TCP/IP

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IP

- Internetworking protocol
    - Network layer
- Common packet format for the Internet
    - Specifies what packets look like
    - *Fragments* long packets into shorter packets
    - *Reassembles* fragments into original shape
- Some parts are fundamental, and some are arbitrary
    - IPv4 is what most people use
    - IPv6 clears up some of the messy parts, but is not yet in wide use

IPv4 packet layout

<table>
<tr><td>Version</td><td>IHL</td><td>TOS</td><td colspan="2">Total Length</td></tr>
<tr><td colspan="3">Identification</td><td>Flags</td><td>Fragment Offset</td></tr>
<tr><td colspan="2">TTL</td><td>Protocol</td><td colspan="2">Header Checksum</td></tr>
<tr><td colspan="5">Source Address</td></tr>
<tr><td colspan="5">Destination Address</td></tr>
<tr><td colspan="5">Options</td></tr>
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IP Fragmentation

- Networks have different maximum packet sizes
  - Big packets are sometimes desirable – less overhead
  - Huge packets are not desirable – reduced response time for others
- Higher level protocols (e.g. TCP or UDP) could figure out the max transfer unit and chop data into smaller packets
  - The endpoints do not necessarily know what the MTU is on the path
  - The route can change underneath
- Consequently, IP transparently fragments and reassembles packets

IP Fragmentation Mechanics

- IP divides a long datagram into N smaller datagrams
- Copies the header
- Assigns a Fragment ID to each part
- Sets the More Fragments bit
- Receiving end puts the fragments together based on the new IP headers
- Throws out fragments after a certain amount of time if they have not be reassembled

IP Options

- Source Routing: The source specifies the set of hosts that the packet should traverse
- Record Route: If this option appears in a packet, every router along a path attaches its own IP address to the packet
- Timestamp: Every router along the route attaches a timestamp to the packet
- Security: Packets are marked with user info, and the security classification of the person on whose behalf they travel on the network
    - Most of these options pose security holes and are generally not implemented

UDP

- Unreliable Datagram Protocol
- IP goes from host to host
- We need a way to get datagrams from one application to another
- How do we identify applications on the hosts ?
    - Assign *port numbers*
    - E.g. port 13 belongs to the time service

UDP Packet Layout

<table>
<tr><td rowspan="5">IP</td><td>Version</td><td>IHL</td><td>TOS</td><td colspan="3">Total Length</td></tr>
<tr><td colspan="3">Identification</td><td colspan="2">Flags</td><td>Fragment Offset</td></tr>
<tr><td colspan="2">TTL</td><td>Protocol</td><td colspan="3">Header Checksum</td></tr>
<tr><td colspan="6">Source Address</td></tr>
<tr><td colspan="6">Destination Address</td></tr>
<tr><td rowspan="2">UDP</td><td colspan="4">Source Port</td><td colspan="2">Destination Port</td></tr>
<tr><td colspan="4">Length</td><td colspan="2">Checksum</td></tr>
<tr><td></td><td colspan="6">Data</td></tr>
</table>

- UDP adds Ports, Data Length and Data checksum

UDP

- UDP is unreliable
  - A UDP packet may get dropped at any time
  - It may get duplicated
  - A series of UDP packets may get reordered
- Applications need to deal with reordering, duplicate suppression, reliable delivery
  - Some apps can ignore these effects and still function
- Unreliable datagrams are the bare-bones network service
  - Good to build on, esp for multimedia applications

TCP

- Transmission Control Protocol
    - Reliable, ordered communication
- Enough applications demand reliable ordered delivery that they should not have to implement their own protocol
- A standard, adaptive protocol that delivers good-enough performance and deals well with congestion
- All web traffic travels over TCP/IP

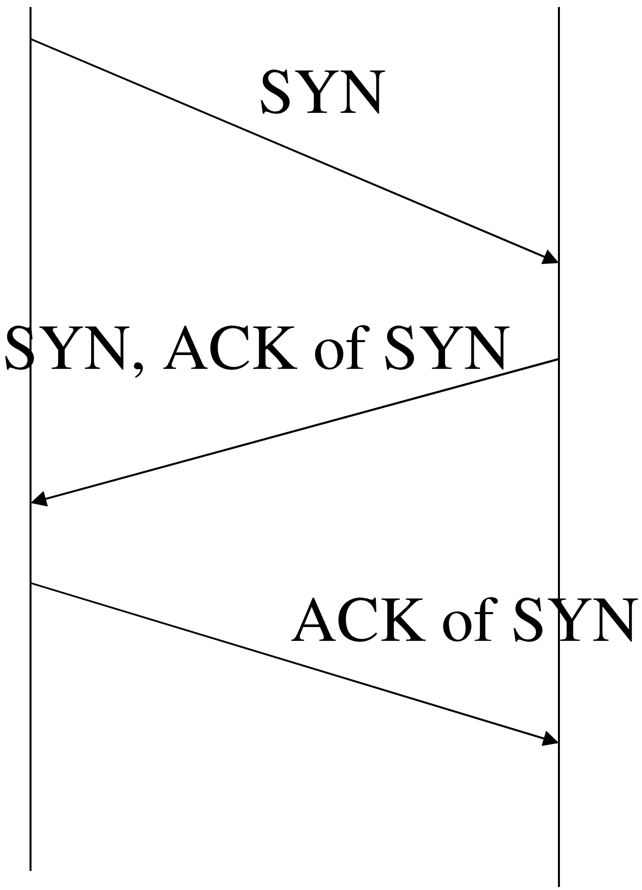
TCP/IP Packets

IP	Version	IHL	TOS	Total Length
	Identification		Flags	Fragment Offset
	TTL	Protocol	Header Checksum	
	Source Address			
	Destination Address			
TCP	Source Port		Destination Port	
	Sequence Number			
	Acknowledgement Number			
	Offset	ACK|URG|SYN|FIN|RST	Window	
	Checksum		Urgent Pointer	
	Options		Padding	
	Data			

TCP Packets

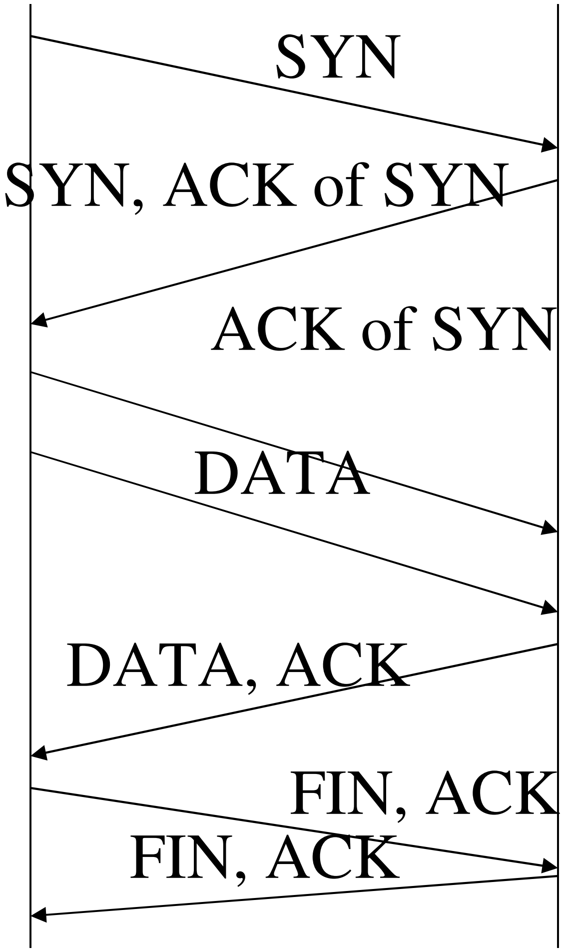
- Each packet carries a unique ID
  - The initial number is chosen randomly
  - The ID is incremented by the data length
- Each packet carries an acknowledgement
  - Can acknowledge a set of packets by ack’ing the latest one received
- Reliable transport is implemented using these identifiers

TCP Connections



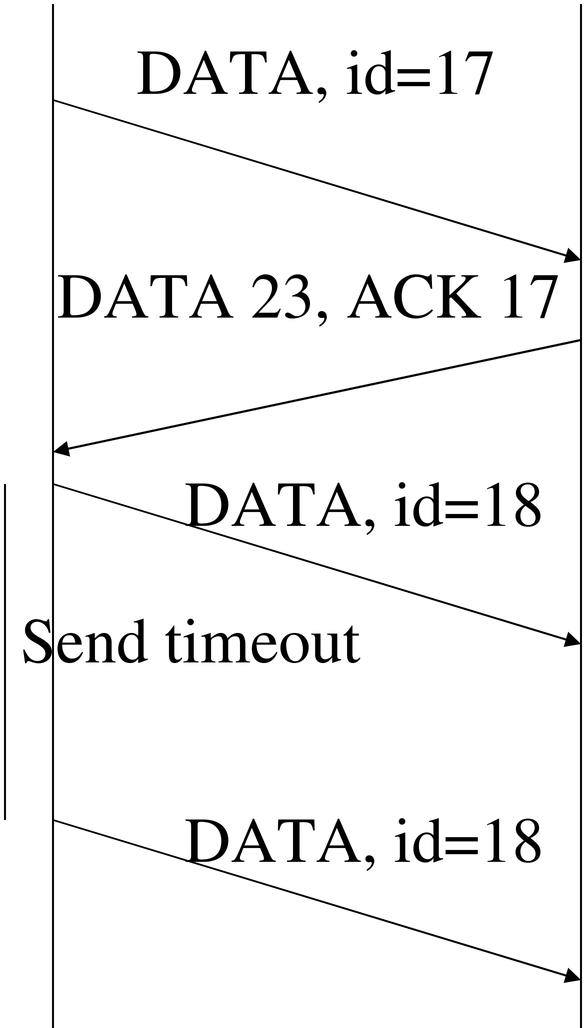
- TCP is *connection* oriented
- A connection is initiated with a three-way handshake
- Three-way handshake ensures against duplicate SYN packets
- Takes 3 packets, 1.5 RTT

Typical TCP Usage



- Three round-trips to set up a connection, send a data packet, receive a response, tear down connection
- FINs work (mostly) like SYNs to tear down connection
    - Need to wait after a FIN for straggling packets

Reliable transport



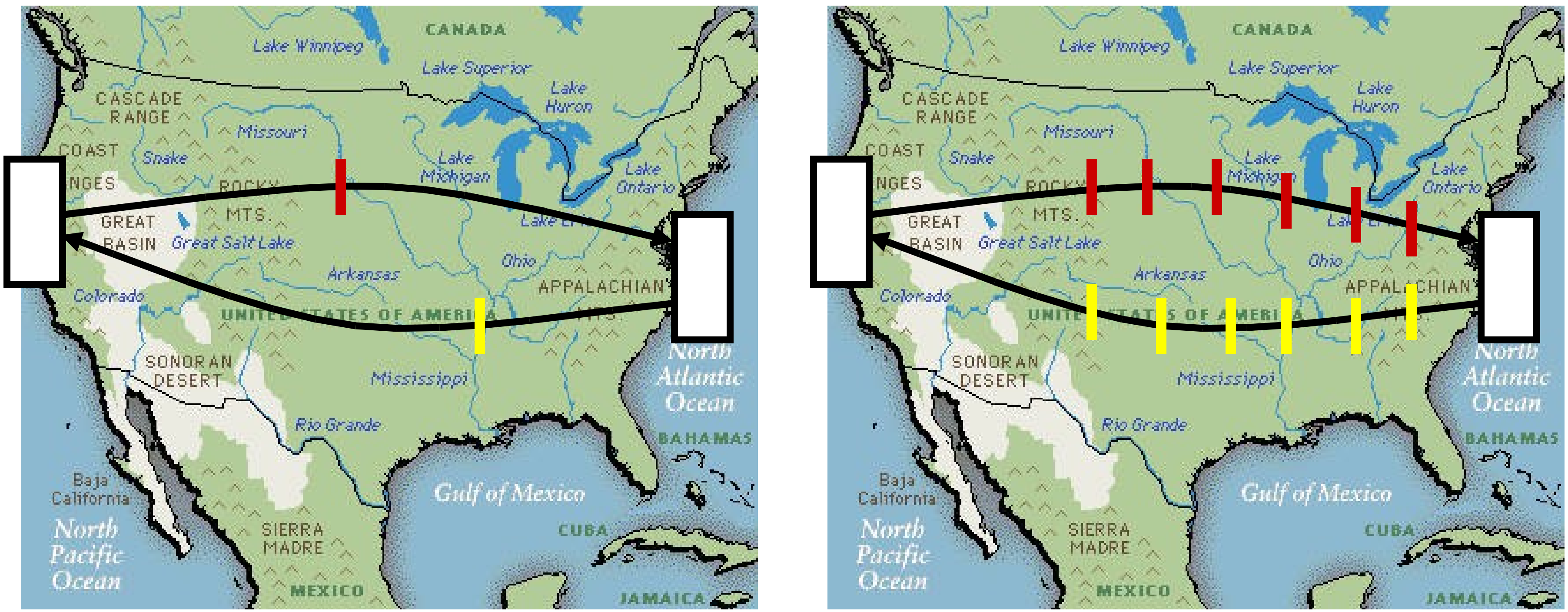
- TCP keeps a copy of all sent, but unacknowledged packets
- If acknowledgement does not arrive within a “send timeout” period, packet is resent
- Send timeout adjusts to the round-trip delay

TCP timeouts

- Timeout is thus a function of RTT and deviation

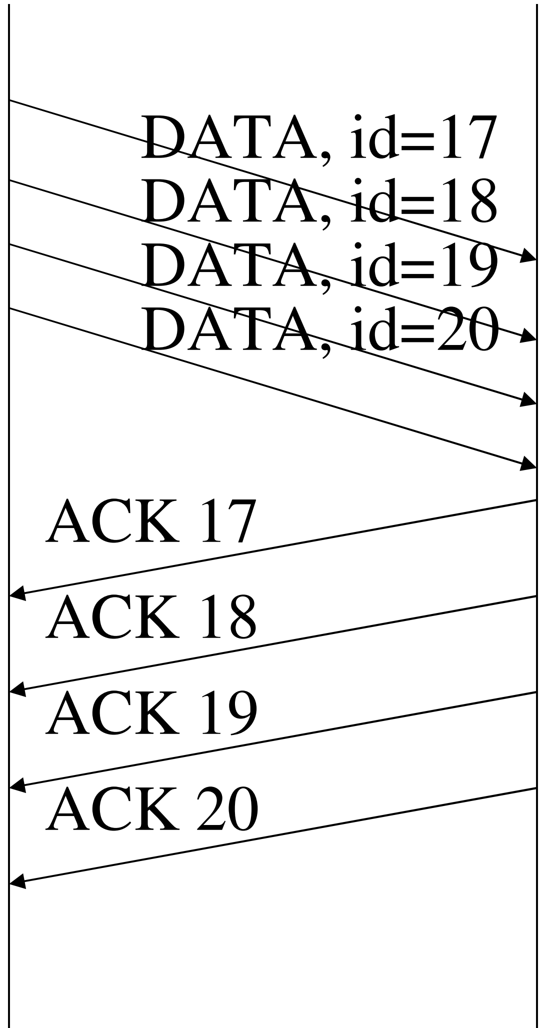
NewAverageRTT = (1 - α) OldAverageRTT + α LatestRTT
NewAverageDev = (1 - α) OldAverageDev + α LatestDev
where LatestRTT = (ack_receive_time – send_time),
LatestDev = |LatestRTT – AverageRTT|,
α = 1/8, typically.
Timeout = AverageRTT + 4*AverageDev

TCP Windows



- Multiple outstanding packets can increase throughput

TCP Windows

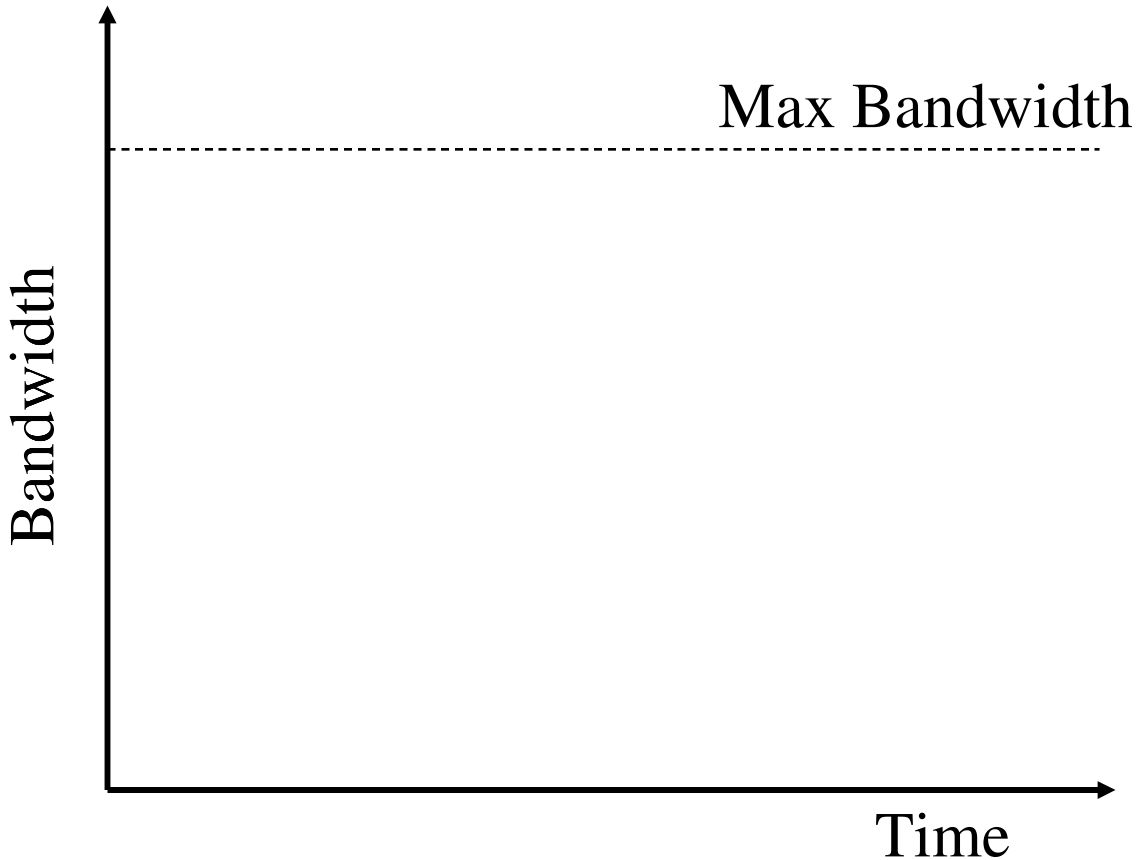


- Can have more than one packet in transit
- Especially over fat pipes, e.g. satellite connection
- Need to keep track of all packets within the window
- Need to adjust window size

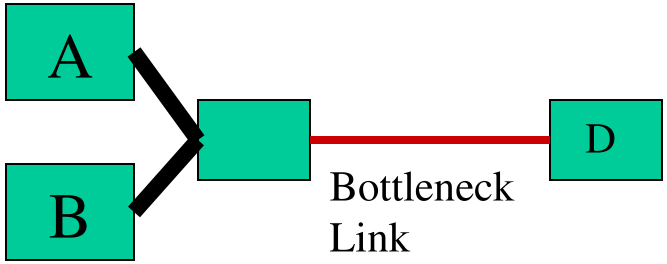
TCP Congestion Control

- TCP Increases its window size as long as no packets are dropped
- It halves the window size when a packet drop occurs
    - A packet drop is evident from the acknowledgements
- Therefore, it will slowly build up to the max bandwidth, and hover around the max
    - It doesn't achieve the max possible though
    - Instead, it shares the bandwidth well with other TCP connections
- This linear-increase, exponential backoff in the face of congestion is termed *TCP-friendliness*

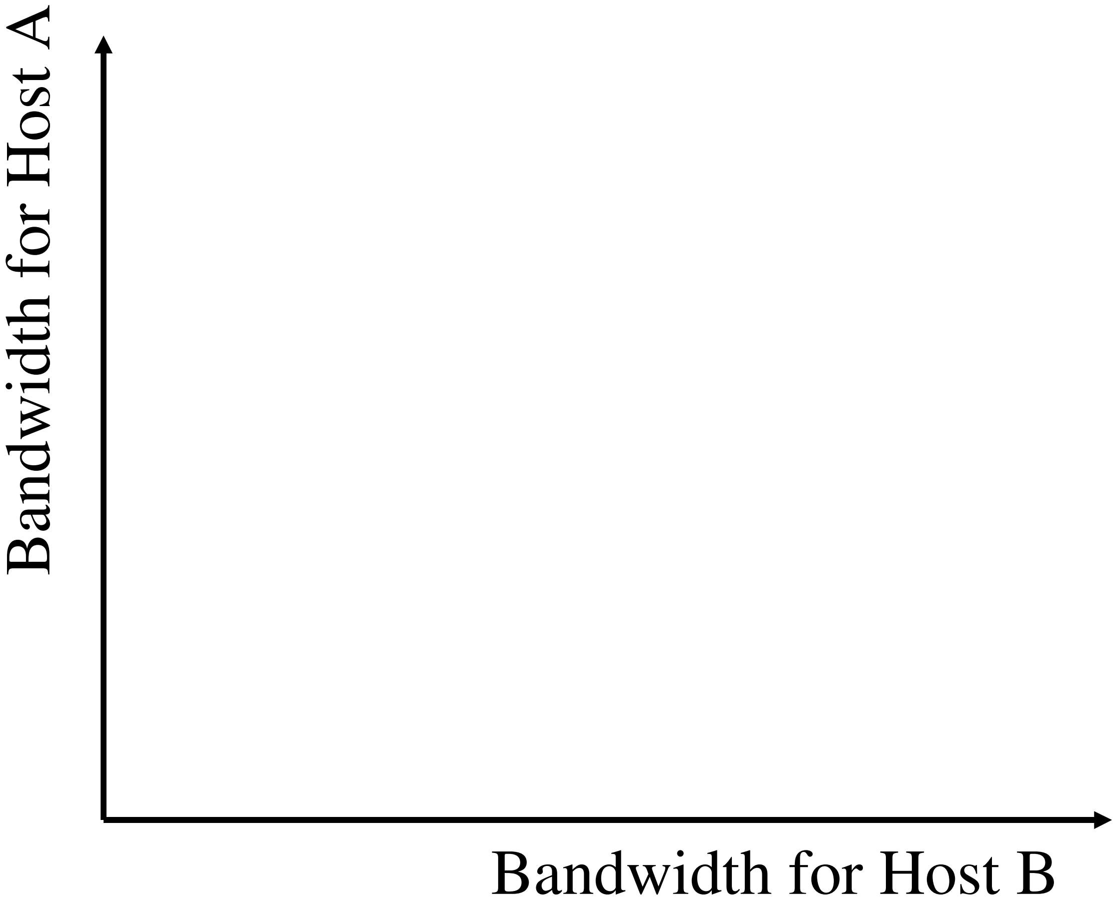
TCP Window Size



- Linear increase
- Exponential backoff
- Assuming no other losses in the network except those due to bandwidth



TCP Fairness

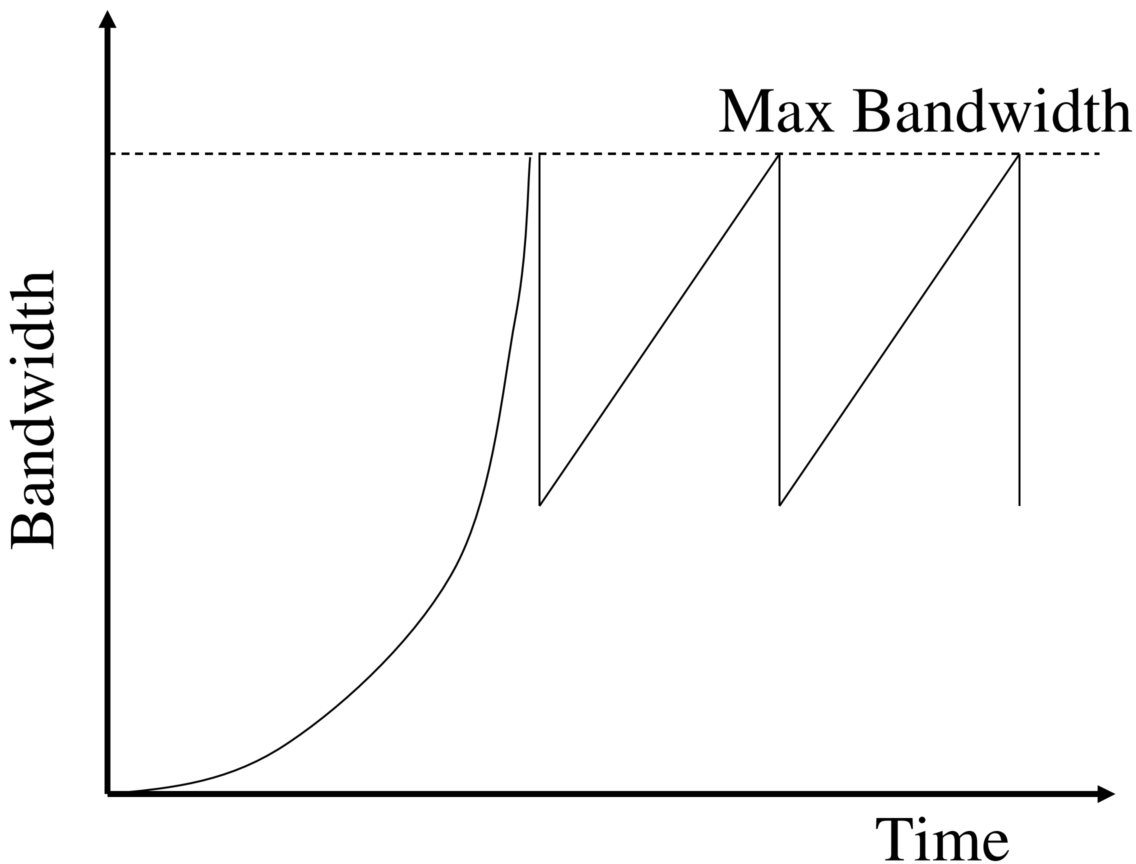


- Want to share the bottleneck link fairly between two flows

TCP Slow Start

- Linear increase takes a long time to build up a window size that matches the link bandwidth*delay
- Most file transactions are not long enough
- Consequently, TCP can spend a lot of time with small windows, never getting the chance to reach a sufficiently large window size
- Fix: Allow TCP to build up to a large window size initially by doubling the window size until first loss

TCP Slow Start



- Initial phase of exponential increase
- Assuming no other losses in the network except those due to 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