

## Reliable Byte-Stream (TCP)

### Outline

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

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## End-to-End Protocols

- Underlying best-effort network
  - drop messages
  - re-orders messages
  - delivers duplicate copies of a given message
  - limits messages to some finite size
  - delivers messages after an arbitrarily long delay
- Common end-to-end services
  - 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

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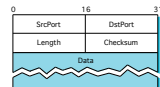
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## Simple Demultiplexor (UDP)

- Unreliable and unordered datagram service
- Adds multiplexing
- No flow control
- Endpoints identified by ports
  - servers have *well-known* ports
  - see `/etc/services` on Unix
- Header format
- Optional checksum
  - pseudo header + UDP header + data



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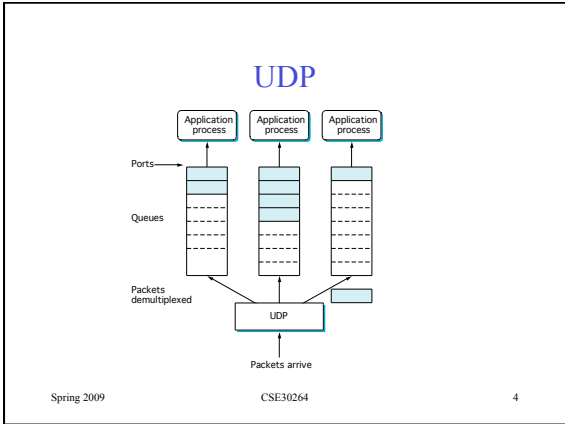
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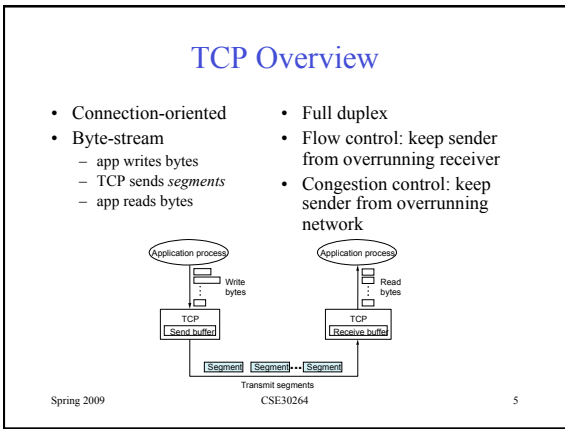
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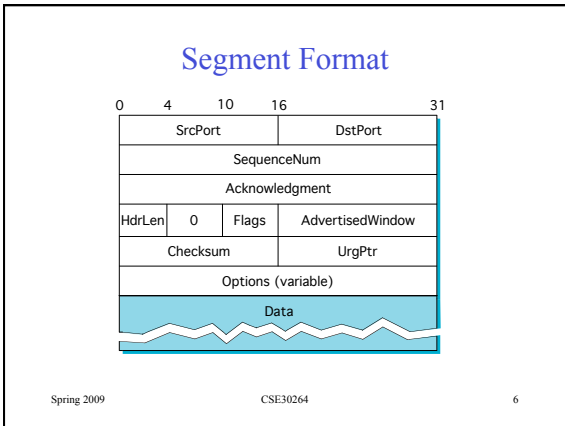
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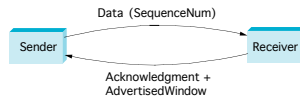
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## Segment Format (cont)

- Each connection identified with 4-tuple:
  - (SrcPort, SrcIPAddr, DstPort, DstIPAddr)
- Sliding window + flow control
  - ACK, SequenceNum, AdvertisedWindow



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

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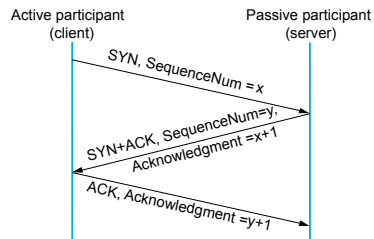
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## Connection Establishment



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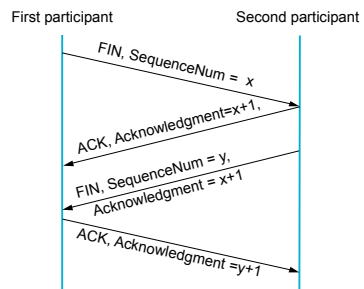
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## Connection Termination



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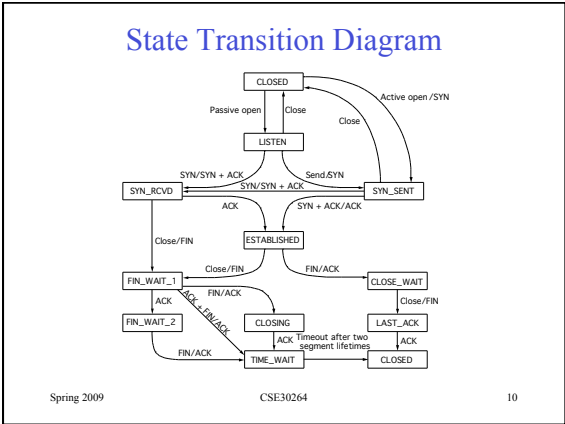
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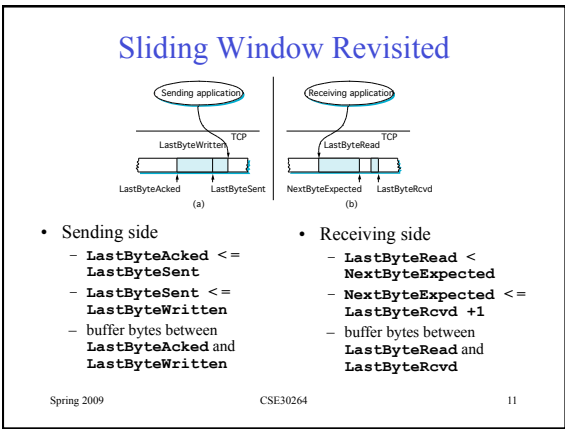
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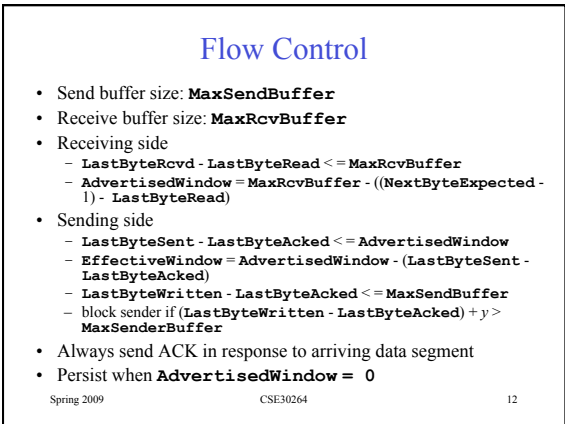
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## Protection Against Wrap Around

- 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

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## Silly Window Syndrome

- How aggressively does sender exploit open window?

The diagram shows a Sender and a Receiver connected by a line. Above the line, several blue rectangles represent data segments. A curved arrow points from the Sender to the Receiver, indicating data transmission. Below the line, three white rectangles represent segments that have been received but are not yet acknowledged, illustrating the receiver's buffer being full and the sender's window being stuck.

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

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## 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

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when application produces data to send
  if both the available data and the window >= MSS
    send a full segment
  else
    if there is unACKed data in flight
      buffer the new data until an ACK arrives
    else
      send all the new data now
            
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## Adaptive Retransmission

- Round-Trip Time Estimation:
  - wait at least one RTT before retransmitting
  - importance of accurate RTT estimators:
    - Low RTT -> unneeded retransmissions
    - High RTT -> poor throughput
  - RTT estimator must adapt to change in RTT
    - But not too fast, or too slow!
- problem: If the instantaneously calculated RTT is 10, 20, 5, 12, 3, 5, 6; what RTT should we use for calculations?
- $\text{EstimatedRTT} = \alpha * \text{EstimatedRTT} + (1 - \alpha) \text{SampleRTT}$
- recommended value for  $\alpha$ : 0.8 - 0.9
- retransmit timer set to  $\beta$  RTT, where  $\beta = 2$

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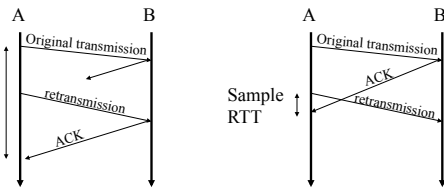
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## Retransmission Ambiguity



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## Karn/Partridge Algorithm

- Accounts for retransmission ambiguity
- If a segment has been retransmitted:
  - don't count RTT sample on ACKs for this segment
  - reuse RTT estimate only after one successful transmission
  - double timeout after each retransmission

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## Jacobson/Karels Algorithm

- Key observation:
  - using  $\beta$  RTT for timeout doesn't work
  - at high loads round trip variance is high
- Solution:
  - if  $D$  denotes mean variation
  - timeout = RTT + 4D

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## Jacobson/Karels Algorithm

- New Calculations for average RTT
- Diff = SampleRTT - EstimatedRTT
- EstimatedRTT = EstimatedRTT + (d \* Diff)
- Dev = Dev + d \* (|Diff| - Dev)
  - where d is a factor between 0 and 1
- Consider variance when setting timeout value
- TimeOut = m \* EstimatedRTT + f \* Dev
  - where m = 1 and f = 4

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## Record Boundaries

- Byte-stream protocol: write 8+2+20 bytes and read 5+5+5+5+5+5 (loop).
- TCP offers two features to insert record boundaries:
  - URG flag
  - push operation

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## TCP Extensions

- Implemented as header options
- Better way to measure RTT (use actual system clock for sending time and add timestamp to segment).
- 64-bit sequence numbers: 32-bit sequence number in low-order 32 bits, timestamp in high-order 32 bits.
- Shift (scale) advertised window.

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