Advanced Computer Networks

TCP and Fair Queuing

Introduction to TCP

- Communication abstraction:
  - Reliable
  - Ordered
  - Point-to-point
  - Byte-stream
  - Full duplex
  - Flow and congestion controlled

- Protocol implemented entirely at the ends
  - Fate sharing

- Sliding window with cumulative acks
  - Ack field contains last in-order packet received
  - Duplicate acks sent when out-of-order packet received

Key Things You Should Know Already

- Port numbers
- TCP/UDP checksum
- Sliding window flow control
  - Sequence numbers
- TCP connection setup
- TCP reliability
  - Timeout
  - Data-driven

TCP Congestion Control

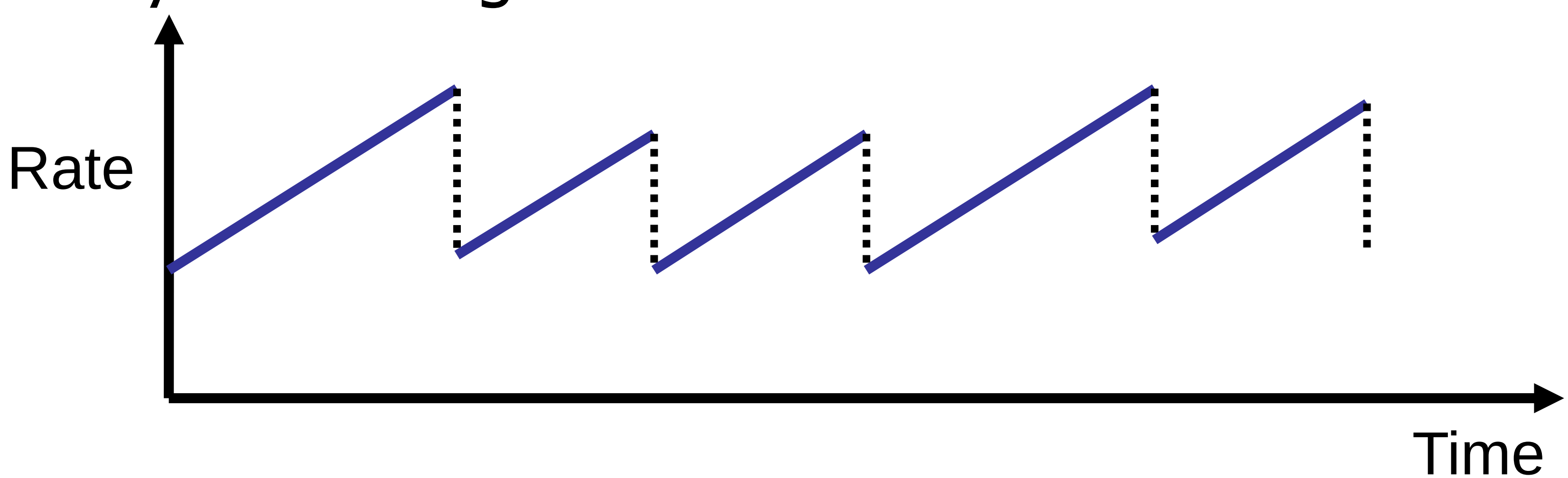
- Changes to TCP motivated by ARPANET congestion collapse
- Basic principles
  - Additive increase/multiplicative decrease (AIMD)
  - Packet conservation
  - Reaching steady state quickly
  - ACK clocking

TCP Congestion Control - Solutions

- Reaching equilibrium
  - Slow start
- Eliminates spurious retransmissions
  - Accurate RTO estimation
  - Fast retransmit
- Adapting to resource availability
  - Congestion avoidance

AIMD

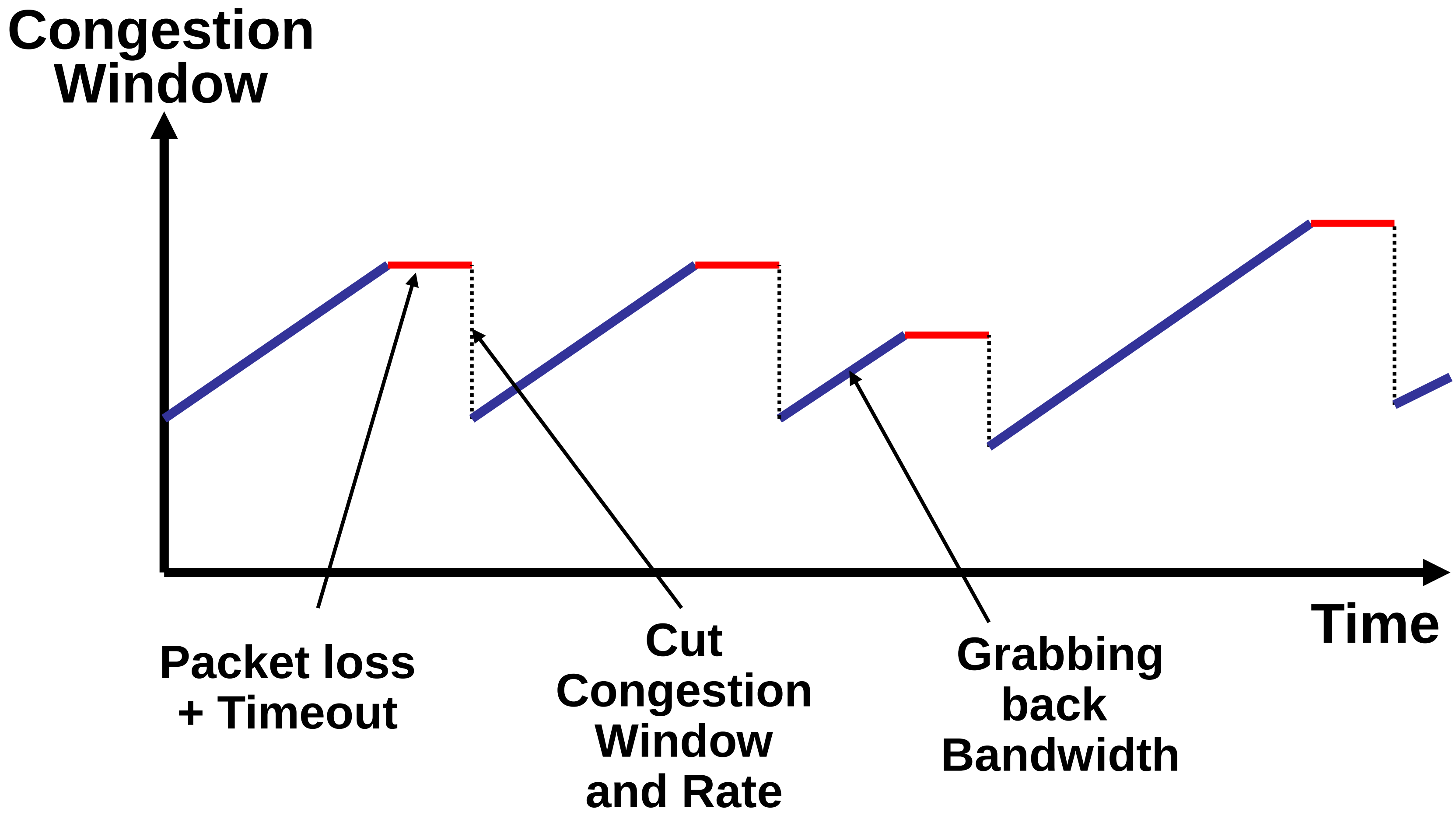
- Distributed, fair and efficient
- Packet loss is seen as sign of congestion and results in a multiplicative rate decrease
  - Factor of 2
- TCP periodically probes for available bandwidth by increasing its rate



Congestion Avoidance

- If loss occurs when cwnd = W
  - Network can handle 0.5W ~ W segments
  - Set cwnd to 0.5W (multiplicative decrease)
- Upon receiving ACK
  - Increase cwnd by (1 packet)/cwnd
    - What is 1 packet? → 1 MSS worth of bytes
    - After cwnd packets have passed by → approximately increase of 1 MSS
- Implements AIMD

Congestion Avoidance Behavior

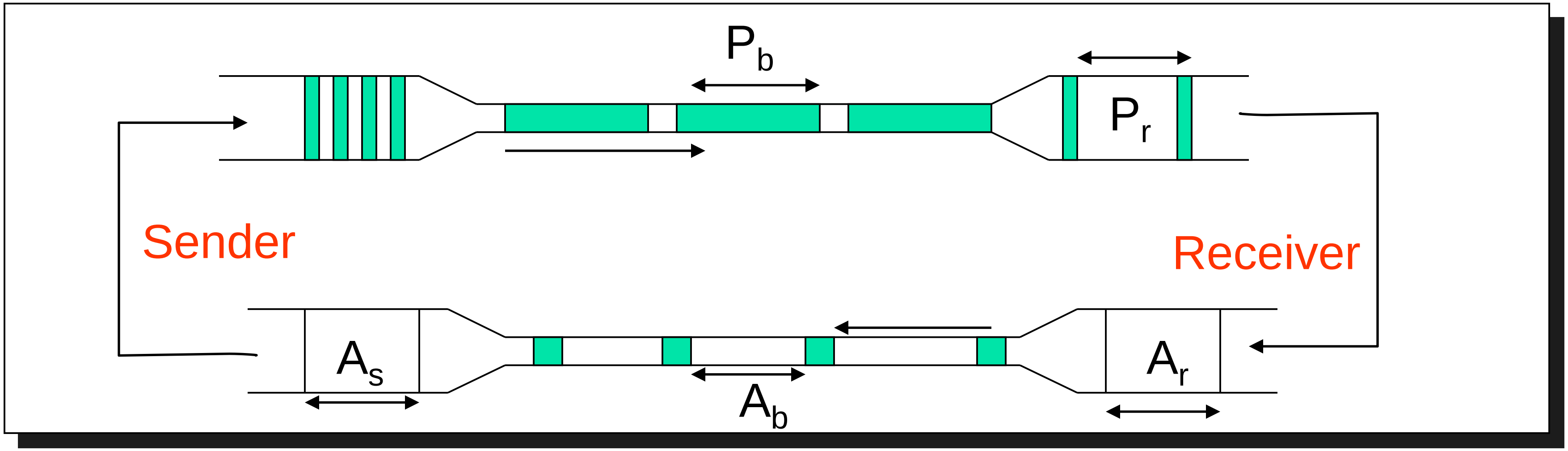


Packet Conservation

- At equilibrium, inject packet into network only when one is removed
  - Sliding window and not rate controlled
  - But still need to avoid sending burst of packets → would overflow links
    - Need to carefully pace out packets
    - Helps provide stability
- Need to eliminate spurious retransmissions
  - Accurate RTO estimation
  - Better loss recovery techniques (e.g. fast retransmit)

TCP Packet Pacing

- Congestion window helps to “pace” the transmission of data packets
- In steady state, a packet is sent when an ack is received
  - Data transmission remains smooth, once it is smooth
  - Self-clocking behavior

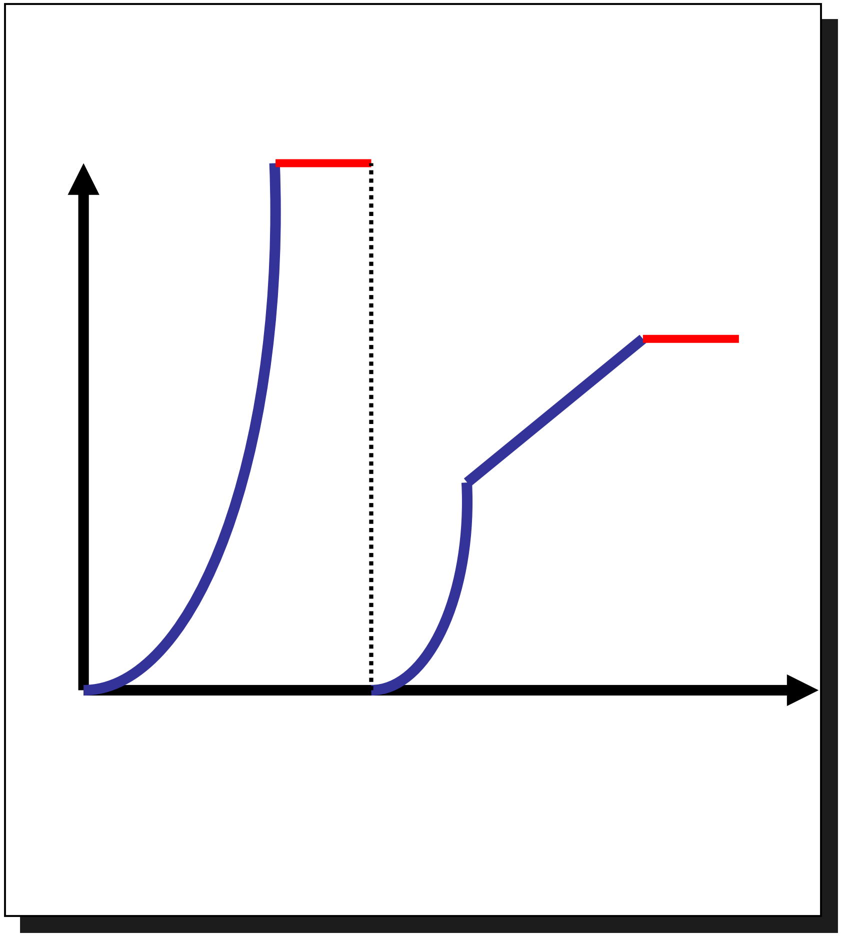


Reaching Steady State

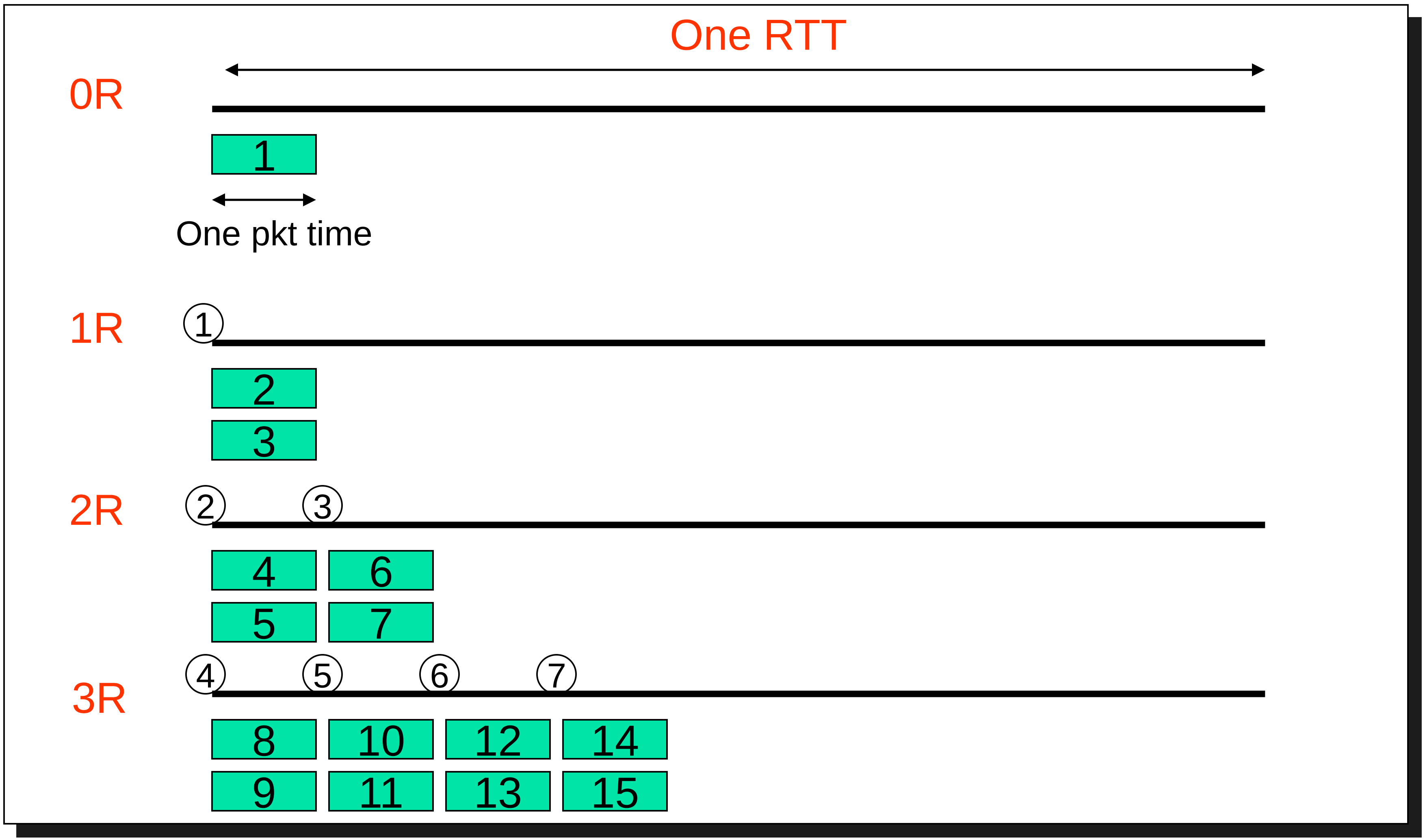
- Doing AIMD is fine in steady state but slow…
- How does TCP know what is a good initial rate to start with?
  - Should work both for a CDPD (10s of Kbps or less) and for supercomputer links (10 Gbps and growing)
- Quick initial phase to help get up to speed (slow start)

Slow Start Packet Pacing

- How do we get this clocking behavior to start?
  - Initialize cwnd = 1
  - Upon receipt of every ack, cwnd = cwnd + 1
- Implications
  - Window actually increases to W in RTT * $\log_2(W)$
  - Can overshoot window and cause packet loss



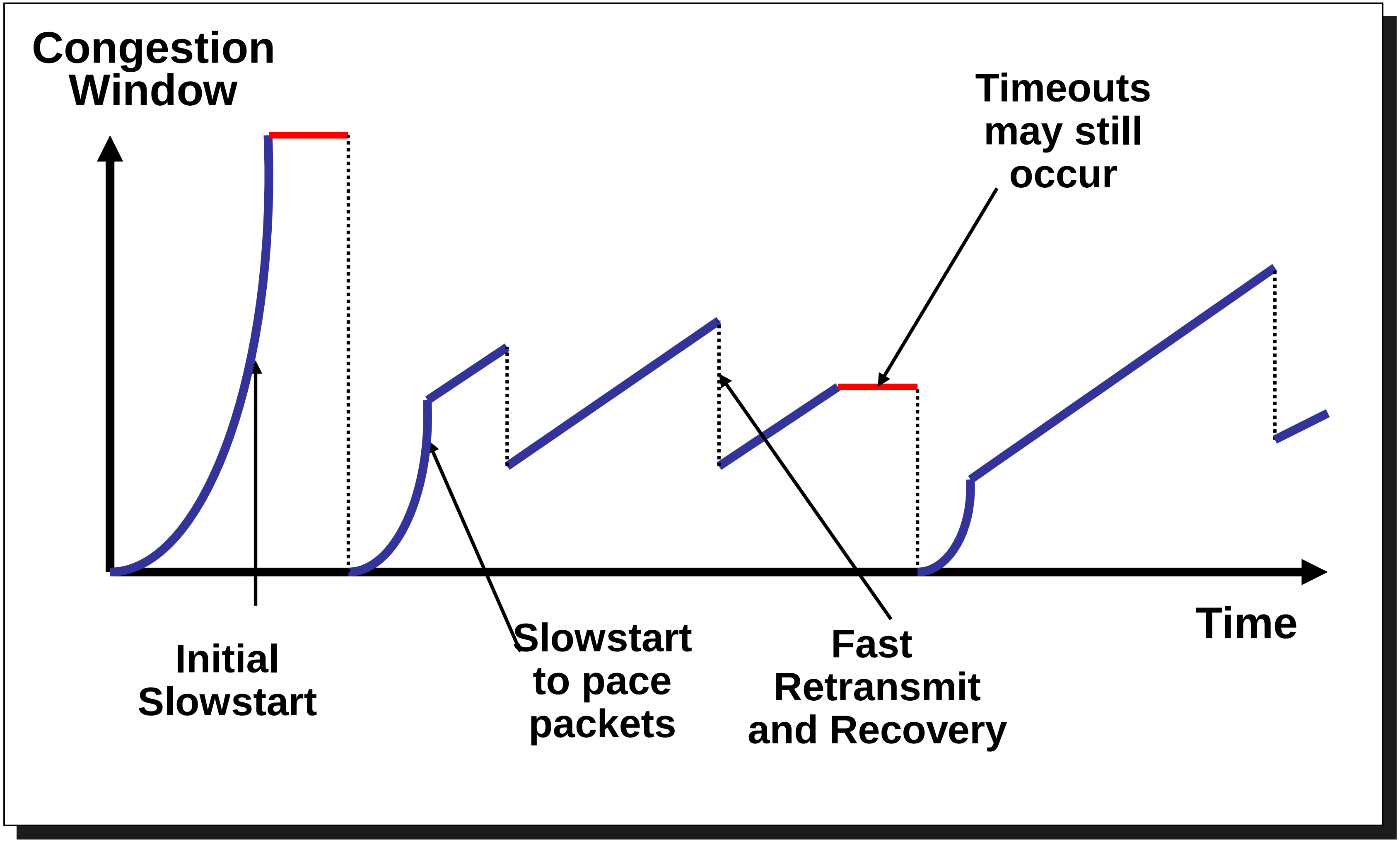
Slow Start Example



Return to Slow Start

- If packet is lost we lose our self clocking as well
  - Need to implement slow-start and congestion avoidance together
- When timeout occurs set ssthresh to 0.5w
  - If cwnd < ssthresh, use slow start
  - Else use congestion avoidance

TCP Saw Tooth Behavior



Changing Workloads

- New applications are changing the way TCP is used
- 1980's Internet
  - Telnet & FTP → long lived flows
  - Well behaved end hosts
  - Homogenous end host capabilities
  - Simple symmetric routing
- 2000's Internet
  - Web & more Web → large number of short xfers
  - Wild west – everyone is playing games to get bandwidth
  - Cell phones and toasters on the Internet
  - Policy routing
- How to accommodate new applications?

TCP Friendliness

- What does it mean to be TCP friendly?
  - TCP is not going away
  - Any new congestion control must compete with TCP flows
    - Should not clobber TCP flows and grab bulk of link
    - Should also be able to hold its own, i.e. grab its fair share, or it will never become popular
- How is this quantified/shown?
  - Has evolved into evaluating loss/throughput behavior
  - If it shows 1/sqrt(p) behavior it is ok

TCP Friendly Rate Control (TFRC)

- Equation 1 – real TCP response

$$T = \frac{s}{R\sqrt{\frac{2p}{3}} + t_{RTO}(3\sqrt{\frac{3p}{8}})p(1+32p^2)}$$

  - 1^{st} term corresponds to simple derivation
  - 2^{nd} term corresponds to more complicated timeout behavior
    - Is critical in situations with > 5% loss rates → where timeouts occur frequently
- Key parameters
  - RTO
  - RTT
  - Loss rate

TCP Performance

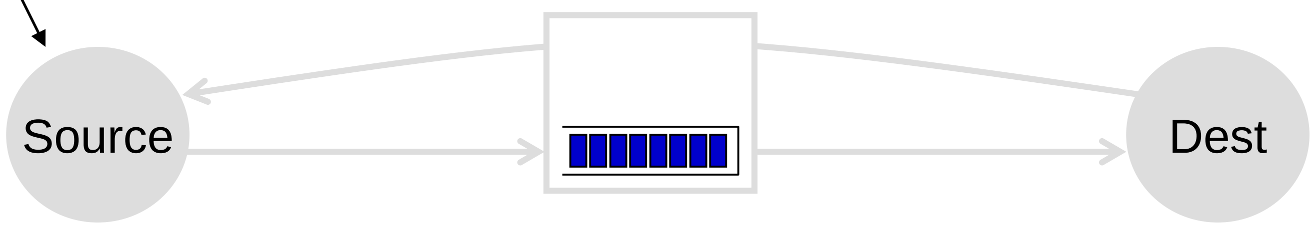
- Can TCP saturate a link?
- Congestion control
  - Increase utilization until… link becomes congested
  - React by decreasing window by 50%
  - Window is proportional to rate * RTT
- Doesn’t this mean that the network oscillates between 50 and 100% utilization?
  - <u>Average utilization = 75%??</u>
  - No…this is *not* right!

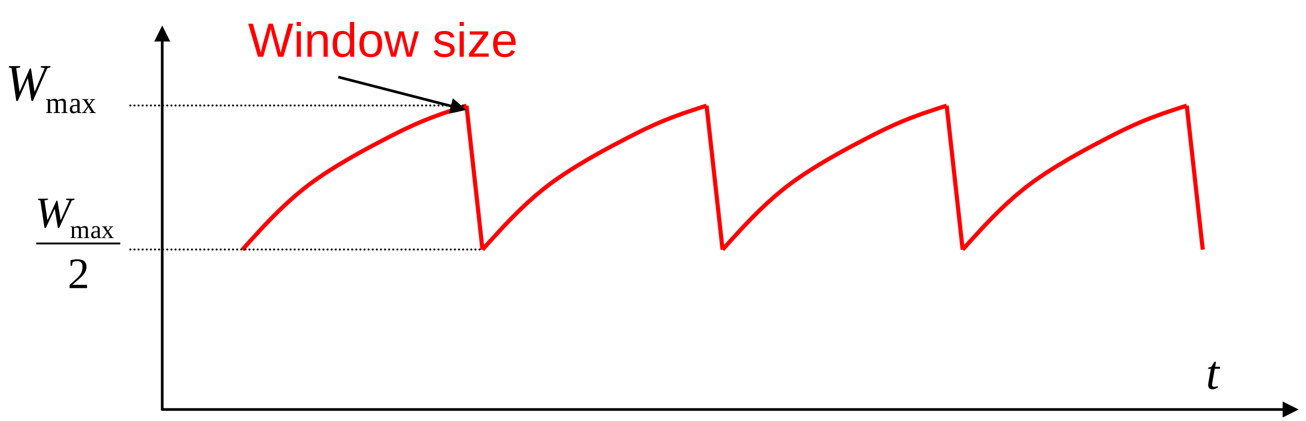
TCP Congestion Control

Rule for adjusting W

- If an ACK is received: W ← W+1/W
- If a packet is lost: W ← W/2

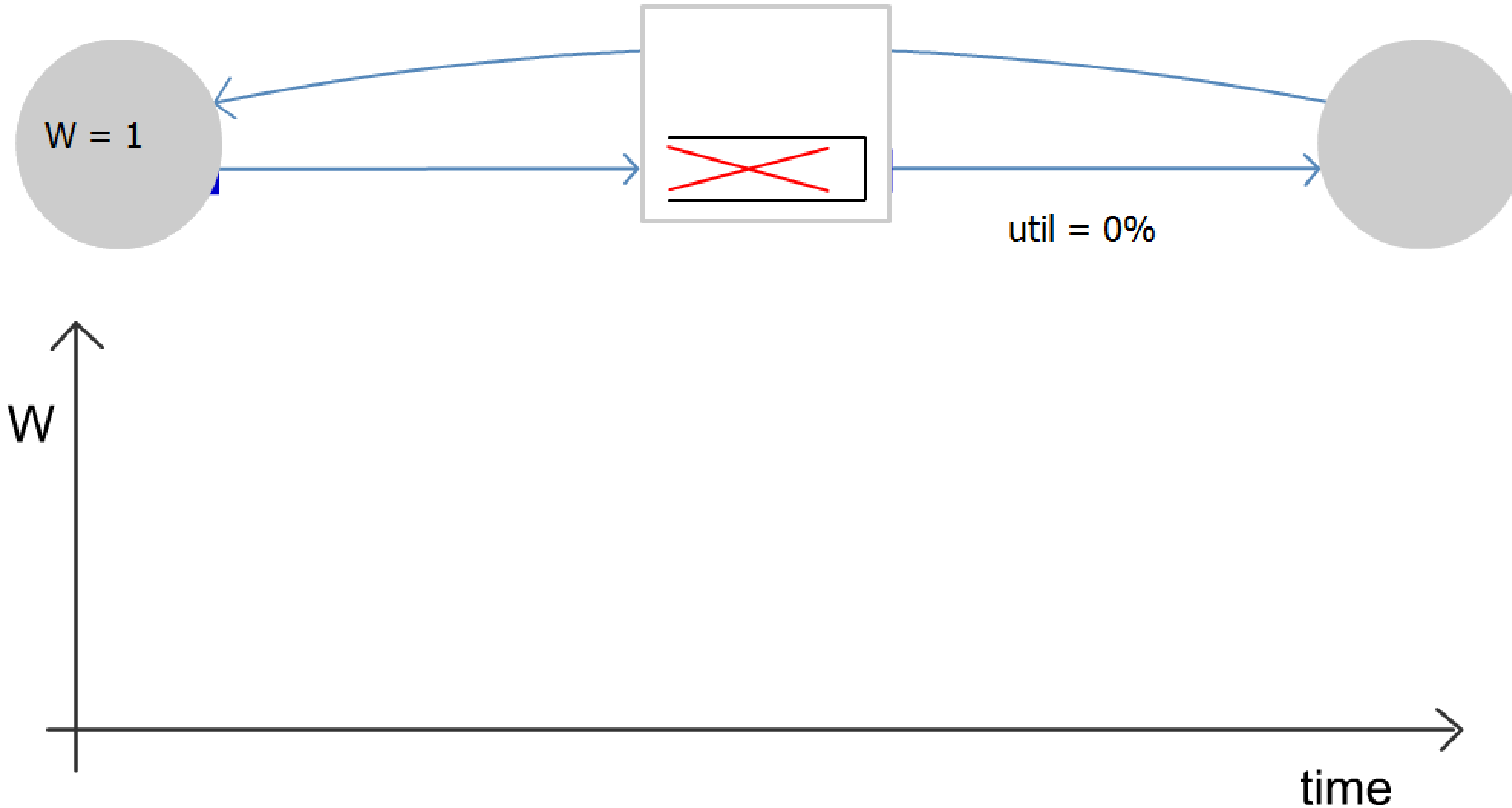
Only W packets may be outstanding



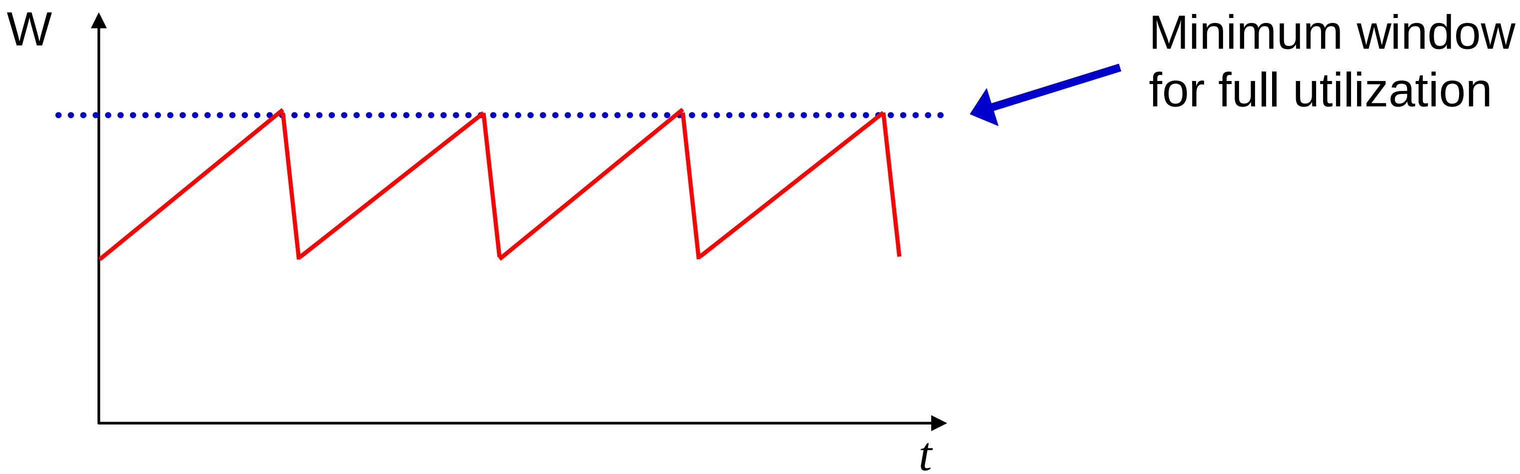


Single TCP Flow

Router without buffers



Summary Unbuffered Link



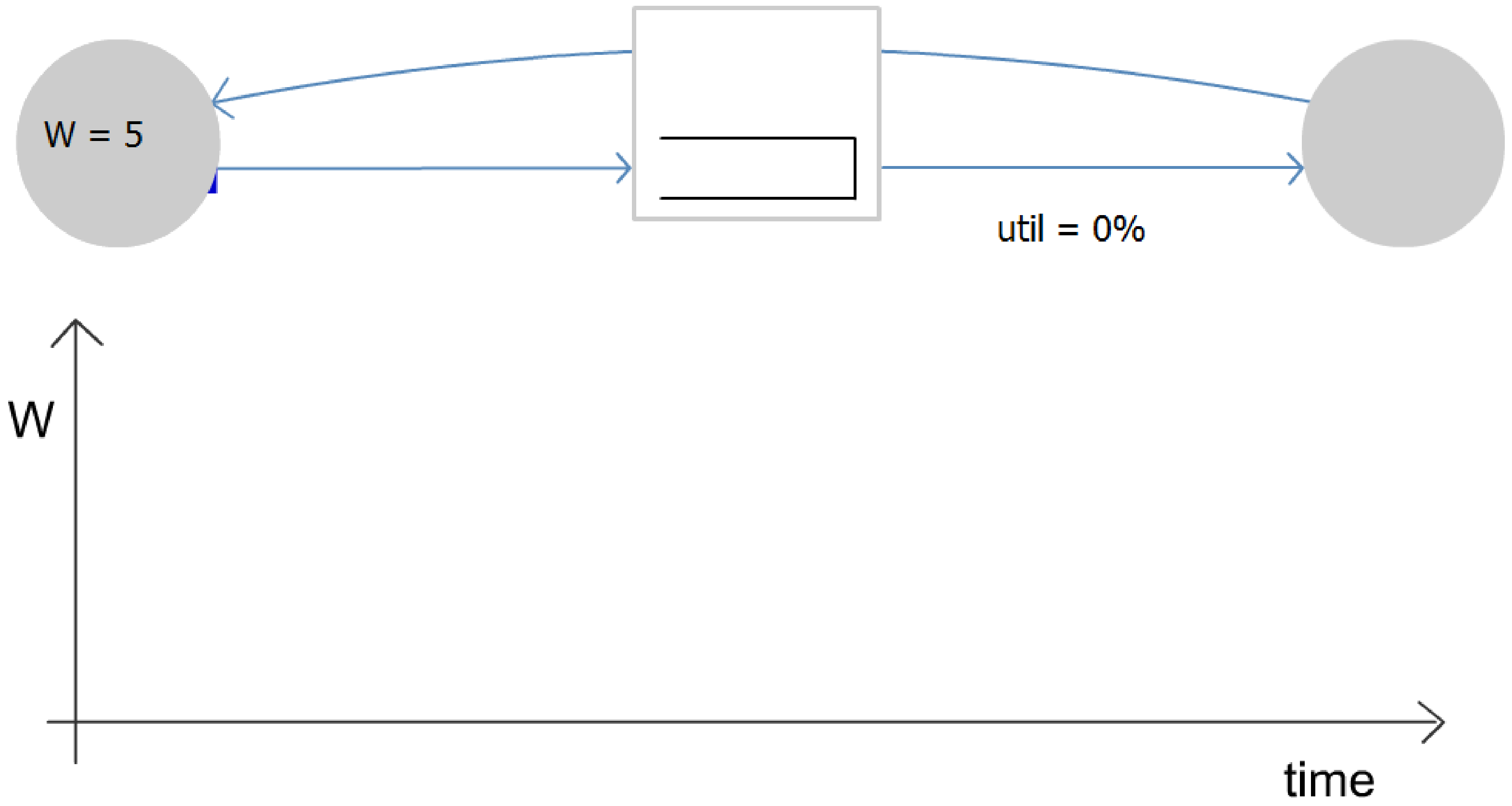
- The router can’t fully utilize the link
  - If the window is too small, link is not full
  - If the link is full, next window increase causes drop
  - With no buffer it still achieves 75% utilization

TCP Performance

- In the real world, router queues play important role
  - Window is proportional to rate * RTT
    - But, RTT changes as well the window
  - Window to fill links = propagation RTT * bottleneck bandwidth
    - If window is larger, packets sit in queue on bottleneck link

Single TCP Flow

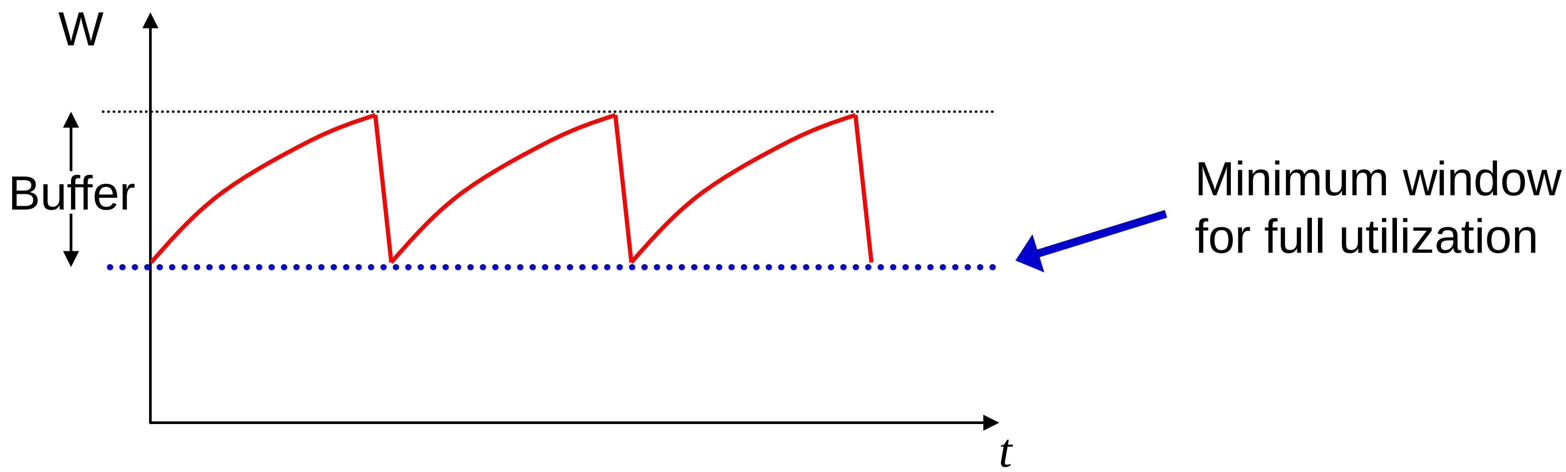
Router with large enough buffers for full link utilization



TCP Performance with Queue

- If we have a large router queue → can get <u>100% utilization</u>
  - But, router queues can cause large delays
- How big does the queue need to be?
  - Windows vary from W → W/2
    - Must make sure that link is always full
    - W/2 > RTT * BW
    - W = RTT * BW + Qsize
    - Therefore, Qsize > RTT * BW
  - <u>Ensures 100% utilization</u>
  - Delay?
    - Varies between RTT and 2 * RTT

Summary Buffered Link



- With sufficient buffering we achieve full link utilization
  - The window is always above the critical threshold
  - Buffer absorbs changes in window size
    - Buffer Size = Height of TCP Sawtooth
    - Minimum buffer size needed is 2T*C, where T=RTT, C=BW
  - This is the origin of the rule-of-thumb

Queuing Disciplines

- Each router must implement some queuing discipline
- Queuing allocates both bandwidth and buffer space:
  - Bandwidth: which packet to serve (transmit) next
  - Buffer space: which packet to drop next (when required)
- Queuing also affects latency

Typical Internet Queuing

- FIFO + drop-tail
  - Simplest choice
  - Used widely in the Internet
- FIFO (first-in-first-out)
  - Implies single class of traffic
- Drop-tail
  - Arriving packets get dropped when queue is full regardless of flow or importance
- Important distinction:
  - FIFO: scheduling discipline
  - Drop-tail: drop policy