

Transport Layer

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COS 461: Computer Networks

http://www.cs.princeton.edu/courses/archive/spr14/cos461/

IP Protocol Stack: Key Abstractions



- Transport layer is where we "pay the piper"
 - Provide applications with good abstractions
 - Without support or feedback from the network

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Transport Protocols

- Logical communication between processes
 - -Sender divides a message into segments
 - Receiver reassembles segments into message
- Transport services
 - -(De)multiplexing packets
 - Detecting corrupted data
 - Optionally: reliable delivery, flow control, ...

Two Basic Transport Features

• Demultiplexing: port numbers

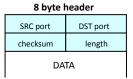
Service request for 128.2.194.242:80 (i.e., the Web server)

OS Echo server (port 7)

• Error detection: checksums

User Datagram Protocol (UDP)

- Lightweight communication between processes
 - Send and receive messages
 - Avoid overhead of ordered, reliable delivery
 - No connection setup delay, no in-kernel connection state
- Used by popular apps
 - Query/response for DNS
 - Real-time data in VoIP



Advantages of UDP

- Fine-grain control
 - UDP sends as soon as the application writes
- No connection set-up delay
 - UDP sends without establishing a connection
- No connection state
 - No buffers, parameters, sequence #s, etc.
- · Small header overhead
 - UDP header is only eight-bytes long

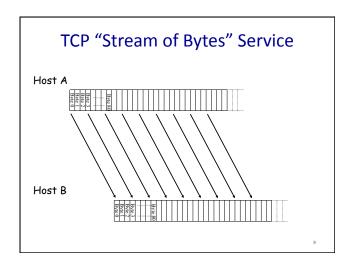
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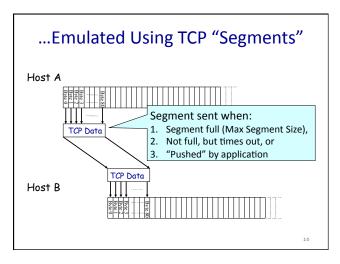
Transmission Control Protocol (TCP)

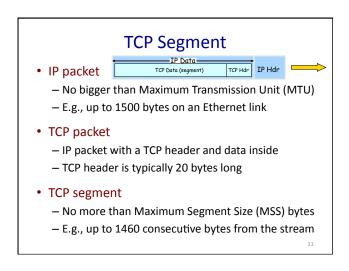
- · Stream-of-bytes service
 - Sends and receives a stream of bytes
- · Reliable, in-order delivery
 - Corruption: checksums
 - Detect loss/reordering: sequence numbers
 - Reliable delivery: acknowledgments and retransmissions
- · Connection oriented
 - Explicit set-up and teardown of TCP connection
- Flow control
 - Prevent overflow of the receiver's buffer space
- Congestion control
 - Adapt to network congestion for the greater good

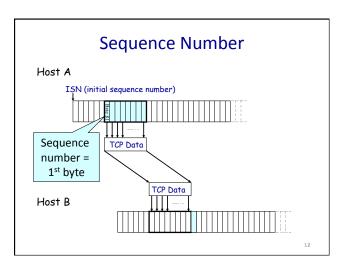
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Breaking a Stream of Bytes into TCP Segments









Initial Sequence Number (ISN)

- Sequence number for the very first byte
 - E.g., Why not a de facto ISN of 0?
- Practical issue: reuse of port numbers
 - Port numbers must (eventually) get used again
 - ... and an old packet may still be in flight
 - ... and associated with the new connection
- So, TCP must change the ISN over time
 - Set from a 32-bit clock that ticks every 4 microsec
 - ... which wraps around once every 4.55 hours!

Reliable Delivery on a Lossy **Channel With Bit Errors**

Challenges of Reliable Data Transfer

- Over a perfectly reliable channel: Done
- Over a channel with bit errors
 - Receiver detects errors and requests retransmission
- Over a lossy channel with bit errors
 - Some data missing, others corrupted
 - Receiver cannot easily detect loss
- Over a channel that may reorder packets
 - Receiver cannot easily distinguish loss vs. out-of-order

An Analogy

- · 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? Has she 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?"

Take-Aways from the Example

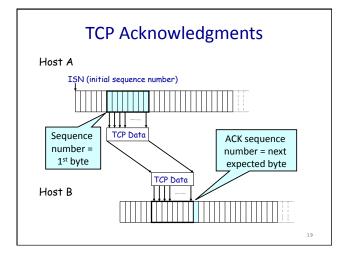
- · Acknowledgments from receiver
 - Positive: "okay" or "uh huh" or "ACK"
 - Negative: "please repeat that" or "NACK"
- Retransmission by the sender
 - After not receiving an "ACK"
 - After receiving a "NACK"
- Timeout by the sender ("stop and wait")
 - Don't wait forever without some acknowledgment

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TCP Support for Reliable Delivery

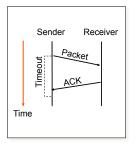
- · Detect bit errors: checksum
 - Used to detect corrupted data at the receiver
 - ...leading the receiver to drop the packet
- Detect missing data: sequence number
 - Used to detect a gap in the stream of bytes
 - ... and for putting the data back in order
- Recover from lost data: retransmission
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

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Automatic Repeat reQuest (ARQ)

- ACK and timeouts
 - Receiver sends ACK when it receives packet
 - Sender waits for ACK and times out
- Simplest ARQ protocol
 - Stop and wait
 - Send a packet, stop and wait until ACK arrives



Quick TCP Math

• Initial Seq No = 501. Sender sends 4500 bytes successfully acknowledged. Next sequence number to send is:

(A) 5000 (B) 5001 (C) 5002

• Next 1000 byte TCP segment received. Receiver acknowledges with ACK number:

(A) 5001 (B) 6000 (C) 6001

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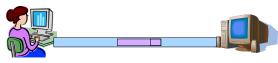
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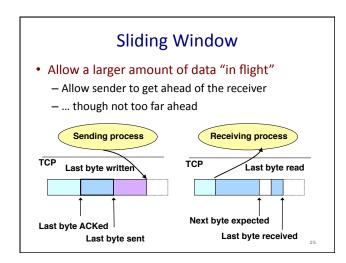
Flow Control: TCP Sliding Window

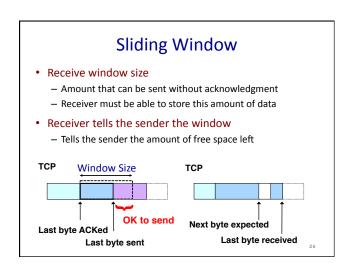
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Motivation for Sliding Window

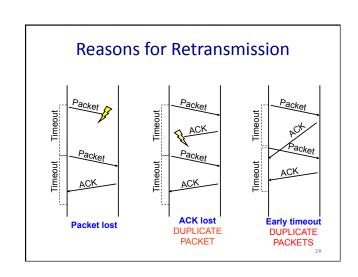
- Stop-and-wait is inefficient
 - Only one TCP segment is "in flight" at a time
- Consider: 1.5 Mbps link with 50 ms round-trip-time (RTT)
 - Assume TCP segment size of 1 KB (8 Kbits)
 - 8 Kbits/segment at 50 msec/segment → 160 Kbps
 - That's 11% of the capacity of 1.5 Mbps link







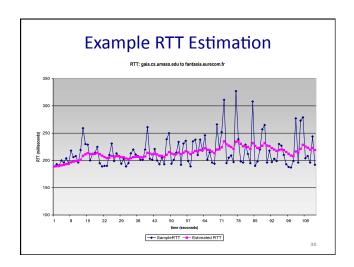
Optimizing Retransmissions



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?
 - Running average of delay to receive an ACK

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Still, timeouts are slow (≈RTT)

- When packet n is lost...
 - ... packets n+1, n+2, and so on may get through
- Exploit the ACKs of these packets
 - ACK says receiver is still awaiting nth packet
 - Duplicate ACKs suggest later packets arrived
 - Sender uses "duplicate ACKs" as a hint
- Fast retransmission
 - Retransmit after "triple duplicate ACK"

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Effectiveness of Fast Retransmit

- When does Fast Retransmit work best?
 - High likelihood of many packets in flight
 - -Long data transfers, large window size, ...
- Implications for Web traffic
 - Most Web transfers are short (e.g., 10 packets)
 - So, often there aren't many packets in flight
 - Making fast retransmit is less likely to "kick in"
 - Forcing users to click "reload" more often...

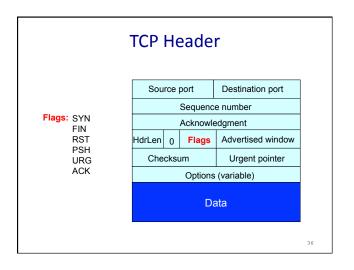
Effectiveness of Fast Retransmit

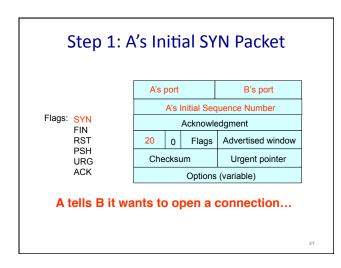
- When does Fast Retransmit work best?
 - (A) Short data transfers
 - (B) Large window size
 - (C) Small RTT networks

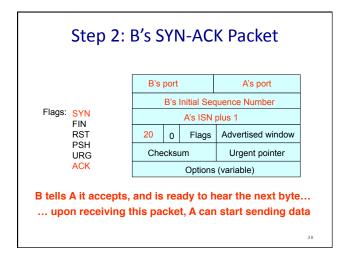
Starting and Ending a Connection: TCP Handshakes

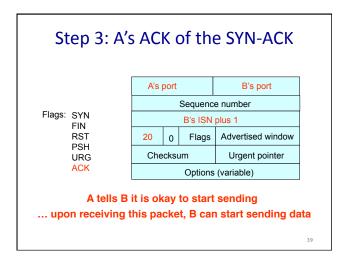
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Establishing a TCP Connection A SYN B SYN ACK Bala Each host tells its ISN to the other host. • Three-way handshake to establish connection — Host A sends a SYN (open) to the host B — Host B returns a SYN acknowledgment (SYN ACK) — Host A sends an ACK to acknowledge the SYN ACK





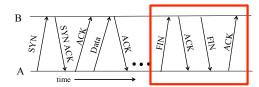




SYN Loss and Web Downloads

- Upon sending SYN, sender sets a timer
 - If SYN lost, timer expires before SYN-ACK received
 - Sender retransmits SYN
- How should the TCP sender set the timer?
 - No idea how far away the receiver is
 - Some TCPs use default of 3 or 6 seconds
- · Implications for web download
 - User gets impatient and hits reload
 - ... Users aborts connection, initiates new socket
 - Essentially, forces a fast send of a new SYN!

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

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Sending/Receiving the FIN Packet

- Sending a FIN: close()
 - Process is done sending data via socket
 - Process invokes "close()"
 - Once TCP has sent all the outstanding bytes...
 - ... then TCP sends a FIN
- Receiving a FIN: EOF
 - Process is reading data from socket
 - Eventually, read call returns an EOF

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Conclusions

- Transport protocols
 - Multiplexing and demultiplexing
 - Checksum-based error detection
 - -Sequence numbers
 - Retransmission
 - Window-based flow control