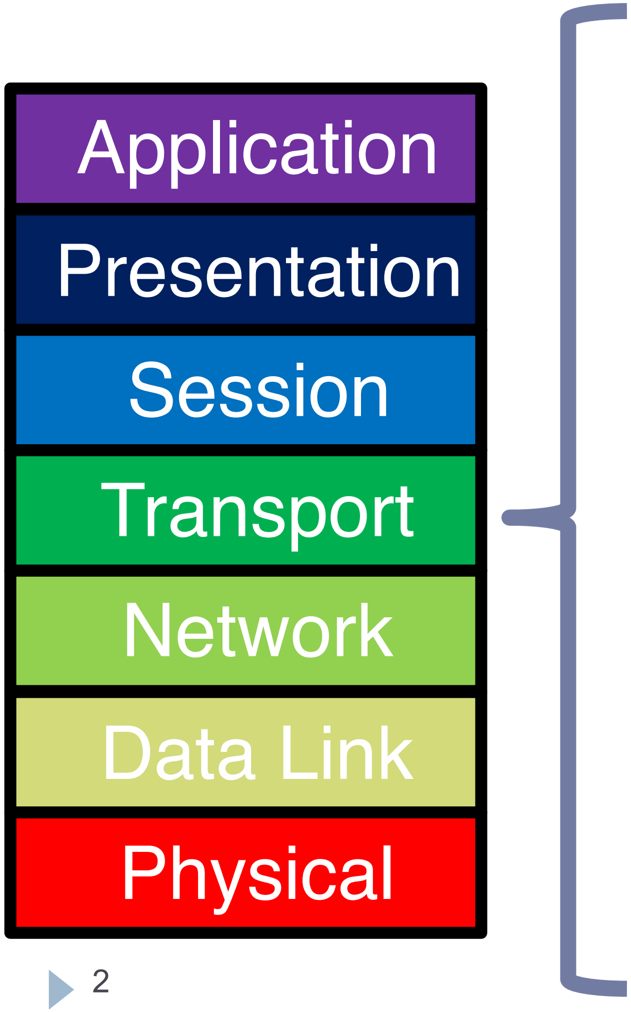
Cristina Nita-Rotaru

CS4700/5700: Network fundamentals

Transport.

Transport Layer



Application

Presentation

Session

Transport

Network

Data Link

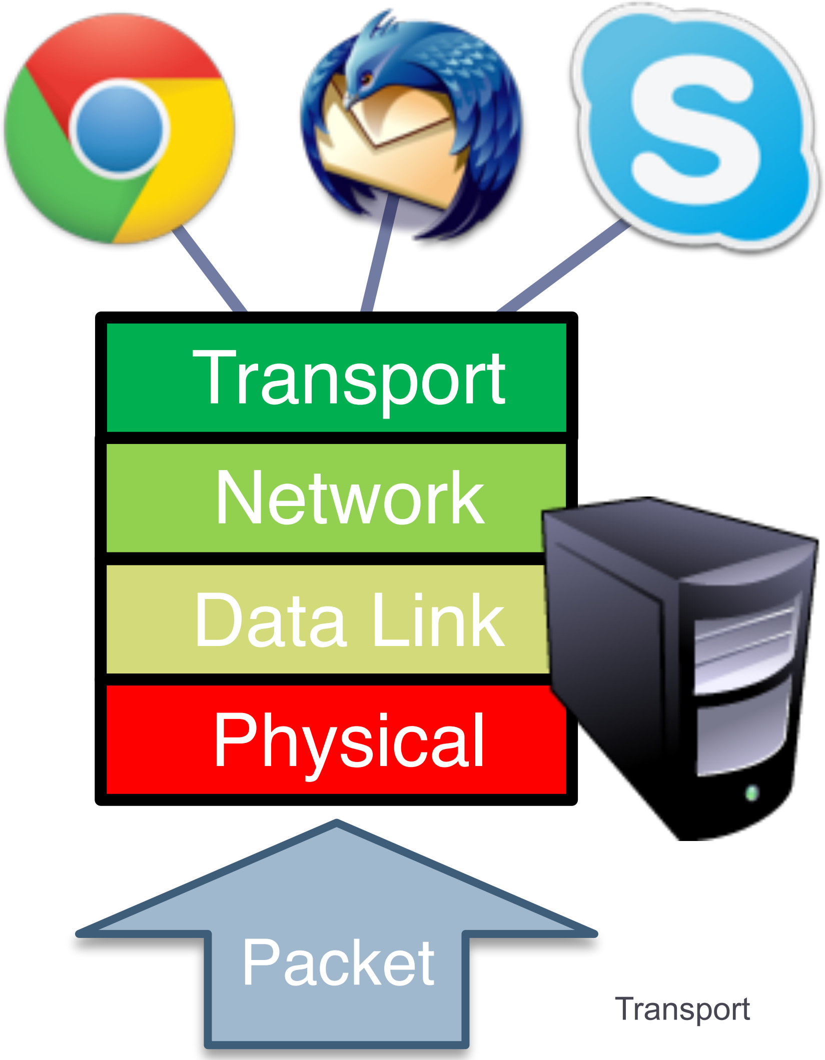
Physical

- Function:
  - Demultiplexing of data streams
- Optional functions:
  - Creating long lived connections
  - Reliable, in-order packet delivery
  - Error detection
  - Flow and congestion control
- Key challenges:
  - Detecting and responding to congestion
  - Balancing fairness against high utilization

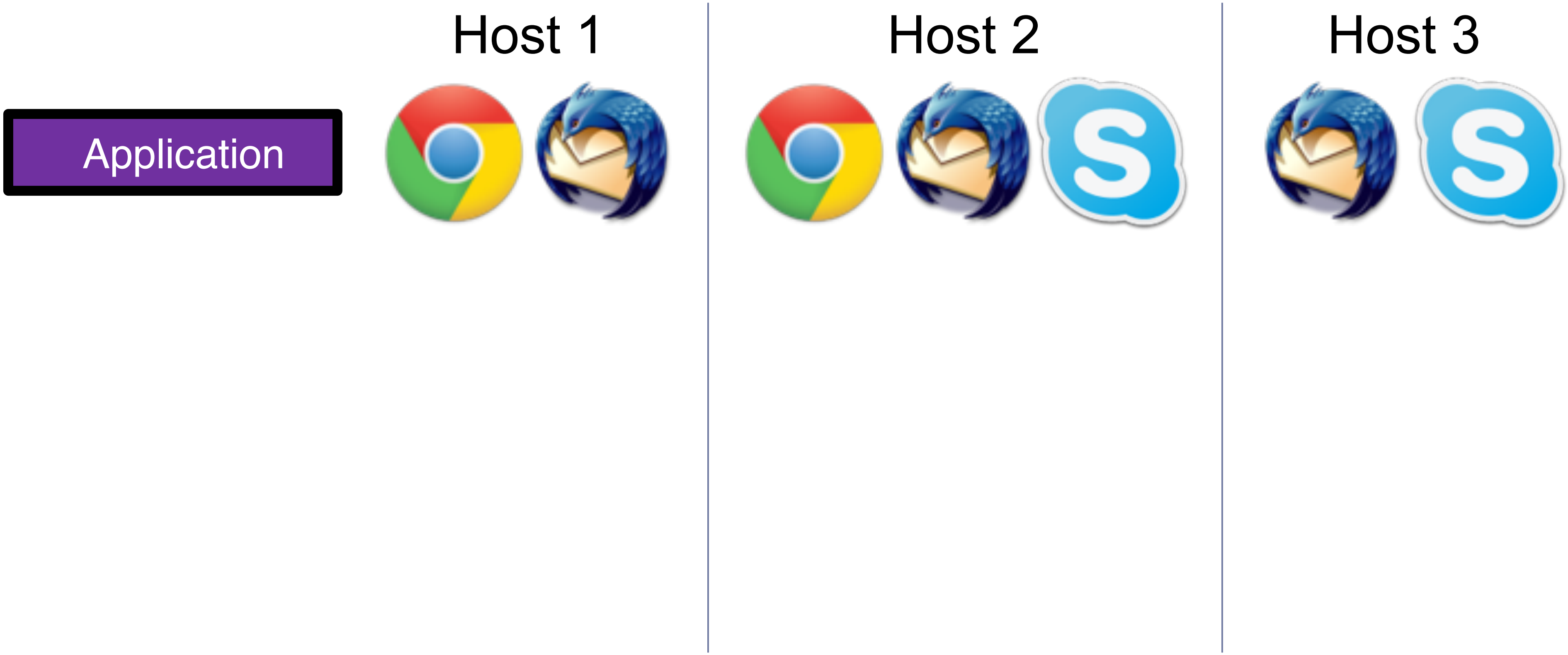
1: UDP

The Case for Multiplexing

- Datagram network
  - No circuits
  - No connections
- Clients run many applications at the same time
  - Who to deliver packets to?
- IP header “protocol” field
  - 8 bits = 256 concurrent streams
- Insert Transport Layer to handle demultiplexing



Demultiplexing Traffic



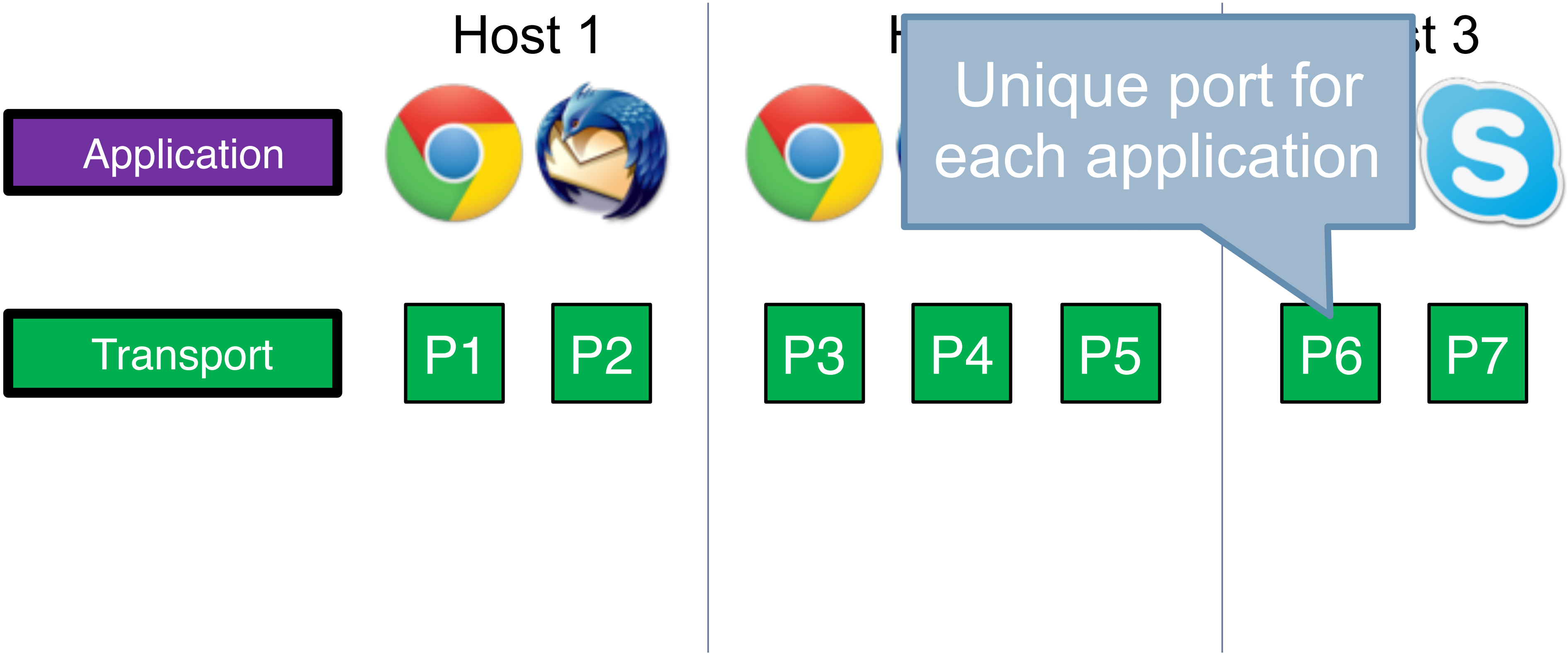
Application

Host 1

Host 2

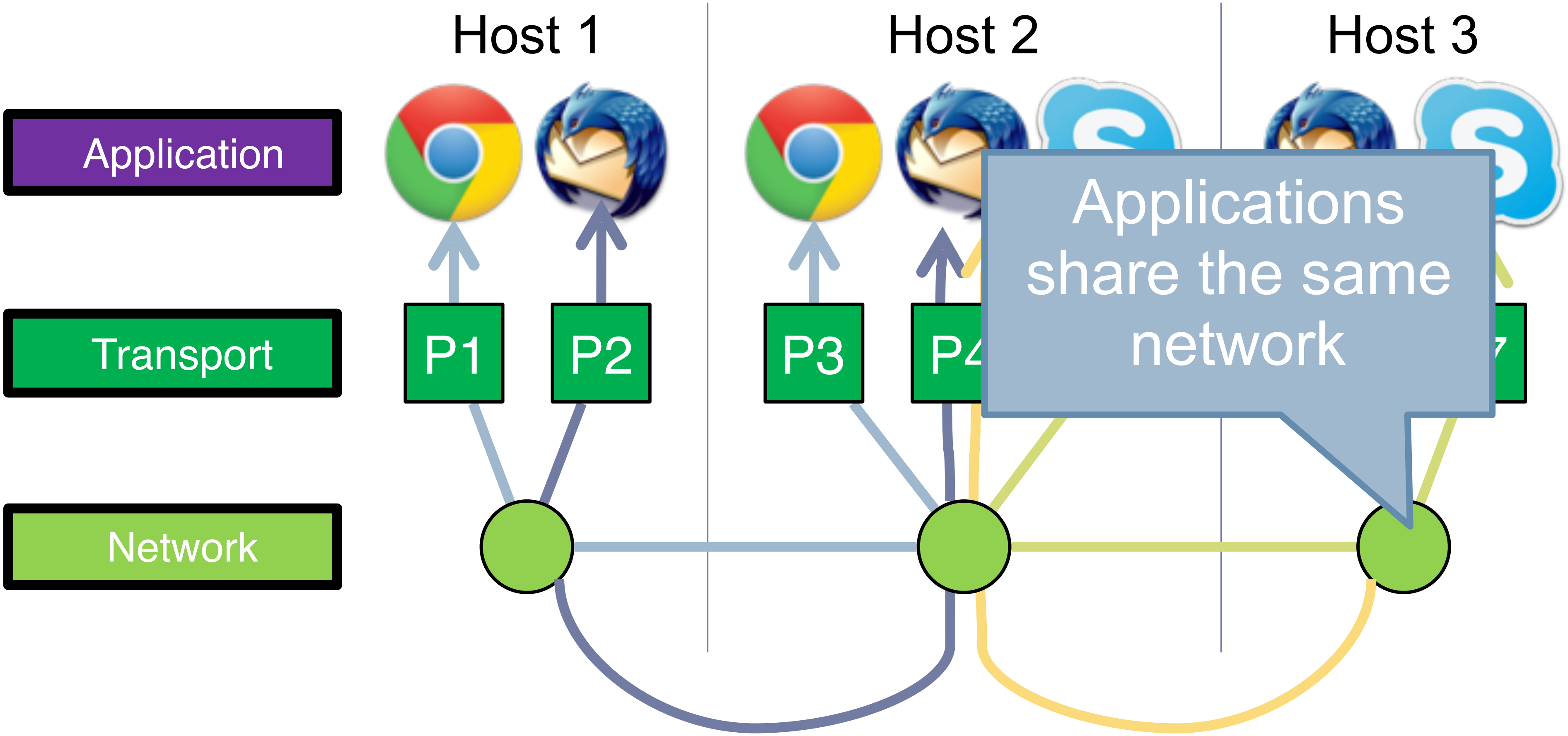
Host 3

Demultiplexing Traffic



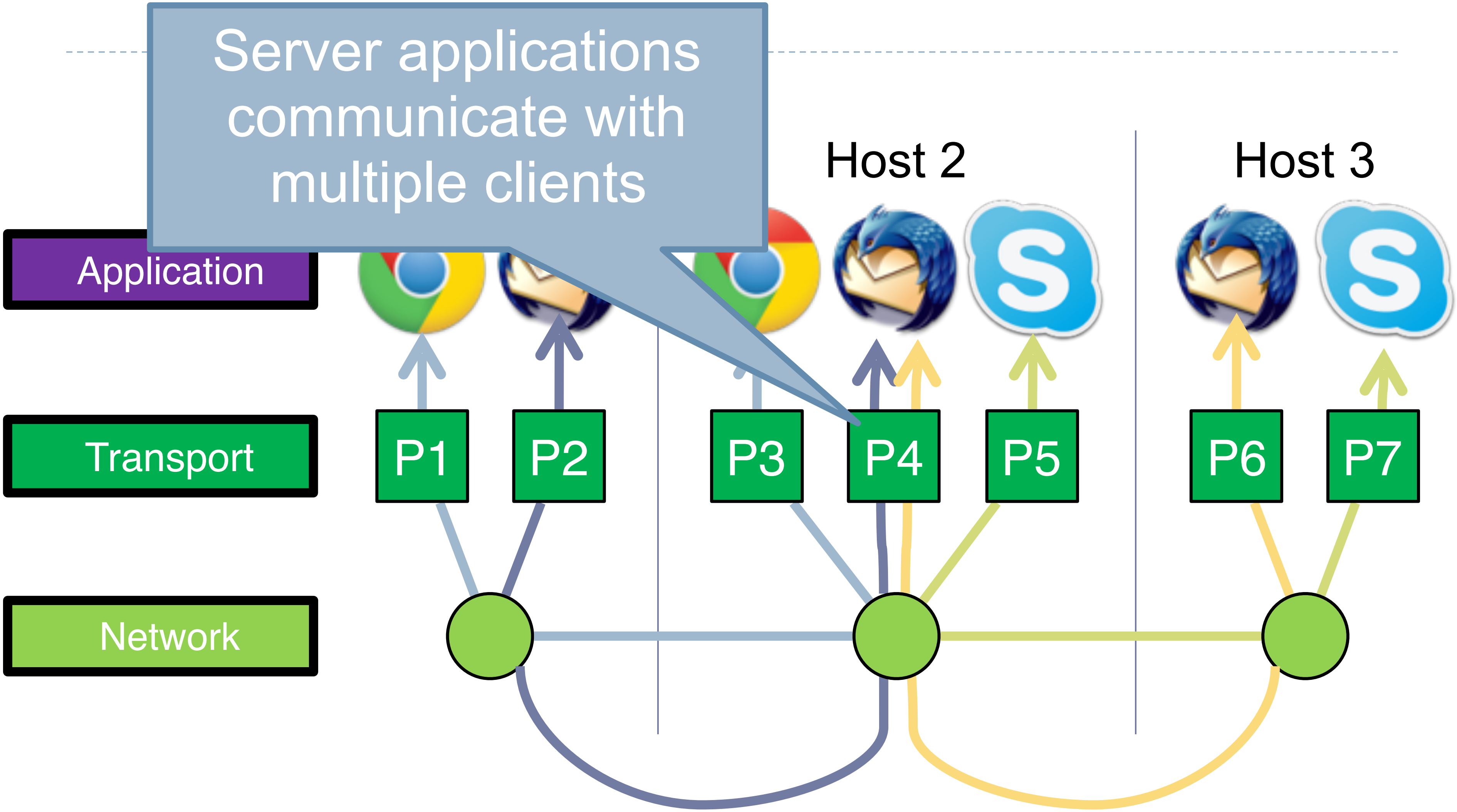
Unique port for each application

Demultiplexing Traffic



Applications share the same network

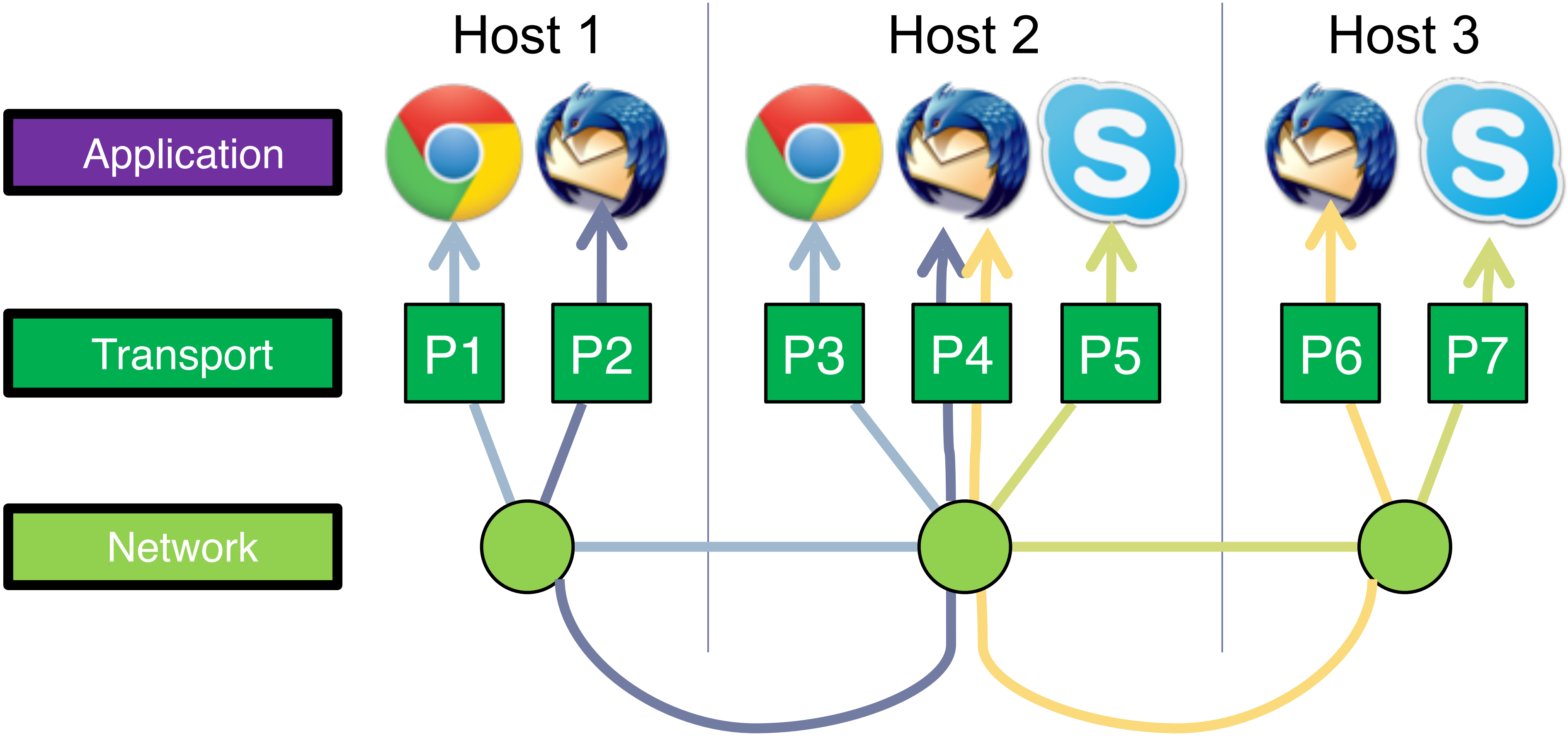
Demultiplexing Traffic



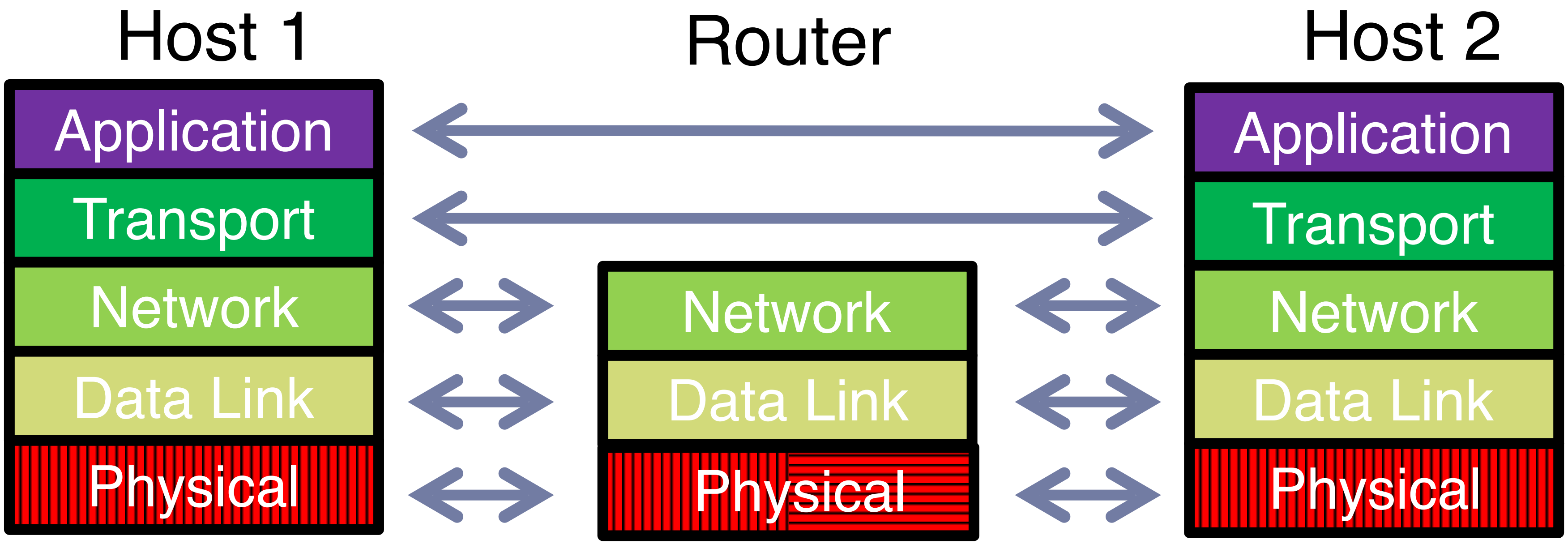
Server applications communicate with multiple clients

Demultiplexing Traffic

Endpoints identified by *<src_ip, src_port, dest_ip, dest_port>*



Layering, Revisited



- Lowest level end-to-end protocol (in theory)
  - Transport header only read by source and destination
  - Routers view transport header as payload

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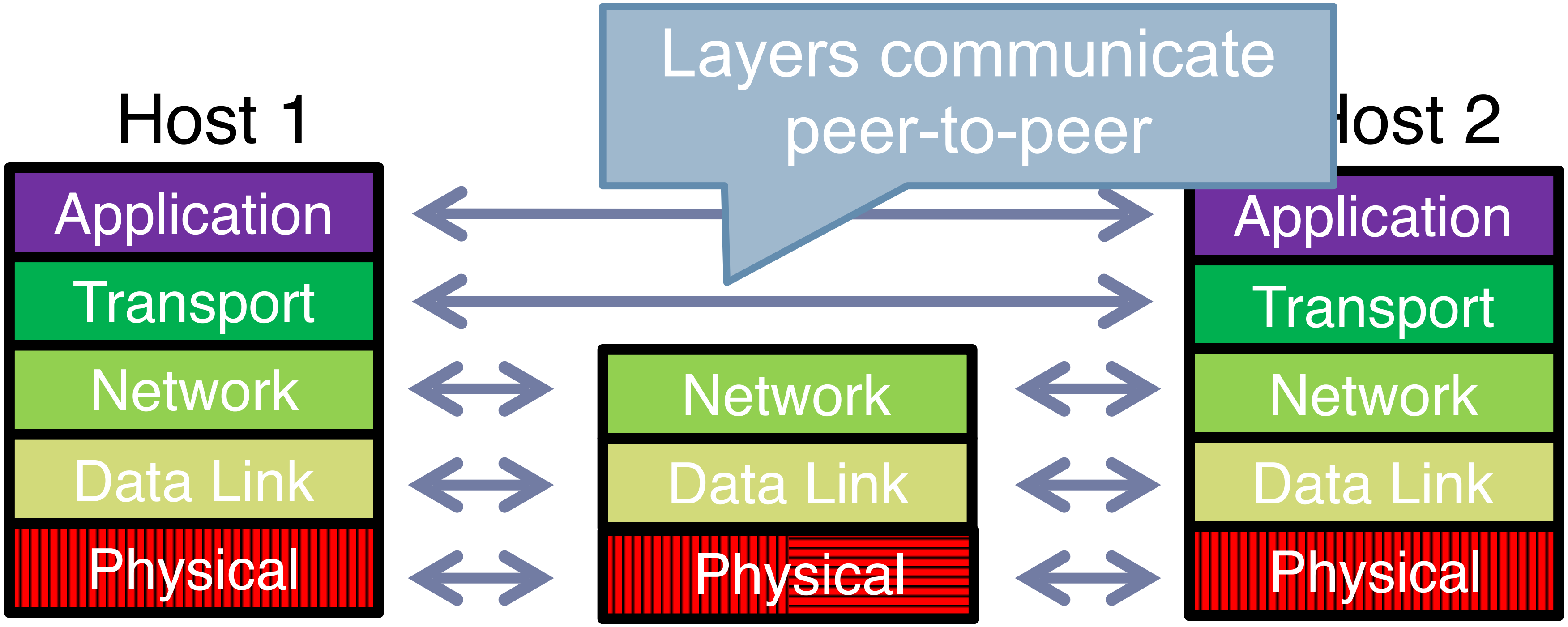

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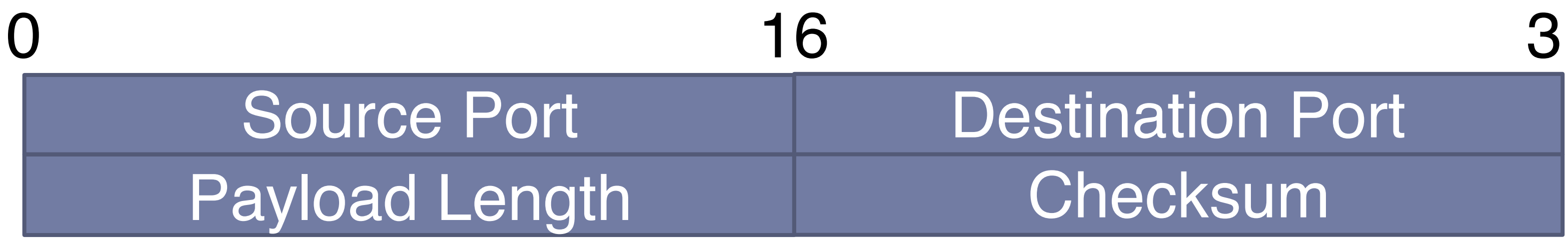
User Datagram Protocol (UDP)



0 16 31

Source Port | Destination Port

Payload Length | Checksum

- Simple, connectionless datagram
  - C sockets: SOCK_DGRAM
- Port numbers enable demultiplexing
  - 16 bits = 65535 possible ports
  - Port 0 is invalid
- Checksum for error detection
  - Detects (some) corrupt packets
  - Does not detect dropped, duplicated, or reordered packets

Uses for UDP

- Invented after TCP
  - Why?
- Not all applications can tolerate TCP
- Custom protocols can be built on top of UDP
  - Reliability? Strict ordering?
  - Flow control? Congestion control?
- Examples
  - RTMP, real-time media streaming (e.g. voice, video)
  - Facebook datacenter protocol

2: TCP

Transmission Control Protocol

- Reliable, in-order, bi-directional byte streams
  - Port numbers for demultiplexing
  - Virtual circuits (connections)
  - Flow control
  - Congestion control, approximate fairness

<table>
<tr><th>0</th><th>4</th><th>16</th><th>31</th></tr>
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Transmission Control Protocol

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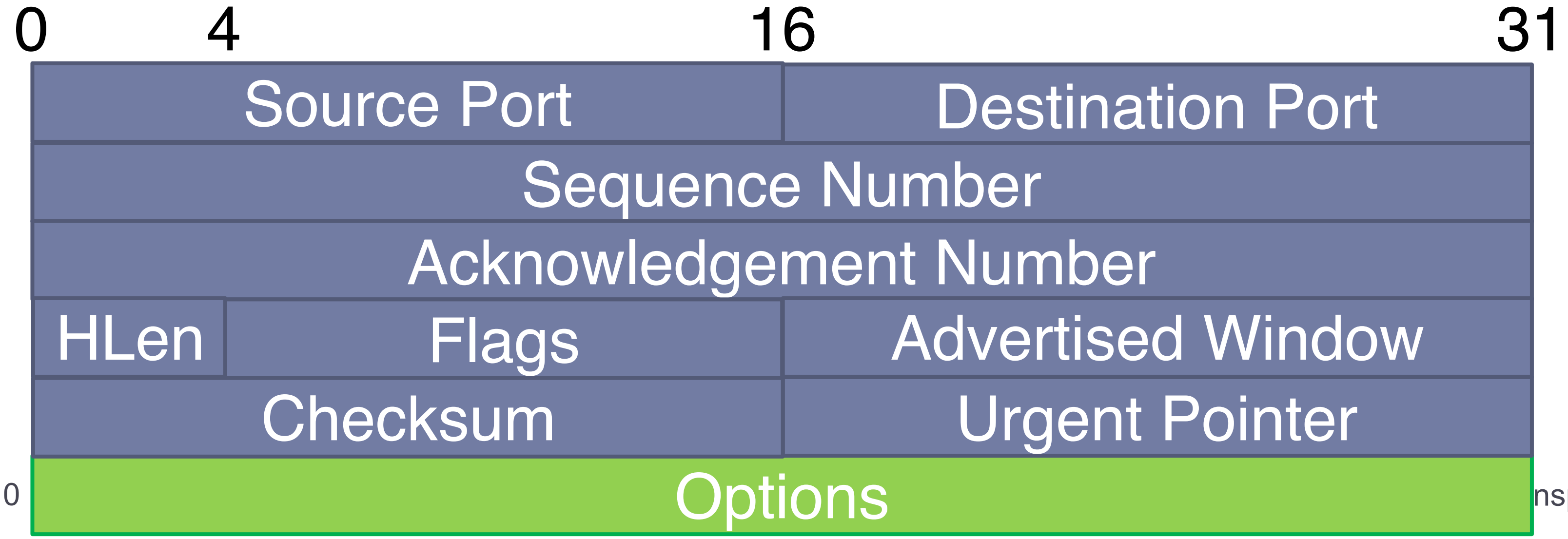
Why these features?



Connection Setup

- Why do we need connection setup?
  - To establish state on both hosts
  - Most important state: sequence numbers
    - Count the number of bytes that have been sent
    - Initial value chosen at random
    - Why?
- Important TCP flags (1 bit each)
  - SYN – synchronization, used for connection setup
  - ACK – acknowledge received data
  - FIN – finish, used to tear down connection

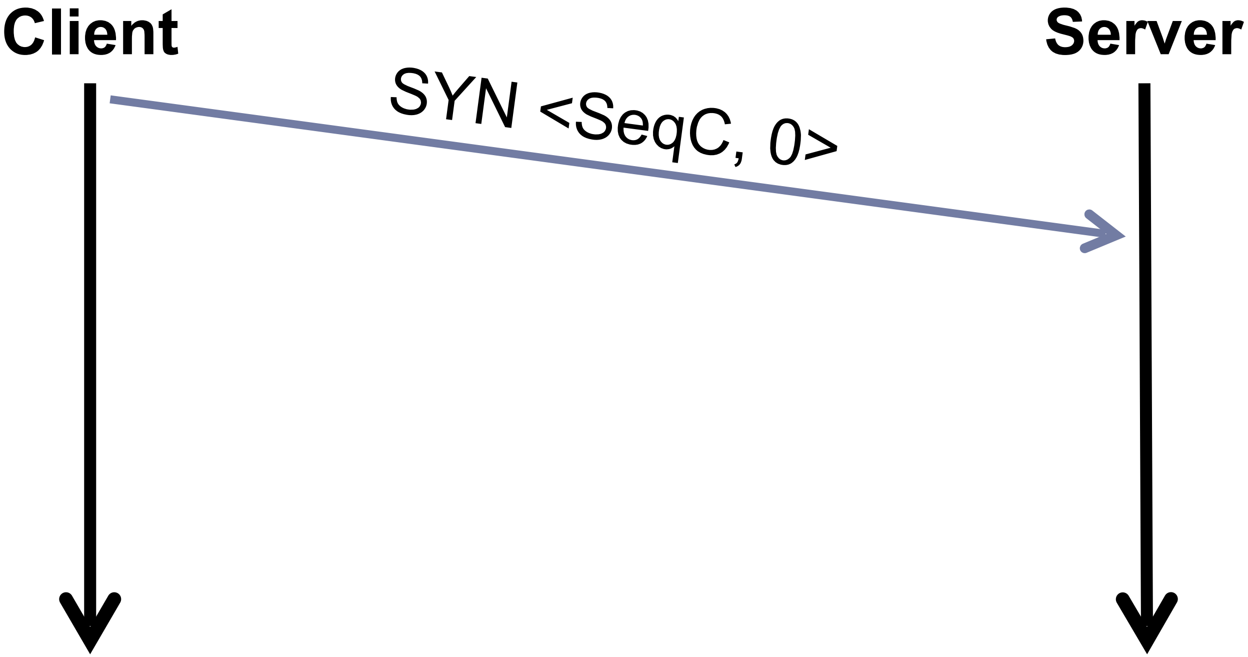
Three Way Handshake

Client

Server

- Each side:
  - Notifies the other of starting sequence number
  - ACKs the other side’s starting sequence number

Three Way Handshake



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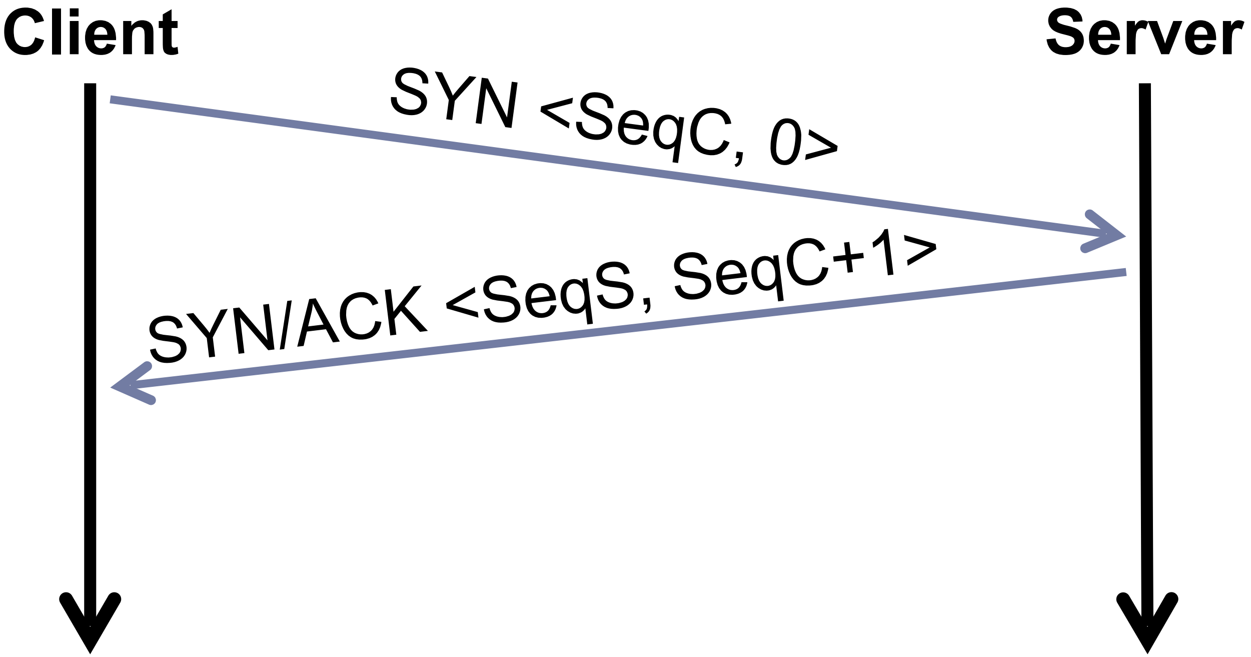

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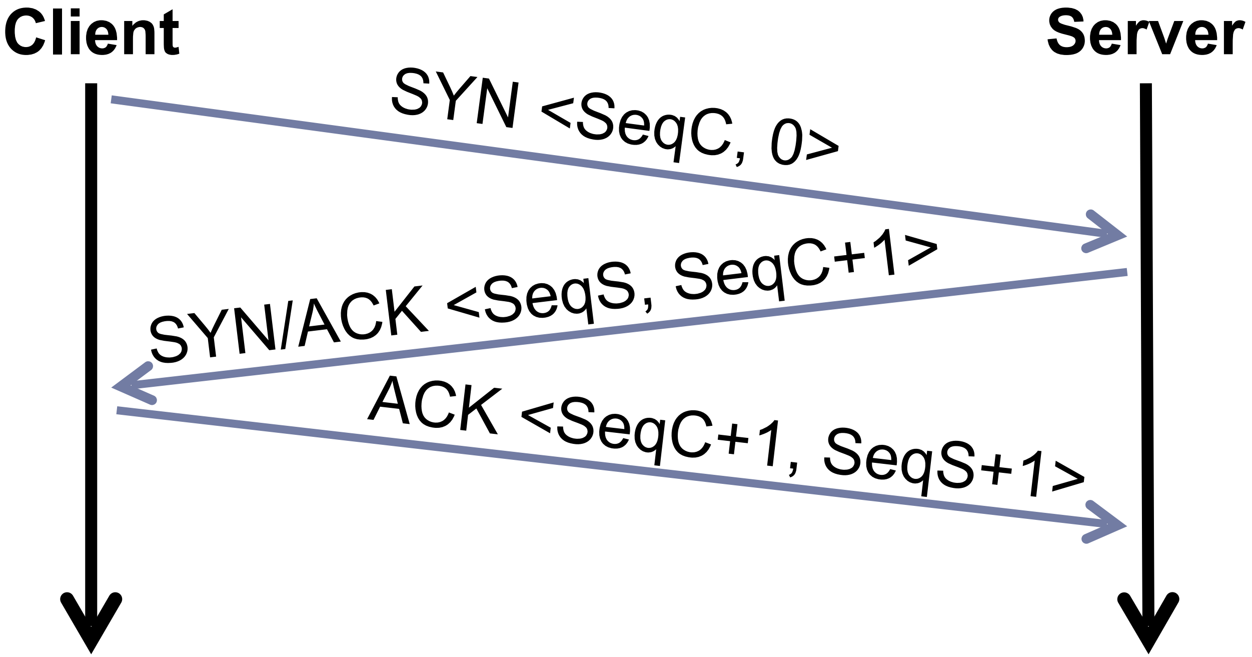

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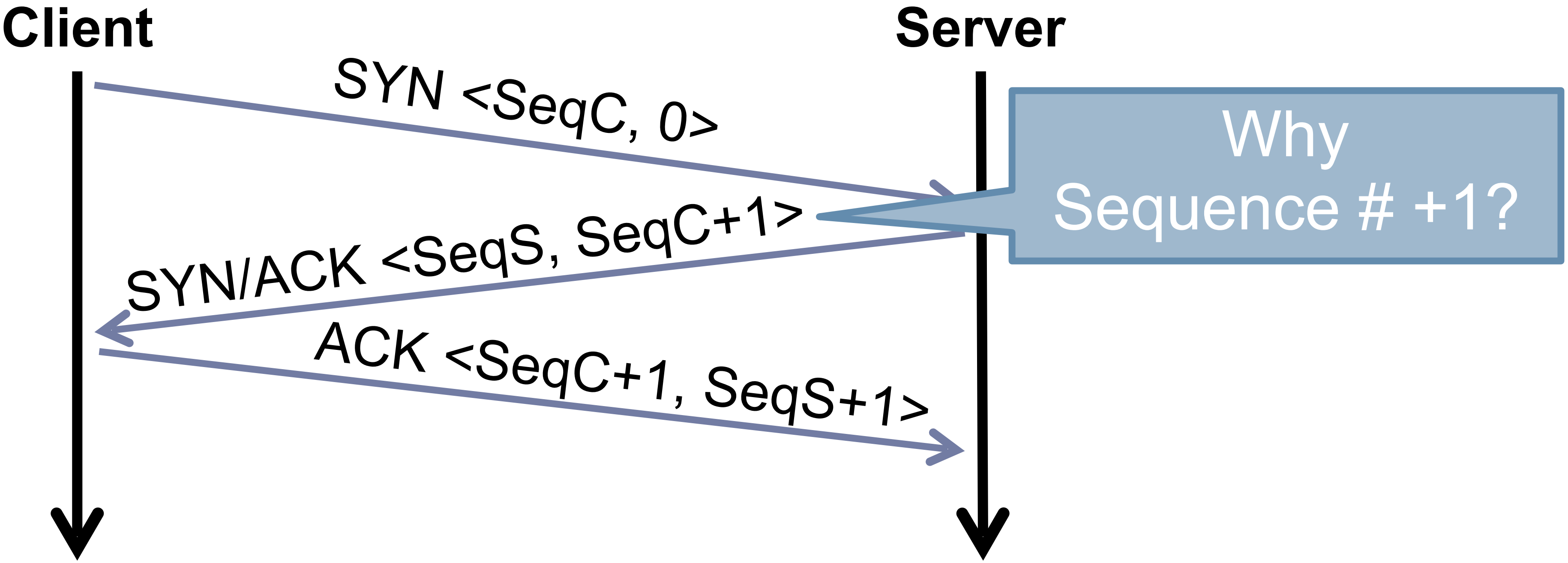

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- Source spoofing
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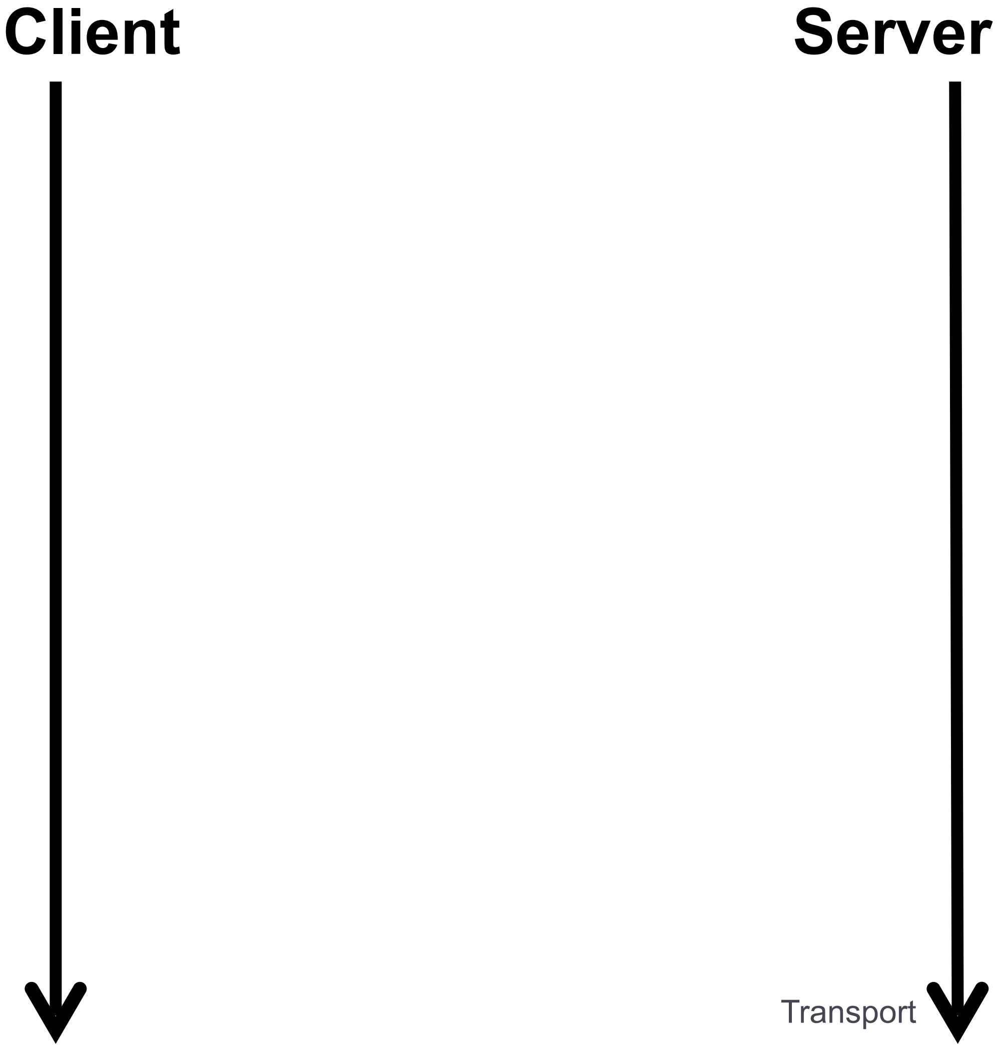
- Connection state management
  - Each SYN allocates state on the server
  - SYN flood = denial of service attack
  - Solution: SYN cookies

Connection Tear Down

- Either side can initiate tear down



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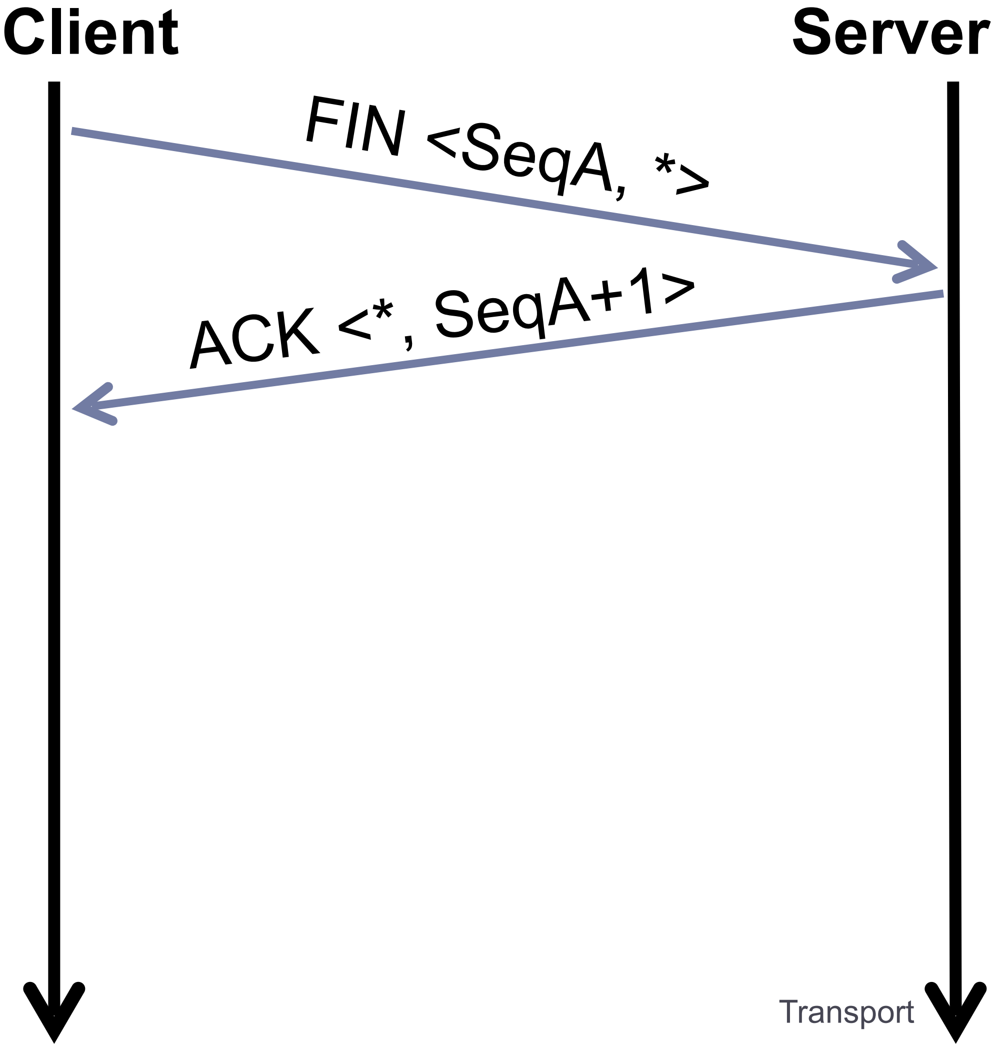

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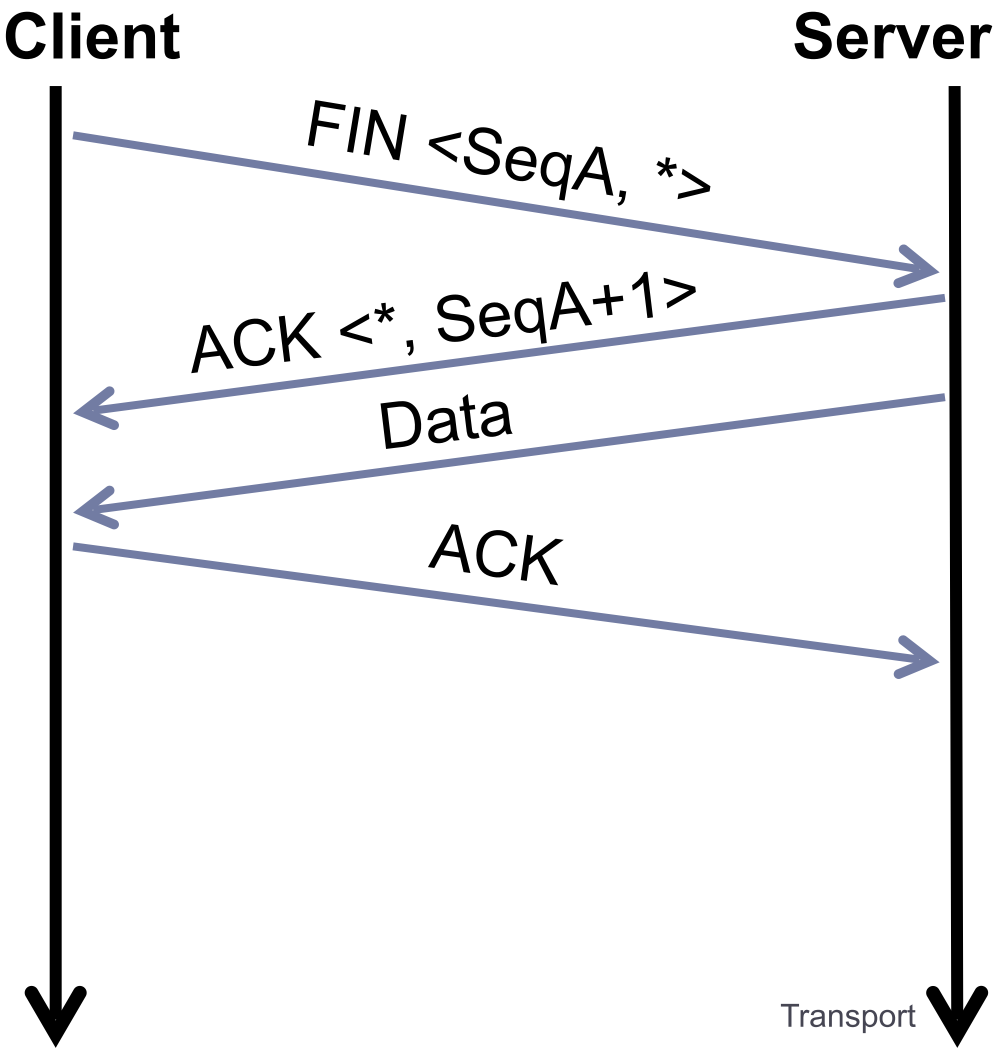
  - Half open connection
  - *shutdown()*



Connection Tear Down

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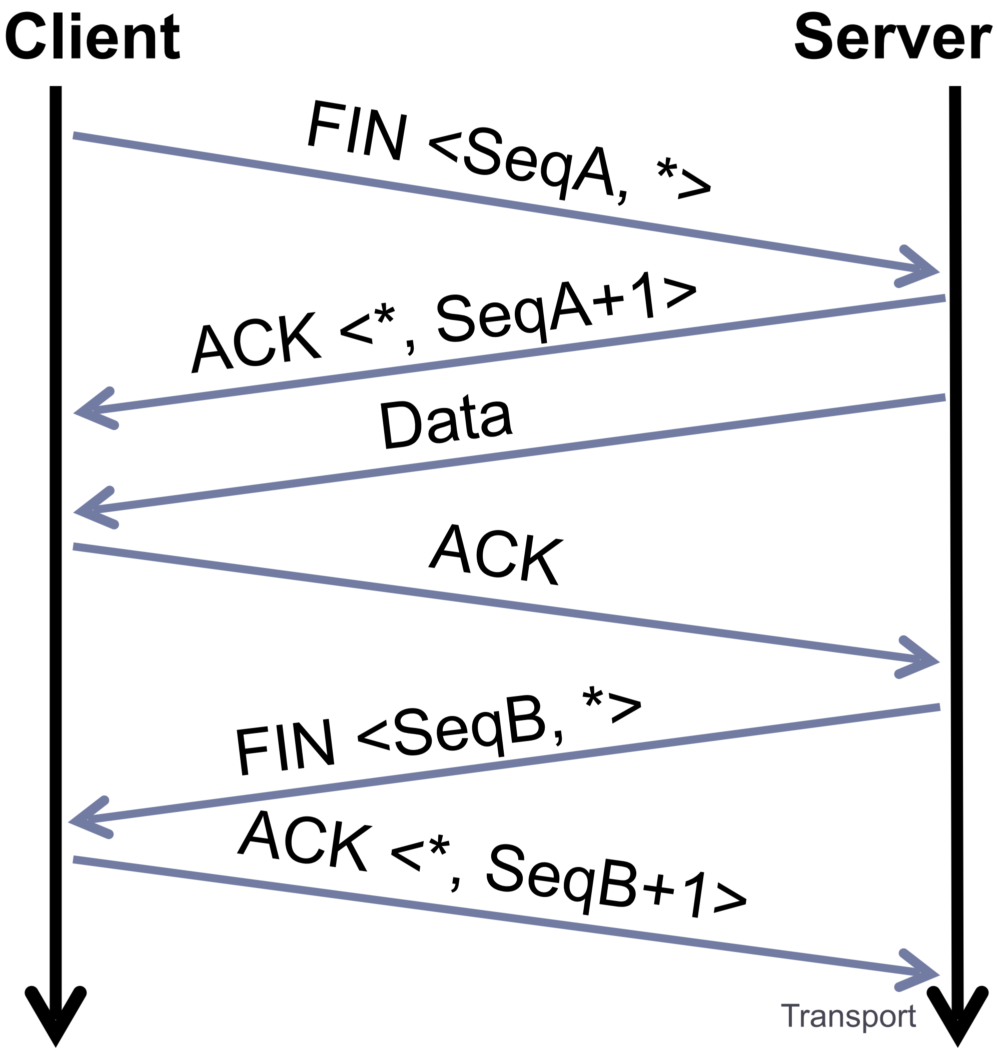
- Acknowledge the last FIN
  - Sequence number + 1



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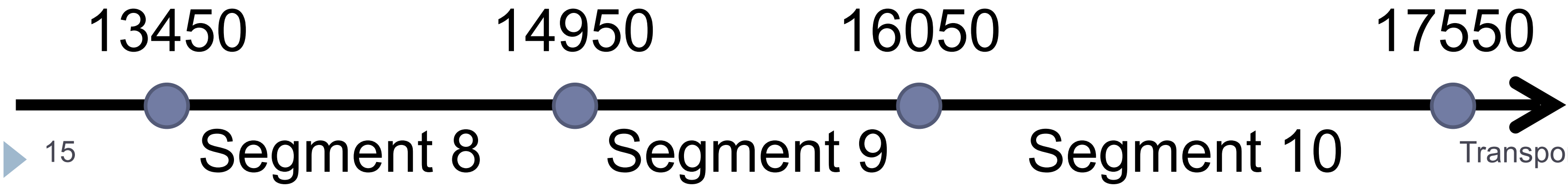
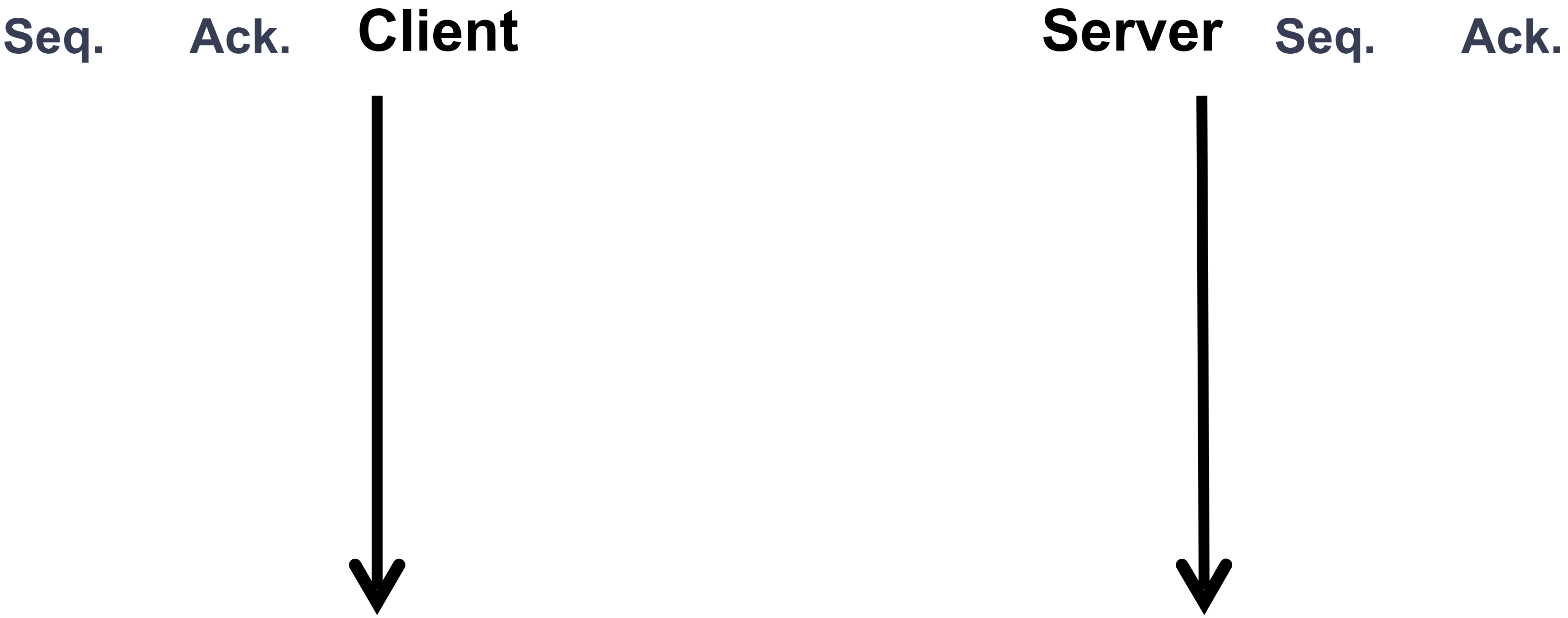


13450 14950 16050 17550

Segment 8 Segment 9 Segment 10

Bidirectional Communication



- Each side of the connection can send and receive
  - Different sequence numbers for each direction

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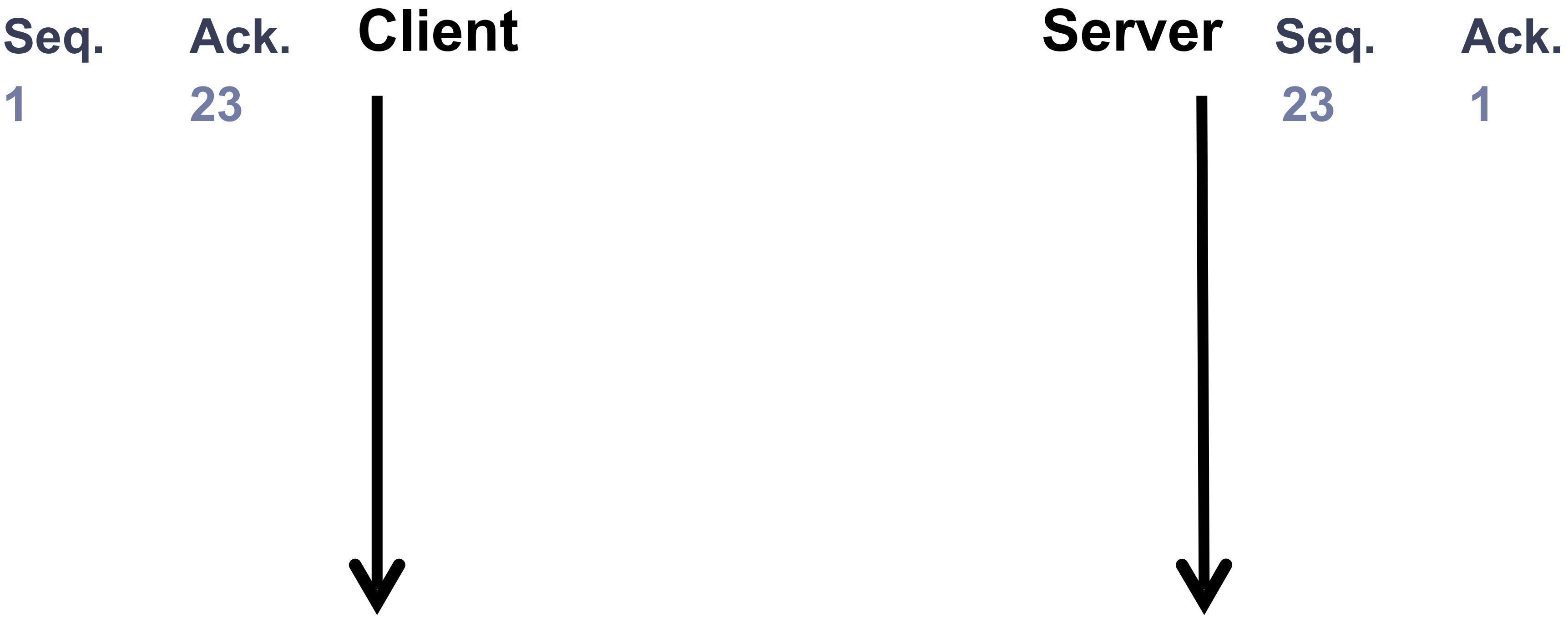
Seq. Ack. Client

1 23

Server Seq. Ack.

23 1

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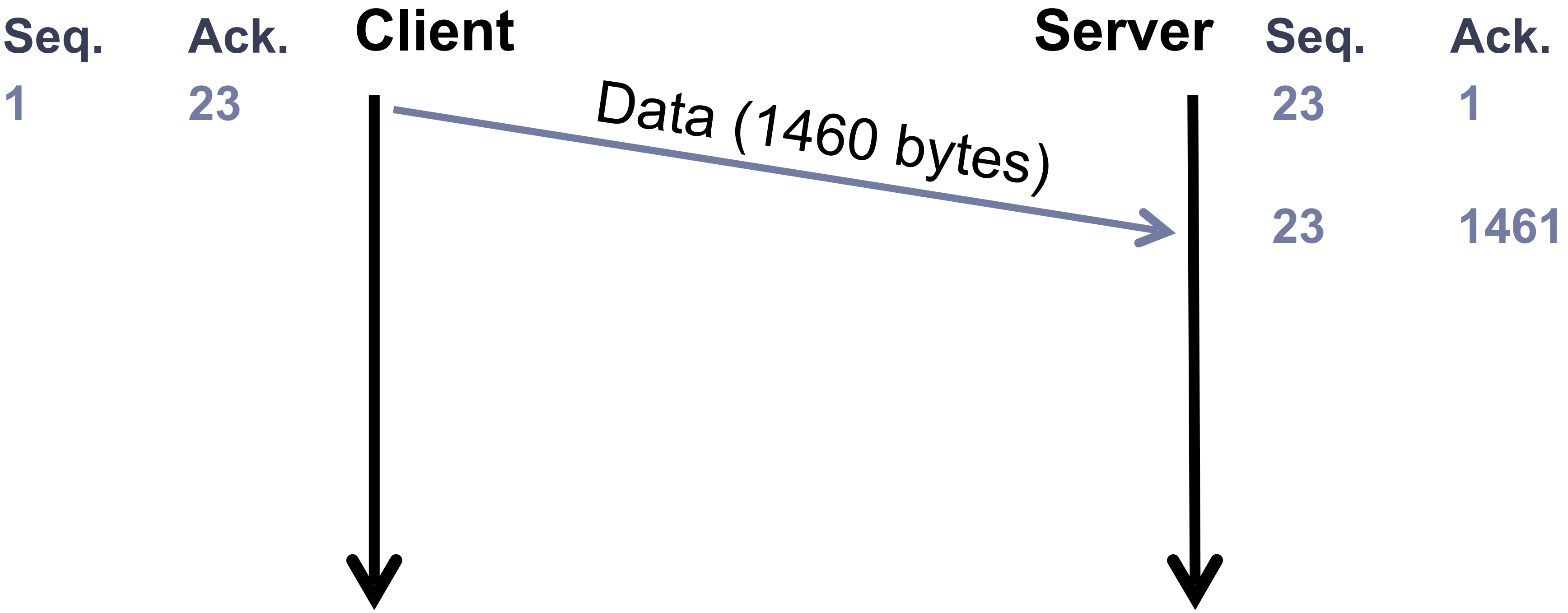

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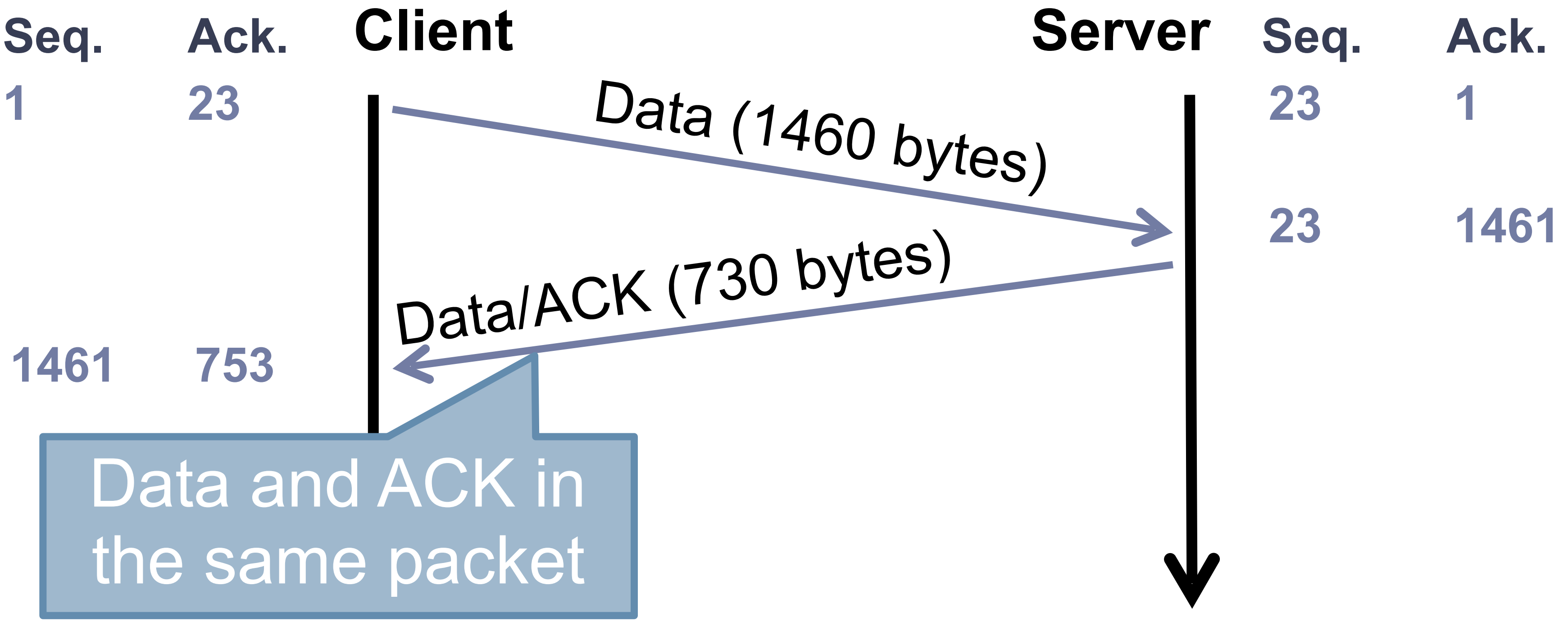

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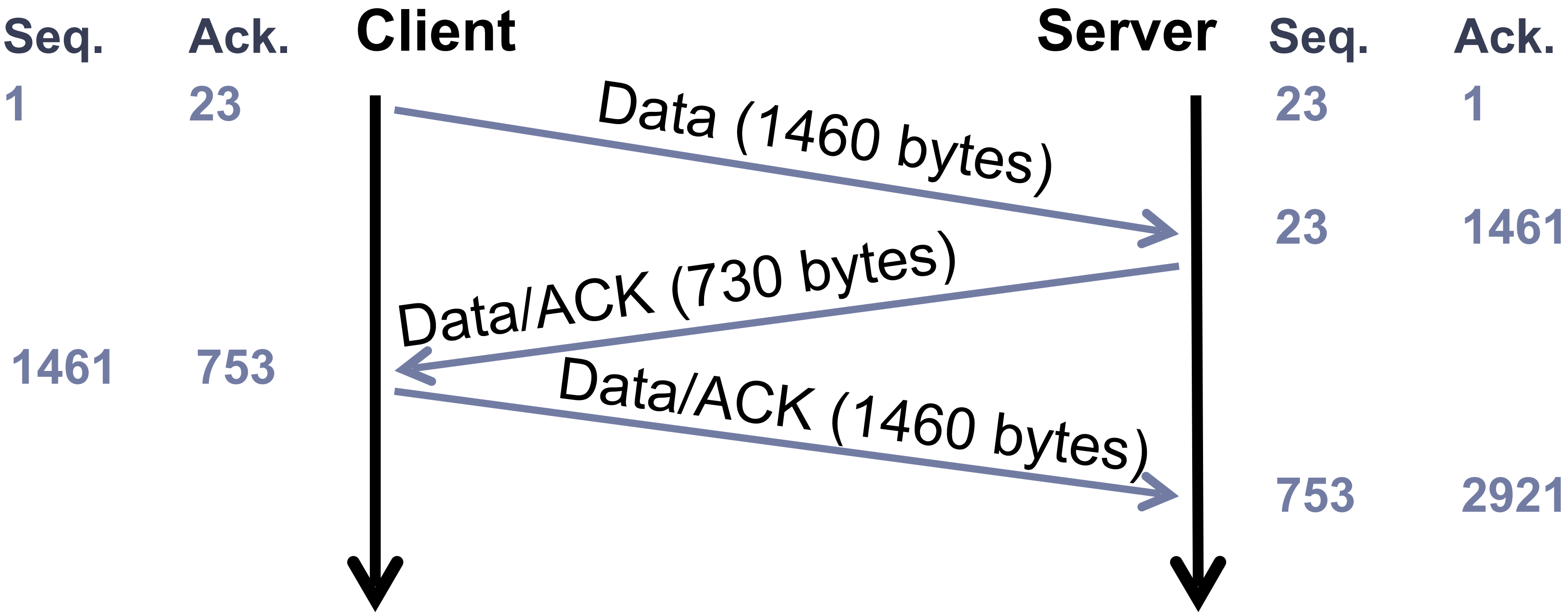

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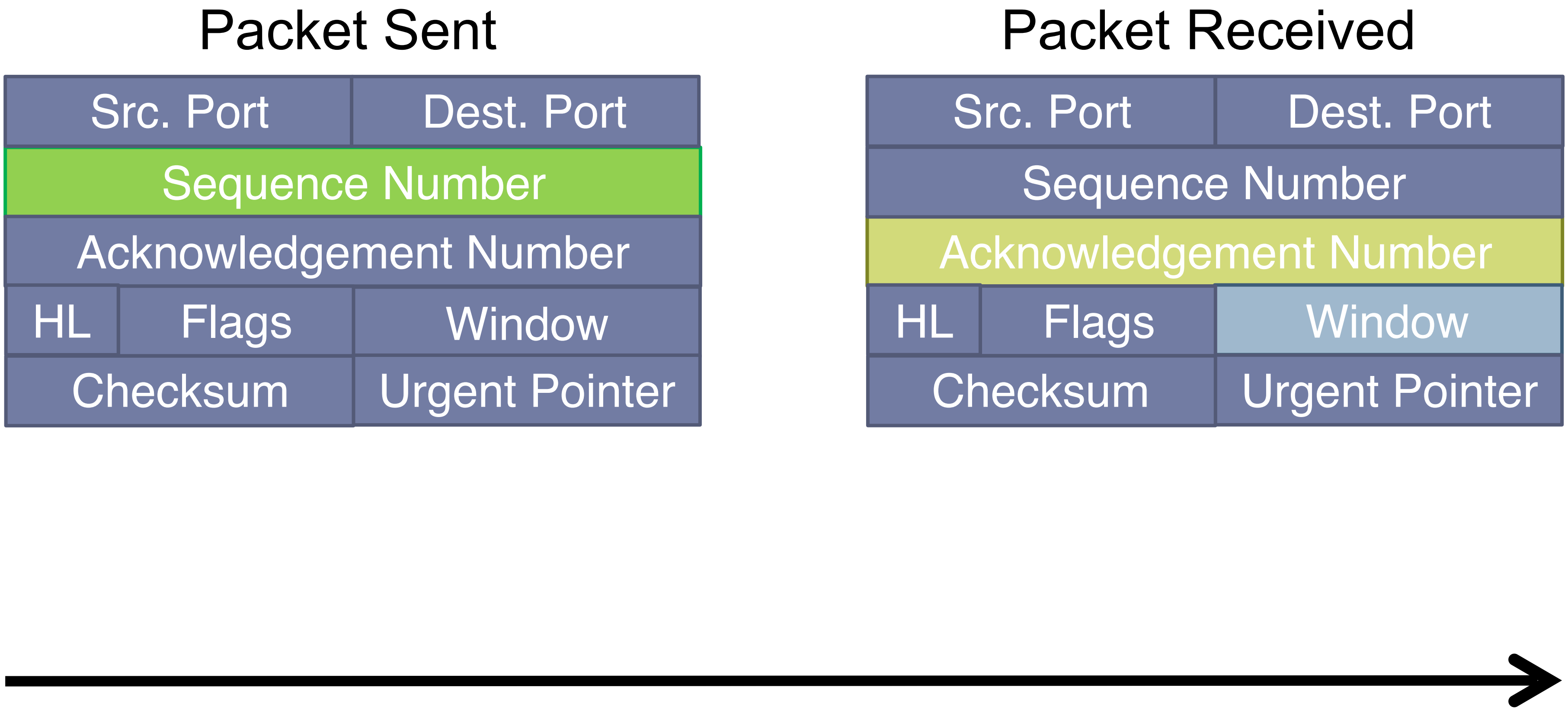
- Window may go to zero!

Flow Control: Sender Side



Packet Sent

Src. Port		Dest. Port	
Sequence Number			
Acknowledgement Number			
HL	Flags	Window	
Checksum		Urgent Pointer	

Packet Received

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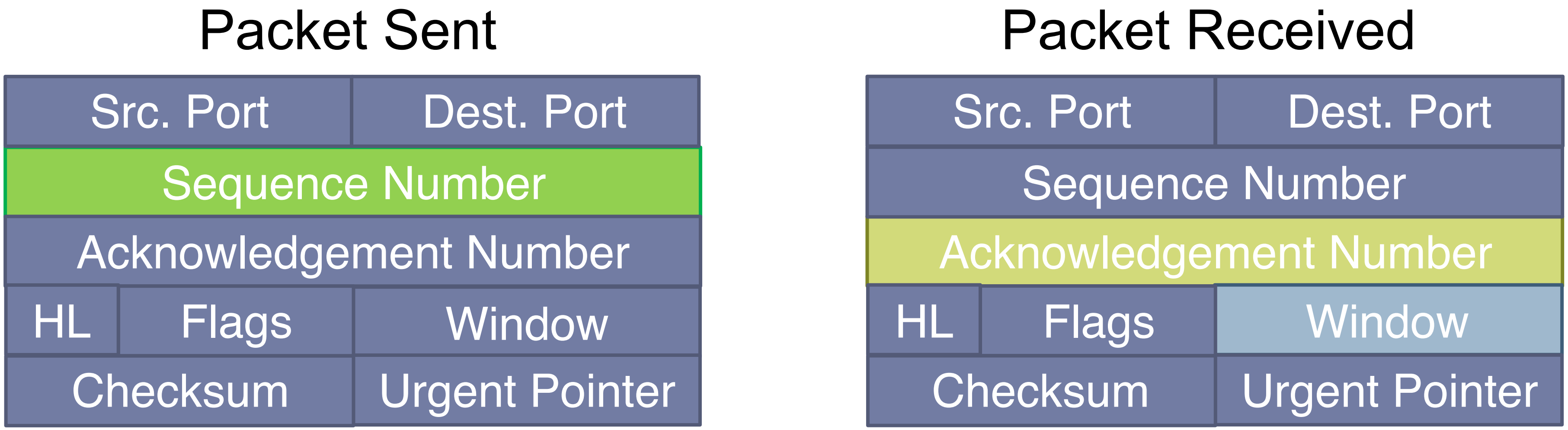

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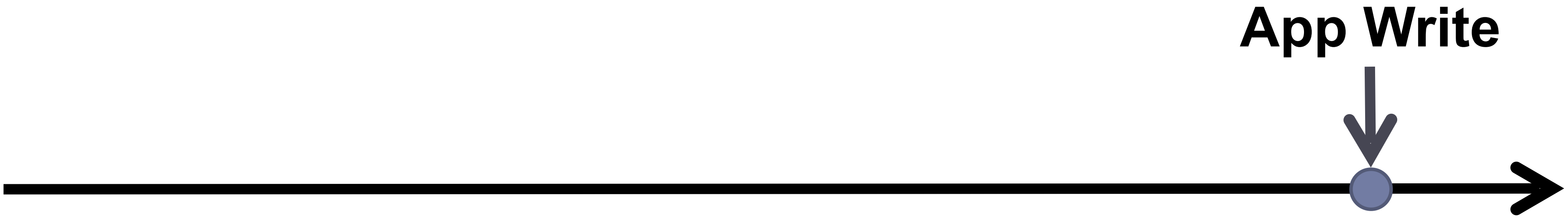

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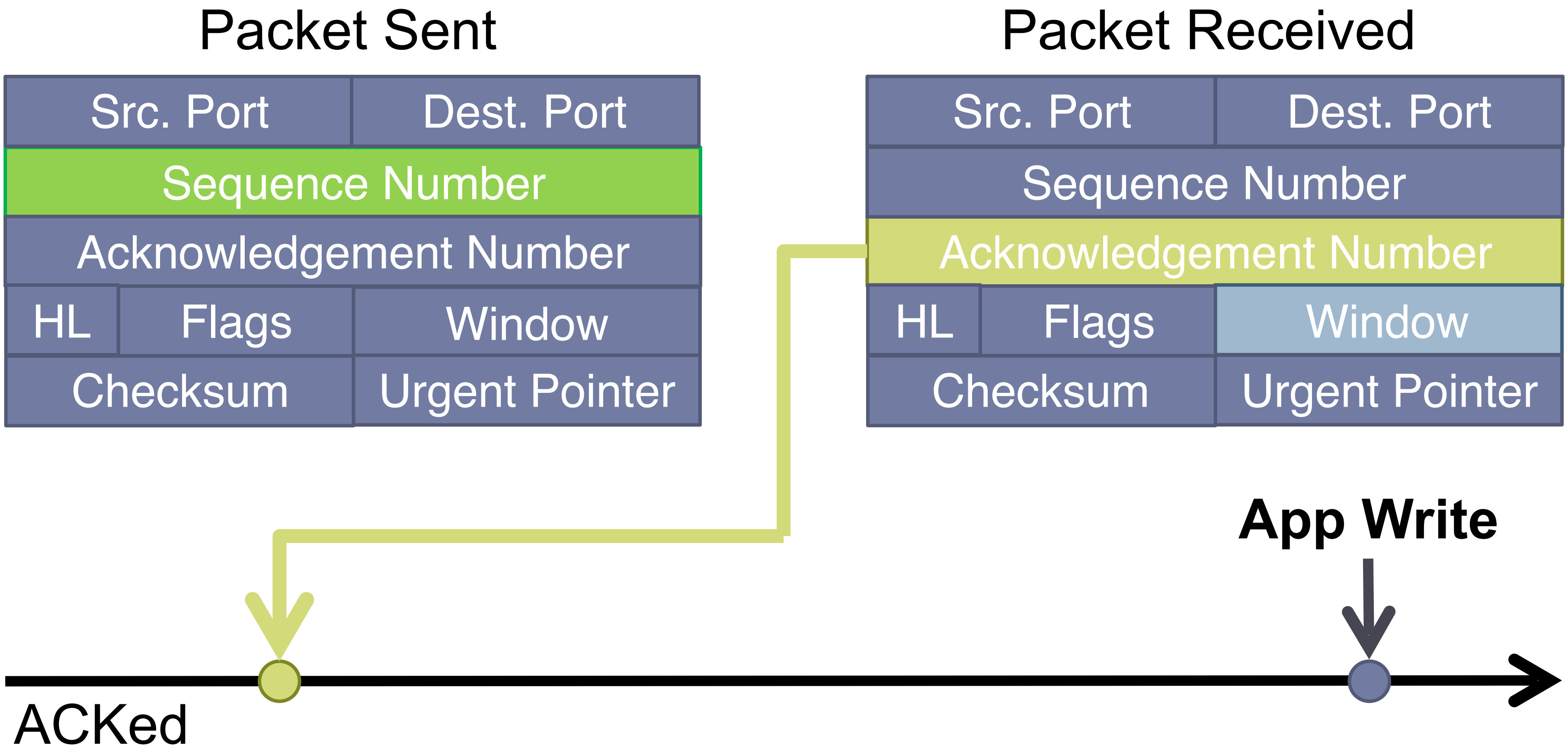

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App Write

ACKed

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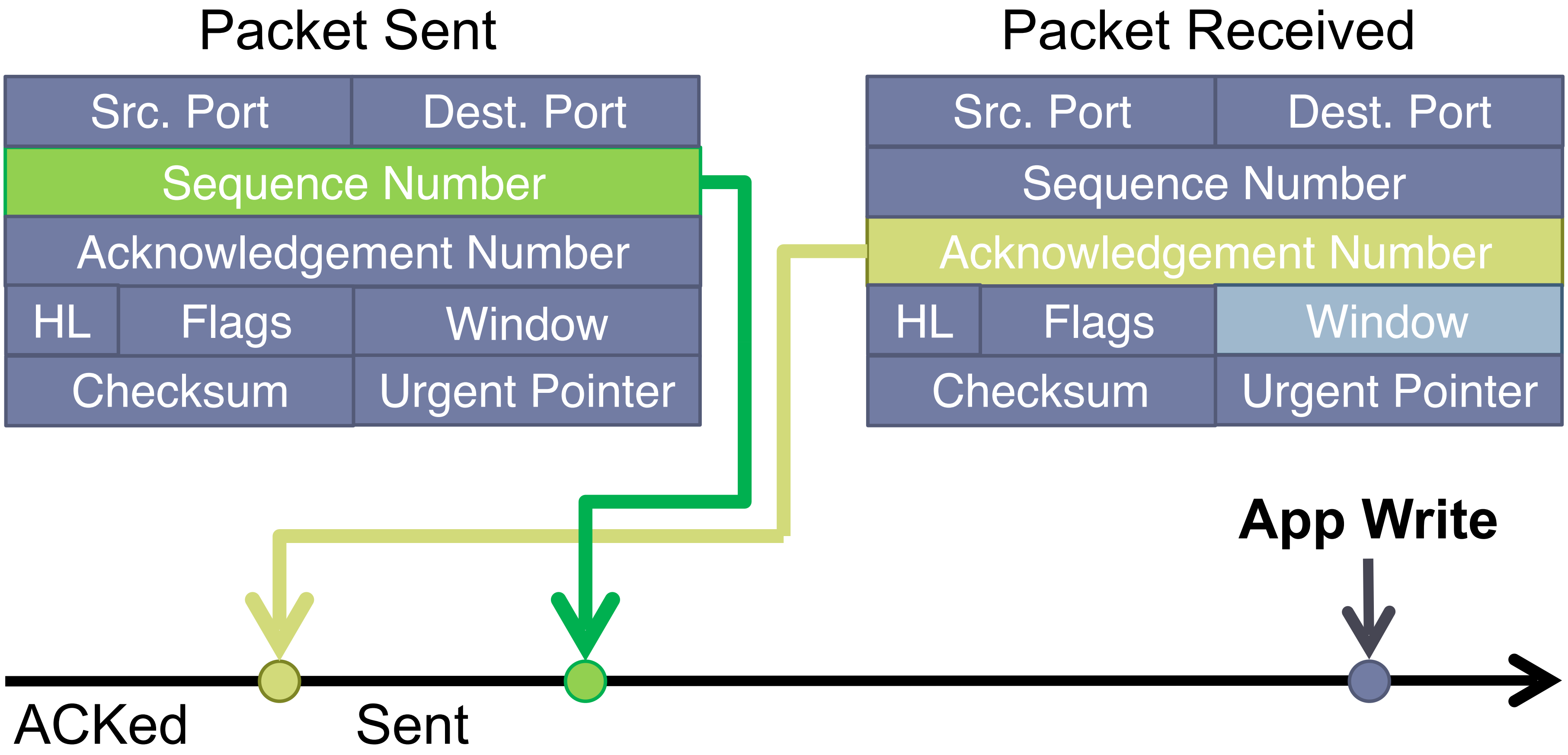

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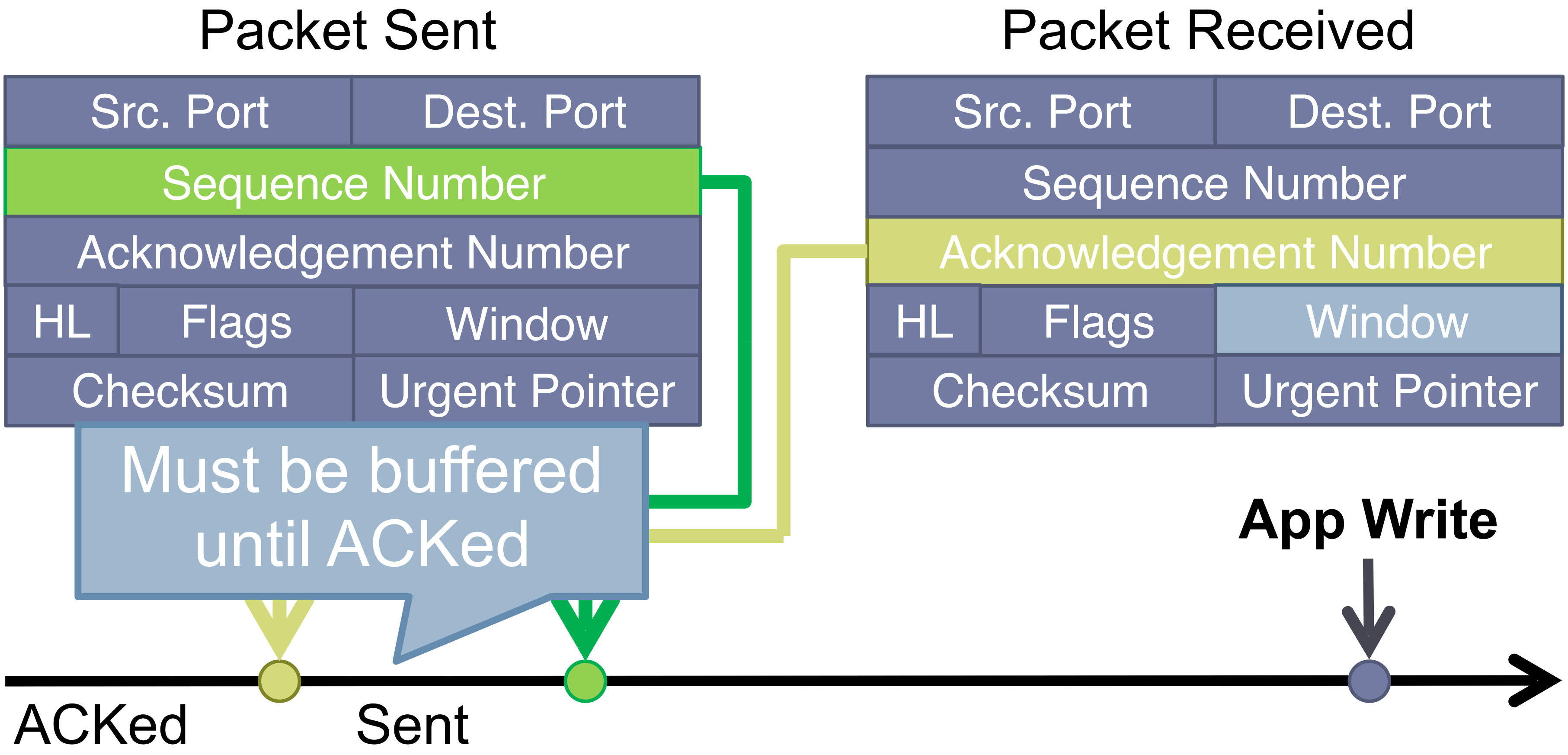

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Must be buffered until ACKed

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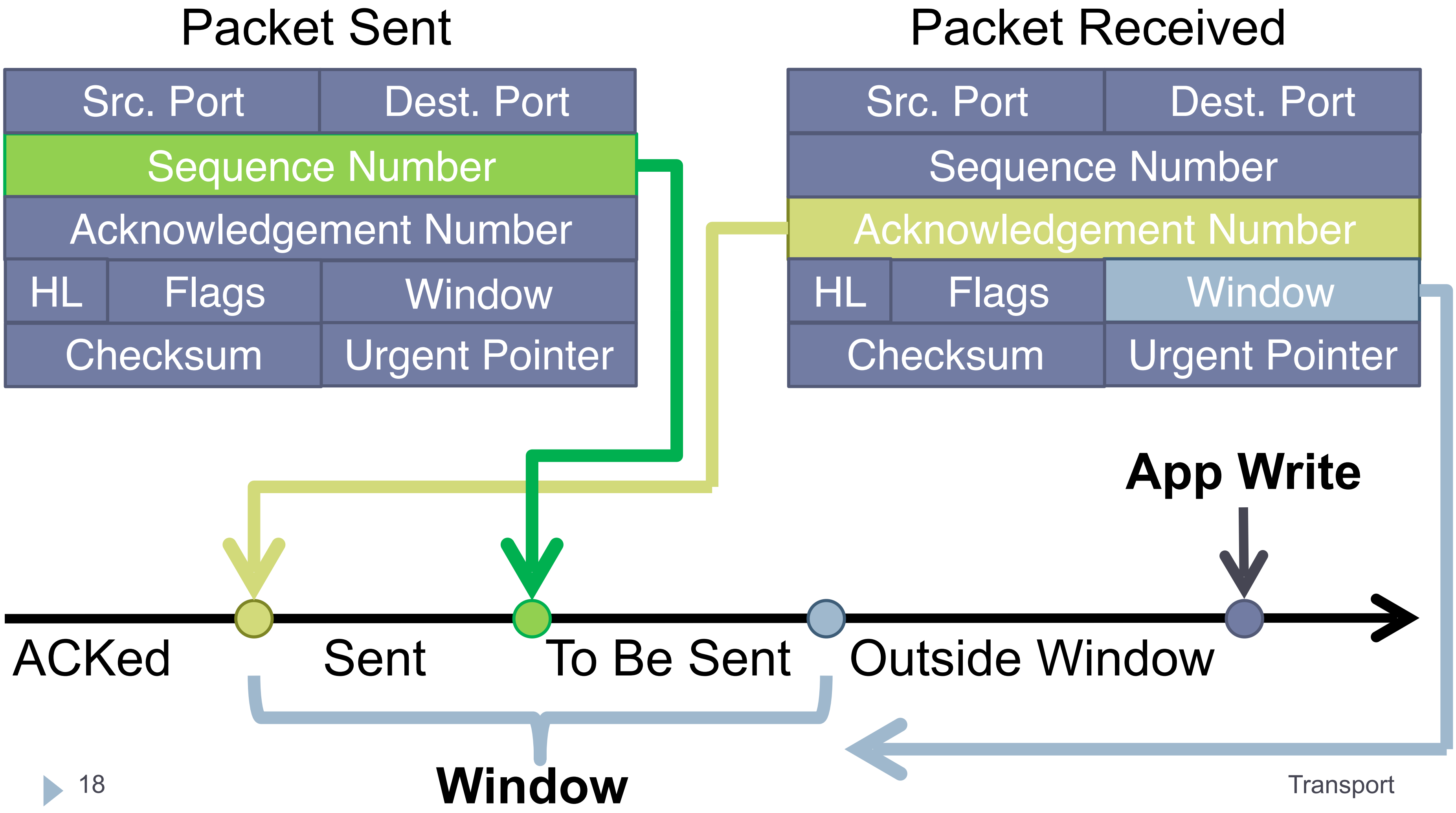

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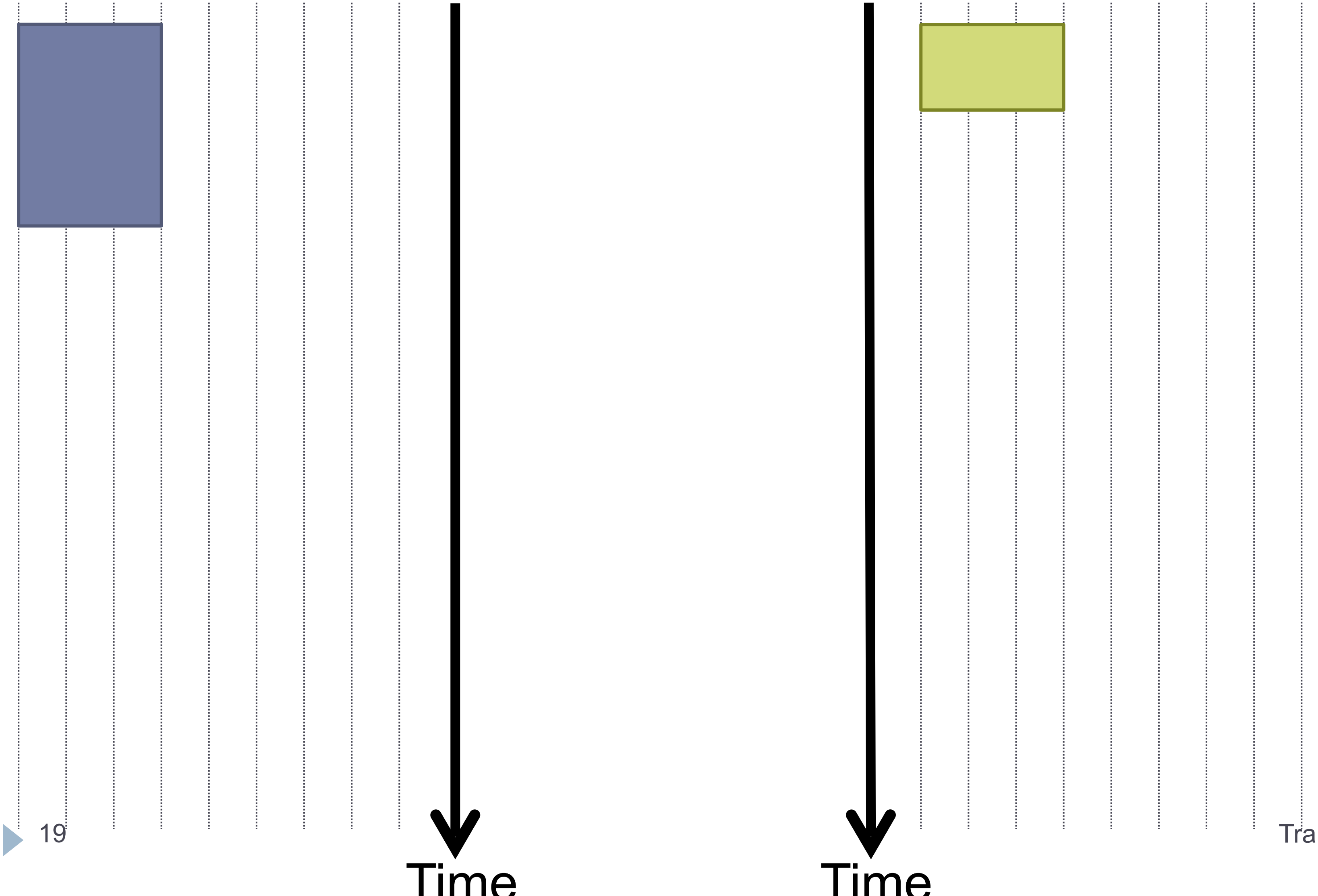
Sliding Window Example



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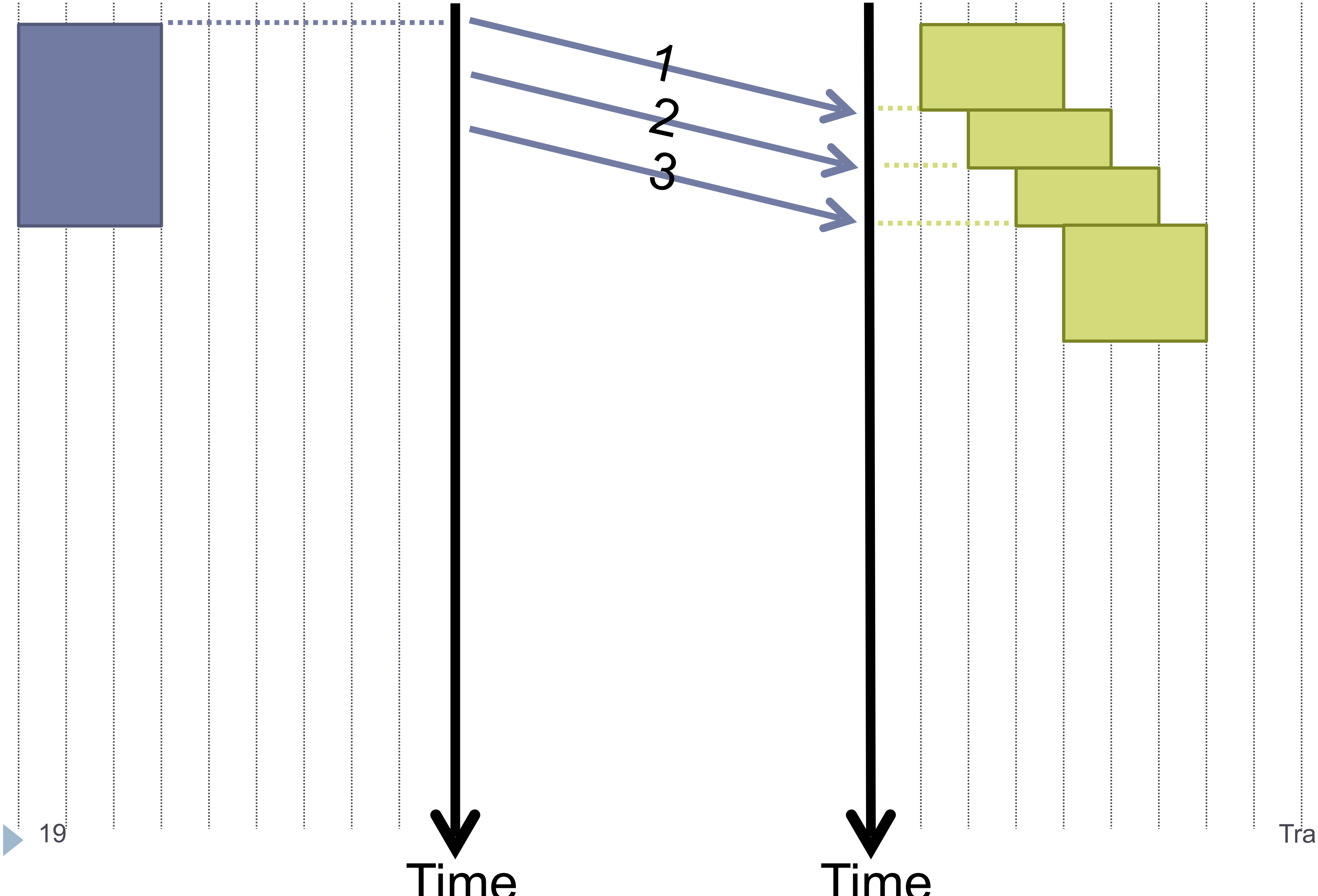

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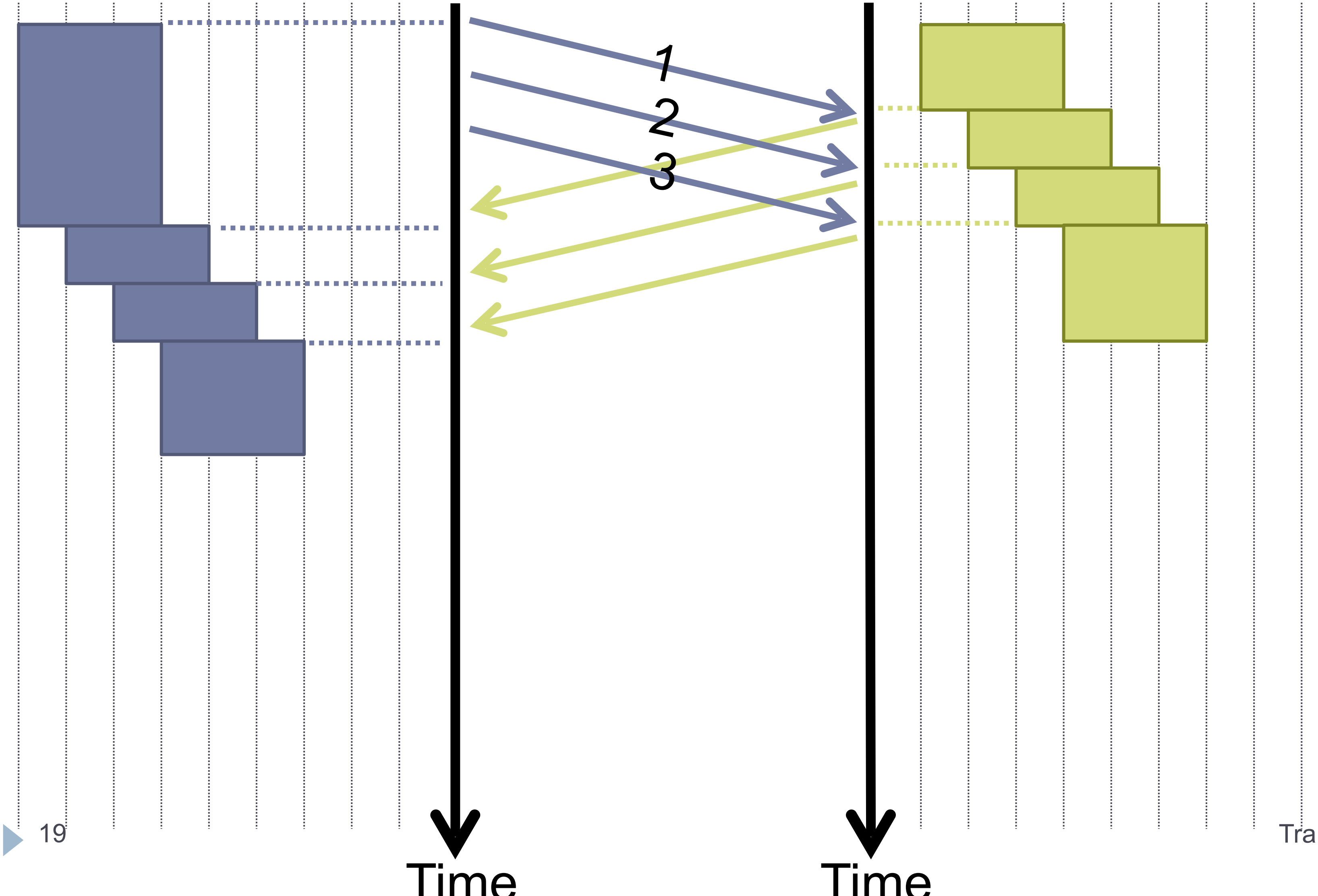

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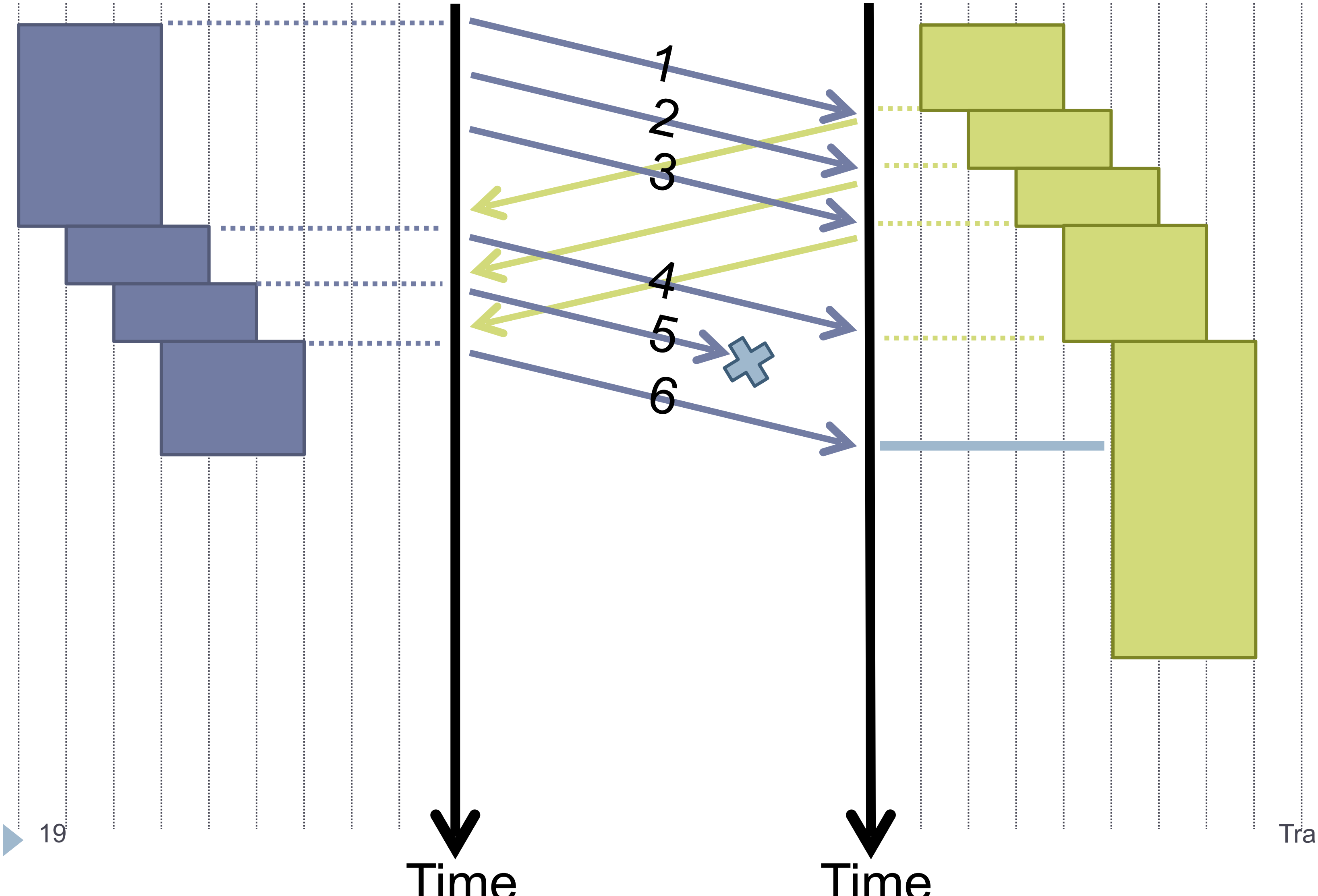

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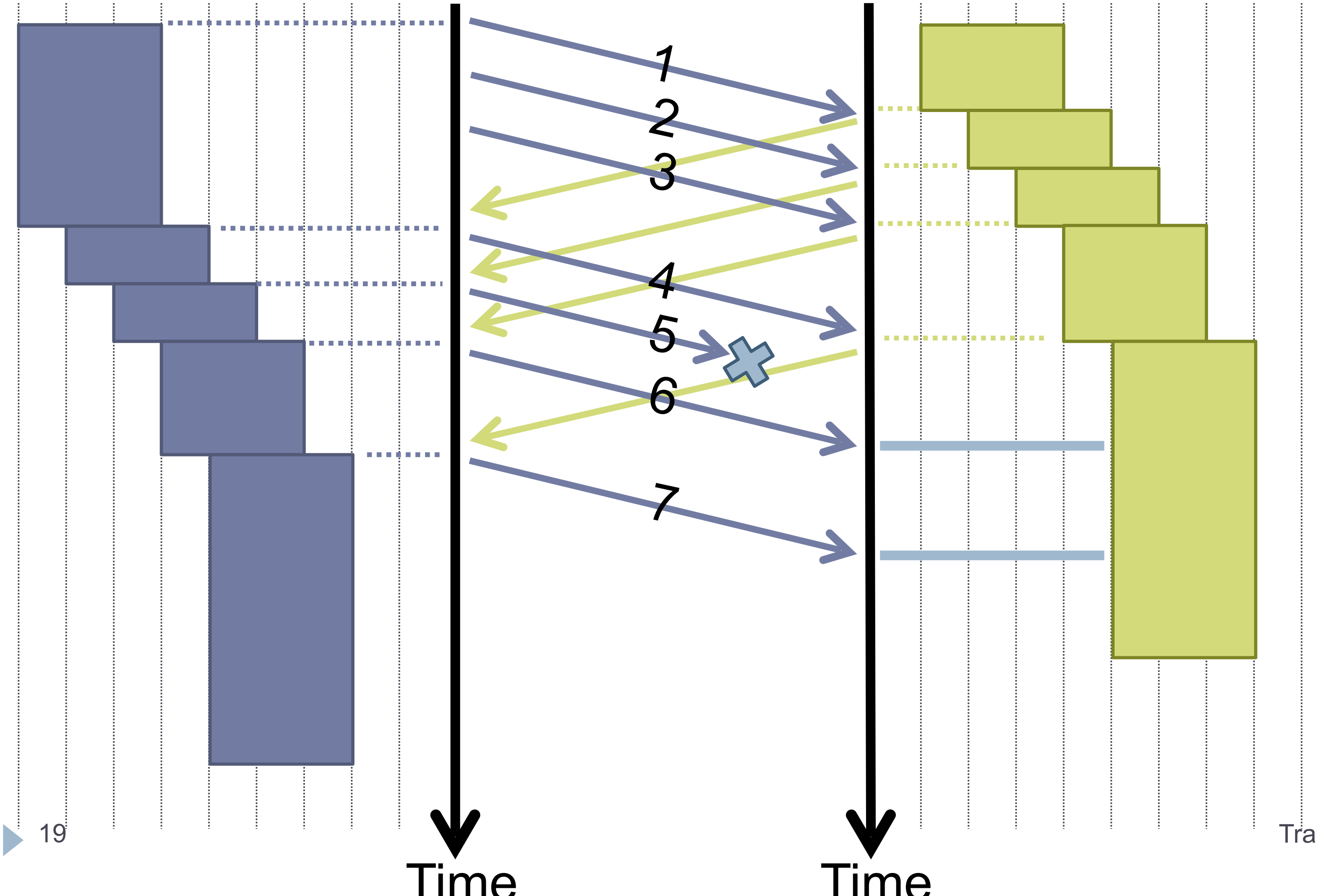

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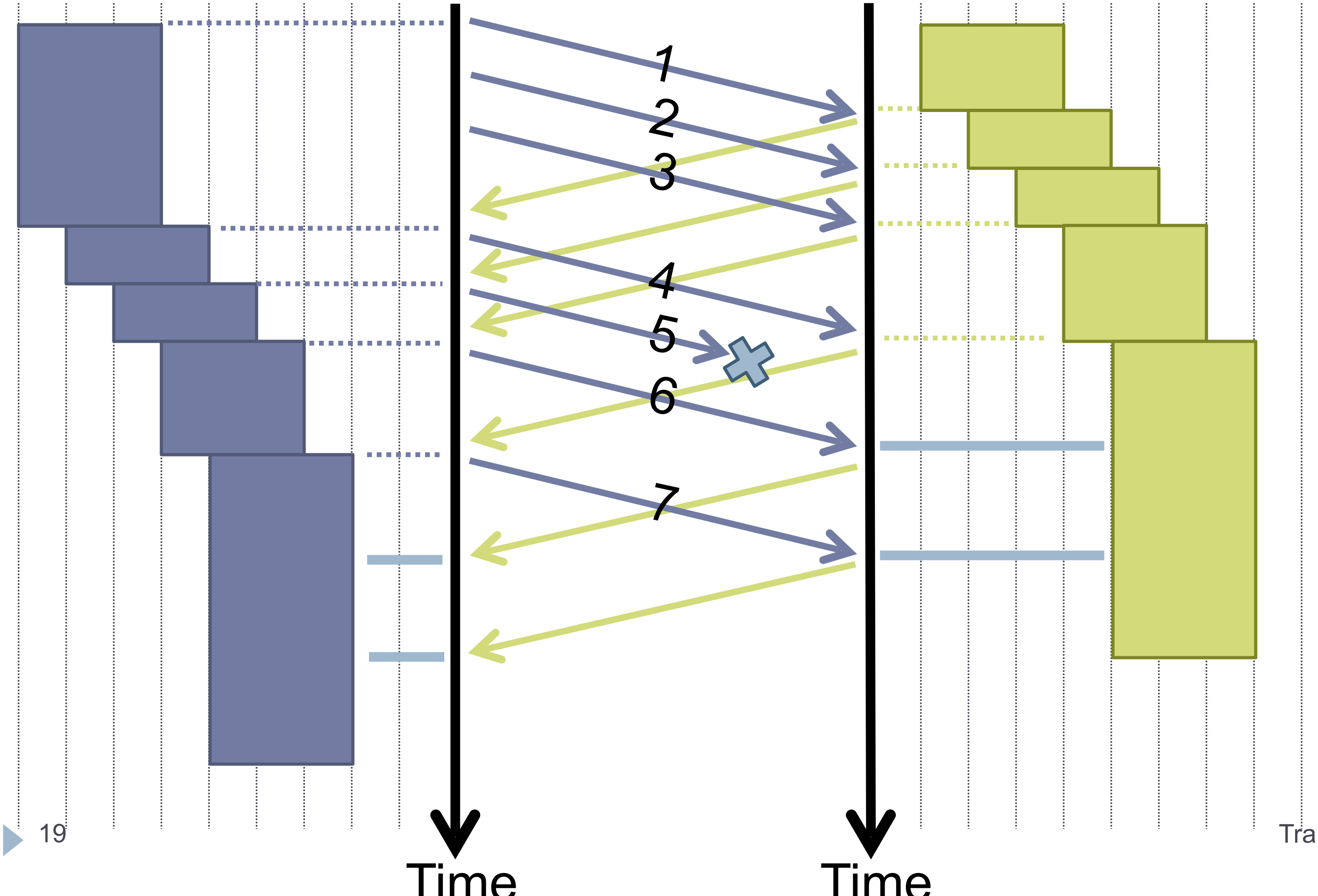

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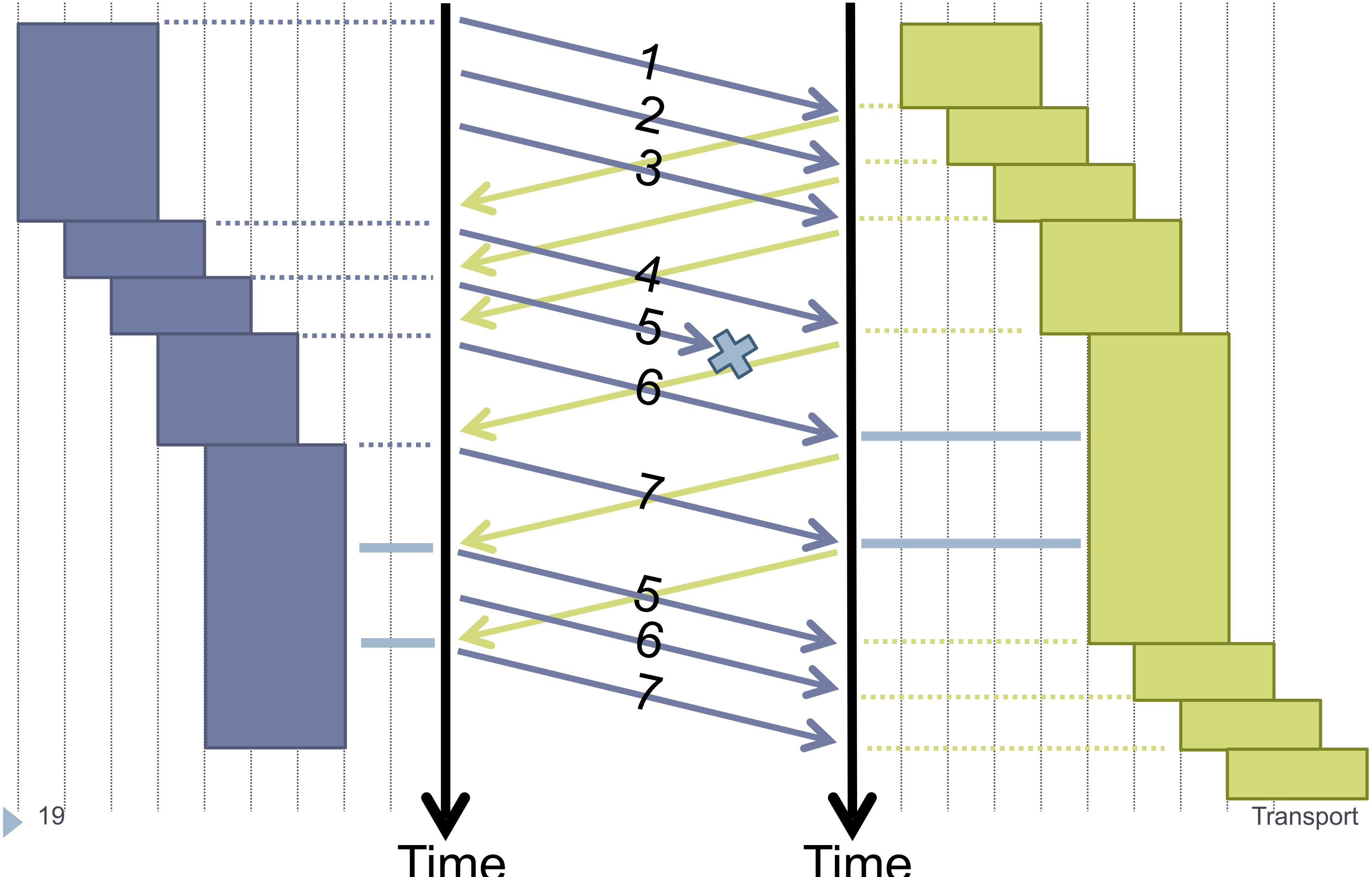

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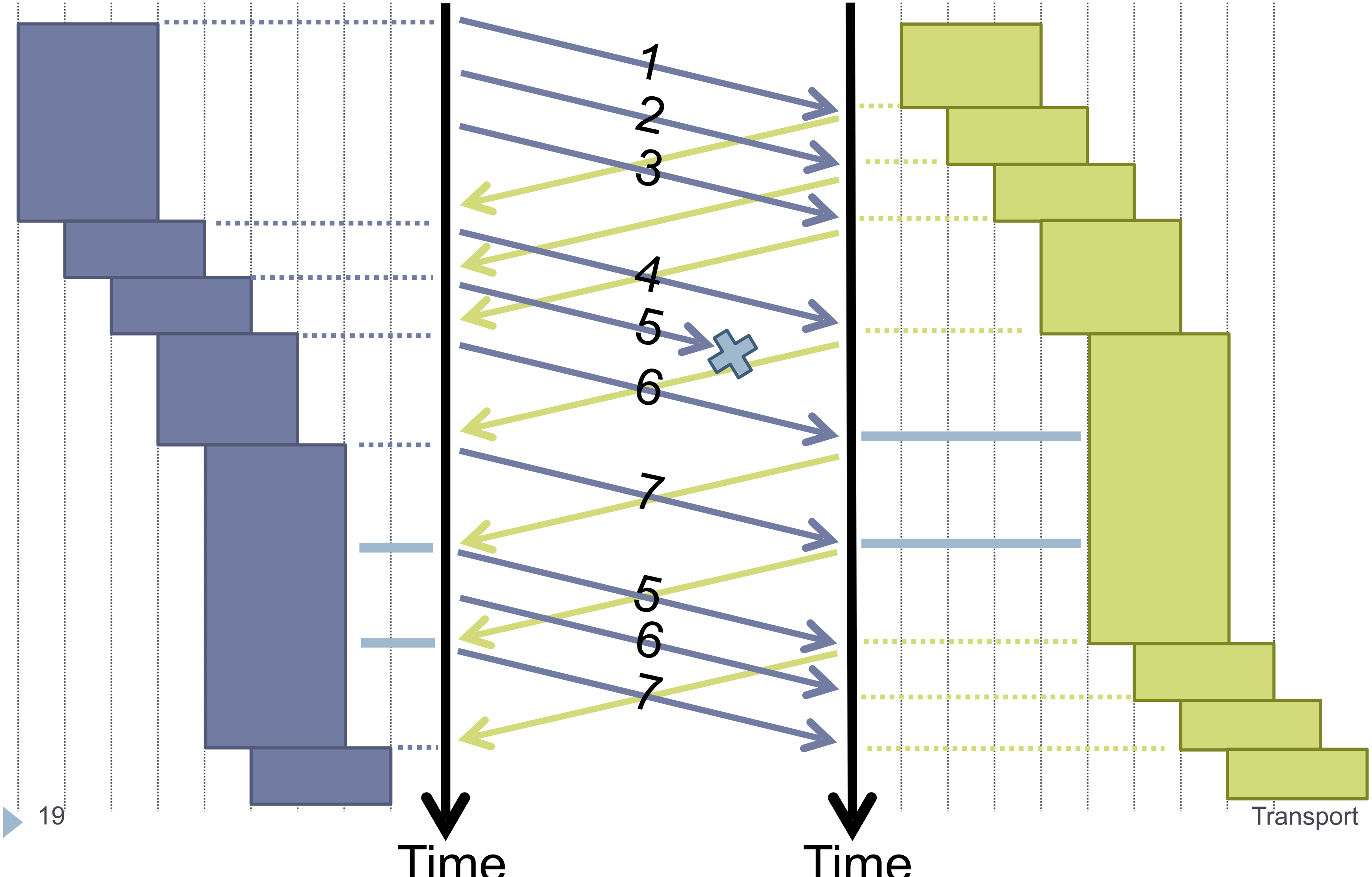

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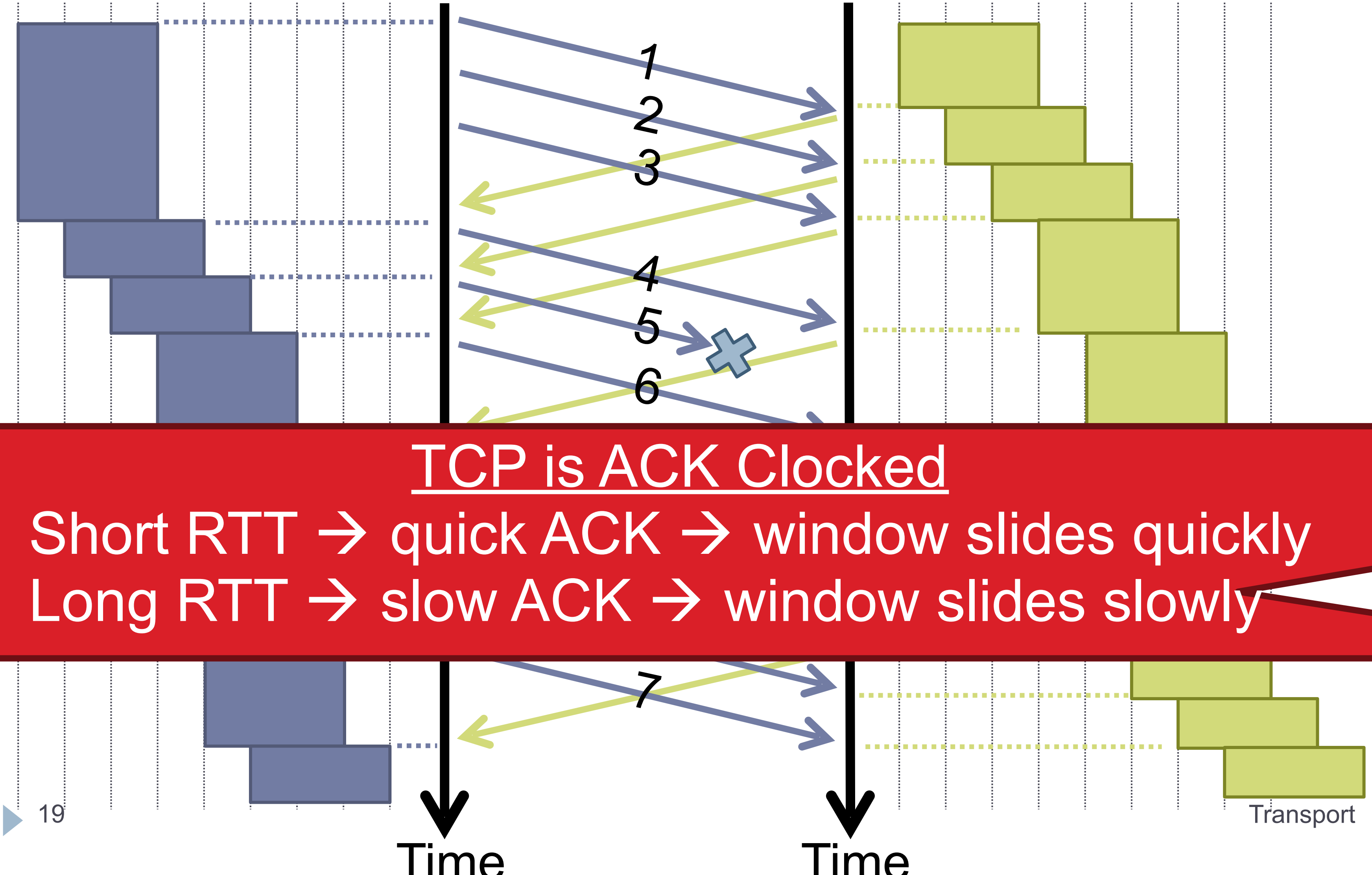

TCP is ACK Clocked

- Short RTT → quick ACK → window slides quickly
- Long RTT → slow ACK → window slides slowly

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Sequence Numbers, Revisited

- 32 bits, unsigned
  - Why so big?
- For the sliding window you need…
  - lSequence # Spacel > 2 * lSending Window Sizel
  - $2^{32} > 2 * 2^{16}$
- Guard against stray packets
  - IP packets have a maximum segment lifetime (MSL) of 120 seconds
    - i.e. a packet can linger in the network for 2 minutes
  - Sequence number would wrap around at 286Mbps
    - What about GigE? PAWS algorithm + TCP options (timestamp)

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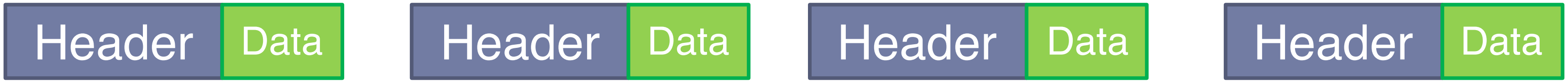

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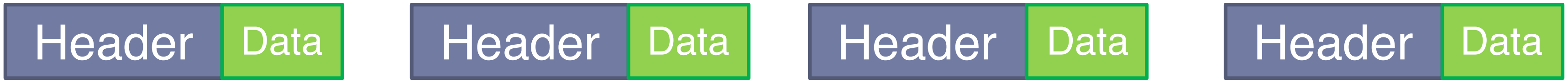

Header Data Header Data Header Data Header Data

- Equivalent problem: sender transmits packets one byte at a time

1. for (int x = 0; x < strlen(data); ++x)
2.   write(socket, data + x, 1);

Nagle's Algorithm

1. If the window >= MSS and available data >= MSS:

   Send the data

2. Elif there is unACKed data:

   Enqueue data in a buffer (send after a timeout)

3. Else: send the data

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Send a full packet

Send a non-full packet if nothing else is happening

- Problem: Nagle's Algorithm delays transmissions
  - What if you need to send a packet immediately?
    1. int flag = 1;
    2. setsockopt(sock, IPPROTO_TCP, TCP_NODELAY,
                  (char *) &flag, sizeof(int));

Error Detection

- Checksum detects (some) packet corruption
  - Computed over IP header, TCP header, and data
- Sequence numbers catch sequence problems
  - Duplicates are ignored
  - Out-of-order packets are reordered or dropped
  - Missing sequence numbers indicate lost packets
- Lost segments detected by sender
  - Use timeout to detect missing ACKs
  - Need to estimate RTT to calibrate the timeout
  - Sender must keep copies of all data until ACK

Retransmission Time Outs (RTO)

- Problem: time-out is linked to round trip time

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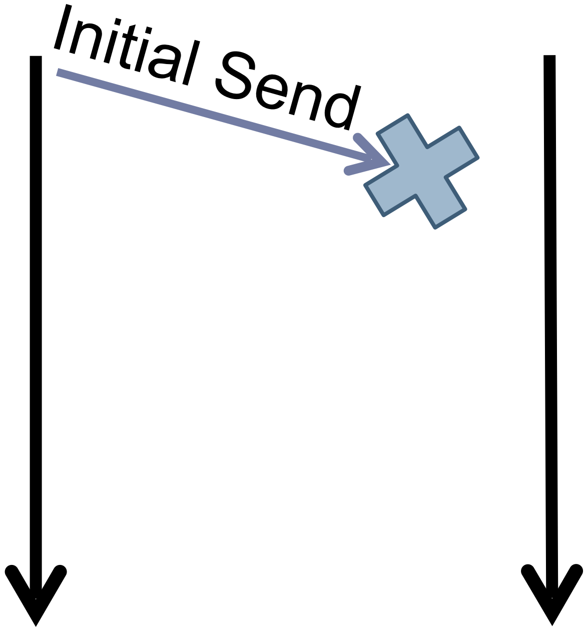

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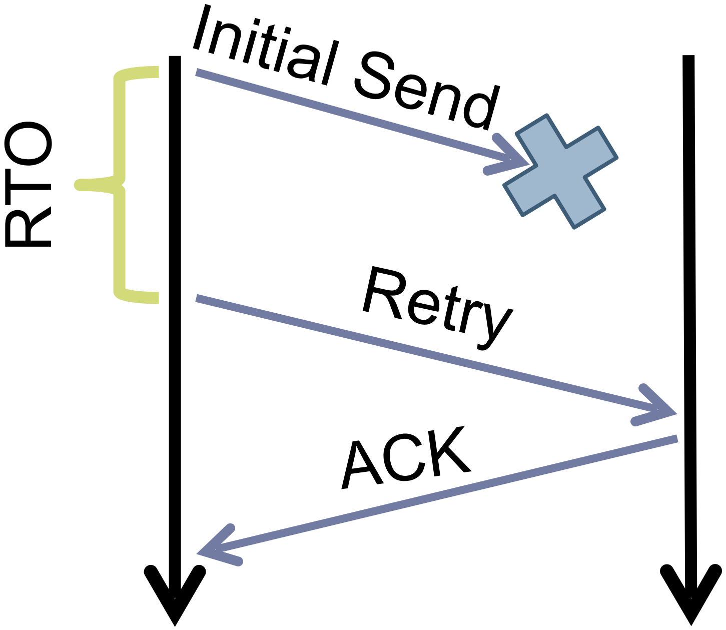

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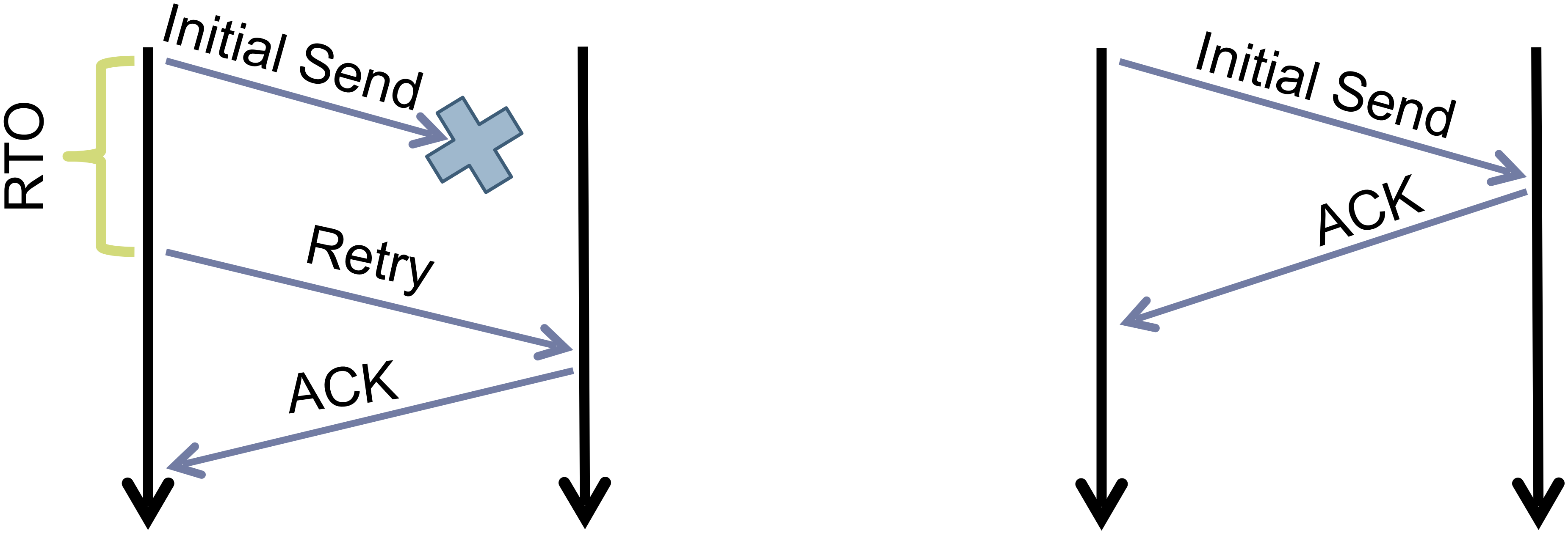

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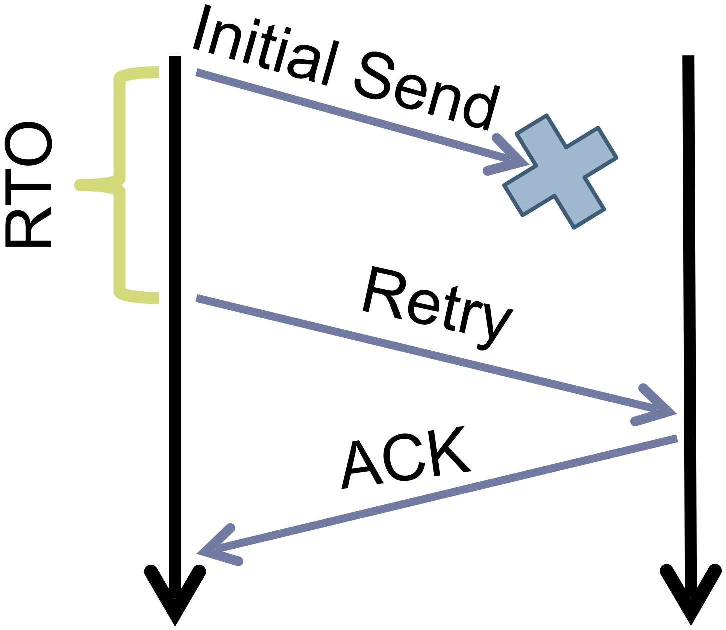

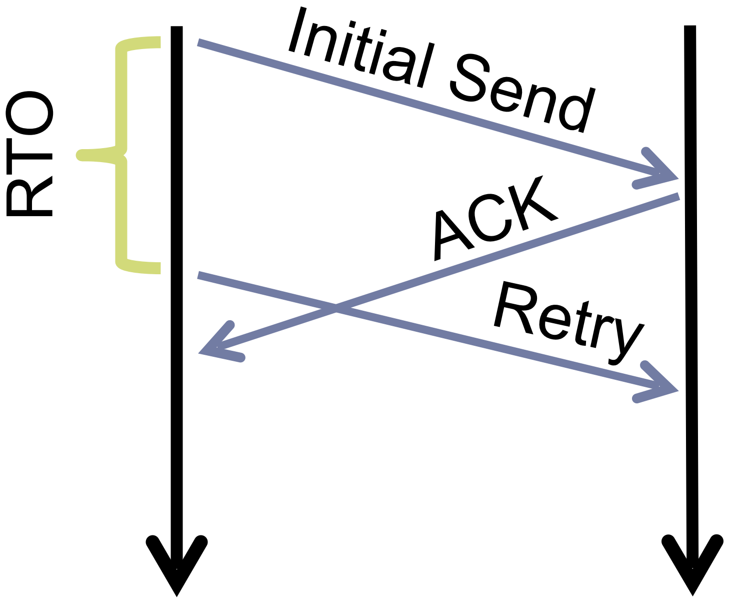

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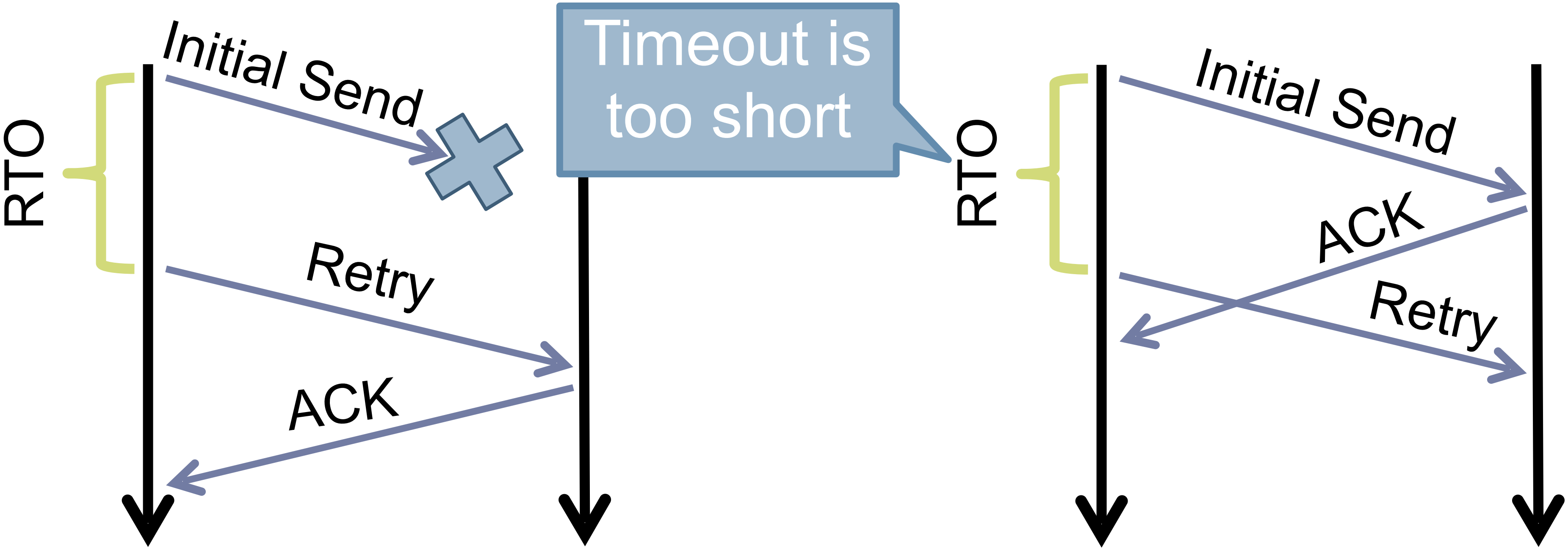

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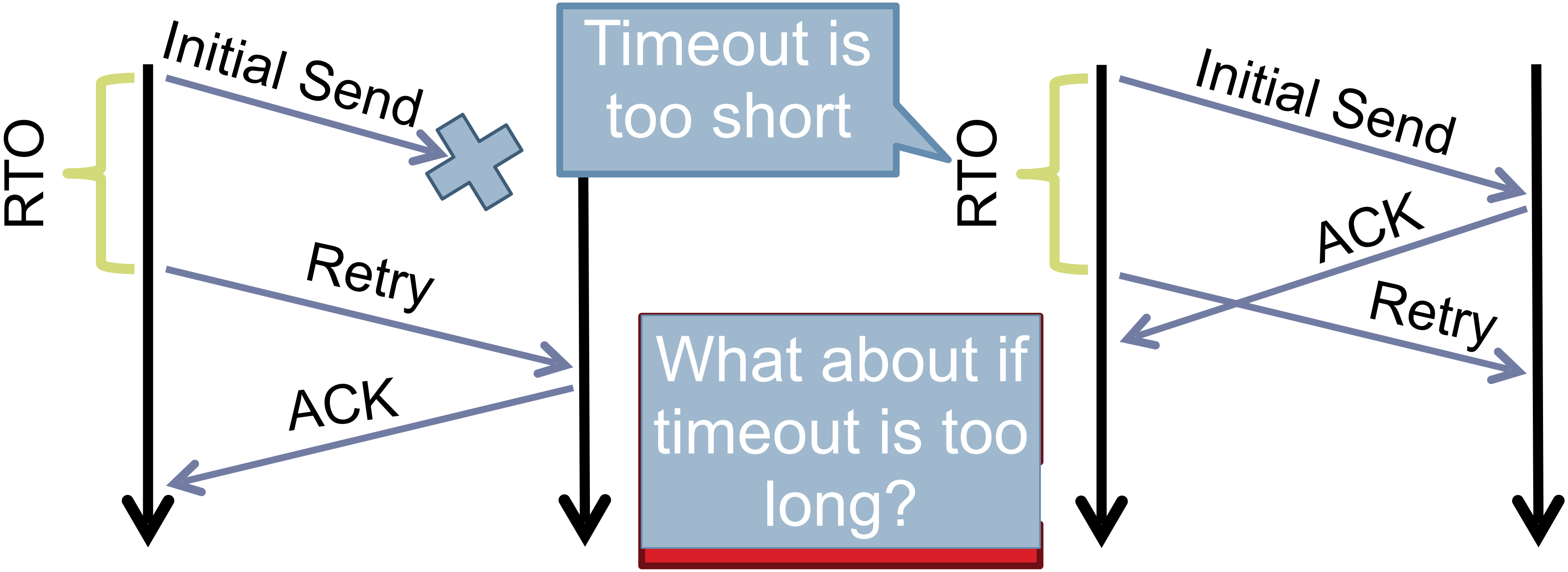
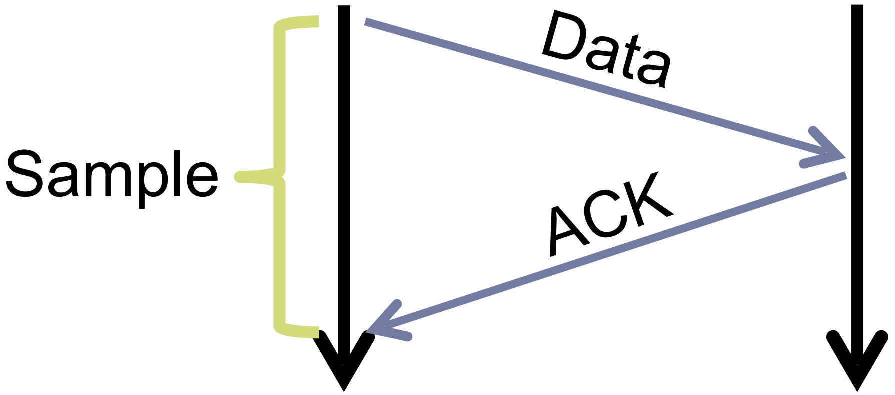


Timeout is too short

What about if timeout is too long?

Round Trip Time Estimation



- Original TCP round-trip estimator
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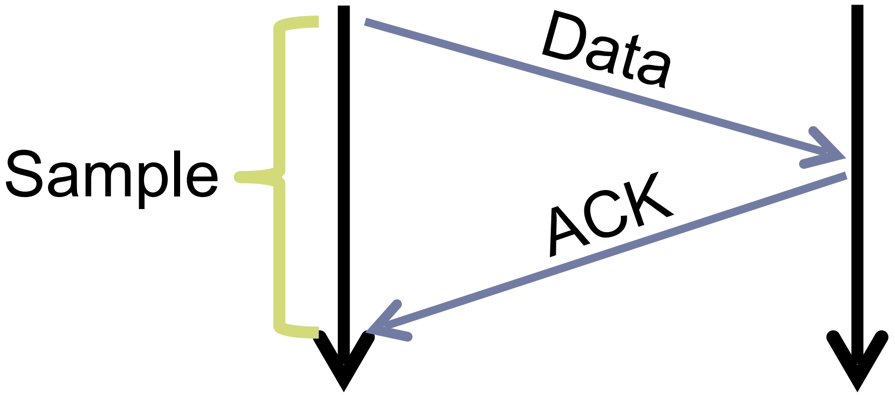

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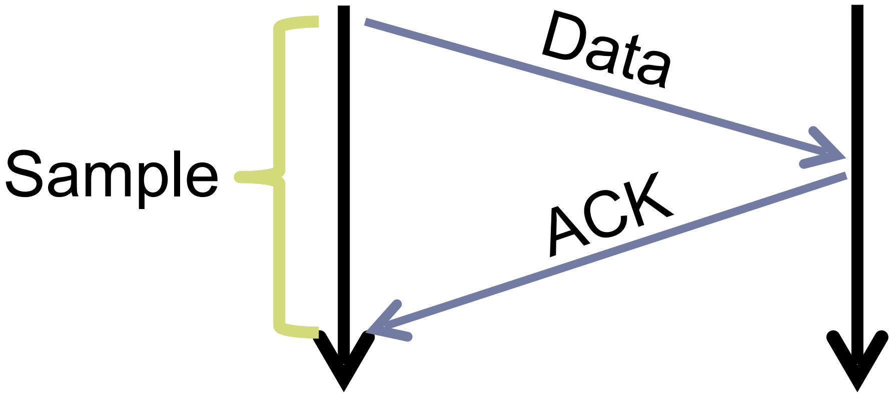

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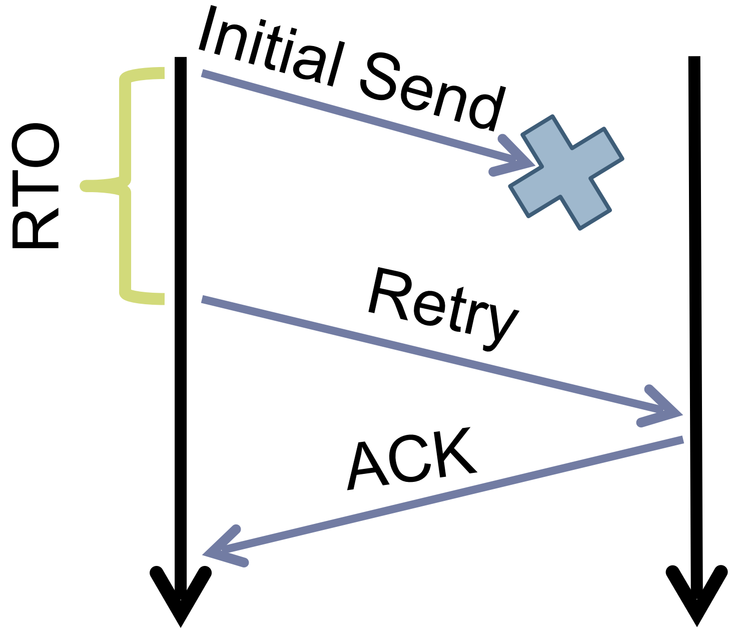
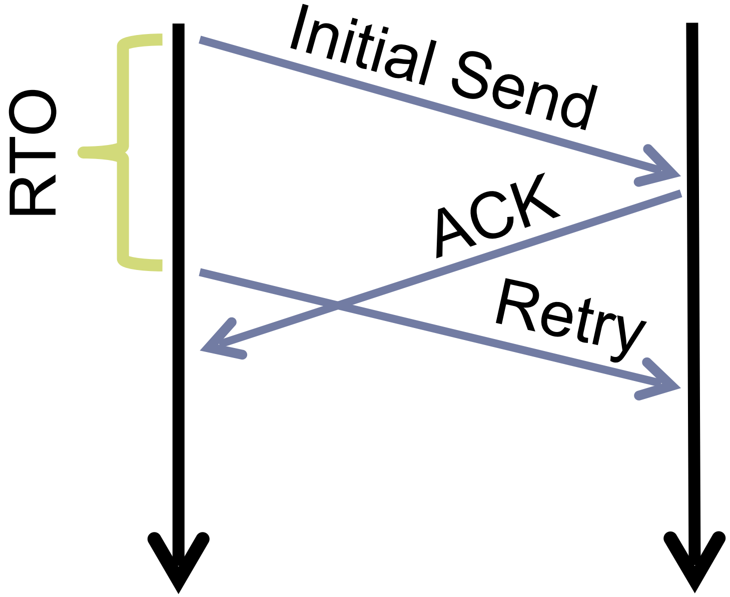
  - Today: RTO = new_rtt + 4*DevRTT

RTT Sample Ambiguity





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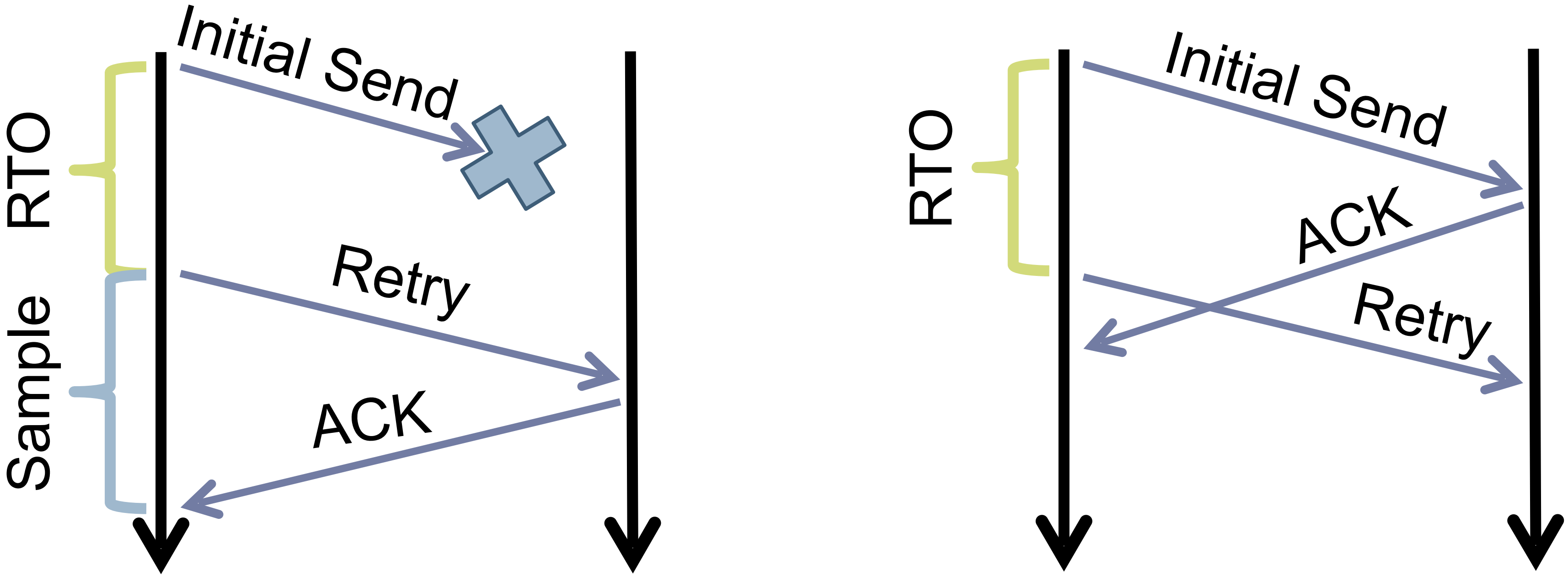

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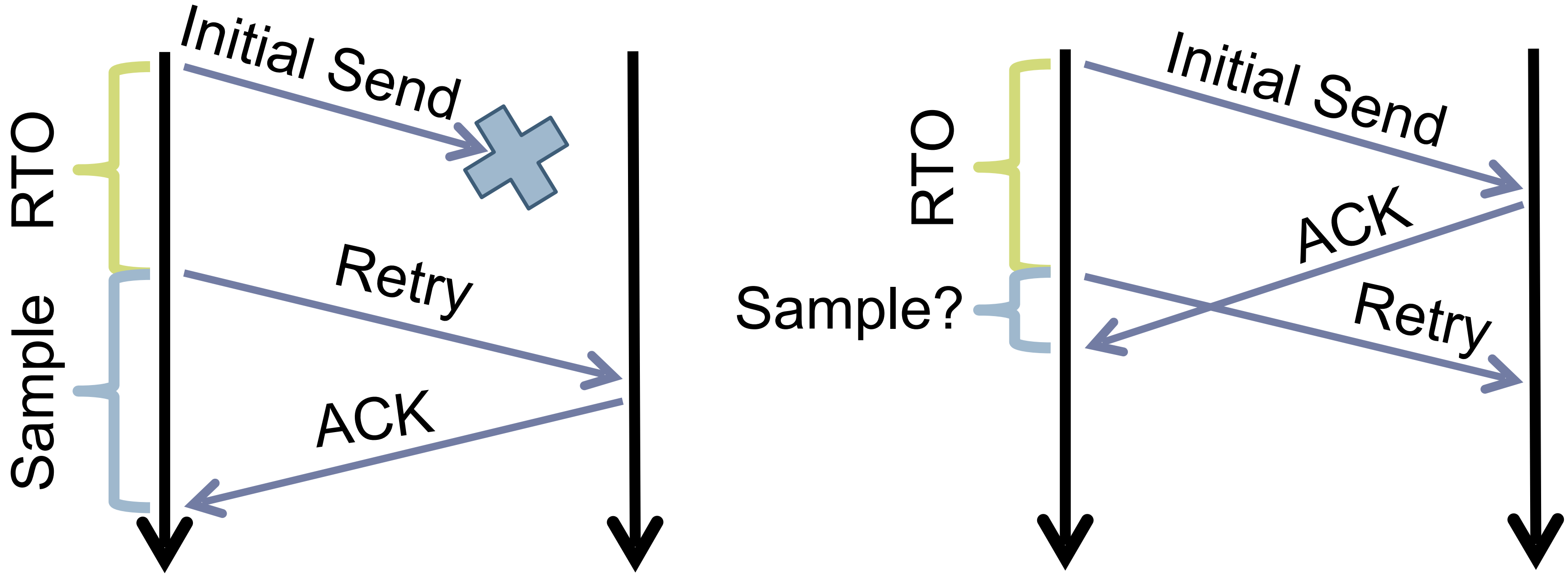

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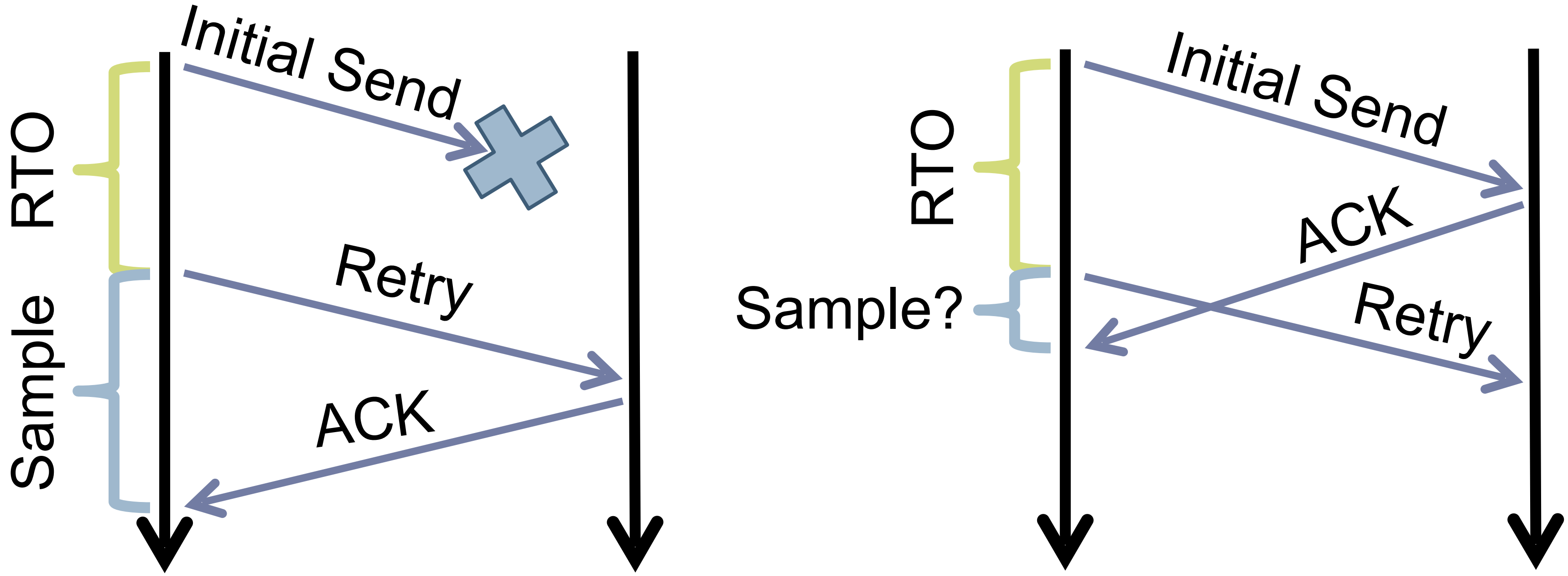

- Karn’s algorithm: ignore samples for retransmitted segments

3: Congestion control.

What is Congestion?

- Load on the network is higher than capacity
  - Capacity is not uniform across networks
    - Modem vs. Cellular vs. Cable vs. Fiber Optics
  - There are multiple flows competing for bandwidth
    - Residential cable modem vs. corporate datacenter
  - Load is not uniform over time
    - 10pm, Sunday night = Bittorrent Game of Thrones

Why is Congestion Bad?

- Results in packet loss
  - Routers have finite buffers
  - Internet traffic is ***self similar***, no buffer can prevent all drops
  - When routers get overloaded, packets will be dropped
- Practical consequences
  - Router queues build up, delay increases
  - Wasted bandwidth from retransmissions
  - Low network goodput

The Danger of Increasing Load

- Knee – point after which
  - Throughput increases very slow
  - Delay increases fast
- In an M/M/1 queue
  - Delay = 1/(1 – utilization)
- Cliff – point after which
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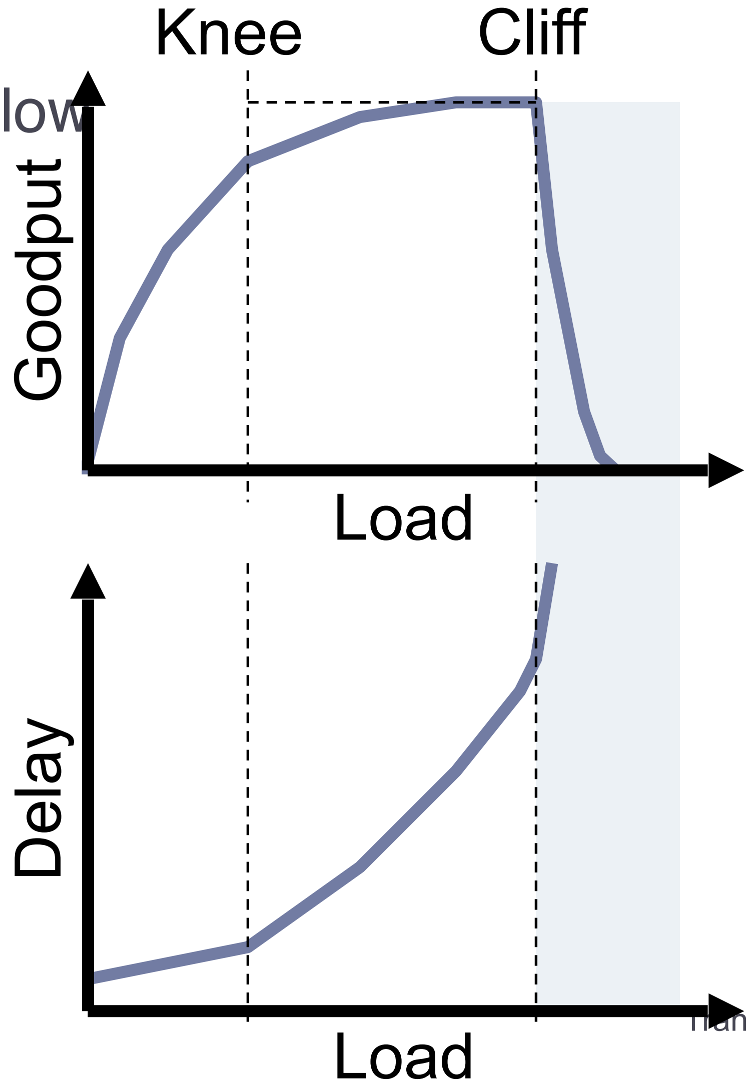

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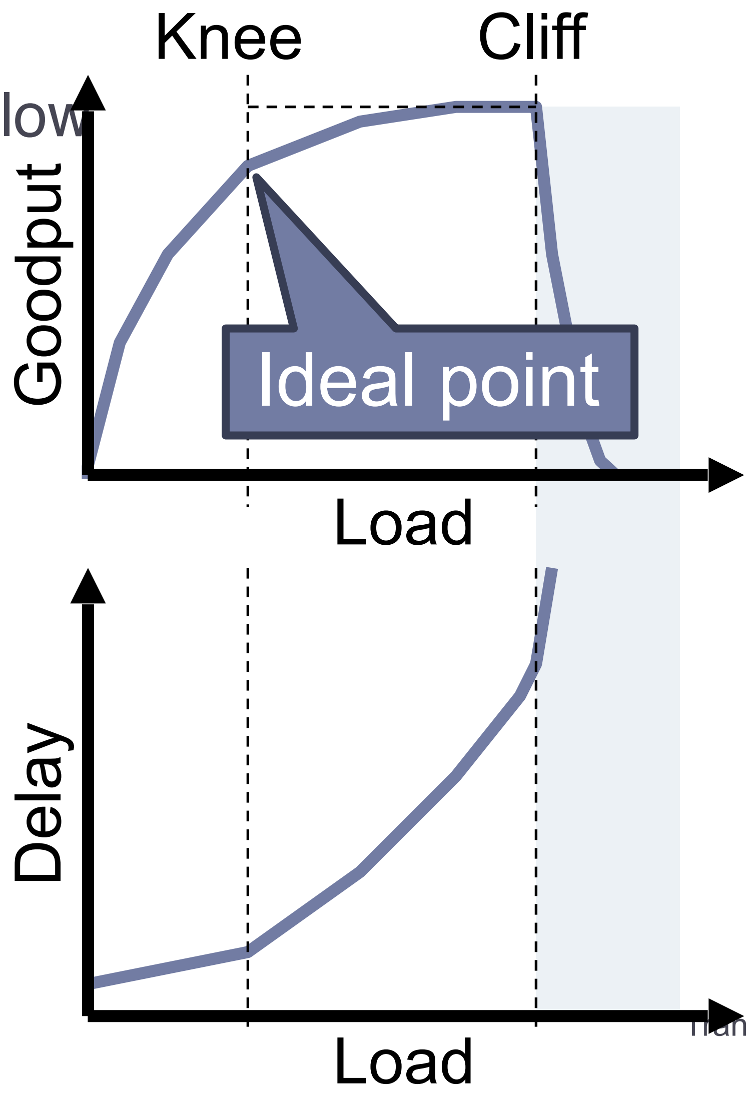

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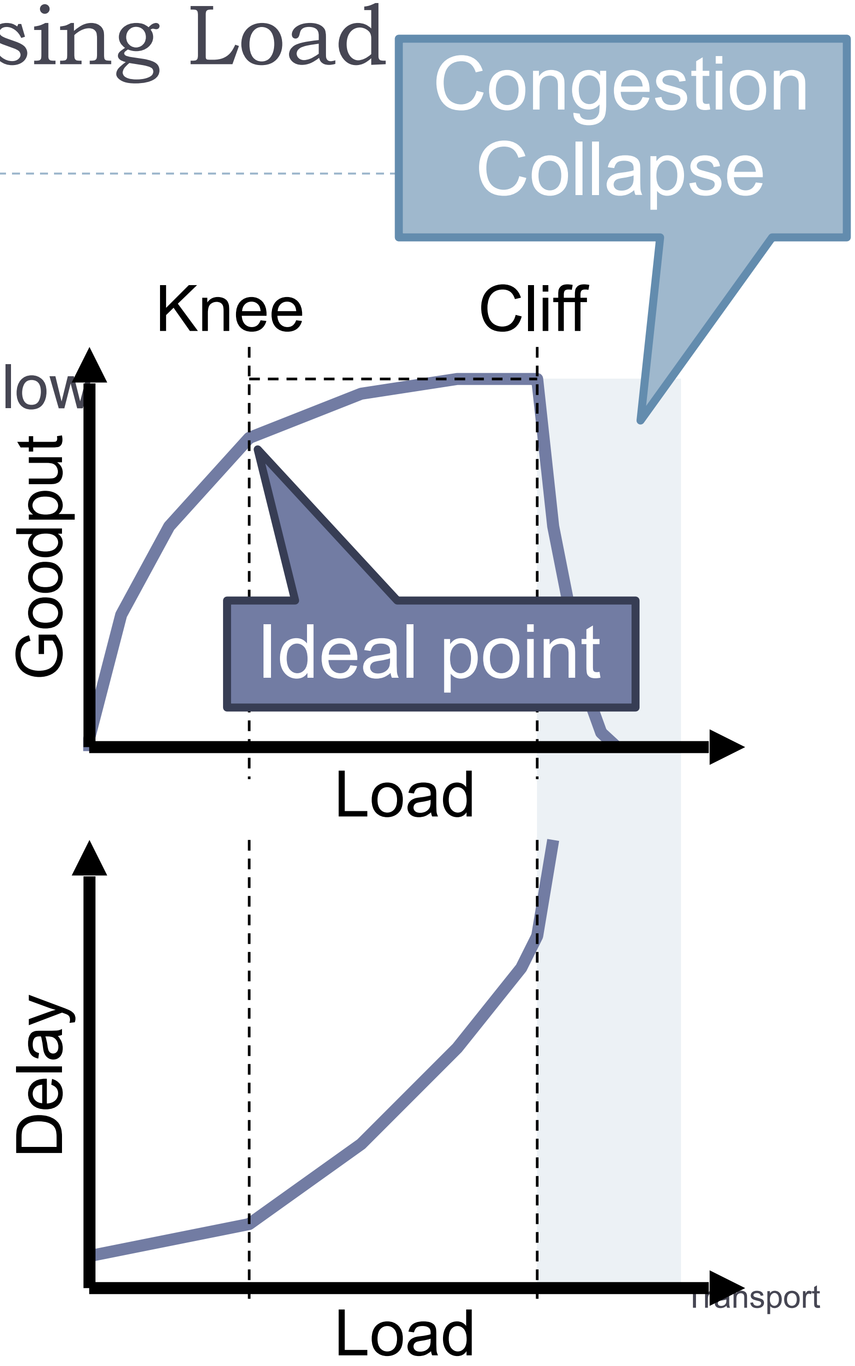
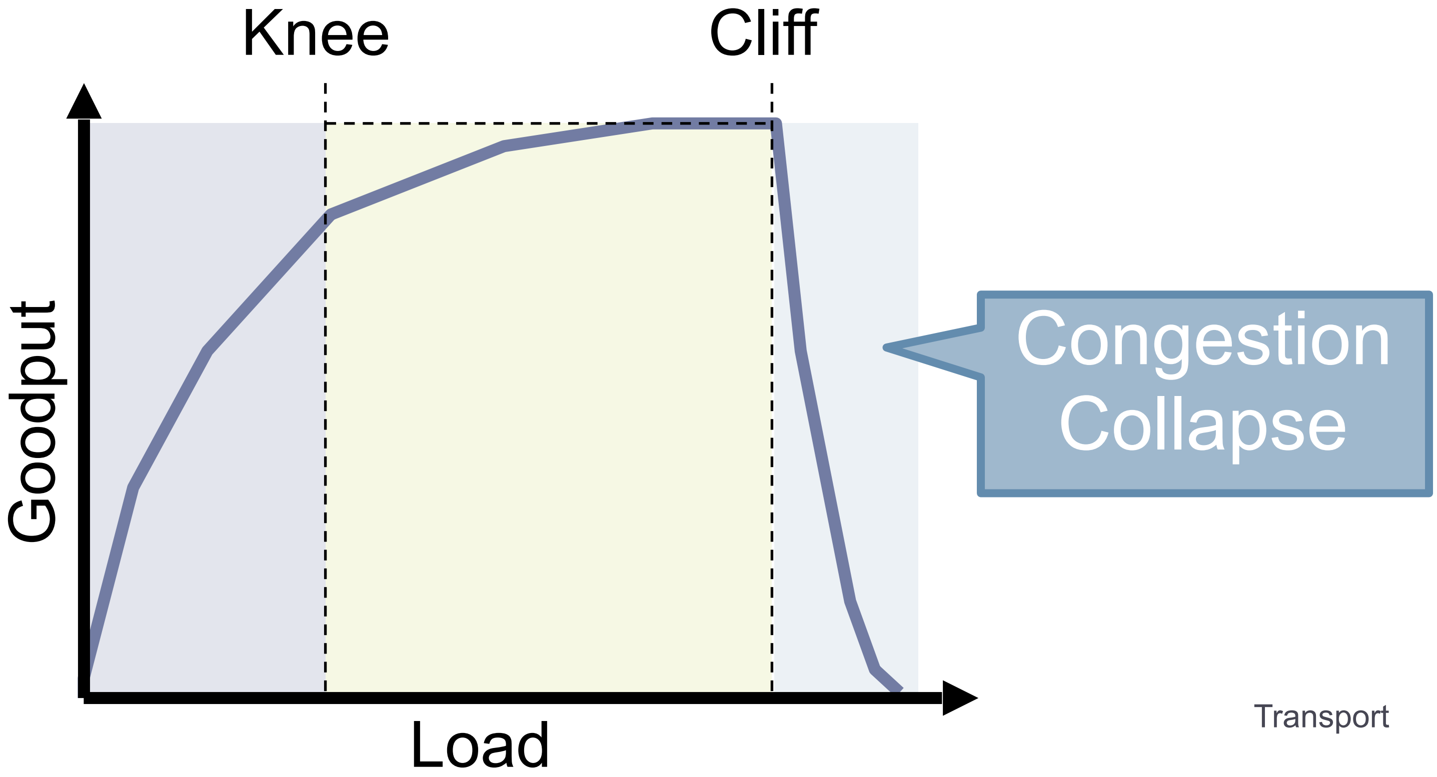


Cong. Control vs. Cong. Avoidance



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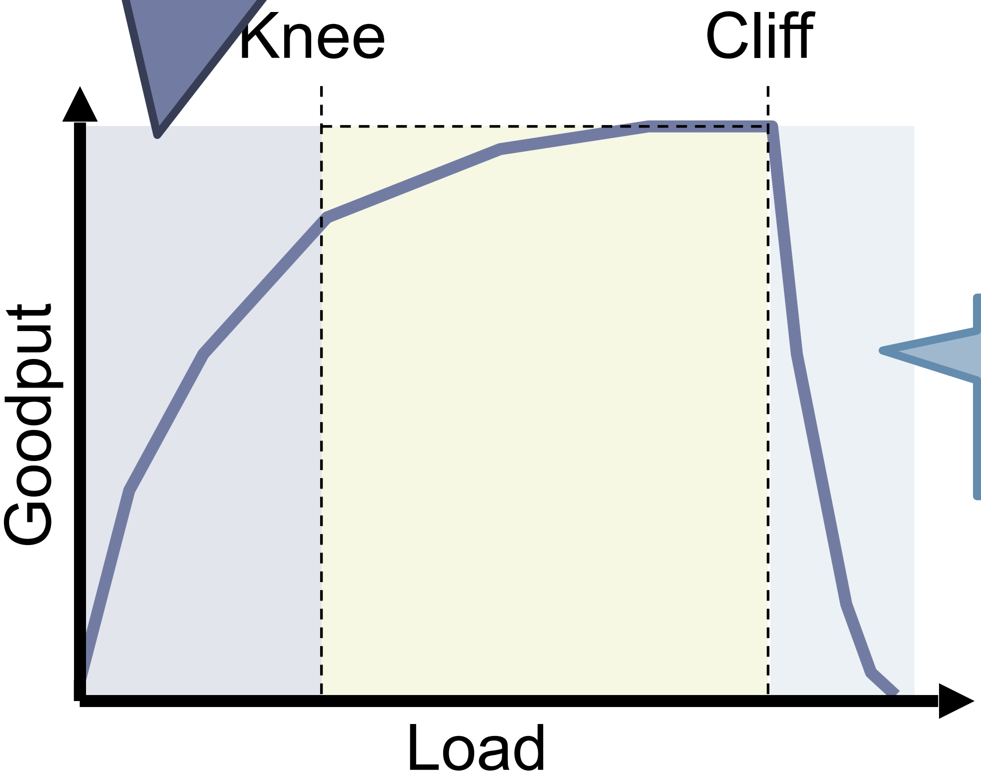

Congestion Avoidance:
Stay left of the knee

Congestion
Collapse

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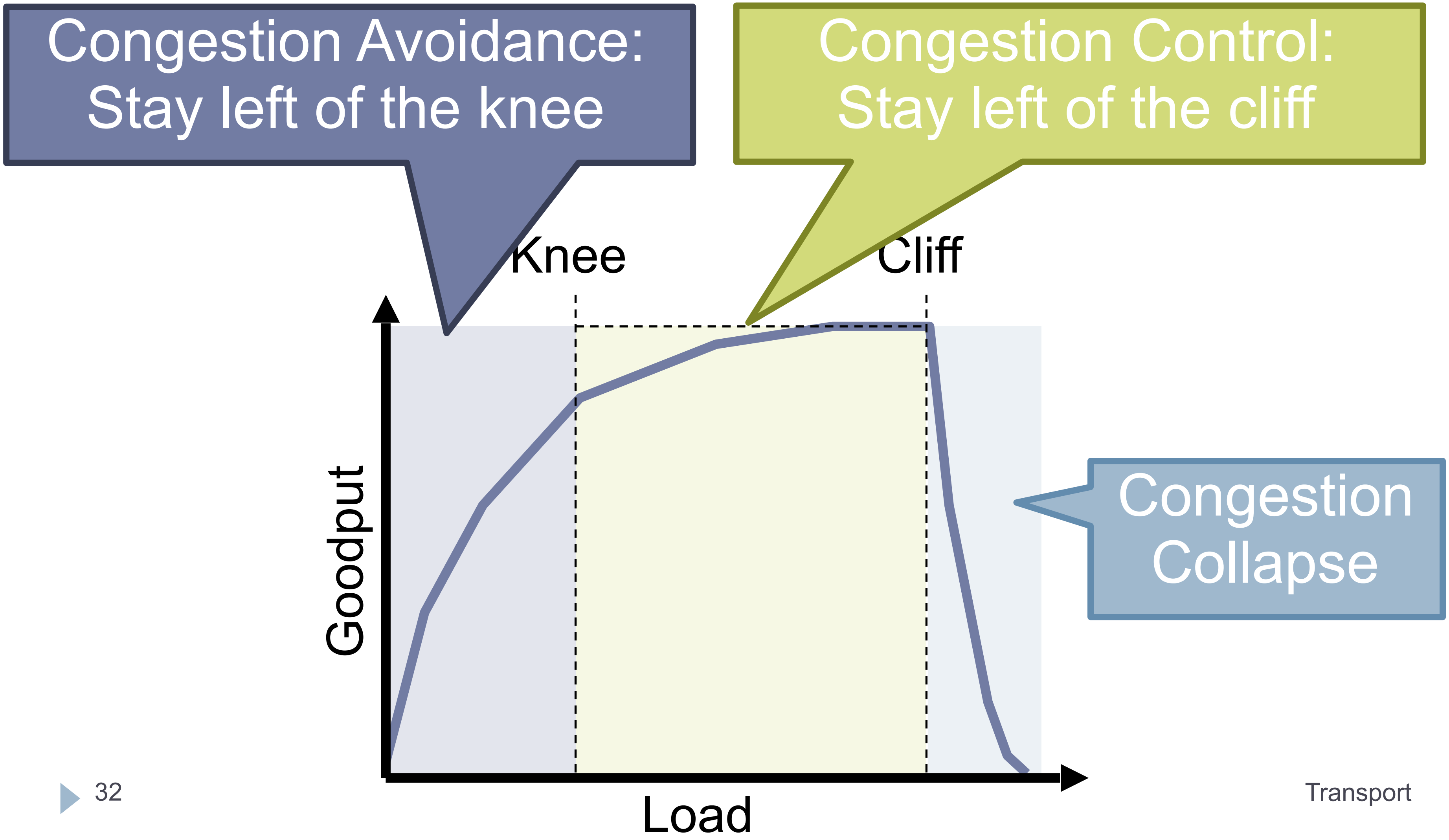

Congestion Avoidance: Stay left of the knee

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Congestion Collapse

Advertised Window, Revisited

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- The advertised window only protects the receiver
- A sufficiently fast receiver can max the window
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- Key points
  - Window size determines send rate
  - Window must be adjusted to prevent congestion collapse

Goals of Congestion Control

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1. Adjusting to the bottleneck bandwidth
2. Adjusting to variations in bandwidth
3. Sharing bandwidth between flows
4. Maximizing throughput

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- Do nothing, send packets indiscriminately
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- Introduce a congestion window at the sender
  - Congestion control is sender-side problem

Congestion Window (*cwnd*)

- Limits how much data is in transit
- Denominated in bytes

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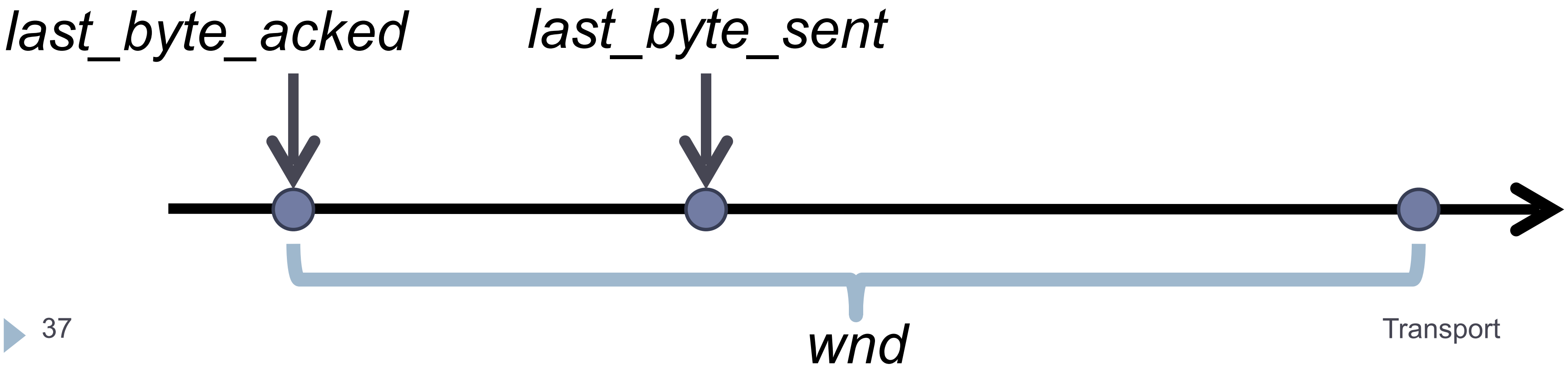

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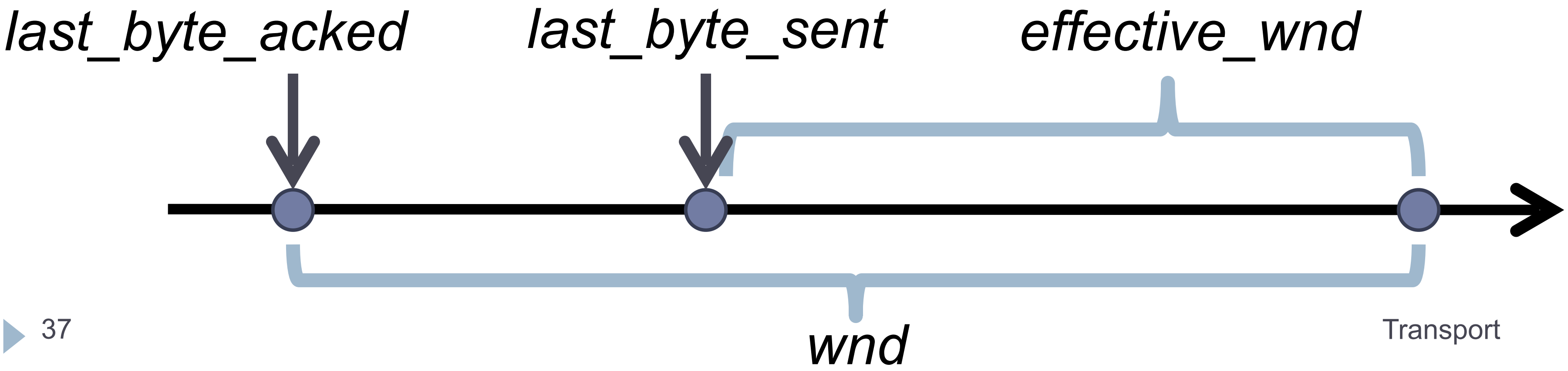

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2. Rate adjustment algorithm
   - Modify *cwnd*
   - Probe for bandwidth
   - Responding to congestion

Except on wireless networks

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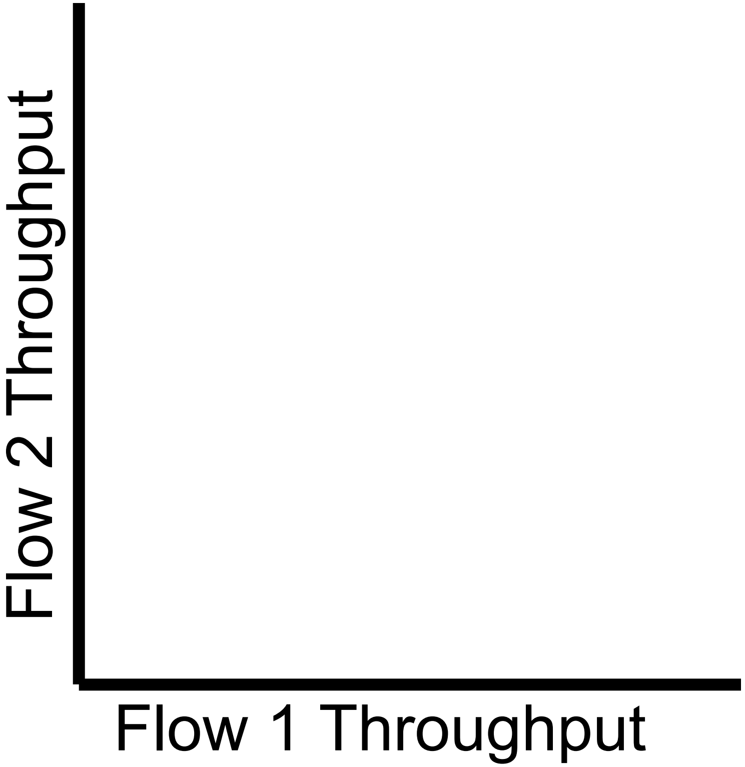
- Question: increase/decrease functions to use?

Utilization and Fairness



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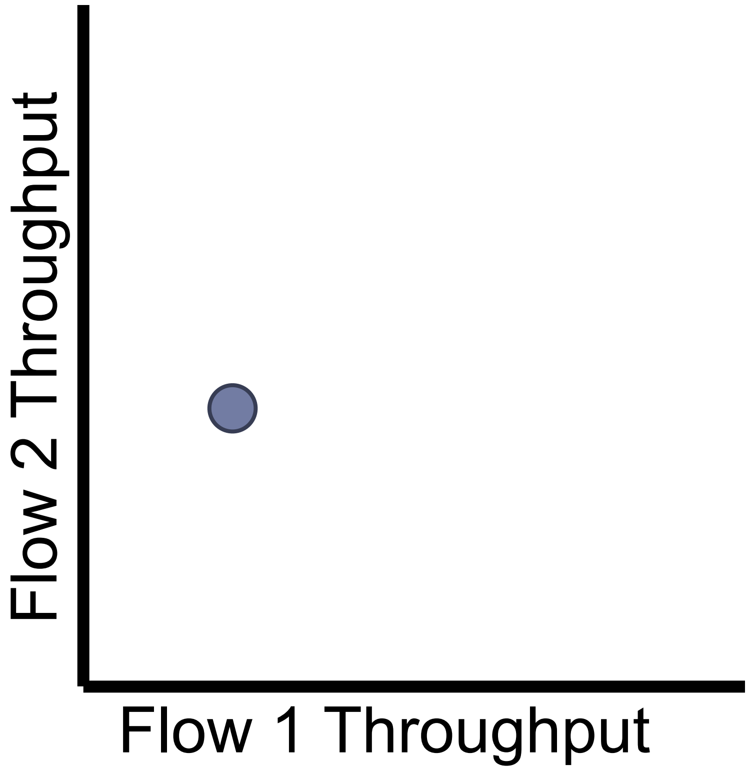

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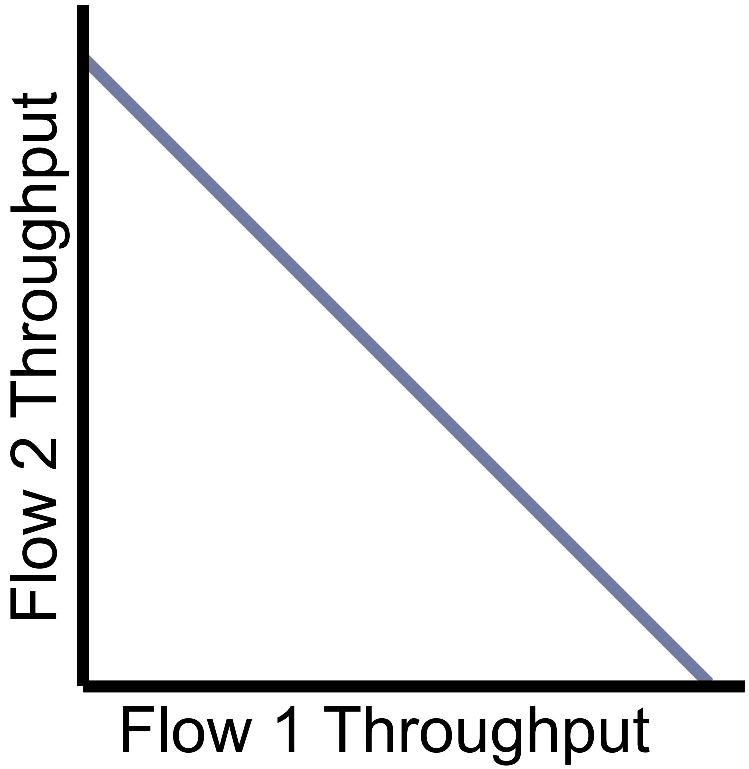

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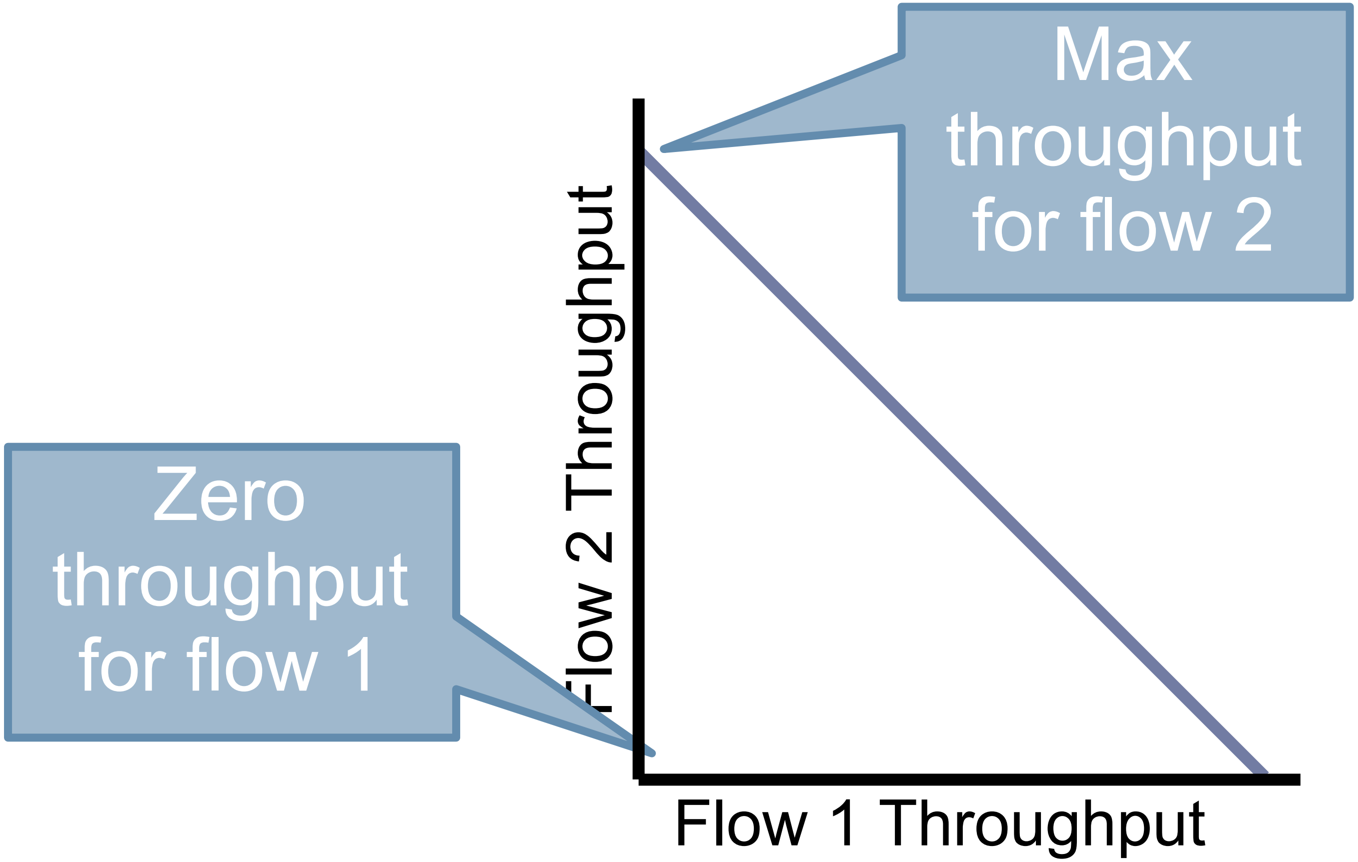

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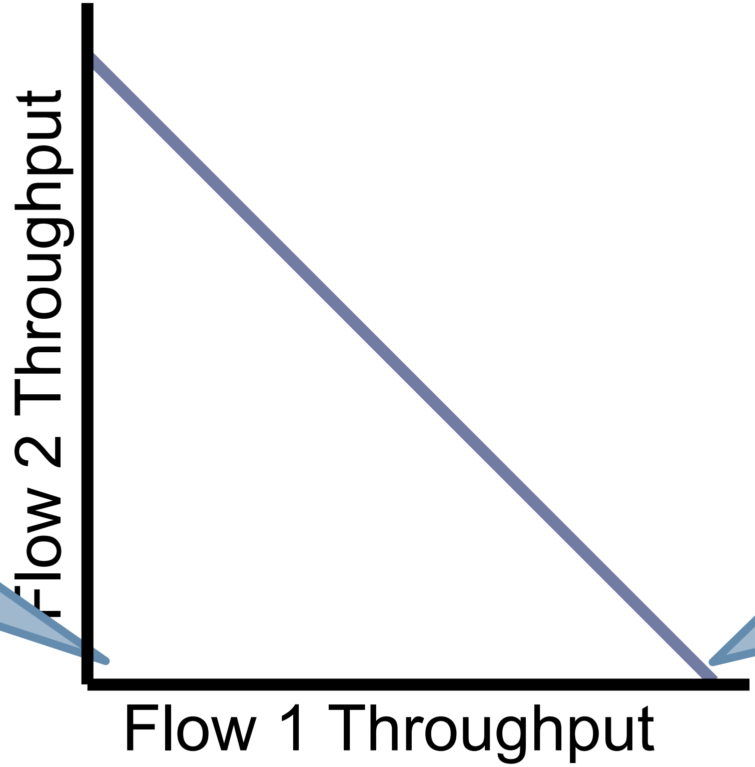
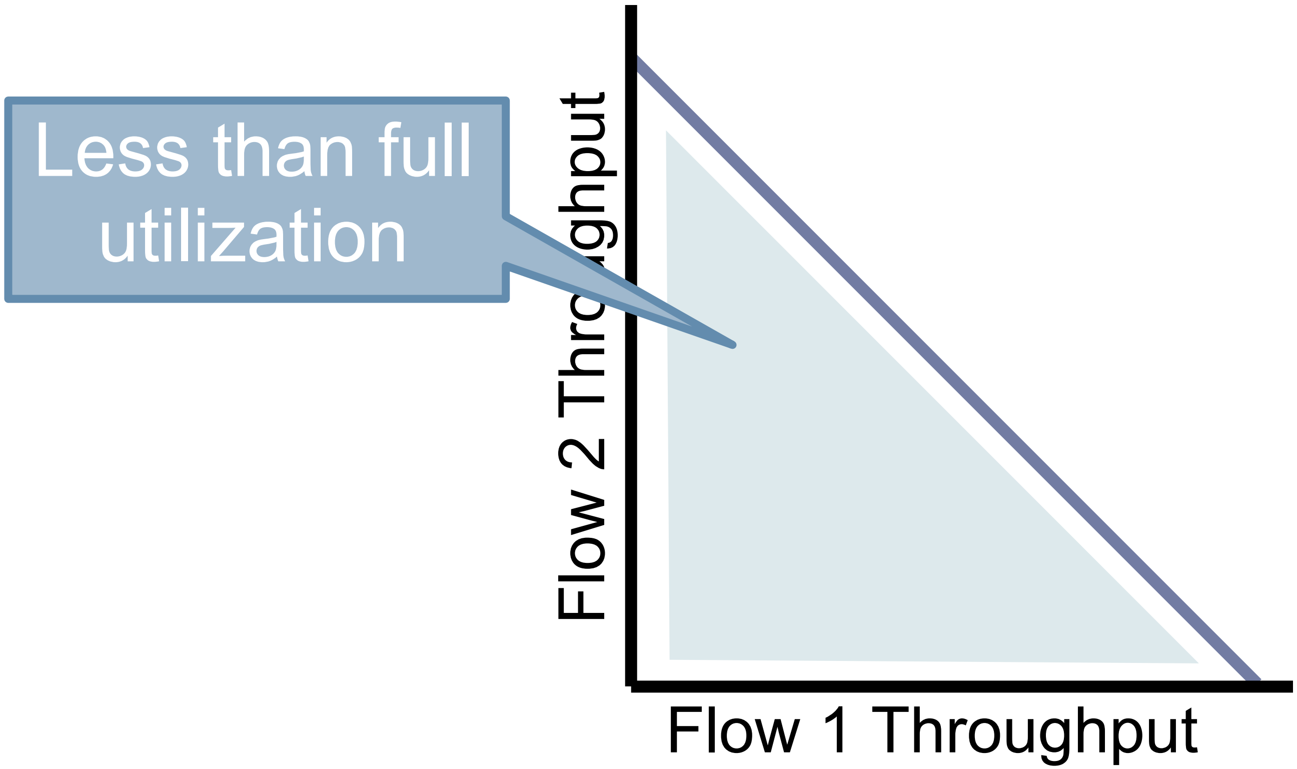


Zero throughput for flow 2

Max throughput for flow 1

Utilization and Fairness



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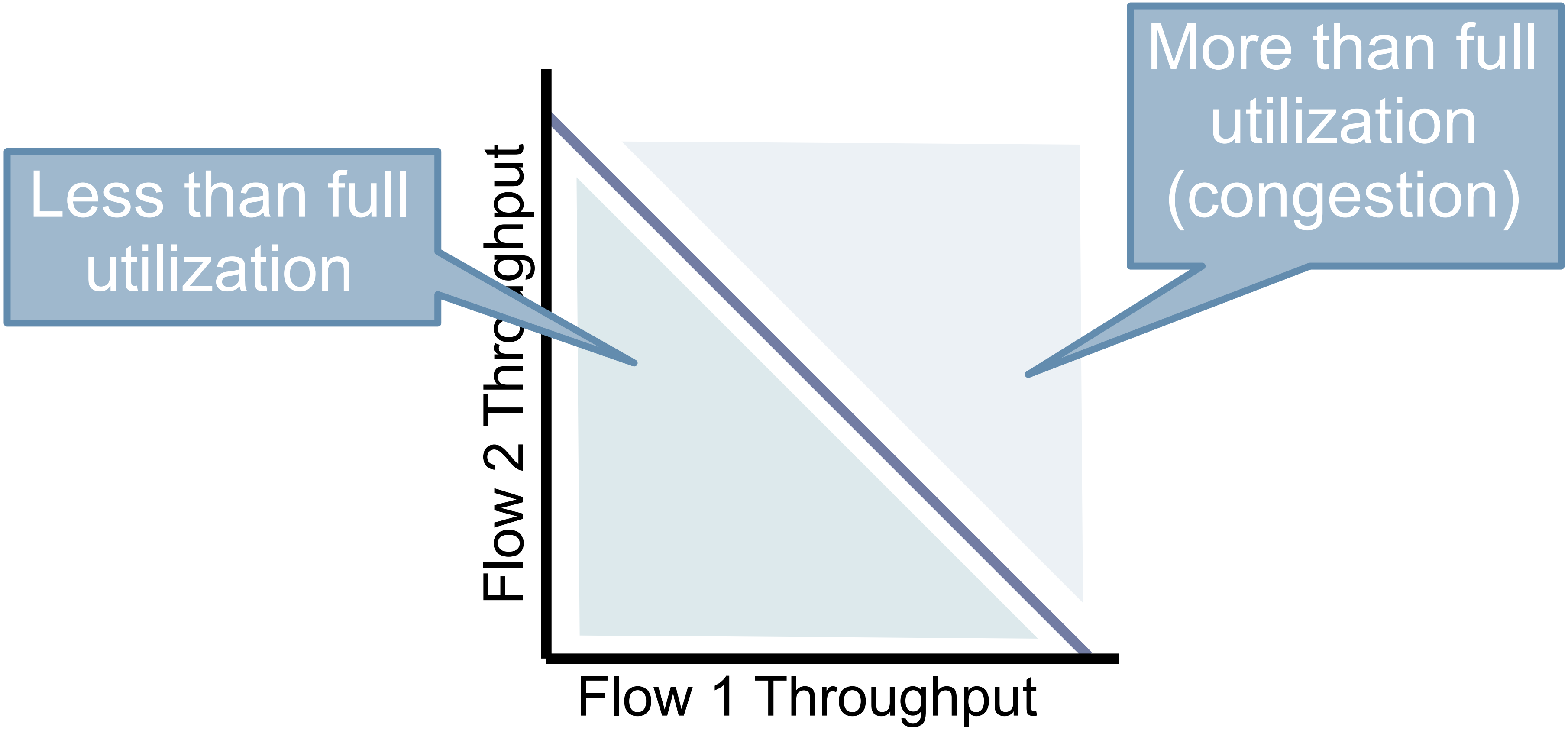
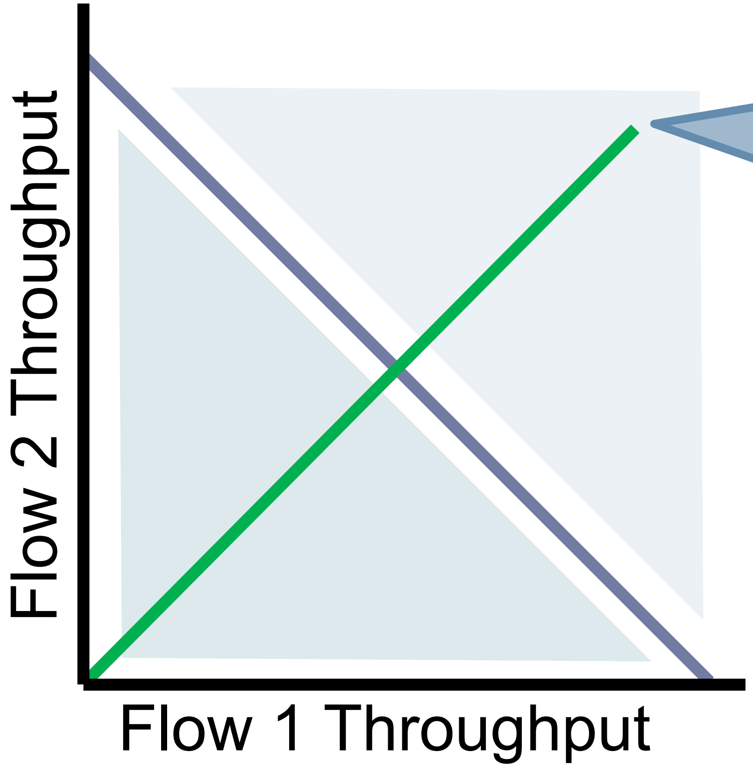


Less than full utilization

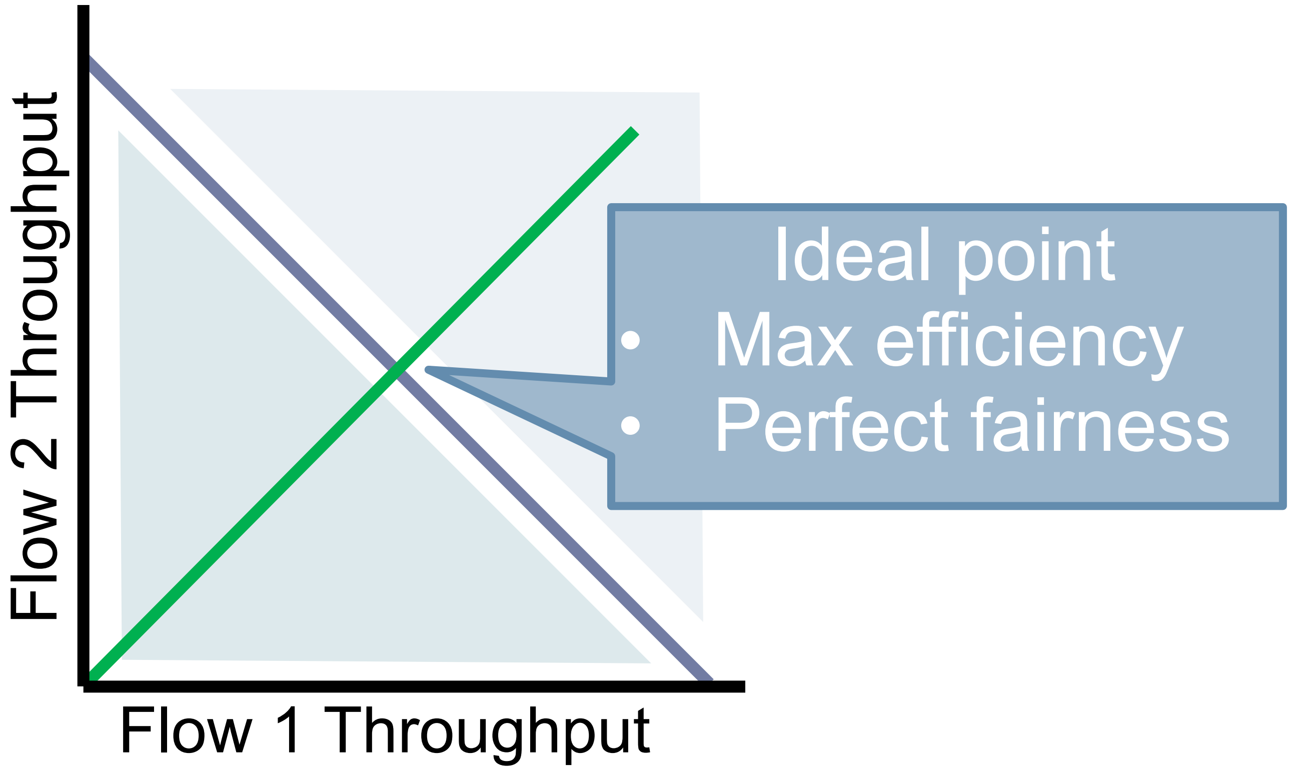
More than full utilization (congestion)

Utilization and Fairness



Equal throughput (fairness)

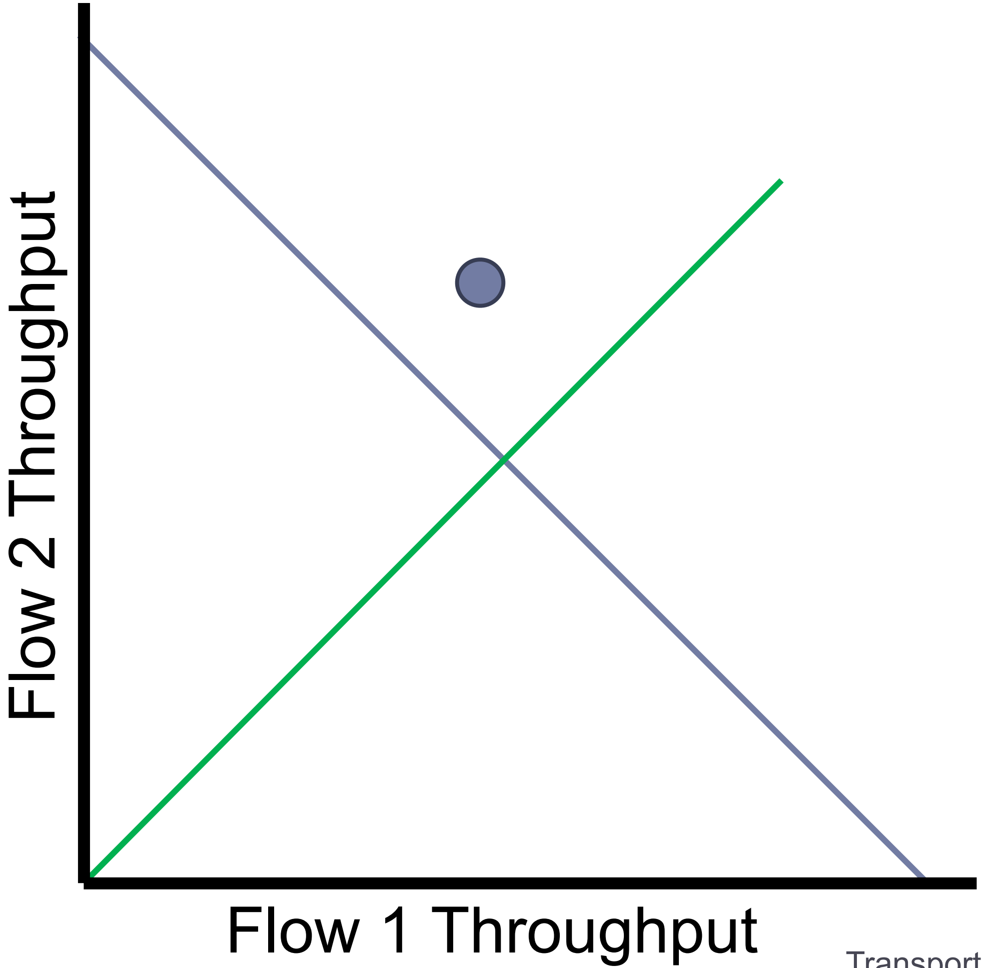
Utilization and Fairness



Ideal point

- Max efficiency
- Perfect fairness

Multiplicative Increase, Additive Decrease



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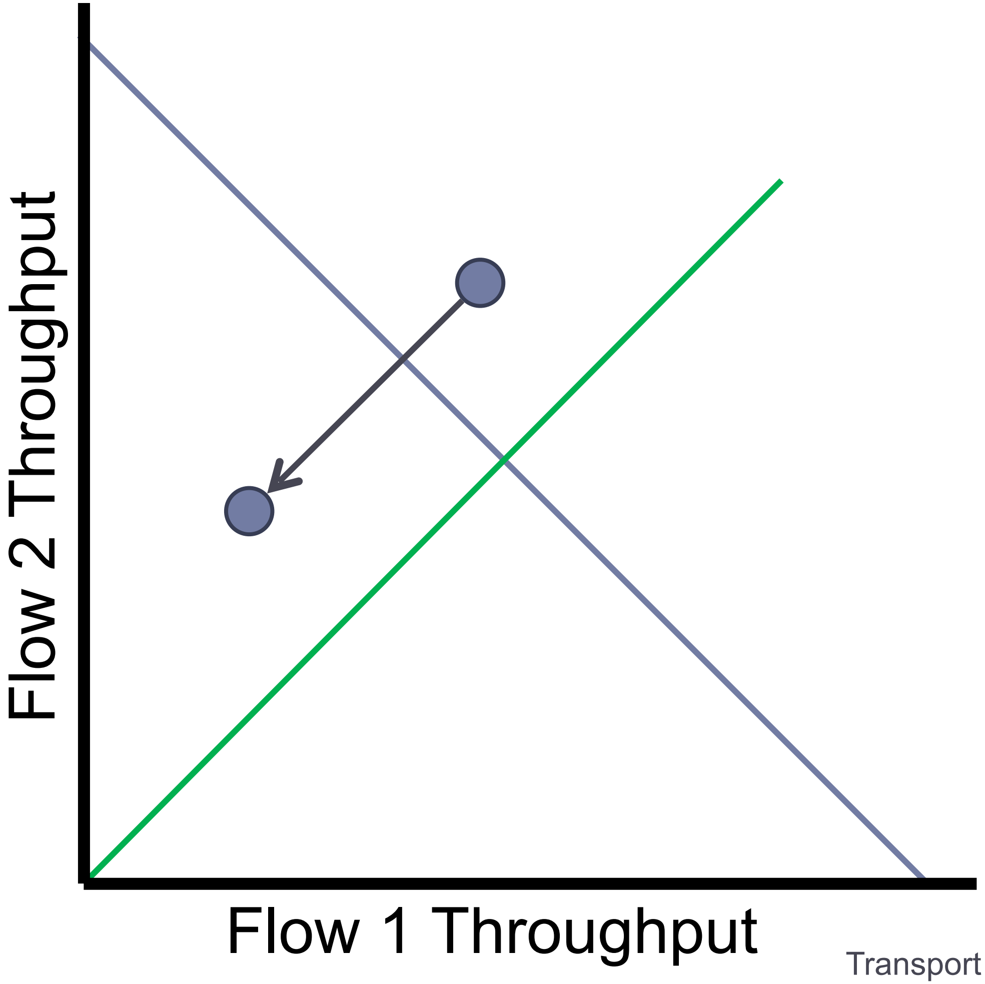

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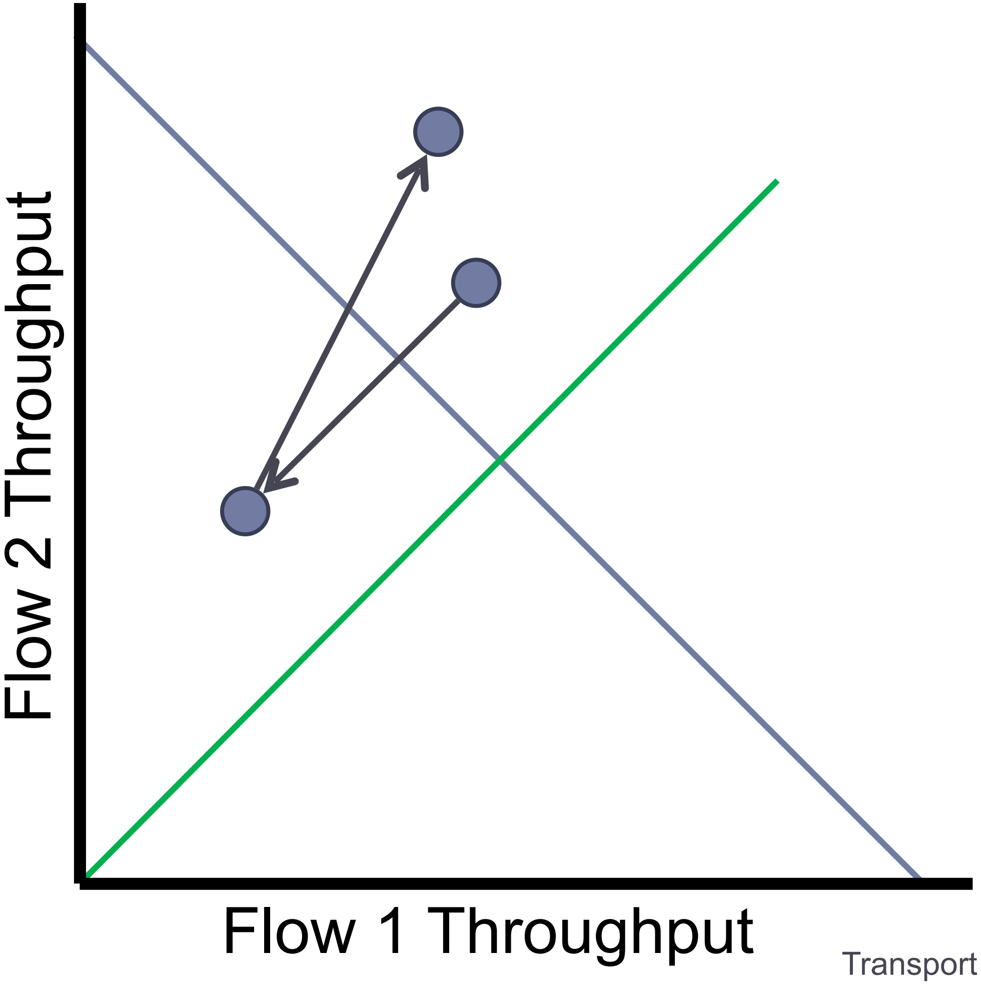

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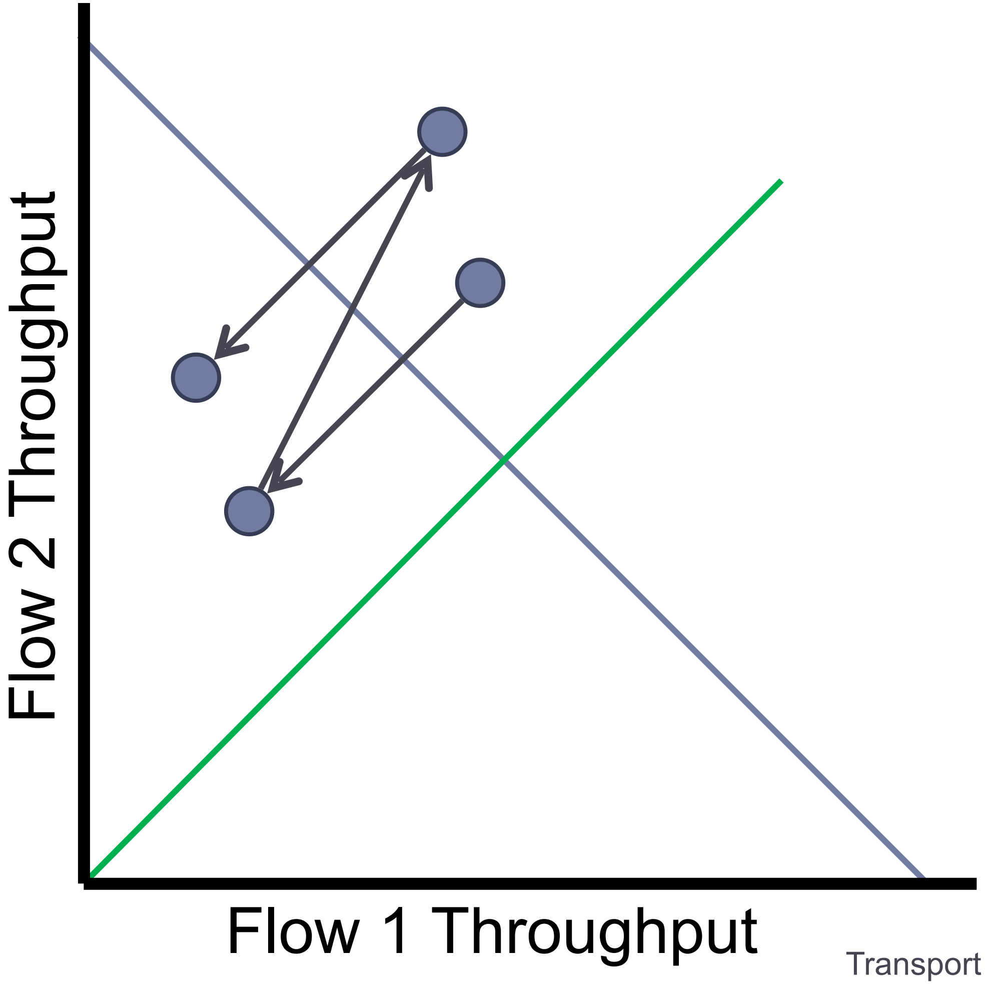

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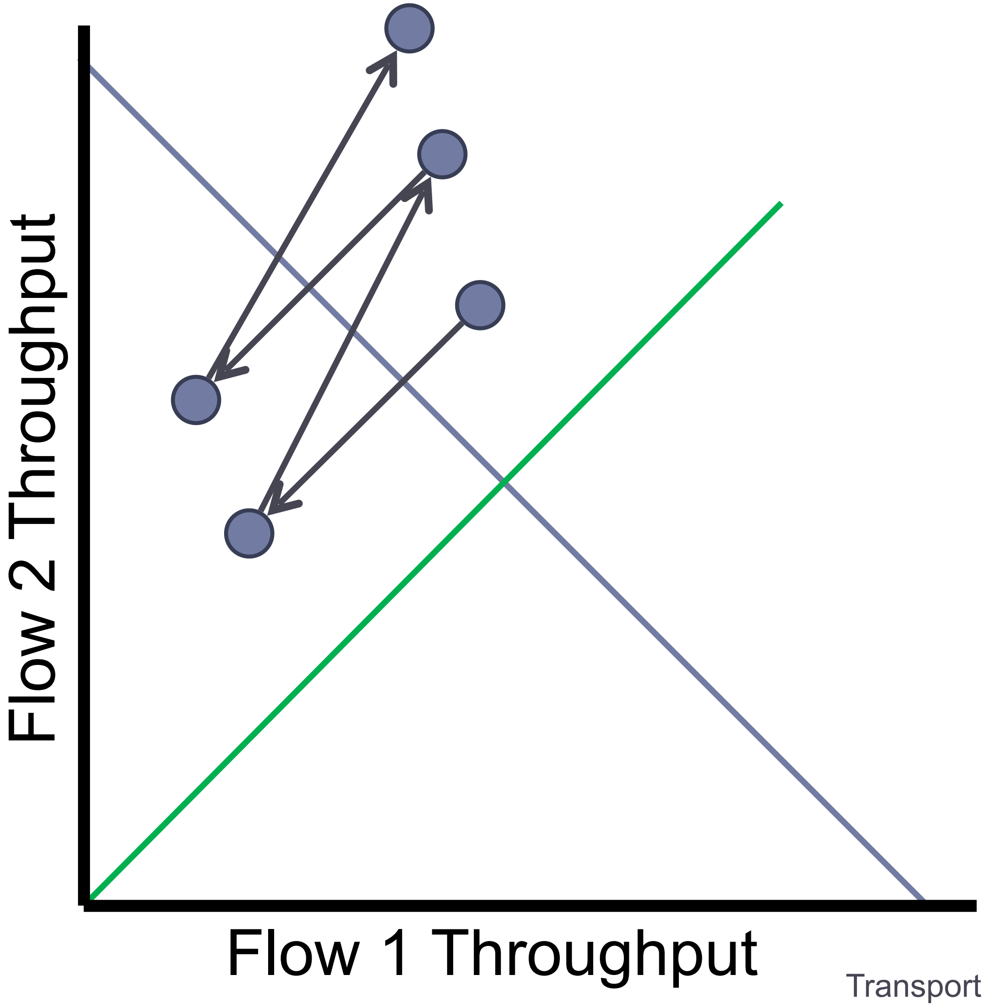
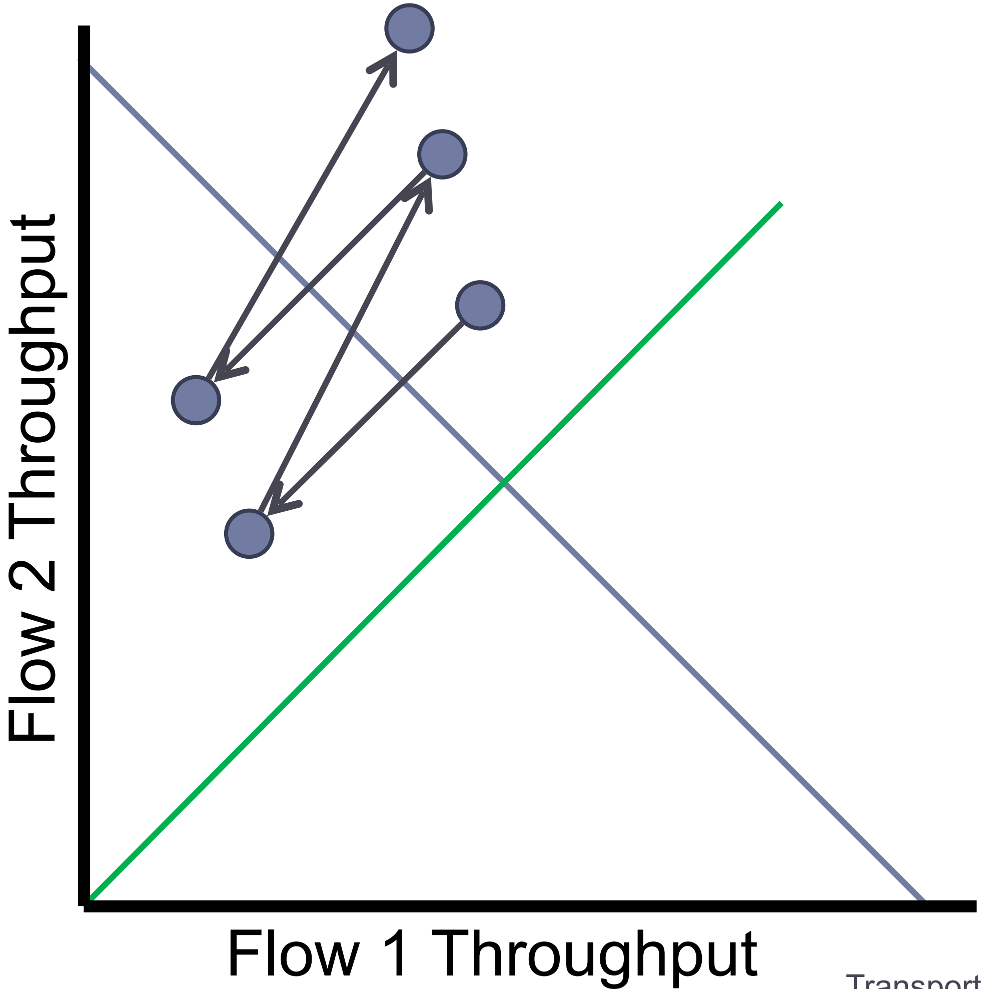


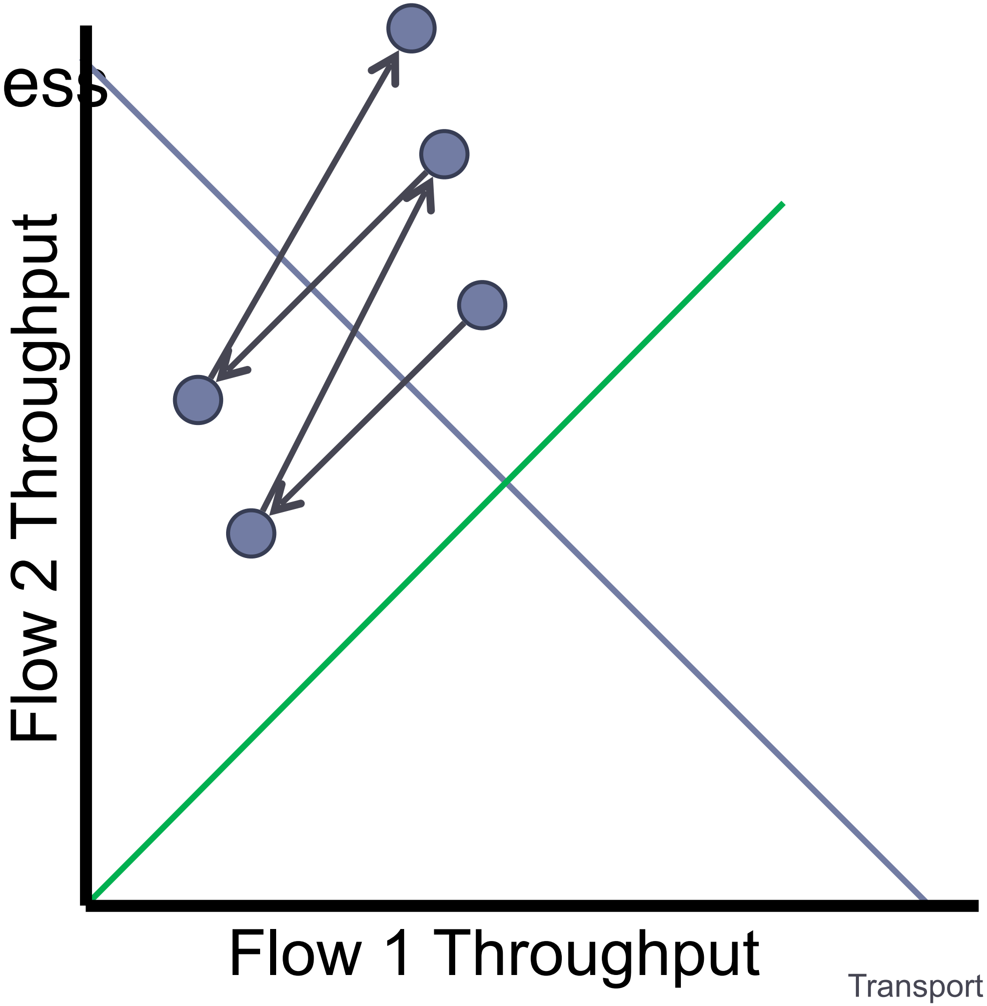
Multiplicative Increase, Additive Decrease

- Not stable!



Multiplicative Increase, Additive Decrease

- Not stable!
- Veers away from fairness



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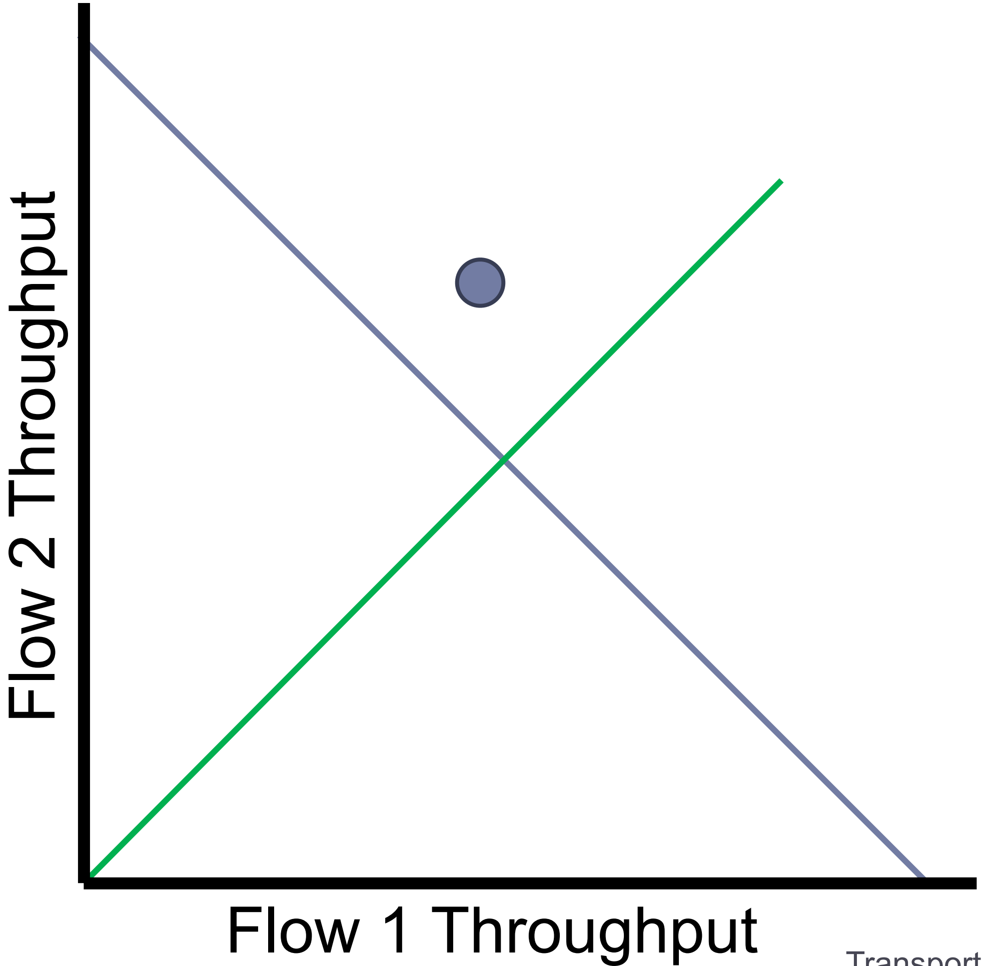

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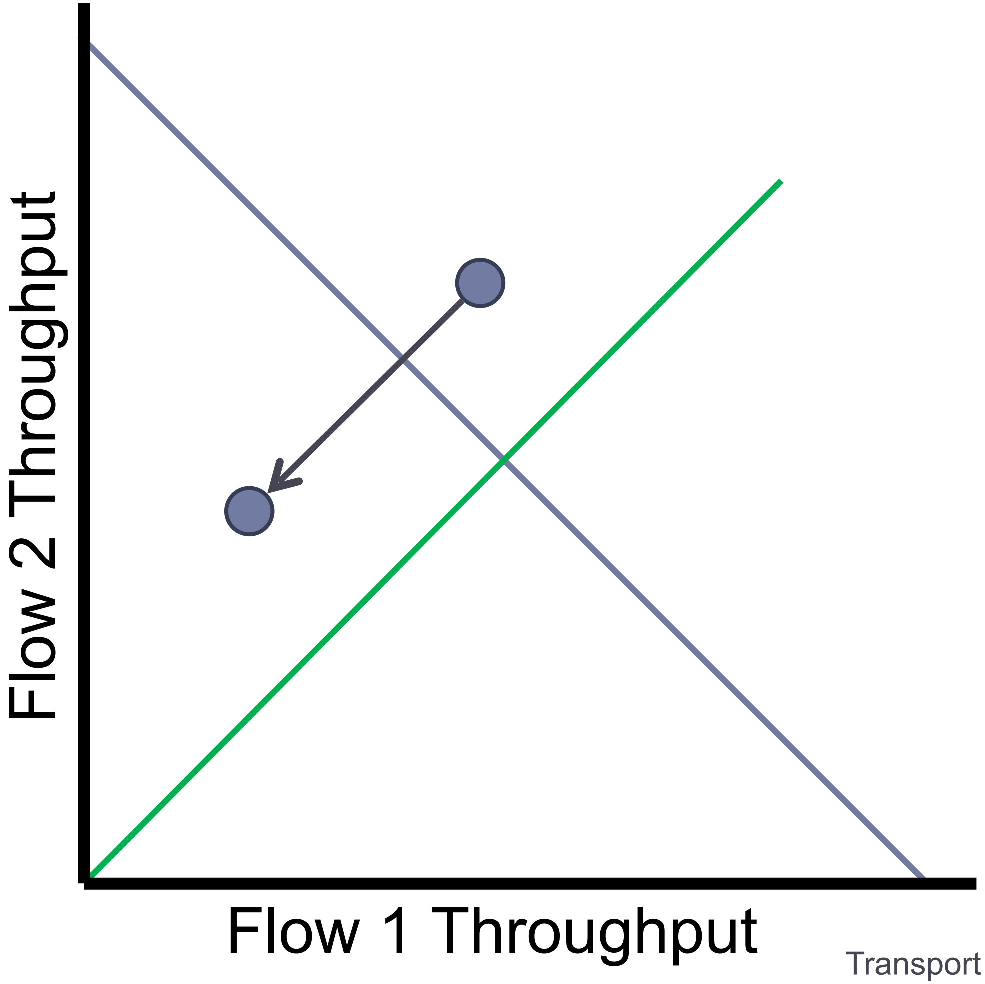

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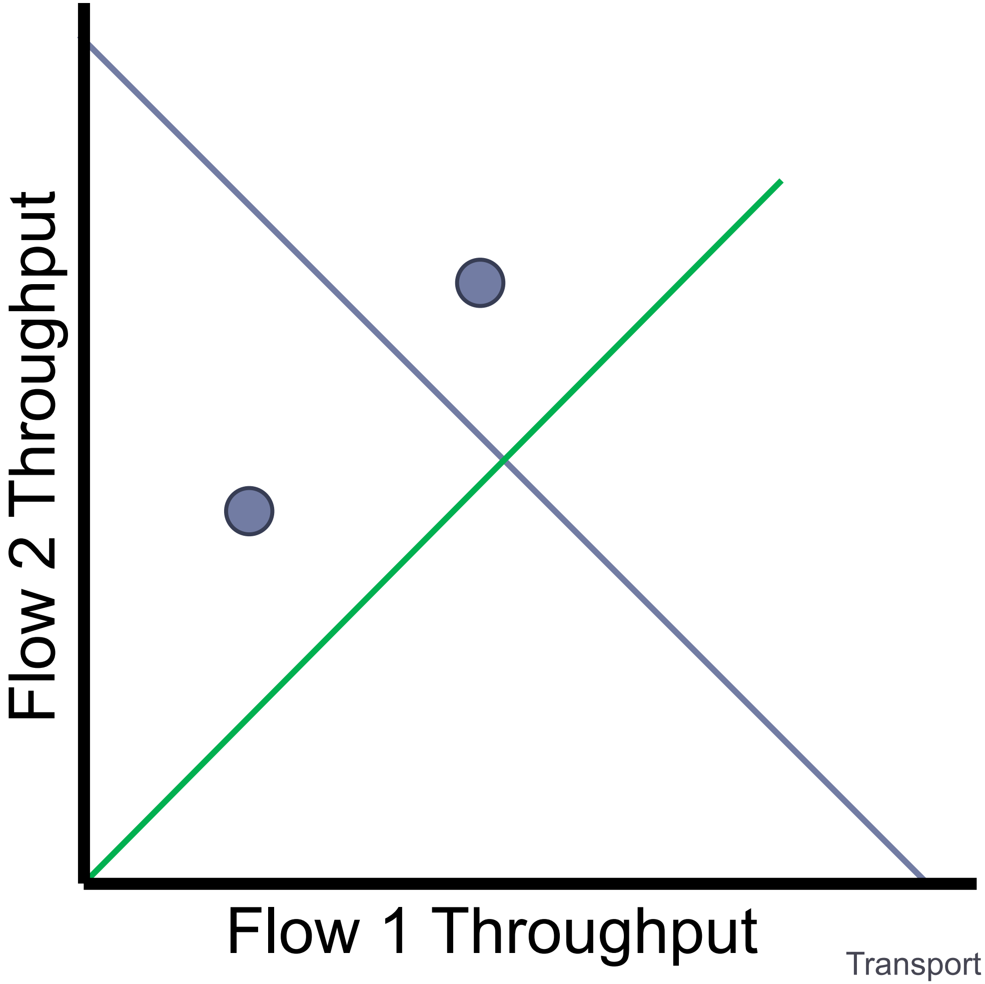

Additive Increase, Additive Decrease

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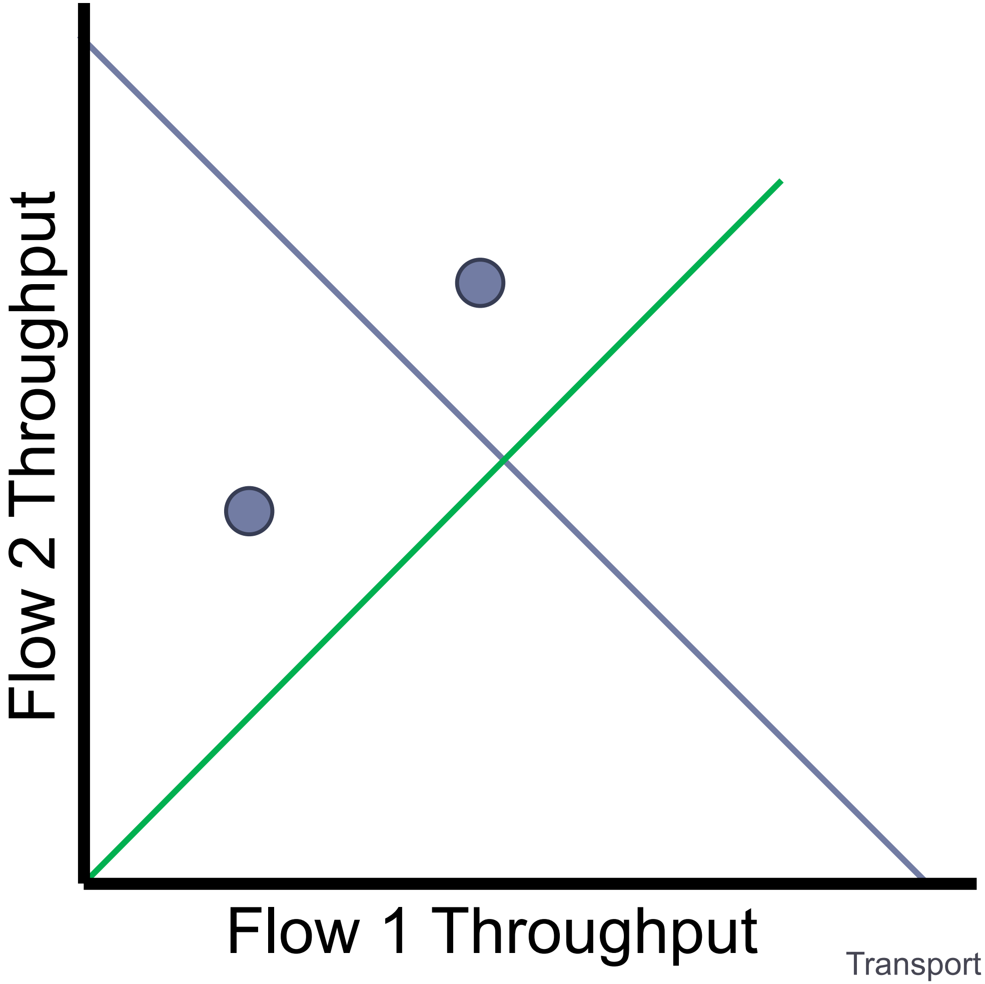
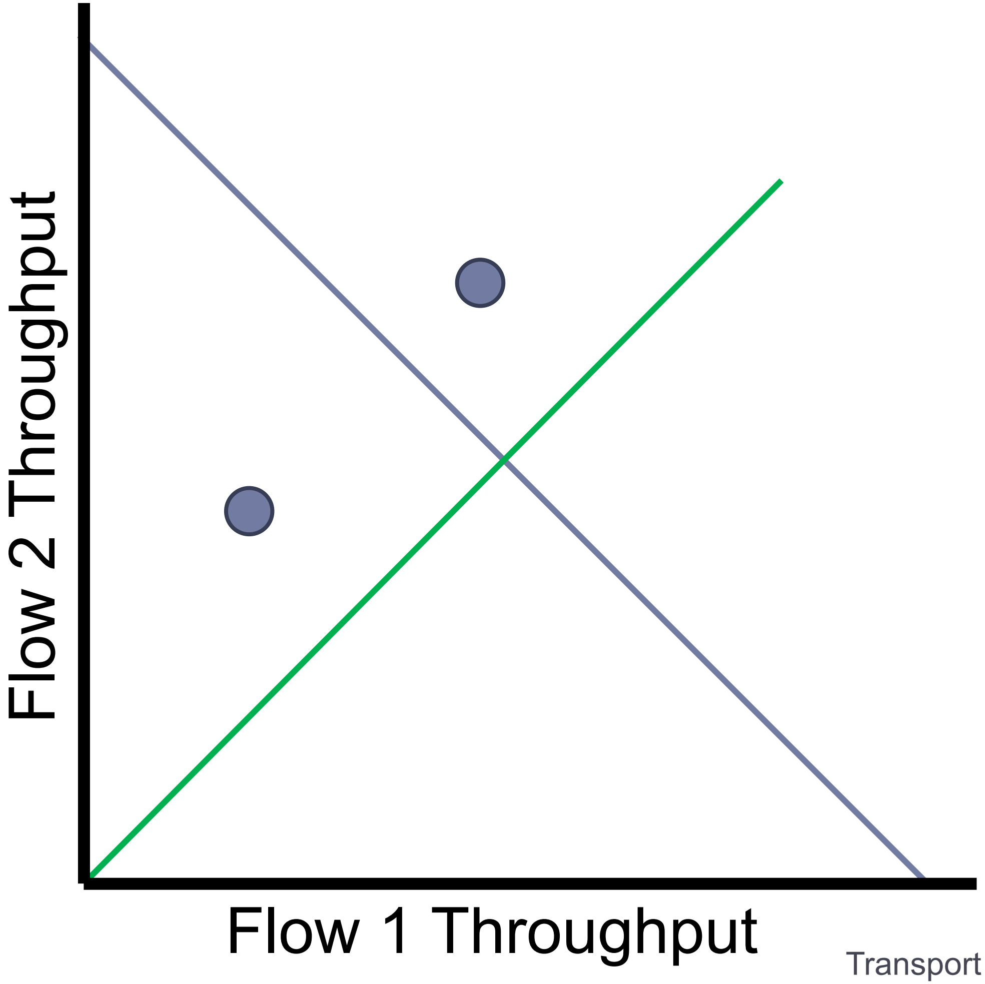


Additive Increase, Additive Decrease

- Stable
- But does not converge to fairness



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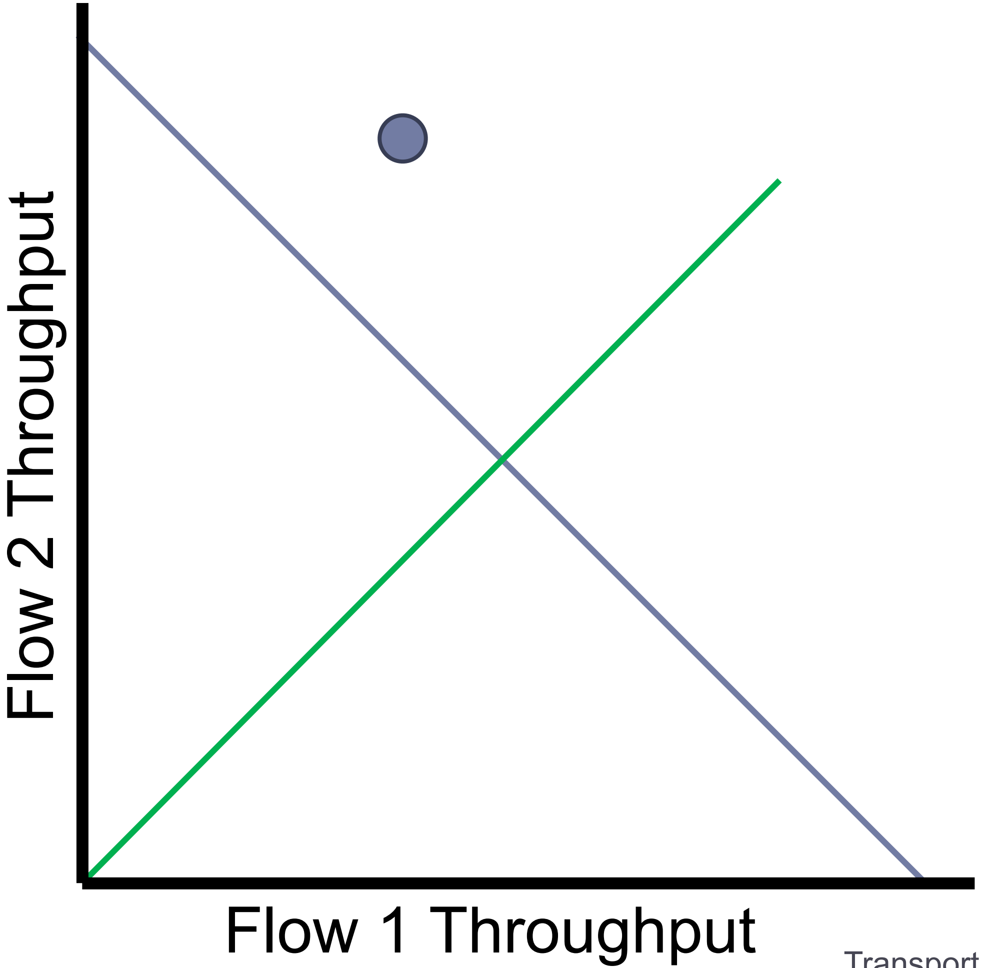

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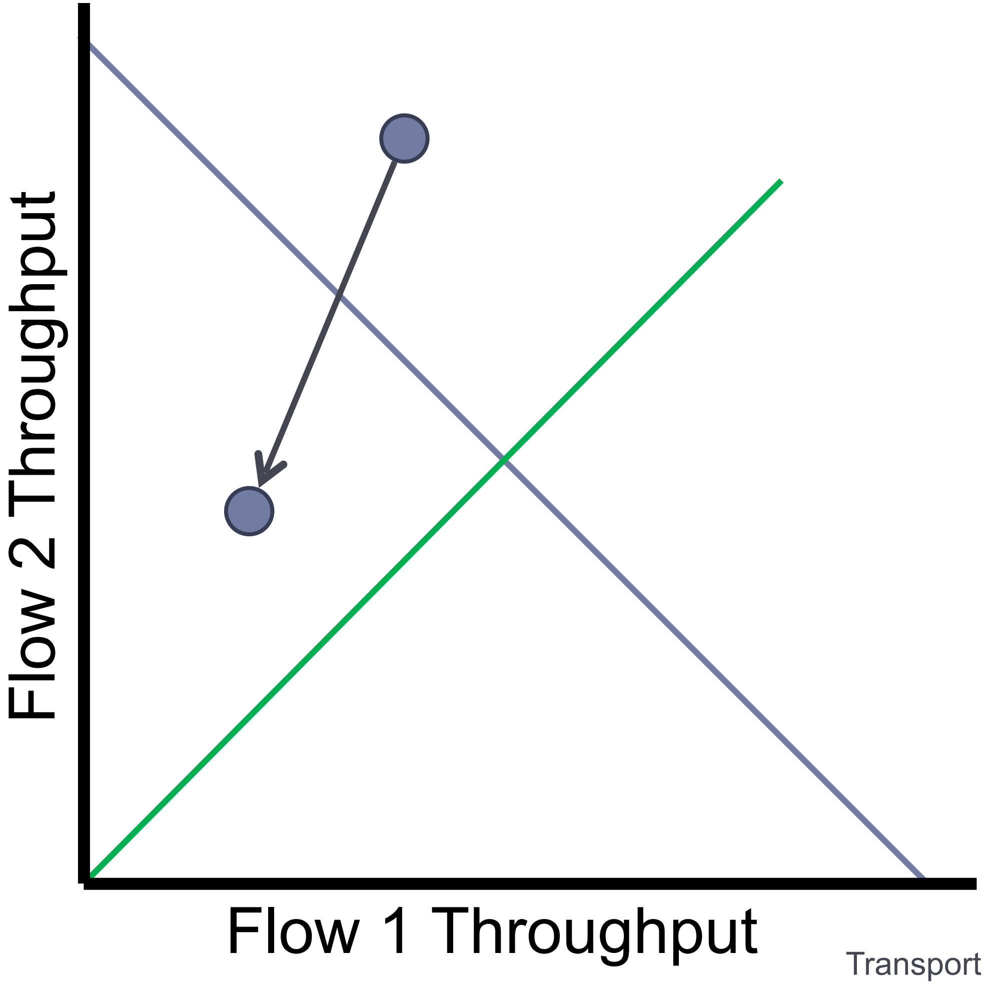

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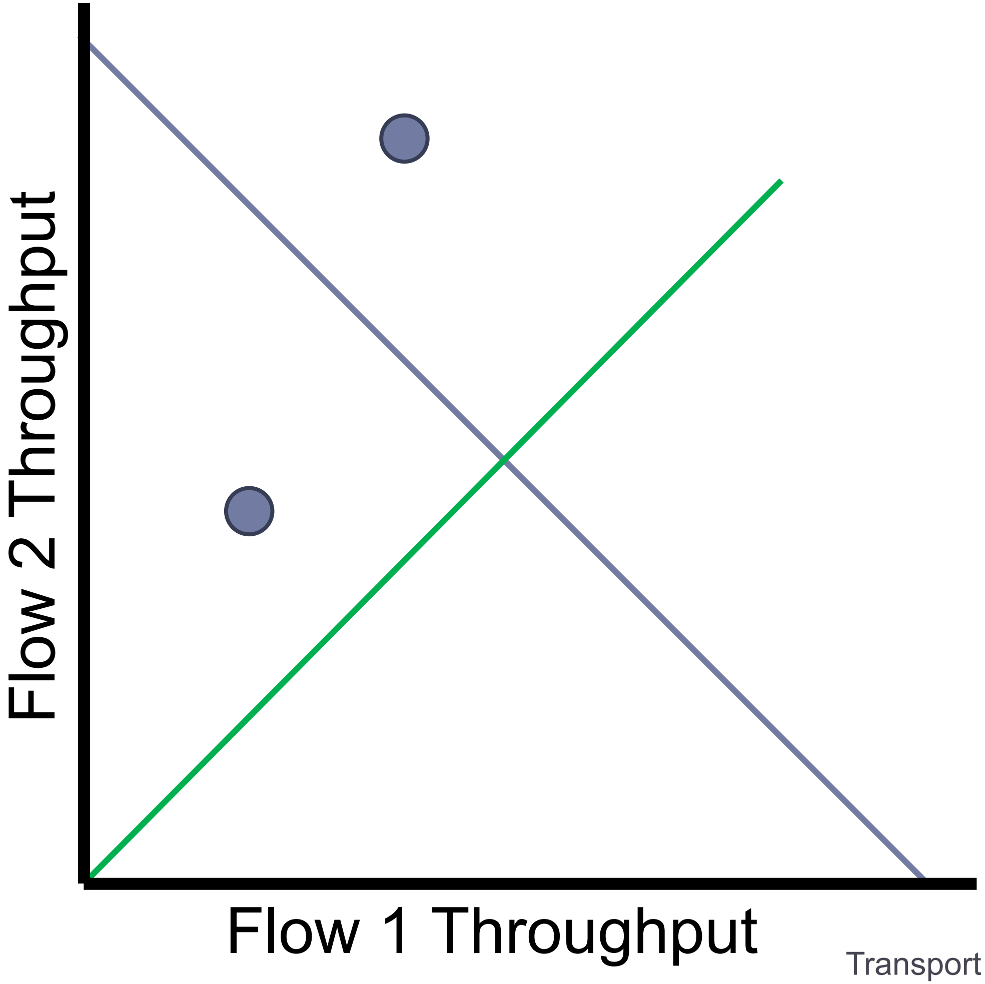
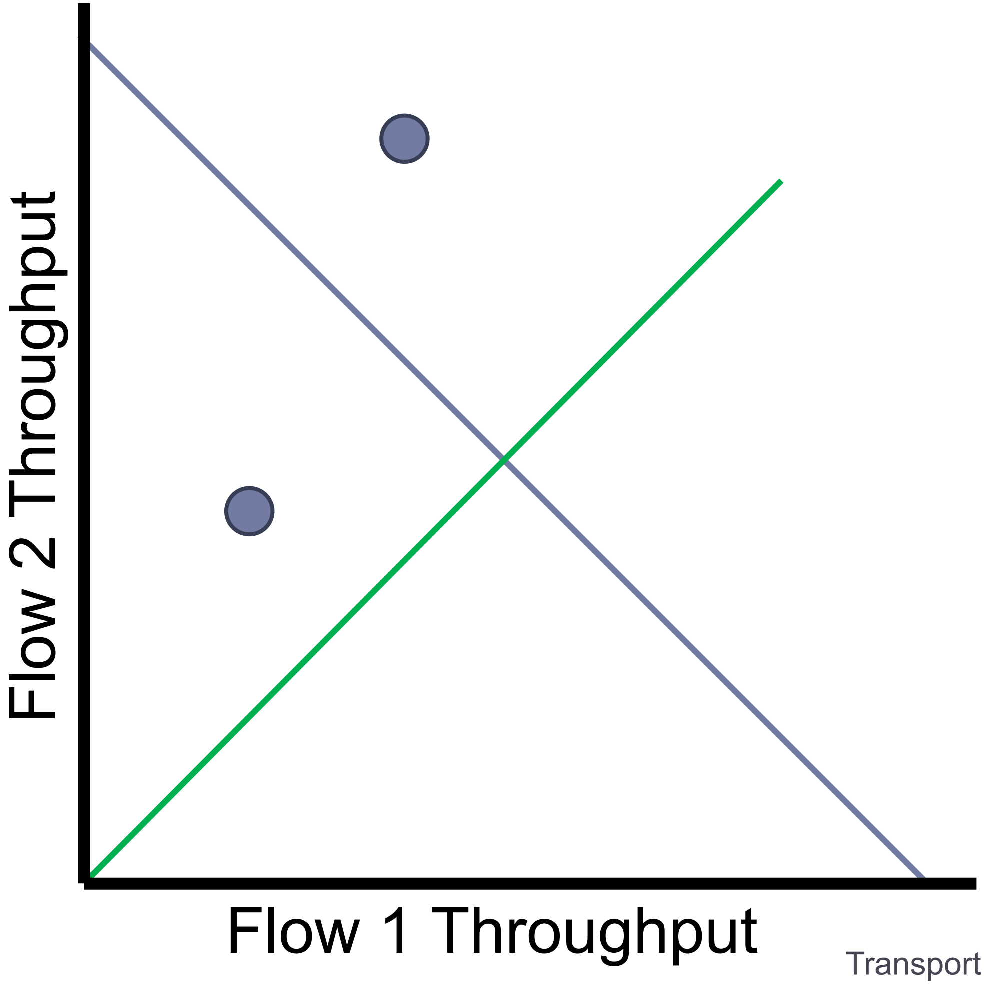


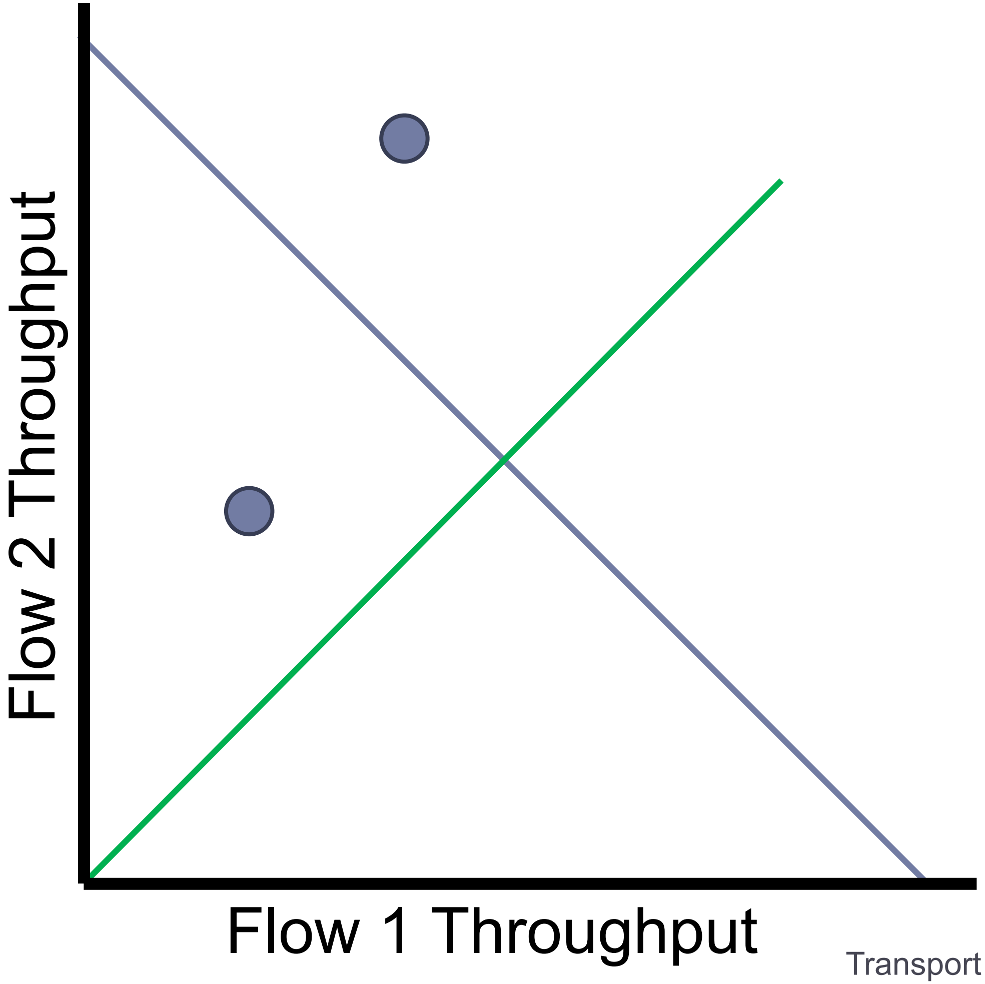
Multiplicative Increase, Multiplicative Decrease

- Stable



Multiplicative Increase, Multiplicative Decrease

- Stable
- But does not converge to fairness



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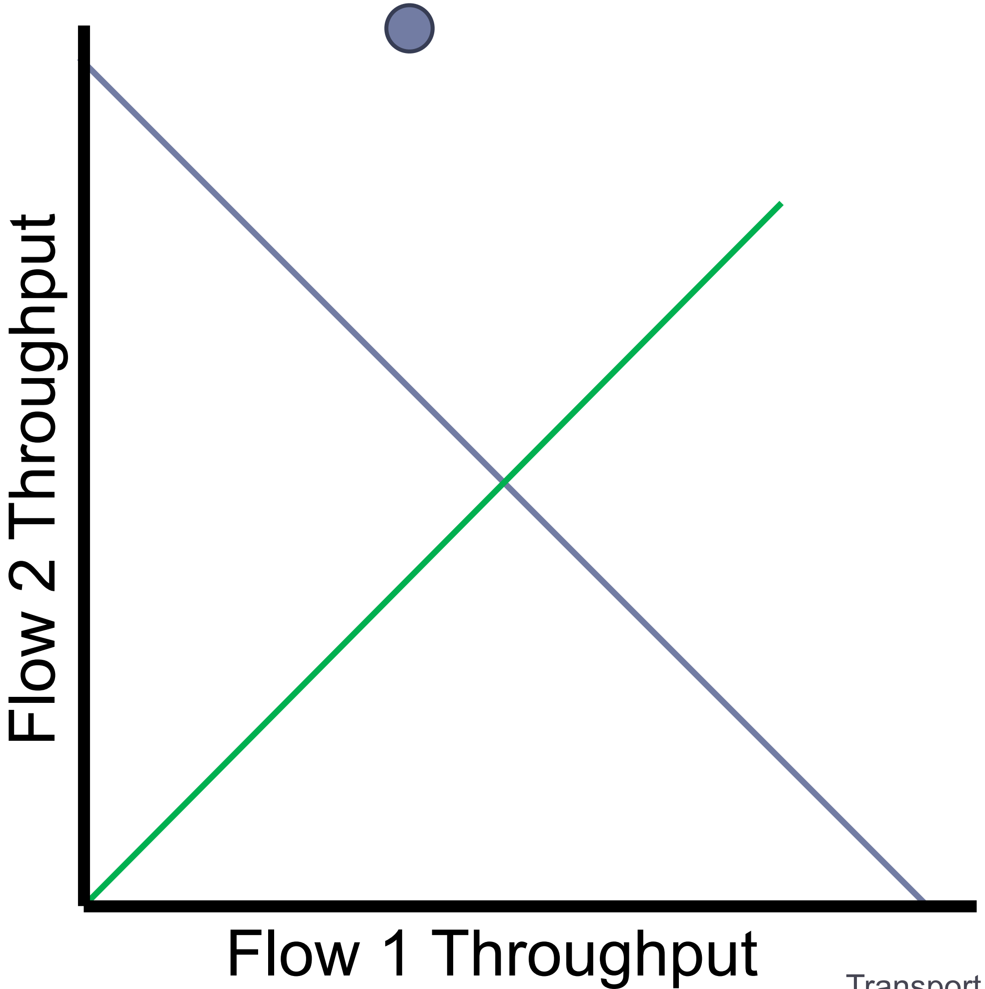

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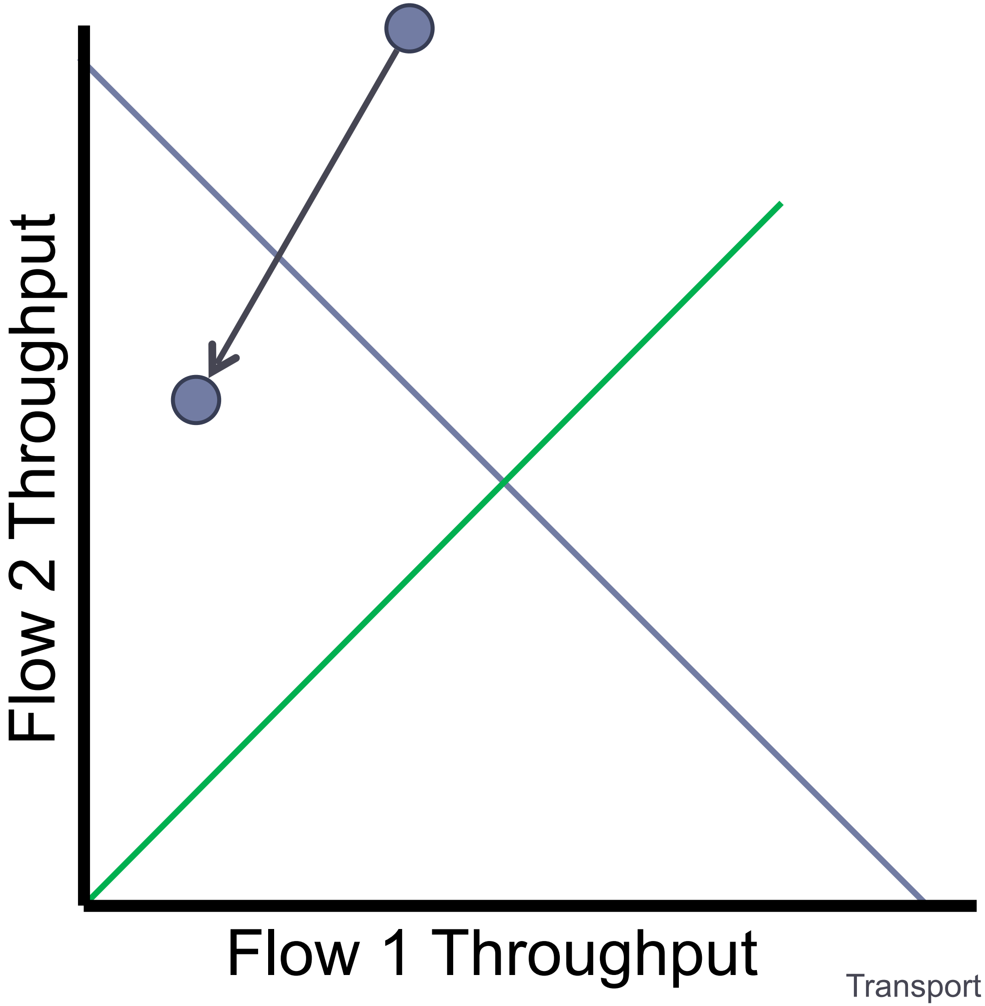

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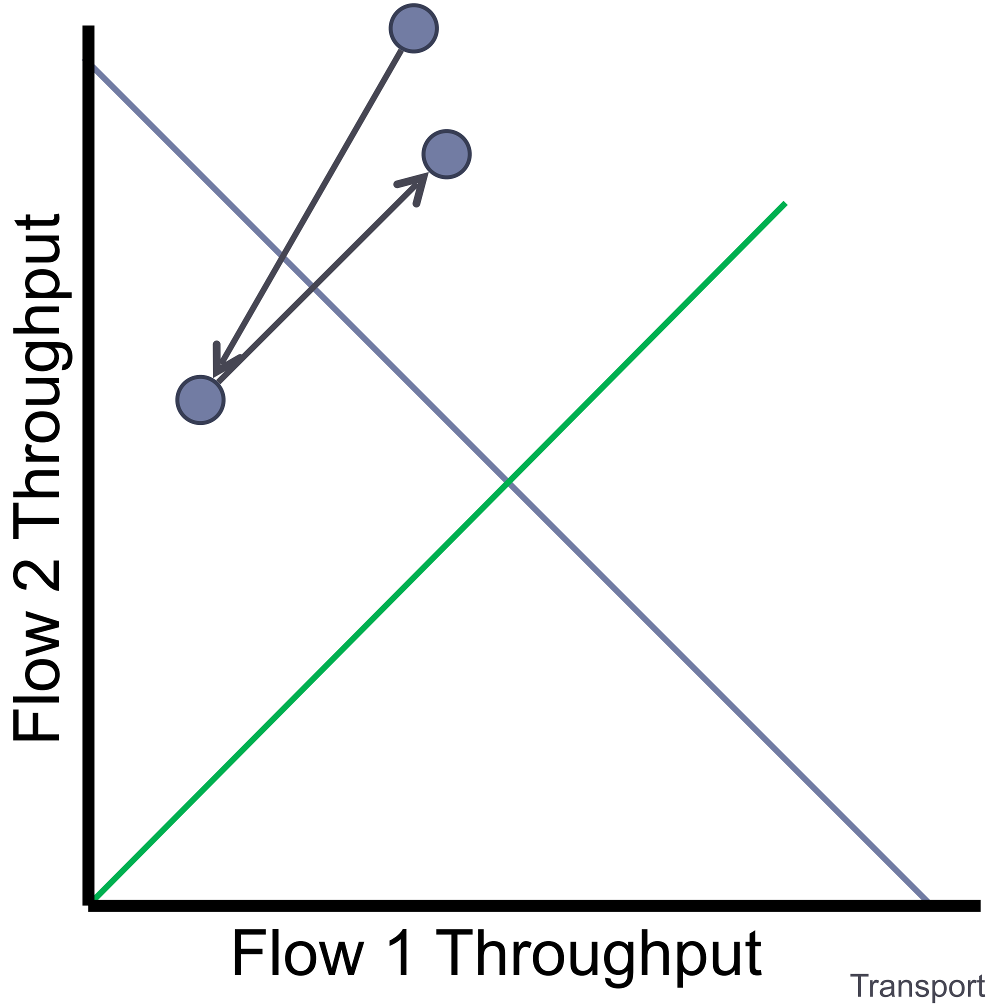

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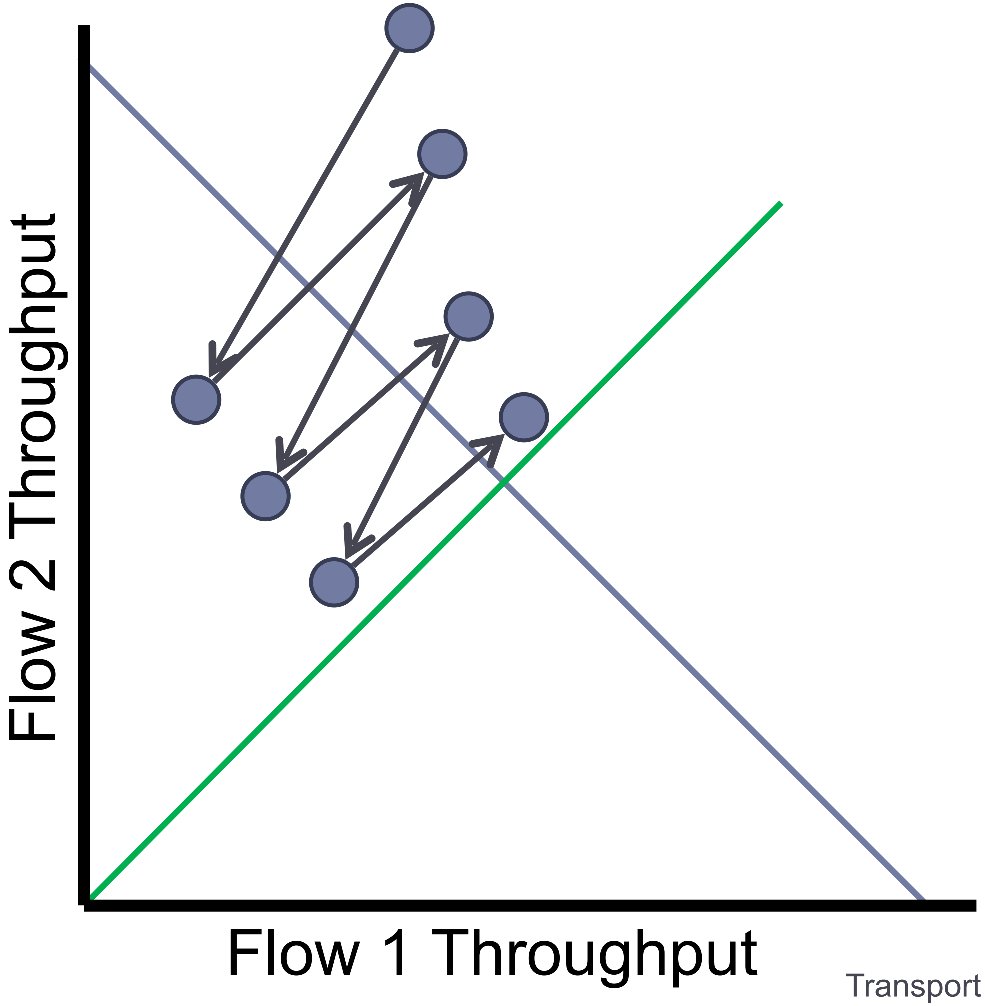

Additive Increase, Multiplicative Decrease

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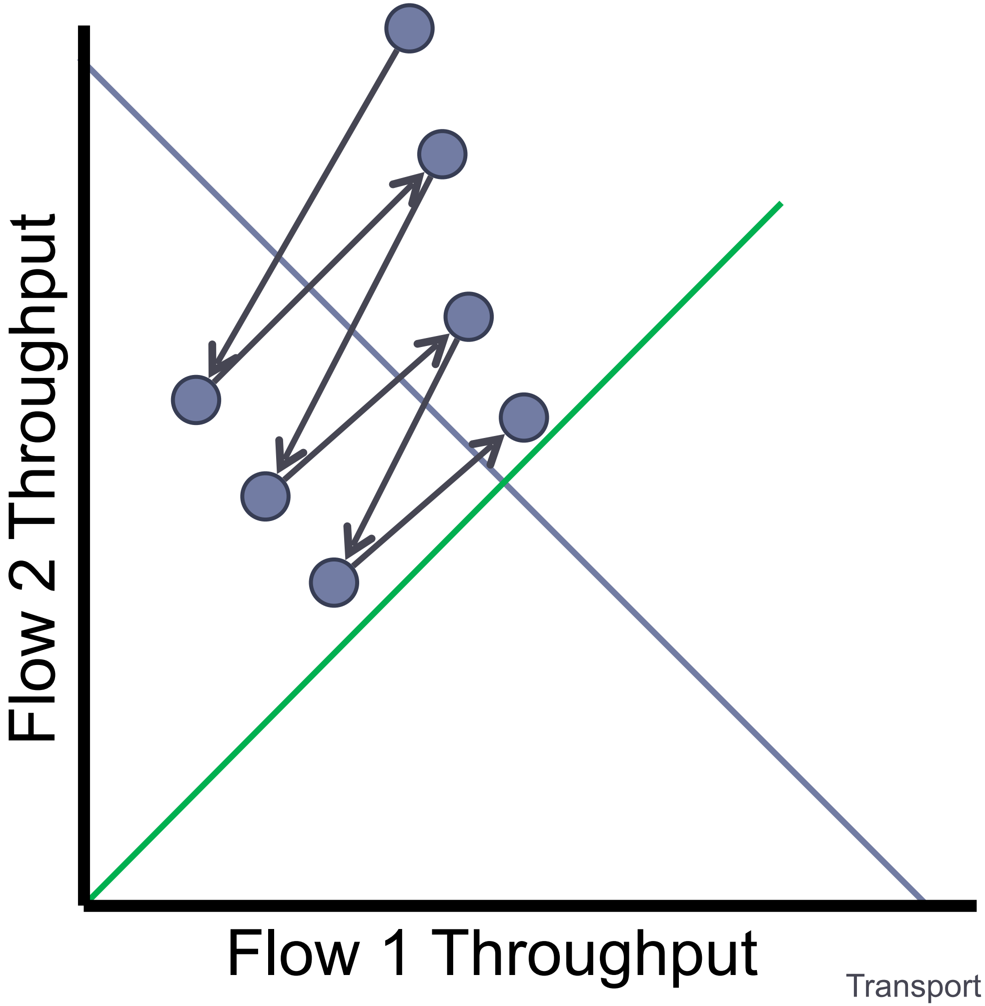

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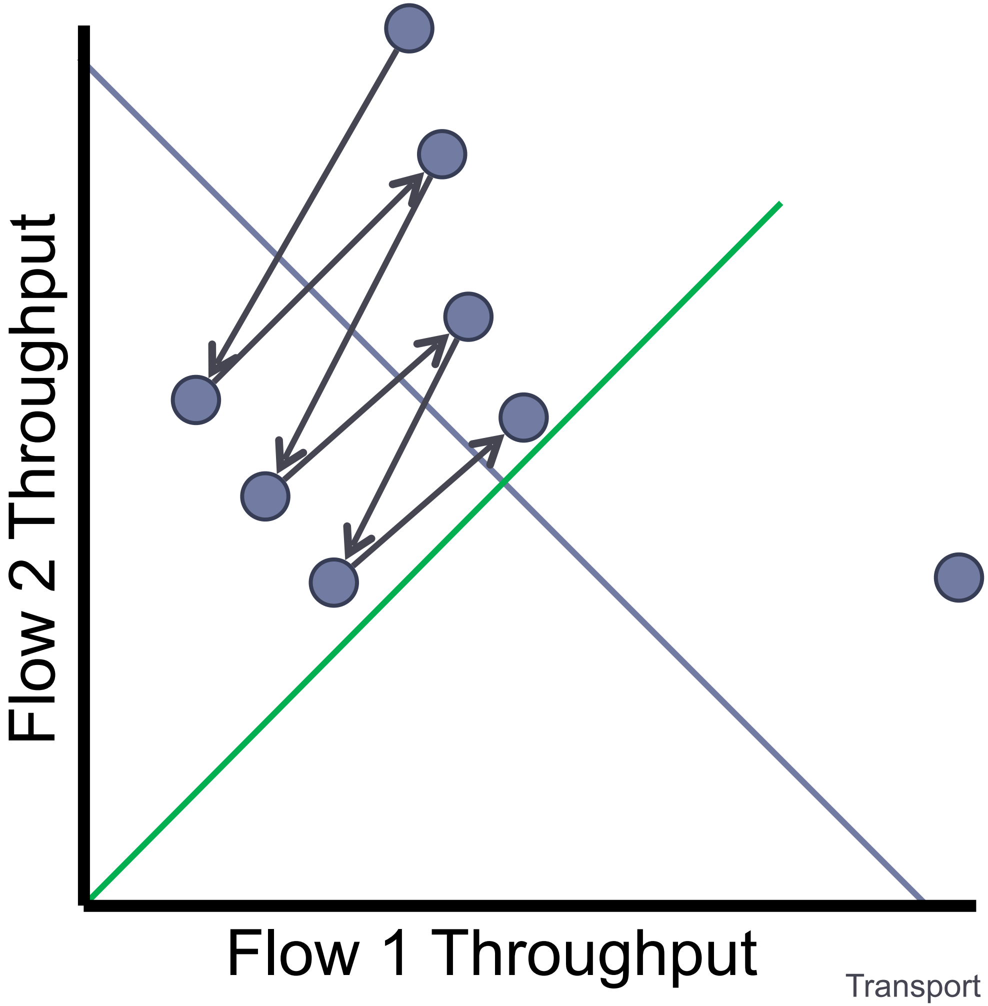
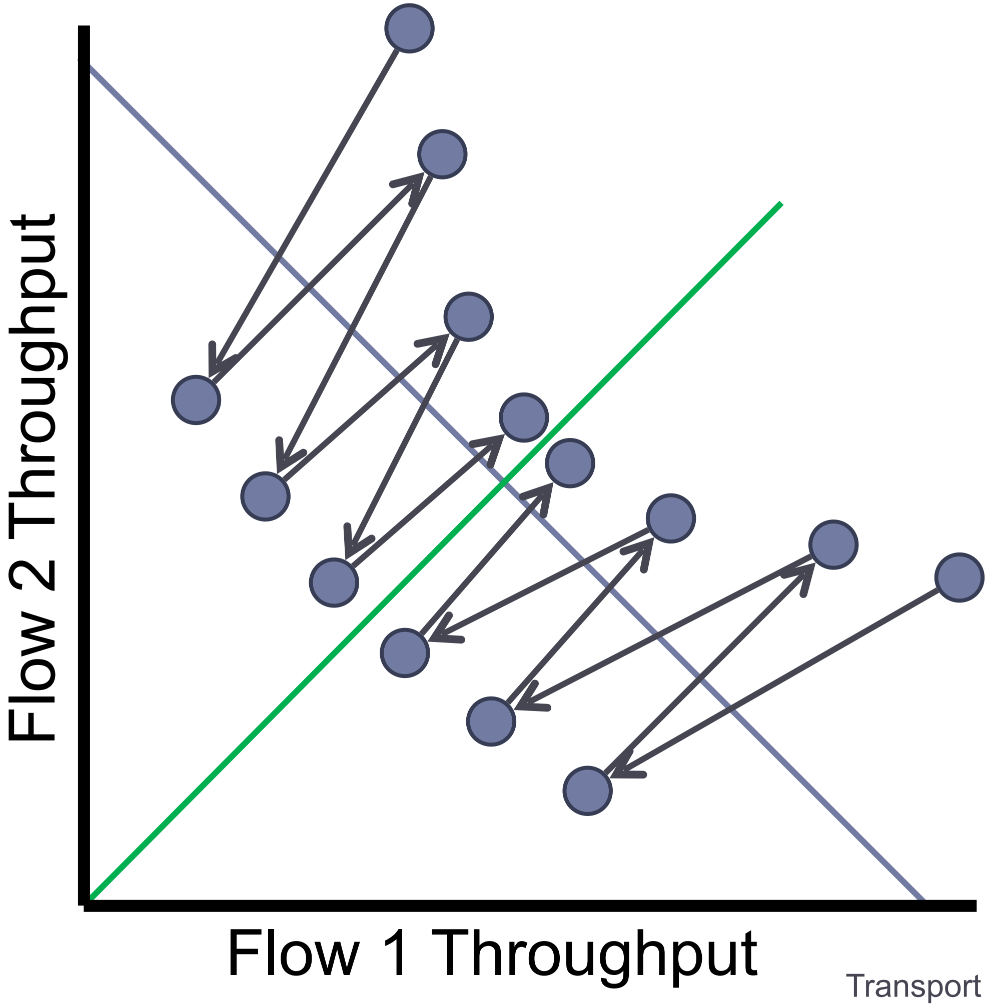


Additive Increase, Multiplicative Decrease

- Converges to stable and fair cycle
- Symmetric around *y*=*x*



Implementing Congestion Control

- Maintains three variables:
  - *cwnd*: congestion window
  - *adv_wnd*: receiver advertised window
  - *ssthresh*: threshold size (used to update *cwnd*)
- For sending, use: *wnd = min(cwnd, adv_wnd*)

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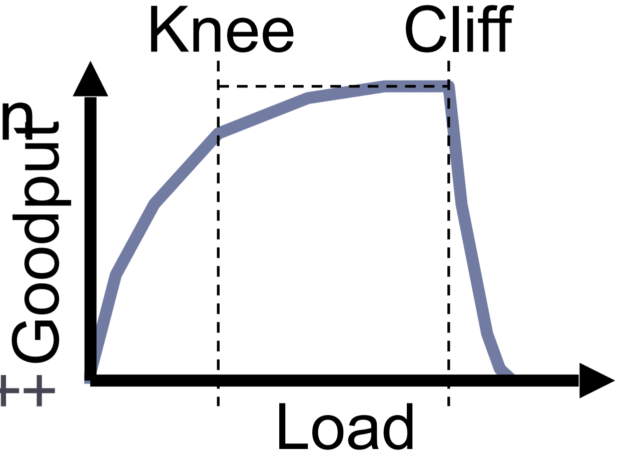
- For sending, use: *wnd = min(cwnd, adv_wnd*)
- Two phases of congestion control
  1. Slow start (*cwnd < ssthresh*)
     - Probe for bottleneck bandwidth
  2. Congestion avoidance (*cwnd >= ssthresh*)
     - AIMD

Slow Start

- Goal: reach knee quickly
- Upon starting/restarting a connection
  - *cwnd* =1
  - *ssthresh* = *adv_wnd*
  - Each time a segment is ACKed, *cwnd*++



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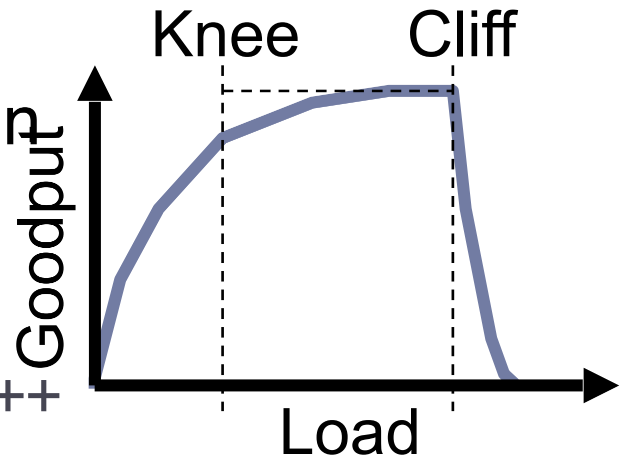
  - *ssthresh* is reached
  - Or a packet is lost



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- Continues until…
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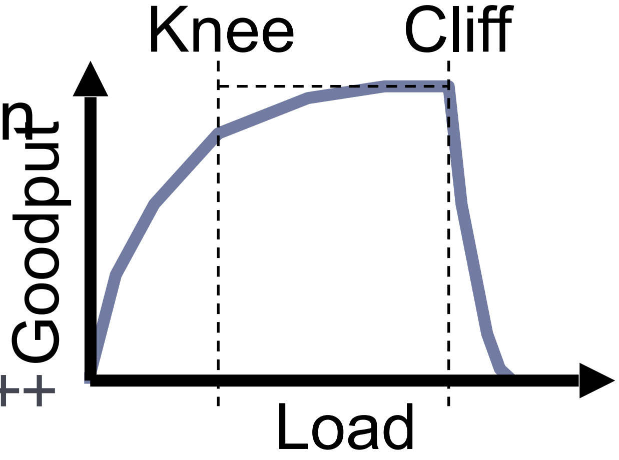
  - Or a packet is lost
- Slow Start is not actually slow
  - *cwnd* increases exponentially



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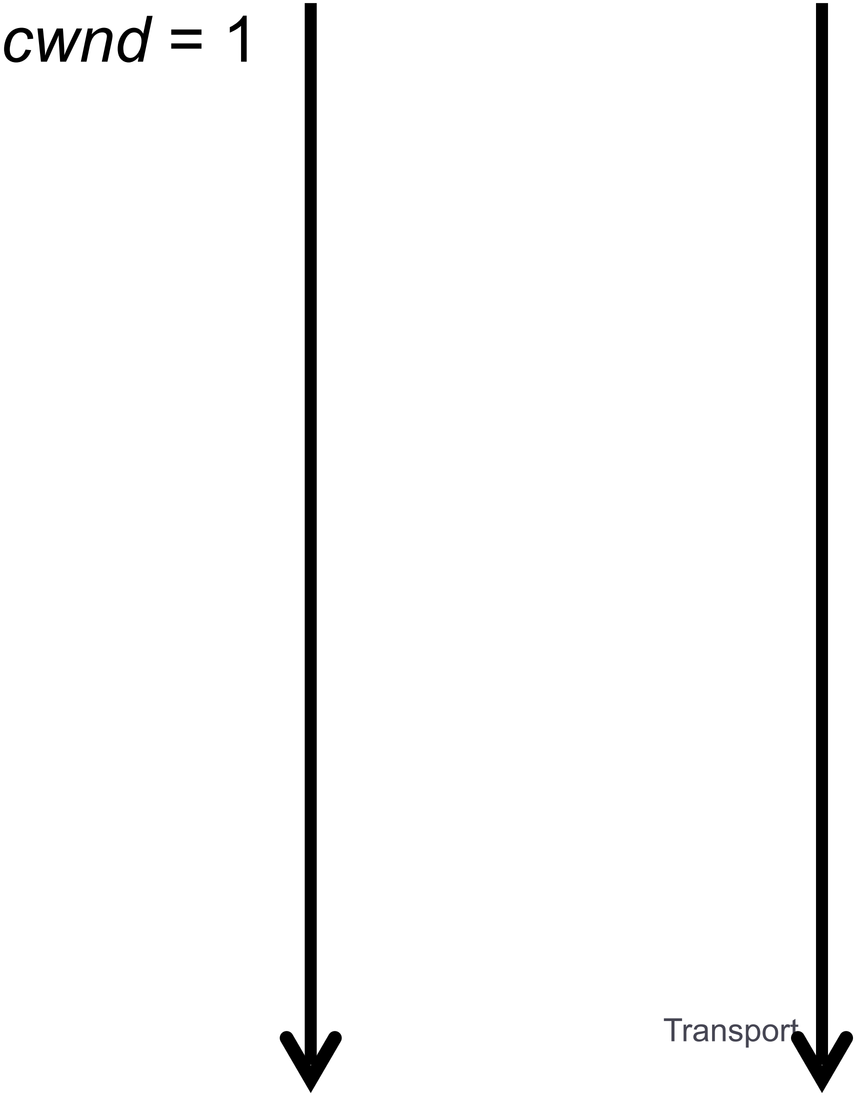
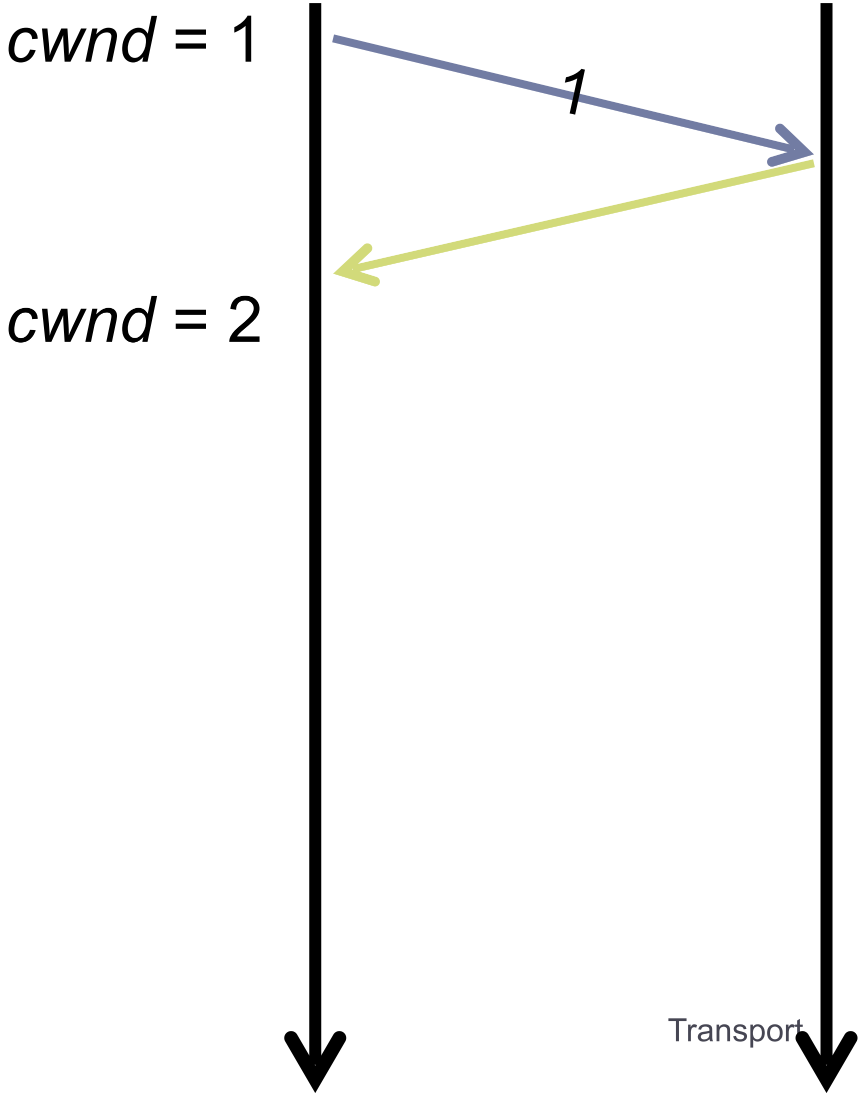


cwnd = 1

Slow Start Example

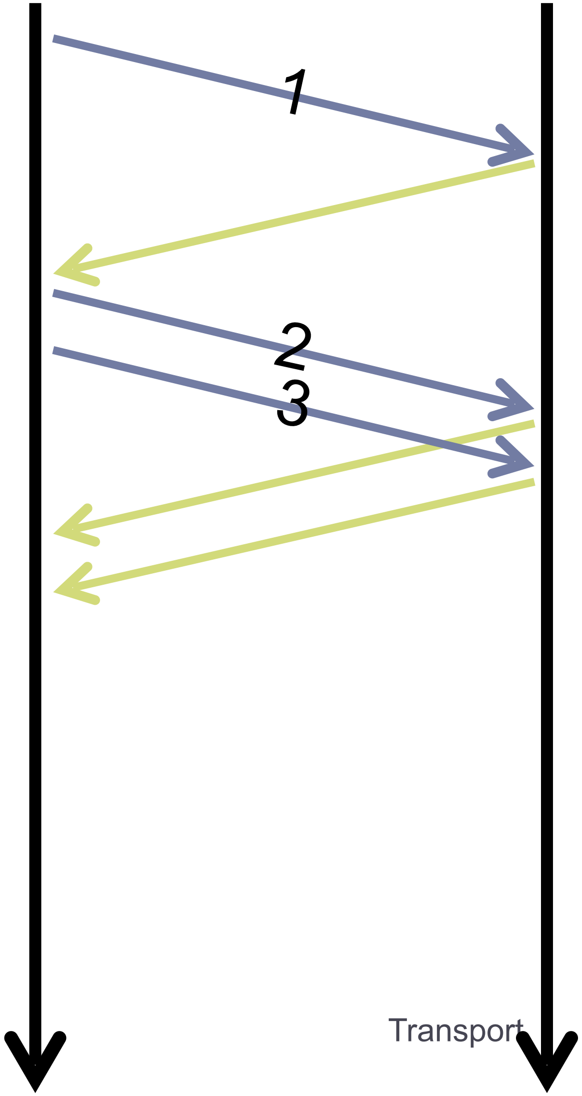


Slow Start Example

cwnd = 1

cwnd = 2

cwnd = 4



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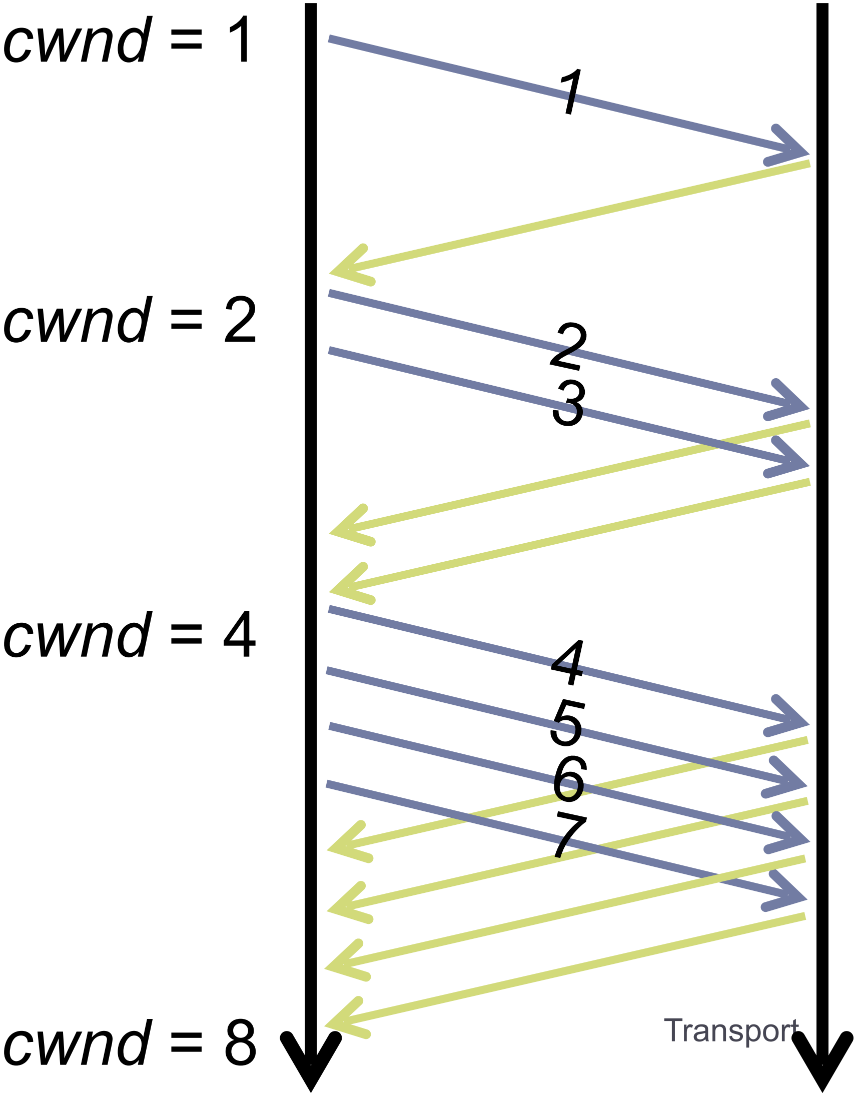


cwnd = 1

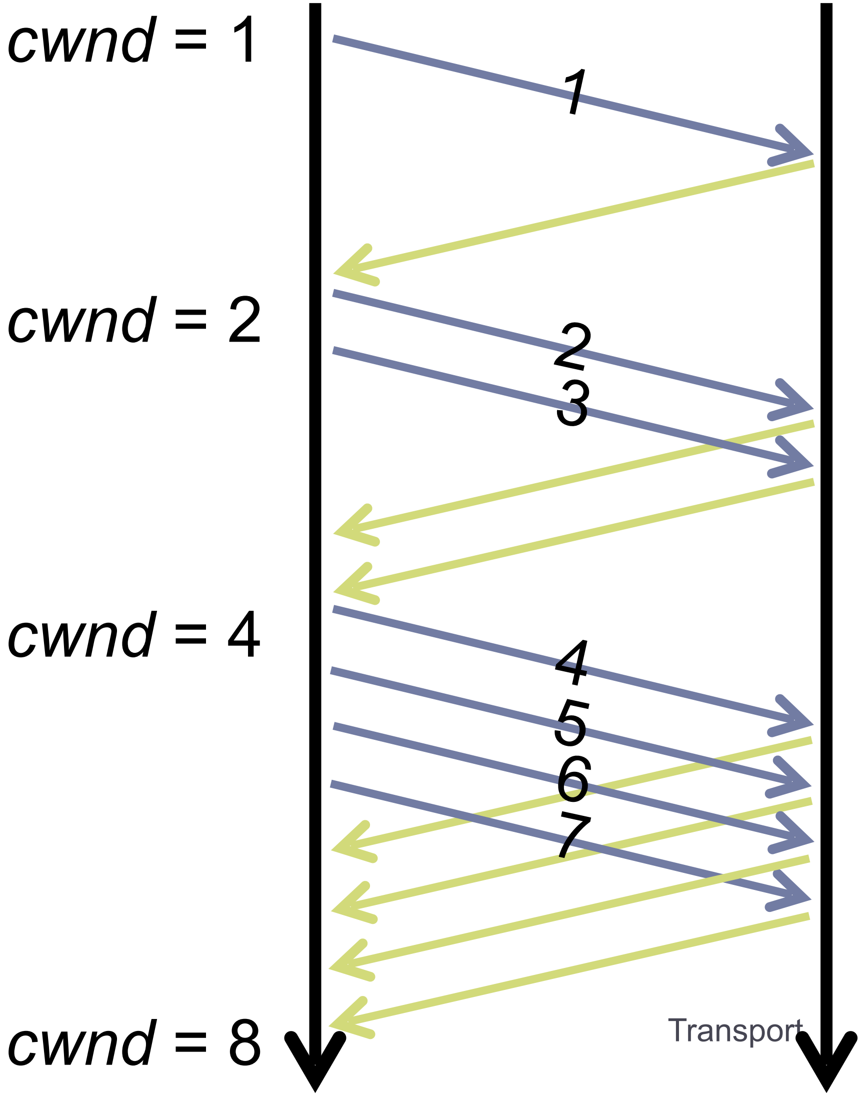
cwnd = 2

cwnd = 4

cwnd = 8

Slow Start Example

- *cwnd* grows rapidly
- Slows down when…
  - *cwnd >= ssthresh*
  - Or a packet drops



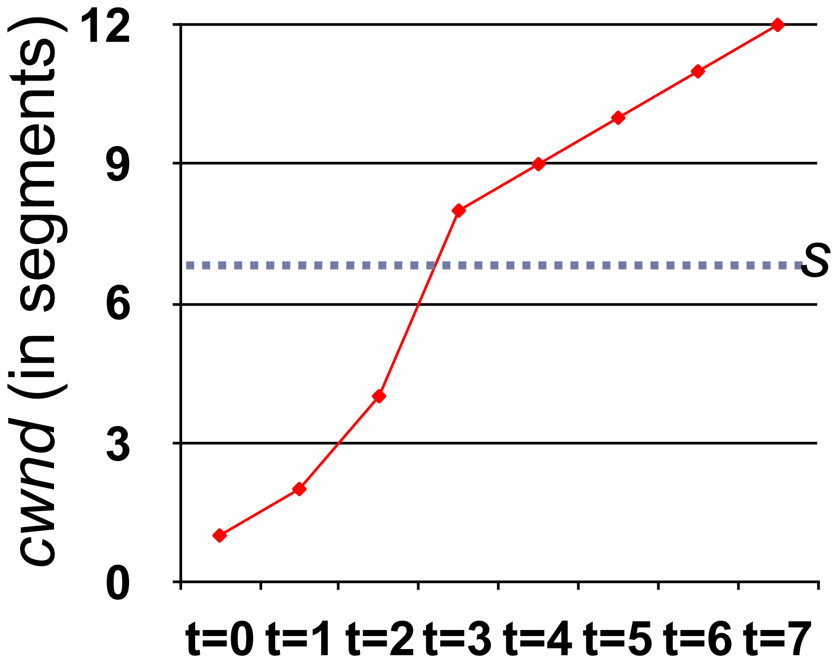
Congestion Avoidance

- AIMD mode
- *ssthresh* is lower-bound guess about location of the knee
- **If** *cwnd >= ssthresh* **then**

  each time a segment is ACKed

  increment *cwnd by 1/cwnd (cwnd += 1/cwnd).*
- So *cwnd* is increased by one only if all segments have been acknowledged

Congestion Avoidance Example



Round Trip Times

ssthresh = 8

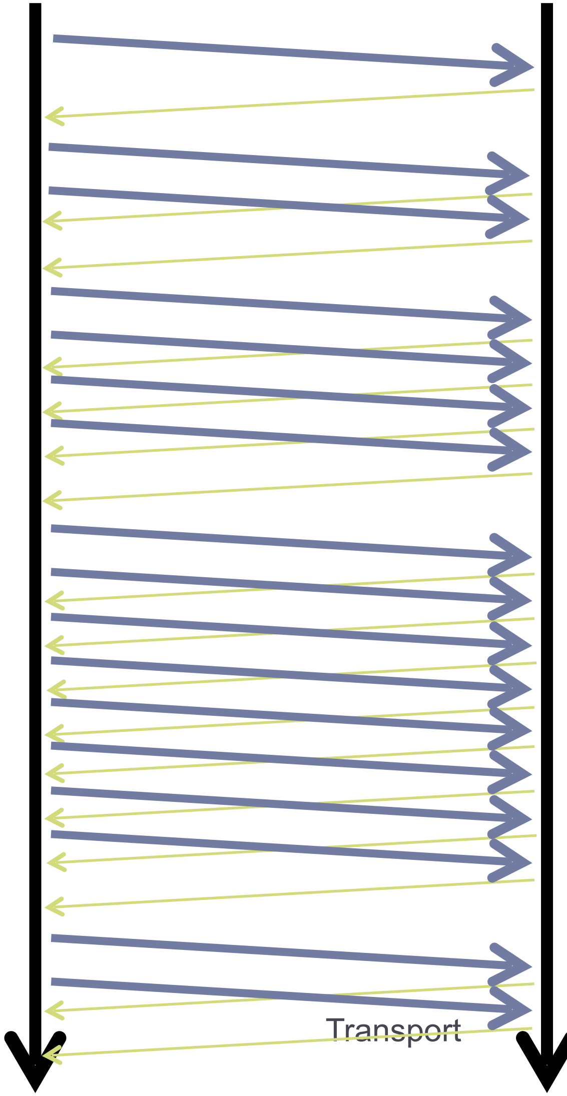
cwnd = 1

cwnd = 2

cwnd = 4

cwnd = 8

cwnd = 9



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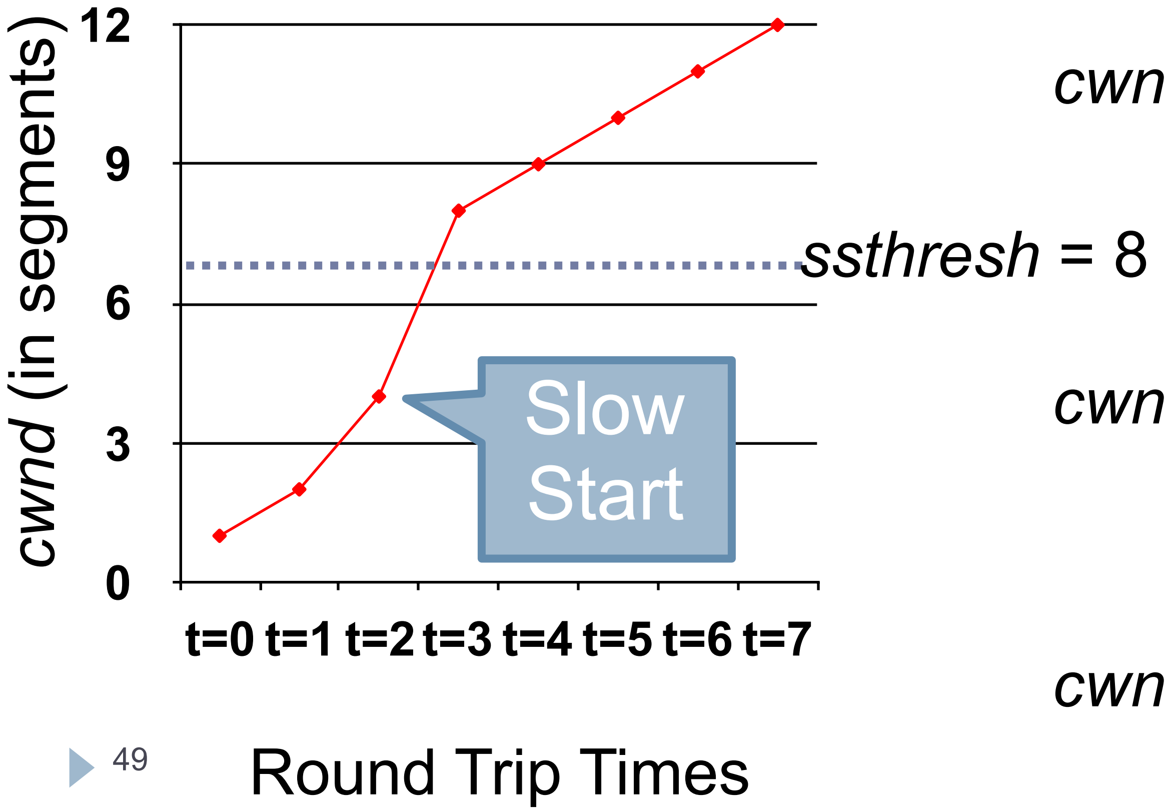

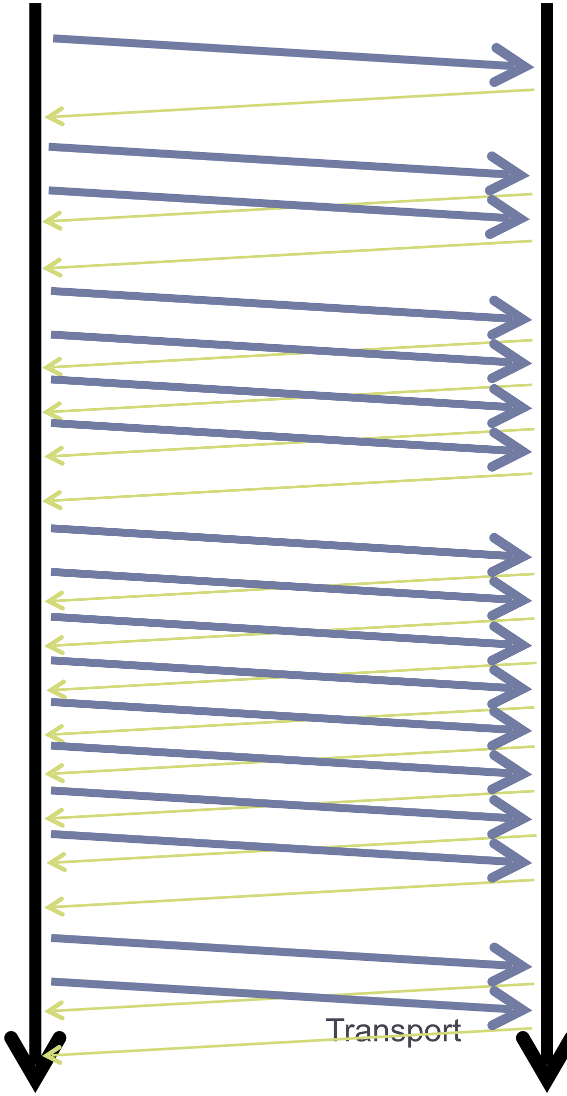

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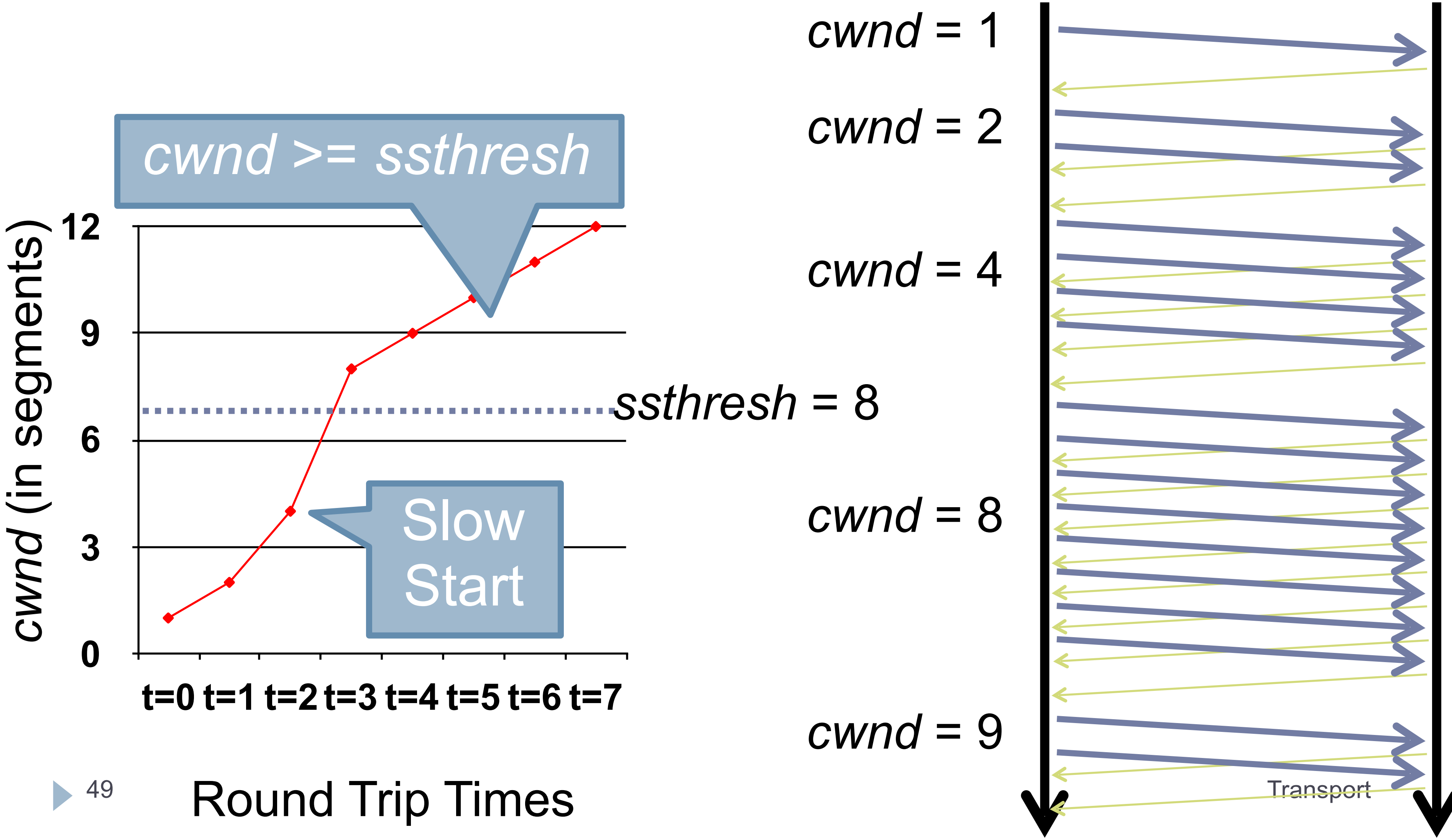
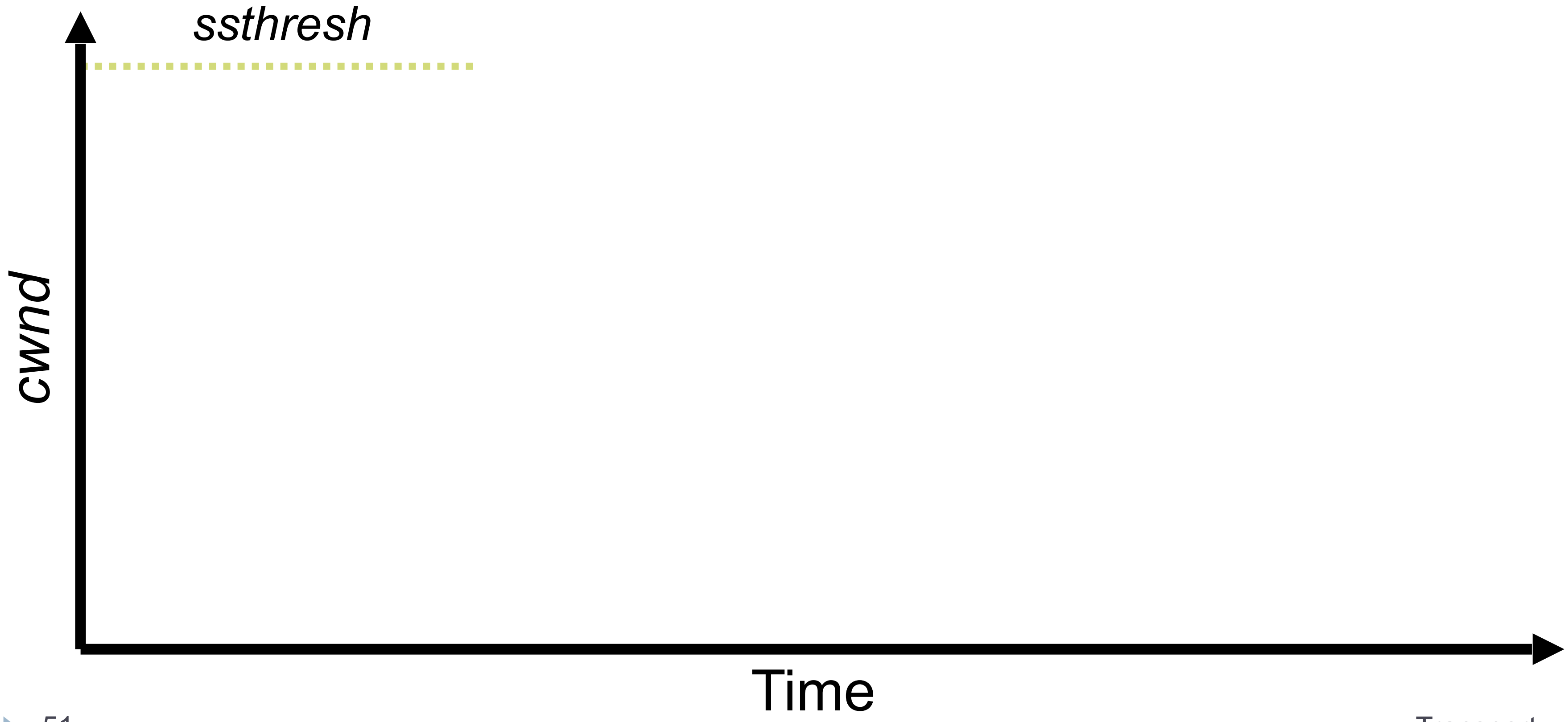


TCP Pseudocode

```
Initially:
        cwnd = 1;
        ssthresh = adv_wnd;
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 Big Picture



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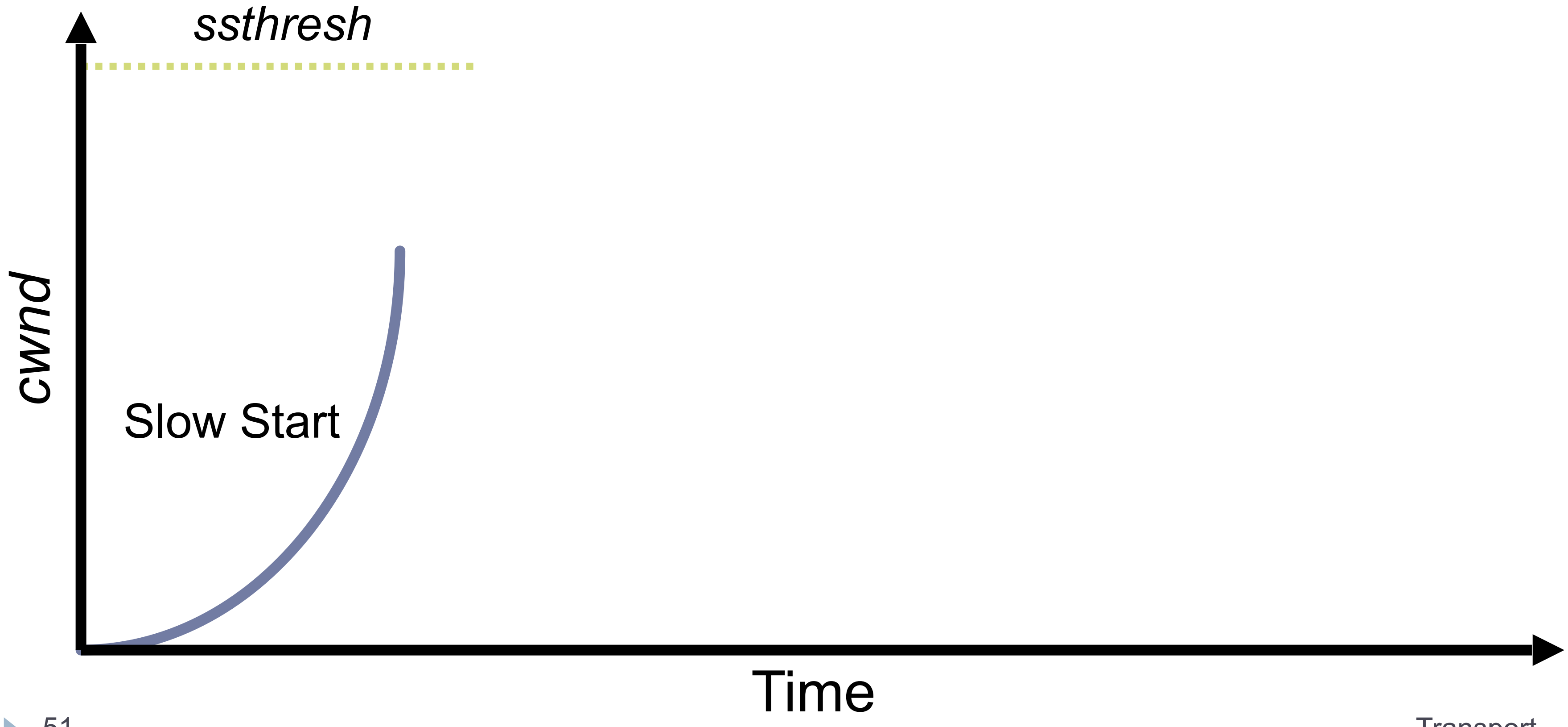

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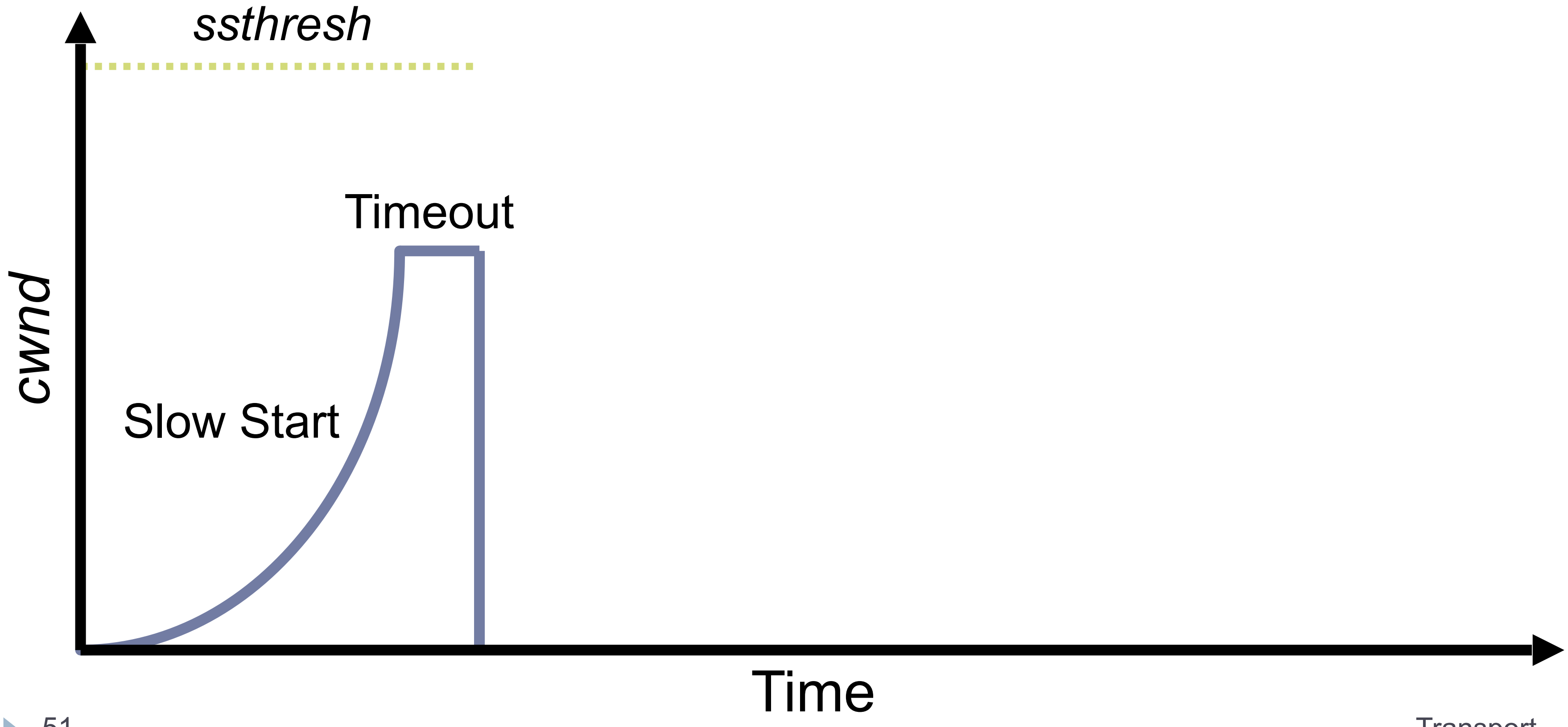

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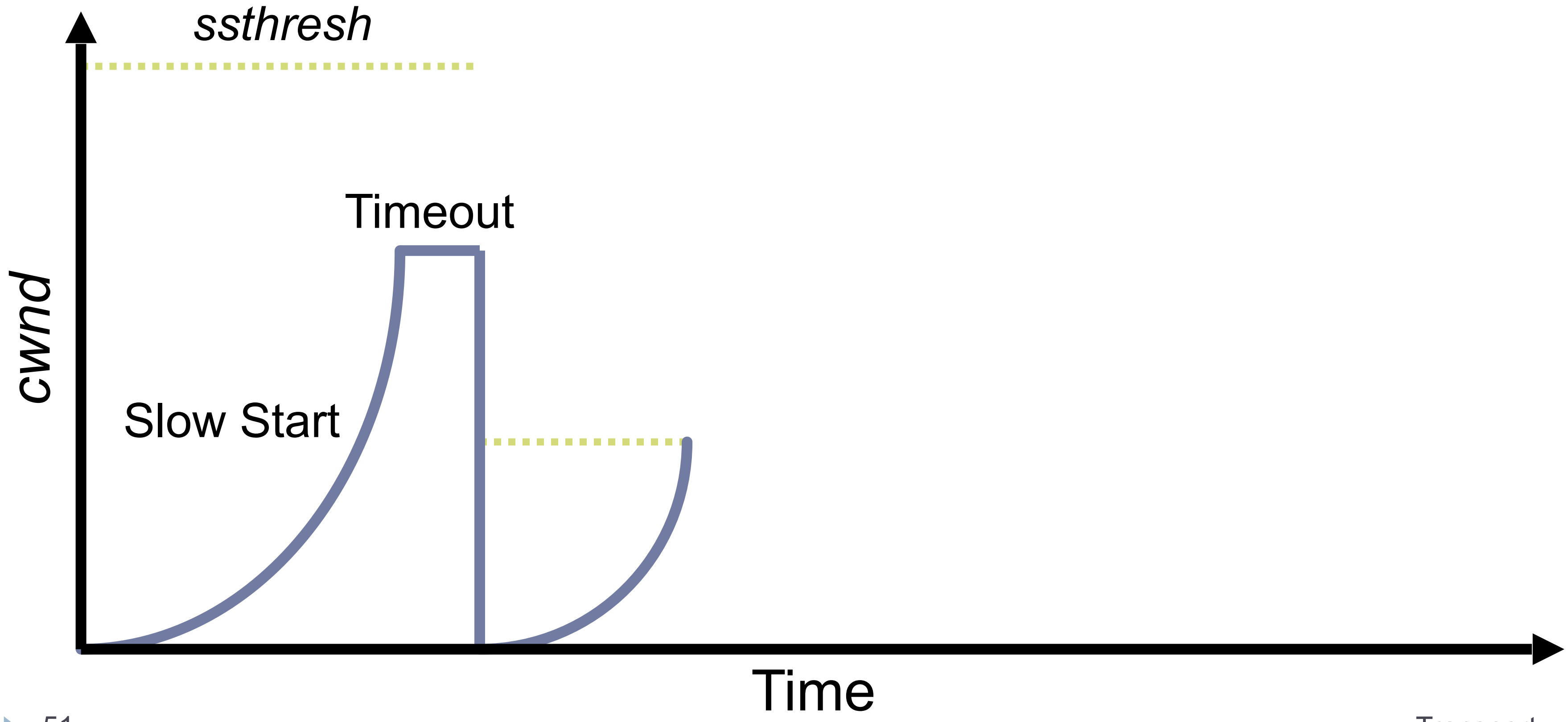

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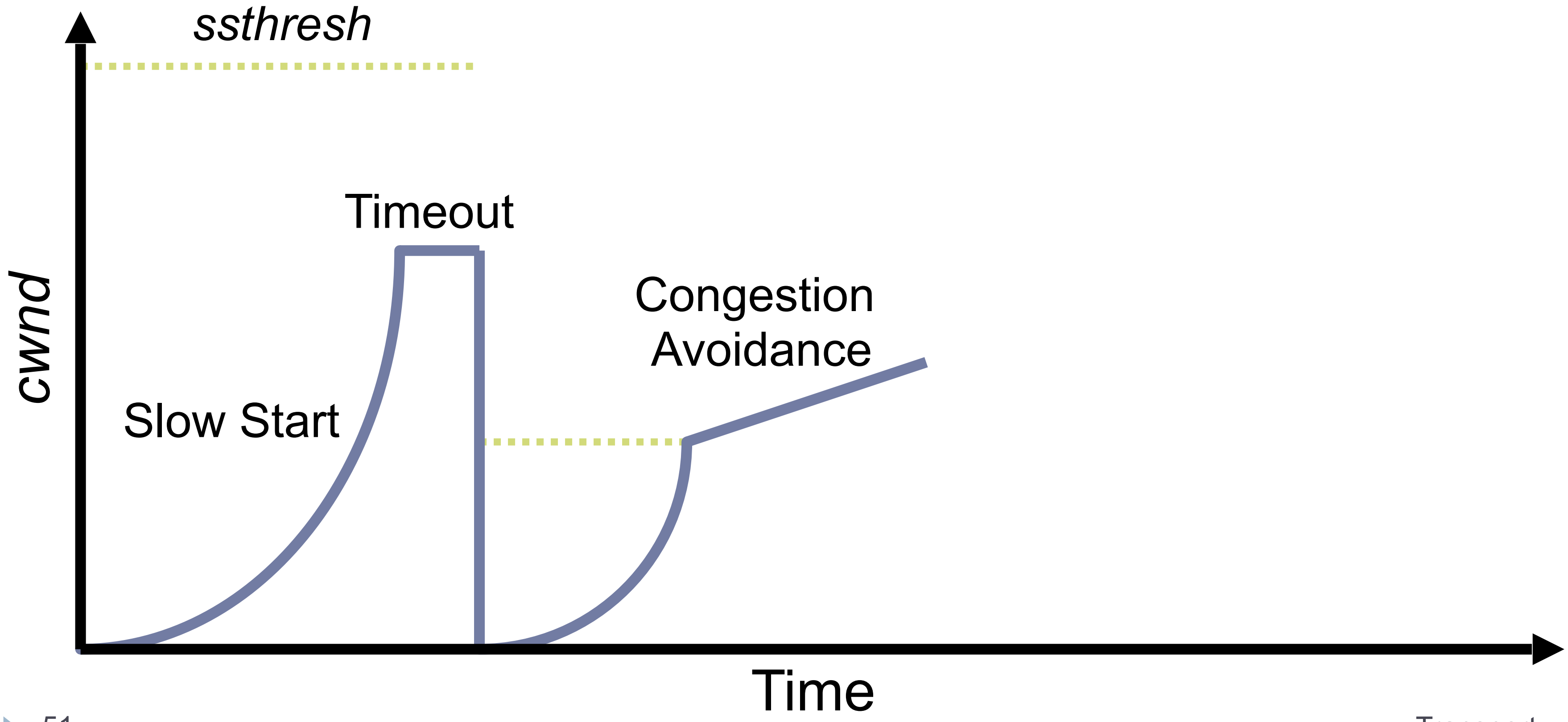

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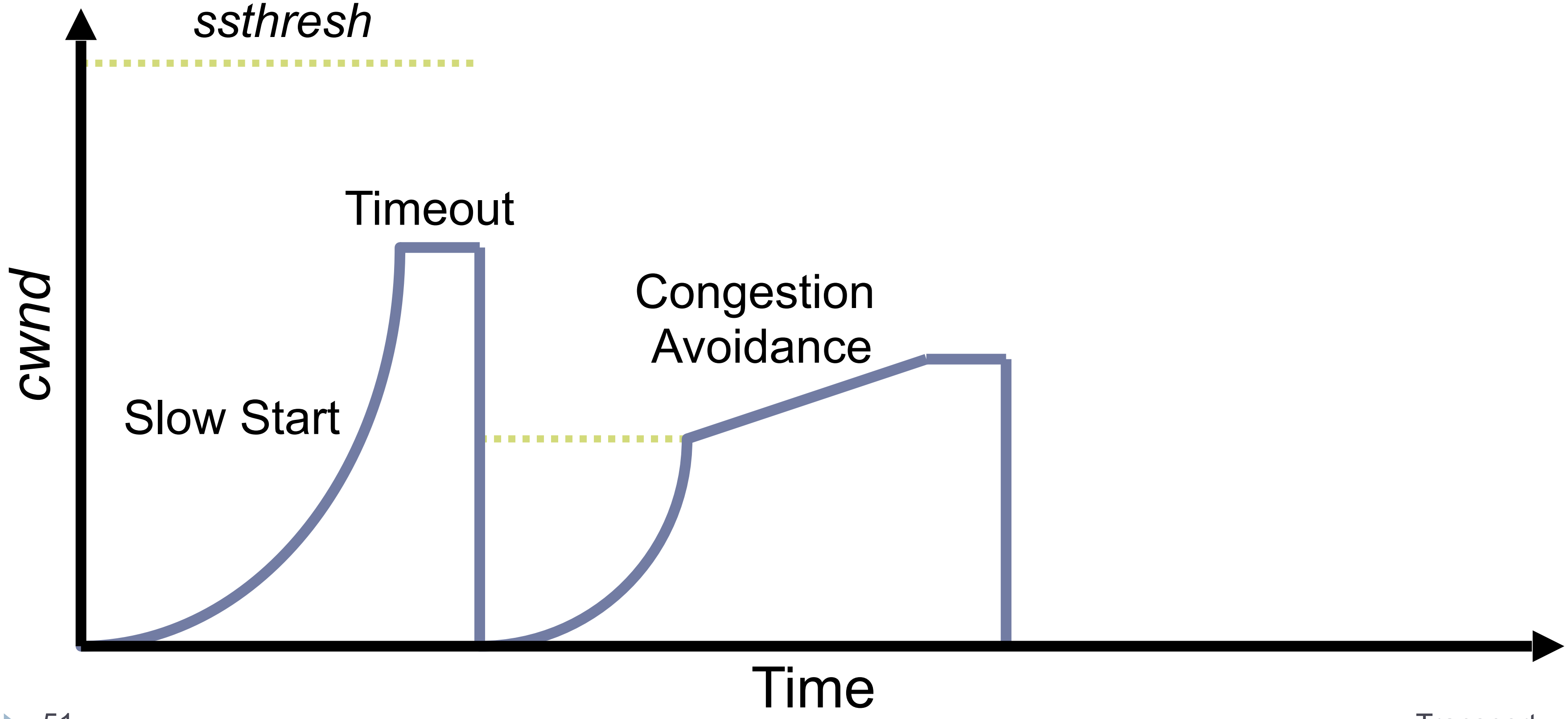

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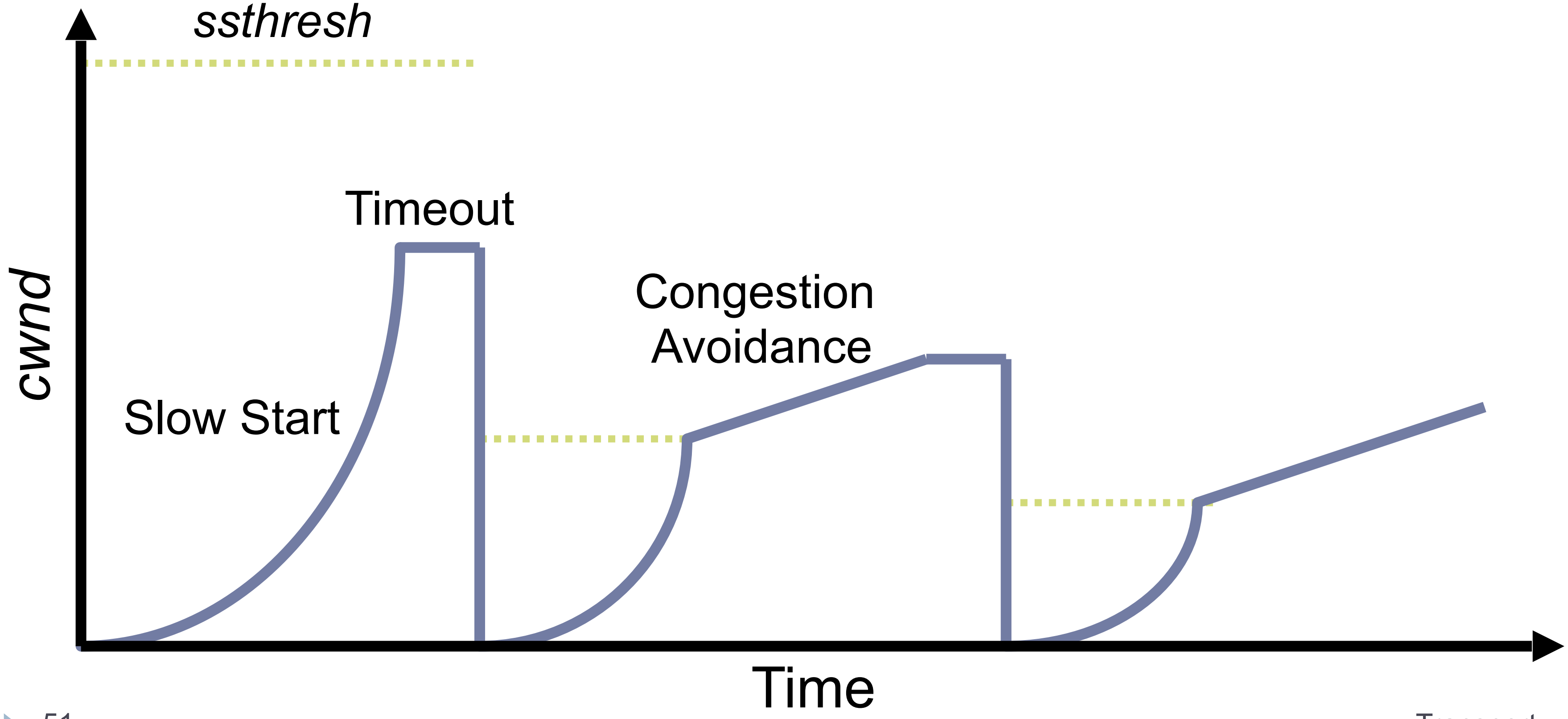

4: Evolution of TCP

The Evolution of TCP

- Thus far, we have discussed TCP Tahoe
  - Original version of TCP
- However, TCP was invented in 1974!
  - Today, there are many variants of TCP

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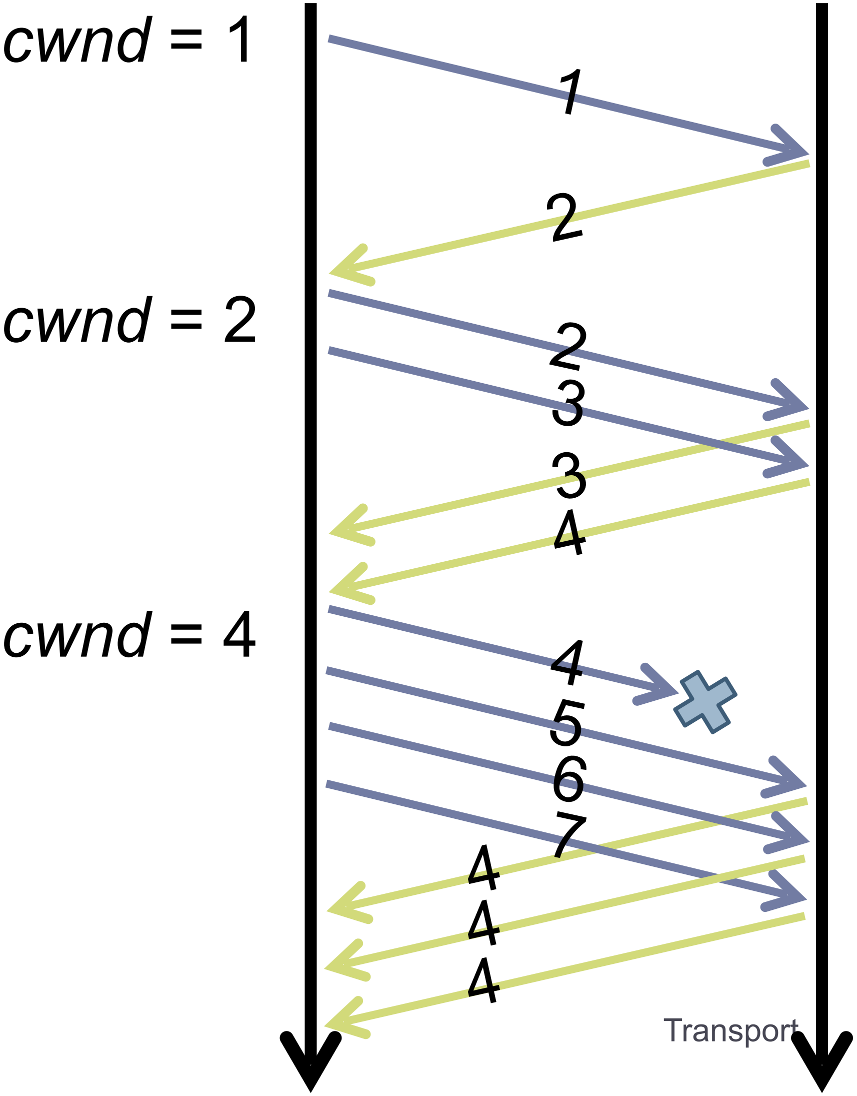
  - Original version of TCP
- However, TCP was invented in 1974!
  - Today, there are many variants of TCP
- Early, popular variant: TCP Reno
  - Tahoe features, plus…
  - Fast retransmit
  - Fast recovery

TCP Reno: Fast Retransmit

- Problem: in Tahoe, if segment is lost, there is a long wait until the RTO
- Reno: retransmit after 3 duplicate ACKs



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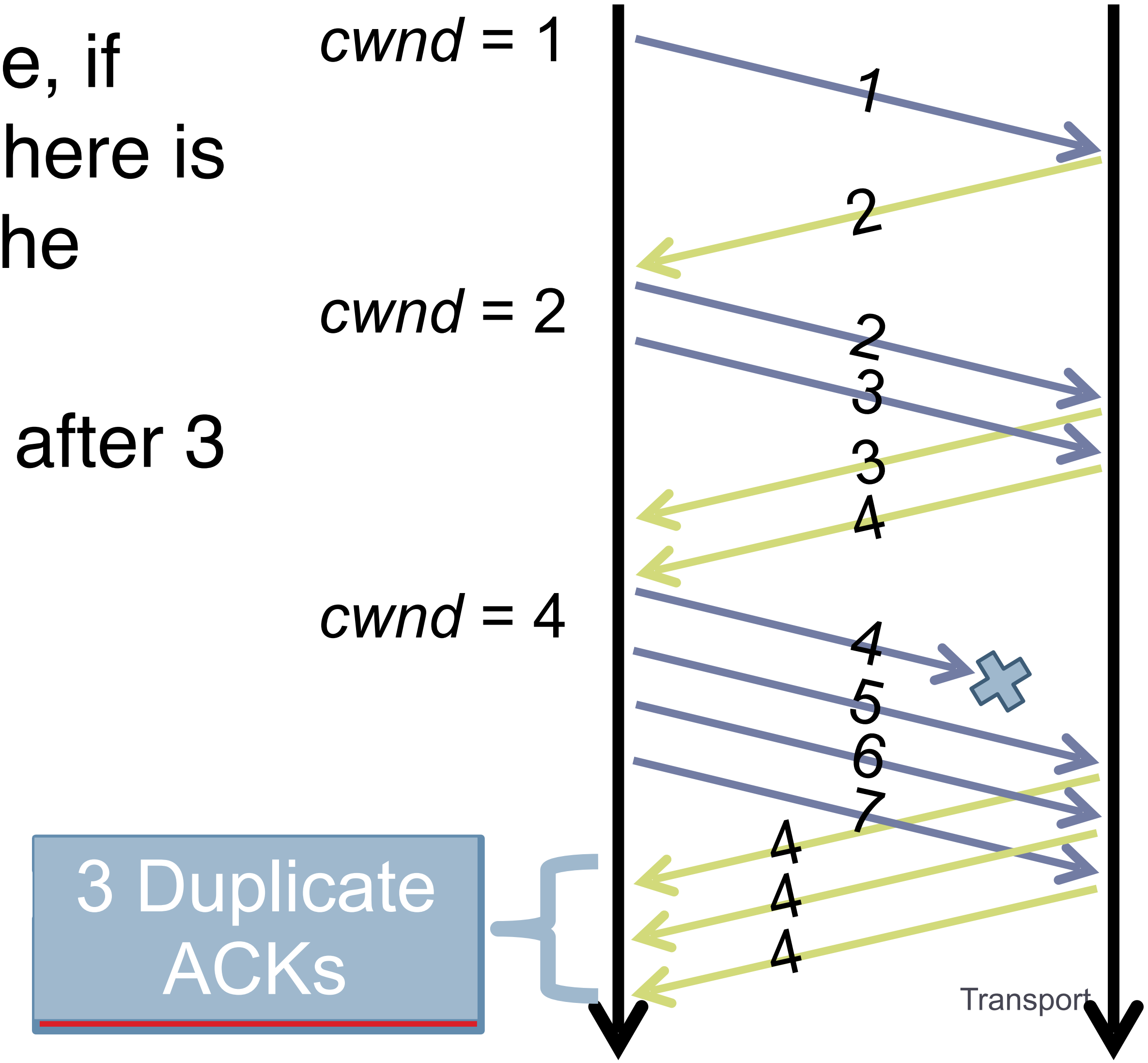


cwnd = 1

cwnd = 2

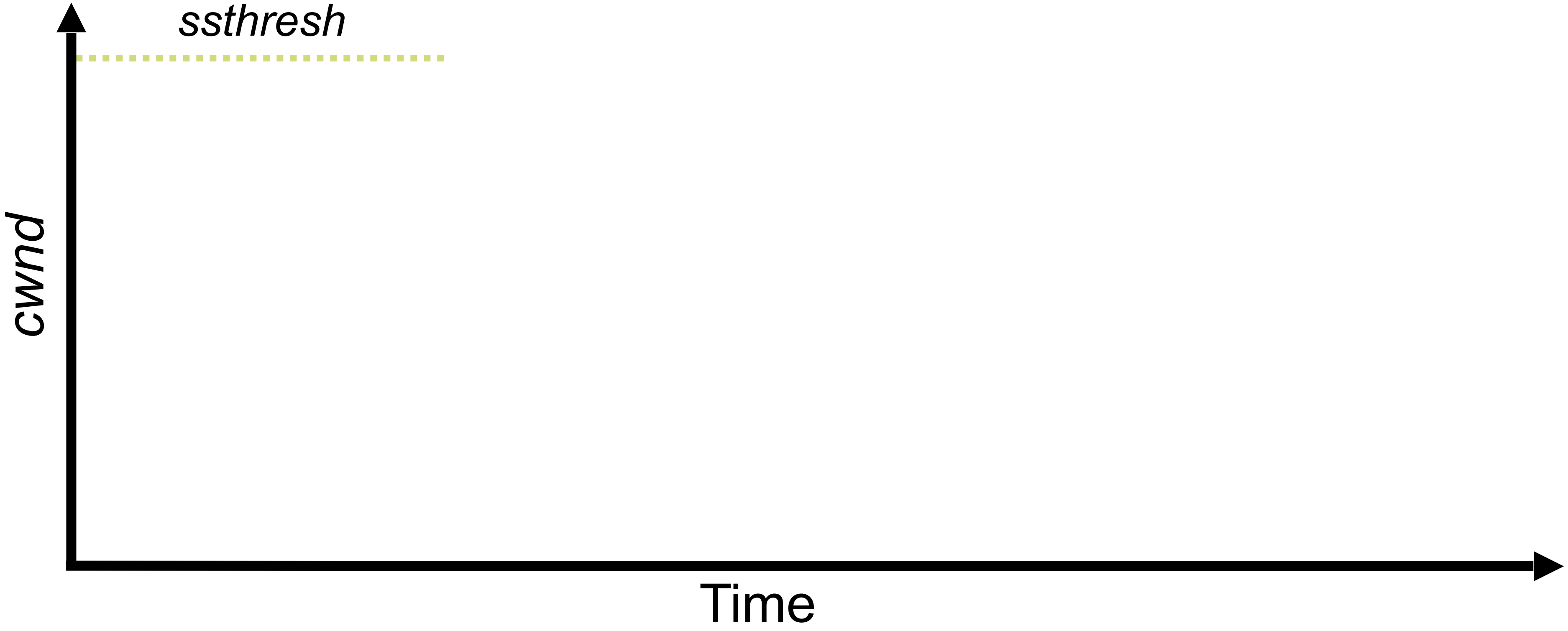
cwnd = 4

3 Duplicate ACKs

TCP Reno: Fast Recovery

- After a fast-retransmit set *cwnd* to *ssthresh/2*
  - i.e. don't reset *cwnd* to 1
  - Avoid unnecessary return to slow start
  - Prevents expensive timeouts
- But when RTO expires still do *cwnd* = 1
  - Return to slow start, same as Tahoe
  - Indicates packets aren't being delivered at all
  - i.e. congestion must be really bad

Fast Retransmit and Fast Recovery



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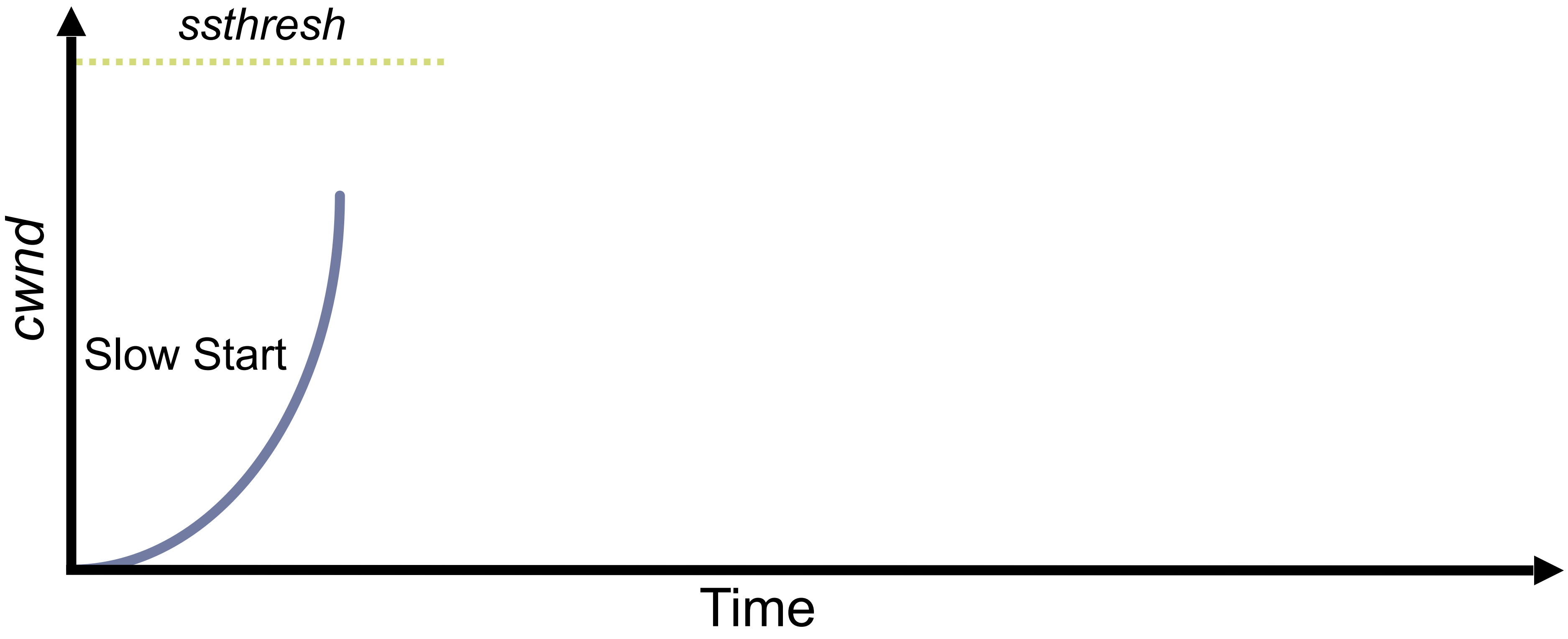

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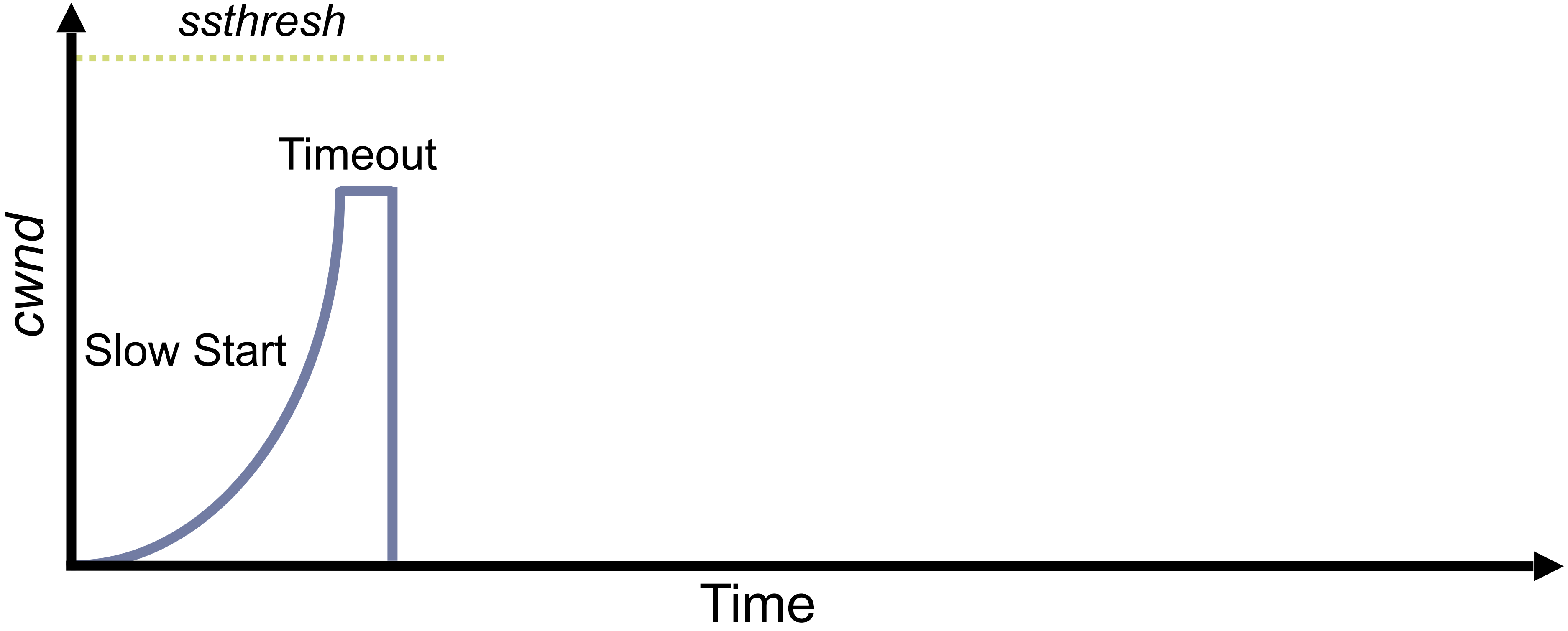

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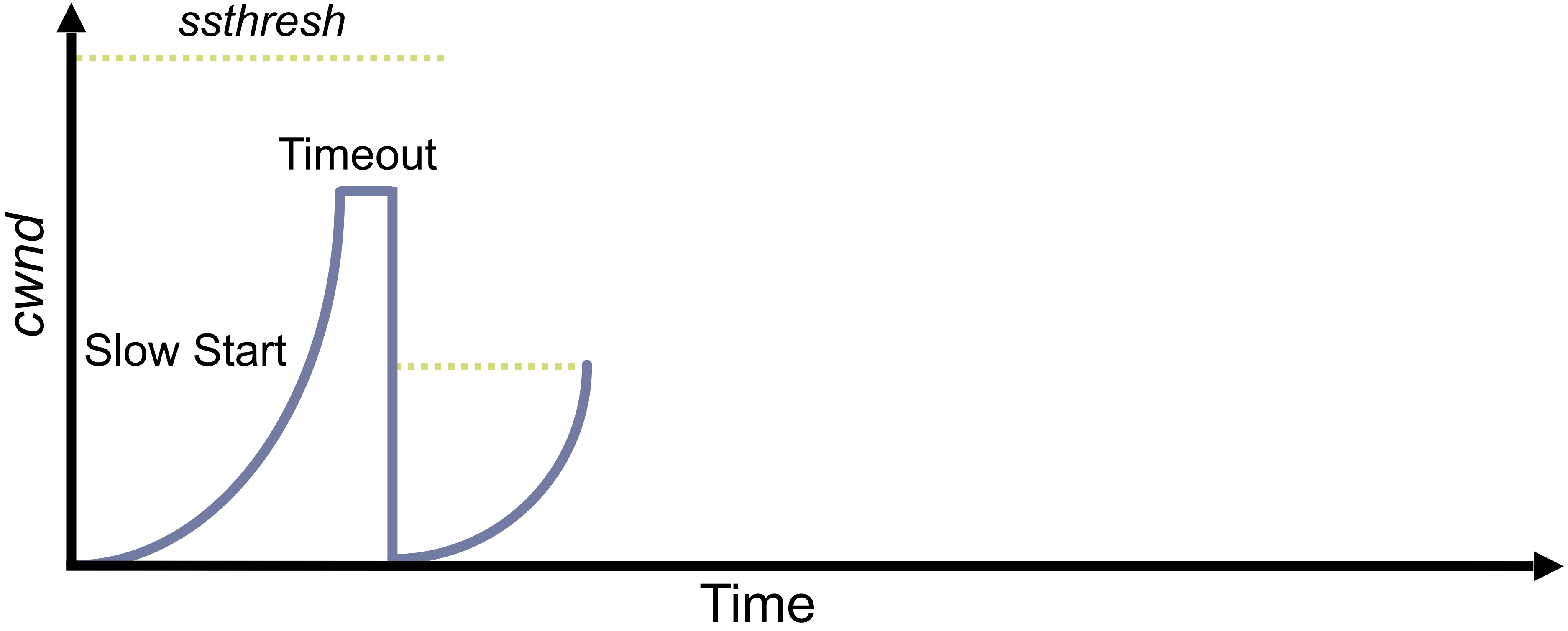

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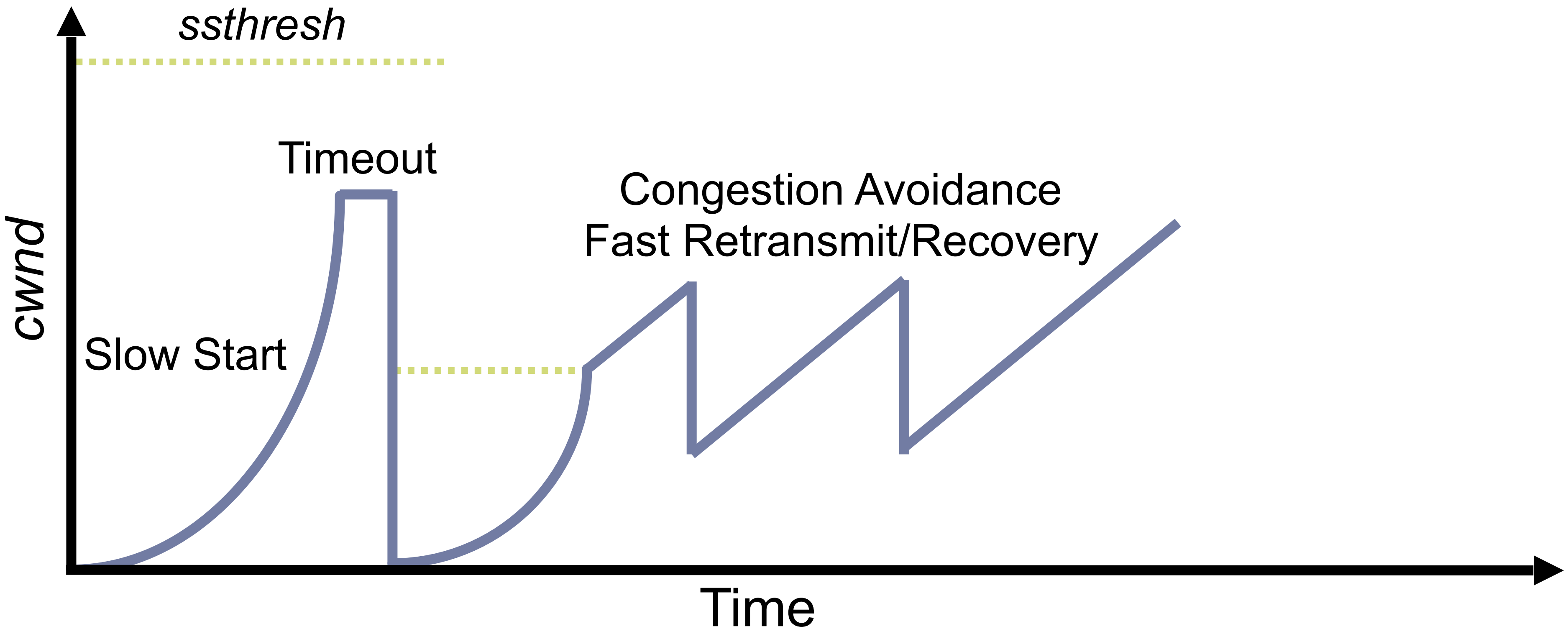

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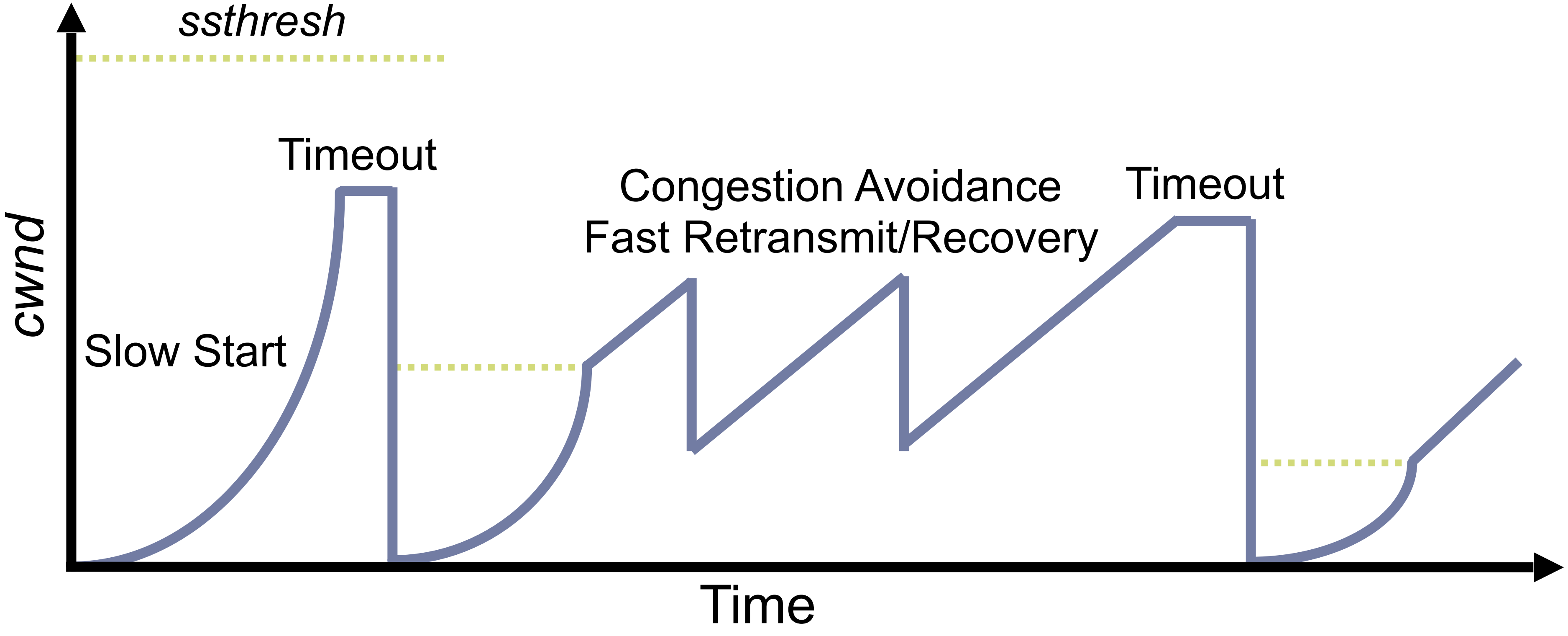

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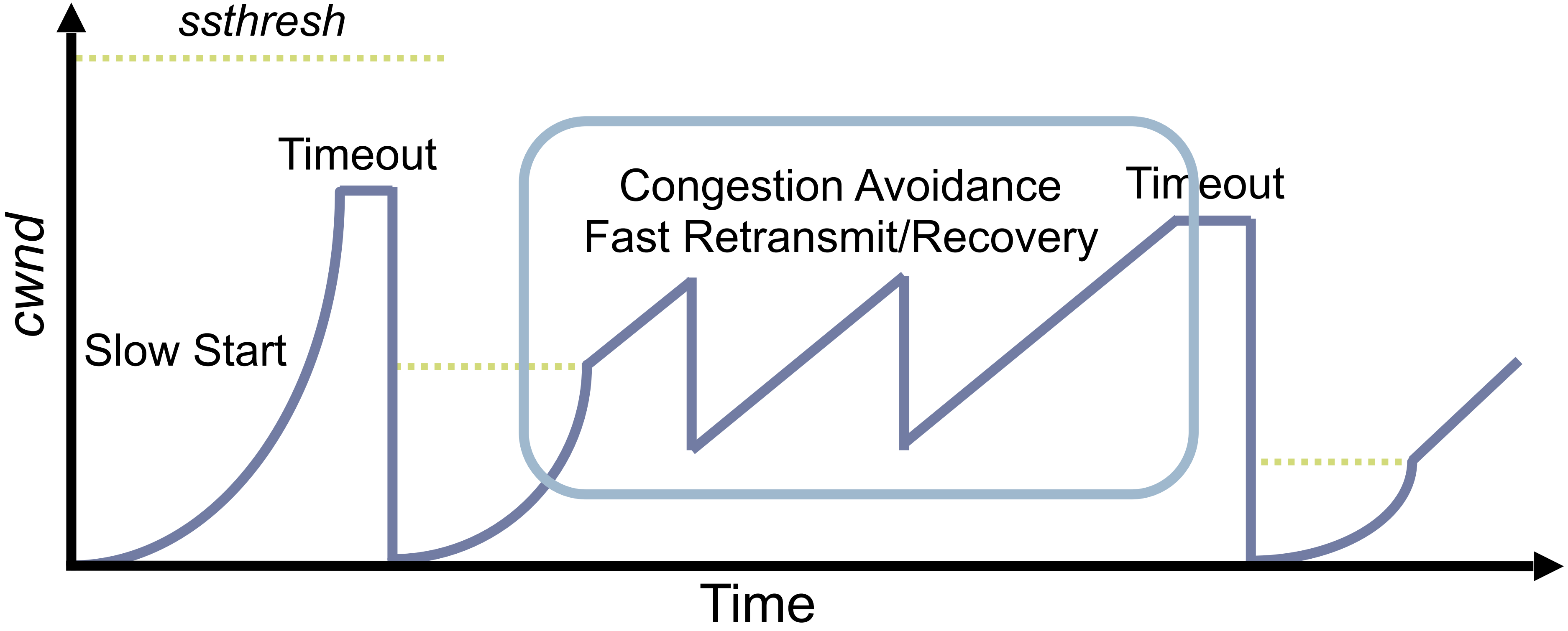

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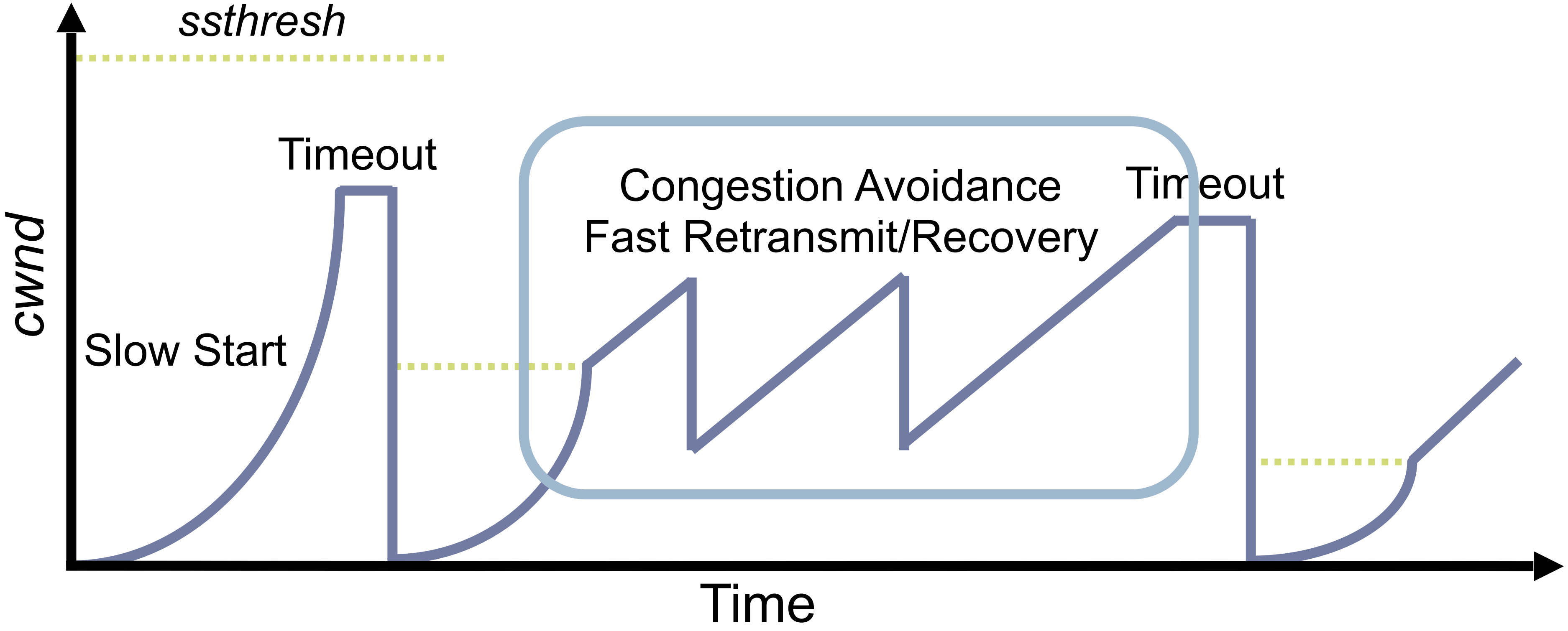

- At steady state, *cwnd* oscillates around the optimal window size
- TCP always forces packet drops

Many TCP Variants…

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- Vegas: delay-based congestion avoidance
- And many, many, many more…

TCP in the Real World

- What are the most popular variants today?
  - Key problem: TCP performs poorly on high bandwidth-delay product networks (like the modern Internet)
  - Compound TCP (Windows)
    - Based on Reno
    - Uses two congestion windows: delay based and loss based
    - Thus, it uses a *compound* congestion controller
  - TCP CUBIC (Linux)
    - Enhancement of BIC (Binary Increase Congestion Control)
    - Window size controlled by cubic function
    - Parameterized by the time *T* since the last dropped packet

High Bandwidth-Delay Product

- Key Problem: TCP performs poorly when
  - The capacity of the network (bandwidth) is large
  - The delay (RTT) of the network is large
  - Or, when bandwidth * delay is large
    - b * d = maximum amount of in-flight data in the network
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    - b * d = maximum amount of in-flight data in the network
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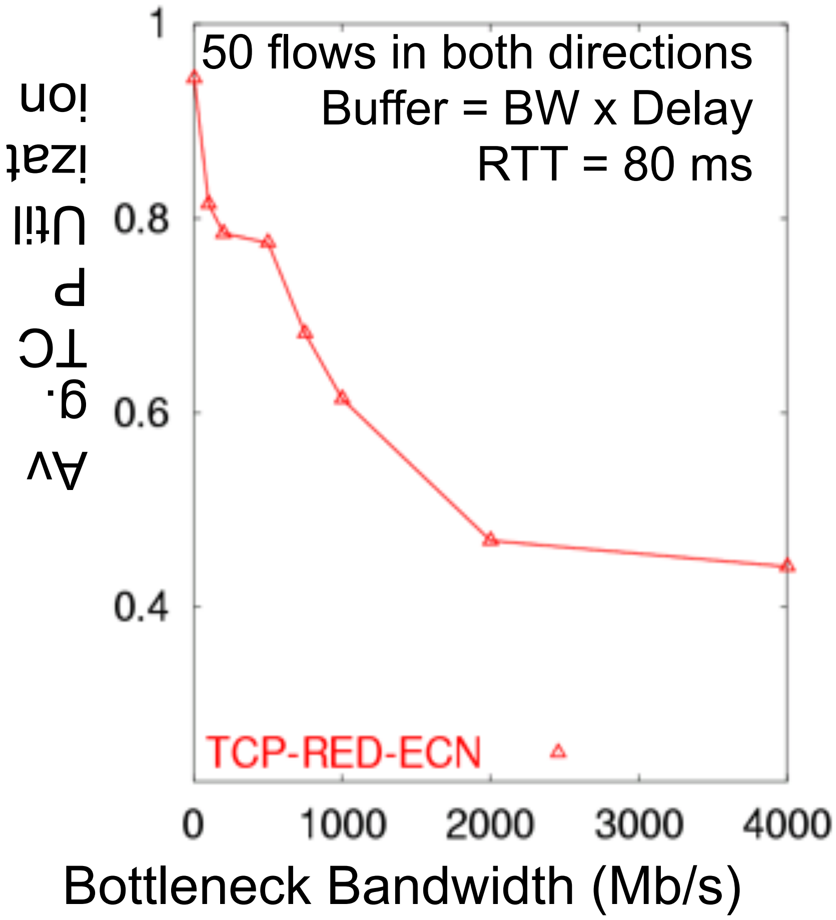
- Why does TCP perform poorly?
  - Slow start and additive increase are slow to converge
  - TCP is ACK clocked
    - i.e. TCP can only react as quickly as ACKs are received
    - Large RTT → ACKs are delayed → TCP is slow to react

Poor Performance of TCP Reno CC



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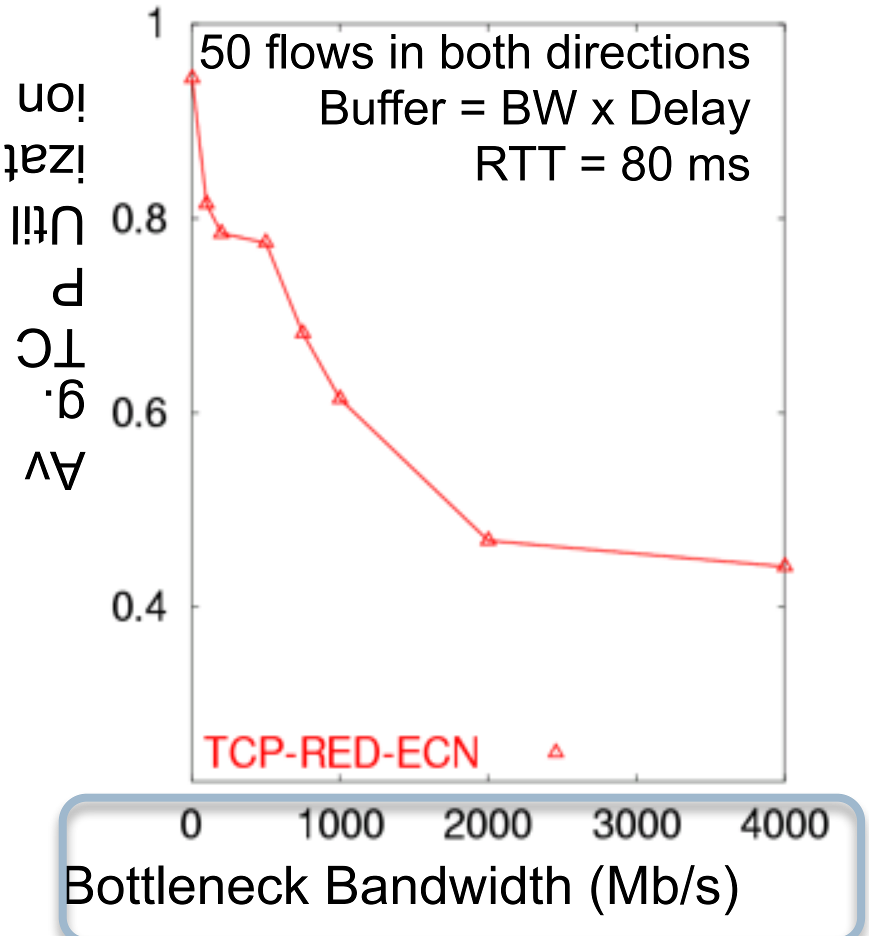

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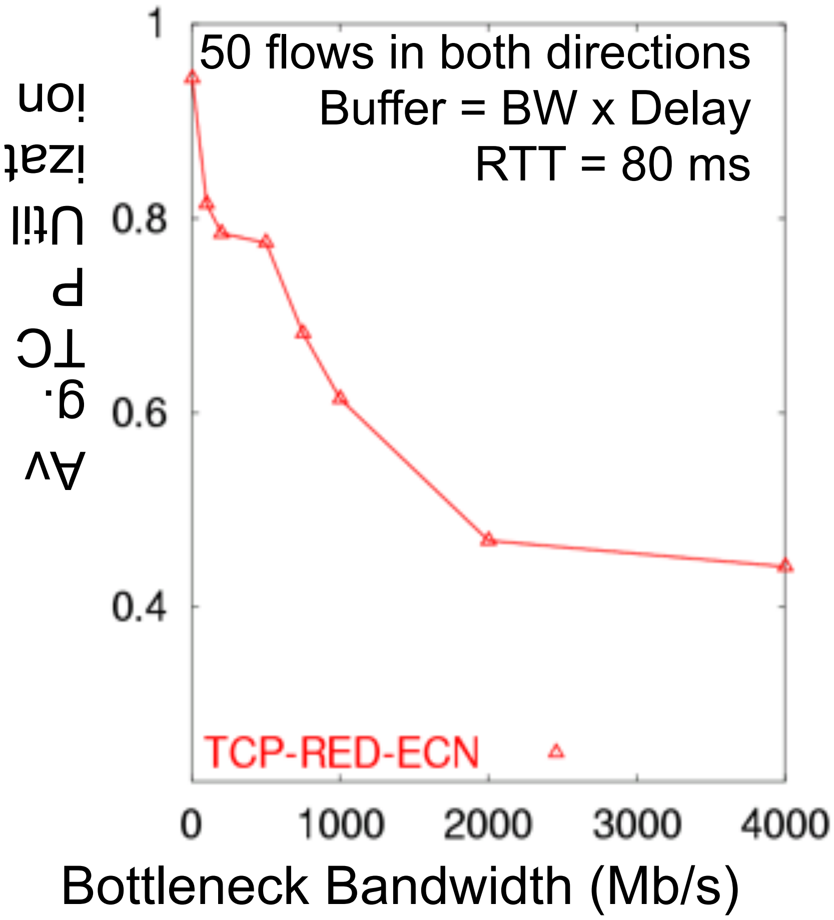

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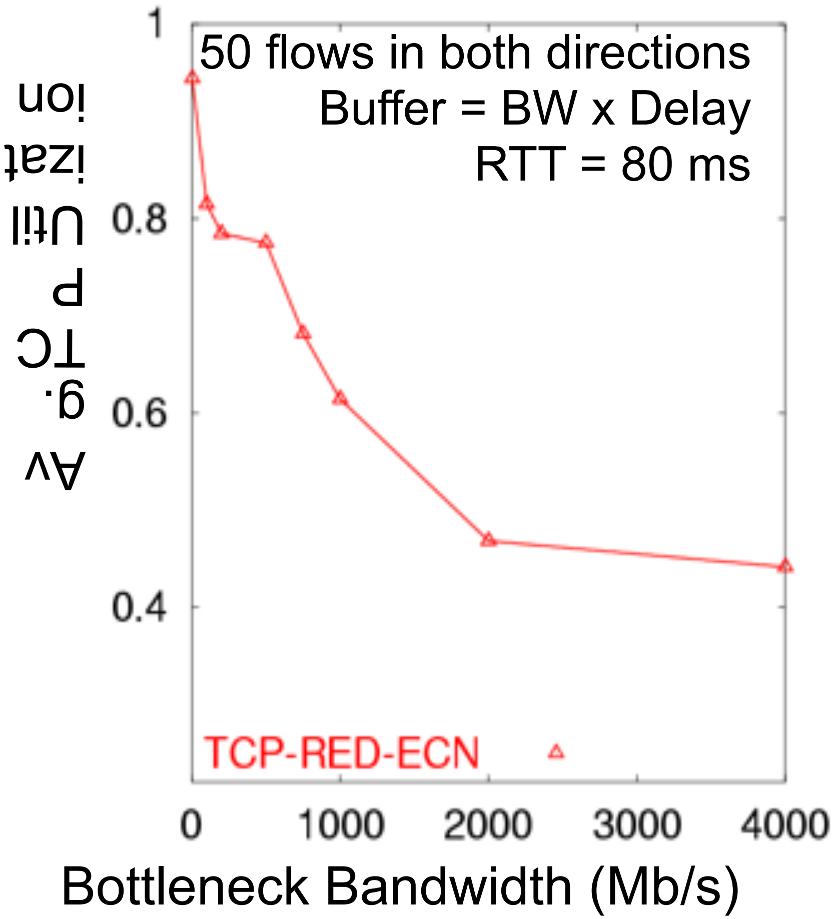

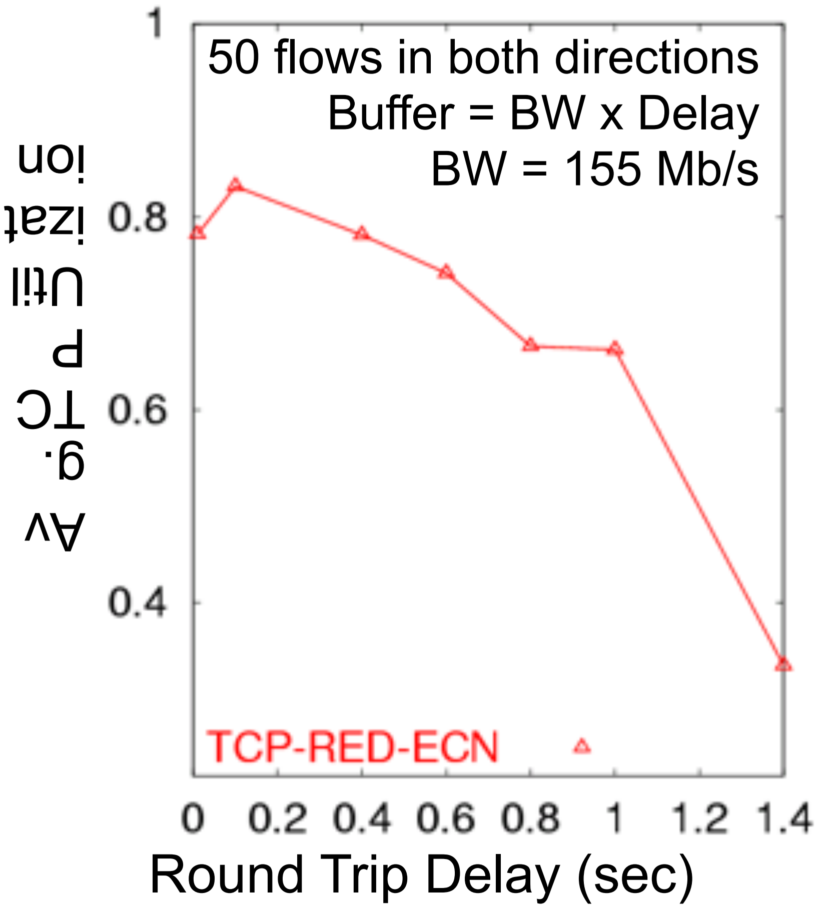

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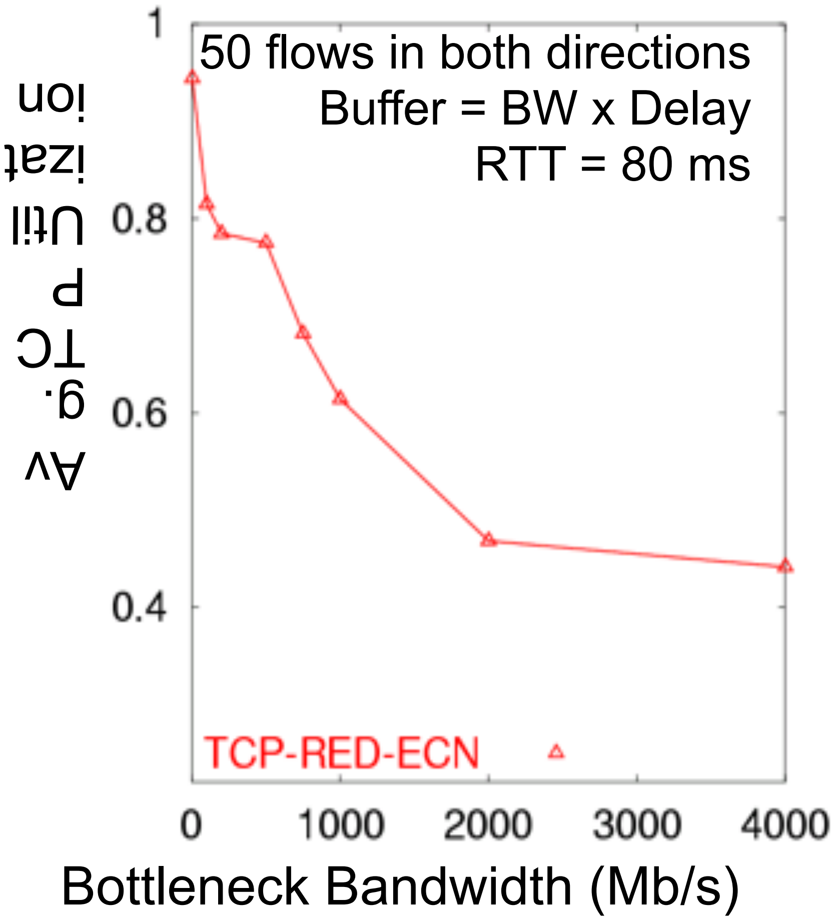

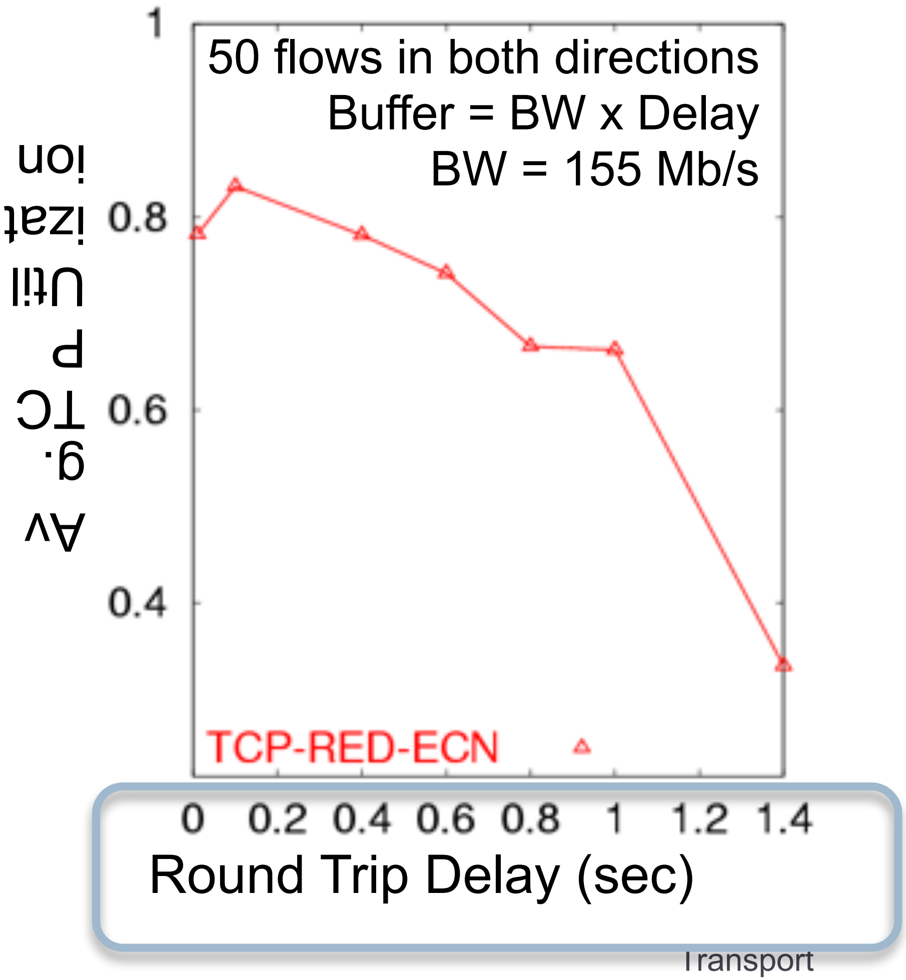

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- Simple implementation

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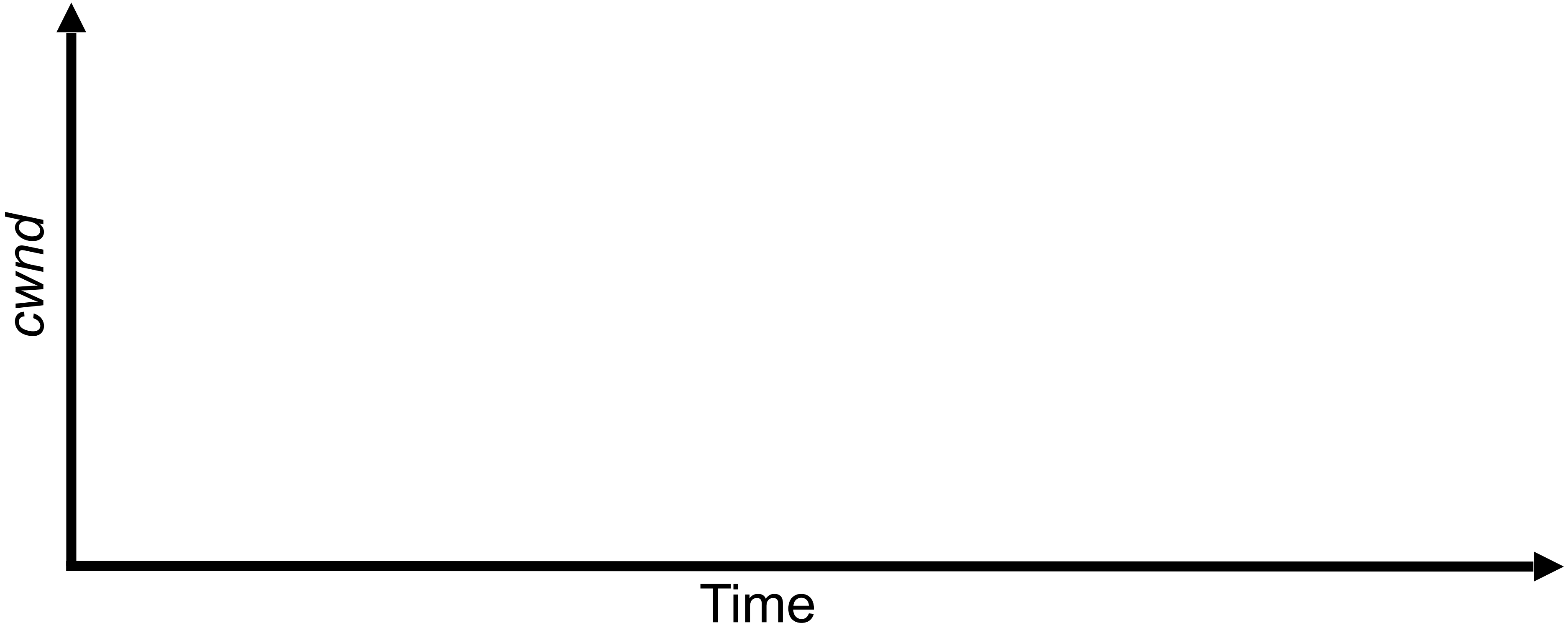
- *wnd* = min(*cwnd* + *dwnd*, *adv_wnd*)
  - *cwnd* is controlled by AIMD
  - *dwnd* is the delay window
- Rules for adjusting *dwnd:*
  - If RTT is increasing, decrease *dwnd* (*dwnd* >= 0)
  - If RTT is decreasing, increase *dwnd*
  - Increase/decrease are proportional to the rate of change

Compound TCP Example



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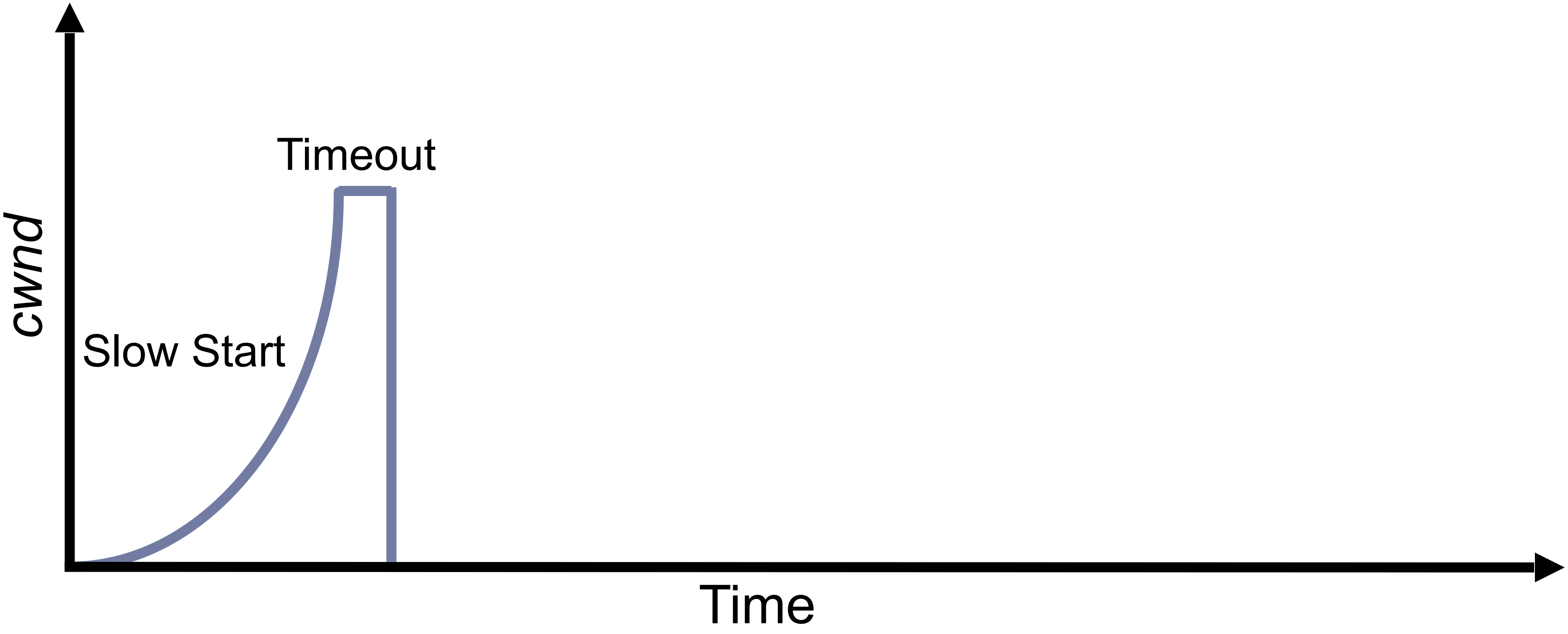

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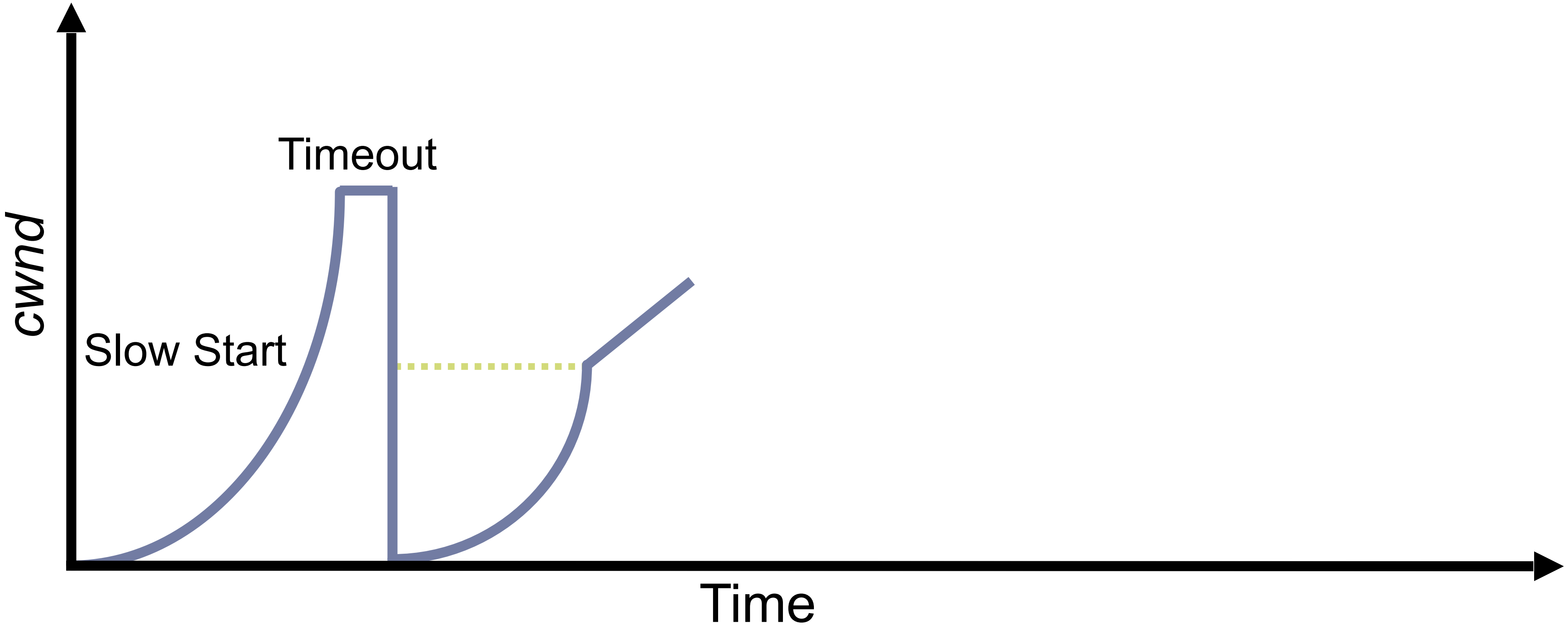

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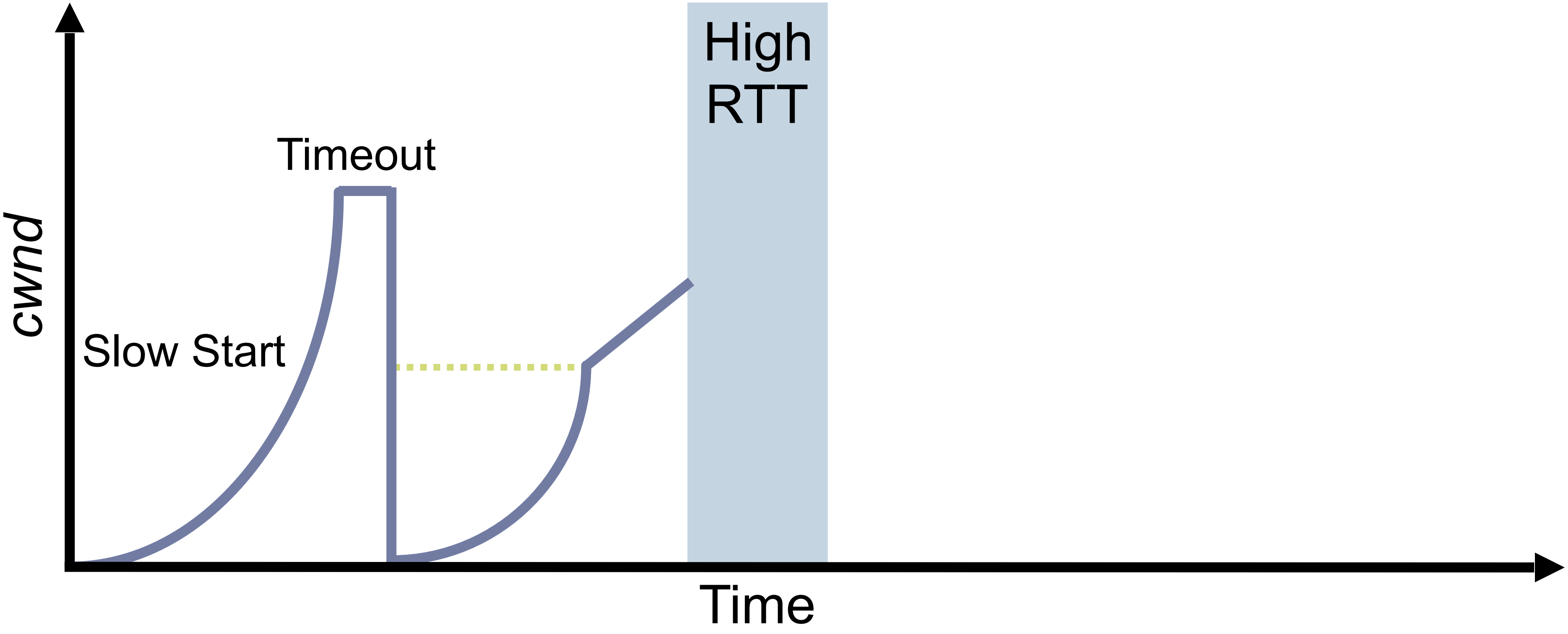

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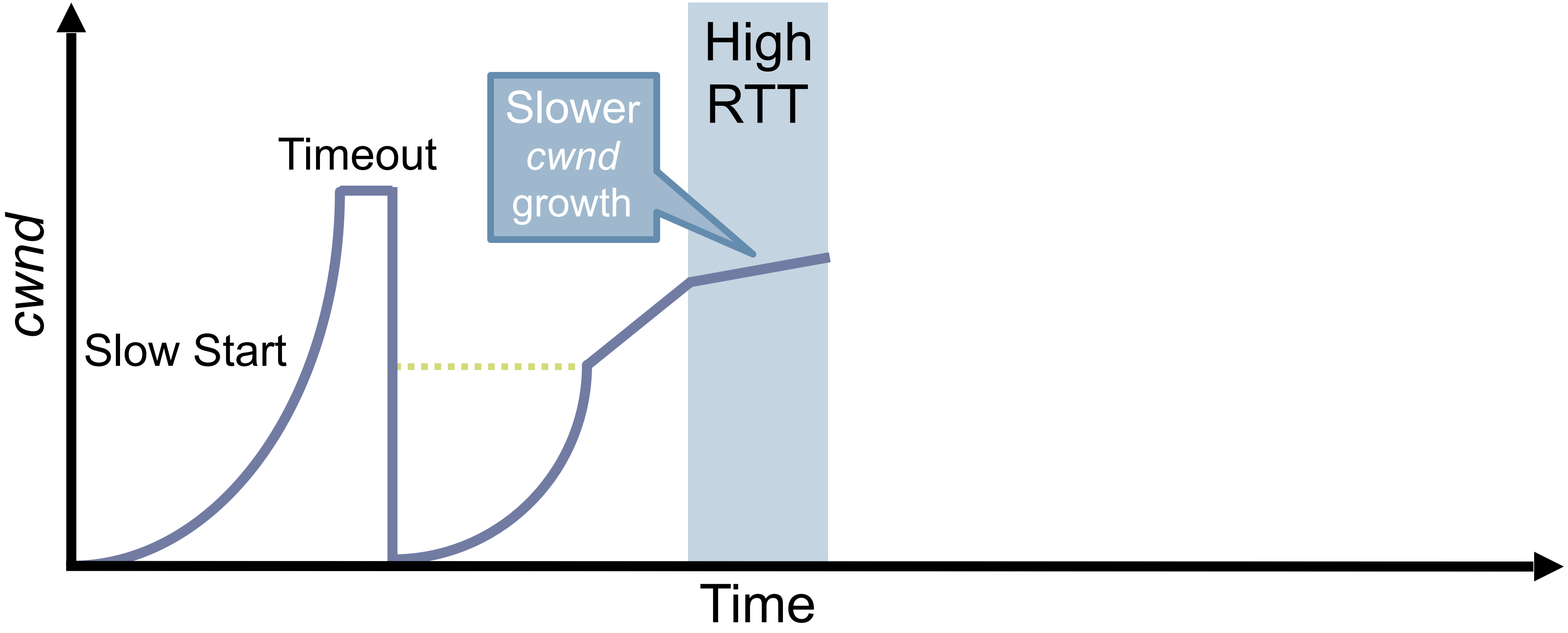

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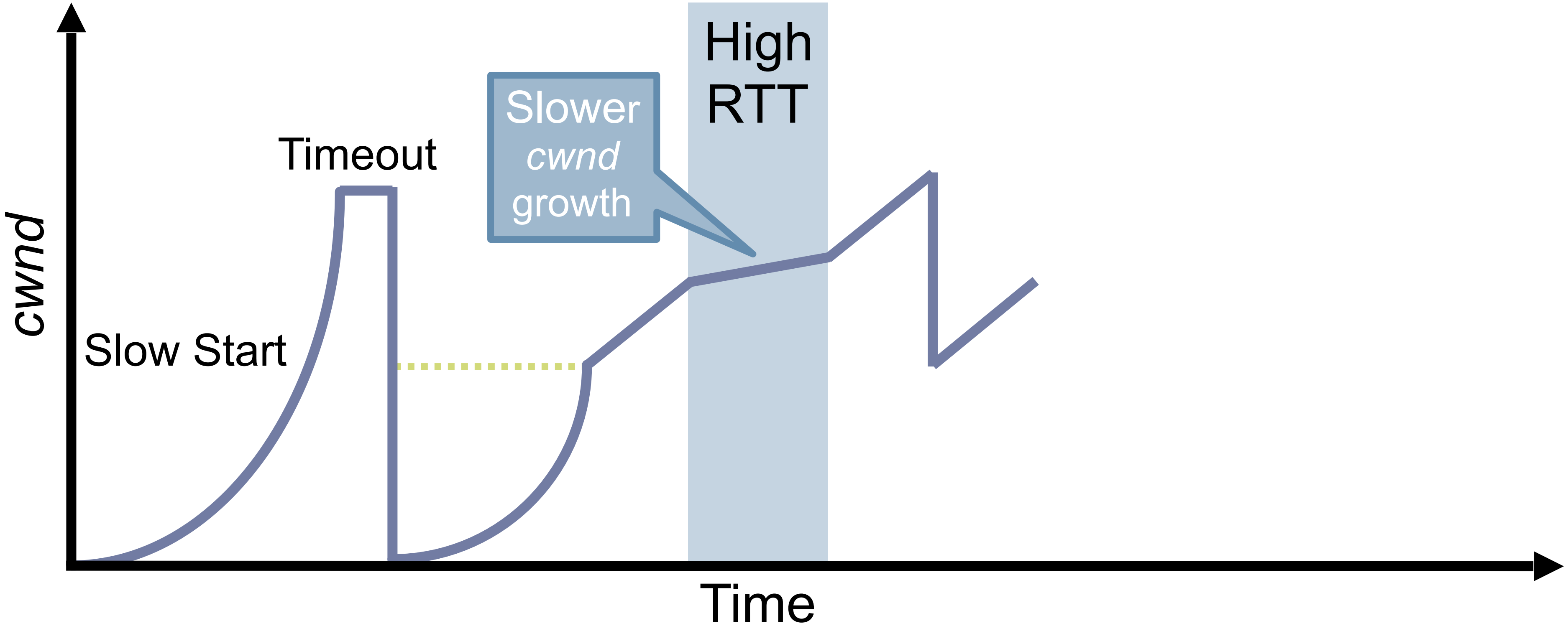

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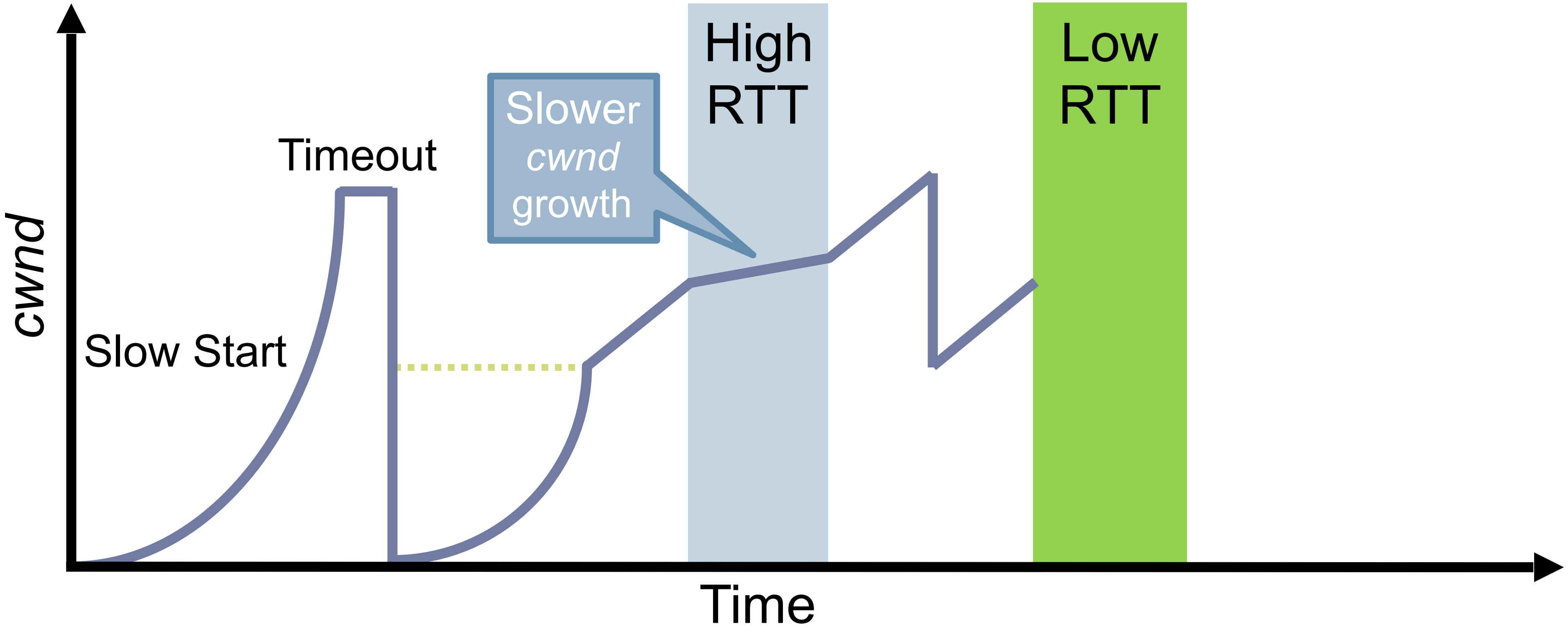

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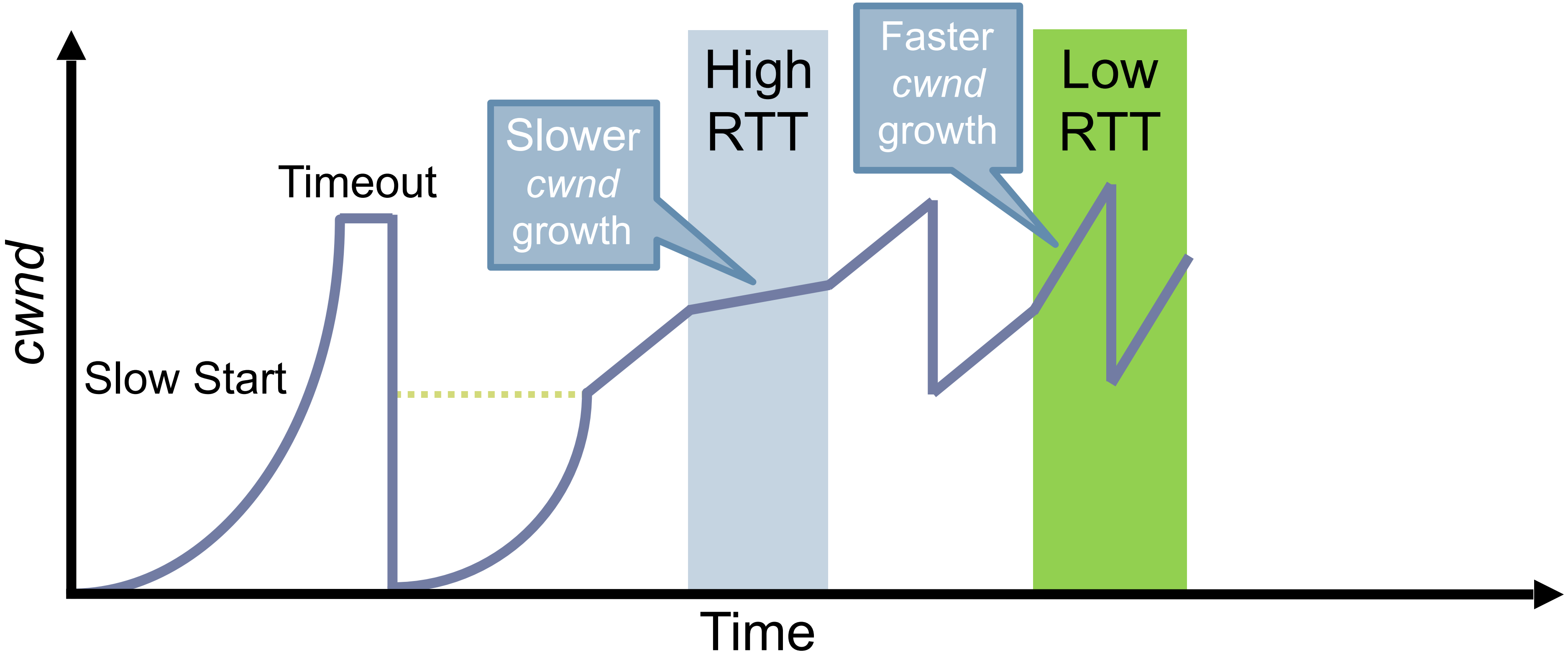

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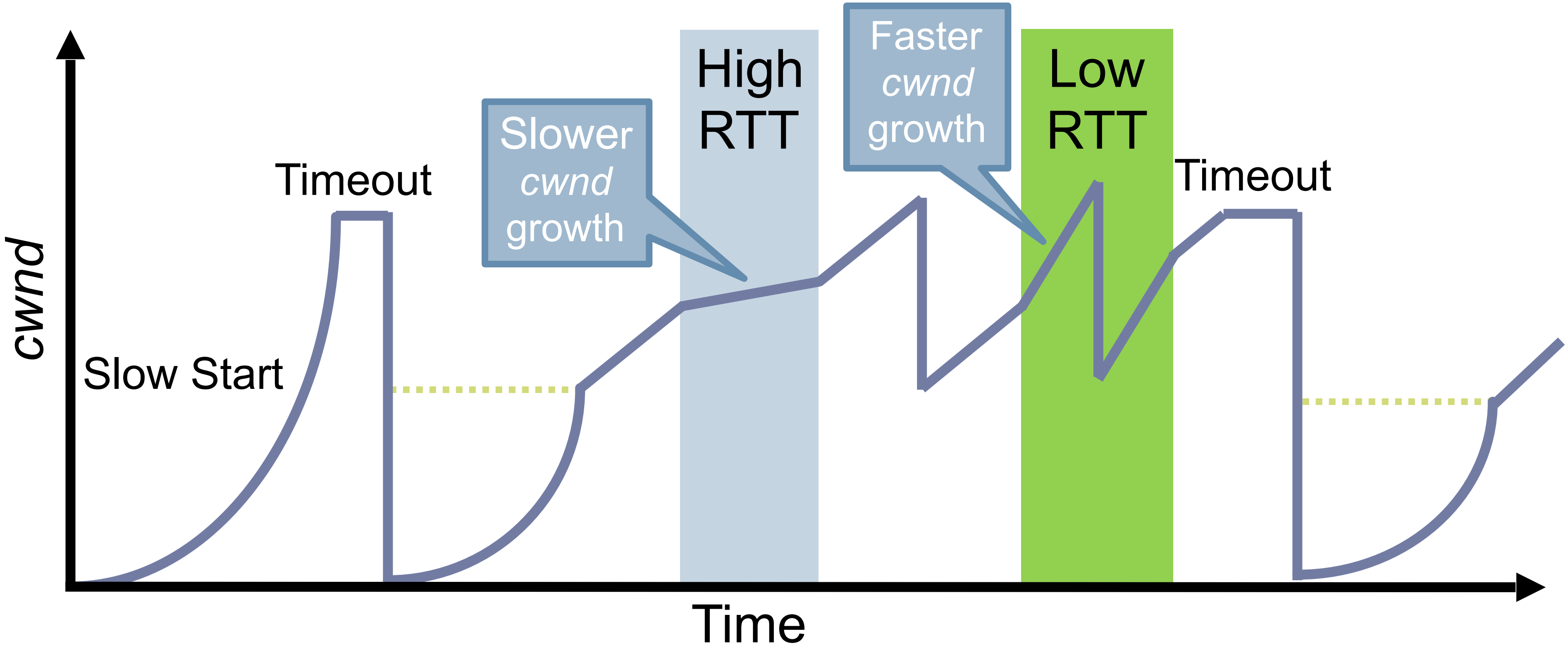

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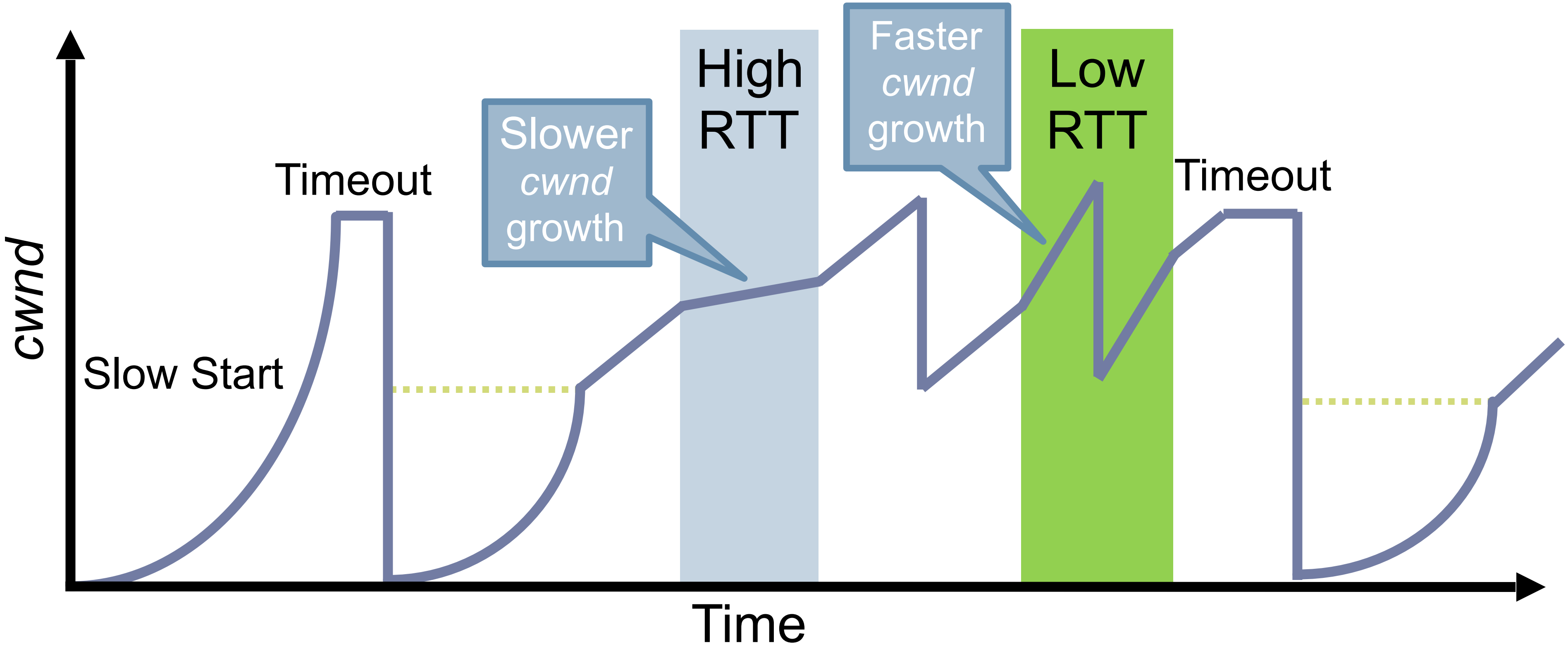

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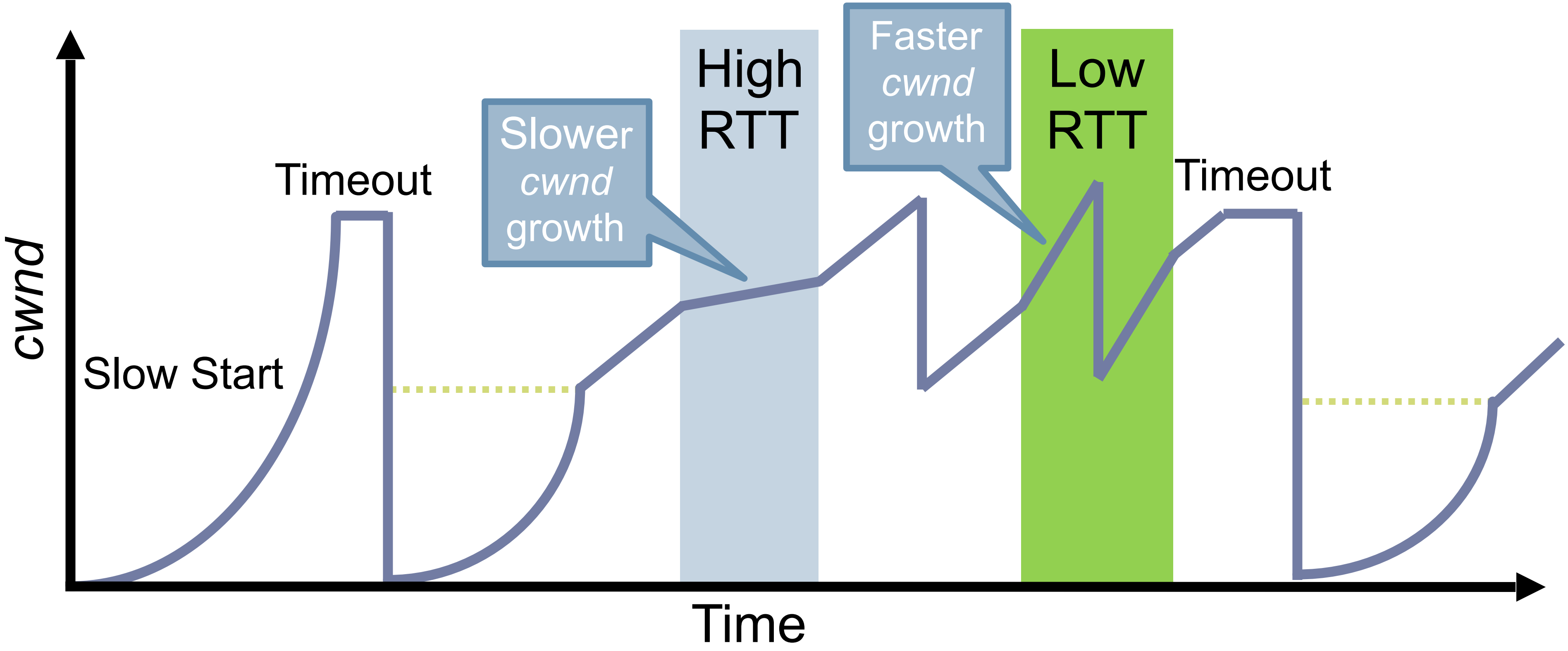

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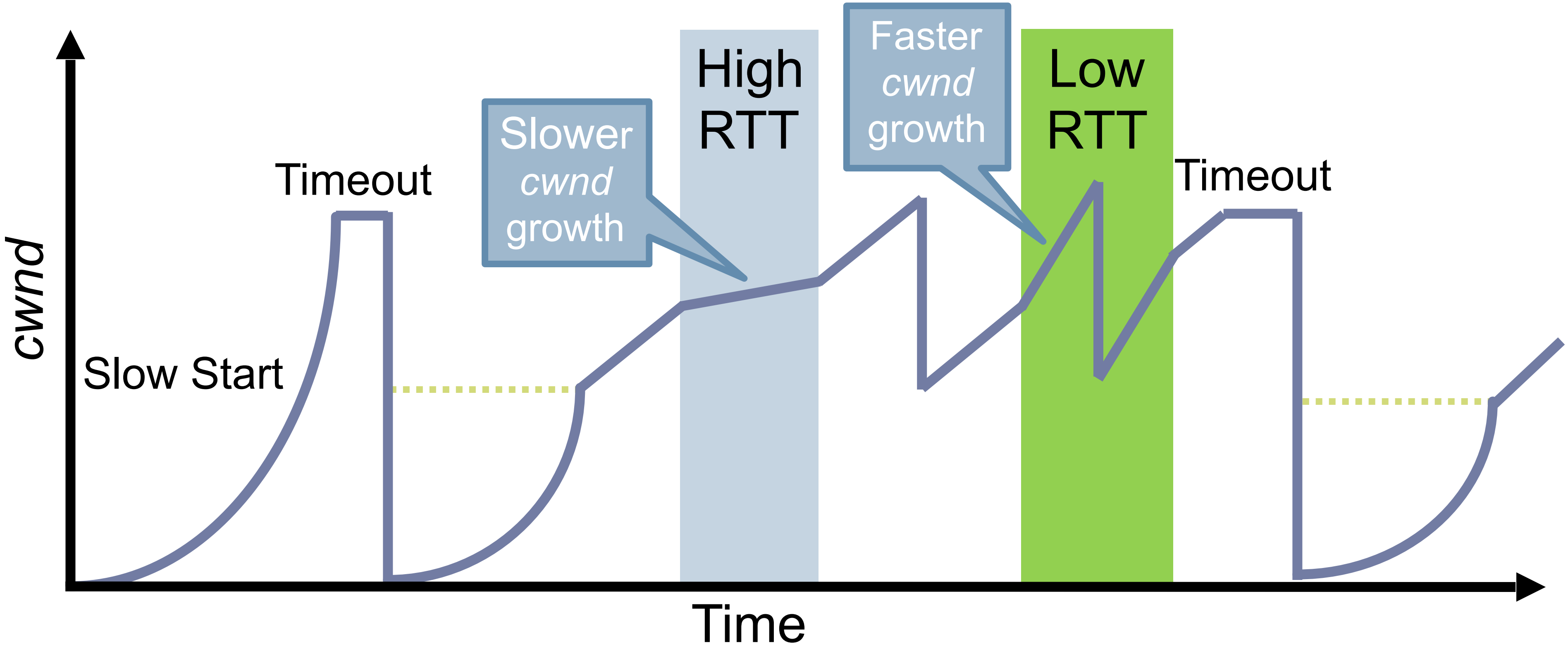


- Aggressiveness corresponds to changes in RTT
- Advantages: fast ramp up, more fair to flows with different RTTs
- Disadvantage: must estimate RTT, which is very challenging

TCP CUBIC Implementation

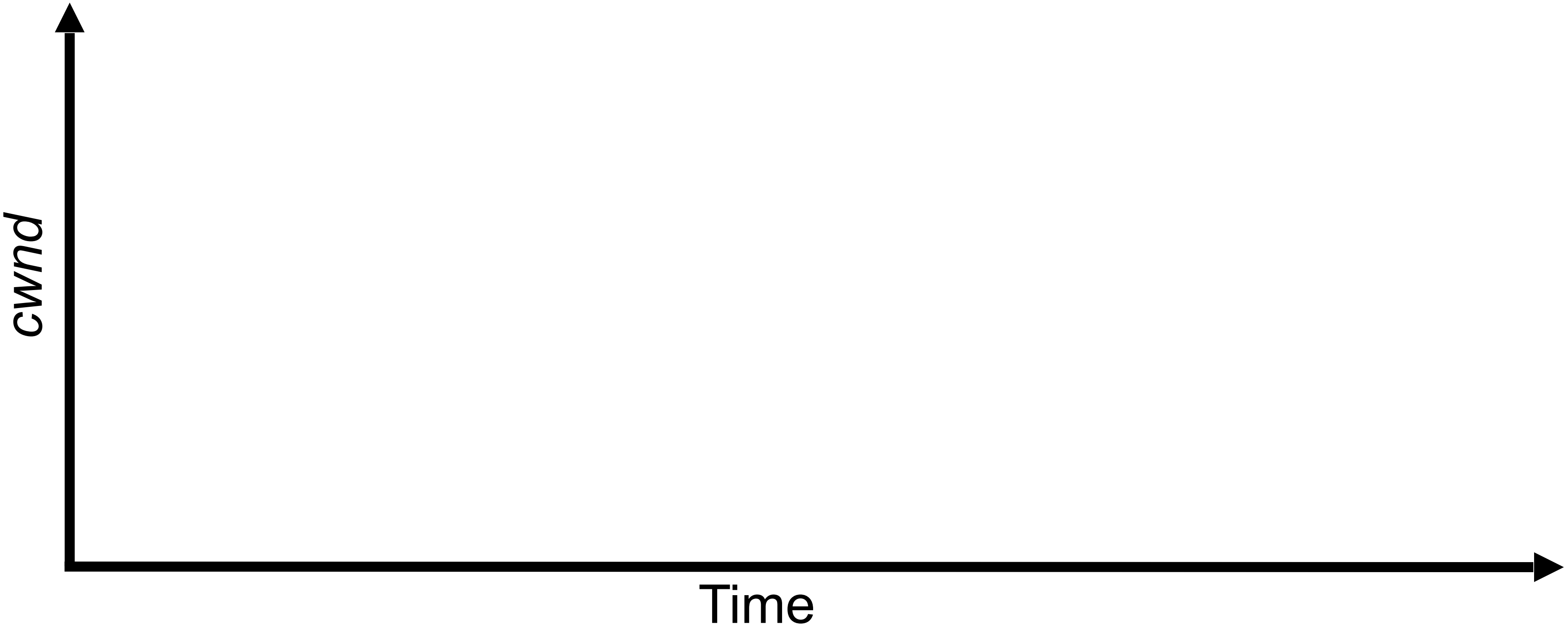
- Default TCP implementation in Linux
- Replace AIMD with cubic function

$$W(t) = C(t - K)^3 + W_{max}$$

$$K = \sqrt[3]{\frac{W_{max}\beta}{C}}$$

  - B → a constant fraction for multiplicative increase
  - t → time since last packet drop
  - W_{max} → cwnd when last packet dropped
  - C → scaling constant

TCP CUBIC Example



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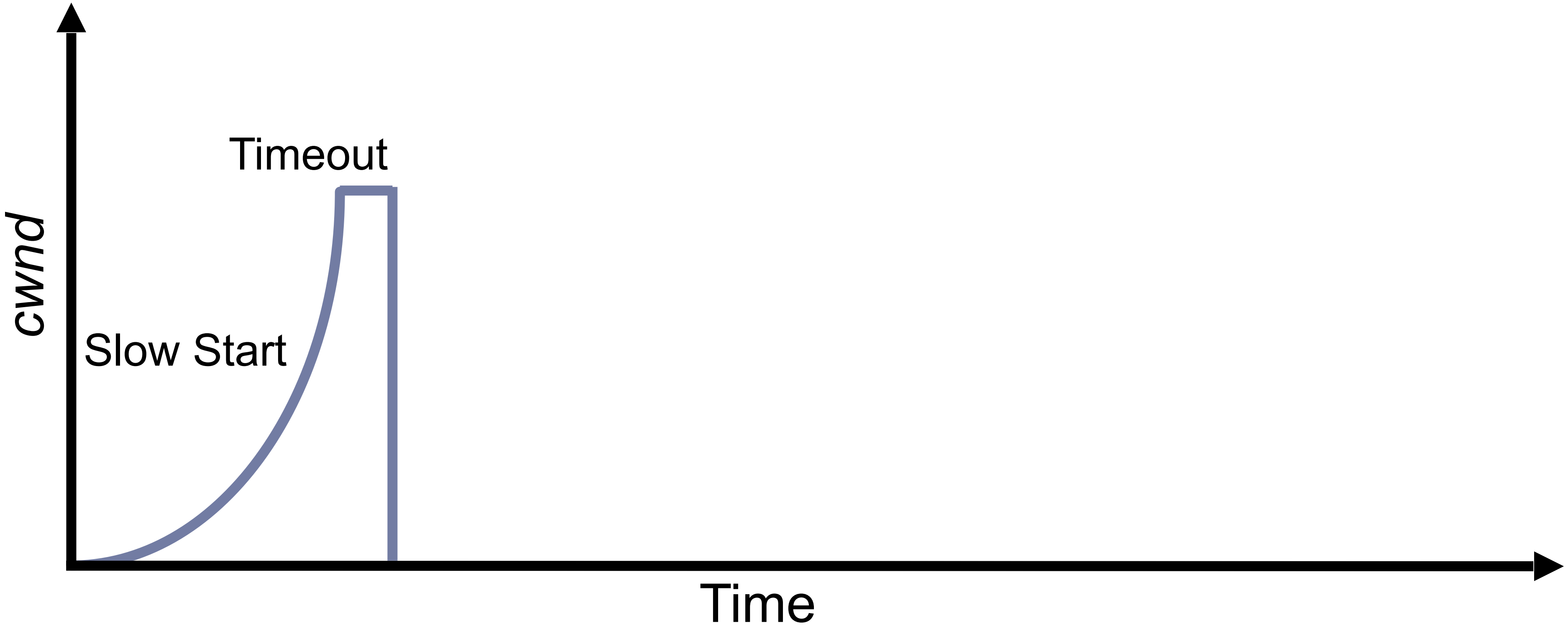

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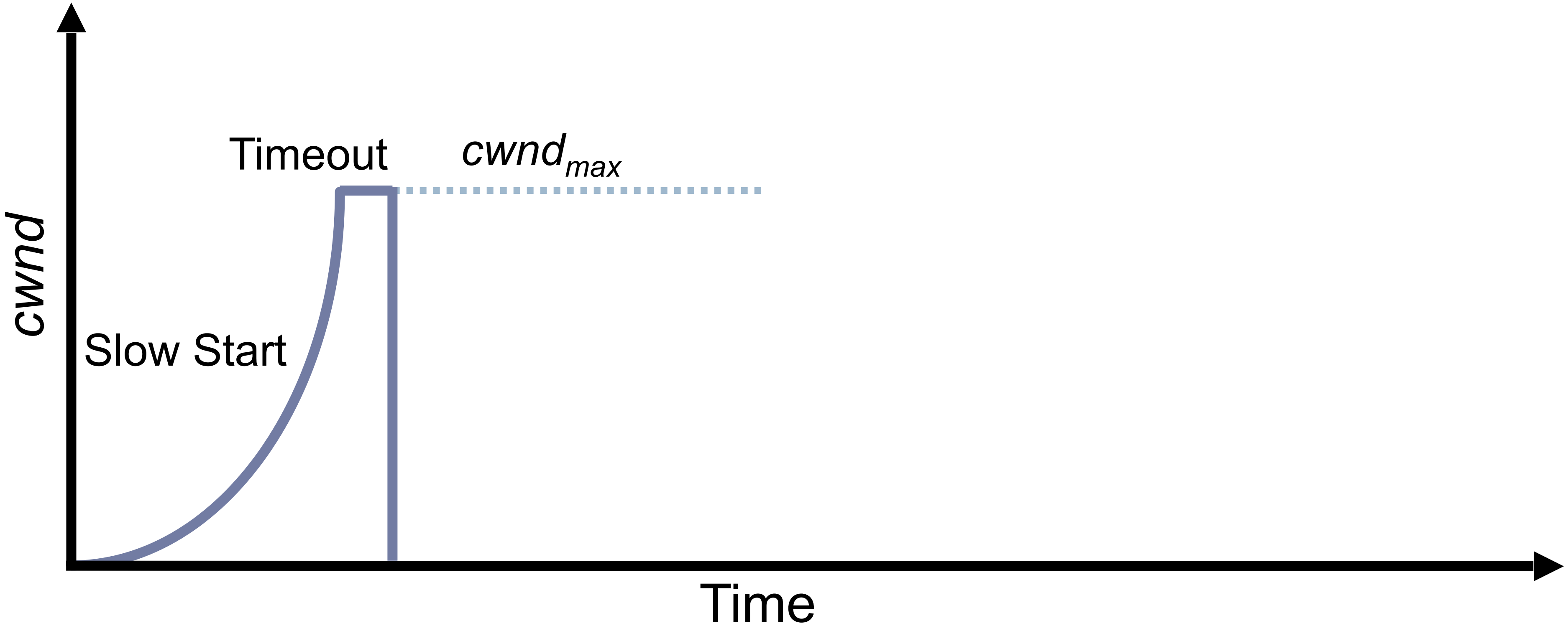

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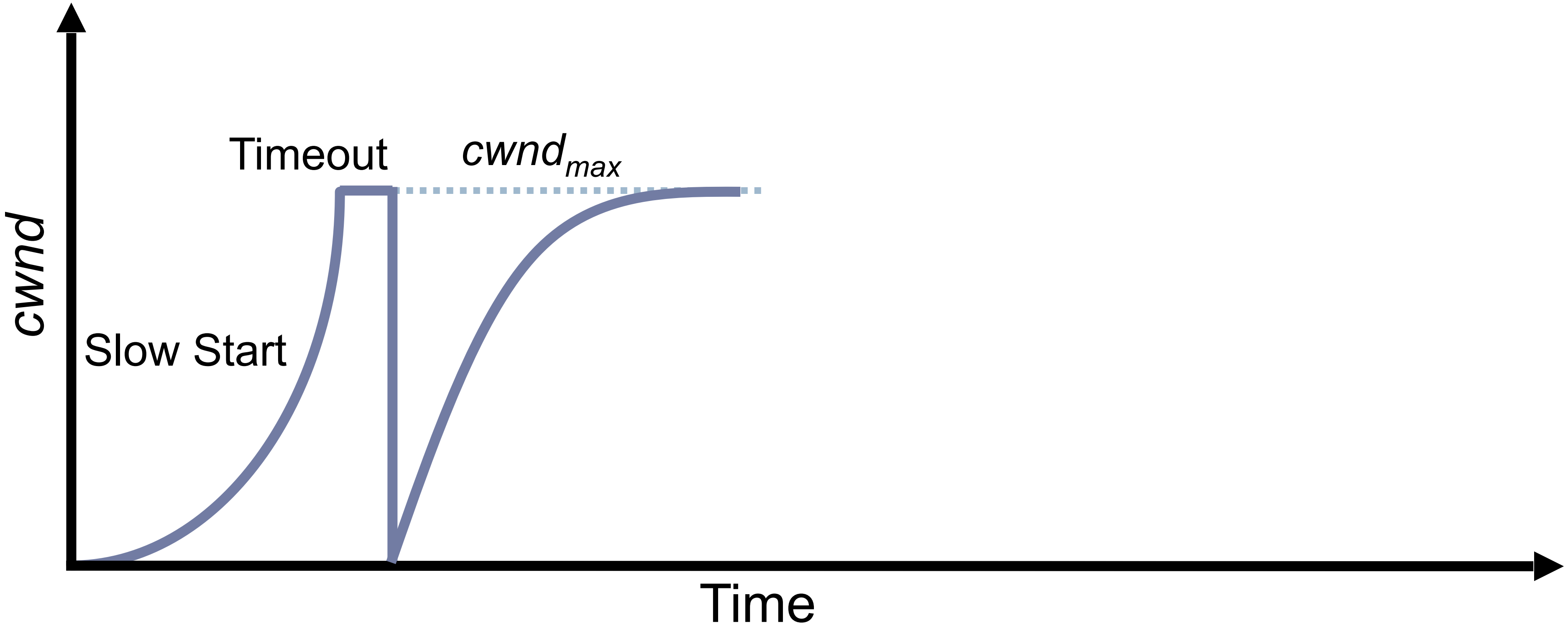

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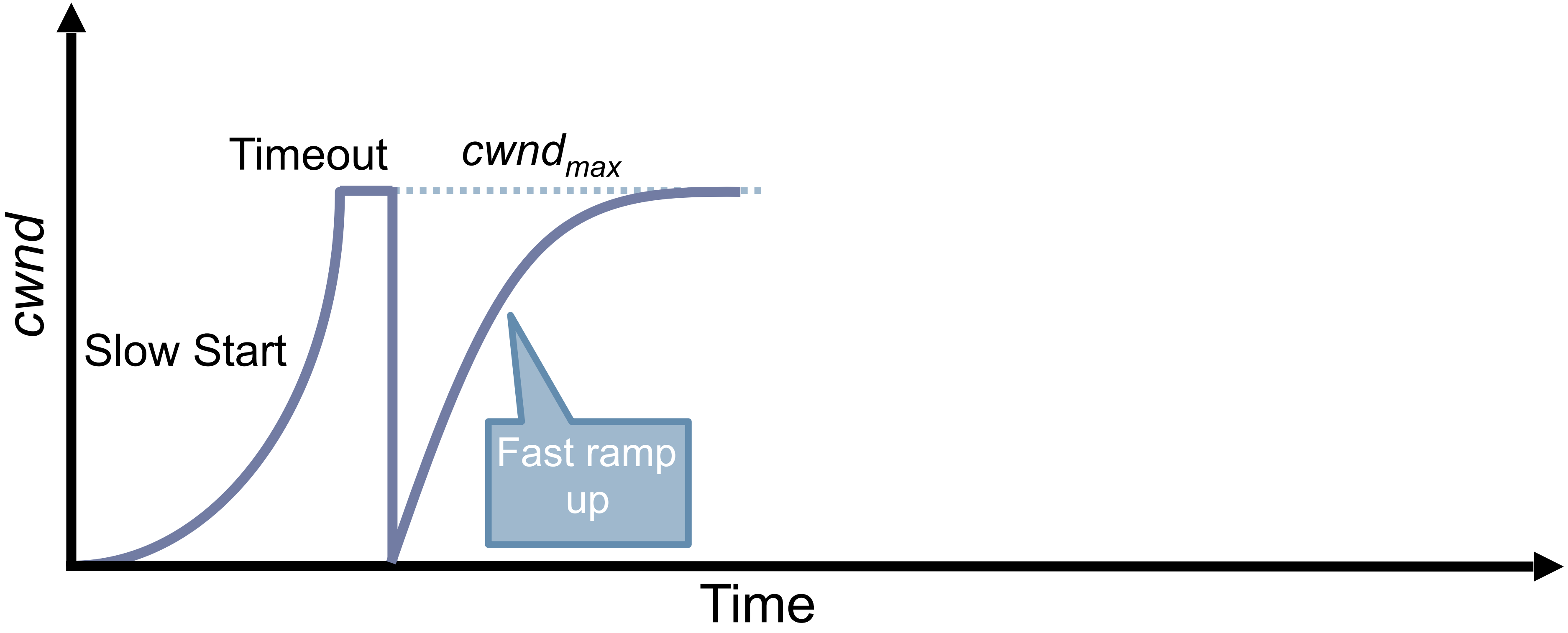

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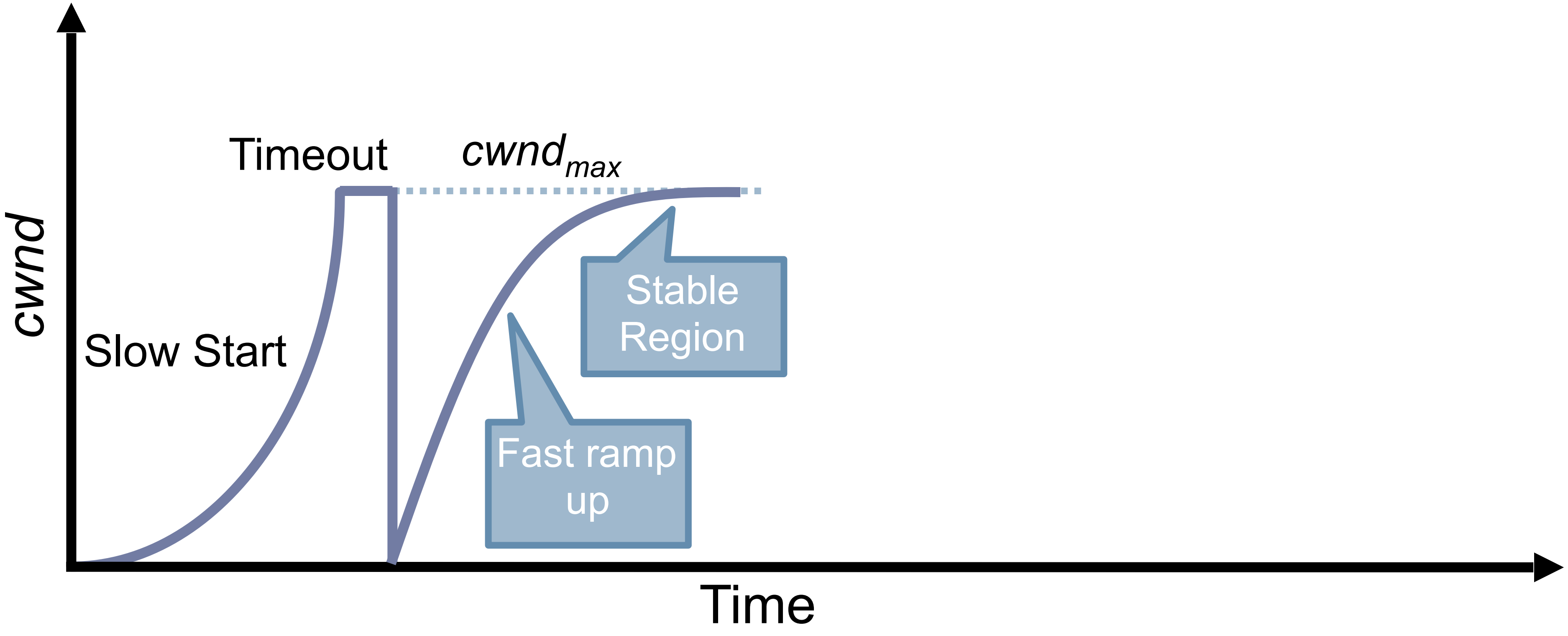

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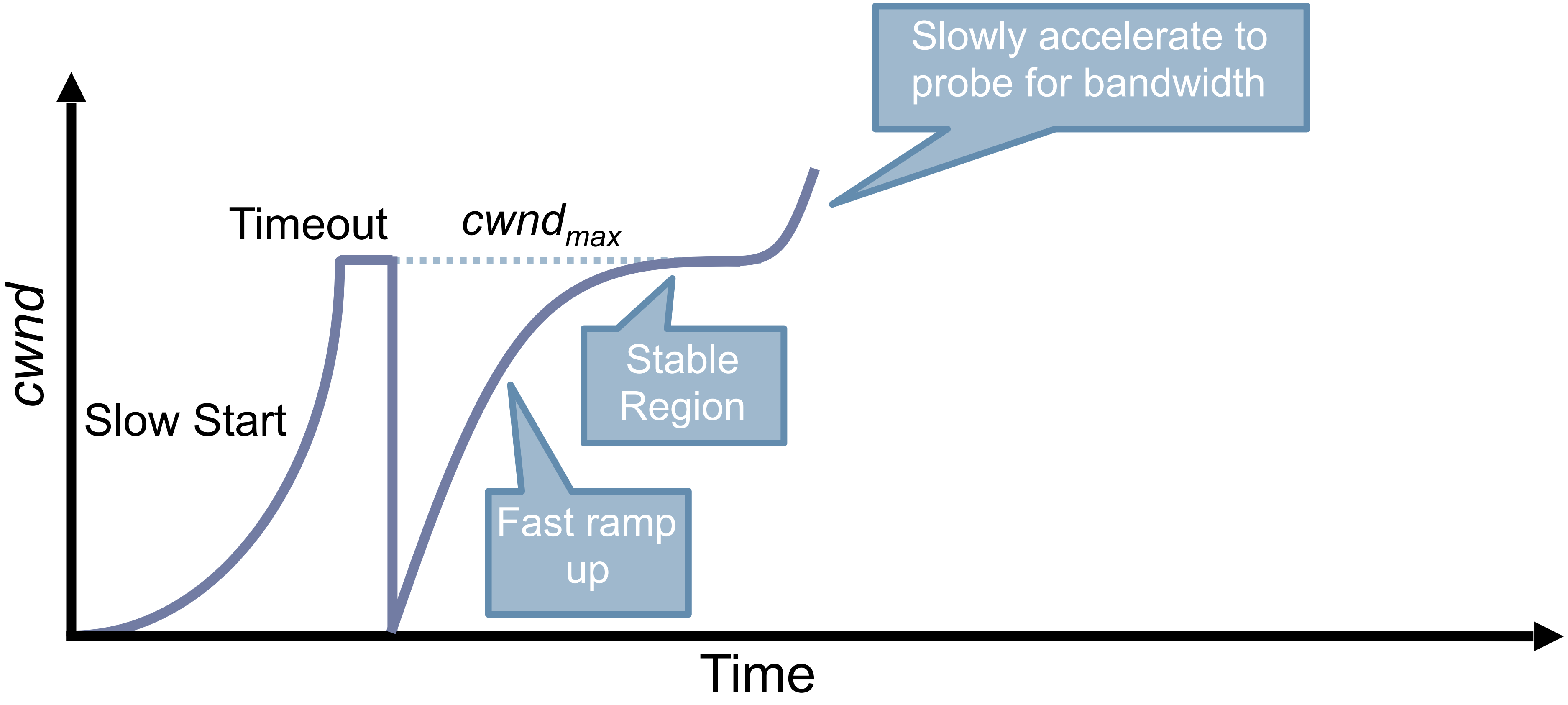

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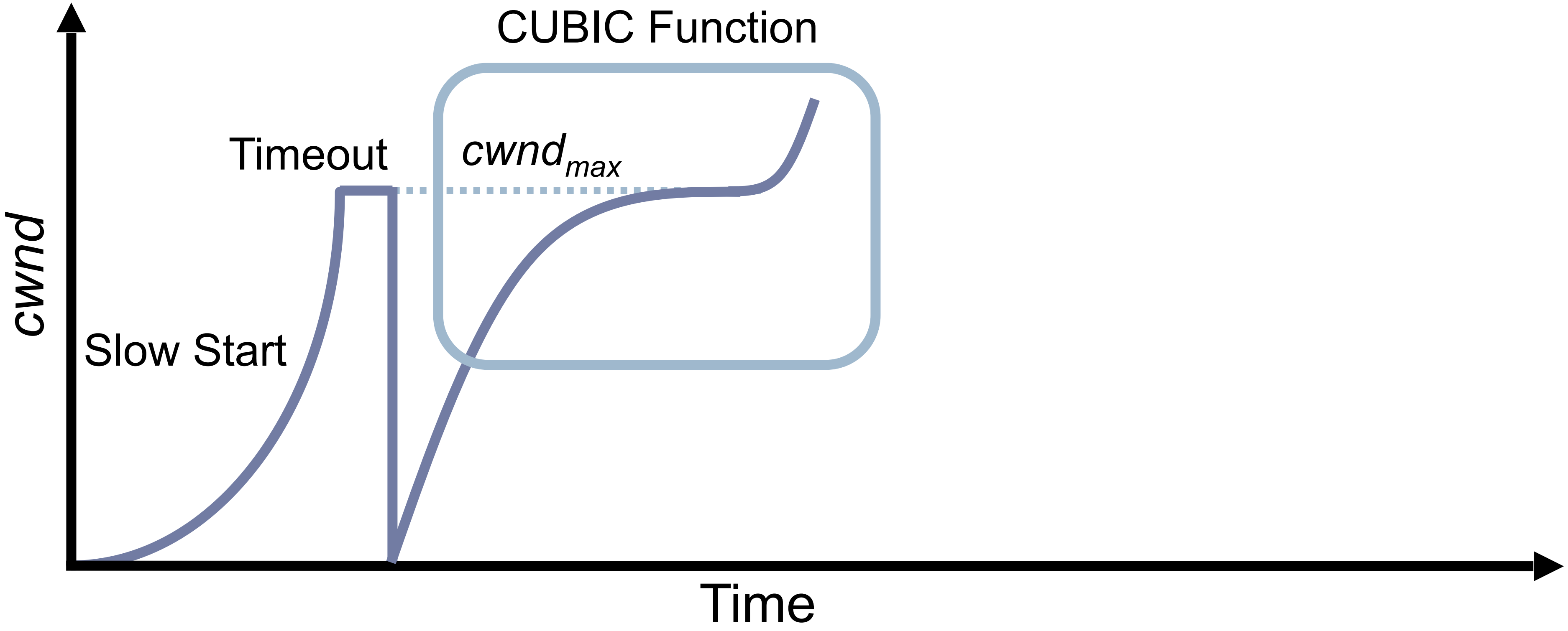

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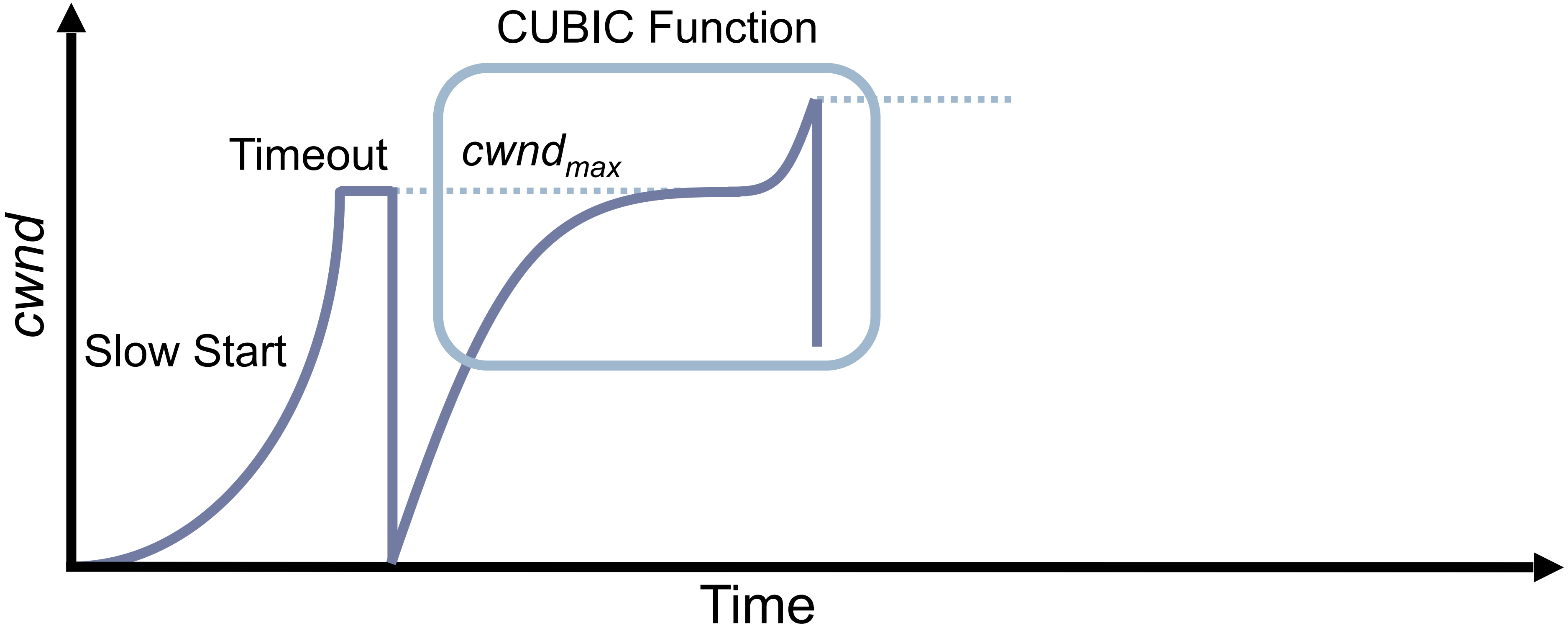

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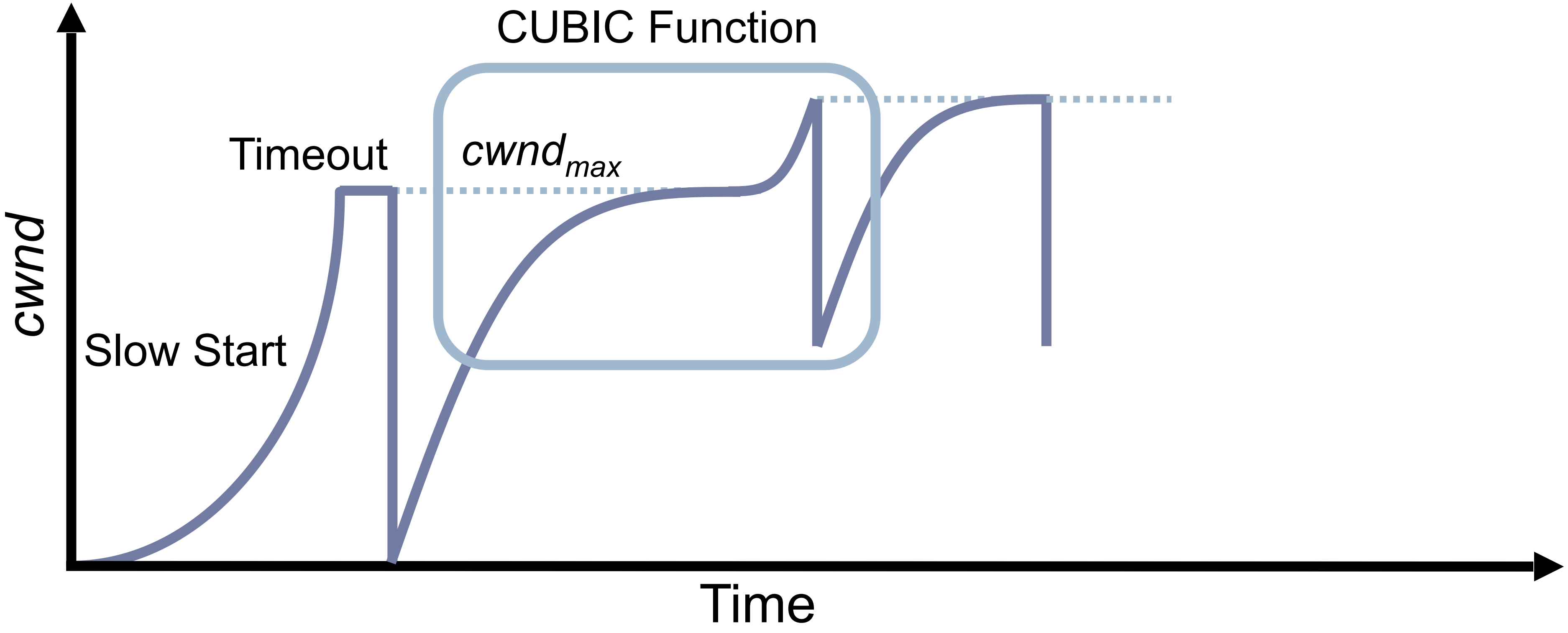

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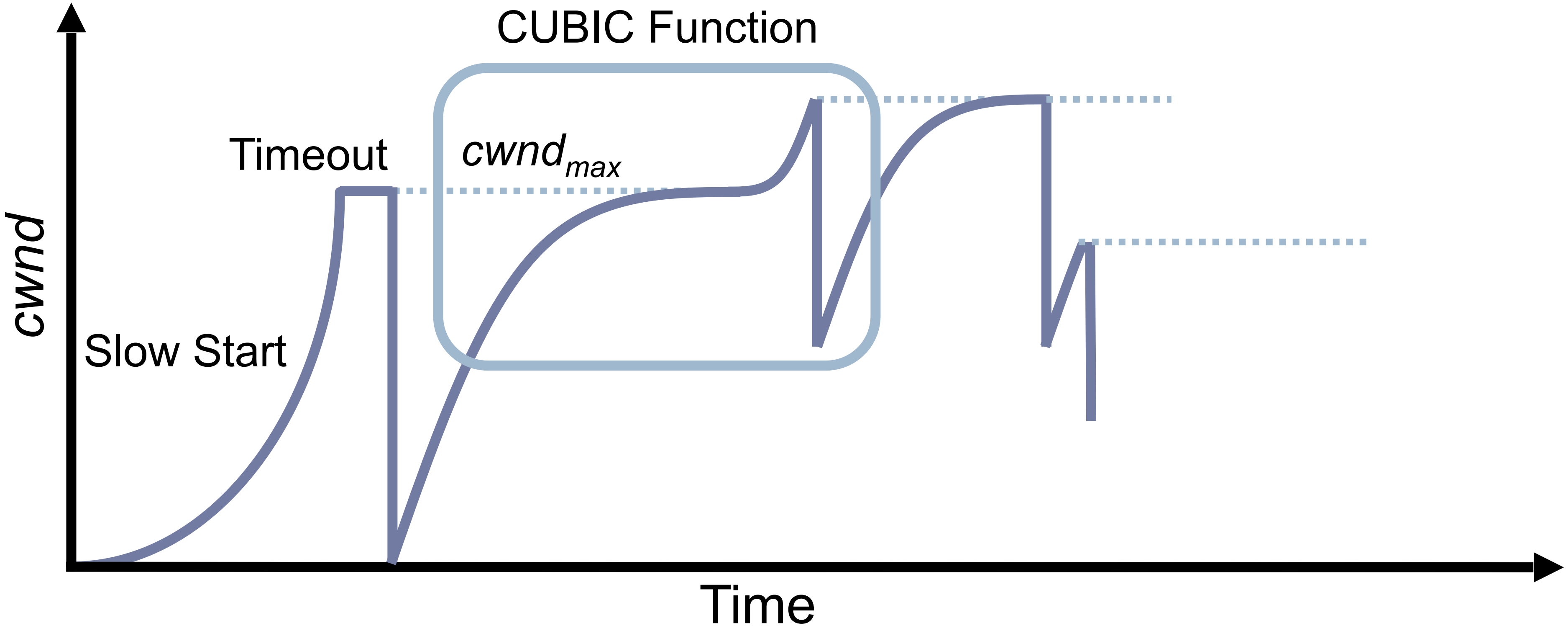

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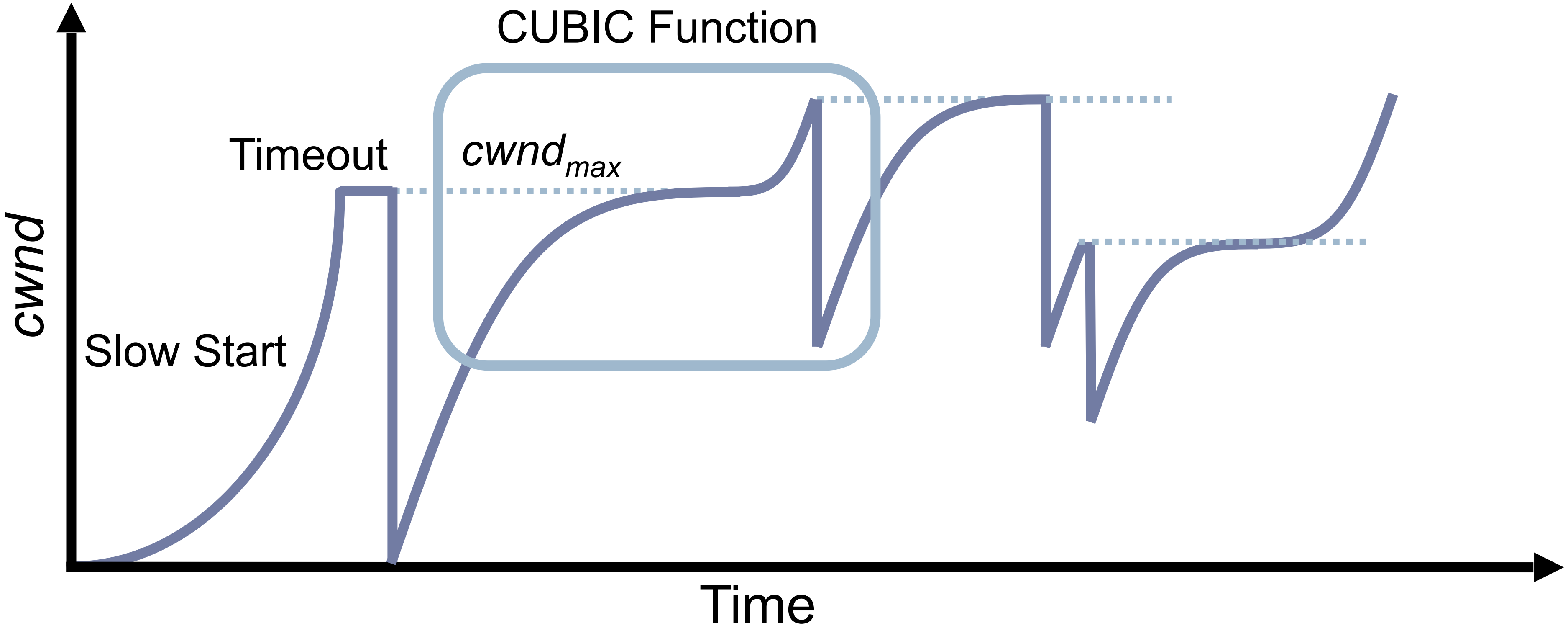

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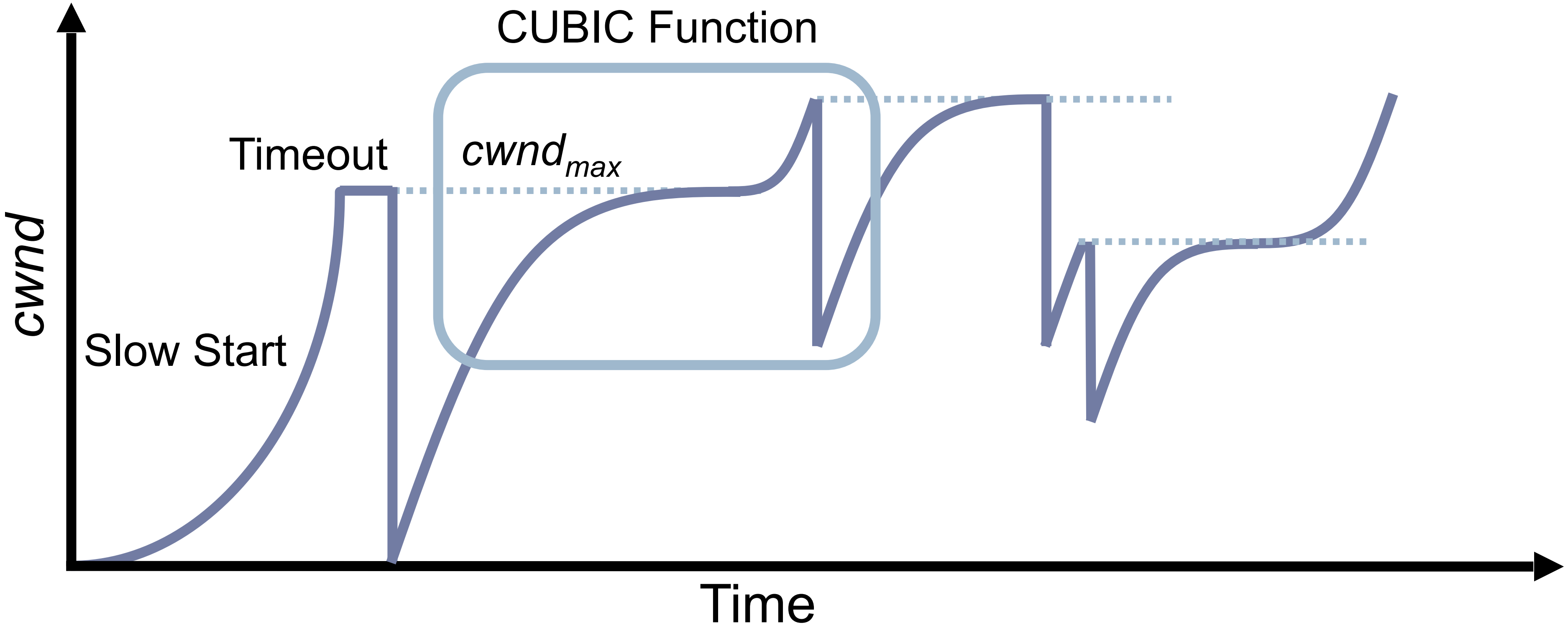

- Less wasted bandwidth due to fast ramp up

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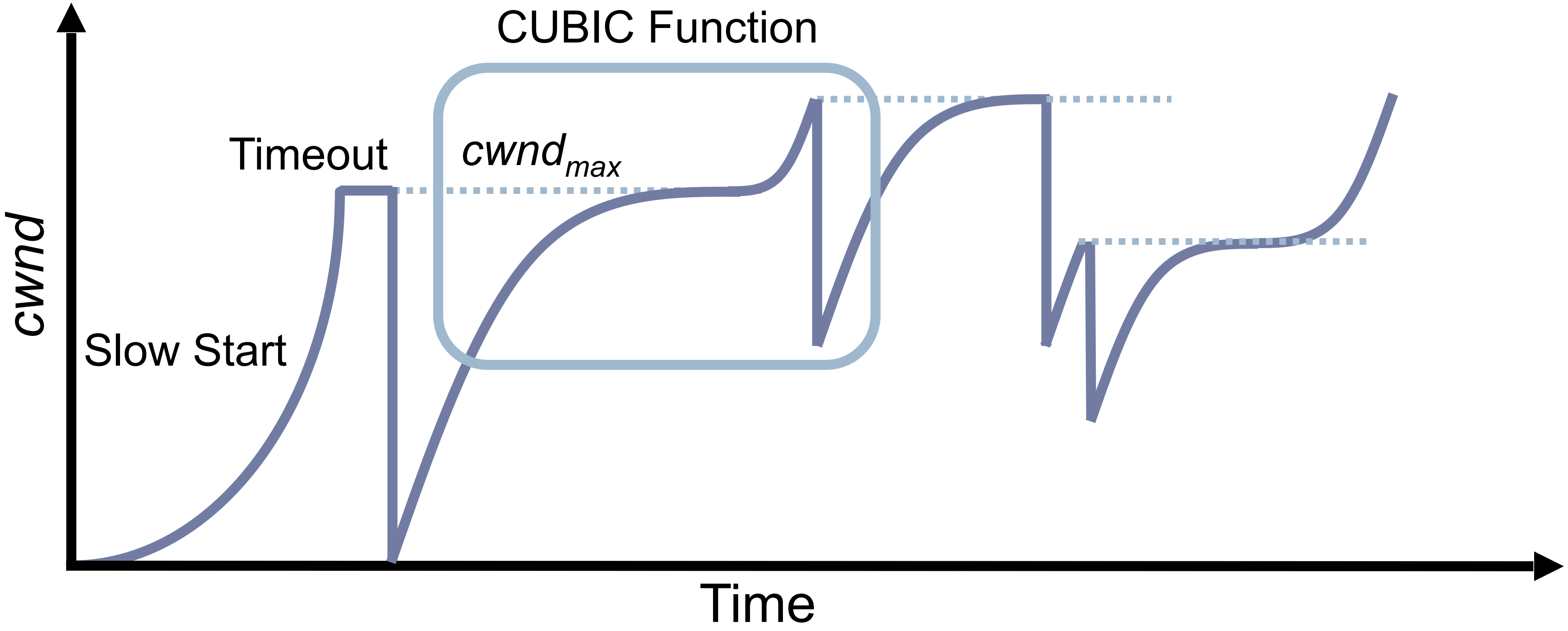
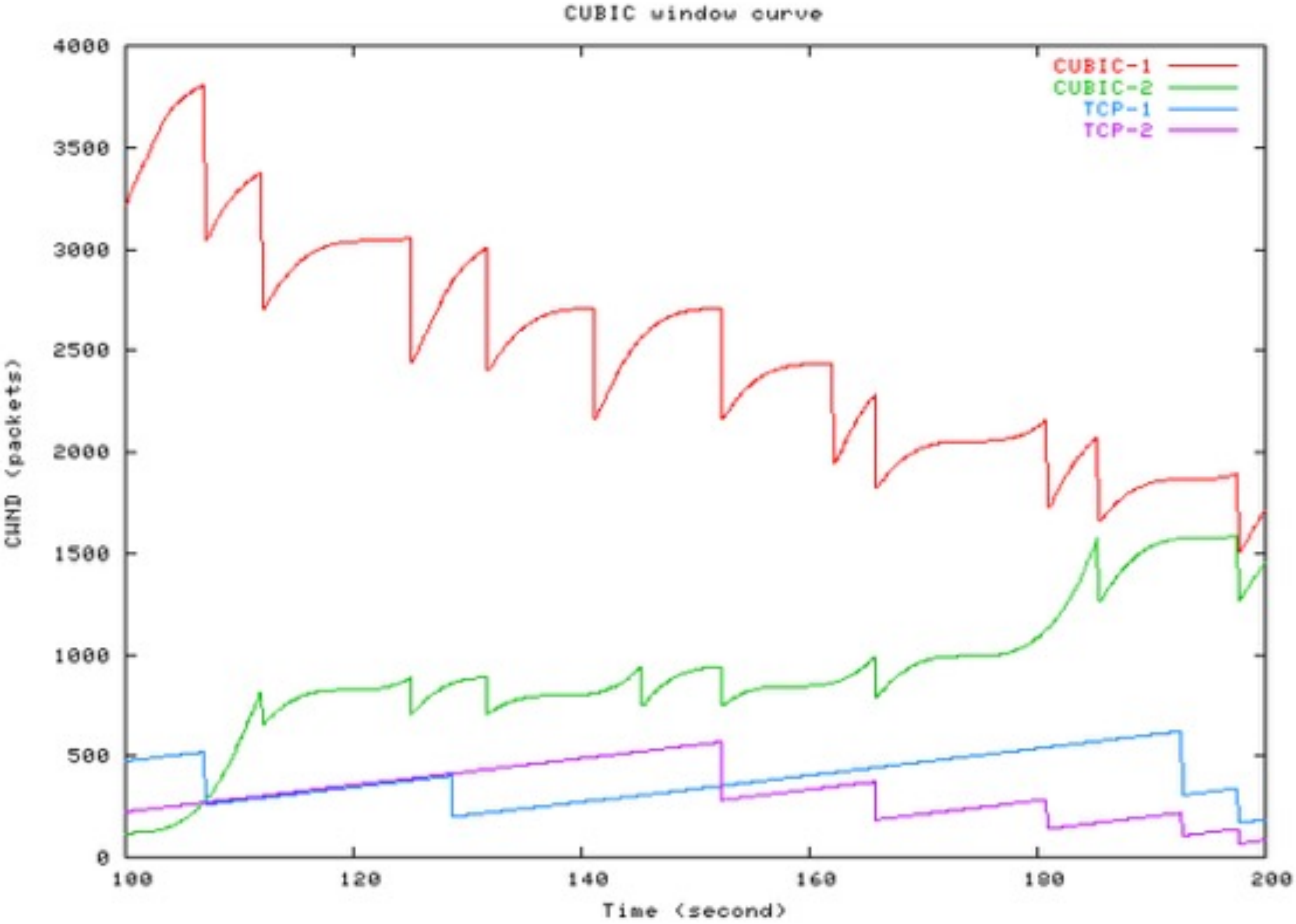


- Less wasted bandwidth due to fast ramp up
- Stable region and slow acceleration help maintain fairness
  - Fast ramp up is more aggressive than additive increase
  - To be fair to Tahoe/Reno, CUBIC needs to be less aggressive

Simulations of CUBIC Flows



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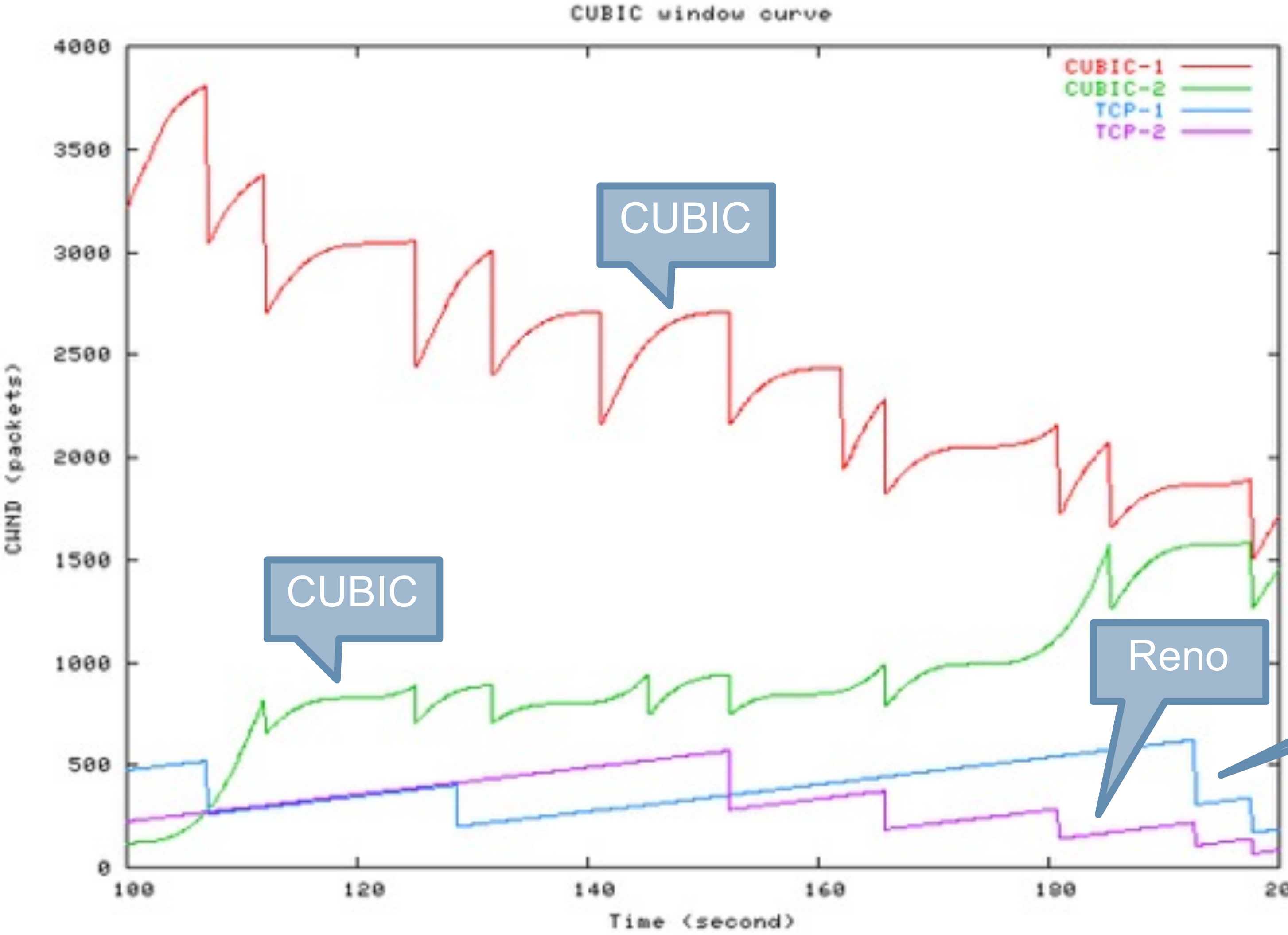

CUBIC

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- TCP assumes all flows employ TCP-like congestion control
  - TCP-friendly or TCP-compatible
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- TCP assumes all flows employ TCP-like congestion control
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- If new congestion control algorithms are developed, they must be TCP-friendly
- Be wary of unforeseen interactions
  - Variants work well with others like themselves
  - Different variants competing for resources may trigger unfair, pathological behavior

TCP Perspectives

- Cerf/Kahn
  - Provide flow control
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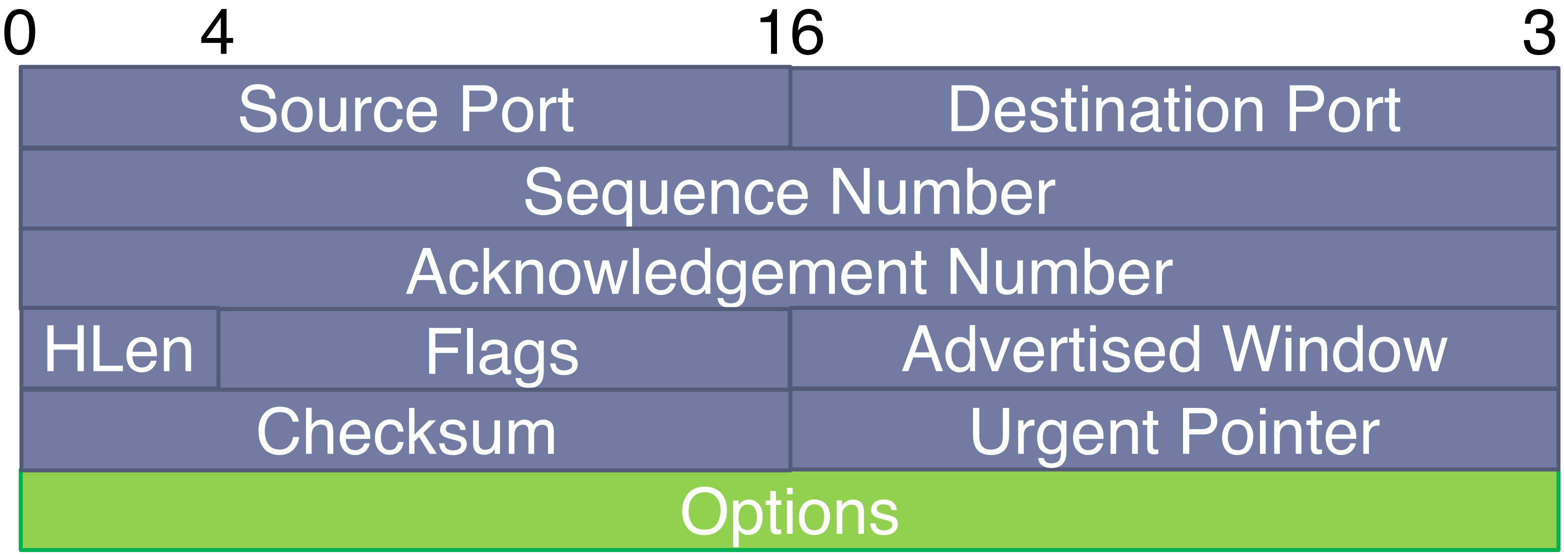
- Winstein/Balakrishnan
  - TCP is maximizing an objective function
    - Fairness/efficiency
    - Throughput/delay
  - Let a machine pick the best fit for your environment

5: Problems with TCP

Common TCP Options



0 4 16 31

Source Port | Destination Port

Sequence Number

Acknowledgement Number

HLen | Flags | Advertised Window

Checksum | Urgent Pointer

Options

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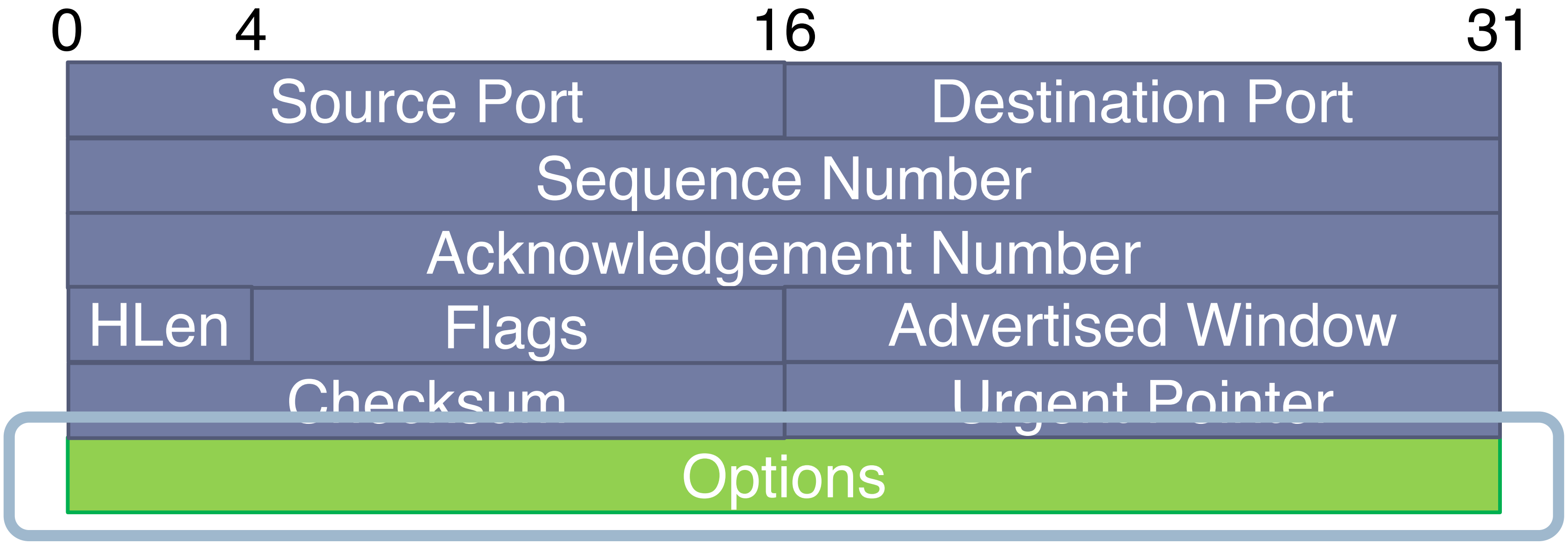

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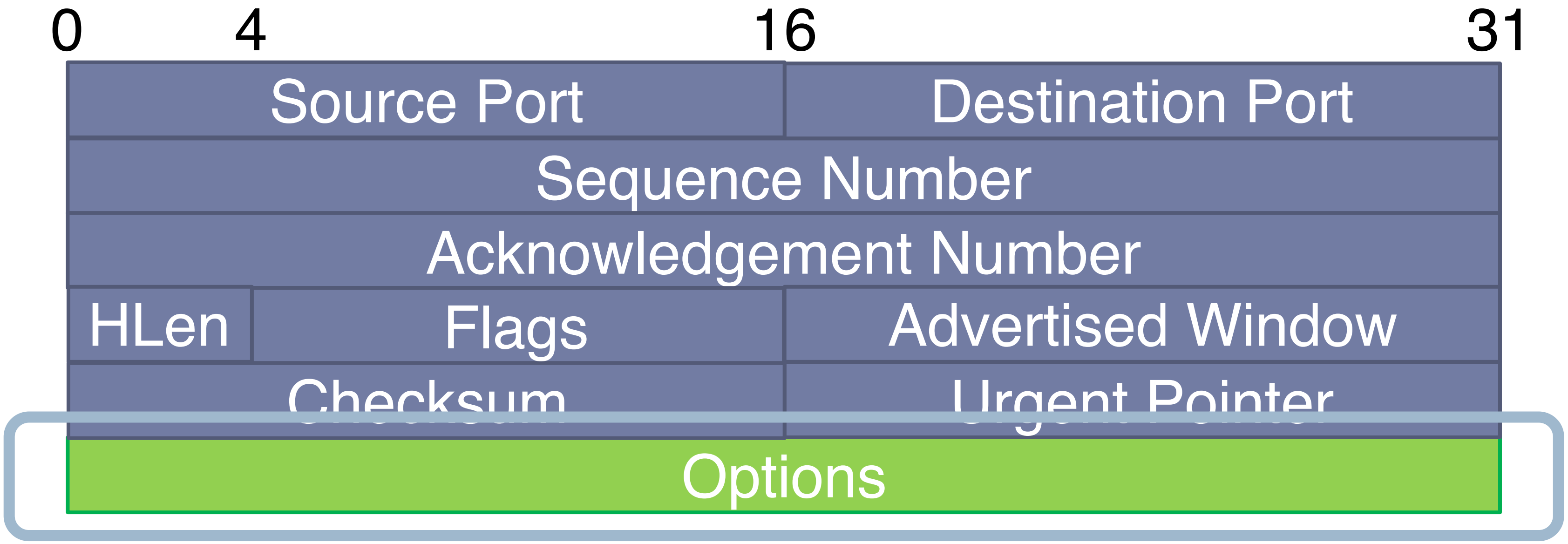

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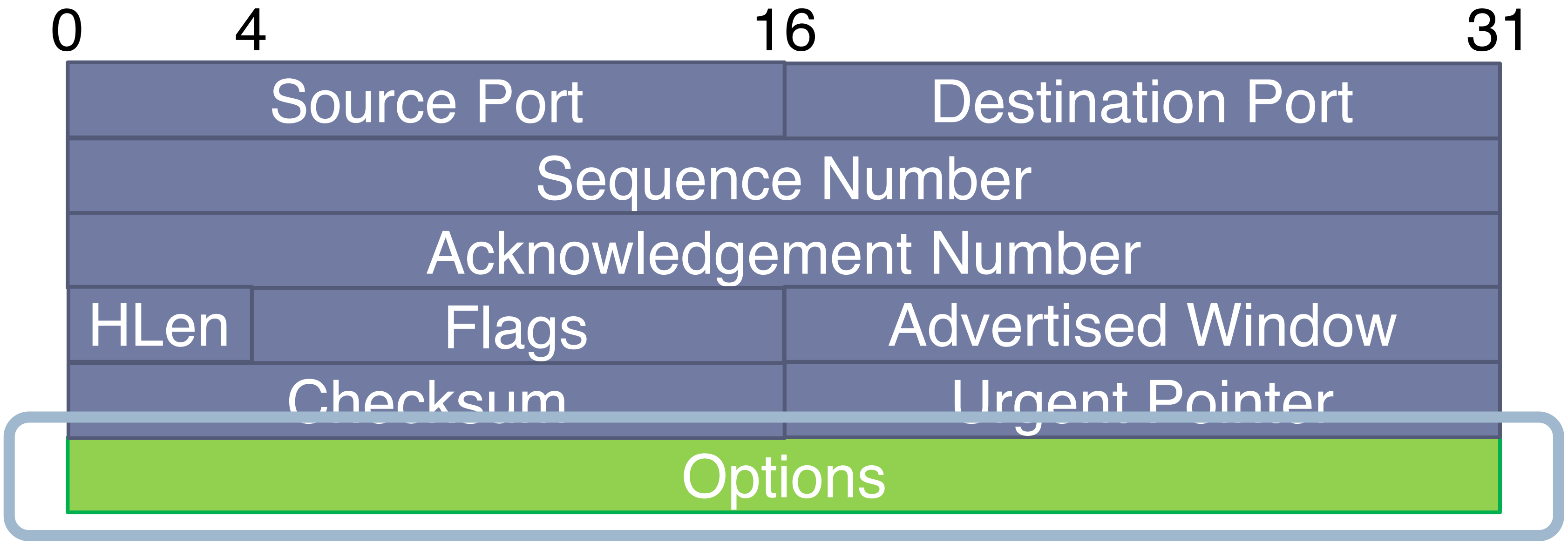

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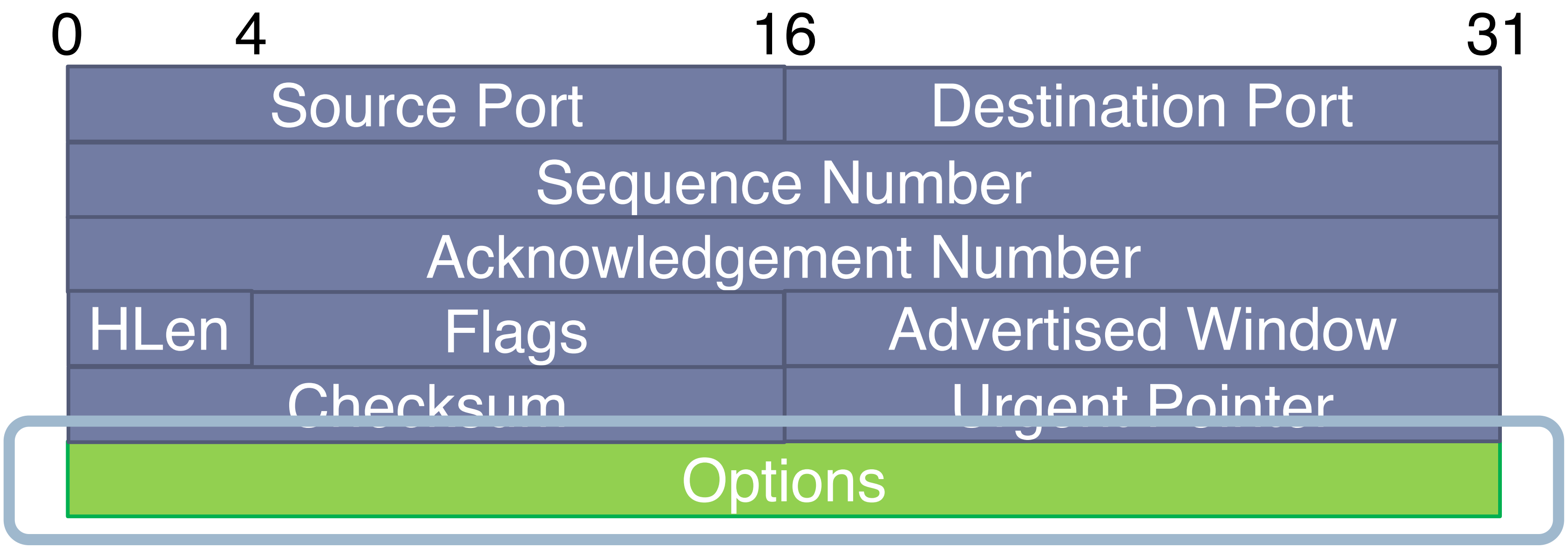

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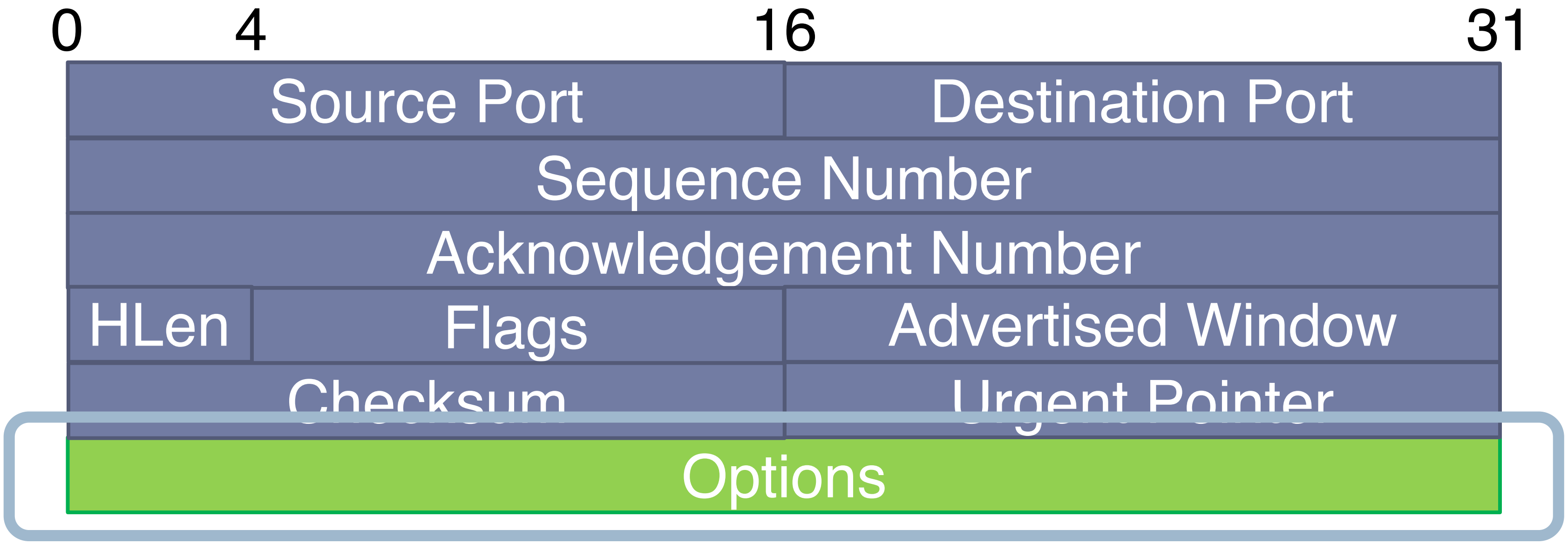

- Window scaling
- SACK: selective acknowledgement
- Maximum segment size (MSS)
- Timestamp

Window Scaling

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  - Effectively caps the window at 65536B, 64KB
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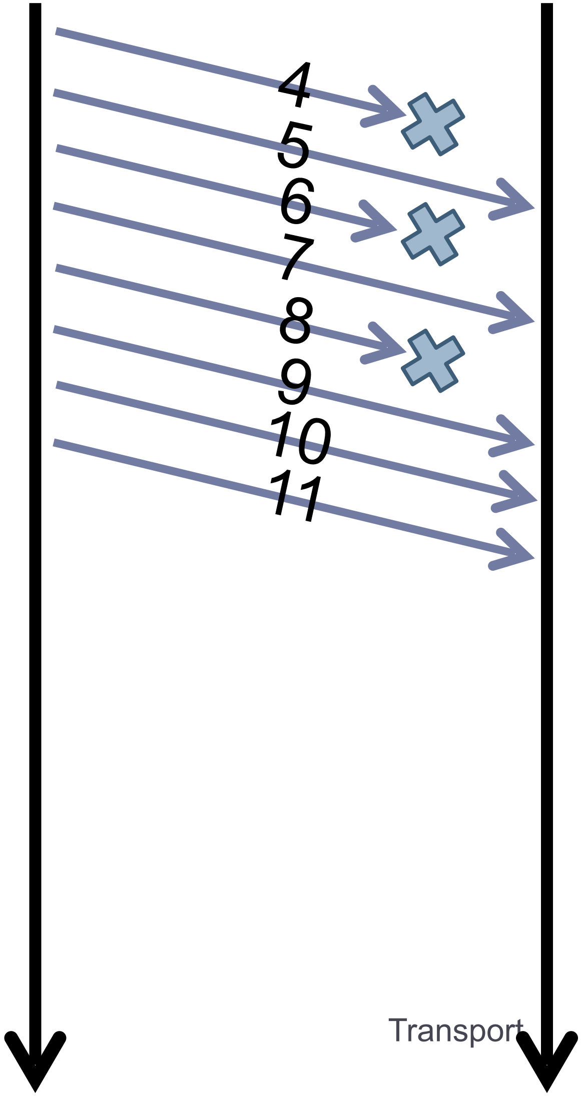
- Solution: introduce a window scaling value
  - *wnd* = *adv_wnd* << *wnd_scale*;
  - Maximum shift is 14 bits, 1GB maximum window

SACK: Selective Acknowledgment



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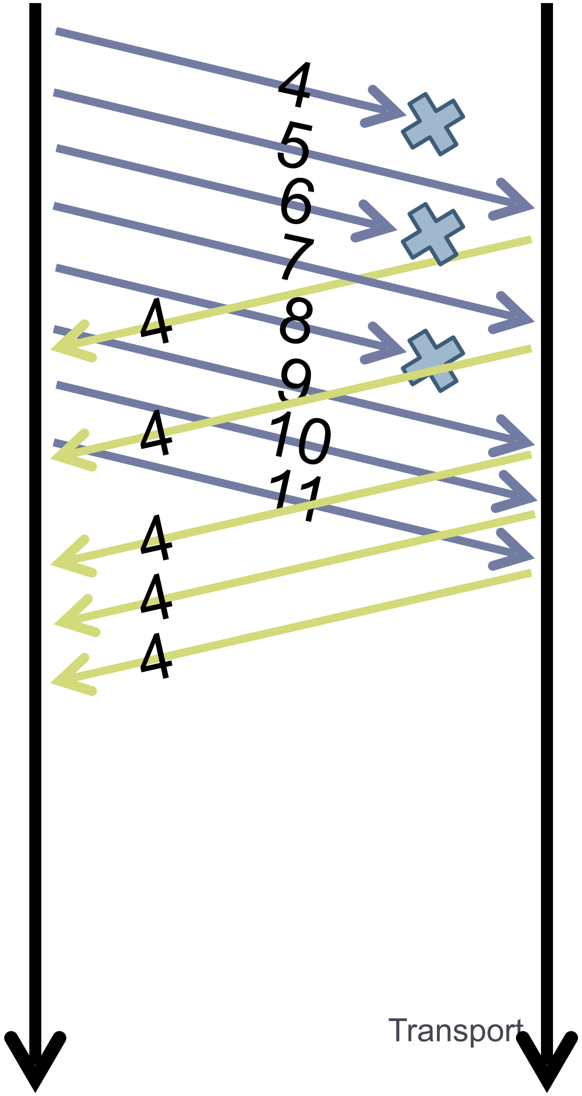
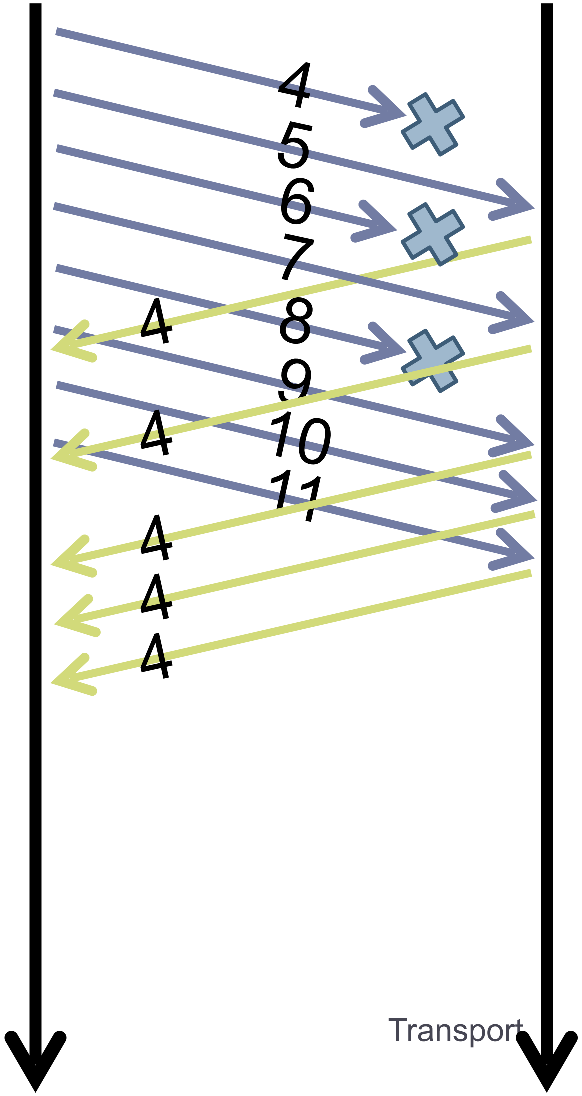


SACK: Selective Acknowledgment

- Problem: duplicate ACKs only tell us about 1 missing packet
  - Multiple rounds of dup ACKs needed to fill all holes
- Solution: selective ACK
  - Include received, out-of-order sequence numbers in TCP header
  - Explicitly tells the sender about holes in the sequence



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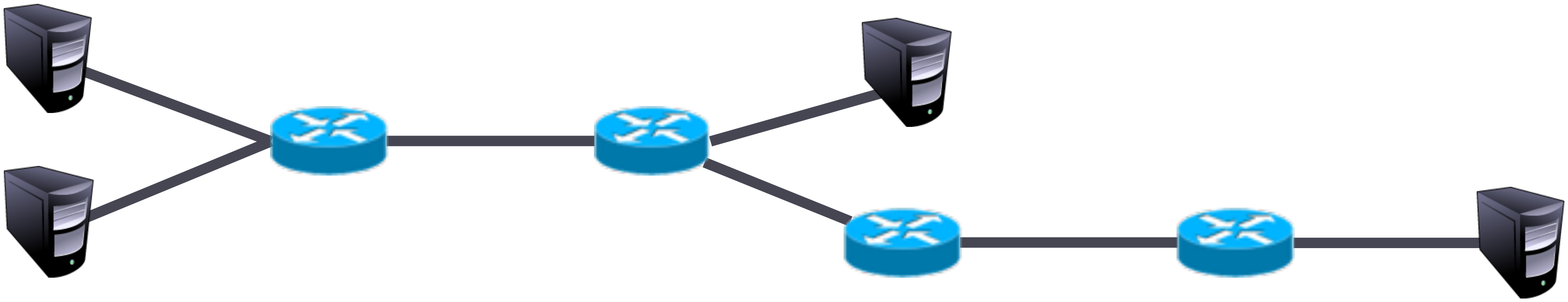
  - Essentially, what is the hosts MTU
  - Saves on path discovery overhead
- Timestamp
  - When was the packet sent (approximately)?
  - Used to prevent sequence number wraparound
  - PAWS algorithm

Issues with TCP

- The vast majority of Internet traffic is TCP
- However, many issues with the protocol
  - Lack of fairness
  - Synchronization of flows
  - Poor performance with small flows
  - Really poor performance on wireless networks
  - Susceptibility to denial of service

Fairness

- Problem: TCP throughput depends on RTT



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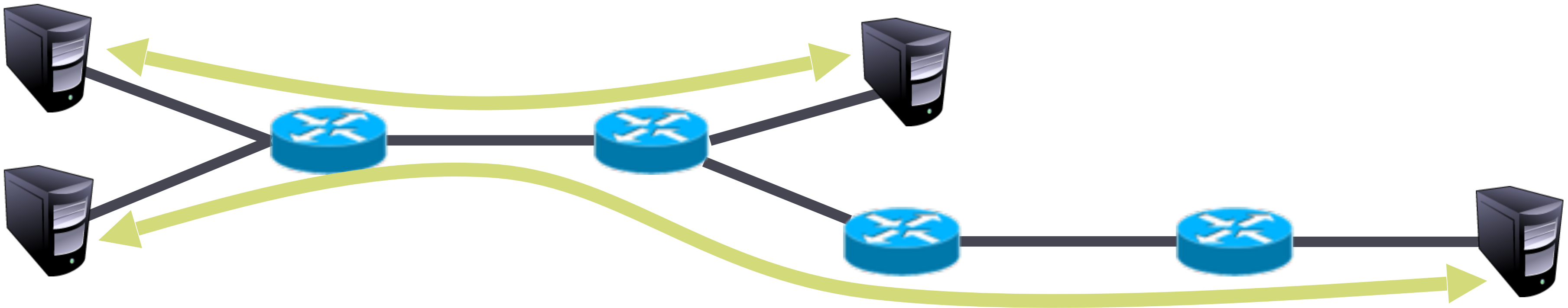

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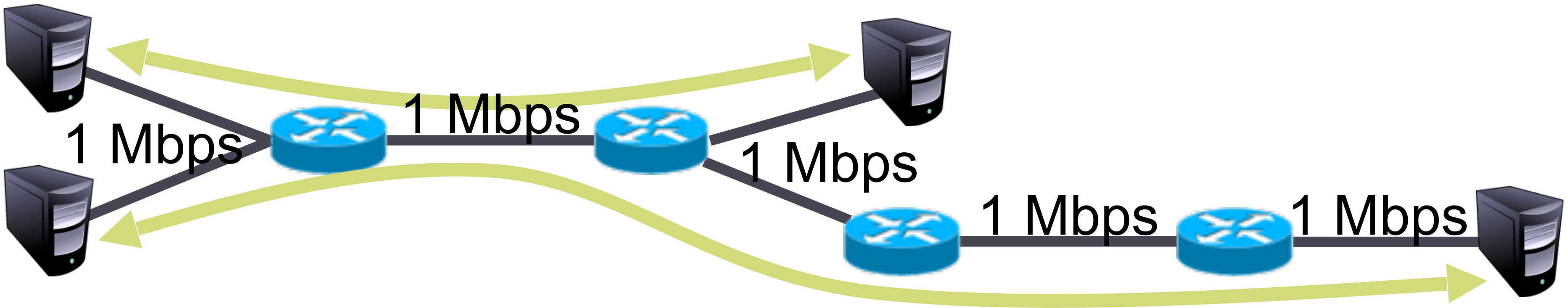

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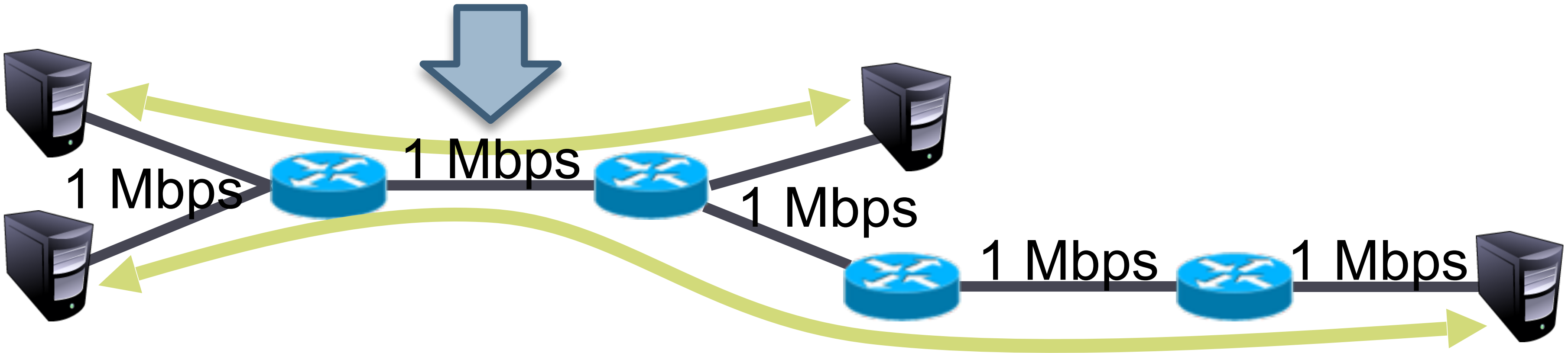

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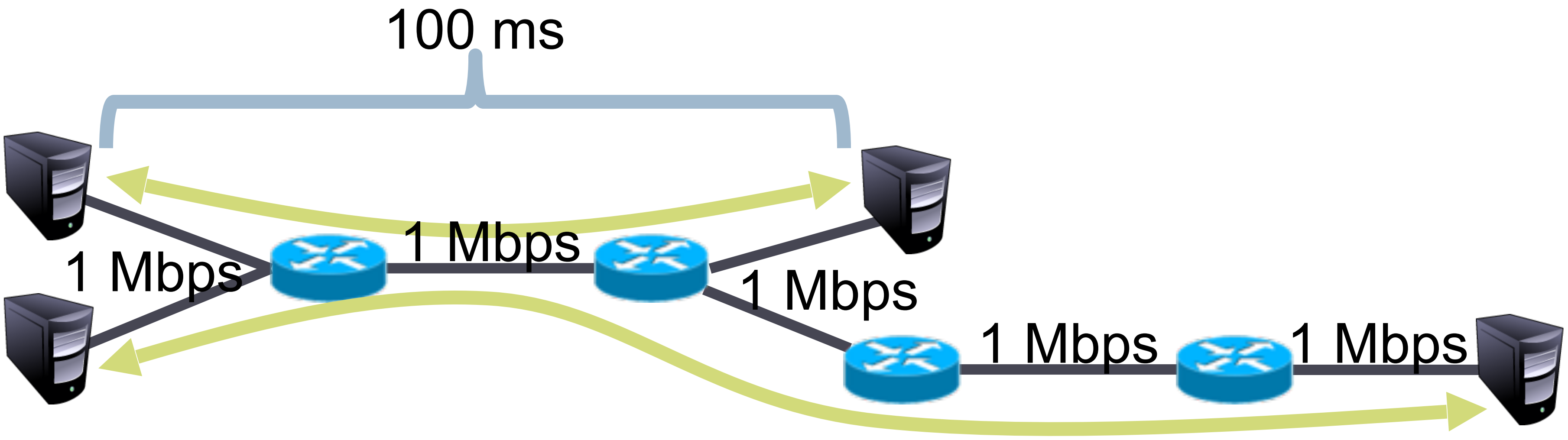

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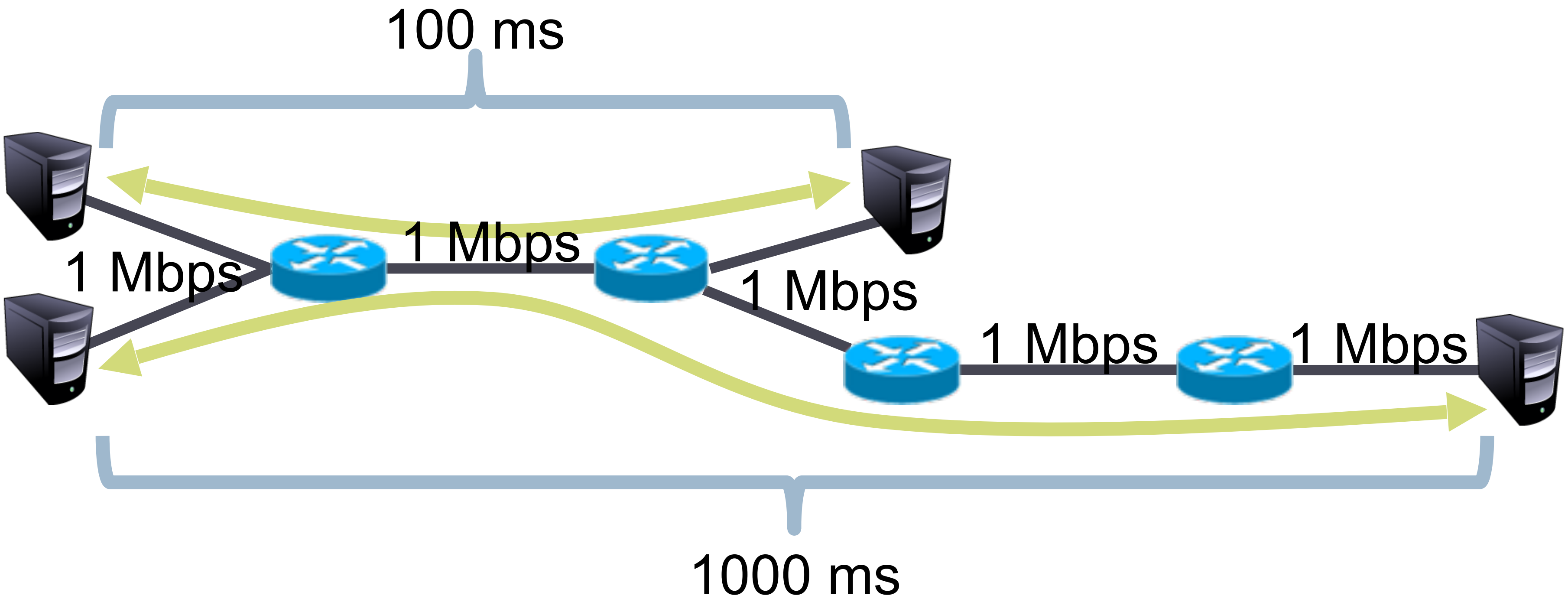

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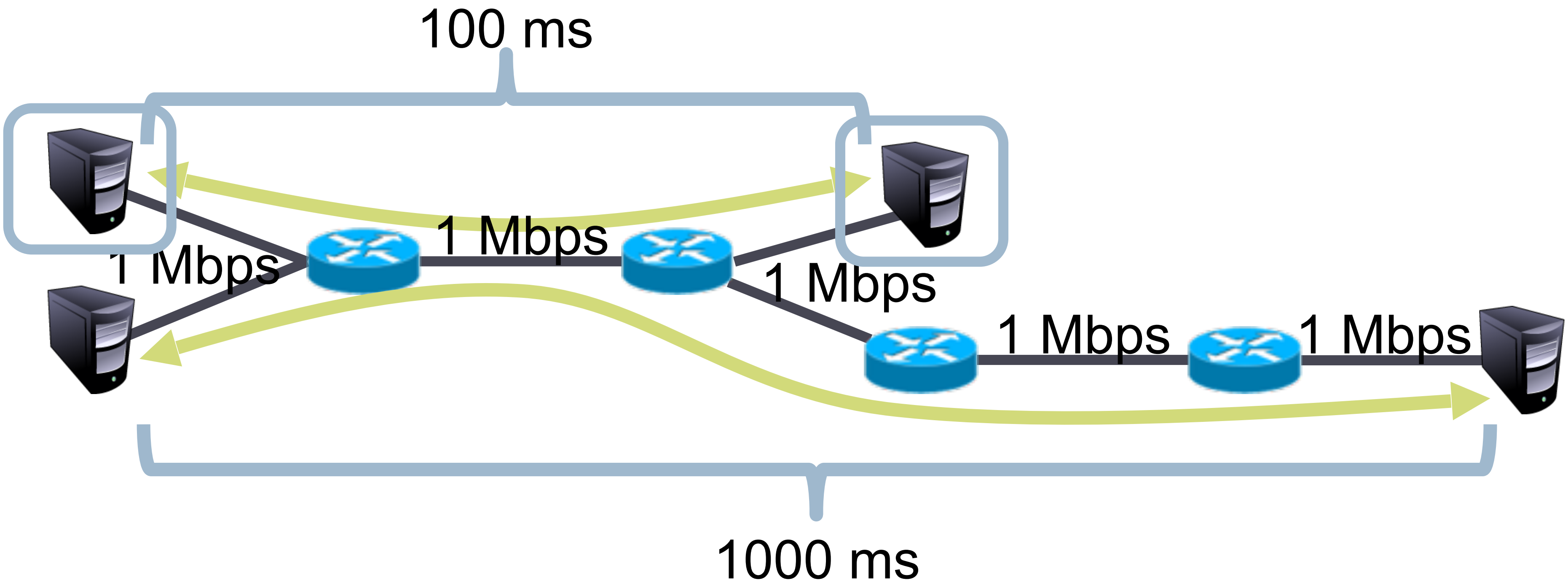

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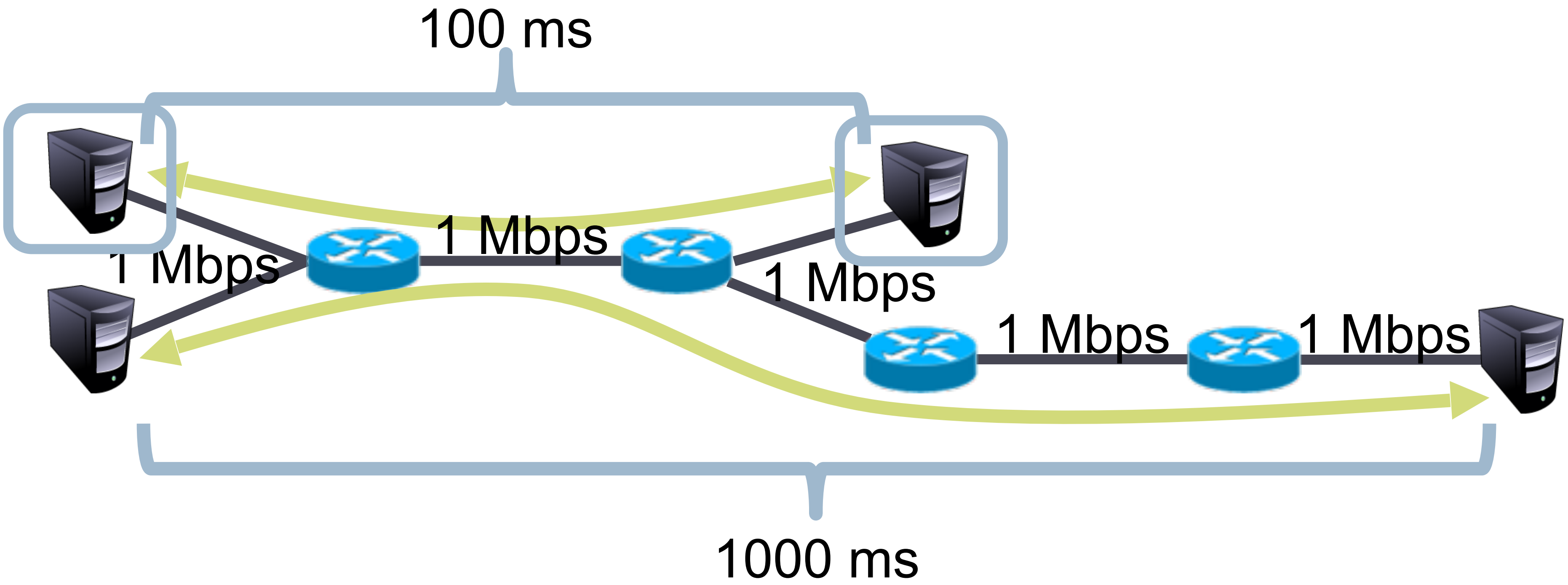
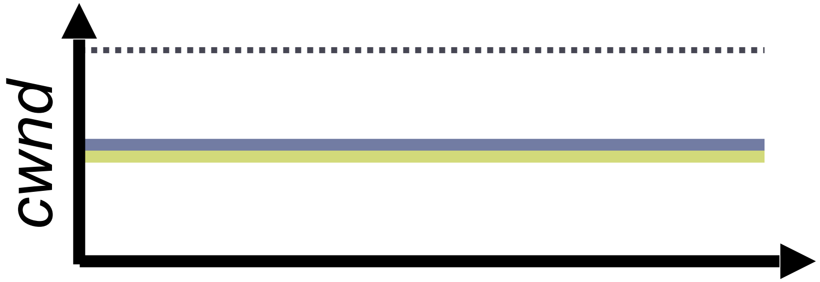


- ACK clocking makes TCP inherently unfair
- Possible solution: maintain a separate delay window
  - Implemented by Microsoft's Compound TCP

Synchronization of Flows

- Ideal bandwidth sharing



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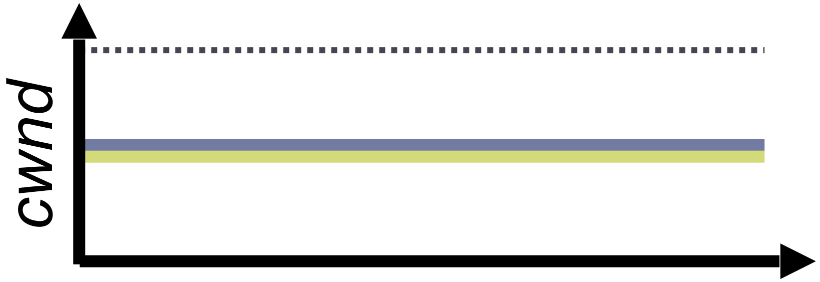
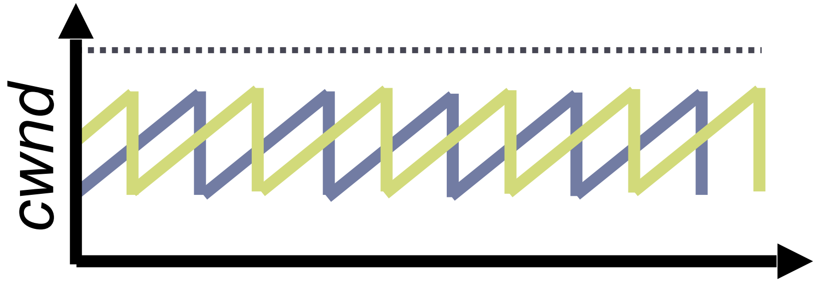


□ Oscillating, but high overall utilization



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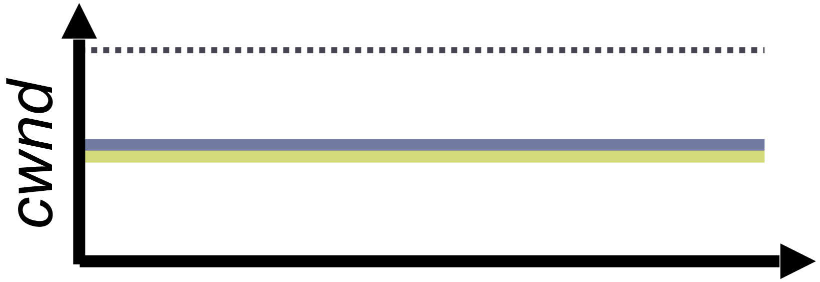

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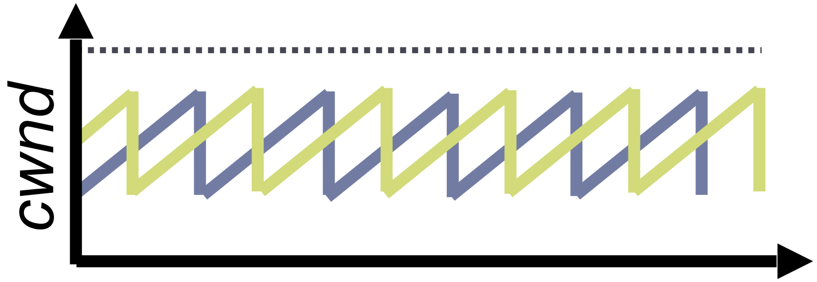

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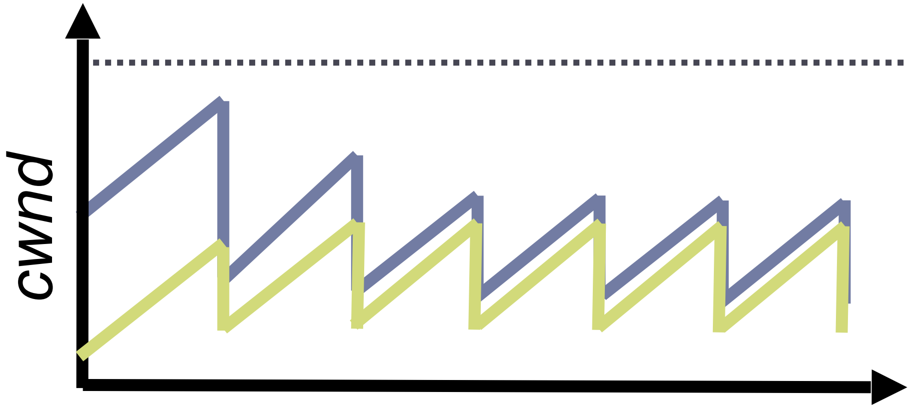

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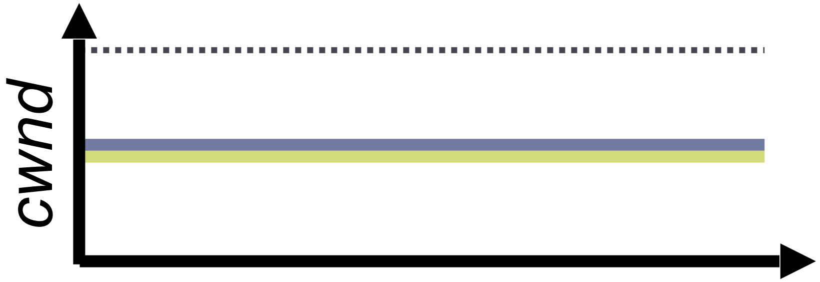

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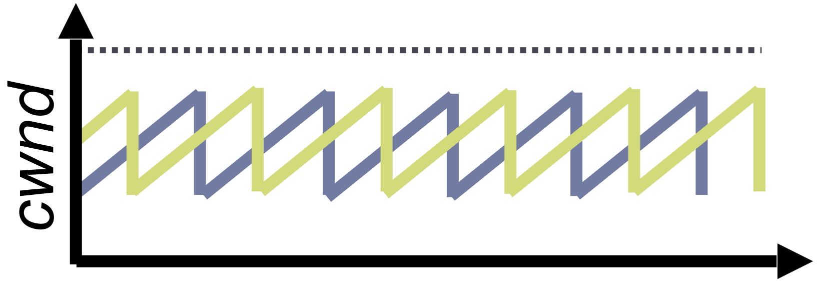

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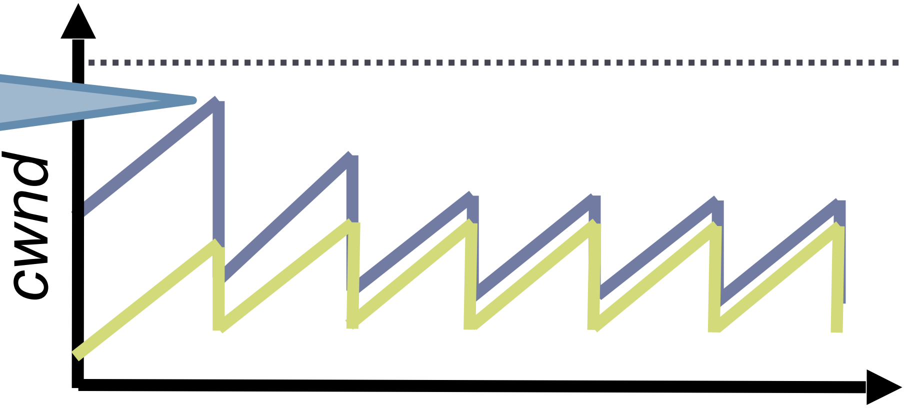

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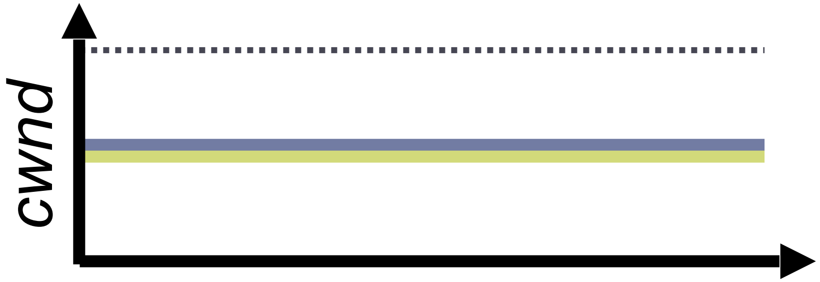

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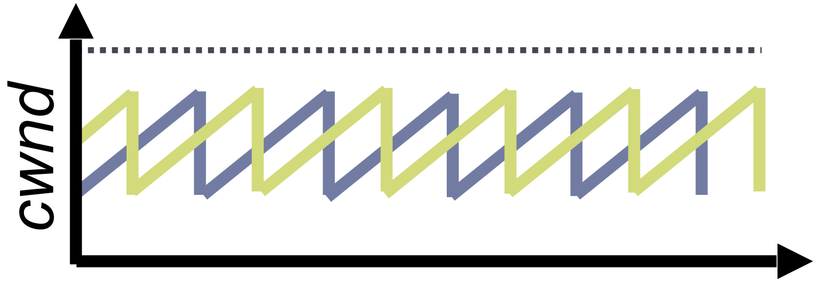

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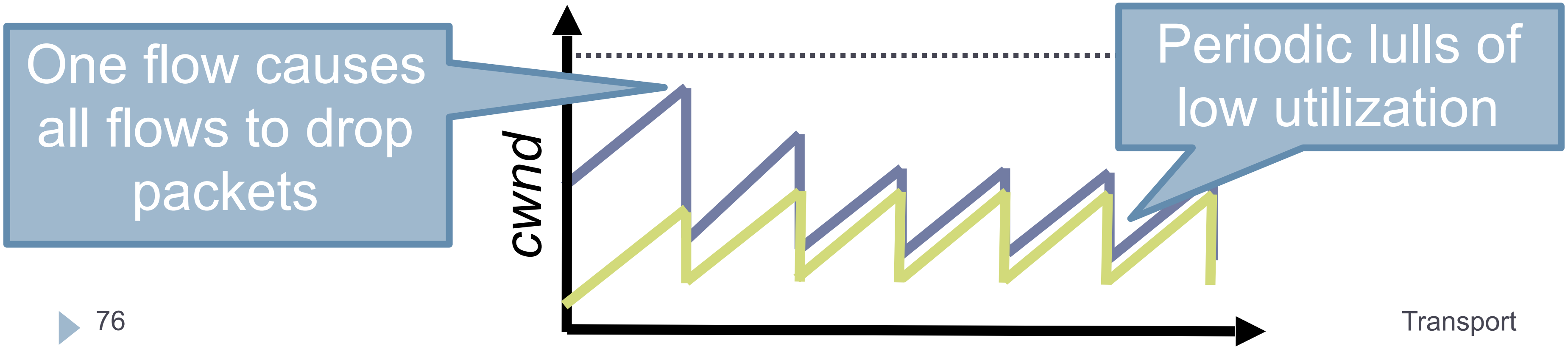

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Periodic lulls of low utilization

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- Problem: TCP is biased against short flows
  - 1 RTT wasted for connection setup (SYN, SYN/ACK)
  - *cwnd* always starts at 1

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  - *cwnd* always starts at 1
- Vast majority of Internet traffic is short flows
  - Mostly HTTP transfers, <100KB
  - Most TCP flows never leave slow start!

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- Problem: TCP is biased against short flows
  - 1 RTT wasted for connection setup (SYN, SYN/ACK)
  - *cwnd* always starts at 1
- Vast majority of Internet traffic is short flows
  - Mostly HTTP transfers, <100KB
  - Most TCP flows never leave slow start!
- Proposed solutions (driven by Google):
  - Increase initial *cwnd* to 10
  - TCP Fast Open: use cryptographic hashes to identify receivers, eliminate the need for three-way handshake
  - QUIC 0-RTT handshake

Wireless Networks

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- TCP throughput ~ 1/sqrt(drop rate)
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- Possible solutions:
  - Break layering, push data link info up to TCP
  - Use delay-based congestion detection (TCP Vegas)
  - Explicit congestion notification (ECN)

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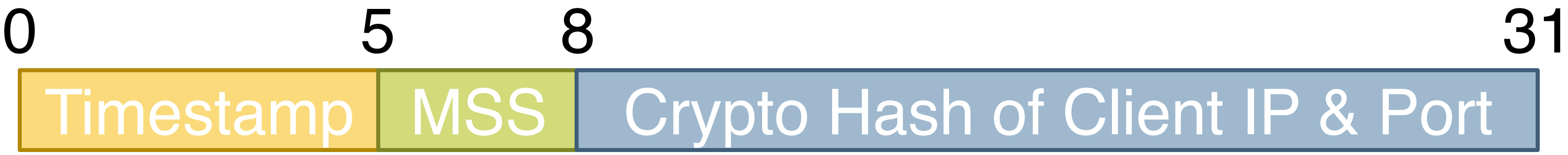
- Solution: SYN cookies
  - Idea: don’t store initial state on the server
  - Securely insert state into the SYN/ACK packet
  - Client will reflect the state back to the server

SYN Cookies

0

Sequence Number

SYN Cookies



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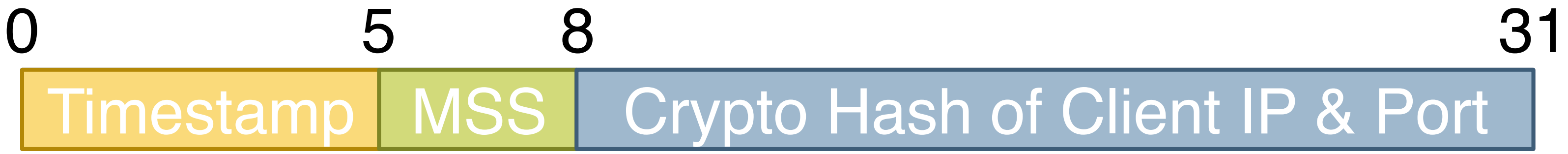
Timestamp | MSS | Crypto Hash of Client IP & Port

SYN Cookies



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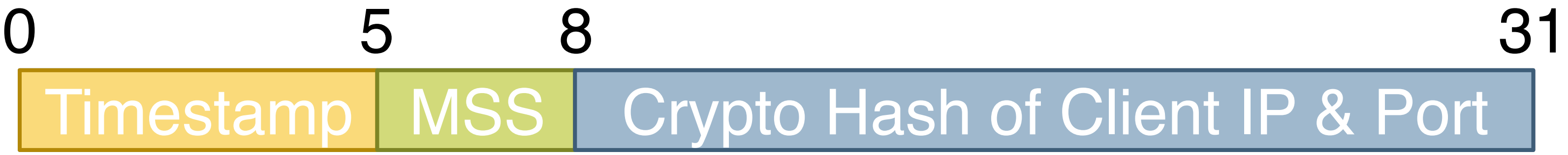

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Timestamp | MSS | Crypto Hash of Client IP & Port

- Did the client really send me a SYN recently?
  - Timestamp: freshness check
  - Cryptographic hash: prevents spoofed packets
- Maximum segment size (MSS)
  - Usually stated by the client during initial SYN
  - Server should store this value…
  - Reflect the clients value back through them

SYN Cookies in Practice

- Advantages
  - Effective at mitigating SYN floods
  - Compatible with all TCP versions
  - Only need to modify the server
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- Disadvantages
  - MSS limited to 3 bits, may be smaller than clients actual MSS
  - Server forgets all other TCP options included with the client's SYN
    - SACK support, window scaling, etc.