UNIT I

INTRODUCTION

BASICS OF COMPUTATIONAL THINGKING (CT)

DATA TRANSMISSION & NETWORK CAPACITY REQUIREMENTS

- ❖ Effective and efficient data communication and networking facilities are vital to any enterprise.
- ❖ In this section, we first look at trends that are increasing the challenge for the business manager in planning and managing such facilities.
- ❖ Then we look specifically at the requirement for ever-greater transmission speeds and network capacity.
- ❖ Momentous changes in the way organizations do business and process information have been driven by changes in networking technology and at the same time have driven those changes.
- ❖ It is hard to separate chicken and egg in this field.

DATA TRANSMISSION & NETWORK CAPACITY REQUIREMENTS

- Similarly, the use of the Internet by both businesses and individuals reflects this cyclic dependency:
- The availability of new image-based services on the Internet (Web) has resulted in an increase in the total number of users and the traffic volume generated by each user.
- This, in turn, has resulted in a need to increase the speed and efficiency of the Internet.
- On the other hand, it is only such increased speed that makes the use of Web-based applications palatable to the end user.

DATA TRANSMISSION & NETWORK CAPACITY REQUIREMENTS

- **Emergence of High-Speed LANs**

- PCs, microcomputer workstations begin getting widespread acceptance in business computing in early 1980s & have achieved status of a telephone: an essential tool for office workers.
- Office LANs provided basic connectivity services - connecting personal computers and terminals to mainframes and midrange systems that ran corporate applications, and providing workgroup connectivity at the departmental or divisional level.
- In both cases, traffic patterns were relatively light, with an emphasis on file transfer and electronic mail.

DATA TRANSMISSION & NETWORK CAPACITY REQUIREMENTS

- ❖ LANs available for this type of workload such as Ethernet and token ring, are well suited to this environment.
- ❖ In the last 20 years, two significant trends altered the role of the personal computer and therefore the requirements on the LAN:

1. The speed and computing power of personal computers continued to enjoy explosive growth. These more powerful platforms support graphics-intensive applications and ever more elaborate graphical user interfaces to the operating system.
2. Management information system (MIS) organizations have recognized the LAN as a viable and essential computing platform, resulting in the focus on network computing.

DATA TRANSMISSION & NETWORK CAPACITY REQUIREMENTS

- ❖ This trend began with client/server computing, which has become a dominant architecture in the business environment and the more recent Web-focused intranet trend.
- ❖ Both of these approaches involve the frequent transfer of potentially large volumes of data in a transaction-oriented environment.
- ❖ The effect of these trends has been to increase the volume of data to be handled over LANs and, because applications are more interactive, to reduce the acceptable delay on data transfers.
- ❖ The earlier generation of 10-Mbps Ethernets and 16-Mbps token rings was simply not up to the job of supporting these requirements.

DATA TRANSMISSION & NETWORK CAPACITY REQUIREMENTS

- The examples that call for higher-speed LANs are - Centralized server farms, power workgroups and high-speed local backbone.

- **Centralized server farms**

- In many applications, there is a need for user, or client, systems to be able to draw huge amount of data from centralized servers, called server farms.
- E.g. Color publishing operation where the servers contain tens of GBs of image data that must be downloaded to imaging workstations.
- As the performance of the servers themselves has increased, the bottleneck has shifted to the network.

DATA TRANSMISSION & NETWORK CAPACITY REQUIREMENTS

- **Power Workgroups**

- These groups typically consist of a small number of cooperating users who need to draw massive data files across the network.
- E.g. Software development group that tests a new software version, or a computer-aided design (CAD) company that runs new design simulations.
- A large amount of data are distributed to several workstations, processed, and updated at very high speed for multiple iterations.

- **High Speed Local Backbone**

- As processing demand grows, LANs proliferate at a site, and high-speed interconnection is necessary.

DATA TRANSMISSION & NETWORK CAPACITY REQUIREMENTS

❑ Corporate Wide Area Networking Needs

- ❖ In early 1990s, there was emphasis on centralized data processing model in many organizations .
- ❖ Typically, there might be significant computing facilities at a few regional offices, consisting of mainframes or well-equipped midrange systems.
- ❖ These centralized facilities could handle most corporate applications, including basic finance, accounting, and personnel programs, as well as many of the business-specific applications.
- ❖ Smaller, outlying offices could be equipped with terminals or basic PCs linked to one of the regional centers in a transaction-oriented environment.

DATA TRANSMISSION & NETWORK CAPACITY REQUIREMENTS

❑ Digital Electronics

- ❖ The rapid conversion of consumer electronics to digital technology is having an impact on both the Internet and corporate intranets.
- ❖ As these new gadgets come into view and proliferate, they dramatically increase the amount of image and video traffic carried by networks.
- ❖ A related product development is the digital camcorder.
- ❖ This product has made it easier for individuals and companies to make digital video files to be placed on corporate and Internet Web sites, again adding to the traffic burden.
- ❖ Two more examples are digital versatile disks (DVDs) & digital cameras.

CONVERGENCE

- Convergence refers to the merger of previously distinct telephony and information technologies and markets.
- It involves moving voice into a data infrastructure, integrating all the voice and data networks inside a user organization into a single data network infrastructure, and then extending that into the wireless arena.
- The foundation of this convergence is packet-based transmission using the Internet Protocol (IP).
- Convergence increases the function and scope of both the infrastructure and the application base
- The convergence is a three layer model of enterprise communications:

CONVERGENCE

❑ Applications

- ❖ Convergence integrates communications applications, such as voice calling (telephone), voice mail, e-mail, and instant messaging, with business applications, such as workgroup collaboration, customer relationship management and other back-office functions.
- ❖ With convergence, applications provide features that incorporate voice, data, and video in a seamless, organized, and value-added manner.
- ❖ One example is multimedia messaging, which enables a user to employ a single interface to access messages from a variety of sources
- ❖ E.g. Office voice mail, office e-mail, beeper and fax.

CONVERGENCE

❑ Enterprise Services

- ❖ At this level, the manager deals with the information network in terms of the services it provides to support applications.
- ❖ The network manager needs design, maintenance, and support services related to the deployment of convergence-based facilities.
- ❖ Also at this level, network managers deal with the enterprise network as a function-providing system.
- ❖ Such management services may include setting up authentication schemes; capacity management for various users, groups, and applications and QoS provision.

CONVERGENCE

❑ Infrastructure

- ❖ The infrastructure plays an important role in the field of computer networks and data communication.
- ❖ The network & communication infrastructure consists of communication links, LANs, WANs and Internet connections available to the enterprise.
- ❖ Enterprise n/w infrastructure includes private/public cloud connections to data centers which host high-volume data storage & Web services.
- ❖ A key aspect of convergence at this level is the ability to carry voice, image and video over networks that were originally designed to carry data traffic.

A DATA COMMUNICATION MODEL

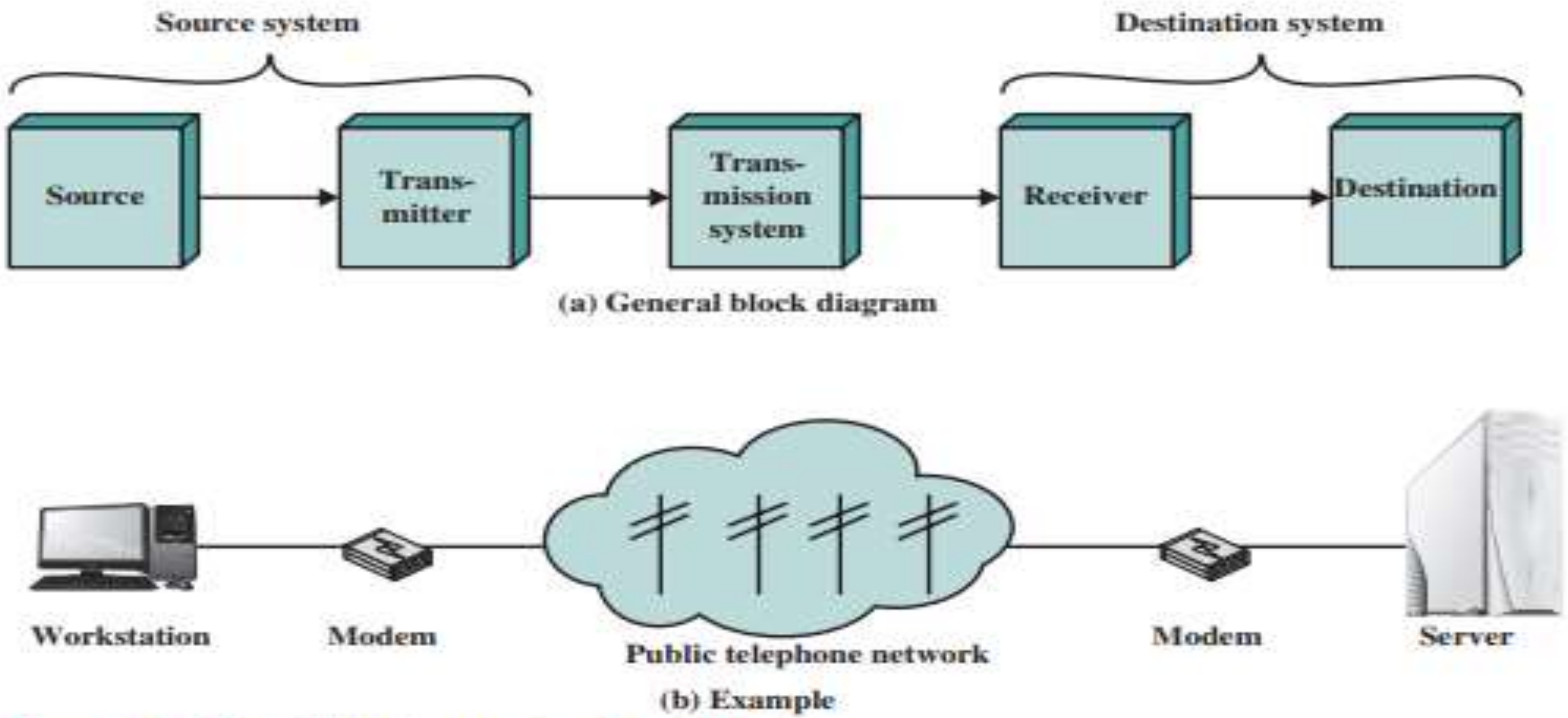


Figure 1.3 Simplified Communications Model

A DATA COMMUNICATION MODEL

- The fundamental purpose of a communications system is the exchange of data between two parties.
- Figure 1.3 presents one particular example, which is communication between a workstation and a server over a public telephone network.
- Another example is the exchange of voice signals between two telephones over the same network.
- The following are key elements of the model:

Source:

- This device generates the data to be transmitted.
- Examples are telephones and personal computers.

A DATA COMMUNICATION MODEL

- **Transmitter:**

- Usually, the data generated by a source system are not transmitted directly in the form in which they were generated.
- Rather, a transmitter transforms and encodes the information in such a way as to produce electromagnetic signals that can be transmitted across some sort of transmission system.
- For example, a modem takes a digital bit stream from an attached device such as a personal computer and transforms that bit stream into an analog signal that can be handled by the telephone network.

A DATA COMMUNICATION MODEL

- **Transmission system:**
  - This can be a single transmission line or a complex network connecting source and destination.
- **Receiver:**
  - The receiver accepts the signal from the transmission system and converts it into a form that can be handled by the destination device.
  - For example, a modem will accept an analog signal coming from a network or transmission line and convert it into a digital bit stream.

A DATA COMMUNICATION MODEL

- **Destination**
  - Takes the incoming data from the receiver.

Table 1.1 Communications Tasks

Transmission system utilization	Addressing
Interfacing	Routing
Signal generation	Recovery
Synchronization	Message formatting
Exchange management	Security
Error detection and correction	Network management
Flow control	

THE TRANSMISSION OF DATA COMMUNICATION

❑ The Transmission of Information

- ❖ The basic building block of any enterprise network infrastructure is the transmission line.
- ❖ Much of the technical detail of how information is encoded and transmitted across a line is of no real interest to the business manager.
- ❖ The manager is concerned with whether the particular facility provides the required capacity, with acceptable reliability, at minimum cost.
- ❖ However, there are certain aspects of transmission technology that a manager must understand to ask the right questions and make informed decisions.

THE TRANSMISSION OF DATA COMMUNICATION

- A basic choice faced by the business user is about transmission medium.
- For use within the business premises, this choice is generally completely up to the business.
- For long-distance communications, the choice is generally but not always made by the long-distance carrier.
- In either case, changes in technology are rapidly changing the mix of media used.
- Of particular note are fiber optic transmission and wireless transmission (e.g., satellite and cellular communications).
- These media are driving evolution of data communications transmission.

THE TRANSMISSION OF DATA COMMUNICATION

- Despite the growth in the capacity and the drop in cost of transmission facilities, transmission services remain the most costly component of a communications budget for most businesses.
- Thus, the manager needs to be aware of techniques that increase the efficiency of the use of these facilities.
- Major approaches to greater efficiency are multiplexing compression.
- Multiplexing - Ability of a no. of devices to share a transmission facility.
- If each device needs the facility only a fraction of the time, then a sharing arrangement allows the cost of the facility to be spread over many users.

THE TRANSMISSION OF DATA COMMUNICATION

- Compression, as the name indicates, involves squeezing the data down so that a lower-capacity, cheaper transmission facility can be used to meet a given demand.
- These two techniques show up separately and in combination in a number of types of communications equipment.
- Manager needs to understand the technologies to assess appropriateness and cost-effectiveness of the various products on the market.

THE TRANSMISSION OF DATA COMMUNICATION

❑ Transmission and Transmission Media

- ❖ Information is communicated by converting it into an electromagnetic signal and transmitting that signal over some medium, such as a twisted pair telephone line.
- ❖ The most commonly used transmission media are twisted-pair lines, coaxial cable, optical fiber cable, and terrestrial and satellite microwave.
- ❖ The data rates that can be achieved and the rate at which errors can occur depend on the nature of the signal and the type of medium.

THE TRANSMISSION OF DATA COMMUNICATION

❑ Communication Techniques

- ❖ The transmission of information across a transmission medium involves more than simply inserting a signal on the medium.
- ❖ The technique used to encode the information into an electromagnetic signal must be determined.
- ❖ There are various ways in which the encoding can be done, and the choice affects performance and reliability.
- ❖ Furthermore, the successful transmission of information involves a high degree of cooperation between the various components.

THE TRANSMISSION OF DATA COMMUNICATION

- The interface between a device and the transmission medium must be agreed on.
- Some means of controlling the flow of information and recovering from its loss or corruption must be used.
- These latter functions are performed by a data link control protocol.

THE TRANSMISSION OF DATA COMMUNICATION

❑ Transmission Efficiency

- ❖ Major cost in any computer/communication facility is transmission cost.
- ❖ Hence, it is important to maximize the amount of information that can be carried over a given resource or to minimize transmission capacity needed to satisfy a given information communications requirement.
- ❖ Two ways of achieving this objective are multiplexing and compression.
- ❖ The two techniques can be used separately or in combination.
- ❖ Three most common multiplexing techniques must be examined namely frequency division, synchronous time division, and statistical time division, as well as the important compression techniques.

TYPES OF NETWORKS

1) Wide Area Networks

- Wide area networks generally cover a large geographical area.
- They often require a crossing of public right of ways & typically rely at least in part on circuits provided by one or more common carriers.
- The companies that offer communication services to the general public.
- WAN consists of a number of interconnected switching nodes.
- A transmission from any one device is routed through these internal nodes to the specified destination device.
- These nodes including the boundary nodes are not concerned with the content of the data.

TYPES OF NETWORKS

- ❖ Rather, their purpose is to provide a switching facility that will move the data from node to node until they reach their destination.
- ❖ WANs have been implemented using one of two technologies - circuit switching and packet switching.
- ❖ Subsequently, frame relay and ATM networks assumed major roles.
- ❖ While ATM and, to some extent frame relay, are still widely used, their use is gradually being supplanted by services based on gigabit Ethernet and Internet Protocol technologies.
- ❖ Let us discuss circuit switching and packet switching techniques as follows:

TYPES OF NETWORKS

- **Circuit Switching**
- A dedicated communication path is established between two stations through the nodes of the network in a circuit switching network.
- That path is a connected sequence of physical links between nodes.
- On each link, a logical channel is dedicated to the connection.
- Data generated by the source station are transmitted along the dedicated path as rapidly as possible.
- At each node, incoming data are routed or switched to the appropriate outgoing channel without delay.
- Most common example of circuit switching is the telephone network.

TYPES OF NETWORKS

❑ Packet Switching

- ❖ In packet-switching network, it’s not necessary to dedicate transmission capacity along a path through the network.
- ❖ Rather, data are sent out in a sequence of small chunks, called packets.
- ❖ Each packet is passed through the network from node to node along some path leading from source to destination.
- ❖ At each node, the entire packet is received, stored briefly, and then transmitted to the next node.
- ❖ Packet-switching networks are commonly used for terminal-to-computer and computer-to-computer communications.

TYPES OF NETWORKS

❑ Frame Relay

- ❖ Packet switching was developed when digital long-distance transmission facilities had a higher error rate as compared to today's facilities.
- ❖ As a result, there is a considerable amount of overhead built into packet switching schemes to compensate for errors.
- ❖ The overhead includes additional bits added to each packet to introduce redundancy & additional processing at the end stations & intermediate switching nodes to detect and recover from errors.
- ❖ With modern high-speed telecommunications systems, this overhead is unnecessary and counterproductive.

TYPES OF NETWORKS

- It is unnecessary because the rate of errors has been dramatically lowered and any remaining errors can easily be caught in the end systems by logic that operates above the level of the packet-switching logic.
- It is counterproductive as the overhead involved soaks up a significant fraction of the high capacity provided by the network.
- Frame relay can take advantage of high data rates & low error rates.
- Whereas the original packet-switching networks were designed with a data rate to the end user of about 64 kbps, frame relay networks are designed to operate efficiently at user data rates of up to 2 Mbps.
- High data rates are achieved by stripping the overhead with error control.

TYPES OF NETWORKS

- **Asynchronous Transfer Mode (ATM)**
- Asynchronous transfer mode, sometimes referred to as cell relay, is a culmination of developments in circuit switching and packet switching.
- ATM can be viewed as an evolution from frame relay.
- **Frame Relay vs. ATM:**
- The frame relay uses variable-length packets called frames while the ATM uses fixed-length packets called cells.
- As with frame relay, ATM provides little overhead for error control.
- By using a fixed packet length, the processing overhead is reduced even further for ATM compared to frame relay.

TYPES OF NETWORKS

- In general, ATM is designed to work in the ranges of Tens and Hundreds of MBPS and GBPS.
- ATM can also be viewed as an evolution from circuit switching.
- With circuit switching, fixed-data-rate circuits are available to end system.
- ATM allows the definition of multiple virtual channels with data rates that are dynamically defined at the time the virtual channel is created.
- With small, fixed-size cells, ATM is so efficient that it can offer a constant-data-rate channel even though it is using a packet-switching technique.
- Thus, ATM extends circuit switching to allow multiple channels with the data rate on each channel dynamically set on demand.

TYPES OF NETWORKS

2) Local Area Networks (LANs)

- ❖ LAN is a communications network that interconnects a variety of devices and provides a means for information exchange among those devices.
- ❖ There are several key distinctions between LANs and WANs:
- ➢ Scope of LAN is small, typically a single building or a cluster of buildings.
- ➢ This difference in geographic scope leads to different technical solutions.
- ➢ It is usually the case that the LAN is owned by the same organization that owns the attached devices.
- ❖ For WANs, this is less often the case, or at least a significant fraction of the network assets is not owned. This has two implications.

TYPES OF NETWORKS

- First, care must be taken in the choice of LAN, because there may be a substantial capital investment (compared to dial-up or leased charges for WANs) for both purchase and maintenance. Second, the network management responsibility for a LAN falls solely on the user.
- Internal data rates of LANs are typically much greater than those of WANs.
- LANs come in a number of different configurations.
- The most common are switched LANs and wireless LANs.
- The most common switched LAN is a switched Ethernet LAN, which has a single switch with a no. of attached devices or interconnected switches.
- The most common type of wireless LANs are Wi-Fi LANs.

TYPES OF NETWORKS

3) Wireless Networks

- As mentioned, wireless LANs are widely used in business environments.
- Wireless technology is common for both the wide area voice as well as data networks.
- Wireless networks provide advantages in the areas of mobility and ease of installation and configuration.

THE INTERNET

❑ Origin of Internet

- ❖ The Internet evolved from the ARPANET
- ❖ The ARPANET was developed in the year 1969 by Advanced Research Projects Agency (ARPA) of U.S. Department of Defense.
- ❖ It was the first operational packet-switching network.
- ❖ The ARPANET began operations in four locations.
- ❖ Today the no. of hosts is in the thousands of millions, the no. of users in the billions & the number of countries participating nearing 200.
- ❖ The no. of connections to the Internet continues to grow exponentially along with the websites.

THE INTERNET

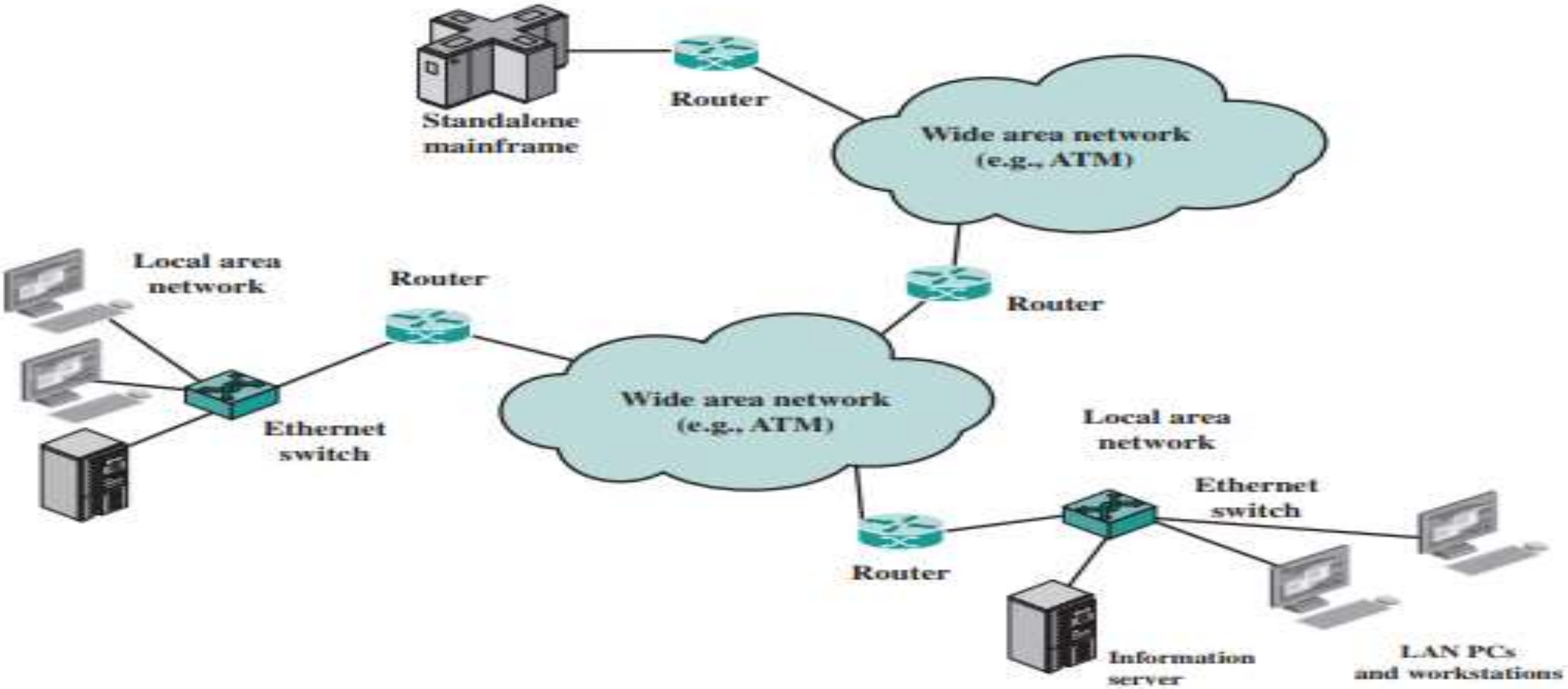


Figure 1.5 Key Elements of the Internet

THE INTERNET

❑ Key Elements

- ❖ Figure 1.5 illustrates the key elements that comprise the Internet.
- ❖ The purpose of Internet is to interconnect end systems called hosts which include PCs, workstations, servers, mainframes, and so on.
- ❖ Most hosts that use the Internet are connected to a network, such as a local area network or a wide area network (WAN).
- ❖ These networks are in turn connected by routers. Each router attaches to two or more networks.
- ❖ Some hosts, such as mainframes or servers, connect directly to a router rather than through a network.

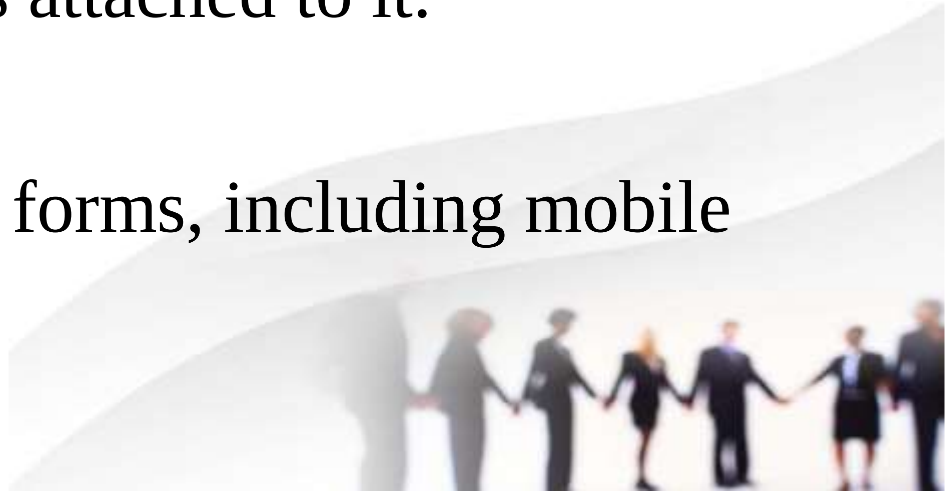
THE INTERNET

- ❖ In essence, the Internet operates as follows:
- ❖ A host may send data to another host anywhere on the Internet.
- ❖ The source host breaks the data to be sent into a sequence of packets, called IP datagrams or IP packets.
- ❖ Each packet includes a unique numeric address of the destination host.
- ❖ This address is known as an IP address since it is carried in an IP packet.
- ❖ Based on this destination address, each packet travels through a series of routers and networks from source to destination.
- ❖ Each router, as it receives a packet, makes a routing decision and forwards the packet along its way to the destination.

THE INTERNET

❑ Internet Architecture

- ❖ Internet is made up of thousands of overlapping hierarchical networks.
- ❖ Because of this, it is not practical to attempt a detailed description of the exact architecture or topology of the Internet.
- ❖ An overview of the common, general characteristics can be made.
- ❖ A key element of the Internet is the set of hosts attached to it.
- ❖ Simply put, a host is a computer.
- ❖ In today's world, the computers come in many forms, including mobile phones and even cars.
- ❖ All of these forms can be hosts on the Internet.



THE INTERNET

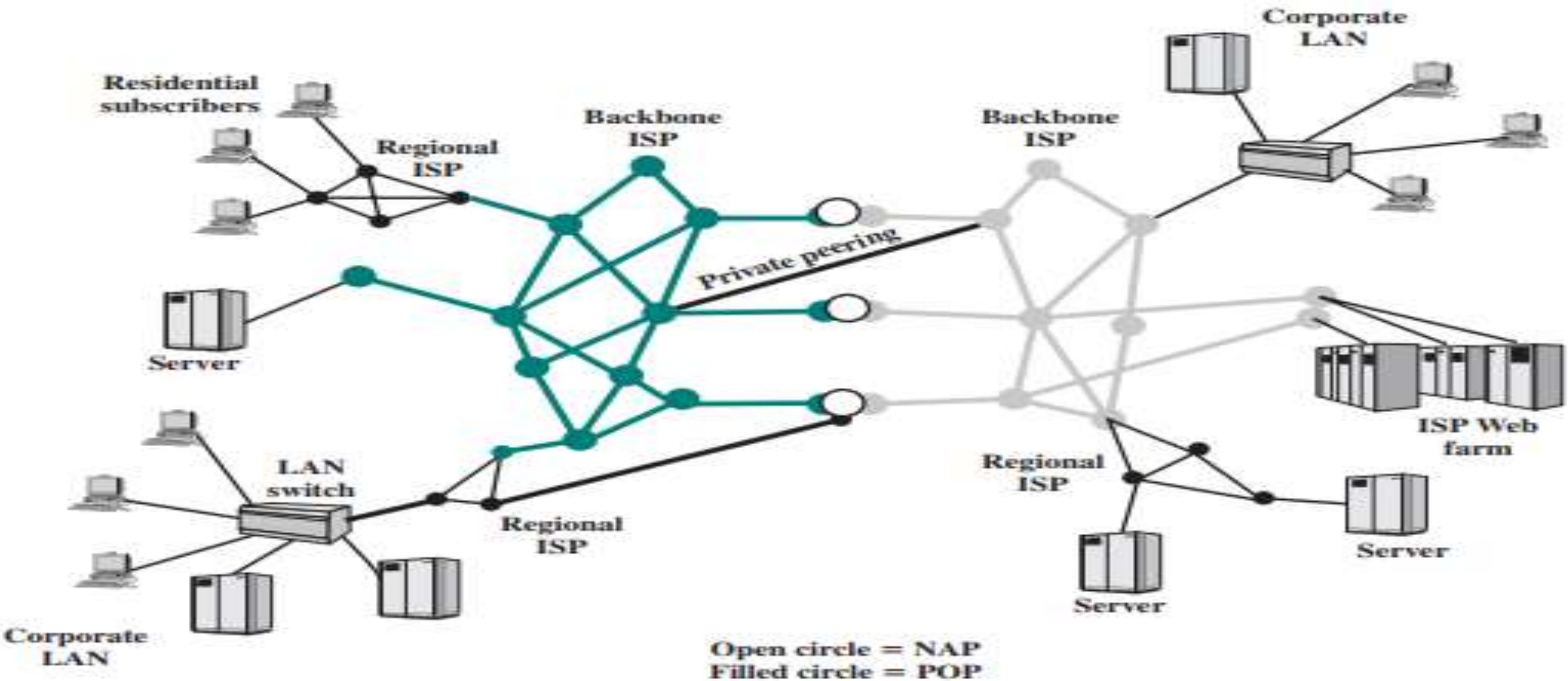


Figure 1.6 Simplified View of Portion of Internet

THE INTERNET

- Figure 1.6 given above illustrates the discussion and Table 1.2 given below summarizes the terminology.
- Hosts are sometimes grouped together in a LAN.
- This is the typical configuration in a corporate environment.
- Individual hosts and LANs are connected to an Internet service provider (ISP) through a point of presence (POP).
- The connection is made in a series of steps starting with the customer premises equipment (CPE).
- The CPE is the communications equipment located onsite with the host.

THE INTERNET

Table 1.2 Internet Terminology

Central Office (CO)

The place where telephone companies terminate customer lines and locate switching equipment to interconnect those lines with other networks.

Customer Premises Equipment (CPE)

Telecommunications equipment that is located on the customer's premises (physical location) rather than on the provider's premises or in between. Telephone handsets, modems, cable TV set-top boxes, and digital subscriber line routers are examples. Historically, this term referred to equipment placed at the customer's end of the telephone line and usually owned by the telephone company. Today, almost any end-user equipment can be called customer premises equipment and it can be owned by the customer or by the provider.

Internet Service Provider (ISP)

A company that provides other companies or individuals with access to, or presence on, the Internet. An ISP has the equipment and the telecommunication line access required to have a POP on the Internet for the geographic area served. The larger ISPs have their own high-speed leased lines so that they are less dependent on the telecommunication providers and can provide better service to their customers.

Network Access Point (NAP)

In the United States, a network access point (NAP) is one of several major Internet interconnection points that serve to tie all the ISPs together. Originally, four NAPs—in New York, Washington, D.C., Chicago, and San Francisco—were created and supported by the National Science Foundation as part of the transition from the original U.S. government–financed Internet to a commercially operated Internet. Since that time, several new NAPs have arrived, including WorldCom's "MAE West" site in San Jose, California, and ICS Network Systems' "Big East."

The NAPs provide major switching facilities that serve the public in general. Companies apply to use the NAP facilities. Much Internet traffic is handled without involving NAPs, using peering arrangements and interconnections within geographic regions.

Network Service Provider (NSP)

A company that provides backbone services to an Internet service provider. Typically, an ISP connects at a point called an Internet exchange (IX) to a regional ISP that in turn connects to an NSP backbone.

Point of Presence (POP)

A site that has a collection of telecommunications equipment, usually refers to ISP or telephone company sites. An ISP POP is the edge of the ISP's network; connections from users are accepted and authenticated here. An Internet access provider may operate several POPs distributed throughout its area of operation to increase the chance that their subscribers will be able to reach one with a local telephone call. The largest national ISPs have POPs all over the country.

TRADITIONAL INTERNET BASED APPLICATIONS

- A no. of applications have been standardized to operate on top of TCP.
- We mention three of the most common here.
- The Simple Mail Transfer Protocol (SMTP) provides a basic electronic mail transport facility.
- It provides mechanism for transferring messages among separate hosts.
- Features of SMTP include mailing lists, return receipts, and forwarding.
- SMTP does not specify the way in which messages are to be created.
- Some local editing or native electronic mail facility is required.
- Once a message is created, SMTP accepts the message and makes use of TCP to send it to an SMTP module on another host.

TRADITIONAL INTERNET BASED APPLICATIONS

- The target SMTP module will make use of a local electronic mail package to store the incoming message in a user’s mailbox.
- The File Transfer Protocol (FTP) is used to send files from one system to another under user command.
- Both text and binary files are accommodated, and the protocol provides features for controlling user access.
- When a user wishes to engage in file transfer, FTP sets up a TCP connection to the target system for the exchange of control messages.
- This connection allows user ID and password to be transmitted and allows the user to specify the file and file actions desired.

TRADITIONAL INTERNET BASED APPLICATIONS

- ❖ Once a file transfer is approved, a second TCP connection is set up for the data transfer.
- ❖ The file is transferred over the data connection, without the overhead of any headers or control information at the application level.
- ❖ When the transfer is complete, the control connection is used to signal the completion and to accept new file transfer commands.
- ❖ SSH (Secure Shell) provides a secure remote logon capability, which enables a user at a terminal or personal computer to log on to a remote computer and function as if directly connected to that computer.
- ❖ SSH also supports file transfer between a local host and a remote server.

TRADITIONAL INTERNET BASED APPLICATIONS

- SSH enables the user and the remote server to authenticate each other.
- It also encrypts all traffic in both directions.
- SSH traffic is carried on a TCP connection.

MULTIMEDIA

- With increasing availability of broadband access to Internet has come an increased interest in web-based/Internet-based multimedia applications.
- The terms multimedia and multimedia applications are used rather loosely in the literature and in commercial publications, and no single definition of the term multimedia has been agreed.
- One way to organize the concepts associated with multimedia is to look at a taxonomy that captures a number of dimensions of this field.
- For our purpose, definitions in Table 2.2 provide a starting point while the figure 2.11 looks at multimedia from three different dimensions: type of media, applications & technology required to support the applications.

MULTIMEDIA

Table 2.2 Multimedia Terminology

Media
Refers to the form of information and includes text, still images, audio, and video.
Multimedia
Human–computer interaction involving text, graphics, voice, and video. Multimedia also refers to storage devices that are used to store multimedia content.
Streaming media
Refers to multimedia files, such as video clips and audio, that begin playing immediately or within seconds after it is received by a computer from the Internet or Web. Thus, the media content is consumed as it is delivered from the server rather than waiting until an entire file is downloaded.

MULTIMEDIA

❑ Media Types

- ❖ Typically, the term multimedia refers to four distinct types of media: text, audio, graphics, and video.
- ❖ From a communications perspective, the term text is self-explanatory, referring to information that can be entered via a keyboard and is directly readable and printable.
- ❖ Text messaging, instant messaging, and text (non-html) e-mail are common examples, as are chat rooms and message boards.
- ❖ It is used in the broader sense of data that can be stored in files and databases and that does not fit into the other three categories.

MULTIMEDIA

- ❖ E.g. An organization’s database may contain files of numerical data, in which data are stored in a more compact form than printable characters.
- ❖ The term audio generally encompasses two different ranges of sound.
- ❖ Voice or speech, refers to sounds produced by human speech mechanism.
- ❖ A modest bandwidth (under 4 kHz) is required to transmit voice.
- ❖ Telephony and related applications (voice mail, audio teleconferencing and telemarketing) are the most common traditional applications of voice communications technology.
- ❖ A broader frequency spectrum is needed to support music applications, including the download of music files.

MULTIMEDIA

- The image service supports the communication of individual pictures, charts, or drawings.
- Image-based applications include facsimile, computer-aided design (CAD), publishing, and medical imaging.
- Images can be represented in a vector graphics format, such as is used in drawing programs and PDF files.
- Image is represented as a two-dimensional array of spots, called pixels.
- The compressed JPG format is derived from a raster graphics format.
- The video service carries sequences of pictures in time.
- In essence, video makes use of a sequence of raster-scan images.

MULTIMEDIA

❑ Multimedia Applications

- ❖ The Internet, until recently, has been dominated by information retrieval applications, e-mail, and file transfer, plus Web interfaces that emphasized text and images.
- ❖ Internet is being used for multimedia applications that involve massive amounts of data for visualization and support of real-time interactivity.
- ❖ Streaming audio and video are perhaps the best known of such kind of applications.
- ❖ An example of interactive application is a virtual training environment involving distributed simulations and real-time user interaction.

MULTIMEDIA

Table 2.3 Domains of Multimedia Systems and Example Applications

Domain	Example Application
Information management	Hypermedia, multimedia-capable databases, content-based retrieval
Entertainment	Computer games, digital video, audio (MP3)
Telecommunication	Videoconferencing, shared workspaces, virtual communities
Information publishing/delivery	Online training, electronic books, streaming media

MULTIMEDIA

❑ Multimedia Applications

❖ There are following multimedia application domains:

➢ **Information systems:**

❖ These applications present information using multimedia.

❖ Examples include information kiosks, electronic books that include audio and video, and multimedia expert systems.

➢ **Communication systems:**

❖ These applications support collaborative work mostly.

❖ E.g. Video conferencing.

MULTIMEDIA

- **Entertainment systems:**

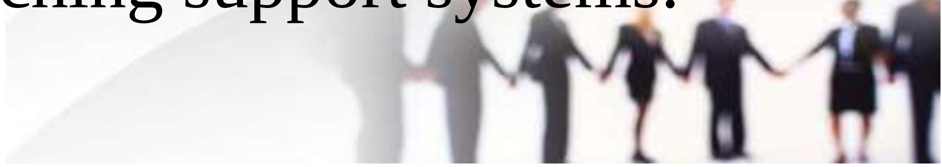
❖ These applications include computer and network games and other forms of audiovisual entertainment.

- **Business systems:**

❖ These applications include business-oriented multimedia presentation, video brochures and online shopping.

- **Educational systems:**

❖ The applications include e-books with a multimedia component, simulation and modeling applets, and other teaching support systems.



MULTIMEDIA

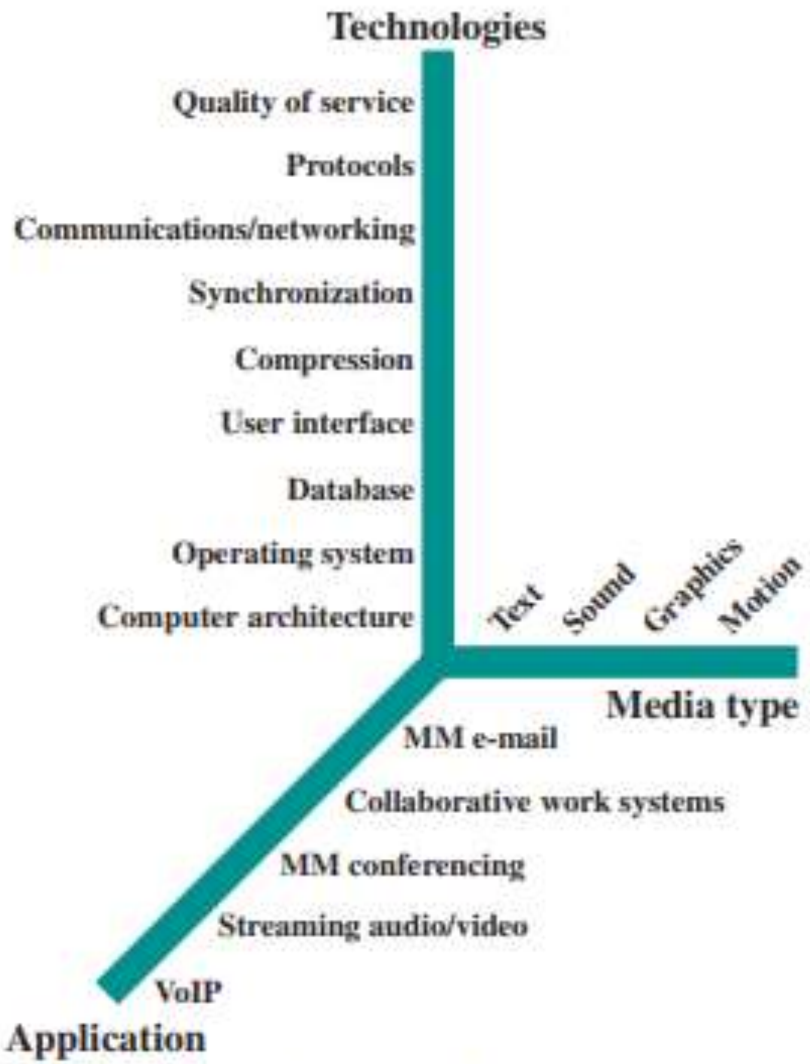


Figure 2.11 A Multimedia Taxonomy

MULTIMEDIA

- ❖ Figure 2.11 lists some of the technologies that are relevant to the support of multimedia applications.
- ❖ As can be seen, a wide range of technologies is involved.
- ❖ The lowest four items on list are beyond the scope of this book.
- ❖ The other items represent only a partial list of communications and networking technologies for multimedia.

➢ **Compression:**

- ❖ Digitized video, and to a much lesser extent audio, can generate an enormous amount of traffic on a network.
- ❖ A streaming application delivered to many users, magnifies the traffic.

MULTIMEDIA

- ❖ Accordingly, standards have been developed for producing significant savings through compression.
- ❖ The most notable standards are JPG for still images and MPG for video.

- ➢ **Communications / networking:**

- ❖ This category refers to the transmission & networking technologies (e.g. SONET, ATM) that can support high volume multimedia traffic.

- ➢ **Protocols:**

- ❖ A number of protocols are instrumental in supporting multimedia traffic.
- ❖ One example is the Real-time Transport Protocol (RTP).

MULTIMEDIA

- **Quality of service (QoS):**
  - The Internet and its underlying local area and wide area networks must include a QoS capability to provide differing levels of service to the different types of application traffic.
  - A QoS capability can deal with priority, delay constraints, delay variability constraints, and other similar requirements.

SOCKET PROGRAMMING

- The concept of sockets and sockets programming was developed in the 1980s in the UNIX environment as the Berkeley Sockets Interface.
- In essence, a socket enables communication between a client and server process and may be either connection oriented or connectionless.
- A socket can be considered an end point in a communication.
- A client socket in one computer uses an address to call a server socket on another computer.
- Once appropriate sockets are engaged, the computers can exchange data.
- Typically, computers with server sockets keep a TCP or UDP port open, ready for unscheduled incoming calls.

SOCKET PROGRAMMING

- The client typically determines the socket identification of the desired server by finding it in a Domain Name System (DNS) database.
- Once connection is made, server switches the dialogue to different port number to free up the main port number for additional incoming calls.
- Internet applications, such as TELNET and remote login (rlogin), make use of sockets, with the details hidden from the user.
- The sockets can be constructed from within a program (in a language such as C, Java, or Python), enabling the programmer to easily support networking functions and applications.

SOCKET PROGRAMMING

- The socket is used to define an API, which is a generic communication interface for writing programs that use TCP or UDP.
- In practice, when used as an API, a socket is identified by the triple which is protocol, local address, local process.
- The local address is an IP address and the local process is a port number.
- Because port numbers are unique within a system, the port number implies the protocol (TCP or UDP).
- For clarity & ease of implementation, the API sockets include protocol as well as the IP address and port number in defining a unique socket.

THE NEED FOR A PROTOCOL ARCHITECTURE

- When computers, terminals, and/or other data processing devices exchange data, the procedures involved can be quite complex.
- Consider, for example, the transfer of a file between two computers.
- There must be a data path between the two computers, either directly or via a communication network.
- Typical tasks to be performed are as follows:

1. The source system must either activate the direct data communication path or inform the communication network of the identity of the desired destination system.
2. Source system to ensure destination system is prepared to receive data.

THE NEED FOR A PROTOCOL ARCHITECTURE

3. The file transfer application on the source system must ascertain that the file management program on the destination system is prepared to accept and store the file for this particular user.
4. If the file formats used on the two systems are different, one or the other system must perform a format translation function.

- ❖ It is clear that there must be a high degree of cooperation between the two computer systems.
- ❖ Instead of implementing the logic for this as a single module, the task is broken up into subtasks, each of which is implemented separately.
- ❖ In a protocol architecture, the modules are arranged in a vertical stack.

THE NEED FOR A PROTOCOL ARCHITECTURE

- Each layer in the stack performs a related subset of the functions required to communicate with another system.
- It relies on the next lower layer to perform more primitive functions and to conceal the details of those functions.
- It provides services to the next higher layer with layers to be defined so that changes in one layer do not require changes in other layers.
- Of course, it takes two to communicate, so the same set of layered functions must exist in two systems.
- Communication is achieved by having the corresponding, or peer, layers in two systems communicate.

THE NEED FOR A PROTOCOL ARCHITECTURE

- The peer layers communicate by means of formatted blocks of data that obey a set of rules or conventions known as a protocol.
- The key features of a protocol are as follows:

• Syntax: Concerns the format of the data blocks

• Semantics:

- Includes control information for coordination and error handling

• Timing:

- Includes speed matching and sequencing.

THE TCP/IP PROTOCOL ARCHITECTURE

- TCP/IP is a result of protocol research and development conducted on the experimental packet-switched network, ARPANET, funded by the Defense Advanced Research Projects Agency (DARPA) & is generally referred to as the TCP/IP protocol suite.
- This protocol suite consists of a large collection of protocols that have been issued as Internet standards by the Internet Activities Board (IAB).
- The TCP/IP Layers In general terms, computer communications can be said to involve three agents: applications, computers, and networks.
- The applications that we are concerned with are distributed applications that involve the exchange of data between two computer systems.

THE TCP/IP PROTOCOL ARCHITECTURE

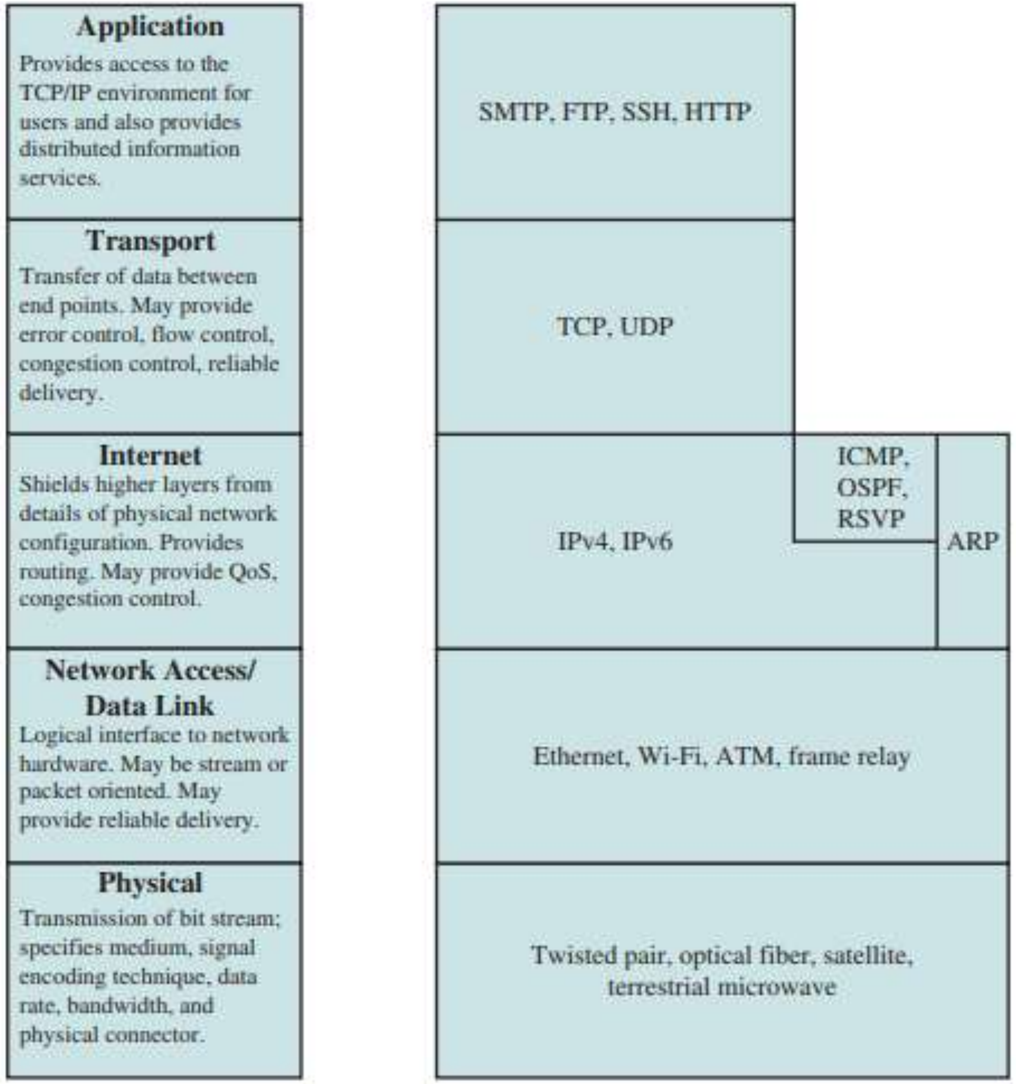


Figure 2.3 The TCP/IP Layers and Example Protocols

THE TCP/IP PROTOCOL ARCHITECTURE

- ❖ These applications, and others, execute on computers that can often support multiple simultaneous applications.
- ❖ Computers are connected to networks, and the data to be exchanged are transferred by the network from one computer to another.
- ❖ Thus, the transfer of data from one application to another involves first getting the data to the computer in which the application resides and then getting the data to the intended application within the computer.
- ❖ With these concepts in mind, we can organize the communication task into five relatively independent layers:
  - Physical layer

THE TCP/IP PROTOCOL ARCHITECTURE

- Network access/data link layer
- Internet layer
- Host-to-host, or transport layer
- Application layer

❖ The physical layer covers physical interface between a data transmission device (e.g., workstation, computer) & transmission medium or network.

❖ This layer is concerned with specifying characteristics of transmission medium, the nature of the signals, the data rate, and related matters.

❖ This layer is concerned with access to and routing data across a network for two end systems attached to the same network.

THE TCP/IP PROTOCOL ARCHITECTURE

- Wherever two devices are attached to different networks, the procedures are needed to allow data to traverse multiple interconnected networks.
- This is the function of the internet layer.
- Internet Protocol (IP) provides routing function across multiple networks.
- This protocol is implemented not only in end systems but also in routers.
- A router is a processor that connects two networks and whose primary function is to relay data from one network to the other on its route from the source to the destination end system.
- The host-to-host layer or transport layer, may provide reliable end-to-end service or end-to-end delivery service without reliability mechanisms.

THE TCP/IP PROTOCOL ARCHITECTURE

- The Transmission Control Protocol (TCP) is the most commonly used protocol to provide this functionality.
- Finally, the application layer contains the logic needed to support the various user applications.
- For each different type of application, such as file transfer, a separate module is needed that is peculiar to that application.

THE TCP/IP PROTOCOL ARCHITECTURE

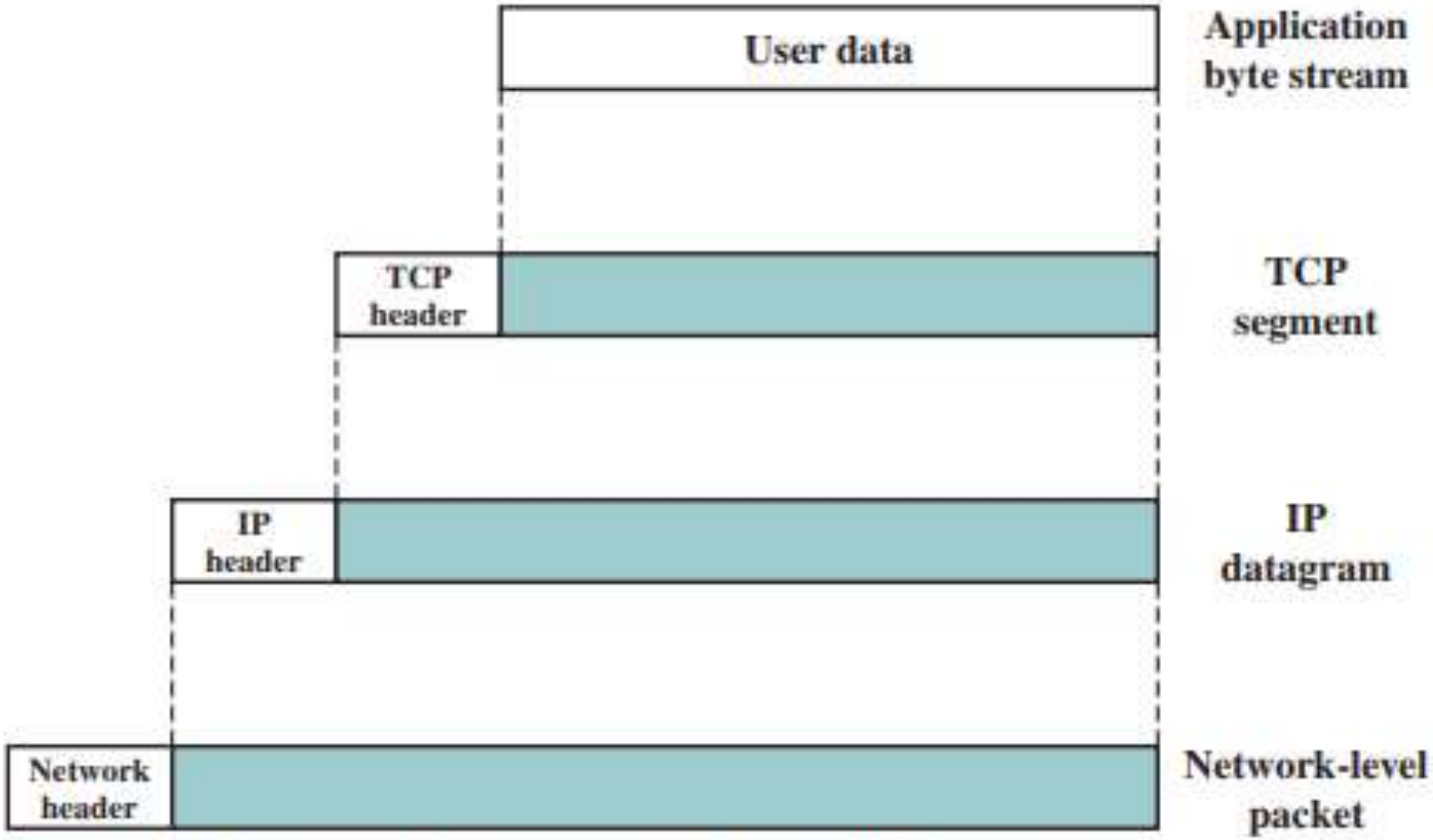


Figure 2.5 Protocol Data Units (PDUs) in the TCP/IP Architecture

THE TCP/IP PROTOCOL ARCHITECTURE

❑ Operation of TCP and IP

- ❖ To control this operation, control information, as well as user data, must be transmitted, as suggested in Figure 2.5.
- ❖ Let us say that the sending process generates a block of data and passes this to TCP.
- ❖ TCP may break a block into smaller pieces to make it more manageable.
- ❖ To each of these pieces, TCP appends control information known as the TCP header, forming a TCP segment.
- ❖ The control information is to be used by the peer TCP entity at host B.
- ❖ Examples of items in this header include:

THE TCP/IP PROTOCOL ARCHITECTURE

- **Destination port:**

❖ When the TCP entity at B receives the segment, it must know to whom the data are to be delivered.

- **Sequence number:**

❖ TCP numbers sends to a particular destination port sequentially, so that if they arrive out of order, the TCP entity at B can reorder them.

- **Checksum:**

❖ The sending TCP includes a code that is a function of the contents of the remainder of the segment.

THE TCP/IP PROTOCOL ARCHITECTURE

- The receiving TCP performs the same calculation and compares a result with the incoming code.
- A discrepancy results if there has been some error in transmission.
- TCP hands each segment over to IP, with instructions to transmit it to B.
- These segments must be transmitted across one or more subnetworks and relayed through one or more intermediate routers.
- This operation, too, requires the use of control information.
- IP appends control information header to segments to form IP datagram.
- An example of an item stored in the IP header is the destination host address (in this example, B).

UNIT II

DATA TRANSMISSION

BASIC TERMINOLOGIES

- Data transmission occurs between transmitter and receiver over some transmission medium.
- Transmission media may be classified as guided or unguided.
- In both cases, communication is in the form of electromagnetic waves.
- With guided media, the waves are guided along a physical path.
- E.g. twisted pair, coaxial cable, and optical fiber.
- Unguided media, also called wireless, provide a means for transmitting electromagnetic waves but do not guide them;
- E.g. Propagation through air, vacuum and seawater.

BASIC TERMINOLOGIES

- The term direct link is used to refer to the transmission path between two devices in which signals propagate directly from transmitter to receiver
- However, there should not be any intermediate devices, other than amplifiers or repeaters used to increase signal strength.
- Note that this term can apply to both guided and unguided media.
- A guided transmission medium is point to point if it provides a direct link between two devices & those are the only two devices sharing the medium.
- In a multipoint guided configuration, more devices share the same medium.
- A transmission may be simplex, half duplex, or full duplex.
- In simplex transmission, signals are transmitted in only one direction.

BASIC TERMINOLOGIES

- In half-duplex, both stations may transmit, but only one at a time.
- In full-duplex operation, both stations may transmit simultaneously.
- The medium is carrying signals in both directions at the same time.

FREQUENCY, SPECTRUM AND BANDWIDTH

- We are more concerned with electromagnetic signals used as a means to transmit data.
- In the Simplified Data Communications Model, we have seen that an analog signal is created after the input data is processed by a transmitter.
- A signal is generated by the transmitter and transmitted over a medium.
- The signal is a function of time, but can be expressed as a function of frequency i.e. signal has components of different frequencies.
- It turns out that the frequency domain view of a signal is more important to an understanding of data transmission than a time domain view.

ANALOG AND DIGITAL DATA

- The concepts of analog and digital data are simple enough.
- Analog data take on continuous values in some interval.
- E.g. Voice and video are continuously varying patterns of intensity.
- Most data collected by sensors are continuous valued.
- Digital data take on discrete values; examples are text and integers.
- The most familiar example of analog data is audio, which, in the form of acoustic sound waves, can be perceived directly by human beings.
- Frequency components of typical speech may be found between approximately 100 Hz and 7 kHz.
- Typical speech has a dynamic range of about 25 dB.

ANALOG AND DIGITAL DATA

- Another common example of analog data is video.
- Easier to characterize the data in terms of the TV screen (destination) rather than the original scene (source) recorded by the TV camera.
- To produce a picture on the screen, an electron beam scans across the surface of the screen from left to right and top to bottom.
- For black-and-white television, the amount of illumination produced (on a scale from black to white) at any point is proportional to the intensity of the beam as it passes that point.
- Thus at any instant in time the beam takes on an analog value of intensity to produce the desired brightness at that point on the screen.

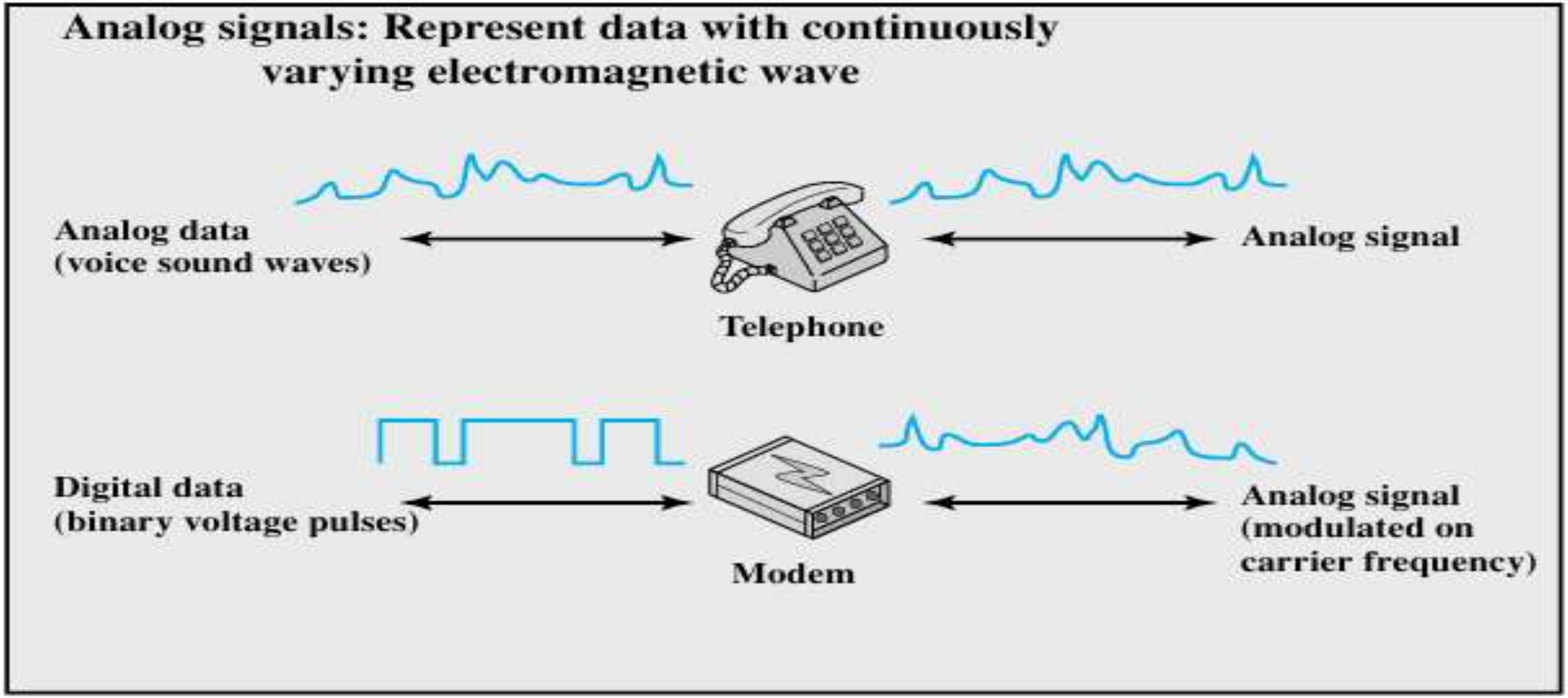
ANALOG AND DIGITAL DATA

- A familiar example of digital data is text or character strings.
- While textual data are most convenient for human beings, they cannot, in character form, be easily stored or transmitted by data processing and communications systems.
- Such systems are designed for binary data.

ANALOG AND DIGITAL SIGNALS

- In a communications system, data are propagated from one point to another by means of electromagnetic signals.
- An analog signal is a continuously varying electromagnetic wave that may be propagated over a variety of media, depending on spectrum.
- E.g. Wired media such as twisted pair, coaxial cable and fiber optic cable and the unguided media such as atmosphere or space propagation.
- A digital signal is a sequence of voltage pulses that may be transmitted over a wire medium
- E.g. A constant positive voltage level may represent binary 0 and a constant negative voltage level may represent binary 1.

ANALOG AND DIGITAL SIGNALS



ANALOG AND DIGITAL SIGNALS

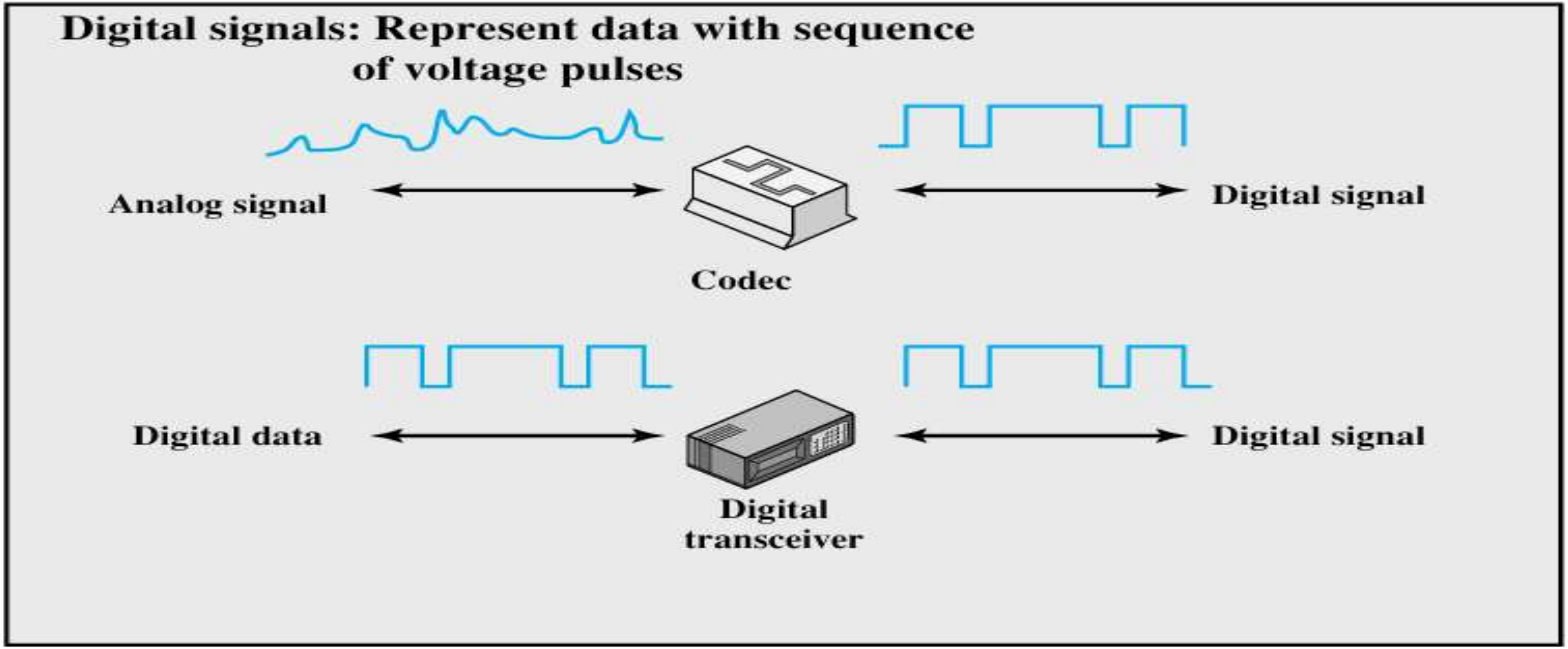


Figure 3.14 Analog and Digital Signaling of Analog and Digital Data

DATA AND SIGNALS

- Both data and signals are the most significant constituents of any kind of communication process.
- In the foregoing discussion, we have looked at analog signals used to represent analog data and digital signals used to represent digital data.
- Generally, analog data are a function of time and occupy a limited frequency spectrum.
- Such data can be represented by an electromagnetic signal occupying the same spectrum.
- Digital data can be represented by digital signals, with a different voltage level for each of the two binary digits.

ANALOG AND DIGITAL TRANSMISSION

- ❖ Analog & digital signals are transmitted on suitable transmission media.
- ❖ Table 3.1 summarizes the methods of data transmission.
- ❖ Analog transmission is a means of transmitting analog signals without regard to their content.
- ❖ The signals may represent analog data (e.g., voice) or digital data (e.g., binary data that pass through a modem).
- ❖ In either case, the analog signal will become weaker (attenuate) after a certain distance.
- ❖ To achieve longer distances, the analog transmission system includes amplifiers that boost the energy in the signal.

ANALOG AND DIGITAL TRANSMISSION

Table 3.1 Analog and Digital Transmission

(a) Data and Signals

	Analog Signal	Digital Signal
Analog Data	Two alternatives: (1) signal occupies the same spectrum as the analog data; (2) analog data are encoded to occupy a different portion of spectrum.	Analog data are encoded using a codec to produce a digital bit stream.
Digital Data	Digital data are encoded using a modem to produce analog signal.	Two alternatives: (1) signal consists of two voltage levels to represent the two binary values; (2) digital data are encoded to produce a digital signal with desir ed properties.

(b) Treatment of Signals

	Analog Transmission	Digital Transmission
Analog Signal	Is propagated through amplifiers; same treatment whether signal is used to represent analog data or digital data.	Assumes that the analog signal represents digital data. Signal is propagated through repeaters; at each repeater, digital data are recovered from inbound signal and used to generate a new analog outbound signal.
Digital Signal	Not used	Digital signal represents a stream of 1s and 0s, which may represent digital data or may be an encoding of analog data. Signal is propagated through repeaters; at each repeater, stream of 1s and 0s is recovered from inbound signal and used to generate a new digital outbound signal.

ANALOG AND DIGITAL TRANSMISSION

- Digital transmission, in contrast, assumes a binary content to the signal.
- Digital signal can be transmitted over only a limited distance before the attenuation, noise & other impairments endanger the integrity of the data.
- To achieve greater distances, repeaters are used.
- A repeater receives the digital signal, recovers the pattern of 1s and 0s, and retransmits a new signal.
- Thus the attenuation is overcome.

ASYNCHRONOUS AND SYNCHRONOUS TRANSMISSION

- The transmission of a stream of bits from one device to another across a transmission link involves a great deal of cooperation and agreement between the two sides.
- One of the most fundamental requirements is synchronization.
- The receiver must know the rate at which bits are being received so that it can sample the line at appropriate intervals to determine the value of each received bit.
- Two techniques are in common use for this purpose:
    - Asynchronous transmission
    - Synchronous transmission

ASYNCHRONOUS AND SYNCHRONOUS TRANSMISSION

- The receiver samples each bit in the character and then looks for the beginning of the next character.
- This technique would not work well for long blocks of data because the receiver’s clock might not be synchronized with the transmitter’s clock.
- However, sending data in large blocks is more efficient than sending data one character at a time.
- For large blocks, synchronous transmission is used.
- Each block of data can be treated as a frame that includes a starting and an ending flag.
- E.g. Manchester encoding

TRANSMISSION IMPAIRMENTS

- ❖ In a communications system, the signal that is received may differ from the signal that is transmitted due to various transmission impairments.
- ❖ For analog signals, these impairments can degrade the signal quality.
- ❖ For digital signals, bit errors may be introduced, such that a binary 1 is transformed into a binary 0 or vice versa.
- ❖ In this section, we examine the various impairments and how they may affect the information-carrying capacity of a communication link.
- ❖ The most significant impairments are:
  - Attenuation and attenuation distortion
  - Delay distortion
  - Noise

ATTENUATION

- Strength of signal falls off with distance over any transmission medium.
- For guided media, this reduction in strength (attenuation) is exponential.
- For unguided media, attenuation is a more complex function of distance and the makeup of the atmosphere.
- Attenuation introduces three considerations for a transmission engineer.
- First, a received signal must have sufficient strength so that the electronic circuitry in the receiver can detect the signal.
- Second, the signal must maintain a level sufficiently higher than noise to be received without error.
- Third, attenuation varies with frequency.

DELAY DISTORTION

- Delay distortion occurs because the velocity of propagation of a signal through a guided medium varies with frequency.
- For a band limited signal, the velocity tends to be highest near the center frequency and fall off toward the two edges of the band.
- Thus various frequency components of a signal arrive at a receiver at different times resulting in phase shifts between the different frequencies.
- This effect is referred to as delay distortion because the received signal is distorted due to varying delays experienced at its constituent frequencies.
- Delay distortion is particularly critical for digital data.
- Consider a sequence of bits transmitted using an analog or digital

DELAY DISTORTION

- Because of delay distortion, some of the signal components of one bit position will spill over into other bit positions.
- This will cause an intersymbol interference, which is a major limitation to maximum bit rate over a transmission channel.
- Equalizing techniques can also be used for delay distortion.

NOISE

- ❖ Noise is a major limiting factor in communications system performance.
- ❖ For any data transmission event, the received signal will consist of the transmitted signal, modified by the various distortions imposed by the transmission system, plus additional unwanted signals that are inserted somewhere between transmission and reception.
- ❖ The latter undesired signals are known as “noise” having four categories:
  - Thermal noise
  - Intermodulation noise
  - Crosstalk
  - Impulse noise

NOISE

- Thermal noise is due to thermal agitation of electrons.
- It is present in all electronic devices and transmission media and is a function of temperature.
- Thermal noise is uniformly distributed across bandwidths typically used in communications systems and hence is often referred to as white noise.
- Thermal noise cannot be eliminated and therefore places an upper bound on communications system performance.
- Because of the weakness of the signal received by satellite earth stations, thermal noise is particularly significant for satellite communication.

NOISE

- When signals at different frequencies share the same transmission medium, the result may be intermodulation noise.
- The effect of intermodulation noise is to produce signals at a frequency that is the sum or difference of the two original frequencies or multiples of those frequencies.
- For example, if two signals, one at 4000 Hz and one at 8000 Hz, share the same transmission facility, they might produce energy at 12,000 Hz.
- This noise could interfere with an intended signal at 12,000 Hz.
- Intermodulation noise is produced by nonlinearities in the transmitter, receiver, and/or intervening transmission medium.

NOISE

- Ideally, these components behave as linear systems; that is, the output is equal to the input times a constant.
- In any real system, the output is a more complex function of the input.
- Excessive nonlinearity can be caused by component malfunction or overload from excessive signal strength.
- It is under these circumstances that the sum and difference frequency terms occur.
- Crosstalk has been experienced by anyone who, while using a telephone, has been able to hear another conversation; it is an unwanted coupling between signal paths.

NOISE

- It can occur by electrical coupling between nearby twisted pairs or, rarely,
  coax cable lines carrying multiple signals.
- Crosstalk can also occur when the microwave antennas pick up some unwanted signals from the air.
- Although highly directional antennas are used, microwave energy does spread during propagation.
- A crosstalk is of the same order of magnitude as a thermal noise.
- All of the types of noise discussed so far have reasonably predictable and relatively constant magnitudes.
- Thus it is possible to engineer a transmission system to cope with them.

NOISE

- Impulse noise, however, is noncontinuous, consisting of irregular pulses or noise spikes of short duration and of relatively high amplitude.
- It is generated from a variety of causes, including external electromagnetic disturbances, such as lightning, and faults and flaws in the communications system.
- Impulse noise is generally only a minor annoyance for analog data.
- For example, voice transmission may be corrupted by short clicks and crackles with no loss of intelligibility.
- However, the impulse noise is the primary source of error in any knid of digital data communication.

CHANNEL CAPACITY

- We have seen that there are a variety of impairments that distort or corrupt a signal.
- For digital data, the question that then arises is to what extent these impairments limit the data rate that can be achieved.
- The maximum rate at which the data can be transmitted over the given communication path or channel, under given conditions, is referred to as the channel capacity.
- There are four concepts here that we are trying to relate to one another.

- **Data rate:**

- The rate, in bits per second (bps), at which data can be communicated.

CHANNEL CAPACITY

- **Bandwidth:**
  - The bandwidth of the transmitted signal as constrained by the transmitter and the nature of the transmission medium expressed either in cycles per second or hertz.
- **Noise:**
  - The average level of noise over the communications path
- **Error rate:**
  - The rate at which errors occur, where an error is the reception of a ‘bit 1’ when a ‘bit 0’ was transmitted or the reception of a ‘bit 0’ when a ‘bit 1’ was transmitted.

NYQUIST BANDWIDTH

- Consider a channel that is noise free having a limitation on data rate.
- A formulation of this limitation, due to Nyquist, states that if the rate of signal transmission is 2B, then a signal with frequencies no greater than B is sufficient to carry the signal rate.
- Given a bandwidth of B, the highest signal rate that can be carried is 2B.
- This limitation is due to the effect of intersymbol interference, such as is produced by delay distortion.
- The result is useful in the development of digitalto-analog encoding schemes and is, in essence, based on the same derivation as that of the sampling theorem.

FRAMEWORK FOR PROBLEM SOLVING

- Note that in the preceding paragraph, we referred to signal rate.
- If the signals to be transmitted are binary (two voltage levels), then the data rate that can be supported by B Hz is 2B bps.
- However, the signals with more than two levels can be used i.e. each signal element can represent more than 1 bit.
- For example, if four possible voltage levels are used as signals, then each signal element can represent 2 bits.
- With multilevel signaling, the Nyquist formulation becomes

$$C = 2.B.\log_2 M$$

where M is the number of discrete signal or voltage levels.

FRAMEWORK FOR PROBLEM SOLVING

- So, for a given bandwidth, the data rate can be increased by increasing the number of different signal elements.
- However, this places an increased burden on the receiver.
- Instead of distinguishing one of two possible signal elements during each signal time, it must distinguish one of M possible signal elements.
- Noise and other impairments on the transmission line, however, will limit the practical value of M.

SHANNON CAPACITY

- Nyquist’s formula indicates that, all other things being equal, doubling the bandwidth doubles the data rate.
- Now consider the relationship among data rate, noise, and error rate.
- The presence of noise can corrupt 1 or more bits.
- If the data rate is increased, then the bits become “shorter” so that more bits are affected by a given pattern of noise.
- If the data rate is increased, then more bits will occur during the interval of a noise spike, and hence more errors will occur.
- All of these concepts can be tied together neatly in a formula developed by the mathematician Claude Shannon.

SHANNON CAPACITY

- As we have just illustrated, the higher the data rate, the more damage that unwanted noise can do.
- We would expect that a greater signal strength would improve the ability to receive data correctly in the presence of noise.
- The key parameter involved is the signal-to-noise ratio (SNR, or S/N),12 which is the ratio of the power in a signal to the power contained in the noise that is present at a particular point in transmission.
- Typically, this ratio is measured at a receiver, because it is at this point that an attempt is made to process the signal and recover the data.

GUIDED TRANSMISSION MEDIA

- For guided transmission media, the transmission capacity, in terms of either data rate or bandwidth, depends critically on the distance and on whether the medium is point-to-point or multipoint.
- Table 4.1 indicates the characteristics typical for the common guided media for long-distance point-to-point applications;
- We defer a discussion of the use of these media for local area networks (LANs) to Part Four.
- The three guided media commonly used for data transmission are twisted pair, coaxial cable, and optical fiber.
- We examine each of these in turn.

TWISTED PAIR

- The least expensive and most widely used guided transmission medium is twisted pair.
- A twisted pair consists of two insulated copper wires arranged in a regular spiral pattern.
- A wire pair acts as a single communication link.
- Typically, a number of these pairs are bundled together into a cable by wrappingthem in a tough protective sheath, or jacket.
- The wires in a pair have thicknesses of from 0.4 to 0.9 mm.
- By far the most common guided transmission medium for both analog and digital signals is twisted pair.

TWISTED PAIR

- It is the most commonly used medium in the telephone network and is the workhorse for communications within buildings.
- Twisted pair is used to transmit both analog and digital transmission.
- Twisted pair comes in two varieties: unshielded and shielded.
- The unshielded twisted pair (UTP) consists of one or more twisted-pair cables, typically enclosed within an overall thermoplastic jacket, which provides no electromagnetic shielding.
- In case of a shielded twisted pair, each pair of wires is individually shielded with metallic foil, generally referred to as foil twisted pair.

COAXIAL CABLE

- Coaxial cable consists of two conductors but is constructed differently to permit it to operate over a wider range of frequencies.
- It consists of a hollow outer cylindrical conductor that surrounds a single inner wire conductor.
- The inner conductor is held in place by either regularly spaced insulating rings or a solid dielectric material.
- The outer conductor is covered with a jacket or shield.
- A single coaxial cable has a diameter of from 1 to 2.5 cm.
- Coaxial cable can be used over longer distances and support more stations on a shared line than twisted pair.

COAXIAL CABLE

- Coaxial cable is a versatile transmission medium.
- It is available in abudance everywhere.
- Thus, it is used in a wide variety of applications.
- The most important of these are:
  - Television distribution
  - Long-distance telephone transmission
  - Short-run computer system links
  - Local area networks (LANs)
- Coaxial cable is used to transmit both analog and digital signals.
- However, it has certain limitations in case of long-term communication.

OPTICAL FOBRE

- Coaxial cable is used to transmit both analog and digital signals.
- Optical fiber already enjoys considerable use in long-distance telecommunications, and its use in military applications is growing.
- The continuing improvements in performance and decline in prices, together with the inherent advantages of optical fiber, have made it increasingly attractive for local area networking.
- Five basic categories of application for optical fiber are:
  - Long-haul trunks
  - Metropolitan trunks
  - Rural exchange trunks

OPTICAL FOBRE

- ❑ Subscriber loops
- ❑ Local area networks

- ❖ A fiber optic link consists of a transmitter on one end of a fiber and a receiver on the other end.
- ❖ Different types of light source that are used in fiber optic systems are: lightemitting diode (LED) & injection laser diode (ILD).
- ❖ The LED is less costly, operates over a greater temperature range and has a longer operational life while the ILD which operates on the laser principle, is more efficient and can sustain greater data rates

WIRELESS TRANSMISSION

- Three general ranges of frequencies are of interest in our discussion of wireless transmission.
- Frequencies in the range of about 1 GHz (gigahertz = 109 hertz) to 40 GHz are referred to as microwave frequencies.
- At these frequencies, highly directional beams are possible, and microwave is quite suitable for point-to-point transmission.
- Microwave is also used for satellite communications.
- Frequencies in the range of 30 MHz to 1 GHz are suitable mostly for the omnidirectional applications.
- We refer to this range as the radio range.

WIRELESS TRANSMISSION

- Another important frequency range, for local applications, is the infrared portion of the spectrum.
- This covers, roughly, from $3 * 10^{11}$ to $2 * 10^{14}$Hz.
- Infrared is useful to local point-to-point and multipoint applications within confined areas, such as a single room.
- For unguided media, transmission and reception are achieved by means of an antenna.
- An antenna can be defined as an electrical conductor or system of conductors used either for radiating electromagnetic energy or for collecting electromagnetic energy.

TERRESTRIAL MICROWAVE

- The most common type of microwave antenna is the parabolic “dish.”
- A typical size is about 3 m in diameter.
- The antenna is fixed rigidly and focuses a narrow beam to achieve line-of-sight transmission to the receiving antenna.
- Microwave antennas are usually located at substantial heights above ground level to extend the range between antennas and to be able to transmit over intervening obstacles.
- To achieve long-distance transmission, a series of microwave relay towers is used, with point-to-point microwave links strung together over the desired distance.

TERRESTRIAL MICROWAVE

- The primary use is in long-haul telecommunications service as an alternative to coaxial cable or optical fiber.
- It requires far fewer amplifiers or repeaters than coaxial cable over the same distance, but requires line-of-sight transmission.
- Microwave is commonly used for both voice and television transmission.
- Another increasingly common use of microwave is for short point-to-point links between buildings.
- This can be used for closed-circuit TV or as a data link between LANs.
- Another important use of microwave is in cellular systems.
- Common frequencies used for transmission are 1 to 40 GHz.

SATELLITE MICROWAVE

- A communication satellite is, in effect, a microwave relay station.
- It is used to link two or more ground-based microwave transmitters / receivers, known as earth stations, or ground stations.
- The satellite receives transmissions on one frequency band (uplink), amplifies or repeats the signal, and transmits it on another frequency.
- A single orbiting satellite will operate on a number of frequency bands, called transponder channels, or simply transponders.
- The following are the most important applications for satellites:
    - Television distribution
    - Long-distance telephone transmission

SATELLITE MICROWAVE

- ❑ Private business networks
- ❑ Global positioning

❖ Satellites are well suited to television distribution and are being used extensively throughout the world for this purpose.

❖ Satellite transmission is also used for point-to-point trunks between telephone exchange offices in public telephone networks.

BROADCAST RADIO

- The principal difference between broadcast radio and microwave is that the former is omnidirectional and the latter is directional.
- Thus broadcast radio does not require dish-shaped antennas, and the antennas need not be rigidly mounted to a precise alignment.
- Radio is a general term used to encompass frequencies in the range of 3 kHz to 300 GHz.
- We are using the informal term broadcast radio to cover the VHF and part of the UHF band: 30 MHz to 1 GHz.
- This range covers FM radio and UHF and VHF television.
- This range is also used for a number of data networking applications.

LINE OF SIGHT TRANSMISSION

- ❖ In this section, let us examine some impairments specific to wireless line-of-sight transmission.

- ❑ **Free Space Loss**

- ❖ For wireless communication, the signal disperses with distance.
- ❖ Therefore, an antenna with a fixed area receives less signal power the farther it is from the transmitting antenna.
- ❖ For satellite communication this is the primary mode of signal loss.
- ❖ A transmitted signal attenuates over distance because the signal is being spread over a larger and larger area.
- ❖ This form of attenuation is known as free space loss.

LINE OF SIGHT TRANSMISSION

- **Atmospheric Absorption**

- An additional loss between the transmitting and receiving antennas is atmospheric absorption.
- Water vapor and oxygen contribute most to attenuation.
- A peak attenuation occurs in the vicinity of 22 GHz due to water vapor.
- At frequencies below 15 GHz, the attenuation is less.
- The presence of oxygen results in an absorption peak in the vicinity of 60 GHz but contributes less at frequencies below 30 GHz.
- Rain and fog (suspended water droplets) cause scattering of radio waves that results in attenuation.

LINE OF SIGHT TRANSMISSION

❑ Multipath

- ❖ For wireless facilities where there is a relatively free choice of where antennas are to be located, they can be placed so that if there are no nearby interfering obstacles, there is a direct line-of-sight path from transmitter to receiver.
- ❖ This is generally the case for many satellite facilities and for point-to-point microwave.
- ❖ In other cases like mobile telephony, there are obstacles in abundance.
- ❖ In fact, in extreme cases, there may be no direct signal.

LINE OF SIGHT TRANSMISSION

❑ Refraction

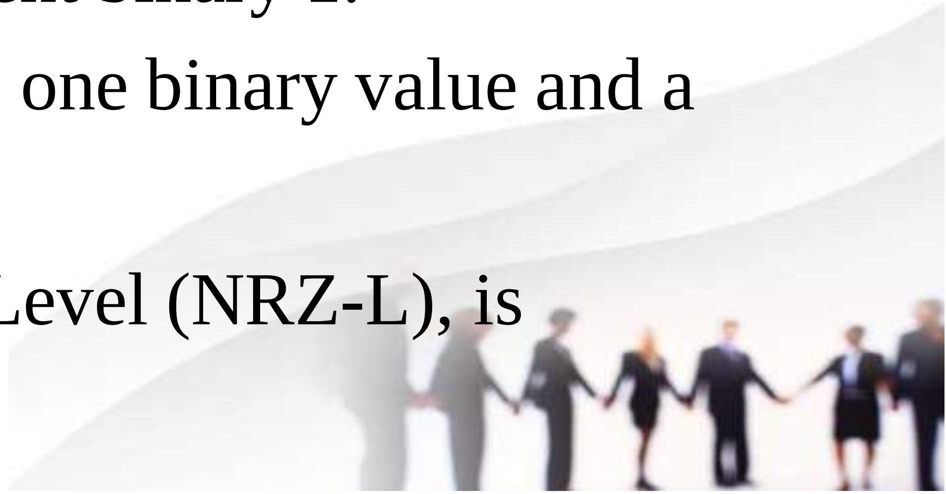
- ❖ Radio waves are refracted on propagating through the atmosphere.
- ❖ The refraction is caused by changes in the speed of the signal with altitude or by other spatial changes in the atmospheric conditions.
- ❖ Normally, the speed of the signal increases with altitude, causing radio waves to bend downward.
- ❖ However, on occasion, weather conditions may lead to variations in speed with height that differ significantly from the typical variations.
- ❖ This may result in a situation in which only a fraction or no part of the line-of-sight wave reaches the receiving antenna.

UNIT III

SIGNAL ENCODING TECHNIQUES

NONRETURN TO ZERO (NRZ)

- ❖ The most common, and easiest, way to transmit digital signals is to use two different voltage levels for the two binary digits.
- ❖ Codes that follow this strategy share the property that the voltage level is constant during a bit interval; there is no transition.
- ❖ For example, the absence of voltage can be used to represent binary 0, with a constant positive voltage used to represent binary 1.
- ❖ More commonly, a negative voltage represents one binary value and a positive voltage represents the other.
- ❖ This latter code, known as Nonreturn to Zero-Level (NRZ-L), is illustrated in Figure 5.2



NONRETURN TO ZERO (NRZ)

- NRZ-L is typically the code used to generate or interpret digital data by terminals and other devices.
- If a different code is to be used for transmission, it is generated from an NRZ-L signal by the transmission system.
- A variation of NRZ is known as NRZI (Nonreturn to Zero, invert on 1s).
- As with NRZ-L, NRZI maintains a constant voltage pulse for the duration of a bit time.
- The data themselves are encoded as the presence or absence of a signal transition at the beginning of the bit time.
- A transition denotes a binary 1 while no transition indicates a binary 0.

NONRETURN TO ZERO (NRZ)

- ❖ The NRZ codes are the easiest to engineer and, in addition, make efficient use of bandwidth.
- ❖ Due of their simplicity and low-frequency response chararacteristics, NRZ codes are commonly used for digital magnetic recording.
- ❖ However, their limitations make these codes unattractive for the signal transmission applications.

MULTILEVEL BINARY

- Multilevel binary encoding techniques address some of the deficiencies of the NRZ codes.
- These codes use more than two signal levels.
- Two examples of this scheme are illustrated in Figure 5.2, bipolar-AMI (alternate mark inversion) and pseudoternary.
- In the case of the bipolar-AMI scheme, a binary 0 is represented by no line signal, and a binary 1 is represented by a positive or negative pulse.
- The binary 1 pulses must alternate in polarity.
- There are several advantages to this approach.
- First, there will be no loss of synchronization if a long string of 1s occurs.

MULTILEVEL BINARY

- Each 1 introduces a transition & receiver can resynchronize on it.
- A long string of 0s would still be a problem.
- Second, because the 1 signals alternate in voltage from positive to negative, there is no net dc component.
- Also, the bandwidth of the resulting signal is considerably less than the bandwidth for NRZ (Figure 5.3).
- Finally, the pulse alternation property provides a simple means of the error detection process.
- Any isolated error, whether it deletes a pulse or adds a pulse, causes a violation of this property.

BINARY

- There is another set of coding techniques, grouped under the term biphase, that overcomes the limitations of NRZ codes.
- Two of these techniques, Manchester and differential Manchester, are in common use.
- In Manchester code, there is a transition at the middle of each bit period.
- The midbit transition serves as a clocking mechanism and also as data.
- A low-to-high transition represents a 1 and a high-to-low transition represents a 0.4
- In differential Manchester, the midbit transition is used in order to provide a clocking.

BINARY

- The encoding of a 0 is represented by the presence of a transition at the beginning of a bit period, and a 1 is represented by the absence of a transition at the beginning of a bit period.
- Differential Manchester has the added advantage of employing differential encoding.
- Biphase codes are popular techniques for data transmission.
- A common Manchester code has been specified for the IEEE 802.3 standard for baseband coaxial cable and twisted-pair bus LANs.
- Differential Manchester has been specified for the IEEE 802.5 token ring LAN, using shielded twisted pair

MODULATION RATE

- When signal encoding techniques are used, a distinction needs to be made between data rate (bps) and modulation rate (baud).
- The data rate, or bit rate, is given as,

  1/Tb, where Tb = bit duration.
- The modulation rate is the rate at which signal elements are generated.
- The minimum size signal element is a pulse of one-half the duration of a bit interval.
- For a string of all binary 0s or all binary 1s, a continuous stream of such pulses is generated.
- Hence, the maximum modulation rate for Manchester is 2/Tb.

SCRAMBLING TECHNIQUES

- Although the biphase techniques have achieved widespread use in local area network applications at relatively high data rates (up to 10 Mbps), they have not been widely used in long-distance applications.
- The principal reason for this is that they require a high signaling rate relative to the data rate.
- This sort of inefficiency is more costly in a long-distance application.
- Another approach is to make use of some sort of scrambling scheme.
- Sequences that would result in a constant voltage level on the line are replaced by filling sequences that will provide sufficient transitions for the receiver’s clock to maintain synchronization.

SCRAMBLING TECHNIQUES

- The filling sequence must be recognized by the receiver and replaced with the original data sequence.
- The filling sequence is the same length as the original sequence, so there is no data rate penalty.
- The design goals for this approach can be summarized as follows:
  - No dc component
  - No long sequences of zero-level line signals
  - No reduction in data rate
  - Error-detection capability
- Two of the major scrambling techniques are: B8ZS and HDB3.

AMPLITUDE SHIFT KEYING

- In ASK, the two binary values are represented by two different amplitudes of the carrier frequency.
- One amplitude is 0 i.e. one binary digit is represented by the presence at constant amplitude of the carrier the other by the absence of the carrier.
- On voice-grade lines, it is typically used only up to 1200 bps.
- The ASK technique is used to transmit digital data over optical fiber.
- Laser transmitters normally have a fixed “bias” current that causes the device to emit a low light level.
- This low level represents one signal element, while a higher-amplitude light wave represents another signal element.

AMPLITUDE SHIFT KEYING

- In ASK, the resulting transmitted signal for one bit time is represented with the help of following equation:

  ASK $s(t) = A\cos 2\Pi f_c t$ … binary 1

  $= 0$ … binary 0

  where the carrier signal is $A\cos 2\Pi f_c t$.

- ASK is susceptible to sudden gain changes and is a rather inefficient modulation technique.
- On voice-grade lines, it is typically used only up to 1200 bps.

FREQUENCY SHIFT KEYING

- The most common form of Frequency Shift Keying is the binary FSK (BFSK), in which the two binary values are represented by two different frequencies near a carrier frequency.
- The resulting transmitted signal for one bit time is

$$\textbf{BFSK}\ s(t) = A\cos 2\Pi f_1 t \qquad \ldots \text{ binary } 1$$

$$= A\cos 2\Pi f_2 t \qquad \ldots \text{ binary } 0$$

where f1 and f2 are typically offset from the carrier frequency fc by equal but opposite amounts.

FREQUENCY SHIFT KEYING

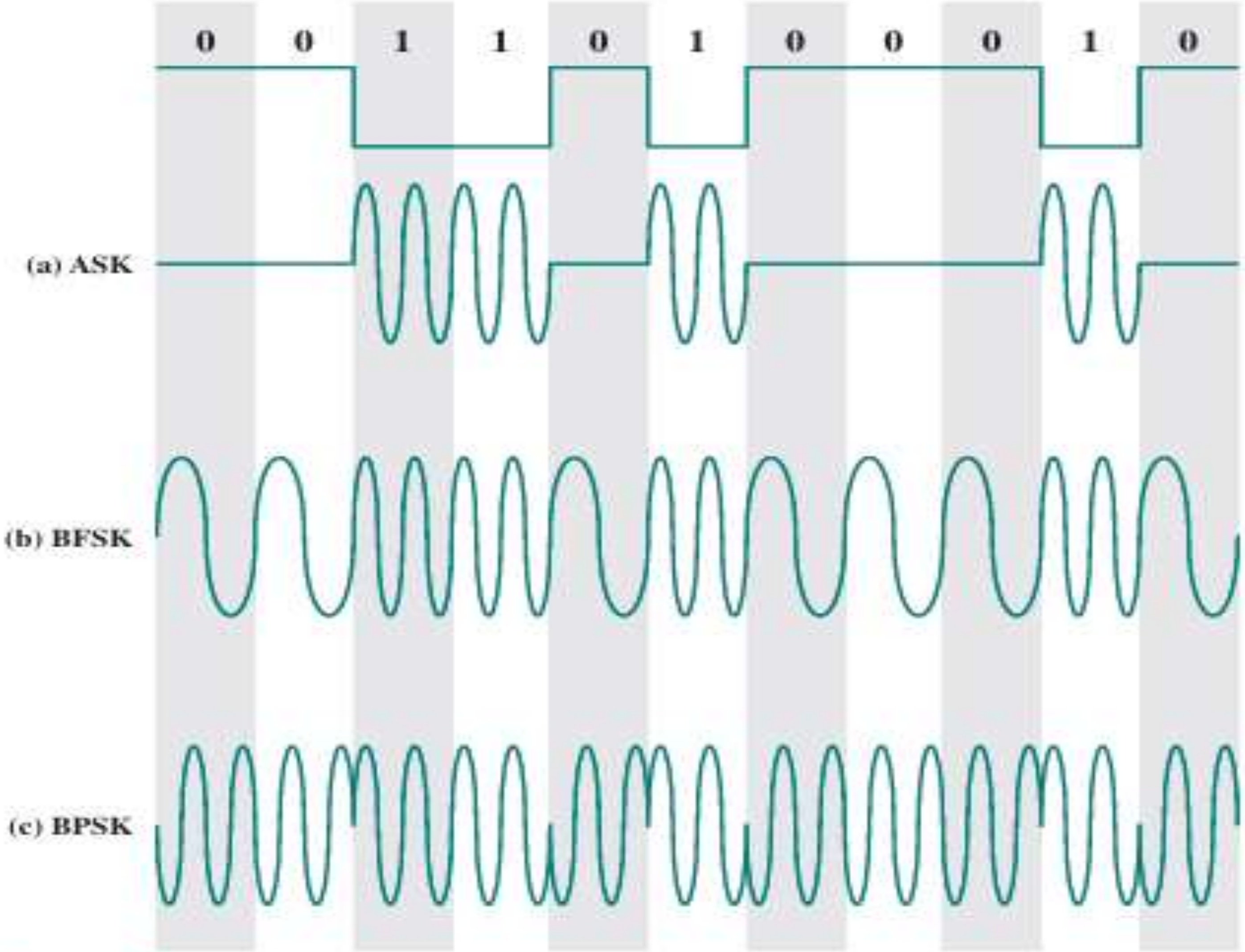


Figure 5.7 Modulation of Analog Signals for Digital Data

FREQUENCY SHIFT KEYING

- Figure 5.8 shows an example of the use of BFSK.
- However, this BFSK is for full-duplex operation over a voice-grade line.
- The figure is a specification for the Bell System 108 series modems.
- Recall that a voice-grade line will pass frequencies in the approximate range 300 to 3400 Hz.
- Also recall that a full duplex means the signals are transmitted in both the directions at the same time.
- To achieve full-duplex transmission, this band width is split.
- In one direction (transmit or receive), the frequencies used to represent 1 and 0 are centered on 1170 Hz, with a shift of 100 Hz on either side.

FREQUENCY SHIFT KEYING

- The effect of alternating between those two frequencies is to produce a signal whose spectrum is indicated as a shaded area on the left in Fig. 5.8.
- Similarly, for the other direction (receive or transmit) the modem uses frequencies shifted 100 Hz to each side of a center frequency of 2125 Hz.
- This signal is indicated by the shaded area on the right in Figure 5.8.
- Note that there is little overlap and thus little interference.
- BFSK is less susceptible to error than ASK.
- On voice-grade lines, it is typically used up to 1200 bps.
- It is commonly used for high-frequency (3-30 MHz) radio transmission.
- It can be used at even higher frequencies on LANs that use coaxial cable.

FREQUENCY SHIFT KEYING

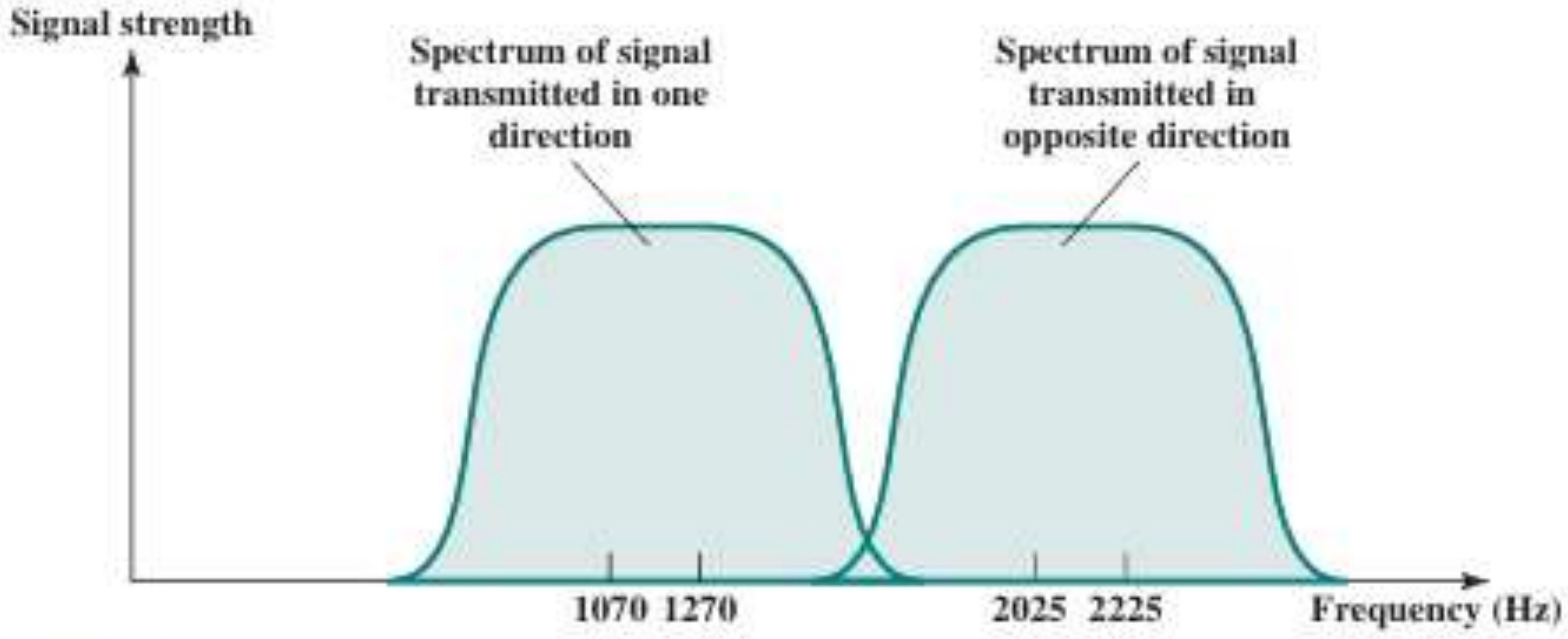


Figure 5.8 Full-Duplex FSK Transmission on a Voice-Grade Line

PHASE SHIFT KEYING

- In PSK, the phase of the carrier signal is shifted to represent data.
- The simplest scheme uses two phases to represent the two binary digits and is known as binary phase shift keying.
- The resulting transmitted signal for one bit time is

$$\text{BPSK } s(t) = A\cos 2\Pi f_c t \quad = \quad A\cos 2\Pi f_c t \qquad \text{...binary 1}$$

$$\text{BPSK } s(t) = A\cos(2\Pi f_c t + \Pi) \quad = -A\cos 2\Pi f_c t \qquad \text{...binary 0}$$

- Because a phase shift of 180° or Π is equivalent to flipping sine wave or multiplying it by -1, the rightmost expressions in Equation can be used.
- This leads to a convenient formulation.

PHASE SHIFT KEYING

- If we have a bit stream, and we define d(t) as the discrete function that takes on the value of +1 for one bit time.
- If the corresponding bit in the bit stream is 1 and the value of -1 for one bit time if the corresponding bit in the bit stream is 0.
- Then we can define the transmitted signal as

BPSK $s_d(t) = A\ d(t)\ A\cos 2\Pi f_c t \qquad \ldots 1$

- An alternative form of two-level PSK is differential PSK (DPSK).

QUADRATURE AMPLITUDE MODULATION

- More efficient use of bandwidth can be achieved if each signaling element represents more than one bit.
- For example, instead of a phase shift of 180° as allowed in BPSK, a common encoding technique known as quadrature phase shift keying (QPSK) uses phase shifts separated by multiples of p/2 or 90°

$$
\textbf{QPSK } s(t) = \begin{cases} A\cos(2\Pi f_c t + \Pi/4) & \ldots 11 \\ A\cos(2\Pi f_c t + 3\Pi/4) & \ldots 01 \\ A\cos(2\Pi f_c t - 3\Pi/4) & \ldots 00 \\ A\cos(2\Pi f_c t - \Pi/4) & \ldots 10 \end{cases}
$$

- Thus each signal element represents two bits rather than one.

QUADRATURE AMPLITUDE MODULATION

- Figure 5.11 shows the QPSK modulation scheme in general terms.
- The input is a stream of binary digits with a data rate of R = 1/Tb
- Here Tb is the width of each bit.
- This stream is converted into two separate bit streams of R/2 bps each, by taking alternate bits for the two streams.
- Two data streams are referred to as the I (in-phase) and Q (quadrature phase) streams.
- In the diagram, the upper stream is modulated on a carrier of frequency f_c by multiplying the bit stream by the carrier.

PULSE CODE MODULATION

- Pulse code modulation (PCM) is based on the sampling theorem:
- **Statement of Sampling Theorem:**
- If a signal f(t) is sampled at regular intervals of time and at a rate higher than twice the highest signal frequency, then the samples contain all the information of the original signal.
- The function f(t) may be reconstructed from these samples by the use of a low-pass filter.
- If voice data are limited to frequencies below 4000 Hz, a conservative procedure for intelligibility, 8000 samples per second would be sufficient to characterize the voice signal completely.

PULSE CODE MODULATION

- ❖Note, however, that these are analog samples, called pulse amplitude modulation (PAM) samples.
- ❖To convert to digital, each of these analog samples must be assigned a binary code.
- ❖Thus, PCM starts with a continuous-time, continuous-amplitude (analog) signal, from which a digital signal is produced (Figure 5.18).

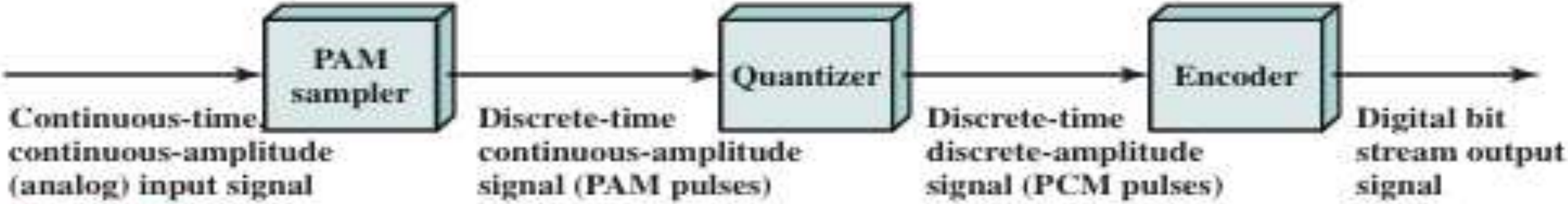
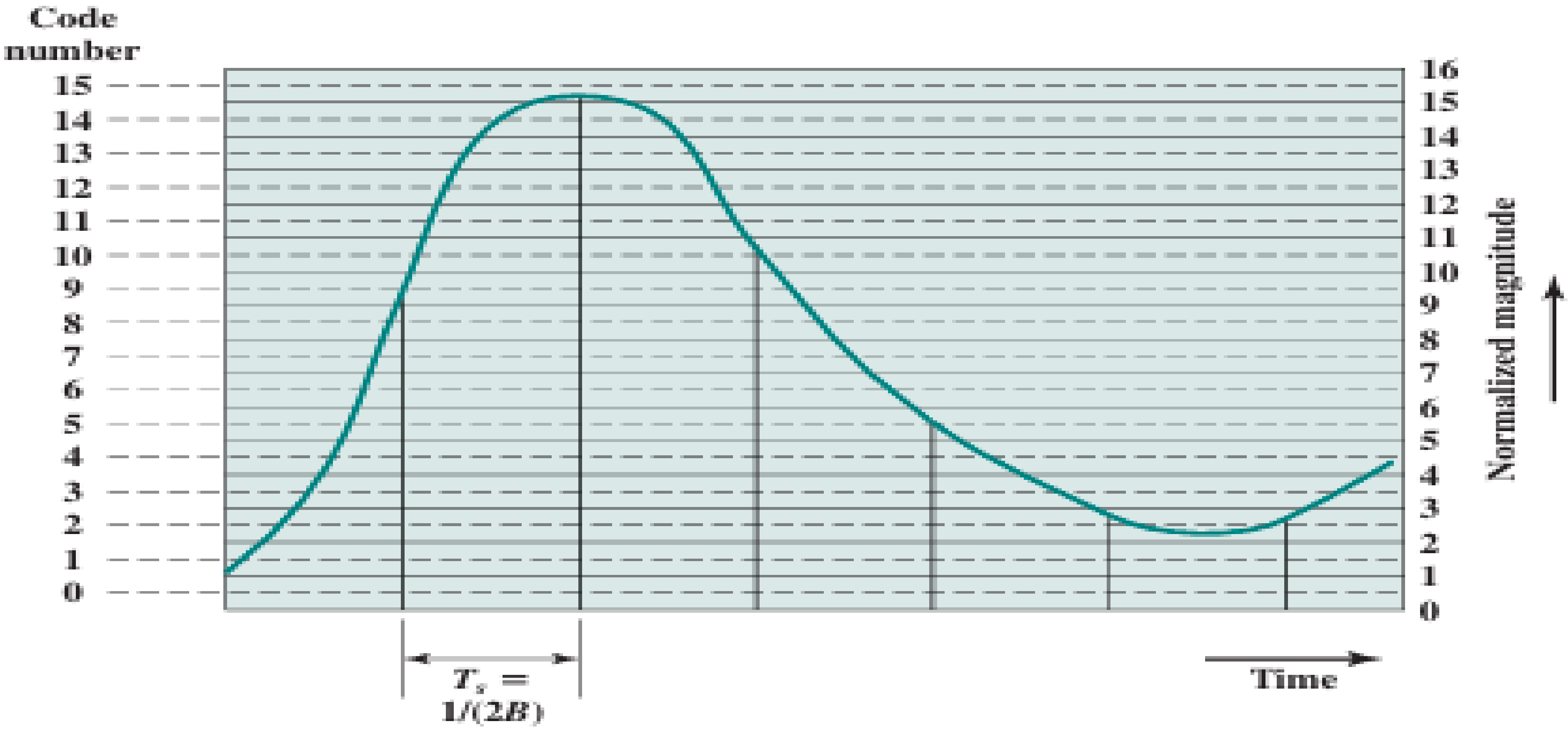


Figure 5.18 PCM Block Diagram

PULSE CODE MODULATION

- The digital signal consists of blocks of n bits, where each n-bit number is the amplitude of a PCM pulse.
- On reception, the process is reversed to reproduce the analog signal.
- Notice, however, that this process of pulse code modulation violates the terms of the sampling theorem.
- By quantizing PAM pulse, the original signal is now only approximated and cannot be recovered exactly.
- This effect is known as quantizing error or quantizing noise.
- Figure 5.17 shows an example in which the original signal is assumed to be band limited with a bandwidth of B.

PULSE CODE MODULATION



PAM value	1.1	9.2	15.2	10.8	5.6	2.8	2.7
Quantized code number	1	9	15	10	5	2	2
PCM code	0001	1001	1111	1010	0101	0010	0010

Figure 5.17 Pulse Code Modulation Example

DELTA MODULATION

- A variety of techniques have been used to improve the performance of PCM or to reduce its complexity.
- One of the most popular alternatives to PCM is delta modulation (DM).
- With delta modulation, an analog input is approximated by a staircase function that moves up or down by one quantization level (d) at each sampling interval (Ts).
- The important characteristic of this staircase function is that its behavior is binary.
- At each sampling time, the function moves up or down a constant amount δ.

DELTA MODULATION

- The output of the delta modulation process can be represented as a single binary digit for each sample.
- In essence, a bit stream is produced by approximating the derivative of an analog signal rather than its amplitude.
- We can observe that 1 is generated if the staircase function is to go up during the next interval while 0 is generated otherwise.
- But the principal advantage of Delta Modulation over Pulse Code Modulation is the simplicity of its implementation.
- In general, PCM exhibits better SNR characteristics at the same data rate.

ANGLE MODULATION

- Frequency & Pulse modulation are special cases of angle modulation.
- The modulated signal is expressed as

Angle modulation $\qquad s(t) = A_c \cos[2\Pi f_c t + \Phi(t)] \qquad \ldots 2$

- For phase modulation, the phase is proportional to the modulating signal:

PM $\qquad \Phi(t) = n_p m(t) \qquad \ldots 3$

where n_p is the phase modulation index

- For frequency modulation, the derivative of the phase is proportional to the modulating signal:

Fm $\qquad \Phi'(t) = n_f m(t) \qquad \ldots 4$

where $f'(t)$ is the derivative of $f(t)$

ANGLE MODULATION

- As with AM, both FM and PM result in a signal whose bandwidth is centered at f_c.
- We can now see that the magnitude of that bandwidth is very different.
- Amplitude modulation is a linear process.
- Amplitude modulation produces frequencies that are the sum/difference of the carrier signal and the components of a modulating signal.
- Hence, for AM,

$$\mathbf{B_T = 2B}$$

- However, angle modulation includes a term of the form cos $\Phi(t)$, which is non linear and will produce a wide range of frequencies.

UNIT I

INTRODUCTION

TYPES OF ERRORS

- In digital transmission systems, an error occurs when a bit is altered between transmission and reception.
- That is, a binary 1 is transmitted and a binary 0 is received, or a binary 0 is transmitted and a binary 1 is received.
- Two general types of errors can occur: single-bit errors and burst errors.
- A single-bit error is an isolated error condition that alters one bit but does not affect nearby bits.
- A burst error of length B is a contiguous sequence of B bits in which the first and last bits and any number of the intermediate bits are received in the error.

TYPES OF ERRORS

- IEEE Standard 100 and ITU-T both define an error burst as follows:
- Error burst is a group of bits in which two successive erroneous bits are always separated by less than a given number x of correct bits.
- Last erroneous bit in a burst with the first erroneous bit in the following burst are accordingly separated by x correct bits or more.
- There is a cluster of bits in which a number of errors occur, although not necessarily all of the bits in the cluster suffer an error.
- A single-bit error can occur in the presence of white noise, when a slight random deterioration of the signal-to-noise ratio is sufficient to confuse the receiver’s decision of a single bit.

ERROR DETECTION

- Regardless of the design of the transmission system, there will be errors.
- Data transmitted as one or more contiguous sequences of bits i.e. frames.
- We define these probabilities with respect to errors in transmitted frames:

  Pb: Probability that bit is received in error i.e. bit error rate (BER)

  P1: Probability that a frame arrives with no bit errors

  P2: Probability that, with an error-detecting algorithm in use, a frame arrives with one or more undetected errors

  P3: Probability that, with an error-detecting algorithm in use, a frame arrives with one or more detected bit errors and also with no undetected bit errors

ERROR DETECTION

- First consider the case in which no means are taken to detect errors.
- Then the probability of detected errors (P3) is zero.
- To express the remaining probabilities, assume the probability that any bit is in error (Pb) is constant and independent for each bit.
- Then we have

$$P_1 = (1- Pb)^F$$

$$P_2 = 1- P_1$$

where F is the number of bits per frame.

- In words, the probability that a frame arrives with no bit errors decreases when the probability of a single bit error increases, as you would expect.

ERROR DETECTION

- The probability that a frame arrives with no bit errors decreases with the increasing frame length.
- The longer the frame, the more bits it has and the higher the probability that one of these is in error.
- This kind of result motivates the use of error-detecting techniques to achieve a desired frame error rate on a connection with a given BER.
- All of these techniques operate on the following principle Figure 6.2

ERROR DETECTION

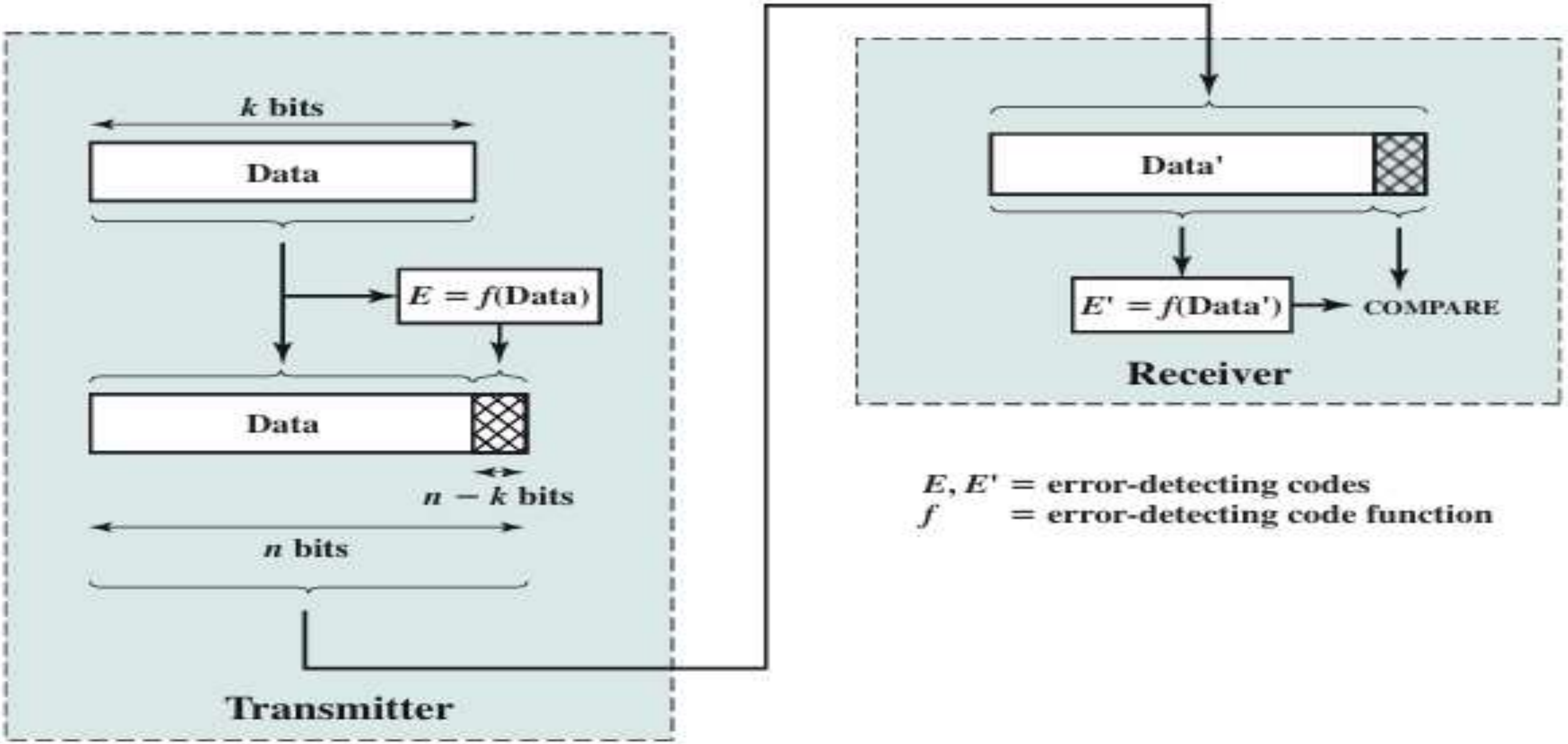


Figure 6.2 Error-Detection Process

PARITY CHECK

- **Parity Bit**
- A simple scheme is to append a parity bit to the end of a block of data.
- A typical example is character transmission, in which a parity bit is attached to each 7-bit IRA character.
- A value is selected so that the character has even no. 1s or odd no. of 1s.
- If two (or any even number) of bits are inverted due to error, an undetected error occurs.
- Typically, even parity is used for synchronous transmission and odd parity for asynchronous transmission.
- The use of the parity bit is not foolproof though.

PARITY CHECK

- Since the noise impulses are often long enough to destroy more than one bit particularly at high data rates.
- The two-dimensional parity scheme is more robust than single parity bit.
- A string of data bits to be checked is arranged in a two dimensional array.
- Appended to each row i is an even parity bit r_i for that row, and appended to each column j is an even parity bit c_j for that column.
- An overall parity bit p completes the matrix.
- Thus the error-detecting code consists of i + j + 1 parity bits.
- In this scheme, every bit participates in two parity checks.
- As with a simple parity bit, any odd no. of bit errors is detected.

THE INTERNET CHECKSUM

- The Internet checksum is an error-detecting code used in many Internet standard protocols, including IP, TCP, and UDP.
- The calculation makes use of 1's Complement operation with addition.
- To perform the ones complement operation on a set of binary digits, replace 0 digits with 1 digits and 1 digits with 0 digits.
- The ones-complement addition of two binary integers of equal bit length is performed as follows:

1. The two numbers are treated as unsigned binary integers and added.
2. If there is a carry out of the leftmost bit, add 1 to the sum. This is called an end around carry.

THE INTERNET CHECKSUM

- ❖ Typically, the checksum is included as a field in the header of a protocol data unit, such as in IP datagram.
- ❖ To compute the checksum, the checksum field is first set to all zeros.
- ❖ The checksum is then calculated by performing the ones-complement addition of all the words in the header, and then taking the ones-complement operation of the result.
- ❖ This result is placed in the checksum field.
- ❖ To verify a checksum, the ones-complement sum is computed over the same set of octets, including the checksum field.
- ❖ If the result is all 1 bits, the check succeeds.

THE INTERNET CHECKSUM

- The Internet checksum provides greater error-detection capability than a parity bit or two-dimensional parity scheme but is considerably less effective than the cyclic redundancy check (CRC).
- The primary reason for its adoption in Internet protocols is efficiency.
- Most of these protocols are implemented in software and the Internet checksum, involving simple addition and comparison operations, causes very little overhead.
- It is assumed that at the lower link level, a strong error-detection code such as CRC is used & so the Internet checksum is simply an additional end-to-end check for errors.

CYCLIC REDUNDANCY CHECK (CRC)

- CRC is one of the most common, most powerful error-detecting code.
- For k-bit block of bits (message), a transmitter generates an (n – k) bit sequence known as frame check sequence (FCS) such that the resulting frame consisting of n bits is exactly divisible by some predetermined no.
- The receiver then divides the incoming frame by that number and, if there is no remainder, assumes there was no error.
- To clarify this, we present the procedure in three equivalent ways:

  (a) modulo 2 arithmetic

  (b) polynomials and

  (c) digital logic

CYCLIC REDUNDANCY CHECK (CRC)

❑ Modulo 2 Arithmetic

❖ Modulo 2 arithmetic uses binary addition with no carries, which is just the exclusive OR (XOR) operation.

❖ Binary subtraction with no carries is interpreted as the XOR operation.

❖ Now define

T = n-bit frame to be transmitted

D = k-bit block of data, or message, the first k bits of T

F = (n - k)-bit FCS, the last (n - k) bits of T

P = pattern of n - k + 1 bits; this is the predetermined divisor

❖ It should be clear that $T = 2^{(n-k)}D + F$ with T/P to have no remainder..

CYCLIC REDUNDANCY CHECK (CRC)

Polynomials

- A second way of viewing the CRC process is to express all values as polynomials in a dummy variable X, with binary coefficients.
- The coefficients correspond to the bits in the binary number.
- Thus, for D = 110011, we have D(X) = X5 + X4 + X + 1, and for P = 11001, we have P(X) = X4 + X3 + 1.
- Arithmetic operations are again modulo 2.
- The CRC process can now be described as

$$\frac{X^{n-k}D(X)}{P(X)} = Q(X) + \frac{R(X)}{P(X)}$$

$$T(X) = X^{n-k}D(X) + R(X)$$

CYCLIC REDUNDANCY CHECK (CRC)

Digital Logic

- The CRC process can be represented by, and indeed implemented as, a dividing circuit consisting of XOR gates and a shift register.
- The shift register is a string of 1-bit storage devices.
- Each device has an output line, which indicates the value currently stored, and an input line.
- At discrete time instants, known as clock times, the value in the storage device is replaced by the value indicated by its input line.
- The entire register is clocked simultaneously, causing a 1-bit shift along the entire register.

FORWARD ERROR CORRECTION

- Error detection is a useful technique,
- It is usually found in data link control protocols such as HDLC.
- It is also found in transport protocols such as TCP.
- An error detecting code is used as a part of a protocol that corrects errors in transmitted data by requiring the blocks of data be retransmitted.
- For wireless applications this approach is inadequate for two reasons:

1. The BER on a wireless link can be quite high, which would result in a large number of retransmissions.
2. In some cases, especially satellite links, the propagation delay is very long com pared to the transmission time of a single frame.

FORWARD ERROR CORRECTION

- The result is a very inefficient system.
- The common approach to retransmission is to retransmit the frame in error plus all subsequent frames.
- With a long data link, an error in single frame necessitates retransmitting many frames.
- Instead, it would be desirable to enable the receiver to correct errors in an incoming transmission on the basis of the bits in that transmission.
- Figure 6.8 shows in general how this is done.
- On the transmission end, each k-bit block of data is mapped into an n-bit block (n>k) called codeword using FEC (f/w error correction) encoder.

FORWARD ERROR CORRECTION

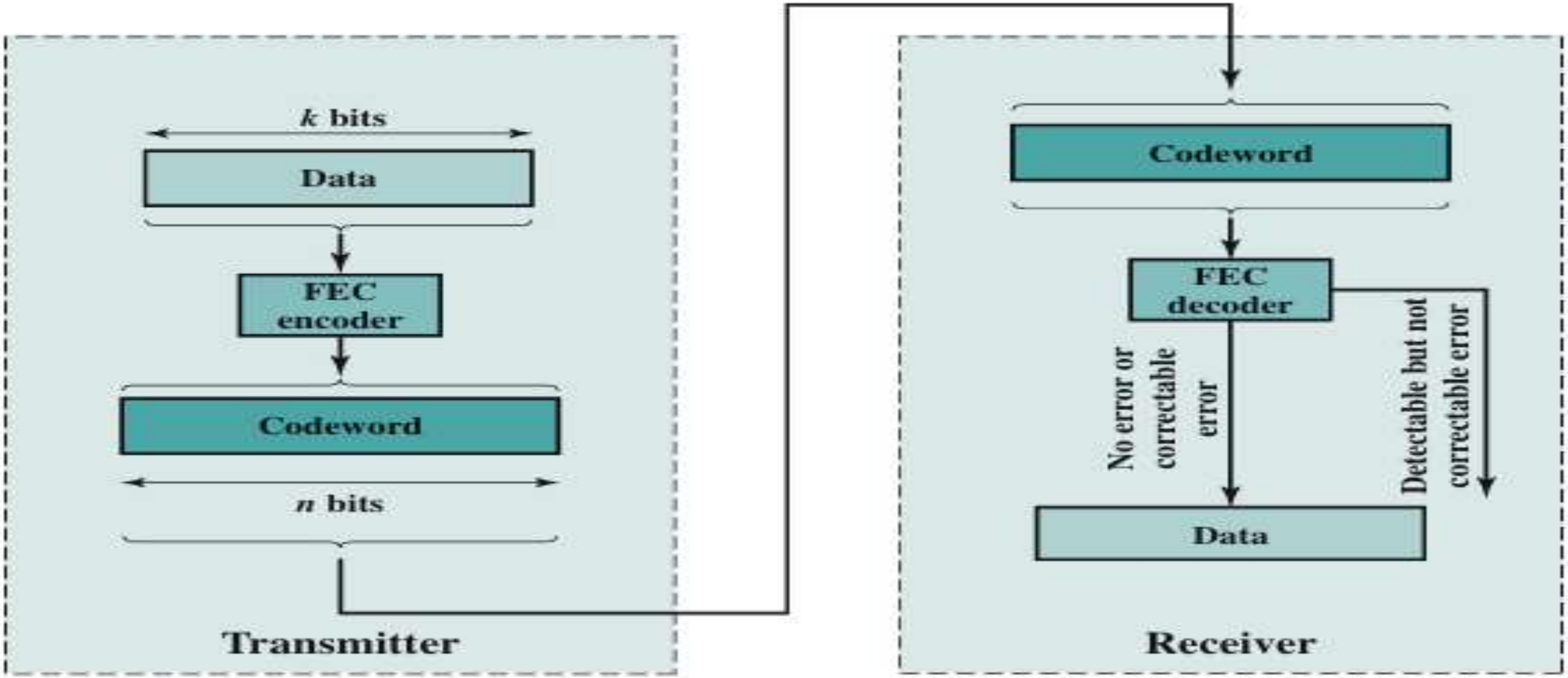


Figure 6.8 Error-Correction Process

FORWARD ERROR CORRECTING CODES

- To begin, we define a term that shall be of use to us.
- The Hamming distance d(v1, v2) between two n-bit binary sequences v1 and v2 is the number of bits in which v1 and v2 disagree.
- For example, if v1=011011, v2=110001 then d(v1, v2) = 3.
- Now let us consider the block code technique for error correction.
- Suppose we wish to transmit blocks of data of length k bits.
- Instead of transmitting each block as k bits, we map each k-bit sequence into a unique n-bit codeword.
- The preceding example illustrates the essential properties of a block error correcting code.

FORWARD ERROR CORRECTING CODES

- An (n, k) block code encodes k data bits into n-bit codewords.
- Typically, each valid codeword reproduces the original k data bits and adds to them (n - k) check bits to form the n-bit codeword.
- Thus the design of a block code is equivalent to the design of a function of the form $v_c = f(v_d)$, where v_d is a vector of k data bits and v_c is a vector of n codeword bits.
- With an (n, k) block code, there are 2^k valid codewords out of a total of 2^n possible codewords.
- The ratio of redundant bits to data bits, (n-k)/k, is called redundancy of the code & the ratio of data bits to total bits, k/n, is called the code rate.

FORWARD ERROR CORRECTING CODES

- The code rate is a measure of how much additional bandwidth is required to carry data at the same data rate as without the code.
- For example, a code rate of 1/2 requires double the transmission capacity of an uncoded system to maintain the same data rate.
- Our example has a code rate of 2/5 and so requires 2.5 times the capacity of an uncoded system.
- For example, if the data rate input to the encoder is 1 Mbps, then the output from the encoder must be at a rate of 2.5 Mbps to keep up.

FLOW CONTROL

- Flow control is a technique for assuring that a transmitting entity does not over whelm a receiving entity with data.
- The receiving entity typically allocates a data buffer of some maximum length for a transfer.
- When data are received, the receiver must do a certain amount of processing before passing the data to the higher level software.
- In the absence of flow control, the receiver's buffer may fill up and over flow while it is processing old data.
- To begin, we should examine mechanisms for flow control in the absence of errors.

FLOW CONTROL

- The model we will use is a vertical time sequence diagram.
- It has the advantages of showing time dependencies and illustrating the correct send–receive relationship.
- Each arrow represents a single frame transiting a data link between the two stations.
- The data are sent in a sequence of frames.
- Each frame contains a portion of the data and some control information.
- The time it takes for a station to emit all of the bits of a frame onto the medium is nothing but the transmission time.
- This is proportional to the length of the frame.

FLOW CONTROL

- **Stop-and-Wait Flow Control**
- The simplest form of flow control, known as stop-and-wait flow control, works as follows.
- A source entity transmits a frame.
- After the destination entity receives a frame, it indicates its willingness to accept another frame by sending an acknowledgment to that frame.
- The source must wait until it receives the acknowledgment before sending the next frame.
- The destination can thus stop the flow of data simply by withholding acknowledgment.

FLOW CONTROL

❑ Sliding Window Flow Control

- ❖ The essence of the problem described so far is that only one frame at a time can be in transit.
- ❖ In situations where the bit length of the link is greater than the frame length (a>1), serious inefficiencies result.
- ❖ Efficiency can be greatly improved by allowing multiple frames to be in transit at the same time.
- ❖ Let us examine how this might work for two stations A & B connected via a full-duplex link.
- ❖ Station B allocates buffer space for W frames.

ERROR CONTROL

- ❖ Error control refers to mechanisms to detect and correct errors that occur in the transmission of frames.
- ❖ The model covering the typical case is illustrated in Figure 7.1b.
- ❖ The data are sent as a sequence of frames.
- ❖ The frames arrive in the same order in which they are sent
- ❖ Each transmitted frame suffers an arbitrary and potentially variable amount of delay before reception.
- ❖ In addition, we admit the possibility of two types of errors:
- **Lost frame:** A frame fails to arrive at the other side.
- ❖ In the case of a network, the network may simply fail to deliver a frame.

ERROR CONTROL

- ❖ In a direct point to-point data link, a noise burst may damage a frame to the extent that the receiver is not aware of a frame which is transmitted.

- **Damaged frame:**

- ❖ A recognizable frame does arrive, but some of the bits are in error.
- ❖ The most common techniques for error control are based on some or all of the following ingredients:

- **Error detection:**

- ❖ The destination detects frames that are in error, using the techniques described in the preceding chapter, and discards those frames.

ERROR CONTROL

- **Positive Acknowledgement:**
  - ❖ The destination returns a positive acknowledgment to successfully received, error-free frames.
- **Retransmission after timeout:**
  - ❖ The source retransmits a frame that has not been acknowledged after a predetermined amount of time.
- **Negative acknowledgment and retransmission:**
  - ❖ The destination returns a negative acknowledgment to frames in which an error is detected.
  - ❖ The source retransmits such frames.

ERROR CONTROL

- These mechanisms are all referred to as automatic repeat request (ARQ).
- Effect of ARQ is to turn an unreliable data link into a reliable one.
- Three versions of ARQ have been standardized:
  - Stop-and-wait ARQ
  - Go-back-N ARQ
  - Selective-reject ARQ

ERROR CONTROL

❑ Stop-And-Wait ARQ

- ❖ Stop-and-wait ARQ is based on stop-and-wait flow control technique.
- ❖ The source station transmits a single frame and then must await an acknowledgment (ACK).
- ❖ No other data frames can be sent until the destination station’s reply arrives at the source station.
- ❖ Two sorts of errors could occur.
- ❖ First, the frame that arrives at the destination could be damaged.
- ❖ The receiver detects this by using the error-detection technique referred to earlier and simply discards the frame.

ERROR CONTROL

- To check the possibility, the source station is equipped with a timer.
- After a frame is transmitted, source station waits for an acknowledgment.
- If no acknowledgment is received by the time that the timer expires, then the same frame is sent again.
- Note that this method requires that the transmitter maintain a copy of a transmitted frame until an acknowledgment is received for that frame.
- The second sort of error is a damaged acknowledgment.
- Consider the following situation. Station A sends a frame.
- The frame is received correctly by station B, which responds with an acknowledgment.

ERROR CONTROL

- ❖ The ACK is damaged in transit and is not recognizable by A, which will therefore time out and resend the same frame.
- ❖ This duplicate frame arrives and is accepted by B.
- ❖ Thus B has accepted two copies of same frame as if they were separate.
- ❖ To avoid this problem, frames are alternately labeled with 0 or 1, and positive acknowledgments are of the form ACK0 and ACK1.
- ❖ In keeping with the sliding-window convention, an ACK0 acknowledges receipt of a frame numbered 1 and indicates that the receiver is ready for a frame numbered 0.

ERROR CONTROL

Go-Back-N ARQ

- The form of error control based on sliding-window flow control that is most commonly used is called go-back-N ARQ.
- Station sends many frames serially numbered modulo a maximum value.
- The number of unacknowledged frames outstanding is determined by window size, using the sliding-window flow control technique.
- With no errors, a destination will acknowledge incoming frames as usual
- If the destination station detects an error in the frame, it might send a negative acknowledgment (REJ = reject) for that frame.

ERROR CONTROL

- ❖ The destination station will discard that frame and all future incoming frames until the frame in error is correctly received.
- ❖ The source station, when it receives a REJ, must retransmit the frame in error plus all succeeding frames that were transmitted in the interim.
- ❖ Suppose that station A is sending frames to station B.
- ❖ A sets an acknowledgment timer for each transmitted frame.
- ❖ Suppose that B has previously successfully received frame (i-1) and A has just transmitted frame i.
- ❖ The major contingencies that may occur - Damaged frame, Damaged RR and Damaged REJ.

ERROR CONTROL

❑ Selective-Reject ARQ

- ❖ Selective reject would appear to be more efficient than go-back-N, because it minimizes the amount of retransmission.
- ❖ On the other hand, the receiver must maintain a buffer large enough to save post-SREJ frames until the frame in error is retransmitted and must contain logic for reinserting that frame in the proper sequence.
- ❖ The transmitter, too, requires more complex logic to be able to send a frame out of sequence.
- ❖ Because of such complications, selective-reject ARQ is much less widely used than go-back-N ARQ.

ERROR CONTROL

- ❖ Selective reject is a useful choice for a satellite link because of the long propagation delay involved.
- ❖ The window size limitation is more restrictive for selective-reject than for go-back-N.
- ❖ Consider the case of a 3-bit sequence number size for selective-reject.
- ❖ Allow a window size of seven, and consider the following scenario:

1. Station A sends frames 0 through 6 to station B.
2. Station B receives all seven frames and acknowledges with RR 7.
3. Because of a noise burst, the RR 7 is lost.
4. A times out and retransmits frame 0.

ERROR CONTROL

5. B has advanced its receive window to accept 7,0,1,2,3,4,5. It assumes frame 7 has been lost & this is a new frame 0, which it accepts.

- ❖ The problem with the preceding scenario is that there is an overlap between the sending and receiving windows.
- ❖ To overcome the problem, the maximum window size should be no more than half the range of sequence numbers.
- ❖ In the preceding scenario, if only four unacknowledged frames may be outstanding, no confusion can result.
- ❖ In general, for a k-bit sequence number field, which provides a sequence number range of 2k, the maximum window size is limited to 2k-1.

HIGH-LEVEL DATA LINK CONTROL (HDLC)

- Most important data link control protocol is HDLC (ISO 3009, ISO 4335).
- Not only is HDLC widely used, but it is the basis for many other important data link control protocols, which use the same or similar formats and the same mechanisms as employed in HDLC.
- To satisfy a variety of applications, HDLC defines three types of stations, two link configurations, and three data transfer modes of operation.
- The three station types are:

Primary station:

- Responsible for controlling the operation of the link.
- Frames issued by the primary are called commands.

HIGH-LEVEL DATA LINK CONTROL (HDLC)

- **Secondary station:**
  - Operates under the control of the primary station.
  - Frames issued by a secondary are called responses.
  - The primary maintains a separate logical link with each secondary station on the line.
- **Combined station:**
  - Combines the features of primary and secondary.
  - A combined station may issue both commands and responses.
  - The two link configurations are unbalanced configuration and balanced configuration respectively.

HIGH-LEVEL DATA LINK CONTROL (HDLC)

- **Unbalanced configuration:**
- Consists of one primary and one or more secondary stations and supports both full-duplex and half-duplex transmission.
- **Balanced configuration:**
- Consists of two combined stations and supports both full-duplex and half-duplex transmission.
- The three data transfer modes are
  - Normal response mode
  - Asynchronous balanced mode
  - Asynchronous response mode

HIGH-LEVEL DATA LINK CONTROL (HDLC)

- **Normal response mode (NRm):**

❖ Used with an unbalanced configuration.

❖ The primary may initiate data transfer to a secondary, but a secondary may only transmit data in response to a command from the primary.

- **Asynchronous balanced mode (ABm):**

❖ Used with a balanced configuration.

❖ Either combined station may initiate transmission without receiving permission from the other combined station.

- **Asynchronous response mode (ARm):**

❖ Used with an unbalanced configuration.

HIGH-LEVEL DATA LINK CONTROL (HDLC)

- ❖ Secondary initiates transmission without explicit permission of primary.
- ❖ The primary still retains responsibility for the line, including initialization, error recovery, and logical disconnection.
- ❖ NRM is used on multidrop lines, in which a number of terminals are connected to a host computer.
- ❖ The computer polls each terminal for input.
- ❖ NRM is also sometimes used on point-to-point links, particularly if the link connects a terminal or other peripheral to a computer.
- ❖ ABM is the most widely used of the three modes.
- ❖ It makes more efficient use of a full-duplex point-to-point link.

UNIT I

INTRODUCTION

FREQUENCY DIVISION MULTIPLEXING (FDM)

- Two communicating stations may not utilize full capacity of a data link.
- For efficiency, it should be possible to share that capacity.
- A generic term for such sharing is multiplexing.
- A common application of multiplexing is in long-haul communications.
- Trunks on long-haul networks are high-capacity fiber, coaxial, or microwave links.
- These links can carry large numbers of voice and data transmissions simultaneously using multiplexing.
- Figure 8.1 depicts the multiplexing function in its simplest form. There are n inputs to a multiplexer.

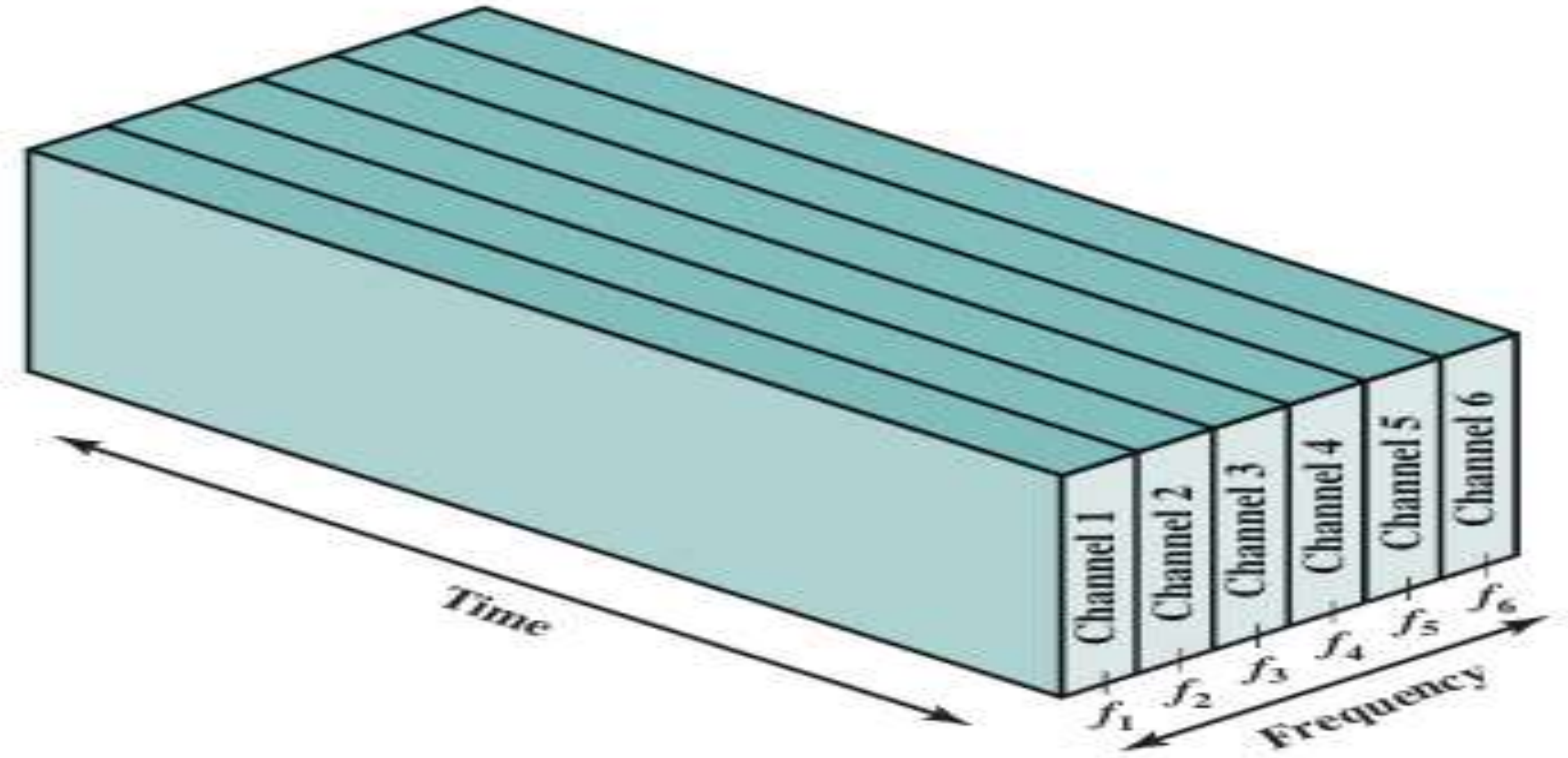
FREQUENCY DIVISION MULTIPLEXING (FDM)

- The multiplexer is connected by a single data link to a demultiplexer.
- The link is able to carry n separate channels of data.
- The multiplexer combines (multiplexes) data from the n input lines and transmits over a higher-capacity data link.
- The demultiplexer accepts the multiplexed data stream, separates data according to channel, and delivers data to the appropriate output lines.
- The widespread use of multiplexing in communication can be easily understood by the extensive need for higher data rates in daily life.
- FDM is possible when the useful bandwidth of the transmission medium exceeds the required bandwidth of signals to be transmitted.

FREQUENCY DIVISION MULTIPLEXING (FDM)

- ❖ A number of signals can be carried simultaneously if each signal is modulated onto a different carrier frequency
- ❖ The carrier frequencies are sufficiently separated that the bandwidths of the signals do not significantly overlap.
- ❖ A general case of FDM is shown in figure (a) given below.
- ❖ Six signal sources are fed into a multiplexer, which modulates each signal onto a different frequency (f1, . . . , f6).
- ❖ Each modulated signal requires a certain bandwidth centered on its carrier frequency, referred to as a channel.
- ❖ Channels are separated by guard bands (unused portions of a spectrum).

FREQUENCY DIVISION MULTIPLEXING (FDM)



(a) Frequency-division multiplexing

FREQUENCY DIVISION MULTIPLEXING (FDM)

- ❖The composite signal transmitted across the medium is analog.
- ❖Note, how ever, that the input signals may be either digital or analog.
- ❖In the case of digital input, the input signals must be passed through modems to be converted to analog.
- ❖In either case, each input analog signal must then be modulated to move it to the appropriate frequency band.
- ❖A generic depiction of an FDM system is shown in Figure 8.3.
- ❖A number of analog or digital signals [mi(t), i = 1, n] are to be multiplexed onto the same trans mission medium.
- ❖Each signal mi(t) is modulated onto a carrier f_i because multiple carriers are to be used, each is referred to as a subcarrier.
- ❖Any type of modulation may be used.

FREQUENCY DIVISION MULTIPLEXING (FDM)

- The resulting analog, modulated signals are then summed to produce a composite baseband1 signal $m_b(t)$.
- Figure 8.3b shows the result.
- The spectrum of signal mi(t) is shifted to be centered on f_i.
- For this scheme to work, f_i must be chosen so that the bandwidths of the various signals do not significantly overlap.
- Otherwise, it will be impossible to recover the original signals.

FREQUENCY DIVISION MULTIPLEXING (FDM)

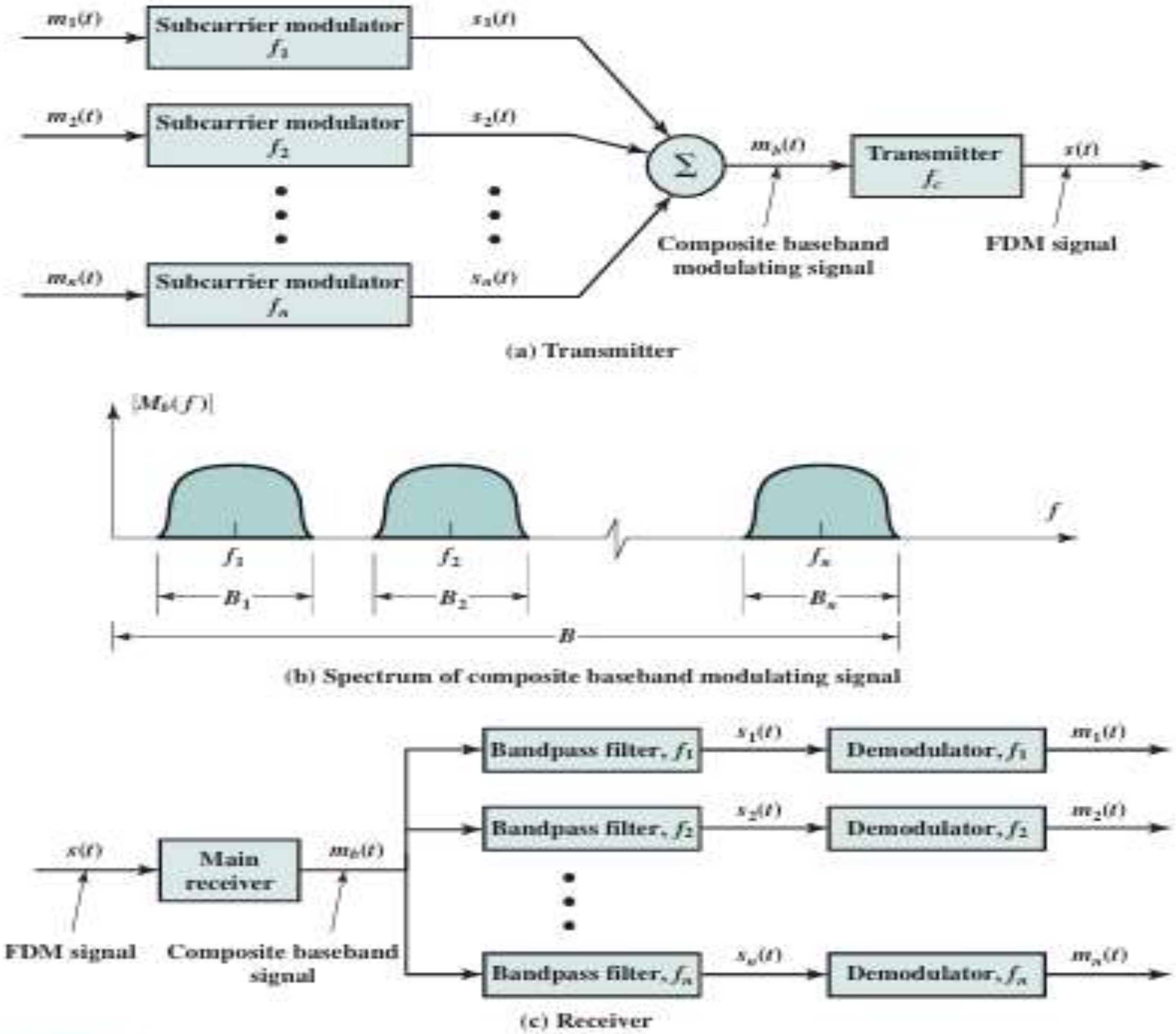


Figure 8.3 FDM System

ANALOG CARRIER SYSTEMS

- The long-distance carrier system provided in the United States and throughout the world is designed to transmit voiceband signals over high-capacity transmission links (coaxial cable & microwave systems).
- The earliest technique for utilizing high-capacity links is FDM.
- In US, AT&T has designated a hierarchy of FDM schemes to accommodate transmission systems of various capacities.
- A similar system has been adopted internationally by the name of ITU-T.
- At the first level of the AT&T hierarchy, 12 voice channels are combined to produce a group signal with a bandwidth of 12 * 4 kHz = 48 kHz, in the range 60 to 108 kHz.

ANALOG CARRIER SYSTEMS

- The signals are produced in a fashion similar to that described previously, using subcarrier frequencies of from 64 to 108 kHz in increments of 4 kHz.
- The next basic building block is the 60-channel supergroup, which is formed by frequency division multiplexing five group signals.
- At this step, each group is treated as a single signal with a 48-kHz bandwidth and is modulated by a subcarrier.
- Subcarriers have frequencies of 420 to 612 kHz in increments of 48 kHz.
- The resulting signal occupies 312 to 552 kHz.
- There are several variations to supergroup formation.

ANALOG CARRIER SYSTEMS

- Each of the five inputs to the supergroup multiplexer may be a group channel containing 12 multiplexed voice signals.
- Any signal up to 48 kHz wide whose bandwidth is contained within 60 to 108 kHz may be used as input to the supergroup multiplexer.
- As another variation, it is possible to combine 60 voiceband channels into a super group.
- This may reduce multiplexing costs where an interface with existing group multiplexer is not required.
- The next level of the hierarchy is the mastergroup, which combines 10 super group inputs.

ANALOG CARRIER SYSTEMS

- ❖ Again, any signal with a bandwidth of 240 kHz in the range 312 to 552 kHz can serve as input to the mastergroup multiplexer.
- ❖ The mastergroup has a bandwidth of 2.52 MHz and can support 600 voice frequency (VF) channels.
- ❖ Higher-level multiplexing is defined above the mastergroup.
- ❖ Note that the original voice or data signal may be modulated many times.
- ❖ A data signal encoded using QPSK can form an analog voice signal.
- ❖ This signal could then be used to modulate a 76-kHz carrier to form a component of a group signal.
- ❖ This group signal could then be used to modulate a 516-kHz carrier.

WAVELENGTH DIVISION MULTIPLEXING (WDM)

- The true potential of optical fiber is fully exploited when multiple beams of light at different frequencies are transmitted on the same fiber.
- This is a form of frequency division multiplexing but is commonly called wavelength division multiplexing (WDM).
- With WDM, the light streaming through the fiber consists of many colors, or wavelengths, each carrying a separate channel of data.
- In 1997, a landmark was reached when Bell Laboratories was able to demonstrate a WDM system with 100 beams each operating at 10 Gbps, for a total data rate of 1 trillion bits per second i.e. 1 TBPS.
- Commercial systems with 160 channels of 10 Gbps are now available.

WAVELENGTH DIVISION MULTIPLEXING (WDM)

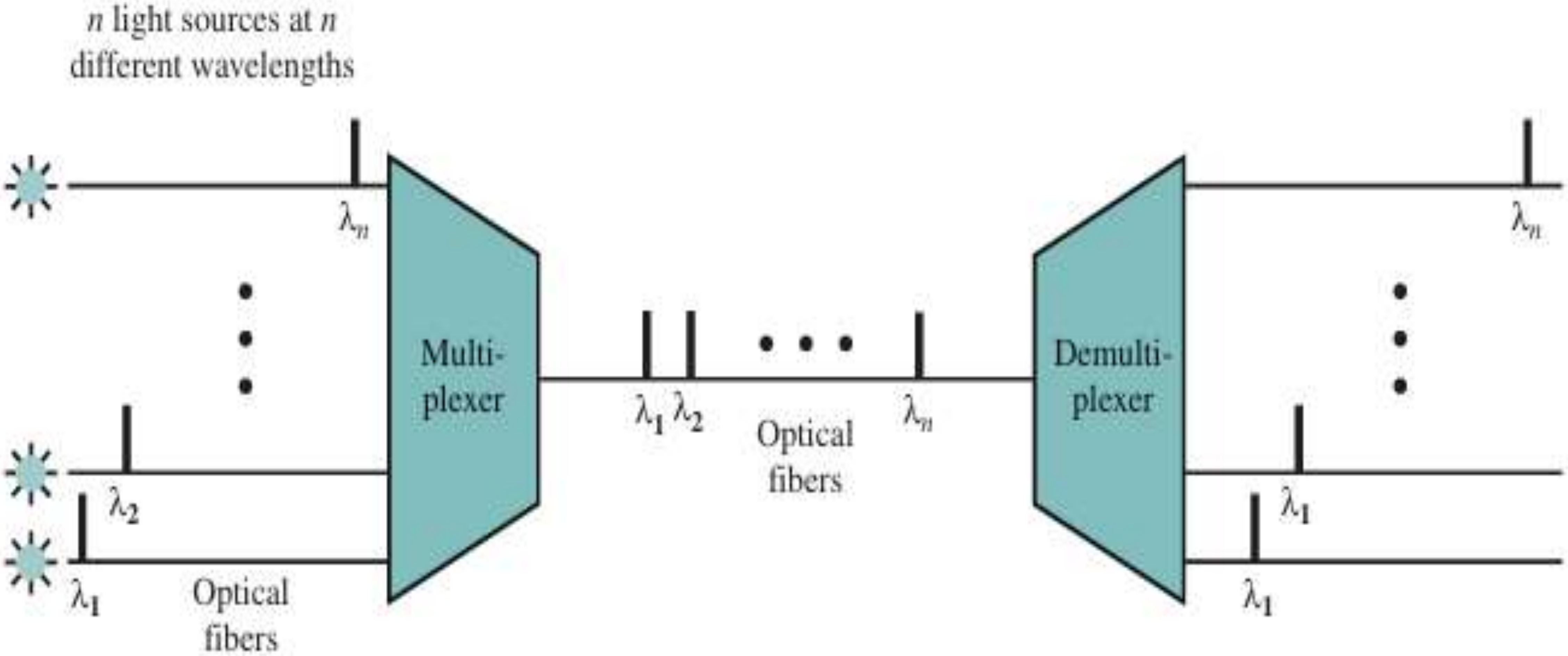


Figure 8.5 Wavelength Division Multiplexing

WAVELENGTH DIVISION MULTIPLEXING (WDM)

- ❖ In a lab environment, Alcatel has carried 256 channels at 39.8 Gbps each, a total of 10.1 TBPS, over a 100-km span.
- ❖ A typical WDM system has the same general architecture as some of the other FDM systems.
- ❖ A number of sources generate a laser beam at different wavelengths.
- ❖ These are sent to a multiplexer, which consolidates the sources for transmission over a single fiber line.
- ❖ Optical amplifiers, typically spaced tens of kilometers apart, amplify all of the wavelengths simultaneously.

WAVELENGTH DIVISION MULTIPLEXING (WDM)

- ❖ Finally, the composite signal arrives at a demultiplexer.
- ❖ At the multiplexer, the component channels are separated and sent to receivers at the destination point (Figure 8.5).
- ❖ Most WDM systems operate in the 1550-nm range.
- ❖ In early systems, 200 GHz was allocated to each channel, but today most WDM systems use 50-GHz spacing.
- ❖ The term dense wavelength division multiplexing (DWDM) connotes the use of more channels, more closely spaced, than ordinary WDM.
- ❖ A channel spacing of 200 GHz or less could be considered dense.

SYNCHRONOUS TIME DIVISION MULTIPLEXING

- Synchronous TDM is possible when the achievable data rate of the medium exceeds the data rate of digital signals to be transmitted.
- Multiple digital signals can be carried on a single transmission path by interleaving portions of each signal in time.
- Interleaving can be at bit level or in blocks of bytes or larger quantities.
- E.g. Multiplexer in Fig. has six inputs that might each be, say, 1 Mbps.
- A single line with capacity of 6 Mbps can accommodate all six sources.
- A generic depiction of a synchronous TDM is provided in Figure 8.6.
- A number of signals [$m_i(t)$, i = 1, n] are to be multiplexed onto the same transmission medium.

SYNCHRONOUS TIME DIVISION MULTIPLEXING

- ❖ The signals carry digital data and are generally digital signals.
- ❖ The incoming data from each source are briefly buffered.
- ❖ Each buffer is typically one bit or one character in length.
- ❖ Buffers are scanned serially to form composite digital data stream $m_c(t)$.
- ❖ The scan operation is sufficiently rapid so that each buffer is emptied before more data can arrive.
- ❖ Thus, the data rate of mc(t) must at least equal the sum of the data rates of the mi(t).
- ❖ The digital signal mc(t) may be transmitted directly, or passed through a modem so that an analog signal is trans mitted.

SYNCHRONOUS TIME DIVISION MULTIPLEXING

- In either case, transmission is typically synchronous.
- The transmitted data may have a format something like Figure 8.6b.
- The data are organized into frames.
- Each frame contains a cycle of time slots.
- In each frame, one or more slots are dedicated to each data source.
- The sequence of slots dedicated to one source, from frame to frame, is called a channel.
- Generally, the slot length equals the transmitter buffer length which is typically a bit or a byte (character).
- Each time slot contains one character of data.

SYNCHRONOUS TIME DIVISION MULTIPLEXING

- Typically, the start and stop bits of each character are eliminated before transmission and reinserted by the receiver, thus improving efficiency.
- The bit-interleaving technique is used with synchronous sources and may also be used with asynchronous sources.
- Each time slot contains just one bit. At the receiver, the interleaved data are demultiplexed and routed to the appropriate destination buffer.
- For each input source $m_i(t)$, there is an identical output destination that will receive the output data at the same rate at which it was generated.
- This TDM is called synchronous not because synchronous transmission is used, but because the time slots are preassigned to sources and fixed.

SYNCHRONOUS TIME DIVISION MULTIPLEXING

- The time slots for each source are transmitted whether or not the source has data to send.
- This is, of course, also the case with FDM.
- Thus, the capacity is wasted to achieve simplicity of implementation.
- Even when fixed assignment is used, how ever, it is possible for a synchronous TDM device to handle sources of different data rates.
- For example, the slowest input device could be assigned one slot per cycle, while faster devices are assigned multiple slots per cycle.
- An alternative to synchronous TDM is statistical TDM.
- The statistical multiplexer dynamically allocates time slots on demand.

SYNCHRONOUS TIME DIVISION MULTIPLEXING

- As with a synchronous TDM, the statistical multiplexer has a number of I/O lines on one side and a higher speed multiplexed line on the other.
- Each I/O line has a buffer associated with it.
- In the case of the statistical multiplexer, there are n I/O lines, but only k, where k 6 n, time slots available on the TDM frame.
- For input, the function of the multiplexer is to scan the input buffers, collecting data until a frame is filled, and then send the frame.
- On output, the multiplexer receives a frame and distributes the slots of data to the appropriate output buffers.
- Packet switching is, in effect, a form of statistical TDM.

SYNCHRONOUS TIME DIVISION MULTIPLEXING

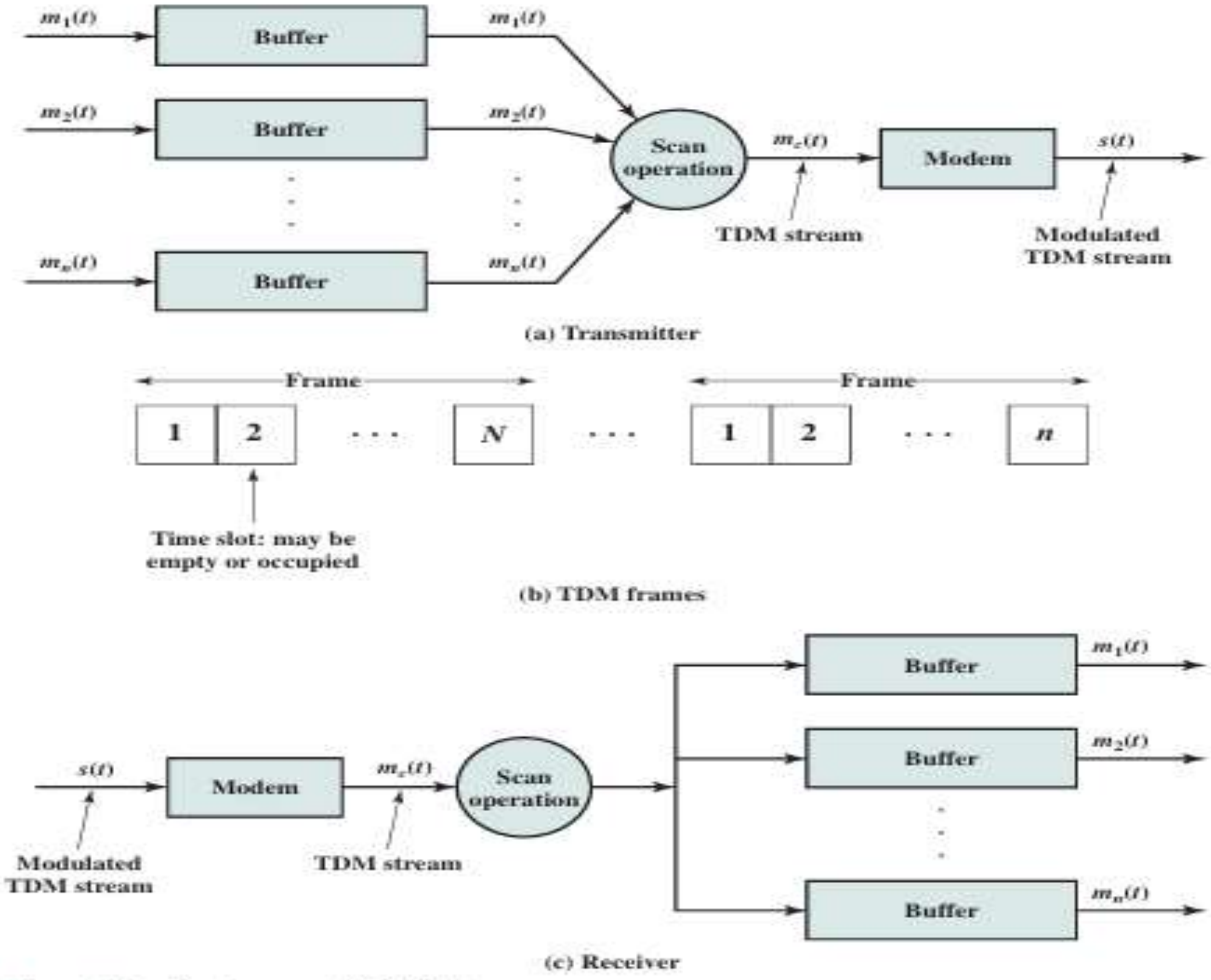


Figure 8.6 Synchronous TDM System

TDM LINK CONTROL

- Transmitted data stream does not contain the headers and trailers that we have come to associate with synchronous transmission.
- The control mechanisms provided by a data link protocol are not needed.
- It is instructive to ponder this point, and we do so by considering two key data link control mechanisms: flow control and error control.
- For both the multiplexer and demultiplexer, flow control is not needed.
- The data rate on the multiplexed line is fixed, and the multiplexer and demultiplexer are designed to operate at that rate.
- But suppose that one of the individual output lines attaches to a device that is temporarily unable to accept data.

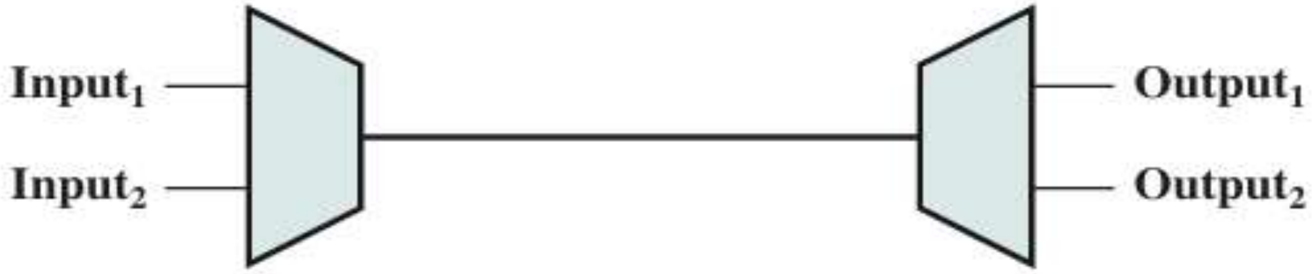
TDM LINK CONTROL

- ❖ Should the transmission of TDM frames cease?
- ❖ Clearly not, because the remaining output lines are expecting to receive data at predetermined times.
- ❖ The solution is for the saturated output device to cause the flow of data from the corresponding input device to cease.
- ❖ Thus, for a while, the channel in question will carry empty slots, but the frames as a whole will maintain the same transmission rate.
- ❖ The reasoning for error control is the same.
- ❖ It would not do to request retransmission of an entire TDM frame because an error occurs on one channel.

TDM LINK CONTROL

- ❖ The devices using the other channels do not want a retransmission nor would they know that a retransmission has been requested by some other device on another channel.
- ❖ Again, the solution is to apply error control on a per-channel basis.
- ❖ Flow control and error control can be provided on a per-channel basis by using a data link control protocol such as HDLC on a per-channel basis.

TDM LINK CONTROL



(a) Configuration

Input$_1$·········· $F_1\ f_1\ f_1\ d_1\ d_1\ d_1\ C_1\ A_1\ F_1\ f_1\ f_1\ d_1\ d_1\ d_1\ C_1\ A_1\ F_1$

Input$_2$··· $F_2\ f_2\ f_2\ d_2\ d_2\ d_2\ d_2\ C_2\ A_2\ F_2\ f_2\ f_2\ d_2\ d_2\ d_2\ d_2\ C_2\ A_2\ F_2$

(b) Input data streams

$\cdots\ f_2\ F_1\ d_2\ f_1\ d_2\ f_1\ d_2\ d_1\ d_2\ d_1\ C_2\ d_1\ A_2\ C_1\ F_2\ A_1\ f_2\ F_1\ f_2\ f_1\ d_2\ f_1\ d_2\ d_1\ d_2\ d_1\ d_2\ d_1\ C_2\ C_1\ A_2\ A_1\ F_2\ F_1$

(c) Multiplexed data stream

Legend: F = flag field; d = one octet of data field
A = address field; f = one octet of FCS field
C = control field

Figure 8.7 Use of Data Link Control on TDM Channels

DIGITAL CARRIER SYSTEMS

- ❖ Long-distance carrier system provided in the US & throughout the world was designed to transmit voice signals over high-capacity transmission links, such as optical fiber, coaxial cable, and microwave.
- ❖ Part of the evolution of telecom networks to digital technology has been the adoption of synchronous TDM transmission structures.
- ❖ AT&T developed hierarchy of TDM structures of large capacities in US.
- ❖ This structure is used in Canada and Japan as well as the United States.
- ❖ A similar, but unfortunately not identical, hierarchy has been adopted internationally under the auspices of ITU-T.
- ❖ TDM hierarchy use DS-1 transmission format multiplexing 24 channels.

DIGITAL CARRIER SYSTEMS

Table 8.3 North American and International TDM Carrier Standards

North American			International (ITU-T)		
Designation	**Number of Voice Channels**	**Data Rate (Mbps)**	**Level**	**Number of Voice Channels**	**Data Rate (Mbps)**
DS-1	24	1.544	1	30	2.048
DS-1C	48	3.152	2	120	8.448
DS-2	96	6.312	3	480	34.368
DS-3	672	44.736	4	1920	139.264
DS-4	4032	274.176	5	7680	565.148

DIGITAL CARRIER SYSTEMS

- Each frame contains 8 bits per channel plus a framing bit.

  i.e. 24 * 8 + 1 = 193 bits.
- For voice transmission, the following rules apply.
- Each channel contains one word of digitized voice data.
- The original analog voice signal is digitized using pulse code modulation at a rate of 8000 samples per second.
- Therefore, each channel slot and hence each frame must repeat 8000 times per second.
- With a frame length of 193 bits, we have a data rate of 8000 * 193 = 1.544 Mbps.

DIGITAL CARRIER SYSTEMS

- ❖ For five of every six frames, 8-bit PCM samples are used.
- ❖ For every sixth frame, each channel contains a 7-bit PCM word plus a signaling bit.
- ❖ The signaling bits form a stream for each voice channel that contains network control and routing information.
- ❖ For example, control signals are used to establish a connection or terminate a call.
- ❖ The same DS-1 format is used to provide digital data service.
- ❖ The DS-1 format can be used to carry a mixture of both voice as well as data channels.

SONET/SDH3 CABLE MODEMS

- SONET (Synchronous Optical Network) is an optical transmission interface originally proposed by BellCore and standardized by ANSI.
- A compatible version, referred to as Synchronous Digital Hierarchy (SDH), has been published by ITU-T.
- SONET is intended to provide a specification for taking advantage of the high-speed digital transmission capability of optical fiber.

Signal Hierarchy

- The SONET specification defines a hierarchy of standardized digital data rates (Table 8.4).
- The lowest level is at 51.84 Mbps.

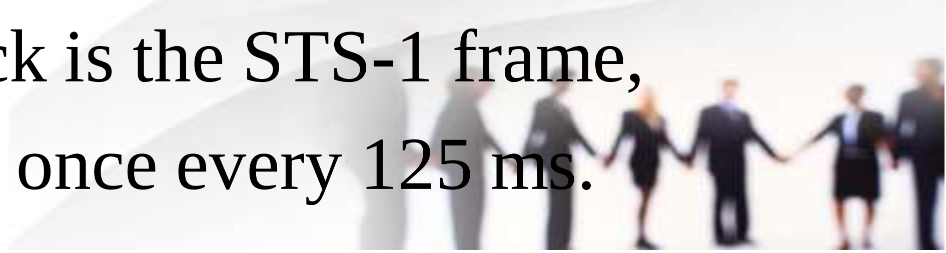
SONET/SDH3 CABLE MODEMS

Table 8.4 SONET/SDH Signal Hierarchy

SONET Designation	ITU-T Designation	Data Rate	Payload Rate (Mbps)
STS-1/OC-1		51.84 Mbps	50.112 Mbps
STS-3/OC-3	STM-1	155.52 Mbps	150.336 Mbps
STS-12/OC-12	STM-4	622.08 Mbps	601.344 Mbps
STS-48/OC-48	STM-16	2.48832 Gbps	2.405376 Gbps
STS-192/OC-192	STM-64	9.95328 Gbps	9.621504 Gbps
STS-768	STM-256	39.81312 Gbps	38.486016 Gbps
STS-3072		159.25248 Gbps	153.944064 Gbps

SONET/SDH3 CABLE MODEMS

- This rate can be used to carry a single DS-3 signal or a group of lower-rate signals, such as DS1, DS1C, DS2 at ITU-T rates (e.g., 2.048 Mbps).
- Multiple STS-1 signals can be combined to form an STS-N signal.
- The signal is created by interleaving bytes from N STS-1 signals that are mutually synchronized.
- For the ITU-T Synchronous Digital Hierarchy, the lowest rate is 155.52 Mbps, which is designated STM-1.
- This corresponds to SONET STS-3.
- Frame Format The basic SONET building block is the STS-1 frame, which consists of 810 octets and is transmitted once every 125 ms.



ASYMMETRIC DIGITAL SUBSCRIBER LINE (ADSL)

- In the implementation and deployment of a high-speed wide area public digital network, the most challenging part is the link between subscriber and network: the digital subscriber line.
- With billions of potential endpoints worldwide, the prospect of installing new cable for each new customer is daunting.
- Instead, network design ers have sought ways of exploiting the installed base of twisted-pair wire that links virtually all residential and business customers to telephone networks.
- These links were installed to carry voice-grade signals in a bandwidth from 0 to 4 kHz.

ASYMMETRIC DIGITAL SUBSCRIBER LINE (ADSL)

- However, the wires are capable of transmitting signals over a far broader spectrum - 1 MHz or more.
- ADSL is the most widely publicized family of new modem technologies designed to provide high-speed digital data transmission over ordinary telephone wire.
- ADSL is now being offered by a number of carriers and is defined in an ANSI standard.
- In this following section, we first look at the overall design of ADSL and then examine the key underlying technology, known as DMT.

ADSL DESIGN

- The term asymmetric refers to the fact that ADSL provides more capacity down stream than upstream.
- ADSL was originally targeted at the expected need for video on demand and related services.
- This application has not materialized.
- However, since the introduction of ADSL technology, the demand for high-speed access to the Internet has grown.
- User needs higher capacity for downstream than upstream transmission.
- Most of the user transmissions are in the form of keyboard strokes or in transmission of short e-mail messages.

ADSL DESIGN

- The incoming traffic, especially web traffic, can involve large amounts of data and include images or even video.
- Thus, ADSL provides a perfect fit for the Internet requirement.
- ADSL uses frequency-division multiplexing in a novel way to exploit the 1-MHz capacity of twisted pair.
- There are three elements of the ADSL strategy:
- Reserve lowest 25 kHz for the voice which is known as Plain Old Telephone Service or POTS, in short.
- The voice is carried only in the 0 to 4 kHz band
- Additional bandwidth prevents crosstalk between voice & data channels.

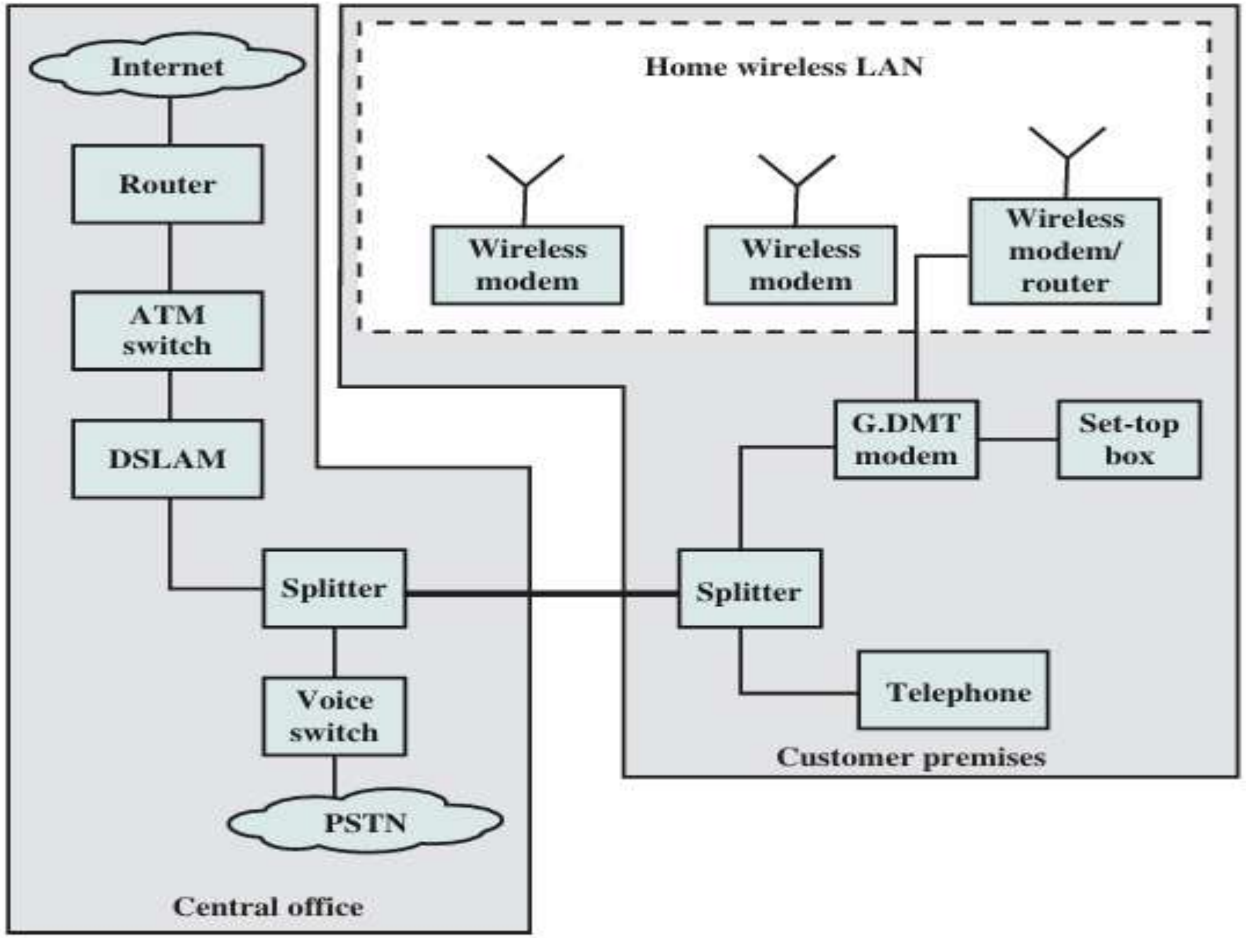
ADSL DESIGN

- ❖ Use either echo cancellation4 or FDM to allocate two bands, a smaller upstream band and a larger downstream band.
- • Use FDM within the upstream and downstream bands.
- ❖ In this case, a single bit stream is split into multiple parallel bit streams and each portion is carried in a separate frequency band.
- ❖ When echo cancellation is used, the entire frequency band for the upstream channel overlaps the lower portion of the downstream channel.
- ❖ This has two advantages compared to the use of distinct frequency bands for upstream and downstream.

BROADBAND ACCESS CONFIGURATION

- Figure 8.17 shows the configuration for broadband service using DSL.
- The DSL link is between the provider central office and the residential or business premises.
- On customer side, a splitter allow simultaneous telephone & data service.
- The data service makes use of a DSL modem.
- A DSL data signal can be divided into a video stream and a data stream.
- It connects modem to either a single local computer or to a wireless modem/router which enables the customer to support a wireless LAN.
- On the provider side, a splitter is also used to separate the telephone service from the Internet service.

BROADBAND ACCESS CONFIGURATION



ATM = asynchronous transfer mode
DSLAM = digital subscriber line access multiplexer
PSTN = public switched telephone network
G.DMT = G.992.1 discrete multitone

Figure 8.17 DSL Broadband Access

BROADBAND ACCESS CONFIGURATION

- ❖ The voice traffic is connected to a public s/w telephone network (PSTN) providing same service as an ordinary telephone line to a subscriber.
- ❖ The data traffic connects to a DSL access multiplexer which multiplexes multiple customer DSL connections on to a single high-speed ATM line.
- ❖ The ATM line connects via one or more ATM switches to a router that provides an entry point to the Internet.

MULTIPLE CHANNEL ACCESS

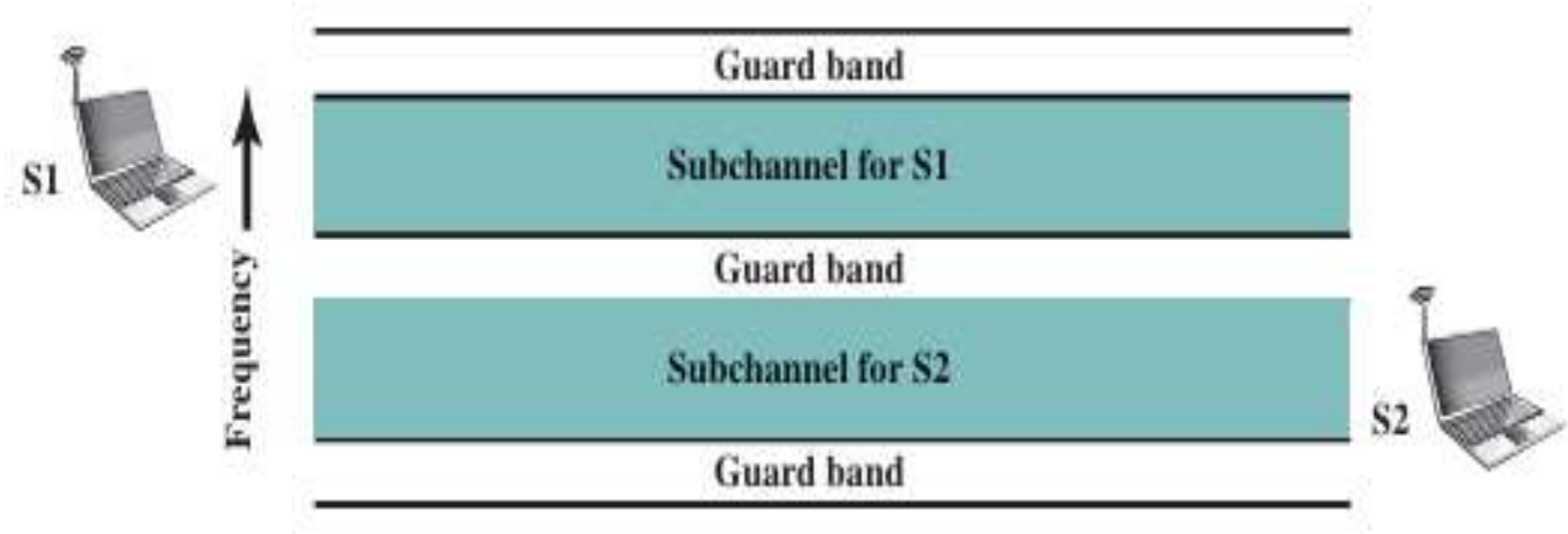
- ❖ In this section, we look at four multiplexing techniques used for sharing channel capacity among multiple transmitter/receiver stations.
- ❖ These techniques differ from the FDM and TDM techniques so far discussed, because no physical multiplexer is involved.
- ❖ Individual stations are assigned a frequency band or a sequence of time slots and transmit directly on the channel and not through a multiplexer.
- ❖ The techniques that are used as building blocks in a number of wireless schemes include wireless LANs (Wi-Fi), cellular networks, satellite networks and wireless broadband Internet access, such as WiMAX.

FREQUENCY DIVISION DUPLEX (FDD)

- **Frequency-Division Duplex (FDD)**

- Frequency-Division Duplex (FDD) is not a particularly interesting case.
- FDD simply means that two stations have a full-duplex connection in which each station transmits on a separate frequency band.
- Two frequency bands are separated from each other & from other bands on the network by guard bands, to prevent interference (Figure a).
- The combination of the two frequency bands is often referred to as a subchannel, with the combination of the two subchannels viewed as a full-duplex channel between the stations.

FREQUENCY DIVISION DUPLEX (FDD)



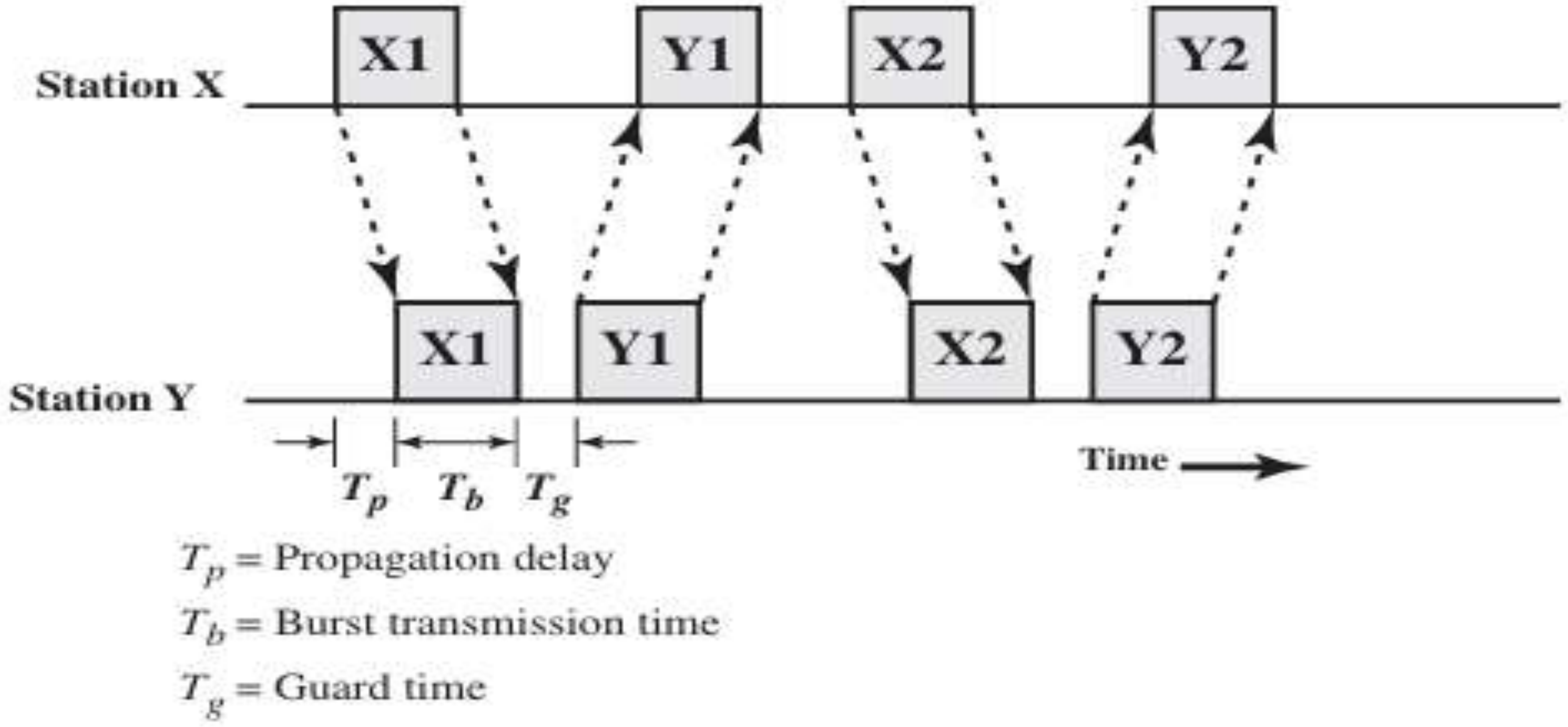
(a) Frequency-division duplex (TDD)

TIME DIVISION DUPLEX (TDD)

❑ Time Division Duplex (TDD)/Time Compression MUX (TCM)

- ❖ In TDD, data are transmitted in one direction at a time.
- ❖ To achieve the desired subscriber data rate with simple TDD, the transmitter's bit stream is divided into equal segments
- ❖ It is compressed in time to a higher transmission rate and transmitted in bursts which are expanded at the other end to the original rate.
- ❖ A short quiescent period is used between bursts going in opposite directions to allow the channel to settle down.
- ❖ Thus, the actual data rate on the channel must be greater than twice the data rate required by the two end systems.

TIME DIVISION DUPLEX (TDD)



(b) Time-division duplex (TDD)

UNIT VI

CELLULAR TELEPHONY AND SATELLITE NETWORKS

CELLULAR TELEPHONY

FREQUENCY REUSE PRINCIPLE

- ❖ In a cellular system, each cell has a base transceiver.
- ❖ The trans mission power is carefully controlled to allow communication within the cell using a given frequency, while limiting the power at that frequency that escapes the cell into adjacent ones.
- ❖ The objective is to use same frequency in other nearby cells allowing the frequency to be used for multiple simultaneous conversations.
- ❖ Depending on the traffic, 10 to 50 frequencies are assigned to each cell.
- ❖ The essential issue is to determine how many cells must intervene between two cells using the same frequency so that the two cells do not interfere with each other.

FREQUENCY REUSE PRINCIPLE

- Various patterns of frequency reuse are possible.
- Figure 10.2 shows some examples.
- If the pattern consists of N cells and each cell is assigned the same number of frequencies, each cell can have K>N frequencies, where K is the total number of frequencies allotted to the system.
- For frequency reuse, following parameters are commonly used:

  D = Min. Dist. Bet^n centers of cells using same frequency band

  R = radius of a cell

  d = distance between centers of adjacent cells (d = 13R)

  N = number of cells in a repetitious pattern = Reuse factor

FREQUENCY REUSE PRINCIPLE

- In a hexagonal cell pattern, only the following values of N are possible:

  $N = I^2 + J^2 + (I * J)$ $\qquad\qquad\qquad$ $I, J = 0, 1, 2, 3, c$

- Hence, possible values of N are 1, 3, 4, 7, 9, 12, 13, 16, 19, 21 & so on.
- Also, the following relationship holds:

$$\frac{D}{R} = \sqrt{3N}$$

- This can also be expressed as $D/d = \sqrt{N}$

ROAMING

- Roaming refers to a wireless network service extension in an area that differs from the registered home network location.
- Roaming enables a mobile device to access the Internet and other mobile services when out of its normal coverage area.
- It gives mobile user an ability to move from one access point to another.
- Roaming is derived from real-time optimally adapting mesh (ROAM).
- Roaming services are usually provided by cellular service providers as well as Internet service providers (ISP) via a cooperative agreement.
- The cellular roaming services are provided by both Global System for Mobile Communications (GSM) & CDMA operators.

ROAMING

- Services are either free or billed according to local area rates.
- Wireless telecommunication roaming services are included in the mobile or cellphone subscriber service package for use outside local n/w zones.
- GSM/WLAN roaming services are supplied in two different scenarios.
- One is SIM-based roaming & other is username/password base roaming.
- WLAN roaming services can be segmented as follows:

- **Internal Roaming**

- Implemented when a mobile station is transferred with a strong signal between access points preventing network blockage/interruption from the weak signals.

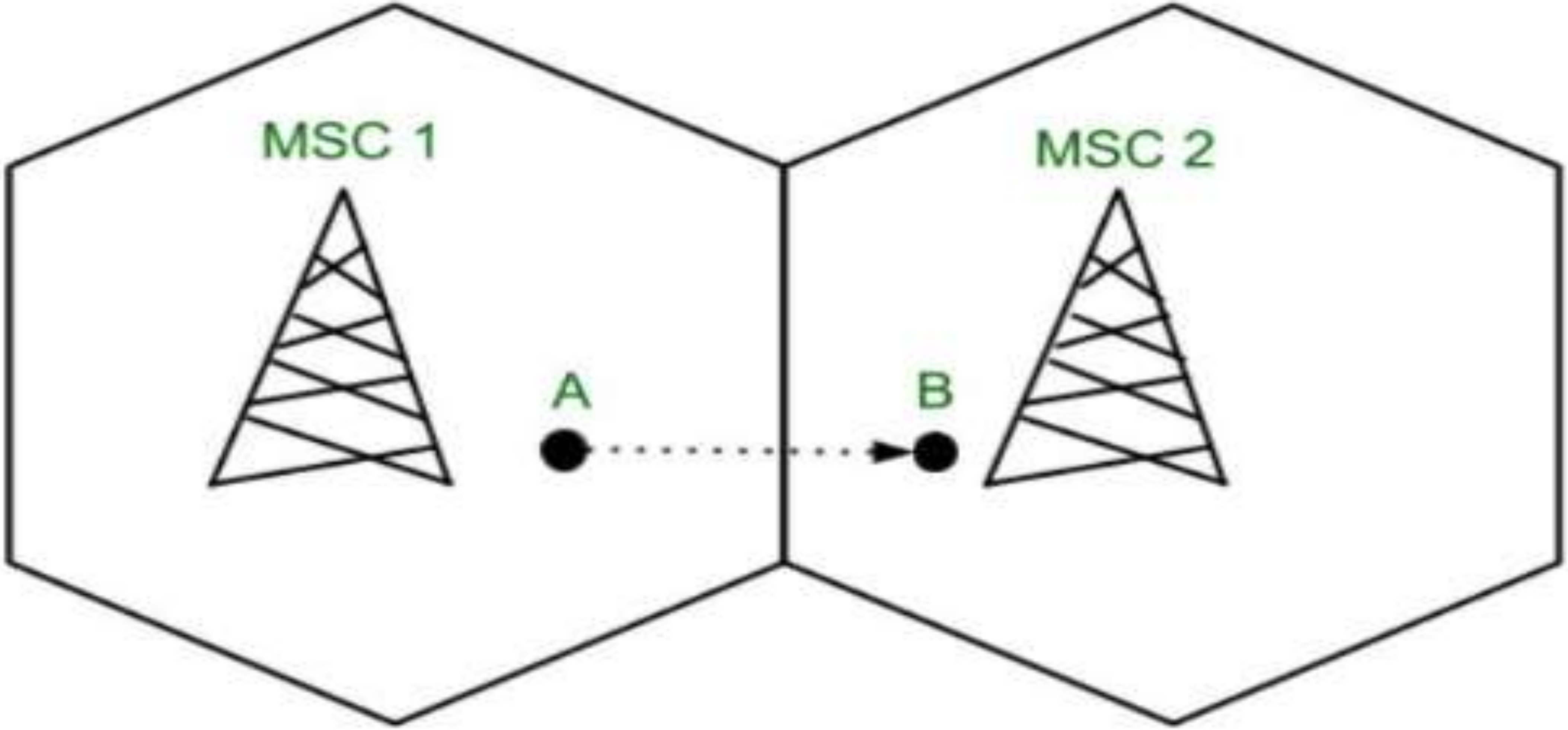
ROAMING

- **External Roaming:**
  - Implemented when a mobile station shifts to a wireless LAN or other foreign Wireless Internet Service Provider (WISP) to access service.
  - WISP allows users maintain net connection when moving in a local area.
- ISP use special software to track roaming usage & corresponding billing.
  - The subscribers should have an ISP connection that supports roaming.
  - A traveling user may route calls to the ISP's locally assigned number after logging into a foreign ISP through a computer modem.
  - The foreign ISP provides an Internet access after validating the user's home mail server.

HANDOFF

- In cellular telecommunications, the terms handover or handoff refers to the process of transferring an ongoing call or data connectivity from one Base Station to another Base Station.
- When a mobile moves into a different cell while the conversation is in progress then the MSC (Mobile Switching Center) transfers the call to a new channel belonging to the new Base Station.
- When a mobile user A moves from one cell to another cell then BSC 1 signal strength loses for the mobile User A and the signal strength of BSC 2 increases and thus ongoing calls or data connectivity for mobile users goes on without interrupting.

HANDOFF



TYPES OF HANDOFF

❑ Hard Handoff

- ❖ When there is an actual break in the connectivity while switching from one Base Station to another Base Station.
- ❖ There is no burden on the Base Station and MSC because the switching takes place so quickly that it can hardly be noticed by the users.
- ❖ The connection quality is not that good.
- ❖ Hard Handoff adopted the ‘break before make’ policy.
- ❖ It is generally implemented in TDM and FDM when a user connects to the base station with a fluctuating radio frequency.
- ❖ Hard Handoff is efficient, cheaper in cost as compared to soft Handoff.

TYPES OF HANDOFF

❑ Soft Handoff

- ❖ Soft Handoff is a mechanism in which the device gets connected with two or more base stations at the same time.
- ❖ At least one of the links is kept when radio signals are added or removed to the Base Station.
- ❖ Soft Handoff adopted the ‘make before break’ policy.
- ❖ If a channel is in power loss then another channel will always be on standby mode so this makes it best in terms of quality as compared to Hard handoff.
- ❖ Soft handoffs are used in devices supporting CDMA/WDMA networks

TYPES OF HANDOFF

- ❖ High Transmission speed as more than one repeater can transmit signals.
- ❖ It has a very low delay in signals.
- ❖ It can’t be implemented on devices supporting GSM or LTE networks.

❑ **Delayed Handoff**

- ❖ Delayed handoff occurs when no base station is available for accepting the transfer.
- ❖ The call continues until the signal strength reaches a threshold, and after that, the call is dropped.
- ❖ Generally, it happens when the user is out of the network coverage area, or at some dead spots where network reach is very low.

TYPES OF HANDOFF

❑ Mobile Assisted Handoff

- ❖ Mobile Assisted handoff is generally used when a mobile phone helps one base station to transfer the call to another base station with a better improvised connectivity and more signal strength.
- ❖ This handoff is used in TDMA technique-based GSM devices.

GENERATIONS OF CELLULAR TELEPHONY

- Since their introduction in the mid-1980s, cellular networks have evolved rapidly.
- For convenience, industry and standards bodies group the technical advances into “generations.”
- We are now using up to the fourth/fifth generation (4G/5G) of cellular network technology.
- In this section, we give a brief overview of the four generations.
- The following section is devoted to 4G.
- Table 10.1 lists some of the key characteristics of the cellular network generations.

GENERATIONS OF CELLULAR TELEPHONY

Table 10.1 Wireless Network Generations

Technology	1G	2G	2.5G	4G
Design began	1970	1980	1985	2000
Implementation	1984	1991	1999	2012
Services	Analog voice	Digital voice	Higher capacity packetized data	Completely IP based
Data rate	1.9. kbps	14.4 kbps	384 kbps	200 Mbps
Multiplexing	FDMA	TDMA, CDMA	TDMA, CDMA	OFDMA, SC-FDMA
Core network	PSTN	PSTN	PSTN, packet network	IP backbone

FIRST GENERATION

- ❖ The original cellular networks (1G) provided analog traffic channels and were designed to be an extension of public switched telephone networks.
- ❖ Users with brick-sized cell phones placed and received calls in the same fashion as landline subscribers.
- ❖ The most widely deployed 1G system was the Advanced Mobile Phone Service (AMPS), developed by AT&T.
- ❖ This approach was also common in South America, Australia, and China.
- ❖ In North America, two 25-MHz bands were allocated to AMPS, one for transmission from a base station to the mobile unit (869–894 MHz) & other for transmission from a mobile to the base station (824–849 MHz).

FIRST GENERATION

- Each of these bands is split in two to encourage competition.
- That means, two operators can be accommodated in each market.
- An operator is allocated only 12.5 MHz in each direction for its system.
- The channels are spaced 30 kHz apart allowing 416 channels/operator.
- Twenty-one channels are allocated for control, leaving 395 to carry calls.
- The control channels are data channels operating at 10 kbps.
- The conversation channels carry the conversations in analog using the frequency modulation (FM).
- Simple FDMA is used to provide multiple access.

FIRST GENERATION

- Control information is sent on conversation channels in bursts as data.
- This number of channels is inadequate for most major markets, so some way must be found either to use less bandwidth per conversation or to reuse frequencies.
- Both approaches are considered in various approaches to 1G telephony.
- For AMPS, frequency reuse is exploited.

SECOND GENERATION

- First-generation cellular networks, such as AMPS, quickly became highly popular, threatening to swamp available capacity.
- Second-generation systems (2G) provided higher-quality signals, higher data rates for support of digital services & greater capacity.
- Key differences between 1G and 2G networks include:

- **Digital traffic channels:**

- The most notable difference between the two generations is that 1G systems are almost purely analog, whereas 2G systems are digital.
- In particular, the First Generation systems are designed to support voice channels using FM.

SECOND GENERATION

- ❖ The digital traffic is supported only by the use of a modem that converts the digital data into analog form.
- ❖ 2G systems provide digital traffic channels and readily support digital data while the voice traffic is encoded in digital form before transmitting.

- **Encryption:**

- ❖ Since all of the user traffic, as well as the control traffic, is digitized in 2G systems, it is relatively simple to encrypt all the traffic present in the system to prevent eavesdropping.
- ❖ All 2G systems provide this capability, whereas 1G systems send user traffic in the clear, providing no security.

SECOND GENERATION

- **Error detection and correction:**

❖ The digital traffic stream of 2G systems also lends itself to the use of error detection and correction techniques.

❖ The result can be very clear voice reception.

- **Channel access:**

❖ In 1G systems, each cell supports a number of channels.

❖ At any given time a channel is allocated to only one user.

❖ 2G systems also provide multiple channels per cell, but each channel is dynamically shared by a number of users using TDMA (time-division multiple access) or CDMA (code division multiple access).

THIRD GENERATION

- The objective of third generation (3G) of wireless communication is to provide fairly high-speed wireless communications to support multimedia, data, and video in addition to voice.
- The ITU's International Mobile Telecommunications for year 2000 (IMT-2000) initiative has defined 3G capabilities as follows:

- Voice quality comparable to the public switched telephone network
- 144 kbps data rate available to users in motor vehicles over large areas
- 384 kbps available to pedestrians standing/moving over small areas
- Support (to be phased in) for 2.048 Mbps for office use
- Symmetrical and asymmetrical data transmission rates

THIRD GENERATION

- Support for both packet-switched and circuit-switched data services
- An adaptive interface to the Internet to reflect efficiently the common asymmetry between inbound and outbound traffic
- More efficient use of the available spectrum in general
- Support for a wide variety of mobile equipment
- Flexibility to allow the introduction of new services and technologies

❖ The dominant technology for 3G systems is CDMA.

❖ All in all, three different CDMA schemes have been adopted in case of the Third Generation systems.

❖ The 3G systems share the following design features:

THIRD GENERATION

- **Bandwidth:**

❖ A major design goal for a 3G system is to limit channel usage to 5 MHz.

❖ There are several reasons for this goal.

❖ On the one hand, a bandwidth of 5 MHz or more improves the receiver’s ability to resolve multipath when compared to narrower bandwidths.

❖ On the other hand, available spectrum is limited by competing needs, and 5 MHz is a reasonable upper limit on what can be allocated for 3G.

❖ Finally, 5 MHz is adequate for supporting data rates of 144 and 384 kHz, the main targets for 3G services.

THIRD GENERATION

- **Chip rate:**
  - ❖ Given the bandwidth, the chip rate depends on desired data rate, the need for error control, and bandwidth limitations.
  - ❖ A chip rate of 3 Mcps or more is reasonable for these design parameters.
- **Multirate:**
  - ❖ The term multirate refers to the provision of multiple fixed-data-rate logical channels to a given user in which different data rates are provided on different logical channels.
  - ❖ The traffic on each logical channel can be switched independently through the wireless & fixed networks to different destinations.

FOURTH GENERATION

- An advantage of multirate is that the system can flexibly support multiple simultaneous applications from a given user & can efficiently use available capacity by providing a capacity required for each service.
- Evolution of smartphones & cellular networks has ushered in a new generation of capabilities and standards, which is collectively called 4G.
- 4G systems provide ultra-broadband Internet access for a variety of mobile devices including laptops, smartphones, and tablets.
- They support mobile web access & high-bandwidth applications such as HD mobile TV, mobile video conferencing, and gaming services.

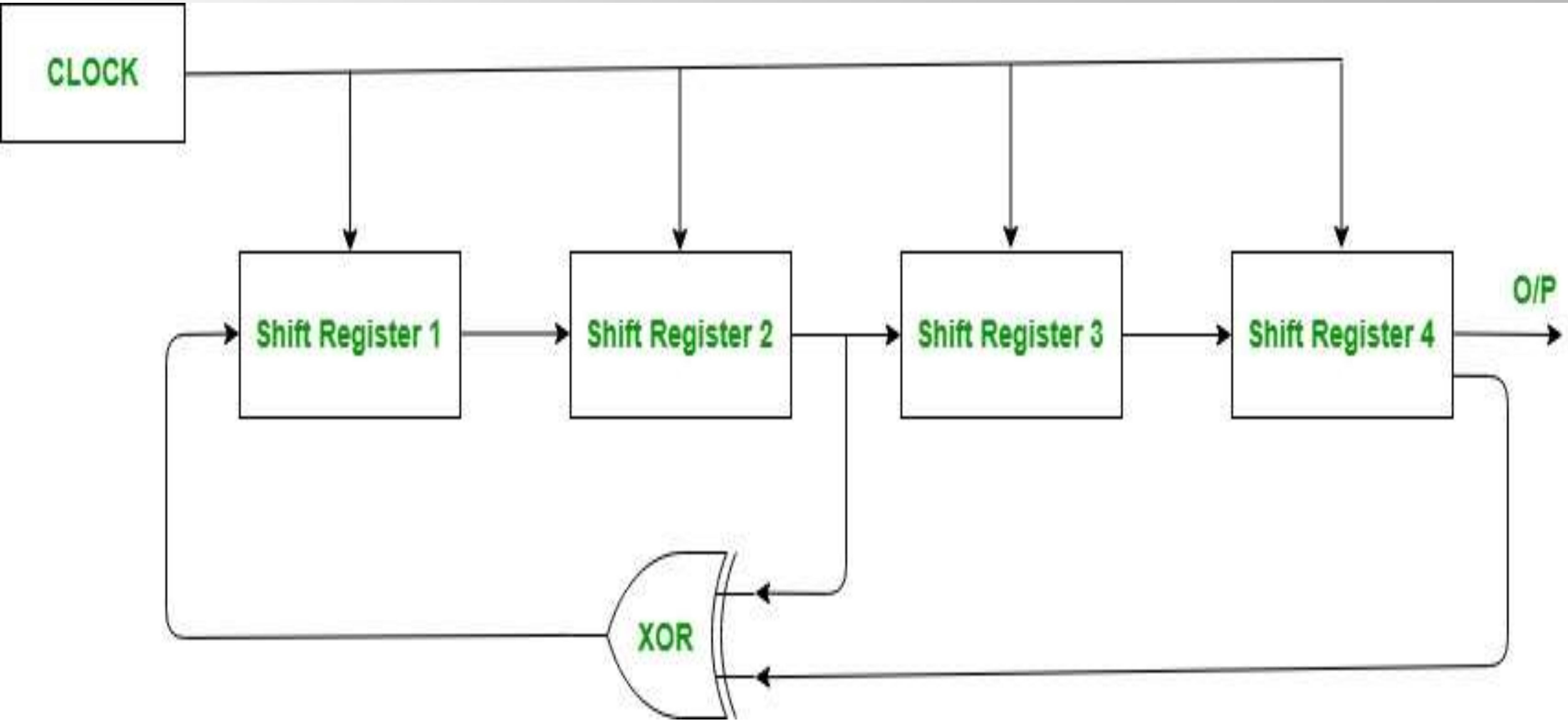
IS 95 STANDARD

- ❖ IS-95 stands for **Interim Standard 95** and is also known as **CDMAOne**.
- ❖ It was the first ever CDMA-based digital cellular technology and was developed by Qualcomm.
- ❖ It is an 2G cellular system based on DS-CDMA.
- ❖ To understand IS-95 we need to understand DS and CDMA separately.
- ❖ **DSSS** is Direct Sequence Spread Spectrum Technique which is a spread spectrum technique in which the data to be transmitted is encoded using spreading code and received and then decoded using the same code.
- ❖ It is used to avoid interference, spying and jamming.
- ❖ The spreading code used is known to transmitter and receiver only.

IS 95 STANDARD

- **CDMA** stands for Code Division Multiple Access.
- It uses the same bandwidth for all the users.
- However, each user is assigned a separate code which differentiates the from each other.
- Narrow bandwidth signals are multiplied with a very large bandwidth signals called Pseudo Noise Code Sequence (PN code).
- Each user has its own PN code which is orthogonal to each other.
- Auto-correlation is maximum & cross-correlation is zero for PN codes.
- Repeats itself after a very large time period & thus appears to be random.
- PN Sequence is generated by *Linear Feedback Shift Register*.

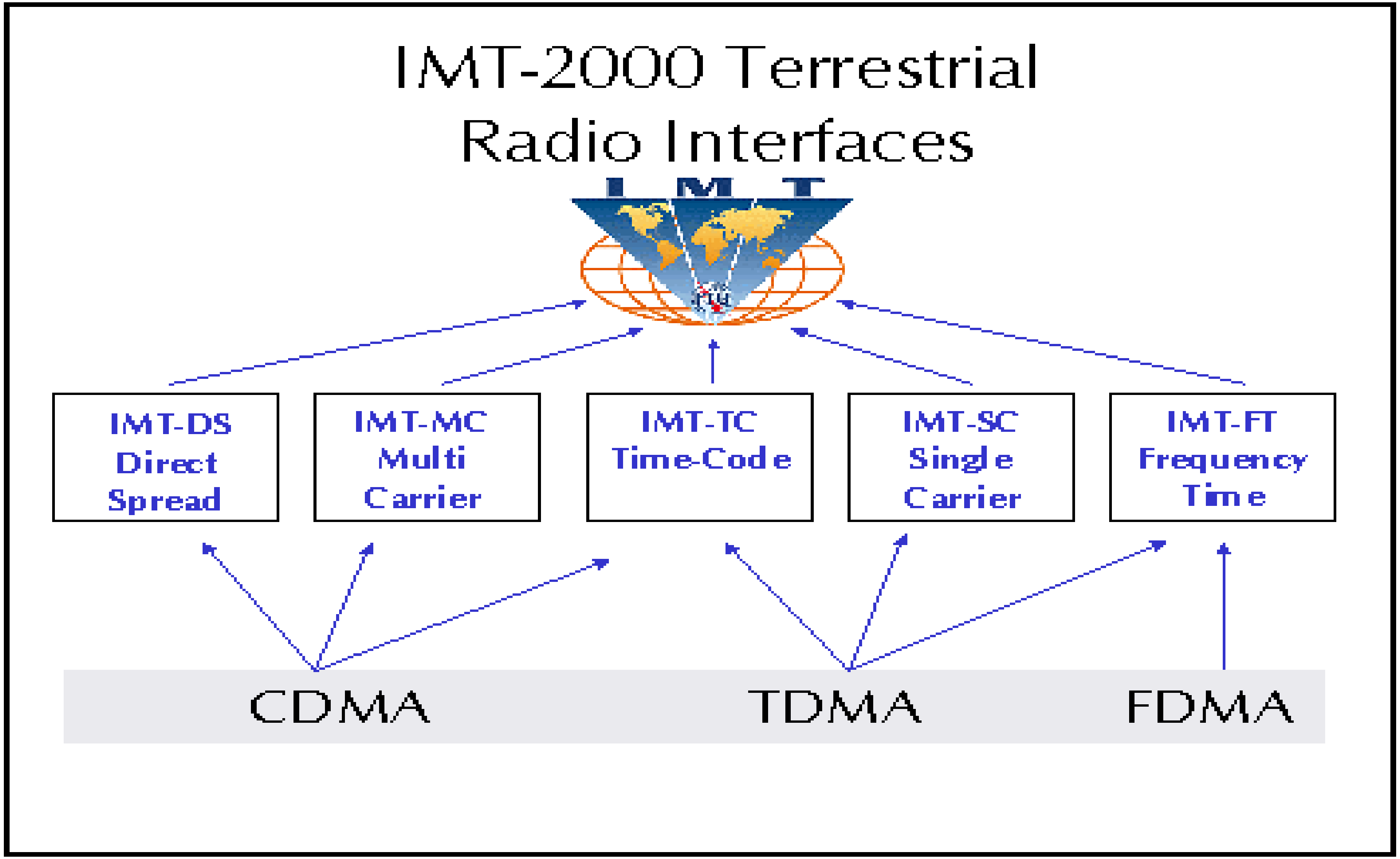
IS 95 STANDARD



IMT 2000 RADIO INTERFACE

- The group of radio experts on IMT-2000 meeting in Helsinki from 25 October to 5 November 1999 approved a comprehensive set of terrestrial and satellite radio interface specifications for IMT-2000.
- They incorporate the flexibility required by existing mobile operators to seamlessly evolve their pre-IMT-2000 networks towards third generation service capabilities as well as meeting the various specific needs of operators of new satellite and terrestrial systems.
- The IMT-2000 terrestrial standard consists of a set of radio interfaces which allow performance optimization in a wide range of radio operating environments (see figure given below).

IMT 2000 RADIO INTERFACE



IMT 2000 RADIO INTERFACE

- IMT-2000 provides wireless multimedia service capabilities, harnessing the power of the Internet for e-commerce while on the move, instant access to personal or business information and for entertainment.
- Economic benefits of new global standards are felt by all ITU Member States as wireless access provides a cost-effective solution to the "telecommunications gap" between developing & developed countries.
- "It is particularly important to narrow this gap as we move into the 21st century where timely access to information will be essential for economic progress" Mr. Yoshio Utsumi, ITU Secretary-General said at the close of the meeting.

IMT 2000 RADIO INTERFACE

- Agreement on IMT-2000 radio specifications mean a full interoperability and interworking of mobile systems can be achieved.
- The mobile terminal manufacturers will build units to work anywhere in the world irrespective of the specific network or radio options chosen by the operator providing mobile services at that specific location.
- “With IMT-2000, mobile users should see an increasing range of service capabilities which they are likely to want to enjoy anywhere they go”.
- "The ITU has delivered on its commitment to provide the international community with a standard that enables full interoperability of third generation mobile systems", Mr. Yoshio Utsumi said.

IMT 2000 RADIO INTERFACE

- "The extent to which the promises of this new wave of communication can become a reality for every consumer now depends on the delivery of the IMT-2000 implementation" he concluded.
- Only if all IMT-2000 radio options are available everywhere, can operators truly compete globally and manufacturers achieve maximum possible economies of scale.
- IMT-2000 radio specifications, additional IMT-2000 recommendations on QoS & the use of High Altitude Platform Stations was presented to ITU-R Study Group 8 for endorsement at its meeting in Geneva that took place during10-12 November 1999.

SATELLITE NETWORKS

ORBITS AND FOOTPRINTS IN SATELLITE NETWORKS

- Satellite orbits have fundamental characteristics that principally vary with their radius & orientation with respect to the Earth's axis of rotation.
- The geostationary orbit has an altitude of 36,000 km orthogonal to rotational axis,
- This implies that satellites in this orbit have the same angular speed as the Earth and thus appear stationary to one portion of the Earth's surface.
- This orbit is seldom used in geology, but is important for communication and weather satellites for obvious reasons.
- The large majority of imaging sensors are on sun-synchronous, near-polar orbits.

ORBITS AND FOOTPRINTS IN SATELLITE NETWORKS

- ❖ Understanding the concept of a satellite footprint is crucial for anyone involved in satellite communications, broadcasting, or even just curious about how global connectivity works.
- ❖ It's the area on Earth's surface that a satellite's signals can reach and knowing its intricacies can unlock a world of technological possibilities.
- ❖ From ensuring your favorite channels are crystal clear to maintaining seamless global communication networks, the footprint of a satellite plays a pivotal role.
- ❖ It is fundamental in ensuring that the communication and positioning information is accurately transmitted to & from devices across the globe.

ORBITS AND FOOTPRINTS IN SATELLITE NETWORKS

- ❖ First, satellite footprints determine the coverage area where signals from a satellite can be received.
- ❖ For GNSS/GPS antennas, this means identifying the specific geographic regions where devices can connect to satellite signals for navigation and timing information.
- ❖ Imagine satellite footprint as an invisible net cast over an earth's surface.
- ❖ The size and shape of this net are carefully calculated to optimize signal strength and ensure consistent connectivity.
- ❖ This is particularly critical for applications ranging from navigation in our cars to timing in financial transactions.

ORBITS AND FOOTPRINTS IN SATELLITE NETWORKS

- ❖ Second, the satellite footprint impacts the **Quality of Service (QoS)**.
- ❖ The GNSS/GPS antennas receive strong, clear signals, leading to accurate positioning and reliable data transmission.
- ❖ Outside the footprint, signal quality drops, leading to the inaccuracies.
- ❖ This is why understanding the satellite footprint is indispensable for anyone deploying or using GNSS/GPS antennas, including military operations, search and rescue teams, and commercial aviation.
- ❖ Furthermore, the satellite footprint guides the placement and orientation of GNSS/GPS antennas.
- ❖ For optimal signal reception, antennas are aligned with satellite footprint.

ORBITS AND FOOTPRINTS IN SATELLITE NETWORKS

- This ensures that they are always within the path of the strongest signals, enhancing the efficiency and reliability of the technology in use.
- For surveying, agriculture, or urban planning, a correct alignment with satellite footprint is a cornerstone for successful GNSS/GPS operation.
- In essence, the satellite footprints play a pivotal role in the global networking of GNSS/GPS technologies.
- By enabling precise positioning, enhancing signal quality and guiding the deployment of GNSS/GPS infrastructure, these footprints ensure that we remain connected and on course, no matter where we are on Earth.

TYPES OF SATELLITES

- This orbit allows the satellite to fly over the same area at the same time of the day and provides a nearly global coverage of the Earth over a period of typically 1–2 weeks.
- Four different types of orbits can be identified as shown in the Figure given below:

- **Geostationary (or geosynchronous) earth orbit (GEO):**

- GEO satellites have a distance of almost 36,000 km to the earth.
- TV and radio broadcast weather satellites and satellites operating as backbones for the telephone network are some of the examples.

TYPES OF SATELLITES

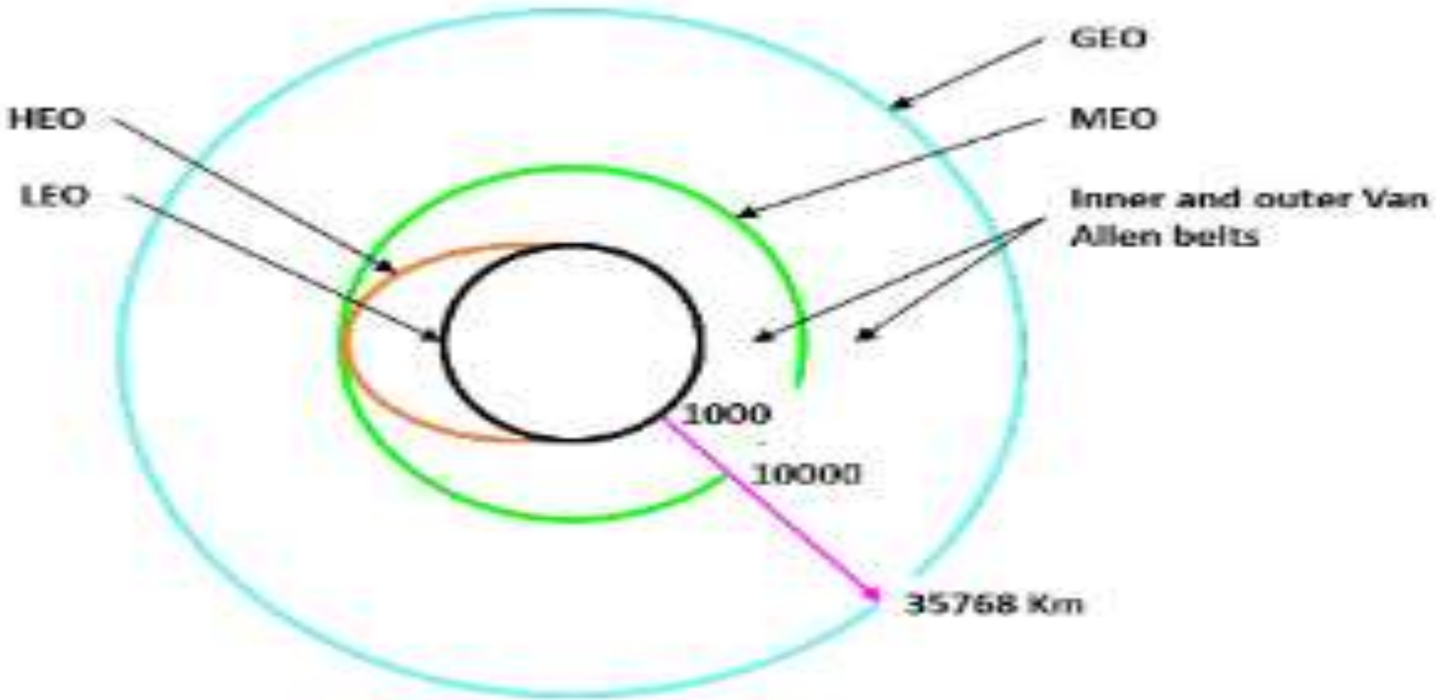


Figure 9 : Different types of satellite orbits

TYPES OF SATELLITES

- **Medium earth orbit (MEO):**
  - MEOs operate at a distance of about 5,000–12,000 km some upcoming systems ICO are example of this type of orbit.
- **Low earth orbit (LEO):**
  - Initially LEO satellites were mainly used for espionage, nowadays many satellites use this class with altitudes of 500–1,500 km.
  - Globalstar, Iradium are examples of LEO satellites.
- **Highly elliptical orbit (HEO):**
  - All the satellites with noncircular orbits belong to this class.
  - Currently, very few satellites using the elliptical orbits are planned.

TRILATERATION IN GPS

- Have you ever wondered how your GPS receiver works?
- They use a technique called trilateration.
- Despite how GPS receivers are often confused with triangulation (which measures angles), they really don’t use angles at all.
- Trilateration involves measuring distances.
- How does the GPS system pinpoint your location using trilateration?
- Using a simple two-dimensional example, let’s imagine we have three GPS satellites each with a known position in space.
- Really, all that satellites do is broadcast a signal for your GPS receiver to pick up with a specific time and distance.

TRILATERATION IN GPS

- For example, the first satellite broadcasts a signal that eventually hits your GPS receiver.
- We don't know the angle, but we do know the distance.
- That's why this distance forms a circle equal in all directions.
- This means that your GPS position could be **anywhere on this circle** at this specific radius.
- What happens when your GPS receives a second signal?
- Again, this distance is equally broadcasted in all directions until it hits your GPS receiver.
- This means that the distance could be anywhere on that circle.

TRILATERATION IN GPS

- But this time, we have two known distances from two satellites.
- With two signals, the precise position could be any of the two points where the circles intersect.
- Because we have a third satellite, it reveals your true location where all three circles intersect.
- Using three distances, trilateration can pinpoint a precise location.
- Each satellite is at the center of a sphere and where they all intersect is the position of the GPS receiver.
- As the position of the GPS receiver moves, the radius of each circle (distance) will also change.

TRILATERATION IN GPS

- But the reality is in our three-dimensional world that GPS satellites broadcast signals as a sphere.
- Each satellite is at the center of a sphere.
- Where all spheres intersect determines the position of the GPS receiver.