Transport Layer Protocols

EE3204: Computer Communication Networks I
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Outline

- Principles underlying transport-layer services
  - (De)multiplexing
  - Detecting corruption
  - Reliable delivery
  - Flow control
- Transport-layer protocols in the Internet
  - User Datagram Protocol (UDP)
  - Transmission Control Protocol (TCP)

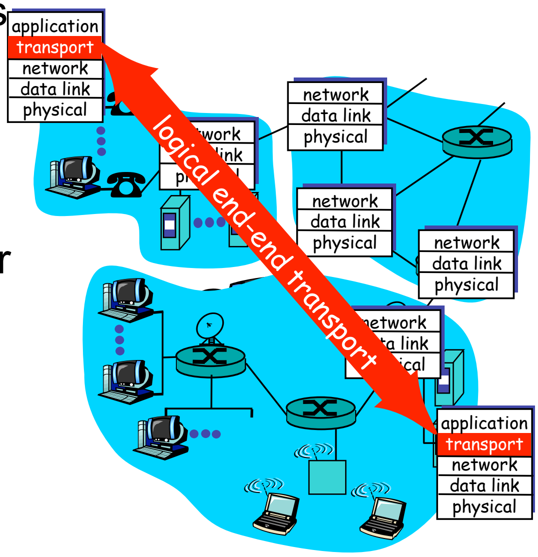
Note: Some slides & graphics adapted from Kurose & Ross, Computer Networking

Role of Transport Layer

- Application layer
  - Communication for specific applications
  - E.g., HyperText Transfer Protocol (HTTP), File Transfer Protocol (FTP), Network News Transfer Protocol (NNTP)
- Transport layer
  - Communication between processes (e.g., socket)
  - Relies on network layer and serves the application layer
  - E.g., TCP and UDP
- Network layer
  - Logical communication between nodes
  - Hides details of the link technology
  - E.g., IP

Transport Protocols

- Provide *logical communication* between application processes running on different hosts
- Run on end hosts
  - Sender: breaks application messages into segments, and passes to network layer
  - Receiver: reassembles segments into messages, passes to application layer
- Multiple transport protocol available to applications
  - Internet: TCP and UDP

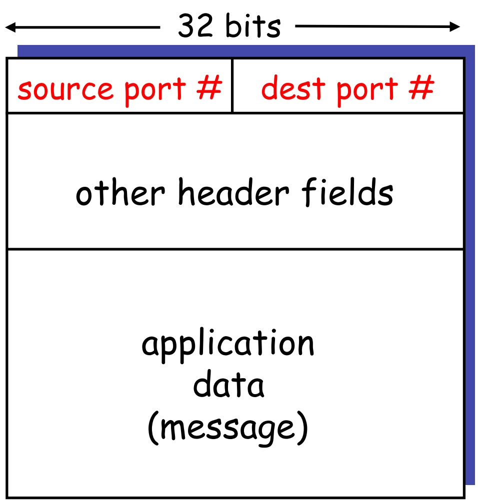


Internet Transport Protocols

- Datagram messaging service (UDP)
  - No-frills extension of “best-effort” IP
- Reliable, in-order delivery (TCP)
  - Connection set-up
  - Discarding of corrupted packets
  - Retransmission of lost packets
  - Flow control
  - Congestion control
- Other services not available
  - Delay guarantees
  - Bandwidth guarantees

Multiplexing and Demultiplexing

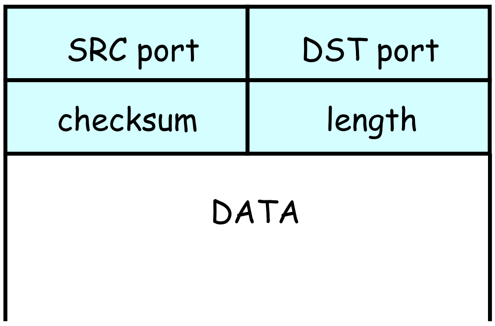
- Host receives IP datagrams
  - Each datagram has source and destination IP address,
  - Each datagram carries one transport-layer segment
  - Each segment has source and destination port number
- Host uses IP addresses and port numbers to direct the segment to appropriate socket



TCP/UDP segment format

Unreliable Message Delivery Service

- Lightweight communication between processes
  - Avoid overhead and delays of ordered, reliable delivery
  - Send messages to and receive them from a socket
- User Datagram Protocol (UDP)
  - IP plus port numbers to support (de)multiplexing
  - Optional error checking on the packet contents

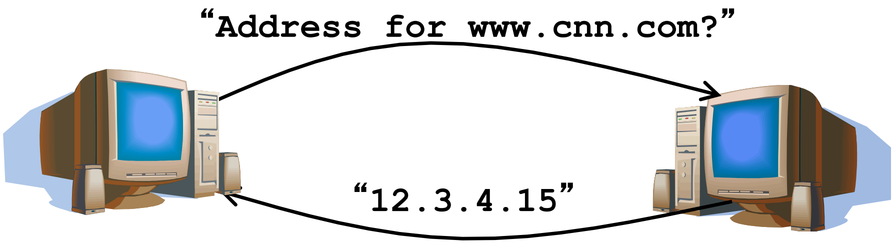


Why Would Anyone Use UDP?

- Finer control over what data is sent and when
  - As soon as an application process writes into the socket
  - … UDP will package the data and send the packet
- No delay for connection establishment
  - UDP just blasts away without any formal preliminaries
  - … which avoids introducing any unnecessary delays
- No connection state
  - No allocation of buffers, parameters, sequence #s, etc.
  - … making it easier to handle many active clients at once
- Small packet header overhead
  - UDP header is only eight-bytes long

Popular Applications That Use UDP

- Multimedia streaming
  - Retransmitting lost/corrupted packets is not worthwhile
  - By the time the packet is retransmitted, it's too late
  - E.g., telephone calls, video conferencing, gaming
- Simple query protocols like Domain Name System
  - Overhead of connection establishment is overkill
  - Easier to have application retransmit if needed



Transmission Control Protocol (TCP)

- Connection oriented
  - Explicit set-up and tear-down of TCP session
- Stream-of-bytes service
  - Sends and receives a stream of bytes, not messages
- Reliable, in-order delivery
  - Checksums to detect corrupted data
  - Acknowledgments & retransmissions for reliable delivery
  - Sequence numbers to detect losses and reorder data
- Flow control
  - Prevent overflow of the receiver's buffer space
- Congestion control
  - Adapt to network congestion for the greater good

An Analogy: Talking on a Cell Phone

- Alice and Bob on their cell phones
  - Both Alice and Bob are talking
- What if Alice couldn't understand Bob?
  - Bob asks Alice to repeat what she said
- What if Bob hasn't heard Alice for a while?
  - Is Alice just being quiet?
  - Or, have Bob and Alice lost reception?
  - How long should Bob just keep on talking?
  - Maybe Alice should periodically say "uh huh"
  - … or Bob should ask "Can you hear me now?" ☺

Some Reflections on the Example

- Acknowledgments from receiver
  - Positive: "okay" or "ACK"
  - Negative: "please repeat that" or "NACK"
- Timeout by the sender ("stop and wait")
  - Don't wait indefinitely without receiving some response
  - … whether a positive or a negative acknowledgment
- Retransmission by the sender
  - After receiving a "NACK" from the receiver
  - After receiving no feedback from the receiver

Challenges of Reliable Data Transfer

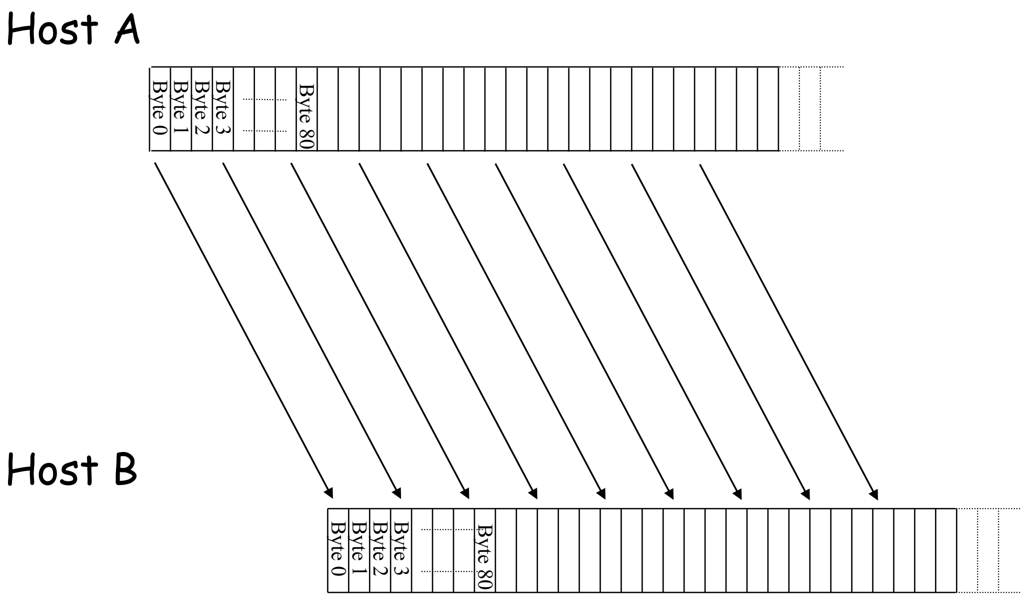
- Over a perfectly reliable channel
  - All of the data arrives in order, just as it was sent
  - Simple: sender sends data, and receiver receives data
- Over a channel with bit errors
  - All of the data arrives in order, but some bits corrupted
  - Receiver detects errors and says "please repeat that"
  - Sender retransmits the data that were corrupted
- Over a lossy channel with bit errors
  - Some data are missing, and some bits are corrupted
  - Receiver detects errors but cannot always detect loss
  - Sender must wait for acknowledgment ("ACK" or "OK")
  - … and retransmit data after some time if no ACK arrives

TCP Support for Reliable Delivery

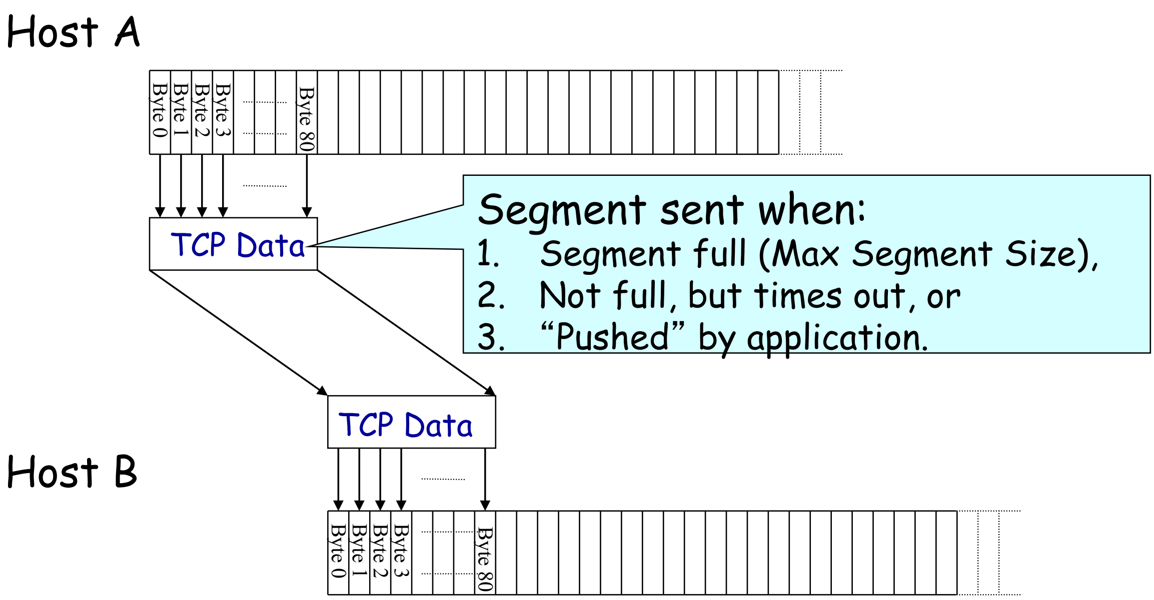
- Checksum
  - Used to detect corrupted data at the receiver
  - …leading the receiver to drop the packet
- Sequence numbers
  - Used to detect missing data
  - ... and for putting the data back in order
- Retransmission
  - Sender retransmits lost or corrupted data
  - Timeout based on estimates of round-trip time
  - Fast retransmit algorithm for rapid retransmission

TCP Segments

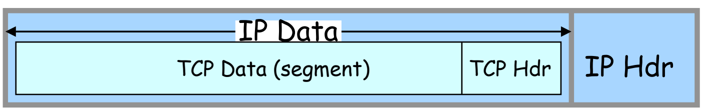
TCP "Stream of Bytes" Service



…Emulated Using TCP "Segments"

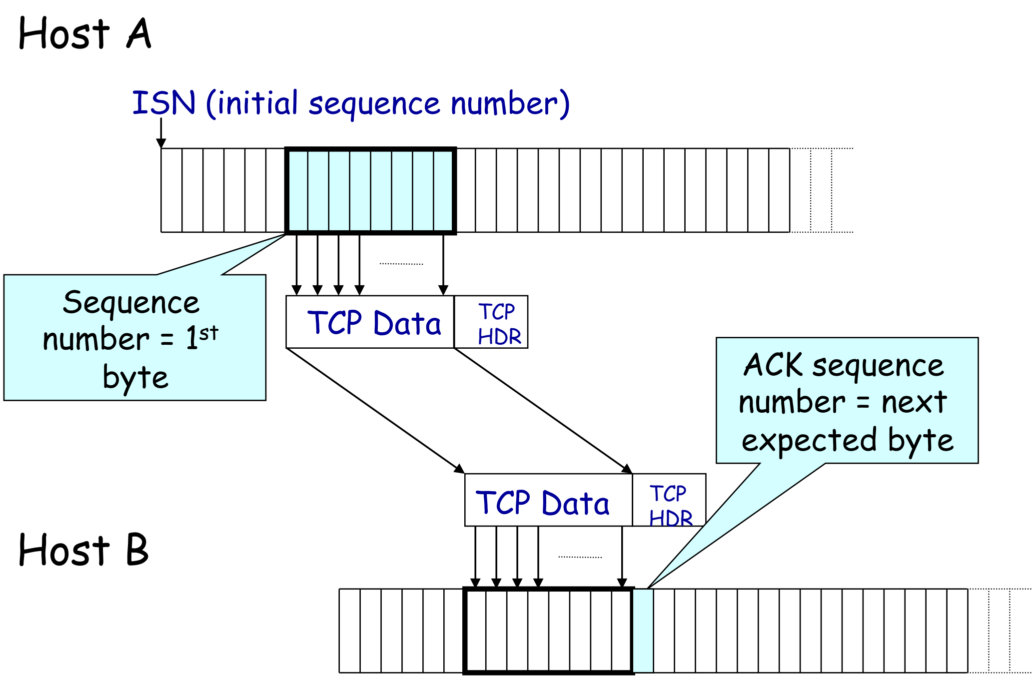


TCP Segment



- IP packet
  - No bigger than Maximum Transmission Unit (MTU)
  - E.g., up to 1500 bytes on an Ethernet
- TCP packet
  - IP packet with a TCP header and data inside
  - TCP header is typically 20 bytes long
- TCP segment
  - No more than Maximum Segment Size (MSS) bytes
  - E.g., up to 1460 consecutive bytes from the stream

Sequence Numbers

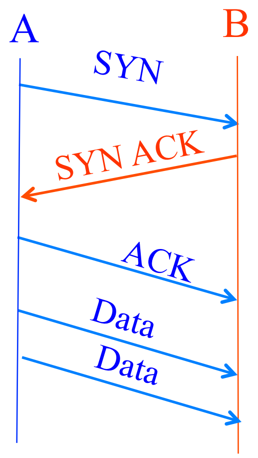


Initial Sequence Number (ISN)

- Sequence number for the very first byte
  - E.g., Why not a de facto ISN of 0?
- Practical issue
  - IP addresses and port #s uniquely identify a connection
  - Eventually, though, these port #s do get used again
  - … and there is a chance an old packet is still in flight
  - … and might be associated with the new connection
- So, TCP requires changing the ISN over time
  - Set from a 32-bit clock that ticks every 4 microseconds
  - … which only wraps around once every 4.55 hours!
- But, this means the hosts need to exchange ISNs

TCP Three-Way Handshake

Establishing a TCP Connection



Each host tells its ISN to the other host.

- Three-way handshake to establish connection
  - Host A sends a **SYN** (open) to the host B
  - Host B returns a SYN acknowledgment (**SYN ACK**)
  - Host A sends an **ACK** to acknowledge the SYN ACK

TCP Header

Flags: SYN
FIN
RST
PSH
URG
ACK

Source port			Destination port
Sequence number			
Acknowledgment			
HdrLen	0	Flags	Advertised window
Checksum			Urgent pointer
Options (variable)			
Data			

Step 1: A’s Initial SYN Packet

Flags: SYN
FIN
RST
PSH
URG
ACK

A’s port			B’s port
A’s Initial Sequence Number			
Acknowledgment			
20	0	Flags	Advertised window
Checksum			Urgent pointer
Options (variable)			

A tells B it wants to open a connection…

Step 2: B's SYN-ACK Packet

B's port		A's port	
B's Initial Sequence Number			
A's ISN plus 1			
20	0	Flags	Advertised window
Checksum		Urgent pointer	
Options (variable)			

Flags: SYN
FIN
RST
PSH
URG
ACK

B tells A it accepts, and is ready to hear the next byte…

… upon receiving this packet, A can start sending data

Step 3: A's ACK of the SYN-ACK

A's port		B's port	
Sequence number			
B's ISN plus 1			
20	0	Flags	Advertised window
Checksum		Urgent pointer	
Options (variable)			

Flags: SYN
FIN
RST
PSH
URG
ACK

A tells B it wants is okay to start sending

… upon receiving this packet, B can start sending data

What if the SYN Packet Gets Lost?

- Suppose the SYN packet gets lost
  - Packet is lost inside the network, or
  - Server rejects the packet (e.g., listen queue is full)
- Eventually, no SYN-ACK arrives
  - Sender sets a timer and wait for the SYN-ACK
  - … and retransmits the SYN-ACK if needed
- How should the TCP sender set the timer?
  - Sender has no idea how far away the receiver is
  - Hard to guess a reasonable length of time to wait
  - Some TCPs use a default of 3 or 6 seconds

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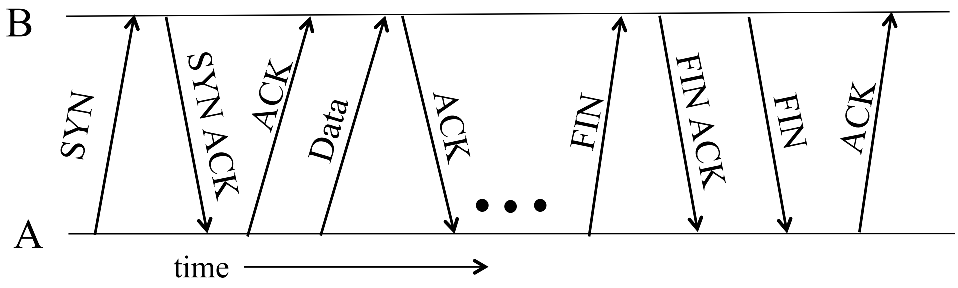
SYN Loss and Web Downloads

- User clicks on a hypertext link
  - Browser creates a socket and does a "connect"
  - The "connect" triggers the OS to transmit a SYN
- If the SYN is lost…
  - The 3-6 seconds of delay may be very long
  - The user may get impatient
  - … and click the hyperlink again, or click "reload"
- User triggers an "abort" of the "connect"
  - Browser creates a new socket and does a "connect"
  - Essentially, forces a faster send of a new SYN packet!
  - Sometimes very effective, and the page comes fast

Tearing Down the Connection

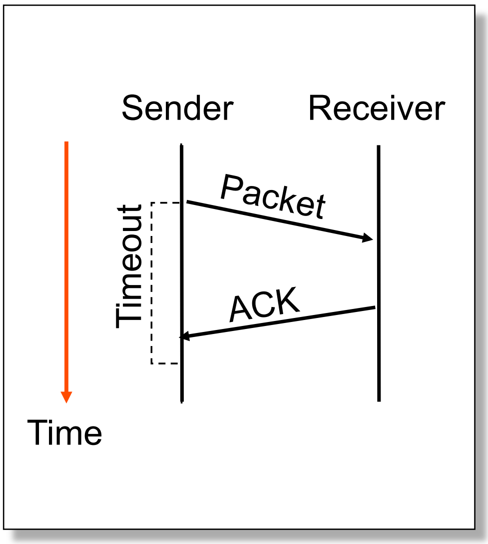


- Closing the connection
  - Finish (FIN) to close and receive remaining bytes
  - And other host sends a FIN ACK to acknowledge
  - Reset (RST) to close and not receive remaining bytes

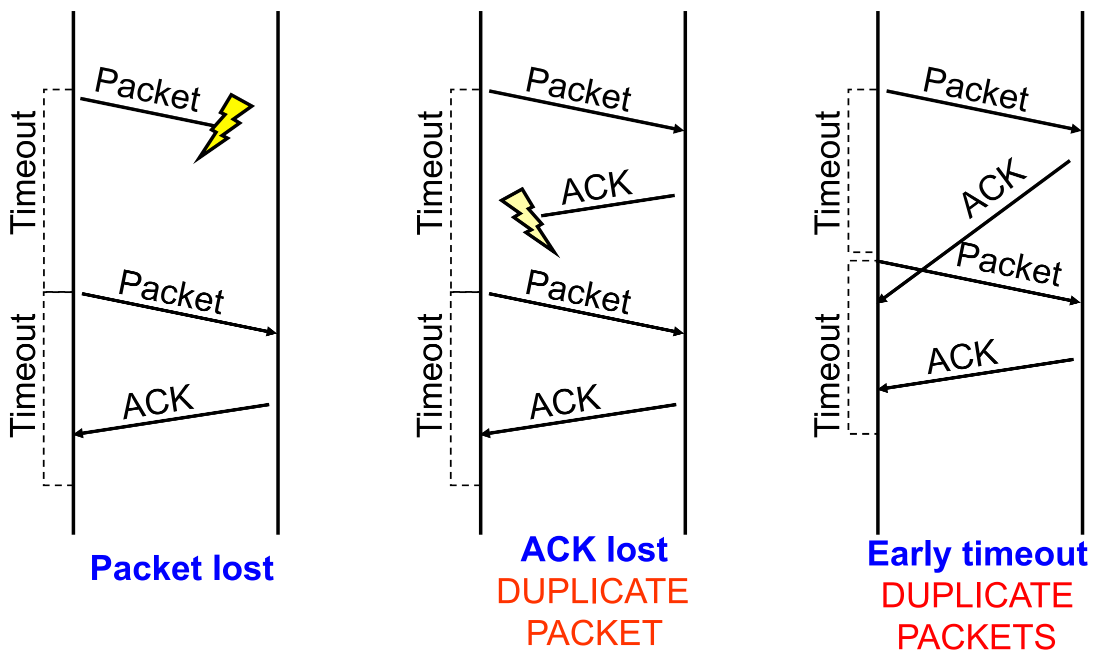
TCP Retransmissions

Automatic Repeat reQuest (ARQ)

- Automatic Repeat Request
  - Receiver sends acknowledgment (ACK) when it receives packet
  - Sender waits for ACK and timeouts if it does not arrive within some time period
- Simplest ARQ protocol
  - Stop and wait
  - Send a packet, stop and wait until ACK arrives



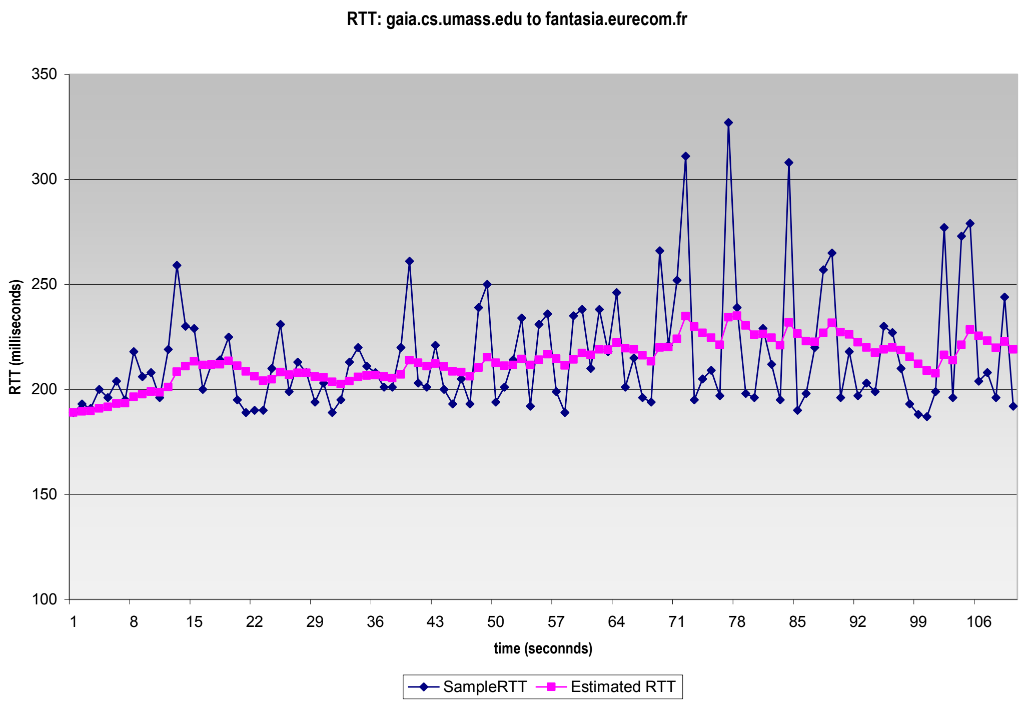
Reasons for Retransmission



How Long Should Sender Wait?

- Sender sets a timeout to wait for an ACK
  - Too short: wasted retransmissions
  - Too long: excessive delays when packet lost
- TCP sets timeout as a function of the RTT
  - Expect ACK to arrive after an RTT
  - … plus a fudge factor to account for queuing
- But, how does the sender know the RTT?
  - Can estimate the RTT by watching the ACKs
  - Smooth estimate: keep a running average of the RTT
    - EstimatedRTT = a * EstimatedRTT + (1 –a) * SampleRTT
  - Compute timeout: TimeOut = 2 * EstimatedRTT

Example RTT Estimation



A Flaw in This Approach

- An ACK doesn't really acknowledge a transmission
  - Rather, it acknowledges receipt of the data
- Consider a retransmission of a lost packet
  - If you assume the ACK goes with the 1st transmission
  - … the SampleRTT comes out way too large
- Consider a duplicate packet
  - If you assume the ACK goes with the 2nd transmission
  - … the Sample RTT comes out way too small
- Simple solution in the Karn/Partridge algorithm
  - Only collect samples for segments sent one single time

Yet Another Limitation…

- Doesn't consider variance in the RTT
  - If variance is small, the EstimatedRTT is pretty accurate
  - … but, if variance is large, the estimate isn't all that good
- Better to directly consider the variance
  - Consider difference: SampleRTT – EstimatedRTT
  - Boost the estimate based on the difference
- Jacobson/Karels algorithm
  - See Section 5.2 of the Peterson/Davie book for details

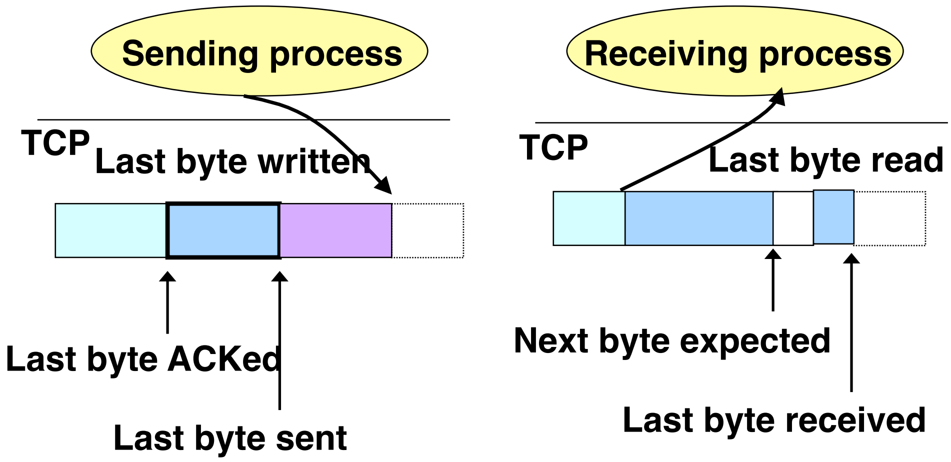
TCP Sliding Window

Motivation for Sliding Window

- Stop-and-wait is inefficient
  - Only one TCP segment is “in flight” at a time
  - Especially bad when delay-bandwidth product is high
- Numerical example
  - 1.5 Mbps link with a 45 msec round-trip time (RTT)
    - Delay-bandwidth product is 67.5 Kbits (or 8 KBytes)
  - But, sender can send at most one packet per RTT
    - Assuming a segment size of 1 KB (8 Kbits)
    - … leads to 8 Kbits/segment / 45 msec/segment ➔ 182 Kbps
    - That’s just one-eighth of the 1.5 Mbps link capacity

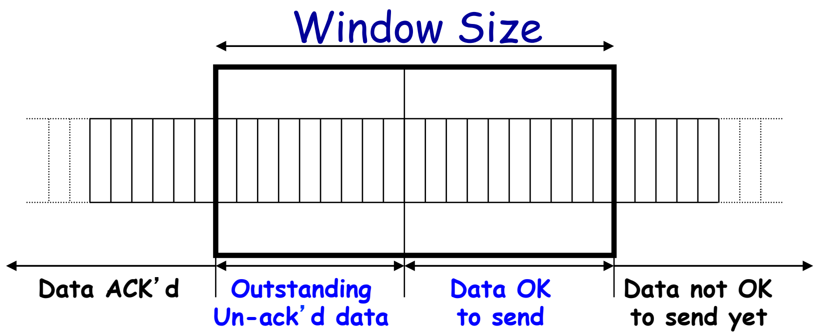
Sliding Window

- Allow a larger amount of data "in flight"
  - Allow sender to get ahead of the receiver
  - … though not *too far* ahead

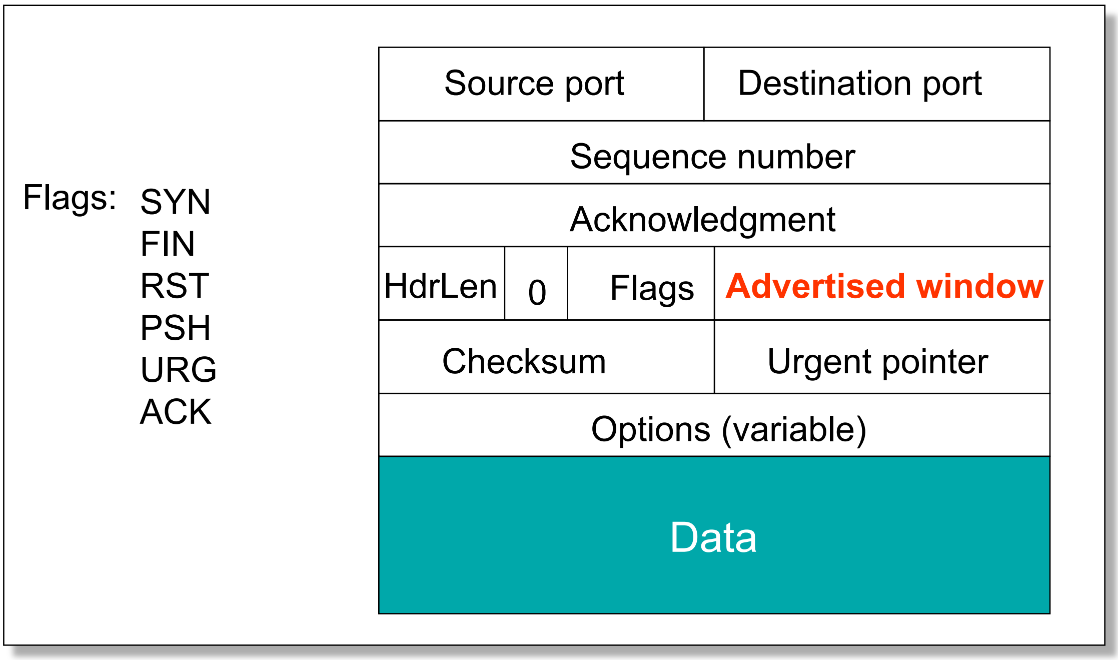


Receiver Buffering

- Window size
  - Amount that can be sent without acknowledgment
  - Receiver needs to be able to store this amount of data
- Receiver advertises the window to the receiver
  - Tells the receiver the amount of free space left
  - … and the sender agrees not to exceed this amount



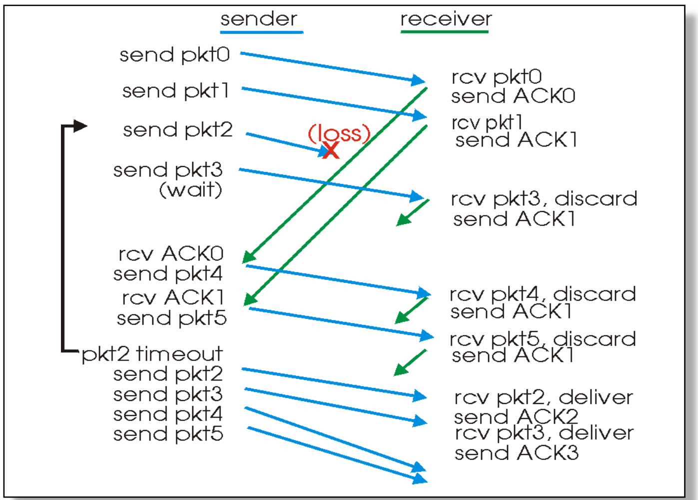
TCP Header for Receiver Buffering



Fast Retransmission

Timeout is Inefficient

- Timeout-based retransmission
  - Sender transmits a packet and waits until timer expires
  - … and then retransmits from the lost packet onward



Fast Retransmission

- Better solution possible under sliding window
  - Although packet n might have been lost
  - … packets n+1, n+2, and so on might get through
- Idea: have the receiver send ACK packets
  - ACK says that receiver is still awaiting n^{th} packet
    - And *repeated* ACKs suggest later packets have arrived
  - Sender can view the “duplicate ACKs” as an early hint
    - … that the n^{th} packet must have been lost
    - … and perform the retransmission early
- Fast retransmission
  - Sender retransmits data after the triple duplicate ACK

Effectiveness of Fast Retransmit

- When does Fast Retransmit work best?
  - Long data transfers
    - High likelihood of many packets in flight
  - High window size
    - High likelihood of many packets in flight
  - Low burstiness in packet losses
    - Higher likelihood that later packets arrive successfully
- Implications for Web traffic
  - Most Web transfers are short (e.g., 10 packets)
    - Short HTML files or small images
  - So, often there aren't many packets in flight
  - … making fast retransmit less likely to "kick in"
  - Forcing users to like "reload" more often…

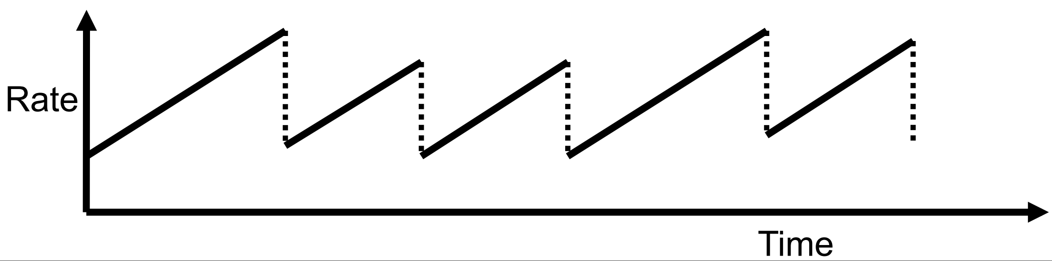
TCP Congestion Control

TCP Congestion Control

- Flow control addresses congestion at receiver, not in middle of network (for example, at intermediate routers)
- Suppose you initially send up to the size of flow control window (FW)
  - Intermediate routers may not be able to handle so much traffic
  - Congestion overflows router buffers causing lost packets causing retransmissions causing more congestion … → congestion collapse
- Idea behind TCP Congestion Control:
  - Send only enough packets into the network that the network has the capacity to handle without loss
- Define a congestion window (CW) that can be used to respond to network congestion
  - Distinct from flow control window FW
  - Actual window size W = min (CW, FW)
    - # of data bytes that can be on the link
    - Send no more data than the bottleneck can handle

TCP Congestion Control

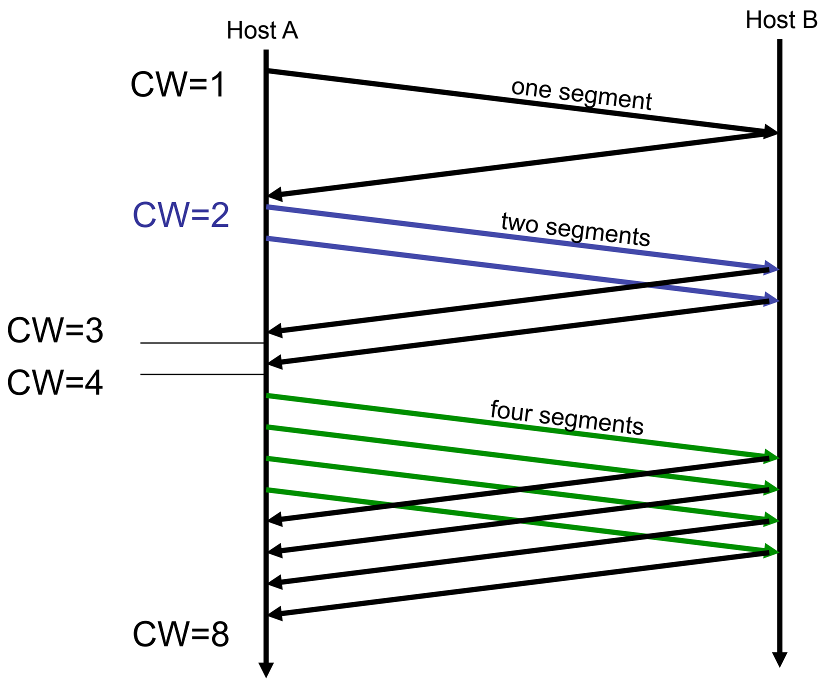
- How does sender set CW?
  - Adaptively probe the network with data segments
  - Keep expanding CW until a segment is lost, then contract CW.
  - Continue with expand/contract cycle throughout connection – "sawtooth" behavior'
- Additive Increase / Multiplicative Decrease (AIMD)
  - Increment CW by one packet per RTT (*linear increase*)
  - Divide CW by two whenever a timeout occurs (*multiplicative decrease*)



TCP Slow Start

- The rate at which new packets should be injected into network is the rate at which ACKs are returned by other end
  - Use ACK's to pace transmission of packets: "self-clocking"
  - Start by setting CW = 1 segment (in bytes)
    - Initial segment size set by receiver
  - For each ACK that returns, increment CW by one.
    - Send 1 packet. When ACK returns, increment CW, CW=2
    - Send 2 packets. When 1st ACK returns, increment CW to CW=3, when 2nd ACK returns, increment CW to CW=4
    - Can send 4 packets. After 4 ACKs return, CW will be up to 8
  - Exponential increase – not "slow", quickly reach window size that the network can accommodate

TCP Slow Start

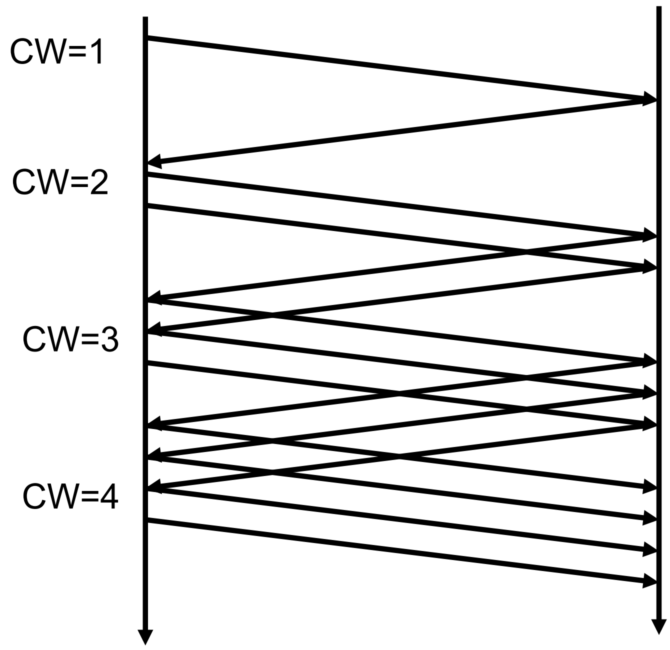


TCP Additive Increase/ Multiplicative Decrease

- How does a sender detect that CW is too large?
  - It starts to see timeouts, which are interpreted as packet loss due to congestion
- After a timeout:
  - TCP drastically resets CW=1 and slow starts again
  - TCP remembers that congestion occurred near CW by storing CW/2 in *ssthresh* = CW/2
    - ssthresh = Slow Start Threshold
  - But TCP exponentially increases only to ssthresh, halfway to old congestion mark
    - After CW>ssthresh, *additively increase* CW
    - Rationale: Be cautious about sending new data packets once you get near old mark that caused timeouts/congestion

TCP Additive Increase

- If entire window’s worth (CW) of packets in a RTT is ACKed w/o error, then increment CW by one
- In practice, TCP adds α/CW to CW as each ACK returns, rather than waiting for a full CW of ACKs to return



Rationale behind AIMD

- Why not just slow start exclusively (exponential increase) after timeout, instead of additive increase?
  - Be more cautious about adding new packets once you're near old congestion point.
- Each time a timeout occurs, divide CW by half and store in ssthresh: multiplicative decrease
  - Minimum ssthresh and minimum CW is one
- Why not additive decrease instead of multiplicative decrease after congestion?
  - Consequences of having a too-large congestion window are worse than having a too-small CW
  - Additive decrease can keep CW too large for too long compared to multiplicative decrease

More on TCP AIMD

- What happens if the amount of unacknowledged data is greater than CW?
  - Can't send new data
  - Retransmit unacknowledged data
  - Wait for ACKs for unacknowledged data to increase CW above size of unacknowledged data, then can send new data
- After a timeout, TCP slows down in two ways:
  - Congestion window collapses, restricting new data
  - RTO backs off exponentially, slowing down retransmission of old unacknowledged data

TCP Saw Tooth Behavior

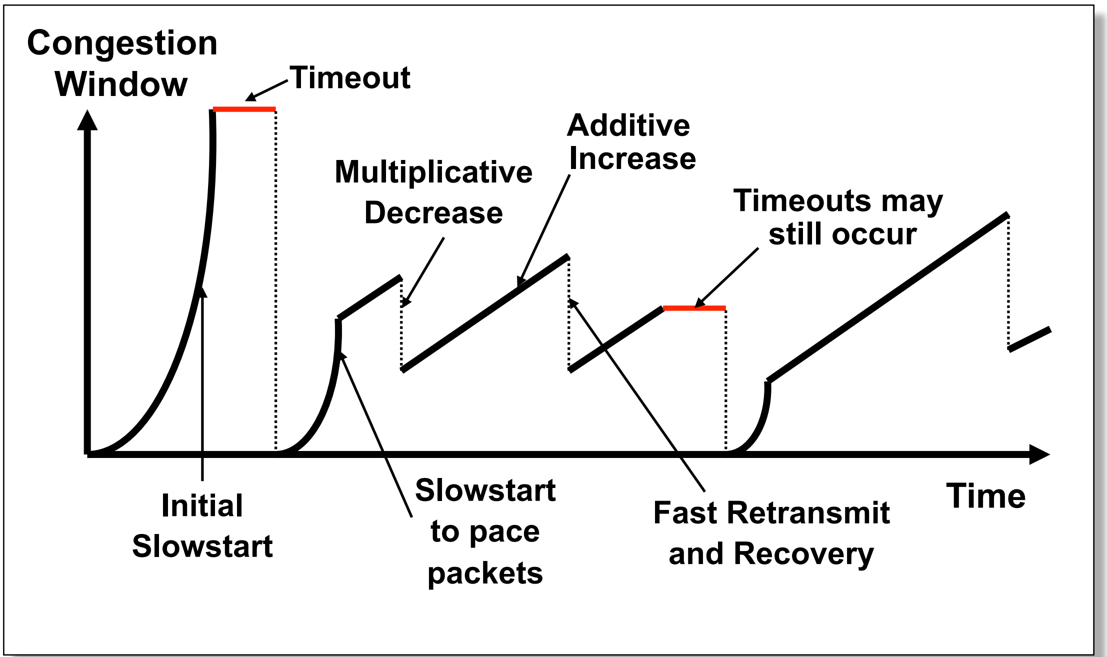
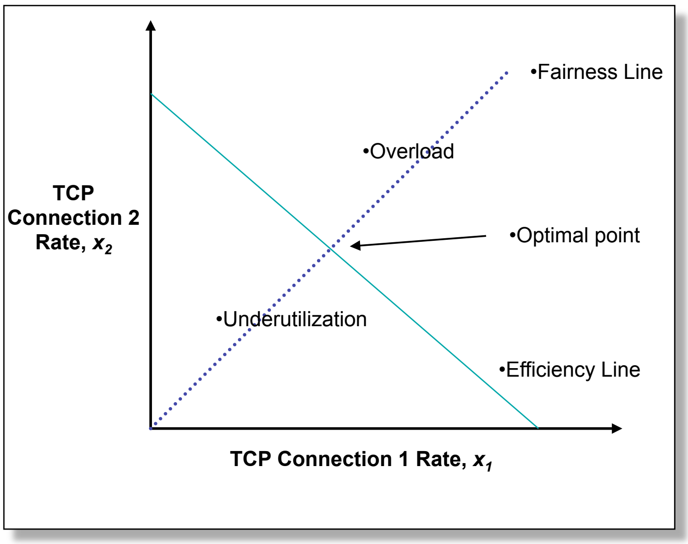
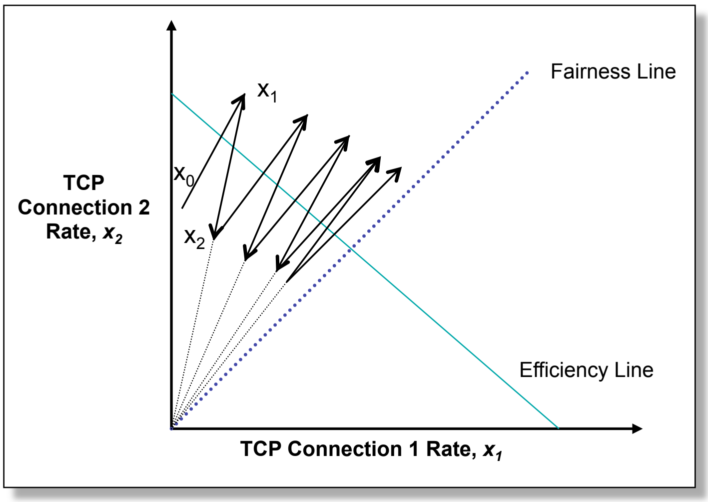


Diagram courtesy of Srini Seshan

Visualizing behavior of competing TCP connections over time



Visualizing TCP’s AIMD



Q: Is Additive Increase / Multiplicative Decrease fair?

Q: What if we had used MIMD? Or AIAD?

Congestion Avoidance

- Congestion control:
  - Cycle of actively probing, transmitting more than the network can handle, then backing off
- Congestion avoidance:
  - Back off before there are packet losses
  - How can you tell that congestion is increasing?
    - Look at RTT – is it expanding?
    - Implicit: Random Early Dropping (RED) of packets by intermediate routers
    - Explicit: Intermediate routers indicate that there is congestion by setting a bit in the packet and receiver send that information back in the ACK (DECbit)

Summary of TCP

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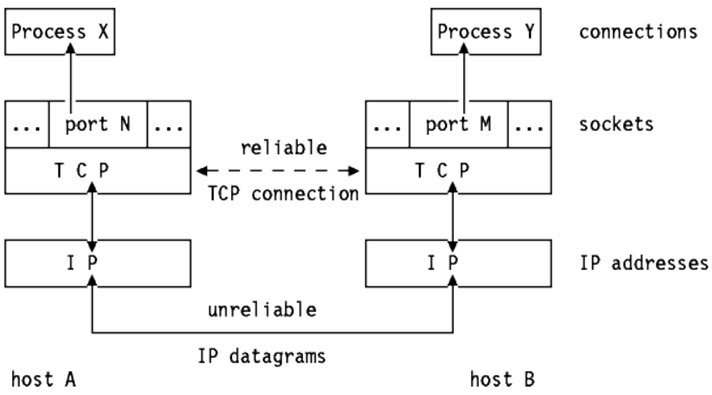
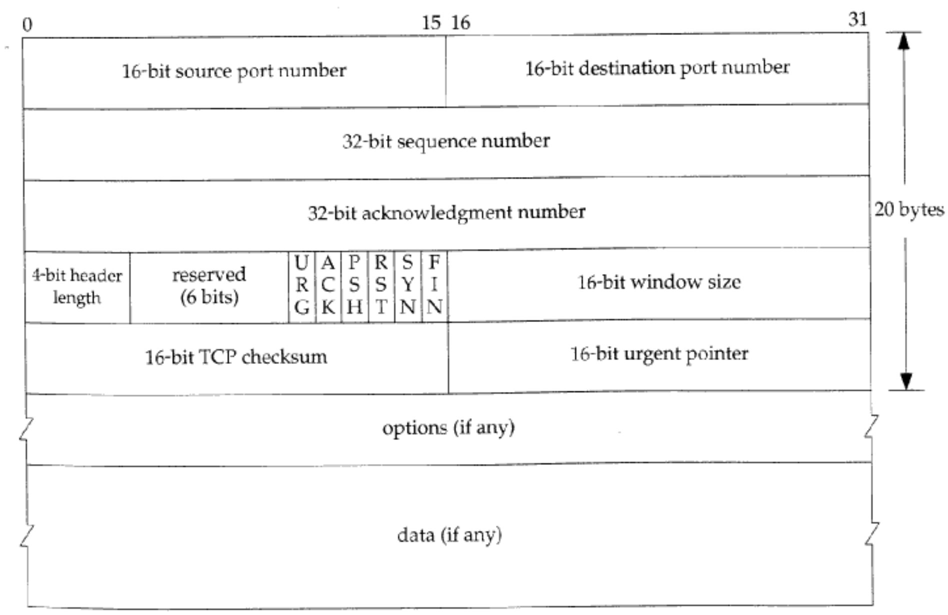
Summary of Transport Layer Services

- Connection-oriented communication
- Multiplexing
- Byte orientation
- In-order delivery
- Reliability: Error Detection & Re-transmissions
- Flow control
- Congestion control and congestion avoidance

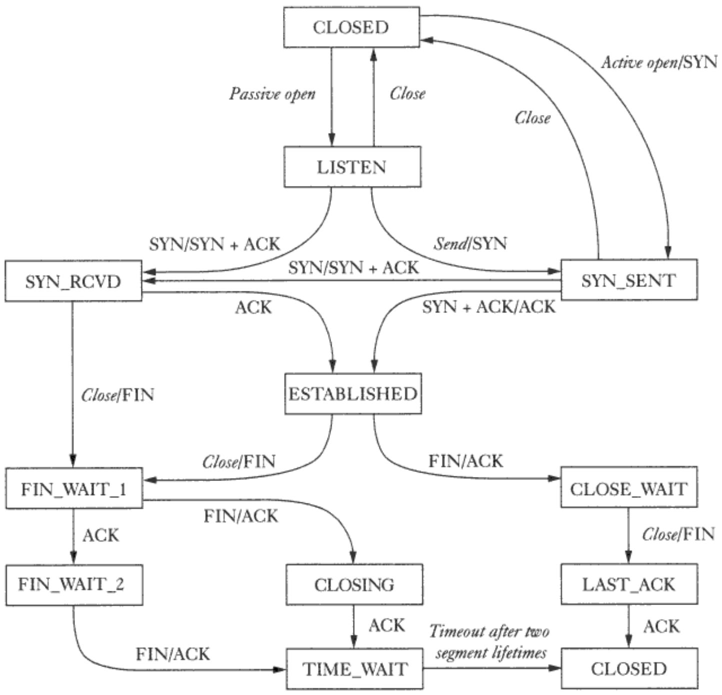
TCP Socket



TCP Header



TCP Finite State Machine Diagram



Conclusion

- Transport protocols
  - Multiplexing and demultiplexing
  - Sequence numbers
  - Window-based flow control
  - Timer-based retransmission
  - Checksum-based error detection
- TCP – provides end-to-end reliability
  - TCP also encompasses congestion control
  - AIMD: Additive Increase Multiplicative Decrease
- References for transport layer protocols
  - Chapter 6/7 of Peterson & Davie
  - Chapter 3 of Kurose & Ross