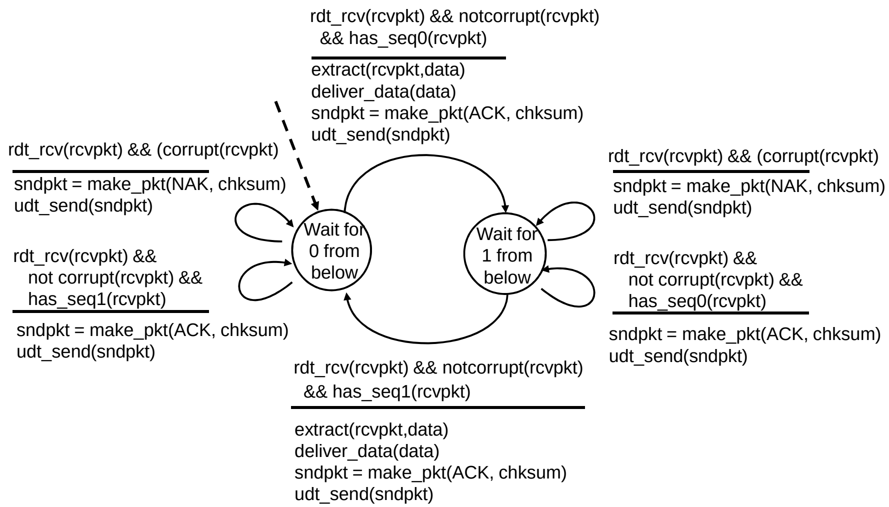
- sender adds *sequence number* to each pkt
- sender retransmits current pkt if ACK/NAK garbled
- receiver discards (doesn't deliver up) duplicate pkt

> **stop and wait**
> Sender sends one packet, then waits for receiver response

rdt2.1: sender, handles garbled ACK/NAKs



rdt2.1: receiver, handles garbled ACK/NAKs



rdt2.1: discussion

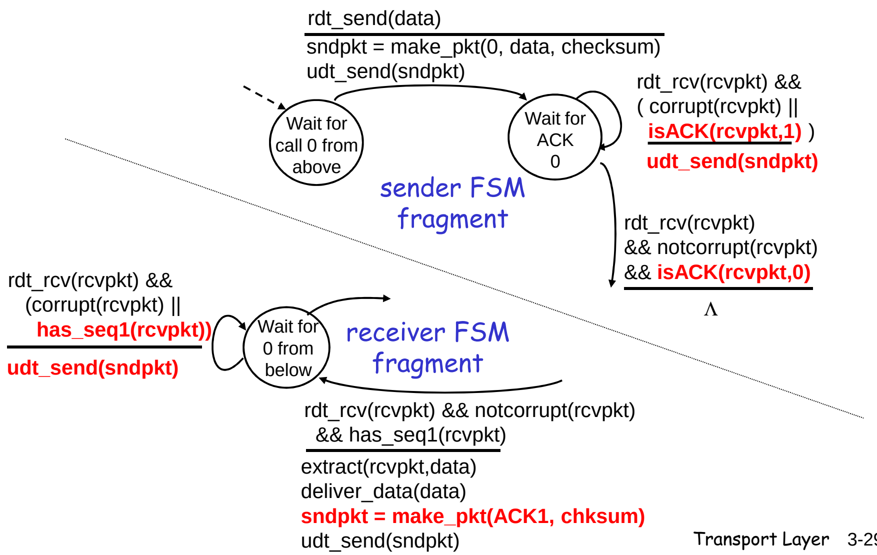
Sender:
- seq # added to pkt
- two seq. #'s (0,1) will suffice. Why?
- must check if received ACK/NAK corrupted
- twice as many states
  - state must "remember" whether "current" pkt has 0 or 1 seq. #

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 ACKs 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



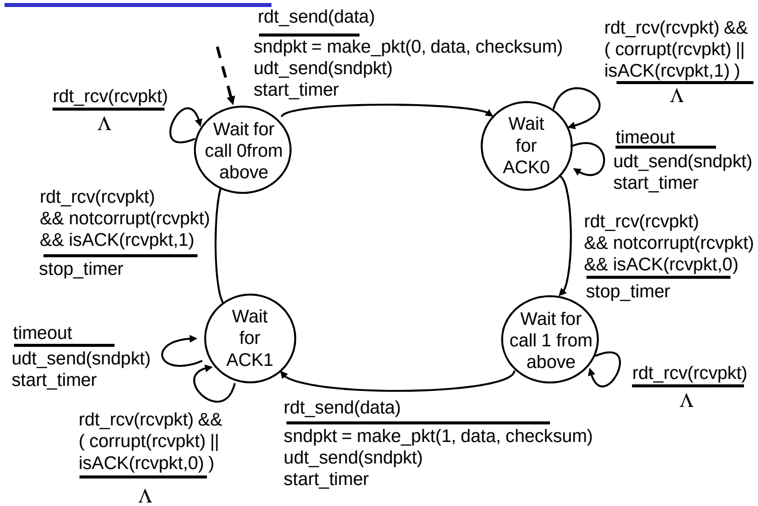
rdt3.0: channels with errors *and* loss

New assumption: underlying channel can also lose packets (data or ACKs)
- checksum, seq. #, ACKs, retransmissions will be of help, but not enough

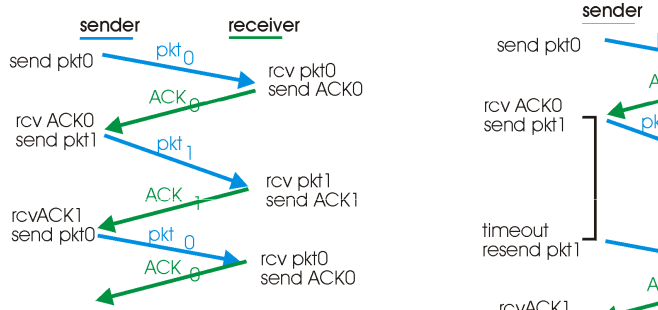
Q: how to deal with loss?
- sender waits until certain data or ACK lost, then retransmits
- yuck: drawbacks?

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

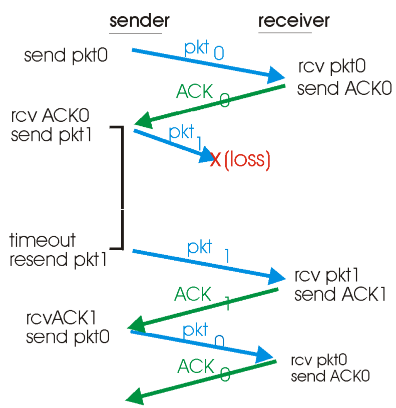
rdt3.0 sender



rdt3.0 in action

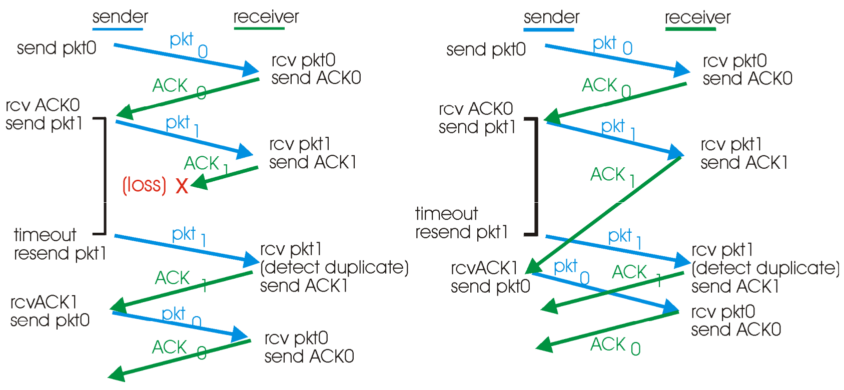


(a) operation with no loss



(b) lost packet

rdt3.0 in action



(c) lost ACK (d) premature timeout

Performance of rdt3.0

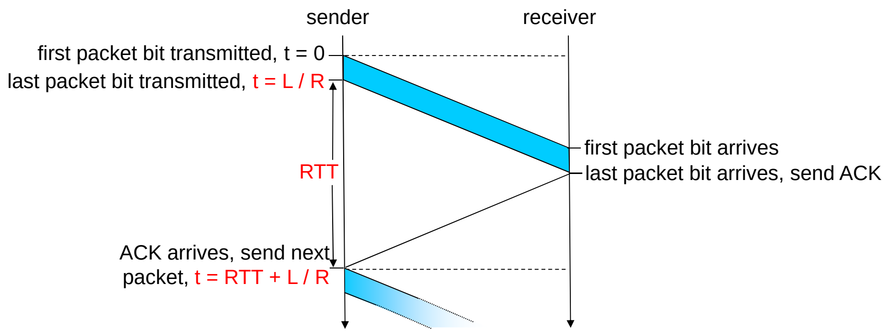
- rdt3.0 works, but performance stinks
- example: 1 Gbps link, 15 ms e-e prop. delay, 1KB packet:

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

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

  - U_{sender}: utilization – fraction of time sender busy sending
  - 1KB pkt every 30 msec -> 267kbps thruput over 1 Gbps link
  - network protocol limits use of physical resources!

rdt3.0: stop-and-wait operation

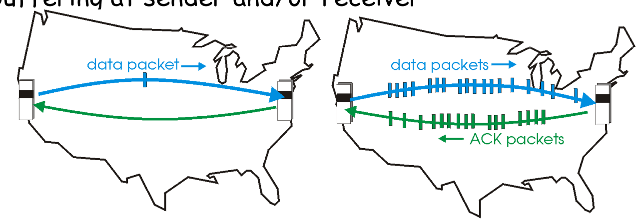


$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols

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

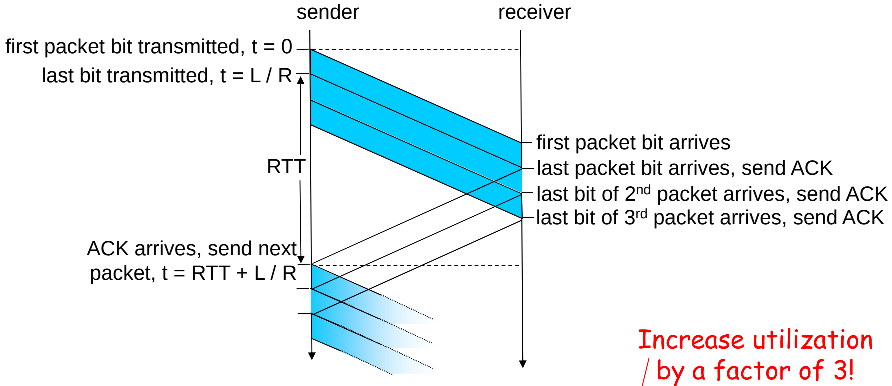
- range of sequence numbers must be increased
- buffering at sender and/or receiver



(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



$$U_{sender} = \frac{3 * L / R}{RTT + 3 L /} = \frac{.024}{30.024} = 0.0008$$

Pipelining Protocols

Go-back-N: big picture:

- Sender can have up to N unacked packets in pipeline
- Rcvr only sends cumulative acks
  - Doesn't ack packet if there's a gap
- Sender has timer for oldest unacked packet
  - If timer expires, retransmit all unacked packets

Selective Repeat: big pic

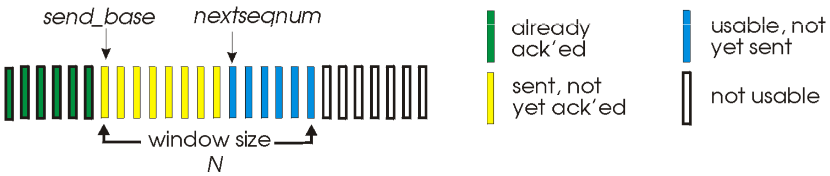
- Sender can have up to N unacked packets in pipeline
- Rcvr acks individual packets
- Sender maintains timer for each unacked packet
  - When timer expires, retransmit only unack packet

Go-Back-N

- Trasmit multiple packets (up to N) without waiting for ACK

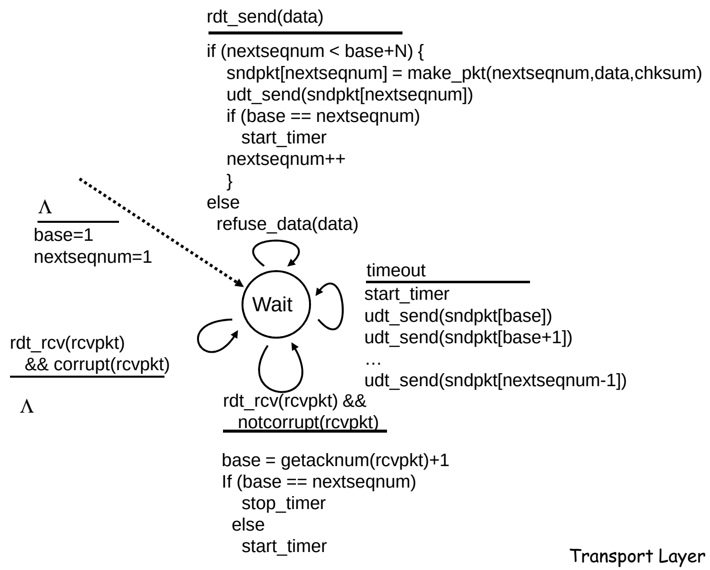
Sender:

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

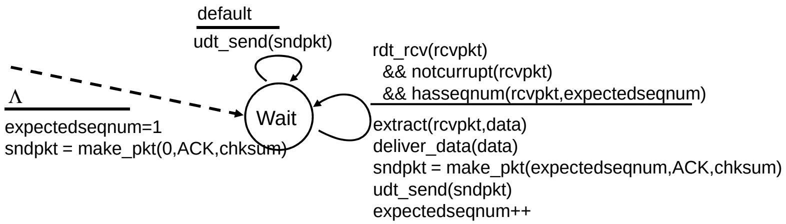


- ACK(n): ACKs all pkts up to, including seq # n - "cumulative ACK"
  - may receive duplicate ACKs (see receiver)
- *timeout(n):* retransmit pkt n and all higher seq # pkts in window

GBN: sender extended FSM (1 timer)



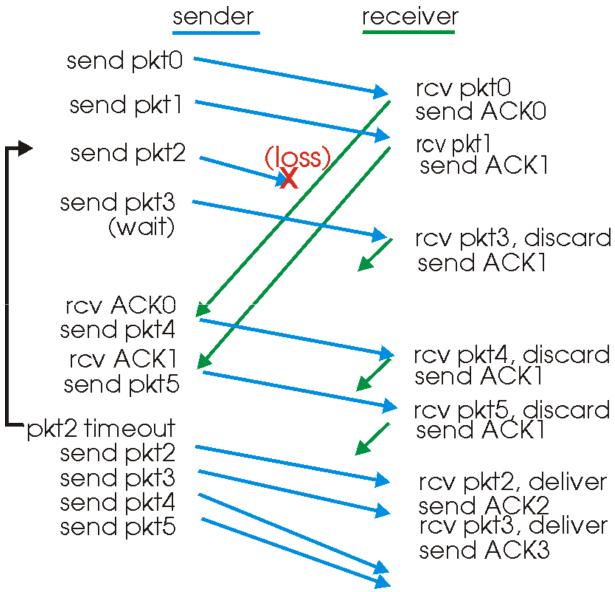
GBN: receiver extended FSM



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 #

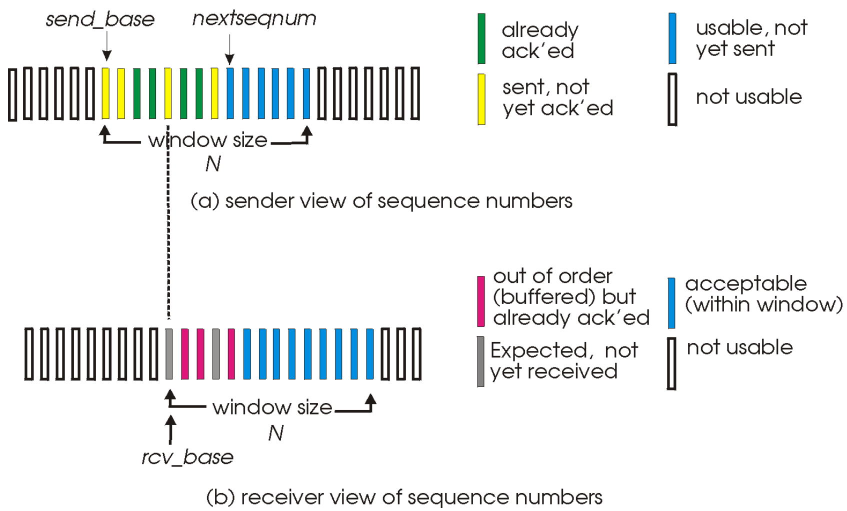
GBN in action



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



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

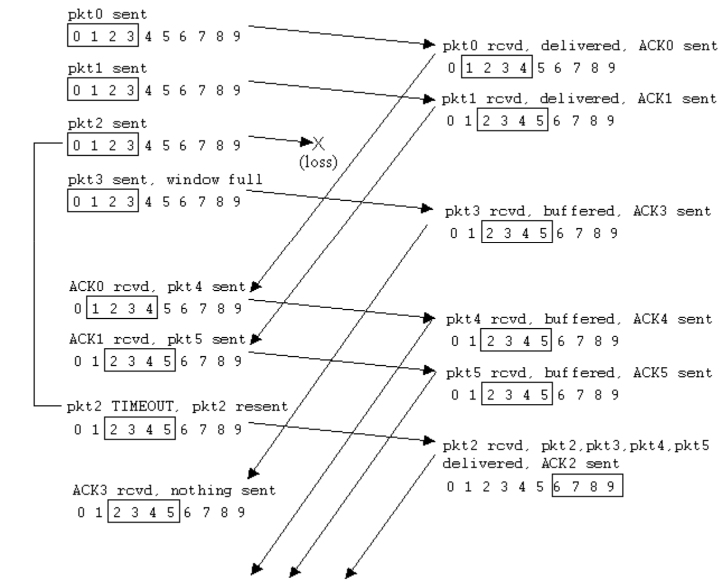
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore

Selective repeat in action



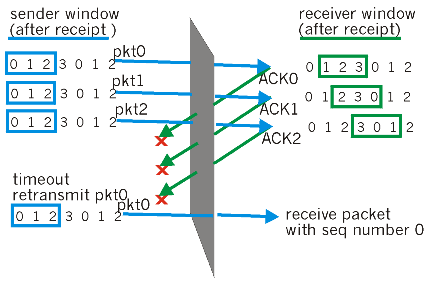
Selective repeat: dilemma

Example:

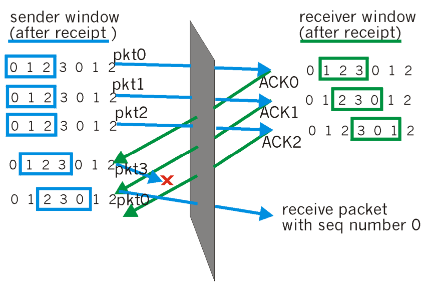
- seq #'s: 0, 1, 2, 3
- window size=3

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

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



(a)

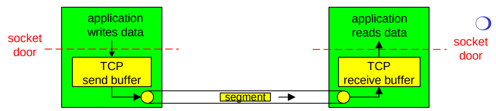


(b)

TCP: Overview

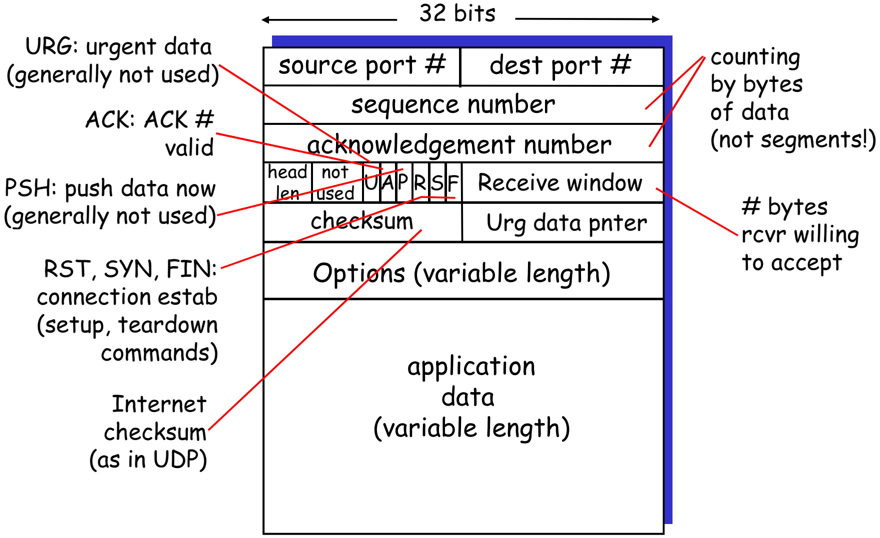
RFCs: 793, 1122, 1323, 2018, 2581

- 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



Sequence and Acknowledgement Numbers

- TCP views data as unstructured, but ordered, data

In a segment:

- Sequence number: is the byte-stream number of the first byte in the segment
  - Initial sequence number is randomly chosen
- Ack number: is the number of the next byte expected from the other side
  - TCP uses cumulative acknowledgements

Q: how receiver handles out-of-order segments

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

TCP seq. numbers, ACKs

sequence numbers:

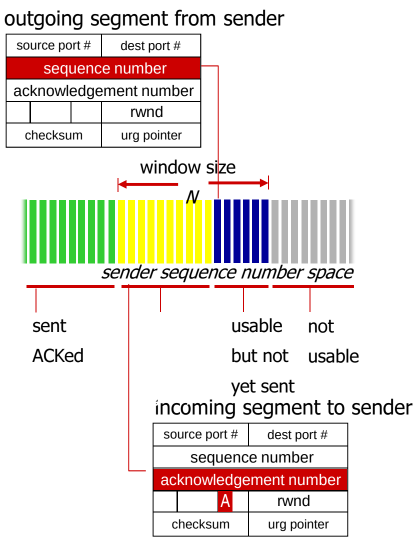
- byte stream "number" of first byte in segment's data

acknowledgements:

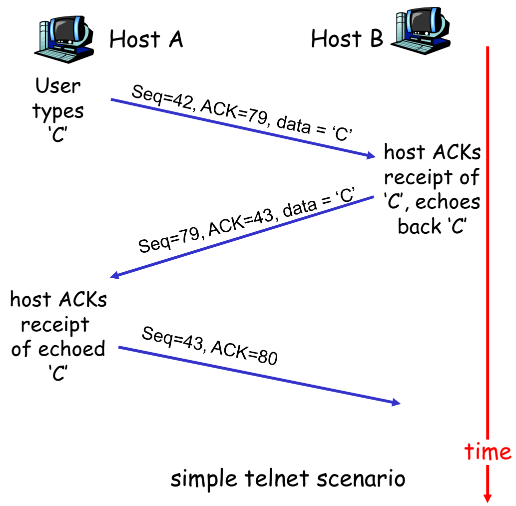
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

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



TCP seq. #'s and ACKs



simple telnet scenario

TCP reliable data transfer

- TCP creates rdt service on top of IP's unreliable service
- Pipelined segments
- Cumulative acks
- TCP should use a 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 for that segment
- expiration interval: `TimeOutInterval`

timeout:

- retransmit segment that caused timeout
- restart timer

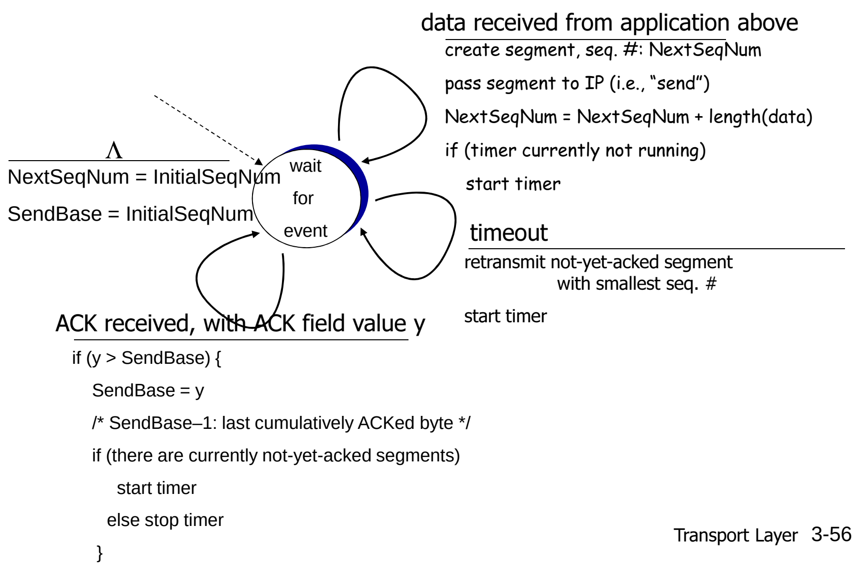
Ack rcvd:

- If acknowledges previously unacked segments
  - update what is known to be acked

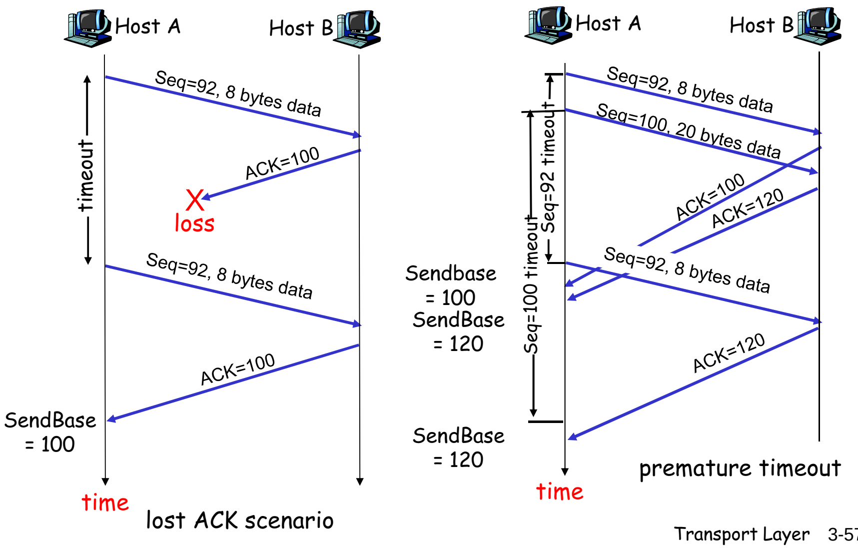
TCP sender (simplified)

```
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)
        break
    event: timer timeout
        retransmit not-yet acked segment with smallest sequence
                    number
        start timer
        break
    event: ACK received, with ACK field value of y
        if (y > SendBase) {
            Sendbase=y;
            if( there are currently any not yet acked segment)
                    start timer
        break
  }  /* end of loop forever */
```

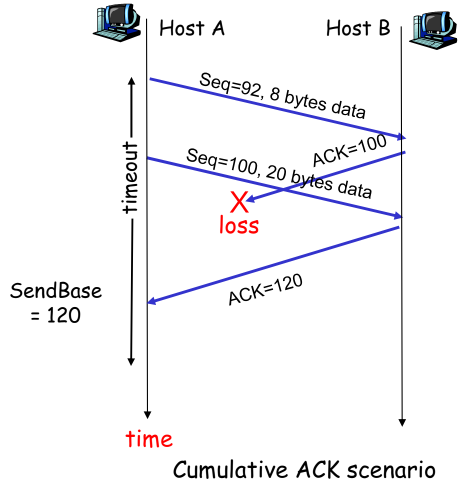

TCP sender (simplified)



TCP: retransmission scenarios



TCP retransmission scenarios (more)



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

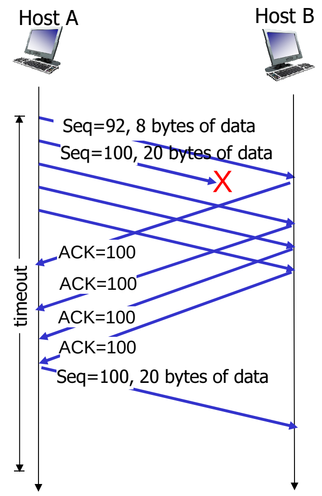
Fast retransmit algorithm:

event: ACK received, with ACK field value of y
    if (y > SendBase) {
        Sendbase=y;
        if(there are currently any not yet acked segment)
            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
        }
    break

a duplicate ACK for already ACKed segment

fast retransmit

TCP fast retransmit



fast retransmit after sender receipt of triple duplicate ACK

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

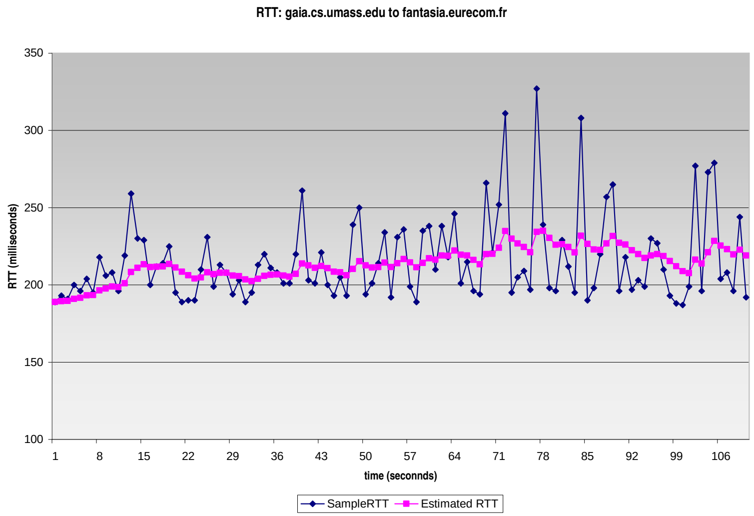
Q: how to estimate RTT?
- **SampleRTT**: measured time from segment transmission until ACK receipt
  - ignore retransmitted segments
- **SampleRTT** will vary, want estimated RTT "smoother"
  - average several recent measurements, not just current **SampleRTT**

TCP Round Trip Time and Timeout

$$\texttt{EstimatedRTT = (1-}\ \alpha\texttt{)*EstimatedRTT +}\ \alpha\texttt{*SampleRTT}$$

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

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:

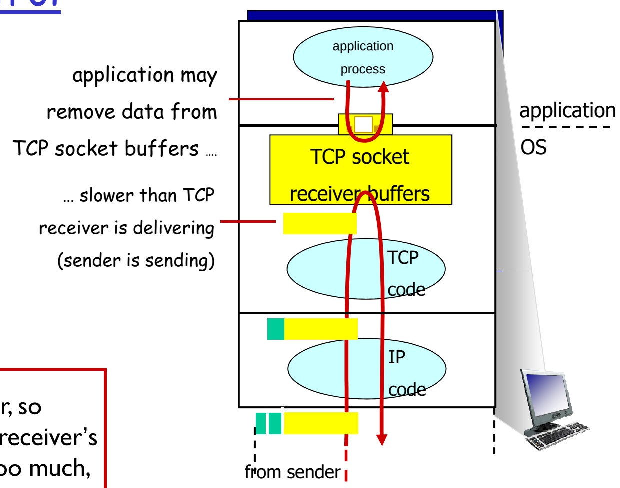
DevRTT = (1-β)*DevRTT +
β*|SampleRTT-EstimatedRTT|

(typically, β = 0.25)

Then set timeout interval:

TimeoutInterval = EstimatedRTT + 4*DevRTT

TCP flow control

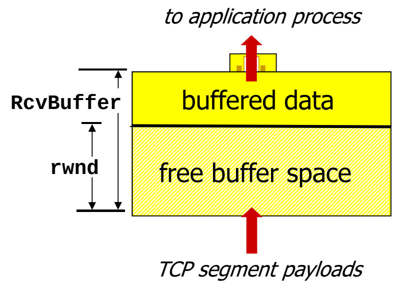


receiver protocol stack

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

TCP flow control

- receiver "advertises" free buffer space by including **rwnd** value in TCP header of receiver-to-sender segments
  - **RcvBuffer** size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust **RcvBuffer**
- sender limits amount of unacked ("in-flight") data to receiver's **rwnd** value



receiver-side buffering

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");
```

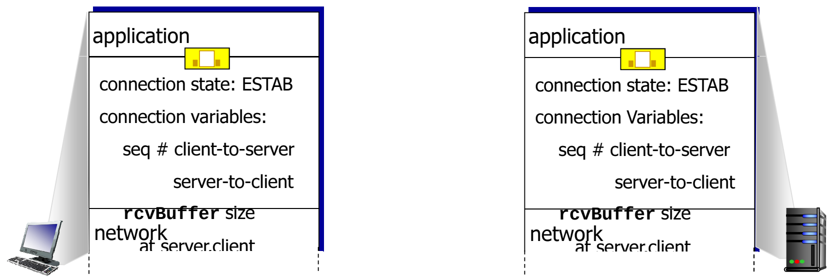
- *server:* contacted by client

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


Connection Management

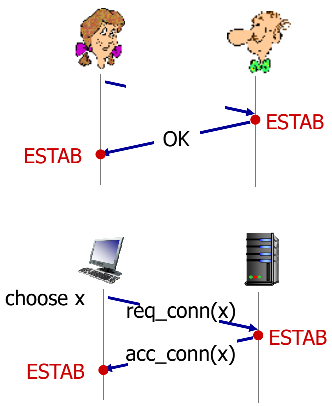
before exchanging data, sender/receiver "handshake":

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



Agreeing to establish a connection

2-way handshake:

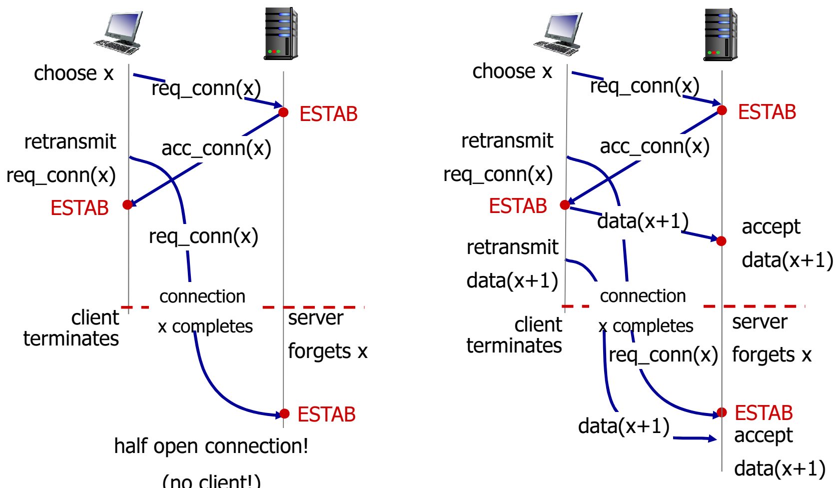


Q: will 2-way handshake always work in network?

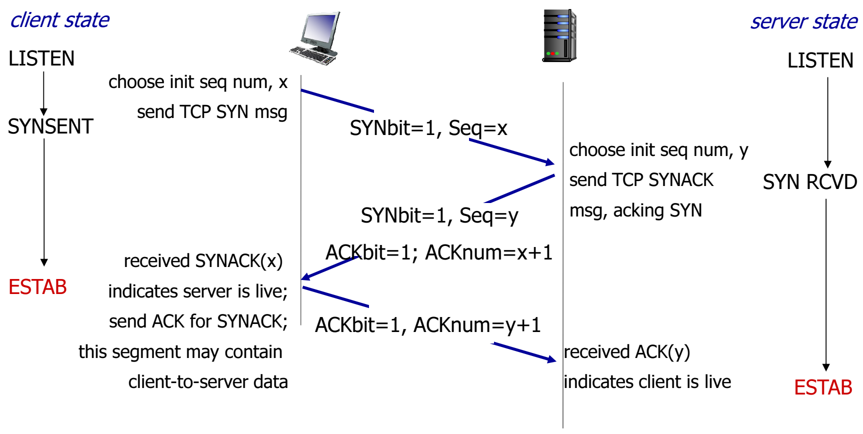
- variable delays
- retransmitted messages (e.g. req_conn(x)) due to message loss
- message reordering
- can't "see" other side

Agreeing to establish a connection

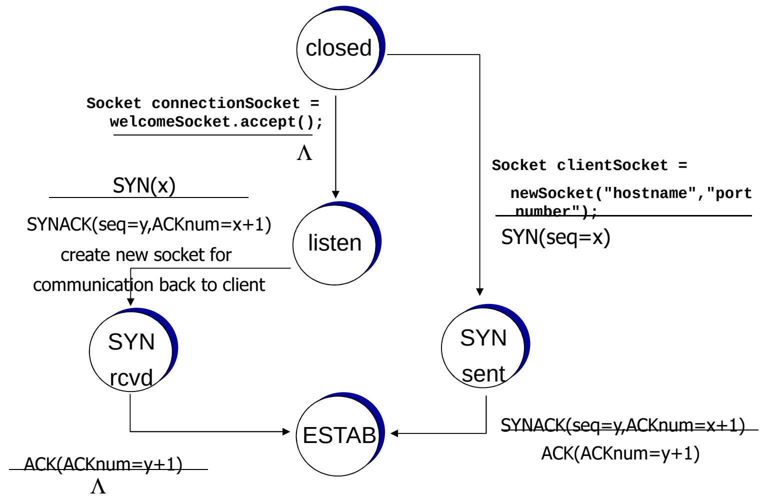
2-way handshake failure scenarios:



TCP 3-way handshake



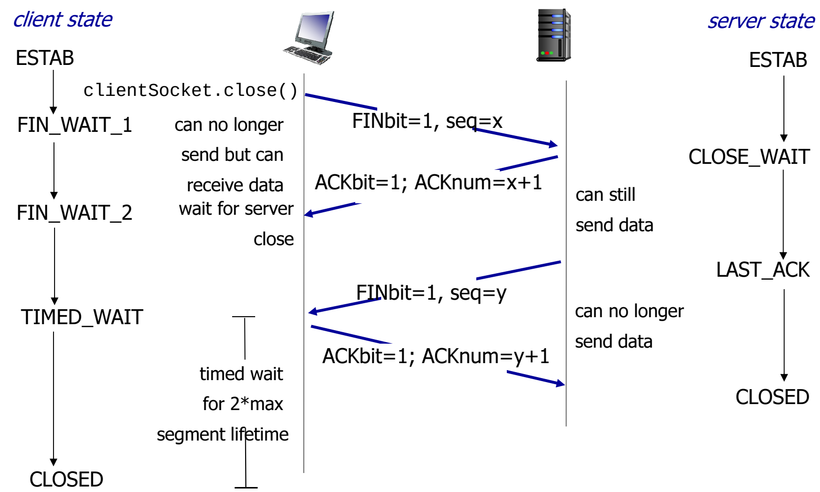
TCP 3-way handshake: FSM



TCP: closing a connection

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

TCP: closing a connection



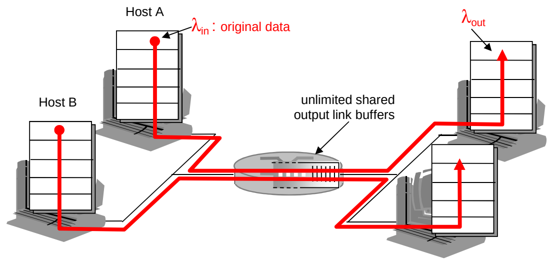
Principles of Congestion Control

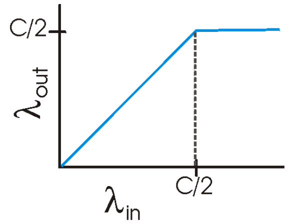
Congestion:

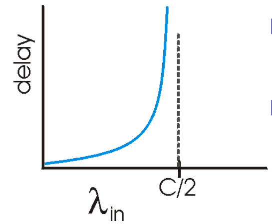
- informally: "too many sources sending too much data too fast for *network* to handle"
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Causes/costs of congestion: scenario 1

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



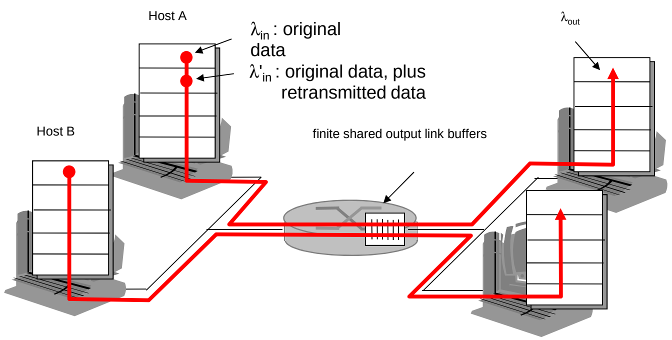




- large delays when congested
- maximum achievable throughput

Causes/costs of congestion: scenario 2

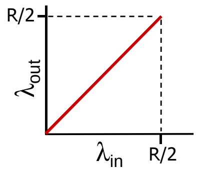
- one router, *finite* buffers
- sender retransmission of lost packet

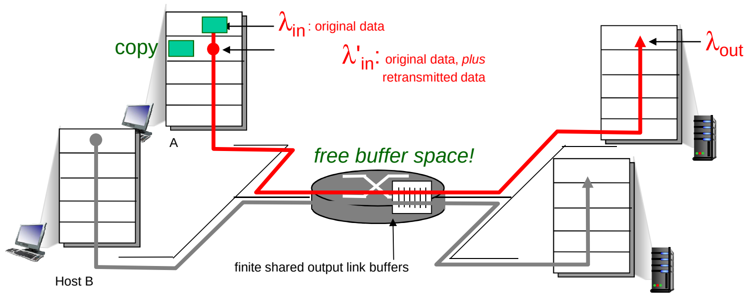


Causes/costs of congestion: scenario 2

idealization: perfect knowledge

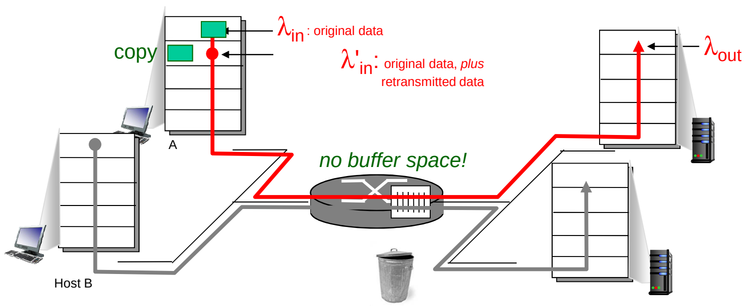
- sender sends only when router buffers available





Causes/costs of congestion: scenario 2

- *Idealization: known loss* packets can be lost, dropped at router due to full buffers
- sender only resends if packet *known* to be lost



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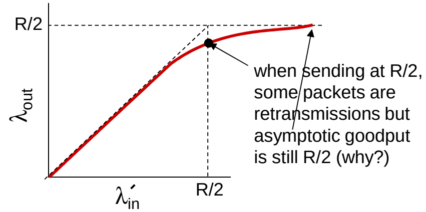
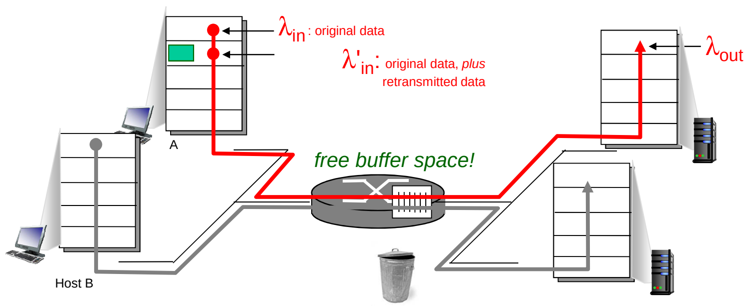
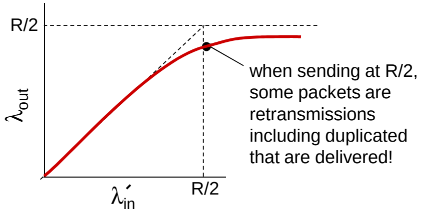
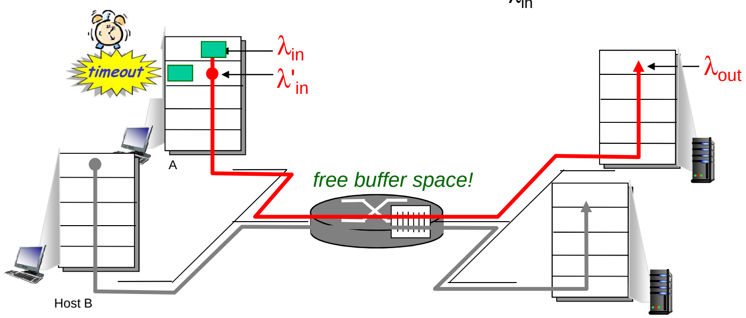




Causes/costs of congestion: scenario 2

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





Causes/costs of congestion: scenario 2

Realistic: duplicates
- packets can be lost, dropped at router due to full buffers
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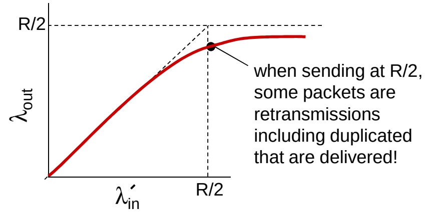
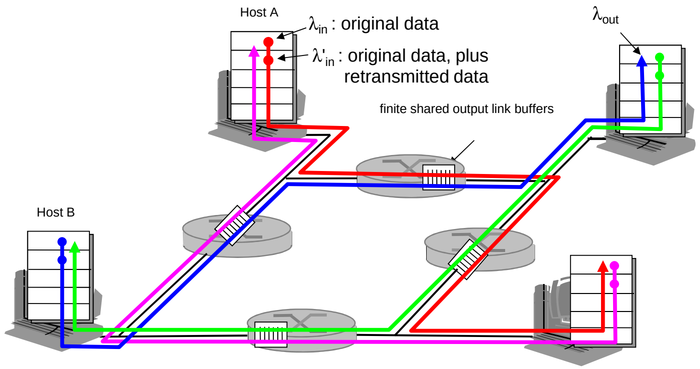


- "costs" of congestion:
- more work (retrans) for given "goodput"
- unneeded retransmissions: link carries multiple copies of pkt
  - decreasing goodput

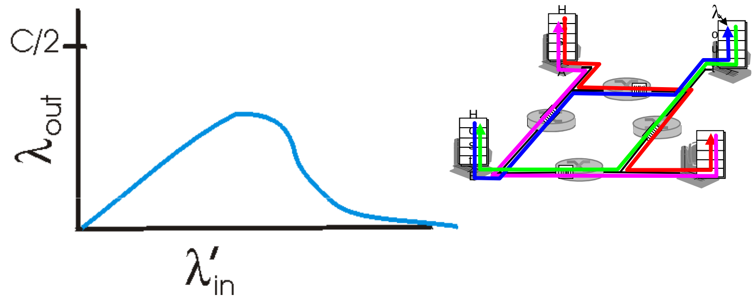
Causes/costs of congestion: scenario 3

- four senders
- multihop paths
- timeout/retransmit

Q: what happens as λ_{in} 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

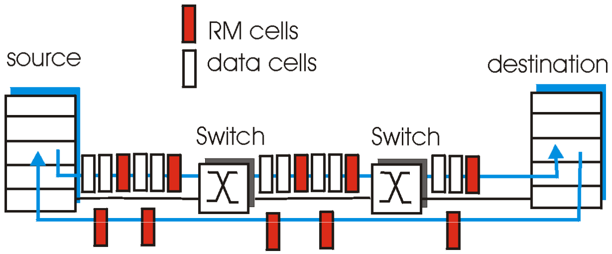
ABR: available bit rate:

- "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:

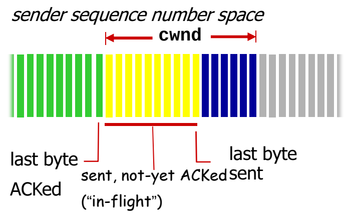
- sent by sender, interspersed with data cells
- bits in RM cell set by switches ("*network-assisted*")
  - NI bit: no increase in rate (mild congestion)
  - CI bit: congestion indication
- RM cells returned to sender by receiver, with bits intact

Case study: ATM ABR congestion control



- two-byte ER (explicit rate) field in RM cell
  - congested switch may lower ER value in cell
  - sender' send rate thus maximum supportable rate on path
- EFCI bit in data cells: set to 1 in congested switch
  - if data cell preceding RM cell has EFCI set, sender sets CI bit in returned RM cell

TCP Congestion Control



- end-end control (no network assistance)
- sender limits transmission:
  LastByteSent-LastByteAcked ≤ min{CongWin, RcvWindow}
- Roughly,

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

- **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

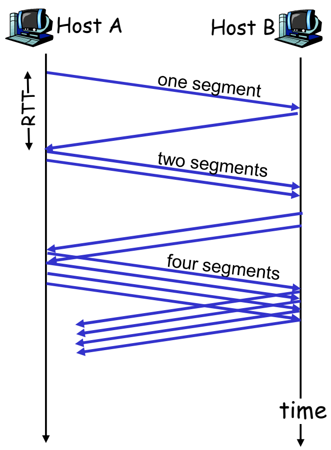
three mechanisms:
  - slow start
  - AIMD
  - conservative after timeout events

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)

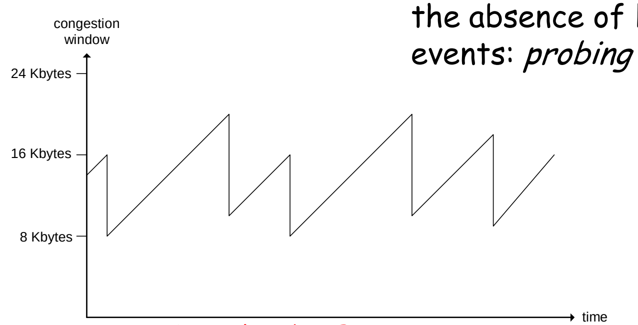
- When connection begins, increase rate exponentially until first loss event:
  - double **CongWin** every RTT
  - done by incrementing **CongWin** for every ACK received
- Summary: initial rate is slow but ramps up exponentially fast



TCP AIMD

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

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



Long-lived TCP connection

Refinement: inferring loss

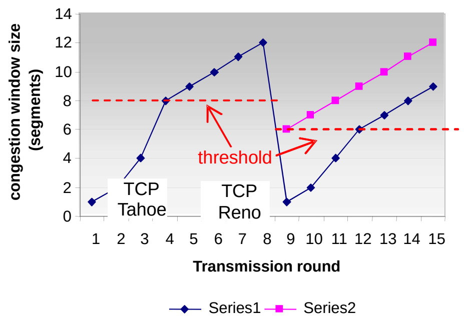
- After 3 dup ACKs:
  - **CongWin** is cut in half
  - window then grows linearly
- But after timeout event:
  - **CongWin** instead set to 1 MSS;
  - window then grows exponentially
  - to a threshold, then grows linearly

Philosophy:
- 3 dup ACKs indicates network capable of delivering some segments
- timeout before 3 dup ACKs is "more alarming"

Refinement (more)

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

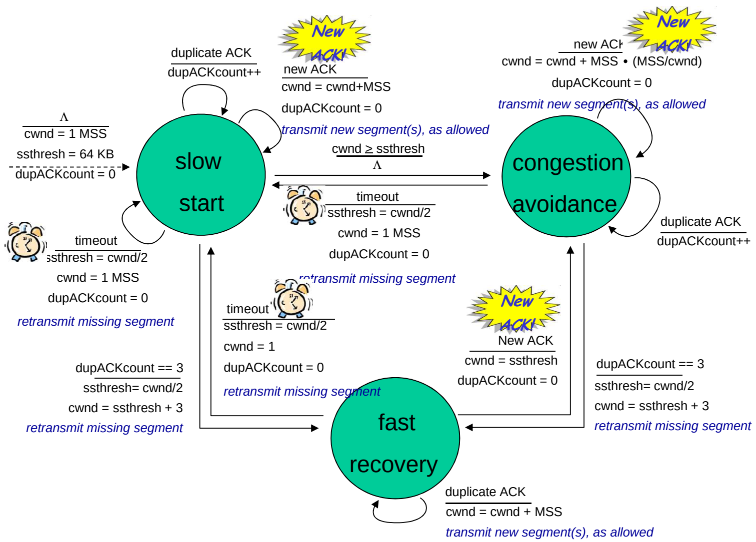
Summary: TCP Congestion Control

- 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

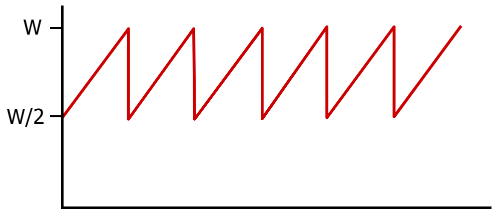
Summary: TCP Congestion Control



TCP throughput

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

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



TCP Futures: TCP over "long, fat pipes"

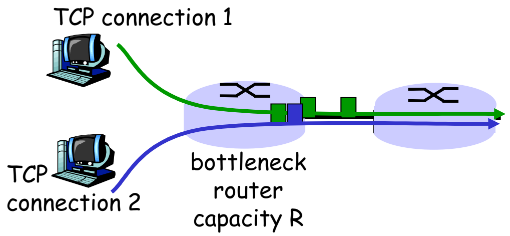
- Example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput
- Requires window size W = 83,333 in-flight segments
- Throughput in terms of loss rate:

$$\frac{1.22 \cdot MSS}{RTT\sqrt{L}}$$

- ➜ L = $2 \cdot 10^{-10}$ *Wow*
- New versions of TCP for high-speed

TCP Fairness

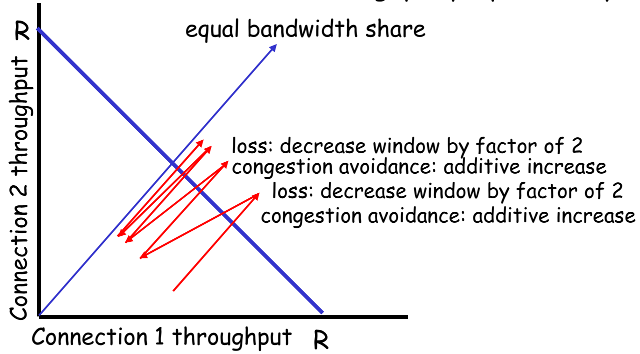
Fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



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 cnctions between 2 hosts.
- Web browsers do this
- Example: link of rate R supporting 9 cnctions;
  - new app asks for 1 TCP, gets rate R/10
  - new app asks for 11 TCPs, gets R/2 !