User Datagram Protocol (UDP)
Transmission Control Protocol (TCP)

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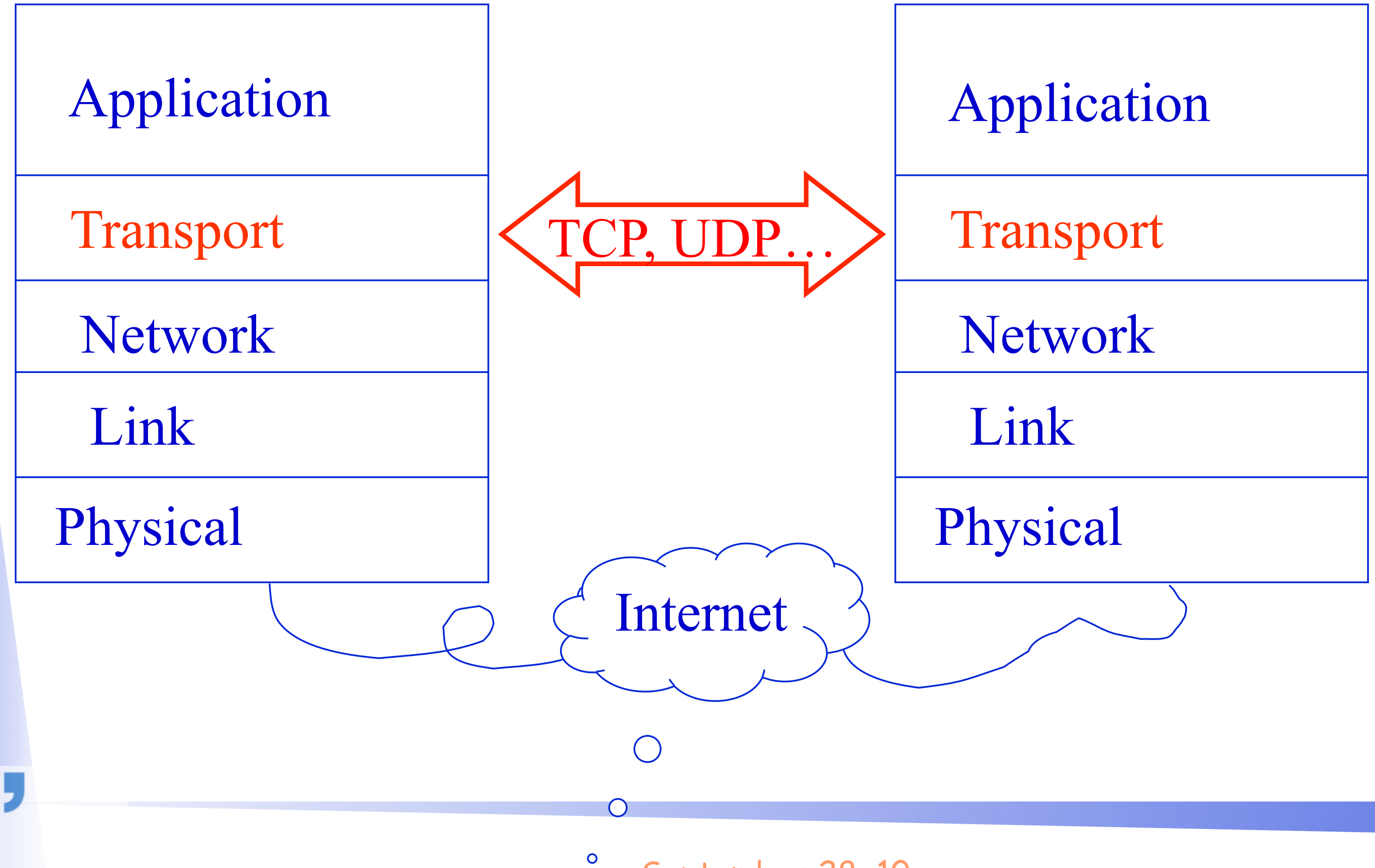
28.09.2010

Some material from "Computer Networking: A Top Down Approach" by Jim Kurose, Keith Ross.

Outline

- Background
- UDP
  - Role and Functioning
- TCP
  - Basics
  - Error control
  - Flow control
  - Congestion control

Transport layer



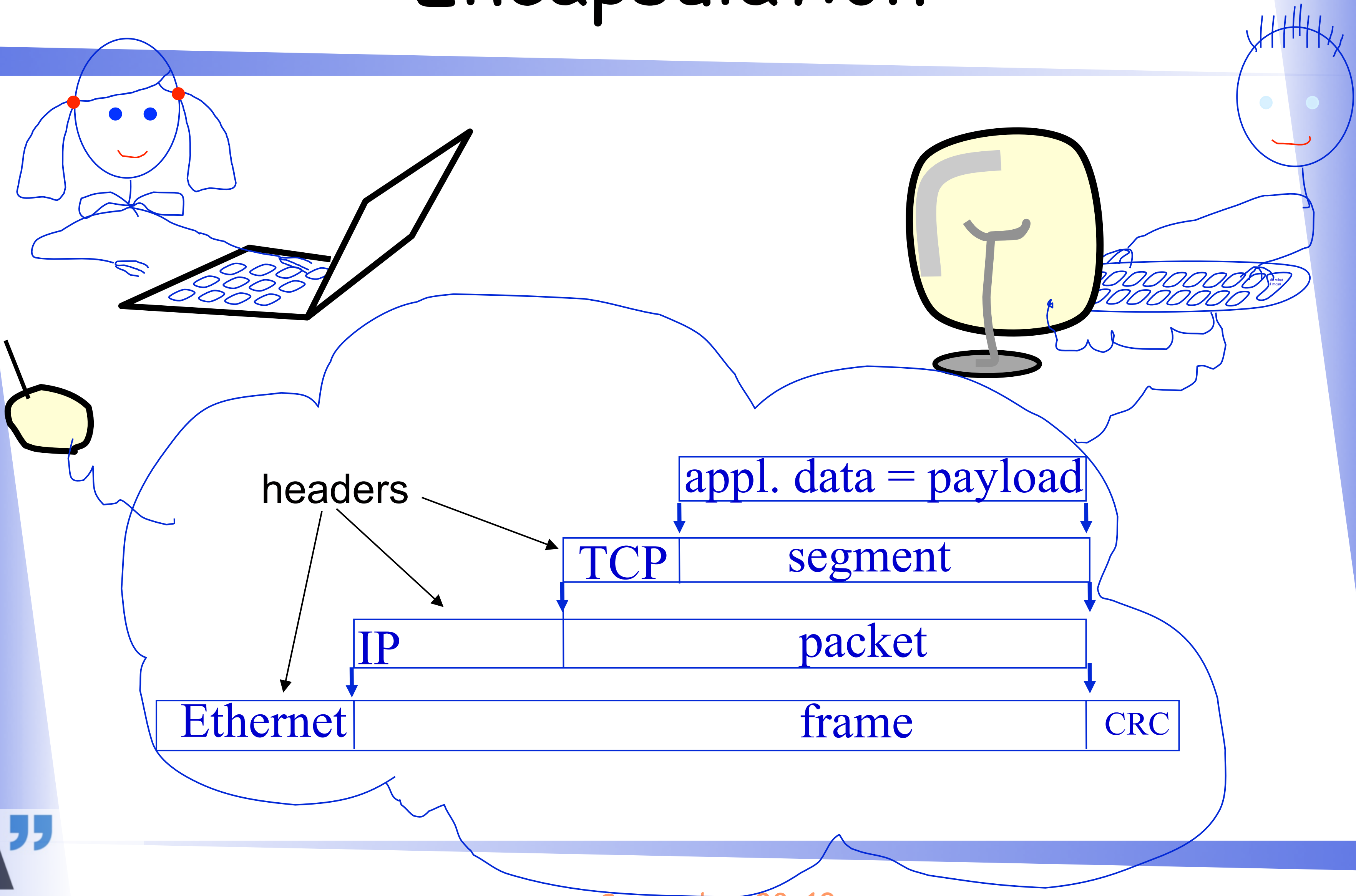
Transport layer (cont.)

- Offers end-to-end transport of data for applications
- Different characteristics
  - Reliable vs. unreliable
  - Forward error correction (FEC) vs. Automatic Repeat-reQuest (ARQ)
  - TCP friendly or not
  - Structured vs. unstructured stream
  - …

Reliable vs. best effort

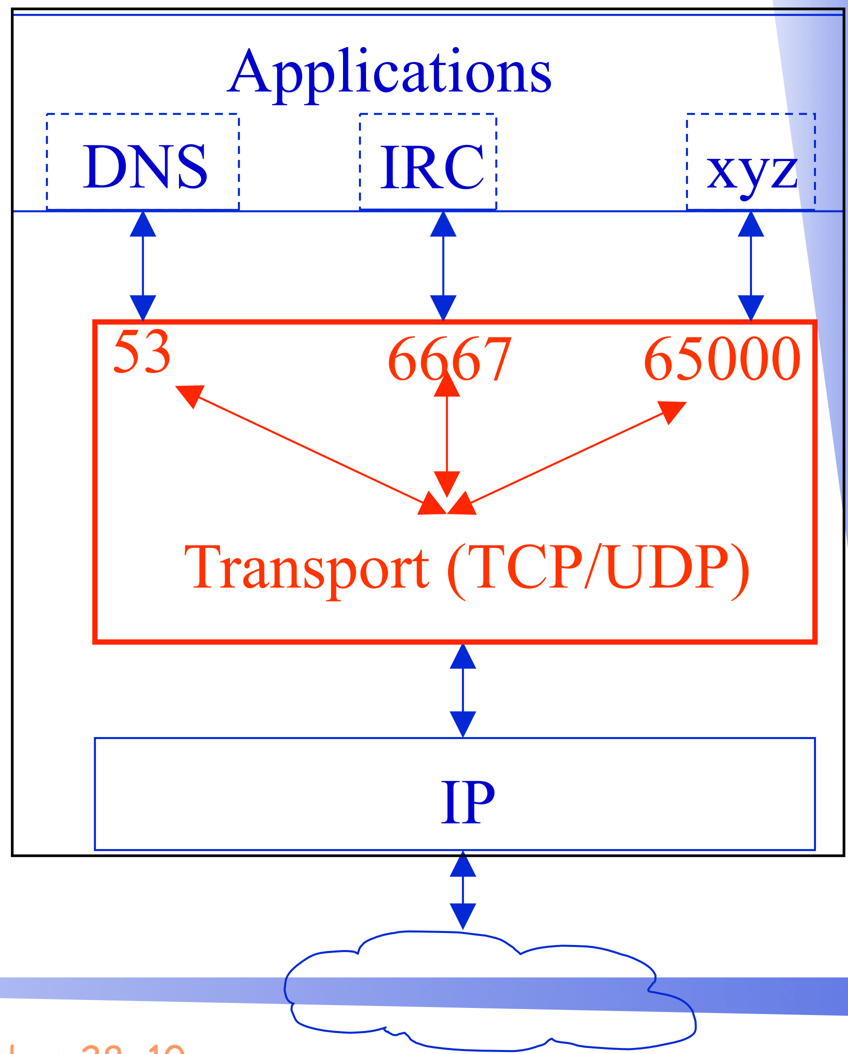
- TCP – reliable transport
  - Guarantees ordered delivery of packets
  - Important for e.g.
    - Signaling messages
    - File transfer
- UDP – best effort transport
  - No guarantees of packet delivery
  - Non-critical data delivery, e.g. VoIP

Encapsulation



Role of ports

- Well-known port numbers
  - RFC 2780 (&4443)
  - 0-1023
- Registered port numbers
  - 1024-49151
- Other port numbers
  - 49152-65535



Checksums

- For detecting damaged packets
  - Compute at sender, check at receiver
- Computed from pseudo-header and transport segment
  - Pseudo header includes
    - source and destination IP addresses
    - protocol number
    - TCP/UDP length
    - Slightly different method for IPv4 (RFC 768/793) and IPv6 (RFC 2460)
    - Included for protection against misrouted segments
  - Divide into 16-bit words and compute one's complement of the one's complement sum of all the words

Part 2: UDP - User Datagram Protocol

User Datagram Protocol (UDP)

- Lightweight protocol
  - Just add port numbering and integrity checking (checksums) to IP
  - No segmentation
- Unreliable connectionless transport service
  - No acknowledgments and no retransmissions
  - Checksum optional in IPv4 and mandatory in IPv6

UDP datagram

0	16 31
UDP SOURCE PORT	UDP DESTINATION PORT
UDP MSG LENGTH	UDP CHECKSUM
DATA ...	

- Source port and checksum are optional
  - Checksum mandatory with IPv6
- Length: header and data in bytes
- Ports provide application multiplexing within a host (single IP)

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Part 3: TCP - Transmission Control Protocol

Outline

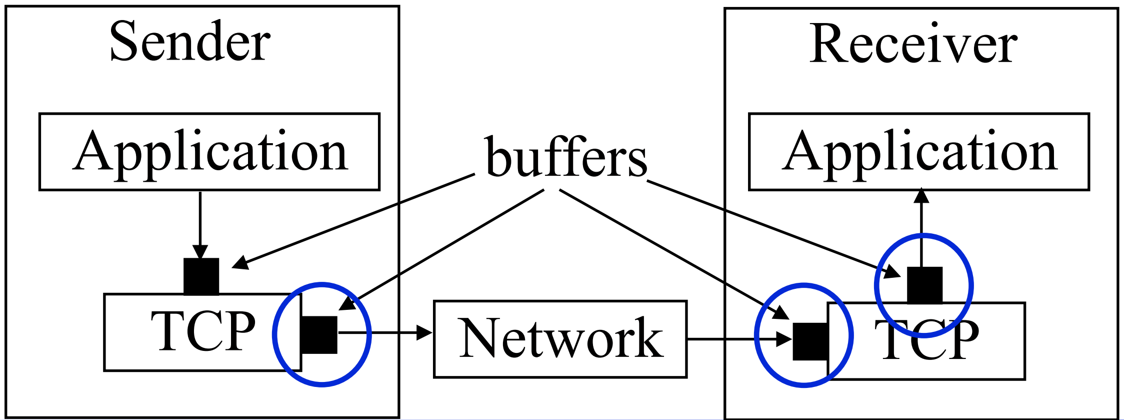
- TCP general overview
- TCP-header
- Connection management
- Error control
- Flow control
- Congestion control

TCP properties

- End-to-end
- Connection oriented
  - State maintained at both ends
  - Identified by a four-tuple
    - Formed by the two end point's IP address and TCP port number
- Reliable
  - Try to guarantee in order delivery of each packet
  - Buffered transfer
- Full Duplex
  - Data transfer simultaneously in both directions

TCP properties

- Three main functionalities for active connection
  1. Error control
     - Deal with the best effort unreliable network
  2. Flow control
     - Do not overload the receiving application
  3. Congestion control
     - Do not overload the network itself



TCP-header (RFC 793)

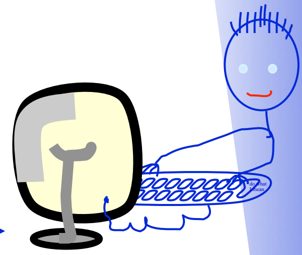
```
 0                   10                  20                        31
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|    Source port                |       Destination port        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                         Sequence number                       |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                         Acknowledgment number                 |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
| hdr   |           |U|A|P|R|S|F|                               |
| length| Varattu   |R|C|S|S|Y|I| Advertized receiver window    |
|       |           |G|K|H|T|N|N|                               |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|     Checksum                  |        Urgent-pointer       |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                    Options                    |    Padding    |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                              data                             |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

TCP options

- 3 = window scaling
- 8,10 = Timestamp and echo of previous timestamp
  - Improve accuracy of RTT computation
  - Protect against wrapped sequence numbers
- 2 = Maximum Segment Size (MSS)
  - Negotiated while establishing connection
  - Try to avoid fragmentation
- 1 = No-operation
  - Sometimes between options, align option fields
- 0 = End of options

Connection establishment

- Three-way handshake



<SEQ=100><SYN>

<SEQ=300><ACK=101><SYN><ACK>

<SEQ=101><ACK=301><ACK>

Third packet may contain data:

<SEQ=101><ACK=301><ACK><DATA>

Terminating connection

- Modified three-way handshake
- If other end has no more data to send, can be terminated one way:
  - Send a packet with FIN flag set
  - Recipient acks the FIN packet
- After done with the data transfer to the other direction
  - FIN packet and ack to the inverse direction

Outline

- TCP general overview
- TCP-header
- Connection management
- Error control
- Flow control
- Congestion control

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Error control

- Mechanisms to detect and recover from lost packets
- Sequence numbers
  - Used in acknowledgments
  - Identify the packets that are acknowledged
- Positive acknowledgments (ARQ)
- Error detection and correction
  - Timers
  - Checksums
- Retransmissions

Cumulative Acknowledgments

- Acknowledge only the next expected packet in sequence
  - E.g. received 1,2,3,4,6 -> ACK 5
- Advantages
  - Single ACK for multiple packets
    - Delayed ACKs scheme = one ACK for 2*MSS data
  - Lost ACK does not necessarily trigger retransmission
- Drawback
  - Cannot tell if lost only first or all of a train of packets
  - => Selective ACK

Selective Acknowledgments (SACK)

- RFC 2018
- Helps recovery when multiple packets are lost
- Receiver reports which segments were lost using TCP SACK (Selective Acknowledgment) options
- Sender can retransmit several packets per RTT

Retransmission timeout (RTO)

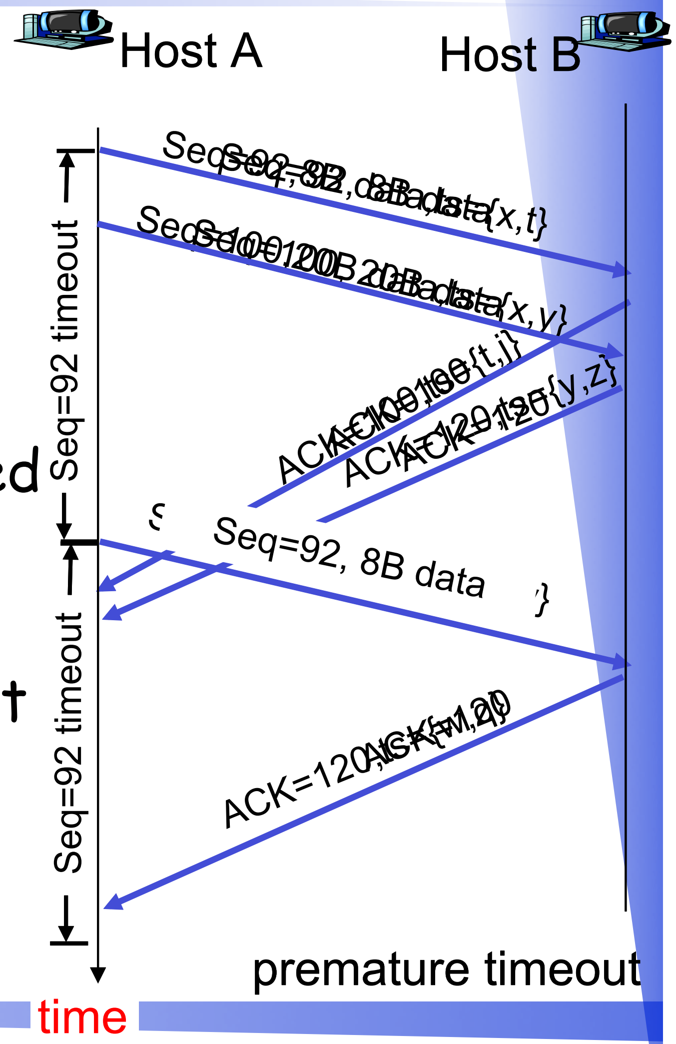
- RTO associated to each transmitted packet
- Retransmit packet if no ACK is received before RTO has elapsed
- Adjusting RTO (original algorithm):
  - RTT = (α*oldRTT)+((1-α)*newRTTsample) (recommeded α=0,9)
  - RTO: β*RTT, β>1 (recommended β=2)
- Problem?
  - Does not take into account large variation in RTT

Modified algorithm

- Initialize: RTO = 3
- Two variables: SRTT (smoothed round-trip time) and RTTVAR (round-trip time variation)
  - First measurement R:
    - SRTT = R
    - RTTVAR = R/2
  - For subsequent measurement R:
    - RTTVAR = (1 - beta) * RTTVAR + beta * |SRTT - R|
    - SRTT = (1 - alpha) * SRTT + alpha * R
    - Use alpha=1/8, beta=1/4
- RTO = SRTT + 4*RTTVAR
- If computed RTO < 1s –> round it up to 1s

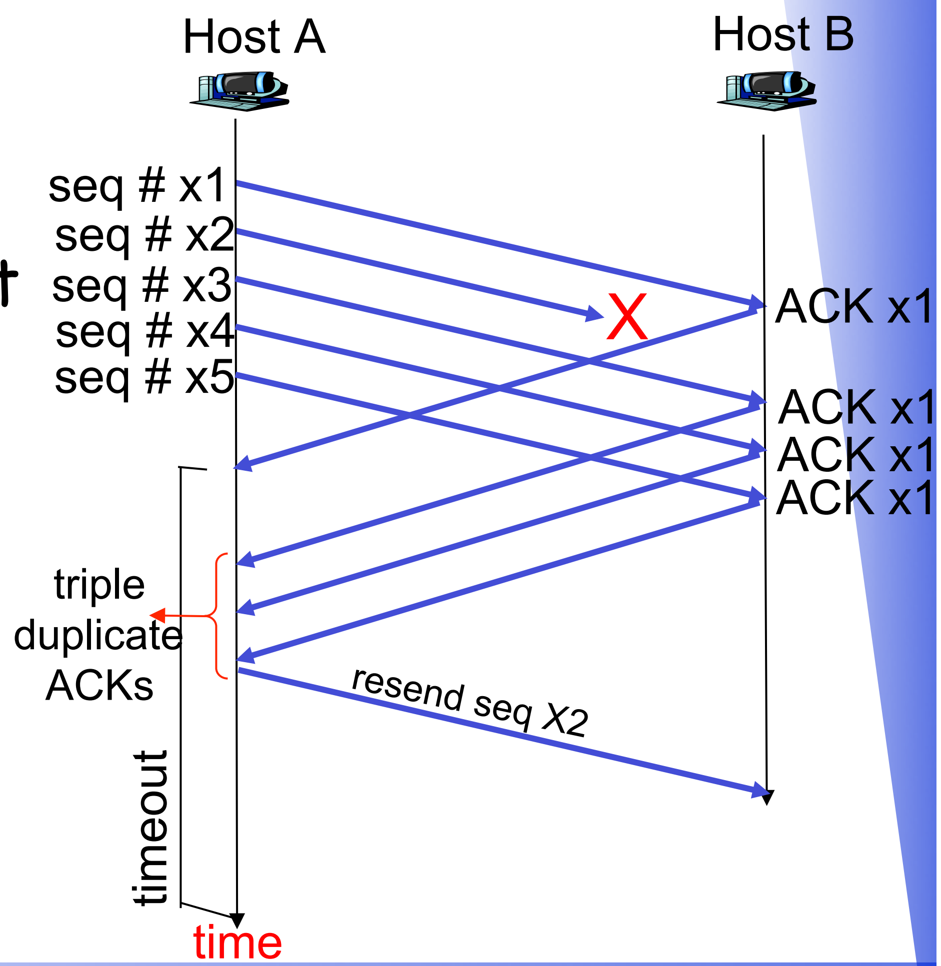
Karn's algorithm

- Receiving ACK for retransmitted packet
  - Is the ACK for original packet or retransmission??
  - No way to know...
  - ➔ Do not update RTO for retransmitted packets
- Timer backoff also needed
  - At timeout: new_timeout = 2*timeout (exponential backoff)
- TCP timestamps can also help disambiguate ACKs



Fast Retransmit

- Introduced by Van Jacobson 1988
- TCP ACKs the next expected missing packet
- Duplicate ACKs indicate lost packet(s)
- Do not wait for timeout but retransmit after 3 duplicate ACKs
  - Wait for reordered packets, don't do go-back-n



Outline

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- Flow control
- Congestion control

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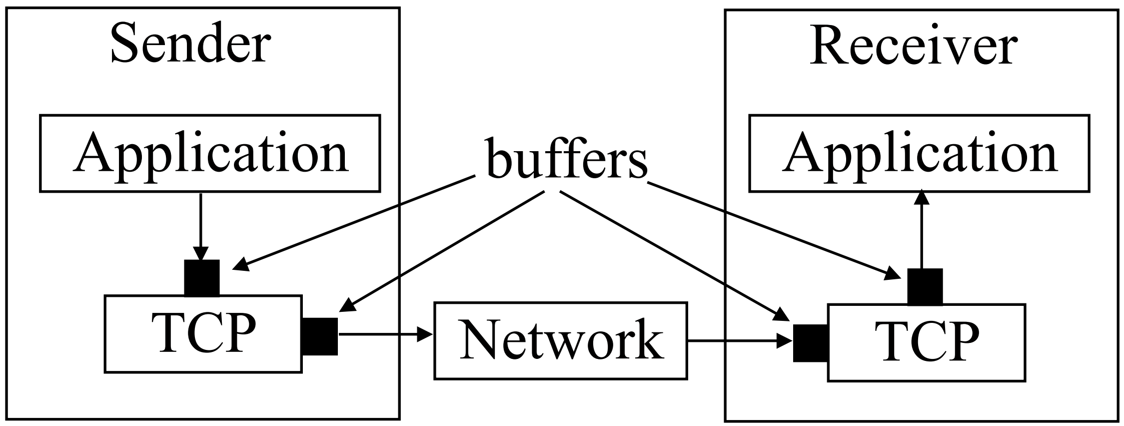
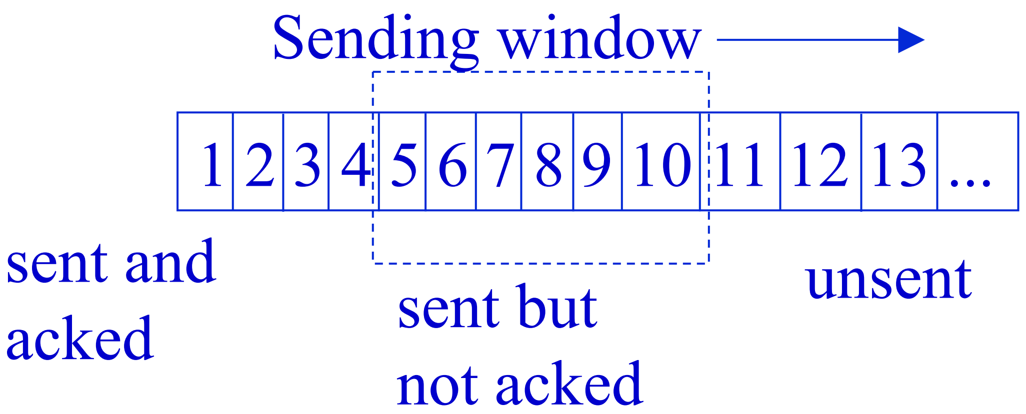
Flow control

- Goal: do not overflow the receiving application
- Window based mechanism to limit transmission rate
- Receiver advertised window



Sliding Window



- Multiple packets simultaneously "in flight", i.e. outstanding
  - Improve efficiency
- Buffer sent unacked packets

Receiver advertised window

- Receiver advertises the maximum window size the sender is allowed to use
- Enables receiver TCP to signal sending TCP to backoff
  - Receiving application not consuming received data fast enough
- Value is included in each ACK
  - Can change dynamically

Silly Window Syndrome

- Problem in worst case:
  - Receiver buffer between TCP and application fills up
  - Receiving application read a single byte -> TCP advertises a receiver window of size one
  - Sender transmits a single byte
- Lot of overhead due to packet headers

Avoiding Silly Window Syndrome

- Window update only with significant size
  - At least MSS worth of data or
  - Half of its buffer
- Analogy at sender side
  - Application gives small chunks of data to TCP -> send small packets
  - Nagle's algorithm: Delay sending data until have MSS worth of it
    - Does not work for all applications, e.g. delay sensitive apps
    - Need also mechanism to tell TCP to transmit immediately -> Push flag

Large Receiver Windows

- Receiver window hdr field size is 16 bits
  - => max size is about 65KBytes
- Example: 10Mbit/s path from Europe to US west coast
  - 0.15s * 10^7/8 ≈ 190KBytes
  - 16 bits not enough!
- Use Window Scaling option
  - Both ends set a factor during handshake (SYN segments)
  - Multiply window field value with this factor

bandwidth

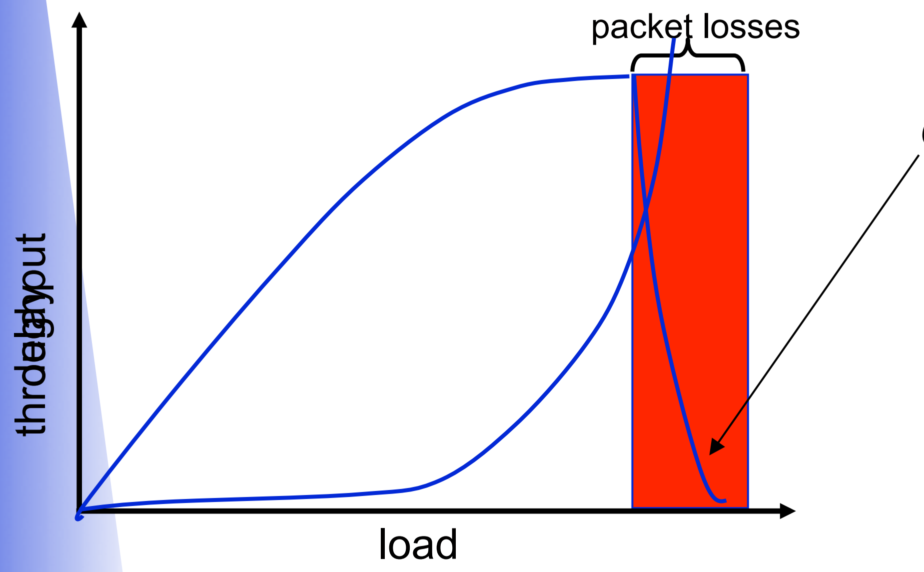
delay=RTT

Outline

- TCP general overview
- TCP-header
- Connection management
- Error control
- Flow control
- Congestion control
  - Background and motivation
  - Basic TCP congestion control
  - Fairness
  - Other TCP versions and recent developments
- Conclusions

Why we need congestion control

- Flow control in TCP prevents overwhelming the receiving application
- Problem: Multiple TCP senders sharing a link can still overwhelm it

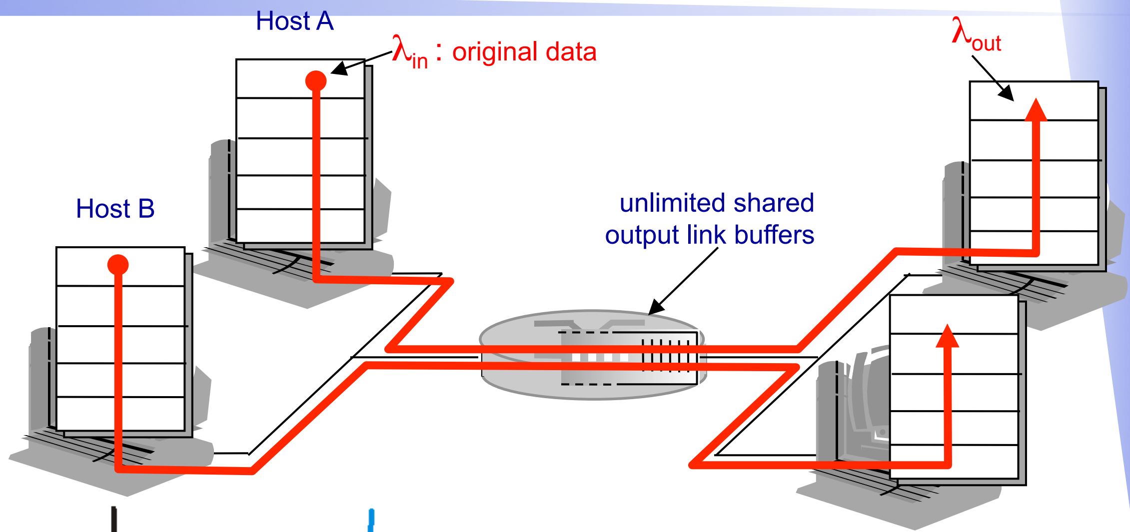


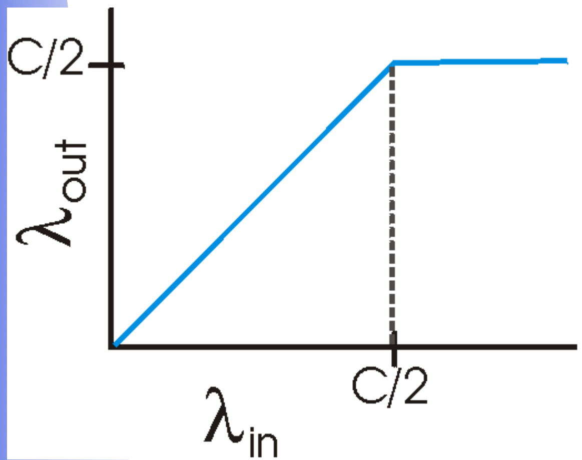
Congestion collapse due to:

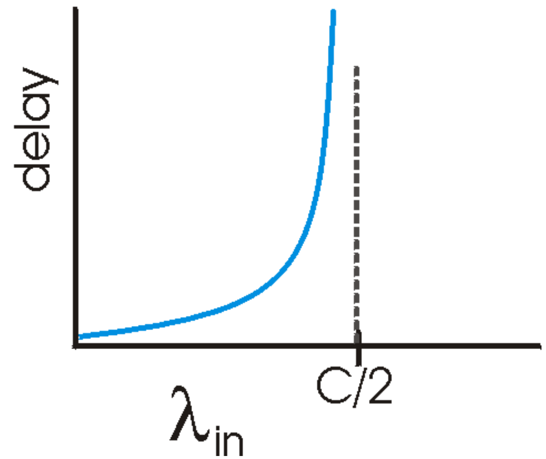
- Retransmitting lost packets
  - Further increases the load
- Spurious retransmissions of packets still in flight
  - Unnecessary retransmissions lead to even more load!
  - Like pouring gasoline on a fire

Causes/costs of congestion: scenario 1

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



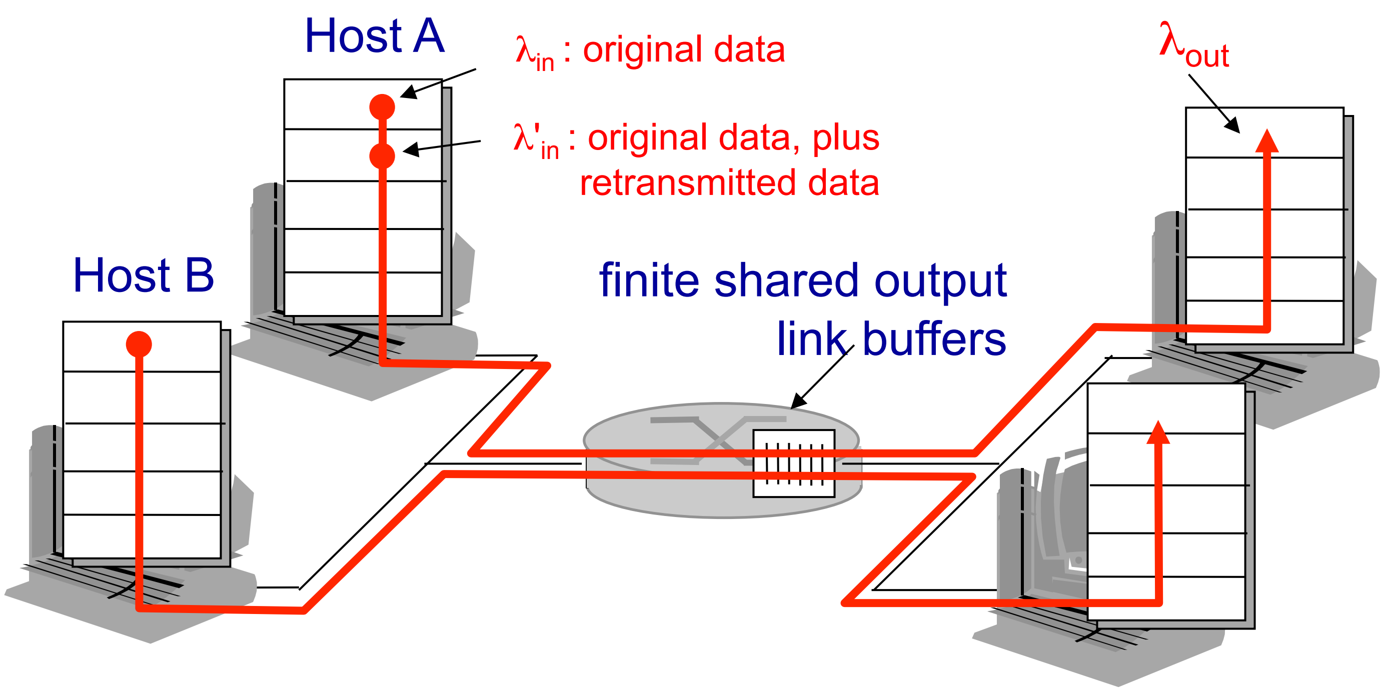




- large delays when congested
- maximum achievable throughput

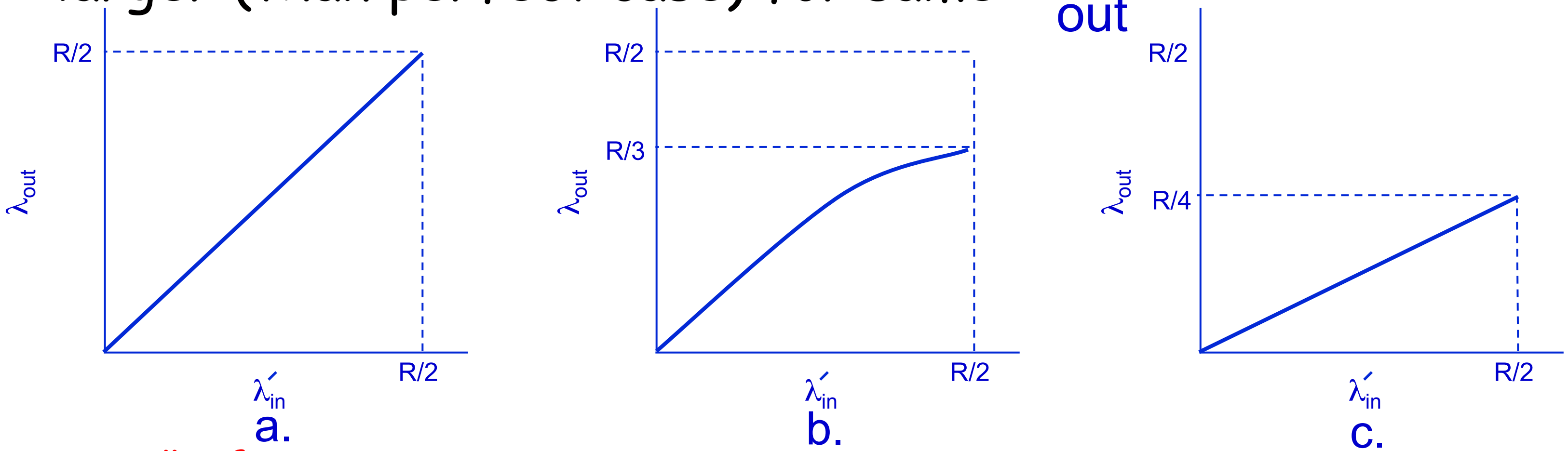
Causes/costs of congestion: scenario 2

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



Causes/costs of congestion: scenario 2

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



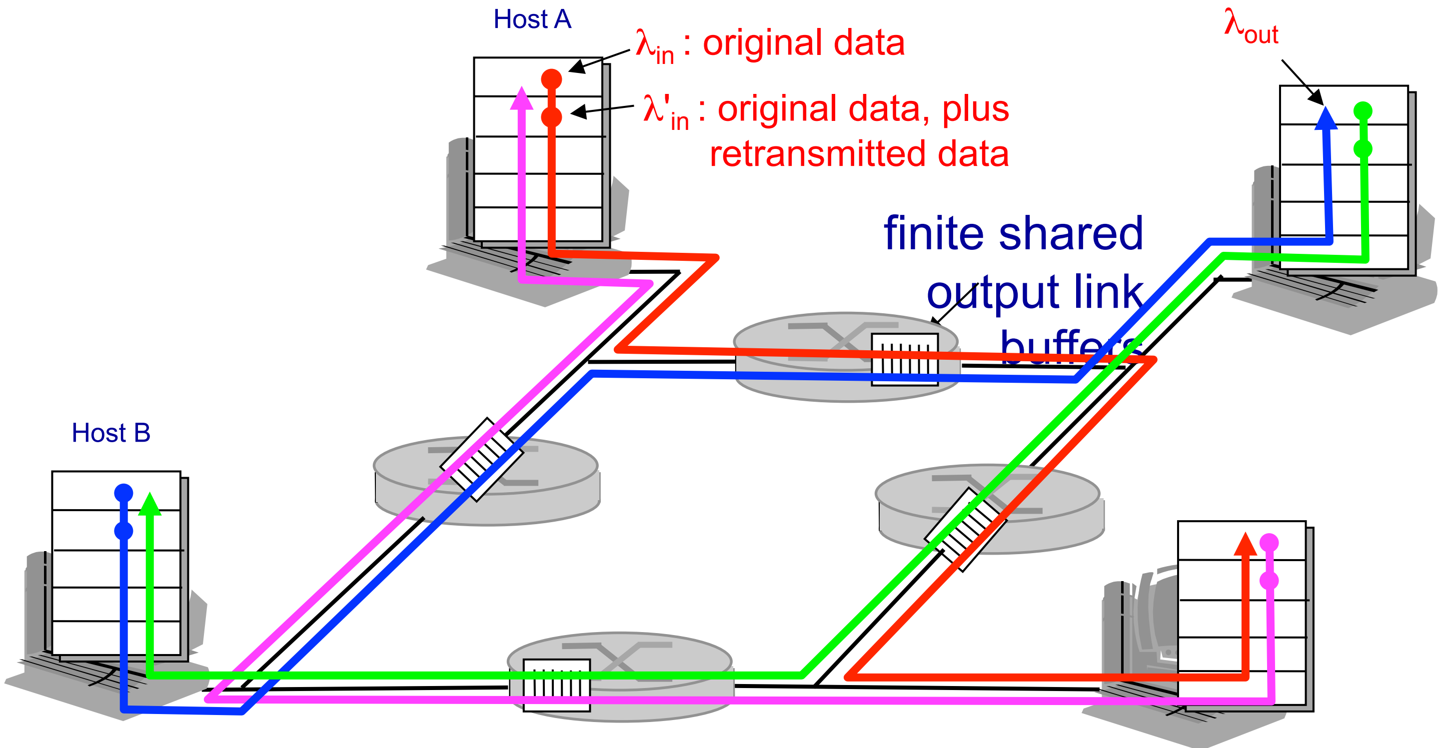
"costs" of congestion:

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

Causes/costs of congestion: scenario 3

- four senders
- multihop paths
- timeout/retransmit

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



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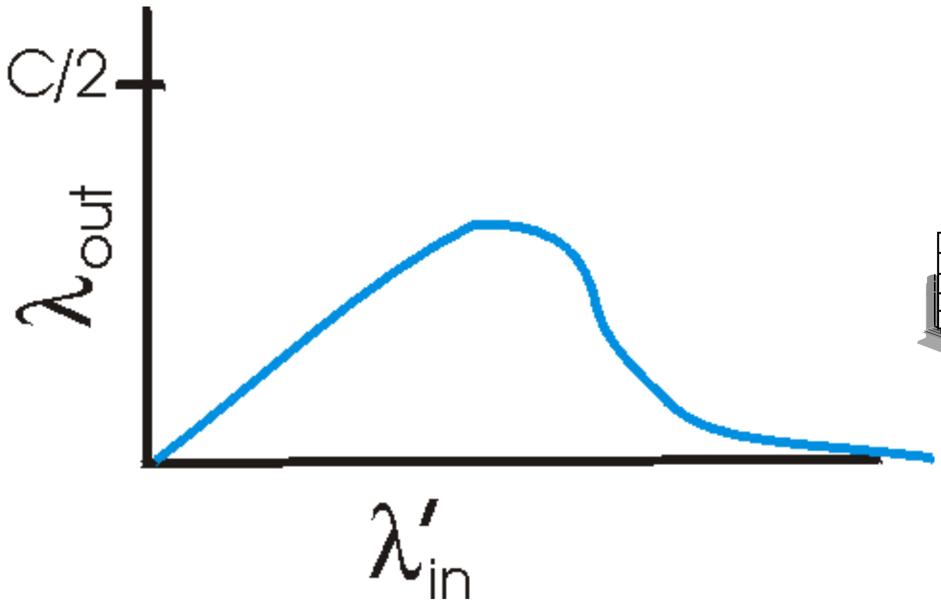
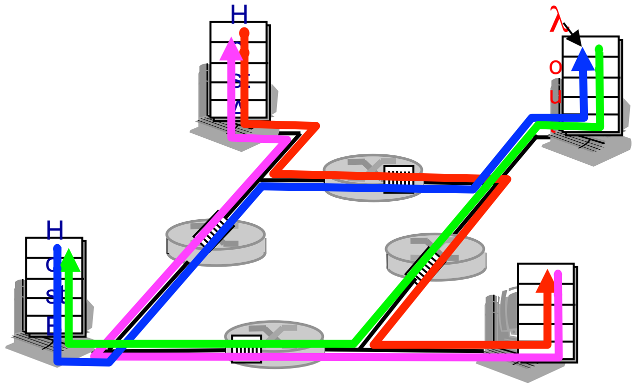
Causes/costs of congestion: scenario 3





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

Explicit Congestion Notification (ECN)

- Routers flag packets upon congestion
  - Active queue management
- Sender consequently adjusts sending rate
- Supported by routers but not widely used
  - Fear of software bugs
  - Running with default configurations
- Most OSs (Win7, Ubuntu, Fedora) ship with ECN disabled
  - Tuning for bugs (e.g. popular Cisco PIX firewall)

TCP Congestion control

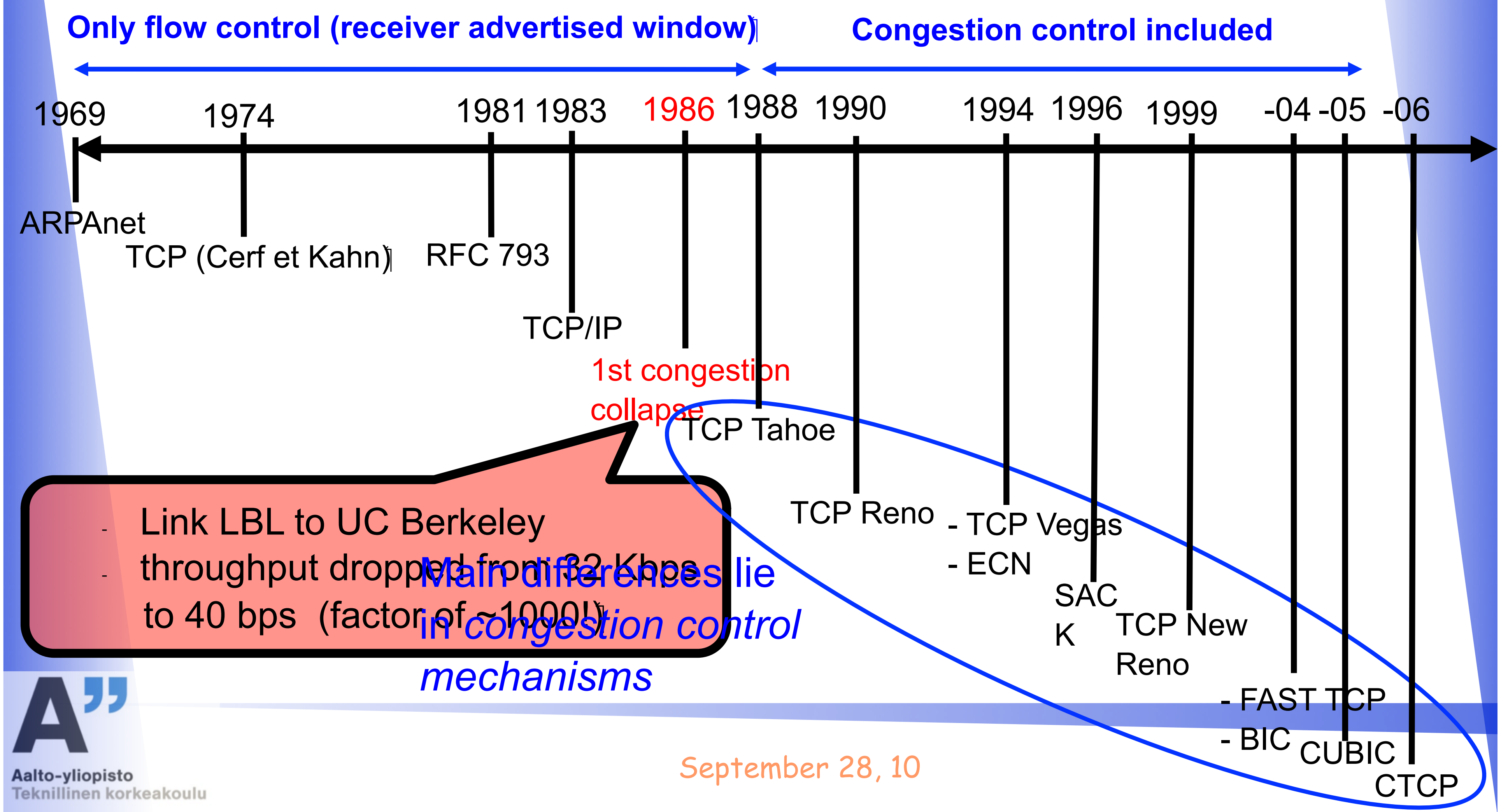
- Principle:
  - Continuously throttle TCP sender's transmission rate
  - Probe the network by increasing the rate when all is fine
  - Decrease rate when signs of congestion (e.g. packet loss)

- How?
  - Introduce *congestion window* (cwnd):

    flow control

    #outstanding bytes = min(cwnd, rwnd)
  - Adjust cwnd size to control the transmission rate
    - Adjustment strategy depends on TCP version

Glimpse into the past

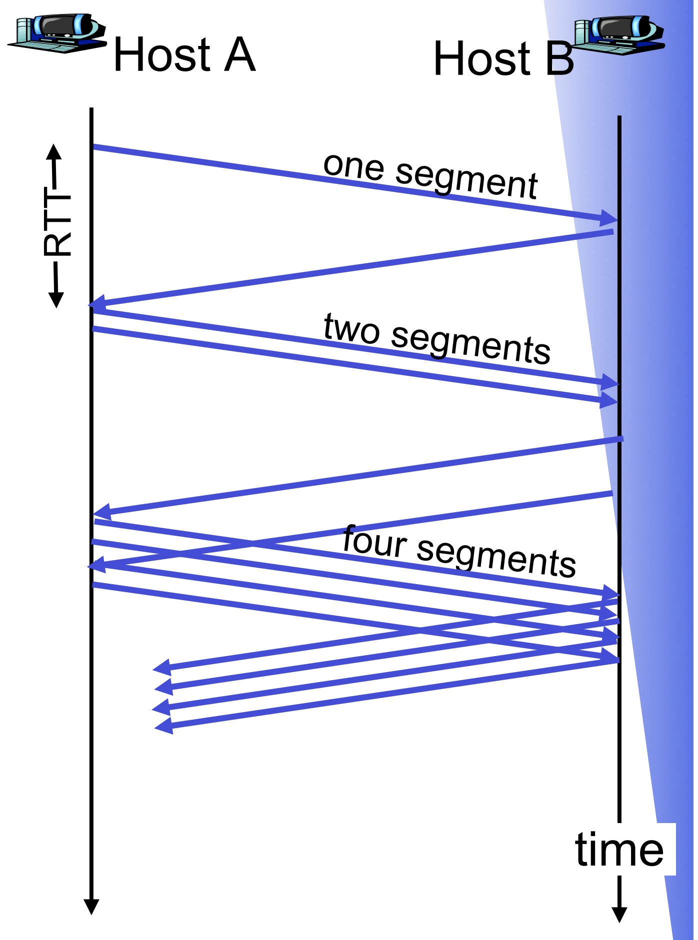


TCP Tahoe

- 1988 Van Jacobson
- The basis for TCP congestion control
- Lost packets are sign of congestion
  - Detected with *timeouts:* no ACK received in time
- Two modes:
  - Slow Start
  - Congestion Avoidance
- New retransmission timeout (RTO) calculation
  - Incorporates variance in RTT samples
  - Timeout really means a lost packet (=congestion)
- Fast Retransmit

Slow Start (SS)

- On each ACK for new data, increase cwnd by 1 packet
  - Exponential increase in the size of cwnd
  - Ramp up a new TCP connection fast (not slow!)
    - Kind of a misnomer…
- In two cases:
  - Beginning of connection
  - After a timeout



Congestion Avoidance (CA)

- Approach the rate limit of the network more conservatively
- Easy to drive the net into saturation but hard for the net to recover
- Increase cwnd by 1 for cwnd worth of ACKs (i.e. per RTT)

Combining SS and CA

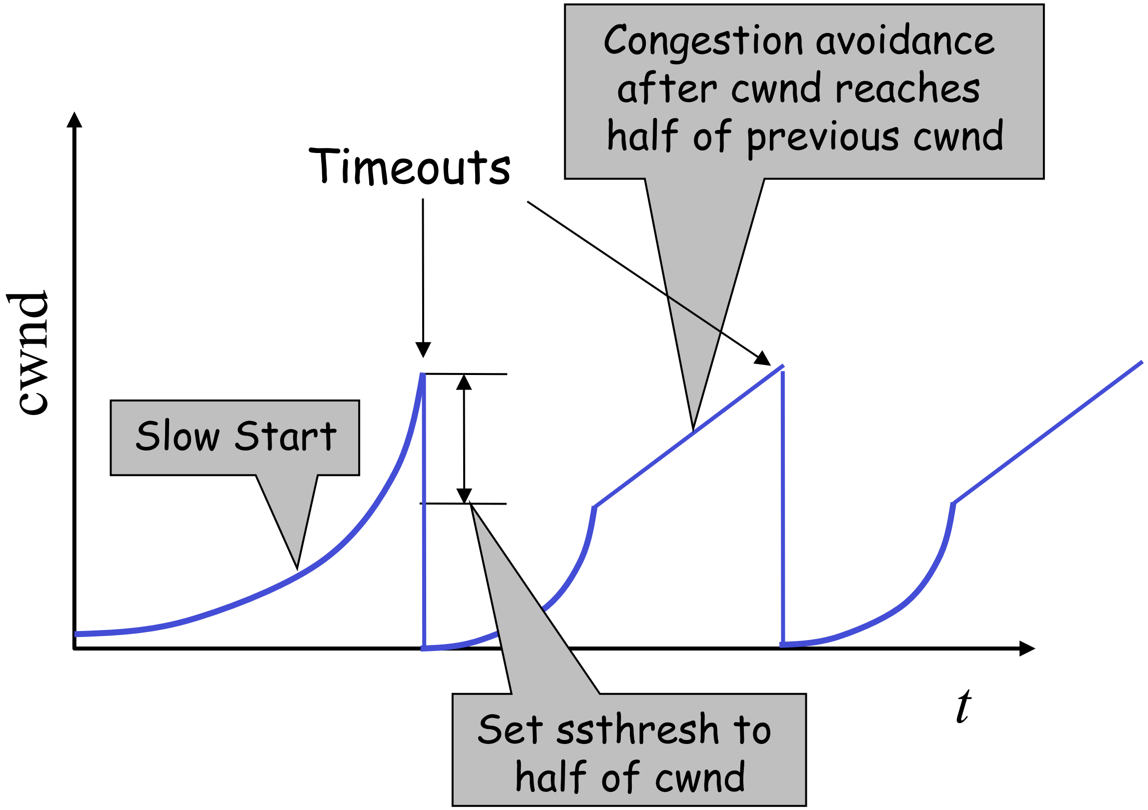
- Introduce Slow start threshold (ssthresh)
- On timeout:
  - ssthresh = 0.5 x cwnd
  - cwnd = 1 packet
- On new ACK:
  - If cwnd < ssthresh: do Slow Start
  - Else: do Congestion Avoidance

AIMD

- ACKs: increase **cwnd** by 1 MSS per RTT: additive increase
- loss: cut **cwnd** in half (non-timeout-detected loss): multiplicative decrease

AIMD: Additive Increase Multiplicative Decrease

TCP Tahoe: adjusting cwnd



TCP Reno

- Van Jacobson 1990
- Fast retransmit with Fast recovery
  - Duplicate ACKs tell sender that packets still go through
  - Do less aggressive back-off:
    - ssthresh = 0.5 x cwnd
    - cwnd = ssthresh + 3 packets
    - Increment cwnd by one for each additional duplicate ACK
    - When the next new ACK arrives: cwnd = ssthresh

Fast recovery

Nb of packets that were delivered

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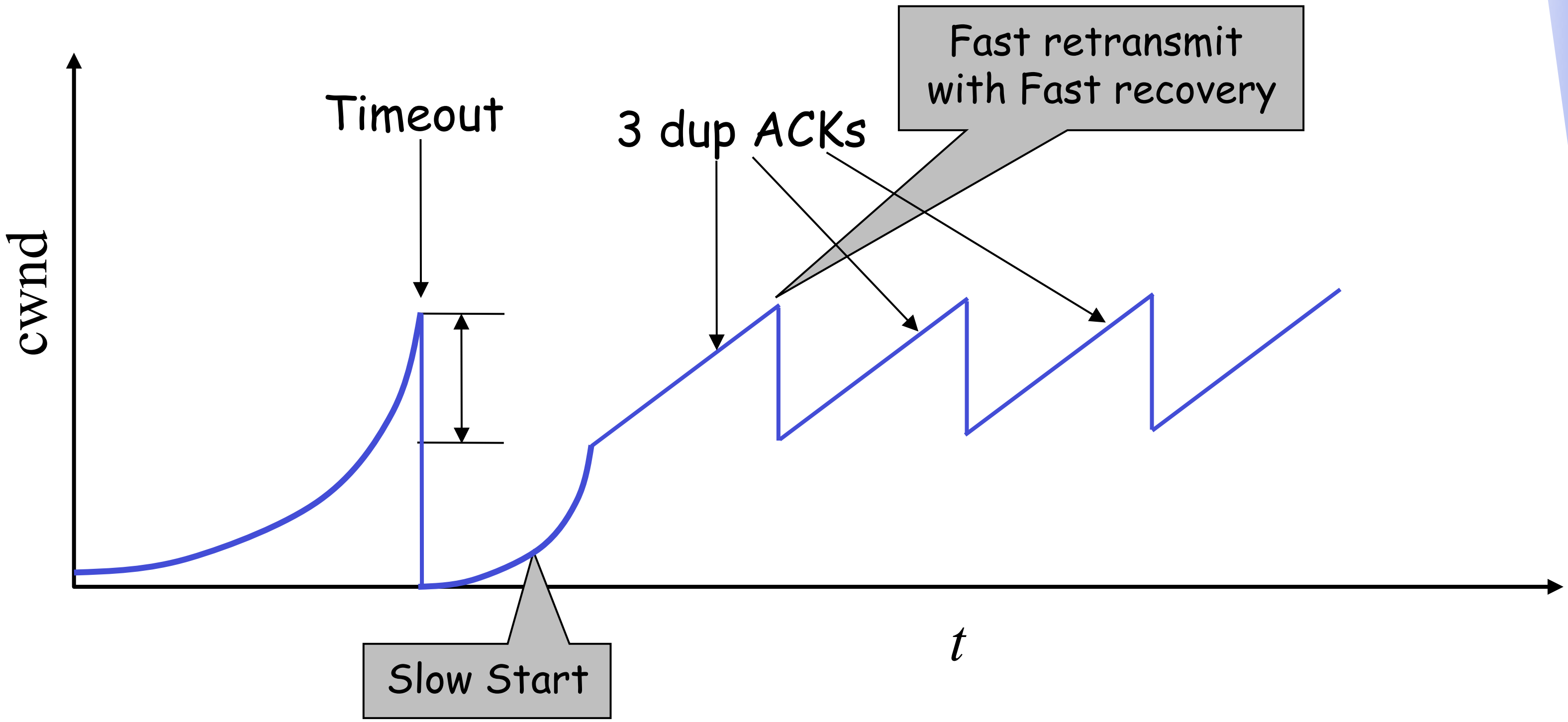
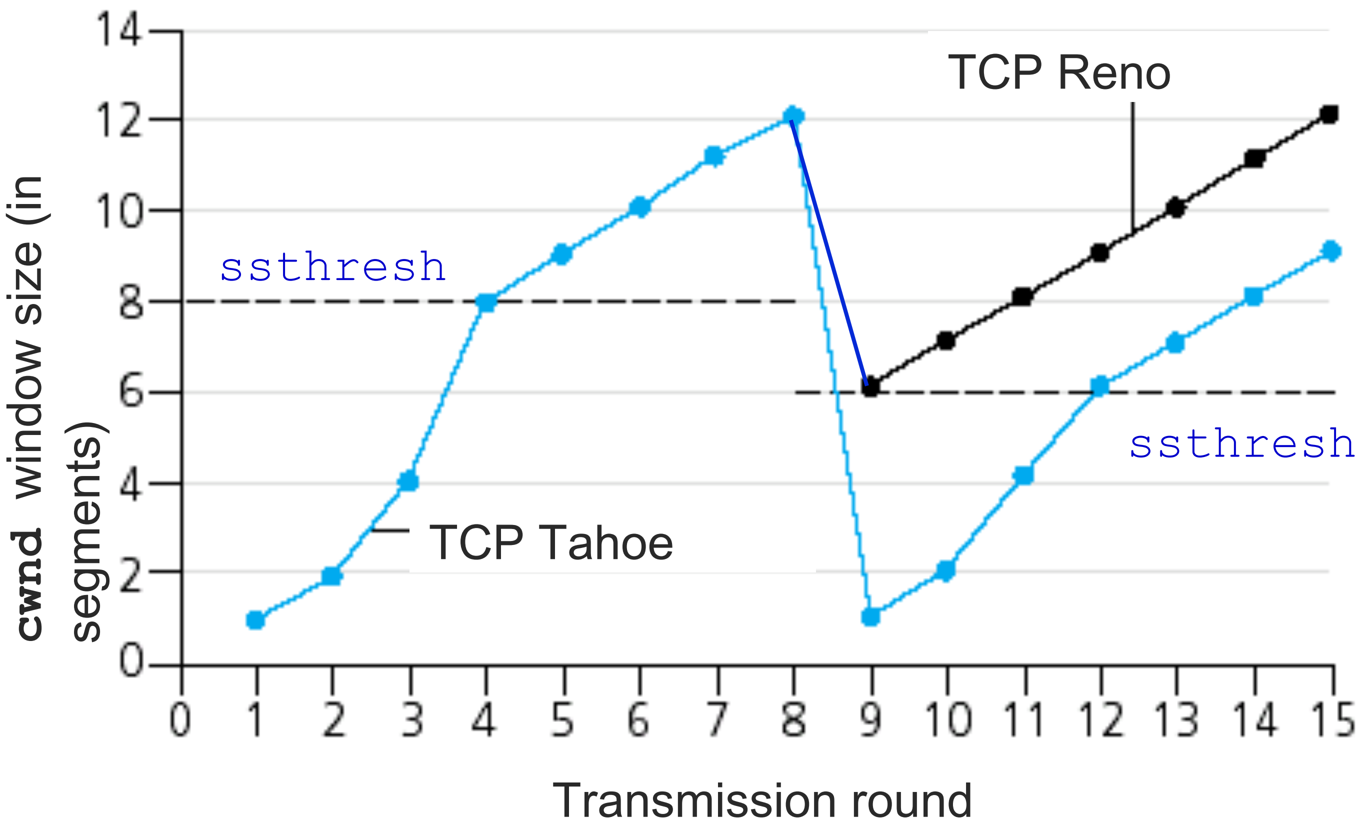
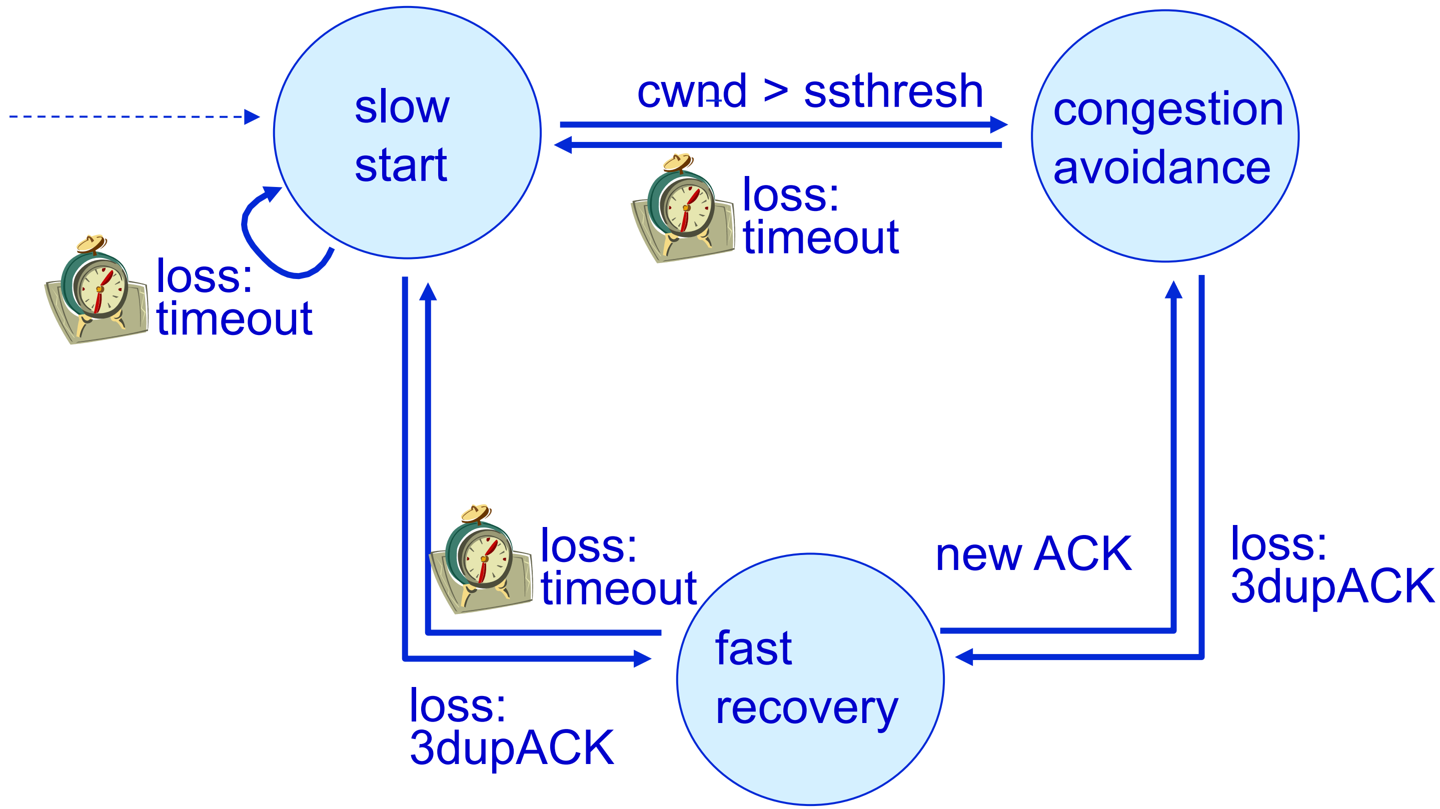
TCP Reno: adjusting cwnd



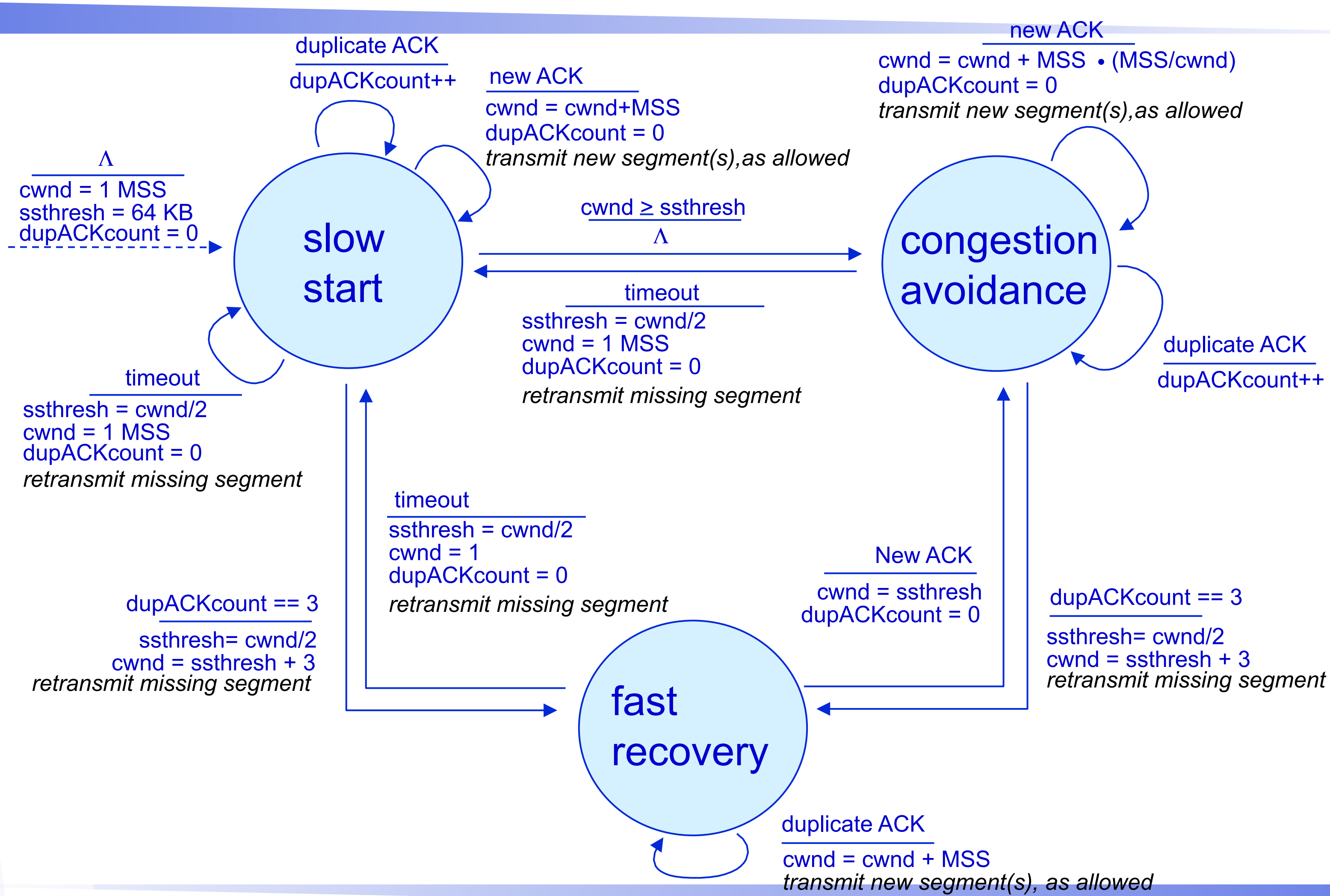
Tahoe vs. Reno



Congestion control FSM



Congestion control FSM: details



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TCP New Reno

- 1999 by Sally Floyd
- Modification to Reno's Fast Recovery phase
- Problem with Reno:
  - Multiple packets lost in a window
  - Sender out of Fast Recovery after retransmission of only one packet

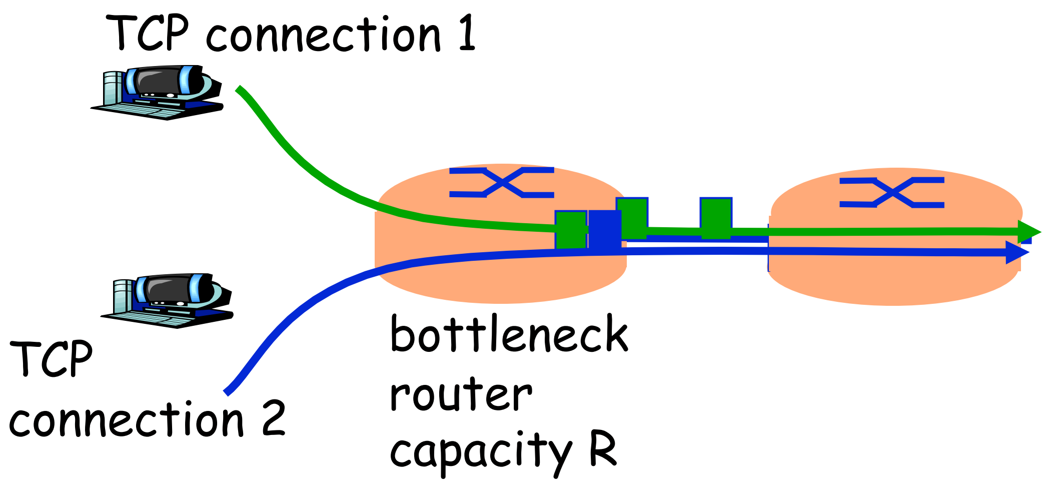
    ➔ cwnd closed up

    ➔ no room in cwnd to generate duplicate ACKs for additional Fast Retransmits

    ➔ eventual timeout
- New Reno continues Fast Recovery until all lost packets from that window are recovered

TCP Fairness

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

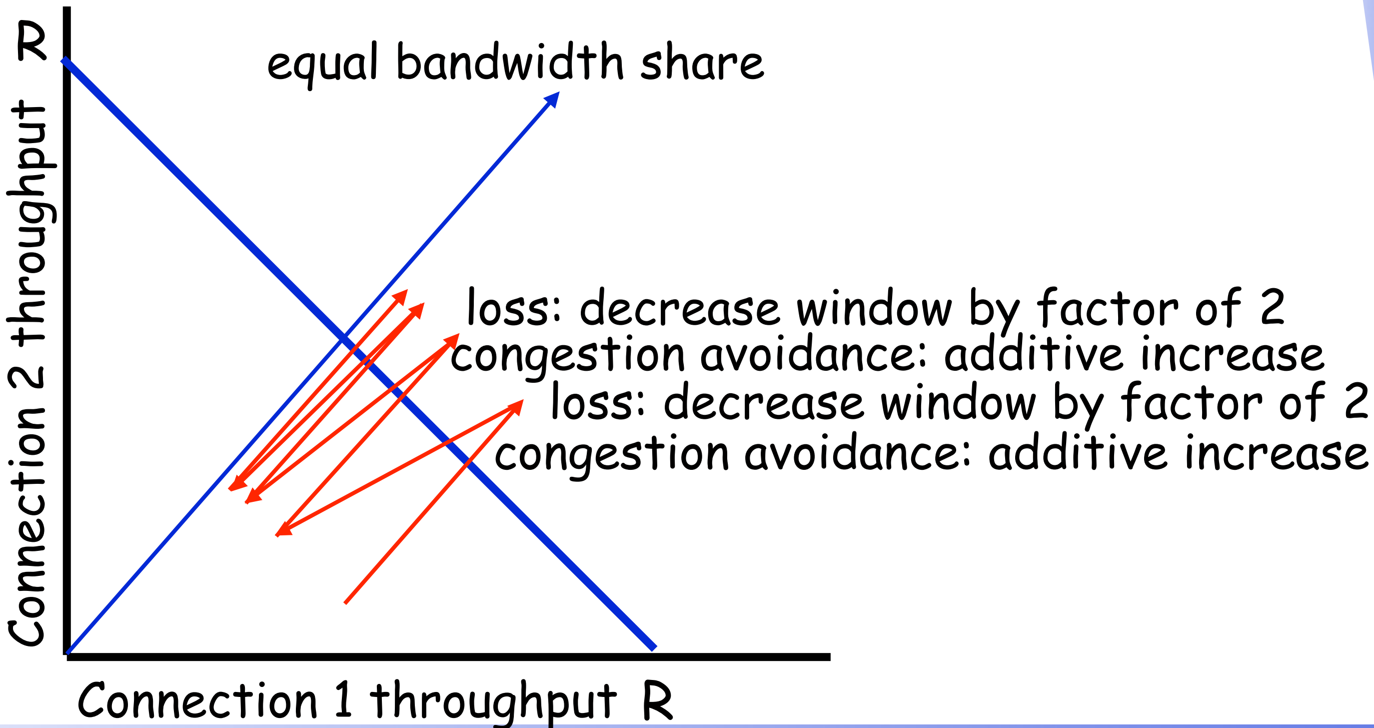


Is TCP fair?

Why is TCP fair?

Two competing sessions:

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



TCP Fairness Issues (cont.)

RTT Fairness

- What if two connections have different RTTs?
  - "Faster" connection grabs larger share
- Reno's (AIMD) fairness is RTT biased

Fairness and parallel TCP connections

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

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

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Other TCP versions

- Delay-based congestion control
  - TCP Vegas
- Wireless networks
  - Take into account random packet loss due to bit errors (not congestion!)
  - E.g. TCP Veno
- Paths with high *bandwidth*delay*
  - These *"long fat pipes"* require large cwnd to be saturated
  - SS and CA provide too slow response
  - TCP CUBIC
  - Compound TCP (CTCP)

TCP Vegas

- 1994 by Brakmo et Peterson
- Issue: Tahoe and Reno RTO clock is very coarse grained
  - "ticks" each 500ms
- Increasing delay is a sign of congestion
  - Packets start to fill up queues
  - Expected throughput = cwnd / BaseRTT
  - Compare expected to actual throughput
  - Adjust rate accordingly before packets are lost
- Also some modifications to Slow start and Fast Retransmit
- Potentially up to 70% better throughput than Reno
- Fairness with Reno?
  - Reno grabs larger share due to late congestion detection

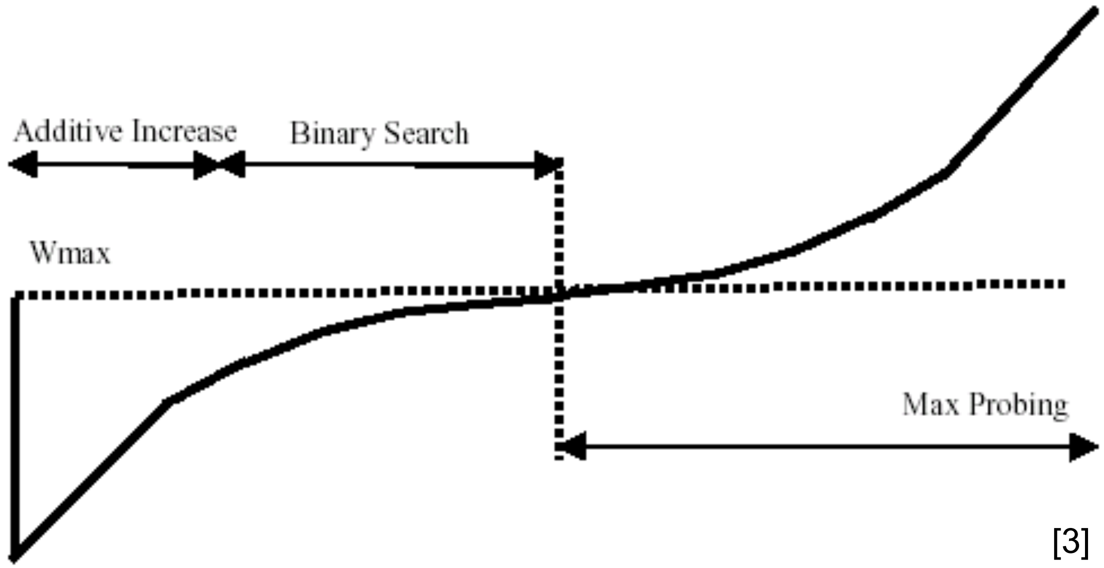
minimum of all measured round trip times

BIC and CUBIC

- 2004, 2005 by Xu and Rhee
- Both for paths with high (*bandwidth* x *delay*)
  - These "*long fat pipes*" lead to large cwnd
  - SS and CA provide too slow response
  - Scale up to tens of Gb/s
- BIC TCP
  - No AIMD
  - Window growth function is combination of *binary search* and *linear increase*
  - Aim for TCP friendliness and RTT fairness

BIC and CUBIC

- BIC window growth function



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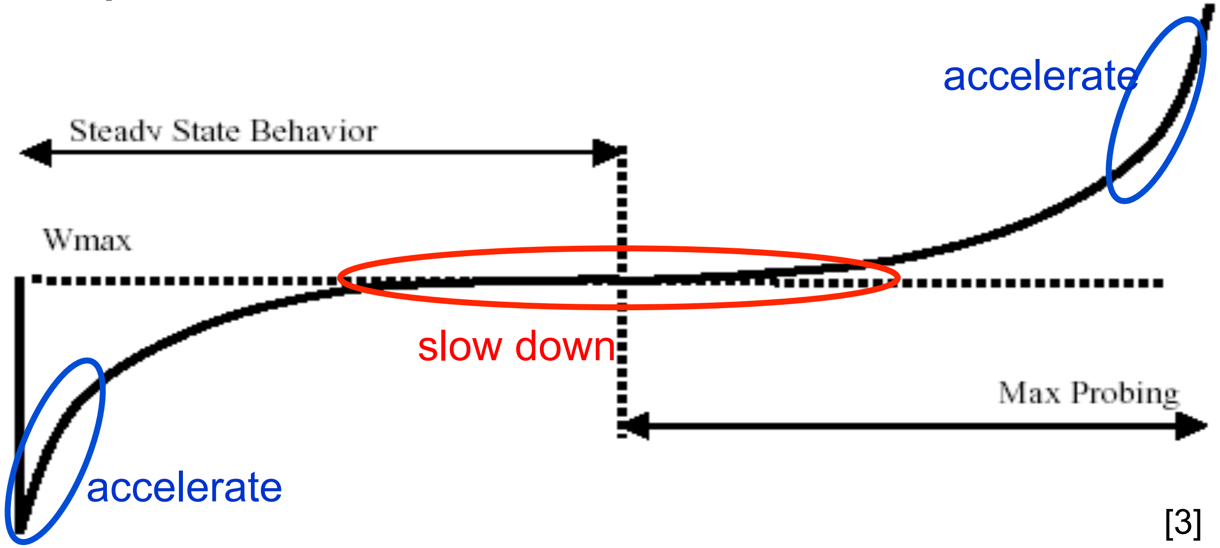
BIC and CUBIC (cont.)

- CUBIC TCP
  - Enhanced version of BIC
  - Simplifies BIC window control using a *cubic function*
  - Improves its TCP friendliness & RTT fairness



$$W_{cubic} = C(t-K)^3 + W_{max} \qquad K = \sqrt[3]{W_{max}\beta / C}$$

Compound TCP (CTCP)

- From Microsoft research, 2006
- Tackles same problems as BIC and CUBIC
  - High speed and long distance networks
  - RTT fairness, TCP friendliness
- Loss-based vs. delay-based approaches
  - Loss-based (e.g. HSTCP, BIC...) too aggressive
  - Delay-based (e.g. Vegas) too timid
- Compound approach
  - Use delay metric to sense the network congestion
  - Adaptively adjust aggressiveness based on network congestion level
  - Loss-based component: *cwnd* (standard TCP Reno)
  - Scalable delay-based component: *dwnd*
  - TCP sending window is *Win = cwnd + dwnd*

Deployment

- Windows
  - Server 2008 uses Compound TCP (CTCP) by default
  - Vista, XP support CTCP, New Reno by default
- Linux
  - TCP BIC default in kernels 2.6.8 through 2.6.18
  - TCP CUBIC since 2.6.19

Conclusions

- Transport layer
  - End-to-end transport of data for applications
  - Application multiplexing through port numbers
  - Reliable (TCP) vs. unreliable (UDP)
- UDP
  - Unreliable, no state
  - Optionally integrity checking
- TCP
  - Connection management
  - Error control: deal with unreliable network path
  - Flow control: Prevent overwhelming receiving application
  - Congestion control: Prevent overwhelming the network
    - Loss-based and delay-based congestion detection
    - More and less aggressive rate control
    - Suitable for different network types
    - Fairness is important

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