CSC4200/5200 – COMPUTER NETWORKING

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CONGESTION CONTROL

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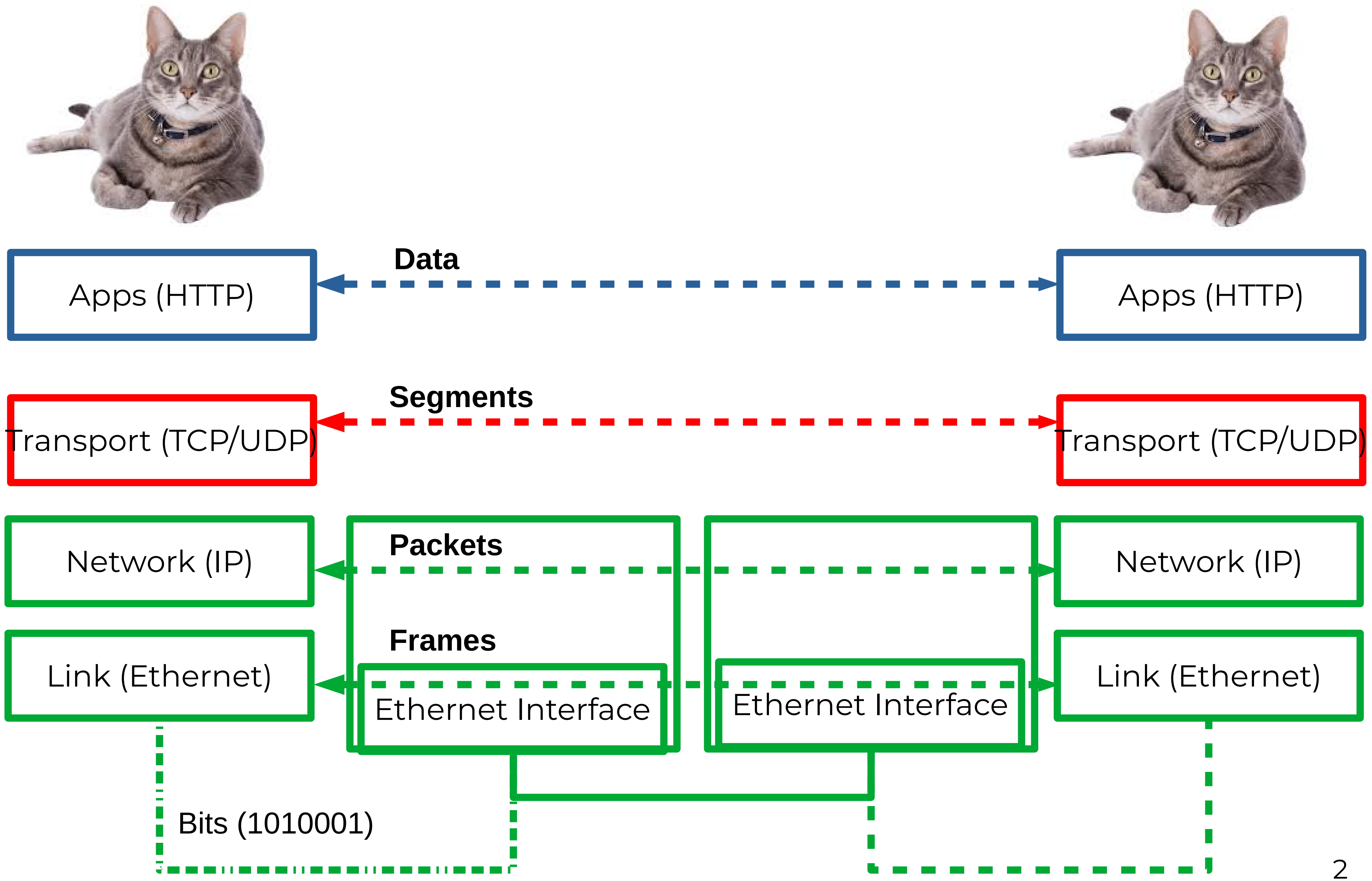
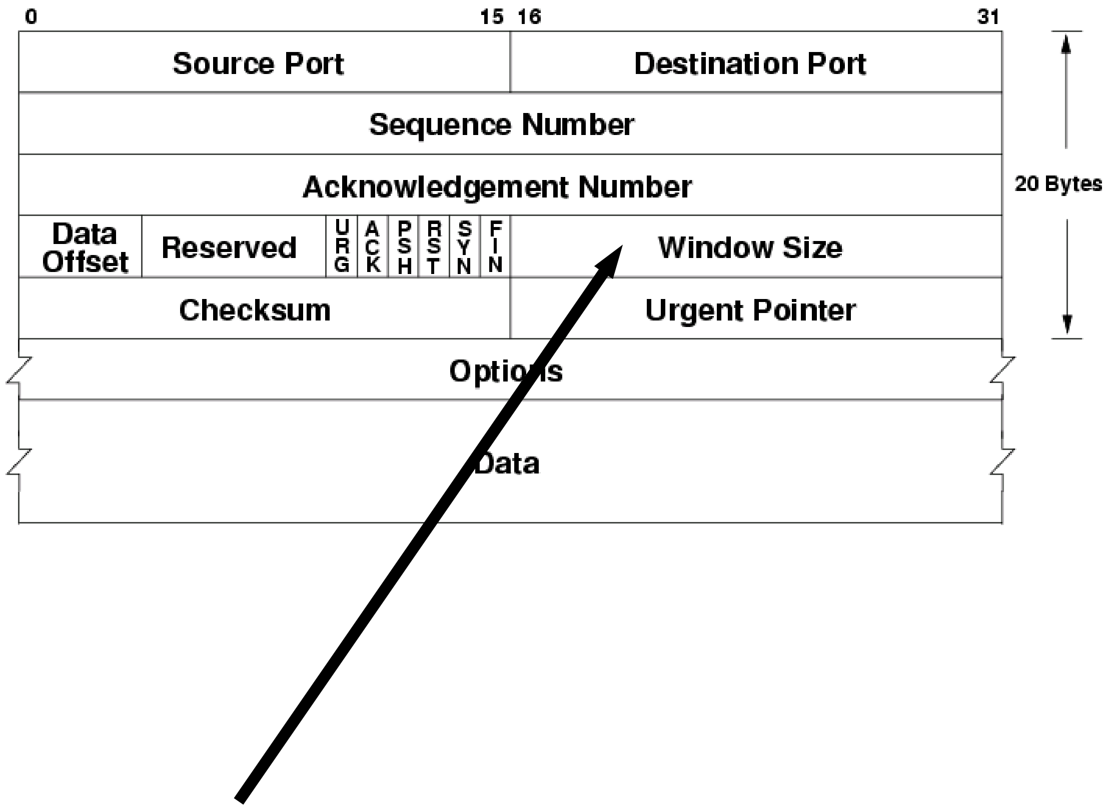
Tennessee TECH


Data
Apps (HTTP)
Apps (HTTP)
Segments
Transport (TCP/UDP)
Transport (TCP/UDP)
Packets
Network (IP)
Network (IP)
Frames
Link (Ethernet)
Ethernet Interface
Ethernet Interface
Link (Ethernet)
Bits (1010001)

TCP flow control

- receiver "advertises" free buffer space in the header
- sender limits amount of unacked ("in-flight") data to receiver's **`rwnd`** value
- guarantees receive buffer will not overflow



Congestion Control

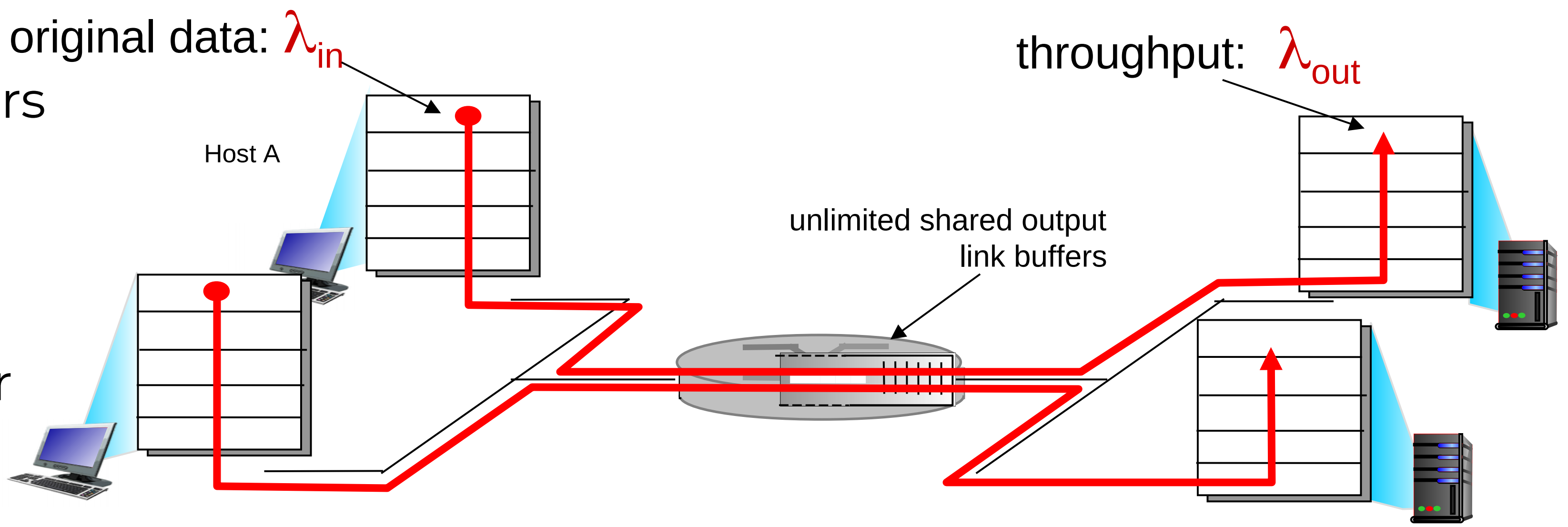
Principles of congestion control

congestion:

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

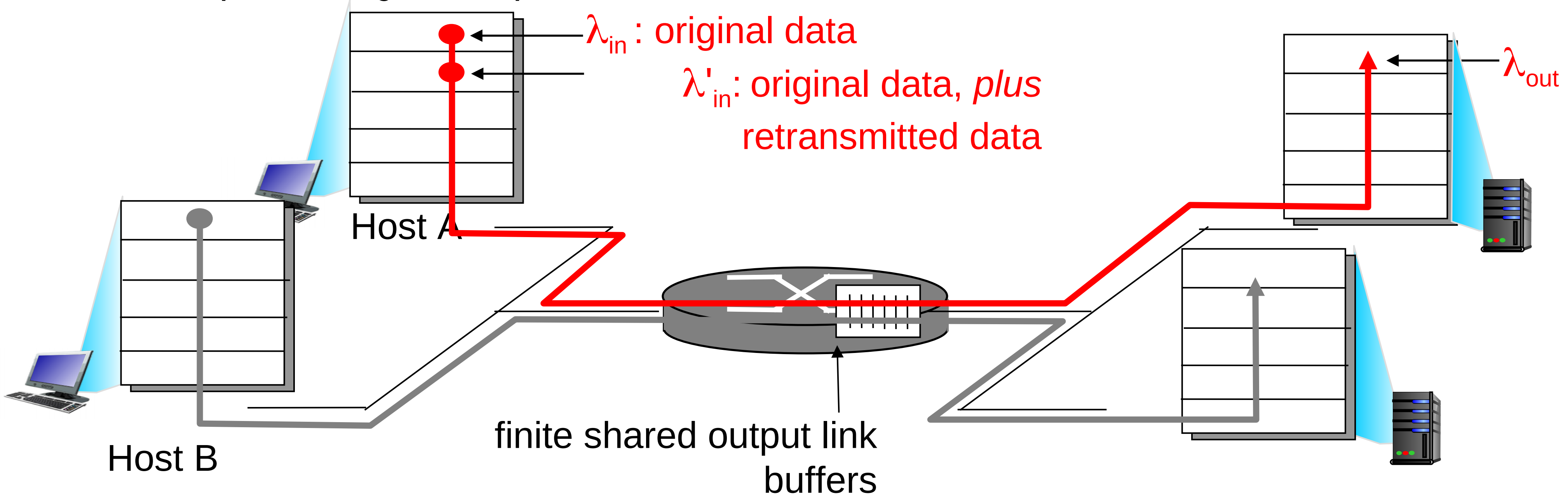
Congestion: scenario 1

- three senders, two receivers
- one router, infinite buffers
- output link capacity: R
- The router can only transmit one –... and either buffer or drop the other
- If many packets arrive,
- Buffer overflow


original data: λ_{in}
Host A
Host B
unlimited shared output link buffers
throughput: λ_{out}

Causes/costs of congestion: scenario 2

- one router, *finite* buffers
- sender retransmission of timed-out packet
  - application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$ ≥
  - transport-layer input includes *retransmissions* : λ_{in} λ_{in}



λ_{in} : original data
λ'_{in}: original data, *plus* retransmitted data
λ_{out}
Host A
Host B
finite shared output link buffers

Metrics: Throughput vs Delay

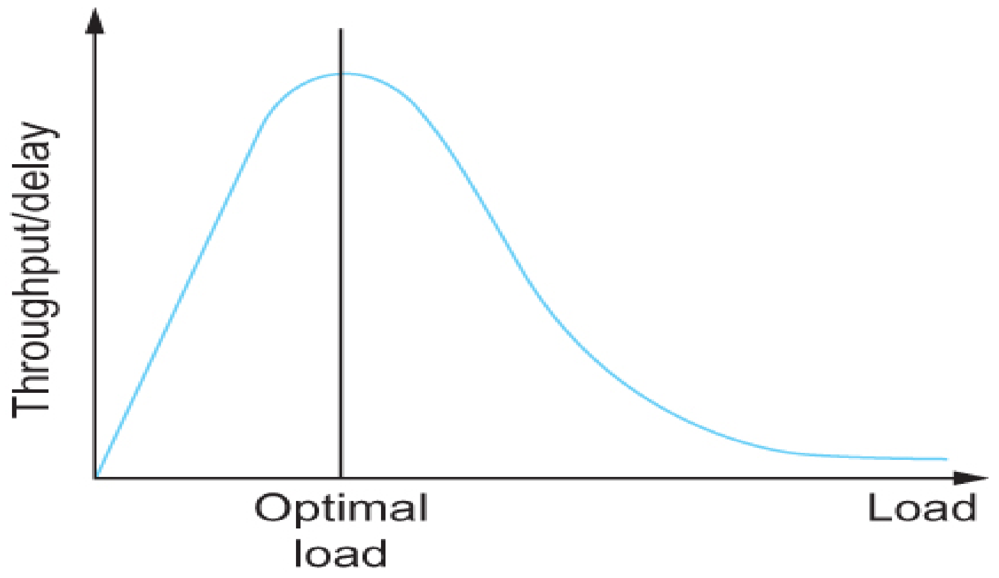
High throughput –

- Throughput: measured performance of a system –E.g., number of bits/second of data that get through
- Low delay –
- Delay: time required to deliver a packet or message –E.g., number of ms to deliver a packet •
- These two metrics are sometimes at odds –
  - More packets = more queuing

Issues in Resource Allocation

- Evaluation Criteria
  - Effective Resource Allocation

power of the network.
Power = Throughput/Delay



Ratio of throughput to delay as a function of load

Issues in Resource Allocation

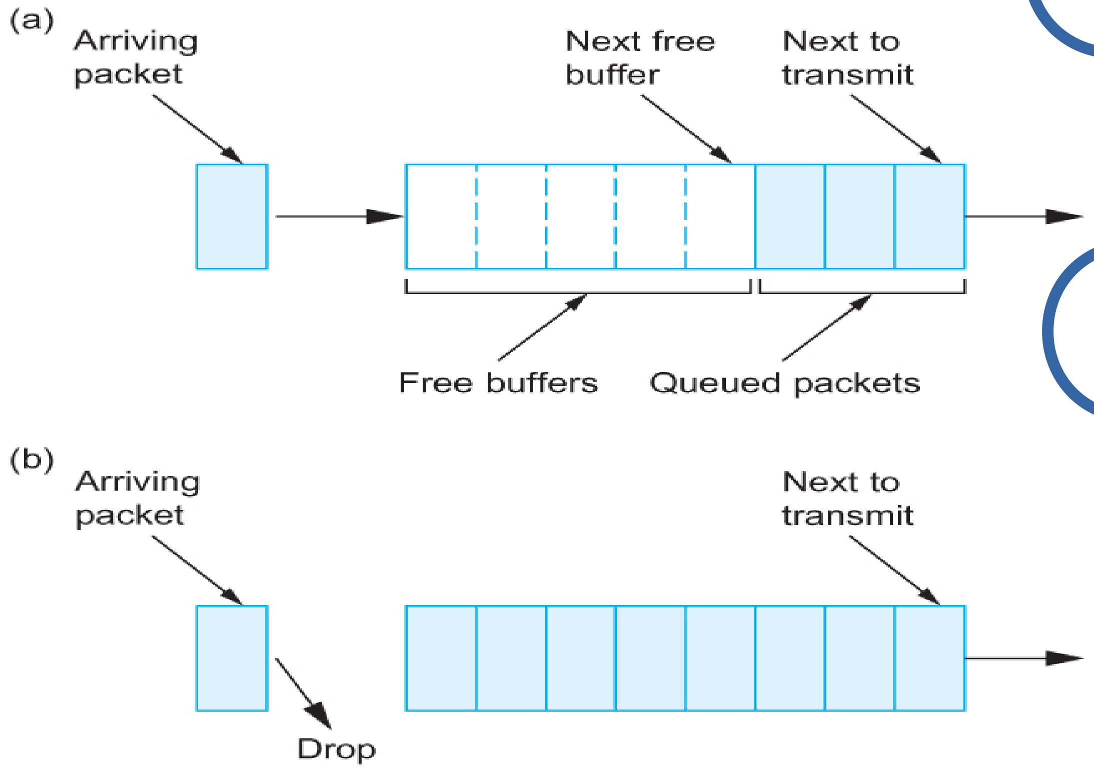
- Evaluation Criteria
  - Fair Resource Allocation
    - The effective utilization of network resources is not the only criterion for judging a resource allocation scheme.
    - We want to be "fair"
    - Equal share of bandwidth

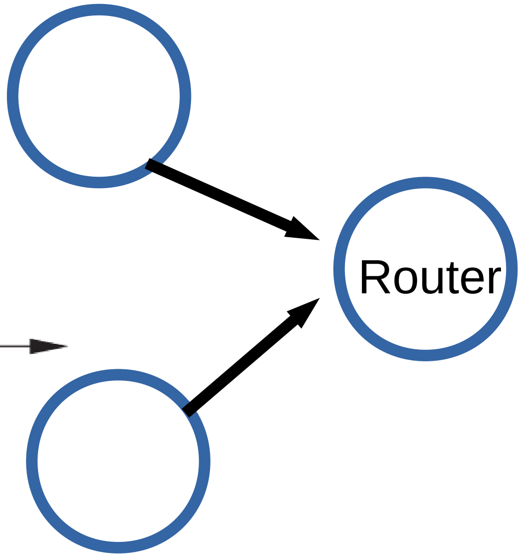
But, what if the flows traverse different paths?

Open problem, often determined by economics

Queuing Disciplines

Simplest – FIFO and drop tail





(a) FIFO queuing; (b) tail drop at a FIFO queue.

What are the problems?

Defining Fairness: Flows

“fair” to whom?   – Should be Fair to a Flow

What is a flow?

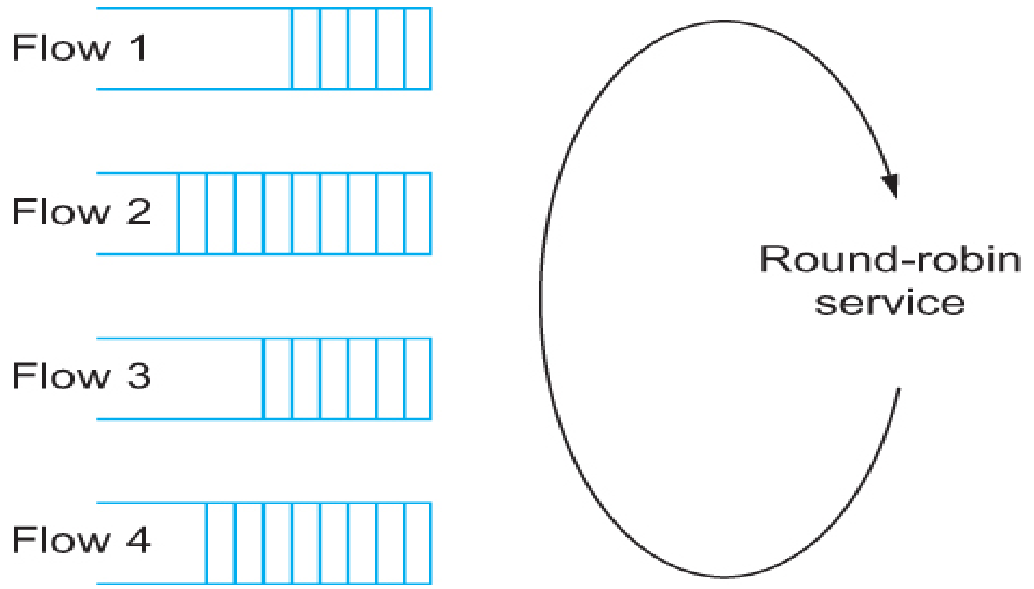
Combination of <Src IP, Src Port, Dst IP, Dst Port>

Fair Queuing

- Fair Queuing
  - FIFO does not discriminate between different traffic sources, or
  - it does not separate packets according to the flow to which they belong.
  - Fair queuing (FQ) maintains a separate queue for each flow

Queuing Disciplines

- Fair Queuing



Round-robin service of four flows at a router

Min Max Fair queuing

- Assume ***n*** clients
- Channel capacity **C**
- Give **c/n** to each client
  - If C1 does not want c/n
  - Divide the excess capacity equally among others
  - So everyone else gets c/n + (c/n – c1)/n-1
  - Repeat for C2 and others

TCP Congestion Control

- Each source determines available capacity
- Max many packets is allowed to have in transit - window
- Congestion window = # of unacked bytes
- MaxSendWindow = min(congestion window,   receiver window)

- How do you change congestion window?
  - Decrease on losing a packet (back off)
  - Increase on successful send

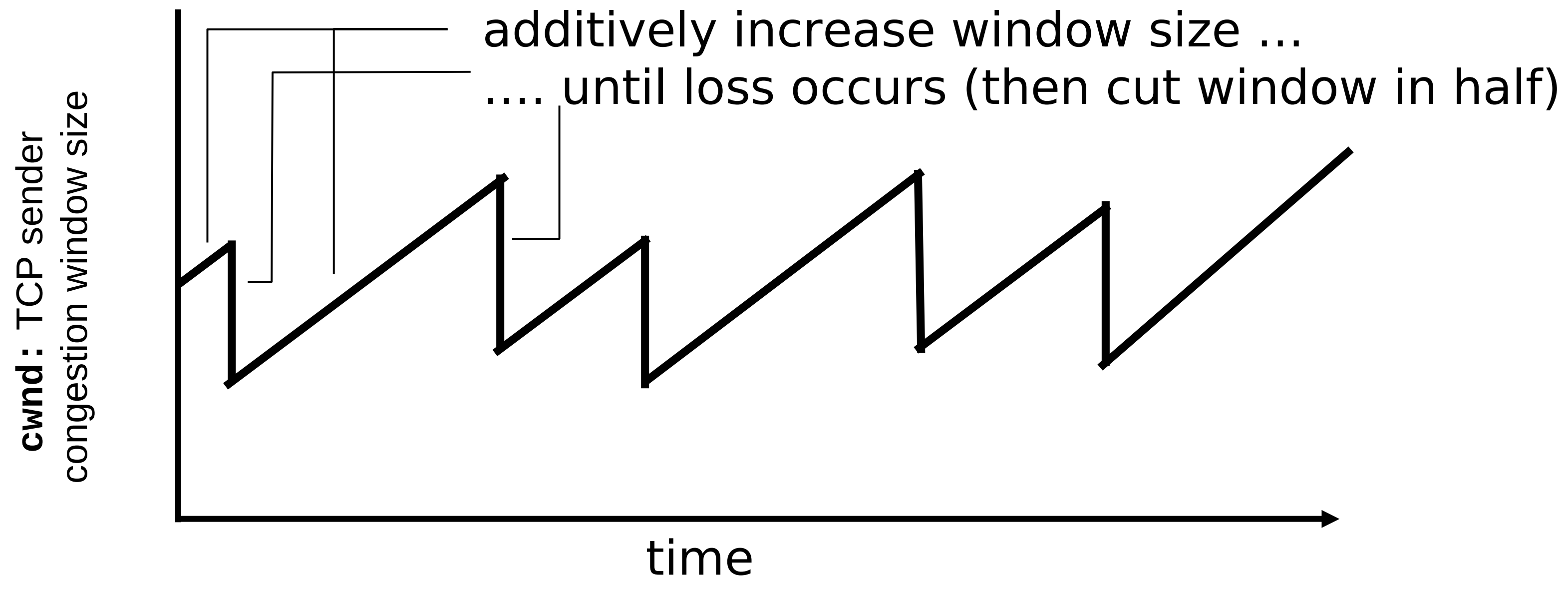
How much to increase and decrease?

- Additive Increase, Multiplicative Decrease (AIMD)

How much to increase and decrease?

- *approach:* sender increases transmission rate (window size), probing for usable bandwidth, until loss occurs
  - *additive increase:* increase **cwnd** by 1 MSS every RTT until loss detected
  - *multiplicative decrease*: cut **cwnd** in half after loss

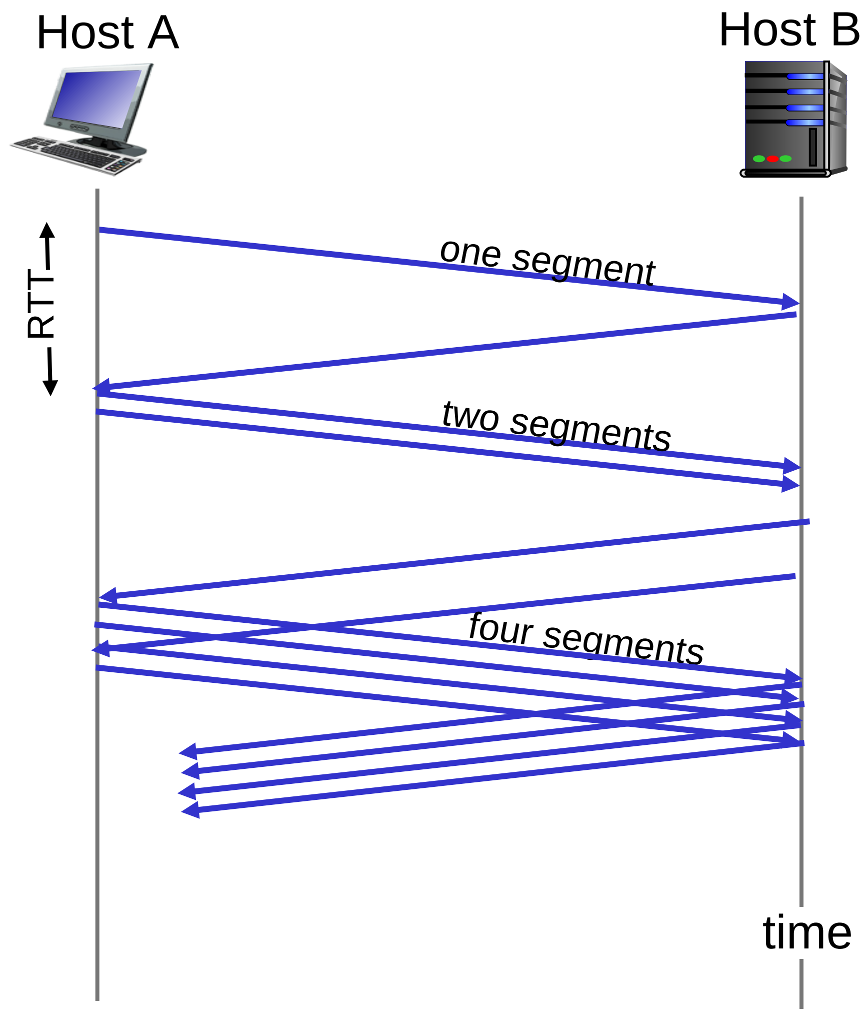
AIMD saw tooth behavior: probing for bandwidth



TCP Slow Start

- when connection begins, increase rate exponentially until first loss event:
  - initially **cwnd** = 1 MSS
  - double **cwnd** every RTT
  - done by incrementing **cwnd** for every ACK received

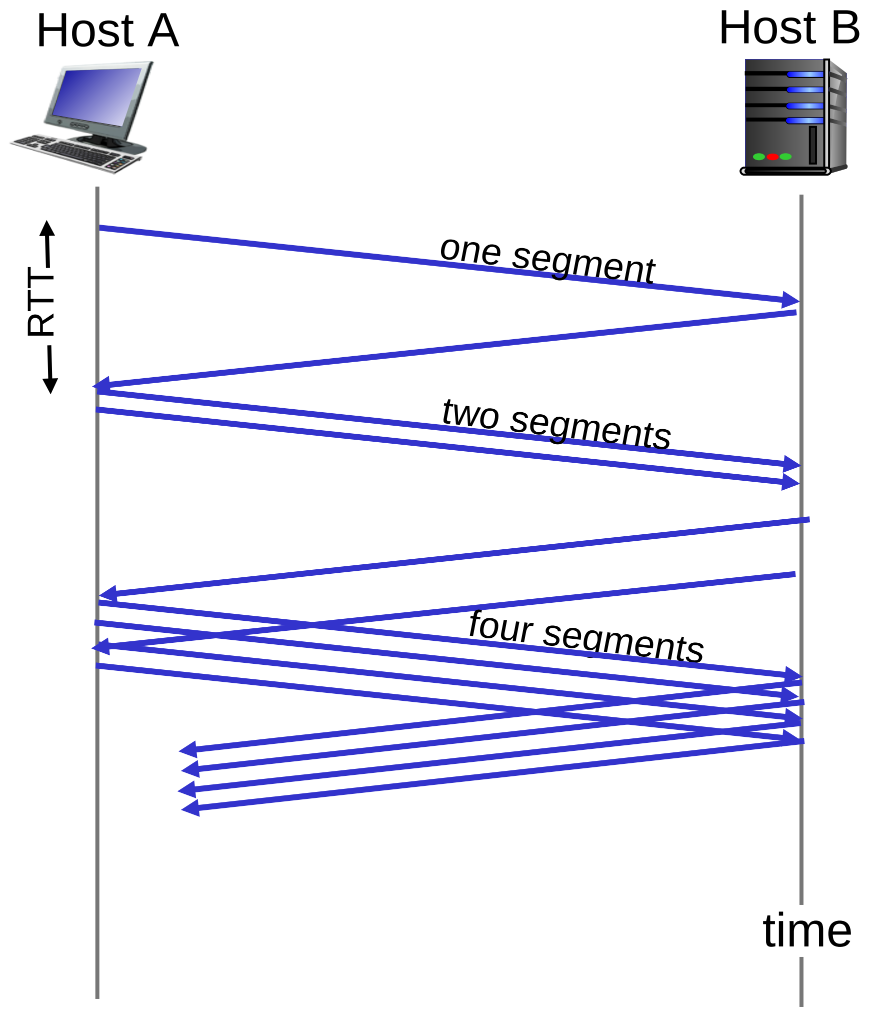
- *summary:* initial rate is slow but ramps up exponentially fast



TCP Slow Start

Why not start with a large window?

Why not increase one by one?



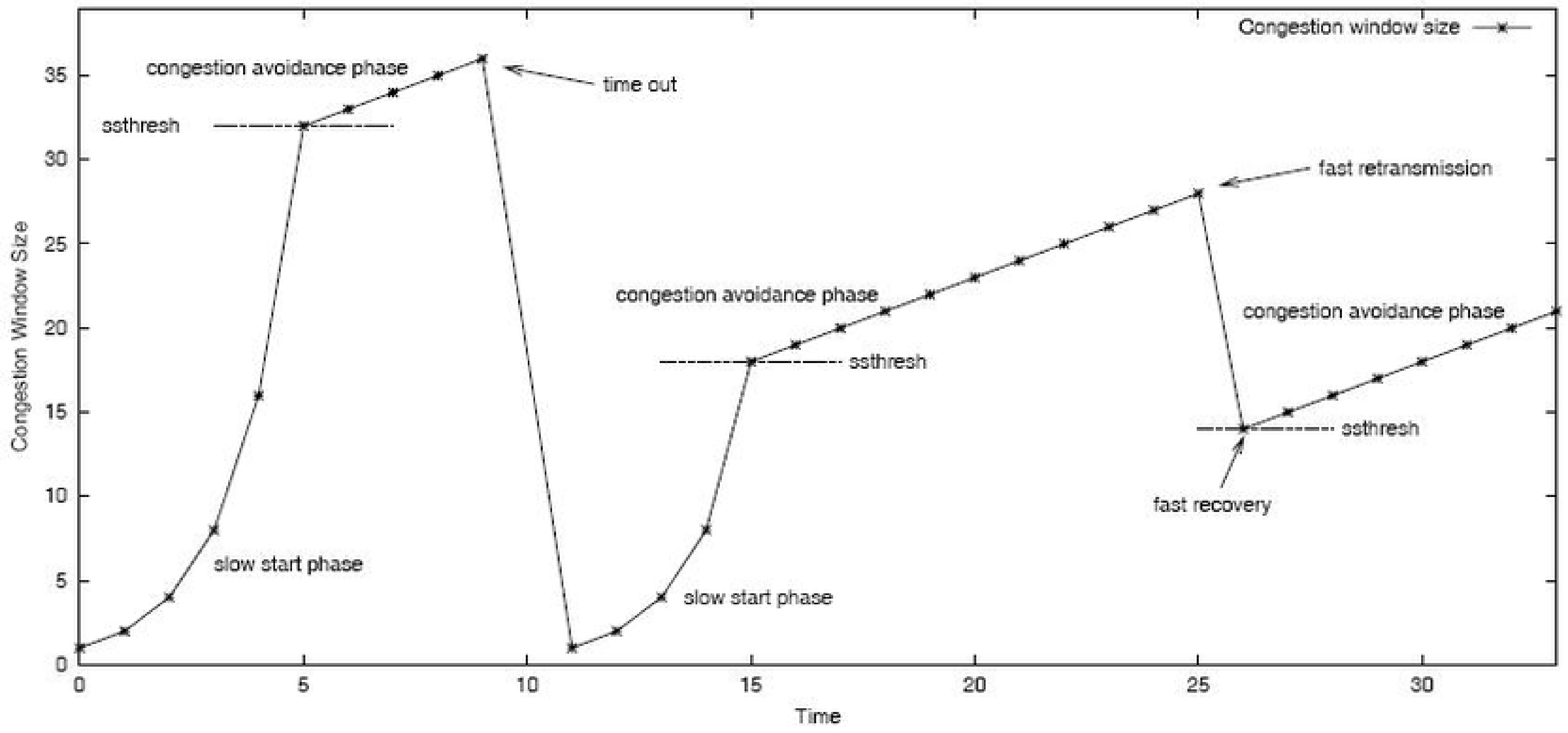
TCP: detecting, reacting to loss

- loss indicated by timeout:
  - **cwnd** set to 1 MSS;
  - window then grows exponentially (as in slow start) to threshold, then grows linearly
- loss indicated by 3 duplicate ACKs: TCP RENO
  - dup ACKs indicate network capable of delivering some segments
  - **cwnd** is cut in half window then grows linearly
- TCP Tahoe always sets **cwnd** to 1 (timeout or 3 duplicate acks)

TCP:Two types of loss

- Triple duplicate ack
  - Do a multiplicative decrease, keep going
- Timeout
  - Reset CWND to 1
  - Take advantage of

TCP Slow Start and congestion avoidance



How to set ssthresh?

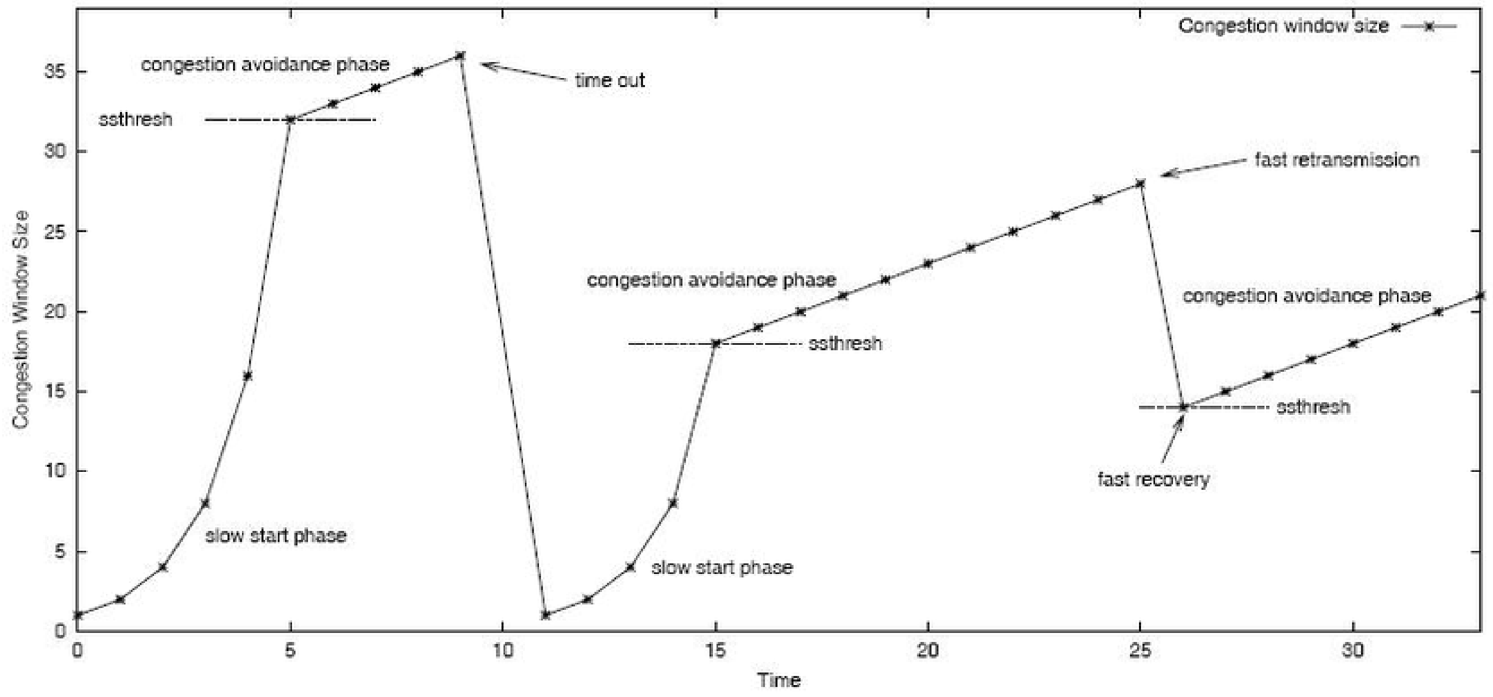
Initially – Randomly high

Later – adjusted as congestion happens

https://www.researchgate.net/figure/3-TCP-slow-start-phase-and-congestion-avoidance-phase_fig3_225731524

TCP Congestion Summary

CWND < Threshold → Slow Start, Exponential increase
CWND > Threshold → Congestion Avoidance, Linear increase
Triple Duplicate ACK → Threshold = CWND/2, CWND = CWND/2
Timeout → Threshold = CWND/2, CWDN = 1 (or 3)



TCP Throughput

TCP average throughput as a function of window size and RTT?

Ignore slow start, assume long TCP flow

Let W be the window size

Throughput = W/RTT

After loss, throughput = W/2*RTT

Average throughput = 0.75W/RTT

Problems with Fast Links

Consider the high speed link:

9000 byte segments

100ms RTT

100Gbps/second throughput

Throughput = 0.75W/RTT

So, WindowSize (w) = Throughput * RTT / 0.75

W = 1,481,481,444 segments

Problems with Fast Links

TCP assumes all losses are due to congestion

Throughput = (1.22*MSS)*(RTT/sqrt(Loss))

What is the loss rate to maximize 100Gbps pipe with
9000 bytes segments and 100ms RTT? Hint – must be very very low

https://www.switch.ch/network/tools/tcp_throughput/