CS2105

AY22/23 Sem 2

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01. Introduction

- **Network Edge** - Hosts (Clients and servers)
- **Access Networks** - Wired and wireless communication links
- **Network Core** - Network of interconnected routers

Network Core

Packet-Switching

- Host breaks messages into packets of L bits
- Transmits packets into access network at transmission rate R (aka Link bandwidth, capacity)

$$\text{Packet Transmission Delay} = \frac{\text{Packet Size (bits)}}{\text{Transmission Rate (bits/sec)}}$$

- **Store and Forward** - Entire packet must arrive at router before being transmitted to next link

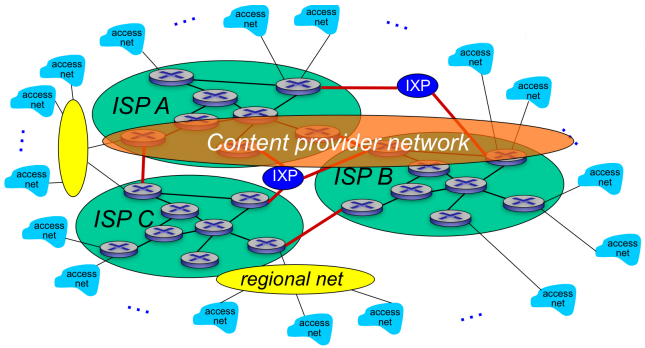
Key Functions of Network Core

- **Routing** - Determines source-destination routes taken by packets (How we get the hashtable)
- **Forwarding** - Move packets from router's input to correct router output

Circuit Switching

- Resources reserved for call between source and destination
- Pros: Better performance
- Cons: More resources

Internet Structure



- End systems connect to Internet via **Access Internet Service Providers (ISPs)**
- ISPs connect to larger global ISPs (usually competitors)
- Large ISPs connect via **peering links** or **internet exchange points (IXP)**
- **IXP** - Physical place with routers from different ISPs
- **Regional Networks** - Smallers ISPs
- **Content Provider Networks** - Provide content close to end users

Loss, Delay, and Throughput

Packet Loss

- If Arrival Rate $>$ Transmission Rate, packets will queue and can be dropped if queue fills up
- Solutions: Lost packets can be retransmitted

Packet Delay

$$d_{\text{nodal}} = d_{\text{proc}} + d_{\text{queue}} + d_{\text{trans}} + d_{\text{prop}}$$

- **Nodal Processing** - (d_{proc}) Check for bit errors and determine output link
- **Queueing Delay** - (d_{queue}) Time at queue waiting for transmission
- **Transmission Delay** - (d_{trans}) Time to load packet onto link
  - $d_{\text{trans}} = \frac{L}{R}$ where L is packet length and R is link bandwidth
- **Propagation Delay** - (d_{prop}) Time for 1 bit to reach end of link
  - $d_{\text{prop}} = \frac{d}{s}$ where d is length of link and s is propagation speed

Throughput

- Rate at which bits transferred between hosts
  - Different from transmission rate, which is the theoretical upper bound
- Average: Rate over long period of time
- Instantaneous: Rate at given point in time

Protocol Layers and Service Models

- **Protocol** - Defines format, order of messages sent and received, and actions taken on message transmission
- Networks are complex with many components. How can we organize its structure?
- **Layering** - Each layer implements a service by doing something within layer and relying on services provided by layer below it
  - Explicit structure allows us to make sense of complex components
  - Easy maintenance (Like OOP, change in 1 layer should not affect others)

Internet Protocol Stack

1. Application
2. Transport
3. Network
4. Link
5. Physical

02. Application Layer

- Programs that run on end systems, and not on network-core devices

Client-server Architecture

- Server: Always-on host, Permanent IP address
- Clients: Communicates with server, Intermittently connected, Dynamic IP addresses, Do not communicate with each other directly

P2P Architecture

- Peers request service from other peers and provide service in return
- No always-on server, Intermittently connected, Dynamic IP addresses
- **Self Scalability** - New peers offer new services and demands

Process

- **Process** - Program running in host
- **Inter-process Communication** - How 2 processes in 1 host communicate
- **Messages** - Processes in different hosts communicate by exchanging this
- **Client Process** - Process that initiates communication
- **Server Process** - Process that waits to be contacted
- **Socket** - Process sends/receives messages to/from its socket (like a door)
  - Outside of socket, transport layer delivers message

Addressing Processes

- Motivation: IP address is not enough to address process, since many processes can be running on same host
  - **Identifier** - IP address and port number
  - **Port Number:** - Associated with process on host

Transport Protocol Services

1. **TCP** - Transmission Control Protocol
   - Reliable transport
   - Flow control: Sender does not overwhelm receiver
   - Congestion control
   - Connection-oriented: Setup required between client and server
2. **UDP** - User Datagram Protocol
   - Unreliable data transfer
   - Fast

App-layer Protocol

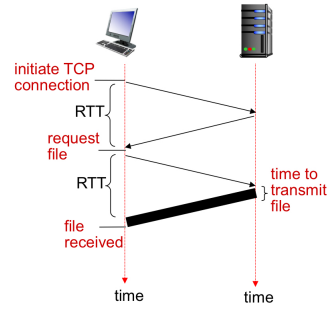
- Types of messages exchanged (e.g. Request or response)
- Message syntax: How fields are delineated
- Messages semantics: Meaning of information in fields

HTTP

- **Hypertext Transfer Protocol** - Web's application layer protocol
- Motivation: Web page consists of objects (HTML, images). Need method to request/send web objects.
- Follows client/server model
- Uses TCP
- **Stateless** - Server maintains no information about past requests

Non-persistent HTTP

- At most 1 object sent over TCP connection
- Downloading multiple objects requires multiple TCP connections

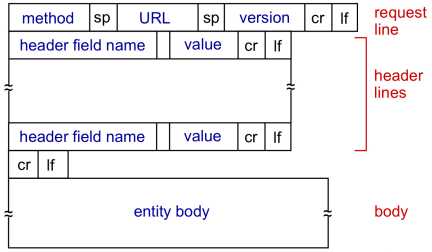


- Server closes TCP connection after sending file
- **Return Trip Time** - (RTT) Time for small packet to travel from client to server and back
- Response Time: 2 RTT + File transmission time

Persistent HTTP

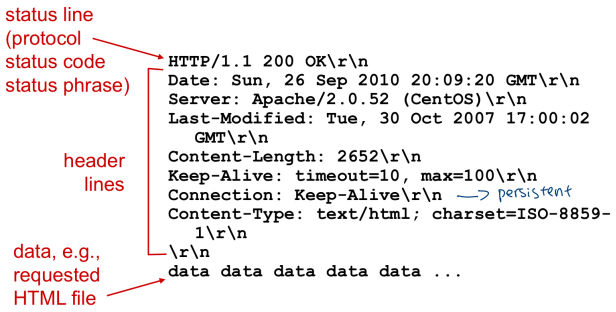
- Multiple objects can be sent over single TCP connection
- Server leaves TCP connection open after sending response
- Subsequent objects can use same TCP connection and be sent using 1 RTT + File transmission time

HTTP Request Message



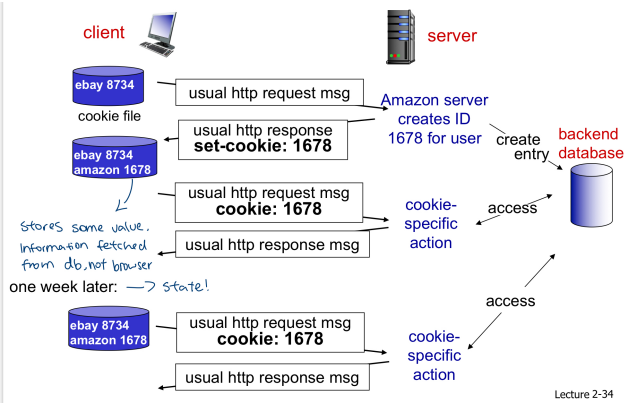
- To upload form input: **POST method** - Input uploaded via entity body and **URL method** - Input uploaded in URL field of GET method
- **HTTP/1.0** - GET, POST, HEAD (Ask server to leave request object out of response). Non-persistent by default.
- **HTTP/1.1** - GET, POST, HEAD, PUT, DELETE. Persistent by default.

HTTP Response Message



Cookies

- Maintains state on client side
- Components: Cookie header for HTTP response, Cookie header for HTTP request, Cookie file on user's host (Key-value pair), Database on server

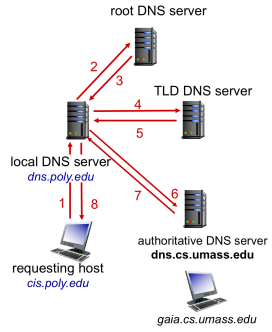


Web Cache (Proxy Server)

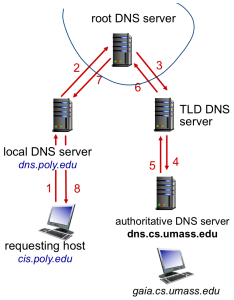
- Goal: Fulfill request without involving origin server via caching
- Browser sends all HTTP requests to cache
- Pros: Faster, Reduces traffic to origin server
- Cons: What if origin updates?
  - **Conditional GET** - Origin server doesn't send object if cache has updated version
  - Cache: Specifies date of cached copy in HTTP request to origin
  - Origin Server: Response contains no object if cached object is updated

Domain Name System

- Maps between IP address and name (e.g. yahoo.com)
- Implemented using distributed and hierarchical databases
- Application-layer protocol
- Uses UDP
- **Local DNS Name Server** - Local cache of name-to-address mapping. Forwards query into hierarchy.
  - **Time to Live** - (TTL) Cached mappings disappear after some time
- **Root Name Server** - Contacted by local name server that cannot resolve name. Provides IP address of TLD servers.
- **Top-level Domain Server** - (TLD) Provides IP address of authoritative server
- **Authoritative DNS Server** - Organization's own DNS server. Provides mappings for organization's named hosts.
- Iterated query: "Not sure, ask this server"

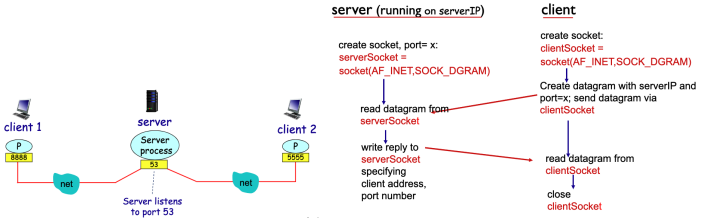


- Recursive query: "Okay, let me find for you"
  - Heavy load on upper levels of hierarchy



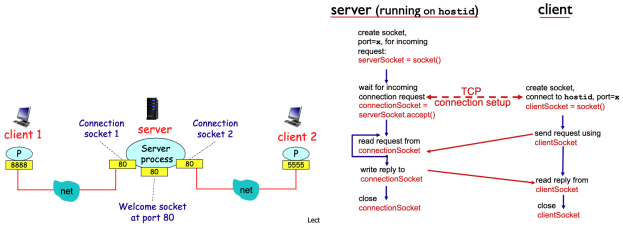
03. Socket Programming with UDP and TCP

UDP Socket



- No connection beforehand. Just send it.
- Server has 1 socket to serve all clients
- Sender attaches destination IP address and port number (**Stateless**)
- Unreliable datagram: Data may be lost or out-of-order
  - **Datagram** - Group of bytes

TCP Socket



- Client establishes connection to server via welcome socket
- Server makes new socket for each client
- Server identifies client via connection (**Stateful**)
- Reliable stream pipe: Data always in order

04. UDP and Reliable Protocol

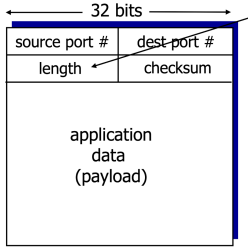
Transport Layer Services

- Transport layer: Logical communication between **processes**
- Network layer: Logical communication between **hosts** (Unreliable)

UDP

- On top of network layer, UDP adds:
  1. Connectionless multiplexing/de-multiplexing
     - UDP segments contain both source and destination ports
     - Multiplexing: Sent to target processes
  2. Checksum

UDP Segment Header



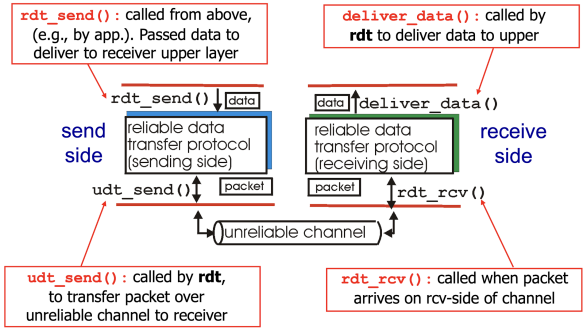
UDP segment format

Checksum

- Goal: Detect errors in received segment
  1. Treat UDP segment as sequence of 16-bit integers
  2. Add every 16-bit integer (Carry added back to result)
  3. Invert to get UDP checksum (1's complement)
  4. When receiving, sum segment again. All 1s if correct.

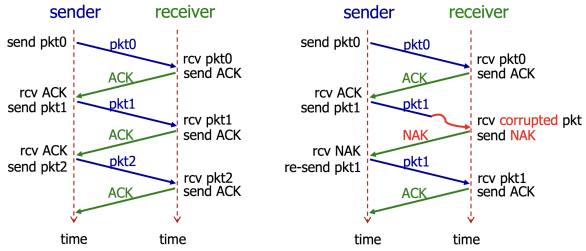
Reliable Data Transfer (rdt)

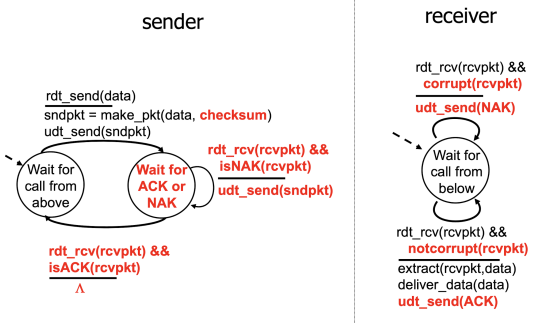
- Characteristics of unreliable channel will determine services provided by rdt



rdt 2.0

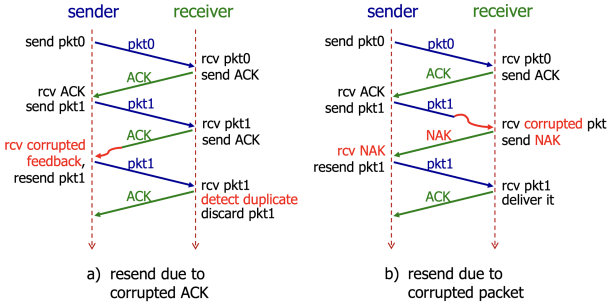
- New problem: **Bit error in data**
- Solution:
  - Recipient perform checksum to detect bit errors
  - **ACK** - Receiver tells sender that packet received is ok
  - **NAK** - Receiver tells sender that packet received has errors
  - After receiving NAK, sender retransmits packet

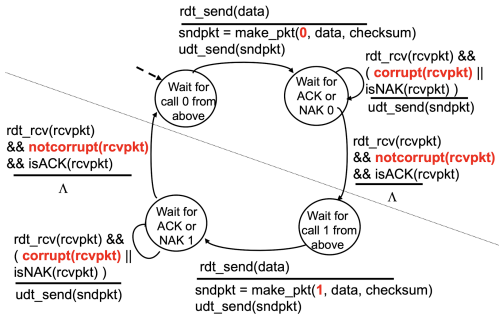


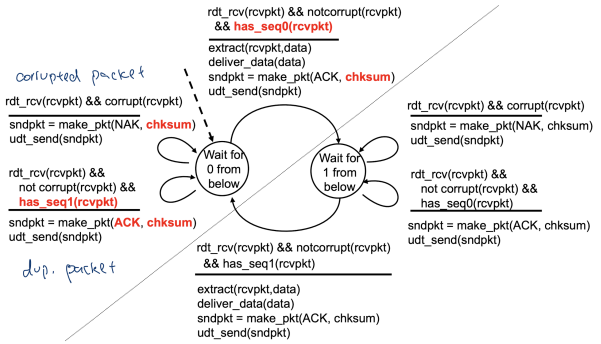


rdt 2.1

- New problem: ACK/NAK may be corrupted
- Solution: Sender does checksum and retransmits packet if ACK/NAK is corrupted
- New problem: If ACK is corrupted, sender will send a **duplicate** packet
- Solution: Sender adds **sequence number** to each packet and receiver discards duplicate packet (Only 0 and 1 needed)





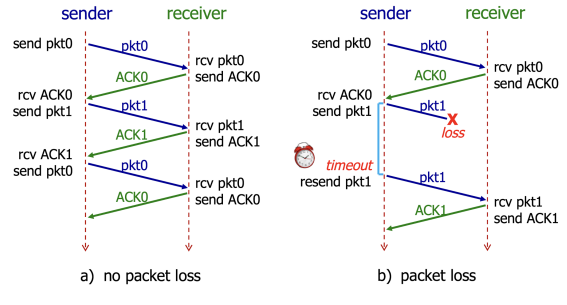


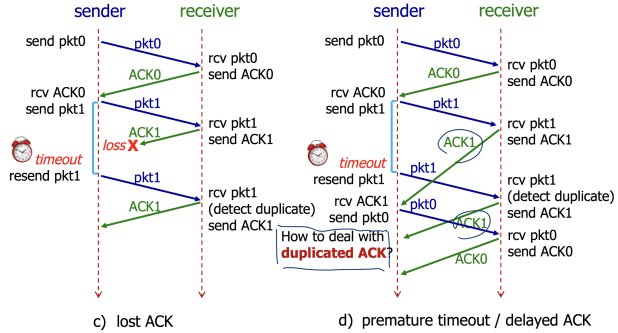
rdt 2.2

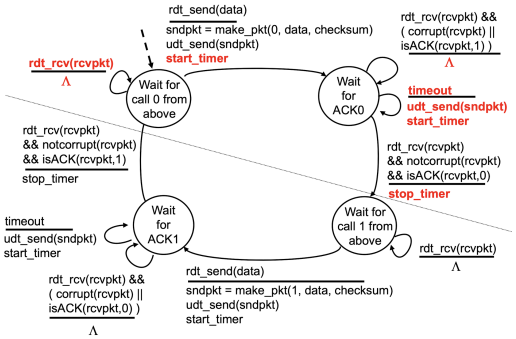
- New problem: Use ACKs only. No NAK.
- Solution: Receiver sends ACK for last packet received. **ACK must include seq number of packet**.

rdt 3.0

- New problem: Lost packets
- Solution: Sender waits for some time for ACK and retransmits packet
  - What if duplicate packet? Sequence number handles this.
  - What if duplicate ACK? Do nothing. Only retransmit after timeout.
  - What if packet out of order? Have more sequence numbers.







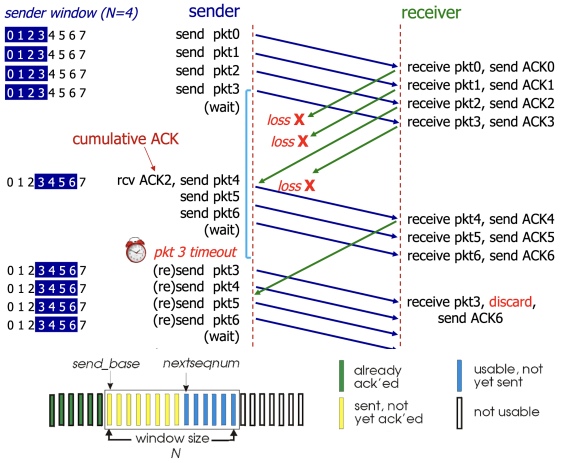
Performance of rdt 3.0

- **Stop-and-wait protocol** - Sender sends 1 packet at a time, then waits for receiver response
- Performance is bad. Stop-and-wait protocol limits use of resources
- **Utilization** - Fraction of time sender is busy sending
  - Given: 1 Gbps link, 15 ms prop. delay, 8000 bit packet
  - $D_{trans} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 0.008 \text{ ms}$
  - $U_{sender} = \frac{D_{\text{trans}}}{RTT + D_{\text{trans}}} = \frac{0.008}{30.008} = 0.027\%$

Pipelined Protocols

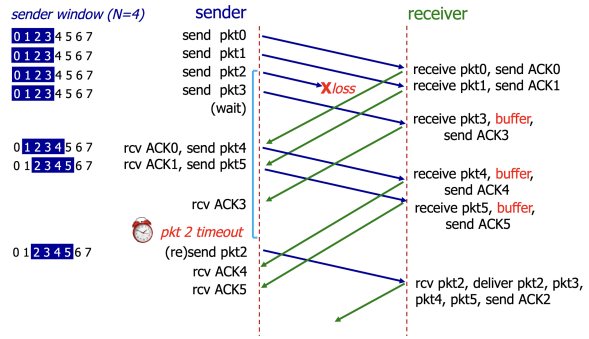
- **Pipelining** - Sender allows sending of multiple not-yet-ACKed packets
  - Need more sequence numbers
  - Buffering at sender and receiver

Go-Back-N



- Intuition: Sliding window
- Sender:
  - Sends when sliding window reaches packet
  - 1 timer to receive ACK for packet at `send_base`
  - If timeout(n), retransmit packet n and other pkts in window
  - If receive ACK n, check if can shift sliding window. Previous un-ACKed packets guaranteed to be received.
  - If duplicate ACK, ignore like in rdt 3.0. Retransmit only on timeout.
- Receiver:
  - **Cumulative ACK** - ACK for correct pkt with highest in-order sequence
  - Discard out-of-order packets
- Not efficient, since future packets discarded if pkt lost and out-of-order

Selective Repeat



- Receiver individually ACKs correct pkts (Not accumulative) and sender maintains timer for each unACKed pkt
- Sender:
  - If timeout(n), retransmit packet n only
  - If ACK(n) and n is smallest unACKed pkt, slide window
- Receiver:
  - Once receive pkt n in window, send ACK(n). If out-of-order, buffer. If in-order, deliver and slide window
  - Once receive pkt n outside of window, still send ACK(n)

Summary

rdt	New problems	New features
1.0	n/a	n/a
2.0	Bit error in data	Sender: Checksum, Receiver: ACK/NAK
2.1	Bit error in feedback, Duplicate packets	Receiver: Checksum, Sender: Sequence number
2.2	Remove NAK	Receiver: Sequence number
3.0	Packet loss in data and feedback	Timeout, Re-transmission

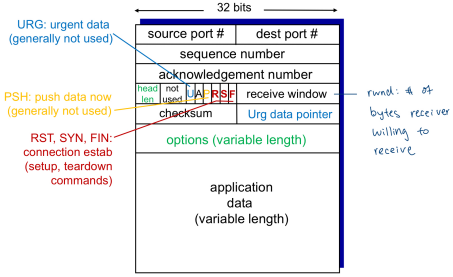
05. TCP

- Point-to-point: 1 sender and 1 receiver
- Connection-oriented
- Full duplex data: Data and feedback can be sent together both ways
  - Before in UDP: Data only goes 1 way
- Reliable, in-order **byte stream** (segment)
  - Before in UDP: Send packet by packet
- Pipelined: **Dynamic window size** set by flow control

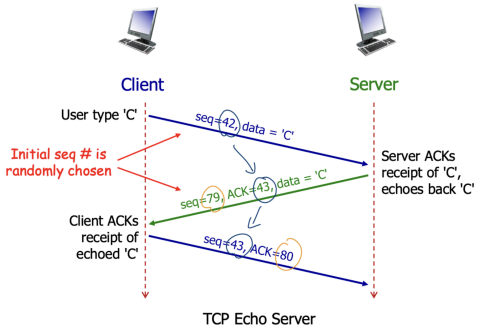
Buffers and Segments

- Sender and receiver both have 2 buffers to send and receive data
- **Maximum Segment Size** - (MSS) Usually 1460 bytes. Limited by max. transmission unit (MTU)

Segment Structure



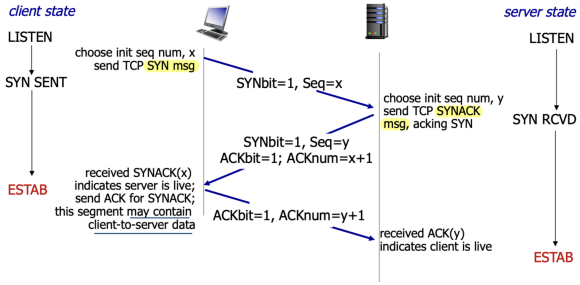
- Source/Dest. Port Number: Same as UDP, except ports can be demultiplexed to different sockets
- **Sequence Number** - Byte number of first byte of data in segment
- **Acknowledgement Number** - Sequence number of **next byte expected** from other side (Diff. from UDP)



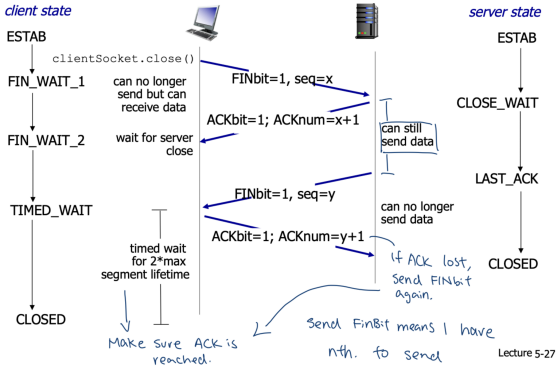
- **Receive Window** - For flow control. Receiver controls sender, so sender won't overflow receiver's buffer. Constrains sliding window.

Connection Management

3 Way Handshake



Closing a Connection

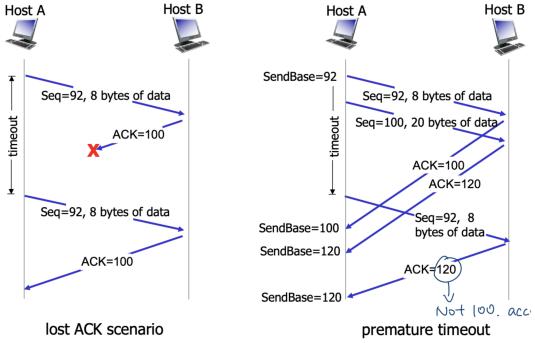


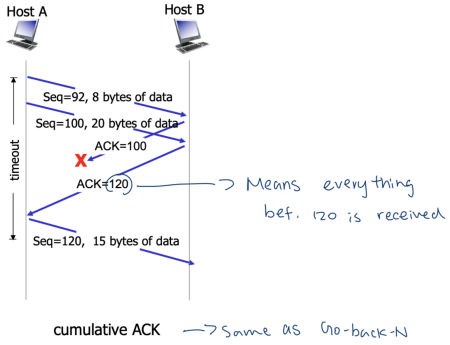
Reliable Data Transfer

- On top of UDT service, TCP adds:
  - Pipelined segments
  - Cumulative ACKs
  - Single retransmission timer
- Out-of-order packets not specified in TCP. Up to implementer.
- Retransmits on timeout or 4 duplicate ACKs

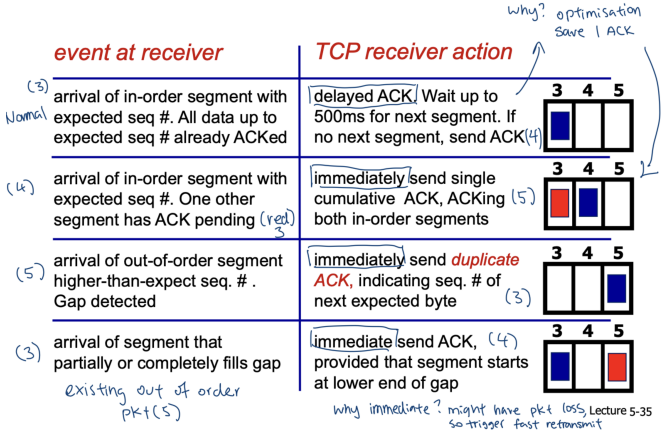
Sender Events

- If data received from application:
  - Create segment with sequence number
  - Start timer for **oldest un-ACKed segment**
- If timeout:
  - Retransmit segment causing timeout (Similar to selective repeat)
  - Restart timer
- If ACK received:
  - If ACK acknowledges previously un-ACKed segments, update window and start timer.





Receiver Events



- Delayed ACK for in-order segment
  - If no next segment received in time, send ACK
  - If next segment received in time, send ACK for 2nd segment (Saves 1 ACK. Works due to cumulative ACK.)
- Immediate ACK for out-of-order segment (No matter if it creates or fills gap)
  - Why immediate ACK? Might have packet loss, so can trigger fast retransmit earlier.

Timeout

- Motivation: What should timeout value be? Too short results in premature timeout. Too long results to slow reaction.
- **Sample RTT** - Measured time from segment transmission until ACK receipt. If retransmission, forget it.
  - Sample RTT varies a lot, so not accurate. How to get better estimate?
- **Estimated RTT** - Average of recent measurements
  - Uses previous Estimated RTT
  - Usually, $\alpha = 1/8$

$$\text{Estimated RTT} = (1-\alpha) * \text{Estimated RTT} + \alpha * \text{Sample RTT}$$

- **Timeout Interval** - Estimated RTT + "Safety margin" (Deviation of estimate)
  - Usually, $\beta = 1/4$

$$\text{Dev RTT} = (1-\beta) * \text{Dev RTT} + \beta * |\text{Sample RTT} - \text{Estimated RTT}|$$

$$\text{Timeout Interval} = \text{Estimated RTT} + 4 * \text{Dev RTT}$$

Fast Retransmit

- Motivation: Timeout period often quite long. Long delay before resending lost pkt.
- **Fast Retransmit** - If sender receives 4 ACKs for same data, resend un-ACKed segment with smallest sequence number