DATA COMMUNICATION

COMPUTER NETWORKS

(Common to CSE & ISE)

Subject Code: 21CS62	**CIE Marks : 50**
Hours/Week : 04	**Exam Hours : 03**
Total Hours : 50	**SEE Marks: 80**

MODULE – 1 **08 Hours**

Introduction: Data Communications, Networks, Network Types, Internet History, Standards and Administration
Networks Models: Protocol Layering, TCP/IP Protocol suite, The OSI model
Introduction to Physical Layer-1: Data and Signals, Digital Signals, Transmission Impairment, Data Rate limits, Performance **Digital Transmission:** Digital to digital conversion (Only Line coding: Polar, Bipolar and Manchester coding)
Physical Layer-2: Analog to digital conversion (only PCM), Transmission Modes
Analog Transmission: Digital to analog conversion

MODULE – 2 **08 Hours**

Data link control: DLC services, Data link layer protocols, HDLC, and Point to Point protocol (Framing, Transition phases only). **Error Detection and correction:** Introduction, Block coding, Cyclic codes, Checksum, Forward error correction**, Media Access control:** Random Access, Controlled Access and Channelization

MODULE 1: TABLE OF CONTENTS

1.1 DATA COMMUNICATIONS
- 1.1.1 Components
  - 1.1.2 Data Representation
  - 1.1.3 Direction of Data Flow

1.2 NETWORKS
- 1.2.1 Network Criteria
- 1.2.2 Physical Structures
  - 1.2.2.1 Type of Connection
  - 1.2.2.2 Physical Topology
    - 1.2.2.2.1 Bus Topology
    - 1.2.2.2.2 Star Topology
    - 1.2.2.2.3 Ring Topology
    - 1.2.2.2.4 Mesh Topology

1.3 NETWORK TYPES
- 1.3.1 Local Area Network
- 1.3.2 Wide Area Network
  - 1.3.2.1 Internetwork
- 1.3.3 LAN vs. WAN
- 1.3.4 Switching
  - 1.3.4.1 Circuit-Switched Network
  - 1.3.4.2 Packet-Switched Network
- 1.3.5 The Internet

1.4 PROTOCOL LAYERING
- 1.4.1 Scenarios
  - 1.4.1.1 Protocol Layering
- 1.4.2 Principles of Protocol Layering
- 1.4.3 Logical Connections

1.5 TCP/IP PROTOCOL SUITE
- 1.5.1 Layered Architecture
- 1.5.2 Layers in the TCP/IP Protocol Suite
- 1.5.3 Description of Each Layer
- 1.5.4 Encapsulation and Decapsulation
- 1.5.5 Addressing 42
- 1.5.6 Multiplexing and Demultiplexing

1.6 THE OSI MODEL
- 1.6.1 OSI versus TCP/IP

1.7 DATA AND SIGNALS
- 1.7.1 Analog and Digital Data
- 1.7.2 Analog and Digital Signals
- 1.7.3 Periodic and Nonperiodic

1.8 DIGITAL SIGNALS
- 1.8.1 Bit Rate
- 1.8.2 Bit Length
- 1.8.3 Digital Signal as a Composite Analog Signal

1.8.4 Transmission of Digital Signals
1.8.4.1 Baseband Transmission
1.8.4.2 Broadband Transmission (Using Modulation)
1.9 TRANSMISSION IMPAIRMENT
1.9.1 Attenuation
1.9.1.1 Decibel
1.9.2 Distortion
1.9.3 Noise
1.9.3.1 Signal-to-Noise Ratio (SNR)
1.10 DATA RATE LIMITS
1.10.1 Noiseless Channel: Nyquist Bit Rate
1.10.2 Noisy Channel: Shannon Capacity
1.11 PERFORMANCE
1.11.1 Bandwidth
1.11.2 Throughput
1.11.3 Latency (Delay)
1.11.4 Bandwidth-Delay Product
1.11.5 Jitter
1.12 DIGITAL-TO-DIGITAL CONVERSION
1.12.1 Line Coding
1.12.1.1 Characteristics
1.12.2 Line Coding Schemes
1.12.2.1 Unipolar Scheme
1.12.2.2 Polar Schemes
1.12.2.3 Bipolar Schemes (or Multilevel Binary)

2.1 ANALOG-TO-DIGITAL CONVERSION
2.1.1 PCM
2.1.1.1 Sampling
2.1.1.1.1 Sampling Rate
2.1.2 Quantization
2.1.2.1 Quantization Levels
2.1.2.2 Quantization Error
2.1.2.3 Uniform vs. Non Uniform Quantization
2.1.3 Encoding
2.1.3.1 Original Signal Recovery
2.1.3.2 PCM Bandwidth
2.1.3.3 Maximum Data Rate of a Channel
2.1.3.4 Minimum Required Bandwidth
2.2 TRANSMISSION MODES
2.2.1 PARALLEL TRANSMISSION
2.2.2 SERIAL TRANSMISSION
2.2.2.1 Asynchronous Transmission
2.2.2.2 Synchronous Transmission
2.2.2.3 Isochronous
2.3 DIGITAL TO ANALOG CONVERSION
2.3.1 Aspects of Digital to Analog Conversion
2.3.2 Amplitude Shift Keying (ASK)
2.3.2.1 Binary ASK (BASK)
2.3.2.1.1 Implementation of BASK
2.3.2.1.2 Bandwidth for ASK
2.3.3 Frequency Shift Keying (FSK)
2.3.3.1 Binary FSK (BFSK)
2.3.3.1.1 Implementation of BFSK
2.3.3.1.2 Bandwidth for BFSK
2.3.4 Phase Shift Keying (PSK)
2.3.4.1 Binary PSK (BPSK)

2.3.4.1.1 Implementation of BPSK
2.3.4.1.2 Bandwidth for BPSK
2.3.4.2 Quadrature PSK (QPSK)
2.3.4.3 Constellation Diagram
2.3.5 Quadrature Amplitude Modulation (QAM)
2.3.5.1 Bandwidth for QAM
2.3.5.2 TDM

QUESTIONS(MODULE 1)

MODULE 1: INTRODUCTION

1) Define data communications. Explain its 4 fundamental characteristics. (4*)
2) Explain different components of data communication system. (6*)
3) Explain different forms of information. (4)
4) Describe simplex, half-duplex and full duplex methods of data flow. (6*)
5) Explain the 3 criteria necessary for an effective and efficient network. (4*)
6) Explain point to point and multipoint connection. (4*)
7) Explain the following topologies:
   i) Mesh ii) Star iii) Bus iv) Ring (12*)
8) Explain in detail LAN & WAN. List the differences between LAN & WAN. (10*)
9) Explain circuit-switched and packet-switched networks. (6*)

MODULE 1(CONT.): NETWORK MODELS

1) Explain TCP/IP architecture with a layer diagram. (4*)
2) List the 5 layers and its functionality in TCP/IP model. (8*)
3) With respect to in TCP/IP model, explain the following:
   i) Encapsulation and decapsulation. ii) Multiplexing and demultiplexing. (8)
4) Explain four levels of addressing employed in TCP/IP protocol. (6*)
5) What are the uses of a layered network model? Compare OSI and TCP/IP models. (4)

MODULE 1(CONT.): DATA AND SIGNALS

1) Compare the following:
   i) Analog signal vs. Digital signal. ii) Periodic signal vs. Non-periodic signal. (4)
2) Describe digital signal as a composite analog signal. (4)
3) Explain 2 methods for transmitting a digital signal (8*)
4) What do you mean by transmission impairment? Explain causes of transmission impairment? (6*)
5) What are the three factors data rate is dependent on? Explain the theoretical formula which was developed to calculate the data rate. (8*)
6) Explain 4 performance parameters of network. (8*)

MODULE 1(CONT.): DIGITAL TRANSMISSION

1) Explain in detail any 6 characteristics of digital signal. (6*)
2) Compare the following:
   i) Data element vs. Signal element. ii) Data rate vs. Signal rate. (4)
3) Explain following encoding schemes with example:
   i) Unipolar Scheme ii) Polar Schemes iii) Bipolar Schemes (8*)
4) Represent the following sequences using different line coding schemes. i) 101011100. ii) 10110011. iii) 00110101. (6*)
5) Define the following:
   i) Network ii) Internet
   ii)iii) Protocol iv) Decibel v) SNR vi) Line coding (6*)

DATA COMMUNICATION

MODULE 1: DIGITAL TRANSMISSION (CONT.)

1) Explain the PCM encoder with neat diagram. (8*)
2) What do you mean by Sampling? Explain three sampling methods with a neat diagram. (4)
3) Explain non-uniform quantization and how to recover original signal using PCM decoder. (4)
4) Explain different types of transmission modes. (8*)
5) What is sampling and quantization? Explain briefly. (6)

MODULE 1(CONT.): ANALOG TRANSMISSION

1) Define digital to analog conversion? List different types of digital to analog conversion. (2)
2) Describe ASK, FSK and PSK mechanisms and apply them over the digital data 101101. (4)
3) Discuss the bandwidth requirement for ASK, FSK and PSK. (4*)
4) Explain different aspects of digital-to-analog conversion? (6*)
5) Define ASK. Explain BASK. (6*)
6) Define FSK. Explain BFSK. (6*)
7) Define PSK. Explain BPSK. (6*)
8) Explain QPSK. (6)
9) Explain the concept of constellation diagram. (6)
10) Explain QAM. (6)

MODULE 1: INTRODUCTION

1.1 DATA COMMUNICATIONS

- Data communication is defined as exchange of data between 2 devices over a transmission-medium.
- A communication-system is made up of
  - → hardware (physical equipment) and
  - → software (programs)
- For data-communication, the communicating-devices must be part of a communication-system.
- Four attributes of a communication-system:

1) Delivery

- The system must deliver data to the correct destination.

2) Accuracy

- The system must deliver the data accurately.
- Normally, the corrupted-data are unusable.

3) Timeliness

- The system must deliver audio/video data in a timely manner.
- This kind of delivery is called real-time transmission.
- Data delivered late are useless.

4) Jitter

- Jitter refers to the variation in the packet arrival-time.
- In other words, jitter is the uneven delay in the delivery of audio/video packets.

1.1.1 Components of Communication System

• Five components of a communication-system (Figure 1.1):

1) Message
2) Sender
3) Receiver
4) Transmission-Medium
5) Protocol

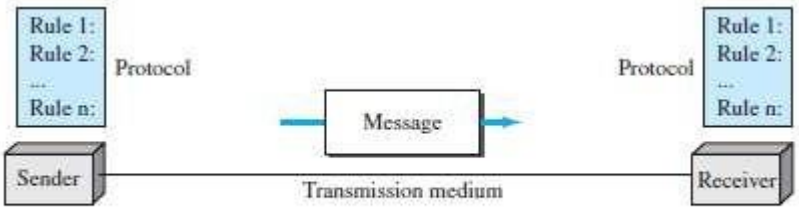


Figure 1.1 *Five components of data communication*

1) Message

- Message is the information (or data) to be communicated.
- Message may consist of
  - → number/text
  - → picture or
  - → audio/video

2) Sender

- Sender is the device that sends the data-message.
- Sender can be
  - → computer and
  - → mobile phone

3) Receiver

- Receiver is the device that receives the message.
- Receiver can be
  - → computer and
  - → mobile phone

4) Transmission Medium

- Transmission-medium is physical-path by which a message travels from sender to receiver.
- Transmission-medium can be wired or wireless.
- Examples of wired medium:
  - → twisted-pair wire (used in landline telephone)
  - → coaxial cable (used in cable TV network)
  - → fiber-optic cable
- Examples of wireless medium:
  - → radio waves
  - → microwaves
  - → infrared waves (ex: operating TV using remote control)

5) Protocol

- A protocol is a set of rules that govern data-communications.
- In other words, a protocol represents an agreement between the communicating-devices.
- Without a protocol, 2 devices may be connected but not communicating.

1.1.2 Data Representation

• Five different forms of information:

1) Text

- Text is represented as a bit-pattern. (Bit-pattern → sequence of bits: 0s or 1s).
- Different sets of bit-patterns are used to represent symbols (or characters).
- Each set is called a code.
- The process of representing symbols is called encoding.
- Popular encoding system: ASCII, Unicode.

2) Number

- Number is also represented as a bit-pattern.
- ASCII is not used to represent number. Instead, number is directly converted to binary-form.

3) Image

- Image is also represented as a bit-pattern.
- An image is divided into a matrix of pixels (picture-elements).
- A pixel is the smallest element of an image. (Pixel → Small dot)
- The size of an image depends upon number of pixels (also called resolution).
  For example: An image can be divided into 1000 pixels or 10,000 pixels.
- Two types of images:
  - **i) Black & White Image**
    - ¤ If an image is black & white, each pixel can be represented by a value either 0 or 1.
    - ¤ For example: Chessboard
  - **ii) Color Image**
    - ¤ There are many methods to represent color images.
    - ¤ RGB is one of the methods to represent color images.
    - ¤ RGB is called so called '.' each color is combination of 3 colors: red, green & blue.

4) Audio

- Audio is a representation of sound.
- By nature, audio is different from text, numbers, or images. Audio is continuous, not discrete.

5) Video

- Video is a representation of movie.
- Video can either
  → be produced as a continuous entity (e.g., by a TV camera), or
  → be a combination of images arranged to convey the idea of motion.

1.1.3 Direction of Data Flow

• Three ways of data-flow between 2 devices (Figure 1.2):
1) Simplex
2) Half-duplex
3) Full-duplex

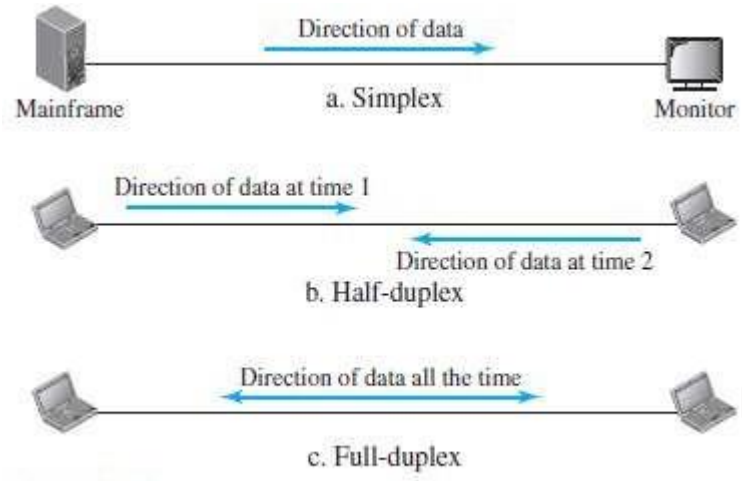


Figure 1.2 *Data flow (simplex, half-duplex, and full-duplex)*

1) Simplex
- The communication is unidirectional
  (For ex: The simplex mode is like a one-way street).
- On a link, out of 2 devices:
  i) Only one device can transmit.
  ii) Another device can only receive.
- For example (Figure 1.2a):
  The monitor can only accept output.
- Entire-capacity of channel is used to send the data in one direction.

2) Half Duplex
- Both the stations can transmit as well as receive but not at the same time.
  (For ex: The half-duplex mode is like a one-lane road with 2 directional traffic).
- When one station is sending, the other can only receive and vice-versa.
- For example (Figure 1.2b): Walkie-talkies
- Entire-capacity of a channel is used by one of the 2 stations that are transmitting the data.

3) Full Duplex
- Both stations can transmit and receive at the same time.
  (For ex: The full-duplex is like a 2-way street with traffic flowing in both directions at the same time).
- For example (Figure 1.2c):
  Mobile phones (When 2 people are communicating by a telephone line, both can listen and talk at the same time)
- Entire-capacity of a channel is shared by both the stations that are transmitting the data.

1.2 NETWORKS

- A network is defined as a set of devices interconnected by communication-links.
- This interconnection among computers facilitates information sharing among them.
- Computers may connect to each other by either wired or wireless media.
- Often, devices are referred to as nodes.
- A node can be any device capable of sending/receiving data in the network.
- For example: Computer & Printer
- The best-known computer network is the Internet.

1.2.1 Network Criteria

- A network must meet following 3 criteria's:

1) Performance

- Performance can be measured using i) Transit-time or ii) Response-time.
  - **i) Transit Time** is defined as time taken to travel a message from one device to another.
  - **ii) Response Time** is defined as the time elapsed between enquiry and response.
- The network-performance depends on following factors:
  - i) Number of users
  - ii) Type of transmission-medium
  - iii) Efficiency of software
- Often, performance is evaluated by 2 networking-metrics: i) throughput and ii) delay.
- Good performance can be obtained by achieving higher throughput and smaller delay times

2) Reliability

- Reliability is measured by
  - → frequency of network-failure
  - → time taken to recover from a network-failure
  - → network's robustness in a disaster
- More the failures are, less is the network's reliability.

3) Security

- Security refers to the protection of data from the unauthorized access or damage.
- It also involves implementing policies for recovery from data-losses.

1.2.2 Physical Structures

1.2.2.1 Type of Connection

• Two types of connections (Figure 1.3):

1) Point-to-Point

- Only two devices are connected by a dedicated-link (Figure 1.3a).
- Entire-capacity of the link is reserved for transmission between those two devices.
- For example: Point-to-Point connection b/w remote-control & TV for changing the channels.

2) Multipoint (Multi-Drop)

- Three or more devices share a single link.
- The capacity of the channel is shared, either spatially or temporally (Figure 1.3b).
  - i) If link is used simultaneously by many devices, then it is spatially shared connection.
  - ii) If user takes turns while using the link, then it is time shared (temporal) connection. (spatially→space or temporally→time)

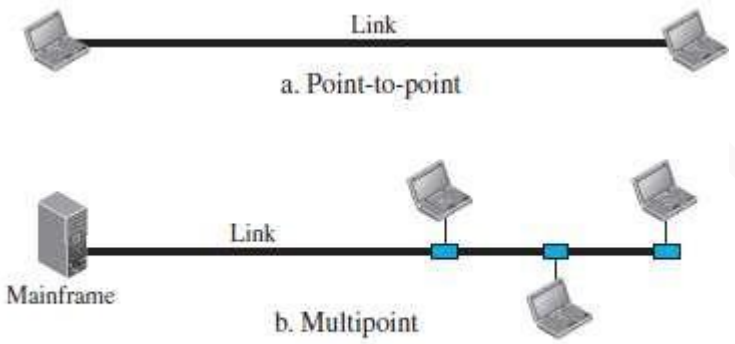


Figure 1.3 *Types of connections: point-to-point and multipoint*

1.2.2.2 Physical Topology

- The physical-topology defines how devices are connected to make a network.
- Four basic topologies are:
    1) Mesh
    2) Star
    3) Bus and
    4) Ring

1.2.2.2.1 Bus Topology

- All the devices are connected to the single cable called bus (Figure 1.4).
- Every device communicates with the other device through this bus.
- A data from the source is broadcasted to all devices connected to the bus.
- Only the intended-receiver, whose physical-address matches, accepts the data.

Figure 1.4 *A bus topology connecting three stations*

- Devices are connected to the bus by drop-lines and taps.
- A drop-line is a connection running between the device and the bus.
- A tap is a connector that links to the bus or
- Advantages:
    1) Easy installation.
    2) Cable required is the least compared to mesh/star topologies.
    3) Redundancy is eliminated.
    4) Costs less (Compared to mesh/star topologies).
    5) Mostly used in small networks. Good for LAN.
- Disadvantages:
    1) Difficult to detect and troubleshoot fault.
    2) Signal reflection at the taps can cause degradation in quality.
    3) A fault/break in the cable stops all transmission.
    4) There is a limit on
        i) Cable length
        ii) Number of nodes that can be connected.
    5) Security is very low because all the devices receive the data sent from the source.

1.2.2.2.2 Star Topology

- All the devices are connected to a central controller called a hub (Figure 1.5).
- There exists a dedicated point-to-point link between a device & a hub.
- The devices are not directly linked to one another. Thus, there is no direct traffic between devices.
- The hub acts as a junction:

  If device-1 wants to send data to device-2,
  the device-1 sends the data to the hub,
  then the hub relays the data to the device-2.

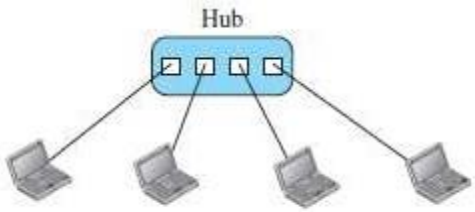


Figure 1.5 *A star topology connecting four stations*

- Advantages:
  1) Less expensive: Each device needs only one link & one I/O port to connect it to any devices.
  2) Easy installation & reconfiguration: Nodes can be added/removed w/o affecting the network.
  3) Robustness: If one link fails, it does not affect the entire system.
  4) Easy to detect and troubleshoot fault.
  5) Centralized management: The hub manages and controls the whole network.
- Disadvantages:
  1) Single point of failure: If the hub goes down, the whole network is dead.
  2) Cable length required is the more compared to bus/ring topologies.
  3) Number of nodes in network depends on capacity of hub.

1.2.2.2.3 Ring Topology

- Each device is connected to the next, forming a ring (Figure 1.6).
- There are only two neighbors for each device.
- Data travels around the network in one direction till the destination is reached.
- Sending and receiving of data takes place by the help of token.
- Each device has a repeater.
- A repeater
  → receives a signal on transmission-medium &
  → regenerates & passes the signal to next device.

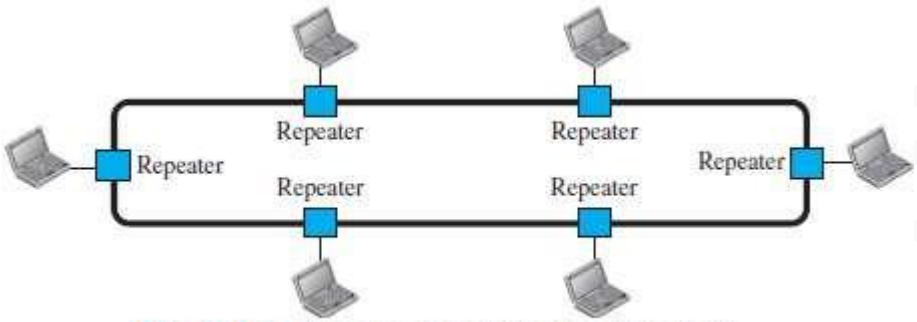


Figure 1.6 *A ring topology connecting six stations*

- Advantages:
  1) Easy installation and reconfiguration.
     To add/delete a device, requires changing only 2 connections.
  3) Fault isolation is simplified.
     If one device does not receive a signal within a specified period, it can issue an alarm.
     The alarm alerts the network-operator to the problem and its location.
  3) Congestion reduced: Because all the traffic flows in only one direction.
- Disadvantages:
  1) Unidirectional traffic.
  2) A fault in the ring/device stops all transmission.
     The above 2 drawbacks can be overcome by using dual ring.
  3) There is a limit on
     i) Cable length &
     ii) Number of nodes that can be connected.
  4) Slower: Each data must pass through all the devices between source and destination.

1.2.2.2.4 Mesh Topology

- All the devices are connected to each other (Figure 1.7).
- There exists a dedicated point-to-point link between all devices.
- There are n(n-1) physical channels to link n devices.
- Every device not only sends its own data but also relays data from other nodes.
- For 'n' nodes,
  → there are n(n-1) physical-links
  → there are n(n-1)/2 duplex-mode links
- Every device must have (n–1) I/O ports to be connected to the other (n-1) devices.

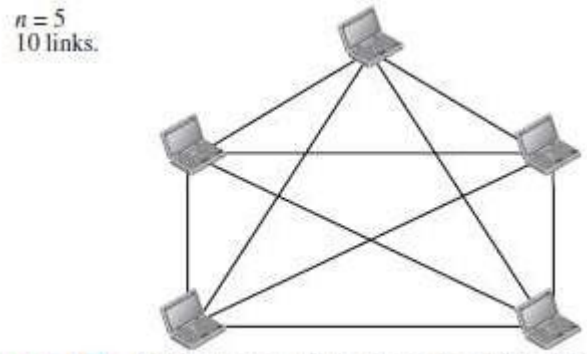


Figure 1.7 *A fully connected mesh topology (five devices)*

- Advantages:
  1) Congestion reduced: Each connection can carry its own data load.
  2) Robustness: If one link fails, it does not affect the entire system.
  3) Security: When a data travels on a dedicated-line, only intended-receiver can see the data.
  4) Easy fault identification & fault isolation: Traffic can be re-routed to avoid problematic links.
- Disadvantages:
  1) Difficult installation and reconfiguration.
  2) Bulk of wiring occupies more space than available space.
  3) Very expensive: as there are many redundant connections.
  4) Not mostly used in computer networks. It is commonly used in wireless networks.
  5) High redundancy of the network-connections.

1.3 Network Types

• Two popular types of networks:
1) LAN (Local Area Network) &
2) WAN (Wide Area Network)

1.3.1 LAN

• LAN is used to connect computers in a single office, building or campus (Figure 1.8).
• LAN is usually privately owned network.
• A LAN can be simple or complex.
1) Simple: LAN may contain 2 PCs and a printer.
2) Complex: LAN can extend throughout a company.
• Each host in a LAN has an address that uniquely defines the host in the LAN.
• A packet sent by a host to another host carries both source host's and destination host's addresses.
• LANs use a smart connecting switch.
• The switch is able to
→ recognize the destination address of the packet &
→ guide the packet to its destination.
• The switch
→ reduces the traffic in the LAN &
→ allows more than one pair to communicate with each other at the same time.
• Advantages:
1) Resource Sharing
➢ Computer resources like printers and hard disks can be shared by all devices on the network.
2) Expansion
➢ Nowadays, LANs are connected to WANs to create communication at a wider level.

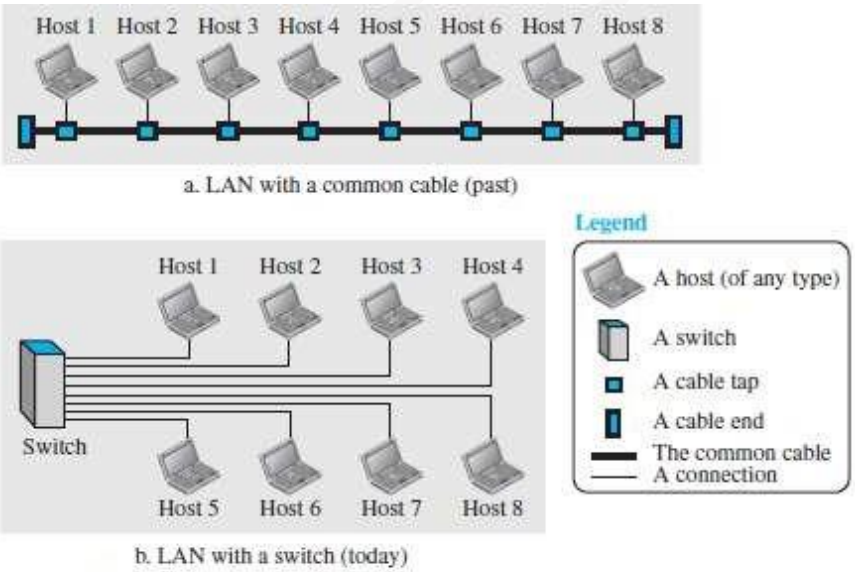


Figure 1.8 *An isolated LAN in the past and today*

1.3.2 WAN

- WAN is used to connect computers anywhere in the world.
- WAN can cover larger geographical area. It can cover cities, countries and even continents.
- WAN interconnects connecting devices such as switches, routers, or modems.
- Normally, WAN is
  - → created & run by communication companies (Ex: BSNL, Airtel)
  - → leased by an organization that uses it.
- A WAN can be of 2 types:

1) Point-to-Point WAN

➢ A point-to-point WAN is a network that connects 2 communicating devices through a transmission media (Figure 1.9).

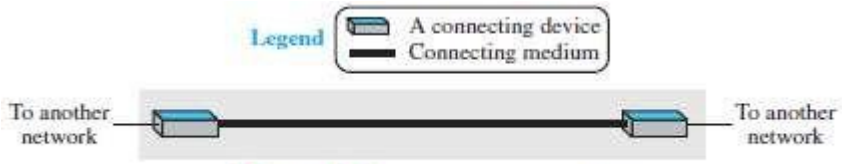


Figure 1.9 *A point-to-point WAN*

2) Switched WAN

➢ A switched WAN is a network with more than two ends.

➢ The switched WAN can be the backbones that connect the Internet.

➢ A switched WAN is a combination of several point-to-point WANs that are connected by switches (Figure 1.10).

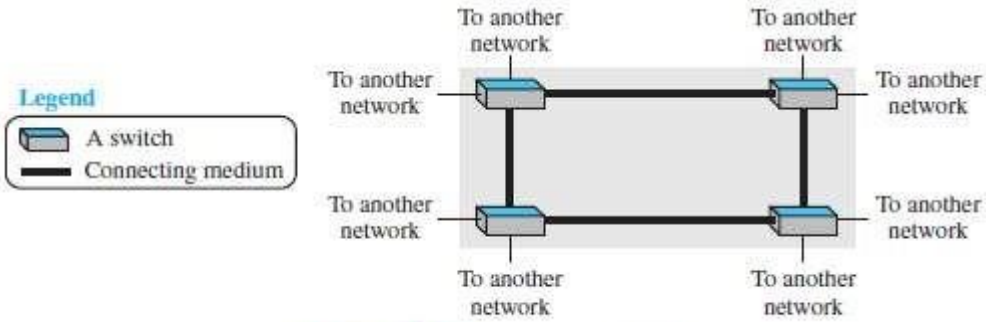


Figure 1.10 *A switched WAN*

1.3.2.1 Internetwork

- A network of networks is called an internet. (Internet → inter-network) (Figure 1.12).
- For example (Figure 1.11):
  - Assume that an organization has two offices,
    - i) First office is on the east coast &
    - ii) Second office is on the west coast.
  - Each office has a LAN that allows all employees in the office to communicate with each other.
  - To allow communication between employees at different offices, the management leases a point-to-point dedicated WAN from a ISP and connects the two LANs.
    (ISP → Internet service provider such as a telephone company ex: BSNL).

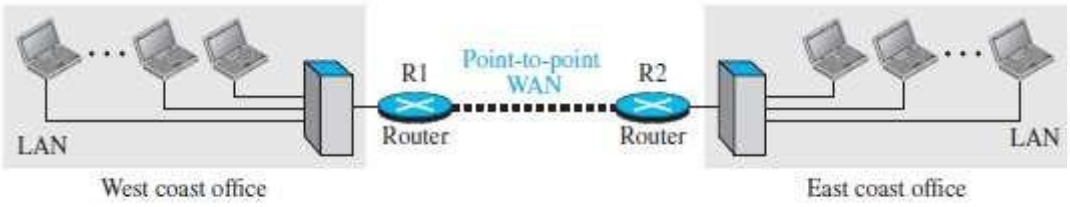


Figure 1.11 *An internetwork made of two LANs and one point-to-point WAN*

- When a host in the west coast office sends a message to another host in the same office, the router blocks the message, but the switch directs the message to the destination.
- On the other hand, when a host on the west coast sends a message to a host on the east coast, router R1 routes the packet to router R2, and the packet reaches the destination.

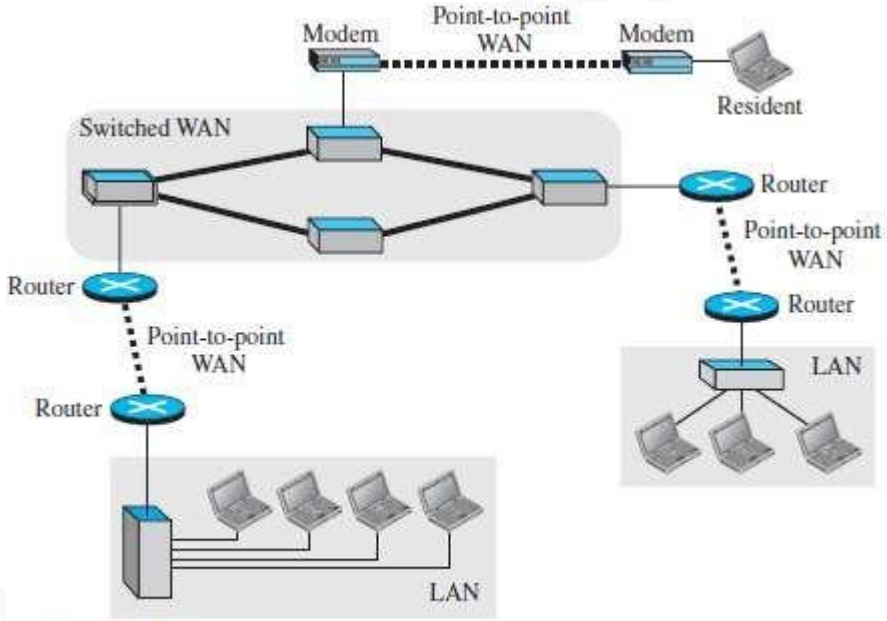


Figure 1.12 *A heterogeneous network made of four WANs and three LANs*

1.3.3 LAN vs. WAN

Parameters	LAN	WAN
Expands to	Local Area Network	Wide Area Network
Meaning	LAN is used to connect computers in a single office, building or campus	WAN is used to connect computers in a large geographical area such as countries
Ownership of network	Private	Private or public
Range	Small: up to 10 km	Large: Beyond 100 km
Speed	High: Typically 10, 100 and 1000 Mbps	Low: Typically 1.5 Mbps
Propagation Delay	Short	Long
Cost	Low	High
Congestion	Less	More
Design & maintenance	Easy	Difficult
Fault Tolerance	More Tolerant	Less Tolerant
Media used	Twisted pair	Optical fiber or radio waves
Used for	College, Hospital	Internet
Interconnects	LAN interconnects hosts	WAN interconnects connecting devices such as switches, routers, or modems

1.3.4 Switching

- An internet is a switched network in which a switch connects at least two links together.
- A switch needs to forward data from a network to another network when required.
- Two types of switched networks are 1) circuit-switched and 2) packet-switched networks.

1.3.4.1 Circuit Switched Network

- A dedicated connection, called a circuit, is always available between the two end systems.
- The switch can only make it active or inactive.

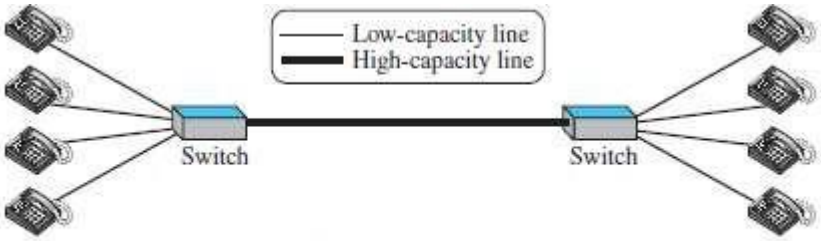


Figure 1.13 *A circuit-switched network*

- As shown in Figure 1.13, the 4 telephones at each side are connected to a switch.
- The switch connects a telephone at one side to a telephone at the other side.
- A high-capacity line can handle 4 voice communications at the same time.
- The capacity of high line can be shared between all pairs of telephones.
- The switch is used for only forwarding.

- Advantage:
  A circuit-switched network is efficient only when it is working at its full capacity.
- Disadvantage:
  Most of the time, the network is inefficient because it is working at partial capacity.

1.3.4.2 Packet Switched Network

- In a computer network, the communication between the 2 ends is done in blocks of data called packets.
- The switch is used for both storing and forwarding because a packet is an independent entity that can be stored and sent later.

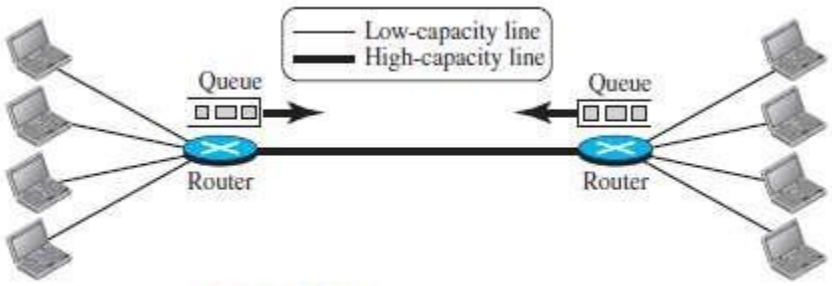


Figure 1.14 *A packet-switched network*

- As shown in Figure 1.14, the 4 computers at each side are connected to a router.
- A router has a queue that can store and forward the packet.
- The high-capacity line has twice the capacity of the low-capacity line.
- If only 2 computers (one at each site) need to communicate with each other, there is no waiting for the packets.
- However, if packets arrive at one router when high-capacity line is at its full capacity, the packets should be stored and forwarded.

- Advantages:
  A packet-switched network is more efficient than a circuit switched network.
- Disadvantage:
  The packets may encounter some delays.

1.3.5 The Internet Today

- A network of networks is called an internet. (Internet → inter-network)
- Internet is made up of (Figure 1.15)
  - **1)** Backbones
  - **2)** Provider networks &
  - **3)** Customer networks

1) Backbones

- Backbones are large networks owned by communication companies such as BSNL and Airtel.
- The backbone networks are connected through switching systems, called peering points.

2) Provider Networks

- Provider networks use the services of the backbones for a fee.
- Provider networks are connected to backbones and sometimes to other provider networks.

3) Customer Networks

- Customer networks actually use the services provided by the Internet.
- Customer networks pay fees to provider networks for receiving services.

- Backbones and provider networks are also called Internet Service Providers (ISPs).
- The backbones are often referred to as international ISPs.
  The provider networks are often referred to as national or regional ISPs.

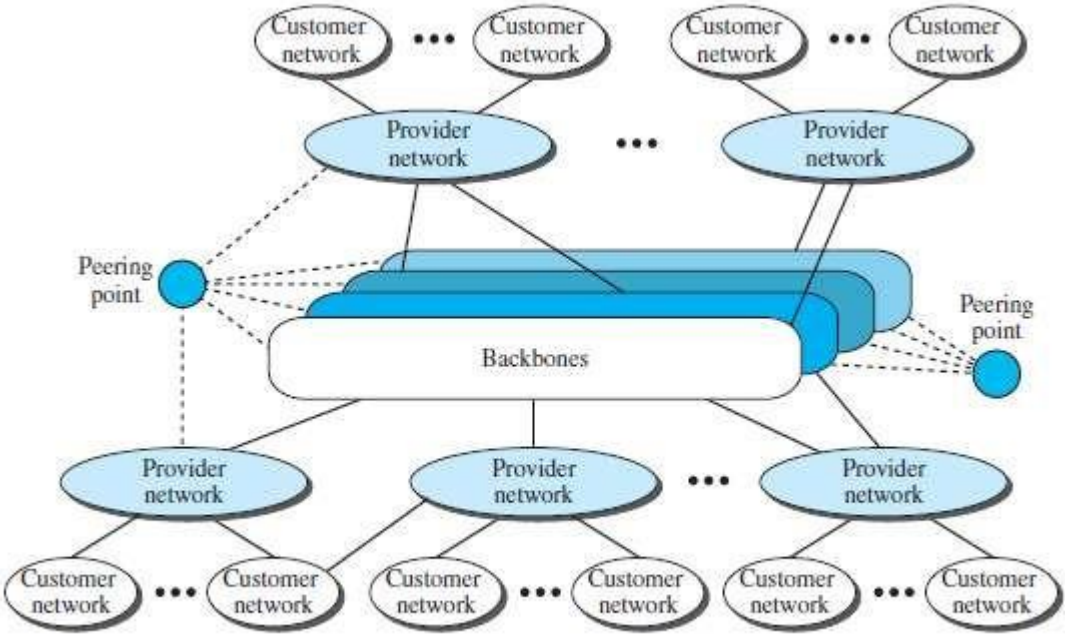


Figure 1.15 *The Internet today*

MODULE 1(CONT.): NETWORK MODELS

1.4 PROTOCOL LAYERING

• A protocol defines the rules that both the sender and receiver and all intermediate devices need to follow to be able to communicate effectively.
• When communication is simple, we may need only one simple protocol.
When communication is complex, we need to divide the task b/w different layers. We need a protocol at each layer, or protocol layering.

1.4.1 Scenarios

First Scenario

• In the first scenario, communication is so simple that it can occur in only one layer (Figure 2.1).
• Assume Maria and Ann are neighbors with a lot of common ideas.
• Communication between Maria and Ann takes place in one layer, face to face, in the same language

Figure 2.1 *A single-layer protocol*

Second Scenario

• Maria and Ann communicate using regular mail through the post office (Figure 2.2).
• However, they do not want their ideas to be revealed by other people if the letters are intercepted.
• They agree on an encryption/decryption technique.
• The sender of the letter encrypts it to make it unreadable by an intruder; the receiver of the letter decrypts it to get the original letter.

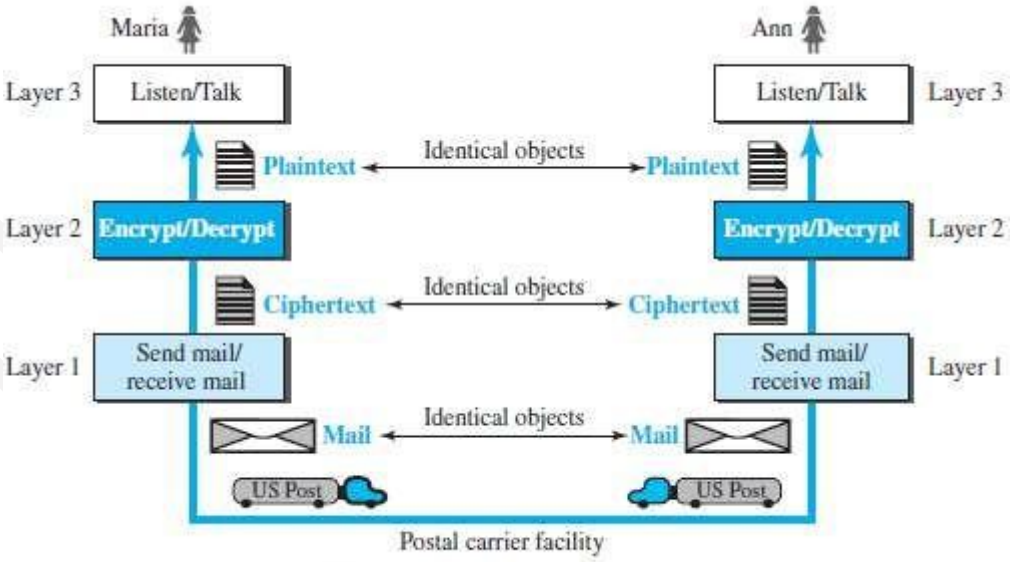


Figure 2.2 *A three-layer protocol*

1.4.1.1 Protocol Layering

- Protocol layering enables us to divide a complex task into several smaller and simpler tasks.
- Modularity means independent layers.
- A layer (module) can be defined as a black box with inputs and outputs, without concern about how inputs are changed to outputs.
- If two machines provide the same outputs when given the same inputs, they can replace each other.
- Advantages:
  - **1)** It allows us to separate the services from the implementation.
  - **2)** There are intermediate systems that need only some layers, but not all layers.
- Disadvantage:
  - 1) Having a single layer makes the job easier. There is no need for each layer to provide a service to the upper layer and give service to the lower layer.

1.4.2 Principles of Protocol Layering

1) First Principle

- If we want bidirectional communication, we need to make each layer able to perform 2 opposite tasks, one in each direction.
- For example, the third layer task is to listen (in one direction) and talk (in the other direction).

2) Second Principle

- The two objects under each layer at both sites should be identical.
- For example, the object under layer 3 at both sites should be a plaintext letter.

1.4.3 Logical Connections

- We have layer-to-layer communication (Figure 2.3).
- There is a logical connection at each layer through which 2 end systems can send the object created from that layer.

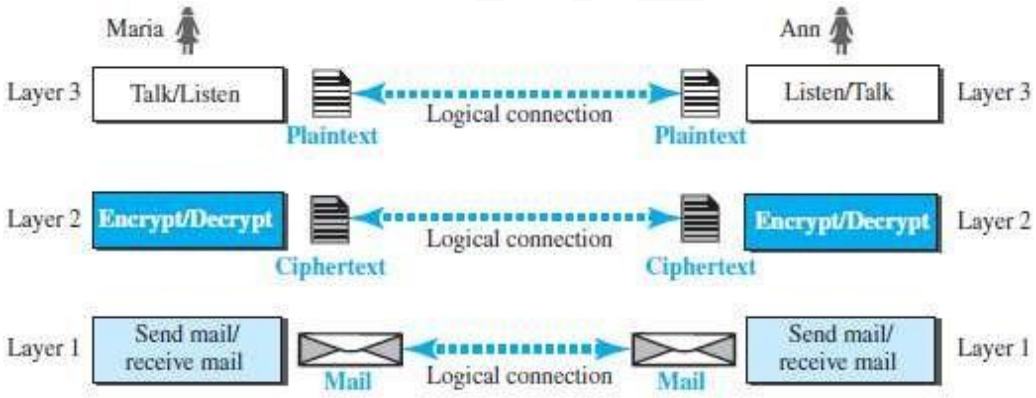


Figure 2.3 *Logical connection between peer layers*

1.5 TCP/IP PROTOCOL SUITE

• TCP/IP is a protocol-suite used in the Internet today.
• Protocol-suite refers a set of protocols organized in different layers.
• It is a hierarchical protocol made up of interactive modules, each of which provides a specific functionality.
• The term hierarchical means that each upper level protocol is supported by the services provided by one or more lower level protocols.

1.5.1 Layered Architecture

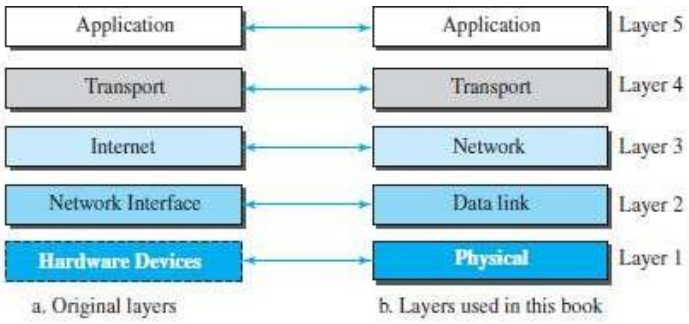


Figure 2.4 *Layers in the TCP/IP protocol suite*

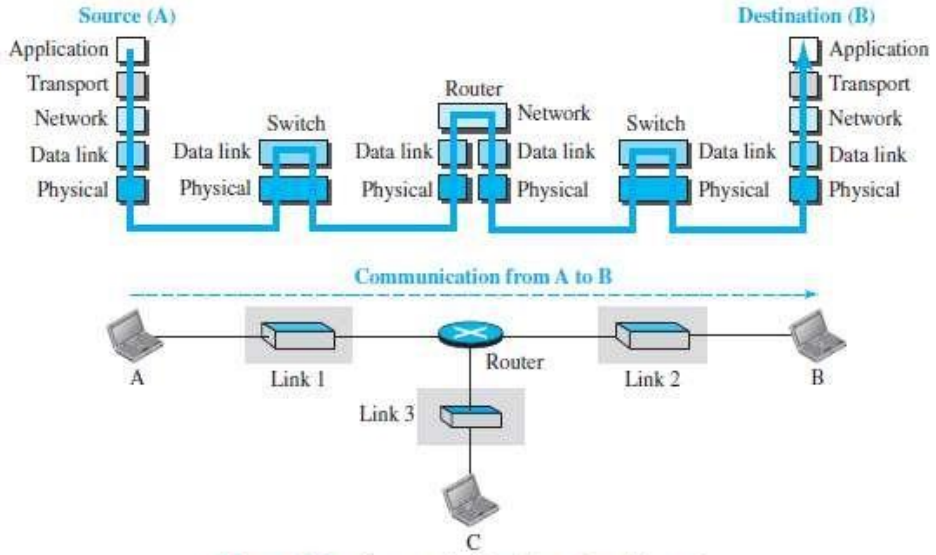


Figure 2.5 *Communication through an internet*

• Let us assume that computer A communicates with computer B (Figure 2.4).
• As the Figure 2.5 shows, we have five communicating devices:
  1) Source host(computer A)
  2) Link-layer switch in link 1
  3) Router
  4) Link-layer switch in link 2
  5) Destination host (computer B).
• Each device is involved with a set of layers depending on the role of the device in the internet.
• The two hosts are involved in all five layers.
• The source host
  → creates a message in the application layer and
  → sends the message down the layers so that it is physically sent to the destination host.
• The destination host
  → receives the message at the physical layer and
  → then deliver the message through the other layers to the application layer.
• The router is involved in only three layers; there is no transport or application layer.
• A router is involved in n combinations of link and physical layers.
  where n = number of links the router is connected to.
• The reason is that each link may use its own data-link or physical protocol.
• A link-layer switch is involved only in two layers: i) data-link and ii) physical.

1.5.2 Layers in the TCP/IP Protocol Suite

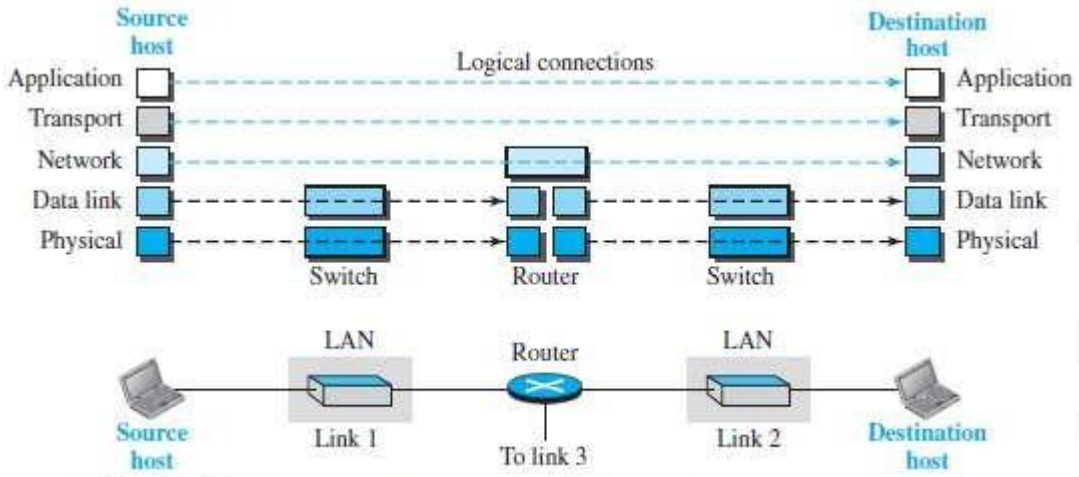


Figure 2.6 *Logical connections between layers of the TCP/IP protocol suite*

- As shown in the figure 2.6, the duty of the application, transport, and network layers is end-to-end.
- However, the duty of the data-link and physical layers is hop-to-hop. A hop is a host or router.
- The domain of duty of the top three layers is the internet.
  The domain of duty of the two lower layers is the link.
- In top 3 layers, the data unit should not be changed by any router or link-layer switch.
  In bottom 2 layers, the data unit is changed only by the routers, not by the link-layer switches.

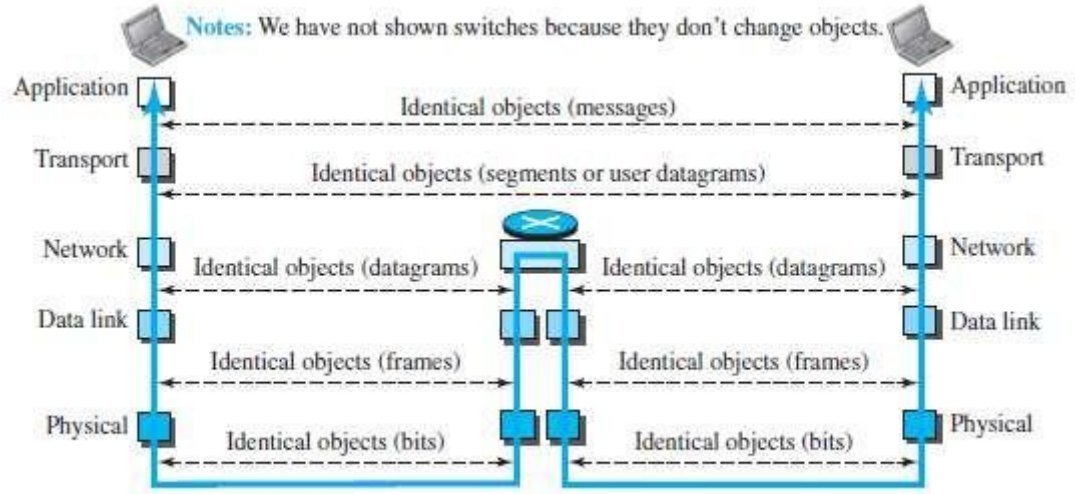


Figure 2.7 *Identical objects in the TCP/IP protocol suite*

- Identical objects exist between two hops. Because router may fragment the packet at the network layer and send more packets than received (Figure 2.7).
- The link between two hops does not change the object.

1.5.3 Description of Each Layer

Physical Layer

- The physical layer is responsible for movements of individual bits from one node to another node.
- Transmission media is another hidden layer under the physical layer.
- Two devices are connected by a transmission medium (cable or air).
- The transmission medium does not carry bits; it carries electrical or optical signals.
- The physical layer
  - → receives bits from the data-link layer &
  - → sends through the transmission media.

Data Link Layer

- Data-link-layer (DLL) is responsible for moving frames from one node to another node over a link.
- The link can be wired LAN/WAN or wireless LAN/WAN.
- The data-link layer
  - → gets the datagram from network layer
  - → encapsulates the datagram in a packet called a frame.
  - → sends the frame to physical layer.
- TCP/IP model does not define any specific protocol.
- DLL supports all the standard and proprietary protocols.
- Each protocol may provide a different service.
- Some protocols provide complete error detection and correction; some protocols provide only error correction.

Network Layer

- The network layer is responsible for source-to-destination transmission of data.
- The network layer is also responsible for routing the packet.
- The routers choose the best route for each packet.
- Why we need the separate network layer?
  1) The separation of different tasks between different layers.
  2) The routers do not need the application and transport layers.
- TCP/IP model defines 5 protocols:
  1) IP (Internetworking Protocol)
  2) ARP (Address Resolution Protocol)
  3) ICMP (Internet Control Message Protocol)
  4) IGMP (Internet Group Message Protocol)

1) IP

- IP is the main protocol of the network layer.
- IP defines the format and the structure of addresses.
- IP is also responsible for routing a packet from its source to its destination.
- It is a connection-less & unreliable protocol.
  - i) Connection-less means there is no connection setup b/w the sender and the receiver.
  - ii) Unreliable protocol means
    - → IP does not make any guarantee about delivery of the data.
    - → Packets may get dropped during transmission.
- It provides a best-effort delivery service.
- Best effort means IP does its best to get the packet to its destination, but with no guarantees.
- IP does not provide following services
  - → flow control
  - → error control
  - → congestion control services.
- If an application requires above services, the application should rely only on the transport-layer protocol.

2) ARP

- ARP is used to find the physical-address of the node when its Internet-address is known.
- Physical address is the 48-bit address that is imprinted on the NIC or LAN card.
- Internet address (IP address) is used to uniquely & universally identify a device in the internet.

3) ICMP

- ICMP is used to inform the sender about datagram-problems that occur during transit.

4) IGMP

- IGMP is used to send the same message to a group of recipients.

Transport Layer

- TL protocols are responsible for delivery of a message from a process to another process.
- The transport layer
  - → gets the message from the application layer
  - → encapsulates the message in a packet called a segment and
  - → sends the segment to network layer.
- TCP/IP model defines 3 protocols:1) TCP (Transmission Control Protocol)
  2) UDP (User Datagram Protocol) &
  3) SCTP (Stream Control Transmission Protocol)

1) TCP

- TCP is a reliable connection-oriented protocol.
- A connection is established b/w the sender and receiver before the data can be transmitted.
- TCP provides
  - → flow control
  - → error control and
  - → congestion control

2) UDP

- UDP is the simplest of the 3 transport protocols.
- It is an unreliable, connectionless protocol.
- It does not provide flow, error, or congestion control.
- Each datagram is transported separately & independently.
- It is suitable for application program that
  - → needs to send short messages &
  - → cannot afford the retransmission.

3) SCTP

- SCTP provides support for newer applications such as voice over the Internet.
- It combines the best features of UDP and TCP.

Application Layer

- The two application layers exchange messages between each other.
- Communication at the application layer is between two processes (two programs running at this layer).
- To communicate, a process sends a request to the other process and receives a response.
- Process-to-process communication is the duty of the application layer.
- TCP/IP model defines following protocols:
  1) SMTP is used to transport email between a source and destination.
  2) TELNET is used for accessing a site remotely.
  3) FTP is used for transferring files from one host to another.
  4) DNS is used to find the IP address of a computer.
  5) SNMP is used to manage the Internet at global and local levels.
  6) HTTP is used for accessing the World Wide Web (WWW).

(FTP → File Transfer Protocol
(DNS → Domain Name System
(SNMP → Simple Network Management Protocol
SMTP → Simple Mail Transfer Protocol)
HTTP → Hyper Text Transfer Protocol)
TELNET → Terminal Network)

1.5.4 Encapsulation and Decapsulation

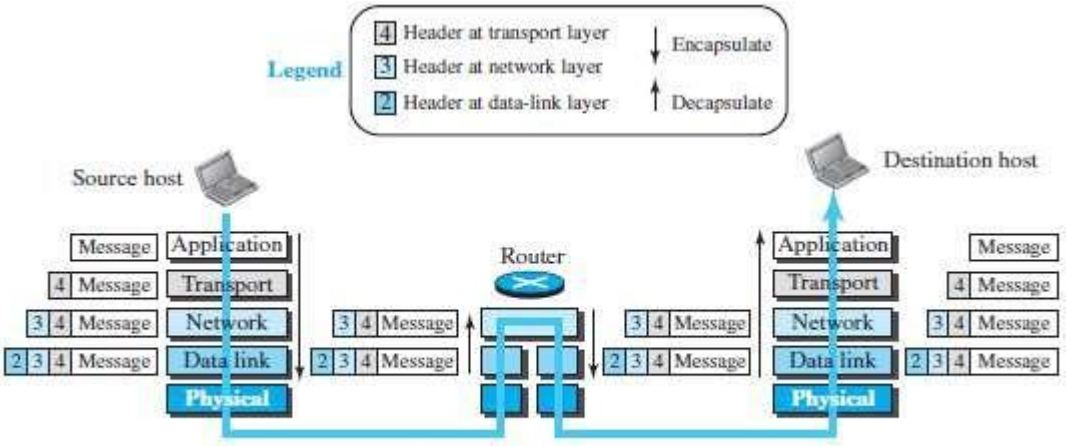


Figure 2.8 *Encapsulation/Decapsulation*

A) Encapsulation at the Source Host

• At the source, we have only encapsulation (Figure 2.8).

1) At the application layer, the data to be exchanged is referred to as a message.
- A message normally does not contain any header or trailer.
- The message is passed to the transport layer.

2) The transport layer takes the message as the payload.
- TL adds its own header to the payload.
- The header contains
  - → identifiers of the source and destination application programs
  - → information needed for flow, error control, or congestion control.
- The transport-layer packet is called the segment (in TCP) and the user datagram (in UDP).
- The segment is passed to the network layer.

3) The network layer takes the transport-layer packet as payload.
- NL adds its own header to the payload.
- The header contains
  - → addresses of the source and destination hosts
  - → some information used for error checking of the header &
  - → fragmentation information.
- The network-layer packet is called a datagram.
- The datagram is passed to the data-link layer.

4) The data-link layer takes the network-layer packet as payload.
- DLL adds its own header to the payload.
- The header contains the physical addresses of the host or the next hop (the router).
- The link-layer packet is called a frame.
- The frame is passed to the physical layer for transmission

B) Decapsulation and Encapsulation at the Router

• At the router, we have both encapsulation & encapsulation and because the router is connected to two or more links.

1) Data-link layer
- → receives frame from physical layer
- → decapsulates the datagram from the frame and
- → passes the datagram to the network layer.

2) The network layer
- → inspects the source and destination addresses in the datagram header and
- → consults forwarding table to find next hop to which the datagram is to be delivered.

- The datagram is then passed to the data-link layer of the next link.

3) The data-link layer of the next link
- → encapsulates the datagram in a frame and
- → passes the frame to the physical layer for transmission.

C) Decapsulation at the Destination Host

• At the destination host, each layer
→ decapsulates the packet received from lower layer
→ removes the payload and
→ delivers the payload to the next-higher layer

1.5.5 Addressing

• We have logical communication between pairs of layers.
• Any communication that involves 2 parties needs 2 addresses: source address and destination address.
• We need 4 pairs of addresses (Figure 2.9):

1) At the application layer, we normally use names to define
→ site that provides services, such as bitm.edu.in, or
→ e-mail address, such as forouzan@gmail.com.

2) At the transport layer, addresses are called port numbers.
➢ Port numbers define the application-layer programs at the source and destination.
➢ Port numbers are local addresses that distinguish between several programs running at the same time.

3) At the network-layer, addresses are called IP addresses.
➢ IP address uniquely defines the connection of a device to the Internet.
➢ The IP addresses are global, with the whole Internet as the scope.

4) At the data link-layer, addresses are called MAC addresses
➢ The MAC addresses defines a specific host or router in a network (LAN or WAN).
➢ The MAC addresses are locally defined addresses.

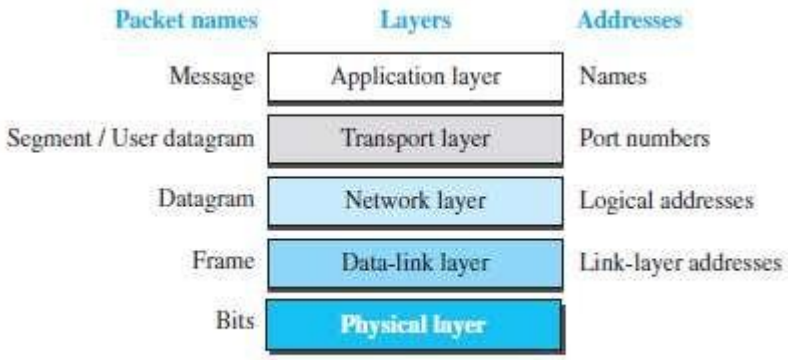


Figure 2.9 *Addressing in the TCP/IP protocol suite*

1.6 OSI MODEL

- OSI model was developed by ISO.
- ISO is the organization, OSI is the model.
- Purpose: OSI was developed to allow systems with diff. platforms to communicate with each other.
- Platform means hardware, software or operating system.
- OSI is a network-model that defines the protocols for network communications.
- OSI has 7 layers as follows (Figure 2.11):
    1) Application Layer
    2) Presentation Layer
    3) Session Layer
    4) Transport Layer
    5) Network Layer
    6) Data Link Layer
    7) Physical Layer
- Each layer has specific duties to perform and has to co-operate with the layers above & below it.

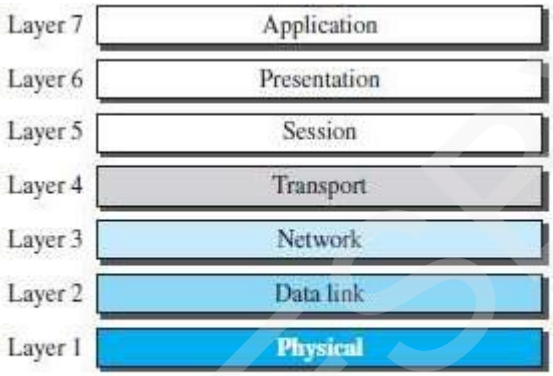


Figure 2.11 *The OSI model*

1.6.1 OSI vs. TCP/IP

1) The four bottommost layers in the OSI model & the TCP/IP model are same (Figure 2.12). However, the Application-layer of TCP/IP model corresponds to the Session, Presentation & Application Layer of OSI model.
    Two reasons for this are:
    1) TCP/IP has more than one transport-layer protocol.
    2) Many applications can be developed at Application layer
2) The OSI model specifies which functions belong to each of its layers.
    In TCP/IP model, the layers contain relatively independent protocols that can be mixed and matched depending on the needs of the system.

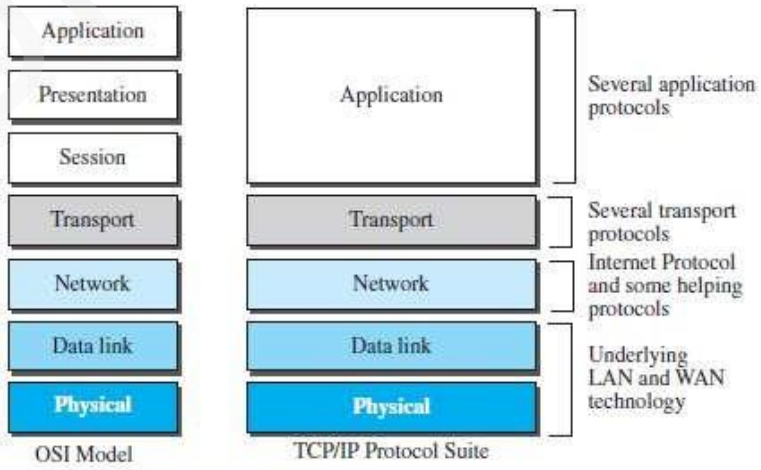


Figure 2.12 *TCP/IP and OSI model*

MODULE 1(CONT.): DATA AND SIGNALS

1.7 DATA AND SIGNALS

1.7.1 Analog & Digital Data

- To be transmitted, data must be transformed to electromagnetic-signals.
- Data can be either analog or digital.

1) Analog Data refers to information that is continuous.

- For example:

  The sounds made by a human voice.

2) Digital Data refers to information that has discrete states.

- For example:

  Data are stored in computer-memory in the form of 0s and 1s.

1.7.2 Analog & Digital Signals

- Signals can be either analog or digital (Figure 3.2).

1) Analog Signal has infinitely many levels of intensity over a period of time.

2) Digital Signal can have only a limited number of defined values.

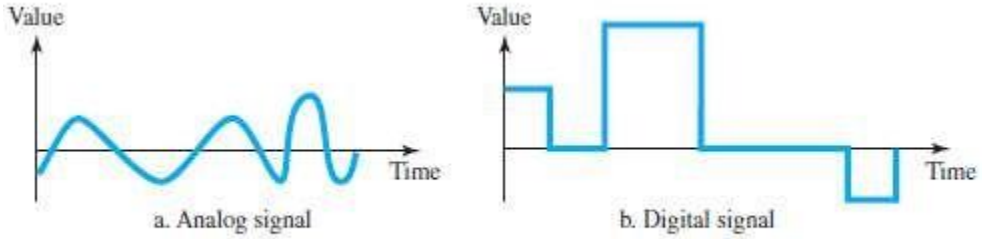


Figure 3.2 *Comparison of analog and digital signals*

1.7.3 Periodic & Non-Periodic Signals

- The signals can take one of 2 forms: periodic or non-periodic.

1) Periodic Signal

- Signals which repeat itself after a fixed time period are called Periodic Signals.
- The completion of one full pattern is called a cycle.

2) Non-Periodic Signal

- Signals which do not repeat itself after a fixed time period are called Non-Periodic Signals.

1.8 DIGITAL SIGNALS

- Information can be represented by a digital signal.
- For example:
  1) 1 can be encoded as a positive voltage.
     0 can be encoded as a zero voltage (Figure 3.17a).
  2) A digital signal can have more than 2 levels (Figure 3.17b).

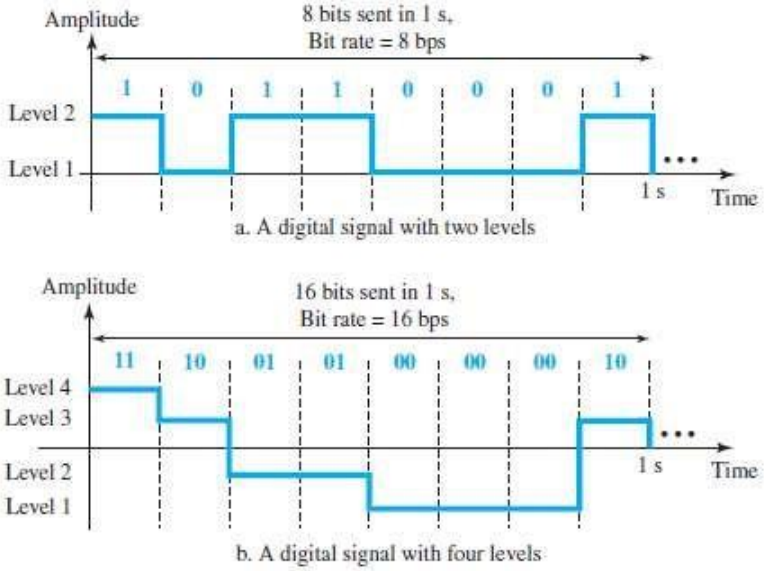


Figure 3.17 *Two digital signals: one with two signal levels and the other with four signal levels*

Example 1.1

of bits from the following formula. Each signal level is represented by 3 bits.

$$\text{Number of bits per level} = \log_2 8 = 3$$

c

1.8.1 Bit Rate

- The bit rate is the number of bits sent in 1s.
- The bit rate is expressed in bits per second (bps).

Example 1.2

Assume we need to download text documents at the rate of 100 pages per second. What is the required bit rate of the channel?

Solution

A page is an average of 24 lines with 80 characters in each line. If we assume that one character requires 8 bits, the bit rate is

$$100 \times 24 \times 80 \times 8 = 1{,}536{,}000 \text{ bps} = 1.536 \text{ Mbps}$$

Example 1.3

A digitized voice channel, as we will see in Chapter 4, is made by digitizing a 4-kHz bandwidth analog voice signal. We need to sample the signal at twice the highest frequency (two samples per hertz). We assume that each sample requires 8 bits. What is the required bit rate?

Solution

The bit rate can be calculated as

$$2 \times 4000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}$$

Example 1.4

What is the bit rate for high-definition TV (HDTV)?

Solution

HDTV uses digital signals to broadcast high quality video signals. The HDTV screen is normally a ratio of 16 : 9 (in contrast to 4 : 3 for regular TV), which means the screen is wider. There are 1920 by 1080 pixels per screen, and the screen is renewed 30 times per second. Twenty-four bits represents one color pixel. We can calculate the bit rate as

$$1920 \times 1080 \times 30 \times 24 = 1{,}492{,}992{,}000 \approx 1.5 \text{ Gbps}$$

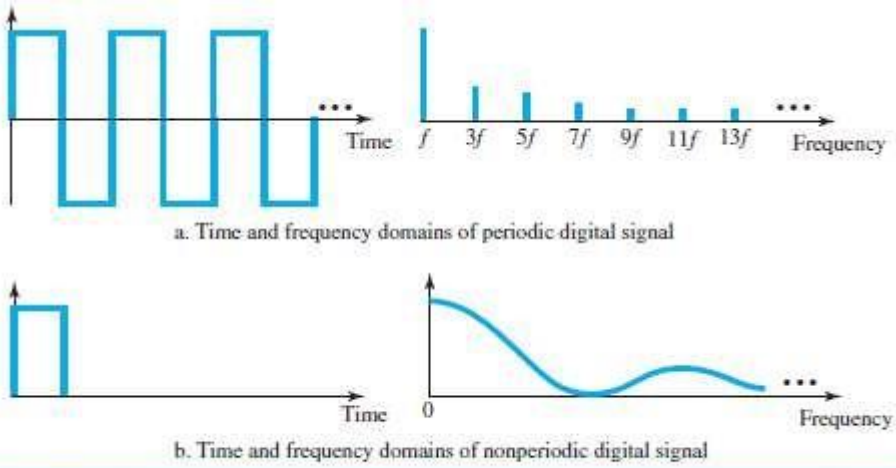
1.8.2 Bit Length

- The bit length is the distance one bit occupies on the transmission medium.

Bit length = propagation speed × bit duration

1.8.3 Digital Signal as a Composite Analog Signal

- A digital signal is a composite analog signal.
- A digital signal, in the time domain, comprises connected vertical and horizontal line segments.
  1) A vertical line in the time domain means a frequency of infinity (sudden change in time);
  2) A horizontal line in the time domain means a frequency of zero (no change in time).
- Fourier analysis can be used to decompose a digital signal.
  1) If the digital signal is periodic, the decomposed signal has a frequency domain representation with an infinite bandwidth and discrete frequencies (Figure 3.18a).
  2) If the digital signal is non-periodic, the decomposed signal has a frequency domain representation with an infinite bandwidth and continuous frequencies (Figure 3.18b).



a. Time and frequency domains of periodic digital signal

b. Time and frequency domains of nonperiodic digital signal

Figure 3.18 *The time and frequency domains of periodic and nonperiodic digital signals*

1.8.4 Transmission of Digital Signals

• Two methods for transmitting a digital signal:
1) Baseband transmission 2) Broadband transmission (using modulation).

1.8.4.1 Baseband Transmission

• Baseband transmission means sending a digital signal over a channel without changing the digital signal to an analog signal (Figure 3.19).

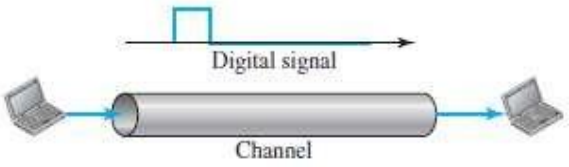


Figure 3.19 *Baseband transmission*

• Baseband transmission requires that we have a low-pass channel.
• Low-pass channel means a channel with a bandwidth that starts from zero.
• For example, we can have a dedicated medium with a bandwidth constituting only one channel.

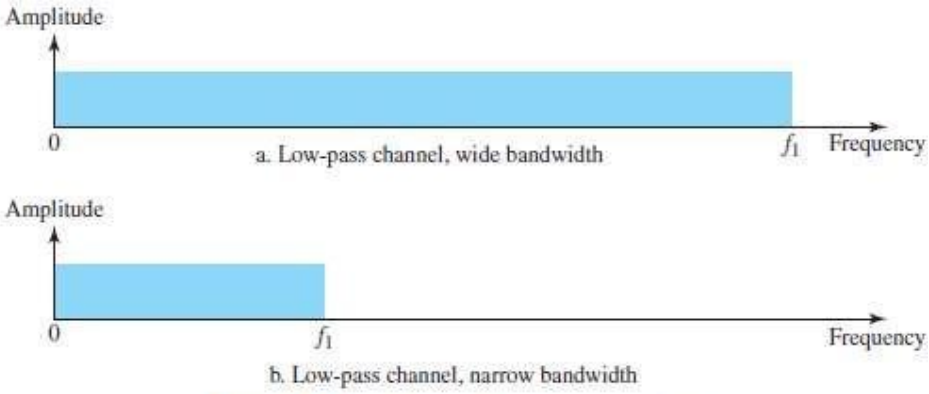


Figure 3.20 *Bandwidths of two low-pass channels*

• Two cases of a baseband communication:
Case 1: Low-pass channel with a wide bandwidth (Figure 3.20a)
Case 2: Low-pass channel with a limited bandwidth (Figure 3.20b)

Case 1: Low-Pass Channel with Wide Bandwidth

- To preserve the shape of a digital signal, we need to send the entire spectrum i.e. the continuous range of frequencies between zero and infinity.
- This is possible if we have a dedicated medium with an infinite bandwidth between the sender and receiver.
- If we have a medium with a very wide bandwidth, 2 stations can communicate by using digital signals with very good accuracy (Figure 3.21).
- Although the output signal is not an exact replica of the original signal, the data can still be deduced from the received signal.

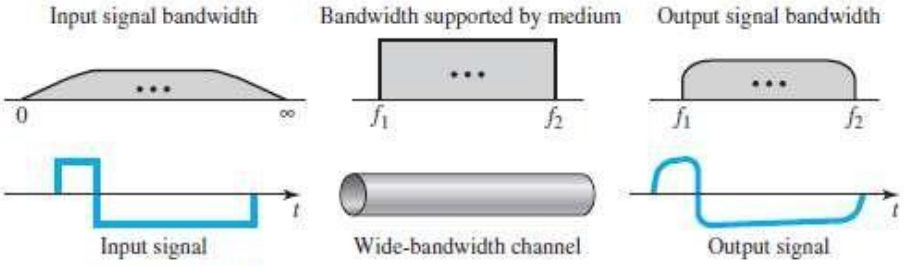


Figure 3.21 *Baseband transmission using a dedicated medium*

Case 2: Low-Pass Channel with Limited Bandwidth

- In a low-pass channel with limited bandwidth, we approximate the digital signal with an analog signal.
- The level of approximation depends on the bandwidth available.

A) Rough Approximation

- Assume that we have a digital signal of bit rate N (Figure 3.22).
- If we want to send analog signals to roughly simulate this signal, we need to consider the worst case, a maximum number of changes in the digital signal.
- This happens when the signal carries the sequence 01010101 . . or 10101010. . . .
- To simulate these two cases, we need an analog signal of frequency f = N/2.
- Let 1 be the positive peak value and 0 be the negative peak value.
- We send 2 bits in each cycle; the frequency of the analog signal is one-half of the bit rate, or N/2.
- This rough approximation is referred to as using the first harmonic (N/2) frequency. The required bandwidth is

$$\text{Bandwidth} = \frac{N}{2} - 0 = \frac{N}{2}$$

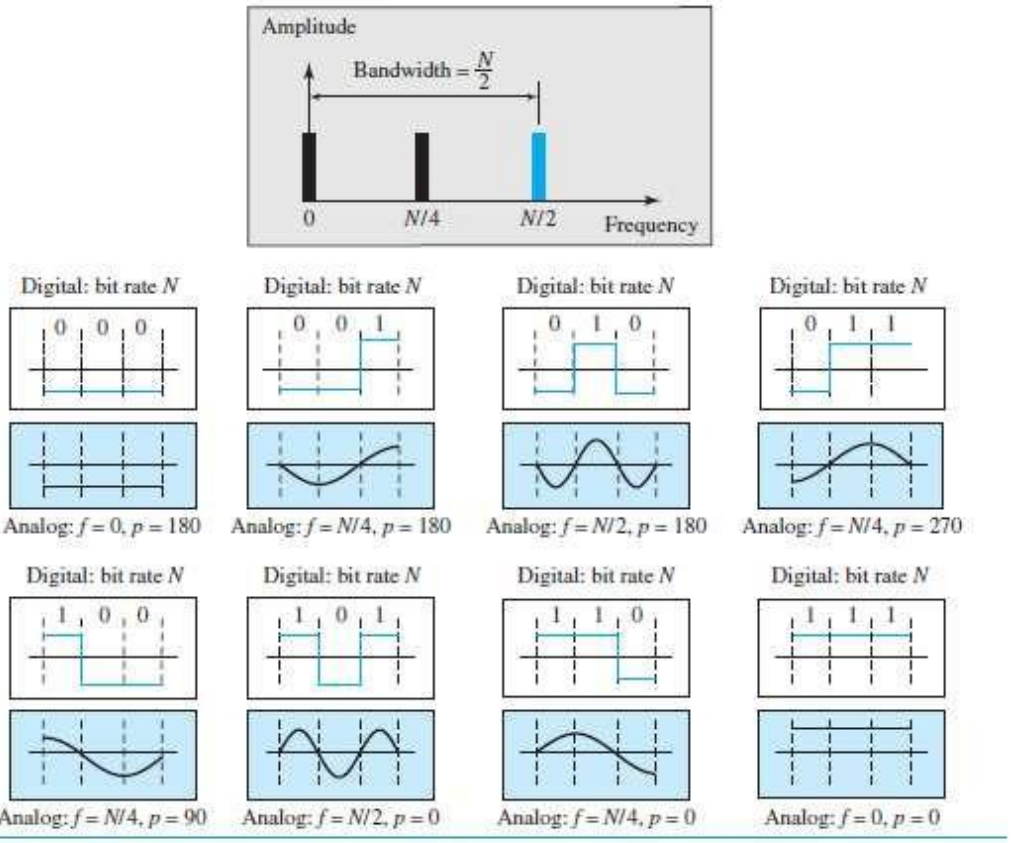


Figure 3.22 *Rough approximation of a digital signal using the first harmonic for worst case*

B) Better Approximation

- To make the shape of the analog signal look more like that of a digital signal, we need to add more harmonics of the frequencies (Figure 3.23).
- We can increase the bandwidth to 3N/2, 5N/2, 7N/2, and so on.
- In baseband transmission, the required bandwidth is proportional to the bit rate; If we need to send bits faster, we need more bandwidth.

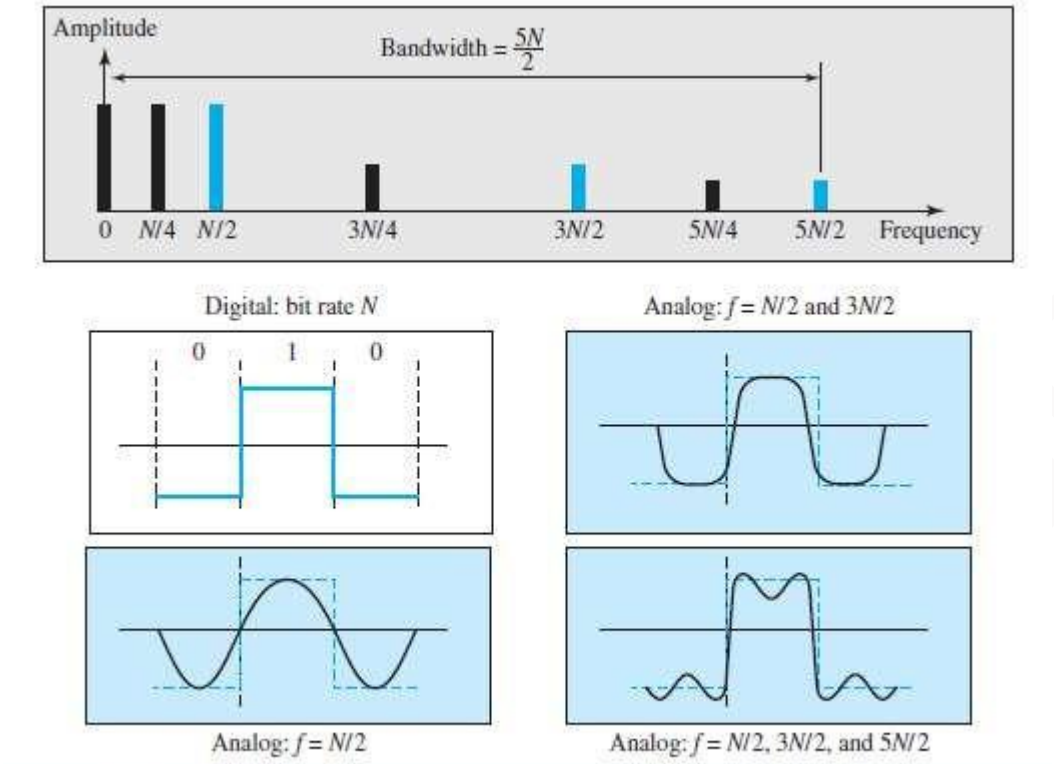


Figure 3.23 *Simulating a digital signal with first three harmonics*

Table 3.2 *Bandwidth requirements*

Bit Rate	*Harmonic 1*	*Harmonics 1, 3*	*Harmonics 1, 3, 5*
n = 1 kbps	B = 500 Hz	B = 1.5 kHz	B = 2.5 kHz
n = 10 kbps	B = 5 kHz	B = 15 kHz	B = 25 kHz
n = 100 kbps	B = 50 kHz	B = 150 kHz	B = 250 kHz

Example 1.5

What is the required bandwidth of a low-pass channel if we need to send 1 Mbps by using base-band transmission?

Solution

The answer depends on the accuracy desired.

a. The minimum bandwidth, a rough approximation, is B = bit rate /2, or 500 kHz. We need a low-pass channel with frequencies between 0 and 500 kHz.

b. A better result can be achieved by using the first and the third harmonics with the required bandwidth $B = 3 \times 500$ kHz = 1.5 MHz.

c. A still better result can be achieved by using the first, third, and fifth harmonics with $B = 5 \times 500$ kHz = 2.5 MHz.

Example 1.6

We have a low-pass channel with bandwidth 100 kHz. What is the maximum bit rate of this channel?

Solution

The maximum bit rate can be achieved if we use the first harmonic. The bit rate is 2 times the available bandwidth, or 200 kbps.

1.8.4.2 Broadband Transmission (Using Modulation)

• Broadband transmission or modulation means changing the digital signal to an analog signal for transmission.
• Modulation allows us to use a bandpass channel (Figure 3.24).
• Bandpass channel means a channel with a bandwidth that does not start from zero.
• This type of channel is more available than a low-pass channel.

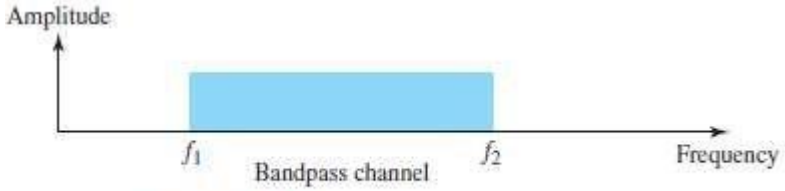


Figure 3.24 *Bandwidth of a bandpass channel*

• If the available channel is a bandpass channel,
We cannot send the digital signal directly to the channel;
We need to convert the digital signal to an analog signal before transmission (Figure 3.25).

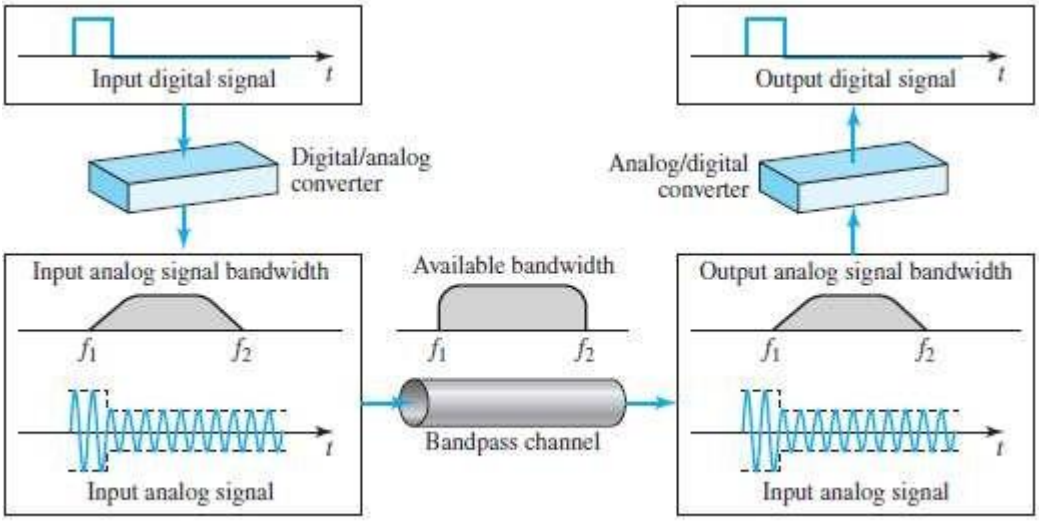


Figure 3.25 *Modulation of a digital signal for transmission on a bandpass channel*

1.9 TRANSMISSION IMPAIRMENT

- Signals travel through transmission media, which are not perfect.
- The imperfection causes signal-impairment.
- This means that signal at beginning of the medium is not the same as the signal at end of medium.
- What is sent is not what is received.
- Three causes of impairment are (Figure 3.26):
  1) Attenuation
  2) Distortion &
  3) Noise.

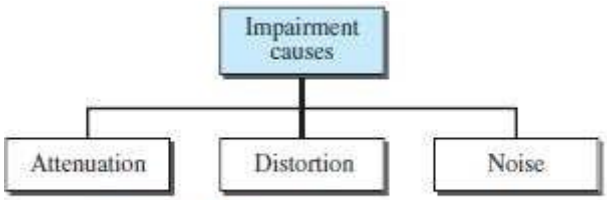


Figure 3.26 *Causes of impairment*

1.9.1 Attenuation

- As signal travels through the medium, its strength decreases as distance increases. This is called attenuation (Figure 3.27).
- As the distance increases, attenuation also increases.
- For example:
  Voice-data becomes weak over the distance & loses its contents beyond a certain distance.
- To compensate for this loss, amplifiers are used to amplify the signal.

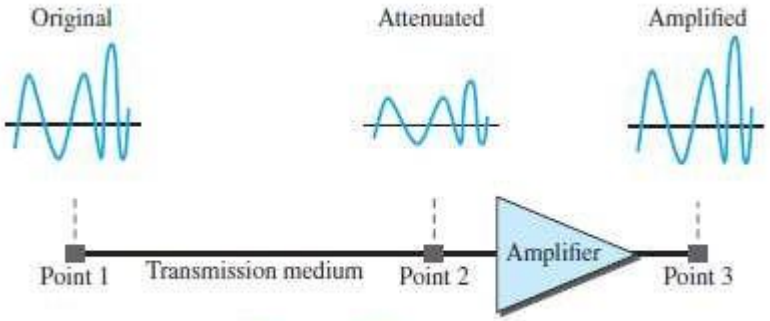


Figure 3.27 *Attenuation*

1.9.1.1 Decibel

- The decibel (dB) measures the relative strengths of
  → 2 signals or
  → one signal at 2 different points.
- The decibel is negative if a signal is attenuated.
  The decibel is positive if a signal is amplified.

$$dB = 10 \log_{10} \frac{P_2}{P_1}$$

- Variables P_1 and P_2 are the powers of a signal at points 1 and 2, respectively.
- To show that a signal has lost or gained strength, engineers use the unit of decibel.

Example 1.7

Suppose a signal travels through a transmission medium and its power is reduced to one-half. This means that $P_2 = \frac{1}{2} P_1$. In this case, the attenuation (loss of power) can be calculated as

$$10 \log_{10} \frac{P_2}{P_1} = 10 \log_{10} \frac{0.5P_1}{P_1} = 10 \log_{10} 0.5 = 10(-0.3) = -3 \text{ dB}$$

Example 1.8

A signal travels through an amplifier, and its power is increased 10 times. This means that $P_2 = 10P_1$. In this case, the amplification (gain of power) can be calculated as

$$10 \log_{10} \frac{P_2}{P_1} = 10 \log_{10} \frac{10P_1}{P_1} = 10 \log_{10} 10 = 10(1) = 10 \text{ dB}$$

Example 1.9

Sometimes the decibel is used to measure signal power in milliwatts. In this case, it is referred to as dB_m and is calculated as $dB_m = 10 \log_{10} P_m$, where P_m is the power in milliwatts. Calculate the power of a signal if its $dB_m = -30$.

Solution

We can calculate the power in the signal as

$$dB_m = 10 \log_{10} \longrightarrow dB_m = -30 \longrightarrow \log_{10} P_m = -3 \longrightarrow P_m = 10^{-3} \text{ mW}$$

Example 1.10

The loss in a cable is usually defined in decibels per kilometer (dB/km). If the signal at the beginning of a cable with –0.3 dB/km has a power of 2 mW, what is the power of the signal at 5 km?

Solution

The loss in the cable in decibels is $5 \times (-0.3) = -1.5$ dB. We can calculate the power as

$$dB = 10 \log_{10} (P_2 / P_1) = -1.5 \longrightarrow (P_2 / P_1) = 10^{-0.15} = 0.71$$

$$P_2 = 0.71P_1 = 0.7 \times 2 \text{ mW} = 1.4 \text{ mW}$$

1.9.2 Distortion

- Distortion means that the signal changes its form or shape (Figure 3.29).
- Distortion can occur in a composite signal made of different frequencies.
- Different signal-components
  - → have different propagation speed through a medium.
  - → have different delays in arriving at the final destination.
- Differences in delay create a difference in phase if delay is not same as the period-duration.
- Signal-components at the receiver have phases different from what they had at the sender.
- The shape of the composite signal is therefore not the same.

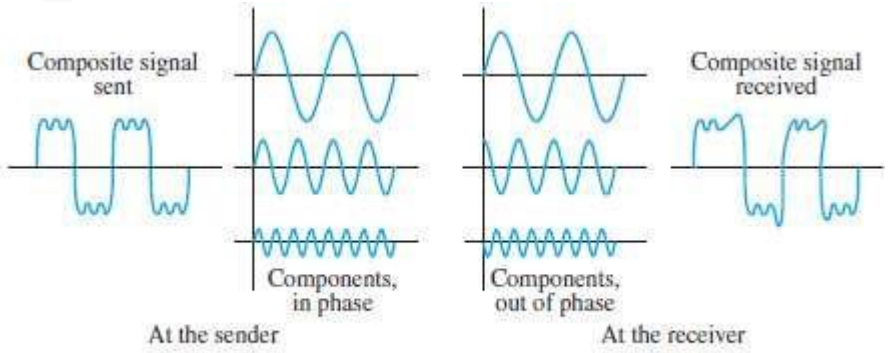


Figure 3.29 *Distortion*

1.9.3 Noise

- Noise is defined as an unwanted data (Figure 3.30).
- In other words, noise is the external energy that corrupts a signal.
- Due to noise, it is difficult to retrieve the original data/information.
- Four types of noise:

i) Thermal Noise

➢ It is random motion of electrons in wire which creates extra signal not originally sent by transmitter.

ii) Induced Noise

➢ Induced noise comes from sources such as motors & appliances.
➢ These devices act as a sending-antenna.
The transmission-medium acts as the receiving-antenna.

iii) Crosstalk

➢ Crosstalk is the effect of one wire on the other.
➢ One wire acts as a sending-antenna and the other as the receiving-antenna.

iv) Impulse Noise

➢ Impulse Noise is a spike that comes from power-lines, lightning, and so on.
(spike → a signal with high energy in a very short time)

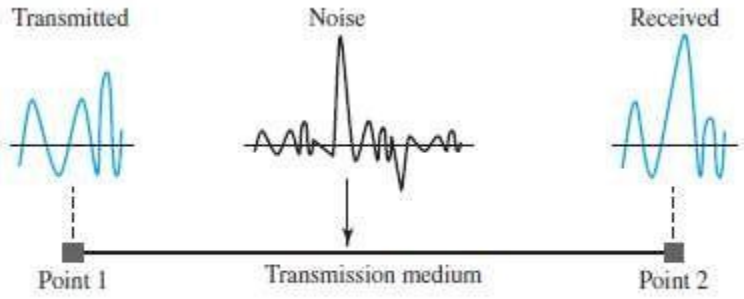


Figure 3.30 *Noise*

1.9.3.1 Signal-to-Noise Ratio (SNR)

- SNR is used to find the theoretical bit-rate limit.
- SNR is defined as

$$SNR = \frac{\text{average signal power}}{\text{average noise power}}$$

- SNR is actually the ratio of what is wanted (signal) to what is not wanted (noise).
- A high-SNR means the signal is less corrupted by noise.
A low-SNR means the signal is more corrupted by noise.
- Because SNR is the ratio of 2 powers, it is often described in decibel units, SNR_{dB}, defined as

$$SNR_{dB} = 10\log_{10} SNR$$

Example 1.11

The power of a signal is 10 mW and the power of the noise is 1 μW; what are the values of SNR and SNR_{dB}?

Solution

The values of SNR and SNR_{dB} can be calculated as follows:

$$SNR = (10{,}000\ \mu w) / (1\ \mu w) = 10{,}000 \qquad SNR_{dB} = 10 \log_{10} 10{,}000 = 10 \log_{10} 10^4 = 40$$

1.10 DATA RATE LIMITS

- Data-rate depends on 3 factors:
  1) Bandwidth available
  2) Level of the signals
  3) Quality of channel (the level of noise)
- Two theoretical formulas can be used to calculate the data-rate:
  1) Nyquist for a noiseless channel and
  2) Shannon for a noisy channel.

1.10.1 Noiseless Channel: Nyquist Bit Rate

- For a noiseless channel, the Nyquist bit-rate formula defines the theoretical maximum bit-rate

$$\text{Bitrate} = 2 \times \text{Bandwidth} \times \log_2 L$$

where bandwidth = bandwidth of the channel
L = number of signal-levels used to represent data
BitRate = bitrate of channel in bps

- According to the formula,
  - By increasing number of signal-levels, we can increase the bit-rate.
  - Although the idea is theoretically correct, practically there is a limit.
  - When we increase the number of signal-levels, we impose a burden on the receiver.
  - If no. of levels in a signal is 2, the receiver can easily distinguish b/w 0 and 1.
  - If no. of levels is 64, the receiver must be very sophisticated to distinguish b/w 64 different levels.
  - In other words, increasing the levels of a signal reduces the reliability of the system.

Example 1.12

Consider a noiseless channel with a bandwidth of 3000 Hz transmitting a signal with two signal levels. The maximum bit rate can be calculated as

$$\text{BitRate} = 2 \times 3000 \times \log_2 2 = 6000 \text{ bps}$$

Example 1.13

Consider the same noiseless channel transmitting a signal with four signal levels (for each level, we send 2 bits). The maximum bit rate can be calculated as

$$\text{BitRate} = 2 \times 3000 \times \log_2 4 = 12{,}000 \text{ bps}$$

Example 1.14

We need to send 265 kbps over a noiseless channel with a bandwidth of 20 kHz. How many signal levels do we need?

Solution

We can use the Nyquist formula as shown:

$$265{,}000 = 2 \times 20{,}000 \times \log_2 L \longrightarrow \log_2 L = 6.625 \longrightarrow L = 2^{6.625} = 98.7 \text{ levels}$$

1.10.2 Noisy Channel: Shannon Capacity

- In reality, we cannot have a noiseless channel; the channel is always noisy.
- For a noisy channel, the Shannon capacity formula defines the theoretical maximum bit-rate.

$$\text{Capacity} = \text{bandwidth} \times \log_2(1 + \text{SNR})$$

where bandwidth = bandwidth of channel in bps.
SNR = signal-to-noise ratio and
Capacity = capacity of channel in bps.

- This formula does not consider the no. of levels of signals being transmitted (as done in the Nyquist bit rate).

  This means that no matter how many levels we have, we cannot achieve a data-rate higher than the capacity of the channel.

- In other words, the formula defines a characteristic of the channel, not the method of transmission.

Example 1.15

We can calculate the theoretical highest bit rate of a regular telephone line. A telephone line normally has a bandwidth of 3000 Hz (300 to 3300 Hz) assigned for data communications. The signal-to-noise ratio is usually 3162. For this channel the capacity is calculated as

$$C = B \log_2(1 + \text{SNR}) = 3000 \log_2(1 + 3162) = 3000 \times 11.62 = 34{,}860 \text{ bps}$$

Example 1.16

The signal-to-noise ratio is often given in decibels. Assume that $\text{SNR}_{\text{dB}} = 36$ and the channel bandwidth is 2 MHz. The theoretical channel capacity can be calculated as

$$\text{SNR}_{\text{dB}} = 10 \log_{10} \text{SNR} \longrightarrow \text{SNR} = 10^{\text{SNR}_{\text{dB}}/10} \longrightarrow \text{SNR} = 10^{3.6} = 3981$$

$$C = B \log_2(1 + \text{SNR}) = 2 \times 10^6 \times \log_2 3982 = 24 \text{ Mbps}$$

Example 1.17

We have a channel with a 1-MHz bandwidth. The SNR for this channel is 63. What are the appropriate bit rate and signal level?

Solution

First, we use the Shannon formula to find the upper limit.

$$C = B \log_2(1 + \text{SNR}) = 10^6 \log_2(1 + 63) = 10^6 \log_2 64 = 6 \text{ Mbps}$$

The Shannon formula gives us 6 Mbps, the upper limit. For better performance we choose something lower, 4 Mbps, for example. Then we use the Nyquist formula to find the number of signal levels.

$$4 \text{ Mbps} = 2 \times 1 \text{ MHz} \times \log_2 L \longrightarrow L = 4$$

1.11 PERFORMANCE

1.11.1 Bandwidth

- One characteristic that measures network-performance is bandwidth.
- Bandwidth of analog and digital signals is calculated in separate ways:

(1) Bandwidth of an Analog Signal (in hz)

- Bandwidth of an analog signal is expressed in terms of its frequencies.
- Bandwidth is defined as the range of frequencies that the channel can carry.
- It is calculated by the difference b/w the maximum frequency and the minimum frequency.

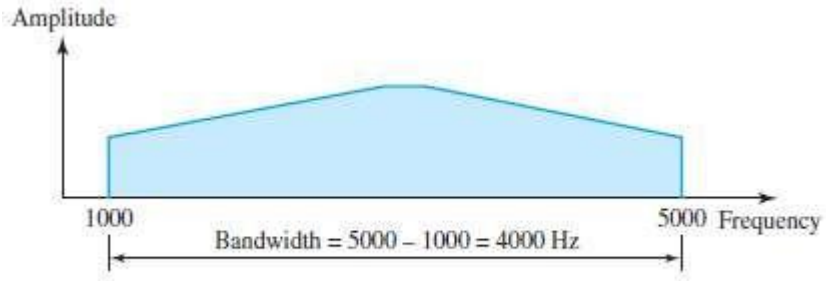


Figure 3.13 *The bandwidth of signals*

In figure 3.13, the signal has a minimum frequency of F1 = 1000Hz and maximum frequency of F2 = 5000Hz.

Hence, the bandwidth is given by F2 – F1= 5000 - 1000 = 4000 Hz

(2) Bandwidth of a Digital Signal (in bps)

- Bandwidth refers to the number of bits transmitted in one second in a channel (or link).
- For example:

The bandwidth of a Fast Ethernet is a maximum of 100 Mbps. (This means that this network can send 100 Mbps).

Relationship between (1) and (2)

- There is an explicit relationship between the bandwidth in hertz and bandwidth in bits per seconds.
- Basically, an increase in bandwidth in hertz means an increase in bandwidth in bits per second.
- The relationship depends on
  → baseband transmission or
  → transmission with modulation.

1.11.2 Throughput

- The throughput is a measure of how fast we can actually send data through a network.
- Although, bandwidth in bits per second and throughput seem the same, they are actually different.
- A link may have a bandwidth of B bps, but we can only send T bps through this link with T always less than B.
- In other words,
    1) The bandwidth is a potential measurement of a link.
    2) The throughput is an actual measurement of how fast we can send data.
        For example:
        - ¤ We may have a link with a bandwidth of 1 Mbps, but the devices connected to the end of the link may handle only 200 kbps.
        - ¤ This means that we cannot send more than 200 kbps through this link.

Example 1.18

A network with bandwidth of 10 Mbps can pass only an average of 12,000 frames per minute with each frame carrying an average of 10,000 bits. What is the throughput of this network?

Solution

We can calculate the throughput as

$$\text{Throughput} = (12{,}000 \times 10{,}000) / 60 = 2 \text{ Mbps}$$

1.11.3 Latency (Delay)

• The latency defines how long it takes for an entire message to completely arrive at the destination from the time the first bit is sent out from the source.

Latency = propagation time + transmission time + queuing time + processing delay

1) Propagation Time

- Propagation time is defined as the time required for a bit to travel from source to destination.
- Propagation time is given by

Propagation time = Distance / (Propagation Speed)

- Propagation speed of electromagnetic signals depends on
  - → medium and
  - → frequency of the signal.

Example 1.19

propagation speed to be 2.4×10^8 m/s in cable.

Solution

We can calculate the propagation time as

$$\text{Propagation time} = (12{,}000 \times 10{,}000) / (2.4 \times 2^8) = 50 \text{ ms}$$

b

2) Transmission Time

- The time required for transmission of a message depends on
  - → size of the message and
  - → bandwidth of the channel.
- The transmission time is given by

Transmission time = (Message size) / Bandwidth

Example 1.20

What are the propagation time and the transmission time for a 2.5-KB (kilobyte) message (an e-mail) if the bandwidth of the network is 1 Gbps? Assume that the distance between the sender and the receiver is 12,000 km and that light travels at 2.4×10^8 m/s.

Solution

We can calculate the propagation and transmission time as

$$\text{Propagation time} = (12{,}000 \times 1000) / (2.4 \times 10^8) = 50 \text{ ms}$$

$$\text{Transmission time} = (2500 \times 8) / 10^9 = 0.020 \text{ ms}$$

c

Example 1.21

What are the propagation time and the transmission time for a 5-MB (megabyte) message (an image) if the bandwidth of the network is 1 Mbps? Assume that the distance between the sender and the receiver is 12,000 km and that light travels at 2.4×10^8 m/s.

Solution

We can calculate the propagation and transmission times as

$$\text{Propagation time} = (12{,}000 \times 1000) / (2.4 \times 10^8) = 50 \text{ ms}$$

$$\text{Transmission time} = (5{,}000{,}000 \times 8) / 10^6 = 40 \text{ s}$$

3) Queuing Time

- Queuing-time is the time needed for each intermediate-device to hold the message before it can be processed.
  (Intermediate device may be a router or a switch)
- The queuing-time is not a fixed factor. This is because
  i) Queuing-time changes with the load imposed on the network.
  ii) When there is heavy traffic on the network, the queuing-time increases.
- An intermediate-device
  → queues the arrived messages and
  → processes the messages one by one.
- If there are many messages, each message will have to wait.

4) Processing Delay

- Processing delay is the time taken by the routers to process the packet header.

1.11.4 Bandwidth Delay Product

- Two performance-metrics of a link are 1) Bandwidth and 2) Delay
- The bandwidth-delay product is very important in data-communications.
- Let us elaborate on this issue, using 2 hypothetical cases as examples.

Case 1: The following figure shows case 1 (Figure 3.32).

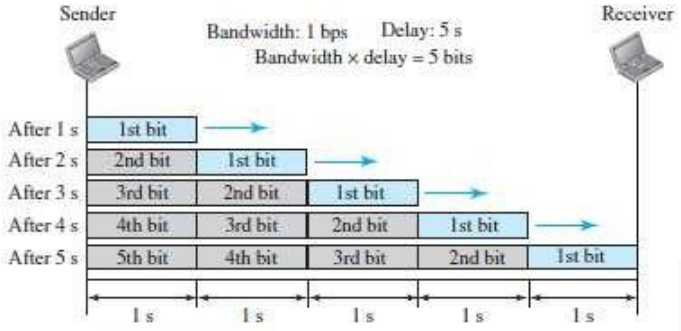


Figure 3.32 *Filling the link with bits for case 1*

- Let us assume,
  Bandwidth of the link = 1 bps Delay of the link = 5s.
- From the figure 3.32, bandwidth-delay product is 1 x 5 = 5. Thus, there can be maximum 5 bits on the line.
- There can be no more than 5 bits at any time on the link.

Case 2: The following figure shows case 2 (Figure 3.33).

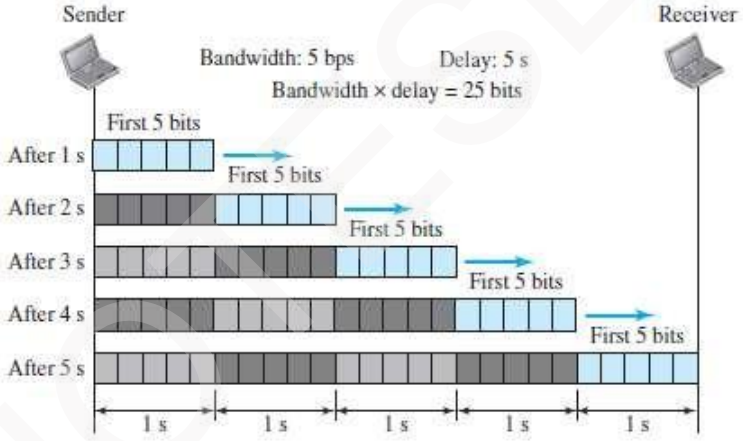


Figure 3.33 *Filling the link with bits in case 2*

- Let us assume,
  Bandwidth of the link = 4 bps Delay of the link = 5s.
- From the figure 3.33, bandwidth-delay product is 5 x 5 = 25. Thus, there can be maximum 25 bits on the line.
- At each second, there are 5 bits on the line, thus the duration of each bit is 0.20s.

- The above 2 cases show that the (bandwidth X delay) is the number of bits that can fill the link.
- This measurement is important if we need to
  → send data in bursts and
  → wait for the acknowledgment of each burst.
- To use the maximum capability of the link
  → We need to make the burst-size as (2 x bandwidth x delay).
  → We need to fill up the full-duplex channel (two directions).
- Amount (2x bandwidth x delay) is the number of bits that can be in transition at any time (Fig 3.34).

Figure 3.34 *Concept of bandwidth-delay product*

1.11.5 Jitter

- Another performance issue that is related to delay is jitter.
- We can say that jitter is a problem
  → if different packets of data encounter different delays and
  → if the application using the data at the receiver site is time-sensitive (for ex: audio/video).
- For example:
  If the delay for the first packet is 20 ms
  the delay for the second is 45 ms and
  the delay for the third is 40 ms
  then the real-time application that uses the packets suffers from jitter.

MODULE 1(CONT.): DIGITAL TRANSMISSION

1.12 DIGITAL TO DIGITAL CONVERSION

- Data can be analog or digital, so can be the signal that represents it.
- Signal encoding is the conversion from analog/digital data to analog/digital signal.
- The possible encodings are:
    1) Digital data to digital signal
    2) Digital data to analog signal
    3) Analog data to digital signal
    4) Analog data to analog signal

1.12.1 LINE CODING

- Line-coding is the process of converting digital-data to digital-signals (Figure 4.1).
- The data may be in the form of text, numbers, graphical images, audio, or video
- The data are stored in computer memory as sequences of bits (0s or 1s).
- Line-coding converts a sequence of bits to a digital-signal.
- At the sender, digital-data is encoded into a digital-signal.
  At the receiver, digital-signal is decoded into a digital-data.

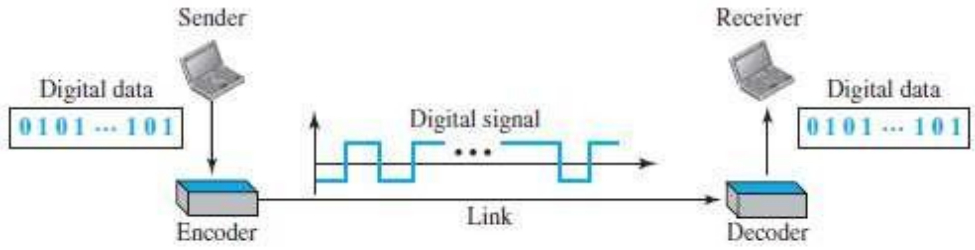


Figure 4.1 *Line coding and decoding*

1.12.1.1 Characteristics

- Different characteristics of digital signal are
  - **1)** Signal Element Vs Data Element
  - **2)** Data Rate Vs Signal Rate
  - **3)** Bandwidth
  - **4)** Baseline Wandering
  - **5)** DC Components
  - **6)** Built-in Error Detection
  - **7)** Self-synchronization
  - **8)** Immunity to Noise and Interference
  - **9)** Complexity

1) Data Element vs. Signal Element

Data Element	Signal Element
A data-element is the smallest entity that can represent a piece of information (Figure 4.2).	A signal-element is shortest unit (timewise) of a digital-signal.
A data-element is the bit.	A signal-element carries data-elements.
Data-elements are being carried.	Signal-elements are the carriers.

➢ Ratio r is defined as number of data-elements carried by each signal-element.

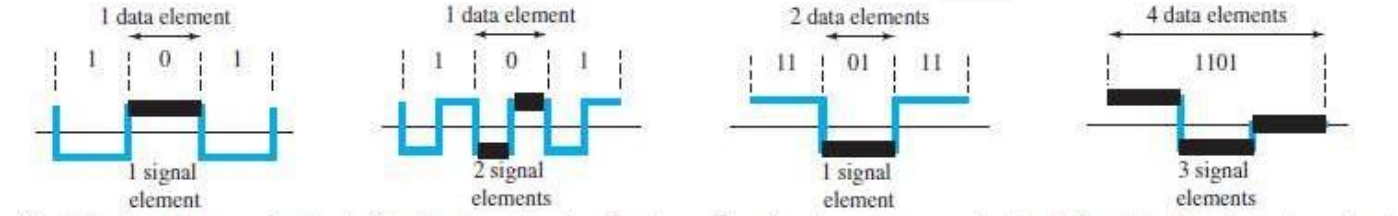


Figure 4.2 *Signal element versus data element*

2) Data Rate vs. Signal Rate

Data Rate	Signal Rate
The data-rate defines the number of data-elements (bits) sent in 1 sec.	The signal-rate is the number of signal-elements sent in 1 sec.
The unit is bits per second (bps).	The unit is the baud.
The data-rate is sometimes called the bit-rate.	The signal-rate is sometimes called the pulse rate, the modulation rate, or the baud rate
Goal in data-communications: increase the data-rate.	Goal in data-communications: decrease the signal-rate.
Increasing the data-rate increases the speed of transmission.	Decreasing the signal-rate decreases the bandwidth requirement.

➢ The relationship between data-rate and signal-rate is given by

$$S_{ave} = c \times N \times (1/r) \quad \text{baud}$$

where N = data-rate (in bps)
c = case factor, which varies for each case
S = number of signal-elements and
r = previously defined factor.

➢ This relationship depends on
→ value of r.
→ data pattern.

(If we have a data pattern of all 1s or all 0s, the signal-rate may be different from a data pattern of alternating Os and 1s).

3) Bandwidth

- Digital signal that carries information is non-periodic.
- The bandwidth of a non-periodic signal is continuous with an infinite range.
- However, most digital-signals we encounter in real life have a bandwidth with finite values.
- The effective bandwidth is finite.
- The baud rate, not the bit-rate, determines the required bandwidth for a digital-signal.
- More changes in the signal mean injecting more frequencies into the signal.
  (Frequency means change and change means frequency.)
- The bandwidth refers to range of frequencies used for transmitting a signal.
- Relationship b/w baud rate (signal-rate) and the bandwidth (range of frequencies) is given as

$$B_{min} = c \times N \times (1/r)$$

where N = data-rate (in bps)
c = case factor, which varies for each case
r = previously defined factor
B_{min} = minimum bandwidth

4) Baseline Wandering

- While decoding, the receiver calculates a running-average of the received signal-power. This average is called the baseline.
- The incoming signal-power is estimated against this baseline to determine the value of the data-element.
- A long string of 0s or 1s can cause a drift in the baseline (baseline wandering).
  Thus, make it difficult for the receiver to decode correctly.
- A good line-coding scheme needs to prevent baseline wandering.

5) DC Components

- When the voltage-level in a digital-signal is constant for a while, the spectrum creates very low frequencies.
- These frequencies around zero are called DC (direct-current) components.
- DC components present problems for a system that cannot pass low frequencies.
- For example: Telephone line cannot pass frequencies below 200 Hz.
- For Telephone systems, we need a scheme with no DC component.

6) Built-in Error Detection

- Built-in error-detecting capability has to be provided to detect the errors that occurred during transmission.

7) Self Synchronization

- To correctly interpret the signals received from the sender, the receiver's bit intervals must correspond exactly to the sender's bit intervals.
- If the receiver clock is faster or slower, the bit intervals are not matched and the receiver might misinterpret the signals.

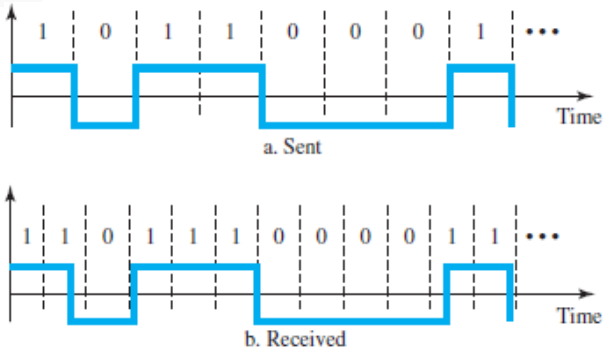


Figure 4.3 *Effect of lack of synchronization*

- As shown in figure 4.3, we have a situation where the receiver has shorter bit duration.
- The sender sends 10110001, while the receiver receives 110111000011.
- A self-synchronizing digital-signal includes timing-information in the data being transmitted.
  - This can be achieved if there are transitions in the signal that alert the receiver to the beginning, middle, or end of the pulse.
  - If the receiver's clock is out-of-synchronization, these points can reset the clock.

8) Immunity to Noise & Interference

- The code should be immune to noise and other interferences.

9) Complexity

- A complex scheme is more costly to implement than a simple one.
- For ex: A scheme that uses 4 signal-levels is more difficult to interpret than one that uses only 2 levels.

Example 1.22

A signal is carrying data in which one data element is encoded as one signal element ($r = 1$). If the bit rate is 100 kbps, what is the average value of the baud rate if c is between 0 and 1?

Solution

We assume that the average value of c is 1/2. The baud rate is then

$$S = c \times N \times (1 / r) = 1/2 \times 100{,}000 \times (1/1) = 50{,}000 = 50 \text{ kbaud}$$

Example 1.23

In a digital transmission, the receiver clock is 0.1 percent faster than the sender clock. How many extra bits per second does the receiver receive if the data rate is 1 kbps? How many if the data rate is 1 Mbps?

Solution

At 1 kbps, the receiver receives 1001 bps instead of 1000 bps.

1000 bits sent → 1001 bits received → 1 extra bps

At 1 Mbps, the receiver receives 1,001,000 bps instead of 1,000,000 bps.

1,000,000 bits sent → 1,001,000 bits received → 1000 extra bps

1.12.2 LINE CODING SCHEMES

• The Line Coding schemes are classified into 3 broad categories (Figure 4.4):

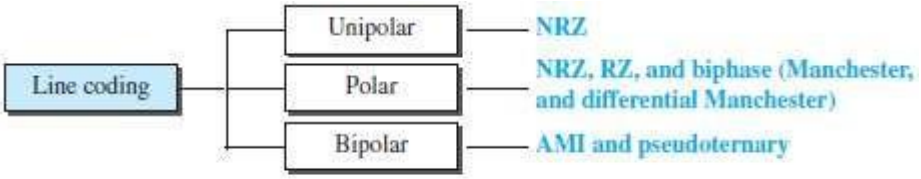


Figure 4.4 *Line coding schemes*

1.12.2.1 Unipolar Scheme

• All signal levels are either above or below the time axis.

NRZ (Non-Return-to-Zero)

- The positive voltage defines bit 1 and the zero voltage defines bit 0 (Figure 4.5).
- It is called NRZ because the signal does not return to 0 at the middle of the bit.

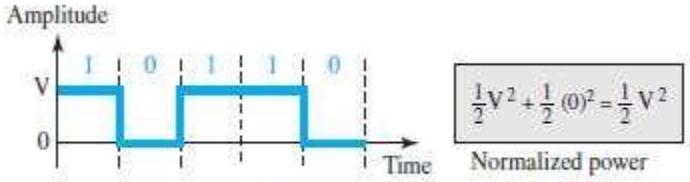


Figure 4.5 *Unipolar NRZ scheme*

- Disadvantages:
  1) Compared to polar scheme, this scheme is very costly.
  2) Also, the normalized power is double that for polar NRZ.
  3) Not suitable for transmission over channels with poor performance around zero frequency.
  (Normalized power → power needed to send 1 bit per unit line resistance)

1.12.2.2 Polar Schemes

- The voltages are on the both sides of the time axis.
- Polar NRZ scheme can be implemented with two voltages (V).

For example: -V for bit 1
+V for bit 0.

a) Non-Return-to-Zero (NRZ)

- We use 2 levels of voltage amplitude.
- Two versions of polar NRZ (Figure 4.6):

i) NRZ-L (NRZ-Level)

¤ The level of the voltage determines the value of the bit.
¤ For example: i) Voltage-level for 0 can be positive and
ii) Voltage-level for 1 can be negative.

ii) NRZ-I (NRZ-Invert)

¤ The change or lack of change in the level of the voltage determines the value of the bit.
¤ If there is no change, the bit is 0;
If there is a change, the bit is 1.

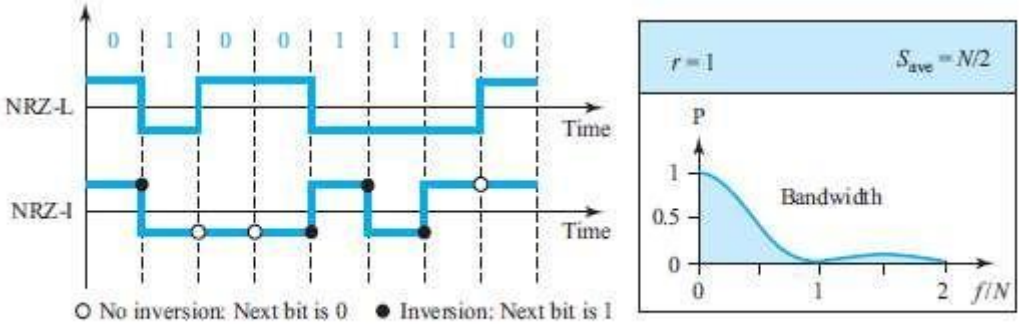


Figure 4.6 *Polar NRZ-L and NRZ-I schemes*

- Disadvantages:

1) **Baseline wandering** is a problem for both variations (NRZ-L NRZ-I).
i) In NRZ-L, if there is a long sequence of 0s or 1s, the average signal-power becomes skewed.
The receiver might have difficulty discerning the bit value.
ii) In NRZ-I, this problem occurs only for a long sequence of 0s.
If we eliminate the long sequence of 0s, we can avoid baseline wandering.
2) The **synchronization problem** also exists in both schemes.
→ A long sequence of 0s can cause a problem in both schemes.
→ A long sequence of 1s can cause a problem in only NRZ-L.
3) In NRZ-L, problem occurs when there is a sudden **change of polarity** in the system.
¤ For example:
In twisted-pair cable, a change in the polarity of the wire results in
→ all 0s interpreted as 1s and
→ all 1s interpreted as 0s.
¤ NRZ-I does not have this problem.
¤ Both schemes have an average signal-rate of N/2 Bd.
4) NRZ-L and NRZ-I both have a **DC component problem**.

Example 1.24

A system is using NRZ-I to transfer 10-Mbps data. What are the average signal rate and minimum bandwidth?

Solution

The average signal rate is $S = N/2 = 500$ kbaud. The minimum bandwidth for this average baud rate is $B_{min} = S = 500$ kHz.

b) Return-to-Zero (RZ)

- In NRZ encoding, problem occurs when the sender-clock and receiver-clock are not synchronized.
- Solution: Use return-to-zero (RZ) scheme (Figure 4.7).
- RZ scheme uses 3 voltages: positive, negative, and zero.
- There is always a transition at the middle of the bit. Either
  - i) from high to zero (for 1) or
  - ii) from low to zero (for 0)

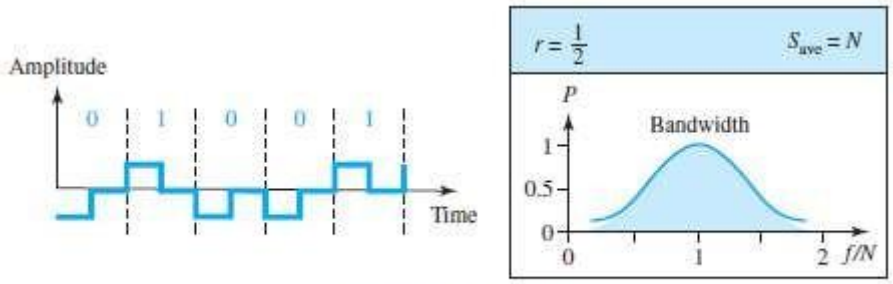


Figure 4.7 *Polar RZ scheme*

- Disadvantages:
  1) RZ encoding requires 2 signal-changes to encode a bit & .'. occupies greater bandwidth.
  2) Complexity: RZ uses 3 levels of voltage, which is more complex to create and detect.
  3) Problem occurs when there is a sudden change of polarity in the system. This result in
     → all 0s interpreted as 1s &
     → all 1s interpreted as 0s.

c) Biphase: Manchester & Differential Manchester

i) Manchester Encoding

- This is a combination of NRZ-L & RZ schemes (RZ→transition at the middle of the bit).
- There is always a transition at the middle of the bit. Either
  - i) from high to low (for 0) or
  - ii) from low to high (for 1).
- It uses only two voltage levels (Figure 4.8).
- The duration of the bit is divided into 2 halves.
- The voltage
  - → remains at one level during the first half &
  - → moves to the other level in the second half.
- The transition at the middle of the bit provides synchronization.

ii) Differential Manchester

- This is a combination of NRZ-I and RZ schemes.
- There is always a transition at the middle of the bit, but the bit-values are determined at the beginning of the bit.
- If the next bit is 0, there is a transition. If the next bit is 1, there is none.

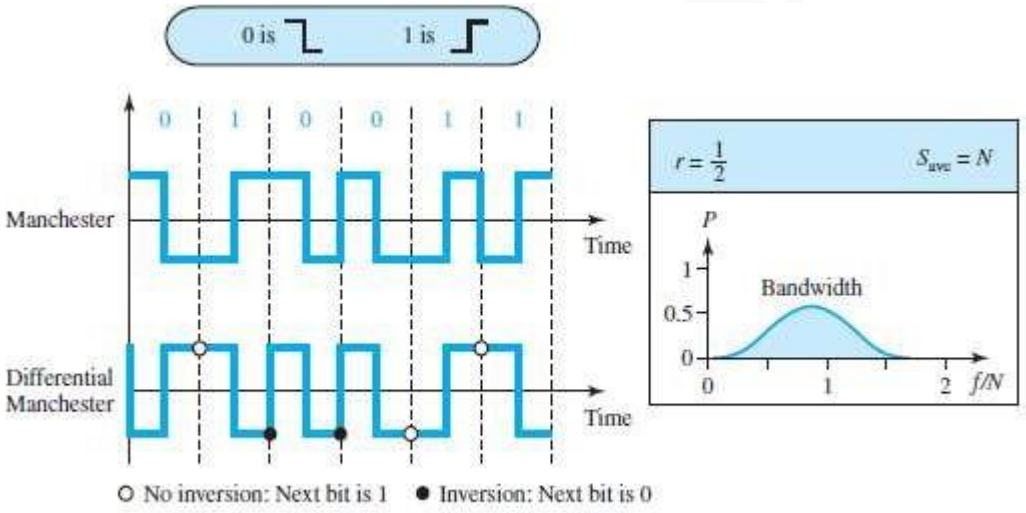


Figure 4.8 *Polar biphase: Manchester and differential Manchester schemes*

- Advantages:
  1) The Manchester scheme overcomes problems associated with NRZ-L.
     Differential Manchester overcomes problems associated with NRZ-I.
  2) There is no baseline wandering.
  3) There is no DC component '.' each bit has a positive & negative voltage contribution.
- Disadvantage:
  1) Signal-rate: Signal-rate for Manchester & diff. Manchester is double that for NRZ.

1.12.2.3 Bipolar Schemes (or Multilevel Binary)

- This coding scheme uses 3 voltage levels (Figure 4.9):
  - i) positive
  - ii) negative &
  - iii) zero.
- Two variations of bipolar encoding:
  - i) AMI (Alternate Mark Inversion)
  - ii) Pseudoternary

i) AMI

- Binary 0 is represented by a neutral 0 voltage (AMI → Alternate 1 Inversion).
- Binary 1s are represented by alternating positive and negative voltages.

ii) Pseudoternary

- Binary 1 is represented by a neutral 0 voltage.
- Binary 0s are represented by alternating positive and negative voltages.

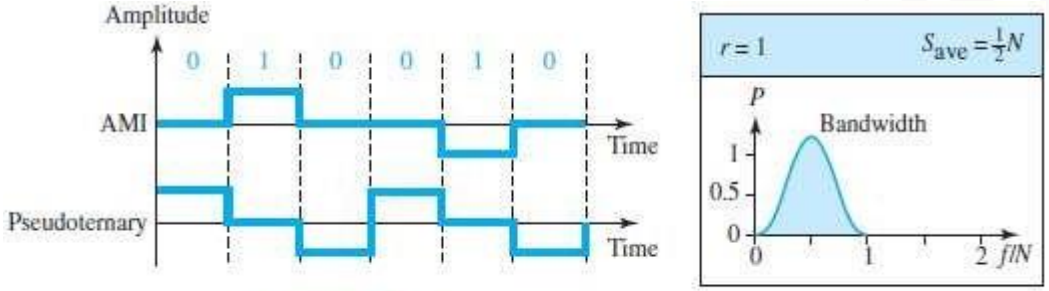


Figure 4.9 *Bipolar schemes: AMI and pseudoternary*

- Advantages:
  1) The bipolar scheme has the same signal-rate as NRZ.
  2) There is no DC component '.' each bit has a positive & negative voltage contribution.
  3) The concentration of the energy is around frequency N/2.
- Disadvantage:
  1) AMI has a synchronization problem when a long sequence of 0s is present in the data.

Table 4.1 *Summary of line coding schemes*

Category	*Scheme*	*Bandwidth (average)*	*Characteristics*
Unipolar	NRZ	$B = N/2$	Costly, no self-synchronization if long 0s or 1s, DC
Polar	NRZ-L	$B = N/2$	No self-synchronization if long 0s or 1s, DC
	NRZ-I	$B = N/2$	No self-synchronization for long 0s, DC
	Biphase	$B = N$	Self-synchronization, no DC, high bandwidth
Bipolar	AMI	$B = N/2$	No self-synchronization for long 0s, DC

QUESTIONS

MODULE 1: INTRODUCTION

10) Define data communications. Explain its 4 fundamental characteristics. (4*)
11) Explain different components of data communication system. (6*)
12) Explain different forms of information. (4)
13) Describe simplex, half-duplex and full duplex methods of data flow. (6*)
14) Explain the 3 criteria necessary for an effective and efficient network. (4*)
15) Explain point to point and multipoint connection. (4*)
16) Explain the following topologies:
i) Mesh ii) Star iii) Bus iv) Ring (12*)
17) Explain in detail LAN & WAN. List the differences between LAN & WAN. (10*)
18) Explain circuit-switched and packet-switched networks. (6*)

MODULE 1(CONT.): NETWORK MODELS

6) Explain TCP/IP architecture with a layer diagram. (4*)
7) List the 5 layers and its functionality in TCP/IP model. (8*)
8) With respect to in TCP/IP model, explain the following:
i) Encapsulation and decapsulation. ii) Multiplexing and demultiplexing. (8)
9) Explain four levels of addressing employed in TCP/IP protocol. (6*)
10) What are the uses of a layered network model? Compare OSI and TCP/IP models. (4)

MODULE 1(CONT.): DATA AND SIGNALS

7) Compare the following:
i) Analog signal vs. Digital signal. ii) Periodic signal vs. Non-periodic signal. (4)
8) Describe digital signal as a composite analog signal. (4)
9) Explain 2 methods for transmitting a digital signal (8*)
10) What do you mean by transmission impairment? Explain causes of transmission impairment? (6*)
11) What are the three factors data rate is dependent on? Explain the theoretical formula which wasdeveloped to calculate the data rate. (8*)
12) Explain 4 performance parameters of network. (8*)

MODULE 1(CONT.): DIGITAL TRANSMISSION

6) Explain in detail any 6 characteristics of digital signal. (6*)
7) Compare the following:
i) Data element vs. Signal element. ii) Data rate vs. Signal rate. (4)
8) Explain following encoding schemes with example:
i) Unipolar Scheme ii) Polar Schemes iii) Bipolar Schemes (8*)
9) Represent the following sequences using different line coding schemes.
i) 101011100. ii) 10110011. iii) 00110101. (6*)
10) Define the following:
i) Network ii) Internet
ii)iii) Protocol iv) Decibel v) SNR vi) Line coding (6*)

MODULE 1: DIGITAL TRANSMISSION (CONT.)

6) Explain the PCM encoder with neat diagram. (8*)
7) What do you mean by Sampling? Explain three sampling methods with a neat diagram. (4)
8) Explain non-uniform quantization and how to recover original signal using PCM decoder. (4)
9) Explain different types of transmission modes. (8*)
10) What is sampling and quantization? Explain briefly. (6)

MODULE 1(CONT.): ANALOG TRANSMISSION

10) Define digital to analog conversion? List different types of digital to analog conversion. (2)
11) Describe ASK, FSK and PSK mechanisms and apply them over the digital data 101101. (4)
12) Discuss the bandwidth requirement for ASK, FSK and PSK. (4*)
13) Explain different aspects of digital-to-analog conversion? (6*)
14) Define ASK. Explain BASK. (6*)
15) Define FSK. Explain BFSK. (6*)
16) Define PSK. Explain BPSK. (6*)
17) Explain QPSK. (6)
18) Explain the concept of constellation diagram. (6)
10)Explain QAM. (6)

MODULE 1: DIGITAL TRANSMISSION (CONT.)

2.1 ANALOG TO DIGITAL CONVERSION

- An analog-signal may created by a microphone or camera.
- To change an analog-signal to digital-data, we use PCM (pulse code modulation).
- After the digital-data are created (digitization), then we convert the digital-data to a digital-signal.

2.1.1 PCM

- PCM is a technique used to change an analog signal to digital data (digitization).
- PCM has encoder at the sender and decoder at the receiver.
- The encoder has 3 processes (Figure 4.21):
  1) Sampling
  2) Quantization &
  3) Encoding.

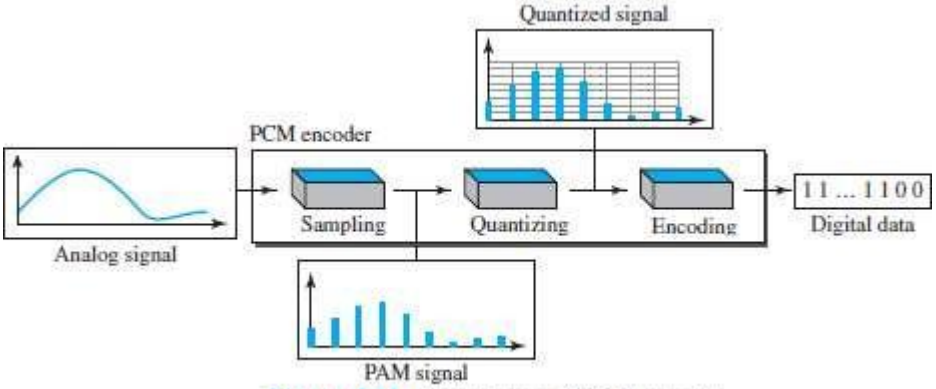


Figure 4.21 *Components of PCM encoder*

2.1.1.1 Sampling

- We convert the continuous time signal (analog) into the discrete time signal (digital).
- Pulses from the analog-signal are sampled every T_s sec
  where T_s is the sample-interval or period.
- The inverse of the sampling-interval is called the sampling-frequency (or sampling-rate).
- Sampling-frequency is given by

$$f_s = 1/T_s$$

- Three sampling methods (Figure 4.22):

1) Ideal Sampling

- This method is difficult to implement.

2) Natural Sampling

- A high-speed switch is turned ON for only the small period of time when the sampling occurs.
- The result is a sequence of samples that retains the shape of the analog-signal.

3) Flat Top Sampling

- The most common sampling method is sample and hold.
- Sample and hold method creates flat-top samples.
- This method is sometimes referred to as *PAM* (pulse amplitude modulation).

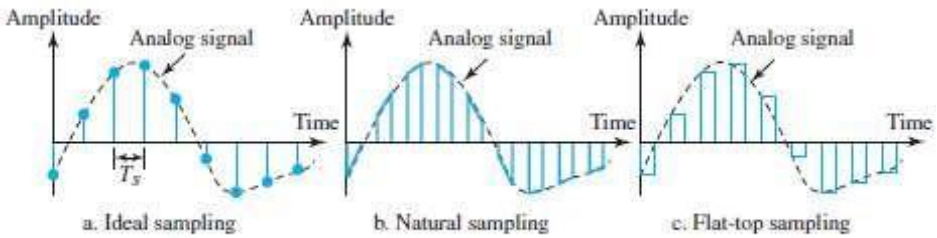


Figure 4.22 *Three different sampling methods for PCM*

2.1.1.1.1 Sampling Rate

- According to Nyquist theorem,

"The sampling-rate must be at least 2 times the highest frequency, not the bandwidth".

i) If the analog-signal is **low-pass**, the bandwidth and the highest frequency are the same value (Figure 4.23a).

ii) If the analog-signal is **bandpass**, the bandwidth value is lower than the value of the maximum frequency (Figure 4.23b).

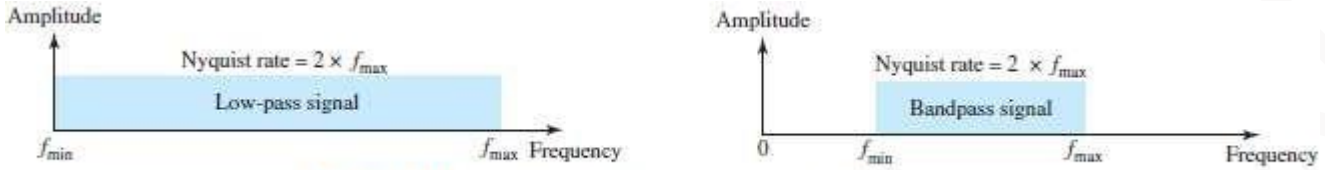


Figure 4.23 *Nyquist sampling rate for low-pass and bandpass signals*

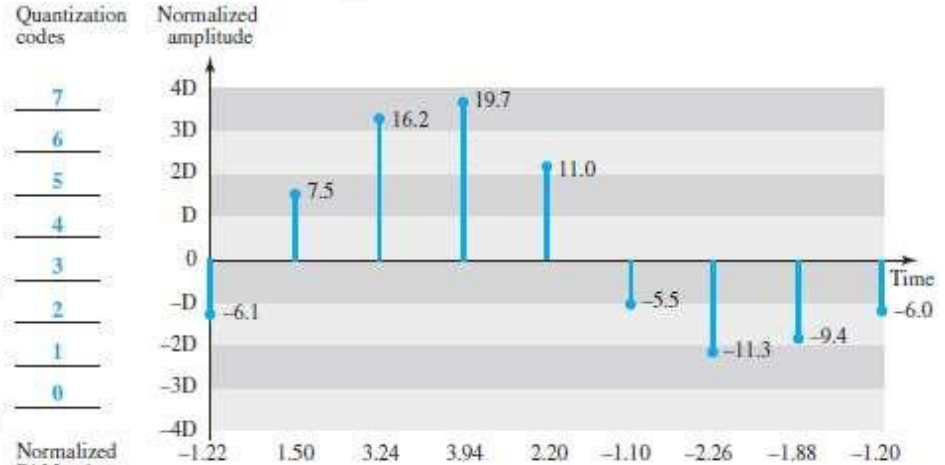
2.1.2 Quantization

- The sampled-signal is quantized.
- Result of sampling is a set of pulses with amplitude-values b/w max & min amplitudes of the signal.
- Four steps in quantization:
  1) We assume that the original analog-signal has amplitudes between V_{min} & V_{max}.
  2) We divide the range into L zones, each of height Δ(delta).

$$\Delta = \frac{V_{max} - V_{min}}{L}$$

where L = number of levels.

  3) We assign quantized values of 0 to (L-1) to the midpoint of each zone.
  4) We approximate the value of the sample amplitude to the quantized values.
- For example: Let V_{min}=-20 V_{max} =+20 V L = 8 Therefore, Δ = [+20-(-20)]/8= 5 V
- In the chart (Figure 4.26),
  1) First row is normalized-PAM-value for each sample.
  2) Second row is normalized-quantized-value for each sample.
  3) Third row is normalized error (which is diff. b/w normalized PAM value & quantized values).
  4) Fourth row is quantization code for each sample.
  5) Fifth row is the encoded words (which are the final products of the conversion).



Normalized PAM values	–1.22	1.50	3.24	3.94	2.20	–1.10	–2.26	–1.88	–1.20
Normalized quantized values	–1.50	1.50	3.50	3.50	2.50	–1.50	–2.50	–1.50	–1.50
Normalized error	–0.38	0	+0.26	–0.44	+0.30	–0.40	–0.24	+0.38	–0.30
Quantization code	2	5	7	7	6	2	1	2	2
Encoded words	010	101	111	111	110	010	001	010	010

Figure 4.26 *Quantization and encoding of a sampled signal*

2.1.2.1 Quantization Level

- Let L = number of levels.
- The choice of L depends on
  → range of the amplitudes of the analog-signal and
  → how accurately we need to recover the signal.
- If the signal has only 2 amplitude values, we need only 2 quantization-levels.
  If the signal (like voice) has many amplitude values, we need more quantization-levels.
- In audio digitizing, L is normally chosen to be 256.
  In video digitizing, L is normally thousands.
- Choosing lower values of L increases the quantization-error.

2.1.2.2 Quantization Error

- Quantization-error is the difference b/w normalized PAM value & quantized values
- Quantization is an approximation process.
- The input values to the quantizer are the real values.
  The output values from the quantizer are the approximated values.
- The output values are chosen to be the middle value in the zone.
- If the input value is also at the middle of the zone,
  Then, there is no error.
  Otherwise, there is an error.
- In the previous example,
  The normalized amplitude of the third sample is 3.24, but the normalized quantized value is 3.50. This means that there is an error of +0.26.

2.1.2.3 Uniform vs. Non Uniform Quantization

- Non-uniform quantization can be done by using a process called companding and expanding.
  1) The signal is companded at the sender before conversion.
  2) The signal is expanded at the receiver after conversion.
- Companding means reducing the instantaneous voltage amplitude for large values.
  Expanding means increasing the instantaneous voltage amplitude for small values.
- It has been proved that non-uniform quantization effectively reduces the SNR_{dB} of quantization.

2.1.3 Encoding

- The quantized values are encoded as n-bit code word.
- In the previous example,
  A quantized value 2 is encoded as 010.
  A quantized value 5 is encoded as 101.
- Relationship between number of quantization-levels (L) & number of bits (n) is given by
  $n=\log_2 L$ or $2^n=L$
- The bit-rate is given by:

$$\text{Bit rate} = \text{sampling rate} \times \text{number of bits per sample} = f_s \times n$$

Example 2.1

A complex low-pass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

Solution

The bandwidth of a low-pass signal is between 0 and f, where f is the maximum frequency in the signal. Therefore, we can sample this signal at 2 times the highest frequency (200 kHz). The sampling rate is therefore 400,000 samples per second.

Example 2.2

What is the SNR_{dB} in the example of Figure 4.26?

Solution

We can use the formula to find the quantization. We have eight levels and 3 bits per sample, so $SNR_{dB} = 6.02(3) + 1.76 = 19.82$ dB. Increasing the number of levels increases the SNR.

Example 2.3

A telephone subscriber line must have an SNR_{dB} above 40. What is the minimum number of bits per sample?

Solution

We can calculate the number of bits as

$$SNR_{dB} = 6.02n_b + 1.76 = 40 \rightarrow n = 6.35$$

Telephone companies usually assign 7 or 8 bits per sample.

Example 2.4

We want to digitize the human voice. What is the bit rate, assuming 8 bits per sample?

Solution

The human voice normally contains frequencies from 0 to 4000 Hz. So the sampling rate and bit rate are calculated as follows:

$$\text{Sampling rate} = 4000 \times 2 = 8000 \text{ samples/s}$$

$$\text{Bit rate} = 8000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}$$

2.1.3.1 Original Signal Recovery

- PCM decoder is used for recovery of the original signal.
- Here is how it works (Figure 4.27):
  1) The decoder first uses circuitry to convert the code words into a pulse that holds the amplitude until the next pulse.
  2) Next, the staircase-signal is passed through a low-pass filter to smooth the staircase signal into an analog-signal.
- The filter has the same cut-off frequency as the original signal at the sender.
- If the signal is sampled at the Nyquist sampling-rate, then the original signal will be re-created.
- The maximum and minimum values of the original signal can be achieved by using amplification.

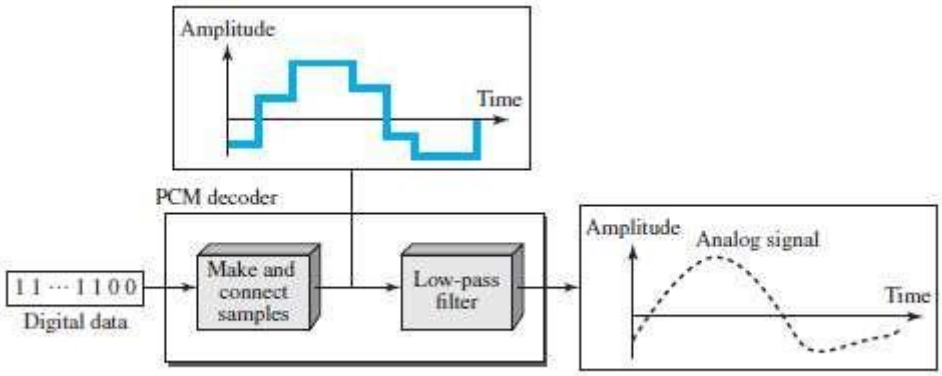


Figure 4.27 *Components of a PCM decoder*

2.1.3.2 PCM Bandwidth

- The minimum bandwidth of a line-encoded signal is

$$B_{min} = c \times N \times \frac{1}{r}$$

- We substitute the value of N in above formula:

$$B_{min} = c \times N \times \frac{1}{r} = c \times n_b \times f_s \times \frac{1}{r} = c \times n_b \times 2 \times B_{analog} \times \frac{1}{r}$$

- When 1/r = 1 (for a NRZ or bipolar signal) and c = (1/2) (the average situation), the minimum bandwidth is

$$B_{min} = n_b \times B_{analog}$$

- This means the minimum bandwidth of the digital-signal is n_b times greater than the bandwidth of the analog-signal.

2.1.3.3 Maximum Data Rate of a Channel

• The Nyquist theorem gives the data-rate of a channel as

$$N_{max} = 2 \times B \times \log_2 L$$

• We can deduce above data-rate from the Nyquist sampling theorem by using the following arguments.

1) We assume that the available channel is low-pass with bandwidth B.
2) We assume that the digital-signal we want to send has L levels, where each level is a signal-element. This means r = $1/\log_2 L$.
3) We first pass digital-signal through a low-pass filter to cut off the frequencies above B Hz.
4) We treat the resulting signal as an analog-signal and sample it at 2 x B samples per second and quantize it using L levels.
5) The resulting bit-rate is

$$N = f_s \times n_b = 2 \times B \times \log_2 L$$

This is the maximum bandwidth; if the case factor c increases, the data-rate is reduced.

$$N_{max} = 2 \times B \times \log_2 L \quad \text{bps}$$

2.1.3.4 Minimum Required Bandwidth

• The previous argument can give us the minimum bandwidth if the data-rate and the number of signal-levels are fixed. We can say

$$B_{min} = \frac{N}{(2 \times \log_2) L} \quad \text{Hz}$$

2.2 TRANSMISSION MODES

• Two ways of transmitting data over a link (Figure 4.31): 1) Parallel mode & 2) Serial mode.

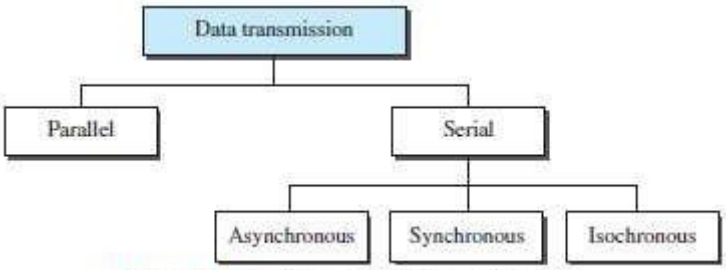


Figure 4.31 *Data transmission and modes*

2.1.1 PARALLEL TRANSMISSION

• Multiple bits are sent with each clock-tick (Figure 4.32).
• „n" bits in a group are sent simultaneously.
• „n" wires are used to send „n" bits at one time.
• Each bit has its own wire.
• Typically, the 8 wires are bundled in a cable with a connector at each end.

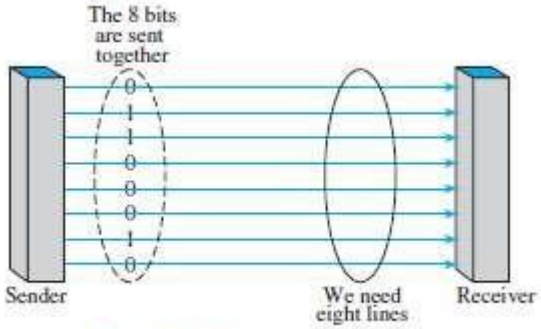


Figure 4.32 *Parallel transmission*

• Advantage:
  1) Speed: Parallel transmission can increase the transfer speed by a factor of n over serial transmission.
• Disadvantage:
  1) Cost: Parallel transmission requires n communication lines just to transmit the data-stream. Because this is expensive, parallel transmission is usually limited to short distances.

2.2.2 SERIAL TRANSMISSION

• One bit is sent with each clock-tick using only a single link (Figure 4.33).

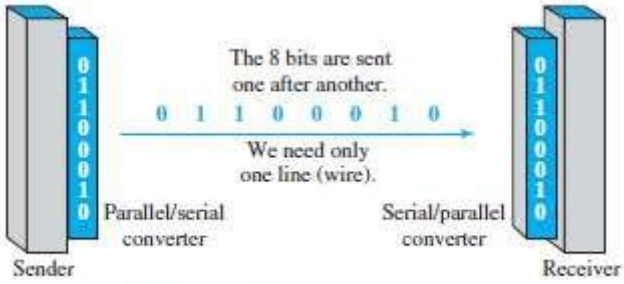


Figure 4.33 *Serial transmission*

• Advantage:
  1) Cost: Serial transmission reduces cost of transmission over parallel by a factor of n.
• Disadvantage:
  1) Since communication within devices is parallel, following 2 converters are required at interface:
     i) Parallel-to-serial converter
     ii) Serial-to-parallel converter
• Three types of serial transmission: asynchronous, synchronous, and isochronous.

2.2.2.1 Asynchronous Transmission

- Asynchronous transmission is so named because the timing of a signal is not important (Figure 4.34).
- Prior to data transfer, both sender & receiver agree on pattern of information to be exchanged.
- Normally, patterns are based on grouping the bit-stream into bytes.
- The sender transmits each group to the link without regard to a timer.
- As long as those patterns are followed, the receiver can retrieve the info. without regard to a timer.
- There may be a gap between bytes.
- We send
  → 1 start bit (0) at the beginning of each byte
  → 1 stop bit (1) at the end of each byte.
- Start bit alerts the receiver to the arrival of a new group.
  Stop bit lets the receiver know that the byte is finished.
- Here, the term asynchronous means "asynchronous at the byte level".
  However, the bits are still synchronized & bit-durations are the same.

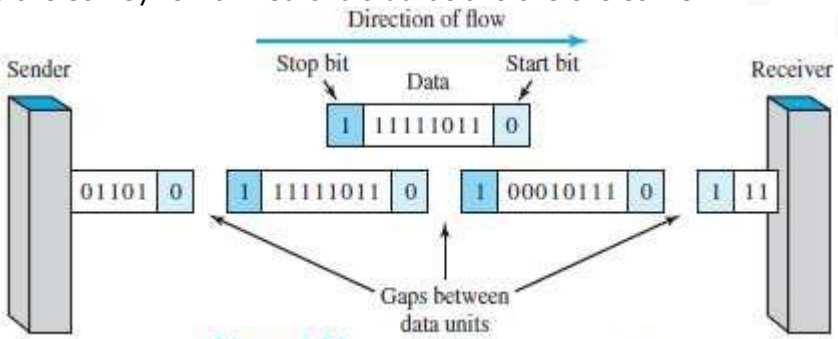


Figure 4.34 *Asynchronous transmission*

- Disadvantage:
  1) Slower than synchronous transmission. (Because of stop bit, start bit and gaps)
- Advantages:
  1) Cheap & effective.
  2) Useful for low-speed communication.

2.2.2.2 Synchronous Transmission

- We send bits one after another without start or stop bits or gaps (Figure 4.35).
- The receiver is responsible for grouping the bits.
- The bit-stream is combined into longer "frames," which may contain multiple bytes.
- If the sender wants to send data in separate bursts, the gaps b/w bursts must be filled with a special sequence of 0s & 1s (that means idle).

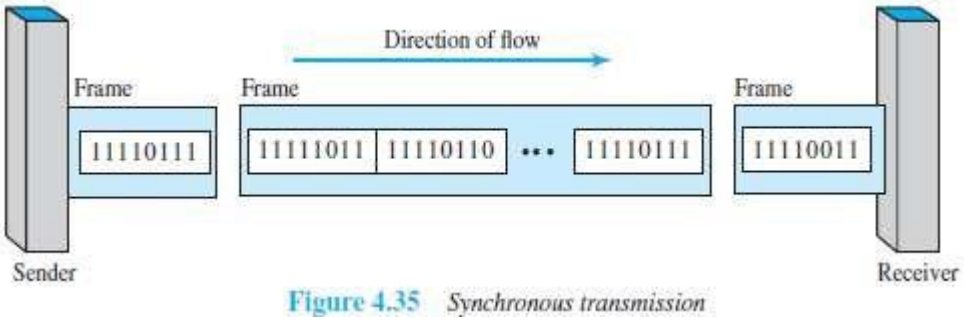


Figure 4.35 *Synchronous transmission*

- Advantages:
  1) Speed: Faster than asynchronous transmission. („." of no stop bit, start bit and gaps).
  2) Useful for high-speed applications such as transmission of data from one computer to another.

2.2.2.3 Isochronous

- Synchronization between characters is not enough; the entire stream of bits must be synchronized.
- The isochronous transmission guarantees that the data arrive at a fixed rate.
- In real-time audio/video, jitter is not acceptable. Therefore, synchronous transmission fails.
- For example: TV images are broadcast at the rate of 30 images per second. The images must be viewed at the same rate.

MODULE 1(CONT.): ANALOG TRANSMISSION

2.3 DIGITAL TO ANALOG CONVERSION

• Digital-to-analog conversion is the process of changing one of the characteristics of an analog-signal based on the information in digital-data (Figure 5.1).

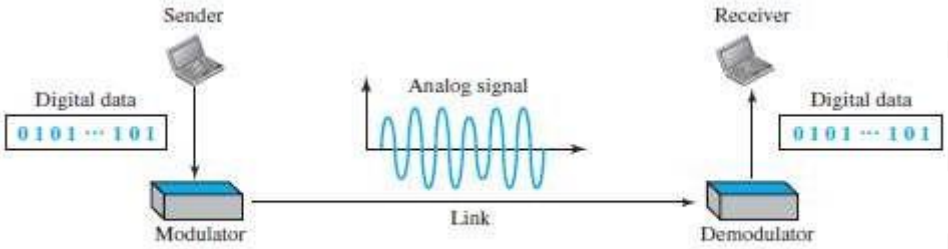


Figure 5.1 *Digital-to-analog conversion*

• A sine wave can be defined by 3 attributes:
  1) Amplitude
  2) Frequency &
  3) Phase.
• When anyone of the 3 attributes of a wave is varied, a different version of the wave will be created.
• So, by changing one attribute of an analog signal, we can use it to represent digital-data.
• Four methods of digital to analog conversion (Figure 5.2):
  1) Amplitude shift keying (ASK)
  2) Frequency shift keying (FSK)
  3) Phase shift keying (PSK)
  4) Quadrature amplitude modulation (QAM).
• QAM is a combination of ASK and PSK i.e. QAM combines changing both the amplitude and phase.
  QAM is the most efficient of these 4 methods.
    QAM is the method commonly used today.

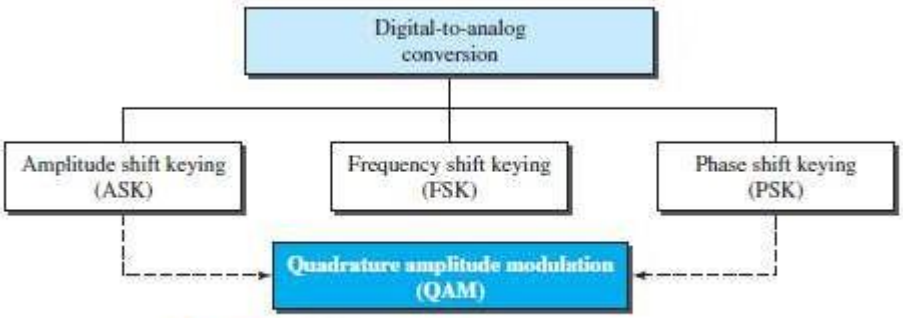


Figure 5.2 *Types of digital-to-analog conversion*

2.3.1 Aspects of Digital-to-Analog Conversion

1) Data Element vs. Signal Element

- A data-element is the smallest piece of information to be exchanged i.e. the bit.
- A signal-element is the smallest unit of a signal that is transmitted.

2) Data Rate vs. Signal Rate

- Data rate (Bit rate) is the number of bits per second.

  Signal-rate (Baud rate) is the number of signal elements per second.
- The relationship between data-rate(N) and the signal-rate(S) is

$$S = N \times \frac{1}{r} \quad \text{baud}$$

where r = number of data-elements carried in one signal-element.

- The value of r is given by

  $r = \log_2 L$ or $2^r = L$

  where L = type of signal-element (not the level)

  (In transportation,

  → a baud is analogous to a vehicle, and

  → a bit is analogous to a passenger.

  We need to maximize the number of people per car to reduce the traffic).

3) Carrier Signal

- The sender produces a high-frequency signal that acts as a base for the information-signal.
- This base-signal is called the carrier-signal (or carrier-frequency).
- The receiver is tuned to the frequency of the carrier-signal that it expects from the sender.
- Then, digital-information changes the carrier-signal by modifying its attributes (amplitude, frequency, or phase). This kind of modification is called modulation (shift keying).

4) Bandwidth

- In both ASK & PSK, the bandwidth required for data transmission is proportional to the signal-rate.
- In FSK, the bandwidth required is the difference between the two carrier-frequencies.

Example 2.5

An analog signal carries 4 bits per signal element. If 1000 signal elements are sent per second, find the bit rate.

Solution

In this case, $r = 4$, $S = 1000$, and N is unknown. We can find the value of N from

$$S = N \times (1/r) \quad \text{or} \quad N = S \times r = 1000 \times 4 = 4000 \text{ bps}$$

Example 2.6

An analog signal has a bit rate of 8000 bps and a baud rate of 1000 baud. How many data elements are carried by each signal element? How many signal elements do we need?

Solution

In this example, $S = 1000$, $N = 8000$, and r and L are unknown. We first find the value of r and then the value of L.

$$S = N \times 1/r \longrightarrow r = N/S = 8000/10{,}000 = 8 \text{ bits/baud}$$

$$r = \log_2 L \longrightarrow L = 2^r = 2^8 = 256$$

2.3.2 Amplitude Shift Keying (ASK)

•The amplitude of the carrier-signal is varied to represent different signal-elements.
• Both frequency and phase remain constant for all signal-elements.

2.3.2.1 Binary ASK (BASK)

• BASK is implemented using only 2 levels. (Figure 5.3)
• This is also referred to as OOK (On-Off Keying).

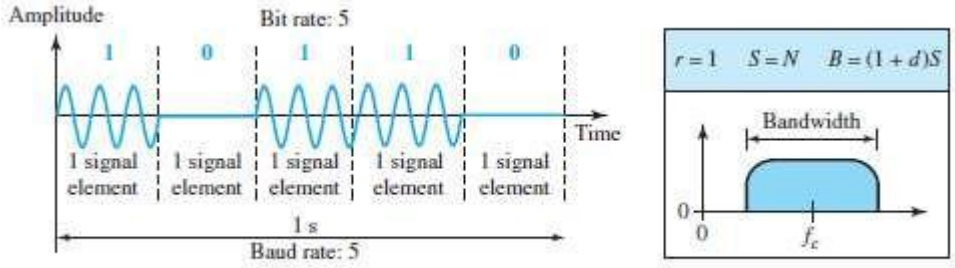


Figure 5.3 *Binary amplitude shift keying*

2.3.2.1.1 Implementation of BASK

• Here, line coding method used = unipolar NRZ (Figure 5.4).
• The unipolar NRZ signal is multiplied by the carrier-frequency coming from an oscillator.
1) When amplitude of the NRZ signal = 0, amplitude of the carrier-signal = 0.
2) When amplitude of the NRZ signal = 1, the amplitude of the carrier-signal is held.

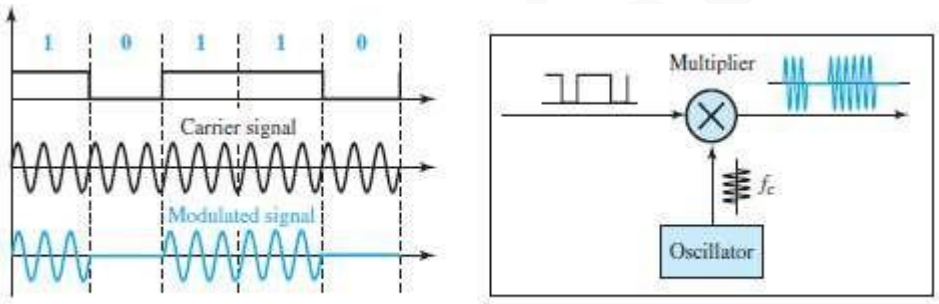


Figure 5.4 *Implementation of binary ASK*

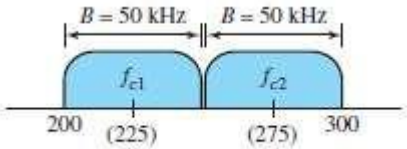


Figure 5.5 *Bandwidth of full-duplex ASK*

2.3.2.1.2 Bandwidth for ASK

• Here, the bandwidth (B) is proportional to the signal-rate (S) (Figure 5.5)
• The bandwidth is given by

$$B = (1 + d) \times S$$

where d(0<d<1)= this factor depends on modulation and filtering-process.

Example 2.7

We have an available bandwidth of 100 kHz which spans from 200 to 300 kHz. What are the carrier frequency and the bit rate if we modulated our data by using ASK with $d = 1$?

Solution

The middle of the bandwidth is located at 250 kHz. This means that our carrier frequency can be at $f_c = 250$ kHz. We can use the formula for bandwidth to find the bit rate (with $d = 1$ and $r = 1$).

$$B = (1 + d) \times S = 2 \times N \times (1/r) = 2 \times N = 100 \text{ kHz} \longrightarrow N = 50 \text{ kbps}$$

2.3.3 Frequency Shift Keying (FSK)

- The frequency of the carrier-signal is varied to represent different signal-elements.
- The frequency of the modulated-signal is constant for the duration of one signal-element, but changes for the next signal-element if the data-element changes.
- Both amplitude and phase remain constant for all signal-elements.

2.3.3.1 Binary FSK (BFSK)

- This uses 2 carrier-frequencies: f1 and f2. (Figure 5.6)
  1) When data-element = 1, first carrier frequency(f1) is used.
  2) When data-element = 0, second carrier frequency(f2) is used.

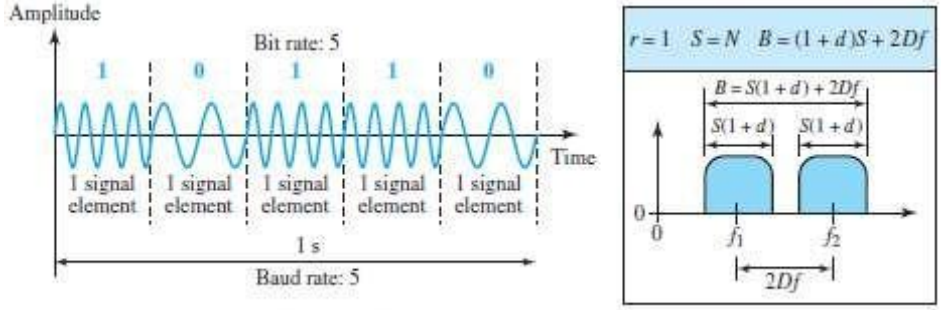


Figure 5.6 *Binary frequency shift keying*

2.3.3.1.1 Implementation

- Here, line coding method used = unipolar NRZ.
- Two implementations of BFSK: i) Coherent and ii) Non-Coherent.

Coherent BFSK	Non Coherent BFSK
The phase continues through the boundary of two signal-elements (Figure 5.7).	There may be discontinuity in the phase when one signal-element ends and the next begins.
This is implemented by using one voltage-controlled oscillator (VCO). VCO changes frequency according to the input voltage.	This is implemented by → treating BFSK as 2 ASK modulations and → using 2 carrier-frequencies
When the amplitude of NRZ signal = 0, the VCO keeps its regular frequency. When the amplitude of NRZ signal = 0, the VCO increases its frequency.	

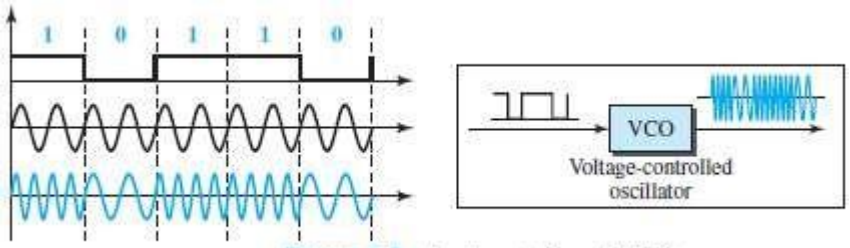


Figure 5.7 *Implementation of BFSK*

2.3.3.1.2 Bandwidth for BFSK

- FSK has two ASK signals, each with its own carrier-frequency f1 or f2. (Figure 5.6)
- The bandwidth is given by

$B = (1 + d) \times S + 2\Delta f$ where 2Δf is the difference between f1 and f2,

Example 2.8

We have an available bandwidth of 100 kHz which spans from 200 to 300 kHz. What should be the carrier frequency and the bit rate if we modulated our data by using FSK with $d = 1$?

Solution

This problem is similar to Example 5.3, but we are modulating by using FSK. The midpoint of the band is at 250 kHz. We choose $2\Delta_f$ to be 50 kHz; this means

$$B = (1 + d) \times S + 2\Delta_f = 100 \longrightarrow 2S = 50\ \text{kHz} \longrightarrow S = 25\ \text{kbaud} \longrightarrow N = 25\ \text{kbps}$$

Example 2.9

We need to send data 3 bits at a time at a bit rate of 3 Mbps. The carrier frequency is 10 MHz. Calculate the number of levels (different frequencies), the baud rate, and the bandwidth.

Solution

We can have $L = 2^3 = 8$. The baud rate is $S = 3$ MHz/3 = 1 Mbaud. This means that the carrier frequencies must be 1 MHz apart ($2\Delta_f = 1$ MHz). The bandwidth is $B = 8 \times 1 = 8$ MHz. Figure 5.8 shows the allocation of frequencies and bandwidth.

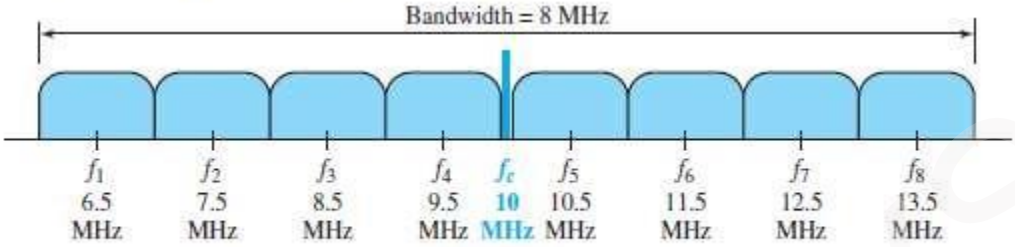


Figure 5.8 *Bandwidth of MFSK used*

2.3.4 Phase Shift Keying (PSK)

- The phase of the carrier-signal is varied to represent different signal-elements.
- Both amplitude and frequency remain constant for all signal-elements.

2.3.4.1 Binary PSK (BPSK)

- We have only two signal-elements:
  1) First signal-element with a phase of 0°.
  2) Second signal-element with a phase of 180° (Figure 5.9).
- ASK vs. PSK
  - In ASK, the criterion for bit detection is the amplitude of the signal.
  - In PSK, the criterion for bit detection is the phase.
- Advantages:
  1) PSK is less susceptible to noise than ASK.
  2) PSK is superior to FSK because we do not need 2 carrier-frequencies.
- Disadvantage:
  1) PSK is limited by the ability of the equipment to distinguish small differences in phase.

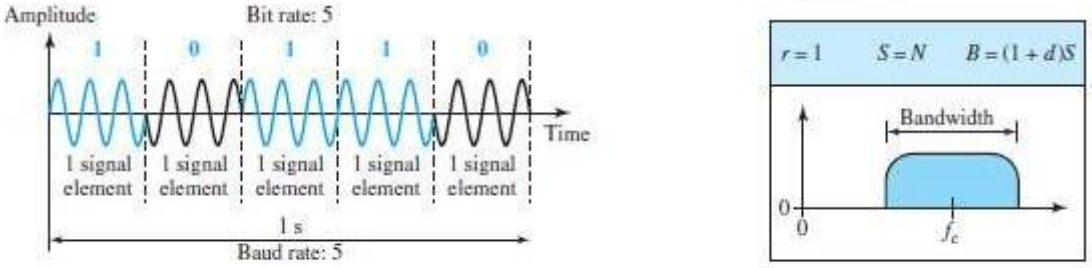


Figure 5.9 *Binary phase shift keying*

2.3.4.1.1 Implementation

- The implementation of BPSK is as simple as that for ASK. (Figure 5.10).
- The signal-element with phase 180° can be seen as the complement of the signal-element with phase 0°.
- Here, line coding method used: polar NRZ.
- The polar NRZ signal is multiplied by the carrier-frequency coming from an oscillator.
  1) When data-element = 1, the phase starts at 0°.
  2) When data-element = 0, the phase starts at 180°.

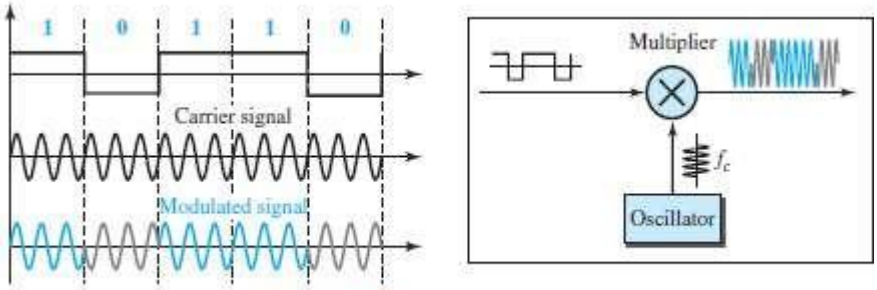


Figure 5.10 *Implementation of BASK*

2.3.4.1.2 Bandwidth for BPSK

- The bandwidth is the same as that for BASK, but less than that for BFSK. (Figure 5.9b)
- No bandwidth is wasted for separating 2 carrier-signals.

2.3.4.2 Quadrature PSK (QPSK)

- The scheme is called QPSK because it uses 2 separate BPSK modulations (Figure 5.11):
  1) First modulation is in-phase,
  2) Second modulation is quadrature (out-of-phase).
- A serial-to-parallel converter
  → accepts the incoming bits
  → sends first bit to first modulator and
  → sends second bit to second modulator.
- The bit to each BPSK signal has one-half the frequency of the original signal.
- Advantages:
  1) Decreases the baud rate.
  2) Decreases the required bandwidth.

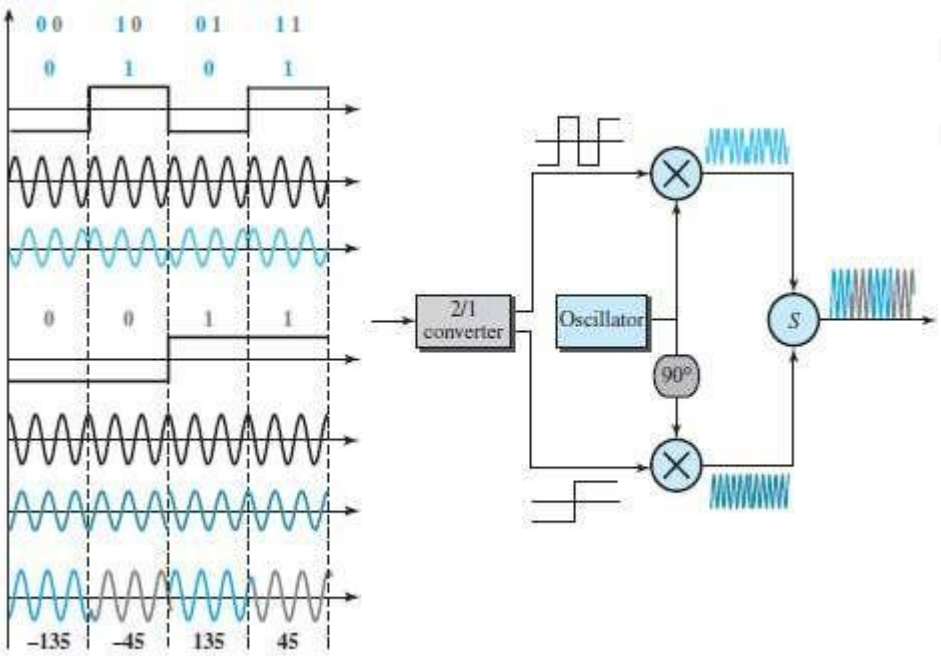


Figure 5.11 *QPSK and its implementation*

- As shown in Figure 5.11, the 2 composite-signals created by each multiplier are 2 sine waves with the same frequency, but different phases.
- When the 2 sine waves are added, the result is another sine wave, with 4 possible phases: 45°, -45°, 135°, and -135°.
- There are 4 kinds of signal-elements in the output signal (L=4), so we can send 2 bits per signal-element (r=2).

Example 2.10

Find the bandwidth for a signal transmitting at 12 Mbps for QPSK. The value of $d = 0$.

Solution

For QPSK, 2 bits are carried by one signal element. This means that $r = 2$. So the signal rate (baud rate) is $S = N \times (1/r) = 6$ Mbaud. With a value of $d = 0$, we have $B = S = 6$ MHz.

2.3.4.3 Constellation Diagram

- A constellation diagram can be used to define the amplitude and phase of a signal-element.
- This diagram is particularly useful
  - → when 2 carriers (one in-phase and one quadrature) are used.
  - → when dealing with multilevel ASK, PSK, or QAM.
- In a constellation diagram, a signal-element type is represented as a dot.
- The diagram has 2 axes (Figure 5.12):
  1) The horizontal X axis is related to the in-phase carrier.
  2) The vertical Y axis is related to the quadrature carrier.
- For each point on the diagram, 4 pieces of information can be deduced.
  1) The projection of point on the X axis defines the peak amplitude of the in-phase component.
  2) The projection of point on Y axis defines peak amplitude of the quadrature component.
  3) The length of the line that connects the point to the origin is the peak amplitude of the signal-element (combination of the X and Y components);
  4) The angle the line makes with the X axis is the phase of the signal-element.

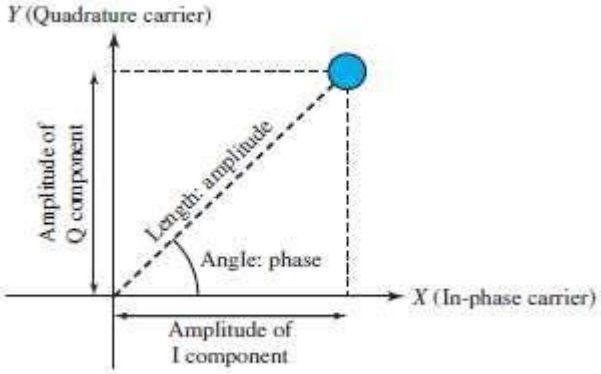


Figure 5.12 *Concept of a constellation diagram*

Example 2.11

Show the constellation diagrams for ASK (OOK), BPSK, and QPSK signals.

Solution

Figure 5.13 shows the three constellation diagrams. Let us analyze each case separately:

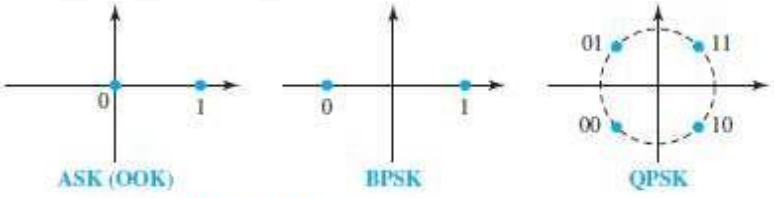


Figure 5.13 *Three constellation diagrams*

2.3.5 Quadrature Amplitude Modulation (QAM)

• This is a combination of ASK and PSK.

• Main idea: Using 2 carriers, one in-phase and the other quadrature, with different amplitude levels for each carrier.

• There are many variations of QAM (Figure 5.14).

A) Figure 5.14a shows the 4-QAM scheme using a unipolar NRZ signal. This is same as BASK.

B) Figure 5.14b shows another QAM using polar NRZ. This is the same as QPSK.

C) Figure 5.14c shows another 4-QAM in which we used a signal with 2 positive levels to modulate each of the 2 carriers.

D) Figure 5.14d shows a 16-QAM constellation of a signal with 8 levels, 4 positive & 4 negative.

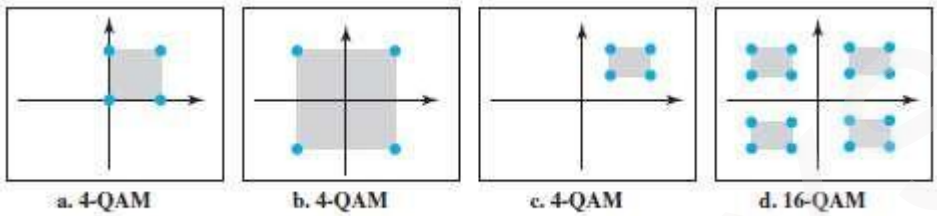


Figure 5.14 *Constellation diagrams for some QAMs*

2.3.5.1 Bandwidth for QAM

• The bandwidth is same as in ASK and PSK transmission.

• QAM has the same advantages as PSK over ASK.

MODULE 2: TABLE OF CONTENTS

3.1 INTRODUCTION
- 3.1.1 Types of Errors
- 3.1.2 Redundancy
- 3.1.3 Detection versus Correction
- 3.1.4 Coding

3.2 BLOCK CODING
- 3.2.1 Error Detection
  - 3.2.1.1 Hamming Distance
    - 3.2.1.1.1 Minimum Hamming Distance for Error Detection
  - 3.2.1.2 Linear Block Codes
    - 3.2.1.2.1 Minimum Distance for Linear Block Codes
  - 3.2.1.3 Parity-Check Code

3.3 CYCLIC CODES
- 3.3.1 Cyclic Redundancy Check (CRC)
- 3.3.2 Polynomials
- 3.3.3 Cyclic Code Encoder Using Polynomials
- 3.3.4 Cyclic Code Analysis
- 3.3.5 Advantages of Cyclic Codes

3.4 CHECKSUM
- 3.4.1 Concept of Checksum
  - 3.4.1.1 One's Complement
  - 3.4.1.2 Internet Checksum
  - 3.4.1.3 Algorithm
- 3.4.2 Other Approaches to the Checksum
  - 3.4.2.1 Fletcher Checksum
  - 3.4.2.2 Adler Checksum

3.5 FORWARD ERROR CORRECTION
- 3.5.1 Using Hamming Distance
- 3.5.2 Using XOR
- 3.5.3 Chunk Interleaving
- 3.5.4 Combining Hamming Distance and Interleaving
- 3.5.5 Compounding High- and Low-Resolution Packets

3.6 DLC SERVICES
- 3.6.1 Framing
  - 3.6.1.1 Frame Size
  - 3.6.1.2 Character-Oriented Framing
  - 3.6.1.3 Bit-Oriented Framing
- 3.6.2 Flow and Error Control
  - 3.6.2.1 Flow-control
    - 3.6.2.1.1 Buffers
  - 3.6.2.2 Error-control
    - 3.6.2.2.1 Combination of Flow and Error Control
- 3.6.3 Connectionless and Connection-Oriented

3.7 DATA-LINK LAYER PROTOCOLS
- 3.7.1 Simple Protocol
  - 3.7.1.1 Design
  - 3.7.1.2 FSMs
- 3.7.2 Stop-and-Wait Protocol
  - 3.7.2.1 Design
  - 3.7.2.2 FSMs

3.7.2.3 Sequence and Acknowledgment Numbers

3.7.3 piggybacking
3.8 High-level Data Link Control (HDLC)
3.8.1 Configurations and Transfer Modes
3.8.2 Framing
3.8.2.1 Frame Format
3.8.2.1.1 Control Fields of HDLC Frames
3.9 POINT-TO-POINT PROTOCOL (PPP)
3.9.1 Framing
3.9.1.1 Byte Stuffing
3.9.2 Transition Phases
3.10 INTRODUCTION
3.11 RANDOM ACCESS PROTOCOL
3.11.1 ALOHA
3.11.1.1 Pure ALOHA
3.11.1.1.1 Vulnerable time
3.11.1.1.2 Throughput
3.11.1.2 Slotted ALOHA
3.11.1.2.1 Throughput
3.11.2 CSMA
3.11.2.1 Vulnerable Time
3.11.2.2 Persistence Methods
3.11.3 CSMA/CD
3.11.3.1 Minimum Frame-size
3.11.3.2 Procedure
3.11.3.3 Energy Level
3.11.3.4 Throughput
3.11.4 CSMA/CA
3.11.4.1 Frame Exchange Time Line
3.11.4.2 Network Allocation Vector
3.11.4.3 Collision During Handshaking
3.11.4.4 Hidden-Station Problem
3.11.4.5 CSMA/CA and Wireless Networks
3.12 CONTROLLED ACCESS PROTOCOL
3.12.1 Reservation
3.12.2 Polling
3.12.3 Token Passing
3.12.3.1 Logical Ring
3.13 CHANNELIZATION
3.13.1 FDMA
3.13.2 TDMA
3.13.3 CDMA
3.13.3.1 Implementation
3.13.3.2 Chips
3.13.3.3 Data Representation
3.13.3.4 Encoding and Decoding
Sequence Generation

QUESTIONS(MODULE 2)

MODULE 2: ERROR-DETECTION AND CORRECTION

1) Explain two types of errors (4*)
2) Compare error detection vs. error correction (2)
3) Explain error detection using block coding technique. (10*)
4) Explain hamming distance for error detection (6*)
5) Explain parity-check code with block diagram. (6*)
6) Explain CRC with block diagram & an example. (10*)
7) Write short notes on polynomial codes. (5*)
8) Explain internet checksum algorithm along with an example. (6*)
9) Explain the following:
   i) Fletcher checksum and ii) Adler checksum (8)
10) Explain various FEC techniques. (6)

MODULE 2: DATA LINK CONTROL

1) Explain two types of frames. (2)
2) Explain character oriented protocol. (6*)
3) Explain the concept of byte stuffing and unstuffing with example. (6*)
4) Explain bit oriented protocol. (6*)
5) Differentiate between character oriented and bit oriented format for Framing. (6*)
6) Compare flow control and error control. (4)
7) With a neat diagram, explain the design of the simplest protocol with no flow control. (6)
8) Write algorithm for sender site and receiver site for the simplest protocol. (6)
9) Explain Stop-and-Wait protocol (8*)
10) Explain the concept of Piggybacking (2*)
11) Explain in detail HDLC frame format. (8*)
12) Explain 3 type of frame used in HDLC (8*)
13) With a neat schematic, explain the frame structure of PPP protocol. (8*)
14) Explain framing and transition phases in Point-to-Point Protocol. (8*)

MODULE 2: RANDOM ACCESS(MULTIPLE ACCESS)

1) Explain random access protocol. (4)
2) Explain pure ALOHA. (6*)
3) Explain slotted ALOHA. (4*)
4) Explain CSMA. (6*)
5) Explain different persistence methods of CSMA. (6*)
6) Explain CSMA/CA. (6*)
7) Explain CSMA/CD. (10*)
8) List & explain different controlled access protocols. (10*)
9) Explain reservation access method. (4*)
10) Explain polling access method. (6*)
11) Explain token passing access method. (6*)
12) List & explain channelization protocols. (10*)
13) Explain FDMA. (6*)
14) Explain TDMA. (6*)
15) Explain CDMA. (8*)

MODULE 2: ERROR-DETECTION AND CORRECTION

3.1 INTRODUCTION

3.1.1 Types of Errors

• When bits flow from 1 point to another, they are subject to unpredictable-changes '.' of interference.
• The interference can change the shape of the signal.
• Two types of errors: 1) Single-bit error 2) Burst-error.

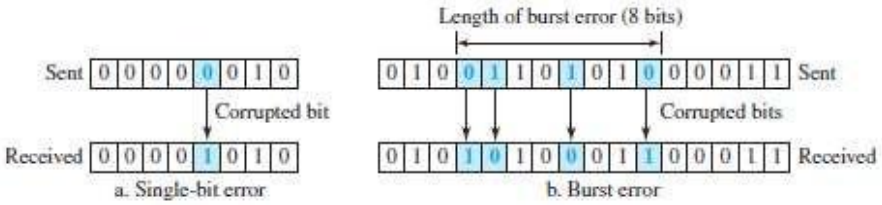


Figure 10.1 *Single-bit and burst error*

1) Single-Bit Error

➢ Only 1 bit of a given data is changed
→ from 1 to 0 or
→ from 0 to 1 (Figure 10.1a).

2) Burst Error

➢ Two or more bits in the data have changed
→ from 1 to 0 or
→ from 0 to 1 (Figure 10.1b).
➢ A burst-error occurs more than a single-bit error. This is because:
Normally, the duration of noise is longer than the duration of 1-bit.
➢ When noise affects data, the noise also affects the bits.
➢ The no. of corrupted-bits depends on
→ data-rate and
→ duration of noise.

3.1.2 Redundancy

• The central concept in detecting/correcting errors is *redundancy*.
• Some extra-bits along with the data have to be sent to detect/correct errors. These extra bits are called redundant-bits.
• The redundant-bits are
→ added by the sender and
→ removed by the receiver.
• The presence of redundant-bits allows the receiver to detect/correct errors.

3.1.3 Error Detection vs. Error Correction

• Error-correction is more difficult than error-detection.

1) Error Detection

➢ Here, we are checking whether any error has occurred or not.
➢ The answer is a simple YES or NO.
➢ We are not interested in the number of corrupted-bits.

2) Error Correction

➢ Here, we need to know
→ exact number of corrupted-bits and
→ location of bits in the message.
➢ Two important factors to be considered:
1) Number of errors and
2) Message-size.

3.1.4 Coding

• Redundancy is achieved through various coding-schemes.
  1) Sender adds redundant-bits to the data-bits. This process creates a relationship between
    → redundant-bits and
    → data-bits.
  2) Receiver checks the relationship between redundant-bits & data-bits to detect/correct errors.
• Two important factors to be considered:
  1) Ratio of redundant-bits to the data-bits and
  2) Robustness of the process.
• Two broad categories of coding schemes: 1) Block-coding and 2) Convolution coding.

3.2 Block Coding

• The message is divided into k-bit blocks. These blocks are called data-words.
• Here, r-redundant-bits are added to each block to make the length n=k+r.
• The resulting n-bit blocks are called code-words.
• Since n>k, the number of possible code-words is larger than the number of possible data-words.
• Block-coding process is 1-to-1; the same data-word is always encoded as the same code-word.
• Thus, we have 2^n-2^k code-words that are not used. These code-words are invalid or illegal.

3.2.1 Error Detection

• If the following 2 conditions are met, the receiver can detect a change in the original code-word:
  1) The receiver has a list of valid code-words.
  2) The original code-word has changed to an invalid code-words.

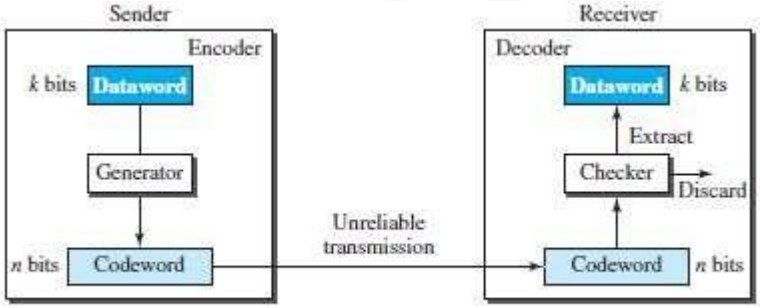


Figure 10.2 *Process of error detection in block coding*

• Here is how it works (Figure 10.2):
  1) At Sender
  i) The sender creates code-words out of data-words by using a generator.
    The generator applies the rules and procedures of encoding.
  ii) During transmission, each code-word sent to the receiver may change.
  2) At Receiver
  i) a) If the received code-word is the same as one of the valid code-words,
    the code-word is accepted;
      the corresponding data-word is extracted for use.
    b) If the received code-word is invalid, the code-word is discarded.
  ii) However, if the code-word is corrupted but the received code-word still matches a valid code-word, the error remains undetected.
• An error-detecting code can detect only the types of errors for which it is designed; other types of errors may remain undetected.

Example 3.1

Let us assume that $k = 2$ and $n = 3$. Table 10.1 shows the list of datawords and codewords.

Table 10.1 *A code for error detection*

Dataword	*Codeword*	*Dataword*	*Codeword*
00	000	10	101
01	011	11	110

Assume the sender encodes the dataword 01 as 011 and sends it to the receiver. Consider the following cases:

1. The receiver receives 011. It is a valid codeword. The receiver extracts the dataword 01 from it.
2. The codeword is corrupted during transmission, and 111 is received (the leftmost bit is corrupted). This is not a valid codeword and is discarded.
3. The codeword is corrupted during transmission, and 000 is received (the right two bits are corrupted). This is a valid codeword. The receiver incorrectly extracts the dataword 00. Two corrupted bits have made the error undetectable.

3.2.1.1 Hamming Distance

- The main concept for error-control: Hamming distance.
- The Hamming distance b/w 2 words is the number of differences between the corresponding bits.
- Let d(x,y) = Hamming distance b/w 2 words x and y.
- Hamming distance can be found by
  - → applying the XOR operation on the 2 words and
  - → counting the number of 1s in the result.
- For example:
  - **1)** The Hamming distance d(000, 011) is 2 because 000⊕011= 011 (two 1s).
  - 2) The Hamming distance d(10101, 11110) is 3 because 10101⊕11110 = 01011 (three 1s).

Hamming Distance and Error

➢ Hamming distance between the received word and the sent code-word is the number of bits that are corrupted during transmission.

➢ For example: Let Sent code-word = 00000
Received word = 01101
Hamming distance = d(00000, 01101) =3. Thus, 3 bits are in error.

3.2.1.1.1 Minimum Hamming Distance for Error Detection

- Minimum Hamming distance is the smallest Hamming distance b/w all possible pairs of code-words.
- Let d_{min} = minimum Hamming distance.
- To find d_{min} value, we find the Hamming distances between all words and select the smallest one.

Minimum-distance for Error-detection

➢ If 's' errors occur during transmission, the Hamming distance b/w the sent code-word and received code-word is 's' (Figure 10.3).

➢ If code has to detect upto 's' errors, the minimum-distance b/w the valid codes must be 's+1' i.e. $d_{min}=s+1$.

➢ We use a geometric approach to define $d_{min}=s+1$.

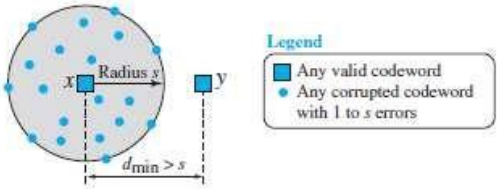


Figure 10.3 *Geometric concept explaining d_{min} in error detection*

- ¤ Let us assume that the sent code-word x is at the center of a circle with radius s.
- ¤ All received code-words that are created by 0 to s errors are points inside the circle or on the perimeter of the circle.
- ¤ All other valid code-words must be outside the circle

- For example: A code scheme has a Hamming distance d_{min} = 4.
  This code guarantees the detection of upto 3 errors (d = s + 1 or s = 3).

3.2.1.2 Linear Block Codes

- Almost all block codes belong to a subset of block codes called linear block codes.
- A linear block code is a code in which the XOR of 2 valid code-words creates another valid code-word.
  (XOR → Addition modulo-2)

Table 10.1 *A code for error detection*

Dataword	*Codeword*	*Dataword*	*Codeword*
00	000	10	101
01	011	11	110

- The code in Table 10.1 is a linear block code because the result of XORing any code-word with any other code-word is a valid code-word.
  For example, the XORing of the 2nd and 3rd code-words creates the 4th one.

3.2.1.2.1 Minimum Distance for Linear Block Codes

- Minimum Hamming distance is no. of 1s in the nonzero valid code-word with the smallest no. of 1s.
- In Table 10.1,
  The numbers of 1s in the nonzero code-words are 2, 2, and 2.
  So the minimum Hamming distance is $d_{min} = 2$.

3.2.1.3 Parity Check Code

- This code is a linear block code. This code can detect an odd number of errors.
- A k-bit data-word is changed to an n-bit code-word where n=k+1.
- One extra bit is called the parity-bit.
- The parity-bit is selected to make the total number of 1s in the code-word even.
- Minimum hamming distance d_{min} = 2. This means the code is a single-bit error-detecting code.

Table 10.2 *Simple parity-check code C(5, 4)*

Dataword	Codeword	Dataword	Codeword
0000	**00000**	1000	**10001**
0001	**00011**	1001	**10010**
0010	**00101**	1010	**10100**
0011	**00110**	1011	**10111**
0100	**01001**	1100	**11000**
0101	**01010**	1101	**11011**
0110	**01100**	1110	**11101**
0111	**01111**	1111	**11110**

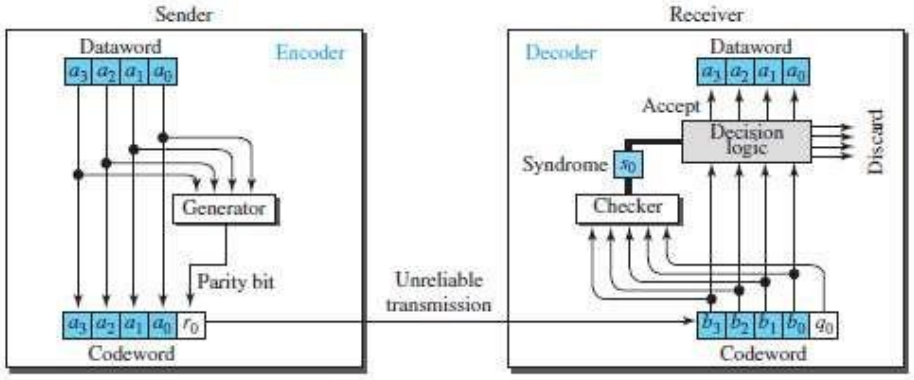


Figure 10.4 *Encoder and decoder for simple parity-check code*

- Here is how it works (Figure 10.4):

1) At Sender

- The encoder uses a generator that takes a copy of a 4-bit data-word (a_0, a_1, a_2, and a_3) and generates a parity-bit r_0.
- The encoder
  - → accepts a copy of a 4-bit data-word (a_0, a_1, a_2, and a_3) and
  - → generates a parity-bit r_0 using a generator
  - → generates a 5-bit code-word
- The parity-bit & 4-bit data-word are added to make the number of 1s in the code-word even.
- The addition is done by using the following:

$$r_0 = a_3 + a_2 + a_1 + a_0 \quad \text{(modulo-2)}$$

- The result of addition is the parity-bit.
  1) If the no. of 1s in data-word = even, result = 0. (r_0=0)
  2) If the no. of 1s in data-word = odd, result = 1. (r_0=1)
  3) In both cases, the total number of 1s in the code-word is even.
- The sender sends the code-word, which may be corrupted during transmission.

2) At Receiver

- The receiver receives a 5-bit word.
- The checker performs the same operation as the generator with one exception:
  The addition is done over all 5 bits.

$$s_0 = b_3 + b_2 + b_1 + b_0 + q_0 \quad \text{(modulo-2)}$$

- The result is called the syndrome bit (s_o).
- Syndrome bit = 0 when the no. of 1s in the received code-word is even; otherwise, it is 1.
- The syndrome is passed to the decision logic analyzer.
  1) If s_0=0, there is no error in the received code-word. The data portion of the received code-word is accepted as the data-word.
  2) If s_0=1, there is error in the received code-word. The data portion of the received code-word is discarded. The data-word is not created.

Example 3.2

Let us look at some transmission scenarios. Assume the sender sends the dataword 1011. The codeword created from this dataword is 10111, which is sent to the receiver. We examine five cases:

1. No error occurs; the received codeword is 10111. The syndrome is 0. The dataword 1011 is created.
2. One single-bit error changes a_1. The received codeword is 10011. The syndrome is 1. No dataword is created.
3. One single-bit error changes r_0. The received codeword is 10110. The syndrome is 1. No dataword is created. Note that although none of the dataword bits are corrupted, no dataword is created because the code is not sophisticated enough to show the position of the corrupted bit.
4. An error changes r_0 and a second error changes a_3. The received codeword is 00110. The syndrome is 0. The dataword 0011 is created at the receiver. Note that here the dataword is wrongly created due to the syndrome value. The simple parity-check decoder cannot detect an even number of errors. The errors cancel each other out and give the syndrome a value of 0.
5. Three bits—a_3, a_2, and a_1—are changed by errors. The received codeword is 01011. The syndrome is 1. The dataword is not created. This shows that the simple parity check, guaranteed to detect one single error, can also find any odd number of errors.

3.3 Cyclic Codes

- Cyclic codes are special linear block codes with one extra property:
  If a code-word is cyclically shifted (rotated), the result is another code-word.
  For ex: if code-word = 1011000 and we cyclically left-shift, then another code-word = 0110001.
- Let First-word = a_0 to a_6 and Second-word = b_0 to b_6, we can shift the bits by using the following:

$$b_1 = a_0 \quad b_2 = a_1 \quad b_3 = a_2 \quad b_4 = a_3 \quad b_5 = a_4 \quad b_6 = a_5 \quad b_0 = a_6$$

3.3.1 Cyclic Redundancy Check (CRC)

- CRC is a cyclic code that is used in networks such as LANs and WANs.

Table 10.3 *A CRC code with C(7, 4)*

Dataword	Codeword	Dataword	Codeword
0000	0000000	1000	1000101
0001	0001011	1001	1001110
0010	0010110	1010	1010011
0011	0011101	1011	1011000
0100	0100111	1100	1100010
0101	0101100	1101	1101001
0110	0110001	1110	1110100
0111	0111010	1111	1111111

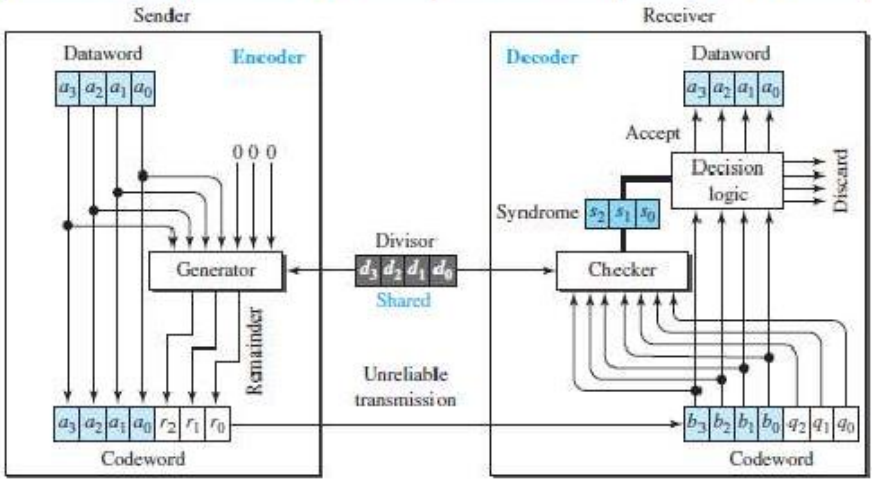


Figure 10.5 *CRC encoder and decoder*

- Let Size of data-word = k bits (here k=4).
  Size of code-word = n bits (here n=7).
  Size of divisor = n-k+1 bits (here n-k+1=4). (Augmented → increased)
- Here is how it works (Figure 10.5):

1) At Sender

- n-k 0s is appended to the data-word to create augmented data-word. (here n-k=3).
- The augmented data-word is fed into the generator (Figure 10.6).
- The generator divides the augmented data-word by the divisor.
- The remainder is called check-bits ($r_2r_1r_0$).
- The check-bits ($r_2r_1r_0$) are appended to the data-word to create the code-word.

2) At Receiver

- The possibly corrupted code-word is fed into the checker.
- The checker is a replica of the generator.
- The checker divides the code-word by the divisor.
- The remainder is called syndrome bits ($r_2r_1r_0$).
- The syndrome bits are fed to the decision-logic-analyzer.
- The decision-logic-analyzer performs following functions:

i) For No Error

- If all syndrome-bits are 0s, the received code-word is accepted.
- Data-word is extracted from received code-word (Figure 10.7a).

ii) For Error

- If all syndrome-bits are not 0s, the received code-word is discarded (Figure 10.7b).

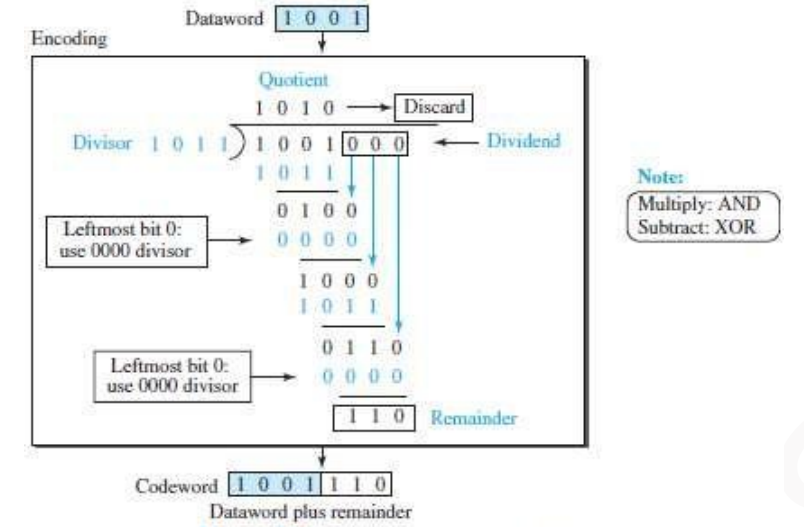


Figure 10.6 *Division in CRC encoder*

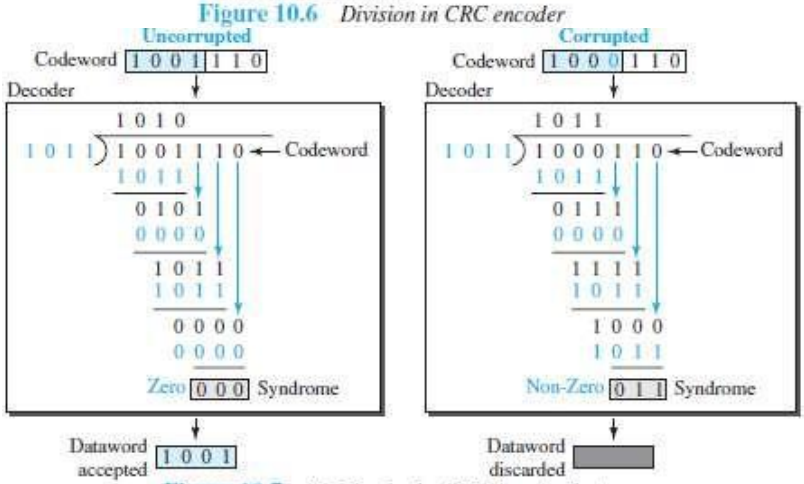
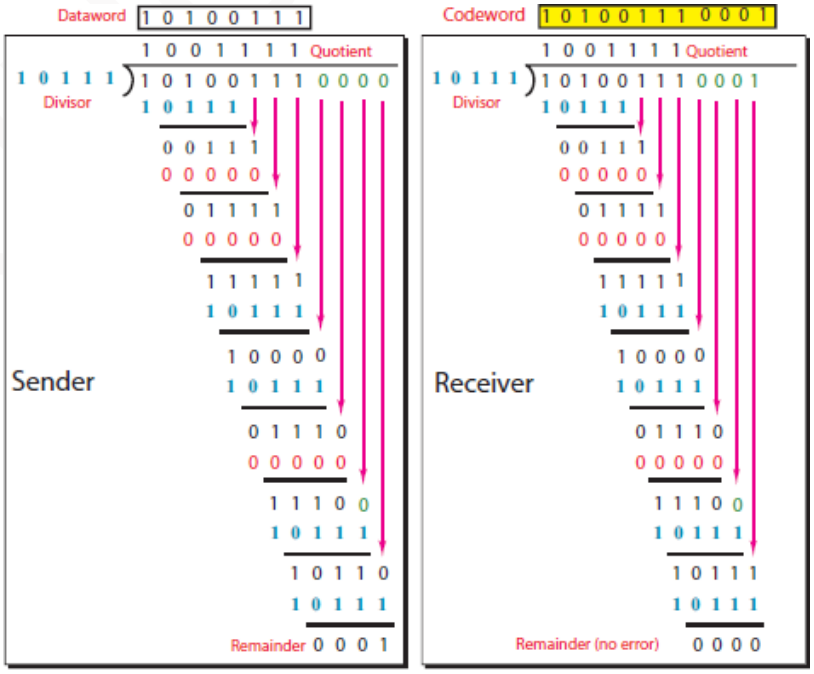


Figure 10.7 *Division in the CRC decoder for two cases*

Example 3.3

Given the dataword 10100111 and the divisor 10111, show the generation of the CRC codeword at the sender site (using binary division).



3.3.2 Polynomials

- A pattern of 0s and 1s can be represented as a polynomial with coefficients of 0 and 1 (Figure 10.8).
- The power of each term shows the position of the bit; the coefficient shows the value of the bit.

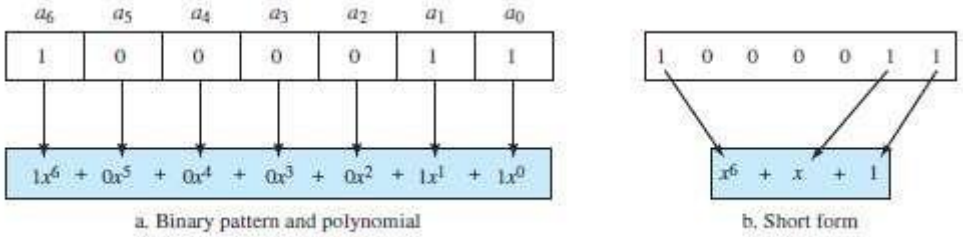


Figure 10.8 *A polynomial to represent a binary word*

3.3.3 Cyclic Code Encoder Using Polynomials

- Let Data-word = 1001 = x^3+1.
  Divisor = 1011 = x^3+x+1.
- In polynomial representation, the divisor is referred to as generator polynomial t(x) (Figure 10.9).

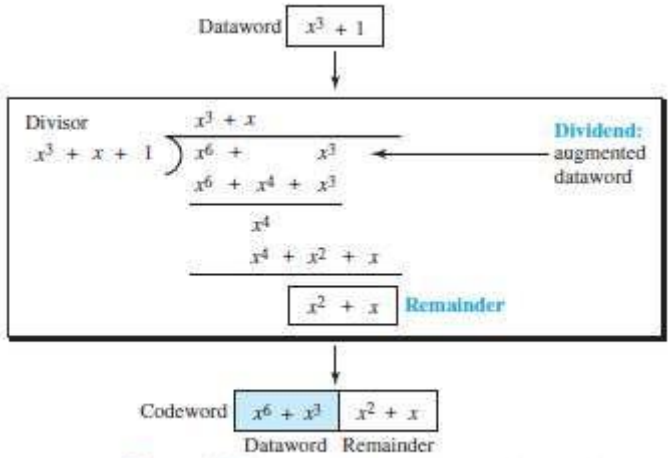


Figure 10.9 *CRC division using polynomials*

3.3.4 Cyclic Code Analysis

- We define the following, where *f*(*x*) is a polynomial with binary coefficients:

Dataword: *d*(*x*) **Codeword: *c*(*x*)** **Generator: *g*(*x*)** **Syndrome: *s*(*x*)** **Error: *e*(*x*)**

In a cyclic code,

1. If $s(x) \neq 0$, one or more bits is corrupted.
2. If $s(x) = 0$, either
   a. No bit is corrupted, or
   b. Some bits are corrupted, but the decoder failed to detect them.

Single Bit Error

- If the generator has more than one term and the coefficient of x^0 is 1, all single-bit errors can be caught.

Two Isolated Single-Bit Errors

- If a generator cannot divide x^i+1 (t between 0 & n-1), then all isolated double errors can be detected (Figure 10.10).

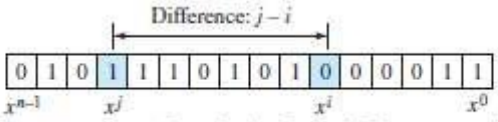


Figure 10.10 *Representation of two isolated single-bit errors using polynomials*

Odd Numbers of Errors

• A generator that contains a factor of x+1 can detect all odd-numbered errors.

A good polynomial generator needs to have the following characteristics:

1. **It should have at least two terms.**
2. **The coefficient of the term x^0 should be 1.**
3. **It should not divide $x^t + 1$, for t between 2 and $n - 1$.**
4. **It should have the factor $x + 1$.**

Standard Polynomials

Table 10.4 *Standard polynomials*

Name	*Polynomial*	*Used in*
CRC-8	x^8+x^2+x+1 **100000111**	ATM header
CRC-10	$x^{10}+x^9+x^5+x^4+x^2+1$ **11000110101**	ATM AAL
CRC-16	$x^{16}+x^{12}+x^5+1$ **10001000000100001**	HDLC
CRC-32	$x^{32}+x^{26}+x^{23}+x^{22}+x^{16}+x^{12}+x^{11}+x^{10}+x^8+x^7+x^5+x^4+x^2+x+1$ **10000010011000001000111011011 0111**	LANs

3.3.5 Advantages of Cyclic Codes

• The cyclic codes have a very good performance in detecting
→ single-bit errors
→ double errors
→ odd number of errors and
→ burst-errors.

• They can easily be implemented in hardware and software. They are fast when implemented in hardware.

3.4 Checksum

- Checksum is an error-detecting technique.
- In the Internet,
  - → The checksum is mostly used at the network and transport layer.
  - → The checksum is not used in the data link layer.
- Like linear and cyclic codes, the checksum is based on the concept of redundancy.
- Here is how it works (Figure 10.15):

1) At Source

- Firstly the message is divided into m-bit units.
- Then, the generator creates an extra m-bit unit called the checksum.
- The checksum is sent with the message.

2) At Destination

- The checker creates a new checksum from the combination of the message and sent-checksum.
  - i) If the new checksum is all 0s, the message is accepted.
  - ii) If the new checksum is not all 0s, the message is discarded.

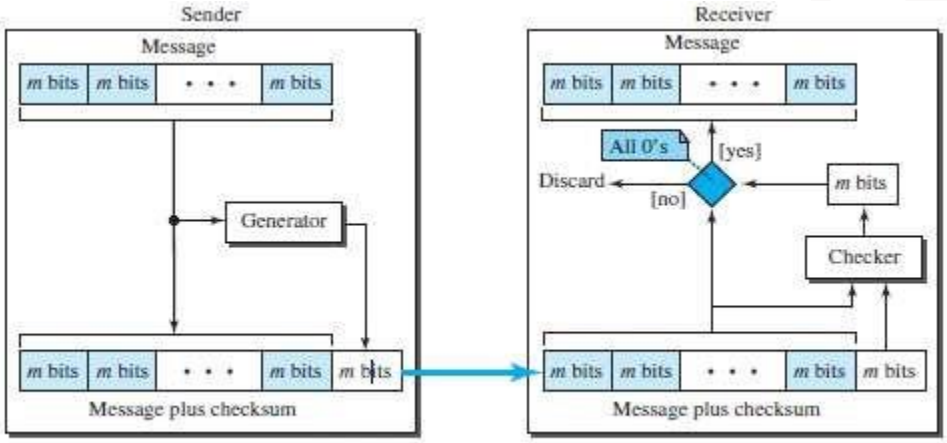


Figure 10.15 *Checksum*

3.4.1 Concept of Checksum

Consider the following example:

Example 3.4

- Our data is a list of five 4-bit numbers that we want to send to a destination.
- In addition to sending these numbers, we send the sum of the numbers.
- For example:
  - Let set of numbers = (7, 11, 12, 0, 6).
  - We send (7, 11, 12, 0, 6, 36), where 36 is the sum of the original numbers.
- The receiver adds the five numbers and compares the result with the sum.
- If the result & the sum are the same,
  - The receiver assumes no error, accepts the five numbers, and discards the sum.
  - Otherwise, there is an error somewhere and the data are not accepted.

Example 3.5

- To make the job of the receiver easy if we send the negative (complement) of the sum, called the checksum.
- In this case, we send (7, 11, 12, 0, 6, -36).
- The receiver can add all the numbers received (including the checksum).
- If the result is 0, it assumes no error; otherwise, there is an error.

3.4.1.1 One's Complement

- The previous example has one major drawback.
  - All of our data can be written as a 4-bit word (they are less than 15) except for the checksum.
- Solution: Use one's complement arithmetic.
  - ➢ We can represent unsigned numbers between 0 and 2^n-1 using only n bits.
  - ➢ If the number has more than n bits, the extra leftmost bits need to be added to the n rightmost bits (wrapping).
  - ➢ A negative number can be represented by inverting all bits (changing 0 to 1 and 1 to 0).
  - ➢ This is the same as subtracting the number from 2^n-1.

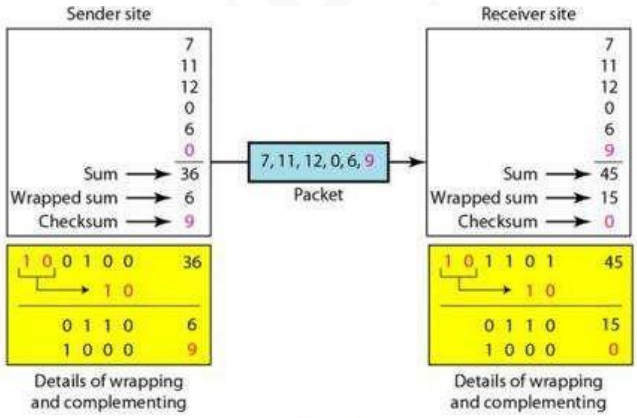


Figure 10.16

- Here is how it works (Figure 10.16):

1) At Sender

- ➢ The sender initializes the checksum to 0 and adds all data items and the checksum.
- ➢ The result is 36.
- ➢ However, 36 cannot be expressed in 4 bits.
- ➢ The extra two bits are wrapped and added with the sum to create the wrapped sum value 6.
- ➢ The sum is then complemented, resulting in the checksum value 9 (15 - 6 = 9).
- ➢ The sender now sends six data items to the receiver including the checksum 9.

2) At Receiver

- ➢ The receiver follows the same procedure as the sender.
- ➢ It adds all data items (including the checksum); the result is 45.
- ➢ The sum is wrapped and becomes 15. The wrapped sum is complemented and becomes 0.
- ➢ Since the value of the checksum is 0, this means that the data is not corrupted. The receiver drops the checksum and keeps the other data items.
- ➢ If the checksum is not zero, the entire packet is dropped.

3.4.1.2 Internet Checksum

- Traditionally, the Internet has been using a 16-bit checksum.
- The sender or the receiver uses five steps.

Table 10.5 *Procedure to calculate the traditional checksum*

Sender	*Receiver*
1. The message is divided into 16-bit words.	1. The message and the checksum are received.
2. The value of the checksum word is initially set to zero.	2. The message is divided into 16-bit words.
3. All words including the checksum are added using one's complement addition.	3. All words are added using one's complement addition.
4. The sum is complemented and becomes the checksum.	4. The sum is complemented and becomes the new checksum.
5. The checksum is sent with the data.	5. If the value of the checksum is 0, the message is accepted; otherwise, it is rejected.

3.4.1.3 Algorithm

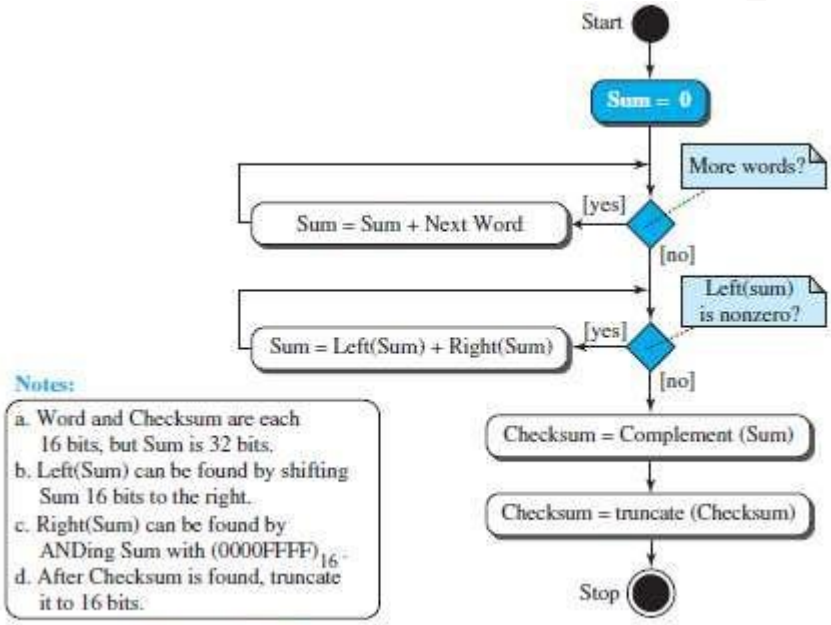


Figure 10.17 *Algorithm to calculate a traditional checksum*

3.4.2 Other Approaches to the Checksum

- If two 16-bit items are transposed in transmission, the checksum cannot catch this error.
- The reason is that the traditional checksum is not weighted: it treats each data item equally.
- In other words, the order of data items is immaterial to the calculation.
- Two approaches have been used to prevent this problem: 1)Fletcher and 2)Adler

3.4.2.1 Fletcher Checksum

- The Fletcher checksum was devised to weight each data item according to its position.
- Fletcher has proposed two algorithms: 8-bit and 16-bit (Figure 10.18).
- The first, 8-bit Fletcher, calculates on 8-bit data items and creates a 16-bit checksum.
  The second, 16-bit Fletcher, calculates on 16-bit data items and creates a 32-bit checksum.
- The 8-bit Fletcher is calculated over data octets (bytes) and creates a 16-bit checksum.
- The calculation is done modulo 256 (2^8), which means the intermediate results are divided by 256 and the remainder is kept.
- The algorithm uses two accumulators, L and R.
- The first simply adds data items together;
  The second adds a weight to the calculation.

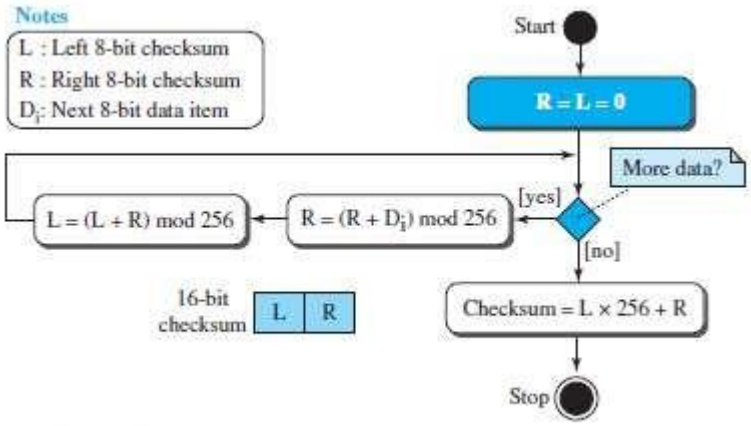


Figure 10.18 *Algorithm to calculate an 8-bit Fletcher checksum*

3.4.2.2 Adler Checksum

- The Adler checksum is a 32-bit checksum.
- It is similar to the 16-bit Fletcher with three differences (Figure 10.19).
  1) Calculation is done on single bytes instead of 2 bytes at a time.
  2) The modulus is a prime number (65,521) instead of 65,536.
  3) L is initialized to 1 instead of 0.
- A prime modulo has a better detecting capability in some combinations of data.

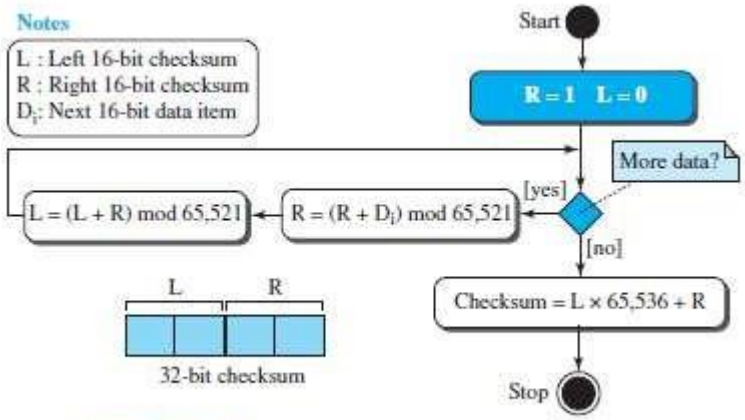


Figure 10.19 *Algorithm to calculate an Adler checksum*

3.5 FORWARD ERROR CORRECTION

• Retransmission of corrupted and lost packets is not useful for real-time multimedia transmission because it creates an unacceptable delay in reproducing: we need to wait until the lost or corrupted packet is resent.
• We need to correct the error or reproduce the packet immediately.
• Several schemes have been designed and used that are collectively referred to as forward error correction (FEC) techniques.

3.5.1 Using Hamming Distance

• To detect t errors, we need to have $d_{min} = 2t + 1$ (Figure 10.20).
• In other words, if we want to correct 10 bits in a packet, we need to make the minimum hamming distance 21 bits, which means a lot of redundant bits need to be sent with the data.

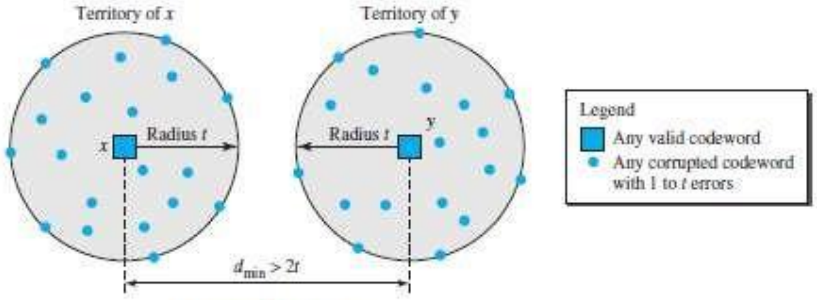


Figure 10.20 *Hamming distance for error correction*

3.5.2 Using XOR

• Use the property of the exclusive OR operation as shown below.

$$\mathbf{R} = \mathbf{P}_1 \oplus \mathbf{P}_2 \oplus \ldots \oplus \mathbf{P}_i \oplus \ldots \oplus \mathbf{P}_N \quad \rightarrow \quad \mathbf{P}_i = \mathbf{P}_1 \oplus \mathbf{P}_2 \oplus \ldots \oplus \mathbf{R} \oplus \ldots \oplus \mathbf{P}_N$$

• We divide a packet into N chunks, create the exclusive OR of all the chunks and send $N + 1$ chunks.
• If any chunk is lost or corrupted, it can be created at the receiver site.
• If N = 4, it means that we need to send 25 percent extra data and be able to correct the data if only one out of four chunks is lost.

3.5.3 Chunk Interleaving

• Another way to achieve FEC in multimedia is to allow some small chunks to be missing at the receiver.
• We cannot afford to let all the chunks belonging to the same packet be missing.
However, we can afford to let one chunk be missing in each packet.

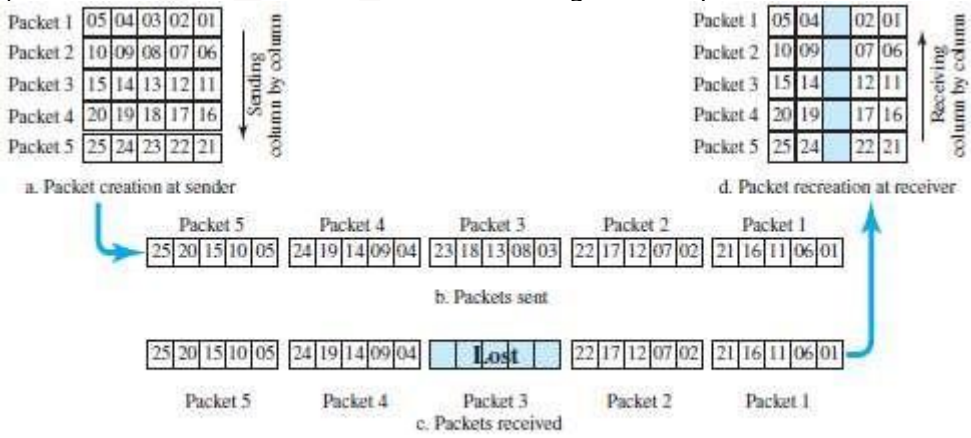


Figure 10.21 *Interleaving*

• In Figure 10.21, each packet is divided into 5 chunks (normally the number is much larger).
• Then, we can create data chunk-by-chunk (horizontally), but combine the chunks into packets vertically.
• In this case, each packet sent carries a chunk from several original packets.
• If the packet is lost, we miss only one chunk in each packet.
• Normally, missing of a chunk is acceptable in multimedia communication.

3.5.4 Combining Hamming Distance and Interleaving

• Hamming distance and interleaving can be combined.
• Firstly, we can create n-bit packets that can correct t-bit errors.
• Then, we interleave m rows and send the bits column-by-column.
• In this way, we can automatically correct burst-errors up to m×t bit errors.

3.5.5 Compounding High- and Low-Resolution Packets

• Another solution is to create a duplicate of each packet with a low-resolution redundancy and combine the redundant version with the next packet.
• For example (Figure 10.22):
We can create 4 low-resolution packets out of 5 high-resolution packets and send them (Fig 10.22).
• If a packet is lost, we can use the low-resolution version from the next packet.
• In this method, if the last packet is lost, it cannot be recovered, but we use the low-resolution version of a packet if the lost packet is not the last one.
• The audio and video reproduction does not have the same quality, but the lack of quality is not recognized most of the time.

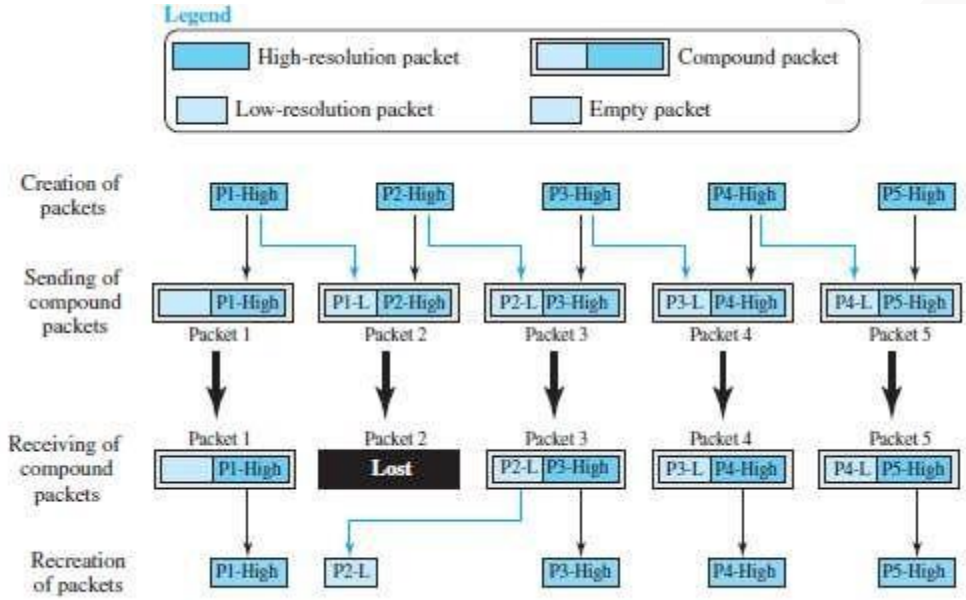


Figure 10.22 *Compounding high- and low-resolution packets*

MODULE 2(CONT.): DATA LINK CONTROL

3.6 DLC SERVICES

• The data link control (DLC) deals with procedures for communication between two adjacent nodes i.e. node-to-node communication.
• Data link control functions include 1) Framing and 2) Flow control and 3) Error control.

3.6.1 Framing

• A frame is a group of bits.
• Framing means organizing the bits into a frame that are carried by the physical layer.
• The data-link-layer needs to form frames, so that each frame is distinguishable from another.
• Framing separates a message from other messages by adding sender-address & destination-address.
• The destination-address defines where the packet is to go.
The sender-address helps the recipient acknowledge the receipt.
• Q: Why the whole message is not packed in one frame?
Ans: Large frame makes flow and error-control very inefficient.
Even a single-bit error requires the re-transmission of the whole message.
When a message is divided into smaller frames, a single-bit error affects only that small frame.

(Our postal system practices a type of framing. The simple act of inserting a letter into an envelope separates one piece of information from another; the envelope serves as the delimiter. In addition, each envelope defines the sender and receiver addresses since the postal system is a many-to-many carrier facility).

3.6.1.1 Frame Size

• Two types of frames:

1) Fixed Size Framing

- There is no need for defining boundaries of frames; the size itself can be used as a delimiter.
- For example: ATM WAN uses frames of fixed size called cells.

2) Variable Size Framing

- We need to define the end of the frame and the beginning of the next frame.
- Two approaches are used: 1) Character-oriented approach
  2) Bit-oriented approach.

3.6.1.2 Character Oriented Framing

• Data to be carried are 8-bit characters from a coding system such as ASCII (Figure 11.1).
• The header and the trailer are also multiples of 8 bits.
  1) Header carries the source and destination-addresses and other control information.
  2) Trailer carries error-detection or error-correction redundant bits.
• To separate one frame from the next frame, an 8- bit (I-byte) flag is added at the beginning and the end of a frame.
• The flag is composed of protocol-dependent special characters.
• The flag signals the start or end of a frame.

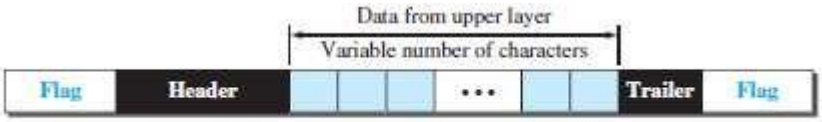


Figure 11.1 *A frame in a character-oriented protocol*

• Problem:
  ➢ Character-oriented framing is suitable when only text is exchanged by the data-link-layers.
  ➢ However, if we send other type of information (say audio/video), then any pattern used for the flag can also be part of the information.
  ➢ If the flag-pattern appears in the data-section, the receiver might think that it has reached the end of the frame.

Solution: A byte-stuffing is used.
  (Byte stuffing → character stuffing)

• In byte stuffing, a special byte is added to the data-section of the frame when there is a character with the same pattern as the flag.
• The data-section is stuffed with an extra byte. This byte is called the escape character (ESC), which has a predefined bit pattern.
• When a receiver encounters the ESC character, the receiver
  → removes ESC character from the data-section and
  → treats the next character as data, not a delimiting flag.
• Problem:
  ➢ What happens if the text contains one or more escape characters followed by a flag?
  ➢ The receiver removes the escape character, but keeps the flag, which is incorrectly interpreted as the end of the frame.

Solution:
  ➢ Escape characters part of the text must also be marked by another escape character (Fig 11.2).

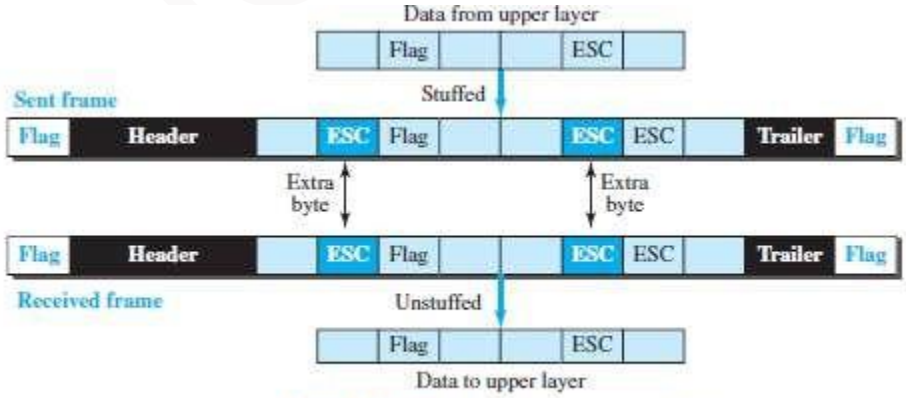


Figure 11.2 *Byte stuffing and unstuffing*

• In short, byte stuffing is the process of adding one extra byte whenever there is a flag or escape character in the text.

3.6.1.3 Bit Oriented Framing

• The data-section of a frame is a sequence of bits to be interpreted by the upper layer as text, audio, video, and so on.

• However, in addition to headers and trailers, we need a delimiter to separate one frame from the other.

• Most protocols use a special 8-bit pattern flag 01111110 as the delimiter to define the beginning and the end of the frame (Figure 11.3).

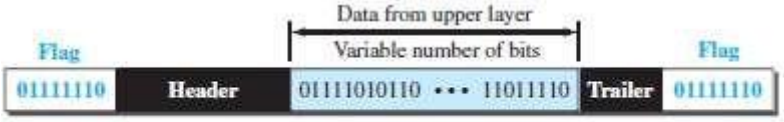


Figure 11.3 *A frame in a bit-oriented protocol*

• Problem:

- If the flag-pattern appears in the data-section, the receiver might think that it has reached the end of the frame.

Solution: A bit-stuffing is used.

• In bit stuffing, if a 0 and five consecutive 1 bits are encountered, an extra 0 is added. This extra stuffed bit is eventually removed from the data by the receiver. (Figure 11.4).

• This guarantees that the flag field sequence does not inadvertently appear in the frame.

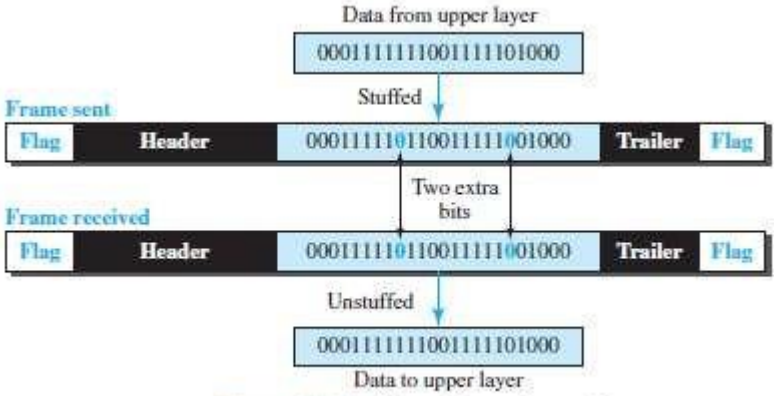


Figure 11.4 *Bit stuffing and unstuffing*

• In short, bit stuffing is the process of adding one extra 0 whenever five consecutive 1s follow a 0 in the data, so that the receiver does not mistake the pattern 0111110 for a flag.

3.6.2 Flow Control and Error Control

• One of the responsibilities of the DLC sublayer is flow and error control at the data-link layer.

3.6.2.1 Flow Control

• Whenever an entity produces items and another entity consumes them, there should be a balance between production and consumption rates.

• If the items are produced faster than they can be consumed, the consumer can be overwhelmed and may need to discard some items.

• We need to prevent losing the data items at the consumer site.

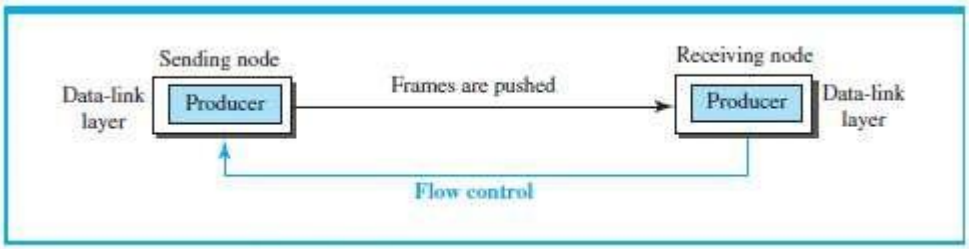


Figure 11.5 *Flow control at the data-link layer*

• At the sending node, the data-link layer tries to push frames toward the data-link layer at the receiving node (Figure 11.5).

• If the receiving node cannot process and deliver the packet to its network at the same rate that the frames arrive, it becomes overwhelmed with frames.

• Here, flow control can be feedback from the receiving node to the sending node to stop or slow down pushing frames.

3.6.2.1.1 Buffers

• Flow control can be implemented by using buffer.

• A buffer is a set of memory locations that can hold packets at the sender and receiver.

• Normally, two buffers can be used.

1) First buffer at the sender.
2) Second buffer at the receiver.

• The flow control communication can occur by sending signals from the consumer to the producer.

• When the buffer of the receiver is full, it informs the sender to stop pushing frames.

3.6.2.2 Error Control

• Error-control includes both error-detection and error-correction.

• Error-control allows the receiver to inform the sender of any frames lost/damaged in transmission.

• A CRC is

→ added to the frame header by the sender and
→ checked by the receiver.

• At the data-link layer, error control is normally implemented using one of the following two methods.

1) First method: If the frame is corrupted, it is discarded;
   If the frame is not corrupted, the packet is delivered to the network layer.
   This method is used mostly in wired LANs such as Ethernet.
2) Second method: If the frame is corrupted, it is discarded;
   If the frame is not corrupted, an acknowledgment is sent to the sender.
   Acknowledgment is used for the purpose of both flow and error control.

3.6.2.2.1 Combination of Flow and Error Control

• Flow and error control can be combined.

• The acknowledgment that is sent for flow control can also be used for error control to tell the sender the packet has arrived uncorrupted.

• The lack of acknowledgment means that there is a problem in the sent frame.

• A frame that carries an acknowledgment is normally called an ACK to distinguish it from the data frame.

3.6.3 Connectionless and Connection-Oriented

• A DLC protocol can be either connectionless or connection-oriented.

1) Connectionless Protocol

- Frames are sent from one node to the next without any relationship between the frames; each frame is independent.
- The term connectionless does not mean that there is no physical connection (transmission medium) between the nodes; it means that there is no connection between frames.
- The frames are not numbered and there is no sense of ordering.
- Most of the data-link protocols for LANs are connectionless protocols.

2) Connection Oriented Protocol

- A logical connection should first be established between the two nodes (setup phase).
- After all frames that are somehow related to each other are transmitted (transfer phase), the logical connection is terminated (teardown phase).
- The frames are numbered and sent in order.
- If the frames are not received in order, the receiver needs to wait until all frames belonging to the same set are received and then deliver them in order to the network layer.
- Connection oriented protocols are rare in wired LANs, but we can see them in some point-to-point protocols, some wireless LANs, and some WANs.

3.7 DATA LINK LAYER PROTOCOLS

• Traditionally 2 protocols have been defined for the data-link layer to deal with flow and error control:
1) Simple Protocol and 2) Stop-and-Wait Protocol.
• The behavior of a data-link-layer protocol can be better shown as a finite state machine (FSM).
• An FSM is a machine with a finite number of states (Figure 11.6).
• The machine is always in one of the states until an event occurs.
• Each event is associated with 2 reactions:
1) Defining the list (possibly empty) of actions to be performed.
2) Determining the next state (which can be the same as the current state).
• One of the states must be defined as the initial state, the state in which the machine starts when it turns on.

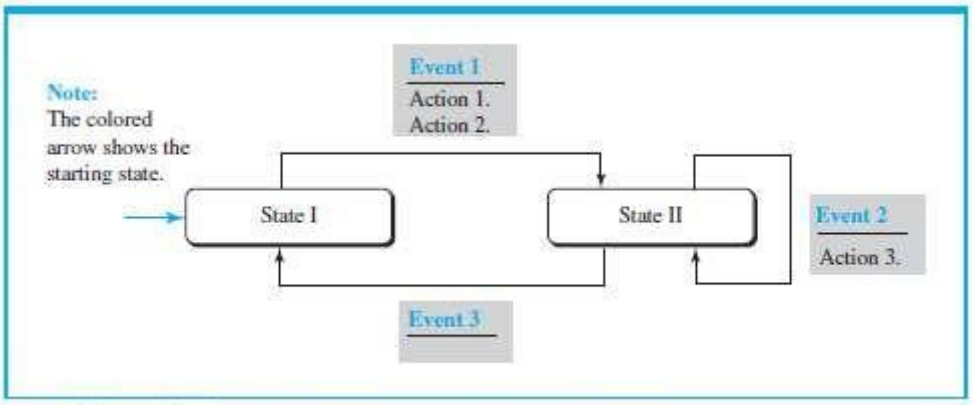


Figure 11.6 *Connectionless and connection-oriented service represented as FSMs*

3.7.1 Simplest Protocol

• Assumptions:
- The protocol has no flow-control or error-control.
- The protocol is a unidirectional protocol (in which frames are traveling in only one direction).
- The receiver can immediately handle any frame it receives.

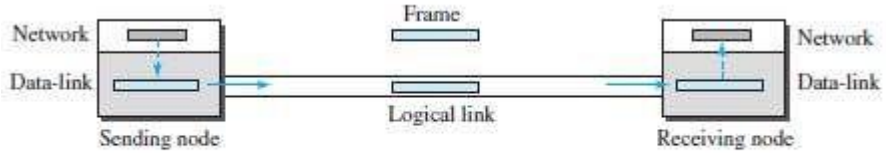


Figure 11.7 *Simple protocol*

3.7.1.1 Design

• Here is how it works (Figure 11.7):

1) At Sender

¤ The data-link-layer
→ gets data from its network-layer
→ makes a frame out of the data and
→ sends the frame.

2) At Receiver

¤ The data-link-layer
→ receives a frame from its physical layer
→ extracts data from the frame and
→ delivers the data to its network-layer.

- Data-link-layers of sender & receiver provide transmission services for their network-layers.
- Data-link-layers use the services provided by their physical layers for the physical transmission of bits.

3.7.1.2 FSMs

- Two main requirements:
  1) The sender-site cannot send a frame until its network-layer has a data packet to send.
  2) The receiver-site cannot deliver a data packet to its network-layer until a frame arrives.
- These 2 requirements are shown using two FSMs.
- Each FSM has only one state, the ready state.

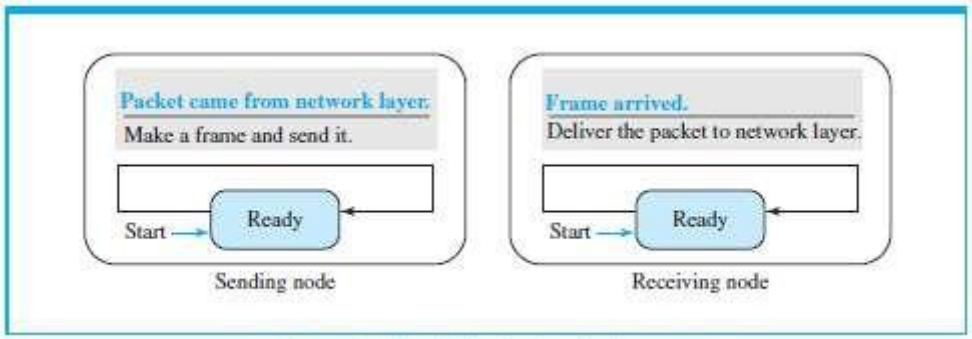


Figure 11.8 *FSMs for the simple protocol*

- Here is how it works (Figure 11.8):

1) At Sending Machine

➢ The sending machine remains in the ready state until a request comes from the process in the network layer.

➢ When this event occurs, the sending machine encapsulates the message in a frame and sends it to the receiving machine.

2) At Receiving Machine

➢ The receiving machine remains in the ready state until a frame arrives from the sending machine.

➢ When this event occurs, the receiving machine decapsulates the message out of the frame and delivers it to the process at the network layer.

Example 3.6

Figure 11.9 shows an example of communication using this protocol. It is very simple. The sender sends frames one after another without even thinking about the receiver.

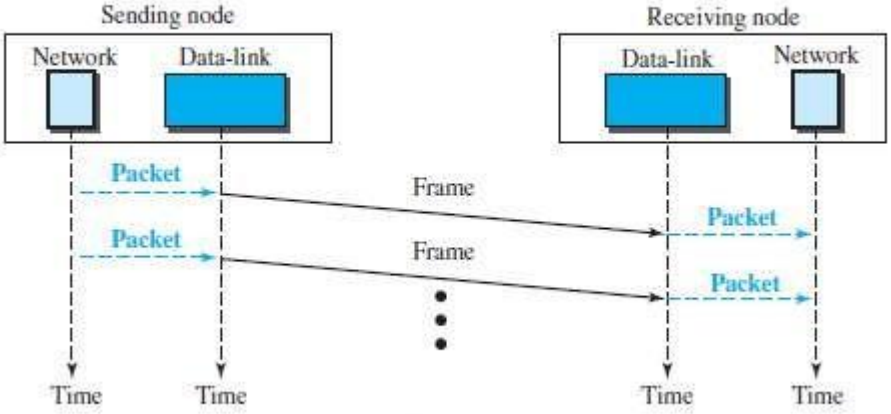


Figure 11.9 *Flow diagram*

3.7.2 Stop & Wait Protocol

- This uses both flow and error control.
- Normally, the receiver has limited storage-space.
- If the receiver is receiving data from many sources, the receiver may
  → be overloaded with frames &
  → discard the frames.
- To prevent the receiver from being overloaded with frames, we need to tell the sender to slow down.

3.7.2.1 Design

1) At Sender

- The sender
  → sends one frame & starts a timer
  → keeps a copy of the sent-frame and
  → waits for ACK-frame from the receiver (okay to go ahead).
- Then,
  1) If an ACK-frame arrives before the timer expires, the timer is stopped and the sender sends the next frame.
  Also, the sender discards the copy of the previous frame.
  2) If the timer expires before ACK-frame arrives, the sender resends the previous frame and restarts the timer

2) At Receiver

- To detect corrupted frames, a CRC is added to each data frame.
- When a frame arrives at the receiver-site, the frame is checked.
- If frame's CRC is incorrect, the frame is corrupted and discarded.
- The silence of the receiver is a signal for the sender that a frame was either corrupted or lost.

3.7.2.2 FSMs

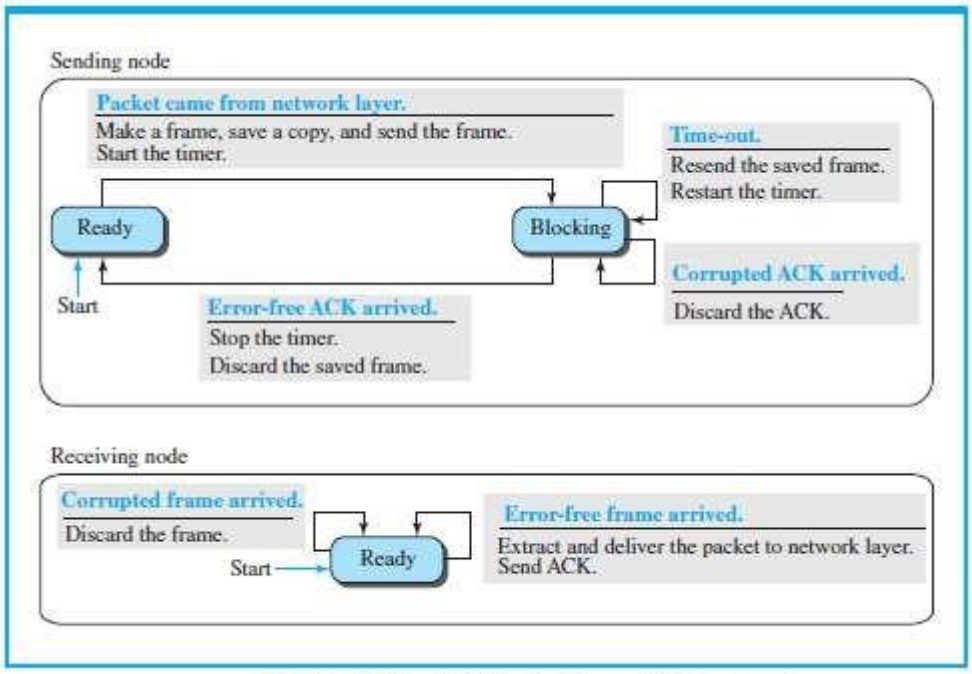


Figure 11.11 *FSM for the Stop-and-Wait protocol*

- Here is how it works (Figure 11.11):

1) Sender States

- Sender is initially in the ready state, but it can move between the ready and blocking state.

i) Ready State: When the sender is in this state, it is only waiting for a packet from the network layer.

If a packet comes from the network layer, the sender creates a frame, saves a copy of the frame, starts the only timer and sends the frame. The sender then moves to the blocking state.

ii) Blocking State: When the sender is in this state, three events can occur:

a) If a time-out occurs, the sender resends the saved copy of the frame and restarts the timer.

b) If a corrupted ACK arrives, it is discarded.

c) If an error-free ACK arrives, the sender stops the timer and discards the saved copy of the frame. It then moves to the ready state.

2) Receiver

• The receiver is always in the ready state. Two events may occur:

a) If an error-free frame arrives, the message in the frame is delivered to the network layer and an ACK is sent.

b) If a corrupted frame arrives, the frame is discarded.

Example 3.7

Figure 11.12 shows an example. The first frame is sent and acknowledged. The second frame is sent, but lost. After time-out, it is resent. The third frame is sent and acknowledged, but the acknowledgment is lost. The frame is resent. However, there is a problem with this scheme. The network layer at the receiver site receives two copies of the third packet, which is not right. In the next section, we will see how we can correct this problem using sequence numbers and acknowledgment numbers.

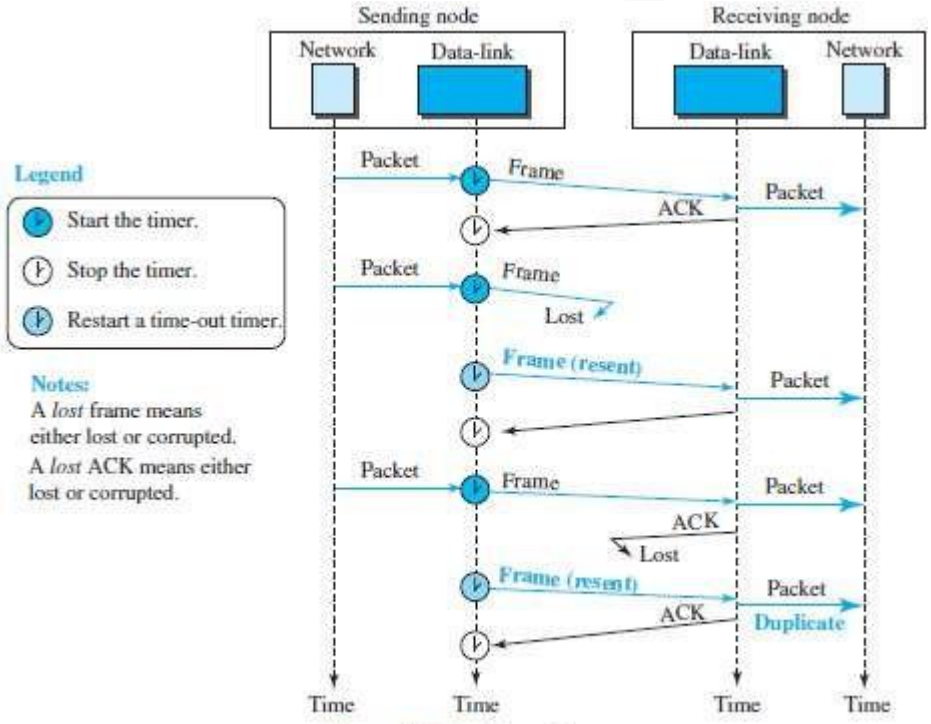


Figure 11.12 *Flow diagram*

3.7.2.3 Sequence and Acknowledgment Numbers

• Q: How to deal with corrupted-frame?
Ans: If the corrupted-frame arrives at the receiver-site, then the frame is simply discarded.
• Q: How to deal with lost-frames?
Ans: If the receiver receives out-of-order data-frame, then it means that frames were lost. .'. The lost-frames need to be resent.
• Problem in Stop and Wait protocols:
1) There is no way to identify a frame.
2) The received-frame could be the correct one, or a duplicate, or a frame out of order.
Solution: 1) Use sequence-number for each data frame.
2) Use Acknowledgment-number for each ACK frame.

Sequence Numbers

- Frames need to be numbered. This is done by using sequence-numbers.
- A sequence-number field is added to the data-frame.

Acknowledgment Numbers

- An acknowledgment-number field is added to the ACK-frame.
- Sequence numbers are 0, 1, 0, 1, 0, 1, . . .
  The acknowledgment numbers can also be 1, 0, 1, 0, 1, 0, …
- The acknowledgment-numbers always announce the sequence-number of the next frame expected by the receiver.
- For example,
  If frame-0 has arrived safely, the receiver sends an ACK-frame with acknowledgment-1 (meaning frame-1 is expected next).

Example 3.8

Figure 11.13 shows how adding sequence numbers and acknowledgment numbers can prevent duplicates. The first frame is sent and acknowledged. The second frame is sent, but lost. After time-out, it is resent. The third frame is sent and acknowledged, but the acknowledgment is lost. The frame is resent.

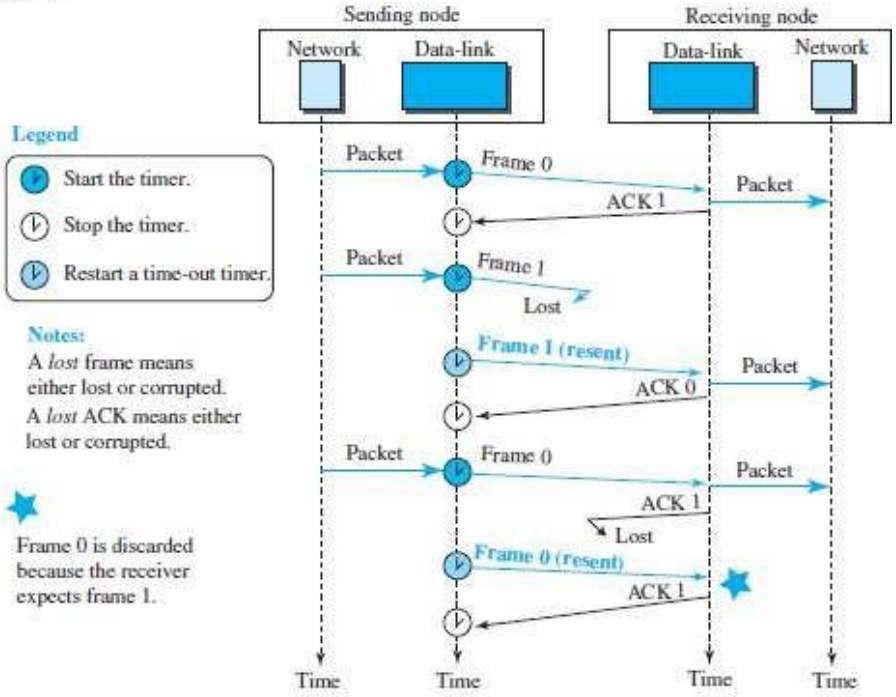


Figure 11.13 *Flow diagram*

3.7.3 Piggybacking

• A technique called piggybacking is used to improve the efficiency of the bidirectional protocols.
• The data in one direction is piggybacked with the acknowledgment in the other direction.
• In other words, when node A is sending data to node B, Node A also acknowledges the data received from node B.

3.8 High-Level Data Link Control (HDLC)

• HDLC is a bit-oriented protocol for communication over point-to-point and multipoint links.
• HDLC implements the ARQ mechanisms.

3.8.1 Configurations and Transfer Modes

• HDLC provides 2 common transfer modes that can be used in different configurations:
1) Normal response mode (NRM)
2) Asynchronous balanced mode (ABM).

NRM

• The station configuration is unbalanced (Figure 11.14).
• We have one primary station and multiple secondary stations.
• A primary station can send commands, a secondary station can only respond.
• The NRM is used for both point-to-point and multiple-point links.

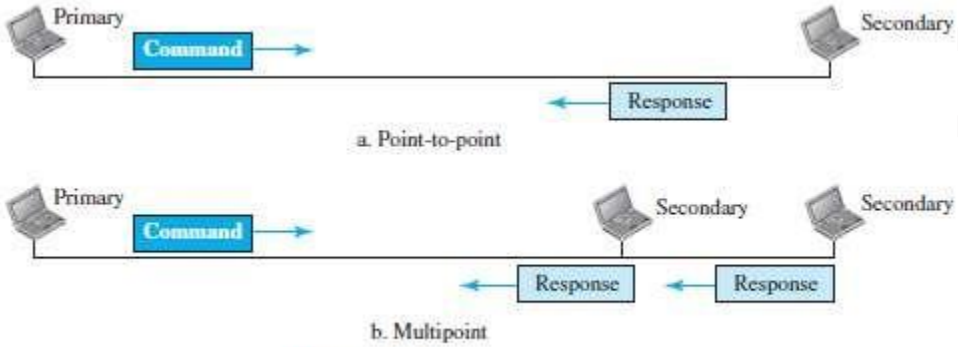


Figure 11.14 *Normal response mode*

ABM

• The configuration is balanced (Figure 11.15).
• Link is point-to-point, and each station can function as a primary and a secondary (acting as peers).
• This is the common mode today.

Figure 11.15 *Asynchronous balanced mode*

3.8.2 Framing

• To provide the flexibility necessary to support all the options possible in the modes and configurations, HDLC defines three types of frames:

1) Information frames (I-frames): are used to transport user data and control information relating to user data (piggybacking).

2) Supervisory frames (S-frames): are used only to transport control information.

3) Unnumbered frames (U-frames): are reserved for system management.
Information carried by U-frames is intended for managing the link itself.

• Each type of frame serves as an envelope for the transmission of a different type of message.

3.8.2.1 Frame Format

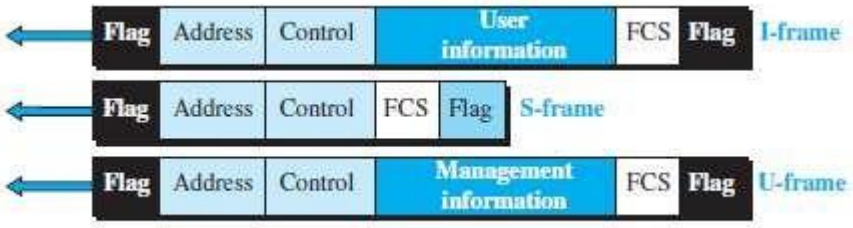


Figure 11.16 *HDLC frames*

• Various fields of HDLC frame are:

1) Flag Field
- This field has a synchronization pattern 01111110.
- This field identifies both the beginning and the end of a frame.

2) Address Field
- This field contains the address of the secondary station.
- If a primary station created the frame, it contains a to-address.
- If a secondary creates the frame, it contains a from-address.
- This field can be 1 byte or several bytes long, depending on the needs of the network.

3) Control Field
- This field is one or two bytes used for flow and error control.

4) Information Field
- This field contains the user's data from the network-layer or management information.
- Its length can vary from one network to another.

5) FCS Field
- This field is the error-detection field. (FCS → Frame Check Sequence)
- This field can contain either a 2- or 4-byte standard CRC.

3.8.2.1.1 Control Fields of HDLC Frames

• The control field determines the type of frame and defines its functionality (Figure 11.17).

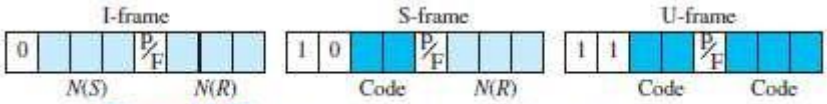


Figure 11.17 *Control field format for the different frame types*

1) Control Field for I-Frames

• I-frames are designed to carry user data from the network-layer.
• In addition, they can include flow and error-control information (piggybacking).
• The subfields in the control field are:

1) The first bit defines the type.
   If the first bit of the control field is 0, this means the frame is an I-frame.
2) The next 3 bits N(S) define the sequence-number of the frame.
   With 3 bits, we can define a sequence-number between 0 and 7
3) The last 3 bits N(R) correspond to the acknowledgment-number when piggybacking is used.
4) The single bit between N(S) and N(R) is called the P/F bit.
   The P/F field is a single bit with a dual purpose. It can mean poll or final.
   i) It means poll when the frame is sent by a primary station to a secondary (when the address field contains the address of the receiver).
   ii) It means final when the frame is sent by a secondary to a primary (when the address field contains the address of the sender).

2) Control Field for S-Frames

• Supervisory frames are used for flow and error-control whenever piggybacking is either impossible or inappropriate (e.g., when the station either has no data of its own to send or needs to send a command or response other than an acknowledgment).
• S-frames do not have information fields.
• The subfields in the control field are:

1) If the first 2 bits of the control field is 10, this means the frame is an S-frame.
2) The last 3 bits N(R) corresponds to the acknowledgment-number (ACK) or negative acknowledgment-number (NAK).
3) The 2 bits called code is used to define the type of S-frame itself.
   With 2 bits, we can have four types of S-frames:
   1) Receive Ready (RR) = 00
   ¤ This acknowledges the receipt of frame or group of frames.
   ¤ The value of N(R) is the acknowledgment-number.
   2) Receive Not Ready (RNR) = 10
   ¤ This is an RR frame with 1 additional function:
      i) It announces that the receiver is busy and cannot receive more frames.
   ¤ It acts as congestion control mechanism by asking the sender to slow down.
   ¤ The value of N(R) is the acknowledgment-number.
   3) ReJect (REJ) = 01
   ¤ It is a NAK frame used in Go-Back-N ARQ to improve the efficiency of the process.
   ¤ It informs the sender, before the sender time expires, that the last frame is lost or damaged.
   ¤ The value of N(R) is the negative acknowledgment-number.
   4) Selective REJect (SREJ) = 11
   ¤ This is a NAK frame used in Selective Repeat ARQ.
   ¤ The value of N(R) is the negative acknowledgment-number.

3) Control Field for U-Frames

• Unnumbered frames are used to exchange session management and control information between connected devices.
• U-frames contain an information field used for system management information, but not user data.
• Much of the information carried by U-frames is contained in codes included in the control field.
• U-frame codes are divided into 2 sections:
  i) A 2-bit prefix before the P/F bit
  ii) A 3-bit suffix after the P/F bit.
• Together, these two segments (5 bits) can be used to create up to 32 different types of U-frames.

Example 3.9

Figure 11.18 shows how U-frames can be used for connection establishment and connection release. Node A asks for a connection with a set asynchronous balanced mode (SABM) frame; node B gives a positive response with an unnumbered acknowledgment (UA) frame. After these two exchanges, data can be transferred between the two nodes (not shown in the figure). After data transfer, node A sends a DISC (disconnect) frame to release the connection; it is confirmed by node B responding with a UA (unnumbered acknowledgment).

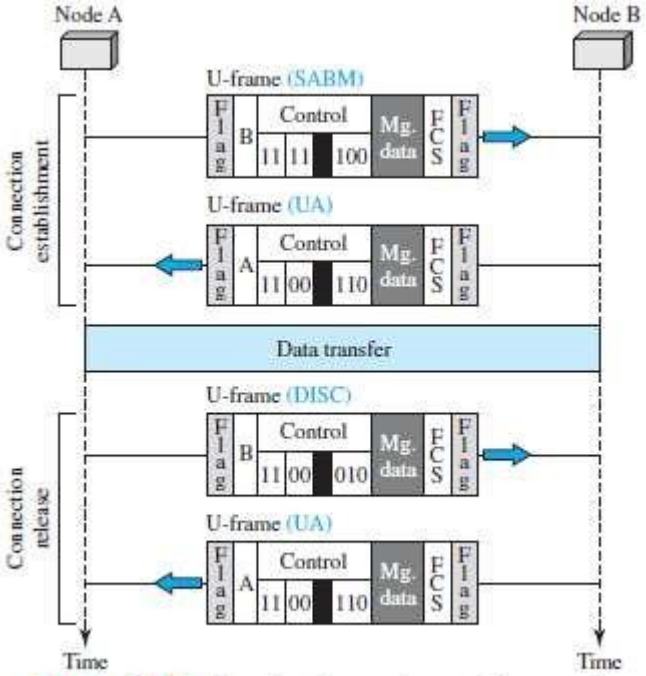


Figure 11.18 *Example of connection and disconnection*

3.9 POINT-TO-POINT PROTOCOL (PPP)

• PPP is one of the most common protocols for point-to-point access.
• Today, millions of Internet users who connect their home computers to the server of an ISP use PPP.

3.9.1 Framing

• PPP uses a character-oriented (or byte-oriented) frame (Figure 11.20).

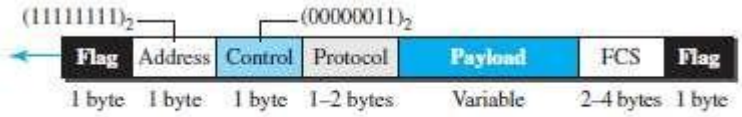


Figure 11.20 *PPP frame format*

• Various fields of PPP frame are:

1) Flag
- This field has a synchronization pattern 01111110.
- This field identifies both the beginning and the end of a frame.

2) Address
- This field is set to the constant value 11111111 (broadcast address).

3) Control
- This field is set to the constant value 00000011 (imitating unnumbered frames in HDLC).
- PPP does not provide any flow control.
- Error control is also limited to error detection.

4) Protocol
- This field defines what is being carried in the payload field.
- Payload field carries either i) user data or ii) other control information.
- By default, size of this field = 2 bytes.

5) Payload field
- This field carries either i) user data or ii) other control information.
- By default, maximum size of this field = 1500 bytes.
- This field is byte-stuffed if the flag-byte pattern appears in this field.
- Padding is needed if the payload-size is less than the maximum size.

6) FCS
- This field is the PPP error-detection field.
- This field can contain either a 2- or 4-byte standard CRC.

3.9.1.1 Byte Stuffing

• Since PPP is a byte-oriented protocol, the flag in PPP is a byte that needs to be escaped whenever it appears in the data section of the frame.
• The escape byte is 01111101, which means that every time the flag like pattern appears in the data, this extra byte is stuffed to tell the receiver that the next byte is not a flag.
• Obviously, the escape byte itself should be stuffed with another escape byte.

3.9.2 Transition Phases

• The transition diagram starts with the dead state (Figure 11.21).

1) Dead State

➢ In dead state, there is no active carrier and the line is quiet.

2) Establish State

➢ When 1 of the 2 nodes starts communication, the connection goes into the establish state.

➢ In establish state, options are negotiated between the two parties.

3) Authenticate State

➢ If the 2 parties agree that they need authentication,

Then the system needs to do authentication;

Otherwise, the parties can simply start communication.

4) Open State

➢ Data transfer takes place in the open state.

5) Terminate State

➢ When 1 of the endpoints wants to terminate connection, the system goes to terminate state.

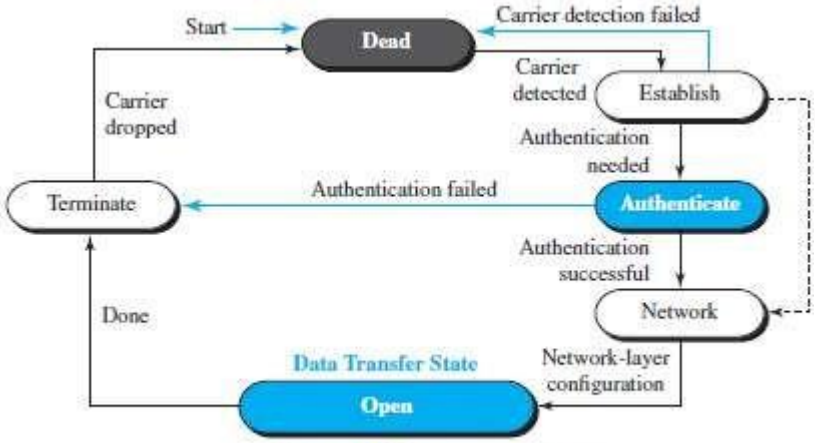


Figure 11.21 *Transition phases*

MODULE 2: MEDIA ACCESS CONTROL(MULTIPLE ACCESS)

4.1 Introduction

- When nodes use shared-medium, we need multiple-access protocol to coordinate access to medium.
- Analogy:
  - This problem is similar to the rules of speaking in an assembly.
  - We need to ensure
    - → Each people has right to speak.
    - → Two people do not speak at the same time
    - → Two people do not interrupt each other (i.e. Collision Avoidance)
- Many protocols have been designed to handle access to a shared-link (Figure 12.1).
- These protocols belong to a sublayer in the data-link layer called Media Access Control (MAC).

1) Four random-access protocols (or Contention Methods):
   i) ALOHA ii) CSMA iii) CSMA/CD iv) CSMA/CA
   These protocols are mostly used in LANs and WANs.
2) Three controlled-access protocols:
   i) Reservation ii) Polling iii) Token-passing
   Some of these protocols are used in LANs.
3) Three channelization protocols:
   i) FDMA ii) TDMA iii) CDMA
   These protocols are used in cellular telephony.

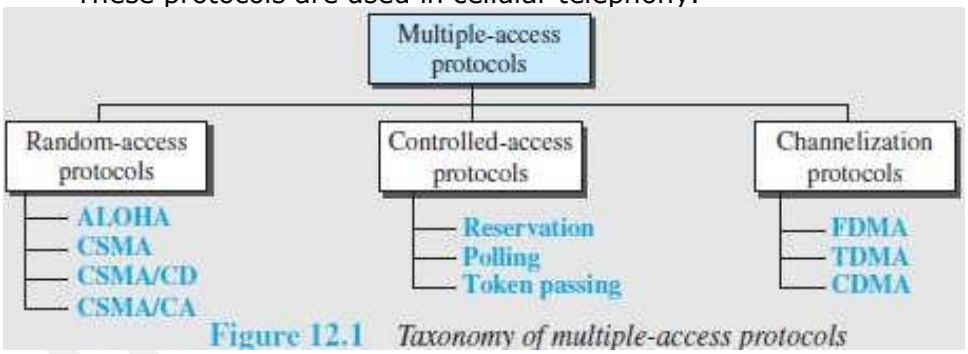


Figure 12.1 *Taxonomy of multiple-access protocols*

4.2 RANDOM ACCESS PROTOCOL

- No station is superior to another station.
- No station is assigned control over other station.
- To send the data, a station uses a procedure to make a decision on whether or not to send.
- This decision depends on the state of the medium: idle or busy.
- This is called Random Access because
  - → Transmission is random among the stations.
  - → There is no scheduled-time for a station to transmit.
- This is called Contention Method because
  - → Stations compete with one another to access the medium.
- If more than one station tries to send,
  there is an access-conflict (i.e. collision) and the frames will be destroyed.
- Each station follows a procedure that answers the following questions:
  1) When can the station access the medium?
  2) What can the station do if the medium is busy?
  3) How can the station determine the success or failure of the transmission?
  4) What can the station do if there is a collision?
- Four random-access protocols (or Contention methods):

DATA COMMUNICATION

1) ALOHA
2) CSMA (Carrier Sense Multiple Access)
3) CSMA/CD (Carrier Sense Multiple Access with Collision-detection)
4) CSMA/CA (Carrier Sense Multiple Access with Collision Avoidance)

4.2.1 ALOHA

- ALOHA was designed for a wireless LAN, but it can be used on any shared medium.
- Since the medium is shared between the stations, there is possibility of collisions.
- When 2 or more stations send the data simultaneously, there is possibility of collision & data loss.

4.2.1.1 Pure ALOHA

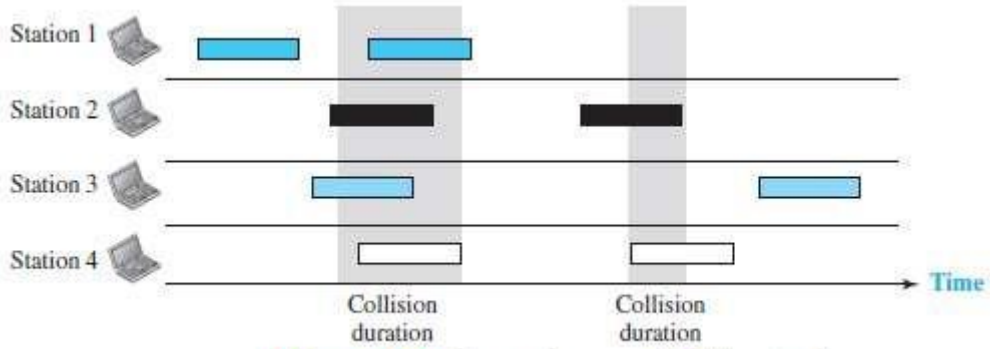


Figure 12.2 *Frames in a pure ALOHA network*

- Here is how it works (Figure 12.2):
  1) The sender sends a frame & starts the timer.
  2) The receiver receives the frame and responds with an acknowledgment.
  3) If the acknowledgment does not arrive after a time-out period, the sender resends the frame. The sender assumes that the frame (or the acknowledgment) has been destroyed.
  4) Since the medium is shared between the stations, there is possibility of collisions.
  5) If two stations try to resend the frames after the time-out, the frames will collide again.
  6) Two methods to deal with collision:
     1) Randomness
     ¤ When the time-out period passes, each station waits a random amount of time before resending the frame. This time is called back-off time T_B.
     ¤ The randomness will help avoid more collisions.
     2) Limit Maximum Retransmission
     ¤ This method prevents congestion by reducing the number of retransmitted frames.
     ¤ After a maximum number of retransmission-attempts K_{max}, a station must give up and try later (Figure 12.3).

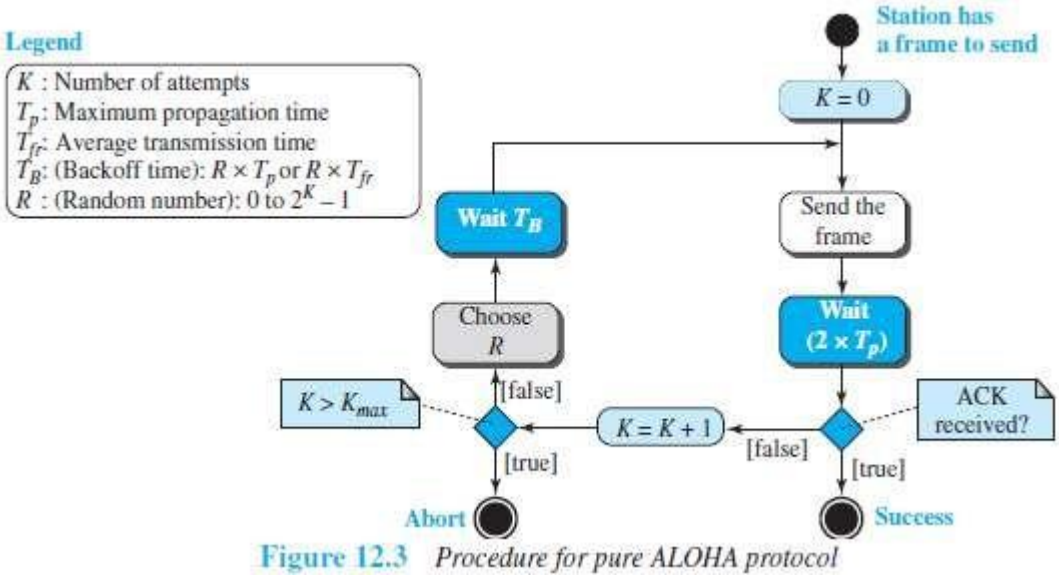


Figure 12.3 *Procedure for pure ALOHA protocol*

4.2.1.1.1 Vulnerable Time

- The vulnerable-time is defined as a time during which there is a possibility of collision.

Pure ALOHA vulnerable time = $2 \times T_{fr}$

where T_{fr} = Frame transmission time

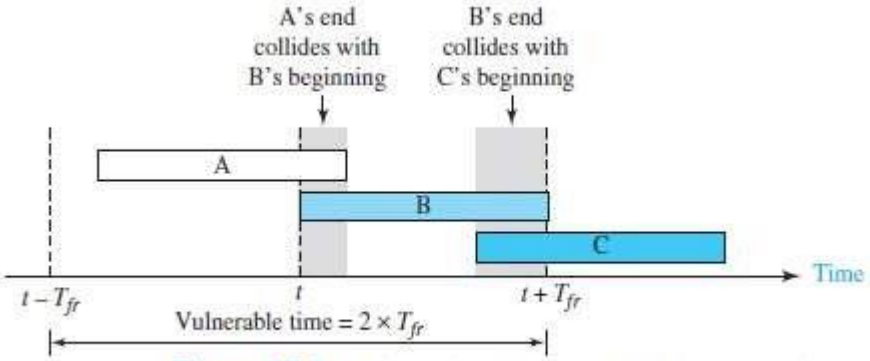


Figure 12.4 *Vulnerable time for pure ALOHA protocol*

- In Figure 12.4,
  - If station B sends a frame between t-T_{fr} and t, this leads to a collision between the frames from station A and station B.
  - If station C sends a frame between t and t+T_{fr}, this leads to a collision between the frames from station A and station C.

Example 4.1

A pure ALOHA network transmits 200-bit frames on a shared channel of 200 kbps. What is the requirement to make this frame collision-free?

Solution

Average frame transmission time T_{fr} is 200 bits/200 kbps or 1 ms. The vulnerable time is 2×1 ms = 2 ms. This means no station should send later than 1 ms before this station starts transmission and no station should start sending during the period (1 ms) that this station is sending.

4.2.1.1.2 Throughput

- The average number of successful transmissions is given by

$$S = G \times e^{-2G}$$

where G = average no. of frames in one frame transmission time (T_{fr})

- For G = 1, the maximum throughput S_{max} = 0.184.
- In other words, out of 100 frames, 18 frames reach their destination successfully.

Example 4.2

A pure ALOHA network transmits 200-bit frames on a shared channel of 200 kbps. What is the throughput if the system (all stations together) produces

a. 1000 frames per second? b. 500 frames per second? c. 250 frames per second?

Solution

The frame transmission time is 200/200 kbps or 1 ms.

a. If the system creates 1000 frames per second, or 1 frame per millisecond, then $G = 1$. In this case $S = G \times e^{-2G} = 0.135$ (13.5 percent). This means that the throughput is $1000 \times 0.135 = 135$ frames. Only 135 frames out of 1000 will probably survive.

b. If the system creates 500 frames per second, or 1/2 frames per millisecond, then $G = 1/2$. In this case $S = G \times e^{-2G} = 0.184$ (18.4 percent). This means that the throughput is $500 \times 0.184 = 92$ and that only 92 frames out of 500 will probably survive. Note that this is the *maximum* throughput case, percentagewise.

c. If the system creates 250 frames per second, or 1/4 frames per millisecond, then $G = 1/4$. In this case $S = G \times e^{-2G} = 0.152$ (15.2 percent). This means that the throughput is $250 \times 0.152 = 38$. Only 38 frames out of 250 will probably survive.

4.2.1.2 Slotted ALOHA

- Slotted ALOHA was invented to improve the efficiency of pure ALOHA.
- The time is divided into time-slots of T_{fr} seconds (Figure 12.5).
- The stations are allowed to send only at the beginning of the time-slot.

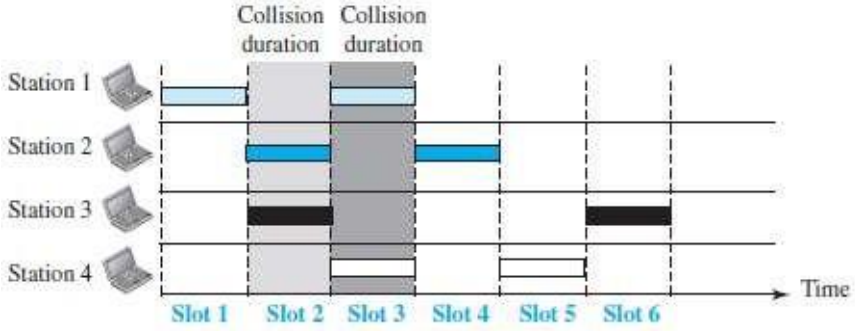


Figure 12.5 *Frames in a slotted ALOHA network*

- If a station misses the time-slot, the station must wait until the beginning of the next time-slot.
- If 2 stations try to resend at beginning of the same time-slot, the frames will collide again (Fig 12.6).

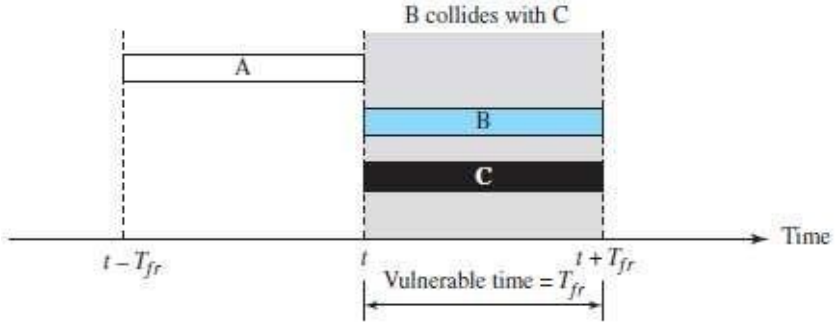


Figure 12.6 *Vulnerable time for slotted ALOHA protocol*

- The vulnerable time is given by:

vulnerable time = T_{fr}

4.2.1.2.1 Throughput

- The average number of successful transmissions is given by

$$S = G \times e^{-G}$$

- For G = 1, the maximum throughput S_{max} = 0.368.
- In other words, out of 100 frames, 36 frames reach their destination successfully.

Example 4.3

A slotted ALOHA network transmits 200-bit frames using a shared channel with a 200-kbps bandwidth. Find the throughput if the system (all stations together) produces

a. 1000 frames per second.
b. 500 frames per second.
c. 250 frames per second.

Solution

This situation is similar to the previous exercise except that the network is using slotted ALOHA instead of pure ALOHA. The frame transmission time is 200/200 kbps or 1 ms.

a. In this case G is 1. So $S = G \times e^{-G} = 0.368$ (36.8 percent). This means that the throughput is $1000 \times 0.0368 = 368$ frames. Only 368 out of 1000 frames will probably survive. Note that this is the maximum throughput case, percentagewise.
b. Here G is 1/2. In this case $S = G \times e^{-G} = 0.303$ (30.3 percent). This means that the throughput is $500 \times 0.0303 = 151$. Only 151 frames out of 500 will probably survive.
c. Now G is 1/4. In this case $S = G \times e^{-G} = 0.195$ (19.5 percent). This means that the throughput is $250 \times 0.195 = 49$. Only 49 frames out of 250 will probably survive.

4.2.2 CSMA

- CSMA was developed to minimize the chance of collision and, therefore, increase the performance.
- CSMA is based on the principle "sense before transmit" or "listen before talk."
- Here is how it works:
  1) Each station checks the state of the medium: idle or busy.
  2) i) If the medium is idle, the station sends the data.
     i) If the medium is busy, the station defers sending.
- CSMA can reduce the possibility of collision, but it cannot eliminate it.

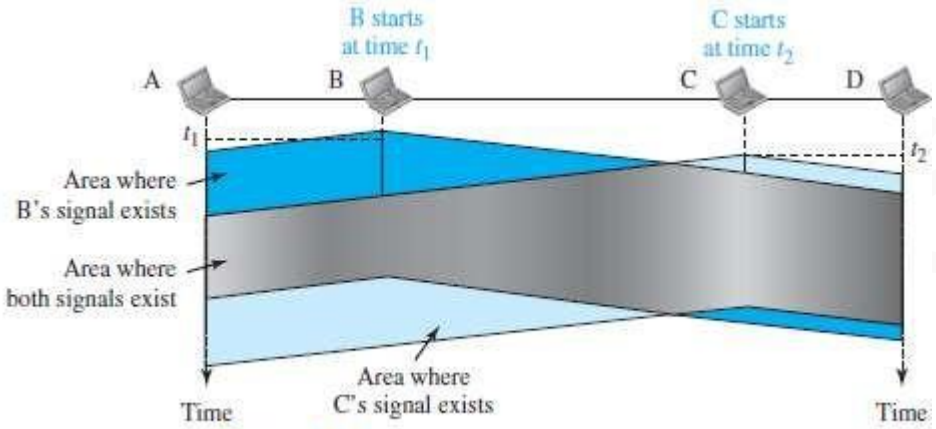


Figure 12.7 *Space/time model of a collision in CSMA*

- The possibility of collision still exists.
  For example:
  When a station sends a frame, it still takes time
  → for the first bit to reach every station and
  → for every station to sense it.
- For example: In Figure 12.7,
  - At time t1, station B senses & finds the medium idle, so sends a frame.
  - At time t2, station C senses & finds the medium idle, so sends a frame.
  - The 2 signals from both stations B & C collide and both frames are destroyed.

4.2.2.1 Vulnerable Time

- The vulnerable time is the propagation time T_p (Figure 12.8).
- The propagation time is the time needed for a signal to propagate from one end of the medium to the other.
- Collision occurs when
  → a station sends a frame, and
  → other station also sends a frame during propagation time
- If the first bit of the frame reaches the end of the medium, every station will refrain from sending.

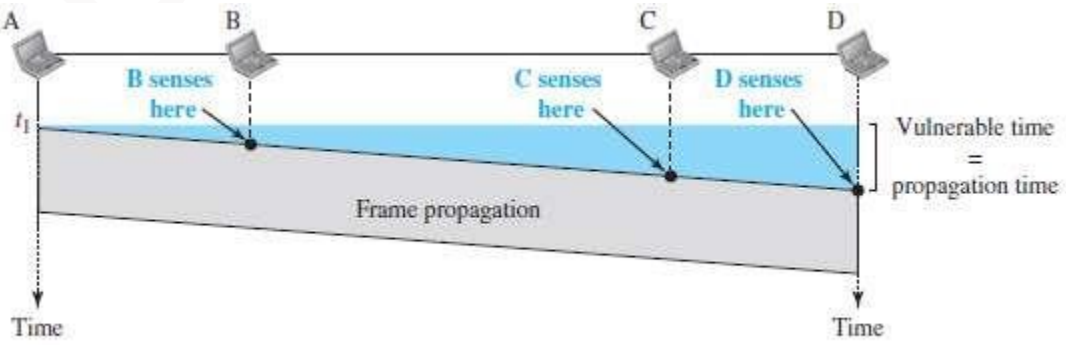


Figure 12.8 *Vulnerable time in CSMA*

4.2.2.2 Persistence Methods

- Q: What should a station do if the channel is busy or idle?

Three methods can be used to answer this question:

1) 1-persistent method 2) Non-persistent method and 3) p-persistent method

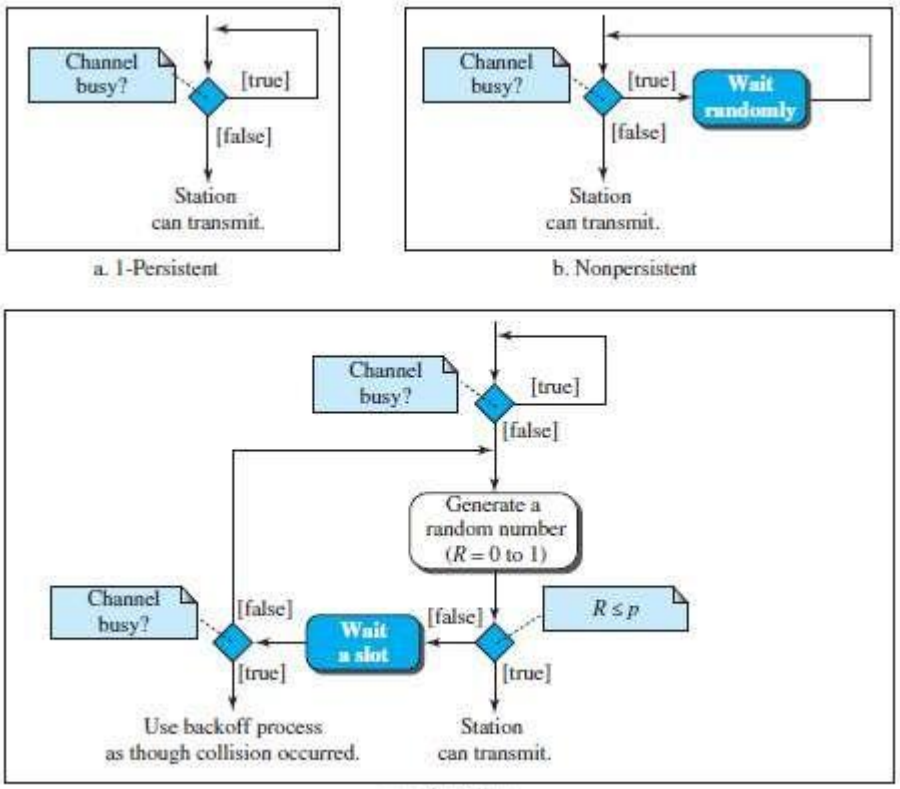


Figure 12.10 *Flow diagram for three persistence methods*

1) 1-Persistent

- Before sending a frame, a station senses the line (Figure 12.10a).
  - i) If the line is idle, the station sends immediately (with probability = 1).
  - ii) If the line is busy, the station continues sensing the line.
- This method has the highest chance of collision because 2 or more stations:
  - → may find the line idle and
  - → send the frames immediately.

2) Non-Persistent

- Before sending a frame, a station senses the line (Figure 12.10b).
  - i) If the line is idle, the station sends immediately.
  - ii) If the line is busy, the station waits a random amount of time and then senses the line again.
- This method reduces the chance of collision because 2 or more stations:
  - → will not wait for the same amount of time and
  - → will not retry to send simultaneously.

3) P-Persistent

- This method is used if the channel has time-slots with a slot-duration equal to or greater than the maximum propagation time (Figure 12.10c).
- Advantages:
  - i) It combines the advantages of the other 2 methods.
  - ii) It reduces the chance of collision and improves efficiency.
- After the station finds the line idle, it follows these steps:
  - 1) With probability p, the station sends the frame.
  - 2) With probability q=1-p, the station waits for the beginning of the next time-slot and checks the line again.
    - i) If line is idle, it goes to step 1.
    - ii) If line is busy, it assumes that collision has occurred and uses the back off procedure.

4.2.3 CSMA/CD

- Disadvantage of CSMA: CSMA does not specify the procedure after a collision has occurred.
  Solution: CSMA/CD enhances the CSMA to handle the collision.
- Here is how it works (Figure 12.12):
  1) A station
     → sends the frame &
     → then monitors the medium to see if the transmission was successful or not.
  2) If the transmission was unsuccessful (i.e. there is a collision), the frame is sent again.

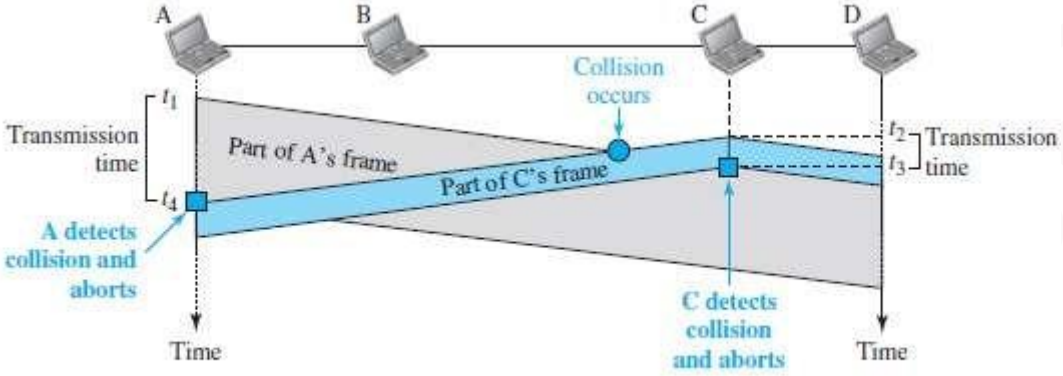


Figure 12.12 *Collision and abortion in CSMA/CD*

- In the Figure 12.11,
  - At time t_1, station A has executed its procedure and starts sending the bits of its frame.
  - At time t_2, station C has executed its procedure and starts sending the bits of its frame.
  - The collision occurs sometime after time t_2.
  - Station C detects a collision at time t_3 when it receives the first bit of A's frame.
    Station C immediately aborts transmission.
  - Station A detects collision at time t_4 when it receives the first bit of C's frame.
    Station A also immediately aborts transmission.
- Station A transmits for the duration $t_4 - t_1$.
  Station C transmits for the duration $t_3 - t_2$.
- For the protocol to work:
  The length of any frame divided by the bit rate must be more than either of these durations.

4.2.3.1 Minimum Frame Size

- For CSMA/CD to work, we need to restrict the frame-size.
- Before sending the last bit of the frame, the sender must
  → detect a collision and
  → abort the transmission.
- This is so because the sender
  → does not keep a copy of the frame and
  → does not monitor the line for collision-detection.
- Frame transmission time T_{fr} is given by
  $T_{fr} = 2T_p$ where T_p=maximum propagation time

Example 4.4

A network using CSMA/CD has a bandwidth of 10 Mbps. If the maximum propagation time (including the delays in the devices and ignoring the time needed to send a jamming signal, as we see later) is 25.6 μs, what is the minimum size of the frame?

Solution

The minimum frame transmission time is $T_{fr} = 2 \times T_p = 51.2$ μs. This means, in the worst case, a station needs to transmit for a period of 51.2 μs to detect the collision. The minimum size of the frame is 10 Mbps × 51.2 μs = 512 bits or 64 bytes. This is actually the minimum size of the frame for Standard Ethernet, as we will see later in the chapter.

4.2.3.2 Procedure

- CSMA/CD is similar to ALOHA with 2 differences (Figure 12.13):
  1) Addition of the persistence process.
     ¤ We need to sense the channel before sending the frame by using non-persistent, 1-persistent or p-persistent.
  2) Frame transmission.
     i) In ALOHA, first the entire frame is transmitted and then acknowledgment is waited for.
     ii) In CSMA/CD, transmission and collision-detection is a continuous process.

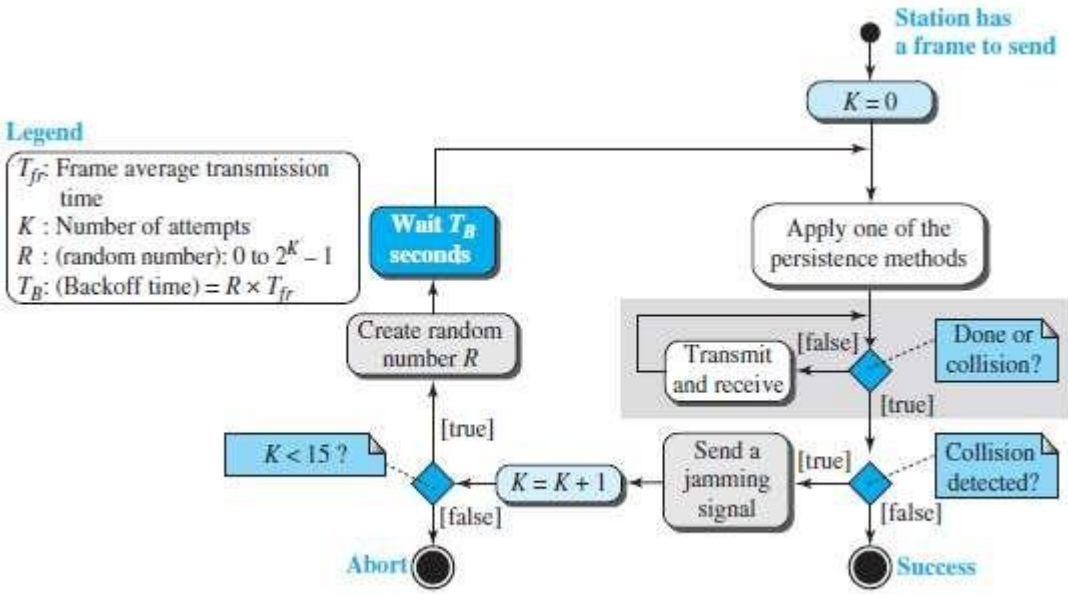


Figure 12.13 *Flow diagram for the CSMA/CD*

4.2.3.3 Energy Level

- In a channel, the energy-level can have 3 values: 1) Zero 2) Normal and 3) Abnormal.
  1) At zero level, the channel is idle (Figure 12.14).
  2) At normal level, a station has successfully captured the channel and is sending its frame.
  3) At abnormal level, there is a collision and the level of the energy is twice the normal level.
- A sender needs to monitor the energy-level to determine if the channel is
  → Idle
  → Busy or
  → Collision mode

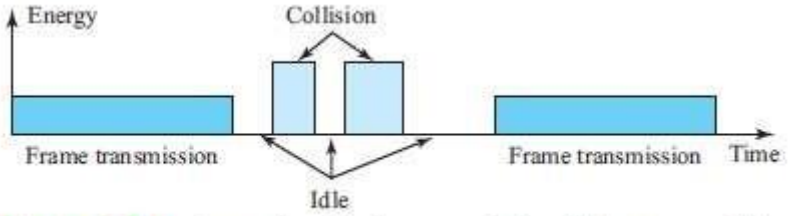


Figure 12.14 *Energy level during transmission, idleness, or collision*

4.2.3.4 Throughput

- The throughput of CSMA/CD is greater than pure or slotted ALOHA.
- The maximum throughput is based on
  → different value of G
  → persistence method used (non-persistent, 1-persistent, or p-persistent) and
  → 'p" value in the p-persistent method.
- For 1-persistent method, the maximum throughput is 50% when G =1.
- For non-persistent method, the maximum throughput is 90% when G is between 3 and 8.

4.2.4 CSMA/CA

• Here is how it works (Figure 12.15):

1) A station needs to be able to receive while transmitting to detect a collision.
   i) When there is no collision, the station receives one signal: its own signal.
   ii) When there is a collision, the station receives 2 signals:
      a) Its own signal and
      b) Signal transmitted by a second station.
2) To distinguish b/w these 2 cases, the received signals in these 2 cases must be different.

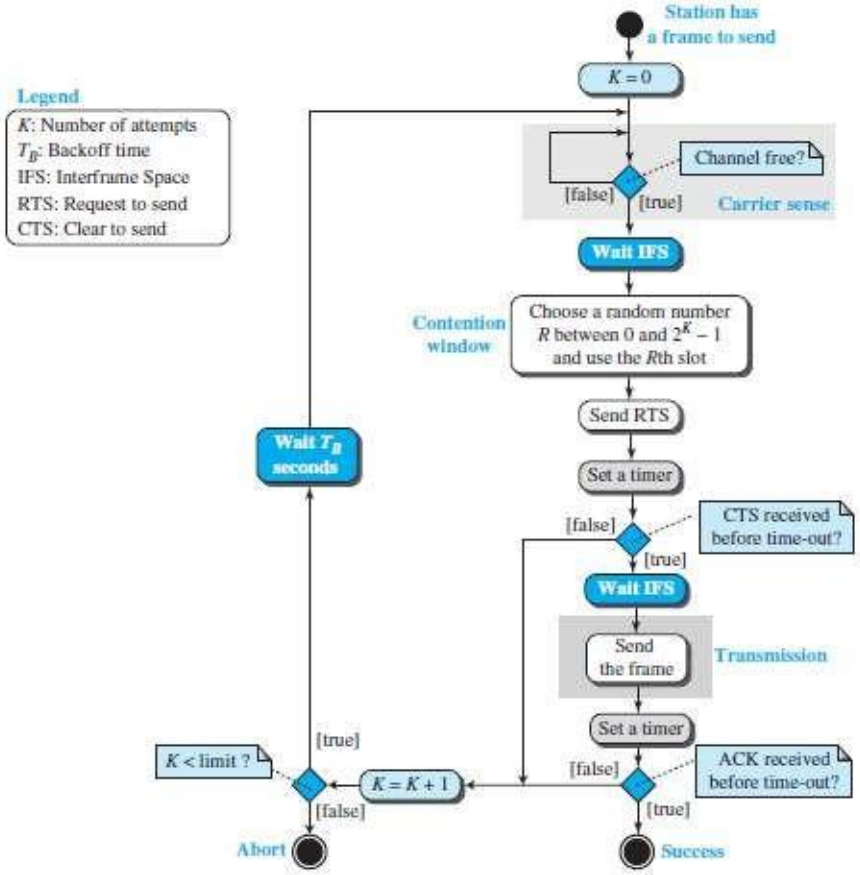


Figure 12.15 *Flow diagram of CSMA/CA*

• CSMA/CA was invented to avoid collisions on wireless networks.
• Three methods to avoid collisions (Figure 12.16):

1) Interframe space 2) Contention window and 3) Acknowledgments

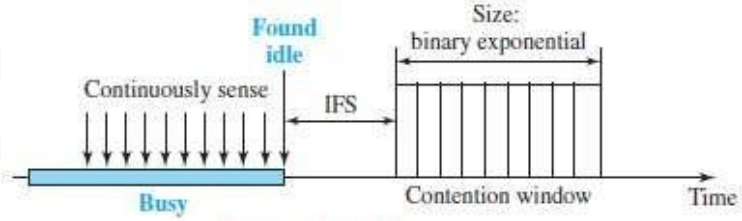


Figure 12.16 *Contention window*

1) Interframe Space (IFS)

• Collisions are avoided by deferring transmission even if the channel is found idle.
• When the channel is idle, the station does not send immediately.
  Rather, the station waits for a period of time called the inter-frame space or IFS.
• After the IFS time,
  if the channel is still idle,
    then, the station waits for the contention-time &
      finally, the station sends the frame.
• IFS variable can also be used to prioritize stations or frame types.
  For example, a station that is assigned a shorter IFS has a higher priority.

2) Contention Window

- The contention-window is an amount of time divided into time-slots.
- A ready-station chooses a random-number of slots as its wait time.
- In the window, the number of slots changes according to the binary exponential back-off strategy.
- For example:
  At first time, number of slots is set to one slot and
  Then, number of slots is doubled each time if the station cannot detect an idle channel.

3) Acknowledgment

- There may be a collision resulting in destroyed-data.
- In addition, the data may be corrupted during the transmission.
- To help guarantee that the receiver has received the frame, we can use
  i) Positive acknowledgment and
  ii) Time-out timer

4.2.4.1 Frame Exchange Time Line

- Two control frames are used:
  1) Request to send (RTS)
  2) Clear to send (CTS)
- The procedure for exchange of data and control frames in time (Figure 12.17):
  1) The source senses the medium by checking the energy level at the carrier frequency.
     ii) If the medium is idle,
     then the source waits for a period of time called the DCF interframe space (DIFS);
     finally, the source sends a RTS.
  2) The destination
     → receives the RTS
     → waits a period of time called the short interframe space (SIFS)
     → sends a control frame CTS to the source.
     CTS indicates that the destination station is ready to receive data.
  3) The source
     → receives the CTS
     → waits a period of time SIFS
     → sends a data to the destination
  4) The destination
     → receives the data
     → waits a period of time SIFS
     → sends a acknowledgment ACK to the source.
     ACK indicates that the destination has been received the frame.

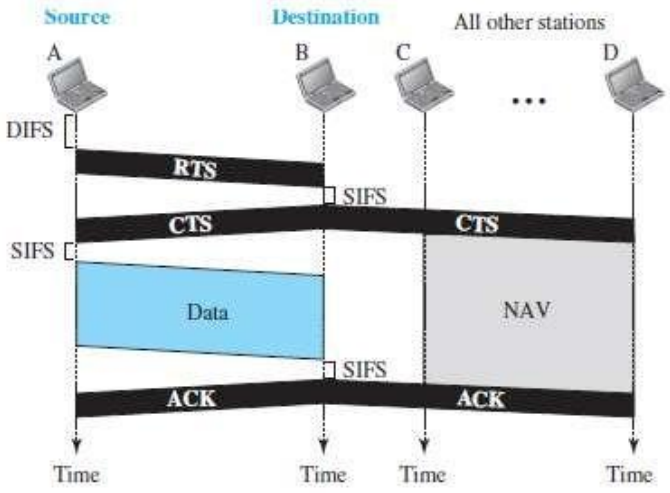


Figure 12.17 *CSMA/CA and NAV*

4.2.4.2 Network Allocation Vector

• When a source-station sends an RTS, it includes the duration of time that it needs to occupy the channel.
• The remaining stations create a timer called a network allocation vector (NAV).
• NAV indicates waiting time to check the channel for idleness.
• Each time a station accesses the system and sends an RTS frame, other stations start their NAV.

4.2.4.3 Collision during Handshaking

• Two or more stations may try to send RTS at the same time.
• These RTS may collide.
• The source assumes there has been a collision if it has not received CTS from the destination.
• The backoff strategy is employed, and the source tries again.

4.2.4.4 Hidden Station Problem

• Figure 12.17 also shows that the RTS from B reaches A, but not C.
• However, because both B and C are within the range of A, the CTS reaches C.
• Station C knows that some hidden station is using the channel and refrains from transmitting until that duration is over.

4.2.4.5 CSMA/CA and Wireless Networks

• CSMA/CA was mostly intended for use in wireless networks.
• However, it is not sophisticated enough to handle some particular issues related to wireless networks, such as hidden terminals or exposed terminals.

4.3 CONTROLLED ACCESS PROTOCOLS

- Here, the stations consult one another to find which station has the right to send.
- A station cannot send unless it has been authorized by other stations.
- Three popular controlled-access methods are: 1) Reservation 2) Polling 3) Token Passing

4.3.1 Reservation

- Before sending data, each station needs to make a reservation of the medium.
- Time is divided into intervals.
- In each interval, a reservation-frame precedes the data-frames.
- If no. of stations = N, then there are N reservation mini-slots in the reservation-frame.
- Each mini-slot belongs to a station.
- When a station wants to send a data-frame, it makes a reservation in its own minislot.
- The stations that have made reservations can send their data-frames.

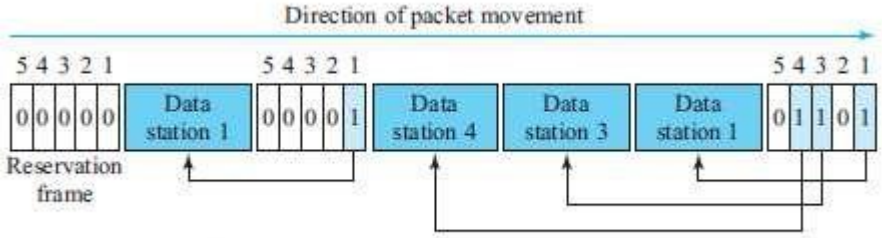


Figure 12.18 *Reservation access method*

- For example (Figure 12.18):
  - 5 stations have a 5-minislot reservation-frame.
  - In the first interval, only stations 1, 3, and 4 have made reservations.
  - In the second interval, only station-1 has made a reservation.

4.3.2 Polling

- In a network,
  - One device is designated as a primary station and
  - Other devices are designated as secondary stations.
- Functions of primary-device:
  1) The primary-device controls the link.
  2) The primary-device is always the initiator of a session.
  3) The primary-device is determines which device is allowed to use the channel at a given time.
  4) All data exchanges must be made through the primary-device.
- The secondary devices follow instructions of primary-device.
- Disadvantage: If the primary station fails, the system goes down.
- Poll and select functions are used to prevent collisions (Figure 12.19).

1) Select

➢ If the primary wants to send data, it tells the secondary to get ready to receive; this is called select function.

➢ The primary
- → alerts the secondary about upcoming transmission by sending select frame (SEL)
- → then waits for an acknowledgment (ACK) from secondary
- → then sends the data frame and
- → finally waits for an acknowledgment (ACK) from the secondary.

2) Poll

➢ If the primary wants to receive data, it asks the secondaries if they have anything to send; this is called poll function.

➢ When the first secondary is approached, it responds either
- → with a NAK frame if it has no data to send or
- → with data-frame if it has data to send.

i) If the response is negative (NAK frame), then the primary polls the next secondary in the same manner.

ii) When the response is positive (a data-frame), the primary
- → reads the frame and
- → returns an acknowledgment (ACK frame).

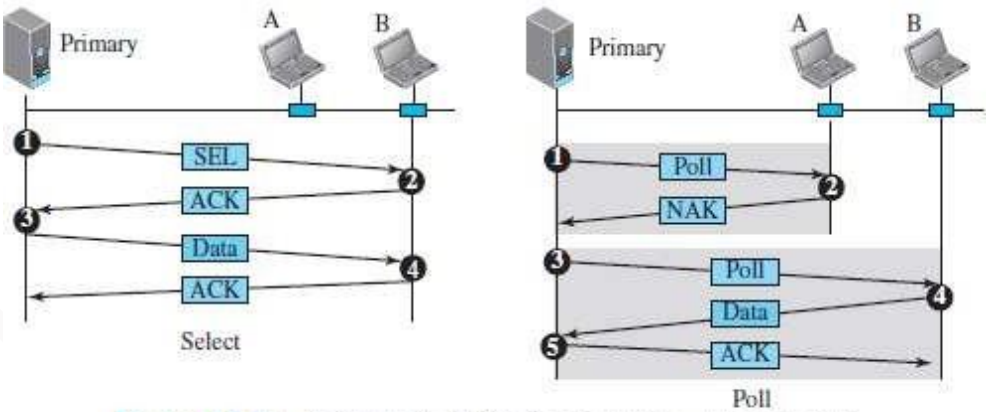


Figure 12.19 *Select and poll functions in polling-access method*

4.3.3 Token Passing

• In a network, the stations are organized in a ring fashion i.e. for each station; there is a predecessor and a successor.
  1) The predecessor is the station which is logically before the station in the ring.
  2) The successor is the station which is after the station in the ring.
• The current station is the one that is accessing the channel now.
• A token is a special packet that circulates through the ring.
• Here is how it works:
  - A station can send the data only if it has the token.
  - When a station wants to send the data, it waits until it receives the token from its predecessor.
  - Then, the station holds the token and sends its data.
  - When the station finishes sending the data, the station
    → releases the token
    → passes the token to the successor.
• Main functions of token management:
  1) Stations must be limited in the time they can hold the token.
  2) The token must be monitored to ensure it has not been lost or destroyed.
     For ex: if a station that is holding the token fails, the token will disappear from the network
  3) Assign priorities
     → to the stations and
     → to the types of data being transmitted.
  4) Make low-priority stations release the token to high priority stations.

4.3.3.1 Logical Ring

• In a token-passing network, stations do not have to be physically connected in a ring; the ring can be a logical one.
• Four physical topologies to create a logical ring (Figure 12.20):
  1) Physical ring
  2) Dual ring
  3) Bus ring
  4) Star ring

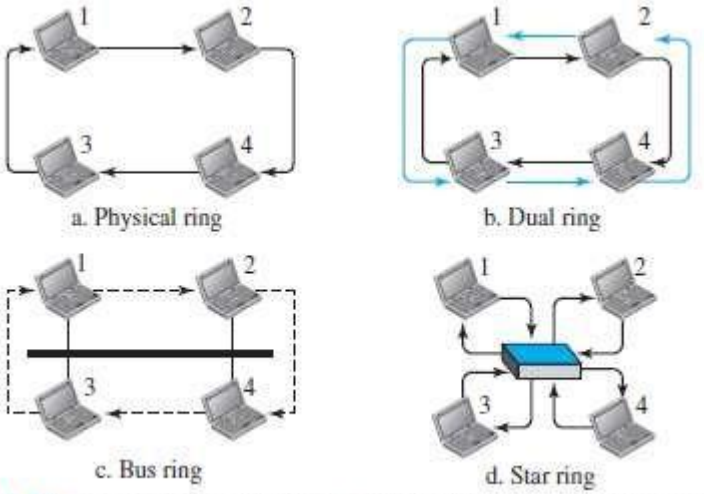


Figure 12.20 *Logical ring and physical topology in token-passing access method*

1) Physical Ring Topology

• When a station sends token to its successor, token cannot be seen by other stations. (Figure 12.20a)
• This means that the token does not have the address of the next successor.
• Disadvantage: If one of the links fails, the whole system fails.

2) Dual Ring Topology

- A second (auxiliary) ring
  - → is used along with the main ring (Figure 12.20b).
  - → operates in the reverse direction compared with the main ring.
  - → is used for emergencies only (such as a spare tire for a car).
- If the main ring fails, the system automatically combines the 2 rings to form a temporary ring.
- After the failed link is restored, the second ring becomes idle again.
- Each station needs to have 2 transmitter-ports and 2 receiver-ports.
- This topology is used in
  - i) FDDI (Fiber Distributed Data Interface) and
  - ii) CDDI (Copper Distributed Data Interface).

3) Bus Ring Topology

- The stations are connected to a single cable called a bus (Figure 12.20c).
- This makes a logical ring, because each station knows the address of its successor and predecessor.
- When a station has finished sending its data, the station
  - → releases the token and
  - → inserts the address of its successor in the token.
- Only the station gets the token to access the shared media.
- This topology is used in the Token Bus LAN.

4) Star Ring Topology

- The physical topology is a star (Figure 12.20d).
- There is a hub that acts as the connector.
- The wiring inside the hub makes the ring i.e. the stations are connected to the ring through the 2 wire connections.
- Disadvantages:
  1) This topology is less prone to failure because
     If a link goes down,
     then the link will be bypassed by the hub and
     the rest of the stations can operate.
  2) Also adding and removing stations from the ring is easier.
- This topology is used in the Token Ring LAN.

4.4 CHANNELIZATION PROTOCOLS

- Channelization is a multiple-access method.
- The available bandwidth of a link is shared b/w different stations in time, frequency, or through code.
- Three channelization protocols:
  1) FDMA (Frequency Division Multiple Access)
  2) TDMA (Time Division Multiple Access) and
  3) CDMA (Code Division Multiple Access)

4.4.1 FDMA

- The available bandwidth is divided into frequency-bands (Figure 12.21).
- Each band is reserved for a specific station.
- Each station can send the data in the allocated band.
- Each station also uses a bandpass filter to confine the transmitter frequencies.
- To prevent interferences, small guard bands are used to separate the allocated bands from one another.

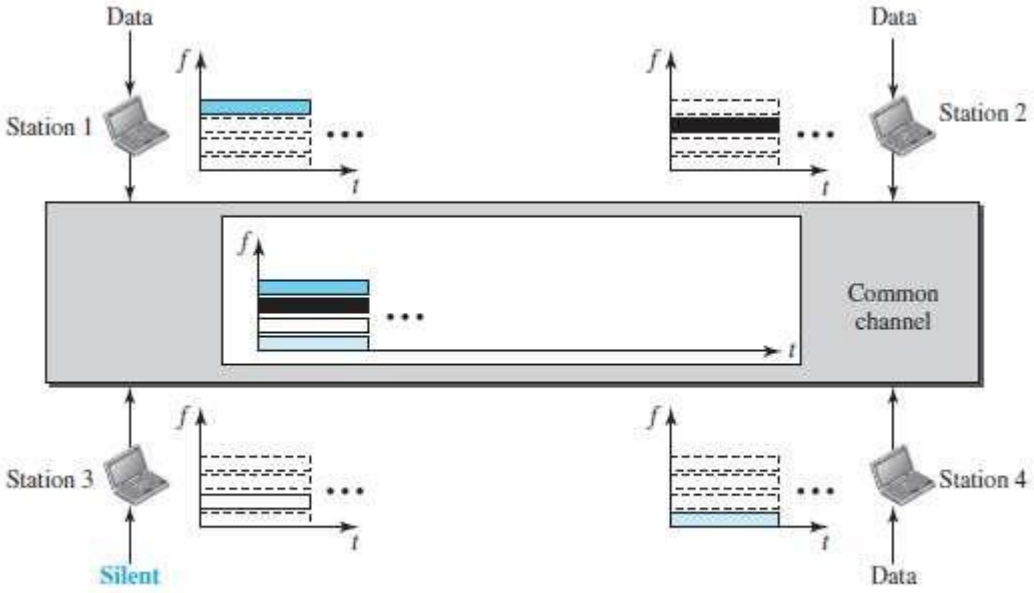


Figure 12.21 *Frequency-division multiple access (FDMA)*

- FDM vs. FDMA

1) FDM

- FDM is a multiplexing method in the physical layer.
- FDM
  → combines individual-loads from low-bandwidth channels and
  → transmits aggregated-load by using a high-bandwidth channel.
- The channels that are combined are low-pass.
- The multiplexer
  → modulates & combines the signals and
  → creates a bandpass signal.
- The bandwidth of each channel is shifted by the multiplexer.

2) FDMA

- FDMA is an access method in the data link layer.
- In each station, the data link layer tells the physical layer to make a bandpass signal from the data passed to it.
- The signal must be created in the allocated band.
- There is no physical multiplexer at the physical layer.
- The signals created at each station are automatically bandpass-filtered.
- They are mixed when they are sent to the common channel.

4.4.2 TDMA

- The stations share the bandwidth of the channel in time (Figure 12.22).
- Each time-slot is reserved for a specific station.
- Each station can send the data in the allocated time-slot.

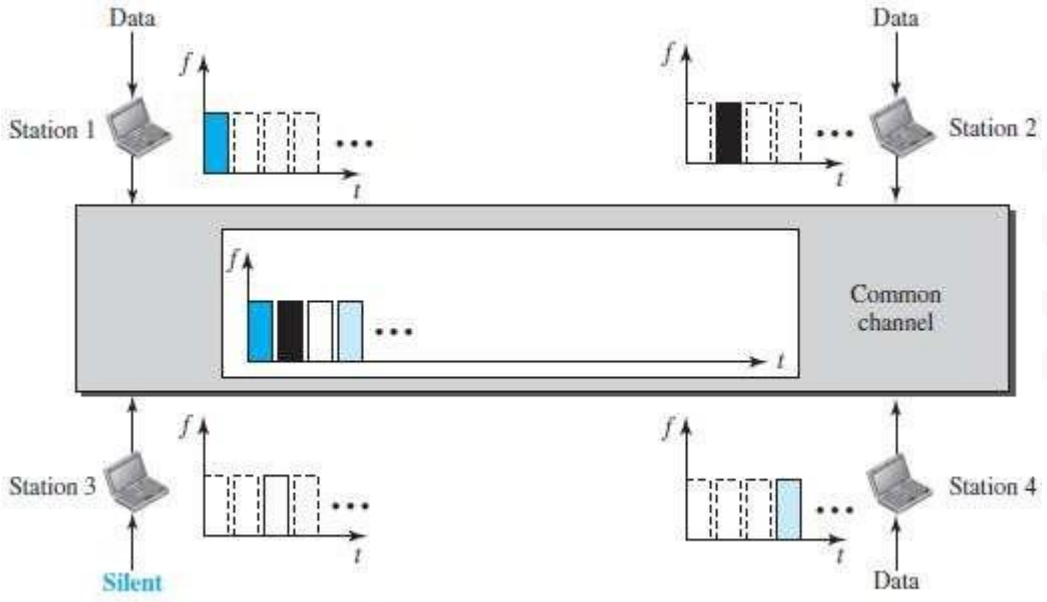


Figure 12.22 *Time-division multiple access (TDMA)*

- Main problem: Achieving synchronization between the different stations.
  i.e. each station needs to know the beginning of its slot and the location of its slot.
  This may be difficult because of propagation delays introduced in the system.
- To compensate for the delays, we can insert guard-times.
- Normally, synchronization is accomplished by having some synchronization bits at the beginning of each slot.
- TDMA vs. TDM

1) TDM

- TDM is a multiplexing method in the physical layer.
- TDM
  → combines the individual-data from slower channels and
  → transmits the aggregated- data by using a faster channel.
- The multiplexer interleaves data units from each channel.

2) TDMA

- TDMA is an access method in the data link layer.
- In each station, the data link layer tells the physical layer to use the allocated time-slot.
- There is no physical multiplexer at the physical layer.

4.4.3 CDMA

- CDMA simply means communication with different codes.
- CDMA differs from FDMA because
  → only one channel occupies the entire bandwidth of the link.
- CDMA differs from TDMA because
  → all stations can send data simultaneously; there is no timesharing.

(Analogy: CDMA simply means communication with different codes.
For example, in a large room with many people, 2 people can talk privately in English if nobody else understands English. Another 2 people can talk in Chinese if they are the only ones who understand Chinese, and so on).

4.4.3.1 Implementation

- Let us assume we have four stations 1, 2, 3, and 4 connected to the same channel.
- The data from station-1 are d_1, from station-2 are d_2, and so on.
- The code assigned to the first station is c_1, to the second is c_2, and so on.
- We assume that the assigned codes have 2 properties.
  1) If we multiply each code by another, we get 0.
  2) If we multiply each code by itself, we get 4 (the number of stations).

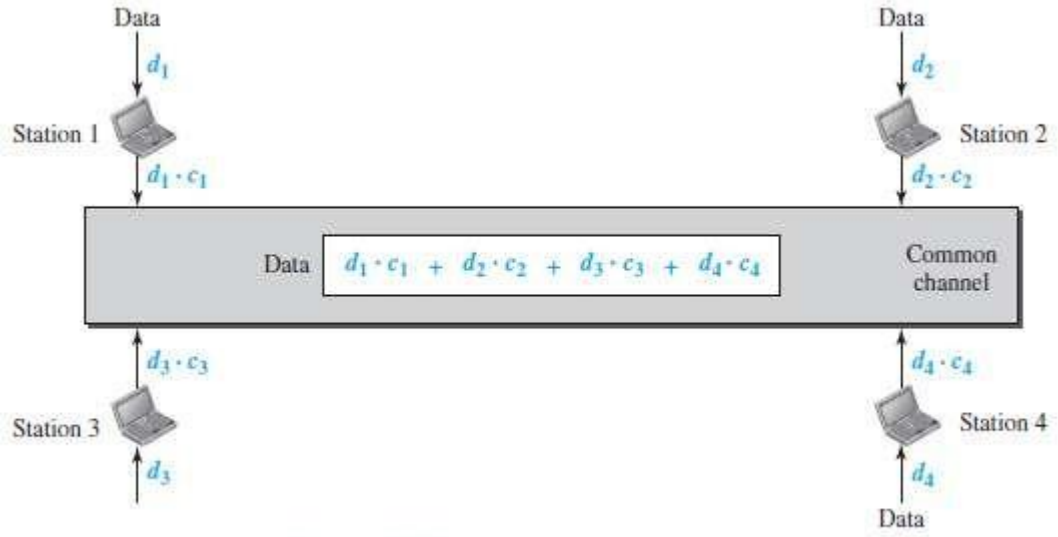


Figure 12.23 *Simple idea of communication with code*

- Here is how it works (Figure 12.23):
  - Station-1 multiplies the data by the code to get $d_1.c_1$.
  - Station-2 multiplies the data by the code to get $d_2.c_2$. And so on.
  - The data that go on the channel are the sum of all these terms.

$$d_1 \cdot c_1 + d_2 \cdot c_2 + d_3 \cdot c_3 + d_4 \cdot c_4$$

  - The receiver multiplies the data on the channel by the code of the sender.
  - For example, suppose stations 1 and 2 are talking to each other.
  - Station-2 wants to hear what station-1 is saying.
  - Station-2 multiplies the data on the channel by c_1 the code of station-1.
    $(c_1.c_1)=4$, $(c_2.c_1)=0$, $(c_3.c_1)=0$, and $(c_4.c_1)=0$,
    Therefore, station-2 divides the result by 4 to get the data from station-1.

$$\begin{aligned}\text{data} &= (d_1 \cdot c_1 + d_2 \cdot c_2 + d_3 \cdot c_3 + d_4 \cdot c_4) \cdot c_1 \\ &= d_1 \cdot c_1 \cdot c_1 + d_2 \cdot c_2 \cdot c_1 + d_3 \cdot c_3 \cdot c_1 + d_4 \cdot c_4 \cdot c_1 = 4 \times d_1\end{aligned}$$

4.4.3.2 Chips

- CDMA is based on coding theory.
- Each station is assigned a code, which is a sequence of numbers called chips (Figure 12.24).

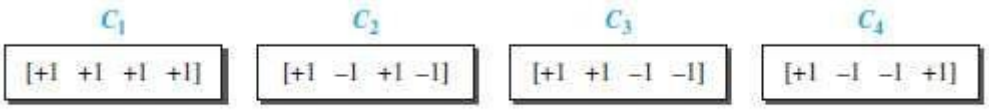


Figure 12.24 *Chip sequences*

- These sequences were carefully selected & are called orthogonal sequences
- These sequences have the following properties:
    1) Each sequence is made of N elements, where N is the number of stations.
    2) Multiplication of a sequence by a scalar:
        If we multiply a sequence by a number i.e. every element in the sequence is multiplied by that element.
        For example,
        $$2 \cdot [+1 +1 -1 -1] = [+2 +2 -2 -2]$$
    3) Inner product of 2 equal sequences:
        If we multiply 2 equal sequences, element by element, and add the results, we get N, where N is the number of elements in the each sequence.
        For example,
        $$[+1 +1 -1 -1] \cdot [+1 +1 -1 -1] = 1 + 1 + 1 + 1 = 4$$
    4) Inner product of 2 different sequences:
        If we multiply 2 different sequences, element by element, and add the results, we get 0.
        For example,
        $$[+1 +1 -1 -1] \cdot [+1 +1 +1 +1] = 1 + 1 - 1 - 1 = 0$$
    5) Adding 2 sequences means adding the corresponding elements. The result is another sequence.
        For example,
        $$[+1 +1 -1 -1] + [+1 +1 +1 +1] = [+2 +2 \ 0 \ 0]$$

4.4.3.3 Data Representation

- We follow the following rules for encoding:
    1) To send a 0 bit, a station encodes the bit as -1
    2) To send a 1 bit, a station encodes the bit as +1
    3) When a station is idle, it sends no signal, which is interpreted as a 0.

4.4.3.4 Encoding and Decoding

- We assume that
  - Stations 1 and 2 are sending a 0 bit.
  - Station-4 is sending a 1 bit.
  - Station-3 is silent.
- Here is how it works (Figure 12.26):
  - At the sender-site, the data are translated to -1, -1, 0, and +1.
  - Each station multiplies the corresponding number by its chip (its orthogonal sequence).
  - The result is a new sequence which is sent to the channel.
  - The sequence on the channel is the sum of all 4 sequences.
  - Now imagine station-3, which is silent, is listening to station-2.
  - Station-3 multiplies the total data on the channel by the code for station-2, which is [+1 -1 +1-1], to get

$$[-1\ -1\ -3\ +1] \bullet [+1\ -1\ +1\ -1] = -4/4 = -1 \rightarrow \text{bit 1}$$

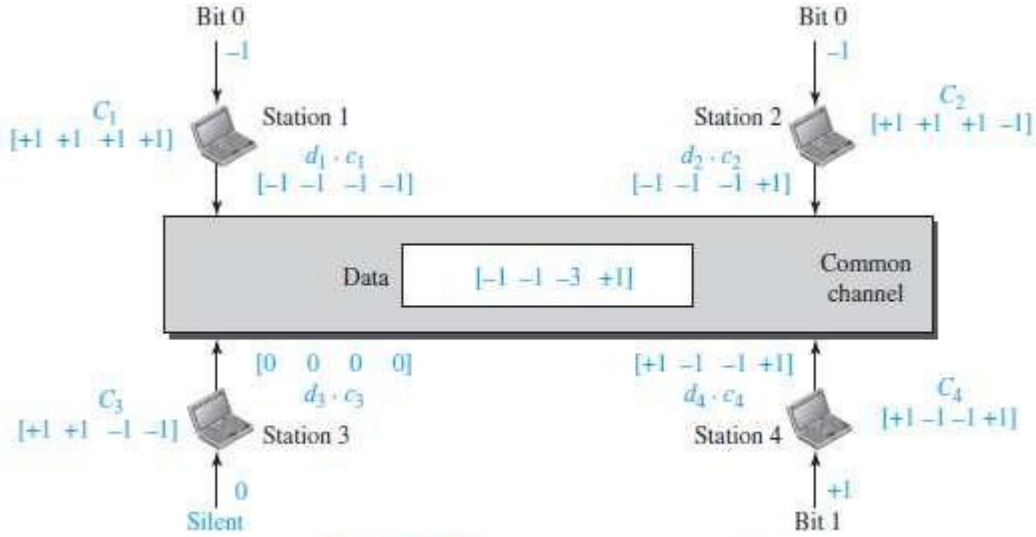
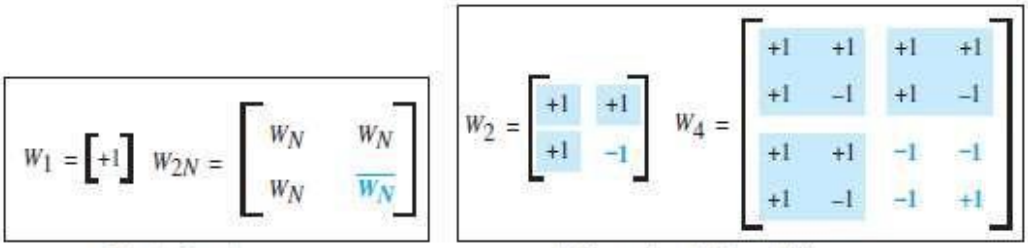


Figure 12.26 *Sharing channel in CDMA*

4.4.3.5 Sequence Generation

- To generate chip sequences, we use a Walsh table (Figure 12.29).
- Walsh table is a 2-dimensional table with an equal number of rows and columns.



a. Two basic rules b. Generation of W_2 and W_4

Figure 12.29 *General rule and examples of creating Walsh tables*

- In the Walsh table, each row is a sequence of chips.
- W_1 for a one-chip sequence has one row and one column. We can choose –1 or +1 for the chip for this trivial table (we chose +1).
- According to Walsh, if we know the table for N sequences W_N, we can create the table for 2N sequences W_{2N} (Figure 12.29).
- The W_N with the overbar W_N stands for the complement of W_N where each +1 is changed to -1 and vice versa.
- After we select W_1, W_2 can be made from four W_1's, with the last one the complement of W_1
- After W_2 is generated, W_4 can be made of four W_2's, with the last one the complement of W_2.
- The number of sequences in a Walsh table needs to be $N = 2^m$.

Example 4.5

Find the chips for a network with

a. Two stations
b. Four stations

Solution

We can use the rows of W_2 and W_4 in Figure 12.29:

a. For a two-station network, we have [+1 +1] and [+1 –1].
b. For a four-station network we have [+1 +1 +1 +1], [+1 –1 +1 –1], [+1 +1 –1 –1], and [+1 –1 –1 +1].

Example 4.6

What is the number of sequences if we have 90 stations in our network?

Solution

The number of sequences needs to be 2^m. We need to choose $m = 7$ and $N = 2^7$ or 128. We can then use 90 of the sequences as the chips.

Example 4.7

Prove that a receiving station can get the data sent by a specific sender if it multiplies the entire data on the channel by the sender's chip code and then divides it by the number of stations.

Solution

Let us prove this for the first station, using our previous four-station example. We can say that the data on the channel $D = (d_1 \cdot c_1 + d_2 \cdot c_2 + d_3 \cdot c_3 + d_4 \cdot c_4)$. The receiver that wants to get the data sent by station 1 multiplies these data by c_1.

$$
\begin{aligned}
D \cdot c_1 &= (d_1 \cdot c_1 + d_2 \cdot c_2 + d_3 \cdot c_3 + d_4 \cdot c_4) \quad c_1 \\
&= d_1 \cdot c_1 \cdot c_1 + d_2 \cdot c_2 \cdot c_1 + d_3 \cdot c_3 \cdot c_1 + d_4 \cdot c_4 \cdot c_1 \\
&= d_1 \times N + d_2 \times 0 + d_3 \times 0 + d_4 \times 0 \\
&= d_1 \times N
\end{aligned}
$$

When we divide the result by N, we get d_1.

MODULE 3: NETWORK LAYER PROTOCOLS

5.3 Network Layer Protocols

• The network layer contains following 4 protocols (Figure 19.1):

1) Internet Protocol (IP)

➢ IP is the main protocol responsible for packetizing, forwarding, and delivery of a packet at the network layer.

2) Internet Control Message Protocol (ICMP)

➢ ICMP helps IP to handle some errors that may occur in the network-layer delivery.

3) Internet Group Management Protocol (IGMP)

➢ IGMP is used to help IPv4 in multicasting.

4) Address Resolution Protocol (ARP)

➢ ARP is used to glue the network and data-link layers in mapping network-layer addresses to link-layer addresses.

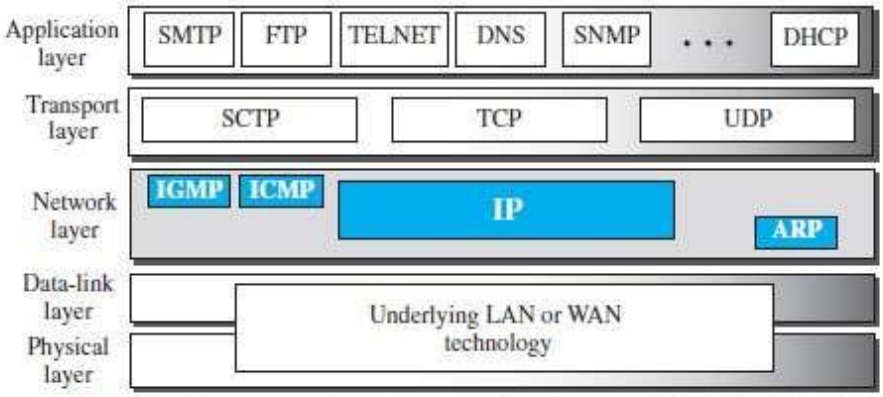


Figure 19.1 *Position of IP and other network-layer protocols in TCP/IP protocol suite*

5.4 INTERNET PROTOCOL (IP)

5.5.1 Internet Protocol (IP)

- IP is main protocol responsible for packetizing, forwarding & delivery of a packet at network layer.
- IP is an unreliable datagram protocol.
- IP provides a best-effort delivery service.
- The term best-effort means that the packets can
  - → be corrupted
  - → be lost or
  - → arrive out-of-order.
- If reliability is important, IP must be paired with a TCP which is reliable transport-layer protocol.
- IP is a connectionless protocol.
- IP uses the datagram approach.
  1) Each datagram is handled independently.
  2) Each datagram can follow a different route to the destination.
  3) Datagrams may arrive out-of-order at the destination.

5.5.2 Datagram Format

- IP uses the packets called datagrams.
- A datagram consist of 2 parts (Figure 19.2): 1) Payload 2) Header.

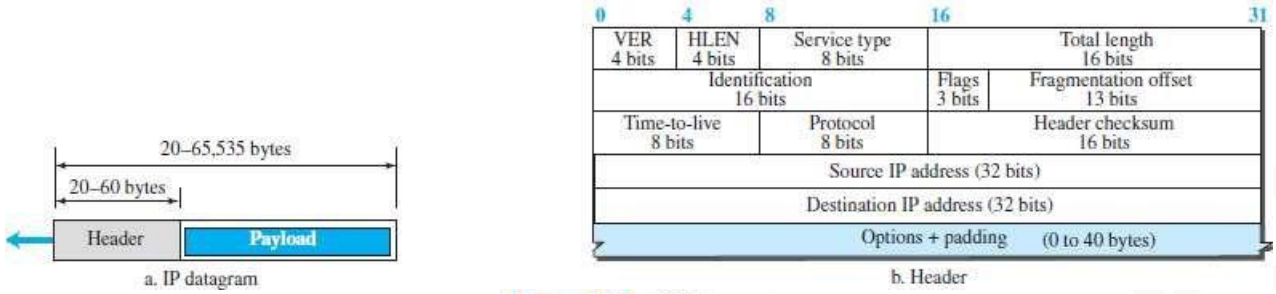


Figure 19.2 *IP datagram*

1) Payload

- Payload (or Data) is the main reason for creating a datagram.
- Payload is the packet coming from other protocols that use the service of IP.

2) Header

- Header contains information essential to routing and delivery.
- IP header contains following fields:

1) Version Number (VER)

➢ This field indicates version number used by the packet. Current version=4

2) Header Length (HLEN)

➢ This field specifies length of header.
➢ When a device receives a datagram, the device needs to know
→ when the header stops and
→ when the data starts.

3) Service Type

➢ This field specifies priority of packet based on delay, throughput, reliability & cost requirements.

4) Total Length

➢ This field specifies the total length of the datagram (header plus data).
➢ Maximum length=65535 bytes.

5) Identification, Flags, and Fragmentation Offset

➢ These 3 fields are used for fragmentation and reassembly of the datagram.
➢ Fragmentation occurs when the size of the datagram is larger than the MTU of the network.

6) Time-to-Live (TTL)

➢ This field is indicates amount of time, the packet is allowed to remain in the network.
➢ If TTL becomes 0 before packet reaches destination, the router
→ discards packet and
→ sends an error-message back to the source.

7) Protocol

➢ This field specifies upper-layer protocol that is to receive the packet at the destination-host.
➢ For example (Figure 19.3):
For TCP, protocol = 6 For UDP, protocol = 17

8) Header Checksum

➢ This field is used to verify integrity of header only.
➢ If the verification process fails, packet is discarded.

9) Source and Destination Addresses

➢ These 2 fields contain the IP addresses of source and destination hosts.

10) Options

➢ This field allows the packet to request special features such as
→ security level
→ route to be taken by packet and
→ timestamp at each router.
➢ This field can also be used for network testing and debugging.

11) Padding

➢ This field is used to make the header a multiple of 32-bit words.

Example 5.3

An IPv4 packet has arrived with the first 8 bits as $(01000010)_2$. The receiver discards the packet. Why?

Solution

There is an error in this packet. The 4 leftmost bits $(0100)_2$ show the version, which is correct. The next 4 bits $(0010)_2$ show an invalid header length $(2 \times 4 = 8)$. The minimum number of bytes in the header must be 20. The packet has been corrupted in transmission.

Example 5.4

In an IPv4 packet, the value of HLEN is $(1000)_2$. How many bytes of options are being carried by this packet?

Solution

The HLEN value is 8, which means the total number of bytes in the header is 8×4, or 32 bytes. The first 20 bytes are the base header, the next 12 bytes are the options.

Example 5.5

In an IPv4 packet, the value of HLEN is 5, and the value of the total length field is $(0028)_{16}$. How many bytes of data are being carried by this packet?

Solution

The HLEN value is 5, which means the total number of bytes in the header is 5×4, or 20 bytes (no options). The total length is $(0028)_{16}$ or 40 bytes, which means the packet is carrying 20 bytes of data $(40 - 20)$.

(Comparing a datagram to a postal package.

1) Payload is the content of the package.
2) Header is only the information written on the package).

5.5.3 Fragmentation

5.5.3.1 Maximum Transfer Unit (MTU)

• Each network imposes a restriction on maximum size of packet that can be carried. This is called the MTU (maximum transmission unit).
• For example:
For Ethernet, MTU = 1500 bytes
For FDDI, MTU = 4464 bytes
• When IP wants send a packet that is larger than MTU of physical-network, IP breaks packet into smaller fragments. This is called fragmentation (Figure 19.5).
• Designers have decided to make the maximum length of IP datagram = 65,535 bytes. This ensures that the IP protocol is independent of the physical network,
• When a datagram is fragmented, each fragment has its own header.
• A fragmented-datagram may itself be fragmented if it encounters a network with an even smaller MTU.
• Source host or router is responsible for fragmentation of original datagram into the fragments.
Destination host is responsible for reassembling the fragments into the original datagram.

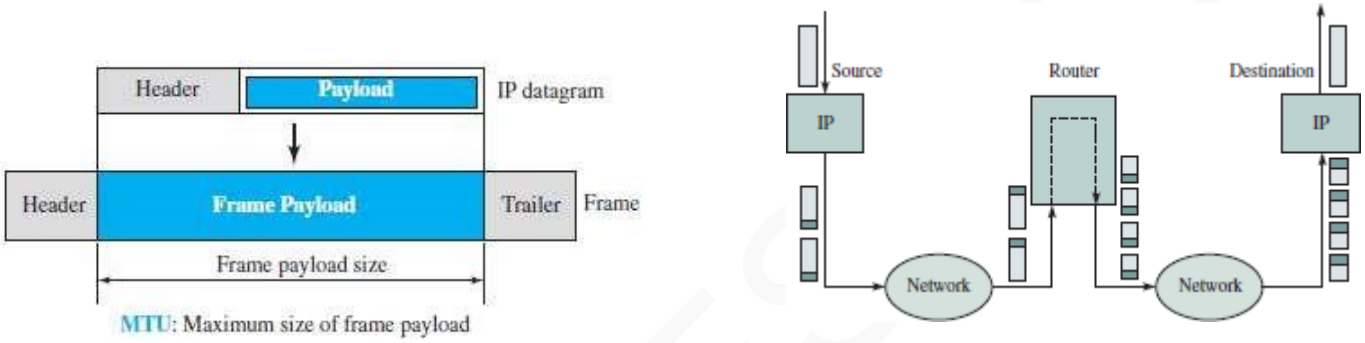


Figure 19.5b: Packet Fragmentation

5.5.3.2 Fields Related to Fragmentation & Reassembly

• Three fields in the IP header are used to manage fragmentation and reassembly:
1) Identification 2) Flags 3) Fragmentation offset.

1) Identification

• This field is used to identify to which datagram a particular fragment belongs to (so that fragments for different packets do not get mixed up).
• To guarantee uniqueness, the IP protocol uses an up-counter to label the datagrams.
• When the IP protocol sends a datagram, IP protocol
→ copies the current value of the counter to the identification field and
→ increments the up-counter by 1.
• When a datagram is fragmented, the value in the identification field is copied into all fragments.
• The identification number helps the destination in reassembling the datagram.

2) Flags

• This field has 3 bits.
1) The leftmost bit is not used.
2) DF bit (Don't Fragment):
i) If DF=1, the router should not fragment the datagram. Then, the router
→ discards the datagram and
→ sends an error-message to the source host.
ii) If DF=0, the router can fragment the datagram if necessary.
3) MF bit (More Fragment):
i) If MF=1, there are some more fragments to come.
ii) If MF=0, this is last fragment.

3) Fragmentation Offset

• This field identifies location of a fragment in a packet.
• This field is the offset of the data in the original datagram.

Example 5.6

A packet has arrived with an M bit value of 0. Is this the first fragment, the last fragment, or a middle fragment? Do we know if the packet was fragmented?

Solution

If the M bit is 0, it means that there are no more fragments; the fragment is the last one. However, we cannot say if the original packet was fragmented or not. A nonfragmented packet is considered the last fragment.

Example 5.7

A packet has arrived with an M bit value of 1. Is this the first fragment, the last fragment, or a middle fragment? Do we know if the packet was fragmented?

Solution

If the M bit is 1, it means that there is at least one more fragment. This fragment can be the first one or a middle one, but not the last one. We don't know if it is the first one or a middle one; we need more information (the value of the fragmentation offset).

Example 5.8

A packet has arrived with an M bit value of 1 and a fragmentation offset value of 0. Is this the first fragment, the last fragment, or a middle fragment?

Solution

Because the M bit is 1, it is either the first fragment or a middle one. Because the offset value is 0, it is the first fragment.

Example 5.9

A packet has arrived in which the offset value is 100. What is the number of the first byte? Do we know the number of the last byte?

Solution

To find the number of the first byte, we multiply the offset value by 8. This means that the first byte number is 800. We cannot determine the number of the last byte unless we know the length of the data.

Example 5.10

A packet has arrived in which the offset value is 100, the value of HLEN is 5, and the value of the total length field is 100. What are the numbers of the first byte and the last byte?

Solution

The first byte number is $100 \times 8 = 800$. The total length is 100 bytes, and the header length is 20 bytes (5×4), which means that there are 80 bytes in this datagram. If the first byte number is 800, the last byte number must be 879.

5.5.4 Options

- This field allows the packet to request special features such as
  - → security level
  - → route to be taken by packet and
  - → timestamp at each router.
- This field can also be used for network testing and debugging.
- As the name implies, options are not required for a datagram.
- The header is made of two parts: 1) Fixed part and 2) Variable part.
  - 1) Maximum size of Fixed part = 20 bytes.
  - 2) Maximum size of Variable part = 40 bytes
- Options are divided into two broad categories: 1) Single-byte options and 2) Multiple-byte options.

1) Single Byte Options

i) No Operation

- This option is used as filler between options.

ii) End of Option

- This option is used for padding at the end of the option field.

2) Multiple Byte Options

i) Record Route

- This option is used to record the routers that handle the datagram.
- This option can list up to 9 router-addresses.

ii) Strict Source Route

- This option is used by the source to pre-determine a route for the datagram.
- Useful purposes: The sender can choose a route with a specific type of service, such as
  - → minimum delay
  - → maximum throughput or
  - → more secure/reliable.
- All the defined-routers must be visited by the datagram.
- If the datagram visits a router that is not on the list, the datagram is discarded.

iii) Loose Source Route

- This option is similar to the strict source route, but it is less rigid.
- Each router in the list must be visited, but the datagram can visit other routers as well.

iv) Timestamp

- This option is used to record the time of datagram processing by a router.
- The time is expressed in milliseconds from midnight GMT (Greenwich Mean Time).
- The recorded-time can help the managers to track the behavior of the routers in the Internet.

5.5.5 Security of IPv4 Datagrams

- Nowadays, the Internet is not secure anymore.
- Three security issues applicable to the IP protocol:
  1) Packet sniffing
  2) Packet modification and
  3) IP spoofing.

1) Packet Sniffing

- Attackers may
  → capture certain packets
  → intercept the packets and
  → make a copy of the packets.
- Packet sniffing is a passive attack.
- Passive attack means the attacker does not modify the contents of the packet.
- The attack is difficult to detect „." sender & receiver may never know that the packet has been copied.
- Solution:
  Although the attack cannot be stopped, encryption of packet may make the attacker"s job difficult.
  The attacker may still sniff the packet, but the content is not detectable (or understandable).

2) Packet Modification

- Attackers may succeed in accessing the content of a packet.
- Then, the attacker can
  → change the address of the packet or
  → change the data of the packet
- Solution:
  The attack can be prevented by data integrity mechanism.
  Data integrity guarantees that the packet is not modified during the transmission.

3) IP Spoofing

- The attacker pretends as a trusted entity and obtains all the secret information.
- For example:
  An attacker sends an IP packet to a bank pretending as legitimate customers.
- Solution:
  The attack can be prevented using an origin-authentication mechanism.

5.5.5.1 IPSec (IP Security)

- IP packets can be protected from the various network-attacks using a protocol called IPSec.
- IPSec protocol & IP protocol can be used to create a connection-oriented service between 2 entities.
- Four services of IPSec:

  1) Defining Algorithms & Keys

  ➢ To create a secure channel b/w two entities, the two entities can agree on some available algorithms and keys.

  2) Packet Encryption

  ➢ To provide privacy, the packets exchanged b/w two parties can be encrypted using the encryption-algorithms and a shared key.
  ➢ This prevents the packet sniffing attack.

  3) Data Integrity

  ➢ Data integrity guarantees that the packet is not modified during the transmission.
  ➢ If the received packet does not pass the data integrity test, the packet is discarded.
  ➢ This prevents the packet modification attack.

  4) Origin Authentication

  ➢ Origin Authentication guarantees that the packet is not created by a pretender.
  ➢ This prevents the IP Spoofing attack.

5.5 ICMP

- ICMP is a network-layer protocol.
- This is used to handle error and other control messages.

5.6.1 MESSAGES

- ICMP messages are divided into 2 broad categories:

1) Error Reporting Messages

➢ These messages report problems that a router or a host may encounter during the processing of datagram.

2) Query Messages

➢ These messages help a host or a network manager get specific information from a router or another host.

➢ For example:

Nodes can discover their neighbors.
Hosts can discover and learn about routers on their network.
Routers can help a node redirect the messages.

- Fields of ICMP messages (Figure 19.8):

1) Type: This field identifies the type of message.

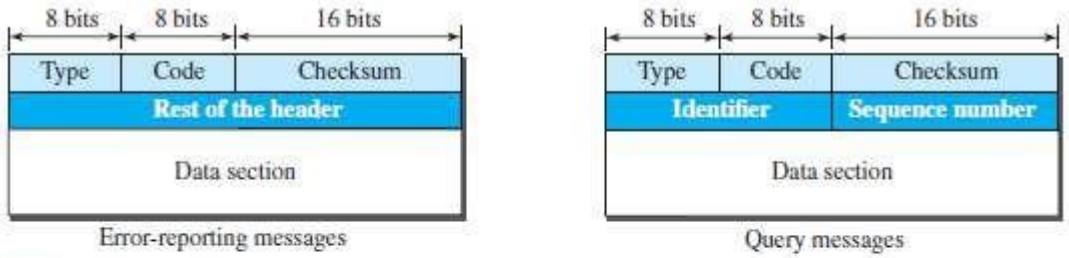
2) Code: This field specifies the reason for the particular message type.

For example,

Type 03 = problem reaching the destinations
Type 11 = problem related to time exceeded.

3) Checksum: This field is used to detect errors in the ICMP message.

4) Data Section: This field can be used for diagnostic purposes by matching the information in the ICMP message with the original data in the IP packet.



Type and code values

Error-reporting messages	Query messages
03: Destination unreachable (codes 0 to 15)	08 and 00: Echo request and reply (only code 0)
04: Source quench (only code 0)	13 and 14: Timestamp request and reply (only code 0)
05: Redirection (codes 0 to 3)	
11: Time exceeded (codes 0 and 1)	
12: Parameter problem (codes 0 and 1)	

Figure 19.8 *General format of ICMP messages*

5.6.1.1 Error Reporting Messages

• Main responsibility of ICMP: To report some errors that may occur during the processing of the datagram (Figure 19.9).
• These messages report problems that a router or a host may encounter during the processing of datagram.
• ICMP does not correct errors; ICMP simply reports the errors to the source.
• Error correction is left to the higher-level protocols (such as TCP or UDP)

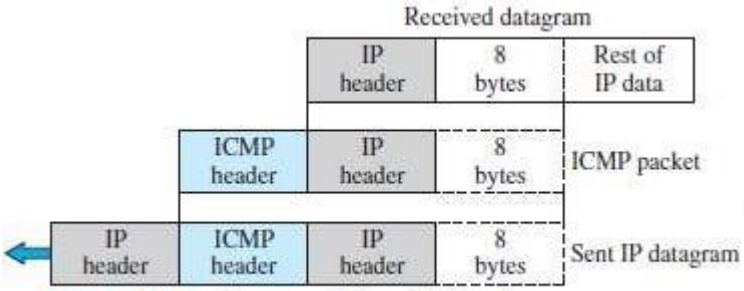


Figure 19.9 *Contents of data field for the error messages*

• Rules for reporting messages:

1) No error-message will be generated for a datagram having a multicast address (or special address).

2) No error-message will be generated in response to a datagram carrying an ICMP error-message.

3) No error-message will be generated for a fragmented datagram that is not the first fragment.

1) Destination Unreachable (Type=3)

• This message is related to problem reaching the destinations.
• This message uses different codes (0 to 15) to define type of error-message.
• Possible values for code field:

Code 0 = network unreachable
Code 1 = host unreachable
Code 2 = protocol unreachable
Code 3 = port unreachable

2) Source Quench (Type=4)

• This message informs the sender that
→ network has encountered congestion and
→ datagram has been dropped.
• The source needs to slow down sending more datagrams.
• In other words, ICMP adds a kind of congestion control mechanism to the IP protocol.

3) Redirection Message (Type=5)

• This is used when the source uses a wrong router to send out its message.
• The router
→ redirects the message to the appropriate router &
→ informs the source to change its default router in the future.
• The IP address of the default router is sent in the message.
• TTL prevents a datagram from being aimlessly circulated in the Internet.
• When TTL becomes 0,
→ the datagram is dropped by the visiting router and
→ a time exceeded message (type 11) is sent to the source.

4) Parameter Problem (Type=12)

• This message can be sent when either
→ there is a problem in the header of a datagram (code 0) or
→ some options are missing or cannot be interpreted (code 1).

5.6.1.2 Query Messages

- These messages help a network manager to get specific information from a router or host.
- Two types of query messages: request (type 8) and reply (type 0).

1) Echo Request & Echo Reply

- These messages are used to determine whether a remote-host is alive.
- A source-host sends an echo request message to destination-host;
  If the destination-host is alive, it responds with an echo reply message.
- Type=8 is used for echo request
  Type=0 is used for echo reply.
- These messages can be used in two debugging tools: ping and traceroute.

2) Timestamp Request & Timestamp Reply

- These messages are used to
  → find the round-trip time between two devices or
  → check whether the clocks in two devices are synchronized.
- The timestamp request sends a number, which defines the time the message is sent.
- The timestamp reply resends another number, which defines the time the message is sent.
- The timestamp reply also includes 2 new numbers representing
  i) the time the request was received and
  ii) the time the response was sent.
- Type=13 is used for timestamp request
  Type=14 is used for timestamp reply.

5.6.2 Debugging Tools

- There are several tools that can be used in the Internet for debugging.
- We can determine the viability of a host or router.
- We can trace the route of a packet.
- Two tools used for debugging: 1) Ping and 2) Traceroute.

5.6.2.1 Ping

- The ping program can be used to find if a host is alive and responding
- Here, ping is used to see how it uses ICMP packets
- The source host sends ICMP echo-request messages;
  The destination, if alive, responds with ICMP echo-reply messages.
- The ping program
  → sets the identifier field in the echo-request and echo-reply message and
  → starts the sequence number from 0; this number is incremented by 1 each time a new message is sent.
- Ping can calculate the round-trip time.
  It inserts the sending time in the data section of the message.
  When the packet arrives, it subtracts the arrival time from the departure time to get the round-trip time (RTT).

5.6.2.2 Traceroute

- The traceroute program can be used to trace the path of a packet from a source to the destination.
- It can find the IP addresses of all the routers that are visited along the path.
- The program is usually set to check for the maximum of 30 hops (routers) to be visited.

Traceroute

- The traceroute program is different from the ping program.
- The ping program gets help from 2 query messages;
  The traceroute program gets help from two error-reporting messages: time-exceeded and destination-unreachable.
- The traceroute is an application layer program, but only the client program is needed.
  In other words, there is no traceroute server program.
- The traceroute application program is encapsulated in a UDP user datagram, but traceroute intentionally uses a port number that is not available at the destination.

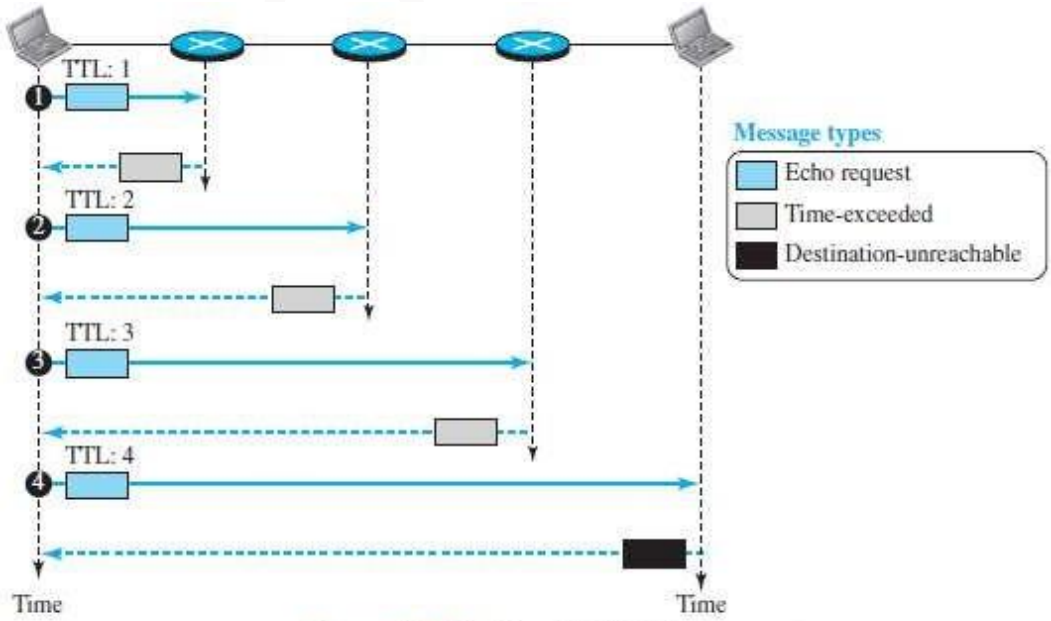


Figure 19.10 *Use of ICMPv4 in traceroute*

5.6 MOBILE IP

- Mobile IP is the extension of IP protocol.
- Mobile IP allows mobile computers to be connected to the Internet.

5.7.1 Addressing

- In Mobile IP, the main problem that must be solved is addressing.

5.7.1.1 Stationary Hosts

- The original IP addressing assumed that a host is stationary.
- A router uses an IP address to route an IP datagram.
- An IP address has two parts: a prefix and a suffix.
- The prefix associates a host with a network.

  For example, the IP address 10.3.4.24/8 defines a host attached to the network 10.0.0.0/8.
- The address is valid only when the host is attached to the network.
- If the network changes, the address is no longer valid.

5.7.1.2 Mobile Hosts

- When a host moves from one network to another, the IP addressing structure needs to be modified.
- The host has two addresses (Figure 19.12):
  1) Home address &
  2) Care-of address

1) Home Address

- Original address of host called the home address.
- The home address is permanent.
- The home address associates the host with its home network.
- Home network is a network that is the permanent home of the host.

2) Care-of-Address

- The care-of address is temporary.
- The care-of address changes as the mobile-host moves from one network to another.
- Care-of address is associated with the foreign network.
- Foreign network is a network to which the host moves.
- When a mobile-host visits a foreign network, it receives its care-of address during the agent discovery and registration phase.

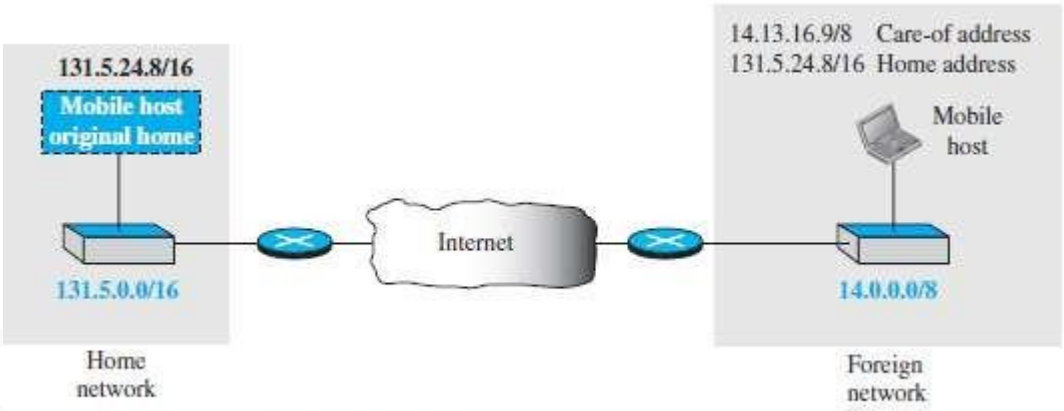


Figure 19.12 *Home address and care-of address*

5.7.2 Agents

• Two agents are required to make change of address transparent to rest of the Internet (Fig 19.13):
1) Home-agent and
2) Foreign-agent.

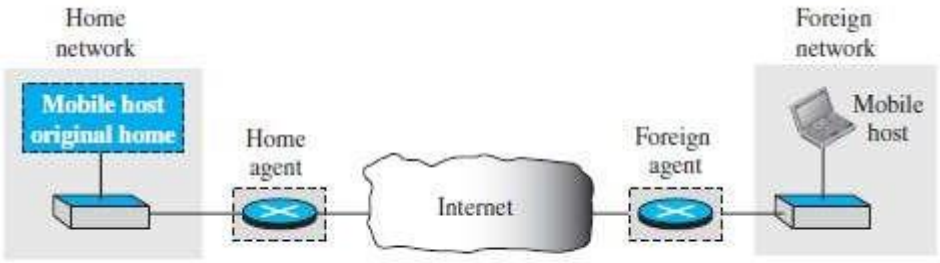


Figure 19.13 *Home agent and foreign agent*

1) Home Agent

• The home-agent is a router attached to the home network.
• The home-agent acts on behalf of mobile-host when a remote-host sends a packet to mobile-host.
• The home-agent receives and delivers packets sent by the remote-host to the foreign-agent.

2) Foreign Agent

• The foreign-agent is a router attached to the foreign network.
• The foreign-agent receives and delivers packets sent by the home-agent to the mobile-host.
• The mobile-host can also act as a foreign-agent i.e. mobile-host and foreign-agent can be the same.
• However, to do this, a mobile-host must be able to receive a care-of address by itself.
• In addition, the mobile-host needs the necessary software to allow it to communicate with the home-agent and to have two addresses: i) its home address and ii) its care-of address.
• This dual addressing must be transparent to the application programs.

Collocated Care-of-Address

➢ When the mobile-host and the foreign-agent are the same, the care-of-address is called a collocated care-of-address.
➢ Advantage:
1) mobile-host can move to any network w/o worrying about availability of a foreign-agent.
➢ Disadvantage:
1) The mobile-host needs extra software to act as its own foreign-agent.

5.7.3 Three Phases

• To communicate with a remote-host, a mobile-host goes through 3 phases (Figure 19.14):

1) Agent Discovery: involves the mobile-host, the foreign-agent, and the home-agent.

2) Registration: involves the mobile-host, the foreign-agent, and the home-agent.

3) Data Transfer: Here, the remote-host is also involved.

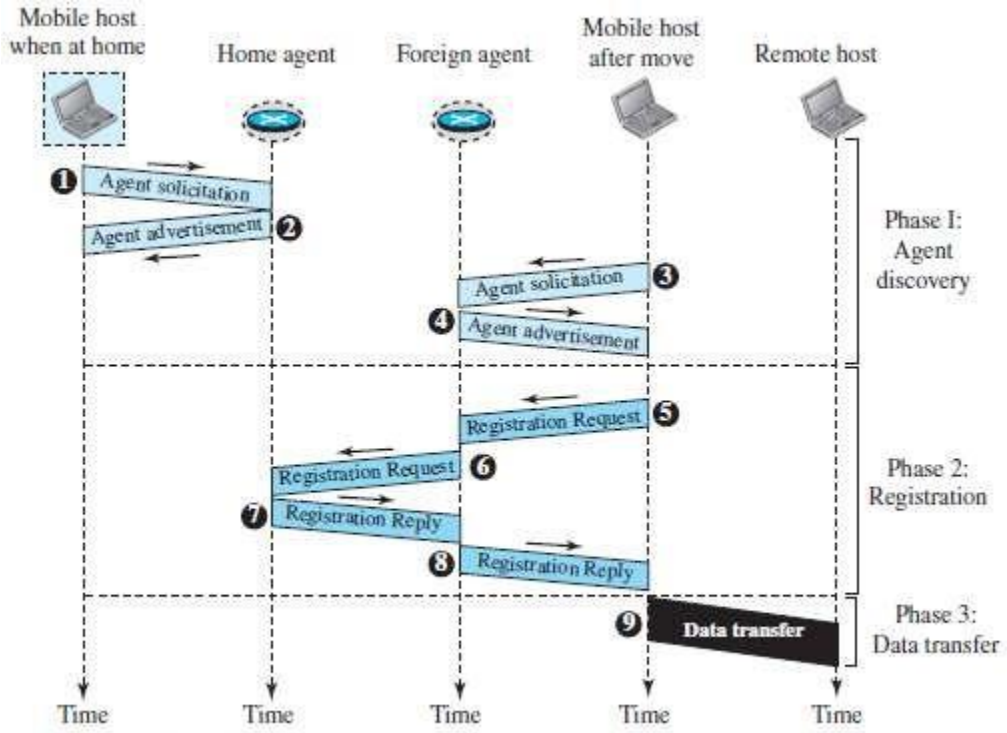


Figure 19.14 *Remote host and mobile host communication*

5.7.3.1 Agent Discovery

• Agent discovery consists of two subphases:

1) A mobile-host must discover (learn the address of) a home-agent before it leaves its home network.

2) A mobile-host must also discover a foreign-agent after it has moved to a foreign network.

• This discovery consists of learning the care-of address as well as the foreign-agent"s address.

• Two types of messages are used: i) advertisement and ii) solicitation.

1) Agent Advertisement

• When a router advertises its presence on a network using an ICMP router advertisement, it can append an agent advertisement to the packet if it acts as an agent.

Figure 19.15 *Agent advertisement*

• Various fields are (Figure 19.15):

1) Type

➢ This field is set to 16.

2) Length

➢ This field defines the total length of the extension message.

3) Sequence Number

➢ This field holds the message number.

➢ The recipient can use the sequence number to determine if a message is lost.

4) Lifetime

➢ This field defines the number of seconds that the agent will accept requests.

➢ If the value is a string of 1s, the lifetime is infinite.

5) Code

➢ This field is a flag in which each bit is set (1) or unset (0) (Table 19.1).

Table 19.1 *Code Bits*

Bit	*Meaning*
0	Registration required. No collocated care-of address.
1	Agent is busy and does not accept registration at this moment.
2	Agent acts as a home agent.
3	Agent acts as a foreign agent.
4	Agent uses minimal encapsulation.
5	Agent uses generic routing encapsulation (GRE).
6	Agent supports header compression.
7	Unused (0).

6) Care-of Addresses

➢ This field contains a list of addresses available for use as care-of addresses.

➢ The mobile-host can choose one of these addresses.

➢ The selection of this care-of address is announced in the registration request.

2) Agent Solicitation

• When a mobile-host has moved to a new network and has not received agent advertisements, it can initiate an agent solicitation.

• It can use the ICMP solicitation message to inform an agent that it needs assistance

5.7.3.2 Registration

• After a mobile-host has moved to a foreign network and discovered the foreign-agent, it must register.

• Four aspects of registration:

1) The mobile-host must register itself with the foreign-agent.

2) The mobile-host must register itself with its home-agent. This is normally done by the foreign-agent on behalf of the mobile-host.

3) The mobile-host must renew registration if it has expired.

4) The mobile-host must cancel its registration (deregistration) when it returns home.

5.7.3.2.1 Request & Reply

• To register with the foreign-agent and the home-agent, the mobile-host uses a registration request and a registration reply.

1) Registration Request

• A registration request is sent from the mobile-host to the foreign-agent

→ to register its care-of address and

→ to announce its home address and home-agent address.

• Foreign-agent, after receiving and registering the request, relays the message to the home-agent.

• The home-agent now knows the address of the foreign-agent because the IP packet that is used for relaying has the IP address of the foreign-agent as the source address.

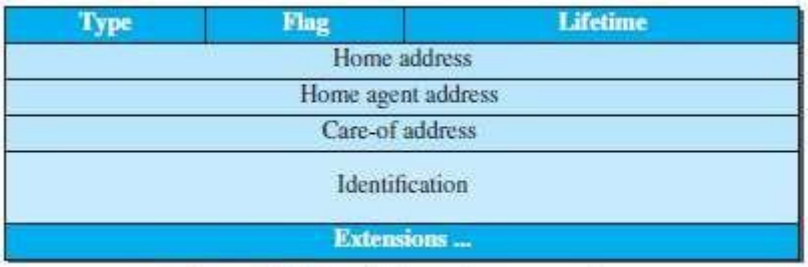


Figure 19.16 *Registration request format*

• Various fields are (Figure 19.16):

1) Type

➢ This field defines the type of message.

➢ For a request message the value of this field is 1.

2) Flag

➢ This field defines forwarding information.

➢ The value of each bit can be set or unset (Table 19.2).

Table 19.2 *Registration request flag field bits*

Bit	*Meaning*
0	Mobile host requests that home agent retain its prior care-of address.
1	Mobile host requests that home agent tunnel any broadcast message.
2	Mobile host is using collocated care-of address.
3	Mobile host requests that home agent use minimal encapsulation.
4	Mobile host requests generic routing encapsulation (GRE).
5	Mobile host requests header compression.
6–7	Reserved bits.

3) Lifetime

➢ This field defines the number of seconds the registration is valid.

i) If the field is a string of 0s, the request message is asking for deregistration.

ii) This field If the field is a string of 1s, the lifetime is infinite.

4) Home Address

➢ This field contains the permanent (first) address of the mobile-host.

5) Home Agent Address

➢ This field contains the address of the home-agent.

6) Care-of-Address

- This field is the temporary (second) address of the mobile-host.

7) Identification

- This field contains a 64-bit number that is inserted into the request by the mobile-host.
- This field matches a request with a reply.

8) Extensions

- This field is used for authentication.
- This field allows a home-agent to authenticate the mobile agent.

2) Registration Reply

- A registration reply is sent from home-agent to foreign-agent and then relayed to the mobile-host.
- The reply confirms or denies the registration request. (Figure 19.17)
- The fields are similar to registration request with the 3 exceptions:

1) The value of the type field is 3.

2) The code field replaces the flag field and shows the result of the registration request (acceptance or denial).

3) The care-of address field is not needed.

Figure 19.17 *Registration reply format*

5.7.3.3 Data Transfer

• After agent discovery & registration, a mobile-host can communicate with a remote-host (Fig 19.17).

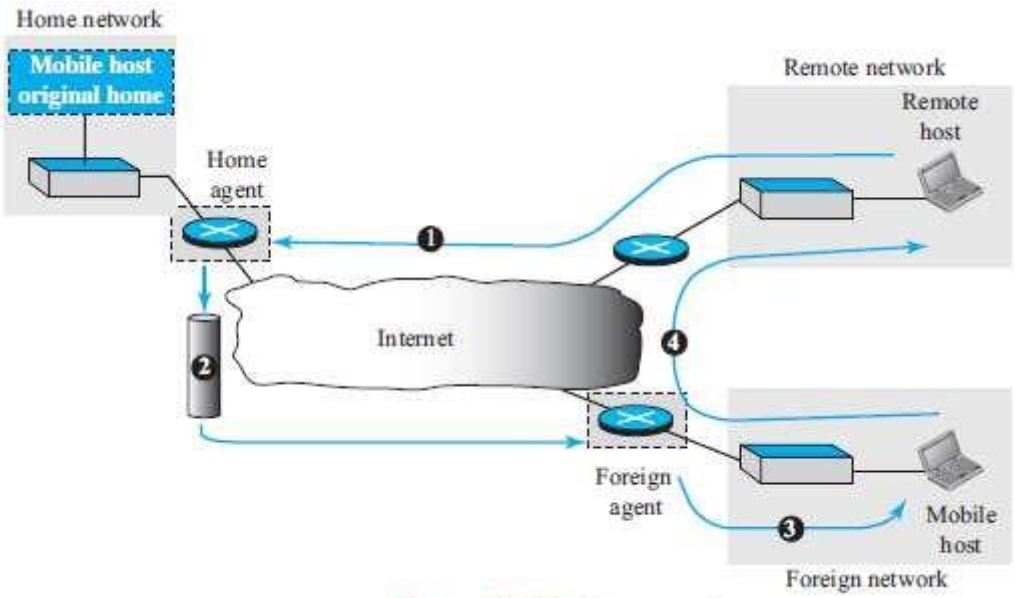


Figure 19.18 *Data transfer*

• Here we have 4 cases (Figure 19.18):

1) From Remote Host to Home Agent

• When a remote-host wants to send a packet to the mobile-host, the remote-host uses
→ address of itself as the source address and
→ home address of the mobile-host as the destination address.
• In other words, the remote-host sends a packet as though the mobile-host is at its home network.
• The packet is intercepted by the home-agent, which pretends it is the mobile-host.
• This is done using the proxy ARP technique (Path 1 of Figure 19.18).

2) From Home Agent to Foreign Agent

• After receiving the packet, the home-agent sends the packet to the foreign-agent, using the tunneling concept.
• The home-agent encapsulates the whole IP packet inside another IP packet using its address as the source and the foreign-agent"s address as the destination. (Path 2 of Figure 19.18).

3) From Foreign Agent to Mobile Host

• When the foreign-agent receives the packet, it removes the original packet.
• However, since the destination address is the home address of the mobile-host, the foreign-agent consults a registry table to find the care-of address of the mobile-host. (Otherwise, the would just be sent back to the home network.)
• The packet is then sent to the care-of address (Path 3 of Figure 19.18).

4) From Mobile Host to Remote Host

• When a mobile-host wants to send a packet to a remote-host (for example, a response to the packet it has received), it sends as it does normally.
• The mobile-host prepares a packet with its home address as the source, and the address of the remote-host as the destination.
• Although the packet comes from the foreign network, it has the home address of the mobile-host (Path 4 of Figure 19.18).

5.7.4 Inefficiency in Mobile IP

• Communication involving mobile IP can be inefficient.
• The inefficiency can be severe or moderate.
1) The severe case is called double crossing or 2X.
2) The moderate case is called triangle routing or dog-leg routing.

5.7.4.1 Double Crossing

• Double crossing occurs when a remote-host communicates with a mobile-host that has moved to the same network (or site) as the remote-host (Figure 19.19).
• When the mobile-host sends a packet to the remote-host, there is no inefficiency; the communication is local.
• However, when remote-host sends a packet to mobile-host, the packet crosses the Internet twice.
• Since a computer usually communicates with other local computers (principle of locality), the inefficiency from double crossing is significant.

5.7.4.2 Triangle Routing

• Triangle routing occurs when the remote-host communicates with a mobile-host that is not attached to the same network (or site) as the mobile-host.
• When the mobile-host sends a packet to the remote-host, there is no inefficiency.

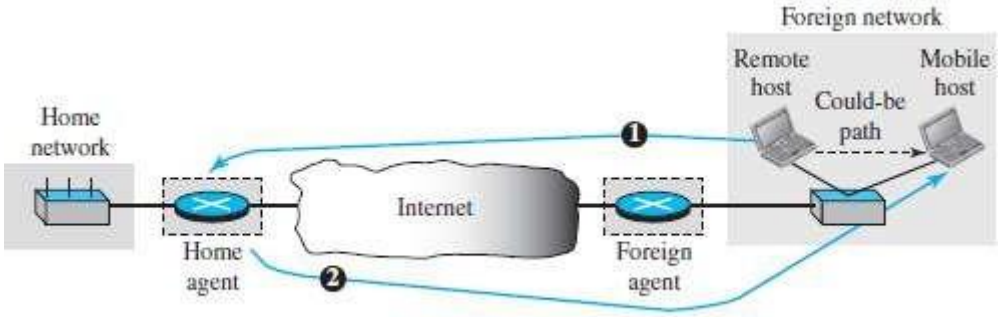


Figure 19.19 *Double crossing*

• However, when the remote-host sends a packet to the mobile-host, the packet goes from the remote-host to the home-agent and then to the mobile-host.
• The packet travels the two sides of a triangle, instead of just one side (Figure 19.20).

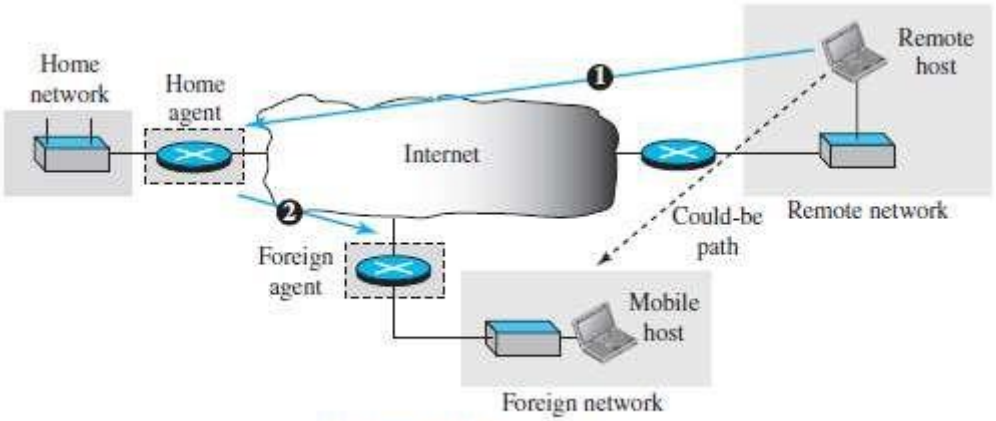


Figure 19.20 *Triangle routing*

Solution

• One solution to inefficiency is for the remote-host to bind the care-of address to the home address of a mobile-host.
• For example, when a home-agent receives the first packet for a mobile-host, it forwards the packet to the foreign-agent; it could also send an update binding packet to the remote-host so that future packets to this host could be sent to the care-of address.
• The remote-host can keep this information in a cache.
• The problem with this strategy is that the cache entry becomes outdated once the mobile-host moves.
• In this case, the home-agent needs to send a warning packet to the remote-host to inform it of the change.

MODULE 5(CONT.): NEXT GENERATION IP

5.7 IPv6 ADDRESSING

- The main reason for migration from IPv4 to IPv6 is the small size of the address-space in IPv4.
- Size of IPv6 address =128 bits (four times the address length in IPv4, which is 32 bits).

5.8.1 Representation

- Two notations can be used to represent IPv6 addresses: 1) binary and 2) colon hexadecimal.

Binary (128 bits)	1111111011110110 ... 1111111100000000
Colon Hexadecimal	FEF6:BA98:7654:3210:ADEF:BBFF:2922:FF00

5.8.2 Address Space

- The address-space of IPv6 contains 2^{128} addresses.

5.8.2.1 Three Address Types

- Three types of destination address: 1) Unicast 2) Anycast and 3) Multicast.

1) Unicast Address

- A unicast address defines a single interface (computer or router).
- The packet with a unicast address will be delivered to the intended recipient.

2) Anycast Address

- An anycast address defines a group of computers that all share a single address.
- A packet with an anycast address is delivered to only one member of the group.
- The member is the one who is first reachable.

3) Multicast Address

- A multicast address also defines a group of computers.
- Difference between anycasting and multicasting.
  - i) In anycasting, only one copy of the packet is sent to one of the members of the group.
  - ii) in multicasting each member of the group receives a copy.

5.8.3 Address Space Allocation

• The address-space is divided into several blocks of varying size.
• Each block is allocated for a special purpose.

Table 22.1 *Prefixes for assigned IPv6 addresses*

Block prefix	*CIDR*	*Block assignment*	*Fraction*
0000 0000	0000::/8	Special addresses	1/256
001	2000::/3	Global unicast	1/8
1111 110	FC00::/7	Unique local unicast	1/128
1111 1110 10	FE80::/10	Link local addresses	1/1024
1111 1111	FF00::/8	Multicast addresses	1/256

5.8.3.1 Global Unicast Addresses

• The block in the address-space used for unicast communication b/w 2 hosts in the Internet is called global unicast address block.
• CIDR for the block is 2000::/3. This means that the three leftmost bits are the same for all addresses in this block (001).
• The size of this block is 2^{125} bits, which is more than enough for Internet expansion for many years to come.
• An address in the block is divided into 3 parts (Figure 22.1):
    1) Global routing prefix (n bits)
    2) Subnet identifier (m bits) and
    3) Interface identifier (q bits).

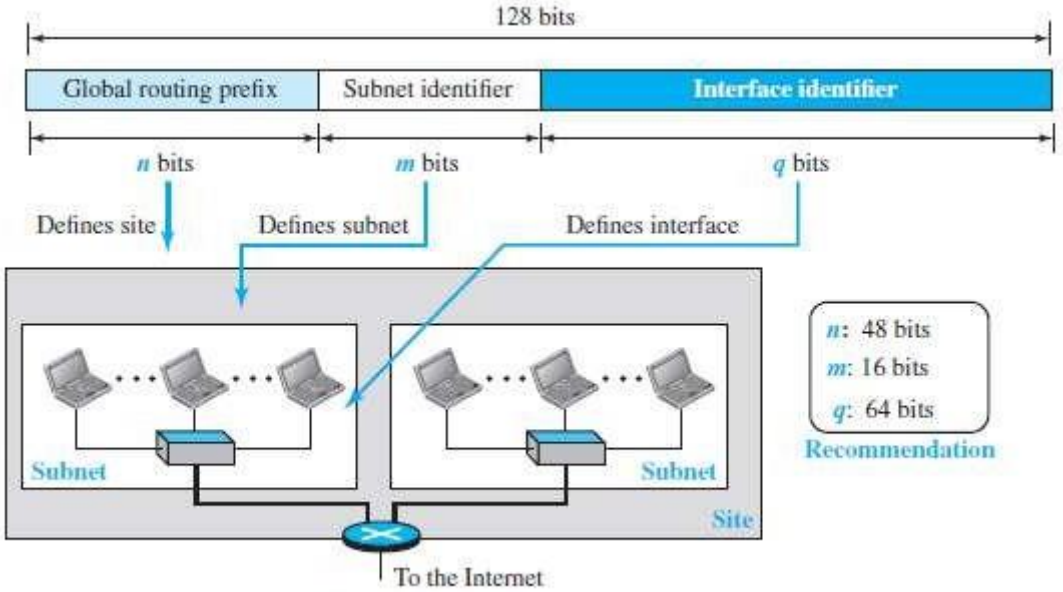


Figure 22.1 *Global unicast address*

• The global routing prefix is used to route the packet through the Internet to the organization site, such as the ISP that owns the block.
• Since the first 3 bits in this part are fixed (001), the rest of the 45 bits can be defined for up to 2^{45} sites (a private organization or an ISP).
    1) The global routers in Internet route a packet to its destination site based on the value of n.
    2) The next m bits define a subnet in an organization.
    3) The last q bits define the interface identifier.
• Two link layer addressing schemes:
    1) 64-bit extended unique identifier (EUI-64) defined by IEEE and
    2) 48-bit link-layer address defined by Ethernet.

1) Mapping EUI-64

• To map a 64-bit physical address, the global/local bit of this format needs to be changed from 0 to 1 (local to global) to define an interface address (Figure 22.2).

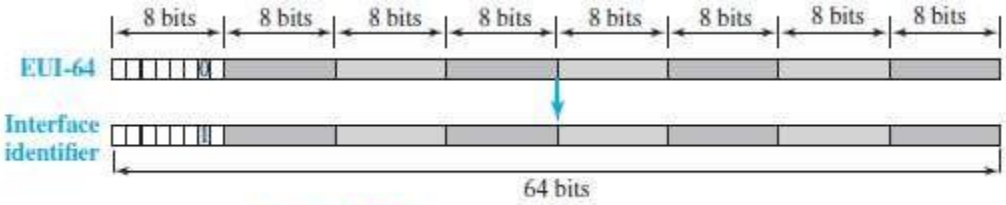


Figure 22.2 *Mapping for EUI-64*

2) Mapping Ethernet MAC Address

• Mapping a 48-bit Ethernet address into a 64-bit interface identifier is more involved.
• We need to change the local/global bit to 1 and insert an additional 16 bits.
• The additional 16 bits are defined as 15 ones followed by one zero, or $FFFE_{16}$ (Figure 22.3).

5.8.3.2 Special Addresses

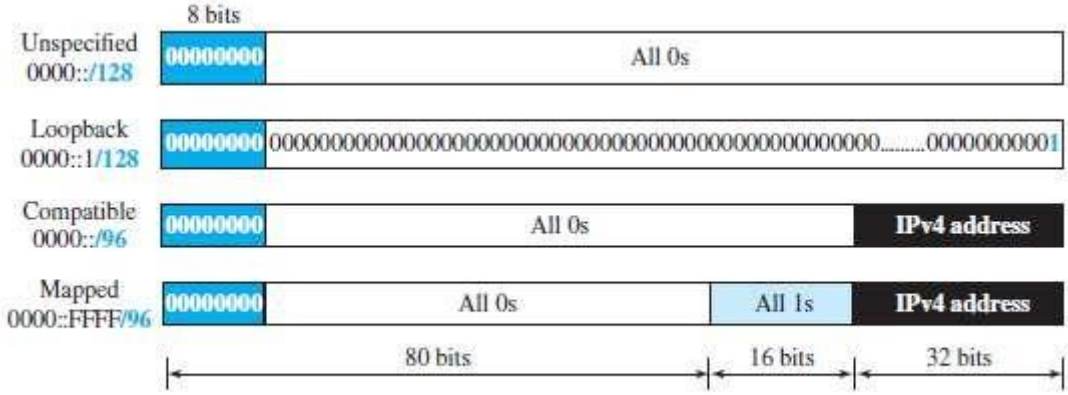


Figure 22.4 *Special addresses*

• Following are different special addresses (Figure 22.4):

1) Unspecified Address

• The unspecified address is a subblock containing only one address.
• This address is used during bootstrap when a host does not know its own address and wants to send an inquiry to find it.

2) Loopback Address

• The loopback address also consists of one address.

3) Transition Address

• During the transition from IPv4 to IPv6, hosts can use their IPv4 addresses embedded in IPv6 addresses.
• Two formats have been designed for this purpose: compatible and mapped.

1) Compatible Address

➢ A compatible address is an address of 96 bits of zero followed by 32 bits of IPv4 address.
➢ It is used when a computer using IPv6 wants to send a message to another computer using IPv6.

2) Mapped Address

➢ A mapped address is used when a computer already migrated to version 6 wants to send an address to a computer still using version 4.

5.8.3.3 Other Assigned Blocks

• IPv6 uses 2 large blocks for private addressing and one large block for multicasting (Figure 22.5).

1) Unique Local Unicast Block

• A subblock in a unique local unicast block can be privately created and used by a site.
• The packet carrying this type of address as the destination address is not expected to be routed.
• This type of address has the identifier 1111 110.
• The next bit can be 0 or 1 to define how the address is selected (locally or by an authority).

2) Link Local Block

• A subblock in link local block can be used as a private address in a network.
• This type of address has the block identifier 1111111010.
• The next 54 bits are set to zero.
• The last 64 bits can be changed to define the interface for each computer.

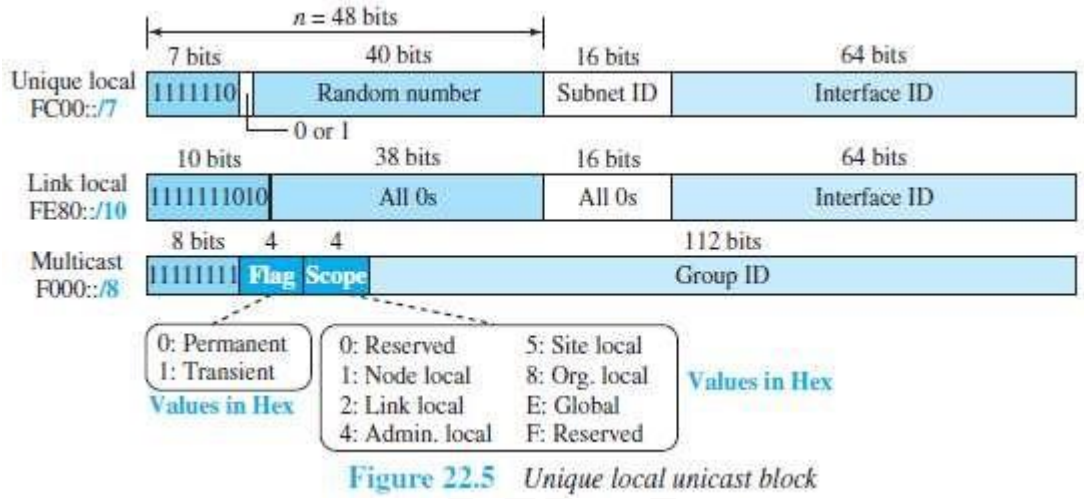


Figure 22.5 *Unique local unicast block*

5.8.4 Autoconfiguration

• When a host in IPv6 joins a network, it can configure itself using the following process:

1) The host first creates a link local address for itself.
- This is done by
  → taking the 10-bit link local prefix (1111 1110 10)
  → adding 54 zeros and
  → adding the 64-bit interface identifier.
- The result is a 128-bit link local address.

2) The host then tests to see if this link local address is unique and not used by other hosts.
- Since the 64-bit interface identifier is supposed to be unique, the link local address generated is unique with a high probability.
- To check uniqueness, the host
  → sends a neighbor solicitation message and
  → waits for a neighbor advertisement message.
- If any host in the subnet is using this link local address, the process fails and the host cannot auto-configure itself.

3) If the uniqueness of the link local address is passed, the host stores this address as its link local address (for private communication), but it still needs a global unicast address.
- The host then sends a router solicitation message to a local router.
- If there is a router running on the network, the host receives a router advertisement message that includes
  → global unicast prefix and
  → subnet prefix that the host needs to add to its interface identifier to generate its global unicast address.
- If the router cannot help the host with the configuration, it informs the host in the router advertisement message (by setting a flag).

5.8 THE IPv6 PROTOCOL

5.9.1 Changes from IPv4 to IPv6 (Advantages of IPv6)

1) Header Format

- IPv6 uses a new header format.
- Options are
  - → separated from the base-header and
  - → inserted between the base-header and the data.
- This speeds up the routing process (because most of the options do not need to be checked by routers).

2) New Options

- IPv6 has new options to allow for additional functionalities.

3) Extension

- IPv6 is designed to allow the extension of the protocol if required by new technologies or applications.

4) Resource Allocation

- In IPv6,
  - → type-of-service (TOS) field has been removed
  - → two new fields: 1) traffic class and 2) flow label, are added to enable the source to request special handling of the packet.
- This mechanism can be used to support real-time audio and video.

5) Security

- The encryption option provides confidentiality of the packet.
- The authentication option provides integrity of the packet.

5.9.2 Packet Format

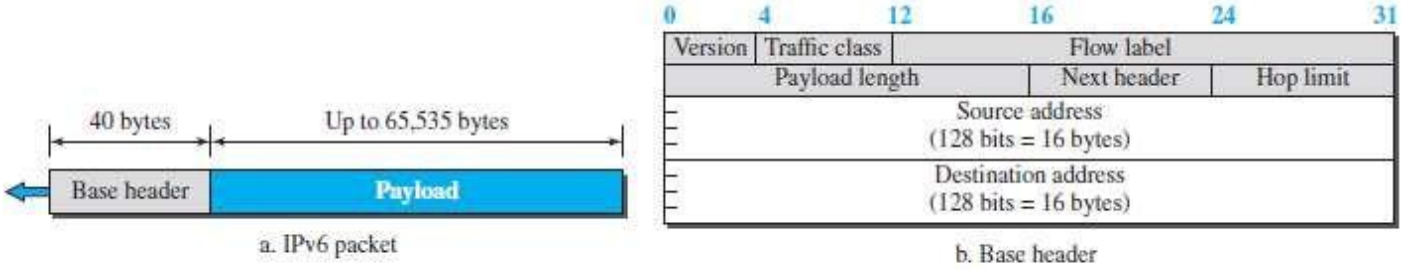


Figure 22.6 *IPv6 datagram*

- IP header contains following fields (Figure 22.6):

1) Version

➢ This specifies version number of protocol. For IPv6, version=6.

2) Traffic Class

➢ This field is used to distinguish different payloads with different delivery requirements.
(Traffic class replaces the type-of-service field in IPv4).

3) Flow Label

➢ This field is designed to provide special handling for a particular flow of data.

4) Payload Length

➢ This indicates length of data (excluding header). Maximum length=65535 bytes.

➢ The length of the base-header is fixed (40 bytes); only the length of the payload needs to be defined.

5) Next Header

➢ This identifies type of extension header that follows the basic header.

6) Hop Limit

➢ This specifies number of hops the packet can travel before being dropped by a router.
(Hop limit serves the same purpose as the TTL field in IPv4).

7) Source and Destination Addresses

➢ These identify source host and destination host respectively.

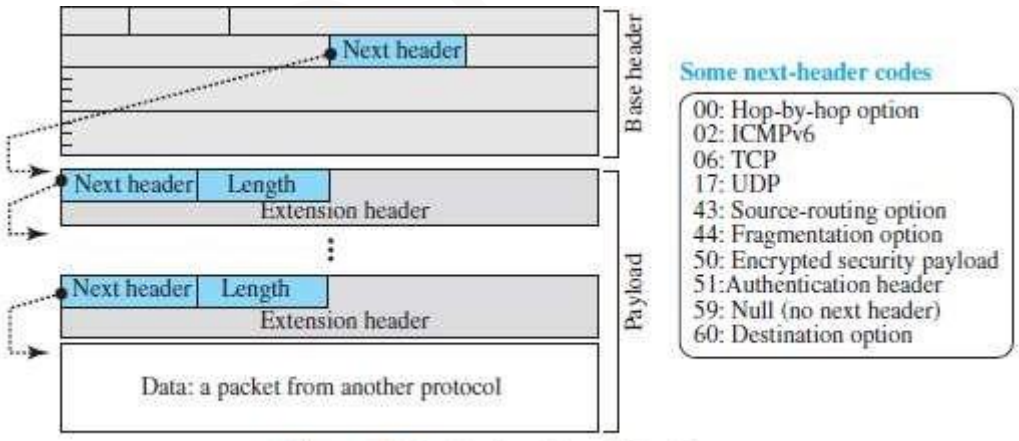


Figure 22.7 *Payload in an IPv6 datagram*

8) Payload

➢ The payload contains zero or more extension headers (options) followed by the data from other protocols (UDP, TCP, and so on).

➢ The payload can have many extension headers as required by the situation.

➢ Each extension header has 2 mandatory fields (Figure 22.7):
1) Next header and
2) Length

- Two mandatory fields are followed by information related to the particular option.

5.9.2.1 Concept of Flow & Priority in IPv6

- To a router, a flow is a sequence of packets that share the same characteristics such as
  - → traveling the same path
  - → using the same resources or
  - → having the same kind of security
- A router that supports the handling of flow labels has a flow label table.
- The table has an entry for each active flow label.
  Each entry defines the services required by the corresponding flow label.
- When a router receives a packet, the router consults its flow label table.
- Then, the router provides the packet with the services mentioned in the entry.
- A flow label can be used to support the transmission of real-time audio/video.
- Real-time audio/video requires resources such as
  - → high bandwidth
  - → large buffers or
  - → long processing time
- Resource reservation guarantees that real-time data will not be delayed due to a lack of resources.

5.9.2.2 Fragmentation & Reassembly

- Fragmentation of the packet is done only by the source, but not by the routers.
  The reassembling is done by the destination.
- At routers, the fragmentation is not allowed to speed up the processing in the router.
- Normally, the fragmentation of a packet in a router needs a lot of processing. This is because
  1) The packets need to be fragmented.
  2) All fields related to the fragmentation need to be recalculated.
- The source will
  - → check the size of the packet and
  - → make the decision to fragment the packet or not.
- If packet-size is greater than the MTU of the network, the router will drop the packet.
- Then, the router sends an error message to inform the source.

5.9.3 Extension Header

- An IP packet is made of
  - → base-header &
  - → some extension headers.
- Length of base header = 40 bytes.
- To support extra functionalities, extension headers can be placed b/w base header and payload.
- Extension headers act like options in IPv4.
- Six types of extension headers (Figure 22.8):
  - 1) Hop-by-hop option
  - 2) Source routing
  - 3) Fragmentation
  - 4) Authentication
  - 5) Encrypted security payload
  - 5) Destination option.

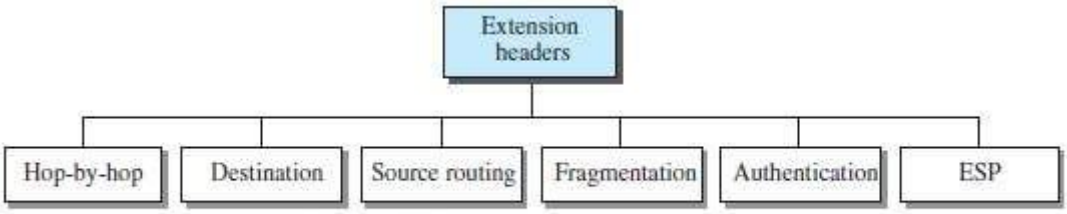


Figure 22.8 *Extension header types*

1) Hop-by-Hop Option

- This option is used when the source needs to pass information to all routers visited by the datagram.
- Three options are defined: i) Pad1, ii) PadN, and iii) Jumbo payload.

i) Pad1

- This option is designed for alignment purposes.
- Some options need to start at a specific bit of the 32-bit word.
- Pad1 is added, if one byte is needed for alignment.

ii) PadN

- PadN is similar in concept to Pad1.
- The difference is that PadN is used when 2 or more bytes are needed for alignment.

iii) Jumbo Payload

- This option is used when larger packet has to be sent. (> 65,535 bytes)
- Large packets are referred to as jumbo packets.
- Maximum length of payload = 65,535 bytes.

2) Destination Option

- This option is used when the source needs to pass information to the destination only.
- Intermediate routers are not allowed to access this information.
- Two options are defined: i) Pad1 & ii) PadN

3) Source Routing

- This option combines the concepts of
  - → strict source routing and
  - → loose source routing.

4) Fragmentation

- In IPv6, only the original source can fragment.
- A source must use a "Path MTU Discovery technique" to find the smallest MTU along the path from the source to the destination.
- Minimum size of MTU = 1280 bytes. This value is required for each network connected to the Internet.
- If a source does not use a Path MTU Discovery technique, the source fragments the datagram to a size of 1280 bytes.

5) Authentication

- This option has a dual purpose:
  - i) Validates the message sender: This is needed so the receiver can be sure that a message is from the genuine sender and not from an attacker.
  - ii) Ensures the integrity of data: This is needed to check that the data is not altered in transition by some attacker.

6) Encrypted Security Payload (ESP)

- This option provides confidentiality and guards against attacker.

5.9.3.1 Comparison of Options between IPv4 and IPv6

1) The no-operation and end-of-option options in IPv4 are replaced by Pad1 and PadN options in IPv6.
2) The record route option is not implemented in IPv6 because it was not used.
3) The timestamp option is not implemented because it was not used.
4) The source route option is called the source route extension header in IPv6.
5) The fragmentation fields in the base-header section of IPv4 have moved to the fragmentation extension header in IPv6.
6) The authentication extension header is new in IPv6.
7) The encrypted security payload extension header is new in IPv6.

5.9 THE ICMPv6 PROTOCOL

• ICMP, ARP & IGMP protocols in IPv4 are combined into one single protocol called ICMPv6 (Fig 22.9).

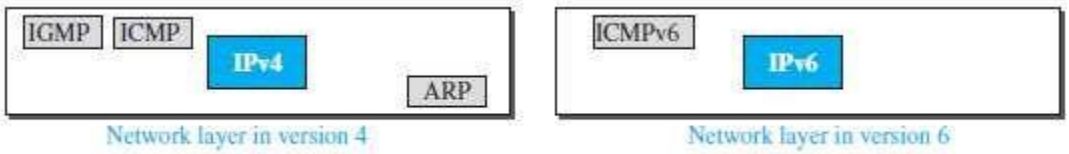


Figure 22.9 *Comparison of network layer in version 4 and version 6*

• Four groups of messages (Figure 22.10):
1) Error-reporting messages
2) Informational messages
3) Neighbor-discovery messages and
4) Group-membership messages.

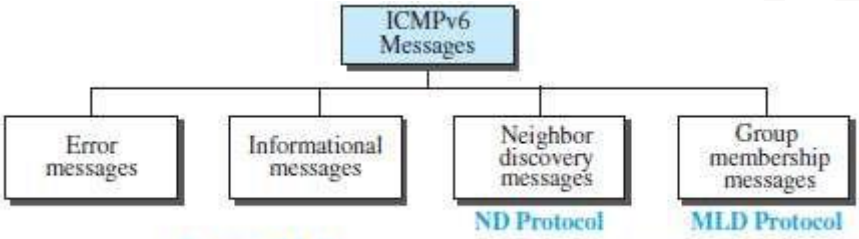


Figure 22.10 *Categories of ICMPv6 messages*

5.10.1 Error-Reporting Messages

• Main responsibility of ICMP: Report errors.
• ICMP forms an error packet, which is then encapsulated in the datagram.
• The encapsulated datagram is delivered to the original source.
• Four types of errors:
1) Destination unreachable
2) Packet too big
3) Time exceeded and
4) Parameter problems.

1) Destination Unreachable Message

➢ Here, a router cannot forward a datagram or a host cannot deliver the datagram to the upper layer protocol.
➢ So, the router/host
→ discards the datagram and
→ sends a destination-unreachable message to the source.

2) Packet Too Big Message

➢ Fragmentation of the packet is done only by the source, but not by the routers.
➢ If a router receives a datagram larger than MTU size of the network, the router
→ discards the datagram and
→ sends a packet-too-big message to the source.

3) Time Exceeded Message

➢ A time-exceeded error message is generated in 2 cases:
i) When the TTL value becomes zero and
ii) When not all fragments of a datagram have arrived in the time-limit.

4) Parameter Problem Message

➢ Any missing value in the datagram-header can create serious problems.
➢ If a router discovers any missing value in any field, the router
→ discards the datagram and
→ sends a parameter-problem message to the source.

5.10.2 Informational Messages

- Two types of messages: i) echo request and ii) echo reply.
- These 2 messages are used to check whether 2 devices can communicate with each other.
- A source-host can send an echo-request message to another host.
  The destination-host can respond with the echo-reply message to the source-host.

5.10.3 Neighbor Discovery Messages

- Two new protocols are used:
  1) Neighbor-Discovery (ND) protocol and
  2) Inverse-Neighbor-Discovery (IND) protocol.
- These 2 protocols are used by nodes on the same link for 3 main purposes:
  1) Hosts use the ND protocol to find routers in the neighborhood that will forward packets for them.
  2) Nodes use the ND protocol to find the link-layer addresses of neighbors.
  3) Nodes use the IND protocol to find the IPv6 addresses of neighbors.
- Seven types of errors:

1) Router Solicitation Message

- A host/router uses router-solicitation message to find a router in n/w that can forward a datagram.
- Physical address of the host/router is included to make the response easier for the router.

2) Router Advertisement Message

- A host/router sends the router-advertisement message in response to a router solicitation message.

3) Neighbor Solicitation Message

- The neighbor solicitation message has the same duty as the ARP request message.
- A host uses the neighbor solicitation message when the host has a message to send to a neighbor.
- The sender knows the IP address of the receiver, but needs the physical address of the receiver.
- The physical address is needed for the datagram to be encapsulated in a frame.

4) Neighbor Advertisement Message

- A host sends the neighbor-advertisement message in response to a neighbor solicitation message.

5) Redirection Message

- The purpose of the redirection message is the same as for version 4.
- However, the format of the packet now accommodates the size of the IP address in version 6.
- Also, an option is added to let the host know the physical address of the target router.

6) Inverse Neighbor Solicitation Message

- A host uses inverse-neighbor-solicitation message to know the physical address of a neighbor, but not the neighbor"s IP address.
- The message is encapsulated in a datagram using a multicast address.
- The node must send the following 2 information in the option field:
  i) Physical address of the sender and
  ii) Physical address of the target node.
- The sender can also include its IP address and the MTU value for the link.

7) Inverse Neighbor Advertisement Message

- A host sends the inverse-neighbor-advertisement message in response to a inverse-neighbor-discovery message.

5.10.4 Group Membership Messages

• The management of multicast delivery handling in IPv4 is given to the IGMPv3 protocol.
• In IPv6, this responsibility is given to the Multicast Listener Delivery protocol.
• MLDv2 has 2 types of messages:
    1) Membership-query message and
    2) Membership-report message.
• The first type can be divided into 3 subtypes: i) General, ii) Group-specific, and iii) Group-and-source specific.

1) Membership Query Message

• A router sends a membership-query message to find active group-members in the network.
• The format of the membership-query in MLDv2 is exactly the same as the one in IGMPv3 three exceptions:
    i) Size of the multicast address & source address has been changed from 32 bits to 128 bits.
    ii) The field size is in the maximum response code field, in which the size has been changed from 8 bits to 16 bits.
    iii) The format of the first 8 bytes matches the format for other ICMPv6 packets because MLDv2 is considered to be part of ICMPv6.

2) Membership Report Message

• The format of the membership-report in MLDv2 is exactly the same as the one in IGMPv3 one exception:
    i) Size of the multicast address & source address has been changed from 32 bits to 128 bits.
• In particular, the record type is the same as the one defined for IGMPv3 (types 1 to 6).

5.10 TRANSITION FROM IPv4 TO IPv6

5.11.1 Strategies

• Three strategies have been devised for transition:
1) Dual stack
2) Tunneling and
3) Header translation.

1) Dual Stack

• Recommended: All hosts must run IPv4 and IPv6 (dual stack) simultaneously until all the Internet uses IPv6 (Figure 22.11).

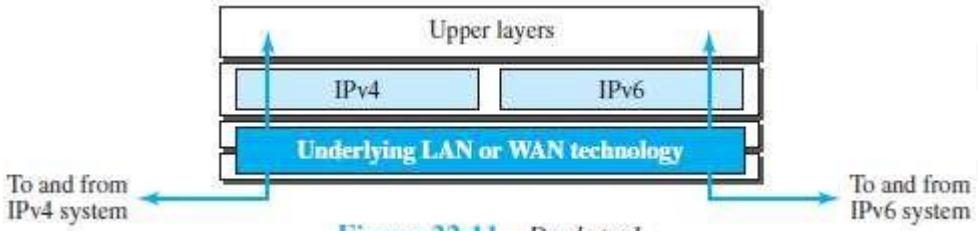


Figure 22.11 *Dual stack*

• To determine which version to use, the source queries the DNS.
i) If the DNS returns an IPv4 address, the source sends an IPv4 packet.
ii) If the DNS returns an IPv6 address, the source sends an IPv6 packet.

2) Tunneling

• Tunneling is a strategy used when
→ two computers using IPv6 want to communicate with each other and
→ the packet must pass through an IPv4 network.
• To pass through IPv4 network, the packet must have an IPv4 address (Figure 22.12).
• So,
i) IPv6 packet is encapsulated in an IPv4 packet when the packet enters the IPv4 network.
ii) IPv6 packet is decapsulated from an IPv4 packet when the packet exits the IPv4 network.

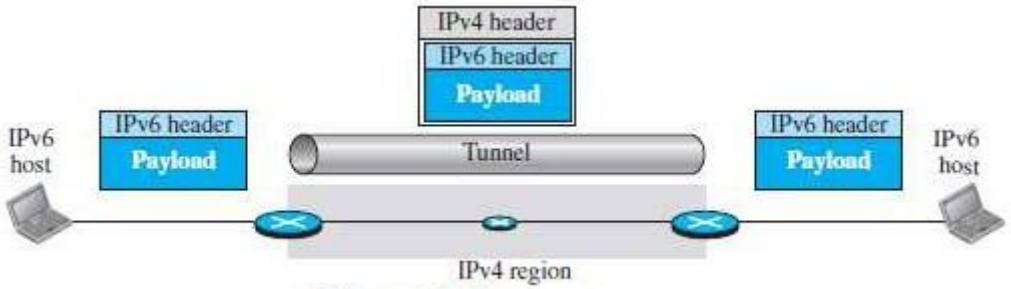


Figure 22.12 *Tunneling strategy*

3) Header Translation

• Header translation is necessary when the majority of the Internet has moved to IPv6 but some systems still use IPv4 (Figure 22.13).
• The sender wants to use IPv6, but the receiver does not understand IPv6.
• Tunneling does not work in this situation because
→ the packet must be in the IPv4 format to be understood by the receiver.
• In this case, the header format must be totally changed through header translation.
• The header of the IPv6 packet is converted to an IPv4 header/

Figure 22.13 *Header translation strategy*

MODULE-WISE QUESTIONS

MODULE 3: NETWORK LAYER PROTOCOLS

1. Explain various field of IPv4. (8*)
2. Explain fragmentation. Explain 3 fields related to fragmentation (6*)
3. Explain options of IPv4. (6*)
4. Explain three network attacks to IP protocol. Also, explain four services of IPSec. (8*)
5. With general format, explain various ICMPv4 messages. (6*)
6. Explain two tools that use ICMP for debugging. (6)
7. Explain the following term with reference to Mobile IP: (4*)
   i) Home address ii) Care-of address iii) Home-agent iv) Foreign-agent
8. Explain three phases for communication in Mobile IP. (8*)

MODULE 5(CONT.): NEXT GENERATION IP

9. Explain 3 address types of IPv6. (6)
10. Explain changes from IPv4 to IPv6. (4*)
11. Explain various field of IPv6. (8*)
12. Explain various extension header of IPv6. (8)
13. Explain various ICMPv6 messages. (6)
14. Explain various group membership messages. (6)
15. Explain 3 ways to make transition from IPv4 to IPv6. (6)

MODEL PAPER-1

1a) List the differences between LAN & WAN. (4 Marks)
1b) Explain the following topologies:
i) Mesh ii) Star (8 Marks)
1c) Explain 4 levels of addressing employed in TCP/IP protocol. (4 Marks)

2a) Explain the theoretical formula which was developed to calculate the data rate. What are the 3 factors on which data rate depends? (10 Marks)
2b) We need to send 265 kbps over a noiseless channel with a bandwidth of 20 kHz. How many signal levels do we need? (2 Marks)
2c) Explain manchester & differential-manchester encoding schemes. Represent the sequence 101011100 using the same encoding schemes. (4 Marks)

3a) Explain non-uniform quantization & how to recover original signal using PCM decoder. (6 Marks)
3b) An analog signal has a bit rate of 8000 bps & a baud rate of 1000 baud. How many data elements are carried by each signal element? How many signal elements do we need? (4 Marks)
3c) Describe ASK, FSK and PSK mechanisms and apply them over the digital data 101101. (6 Marks)

4a) What is multiplexing? Explain the FDM multiplexing and demultiplexing process with neat diagrams. (6 Marks)
4b) Four 1-kbps connections are multiplexed together. A unit is 1 bit. (4 Marks)
Find (i) the duration of 1 bit before multiplexing
(ii) the transmission rate of the link
(iii) the duration of a time slot and
(iv) the duration of a frame.
4c) Explain in detail circuit-switched-network. (6 Marks)

5a) Explain error detection using block coding technique. (10 Marks)
5b) Given the dataword 101001111 and the divisor 10111, show the generation of the CRC codeword at the sender site. (6 Marks)

6a) Differentiate between character oriented and bit oriented format for framing. (4 Marks)
6b) What is PPP? With a neat diagram, explain the frame structure of PPP. Also, explain framing and transition phases in PPP. (12 Marks)

7a) Explain reservation access, polling access & token passing access methods. (12 Marks)
7b) List out 5 goals of fast Ethernet. Explain auto-negotiation. (4 Marks)

8a) Explain MAC sublayer in gigabit-Ethernet (6 Marks)
8b) Explain architecture of IEEE 802.11 (10 Marks)

9a) Explain various components of cellular system with neat diagram. (6 Marks)
9b) Explain the following terms with reference to satellite (4 Marks)
i) Orbits ii) Footprint
9c) Explain various field of IPv4. (6 Marks)

10a) Explain the following term with reference to Mobile IP: (8 Marks)
i) Home address ii) Care-of address iii) Home-agent iv) Foreign-agent
10b) Explain 3 ways to make transition from IPv4 to IPv6. (8 Marks)

MODEL PAPER-2

1a) Explain the following topologies: (8 Marks)
i) Bus ii) Ring
1b) List the 5 layers and its functionality in TCP/IP model. (8 Marks)

2a) Explain 4 performance parameters of network. (8 Marks)
2b) i) A network with bandwidth of 10 Mbps can pass only an average of 12,000 frames per minute with each frame carrying an average of 10,000 bits. What is the throughput of this network? (2 Marks)
ii) Using Shannon theorem, calculate the maximum bit rate of the channel having bandwidth of 3100 Hz & SNR_{db} of 20 db. (2 Marks)
2c) Define the following: (4 Marks)
i) Line coding ii) Internet iii) SNR iv) Decibel

3a) A complex low-pass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal? (2 Marks)
3b) Explain different aspects of digital-to-analog conversion. (8 Marks)
3c) Define ASK. Explain BASK. (6 Marks)

4a) Five channels, each with a 100-kHz bandwidth, are to be multiplexed together. What is the minimum bandwidth of the link if there is a need for a guard band of 10 kHz between the channels to prevent interference? (2 Marks)
4b) Explain in detail synchronous TDM. (8 Marks)
4c) Explain in detail datagram networks. (6 Marks)

5a) Explain checksum with example. Also, write algorithm for Internet Checksum. (12 Marks)
5b) Explain two types of errors. (4 Marks)

6a) Explain HDLC frame format. Also, explain control fields in HDLC. (12 Marks)
6b) Explain the concept of piggybacking. (4 Marks)

7a) Explain CSMA/CA & CSMA/CD. (10 Marks)
7b) Explain frame format of standard Ethernet. (6 Marks)

8a) Explain frame format of IEEE 802.11. (6 Marks)
8b) Explain hidden station problem. (4 Marks)
8c) What is Bluetooth? Explain architecture of Bluetooth. (6 Marks)

9a) What is WiMAX? Explain WiMAX MAC frame format. (6 Marks)
9b) What is cellular telephony? Explain third generation 3G of cellular telephony. (4 Marks)
9c) What is ICMP? With general format, explain various ICMPv4 messages. (6 Marks)

10a) What is Mobile IP? Explain three phases for communication in Mobile IP. (8 Marks)
10b) Explain various field of IPv6. (8 Marks)

MODEL PAPER-3

1a) Define data communications. Explain its 4 fundamental characteristics. (4 Marks)
1b) Explain 3 diff. methods of data flow. Also, explain point to point & multipoint connection. (6 Marks)
1c) Explain in detail LAN. (6 Marks)

2a) What is transmission impairment? Explain causes of transmission impairment. (6 Marks)
2b) i) Consider a noiseless channel with a bandwidth of 3000 Hz transmitting a signal with two signal levels. What is the maximum bit rate? (2 Marks)
ii) The loss in a cable is usually defined in decibels per kilometer (dB/km). If the signal at the beginning of a cable with −0.3 dB/km has a power of 2 mW, what is the power of the signal at 5 km? (2 Marks)
2c) Explain following encoding schemes with example as the sequence 10110011: (8 Marks)
i) Unipolar Scheme ii) Polar Schemes iii) Bipolar Schemes

3a) Explain different types of transmission modes. (8 Marks)
3b) Define FSK. Explain BFSK. (6 Marks)
3c) An analog signal carries 4 bits per signal element. If 1000 signal elements are sent per second, find the bit rate. (2 Marks)

4a) Define and explain the concept of WDM. (4 Marks)
4b) What is spread spectrum? Explain in detail DSSS. (6 Marks)
4c) Explain data transfer phase in Virtual-circuit networks. (6 Marks)

5a) Explain hamming distance for error detection. (6 Marks)
5b) Explain CRC with block diagram & example. (10 Marks)

6a) Explain bit oriented protocol. (6 Marks)
6b) Explain Stop-and-Wait protocol. (10 Marks)

7a) Explain pure ALOHA. A pure ALOHA network transmits 200-bit frames on a shared channel of 200 kbps. What is the requirement to make this frame collision-free? (6 Marks)
7b) Explain TDMA & FDMA. (8 Marks)
7c) Explain addressing in standard Ethernet. (2 Marks)

8a) Explain frame types of IEEE 802.11. (4 Marks)
8b) Explain exposed station problem. (6 Marks)
8c) Explain frame format of Bluetooth. (6 Marks)

9a) What is WiMAX? Explain two types of services of WiMAX. (4 Marks)
9b) Discuss the operation of the cellular telephony. (8 Marks)
9c) Explain fragmentation. Explain 3 fields related to fragmentation (4 Marks)

10a) Explain 3 network attacks to IP protocol. Also, explain four services of IPSec. (8 Marks)
10b) Explain various extension header of IPv6. (8 Marks)

MODEL PAPER-4

1a) Define data communication. Explain different components of data communication system. Explain the 3 criteria necessary for an effective and efficient network. (6 Marks)
1b) Define network. Explain in detail WAN. (6 Marks)
1c) Explain different method of accessing the Internet. (4 Marks)

2a) Explain 2 methods for transmitting a digital signal (8 Marks)
2b) The power of a signal is 10 mW and the power of the noise is 1 μW; what are the values of SNR and SNR_{dB}? (2 Marks)
2c) Explain in detail any 6 characteristics of digital signal. (6 Marks)

3a) Explain the PCM encoder with neat diagram. (10 Marks)
3b) Define PSK. Explain BPSK. (6 Marks)

4a) Explain in detail Statistical TDM. (6 Marks)
4b) Explain in detail FHSS. (6 Marks)
4c) Compare circuit-switched-network, datagram & virtual-circuit. (4 Marks)

5a) Write short notes on polynomial codes. (6 Marks)
5b) Explain parity-check code with block diagram. (10 Marks)

6a) Explain character oriented protocol. (8 Marks)
6b) Explain 3 type of frames used in HDLC. (8 Marks)

7a) Explain slotted ALOHA & CSMA. (10 Marks)
7b) Explain briefly any 2 implementation of standard Ethernet. (6 Marks)

8a) Explain addressing in IEEE 802.11. (8 Marks)
8b) Explain layers of Bluetooth. (8 Marks)

9a) Explain the 3 categories of satellites. (10 Marks)
9b) Explain fourth generation 4G of cellular telephony. (6 Marks)

10a) Explain changes from IPv4 to IPv6. (4 Marks)
10b) Explain 3 address types of IPv6. (4 Marks)
10c) Explain various ICMPv6 messages. (6 Marks)