EECS 489
Computer Networks

Winter 2024

Mosharaf Chowdhury

Material with thanks to Aditya Akella, Sugih Jamin, Philip Levis, Sylvia Ratnasamy, Peter Steenkiste, and many other colleagues.

Agenda

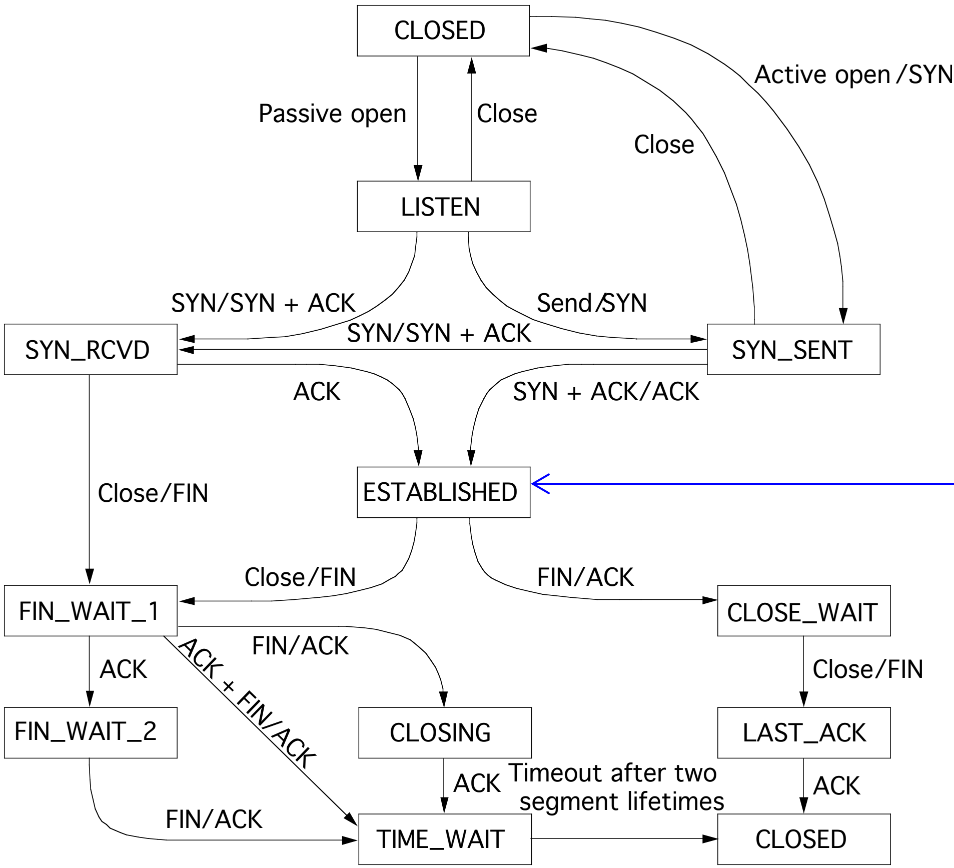
- TCP flow control
- TCP congestion control

RFC 675 ($4.3.2)

```
When establishing a connection, the state of the TCP is represented
by 3 bits --

   S1 S2 R

   S1 = 1 -- SYN sent

   S2 = 1 -- My SYN verified

   R = 1 -- SYN received
```

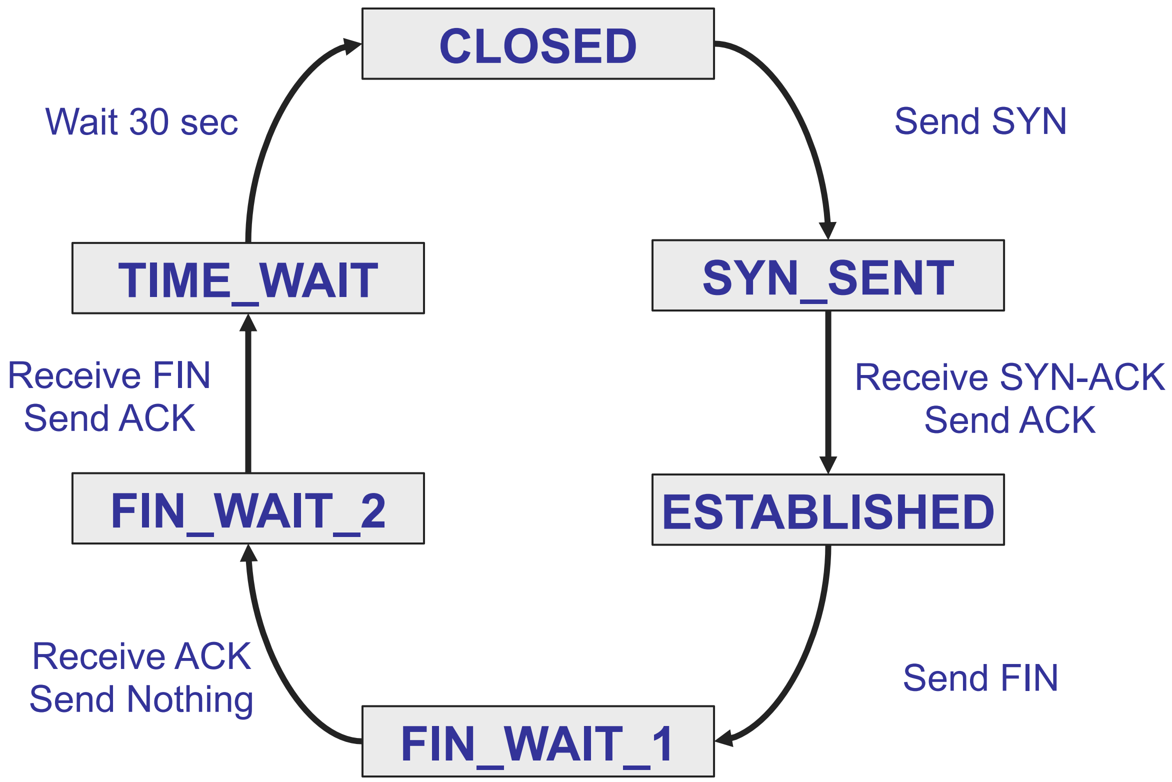
```
The "three way handshake" now looks like --

            A                                               B
    ------------                                     ------------
    S1 S2 R                                               S1 S2 R

    0  0  0                                               0  0  0

           --> <SEQ x><SYN>                          -->

    1  0  0                                               0  0  1

           <-- <SEQ y><SYN, ACK x+1>                 <--

    1  1  1                                               1  0  1

           --> <SEQ x+1><ACK y+1>(DATA OCTETS)       -->

    1  1  1                                               1  1  1
```

TCP state transitions

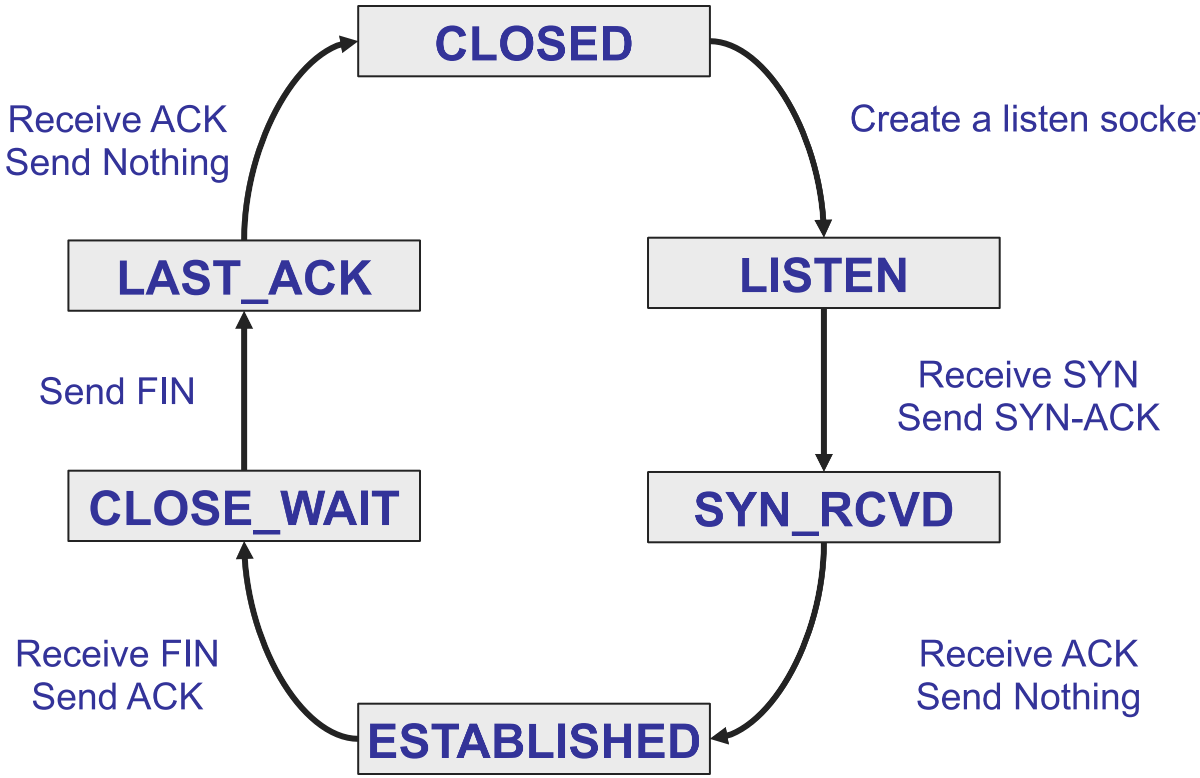


Data, ACK exchanges are in here

TCP client lifecycle



TCP server lifecycle

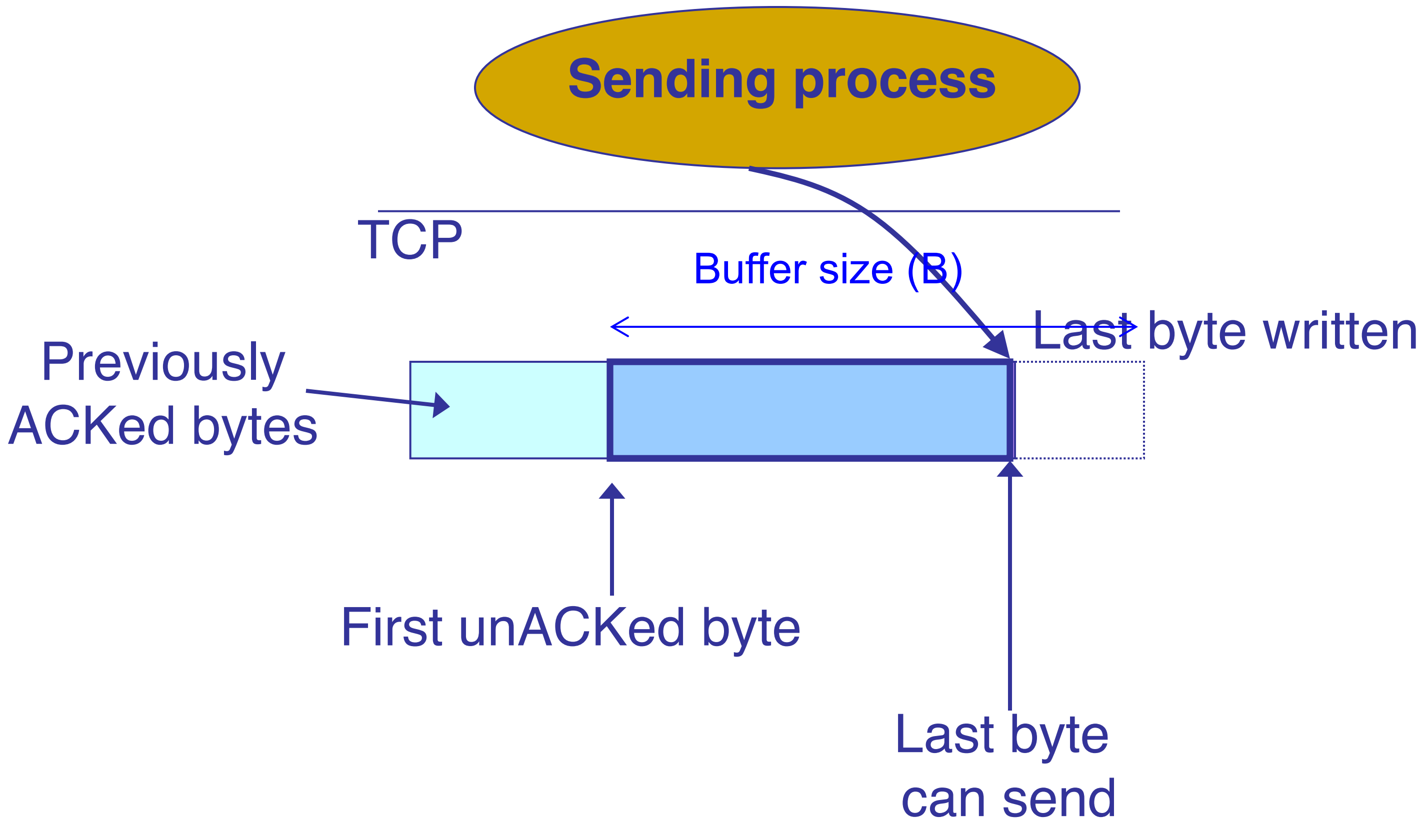


TCP FLOW CONTROL

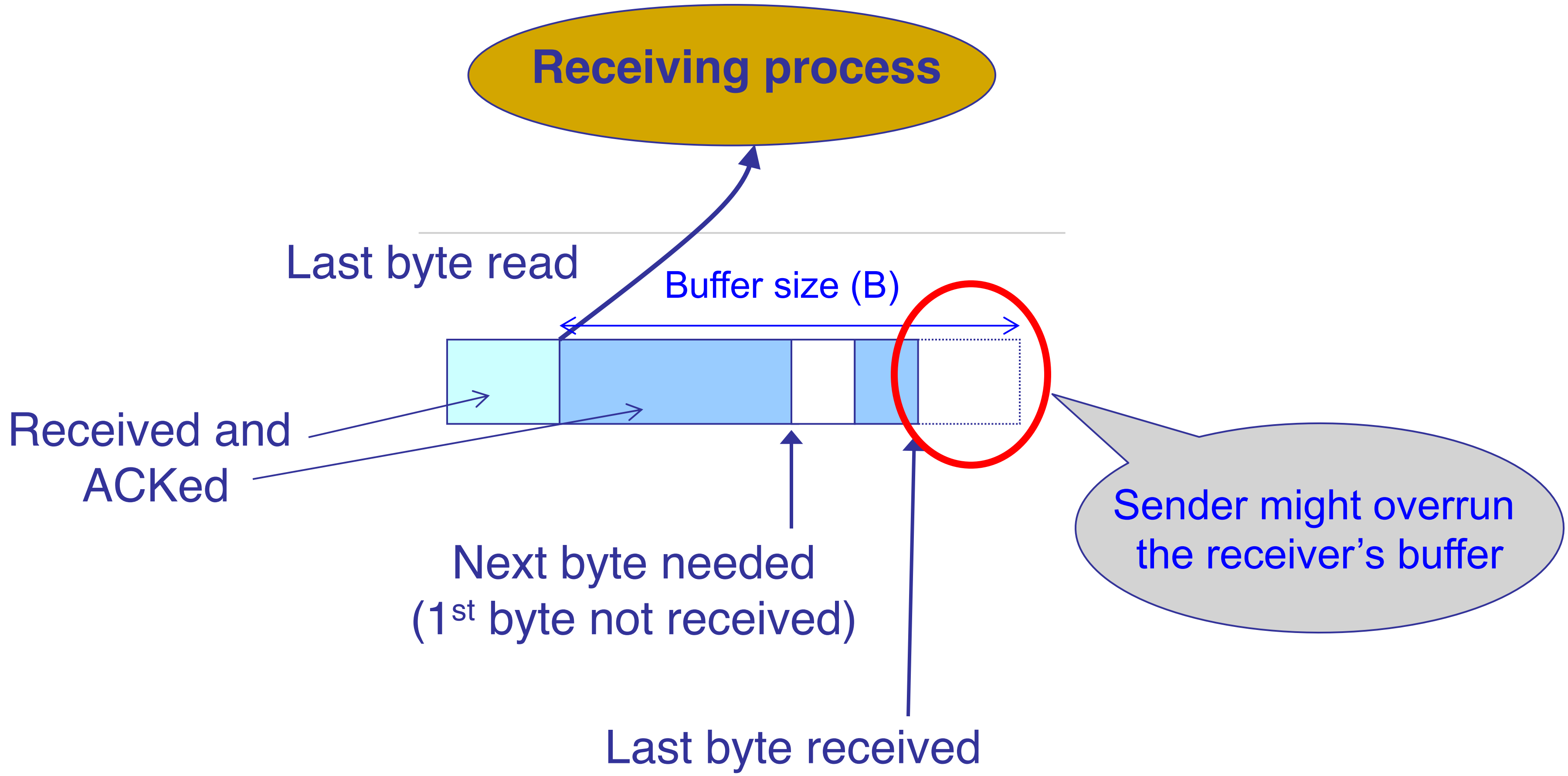
Recap: Sliding window

- Both sender and receiver maintain a window
- Left edge of window:
  - Sender: beginning of unacknowledged data
  - Receiver: beginning of expected data
    - » First "gap" in received data
    - » When sender gets ack, knows that receiver's window has moved
- Right edge: Left edge + constant
  - The constant is only limited by buffer size in the transport layer

Sliding window at sender



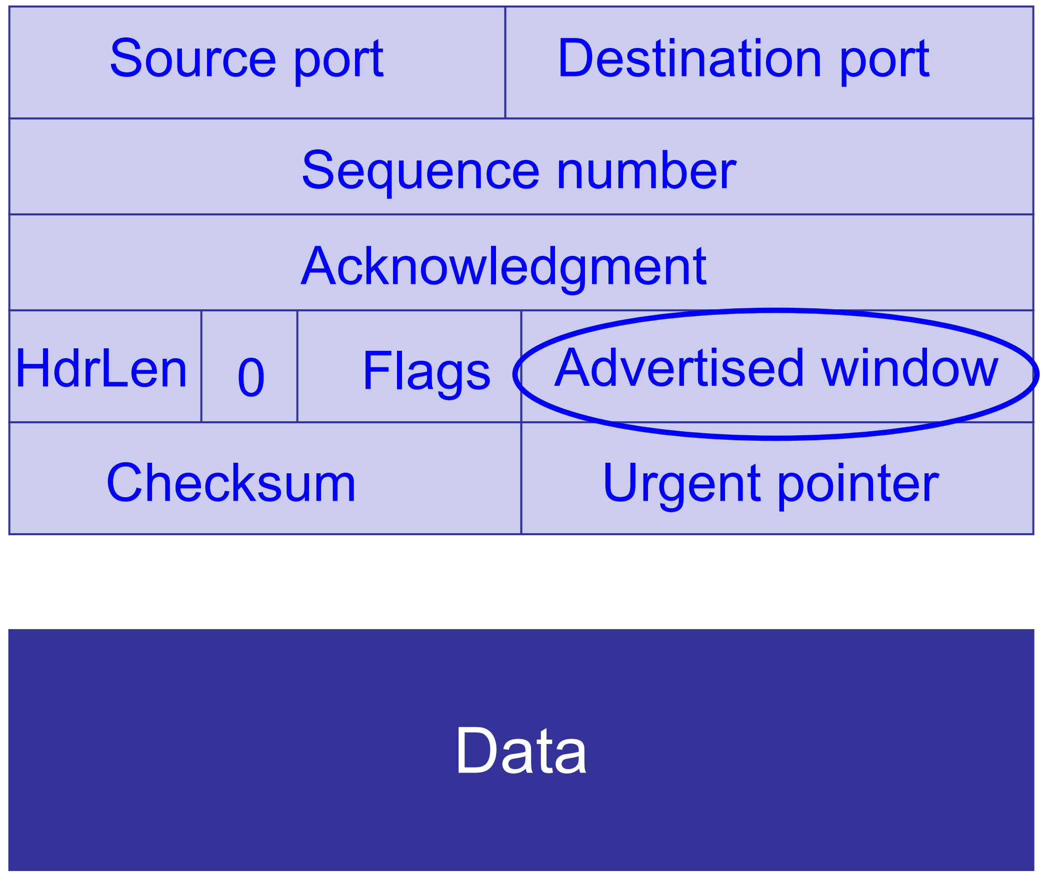
Sliding window at receiver



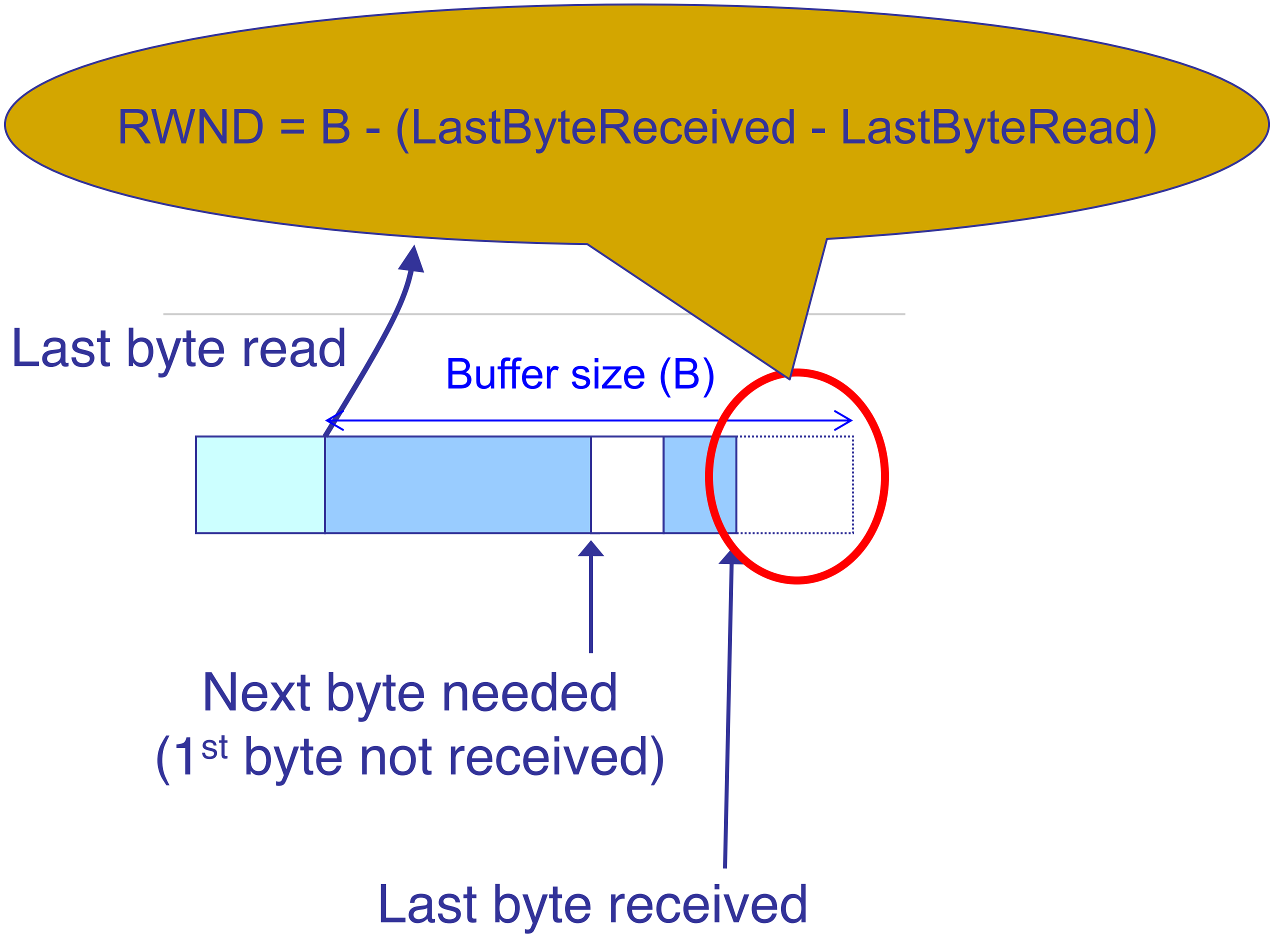
Solution: Advertised window (Flow Control)

- Receiver uses an “Advertised Window” (RWND) to prevent sender from overflowing its window
  - Receiver indicates value of RWND in ACKs
  - Sender ensures that the total number of bytes in flight <= RWND

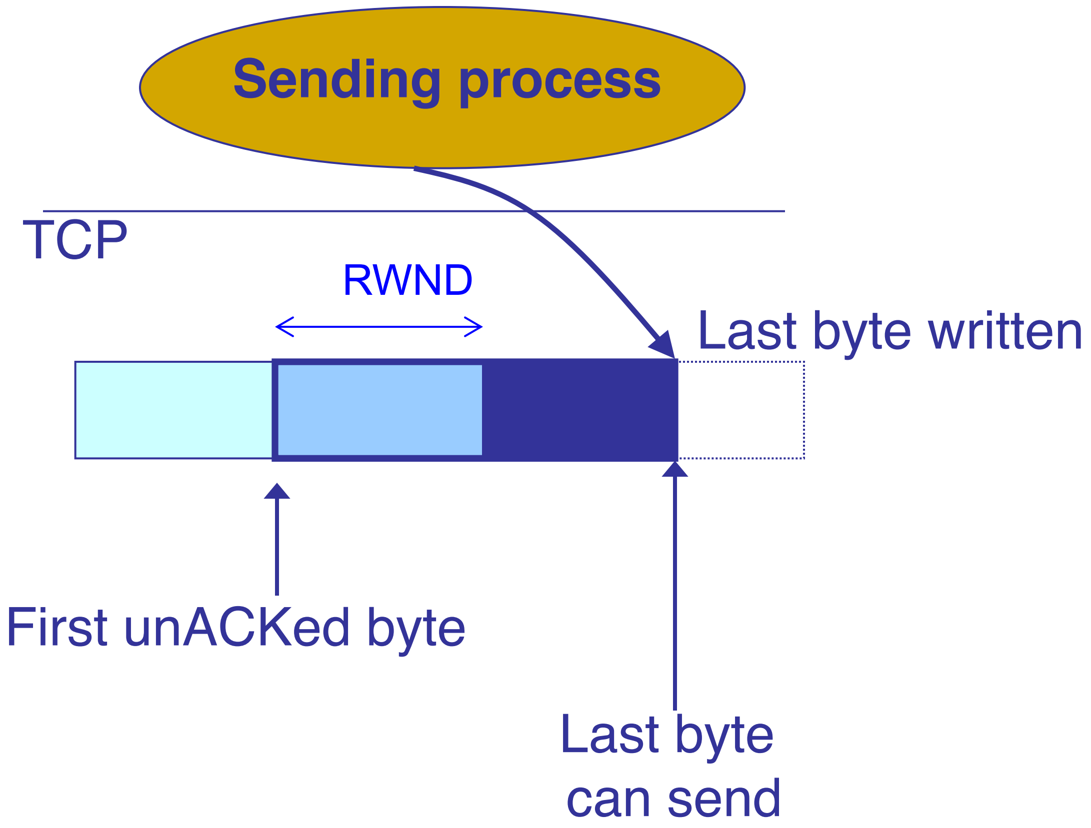
TCP header


Source port
Destination port
Sequence number
Acknowledgment
HdrLen
0
Flags
Advertised window
Checksum
Urgent pointer
Data

Sliding window at receiver



Sliding window at sender



Sliding window with flow control

- Sender: window advances when new data ACK'd
- Receiver: window advances as receiving process consumes data
- Receiver advertises to the sender where the receiver window currently ends ("righthand edge")
  - Sender agrees not to exceed this amount
- UDP does not have flow control
  - Data can be lost due to buffer overflow

Advertised window limits rate

- Sender can send no faster than RWND/RTT bytes/sec
- Receiver only advertises more space when it has consumed old arriving data
- What happens when RWND=0?
  - Sender keeps probing with one data bytes

- In original TCP design, that was the sole protocol mechanism controlling sender's rate
  - What's missing?

TCP CONGESTION CONTROL

What is congestion?

- If two packets arrive at a router at the same time
  - Router will transmit one and buffer/drop the other
- Internet traffic is bursty
  - Many packets can arrive close in time
  - Causes packet delays and drops

- Root cause: statistical multiplexing

Congestion collapse in 1980s

- Sending rate only limited by flow control
  - Dropped packets → senders (repeatedly!) retransmit
- Led to “congestion collapse” in Oct. 1986
  - Throughput on the NSF network dropped from 32Kbits/s to 40bits/sec
- “Fixed” by Van Jacobson’s development of TCP’s congestion control (CC) algorithms

Jacobson's fix to TCP

- Extend TCP's existing window-based protocol but adapt the window size in response to congestion
- A pragmatic and effective solution
  - Required no upgrades to routers or applications!
  - Patch of a few lines of code to TCP implementations
- Extensively researched and improved upon
  - Especially now with datacenters and cloud services

Key design considerations

- How do we know the network is congested?
  - Implicit and/or explicit signals from the network
- Who takes care of congestion?
  - End hosts (may receive some help from the network)
- How do we handle congestion?
  - Continuous adaptation

Three issues to consider

- Discovering the available (bottleneck) bandwidth
- Adjusting to variations in bandwidth
- Sharing bandwidth between flows

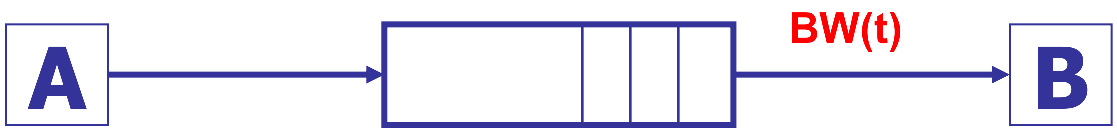
Abstract view

- Ignore internal structure of router and model it as a single queue for a particular input-output pair

Discovering available bandwidth

- Pick sending rate to match bottleneck bandwidth
  - Without any a priori knowledge
  - Could be gigabit link, could be a modem

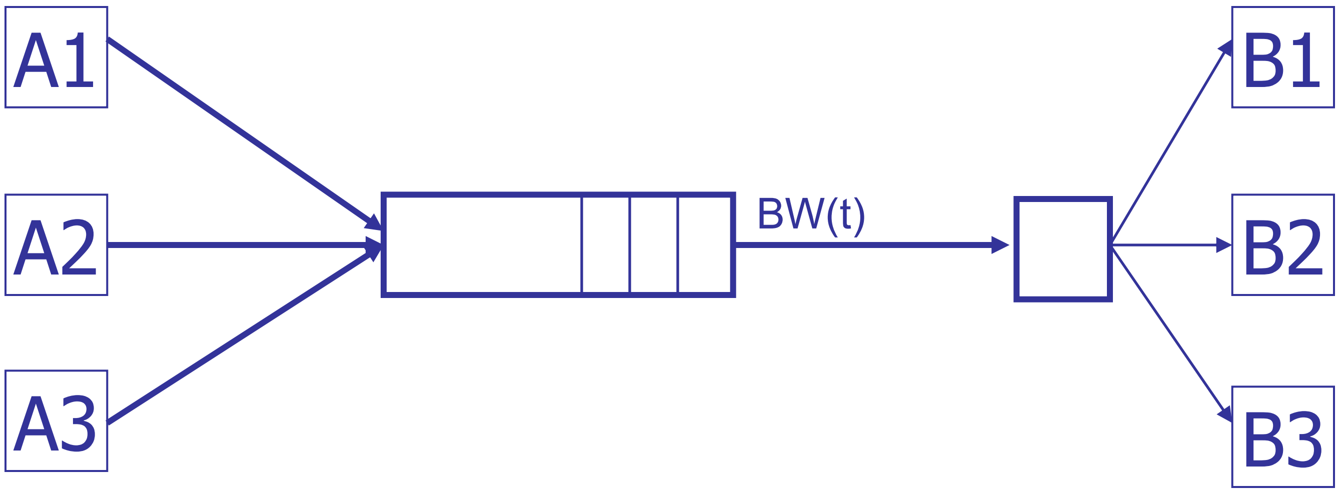
Adjusting to variations in bandwidth



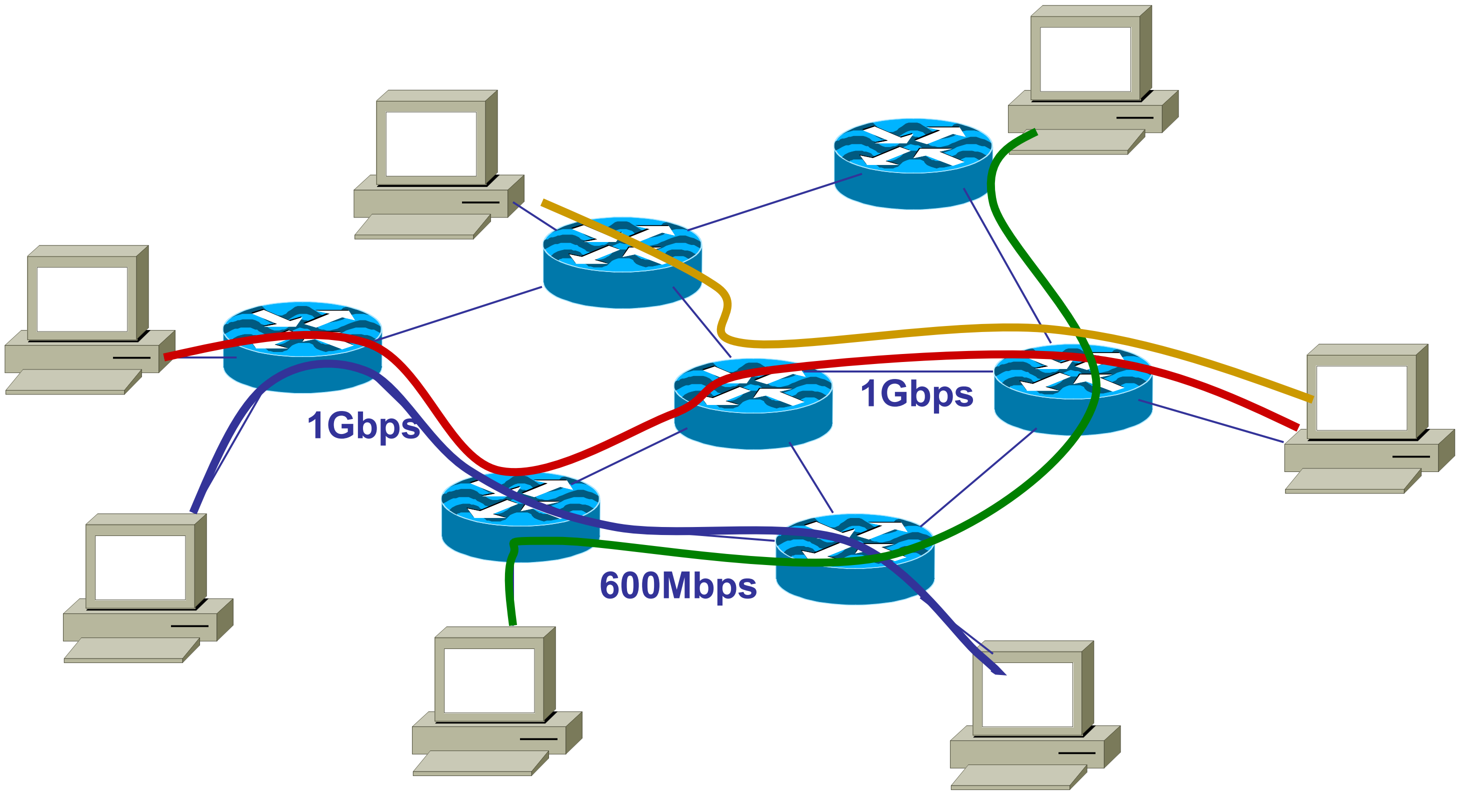
- Adjust rate to match instantaneous bandwidth
  - Assuming you have rough idea of bandwidth

Multiple flows and sharing bandwidth

- Two Issues:
  - Adjust total sending rate to match bandwidth
  - Allocation of bandwidth between flows



Reality



Congestion control is a resource allocation problem involving many flows, many links, and complicated global dynamics

Possible approaches

(0) Send without care

- Many packet drops

Possible approaches

(0) Send without care

(1) Reservations

- Pre-arrange bandwidth allocations
- Requires negotiation before sending packets
- Low utilization

Possible approaches

(0) Send without care

(1) Reservations

(2) Pricing

- Don’t drop packets for the high-bidders
- Requires payment model

Possible approaches

(0) Send without care

(1) Reservations

(2) Pricing

(3) Dynamic Adjustment

- Hosts infer level of congestion; adjust
- Network reports congestion level to hosts; hosts adjust
- Combinations of the above
- Simple to implement but suboptimal, messy dynamics

Possible approaches

(0) Send without care

(1) Reservations

(2) Pricing

(3) Dynamic Adjustment

- Generality of dynamic adjustment has proven to be very powerful
  - Doesn't presume business model, traffic characteristics, application requirements
  - But does assume good citizenship!

5-MINUTE BREAK!

TCP's approach in a nutshell

- Each TCP connection has a window
  - Controls number of packets in flight
- Sending rate ~Window/RTT
- Vary window size to control sending rate

Windows to keep in mind

- Congestion Window: CWND
  - Bytes that can be sent without overflowing routers
  - Computed by sender using congestion control algo.
- Flow control window: RWND
  - Bytes that can be sent without overflowing receiver
  - Determined by the receiver and reported to the sender
- Sender-side window = min {CWND, RWND}
  - Assume for this lecture that RWND >> CWND

Note

- This lecture talks about CWND in units of MSS
  - MSS (Maximum Segment Size): the amount of payload data in a TCP packet
  - This is *only for the simplicity* of presentation
- Real implementations maintain CWND in bytes

Two basic questions

- How does the sender detect congestion?
- How does the sender adjust its sending rate?
  - To address three issues
    - Finding available bottleneck bandwidth
    - Adjusting to bandwidth variations
    - Sharing bandwidth

Detecting congestion

- Packet delays
  - Tricky: noisy signal (delay often varies considerably)
- Routers tell end hosts when they're congested
- Packet loss
  - Fail-safe signal that TCP already has to detect
  - Complication: non-congestive loss (e.g., checksum errors)

Not all losses are the same

- Duplicate ACKs: isolated loss
  - Still getting ACKs
- Timeout: much more serious
  - Not enough dupacks
  - Must have suffered several losses
- Will adjust rate differently for each case

Rate adjustment

- Basic structure
  - Upon receipt of ACK (of new data): increase rate
  - Upon detection of loss: decrease rate
- How we increase/decrease the rate depends on the phase of congestion control we're in:
  - Discovering available bottleneck bandwidth vs.
  - Adjusting to bandwidth variations

Bandwidth discovery with "Slow Start"

- Goal: estimate available bandwidth
  - Start slow (for safety)
  - Ramp up quickly (for efficiency)
- Consider
  - RTT = 100ms, MSS=1000bytes
  - Window size to fill 1Mbps of BW = 12.5 packets
  - Window size to fill 1Gbps = 12,500 packets
  - Either is possible!

Slow Start phase

- Sender starts at a slow rate, but increases exponentially until first loss
- Start with a small congestion window
  - Initially, CWND = 1
  - So, initial sending rate is MSS/RTT
- Double the CWND for each RTT with no loss

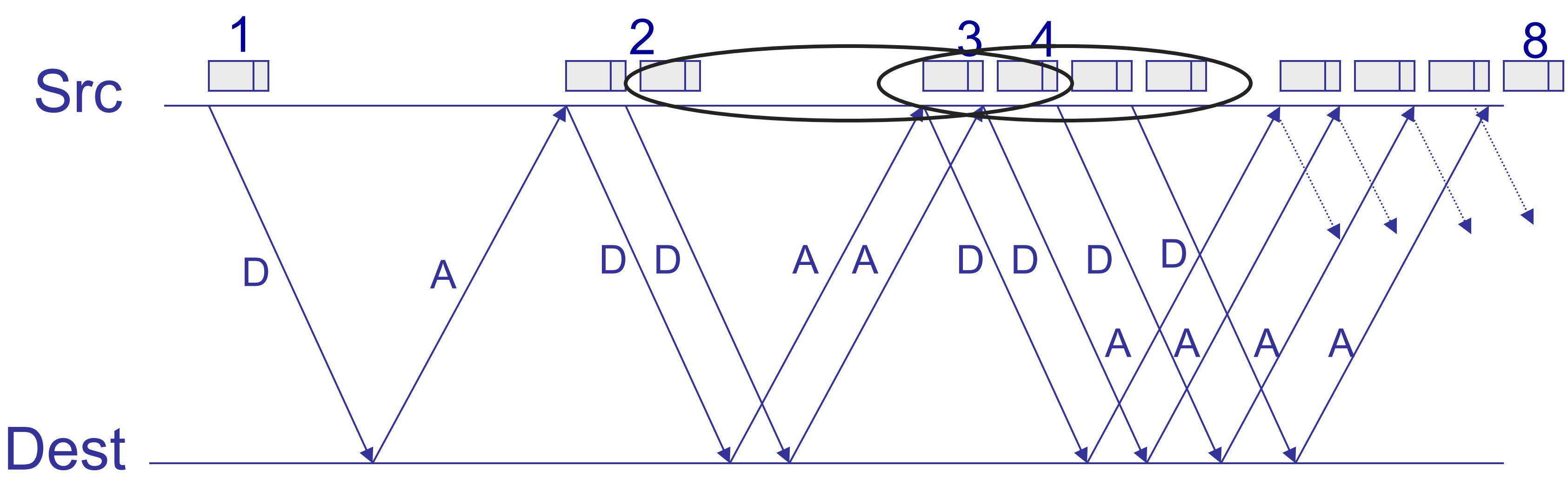
Slow Start in action

- For each RTT: double CWND
  - i.e., for each ACK, CWND += 1

Linear increase per ACK(CWND+1) → exponential increase per RTT (2*CWND)

Slow Start in action

- For each RTT: double CWND
  - i.e., for each ACK, CWND += 1



When does Slow Start stop?

- Slow Start gives an estimate of available bandwidth
  - At some point, there will be loss
- Introduce a “slow start threshold” (ssthresh)
  - Initialized to a large value
- If CWND > ssthresh, stop Slow Start

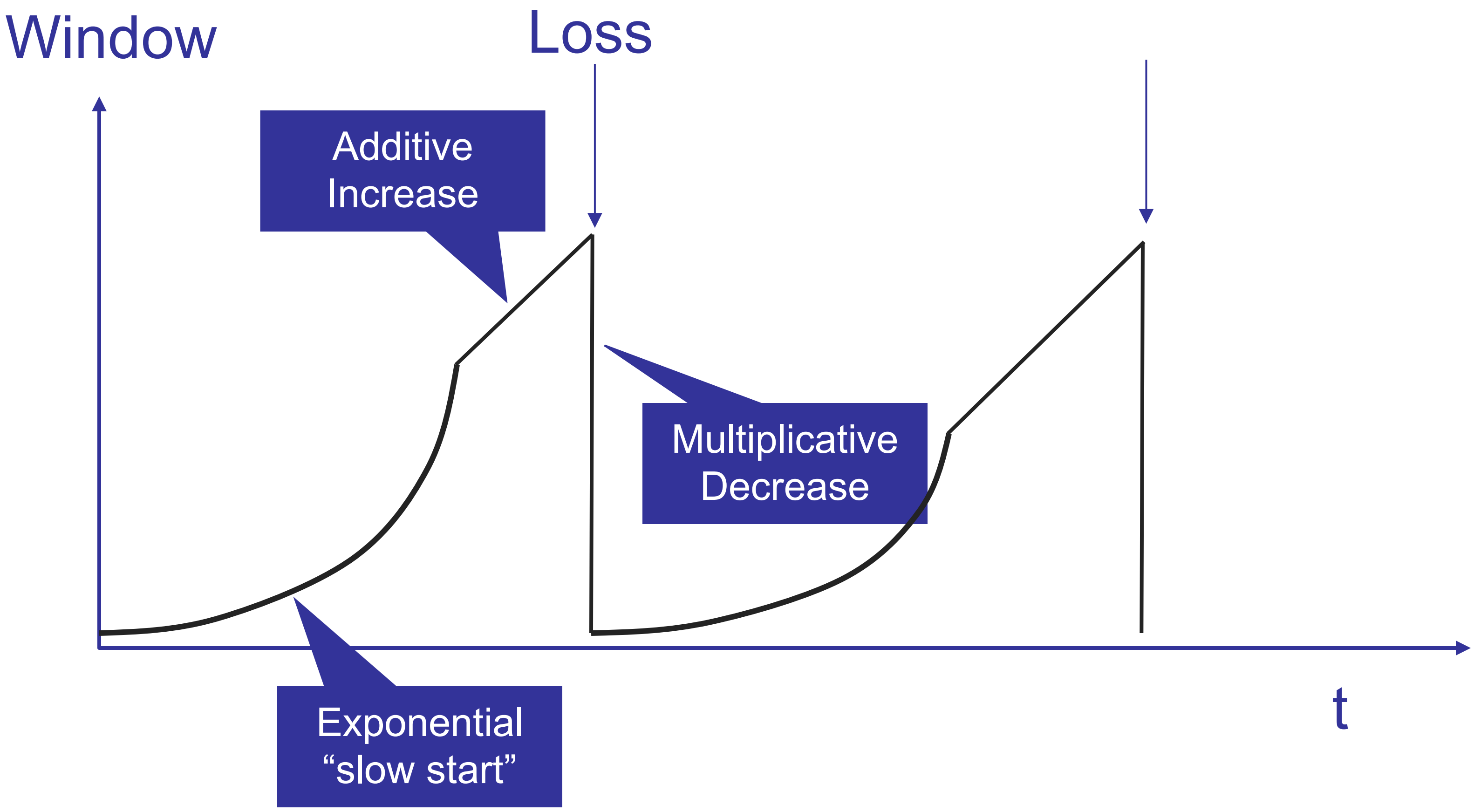
Adjusting to varying bandwidth

- CWND > ssthresh
  - Stop rapid growth and focus on maintenance
- Now, want to track variations in this available bandwidth, oscillating around its current value
  - Repeated probing (rate increase) and backoff (decrease)
- TCP uses: “Additive Increase Multiplicative Decrease” (AIMD)

AIMD

- Additive increase
  - For each ACK, CWND = CWND+ 1/CWND
  - CWND is increased by one only if all segments in a CWND have been acknowledged
- Multiplicative decrease
  - On packet loss, divide ssthresh in half and slow start
    - » ssthresh = CWND/2
    - » CWND = 1
    - » Initiate Slow Start
  - Note that we're ignoring the "dupAck" fix for now

AIMD leads to TCP sawtooth



Why AIMD?

- Recall the three issues
  - Finding available bottleneck bandwidth
  - Adjusting to bandwidth variations
  - Sharing bandwidth
- Two goals for bandwidth sharing
  - Efficiency: High utilization of link bandwidth
  - Fairness: Each flow gets equal share

Why AIMD?

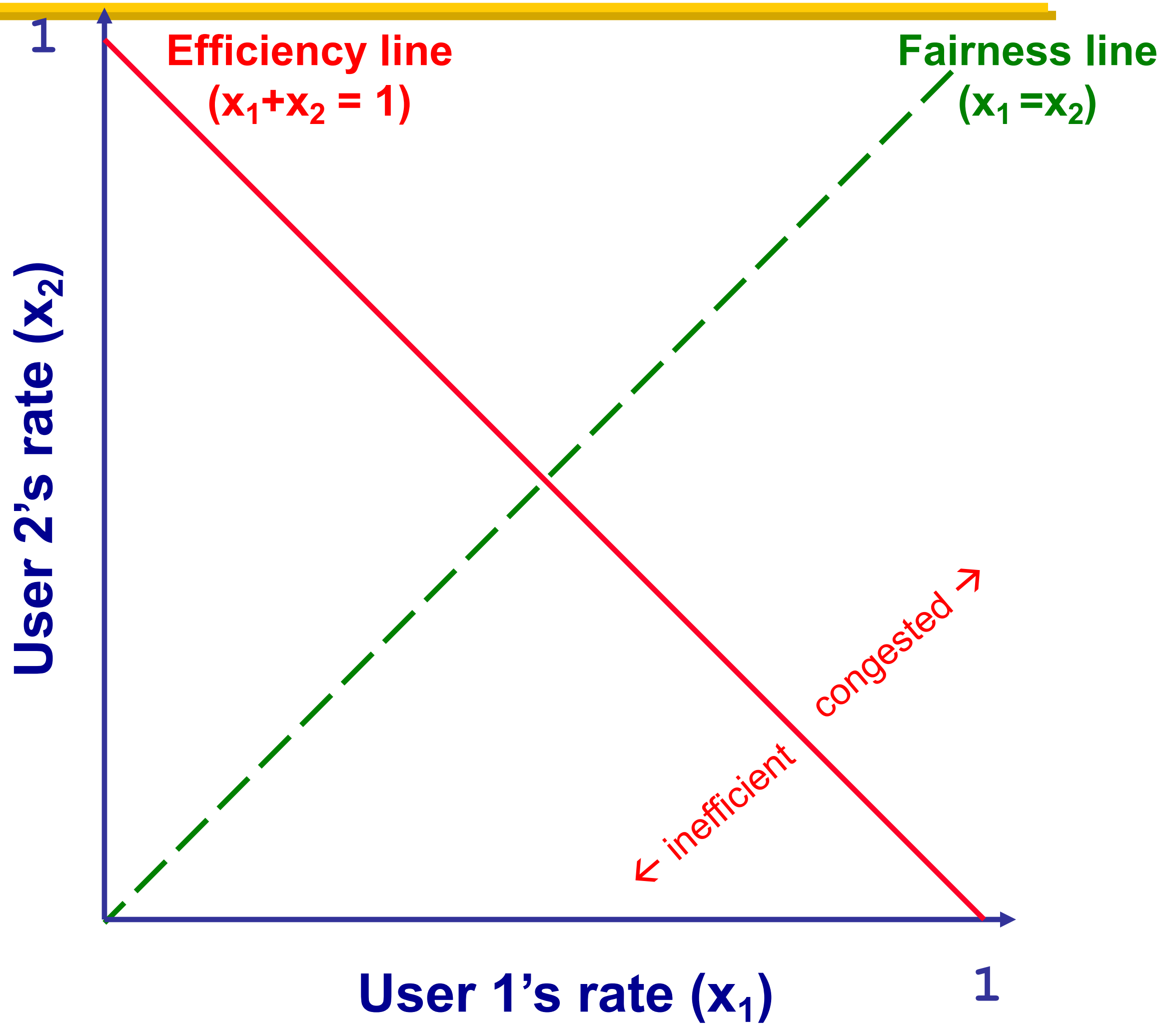
- Every RTT, we can do
  - Multiplicative increase or decrease: CWND$\rightarrow$ a*CWND
  - Additive increase or decrease: CWND$\rightarrow$ CWND + b
- Four alternatives:
  - AIAD: gentle increase, gentle decrease
  - AIMD: gentle increase, drastic decrease
  - MIAD: drastic increase, gentle decrease
  - MIMD: drastic increase and decrease

Simple model of congestion control

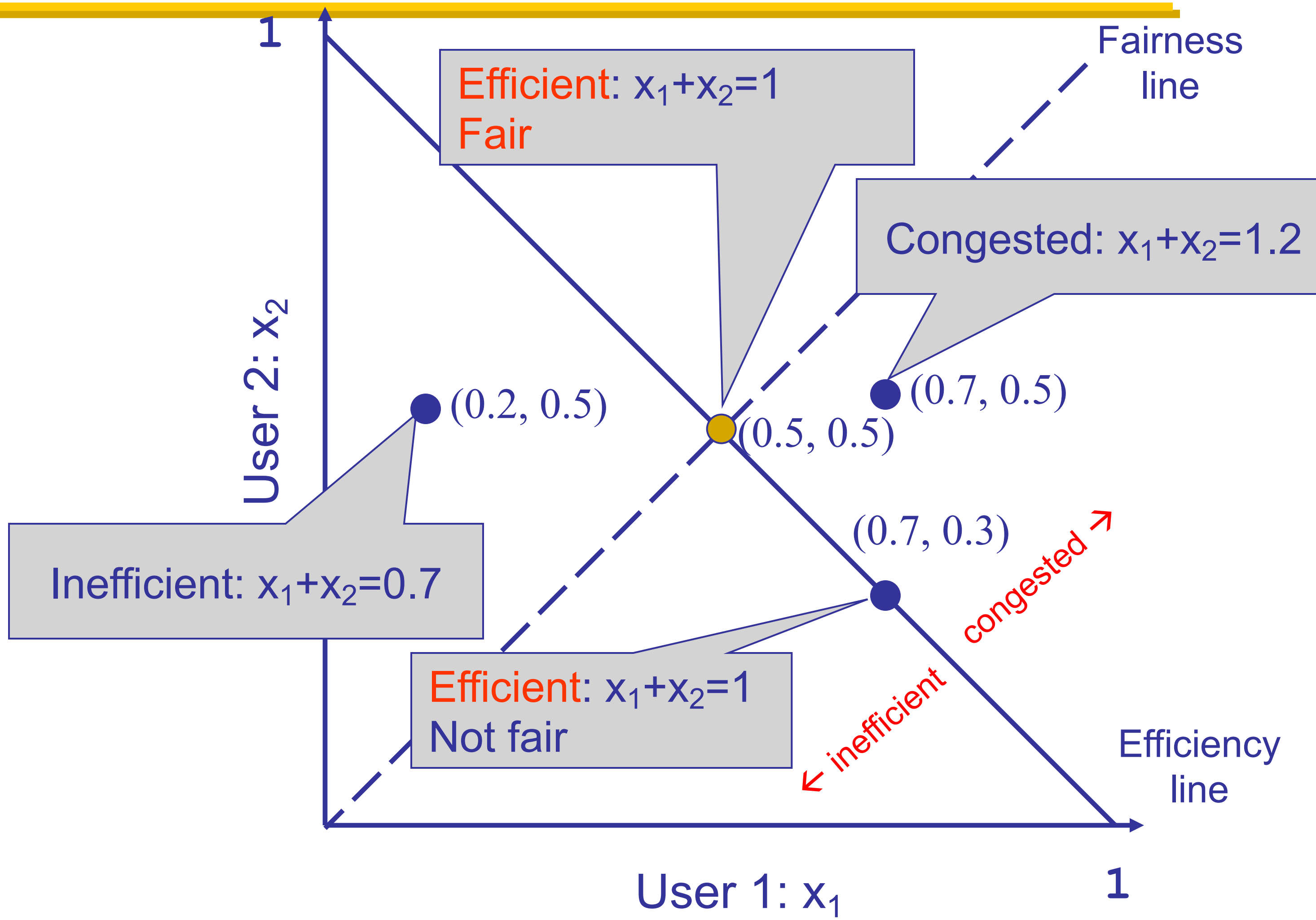
- Two users
  - rates x1 and x2

- Congestion when x1+x2 > 1
- Unused capacity when x1+x2 < 1

- Fair when x1 =x2



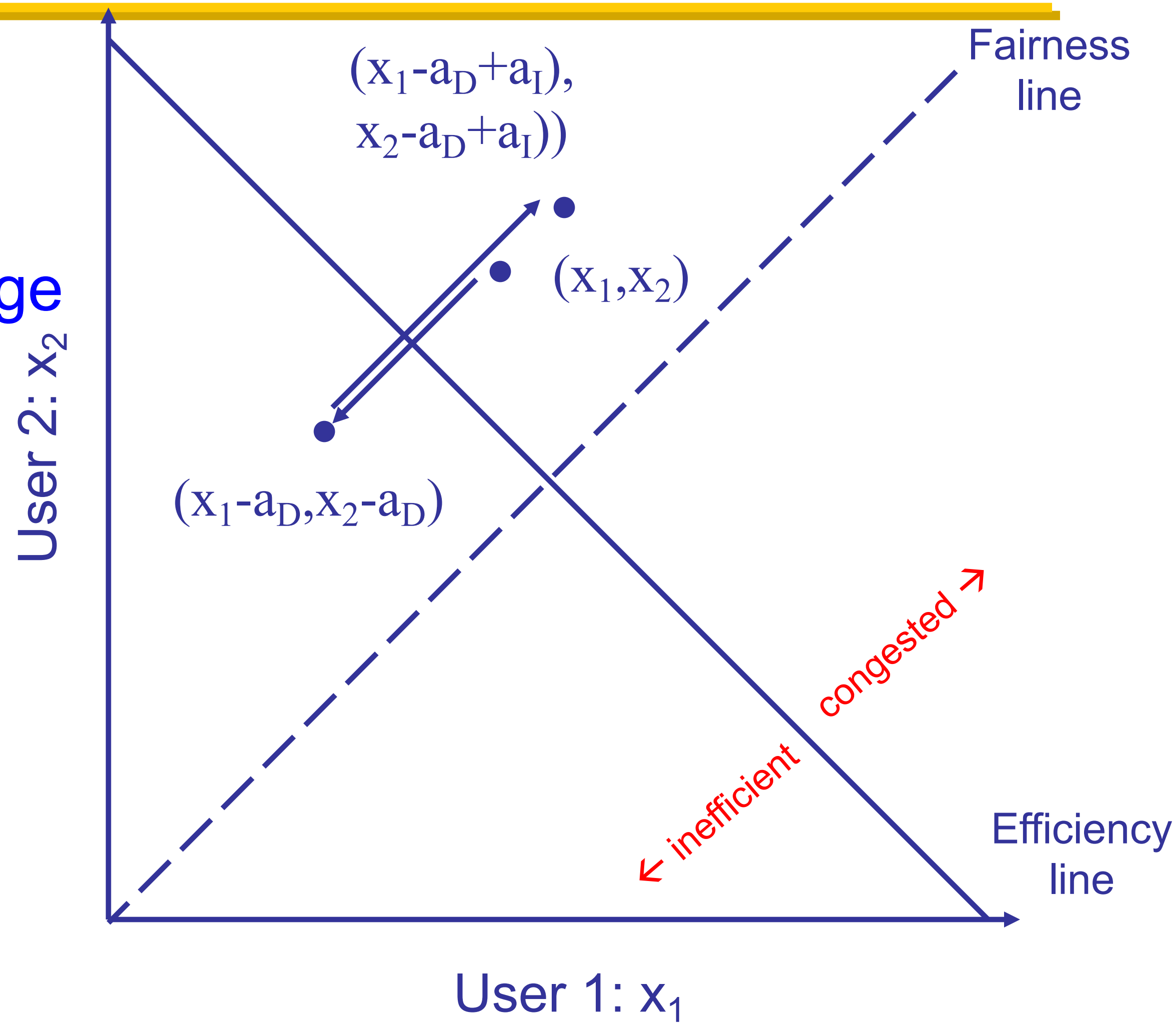
Example



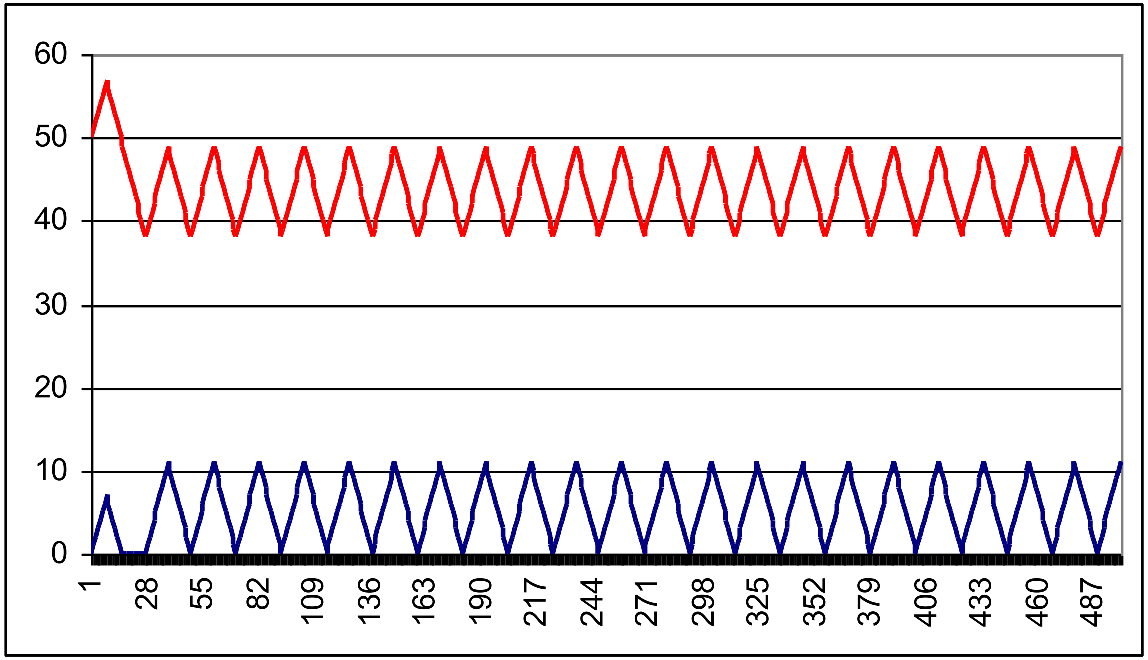
1
Fairness line
Efficient: $x_1+x_2=1$
Fair
Congested: $x_1+x_2=1.2$
User 2: x_2
(0.2, 0.5)
(0.7, 0.5)
(0.5, 0.5)
(0.7, 0.3)
Inefficient: $x_1+x_2=0.7$
congested →
Efficient: $x_1+x_2=1$
Not fair
← inefficient
Efficiency line
User 1: x_1
1

AIAD

- Increase: $x + a_I$
- Decrease: $x - a_D$
- Does not converge to fairness

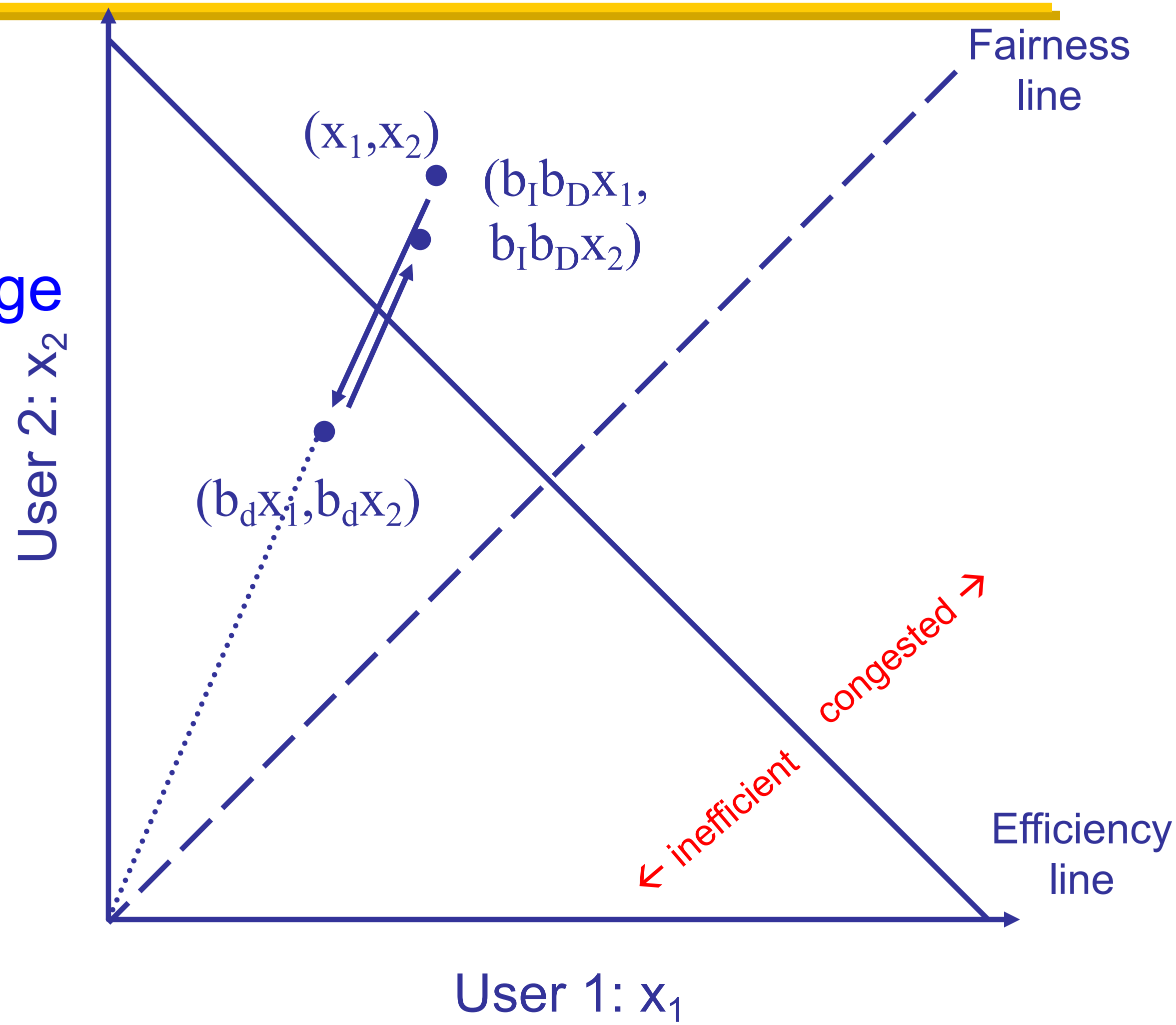


AIAD Sharing Dynamics



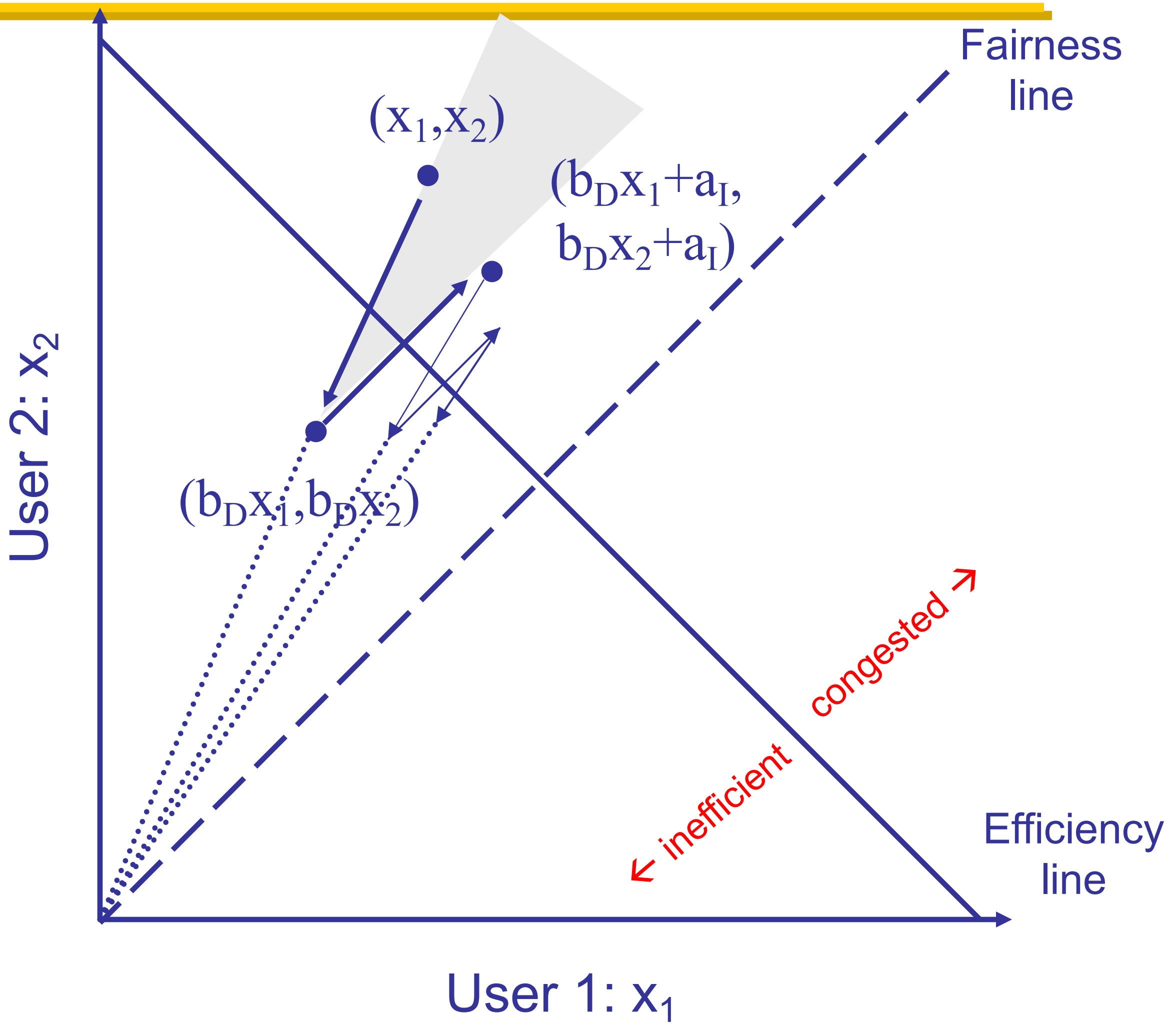
MIMD

- Increase: $x*b_I$
- Decrease: $x*b_D$
- Does not converge to fairness

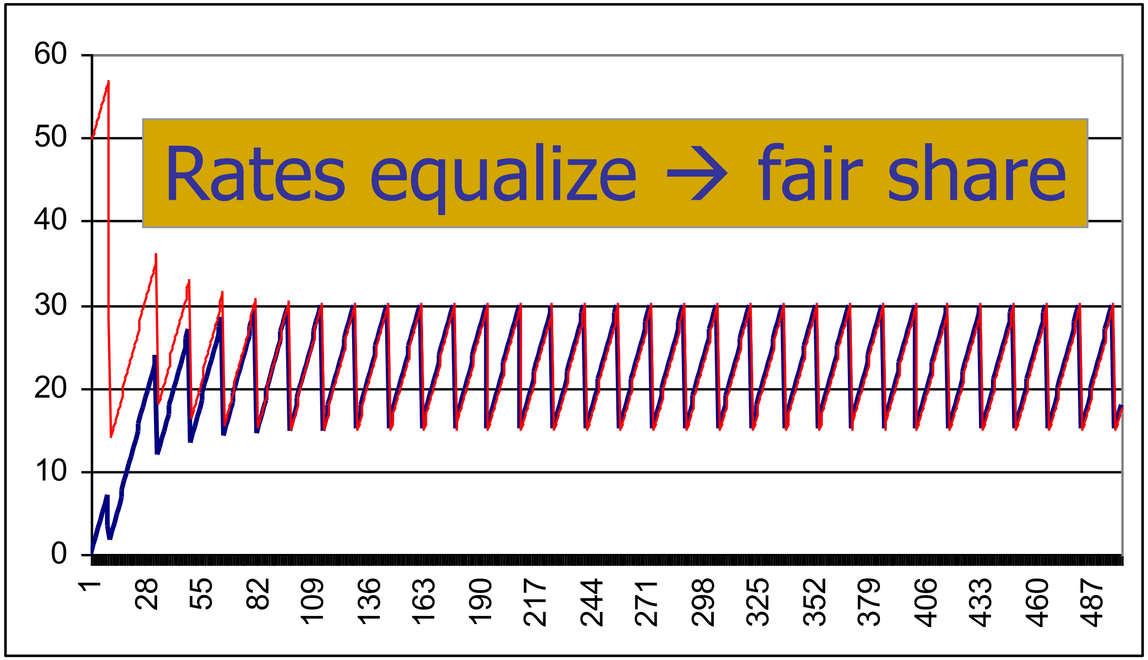


AIMD

- Increase: $x+a_I$
- Decrease: $x*b_D$
- Converges to fairness



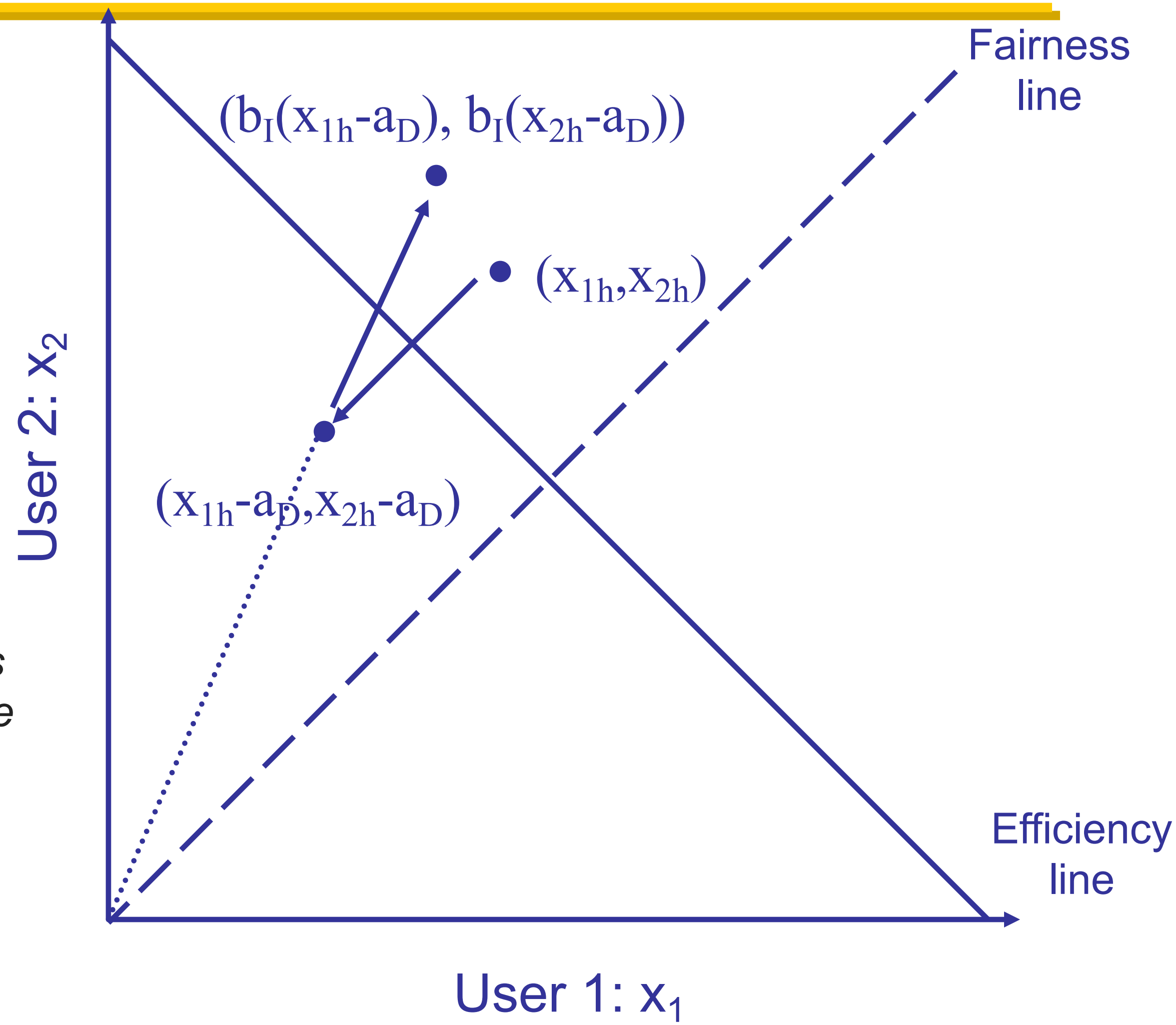
AIMD Sharing Dynamics



MIAD

- Increase: $x*b_I$
- Decrease: $x - a_D$
- Does not converge to fairness
- Does not converge to efficiency

- *"Analysis of the Increase and Decrease Algorithms for Congestion Avoidance in Computer Networks"*
  -- Chiu and Jain



Summary

- Flow control ensures that the sender does not overflow the receiver
- Congestion control ensures that the sender does not overflow the network
  - Discover bandwidth
  - Adjust to conditions
  - Share bandwidth with others