Socket programming

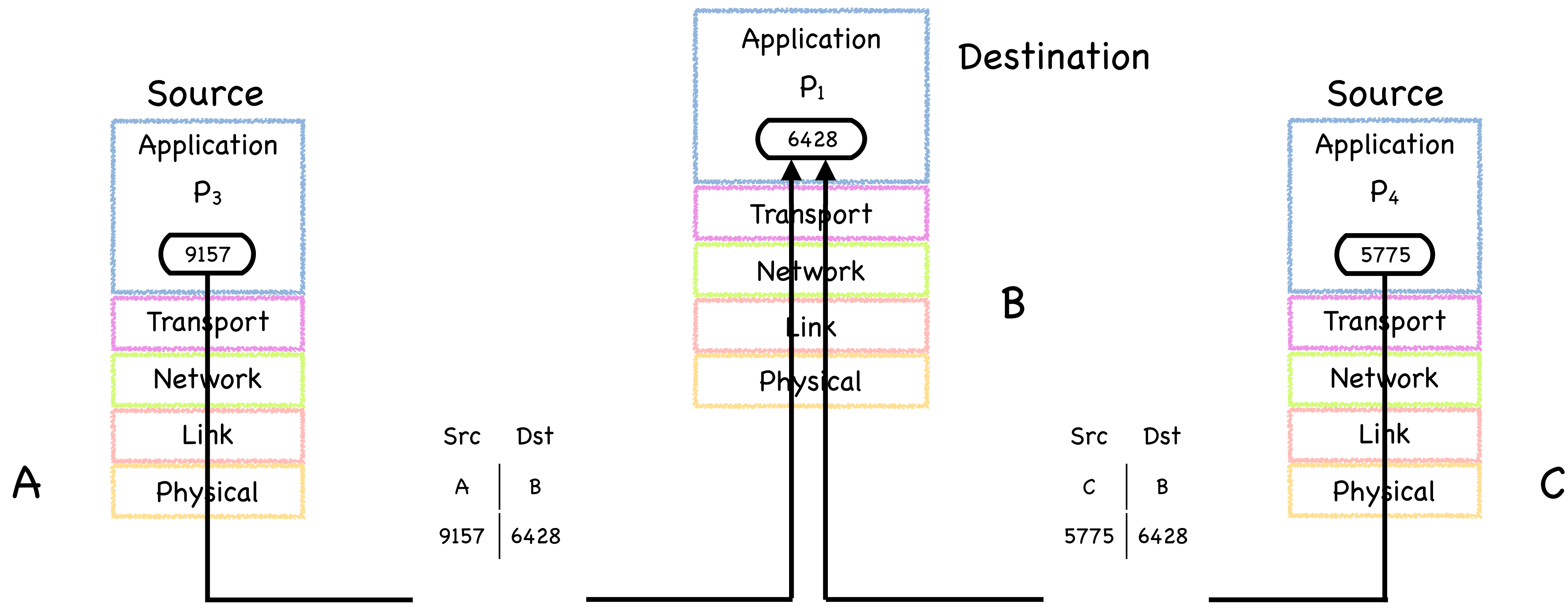
- Two socket types, depending on transport services
  - UDP: unreliable datagram
  - TCP: reliable, byte-stream oriented
- Application at end host distinguished by binding socket to a port number
  - 16 bit unsigned number; 0-1023 are bound to well-know applications
    - web server = 80; mail = 25; telnet = 23

Socket Programming with UDP

- No connection between client and server
  - no handshaking before sending data
  - Sender: explicitly attaches destination IP address and port number to each packet
  - Receiver: extracts sender IP address and port number from received packet
- Best effort: Data may be lost or received out-of-order
- UDP provides applications with unreliable transfer of a group of bytes (“datagram’) between client and server

Connectionless Demux

- Distinct UDP segments with same dest IP address and port, go to the same socket
    - even if they come from different source IP!
- The application must sort things out!



UDP: Perspective

- Speed
  - no connection establishment (takes time)
  - no congestion control: UDP can blast away!
- Simplicity
  - no connection state at sender/receiver
- Extra work for applications
  - reordering, duplicate suppression, missing packets...
  - but some applications may not care!
    - streaming multimedia: loss tolerant, rate sensitive (want constant, fast speeds)

Transmission Control Protocol (TCP)

- Reliable, ordered communication
- Adaptive protocol that delivers good-enough performance and handles congestion well
- All Web traffic travels over TCP/IP
  - Enough applications demand reliable ordered delivery that they should not have to implement their own protocol
  - ..but not really end-to-end (just socket to socket)

Socket Programming with TCP

- Client
  - Creates TCP socket with server's IP address and port number
  - Client TCP establishes connection to server TCP Server

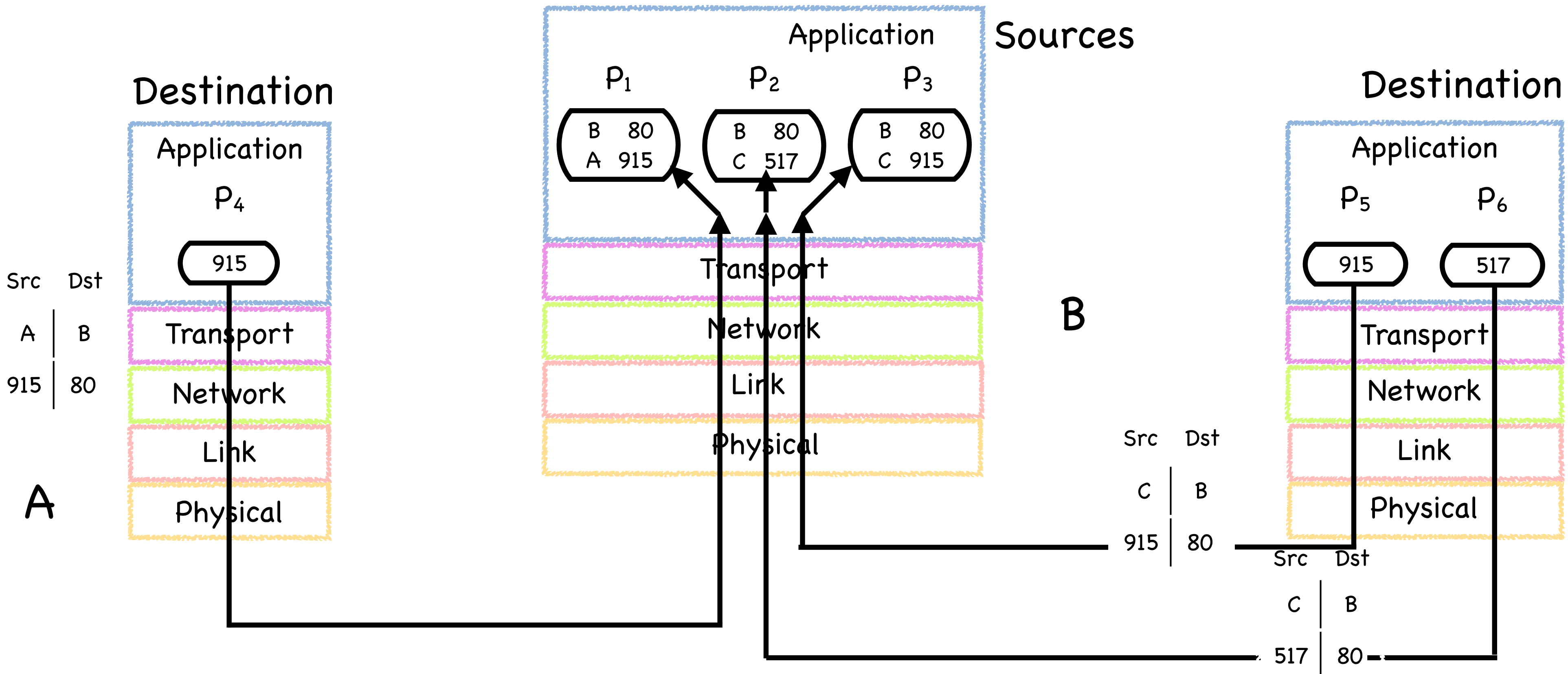
- Server
  - Contacted by client
  - Already running
  - Already created a "welcoming socket"
  - When contacted by client, creates a new TCP socket to communicate just with that client
    - Socket identified by 4-tuple
      - source IP; source port no; dest. IP; dest port no.
    - Server can concurrently serve multiple clients

Connection-Oriented Demux

Host receives three TCP segments

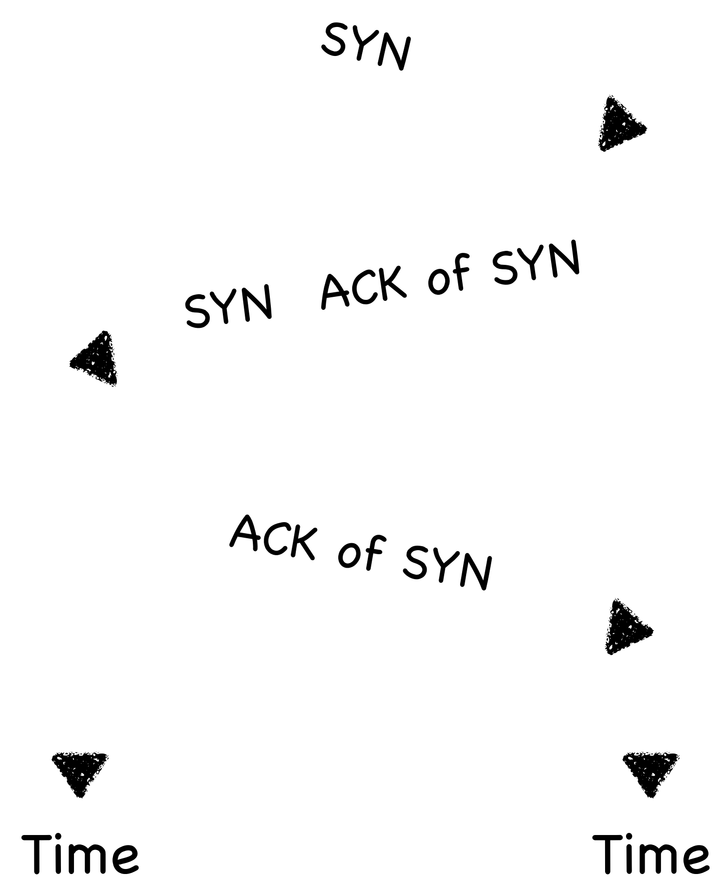
all destined to IP address B, port 80

demuxed to different sockets through socket's 4-tuple



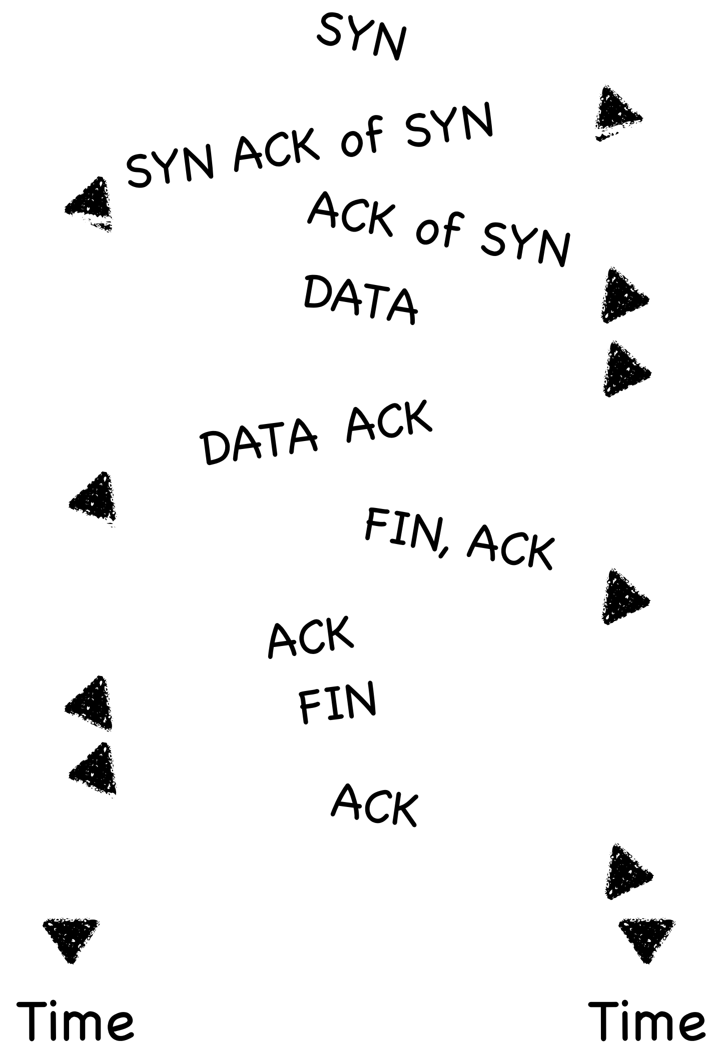
TCP Connections

- Initiated by a three-way handshake
    - 1.5 RTTs
    - create shared state on both side of connection
        - both sides know first sequence number to be used
        - both sides know other side is ready to receive



Typical TCP Usage

- Three round trips to
  - set up a connection
  - send a data packet
  - receive a response
  - tear down connection
- FINs tear down connection
  - Can be piggybacked on Ack



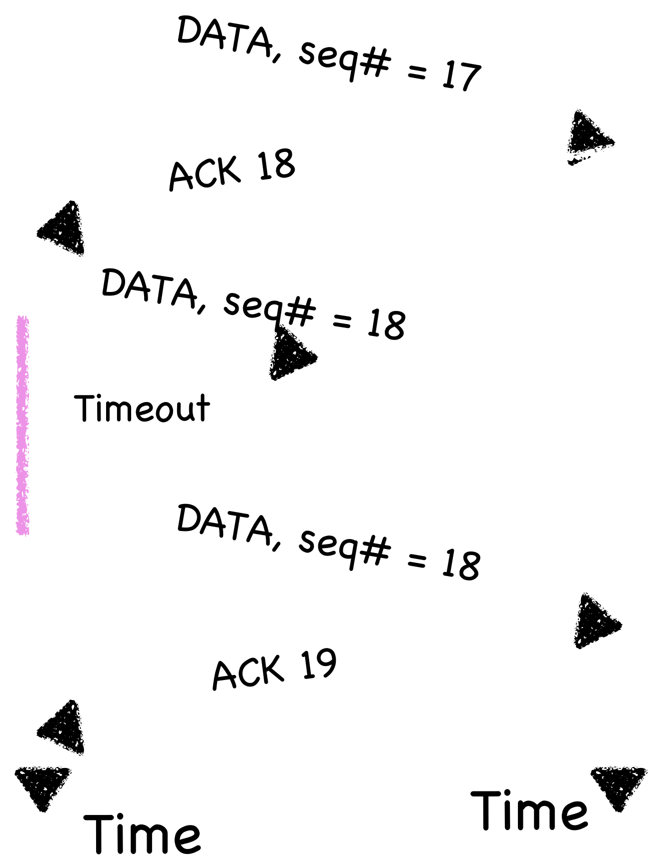
TCP Segments

- Each segment carries SEQ, a unique sequence number
    - initial value of SEQ chosen randomly
        - SEQ incremented by the data length
            - for simplicity, 4410 slides assume payloads of size 1
- Each segment carries an acknowledgement
    - acknowledge a set of packets by acking latest received SEQ
    - the acknowledgment is the sequence number of the next expected packet!

Reliable Transport

Here is joke about TCP.
Did you get it?
Did you get it?
Did you get it?
Did you get it?
Did you get it?

- TCP at sender keeps a copy of all sent, but unacknowledged, packets
- Packet resent if ACK does not arrive within a timeout
- Timeout interval adjusts to round-trip delay



$\text{AverageRTT} = (1 - \alpha)\ \text{OldAverageRTT} + \alpha\ \text{LatestRTT}$
$\text{AverageVar} = (1 - \beta)\ \text{OldAverageVar} + \beta\ \text{LatestVar}$

where $\text{LatestRTT} = (\text{ack\_receive\_time} - \text{send\_time})$,
$\text{LatestVar} = |\text{LatestRTT} - \text{AverageRTT}|$,
$\alpha = 1/8$, $\beta = ¼$ typically.

$\text{Timeout} = \text{AverageRTT} + 4 \times \text{AverageVar}$

How long does it take to send a segment?

Let
L: one-way latency (sec)
b: bandwith (bytes/sec)
S: Size of segment (bytes)

- Time between start sending and end receiving
  - L + S/b sec. (ignoring headers)
- Time before ack is received by sender: L sec
  - assuming acks are small
- End-to-end throughput
  - S/(2L+S/b) bytes/sec [goes to 0 as L grows]

Pipelining

- Sender allows multiple, "in flight", yet-to-be-acknowledged packets (a "window")
  - Increases throughput
  - Needs buffering at sender and receiver

Data packet

Ack packet

Data packet

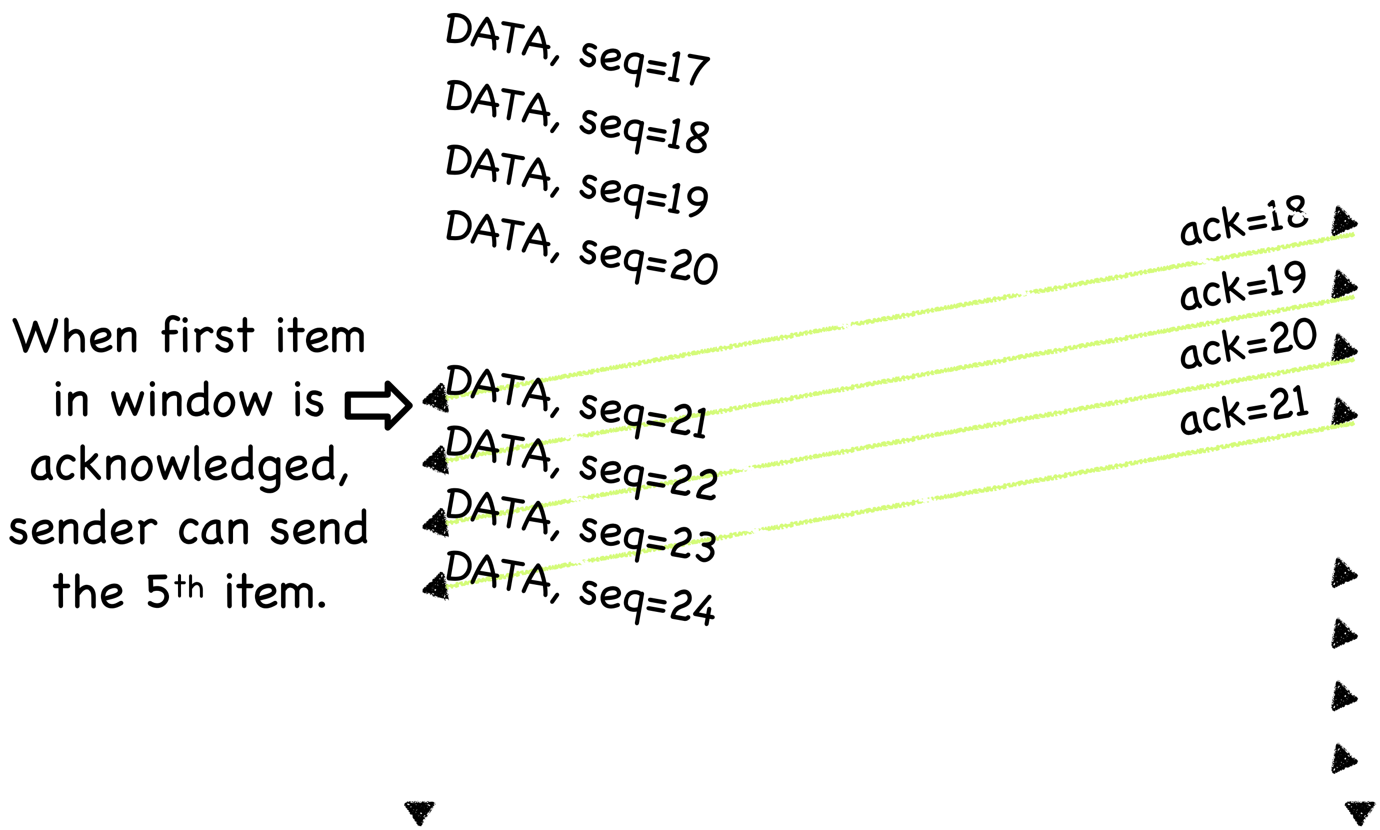
Ack packet

- How large should the window be?
- What if a packet in the middle is missing?

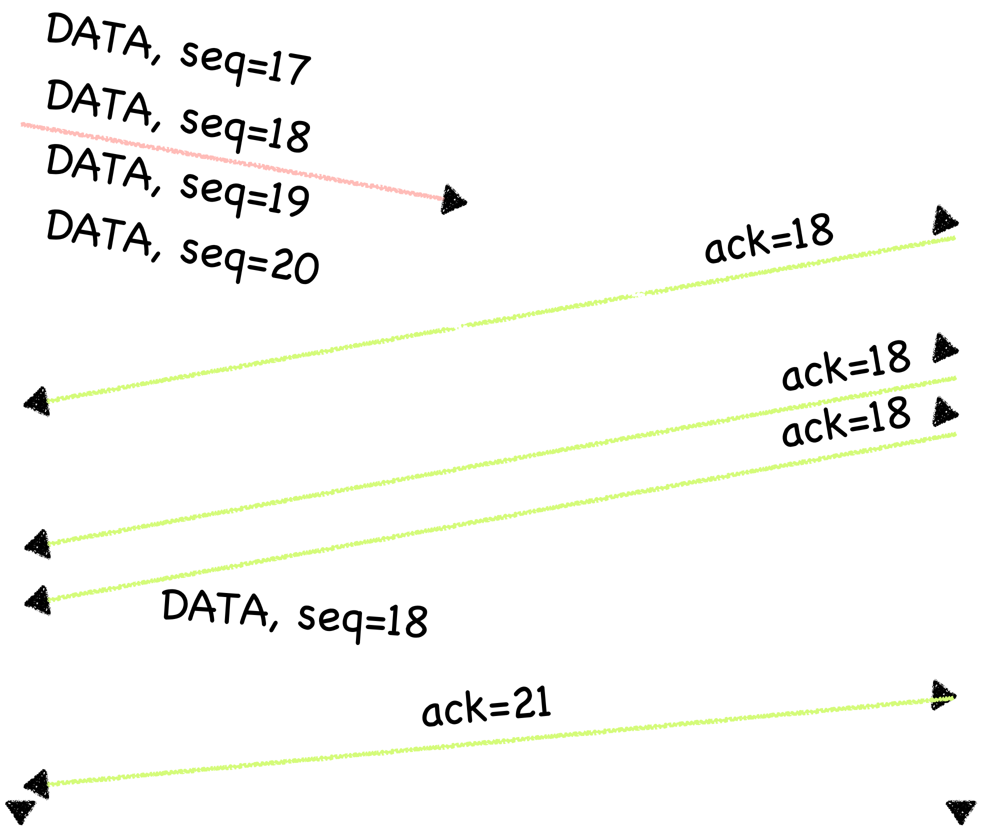
How Much Data "Fits" in a Pipe?

- Suppose
  - bandwidth is b bytes/sec
  - RTT is r seconds
  - ACK is a small message
- Then, can send $b \cdot r$ bytes before receiving ack for first byte...
  - of course, b and r can change over time...

TCP Window, Size 4



TCP Fast Retransmit



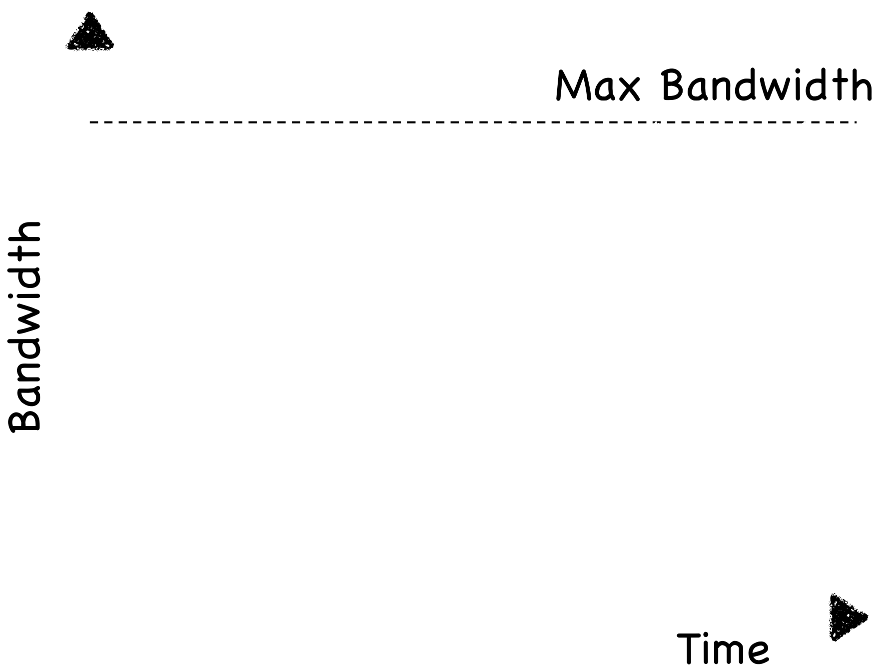
TCP Congestion Control

- Additive Increase/Multiplicative Decrease (AIMD)
  - window_size++ every RTT if no packet dropped
  - window_size/2 if packet is dropped
    - drop detected by acknowledgments
- Slowly builds to max bandwidth, and hovers there
  - – Does not achieve maximum bandwidth
  - + Shares bandwidth well with other TCP connections
- Policy of linear increase, exponential backoff under congestion known as TCP friendliness

TCP Window Size

Linear Increase
Exponential Backoff

Assuming losses in the network
only due to bandwidth



Window Size:

1, 2, 3, 4, 5, 6, 7, 8, 9, 10

5, 6, 7, 8, 9, 10

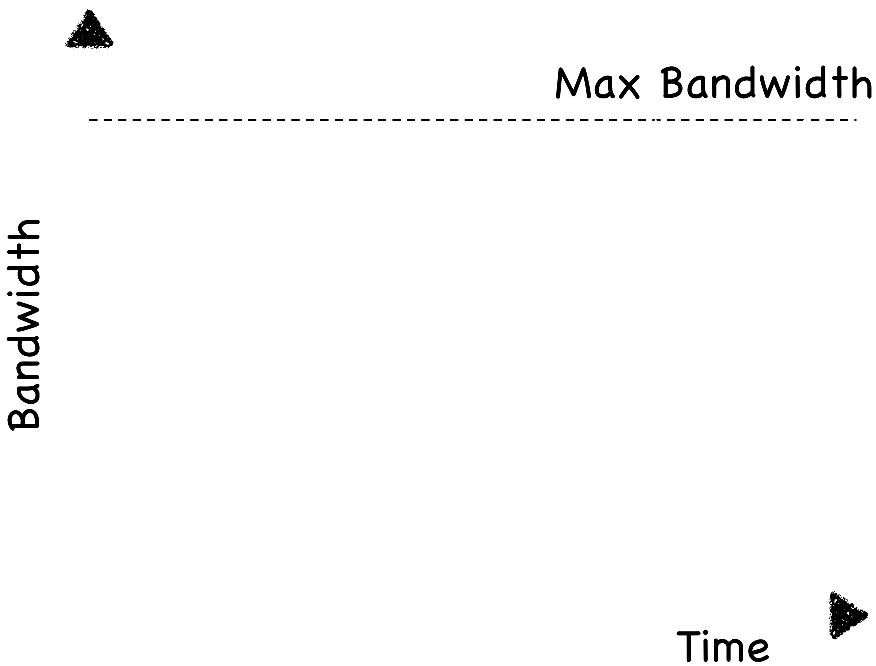
5, 6, 7, 8, 9, 10

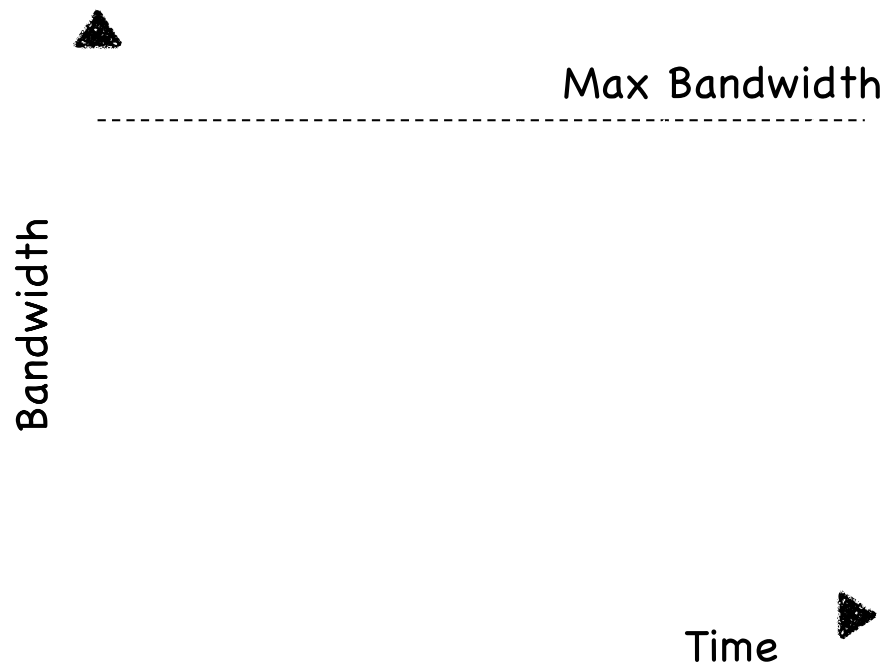
...

TCP Slow Start

- Linear Increase
  - Most file transactions end before that happens...
  - It takes long to reach window size that matches b·r
- Exponential Increase
  - TCP builds large window quickly by doubling window size for each ack received until first loss

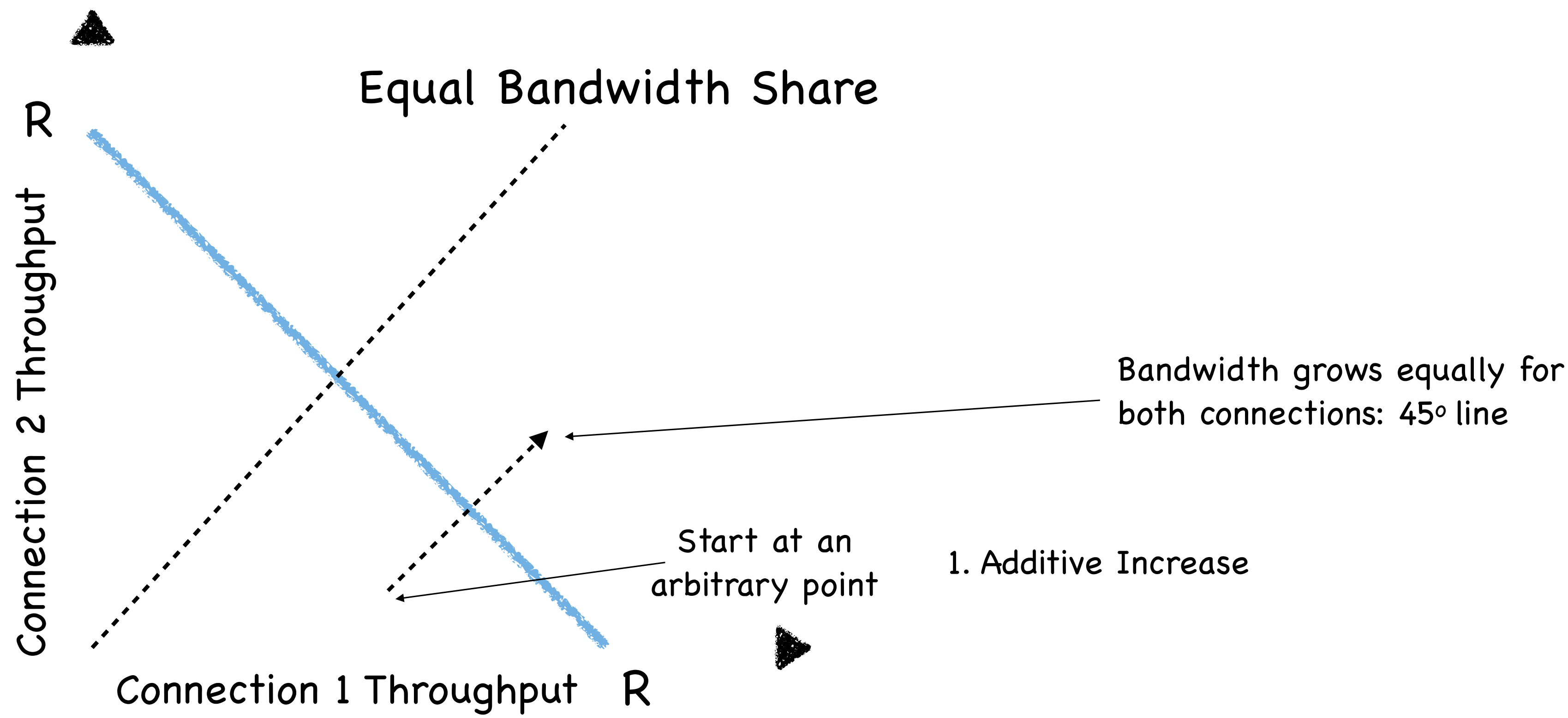
TCP Window Size with Exponential Start





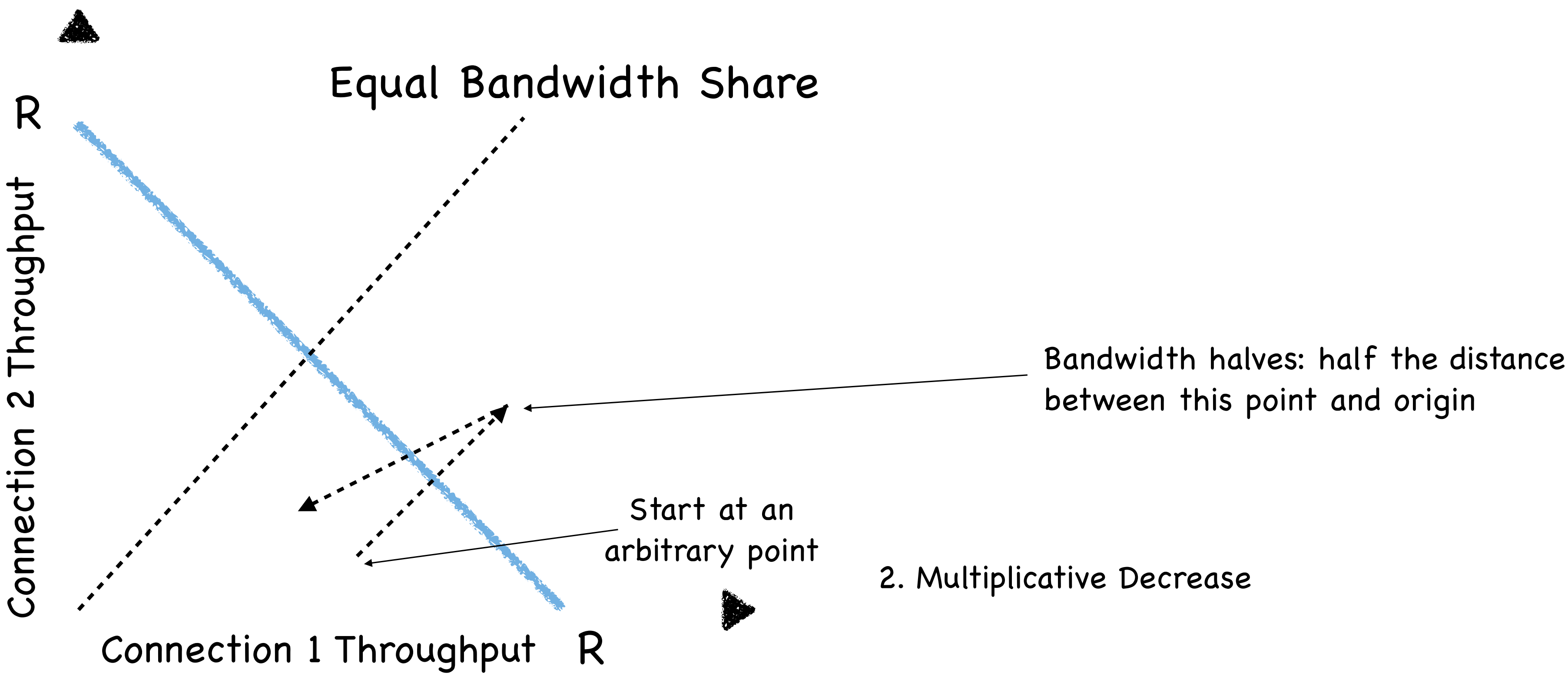
TCP Fairness

- If k TCP sessions share same bottleneck link of bandwidth R, each should have rate R/k
- IS AIMD fair?



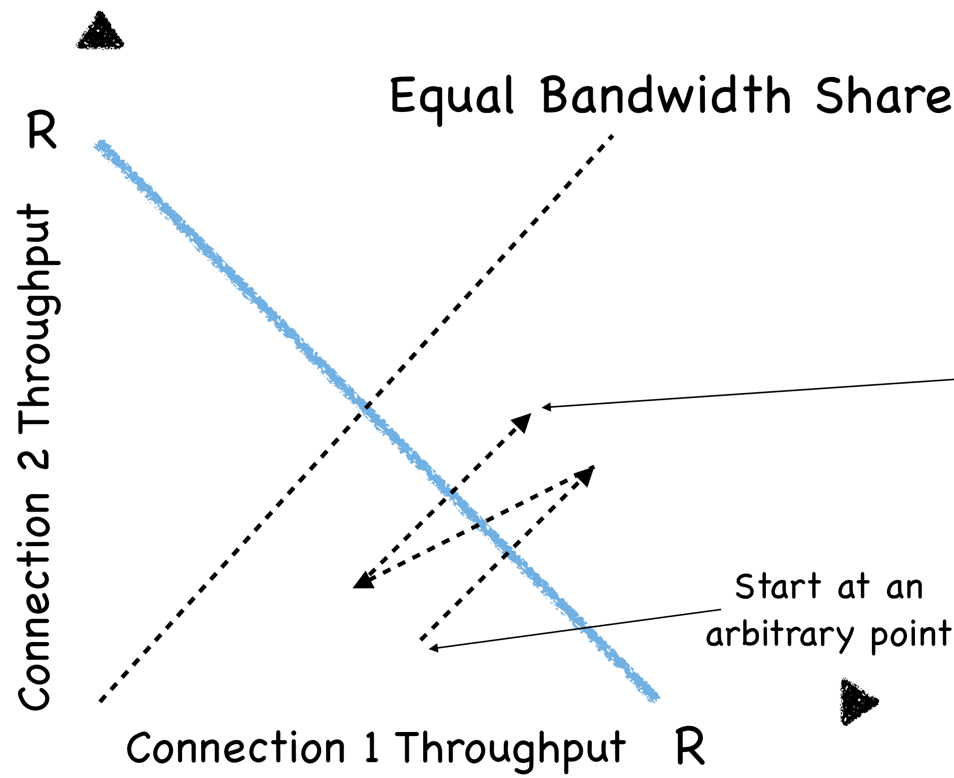
TCP Fairness

- If k TCP sessions share same bottleneck link of bandwidth R, each should have rate R/k
- IS AIMD fair?



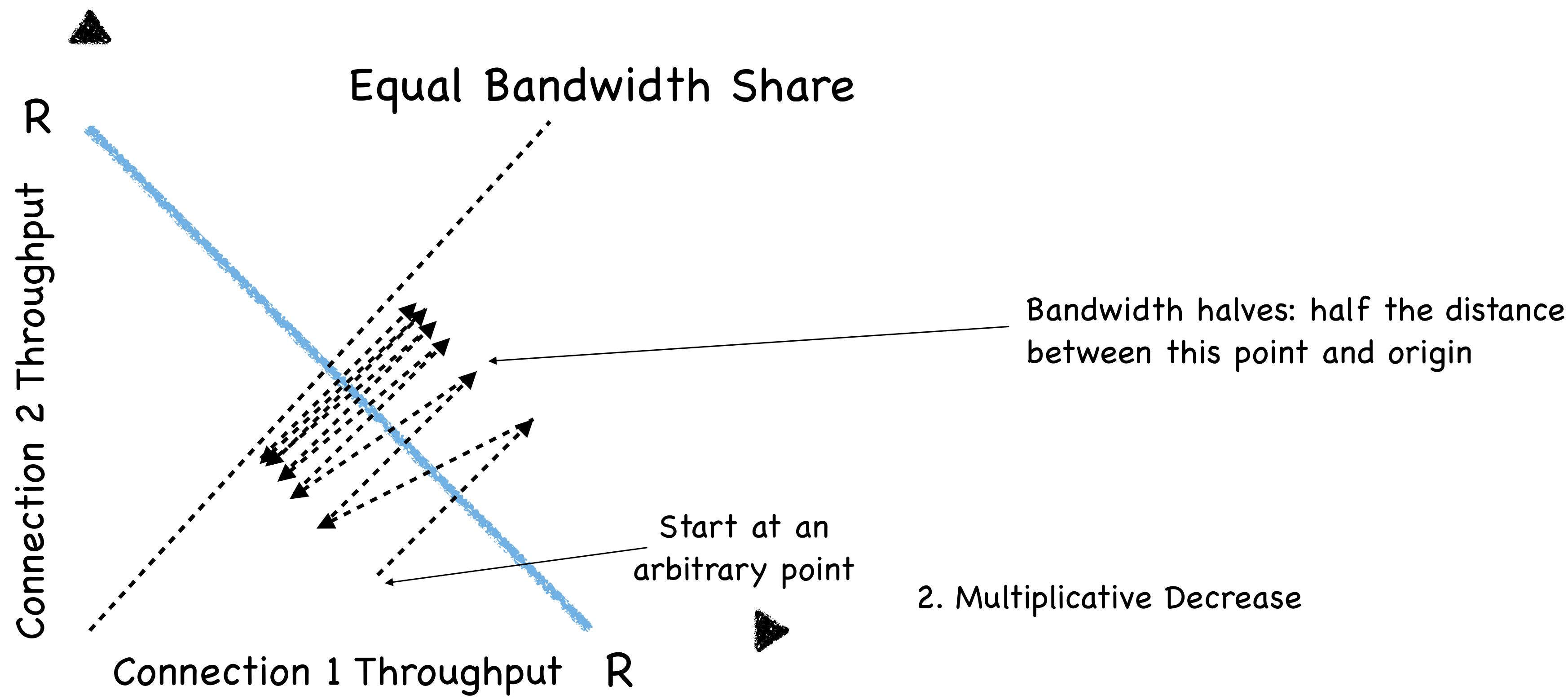
TCP Fairness

- If k TCP sessions share same bottleneck link of bandwidth R, each should have rate R/k
- IS AIMD fair?



TCP Fairness

- If k TCP sessions share same bottleneck link of bandwidth R, each should have rate R/k
- IS AIMD fair? Converges around equal bandwidth



TCP Summary

- Reliable ordered message delivery
  - Connection oriented, 3-way handshake
- Transmission window for better throughput
  - Timeouts based on link parameters
- Congestion control
  - Linear increase, exponential backoff
- Fast adaptation
  - Exponential increase in the initial phase