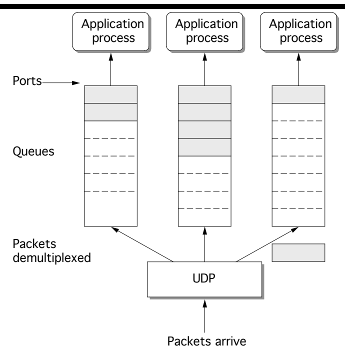
End-to-End Protocols

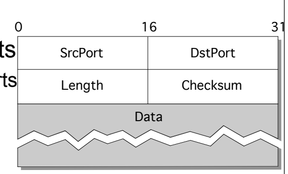
- Underlying best-effort network
  - drop messages
  - re-orders messages
  - delivers duplicate copies of a given message
  - limits messages to some finite size
  - delivers messages after an arbitrarily long delay
- Common end-to-end services
  - guarantee message delivery
  - deliver messages in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

UDP message queue



Simple Demultiplexor (UDP)

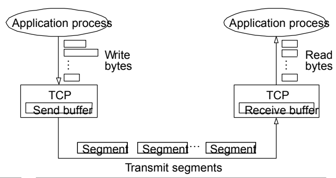
- Unreliable and unordered datagram service
- Adds multiplexing
- No flow control
- Endpoints identified by ports
  - servers have well-known ports
  - see /etc/services on Unix
- Header format



- Optional checksum
  - psuedo header + UDP header + data

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



Data Link Versus Transport

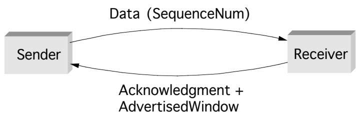
- Potentially connects many different hosts
  - need explicit connection establishment and termination
- Potentially different RTT
  - need adaptive timeout mechanism
- Potentially long delay in network
  - need to be prepared for arrival of very old packets
- Potentially different capacity at destination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

Segment Format

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

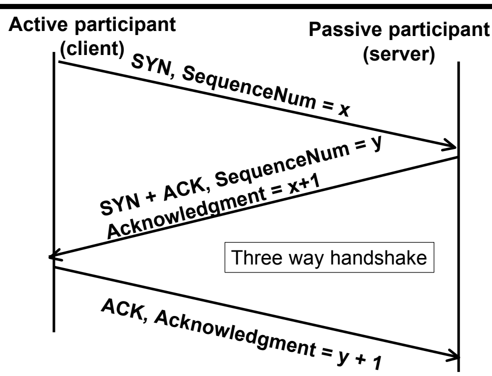
Segment Format (cont)

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



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

Connection Establishment and Termination



Sequence Number Selection

- Initial sequence number (ISN) selection
  - Why not simply chose 0?
  - Must avoid overlap with earlier incarnation
- Requirements for ISN selection
  - Must operate correctly
    - Without synchronized clocks
    - Despite node failures

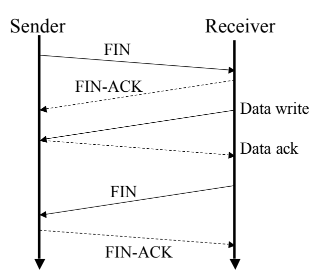
ISN and Quiet Time

- Use local clock to select ISN
  - Clock wraparound must be greater than max segment lifetime (MSL)
- Upon startup, cannot assign sequence numbers for MSL seconds
- Can still have sequence number overlap
  - If sequence number space not large enough for high-bandwidth connections

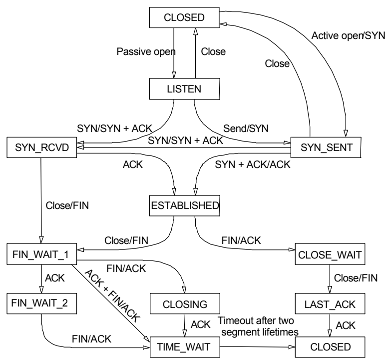
Connection Tear-down

- Normal termination
  - Allow unilateral close
  - Avoid sequence number overlap
- TCP must continue to receive data even after closing
  - Cannot close connection immediately: what if a new connection restarts and uses same sequence number and receives retransmitted FIN from the current session?

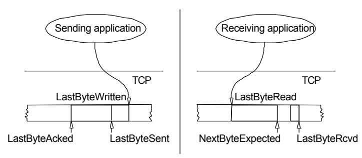
Tear-down Packet Exchange



State Transition Diagram



Sliding Window Revisited



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

Flow Control

- Fast sender can overrun receiver:
  - Packet loss, unnecessary retransmissions
- Possible solutions:
  - Sender transmits at pre-negotiated rate
  - Sender limited to a window's worth of unacknowledged data
- Flow control different from congestion control

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

Round-trip Time Estimation

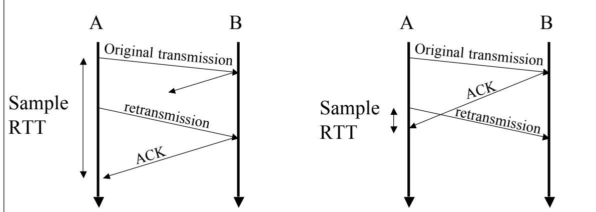
- Wait at least one RTT before retransmitting
- Importance of accurate RTT estimators:
  - Low RTT -> unneeded retransmissions
  - High RTT -> poor throughput
- RTT estimator must adapt to change in RTT
  - But not too fast, or too slow!

Initial Round-trip Estimator

Round trip times exponentially averaged:

- New RTT = α (old RTT) + (1 - α) (new sample)
- Recommended value for α: 0.8 - 0.9
- Retransmit timer set to β RTT, where β = 2
- Every time timer expires, RTO exponentially backed-off

Retransmission Ambiguity

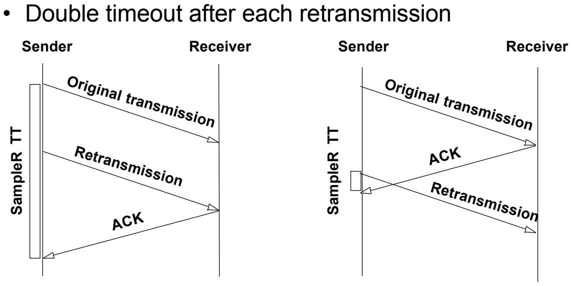


Karn's Retransmission Timeout Estimator

- Accounts for retransmission ambiguity
- If a segment has been retransmitted:
  - Don't count RTT sample on ACKs for this segment
  - Keep backed off time-out for next packet
  - Reuse RTT estimate only after one successful transmission

Karn/Partridge Algorithm

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



Jacobson's Retransmission Timeout Estimator

- Key observation:
  - Using β RTT for timeout doesn't work
  - At high loads round trip variance is high
- Solution:
  - If D denotes mean variation
  - Timeout = RTT + 4D

Jacobson/ Karels Algorithm

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