CS2105
AY22/23 Sem 2
github.com/securespider

01. Introduction

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

Network Core

- **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

Packet-Switching

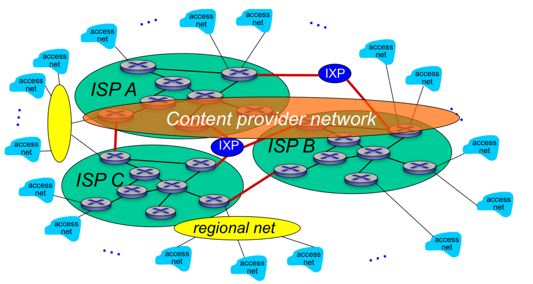
- Host breaks messages into packets of L bits
- Transmits packets into access network at transmission rate R (aka Link bandwidth, capacity)

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

Circuit Switching

- Resources reserved for call between source and destination
- End to end resources allocated and reserved
- Set up and teardown required
- Circuits like (guaranteed) performance cos circuit reserved per usage
- Pros: Better performance
- Cons: More resources (Unable to share with other processes)

Internet Structure



- End systems connect to Internet via **Access Internet Service Providers (ISPs)** (autonomous systems → regional ISP → Access ISP → hosts)
- 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 buffer fills up
- Solutions: Lost packets can be retransmitted

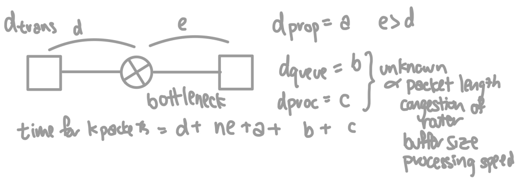
Packet Delay

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

- **Nodal Processing** - (d_{proc}) Check for bit errors and determine output link
- **Queueing Delay** - (d_{queue}) Time at queue waiting for transmission, router congestion
- **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
  - Does not depend on packet size, number of packets or number of intermediate nodes

Throughput

- Rate at which bits transferred between hosts
  - Different from transmission rate (Theoretical upper bound)
  - Measures end-to-end, even through intermediaries
  - Irregardless of the size of packets
  - Dealing with bottlenecks over n packets:



- Average: Rate over long period of time
- Instantaneous: Rate at given point in time

Protocol Layers and Service Models

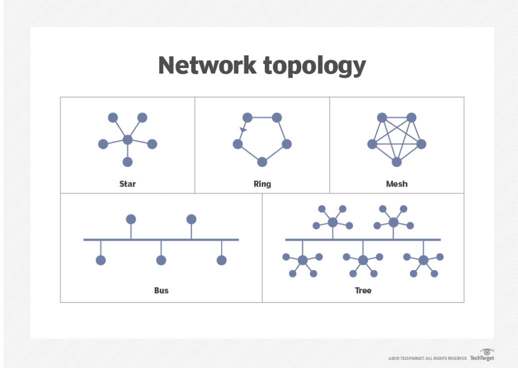
- **Protocol** - Defines format, order of messages sent and received, and actions taken on message transmission
- **Layering** - Each layer implements a service by doing something within layer and relying on services provided by layer below it
  - Explicit structure allows us to make sense of complex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

Internet Protocol Stack

1. **Application** - eg. FTP, SMTP
2. **Transport** - TCP, UDP
3. **Network** - IP, Routing protocols
4. **Link** - Ethernet, WiFi, PPP
5. **Physical** - Wires, bits

Encapsulation - Take information from a higher layer and adds a header to it, treating the higher layer information as data

Topology



- **Star** - Lowest cost and simplest
- **Mesh** - High redundancy, increased cost but highest transmission speeds

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, waits for incoming req and provides service to client
- Clients: Communicates with server, Intermittently connected, Dynamic IP addresses, Do not communicate with each other directly, initate connection with server to request services

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

- **Inter-process Communication** - How 2 processes in 1 host communicate
- **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
  - **IP** - 32-bit address for identifying host
  - **Port Number** - 16-bit to identify specific process on host

Services

- **Data Integrity** - Reliable data transfer
- **Timing** - Low delay/latency
- **Throughput** - Minimum amount of throughput for effectiveness
- **Security** - Encryption, data integrity

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

- Does not guarantee minimum throughput and timing

App-layer Protocol

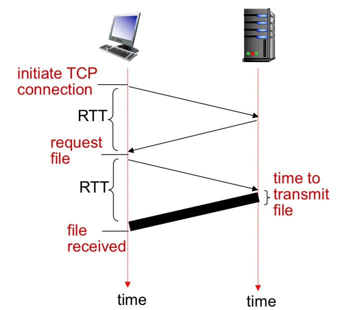
- Types of messages exchanged (e.g. Request or response)
- Message syntax: How fields are delineated
- Messages semantics: Meaning of information in fields

HTTP - 80

- **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 1.0

- At most 1 object sent over TCP connection
- Downloading multiple objects requires multiple TCP connections
- New TCP connection for each web resource/object

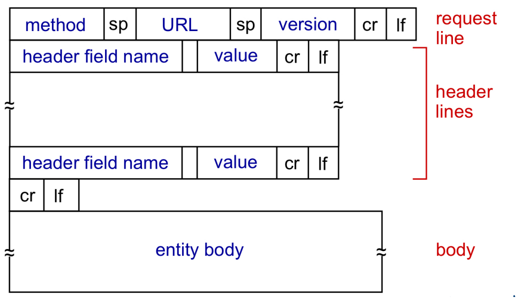


- Server closes TCP connection after sending file
- **Return Trip Time** - (RTT) Time for small packet to travel from client to server and back
- RTT is $2 * d_{prop}$
- Response Time: 2 RTT + File transmission time

Persistent HTTP

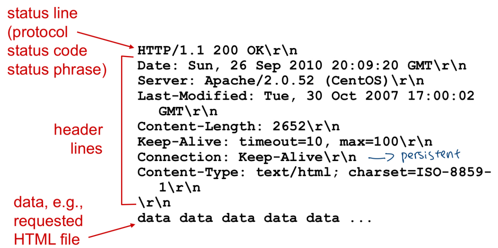
- Multiple objects can be sent over single TCP connection
- Server leaves TCP connection open after sending response
- As little as one RTT for all referenced objects

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)
- **HTTP/1.1** - GET, POST, HEAD, PUT, DELETE

HTTP Response Message



Error codes

200 OK

301 Moved permanently

304 Not modified (during conditional GET)

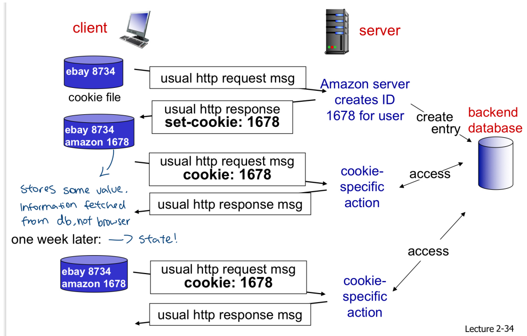
400 Bad request

404 Resource not found

505 HTTP Version Not Supported

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 server 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 (If-modified-since)
  - Origin Server: Response contains no object if cached object is updated (Empty content body)

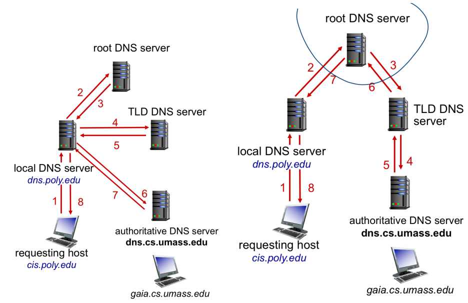
Domain Name System - 53

- Maps between hostname (e.g. yahoo.com) and IP address
- Implemented using distributed and hierarchical databases
- Application-layer protocol (Why?)

  Build complexity in edges to simplify inner core networks
- Uses UDP (Much faster and smaller packet overhead)
- Services provided

  Host/Email aliasing Additional name for complicated hostnames

  Load balancing Providing multiple IP addresses for the same URL
- **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

Local DNS servers does both

DNS Caching

Cache entry - Cache record of recent name mapping

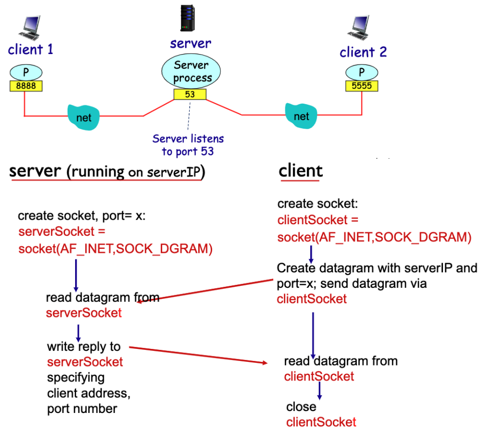
- Timeout after some time based on TTL
- Best effort name-to-address translation (can be out-of-date)
- **Resource record** - Mapping btw hostname and IP
- Format: ¡NAME, TYPE, VALUE, TTL¿

DNS Cache poisoning

- DNS spoofing attack - Rogue DNS records introduced in resolver's cache, leading to incorrect IP address reply

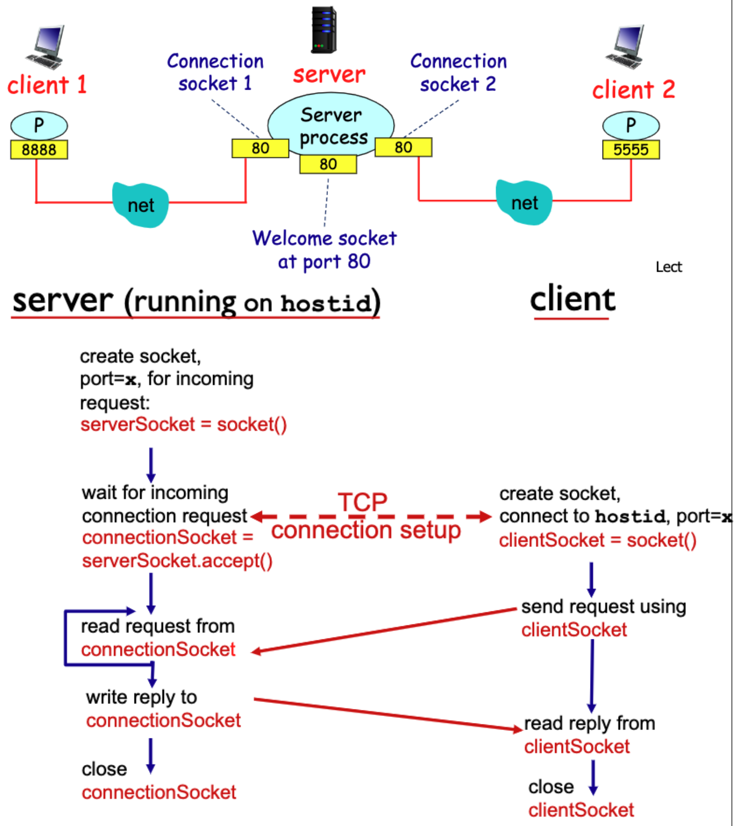
03. Socket Programming with UDP and TCP

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

Remarks

- SOCK_STREAM - TCP, SOCK_DGRAM - UDP
- Can only specify destination port numbers, source port numbers randomly assigned by OS

04. UDP and Reliable Protocol

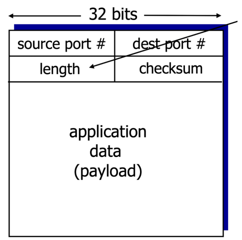
Transport Layer Services

- Provide logical communication between processes in different hosts
- VS network: Process-to-process VS host-to-host
- Sender: Breaks app messages into segment
- Receiver: Reassembles segments

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
     - **Demultiplexing** - Delivering the received segments at the receiver side to the correct app layer processes
  2. Checksum

UDP Segment Header



UDP segment format

Note that length includes 8 byte header + body content

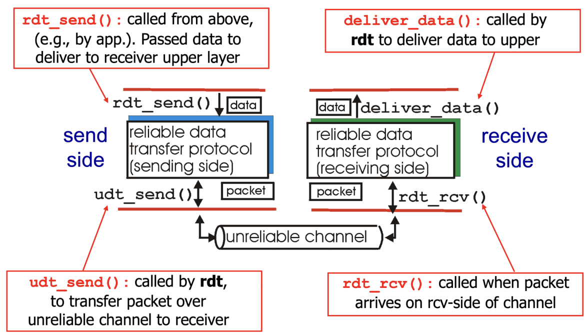
Checksum

- 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.

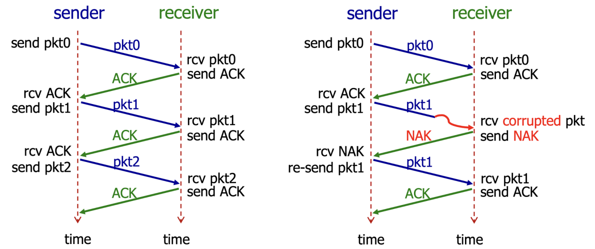
Reliable Data Transfer (rdt)

- Characteristics of unreliable channel will determine services provided by rdt
- Cannot fix unordered packets



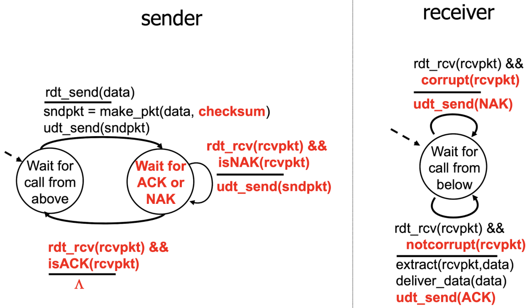
rdt 2.0

- New problem: **Bit error** - May flip bits in packet
- Solution:
  - Perform checksum to detect bit errors
  - **Stop and wait protocol** - Sender sends one packet at a time and wait for response
  - **ACK** - Receiver tells sender that packet received is ok
  - **NAK** - Receiver tells sender that packet received has errors (retransmission)



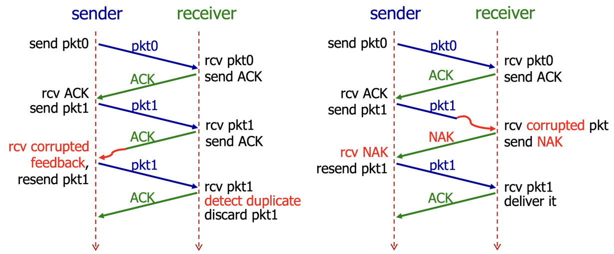
(a) no bit error

(b) with bit error



rdt 2.1

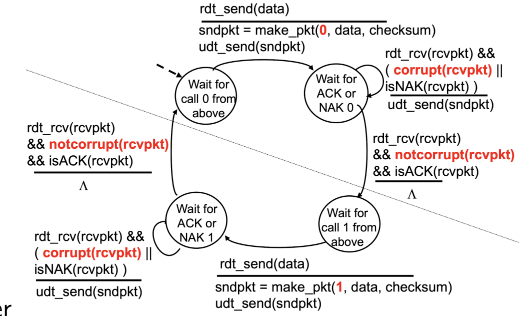
- New problem: ACK/NAK may be corrupted
- Solution: Sender retransmits packet after receiving corrupted ACK/NAK
- New problem: Duplicate packets during retransmission
- Solution: Sender adds **sequence number** to each packet and receiver discards duplicate packet (Only 0 and 1 needed)



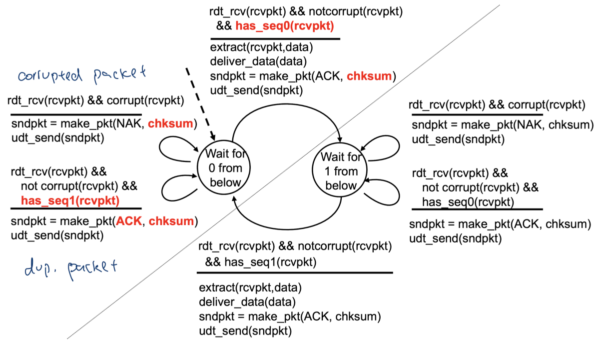
a) resend due to corrupted ACK

b) resend due to corrupted packet

Sender

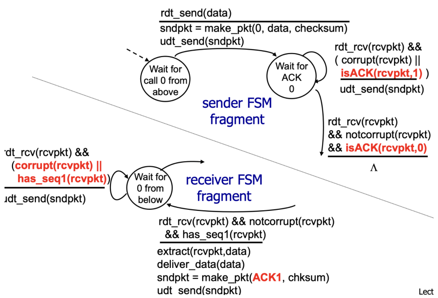


Receiver



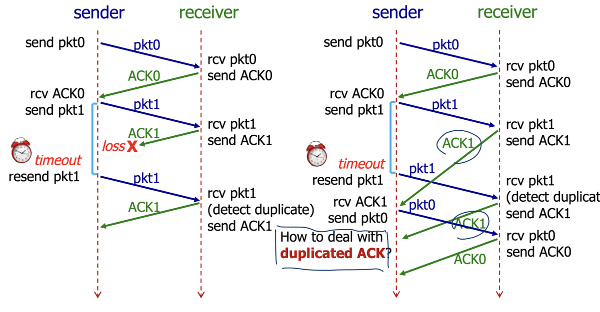
rdt 2.2

- No NAK, only ACKs
- Receiver sends ACK for last packet received. **ACK must include seq number of packet**.
- Receiver send duplicate ACK to retransmit current packet



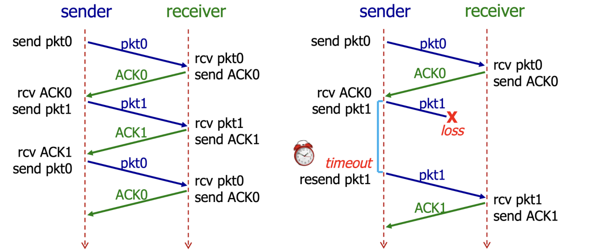
rdt 3.0

- 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.



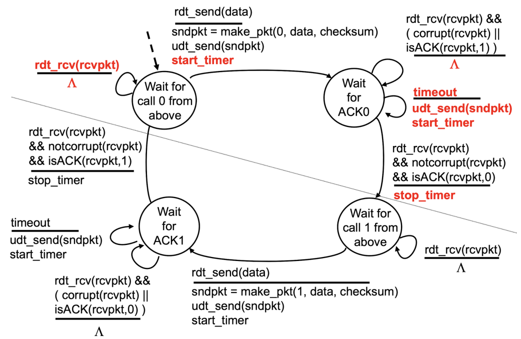
c) lost ACK

d) premature timeout / delayed ACK



a) no packet loss

b) packet loss



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
  - $D_{trans} = \frac{L}{R} = \frac{8000bits}{10^9 bits/sec} = 8microsecs$
  - $U_{sender} = \frac{D_{trans}}{RTT+D_{trans}} = \frac{0.008}{30.008} = 0.027\%$
  - Pipelined Utilisation = U * window_size

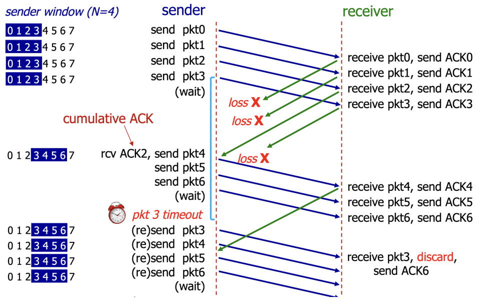
Pipelined Protocols

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

Pipeline vs Parallel (HTTP)

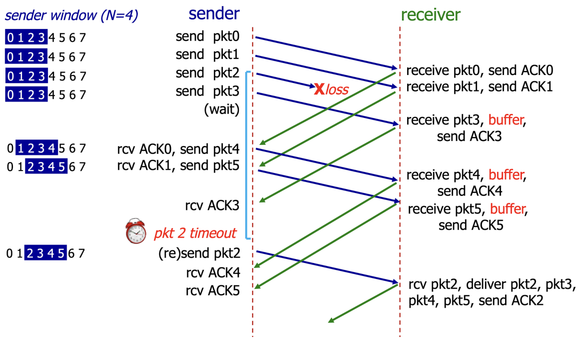
- Parallel connections need to retrieve HTML webpage first before retrieving data elements
- Pipeline can retrieve HTML and data elements all at once

Go-Back-N



- Intuition: Sliding window
- Sender:
  - Timer for oldest in-flight packet
  - If timeout(n), retransmit packet n and other pkts in window
- Receiver:
  - **Cumulative ACK** - ACK for correct pkt with highest in-order sequence
  - Only ACK packets that arrive **in-order**
  - Discard **out-of-order** packets
- Not efficient, since future packets discarded if 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

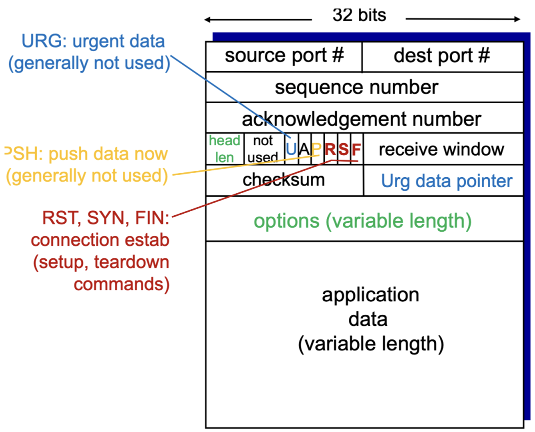
- Receiver:
  - Once receive pkt n in window, send ACK(n). If out-of-order, buffer. If in-order, deliver and slide window
  - Once receive pkt n outside of window, still send ACK(n)
  - ACK m == packet m received but has no implication on receipt of other packets

05. TCP

TCP Features

- **Point-to-point** - 1 sender and 1 receiver
- **Connection-oriented** - Handshaking before exchange data
- **Full duplex** - Bi-directional data flow in connection
- Reliable, in order and pipelined
- Buffers on both hosts
- Limited by **Maximum Segment Size** ≈ 1460B
  - Refers only to content length (no header)
  - Limited by **MTU** - Maximum transmission unit, size of largest IP transaction

Structure

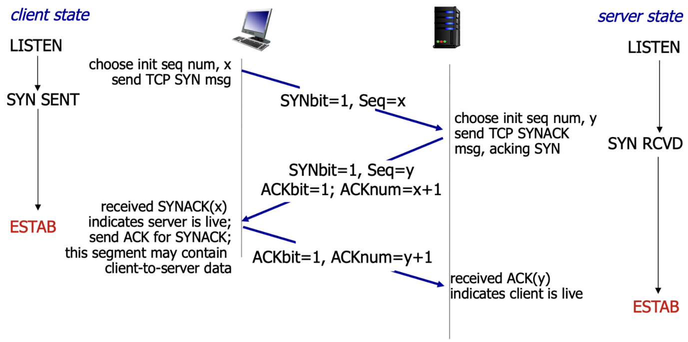


Header is 20-60 bytes long

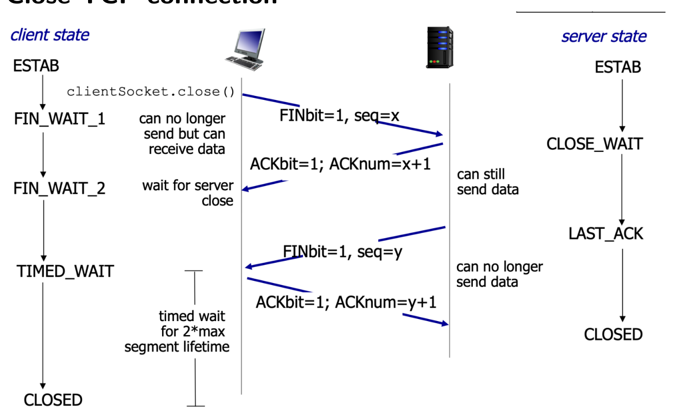
1. Sequence #
   - Let receiver know if packet is retransmission or new data
   - Value only measures the body length in bytes
   - Initial number randomly generated btw $0-(2^{32}-1)$ (Why?)
     - Security
     - Prevent duplicate sequence numbers when link is used extensively
   - ACK # = next byte expected by receiver (inital seq # + data length in bytes)
   - SYN # = byte number of first byte(expected ACK (of sender) - data length)
2. ACK number
3. **rwnd** - Receive window
   - Number of bytes the receiver is willing to accept
   - **Flow control** - Prevent overflow of receiver's buffer
4. Flags - Connection establish or priority

Establish TCP connection

Creates n+1 sockets for n clients (additional 1 for listening)



Close TCP connection



Why TIMED_WAIT (2***MSL** - Maximum segment lifespan)

- Guarantee the port has been closed completely
- Prevent delayed data segments from being received by other TCP connections

RDT

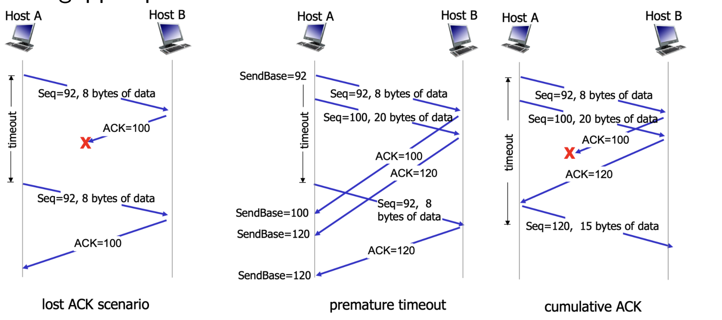
Features

- Pipelined segments
- Cumulative ACKs (like GBN)
- Single retransmission timer
- Retransmit only missing packets (SR)

Retransmission Scenarios

Send ACKs every other packet OR after timeout of packet OR receive gapped packet

event at receiver	*TCP receiver action*
arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed	delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK
arrival of in-order segment with expected seq #. One other segment has ACK pending	immediately send single cumulative ACK, ACKing both in-order segments
arrival of out-of-order segment higher-than-expect seq. # . Gap detected	immediately send *duplicate ACK,* indicating seq. # of next expected byte
arrival of segment that partially or completely fills gap	immediate send ACK, provided that segment starts at lower end of gap



RTT

- Problem: Estimate timeout value for retransmission
- Solution: Average recent **SampleRTT** - Measured time from segment transmission until ACK
- EstimatedRTT $= (1-\alpha)*$ EstimatedRTT $+ \alpha*$ SampleRTT
  - Exponential weighted moving average
- **DevRTT** - Margin due to deviation from Estimate RTT
  - $(1-\beta)*$ DevRTT $+ \beta*$ (Sample RTT - Estimated RTT)
- **Timeout Interval** - EstimatedRTT + 4 * DevRTT
- $\alpha \approx \frac{1}{8}$, $\beta \approx \frac{1}{4}$

Fast retransmission

- Retransmit unacked segment **IMMEDIATELY** with **smallest seq #** after receiving 4 ACKs for same data

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Midterm tips

1. Bits are b and bytes are B (8 bits = byte)

06: IP Addressing

Function

1. **Forwarding** - Move packets from router input to output
2. Routing
   - Determine route taken by packets from source to destination
   - Plan trip from source to destination

Planes

1. Data plane
   - Local per router function
   - Determines how datagram forwarded to output port
2. Control plane
   - Network wide logic
   - Routing algorithms

Subnet

- Network formed by directly interconnected host
- Internal host can communicate without router
- Connect to external networks with router
- Same network prefix
- Valid subnet masks
  - 254, 252, 248, 240, 224, 192, 128

Special subnet

- Private IP addresses - 10.0.0.0/8, 172.16.0.0/12, 192,168.0.0/16
- Loopback address 127.0.0.0/8
- Broadcast 255.255.255.255/32

CIDR

- An IP address logically comprises two parts:

- CIDR: Classless InterDomain Routing
  - arbitrary length for the subnet portion (network prefix)
  - address format: a.b.c.d/x, where x is # bits in subnet portion of address
- Example 200.23.16.0/23

network part			host part
11001000	00010111	0001000**0**	00000000

Subnet mask

- Subnet mask is made by setting all network prefix bits to "1"s and host ID bits to "0"s.
  - example: for IP address 200.23.16.42/23:

	network prefix			host ID
IP address in binary	11001000	00010111	0001000**0**	00101010
Subnet mask	11111111	11111111	1111111**0**	00000000
Subnet mask in decimal	255.255.254.0			

  - used to determine which network an IP address belongs to (use bitwise AND operation)

Hierachial addressing

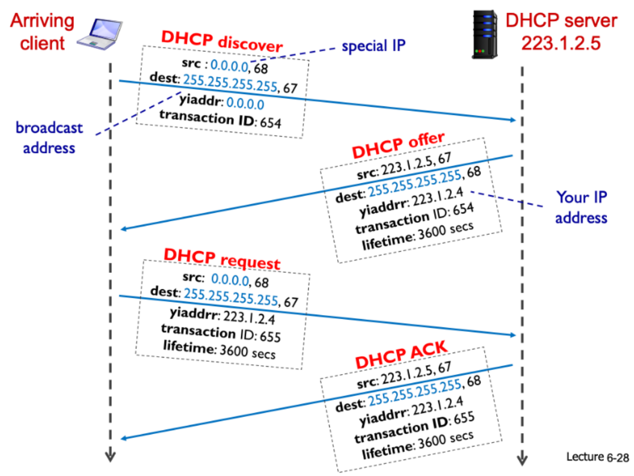
Route aggregation - Advertise routing information by grouping similar IP addresses
Longest Prefix matching

Routes to subnet with the greatest subnet portion of CIDR

DHCP

Dynamic Host Configuration Protocol - Lease IP address from network server when joining network

- Runs over UDP , Client #:68
- Router can act as DHCP
- Server #: 67, sends "DHCP ack" and "DHCP offer"
- Client #: 68, sends "DHCP request" and "DHCP discover"



What happens when multiple servers in network?

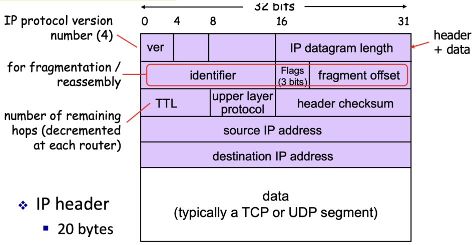
- Transaction ID specifies the DHCP connection

07. IP (cont)

IP fragmentation

MTU - Maximum Transfer Unit, maximum amount of data a link-level frame can carry

Fragmentation - Large datagrams that are broken up by routers



- Frag flag = 1 if there are more fragments from same segment, 0 = last fragment
- Offset - in units of 8-bytes specifying the offset of fragment wrt beginning of unfragmented IP datagram

Problem:
Not enough IP addresses to specify all devices

Network Address Translation

Using private IP addresses to map devices within an organisation and enable communication with the internet

- WAN (Wide area network) vs LAN (Local area network) - private vs public IP address

Implementation

- Routers replace source Ip address, port num of every outgoing datagram to NAT IP address and new port num
- Remember (in NAT translation table) the mapping of the source vs NAT info
- Replace NAT ip address and port num in destination fields of incoming datagrams with corresponding source IP/port stored in NAT translation table

Motivation

- No need to rent a range of public IP addresses from ISP: just 1 for NAT
- Hosts using private IP address can have variable IP addresses without notifying other hosts
- Can change ISP without changing addresses of host in LAN
- Hosts in Lan are not explicitly addressable and visible by outside (Security)

Challenges

- Difficult to have p2p applications across WANs

Routing

Hierachial routing on internet due to size and decentralised administration
Goal: Find the least cost to connect 2 hosts through routers

Intra Autonomous system(AS) routing

- Finds good path between routers within AS
- Single admin so no policy decision needed
- Performance impt in routing
- Protocols - RIP and OSPF

Inter AS routing

- Handles interfaces between AS
- Admin wants control over traffic routing
- Policy performance
- Protocol - BGP

Cost associated along each link based on various factors

- Inversely related to bandwidth
- Related to congestion of router
- Distance, $

Distance vector

Iterative Bellman-Ford computation

1. Starting router queries all the neighbours around it
2. Neighbours could have cached the distance vector to other routers
   (a) Router can send the table of distance vectors to the new router (routing tables)
3. No/incomplete distance vector record: WAit for (change in local link cost or message from neighbour)
4. Recomputes DV estimates using DV received from neighbour
5. Notify neighbours if DV to any destination changes

- Iterative and asynchronous: local iteration caused by
  - Local link cost changes
  - DV update messages from neighbours
- Distributed and self stopping
  - Each node notifies neighbours **only** when DV changes (if necessary)

Link state algorithms

- Periodically braodcasting link costs to each other
  - **All** routers have complete knowledge of topology and link cost
- Use Dijkstra algorithm to compute least cost path locally

Routing Information Protocol(RIP)

- Implements DV algorithm
- Uses hop count as cost metric
- **Self-repair** - If no updates from router for 3 minutes will assume router fails
- Exchanges routing table every 30s (UDP - 520)

Internet Control Message Protocol - ICMP

Used by hosts and routers to communicate network information

When TTL is 0, packet is discarded and ICMP error message is sent to datagram's source address

Error reporting Unreachable host/network/port/protocol

Echo Request

Headers Starts after IP header carried in IP datagrams

Type	Code	Description
8	0	echo request (ping)
0	0	echo reply (ping)
3	1	dest host unreachable
3	3	dest port unreachable
11	0	TTL expired
12	0	bad IP header

Ping

- Used to echo and test connection between different remote hosts
- Format: XX bytes from $client\ ip$: icmp_seq=X ttl=X time=Xms
- Time == time taken for ping command to execute completely
- ttl = number of hops between client and remote host
- icmp_seq = index of the datagrams sent to remote host (ascending order, default 0..6)

Traceroute

- Sends series of small packets with different TTL to display route to get to host
- Mechanism
  - TTL decrements every hop from initial TTL, X
  - Once TTL has reached 0, it will immediately reply with ICMP error message, addressed from the current device to the original source
  - Extract source IP address from the reply to get the device X hops away

08. Link layer

Motivation

- Sending data between N nodes via cable
  - Interconnect the N nodes via broadcast link
  - Every link has to be addressed
  - Define protocol so address to node on shared link (who speaks when and for how long)
  - Need to handle errors (Detection and reliability)
- **Communication channel** - Transmission medium of data signals
- Link layer sends frames transmitted between **physically adjacent** nodes over single link

Services

Framing Encapsulate IP datagram into frame by adding header and trailer

Link Access Control Coordinate which nodes send frames at specific time

Error detection Errors caused by signal attenuation or noise detected by receiver that signals sender for retransmission

Error correction Identifies and correct bit errors without transmission

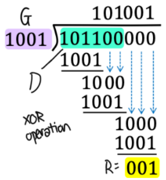
Devices

- Implemented in NIC (network interface card) or on a chip (ethernet card/wifi adapter)
- Adapters are semi-autonomous and implements link and physical layer

Error Detection and correction

Schemes

- Checksum (used in TCP/UDP/IP)
- Parity checking
  - Maintains a parity bit that value dependent on total number of 1s (where d+1 bit is even - even parity scheme)
  - Detect odd number of single bit errors
  - Probability of multiple bit errors is low (if errors are independent)
  - IRL errors are clustered so P(undetected errors) $\approx 50\%$
  - Can be made 2-D where d bits divided into i rows and columns
  - Parity bit along each row and column
  - Detects and corrects bit errors in data by comparing horizontally and vertically for errors
- Cyclic redundancy check (CRC)
  1. Decide generator G and number of bits to append, R
  2. For every info sent, append R bits of 9 to the back of the data, X
  3. Data sent is the largest value less than X divisible by G
  4. At receiver, data has error when it is indivisible by G
  - Easy to implement (Appending 0s and modulo 2 arithmetic xor)



  - Powerful error detection (detects all errors less than r+1 bit + $P(1-0.5^r)$ error $>$ r bits
  - AKA polynomial code

Multiple access links & protocol

Types of links

- Point-to-point
  - Sender-receiver connected by dedicated link (no need multiple access control)
  - Point-to-point protocol, serial line internet protocol
- Broadcast link (shared medium)
  - Multiple nodes in a shared broadcast channel
  - Channel broadcast frames to all nodes in the network
  - Wifi, satellite, bus ethernet

Motivation

- Collision happens when nodes receives two or more signals concurrently
- Need protocol to decide who, when and how long nodes get to communicate

Transmission algorithms

Goals

Collision free

Efficient Node can send at rate R(max transmission rate)

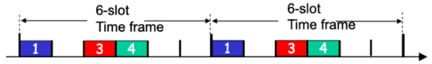
Fairness When M nodes want to transmit, each sends at an average rate R/M

Decentralised No dedicated driver/node to coordinate transmission

Channel partitioning

Time division multiple access (TDMA)

- Access channels in fixed length time slots each round
- Not very efficient as unused slots go idle (max throughput is R/N)
- Collision free, fair and decentralised



Frequency division multiple access (FDMA)

- Channel spectrum divided into frequency bands
- Node assigned to fixed band
- Same pros cons as TDMA

Take turn

Polling protocol

- Taking turn protocol that requires one node to be designated master node
- Master node polls each node in round robin to assign usage
- High efficiency, collision free, fair (proper algo for master node) but not decentralised

Token passing

- Token passed from one node to next sequentially
- Holds token $\iff$ want to transmit frames else forward token to next node
- Collision free, high efficiency (negligible overhead from token passing), fair and decentralised
- Might be disruptive (frame loss, system bugs) or node failure

Random access

Slotted ALOHA

- Split time into slots of equal size(synchronised among all nodes)
- Nodes transmit at full channel transmission rate **ONLY** at the beginning of a slot (randomly)
- Have collisions and partially efficient (maximum efficiency at 37% because collision and empty slots)
- Perfectly fair and decentralised

Unslotted ALOHA

- No time slots/synchronisation and transmit entire frame immediately
- Retransmit frames with probability p everytime there is collision until success
- Chance of collisions increase (max efficiency at 18%)

Carrier Sense Multiple Access (CSMA)

- Node decision to transmit is dependent of activity of other nodes attached to channel
- Node detects if other node is transmitting before transmission (defers transmission when channel is busy)
- Collisions may still occur due to propagation delay (does not detect other nodes transmissions immediately)

CSMA/Collision Detection

- Abort transmission immediately when collision detected
- Frame is retransmitted after random delay
- **Backoff algorithm** - Adapts retransmission attempts to estimated current load
  - More collisions implies heavier load - increase back-off interval with more collisions
  - Binary exponential backoff (Probability of resending = 2^{-m-1} after m collisions)
- Enforce minimum frame size to ensure collision is detected (account for propagation delay)
- Not collision free but efficient, fair and decentralised

09. Link layer LAN

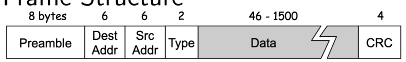
Media Access Control (MAC)

- Used to send and receive link layer frames
- Found on every NIC adapter
- If dest MAC == nic mac, extract datagram and pass to protocol stack — else discard frame without interruption
- 48 bits burnt in NIC ROM (read only memory) administered by IEEE
- First 3 bytes specify vendor of adapter
- Can run ifconfig <interface> on unix systems to get the information

Local Area Network (LAN) - Network that interconnects computers within geographical area

Ethernet

- Dominant wired Lan technology
- 802.3 Standards with different band speeds and physical layer media
- Frame Structure



- Note that hosts can use other network-layer protocol besides IP (Type field)
- Preamble used to synchronise receiver and sender clock rates
  - 7 bytes of 10101010 (0xAA)
  - 1 byte of 10101011 (0xAB) (start of frame)

Data delivery service

- Unreliable (no ACK or NAK to sender NIC)
  - Data in dropped frames will only be recovered if initial sender uses higher layer RDT
- Uses CSMA/CD with binary (exponential) backoff

Topology

- **Bus topology** popular till mid 90s (broadcast Lan)
- Cons:
  - Backbone cable (network fails if damaged)
  - Difficult to troubleshoot problems
  - Slow and not ideal for large networks
- **Star** prevelant today
- **Hubs** - physical layer device
  - Cheap with easy maintenance
  - Slow and may not be ideal for larger networks (Collision)
- **Switch** - Link layer device with no collisions
  - Store and forward packet switch

Switch

Function

- Examine incoming frame Mac address
- Selectively store and forward frame to one or more outgoing links (CSMA/CD)
- Transparent (hosts unaware of switches)
- Plug and play - no need configuration

Mechanism

- Nodes have dedicated direct connection to switch
- Ethernet protocol used without collision due to switching
- Maintains switching forwarding table for **selective forwarding** - only sending frames to the correct interfaces
  - Format *Mac address of host, interface to reach host, TTL*
- **Self learning** - Records sender/location pair in switch table everytime frames are received

1. Record incoming link, Mac address of sending host
2. Index switch table using Mac destination address
3. If entry found for destination,
   (a) if destination is in the same segment as source ? drop : forward frame on destination interface
4. else flood - forward on all interface except arriving interface

vs Router

- Checks MAC vs IP address
- Both store and forward
- Forward frame to link/broadcast vs Route computation

Address Resolution Protocol

- Maps IP address and MAC address of nodes in the same subnet
- TTL - time until address mapping will be forgotten

Resolution when in same subnet

1. Broadcast ARP query with B's ip as dest
   - Dest mac == FF-FF-FF-FF-FF-FF
   - All nodes in same subnet receives but only B replies
2. B replies to A with its MAC address
3. A cache B's IP-Mac address mapping in ARP table until TTL

Different subnet

1. Sends packet with correct MAC and IP but no response because different subnet
2. Resends packet with dest MAC == router MAC
3. Datagram removed passed to IP where routing calculations determine the subnet it is in (at router)
4. Router attaches dest MAC address again or use ARP with correct IPs
5. Receiver receives the information

IP vs MAC address

- 32 bits vs 48 bits
- Network-layer address used to move datagrams from src to dest vs link layer address to move frames over each link
- Dynamic assignment;hierachial vs permanent