Computer Networks and Internet Technology

2021W703033 VO Rechnernetze und Internettechnik
Winter Semester 2021/22

Jan Beutel

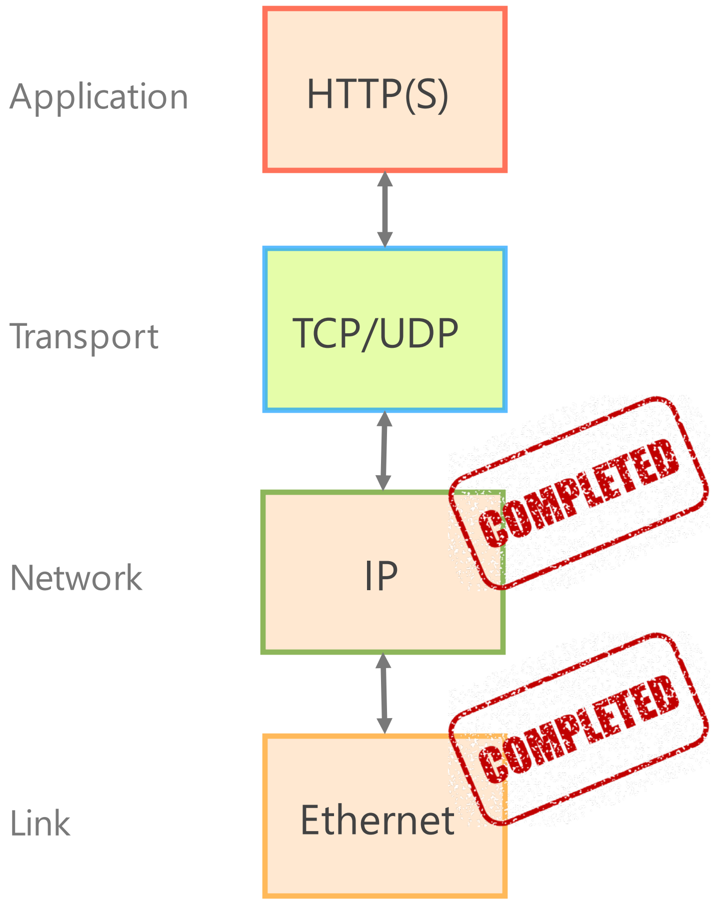
universität innsbruck

With materials and inspiration from Scott Shenker, Jennifer Rexford, Phil Levis, Adrian Perrig and Laurent Vanbever

Communication Networks and Internet Technology

Recap of last weeks lecture

We're continuing our journey up the layers, now looking at the transport layer



What do we need in the Transport layer?

- Functionality implemented in network
  - Keep minimal (easy to build, broadly applicable)

- Functionality implemented in the application
  - Keep minimal (easy to write)
  - Restricted to application-specific functionality

- Functionality implemented in the "network stack"
  - The shared networking code on the host
  - This relieves burden from both app and network
  - The transport layer is a key component here

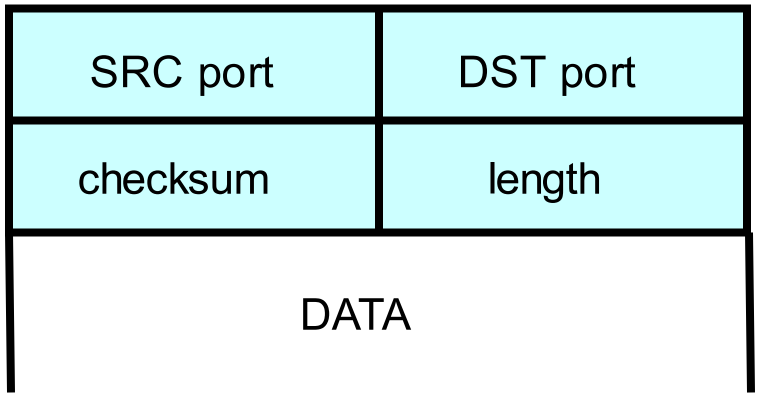
Important Context: Sockets and Ports

- **Sockets**: an operating system abstraction
- **Ports**: a networking abstraction
  - This is not a port on a switch (which is an interface)
  - Think of it as a *logical interface* on a host

UDP: User Datagram Protocol

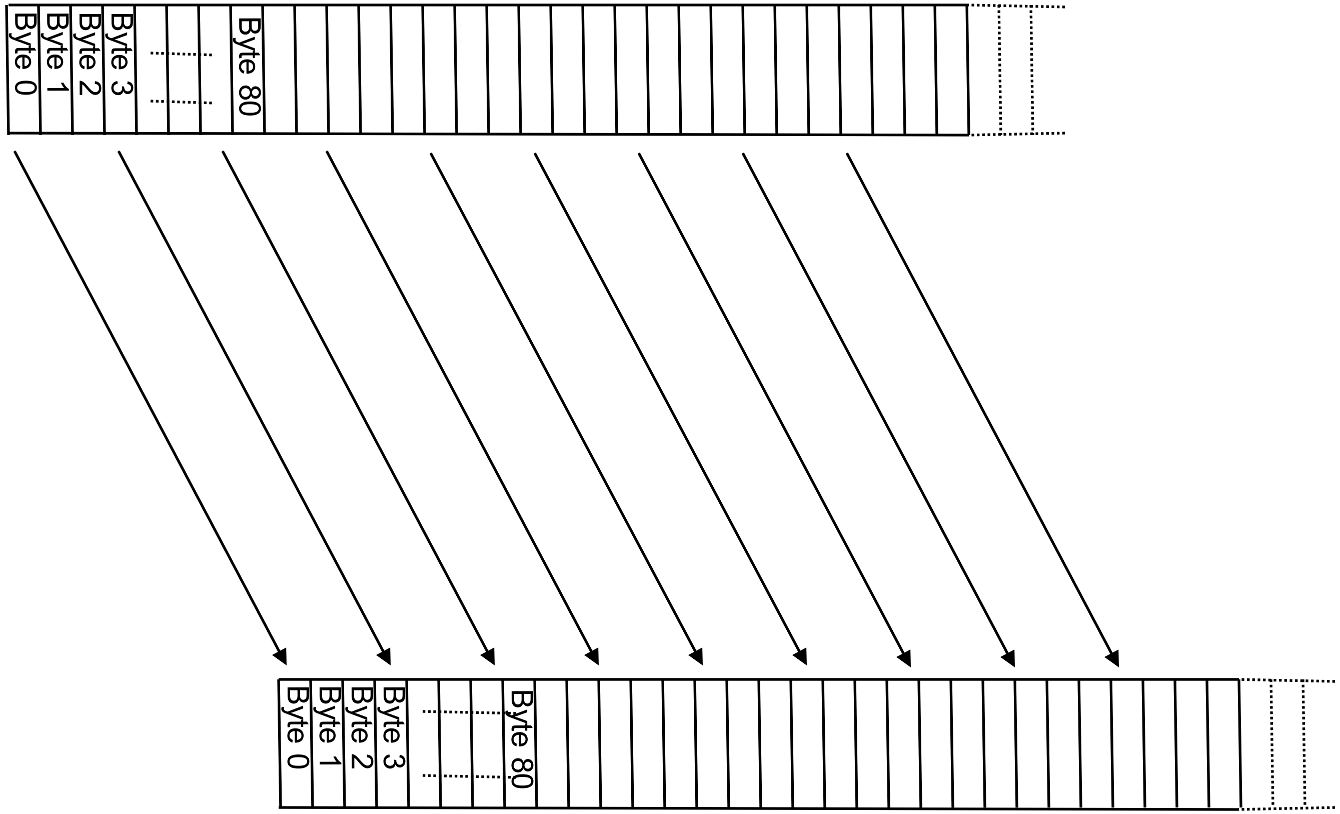
- Lightweight communication between processes
  - Avoid overhead and delays of ordered, reliable delivery
  - Send messages to and receive them from a socket

- UDP described in RFC 768 – (1980!)
  - IP plus port numbers to support (de)multiplexing
  - Optional error checking on the packet contents
    - (checksum field = 0 means “don’t verify checksum”)



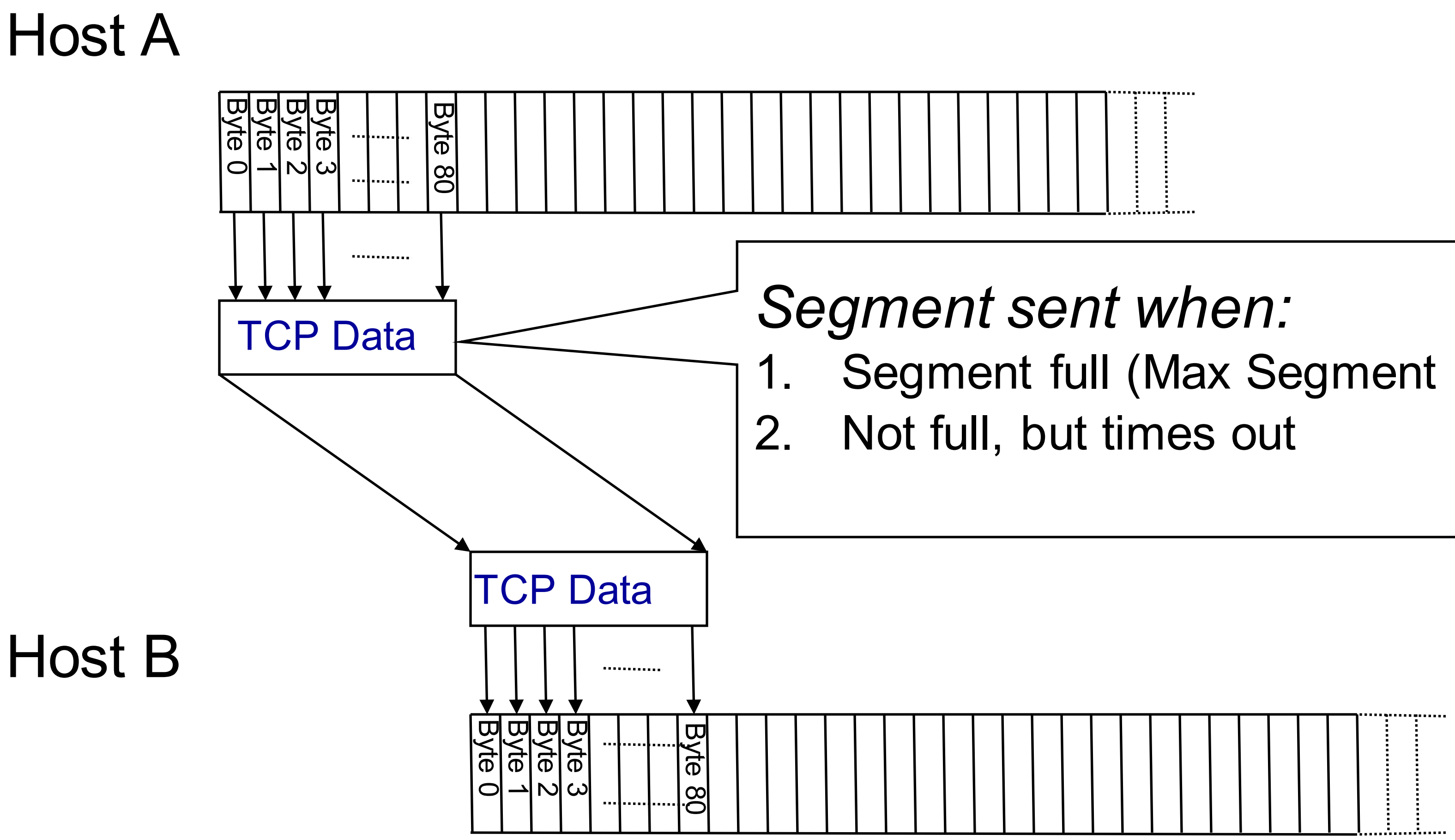
TCP "Stream of Bytes" Service…

Application @ Host A



Application @ Host B

… Provided Using TCP "Segments"

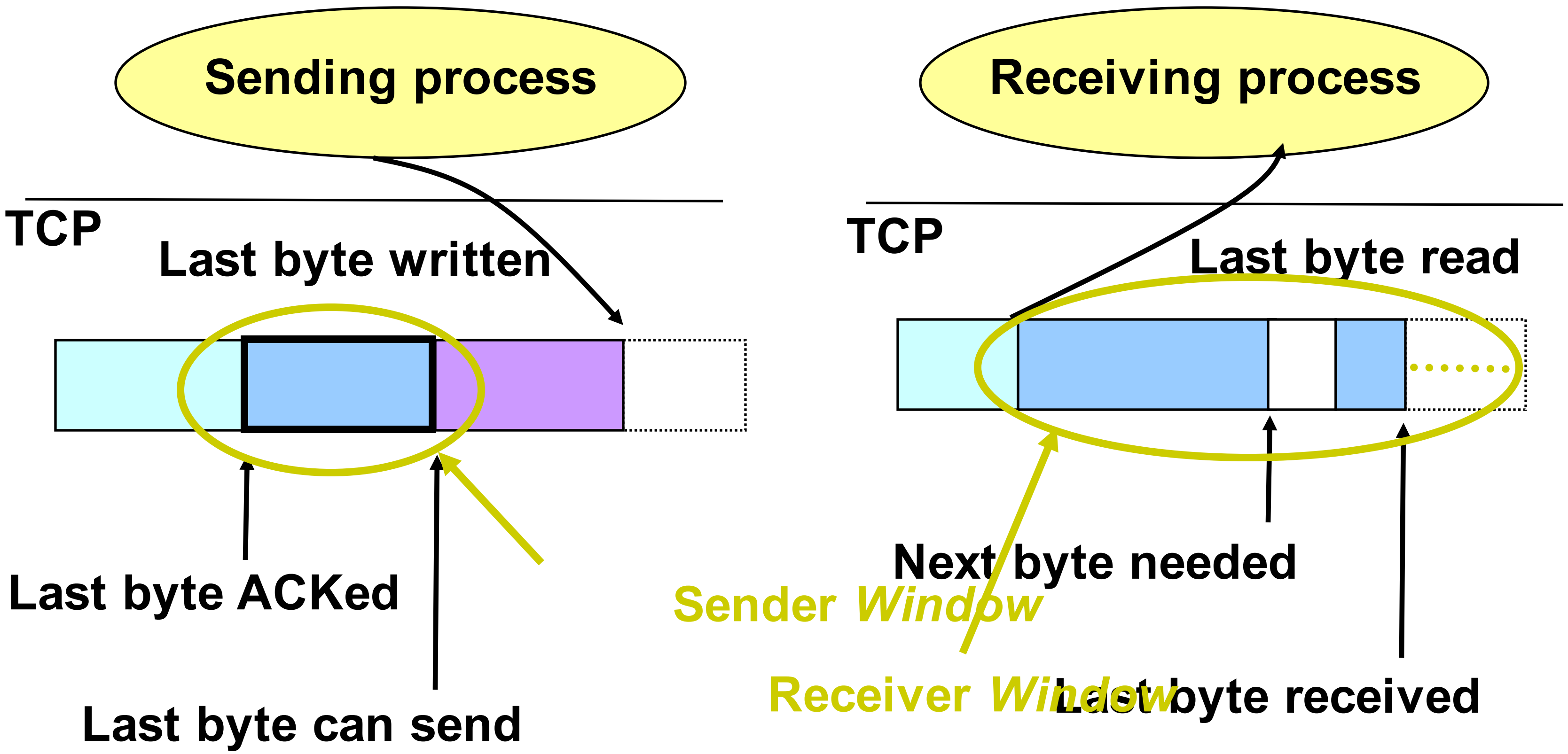


Segment sent when:

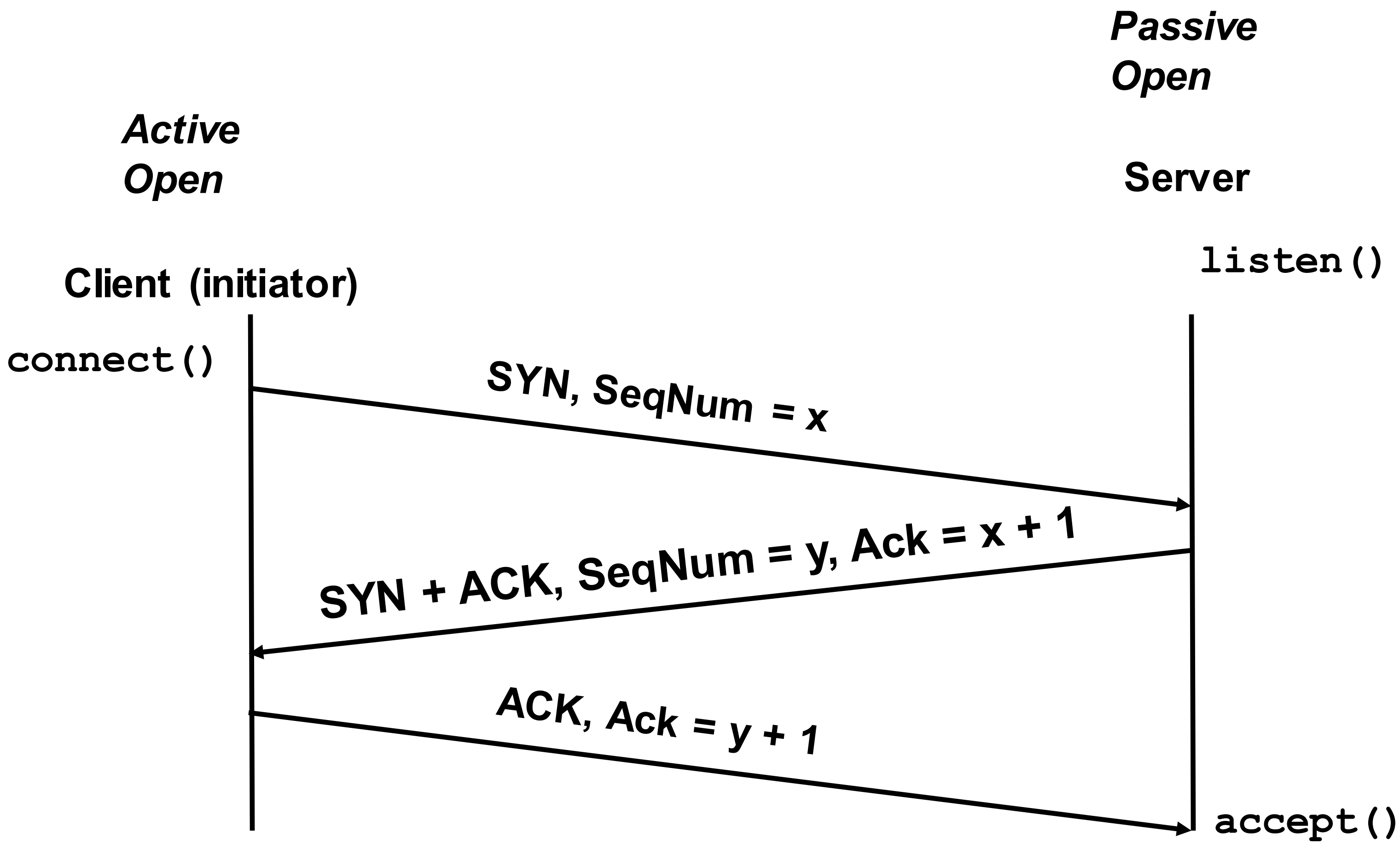
1. Segment full (Max Segment Size),
2. Not full, but times out

Sliding Window

- Allow a larger amount of data "in flight"
  - Allow sender to get ahead of the receiver
  - … though not *too far* ahead



Timing Diagram: 3-Way Handshaking



Communication Networks and Internet Technology

This weeks lecture

UDP / TCP

Congestion
Control

Un/reliable Transport

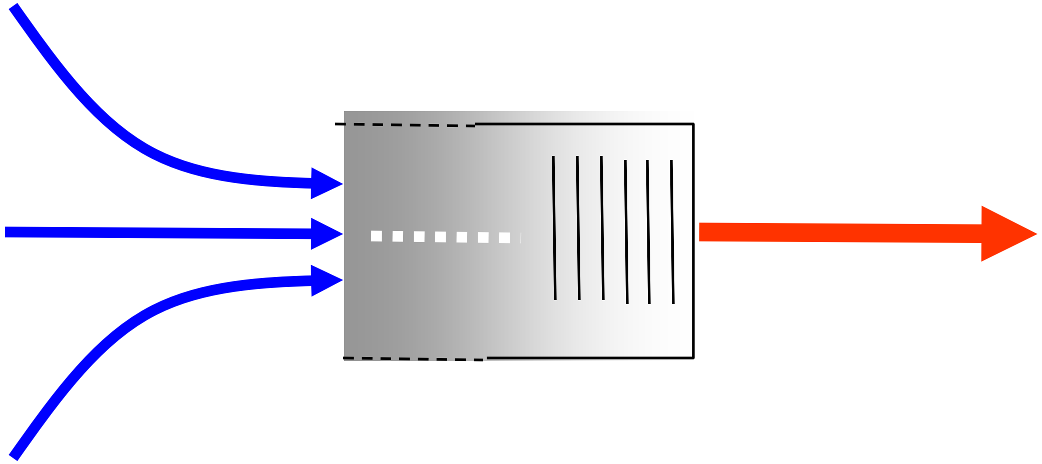
UDP / TCP

Congestion Control

Congestion is harmful

Because of traffic burstiness and lack of BW reservation,

congestion is inevitable



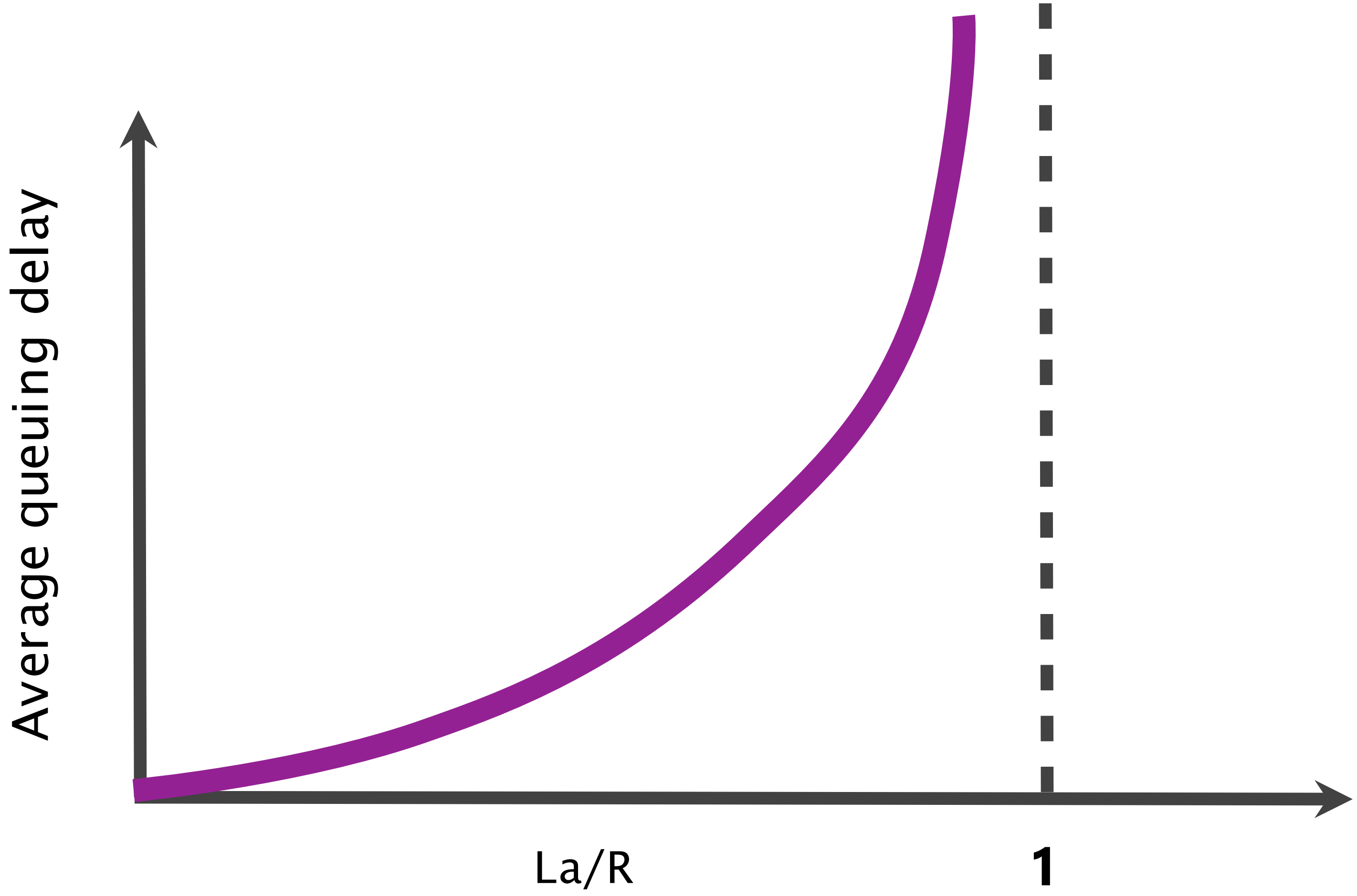
If many packets arrive within

a short period of time

the node cannot keep up anymore

average packet arrival rate	a	[packet/sec]
transmission rate of outgoing link	R	[bit/sec]
fixed packets length	L	[bit
average bits arrival rate	La	[bit/sec]
traffic intensity	La/R	

When the traffic intensity is <=1,
queueing delay depends on the burst size



When the traffic intensity is >1, the queue will increase without bound, and so does the queuing delay

Golden rule

Design your queuing system,
so that it operates far from that point

Congestion is not a new problem

The Internet almost died of congestion in 1986

throughput collapsed from 32 Kbps to... 40 bps

Van Jacobson saved us with Congestion Control

his solution went right into BSD

Recent resurgence of research interest after brief lag

new methods (ML), context (Data centers), requirements

The Internet almost died of congestion in 1986

throughput collapsed from 32 Kbps to… 40 bps

original behavior	On connection, nodes send full window of packets
	Upon timer expiration, retransmit packet immediately
meaning	sending rate only limited by flow control
net effect	window-sized burst of packets

Increase in network load results in a decrease of useful work done

Sudden load increased the round-trip time (RTT)
faster than the hosts' measurements of it

As RTT exceeds the maximum retransmission interval,
hosts begin to retransmit packets

Hosts are sending each packet several times,
eventually some copies arrive at the destination.

This phenomenon is known as congestion collapse

Knee point after which

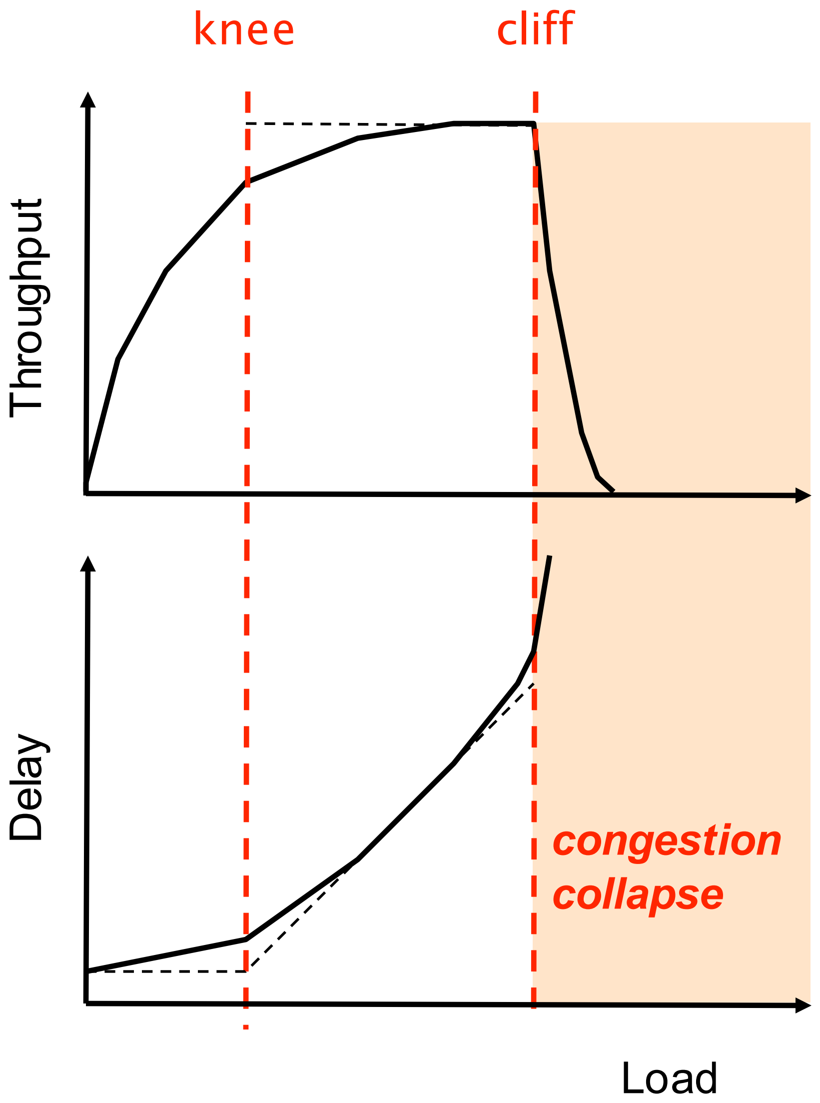
throughput increases slowly

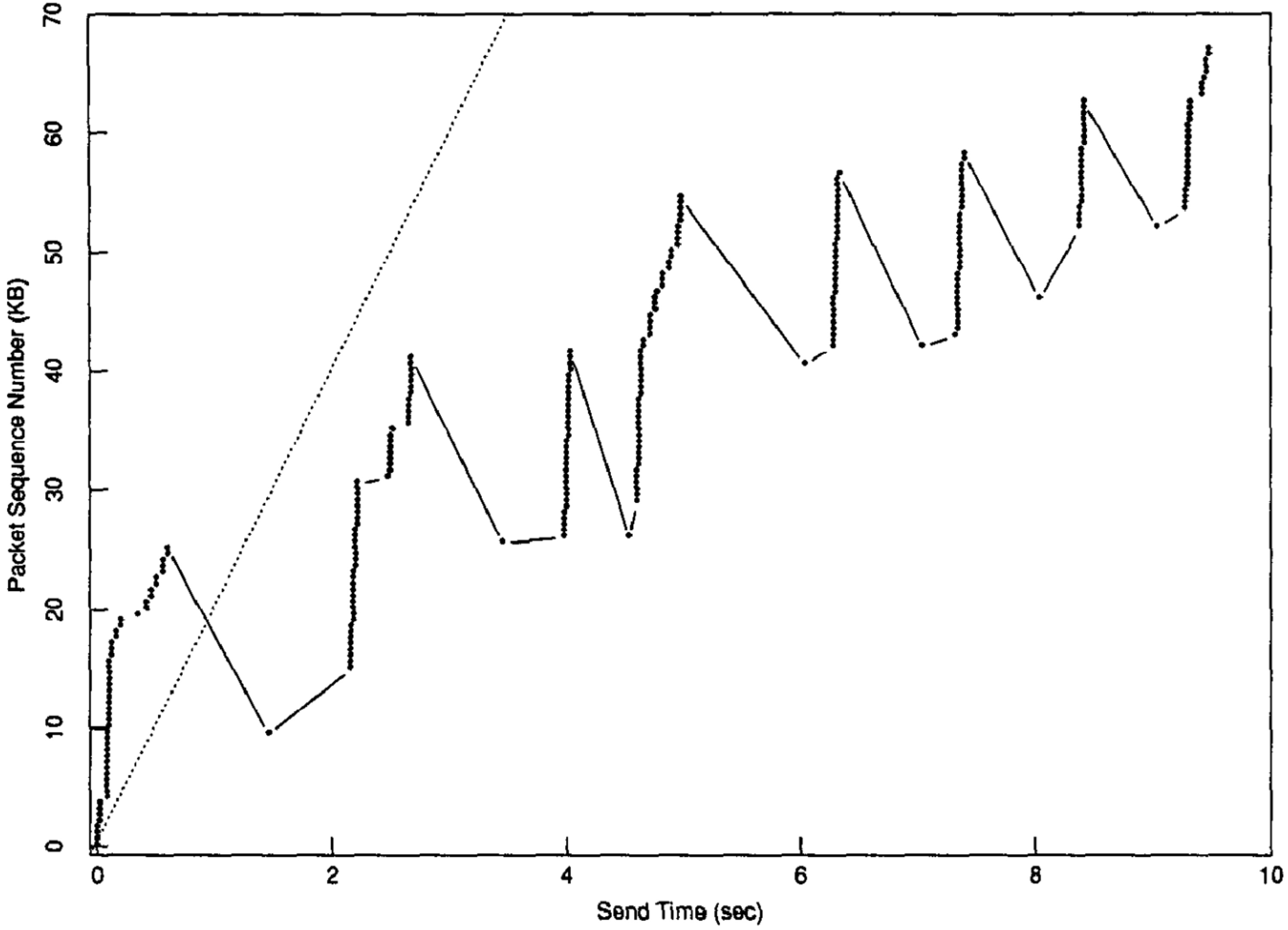
delay increases quickly

Cliff point after which

throughput decreases quickly

delay tends to infinity





Packet Sequence Number (KB)
Send Time (sec)
0
10
20
30
40
50
60
70
0
2
4
6
8
10

Van Jacobson saved us with Congestion Control

his solution went right into BSD

Congestion control aims at solving three problems

#1	bandwidth estimation	How to adjust the bandwidth of a single flow to the bottleneck bandwidth? could be 1 Mbps or 1 Gbps…
#2	bandwidth adaptation	How to adjust the bandwidth of a single flow to variation of the bottleneck bandwidth?
#3	fairness	How to share bandwidth “fairly" among flows, without overloading the network

Congestion control differs from flow control

both are provided by TCP though

Flow control	prevents one fast sender from overloading a slow receiver
Congestion control	prevents a set of senders from overloading the network

TCP solves both using two distinct windows

Flow control

prevents one fast sender from overloading a slow receiver

solved using a **receiving window**

Congestion control

prevents a set of senders from overloading the network

solved using a **“congestion” window**

The sender adapts its sending rate based on these two windows

Receiving Window RWND	How many bytes can be sent without overflowing the receiver buffer? based on the receiver input
Congestion Window CWND	How many bytes can be sent without overflowing the routers? based on network conditions
Sender Window	minimum(CWND, RWND)

The 2 key mechanisms of Congestion Control

detecting
congestion

reacting to
congestion

The 2 key mechanisms of Congestion Control

detecting
congestion

reacting to
congestion

There are essentially three ways to detect congestion

Approach #1	Network could tell the source but signal itself could be lost
Approach #2	Measure packet delay but signal is noisy delay often varies considerably
Approach #3	Measure packet loss fail-safe signal that TCP already has to detect

Packet dropping is the best solution

delay- and signaling-based methods are hard & risky

Approach #3

Measure packet loss

fail-safe signal that TCP already has to detect

Detecting losses can be done using ACKs or timeouts, the two signal differ in their degree of severity

duplicated ACKs	mild congestion signal packets are still making it
timeout	severe congestion signal multiple consequent losses

The 2 key mechanisms of Congestion Control

detecting

congestion

reacting to

congestion

TCP approach is to gently increase when not congested and to rapidly decrease when congested

question

What increase/decrease function should we use?

it depends on the problem we are solving...

Remember that Congestion Control aims at solving three problems

#1	bandwidth **estimation**	How to adjust the bandwidth of a single flow to the bottleneck bandwidth? could be 1 Mbps or 1 Gbps…
#2	bandwidth **adaptation**	How to adjust the bandwidth of a single flow to variation of the bottleneck bandwidth?
#3	**fairness**	How to share bandwidth “fairly" among flows, without overloading the network

#1 bandwidth **estimation**

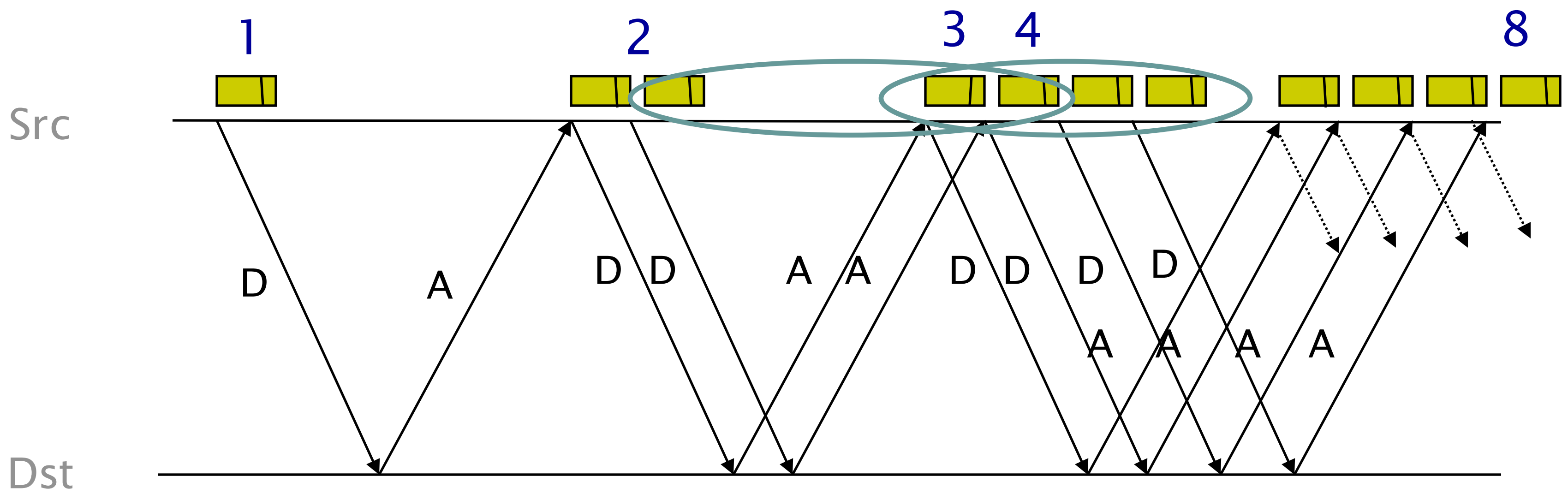
How to adjust the bandwidth of a single flow to the bottleneck bandwidth?

could be 1 Mbps or 1 Gbps…

The goal here is to quickly get a first-order estimate of the available bandwidth

Intuition	Start slow but rapidly increase until a packet drop occurs	
Increase policy	cwnd = 1	initially
	cwnd += 1	upon receipt of an ACK

This increase phase, known as slow start,
corresponds to an… exponential increase of CWND!



slow start is called like this only because of starting point

The problem with slow start is that it can result in a full window of packet losses

Example	Assume that CWND is just enough to “fill the pipe”
	After one RTT, CWND has doubled
	All the excess packets are now dropped
Solution	We need a more gentle adjustment algorithm once we have a rough estimate of the bandwidth

#2	bandwidth adaptation	How to adjust the bandwidth of a single flow to variation of the bottleneck bandwidth?

The goal here is to track the available bandwidth, and oscillate around its current value

Two possible variations

- **M**ultiplicative **I**ncrease or **D**ecrease

  cwnd = a * cwnd

- **A**dditive **I**ncrease or **D**ecrease

  cwnd = b + cwnd

… leading to four alternative design

	increase behavior	decrease behavior
AIAD	gentle	gentle
AIMD	gentle	aggressive
MIAD	aggressive	gentle
MIMD	aggressive	aggressive

To select one scheme, we need to consider the 3rd problem: fairness

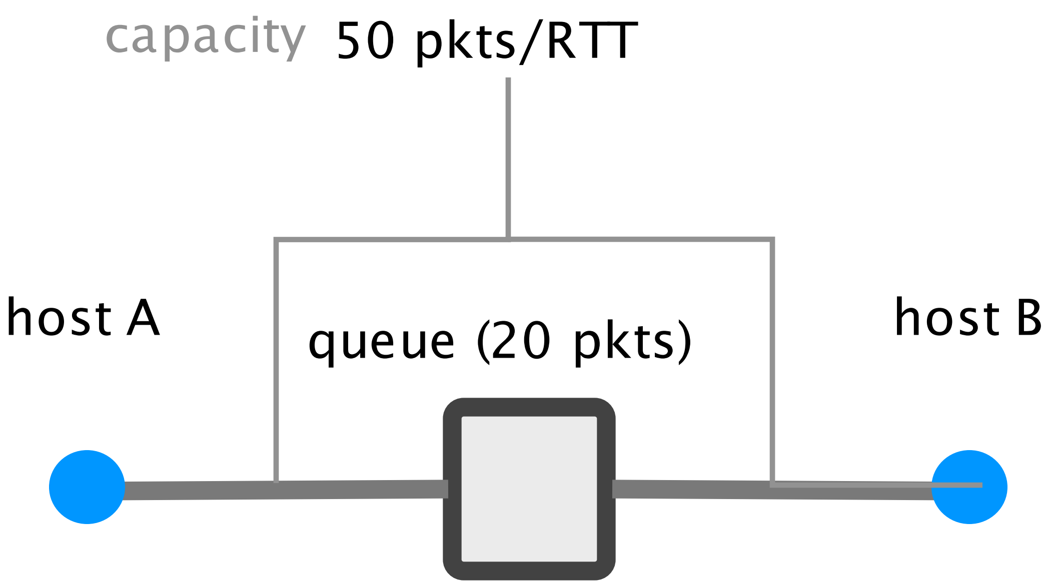
	increase behavior	decrease behavior
AIAD	gentle	gentle
AIMD	gentle	aggressive
MIAD	aggressive	gentle
MIMD	aggressive	aggressive

#3 fairness

How to share bandwidth “fairly" among flows, without overloading the network

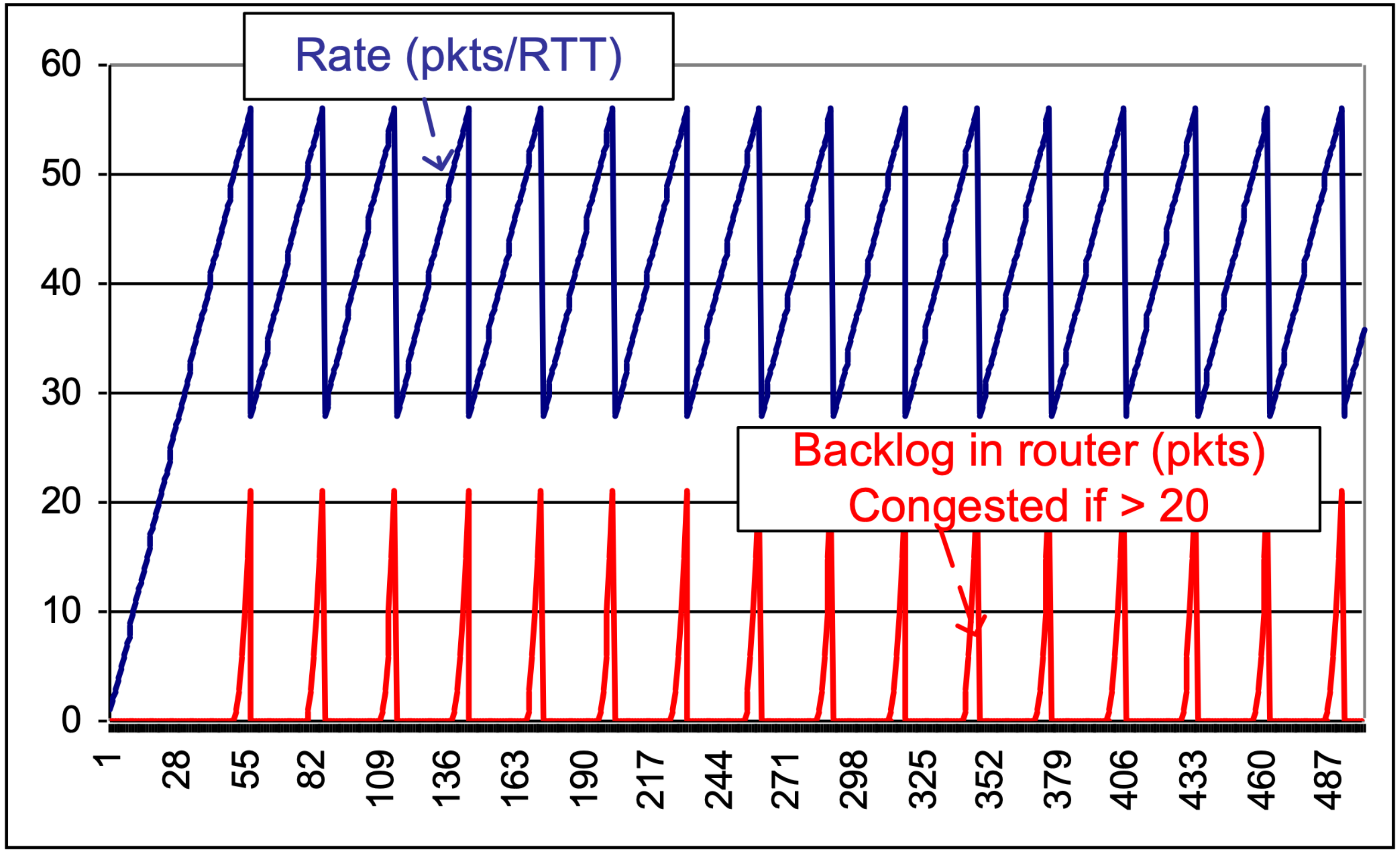
TCP notion of fairness: 2 identical flows should end up with the same bandwidth

Consider first a single flow between A and B and AIMD

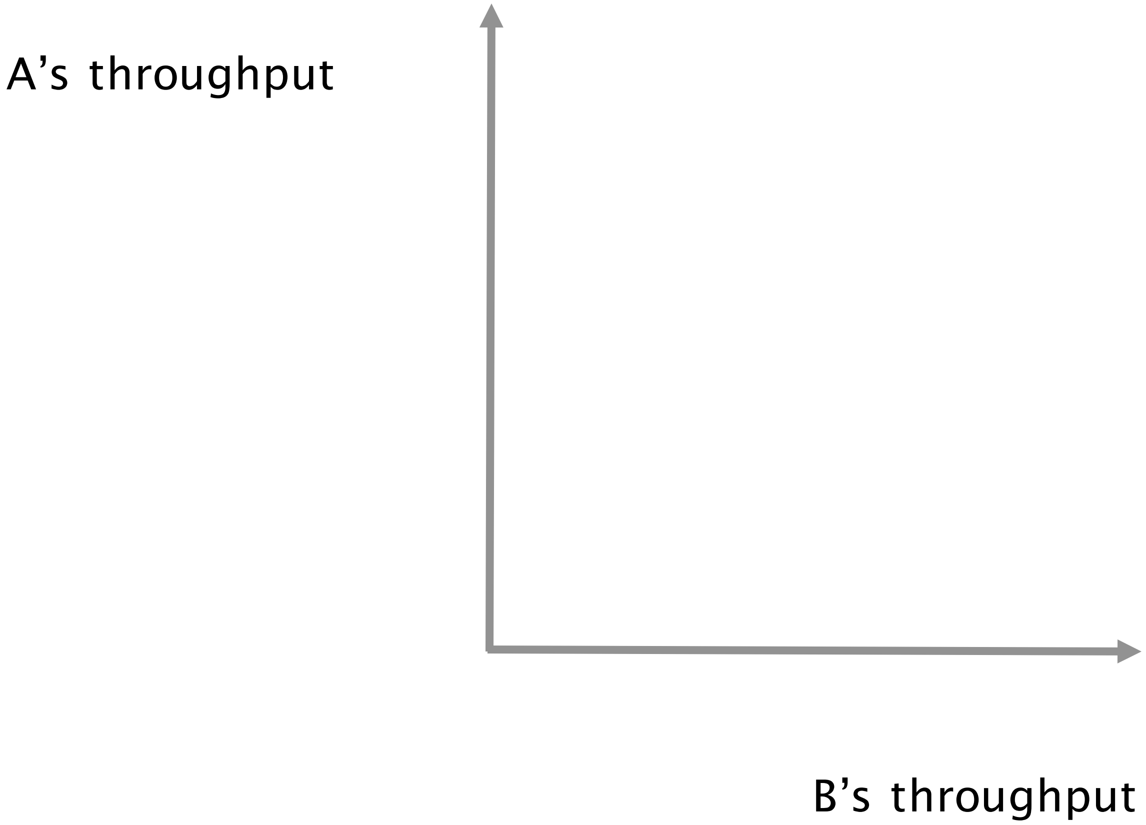


without congestion CWND increases by one packet every ACK

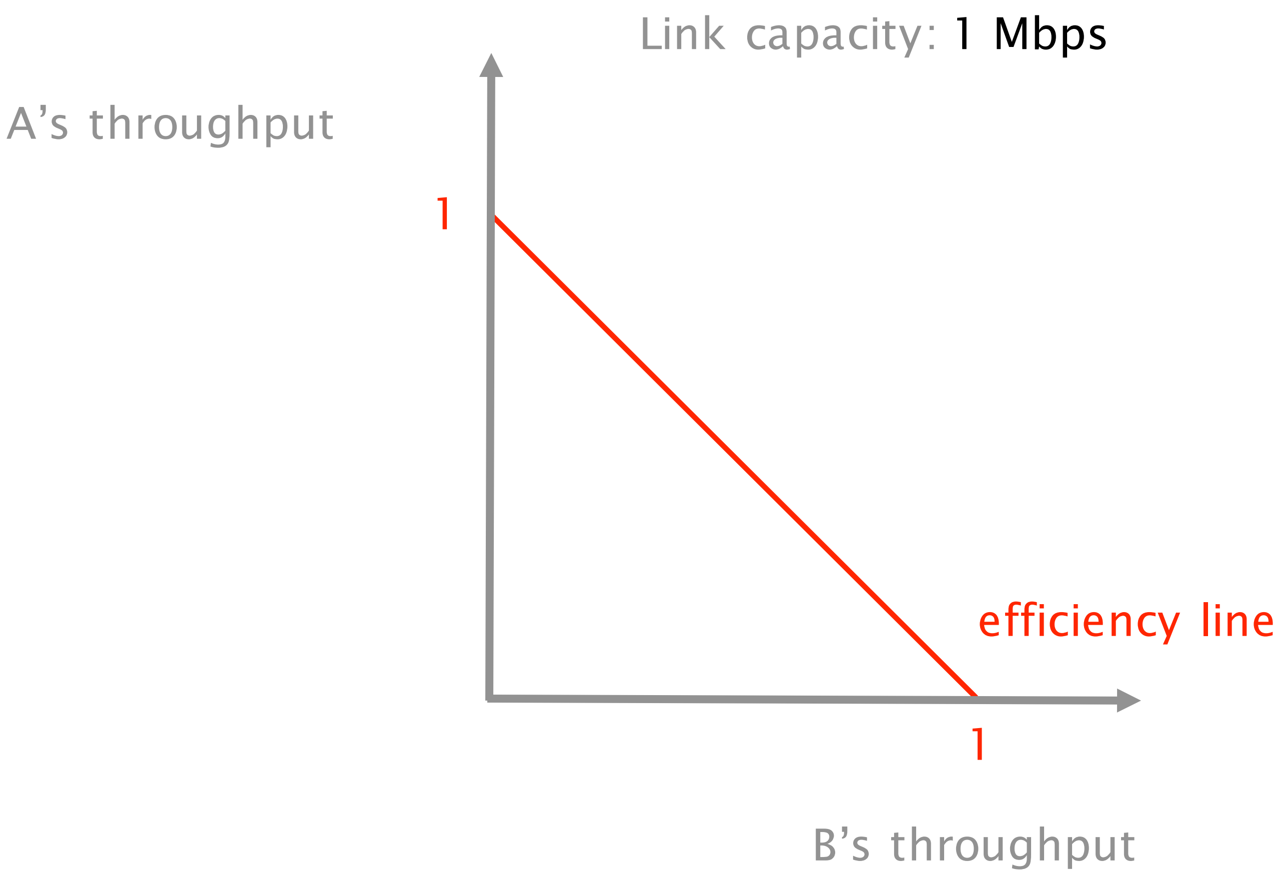
upon congestion CWND decreases by a factor 2


Rate (pkts/RTT)
Backlog in router (pkts)
Congested if > 20
60
50
40
30
20
10
0
1
28
55
82
109
136
163
190
217
244
271
298
325
352
379
406
433
460
487

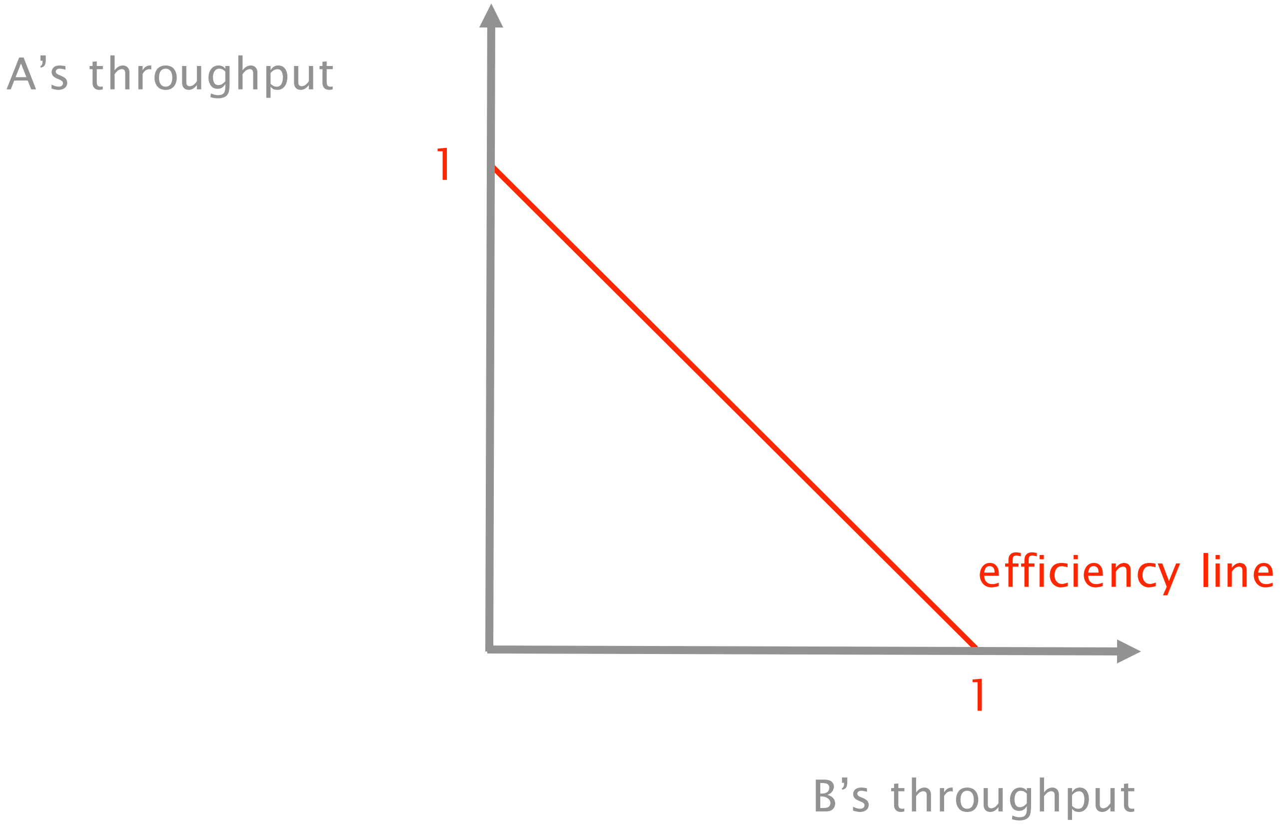
We can analyze the system behavior using a system trajectory plot

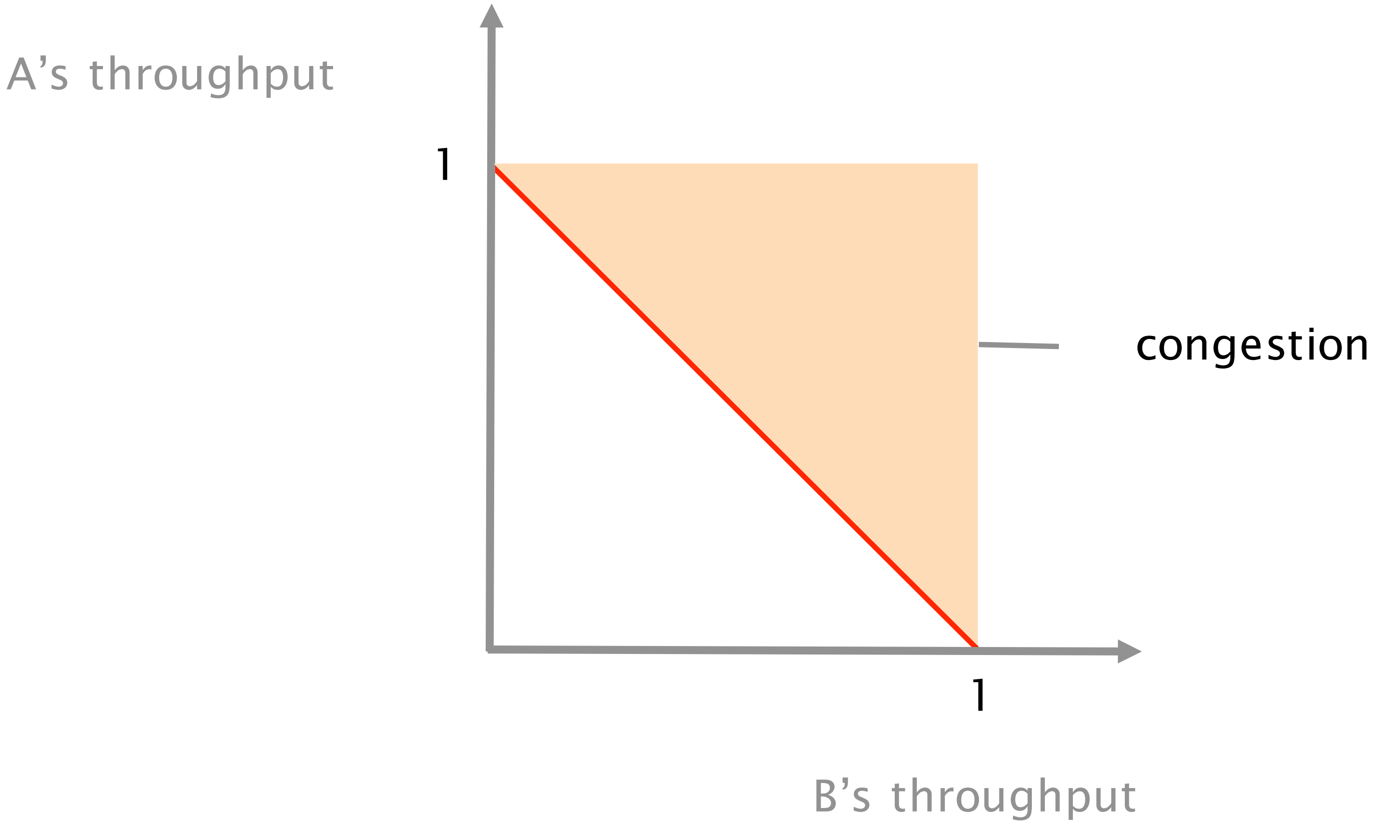


The system is efficient if the capacity is fully used, defining an efficiency line where $a + b = 1$

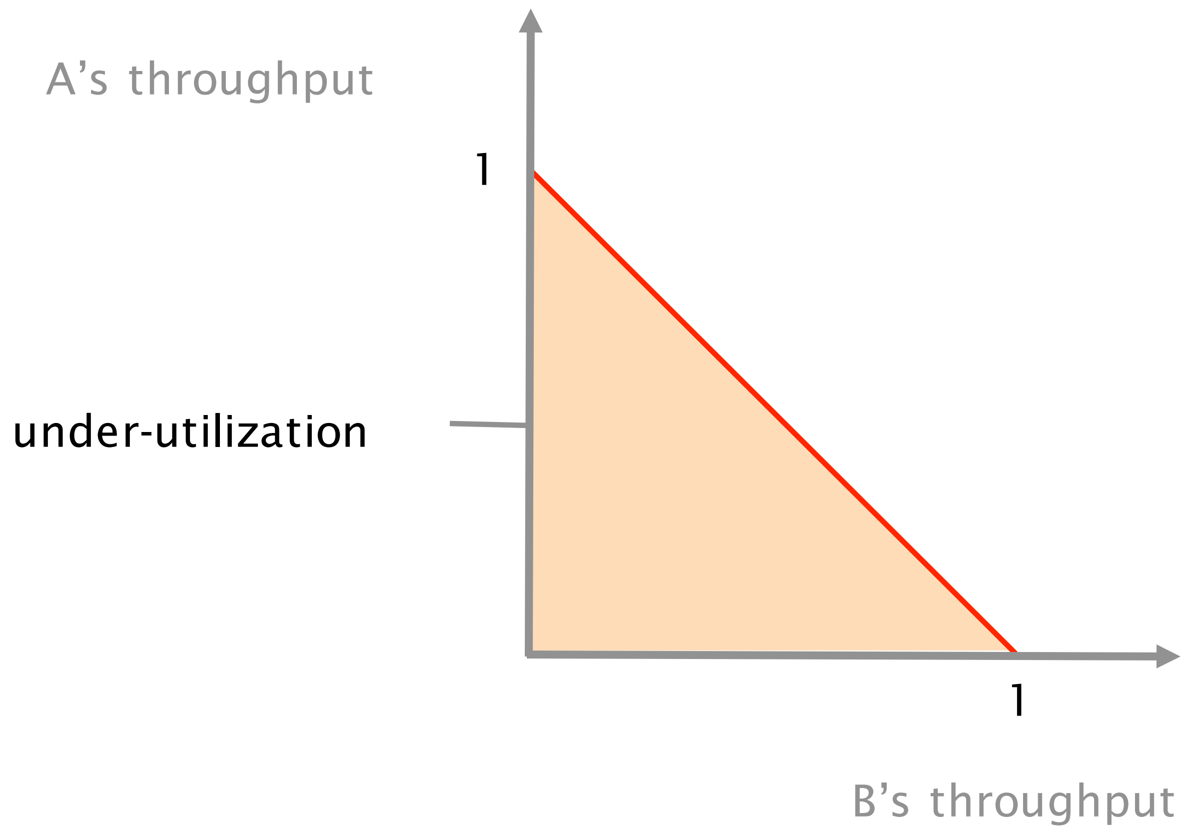


The goal of congestion control is to bring the system as close as possible to this line, and stay there

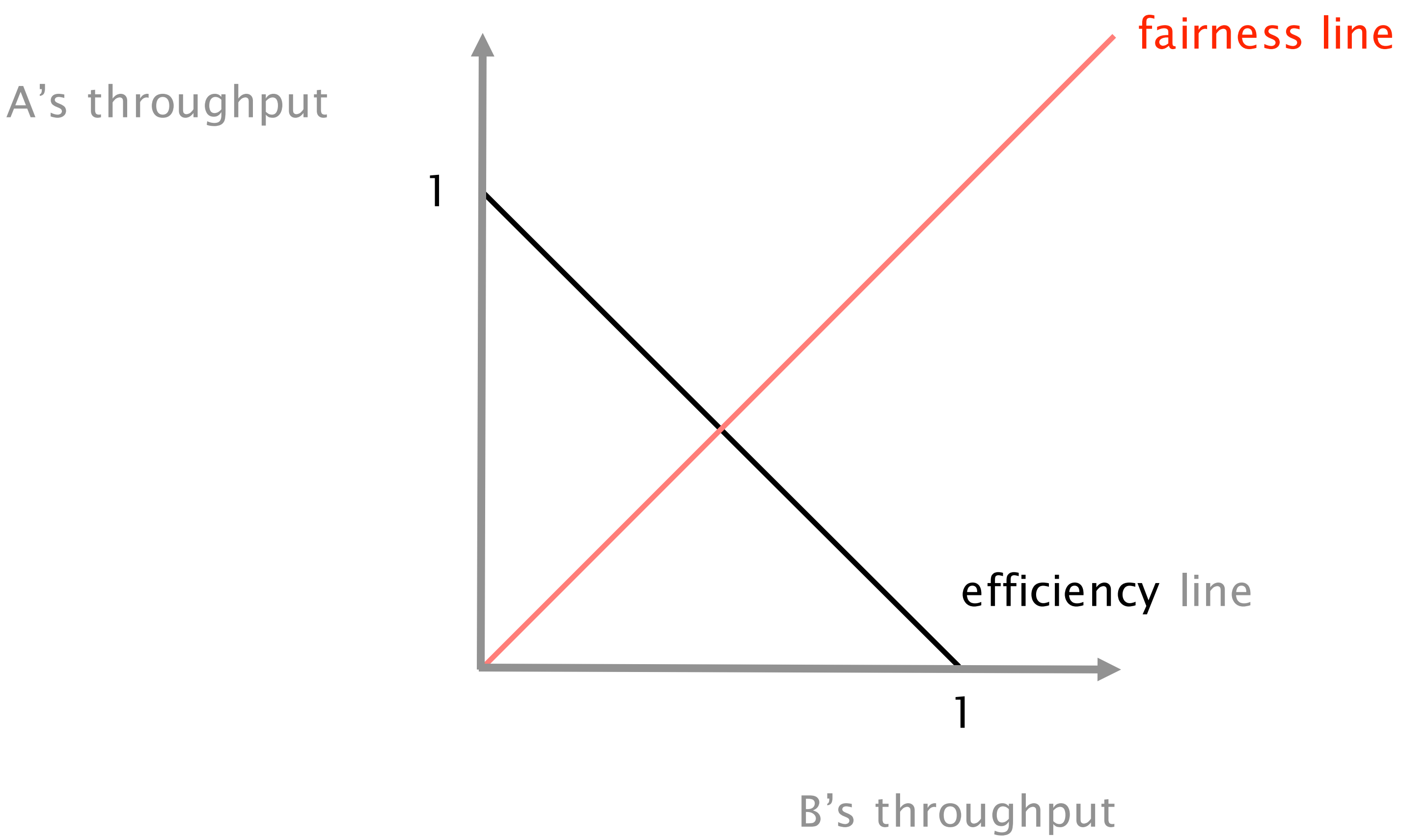


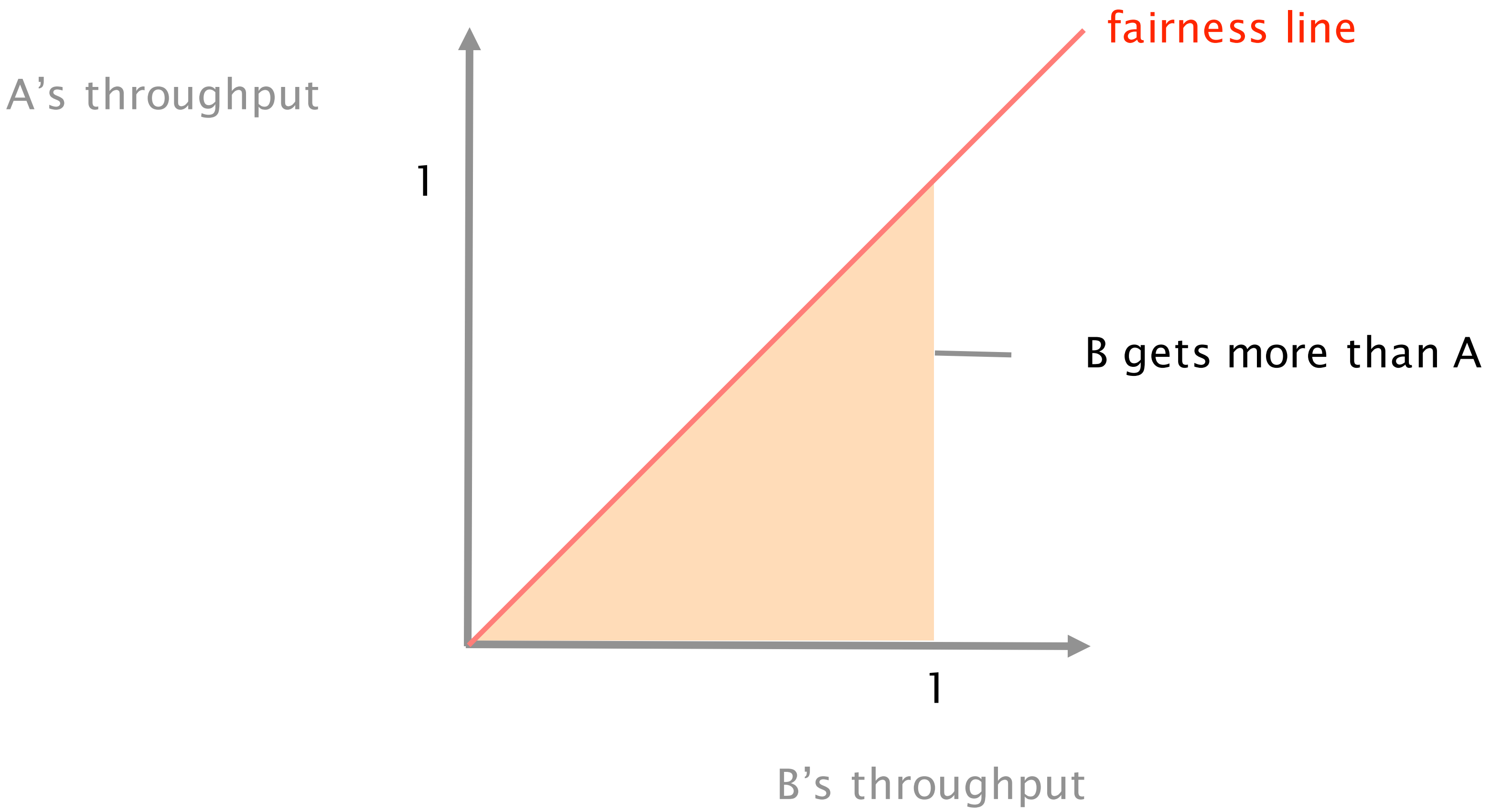


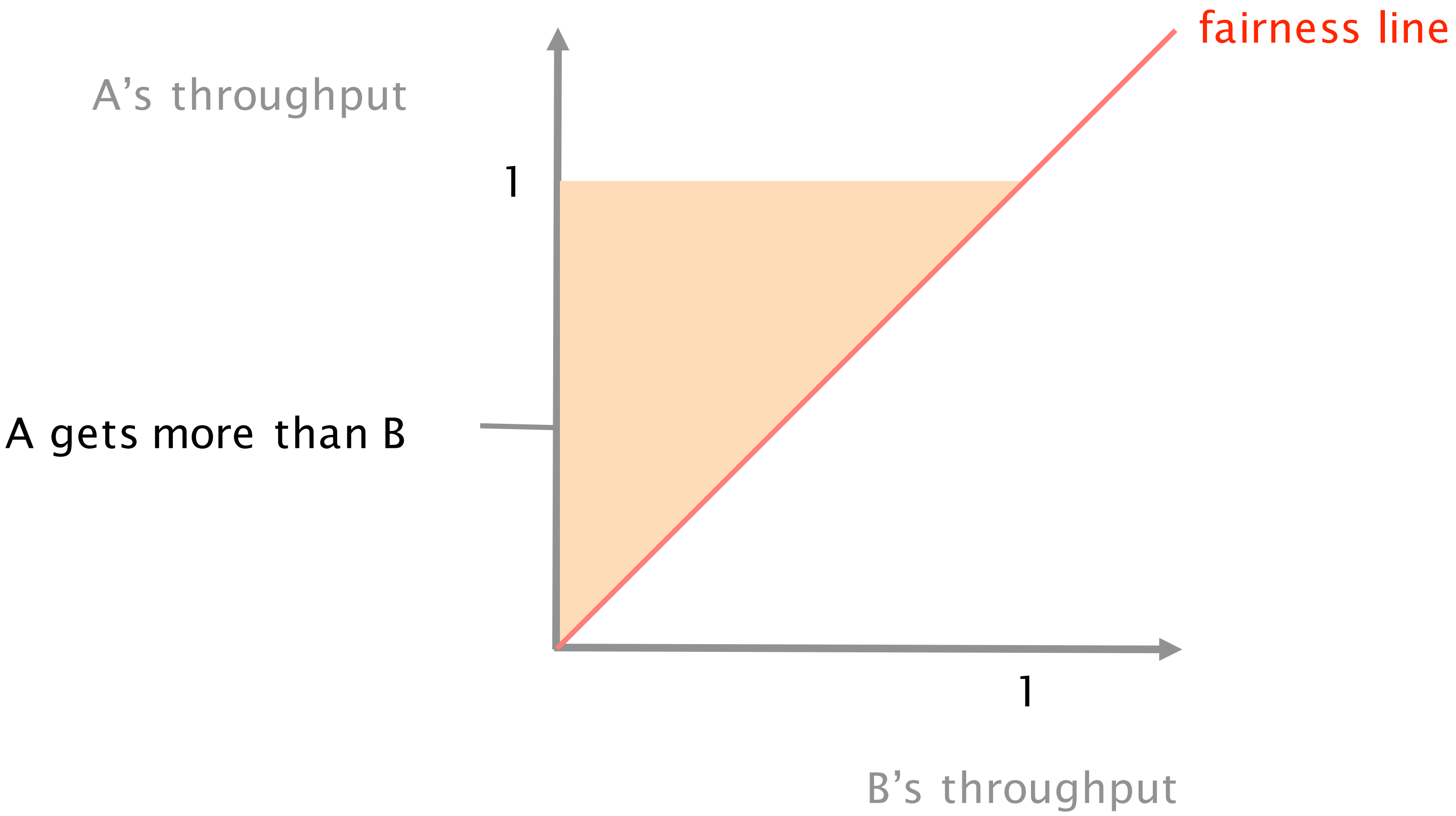
A’s throughput
1
congestion
1
B’s throughput

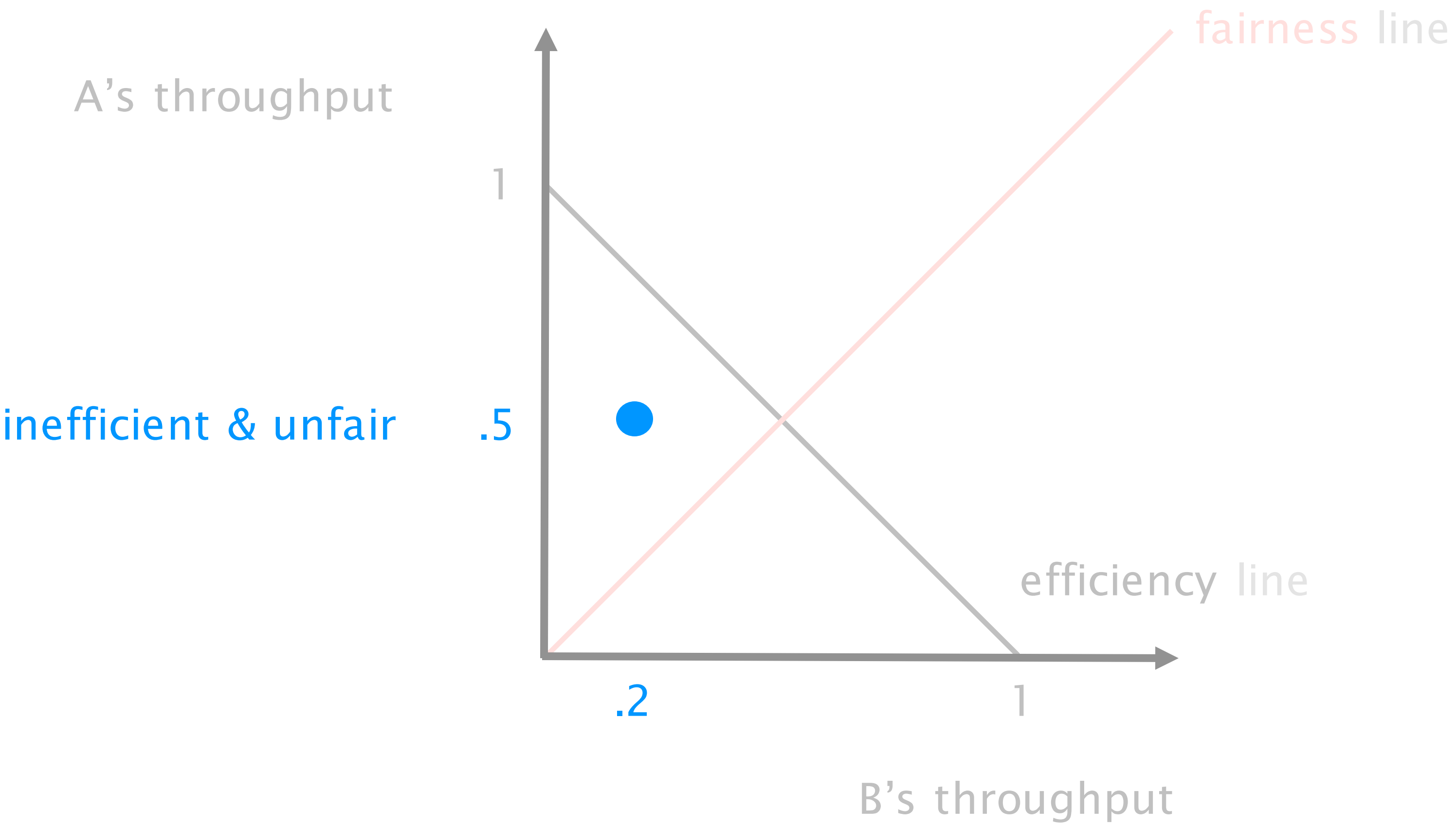

A’s throughput
1
under-utilization
1
B’s throughput

The system is fair whenever A and B have equal throughput, defining a fairness line where $a = b$

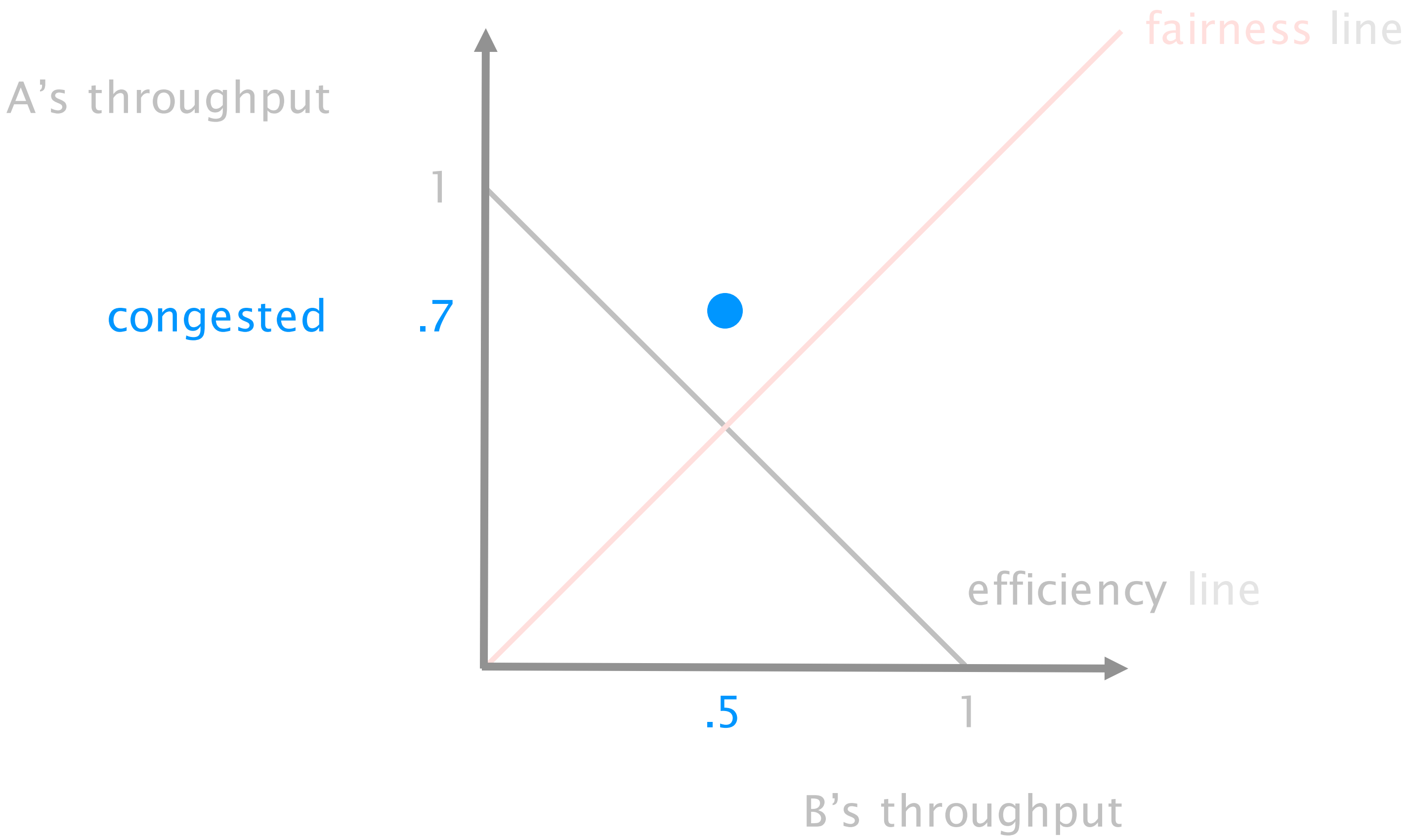



fairness line
A's throughput
1
B gets more than A
1
B's throughput

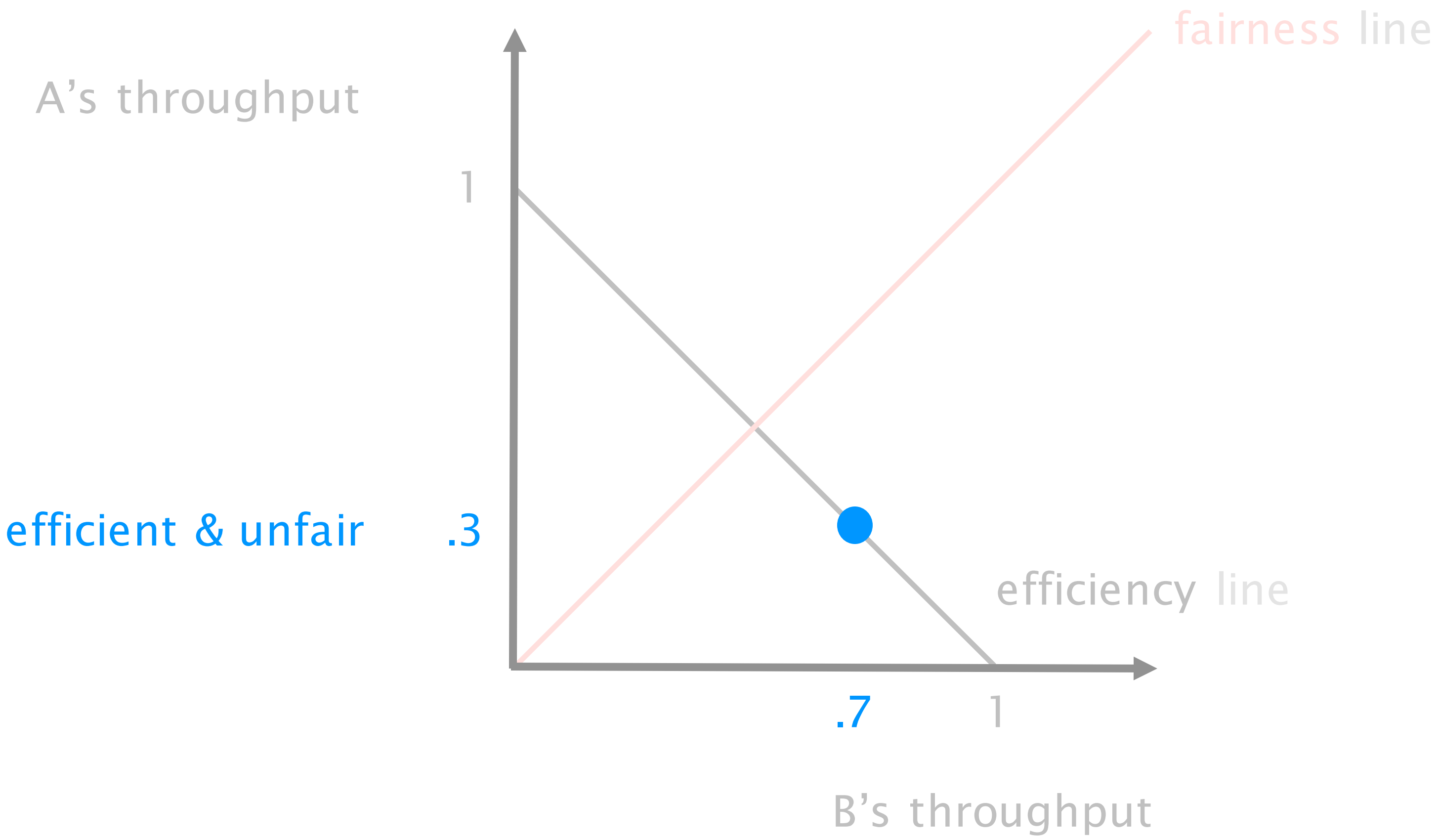

fairness line
A's throughput
1
A gets more than B
1
B's throughput

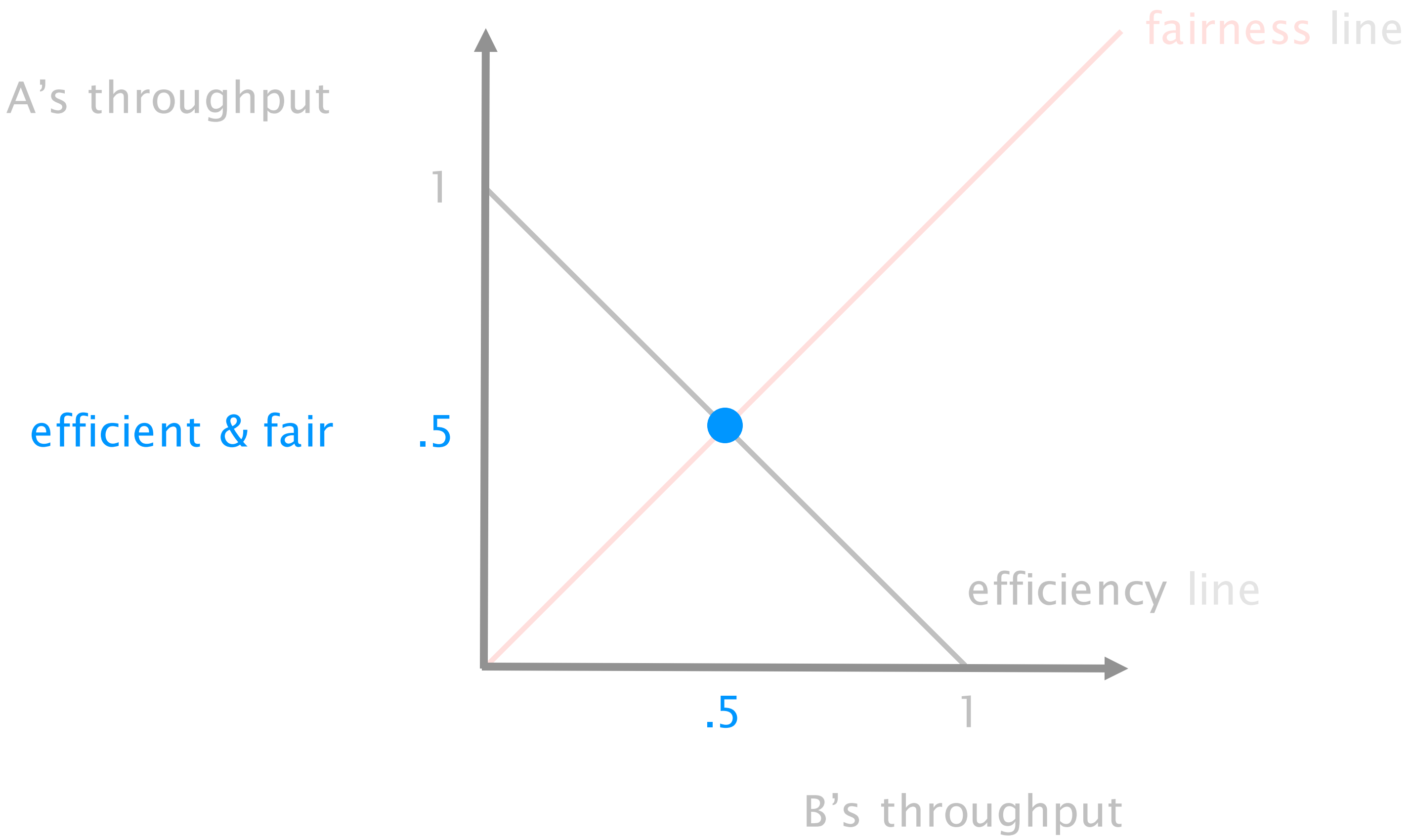


fairness line
A's throughput
1
inefficient & unfair
.5
efficiency line
.2
1
B's throughput



fairness line
A's throughput
1
congested
.7
efficiency line
.5
1
B's throughput

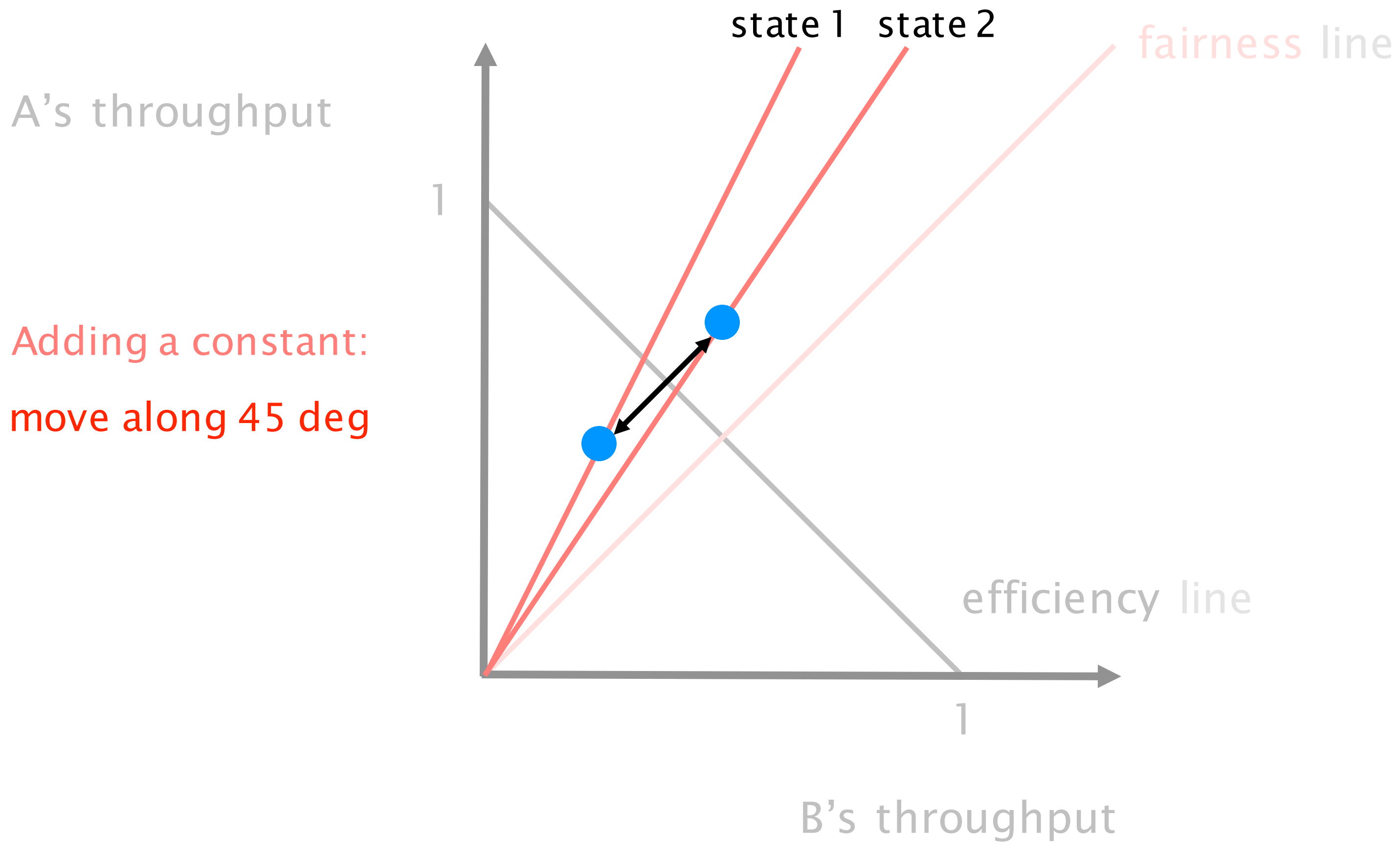

fairness line
A's throughput
1
efficient & unfair
.3
efficiency line
.7
1
B's throughput


fairness line
A's throughput
1
efficient & fair
.5
efficiency line
.5
1
B's throughput

	increase behavior	decrease behavior
AIAD	gentle	gentle
AIMD	gentle	aggressive
MIAD	aggressive	gentle
MIMD	aggressive	aggressive

AIAD does not converge to fairness, nor efficiency:

the system fluctuates between two fairness states



AIAD does not converge to fairness, nor efficiency:

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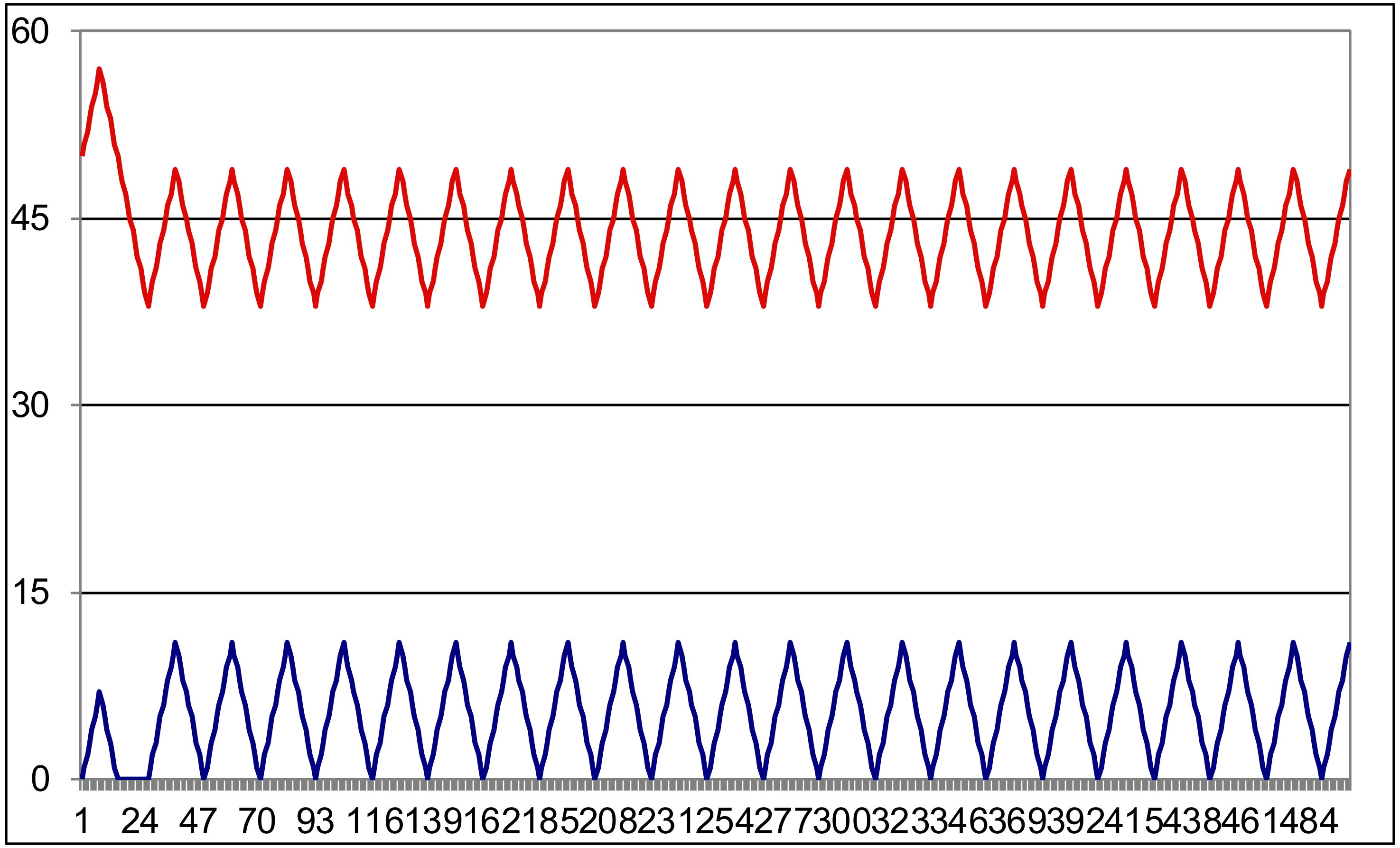
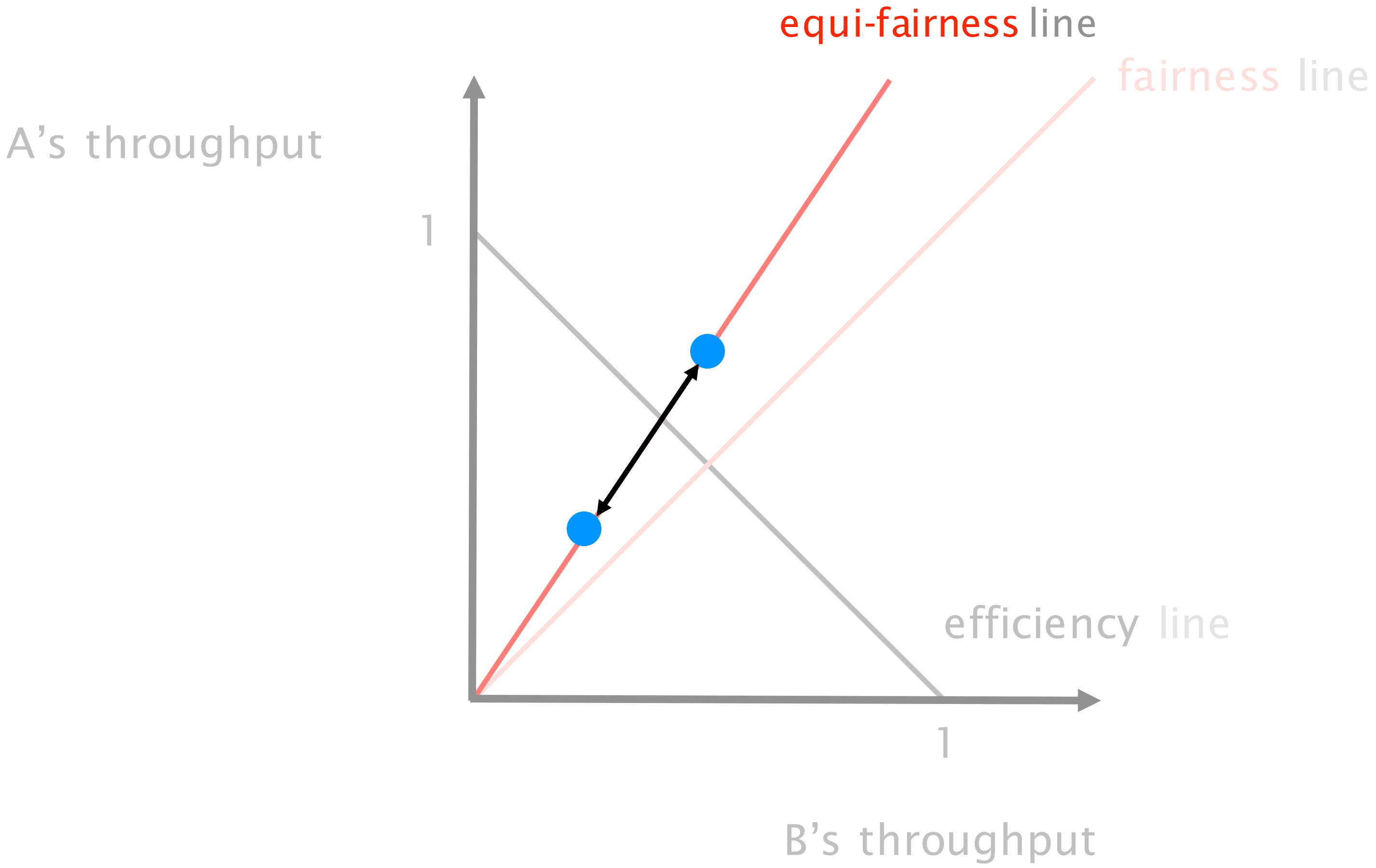


	increase behavior	decrease behavior
AIAD	gentle	gentle
AIMD	gentle	aggressive
MIAD	aggressive	gentle
MIMD	aggressive	aggressive

MIMD does not converge to fairness, nor efficiency:

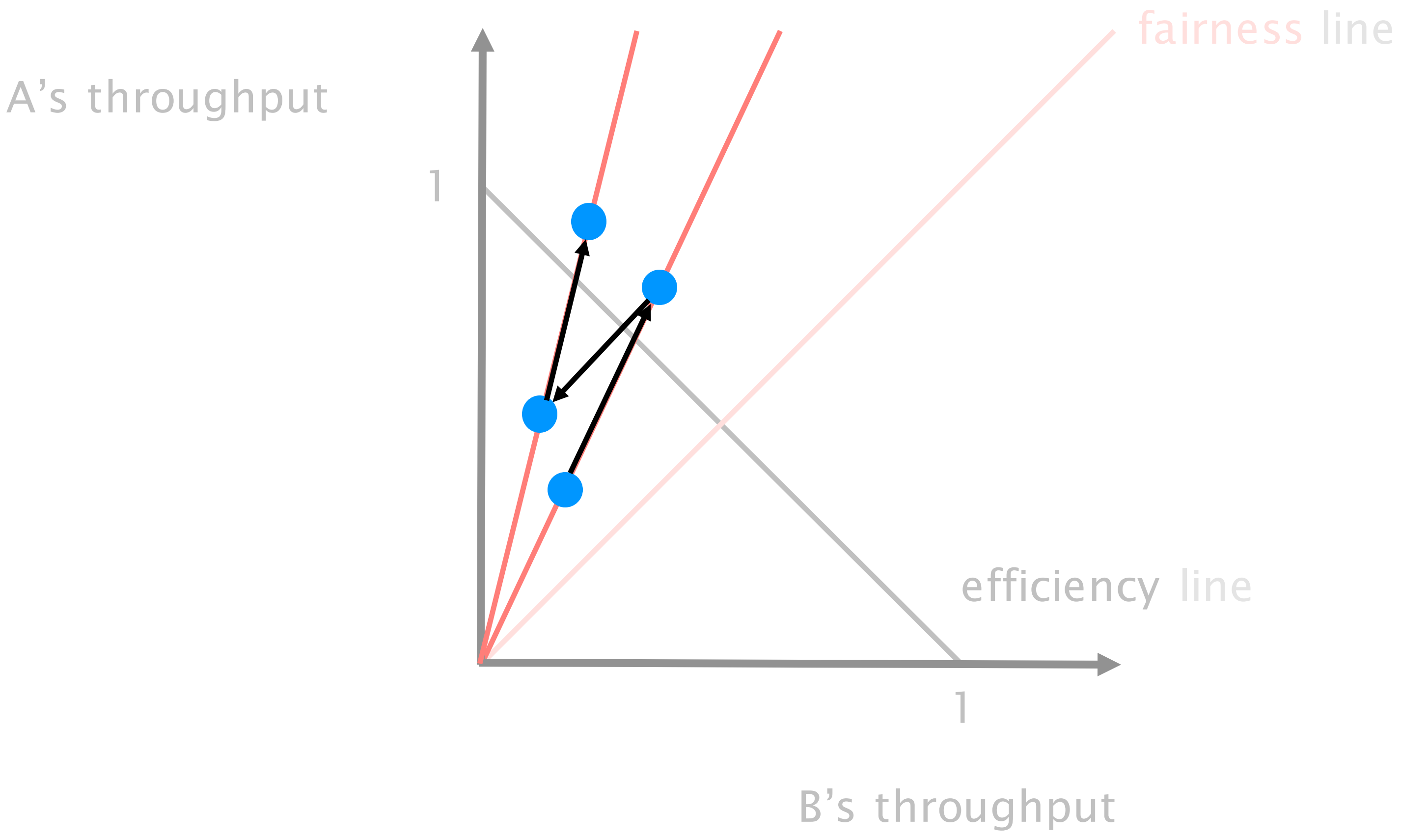
the system fluctuates along a equi-fairness line



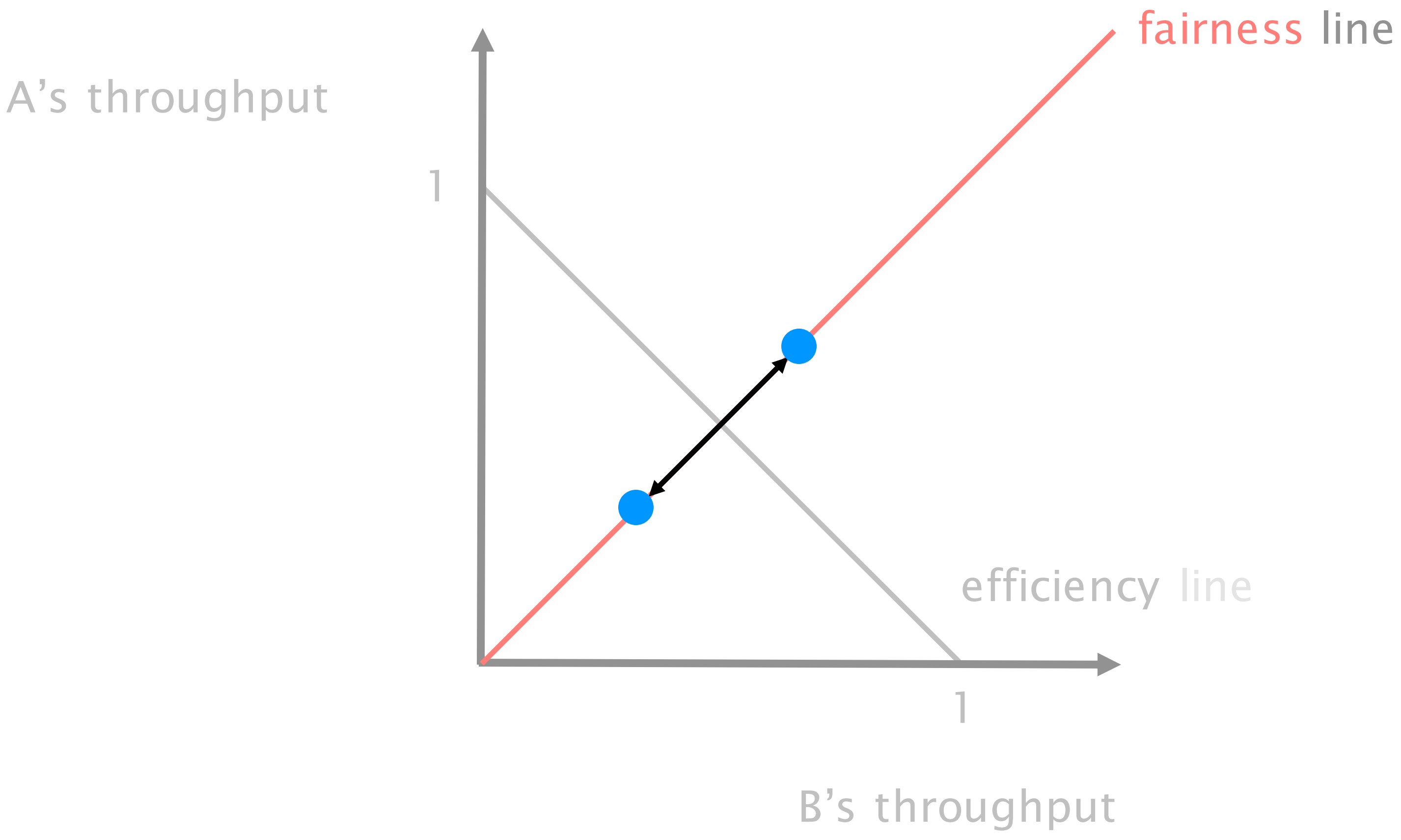
	increase behavior	decrease behavior
AIAD	gentle	gentle
AIMD	gentle	aggressive
MIAD	aggressive	gentle
MIMD	aggressive	aggressive

MIAD converges to a totally unfair allocation,

favoring the flow with a greater rate at the beginning



If flows start along the fairness line, MIAD fluctuates along it, yet deviating from it at the slightest change



	increase behavior	decrease behavior
AIAD	gentle	gentle
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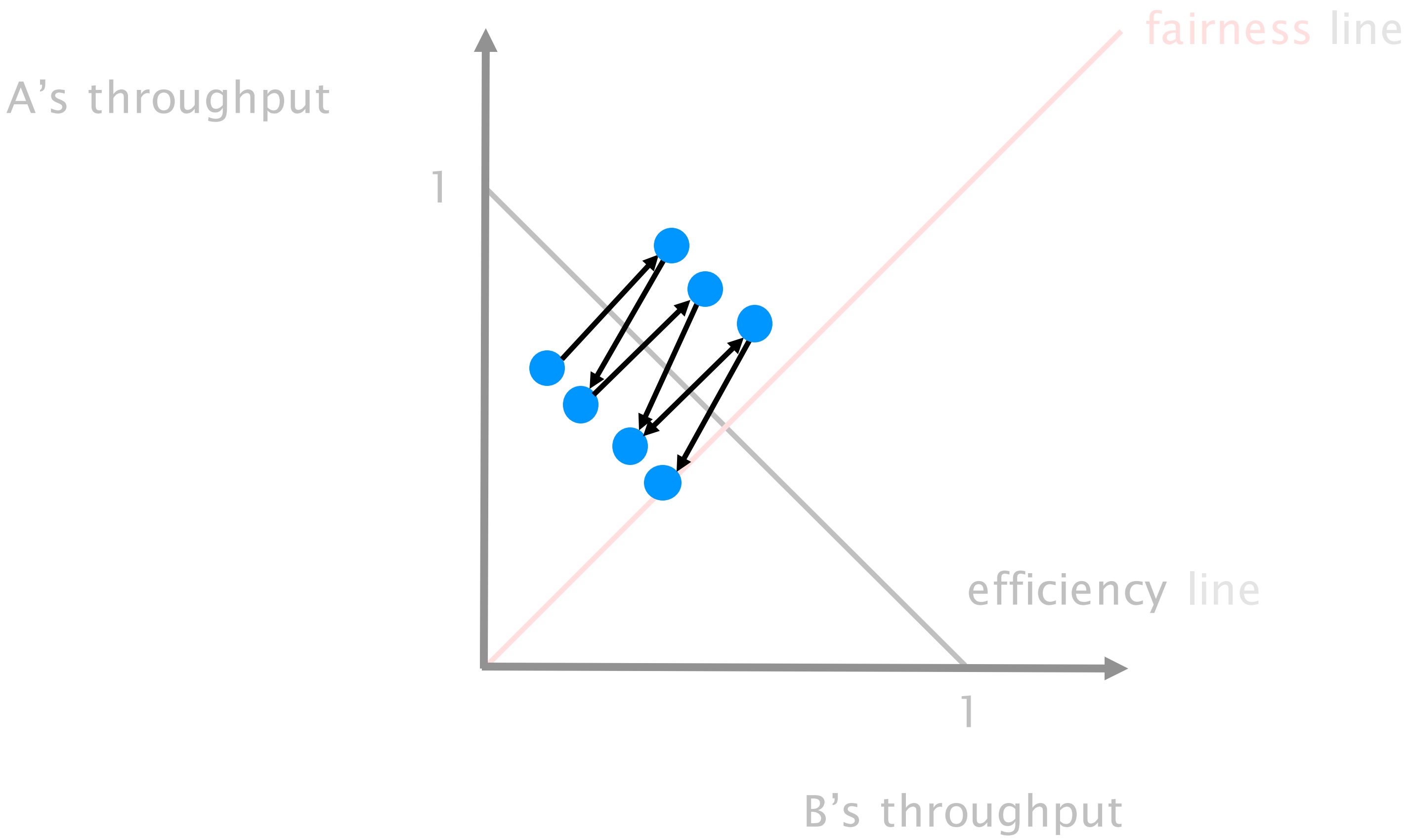
MIMD	aggressive	aggressive

AIMD converge to fairness and efficiency,
it then fluctuates around the optimum (in a stable way)



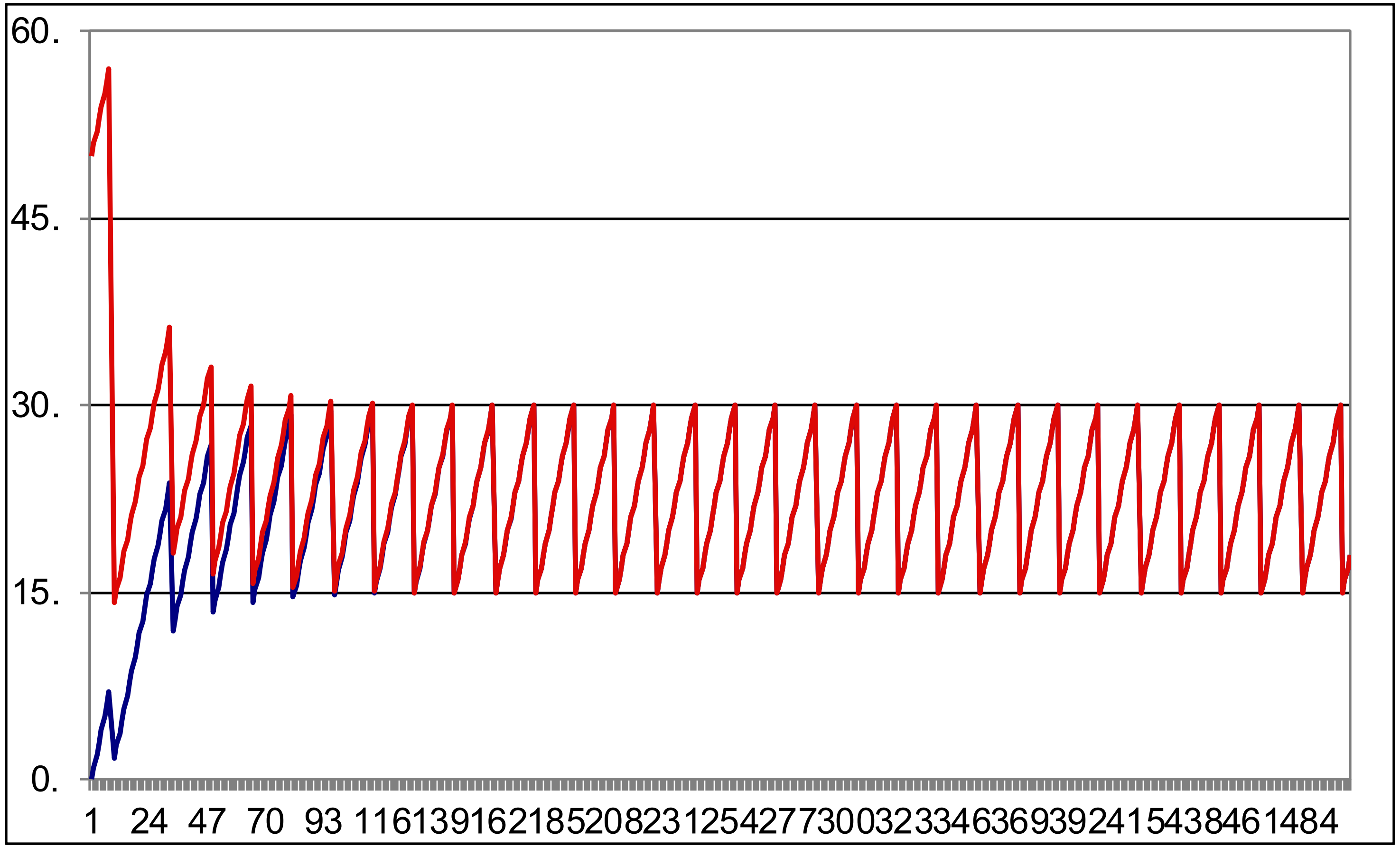
AIMD converge to fairness and efficiency, it then fluctuates around the optimum (in a stable way)

Intuition

During increase,
both flows gain bandwidth at the same rate

During decrease,
the faster flow releases more

AIMD converge to fairness and efficiency,
it then fluctuates around the optimum (in a stable way)



In practice,
TCP implements AIMD

	increase behavior	decrease behavior
AIAD	gentle	gentle
AIMD	gentle	aggressive
MIAD	aggressive	gentle
MIMD	aggressive	aggressive

In practice,
TCP implements AIMD

Implementation

After each ACK,

Increment cwnd by 1/cwnd

linear increase of max. 1 per RTT

Question

When does a sender leave slow-start and start AIMD?

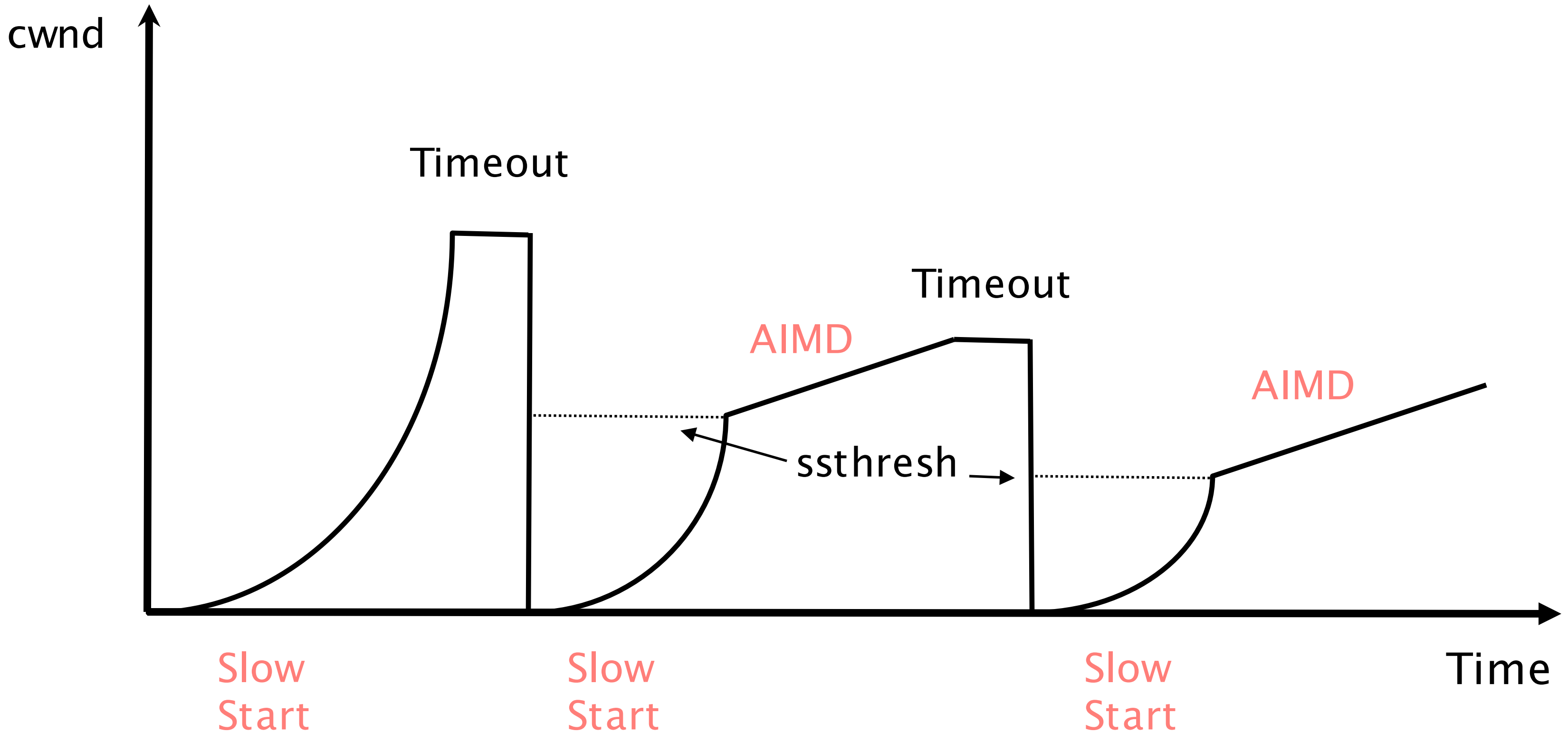
Introduce a slow start treshold, adapt it in function of congestion:

on timeout, sstresh = CWND/2

TCP congestion control in less than 10 lines of code

```
Initially:
    cwnd = 1
    ssthresh = infinite
New ACK received:
    if (cwnd < ssthresh):
        /* Slow Start*/
        cwnd = cwnd + 1
    else:
        /* Congestion Avoidance
    */
        cwnd = cwnd + 1/cwnd
Timeout:
    /* Multiplicative decrease */
    ssthresh = cwnd/2
    cwnd = 1
```

The congestion window of a TCP session typically undergoes multiple cycles of slow-start/AIMD



Going back all the way back to 0 upon timeout completely destroys throughput

solution

Avoid timeout expiration…

which are usually >500ms

Detecting losses can be done using ACKs or timeouts, the two signal differ in their degree of severity

duplicated ACKs	mild congestion signal packets are still making it
timeout	severe congestion signal multiple consequent losses

TCP automatically resends a segment after receiving 3 duplicates ACKs for it

this is known as a "fast retransmit"

After a fast retransmit, TCP switches back to AIMD,

without going all way the back to 0

this is known as “fast recovery”

TCP congestion control (almost complete)

Initially:
```
    cwnd = 1
    ssthresh = infinite
```
New ACK received:
```
    if (cwnd < ssthresh):
        /* Slow Start*/
        cwnd = cwnd + 1
    else:
        /* Congestion Avoidance */
        cwnd = cwnd + 1/cwnd
    dup_ack = 0
```
Timeout:
```
    /* Multiplicative decrease */
    ssthresh = cwnd/2
    cwnd = 1
```
Duplicate ACKs received:
```
    dup_ack ++;
    if (dup_ack >= 3):
        /* Fast Recovery */
        ssthresh = cwnd/2
        cwnd = ssthresh
```

Initially:

```
cwnd = 1
ssthresh = infinite
```

New ACK received:

```
if (cwnd < ssthresh):
    /* Slow Start*/
    cwnd = cwnd + 1
else:
    /* Congestion Avoidance */
    cwnd = cwnd + 1/cwnd
dup_ack = 0
```

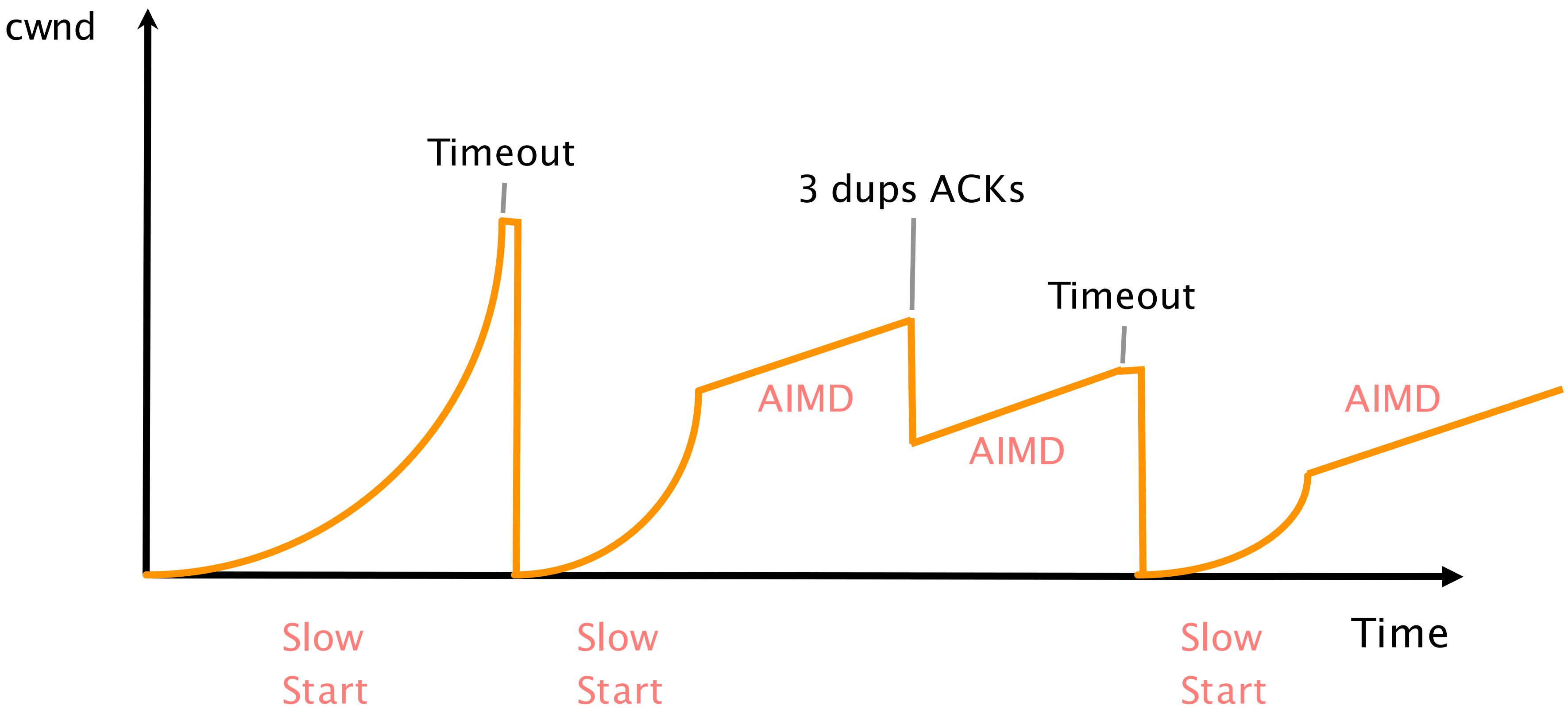
Timeout:

```
/* Multiplicative decrease */
ssthresh = cwnd/2
cwnd = 1
```

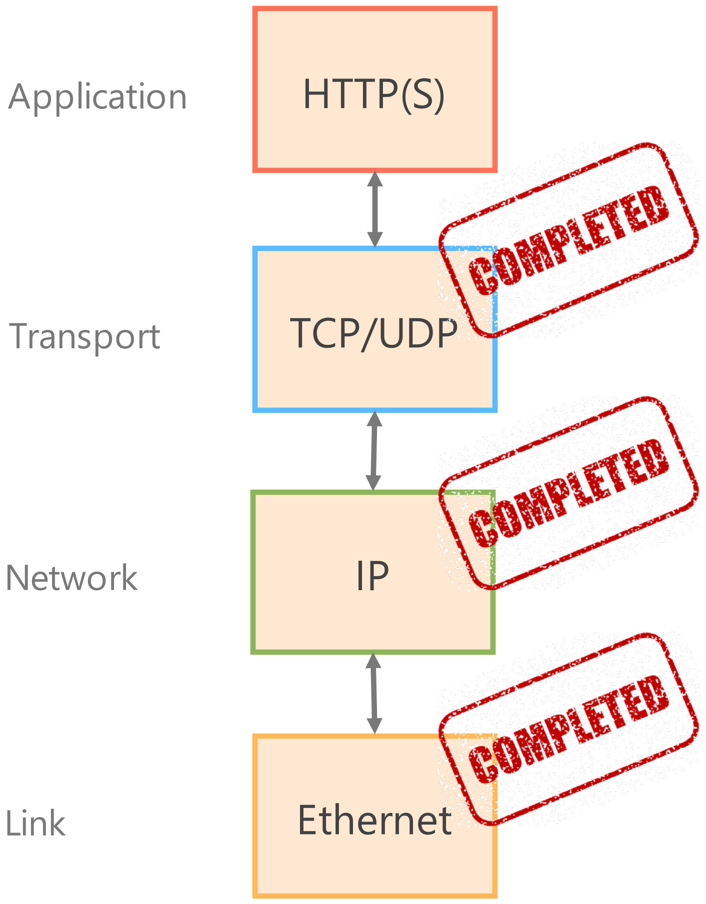
Duplicate ACKs received:

```
dup_ack ++;
if (dup_ack >= 3):
    /* Fast Recovery */
    ssthresh = cwnd/2
    cwnd = ssthresh
```

Congestion control makes TCP throughput look like a “sawtooth”



We now have completed the transport layer (!)



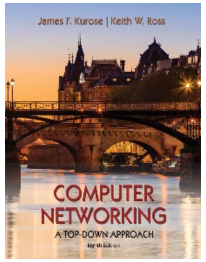
Reliable (or unreliable) transport

Best-effort global packet delivery

Best-effort local packet delivery

Reading: Book Kurose & Ross

Class textbook:
Computer Networking: A Top-Down Approach (8th ed.)
J.F. Kurose, K.W. Ross
Pearson, 2020
http://gaia.cs.umass.edu/kurose_ross



- Week 10
  - 3.5 (Connection-Oriented Transport: TCP)
  - 3.6 (Principles of Congestion Control) and 3.7 (TCP Congestion Control)

Check Your Knowledge

PROBLEM SOLVING HOME

TRY A RANDOM PROBLEM

INTERACTIVE END-OF-CHAPTER EXERCISES

Supplement to Computer Networking: A Top Down Approach 8th Edition

"Tell me and I forget. Show me and I remember. Involve me and I understand." Chinese proverb

CHAPTER 3: TRANSPORT LAYER

- Internet checksum (similar to Chapter 3, P3 and P4)
- Reliable data transfer: rdt22
- TCP sequence and ACK numbers, with segment loss (similar to Chapter 3, P27)
- TCP RTT and timeout (similar to Chapter 3, P31)
- TCP congestion window evolution (similar to Chapter 3, P40)
- TCP retransmissions (reliable data transmission with ACK loss)
- UDP Mux and Demux
- TCP Mux and Demux

http://gaia.cs.umass.edu/kurose_ross/interactive/

Final Exam

- 02.02.2022 – 10:15h
- Online exam
- Duration **120 min**
- Open book
- Individual problem solving
- No (electronic) communication allowed
- Random spot checks