Transport Protocols

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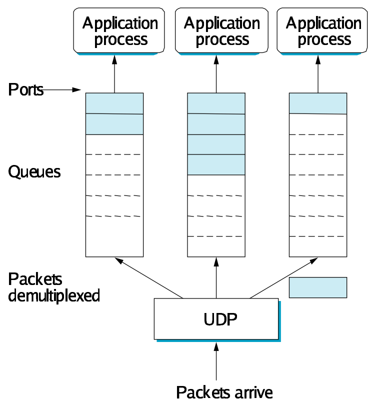
End-to-End Protocols

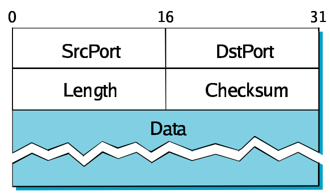
- Convert host-to-host packet delivery service into a process-to-process communication channel
  - Demultiplexing: Multiple applications can share the network
- End points identified by ports
  - Ports are not interpreted globally
  - servers have well defined ports (look at /etc/services)

User Datagram Protocol (UDP)

Demultiplexing



UDP Header



Computes checksum over UDP header, message body and pseudo-header

Application Layer Expectations

- Guaranteed message delivery
- Ordered delivery
- No duplication
- Support arbitrarily large messages
- Synchronization between the sender and receiver
- Support flow control
- Support demultiplexing

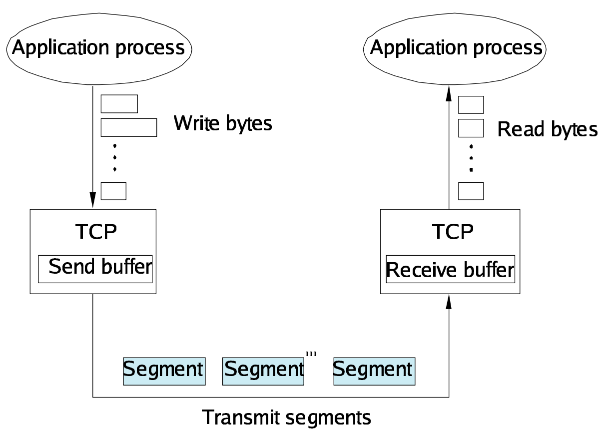
Limitations of Networks

- Packet Losses
- Re-ordering
- Duplicate copies
- Limit on maximum message size
- Long delays

Transmission Control Protocol (TCP)

- Connection oriented
  - Maintains state to provide reliable service
- Byte-stream oriented
  - Handles byte streams instead of messages
- Full Duplex
  - Supports flow of data in each direction
- Flow-control
  - Prevents sender from overrunning the receiver
- Congestion-control
  - Prevents sender from overloading the network

TCP Cont...



Sliding Window: Data Link vs Transport

P2P: Dedicated Link -- Physical Link connects the same two computers
TCP: Connects two processes on any two machines in the Internet
- Needs explicit connection establishment phase to exchange state

P2P: Fixed round trip transmission time (RTT)
TCP: Potentially different and widely varying RTTs
- Timeout mechanism has to be adaptive

P2P: No Reordering
TCP: Scope for reordering due to arbitrary long delays
- Need to be robust against old packets showing up suddenly

Sliding Window: Data Link vs Transport

P2P: End points can be engineered to support the link

TCP: Any kind of computer can be connected to the Internet

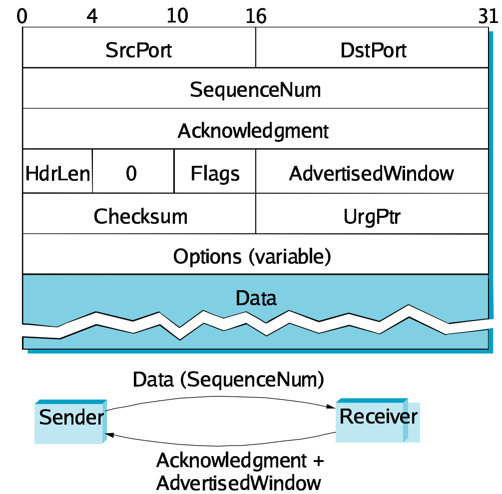
- Need mechanism for each side to learn other side's resources (e.g. buffer space) -- Flow control

P2P: Not possible to unknowingly congest the link

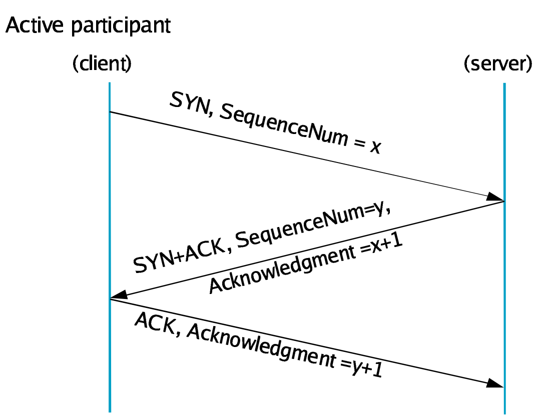
TCP: No idea what links will be traversed, network capacity can dynamically vary due to competing traffic

- Need mechanism to alter sending rate in response to network congestion – Congestion control

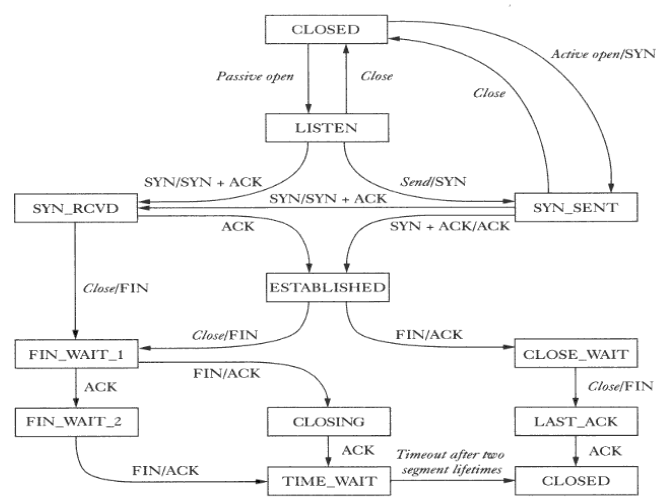
TCP Header Format



Connection Establishment



State Transition Diagram

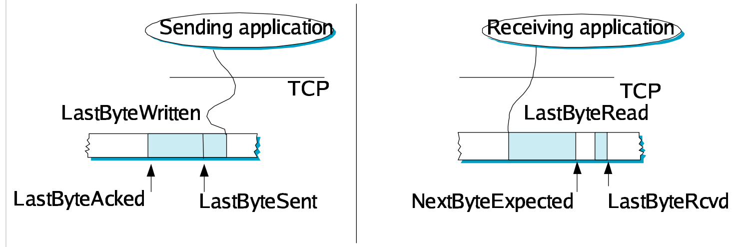


Protection Against Wraparound

- Wraparound occurs because sequence number field is finite
  - 32 bit sequence number space
- Maximum Segment Lifetime (MSL) is 120 sec

Bandwidth	Time until Wraparound
T1 (1.5Mbps)	6.6 hrs
Ethernet (10Mbps)	57 minutes
T3 (45 Mbps)	13 minutes
FDDI (100Mbps)	6 minutes
STS-3 (155Mbps)	4 minutes
STS-12 (622Mbps)	55 seconds
STS-24 (1.2Gbps)	28 seconds

Sliding Window Recap



Sending Side:
- LastByteAcked <= LastByteSent
- LastByteSent <= LastByteWritten
- Buffer bytes between LastByteAcked and LastByteWritten

Receiving Side:
- LastByteRead <= NextByteExpected
- NextByteExpected <= LastByteRcvd+1
- Buffer bytes between LastByteRead and LastByteRcvd

Flow Control

- Buffers are of finite size
  - MaxSendBuffer and MaxRcvBuffer
- Receiving side:
  - LastByteRcvd – LastByteRead <= MaxRcvBuffer
  - AdvertisedWindow = MaxRcvBuffer – ((NextByteExpected – 1) – LastByteRead)
- Sending side:
  - LastByteSent – LastByteAcked <= AdvertisedWindow
  - EffectiveWindow = AdvertisedWindow – (LastByteSent – LastByteAcked)
  - LastByteWritten – LastByteAcked <= MaxSendBuffer
  - Persist when AdvertisedWindow is zero

Congestion Control

- At steady state use Self-clocking
  - Acks pace transmission of packets
- Challenges:
  - How to determine available capacity?
  - How to adjust sending rate to varying capacity?

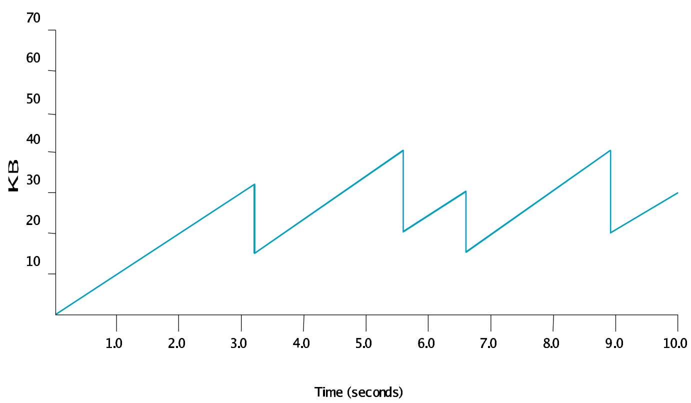
Congestion Avoidance: Additive Increase/Multiplicative Decrease

- Introduce a new variable: CongestionWindow
  - Limits the amount of data in transit
  - MaxWindow = Minimum of (CongestionWindow,AdvertisedWindow)
  - EffectiveWindow = Maxwindow – (LastByteSent – LastByteAcked)
- Adjust CongestionWindow to changes in capacity
  - Decrease CongestionWindow when congestion goes up
  - Increase CongestionWindow when congestion goes down

AIMD Cont...

- Problem: How do we detect congestion?
- Answer: Timeouts
  - TCP interprets timeout as a result of congestion
- Multiplicative decrease: Cut CongestionWindow by half on each timeout
- Additive Increase: Increase CongestionWindow by Maximum Segment Size (MSS) per RTT
  - In practice, increment a little on each ack,
  - CongestionWindow += Increment
  - Increment = MSS * (MSS/CongestionWindow)

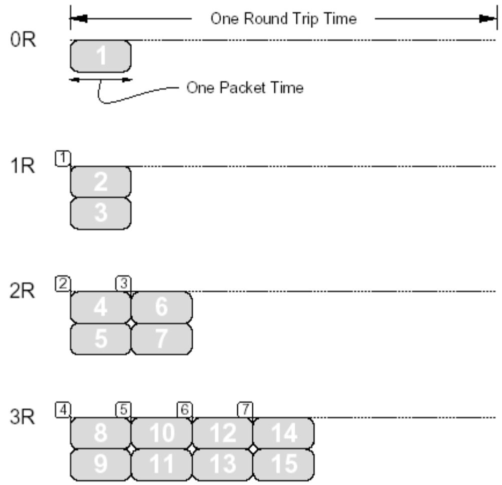
Saw Tooth Pattern



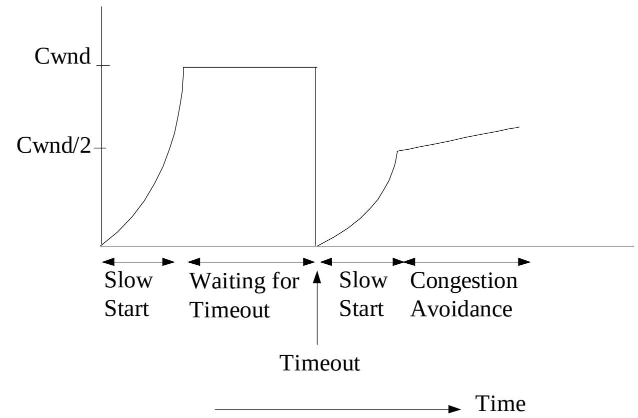
Slow Start

- AIMD approach is used at steady state
- But how to get to steady state?
- Increase Congestion Window exponentially
  - Begin with CongestionWindow = 1
  - Double CongestionWindow every RTT
- “Slow” compared to sending entire advertised window all at once
- Used during beginning of connection
- Used when connection goes dead due to timeout

Figure 2: The Chronology of a Slow-start



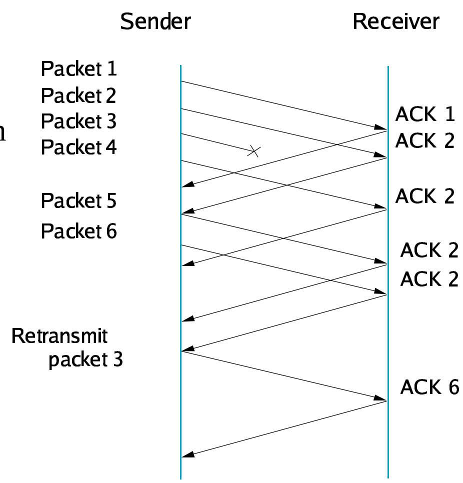
Congestion Window vs Time



Fast Retransmit/Fast Recovery

Fast Retransmit:
Use duplicate acks to trigger retransmission

Fast Recovery:
Peform congestion avoidance instead of slow start



RTT Estimation: Original Algorithm

- Measure SampleRTT for sequence/ack combo
- EstimatedRTT = a*EstimatedRTT + (1-a)*SampleRTT
  - *a* is between 0.8-0.9
  - small *a* heavily influenced by temporary fluctuations
  - large *a* not quick to adapt to real changes
- Timeout = 2 * EstimatedRTT

Jacobson/Karels Algorithm

- Incorrect estimation of RTT worsens congestion
- Algorithm takes into account variance of RTTs
  - If variance is small, EstimatedRTT can be trusted
  - If variance is large, timeout should not depend heavily on EstimatedRTT

Jacobson/Karels Algorithm Cont..

- Difference = SampleRTT - EstimatedRTT
- EstimatedRTT = EstimatedRTT + (d * Difference)
- Deviation = Deviation + d (|Difference| - Deviation)), where d ~ 0.125
- Timeout = u * EstimatedRTT + q * Deviation, where u = 1 and q = 4
- Exponential RTO backoff

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Summary

- Transport protocols essentially demultiplexing functionality
- Examples: UDP, TCP, RTP
- TCP is a reliable connection-oriented byte-stream protocol
  - Sliding window based
  - Provides flow and congestion control