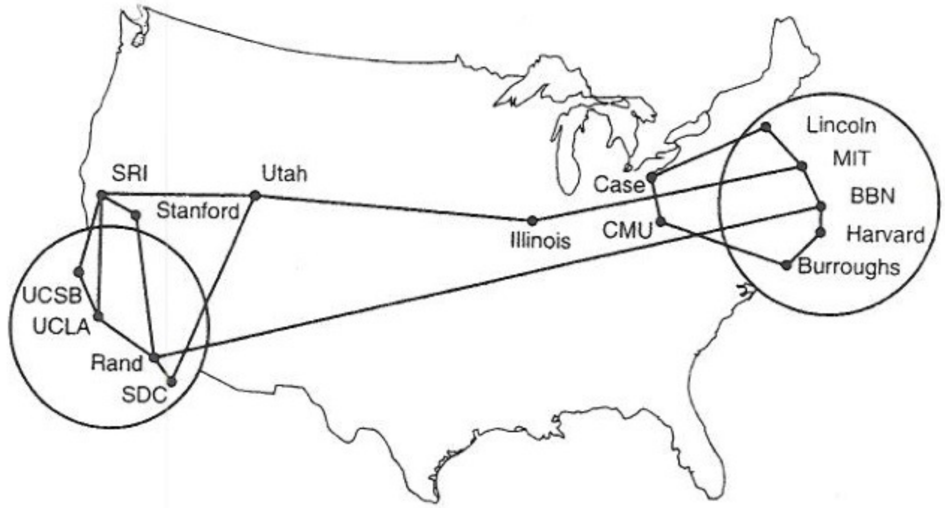


Transport Protocols

Readings: 5.1, & 5.2

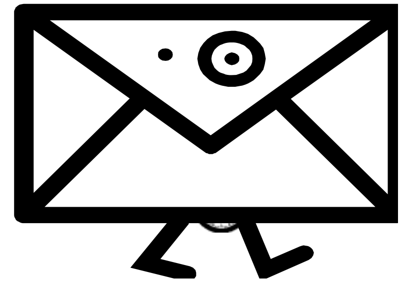
UDP - User Datagram Protocol

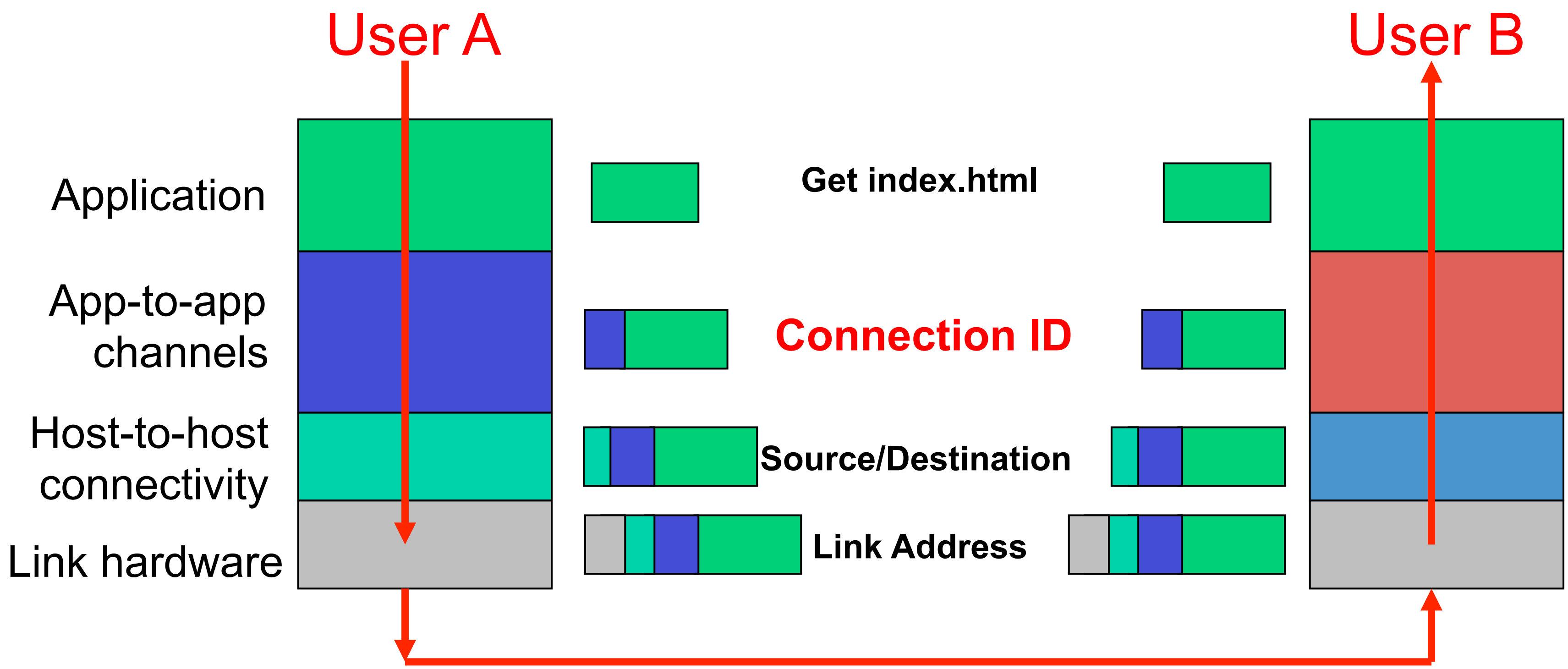
TCP - Transmission Control Protocol

Goals for Today's Lecture

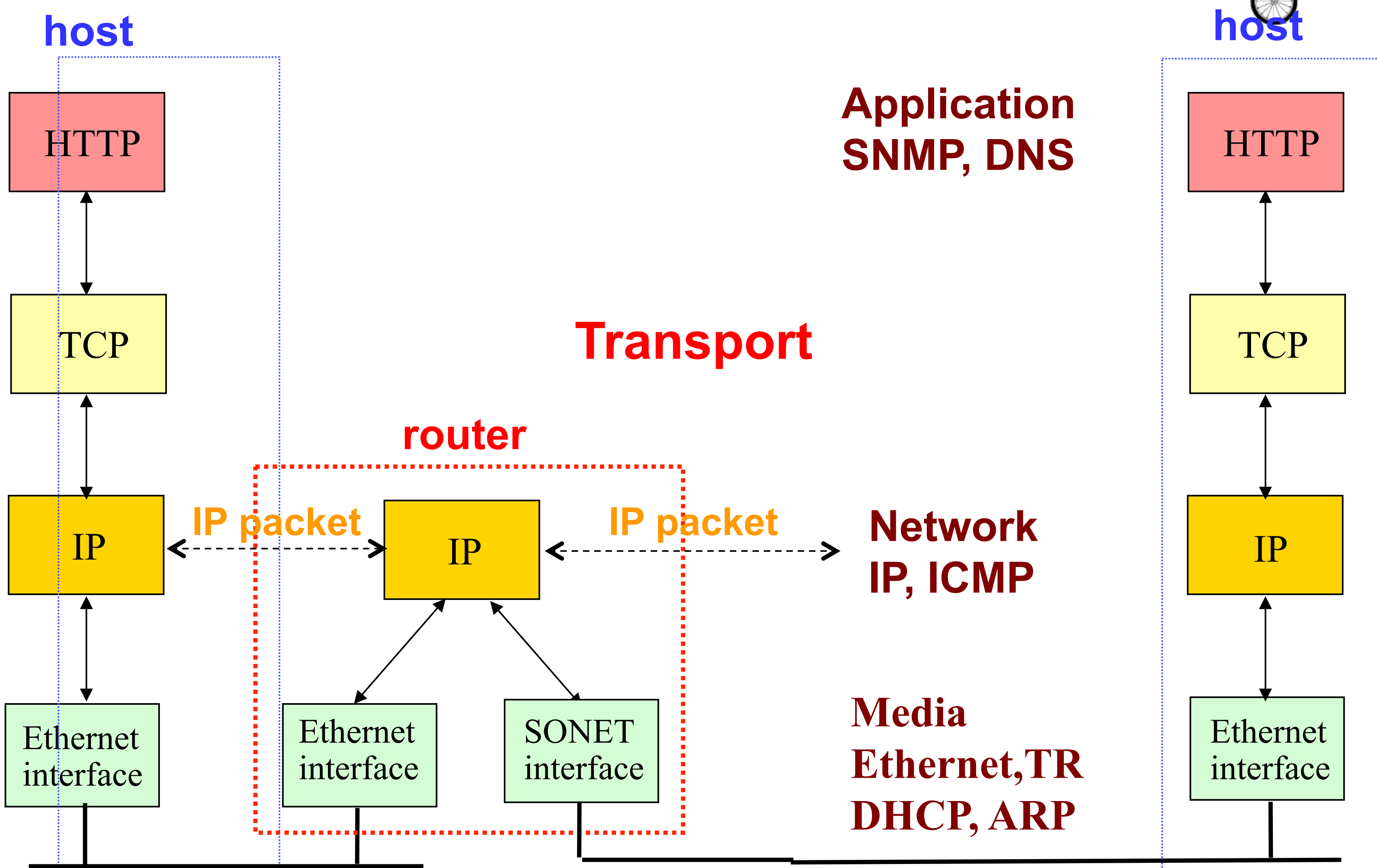
- Principles underlying transport-layer services
  – (De)multiplexing
  – Detecting corruption
  – Reliable delivery
  – Flow control
- (Primary) Transport-layer protocols in the Internet
  – User Datagram Protocol (UDP)
  – Transmission Control Protocol (TCP)

Layer Encapsulation in

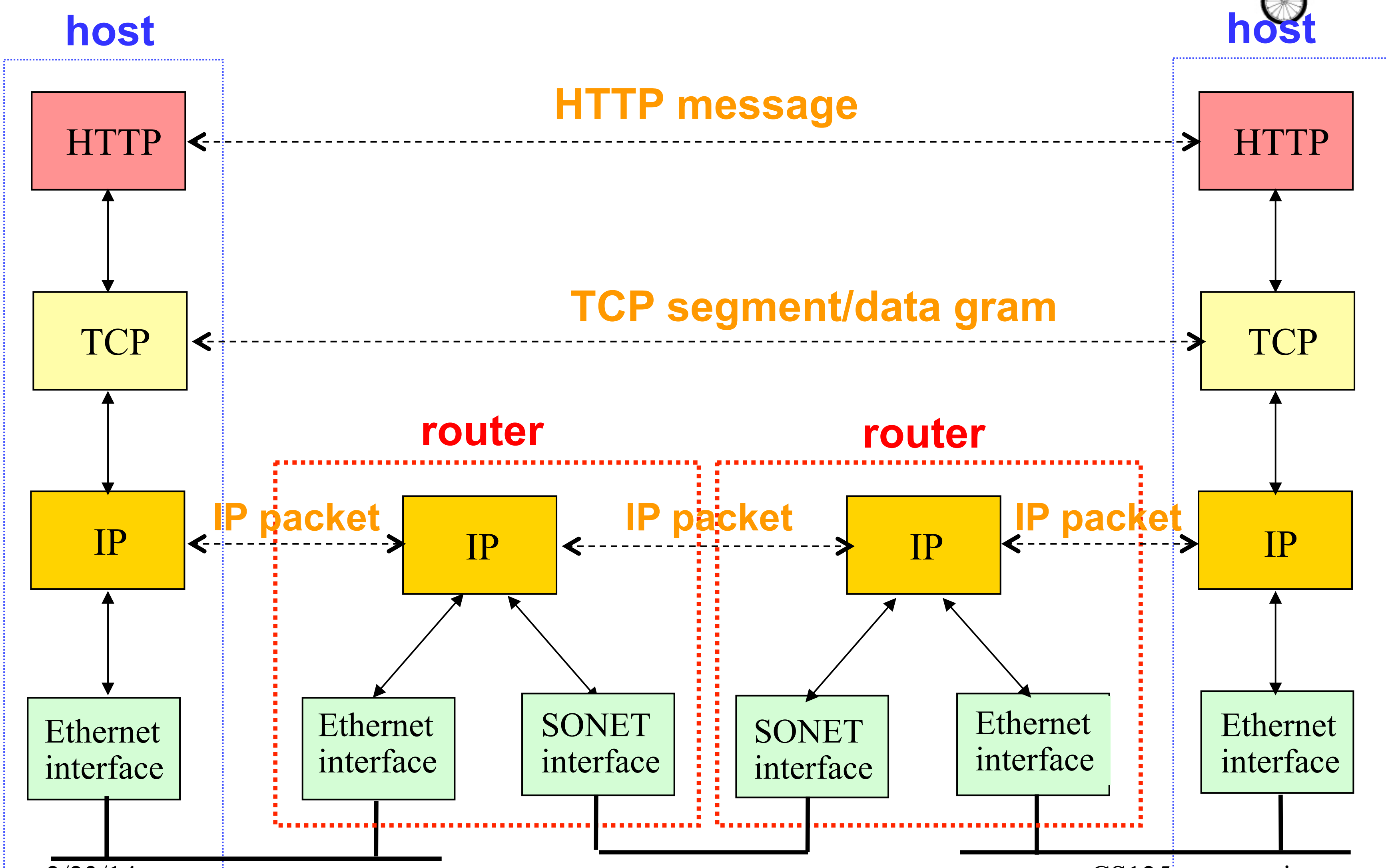




IP Suite In Action: Where We Are!!



IP Suite In Action: End Hosts vs. Routers



End-to-End Protocols

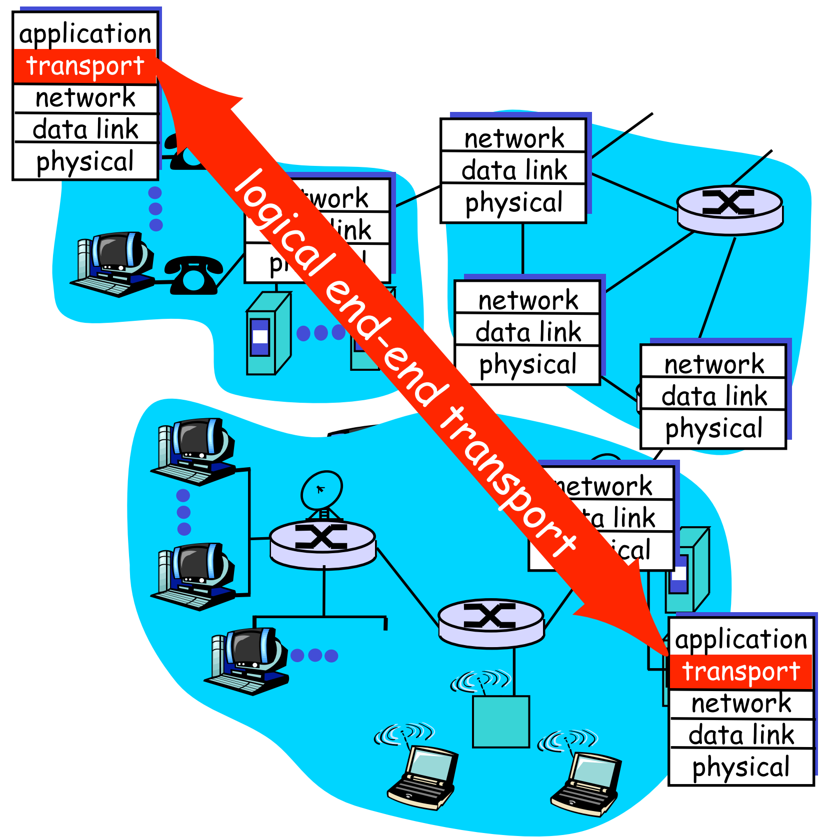
- Working our way up and down Protocol Stack
  - Directly connected networks
  - Switching, Routing and Forwarding
  - Internetworking: IP, Bridges, End-to-End
    - Move from host-to-host
    - To Process-to-process
  - Applications: SNMP, DNS (Top down)
- Transport next layer

Roles of Layers

- Application layer
  - Communication for specific applications
  - E.g., HyperText Transfer Protocol (HTTP), File Transfer Protocol (FTP), Network News Transfer Protocol (NNTP), DNS, SNMP
- 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 – best known



End-to-End Protocols

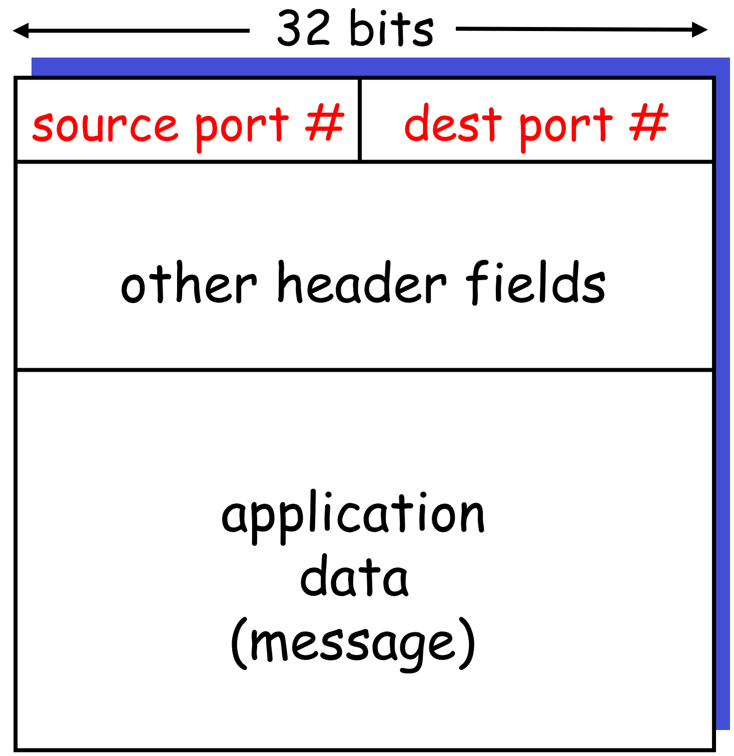
- Underlying best-effort network (IP)
  - Drops messages
  - re-orders messages … different routes, costs
  - delivers duplicate copies of a given message
  - limits messages to some finite size - MTU
  - delivers messages after an arbitrarily long delay
- Common end-to-end services (desired by application)
  - guarantee message delivery
  - deliver messages in the same order they are sent
  - deliver at most one copy of each message
  - support arbitrarily large messages
  - support synchronization
  - allow the receiver to flow control the sender
  - support multiple application processes on each host
  - SECURITY

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 (next lecture)
- Other services not available
  - Delay guarantees
  - Bandwidth guarantees
  - Security

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



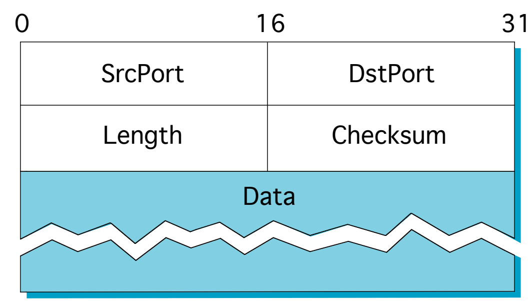
TCP/UDP segment format

UDP - 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

Simple Demultiplexor (UDP)

- Unreliable and unordered datagram service
- No flow control, No Acks, No msg ordering, No feedback
- Endpoints identified by ports
  - servers have *well-known* ports
  - see **`/etc/services`** on Unix
- Header format



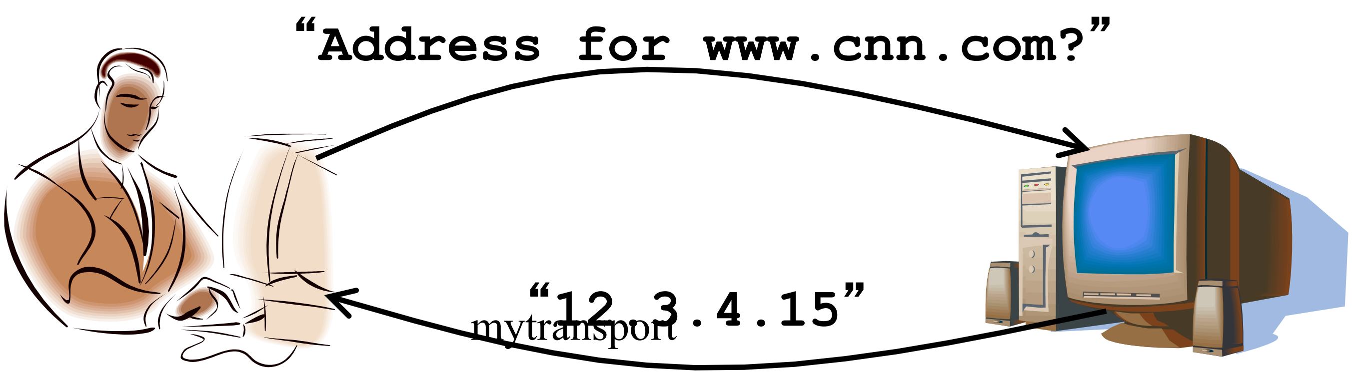
- Optional checksum
  - pseudo header (Source, Destination, Protocol) + UDP header + data
- Application - accepts full responsibility for all other issues

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



UDP PseudoHeader

- Issue - Message Correctness
  - Only have ports in the UDP header, yet the message is directed to an IP address
  - UDP wants to make sure that the message is directed to this host and this port
- PseudoHeader
  - Used in calculation of the checksum
  - Includes Source and Destination IP addresses (from IP header)
  - Also the type field
  - Without reading the RFC, not sure if it is prepended or appended
  - Also some books say UDP length field is used twice in the pseudoheader – read RFC if want to know truth
  - UDP checksum, optional in v4, required in v6

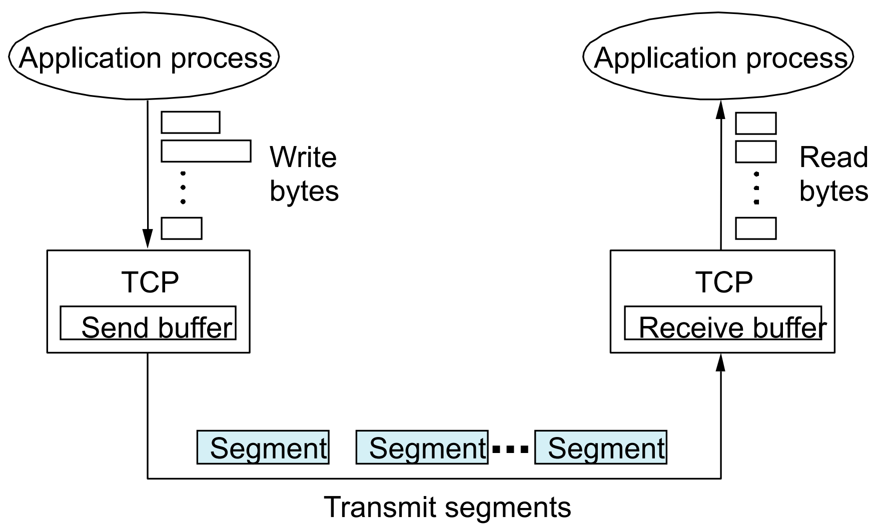
Reliable Byte-Stream (TCP)

Outline

- Connection Establishment/Termination
- Sliding Window
- Flow Control
- Adaptive Timeout
- Extensions

TCP Overview

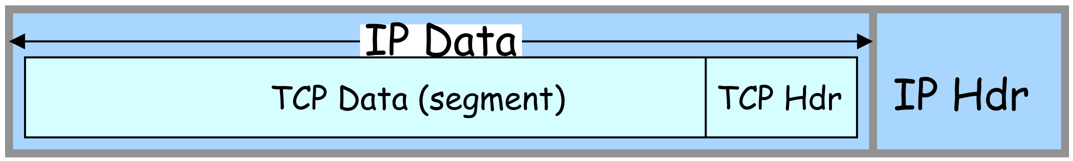
- Connection-oriented
- Byte-stream
  - app writes bytes
  - TCP sends *segments*
  - app reads bytes
- Full duplex
- Flow control: keep sender from overrunning receiver
- Congestion control: keep sender from overrunning network



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 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

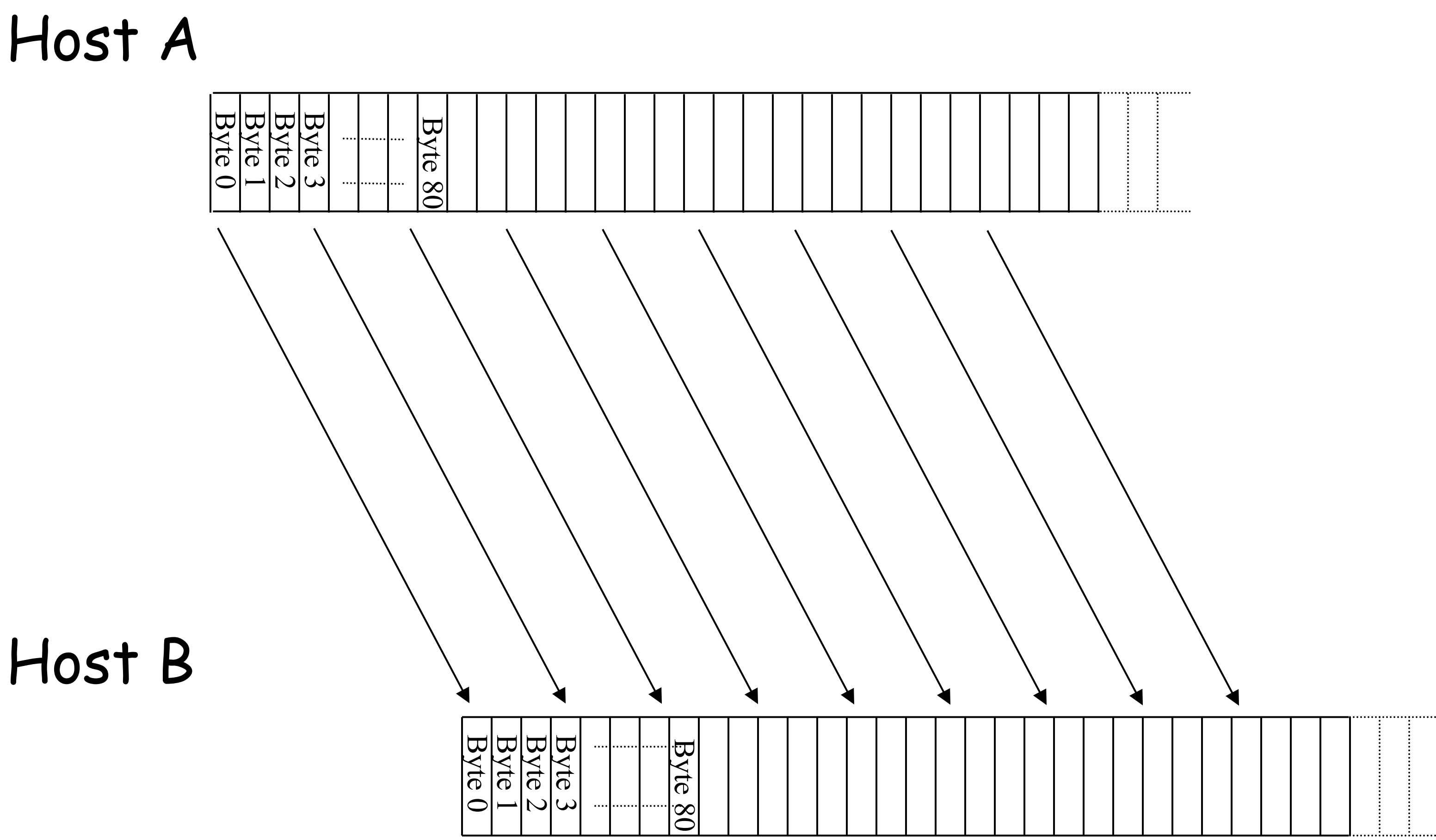
Segment Format

0	4	10	16 … 31
SrcPort			DstPort
SequenceNum			
Acknowledgment			
HdrLen	0	Flags	AdvertisedWindow
Checksum			UrgPtr
Options (variable)			
Data			

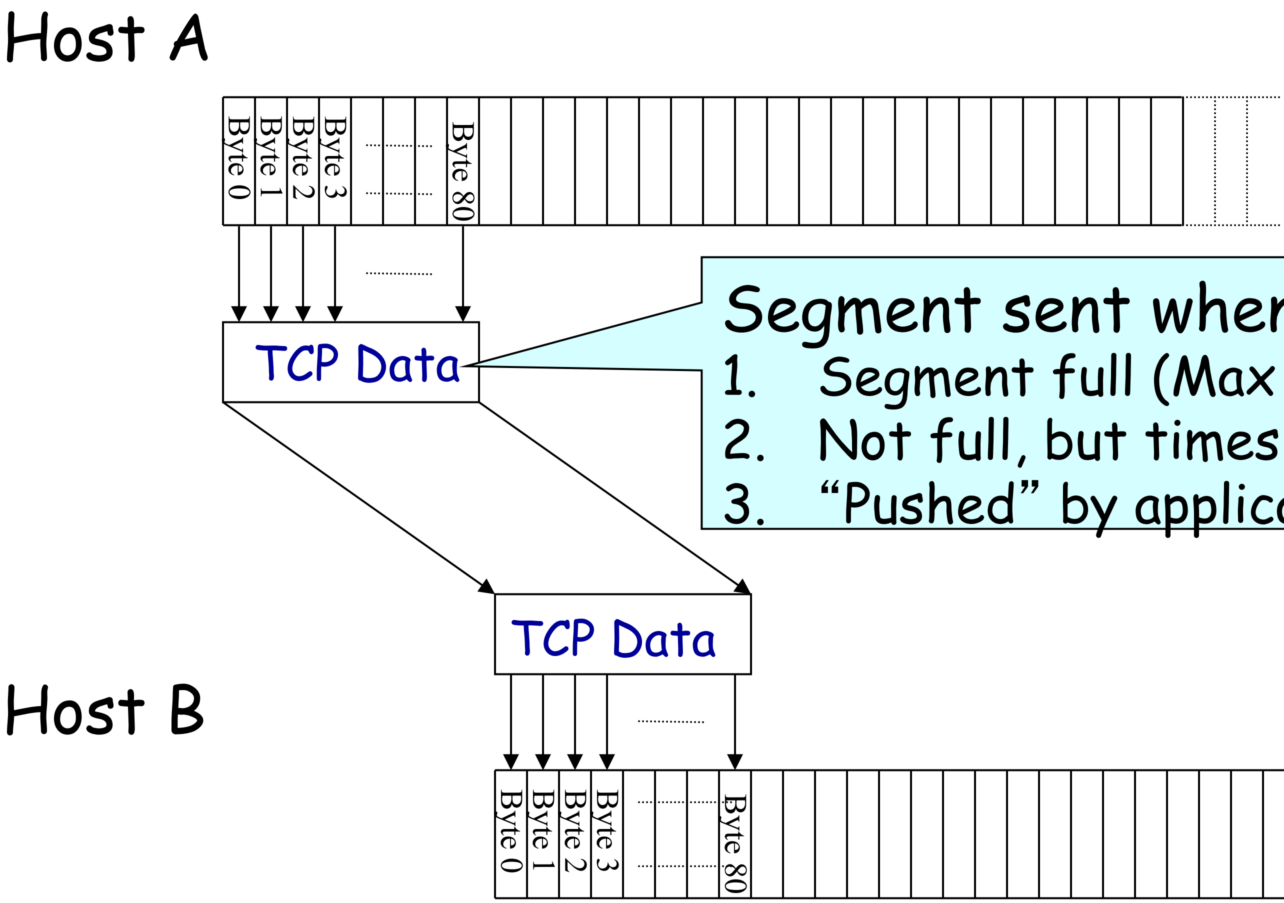
TCP Support for Reliable Delivery

- 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 "Stream of Bytes" Service



…Emulated Using TCP "Segments"

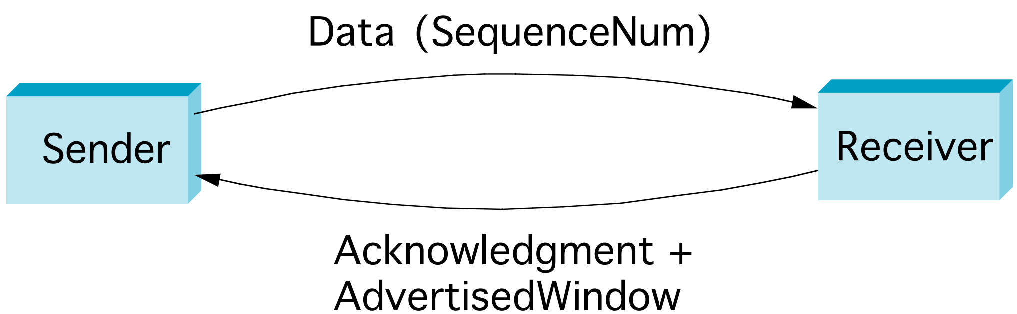


Segment sent when:

1. Segment full (Max Segment Size),
2. Not full, but times out, or
3. "Pushed" by application.

Segment Format (cont)

- Each connection identified with 4-tuple:
  - **`(SrcPort, SrcIPAddr, DsrPort, DstIPAddr)`**
- Sliding window + flow control
  - **`acknowledgment, SequenceNum, AdvertisedWindow`**

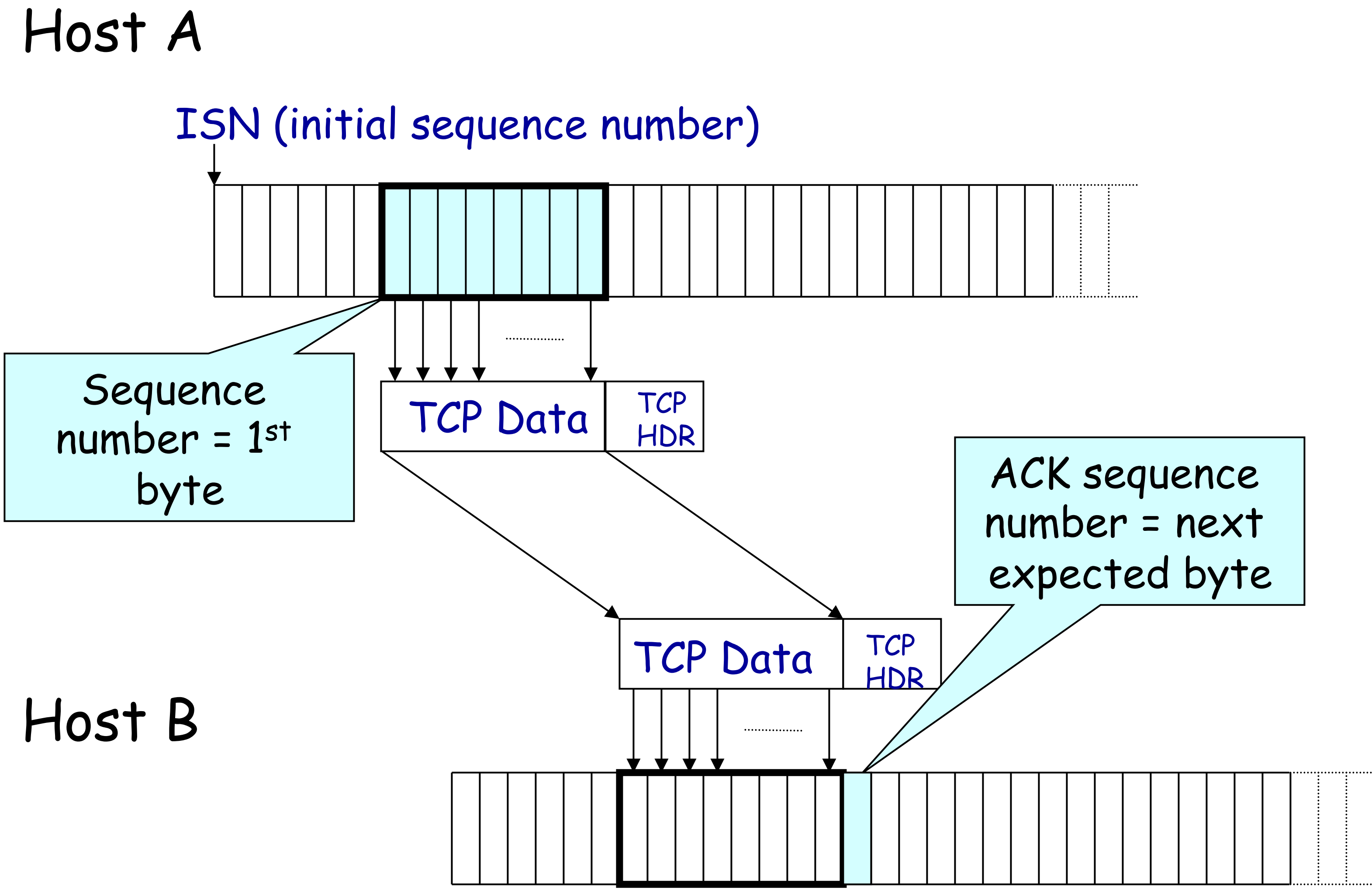


- Flags
  - **`SYN, FIN, RESET, PUSH, URG, ACK`**
- Checksum
  - pseudo header + TCP header + data

Segment Transmission

- MSS - Maximum Segment Size, ~ MTU (-IP Hdrs)
- Push - Sending application/process "pushes" segment, do not wait for full segment, e.g. telnet
- Timer - periodically fires "keep alive"

Sequence Numbers

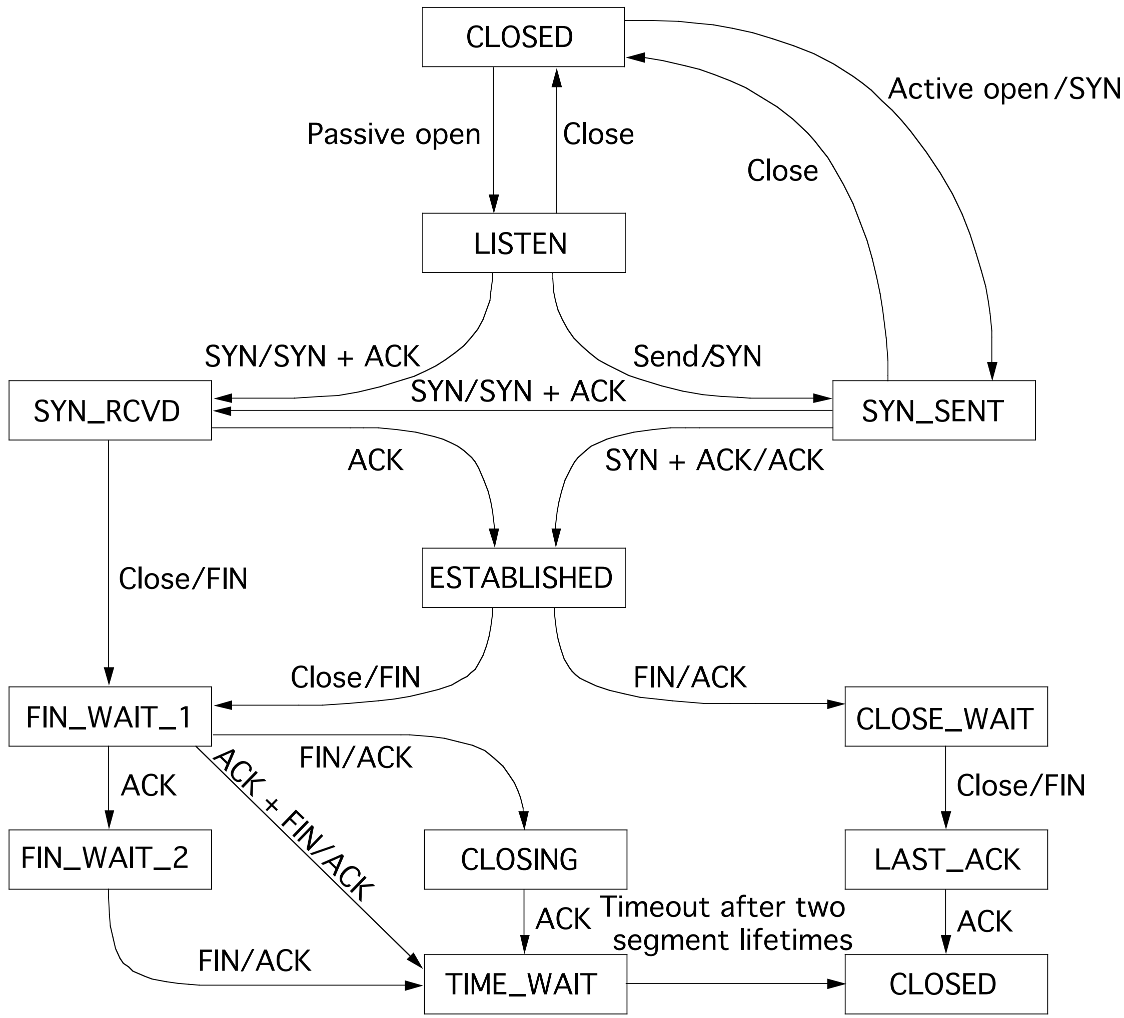


Initial Sequence Number (ISN)

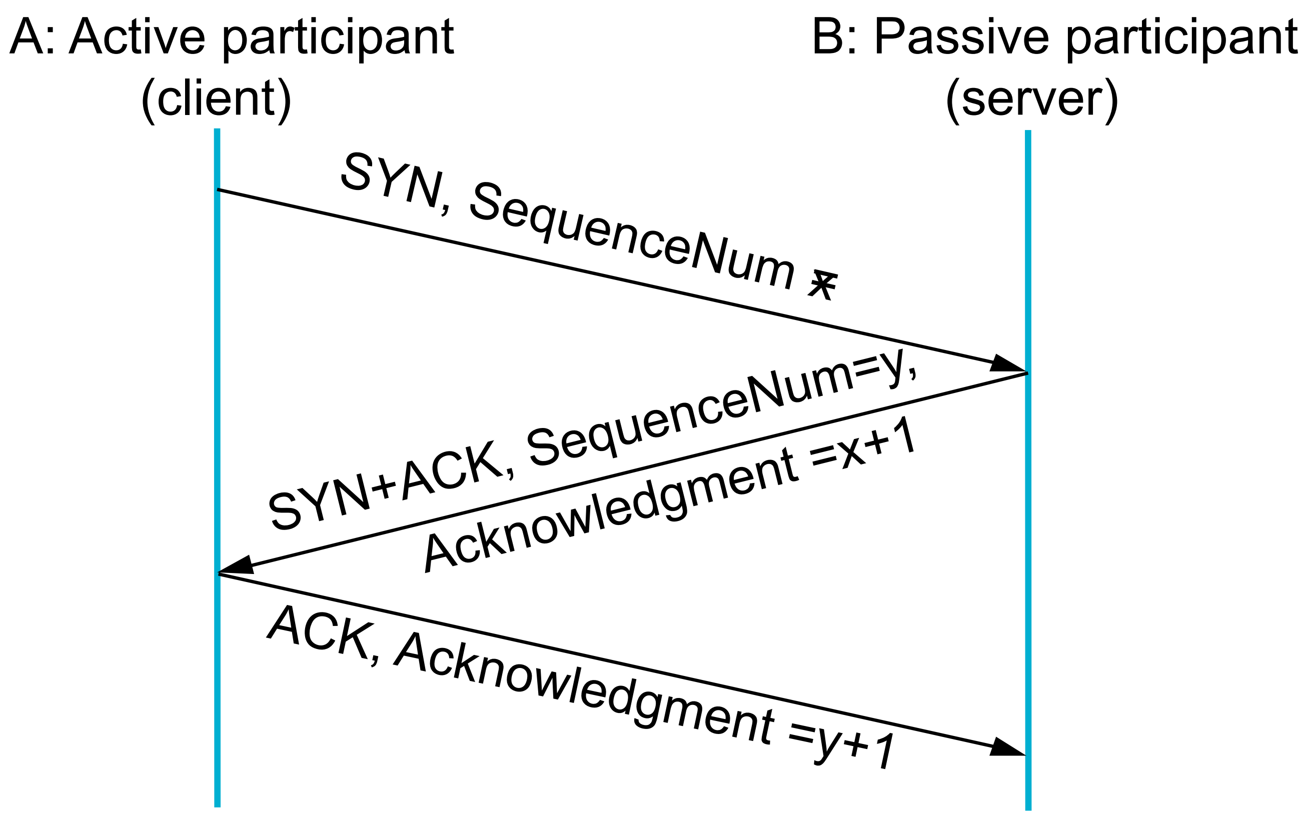
- Sequence number for the very first byte
  - E.g., Why not a de facto ISN of 0?
- Practical issue
  - IP addresses and port #s uniquely identify a connection
  - Eventually, though, these port #s do get used again
  - … and there is a chance an old packet is still in flight
  - … and might be associated 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

State Transition Diagram



Connection Establishment and Termination



TCP Header

Flags: SYN
FIN
RST
PSH
URG
ACK

<table>
<tr><td colspan="2">Source port</td><td colspan="2">Destination port</td></tr>
<tr><td colspan="4">Sequence number</td></tr>
<tr><td colspan="4">Acknowledgment</td></tr>
<tr><td>HdrLen</td><td>0</td><td>Flags</td><td>Advertised window</td></tr>
<tr><td colspan="3">Checksum</td><td>Urgent pointer</td></tr>
<tr><td colspan="4">Options (variable)</td></tr>
<tr><td colspan="4">Data</td></tr>
</table>

Step 1: A’s Initial SYN Packet

Flags: SYN
FIN
RST
PSH
URG
ACK

<table>
<tr><td colspan="3">A’s port</td><td>B’s port</td></tr>
<tr><td colspan="4">A’s Initial Sequence Number</td></tr>
<tr><td colspan="4">Acknowledgment</td></tr>
<tr><td>20</td><td>0</td><td>Flags</td><td>Advertised window</td></tr>
<tr><td colspan="3">Checksum</td><td>Urgent pointer</td></tr>
<tr><td colspan="4">Options (variable)</td></tr>
</table>

A tells B it wants to open a connection…

Step 2: B's SYN-ACK Packet

Flags: SYN
FIN
RST
PSH
URG
ACK

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)			

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

Flags: SYN
FIN
RST
PSH
URG
ACK

<table>
<tr><td colspan="3">A’s port</td><td>B’s port</td></tr>
<tr><td colspan="4">Sequence number</td></tr>
<tr><td colspan="4">B’s ISN plus 1</td></tr>
<tr><td>20</td><td>0</td><td>Flags</td><td>Advertised window</td></tr>
<tr><td colspan="3">Checksum</td><td>Urgent pointer</td></tr>
<tr><td colspan="4">Options (variable)</td></tr>
</table>

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 waits for the SYN-ACK
  - … and retransmits the SYN 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 - ATTACK

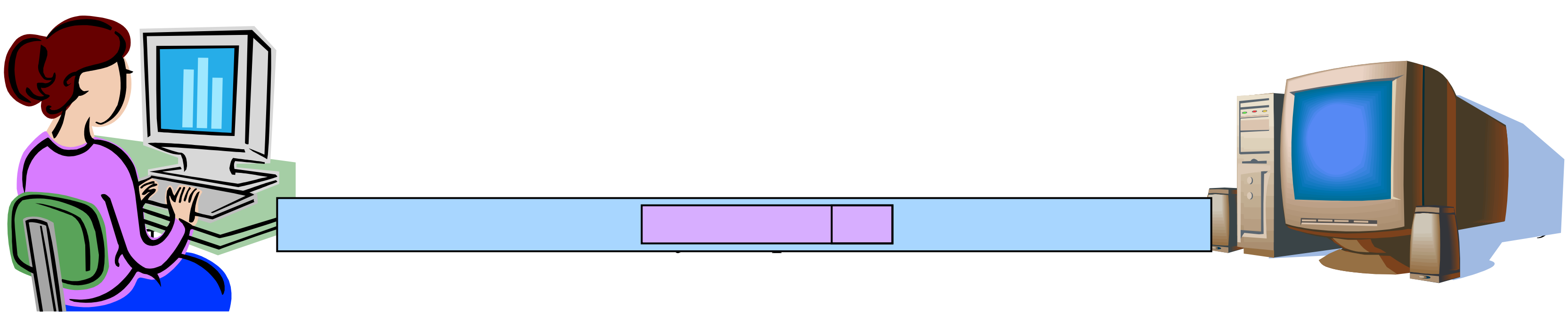
SYN Loss and Web Downloads

- 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

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

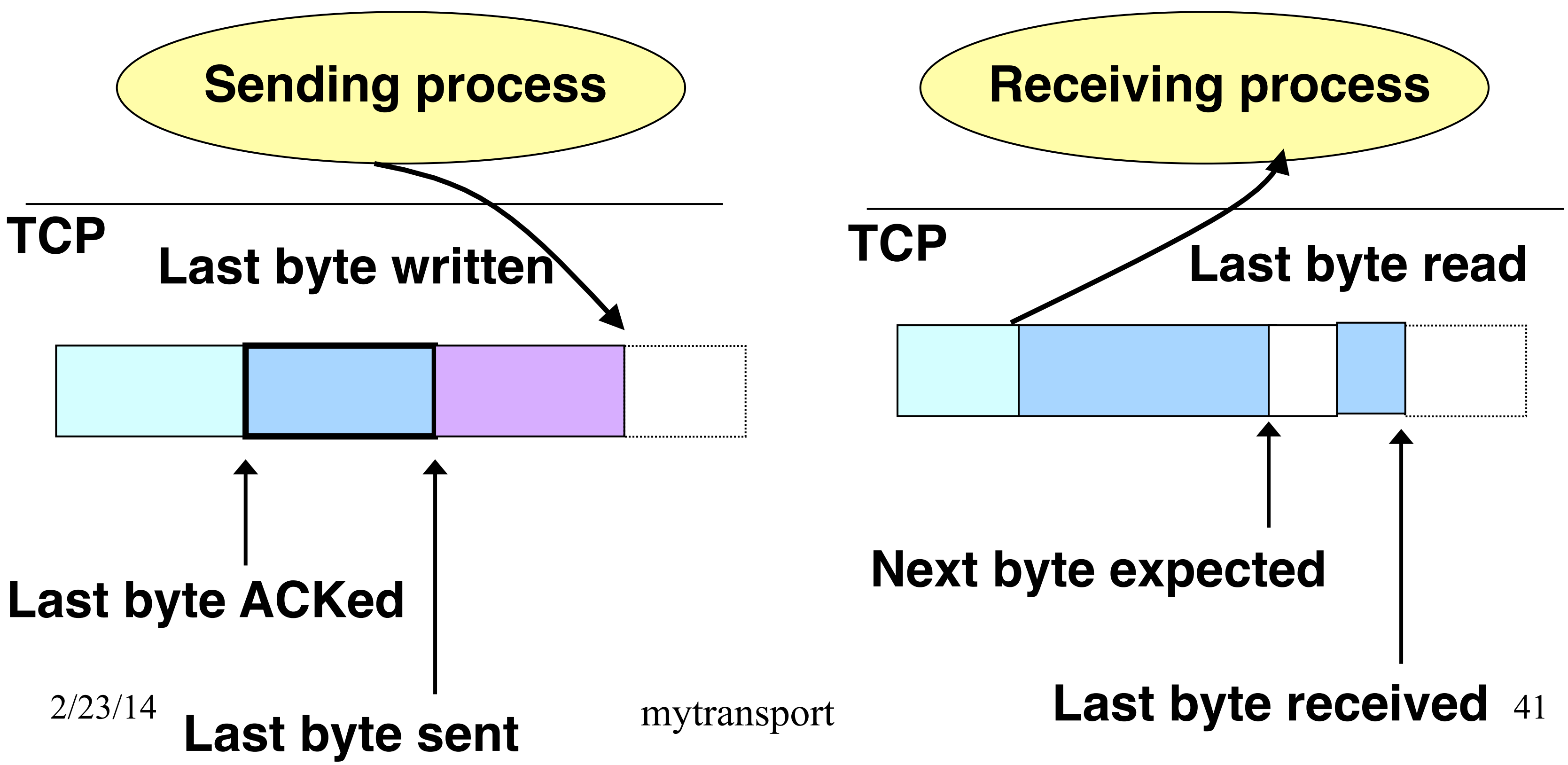


TCP Sliding Window Data Link Versus Transport

- Potentially connects many different hosts
  - need explicit connection establishment and termination
- Potentially different RTT
  - need adaptive timeout mechanism
- Potentially long delay in network
  - need to be prepared for arrival of very old packets
- Potentially different capacity at destination
  - need to accommodate different node capacity
- Potentially different network capacity
  - need to be prepared for network congestion

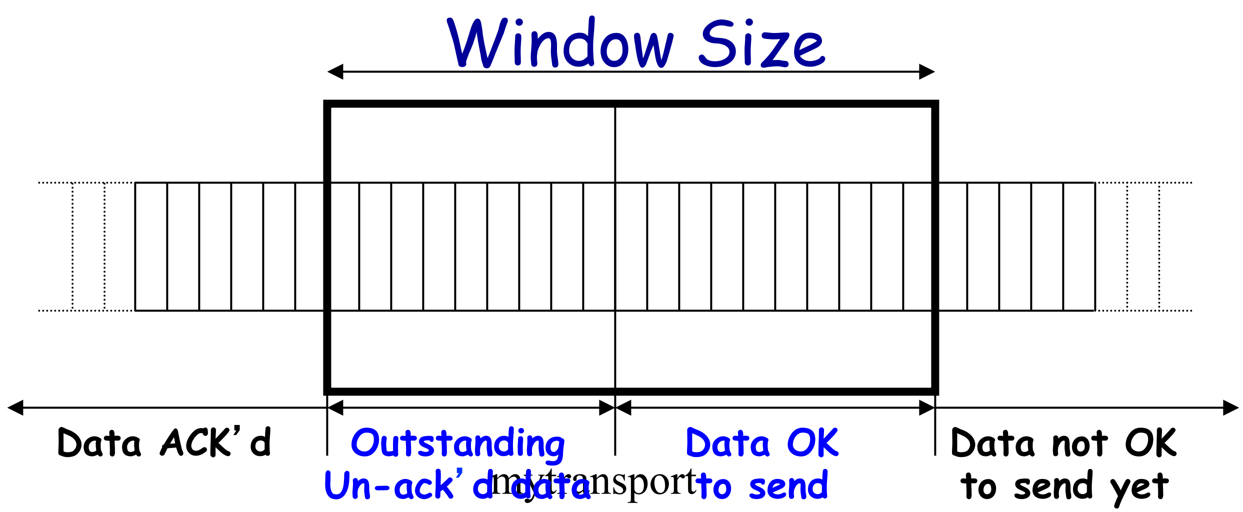
Sliding Window

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



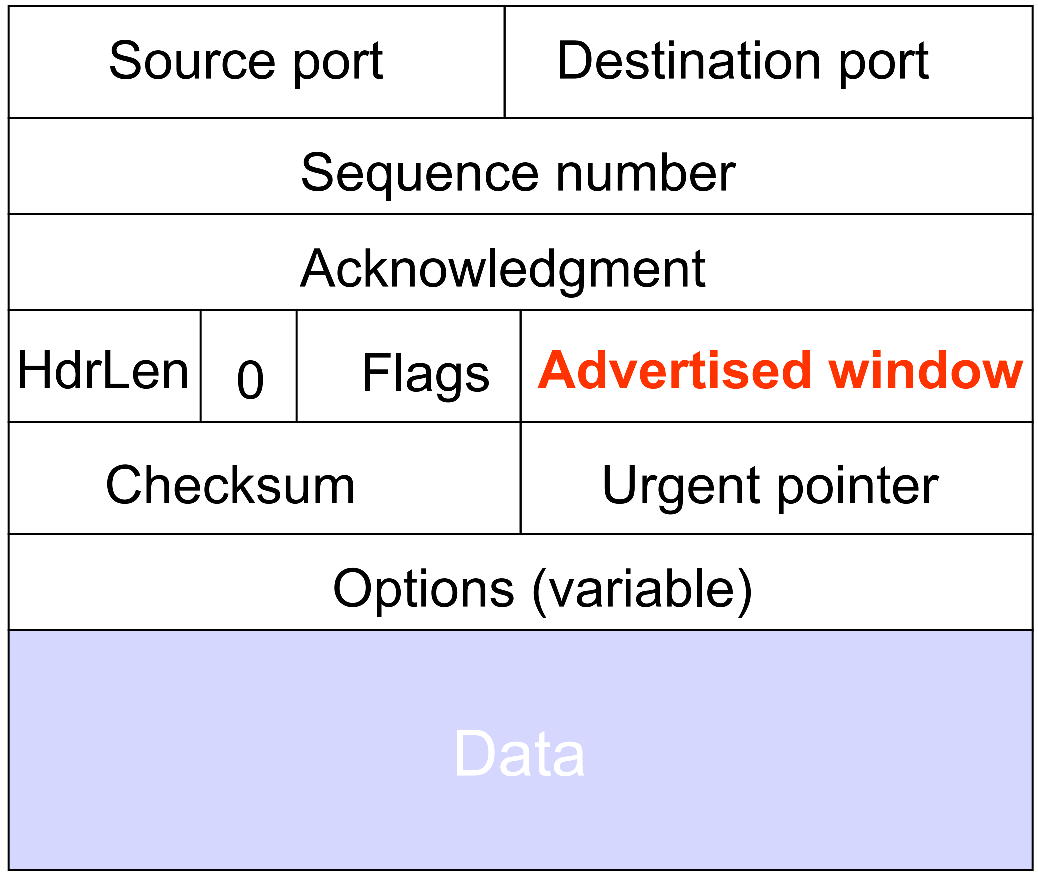
Receiver Buffering

- 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 Sender
- Receiver tells the Sender the amount of free space left
  - … and the Sender agrees not to exceed this amount

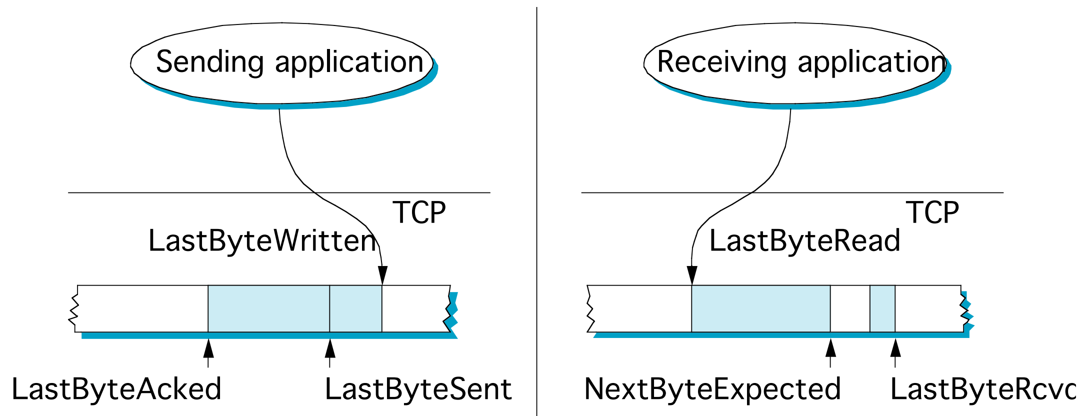


TCP Header for Receiver Buffering

Flags: SYN
FIN
RST
PSH
URG
ACK



Sliding Window Revisited



(a) (b)

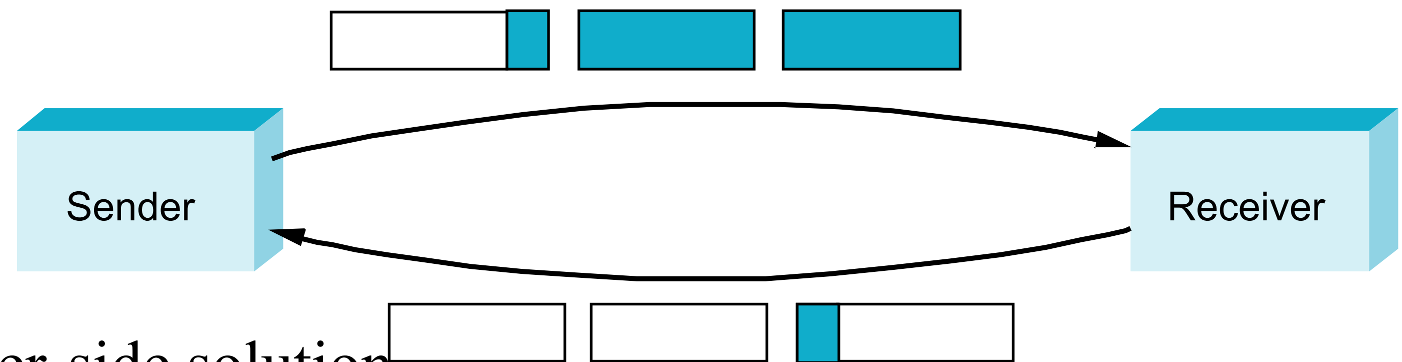
- Sending side
  - **`LastByteAcked <= LastByteSent`**
  - **`LastByteSent <= LastByteWritten`**
  - buffer bytes between **`LastByteAcked`** and **`LastByteWritten`**
- Receiving side
  - **`LastByteRead < NextByteExpected`**
  - **`NextByteExpected <= LastByteRcvd +1`**
  - buffer bytes between **`LastByteRead`** and **`LastByteRcvd`**

Flow Control

- Send buffer size: **`MaxSendBuffer`**
- Receive buffer size: **`MaxRcvBuffer`**
- Receiving side
  - **`LastByteRcvd`** - **`LastByteRead`** < = **`MaxRcvBuffer`**
  - **`AdvertisedWindow`** = **`MaxRcvBuffer`** - (**`NextByteExpected`** - **`NextByteRead`**)
- Sending side
  - **`LastByteSent`** - **`LastByteAcked`** < = **`AdvertisedWindow`**
  - **`EffectiveWindow`** = **`AdvertisedWindow`** - (**`LastByteSent`** - **`LastByteAcked`**)
  - **`LastByteWritten`** - **`LastByteAcked`** < = **`MaxSendBuffer`**
  - block sender if (**`LastByteWritten`** - **`LastByteAcked`**) + y > **`MaxSenderBuffer`**
- Always send ACK in response to arriving data segment
- Persist when **`AdvertisedWindow = 0`**

Silly Window Syndrome

- How aggressively does sender exploit open window?
- Because Ack indicates message size, a short message continues to always be short, Ack tells sender there is that much space available.



- Receiver-side solutions
  - after advertising zero window, wait for space equal to a maximum segment size (MSS) otherwise many small packets
  - delayed acknowledgements

Nagle's Algorithm

- How long does sender delay sending data?
  - too long: hurts interactive applications
  - too short: poor network utilization
  - strategies: timer-based vs self-clocking
- When application generates additional data
  - if new data fills a max segment (and window open): send it
  - Else //partially filled buffer
    - if there is unack'ed data in transit: buffer it until ACK arrives
    - else: send it

Protection Against Wrap Around

- 32-bit **`SequenceNum`**

Bandwidth	Time Until Wrap Around
T1 (1.5 Mbps)	6.4 hours
Ethernet (10 Mbps)	57 minutes
T3 (45 Mbps)	13 minutes
FDDI (100 Mbps)	6 minutes
STS-3 (155 Mbps)	4 minutes
STS-12 (622 Mbps)	55 seconds
STS-24 (1.2 Gbps)	28 seconds
OC48 (2.4Gbps)	14 seconds Solution?

Keeping the Pipe Full

- 16-bit **`AdvertisedWindow`**

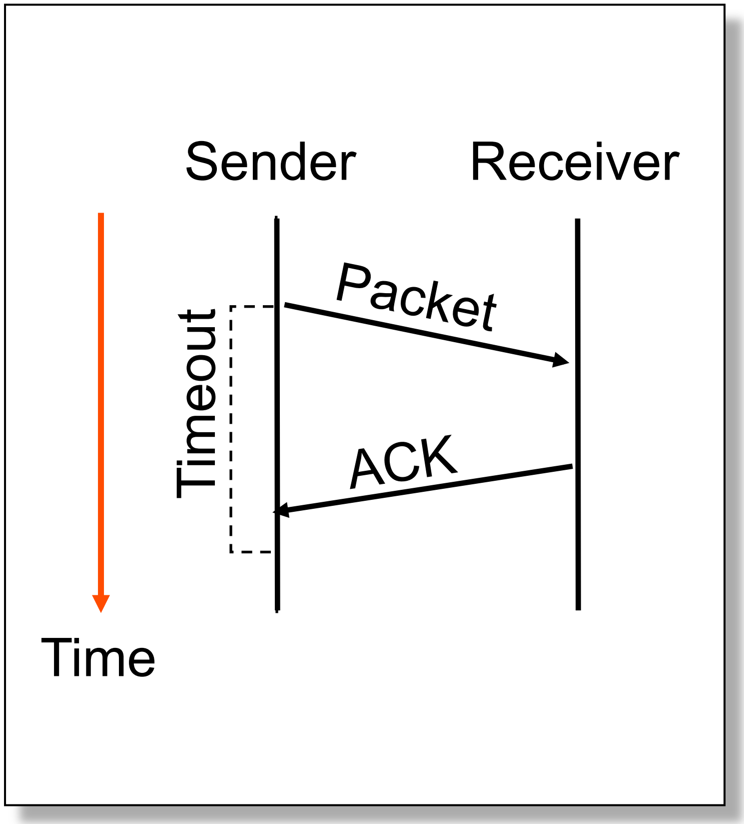
Bandwidth	Delay x Bandwidth Product
T1 (1.5 Mbps)	18KB
Ethernet (10 Mbps)	122KB
T3 (45 Mbps)	549KB
FDDI (100 Mbps)	1.2MB
STS-3 (155 Mbps)	1.8MB
STS-12 (622 Mbps)	7.4MB
STS-24 (1.2 Gbps)	14.8MB
OC48 (2.4Gbps)	29.6MB

assuming 100ms RTT

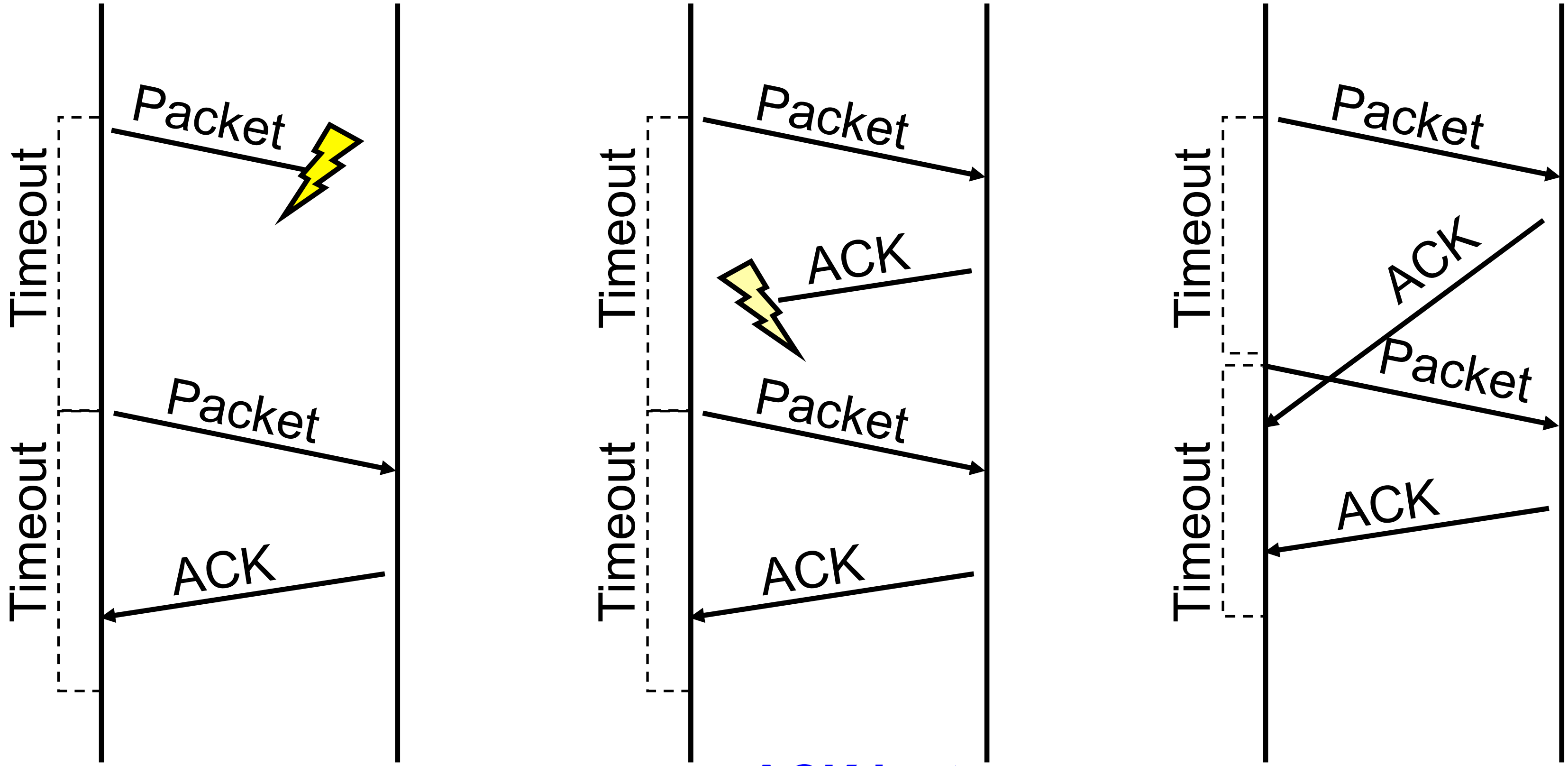
TCP Retransmissions

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



Packet lost

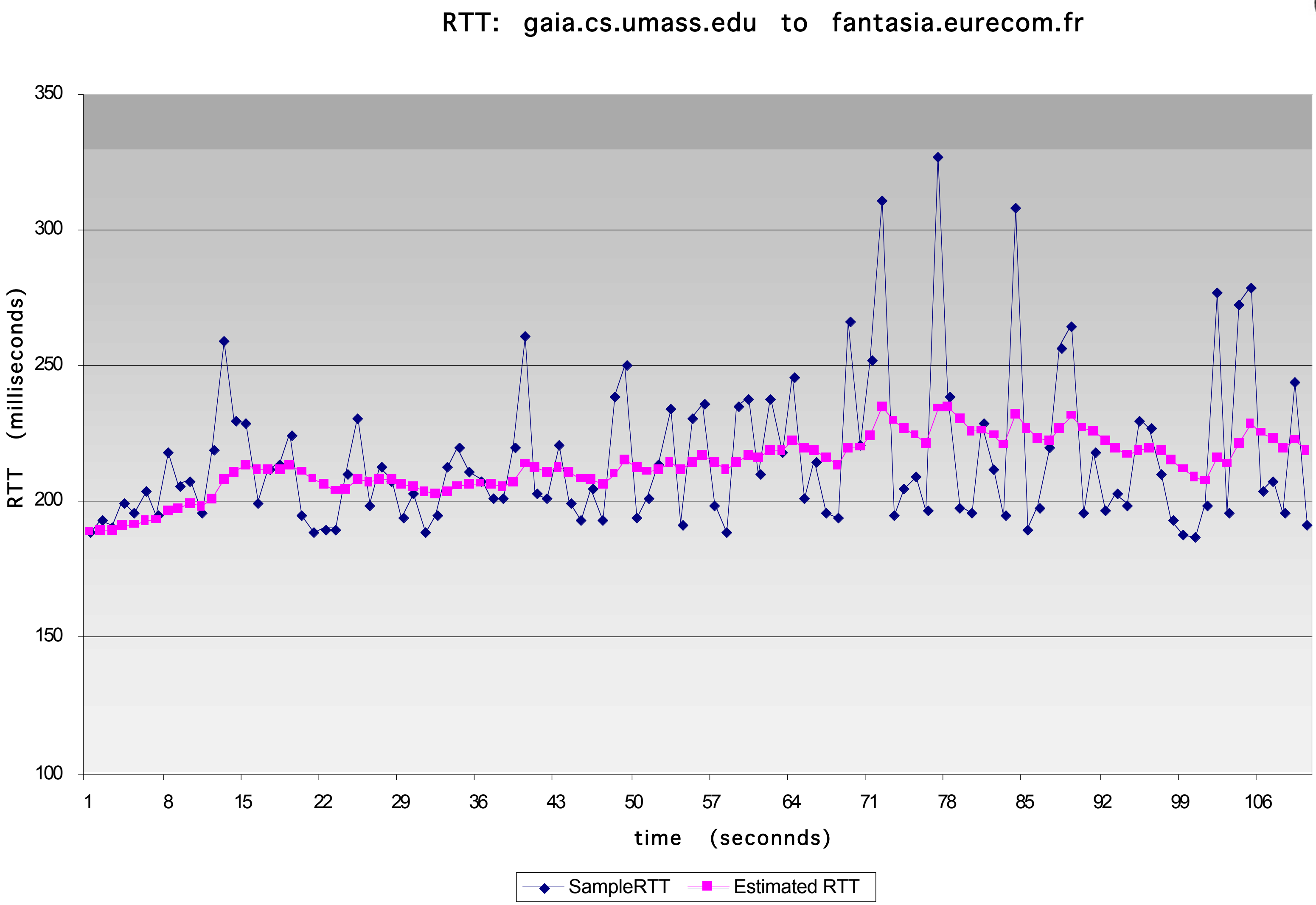
ACK lost
DUPLICATE PACKET

Early timeout
DUPLICATE PACKETS

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 (Dynamic)
  - 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

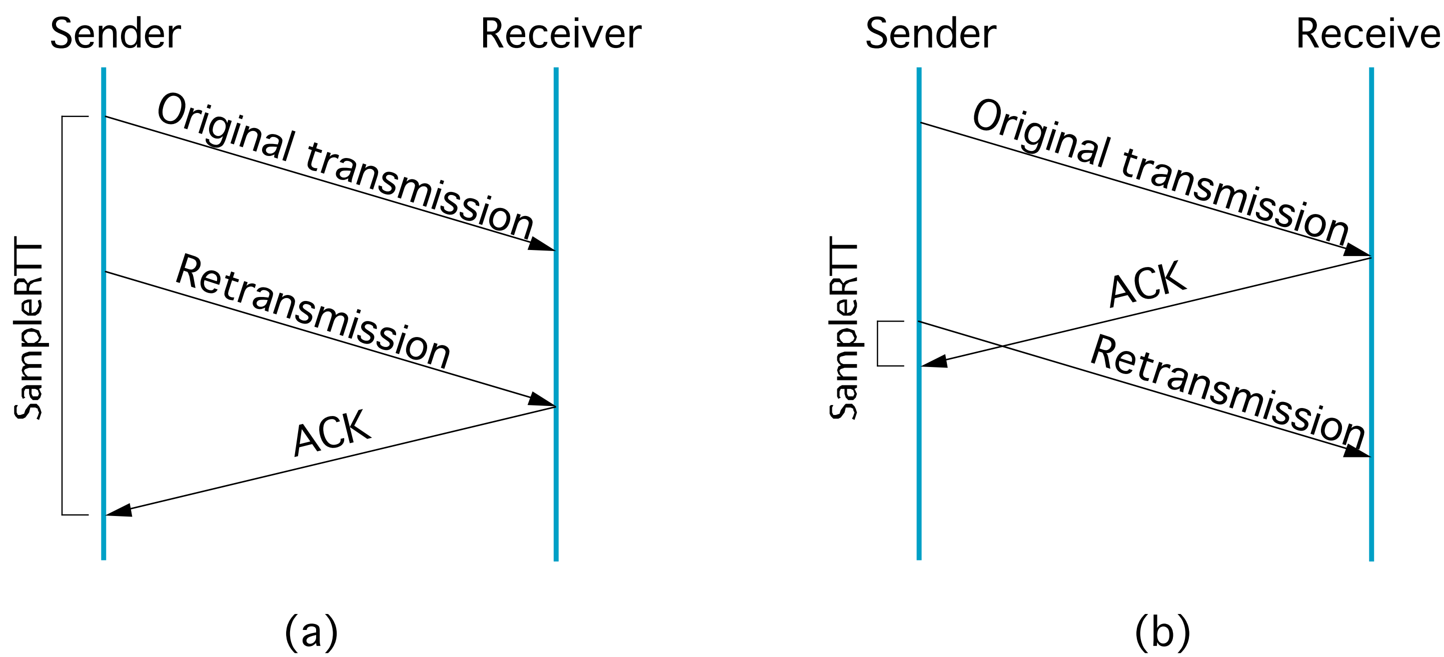
Yet Another Limitation…

- Does NOT consider variance in the RTT
  - If variance is small, the EstimatedRTT is pretty accurate
  - … but, if variance is large, the estimate is NOT good
- Better to directly consider the variance
  - Consider difference: SampleRTT – EstimatedRTT
  - Boost the estimate based on the difference
- Jacobson/Karels algorithm
  - See Peterson/Davie book for details

History:Adaptive Retransmission (Original Algorithm)

- Measure **SampleRTT** for each segment / ACK pair
- Compute weighted average of RTT
  - **EstRTT** = α x **EstRTT** + β x **SampleRTT**
  - where $\alpha + \beta = 1$
  - α between 0.8 and 0.9
  - β between 0.1 and 0.2
- Set timeout based on **EstRTT**
  - **TimeOut** = **2** x **EstRTT**

History: Karn/Partridge Algorithm



- Do not sample RTT when retransmitting
- Double timeout after each retransmission

History: Jacobson/ Karels Algorithm

- New Calculations for average RTT
- **`Diff`** = **`SampleRTT`** - **`EstRTT`**
- **`EstRTT = EstRTT + (`**δ **`x Diff)`**
- **`Dev = Dev +`** δ**`( |Diff| - Dev)`**
  - where δ is a factor between 0 and 1
- Consider variance when setting timeout value
- **`TimeOut`** = μ x **`EstRTT`** + ϕ x **`Dev`**
  - where $\mu = 1$ and $\phi = 4$
- Notes
  - algorithm only as good as granularity of clock (500ms on Unix)
  - accurate timeout mechanism important to congestion control (later)

TCP Extensions

- Implemented as header options
- Store timestamp in outgoing segments
- Extend sequence space with 32-bit timestamp (PAWS)
- Shift (scale) advertised window

TCP Issues

- Messages vs Byte Stream
- Record Boundaries…Database Records
- Solution
  – UrgPtr -- Sending app/process can tell receiver that there is urgent data, I.e., record boundary
  – Push -- app/process can tell TCP "send now"…TCP receiver must inform receiving app…force records

TCP Issues

- TCP is reliable byte stream…what about
  - Reliable, Ordered, etc: Request/Reply protocol
  - Unreliable byte stream: video
- TCP - High overhead to setup/close connection
  - Web experience
- TCP - Window Based
  - How about Rate based…
- TCP - Byte based…not message based

State Transition Diagram