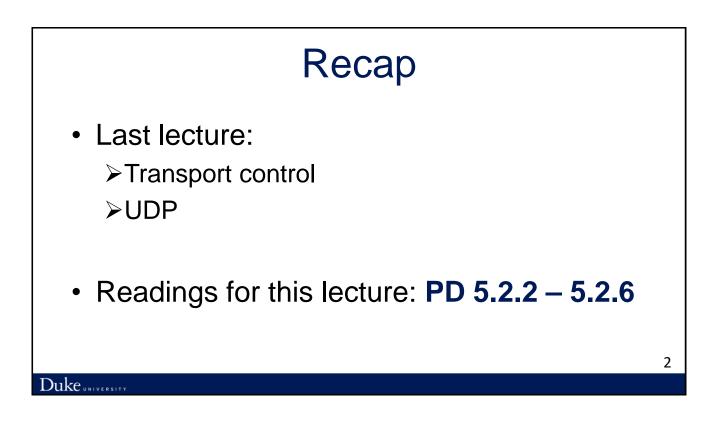
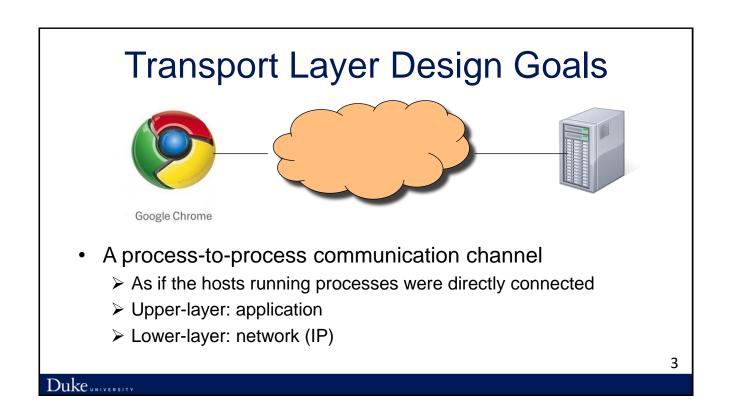
ECE 356/COMPSI 356 Computer Network Architecture

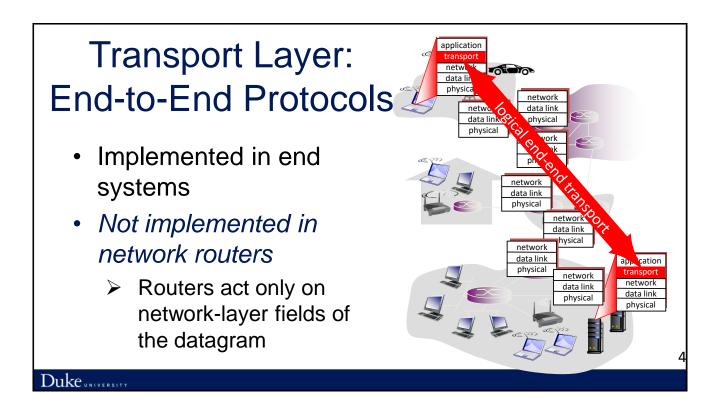
How TCP Achieves Reliable Operation

Monday October 28th, 2019

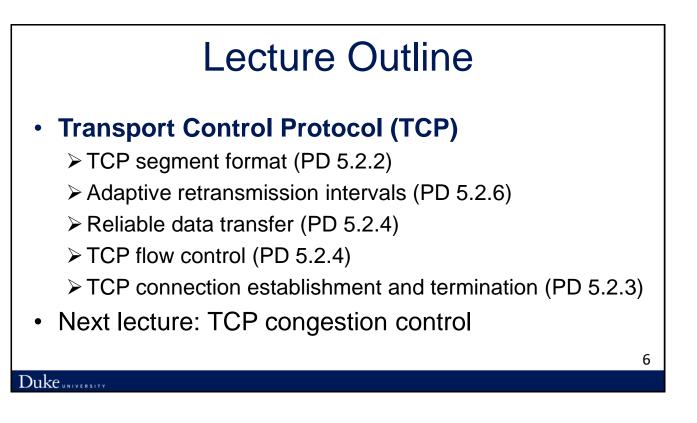
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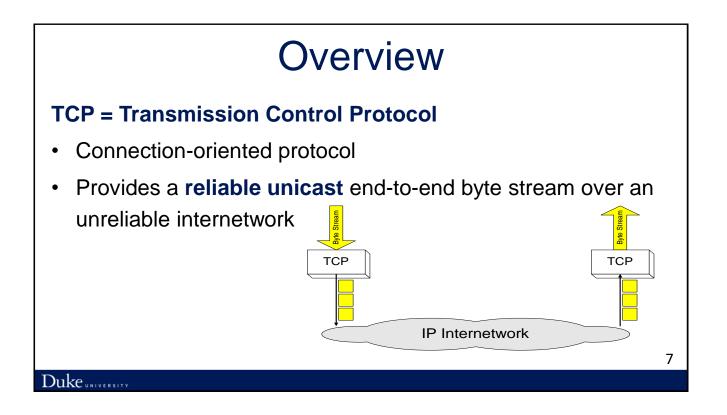


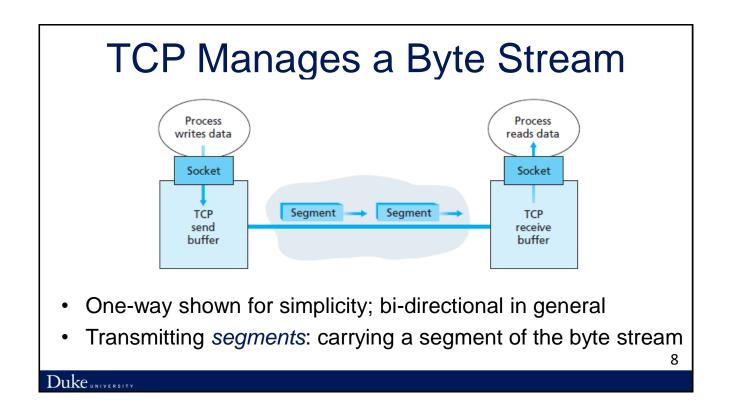




User Datagram Protocol (UDP) Minimal transport service: non-guaranteed datagram delivery "A no-frills, bare-bones transport protocol" Applications Applications "Almost a null protocol" Only provides: UDP UDP Multiplexing by port number IP Checksumming of data Has important advantages over TCP No connection setup: connectionless 5 Duke UNIVERSITY







Unique Design Challenges

We've learned how to reliably transmit over a direct link
 Coding/encoding, framing, sliding window

What's new?

- 1. Process-to-process communication \rightarrow connection setup
- 2. Heterogeneity
 > Bandwidth varies: how fast should the sender send?
 > RTT varies: when should a sender time out?
- 3. Out of order
- 4. Resource sharing
 - Many senders share a link in the middle of the network

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TCP: Connection-Oriented

- Host processes must first "handshake" with each other
 - Exchange messages
 - Establish the parameters of data transfer
- Note: state is established in the *end hosts*, <u>not the</u> <u>intermediate routers</u>
 - Intermediate routers are oblivious to TCP connections
 - > Note the difference with circuit switching
- Full-duplex
- Point-to-point only: no broadcast, no multicast

10

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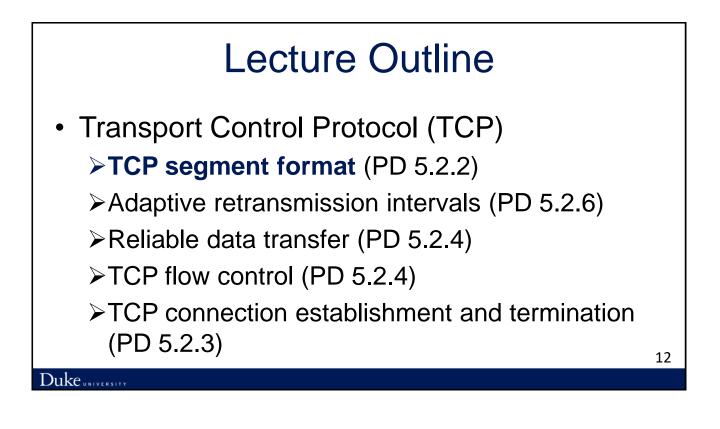
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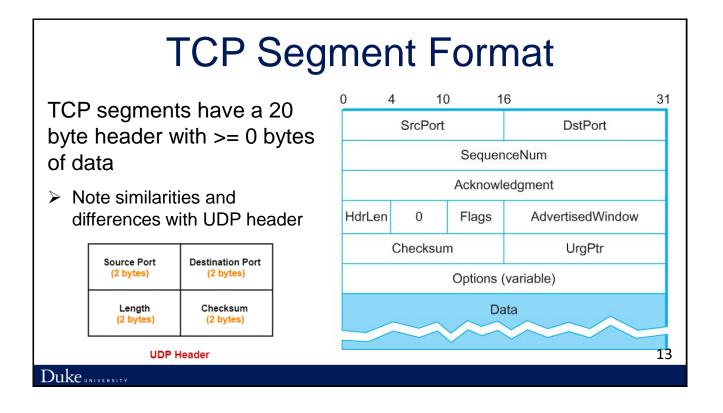
TCP: Key Points to Remember

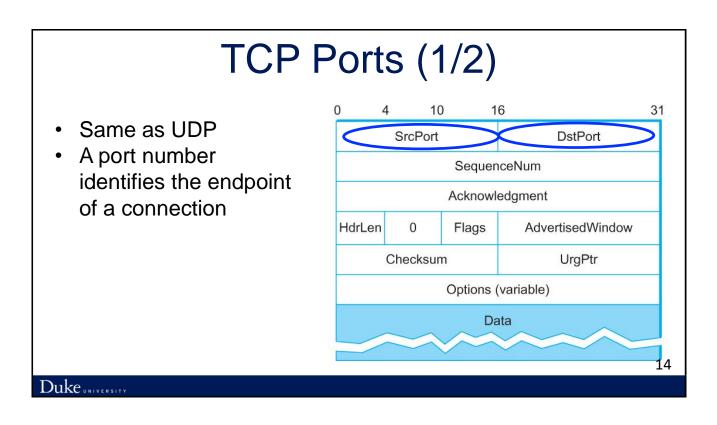
- Connection-oriented unicast operation
- Reliable, in-order byte stream service
- Flow control: not to overrun a receiver

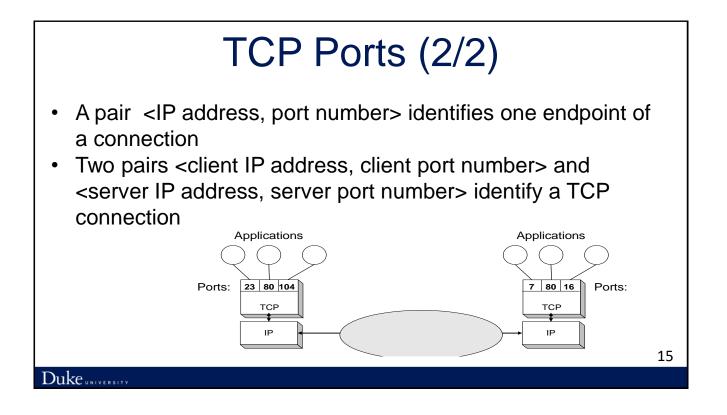
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• Congestion control: not to congest the network



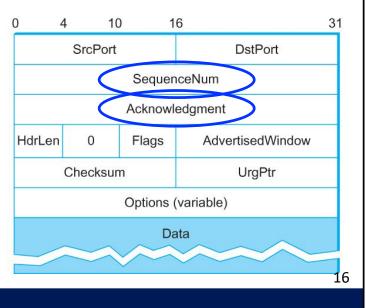






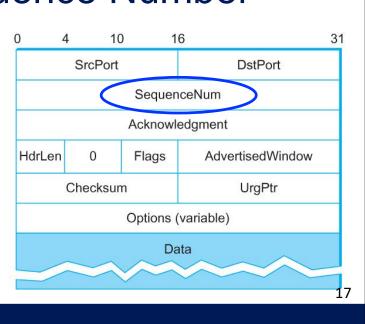
TCP: Reliable Communications

• Via sequence numbers and acknowledgements



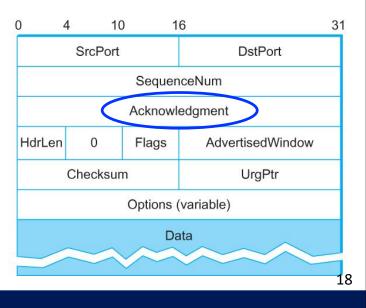
TCP Sequence Number

- Identifies the first byte in the segment
- Initial Sequence Number of a connection is set during connection establishment

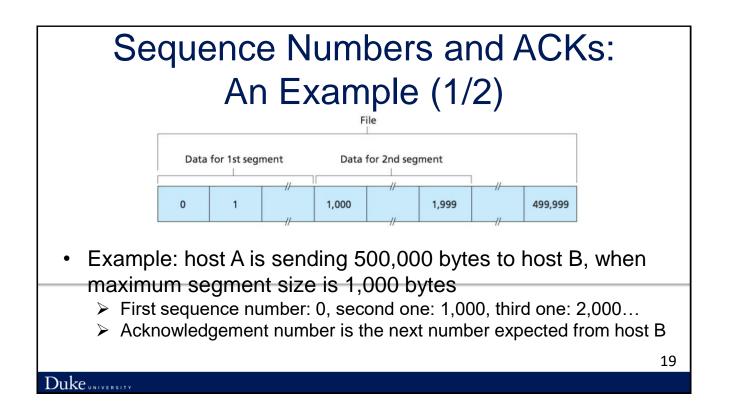


TCP Acknowledgement

- Acknowledgements are piggybacked
- The AckNo contains the next SeqNo that a host is expecting
- ACK is cumulative



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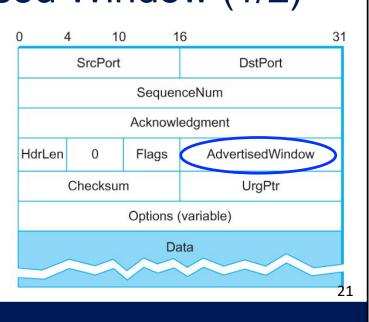


Sequence Numbers and ACKs: An Example (2/2)

- Acknowledgement number is the next number expected from host B
 - E.g., host A received bytes 0 535 from host B, and is about to send a segment: host A puts 536 in its acknowledgement field
- Acknowledgements are cumulative:
 - A received 0 535 and 900 1000: it puts 536 in the acknowledgement

TCP Advertised Window (1/2)

- Used to implement flow control
- Each side of the connection advertises the window size
- Window size is the maximum number of bytes that a receiver can accept

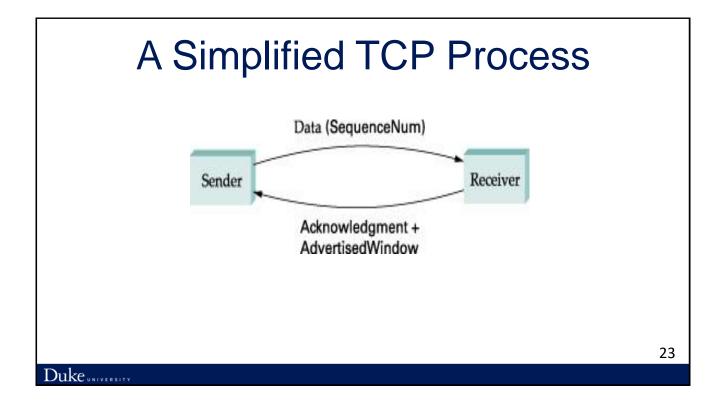


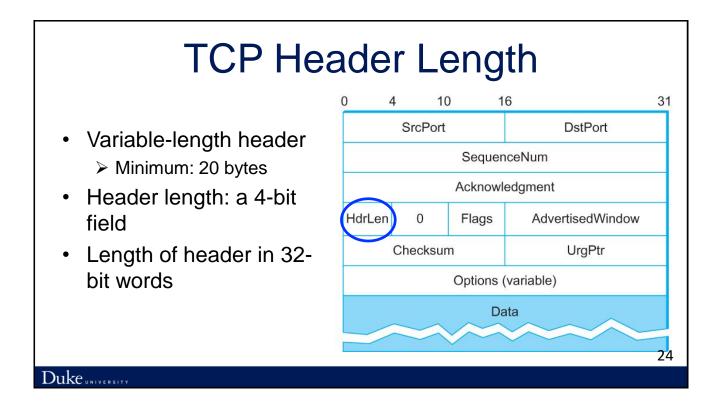
TCP Advertised Window (2/2)

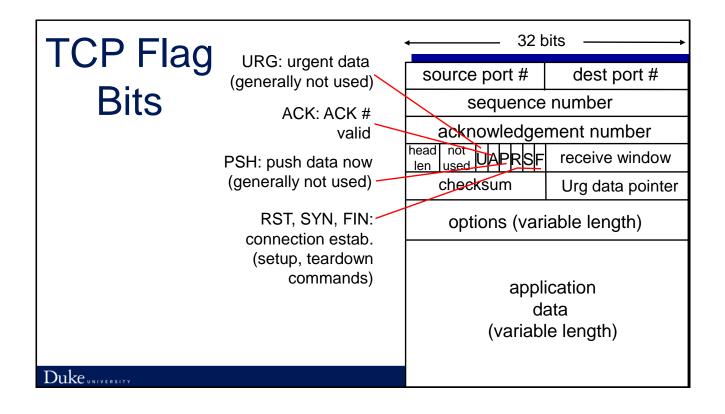
- Included in every segment: dynamic
- Maximum window size is 2¹⁶-1= 65,535 bytes
 - Problematic for highspeed links



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TCP Flag Bits: URG, ACK, PSH

- URG: Urgent pointer is valid (not encouraged to use)
 - If the bit is set, the following bytes contain an urgent message in the range:

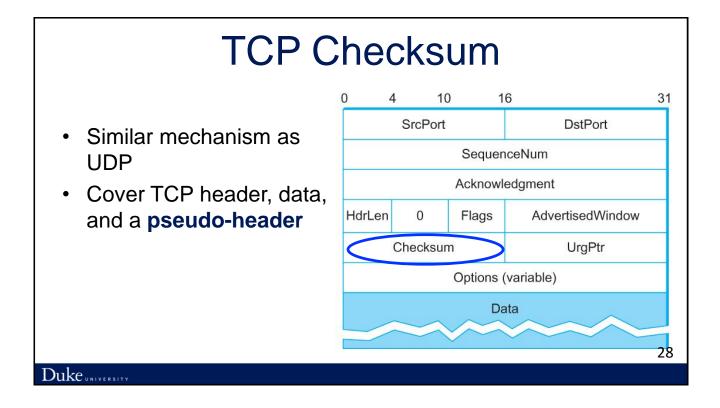
SeqNo <= urgent message < SeqNo+urgent pointer

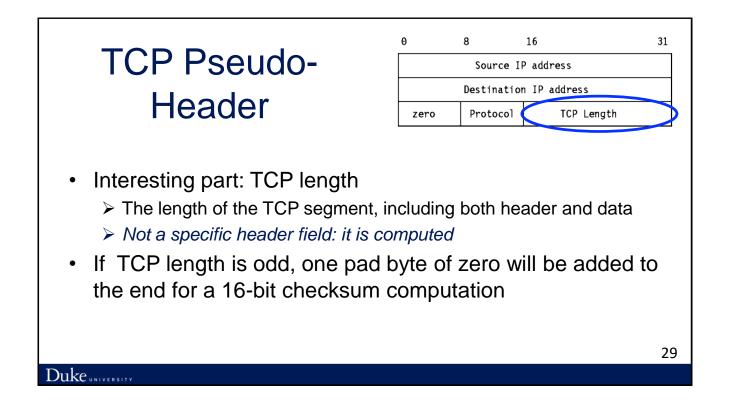
- ACK: Acknowledgement Number is valid
 - Segment contains a valid ACK
- PSH: PUSH Flag (generally not used)
 - Notification from sender to the receiver that the receiver should pass all data that it has to the application.
 - > Normally set by a sender when the sender's buffer is empty

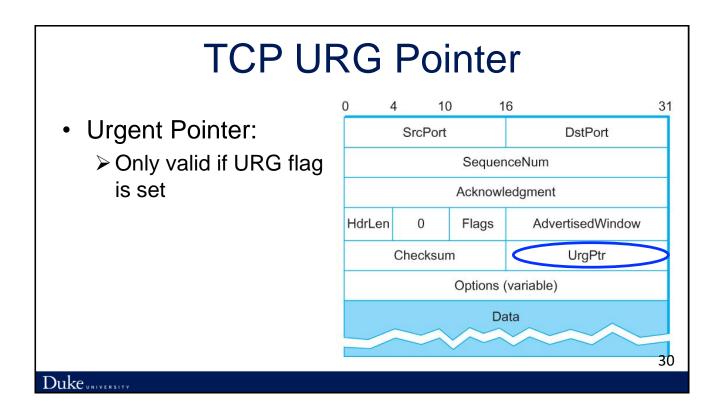
TCP Flag Bits: Connection Establishment

• RST: Reset the connection

- Receiver of a RST terminates the connection and indicates higher layer application about the reset
- SYN: Synchronize sequence numbers
 - > Sent in the first packet when initiating a connection
- FIN: Sender is finished with sending
 > Used for closing a connection







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DstPort

AdvertisedWindow

UrgPtr

TCP Segment Format: Key Points to Remember

10

SequenceNum

Acknowledgment

Flags

SrcPort

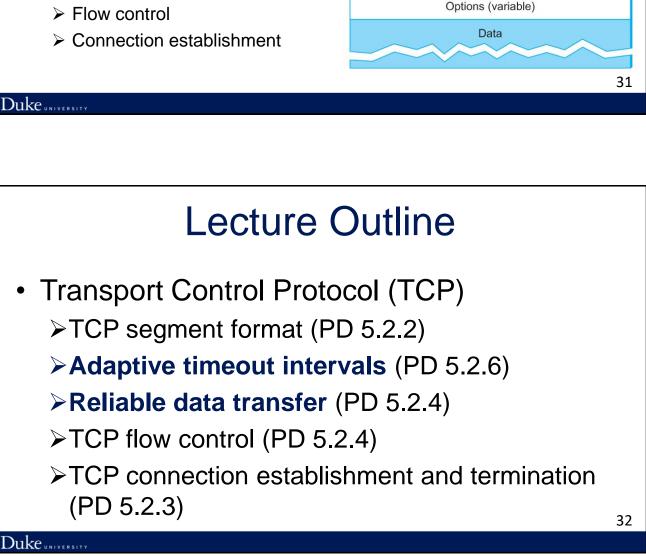
0

Checksum

HdrLen

- Like UDP, provides for demultiplexing and checksumming
- Also has dedicated fields for
 - Reliable communications

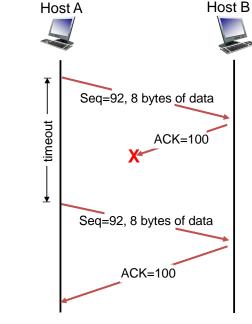
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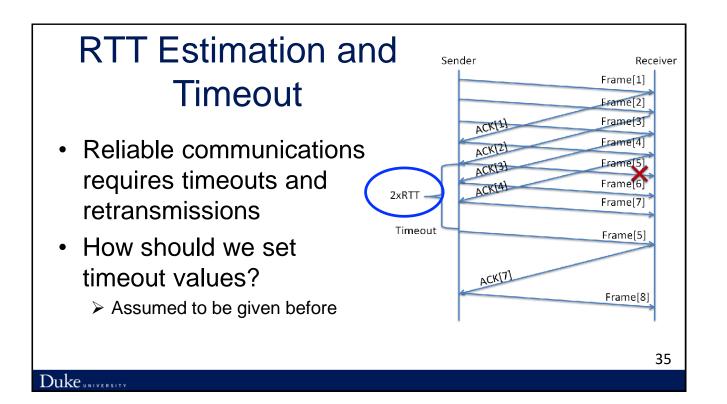


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Same Core Principles as What We Studied for Link Layer Reliability

- Differences include:
 - Adaptive timeout values
 - One timeout variable per connection
 - Retransmissions triggered by timeouts and duplicate acknowledgements





How to Set TCP Timeout Value?

- Definitely needs to be longer than RTT
 > But RTT varies in practice
- If timeout is too short:
 - Premature timeout
 - Unnecessary retransmissions
- If timeout is too long:
 - Slow reaction to segment loss
 - Large average delays when the number of retransmissions is large

RTT Estimation and Timeout

- Set timeout to RTT + "safety margin"
- Two parts:

➤Estimating RTT

Calculating the additional margin

Estimating RTT

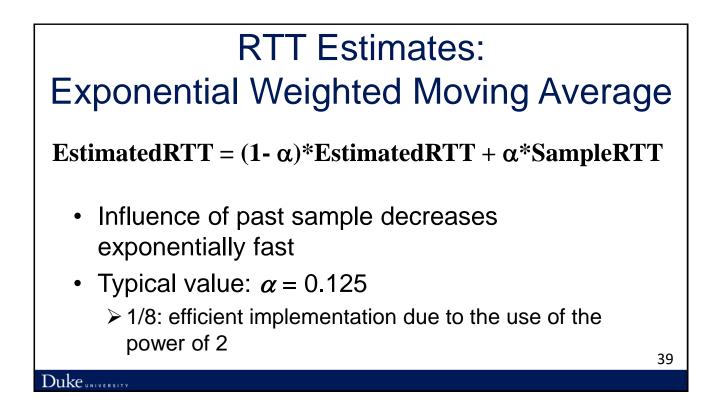
• SampleRTT: measured time from segment transmission until ACK receipt

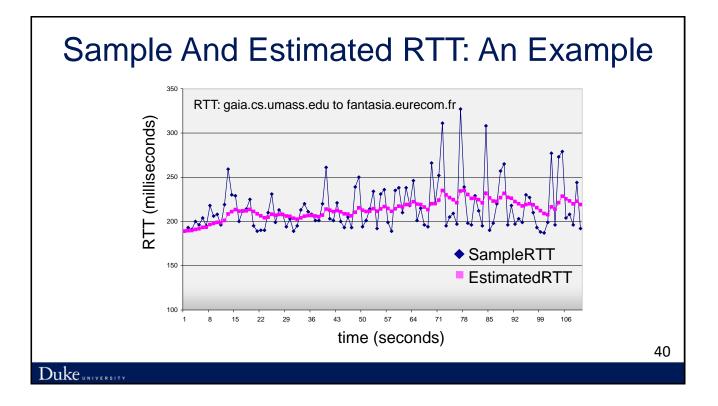
Ignore retransmissions

- SampleRTT varies. We want estimated RTT to be "smoother"
- Solution: average several recent measurements, not just current SampleRTT

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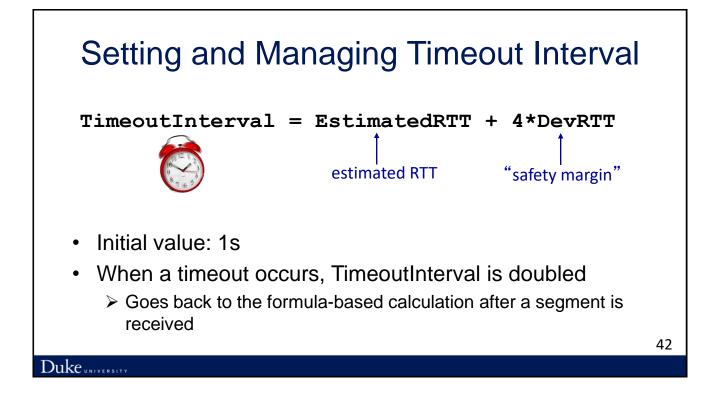




"Safety Margin": Based on the Variability in RTT

 $DevRTT = (1-\beta)*DevRTT + \beta*|SampleRTT-EstimatedRTT|$

- · Also exponentially weighed
- Typical value: β = 0.25
 > Also a power of 2: 1/4

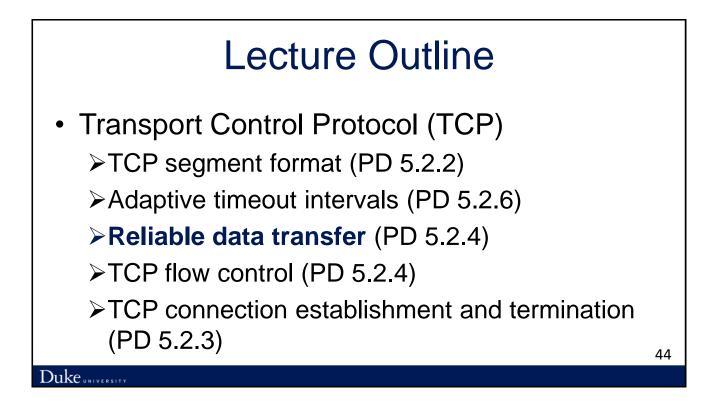


Adaptive Timeout Intervals: Key Points to Remember

Timeout interval is not fixed

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- It is set to a <u>sender-estimated</u> estimated RTT + a safety margin
- \succ Both are dynamic metrics, recalculated with each recorded RTT
- Estimated RTT is calculated based on exponential averaging of sample RTT values
- Safety margin is calculated based on the variability in the RTT



Prevenue Prevenue Prevenue

3 Types of TCP Sender Events

- Data received from the application
- Timeout
- ACK received

TCP Sender Events: Data Received from an Application

- Create segment with seq #
 - Seq # is byte-stream number of first data byte in segment
- · Pass the segment to IP
- Start timer if not already running
 - Think of timer as for oldest unacked segment
 - Expiration interval: TimeOutInterval

TCP Sender Events: Timeout and ACK Received

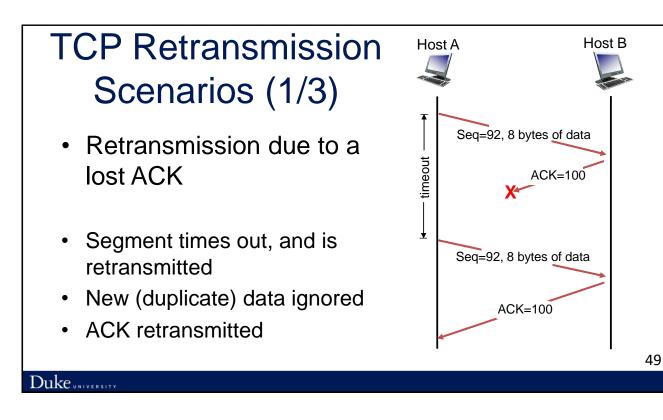
• Timeout:

- Retransmit segment that caused timeout
- ➤ Restart timer

ACK Received:

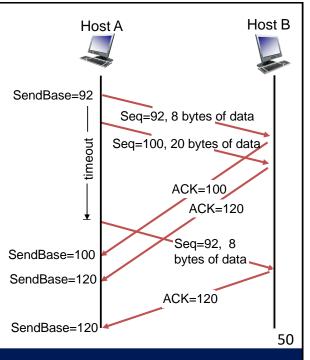
- > If ack acknowledges previously unacked segments:
 - · Update what is known to be ACKed
 - · Start timer if there are still unacked segments

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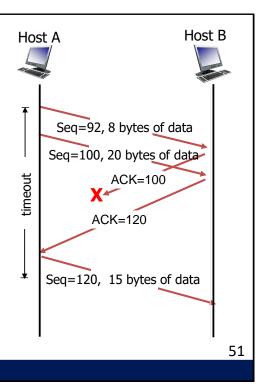
TCP Retransmission Scenarios (2/3)

- Cumulative ACKs
- Both ACKs do not make it back in time
- Segment 92 is retransmitted
- Both segments are acknowledged
 - If ACK arrives before the new timeout value, segment 100 is not retransmitted



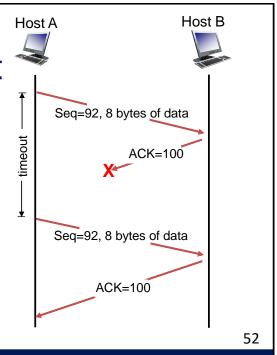
TCP Retransmission Scenarios (3/3)

- Cumulative ACKs
- ACK 100 is lost
- ACK 120 arrives before timeout
- Host A knows that everything was received
 - > No data is retransmitted



Doubling Timeout Interval After a Timeout

- Set to twice the previous value
 - Rather than deriving from RTT measures
- Double for each successive timeout
 - Similar to exponential backoff
- A form of congestion control
 - Assume that losses are due to network being overloaded



TCP ACK G	eneration
-----------	-----------

Event at receiver	TCP receiver action
Arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed.	Delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK.
Arrival of in-order segment with expected seq #. One other segment has ACK pending.	Immediately send single cumulative ACK, ACKing both in-order segments.
Arrival of out-of-order segment higher-than-expect seq. #s. Gap detected.	Immediately send <i>duplicate ACK</i> , indicating seq. # of next expected byte.
Arrival of segment that partially or completely fills gap.	Immediate send ACK, provided that segment starts at lower end of gap.
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TCP Fast Retransmit (1/2)

- Time-out period often relatively long
 Long delay before resending lost packet
- Detect lost segments via duplicate ACKs
 - Sender often sends many segments back-to-back
 - If segment is lost, there will likely be many duplicate ACKs

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TCP Fast Retransmit (2/2)

 If sender receives 3 ACKs for same data ("triple duplicate ACKs") resend unacked segment with smallest seq #

Likely that unacked segment is lost

Do not wait for timeout

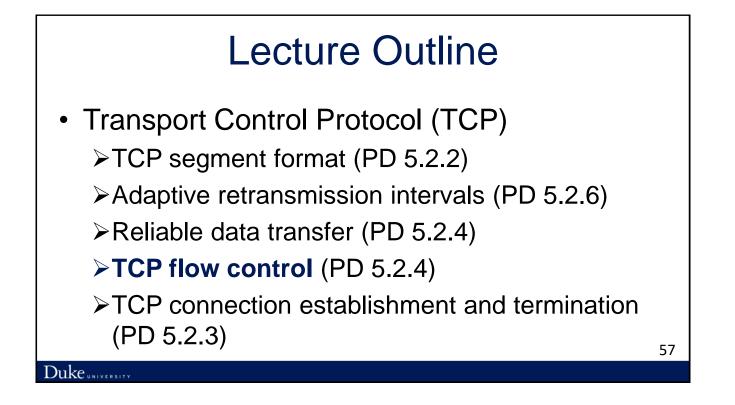
TCP Reliable Data Transfer: Key Points to Remember

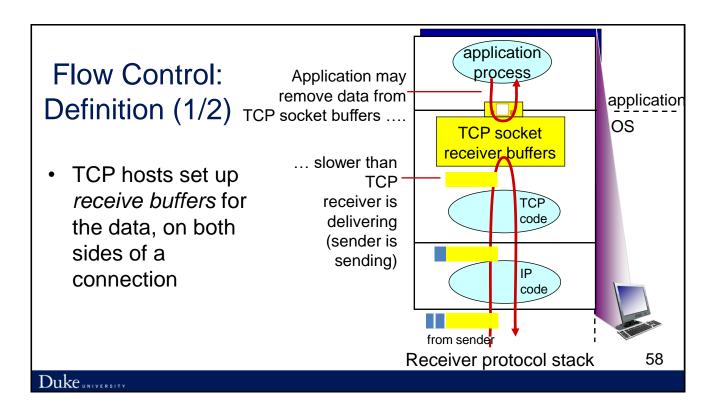
- TCP uses cumulative acknowledgements and retransmissions to ensure reliable in-order data transfer
 Similar to what we have seen for link-layer reliability before
- Uses one retransmission timer per connection
- Doubles timeout interval after a timeout
 Form of congestion control
- Uses *duplicate acknowledgements* to retransmit segments before timeouts

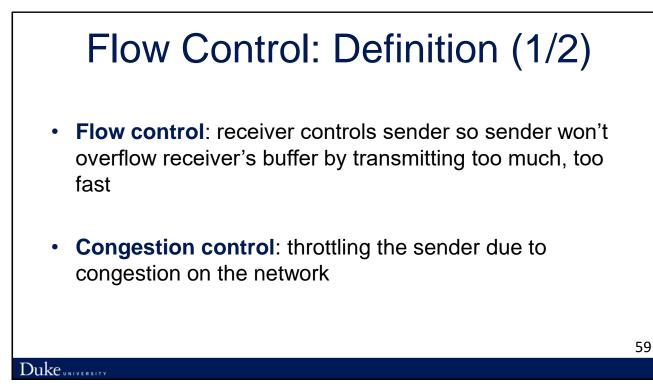
56

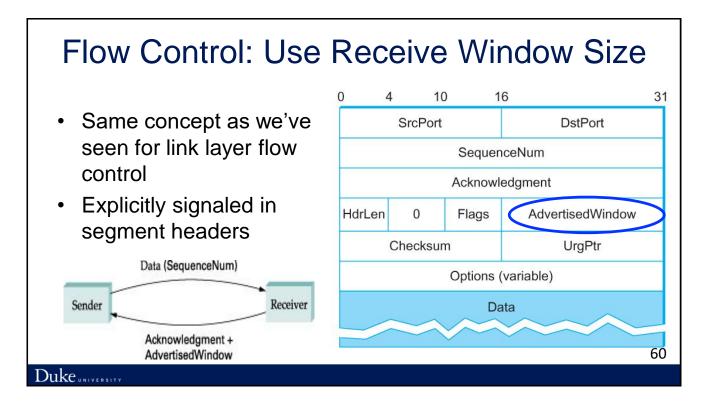
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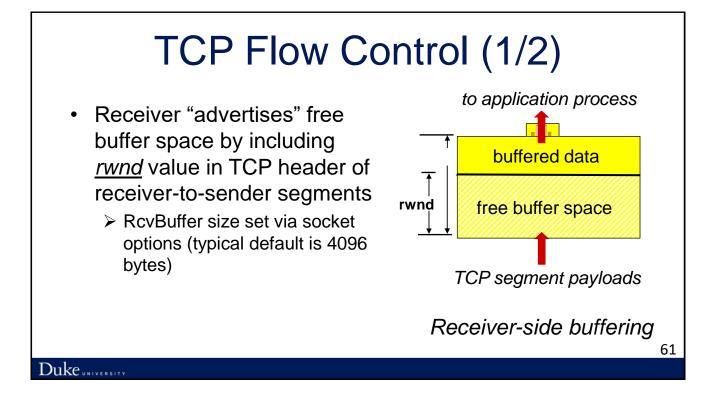
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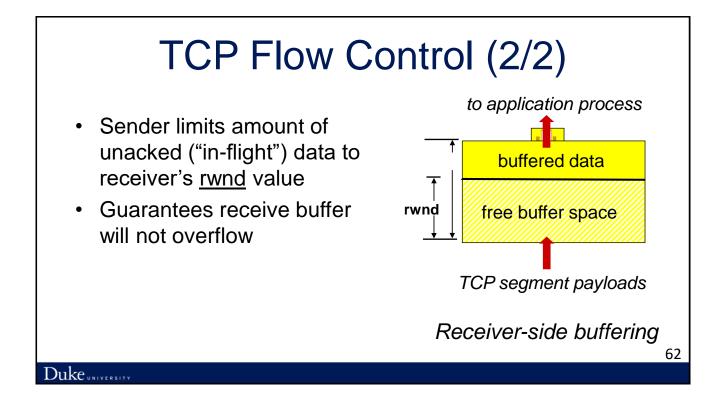






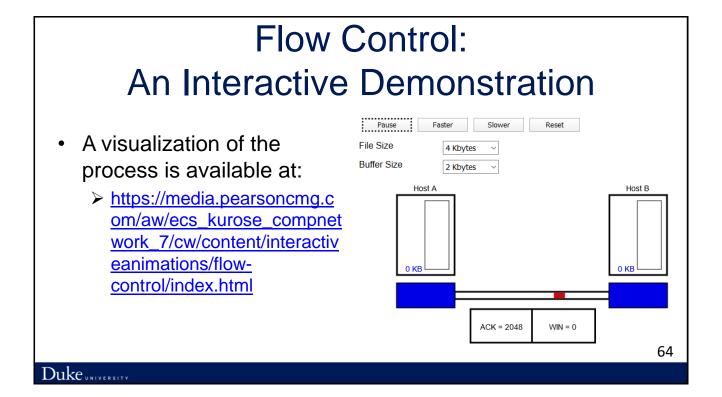






Window Probes

- What if a receiver advertises a window size of zero?
 - Problem: Receiver can't send more ACKs as sender stops sending more data
- Design choices
 - Receivers send duplicate ACKs when window opens
 - Sender sends periodic 1 byte probes



No Flow Control in UDP: What Happens?

- Segments may be lost at the receiver due to buffer overflow
- Typical implementation: UDP appends segments in a finite-sized buffer at the receiving process
- If the process does not read the segments fast enough, the buffer will overflow and the segments will be dropped

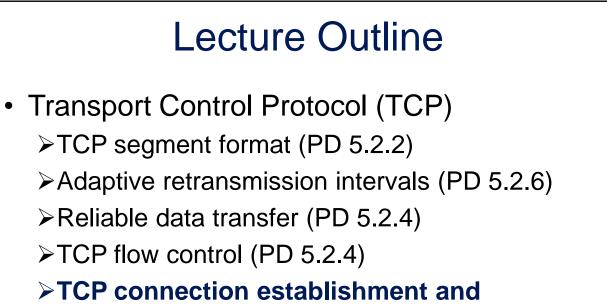
TCP Flow Control: Key Points to Remember

 Receiver controls sender by <u>explicitly stating</u> how much space is available in the receive buffer

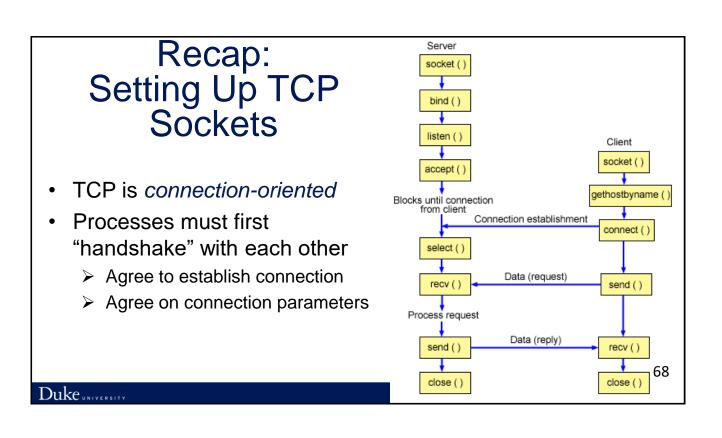
Information included in segment headers

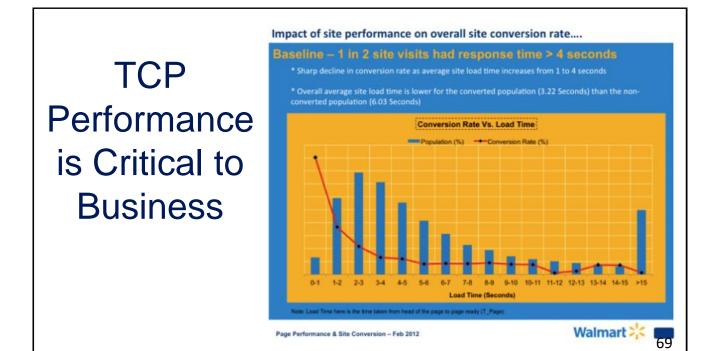
- Highly dynamic
- Approach guarantees that receive buffer will not overflow

66

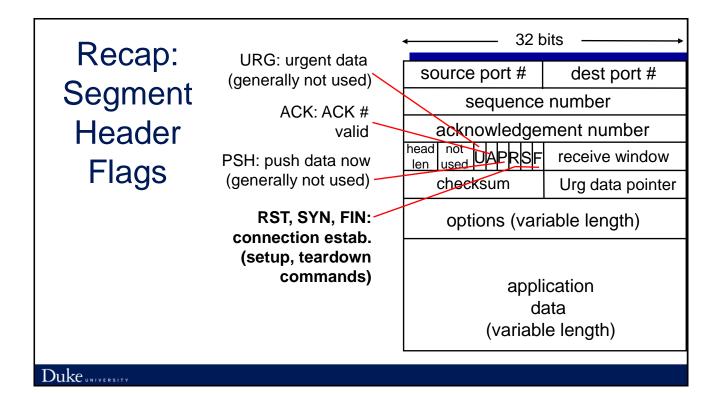


termination (PD 5.2.3)





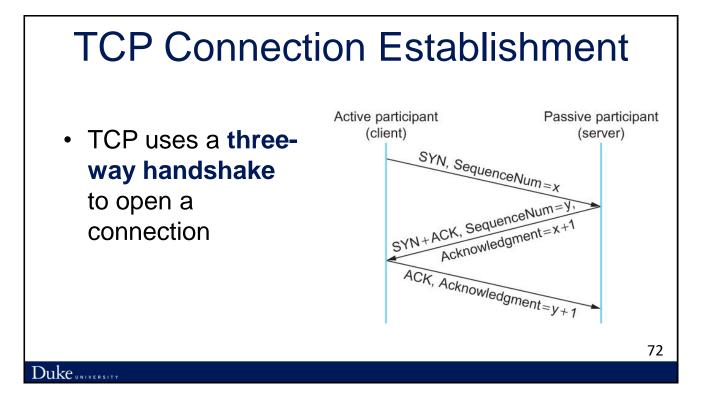
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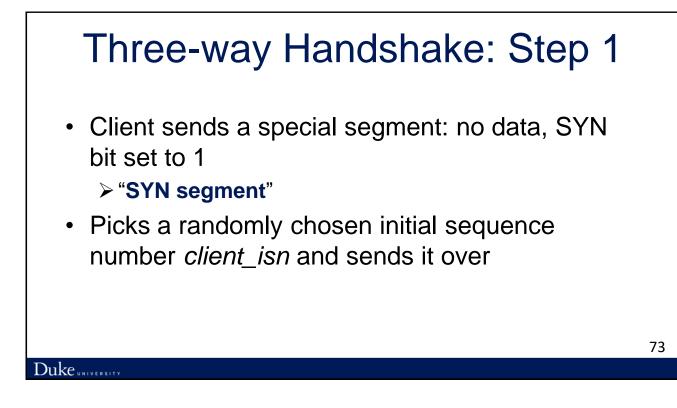


Recap: TCP Flag Bits: Connection Management

RST: Reset the connection

- Receiver of a RST terminates the connection and indicates higher layer application about the reset
- SYN: Synchronize sequence numbers
 Sent in the first packet when initiating a connection
- FIN: Sender is finished with sending
 - Used for closing a connection





Three-way Handshake: Step 2

- SYN segment arrives to the server
- Server allocates TCP buffers and variables
- Sends a "connection granted" segment
 - No application-layer data
 - SYN bit set to 1
 - Acknowledgement is set to client_isn+1
 - Chooses its own initial sequence number server_isn, puts it in the sequence number field
 - SYNACK" segment

Three-way Handshake: Step 3

- Client receives SYNACK segment
- Client allocates buffers and variables
- Client sends to server another segment:
 - Acknowledgement: server_isn+1
 - SYN bit is set to 0
 - May carry payload
- SYN bit set to zero in all subsequent packets

Dress Up as a TCP Packet

- A 3-way handshake example
- More protocol jokes: <u>https://twitter.com/PPathole/st</u> <u>atus/1187371220238508035</u>



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 Kah Zuhl List @kazoolist · Oct 25

 Replying to @PPathole

 I dressed up as UDP packet last year.

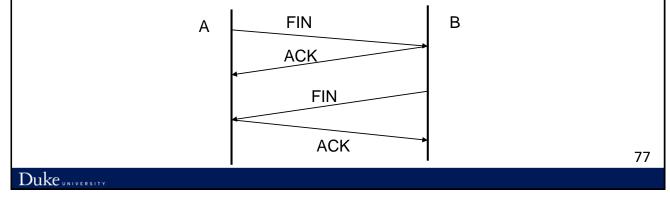
 No one acknowledged me.

 Image: Comparison of the second second



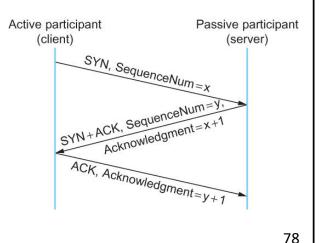
TCP Connection Termination

- Need to de-allocate buffers and variables
- Each end of the data flow must be shut down independently
- If one end is done it sends a FIN segment. The other end sends ACK
- Four messages to completely shut down a connection



TCP Connection Management: Key Points to Remember

- Need to allocate resources on both ends of the communication
- Connection established via a three-way handshake
- Connections need to be torn down, to deallocate the resources



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