UNIT-II MEDIA ACCESS AND INTERNETWORKING

Media Access Protocols – ALOHA - CSMA/CA/CD –Ethernet – Wireless LANs - 802.11- Bluetooth - Switching and Forwarding - Bridges and LAN Switches – Basic Internetworking- IP Service Model – IP fragmentation - Global Addresses – ARP - DHCP – ICMP- Virtual Networks and Tunnels.

Media access control

The data link layer is divided into two sub layers,

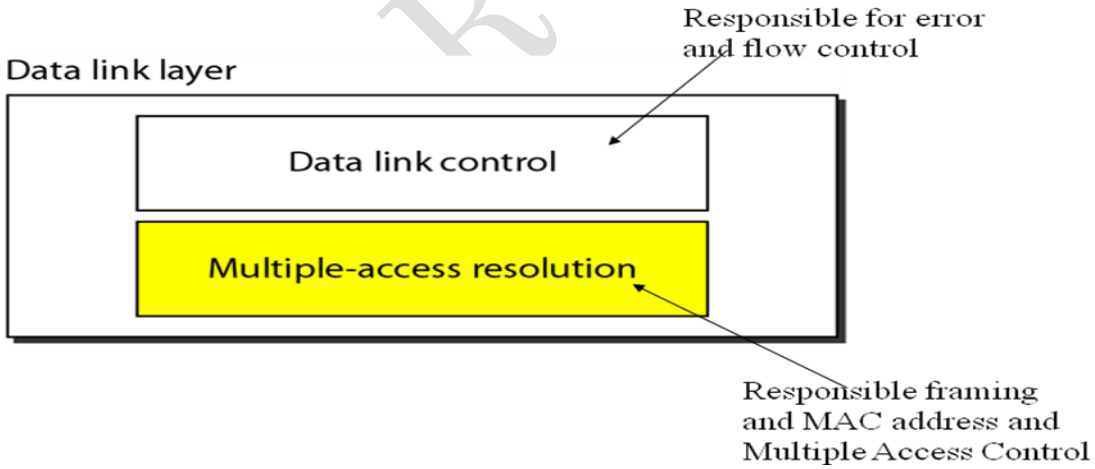
1. Logical link Control Layer
2. Medium Access Control Layer

Logical Link Control Layer:

The LCL layer is concerned with managing traffic (flow and error control) over the physical medium and may also assign sequence numbers to frames and track acknowledgements.

Medium Access Control Layer:

Media Access Control layer is one of two sublayers of the Data Link Control layer and is concerned with sharing the physical connection to the network among several computers.



Medium Access Control:

- **Problem**: When two or more nodes transmit at the same time, their frames will collide and the link bandwidth is **wasted** during collision

  How to coordinate the access of multiple sending/receiving nodes to the shared link**???**

- **Solution**: We need a **protocol** to coordinate the transmission of the active nodes
- These protocols are called **Medium or Multiple Access Control (MAC) Protocols** belong to a **sublayer** of the data link layer called **MAC** (Medium Access Control)

- What is expected from Medium Access Control Protocols:

    1. Main task is to **minimize collisions** in order to **utilize the bandwidth** by:
    2. Determining **when** a station can use the link (medium)
    3. **what** a station should do when the link is **busy**
    4. **what** the station should do when it is involved in **collision**
    5. To avoid the multiple access problem is used in which the station is forced to sense the medium before transmitting. This is called as Carrier Sense Multiple Access (CSMA).

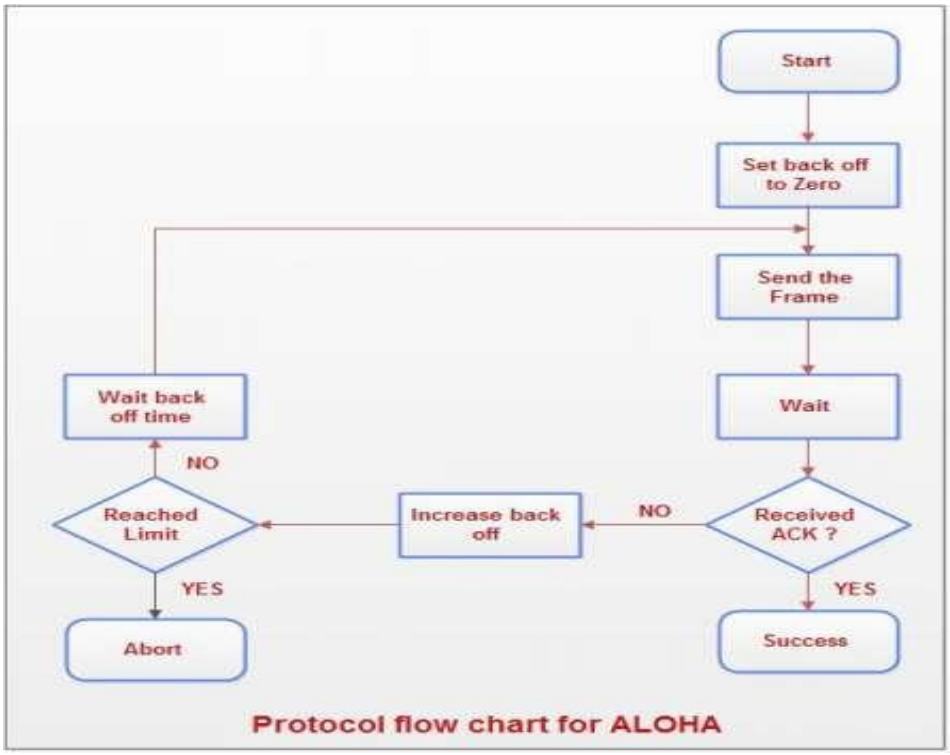
The two commonly used medium access protocols are

1. ALOHA
2. CSMA
3. CSMA/Collision Detection (CSMA/CD)
4. CSMA/Collision Avoidance(CSMA/CA)

ALOHA

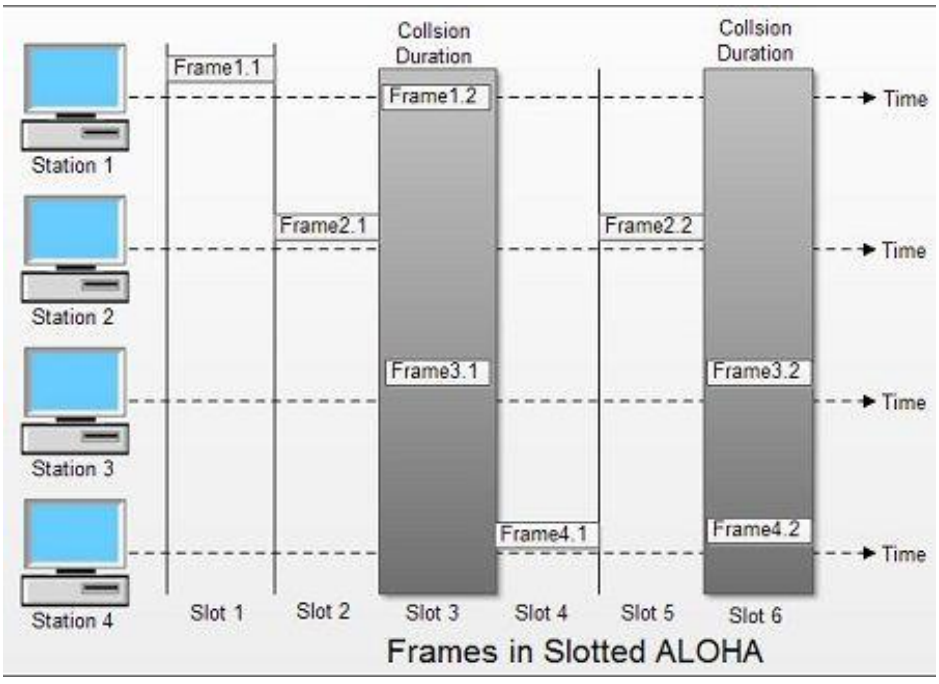
PURE ALOHA

- Each source (transmitter) in a network sends data whenever there is a frame to send.
- If the frame successfully reaches the destination (receiver), the next frame is sent.
- When two or more stations transmit simultaneously, there is collision and the frames are destroyed.
- If acknowledgement is not received within specified time, the station assumes that the frame (or acknowledgement) has been destroyed.
- If the frame is destroyed because of collision the station waits for a random amount of time and sends it again. This waiting time must be random otherwise same frames will collide again and again.
- Therefore, pure ALOHA dictates that when time-out period passes, each station must wait for a random amount of time before resending its frame. This randomness will help avoid more collisions.



SLOTTED ALOHA

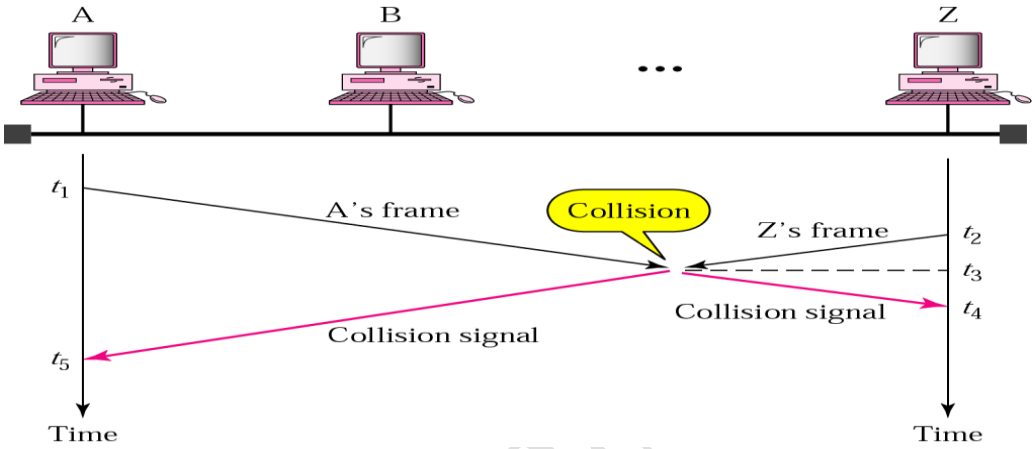
- The time of the shared channel is divided into discrete intervals called slots
- The stations can send a frame only at the beginning of the slot and only one frame is sent in each slot.
- In slotted ALOHA, if any station is not able to place the frame onto the channel at the beginning of the slot *i.e.* it misses the time slot then the station has to wait until the beginning of the next time slot.
- In slotted ALOHA, there is still a possibility of collision if two stations try to send at the beginning of the same time slot



CSMA (Carrier Sense Multiple Access)

CSMA is based on the principle "Sense Before Transmit or Listen Before Talk. Each station must listen before transmitting.

- ➔ If a frame was sent by a station, All stations knows immediately so they can **wait before start sending**
  - ➔ A station with frames to be sent, should **sense the medium** for the presence of another transmission (carrier) before it starts its own transmission
- CSMA can **reduce** the possibility of collision but it cannot eliminate it.
  - Collision can only happen when more than one station begin transmitting within a short time (the **propagation time** period)

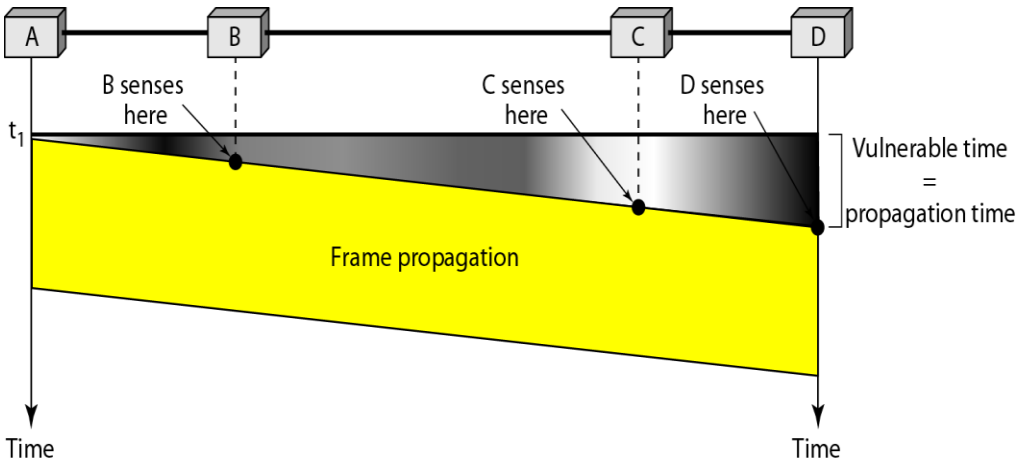


The possibility of collision still exists because of propagation delay; when a station sends a frame, it still takes time (although very short) for the first bit to reach every station and for every station to sense it. In other words, a station may sense the medium and find it idle, only because the first bit sent by another station has not yet been received.

Vulnerable time

Vulnerable time for CSMA is the **maximum propagation time**

- The longer the propagation delay, the worse the performance of the protocol because of the above case.



- When a station sends a frame, and any other station tries to send a frame during this time, a collision will result.
- But if the first bit of the frame reaches the end of the medium, every station will already have heard the bit and will refrain from sending. Figure shows the worst case. The vulnerable time for CSMA is the propagation time tp .
- The leftmost station A sends a frame at time *t1* which reaches the rightmost station D at time *t1+ tp* . The gray area shows the vulnerable area in time and space.

Types of CSMA Protocols

Different CSMA protocols that determine:

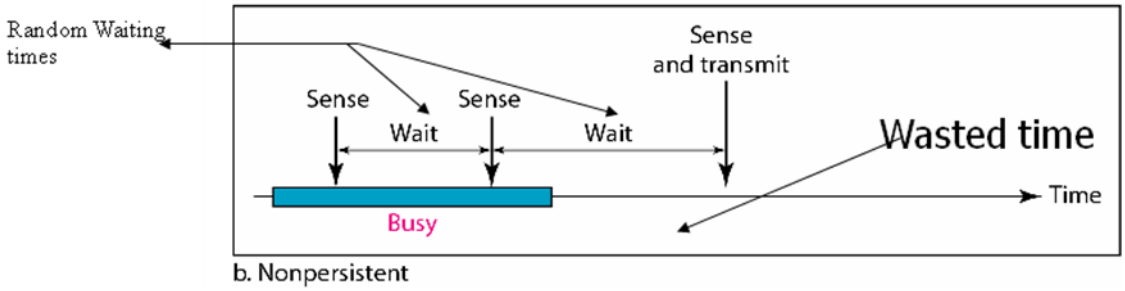
- What a station should do when the medium is **idle**?
- What a station should do when the medium is **busy**?

The channel can access the medium using any of three persistence methods,

1. Non-Persistent CSMA
2. 1-Persistent CSMA
3. p-Persistent CSMA

No persistent CSMA

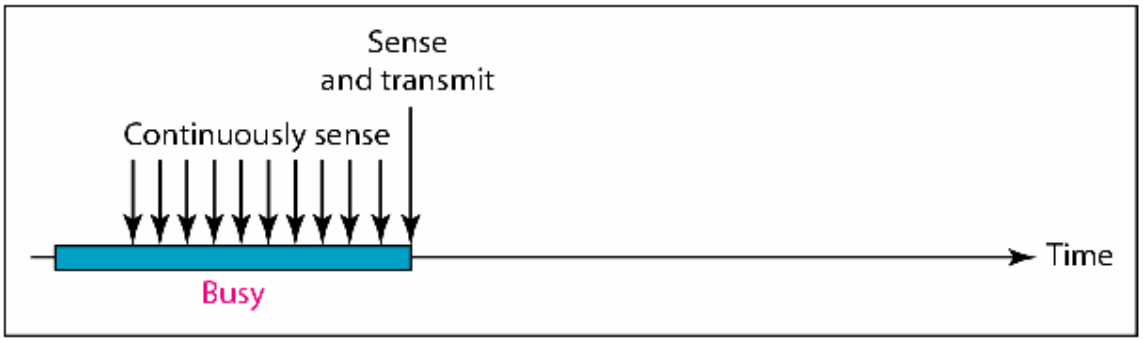
- A station with frames to be sent, should sense the medium
  1. If medium is idle, **transmit**; otherwise, go to 2
  2. If medium is busy, (**backoff**) wait a ***random*** **amount of time** and repeat **1**
- Non-persistent Stations are **deferential (respect others)**
- Performance:
  1. Random delays reduces probability of collisions because two stations with data to be transmitted will wait for different amount of times.
  2. Bandwidth is **wasted** if waiting time (backoff) is large because medium will remain idle following end of transmission even if one or more stations have frames to send



b. Nonpersistent

1-persistent CSMA

- To avoid idle channel time, 1-persistent protocol used
- Station wishing to transmit listens to the medium:
  1. If medium idle, **transmit** immediately;
  2. If medium busy, **continuously listen** until medium becomes idle; then transmit immediately with probability 1
- Performance
  - 1-persistent stations are **selfish**
  - If two or more stations becomes ready at the same time, **collision guaranteed**

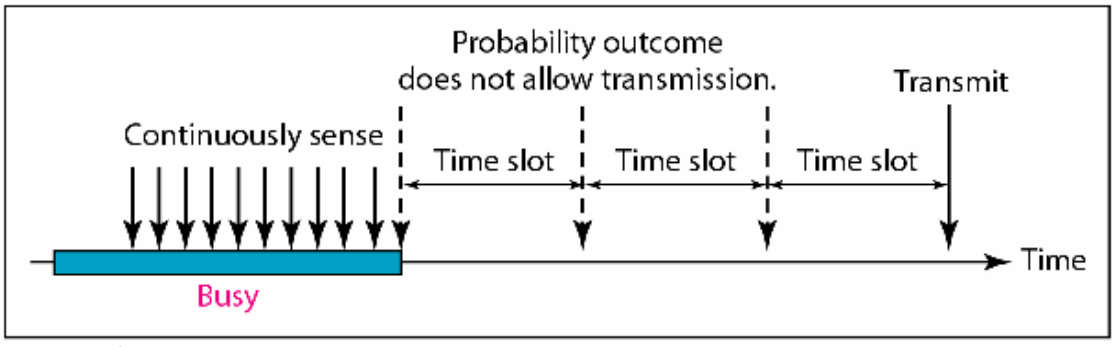


a. 1-persistent

P-persistent CSMA

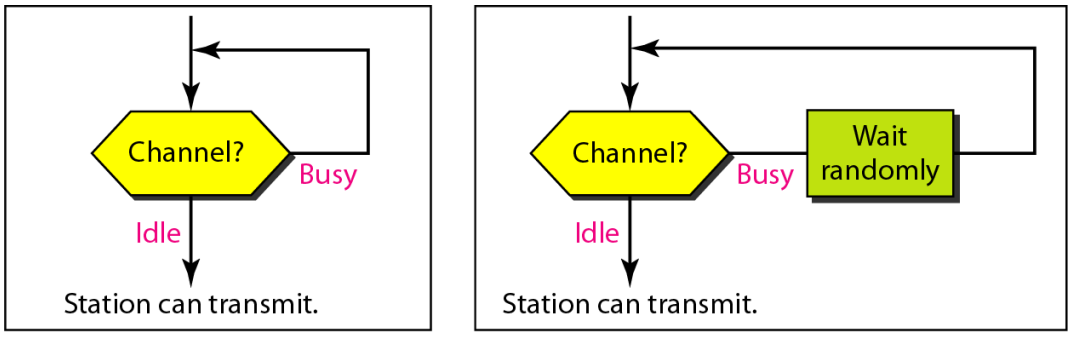
- Time is divided to slots where each Time unit (slot) typically equals **maximum propagation delay**
- Station wishing to transmit listens to the medium:

1. If medium idle,
   - transmit with probability (**p**), OR
   - wait **one time unit (slot)** with probability (**q= 1 – p**), then repeat 1.
2. If medium busy, **continuously listen until idle** and repeat step **1**
3. Performance
   - Reduces the possibility of collisions like **nonpersistent**
   - Reduces channel idle time like **1-persistent**



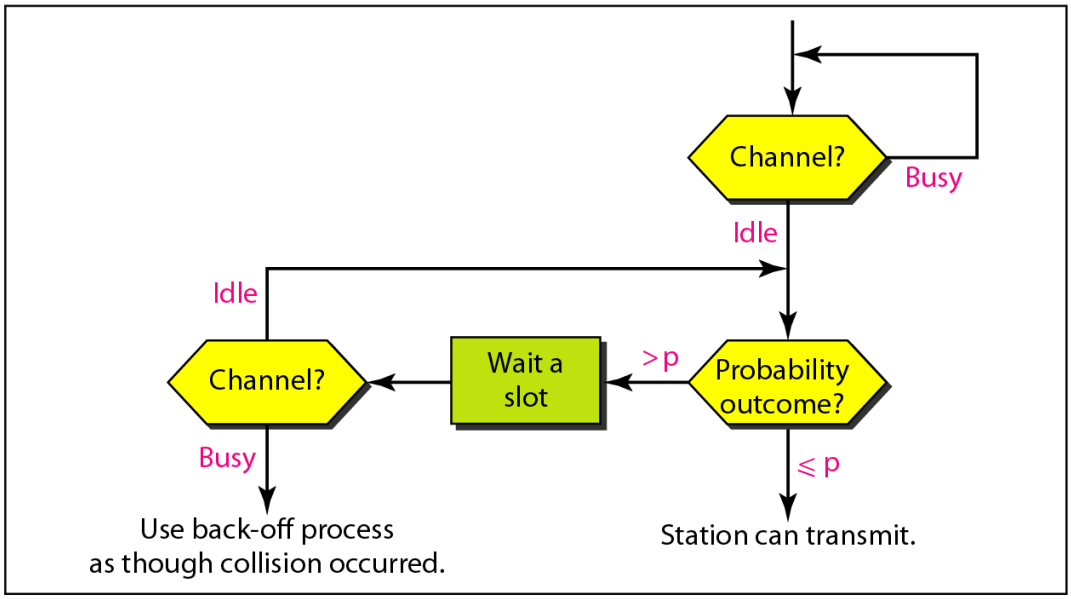
c. p-persistent

Flow diagram for three persistence methods



a. 1-persistent

b. Nonpersistent



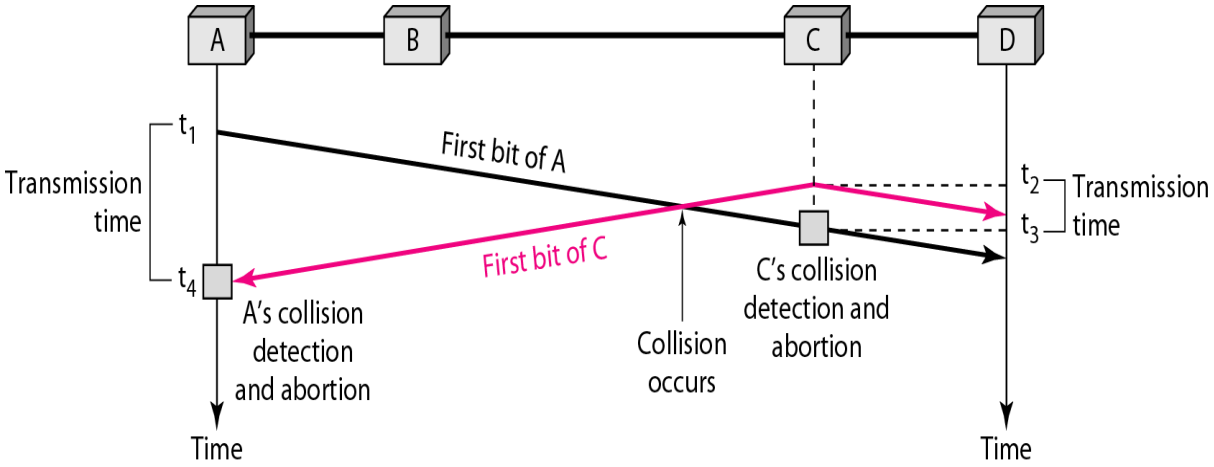
c. p-persistent

CSMA/CD (CSMA - Collision Detection)

- ***CSMA (all previous methods) has an inefficiency:***
  - If a collision has occurred, the channel is **unstable** until colliding packets have **been fully transmitted**
- ***CSMA/CD*** *(Carrier Sense Multiple Access with Collision Detection)* ***overcomes this as follows:***
  - While transmitting, the sender is **listening to medium** for collisions.
  - Sender **stops transmission** if collision has occurred **reducing channel wastage** .

CSMA/CD is Widely used for **bus topology LANs** (IEEE 802.3, **Ethernet).**

Collision of the first bit in CSMA/CD

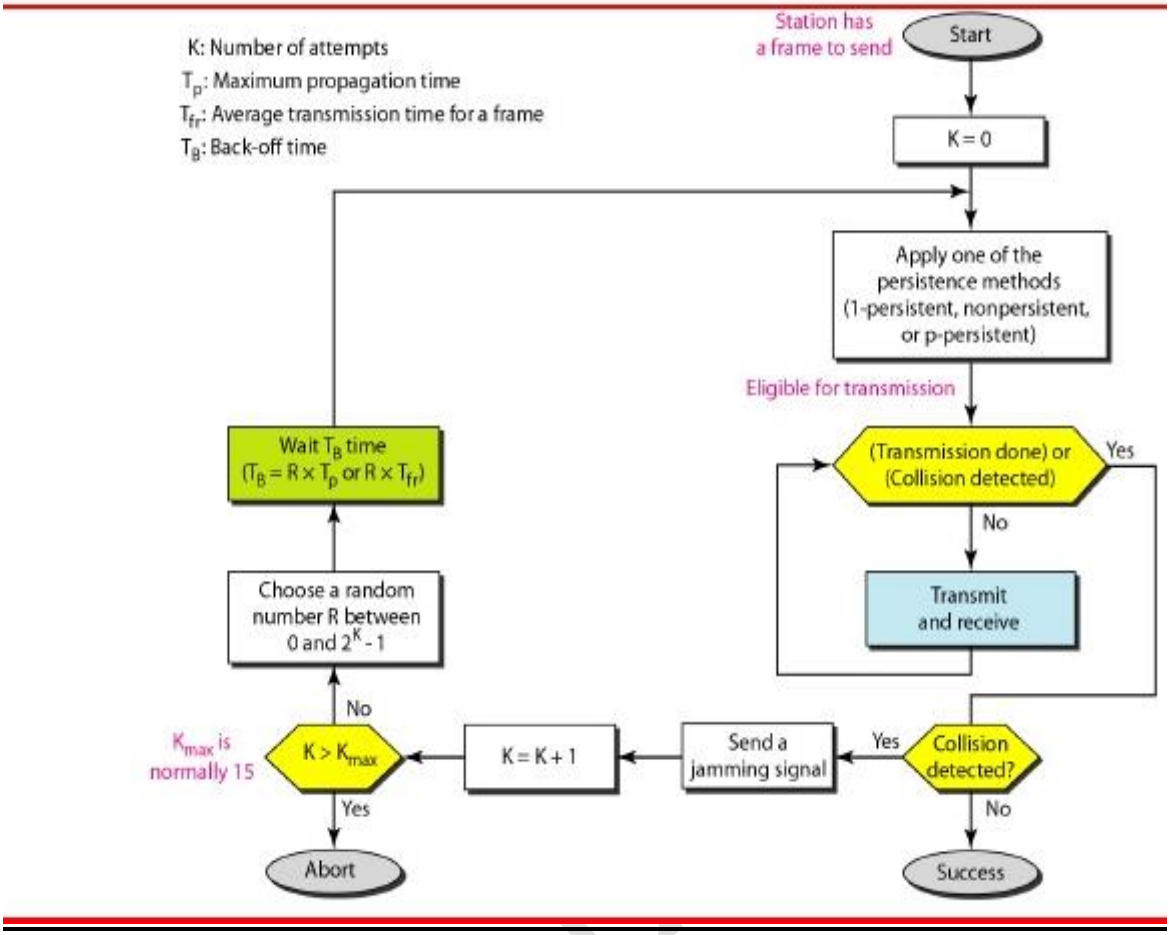


To better understand CSMA/CD, let us look at the first bits transmitted by the two stations involved in the collision. Although each station continues to send bits in the frame until it detects the collision, we show what happens as the first bits collide. In Figure , stations A and C are involved in the collision.

At time *t* 1, station A has executed its persistence procedure and starts sending the bits of its frame. At time *t2,* station C has not yet sensed the first bit sent by A. Station C executes its persistence procedure and starts sending the bits in its frame, which propagate both to the left and to the right. The collision occurs sometime after time *t2'* Station C detects a collision at time *t3* when it receives the first bit ofA's frame. Station C immediately (or after a short time, but we assume immediately) aborts transmission. Station A detects collision at time *t4* when it receives the first bit ofC's frame; it also immediately aborts transmission.

Looking at the figure, we see that A transmits for the duration *t4 - tl;* C transmits for the duration *t3 - t2'* Later we show that, for the protocol to work, the length of any frame divided by the bit rate in this protocol must be more than either of these durations. At time *t4,* the transmission of*A:s* frame, though incomplete, is aborted; at time *t 3* , the transmission ofB's frame, though incomplete, is aborted.

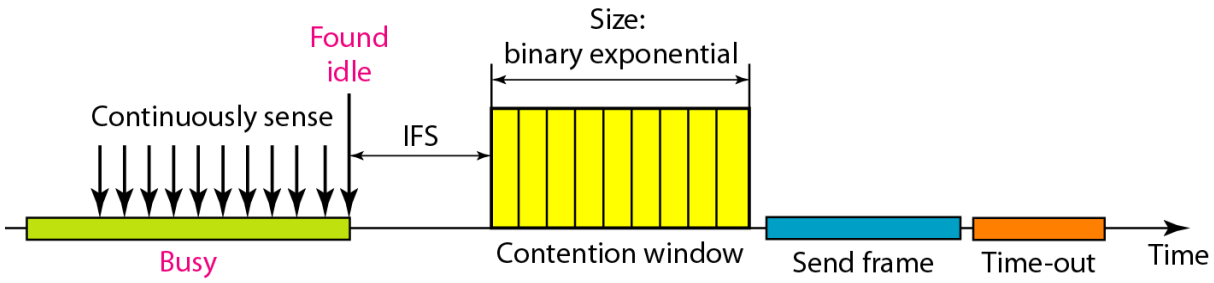
Figure 12.14 *Flow diagram for the CSMA/CD*



CSMA/CA (CSMA – Collision Avoidance)

- **C**arrier **S**ense **M**ultiple **A**ccess with **C**ollision **A**voidance
- Used in a network where collision cannot be detected
  - E.g., wireless LAN

Collisions are avoided in CSMA / CD using three strategies,
1. Inter Frame Space [IFS]
2. Contention window
3. Acknowledgements



Inter Frame Sequence [IFS]

- Collisions are avoided by transmitting if a channel is found idle
- If the channel is found idle, the station does not send immediately
- It waits for a period of time called interframe space (IFS)

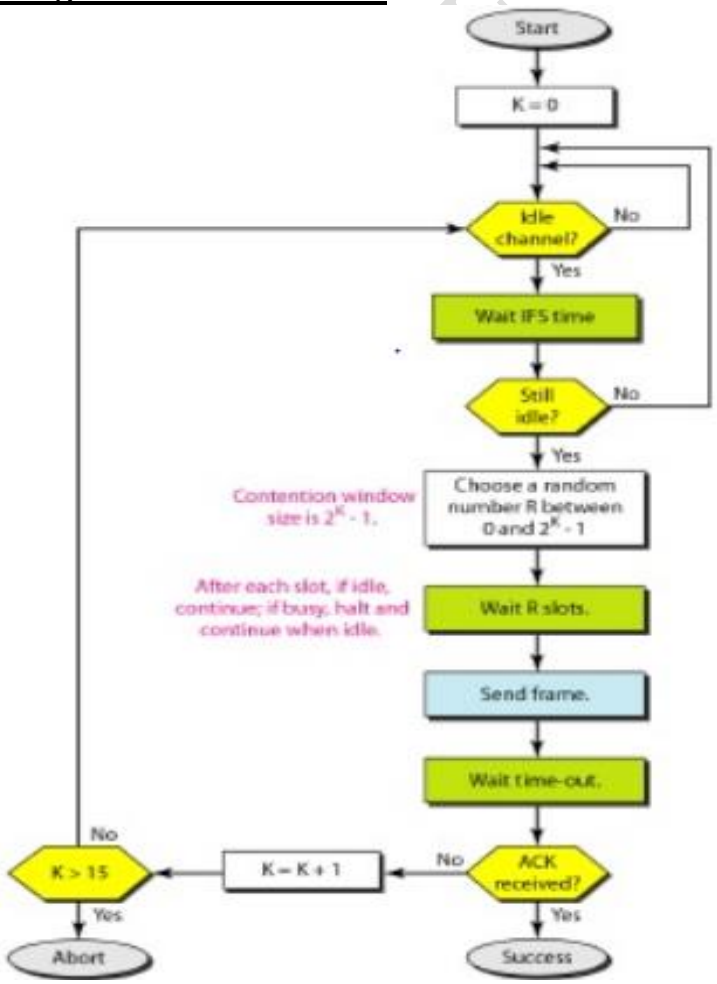
Contention window

- contention window is an amount of time divided into slots.
- The station that is ready to send chooses a random number of slots as its wait time.
- The number of slots in the window changes according to binary exponential back off ie) it is set to one slot the first time and then doubles each time the station cannot detect an idle channel.
- In CSMA/CA, if the station finds the channel busy, it does not restart the timer of the contention window; it stops the timer and restarts it when the channel becomes idle

Acknowledgements

- Collisions can be avoided by setting positive acknowledgements to assure that the receiver has received the frame or not.

Flow Diagram for CSMA / CA:



Ethernet (802.3)

Wired LAN

- Most successful local area networking technology of last 20 years.
- Ethernet is a multiple access network with a set of nodes that send and receive frames over a shared link.

Uses CSMA/CD technology

- Carrier Sense Multiple Access with Collision Detection.
- A set of nodes send and receive frames over a shared link.
- Carrier sense means that all nodes can distinguish between an idle and a busy link.
- Collision detection means that a node listens as it transmits and can therefore detect when a frame it is transmitting has collided with a frame transmitted by another node.

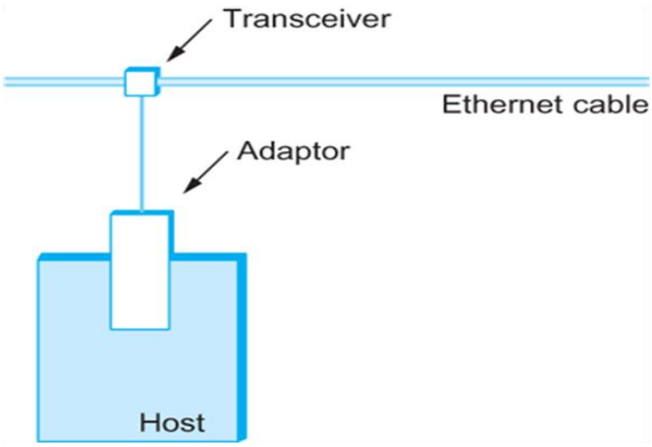
Ethernet can operate at A SPEED

- 100 Mbps (Fast Ethernet)
- 1000 Mbps (Gigabit Ethernet)

Ethernet also used in full duplex, point-to-point configuration

Physical Properties

- An Ethernet segment is implemented on a coaxial cable of up to 500 m.
- Hosts connect to an Ethernet segment by tapping into it.
- A **transceiver** (a small device directly attached to the tap) detects when the line is idle and drives signal when the host is transmitting.
- The transceiver also receives incoming signal.
- The transceiver is connected to an Ethernet adaptor which is plugged into the host.
- The protocol is implemented on the adaptor.



Ethernet transceiver and adaptor

- Multiple Ethernet segments Can be joined together by ***repeaters.***
- A *repeater* is a device that strengthens and forwards weakened digital signals.

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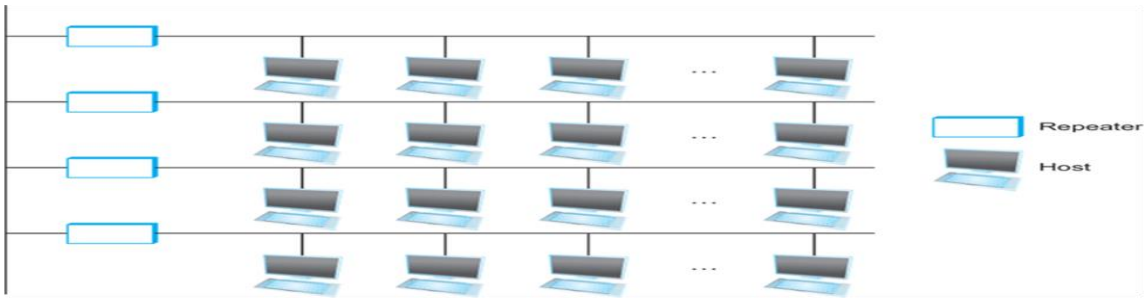


Ethernet repeater

Any signal placed on the Ethernet by a host is broadcast over the entire network

- Signal is propagated in both directions.
- Repeaters forward the signal on all outgoing segments.
- Terminators attached to the end of each segment absorb the signal.

Table : Types of 10 Mbps Ethernets

Name	Cable	Max. segment	Nodes /segment	Signaling technique	Topology used	Cable diameter	Access Method	advantage
10Base5	Thick coax	500 m	100	Baseband (Manchester)	Bus	10	CSMA/CD	Good For back bones
10Base2	Thin coax	200 m	30	Baseband (Manchester)	Bus	5	CSMA/CD	Cheapest system
10Base-T	Twisted pair	100 m	1024	Based band (Manchester)	Star	0.4 to 0.6	CSMA/CD	Easy maintenance
10Base-F	Fiber optics	2000m	1024	Manchester/ on-off	Star	62.5/125 pm	CSMA/CD	Best between buildings

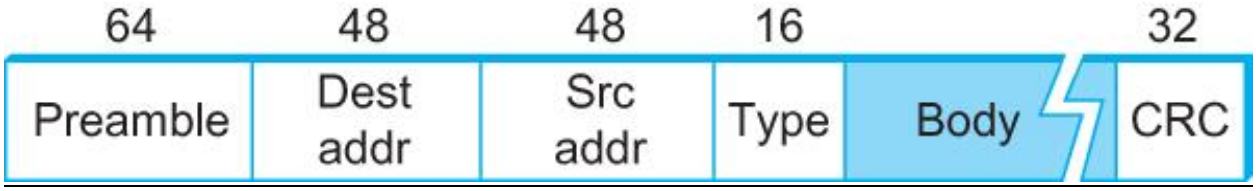
10Base5 cable. 10Base5 cable or **"Thick Ethernet"** or Thicknet is the cable which is the oldest in the category. it is called as thicknet because of the use of thick coaxial cable. The cable is marked after each 2.5 meters.

10Base2 Cable. 10Base2 cable also called **"Thin Ethernet" or Thinnet or cheapnet or cheapernet,** was designed after the thick Ethernet cable. This type of cable is usually thin, flexible and bends easily. It also make use of bus topology. It is also a coaxial cable that is having a smaller diameter than the 10Base5 cable.

10BaseT Cable. l0BaseT cable or **"Twisted Pair"** Cable is cheapest and easiest to maintain. This type of cabling is most popular among local area networks. It make use of unshielded twisted pair and provides maximum segment length of 100 m. It make use of start topology. In this type of network, every station is having a wired link to a central device, called "Hub".

l0BaseF Cable. 10BaseF cable or **"Fiber Optics Cable"** is the most efficient and fastest cable in the category of cables for 802 LANs. The fiber optic cable is very expensive as compared to above discussed cables but it offers a very high data transmission speed and noise immunity. This type of cabling is preferred for running networks between buildings or widely separated hubs. It has the highest length per cable segment i.e. 2000 meters and it can support 1024 nodes per cable segment.

Ethernet Frame Format



Preamble (64bit):
- allows the receiver to synchronize with the signal .
(sequence of alternating 0s and 1s).

Host and Destination Address physical addresses of the nodes(48bit each).

Packet type (16bit): - acts as demux key to identify the higher level protocol

Data (up to 1500 bytes)
- Minimally a frame must contain at least 46 bytes of data.
Frame must be long enough to detect collision.

CRC (32bit) - error detection

Ethernet Addresses

- Each host on an Ethernet (in fact, every Ethernet host in the world) has a unique Ethernet Address.
- The address belongs to the adaptor, not the host.
  - It is usually burnt into ROM.
- Ethernet addresses are typically printed in a human readable format
  - As a sequence of six numbers separated by colons.
  - Each number corresponds to 1 byte of the 6 byte address and is given by a pair of hexadecimal digits, one for each of the 4-bit nibbles in the byte
  - For example, 8:0:2b:e4:b1:2 is
    - 00001000 00000000 00101011 11100100 10110001 00000010
- To summarize, an Ethernet adaptor receives all frames and accepts
  - Frames addressed to its own address
  - Frames addressed to the broadcast address
  - Frames addressed to a multicast addressed if it has been instructed

Ethernet Transmitter Algorithm

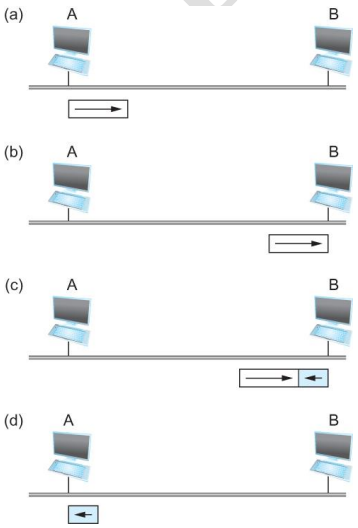
- Algorithm is defined as follows,"When the adaptor has a frame to send and theline is idle, it transmits the frame immediately"

- When the adaptor has a frame to send and the line is busy, it waits for the line to go idle and then transmits immediately.
- The Ethernet is said to be 1-persistent protocol because an adaptor with a frame to send transmits with probability 1 whenever a busy line goes idle.When this happens, the two (or more) frames are said to be *collide* on the network.
- Since Ethernet supports collision detection, each sender is able to determine that a collision is in progress.

- At the moment an adaptor detects that its frame is colliding with another, it first makes sure to transmit a 32-bit jamming sequence and then stops transmission.
    - Thus, a transmitter will minimally send 96 bits in the case of collision 64-bit preamble + 32-bit jamming sequence
- One way that an adaptor will send only 96 bit (called a *runt frame*) is if the two hosts are close to each other.
    - Had they been farther apart,They would have had to transmit longer, and thus send more bits, before detecting the collision.
- The worst case scenario happens when the two hosts are at opposite ends of the Ethernet.
- To know for sure that the frame its just sent did not collide with another frame, the transmitter may need to send as many as 512 bits.
    - Every Ethernet frame must be at least 512 bits (64 bytes) long.
        - 14 bytes of header + 46 bytes of data + 4 bytes of CRC

Ethernet Transmitter Algorithm

- A begins transmitting a frame at time *t*
- *d* denotes the one link latency
- The first bit of A's frame arrives at B at time *t* + *d*
- Suppose an instant before host A's frame arrives, host B begins to transmit its own frame
- B's frame will immediately collide with A's frame and this collision will be detected by host B
- Host B will send the 32-bit jamming sequence
- Host A will not know that the collision occurred until B's frame reaches it, which will happen at *t* + 2 * *d*
- Host A must continue to transmit until this time in order to detect the collision
- Host A must transmit for *2 * d* to be sure that it detects all possible collisions



Worst-case scenario:

(a) A sends a frame at time *t;*

(b) A's frame arrives at B at time *t + d;*

(c) B begins transmitting at time t + d and collides with A's frame;

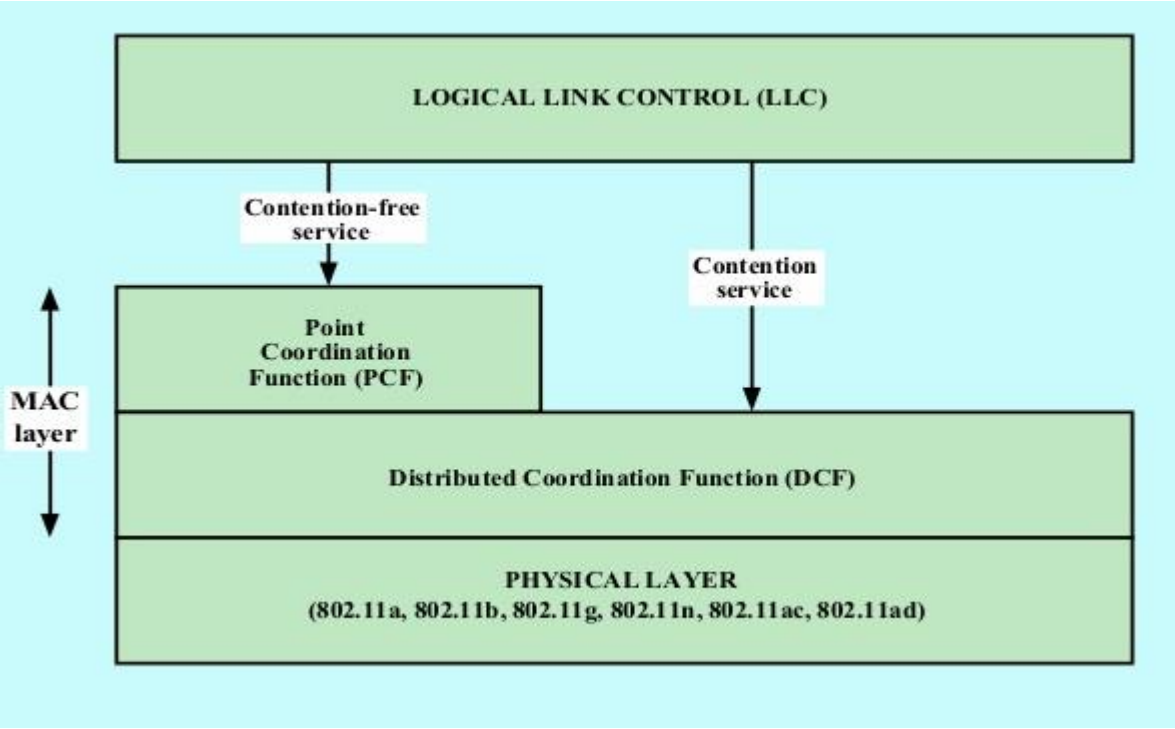
(d) B's runt (32-bit) frame arrives at A at time *t + 2d.*

- Once an adaptor has detected a collision, and stopped its transmission, it waits a certain amount of time and tries again.
- Each time the adaptor tries to transmit but fails, it doubles the amount of time it waits before trying again.
- This strategy of doubling the delay interval between each retransmission attempt is known as *Exponential Backoff.*

Wireless LAN (IEEE 802.11) / Wi-Fi

1. Wireless networking that is Standard IEEE 802.11 is a rapidly evolving technology for connecting computers. It is also known as Wi-Fi.
2. Like Ethernet and token ring siblings, 802.11 is designed for use in a limited geographical area (homes, office buildings, campuses)

Protocol Architecture



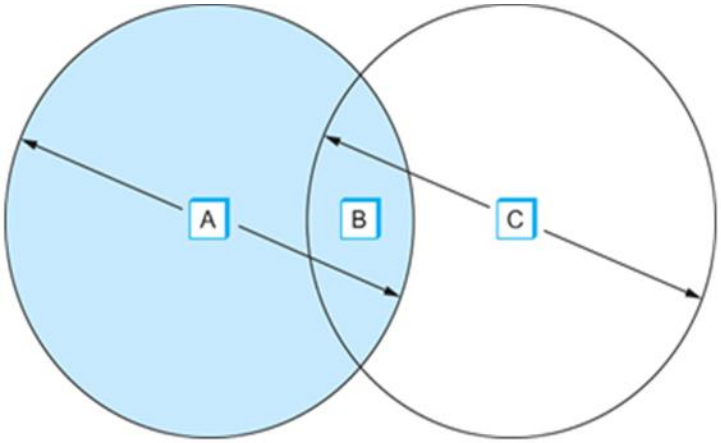
Physical Layer Properties:

- 802.11 runs over six different physical layer protocols (so far). Five are based on spread spectrum radio, and one on diffused infrared. The fastest runs at a maximum of 54 Mbps.

- Original 802.11 standard defined two radio-based physical layer standard
  - One using the frequency hopping
    - Over 79 1-MHz-wide frequency bandwidths
  - Second using direct sequence
    - Using 11-bit chipping sequence
  - Both standards run in the 2.4-GHz and provide up to 2 Mbps

- Then physical layer standard 802.11b was added
  - Using a variant of direct sequence 802.11b provides up to 11 Mbps
  - Uses license-exempt 2.4-GHz band
- Then came 802.11a which delivers up to 54 Mbps using OFDM
  - 802.11a runs on license-exempt 5-GHz band
- Most recent standard is 802.11g which is backward compatible with 802.11b
  - Uses 2.4 GHz band, OFDM and delivers up to 54 Mbps

Example of a wireless network:

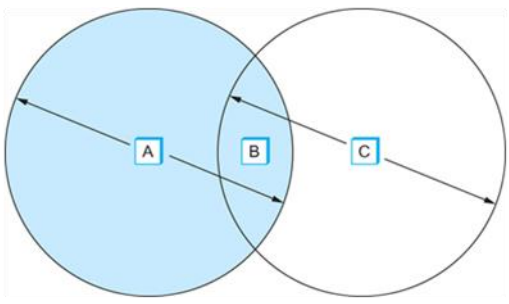


IEEE 802.11 Multiple Access Collision Avoidance (MACA)

- Multiple Access with Collision Avoidance (MACA) is a protocol for slotted media access control used in wireless LAN data transmission.
- MACA is used to avoid data collisions caused by hidden station problems as well as simplifying known station problems.

Hidden nodes problem

- Suppose both A and C want to communicate with B and so they each send it a frame.
  - A and C are unaware of each other since their signals do not carry that far
  - These two frames collide with each other at B
    - But unlike an Ethernet, neither A nor C is aware of this collision
  - A and C are said to ***hidden nodes*** with respect to each other

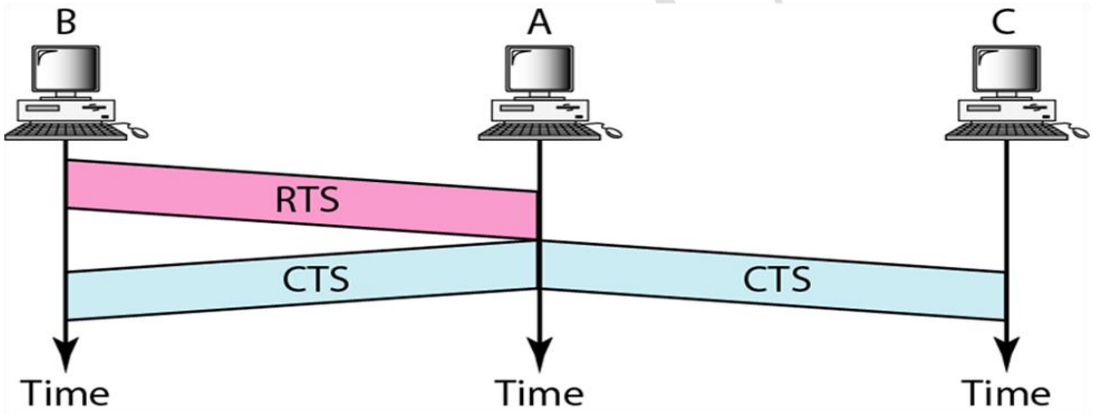


The "Hidden Node" Problem. Although A and C are hidden from each other, their signals can collide at B. (B's reach is not shown.)

- **Key Idea-Solution**
  - Sender and receiver exchange control frames with each other before the sender actually transmits any data.
  - This exchange informs all nearby nodes that a transmission is about to begin
  - Sender transmits a *Request to Send* (RTS) frame to the receiver.
    - The RTS frame includes a field that indicates how long the sender wants to hold the medium

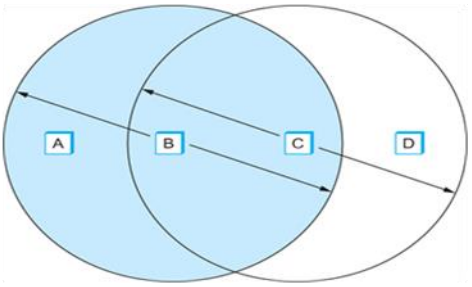
  -Length of the data frame to be transmitted
  - Receiver replies with a *Clear to Send* (CTS) frame
    - This frame echoes this length field back to the sender
- Any node that sees the CTS frame knows that
  - it is close to the receiver, therefore
  - cannot transmit for the period of time it takes to send a frame of the specified length
- Any node that sees the RTS frame but not the CTS frame
  - is not close enough to the receiver to interfere with it, and
  - so is free to transmit

Use of handshaking to prevent hidden station problem



Exposed node problem

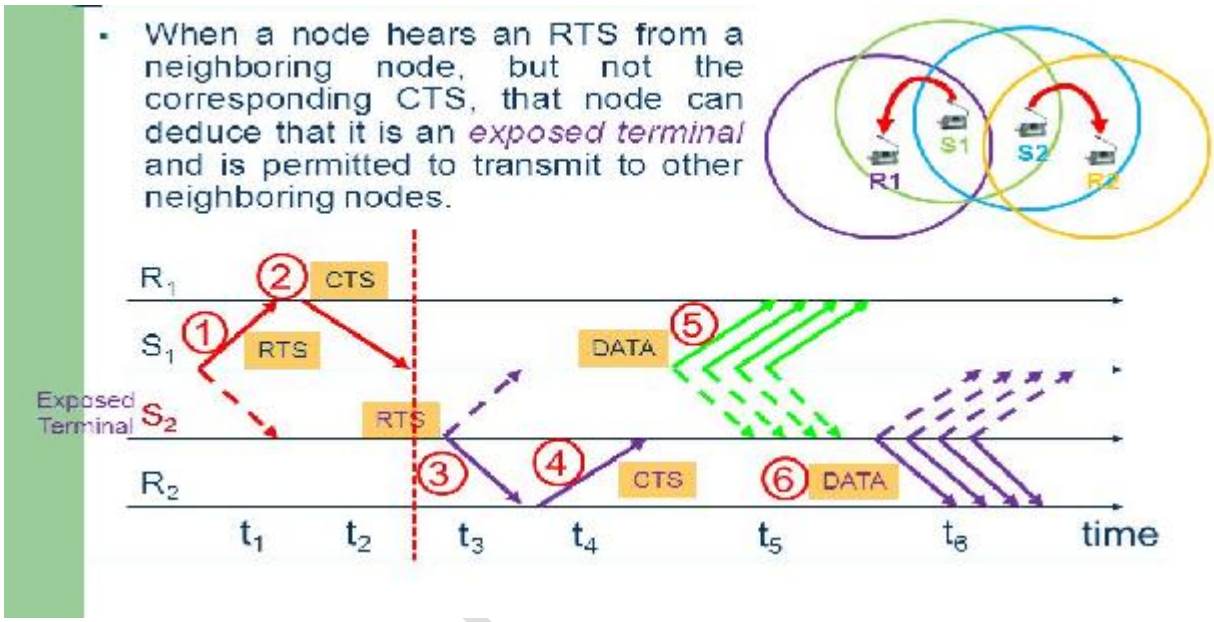
- Another problem called ***exposed node*** problem occurs
  - Suppose B is sending to A. Node C is aware of this communication because it hears B's transmission.
  - It would be a mistake for C to conclude that it cannot transmit to anyone just because it can hear B's transmission.
  - Suppose C wants to transmit to node D.
  - This is not a problem since C's transmission to D will not interfere with A's ability to receive from B.



C->D progress
If B->A not possible
Since C is exposed to B

Although B and C are exposed to each other's signals, there is no interference if B transmits to A while C transmits to D. (A and D's reaches are not shown.)

Solution for exposed node problem:



MACA: Medium Access Collision Avoidance:

MACA is for Wireless LANs

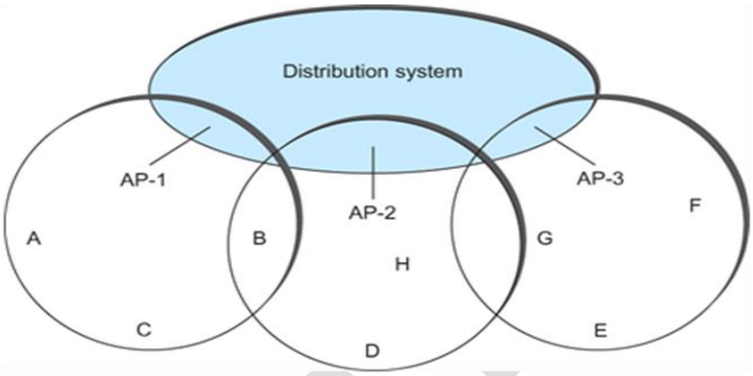
- Receiver sends an ACK to the sender after successfully receiving a frame
- All nodes must wait for this ACK before trying to transmit
- If two or more nodes detect an idle link and try to transmit an RTS frame at the same time
  - Their RTS frame will collide with each other
- 802.11 does not support collision detection
  - So the senders realize the collision has happened when they do not receive the CTS frame after a period of time
  - In this case, they each wait a random amount of time before trying again.
  - The amount of time a given node delays is defined by the same ***exponential backoff*** algorithm used on the Ethernet.
- 802.11 is suitable for an ad-hoc configuration of nodes that may or may not be able to communicate with all other nodes.
- Nodes are free to move around

- The set of directly reachable nodes may change over time
- To deal with this mobility and partial connectivity,
  - 802.11 defines additional structures on a set of nodes
  - Instead of all nodes being created equal,
    - some nodes are allowed to roam
    - some are connected to a wired network infrastructure

- they are called *Access Points* (AP) and they are connected to each other by a so-called *distribution system*

IEEE 802.11 – Distribution System

- Following figure illustrates a distribution system that connects three access points, each of which services the nodes in the same region
- Each of these regions is analogous to a cell in a cellular phone system with the APIs playing the same role as a base station
- The distribution network runs at layer 2 of the ISO architecture

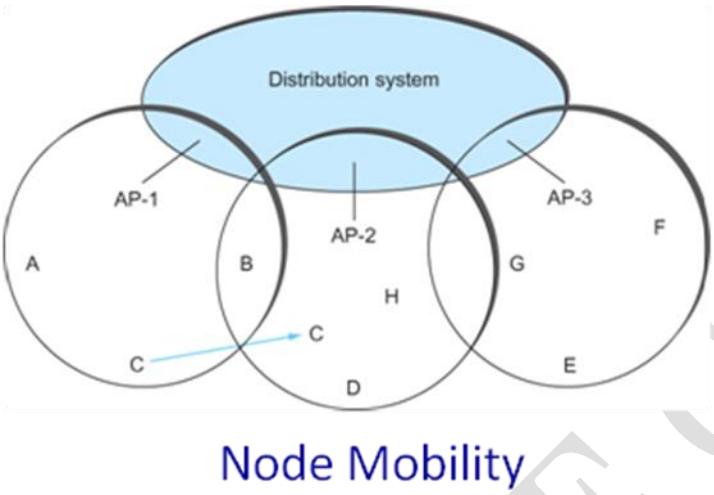


Access points connected to a distribution network

- Although two nodes can communicate directly with each other if they are within reach of each other, the idea behind this configuration is
  - Each nodes associates itself with one access point
  - For node A to communicate with node E, A first sends a frame to its AP-1 which forwards the frame across the distribution system to AP-3, which finally transmits the frame to E
- The technique for selecting an AP is called *scanning*
  - The node sends a *Probe* frame
  - All APs within reach reply with a *Probe Response* frame
  - The node selects one of the access points and sends that AP an *Association Request* frame
  - The AP replies with an *Association Response* frame
- A node engages this protocol whenever
  - it joins the network, as well as
  - when it becomes unhappy with its current AP
    - This might happen, for example, because the signal from its current AP has weakened due to the node moving away from it
    - Whenever a node acquires a new AP, the new AP notifies the old AP of the change via the distribution system

Node Mobility

- Consider the situation shown in the following figure when node C moves from the cell serviced by AP-1 to the cell serviced by AP-2.
- As it moves, it sends *Probe* frames, which eventually result in *Probe Responses* from AP-2.
- At some point, C prefers AP-2 over AP-1 , and so it associates itself with that access point.
  - This is called ***active scanning*** since the node is actively searching for an access point



Node Mobility

- APs also periodically send a *Beacon* frame that advertises the capabilities of the access point; these include the transmission rate supported by the AP
  - This is called ***passive scanning***
  - A node can change to this AP based on the *Beacon* frame simply by sending it an *Association Request* frame back to the access point.

IEEE 802.11 – Frame Format

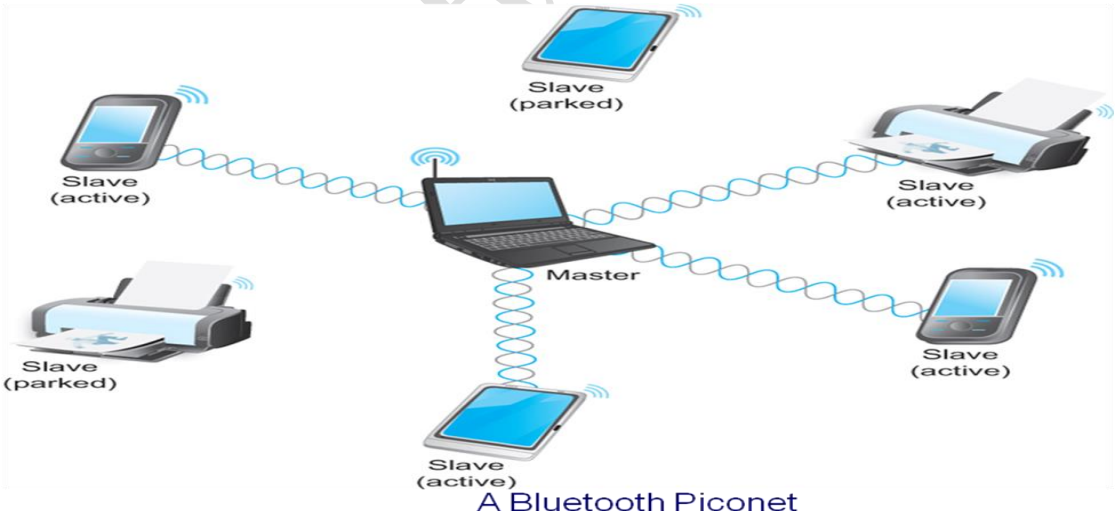
- Source and Destinations addresses: each 48 bits
- Data: up to 2312 bytes
- CRC: 32 bit
- Control field: 16 bits
  - Contains three subfields (of interest)
    - 6 bit **Type** field: indicates whether the frame is an RTS or CTS frame or being used by the scanning algorithm
    - A pair of 1 bit fields : called **ToDS** and **FromDS**
- Addr1 – target node
- Addr2 – immediate sender
- Addr3 – intermediate destination node
- Addr4 – original source

Bluetooth (802.15.1)

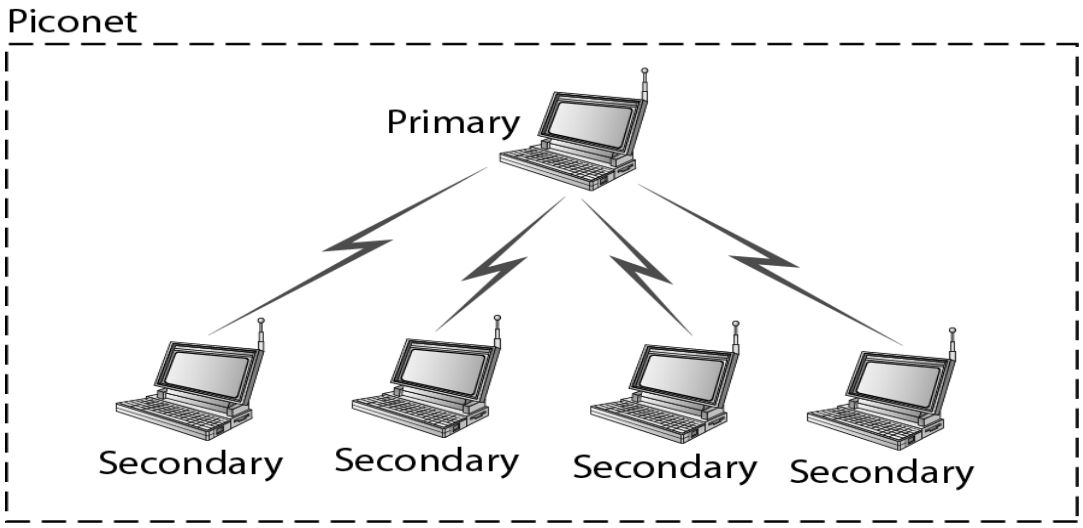
Bluetooth is a wireless LAN technology designed to connect devices of different functions such as telephones, notebooks, computers, cameras, printers, coffee makers, and so on. A Bluetooth LAN is an ad hoc network, which means that the network is formed spontaneously**.** Used for very short range communication between mobile phones, PDAs, notebook computers and other personal or peripheral devices.

Bluetooth is an adhoc network which means the network is formed spontaneously ie the devices find each other and make a network called piconet.

- Has a range of only 10 m
- Communication devices typically belong to one individual or group
  - Sometimes categorized as Personal Area Network (PAN)
- Version 2.0 provides speeds up to 2.1 Mbps
- Power consumption is low
- Bluetooth is specified by an industry consortium called the Bluetooth Special Interest Group
- It specifies an entire suite of protocols, going beyond the link layer to define application protocols, which it calls *profiles*, for a range of applications
  - There is a profile for synchronizing a PDA with personal computer
  - Another profile gives a mobile computer access to a wired LAN
- The basic Bluetooth network configuration is called a *piconet*
  - Consists of a master device and up to seven slave devices
  - Any communication is between the master and a slave
  - The slaves do not communicate directly with each other
  - A slave can be *parked*: set to an inactive, low-power state

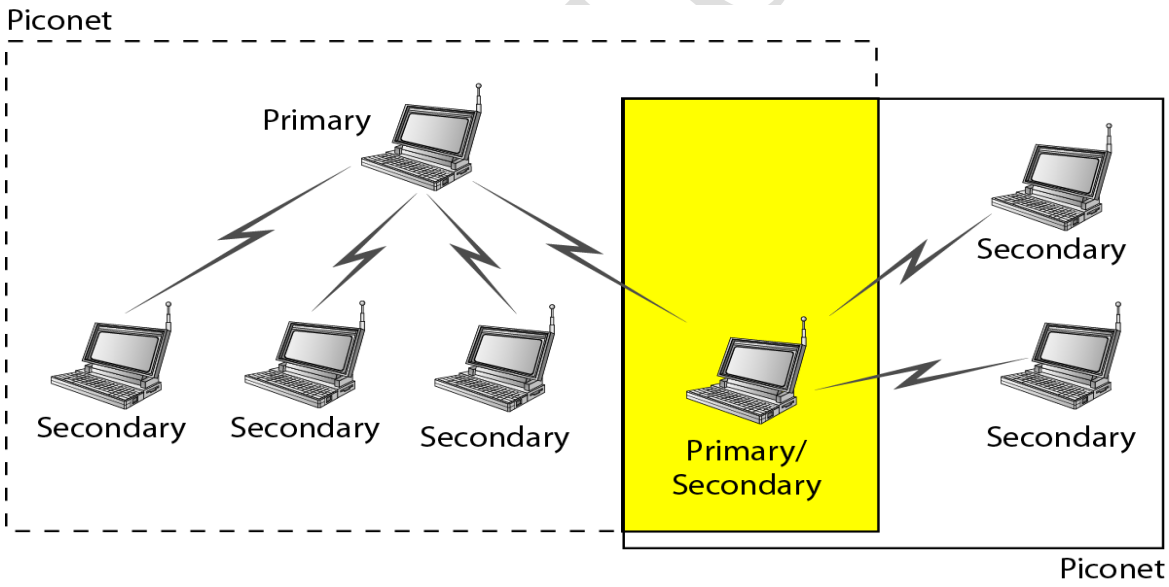


A Bluetooth Piconet

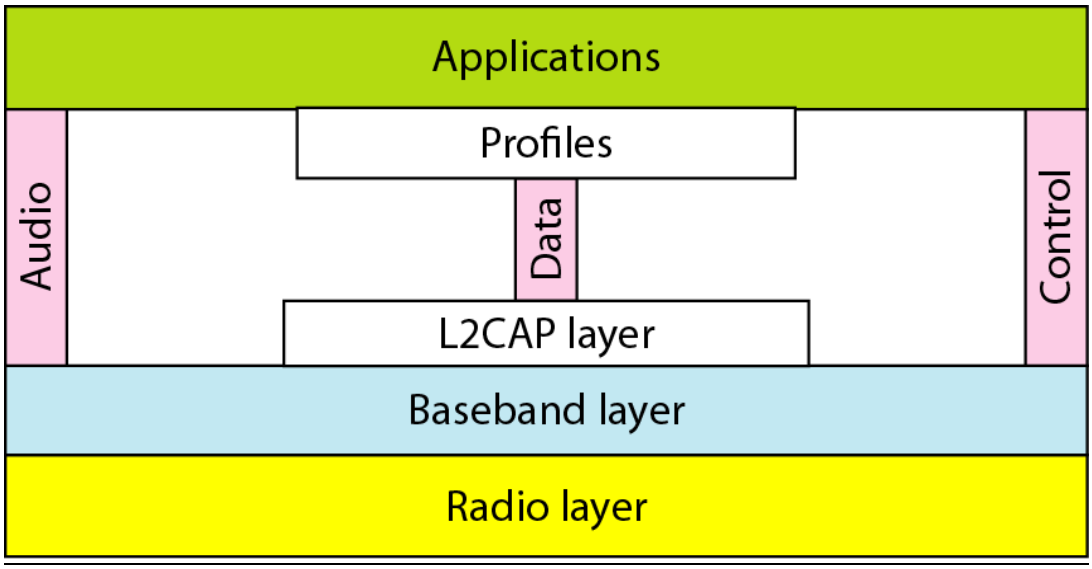


Scatternet

A scatternet is a number of interconnected piconets that supports communication between more than 8 devices. Scatternets can be formed when a member of one piconet (either the master or one of the slaves) elects to participate as a slave in a second, separate piconet. The device participating in both piconets can relay data between members of both ad hoc networks.



Bluetooth layers



L2CAP Layer

- Logical Link Control and Adaptation Layer
- Used for data exchange, It can do multiplexing

Baseband Layers

- Equivalent to MAC sublayer in LAN
- The primary and secondary stations communicate with each other using time slots (TDMA-Time Division Multiple Access)

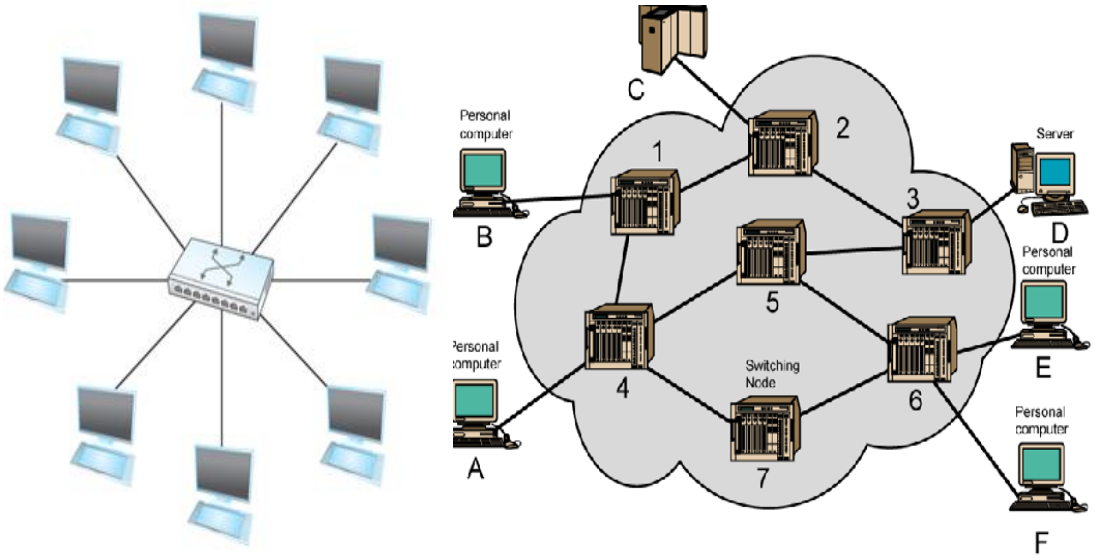
Radio layer

- Equivalent to physical layer of the internet model. Bluetooth devices are low-power and have a range of 10 m

Switch

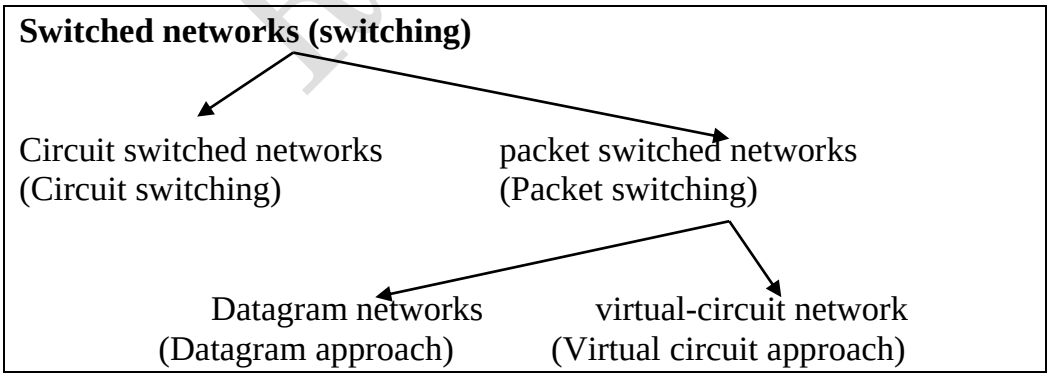
1. A mechanism that allows us to interconnect links(LANs) to form a large network is called **switching.**
2. Switches are devices capable of creating temporary connections between two or more devices linked to the switch.
3. A switch's primary job is to receive incoming packets on one of its links and to transmit them on some other link. This function is referred as switching and forwarding.
4. A multi-input, multi-output device which transfers packets from an input to one or more outputs.

EXAMPLE:



Advantages of switches:

1. Large networks can be built by interconnecting a number of switches.
2. We can connect switches to each other and to hosts using point-to-point links, which typically means that we can build networks of large geographic scope.
3. Adding a new host to the network by connecting it to a switch does not necessarily mean that the hosts already connected will get worse performance from the network.
4. A switch is connected to a set of links and for each of these links, runs the appropriate data link protocol to communicate with each node attached on link.



Circuit switched networks

In circuit switched network, it consists of set of switches connected by physical links. A connection between two stations is a dedicated path made of one or more links. Physical layer is responsible for circuit switching.

The communication in circuit-switched network has three phases,

1) Connection setup
2) Data transfer
3) Teardown (connection termination)

In these types of networks, data are not packetized and there is a continuous flow by the source station to destination station.

Packet switched networks

If the message is passing through a packet switched network, it needs to be divided into packets of fixed or variable size. This type of switching is done by network layer.

The two approaches commonly used are,

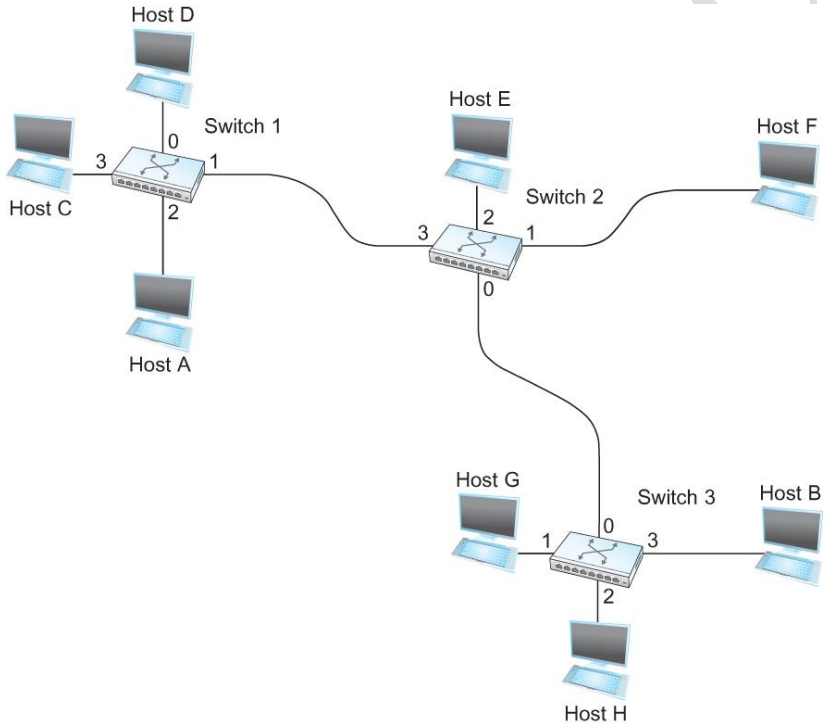
- ***Datagram or Connectionless approach***
- ***Virtual circuit or Connection-oriented approach***

Datagram approach

Every packet contains enough information i.e. destination address that enable any switch to decide where the packet has to go.

Consider An example network with many hosts for datagram forwarding,

- To decide how to forward a packet, a switch consults a *forwarding table* (sometimes called a *routing table*)



Destination	Port
A	3
B	0
C	3
D	3
E	2
F	1
G	0
H	0

Forwarding Table for Switch 2

Fig :
Datagram forwarding : an example network

Characteristics of Connectionless (Datagram) Network

- A host can send a packet anywhere at any time.
- When a host sends a packet, it has no way of knowing if the network is capable of delivering it or if the destination host is even up and running.
- Each packet is forwarded independently of previous packets that might have been sent to the same destination.

- Thus two successive packets from host A to host B may follow completely different paths
- A switch or link failure might not have any serious effect on communication if it is possible to find an alternate route around the failure and update the forwarding table accordingly.

Virtual Circuit Switching

- Widely used technique for packet switching
- Uses the concept of ***virtual circuit*** **(VC)**
- Also called a connection-oriented model
- First set up a virtual connection from the source host to the destination host and then send the data

- Host A wants to send packets to host B

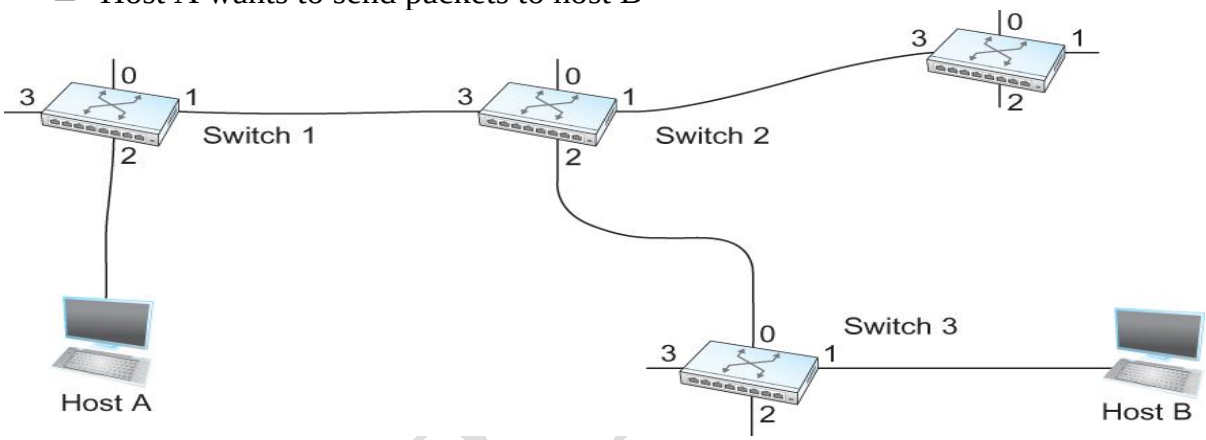


Fig : An example of a virtual circuit network

Two-stage process

- Connection setup
- Data Transfer

Connection setup

- Establish "connection state" in each of the switches between the source and destination hosts
- The connection state for a single connection consists of an entry in the "VC table" in each switch through which the connection passes

One entry in the VC table on a single switch contains

- A virtual circuit identifier (VCI) that uniquely identifies the connection at this switch and that will be carried inside the header of the packets that belong to this connection
- An incoming interface on which packets for this VC arrive at the switch
- An outgoing interface in which packets for this VC leave the switch

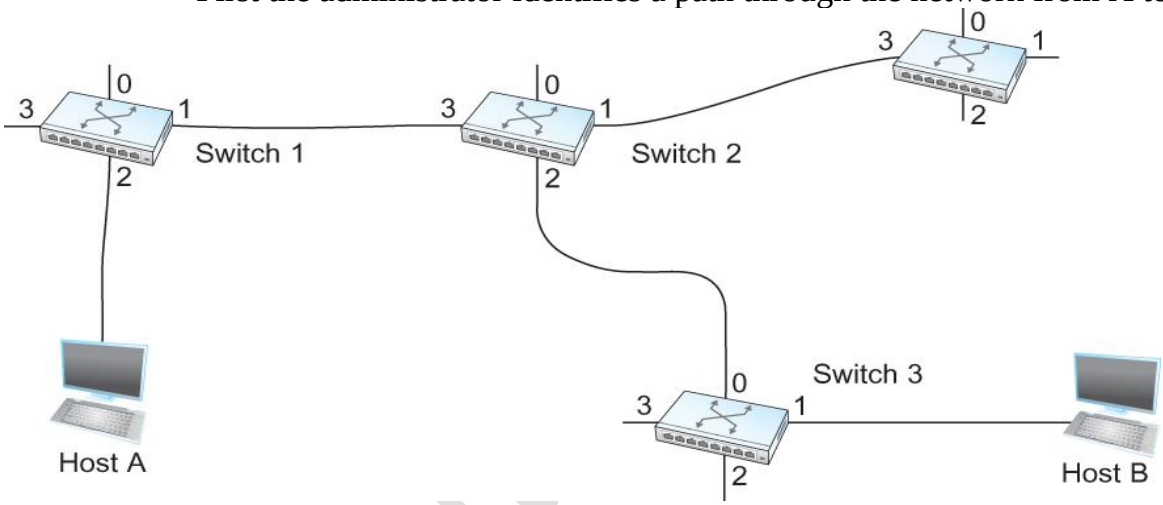
A potentially different VCI that will be used for outgoing packets.

Two broad classes of approach to establishing connection state

- Network Administrator will configure the state
  - The virtual circuit is permanent (PVC)
  - The network administrator can delete this
  - Can be thought of as a long-lived or administratively configured VC
- A host can send messages into the network to cause the state to be established
  - This is referred as signalling and the resulting virtual circuit is said to be switched (SVC)
  - A host may set up and delete such a VC dynamically without the involvement of a network administrator

Let's assume that a network administrator wants to manually create a new virtual connection from host A to host B

- First the administrator identifies a path through the network from A to B



The administrator then picks a VCI value that is currently unused on each link for the connection

- For our example,
  - Suppose the VCI value 5 is chosen for the link from host A to switch 1
  - 11 is chosen for the link from switch 1 to switch 2
  - So the switch 1 will have an entry in the VC table

For switch 1

Incoming Interface	Incoming VC	Outgoing Interface	Outgoing VC
2	5	1	11

Similarly, suppose

- VCI of 7 is chosen to identify this connection on the link from switch 2 to switch 3
- VCI of 4 is chosen for the link from switch 3 to host B
- Switches 2 and 3 are configured with the following VC table

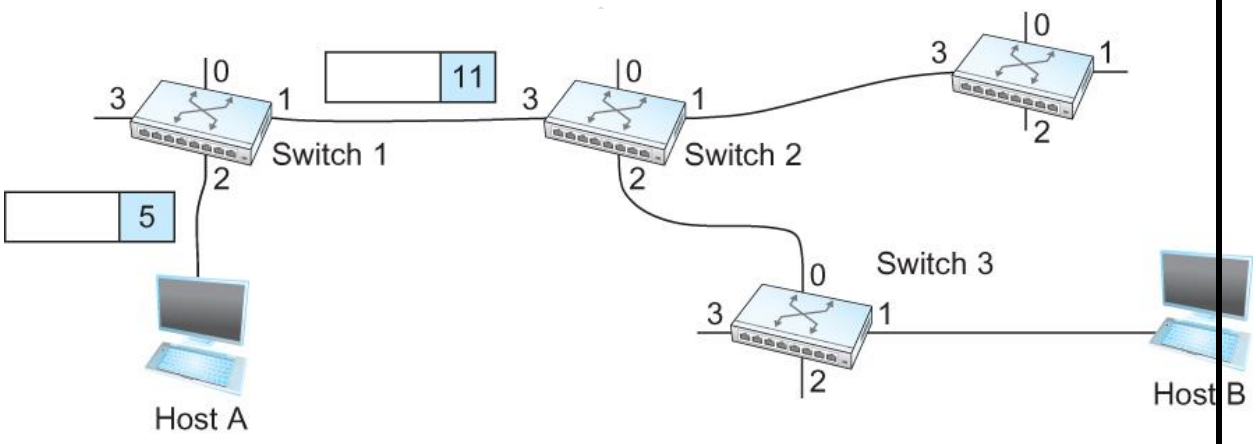
Switch 2

Incoming Interface	Incoming VC	Outgoing Interface	Outgoing VC
3	11	2	7

Switch 3

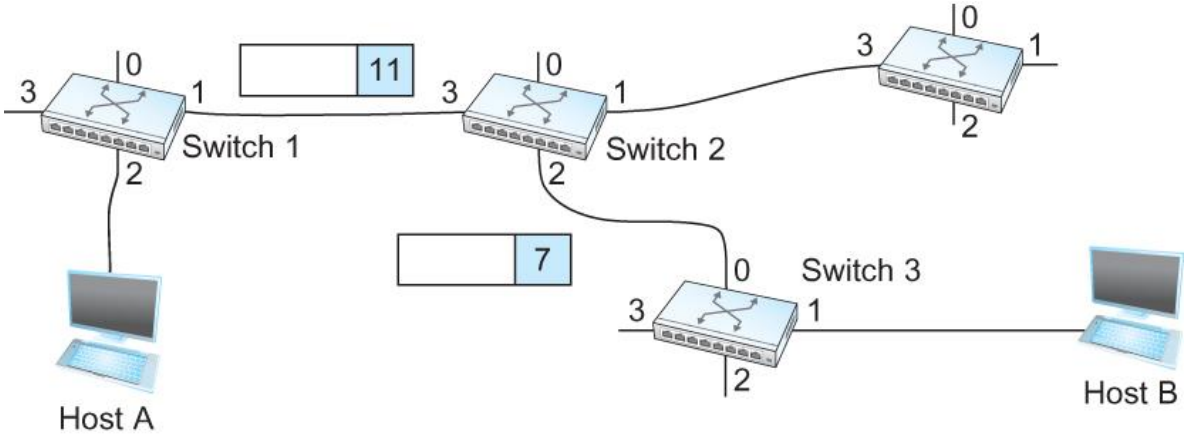
Incoming Interface	Incoming VC	Outgoing Interface	Outgoing VC
0	7	1	4

- For any packet that A wants to send to B, A puts the VCI value 5 in the header of the packet and sends it to switch 1
- Switch 1 receives any such packet on interface 2, and it uses the combination of the interface and the VCI in the packet header to find the appropriate VC table entry.
- The table entry on switch 1 tells the switch to forward the packet out of interface 1 and to put the VCI value 11 in the header



Packet will arrive at switch 2 on interface 3 bearing VCI 11

- Switch 2 looks up interface 3 and VCI 11 in its VC table and sends the packet on to switch 3 after updating the VCI value appropriately
- This process continues until it arrives at host B with the VCI value of 4 in the packet
- To host B, this identifies the packet as having come from host A



Data transfer

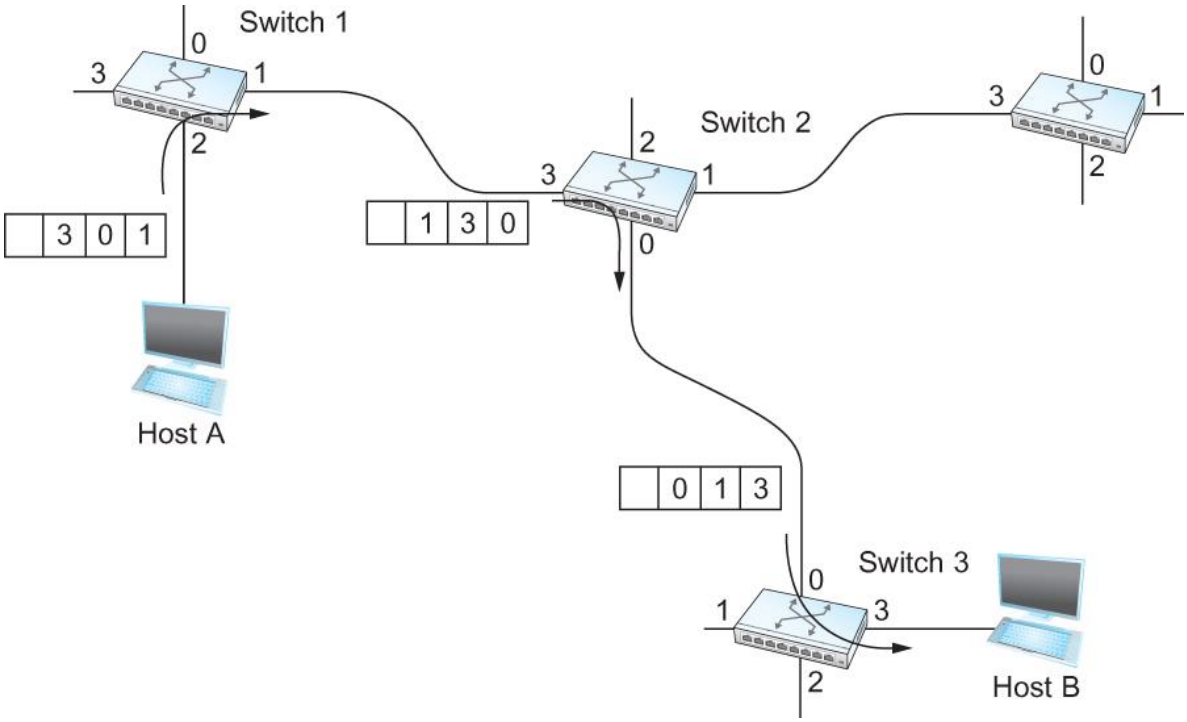
The data transfer phase is active until the source sends all its frame to destination when the connection state is established.

Teardown

In this phase, source A, after sending all frames to B, sends a special frame called teardown request. Destination B responds with a teardown confirmation frame and all switches delete the corresponding entry from their tables.

Source Routing

- Its a type if less commonly used approach in packet switching
- All the information about network topology that is required to switch a packet across the network is provided by the source host



Circuit switching	Packet switching
The connection between two station is a dedicated path using physical links	Virtual connection is established between two station
Physical layer is responsible	Network layer is responsible
Messages are not packetized and there is a continuous flow of messages	Messages are divided into packets of fixed or variable size
Used in telephone companies	Used in switched WANs such as frame relay and ATM networks
-	Types are • Datagram approach • Virtual circuit approach • Source routing

Bridges

- Bridge is a connecting device that is used to connect two or more LANs. It operates in both the physical and the data link layer.
- As a physical layer device, it regenerates the signal it receives.
- As a data link layer device, the bridge can check the physical (MAC) addresses (source and destination) contained in the frame.
- Switch that is used to forward packets between shared-media LANs such as Ethernets.
- It is also called as LAN switches.

Transparent bridges

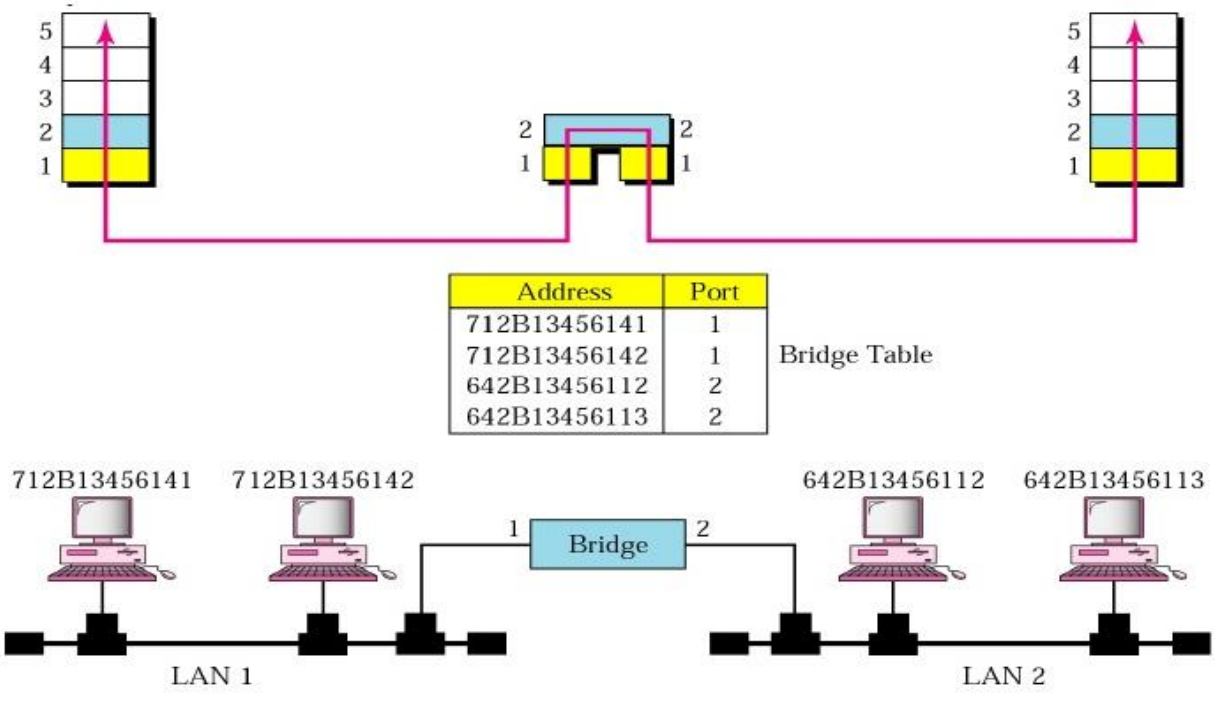
It is a bridge in which the stations are completely unaware of bridge's existence. If a bridge is added or deleted from the system, reconfiguration is not needed.
According to the IEEE 802.1 d specification, a system equipped with transparent bridges must meet three criteria, as follows

Functions of bridges (three criteria)

1. Frame filtering or forwarding
2. Learning
3. Avoidance of loops in system

Frame filtering or forwarding

Bridges has a table which is used for filtering or forwarding decisions. It can check the destination address of frame and decide if the frame has to be forwarded or dropped. If the frame is to be forwarded, the decision must specify the port.



Let us give an example. In Figure two LANs are connected by a bridge. If a frame destined for station 712B13456142 arrives at port 1, the bridge consults its table to find the departing port. According to its table, frames for 7l2B 13456142 leave through port 1; therefore, there is no need for forwarding, and the frame is dropped. On the other hand, if a frame for 712B13456141 arrives at port 2, the departing port is port 1 and the frame is forwarded.

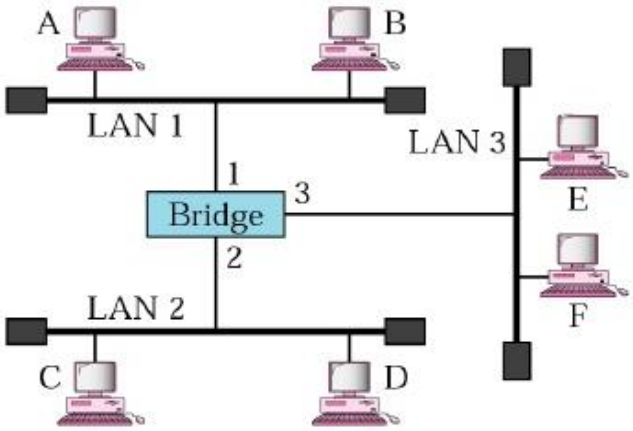
Learning

The earliest bridges had forwarding tables that were static. The systems administrator would manually enter each table entry during bridge setup. Although the process was simple, it was not practical.

If a station was added or deleted, the table had to be modified manually. The same was true if a station's MAC address changed, which is not a rare event.

A better solution to the static table is a dynamic table that maps addresses to ports automatically. To make a table dynamic, we need a bridge that gradually learns from the frame movements.

To do this, the bridge inspects both the destination and the source addresses. The destination address is used for the forwarding decision (table lookup); the source address is used for adding entries to the table and for updating purposes.



Address	Port

a. Original

Address	Port
A	1

b. After A sends a frame to D

Address	Port
A	1
E	3

c. After E sends a frame to A

Address	Port
A	1
E	3
B	1

d. After B sends a frame to C

1. When station A sends a frame to station D, the bridge does not have an entry for either D or A. The frame goes out from all three ports; the frame floods the network. However, by looking at the source address, the bridge learns that station A must be located on the LAN connected to port 1. This means that frames destined for A, in the future, must be sent out through port 1. The bridge adds this entry to its table. The table has its first entry now.

2. When station E sends a frame to station A, the bridge has an entry for A, so it forwards the frame only to port 1. There is no flooding. In addition, it uses the source address of the frame, E, to add a second entry to the table.
3. When station B sends a frame to C, the bridge has no entry for C, so once again it floods the network and adds one more entry to the table.
4. The process of learning continues as the bridge forwards frames.

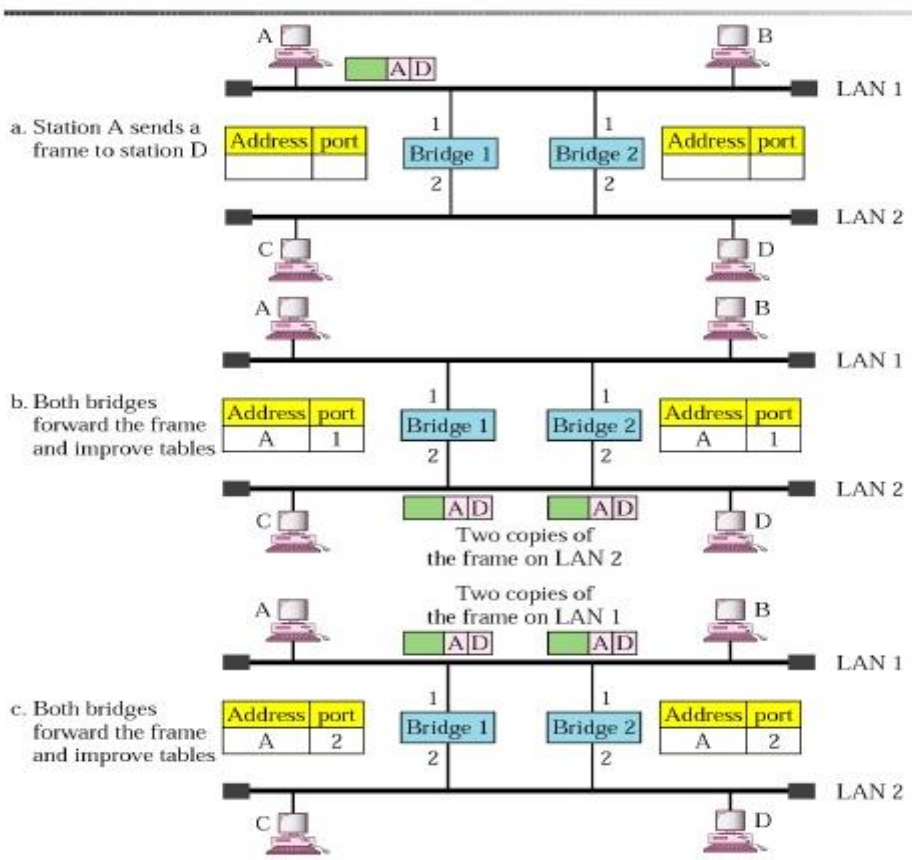
Loop Problem

Loop Problem Transparent bridges work fine as long as there are no redundant bridges in the system. Systems administrators, however, like to have redundant bridges (more than one bridge between a pair of LANs) to make the system more reliable.

If a bridge fails, another bridge takes over until the failed one is repaired or replaced. Redundancy can create loops in the system, which is very undesirable.

Figure shows a very simple example of a loop created in a system with two LANs connected by two bridges.

1. Station A sends a frame to station D. The tables of both bridges are empty. Both forward the frame and update their tables based on the source address A.
2. Now there are two copies of the frame on LAN 2. The copy sent out by bridge 1 is received by bridge 2, which does not have any information about the destination address D; it floods the bridge. The copy sent out by bridge 2 is received by bridge 1 and is sent out for lack of information about D. Note that each frame is handled separately because bridges, as two nodes on a network sharing the medium, use an access method such as CSMA/CD. The tables of both bridges are updated, but still there is no information for destination D.
3. Now there are two copies of the frame on LAN 1. Step 2 is repeated, and both copies flood the network.
4. The process continues on and on. Note that bridges are also repeaters and regenerate frames. So in each iteration, there are newly generated fresh copies of the frames. To solve the looping problem, the IEEE specification requires that bridges use the spanning tree algorithm to create a loopless topology.

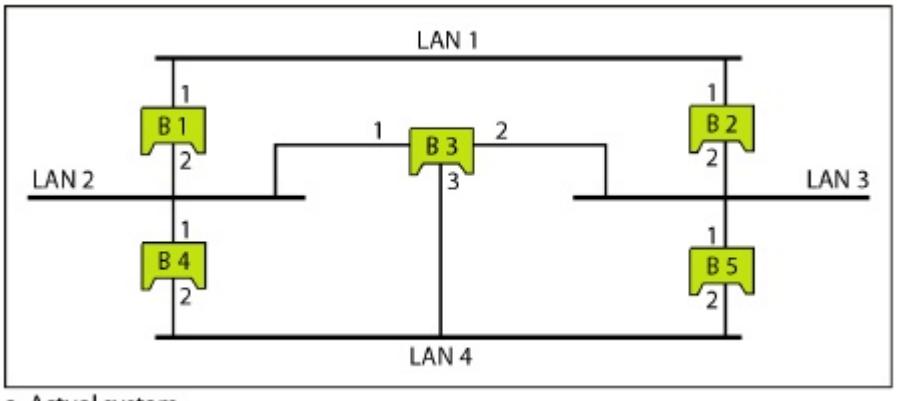


Spanning Tree

In graph theory, a **spanning tree** is a graph in which there is no loop. In a bridged LAN, this means creating a topology in which each LAN can be reached from any other LAN through one path only (no loop).

We cannot change the physical topology of the system because of physical connections between cables and bridges, but we can create a logical topology that overlays the physical one.

The following figure shows a system with four LANs and five bridges.

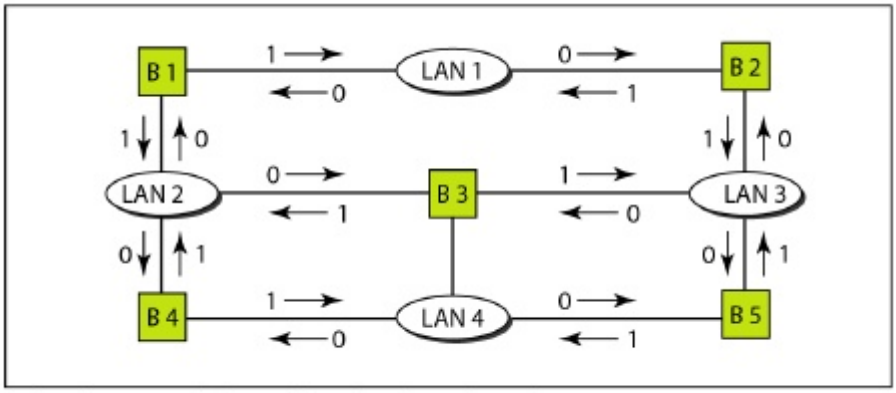


a. Actual system

The graph representation with cost assigned to each arc of above system is as follows.

The connecting arcs show the connection of a LAN to a bridge and vice versa. To find the spanning tree, we need to assign a cost (metric) to each arc.

[Note : assign 1 from Bridge to LAN, assign 0 from LAN to Bridge]

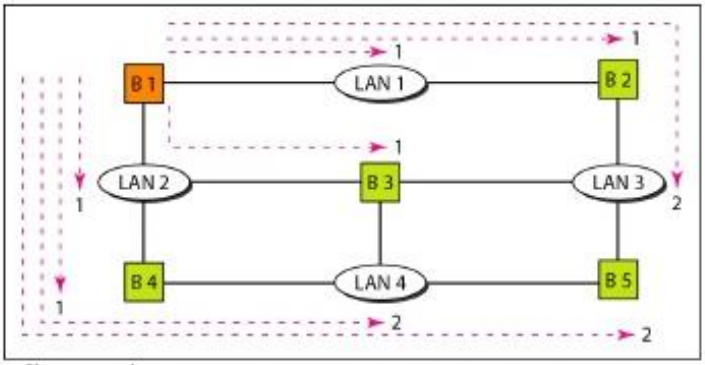


b. Graph representation with cost assigned to each arc

In spanning tree approach, the cost to each arc is assigned by system administrator.

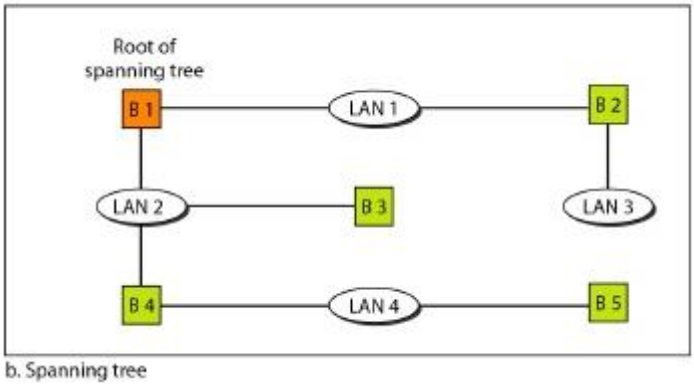
The process to find the spanning tree involves three steps:

1. Every bridge has a built-in ID (normally the serial number, which is unique). Each bridge broadcasts this ID so that all bridges know which one has the smallest ID. The bridge with the smallest ID is selected as the *root* bridge (root of the tree). We assume that bridge B1 has the smallest ID. It is, therefore, selected as the root bridge.
2. The algorithm tries to find the shortest path (a path with the shortest cost) from the root bridge to every other bridge or LAN as follows,



a. Shortest paths

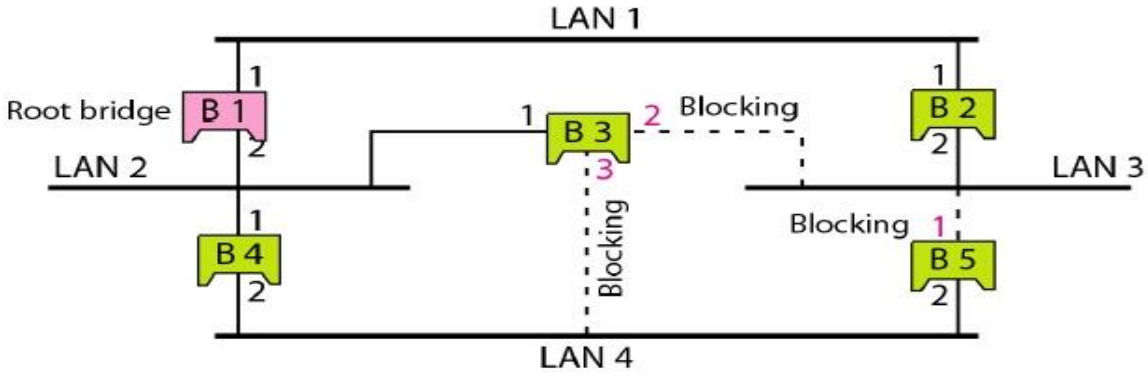
3. The combination of the shortest paths creates the shortest tree, which is also shown in the above figure.



b. Spanning tree

4. Based on the spanning tree approach some ports are forwarding ports, which forward a frame that the bridge receives. Some ports are blocking ports which block the frames received by bridge.

Forwarding and blocking ports after using spanning tree algorithm



Ports 2 and 3 of bridge B3 are blocking ports. Port1 of bridge B5 is also blocking port.

- Spanning tree algorithm is a **dynamic algorithm** .
- Each bridge is equipped with a software process that carries the process dynamically. Each bridge send special messages to one another called bridge protocol data units (BPDUs) to update spanning tree.

Limitations of Bridges

On the issue of scale, it is not realistic to connect more than a few LANs by means of bridges, where in practice "few" typically means "tens of.

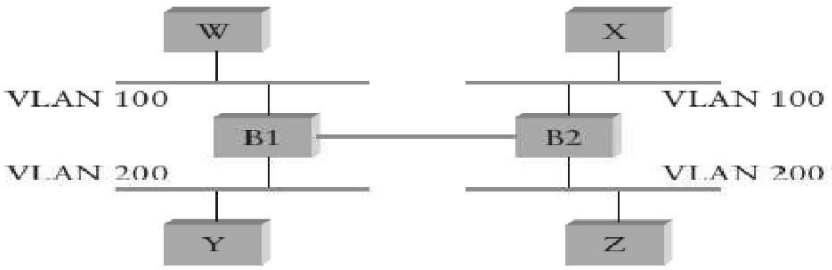
One reason for this is that the spanning tree algorithm scales linearly; that is, there is no provision for imposing a hierarchy on the extended LAN.

A second reason is that bridges forward all broadcast frames.

One approach to increasing the scalability of extended LANs is the *virtual LAN*(VLAN). VLANs allow a single extended LAN to be partitioned into several seemingly separate LANs.

Each virtual LAN is assigned an identifier (sometimes called a *color),*and packets can only travel from one segment to another if both segments have the same identifier.

Two virtual LANs share a common backbone



shows four hosts on four different LAN segments. In the absence of VLANs, any broadcast packet from any host will reach all the other hosts.

Internetworking

The term —internetwork or sometimes just internet refers to an arbitrary collection of networks interconnected to provide some sort of host to- host packet delivery service.

internetwork is often referred to as a "net work of networks" because it is made up of lots of smaller networks.al network built out of a collection of physical networks.

The nodes that interconnect the networks are called routers. They are also sometimes called gateways.

The Internet Protocol is the key tool used today to build scalable, heterogeneous internetworks. It was originally known as the Kahn-Cerf protocol after its inventors.

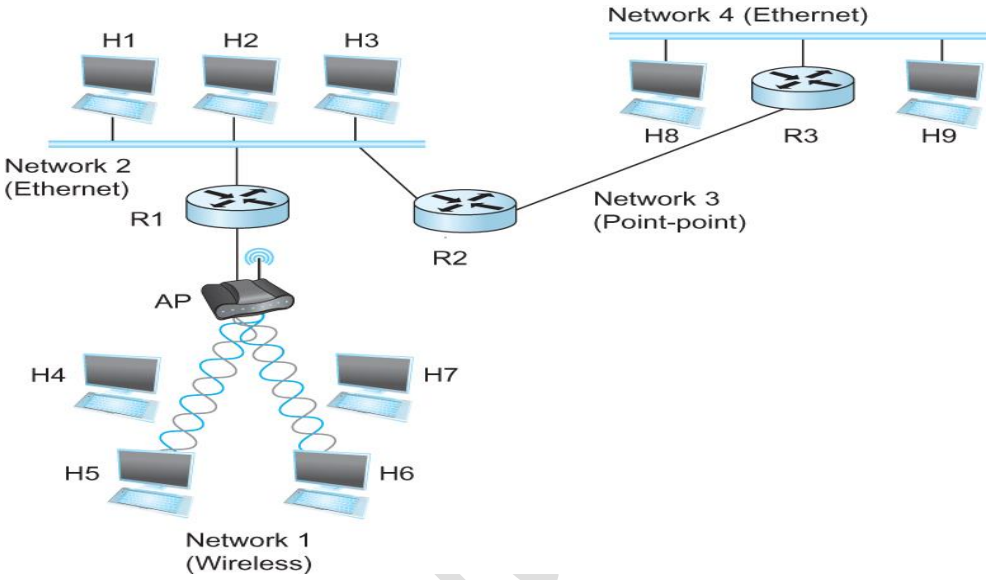


Fig: A simple internetwork. H*n* = host; R*n* = router.

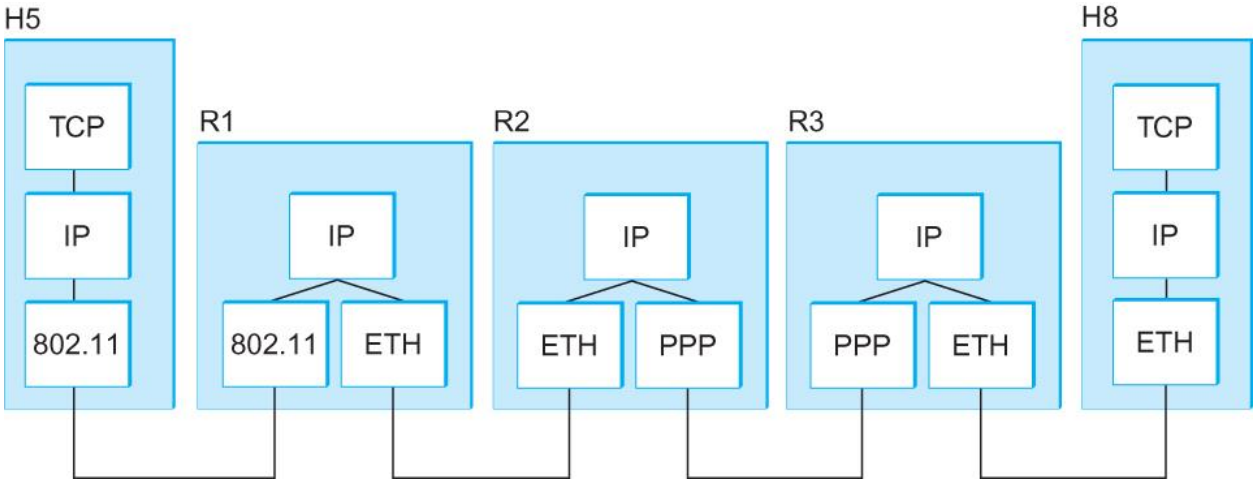


Fig : A simple internetwork, showing the protocol layers used to connect H5 to H8

IP (Internet Protocol)

- IP stands for Internet Protocol
- Key tool used today to build scalable, heterogeneous internetworks
- It runs on all the nodes in a collection of networks and defines the infrastructure that allows these nodes and networks to function as a single logical internetwork

Service Model

Service model is, the host-to-host services you want to provide.

- It has two parts
  1. Datagram Delivery Model
     - Connectionless model for data delivery
  2. Global Addressing Scheme
     - Provides a way to identify all hosts in the network

Datagram Delivery

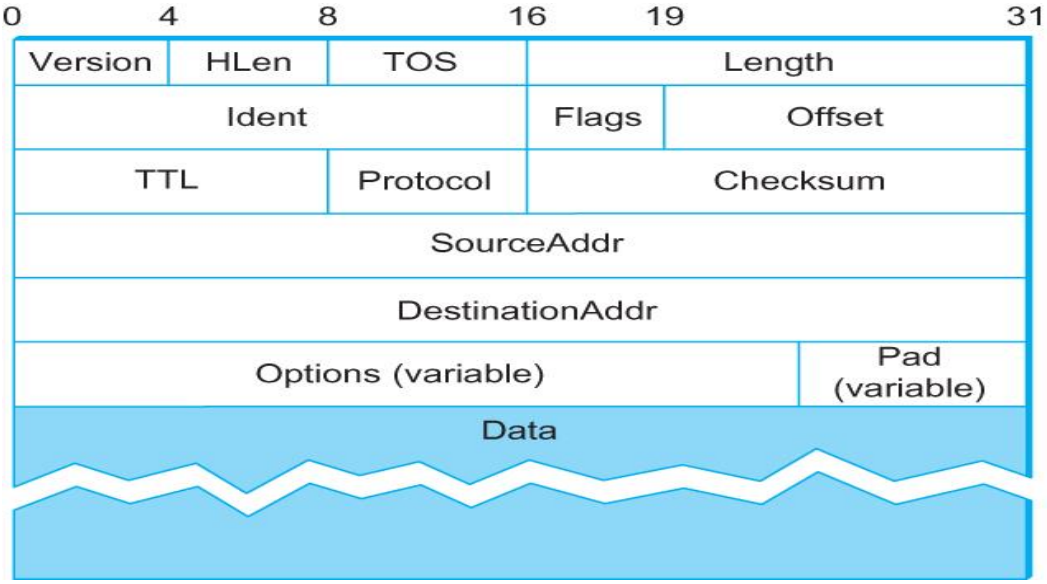
3. Datagram is a type of packet that happens to be sent in a connectionless manner over a network. Every datagram carries enough information to let the network forward the packet to its correct destination; there is no need for any advance setup mechanism to tell the network what to do when the packet arrives.

- Best-effort delivery (unreliable service)
  - » packets are lost or
  - » packets are delivered out of order or
  - » duplicate copies of a packet are delivered or
  - » packets can be delayed for a long time

Advantages of Datagram Delivery Model

- ✓ Best-effort, connectionless service
- ✓ simplest service model
- ✓ asking for a reliable packet delivery service may need to include a lot of extra functionality into the router.
- ✓ It enables IP to "run over anything" ie today IP can run over many network technologies
- ✓ Higher level protocols such as TCP that run over IP is aware of all failure modes.

IPv4 Packet Format



- Version (4): currently 4
- Hlen (4): number of 32-bit words in header
- TOS (8): type of service (not widely used)
- Length (16): number of bytes in this datagram. Max size of IP datagram is 65535 Bytes which is not supported for physical networks. So IP supports fragmentation and reassembly.
- Ident (16): used by fragmentation
- Flags/Offset (16): used by fragmentation
- TTL (8): specifies no of sec that a packet would be allowed to live
- Protocol (8): demux key (TCP=6, UDP=17)
- Checksum (16): for error detection
- DestAddr & SrcAddr (32)

IP Fragmentation and Reassembly

- Each network has some MTU (Maximum Transmission Unit)
  - Ethernet (1500 bytes), FDDI (4500 bytes)
- Strategy
  - Fragmentation occurs in a router when it receives a datagram that it wants to forward over a network which has (MTU < datagram)
  - Reassembly is done at the receiving host
  - All the fragments carry the same identifier in the *Ident* field
  - Fragments are self-contained datagrams
  - IP does not recover from missing fragments

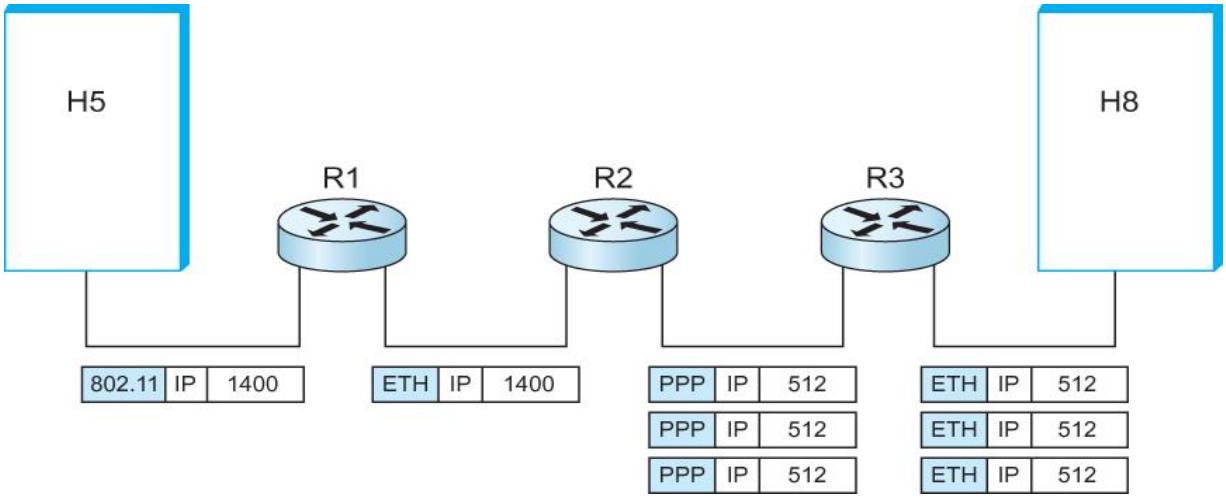


Fig: IP datagrams traversing the sequence of physical networks

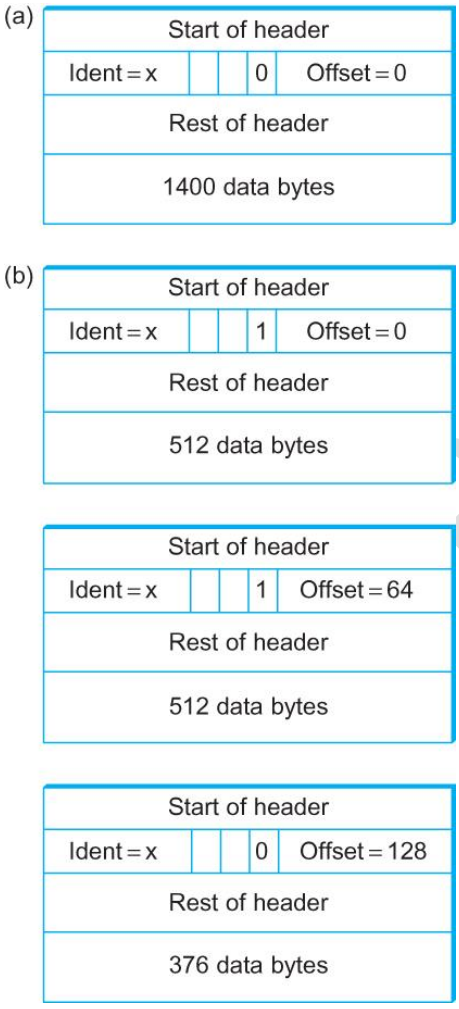


Fig: Header fields used in IP fragmentation. (a) Unfragmented packet; (b) fragmented packets

Global Addresses

- Global uniqueness is the first property that should be provided in an addressing scheme.

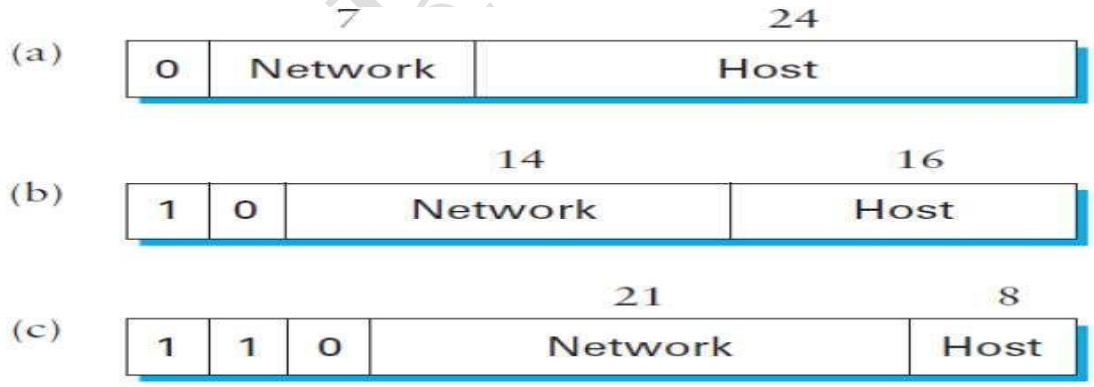
- Ethernet addresses are globally unique, but that alone does not suffice for an addressing scheme in a large internetwork. Ethernet addresses are also *flat*, which means that they have no structure and provide very few clues to routing protocols.
- In contrast, IP addresses are *hierarchical*, by which we mean that they are made up of several parts that correspond to some sort of hierarchy in the internetwork.

- **Specifically, IP addresses consist of two parts, a network part and a host part.** This is a fairly logical structure for an internetwork, which is made up of many interconnected networks.

- The network part of an IP address identifies the network to which the host is attached; all hosts attached to the same network have the same network part in their IP address.

- The host part then identifies each host uniquely on that particular network.
    - IP addresses are divided into three different classes.

- (There are also class D addresses that specify a multicast group, and class E addresses that are currently unused.)



IP addresses : (a)class A;(b)class B;class C.

The class of an IP address is identified in the most significant few bits. If the first bit is 0, it is a class A address. If the first bit is 1 and the second is 0, it is a class B address.

If the first two bits are 1 and the third is 0, it is a class C address. Thus, of the approximately 4 billion possible IP addresses, half are class A, one quarter are class B, and one-eighth are class C.

Each class allocates a certain number of bits for the network part of the address and the rest for the host part.

Class A networks have 7 bits for the network part and 24 bits for the host part, meaning that there can be only 126 class A networks (the values 0 and 127 are reserved), but each of them can accommodate up to 224 − 2 (about 1 6 million) hosts (again, there are two reserved values).

Class B addresses allocate 14 bits for the network and 16 bits for the host, meaning that each class B network has room for 65,534 hosts. Finally, class C addresses have only 8 bits for the host and 21 for the network part.

Therefore, a class C network can have only 256 unique host identifiers, which means only 254 attached hosts (one host identifier, 255, is reserved for broadcast, and 0 is not a valid host number). However, the addressing scheme supports 221 class C networks.

By convention, IP addresses are written as four *decimal* integers separated by dots. Each integer represents the decimal value contained in 1 byte of the address, starting at the most significant. For example, the address of the computer on which this sentence was typed is 171.69.210.245.

Datagram Forwarding in IP

- Strategy
  - every datagram contains destination's address
  - if directly connected to destination network, then forward to host
  - if not directly connected to destination network, then forward to some router
  - forwarding table maps network number into next hop
  - each host has a default router
  - each router maintains a forwarding table

Algorithm

```
if (NetworkNum of destination = NetworkNum of one of my interfaces) then
        deliver packet to destination over that interface
else
        if (NetworkNum of destination is in my forwarding table) then
                deliver packet to NextHop router
        else
                deliver packet to default router
```

For a host with only one interface and only a default router in its forwarding table, this simplifies to

```
if (NetworkNum of destination = my NetworkNum)then
        deliver packet to destination directly
else
        deliver packet to default router
```

SUBNETTING

The original intent of IP addresses was that the network part would uniquely identify exactly one physical network.

It has two drawbacks.

1. Address assignment inefficiency.
2. Assigning network number to every physical network uses the IP address space potentially much faster.

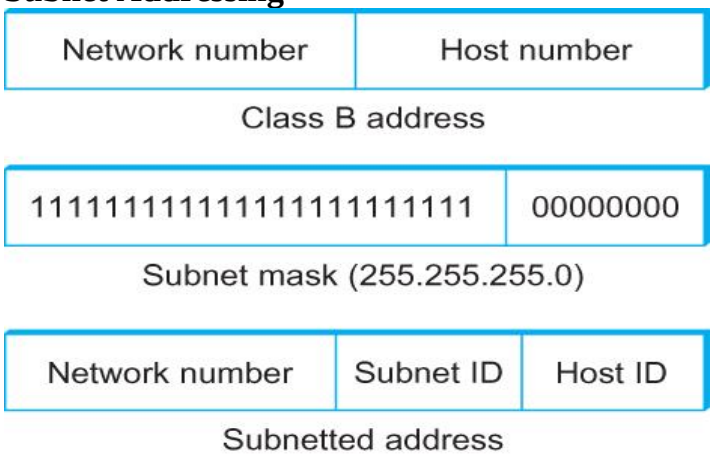
Assigning many network numbers has another drawback that becomes apparent when you think about routing.

***Subnetting provides an elegantly simple way to reduce the total number of network* numbers that are assigned.**

The idea is to take a single IP network number and allocate the IP addresses with that network number to several physical networks, which are now referred to as *subnets.*

- First, the subnets should be close to each other.
- This is because at a distant point in the Internet, they will all look like a single network, having only one network number between them.
- This means that a router will only be able to select one route to reach any of the subnets, so they had better all be in the same general direction.
- A perfect situation in which to use subnetting is a large campus or corporation that has many physical networks.
- From outside the campus, all you need to know to reach any subnet inside the campus is where the campus connects to the rest of the Internet.
- The mechanism by which a single network number can be shared among multiple networks involves configuring all the nodes on each subnet with a *subnet mask.*
- With simple IP addresses, all hosts on the same network must have the same network number.
- The subnet mask enables us to introduce a *subnet number; all hosts on the same* physical network will have the same subnet number, which means that hosts may be on different physical networks but share a single network number.
- For example, suppose that we want to share a single class B address among several physical networks. We could use a subnet mask of 255.255.255.0.
- (Subnet masks are written down just like IP addresses; this mask is therefore all 1s in the upper 24 bits and 0s in the lower 8 bits.)
- In effect, this means that the top 24 bits(where the mask has 1s) are now defined to be the network number, and the lower 8 bits (where the mask has 0s) are the host number.
- Since the top 16 bits identify the network in a class B address, we may now think of the address as having not two parts but three: a network part, a subnet part, and a host part.

Subnet Addressing

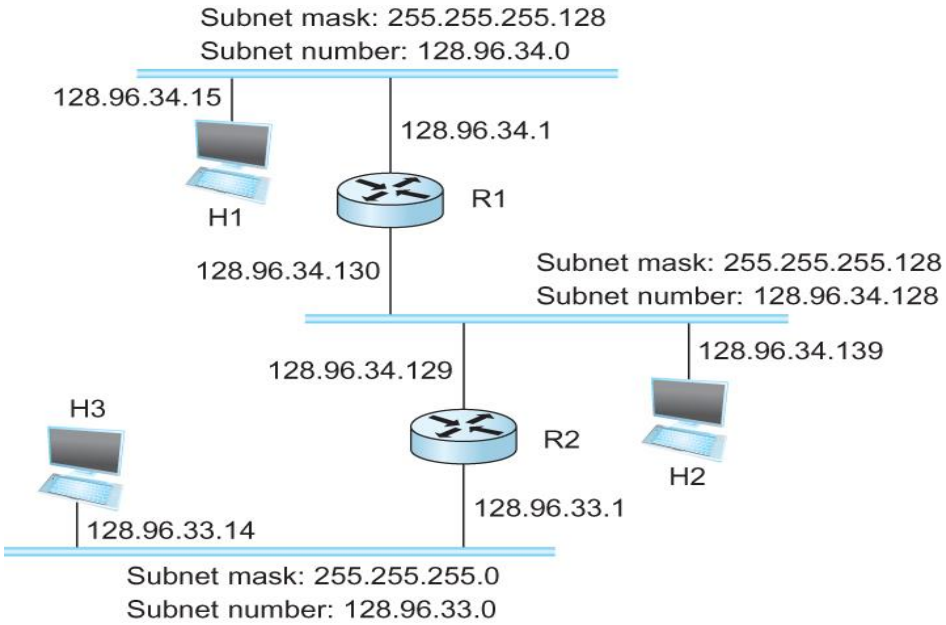


- What subnetting means to a host is that it is now configured with both an IP address and a subnet mask for the subnet to which it is attached.
- For example, host H1 in Figure 4.26 is configured with an address of 128.96.34.15 and a subnet mask of 255.255.255.128.
- The bitwise AND of these two numbers defines the subnet number of the host and of all other hosts on the same subnet.
- In this case, 128.96.34.15 AND 255.255.255.128 equals 128.96.34.0, so this is the subnet number for the topmost subnet.

Forwarding Packet through Subnet

When the host wants to send a packet to a certain IP address, the first thing it does is to perform a bitwise AND between its own subnet mask and the destination IP address. If the results are not equal, the packet needs to be sent to a router to be forwarded to another subnet.

For example, if H1 is sending to H2, then H1 ANDs its subnet mask (255.255.255.128) with the address for H2 (128.96.34.139) to obtain 128.96.34.128. This does not match the subnet number for H1 (128.96.34.0) so H1 knows that H2 is on a different subnet. Since H1 cannot deliver the packet toH2 directly over the subnet, it sends the packet to its default router R1.



To support subnetting, the routing table must now hold entries of the form (SubnetNumber, SubnetMask, NextHop). To find the right entry in the table,the router ANDs the packet's destination address with the SubnetMask for each entry in turn; if the result matches the SubnetNumber of the entry, then this is the right entry to use, and it forwards the packet to the next hop router indicated.

Forwarding Algorithm

```
D = destination IP address
for each entry < SubnetNum, SubnetMask, NextHop>
D1 = SubnetMask & D
if D1 = SubnetNum
if NextHop is an interface
  deliver datagram directly to destination
else
  deliver datagram to NextHop (a router)
```

Example Subnetting Table

SubnetNumber	SubnetMask	NextHop
128.96.34.0	255.255.255.128	Interface 0
128.96.34.128	255.255.255.128	Interface 1
128.96.33.0	255.255.255.0	R2

Classless Routing (CIDR)

Classless Inter Domain Routing (CIDR, pronounced ―cider) is a technique that addresses two scaling concerns in the Internet.

- Do not use classes to determine network ID
- Assign any range of addresses to network
  - Use common part of address as network number
  - E.g., addresses 192.4.16 - 192.4.31 have the first 20 bits in common. Thus, we use these 20 bits as the network number
  - netmask is /20, /xx is valid for almost any xx
- Enables more efficient usage of address space (and router tables)

The growth of backbone routing tables as more and more network numbers need to be stored in them, and the potential for the 32-bit IP address space to be exhausted well before the four-billionth host is attached to the Internet.

This address space exhaustion is called address assignment inefficiency.

The inefficiency arises because the IP address structure, with class A, B, and C addresses, forces us to hand out network address space in fixed-sized chunks of three very different sizes.

- A network with two hosts needs a class C address
  - Address assignment efficiency = 2/255 = 0.78
- A network with 256 hosts needs a class B address
  - Address assignment efficiency = 256/65535 = 0.39

Even though subnetting can help us to assign addresses carefully, it does not get around the fact that any autonomous system with more than 255 hosts, or an expectation of eventually having that many, wants a class B address.
As it turns out, exhaustion of the IP address space centers on exhaustion of the class B network numbers.

One way to deal with that would seem to be saying no to any AS (Autonomous Systems) that requests a class B address unless they can show a need for something close to 64K addresses, and instead giving them an appropriate number of class C addresses to cover the expected number of hosts. Since we would now be handing out address space in chunks of 256 addresses at a time, we could more accurately match the amount of address space consumed to the size of the AS.

For any AS with at least 256 hosts (which means the majority of ASs), we can guarantee an address utilization of at least 50%, and typically much more.

This solution, however, raises a problem that is at least as serious: excessive storage requirements at the routers.

- If a single AS has, say 16 class C network numbers assigned to it,
  - Every Internet backbone router needs 16 entries in its routing tables for that AS
  - This is true, even if the path to every one of these networks is the same
- If we had assigned a class B address to the AS
  - The same routing information can be stored in one entry
  - Efficiency = 16 × 255 / 65, 536 = 6.2%

CIDR, therefore, tries to balance the desire to minimize the number of routes that a router needs to know against the need to hand out addresses efficiently.

To do this, CIDR helps us to *aggregate* routes.

- Uses a single entry in the forwarding table to tell the router how to reach a lot of different networks
- Breaks the rigid boundaries between address classes

To understand how this works, consider our hypothetical AS with 16 class C network numbers. Instead of handing out 16 addresses at random, we can hand out a block of *contiguous* class C addresses.

- Suppose we assign the class C network numbers from 192.4.16 through 192.4.31
- Observe that top 20 bits of all the addresses in this range are the same (11000000 00000100 0001)
  - We have created a 20-bit network number (which is in between class

B network number and class C number in terms of the number of hosts that it can support)

In other words, we get both the high address efficiency of handing out addresses in chunks smaller than a class B network and a single network prefix that can be used in forwarding tables.

Observe that for this scheme to work, we need to hand out blocks of class C addresses that share a common prefix, which means that each block must contain a number of class C networks that is a power of two.

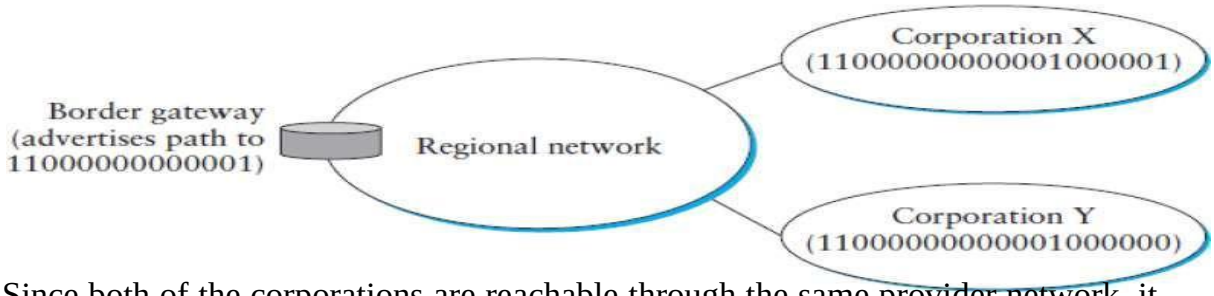
All we need now to make CIDR solve our problems is a routing protocol that can deal with these ―classless addresses, which means that it must understand that a network number may be of any length.

Modern routing protocols do exactly that. The network numbers that are carried in such a routing protocol are represented simply by _length, value_ pairs, where the length gives the number of bits in the network prefix—20 in the above example.

- Requires to hand out blocks of class C addresses that share a common prefix
- The convention is to place a /X after the prefix where X is the prefix length in bits
- For example, the 20-bit prefix for all the networks 192.4.16 through 192.4.31 is represented as 192.4.16/20
- By contrast, if we wanted to represent a single class C network number, which is 24 bits long, we would write it 192.4.16/24
- How do the routing protocols handle this classless addresses
  - It must understand that the network number may be of any length
- Represent network number with a single pair
  <length, value>
- All routers must understand CIDR addressing

Consider the example Figure The two corporations served by the provider network have been assigned adjacent 20- bit network prefixes.

Route Aggregate with CIDR



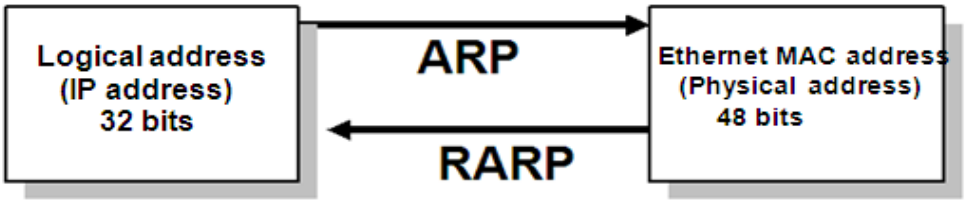
Since both of the corporations are reachable through the same provider network, it can advertise a single route to both of them by just advertising the common 19-bit prefix they share.

In general, it is possible to aggregate routes repeatedly if addresses are assigned carefully.

This means that we need to pay attention to which provider a corporation is attached to before assigning it an address if this scheme is to work. One way to accomplish that is to assign a portion of address space to the provider and then to let the network provider assign addresses from that space to its customers.

ARP Address Resolution Protocol

- The Address Resolution Protocol (ARP) is used to map logical address (IP address) into physical address.
- On a typical physical network, such as a LAN, each device on a link is identified by a physical or station address, usually imprinted on the network interface card (NIC).
- ARP is used to find the physical address of the node when its Internet address is known.



ARP operation

Two operations are

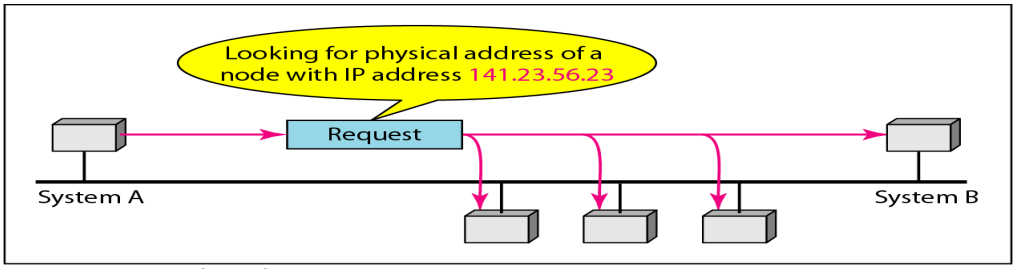
1. ARP request
2. ARP reply

ARP request

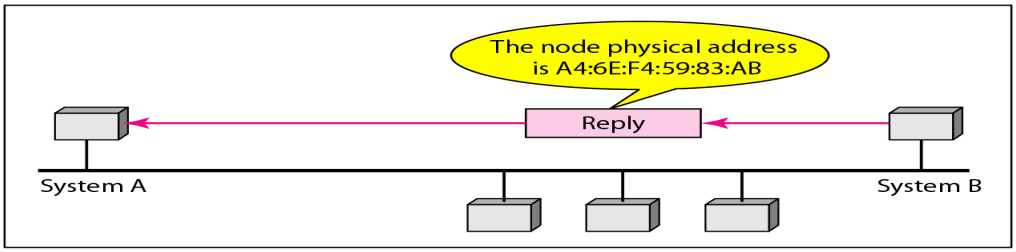
To determine the physical address for a particular IP, the node broadcasts an ARP query or ARP request containing that IP address in the network.

ARP reply

The node with that IP address responds to the sender of the ARP request with an ARP response or ARP reply containing its link-layer address or physical address,



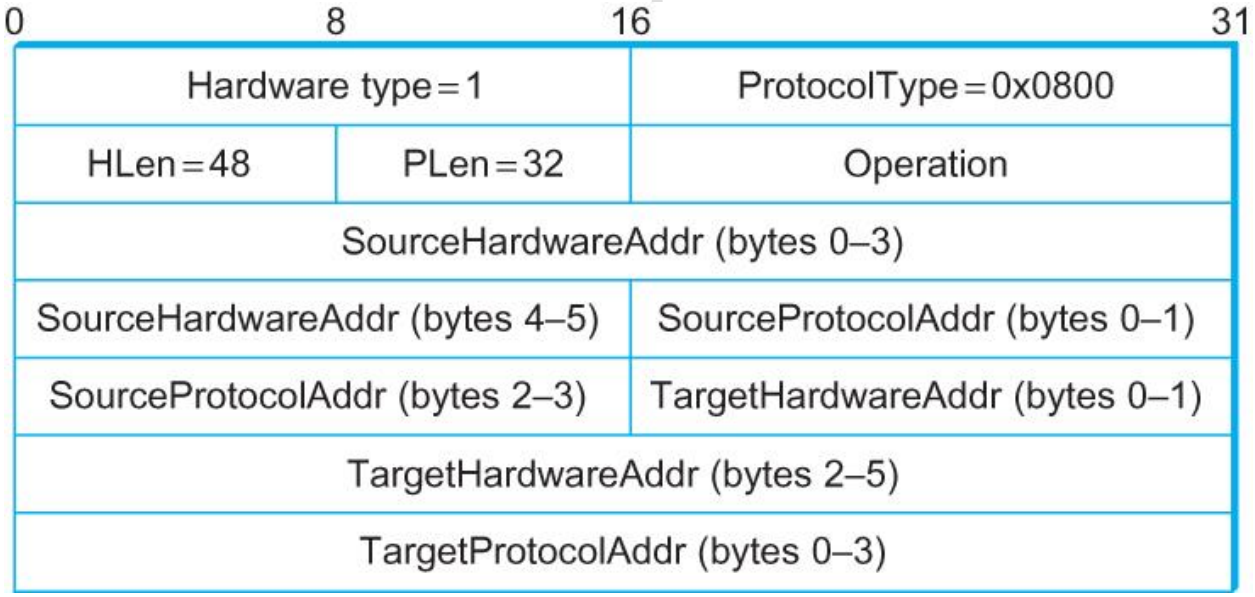
a. ARP request is broadcast



b. ARP reply is unicast

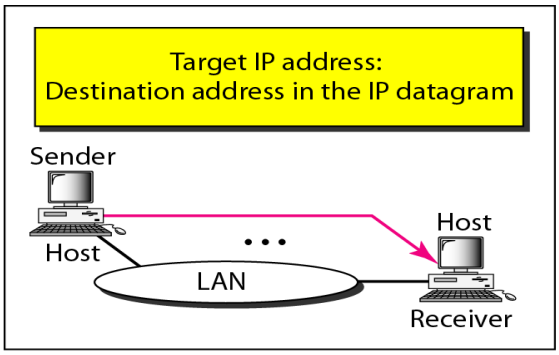
For providing mapping between logical address and physical address, each nodes maintain an ARP cache or ARP table. This table maintains an address pairs that map logical address to physical address.

ARP Packet Format

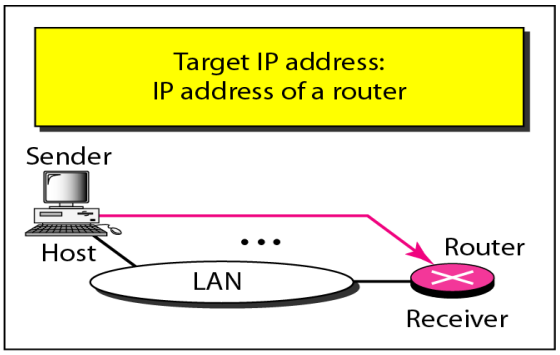


- HardwareType: type of physical network (e.g., Ethernet)
- ProtocolType: type of higher layer protocol (e.g., IP)
- HLEN : length of physical addresses in bytes eg : ethernet – 6 bytes
- PLEN: length of logical addresses in bytes. (eg: IPv4 – 4 bytes)
- Operation: request or response
- Source/Target Physical/Protocol addresses

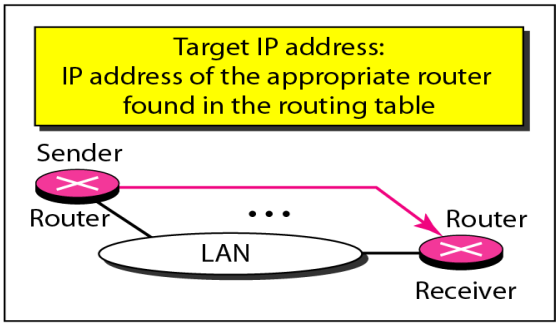
Four cases using ARP



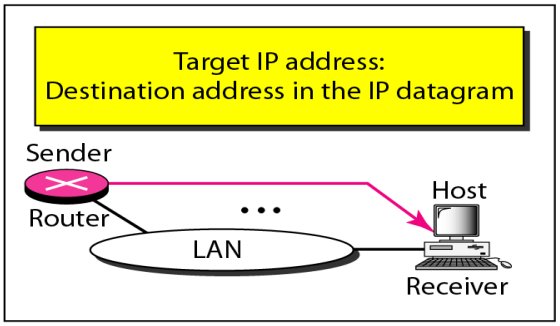
Case 1. A host has a packet to send to another host on the same network.



Case 2. A host wants to send a packet to another host on another network. It must first be delivered to a router.



Case 3. A router receives a packet to be sent to a host on another network. It must first be delivered to the appropriate router.



Case 4. A router receives a packet to be sent to a host on the same network.

RARP

Reverse Address Resolution Protocol (RARP)
Finds the logical address for a machine that knows only its physical address.

DHCP - Dynamic Host Configuration Protocol

- For Mapping Physical to Logical Address can use RARP/BOOTP/DHCP

- DHCP server is responsible for providing configuration information to hosts
- DHCP is a protocol that dynamically assigns IP addresses to host.

Need for dynamic host configuration

1. Ethernet addresses are configured into network by manufacturer and they are unique
2. IP addresses must be unique on a given internetwork but also must reflect the structure of the internetwork
3. Most host Operating Systems provide a way to manually configure the IP information for the host
4. Drawbacks of manual configuration
    i. A lot of work to configure all the hosts in a large network
    ii. Configuration process is error-prune
5. For this reason, automated Configuration Process is required
6. DHCP server is responsible for providing configuration information to hosts. There is at least one DHCP server for an administrative domain
7. DHCP server maintains a pool of available addresses.
8. DHCP is a protocol that dynamically assigns IP addresses to host.
9. DHCP relies on the existence of a DHCP server which is responsible for providing configuration information to host.
10. The use of DHCP avoids the network administrator from assigning addresses to individual host.

The Purpose of DHCP

- Provide dynamic allocation of IP client configuration for a lease period
- Eliminate the work necessary to administer a large IP Network

The first problem faced in DHCP is server discovery. To avoid this, a host undergo the following activities,

1. Newly booted or attached host sends DHCPDISCOVER message to a special IP address (255.255.255.255) which is an IP broadcast address that is received by all hosts and routers on that network.
2. DHCP uses the concept of relay agent, there is at least one relay agent on each network.
3. When a relay agent receives a DHCPDISCOVER message, it unicasts it to DHCP server and awaits for the response, which is then sent back to the requesting client.

The process of relaying a message from a host to remote DHCP server is as follows,

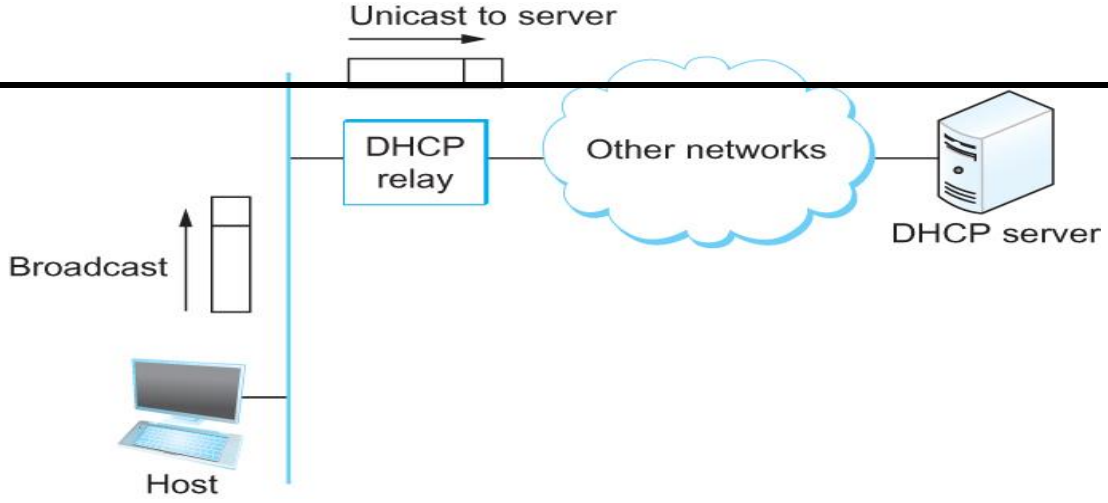
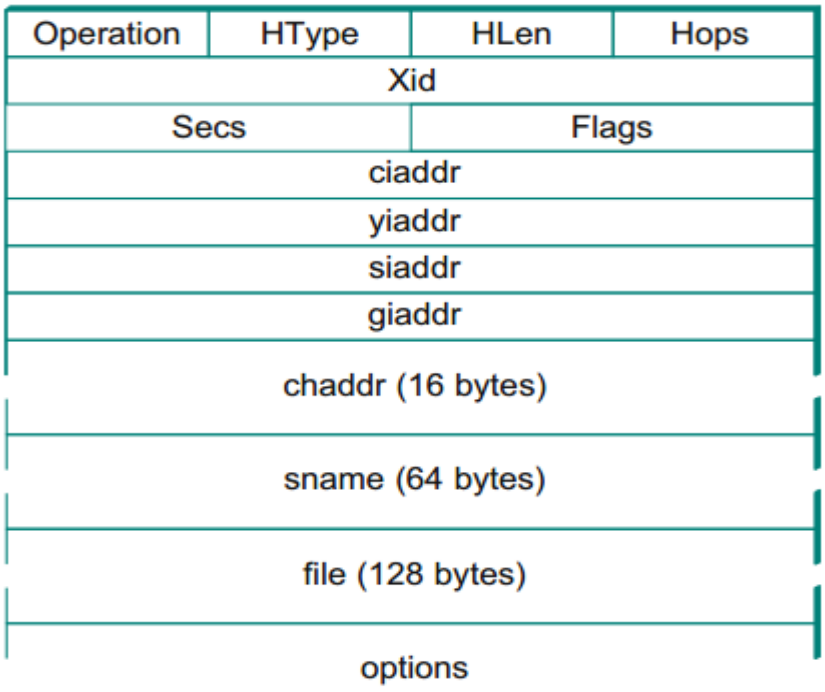


FIGURE : A DHCP relay agent receives a broadcast DHCPDISCOVERmessage from a host and sends a unicast DHCPDISCOVER to the DHCP server.

DHCP packet format

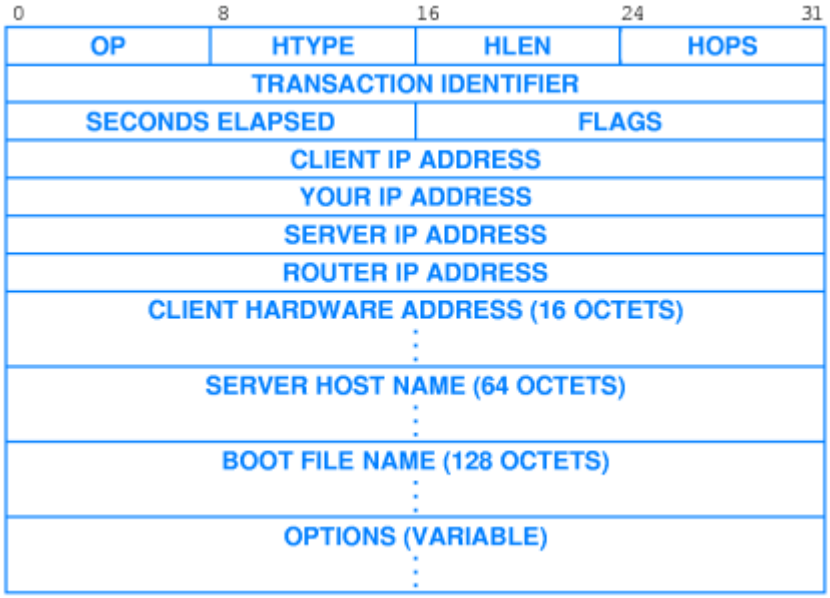
Figure shows the format of a DHCP message. The message is actually sent using a protocol called the User Datagram Protocol (UDP) that runs over IP. UDP does in this context is to provide a demultiplexing key that says, "This is a DHCP packet."

DHCP is derived from an earlier protocol called BOOTP, and some of the packet fields are thus not strictly relevant to host configuration.



- Operation : *1 (Request), 2(Reply)*
- Hardware Type: *1 (for Ethernet)*
- Hardware address length: *6 (for Ethernet)*
- Hop count: *set to 0 by client*
- Transaction ID: *Integer (used to match reply to response)*
- Seconds: *number of seconds since the client started to boot*
- ciaddr : client IP address
- yiaddr : your IP address
- siaddr : server IP address
- giaddr : gateway IP address

- chaddr : client's hardware address(Ethernet address)
- sname : server host name
- file : boot file name
- *client fills in the information that it has, leaves rest blank*



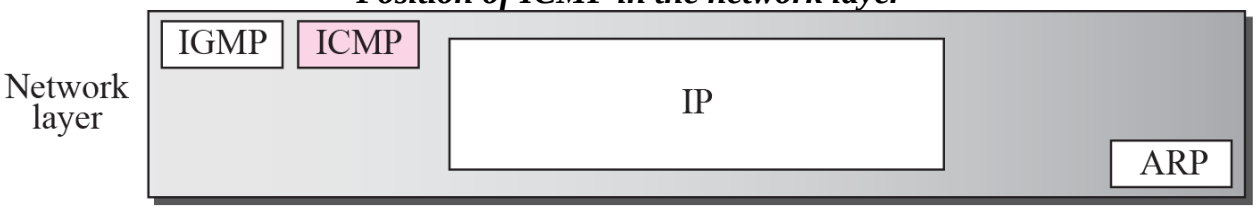
ICMP (Internet Control Message Protocol)

ICMP is used for error and control information

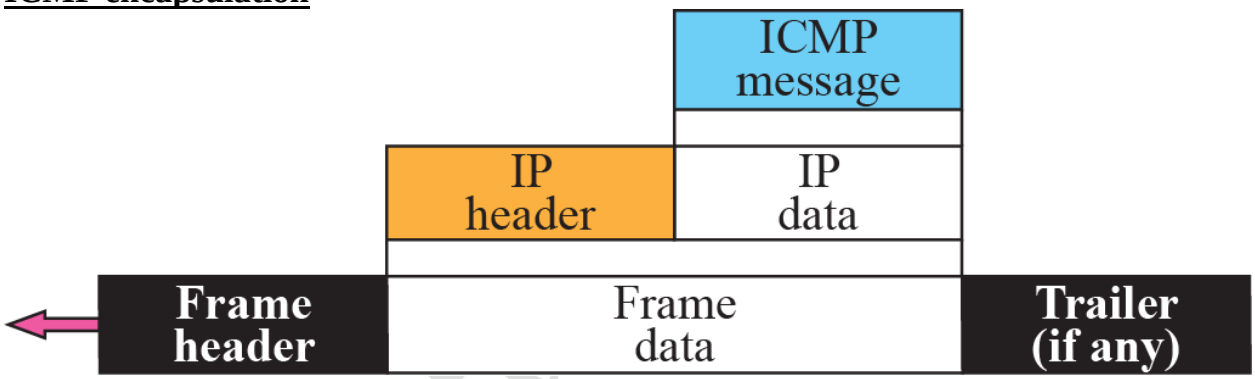
- Defines a collection of error messages that are sent back to the source host whenever a router or host is unable to process an IP datagram successfully

- ***ICMP always reports error messages to the original source.***

Position of ICMP in the network layer



ICMP encapsulation



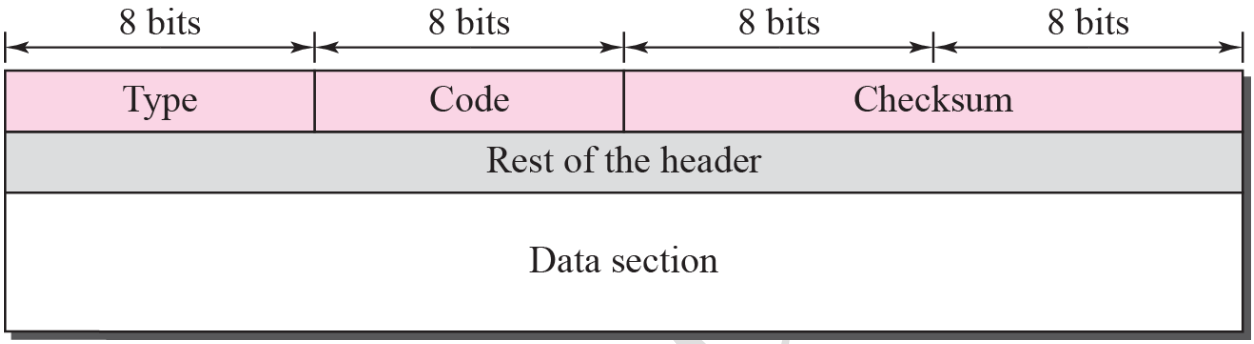
ICMP messages are divided into two broad categories:

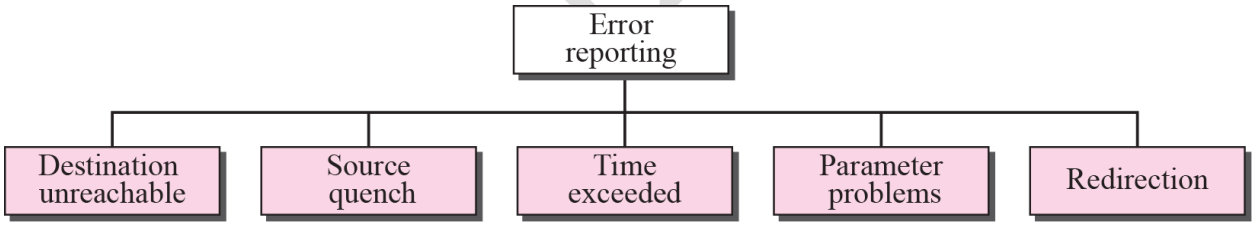
1. error-reporting messages
   - The error-reporting messages report problems that a router or a host (destination) may encounter when it processes an IP packet.
2. query messages.
   - The query messages, which occur in pairs, help a host or a network manager get specific information from a router or another host.

ICMP messages

Category	*Type*	*Message*
Error-reporting messages	3	Destination unreachable
	4	Source quench
	11	Time exceeded
	12	Parameter problem
	5	Redirection
Query messages	8 or 0	Echo request or reply
	13 or 14	Timestamp request or reply

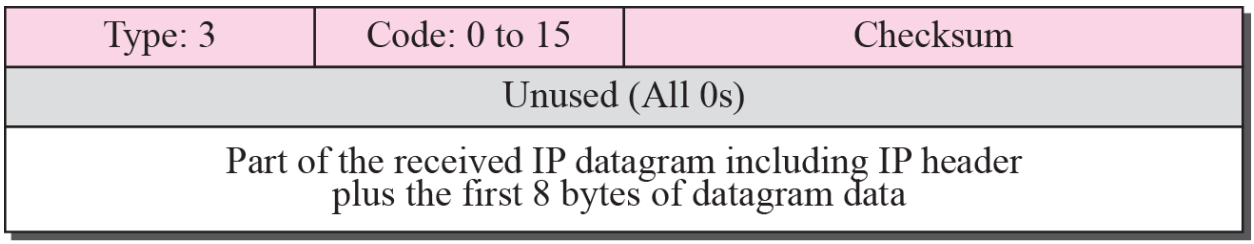
General format of ICMP messages





Destination-unreachable

When a router cannot route a datagram or a host cannot deliver a datagram, the datagram is discarded and the router or the host sends a destination-unreachable message back to the source host that initiated the datagram. Note that destination-unreachable messages can be created by either a router or the destination host



Source Quench

A source-quench message informs the source that a datagram has been discarded due to congestion in a router or the destination host. The source must slow

down the sending of datagram's until the congestion is relieved.

Type: 4	Code: 0	Checksum
Unused (All 0s)		
Part of the received IP datagram including IP header plus the first 8 bytes of datagram data		

Time Exceeded

When the final destination does not receive all of the fragments in a set time, it discards the received fragments and sends a time-exceeded message to the original source.

Type: 11	Code: 0 or 1	Checksum
Unused (All 0s)		
Part of the received IP datagram including IP header plus the first 8 bytes of datagram data		

Parameter-problem

Any ambiguity in the header part of a datagram can Create serious problems as the datagram travels through the Internet. If a router or the destination host discovers an ambiguous or missing value in any field of the datagram, it discards the datagram and sends a parameter-problem message back to the source.

Type: 12	Code: 0 or 1	Checksum
Pointer	Unused (All 0s)	
Part of the received IP datagram including IP header plus the first 8 bytes of datagram data		

Redirection

ICMP Redirect is sent by a router (R1) to the sender of an IP datagram (host) when the datagram should have been sent to a different router (R2)

Type: 5	Code: 0 to 3	Checksum
IP address of the target router		
Part of the received IP datagram including IP header plus the first 8 bytes of datagram data		

Echo-request and echo-reply

- Echo-request and echo-reply messages can be used by network managers to check the operation of the IP protocol.
- Echo-request and echo-reply messages can test the reachability of a host. This

is usually done by invoking the ping command.

Type 8: Echo request
Type 0: Echo reply

Type: 8 or 0	Code: 0	Checksum
Identifier		Sequence number
Optional data Sent by the request message; repeated by the reply message		

Timestamp-request and timestamp-reply message

Timestamp-request and timestamp-reply messages can be used to calculate the round-trip time between a source and a destination machine even if their clocks are not synchronized.

Type 13: request
Type 14: reply

Type: 13 or 14	Code: 0	Checksum
Identifier		Sequence number
Original timestamp		
Receive timestamp		
Transmit timestamp		

Virtual Networks and Tunnels

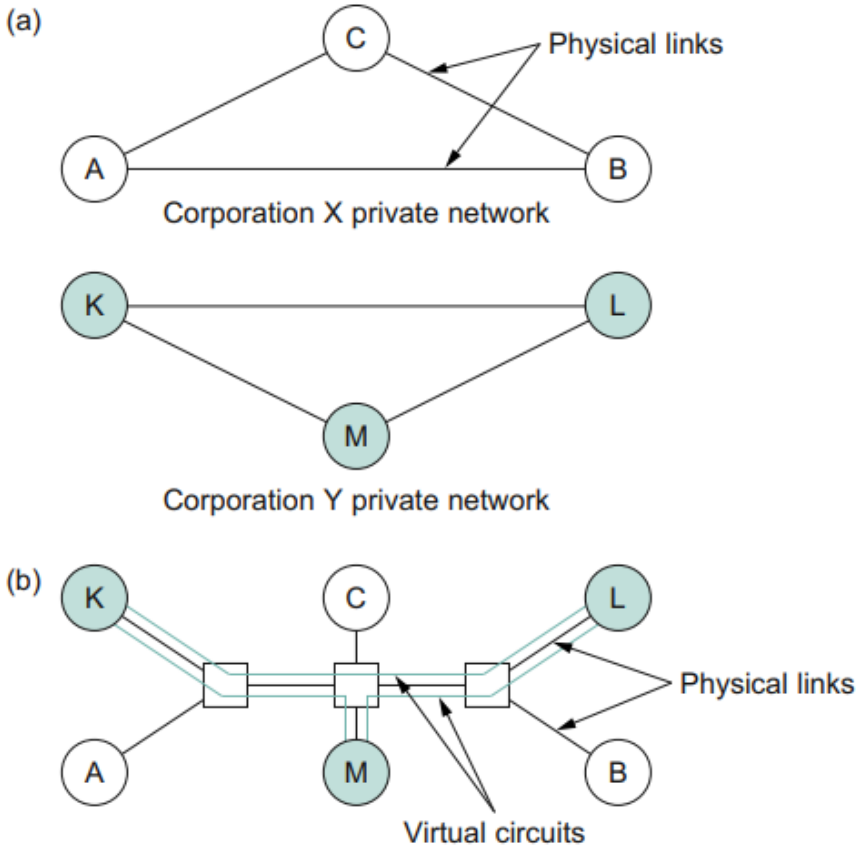
There are many situations where more controlled connectivity is required. An important example of such a situation is the virtual private network (VPN).

The term VPN is heavily overused and definitions vary, but intuitively we can define a VPN by considering first the idea of a private network. Corporations with many sites often build private networks by leasing transmission lines from the phone companies and using those lines to interconnect sites.

In such a network, communication is restricted to take place only among the sites of that corporation, which is often desirable for security reasons. To make a private network virtual, the leased transmission lines—which are not shared with any other corporations—would be replaced by some sort of shared network. A virtual circuit (VC) is a very reasonable replacement for a leased line because it still provides a logical point-to-point connection between the corporation's sites.

For example, if corporation X has a VC from site A to site B, then clearly it can send packets between sites A and B. But there is no way that corporation Y can get its packets delivered to site B without first establishing its own virtual circuit to site B, and the establishment of such a VC can be administratively prevented, thus preventing unwanted connectivity between corporation X and corporation Y.

Figure 3.26(a) shows two private networks for two separate corporations. In Figure 3.26(b) they are both migrated to a virtual circuit network.



■ FIGURE 3.26 An example of virtual private networks: (a) two separate private networks; (b) two virtual private networks sharing common switches.

The limited connectivity of a real private network is maintained, but since the private networks now share the same transmission facilities and switches we say that two virtual private networks have been created. In Figure 3.26, a virtual circuit network (using Frame Relay or ATM, for example) is used to provide the controlled connectivity among sites. It is also possible to provide a similar function using an IP network—an internetwork—to provide the connectivity.

However, we cannot just connect the various corporations' sites to a single internetwork because that would provide connectivity between corporation X and

corporation Y, which we wish to avoid. To solve this problem, we need to introduce a new concept, the IP tunnel.

We can think of an IP tunnel as a virtual point-to-point link between a pair of nodes that are actually separated by an arbitrary number of networks. The virtual link is created within the router at the entrance to the tunnel by providing it with the IP address of the router at the far end of the tunnel.

Whenever the router at the entrance of the tunnel wants to send a packet over this virtual link, it encapsulates the packet inside an IP datagram. The destination address in the IP header is the address of the router at the far end of the tunnel, while the source address is that of the encapsulating router.

In the forwarding table of the router at the entrance to the tunnel, this virtual link looks much like a normal link. Consider, for example, the network in Figure 3.27. A tunnel has been configured from R1 to R2 and assigned a virtual interface number of 0. The forwarding table in R1 might therefore look like Table 3.8

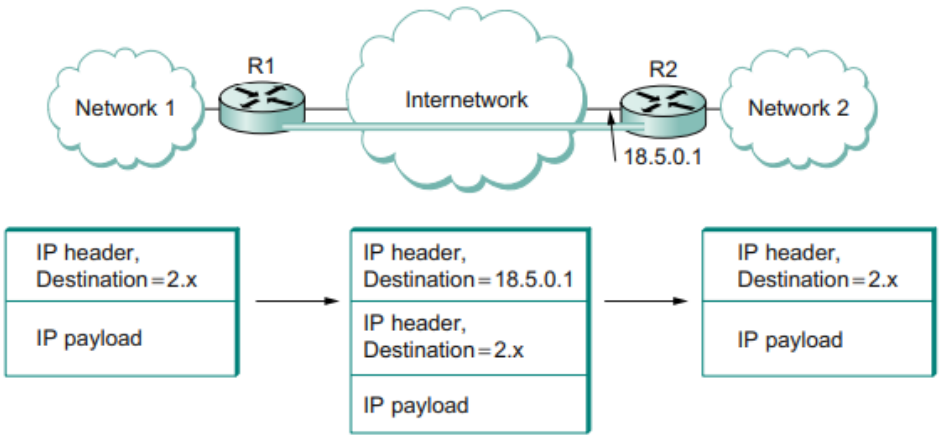


FIGURE 3.27 A tunnel through an internetwork. 18.5.0.1 is the address of R2 that can be reached from R1 across the internetwork.

Table 3.8 Forwarding Table for Router R1 in Figure 3.27

NetworkNum	NextHop
1	Interface 0
2	Virtual interface 0
Default	Interface 1

R1 has two physical interfaces. Interface 0 connects to network 1; interface 1 connects to a large internetwork and is thus the default for all traffic that does not match something more specific in the forwarding table. In addition, R1 has a virtual interface, which is the interface to the tunnel. Suppose R1 receives a packet from network 1 that contains an address in network 2. The forwarding table says this packet should be sent out virtual interface 0. In order to send a packet out this interface, the router takes the packet, adds an IP header addressed to R2, and then proceeds to forward the packet as if it had just been received. R2's address is 18.5.0.1; since the network number of this address is 18, not 1 or 2, a packet destined for R2 will be forwarded out the default interface into the internetwork.

Once the packet leaves R1, it looks to the rest of the world like a normal IP packet destined to R2, and it is forwarded accordingly. All the routers in the internetwork forward it using normal means, until it arrives at R2. When R2 receives the packet, it finds that it carries its own address, so it removes the IP header and looks at the payload of the packet. What it finds is an inner IP packet whose destination address is in network 2. R2 now processes this packet like any other IP packet it receives. Since R2 is directly connected to network 2, it forwards the packet on to that network. Figure 3.27 shows the change in encapsulation of the packet as it moves across the network.

Tunneling does have its downsides.

One is that it increases the length of packets; this might represent a significant waste of bandwidth for short packets. Longer packets might be subject to fragmentation, which has its own set of drawbacks. There may also be performance implications for the routers at either end of the tunnel, since they need to do more work than normal forwarding as they add and remove the tunnel header

Finally, there is a management cost for the administrative entity that is responsible for setting up the tunnels and making sure they are correctly handled by the routing protocols.