Transport

RUTGERS
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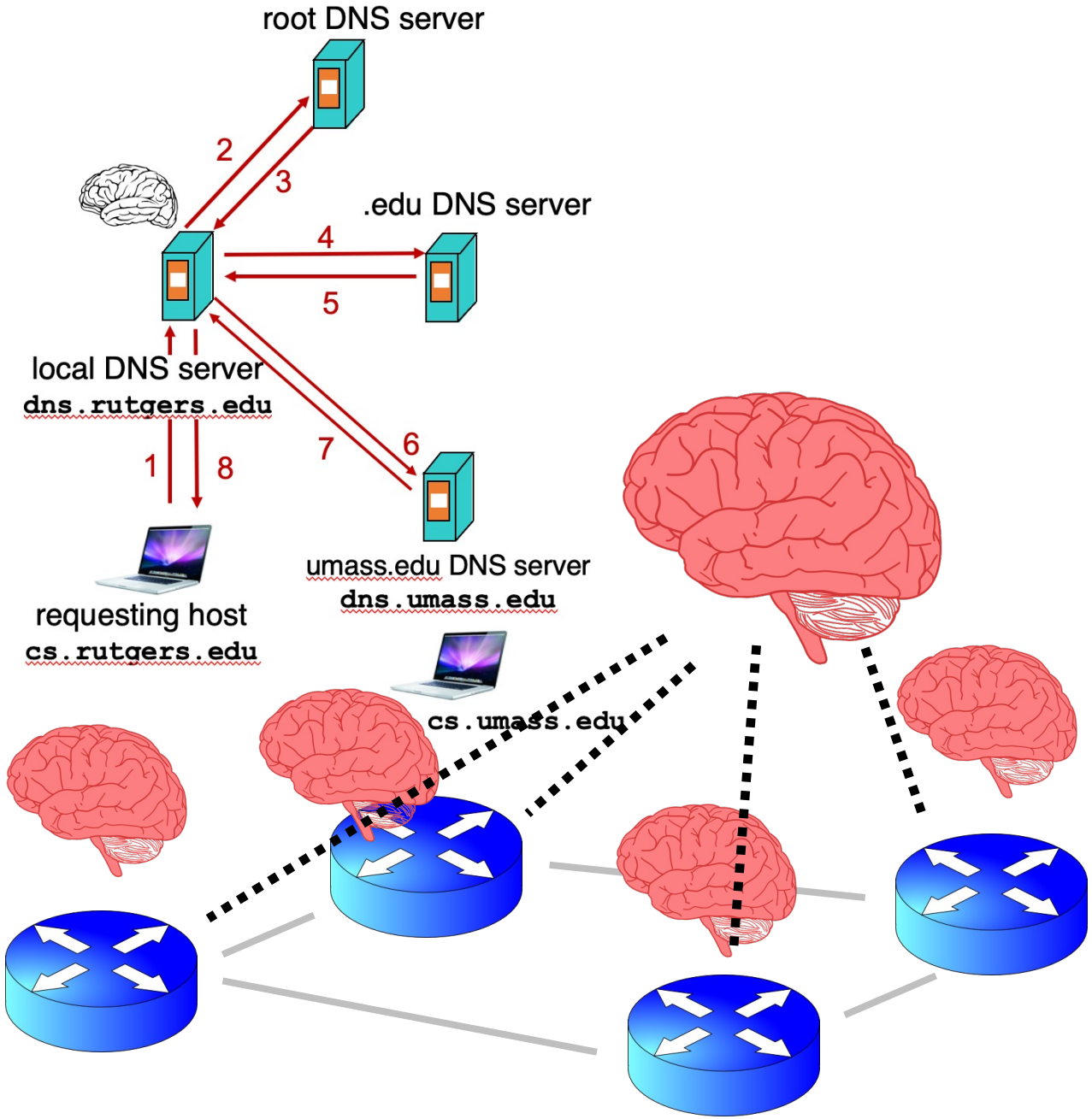
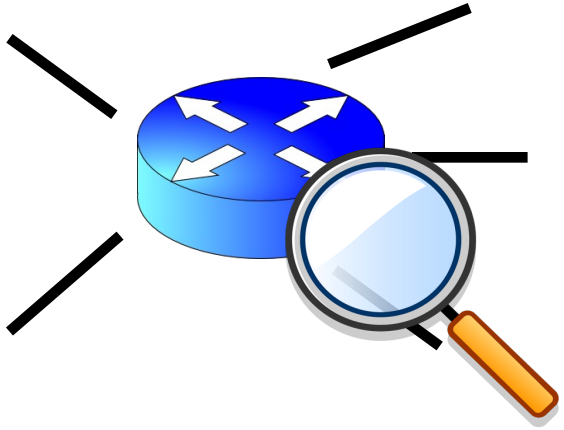
Some fundamental problems

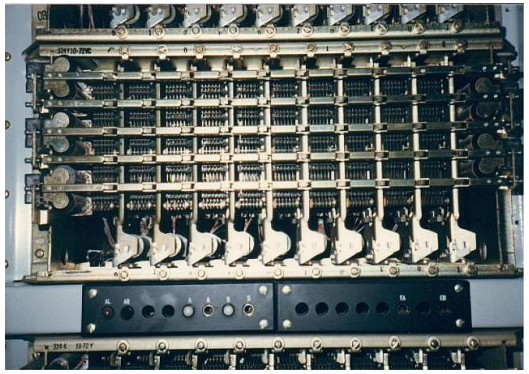
Problems so far

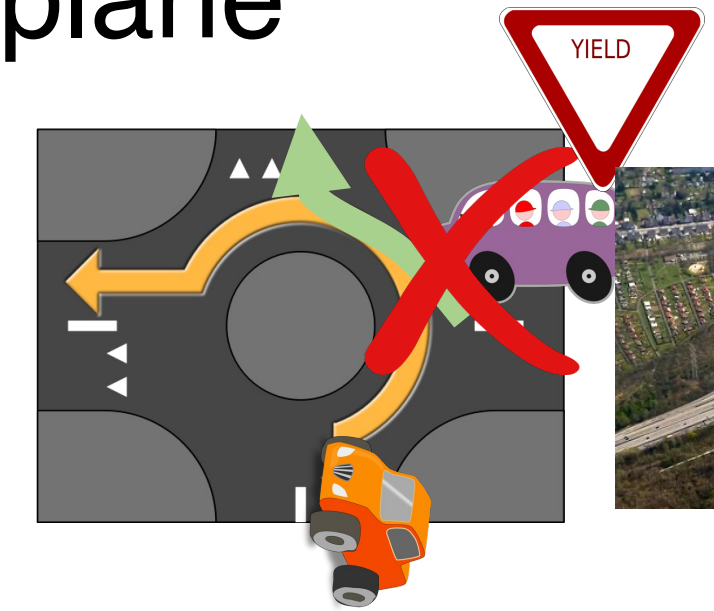
- (0) Name resolution
- (1) Routing
  - Control plane, data plane
  - routing, forwarding

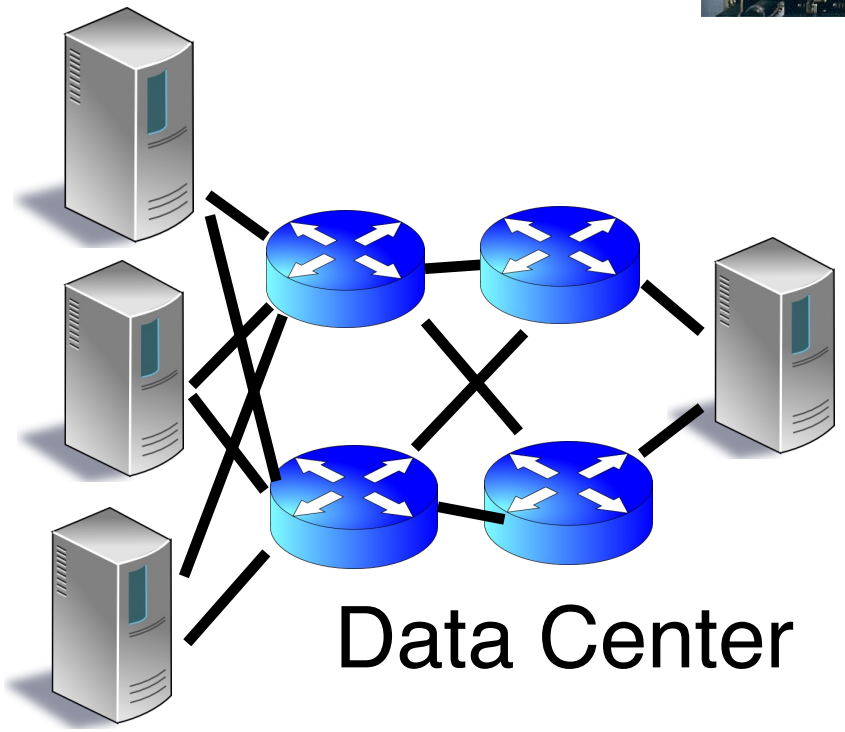


(2) High-speed data plane









Data Center

- Transport won't help if the network has choke points: e.g., routers
- How to design high-speed hardware routers?
- How to design high-speed software routers?
- Data centers, middleboxes

In general, networks give no guarantees

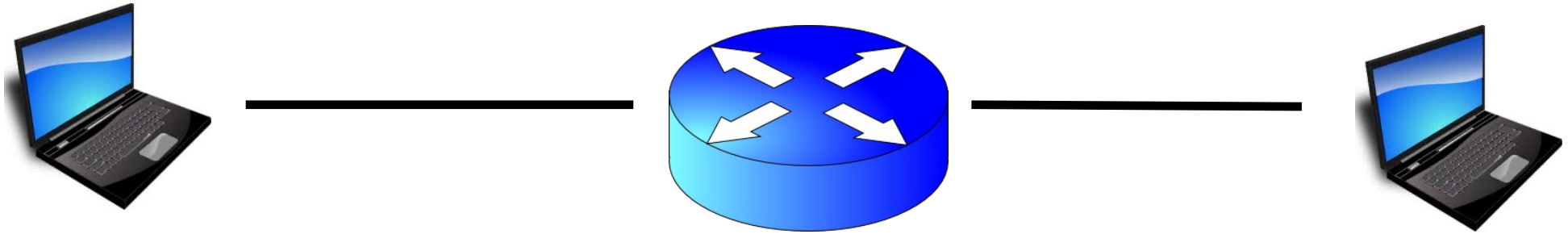
- Packets may be lost, corrupted, reordered, on the way to the destination
  - Best effort delivery

- Advantage: The network becomes very simple to build
  - Don’t have to make it reliable
  - Don’t need to implement any performance guarantees
  - Don’t need to maintain packet ordering
  - Almost any medium can deliver individual packets
    - Example: RFC 1149: “IP Datagrams over Avian Carriers”

- Early Internet thrived: easy to engineer, no guarantees to worry about

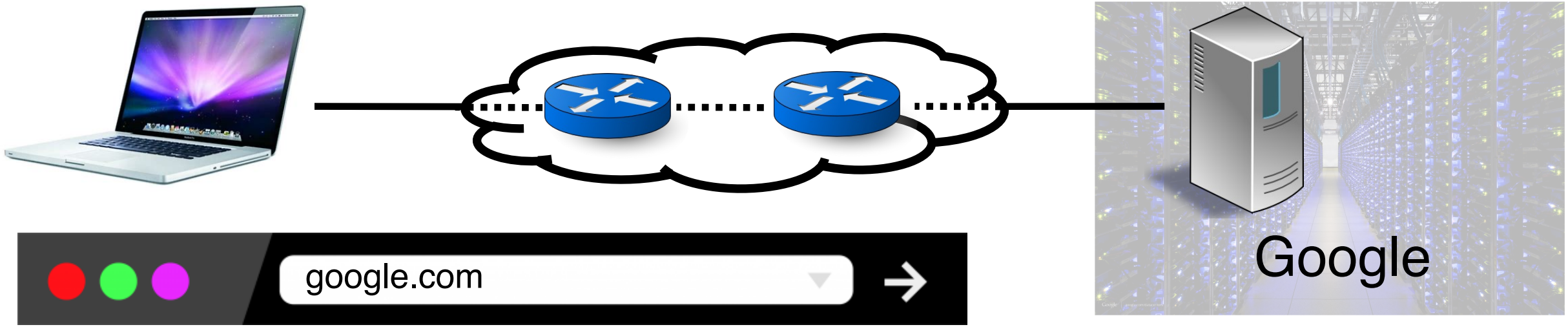
(3) Providing guarantees for applications

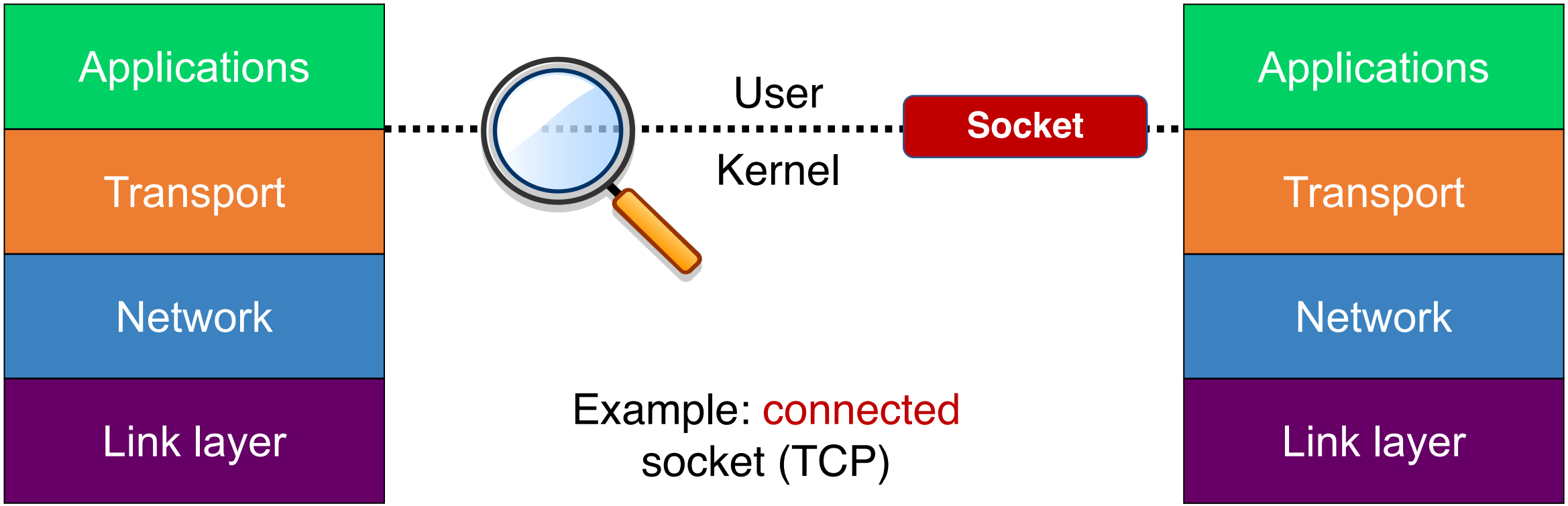
- How should endpoints provide guarantees to applications?



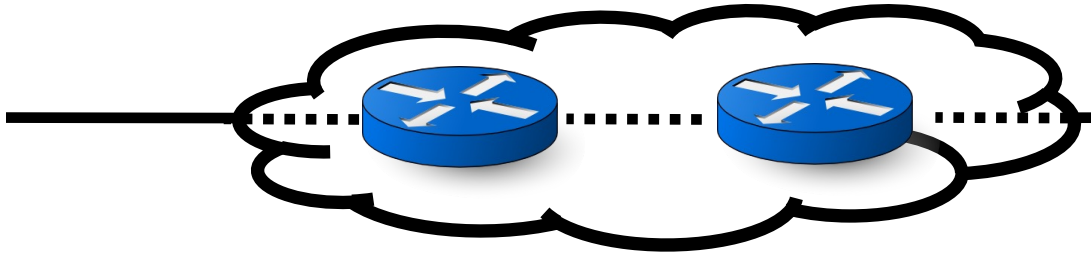
- Transport software on the endpoint oversees implementing guarantees on top of an unreliable network
- Semantics are per “conversation” and agnostic to app data
- Reliable delivery, ordered delivery, fair sharing of resources
- Two popular transports: TCP, UDP
  - (there are others)

Application-OS interface





Example: connected socket (TCP)



google.com

TCP

process

socket

$IP_A + port_A$

connect(

IP_B, $port_B$)

send()

process

socket

$IP_B + port_B$

bind($IPaddr_B$, $port_B$)

listen()

accept()

recv()

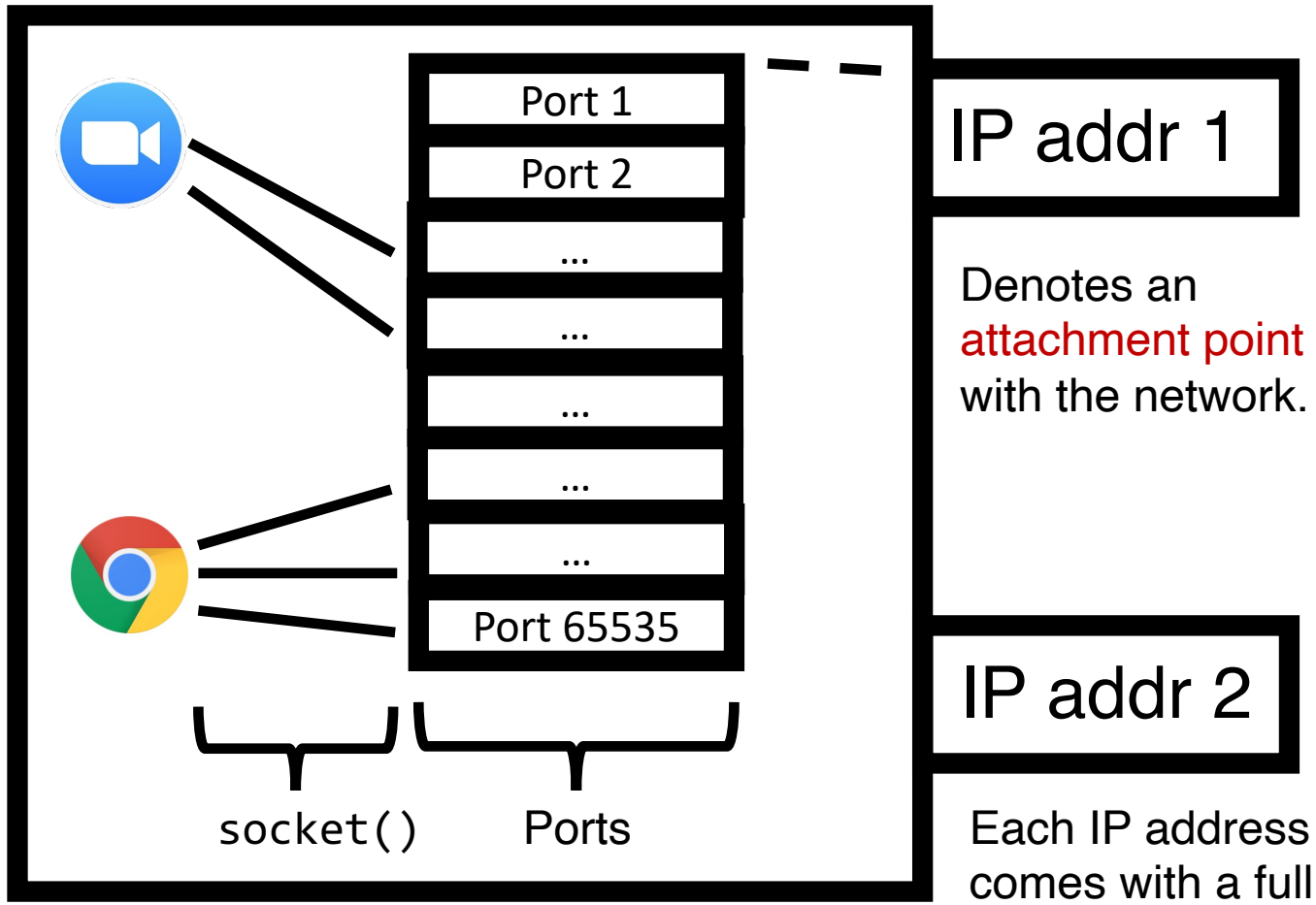
Sample code

- Walk through

- What sockets exist on the machine?
  - `ss`

What does transport do?

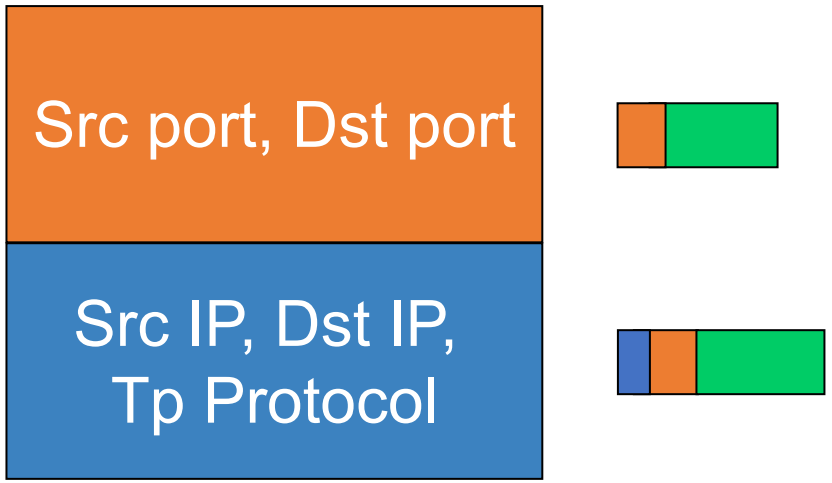
(3.1) App Context



Denotes an attachment point with the network.

Each IP address comes with a full copy of its own ports.

Connection lookup: The operating system does a lookup using these data to determine the right socket and app.



UDP or TCP listening:
(dst IP, dst port, TCP/UDP)

TCP established:
(dst IP, dst port, src IP, src port, TCP)

TCP sockets of different types

Listening (bound but unconnected)

```
# On server side
ls = socket(AF_INET, SOCK_STREAM)
ls.bind(serv_ip, serv_port)
ls.listen() # no accept() yet
```

(dst IP, dst port)

➔

Socket (`ss`)

Enables new connections to be demultiplexed correctly

Connected (Established)

```
# On server side
cs, addr = ls.accept()

# On client side
connect(serv_ip, serv_port)
```

`accept()` creates a new socket with the 4-tuple (established) mapping

(src IP, dst IP, src port, dst port)

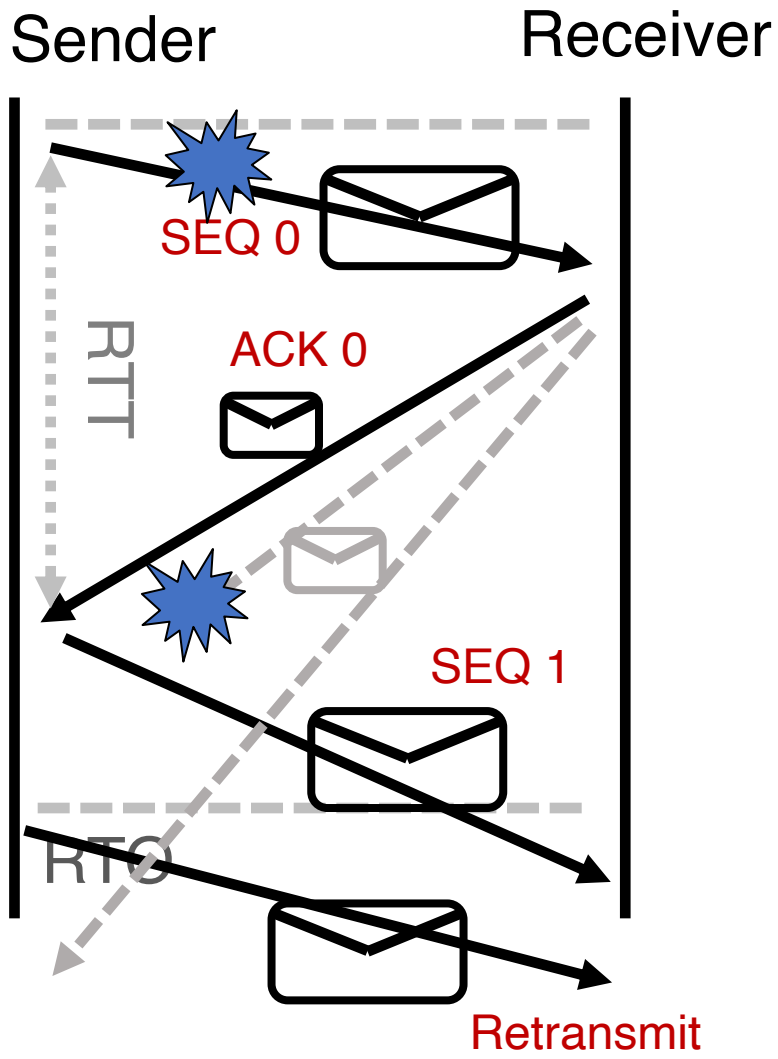
➔

Socket (`cs NOT ls`)

Enables established connections to be demultiplexed correctly

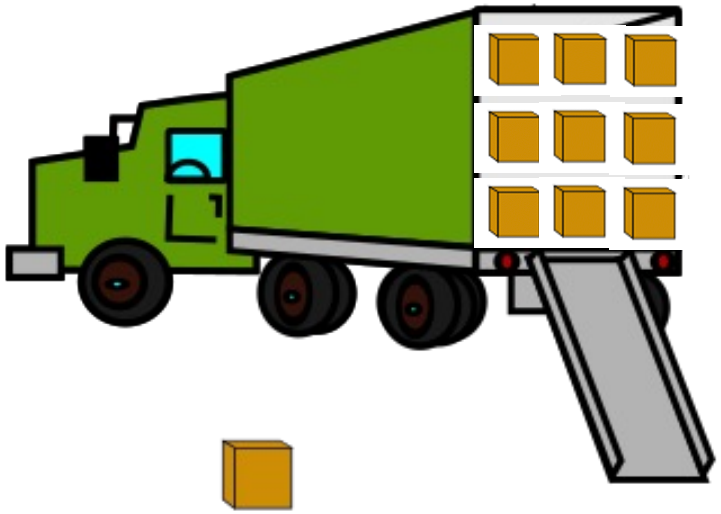
(3.2) Reliability: Stop and Wait. 3 Ideas

- ACKs: Sender sends a single packet, then waits for an ACK to know the packet was successfully received. Then the sender transmits the next packet.
- RTO: If ACK is not received until a timeout, sender retransmits the packet
- Seq: Disambiguate duplicate vs. fresh packets using sequence numbers that change on “adjacent” packets



Stop and wait is reliable, but too slow.

Sending one packet per RTT makes the data transfer rate limited by the time between the endpoints, rather than the bandwidth.



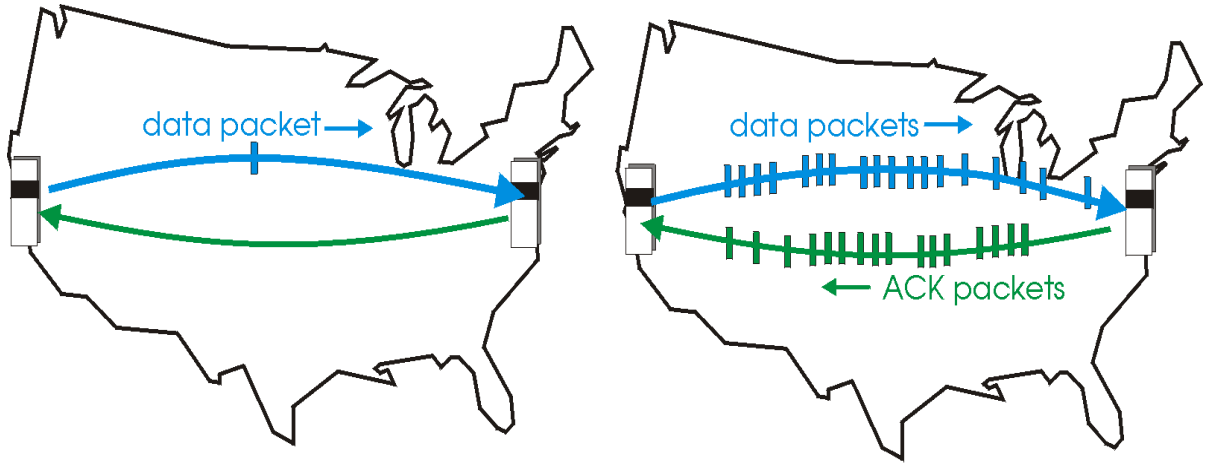
Ensure you got the (one) box safely; make N trips

Ensure you get N boxes safely; make just 1 trip!

Keep many packets in flight

Pipelined reliability

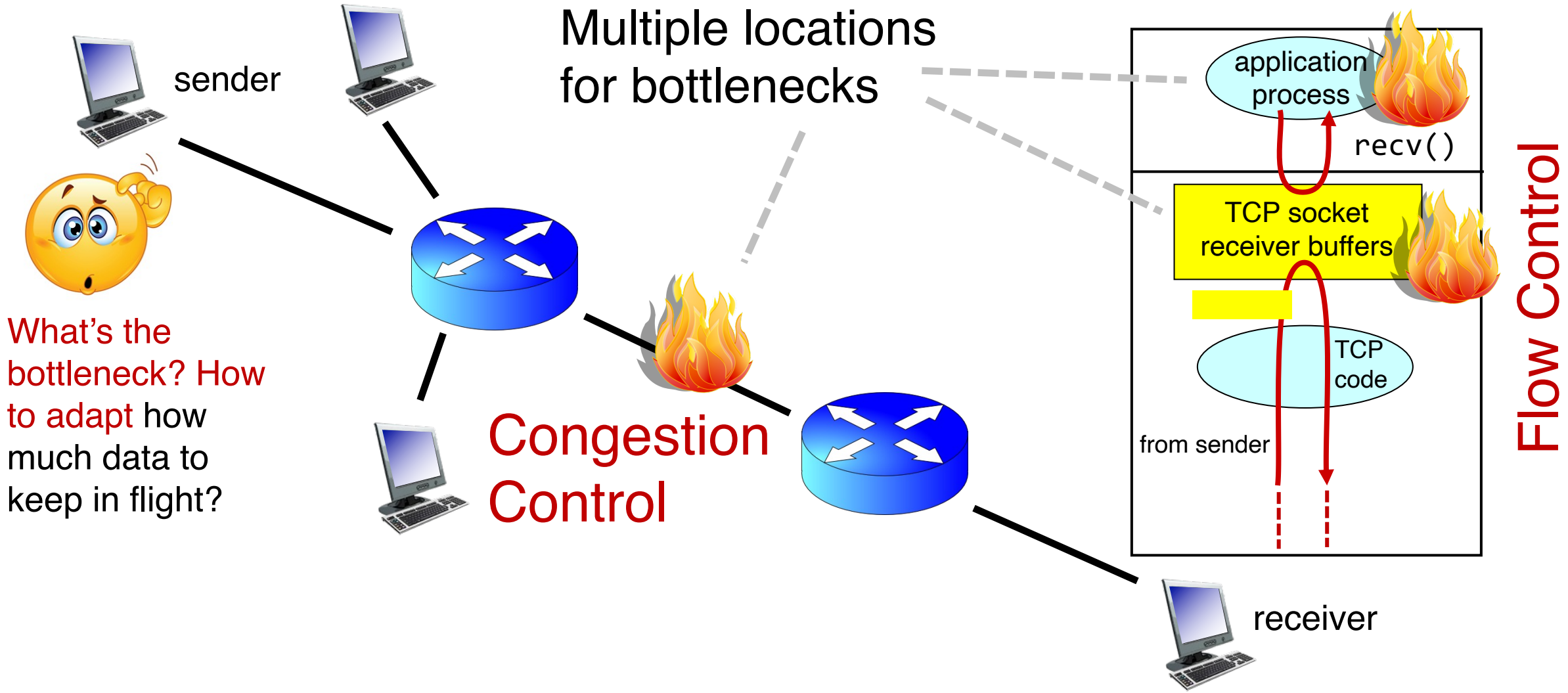
- Data in flight: data that has been sent, but sender hasn't yet received ACKs from the receiver
  - Note: can refer to packets in flight or bytes in flight
- New packets sent at the same time as older ones still in flight
- New packets sent at the same time as ACKs are returning
- More data moving in same time!
- Improves throughput
  - Rate of data transfer
- Window
  - How big should the window be?



(a) a stop-and-wait protocol in operation

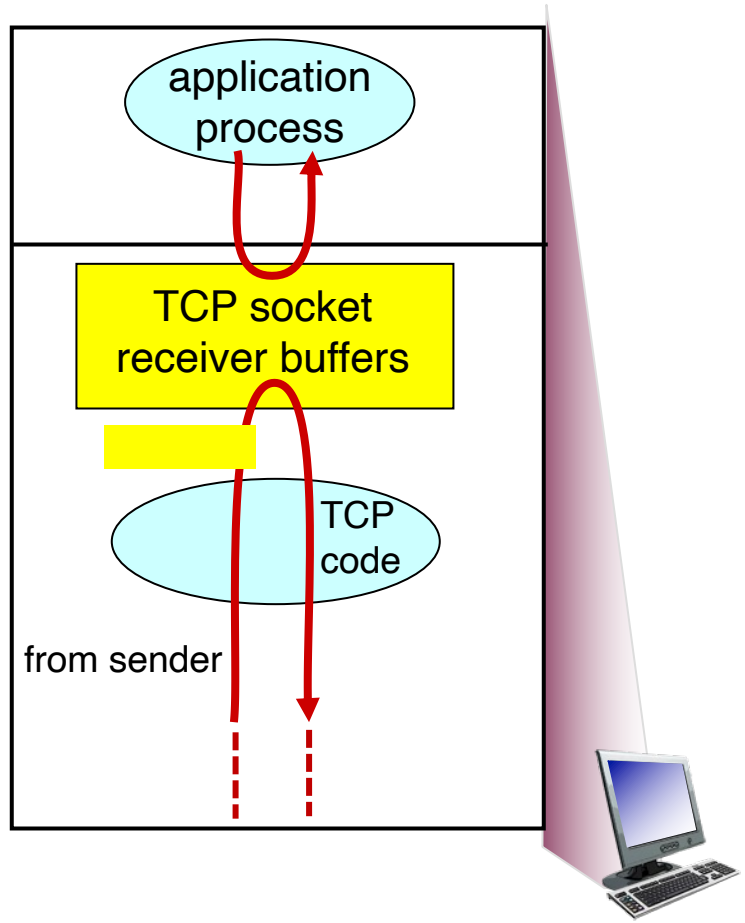
(b) a pipelined protocol in operation

We want to increase throughput, but …



(3.3) Flow Control

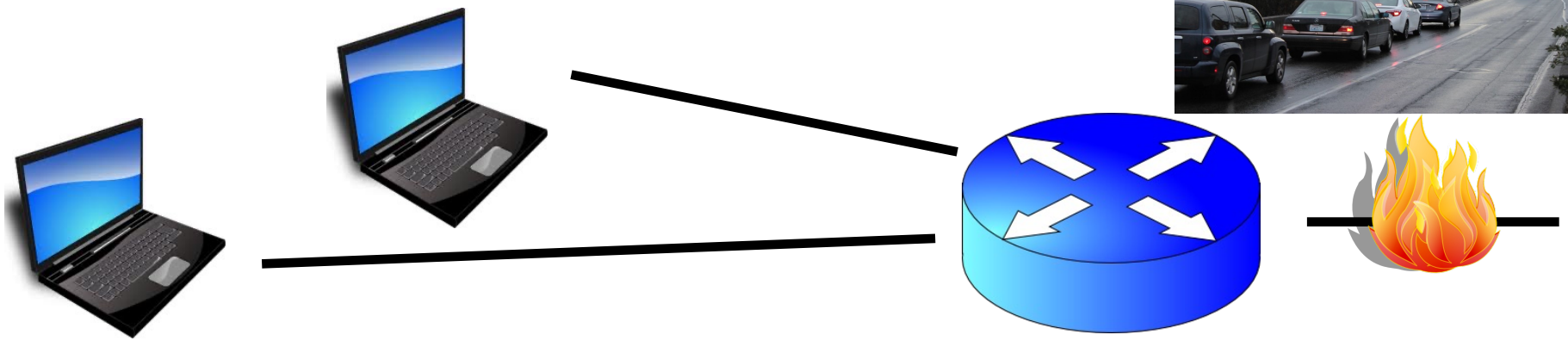
- Have a TCP sender only send as much as the free buffer space available at the receiver.
- *Amount of free buffer varies over time!*
- TCP implements flow control
- Receiver's ACK contains the amount of data the sender can transmit without running out the receiver's socket buffer
- This number is called the advertised window size
- Receiver buffer must be large enough



receiver network stack

(3.4) Congestion control

- How quickly should endpoints send data?



- Known as the congestion control problem
- Congestion control algorithms at source endpoints react to remote network congestion.
- Key question: How to vary the sending rate based on network signals?

A key consequence of the Internet architecture:

Place trust and intelligence in endpoints.

Congestion control is a distributed algorithm (running at endpoints) which attempts to achieve an efficient and fair distribution of bottleneck link resources.

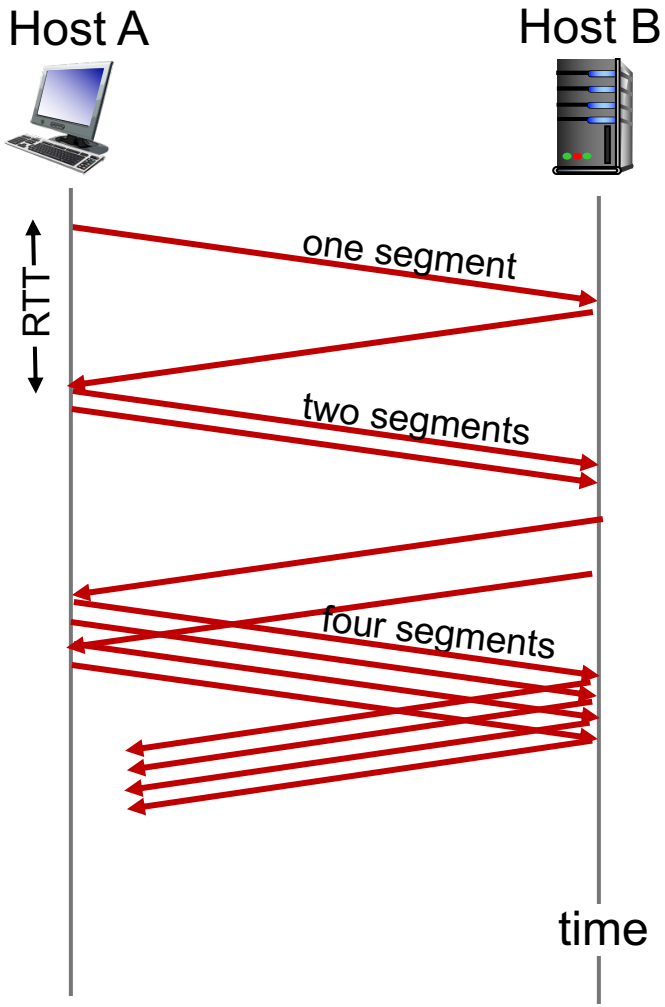
Feedback Control

Finding the right congestion window

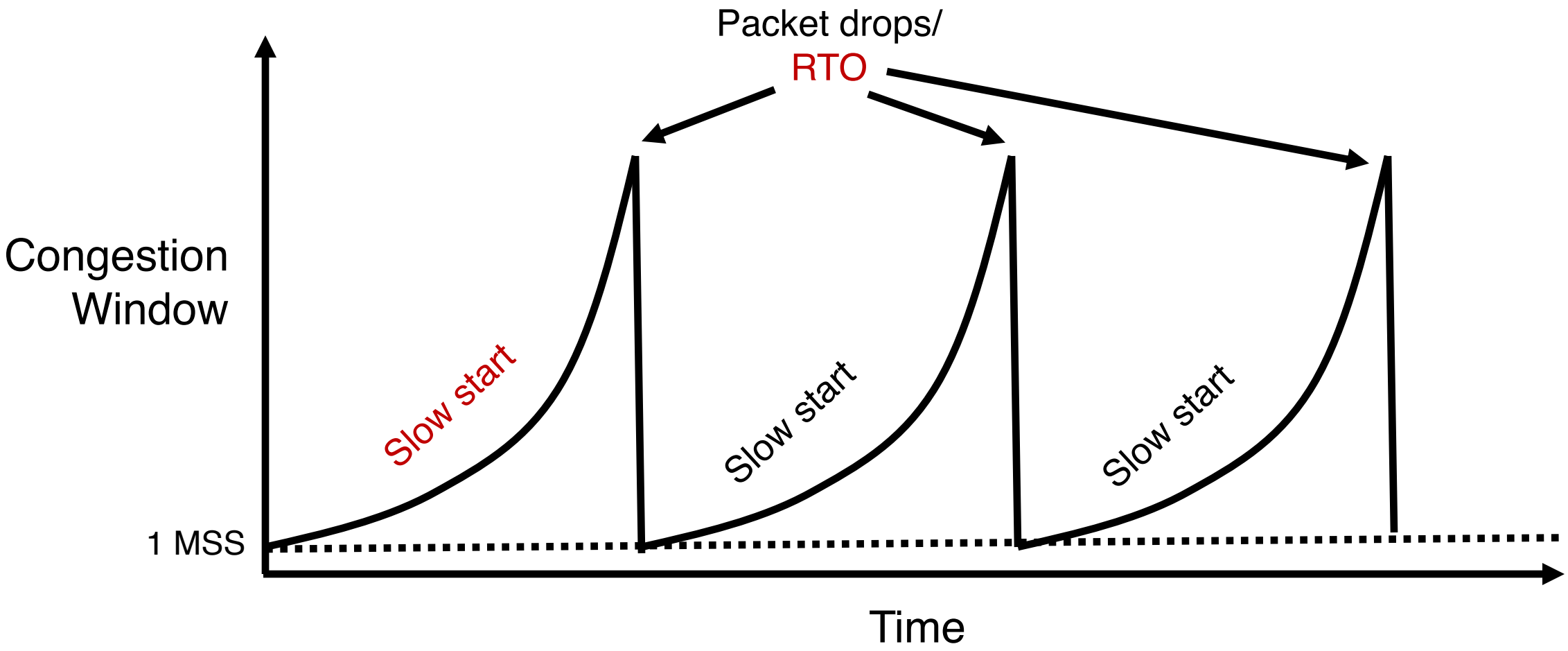
- There is an unknown bottleneck link rate that the sender must match
- If sender sends more than the bottleneck link rate:
  - packet loss, delays, etc.
- If sender sends less than the bottleneck link rate:
  - all packets get through; successful ACKs
- Congestion window (`cwnd`): amount of data in flight

Quickly finding a rate: TCP slow start

- Initially `cwnd` = 1 MSS
  - MSS is “maximum segment size”
- Upon receiving an ACK of each MSS, increase the `cwnd` by 1 MSS
- Effectively, double `cwnd` every RTT
- Initial rate is slow but ramps up exponentially fast
- On loss (RTO), restart from `cwnd := 1 MSS`



Behavior of slow start



Slow start has problems

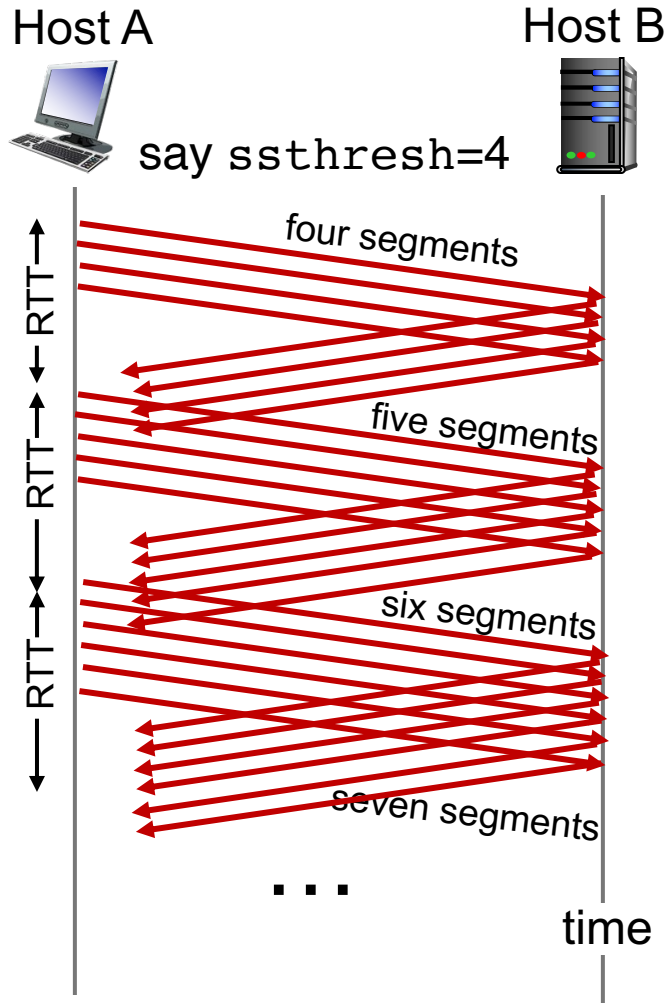
- Congestion window increases too rapidly
  - Example: suppose the “right” window size `cwnd` is 17
  - `cwnd` would go from 16 to 32 and then dropping down to 1
  - Result: massive packet drops

- Congestion window decreases too rapidly
  - Suppose the right `cwnd` is 31, and there is a loss when `cwnd` is 32
  - Slow start will resume all the way back from `cwnd` 1
  - Result: unnecessarily low speed of sending data

- Instead, perform finer adjustments of `cwnd`: congestion avoidance

TCP New Reno: Additive Increase

- Remember the recent past to find a good estimate of link rate
- The last good `cwnd` without packet drop is a good indicator
  - TCP New Reno calls this the slow start threshold (`ssthresh`)
- Increase `cwnd` by 1 MSS every RTT after `cwnd` hits `ssthresh`
  - Effect: increase window additively per RTT



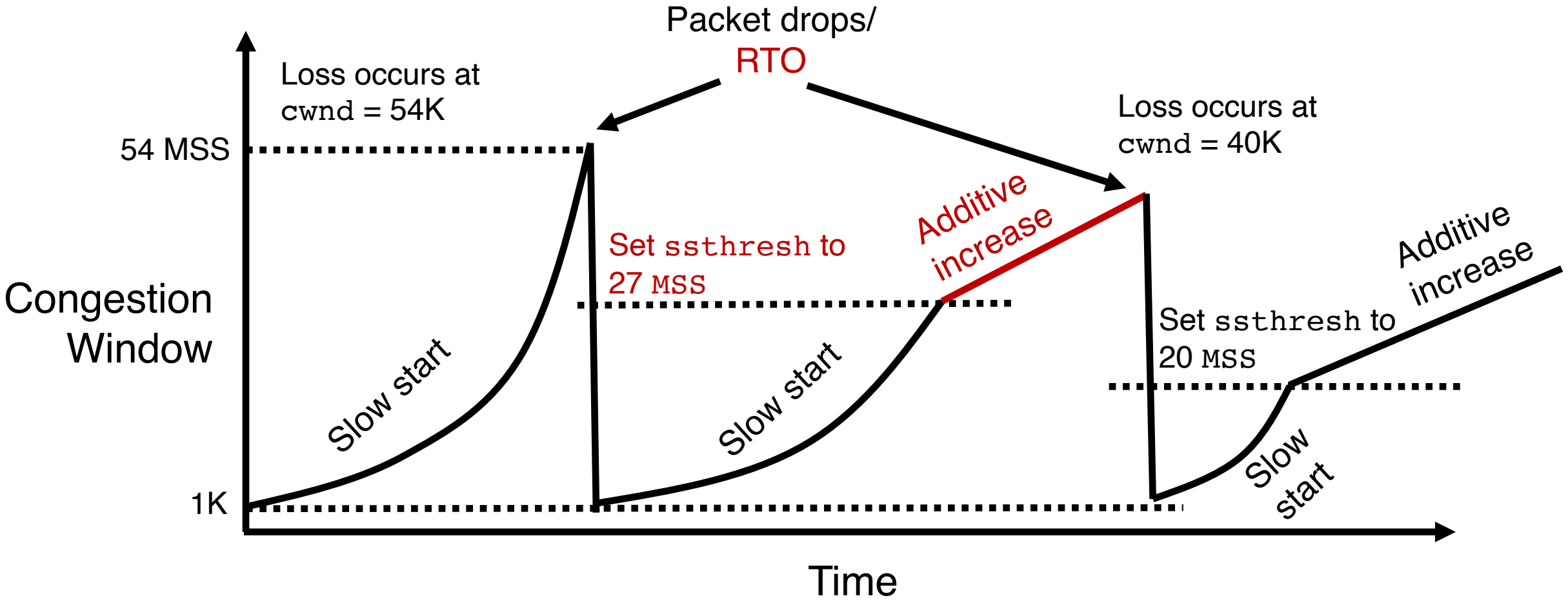
TCP New Reno: Additive increase

- Start with `ssthresh = 64K bytes` (TCP default)
- Do slow start until `ssthresh`
- Once the threshold is passed, do additive increase
  - Add one MSS to `cwnd` for each `cwnd` worth data ACK'ed
  - For each MSS ACK'ed, `cwnd = cwnd + (MSS * MSS) / cwnd`
- Upon a TCP timeout (RTO),
  - Set `cwnd = 1 MSS`
  - Set `ssthresh = max(2 * MSS, 0.5 * cwnd)`
  - i.e., the next linear increase will start at half the current `cwnd`

Behavior of Additive Increase

Say `MSS` = 1 KByte
Default `ssthresh` = 64KB = 64 `MSS`

AI is slow.
Persistent connections
Large window sizes
Different laws to evolve congestion window



Sample code

Bandwidth-Delay Product

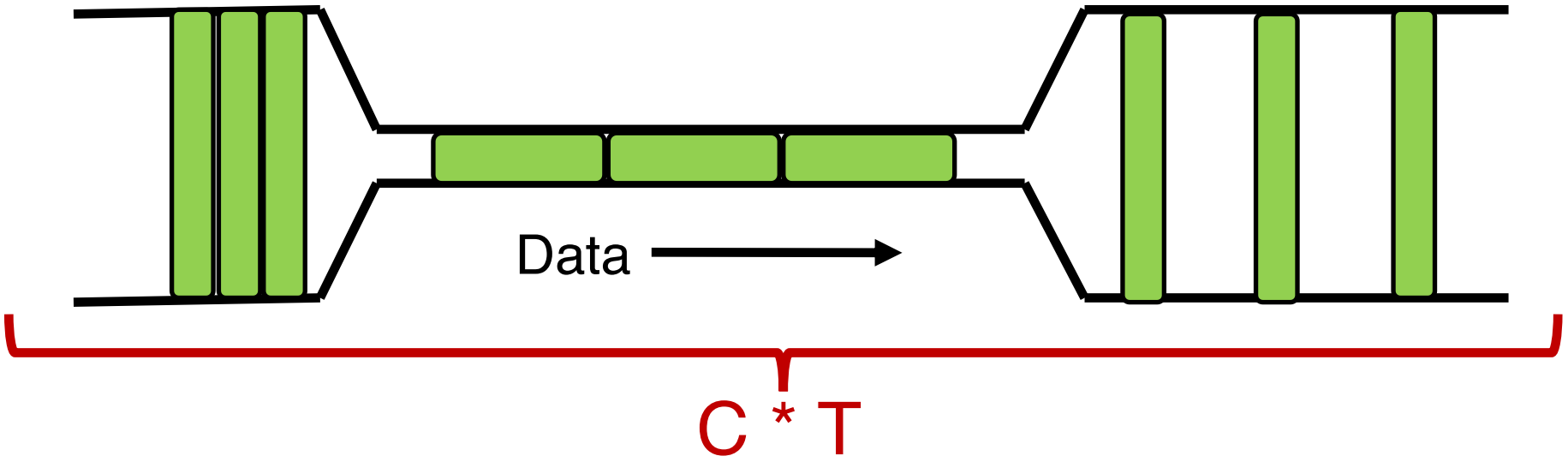
Steady state `cwnd` for a single flow

- Suppose the bottleneck link has rate C
- Suppose the propagation round-trip delay (propRTT) between sender and receiver is T
- Ignore transmission delays for this example;
- Assume steady state: highest sending rate with no bottleneck congestion
- Q: how much data is in flight over a single RTT?
- C * T data i.e., amount of data unACKed at any point in time
- ACKs take time T to arrive (without any queueing). In the meantime, sender is transmitting at rate C

The Bandwidth-Delay Product

- C * T = bandwidth-delay product:
  - The amount of data in flight for a sender transmitting at the ideal rate during the ideal round-trip delay of a packet

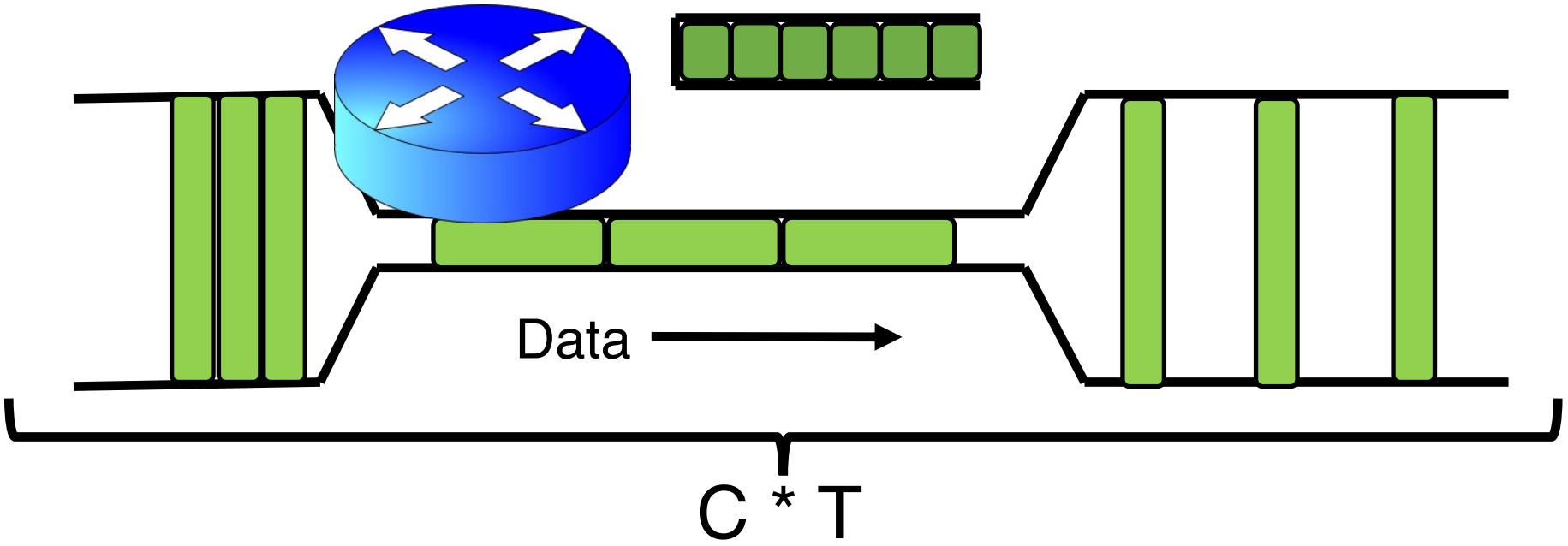
- Note: this is just the amount of data “on the pipe”



The Bandwidth-Delay Product

- Q: What happens if cwnd > C * T?
  - i.e., where are the rest of the in-flight packets?

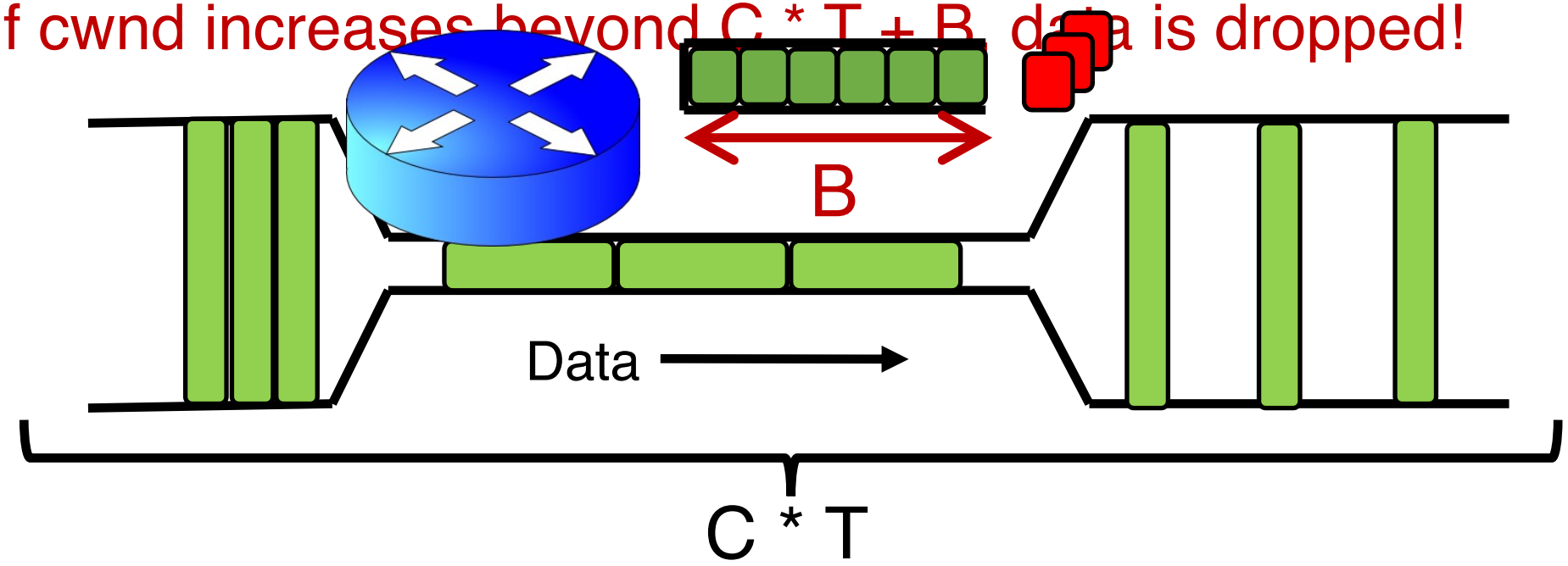
- A: Waiting at the bottleneck router queues



Router buffers and the max `cwnd`

- Router buffer memory is finite: queues can only be so long
  - If the router buffer size is B, there is at most B data waiting in the queue

- If cwnd increases beyond C * T + B, data is dropped!



BDP is a crucial value for a flow

- Bandwidth-Delay Product (BDP) governs the window size of a single flow at steady state
- The bottleneck router buffer size governs how much the `cwnd` can exceed the BDP before packet drops occur
- BDP is the ideal desired window size to use the full bottleneck link, without any queueing.
  - Accommodating flow control, also the min socket buffer size to use the bottleneck link fully

Detecting and Reacting Better to Packet Loss

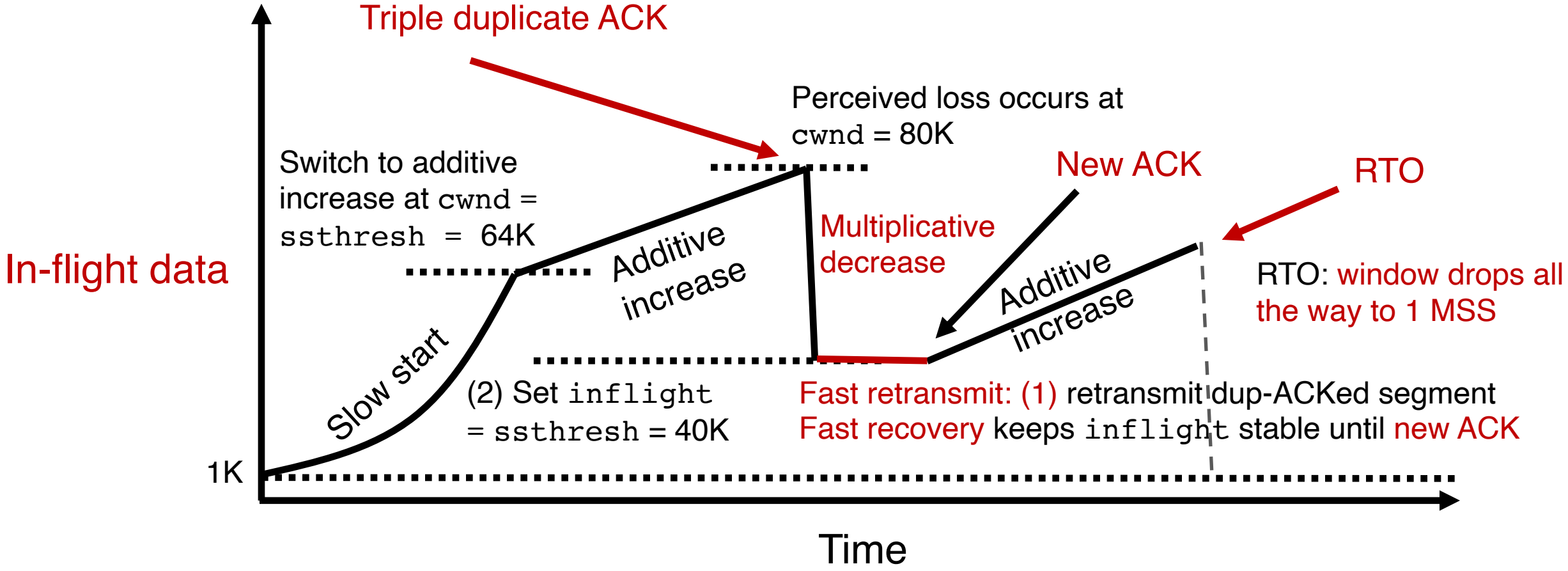
Can we detect loss earlier than RTO?

- Key idea: use the information in the ACKs. How?
- Suppose successive (cumulative) ACKs contain the same ACK#
  - Also called duplicate ACKs
  - Occur when network is reordering packets, or one (but not most) packets in the window were lost
- Reduce `cwnd` when you see many duplicate ACKs
  - Consider many dup ACKs a strong indication that packet was lost
  - Default threshold: 3 dup ACKs, i.e., triple duplicate ACK
  - Make cwnd reduction gentler than setting cwnd = 1; recover faster

Additive Increase/Multiplicative Decrease

Say `MSS` = 1 KByte

Default `ssthresh` = 64KB = 64 `MSS`



TCP New Reno performs additive increase and multiplicative decrease of congestion window.

In short, we often refer to this as AIMD.

Multiplicative decrease is a part of all TCP algorithms. It is necessary for fairness across TCP flows.

Sample code and demo