Transport Protocols

Reading: Sections 2.5, 5.1, and 5.2

CS 375: Computer Networks

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Goals for Today's Lecture

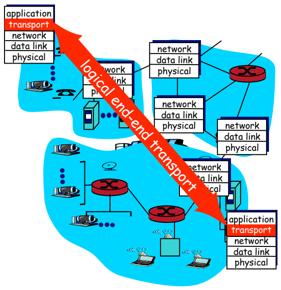
- Principles underlying transport-layer services
  - (De)multiplexing
  - Detecting corruption
  - Reliable delivery
  - Flow control
- Transport-layer protocols in the Internet
  - User Datagram Protocol (UDP)
    - Simple (unreliable) message delivery
    - Realized by a SOCK_DGRAM socket
  - Transmission Control Protocol (TCP)
    - Reliable bidirectional stream of bytes
    - Realized by a SOCK_STREAM socket

Role of Transport Layer

- Application layer
  - Between applications (e.g., browsers and servers)
  - E.g., HyperText Transfer Protocol (HTTP), File Transfer Protocol (FTP), Network News Transfer Protocol (NNTP)
- Transport layer
  - Between processes (e.g., sockets)
  - Relies on network layer and serves the application layer
  - E.g., TCP and UDP
- Network layer
  - Between nodes (e.g., routers and hosts)
  - Hides details of the link technology
  - E.g., IP

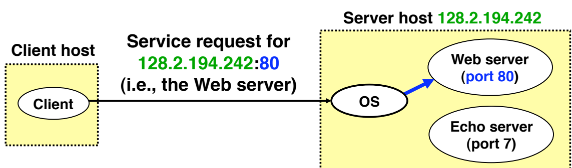
Transport Protocols

- 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 protocols available to applications
  - Internet: TCP and UDP

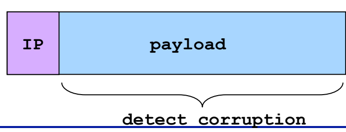


Two Basic Transport Features

- **Demultiplexing:** port numbers



- **Error detection:** checksums



User Datagram Protocol (UDP)

- Datagram messaging service
  - Demultiplexing of messages: port numbers
  - Detecting corrupted messages: checksum
- Lightweight communication between processes
  - Send messages to and receive them from a socket
  - Avoid overhead and delays of ordered, reliable delivery

SRC port	DST port
checksum	length
DATA	

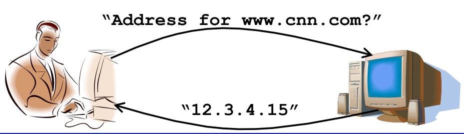
Why Would Anyone Use UDP?

- Fine 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 the application retransmit if needed



Transmission Control Protocol (TCP)

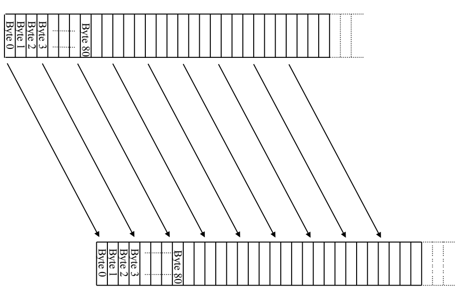
- Stream-of-bytes service
  - Sends and receives a stream of bytes, not messages
- Reliable, in-order delivery
  - Checksums to detect corrupted data
  - Sequence numbers to detect losses and reorder data
  - Acknowledgments & retransmissions for reliable delivery
- Connection oriented
  - Explicit set-up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

Breaking a Stream of Bytes into TCP Segments

TCP "Stream of Bytes" Service

Host A

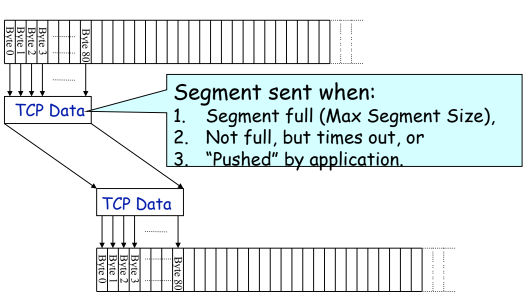
Host B



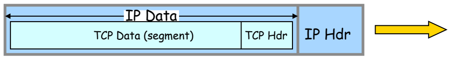
…Emulated Using TCP "Segments"

Host A

Host B

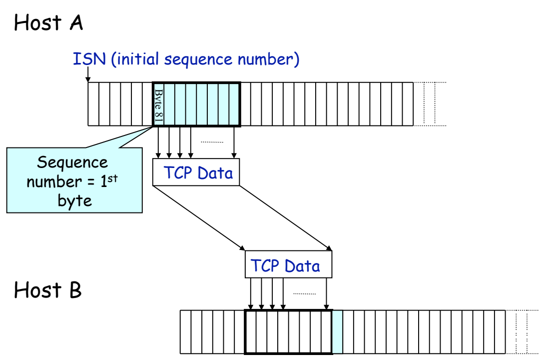


TCP Segment



- 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 Number



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

Reliable Delivery on a Lossy Channel With Bit Errors

16

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?" ☺

17

Some Take-Aways from the Example

- Acknowledgments from receiver
  - Positive: "okay" or "uh huh" 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

18

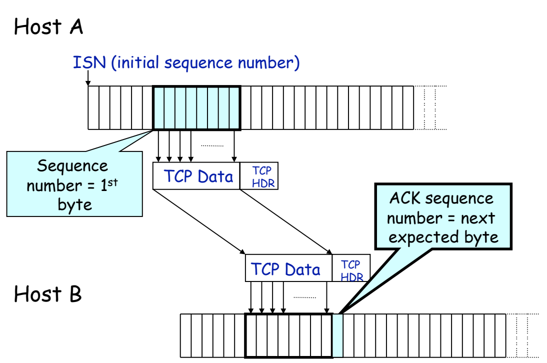
Challenges of Reliable Data Transfer

- 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

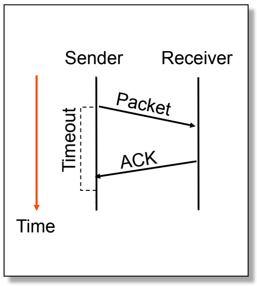
- **Detect bit errors:** checksum
  - Used to detect corrupted data at the receiver
  - …leading the receiver to drop the packet
- **Detect missing data:** sequence number
  - Used to detect a gap in the stream of bytes
  - ... and for putting the data back in order
- **Recover from lost data:** retransmission
  - Sender retransmits lost or corrupted data
  - Two main ways to detect lost packets

TCP Acknowledgments

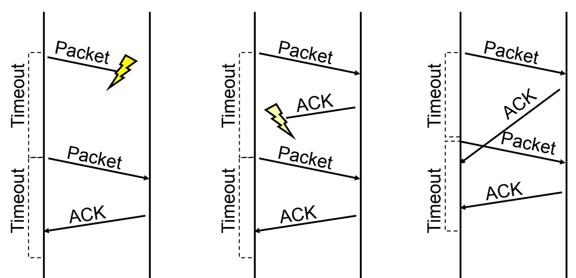


Automatic Repeat reQuest (ARQ)

- 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

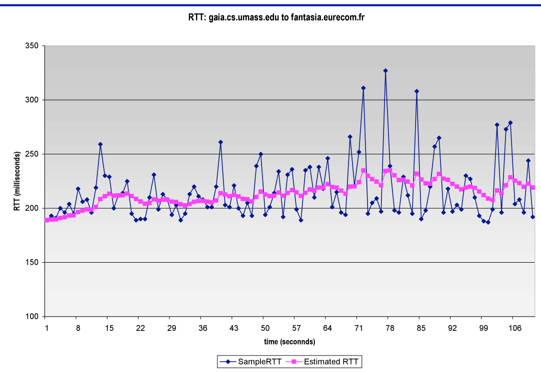
ACK lost
DUPLICATE PACKET

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
  - Expect ACK to arrive after an "round-trip time"
  - … 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

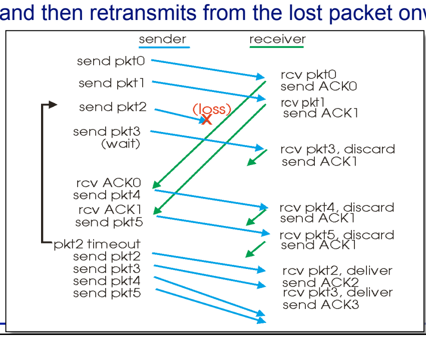


A Flaw in This Approach

- 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

Still, Timeouts are Inefficient

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



Fast Retransmission

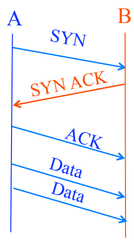
- Better solution possible under sliding window
  - Although packet n might have been lost
  - … packets n+1, n+2, and so on might get through
- Idea: have the receiver send ACK packets
  - ACK says that receiver is still awaiting n^{th} packet
    - And *repeated* ACKs suggest later packets have arrived
  - 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… ☺

Starting and Ending a Connection: TCP Handshakes

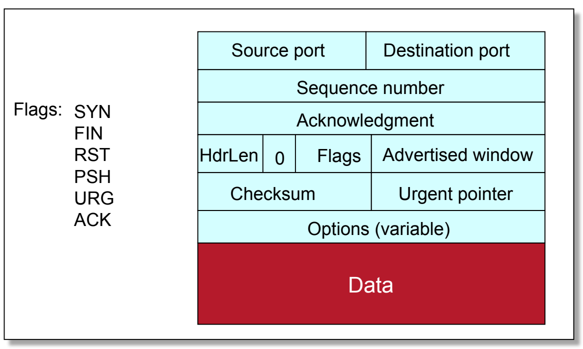
Establishing a TCP Connection



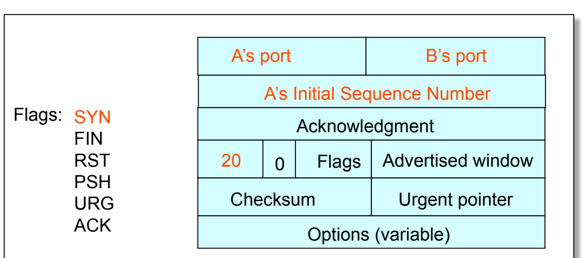
Each host tells its ISN to the other host.

- 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



Step 1: A's Initial SYN Packet



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

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

A tells B it is okay to start sending

… upon receiving this packet, B can start sending data

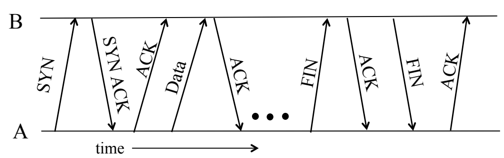
What if the SYN Packet Gets Lost?

- 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 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

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 (each end of) 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

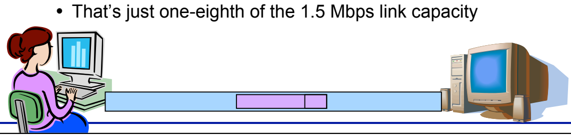
Sending/Receiving the FIN Packet

- Sending a FIN: close()
  - Process is done sending data via the socket
  - Process invokes “close()” to close the socket
  - Once TCP has sent all of the outstanding bytes…
  - … then TCP sends a FIN
- Receiving a FIN: EOF
  - Process is reading data from the socket
  - Eventually, the attempt to read returns an EOF

Flow Control: TCP Sliding Window

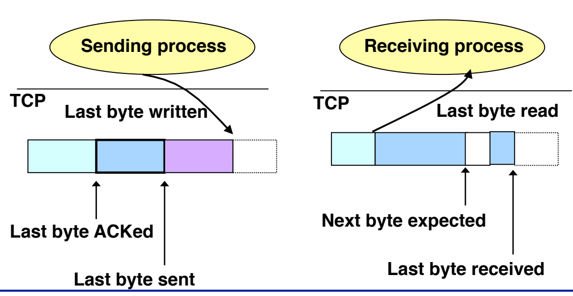
Motivation for Sliding Window

- 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



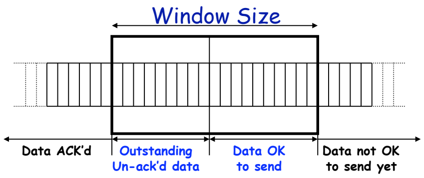
Sliding Window

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

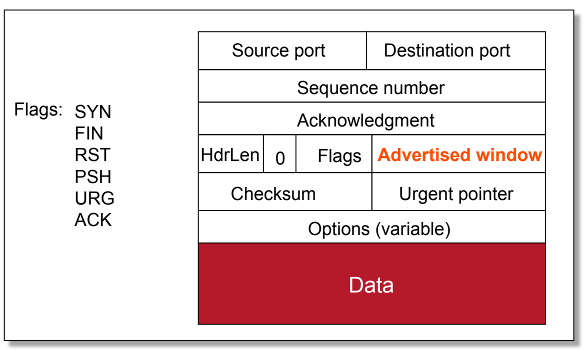


Receiver Buffering

- 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



Conclusions

- Transport protocols
  - Multiplexing and demultiplexing
  - Checksum-based error detection
  - Sequence numbers
  - Retransmission
  - Window-based flow control
- Reading for this week
  - Sections 2.5, 5.1-5.2, and 6.1-6.4
- Next lecture
  - Congestion control