# Communication Networks

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## Communication Networks

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Material inspired from Scott Shenker & Jennifer Rexford

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Last week on

Communication Networks

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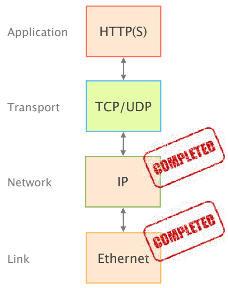
We started to look at the transport layer

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## What Problems Should Be Solved Here?

Data delivering, to the *correct* application

- IP just points towards next protocol
- *Transport needs to demultiplex incoming data (ports)*

Files or bytestreams abstractions for the applications

- Network deals with packets
- *Transport layer needs to translate between them*

Reliable transfer (if needed)

Not overloading the receiver

Not overloading the network

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## What Is Needed to Address These?

*Demultiplexing*: identifier for application process

- Going from host-to-host (IP) to process-to-process

*Translating between bytestreams and packets:*

- Do segmentation and reassembly

*Reliability*: ACKs and all that stuff

*Corruption*: Checksum

*Not overloading receiver*: "Flow Control"

- Limit data in receiver's buffer

*Not overloading network*: "Congestion Control"

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## Sockets

A socket is a software abstraction by which an application process exchanges network messages with the (transport layer in the) operating system

- socketID = socket(…, socket.TYPE)
- socketID.sendto(message, …)
- socketID.recvfrom(…)

Two important types of sockets

- UDP socket: TYPE is SOCK_DGRAM
- TCP socket: TYPE is SOCK_STREAM

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## 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")

| | |
|---|---|
| SRC port | DST port |
| checksum | length |
| DATA | |

## Transmission Control Protocol (TCP)

Reliable, in-order delivery
- Ensures byte stream (eventually) arrives intact
  - In the presence of corruption and loss

Connection oriented
- Explicit set-up and tear-down of TCP session

Full duplex stream-of-bytes service
- Sends and receives a stream of bytes, not messages

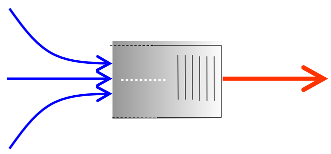
Flow control
- Ensures that sender doesn't overwhelm receiver

Congestion control
- Dynamic adaptation to network path's capacity

This week on
Communication Networks

TCP Congestion Control


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

Congestion is harmful

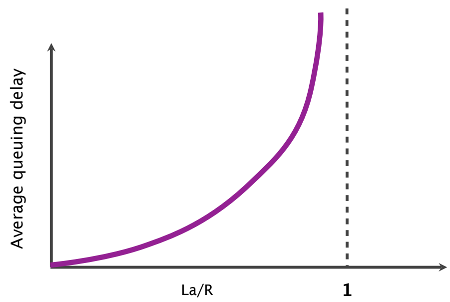
| | | |
|---|---|---|
| average packet arrival rate | *a* | [packet/sec] |
| transmission rate of outgoing link | *R* | [bit/sec] |
| fixed packets length | *L* | [bit |
| average bits arrival rate | *La* | [bit/sec] |
| traffic intensity | *La/R* | |

When the traffic intensity is >1, 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

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



## 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

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The Internet almost died of congestion in 1986

throughput collapsed from 32 Kbps to… 40 bps

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| | |
|---|---|
| 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 |

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## 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

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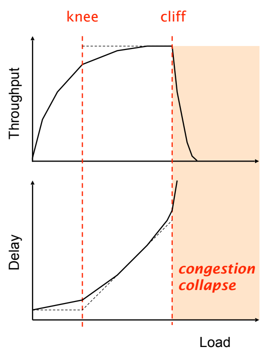
Knee  point after which

| | | |
|---|---|---|
| throughput | increases | slowly |
| delay | increases | quickly |

Cliff  point after which

| | | |
|---|---|---|
| throughput | decreases | quickly |
| delay | tends to | infinity |

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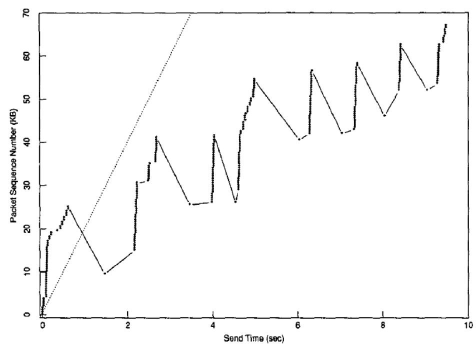


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Van Jacobson saved us with Congestion Control

his solution went right into BSD

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## 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<br>**solved using a receiving window** |
| Congestion control | prevents a set of senders from overloading the network<br>**solved using a "congestion" window** |

## The sender adapts its sending rate based on these two windows

| | |
|---|---|
| Receiving Window<br>RWND | How many bytes can be sent without overflowing the receiver buffer?<br>based on the receiver input |
| Congestion Window<br>CWND | How many bytes can be sent without overflowing the routers?<br>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<br>delay often varies considerably |
| Approach #3 | Measure packet loss<br>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<br>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<br>packets are still making it |
| timeout | severe congestion signal<br>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

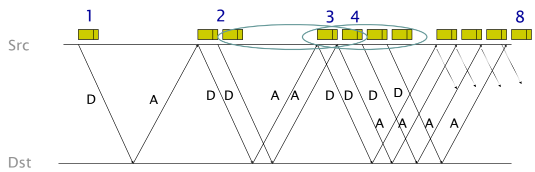
| | | |
|---|---|---|
| #1 | bandwidth estimation | How to adjust the bandwidth of a single flow to the bottleneck bandwidth? could be 1 Mbps or 1 Gbps… |
| #2 | bandwidth adaptation | How to adjust the bandwidth of a single flow to variation of the bottleneck bandwidth? |
| #3 | fairness | How to share 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

- Multiplicative Increase or Decrease
  cwnd = a * cwnd
- Additive Increase of Decrease
  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 |
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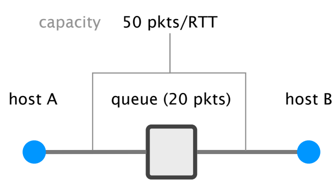
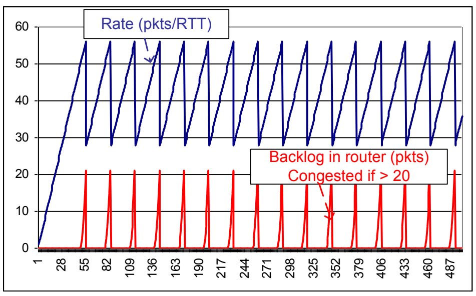
| | | |
|---|---|---|
| #3 | fairness | How to share bandwidth “fairly” among flows, without overloading the network |

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

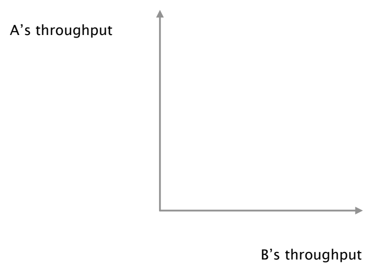
Consider first a single flow between A and B and AIMD



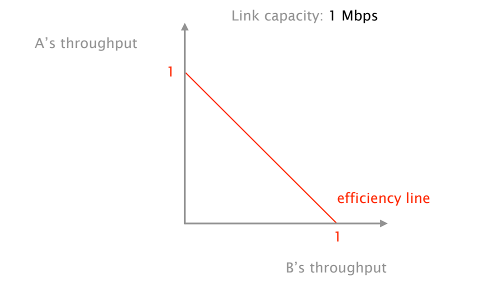

| | |
|---|---|
| without congestion | CWND increases by one packet every ACK |
| upon congestion | CWND decreases by a factor 2 |



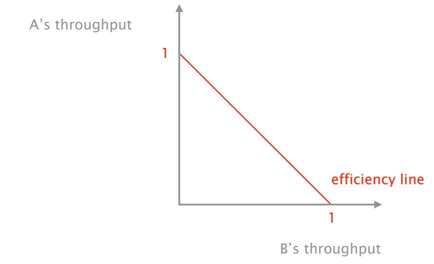

We can analyze the system behavior using a system trajectory plot

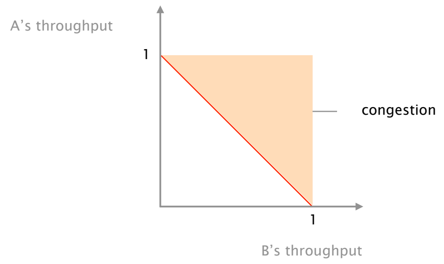


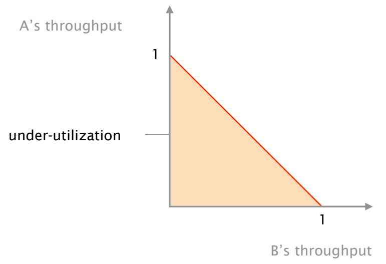
The system is efficient if the capacity is fully used, defining an efficiency line where $a + b = 1$



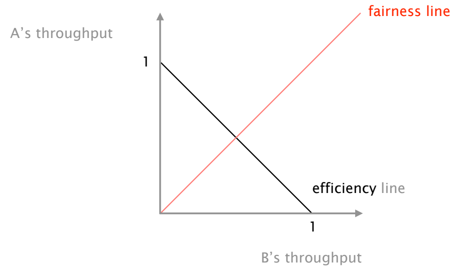

The goal of congestion control is to bring the system as close as possible to this line, and stay there

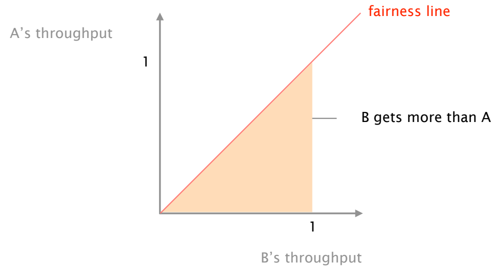


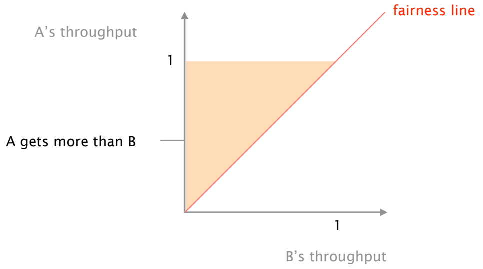



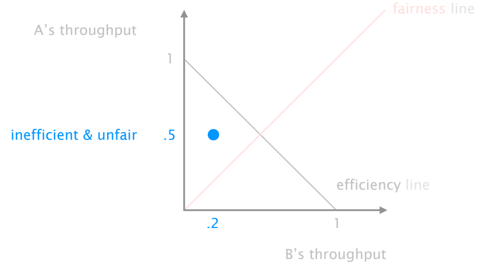



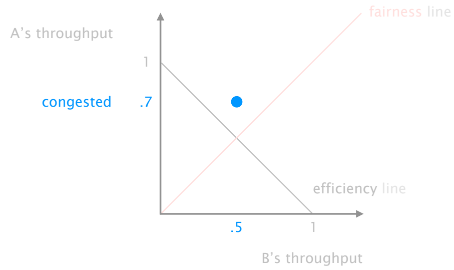

The system is fair whenever A and B have equal throughput, defining a fairness line where $a = b$

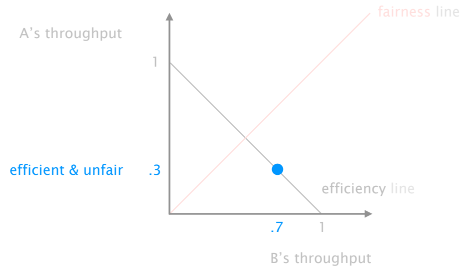


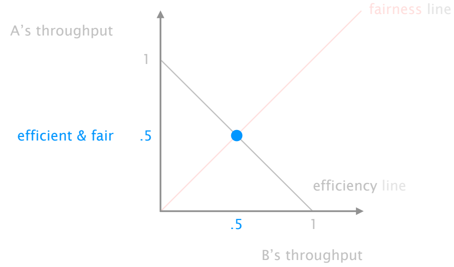






| | increase behavior | decrease behavior |
|---|---|---|
| AIAD | gentle | gentle |
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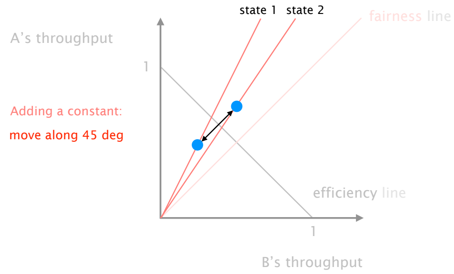
AIAD does not converge to fairness, nor efficiency:
the system fluctuates between two fairness states

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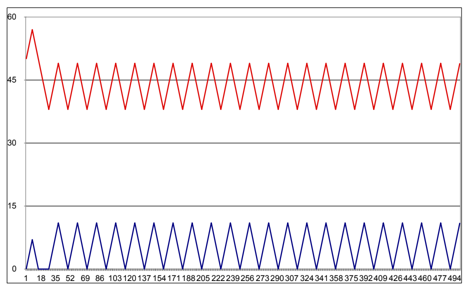


| | increase behavior | decrease behavior |
|---|---|---|
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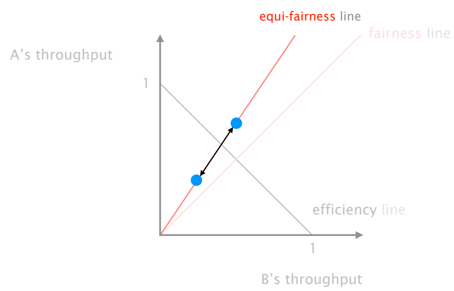
MIMD does not converge to fairness, nor efficiency:
the system fluctuates along a equi-fairness line



| | increase behavior | decrease behavior |
|---|---|---|
| AIAD | gentle | gentle |
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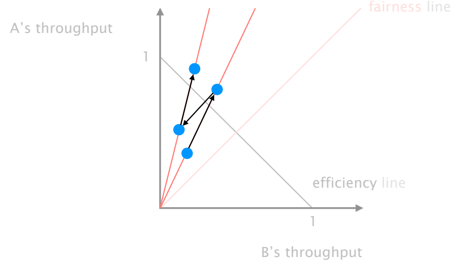
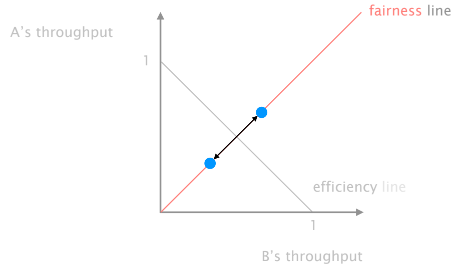
| MIAD | aggressive | gentle |
| MIMD | aggressive | aggressive |

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

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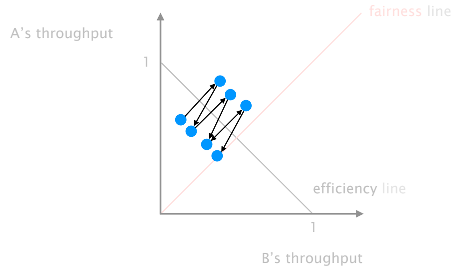
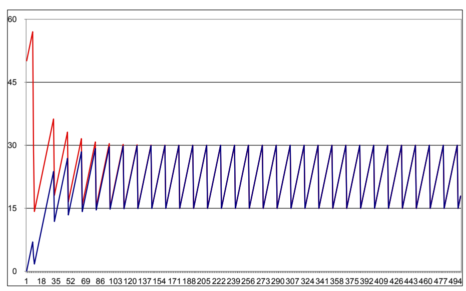
AIMD converge to fairness and efficiency,
it then fluctuates around the optimum (in a stable way)




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?

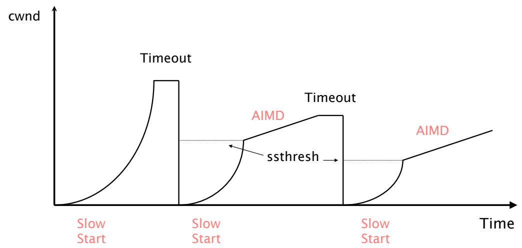
Introduce a slow start treshold, adapt it in function of congestion:

on timeout, sstresh = CNWD/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):
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Timeout:
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    ssthresh = cwnd/2
    cwnd = 1
```
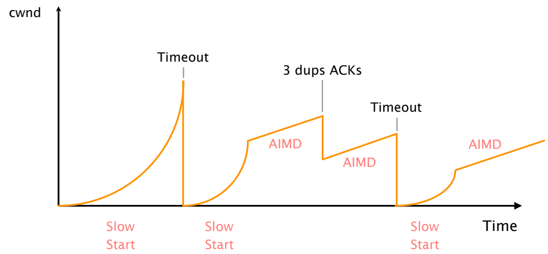

```
Duplicate ACKs received:
    dup_ack ++;
    if (dup_ack >= 3):
        /* Fast Recovery */
        ssthresh = cwnd/2
        cwnd = ssthresh
```

Congestion control makes TCP throughput look like a “sawtooth”

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