

↳ what is the Internet? (recipes)

↳ There are some basic hardware and software components that make Internet.

↳ Hosts ↴

PC, server, routers, wireless devices.

↳ Links ↴

All the lines or links connecting to the hosts / copper wire, coaxial cable, copper wire, radio, satellite...

↳ Routers ↴

marked paths for data, exists in the core of network.

↳ protocols ↴

Implemented by the hosts and useful in many ways to communication, data transfer, etc.

↳ Internet provides communication services like web, games, email, file sharing, best way of transferring data.

↳ best way of "reliable delivery", like TCP vs. unreliable data delivery protocol, IP, HTTP

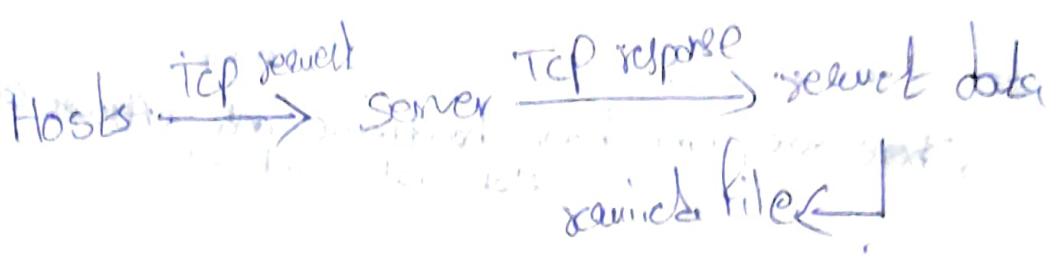
↳ What is protocol? → rules for interaction between the hosts.

↳ format of messages

↳ order of messages

↳ actions of messages

↳ send and receive messages.



①.2

Network edge

Interface

- ↳ The network ~~device~~ that are accessible to end-users. e.g. Applications, skype, web browser, phones, laptop, desktop etc. --

② Network core.

is all routers, and backbone. ~~tier 1~~
 networks that are doing 'bottles' between different networks

Endsystems are .. nothing but hosts. That are accessible to their own.

↳ Two models

↳ client/server model. provides ~~server~~.

The server that provides resources

for the client.

③ services for the web servers

Ex's: web → web servers
 chrome → Apache

↳ peer-to-peer model.

Every body is a client and a server.

Anybody can talk to anybody.

Ex's: skype, Bit torrent etc. --

- ↳ Access Networks and physical media.
 - ↳ the network that is physically connects end system and nodes.
- ↳ How to connect end systems to Router?

Ans :- By Access Networks like ~~and~~ residential access networks. Ex:- cable, fibre etc

- ↳ Ex:- modems, 3g, 4g, College internet access etc

- ↳ Wireless Access networks

↳ wireless LAN's → 11 / 54 / 108 Mbps

↳ wider area → 3G, 4G, LTE / 10's Mbps

- ↳ Two Media

↳ Guided Media → wireless

Ex:- Twisted pair (TP) (LAN)

Coaxial Cable (T.V)

Fiber optic cable (light)

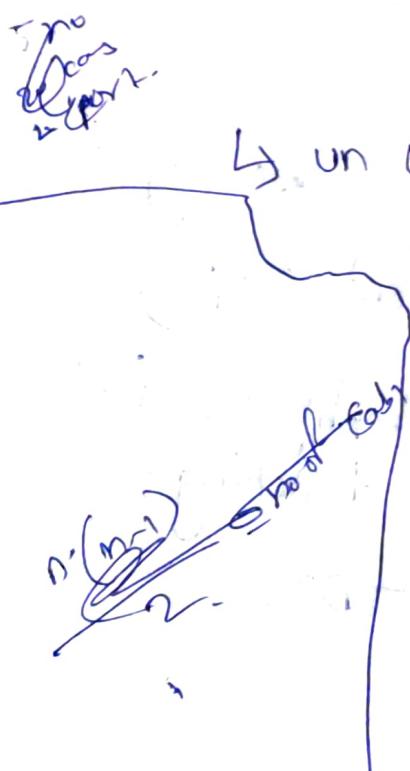
↳ un Guided media → wireless.

Ex:- Terrestrial microwave

LAN (WiFi).

Wide-Area (cellular)

Satellite.



Network Core

↳ How is data transferred through the net?

Ans: ↳ circuit switching.

↳ there is a separate circuit for every connection b/w A and B (sender and receiver).

↳ packet switching.

All the data is chunked and divided into these discrete chunks of data and those chunks of data are independent of each other; we switch that data separately along each router.

End-to-End → all the resources are reserved so that they can quickly transfer.

Reservation → reserving the speech ahead of time & (continuously)

↳ Two simple access techniques that are used in circuit switching:

↳ ~~FDM~~ and TDM

FDM → Frequency division multiplexing

TDM → Time division multiplexing

- getting frequency from particular channels
Ex: Radio -
- getting frequency from particular channels
for particular time -
Ex: phone calls -

packet switching

Advantages

- No bandwidth division.
- No dedicated Allocation.
- No Reservation.

disadvantages

- ↳ Resource contention
- ↳ Congestion? (buffering)
- ↳ Store and forward.

↳ packet switching is also called as statistical multiplexing because if A and B are two hosts sending packets to other clients then if A host sends packets with 100 mbps then B host sees that 100 mbps then there will be A host's packets that will be more.

→ packet switching is a store and forward.

$$\text{Transmission Delay} = \frac{N}{R} L$$

$L \rightarrow$ No of bits

$n \rightarrow$ speech of links

$N \rightarrow$ no of packets

\Rightarrow no of bytes

* Network of Networks

All types of hosts connects into network \hookrightarrow All types of hosts connects into network via an access ISP (Internet service provider)

They can provide by DSL, cable

or wireless cellular

FTTH

To connect billions of hosts to

make up Internet the access

ISPs themselves must be interconnected. This is called network of networks

(H)

How do loss and delay occur?

- ↳ transmission delay occur while router puts the packets in the line ~~on air~~.
- ↳ queuing delay occur while queuing the packets in the line. (waiting)
- ↳ loss - occur when the queue is full then router will drop the ~~packets~~ packets.
- ↳ propagation delay ~~is~~ is the short amount of time that will take according to the medium. ~~to travel~~ for the signal.
- ↳ processing delay the time that will take to find the address that ~~will~~ packets will go.

$$\text{Nodal delay} = d_{\text{proc}} + d_{\text{queue}} + d_{\text{prop}}$$

$$d_{\text{nodal}} = d_{\text{proc}}$$

↳ If average queuing delay increase then traffic intensity increases.

↳ What if buffer is full?

(Ans) packets will drop.

↳ What if packet is lost?

(Ans) packets will be retransmitted at original source

by the previous node. @ original source

host

at which

Throughput is the rate ($\text{bits}/\text{time unit}$) at which

bit transferred between sender/receiver

points of

instantaneous \rightarrow in given time

Average \rightarrow long period of.

why layers?

- ↳ If we layer the layers then it will be well defined, separate and Neat interfaces, Exchange information, Hiding the Details, Similar function to know themselves.

↳ Internet protocol stack @ ISO/OSI

- Application → Implementing network Application
- presentation (encap) } intermediate layer
- section. (syn)
- transport → processing the data buffers ip / UDP.
- network → routing data. / IP
- link → one node to next node. error control / mac address
- physical → bits converter.

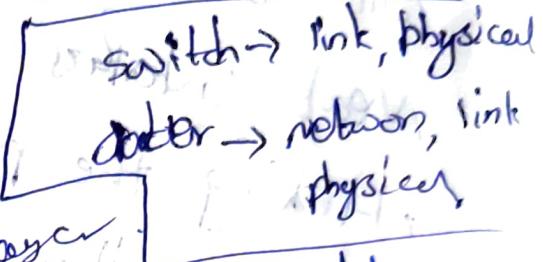
OSI → open system interconnection

~~ISO~~ → transmission control protocol

TCP → user datagram protocol.

UDP → user datagram protocol.

- ↳ End to End \rightarrow client, server
- ↳ point to point \rightarrow switch, Router
client, server



- ↳ Application layer
 - \rightarrow End to End.
 - \rightarrow Implementing network Application
 - \rightarrow User services like, web, Email, file transfer.

- ↳ Transport layer
 - \rightarrow End to end layer
 - \rightarrow End to end data
 - \rightarrow Delete the duplicate data
 - \rightarrow Some data will be present
 - \rightarrow back data will be received
 - \rightarrow reorder the data.

- ↳ Network layer
 - \rightarrow Point to point layer.
 - \rightarrow Addressing the destination node
 - \rightarrow Routing the data through the network
 - \rightarrow provides Administration functionally.

- ↳ Link layer
 - point to point layer
 - Error control.
 - flow control
 - MAC (Medium Access Control layer)
 - ↳ transfers data fairly.

- ↳ physical layer
 - point to point layer
 - transmission of Raw Bits.

- ↳ Encapsulation means adding additional information to a packet so that the packet can be used by the next layer (section layer).

- * datagram is a packet at network layer.
- * segment is a packet at transport layer.
- * frame at link layer.

Malware

↳ virus

↳ worm

↳ Trojan Horse

↳ spyware

↳ Botnet

↳ Denial of Services Attack

↳ By putting fake addresses that

increases the traffic in the network.

↳ It will get you banned.

↳ packet sniffing

↳ wired attack

↳ IP spoofing

↳ pretending to be someone's
addresses

↳ Record and playback attack

↳ recording the packets

↳ Network topology is the arrangement of the various components of a computer network.

1) Bus:

Data transmission in this topology is through a single cable.

↳ Cheaper.

↳ Used in small networks.

↳ Only in one direction.

↳ LAN.

Ex:-

2) Ring:

The hosts are connected in the ring form.

↳ Cheap to install and expand.

↳ Unidirectional.

3) Star:

All the hosts are connected to a single host/hub with single cable in star shape.

↳ Fast performance and low network traffic.

- ④ Mesh topology
- ↳ point to point connection.
 - ↳ fully connected to other hosts
 - ↳ flexible and costly.
 - ↳ more cables.
 - ↳ more security.

- ⑤ Tree topology
- ↳ It has a root node and all other nodes are connected to it forming a hierarchy.

- ↳ Wide Area Network - WAN
- ↳ easy to detect errors.
 - ↳ more costly.
 - ↳ Hybrid topology
- ⑥ Hybrid topology
- ↳ A network structure containing more than one topology to be hybrid topology.
 - ↳ flexible, error detecting.

(2.2)

network Apps

- ↳ e-mail, web, instant messaging, remote login, P2P file sharing, multi user network games, streaming, social networks, video conferencing, grid computing ..

↳ How to Create a network app?

- ↳ no need to write program
- ↳ no need to add extra Network Core Devices
- ↳ Application layer protocols works on end-to-end layer.

↳ Client server Architecture.

- ↳ Google Data centers
- ↳ IP addresses are prominent

↳ P2P Architecture

- ↳ no server
- ↳ Direct communication
- ↳ Intermittent connection
- ↳ IP address may change.

- ↳ Difficult to manage
- ↳ Highly Scalable.
- * Hybridorks and P2P with Ext. Skype.

Socket interface

Socket is programming interface for communication between host processes. (end point)

Transport layer is a layer that transports data that comes out from socket to the other receiving process.

* How can we identify a process.

- ↳ tasklist (Windows)
- ↳ ps -aux (top) (MacLinux)

↳ Identifier ↓ ip address, port number to identify the process.

↳ Application layer protocol defines.

↳ Message Type is a general type of message written by user.

↳ Message syntax of message.
subcomponents of message.

↳ Message ~~syntax~~ semantics
meaning of packets.

↳ Message Rules.
When and how processes
respond to the messages.

protocols

public domain?

proprietary

Skype

↳ HTTP

↳ ppstream.

↳ FTP

↳ SMTP

↳ Bit Torrent

↳ Application layer provides communication
between two processes.

↳ Transport services

↳ recover Data loss

↳ low delay for transporting packets

↳ bandwidth should be more

↳ bandwidth for youtube; streaming

↳ security

Internet Transport protocols

Internet

Transport

↳ TCP

↳ Connection oriented.

↳ Flow control

↳ orderly of datapackets to network

↳ reliable service

↳ congestion control

↳ opposite to TCP

↳ UDP

* http is a hypertext transfer protocol.

- ↳ Non persistent
 - ↳ one object per TCP connection
 - ↳ persistent
 - ↳ multiple objects per TCP connection?
- * In Non persistent HTTP
- ↳ 2 RTT's / object.
 - ↳ overhead in terms of TCP connections
- * In persistent HTTP
- ↳ reuse open connection
 - ↳ send ~~replies~~ ~~replies~~ immediately
 - ↳

 ↳ Electronic Mail
 - ↳ user Agents are used
 - ↳ mail servers store mails
 - ↳ mail servers send mails to user Agents.
 - ↳ SMTP (simple mail transfer protocol)
 - ↳ used to transfer the mails between mail servers

- ↳ SMTP
 - ↳ It uses TCP to reliable data transfer.
 - ↳ port is 25
 - ↳ SMTP allows the message between sending server and receiving server.
 - ↳ There are 3 phases
 - ↳ Handshaking (greets the server)
 - ↳ Transfers b/w servers.
 - ↳ Cyclic.
 - ↳ 7-bit ASCII
 - ↳ persistent connection

⇒ HTTP

Command / Response

ASCII

8-bit

pull from server

single part message

SMTP

Command / Response

ASCII

7-bit

push to server

multi part message

↳ POP (post office protocol) and
IMAP (Internet Mail Access protocol)
are older protocols used to pull
the data from server.

POP3

Download only
Delete
Download and
keep

IMAP

Server

Folders

↳ DNS → Domain Name system.
* How do we map IP addresses and
names?

↳ DNS is to map between names
and IP addresses.

↳ Distributed ~~anywhere~~ everywhere.

↳ Hierarchical distribution.

It is one type of database.

Port no. 53.

* why DNS is distributed?

↳ To decrease the traffic volume.

↳ It could not be disrupted.

↳ easy maintenance.

↳ One server goes down then there will be no problem for other servers.

↳ DNS : Features.

↳ DNS will return the IP Address

↳ DNS will return the Hostname.

↳ DNS supports Common names like www.netflix.com, netflix.

↳ DNS gives unique IP addresses

↳ DNS gives different ones to different ones.

↳ Load distribution.

TLD → Top Level Domains
Ex: com, org, net, edu.

↳ DNS Records: is a database stores
Name, value, type, TTL (Time to live)

↳ Record Types:

Type A → Name → Hostname.
Value → IP Address.

Type NS → Name → Domain.
Value → Hostname of domain
server.

Type CNAME → Alias name.
Value → Canonical name.

Type MX → Name → Hostname.
Value → name of mail server.

3.1

Transport layer is providing logical
communication between processes running
on different hosts.

- ↳ end to end protocol.
- ↳ TCP and UDP protocols are used.

- ↳ Transport layer is nothing but Collects & Distribute to the users.
- ↳ Multiplexing is about ~~of taking~~ pushing data on to one channel.
- ↳ Demultiplexing is about ~~to take~~ pulling data on to one channel.
- ↳ IP datagrams are used to fix the port number, is used to fix the destination on the other host to send the data.
- ↳ In transport layer of demultiplexing
 - ↳ IP datagrams stores Source IP Address, Destination IP Address, and segments.
 - ↳ segment will store source port and destination port.

Demultiplexing :-

- ↳ UDP - ~~dest~~ identifies the 2 things
 - ⇒ dest → ip address
 - ⇒ dest → port
- ↳ In UDP segment - If the source port ~~(C)~~ ip addresses are same and if the Dest port and ip address are different:
- ↳ In UDP socket consists of dest ip address and port number as if some ip addresses and port number. Some ~~dest~~ ip addresses and port number. Then two segments will be directed to some destination socket.
- ↳ port numbers are arranged in 0-1023.
 - Ex:- port 80 → web.
 - FTP → 21.
- ↳ This is how demultiplexing happens in UDP.

- ↳ TCP socket identified by 4 things
 - ↳ Source IP Address
 - ↳ Source port
 - ↳ Dest IP Address
 - ↳ Dest port
- ↳ Different ~~ip address~~ ports ~~can't~~ port ~~can't~~
- ↳ TCP connection pairs are unique.

- * why is UDP there?
- 1) no delay
 - 2) simple & cheap
 - 3) less wasted bandwidth
 - 4) no specific timits.
- streaming multimedia.
- because
- Lost data
inorder.
- Loss tolerant
Rate sensitive.

↳ DNS

↳ SNMP

- ↳ Length → Segment + header is bytes.
- ↳ Checksum is the number ~~that is computed~~ based on data in the packet.
- ↳ The number is used to check whether binary. to check whether the packet is matched to the packet that is received properly.
- ↳ How checksum works?
 - ↳ Checksum is a simple error detection mechanism to determine the integrity of data transmitted over a network.
 - ↳ Add up all the integers or by 1's complement.
 - ↳ Any receiver will also do same.
 - ↳ If the received sum is not equal to stored sum then there must be an error.
 - ↳ If there is an error then it will be detected.
 - ↳ All zeros will be no error.

- ↳ why checksum is at transport level?
 - ↳ link layer has no guarantee that reliable.
 - ↳ it can be corrupted if we don't use checksum in transport layer.

↳ End-to-End principle.

↳ Communication protocol operations occur at the end points of communication system.

* principles of reliable data transfer.

- ↳ transport layer has to implement the protocol to create a reliable channel.
- ↳ `dat.send()` is the function that does reliable data transfer by reliable data transfer protocol.

- ↳ After sending it will divided into packets; Udt_send - ask that only sent to data over unreliable channel. and Udt_rcv will receive the data from unreliable channel and deliver to the deliver_data() function.
- ↳ How these function calls will be implemented?
 - ↳ Finite state machines (FSM)
 - ↳ when these events and actions happen then we will move from 0 state to next state.
- ↳ we incrementally develop the Rdt
 - ↳ Rdt 1.0 is the first version of the protocol \hookrightarrow this is perfectly Reliable over reliable channel
 - ↳ No bit Errors
 - ↳ No packet Loss

↳ on sender side (sdtr-sent)

- ↳ wait for call from Application layer
- and it will make the packet.
- packet = make pkt.
- and it will send the packet.

on receiver side (sdtr-rxv)

↳ on receiver side (sdtr-rxv)

- ↳ wait for call from the
- sender. or it will read back
- the packet and deliver data.

* sdtr 2.0

- ↳ Bits Errors
- ↳ No packet loss
- ↳ How to detect errors?
- ↳ checksum
- ↳ How to recover from errors?
- ↳ By receiver feedback.

- ↳ There are Two types
 - ↳ Acknowledgements : Ack
 - ↳ Negative Acknowledgements : NAK

↳ Then Retransmission takes place.

- ↳ what happens if Ack / NAK is corrupted?
- ↳ sequence number to the packet.

- ↳ Rdt2.1
 - ↳ Implements sequence number to solve corruptible files.
 - ↳ Ack / NAK
 - ↳ sequence is added to packet
 - ↳ 2 sequence (0, 1)

↳ Sdt 2.2

↳ Ack's only. (same as Sdt 2.1)

↳ Sdt 3.0

↳ Bit Errors.

↳ packet loss

↳ Error Detection.

↳ Checksum, Ack's seq#, retransmission.

↳ How to detect packet loss?

↳ How to detect packet loss?

↳ wait for "reasonable" time period.

↳ Retransmit the packet if no Ack received.

↳ Stop-and-wait operation

$$wait = RTT + L/R$$

↳ Here sender waiting most of the time to get Ack for that pipeline protocol is used.

↳ pipelined protocols.

↳ Go-back-N

↳ Selective Repeat

↳ Sender

↳ Up to N packets in pipeline

Receiver:

↳ Sends cumulative ACK's

↳ If there's a gap then it won't ACK.

Sender: set the time.

↳ If timer expires then retransmits.

Sender: Up to N packets in pipe

↳ Up to N packets in pipe

Receiver: Individual packets w/o ACK's

Sender: timer for unACKed.

↳ If timer expires then retransmits.

↳ If timer expires then retransmits.

↳ TCP Deliable data transfer.

↳ TCP creates reliable service on top of IP's unreliable service.

↳ goes in pipelined segments

↳ cumulative Ack's

↳ retransmission timer

↳ when retransmissions occur

↳ timeout events

↳ duplicate Ack's



↳ B. what TCP does in this scenario:

↳ ignores flow control

↳ ignores congestion control

↳ ignore duplicate ACKs

↳ ignore duplicate ACKs

If ACK is lost from first packet and directly see second ACK then it will assume first ACK is also successful

- ↳ If the packet lost then the receiver.
- ↳ sender does not wait for timeout.
It will wait for triple ACK's ack.
It will ~~send~~ the lost packet if
retransmit
called fast retransmit
- ↳ point-to-point
- ↳ connection oriented
- ↳ reliable data; in-order delivery
- ↳ Congestion and flow control.
- ↳ TCP seq number is the number of the first byte in the segment.
- ↳ how to set TCP timeout value?
 - ↳ RTT estimation
 - ↳ too short → transmission waste.
 - ↳ too long → segment loss...
time waste.

↳ how to estimate RTT?

↳ By Sample RTT,

$$\text{EstimatedRTT} = (1-\alpha) * \text{EstimatedRTT}$$

$$+ \alpha * \text{sampleRTT}$$

here $\alpha = 0.125$

↳ Timeout interval = EstimatedRTT +

$$L * \text{DevRTT}$$

$$\text{Here DevRTT} = (1-\beta) * \text{DevRTT} +$$

$$\beta * (\text{sampleRTT} - \text{EstimatedRTT})$$

here $\beta = 0.25$

* Approaches to Congestion Control.

↳ End-to-End help to control.

↳ Routers

↳ By ATM ABR Control.

- * ↳ How TCP controls Congestion.
- ↳ only to End-Connection oriented
- ↳ decreases the send rate.
- ↳ How to perceive Congestion?
- * ↳ How to implement the feedback mechanism.
- ↳ Acks.
- ↳ Acks not Received.
- ↳ How to limit send Rate?
- * ↳ How to limit the unAcked bytes in pipeline.
- $$\text{cwnd} = \frac{\text{LastByteSent} - \text{LastByteAcked}}{\text{sendRate}}$$
- $$\text{sendRate} = \frac{\text{RTT}}{\text{cwnd}}$$

↳ when sending rate increases: more ACKs being received and by that packet will loss so we have to decrease sending rate.

↳ first we have to start slowly and then we use Double Cwnd, then RTT threshold. It is called slow start phase.

4.2

- ↳ Transport layer from sending to receiving host
- ↳ sender: encapsulates segments into the datagrams.
- ↳ network layer protocols in every host, router, and server.
- ↳ two network layer functions:
 - ↳ Forwarding
 - ↳ routing.

- ↳ Routing → moving packets from source's input to router's output.
- ↳ Routing → ~~is~~ finding the route taken by packets from source to dest.
- ↳ Routing algorithm determines end-to-end path through network.
- ↳ Forwarding table determines local forwarding at the router.

- ↳ Datagram networks:
 - ↳ packets forwarded using destination address
 - ↳ source ip address
 - ↳ dest ip address
- ↳ Datagram forwarding table consists of dest ip addresses and route of delivery.

↳ how routers are deciding ip addresses and store them?

↳ There will be address ranges

↳ from proxy 'interface'

↳ In routers, the packets will be forwarded using destination address and longest address prefix.

↳ First 21 bits will match. And last bits will decide the ip address.

↳ If the two ip addresses are same then longest address prefix will decide where it should go.

↳ Router architecture

↳ Input ports → datagrams Enter

↳ Output ports → datagrams exit

↳ Routing determines the routing path

By routing algorithms like (RIP, OSPF, BGP)

↳ Forwards datagrams from input to output ports By high-speed switching fabric.

↳ routing and forwarding data planes
is logical separator.

↳ routing processor will compute
the routes and tells high speed
switching fabric how to forward
the packets from input to output port.



↳ In Input port

↳ line termination (physical layer)

↳ link layer protocol (data link)

↳ queueing

↳ forwarding Table works

↳ if line speed is greater or forwardly
speed is less then packet
will drop.

↳ switching fabric transfers the packet
from input port to output port.

↳ three types of switching fabrics -

↳ memory

↳ bus

↳ crossbar

Memory switching

- ↳ Traditional method for output port
- ↳ packets copied to system memory and then moves to output port
- ↳ limited by memory bandwidth

bus switching

- ↳ by shared bus. It will be faster.
- ↳ limited by bus bandwidth

Crossbar switching

- ↳ fragments datagram into fixed length cells and switches through fabric
- ↳ limited by crossbar bandwidth

Output ports

- ↳ buffering when datagrams arrive.
- ↳ possible bottleneck.

Queuing → packet loss

- ↳ occurs
- ↳ Head of the line (HOL) blocking.

- ↳ what work will done in between routing and forwarding?
 - ↳ Routing Algorithm will tell the ~~packet~~ node to move from which node to output link.
 - ↳ Forwarding tables determine local forwarding at this router.
- * ~~Routing algorithm classification~~

- ↳ IP datagram format
 - ↳ IP protocol version
 - ↳ consists of
 - ↳ header length
 - ↳ time to live
 - ↳ datagram length
 - ↳ check sum
 - ↳ source ip address
 - ↳ dest ip address

↳ If the datagram is large then the IP datagram is divided into several datagrams and they will reassemble only at final destination. (MTU).

↳ IP address is a 32 bit identifier and it is bracketed in two 8 bits.

↳ Interphase is a connector between host and physical link.

↳ Switches have multiple interfaces.
hosts have one @ two interfaces.

↳ maximum value of 18 bits is 223.

↳ Subnets

↳ Isolated network is called subnet
↳ detachly ~~isolate~~ interfaces and creates the island.

CIDR \rightarrow classless Inter Domain Routing

ip address format is a.b.c.d/x

\hookrightarrow x is subnet portion.

Ex:

11001000.00010111.00010000

subnet part

00000000

host part

200.23.16.0/23

\hookrightarrow here 23 is the bits in subnet portion

\hookrightarrow how does host get IP address?

\hookrightarrow By DHCP. subset configuration

\hookrightarrow Dynamic Host Configuration

Protocol.

\hookrightarrow IPv4 and IPv6 are IP versions

\hookrightarrow The length of IPv6 is 128 bit.

- ↳ In IPv4 has 4 billion unique ip addresses available.
- ↳ The range of IPv4 address is 0 to 2³²
Ex: 192.225.108.253
- ↳ It consists of 4 octets each has 8 bits.
- ↳ Octets are separated by dot.
- ↳ Octets are separated by colon.
- ↳ 5 classes
- IPv6:
 - ↳ The length of IPv6 is 128 bits.
 - ↳ It has 340 billion unique ip addresses available.
 - ↳ The range of IPv6 address is 0 to FFFF:FFFF:FFFF:FFFF:FFFF:FFFF:FFFF:FFFF
Ex: 2001:DB8:1902:FB83:1800:1751:CBAA:0000
 - ↳ It consists of 8 octets each has 16 bits.
 - ↳ Octets are separated by (:)
 - ↳ Colon.
 - ↳ No classes.

- DHCP is used to assign IP address to host
 - ↳ It allows host to dynamically obtain its IP address from network.
 - ↳ It allows reuse of addresses.
 - ↳ It allows allocation of IP address to mobile user.
 - ↳ It supports for mobile user.
- ↳ DHCP client server.
 - ↳ DHCP discover
 - ↳ DHCP offer
 - ↳ DHCP request
 - ↳ DHCP ACK
 - ↳ name and address of DNS server.
 - ↳ indicates network versus.
- ↳ How does network get subnet part of IP address?
 - An gets allocated portion of its provider's response's address space.