Good Ideas So Far…

- Flow control
  - Stop & wait
  - Parallel stop & wait
  - Sliding window (e.g., advertised windows)
- Loss recovery
  - Timeouts
  - Acknowledgement-driven recovery (selective repeat or cumulative acknowledgement)
- Congestion control
  - AIMD \(\rightarrow\) fairness and efficiency
- How does TCP actually implement these?

Outline

- The devilish details of TCP
- TCP connection setup and data transfer
- TCP reliability
  - Be nice to your data
- TCP congestion avoidance
  - Be nice to your routers

Sequence Number Space

- Each byte in byte stream is numbered.
  - 32 bit value
  - Wraps around
  - Initial values selected at start up time
- TCP breaks up the byte stream into packets.
  - Packet size is limited to the Maximum Segment Size
- Each packet has a sequence number.
  - Indicates where it fits in the byte stream

```
13450  14950  16050  17550
packet 8  packet 9  packet 10
```
Establishing Connection: Three-Way handshake

- Each side notifies other of starting sequence number it will use for sending
  - Why not simply chose 0?
    - Must avoid overlap with earlier incarnation
  - Security issues
- Each side acknowledges other’s sequence number
  - SYN-ACK: Acknowledge sequence number + 1
- Can combine second SYN with first ACK

TCP State Diagram: Connection Setup

TCP Connection Setup Example

Tearing Down Connection
TCP Connection Teardown Example

- Session
  - Echo client on 128.2.222.198, server on 128.2.210.194
- Client FIN
  - SeqC: 1489294581
  - Server ACK + FIN
    - Ack: 1489294582 (= SeqC+1)
    - SeqS: 1909787689
  - Client ACK
    - Ack: 1909787690 (= SeqS+1)

State Diagram: Connection Tear-down

- Outline
  - TCP connection setup/data transfer
  - TCP reliability

Reliability Challenges

- Congestion related losses
- Variable packet delays
  - What should the timeout be?
- Reordering of packets
  - How to tell the difference between a delayed packet and a lost one?
TCP = Go-Back-N Variant

- Sliding window with cumulative acks
  - Receiver can only return a single “ack” sequence number to the sender.
  - Acknowledges all bytes with a lower sequence number
  - Starting point for retransmission
  - Duplicate acks sent when out-of-order packet received
- But: sender only retransmits a single packet.
  - Reason???
    - Only one that it knows is lost
    - Network is congested → shouldn’t overload it
- Error control is based on byte sequences, not packets.
  - Retransmitted packet can be different from the original lost packet
    - Why?

How to set timeout?

- Wait until sender knows it should have seen an ACK
- How long should this be?

Round-trip Time Estimation

- Wait at least one RTT before retransmitting
- Importance of accurate RTT estimators:
  - Low RTT estimate
    - unneeded retransmissions
  - High RTT estimate
    - poor throughput
- RTT estimator must adapt to change in RTT
  - But not too fast, or too slow!
- Spurious timeouts
  - “Conservation of packets” principle – never more than a window worth of packets in flight

Original TCP Round-trip Estimator

- Round trip times exponentially averaged:
  - New RTT = \( \alpha \) (old RTT) + \((1 - \alpha)\) (new sample)
  - Recommended value for \( \alpha \): 0.8 - 0.9
    - 0.875 for most TCP's
- Retransmit timer set to \((b \times \text{RTT})\), where \(b = 2\)
  - Every time timer expires, RTO exponentially backed-off
  - Not good at preventing spurious timeouts
  - Why?
RTT Sample Ambiguity

- Karn’s RTT Estimator
  - 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

Jacobson’s Retransmission Timeout

- Key observation:
  - At high loads round trip variance is high
- Solution:
  - Base RTO on RTT and standard deviation
    - \( \text{RTO} = \text{RTT} + 4 \times \text{rttvar} \)
    - \( \text{new}\_\text{rttvar} = \beta \times \text{dev} + (1 - \beta) \times \text{old}\_\text{rttvar} \)
      - Dev = linear deviation
      - Inappropriately named – actually smoothed linear deviation

Timestamp Extension

- Used to improve timeout mechanism by more accurate measurement of RTT
- When sending a packet, insert current time into option
  - 4 bytes for time, 4 bytes for echo a received timestamp
- Receiver echoes timestamp in ACK
  - Actually will echo whatever is in timestamp
- Removes retransmission ambiguity
  - Can get RTT sample on any packet

Timer Granularity

- Many TCP implementations set RTO in multiples of 200,500,1000ms
- Why?
  - Avoid spurious timeouts – RTTs can vary quickly due to cross traffic
  - Make timer interrupts efficient
- What happens for the first couple of packets?
  - Pick a very conservative value (seconds)
Fast Retransmit

- What are duplicate acks (dupacks)?
  - Repeated acks for the same sequence
- When can duplicate acks occur?
  - Loss
  - Packet re-ordering
  - Window update – advertisement of new flow control window
- Assume re-ordering is infrequent and not of large magnitude
  - Use receipt of 3 or more duplicate acks as indication of loss
  - Don’t wait for timeout to retransmit packet

TCP (Reno variant)

SACK

- Basic problem is that cumulative acks provide little information
- Selective acknowledgement (SACK) essentially adds a bitmask of packets received
  - Implemented as a TCP option
  - Encoded as a set of received byte ranges (max of 4 ranges/often max of 3)
- When to retransmit?
  - Still need to deal with reordering → wait for out of order by 3pkts
Lecture 18: TCP Details

SACK

Performance Issues

- Timeout >> fast rexmit
- Need 3 dupacks/sacks
- Not great for small transfers
  - Don’t have 3 packets outstanding
- What are real loss patterns like?

Outline

- TCP connection setup/data transfer
- TCP reliability
- TCP congestion avoidance

Additive Increase/Decrease

- Both $X_1$ and $X_2$ increase/decrease by the same amount over time
  - Additive increase improves fairness and additive decrease reduces fairness

Efficiency Line

User 1’s Allocation $x_1$

User 2’s Allocation $x_2$

Fairness Line

$T_1$

$T_2$
**Multiplicative Increase/Decrease**

- Both $X_1$ and $X_2$ increase by the same factor over time.
- Extension from origin – constant fairness.

**What is the Right Choice?**

- Constraints limit us to AIMD.
  - Improves or keeps fairness constant at each step.
  - AIMD moves towards optimal point.

**TCP Congestion Control**

- Changes to TCP motivated by ARPANET congestion collapse.
- Basic principles:
  - AIMD
  - Packet conservation
  - Reaching steady state quickly
  - ACK clocking.

**AIMD**

- Distributed, fair and efficient.
- Packet loss is seen as sign of congestion and results in a multiplicative rate decrease:
  - Factor of 2.
  - TCP periodically probes for available bandwidth by increasing its rate.
### Implementation Issue
- Operating system timers are very coarse – how to pace packets out smoothly?
- Implemented using a congestion window that limits how much data can be in the network.
  - TCP also keeps track of how much data is in transit
  - Data can only be sent when the amount of outstanding data is less than the congestion window.
    - The amount of outstanding data is increased on a "send" and decreased on "ack"
    - (last sent – last acked) < congestion window
  - Window limited by both congestion and buffering
    - Sender’s maximum window = Min (advertised window, cwnd)

### Congestion Avoidance
- If loss occurs when cwnd = W
  - Network can handle 0.5W ~ W segments
  - Set cwnd to 0.5W (multiplicative decrease)
- Upon receiving ACK
  - Increase cwnd by (1 packet)/cwnd
    - What is 1 packet? → 1 MSS worth of bytes
    - After cwnd packets have passed by → approximately increase of 1 MSS
  - Implements AIMD

### Congestion Avoidance Sequence Plot

### Congestion Avoidance Behavior

![Congestion Window vs. Time Graph with labels: Packet loss + retransmit, Cut Congestion Window and Rate, Grabbing back Bandwidth]
Important Lessons

• TCP state diagram → setup/teardown

• TCP timeout calculation → how is RTT estimated

• Modern TCP loss recovery
  • Why are timeouts bad?
  • How to avoid them? → e.g. fast retransmit

Networking is much more fun than physics