Mobile Ad Hoc Networking

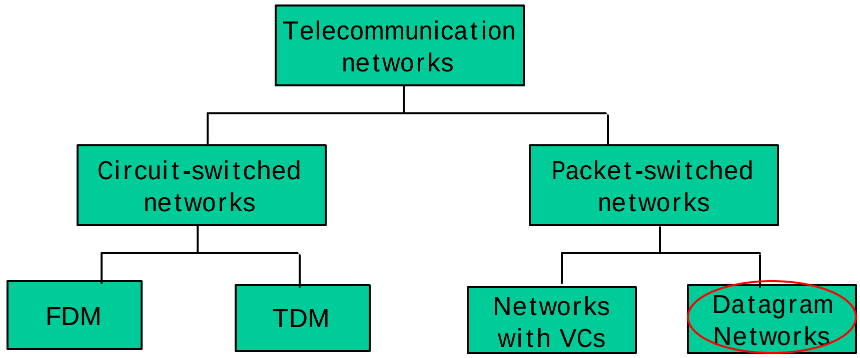
Transport Layer Issues

Transport Layer Issues

Contents:

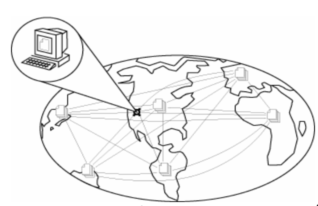
- overview principles behind transport layer services:
  - multiplexing/demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- TCP Performance analysis

But first,
a general overview of networks (and the Internet)



What Is the Internet?

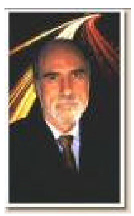
- A network of networks, joining many government, university and private computers together and providing an infrastructure for the use of E-mail, bulletin boards, file archives, hypertext documents, databases and other computational resources
- The vast collection of computer networks which form and act as a single huge network for transport of data and messages across distances which can be anywhere from the same office to anywhere in the world.



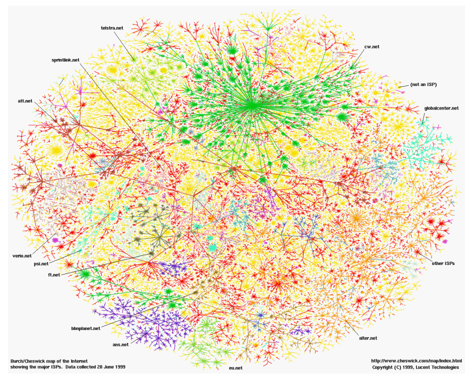
Written by William F. Slater, III
1996
President of the Chicago Chapter of the Internet Society

What is the Internet?

- The largest network of networks in the world.
- Uses TCP/IP protocols and packet switching .
- Runs on any communications substrate.



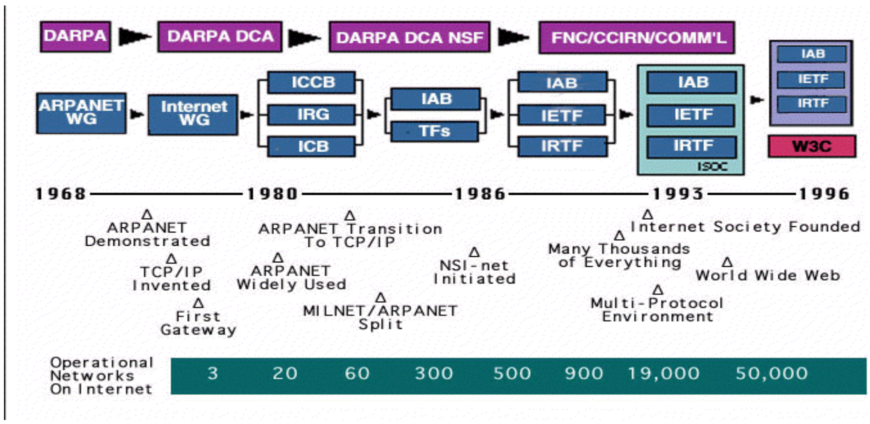
From Dr. Vinton Cerf, Co-Creator of TCP/IP



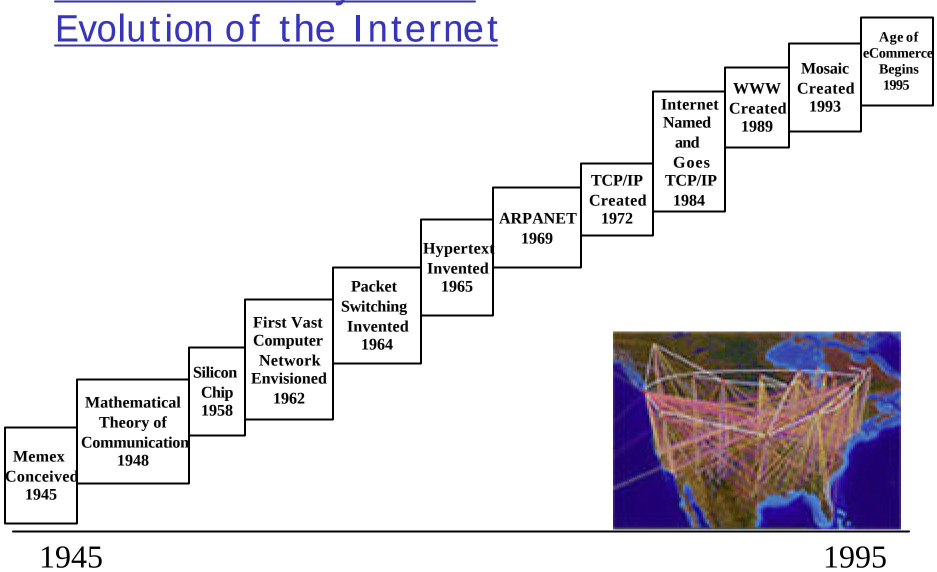
Brief History of the Internet

- 1968 - DARPA (Defense Advanced Research Projects Agency) contracts with BBN (Bolt, Beranek & Newman) to create ARPAnet
- 1970 - First five nodes:
  - UCLA
  - Stanford
  - UC Santa Barbara
  - U of Utah, and
  - BBN
- 1974 - TCP specification by Vint Cerf
- 1984 – On January 1, the Internet with its 1000 hosts converts en masse to using TCP/IP for its messaging

*** Internet History ***



A Brief Summary of the Evolution of the Internet



From Simple, But Significant Ideas Bigger Ones Grow 1940s to 1969

We will prove that packet switching works over a WAN.

Hypertext can be used to allow rapid access to text data

Packet switching can be used to send digitized data though computer networks

We can accomplish a lot by having a vast network of computers to use for accessing information and exchanging ideas

We can do it cheaply by using Digital circuits etched in silicon.

We do it reliably with "bits", sending and receiving data

We can access information using electronic computers

1945 1969

Review: Transport Layer 9

From Simple, But Significant Ideas Bigger Ones Grow 1970s to 1995

Great efficiencies can be accomplished if we use The Internet and the World Wide Web to conduct business.

The World Wide Web is easier to use if we have a browser that To browser web pages, running in a graphical user interface context.

Computers connected via the Internet can be used more easily if hypertext links are enabled using HTML and URLs: it's called World Wide Web

The ARPANET needs to convert to a standard protocol and be renamed to The Internet

We need a protocol for Efficient and Reliable transmission of Packets over a WAN: TCP/IP

Ideas from 1940s to 1969

1970 1995

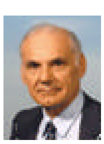
Review: Transport Layer 10

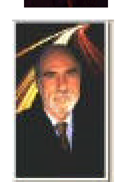
The Creation of the Internet

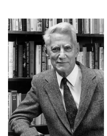
- The creation of the Internet solved the following challenges:
  - Basically inventing digital networking as we know it
  - Survivability of an infrastructure to send / receive high-speed electronic messages
  - Reliability of computer messaging

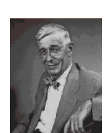
Internet Pioneers

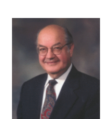
Vannevar Bush (APARNet)	Claude Shannon (Information theory)	Paul Baran (Pakcet switching)
Leonard Kleinrock (Pakcet switching)	Ted Nelson (Hypertext)	Lawrence Roberts (APARNet)
Vinton Cerf (TCP/IP)	Robert Kahn (TCP/IP)	Tim Berners-Lee (WWW)
	Mark Andreesen (Mosaic/Netscape)	

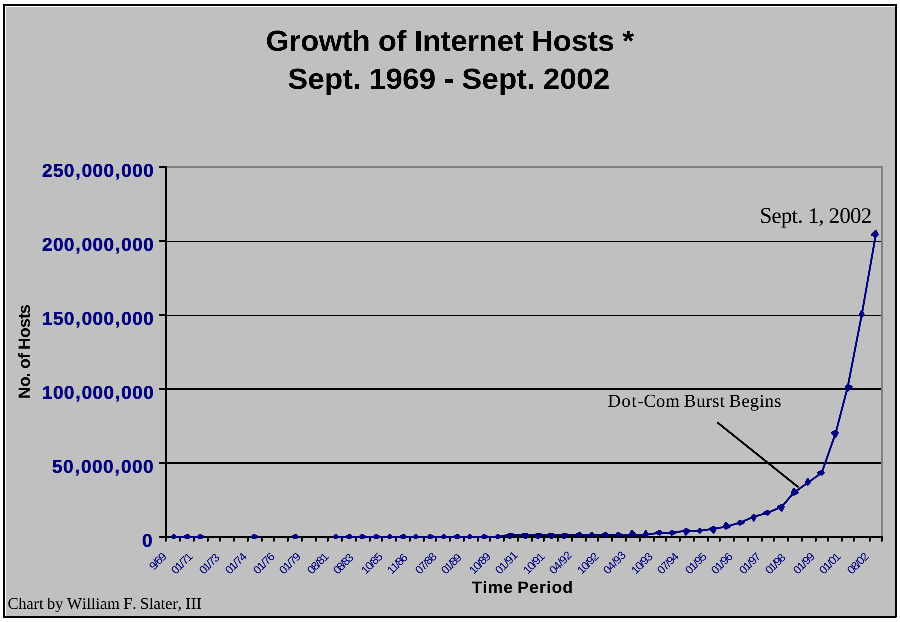






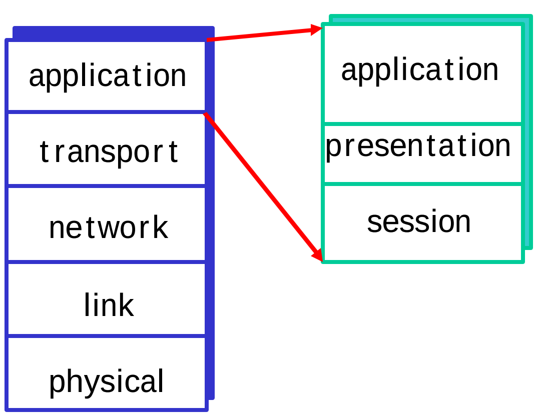






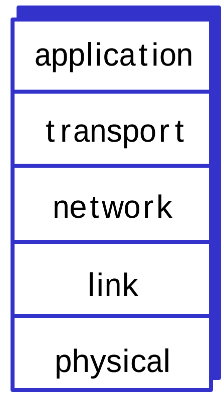
The Internet was not known as "The Internet" until January 1984, at which time there were 1000 hosts that were all converted over to using TCP/IP.

ISO 7-layer reference model



Internet protocol stack

- application: supporting network applications
  - FTP, SMTP, HTTP
- transport: host-host data transfer
  - TCP, UDP
- network: routing of datagrams from source to destination
  - IP, routing protocols e.g. OSPF, BGP
- link: data transfer between neighboring network elements
  - PPP, Ethernet
- physical: bits “on the wire”



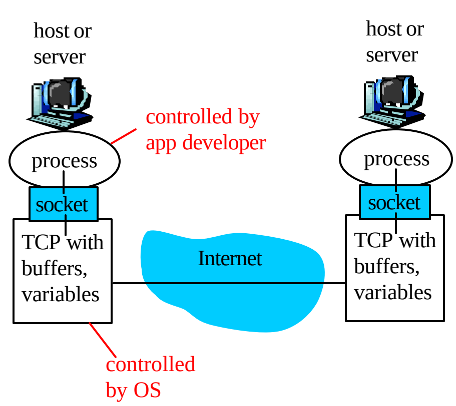
Internet Standardization Process

- All standards of the Internet are published as RFC (Request for Comments)
  - but not all RFCs are Internet Standards !
  - available: http://www.ietf.org
  - Till now: RFC4333
- A typical (but not the only) way of standardization:
  - Internet draft
  - RFC
  - Proposed standard
  - Draft standard (requires 2 working implementations)
  - Internet standard (declared by Internet Architecture Board)

Outline

- 1. Transport-layer services
- 2. Multiplexing and demultiplexing
- 3. Connectionless transport: UDP
- 4. Principles of reliable data transfer
- 5. Connection-oriented transport: TCP
- 6. TCP congestion control
- 7. TCP fairness and delay performance

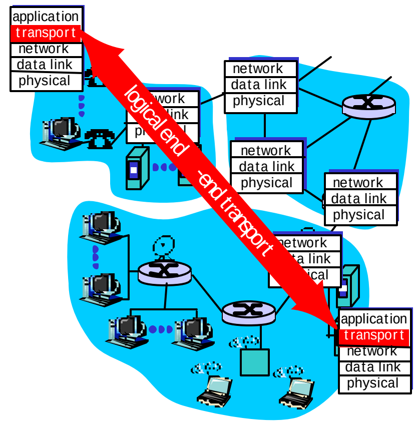
Transport layer – the other side of the door



- API: (1) choose transport protocol; (2) set parameters

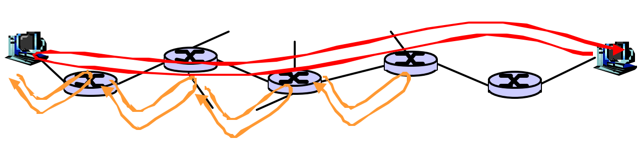
Transport services and protocols

- 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



Transport vs. network layer

- *network layer:* logical communication between hosts
  - Point-to-point
- *transport layer:* logical communication between processes
  - relies on and enhances, network layer services
  - also called “End-to-End”



J. Saltzer, D. Reed, and D. Clark. *End-to-end arguments in system design.* ACM Transactions on Computer Systems, 2(4):277--288, 1984.

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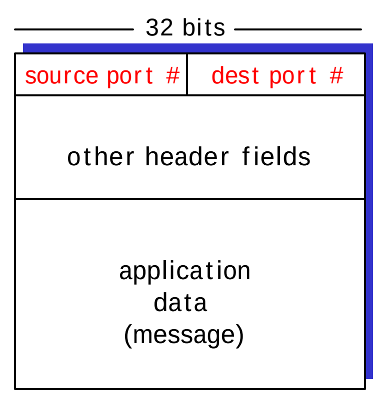
How demultiplexing works

- 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

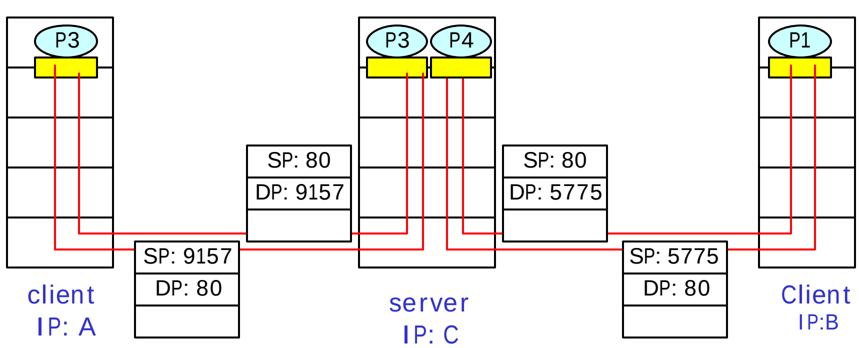


TCP/UDP segment format

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

Connection-oriented demux



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

Q:

- Why use 4-tuple?

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

Examples:

- 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 a different socket for each request

UDP: User Datagram Protocol [RFC 768]

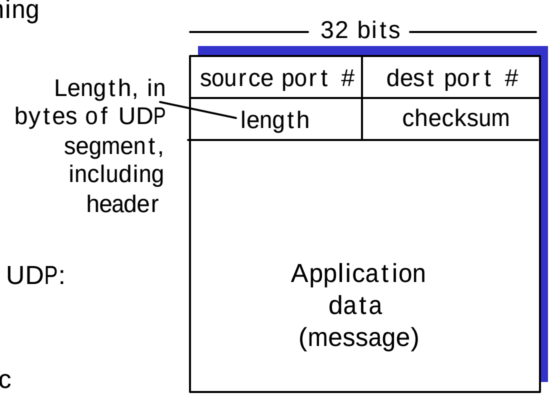
- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out of order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

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 – why ?
- reliable transfer over UDP: add reliability at application layer
  - application-specific error recovery!



UDP segment format

Outline

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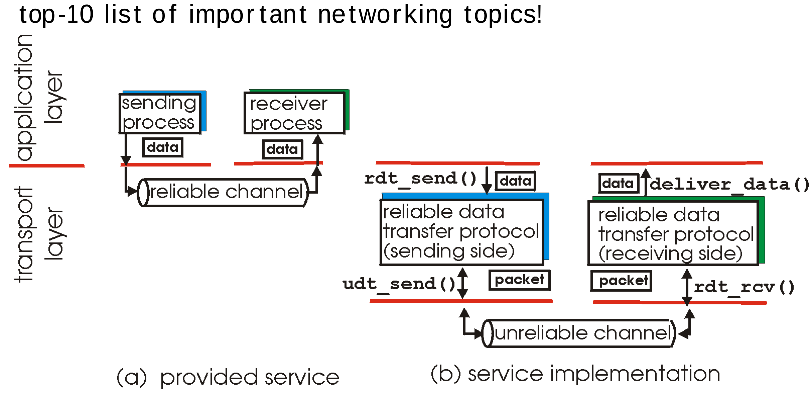
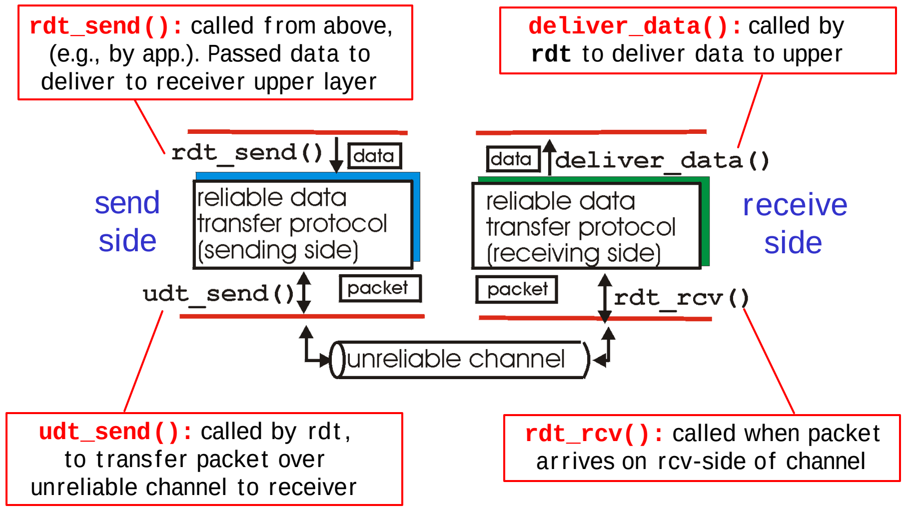
Principles of Reliable data transfer

- 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



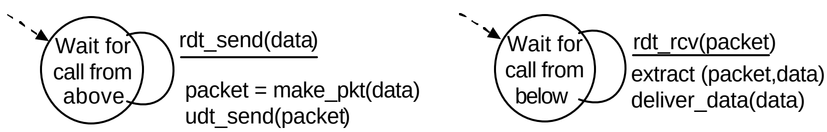
Reliable data transfer: getting started

We'll:

- incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- What is unreliability ?
  - Bit error
  - Packet loss – congestion
  - Delay – too long

Rdt1.0: reliable transfer over a reliable channel

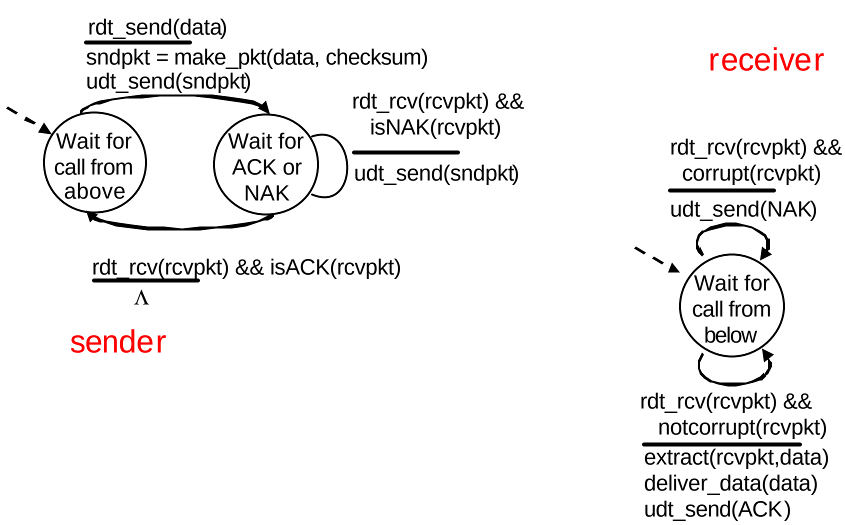
- underlying channel perfectly reliable
  - no bit errors
  - no loss of packets
- separate FSMs for sender, receiver:
  - sender sends data into underlying channel
  - receiver read data from underlying channel



sender

receiver

rdt2.0: channel with bit errors



rdt2.0 has a fatal flaw!

What happens if ACK/NAK corrupted?

- sender doesn't know what happened at receiver!

What to do?

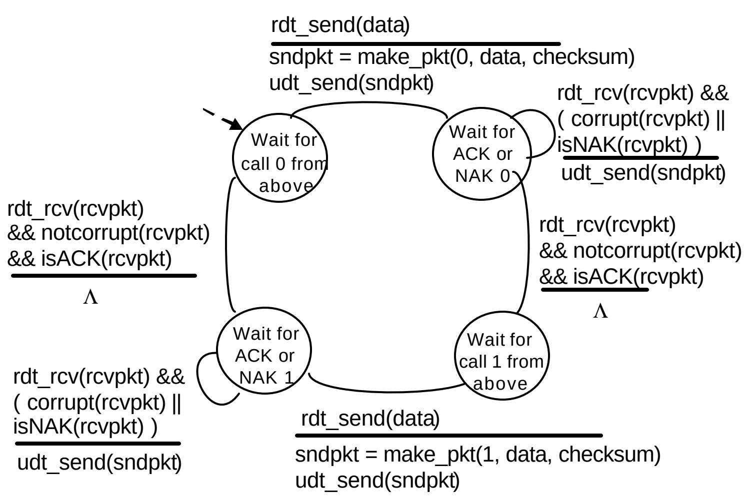
- sender NAKs for receiver's ACK/NAK? What if sender NAK corrupted?
- retransmit, assuming it is NAK …
- but this might cause retransmission of correctly received pkt!
  - packet duplications !

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

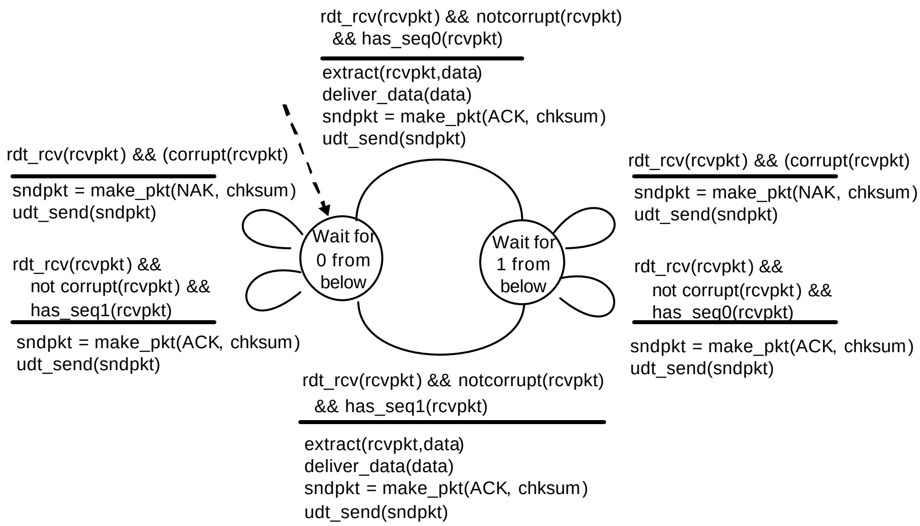
Review: Transport Layer 35

rdt2.1: sender, handles garbled ACK/NAKs

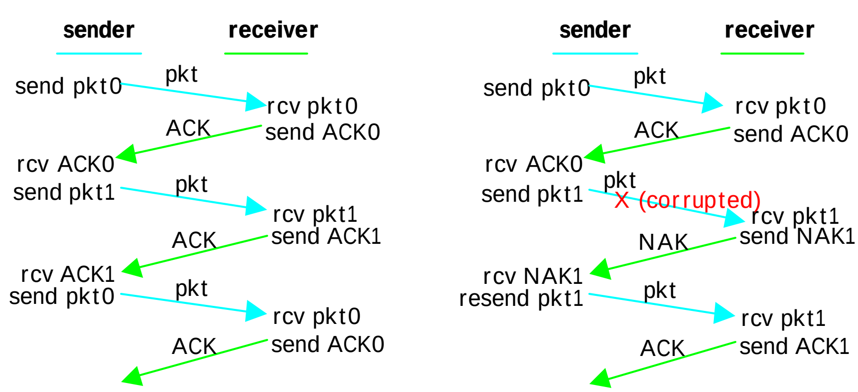


Review: Transport Layer 36

rdt2.1: receiver, handles garbled ACK/NAKs



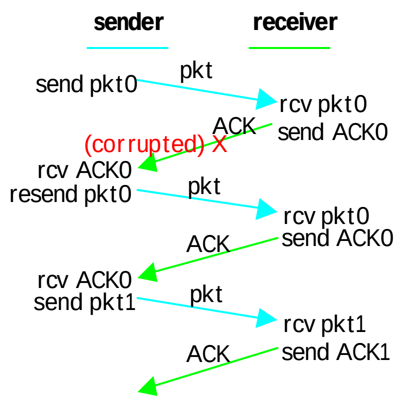
rdt 2.1 in action



a) operation with no corruption

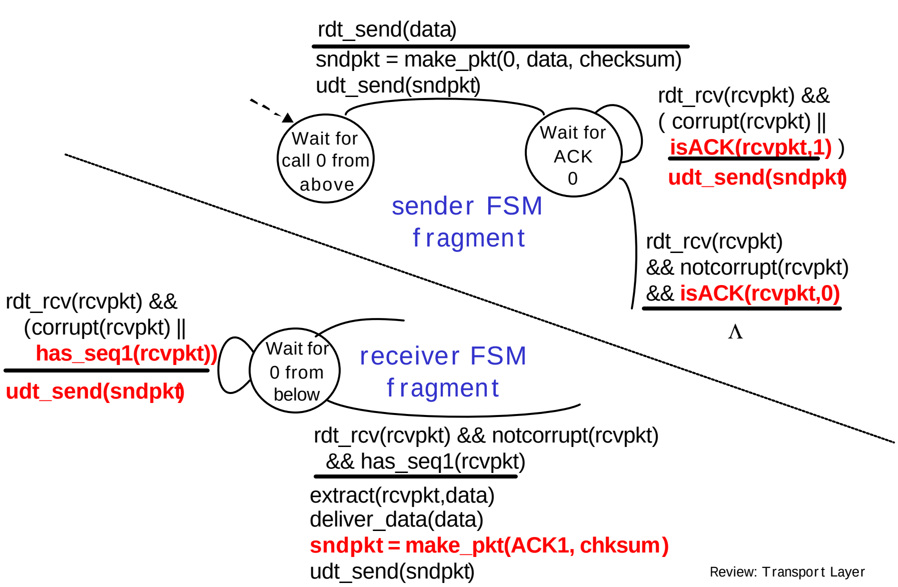
b) packet corrupted

rdt 2.1 in action (cont)

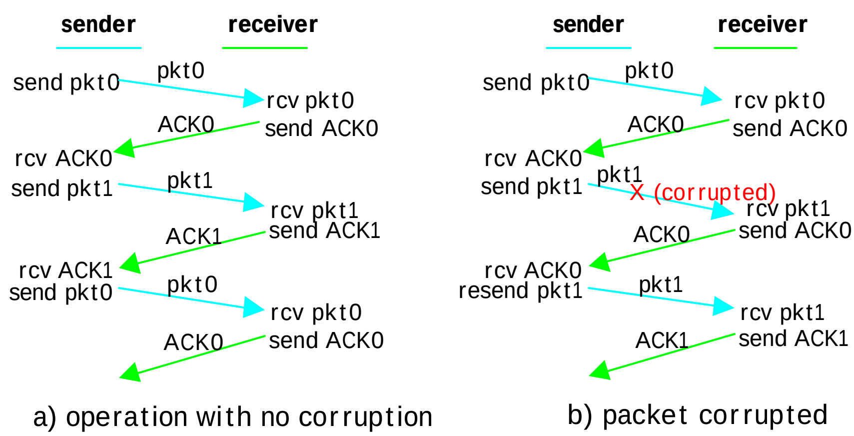


c) ACK corrupted

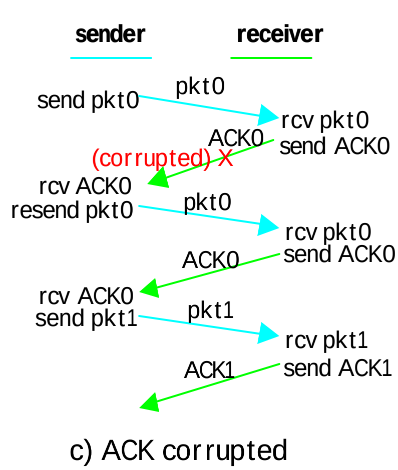
rdt2.2: a NAK-free protocol



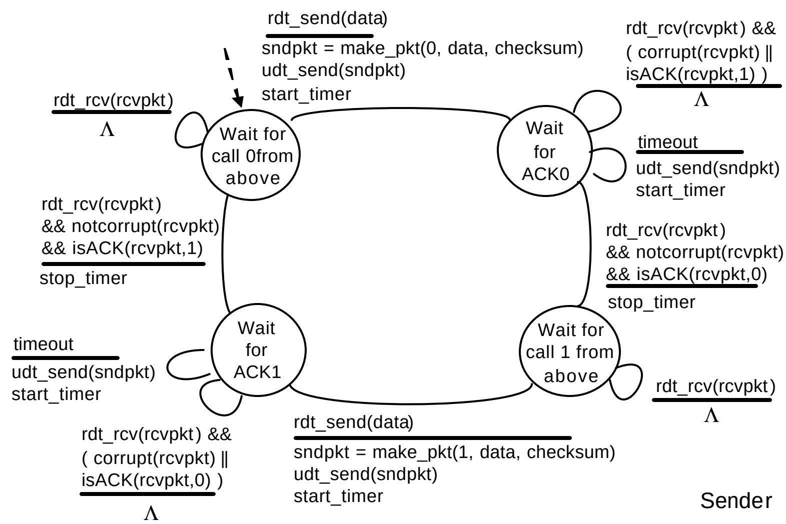
rdt 2.2 in action



rdt 2.2 in action (cont)

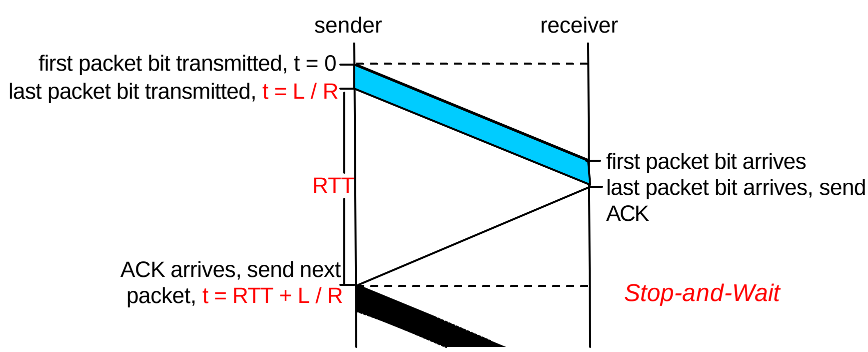


rdt3.0 channels with errors *and* loss



Sender

rdt3.0: Poor performance



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

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

Performance of rdt3.0

- example: 1 Gbps link, 15 ms e-e prop. delay, 1KB packet:

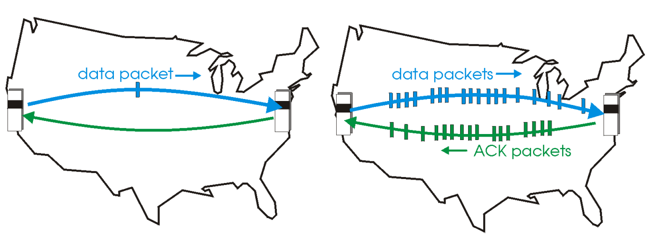
$$T_{transmit} = \frac{L \text{ (packet length in bits)}}{R \text{ (transmission rate, bps)}} = \frac{8\text{kb/pkt}}{10^9 \text{ b/sec}} = 8 \text{ microsec}$$

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

- U_{sender}: utilization – fraction of time sender busy sending
- 1KB pkt every 30 msec -> 33kB/sec thruput over 1 Gbps link
- network protocol limits use of physical resources!
- microsec = 10^{-6}sec millisec=ms=10^{-3}s Gb, Mb, Kb

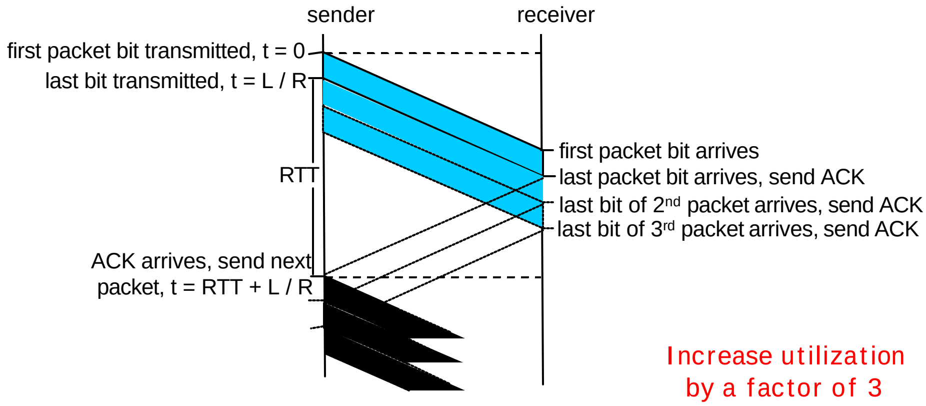
Pipelined protocols

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

Pipelining: increased utilization



Increase utilization by a factor of 3

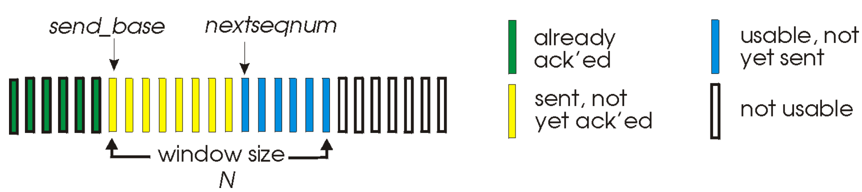
$$U_{sender} = \frac{3 * L / R}{RTT + L / R} = \frac{.024}{30.008} = 0.0008$$

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

Go-Back-N

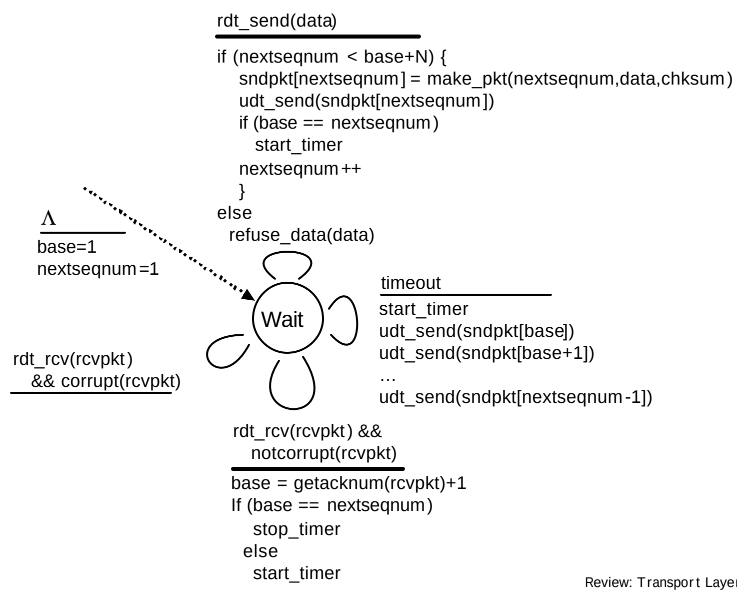
Sender:

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

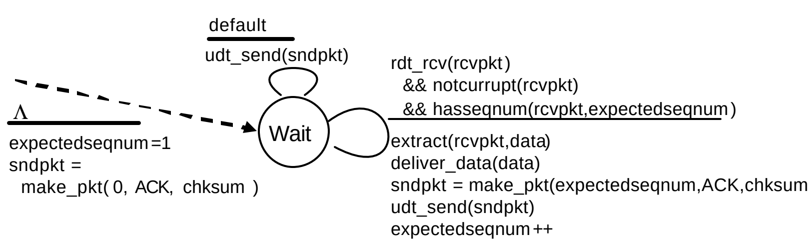


- ACK(n): ACKs all pkts up to, including seq # n - "cumulative ACK"
  - may receive duplicate ACKs (see receiver)
- timer for the packet of send_base
- *timeout(n):* retransmit pkt n and all higher seq # pkts in window

GBN: sender extended FSM



GBN: receiver extended FSM

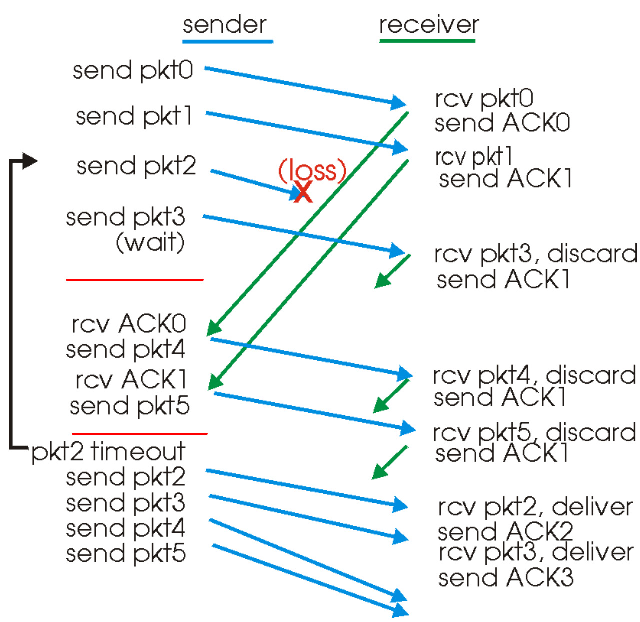


ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- 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



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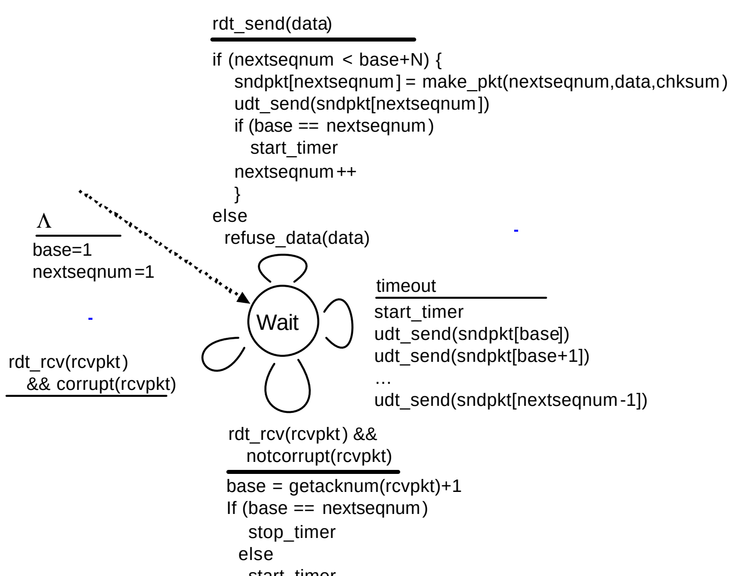
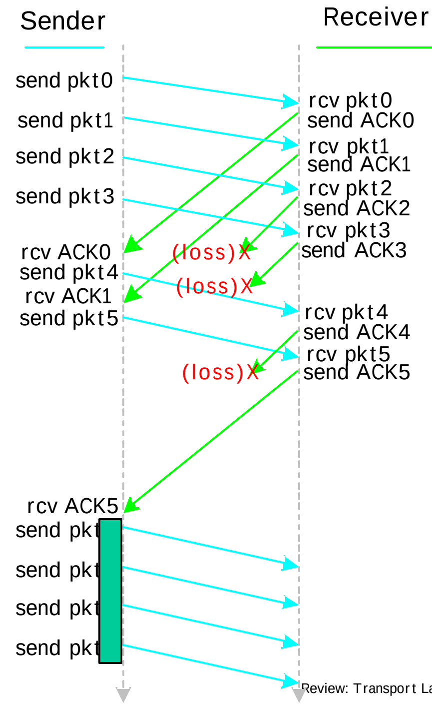


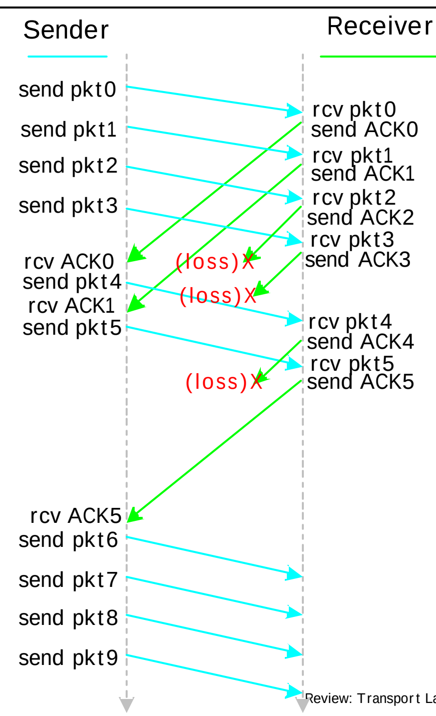
GBN in action

Cumulative ACK



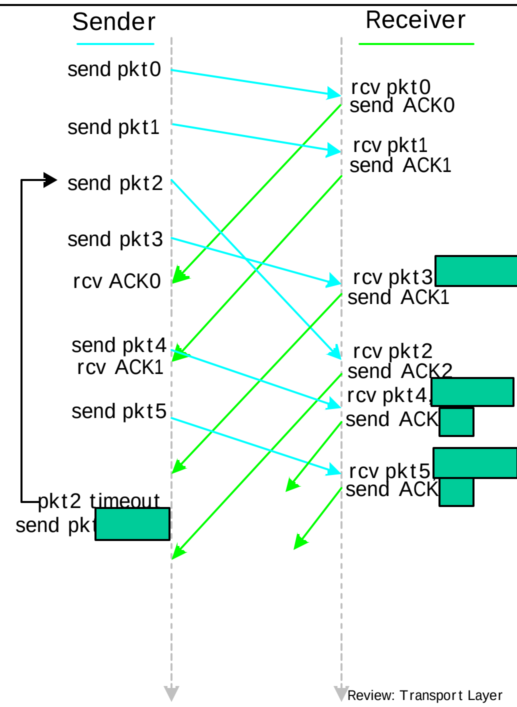
GBN in action

Cumulative ACK



GBN in action

Premature timeout



GBN in action

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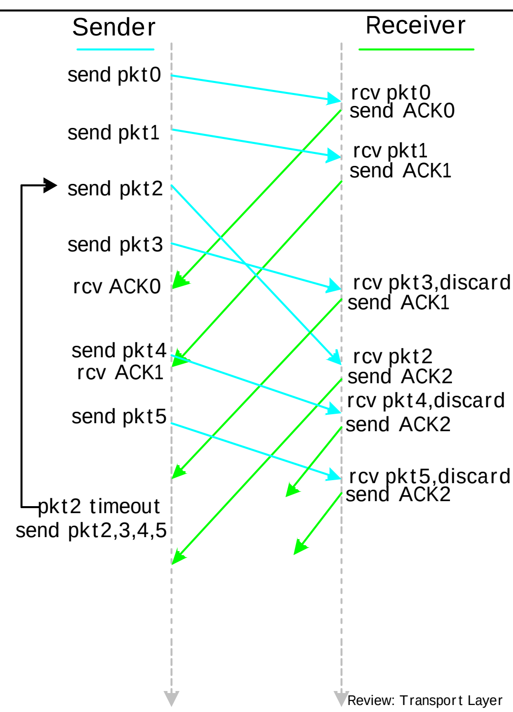
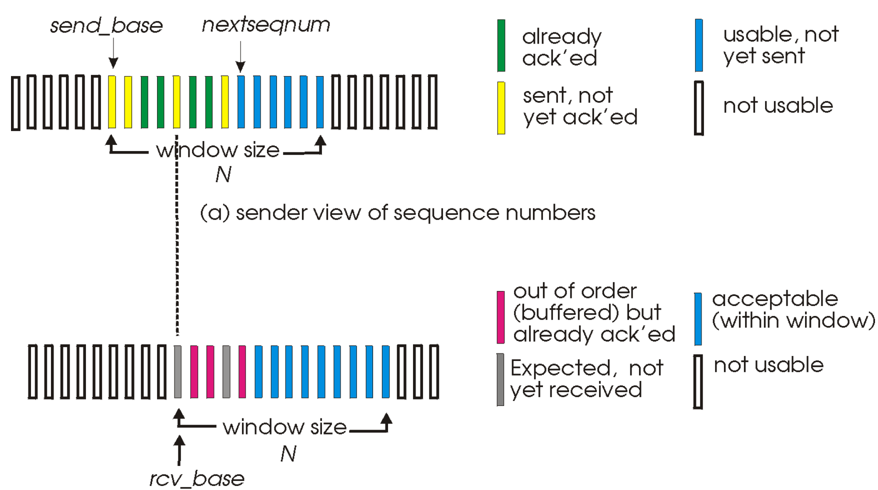


Selective Repeat

- receiver *individually* acknowledges all correctly received pkts
  - buffers pkts, as needed, for eventual in-order delivery to upper layer
- sender only 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

(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

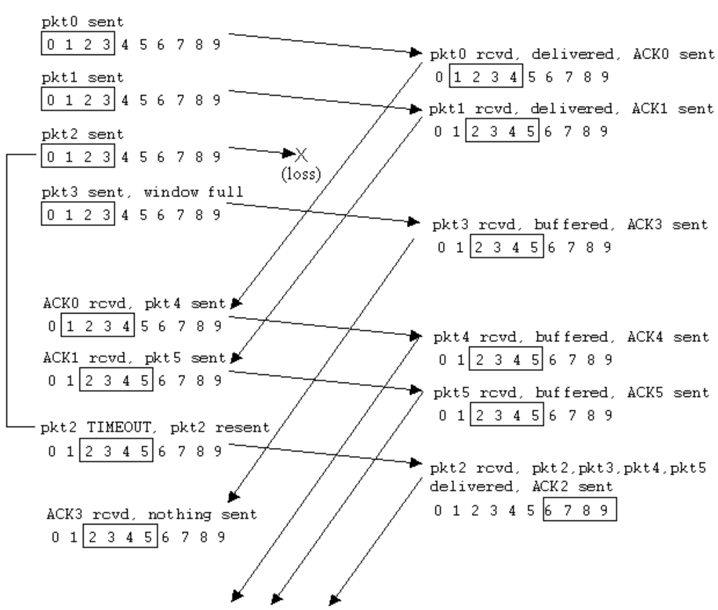
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore

Selective repeat in action



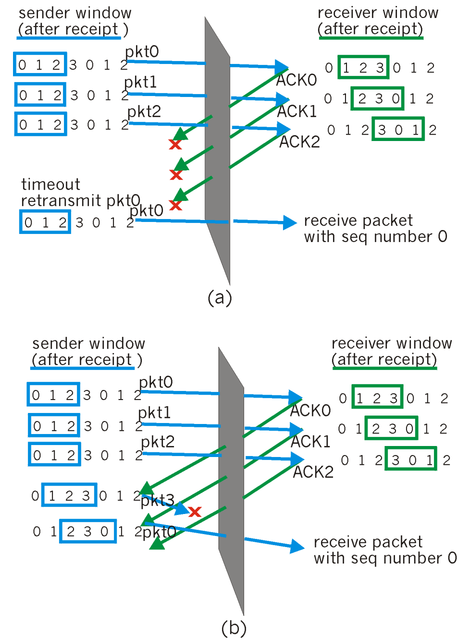
Selective repeat: dilemma

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? Will this happen in GBN ?



Go Back N vs. Selective Repeat

- Efficiency
  - No loss
  - Loss
    - Bursty loss
    - Sporadic loss
- Resource consumption
  - Buffer space
  - Timer
    - How to implement multi-timers ?

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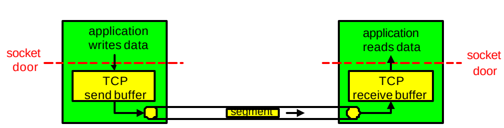
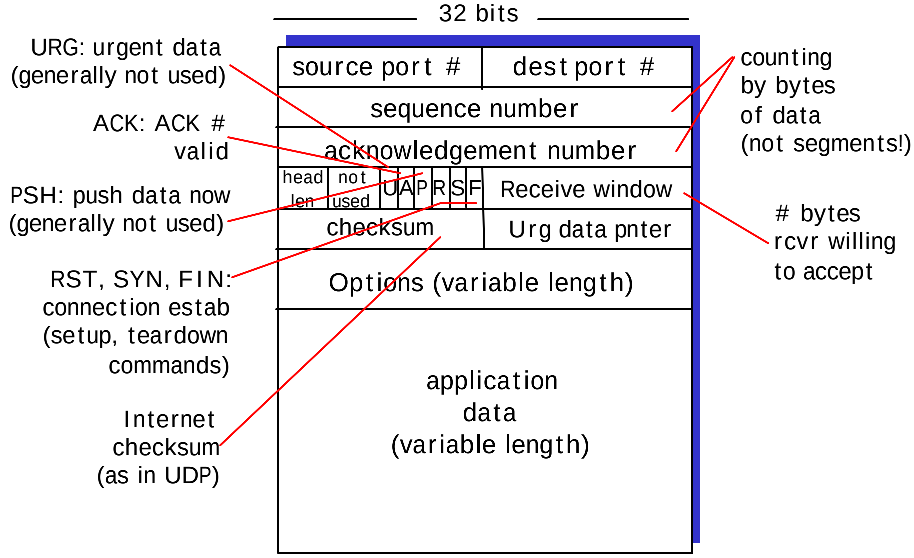
TCP: Overview

RFCs: 793, 1122, 1323, 2018, 2581

- End-to-end, unicast:
  - one sender, one receiver
- reliable, in-order *byte steam:*
  - no "message boundaries"
- Pipelined (not stop-wait):
  - TCP congestion and flow control set window size
  - *send & receive buffers*
- full duplex data:
  - bi-directional data flow in same connection
- 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



TCP Connection Setup

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 – *piggyback*

Q: Is 3-way handshake perfect ?

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 – cumulative ack
  - start timer if there are outstanding segments

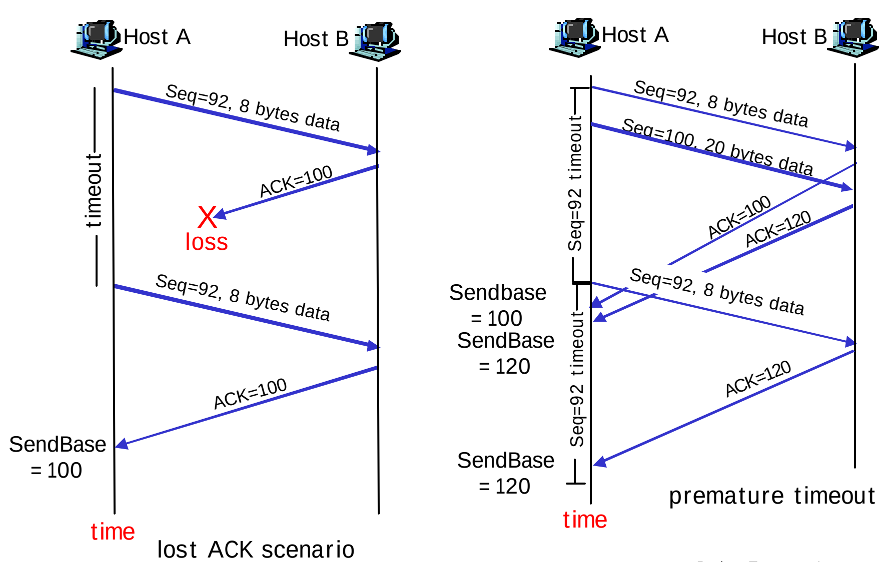
```
NextSeqNum = InitialSeqNum
SendBase = InitialSeqNum

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 */
```

TCP sender (simplified)

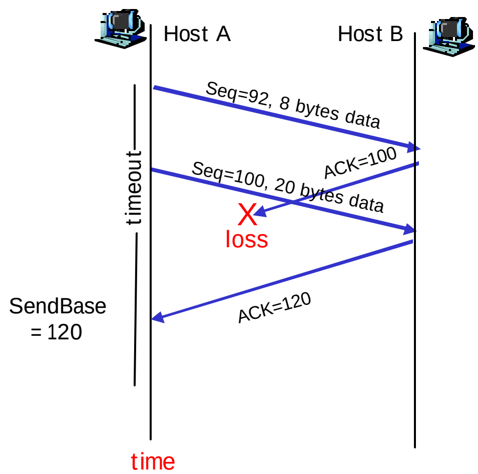
Comment:
- SendBase-1: last cumulatively ack'ed byte

Example:
- 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

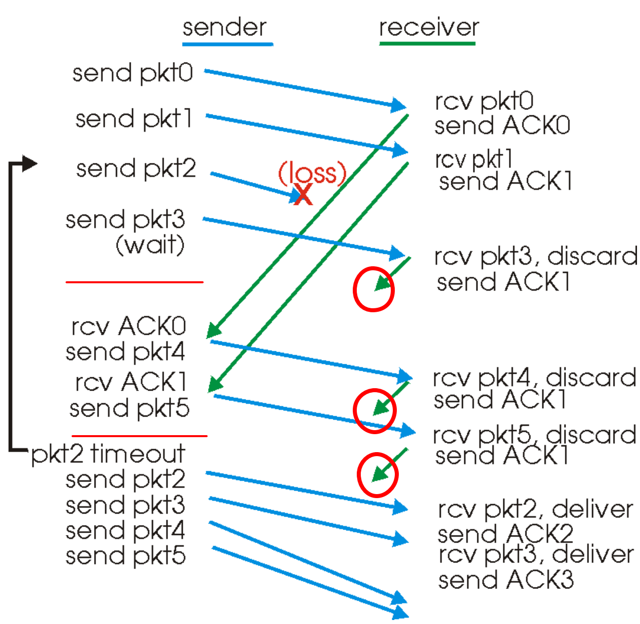
TCP ACK generation [RFC 1122, RFC 2581]

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 segment that partially or completely fills gap	Immediate send ACK, provided that segment starts at lower end of gap
Arrival of out-of-order segment higher-than-expect seq. # . Gap detected	Immediately send duplicate ACK, indicating seq. # of next expected byte

Fast Retransmit

- Time-out period may be relatively long:
  - eRTT+4DevRTT
  - long delay before resending lost packet
- Solution: Fast Retransmit
  - Hint: GBN

GBN in action



Fast Retransmit

- Time-out period may be relatively long:
  - eRTT+4DevRTT
  - 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 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

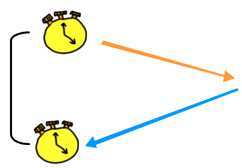
fast retransmit

TCP Round Trip Time and Timeout

Q: how to estimate RTT?

- `SampleRTT`: measured time from segment transmission until ACK receipt

One RTT sample

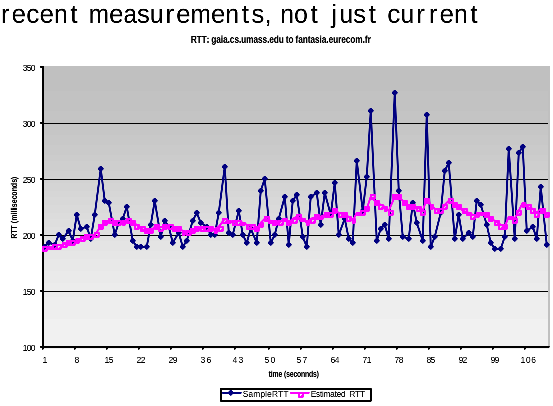


TCP Round Trip Time and Timeout

- **Problem 2:**

  `SampleRTT` will vary -> atypical

  - Need the trend of RTT: history –> future
  - average several recent measurements, not just current **`SampleRTT`**



TCP Round Trip Time and Timeout

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

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

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Principles of Congestion Control

Congestion:

- informally: "too many sources sending too many data too fast for *network* to handle"
- Solution
  - Sender controls sending rate
- different from flow control!
  - Flow control: not overwhelm receiver
  - Congestion control: not overwhelm network
- another top-10 problem!

Approaches towards congestion control

Two broad approaches towards congestion control:

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

Fast, accurate, but expensive

End-end congestion control:

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

TCP Congestion Control

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

`LastByteSent-LastByteAcked`
`£ CongWin`

`RcvWindow`?
`£ min { rcwWindow, CongWin }`

- **`CongWin`** is dynamic, function of perceived network congestion
  - Too high a rate -> congestion
  - Too low a rate -> low network utilization

TCP Congestion Control

How does sender perceive congestion?

- loss event
- TCP sender reduces rate (**`CongWin`**) after loss event

Loss event = timeout *or* 3 duplicate acks

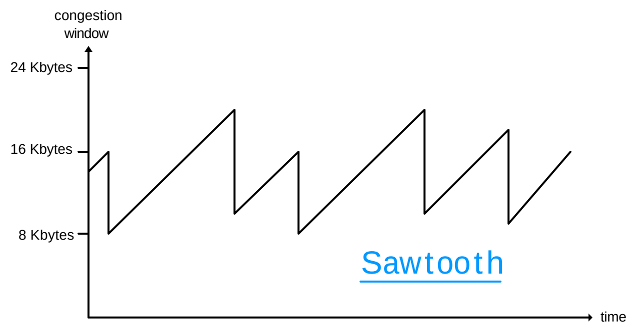
three mechanisms:

- AIMD (additive increase multiplicative decrease)
- slow start
- conservative after timeout events

1. 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



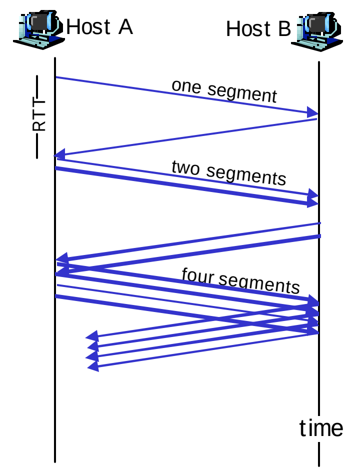
Long-lived TCP connection

2. 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

2. 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



3. Refinement (TCP Reno)

- 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”

TCP versions:

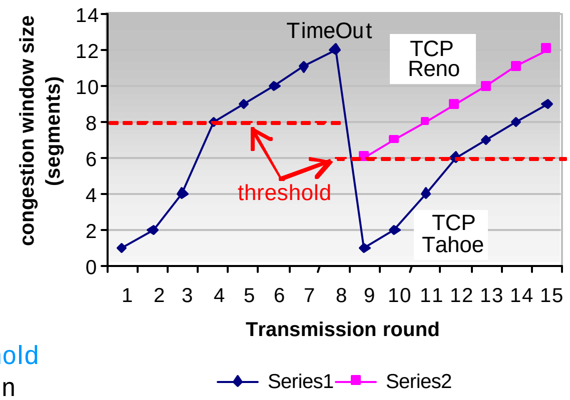
Tahoe -> Reno -> Sack

Vegas, Westwood …
(Nevada)

Refinement (more)

Q: Threshold: When will exponential increase switch to linear?

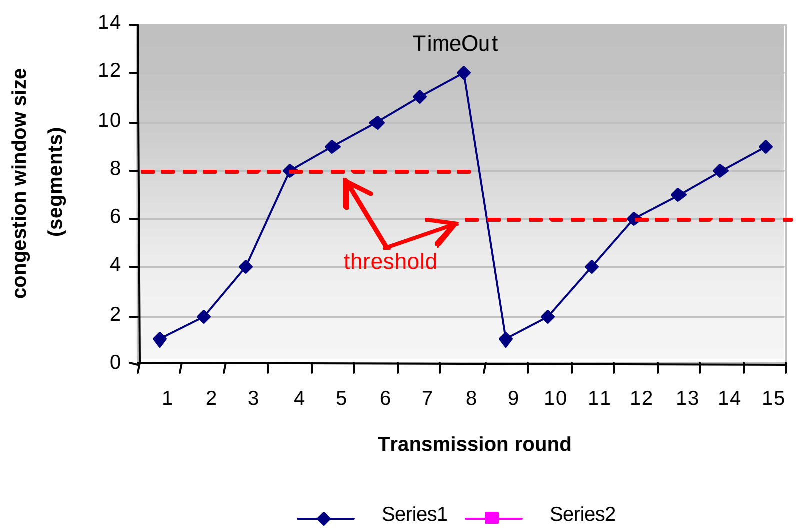
A: When **CongWin** gets to 1/2 of its value before timeout.



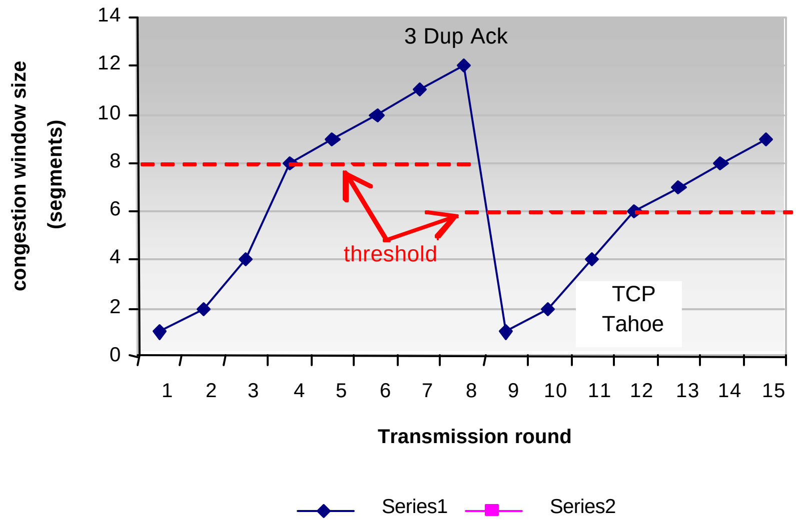
Implementation:

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

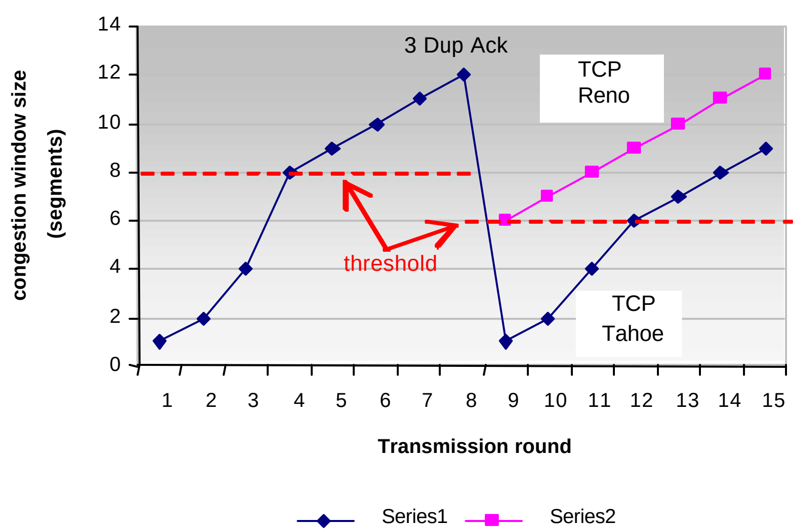
TCP congestion behavior (1)



TCP congestion behavior (2)



TCP congestion behavior (3)



Summary: TCP Congestion Control (Reno)

- 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.

V. Jacobson, Congestion Avoidance and Control. Proceedings of ACM SIGCOMM '88, Aug. 1988.

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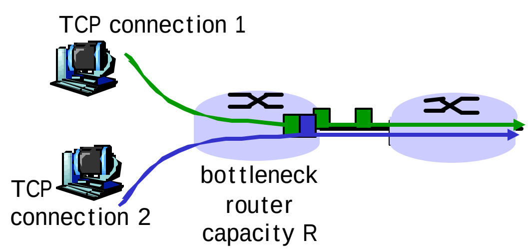
TCP Fairness

Fair: 1. Equal share

2. Full utilization

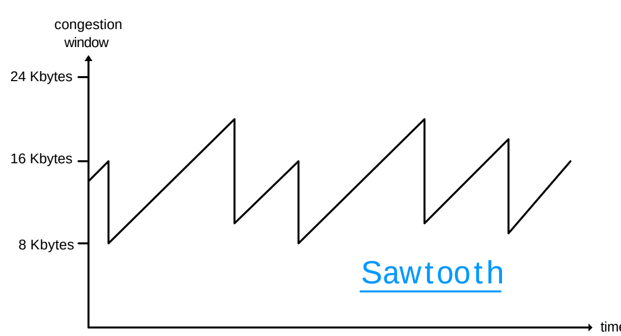
Goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



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

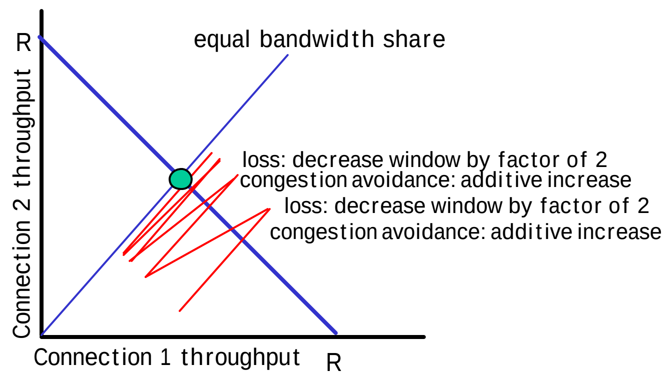


Long-lived TCP connection

Why is TCP fair?

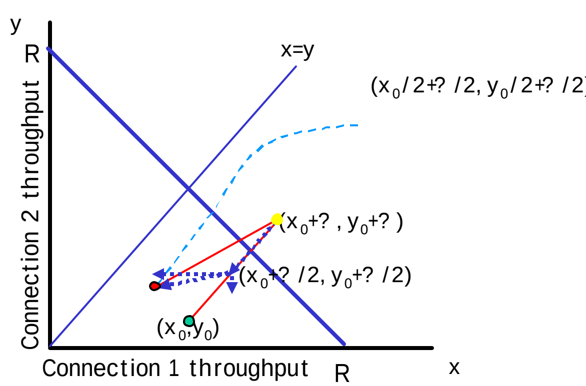
Two competing sessions:

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



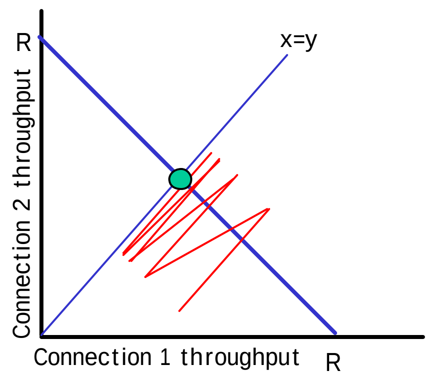
Why is TCP fair?

Known:
$x_0 > y_0$



Why is TCP fair?

D.M. Chiu and R. Jain, "*Analysis of the Increase and Decrease Algorithms for Congestion Avoidance in Computer Networks*," Computer Networks and ISDN Systems, pp. 1-14, 1989.



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, more on later

Fairness and parallel TCP connections

- nothing prevents app from opening parallel connections between 2 hosts.
- Web browsers/FTP client do this
  - **NetAnts, GetRight**
- Example: link of rate R with 9 ongoing Tcp connections;
  - new app asks for 1 TCP, gets rate R/10
  - new app asks for 11 TCPs, gets > R/2 !

Delay performance

Q: How long does it take to receive an object from a Web server after sending a request?

Methods

- Measurement
  - Ping, traceroute
- Simulation
  - Ns-2
- Analytical modeling
  - Math

Delay modeling – No Congestion

Q: How long does it take to receive an object from a Web server after sending a request?

Ignoring congestion, delay is influenced by:

- TCP connection establishment
- data transmission delay
- slow start

Notation, assumptions:

- Assume one link between client and server of rate R
- S: MSS (bits)
- O: object size (bits)
- no retransmissions (no loss, no corruption)

Window size:

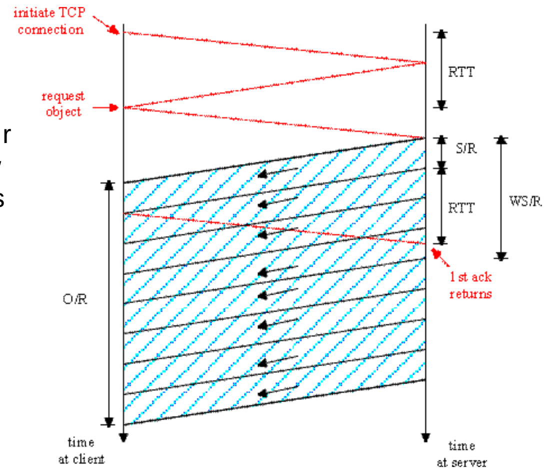
- First assume: fixed congestion window, W segments
- Then dynamic window, modeling slow start

Fixed congestion window (1)

First case:

WS/R > RTT + S/R: ACK for first segment in window returns before window's worth of data sent

delay = ?

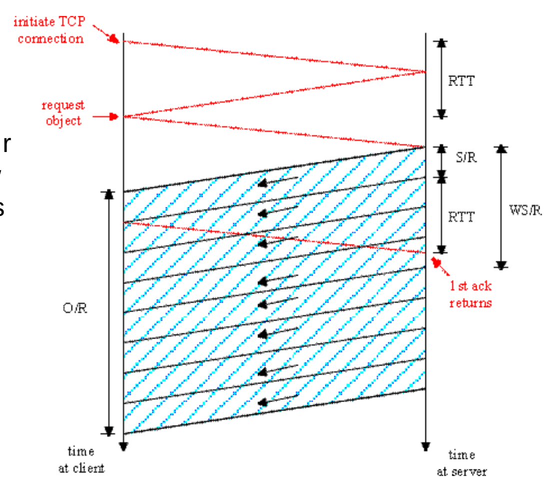


Fixed congestion window (1)

First case:

WS/R > RTT + S/R: ACK for first segment in window returns before window's worth of data sent

delay = 2RTT + O/R

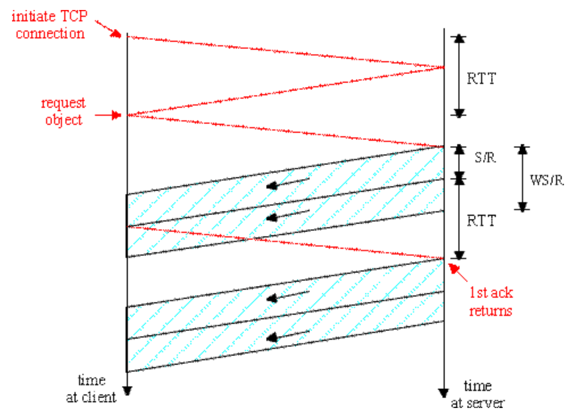


Fixed congestion window (2)

Second case:

- WS/R < RTT + S/R: wait for ACK after sending window's worth of data sent

delay = ?



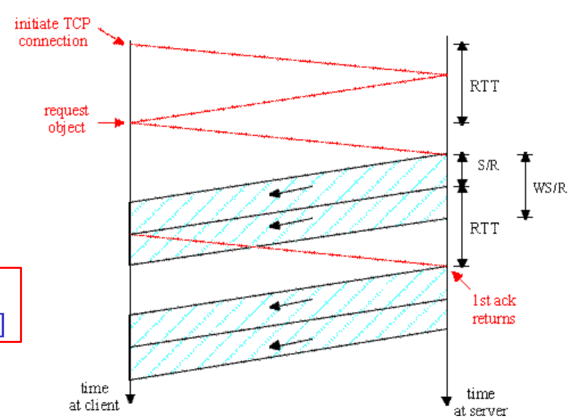
Fixed congestion window (2)

Second case:

- WS/R < RTT + S/R: wait for ACK after sending window's worth of data sent

delay = 2RTT + O/R + (K-1)[S/R + RTT - WS/R]

K ?



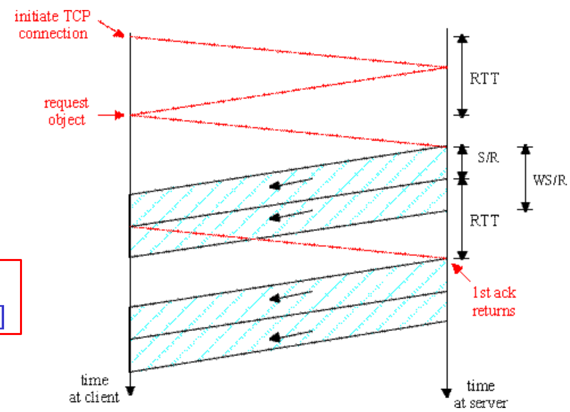
Fixed congestion window (2)

Second case:

- WS/R < RTT + S/R: wait for ACK after sending window's worth of data sent

delay = 2RTT + O/R + (K-1)[S/R + RTT - WS/R]

K =O/(WS)



TCP Delay Modeling: Slow Start (1)

Now suppose window grows according to slow start

But no congestion

Will show that the delay for one object is:

$$Latency = 2RTT + \frac{O}{R} + P\left[RTT + \frac{S}{R}\right] - (2^P - 1)\frac{S}{R}$$

where P is the number of times TCP idles at server:

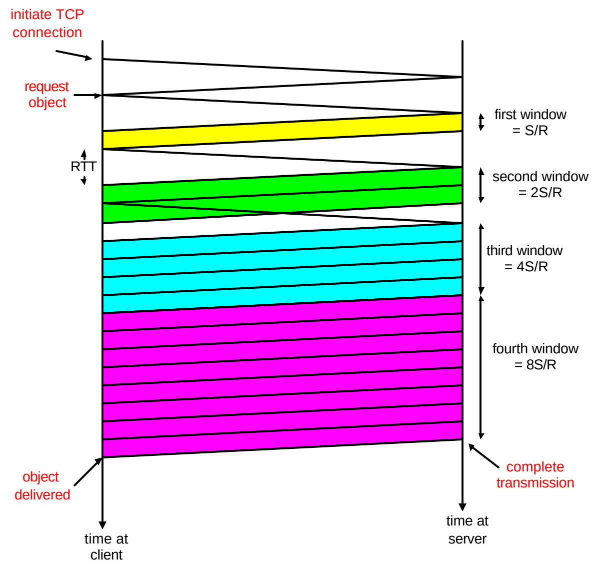
$$P = \min\{Q, K-1\}$$

- Q is the number of times the server idles if the object were of infinite size.
- K is the number of windows that cover the object.

Case 1: P = Q

Delay components:
- 2 RTT for connection estab and request
- O/R to transmit object
- time server idles due to slow start

Server idles:
P = min{K-1,Q} times



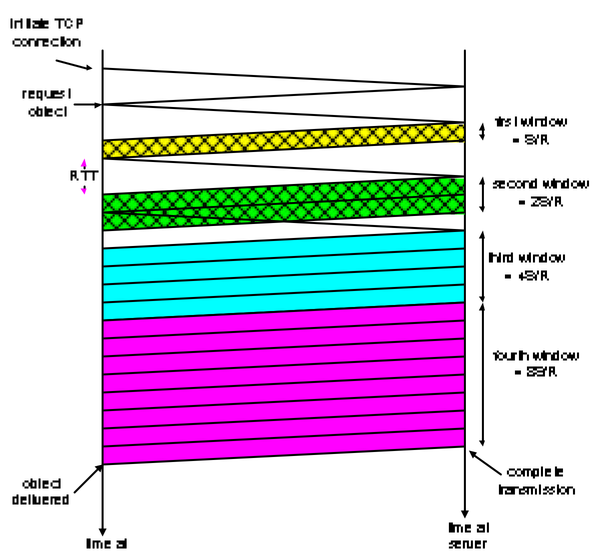
Example:
- O/S = 15 segments
- K = 4 windows
- Q = 2
- P = min{K-1,Q} = 2

Server idles P=2 times

Case 2: P = K-1

Delay components:
- 2 RTT for connection estab and request
- O/R to transmit object
- time server idles due to slow start

Server idles:
P = min{K-1,Q} times



Example:
- O/S = 3 segments
- K = 2 windows
- Q = 2
- P = min{K-1,Q} = 1

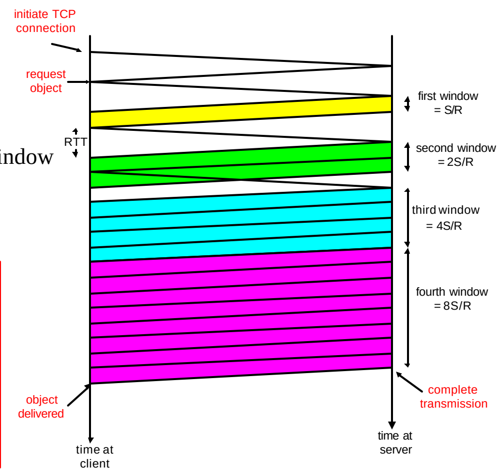
Server idles P=1 times

TCP Delay Modeling (contd)

$$\frac{S}{R} + RTT = \text{time from when server starts to send segment until server receives acknowledgement}$$

$$2^{k-1}\frac{S}{R} = \text{time to transmit the } k\text{th window}$$

$$\left[\frac{S}{R} + RTT - 2^{k-1}\frac{S}{R}\right]^{+} = \text{idle time after the } k\text{th window}$$



$$\boxed{\begin{aligned} \text{delay} &= \frac{O}{R} + 2RTT + \sum_{p=1}^{P} idleTime_p \\ &= \frac{O}{R} + 2RTT + \sum_{k=1}^{P}\left[\frac{S}{R} + RTT - 2^{k-1}\frac{S}{R}\right] \\ &= \frac{O}{R} + 2RTT + P\left[RTT + \frac{S}{R}\right] - (2^P - 1)\frac{S}{R} \end{aligned}}$$

TCP Delay Modeling (contd)

Recall K = number of windows that cover object

How do we calculate K ?

$$\begin{aligned} K &= \min\{k : 2^0 S + 2^1 S + \cdots + 2^{k-1} S \geq O\} \\ &= \min\{k : 2^0 + 2^1 + \cdots + 2^{k-1} \geq O/S\} \\ &= \min\{k : 2^k - 1 \geq \frac{O}{S}\} \\ &= \min\{k : k \geq \log_2(\frac{O}{S} + 1)\} \\ &= \left\lceil \log_2(\frac{O}{S} + 1) \right\rceil \end{aligned}$$

Calculation of Q, number of idles for infinite-size object, is similar

$$\max\{q : 2^{q-1} S/R \leq RTT + S/R\}$$