Transmission Control Protocol

ITS 413 – Internet Technologies and Applications

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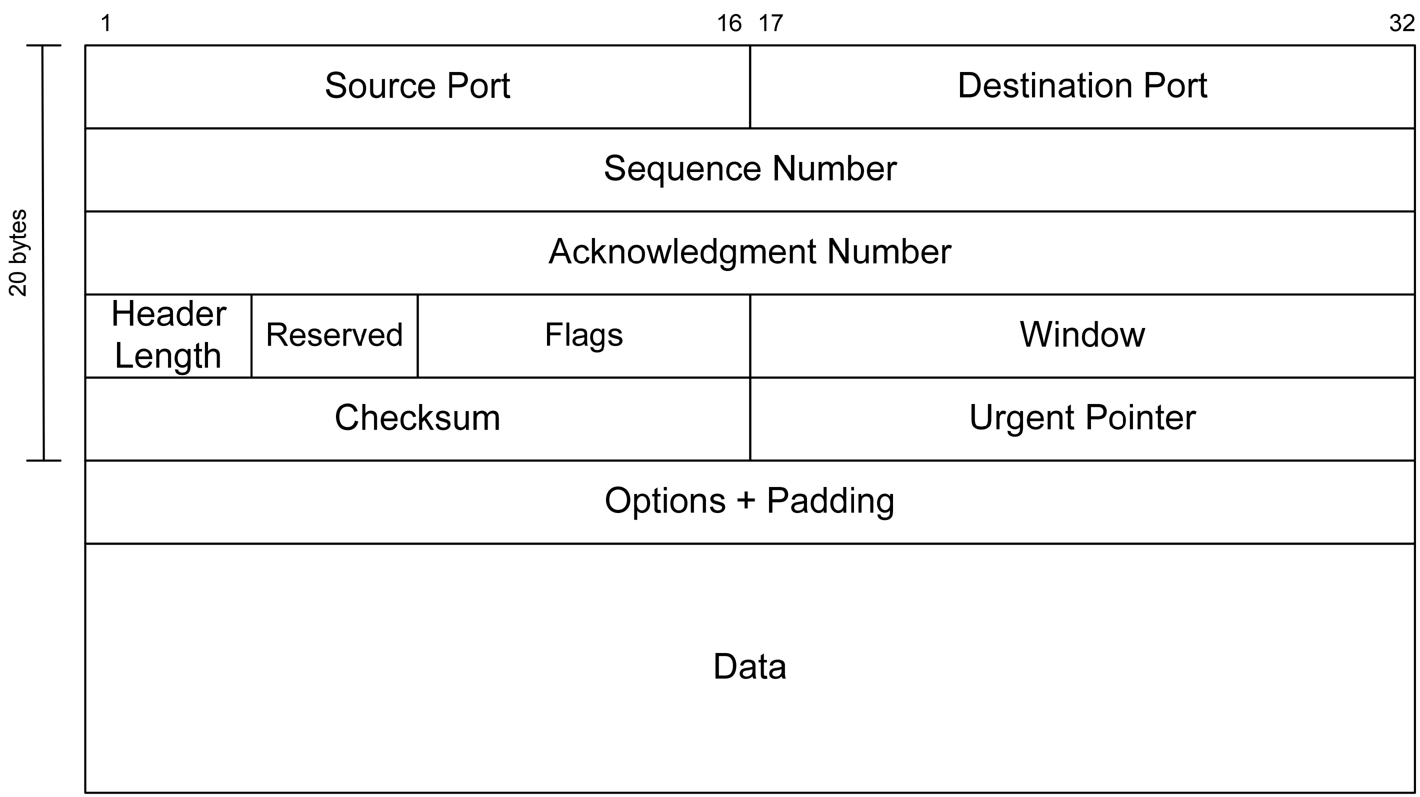
Overview of TCP

Transmission Control Protocol

- The most commonly used transport protocol today
  - Almost all Internet applications that require reliability use TCP
    - Web browsing, email, file sharing, instant messaging, file transfer, database access, proprietary business applications, some multimedia applications (at least for control purposes), …
- TCP provides a reliable, stream-oriented transport service:
  - Stream of bits (or bytes) flow between end-points
    - Stream is unstructured
  - Connection-oriented data transfer
    - Set up a connection before sending data
  - Buffered transfer
    - Applications generate any sized messages
    - TCP may buffer messages until large datagram is formed
    - Option to force (push) the transmission
  - Full duplex connection
    - Once the connection is setup, data can be sent in both directions
  - Reliability
    - Positive acknowledgement with retransmission

TCP Segment

- Header contains 20 bytes, plus optional fields
  - Optional fields must be padded out to multiple of 4 bytes



TCP Segment Fields

- **Source/Destination port**: 16 bit port number of the source/destination
- **Sequence number** of the first data byte in this segment
  - Unless the SYN flag is set, in which case the sequence number is the Initial Sequence Number (ISN)
- **Acknowledgement number**: sequence number of the next data byte TCP expects to receive
- **Header Length**: Size of header (measured in 4 bytes)
- **Reserved** for future use
- **Flags** see next slide
- **Window** contains the number of bytes the receiver is willing to accept (for flow control)
- **Checksum** for detecting errors in the TCP segment
- **Urgent pointer** points to the sequence number of the last byte of urgent data in the segment
- **Options**: such as maximum segment size, window scaling, selective acknowledgement, …

TCP Segment Flags

- Flags (1 bit each, if 1 the flag is true or on):
  - CWR: Congestion Window Reduced
  - ECE: Explicit Congestion Notification Echo
    - CWR and ECE are used on a special congestion control mechanism – we do not cover this in ITS 323
  - URG: segment carries urgent data, use the urgent pointer field; receiver should notify application program of urgent data as soon as possible
  - ACK: segment carries ACK, use the ACK field
  - PSH: push function
  - RST: reset the connection
  - SYN: synchronise the sequence numbers
  - FIN: no more data from sender

- Note
  - There is only one type of TCP packet
    - However the purpose of that packet may differ depending on the flags set
    - If SYN flag is set, we may call it a "SYN packet or TCP SYN"
    - If the ACK flag is set, we may call it a "ACK packet"
    - If the packet carries data, we may call it a "DATA packet"
    - If the packet carries data and the ACK flag is set, it is both a DATA and ACK packet

Main TCP Features

- Connection Management
  - Aim: Initialise parameters for data transfer
  - Setup a connection before sending data
  - Teardown a connection when finished
- Reliability
  - Aim: ensure all data is deliver intact, in-order to receiver
  - Sequence numbers
  - Re-transmission schemes
    - Basic (retransmit after timeout), Fast Retransmit (retransmit after receiving 3 duplicate ACKs)
- Flow Control
  - Aim: ensure the sender does not overflow the receiver
  - Receiver indicates free space in buffer in Advertised Window of ACK
  - Sender cannot send more than the Advertised Window
- Congestion Control
  - Aim: ensure the sender does not overflow the network (routers)

The Causes of Congestion

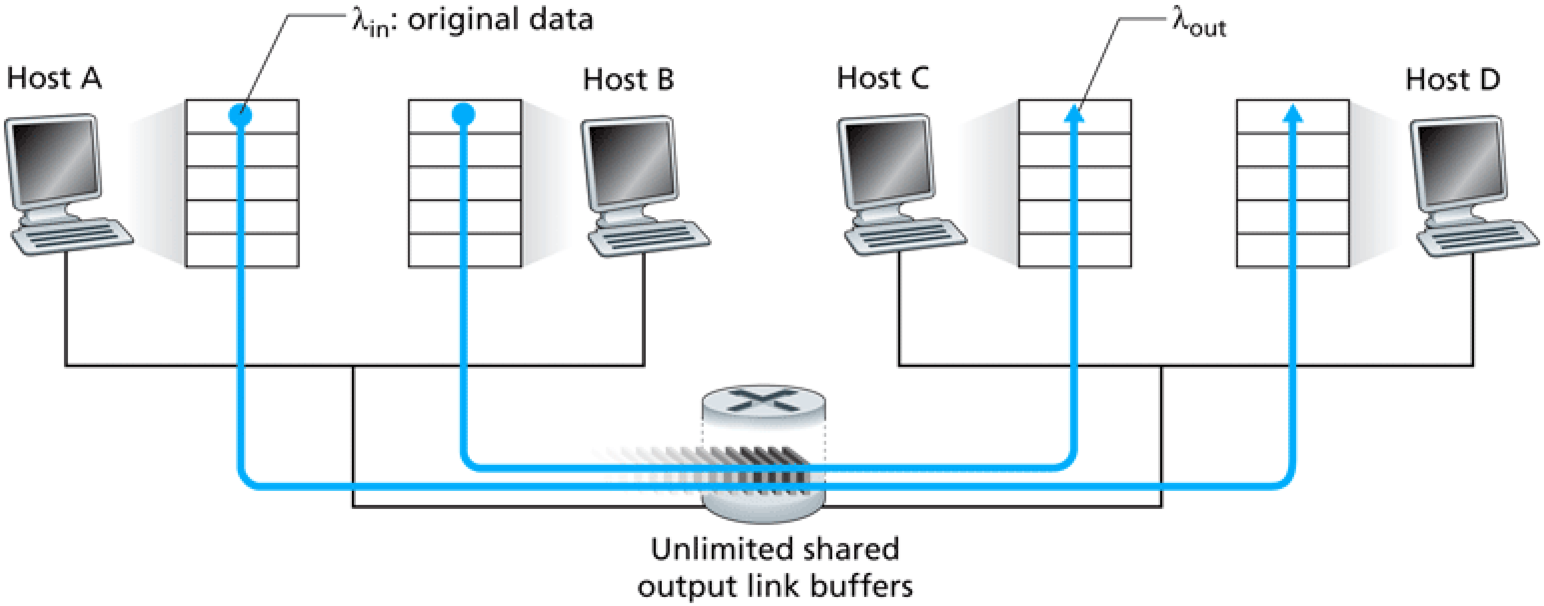
What is Congestion?

- Congestion occurs when the number of packets being transmitted through the network approaches the packet handling capacity of the network
  - What is the packet handling capacity of a network?
  - What happens when capacity is approached?
- Congestion control aims to keep number of packets below a level at which performance falls off dramatically
  - How to keep number of packets below level?

- Congestion is caused by too many sources trying to send data at too high a rate
  - In IP networks, this typically results in routers dropping packets
  - For TCP, lost packets (and larger delay) result in retransmissions
    - Retransmission cause more congestion, and more packet losses, and more retransmissions, …
- Congestion control aims to reduce the rate at which sources send

Congestion Scenario 1

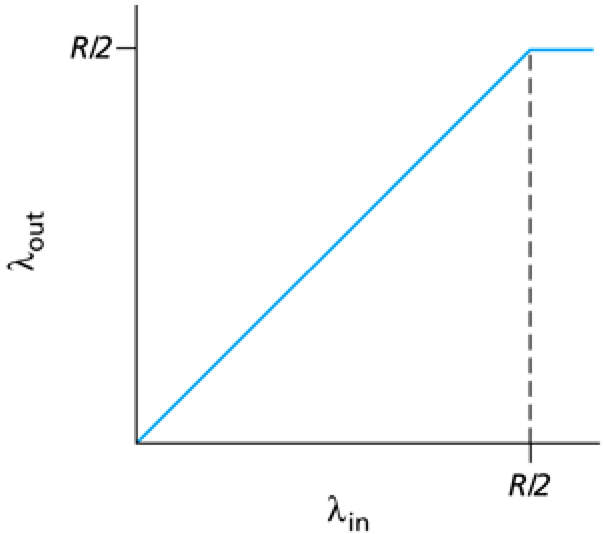
- Two senders (and receivers); a router with infinite buffers

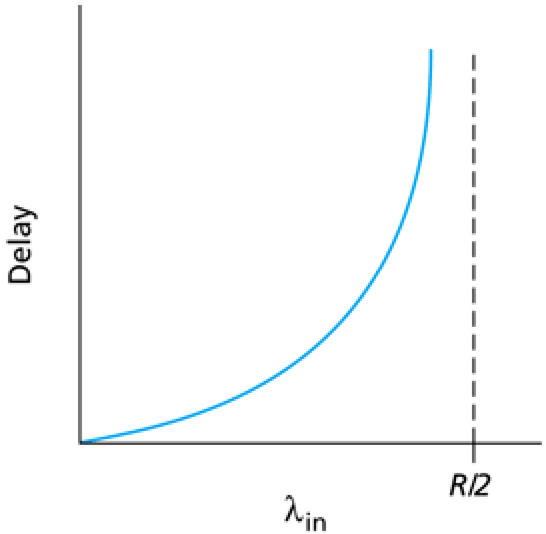


  - Router outgoing link has capacity R
  - Host A sends packets to router at rate λ_{in} bytes per sec; so does Host B
  - Router has infinite buffer space to store packets when input rate exceeds output rate
  - λ_{out} is throughput for a connection

Congestion Scenario 1

- Plot of throughput (λ_{out}) and delay for each connection

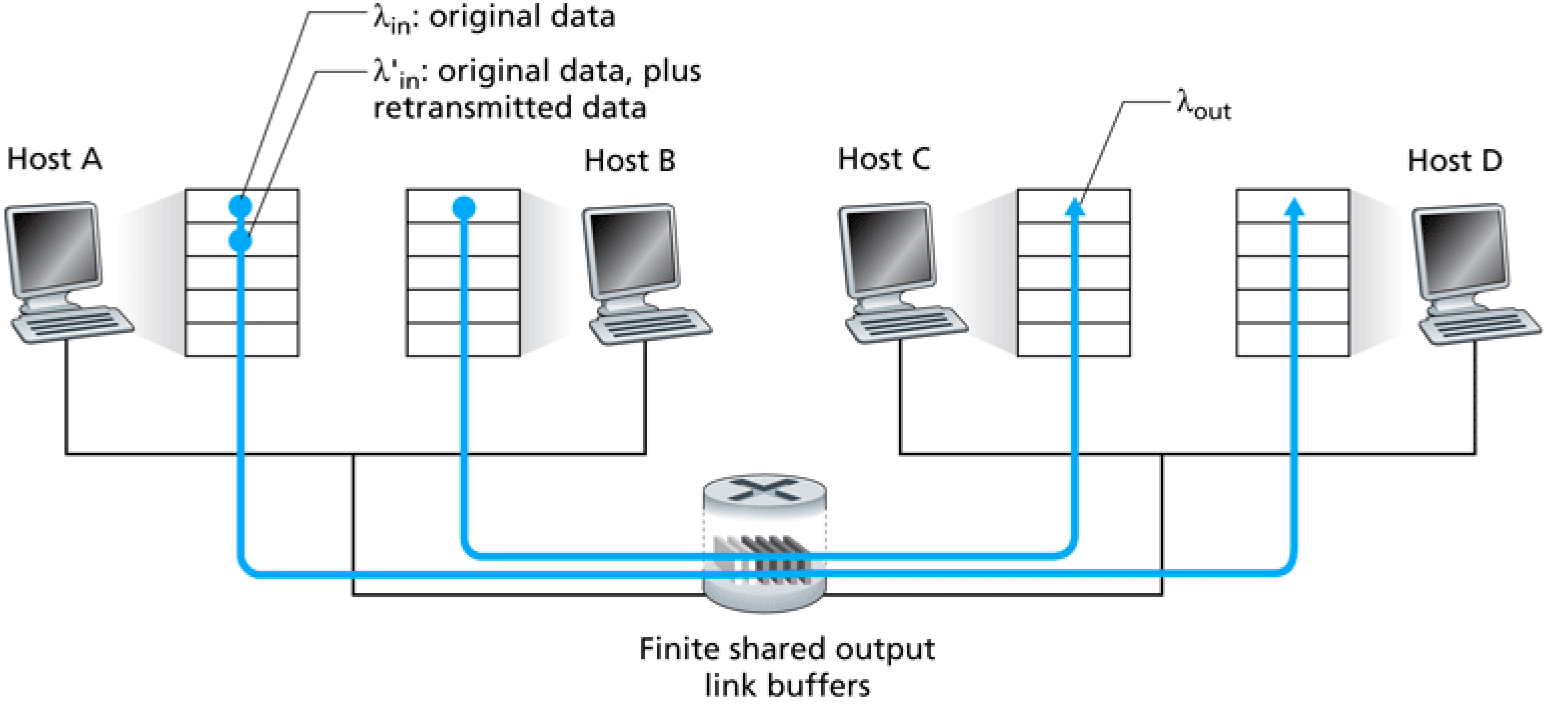




  - While sending rate is less than output capacity at router, each connection achieves full sending rate in throughput
  - When sending rate is greater than output capacity at router, each connection is limited to half of router output capacity
  - However, as sending rate approaches output capacity, delay rises significantly (packets must wait in router buffer)

Congestion Scenario 2

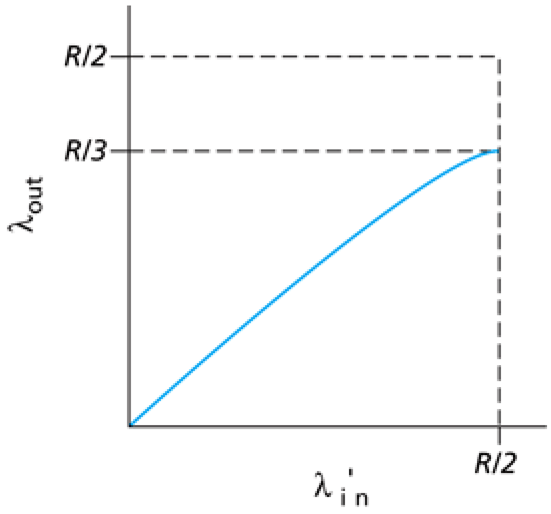
- Two senders (and receivers); a router with finite buffers



  - If buffer is full and new packets arrive, packets will be dropped/lost
  - Lost packets leads to retransmissions by the source hosts
  - λ'_{in} is the *offered load*: original sending rate + retransmission rate

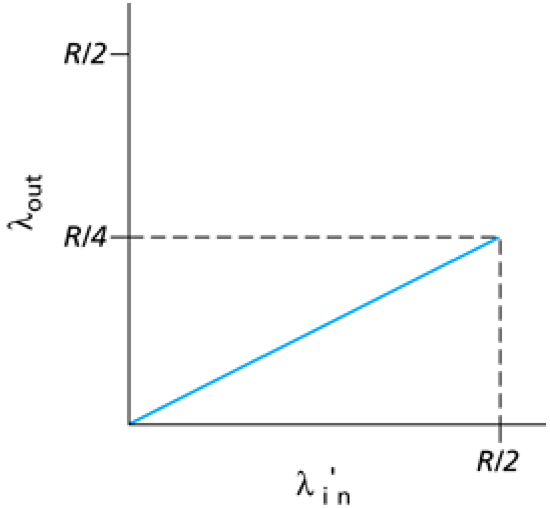
Congestion Scenario 2

- Lets assume source retransmits only when a packet is known to be lost
  - Offered load (λ'_{in}): original sent + retransmissions
  - If every second packet is lost (due to buffer at router being full):
    - For every 2 original packets, 1 retransmitted packet;
    - Offered load is 3 packets
    - 2 packets successfully received at destination

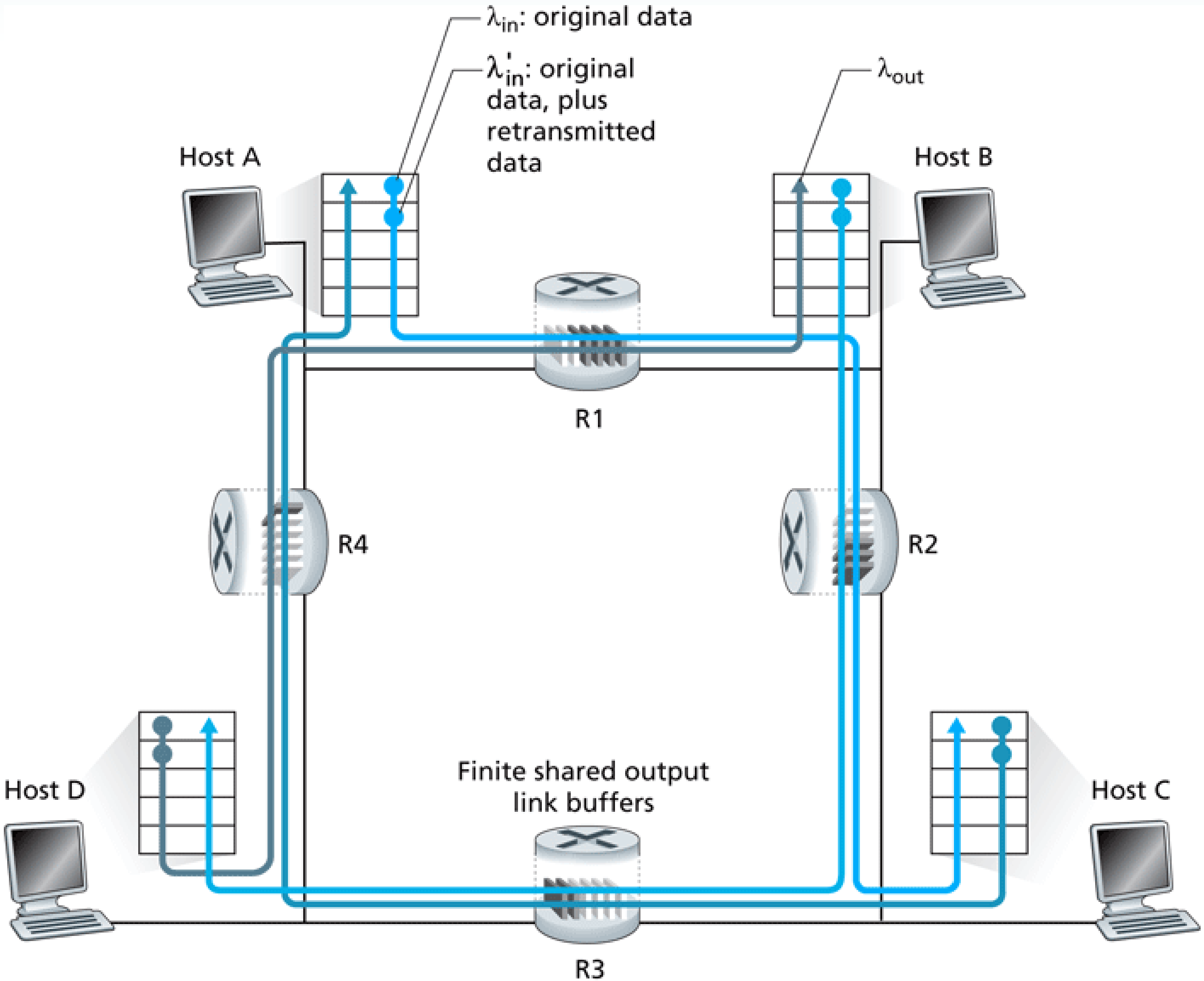


Congestion Scenario 2

- Lets assume a sender times out too early and retransmits a packet even though the original packet was sent by the router to destination
  - The output link from the router will be used to send the original and retransmitted packet
  - But the retransmitted packet will be ignored (discarded) by the destination

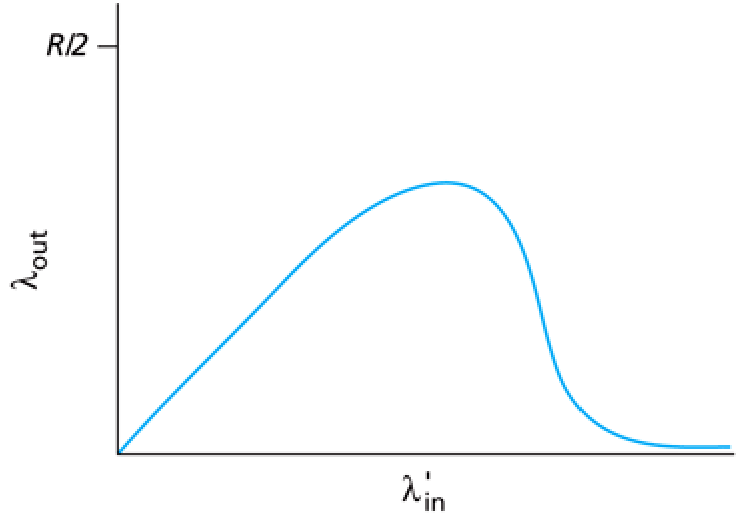


Congestion Scenario 3



Congestion Scenario 3

- Throughput can go to 0 with large amount of traffic
- Network spends all the time sending unneeded/wasted packets



Costs of Congestion

- Large queuing delays are experienced as the sending rate nears the output link capacity at a router
- Sender must perform retransmissions in order to compensate for dropped (lost) packets due to buffer overflow
- Router may send unneeded copies of packets if sender retransmits due to large delays (but not lost packets)
- With multiple routers in a path, if a packet is dropped by a router, all links leading up to that router have been wasted

Approaches to Congestion Control

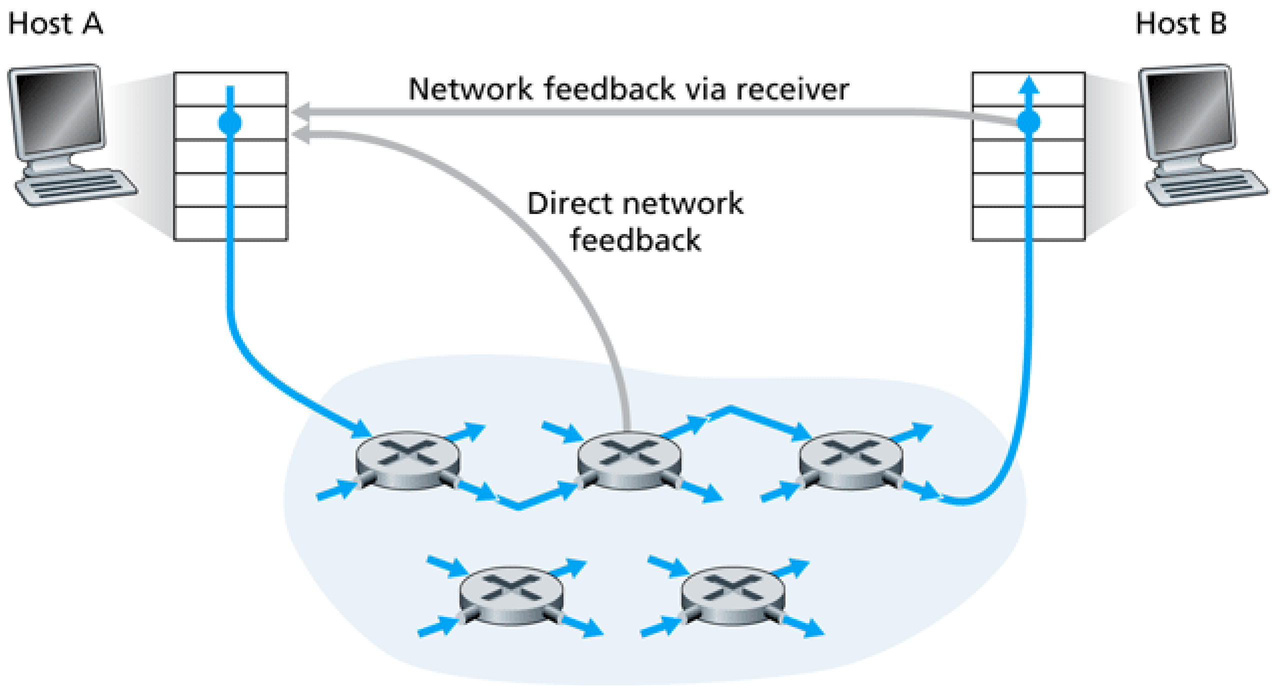
Approaches to Congestion Control

- End-to-end Congestion Control: Transport Layer
  - Network layer provides no feedback on congestion in network
  - End systems (source/destination hosts) infer congestion based on detected events such as packet loss and/or delay

- Network Assisted Congestion Control: Network Layer
  - Network devices (mainly routers) provide explicit feedback to the source host about congestion
    - Routers may provide direct feedback to source
    - Feedback from routers may be provided via the destination host
  - Feedback may be:
    - Backpressure: router A tells previous router B to slow down; router B tells previous router C to slow down; and so on
    - Explicit signalling: routers or destination host send special packets to source informing it of congestion and/or indicate the appropriate rate

Network Assisted Congestion Control

- Feedback may come direct from routers, or via the destination (receiver)



- ATM is an example network technology using Network Assisted Congestion Control

TCP Congestion Control

TCP Congestion Control

- TCP sender limits the rate at which it sends based on perceived network congestion
- How does TCP sender limit its sending rate?
- How does TCP sender perceive there is network congestion?
- How does TCP sender respond to congestion?
  - TCP congestion control algorithm

Limiting the TCP Sending Rate

- Amount of bytes TCP sender can send is limited by Advertised Window from Flow Control
  - Outstanding Bytes ≤ Advertised Window
- In fact, TCP sender also maintains Congestion Window:
  - Outstanding Bytes ≤ min (Advertised Window, Congestion Window)
- When an ACK is received, more bytes can be sent by TCP sender
- Assume the Advertised Window is very large (buffer at receiver is very large)
  - Sending rate ≈ Congestion Window/RTT
- By adjusting the Congestion Window, TCP sender can adjust its sending rate

Perceiving Network Congestion

- TCP sender assumes a loss indicates increased network congestion
  - A loss:
    - TCP sender times out: has not received ACK within timeout period
    - TCP sender receives 3 duplicate ACKs
  - Is this a valid assumption?
    - Most packet losses occur at routers, i.e. congestion
    - However, in some networks (e.g. wireless), packets may be lost due to link errors, not congestion
- TCP sender assumes arrival of ACKs indicates decreased network congestion
  - The faster the arrival rate of ACKs, the large decrease in congestion assumed

TCP Congestion Control Algorithm

- Three main components:
  1. Additive Increase, Multiplicative Decrease (AIMD)
  2. Slow tart (SS)
  3. Reaction to Loss Events

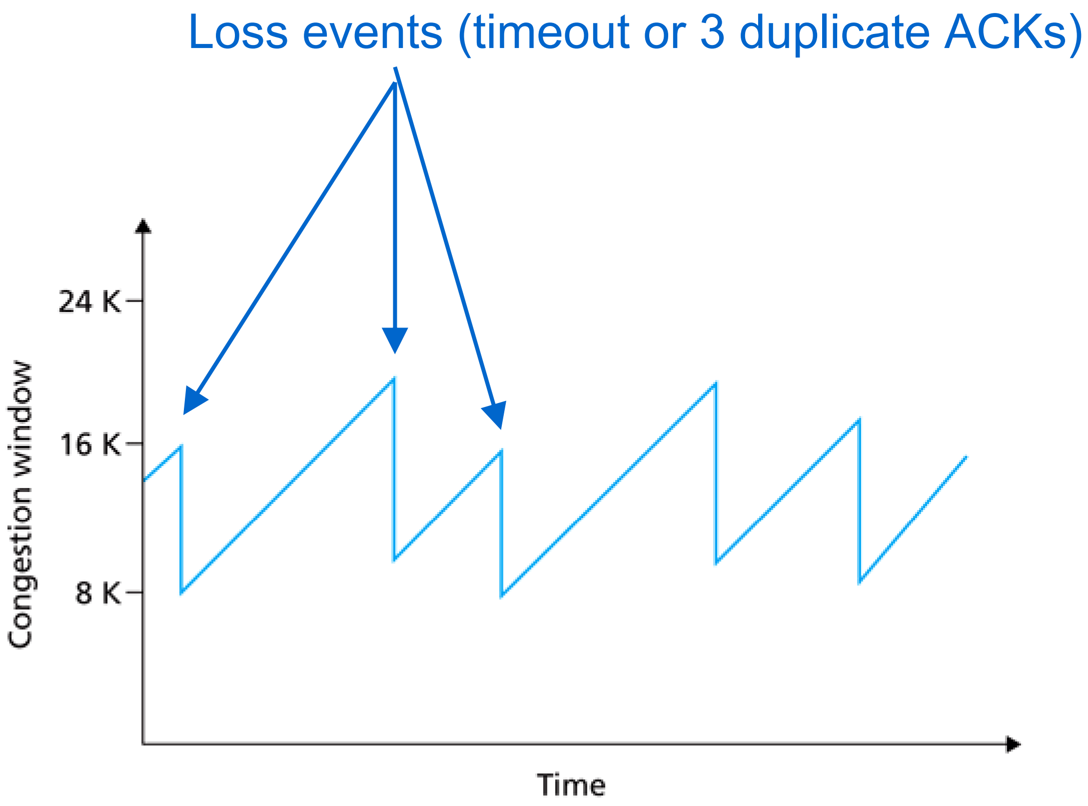
- Terminology
  - Maximum Segment Size (MSS): determined or assumed by TCP sender for network path; measured in bytes
  - Congestion Window (cwnd): measured in bytes
  - Round Trip Time (RTT): time from sending a segment, until corresponding ACK is received

- We will assume:
  - TCP receiver sends an ACK for every segment received

AIMD

- Additive Increase
  - If no congestion detected, then TCP sender assumes there is available (unused) capacity in the network; hence increases its sending rate
    - However, TCP slowly increases its sending rate
  - TCP sender aims to increase Congestion Window by 1 x MSS every RTT
  - One approach:
    - For every new ACK received, increase Congestion Window (cwnd) by:
      - $cwnd_{new} = cwnd_{old} + MSS * MSS / cwnd_{old}$
  - Additive Increase phase is also called Congestion Avoidance
- Multiplicative Decrease
  - If congestion detected, then TCP sender decreases its sending rate
  - TCP sender aims to half the Congestion Window for each loss
    - For each loss detected:
      - $cwnd_{new} = cwnd_{old}/2$
    - cwnd is not decreased to less than 1 MSS
    - (This is not entirely accurate – see Reaction to Loss Events)

AIMD Example

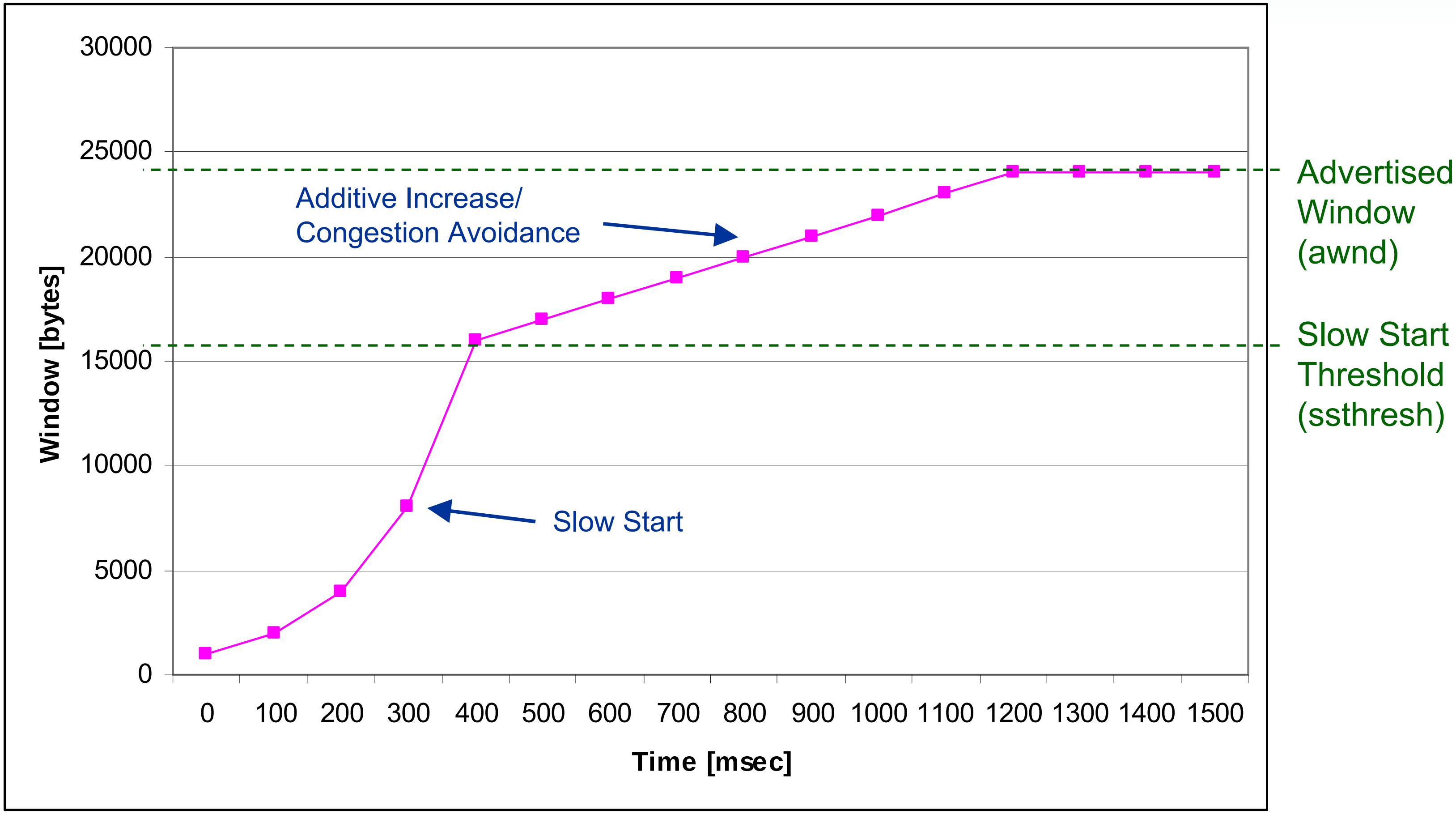


Slow Start Phase

- At start of a TCP connection, the TCP sender sends at a slow rate
  - By default: cwnd = MSS
  - E.g. approximate sending rate for MSS = 1000 bytes, RTT = 200ms is 40kb/s

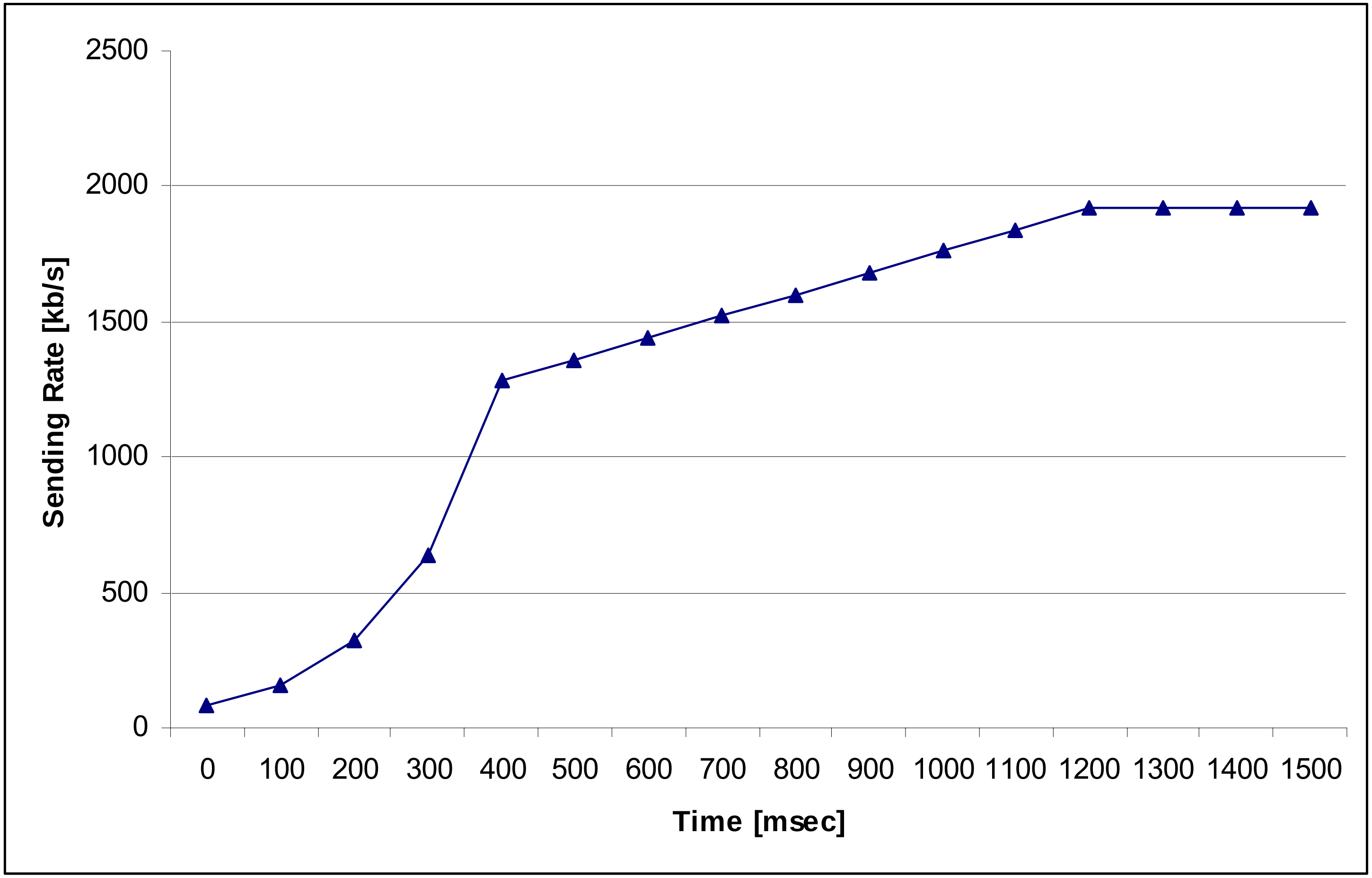
- If large capacity is available for the connection, using additive increase (congestion avoidance) will be too slow
- Therefore Slow Start phase involves *very fast* increase of Congestion Window
  - Cwnd is increased exponentially
  - For every ACK received in Slow Start phase, increase cwnd by 1 MSS
    - $cwnd_{new} = cwnd_{old} + MSS$
  - Slow Start phase is continued until a loss event (then multiplicative decrease) or Congestion Window reaches a threshold (ssthresh) value (then additive increase)

AIMD and Slow Start: Window Size



MSS = 1000 B; RTT=100ms; ssthresh=16000 B; Advertised Window = 24000 B
Window = min (Congestion Window, Advertised Window)

AIMD and Slow Start: Sending Rate

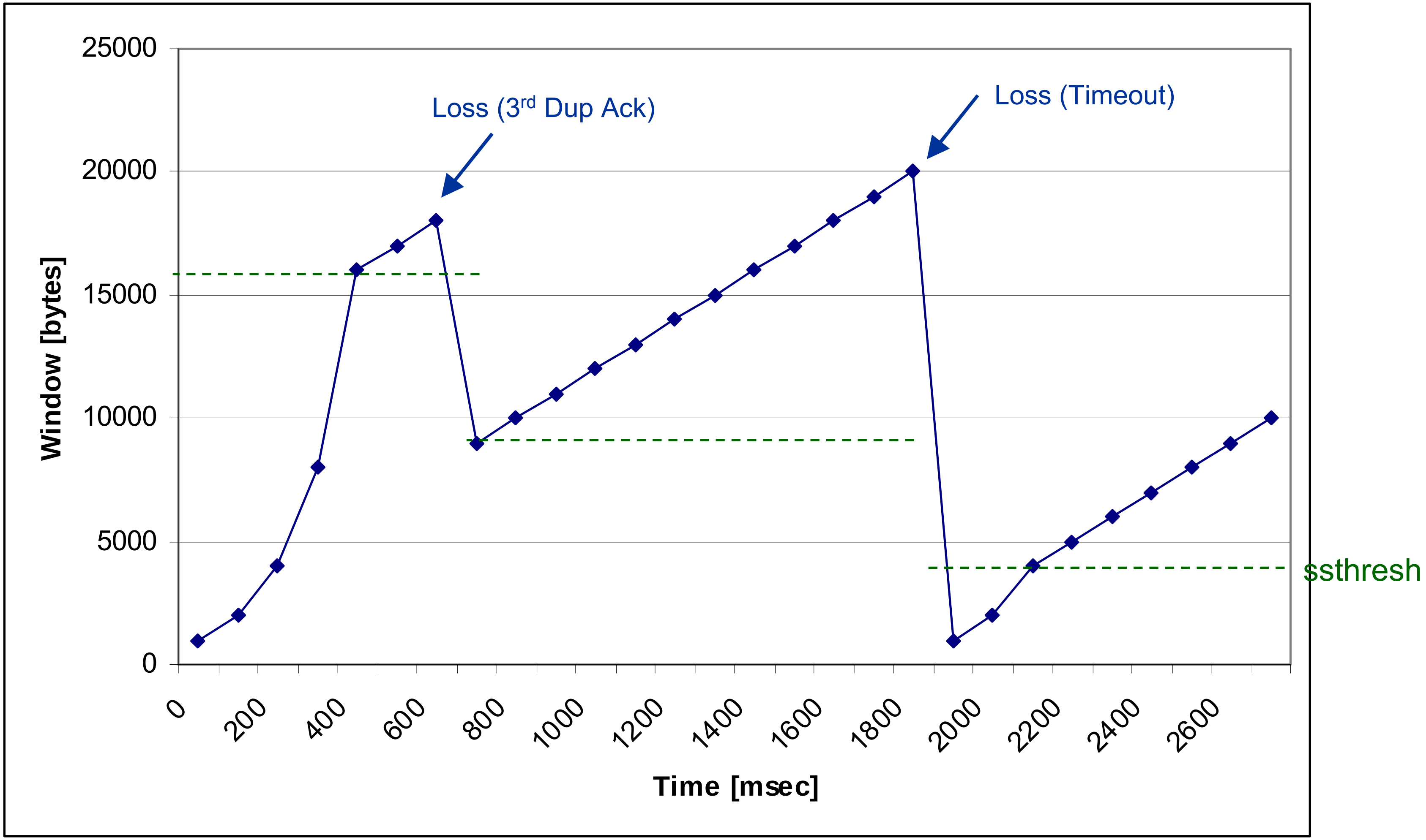


Assumes Sending Rate = Window / RTT

Reaction to Loss Events

- Upon a loss, Multiplicative Decrease halves the current congestion window
- The next action then depends on type of loss event:
  - Loss detected by 3rd Duplicate ACK
    - Slow start threshold is halved
      - $ssthresh_{new} = ssthresh_{old}/2$
    - Congestion window set to slow start threshold
      - $cwnd_{new} = ssthresh$
    - TCP enters Additive Increase (or Congestion Avoidance) phase
  - Loss detected by a timeout
    - Slow start threshold is halved
      - $ssthresh_{new} = ssthresh_{old}/2$
    - Congestion window set to initial value of 1 MSS
      - $cwnd_{new} = MSS$
    - TCP enters Slow Start phase

Reaction to Loss Events



Reaction to Loss Events

- Why?
  - TCP assumes a loss indicates congestion in the network (and therefore slows down)
  - Loss due to 3rd Duplicate ACK
    - Some TCP segments are being delivered (since some ACKs are coming back)
    - TCP assumes small level of congestion, therefore immediately enters Congestion Avoidance phase
  - Loss due to Timeout
    - Most TCP segments were lost (since not even duplicate ACKs are received)
    - TCP assumes heavy congestion, therefore go back to start (of Slow Start) with very slow sending rate

TCP Congestion Control in Practice

- TCP Congestion Control algorithm works well in networks were losses are mainly due to congestion
  - Note that with a congested network, the throughput of TCP connection can be severely limited
- In networks with losses due to errors on links, TCP Congestion Control has problems
  - Example: a wireless link may lose segments due to poor link quality
    - TCP slows down (thinking it is congestion) when it should maintain its sending rate
  - Several variants of TCP have been developed specifically for wireless links
- In high-speed networks (>10Gb/s), TCP may perform poorly even with very few link packet losses

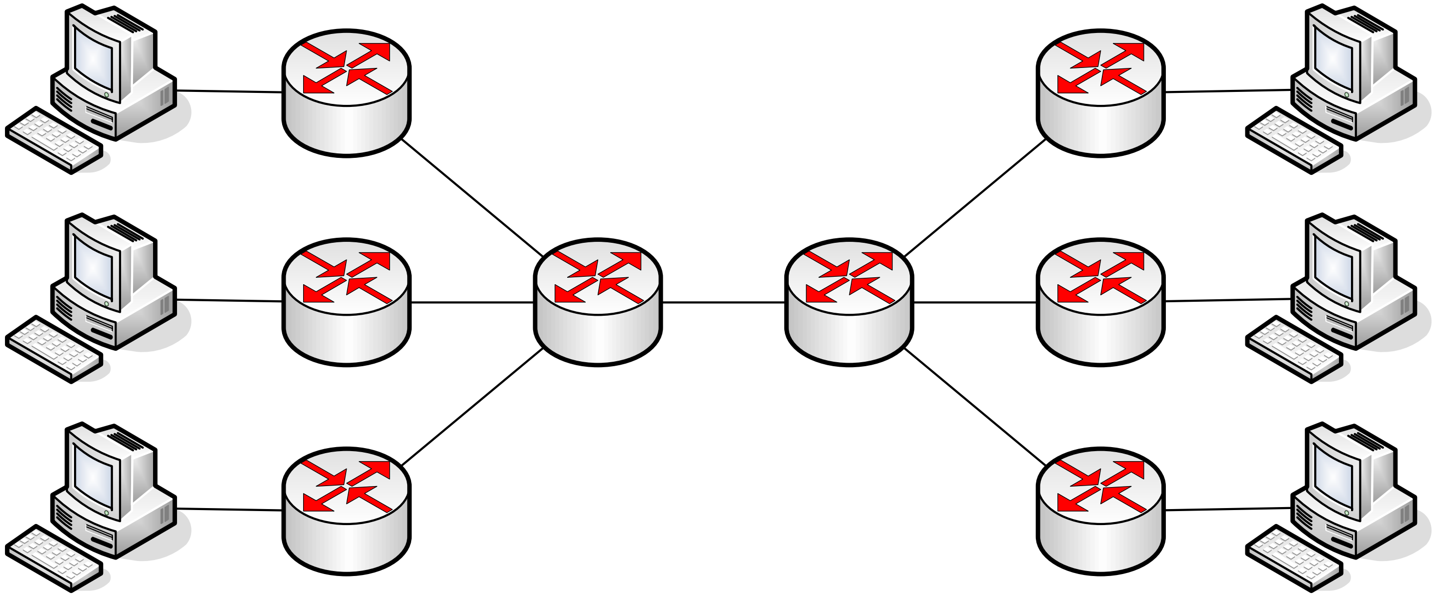
TCP Versions and Options

- TCP RFC 793 (1981)
  - Reliability (sequence numbers), Flow control (receiver window), Connection management
- TCP Tahoe (1988)
  - Adds Slow Start, Congestion Avoidance, Fast Retransmit
- TCP Reno (1990)
  - Adds Fast Recovery
- TCP NewReno (1995)
  - Only halves congestion window once
- Other Options:
  - Selective Acknowledgement (SACK)
  - TCP Vegas

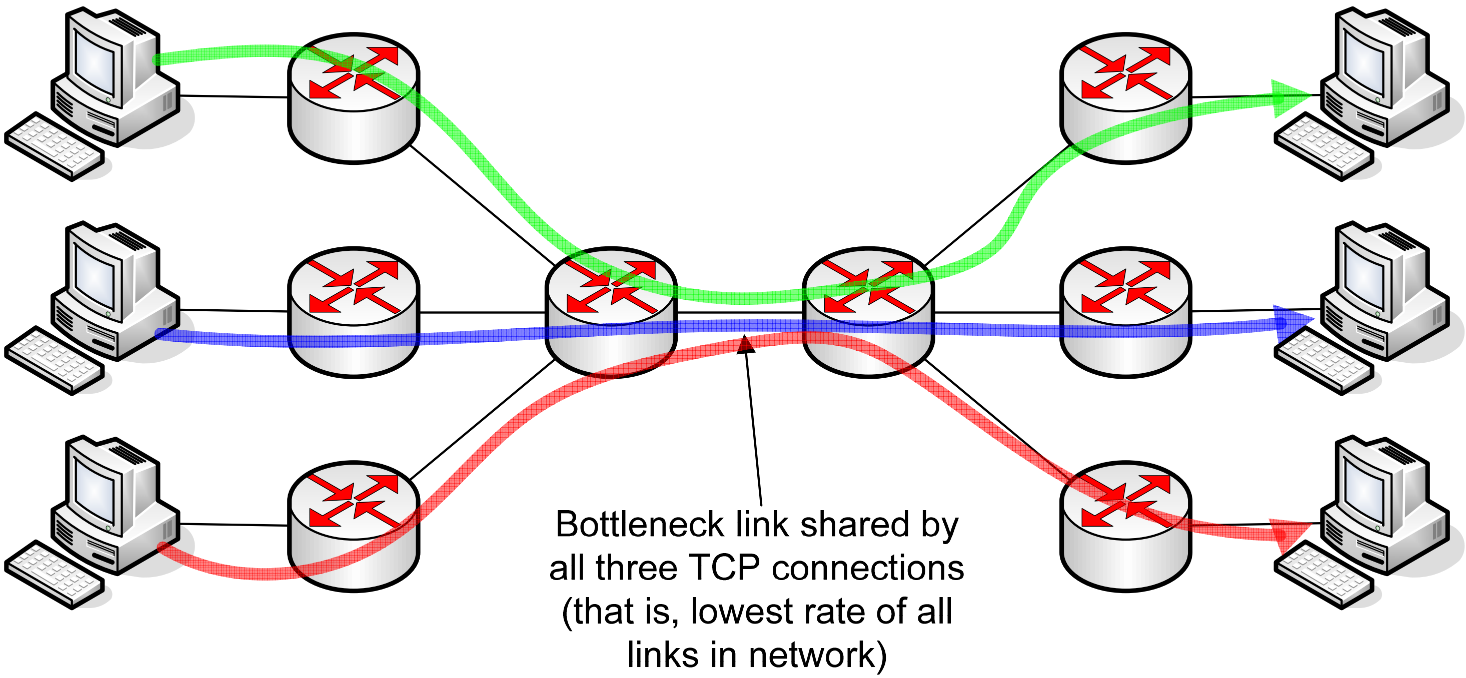
- Some Operating Systems implement their own options/variants

TCP Fairness

Example: TCP Fairness



Example: TCP Fairness

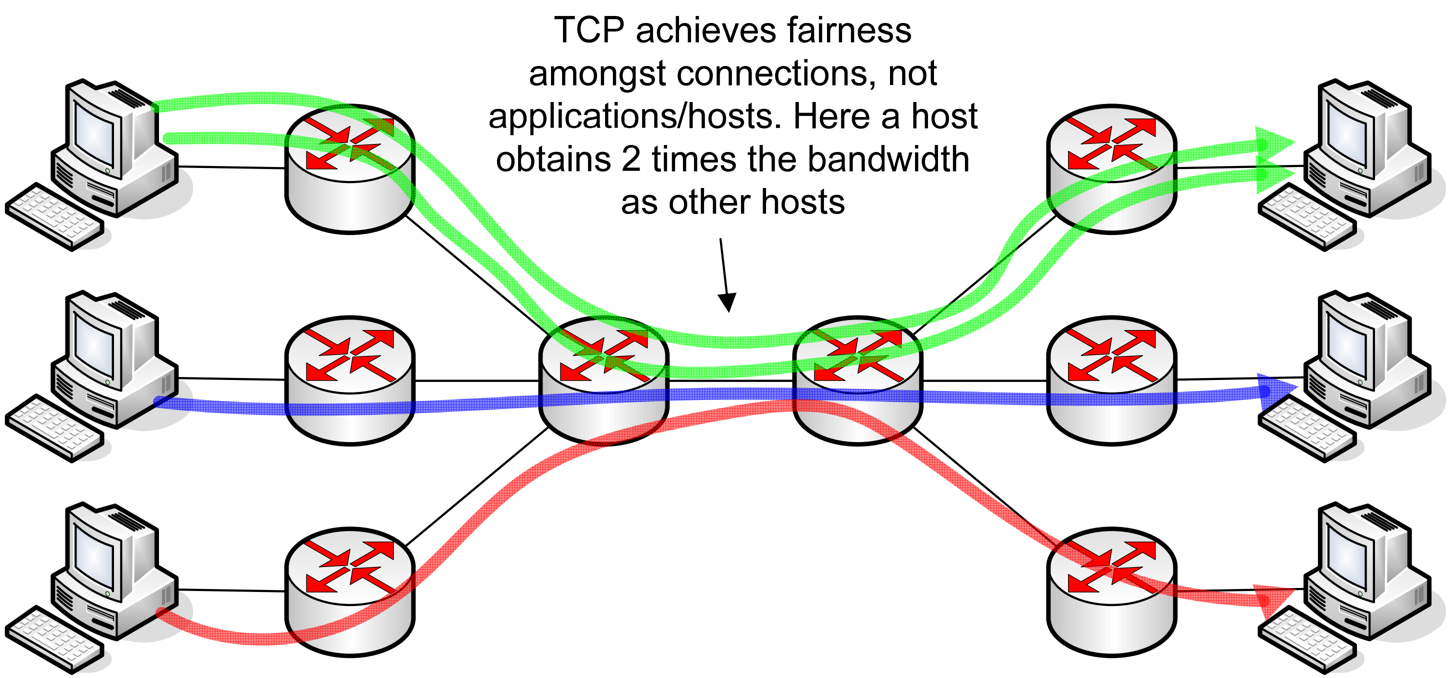


TCP Fairness

- If TCP is fair, with N TCP connections using a R bps link
  - Each TCP connection should achieve R/N bps

- Does TCP achieve fairness?
  - In ideal conditions, yes. If all TCP connections have same RTT and same sized segments, with no other traffic, fairness is achieved
  - In practice:
    - If RTT of connections vary, connections with small RTT are able to higher proportion of bandwidth than connections with large RTT
    - If other non-TCP data is also present (such as multimedia using UDP), then TCP connections receive unfair treatment
    - Applications can use multiple TCP connections: each TCP connection gets fair treatment, but the application using multiple connections gets more bandwidth than application using single connection

Example: TCP Fairness



TCP and The Internet

- IP does not include any built-in congestion control mechanisms
  - If every host sent IP datagrams as fast as possible, the Internet would not work
- The Internet relies on TCP mechanisms to avoid collapse
  - TCP comprises about 90% of all traffic on the Internet
  - As a means for congestion control, TCP has been very successful

- But …
  - If hosts/applications choose not to follow TCP's congestion control rules, then congestion can become a major problem in the Internet
  - Challenges:
    - Web browsers opening many TCP connections at once.
    - Growth of multimedia applications that use UDP.
    - Growth of P2P applications using multiple connections and/or UDP.