COMPUTER NETWORKS

20CS5PCCON

UNIT I

Data Communication

- Communication
  - Sharing of information
  - local (face to face) or remote (telecommunication:tele-far)
  - Telecommunication- telephony, telegraphy, television
  - Data – communicated information
  - Data communications are the exchange of data between two devices via some form of transmission medium such as a wire cable.

Data Communication

The effectiveness of a data communications system depends on four fundamental characteristics: delivery, accuracy, timeliness, and jitter.

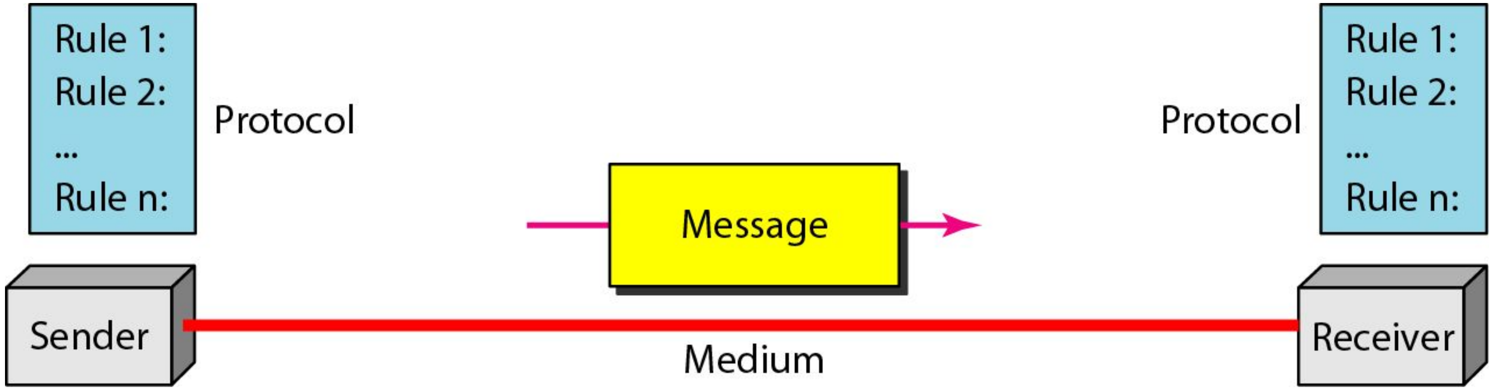
1. Delivery.
   - Data to be delivered to correct destination and correct user
2. Accuracy.
   - Data is to be delivered accurately.
   - If altered in transmission and left uncorrected become unusable.
   - Ex. Text transfers- Email, numerical transfers say amount transferred from one account to another
3. Timeliness.
   - Data need to be delivered in a timely manner.
   - Real-time transmission – Data becomes useless when it arrives late Ex. Audio and video
4. Jitter.
   - Jitter refers to the variation in the packet arrival time.
   - An uneven delay is experienced in the delivery of audio or video packets. Some may arrive in say 10 ms and some in 20 ms.

Data Communication-Components

- A data communications system has five components

1. Message:
   - the information (data) to be communicated.
   - Forms: include text, numbers, pictures, audio, and video.
2. Sender:
   - The device that sends the data message.
   - Ex. computer, workstation, telephone handset, video camera, and so on.
3. Receiver:
   - The device that receives the message.
   - Ex., computer, workstation, telephone handset, television, and so on.
4. Transmission medium:
   - the physical path by which a message travels from sender to receiver.
   - Ex. twisted-pair wire, coaxial cable, fiber-optic cable, and radio waves.
5. Protocol:
   - A protocol is a set of rules that govern data communications.
   - It represents an agreement between the communicating devices.
   - Without a protocol, two devices may be connected but not communicating

Figure 1.1 *Components of a data communication system*



Data Communication-Data Representation

- Different forms of data: text, numbers, images, audio, and video.
- Text :
  - represented as a bit pattern, a sequence of bits (0s or 1s).
  - Different sets of bit patterns to represent text symbols.
    - Code- Each set Ex. ASCII – 7 bits, Unicode – 32 bits
    - Coding- the process of representing symbols.
- Images
  - represented by bit patterns.
  - an image is composed of a matrix of pixels (picture elements), where each pixel is a small dot.
  - The size of the pixel depends on the resolution.
  - Ex., an image can be divided into 1000 pixels (low resolution) or 10,000 pixels (high resolution-needs more memory)

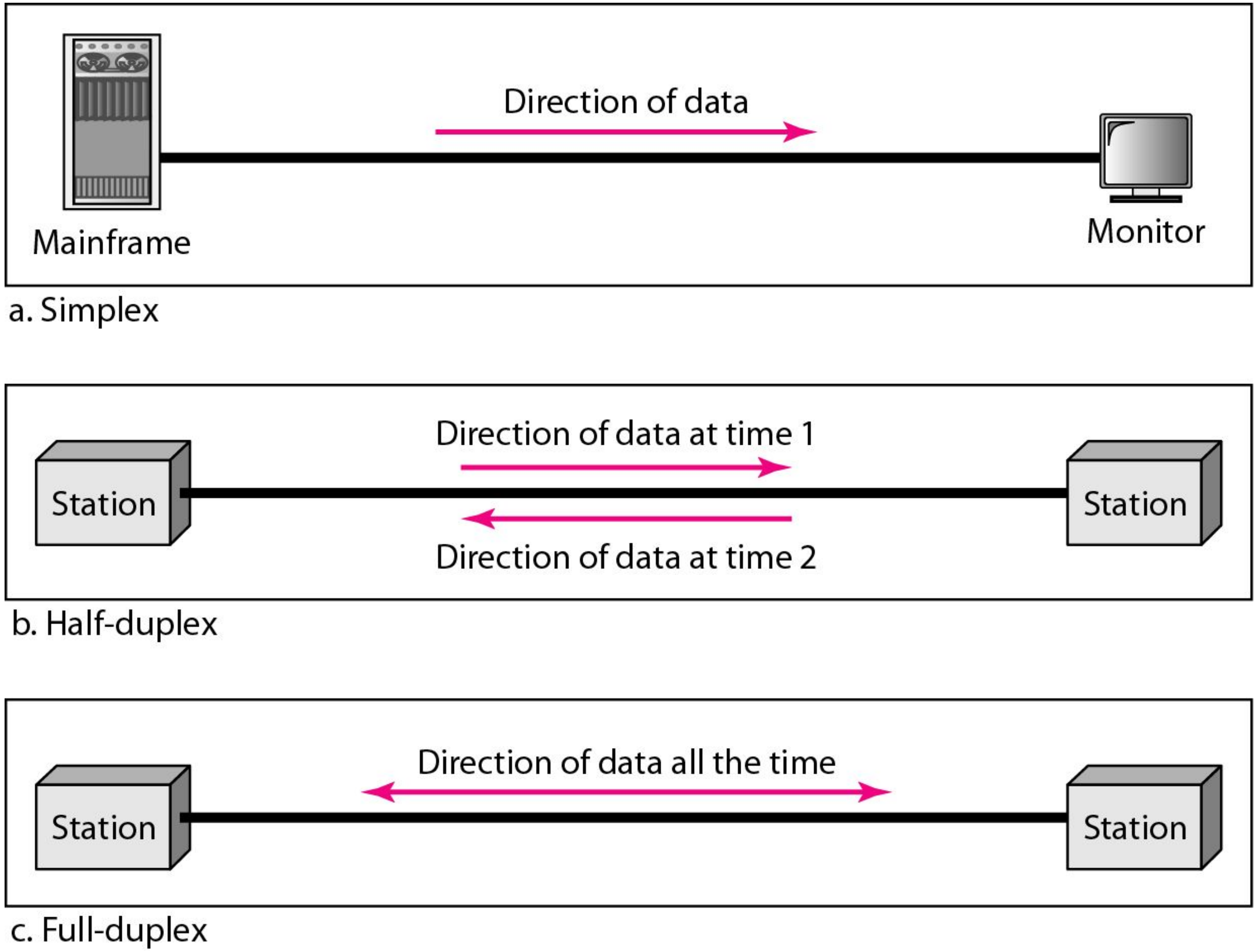
Data Communication-Data Representation

- Images
  - Each pixel is assigned a bit pattern. The size and the value of the pattern depend on the image.
  - Ex. black and-white dots (e.g., a chessboard), a 1-bit pattern is enough to represent a pixel.
  - Gray scale- 2-bit patterns-00 : black pixel, 01- a dark gray pixel, 10- a light gray pixel, 11- white pixel.
  - Color images.
    - RGB - combination of three primary colors: red, green, and blue.
    - The intensity of each color is measured, and a bit pattern is assigned to it.
    - YCM- combination of three other primary colors: yellow, cyan, and magenta
- Audio:
  - Audio refers to the recording or broadcasting of sound or music.
  - Diff: from text, numbers, or images. It is continuous, not discrete.
  - Ex, a microphone to change voice or music to an electric signal, we create a continuous signal.
- Video:
  - the recording or broadcasting of a picture or movie.
  - Video can either be produced as a continuous entity (e.g., by a TV camera), or it can be a combination of images, each a discrete entity, arranged to convey the idea of motion.

Data Communication- Data Flow

- Data Flow Communication between two devices
  - can be simplex, half-duplex, or full-duplex
  - Simplex:
    - the communication is unidirectional, as on a one-way street.
    - Only one of the two devices on a link can transmit; the other can only receive
    - Ex. Keyboards - only introduce input
    - Traditional monitors- only accept output.
    - uses the entire capacity of the channel to send data in one direction.
  - Half-Duplex:
    - each station can both transmit and receive, but not at the same time.
    - When one device is sending, the other can only receive, and vice versa
    - like a one-lane road with traffic allowed in both directions. When cars are traveling in one direction, cars going the other way must wait

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



Data Communication- Data Flow

- Half-Duplex:
  - the entire capacity of a channel is taken over by one of the two devices that is transmitting
  - Ex. Walkie-talkies
  - Used when there is no need for communication in both directions at the same time
  - the entire capacity of the channel can be utilized for each direction.
- Full-Duplex:
  - In full-duplex mode (also called duplex), both stations can transmit and receive
  - like a two-way street with traffic flowing in both directions at the same time.
  - Signals going in one direction share the capacity of the link with signals going in the other direction.
  - This sharing can occur in two ways:
    - Either the link must contain two physically separate transmission paths, one for sending and the other for receiving; or
    - the capacity of the channel is divided between signals traveling in both directions.
    - Ex. One common example of full-duplex communication is the telephone network. When two people are communicating by a telephone line, both can talk and listen at the same time.
    - Used when communication in both directions is required all the time. The capacity of the channel must be divided between the two directions.

1.2 NETWORKS

- A network
  - is the interconnection of a set of devices capable of communication.
  - a device can be a
    - host (end system)
      - large computer, desktop, laptop, workstation, cellular phone, or security system.
    - connecting device
    - router
      - connects the network to other networks
    - switch
      - connects devices together
    - modem (modulator-demodulator), which changes the form of data, and so on.

  - devices are connected using wired or wireless transmission media such as cable or air.

1.2.1 Network Criteria

- The most important of these are performance, reliability, and security.
- Performance
- Performance can be measured in many ways including transit time and response time.
    - Transit time
        - amount of time required for a message to travel from one device to another.
    - Response time
        - elapsed time between an inquiry and a response.
- The performance of a network depends on the
    - number of users
    - type of transmission medium
    - capabilities of the connected hardware
    - the efficiency of the software
- Performance is often evaluated by two networking metrics: throughput and delay.
- We often need more throughput and less delay.
- In practice, this is very challenging as more data is sent into the network, delay increases.

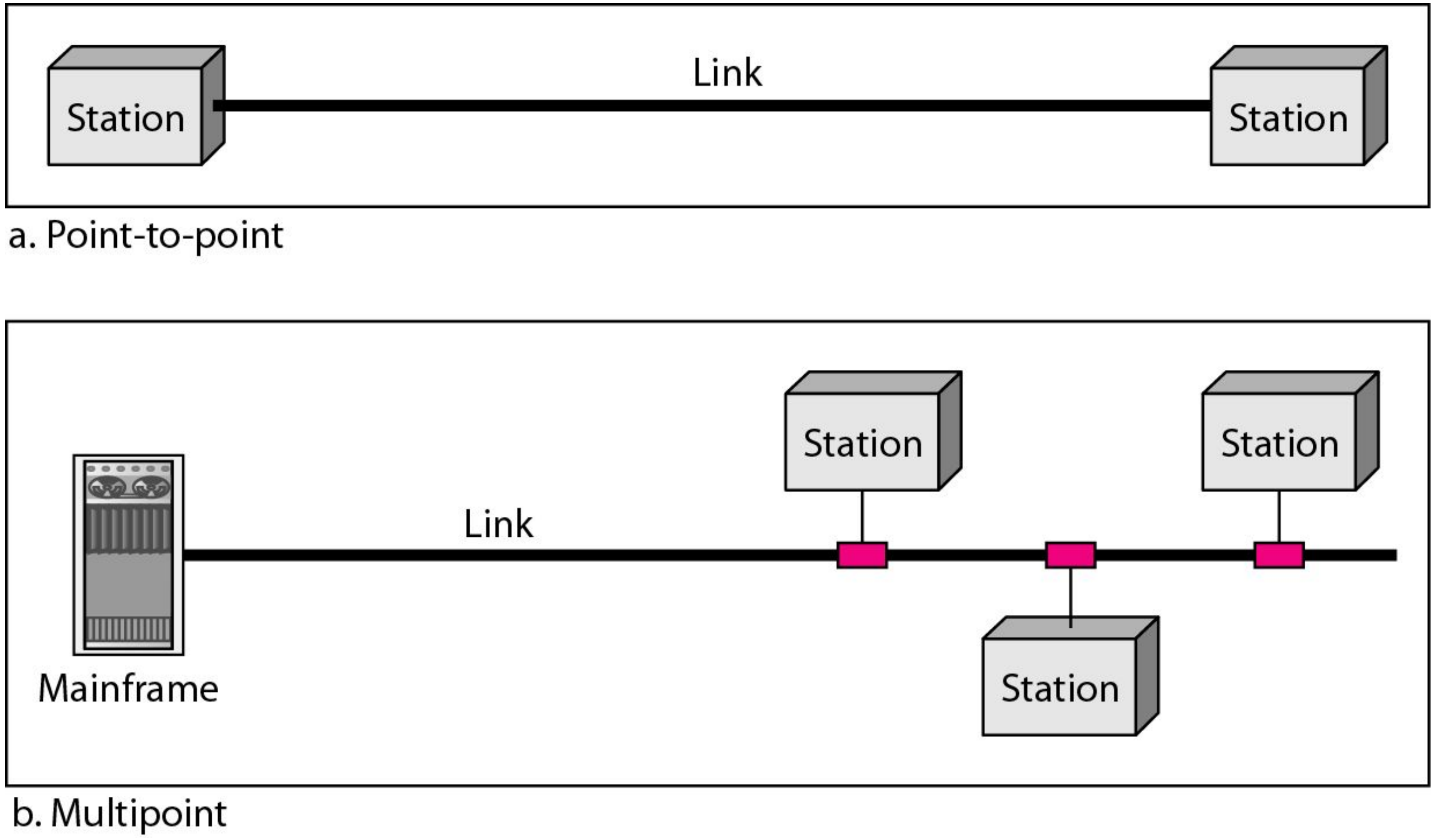
1.2.1 Network Criteria

- Reliability
  - Network reliability is measured by
    - accuracy of delivery
    - the frequency of failure
    - the time it takes a link to recover from a failure
    - network's robustness in a disaster/calamity
- Security
  - Network security issues include protecting data from
    - unauthorized access
    - Protecting data from damage and development
    - implementing policies and procedures for recovery from breaches and data losses.

1.2.2 Physical Structures

- **Type of Connection**
- Two or more devices are connected through links.
- A link is a communications pathway that transfers data from one device to another.
- For communication to occur, two devices must be connected in some way to the same link at the same time.
- There are two possible types of connections: point-to-point and multipoint.
- Point-to-Point
  - provides a dedicated link between two devices.
  - The entire capacity of the link is reserved for transmission between those two devices.
  - use an actual length of wire or cable to connect the two ends, but other options, such as microwave or satellite links
  - changing television channels by infrared remote control -a point-to-point connection is established
- Multipoint
  - A multipoint (also called multidrop) connection is one in which more than two specific devices share a single link
  - the capacity of the channel is shared, either spatially or temporally.
  - If the link is to be used simultaneously, it is a spatially shared connection.
  - If users must take turns, it is a timeshared connection.

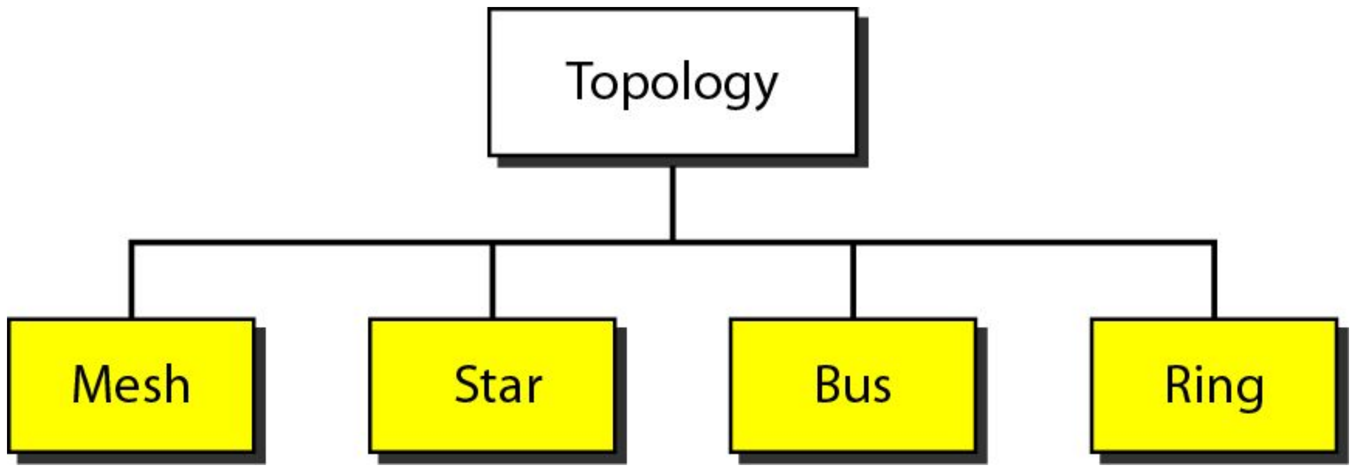
Figure 1.3 *Types of connections: point-to-point and multipoint*



Physical Topology

- Physical topology
  - the way in which a network is laid out physically.
  - Two or more devices connect to a link
  - two or more links form a topology.
  - Topology
    - the geometric representation of the relationship of all the links and linking devices.
    - There are four basic topologies - mesh, star, bus, and ring.

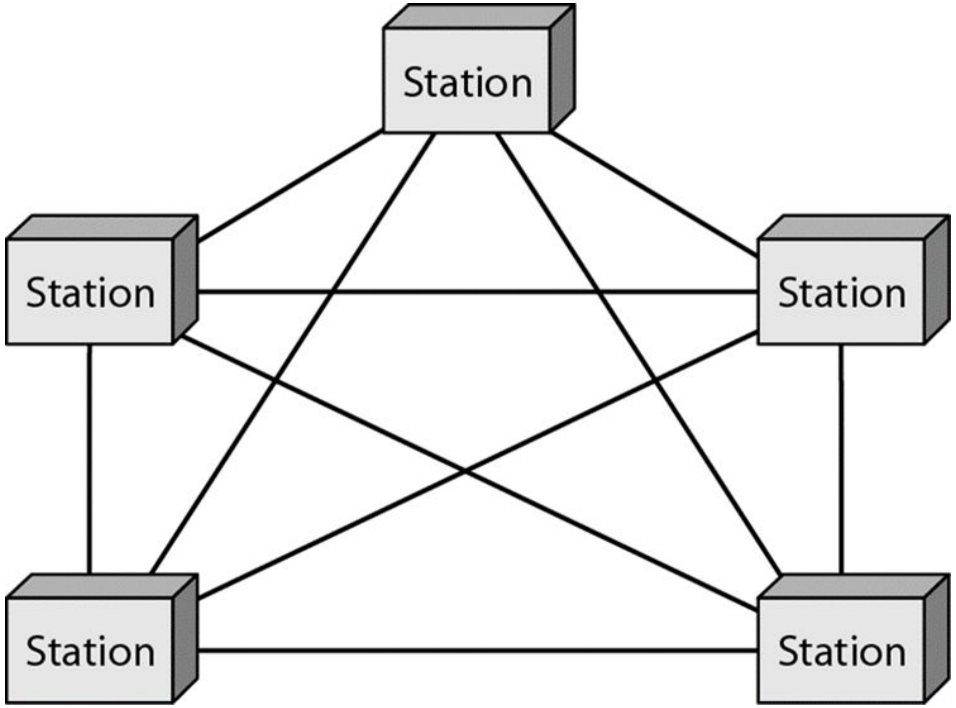
Categories of topology



Physical Topology - Mesh

Mesh:

- Every device has a dedicated point-to-point link to every other device
- We need n (n – 1) physical links.
- If every link allows duplex communication – n(n-1)/2 links are seen
- every device must have n – 1 input/output (I/O) ports



Advantages:

- dedicated links guarantees that each connection can carry its own data load, thus eliminating the traffic problems
- mesh topology is robust – failure of one link does not affect others
- Privacy or security is achieved- dedicated lines
- fault identification and fault isolation easy

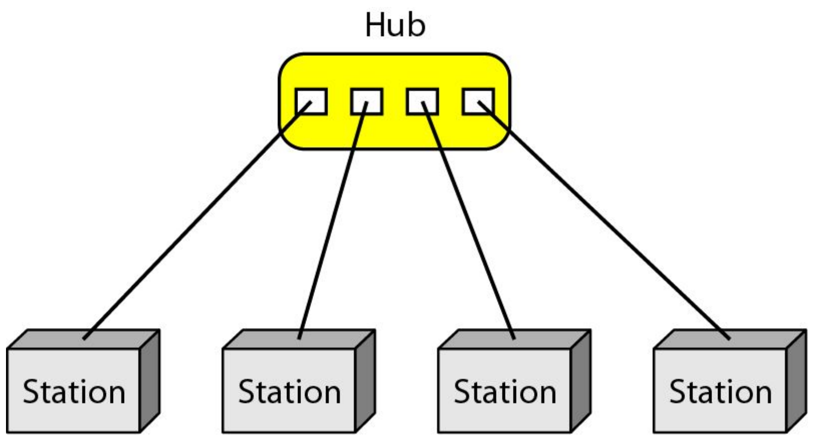
Physical Topology - Mesh

- **Disadvantages:**
  - amount of cabling and the number of I/O ports required.
  - installation and reconnection are difficult.
  - bulk of the wiring can be greater than the available space
  - hardware required to connect each link (I/O ports and cable) expensive.
  - Hence, implemented in a limited fashion, for example, as a backbone connecting the main computers of a hybrid network that can include several other topologies.
  - One practical example - the connection of telephone regional offices in which each regional office needs to be connected to every other regional office

Physical Topology - Star

Star Topology:

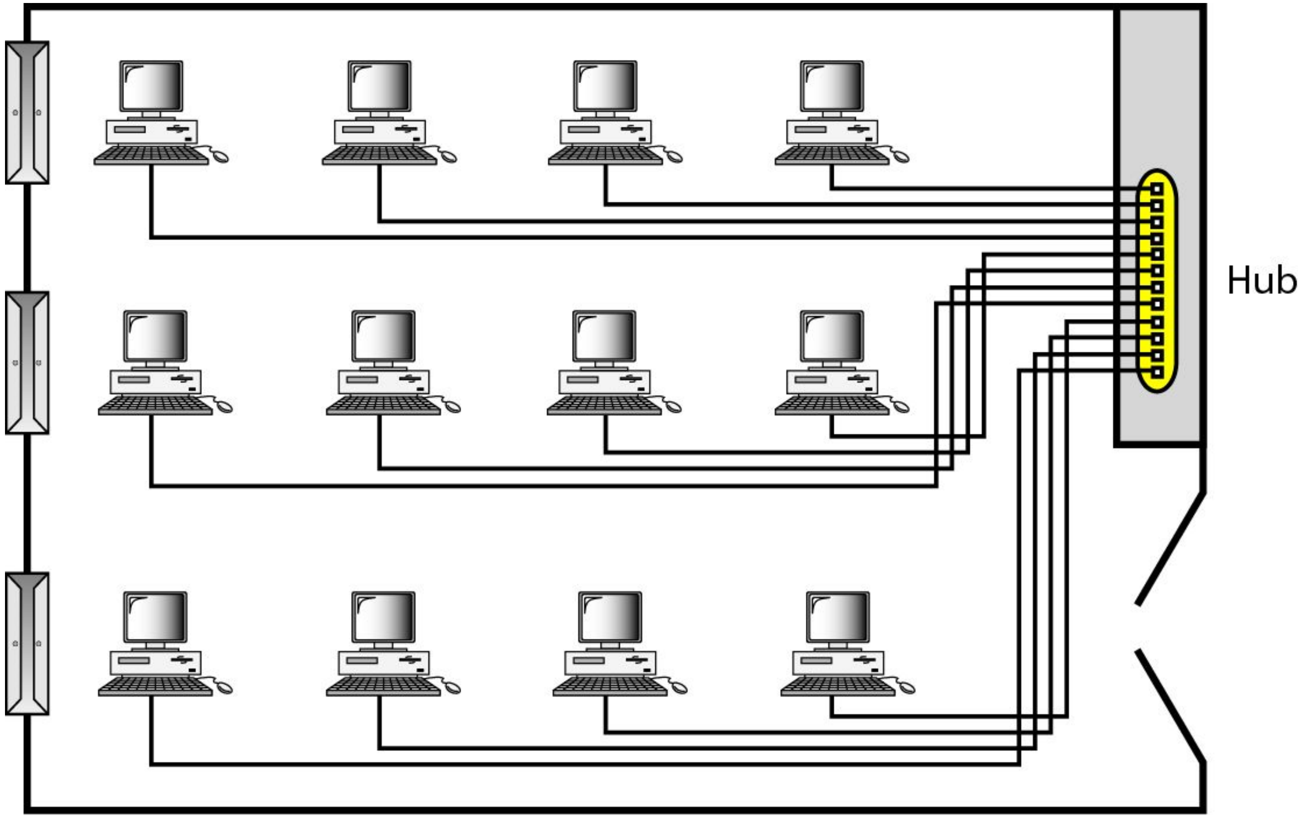
- Each device has a dedicated point-to-point link only to a central controller, called a hub.
- The devices are not directly linked to one another –no direct traffic between devices.
- Controller acts as an exchange:
  - If Device1 wants to communicate to Device2, D1-> controller->D2
- Advantage:
  - less expensive than a mesh topology
    - each device needs only one link and one I/O port to connect it to any number of others (less cabling)
  - easy to install and reconfigure.
  - Additions and deletions involve only one connection: between that device and the hub.



Physical Topology - Star

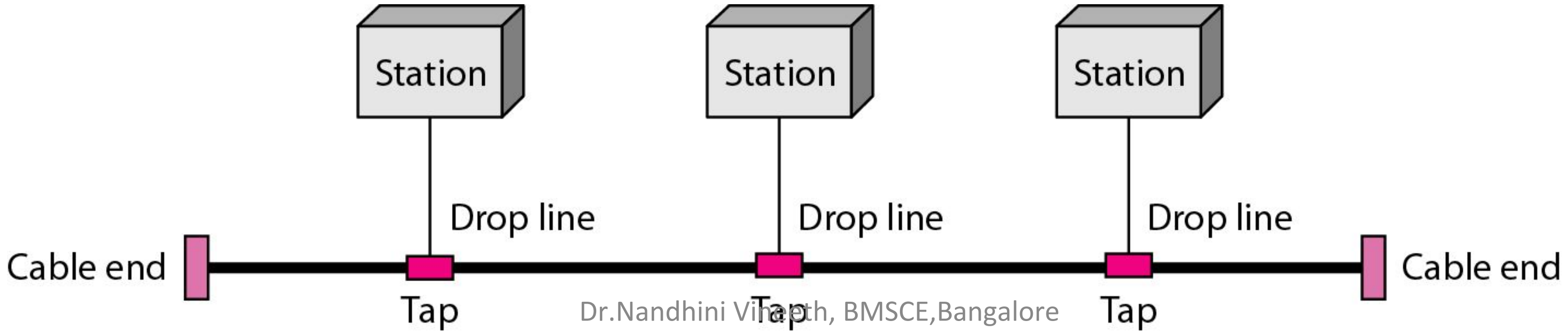
- Robustness
  - if one link fails, only that link is affected.
  - All other links remain active, easy fault identification and fault isolation.
  - When hub is working, it can be used to monitor link problems and bypass defective links.
- Disadvantage :
  - Dependency on one single point, the hub.
  - If the hub goes down, the whole system is dead.
  - Cabling is better than mesh but inefficient compared to ring or bus
  - The star topology is used in local-area networks (LANs)
  - High-speed LANs often use a star topology with a central hub.

Figure 1.10 *An isolated LAN connecting 12 computers to a hub in a closet*



Physical Topology - Bus

- Star and mesh are point to point
- Bus is multipoint
- One long cable acts as a backbone linking all devices in a network
- Nodes are connected to the bus cable by drop lines and taps.
- A drop line is a connection running between the device and the main cable
- A tap is a connector that either splices into the main cable creating a contact with the metallic core
- Limits on the number of taps a bus can support and on the distance between those taps



Physical Topology - Bus

Advantages:

- ease of installation.
- Backbone cable can be laid along the most efficient path and nodes connect to this
- less cabling than mesh or star topologies.
- Used in the design of early local area networks. Traditional Ethernet LANs can use a bus topology

Disadvantages

- include difficult reconnection and fault isolation.
- optimally efficient at installation.
- difficult to add new devices.
- Signal reflection at the taps can cause degradation in quality

can be controlled by limiting the number and spacing of devices connected to a given length of cable.

Adding new devices may therefore require modification or replacement of the backbone.

A fault or break in the bus cable stops all transmission, even between devices on the same side of the problem.

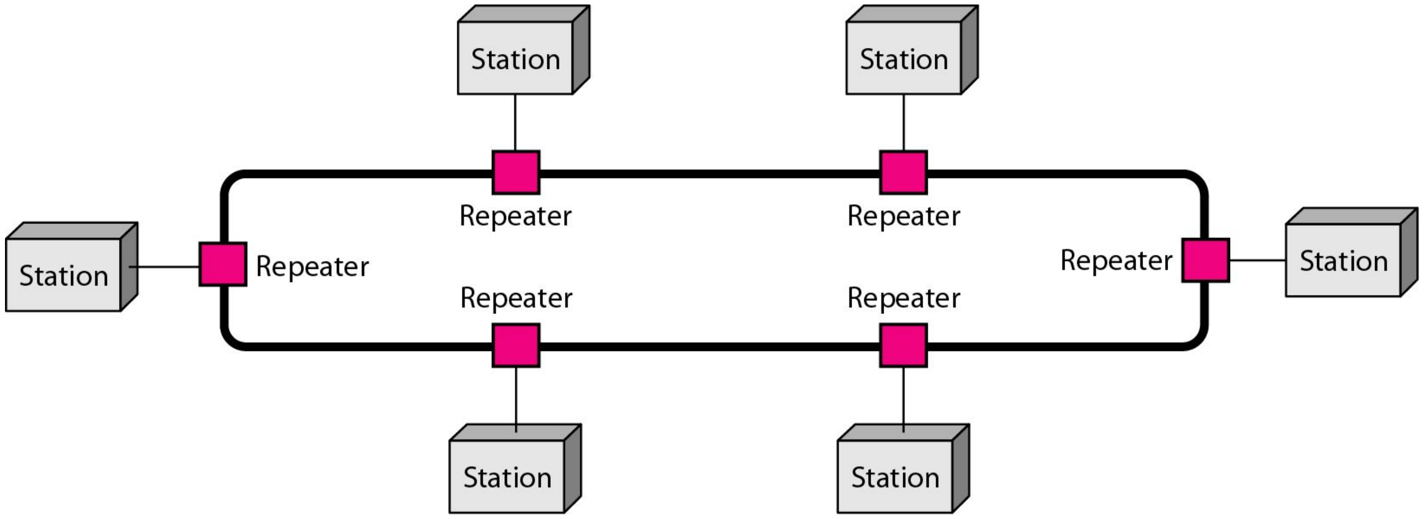
The damaged area reflects signals back in the direction of origin, creating noise in both directions.

Physical Topology - Ring

Each device has a dedicated point-to-point connection with only the two devices on either side of it.

A signal is passed along the ring in one direction, from device to device, until it reaches its destination.

Each device in the ring incorporates a repeater- regenerates the received bits and passes them along



Physical Topology - Ring

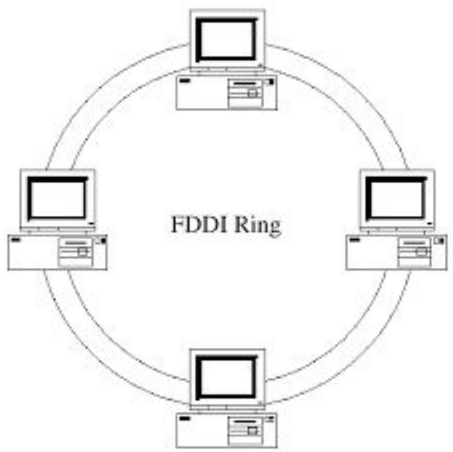
Advantages:

A ring is relatively easy to install and reconfigure.

Each device is linked to only its immediate neighbors (either physically or logically).

To add or delete a device requires changing only two connections.

The only constraints maximum ring length and number of devices.



Disadvantages:

fault isolation is simplified - in a ring a signal is circulating at all times.

Devices raise an alarm to alert the network operator if they does not receive a signal within a specified period

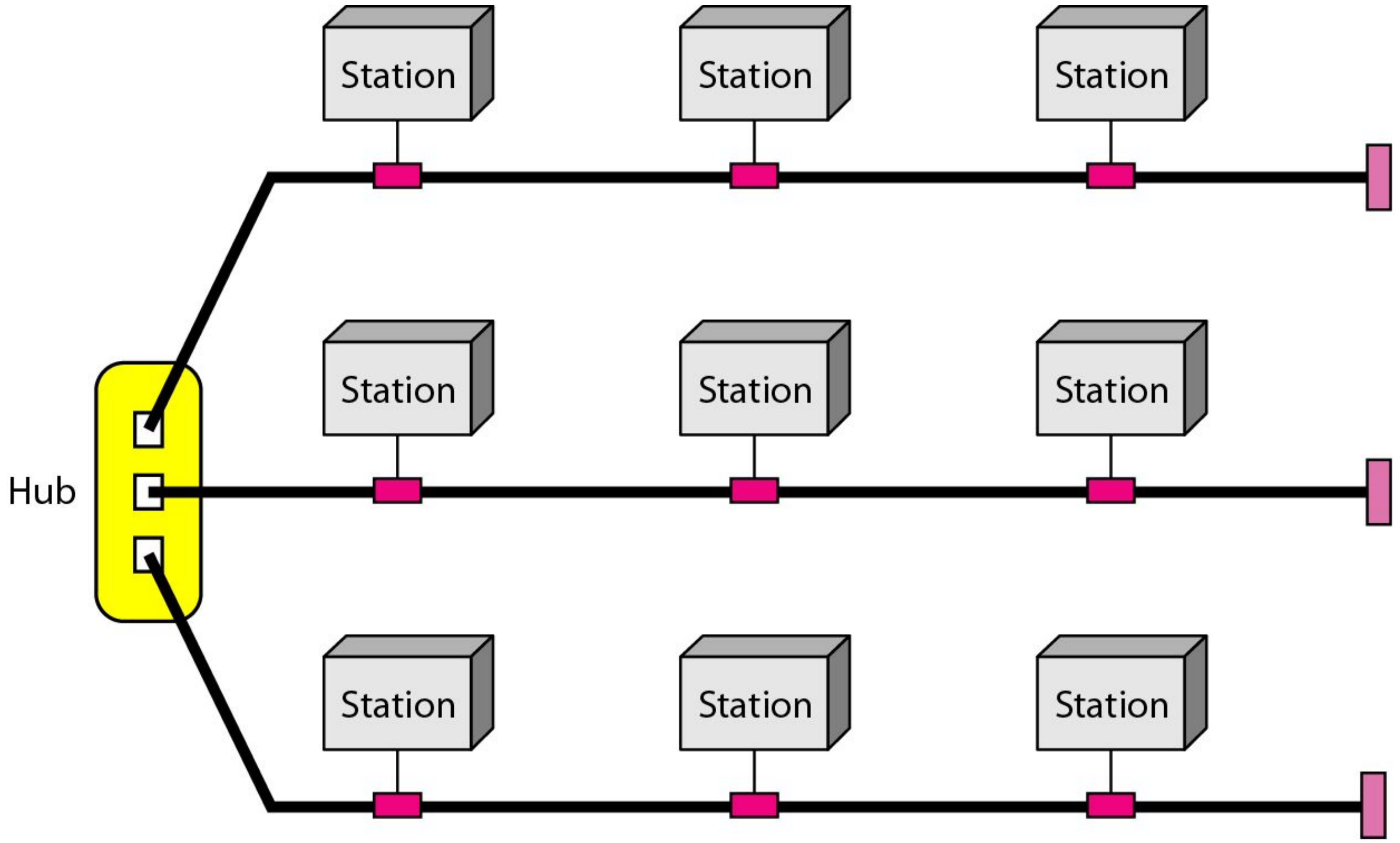
unidirectional traffic can be a disadvantage.

a break in the ring (such as a disabled station) can disable the entire network.

Solution: a dual ring or a switch capable of closing off the break.

Ring topology - IBM introduced its local-area network **Token Ring**

Figure 1.9 *A hybrid topology: a star backbone with three bus networks*



NETWORK TYPES

Based on size, geographical coverage, and ownership, Networks can be divided into

Local Area Network and Wide Area Network

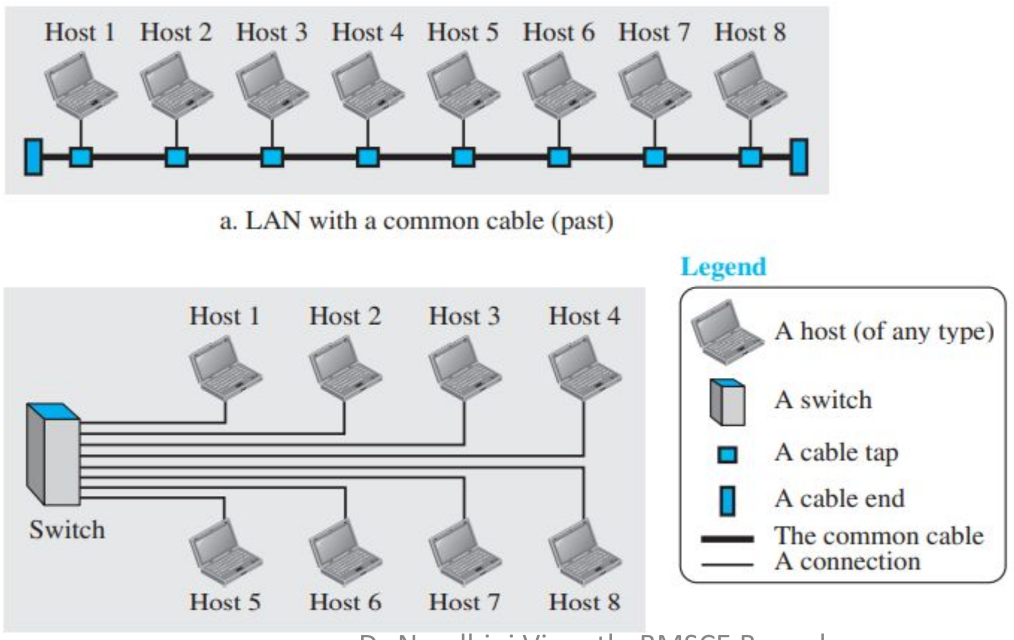
Local Area Network:

- Usually privately owned and connects some hosts in a single office, building, or campus.
- can be as simple as two PCs and a printer in someone's home office
- can extend throughout a company and include audio and video devices.
- Resources are shared
- an identifier, an address uniquely defines the host in the LAN
- Every packet carries both the source host's and the destination host's addresses.
- Earlier, a common cable connected all hosts, a packet sent from one host to another was received by all hosts. The intended recipient kept the packet; the others drop

NETWORK TYPES

- a smart connecting switch, recognizes the destination address of the packet and guide the packet to its destination without sending it to all other hosts.
- The switch allows more than one pair to communicate with each other at the same time if there is no common source and destination among them.

1.8 *An isolated LAN in the past and today*



b. LAN with a switch (today)

NETWORK TYPES

Wide Area Network

- has a wider geographical span, spanning a town, a state, a country, or even the world.

- A LAN interconnects hosts;
- a WAN interconnects connecting devices such as switches, routers, or modems.

- A LAN is normally privately owned by the organization that uses it;
- a WAN is normally created and run by communication companies and leased by an organization that uses it.

- Two distinct examples of WANs today: point-to-point WANs and switched WANs.

Point-to-Point WAN:

Network that connects two communicating devices through a transmission media (cable or air).

Switched WAN:

- A switched WAN is a network with more than two ends.
- used in the backbone of global communication today
- a combination of several point-to-point WANs that are connected by switches.

Figure 1.9 *A point-to-point WAN*

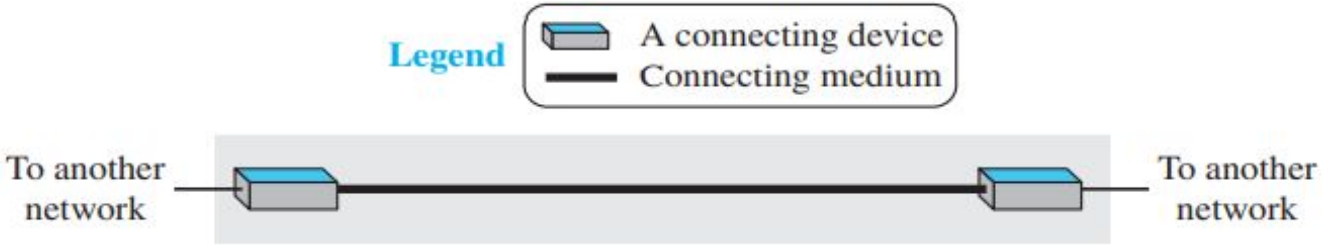
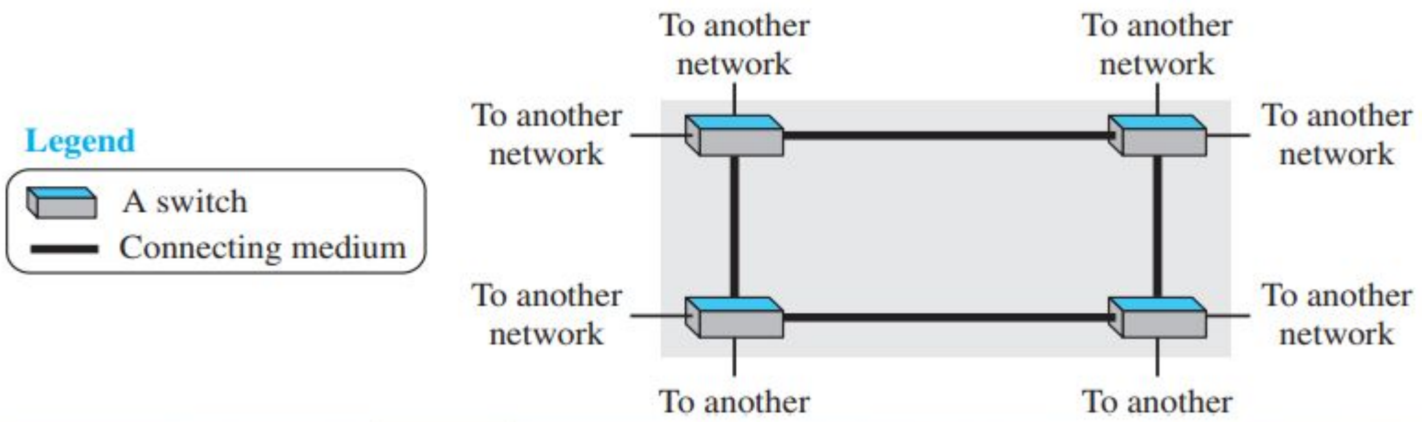


Figure 1.10 *A switched WAN*



Internetwork

- very rare to see a LAN or a WAN in isolation
- they are connected to one another.
- When two or more networks are connected, they make an internetwork, or internet.
- As an example, assume that an organization has two offices, one on the east coast and the other on the west coast.
- Each office has a LAN that allows all employees in the office to communicate with each other.

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

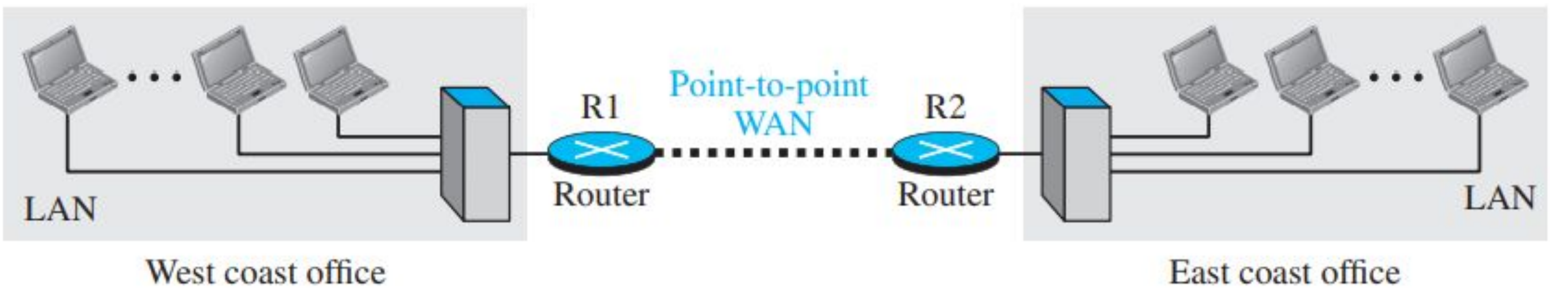
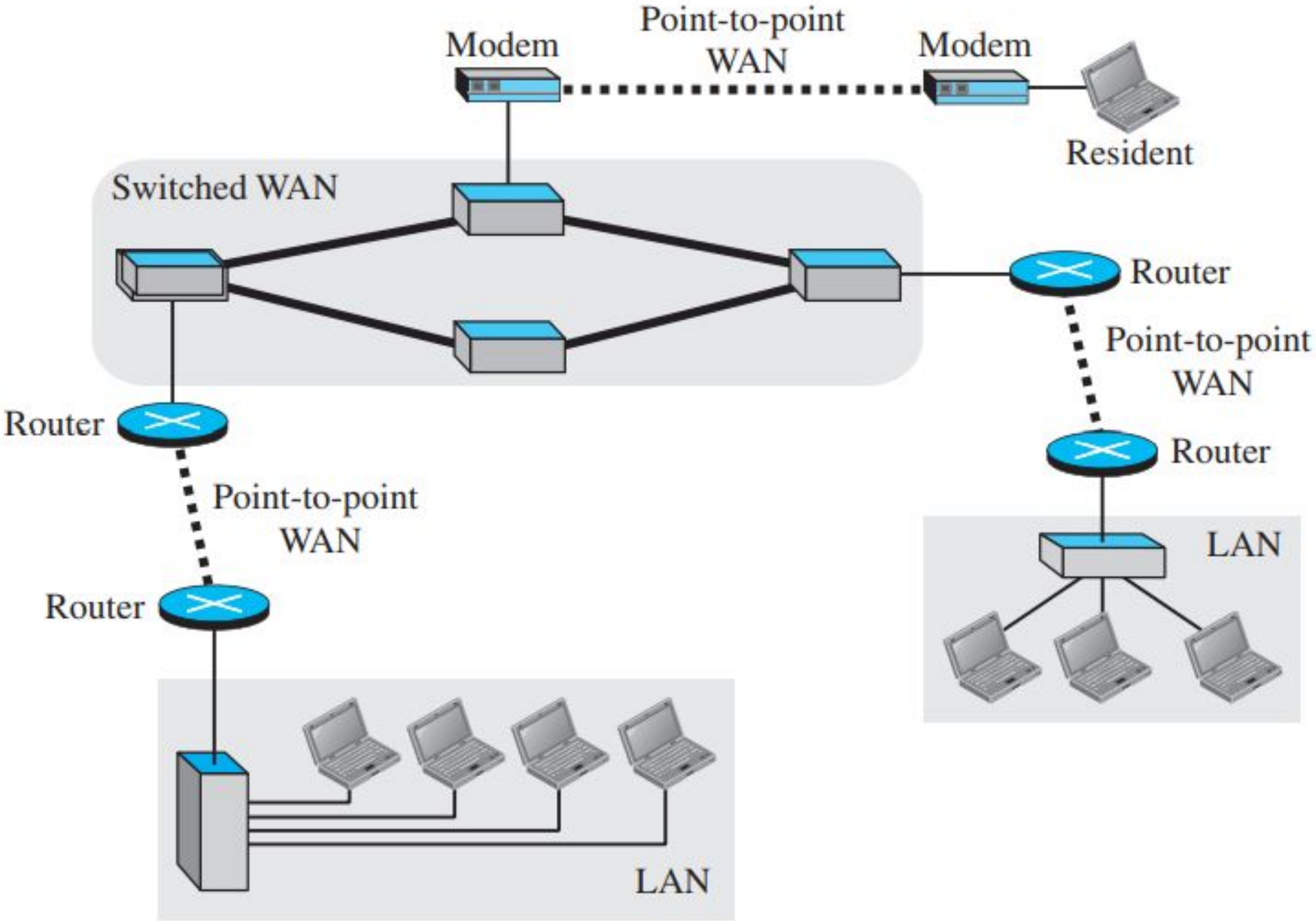


Figure 1.12 *A heterogeneous network made of four WANs and three LANs*



SWITCHING

A switch

connects at least two links together

needs to forward data from a network to another network when required.

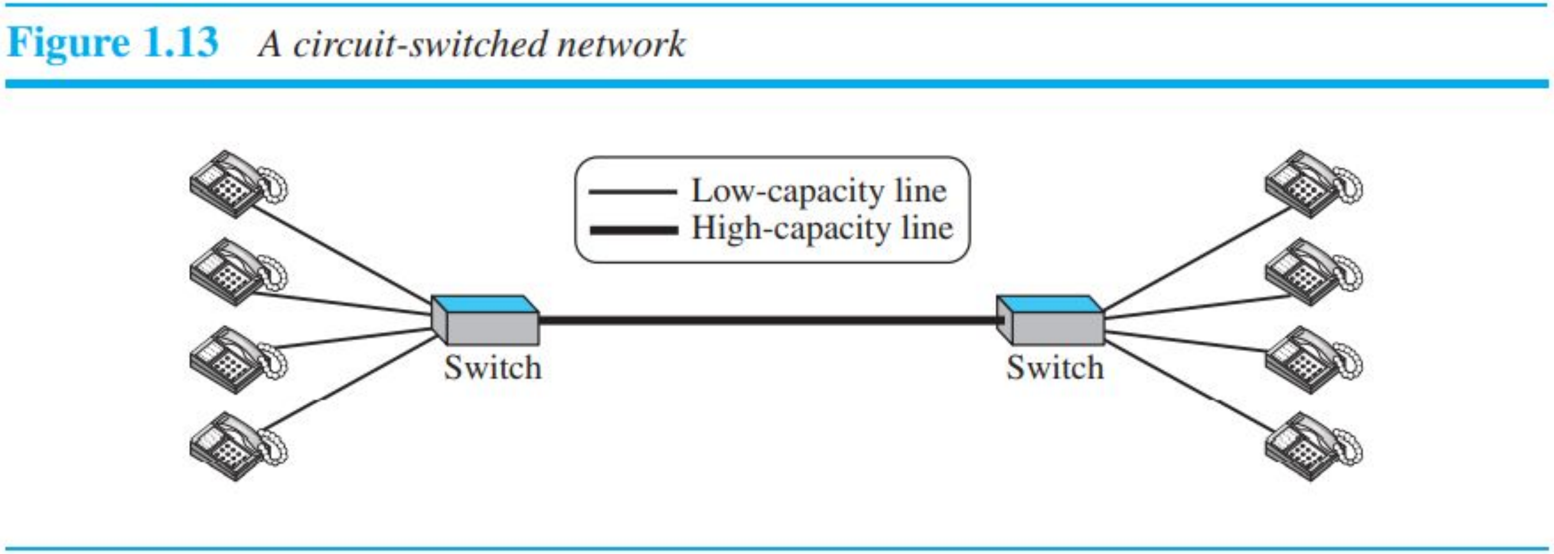
Two most common types of switched networks **- circuit-switched and packet-switched networks**.

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 - connects four telephones to each end.
- Circuit switching was very common in telephone networks in the past
- Explanation of the figure.
- thick line - is a high-capacity communication line that can handle four voice communications at the same time
- the capacity can be shared between all pairs of telephone sets.
- forwarding tasks but no storing capability.
- Two cases.
  - All four to four communication- the capacity of the thick line is fully used.
  - Only one to one - only one-fourth of the capacity of the thick line is used.
- is efficient only when it is working at its full capacity;
- most of the time, it is inefficient because it is working at partial capacity.

- If link capacity is reduced, then communication cannot happen for all

Figure 1.13 *A circuit switched Network*

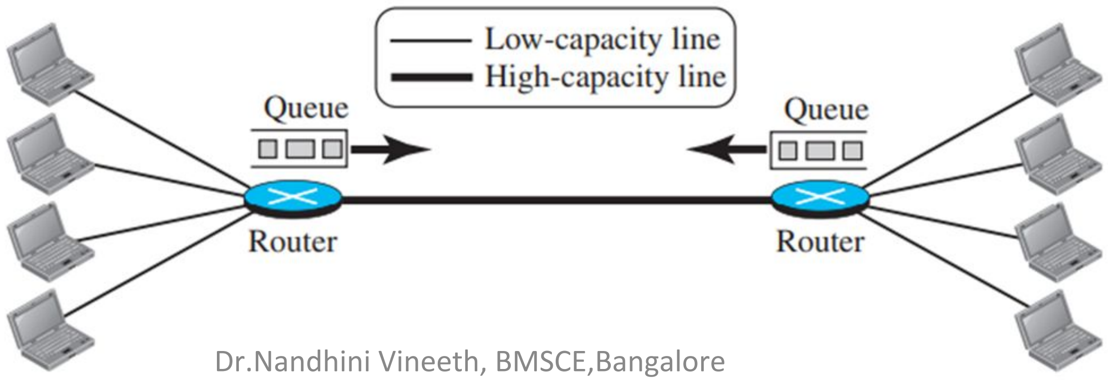


SWITCHING

Packet-Switched Network

- Computer network- the communication between the two ends is done in blocks of data called packets.
- the exchange of individual data packets between the two computers.
- switches function - both storing and forwarding because a packet is an independent entity that can be stored and sent later.
- Figure explanation
- A router in a packet-switched network has a queue that can store and forward the packet.
- When the requirement is less than or equal to , no waiting experience
- When the requirement is more than the capacity of the link, packets should be stored and forwarded
- packet-switched network is more efficient than a circuit switched network, but the packets may encounter some delays.

Figure 1.14 *A packet-switched network*



The Internet

Two or more networks that can communicate with each other

- **Top level**

    the backbones are large networks **owned by some communication companies** such as Sprint, Verizon (MCI), AT&T, and NTT.

    The backbone networks are connected through some complex switching systems -**peering points**.

- Second level

    provider networks

    smaller networks that use the services of the backbones for a fee.

    connected to backbones and sometimes to other provider networks.

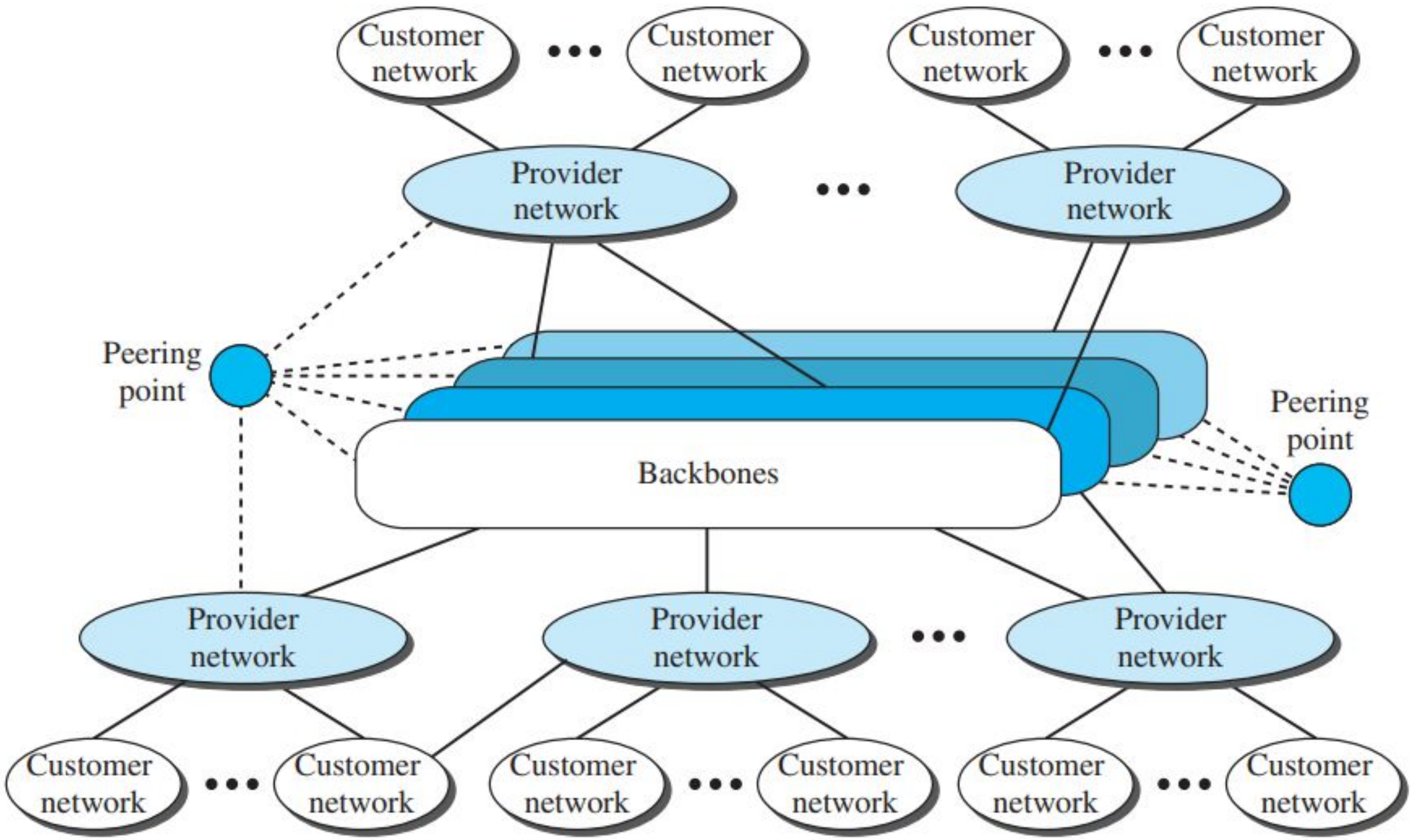
- Third level
- **customer networks** – Service users by paying a fee
- Internet Service Providers (ISPs)-

    Backbones and provider networks

    International ISPs- backbones

    National or regional ISPs - provider networks

Figure 1.15 *The Internet today*



The Internet

Accessing the Internet

The physical connection is normally done through a point-to-point WAN

Using Telephone Networks

Dial Up Service

modem added to telephone line

DSL Service

For high speed internet

Using Cable Networks

Using Wireless Networks

Direct Connection to the Internet

Large organization or corporation can become a local ISP

leases a high-speed WAN from a carrier provider and connects itself to a regional ISP

a large university with several campuses can create an internetwork and then connect the internetwork to the Internet

Network Models - Layering

- Protocol
  - rules that both the sender and receiver and all intermediate devices need to follow to be able to communicate effectively
  - Face to face communication
    - EX1: Two friends communicating
    - EX2: A Lecture hall

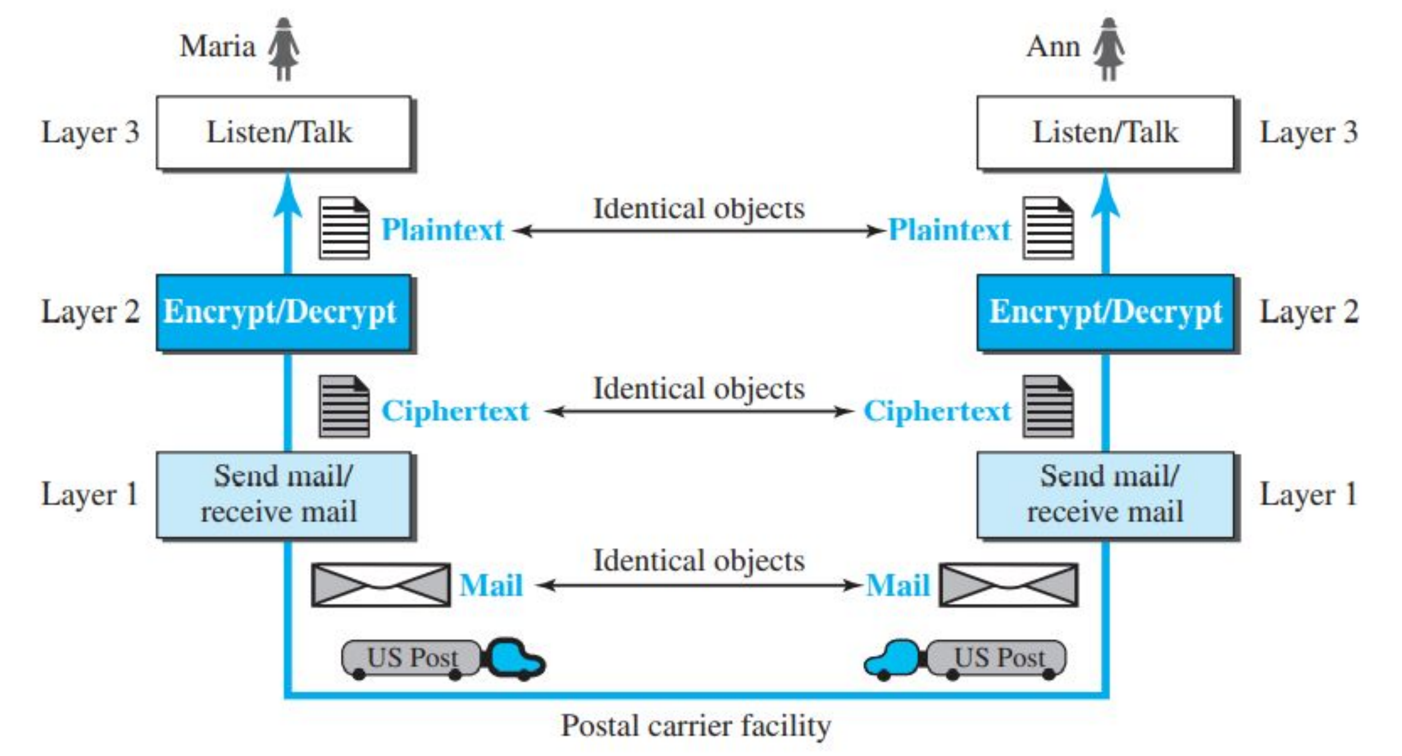
Figure 2.1 *A single-layer protocol*

Network Models - Layering

- Modularity
- A layer (module)
  - a black box with inputs and outputs, without concern about how inputs are changed to outputs.
- Advantages:
  - allows us to separate the services from the implementation. (DIFF VENDORS HAVE DIFF IMPLEMENTATION)
  - Every layer- Service user and service provider
    - A layer needs to be able to receive a set of services from the lower layer and to give the services to the upper layer
  - Method of layer implementation need not be known.
  - Communication involves intermediate systems.
  - Interm system involves only some layers, but not all layers.
  - Otherwise the whole system becomes more expensive.

Network Models - Layering

Figure 2.2 *A three-layer protocol*



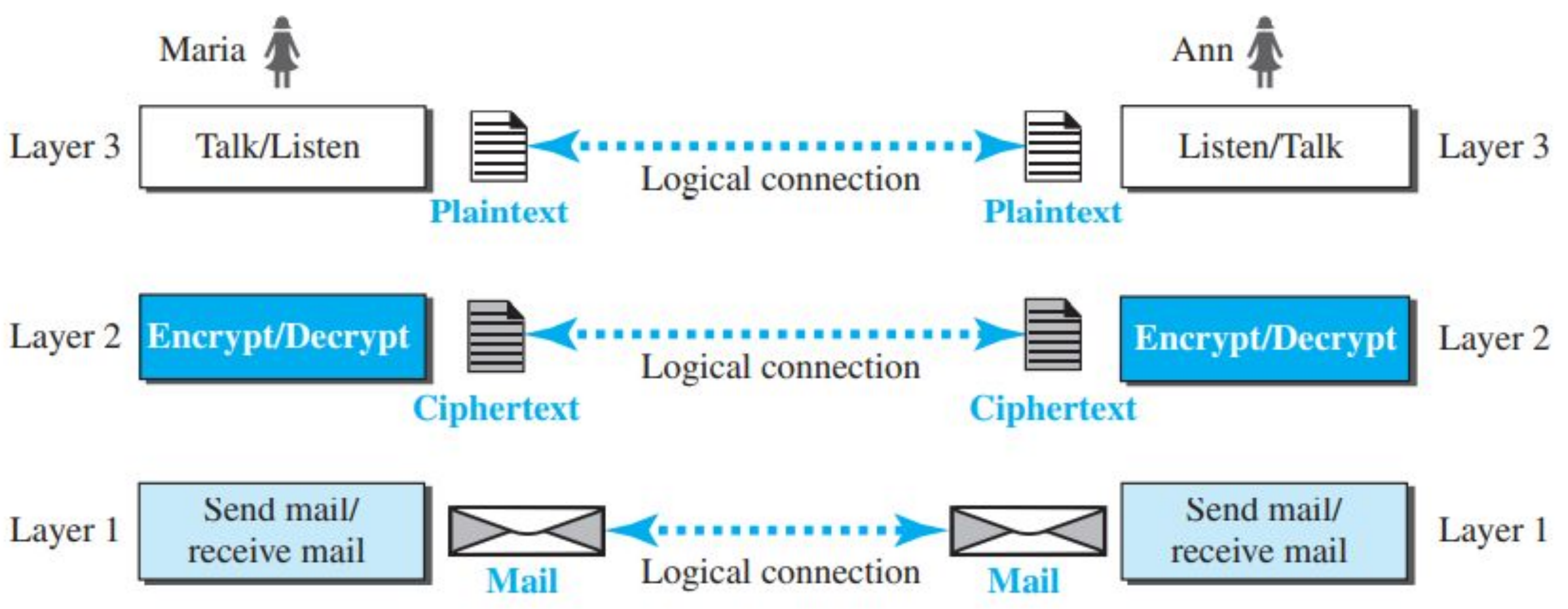
Principles of Protocol Layering

- First principle
  - For bidirectional communication- each layer should be able to perform two opposite tasks, one in each direction.
  - In Fig, listen/talk in layer 3, enc/dec in layer2 and send/rece in layer 1
- Second principle
  - the two objects under each layer at both sites should be identical.
  - In Fig, below layer 2 is ciphertext ….

Logical Connections

- a logical (imaginary) connection at each layer is established through which they can send the object created from that layer

Figure 2.3 *Logical connection between peer layers*



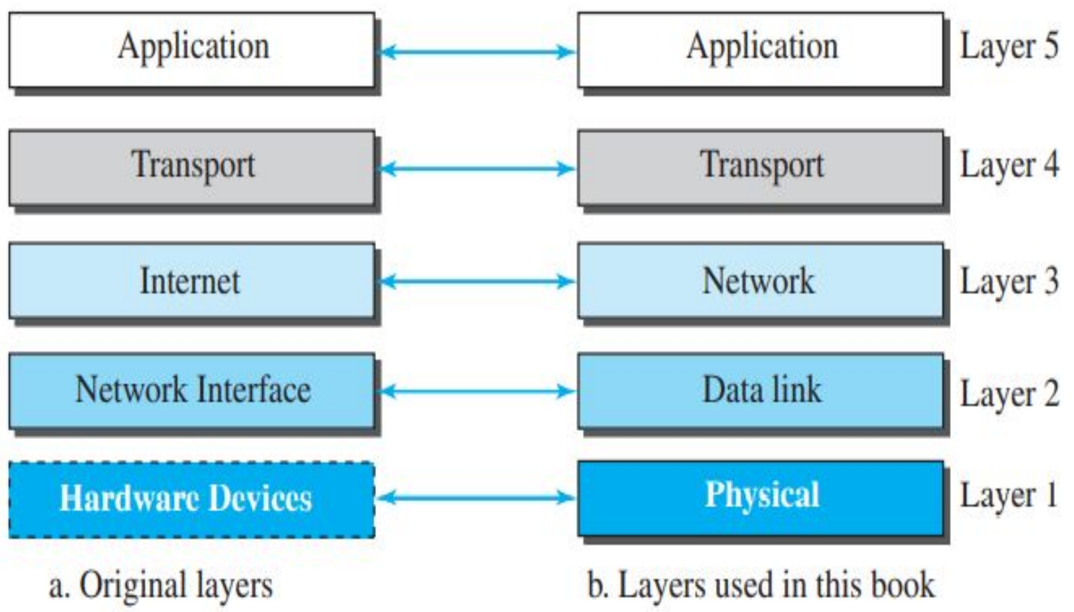
TCP/IP PROTOCOL SUITE

- TCP/IP is a protocol suite (a set of protocols organized in different layers) used in the Internet today.
- It is a hierarchical protocol made up of interactive modules, each of which provides a specific functionality.
- Hierarchical - each upper level protocol is supported by the services provided by one or more lower level protocols.
- original suite -four software layers built upon the hardware. Host-to-network, Internet, Transport and Application.
- Today- a five-layer model.

Figure 2.4 Layers in the TCP/IP protocol Suit

Layered Architecture

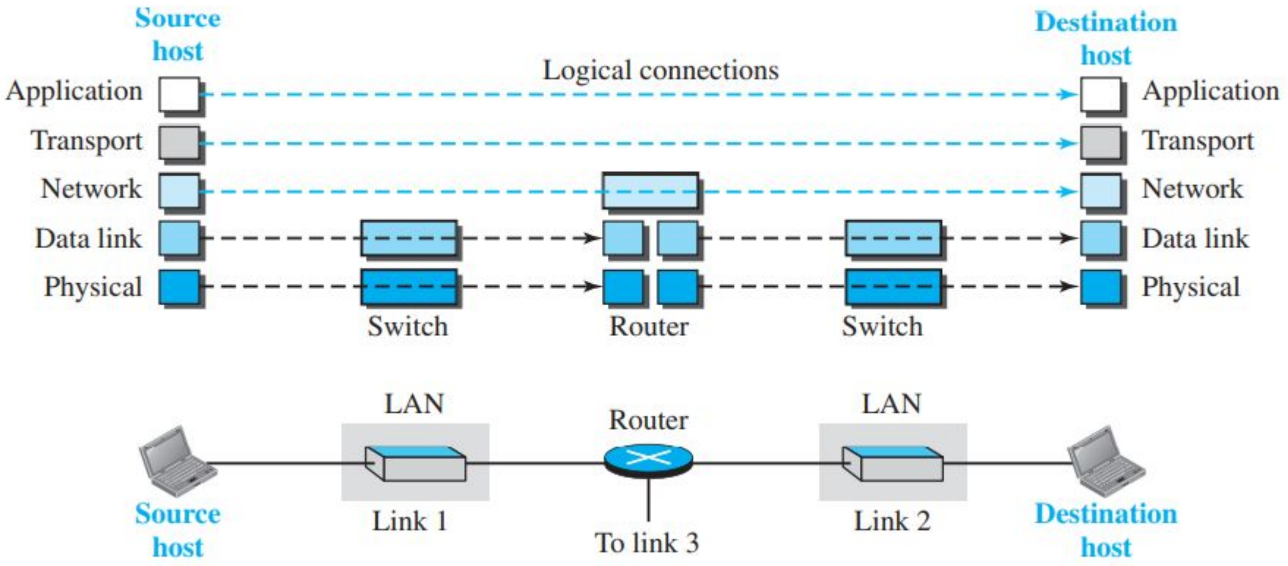
Figure 2.4 Layers in the TCP/IP protocol suite



TCP/IP PROTOCOL SUITE

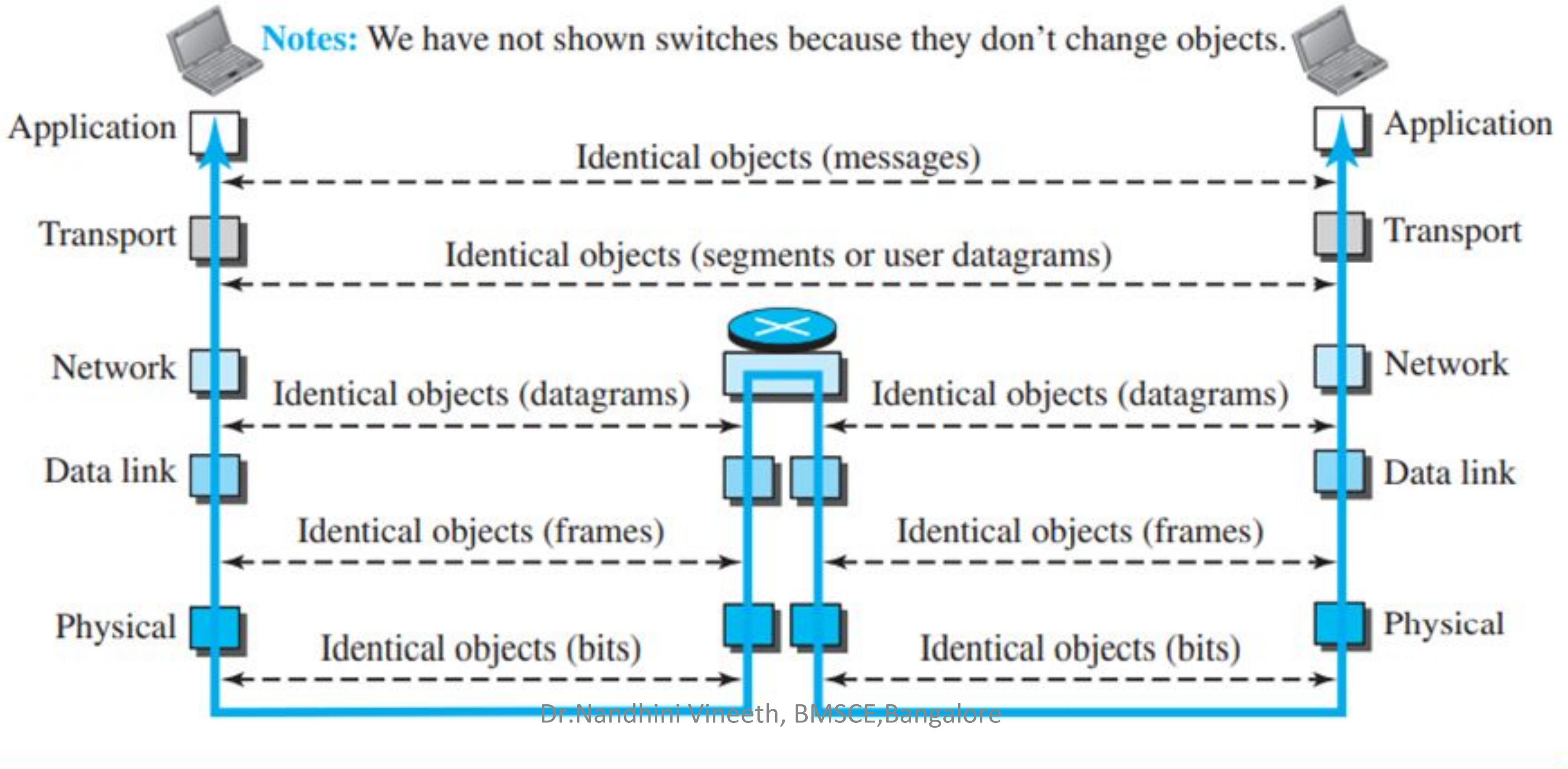
- When A wants to talk to B, two switches and a router are involved
- Switches work with two layers
- The same protocol will be maintained on both sides of the switch
- Router works with three layers
- Different pairs (phy/datalink) will be maintained by both sides

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



Objects in TCP/IP Protocol Suite

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



Layers in the TCP/IP Protocol Suite - Description

Physical layer

- lowest level
- PL comm between two devices is still a logical communication
- Hidden layer, the transmission media, under the physical layer
- Two devices are connected by a transmission medium (cable or air)
- transmission medium does not carry bits; it carries electrical or optical signals.
- Bits received in a frame from the DLL are transformed and sent through the transmission media, the logical unit between two physical layers in two devices is a bit.
- several protocols are used that transform a bit to a signal

Layers in the TCP/IP Protocol Suite - Description

Data-link Layer:

- Multiple links may be available for packets from source to destination
- Routers is responsible to choose the best link
- DLL has the responsibility for taking the datagram and move it across the link
- The link - wired LAN with a link-layer switch, a wireless LAN, a wired WAN, or a wireless WAN.
- Different protocols are used with diff link type.
- TCP/IP does not define any specific protocol for the data-link layer.
- It supports all the standard and proprietary protocols.
- Any protocol that can take the datagram and carry it through the link suffices for the network layer.
- DLL takes a datagram and encapsulates it in a packet called a frame.
- Some link-layer protocols provide complete error detection and correction, some provide only error correction

Layers in the TCP/IP Protocol Suite - Description

Network Layer

- responsible for host-to-host communication and routing the packet through possible routes.
- responsible for creating a connection between the source computer and the destination computer.
- routers in the path are responsible for choosing the best route for each packet.
- not merged with transport layer as fewer protocols on the routers.
- In internet, NL includes the main protocol, Internet Protocol (IP), that defines the format of the packet, called a datagram at the network layer.
- IP also defines the format and the structure of addresses used in this layer.
- IP is also responsible for routing a packet from its source to its destination, which is achieved by each router forwarding the datagram to the next router in its path.

Layers in the TCP/IP Protocol Suite - Description

Network Layer

- IP is a connectionless protocol that provides no flow control, no error control, and no congestion control services
- includes unicast (one-to-one) and multicast (one-to-many) routing protocols.
- IP does routing with the help of a routing protocol that creates forwarding tables
- Other protocols- that help IP in its delivery and routing tasks.
  - Internet Control Message Protocol (ICMP) helps IP to report some problems when routing a packet.
  - Internet Group Management Protocol (IGMP) is another protocol that helps IP in multitasking.
  - Dynamic Host Configuration Protocol (DHCP) helps IP to get the network-layer address for a host.
  - Address Resolution Protocol (ARP) is a protocol that helps IP to find the link-layer address of a host or a router when its network-layer address is given

Layers in the TCP/IP Protocol Suite - Description

Transport Layer

- The logical connection at the transport layer is also end-to-end.
- Source host TL -gets the message from the application layer(application program), encapsulates it in a transport layer packet (called a segment or a user datagram in different protocols) and sends it, through the logical (imaginary) connection, to the transport layer at the destination host (application program).
- more than one protocol in the transport layer- each application program can use the protocol that best matches its requirement.
- In Internet, main protocol, Transmission Control Protocol (TCP), is a connection-oriented protocol – Connection establishment, data transfer and connection release
- It creates a logical pipe between two TCPs for transferring a stream of bytes.

Layers in the TCP/IP Protocol Suite - Description

Transport Layer Contd..

TCP provides

Flow control

matching the sending data rate of the source host with the receiving data rate of the destination host to prevent overwhelming the destination

Error control

to guarantee that the segments arrive at the destination without error and resending the corrupted ones

Congestion control

to reduce the loss of segments due to congestion in the network

User Datagram Protocol

- a connectionless protocol that transmits user datagrams without first creating a logical connection
- each user datagram is an independent entity without being related to the previous or the next one (the meaning of the term connectionless).
- a simple protocol that does not provide flow, error, or congestion control.
- an application program that needs to send short messages and cannot afford the retransmission of the packets involved in TCP, when a packet is corrupted or lost prefers UDP.
- Stream Control Transmission Protocol (SCTP) is designed to respond to new applications that are emerging in the multimedia.

Layers in the TCP/IP Protocol Suite - Description

Application Layer

- the logical connection between the two application layers is end-to-end.
- Exchanging of messages done between each other as though there were a bridge between the two layers.
- Communication is done through all the layers.
- Communication at the application layer is between two processes (two programs running at this layer).
- a process sends a request to the other process and receives a response.
- Process-to-process communication is the duty of the application layer.

Layers in the TCP/IP Protocol Suite - Description

includes many predefined protocols, but a user can also create a pair of processes to be run at the two hosts.

Hypertext Transfer Protocol (HTTP) - a vehicle for accessing the World Wide Web (WWW).

Simple Mail Transfer Protocol (SMTP) - the main protocol used in electronic mail (e-mail) service.

File Transfer Protocol (FTP) - used for transferring files from one host to another.

Terminal Network (TELNET) and Secure Shell (SSH) - used for accessing a site remotely.

Simple Network Management Protocol (SNMP) - used by an administrator to manage the Internet at global and local levels.

Domain Name System (DNS)-used by other protocols to find the network-layer address of a computer.

Internet Group Management Protocol (IGMP) - used to collect membership in a group.

Encapsulation and Decapsulation

- **Encapsulation at the Source Host**
- At the source, we have only encapsulation.
- 1. AL
  - A message normally does not contain any header or trailer, but if it does, we refer to the whole as the message.
  - The message is passed to the transport layer.
- 2. TL
  - takes the message as the payload, adds the transport layer header to the payload.
  - HDR- identifiers of the source and destination application programs + extra info for end-to-end delivery of the message, like information needed for flow, error control, or congestion control.
  - Result - segment (in TCP) and the user datagram (in UDP).
  - The transport layer then passes the packet to the network layer.
- 3. NL
  - gets payload from TL and adds its own header to payload
  - HDR – addresses of the source and destination hosts + info on error checking + fragmentation etc
  - Result – datagram
  - NL passes datagram to DLL

Encapsulation and Decapsulation

Encapsulation at the Source Host Contd..

DLL

- Gets payload from NL and add its own header
- HDR- link-layer addresses of the host or the next hop (the router).
- Result - frame
- DLL passes frames to PL

Encapsulation and Decapsulation

- **Decapsulation and Encapsulation at the Router**
  - both decapsulation and encapsulation because the router is connected to two or more links.
  - DLL
    - receives frames from PL
    - decapsulates the datagram from the frame and passes to NL
  - NL
    - inspects the source and destination addresses in the datagram header
    - consults its forwarding table to find the next hop
    - Contents of datagram will not be changed except when there is fragmentation
    - Datagram is passed to DLL
  - DLL
    - encapsulates the datagram in a frame and passes to PL

- **Decapsulation at the Destination Host**
  - each layer only decapsulates the packet received
  - removes the payload
  - delivers the payload to the next-higher layer protocol
  - decapsulation in the host involves error checking

Figure 2.8 *Encapsulation/Decapsulation*

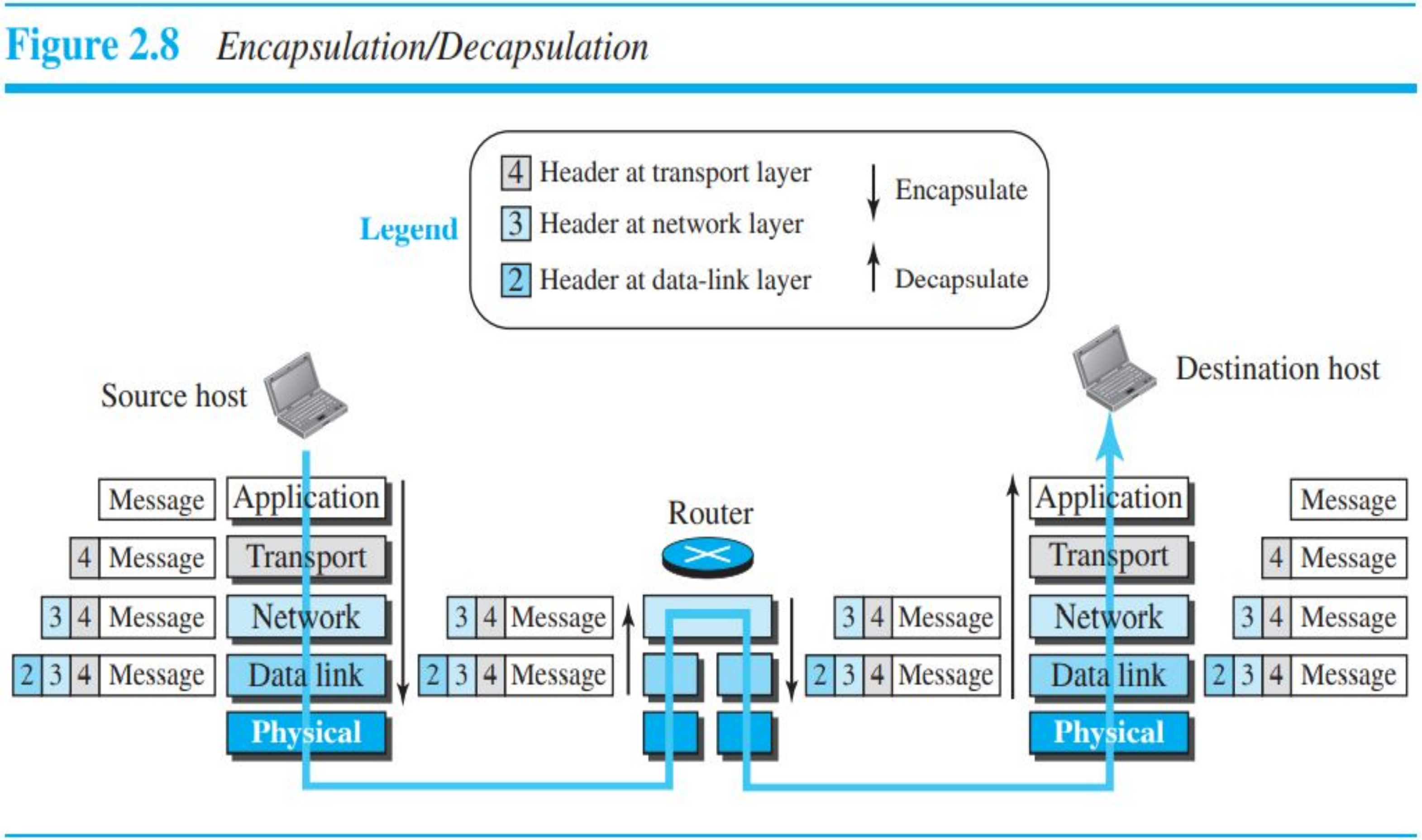


Figure 2.9 *Addressing in the TCP/IP protocol suite*

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

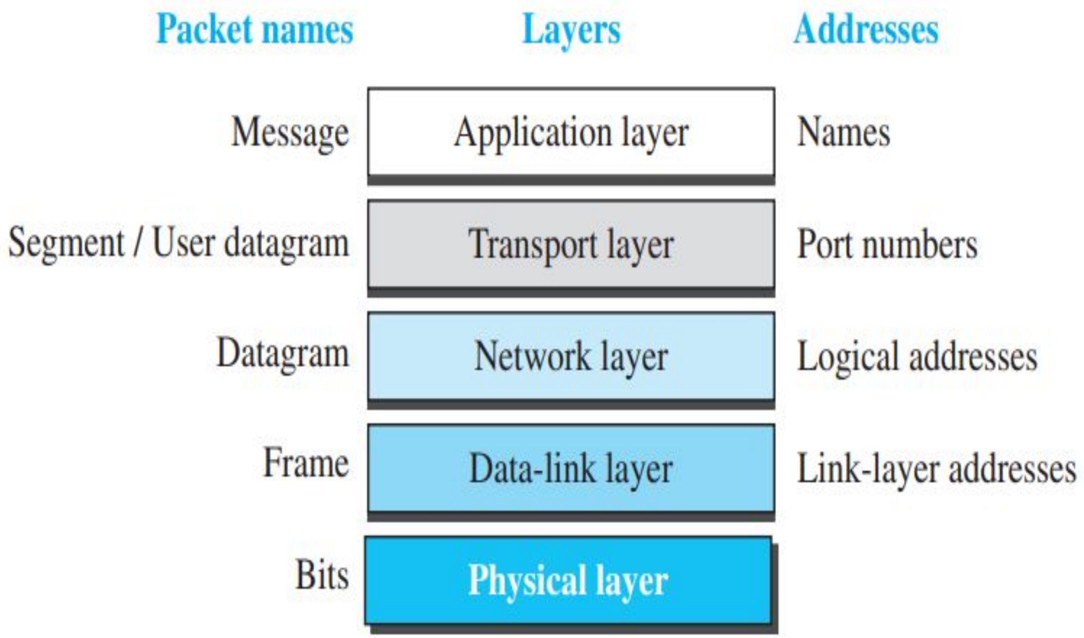
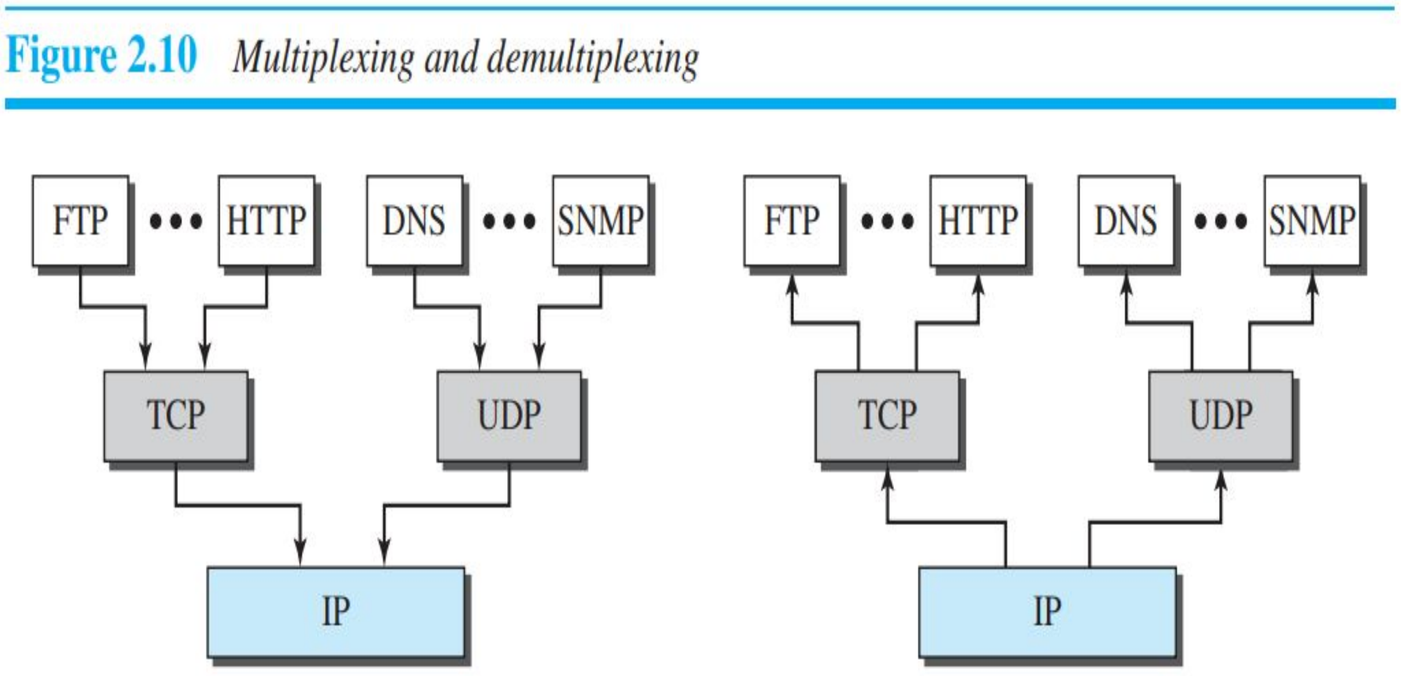


Figure 2.10 *Multiplexing and Demultiplexing*



a. Multiplexing at source

b. Demultiplexing at destination

OSI Layer along with TCP/IP

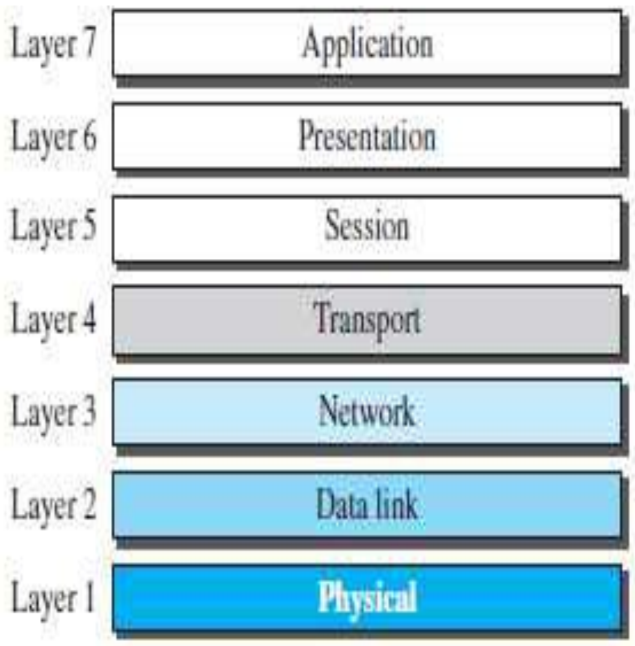


Figure 2.11 *The OSI model*

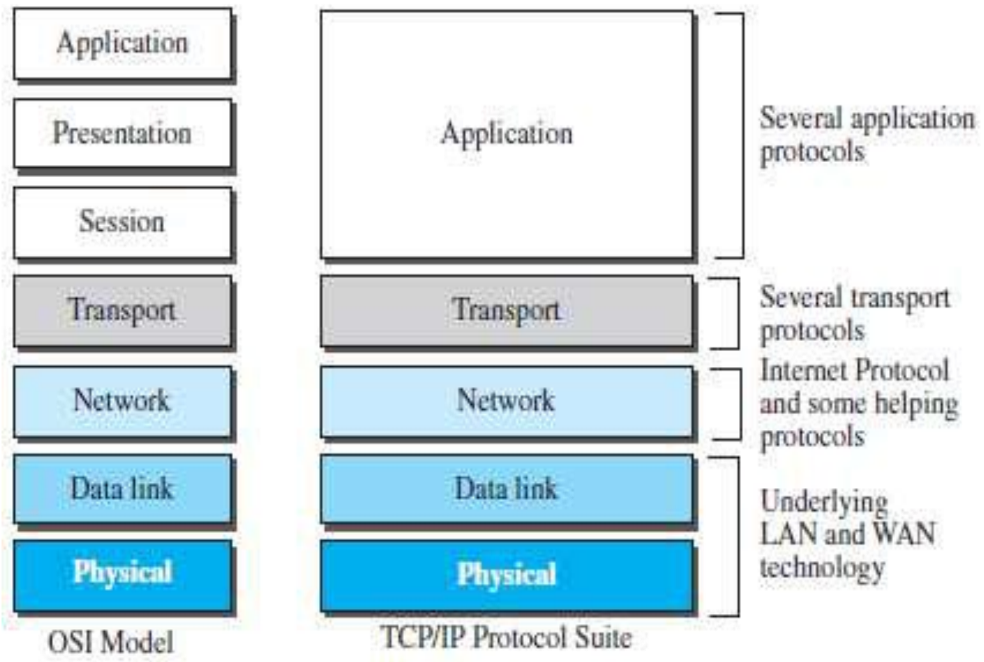


Figure 2.12 *TCP/IP and OSI model*

Limitations of OSI layer

- OSI was completed when TCP/IP was fully in place and a lot of time and money had been spent on the suite; changing it would cost a lot.

- Some layers in the OSI model were never fully defined.

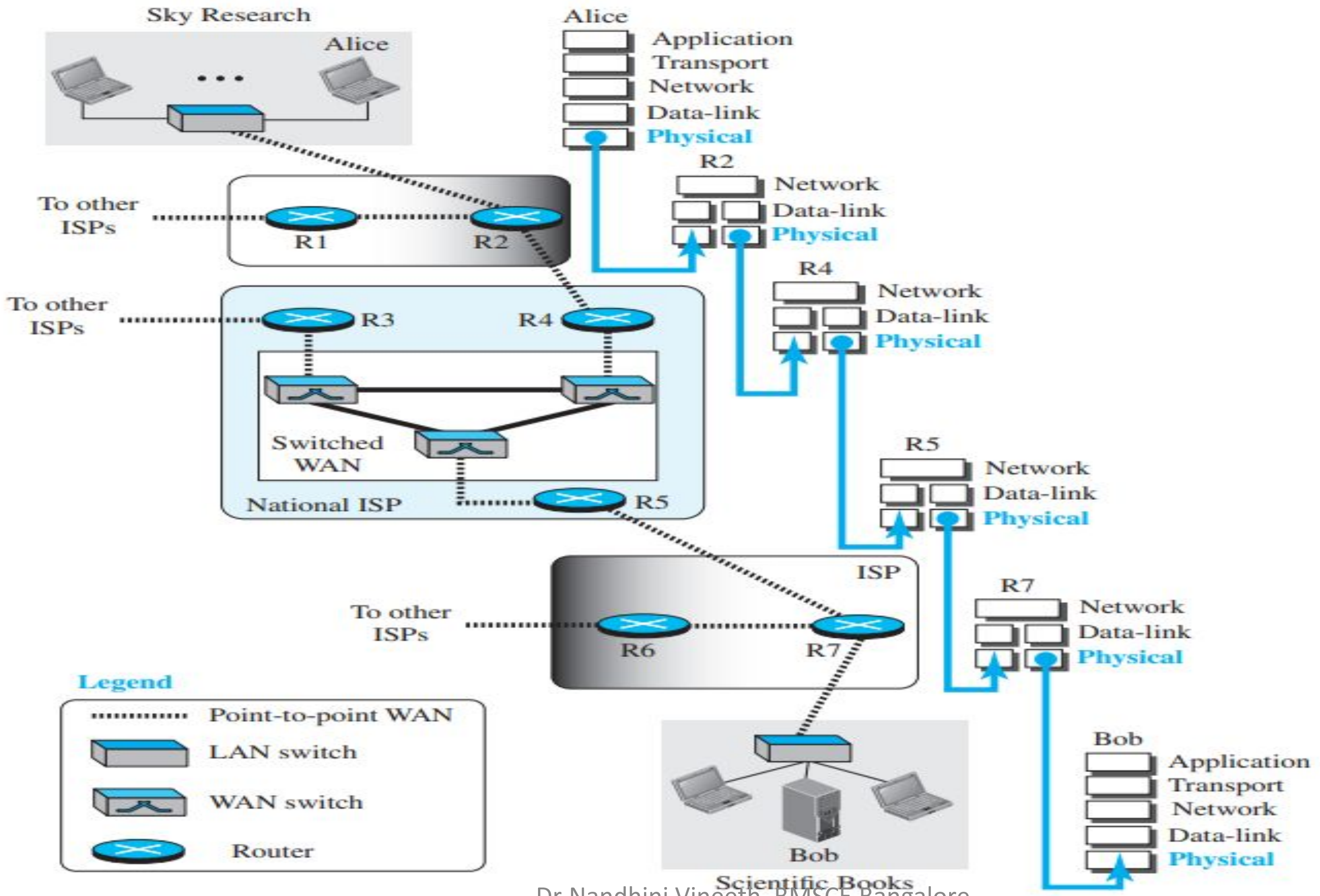
- When OSI was implemented by an organization in a different application, it did not show a high enough level of performance

Physical Layer

- Major Functionality of PL
  - to move data in the form of electromagnetic signals across a transmission medium
  - Transmission media work by conducting energy along a physical path. For transmission, data needs to be changed to signals.

Figure 3.1 Communication at the Physical Layer

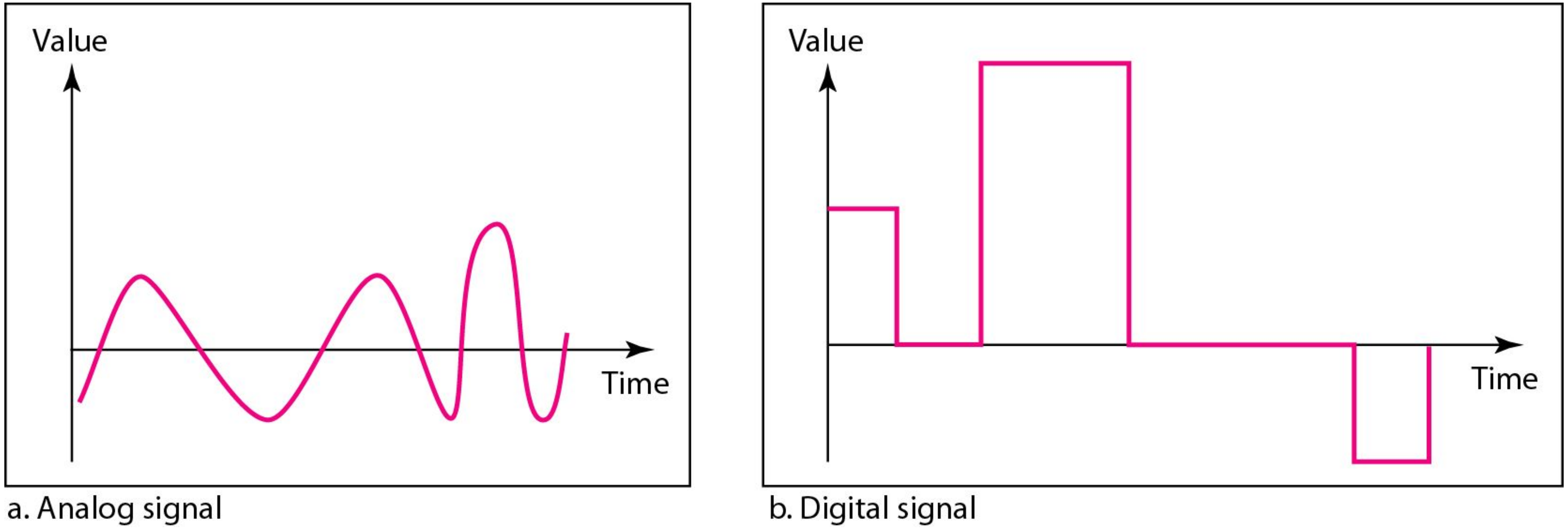
Figure 3.1 *Communication at the physical layer*



ANALOG AND DIGITAL

- Data
  - Can be analog or digital.
  - Analog data are continuous and take continuous values.
  - Digital data have discrete states and take discrete values.
- Signals
  - Can be analog or digital.
  - Analog signals can have an infinite number of values in a range.
  - Digital signals can have only a limited number of values.

Figure 3.1 Comparison of analog and digital signals



Periodic and Nonperiodic

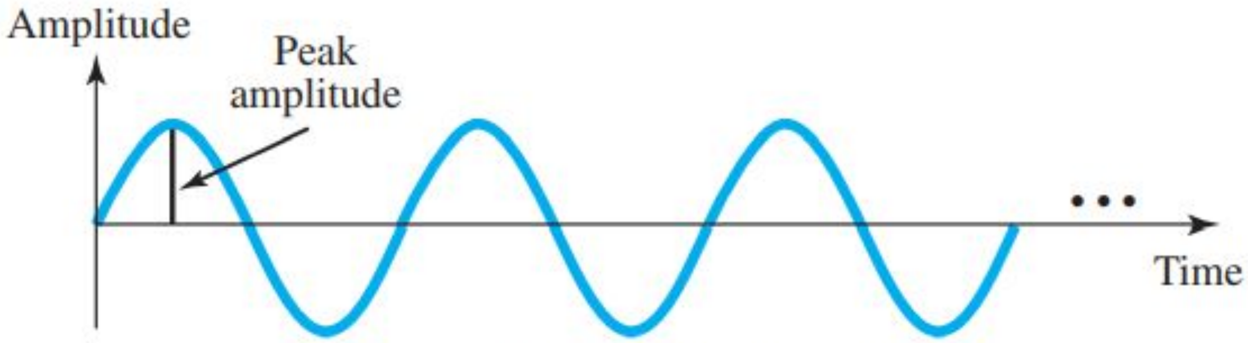
- A periodic signal completes a pattern within a measurable time frame, called a period, and repeats that pattern over subsequent identical periods.

- The completion of one full pattern is called a cycle.

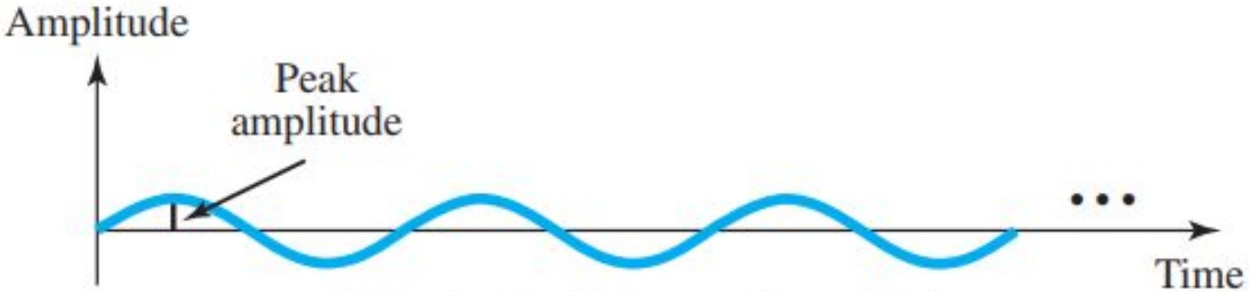
- A nonperiodic signal changes without exhibiting a pattern or cycle that repeats over time

Periodic Analog signal

Figure 3.4 *Two signals with the same phase and frequency, but different amplitudes*

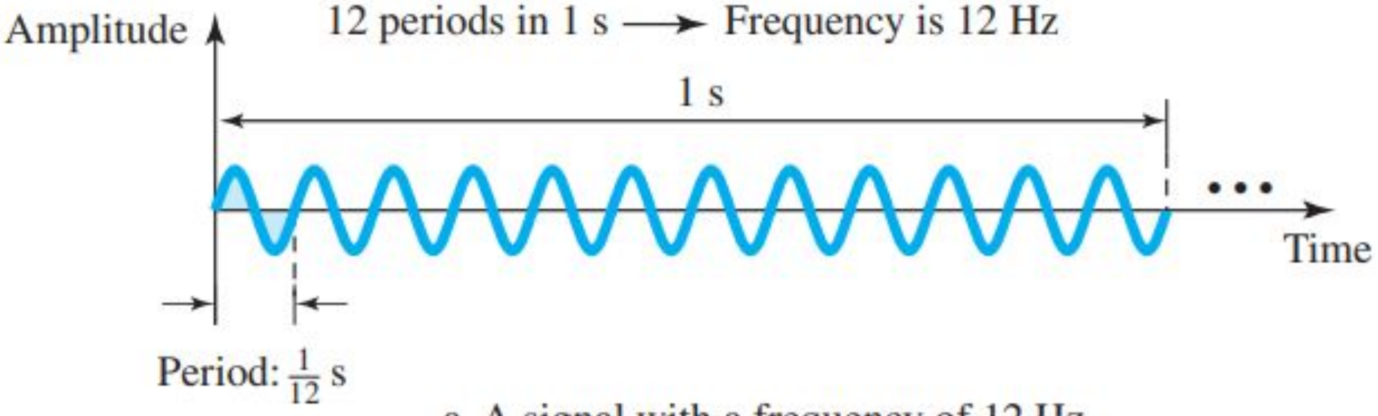


a. A signal with high peak amplitude

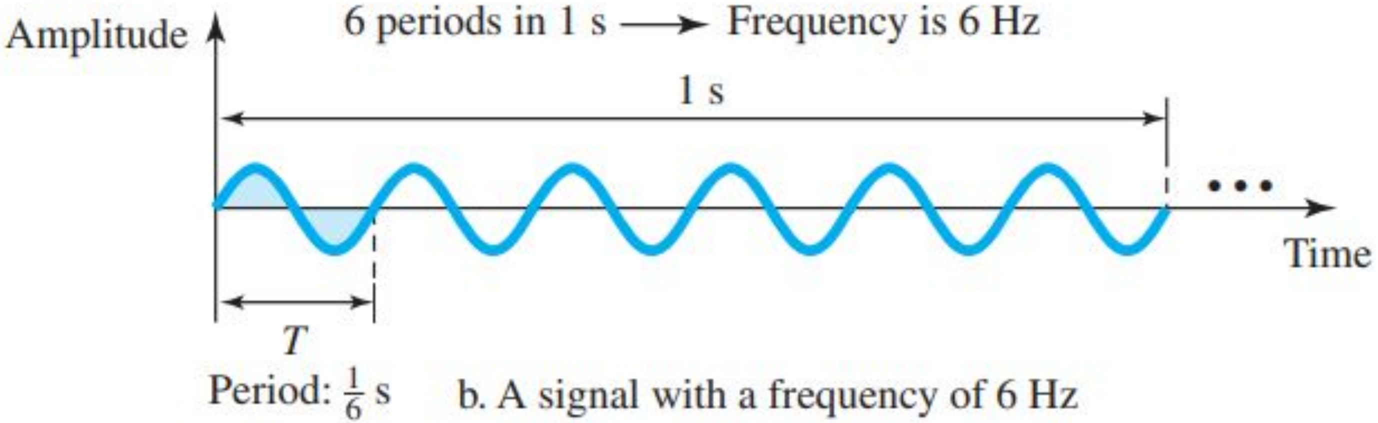


b. A signal with low peak amplitude

Figure 3.5 *Two signals with the same amplitude and phase, but different frequencies*



a. A signal with a frequency of 12 Hz



b. A signal with a frequency of 6 Hz

Table 3.1 *Units of period and frequency*

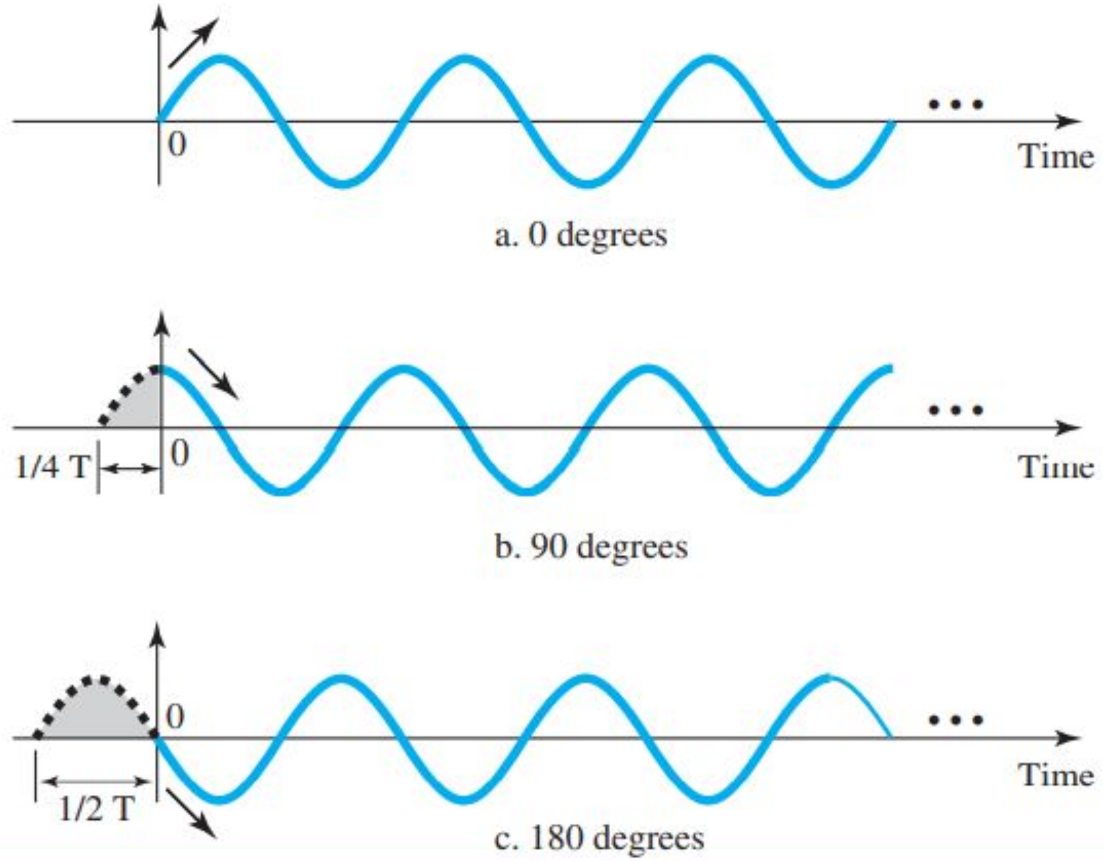
Period		*Frequency*	
Unit	*Equivalent*	*Unit*	*Equivalent*
Seconds (s)	1 s	Hertz (Hz)	1 Hz
Milliseconds (ms)	10^{-3} s	Kilohertz (kHz)	10^3 Hz
Microseconds (μs)	10^{-6} s	Megahertz (MHz)	10^6 Hz
Nanoseconds (ns)	10^{-9} s	Gigahertz (GHz)	10^9 Hz
Picoseconds (ps)	10^{-12} s	Terahertz (THz)	10^{12} Hz

Example 3.3

The power we use at home has a frequency of 60 Hz (50 Hz in Europe). The period of this sine wave can be determined as follows:

$$T = \frac{1}{f} = \frac{1}{60} = 0.0166 \text{ s} = 0.0166 \times 10^3 \text{ ms} = 16.6 \text{ ms}$$

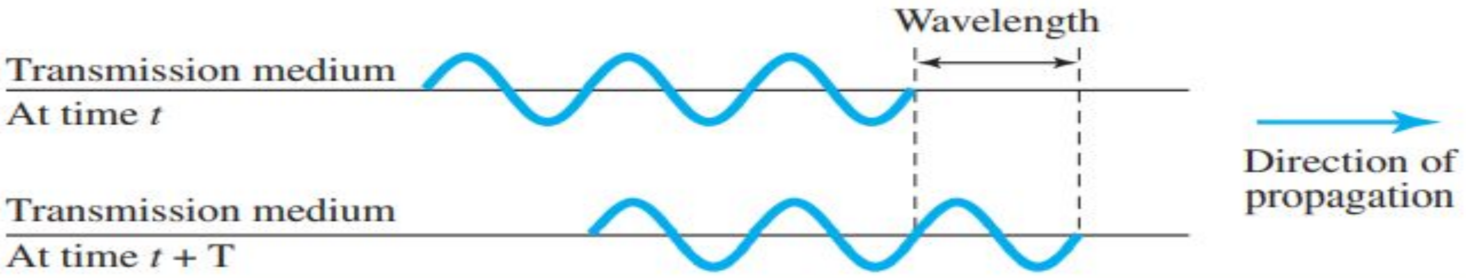
Figure 3.6 *Three sine waves with the same amplitude and frequency, but different phases*



3.2.3 Wavelength

Wavelength is another characteristic of a signal traveling through a transmission medium. Wavelength binds the period or the frequency of a simple sine wave to the **propagation speed** of the medium (see Figure 3.7).

Figure 3.7 *Wavelength and period*

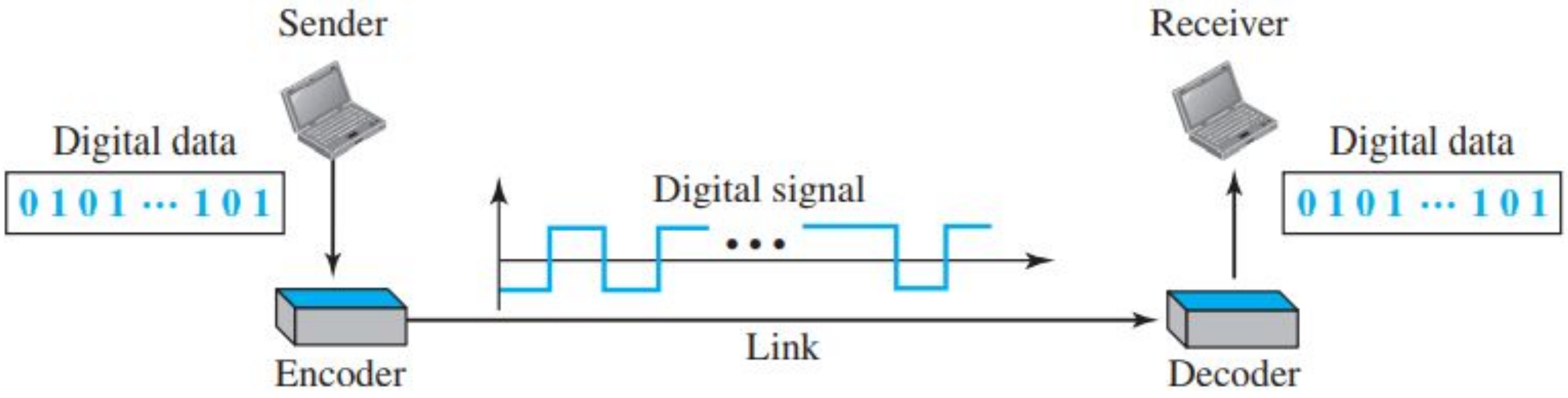


- Frequency of a signal is independent of the medium, the wavelength depends on both the frequency and the medium
- wavelength to describe the transmission of light in an optical fiber. The wavelength is the distance a simple signal can travel in one period.

$$\text{Wavelength} = (\text{propagation speed}) \times \text{period} = \frac{\text{propagation speed}}{\text{frequency}}$$

$$\lambda = \frac{c}{f}$$

Figure 4.1 *Line coding and decoding*



Relationship between data Element and signal Element

- A data Element is the smallest entity that can represent a piece of information.
- A data element is the bit.
- Data elements are being carried

- Signal element is shortest unit (time wise) of a digital signal
- A Signal elements carries data elements

Data rate and Signal rate

Data Rate

- It defines the no of data elements(bits) sent in 1 sec.
- The unit is bits per sec(bps)
- Data rate is also known as bit rate
- In the data communication the data rate should increase
- Increase of data rate improves the transmission

Signal rate

It defines the no of signal elements sent in 1 sec

The unit is baud

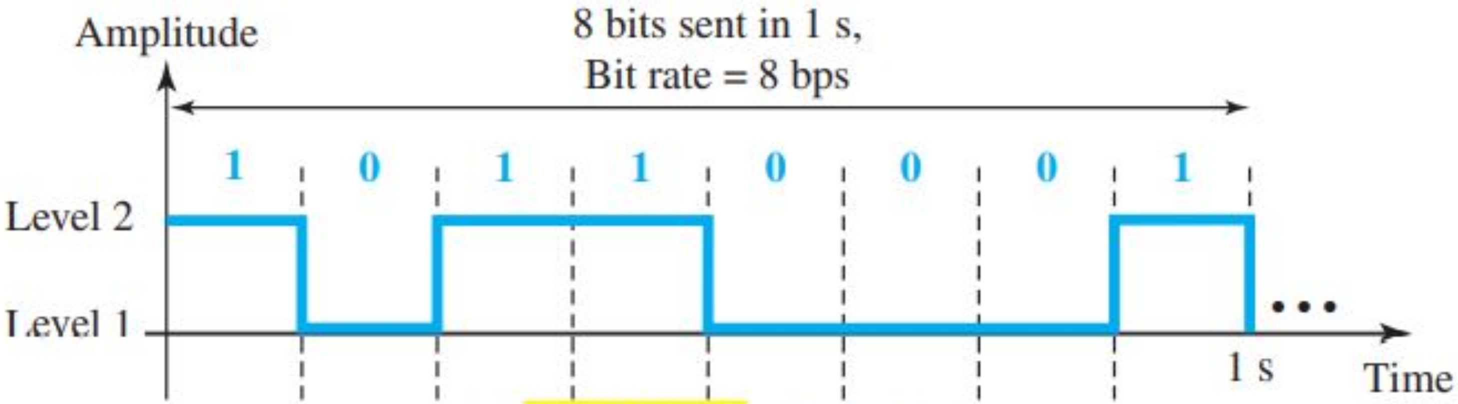
The signal rate is also known as pulse rate, the modulation rate, or the baud rate

The signal rate should decrease

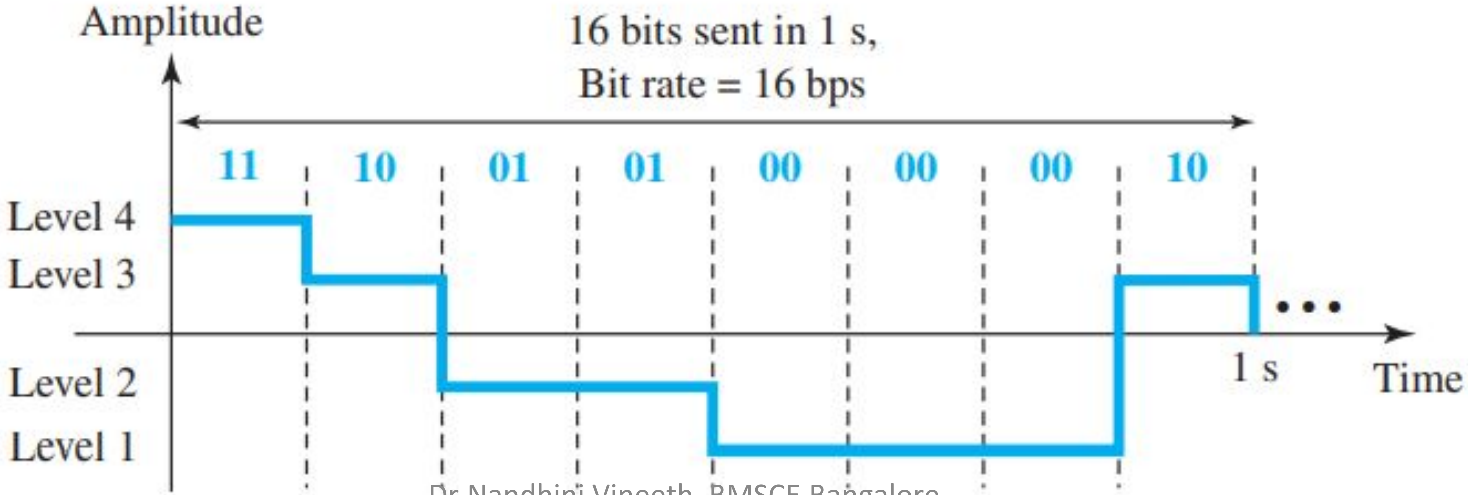
Decreasing the signal rate decreases the bandwidth requirement

DIGITAL SIGNALS

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

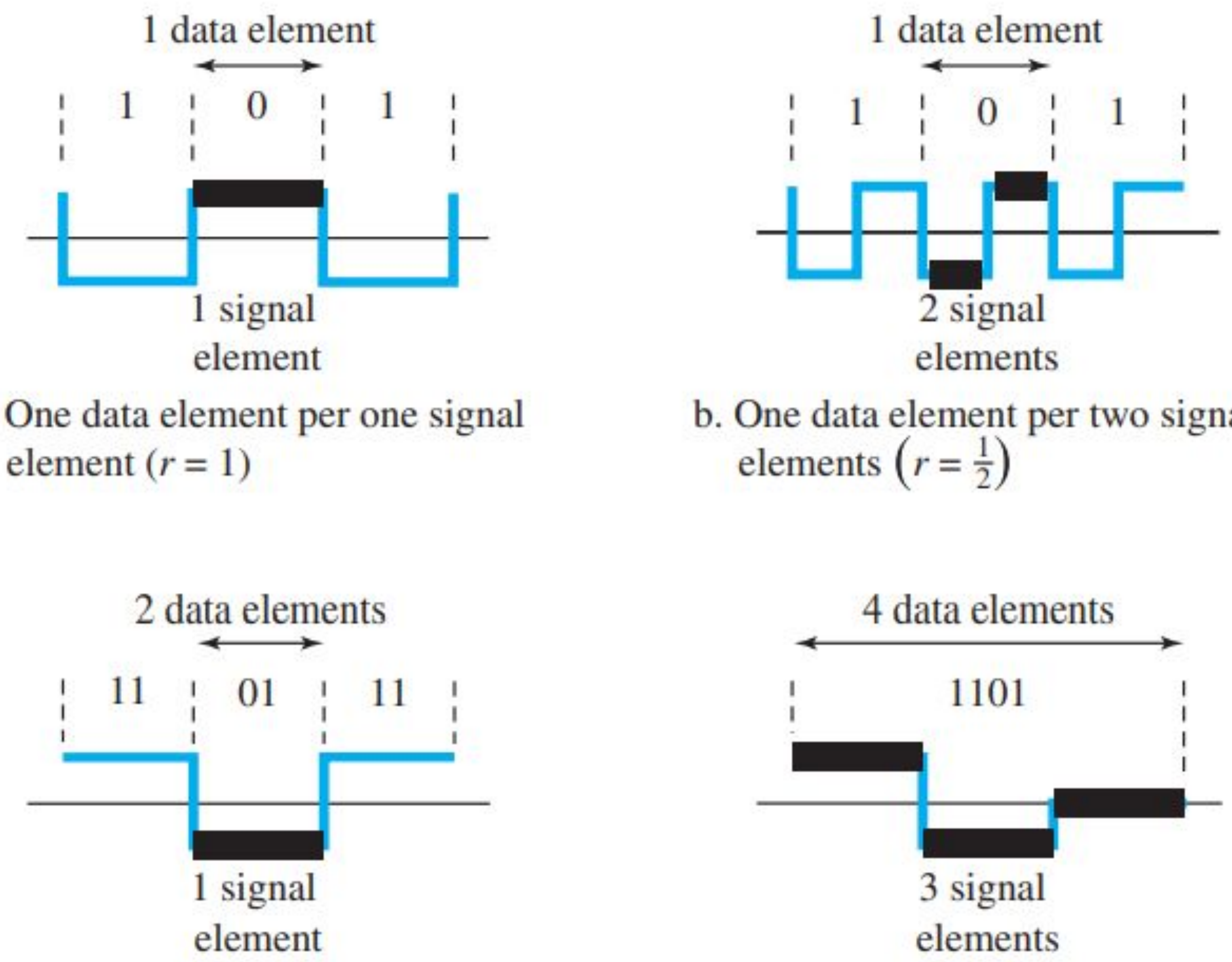


a. A digital signal with two levels



b. A digital signal with four levels

Figure 4.2 *Signal element versus data element*



a. One data element per one signal element ($r = 1$)

b. One data element per two signal elements $\left(r = \frac{1}{2}\right)$

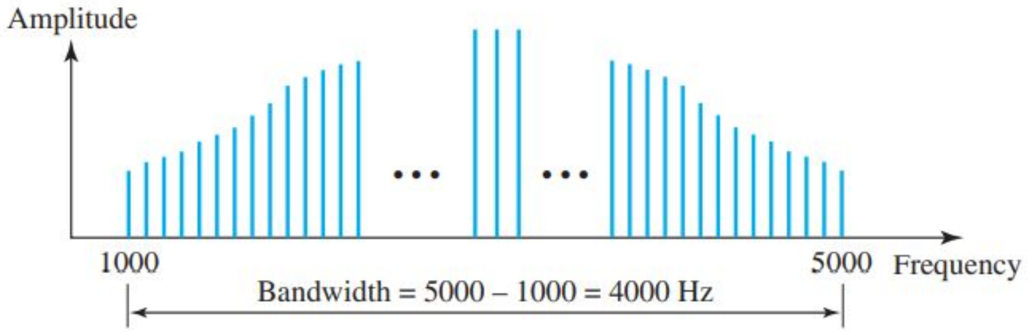
c. Two data elements per one signal element ($r = 2$)

d. Four data elements per three signal elements $\left(r = \frac{4}{3}\right)$

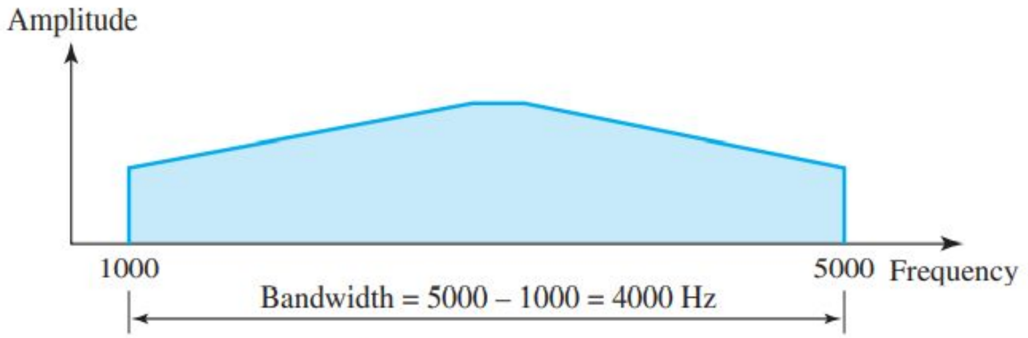
BANDWIDTH-

- is the difference between the highest and the lowest frequencies contained in that signal.
- is proportional to the signal rate (baud rate).
- The minimum bandwidth can be given as
  - **Bmin = c X N X (1 / r)**
- maximum data rate if the bandwidth of the channel is given.
  - **Nmax = (1 / c) X B X r**

Figure 3.13 *The bandwidth of periodic and nonperiodic composite signals*



a. Bandwidth of a periodic signal



b. Bandwidth of a nonperiodic signal

Relationship between Data Rate and Signal Rate

- $S = C * N * (\frac{1}{r})$
- Where
  - N → data rate in bps
  - C → Case factor, which varies for each case
  - S → Number of signal elements
  - R → Previously defined factor

Example 4.1

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

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

Example 4.2

The maximum data rate of a channel is $N_{max} = 2 \times B \times \log_2 L$ (defined by the Nyquist formula). Does this agree with the previous formula for N_{max}?

Solution

A signal with L levels actually can carry $\log_2 L$ bits per level. If each level corresponds to one signal element and we assume the average case ($c = 1/2$), then we have

$$\boxed{N_{\max} = \frac{1}{c} \times B \times r = 2 \times B \times \log_2 L}$$

Baseline wandering

- A receiver will evaluate the running average power of the received signal (called the baseline) and use that to determine the value of the incoming data elements.

- If the incoming signal does not vary over a long period of time, the baseline will drift and thus cause errors in detection of incoming data elements.

- A good line encoding scheme will prevent long runs of fixed amplitude.

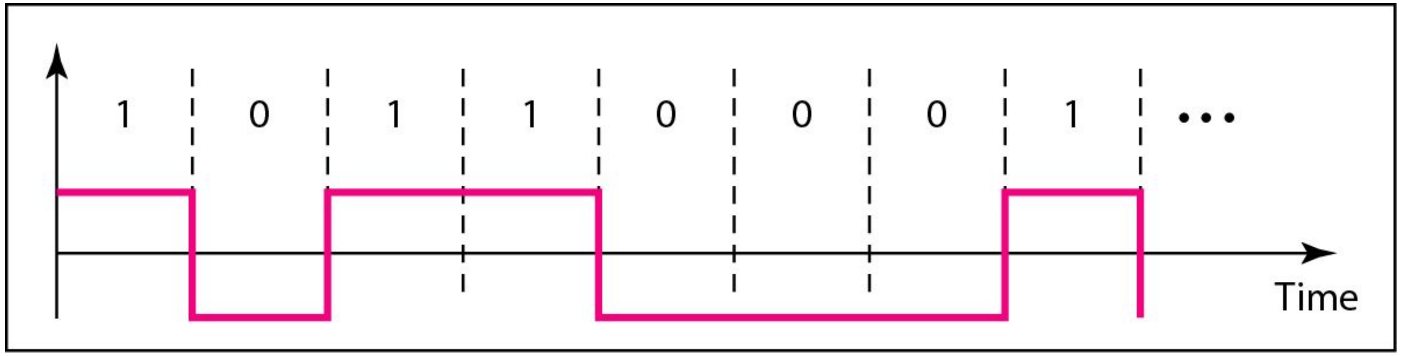
DC components

- When the voltage level remains constant for long periods of time, there is an increase in the low frequencies of the signal (results of Fourier analysis).
- These frequencies around zero are called DC (direct-current) components
- They present problems for a system that cannot pass low frequencies or a system that uses electrical coupling (via a transformer).
- DC component means 0/1 parity that can cause base-line wondering.
- For example, a telephone line cannot pass frequencies below 200 Hz. Also a long-distance link may use one or more transformers to isolate different parts of the line electrically. For these systems, we need a scheme with no DC component.
- Most channels are bandpass and may not support the low frequencies.
- This will require the removal of the dc component of a transmitted signal.

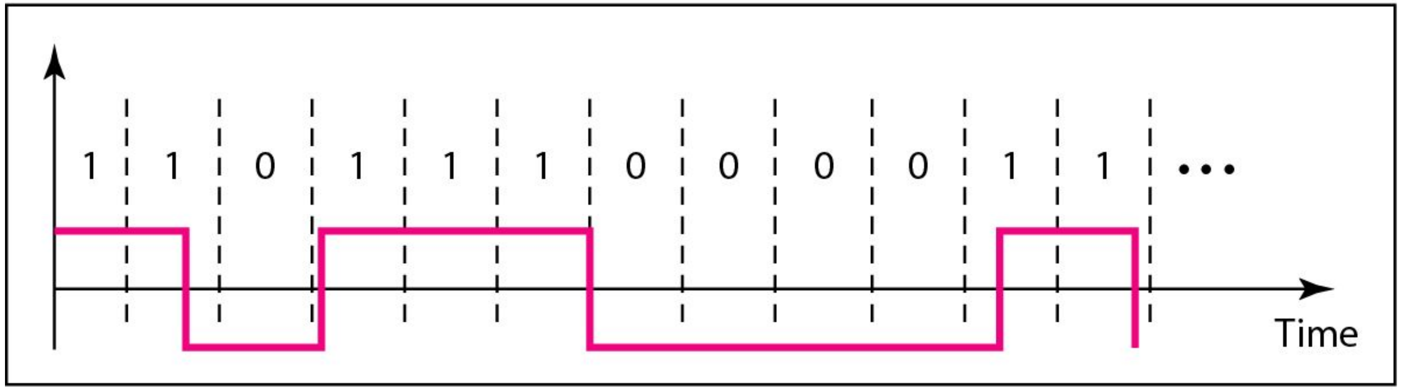
Self synchronization

- The clocks at the sender and the receiver must have the same bit interval.
- If the receiver clock is faster or slower it will misinterpret the incoming bit stream.

Figure 4.3 *Effect of lack of synchronization*



a. Sent



b. Received

Example 4.3

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

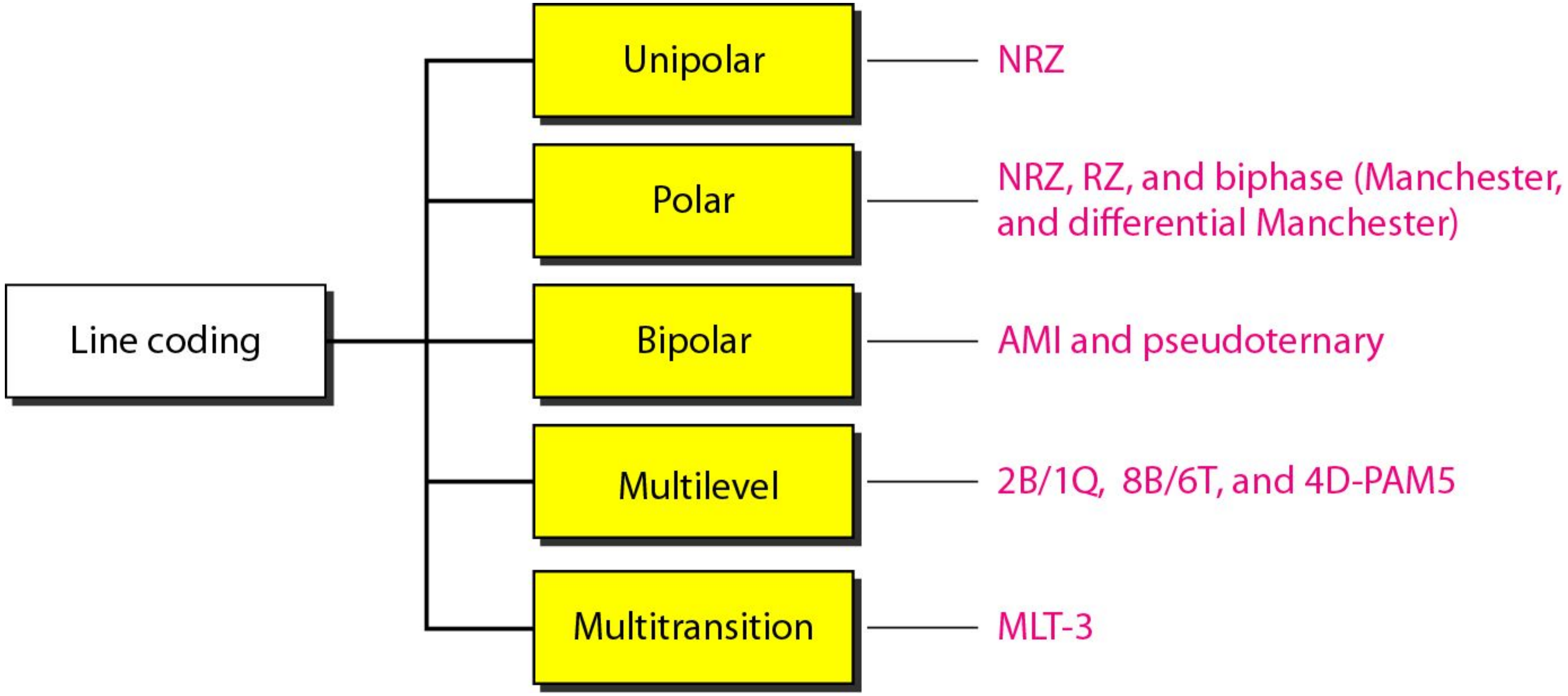
At 1 Mbps, the receiver receives 1,001,000 bps instead of 1,000,000 bps.

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

Built in Error detection, Immunity to noise and Interface , Complexity

- Error detection - errors occur during transmission due to line impairments.
- Some codes are constructed such that when an error occurs it can be detected.
- For example: a particular signal transition is not part of the code. When it occurs, the receiver will know that a symbol error has occurred.
- Noise and interference - there are line encoding techniques that make the transmitted signal “immune” to noise and interference.
- This means that the signal cannot be corrupted, it is stronger than error detection.
- Complexity - A complex scheme is more costly to implement than a simple one. For example, a scheme with four signal levels is more difficult to interpret than one with only two levels.

Figure 4.4 *Line coding schemes*



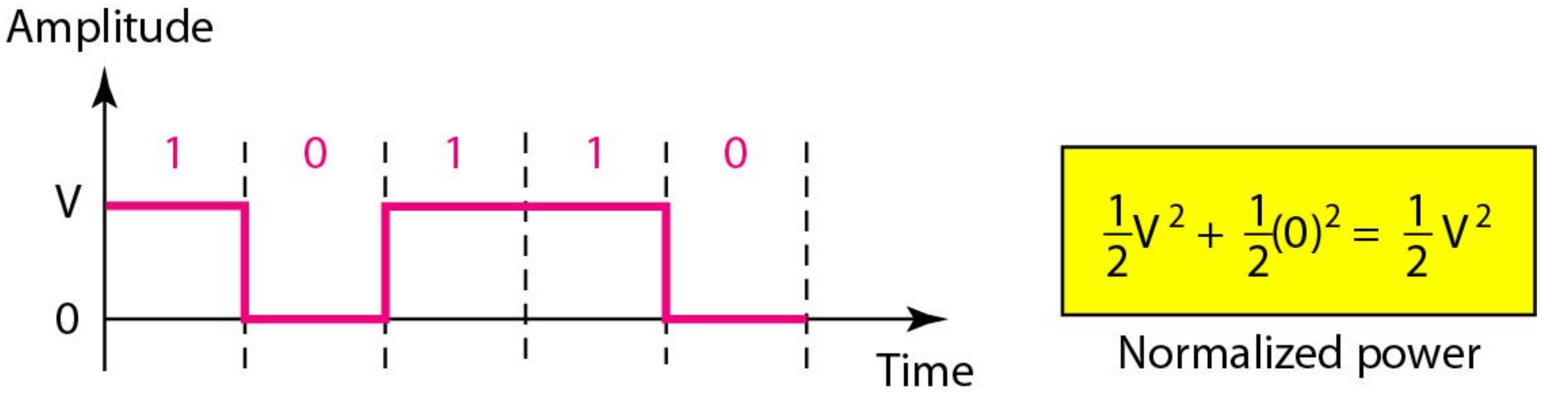
Unipolar

- All signal levels are on one side of the time axis - either above or below

- NRZ - Non Return to Zero scheme is an example of this code. The signal level does not return to zero during a symbol transmission.

- Scheme is prone to baseline wandering and DC components. It has no synchronization or any error detection. It is simple but costly in power consumption.

Figure 4.5 *Unipolar NRZ scheme*

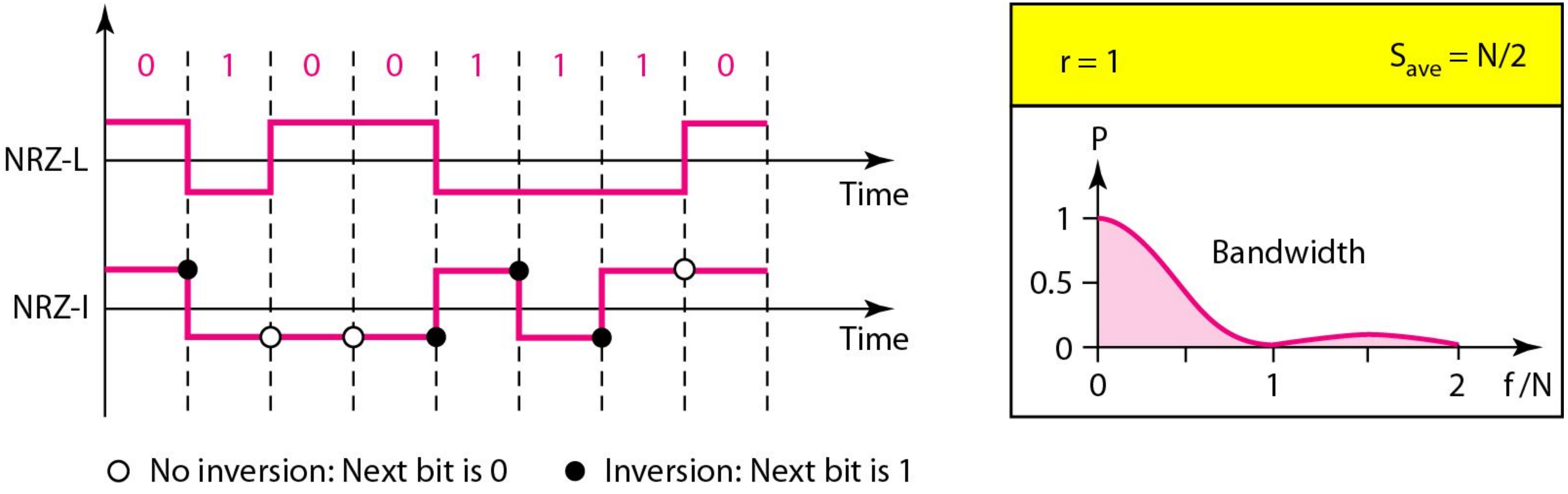


The power of a signal is the sum of the absolute squares of its time-domain samples divided by the signal length,
(Normalized power-the power needed to send 1 bit per unit line resistance)

Polar - NRZ

- The voltages are on both sides of the time axis.
- Polar NRZ scheme can be implemented with two voltages. E.g. +V for 1 and -V for 0.
- There are two versions:
  - NRZ - Level (NRZ-L) - positive voltage for one symbol and negative for the other
  - NRZ - Inversion (NRZ-I) - the change or lack of change in polarity determines the value of a symbol. E.g. a “1” symbol inverts the polarity a “0” does not.

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



Note

In NRZ-L the level of the voltage determines the value of the bit. In NRZ-I the inversion or the lack of inversion determines the value of the bit.

NRZ-L and NRZ-I both have an average signal rate of N/2 Bd.

Note

NRZ-L and NRZ-I both have a DC component problem and baseline wandering, it is worse for NRZ-L. Both have no self synchronization &no error detection. Both are relatively simple to implement.

Example 4.4

A system is using NRZ-I to transfer 1-Mbps data. What are the average signal rate and minimum bandwidth?

Solution

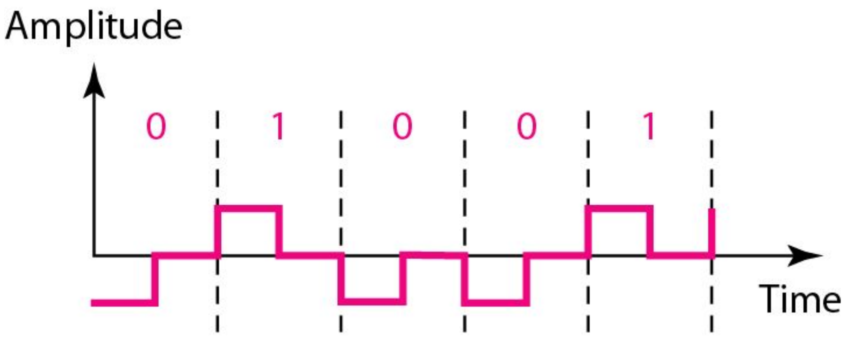
The average signal rate is $S = c \times N \times R = 1/2 \times N \times 1 =$ 500 kbaud. The minimum bandwidth for this average baud rate is $B_{min} = S = 500$ kHz.

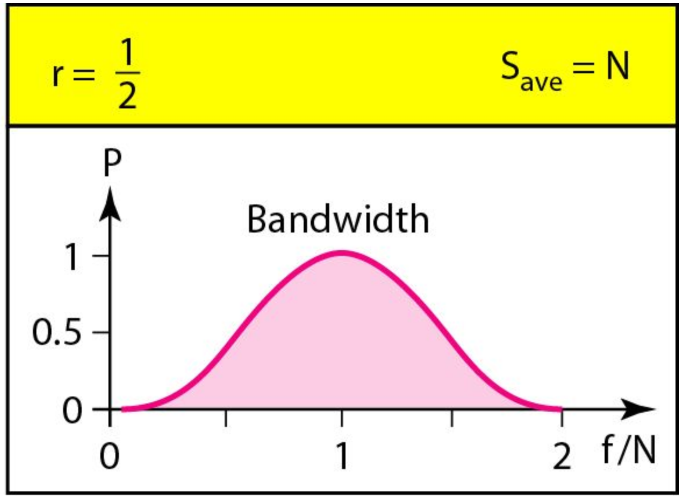
Note c = 1/2 for the avg. case as worst case is 1 and best case is 0

Polar - RZ

- The Return to Zero (RZ) scheme uses three voltage values. +, 0, -.
- Each symbol has a transition in the middle. Either from high to zero or from low to zero.
- This scheme has more signal transitions (two per symbol) and therefore requires a wider bandwidth.
- No DC components or baseline wandering.
- Self synchronization - transition indicates symbol value.
- More complex as it uses three voltage level. It has no error detection capability.

Figure 4.7 *Polar RZ scheme*

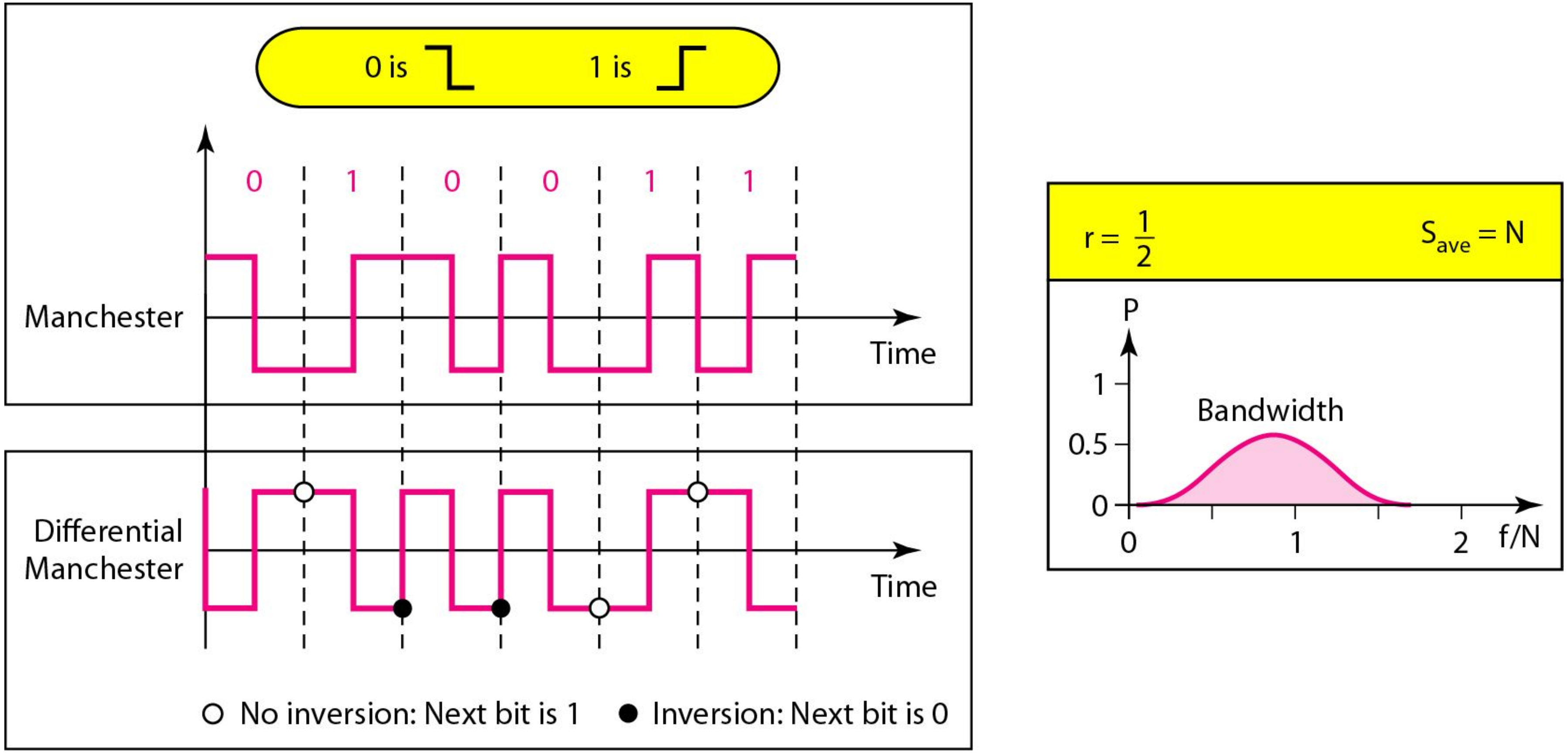




Polar - Biphase: Manchester and Differential Manchester

- Manchester coding consists of combining the NRZ-L and RZ schemes.
  - Every symbol has a level transition in the middle: from high to low or low to high. Uses only two voltage levels.
- Differential Manchester coding consists of combining the NRZ-I and RZ schemes.
  - Every symbol has a level transition in the middle. But the level at the beginning of the symbol is determined by the symbol value. One symbol causes a level change the other does not.

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



Note

In Manchester and differential Manchester encoding, the transition at the middle of the bit is used for synchronization.

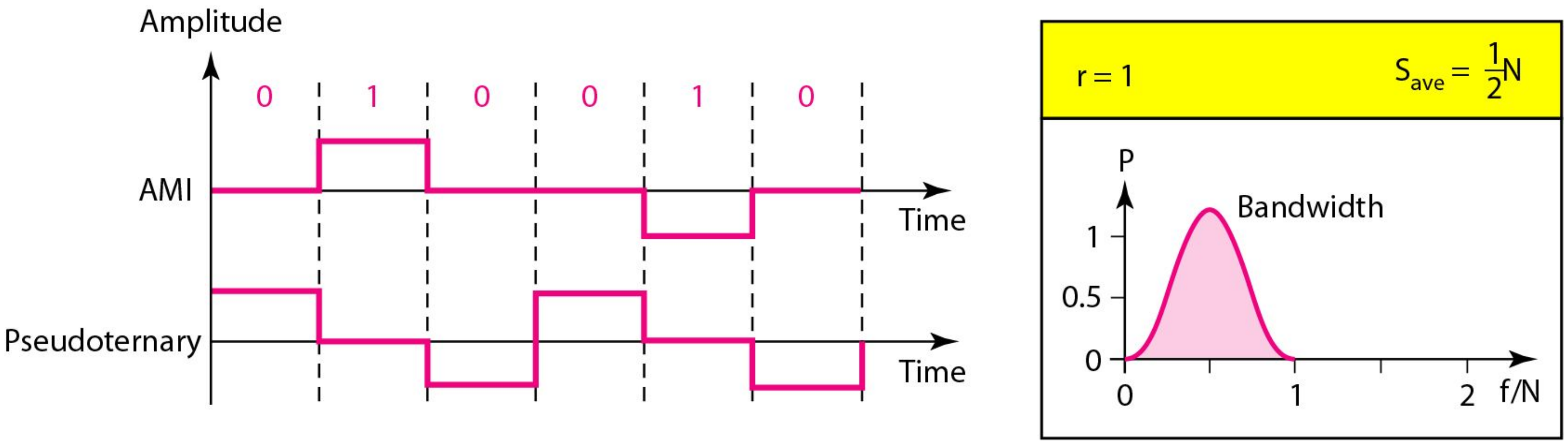
Note

The minimum bandwidth of Manchester and differential Manchester is 2 times that of NRZ. The is no DC component and no baseline wandering. None of these codes has error detection.

Bipolar - AMI and Pseudoternary

- Code uses 3 voltage levels: - +, 0, -, to represent the symbols (note not transitions to zero as in RZ).
- Voltage level for one symbol is at “0” and the other alternates between + & -.
- Bipolar Alternate Mark Inversion (AMI) - the “0” symbol is represented by zero voltage and the “1” symbol alternates between +V and -V.
- Pseudoternary is the reverse of AMI.

Figure 4.9 *Bipolar schemes: AMI and pseudoternary*



Bipolar C/Cs

- It is a better alternative to NRZ.
- Has no DC component or baseline wandering.
- Has no self synchronization because long runs of “0”s results in no signal transitions.
- No error detection.

Multilevel Schemes

- In these schemes we increase the number of data bits per symbol thereby increasing the bit rate.
- Since we are dealing with binary data we only have 2 types of data element a 1 or a 0.
- We can combine the 2 data elements into a pattern of “m” elements to create “2^m” symbols.
- If we have L signal levels, we can use “n” signal elements to create L^n signal elements.

Code C/Cs

- Now we have 2^m symbols and L^n signals.
- If $2^m > L^n$ then we cannot represent the data elements, we don't have enough signals.
- If $2^m = L^n$ then we have an exact mapping of one symbol on one signal.
- If $2^m < L^n$ then we have more signals than symbols and we can choose the signals that are more distinct to represent the symbols and therefore have better noise immunity and error detection as some signals are not valid.

Note

In *m*B*n*L schemes, a pattern of *m* data elements is encoded as a pattern of *n* signal elements in which $2^m \leq L^n$.

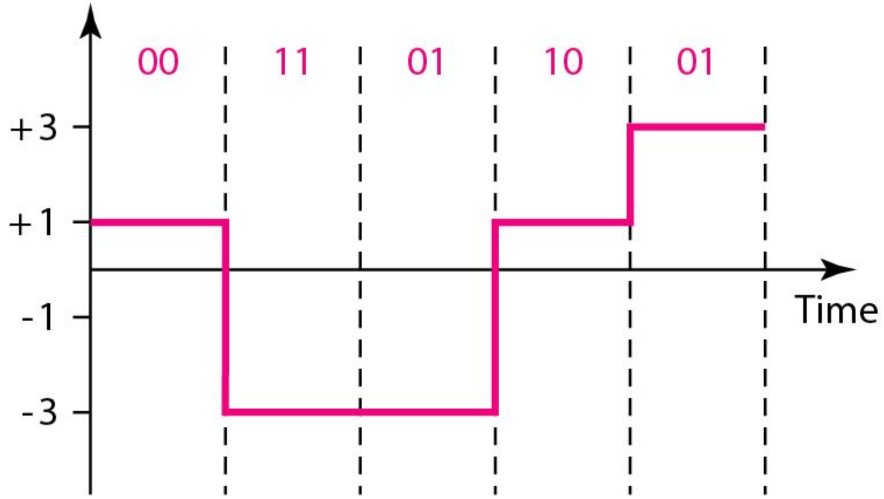
Representing Multilevel Codes

- We use the notation mBnL, where m is the length of the binary pattern, B represents binary data, n represents the length of the signal pattern and L the number of levels.
- L = B binary, L = T for 3 ternary, L = Q for 4 quaternary.

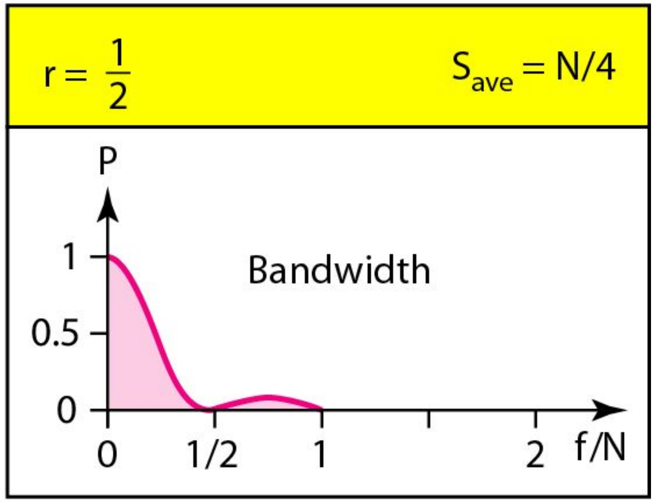
Figure 4.10 *Multilevel: 2B1Q scheme*

	Previous level: positive	Previous level: negative
Next bits	Next level	Next level
00	+1	-1
01	+3	-3
10	-1	+1
11	-3	+3

Transition table



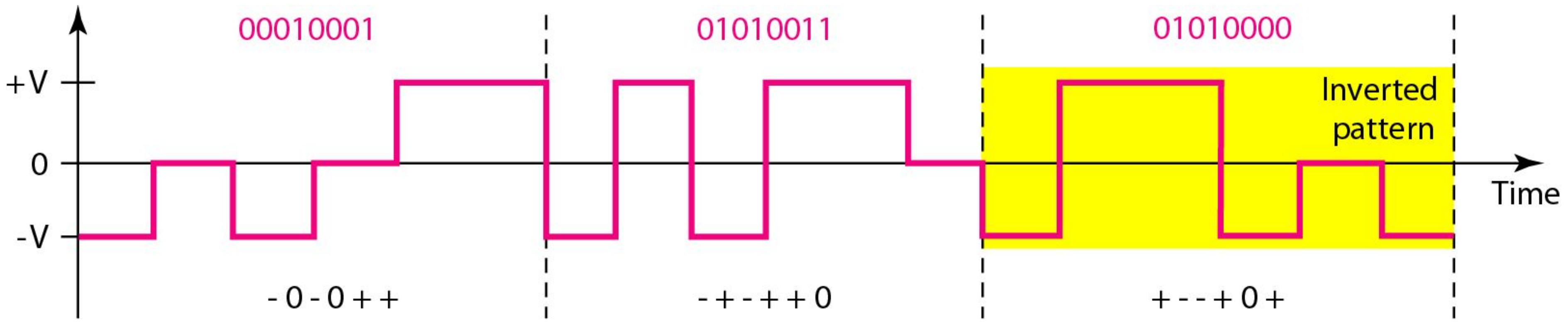
Assuming positive original level



Redundancy

- In the 2B1Q scheme we have no redundancy and we see that a DC component is present.
- If we use a code with redundancy we can decide to use only “0” or “+” weighted codes (more +’s than -’s in the signal element) and invert any code that would create a DC component. E.g. ‘+00++-’ -> ‘-00--+’
- Receiver will know when it receives a “-” weighted code that it should invert it as it doesn’t represent any valid symbol.

Figure 4.11 *Multilevel: 8B6T scheme*

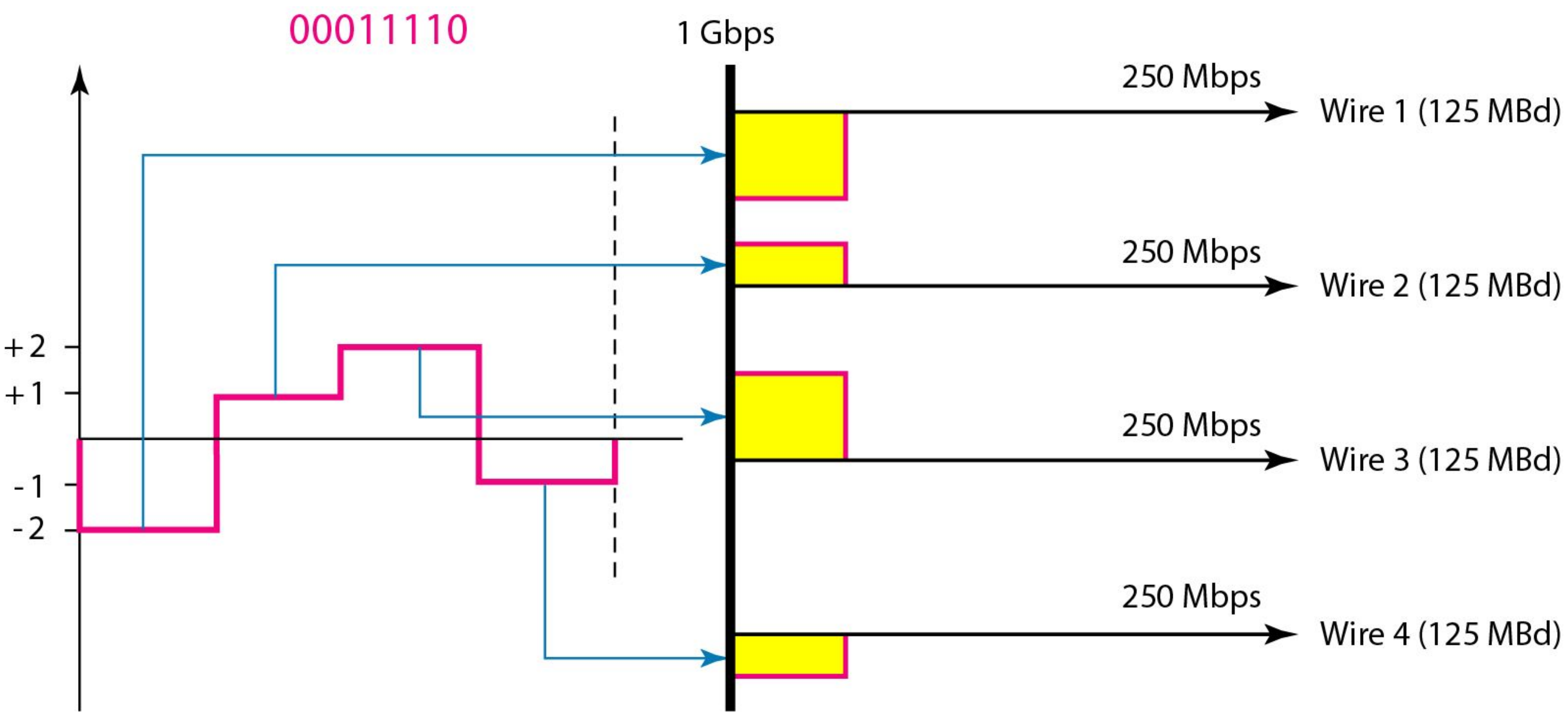


00000000	- + 0 0 - +
00000001	0 - + - + 0
00000010	0 - + 0 - +
00000011	0 - + + 0 -
00000100	- + 0 + 0 -
00000101	+ 0 - - + 0
00000110	+ 0 - 0 - +
00000111	+ 0 - + 0 -
00001000	- + 0 0 + -
00001001	0 - + + - 0
00001010	0 - + 0 + -
00001011	0 - + - 0 +
00001100	- + 0 - 0 +
00001101	+ 0 - + - 0
00001110	+ 0 - 0 + -
00001111	+ 0 - - 0 +
00010000	0 - - + 0 +
00010001	- 0 - 0 + +
00010010	- 0 - + 0 +
00010011	- 0 - + + 0
00010100	0 - - + + 0
00010101	- - 0 0 + +
00010110	- - 0 + 0 +
00010111	- - 0 + + 0
00011000	- + 0 - + 0
00011001	+ - 0 - + 0
00011010	- + + - + 0
00011011	+ 0 0 - + 0
00011100	+ 0 0 + - 0
00011101	- + + + - 0
00011110	+ - 0 + - 0
00011111	- + 0 + - 0
00100000	- + + - 0 0
00100001	+ 0 0 + - -
00100010	- + 0 - + +
00100011	+ - 0 - + +
00100100	+ - 0 + 0 0
00100101	- + 0 + 0 0
00100110	+ 0 0 - 0 0

Multilevel using multiple channels

- In some cases, we split the signal transmission up and distribute it over several links.
- The separate segments are transmitted simultaneously. This reduces the signalling rate per link -> lower bandwidth.
- This requires all bits for a code to be stored.
- xD: means that we use ‘x’ links
- YYYz: We use ‘z’ levels of modulation where YYY represents the type of modulation (e.g. pulse ampl. mod. PAM).
- Codes are represented as: xD-YYYz

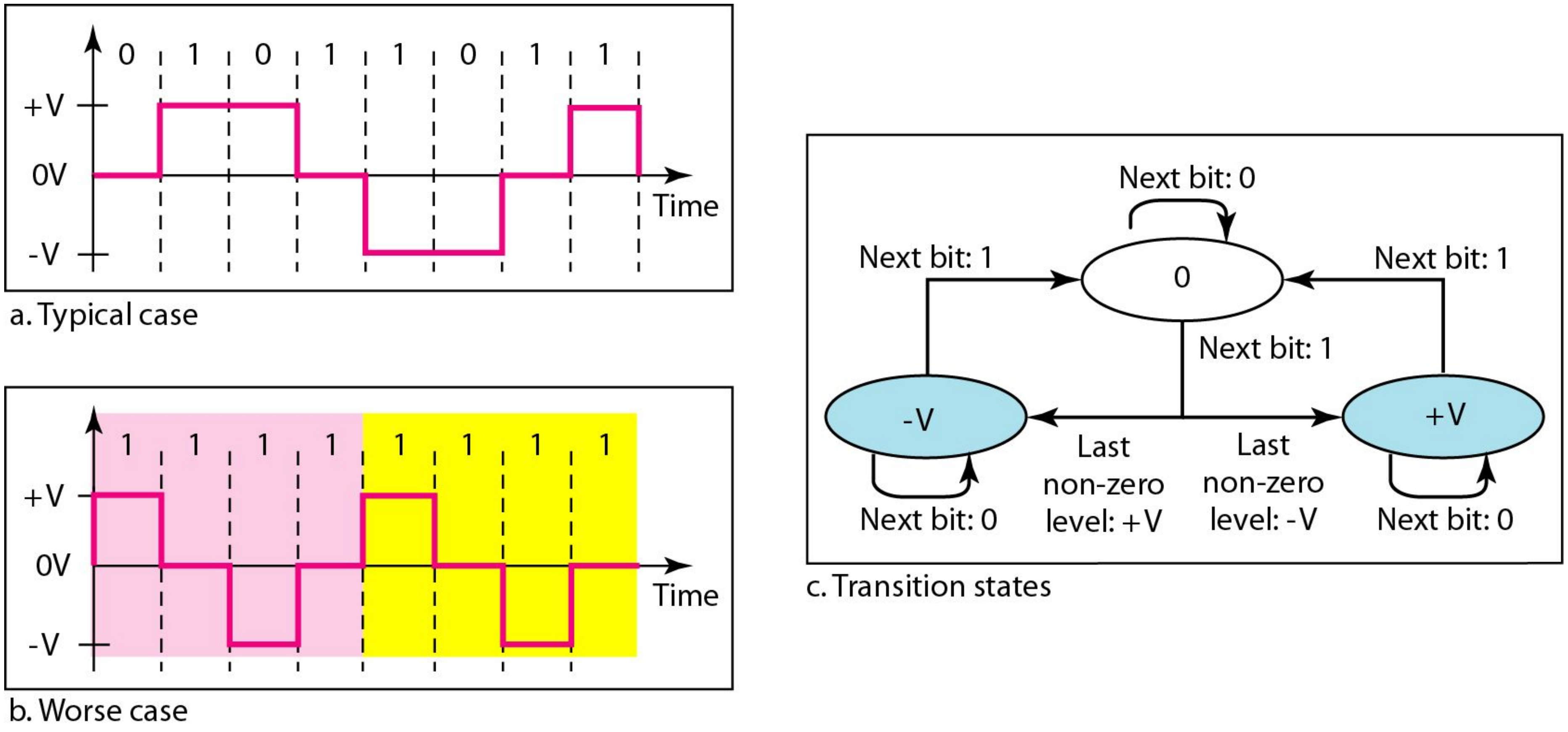
Figure 4.12 *Multilevel: 4D-PAM5 scheme*



Multitransition Coding

- Because of synchronization requirements we force transitions. This can result in very high bandwidth requirements -> more transitions than are bits (e.g. mid bit transition with inversion).
- Codes can be created that are differential at the bit level forcing transitions at bit boundaries. This results in a bandwidth requirement that is equivalent to the bit rate.
- In some instances, the bandwidth requirement may even be lower, due to repetitive patterns resulting in a periodic signal.

Figure 4.13 *Multitransition: MLT-3 scheme*



MLT-3

- Signal rate is same as NRZ-I
- But because of the resulting bit pattern, we have a periodic signal for worst case bit pattern: 1111
- This can be approximated as an analog signal a frequency 1/4 the bit rate!

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
Unipolar	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
Multilevel	2B1Q	$B = N/4$	No self-synchronization for long same double bits
	8B6T	$B = 3N/4$	Self-synchronization, no DC
	4D-PAM5	$B = N/8$	Self-synchronization, no DC
Multiline	MLT-3	$B = N/3$	No self-synchronization for long 0s

Block Coding

- For a code to be capable of error detection, we need to add redundancy, i.e., extra bits to the data bits.
- Synchronization also requires redundancy - transitions are important in the signal flow and must occur frequently.
- Block coding is done in three steps: division, substitution and combination.
- It is distinguished from multilevel coding by use of the slash - xB/yB.
- The resulting bit stream prevents certain bit combinations that when used with line encoding would result in DC components or poor sync. quality.

Note

Block coding is normally referred to as *m*B/*n*B coding;
it replaces each *m*-bit group with an *n*-bit group.

Figure 4.14 *Block coding concept*

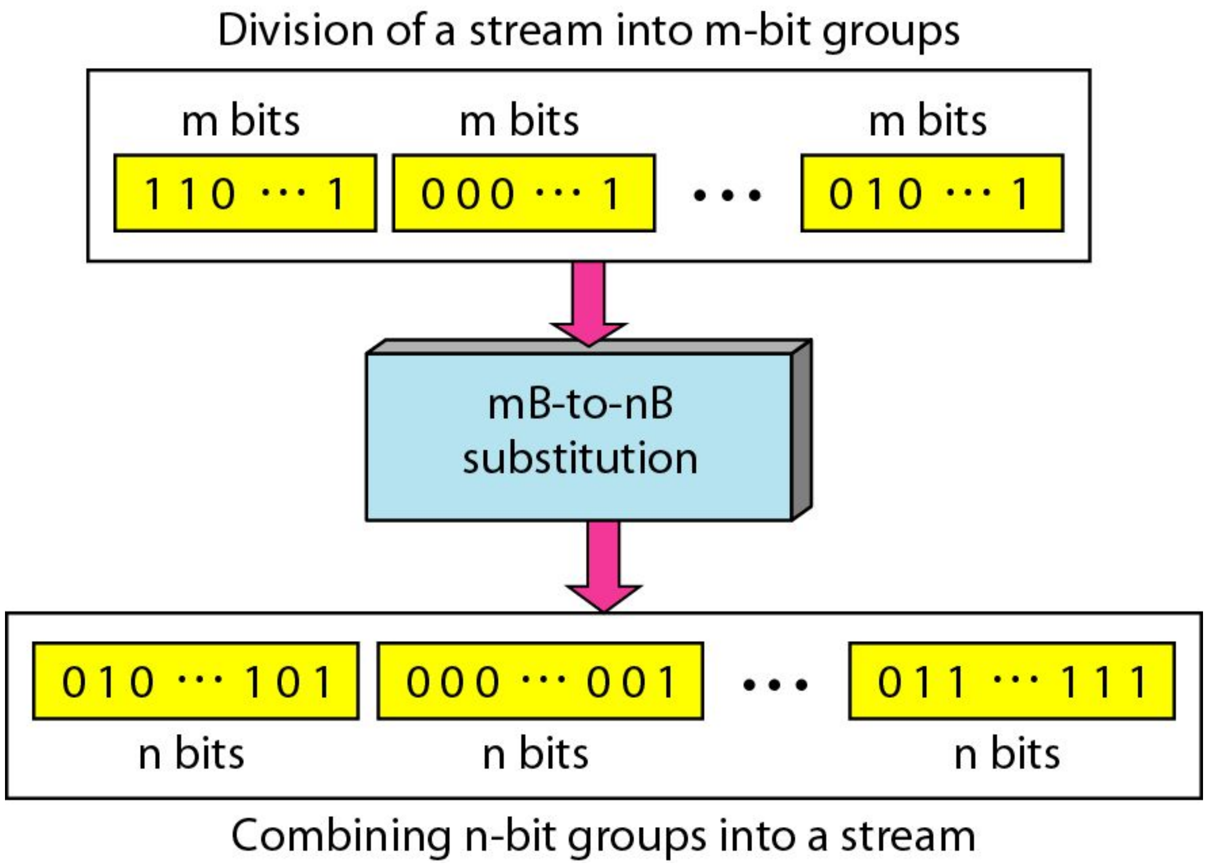
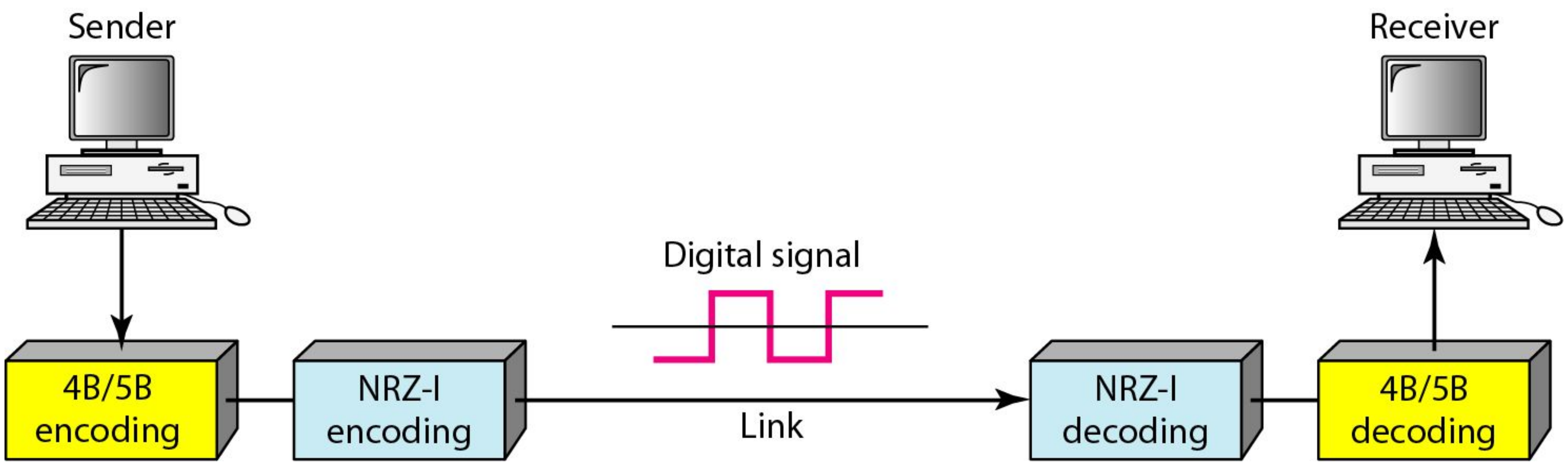


Figure 4.15 *Using block coding 4B/5B with NRZ-I line coding scheme*



OBJECTIVE:

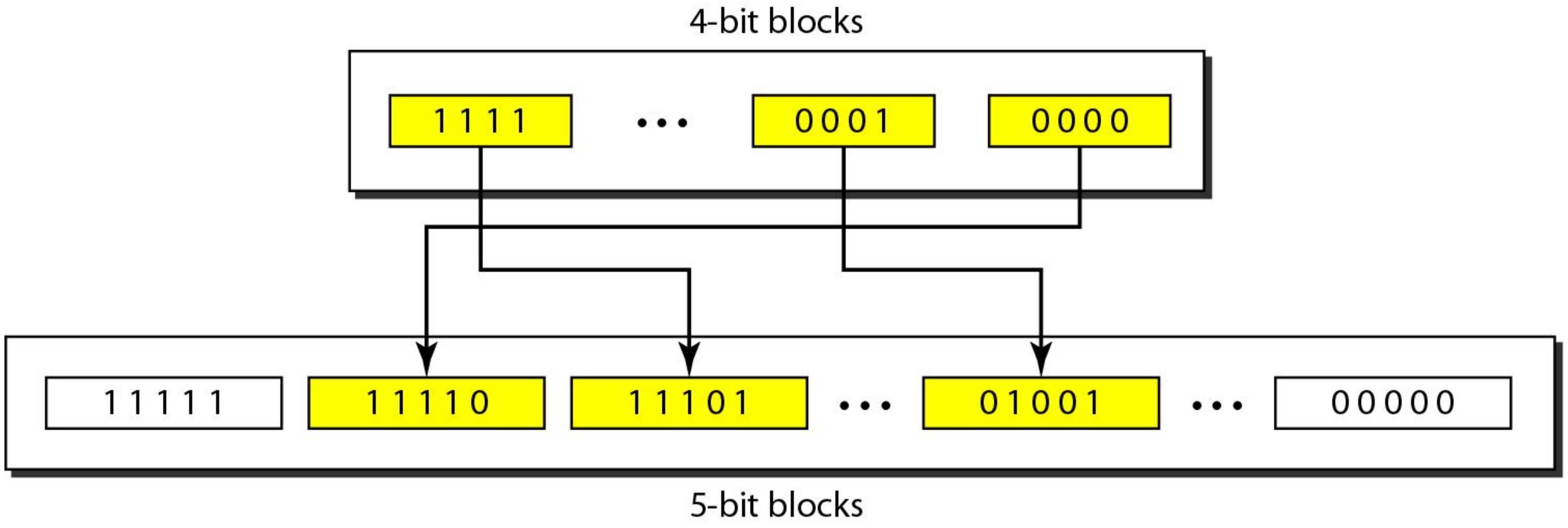
For error detection and synchronization, redundancy is introduced

NRZ-I – Sequence of 0s are cut with the rule with leading zeros are not more than one and trailing zeros not more than two

Table 4.2 *4B/5B mapping codes*

Data Sequence	*Encoded Sequence*	*Control Sequence*	*Encoded Sequence*
0000	11110	Q (Quiet)	00000
0001	01001	I (Idle)	11111
0010	10100	H (Halt)	00100
0011	10101	J (Start delimiter)	11000
0100	01010	K (Start delimiter)	10001
0101	01011	T (End delimiter)	01101
0110	01110	S (Set)	11001
0111	01111	R (Reset)	00111
1000	10010		
1001	10011		
1010	10110		
1011	10111		
1100	11010		
1101	11011		
1110	11100		
1111	11101		

Figure 4.16 *Substitution in 4B/5B block coding*



Redundancy

- A 4 bit data word can have 24 combinations.
- A 5 bit word can have 25=32 combinations.
- We therefore have 32 - 26 = 16 extra words.
- Some of the extra words are used for control/signalling purposes.

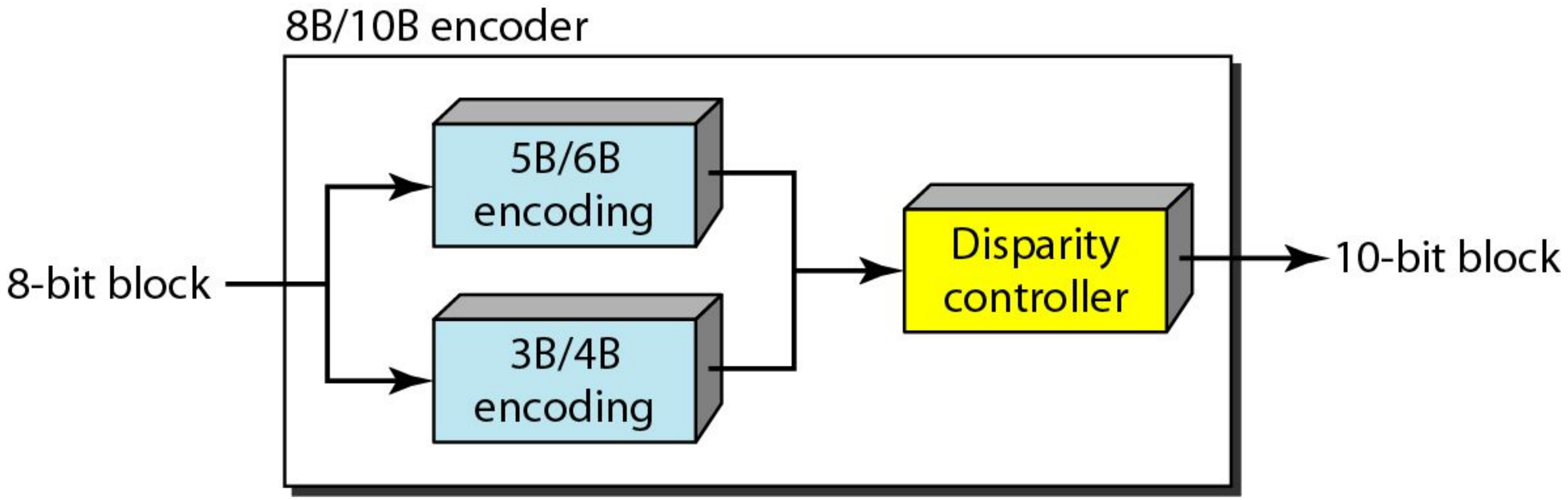
Example 4.5

We need to send data at a 1-Mbps rate. What is the minimum required bandwidth, using a combination of 4B/5B and NRZ-I or Manchester coding?

Solution

First 4B/5B block coding increases the bit rate to 1.25 Mbps. The minimum bandwidth using NRZ-I is N/2 or 625 kHz. The Manchester scheme needs a minimum bandwidth of 1.25 MHz. The first choice needs a lower bandwidth, but has a DC component problem; the second choice needs a higher bandwidth, but does not have a DC component problem.

Figure 4.17 *8B/10B block encoding*



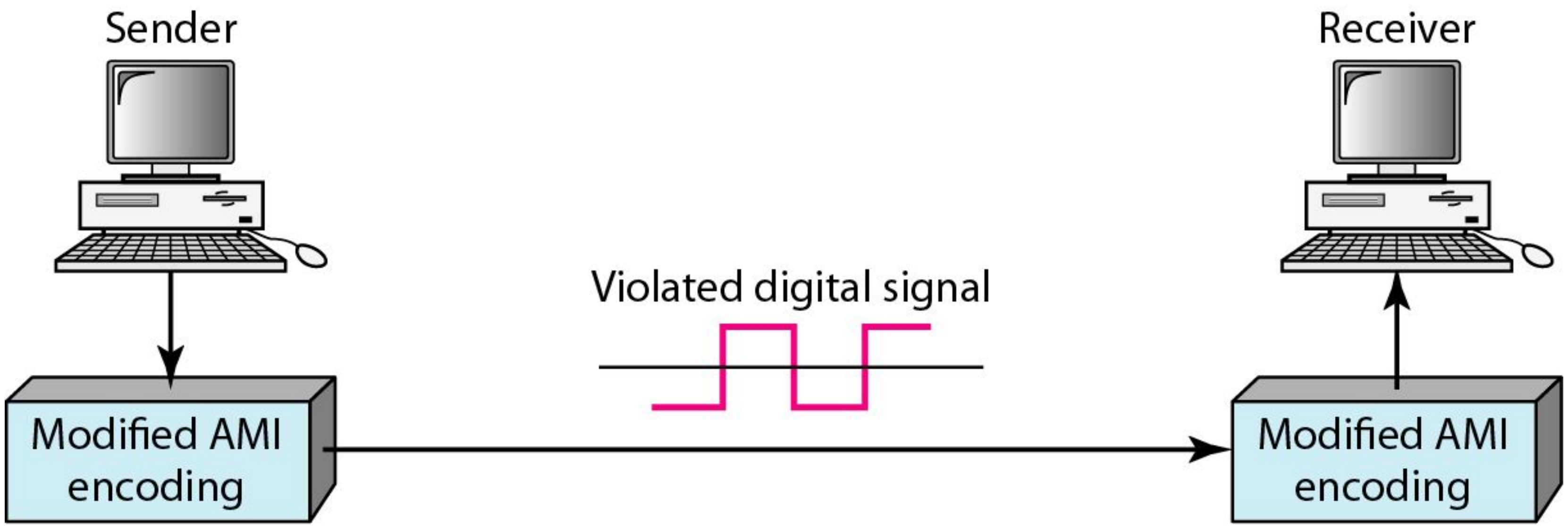
More bits - better error detection

- The 8B10B block code adds more redundant bits and can thereby choose code words that would prevent a long run of a voltage level that would cause DC components.

Scrambling

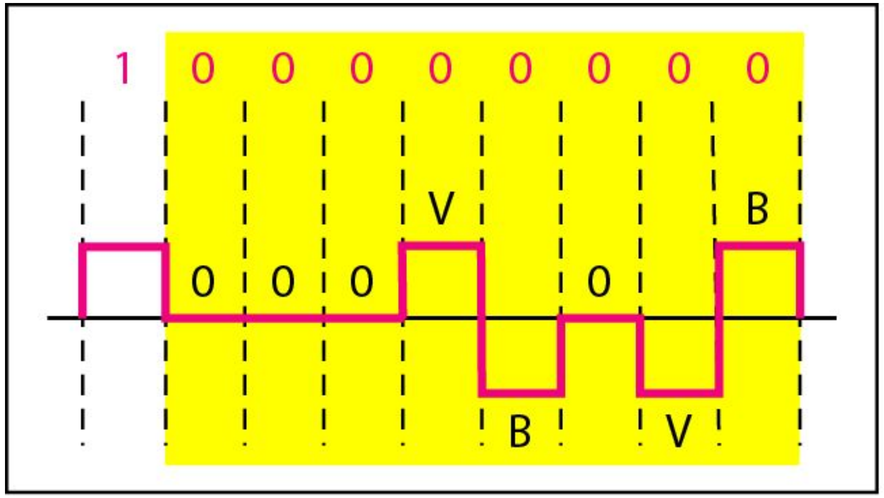
- The best code is one that does not increase the bandwidth for synchronization and has no DC components.
- Scrambling is a technique used to create a sequence of bits that has the required c/c’s for transmission - self clocking, no low frequencies, no wide bandwidth.
- It is implemented at the same time as encoding, the bit stream is created on the fly.
- It replaces ‘unfriendly’ runs of bits with a violation code that is easy to recognize and removes the unfriendly c/c.

Figure 4.18 *AMI used with scrambling*

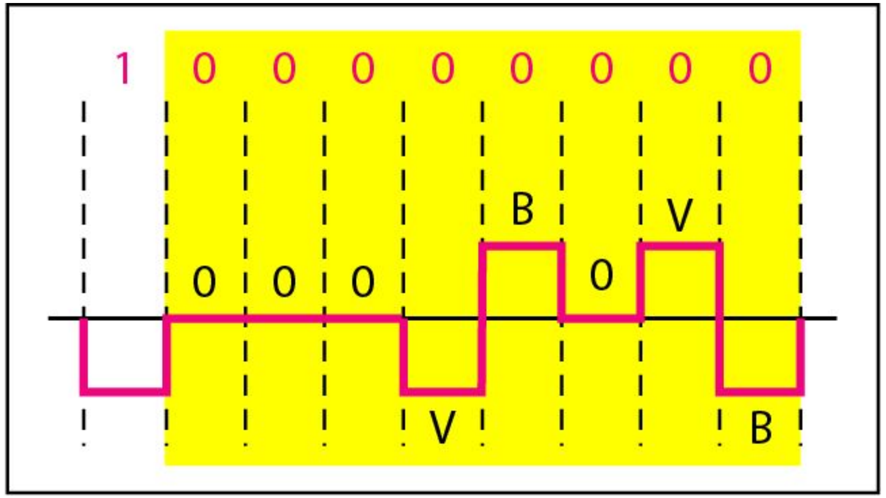


For example: B8ZS substitutes eight consecutive zeros with 000VB0VB.
The V stands for violation, it violates the line encoding rule
B stands for bipolar, it implements the bipolar line encoding rule

Figure 4.19 *Two cases of B8ZS scrambling technique*



a. Previous level is positive.



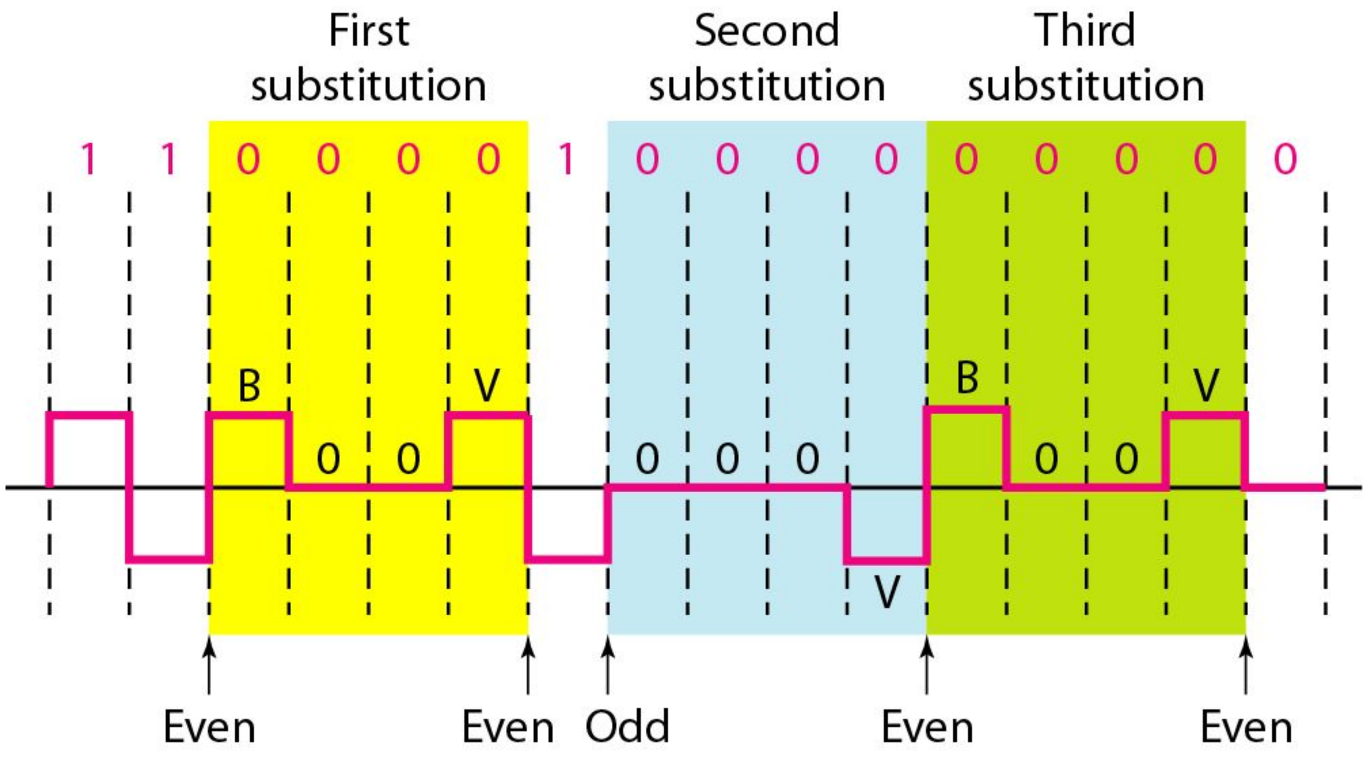
b. Previous level is negative.

HDB3 substitutes four consecutive zeros with 000V or B00V depending on the number of nonzero pulses after the last substitution.

If # of non zero pulses is even the substitution is B00V to make total # of non zero pulse even.

If # of non zero pulses is odd the substitution is 000V to make total # of non zero pulses even.

Figure 4.20 *Different situations in HDB3 scrambling technique*



4-2 ANALOG-TO-DIGITAL CONVERSION

A digital signal is superior to an analog signal because it is more robust to noise and can easily be recovered, corrected and amplified. For this reason, the tendency today is to change an analog signal to digital data. In this section we describe two techniques, pulse code modulation and delta modulation.

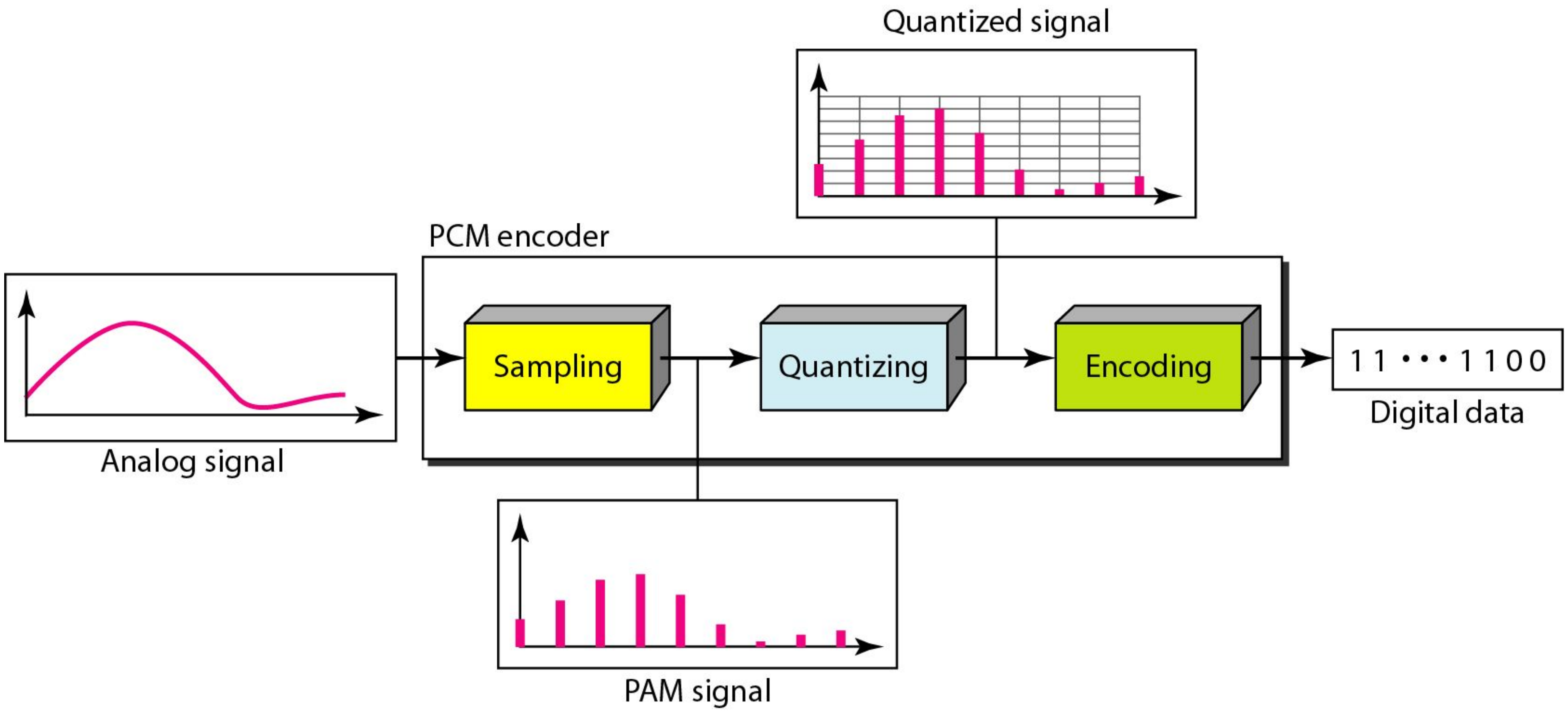
Topics discussed in this section:

- **Pulse Code Modulation (PCM)**
- **Delta Modulation (DM)**

PCM

- PCM consists of three steps to digitize an analog signal:
    1. Sampling
    2. Quantization
    3. Binary encoding
- Before we sample, we have to filter the signal to limit the maximum frequency of the signal as it affects the sampling rate.
- Filtering should ensure that we do not distort the signal, ie remove high frequency components that affect the signal shape.

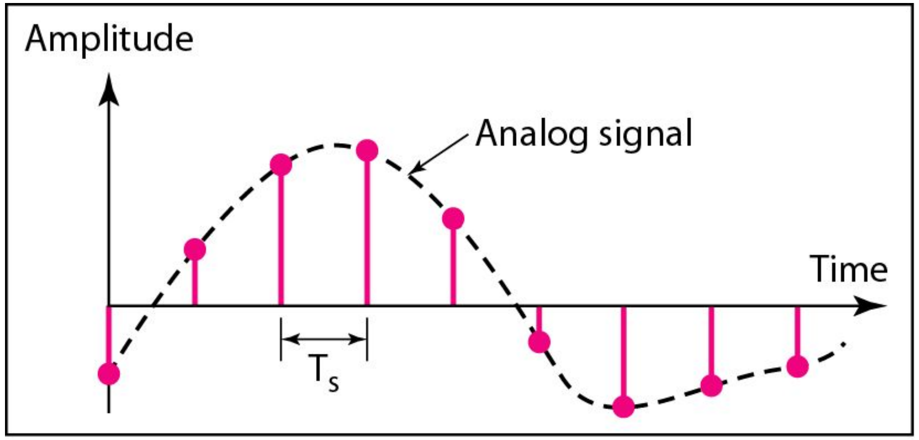
Figure 4.21 *Components of PCM encoder*



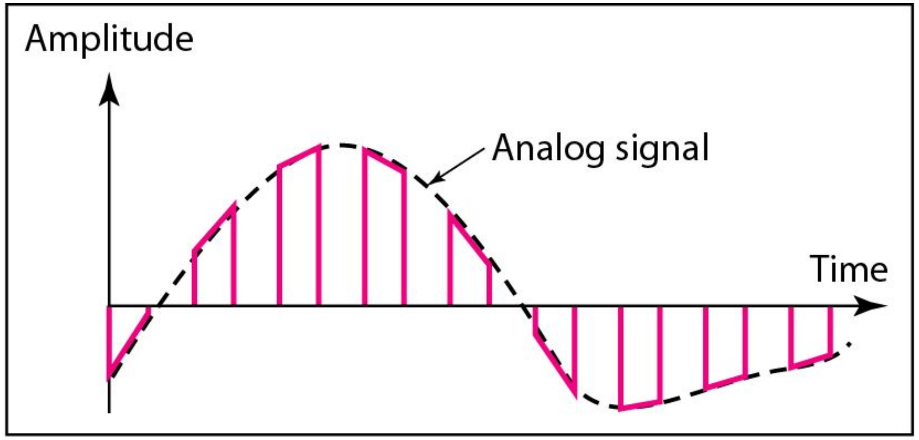
Sampling

- Analog signal is sampled every T_S secs.
- T_s is referred to as the sampling interval.
- $f_s = 1/T_s$ is called the sampling rate or sampling frequency.
- There are 3 sampling methods:
  - Ideal - an impulse at each sampling instant
  - Natural - a pulse of short width with varying amplitude
  - Flattop - sample and hold, like natural but with single amplitude value
- The process is referred to as pulse amplitude modulation PAM and the outcome is a signal with analog (non integer) values

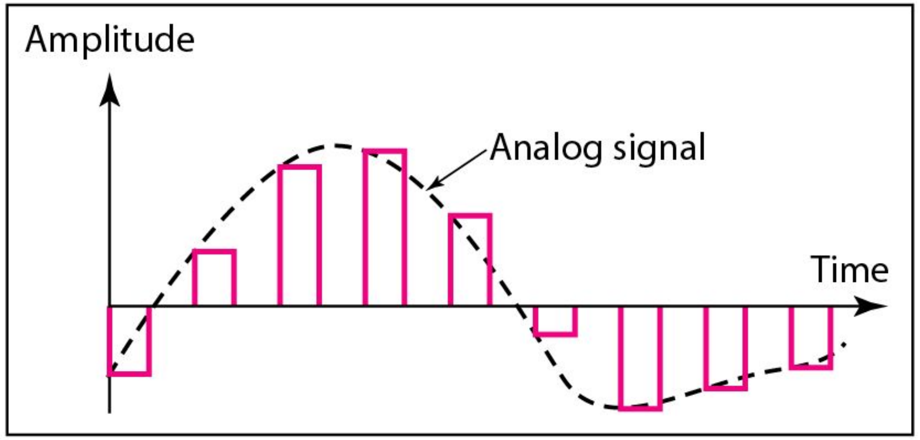
Figure 4.22 *Three different sampling methods for PCM*



a. Ideal sampling



b. Natural sampling

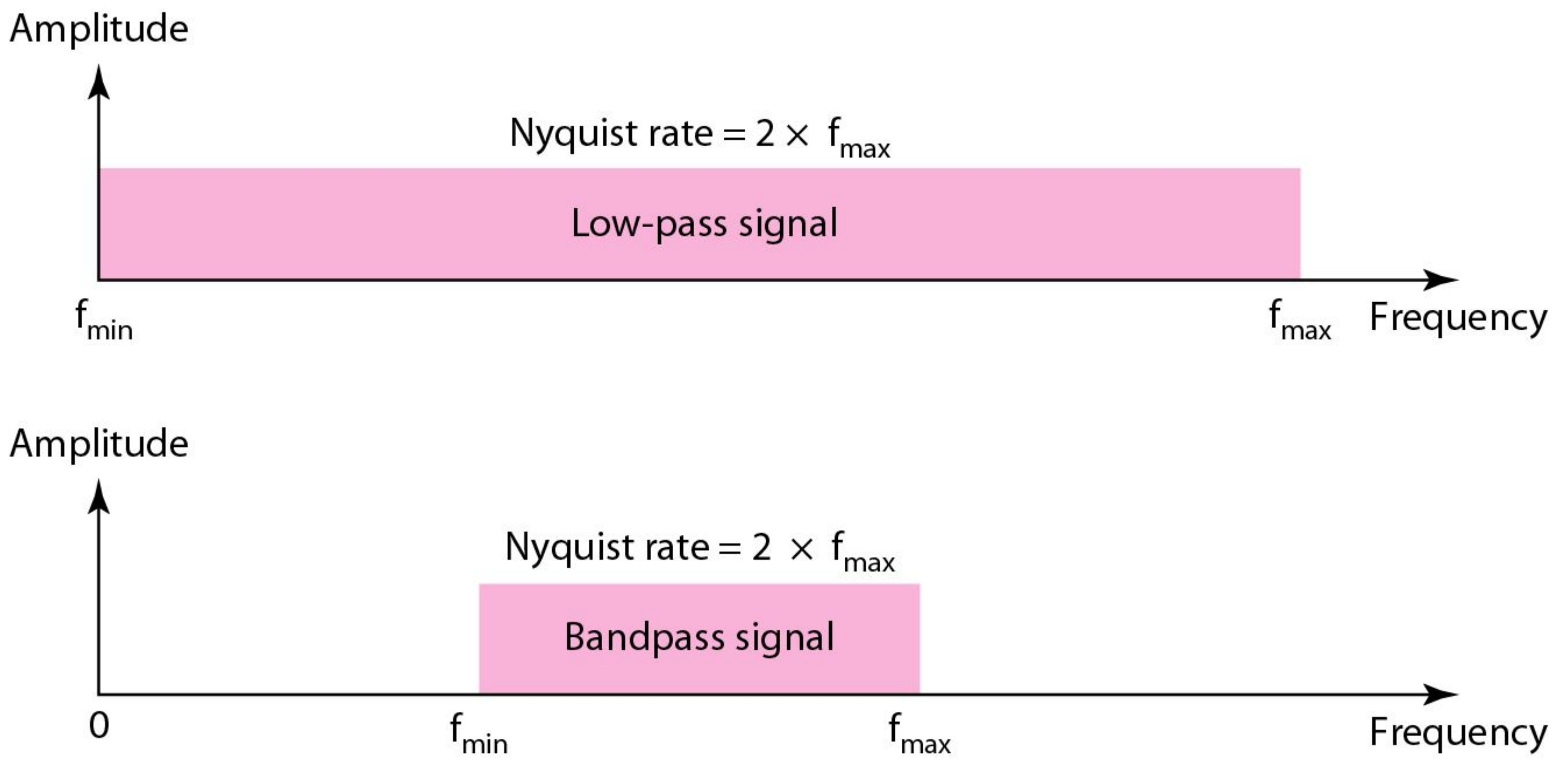


c. Flat-top sampling

Note

According to the Nyquist theorem, the sampling rate must be at least 2 times the highest frequency contained in the signal.

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

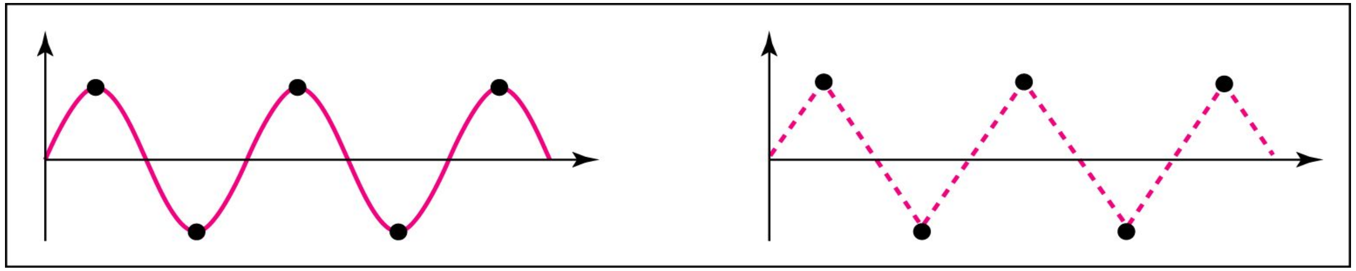


Example 4.6

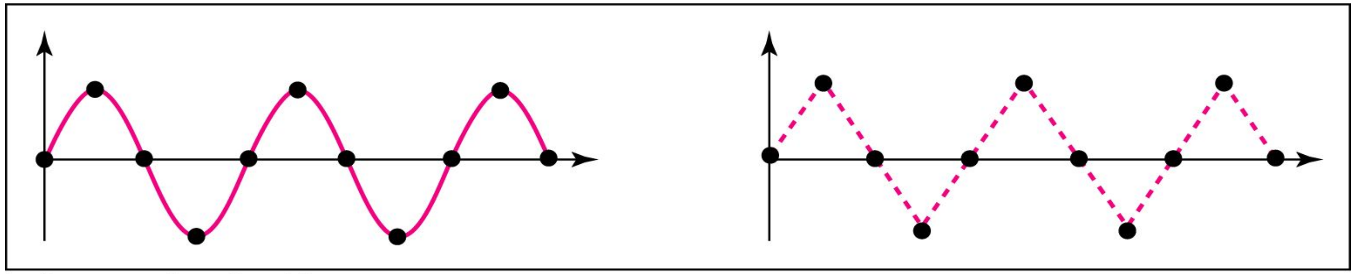
let us sample a simple sine wave at three sampling rates: $f_s = 4f$ (2 times the Nyquist rate), $f_s = 2f$ (Nyquist rate), and $f_s = f$ (one-half the Nyquist rate). Figure 4.24 shows the sampling and the subsequent recovery of the signal.

It can be seen that sampling at the Nyquist rate can create a good approximation of the original sine wave (part a). Oversampling in part b can also create the same approximation, but it is redundant and unnecessary. Sampling below the Nyquist rate (part c) does not produce a signal that looks like the original sine wave.

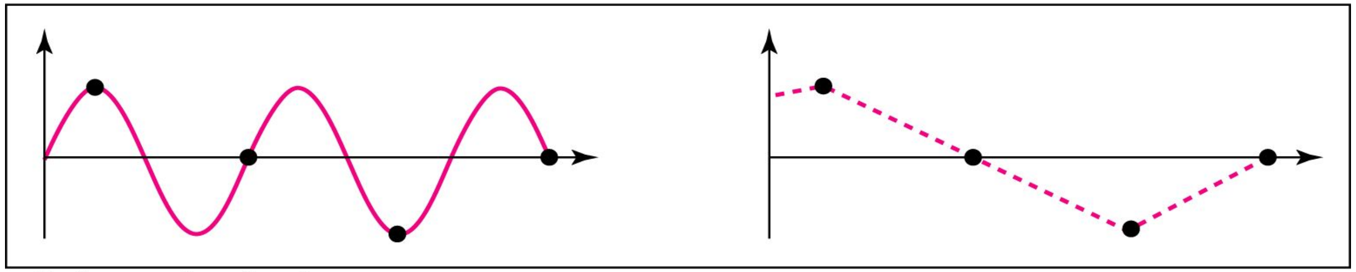
Figure 4.24 *Recovery of a sampled sine wave for different sampling rates*



a. Nyquist rate sampling: $f_s = 2\ f$



b. Oversampling: $f_s = 4\ f$

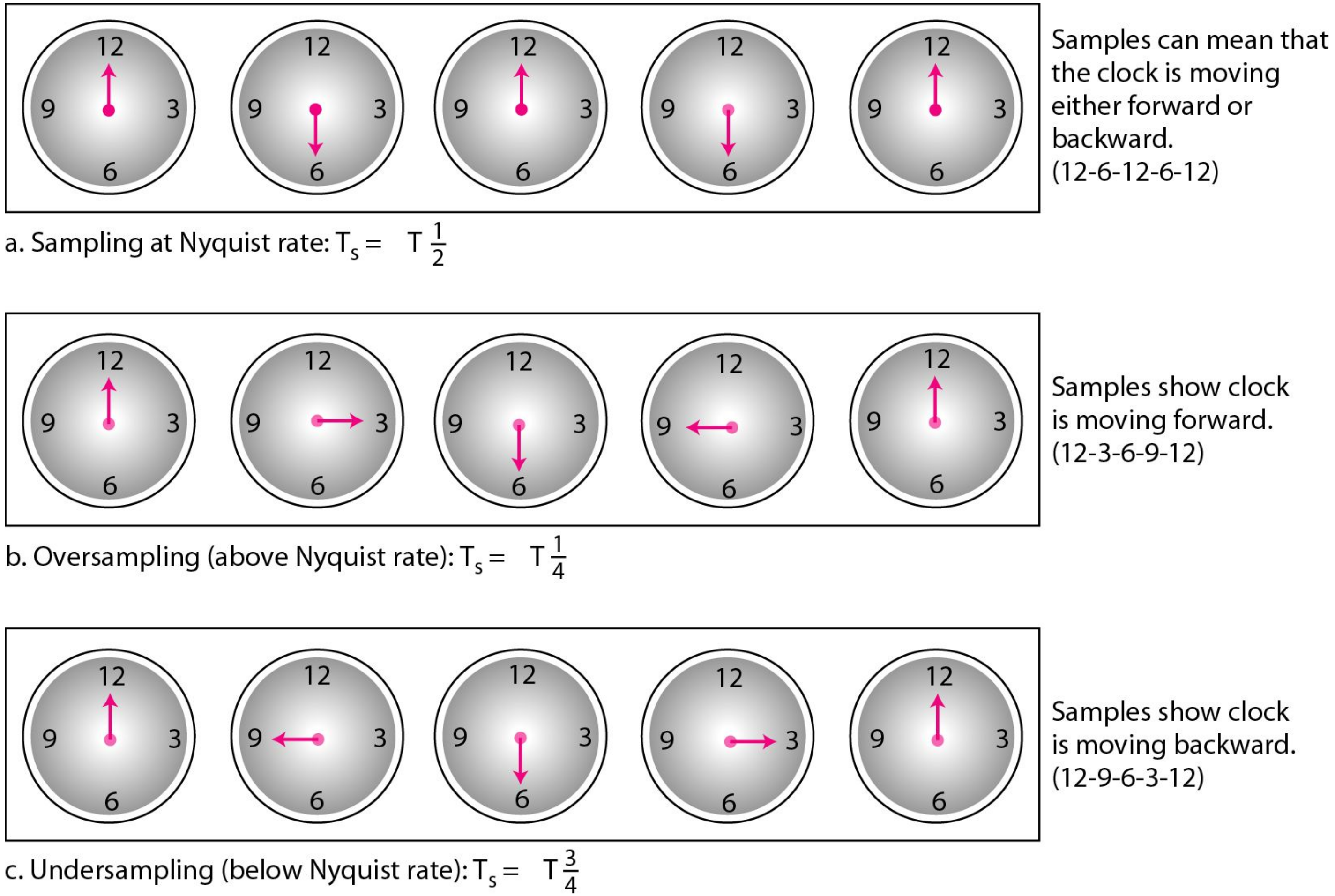


c. Undersampling: $f_s =\ f$

Example 4.7

Consider the revolution of a hand of a clock. The second hand of a clock has a period of 60 s. According to the Nyquist theorem, we need to sample the hand every 30 s ($T_s = T$ or $f_s = 2f$). In Figure 4.25a, the sample points, in order, are 12, 6, 12, 6, 12, and 6. The receiver of the samples cannot tell if the clock is moving forward or backward. In part b, we sample at double the Nyquist rate (every 15 s). The sample points are 12, 3, 6, 9, and 12. The clock is moving forward. In part c, we sample below the Nyquist rate ($T_s = T$ or $f_s = f$). The sample points are 12, 9, 6, 3, and 12. Although the clock is moving forward, the receiver thinks that the clock is moving backward.

Figure 4.25 *Sampling of a clock with only one hand*



Example 4.8

An example related to Example 4.7 is the seemingly backward rotation of the wheels of a forward-moving car in a movie. This can be explained by under-sampling. A movie is filmed at 24 frames per second. If a wheel is rotating more than 12 times per second, the under-sampling creates the impression of a backward rotation.

Example 4.9

Telephone companies digitize voice by assuming a maximum frequency of 4000 Hz. The sampling rate therefore is 8000 samples per second.

Example 4.10

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 4.11

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

Solution

We cannot find the minimum sampling rate in this case because we do not know where the bandwidth starts or ends. We do not know the maximum frequency in the signal.

Quantization

- Sampling results in a series of pulses of varying amplitude values ranging between two limits: a min and a max.
- The amplitude values are infinite between the two limits.
- We need to map the *infinite* amplitude values onto a finite set of known values.
- This is achieved by dividing the distance between min and max into L zones, each of height Δ.

$$\Delta = (\text{max} - \text{min})/L$$

Quantization Levels

- The midpoint of each zone is assigned a value from 0 to L-1 (resulting in L values)
- Each sample falling in a zone is then approximated to the value of the midpoint.

Quantization Zones

- Assume we have a voltage signal with amplitutes V_{min}=-20V and V_{max}=+20V.
- We want to use L=8 quantization levels.
- Zone width Δ = (20 - -20)/8 = 5
- The 8 zones are: -20 to -15, -15 to -10, -10 to -5, -5 to 0, 0 to +5, +5 to +10, +10 to +15, +15 to +20
- The midpoints are: -17.5, -12.5, -7.5, -2.5, 2.5, 7.5, 12.5, 17.5

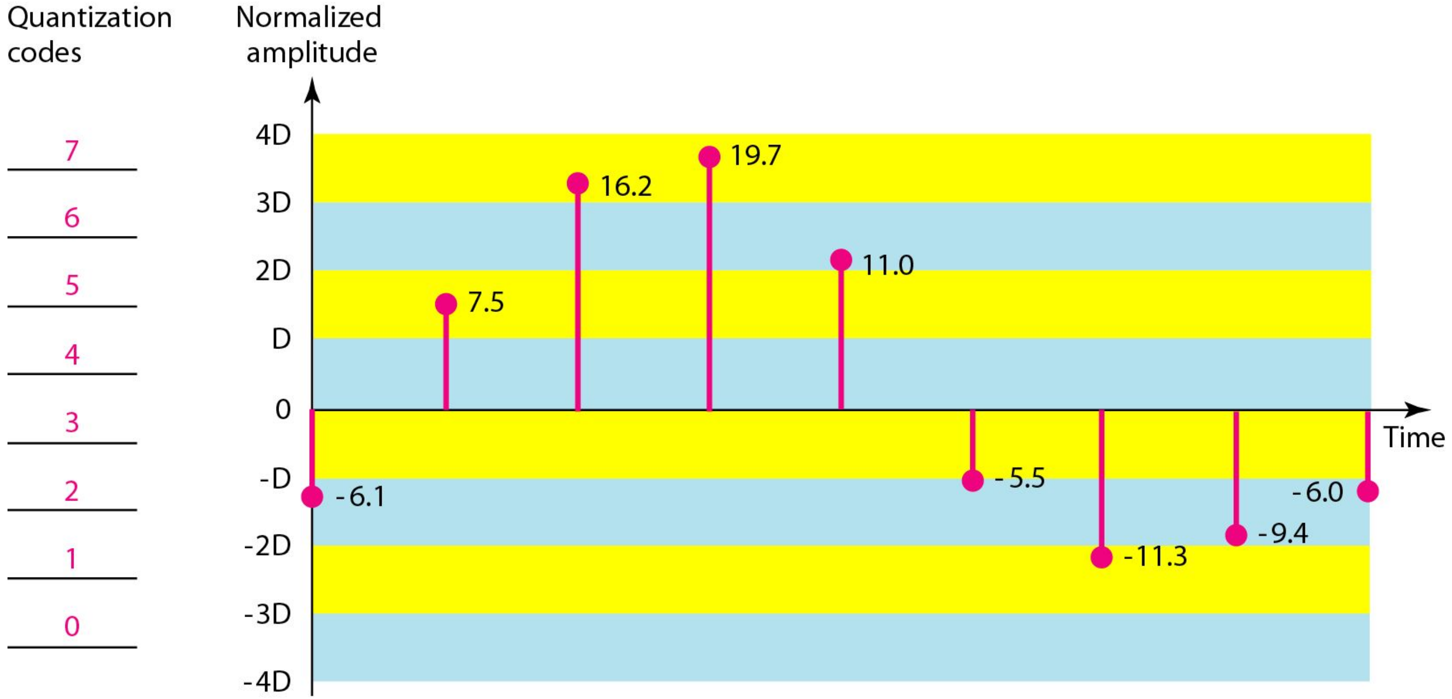
Assigning Codes to Zones

- Each zone is then assigned a binary code.
- The number of bits required to encode the zones, or the number of bits per sample as it is commonly referred to, is obtained as follows:

$$n_b = \log_2 L$$

- Given our example, $n_b = 3$
- The 8 zone (or level) codes are therefore: 000, 001, 010, 011, 100, 101, 110, and 111
- Assigning codes to zones:
  - 000 will refer to zone -20 to -15
  - 001 to zone -15 to -10, etc.

Figure 4.26 *Quantization and encoding of a sampled signal*



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

Quantization Error

- When a signal is quantized, we introduce an error - the coded signal is an approximation of the actual amplitude value.
- The difference between actual and coded value (midpoint) is referred to as the quantization error.
- The more zones, the smaller Δ which results in smaller errors.
- BUT, the more zones the more bits required to encode the samples -> higher bit rate

Quantization Error and SNR

- Signals with lower amplitude values will suffer more from quantization error as the error range: $\Delta/2$, is fixed for all signal levels.

$$SNR_{db} = 0.02n_b + 1.76 \text{ dB}$$

What is the SNR_{db} in the previous example Fig 4.26?

Solution:

$$SNR_{db} = 0.02n_b + 1.76 \text{ dB}$$

Wkt $n_b = 3$

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

Solution:

$$SNR_{db} = 0.02n_b + 1.76 \text{ dB}$$

What is n_b

Encoding

- The bit rate of a PCM signal can be calculated form the number of bits per sample x the sampling rate

$$\text{Bit rate} = n_b \times f_s$$

- The bandwidth required to transmit this signal depends on the type of line encoding used. Refer to previous section for discussion and formulas.
- A digitized signal will always need more bandwidth than the original analog signal. Price we pay for robustness and other features of digital transmission.

Example 4.14

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

Solution

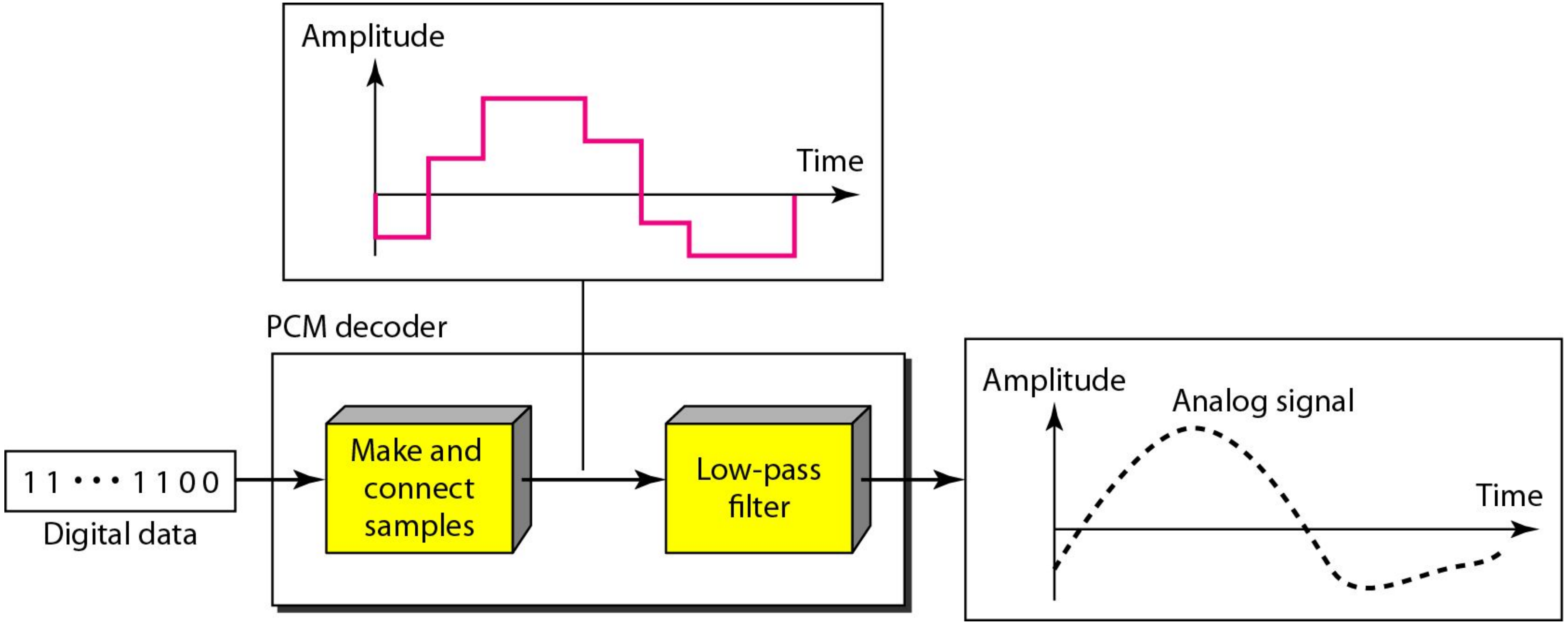
The human voice normally contains frequencies from 0 to 4000 Hz. So the sampling rate and bit rate are calculated as follows:

$$\text{Sampling rate} = 4000 \times 2 = 8000 \text{ samples/s}$$
$$\text{Bit rate} = 8000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}$$

PCM Decoder

- To recover an analog signal from a digitized signal we follow the following steps:
  - We use a hold circuit that holds the amplitude value of a pulse till the next pulse arrives.
  - We pass this signal through a low pass filter with a cutoff frequency that is equal to the highest frequency in the pre-sampled signal.
- The higher the value of L, the less distorted a signal is recovered.

Figure 4.27 *Components of a PCM decoder*

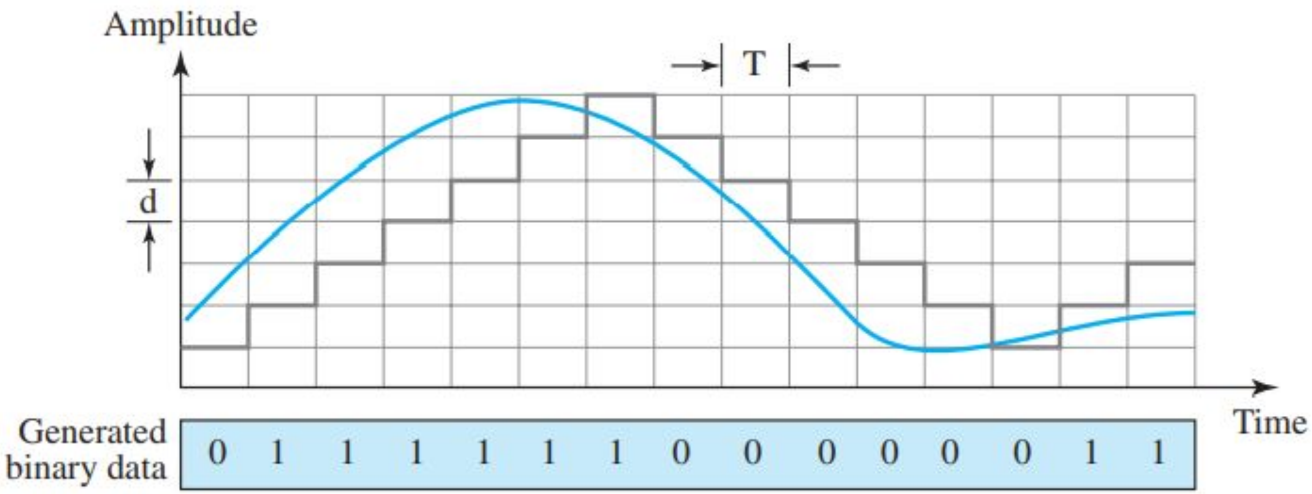


Example 4.15

We have a low-pass analog signal of 4 kHz. If we send the analog signal, we need a channel with a minimum bandwidth of 4 kHz. If we digitize the signal and send 8 bits per sample, we need a channel with a minimum bandwidth of 8 × 4 kHz = 32 kHz.

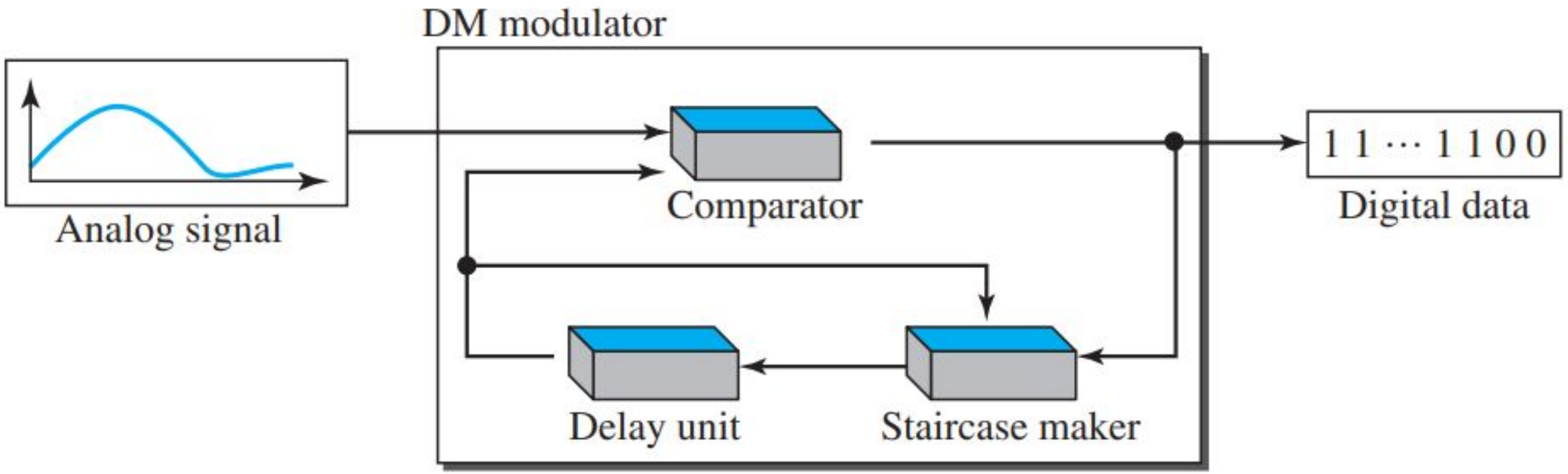
Delta Modulation

Figure 4.28 *The process of delta modulation*



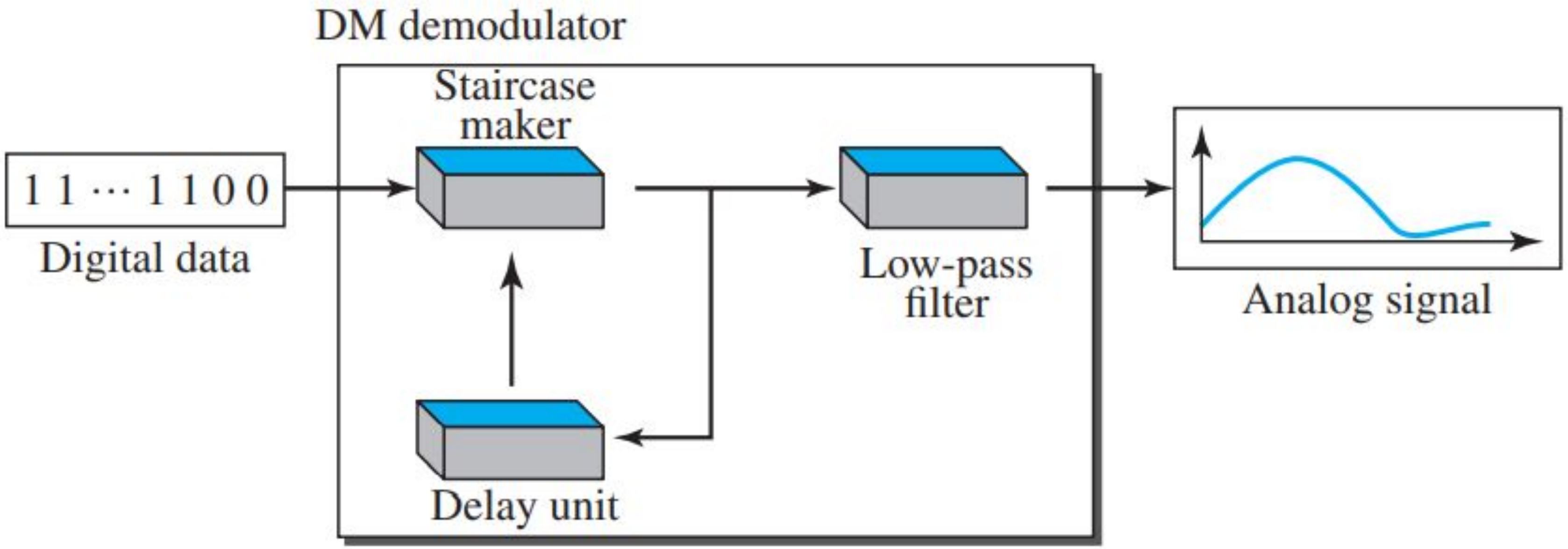
Delta Modulation

Figure 4.29 *Delta modulation components*



Delta Demodulation

Figure 4.30 *Delta demodulation components*



4-3 TRANSMISSION MODES

The transmission of binary data across a link can be accomplished in either parallel or serial mode. In parallel mode, multiple bits are sent with each clock tick. In serial mode, 1 bit is sent with each clock tick. While there is only one way to send parallel data, there are three subclasses of serial transmission: asynchronous, synchronous, and isochronous.

Topics discussed in this section:

- **Parallel Transmission**
- **Serial Transmission**

Figure 4.31 *Data transmission and modes*

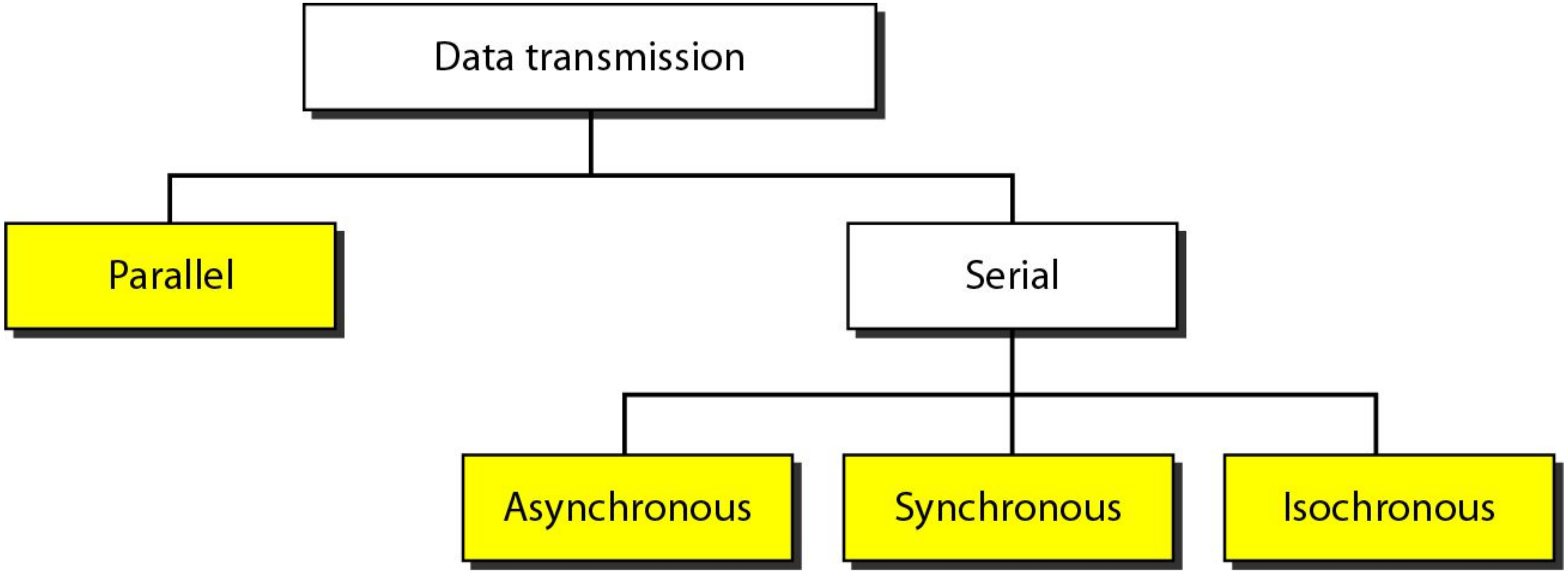


Figure 4.32 *Parallel transmission*

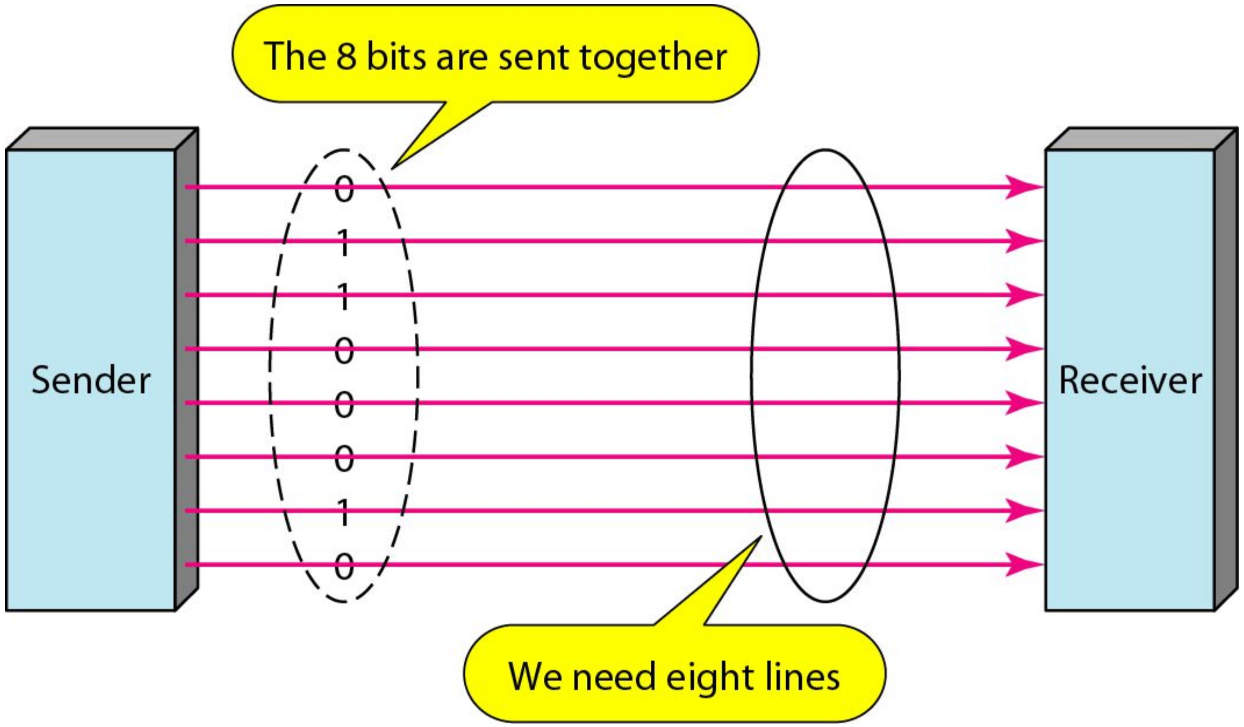
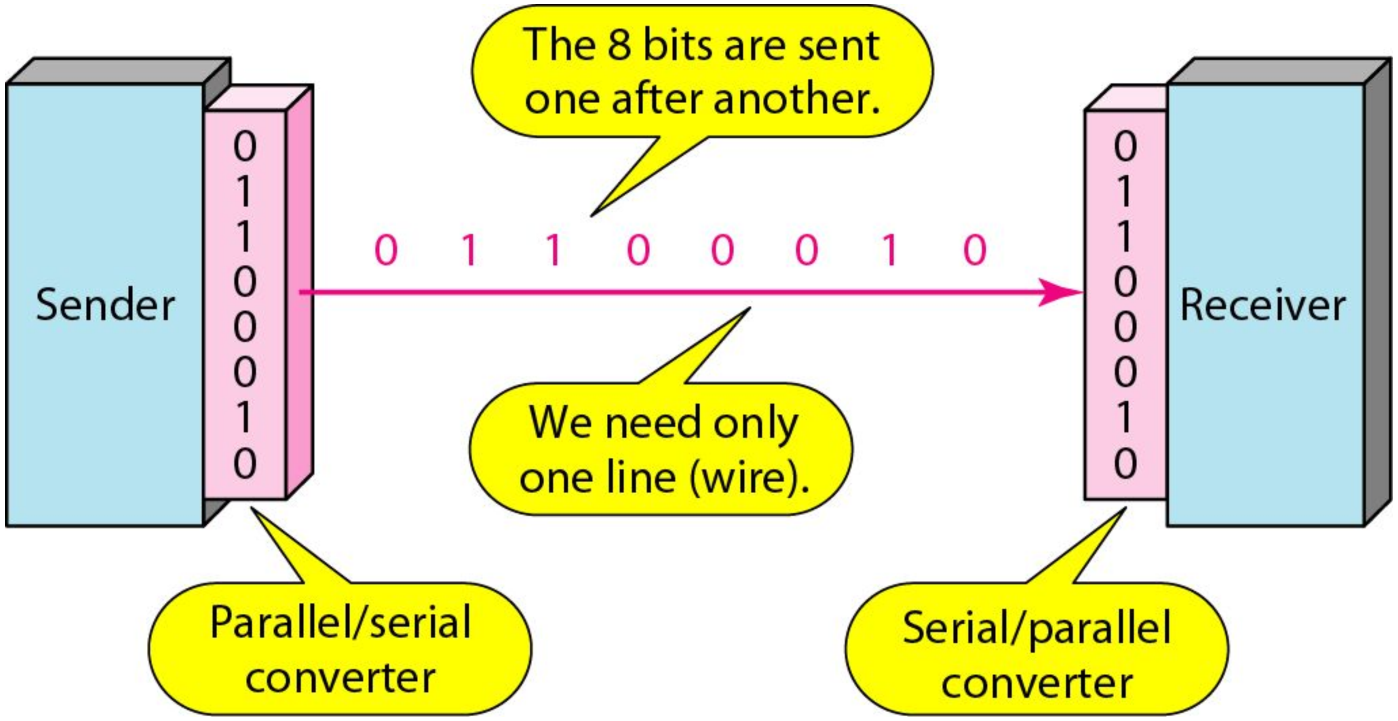


Figure 4.33 *Serial transmission*



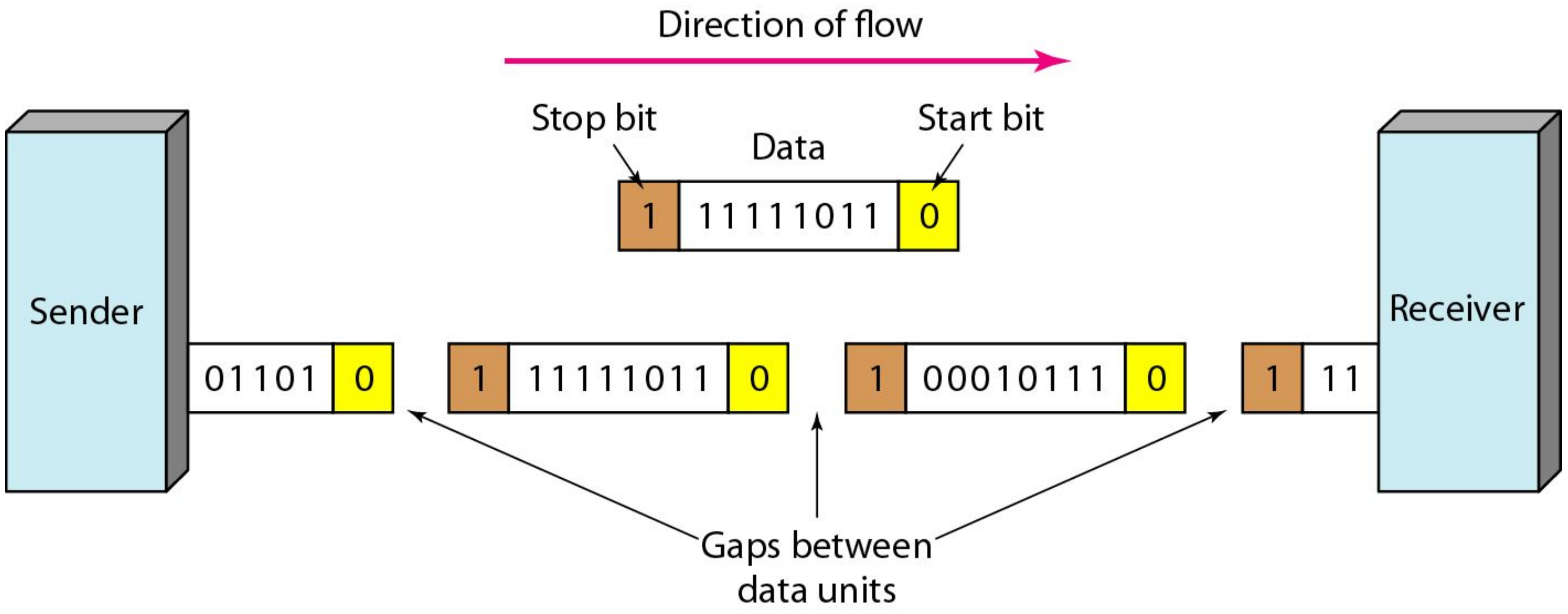
Note

In asynchronous transmission, we send 1 start bit (0) at the beginning and 1 or more stop bits (1s) at the end of each byte. There may be a gap between each byte.

Note

Asynchronous here means “asynchronous at the byte level,” but the bits are still synchronized; their durations are the same.

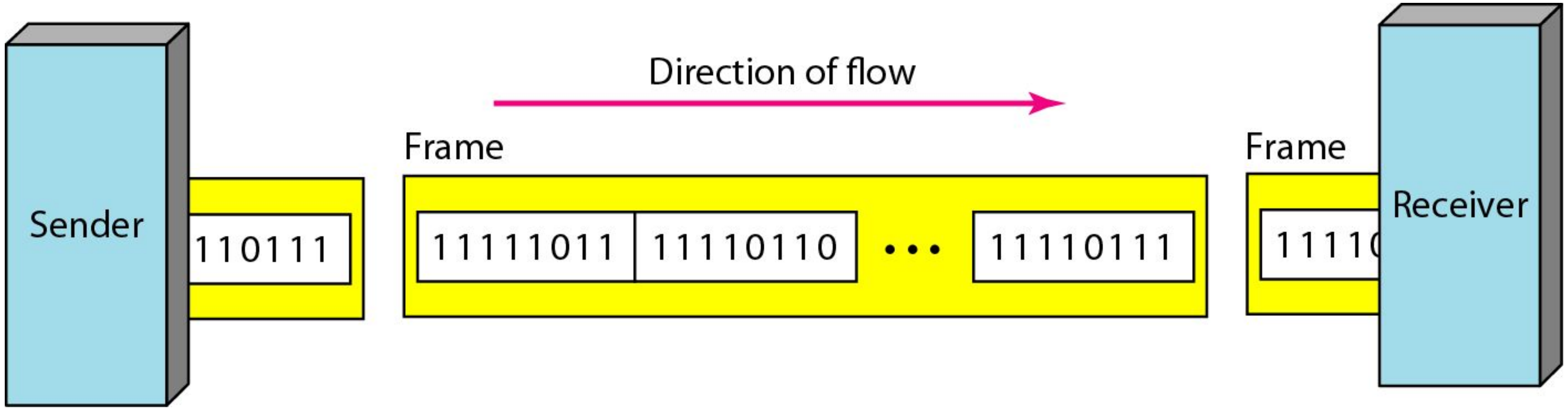
Figure 4.34 *Asynchronous transmission*



Note

In synchronous transmission, we send bits one after another without start or stop bits or gaps. It is the responsibility of the receiver to group the bits. The bits are usually sent as bytes and many bytes are grouped in a frame. A frame is identified with a start and an end byte.

Figure 4.35 *Synchronous transmission*



Isochronous

- In isochronous transmission we cannot have uneven gaps between frames.
- Transmission of bits is fixed with equal gaps.

Chapter 6

Bandwidth Utilization: Multiplexing and Spreading

Note

Bandwidth utilization is the wise use of available bandwidth to achieve specific goals.

Efficiency can be achieved by multiplexing; i.e., sharing of the bandwidth between multiple users.

6-1 MULTIPLEXING

Whenever the bandwidth of a medium linking two devices is greater than the bandwidth needs of the devices, the link can be shared. Multiplexing is the set of techniques that allows the (simultaneous) transmission of multiple signals across a single data link. As data and telecommunications use increases, so does traffic.

Topics discussed in this section:

- **Frequency-Division Multiplexing**
- **Wavelength-Division Multiplexing**
- **Synchronous Time-Division Multiplexing**
- **Statistical Time-Division Multiplexing**

Figure 6.1 *Dividing a link into channels*

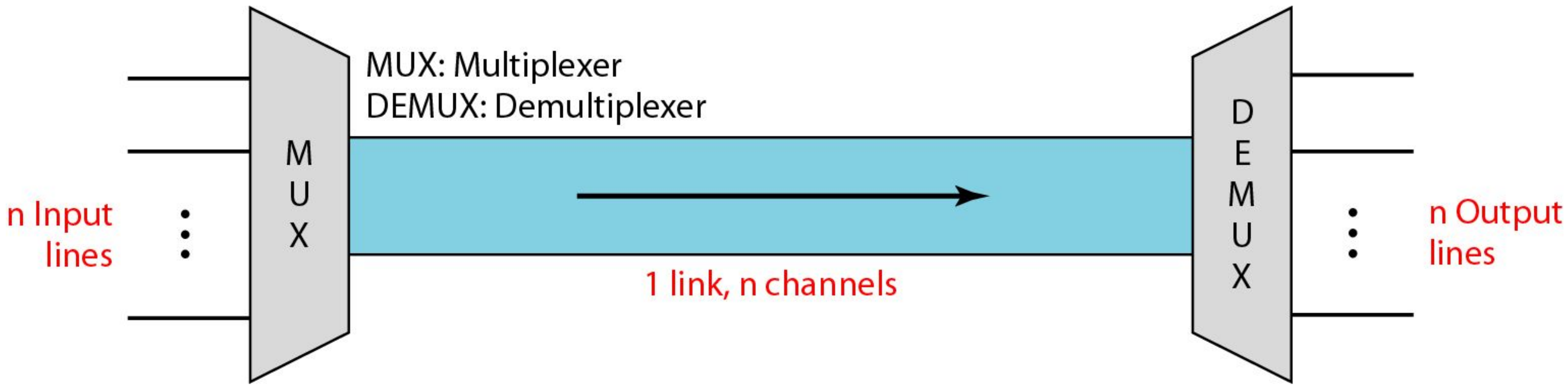


Figure 6.2 *Categories of multiplexing*

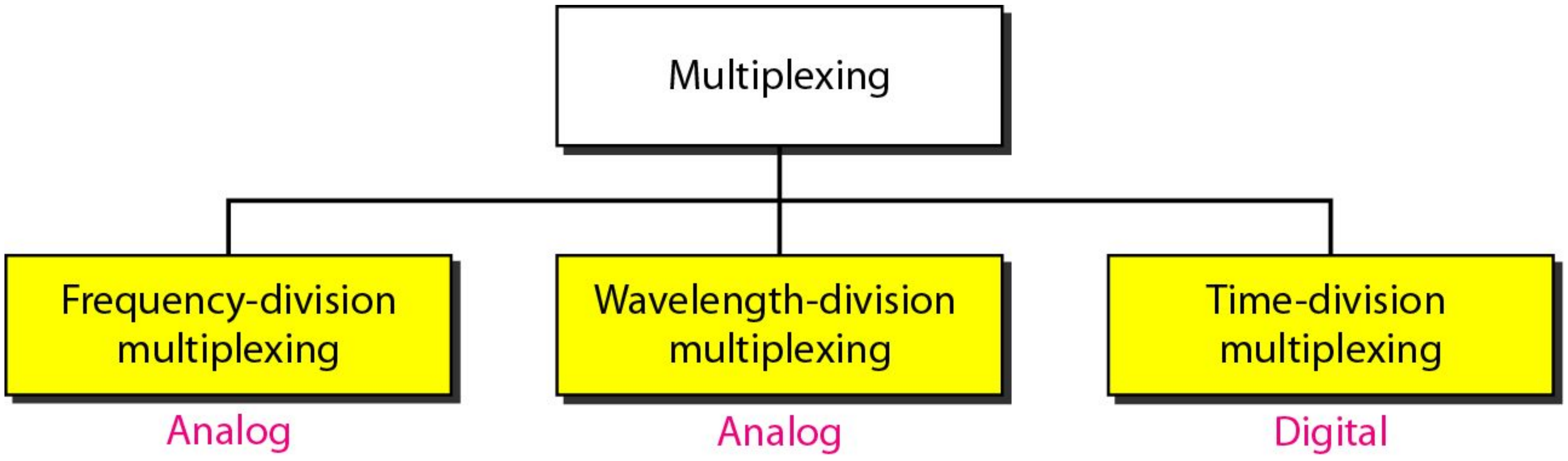
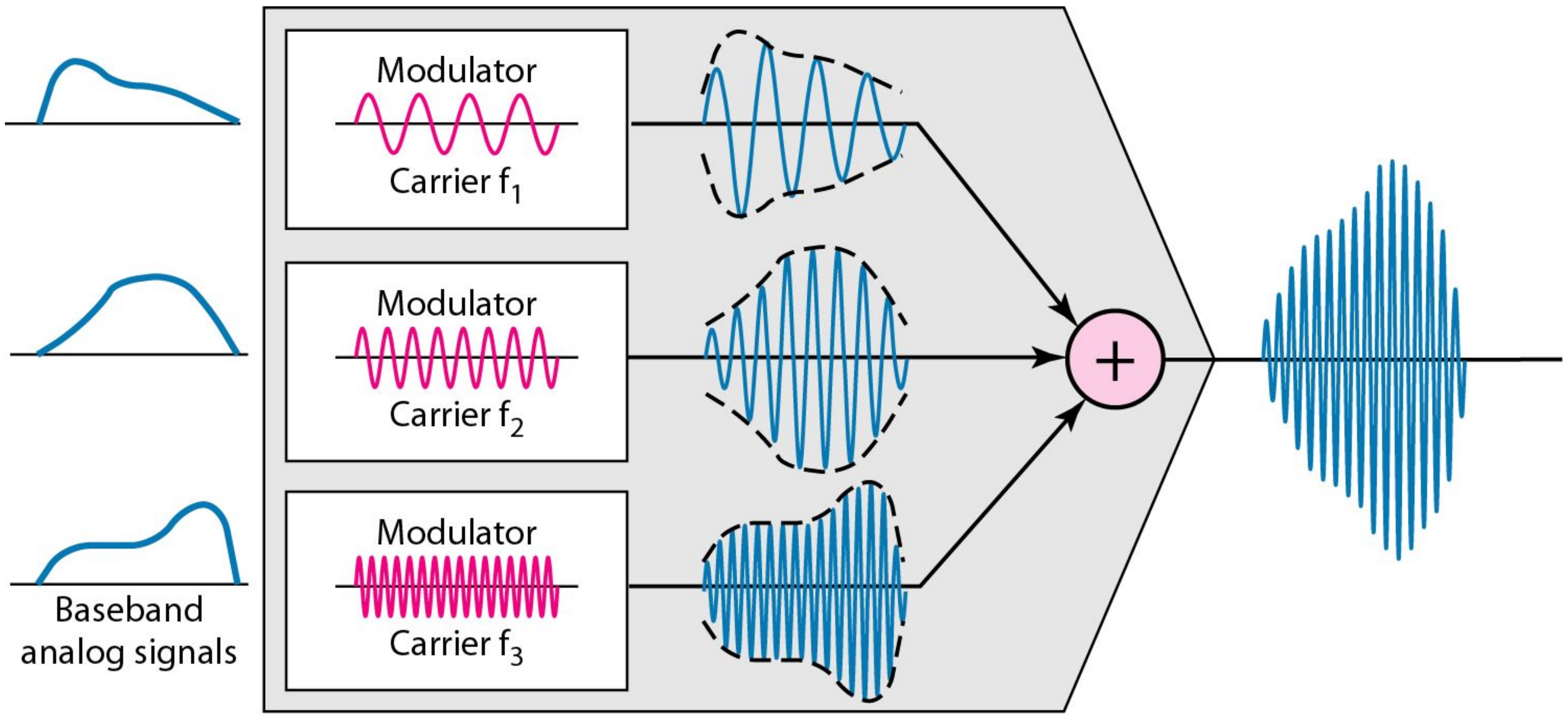


Figure 6.3 *Frequency-division multiplexing (FDM)*

Note

FDM is an analog multiplexing technique that combines analog signals.
It uses the concept of modulation discussed in Ch 5.

Figure 6.4 *FDM process*



FM

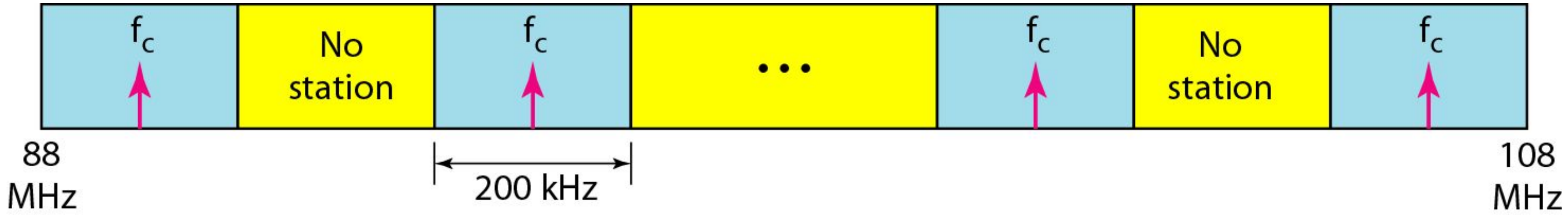
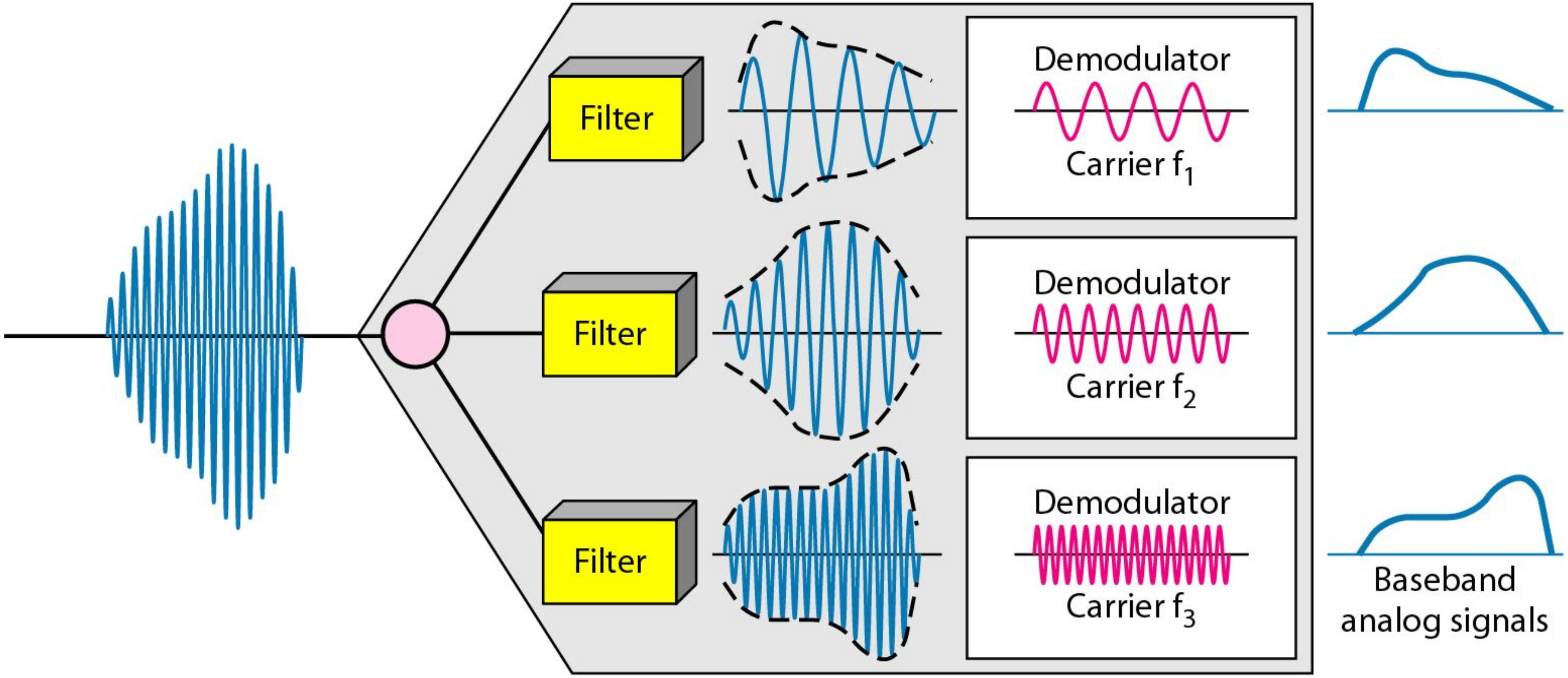


Figure 6.5 *FDM demultiplexing example*



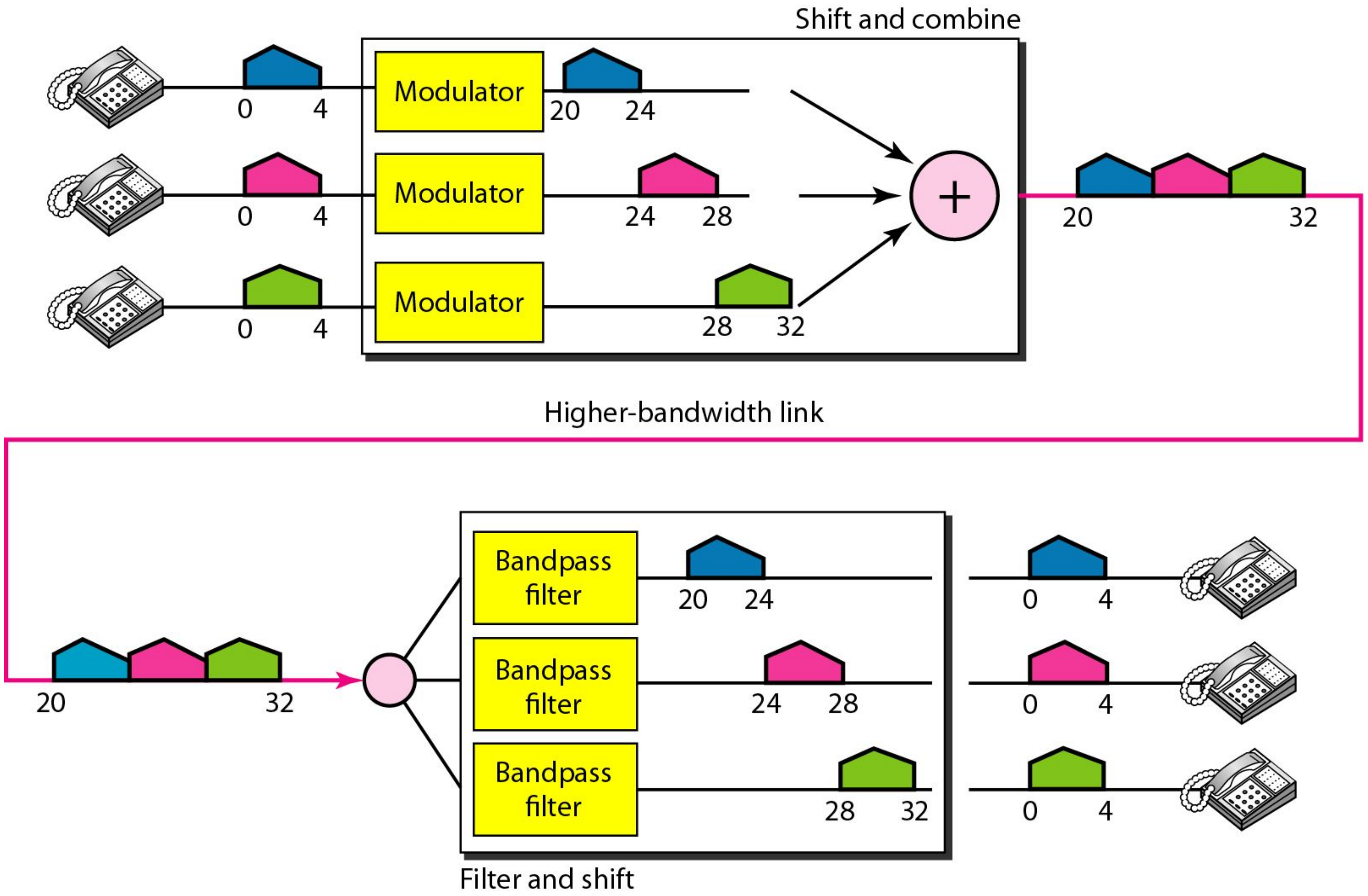
Example 6.1

Assume that a voice channel occupies a bandwidth of 4 kHz. We need to combine three voice channels into a link with a bandwidth of 12 kHz, from 20 to 32 kHz. Show the configuration, using the frequency domain. Assume there are no guard bands.

Solution

We shift (modulate) each of the three voice channels to a different bandwidth, as shown in Figure 6.6. We use the 20- to 24-kHz bandwidth for the first channel, the 24- to 28-kHz bandwidth for the second channel, and the 28- to 32-kHz bandwidth for the third one. Then we combine them as shown in Figure 6.6.

Figure 6.6 *Example 6.1*



Example 6.2

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?

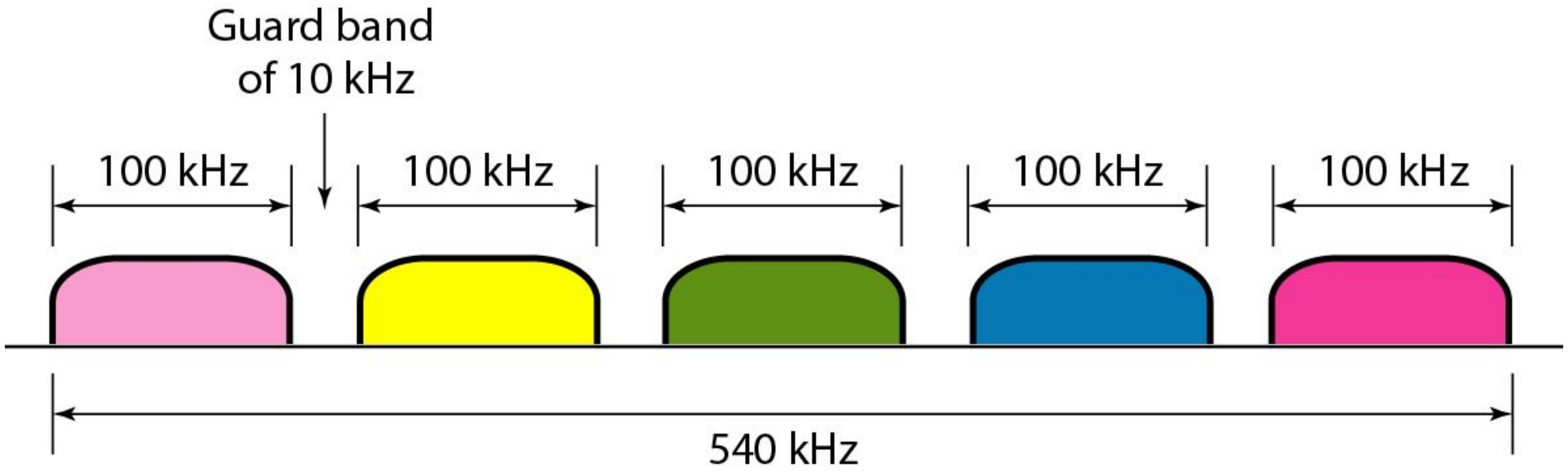
Solution

For five channels, we need at least four guard bands. This means that the required bandwidth is at least

$$5 \times 100 + 4 \times 10 = 540 \text{ kHz},$$

as shown in Figure 6.7.

Figure 6.7 *Example 6.2*



Example 6.3

Four data channels (digital), each transmitting at 1 Mbps, use a satellite channel of 1 MHz. Design an appropriate configuration, using FDM.

Solution

The satellite channel is analog. We divide it into four channels, each channel having 1M/4=250-kHz bandwidth.

Each digital channel of 1 Mbps must be transmitted over a 250KHz channel. Assuming no noise we can use Nyquist to get:

$C = 1Mbps = 2x250K \times \log_2 L \rightarrow L = 4$ or $n = 2$ bits/signal element.

One solution is 4-QAM modulation. In Figure 6.8 we show a possible configuration with $L = 16$.

Figure 6.8 *Example 6.3*

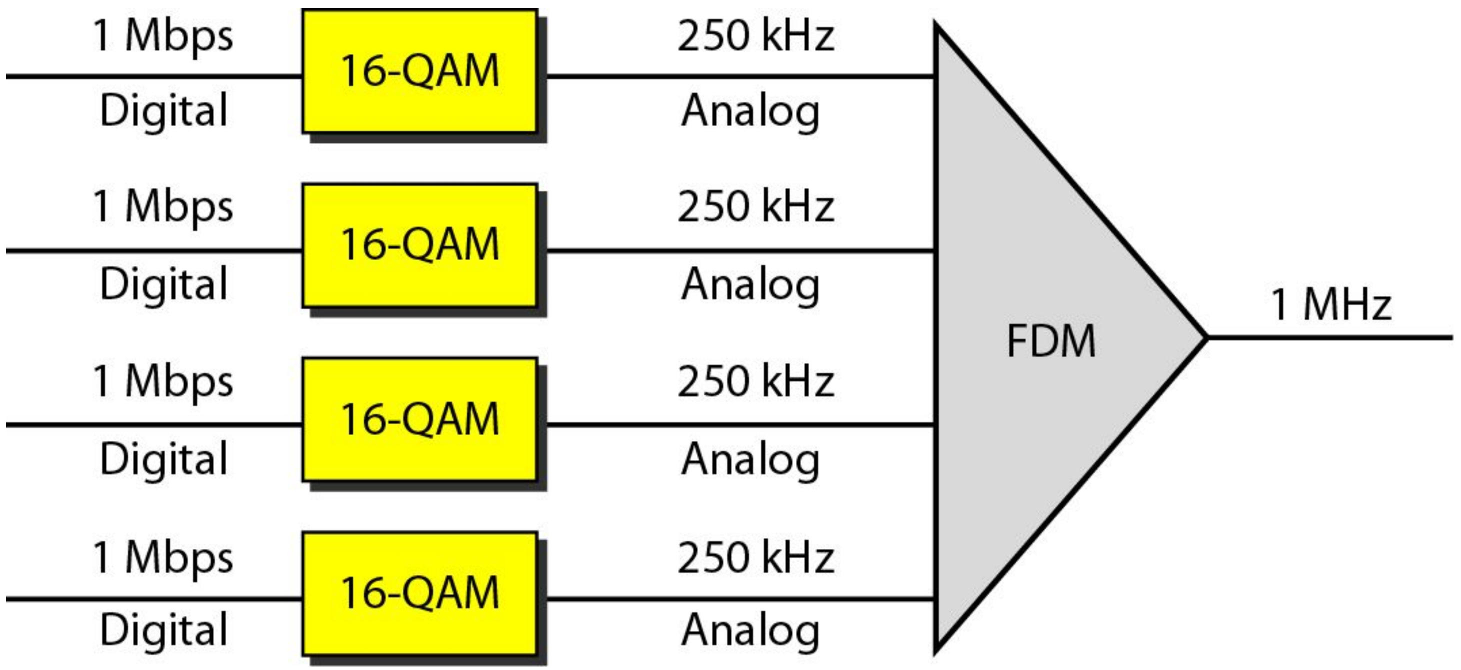
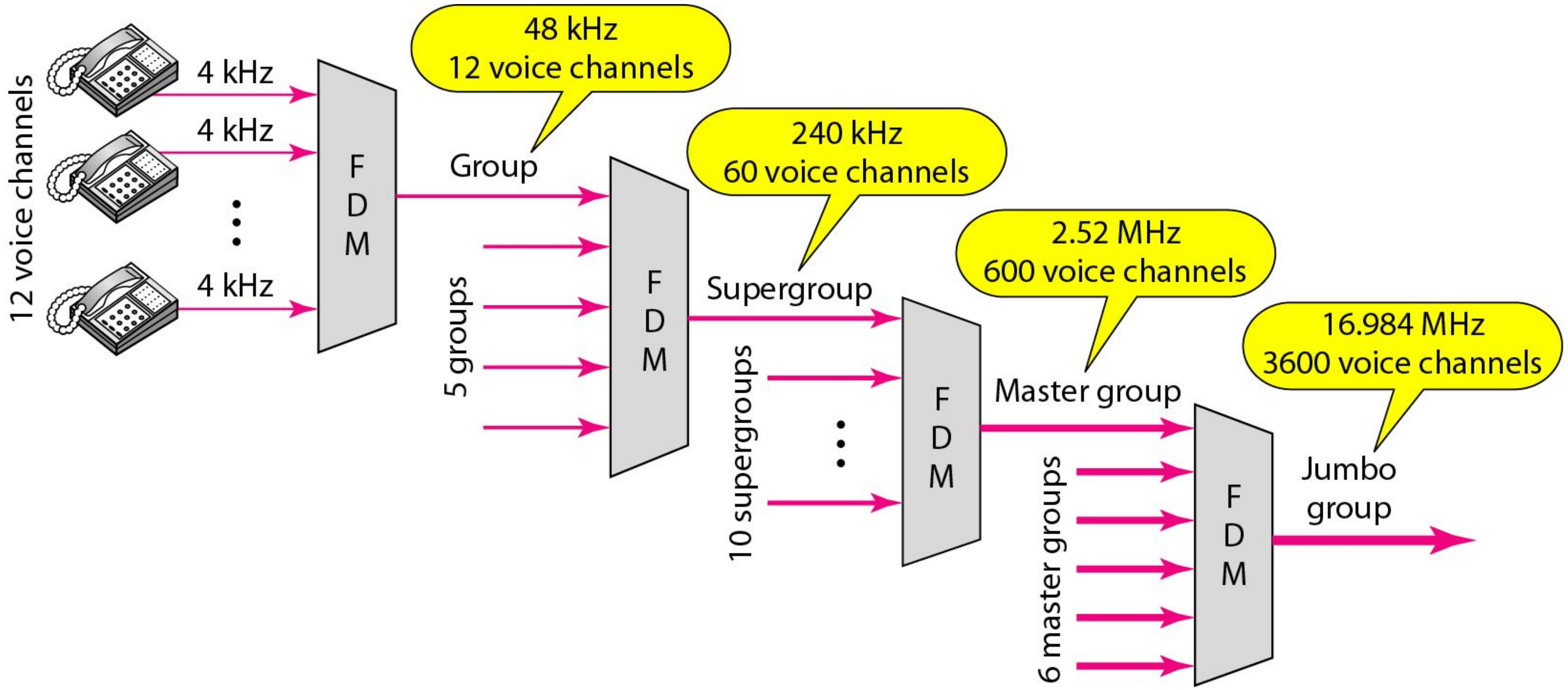


Figure 6.9 *Analog hierarchy*



Other Applications of FDM

A very common application of FDM is AM and FM radio broadcasting.

Radio uses the air as the transmission medium.

A special band from 530 to 1700 kHz is assigned to AM radio.

All radio stations need to share this band.

Each AM station needs 10 kHz of bandwidth.

Without multiplexing, only one AM station could broadcast to the common link, the air.

FM has a wider band of 88 to 108 MHz because each station needs a bandwidth of 200 kHz. Television broadcasting.

Each TV channel has its own bandwidth of 6 MHz.

Cellular telephones also uses FDM.

Each user is assigned two 30-kHz channels, one for sending voice and the other for receiving. The voice signal, which has a bandwidth of 3 kHz (from 300 to 3300 Hz), is modulated by using FM.

Remember that an FM signal has a bandwidth 10 times that of the modulating signal 30 kHz (10 × 3) of bandwidth.

Each user is given, by the base station, a 60-kHz bandwidth in a range available at the time of the call.

Implementation

For cases, such as radio and television broadcasting, there is no need for a physical multiplexer or demultiplexer. As long as the stations agree to send their broadcasts to the air using different carrier frequencies, multiplexing is achieved.

In other cases, such as the cellular telephone system, a base station needs to assign a carrier frequency to the telephone user. There is not enough bandwidth in a cell to permanently assign a bandwidth range to every telephone user. When a user hangs up, her or his bandwidth is assigned to another caller.

Example 6.4

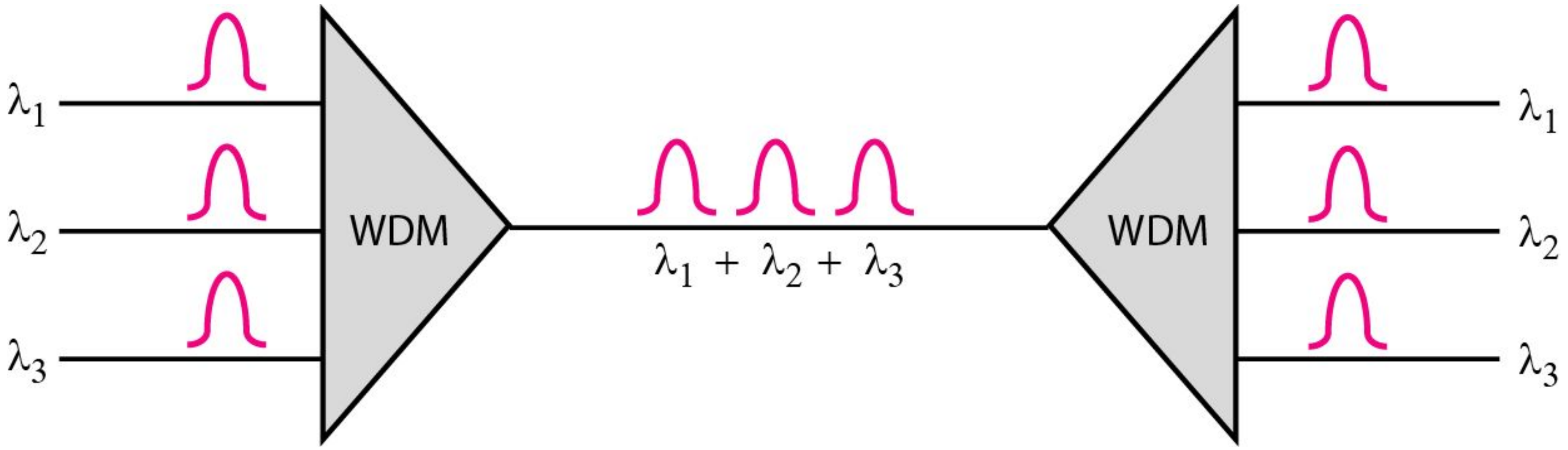
The Advanced Mobile Phone System (AMPS) uses two bands. The first band of 824 to 849 MHz is used for sending, and 869 to 894 MHz is used for receiving. Each user has a bandwidth of 30 kHz in each direction. How many people can use their cellular phones simultaneously?

Solution

Each band is 25 MHz. If we divide 25 MHz by 30 kHz, we get 833.33. In reality, the band is divided into 832 channels. Of these, 42 channels are used for control, which means only 790 channels are available for cellular phone users.

Figure 6.10 *Wavelength-division multiplexing (WDM)*

Wavelength-division multiplexing (WDM) is designed to use the high-data-rate capability of fiber-optic cable.
The optical fiber data rate is higher than the data rate of metallic transmission cable,
but using a fiber-optic cable for a single line wastes the available bandwidth.



Note

WDM is an analog multiplexing technique to combine optical signals.

Figure 6.11 *Prisms in wavelength-division multiplexing and demultiplexing*

Achieved by varying the angle of incidence and frequency

Very narrow bands of light from different sources are combined to make a wider band of light. At the receiver, the signals are separated by the demultiplexer.

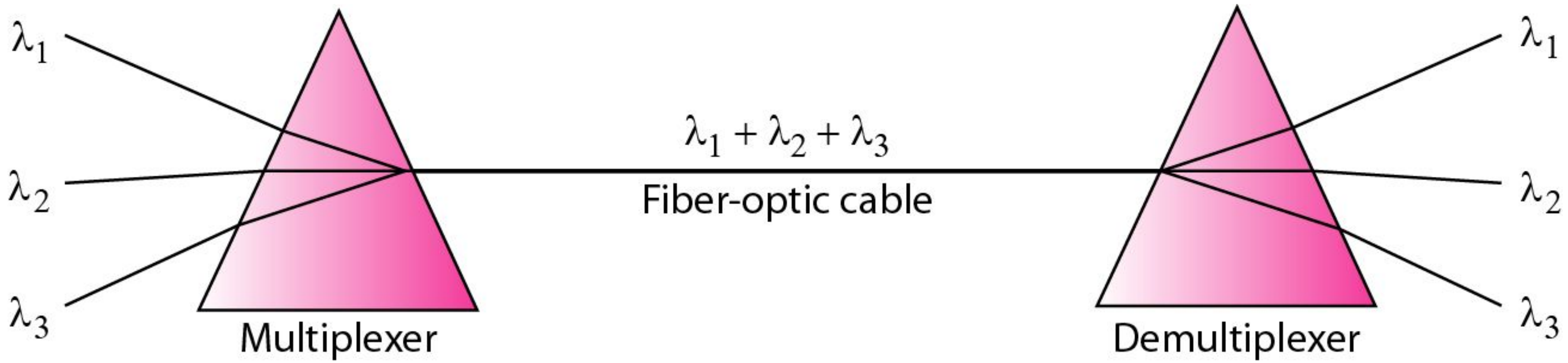
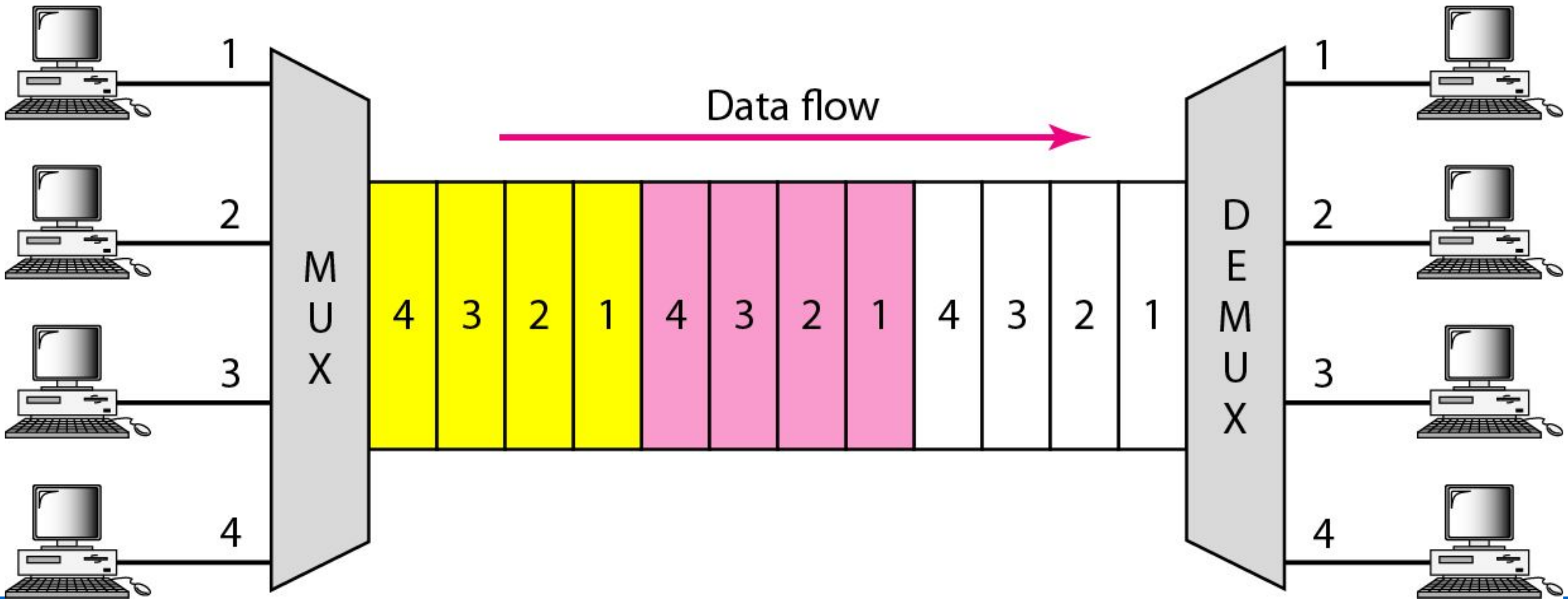


Figure 6.12 Time Division Multiplexing (*TDM)*

TDM is a digital process that allows several connections to share the high bandwidth of a link.

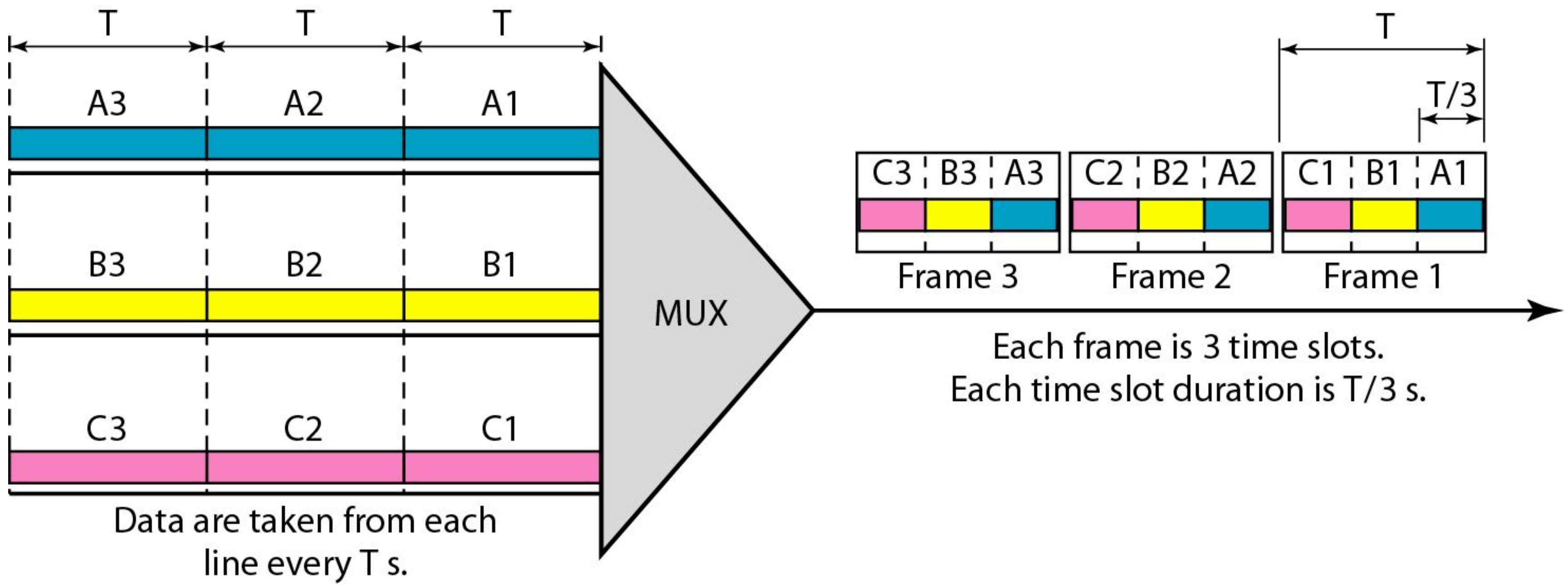
Instead of sharing a portion of the bandwidth as in FDM, time is shared.



Note

TDM is a digital multiplexing technique for combining several low-rate digital channels into one high-rate one.

Figure 6.13 *Synchronous time-division multiplexing*



Note

In synchronous TDM, the data rate of the link is *n* times faster, and the unit duration is *n* times shorter.

Example 6.5

In Figure 6.13, the data rate for each one of the 3 input connection is 1 kbps. If 1 bit at a time is multiplexed (a unit is 1 bit), what is the duration of (a) each input slot, (b) each output slot, and (c) each frame?

Solution

We can answer the questions as follows:

a. The data rate of each input connection is 1 kbps. This means that the bit duration is 1/1000 s or 1 ms. The duration of the input time slot is 1 ms (same as bit duration).

Example 6.5 (continued)

b. ***The duration of each output time slot is one-third of the input time slot. This means that the duration of the output time slot is 1/3 ms.***

c. ***Each frame carries three output time slots. So the duration of a frame is 3 × 1/3 ms, or 1 ms.***

Note: The duration of a frame is the same as the duration of an input unit.

Example 6.6

Figure 6.14 shows synchronous TDM with 4 1Mbps data stream inputs and one data stream for the output. The unit of data is 1 bit. Find (a) the input bit duration, (b) the output bit duration, (c) the output bit rate, and (d) the output frame rate.

Solution

We can answer the questions as follows:

a. The input bit duration is the inverse of the bit rate: 1/1 Mbps = 1 μs.

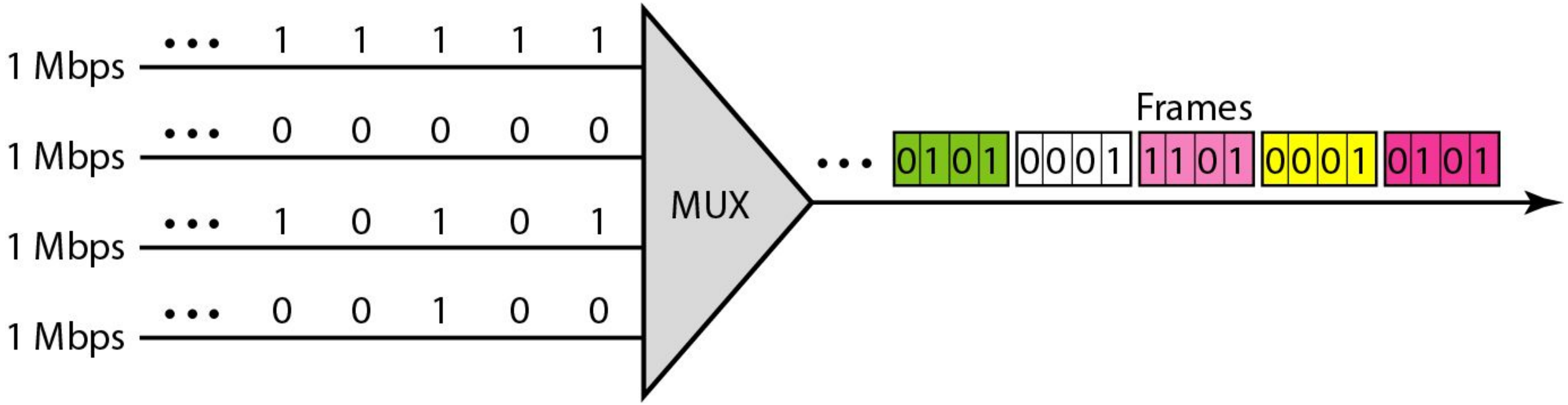
b. The output bit duration is one-fourth of the input bit duration, or ¼ μs.

Example 6.6 (continued)

c. *The output bit rate is the inverse of the output bit duration or 1/(4μs) or 4 Mbps. This can also be deduced from the fact that the output rate is 4 times as fast as any input rate; so the output rate = 4 × 1 Mbps = 4 Mbps.*

d. *The frame rate is always the same as any input rate. So the frame rate is 1,000,000 frames per second. Because we are sending 4 bits in each frame, we can verify the result of the previous question by multiplying the frame rate by the number of bits per frame.*

Figure 6.14 *Example 6.6*



Example 6.7

Four 1-kbps connections are multiplexed together. A unit is 1 bit. Find (a) the duration of 1 bit before multiplexing, (b) the transmission rate of the link, (c) the duration of a time slot, and (d) the duration of a frame.

Solution

We can answer the questions as follows:

a. The duration of 1 bit before multiplexing is 1 / 1 kbps, or 0.001 s (1 ms).

b. The rate of the link is 4 times the rate of a connection, or 4 kbps.

Example 6.7 (continued)

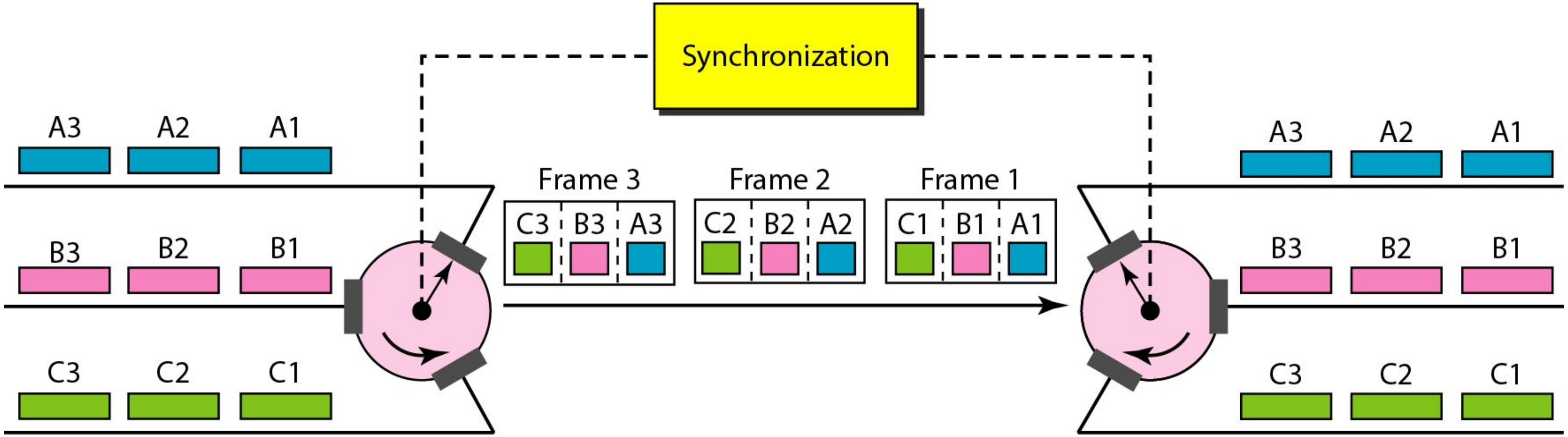
c. *The duration of each time slot is one-fourth of the duration of each bit before multiplexing, or 1/4 ms or 250 μs. Note that we can also calculate this from the data rate of the link, 4 kbps. The bit duration is the inverse of the data rate, or 1/4 kbps or 250 μs.*

d. *The duration of a frame is always the same as the duration of a unit before multiplexing, or 1 ms. We can also calculate this in another way. Each frame in this case has four time slots. So the duration of a frame is 4 times 250 μs, or 1 ms.*

Interleaving

- The process of taking a group of bits from each input line for multiplexing is called interleaving.
- We interleave bits (1 - n) from each input onto one output.

Figure 6.15 *Interleaving*



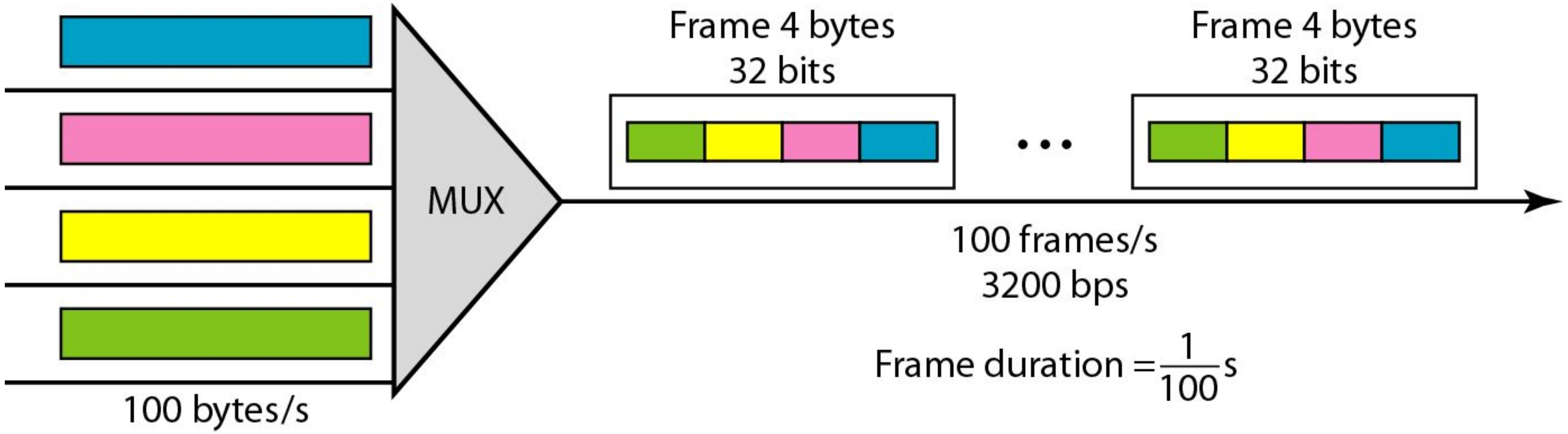
Example 6.8

Four channels are multiplexed using TDM. If each channel sends 100 bytes /s and we multiplex 1 byte per channel, show the frame traveling on the link, the size of the frame, the duration of a frame, the frame rate, and the bit rate for the link.

Solution

The multiplexer is shown in Figure 6.16. Each frame carries 1 byte from each channel; the size of each frame, therefore, is 4 bytes, or 32 bits. Because each channel is sending 100 bytes/s and a frame carries 1 byte from each channel, the frame rate must be 100 frames per second. The bit rate is 100 × 32, or 3200 bps.

Figure 6.16 *Example 6.8*



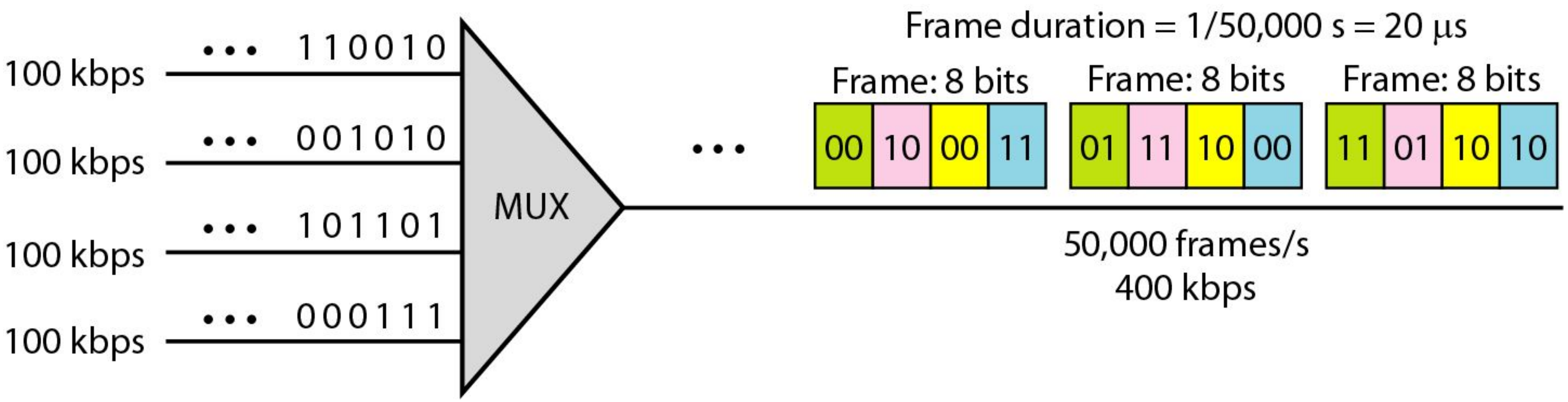
Example 6.9

A multiplexer combines four 100-kbps channels using a time slot of 2 bits. Show the output with four arbitrary inputs. What is the frame rate? What is the frame duration? What is the bit rate? What is the bit duration?

Solution

Figure 6.17 shows the output (4x100kbps) for four arbitrary inputs. The link carries 400K/(2x4)=50,000 2x4=8bit frames per second. The frame duration is therefore 1/50,000 s or 20 μs. The bit duration on the output link is 1/400,000 s, or 2.5 μs.

Figure 6.17 *Example 6.9*



Data Rate Management

- Not all input links maybe have the same data rate.
- Some links maybe slower. There maybe several different input link speeds
- There are three strategies that can be used to overcome the data rate mismatch: multilevel, multislot and pulse stuffing

Data rate matching

- Multilevel: used when the data rate of the input links are multiples of each other.
- Multislot: used when there is a GCD between the data rates. The higher bit rate channels are allocated more slots per frame, and the output frame rate is a multiple of each input link.
- Pulse Stuffing: used when there is no GCD between the links. The slowest speed link will be brought up to the speed of the other links by bit insertion, this is called pulse stuffing.

Figure 6.19 *Multilevel multiplexing*

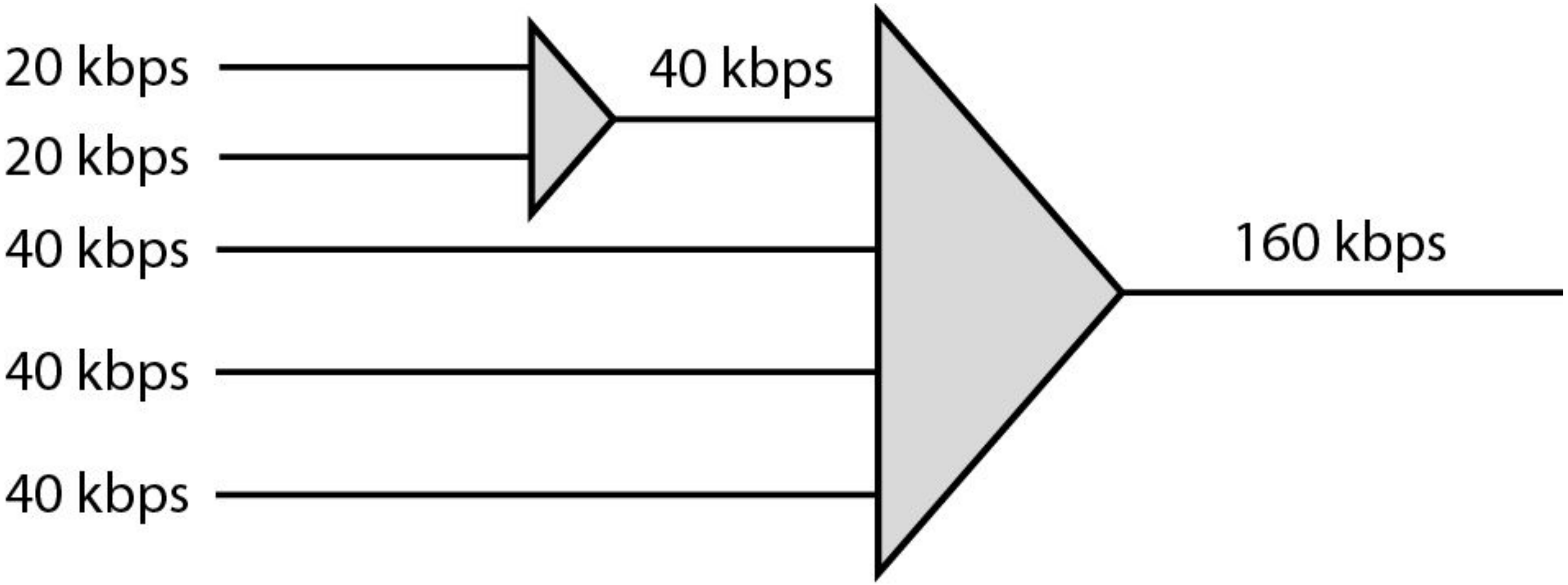


Figure 6.20 *Multiple-slot multiplexing*

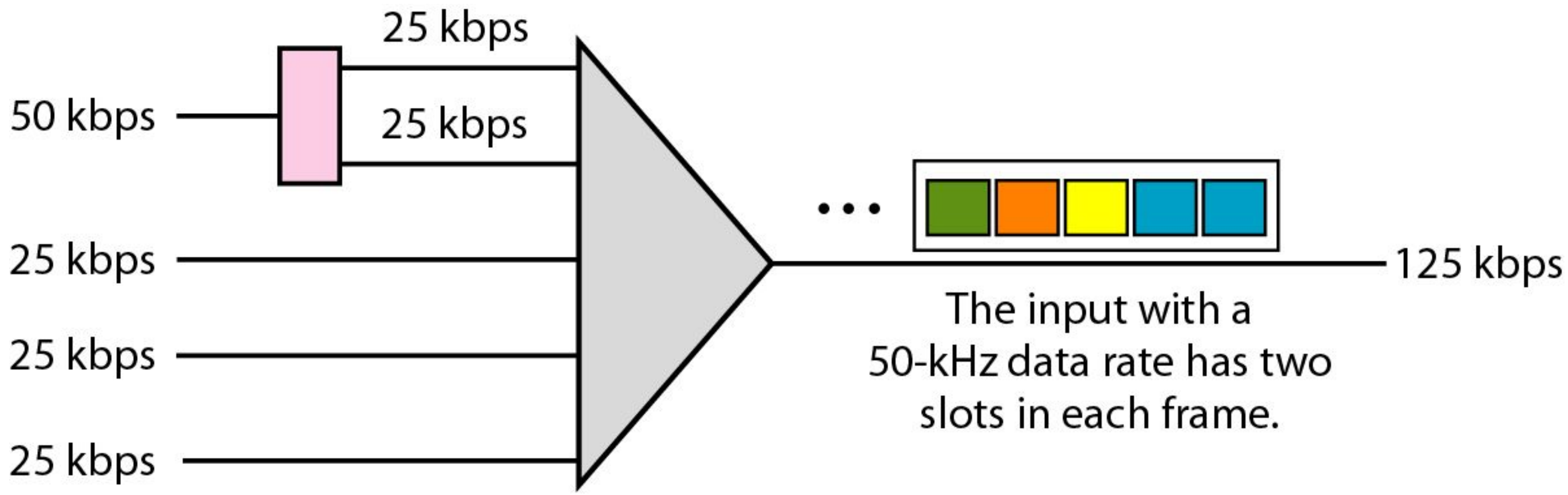
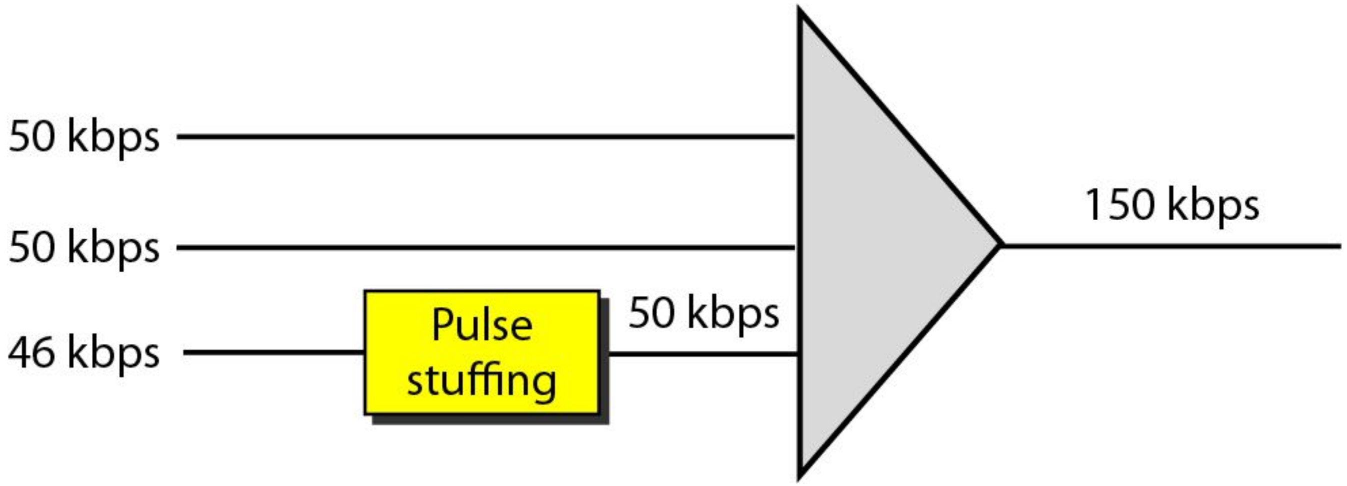


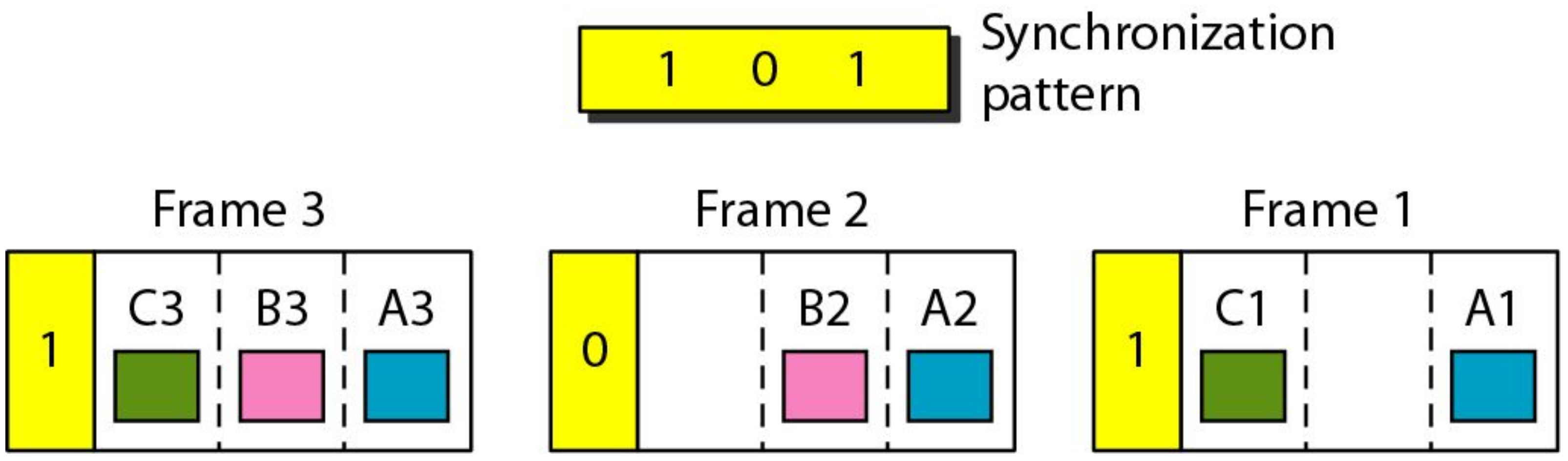
Figure 6.21 *Pulse stuffing*



Synchronization

- To ensure that the receiver correctly reads the incoming bits, i.e., knows the incoming bit boundaries to interpret a “1” and a “0”, a known bit pattern is used between the frames.
- The receiver looks for the anticipated bit and starts counting bits till the end of the frame.
- Then it starts over again with the reception of another known bit.
- These bits (or bit patterns) are called synchronization bit(s).
- They are part of the overhead of transmission.

Figure 6.22 *Framing bits*



Example 6.10

We have four sources, each creating 250 8-bit characters per second. If the interleaved unit is a character and 1 synchronizing bit is added to each frame, find (a) the data rate of each source, (b) the duration of each character in each source, (c) the frame rate, (d) the duration of each frame, (e) the number of bits in each frame, and (f) the data rate of the link.

Solution

We can answer the questions as follows:

a. ***The data rate of each source is 250 × 8 = 2000 bps = 2 kbps.***

Example 6.10 (continued)

b. *Each source sends 250 characters per second; therefore, the duration of a character is 1/250 s, or 4 ms.*
c. *Each frame has one character from each source, which means the link needs to send 250 frames per second to keep the transmission rate of each source.*
d. *The duration of each frame is 1/250 s, or 4 ms. Note that the duration of each frame is the same as the duration of each character coming from each source.*
e. *Each frame carries 4 characters and 1 extra synchronizing bit. This means that each frame is* $4 \times 8 + 1 = 33$ *bits.*

Example 6.11

Two channels, one with a bit rate of 100 kbps and another with a bit rate of 200 kbps, are to be multiplexed. How this can be achieved? What is the frame rate? What is the frame duration? What is the bit rate of the link?

Solution

We can allocate one slot to the first channel and two slots to the second channel. Each frame carries 3 bits. The frame rate is 100,000 frames per second because it carries 1 bit from the first channel. The bit rate is 100,000 frames/s × 3 bits per frame, or 300 kbps.

Figure 6.23 *Digital hierarchy*

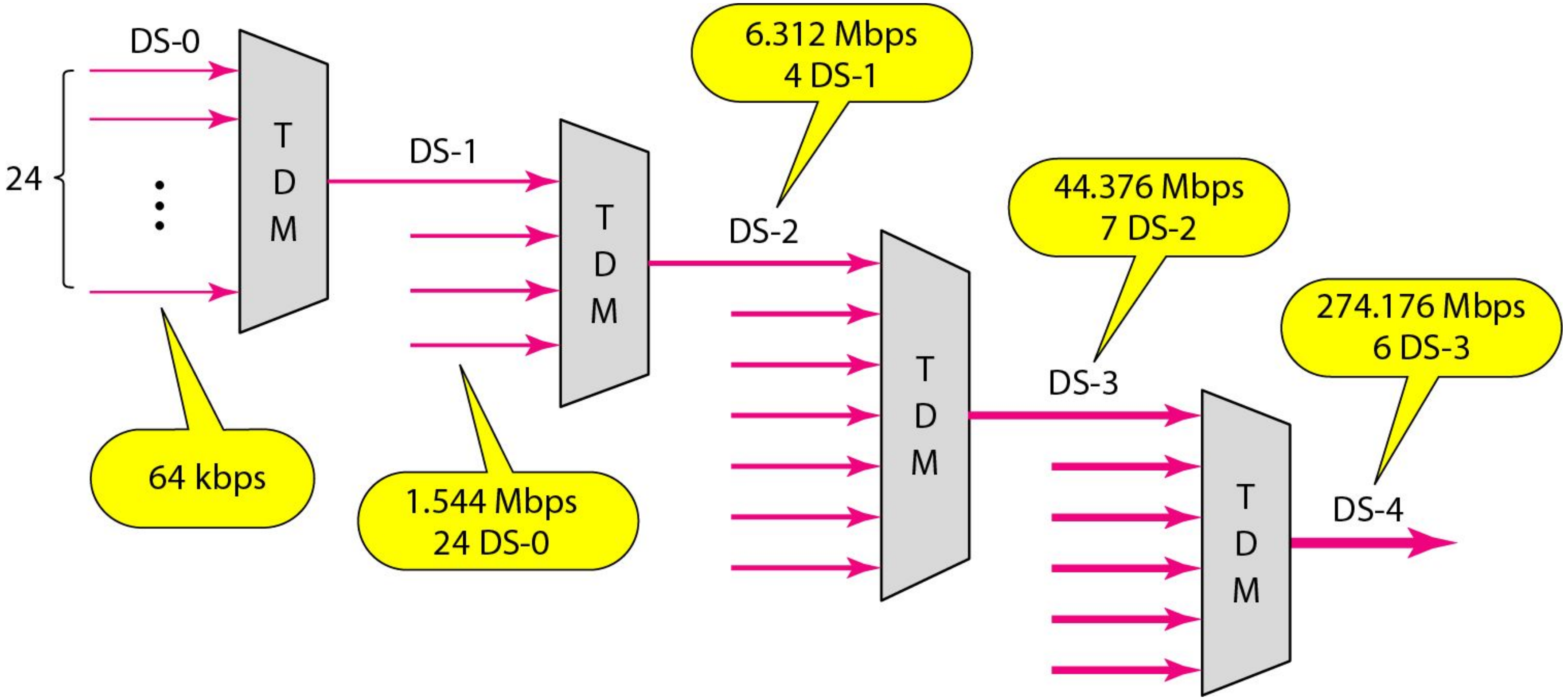


Table 6.1 ***DS and T line rates***

Service	*Line*	*Rate (Mbps)*	*Voice Channels*
DS-1	T-1	1.544	24
DS-2	T-2	6.312	96
DS-3	T-3	44.736	672
DS-4	T-4	274.176	4032

Figure 6.24 *T-1 line for multiplexing telephone lines*

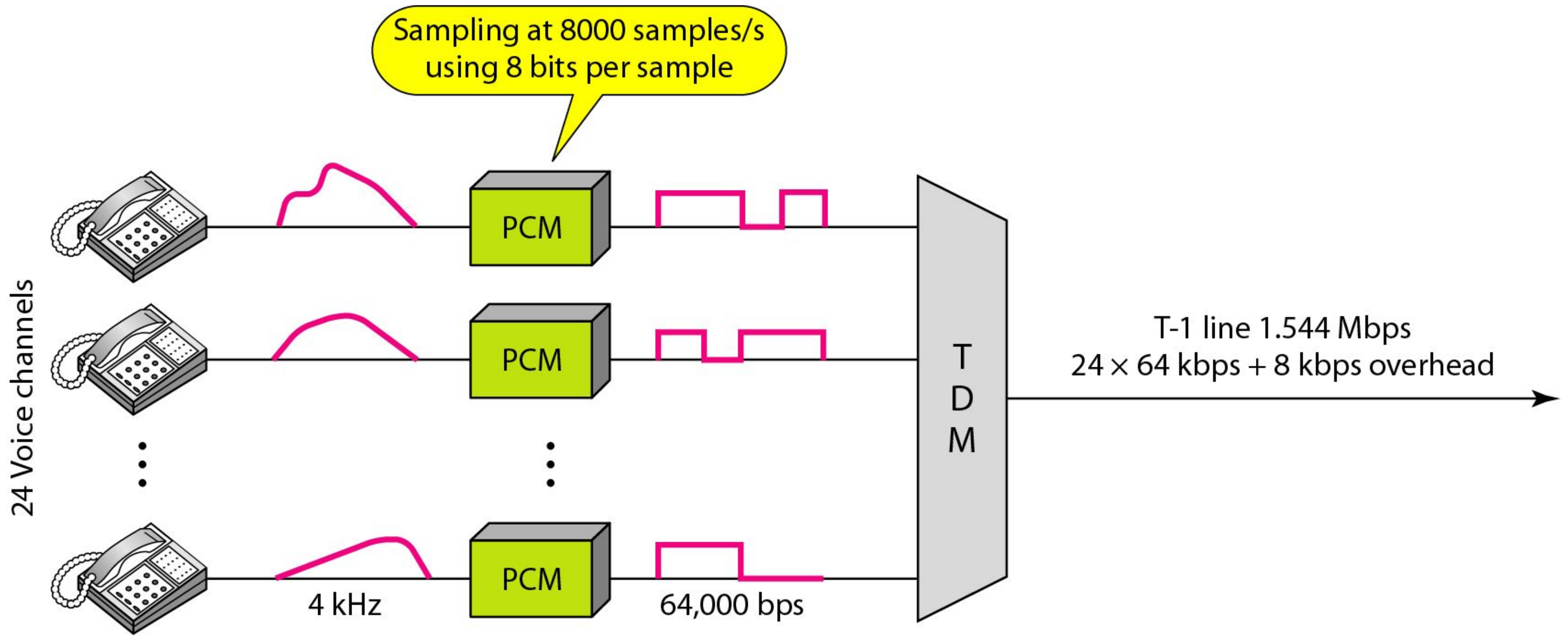


Figure 6.25 *T-1 frame structure*

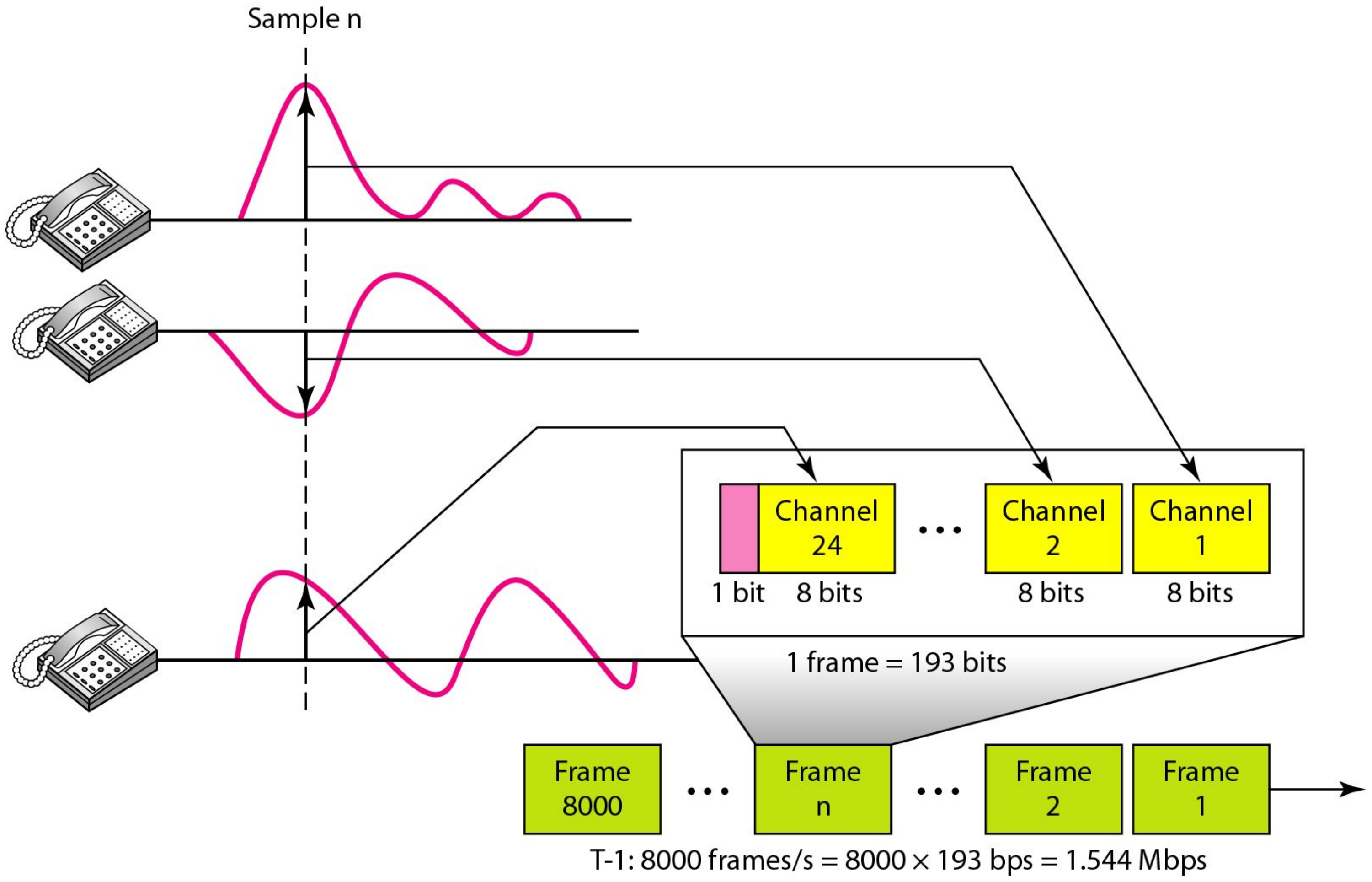


Table 6.2 ***E line rates***

Line	*Rate (Mbps)*	*Voice Channels*
E-1	2.048	30
E-2	8.448	120
E-3	34.368	480
E-4	139.264	1920

Inefficient use of Bandwidth

- Sometimes an input link may have no data to transmit.
- When that happens, one or more slots on the output link will go unused.
- That is wasteful of bandwidth.

Figure 6.18 *Empty slots*

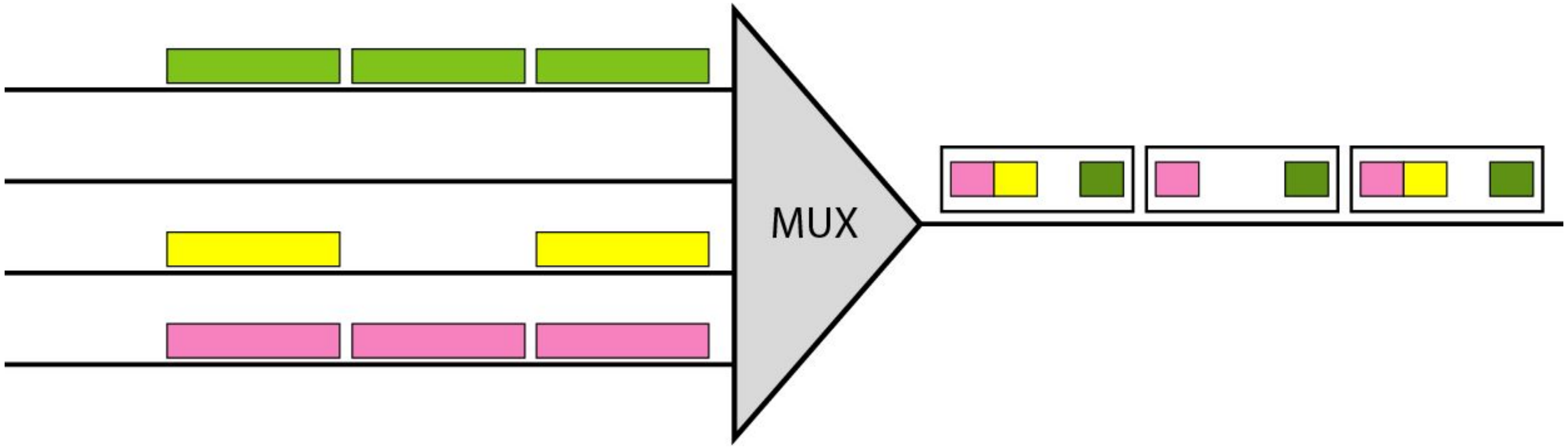
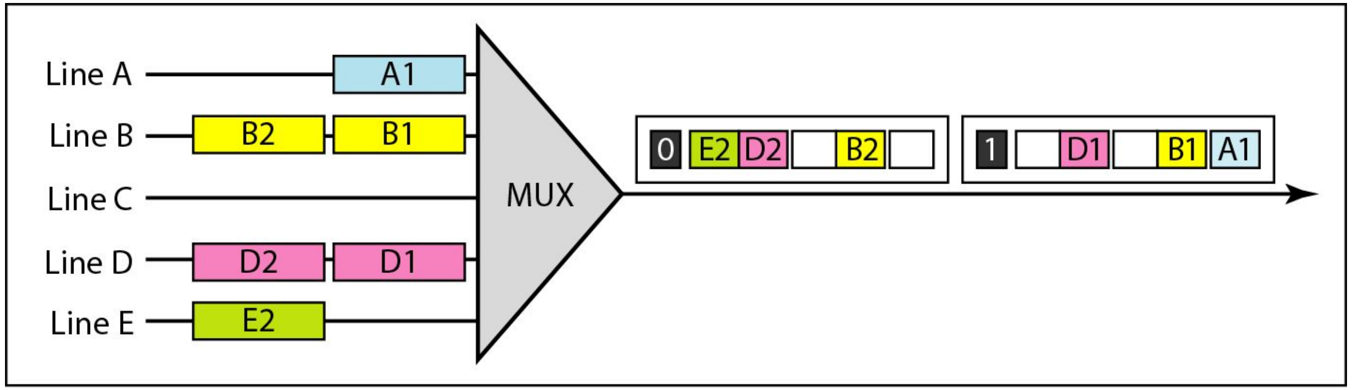
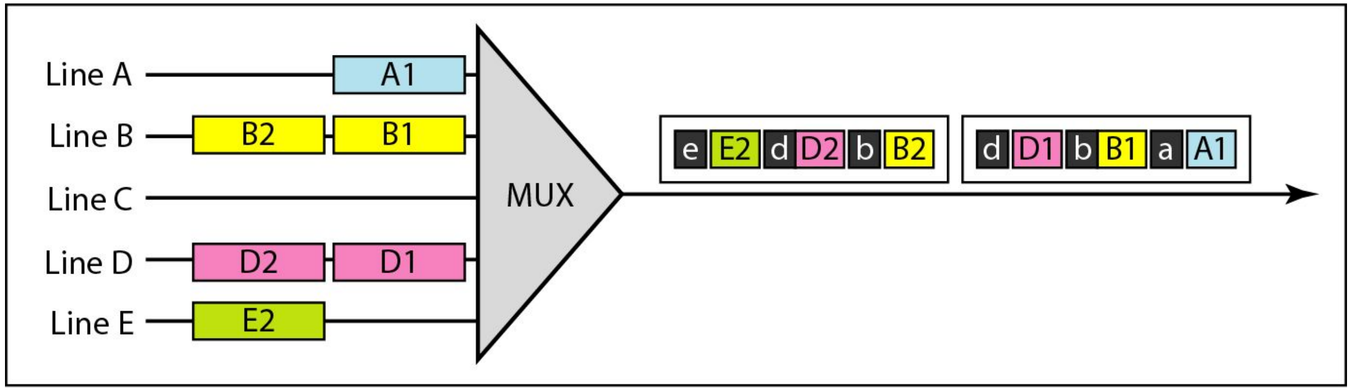


Figure 6.26 *TDM slot comparison*



a. Synchronous TDM



b. Statistical TDM

Addressing

there is no fixed relationship between the inputs and outputs because there are no preassigned or reserved slots. We need to include the address of the receiver inside each slot to show where it is to be delivered. The addressing in its simplest form can be n bits to define N different output lines with $n = \log_2 N$. For example, for eight different output lines, we need a 3-bit address. Slot Size Since a slot carries both data and an address in statistical TDM, the ratio of the data size to address size must be reasonable to make transmission efficient. For example, it would be inefficient to send 1 bit per slot as data when the address is 3 bits. This would mean an overhead of 300 percent. In statistical TDM, a block of data is usually many bytes while the address is just a few bytes.

No Synchronization Bit

There is another difference between synchronous and statistical TDM, but this time it is at the frame level. The frames in statistical TDM need not be synchronized, so we do not need synchronization bits.

Bandwidth

In statistical TDM, the capacity of the link is normally less than the sum of the capacities of each channel. The designers of statistical TDM define the capacity of the link based on the statistics of the load for each channel. If on average only x percent of the input slots are filled, the capacity of the link reflects this. Of course, during peak times, some slots need to wait.

6-2 SPREAD SPECTRUM

In spread spectrum (SS), we combine signals from different sources to fit into a larger bandwidth, but our goals are to prevent eavesdropping and jamming. To achieve these goals, spread spectrum techniques add redundancy.

Topics discussed in this section:

Frequency Hopping Spread Spectrum (FHSS)
Direct Sequence Spread Spectrum Synchronous (DSSS)

Figure 6.27 *Spread spectrum*

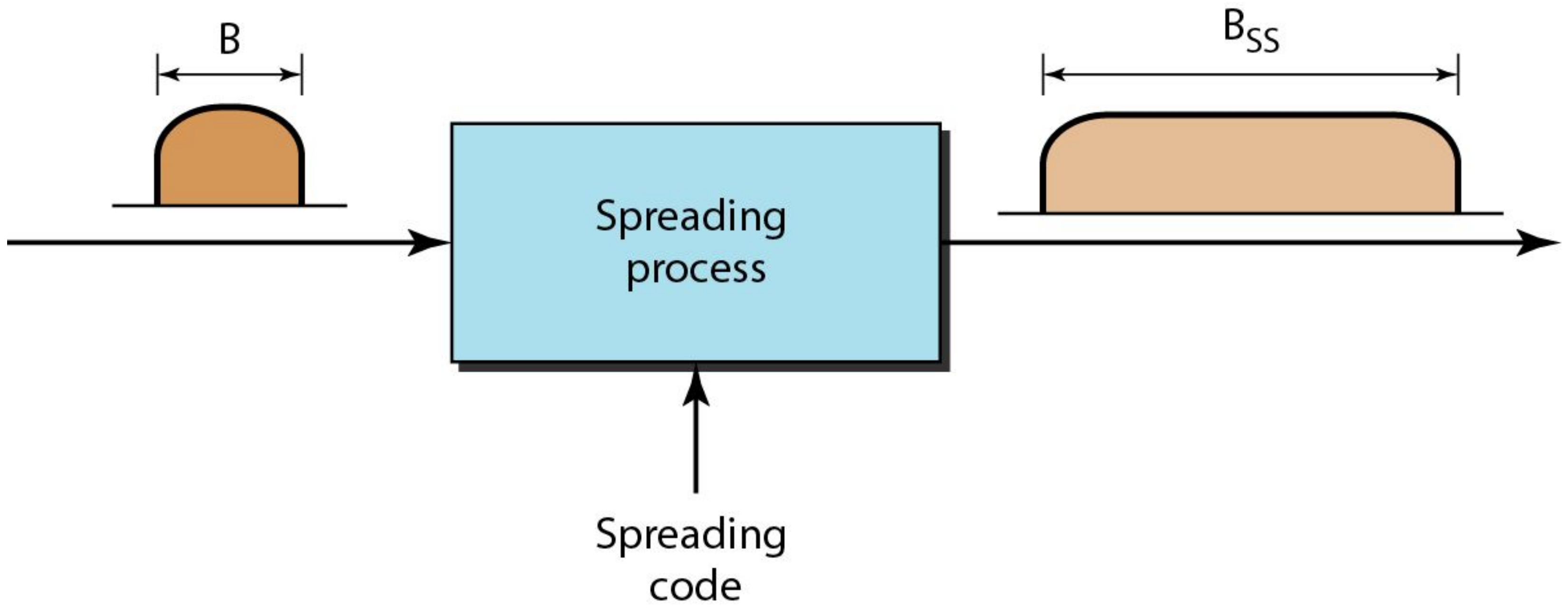


Figure 6.28 *Frequency hopping spread spectrum (FHSS)*

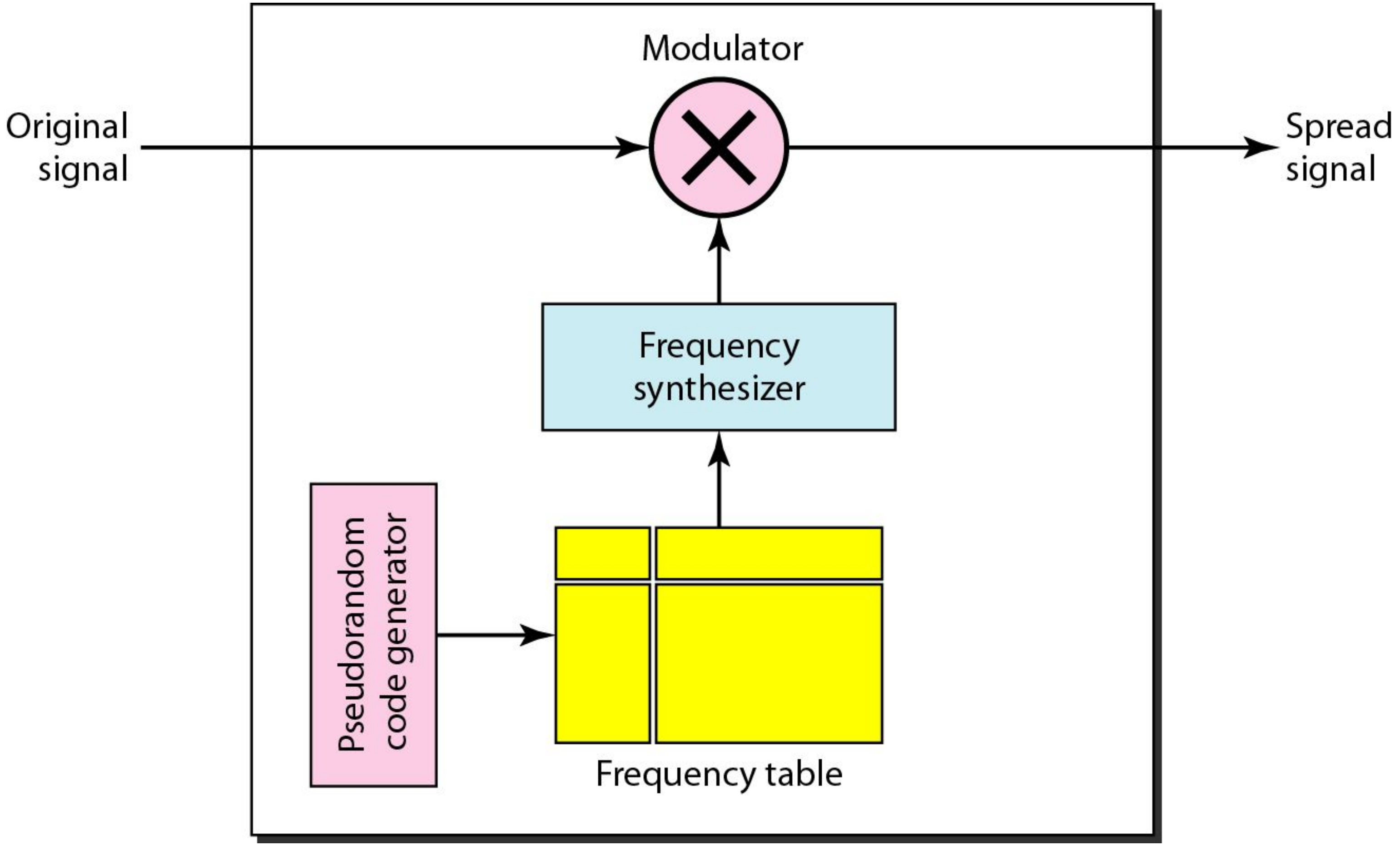


Figure 6.29 *Frequency selection in FHSS*

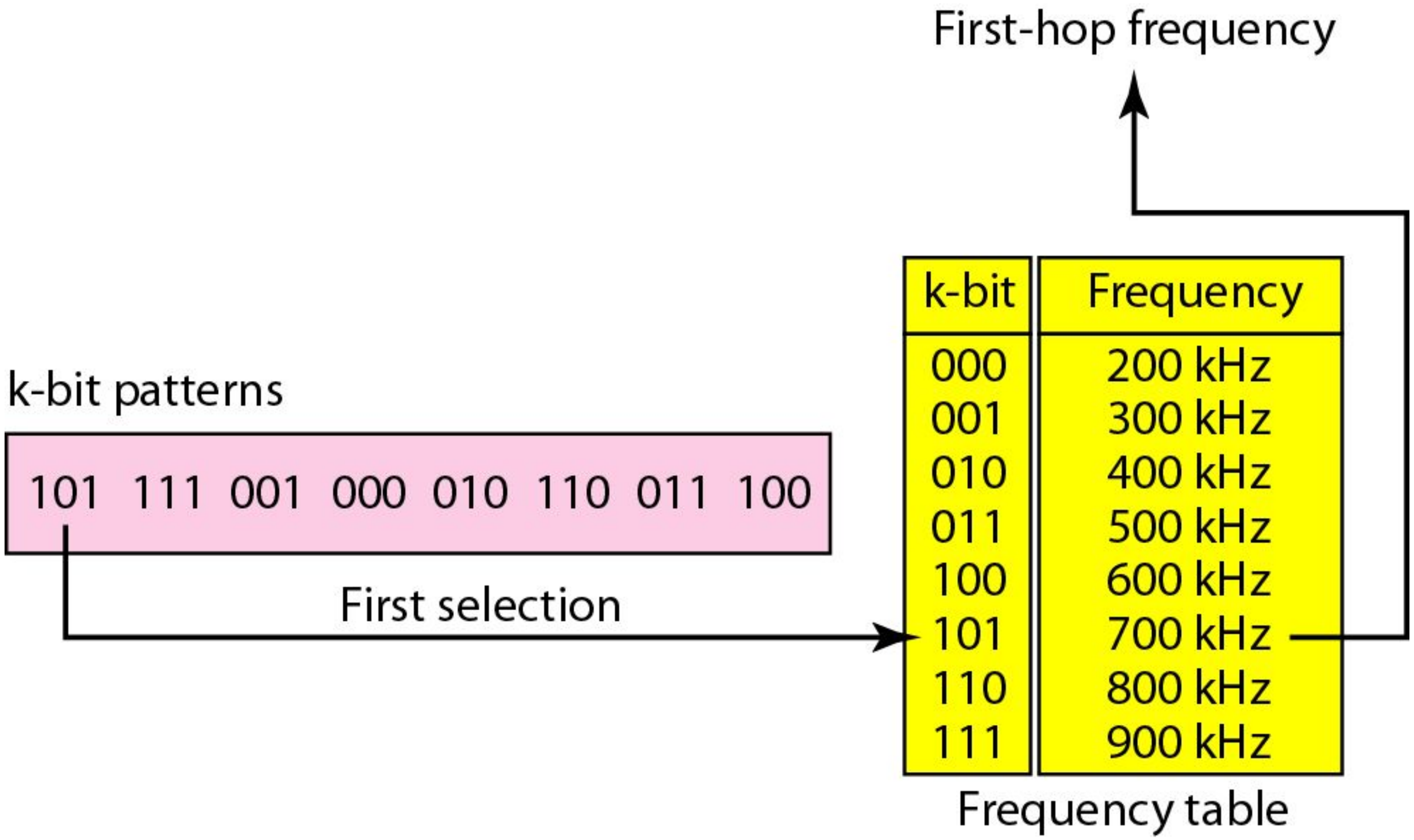


Figure 6.30 *FHSS cycles*

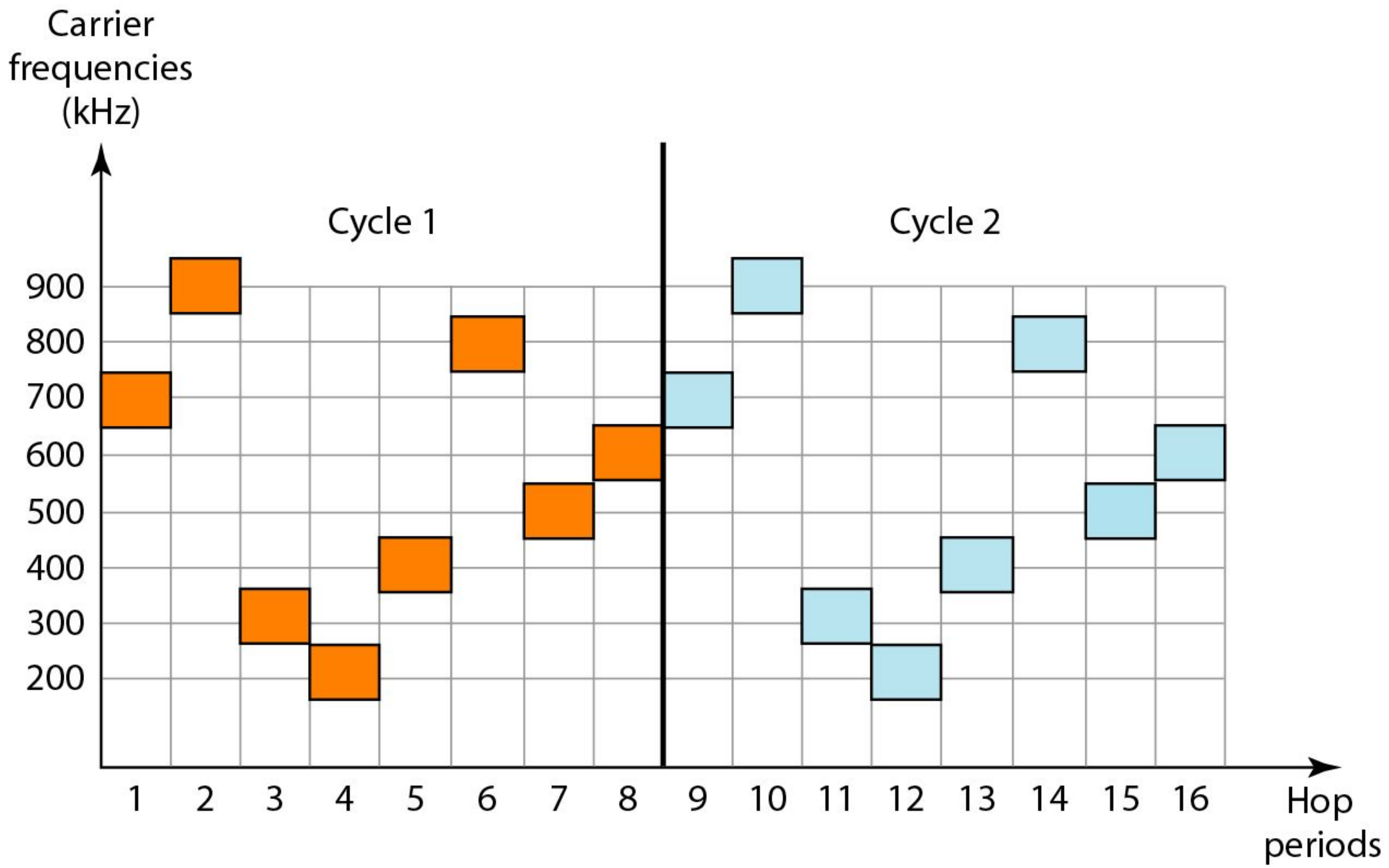


Figure 6.31 *Bandwidth sharing*

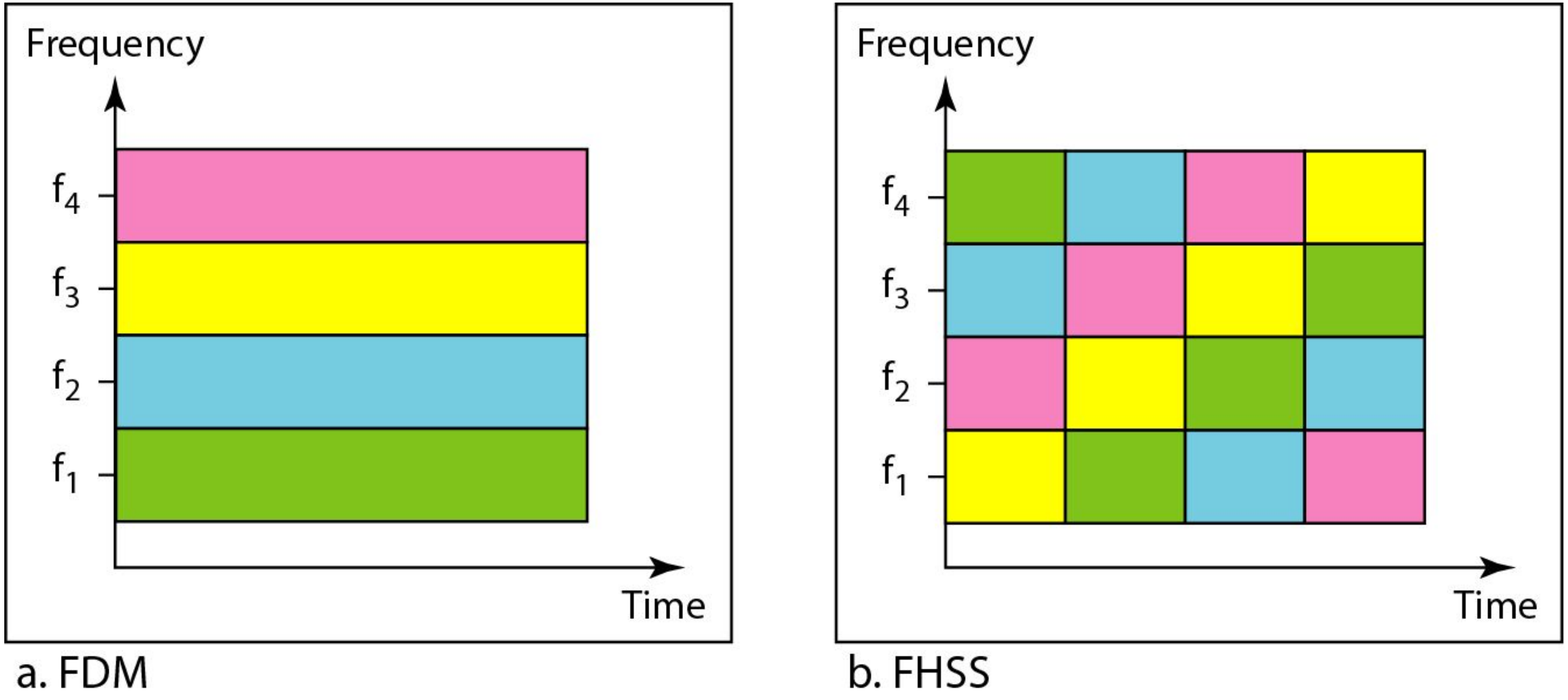


Figure 6.32 *DSSS*

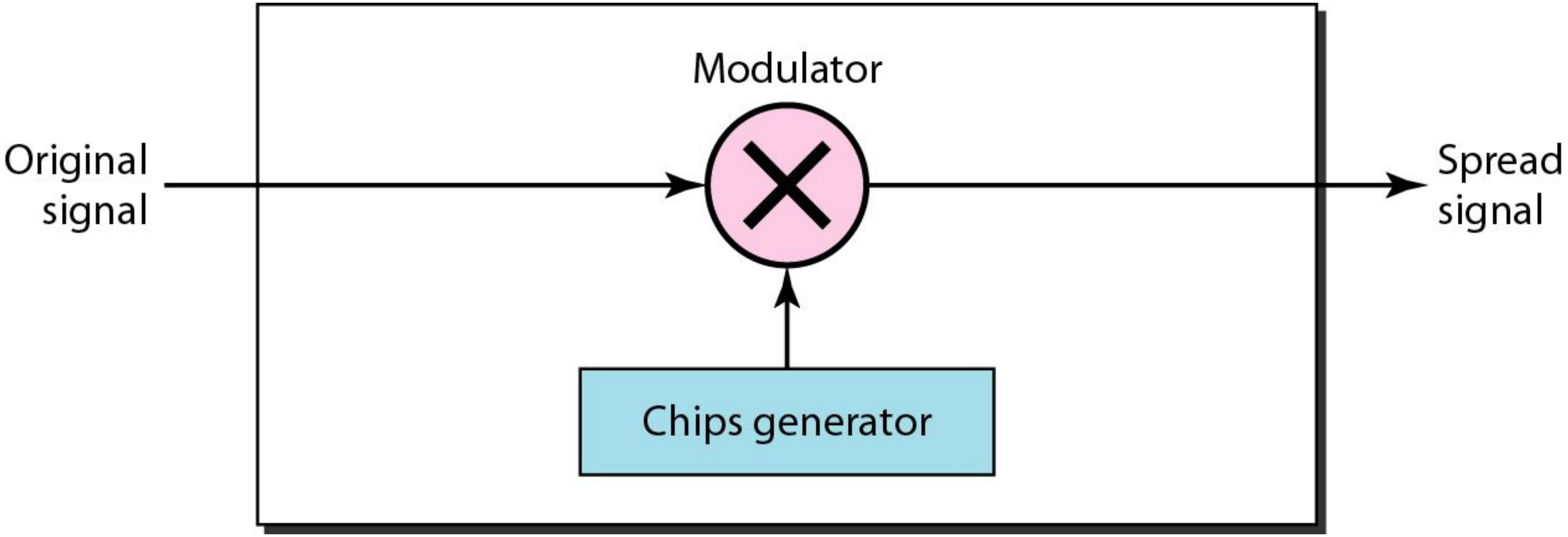
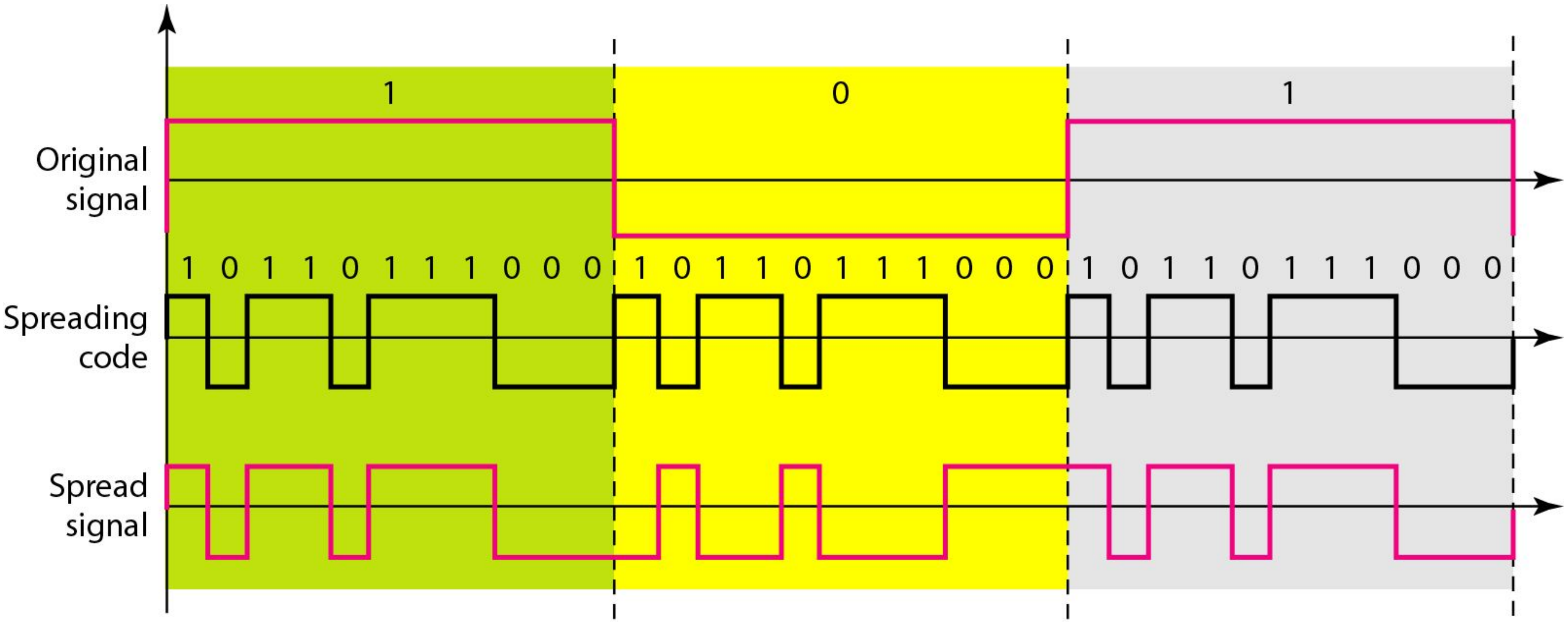


Figure 6.33 Direct Sequence Spread Spectrum (DSSS) example



END OF UNIT I