Transport Layer Congestion Control

Tom Kelliher, CS 325

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1 Administrivia

Announcements

Assignment

Read 4.1–4.3.

From Last Time

TCP Reliability.

Outline

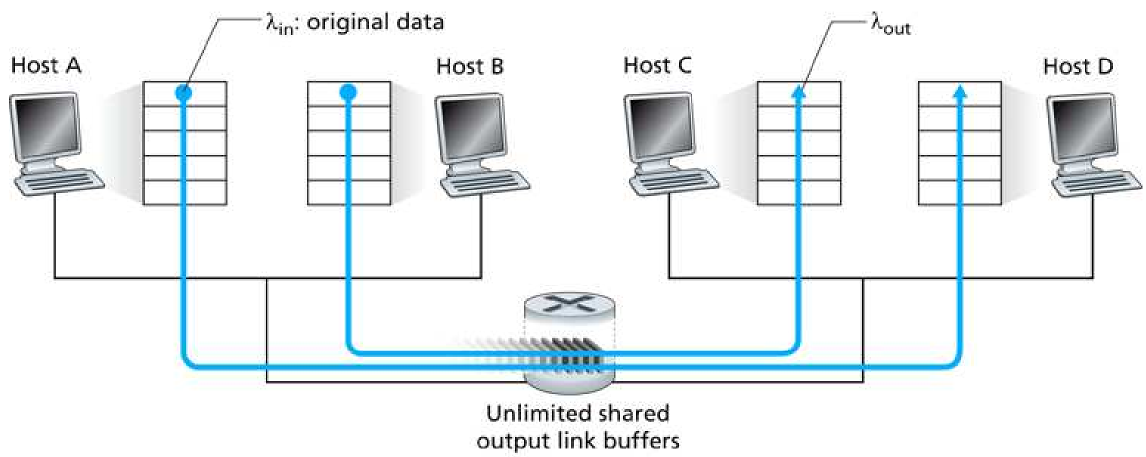
1. Congestion control principles.
2. TCP congestion control.

Coming Up

Network layer introduction.

2 Congestion Control Principles

2.1 Two Senders; Single Router with Infinite Buffers



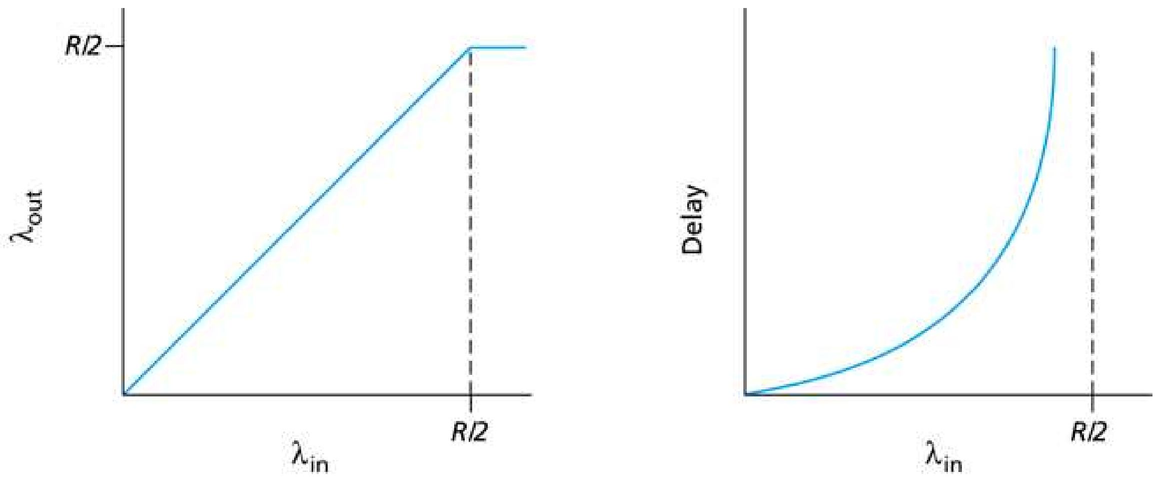
1. Assume no segments are dropped and senders don't time-out any segments — no retransmits.

2. The network load generated by each sender's application layer is λ_{in} bps.

   Assume the senders equally share the available bandwidth.

3. The router's outgoing link has capacity R bps.

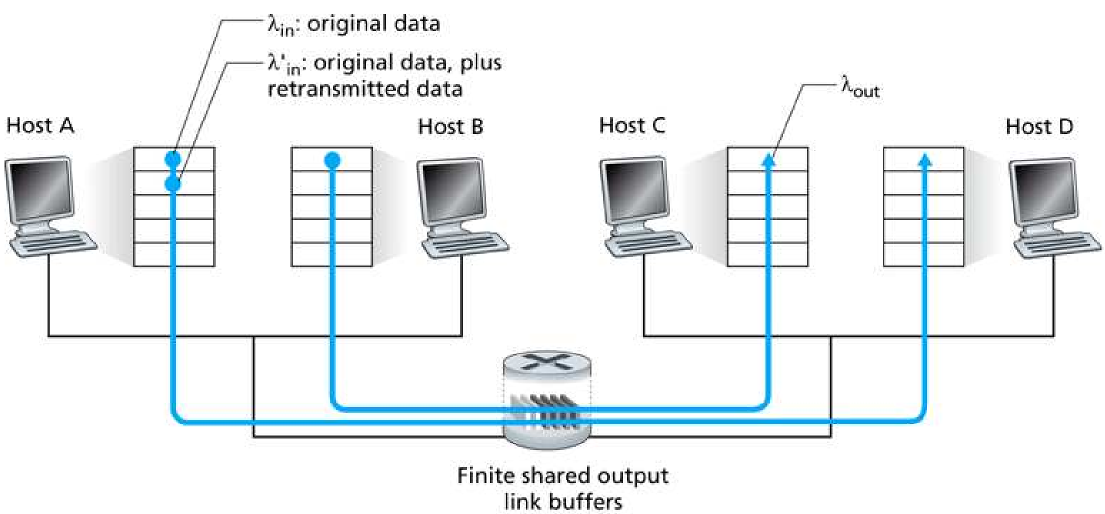
4. The network load seen by the receiver's application layer is λ_{out} bps.



1. $\lambda_{out} = \lambda_{in}$ until we hit the bandwidth limit — $R/2$ shared — $lambda_{out}$ can't exceed this.

2. As λ_{in} approaches $R/2$, the router's segment queue's length — and delay — increases.

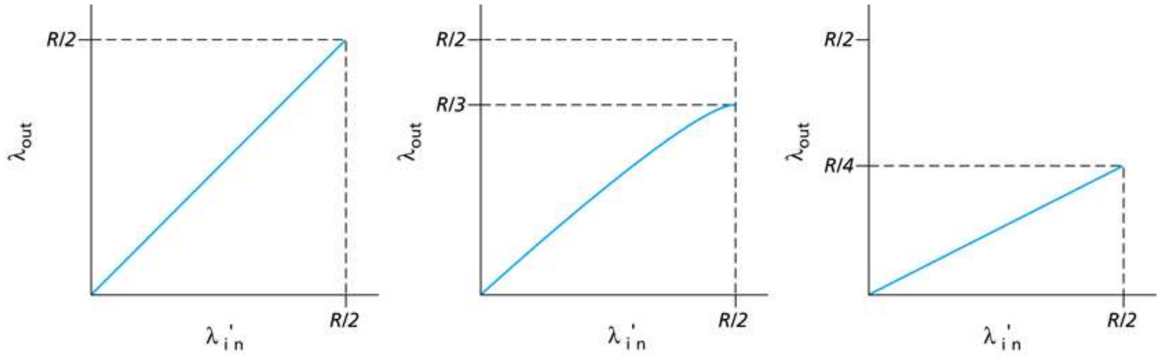
2.2 Two Senders; Single Router with Finite Buffers



1. Now, segments will be be dropped and retransmits will occur.

2. λ'_{in} is the **offered load** of the transport layer.

   Due to retransmits, $\lambda'_{in} \geq \lambda_{in}$.



1. Due to finite buffers, $\lambda'_{\in}$ can't exceed $R/2$ — shared link capacity.

   The graph on the left assumes the sender is omniscient and knows when the router has free buffers and only transmits segments then, avoiding dropped segments. — unrealistic.

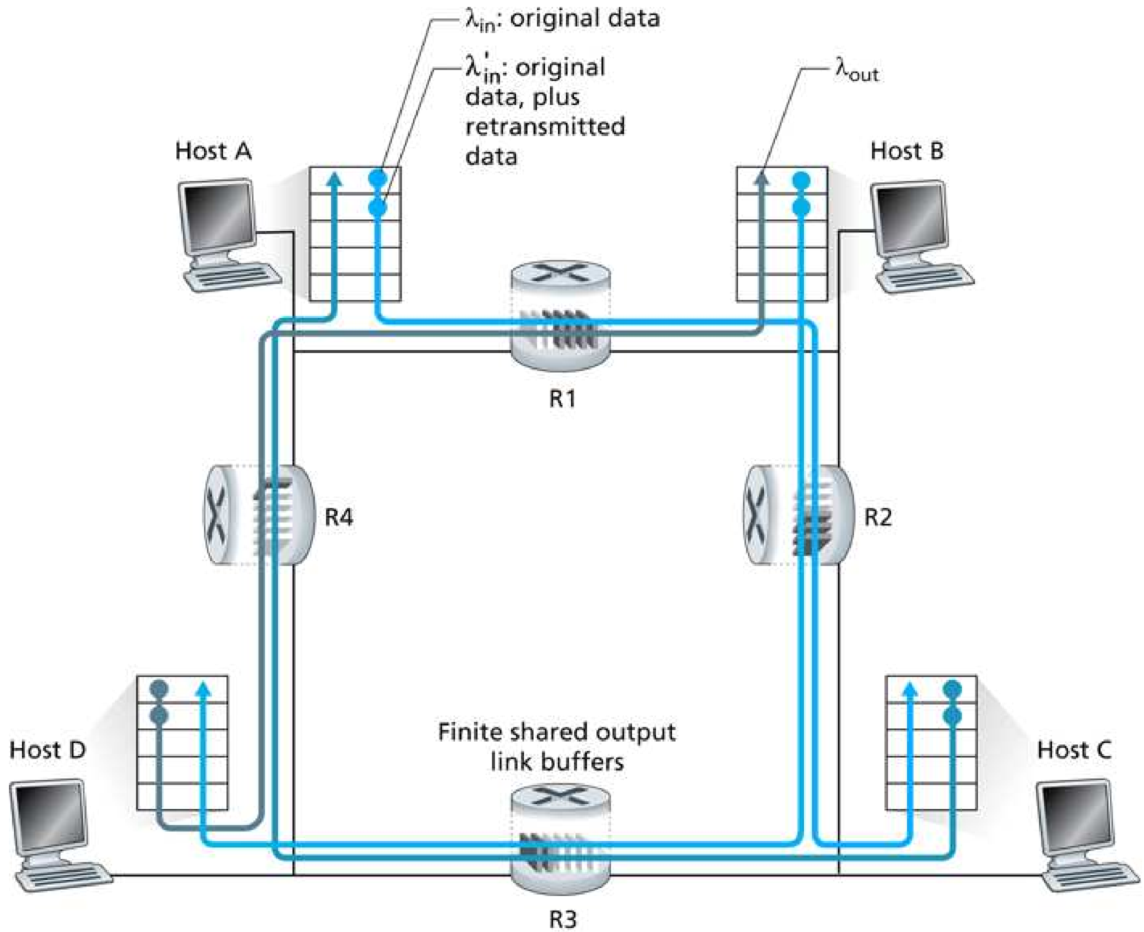
2. Dropped segments mean duplicate, redundant data sent across a link — wasting bandwidth.

   The middle graph shows what could happen if the sender retransmits **only** segments known to be lost — again, unrealistic. Here, we assume 17% of segments are retransmits.

3. Realistically, delays will cause some non-dropped segments to be retransmitted, further wasting bandwidth with redundant segments.

   The graph on the right shows this realistic scenario, assuming each segment is retransmitted once.

2.3 Four Senders; Multiple Routers with Finite Buffers; Multihop Routes



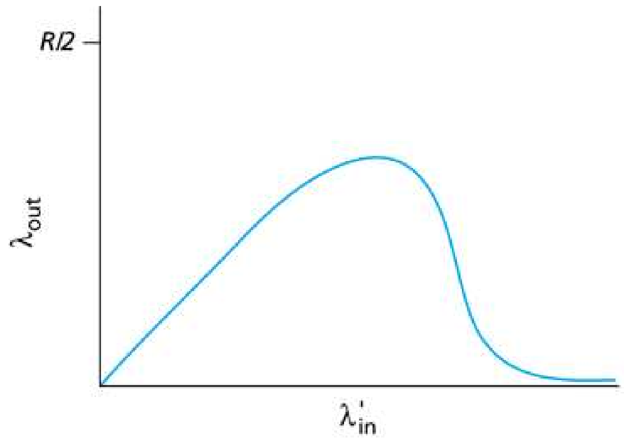
1. Consider what could happen if R2 becomes congested due to B–D traffic:

   (a) R2 begins dropping segments from A–C, wasting bandwidth at R1.

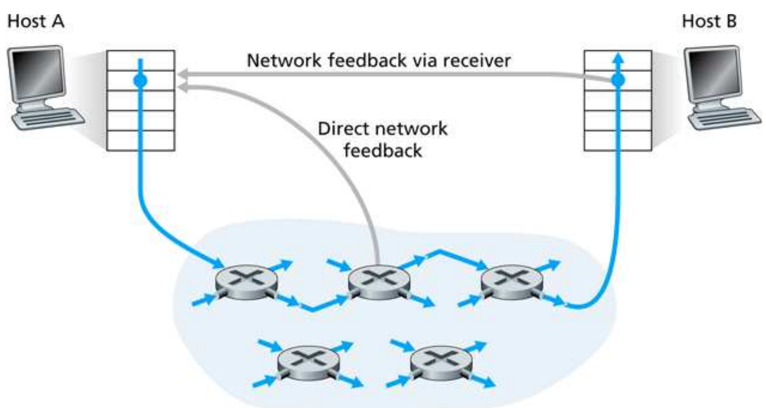
   (b) A's offered load increases, to handle retransmits.

   (c) R1 could become congested, affecting the (initially free) D–B route.

Worst case, B–D traffic could completely lock-out A–C traffic beyond the point at which λ'_{in} saturates the routers' transmit capacity:



2.4 Congestion Control Approaches



Two approaches:

1. End-to-end control:

   (a) No help from network layer —sender/receiver have to intuit congestion on their own.

   (b) TCP intuits congestion through fast retransmits (triple ACKs), not too bad — some bandwidth still available; timeout transmits, really bad — no bandwidth available.

Third idea for TCP: increased RTTs mean congestion is beginning to become a problem.

2. Network assisted. Two approaches here:

   (a) Direct feedback with choke packet — router sends choke command directly to sender.

   (b) Indirect feedback with congestion indication bit in segment — router sets this, when congested, as it forwards a segment to receiver.

   Receiver responsible for getting the indication back to the sender.

3 TCP Congestion Control

1. Recall receive window for flow control:

$$\texttt{LastByteSent} - \texttt{lastByteAcked} \leq \texttt{RcvWindow}$$

2. Recall our transmit model: sender sends n segments each RTT with $n \times \texttt{MSS} = \texttt{RcvWindow}$.

   We therefore have $\lambda'_{\text{in}} = \texttt{RcvWindow}/RTT$.

   So, `RcvWindow`, controlled by receiver, can be used to throttle sender rate.

3. TCP itself defines `CongWindow`, maintained by sender, to throttle sender rate in face of congestion.

   We then have:

$$\texttt{LastByteSent} - \texttt{lastByteAcked} \leq \min(\texttt{RcvWindow}, \texttt{CongWindow})$$

   Sender rate controlled by **both** `RcvWindow` and `CongWindow`.

TCP congestion control algorithm:

1. Multiplicative decrease:

(a) After triple duplicate ACK, cut `CongWindow` in half. `CongWindow` never drops below 1 MSS.

Not as bad as a timeout.

(b) After a timeout, cut `CongWindow` down to 1 MSS.

2. Additive increase:

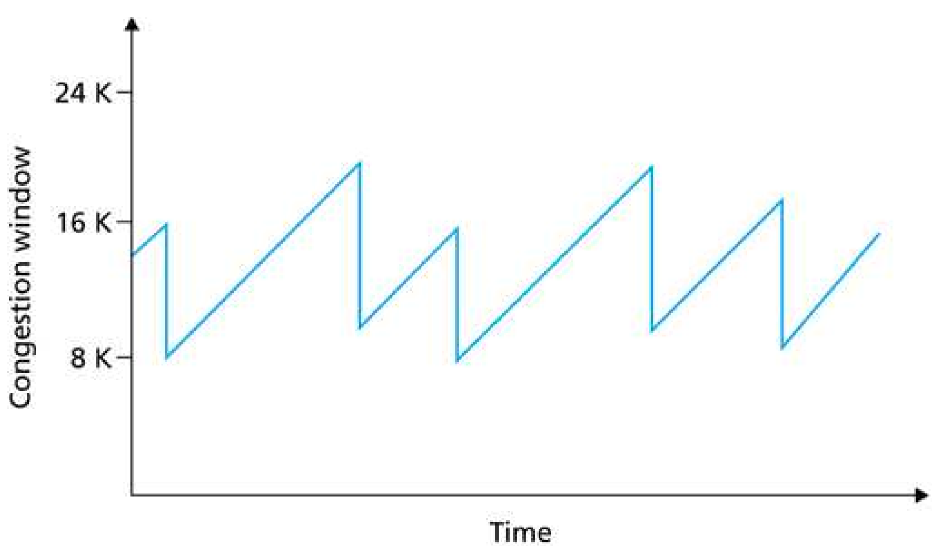
(a) Increase `CongWindow` by 1 MSS each RTT.

Increase rate is controlled by RTT — a lower RTT results in faster `CongWindow` increase rate.

Often implemented by increasing `CongWindow` by 1 MSS $\times$ (MSS/`CongWindow`) for each new ACK.

(b) Actually, `CongWindow` increase is multiplicative until `Threshold` is reached.

(c) Additive increase; multiplicative decrease:

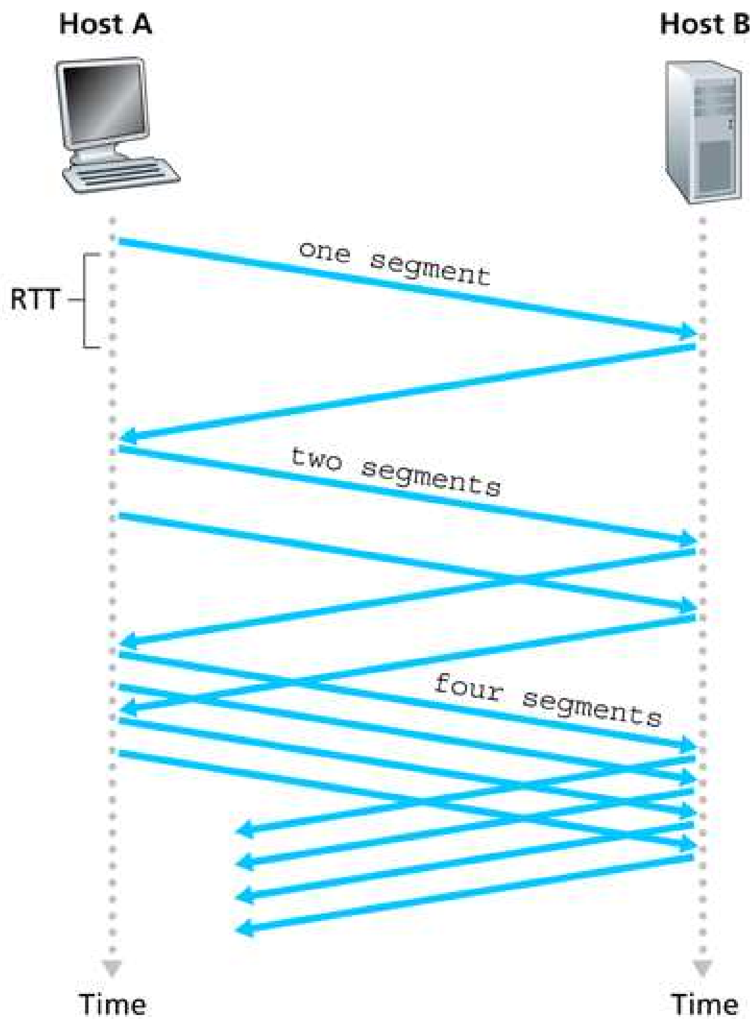


3. Slow start:

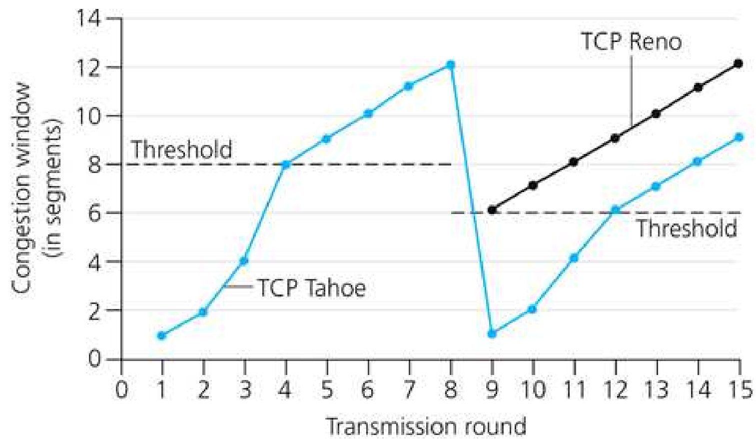
(a) `CongWindow` is set to 1 MSS for a new connection.

(b) `CongWindow` is increased by 1 MSS for each ACK received, until `Threshold` is reached.

Exponential increase in `CongWindow` during SS phase:



(c) Additive increase, once `Threshold` reached:



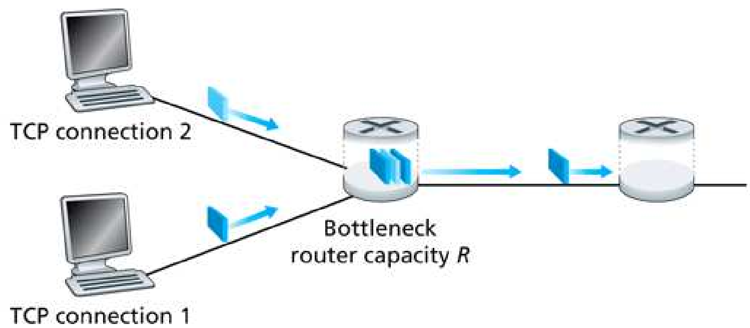
TCP Reno = current algorithm. Is decrease from timeout or fast retransmit?

4. Actual timeout events behavior:

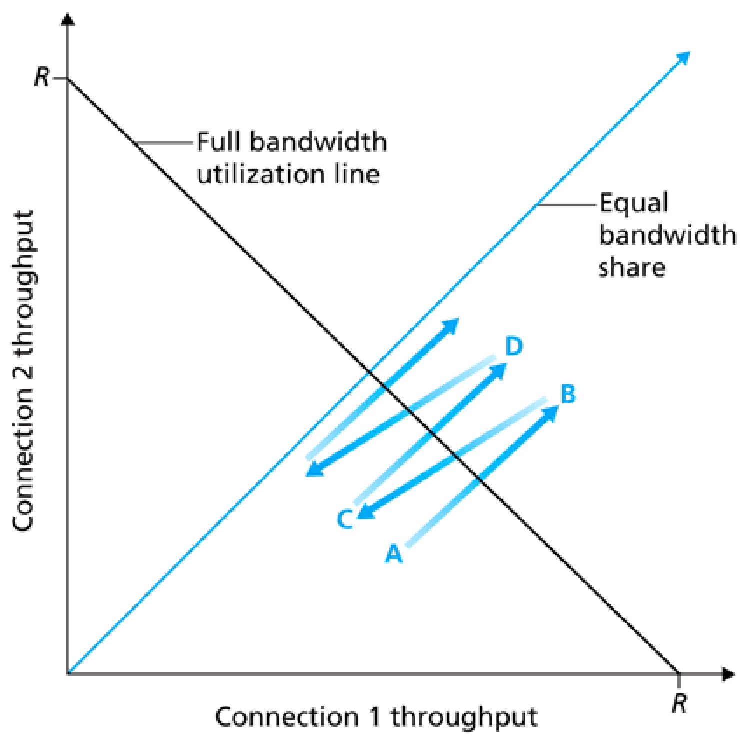
State	Event	Sender Action	Comment
Slow Start (SS)	New ACK received	CongWin = CongWin + MSS. if (CongWin > Threshold) set state to CA.	CongWin doubles every RTT.
Congestion Avoidance (CA)	New ACK received	CongWin = CongWin + MSS(MSS/CongWin)	CongWin increases by 1 MSS every RTT.
SS or CA	Triple Dup ACK	Threshold = CongWin/2. CongWin = Threshold. set state to CA.	Fast recover; multiplicative decrease.
SS or CA	Timeout	Threshold = CongWin/2. CongWin = 1 MSS. Set state to SS.	
SS or CA	Duplicate ACK received	Increment duplicate ACK count for segment	

3.1 Fairness

Is TCP's AIMD algorithm fair? Consider this situation:



Suppose, initially, connection 1 has a higher throughput (`CongWindow`) than connection 2:



This shows what happens during 3DupACK events — multiplicative decrease. Eventually, we converge to equal throughput.

On timeout, both connections will end up with a `CongWindow` of 1 MSS.