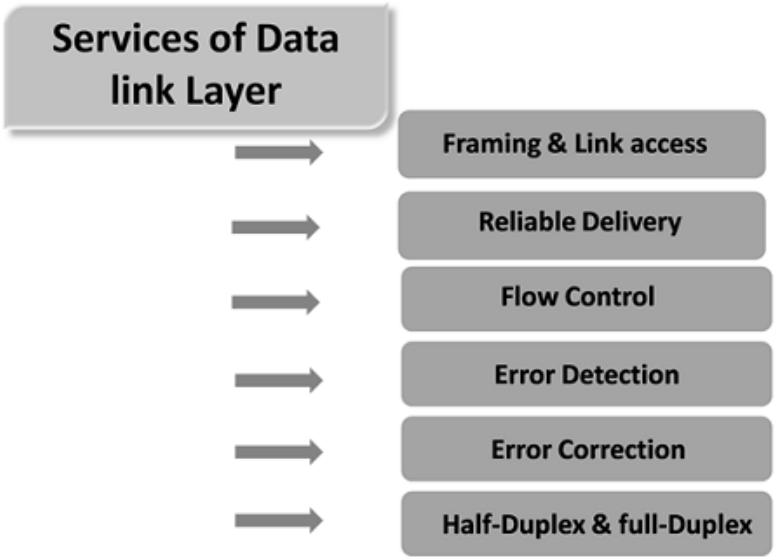
Unit-2

Data Link Layer:

- In the OSI model, the data link layer is a 4th layer from the top and 2nd layer from the bottom.
- The communication channel that connects the adjacent nodes is known as links and in order to move the datagram from source to the destination, the datagram must be moved across an individual link.
- The main responsibility of the Data Link Layer is to transfer the datagram across an individual link.
- The Data link layer protocol defines the format of the packet exchanged across the nodes as well as the actions such as Error detection, retransmission, flow control and random access.
- The Data Link Layer protocols are Ethernet, token ring, FDDI and PPP.
- An important characteristic of a Data Link Layer is that datagram can be handled by different link layer protocols on different links in a path. For example, the datagram is handled by Ethernet on the first link, PPP on the second link.

Following services are provided by the Data Link Layer:



- **Framing & Link access:** Data Link Layer protocols encapsulate each network frame within a Link layer frame before the transmission across the link. A frame consists of a data field in which network layer datagram is inserted and a number of data fields. It specifies the structure of the frame as well as a channel access protocol by which frame is to be transmitted over the link.
- **Reliable delivery:** Data Link Layer provides a reliable delivery service, i.e., transmits the network layer datagram without any error. A reliable delivery service is accomplished with transmissions and acknowledgements. A data link layer mainly provides the reliable delivery service over the links as they have higher error rates and they can be corrected locally, link at which an error occurs rather than forcing to retransmit the data.

- **Flow control:** A receiving node can receive the frames at a faster rate than it can process the frame. Without flow control, the receiver's buffer can overflow, and frames can get lost. To overcome this problem, the data link layer uses the flow control to prevent the sending node on one side of the link from overwhelming the receiving node on another side of the link.
- **Error detection:** Errors can be introduced by signal attenuation and noise. Data Link Layer protocol provides a mechanism to detect one or more errors. This is achieved by adding error detection bits in the frame and then receiving node can perform an error check.
- **Error correction:** Error correction is similar to the Error detection, except that receiving node not only detects the errors but also determine where the errors have occurred in the frame.
- **Half-Duplex & Full-Duplex:** In a Full-Duplex mode, both the nodes can transmit the data at the same time. In a Half-Duplex mode, only one node can transmit the data at the same time.

Data link layer design issues:

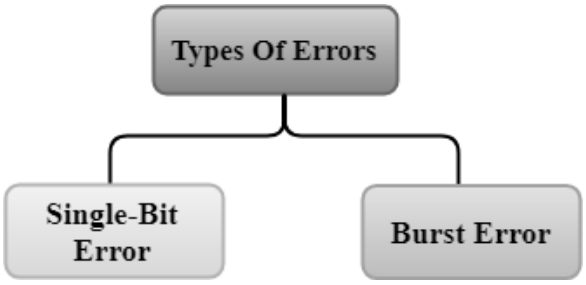
Design issues with data link layer are:

1. **Services provided to the network layer:**
   The data link layer act as a service interface to the network layer. The principle service is transferring data from network layer on sending machine to the network layer on destination machine. This transfer also takes place via DLL (Data link-layer).
2. **Frame synchronization:**
   The source machine sends data in the form of blocks called frames to the destination machine. The starting and ending of each frame should be identified so that the frame can be recognized by the destination machine.
3. **Flow control:**
   Flow control is done to prevent the flow of data frame at the receiver end. The source machine must not send data frames at a rate faster than the capacity of destination machine to accept them.
4. **Error control:**
   Error control is done to prevent duplication of frames. The errors introduced during transmission from source to destination machines must be detected and corrected at the destination machine.

Error Detection:

When data is transmitted from one device to another device, the system does not guarantee whether the data received by the device is identical to the data transmitted by another device. An Error is a situation when the message received at the receiver end is not identical to the message transmitted.

Types of Errors:

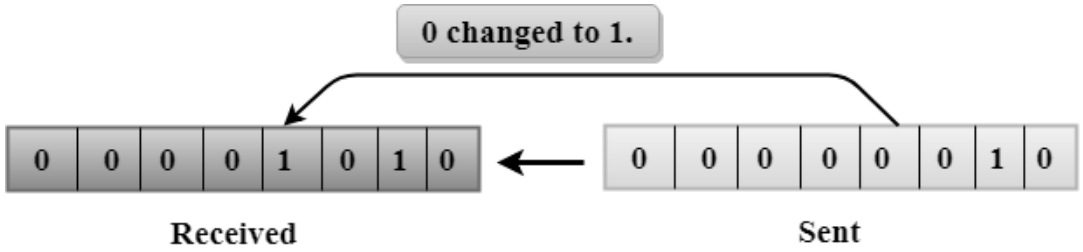


Errors can be classified into two categories:

1. Single-Bit Error
2. Burst Error

Single-Bit Error:

The only one bit of a given data unit is changed from 1 to 0 or from 0 to 1.



In the above figure, the message which is sent is corrupted as single-bit, i.e., 0 bit is changed to 1.

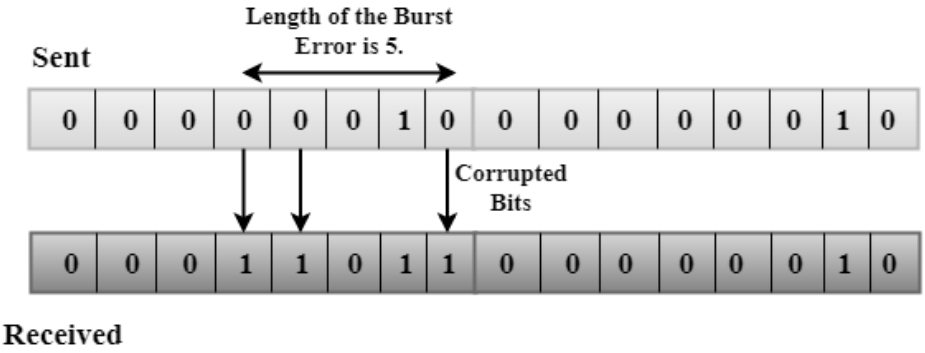
Single-Bit Error does not appear more likely in Serial Data Transmission. For example, Sender sends the data at 10 Mbps, this means that the bit lasts only for 1's and for a single-bit error to occurred, a noise must be more than 1's.

Single-Bit Error mainly occurs in Parallel Data Transmission. For example, if eight wires are used to send the eight bits of a byte, if one of the wire is noisy, then single-bit is corrupted per byte.

Burst Error:

The two or more bits are changed from 0 to 1 or from 1 to 0 is known as Burst Error.

The Burst Error is determined from the first corrupted bit to the last corrupted bit.



The duration of noise in Burst Error is more than the duration of noise in Single-Bit.

Burst Errors are most likely to occur in Serial Data Transmission.

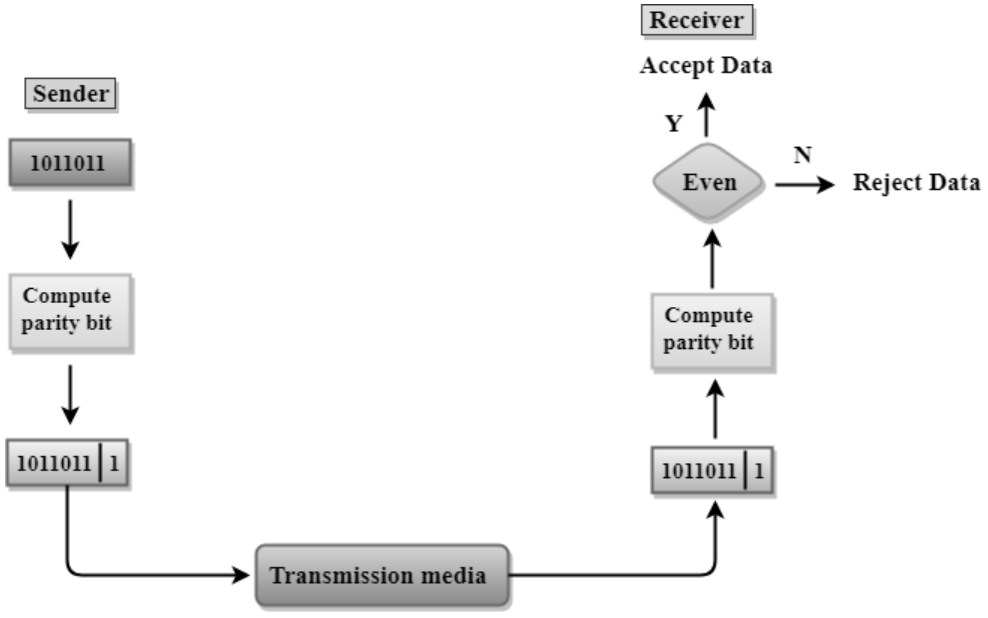
Error Detecting Techniques:

The most popular Error Detecting Techniques are:

1. Single parity check
2. Two-dimensional parity check
3. Checksum
4. Cyclic redundancy check

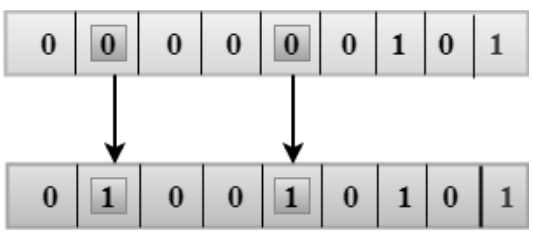
Single Parity Check:

- Single Parity checking is the simple mechanism and inexpensive to detect the errors.
- In this technique, a redundant bit is also known as a parity bit which is appended at the end of the data unit so that the number of 1s becomes even. Therefore, the total number of transmitted bits would be 9 bits.
- If the number of 1s bits is odd, then parity bit 1 is appended and if the number of 1s bits is even, then parity bit 0 is appended at the end of the data unit.
- At the receiving end, the parity bit is calculated from the received data bits and compared with the received parity bit.
- This technique generates the total number of 1s even, so it is known as even-parity checking.



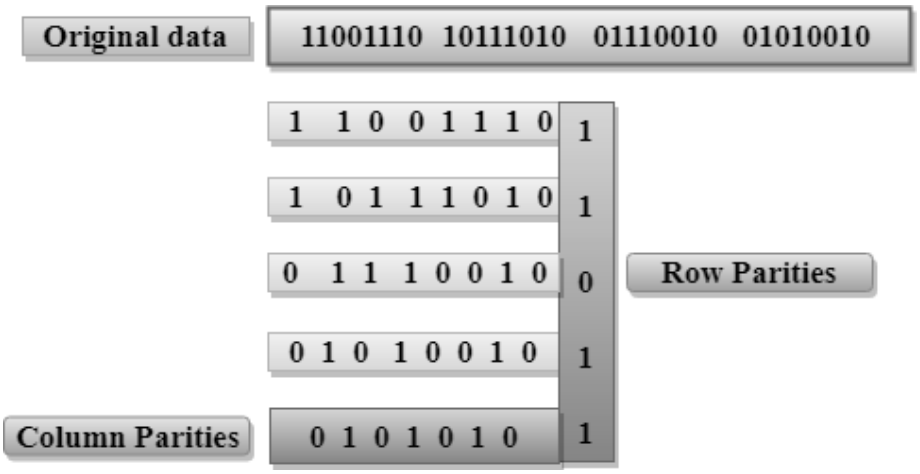
Drawbacks of Single Parity Checking:

- ✓ It can only detect single-bit errors which are very rare.
- ✓ If two bits are interchanged, then it cannot detect the errors.



Two-Dimensional Parity Check:

- Performance can be improved by using Two-Dimensional Parity Check which organizes the data in the form of a table.
- Parity check bits are computed for each row, which is equivalent to the single-parity check.
- In Two-Dimensional Parity check, a block of bits is divided into rows, and the redundant row of bits is added to the whole block.
- At the receiving end, the parity bits are compared with the parity bits computed from the received data.



Drawbacks of 2D Parity Check:

- If two bits in one data unit are corrupted and two bits exactly the same position in another data unit is also corrupted, then 2D Parity checker will not be able to detect the error.
- This technique cannot be used to detect the 4-bit errors or more in some cases.

Checksum:

A Checksum is an error detection technique based on the concept of redundancy.

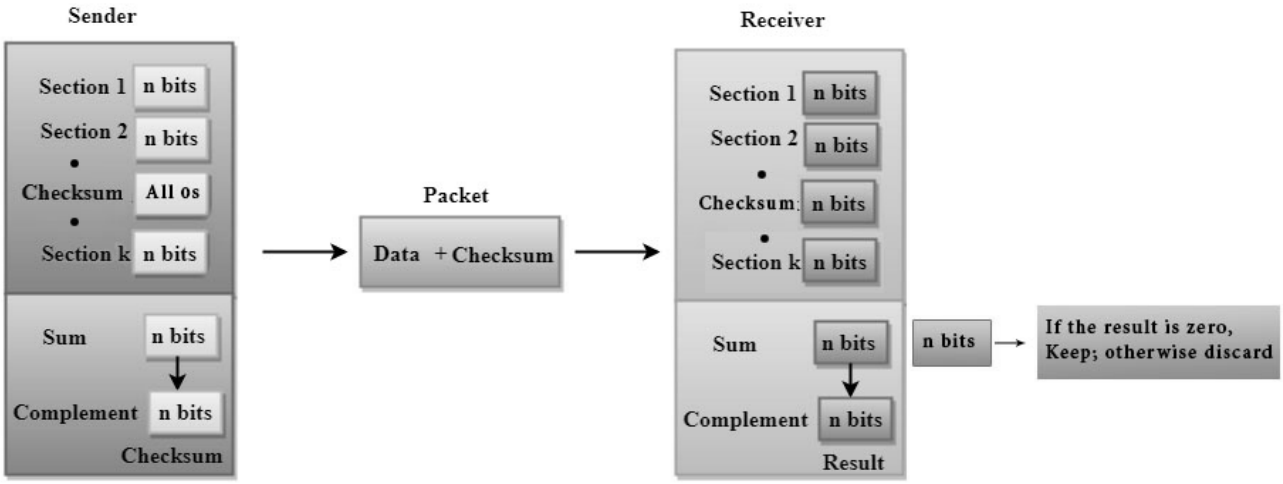
It is divided into two parts:

Checksum Generator:

A Checksum is generated at the sending side. Checksum generator subdivides the data into equal segments of n bits each, and all these segments are added together by using one's complement arithmetic. The sum is complemented and appended to the original data, known as checksum field.

The extended data is transmitted across the network.

Suppose L is the total sum of the data segments, then the checksum would be L.



The Sender follows the given steps:

1. The block unit is divided into k sections, and each of n bits.
2. All the k sections are added together by using one's complement to get the sum.
3. The sum is complemented and it becomes the checksum field.
4. The original data and checksum field are sent across the network.

Checksum Checker:

A Checksum is verified at the receiving side. The receiver subdivides the incoming data into equal segments of n bits each, and all these segments are added together, and then this sum is complemented. If the complement of the sum is zero, then the data is accepted otherwise data is rejected.

The Receiver follows the given steps:

1. The block unit is divided into k sections and each of n bits.
2. All the k sections are added together by using one's complement algorithm to get the sum.
3. The sum is complemented.
4. If the result of the sum is zero, then the data is accepted otherwise the data is discarded.

Cyclic Redundancy Check (CRC):

CRC is a redundancy error technique used to determine the error.

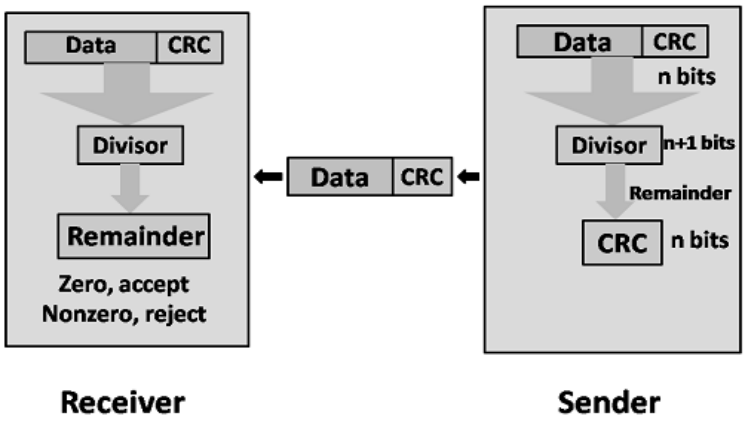
Following are the steps used in CRC for error detection:

- In CRC technique, a string of n 0s is appended to the data unit, and this n number is less than the number of bits in a predetermined number, known as division which is n+1 bits.
- Secondly, the newly extended data is divided by a divisor using a process is known as binary division. The remainder generated from this division is known as CRC remainder.
- Thirdly, the CRC remainder replaces the appended 0s at the end of the original data. This newly generated unit is sent to the receiver.

- The receiver receives the data followed by the CRC remainder. The receiver will treat this whole unit as a single unit, and it is divided by the same divisor that was used to find the CRC remainder.

If the resultant of this division is zero which means that it has no error and the data is accepted.

If the resultant of this division is not zero which means that the data consists of an error. Therefore, the data is discarded.

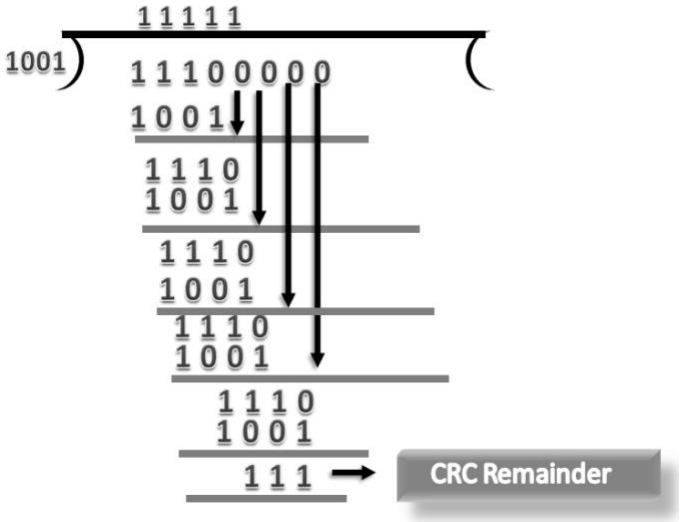


Let's understand this concept through an example:

Suppose the original data is 11100 and divisor is 1001.

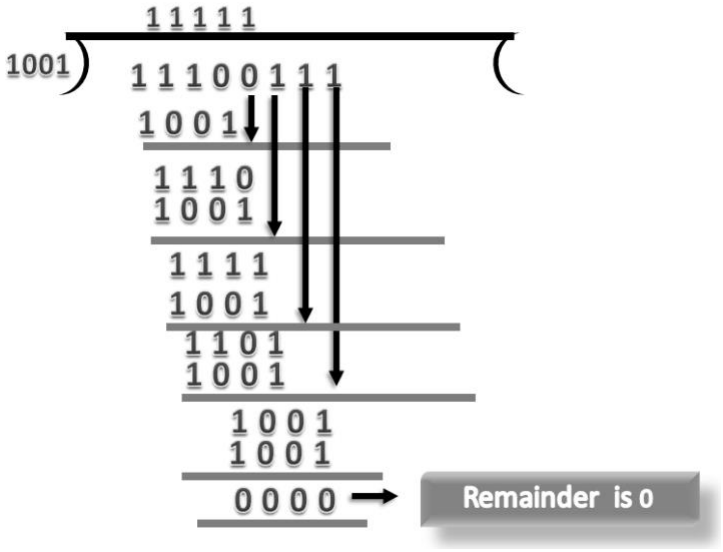
CRC Generator:

- A CRC generator uses a modulo-2 division. Firstly, three zeroes are appended at the end of the data as the length of the divisor is 4 and we know that the length of the string 0s to be appended is always one less than the length of the divisor.
- Now, the string becomes 11100000, and the resultant string is divided by the divisor 1001.
- The remainder generated from the binary division is known as CRC remainder. The generated value of the CRC remainder is 111.
- CRC remainder replaces the appended string of 0s at the end of the data unit, and the final string would be 11100111 which is sent across the network.



CRC Checker:

- The functionality of the CRC checker is similar to the CRC generator.
- When the string 11100111 is received at the receiving end, then CRC checker performs the modulo-2 division.
- A string is divided by the same divisor, i.e., 1001.
- In this case, CRC checker generates the remainder of zero. Therefore, the data is accepted.



Error Correction:

Error Correction codes are used to detect and correct the errors when data is transmitted from the sender to the receiver.

Error Correction can be handled in two ways:

1. **Backward error correction:** Once the error is discovered, the receiver requests the sender to retransmit the entire data unit.
2. **Forward error correction:** In this case, the receiver uses the error-correcting code which automatically corrects the errors.

A single additional bit can detect the error, but cannot correct it.

For correcting the errors, one has to know the exact position of the error. For example, if we want to calculate a single-bit error, the error correction code will determine which one of seven bits is in error. To achieve this, we have to add some additional redundant bits.

Suppose r is the number of redundant bits and d is the total number of the data bits. The number of redundant bits r can be calculated by using the formula:

$$2^r >= d+r+1$$

The value of r is calculated by using the above formula. For example, if the value of d is 4, then the possible smallest value that satisfies the above relation would be 3.

To determine the position of the bit which is in error, a technique developed by R.W Hamming is Hamming code which can be applied to any length of the data unit and uses the relationship between data units and redundant units.

Hamming Code:

Parity bits: The bit which is appended to the original data of binary bits so that the total number of 1s is even or odd.

Even parity: To check for even parity, if the total number of 1s is even, then the value of the parity bit is 0. If the total number of 1s occurrences is odd, then the value of the parity bit is 1.

Odd Parity: To check for odd parity, if the total number of 1s is even, then the value of parity bit is 1. If the total number of 1s is odd, then the value of parity bit is 0.

Algorithm of Hamming code:

- An information of 'd' bits are added to the redundant bits 'r' to form d+r.
- The location of each of the (d+r) digits is assigned a decimal value.
- The 'r' bits are placed in the positions 1,2,.....2^{k-1}.
- At the receiving end, the parity bits are recalculated. The decimal value of the parity bits determines the position of an error.

Relationship b/w Error position & binary number:

Error Position	Binary Number
0	000
1	001
2	010
3	011
4	100
5	101
6	110
7	111

Let's understand the concept of Hamming code through an example:

Suppose the original data is 1010 which is to be sent.

Total number of data bits 'd' = 4

Number of redundant bits r : $2^r >= d+r+1$

$2^r >= 4+r+1$

Therefore, the value of r is 3 that satisfies the above relation.

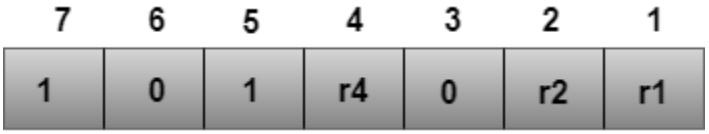
Total number of bits = d+r = 4+3 = 7;

Determining the position of the redundant bits:

The number of redundant bits is 3. The three bits are represented by r1, r2, r4. The position of the redundant bits is calculated with corresponds to the raised power of 2. Therefore, their corresponding positions are 1, 2^1, 2^2.

1. The position of r1 = 1
2. The position of r2 = 2
3. The position of r4 = 4

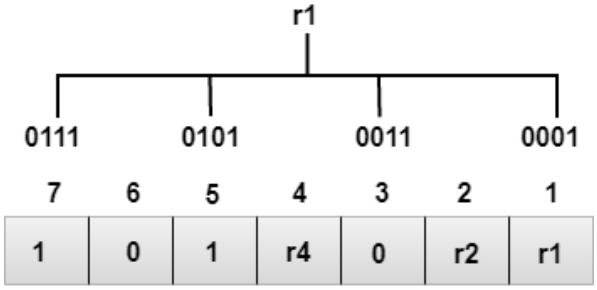
Representation of Data on the addition of parity bits:



Determining the Parity bits:

Determining the r1 bit:

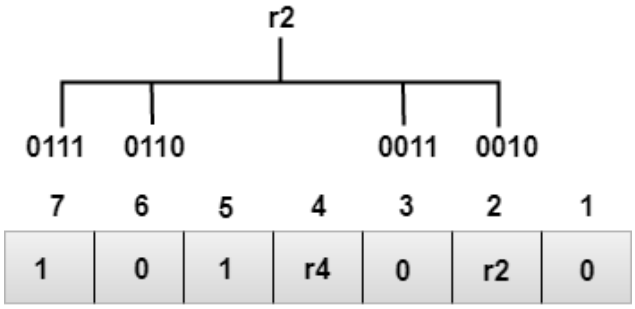
The r1 bit is calculated by performing a parity check on the bit positions whose binary representation includes 1 in the first position.



We observe from the above figure that the bit positions that include 1 in the first position are 1, 3, 5, 7. Now, we perform the even-parity check at these bit positions. The total number of 1 at these bit positions corresponding to r1 is even, therefore, the value of the r1 bit is 0.

Determining r2 bit:

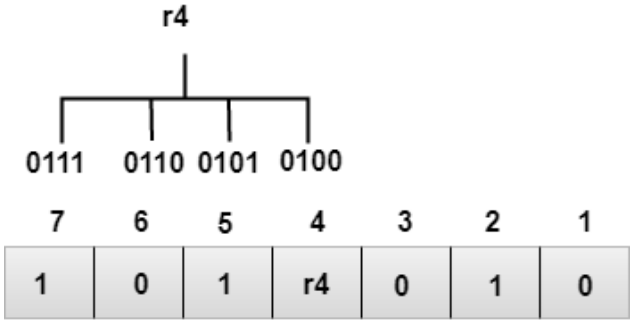
The r2 bit is calculated by performing a parity check on the bit positions whose binary representation includes 1 in the second position.



We observe from the above figure that the bit positions that includes 1 in the second position are 2, 3, 6, 7. Now, we perform the even-parity check at these bit positions. The total number of 1 at these bit positions corresponding to r2 is odd, therefore, the value of the r2 bit is 1.

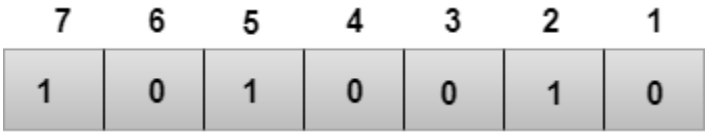
Determining r4 bit:

The r4 bit is calculated by performing a parity check on the bit positions whose binary representation includes 1 in the third position.



We observe from the above figure that the bit positions that includes 1 in the third position are 4, 5, 6, 7. Now, we perform the even-parity check at these bit positions. The total number of 1 at these bit positions corresponding to r4 is even, therefore, the value of the r4 bit is 0.

Data transferred is given below:



Elementary Data Link protocols:

Elementary Data Link protocols are classified into three categories, as given below −

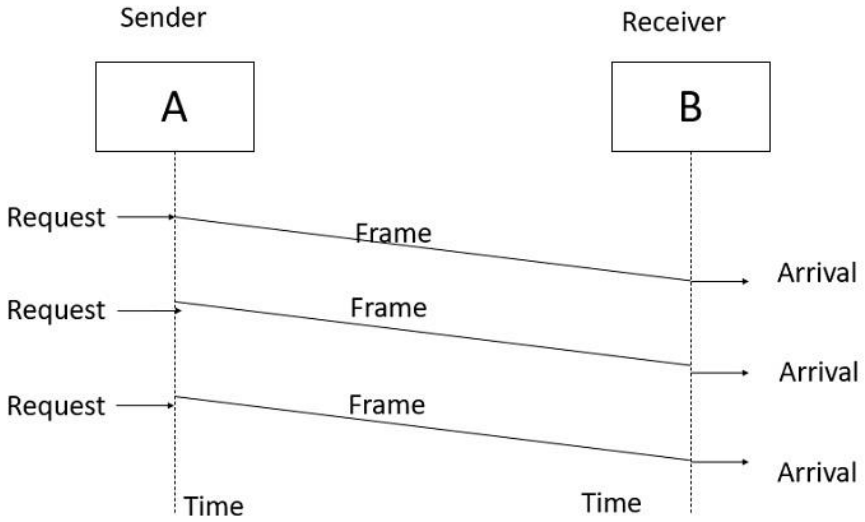
- Protocol 1 − Unrestricted simplex protocol
- Protocol 2 − Simplex stop and wait protocol
- Protocol 3 − Simplex protocol for noisy channels.

Let us discuss each protocol one by one.

Unrestricted Simplex Protocol:

Data transmitting is carried out in one direction only. The transmission (Tx) and receiving (Rx) are always ready and the processing time can be ignored. In this protocol, infinite buffer space is available, and no errors are occurring that is no damage frames and no lost frames.

The Unrestricted Simplex Protocol is diagrammatically represented as follows −



Simplex Stop and Wait protocol:

In this protocol we assume that data is transmitted in one direction only. No error occurs; the receiver can only process the received information at finite rate. These assumptions imply that the transmitter cannot send frames at rate faster than the receiver can process them.

The main problem here is how to prevent the sender from flooding the receiver. The general solution for this problem is to have the receiver send some sort of feedback to sender, the process is as follows:

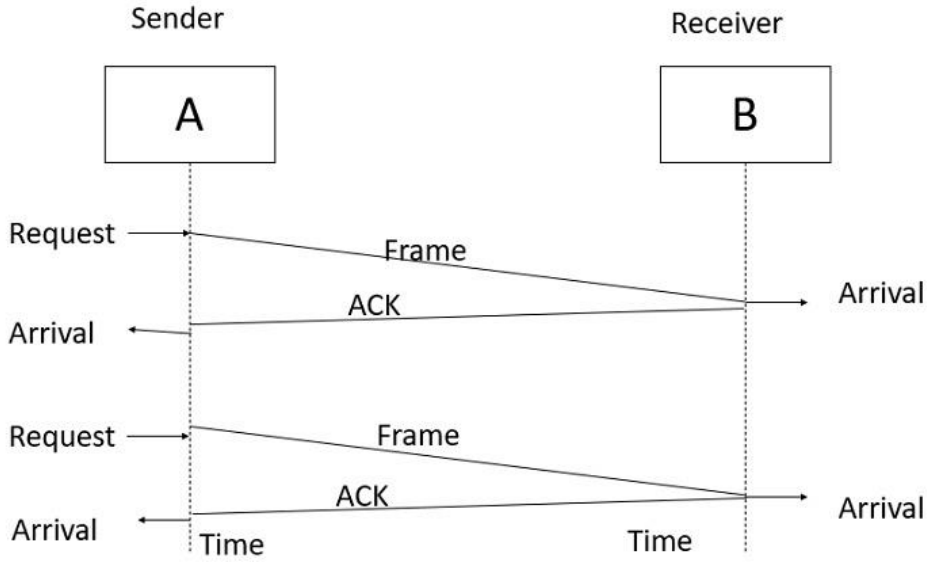
Step1 − The receiver send the acknowledgement frame back to the sender telling the sender that the last received frame has been processed and passed to the host.

Step 2 − Permission to send the next frame is granted.

Step 3 −The sender after sending the sent frame has to wait for an acknowledge frame from the receiver before sending another frame.

This protocol is called Simplex Stop and wait protocol, the sender sends one frame and waits for feedback from the receiver. When the ACK arrives, the sender sends the next frame.

The Simplex Stop and Wait Protocol is diagrammatically represented as follows:

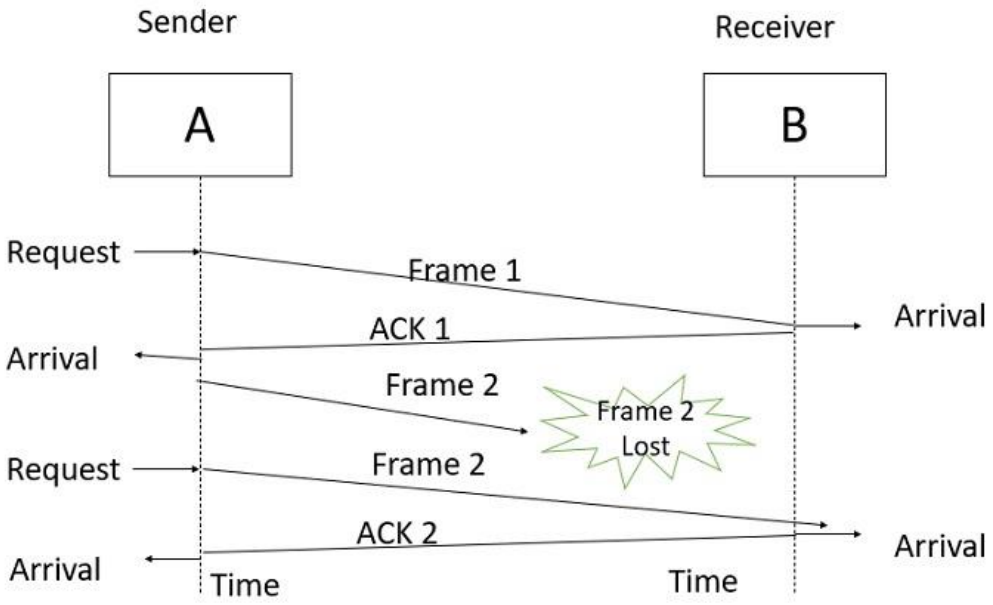


Simplex Protocol for Noisy Channel:

Data transfer is only in one direction, consider separate sender and receiver, finite processing capacity and speed at the receiver, since it is a noisy channel, errors in data frames or acknowledgement frames are expected. Every frame has a unique sequence number.

After a frame has been transmitted, the timer is started for a finite time. Before the timer expires, if the acknowledgement is not received , the frame gets retransmitted, when the acknowledgement gets corrupted or sent data frames gets damaged, how long the sender should wait to transmit the next frame is infinite.

The Simplex Protocol for Noisy Channel is diagrammatically represented as follows −



Sliding Window Protocol:

The sliding window is a technique for sending multiple frames at a time. It controls the data packets between the two devices where reliable and gradual delivery of data frames is needed. It is also used in TCP (Transmission Control Protocol).

In this technique, each frame has sent from the sequence number. The sequence numbers are used to find the missing data in the receiver end. The purpose of the sliding window technique is to avoid duplicate data, so it uses the sequence number.

Types of Sliding Window Protocol:

Sliding window protocol has two types:

1. Go-Back-N ARQ
2. Selective Repeat ARQ

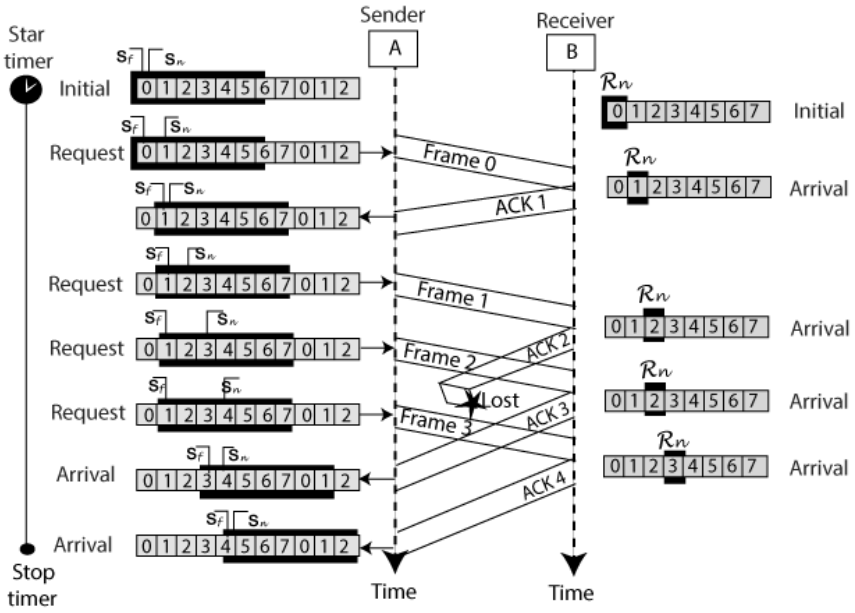
Go-Back-N ARQ:

Go-Back-N ARQ protocol is also known as Go-Back-N Automatic Repeat Request. It is a data link layer protocol that uses a sliding window method. In this, if any frame is corrupted or lost, all subsequent frames have to be sent again.

The size of the sender window is N in this protocol. For example, Go-Back-8, the size of the sender window, will be 8. The receiver window size is always 1.

If the receiver receives a corrupted frame, it cancels it. The receiver does not accept a corrupted frame. When the timer expires, the sender sends the correct frame again. The design of the Go-Back-N ARQ protocol is shown below.

The example of Go-Back-N ARQ is shown below in the figure.

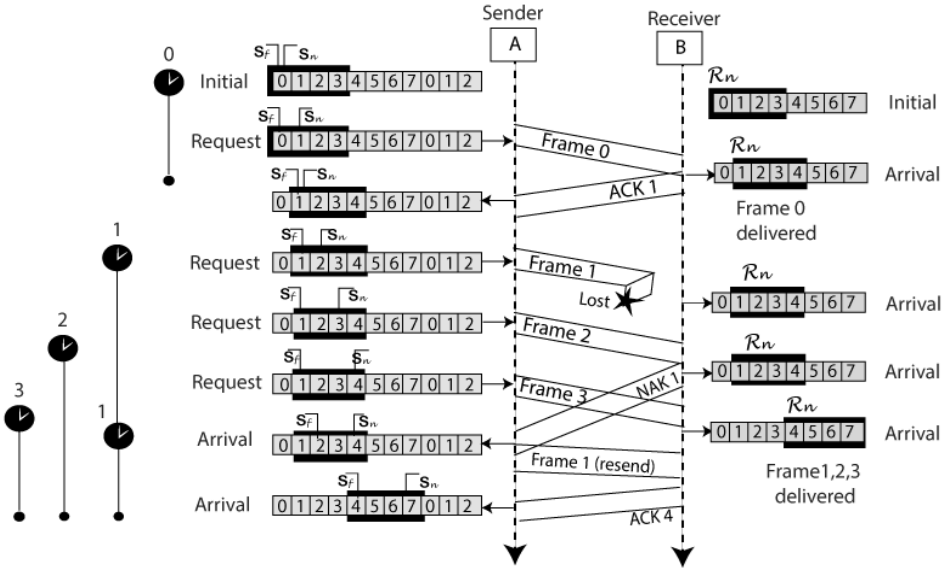


Selective Repeat ARQ:

Selective Repeat ARQ is also known as the Selective Repeat Automatic Repeat Request. It is a data link layer protocol that uses a sliding window method. The Go-back-N ARQ protocol works well if it has fewer errors. But if there is a lot of error in the frame, lots of bandwidth loss in sending the frames again. So, we use the Selective Repeat ARQ protocol. In this protocol, the size of the sender window is always equal to the size of the receiver window. The size of the sliding window is always greater than 1.

If the receiver receives a corrupt frame, it does not directly discard it. It sends a negative acknowledgment to the sender. The sender sends that frame again as soon as on the receiving negative acknowledgment. There is no waiting for any time-out to send that frame. The design of the Selective Repeat ARQ protocol is shown below.

The example of the Selective Repeat ARQ protocol is shown below in the figure.



Difference between the Go-Back-N ARQ and Selective Repeat ARQ:

Go-Back-N ARQ	Selective Repeat ARQ
If a frame is corrupted or lost in it, all subsequent frames have to be sent again.	In this, only the frame is sent again, which is corrupted or lost.
If it has a high error rate, it wastes a lot of bandwidth.	There is a loss of low bandwidth.
It is less complex.	It is more complex because it has to do sorting and searching as well. And it also requires more storage.
It does not require sorting.	In this, sorting is done to get the frames in the correct order.
It does not require searching.	The search operation is performed in it.
It is used more.	It is used less because it is more complex.

Network Layer:

- ✓ The Network Layer is the third layer of the OSI model.
- ✓ It handles the service requests from the transport layer and further forwards the service request to the data link layer.
- ✓ The network layer translates the logical addresses into physical addresses
- ✓ It determines the route from the source to the destination and also manages the traffic problems such as switching, routing and controls the congestion of data packets.
- ✓ The main role of the network layer is to move the packets from sending host to the receiving host.

The main functions performed by the network layer are:

- **Routing:** When a packet reaches the router's input link, the router will move the packets to the router's output link. For example, a packet from S1 to R1 must be forwarded to the next router on the path to S2.
- **Logical Addressing:** The data link layer implements the physical addressing and network layer implements the logical addressing. Logical addressing is also used to distinguish between source and destination system. The network layer adds a header to the packet which includes the logical addresses of both the sender and the receiver.
- **Internetworking:** This is the main role of the network layer that it provides the logical connection between different types of networks.
- **Fragmentation:** The fragmentation is a process of breaking the packets into the smallest individual data units that travel through different networks.

Services Provided by the Network Layer:

- **Guaranteed delivery:** This layer provides the service which guarantees that the packet will arrive at its destination.
- **Guaranteed delivery with bounded delay:** This service guarantees that the packet will be delivered within a specified host-to-host delay bound.
- **In-Order packets:** This service ensures that the packet arrives at the destination in the order in which they are sent.
- **Guaranteed max jitter:** This service ensures that the amount of time taken between two successive transmissions at the sender is equal to the time between their receipt at the destination.
- **Security services:** The network layer provides security by using a session key between the source and destination host. The network layer in the source host encrypts the payloads of datagrams being sent to the destination host. The network layer in the destination host would then decrypt the payload. In such a way, the network layer maintains the data integrity and source authentication services.

Network Layer Design:

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

Before learning about design issues in the network layer, let's learn about it's various functions.

1. **Addressing:**

   Maintains the address at the frame header of both source and destination and performs addressing to detect various devices in network.

2. **Packeting:**

   This is performed by Internet Protocol. The network layer converts the packets from its upper layer.

3. **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.

4. **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:

Through the network/transport layer interface, the network layer transfers it's services to the transport layer. These services are described below.

But before providing these services to the transfer layer following goals must be kept in mind :-

- 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 "datagrams" 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.

Routing algorithm:

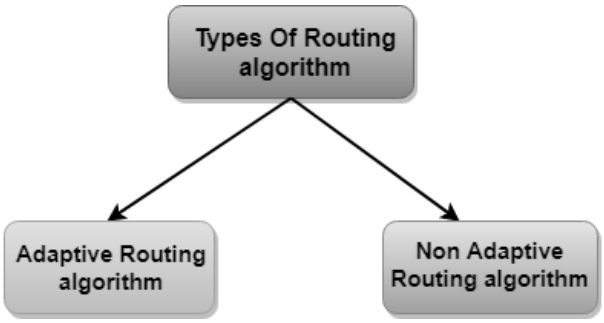
- ✓ In order to transfer the packets from source to the destination, the network layer must determine the best route through which packets can be transmitted.
- ✓ Whether the network layer provides datagram service or virtual circuit service, the main job of the network layer is to provide the best route. 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.
- The main parameters related to this algorithm are hop count, distance and estimated transit time.

An adaptive routing algorithm can be classified into three parts:

- **Centralized algorithm:** It is also known as global routing algorithm as it computes the least-cost path between source and destination by using complete and global knowledge about the network. This algorithm takes the connectivity between the nodes and link cost as input, and this information is obtained before actually performing any calculation. **Link state algorithm** is referred to as a centralized algorithm since it is aware of the cost of each link in the network.
- **Isolation algorithm:** It is an algorithm that obtains the routing information by using local information rather than gathering information from other nodes.
- **Distributed algorithm:** It is also known as decentralized algorithm as it computes the least-cost path between source and destination in an iterative and distributed manner. In the decentralized algorithm, no node has the knowledge about the cost of all the network links. In the beginning, a node contains the information only about its own directly attached links and through an iterative process of calculation computes the least-cost path to the destination. A Distance vector algorithm is a decentralized algorithm as it never knows the complete path from source to the destination; instead it knows the direction through which the packet is to be forwarded along with the least cost path.

Non-Adaptive Routing algorithm:

- Non Adaptive routing algorithm is also known as a static routing algorithm.
- When booting up the network, the routing information stores to the routers.
- Non Adaptive routing algorithms do not take the routing decision based on the network topology or network traffic.

The Non-Adaptive Routing algorithm is of two types:

Flooding: In case of flooding, every incoming packet is sent to all the outgoing links except the one from it has been reached. The disadvantage of flooding is that node may contain several copies of a particular packet.

Random walks: In case of random walks, a packet sent by the node to one of its neighbors randomly. An advantage of using random walks is that it uses the alternative routes very efficiently.

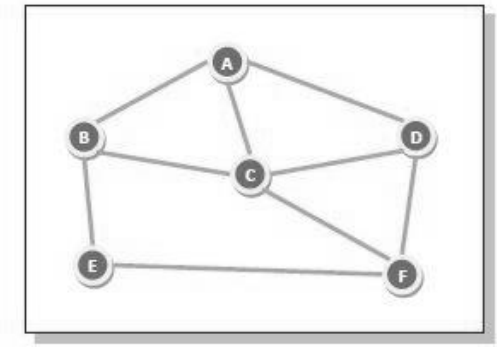
Differences b/w Adaptive and Non-Adaptive Routing Algorithm

Basis Of Comparison	**Adaptive Routing algorithm**	**Non-Adaptive Routing algorithm**
Define	Adaptive Routing algorithm is an algorithm that constructs the routing table based on the network conditions.	The Non-Adaptive Routing algorithm is an algorithm that constructs the static table to determine which node to send the packet.
Usage	Adaptive routing algorithm is used by dynamic routing.	The Non-Adaptive Routing algorithm is used by static routing.
Routing decision	Routing decisions are made based on topology and network traffic.	Routing decisions are the static tables.
Categorization	The types of adaptive routing algorithm, are Centralized, isolation and distributed algorithm.	The types of Non Adaptive routing algorithm are flooding and random walks.
Complexity	Adaptive Routing algorithms are more complex.	Non-Adaptive Routing algorithms are simple.

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.

Types of Flooding:

Flooding may be of three types −

- **Uncontrolled flooding** − Here, each router unconditionally transmits the incoming data packets to all its neighbours.
- **Controlled flooding** − They use some methods to control the transmission of packets to the neighbouring nodes. The two popular algorithms for controlled flooding are Sequence Number Controlled Flooding (SNCF) and Reverse Path Forwarding (RPF).
- **Selective flooding** − Here, the routers don't transmit the incoming packets only along those paths which are heading towards approximately in the right direction, instead of every available paths.

Advantages of Flooding:

- It is very simple to setup and implement, since a router may know only its neighbours.
- It is extremely robust. Even in case of malfunctioning of a large number routers, the packets find a way to reach the destination.
- All nodes which are directly or indirectly connected are visited. So, there are no chances for any node to be left out. This is a main criteria in case of broadcast messages.
- The shortest path is always chosen by flooding.

Limitations of Flooding:

- Flooding tends to create an infinite number of duplicate data packets, unless some measures are adopted to damp packet generation.

- It is wasteful if a single destination needs the packet, since it delivers the data packet to all nodes irrespective of the destination.
- The network may be clogged with unwanted and duplicate data packets. This may hamper delivery of other data packets.

Distance Vector Routing Algorithm:

- **The Distance vector algorithm is iterative, asynchronous and distributed.**
  - **Distributed:** It is distributed in that each node receives information from one or more of its directly attached neighbors, performs calculation and then distributes the result back to its neighbors.
  - **Iterative:** It is iterative in that its process continues until no more information is available to be exchanged between neighbors.
  - **Asynchronous:** It does not require that all of its nodes operate in the lock step with each other.
- The Distance vector algorithm is a dynamic algorithm.
- It is mainly used in ARPANET, and RIP.
- Each router maintains a distance table known as **Vector**.

Three Keys to understand the working of Distance Vector Routing Algorithm:

- **Knowledge about the whole network:** Each router shares its knowledge through the entire network. The Router sends its collected knowledge about the network to its neighbors.
- **Routing only to neighbors:** The router sends its knowledge about the network to only those routers which have direct links. The router sends whatever it has about the network through the ports. The information is received by the router and uses the information to update its own routing table.
- **Information sharing at regular intervals:** Within 30 seconds, the router sends the information to the neighboring routers.

Distance Vector Routing Algorithm:

Let $d_x(y)$ be the cost of the least-cost path from node x to node y. The least costs are related by Bellman-Ford equation,

$$d_x(y) = \min_v\{c(x,v) + d_v(y)\}$$

Where the minv is the equation taken for all x neighbors. After traveling from x to v, if we consider the least-cost path from v to y, the path cost will be $c(x,v)+d_v(y)$. The least cost from x to y is the minimum of $c(x,v)+d_v(y)$ taken over all neighbors.

With the Distance Vector Routing algorithm, the node x contains the following routing information:

- For each neighbor v, the cost c(x,v) is the path cost from x to directly attached neighbor, v.

- The distance vector x, i.e., $D_x = [D_x(y) : y \text{ in } N]$, containing its cost to all destinations, y, in N.
- The distance vector of each of its neighbors, i.e., $D_v = [D_v(y) : y \text{ in } N]$ for each neighbor v of x.

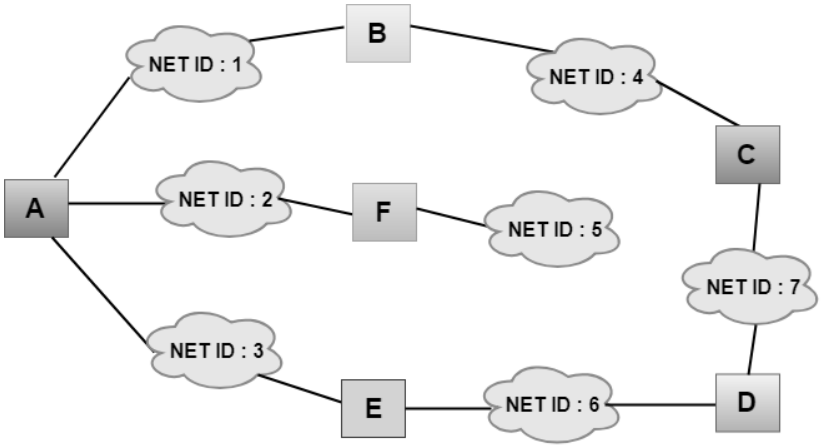
Distance vector routing is an asynchronous algorithm in which node x sends the copy of its distance vector to all its neighbors. When node x receives the new distance vector from one of its neighboring vector, v, it saves the distance vector of v and uses the Bellman-Ford equation to update its own distance vector. The equation is given below:

$$\mathbf{d_x(y) = \min_v\{ c(x,v) + d_v(y)\} \quad \text{for each node y in N}}$$

The node x has updated its own distance vector table by using the above equation and sends its updated table to all its neighbors so that they can update their own distance vectors.

Let's understand through an example:

Sharing Information:



- In the above figure, each cloud represents the network, and the number inside the cloud represents the network ID.
- All the LANs are connected by routers, and they are represented in boxes labeled as A, B, C, D, E, F.
- Distance vector routing algorithm simplifies the routing process by assuming the cost of every link is one unit. Therefore, the efficiency of transmission can be measured by the number of links to reach the destination.
- In Distance vector routing, the cost is based on hop count.

Routing Table:

Two processes occur:

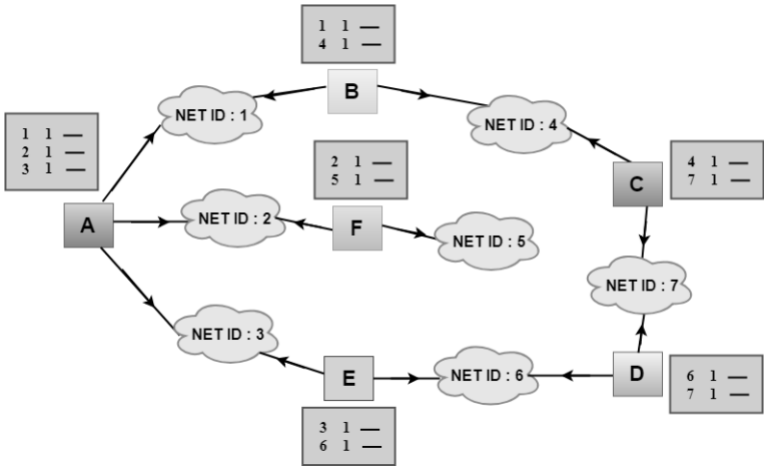
- Creating the Table
- Updating the Table

Creating the Table:

Initially, the routing table is created for each router that contains atleast three types of information such as Network ID, the cost and the next hop.

NET ID	Cost	Next Hop

- **NET ID:** The Network ID defines the final destination of the packet.
- **Cost:** The cost is the number of hops that packet must take to get there.
- **Next hop:** It is the router to which the packet must be delivered.



- In the above figure, the original routing tables are shown of all the routers. In a routing table, the first column represents the network ID, the second column represents the cost of the link, and the third column is empty.
- These routing tables are sent to all the neighbors.

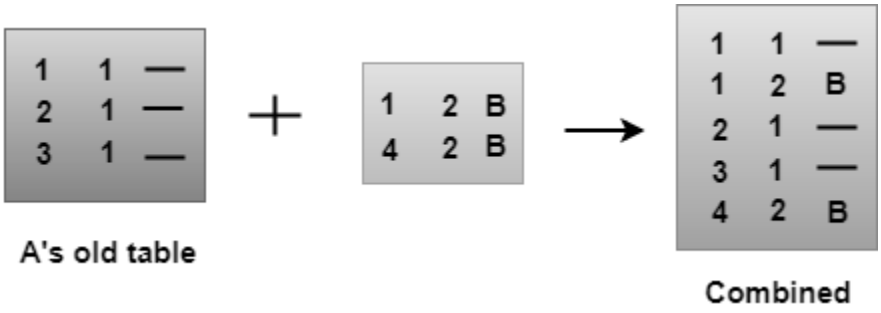
For Example:

1. A sends its routing table to B, F & E.
2. B sends its routing table to A & C.
3. C sends its routing table to B & D.
4. D sends its routing table to E & C.
5. E sends its routing table to A & D.
6. F sends its routing table to A.

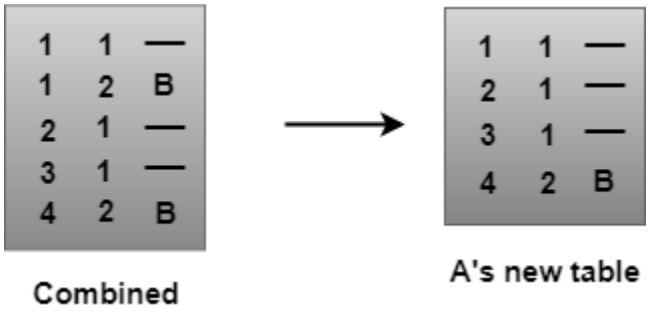
Updating the Table:

- When A receives a routing table from B, then it uses its information to update the table.
- The routing table of B shows how the packets can move to the networks 1 and 4.
- The B is a neighbor to the A router, the packets from A to B can reach in one hop. So, 1 is added to all the costs given in the B's table and the sum will be the cost to reach a particular network.

- After adjustment, A then combines this table with its own table to create a combined table.



- The combined table may contain some duplicate data. In the above figure, the combined table of router A contains the duplicate data, so it keeps only those data which has the lowest cost. For example, A can send the data to network 1 in two ways. The first, which uses no next router, so it costs one hop. The second requires two hops (A to B, then B to Network 1). The first option has the lowest cost, therefore it is kept and the second one is dropped.



- The process of creating the routing table continues for all routers. Every router receives the information from the neighbors, and update the routing table.

Hierarchical routing:

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.

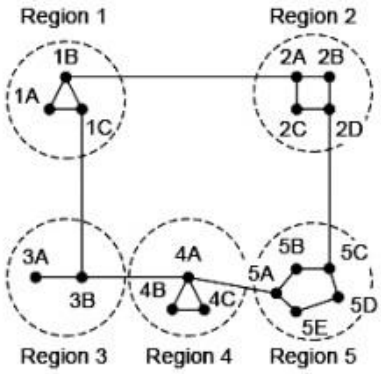
As we know, in both LS and DV algorithms, every router needs to save some information about other routers. When network size is growing, the number of routers in the network will increase. Therefore, the size of routing table increases, then routers cannot handle network traffic as efficiently. To overcome 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.

Explanation:

Step 1 − For example, the best path from 1A to 5C is via region 2, but hierarchical routing of all traffic to region 5 goes via region 3 as it is better for most of the other destinations of region 5.

Step 2 − Consider a subnet of 720 routers. If no hierarchy is used, each router will have 720 entries in its routing table.

Step 3 − Now if the subnet is partitioned into 24 regions of 30 routers each, then each router will require 30 local entries and 23 remote entries for a total of 53 entries.

Example:

If the same subnet of 720 routers is partitioned into 8 clusters, each containing 9 regions and each region containing 10 routers. Then what will be the total number of table entries in each router.

Solution:

10 local entries + 8 remote regions + 7 clusters = 25 entries.

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.

Link State Routing has two phases:

Reliable Flooding:

- **Initial state:** Each node knows the cost of its neighbors.
- **Final state:** Each node knows the entire graph.

Route Calculation:

Each node uses Dijkstra's algorithm on the graph to calculate the optimal routes to all nodes.

- The Link state routing algorithm is also known as Dijkstra's algorithm which is used to find the shortest path from one node to every other node in the network.
- The Dijkstra's algorithm is an iterative, and it has the property that after k^{th} iteration of the algorithm, the least cost paths are well known for k destination nodes.

Let's describe some notations:

- **c(i , j):** Link cost from node i to node j. If i and j nodes are not directly linked, then c(i , j) = ∞.
- **D(v):** It defines the cost of the path from source code to destination v that has the least cost currently.
- **P(v):** It defines the previous node (neighbor of v) along with current least cost path from source to v.
- **N:** It is the total number of nodes available in the network.

Algorithm:

Initialization

N = {A} // **A is a root node**.

for all nodes v

if v adjacent to A

then D(v) = c(A,v)

else D(v) = infinity

loop

find w not in N such that D(w) is a minimum.

Add w to N

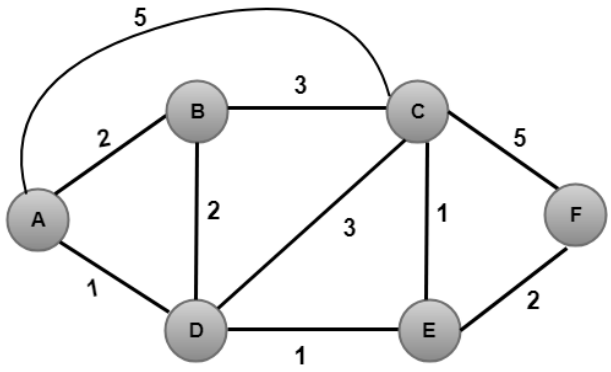
Update D(v) for all v adjacent to w and not in N:

D(v) = min(D(v) , D(w) + c(w,v))

Until all nodes in N

In the above algorithm, an initialization step is followed by the loop. The number of times the loop is executed is equal to the total number of nodes available in the network.

Let's understand through an example:



In the above figure, source vertex is A.

Congestion Control Algorithm:

What is **congestion**?

A state occurring in network layer when the message traffic is so heavy that it slows down network response time.

Effects of Congestion:

- As delay increases, performance decreases.
- If delay increases, retransmission occurs, making situation worse.

Congestion control algorithms:

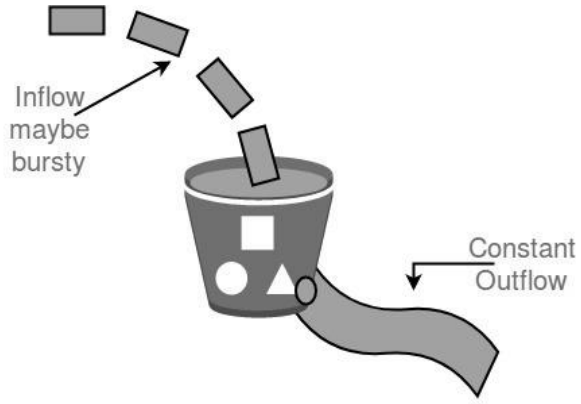
1. Congestion Control is a mechanism that controls the entry of data packets into the network, enabling a better use of a shared network infrastructure and avoiding congestive collapse.
2. Congestive-Avoidance Algorithms (CAA) are implemented at the TCP layer as the mechanism to avoid congestive collapse in a network.
3. There are two congestion control algorithm which are as follows:

Leaky Bucket Algorithm:

- The leaky bucket algorithm discovers its use in the context of network traffic shaping or rate-limiting.
- A leaky bucket execution and a token bucket execution are predominantly used for traffic shaping algorithms.
- This algorithm is used to control the rate at which traffic is sent to the network and shape the burst traffic to a steady traffic stream.
- The disadvantages compared with the leaky-bucket algorithm are the inefficient use of available network resources.
- The large area of network resources such as bandwidth is not being used effectively.

Let us consider an example to understand:

Imagine a bucket with a small hole in the bottom.No matter at what rate water enters the bucket, the outflow is at constant rate.When the bucket is full with water additional water entering spills over the sides and is lost.



Similarly, each network interface contains a leaky bucket and the following **steps** are involved in leaky bucket algorithm:

1. When host wants to send packet, packet is thrown into the bucket.
2. The bucket leaks at a constant rate, meaning the network interface transmits packets at a constant rate.
3. Bursty traffic is converted to a uniform traffic by the leaky bucket.
4. In practice the bucket is a finite queue that outputs at a finite rate.

Token bucket Algorithm:

- ✓ The leaky bucket algorithm has a rigid output design at an average rate independent of the bursty traffic.
- ✓ In some applications, when large bursts arrive, the output is allowed to speed up. This calls for a more flexible algorithm, preferably one that never loses information. Therefore, a token bucket algorithm finds its uses in network traffic shaping or rate-limiting.
- ✓ It is a control algorithm that indicates when traffic should be sent. This order comes based on the display of tokens in the bucket.
- ✓ The bucket contains tokens. Each of the tokens defines a packet of predetermined size. Tokens in the bucket are deleted for the ability to share a packet.
- ✓ When tokens are shown, a flow to transmit traffic appears in the display of tokens.
- ✓ No token means no flow sends its packets. Hence, a flow transfers traffic up to its peak burst rate in good tokens in the bucket.

Need of token bucket Algorithm:-

The leaky bucket algorithm enforces output pattern at the average rate, no matter how bursty the traffic is. So in order to deal with the bursty traffic we need a flexible algorithm so that the data is not lost. One such algorithm is token bucket algorithm.

Steps of this algorithm can be described as follows:

1. In regular intervals tokens are thrown into the bucket. *f*
2. The bucket has a maximum capacity. *f*
3. If there is a ready packet, a token is removed from the bucket, and the packet is sent.
4. If there is no token in the bucket, the packet cannot be sent.

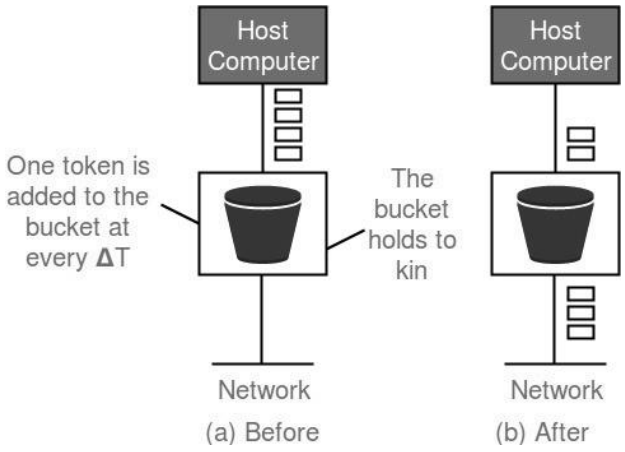
Let's understand with an example,

In figure (A) we see a bucket holding three tokens, with five packets waiting to be transmitted. For a packet to be transmitted, it must capture and destroy one token. In figure (B) We see that three of the five packets have gotten through, but the other two are stuck waiting for more tokens to be generated.

Ways in which token bucket is superior to leaky bucket: The leaky bucket algorithm controls the rate at which the packets are introduced in the network, but it is very conservative in nature. Some flexibility is introduced in the token bucket algorithm. In the token bucket, algorithm tokens are generated at each tick (up to a certain limit). For an incoming packet to be transmitted, it must capture a token and the transmission takes place at the same rate. Hence some of the busty packets are transmitted at the same rate if tokens are available and thus introduces some amount of flexibility in the system.

Formula: M * s = C + ? * s where S – is time taken M – Maximum output rate ? – Token arrival rate C – Capacity of the token bucket in byte

Let's understand with an example,



Admission Control:

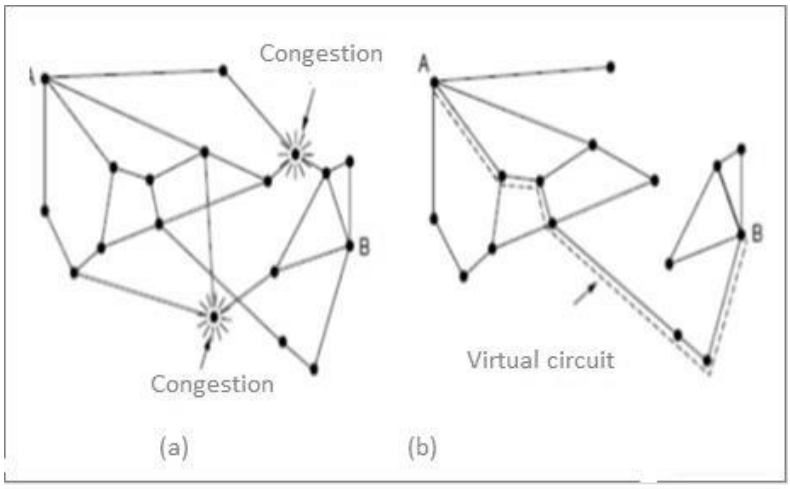
The presence of congestion means the load is greater than the resources available over a network to handle. Generally, we will get an idea to reduce the congestion by trying to increase the resources or decrease the load, but it is not that much of a good idea.

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.

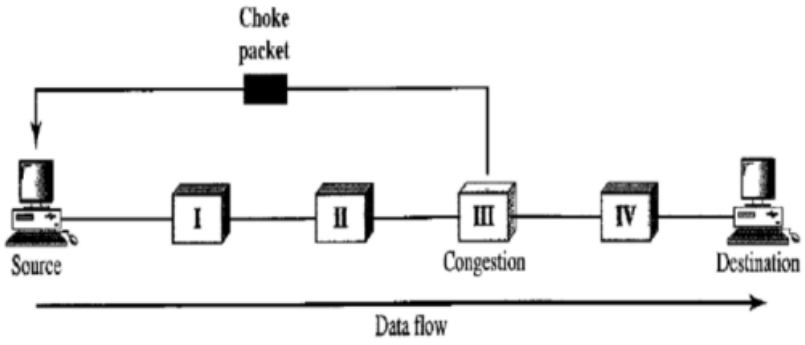
Step 2 − To avoid this situation, we can redraw the network as shown in figure (b), removing the congested routers and all of their lines.

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

Choke Packets:

This approach can be used in virtual circuits as well as in the datagram subnets. In this technique, each router associates a real variable with each of its output lines.

This real variable says “u” has a value between 0and1 and it indicates the percentage utilization of that line. If the value of “u” goes above the threshold, then that output line will enter into a “warning” state.



The router will check each newly arriving packet to see if its output line is in the “warning state”. If it is in the warning state, then the router will send back a choke packet signal to the sending host.

The sender host will not generate any more choke packets. Several variations on the congestion control algorithms have been proposed, depending on the value of thresholds.

Hop-by-hop choke packets:

This technique is an advancement over the Choked packet method. At high speed over long distances, sending a packet back to the source doesn’t help much, because by the time the choke packet reaches the source, already a lot of packets destined for the same original the destination would be out from the source.

So, to help this, Hop-by-Hop Choke packets are used. Over long distances or at high speeds choke p y packets are not very effective. A more efficient the method is to send choke packets hop-by-hop.

This requires each hop to reduce its transmission even before the choke packet arrives at the source.

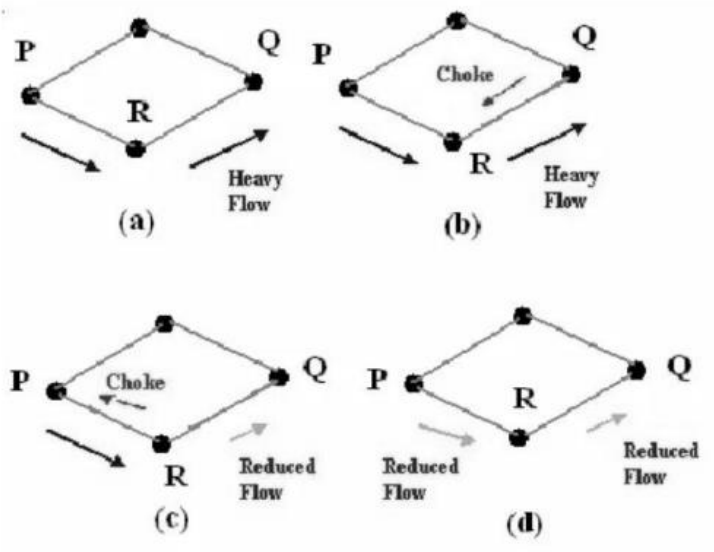


Figure Depicts the functioning of Hop-by-Hop choke packets, (a) Heavy traffic between nodes P and Q, (b) Node Q sends the Choke packet to P, (c) Choke packet reaches R and the flow between R and Q is curtailed down, the Choke packer reaches P and Produces the flow out.