

EECS 489

Computer Networks

Winter 2025

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

S1 S2 R

0 0 0

--> <SEQ x><SYN>

1 0 0

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

1 1 1

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

1 1 1

B

S1 S2 R

0 0 0

-->

0 0 1

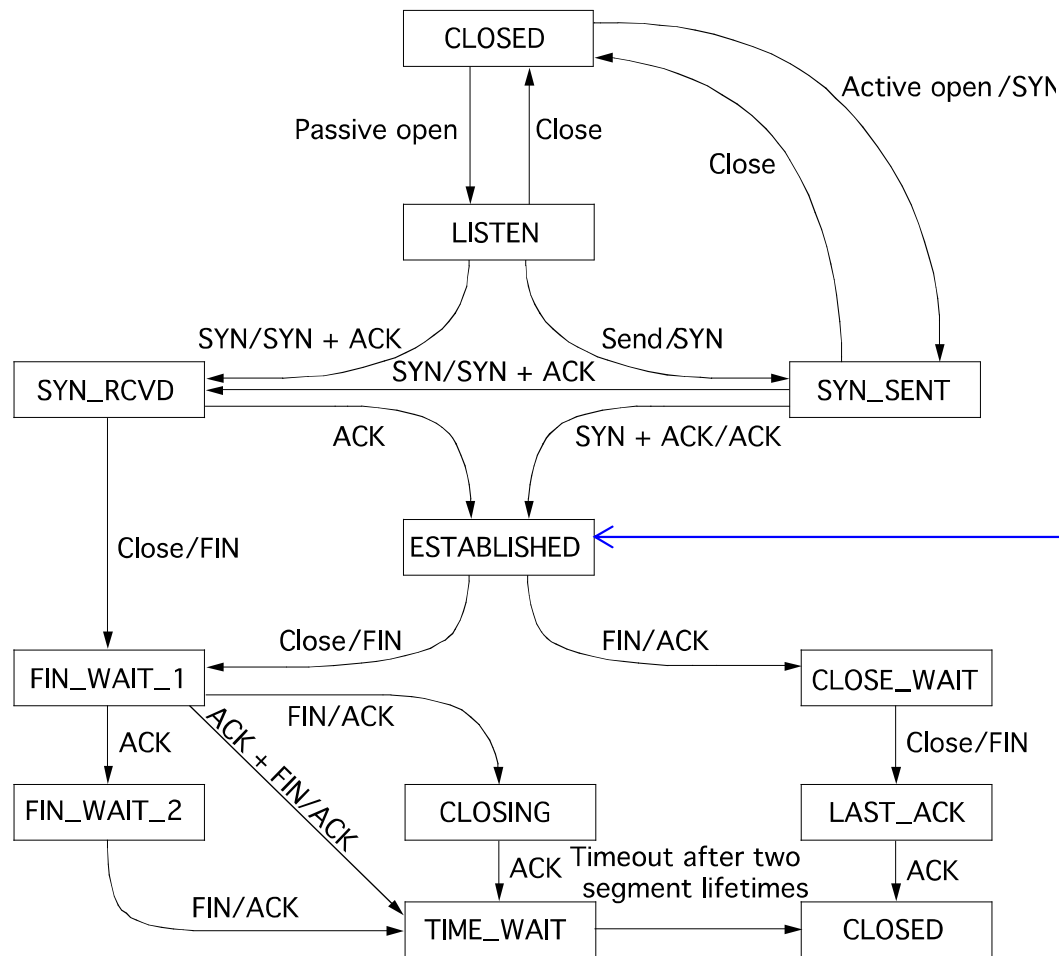
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1 0 1

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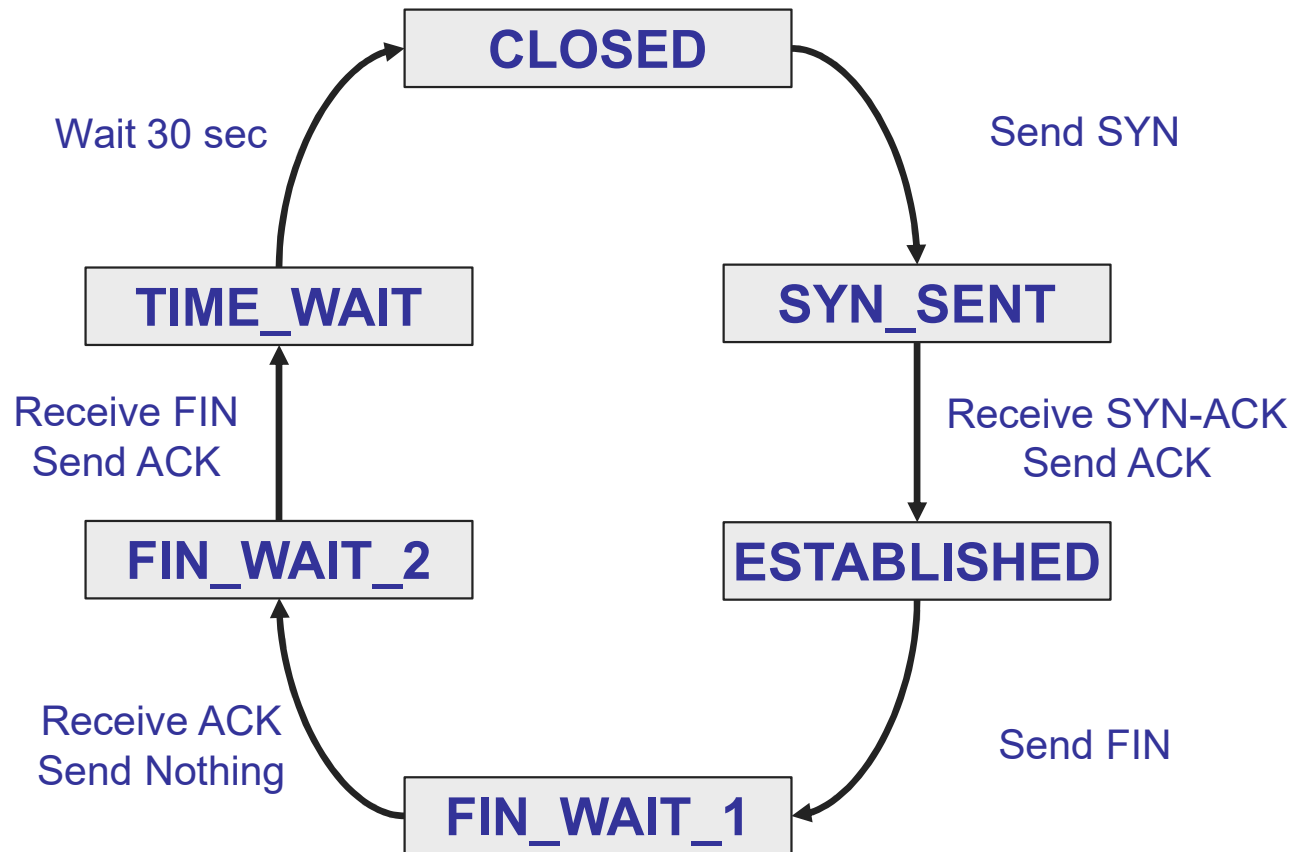
1 1 1

TCP state transitions

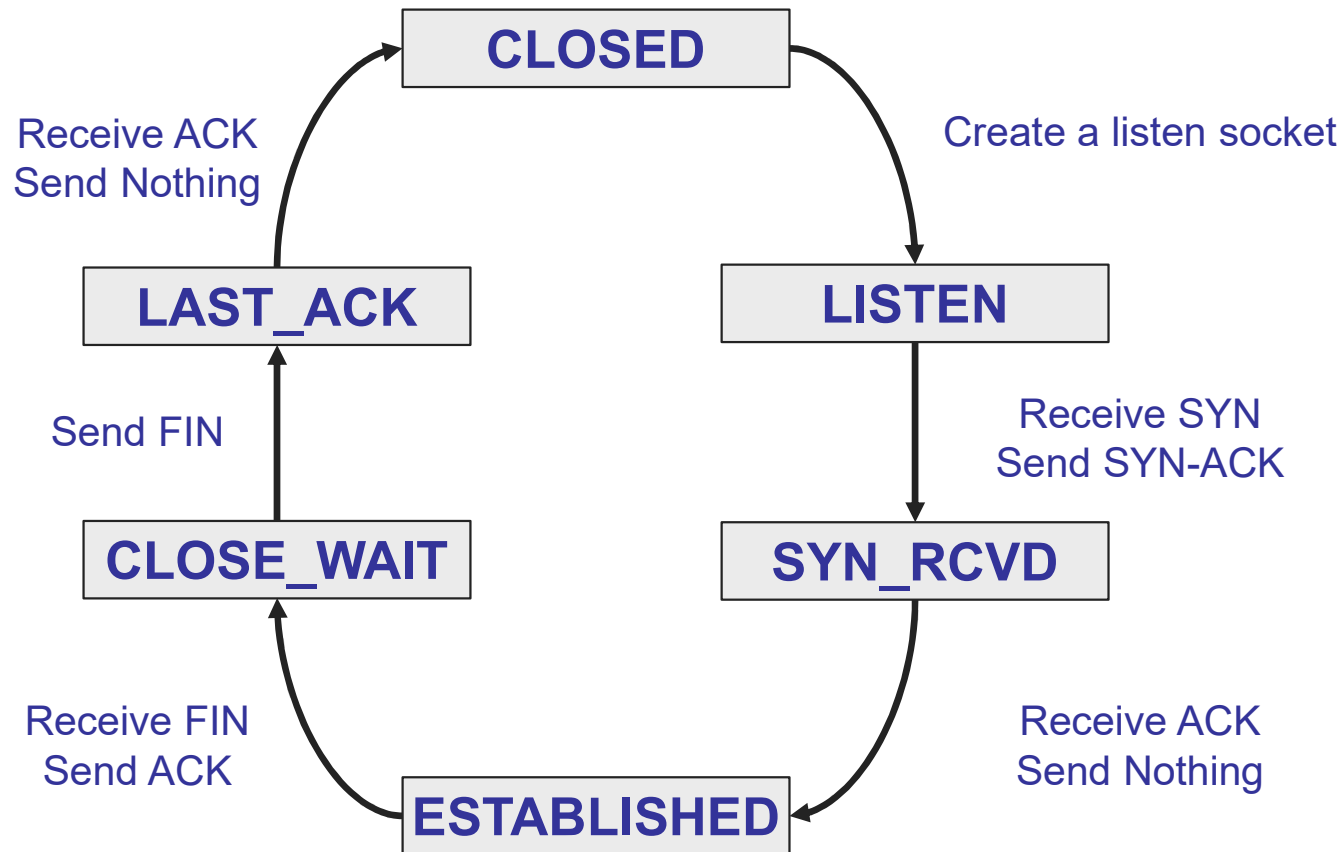


Data, ACK exchanges are in here

TCP client lifecycle



TCP server lifecycle

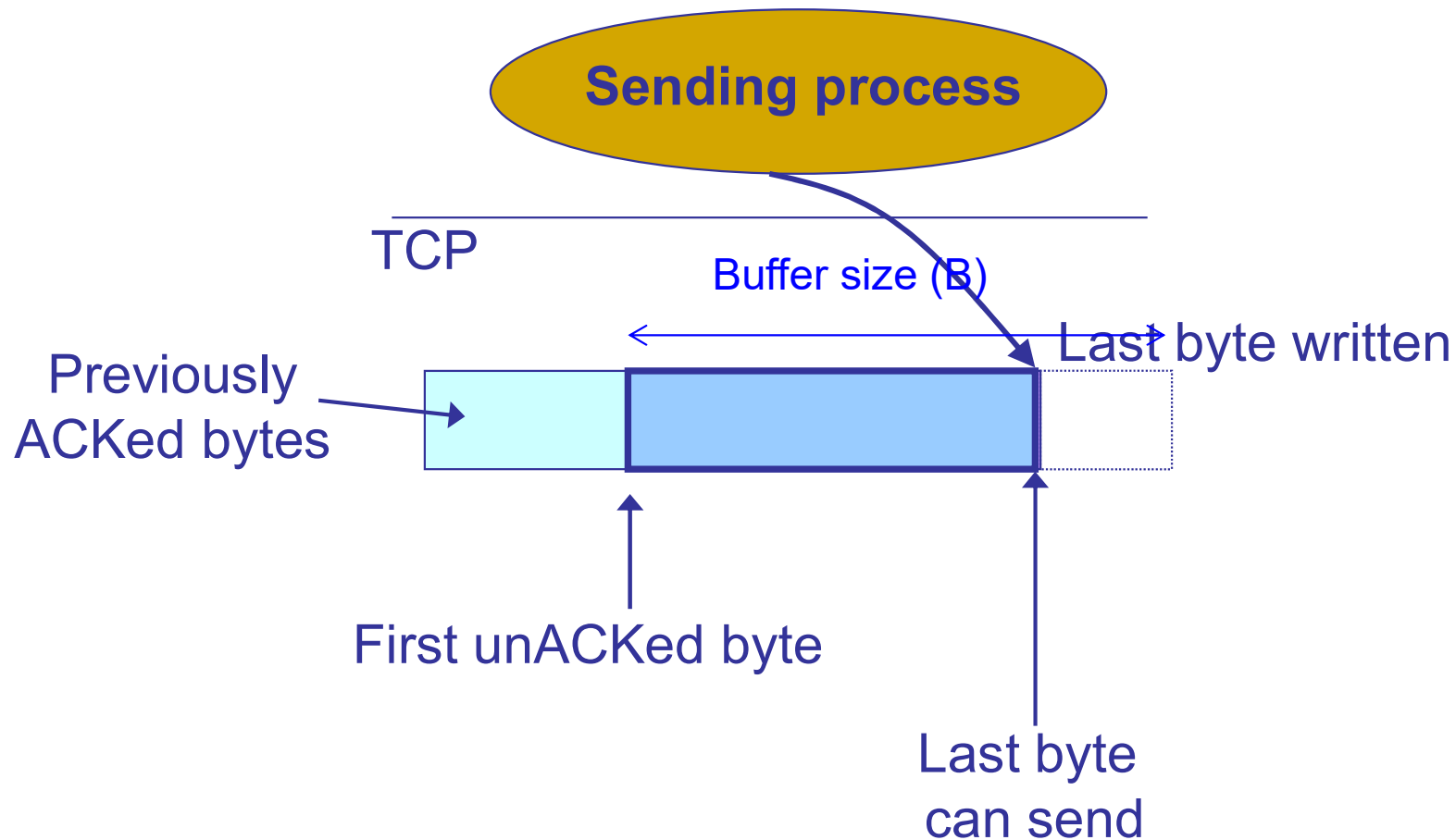


TCP FLOW CONTROL

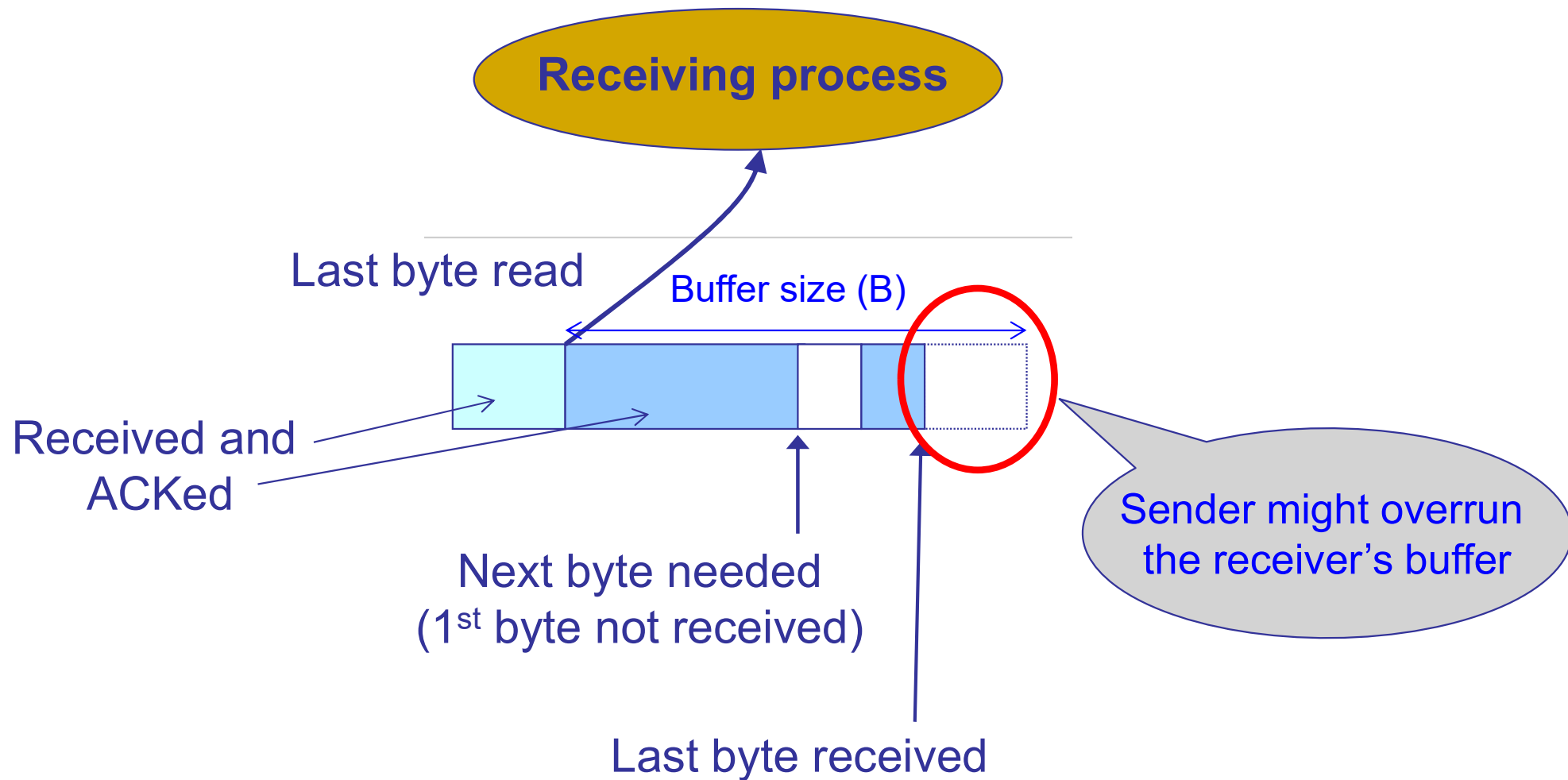
Recap: Sliding window

- 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)

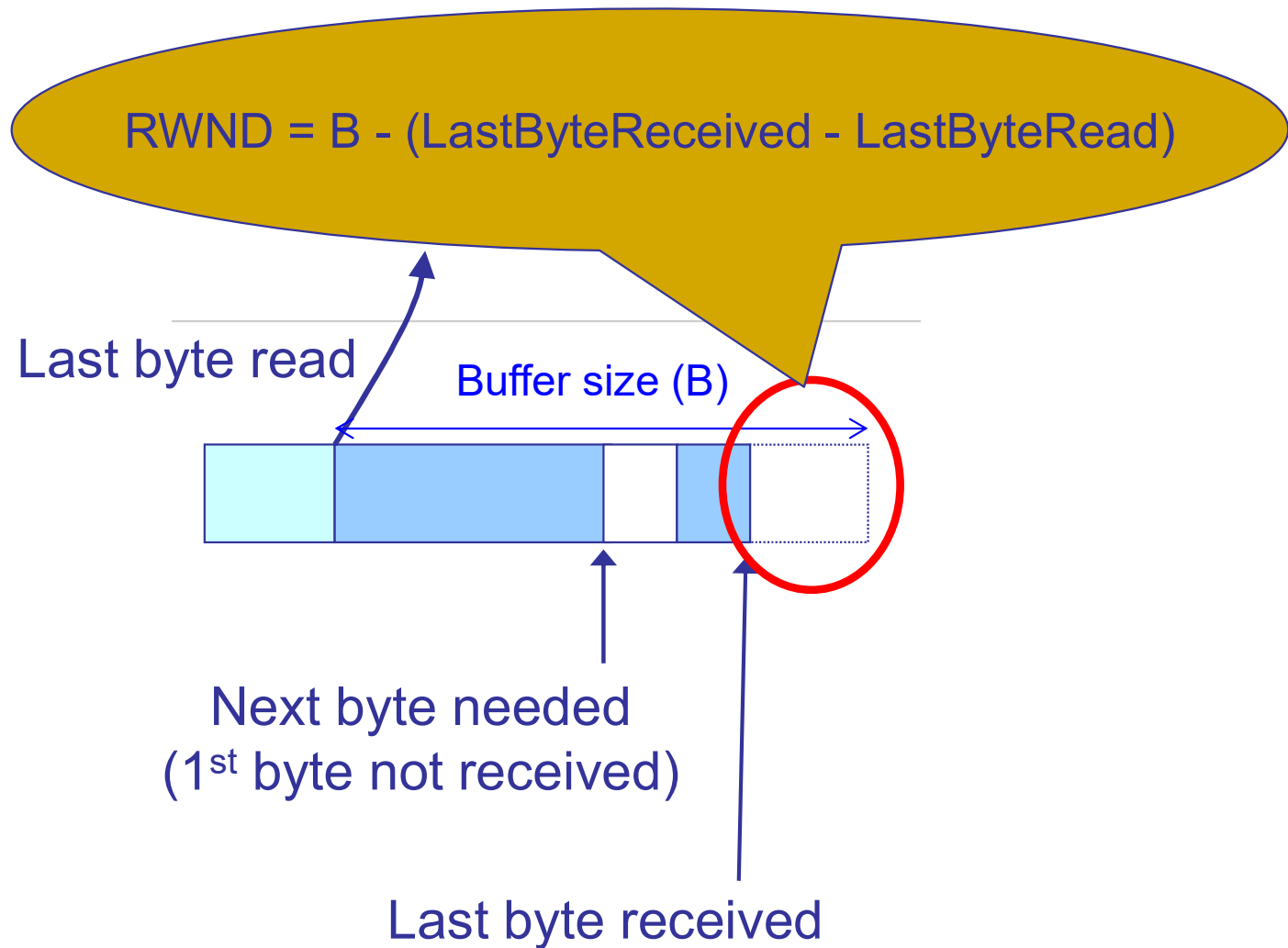
- ❑ Receiver uses an “Advertised Window” (RWND) to prevent sender from overflowing its window
 - Receiver indicates value of RWND in ACKs
 - Sender ensures that the total number of bytes in flight \leq RWND

TCP header

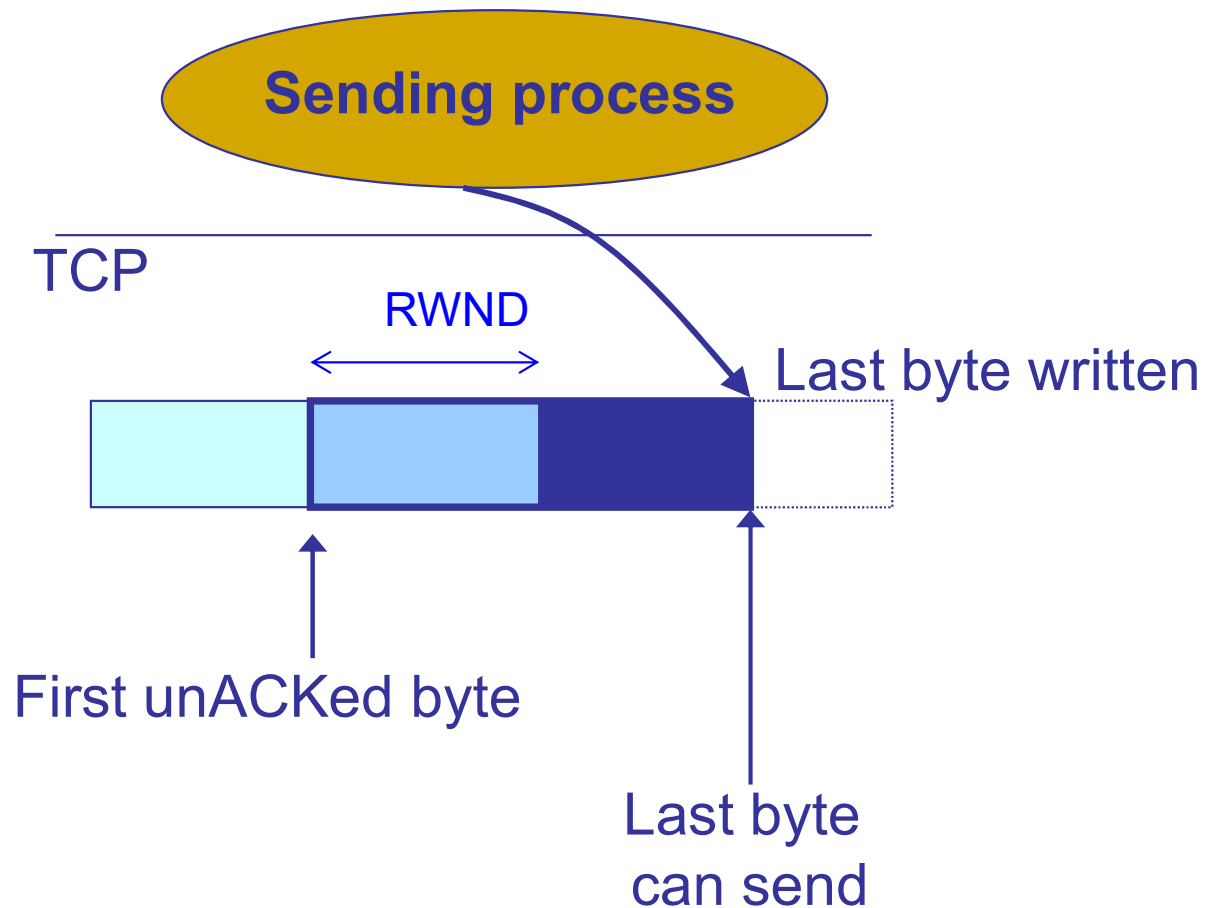
Source port		Destination port	
Sequence number			
Acknowledgment			
HdrLen	0	Flags	Advertised window
Checksum		Urgent pointer	

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

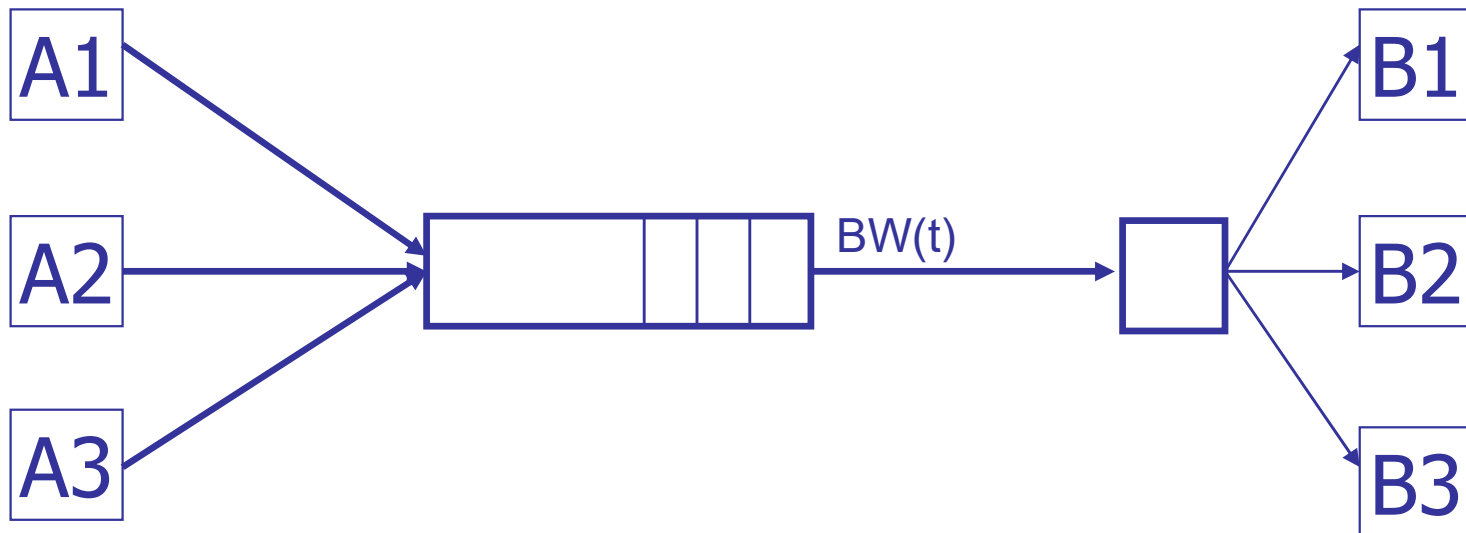


- ❓ Adjust rate to match instantaneous bandwidth
 - Assuming you have rough idea of bandwidth

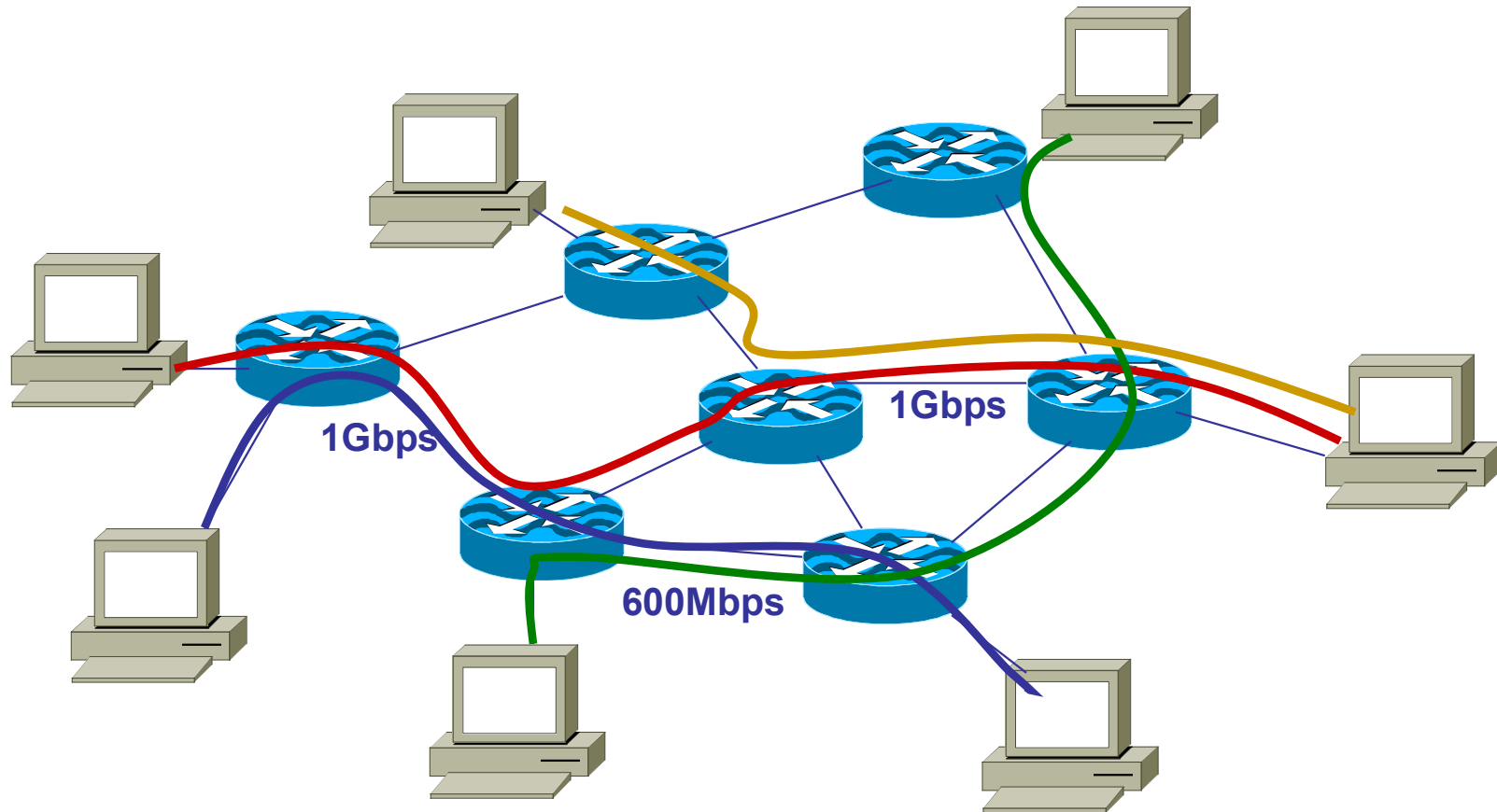
Multiple flows and sharing bandwidth

? 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 $\sim \text{Window}/\text{RTT}$
- 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 = **$\min \{ \text{CWND}, \text{RWND} \}$**
 - Assume for this lecture that $\text{RWND} \gg \text{CWND}$

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
 - $RTT = 100\text{ms}$, $MSS = 1000\text{bytes}$
 - 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

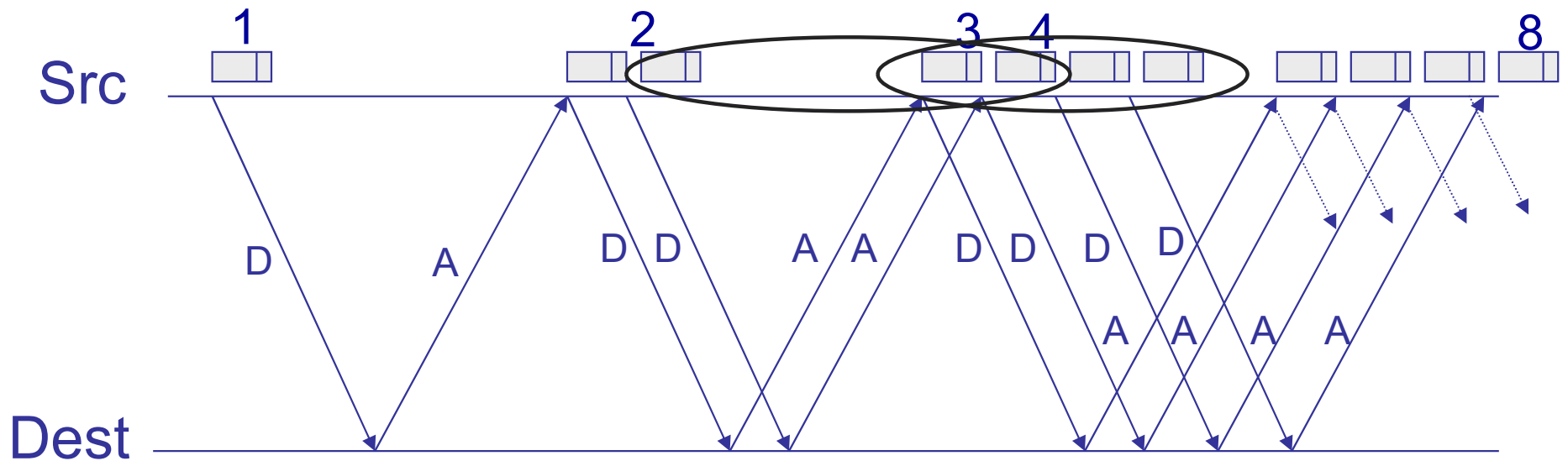
Slow Start in action

- For each RTT: double CWND
 - i.e., for each ACK, $\text{CWND} += 1$

Linear increase per ACK ($\text{CWND}+1$) →
exponential increase per RTT ($2 * \text{CWND}$)

Slow Start in action

- For each RTT: double CWND
 - i.e., for each ACK, $\text{CWND} += 1$



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

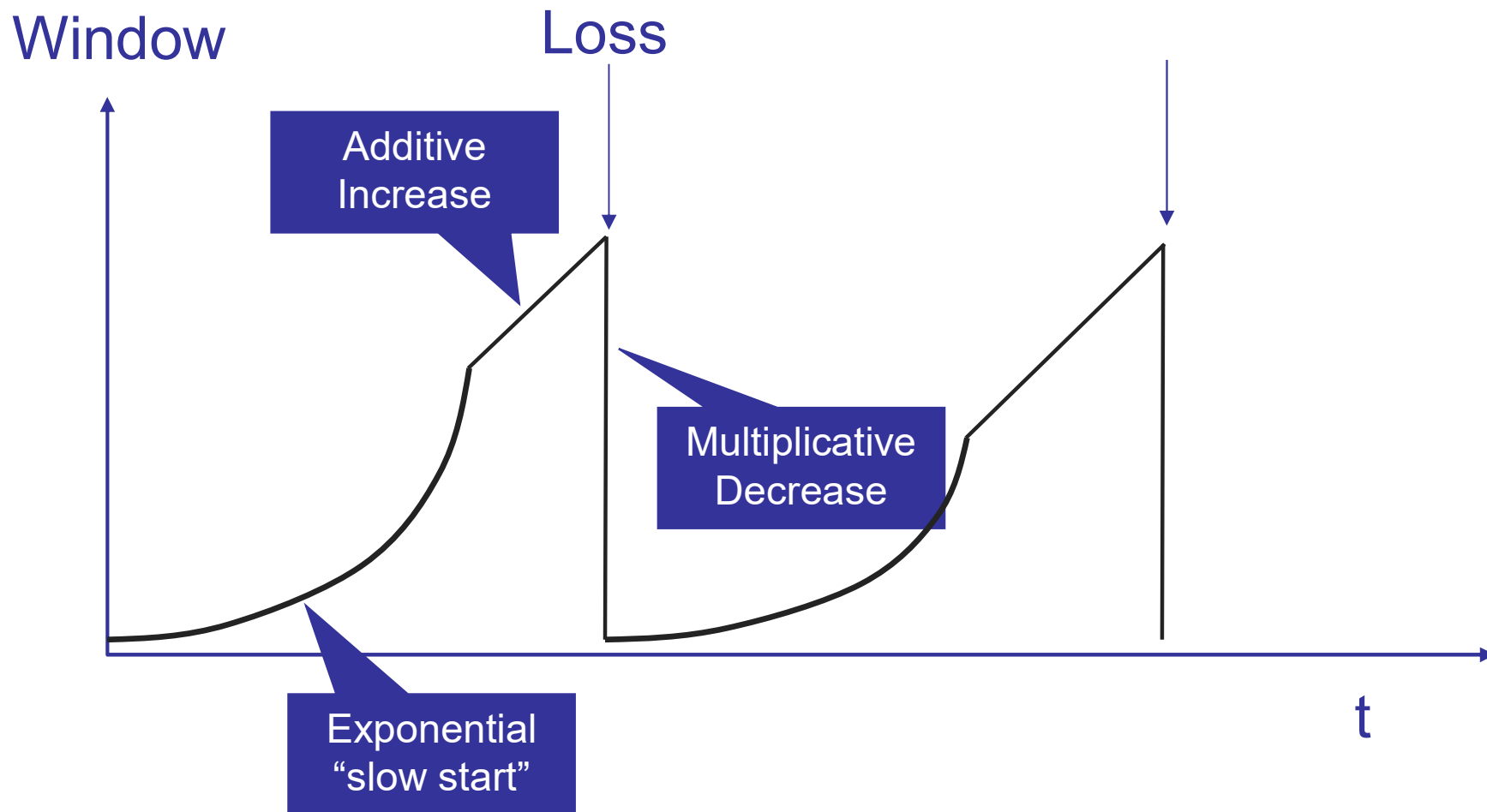
■ Additive increase

- 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**

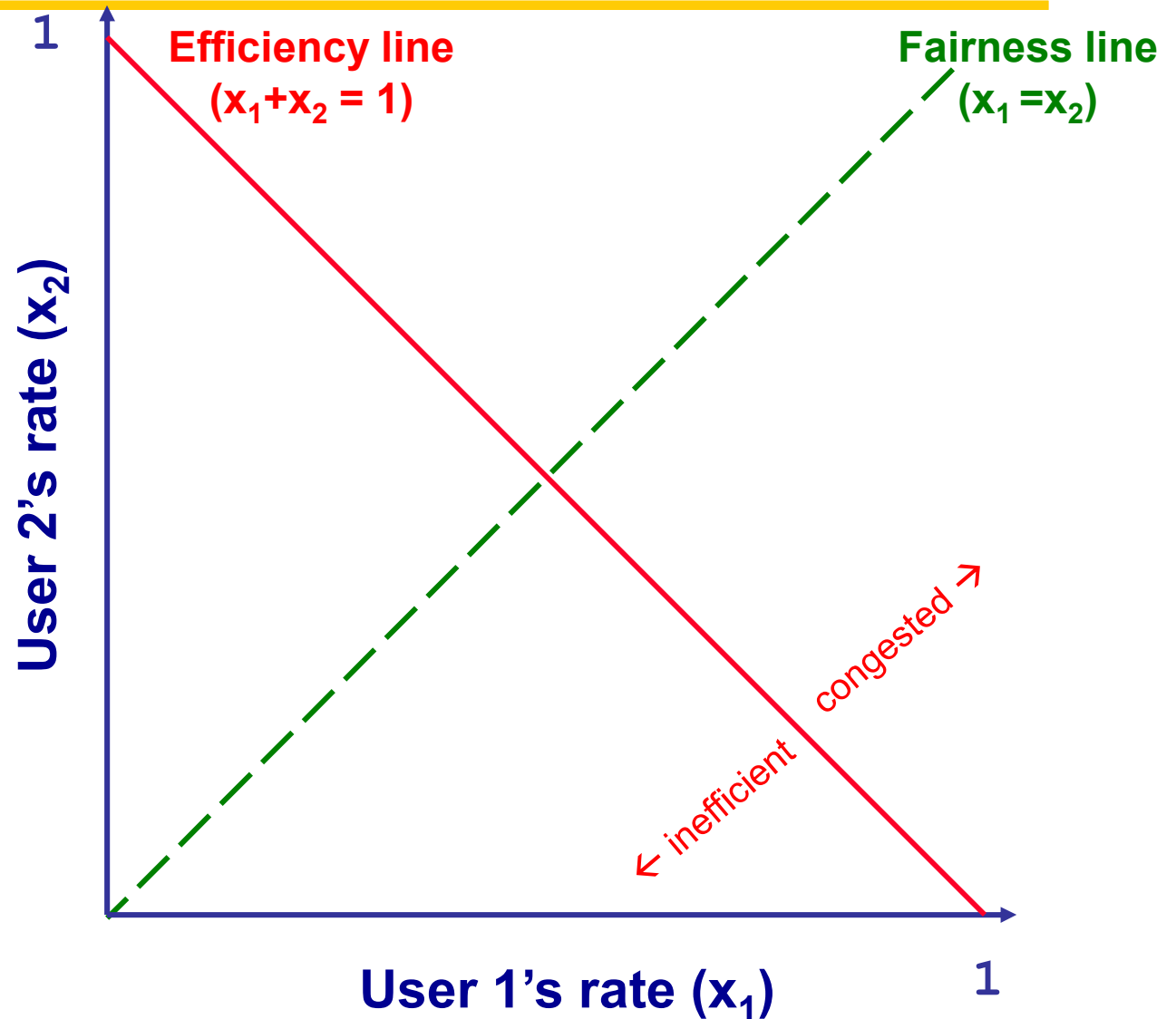
- Two goals for bandwidth sharing
 - **Efficiency**: High utilization of link bandwidth
 - **Fairness**: Each flow gets equal share

Why AIMD?

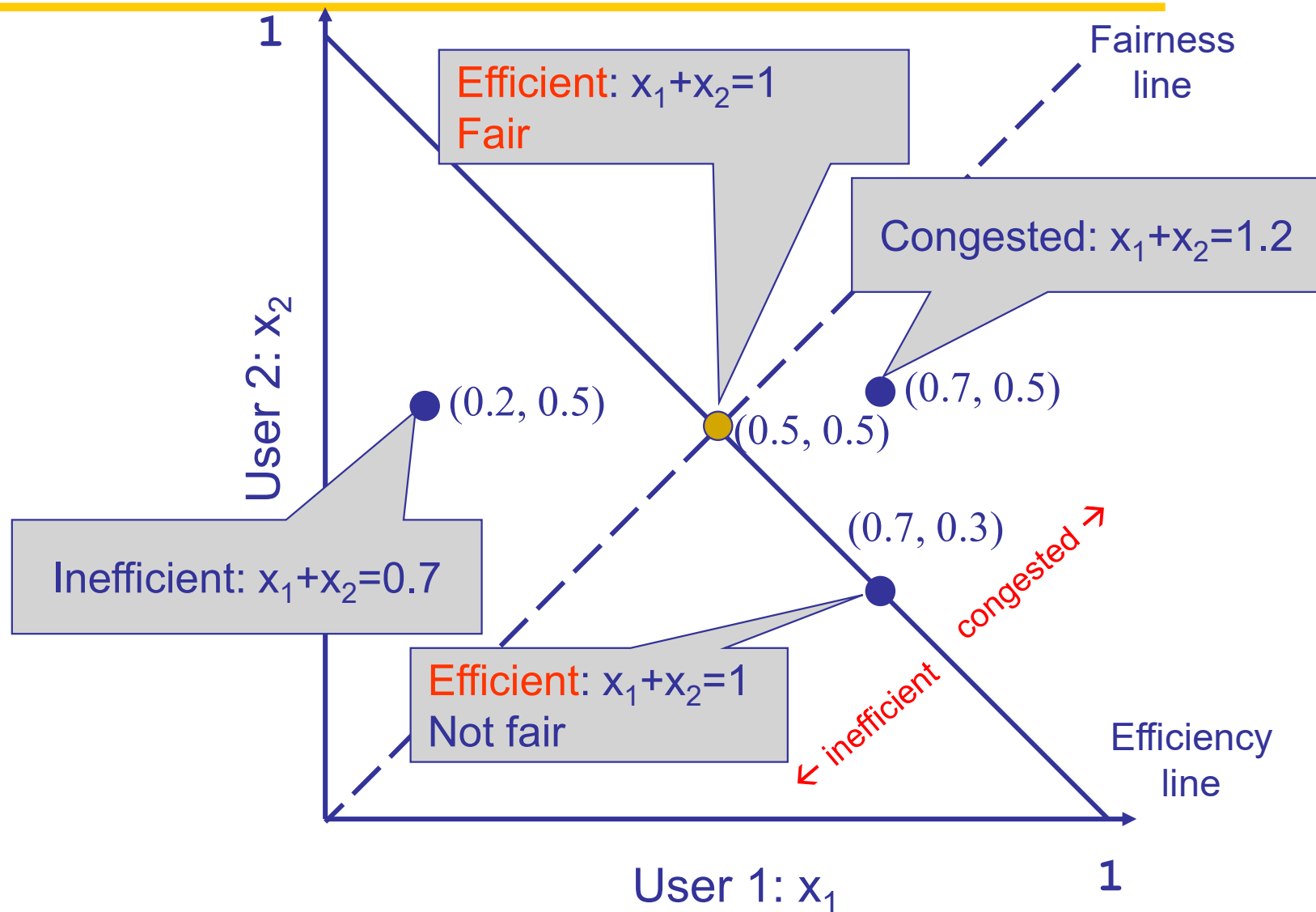
- Every RTT, we can do
 - Multiplicative increase or decrease: $CWND \rightarrow a * CWND$
 - Additive increase or decrease: $CWND \rightarrow CWND + b$
- Four alternatives:
 - AIAD: gentle increase, gentle decrease
 - AIMD: gentle increase, drastic decrease
 - MIAD: drastic increase, gentle decrease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

Simple model of congestion control

- Two users
 - rates x_1 and x_2
- Congestion when $x_1 + x_2 > 1$
- Unused capacity when $x_1 + x_2 < 1$
- Fair when $x_1 = x_2$

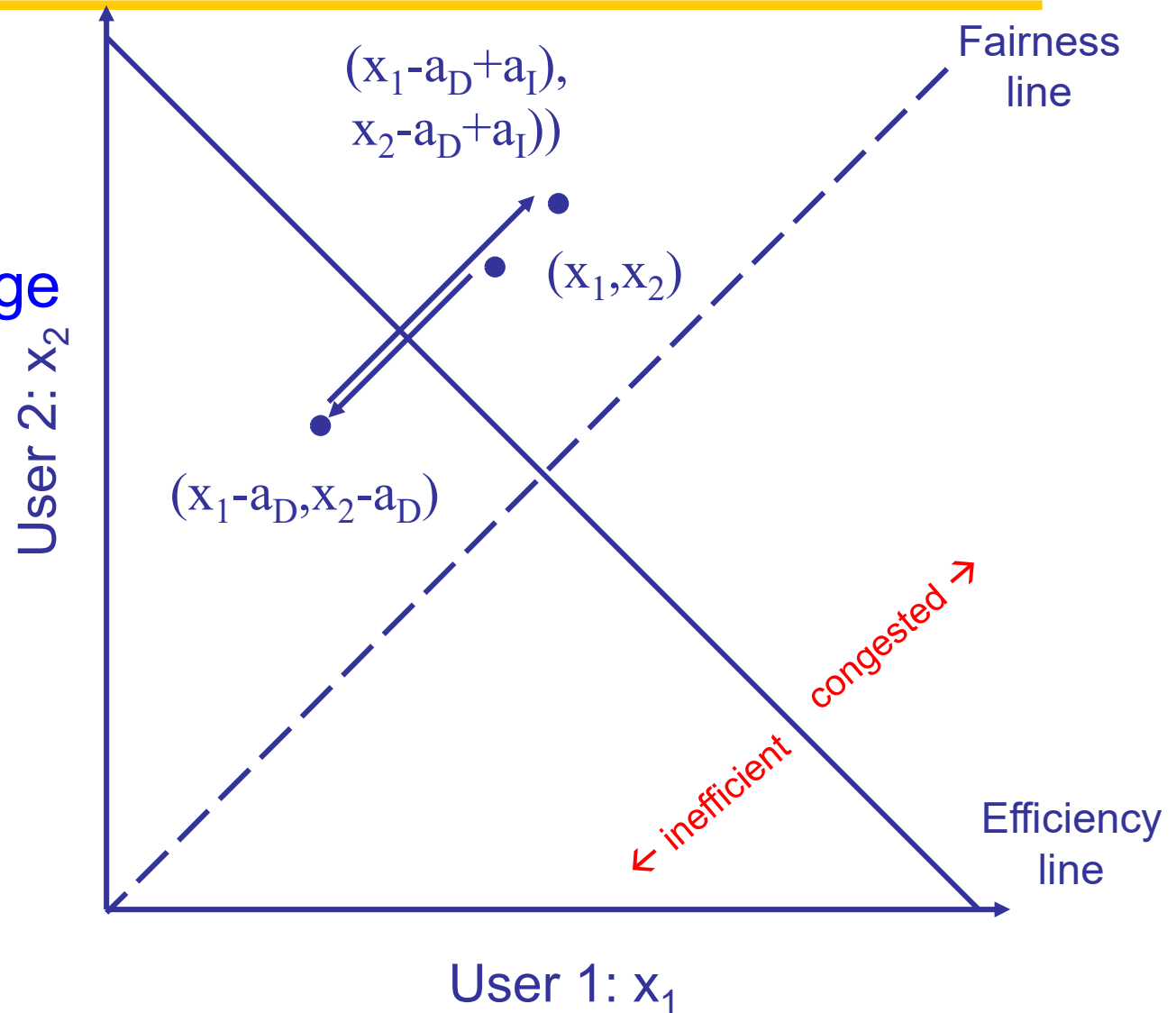


Example

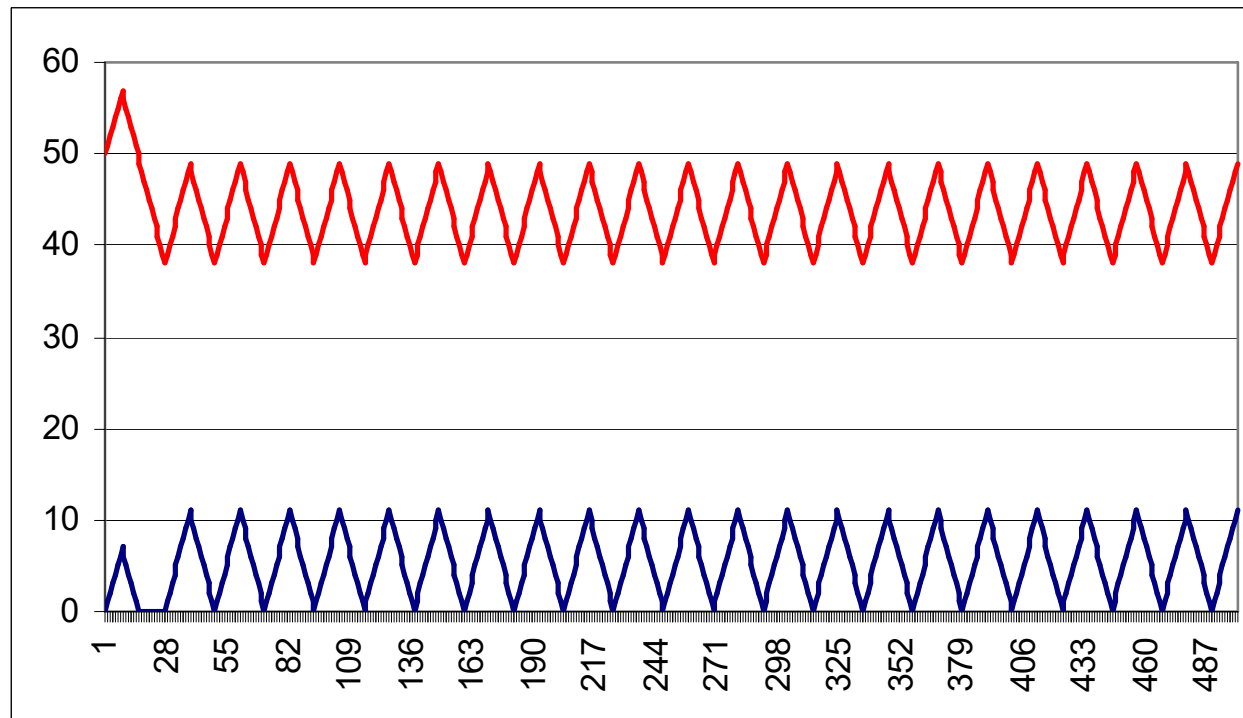
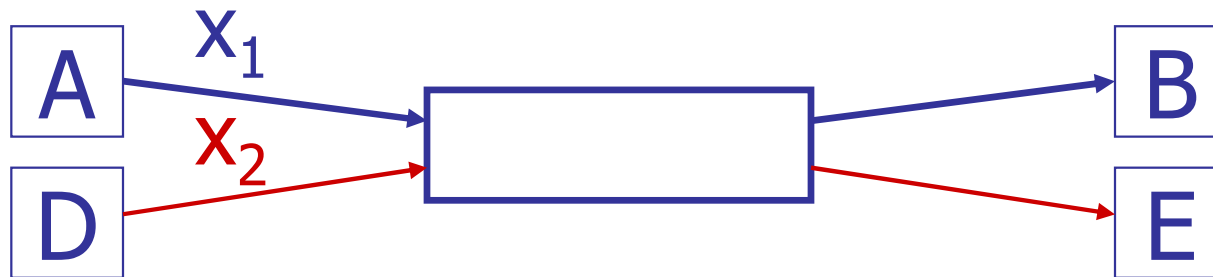


AIAD

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

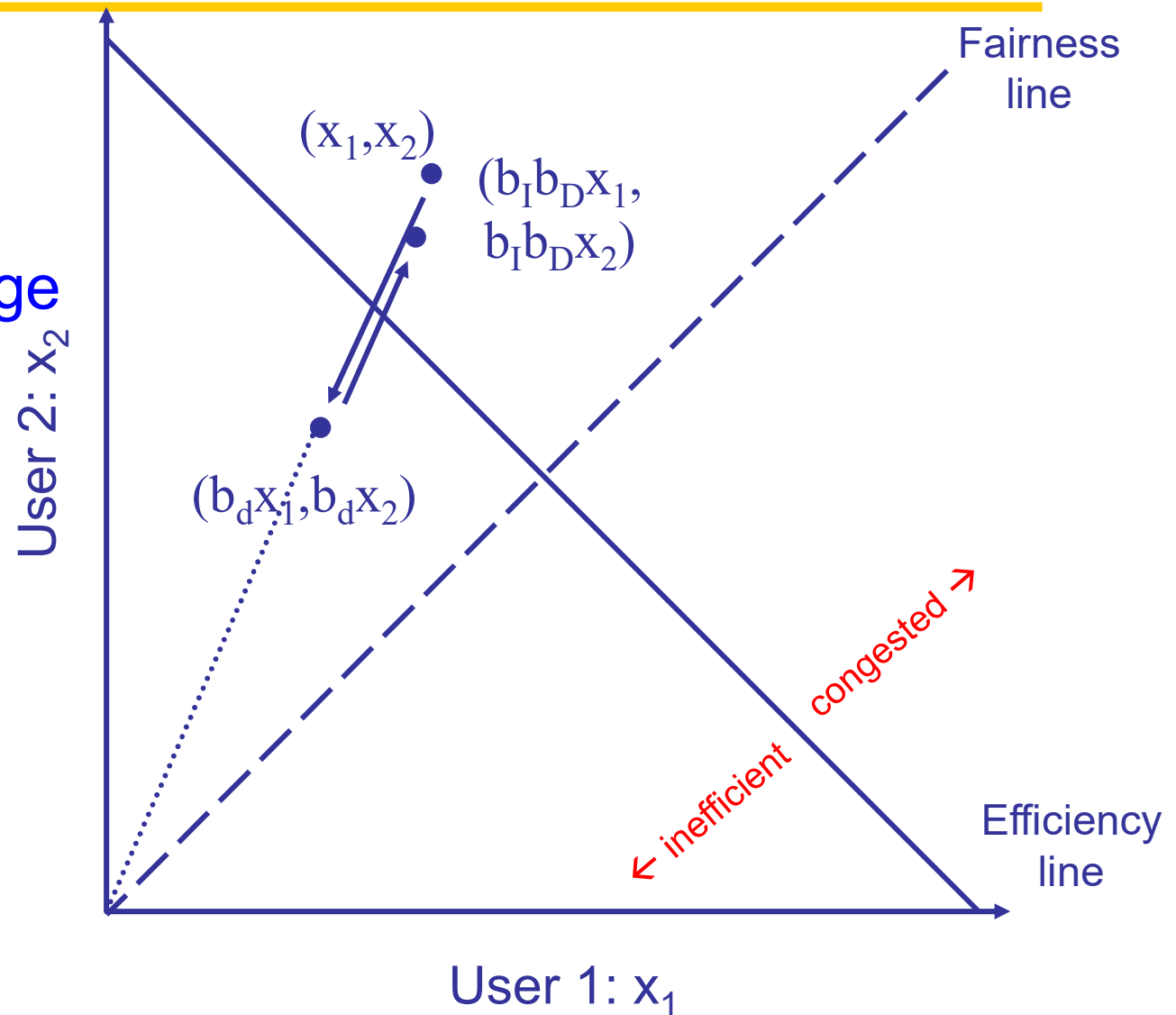


AIAD Sharing Dynamics



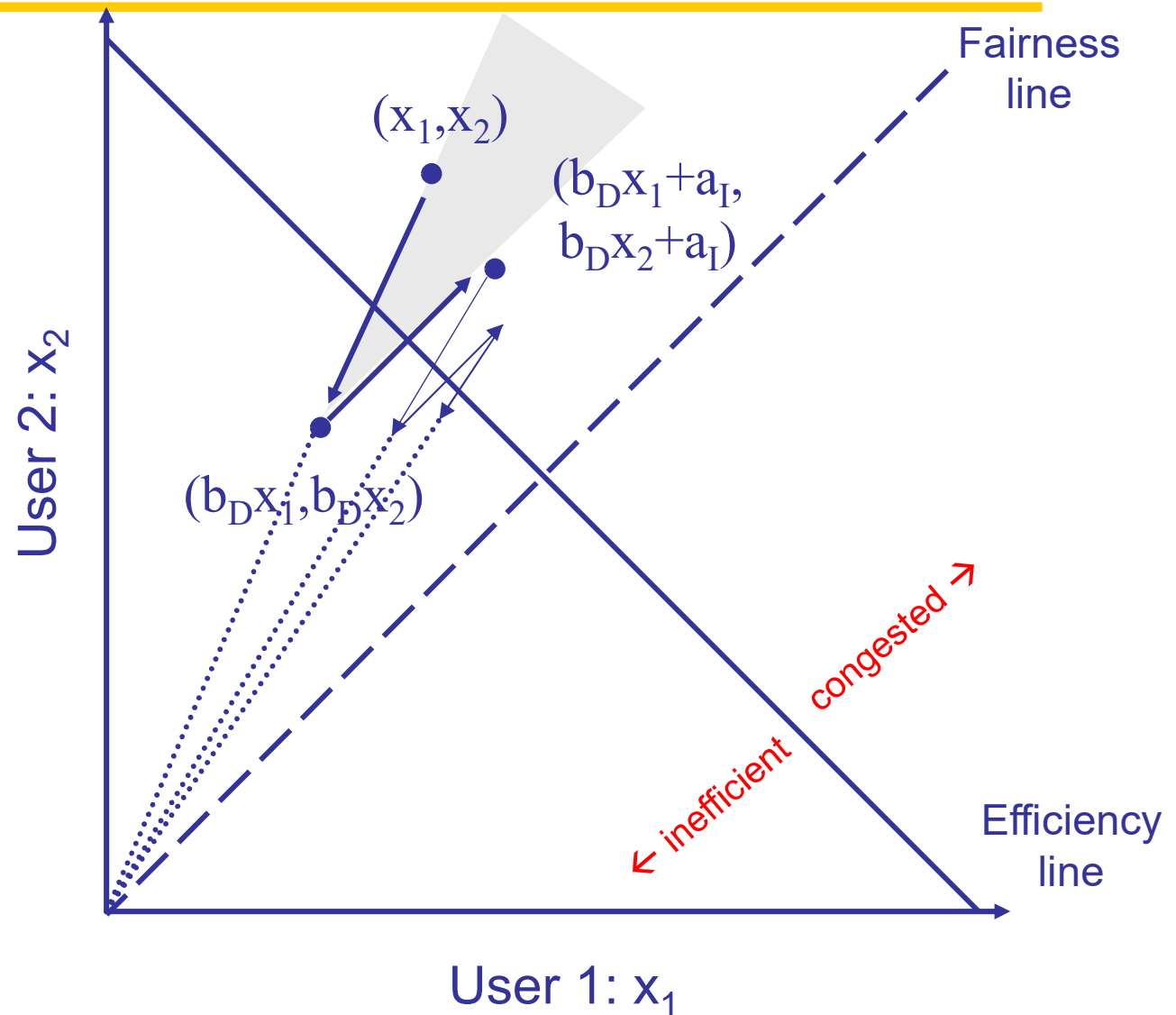
MIMD

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

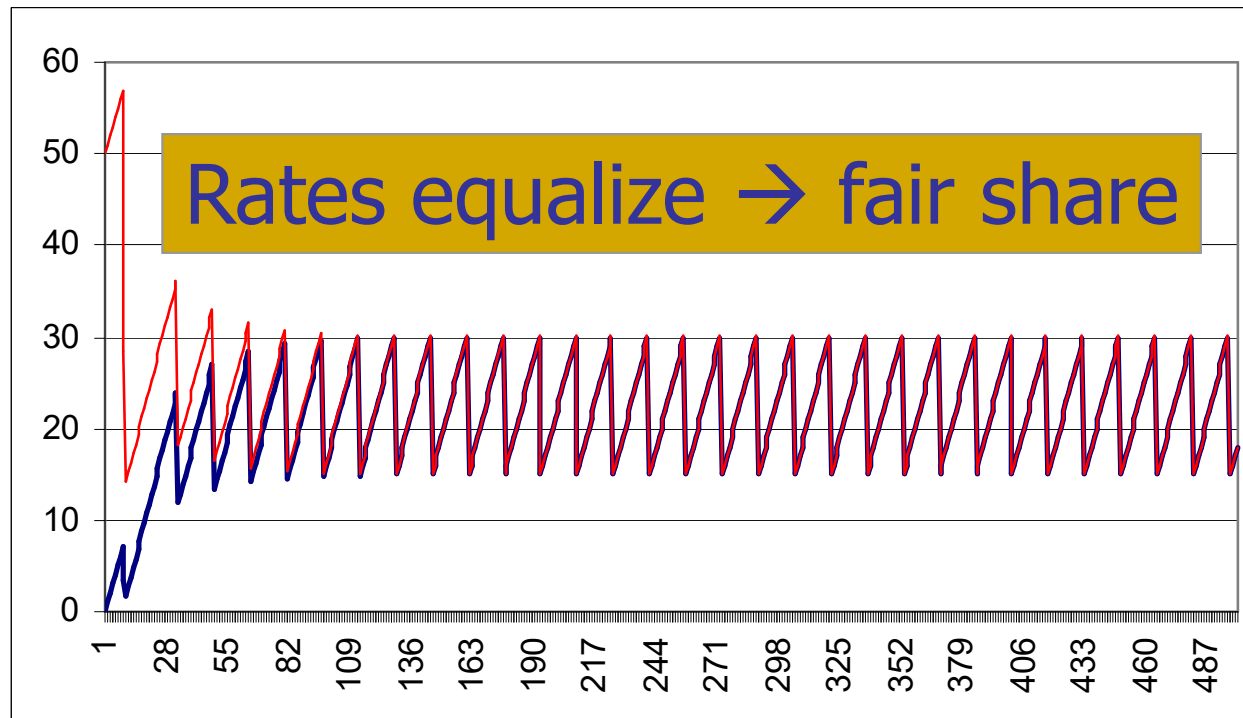


AIMD

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

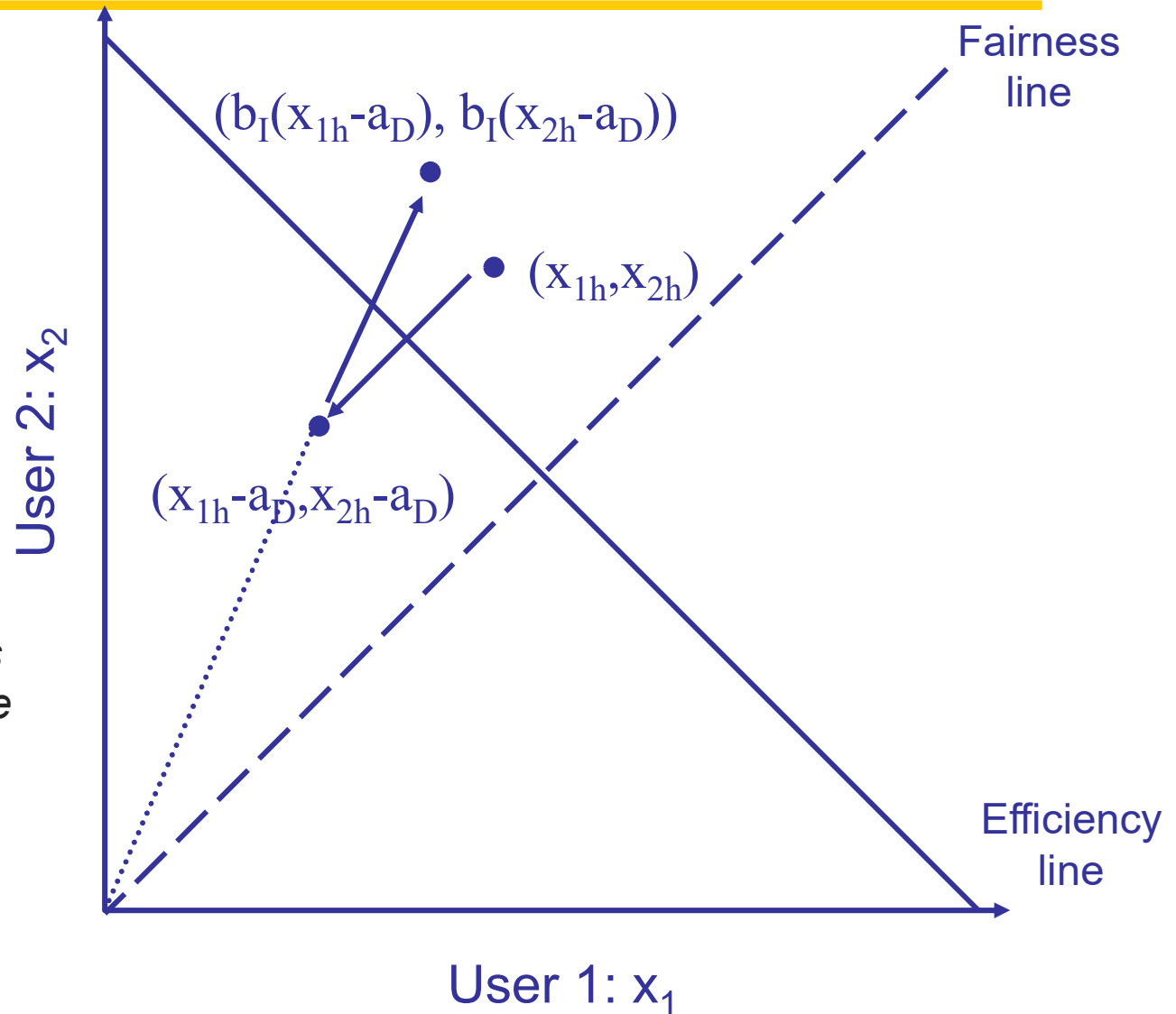


AIMD Sharing Dynamics



MIAD

- Increase: $x * b_l$
- 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