CSE/EE 461 Lecture 13
Connections and Fragmentation

Tom Anderson

tom@cs.washington.edu

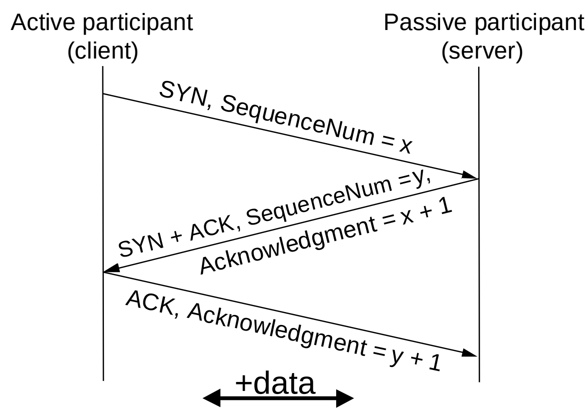
Peterson, Chapter 5.2

TCP Connection Management

- Setup
  - assymetric 3-way handshake
- Transfer
  - sliding window; data and acks in both directions
- Teardown
  - symmetric 2-way handshake
- Client-server model
  - initiator (client) contacts server
  - listener (server) responds, provides service

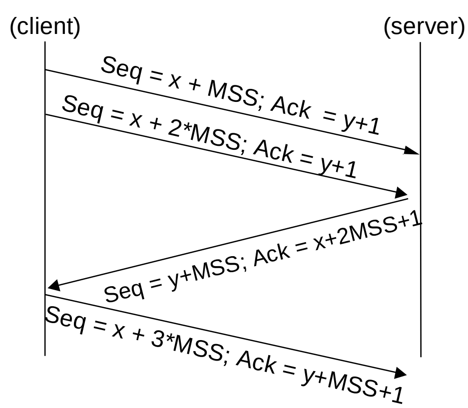
Three-Way Handshake

- Opens both directions for transfer



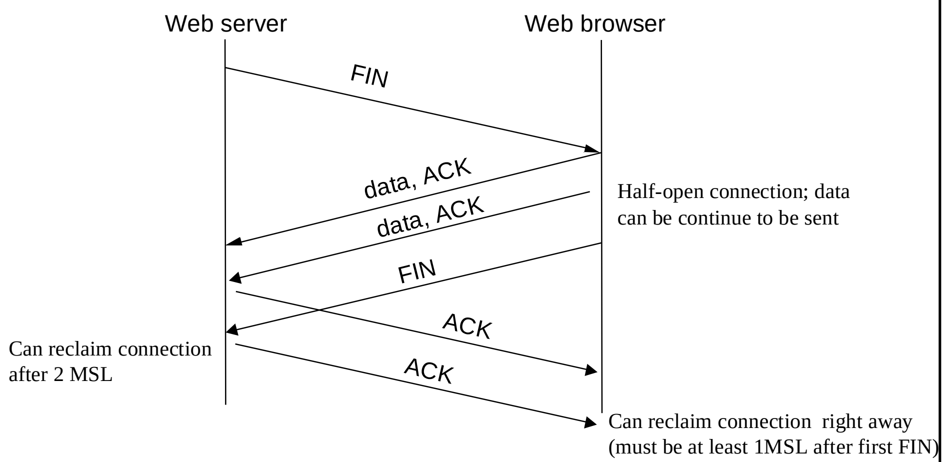
TCP Transfer

- Connection is bi-directional
  - acks can carry response data

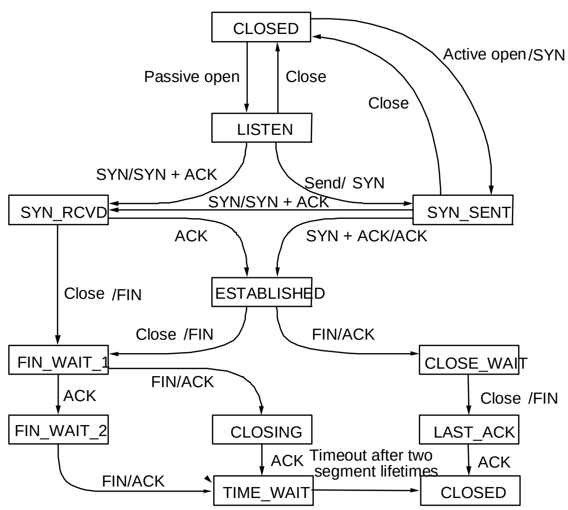


TCP Connection Teardown

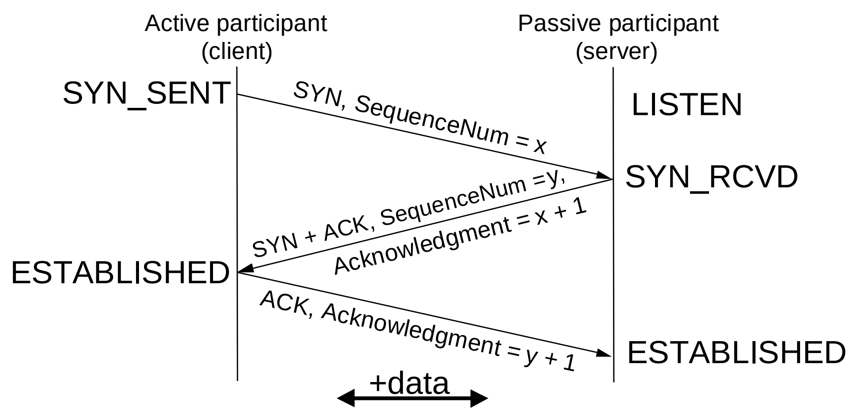
Symmetric: either side can close connection



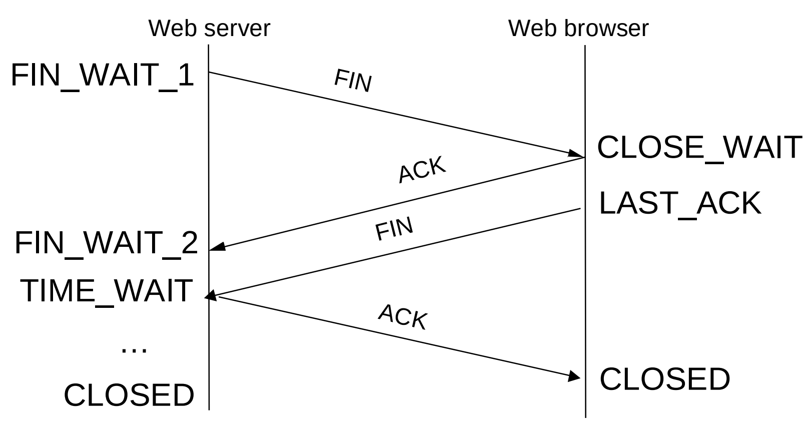
TCP State Transitions



TCP Connection Setup, with States



TCP Connection Teardown

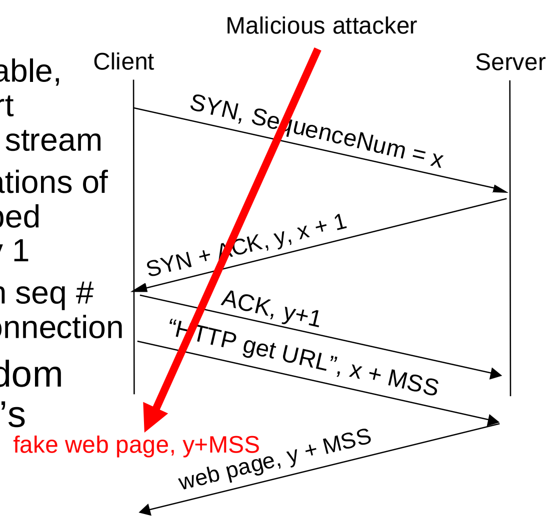


The TIME_WAIT State

- We wait 2MSL (two times the maximum segment lifetime of 60 seconds) before completing the close
- Why?
- ACK might have been lost and so FIN will be resent
- Could interfere with a subsequent connection

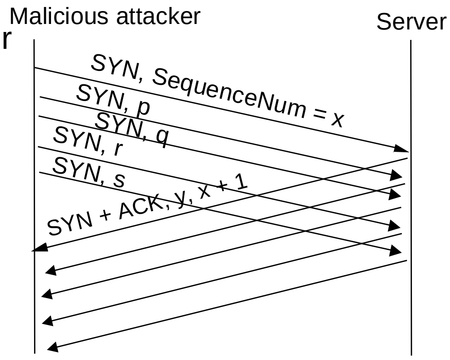
TCP Handshake in an Uncooperative Internet

- TCP Hijacking
  - if seq # is predictable, attacker can insert packets into TCP stream
  - many implementations of TCP simply bumped previous seq # by 1
  - attacker can learn seq # by setting up a connection
- Solution: use random initial sequence #'s
  - weak form of authentication



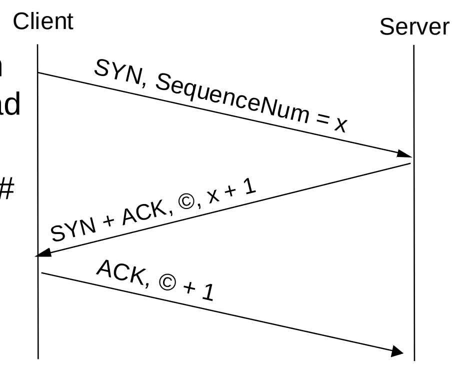
TCP Handshake in an Uncooperative Internet

- TCP SYN flood
  - server maintains state for every open connection
  - if attacker spoofs source addresses, can cause server to open lots of connections
  - eventually, server runs out of memory



TCP SYN cookies

- Solution: SYN cookies
  - Server keeps no state in response to SYN; instead makes client store state
  - Server picks return seq # y = © that encrypts x
  - Gets © +1 from sender; unpacks to yield x

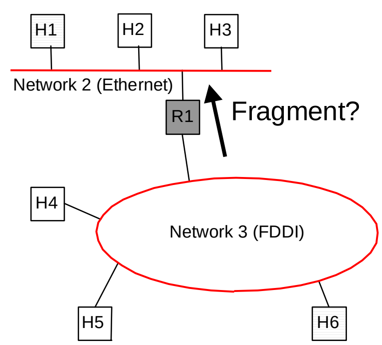


IP Fragmentation

- Both TCP and IP fragment and reassemble packets. Why?
  - IP packets traverse heterogeneous nets
  - Each network has its own max transfer unit
    - Ethernet ~ 1400 bytes; FDDI ~ 4500 bytes
    - P2P ~ 532 bytes; ATM ~ 53 bytes; Aloha ~ 80bytes
  - Path is transparent to end hosts
    - can change dynamically (but usually doesn't)
- IP routers fragment; hosts reassemble

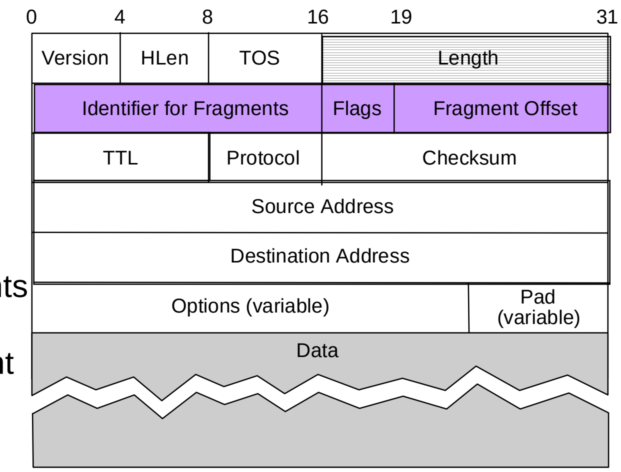
Fragmentation Example

- Different networks may have different frame limits (MTUs)
  - Ethernet 1.5K, FDDI 4.5K
- Don't know if packet will be too big beforehand
  - IPv4: fragment on demand and reassemble at destination
  - IPv6: network returns error message so host can learn limit



Fragment Fields

- Fragments identified by (src, dest, id)
  - Offset gives start, length
- Flags
  - More Fragments (MF)
  - Don't Fragment (DF)



Fragment Considerations

- Relating fragments to original datagram provides:
  - Tolerance of loss, reordering and duplication
  - Ability to fragment fragments
- Consequences of fragmentation:
  - Loss of any fragments causes loss of entire packet
    - possibility of congestion collapse!
  - Need to time-out reassembly when any fragments lost
  - Complicates router hardware => slow path
  - Limits sending speed?
    - To avoid duplicates confusion: 2^16/MSL ~ 500 pkts/sec
- Better to avoid fragmenting if at all possible!

How can TCP choose segment size?

- Pick LAN MTU as segment size?
  - LAN MTU usually larger than WAN MTU

  => Most Internet traffic would be fragmented
- Pick smallest MTU across all networks in Internet?
  - Most traffic is local!
    - Local file server, web proxy, DNS cache, ...
  - Increases packet processing overhead
- Discover MTU to each destination?
  - Path MTU discovery

Path MTU Discovery

- Path MTU is the smallest MTU along path
  - Packets less than this size don't get fragmented
- Hosts send packets with DF (don't fragment) set
  - Routers return ICMP packet to source if too large
    - similar to TTL exceeded
  - Binary search to find largest MTU that doesn't fragment
    - separately for each destination subnet
  - Source TCP uses as segment size
  - Destination TCP reassembles

Layering Revisited

- IP layer "transparent" packet delivery
  - Implementation decisions affect higher layers (and vice versa)
    - Fragmentation => reassembly overhead
      - path MTU discovery
    - Packet loss => congestion or lossy link?
      - link layer retransmission
    - Reordering => packet loss or multipath?
      - router hardware tries to keep packets in order
    - FIFO vs. active queue management

IP Packet Header Limitations

- Fixed size fields in IPv4 packet header
  - source/destination address (32 bits)
    - limits to ~ 4B unique public addresses; about 800M allocated
    - NATs map multiple hosts to single public address
  - IP ID field (16 bits)
    - limits to 65K fragmented packets at once
    - in practice, fewer than 1% of all packets fragment
  - Type of service (8 bits)
    - unused until recently; needed to express priorities
  - TTL (8 bits)
    - limits maximum Internet path length to 255; typical is 30
  - Length (16 bits)
    - Much larger than most link layer MTU's; path MTU discovery

TCP Packet Header Limitations

- Fixed size fields in TCP packet header
  - seq #/ack # -- 32 bits (can't wrap within MSL)
    - T1 ~ 6.4 hours; OC-192 ~ 3.5 seconds
  - source/destination port # -- 16 bits
    - limits # of connections between two machines (NATs)
    - ok to give each machine multiple IP addresses
  - header length
    - limits # of options
  - receive window size -- 16 bits (64KB)
    - rate = window size / delay
    - Ex: 100ms delay => rate ~ 5Mb/sec