Common Conversions

Time: 1 second = 1,000 milliseconds
Data Transfer: 1 gigabit (Gb) = 1,000 megabits (Mb) → 1 megabit (Mb) = 1,000 kilobits (Kb) → 1 kilobit (Kb) = 1,000 bits (b)
Storage: 1 gigabyte(GB) = 1,024 megabytes (MB) → 1 megabyte (MB) = 1,024 kilobytes (KB) → 1 kilobyte = 1,024 bytes

Section 1: Transport Layer

Transport Layer: Responsible for delivering data to applications on host computers.

Multiplexing / De-multiplexing

Multiplexing: Multiple data streams from different sources are combined and transmitted over a single shared medium.

- Handles data from multiple sockets, adding transport header which is used in de-multiplexing.

De-multiplexing: At the receiving end, the reverse occurs, separating data streams from the single channel and routing to corresponding receivers / destination.

- Use the transport header info to deliver received segments to the correct socket.

Connectionless De-multiplexing (UDP)

- IP datagrams with the same dest port but different source IP/port will be directed to the same socket at the dest.

Connection-orientated De-multiplexing (TCP)

- Receiver uses all four values of TCP 4-tuple to direct segments to the appropriate socket.
- A server host may support simultaneous TCP sockets.
- Web servers have different sockets for each client. Non-persistent HTTP will have a different socket for each request.

UDP: User Datagram Protocol

UDP is a communications protocol used primarily for establishing **low-latency** and **loss-tolerating** connections between apps.
It is **connection-less**, where each UDP segment is handled independently of others.
UDP Header (8 bytes): SRC PORT # | DEST PORT # | LEN (BYTES) | CHECKSUM | PAYLOAD
UDP Checksum

- Treat segment content + header fields as a sequence of 16-bit integers.
- <u>Checksum</u> = binary addition of segment contents (sum) → invert the bits to get checksum (complement of sum)
- <u>Sender</u> puts checksum value into UDP checksum field.
- <u>Receiver</u> adds all segment content with the checksum. Result should = 1111 1111 1111 1111, else there are errors.

Reliable Data Transfer (RDT) – STOP AND WAIT PROTOCOLS

With RDT, transferred data is **NOT CORRUPTED | NOT LOST | DELIVERED IN ORDER**. TCP offers this service model to apps.

RDT 1.0 – Transfer over a perfectly reliable channel (not a realistic model)

- All packet flow is from sender → receiver, no need for receiver-side to provide feedback to sender.
- Assume the receiver is able to get data as fast as the sending of data, thus no need for flow/congestion control.

RDT 2.0 – Transfer over a channel with bit errors (more realistic model)

- In this model, we assume packets can be corrupted.
- Recover from errors through **ARQ: Automatic Repeat Requests Protocols**.

<u>ARQ - Stop-and-Wait Protocol:</u> Sender sends packet and waits for an ACK or NACK. ACK = not corrupted | NACK = corrupted.
Sender can't receive more data from upper layer while waiting.
Flaw with Stop-and-Wait: ACK/NACK can be corrupted themselves.
Solution to flaw: Number packets with a sequence number #0 or #1

RDT 2.1 – Protocol includes sequence numbers #0 #1 to track expected packets

- Sender: Check ACK/NACK + Remember whether expected packet = seq #0 or #1
- Receiver: See if packet is a duplicate (duplicate if expected seq# != received seq#).

RDT 2.2 – NAK-free protocol

- Same as 2.1 but only using ACKs. Instead of sending NACK, send the same ACK for the last successfully received packet.
- E.g. Server PKT_0 → Client | Client ACK_0 → Server | Server PKT_1 → Client | **Client ACK_0 → Server [NACK]**

RDT 3.0 – Transfer over a channel with bit errors and loss

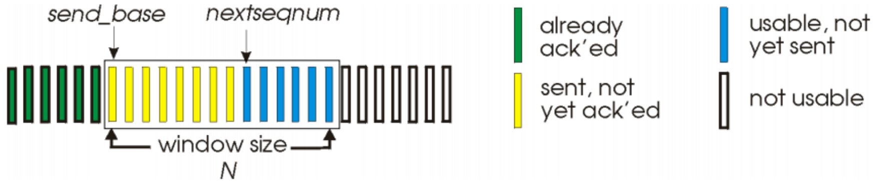
- New assumption: In addition to bit errors, the channel can also lose entire packets.
- New concerns: (1) How to detect packet loss? (2) What to do when packet loss occurs?
- **Time-Based Packet Retransmissions** that can interrupt the sender after an amount of time waiting for an ACK expires.
  The sender will need to:
  (1) Start the timer after each packet
  (2) Respond to timer interrupt – take appropriate actions i.e. retransmit packet
  (3) Stop the timer
- Retransmission is an all-in-one solution: doesn't matter if packet is LOST or LARGE DELAY. Seq #s will handle duplicates.

Reliable Data Transfer (RDT) – PIPELINED PROTOCOLS: Go-Back-N, Selective Repeat

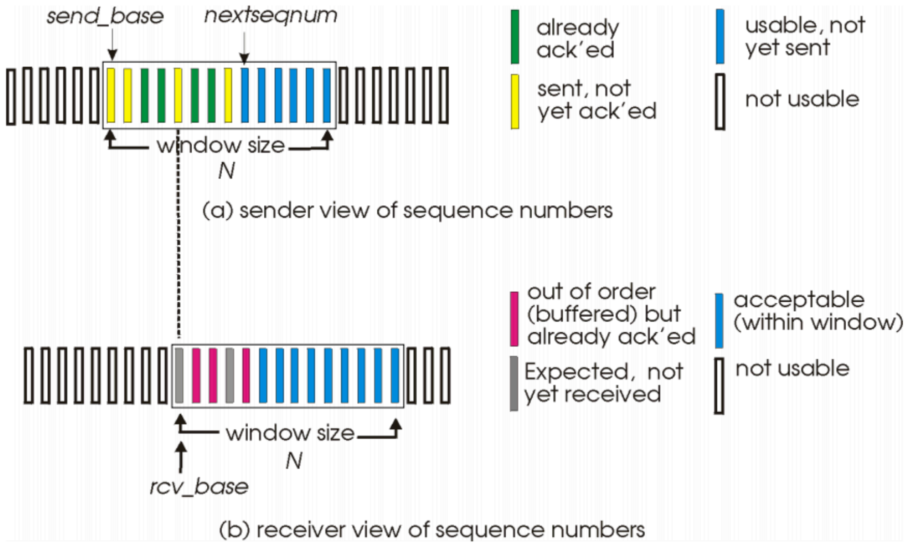
Pipelined protocols allow for multiple “in-flight”, un-acknowledged packets and increases utilisation.

Go-Back-N (GBN) Sender continues to send pkts specified by window-size N without receiving ACKs.	**Selective Repeat (SR)** Receiver individually ACKs all received pkts. Buffers packets for eventual in-order delivery to the upper layer.
• Sender window size N of consecutive un-ACK’d packets. • On timeout/loss of packet P: Receiver – discards P + resend ACK of last successful pkt. Sender - retransmit all pkts of higher seq# in window. • On success: Advance send_base	• Sender window size N consecutive seq #s. • On timeout/loss of packet P Receiver: buffer the out of order pkt. Sender: retransmit P only • On success: Send P + all following in-order packets Advance send_base

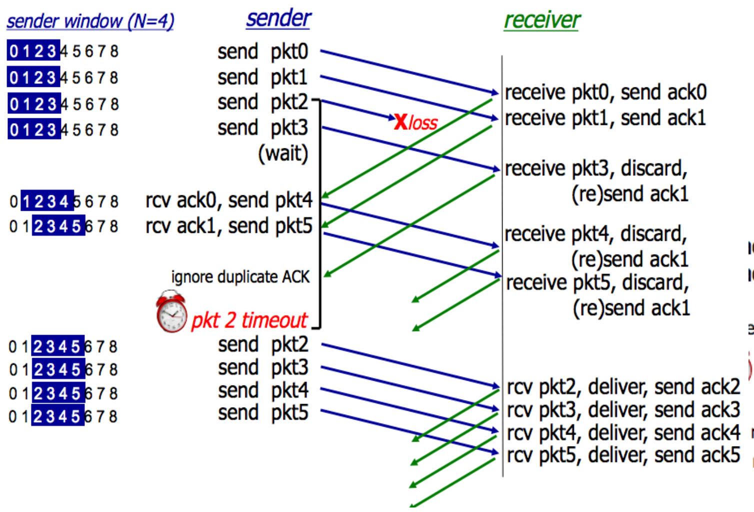
GBN Sender Window:



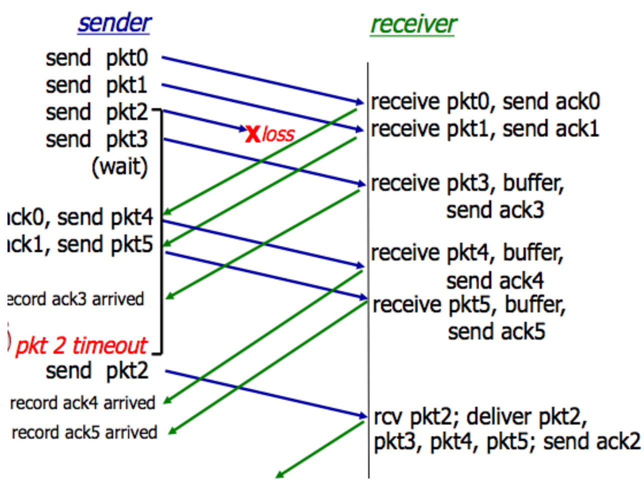
Select Repeat Sender / Receiver Windows:



Go-Back-N



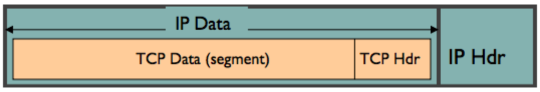
Selective Repeat



Transmission Control Protocol (TCP) – Segment structure

TCP Header (20 bytes): UDP fields + seq#, ack#, receiver window #bytes, connection establishment + teardown, options.

TCP Packet



IP Packet: No bigger than **Max Transmission Unit (MTU)**
TCP Data/Segment: No more than **Max Segment Size (MSS)**

MSS = MTU – IP Header – TCP header

TCP – Process, Timer + Retransmissions

TCP Sender / Receiver Process:

- Sender: Sends packet of SEQ# = X. Packet len = B bytes [X , X+1 , X+2 . . . X + B–1]
- Receiver: If data prior to X has already been received, send ACK# = X+B
  X+B = next expected seq # from the next packet.
  If highest-order byte received is Y, where Y+1 < X → resend ACK Y+1
- Next Seq# = ACK#

What else can TCP do?

- Receivers can buffer out-of-sequence packets like Selective Repeat / NOT drop out-of-seq packets.
- Senders can maintain a single retransmission timer like Go-Back-N and retransmit on timeout.

Set up TCP timeout by choosing a value > RTT.

- Choose value too short: premature timeout, unnecessary retransmission
- Choose value too long: slow reaction to segment loss and lower throughput for connection

1. Measure EstimatedRTT:

Exponential Weighted Moving Average

$$\text{EstimatedRTT}_{\text{CURR}} = (1 - a) * \text{EstimatedRTT}_{\text{PREV}} + a * \text{SampleRTT}_{\text{RECENT}}$$

- SampleRTT
  - Time measured from segment transmission until ACK receipt (ignoring retransmissions)
  - Current value of RTT
- Typical value of a = 0.125

2. Measure timeout interval: EstimatedRTT + "Safety Margin"

RTT Deviation is calculated by

$$\text{DevRTT} = (1 - b) * \text{DevRTT} + B * |\text{SampleRTT} - \text{EstimatedRTT}|$$

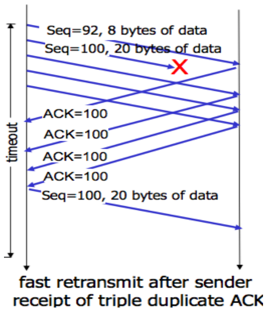
- Typical value of b = 0.25

Timeout Value is calculated by

$$\text{Timeout Interval} = \text{EstimatedRTT} + 4 * \text{DevRTT}$$

- 4 * DevRTT = The "Safety Margin"

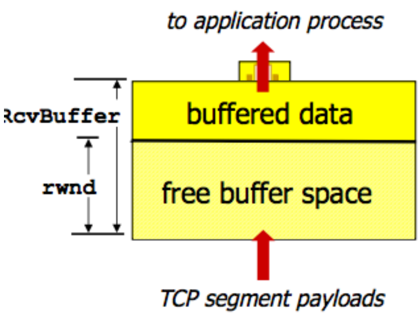
TCP – Fast Retransmission



TCP has a Fast Retransmissions feature that uses duplicate ACKs to trigger early retransmission.

- If sender receives 3 duplicate ACKs for the same data, resend the un-ACK'd data with the smallest sequence #.
- Timeout periods are often long, so there is a long delay before resending lost packets. No need to wait for timeout.

TCP – Flow Control



TCP Flow Control is where the receiver controls the sender, so the sender won't overflow the receiver's buffer by transmitting too much, too fast.

Receiver Advertised Window (RWND): Advertises available recv buffer space in the RWND value in TCP header.

Sender limits the amount of un-ACK'd data to receiver's RWND value.

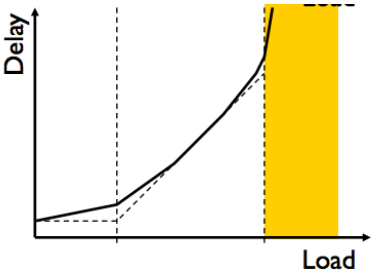
TCP – Connection Management

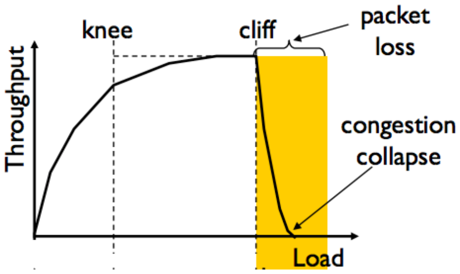
C Establishment: (1) SYN → (2) SYN-ACK → (3) ACK + DATA → Data exchange
C Teardown: Data exchange → (1) FIN → (2) ACK-FIN → (3) ACK → (4) WAIT / Retransmit ACK → (4) CLOSE CONNECTION

RST: Reset Flag. Possibly because application has crashed on one end, or socket is closed, or there is a firewall.

TCP – Congestion Control

Congestion Control is needed if a network node/link/router is taking in more data than it can output, leading to collapse.
Congestion Collapse: Throughput starts to drop to zero, delays approach infinity.





Knee Point
Throughput increases slowly
Delay increases really fast

Cliff Point
Throughput begins to drop to zero
(Congestion collapse)
Delay approaches infinity

CWND: Congestion Window i.e. how many bytes can be sent without overflowing routers?

- Sender varies the window size to control the sending rate.

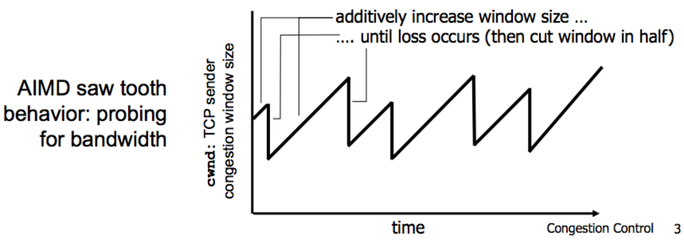
TCP sending rate =~ CWND / RTT bytes per second.
Sender-Side Window: minimum { RWND , CWND }
Basics of sender rate adjustment:

- Upon receiving ACK → increase rate
- Upon detection of loss → decrease rate

(1) Bandwidth Discovery with Slow Start (SS)

- When connection begins, initial rate is slow (for safety) then increase exponentially until the first packet loss event.
- Initial CWND = 1 MSS → Double CWND every RTT or alternate Increment CWND for every ACK received

(2) Additive Increase Multiplicative Decrease (AIMD)



- Slow start gave an estimate on available bandwidth. Now we want to track variations of this bandwidth via. probing.
- Additive Increase: Sender increase CWND / transmission rate, probing until a loss event occurs
- Multiplicative Decrease: Cut CWND in half after a loss occurs

Slow-Start Threshold (SSThresh) is used to determine when a sender should stop slow-start and start AIMD.

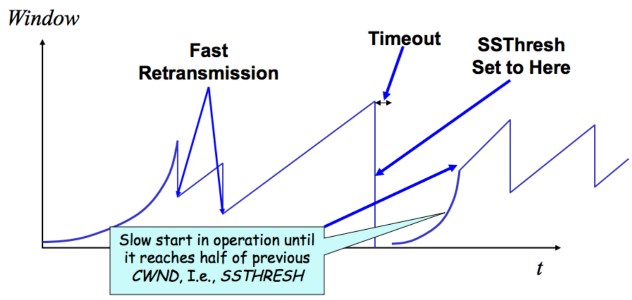
- SSThresh is initialised to a large value. On timeout, SSThresh = CWND/2.
- If CWND < SSThresh = Slow-Start
- If CWND > SSThresh = AIMD / Congestion Avoidance

Congestion Control Rate Increases:

- Slow-Start: CWND += MSS
- Congestion Avoidance/AIMD: CWND += MSS/CWND

Congestion Control Rate Decreases:

- DupACKs:
  SSThresh = CWND/2
  CWND = CWND/2
- Timeout/Loss Event:
  SSThresh = CWND/2
  CWND = 1 MSS

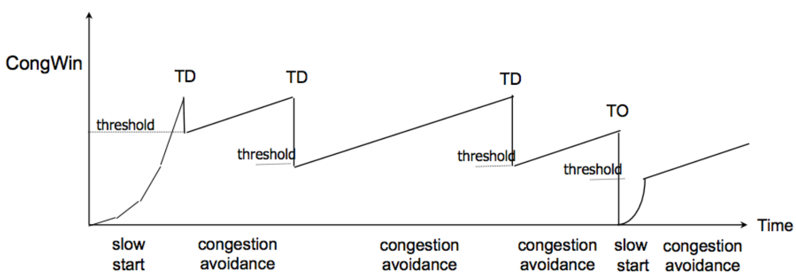


Slow-start restart: Go back to CWND = 1 MSS, but take advantage of knowing the previous value of CWND

TCP - Flavours

TCP Tahoe: CWND = 1 on DupACK and Timeout
TCP Reno: Same as above.
TCP New-Reno: TCP Reno + improved fast recovery

TD = Triple Duplicate ACKs
TO = Timeout



Section 2: Network Layer

The Network Layer focuses on packet forwarding through intermediate routers / networks from source host to destination.

- Service Model: Guaranteed Delivery, Guaranteed Minimum Bandwidth, In-Order Deliveries

Forwarding: Move packets from router's input to appropriate router output. (i.e. station platforms)

- Forwarding Table determines which output link to forward the packet to. **Entry = { K=Header Value | V=Output Link }**
- Generalised Forwarding: Forward packets based off any set of header-field value, using **Longest Prefix Matching**.
  For a given Destination IP Address, use the longest IP prefix that matches the address.
  -> STEP 1: Find the IP ranges / entries in forwarding table which match with the Dest. Address IP.
  -> STEP 2: Choose the IP range / entry with the longest matching prefix address.

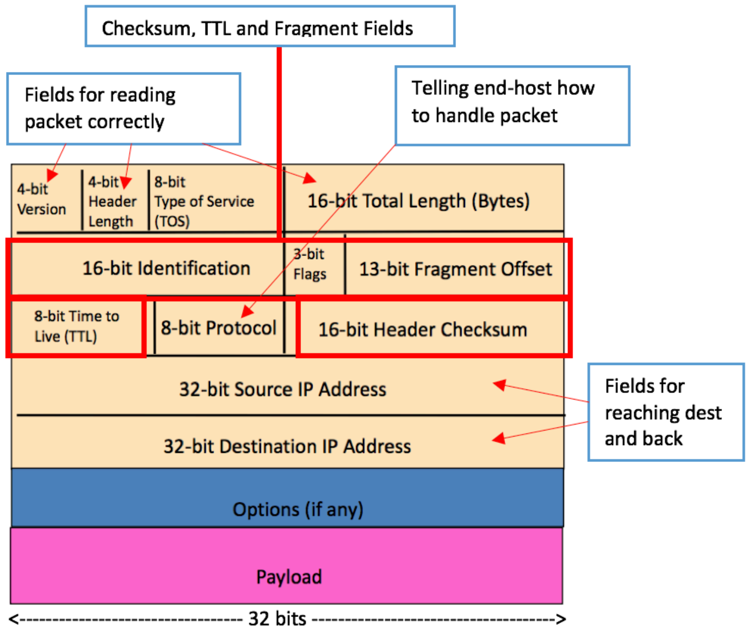
Routing: Determine the route taken by packets from source to destination. (i.e. start → many stations → destination)

The **Data Plane** refers to the functions that determine how packets are forwarded from a router input to its output port.
The **Control Plane** refers to the functions that determine how a packet is routed among routers in the end-to-end path.

- Per-Router Control Plane: Individual routing algos in each router interact in the control plane.
- Logically Centralised Control Plane / Software-Defined Networking (SDN): A distinct controller interacts with local Control Agents (CA's) / centralised servers.

IP – Internet Protocol

IP Packet Structure: 20 bytes of Standard Header, then Options



Version Number (4 bits)

- Indicates version of the IP protocol
- **"4" = IPv4 | "6" = IPv6**

Header Length (4 bits)

- Number of 32-bit words in the header
- Typically "5" for **20-byte** IPv4 header
- Can be more with IP options

Total Datagram Length (16 bits)

- # bytes in the packet
- **Max size = 65,535 bytes (2^{16} – 1)**

Protocol (8 bits)

- Identifies the upper-layer protocols
- Important for de-multiplexing at receiving host

E.g. **Protocol=6 → TCP** | **Protocol=17 → UDP**

Payload (variable length)
Typically a TCP or UDP segment.

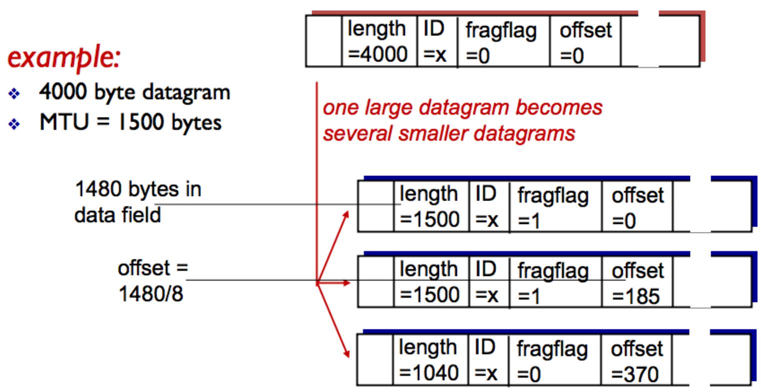
Time To Live (TTL) (8 bits): Max number of remaining hops.

- Value decremented for each hop → packet discarded if value = 0 and a "time exceed" message is sent to the source.
- This mechanism will prevent packet forwarding loops.

IP Fragmentation Reassembly (Frag offset – 13 bits)

- A large IP datagram is divided / fragmented within the network, as datagram can't exceed the **Max Transmission Unit**.
  - **MTU** = size of the largest network layer data that can be sent in a single network transaction.
- One datagram becomes several, which are reassembled at the final destination.
- Fragmentation Header Bits are used to identify the ORDER of the fragments.

The offset is the address or the locator from where the data starts in the original payload. The system/router takes the payload and divides it into smaller parts, keeping track of this offset so that reassembly can be done later.



20 byte header in each packet.

Original packet (4000 bytes)
= 3980 payload + 20 header

Frag pkt #1 (1500 bytes) | Offset = 0 (start of OG data)
= 1480 payload + 20 header

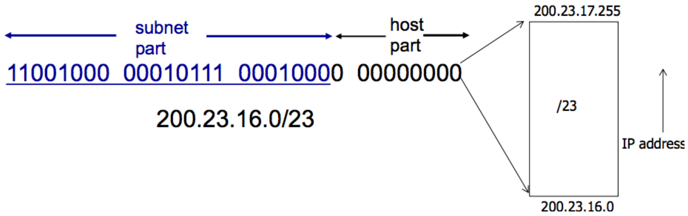
Frag pkt #2 (1500 bytes) | Offset = 1480 / 8 = 185
= 1480 payload + 20 header

Frag pkt #3 (1040 bytes) | Offset = 2960 / 8 = 370
= 1020 payload + 20 header

Original 3980 bytes = 1480 + 1480 + 1020

IPv4 Addressing

Interface is a connection between a host/router at the link-layer, typically with multiple interfaces per host/router.
IP Addresses are associated with each interface.
Subnets are groups of IP addresses that form multiple smaller divisions of a larger/major network. E.g. Companies use subnets.
Classless Inter-Domain Routing (CIDR) – Today's Internet Addressing



Address format: **[255 . 255 . 255 . 255 / X]**

where X = #bits in subnet portion of address.

CIDR uses hierarchical address allocation: addresses are allocated in continuous chunks / prefixes.
What happens when an organisation wants to switch from ISP#1 → ISP#2?

- The organisation keeps its IP address block, ISP#1 and ISP#2 will continue to advertise their address blocks.
- ISP#2 will also advertise the org's more specific address block.
- Routers from the internet will know which ISP to route packets towards, using **Longest-Prefix-Matching.**

A **subnet mask** separates the IP address into the NETWORK PART and HOST PART of the address. Example:

- IP: 200.23.16.2/24 | Subnet Mask: 11111111 11111111 11111111 00000000 (24 most-sig bits set to 1) & IP
- Network: 200.23.16.0 | Host: 0.0.0.2 (remainder bits after MASK & IP)

How many IP addresses belong to the subnet 128.119.254.0/25?

- IP = 10000000 01110111 11111110 00000000
- Subnet Mask = 11111111 11111111 11111111 10000000
- Host has **0b1111111 (127) addresses to use**. Therefore RANGE = 128.119.254.0 to 128.119.254.127

Dynamic Host Configuration Protocol (DHCP)

DHCP allows a host to dynamically obtain its IP from a network server when it joins the network.

- Host can receive same IP address each time it connects to the network or be assigned a temporary IP addr each time.
- Host can renew its lease on address | Allows reuse of an address (only holds addr when "connected" to network)

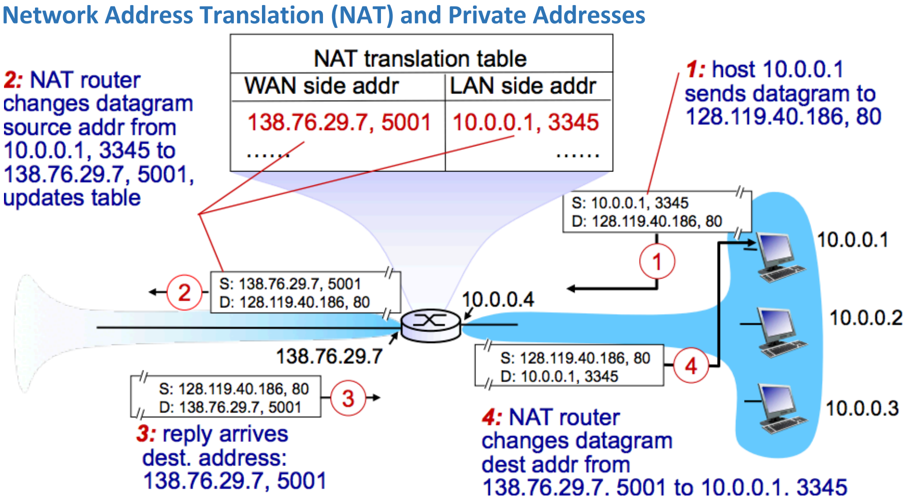
DHCP Steps:

1. Host broadcasts DHCP Discover message
2. DHCP server responds with DHCP Offer message
3. Host requests IP address with DHCP request message
   → Client requires: **[IP] [Address of 1st hop router] [Address of DNS server]**
   → DHCP request encapsulated in UDP → encapsulated in IP → encapsulated in Ethernet frame
   → Broadcast Ethernet frame on LAN
   → Frame is received at the router running the DHCP server
   → Ethernet frame de-multiplex to IP → de-muxed to UDP → de-muxed to DHCP request.
4. DHCP server sends the address with DHCP ACK message
   → DHCP server formulates an ACK: **[client IP] [Address of 1st hop router] [Address + Name of DNS server]**
   → Encapsulate ACK in frames and send to client. Client demuxes back to DHCP
   → Client will now know its **IP Address, Address of 1st hop router, Address + Name of DNS server**.

More about DHCP:

- The **MAC** address is used to identify clients. DHCP server can be configured to accept a list of specified MAC addresses.
- DHCP loopholes: DoS attack by exhausting pool of IP addresses in LAN | Masquerade as a DHCP server

Network Address Translation (NAT) and Private Addresses



NAT allows a device such as a Router, to **act as an agent** between the Internet (public network) and the local (private) network.

Only a single IP address is required to represent an entire group of computers.

Advantages of NAT:

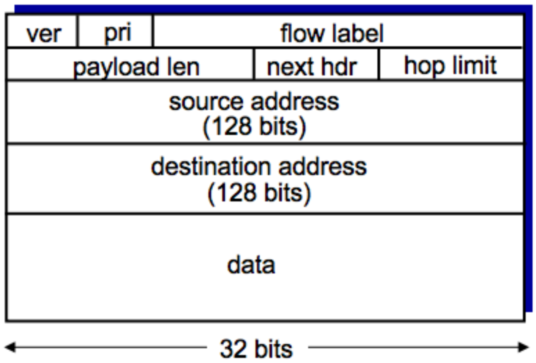
- A range of addresses are not needed from an ISP. Use a single IP address to represent all devices in the LAN.
- Can change addresses in the LAN without having to notify the outside world
- Can change ISP without changing addresses of devices in the LAN.

Disadvantages / Issues of NAT:

- Controversial as routers should only process up to the Network Layer, thus violating end-end agreement.
- NAT modifies port # and IP address → **requires recalculation of TCP and IP checksum**
- Some apps embed IP / port # in their message protocols. Some encrypt the IP / port# fields. How to decrypt on NAT?!
- NAT traversal problems:

IPv6 Addressing

Initial Motivation: 32-bit address space will run out soon.
Additional Motivation: IPv6 header format helps speed up packet processing / forwarding.

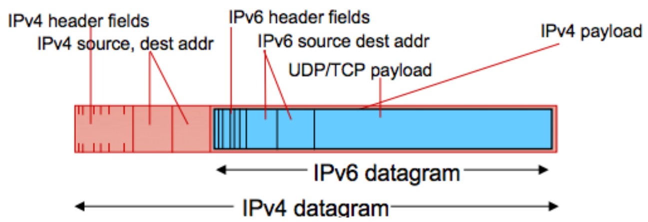


IPv6 Datagram Format:

- Fixed-len 40-byte header, no fragmentation allowed
- **Priority**: identify priority among datagrams in flow (traffic class)
- **Flow Label**: identify datagrams in the same flow
- **Next Header**: Identify the upper layer protocol for data

Changes from IPv4

- **Checksum** removed entirely to reduce processing time at each hop
- **Options** allowed, but outside of the header, indicated by "next header" field
- **ICMPv6**: new version of ICMP



Tunnelling: IPv6 datagram is carried as a payload in IPv4 datagram among IPv4 routers.

← See diagram

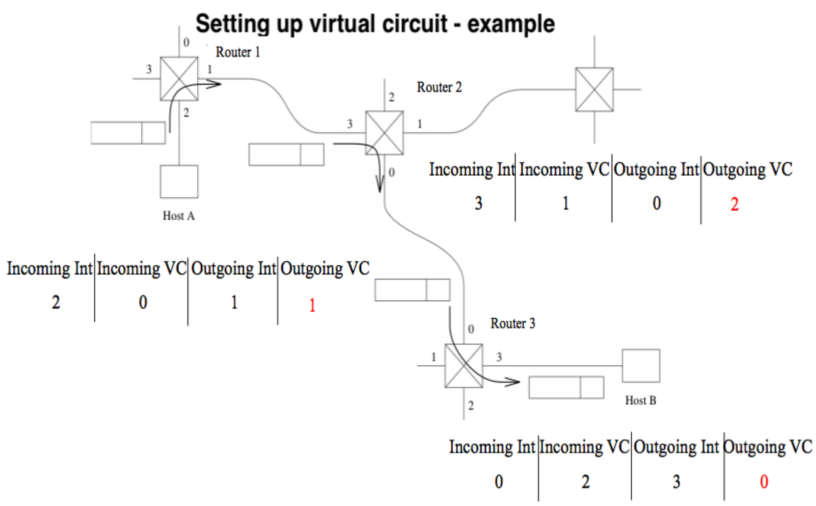
Virtual Circuit Network

Datagram Network: Provides a connection-less network layer service.

- NO SETUP | NO KEEPING STATE ABOUT CONNECTION | PKTS FORWARDED USING DEST HOST ADDRESS ONLY.

Virtual Circuit Network: Provides a connection-based network layer service.

- Signalling protocols used to setup and maintain connection of virtual circuits. Not used in today's internet.

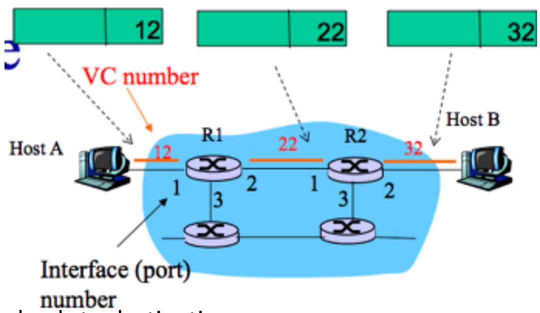


VC SETUP:

1. Source
   - Sends setup msg with dest addr.
2. Intermediate routers
   - Choose VC # (from lowest = 0)
   - Determine outgoing interface from routing table.
   - Create entry in VC table.
   - Forward setup to next hop.
3. Setup reaches dest:
   - Dest chooses incoming VC #
   - Chosen incoming VC # = outgoing VC # of all routers except last.
   - Send ACK along the reverse path to SOURCE
4. Acknowledgement
   - Intermediate routers complete their VC tables

Forwarding Table in a Router: switches/routers maintain connection state info

Incoming interface	Incoming VC #	Outgoing interface	Outgoing VC #
1	12	2	22
2	63	1	18
3	7	2	17
1	97	3	87
…	…	…	…



FORWARDING = Within Router (input → output link) **ROUTING =** Make sure hop leads to destination.

Routing Protocols: Intra-Domain Routing (Link State / Distance Vector) and Inter-Domain Routing

A routing protocol determines the end-to-end path of packets through the network.
The forwarding table determines the local forwarding at this router.
Autonomous Systems (AS) or **Domains** is a region of a network under a single admin authority. E.g. an ISP is an AS.
Internet routing works as two levels:

1. **Intra-Domain Routing Protocol:** AS Establishing routes within its own AS / domain.
    a. Single admin, so no policy decisions are needed. Performance > Policy.
    b. Examples of intra-domain routing:
    <u>Link State</u> → **Open Shortest Path First (OSPF)**
    <u>Distance Vector</u> → **Routing Information Protocol (RIP)**
2. **Inter-Domain Routing Protocol**: AS Establishing routes between other AS / domains
    a. Admin wants control over routing in network + who routes through its network. Policy may > Performance.
    b. Examples of inter-domain routing:
    <u>Path Vector</u> → **Border Gateway Protocol (BGP)**

Link State Routing (Global): All routers have the complete topology and maintain / know the cost of each link in the network.

- <u>How it works</u>: (1) **Link State Advertisement (LSA) Flooding** (2) **Path calculation with Djikstra's**
    - When receiving a new Link State msg, the router forwards it to all neighbours except one that sent the msg.
    - Routers keep a local copy so they don't forward previously seen LSA's.
    - Eventually, each node learns the entire network topology + can use Djikstra's to compute shortest path.
- Eventually, each node learns entire network.
- Characteristics
    - Connectivity / cost changes are flooded to all routers in the network.
    - Converges quickly (less consistency, looping)
    - Limited network sizes, otherwise it will be too costly.
- Challenges:
    - <u>Packet Loss / Out-of-order packets</u> (solved with ACKs, Retransmissions, Seq Numbers, TTL for packets)
    - <u>Scalability</u>: # Messages to flood **O(N*E)** where N = #nodes E = #edges | Djikstra's **O(N^2)**
      # entries in topology database **O(E)** | # entries in forwarding table **O(N)**
    - <u>Transient Disruptions / Infinite Loop problems</u>: Inconsistent link-state database, as some routers know about failures before others. Shortest path is not always consistent, which can cause transient / infinite loops.
    - <u>Oscillations</u>: Costs can change around continuously. For given new costs → new route → new costs and so on.

Distance Vector Routing (Decentralised): Routers only know its neighbours + link cost to neighbours.

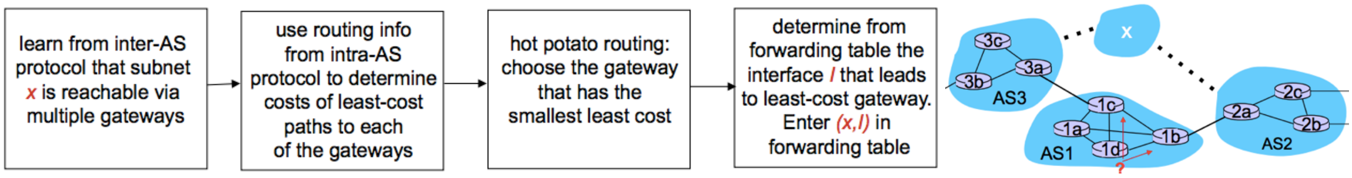
- <u>How it works</u>:
  (1) Each router initialises its DV table based on link costs to immediate neighbours + sends its DV to the neighbours.
  (2) Neighbours process the DV and repeats STEP #1 until the iterative process converges to a set of shortest paths.
  (3) Each node then waits for changes in their local link cost or msg from neighbours.
  (4) If change occurs → recompute costs in DV and notify neighbours if anything changes.
- Initial state: best 1-hop paths | one simultaneous round = best 2-hop | k simultaneous rounds = best (K+1)-hop paths
- Characteristics:
    - Cost changes are <u>iterative</u>, exchanges info from neighbour to neighbour.
    - Requires <u>multiple rounds</u> to converge
    - <u>Scales</u> to large networks.
- **Counting to Infinity Problem ("bad news travels slowly")**: Usually occurs when a node becomes broken.
    - Because of a broken link, nodes keep incorrectly updating their DV table and increasing cost for the broken link until the updates slowly propagates through the network and eventually reaches infinity.
- **Poisoned Reverse Rule** is a method to avoid the Count to Infinity Problem.
    - Routers actively advertise certain links as unreachable (cost=infinity). However, this will significantly increase the number of routing announcements made in the network.

Comparison of Link State vs. Distance Vector

	Link State	**Distance Vector**
Message Complexity	N nodes / E edges = O(N*E) messages sent	Exchange between neighbours only.
Speed of Convergence	O(N^2) algorithm | relatively fast	Convergence time varies Count to Infinity / Routing Loops may occur
Robustness	LS node can advertise incorrect LINK cost. Each node computes only its own table.	DV node can advertise incorrect PATH cost. Each node's table is used by others, errors propagate through the network.

Inter-Domain Routing Protocol

- Gateway Routers are the “edge” of an AS which links to another AS’s gateway in the internet.
- How forwarding works between different AS networks:
  SCENARIO: Router in AS1 needs to determine which gateway to forward a packet to, so it can reach subnet X.



Section 3: Link Layer

The **Data-Link-Layer** has the responsibility of transferring a datagram from one node to a physically adjacent node over a link.
Nodes: Hosts and routers
Links: Communication channels connecting adjacent nodes i.e. Wireless, Wired, LAN
Layer-2 Packet: Frame encapsulating a packet/datagram

Link Layer Services

Framing, link access: Encapsulate datagram into frame, add header/trailer, providing channel access + MAC address to identify.
Reliable delivery between nodes: Low bit-error in some links i.e. Fiber. High error rates in wireless links.
Flow Control: Pacing between adjacent sending and receiving nodes
Error Detection: Errors caused by signal attenuation (reduction of signal strength during transmission) and noise.

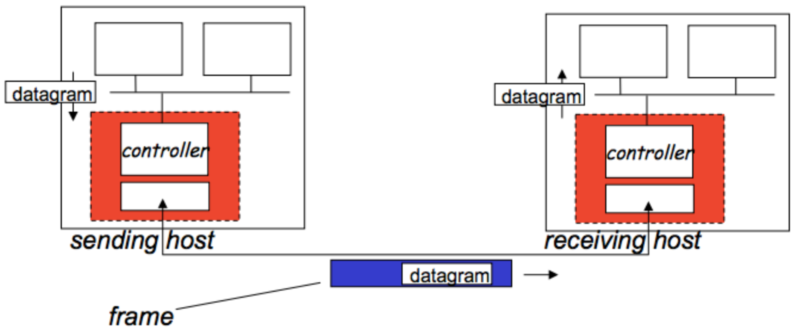
- Receiver signals for retransmission or drops the frame.

Error Correction: Receiver identifies and corrects bit-errors without needing retransmission.
Half-Duplex and Full-Duplex: DUPLEX = ability for two devices to communicate at the same time.

- Wireless WiFi = half-duplex | Wired LAN = full-duplex

Where is the link layer implemented?

- In a **Network Interface Card (NIC)** embedded on each and every host. E.g. Ethernet card, 802.11 card.
- NIC is a combination of hardware, software, firmware.



Adaptors Communicating:
Sending side:

- Encapsulates datagram in frame
- Adds error checking bits, rdt, flow control etc.

Receiving side:

- Looks for errors, rdt, flow control
- Extracts datagram, passes to upper layer at receiving side.

Link Layer Error Detection: Parity Check Method

An error detection method is using **Parity Check Method**. In practise, bit errors occur in bursts.
With Parity Checking, we are willing to trade computational complexity for space efficiency.

- We make detection checking more complex, but without the need to input lots of extra data for checking.

Simple Parity – Sender: For every d_bits add a parity bit.

- Parity Bit = 1 for an odd number of one’s.
- Parity Bit = 0 for an even number of one’s.

Simple Parity – Receiver: For each block of size d_bits, calculate the parity bit and compare with the sent data.
Simple Parity Cost: 1 extra bit for each d_bits.

Example: d = 7 | Result: 001011011101100001100101

Message chunk	Parity bit
0010110	1
1101100	0
0110010	1

Two Dimensional Parity: Compute parity on columns as well as rows.

	Message chunk	Parity bit
	0010110	1
	1101100	0
	0110010	1
Parity byte:	1001000	0

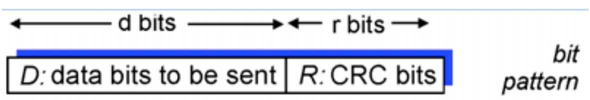
bits

	0	0	1	0	1	1	0	1
	1	0	1	0	0	0	1	0
	1	0	0	1	0	1	1	0
	1	1	1	0	1	1	0	1
Parity byte	1	1	1	1	1	1	0	0

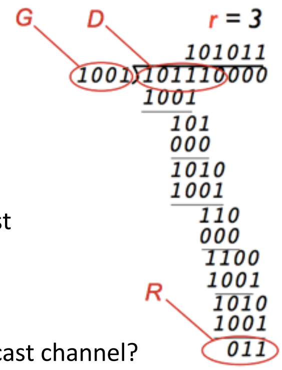
- Exactly ONE-BIT has been flipped in the example: which one is it?

Link Layer Error Detection: Cyclic Redundancy Check (CRC) method
CRC is another error-detection method, widely used in Ethernet, 802.11, WiFi, ATMs.

Goal: choose **R** number of CRC bits && choose G a generator such that:



1. **<D,R>** is exactly divisible by **G**
2. Receiver also knows the value of **G** and divides <D,R> by G to check for errors.
3. If remainder = 0 → NO ERRORS
   If remainder != 0 → ERROR DETECTED



Multiple Access Links / Multiple Access Protocol
There are two types of links:
(1) Point-to-Point: Single wire e.g. Link for dial-up access | Link between Ethernet switch/host
(2) Broadcast (shared wire or medium): e.g. 802.11 Wireless LAN

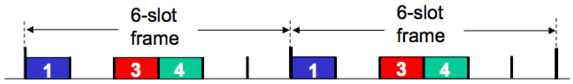
Broadcast links have a **Multiple Access Problem**:
- How to coordinate access from multiple sending/receiving nodes to a shared broadcast channel?
  - **Collision** occurs if nodes receive two or more signals at the same time.
  - All frames in the collision are lost + broadcast channel is wasted during the collision interval.

Multiple Access Protocol is used to determine how nodes share a channel + determine when they can transmit.
- Communication about channel sharing must use the channel itself. i.e. no outside channels allowed for coordination.
- Used by both wired and wireless LAN and satellite networks.

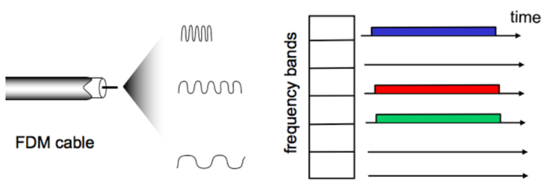
An ideal Multiple Access Protocol: Given a broadcast rate of R bps
1. When one node wants to transmit, it can send at rate R
2. When N nodes want to transmit, it can send at rate R
3. Fully decentralised: no special node to coordinate transmissions
4. Simple

MAC Protocols
The **Medium Access Control** is the lower sub-layer of the data-link-layer, which provides addressing and channel access control mechanisms that make it possible for several nodes to communicate within a multiple access network over a shared medium.

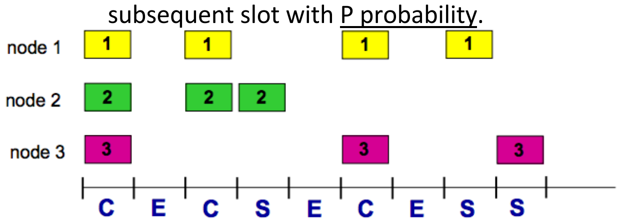
Three classes:
(1) Channel Partitioning Protocols: Divide channel into smaller "pieces" (time slots, frequency, codes)
TDMA (Time Division Multiple Access)
- Each station gets a fixed length slot (len = packet transmission time) in each round. Unused slots go idle.
- Example of TDMA: 6 station LAN, slots 1-3-4 have a packet, slots 2-5-6 are idle



FDMA (Frequency Division Multiple Access
- The channel is divided into frequency bands, where each station has an assigned frequency.
- Unused transmission time in frequency goes idle.
- Example of FDMA: 6 station LAN, slots 1-3-4 have a packet, slots 2-5-6 are idle



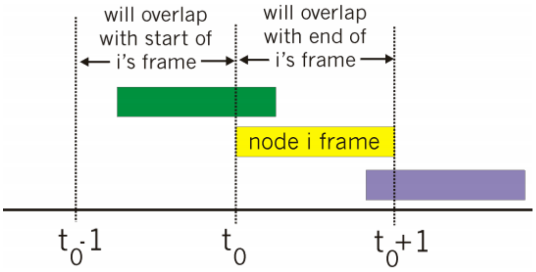
(2) Random Access: Channel not divided, allow collisions to occur and recover from collisions.
Slotted ALOHA: When a node obtains a frame, it transmits in the next slot.
- NO COLLISION: The node can send the frame in the next slot
- COLLISION: The node retransmits the frame in each subsequent slot with P probability.



Max Efficiency: Channel is useful for transmissions **37%** of the time

PROS	CONS
• Single active node can continuously transmit at full rate of channel • Highly decentralised: only slots in nodes need to be in sync • Simple	• Collisions, wasting slots • Idle slots • Nodes maybe able to detect collisions in less than the time to transmit a packet • Clock synchronisation

Pure Un-slotted ALOHA: Simpler, with no synchronisation



- When the first frame arrives, transmit it immediately
- Collision probability will increase:
  frames sent at t_0 collides with frames sent in [$t_0 - 1$, $t_0 + 1$]
- **Max Efficiency = 18%** of the time for useful transmissions.

Carrier Sense Multiple Access (CSMA): Nodes sense / listen before they transmit.

- If channel is sensed to be IDLE: transmit entire frame.
- If channel is sensed to be BUSY: defer transmission.
- Collisions can still occur because propagation delay may cause two nodes to not hear each other's transmissions.
  - Distance between the nodes + propagation delay affect collision probability.
- CSMA reduces but NOT eliminates collisions. This is a problem as collisions still take up an entire full slot.

Carrier Sense Multiple Access + Collision Detection (CSMA / CD): Nodes detect collisions by sensing transmissions from other nodes while transmitting a frame.

- If collision is detected: node instantly terminates transmission of the frame + transmits a jam signal + waits for a random time interval before trying to resend the frame.
- Collisions are detected within a short-time frame
- Collisions are aborted, reducing channel waste
- CD is easy in wired LANs: measure signal strengths, compare transmitted, received signals
- Difficult in wireless LANs: received signal strength overwhelmed by local transmission strength

CSMA/CD Algorithm:

1. NIC receives datagram from network layer: creates a frame + encapsulates datagram
2. **IF** NIC senses channel is IDLE: starts frame transmission.
   ELSE wait until channel is IDLE.
3. **IF** NIC transmits frame without detecting another transmission, transmission is complete.
   ELSE abort transmission + send jam signal to ensure all receivers detect the collision.
4. After abortion, NIC enters **Exponential Back-off**:
   - After m_{th} collision, the NIC chooses a random K from { 0 . . . 2^m-1 }.
   - NIC waits K*512bit times then returns to STEP #2
   - More collisions = longer backoff.

For CSMA/CD to work, place restrictions on min frame size / max distance because transmission / propagation delay can affect collision probability.

(3) Taking Turns: Nodes take turns, but nodes with more to send can take longer turns.

Polling Protocol

- The base station retains total control over the channel + Frame content is no longer fixed, allowing variable sized packets to be sent.
- The base station sends a specific packet (a poll packet) to trigger the transmission by a certain node.
- Nodes wait to receive a poll packet and upon receiving it, transmits the frame.
- How it works: A control token passed from one node to the next sequentially.
- Concerns: Token overhead, latency, single point of failure (token)

MAC Protocols: Channel Partitioning vs. Random Access vs. Taking Turns

- **Channel Partitioning Protocols (TDMA, FDMA)**
  - High load: Shares channels efficiently + fairly
  - Low load: Inefficient, 1/N bandwidth allocated even if only 1 active node in channel.
- **Random Access Protocols (Slotted ALOHA, Un-slotted ALOHA, CSMA, CSMA/CD)**
  - High load: High collision overhead
  - Low load: Single node can fully utilise channel
- **Taking Turns Protocols (Polling)**
  - Best of both Channel Partitioning + Random Access

Local Area Network Addressing + ARP resolution protocol

MAC is a 48-bit address burned in a NIC and is hex-based.

- It is used in a LAN to get a frame from one interface to another physically connected interface.
- Manufacturers of NICs buy a portion of MAC address spaces, administered by the IEEE.

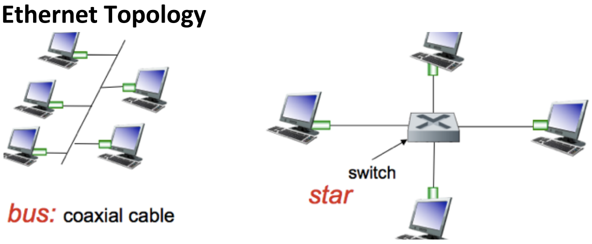
ARP (Address Resolution Protocol) helps determine an interface's MAC address and IP address.

- **ARP Table**: A table in each node in a LAN with entries of: **<IP Address, MAC address, TTL>** where TTL is time after which address is forgotten.
- Scenario #1: Send datagram from A → B within LAN (B's MAC address is not in A's ARP table)
  1. A <u>broadcasts ARP query packet containing B's IP address</u> → All nodes including B receives broadcast
  2. B <u>replies to A's broadcast packet with B's MAC address</u> → Frame is sent to A's MAC address
  3. A <u>caches/saves IP-to-MAC pair as an entry in its ARP table</u> (until TTL is reached)
- Scenario #2: Send datagram from A→B via. R outside of LAN
  1. For A to send a packet to B, A must know: **(1) *B's IP address* (2) *1st hop router R's IP address*: (3) *R's MAC address***
  2. A <u>creates datagram with SRC=(A) | DEST=(B)</u> , then encapsulates in a frame with <u>dest=(R's MAC + datagram)</u>
  3. A <u>sends frame to R</u>: R receives frame, detaches datagram from the frame.
  4. R <u>detaches datagram from the frame</u>, <u>creates link-layer frame</u> with B's MAC address as dest + datagram
  5. R <u>forwards frame to B</u>, then B detaches the datagram from the frame. The data has reached the destination.

Ethernet

There are many different Ethernet standards for different physical media with different speeds: e.g. fibre, cable. However, they all use a common MAC protocol and frame format.

Ethernet Topology



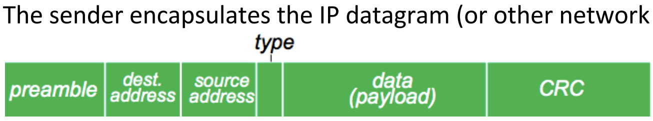
BUS: Popular through mid 90s, where all nodes can collide with each other. CSMA/CD for media access control.

STAR: Used today, where there is an active SWITCH in the centre.

- Each "spoke" runs a separate Ethernet protocol, so nodes do not collide with each other. No sharing, no CSMA/CD.

Ethernet Frame Structure

The sender encapsulates the IP datagram (or other network layer protocol packet) in an **Ethernet Frame**



<u>Preamble (7 bytes)</u>: Used to sync receiver / sender clock rates. | <u>Addresses (6-bytes)</u>: Source / Destination MAC address.
<u>Type</u>: Indicates if there is a higher-layer protocol | <u>CRC</u>: Cyclic Redundancy Check at the receiver.

Ethernet: Unreliable + Connectionless

<u>Connectionless</u>: No handshaking between NICs sending and NICs receiving.
<u>Unreliable</u>: A receiving NIC doesn't respond with ACKs/NAKs to the sender. Data in dropped frames recovered only if the initial sender uses a higher-layer RDT method, otherwise the data is lost.
<u>Ethernet's MAC protocol</u>: Un-slotted CSMA/CD with binary back-off.

LAN: Ethernet Switches

Link-Layer-Devices: Stores and forwards Ethernet frames. Examines incoming frame's MAC → forwards frame to outgoing link.
Transparent: Hosts are unaware of the presence of switches.
Plug-and-play, Self-Learning: Switches do not need to be configured.

Switches: Multiple Simultaneous Transmissions + Forwarding table

Hosts have a dedicated link / connection to the switch. Link has FULL-DUPLEX and no collisions, as each link is its own domain.
Each switch has a **Switch Table: <MAC address of host, interface of host, TTL>**

Switches: Self-Learning

MAC addr	interface	TTL
A	*1*	*60*

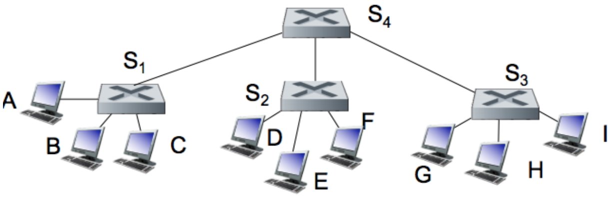
A switch learns which hosts can be reached through which interfaces.

- When a frame is received, a switch <u>learns the location of the sender</u> + records sender/location pair in ^switch table.

When a frame is received at the switch:

1. Record the incoming link, MAC address of the sender.
2. Index the switch table using the MAC destination address.
3. **IF** entry is found **{**
   IF dest MAC on segment exists in the switch's table / comes from same port → drop/filter packet
   ELSE forward frame on the interface indicated by the entry.
   } ELSE flood /* forward frame to all the interfaces except the arriving interface */

Switches: Multiple Interconnected switches
Switches can be connected together i.e. three routers in one house.



Q: Sending from A-to-G, how does S_1 know to forward frame to G via. S_4 and S_3?

A: Self-learning again! Works exactly the same way as in the single-switch case.

Switch Poisoning (DoS): Attacker fills up a switch table with bogus entries by sending large # of frames with bogus source MAC addresses. Since the switch table is full, genuine packets frequently need to be broadcasted as previous real entries are wiped.

Wireless Networks

Two important challenges:
(1) **Communication** over a wireless link (2) **Mobility:** Handling mobile user who changes point of attachment to the network.

Frequency = C / λ , where C = speed of light | λ (lambda) = wavelength
WaveLength = **C / f** , where C = speed of light | f = frequency

Elements of a wireless network

- Wireless Hosts: Laptops, smartphones, running applications (either stationary or mobile)
- Base Station: Cell towers, 802.11 Access Points
  - Typically connected to the Wired Network
  - **Relay:** responsible for sending packets between wired / wireless hosts in its "area"
- Wireless Link: Typically used to connect mobiles to Base Stations
  - Multiple Access Protocol coordinates link access.
  - Various data rates + transmission distances.
- MODE #1: Infrastructure Mode: Base Station connects mobiles into the Wired Network
  - **Handoff**: mobile changes Base Stations that provide them with a connection into the wired network.
- MODE #2: Ad-Hoc Mode: No Base Stations.
  - Nodes can only transmit to others within link coverage. Route among themselves.

Wireless link characteristics

Characteristics make wireless communication difficult:

- Decreased signal strength / path loss: radio signals lose affect as it propagates through matter.
- Interference from others sources: wireless network frequencies shared by other devices e.g. phones can interfere.
- Multipath Propagation: Radio signals reflect off objects, arriving at the dest at slightly different times.

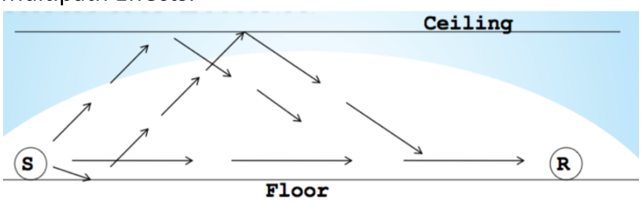
Path loss / Attenuation:

- Things that can affect attenuation:
  Reflection, diffraction, absorption | terrain contours (urban/rural) | humidity

$$\text{Free Space Path Loss} = \left(\frac{4\pi d}{\lambda}\right)^2$$

$$\text{Free Space Path Loss} = \left(\frac{4\pi d}{\lambda}\right)^2$$

Multipath Effects:



Signals bounce off surface and interfere with one another.

Self-interference.

Signal-To-Noise-Ratio (SNR): Ratio between max signal strength that a wireless connection can achieve + noise present in connection. Larger SNR = better signal / easier to extract signals from noise.
Bit Error Rate (BER): #bit errors per unit of time.
SNR vs BER Trade-off:

- Given a physical layer: AIM is to increase SNR + decrease BER
- Given an SNR: AIM is to choose a physical layer that meets BER requirement + giving highest throughput.

Exam Sections

1. Transport Layer (10 marks)
2. Network Layer and Routing (8 marks)
3. Link Layer (10 marks)
4. Wireless / Mobile Networks and Security (7 marks)