15-441 Computer Networking

Lecture 18 – More TCP & Congestion Control

Good Ideas So Far…

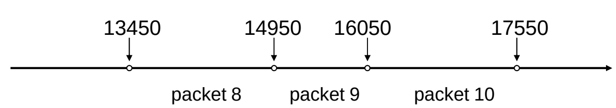
- Flow control
  - Stop & wait
  - Parallel stop & wait
  - Sliding window (e.g., advertised windows)
- Loss recovery
  - Timeouts
  - Acknowledgement-driven recovery (selective repeat or cumulative acknowledgement)
- Congestion control
  - AIMD → fairness and efficiency
- How does TCP actually implement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

Outline

- THE SPOOKY PARTS of TCP
  - If it doesn't scare you now… it will on the Final!
- TCP connection setup/data transfer
  - The Candy-exchange Protocol (TCP)
- TCP reliability
  - How to recover your DEAD packets
- TCP congestion avoidance
  - Avoiding the death-traps of overloaded routers

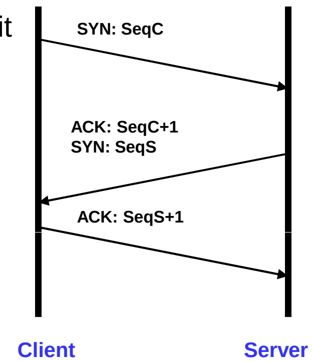
Sequence Number Space

- 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



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 Connection Setup Example

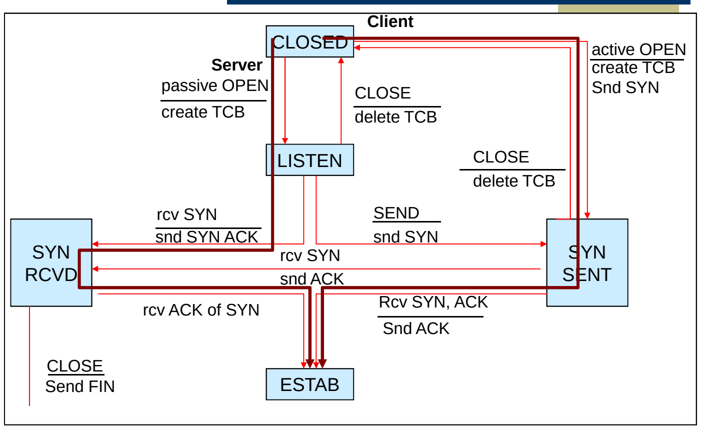
```
09:23:33.042318 IP 128.2.222.198.3123 > 192.216.219.96.80: S
  4019802004:4019802004(0) win 65535 <mss 1260,nop,nop,sackOK>
(DF)

09:23:33.118329 IP 192.216.219.96.80 > 128.2.222.198.3123: S
  3428951569:3428951569(0) ack 4019802005 win 5840 <mss
1460,nop,nop,sackOK> (DF)

09:23:33.118405 IP 128.2.222.198.3123 > 192.216.219.96.80: . ack
  3428951570 win 65535 (DF)
```

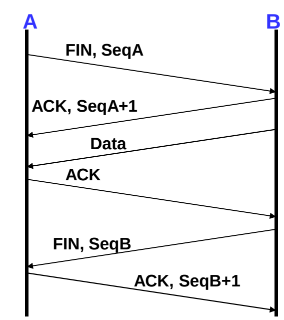
- Client SYN
  - SeqC: Seq. #4019802004, window 65535, max. seg. 1260
- Server SYN-ACK+SYN
  - Receive: #4019802005 (= SeqC+1)
  - SeqS: Seq. #3428951569, window 5840, max. seg. 1460
- Client SYN-ACK
  - Receive: #3428951570 (= SeqS+1)

TCP State Diagram: Connection Setup



Tearing Down Connection

- Either side can initiate tear down
  - Send FIN signal
  - "I'm not going to send any more data"
- Other side can continue sending data
  - Half open connection
  - Must continue to acknowledge
- Acknowledging FIN
  - Acknowledge last sequence number + 1



TCP Connection Teardown Example

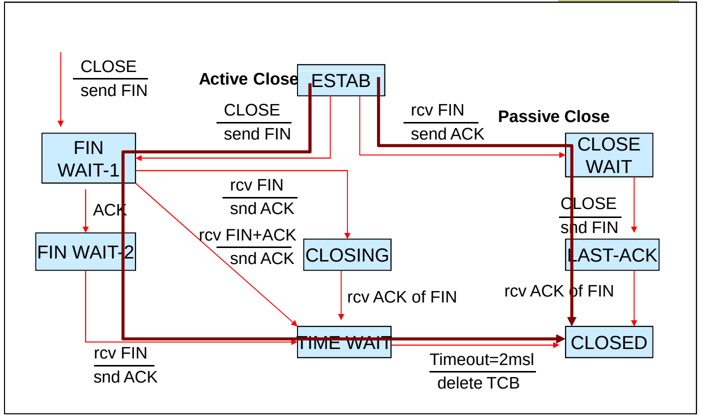
```
09:54:17.585396 IP 128.2.222.198.4474 > 128.2.210.194.6616: F
  1489294581:1489294581(0) ack 1909787689 win 65434 (DF)

09:54:17.585732 IP 128.2.210.194.6616 > 128.2.222.198.4474: F
  1909787689:1909787689(0) ack 1489294582 win 5840 (DF)

09:54:17.585764 IP 128.2.222.198.4474 > 128.2.210.194.6616: . ack
  1909787690 win 65434 (DF)
```

- 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
- TCP congestion avoidance

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

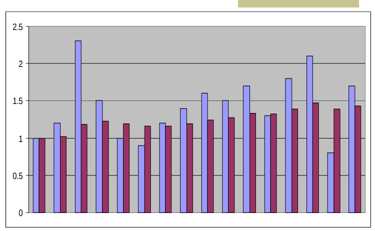
- Sliding window with cumulative acks
  - Receiver can only return a single "ack" sequence number to the sender.
  - Acknowledges all bytes with a lower sequence number
  - Starting point for retransmission
  - Duplicate acks sent when out-of-order packet received
- But: sender only retransmits a single 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?

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

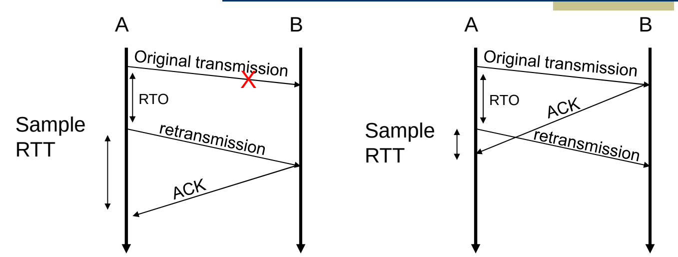
Original TCP Round-trip Estimator

- Round trip times exponentially averaged:
  - New RTT = α (old RTT) + (1 - α) (new sample)
  - Recommended value for α: 0.8 - 0.9
    - 0.875 for most TCP's



- Retransmit timer set to (b * 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
    - RTO = RTT + 4 * rttvar
  - new_rttvar = β * dev + (1- β) old_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

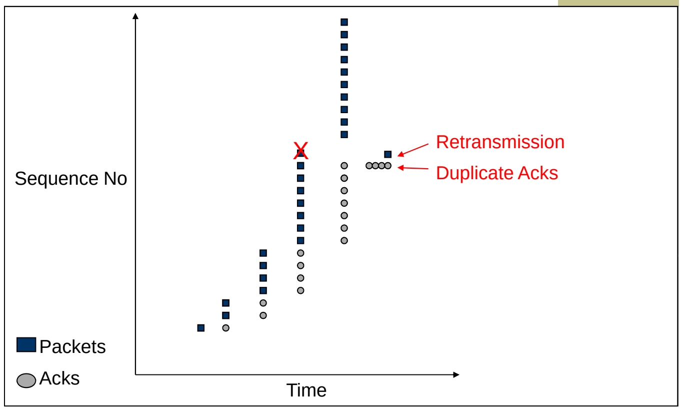
Timer Granularity

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

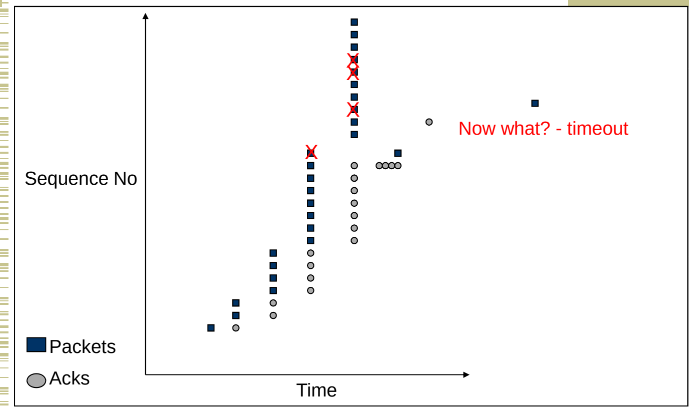
Fast Retransmit

- 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

Fast Retransmit



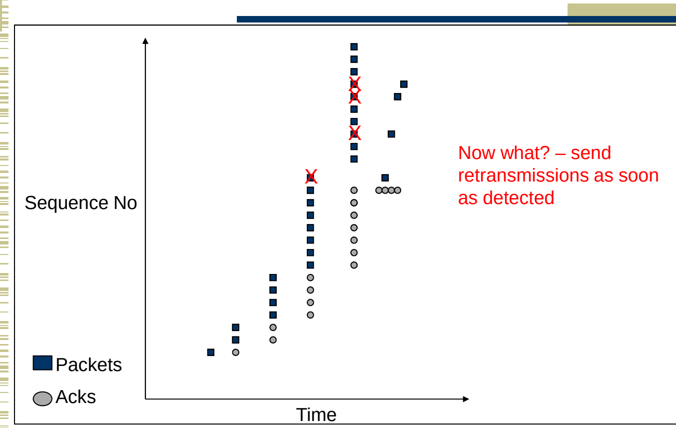
TCP (Reno variant)



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

SACK



Performance Issues

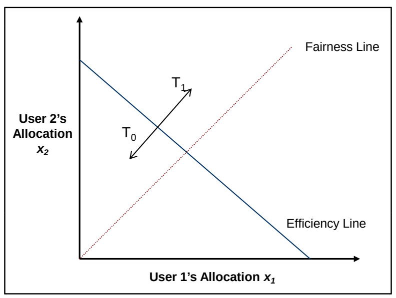
- Timeout >> fast rexmit
- Need 3 dupacks/sacks
- Not great for small transfers
  - Don't have 3 packets outstanding
- What are real loss patterns like?

Outline

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

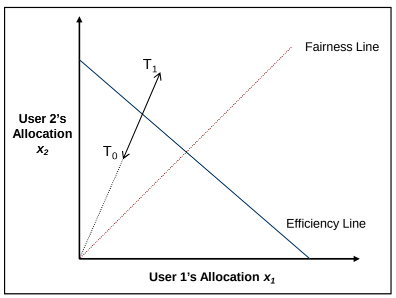
Additive Increase/Decrease

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



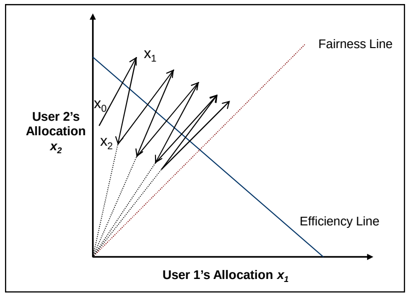
Muliplicative Increase/Decrease

- 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

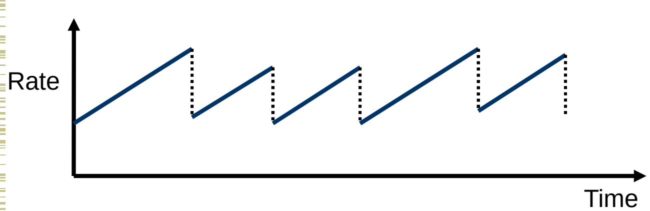


TCP Congestion Control

- 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



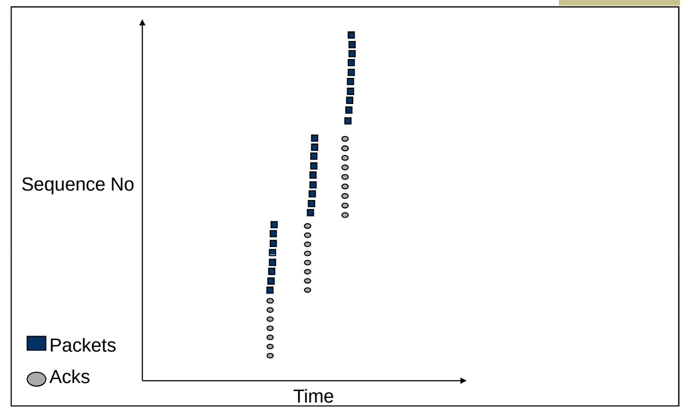
Implementation Issue

- 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)

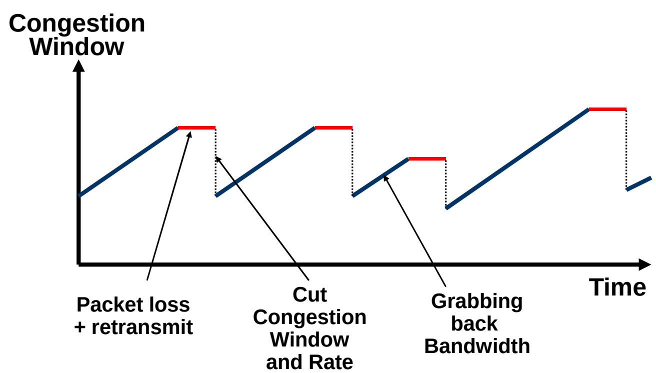
Congestion Avoidance

- 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



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