UNIT-4

The Network Layer Design Issues:

Network Layer is majorly focused on getting packets from the source to the destination, routing, error handling and congestion control.

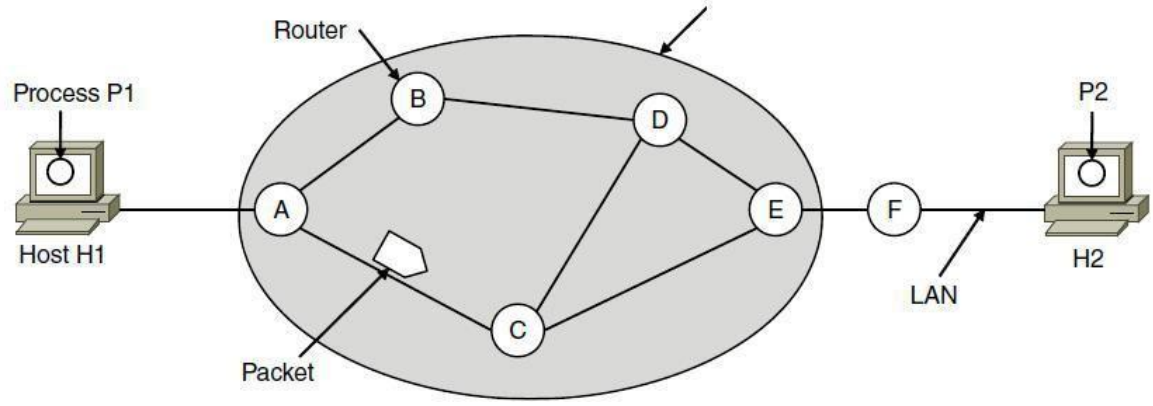
Functions of Network Layer are.

- **Addressing:**
  Maintains the address at the frame header of both source and destination and performs addressing to detect various devices in network.
- **Packeting:**
  This is performed by Internet Protocol. The network layer converts the packets from its upper layer.
- **Routing:**
  It is the most important functionality. The network layer chooses the most relevant and best path for the data transmission from source to destination.
- **Inter-networking:**
  It works to deliver a logical connection across multiple devices.

Network layer Design Issues:

The network layer comes with some design issues they are described as follows:

1. **Store and Forward Packet Switching:**
   The host sends the packet to the nearest router. This packet is stored there until it has fully arrived once the link is fully processed by verifying the checksum then it is forwarded to the next router till it reaches the destination. This mechanism is called "Store and Forward packet switching."



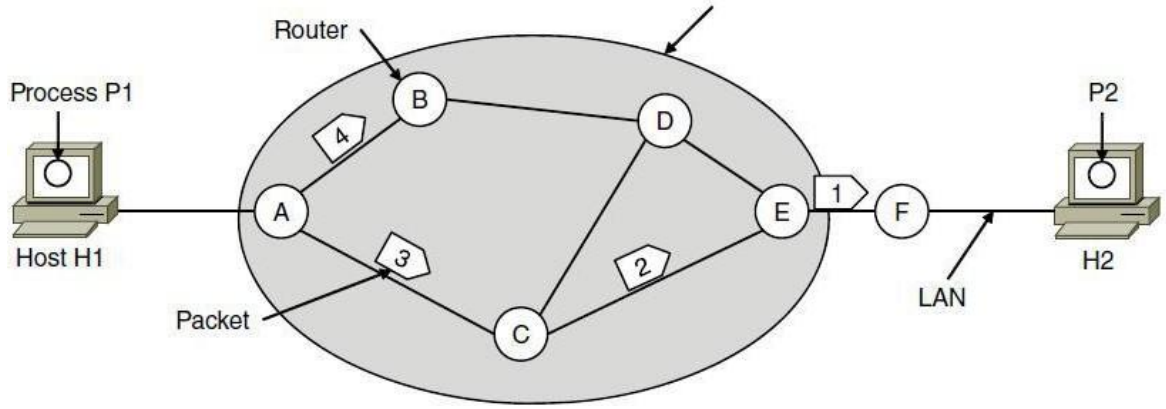
2. **Services provided to Transport Layer:**
   Network layer transfers it's services to the transport layer.

- Offering services must not depend on router technology.
- The transport layer needs to be protected from the type, number and topology of the available router.
- The network addresses for the transport layer should use uniform numbering pattern also at LAN and WAN connections.

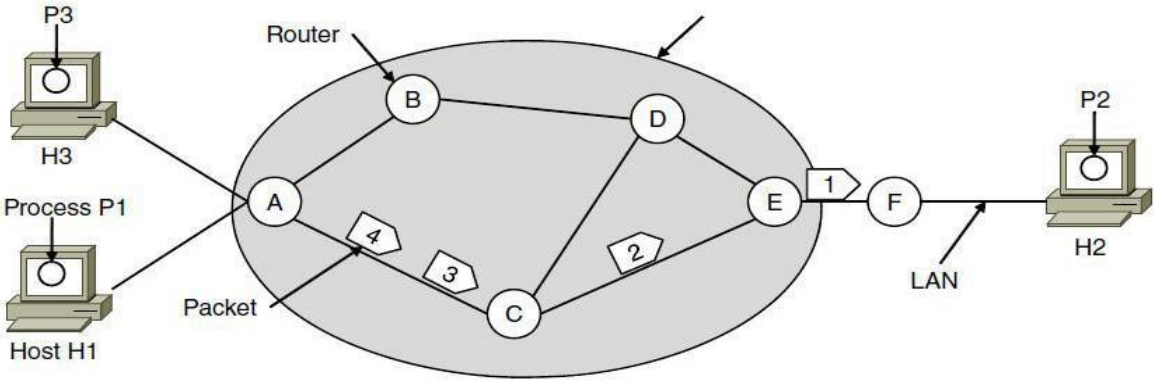
Based on the connections there are 2 types of services provided :

- **Connectionless –** The routing and insertion of packets into subnet is done individually. No added setup is required.
- **Connection-Oriented –** Subnet must offer reliable service and all the packets must be transmitted over a single route.

3. **Implementation of Connectionless Service:**
   Packet are termed as "datagram's" and corresponding subnet as "datagram subnets". When the message size that has to be transmitted is 4 times the size of the packet, then the network layer divides into 4 packets and transmits each packet to router via. a few protocol.Each data packet has destination address and is routed independently irrespective of the packets.



4. **Implementation of Connection Oriented service:**
   To use a connection-oriented service, first we establishes a connection, use it and then release it. In connection-oriented services, the data packets are delivered to the receiver in the same order in which they have been sent by the sender.



It can be done in either two ways:

- **Circuit Switched Connection –** A dedicated physical path or a circuit is established between the communicating nodes and then data stream is transferred.
- **Virtual Circuit Switched Connection –** The data stream is transferred over a packet switched network, in such a way that it seems to the user that there is a dedicated path from the sender to the receiver. A virtual path is established here. While, other connections may also be using the same path.

2. Comparison of Virtual Circuit and Datagram Network:

Issue	Datagram network	Virtual-circuit network
Circuit setup	Not needed	Required
Addressing	Each packet contains the full source and destination address	Each packet contains a short VC number
State information	Routers do not hold state information about connections	Each VC requires router table space per connection
Routing	Each packet is routed independently	Route chosen when VC is set up; all packets follow it
Effect of router failures	None, except for packets lost during the crash	All VCs that passed through the failed router are terminated
Quality of service	Difficult	Easy if enough resources can be allocated in advance for each VC
Congestion control	Difficult	Easy if enough resources can be allocated in advance for each VC

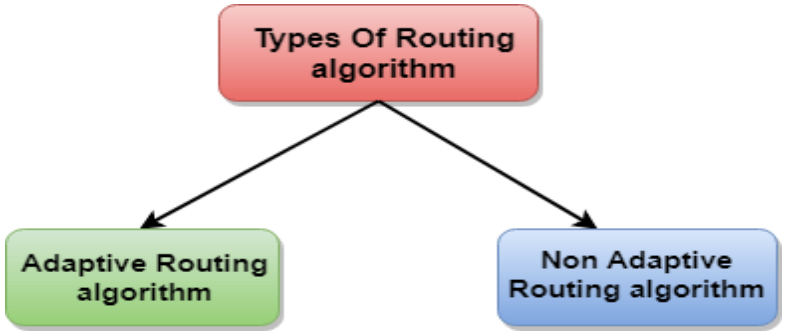
Routing Algorithms:

- In order to transfer the packets from source to the destination, the network layer must determine the best route through which packets can be transmitted.
- Whether the network layer provides datagram service or virtual circuit service, the main job of the network layer is to provide the best route. The routing protocol provides this job.
- The routing protocol is a routing algorithm that provides the best path from the source to the destination. The best path is the path that has the "least-cost path" from source to the destination.
- Routing is the process of forwarding the packets from source to the destination but the best route to send the packets is determined by the routing algorithm.

Classification of a Routing algorithm

The Routing algorithm is divided into two categories:

- Adaptive Routing algorithm
- Non-adaptive Routing algorithm



Adaptive Routing algorithm

- An adaptive routing algorithm is also known as dynamic routing algorithm.
- This algorithm makes the routing decisions based on the topology and network traffic.

Non-Adaptive Routing algorithm

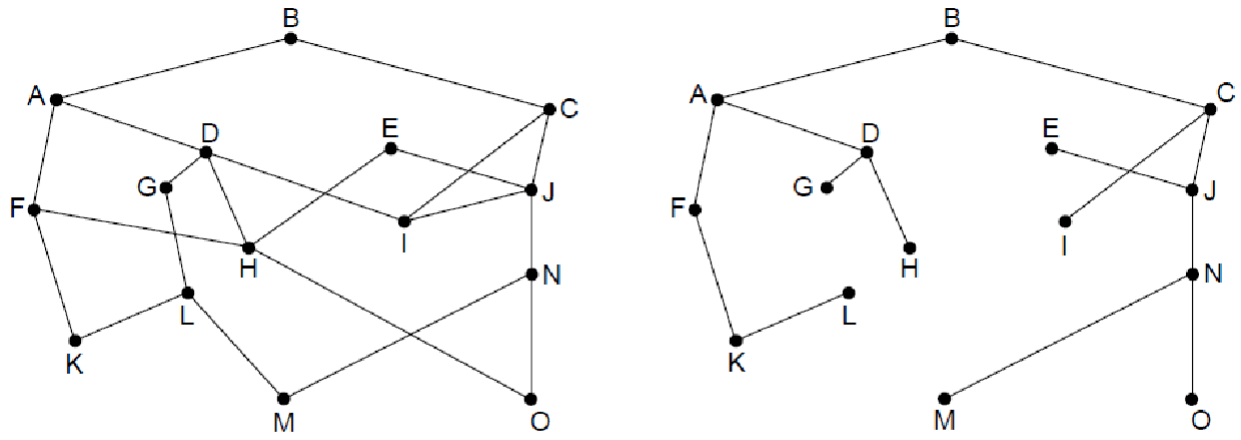
- Non Adaptive routing algorithm is also known as a static routing algorithm.
- Non Adaptive routing algorithms do not take the routing decision based on the network topology or network traffic.
- Different Routing Algorithms
  - Optimality principle
  - Shortest path algorithm
  - Flooding
  - Distance vector routing
  - Link state routing
  - Hierarchical Routing

1. The Optimality Principle

One can make a general statement about optimal routes without regard to network topology or traffic. This statement is known as the optimality principle.

It states that if router J is on the optimal path from router I to router K, then the optimal path from J to K also falls along the same

As a direct consequence of the optimality principle, we can see that the set of optimal routes from all sources to a given destination form a tree rooted at the destination. Such a tree is called a **sink tree**. The goal of all routing algorithms is to discover and use the sink trees for all routers



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2. Shortest Path Routing (Dijkstra's):

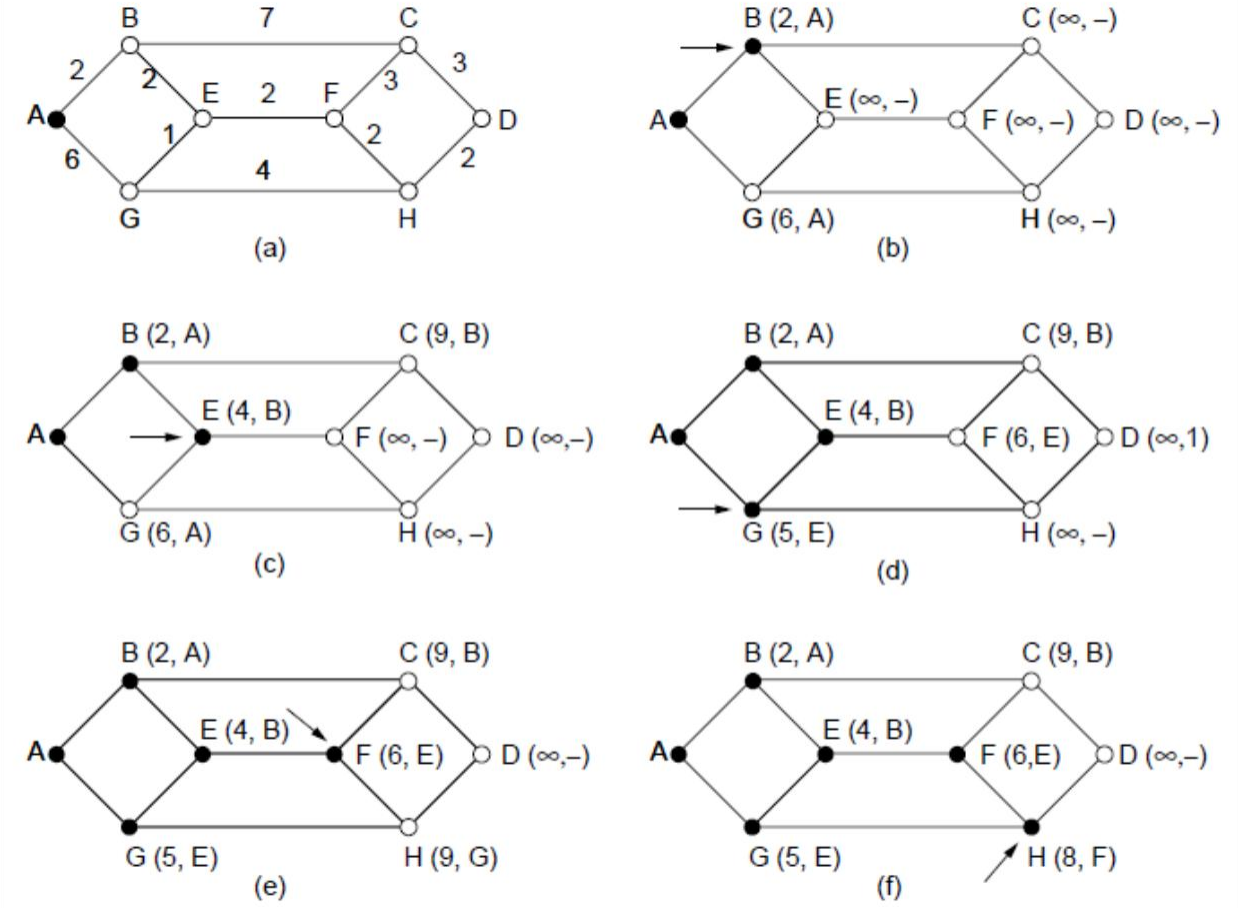
The idea is to build a graph of the subnet, with each node of the graph representing a router And each arc of the graph representing a communication line or link.

To choose a route between a given pair of routers, the algorithm just finds the shortest path Between them on the graph

1. Start with the local node (router) as the root of the tree. Assign a cost of 0 to this node and Make it the first permanent node.
2. Examine each neighbor of the node that was the last permanent node.
3. Assign a cumulative cost to each node and make it tentative
4. among the list of tentative nodes

a. Find the node with the smallest cost and make it Permanent
b. If a node can be reached from more than one route then select the route with the Shortest cumulative cost.

5. Repeat steps 2 to 4 until every node



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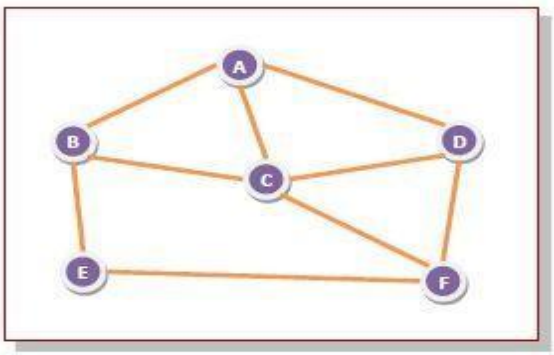
3. Flooding:

Flooding is a non-adaptive routing technique following this simple method: when a data packet arrives at a router, it is sent to all the outgoing links except the one it has arrived on.

For example, let us consider the network in the figure, having six routers that are connected through transmission lines.



Using flooding technique −

- An incoming packet to A, will be sent to B, C and D.
- B will send the packet to C and E.
- C will send the packet to B, D and F.
- D will send the packet to C and F.
- E will send the packet to F.
- F will send the packet to C and E.

4. Distance Vector Routing Algorithm-

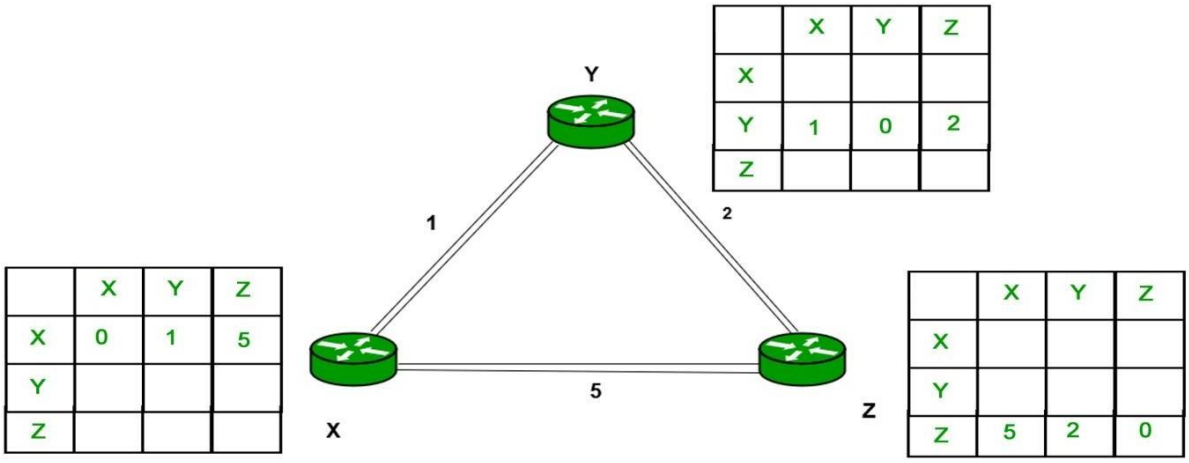
A **distance-vector routing (DVR)** protocol requires that a router inform its neighbors of topology changes periodically. Historically known as the old ARPANET routing algorithm (or known as Bellman-Ford algorithm).

Bellman Ford Basics – Each router maintains a Distance Vector table containing the distance between itself and ALL possible destination nodes. Distances, based on a chosen metric, are computed using information from the neighbors' distance vectors.

Distance Vector Algorithm –

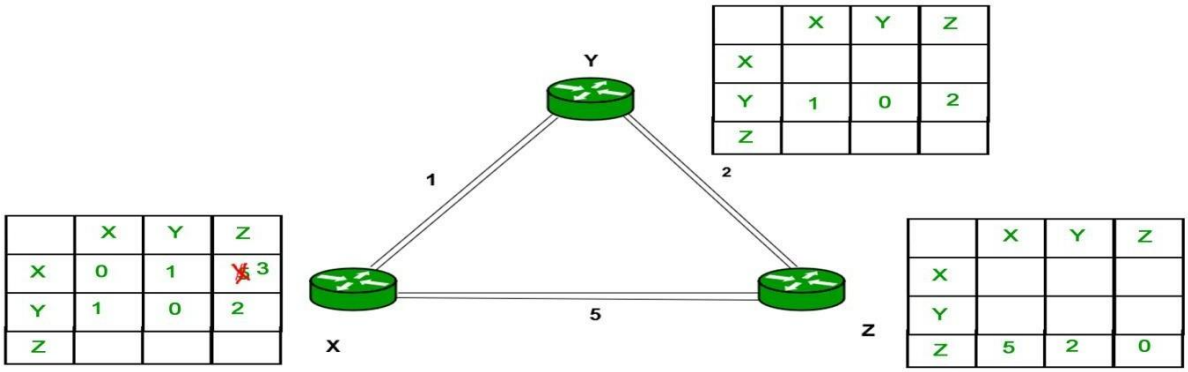
1. A router transmits its distance vector to each of its neighbors in a routing packet.
2. Each router receives and saves the most recently received distance vector from each of its neighbors.
3. A router recalculates its distance vector when:
    - It receives a distance vector from a neighbor containing different information than before.
    - It discovers that a link to a neighbor has gone down.

Example – Consider 3-routers X, Y and Z as shown in figure. Each router have their routing table. Every routing table will contain distance to the destination nodes.

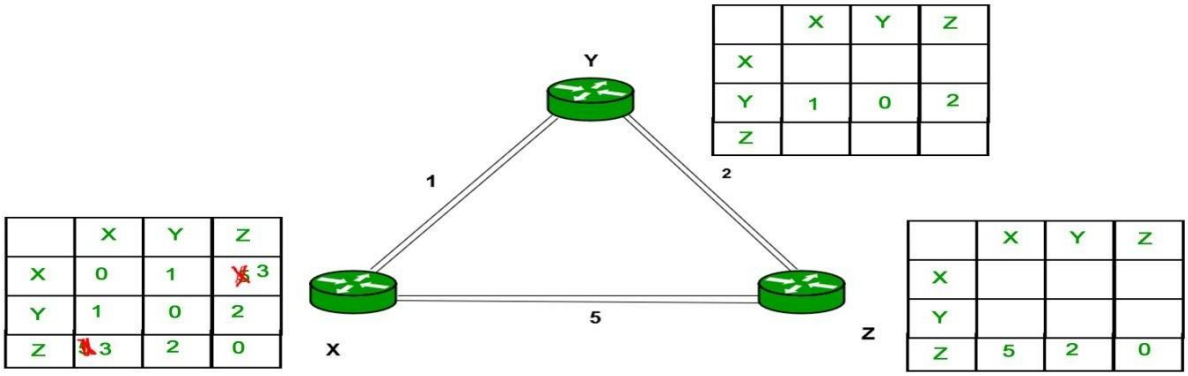


Consider router X , X will share it routing table to neighbors and neighbors will share it routing table to it to X and distance from node X to destination will be calculated using bellmen- ford equation.

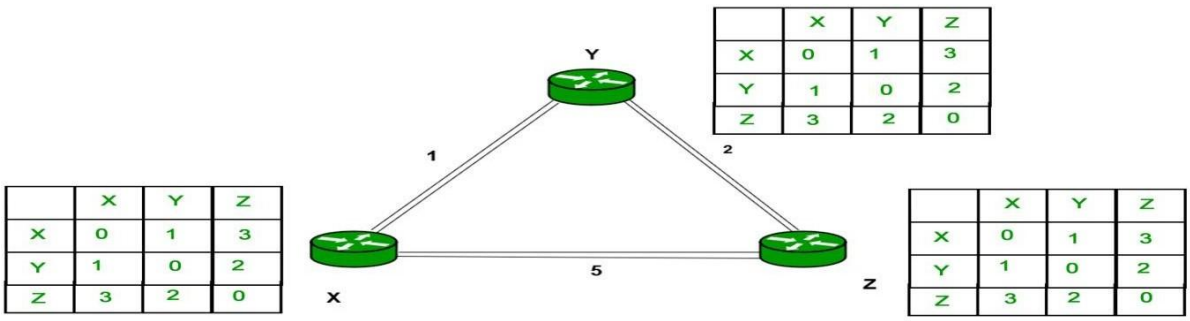
As we can see that distance will be less going from X to Z when Y is intermediate node(hop) so it will be update in routing table X.



Similarly for Z also –



Finally the routing table for all –



Advantages of Distance Vector routing –

- It is simpler to configure and maintain than link state routing.

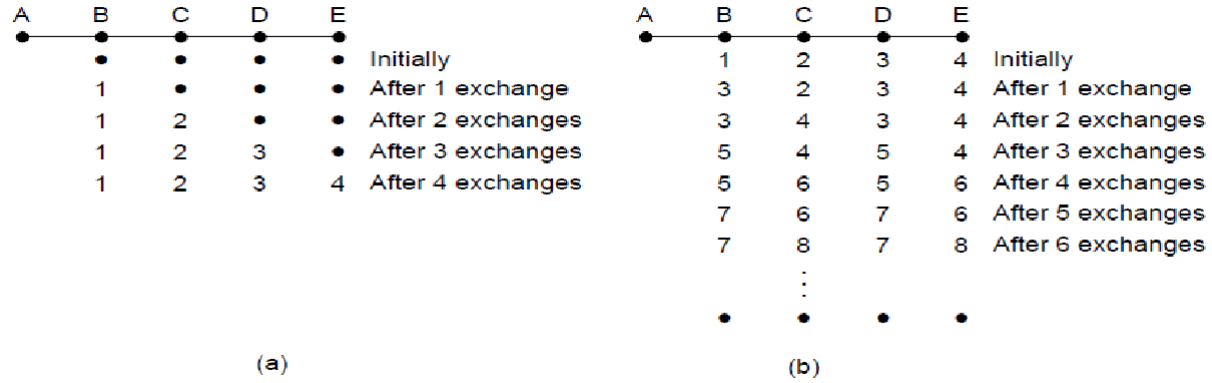
Disadvantages of Distance Vector routing –

- It is slower to converge than link state.

- It is at risk from the count-to-infinity problem.
- It creates more traffic than link state since a hop count change must be propagated to all routers and processed on each router. Hop count updates take place on a periodic basis, even if there are no changes in the network topology, so bandwidth-wasting broadcasts still occur.
- For larger networks, distance vector routing results in larger routing tables than link state since each router must know about all other routers. This can also lead to congestion on WAN links.

5. Count to infinity problem:

1. One of the important issue in Distance Vector Routing is County of Infinity Problem.
2. Counting to infinity is just another name for a routing loop.
3. In distance vector routing, routing loops usually occur when an interface goes down.
4. It can also occur when two routers send updates to each other at the same time.



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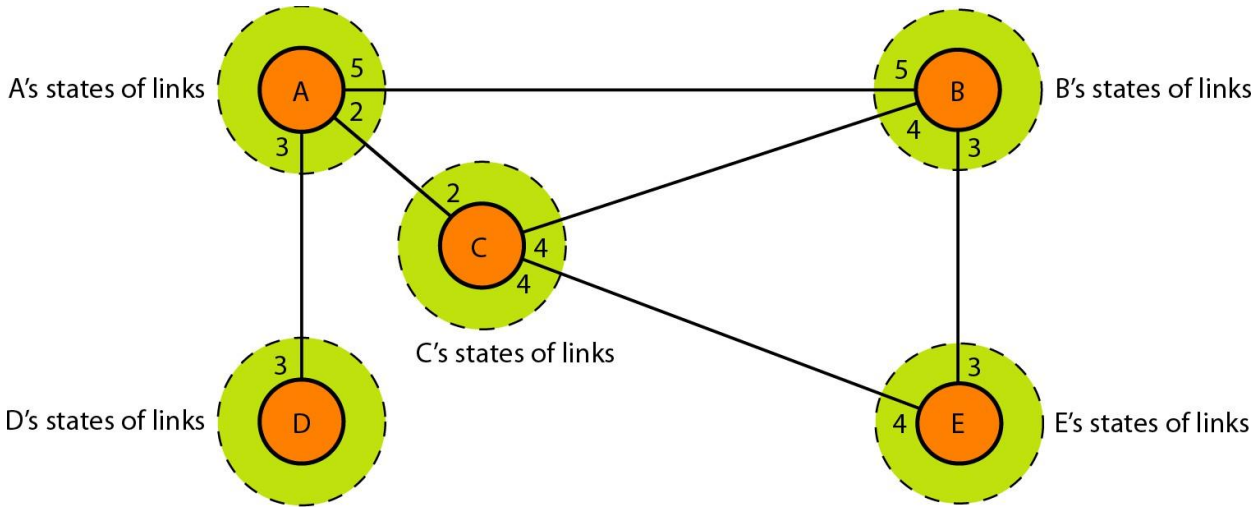
6. Link State Routing

Link state routing is a technique in which each router shares the knowledge of its neighborhood with every other router in the internetwork.

The three keys to understand the Link State Routing algorithm:

- **Knowledge about the neighborhood:** Instead of sending its routing table, a router sends the information about its neighborhood only. A router broadcast its identities and cost of the directly attached links to other routers.
- **Flooding:** Each router sends the information to every other router on the internetwork except its neighbors. This process is known as Flooding. Every router that receives the packet sends the copies to all its neighbors. Finally, each and every router receives a copy of the same information.
- **Information sharing:** A router sends the information to every other router only when the change occurs in the information.



7. Hierarchical Routing Algorithm:

In hierarchical routing, the routers are divided into regions. Each router has complete details about how to route packets to destinations within its own region. But it does not have any idea about the internal structure of other regions.

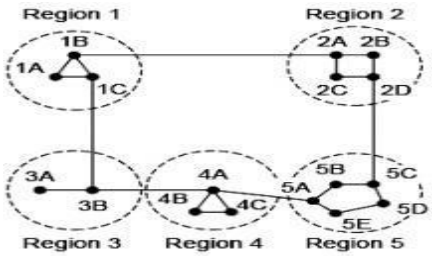
When network size is growing, the number of routers in the network will increase. Therefore, the size of routing table increases, then routers cannot handle network traffic as efficiently. To overcome this problem we are using hierarchical routing.

In hierarchical routing, routers are classified in groups called regions. Each router has information about the routers in its own region and it has no information about routers in other regions. So, routers save one record in their table for every other region.

For huge networks, a two-level hierarchy may be insufficient hence, it may be necessary to group the regions into clusters, the clusters into zones, the zones into groups and so on.

Example

Consider an example of two-level hierarchy with five regions as shown in figure −



Let see the full routing table for router 1A which has 17 entries, as shown below −

Full Table for 1A

Dest.	Line	Hops
1A	-	-
1B	1B	1
1C	1C	1
2A	1B	2
2B	1B	3
2C	1B	3
2D	1B	4
3A	1C	3
3B	1C	2
4A	1C	3
4B	1C	4
4C	1C	4
5A	1C	4
5B	1C	5
5C	1B	5
5D	1C	6
5E	1C	5

When routing is done hierarchically then there will be only 7 entries as shown below −

Hierarchical Table for 1A

Dest.	Line	Hops
1A	-	-
1B	1B	1
1C	1C	1
2	1B	2
3	1C	2
4	1C	3
5	1C	4

Unfortunately, this reduction in table space comes with the increased path length.

General Principles of Congestion Control

Congestion control refers to the techniques used to control or prevent congestion. Congestion control techniques can be broadly classified into two categories:

Open Loop Congestion Control

Open loop congestion control policies are applied to prevent congestion before it happens. The congestion control is handled either by the source or the destination.

Policies adopted by open loop congestion control –

1. **Retransmission Policy:**
   It is the policy in which retransmission of the packets are taken care of. If the sender feels that a sent packet is lost or corrupted, the packet needs to be retransmitted. This transmission may increase the congestion in the network.
   To prevent congestion, retransmission timers must be designed to prevent congestion and also able to optimize efficiency.

2. **Window Policy:**
   The type of window at the sender's side may also affect the congestion. Several packets in the Go-back-n window are re-sent, although some packets may be received successfully at the receiver side. This duplication may increase the congestion in the network and make it worse.
   Therefore, Selective repeat window should be adopted as it sends the specific packet that may have been lost.

3. **Discarding Policy:**
   A good discarding policy adopted by the routers is that the routers may prevent congestion and at the same time partially discard the corrupted or less sensitive packages and also be able to maintain the quality of a message.
   In case of audio file transmission, routers can discard less sensitive packets to prevent congestion and also maintain the quality of the audio file.

4. **Acknowledgment Policy:**
   Since acknowledgements are also the part of the load in the network, the acknowledgment policy imposed by the receiver may also affect congestion. Several approaches can be used to prevent congestion related to acknowledgment.
   The receiver should send acknowledgement for N packets rather than sending acknowledgement for a single packet. The receiver should send an acknowledgment only if it has to send a packet or a timer expires.

5. **Admission Policy:**
   In admission policy a mechanism should be used to prevent congestion. Switches in a flow should first check the resource requirement of a network flow before transmitting it further. If there is a chance of a congestion or there is a congestion in the network, router should deny establishing a virtual network connection to prevent further congestion.

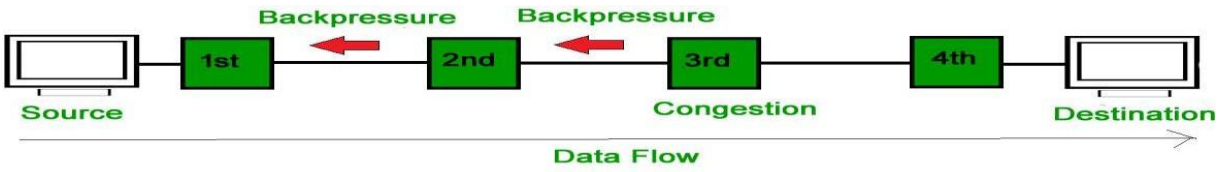
All the above policies are adopted to prevent congestion before it happens in the network.

Closed Loop Congestion Control

Closed loop congestion control techniques are used to treat or alleviate congestion after it happens. Several techniques are used by different protocols; some of them are:

1. Backpressure:

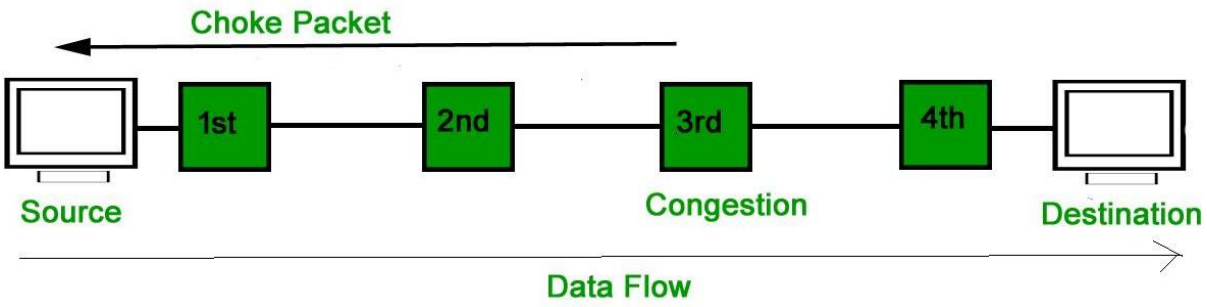
Backpressure is a technique in which a congested node stops receiving packets from upstream node. This may cause the upstream node or nodes to become congested and reject receiving data from above nodes. Backpressure is a node-to-node congestion control technique that propagates in the opposite direction of data flow. The backpressure technique can be applied only to virtual circuit where each node has information of its above upstream node.



In above diagram the 3rd node is congested and stops receiving packets as a result 2nd node may be get congested due to slowing down of the output data flow. Similarly 1st node may get congested and inform the source to slow down.

2. Choke Packet Technique:

Choke packet technique is applicable to both virtual networks as well as datagram subnets. A choke packet is a packet sent by a node to the source to inform it of congestion. Each router monitors its resources and the utilization at each of its output lines. Whenever the resource utilization exceeds the threshold value which is set by the administrator, the router directly sends a choke packet to the source giving it a feedback to reduce the traffic. The intermediate nodes through which the packets has traveled are not warned about congestion.



3. Implicit Signaling:

In implicit signaling, there is no communication between the congested nodes and the source. The source guesses that there is congestion in a network. For example when sender sends several packets and there is no acknowledgment for a while, one assumption is that there is congestion.

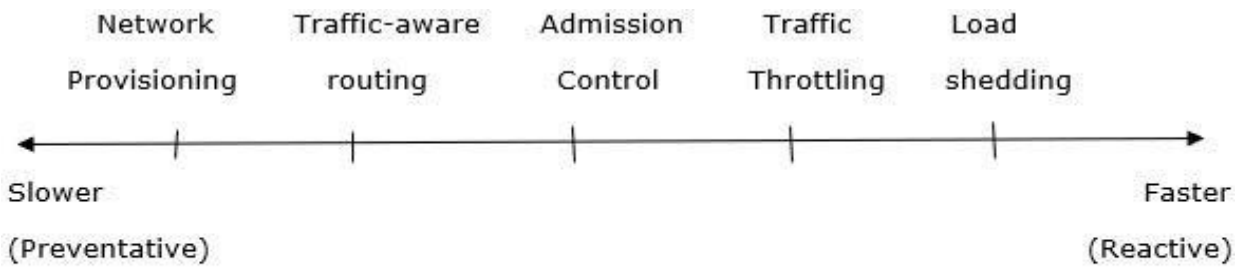
4. Explicit Signaling:

In explicit signaling, if a node experiences congestion it can explicitly sends a packet to the source or destination to inform about congestion. The difference between choke packet and explicit signaling is that the signal is included in the packets that carry data rather than creating a different packet as in case of choke packet technique.

Explicit signaling can occur in either forward or backward direction.

- **Forward Signaling:** In forward signaling, a signal is sent in the direction of the congestion. The destination is warned about congestion. The receiver in this case adopt policies to prevent further congestion.
- **Backward Signaling:** In backward signaling, a signal is sent in the opposite direction of the congestion. The source is warned about congestion and it needs to slow down.

Approaches to Congestion Control-



Time scale of approaches to congestion control

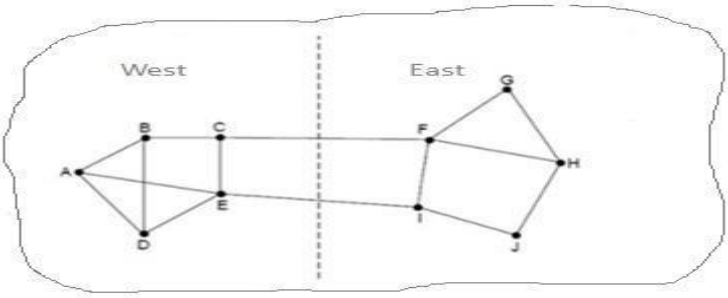
Traffic Aware Routing

Traffic awareness is one of the approaches for congestion control over the network. The basic way to avoid congestion is to build a network that is well matched to the traffic that it carries. If more traffic is directed but a low-bandwidth link is available, congestion occurs.

The main goal of traffic aware routing is to identify the best routes by considering the load, set the link weight to be a function of fixed link bandwidth and propagation delay and the variable measured load or average queuing delay.

Least-weight paths will then favor paths that are more lightly loaded, remaining all are equal.

The traffic aware routing is diagrammatically represented as follows −



Explanation

Step 1 − Consider a network which is divided into two parts, East and West both are connected by links CF and EI.

Step 2 − Suppose most of the traffic in between East and West is using link CF, and as a result CF link is heavily loaded with long delays. Including queueing delay in the weight which is used for shortest path calculation will make EI more attractive.

Step 3 − After installing the new routing tables, most of East-West traffic will now go over the EI link. As a result in the next update CF link will appear to be the shortest path.

Step 4 − As a result the routing tables may oscillate widely, leading to erratic routing and many potential problems.

Step 5 − If we consider only bandwidth and propagation delay by ignoring the load, this problem does not occur. Attempts to include load but change the weights within routing scheme to shift traffic across routes arrow range only to slow down routing oscillations.

Step 6 − Two techniques can contribute for successful solution, which are as follows −

- Multipath routing
- The routing scheme to shift traffic across routes.

Features

- It is one of the congestion control techniques..
- Roots can be changed to shift traffic away because of heavily used paths.
- Network Traffic can be split across multiple paths.

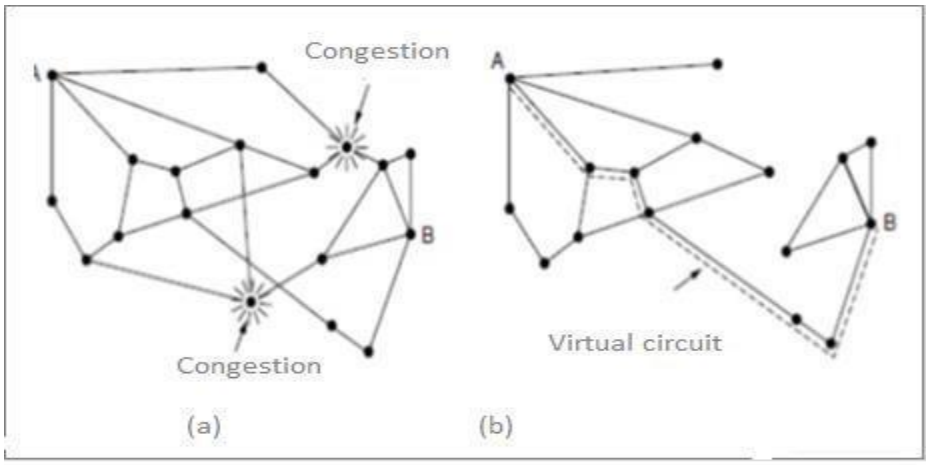
Admission Control

It is one of techniques that is widely used in virtual-circuit networks to keep congestion at bay. The idea is do not set up a new virtual circuit unless the network can carry the added traffic without becoming congested.

Admission control can also be combined with traffic aware routing by considering routes around traffic hotspots as part of the setup procedure.

Example

Take two networks (a) A congestion network and (b) The portion of the network that is not congested. A virtual circuit A to B is also shown below −



Explanation

Step 1 − Suppose a host attached to router A wants to set up a connection to a host attached to router B. Normally this connection passes through one of the congested routers.

Step 2 − To avoid this situation, we can redraw the network as shown in figure (b), removing the congested routers and all of their lines.

Step 3 − the dashed line indicates a possible route for the virtual circuit that avoids the congested routers.

Traffic throttling:

It is one of the approaches for congestion control. In the internet and other computer networks, senders trying to adjust the transmission need to send as much traffic as the network can readily deliver. In this setting the network aim is to operate just before the onset of congestion.

Load Shedding

It is one of the approaches to congestion control. Router contains a buffer to store packets and route it to destination. When the buffer is full, it simply discards some packets. It chooses the packet to be discarded based on the strategy implemented in the data link layer. This is called load shedding

Load shedding will use dropping the old packets than new to avoid congestion. Dropping packets that are part of the difference is preferable because a future packet depends on full frame.

Advantages

The advantages of load shedding are given below −

- It can be used in detection of congestion.
- It can recover from congestion.
- It reduces the network traffic flow.
- Synchronized flow of packets across a network.
- Removes the packets before congestion occurs.

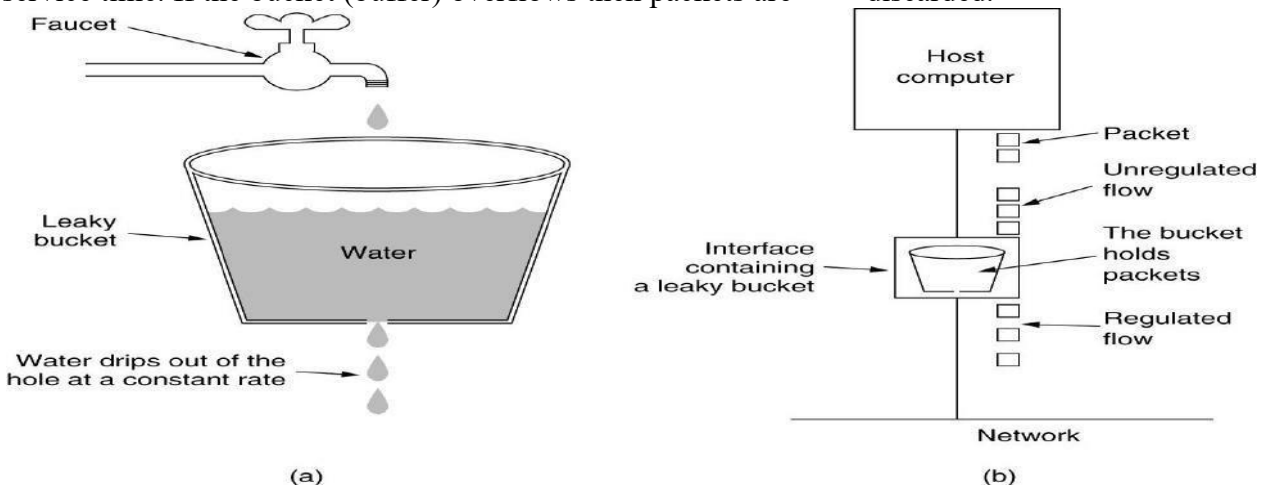
Disadvantages

The disadvantages of load shedding are given below −

- Packets get lost because of discarding by the router.
- If buffer size is less it results in more packets to get discarded.
- Cannot ensure congestion avoidance.
- Overhead for the router to always keep on checking whether the buffer is full.

Leaky Bucket Algorithm:

It is used to control rate in a network. It is implemented as a single server queue with constant service time. If the bucket (buffer) overflows then packets are discarded.



1. The leaky bucket enforces a constant output rate

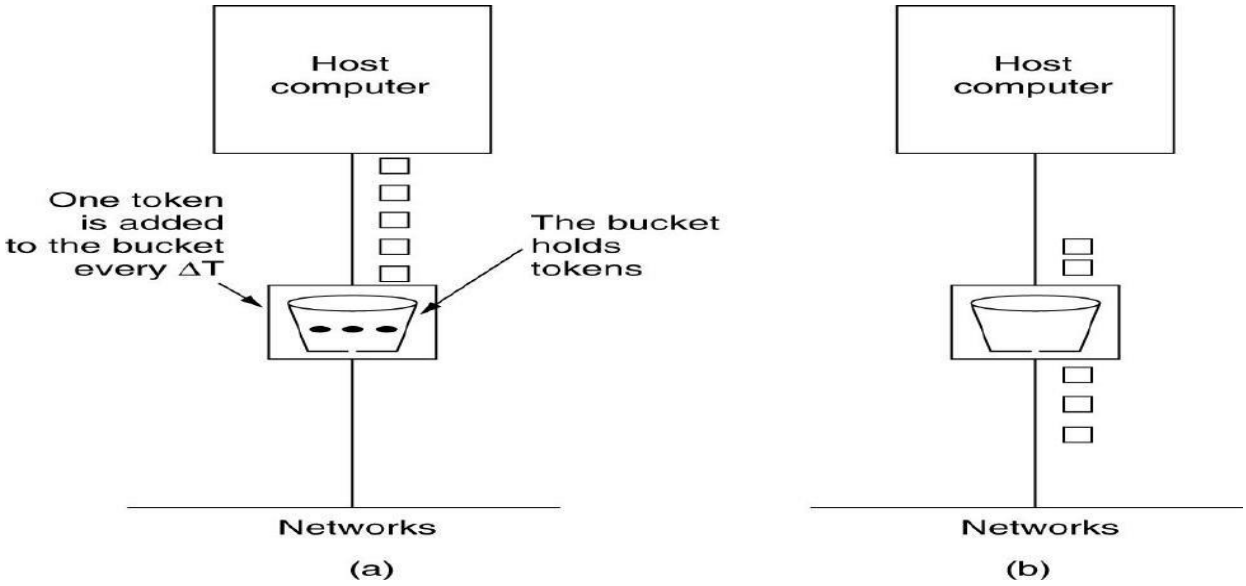
2. The host injects one packet per clock tick onto the network. This results in a uniform flow

of packets, smoothing out bursts and reducing congestion.

3. When packets are the same size the one packet per tick is okay. For variable length packets though, it is better to allow a fixed number of bytes per tick.

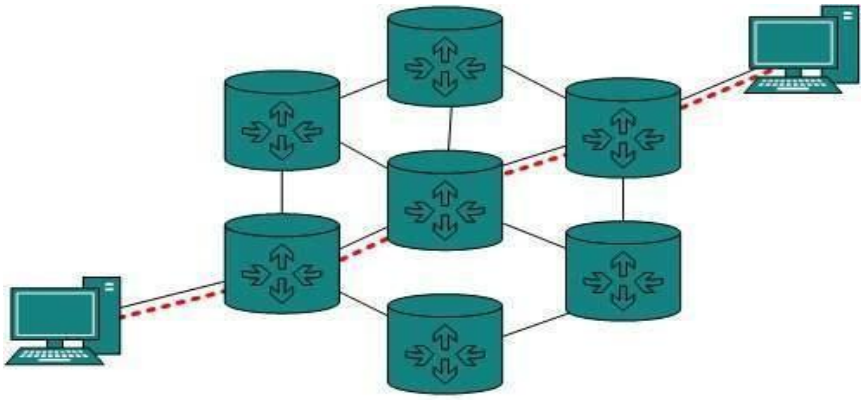
Token Bucket Algorithm

1. In contrast to the LB, the Token Bucket Algorithm, allows the output rate to vary,depending on the size of the burst.
2. In the TB algorithm, the bucket holds tokens. To transmit a packet, the host must capture and destroy one token.
3. Tokens are generated by a clock at the rate of one token every □t sec.
4. Idle hosts can capture and save up tokens (up to the max. size of the bucket) in order to send larger bursts later.



Internetworking:

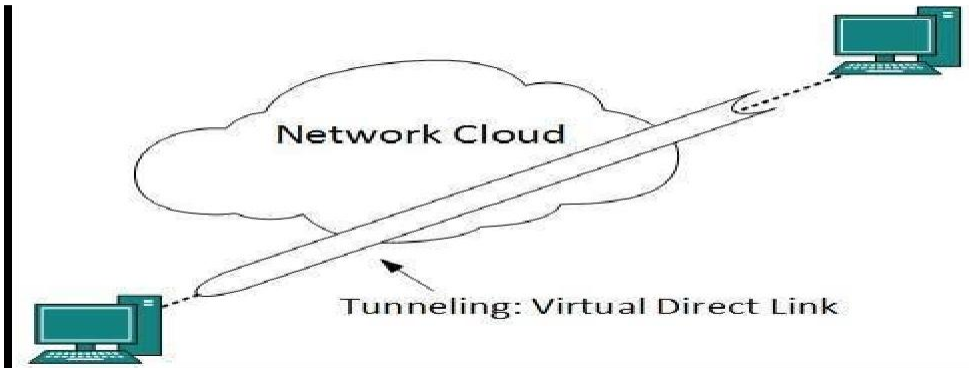
In internetworking, routers have knowledge of each other's address and addresses beyond them. They can be statically configured go on different network or they can learn by using internetworking routing protocol.



Tunneling:

If they are two geographically separate networks, which want to communicate with each other, they may deploy a dedicated line between or they have to pass their data through intermediate networks.

Tunneling is a mechanism by which two or more same networks communicate with each other, by passing intermediate networking complexities. Tunneling is configured at both ends.



When the data enters from one end of Tunnel, it is tagged. This tagged data is then routed inside the intermediate or transit network to reach the other end of Tunnel. When data exists the Tunnel its tag is removed and delivered to the other part of the network.

Both ends seem as if they are directly connected and tagging makes data travel through transit network without any modifications.

Packet Fragmentation:

Most Ethernet segments have their maximum transmission unit (MTU) fixed to 1500 bytes. A data packet can have more or less packet length depending upon the application. Devices in the transit path also have their hardware and software capabilities which tell what amount of data that device can handle and what size of packet it can process.

If the data packet size is less than or equal to the size of packet the transit network can handle, it is processed neutrally. If the packet is larger, it is broken into smaller pieces and then forwarded. This is called packet fragmentation. Each fragment contains the same destination and source address and routed through transit path easily. At the receiving end it is assembled again.

If a packet with DF (don't fragment) bit set to 1 comes to a router which can not handle the packet because of its length, the packet is dropped.

When a packet is received by a router has its MF (more fragments) bit set to 1, the router then knows that it is a fragmented packet and parts of the original packet is on the way.

If packet is fragmented too small, the overhead is increases. If the packet is fragmented too large, intermediate router may not be able to process it and it might get dropped.

Internet Protocol Version 4 (IPv4)

Internet Protocol is one of the major protocols in the TCP/IP protocols suite. This protocol works at the network layer of the OSI model and at the Internet layer of the TCP/IP model. Thus this protocol has the responsibility of identifying hosts based upon their logical addresses and to route data among them over the underlying network.

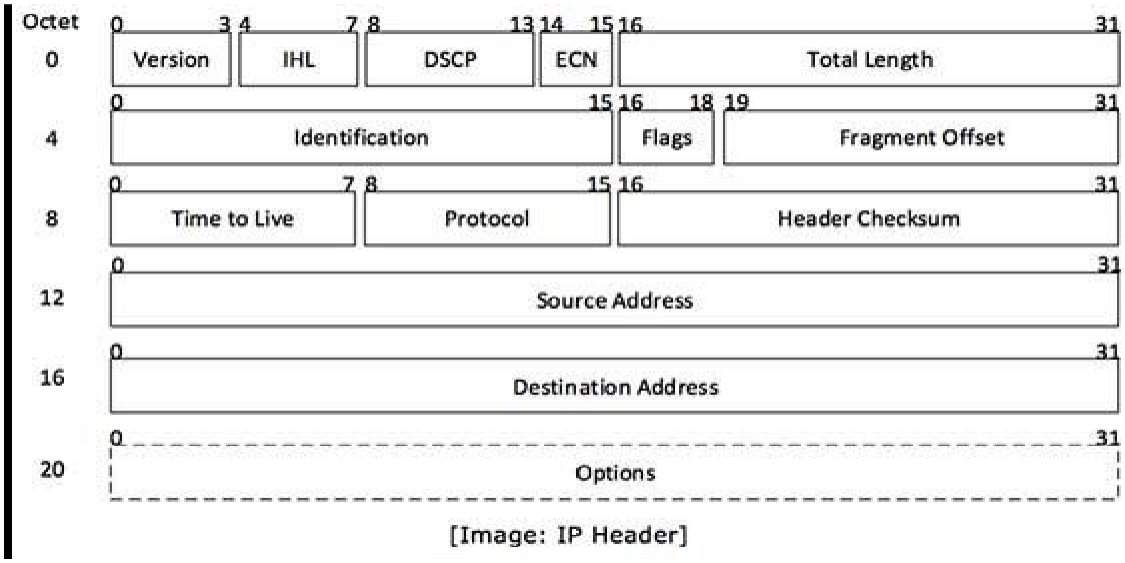
IP provides a mechanism to uniquely identify hosts by an IP addressing scheme. IP uses best effort delivery, i.e. it does not guarantee that packets would be delivered to the destined host, but it will do its best to reach the destination. Internet Protocol version 4 uses 32-bit logical address.

IPv4 - Packet Structure

Internet Protocol being a layer-3 protocol (OSI) takes data Segments from layer-4 (Transport) and divides it into packets. IP packet encapsulates data unit received from above layer and add to its own header information.

(IP Encapsulation)

The encapsulated data is referred to as IP Payload. IP header contains all the necessary information to deliver the packet at the other end.



[Image: IP Header]

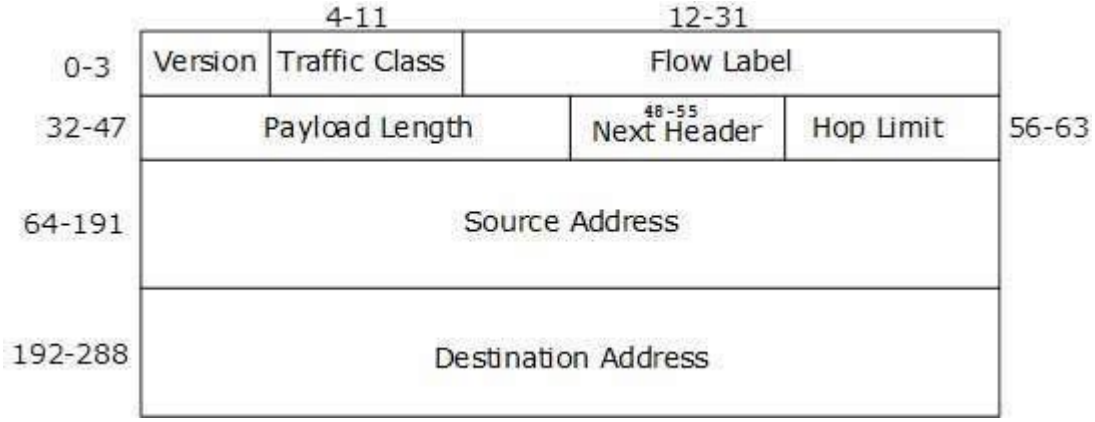
IP header includes many relevant information including Version Number, which, in this context, is 4. Other details are as follows −

- **Version** − Version no. of Internet Protocol used (e.g. IPv4).
- **IHL** − Internet Header Length; Length of entire IP header.
- **DSCP** − Differentiated Services Code Point; this is Type of Service.
- **ECN** − Explicit Congestion Notification; It carries information about the congestion seen in the route.
- **Total Length** − Length of entire IP Packet (including IP header and IP Payload).

- **Identification** − If IP packet is fragmented during the transmission, all the fragments contain same identification number. to identify original IP packet they belong to.
- **Flags** − As required by the network resources, if IP Packet is too large to handle, these 'flags' tells if they can be fragmented or not. In this 3-bit flag, the MSB is always set to '0'.
- **Fragment Offset** − this offset tells the exact position of the fragment in the original IP Packet.
- **Time to Live** − To avoid looping in the network, every packet is sent with some TTL value set, which tells the network how many routers (hops) this packet can cross. At each hop, its value is decremented by one and when the value reaches zero, the packet is discarded.
- **Protocol** − Tells the Network layer at the destination host, to which Protocol this packet belongs to, i.e. the next level Protocol. For example protocol number of ICMP is 1, TCP is 6 and UDP is 17.
- **Header Checksum** − This field is used to keep checksum value of entire header which is then used to check if the packet is received error-free.
- **Source Address** − 32-bit address of the Sender (or source) of the packet.
- **Destination Address** − 32-bit address of the Receiver (or destination) of the packet.
- **Options** − This is optional field, which is used if the value of IHL is greater than 5. These options may contain values for options such as Security, Record Route, Time Stamp, etc.

IPV6 Header:

Fixed Header



IPv6 fixed header is 40 bytes long and contains the following information.

S.N.	Field & Description
1	**Version** (4-bits): It represents the version of Internet Protocol, i.e. 0110.
2	**Traffic Class** (8-bits): These 8 bits are divided into two parts. The most significant 6 bits are used for Type of Service to let the Router Known what services should be provided to this packet. The least significant 2 bits are used for Explicit Congestion Notification (ECN).
3	**Flow Label** (20-bits): This label is used to maintain the sequential flow of the packets belonging to a communication. The source labels the sequence to help the router identify that a particular packet belongs to a specific flow of information. This field helps avoid re-ordering of data packets. It is designed for streaming/real-time media.
4	**Payload Length** (16-bits): This field is used to tell the routers how much information a particular packet contains in its payload. Payload is composed of Extension Headers and Upper Layer data. With 16 bits, up to 65535 bytes can be indicated; but if the Extension

Headers contain Hop-by-Hop Extension Header, then the payload may exceed 65535 bytes and this field is set to 0.

5 **Next Header** (8-bits): This field is used to indicate either the type of Extension Header, or if the Extension Header is not present then it indicates the Upper Layer PDU. The values for the type of Upper Layer PDU are same as IPv4's.

6 **Hop Limit** (8-bits): This field is used to stop packet to loop in the network infinitely. This is same as TTL in IPv4. The value of Hop Limit field is decremented by 1 as it passes a link (router/hop). When the field reaches 0 the packet is discarded.

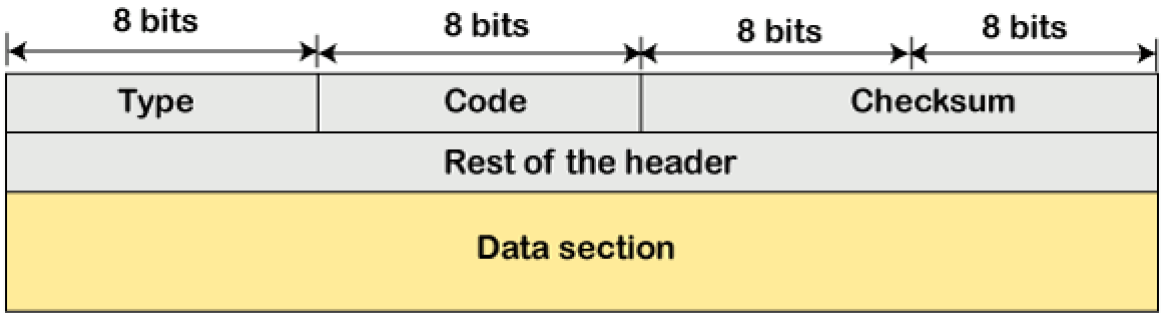
7 **Source Address** (128-bits): This field indicates the address of originator of the packet.

8 **Destination Address** (128-bits): This field provides the address of intended recipient of the packet.

ICMP Protocol

The ICMP stands for Internet Control Message Protocol. It is a network layer protocol. It is used for error handling in the network layer, and it is primarily used on network devices such as routers. As different types of errors can exist in the network layer, so ICMP can be used to report these errors and to debug those errors.

ICMP Message Format:



- **Type:** It is an 8-bit field. It defines the ICMP message type. The values range from 0 to 127 are defined for ICMPv6, and the values from 128 to 255 are the informational messages.
- **Code:** It is an 8-bit field that defines the subtype of the ICMP message
- **Checksum:** It is a 16-bit field to detect whether the error exists in the message or not.

Address Resolution Protocol (ARP):

Address Resolution Protocol (ARP) is a communication protocol used to find the MAC (Media Access Control) address of a device from its IP address. This protocol is used when a device wants to communicate with another device on a Local Area Network or Ethernet.

There are four types of Address Resolution Protocol, which is given below:

- Proxy ARP
- Gratuitous ARP
- Reverse ARP (RARP)
- Inverse ARP

Dynamic Host Configuration Protocol

Dynamic Host Configuration Protocol (DHCP) is a network management protocol used to dynamically assign an IP address to nay device, or node, on a network so they can communicate using IP (Internet Protocol). DHCP automates and centrally manages these configurations. There is no need to manually assign IP addresses to new devices. Therefore, there is no requirement for any user configuration to connect to a DHCP based network.

DHCP can be implemented on local networks as well as large enterprise networks. DHCP is the default protocol used by the most routers and networking equipment. DHCP is also called RFC (Request for comments) 2131.

Components of DHCP

When working with DHCP, it is important to understand all of the components. Following are the list of components:

- **DHCP Server:** DHCP server is a networked device running the DCHP service that holds IP addresses and related configuration information. This is typically a server or a router but could be anything that acts as a host, such as an SD-WAN appliance.
- **DHCP client:** DHCP client is the endpoint that receives configuration information from a DHCP server. This can be any device like computer, laptop, IoT endpoint or anything else that requires connectivity to the network. Most of the devices are configured to receive DHCP information by default.
- **IP address pool:** IP address pool is the range of addresses that are available to DHCP clients. IP addresses are typically handed out sequentially from lowest to the highest.
- **Subnet:** Subnet is the partitioned segments of the IP networks. Subnet is used to keep networks manageable.
- **Lease:** Lease is the length of time for which a DHCP client holds the IP address information. When a lease expires, the client has to renew it.
- **DHCP relay:** A host or router that listens for client messages being broadcast on that network and then forwards them to a configured server. The server then sends responses back to the relay agent that passes them along to the client. DHCP relay can be used to centralize DHCP servers instead of having a server on each subnet.