SCTE Business Data Solutions Course

Working for the Benefit of the Broadband Industry

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Welcome to the SCTE Manual

This handbook is designed as a stand-alone reference manual for technicians working in the broadband telecommunications industry. It may be used either on its own or as an integral part of a classroom course including practical work to enable the student to progress to examination and certification.

We hope you and your career benefit greatly from this handbook and associated training course. Please consider joining the SCTE and taking advantage of the benefits that come from being part of the industry's foremost technical institution.

About the SCTE

Founded in 1945, the SCTE is a non-profit making organisation, managed by an Executive Committee of elected volunteers, whose aim is to raise the standard of broadband engineering in the telecommunications industry. The Society particularly concerns itself with the training and career advancement of technical professionals in this field.

First introduced in 1994, the SCTE training courses have achieved wide acceptance as the standard for young technicians wishing to enter the field of cable telecommunications and for those wishing to advance their knowledge and career prospects. They are used in-house by a number of operating companies and SCTE engineers can be found working in a variety of international organisations.

As a Learned Society, SCTE is able to provide accreditation and certification for its members, giving them professional standing within the industry. Full Members and Fellows are allowed to use the designations MSCTE and FSCTE after their names whilst Technician Members may use TMSCTE. There are also categories for Student and Associate Members which carry the designations SMSCTE and AMSCTE respectively.

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SCTE Business Data Solutions Course

Module 1
Digital Telephony

Section One - Communications Overview

1.1 Introduction

1.1.1 Telephony

The field of technology involving the development, application, and deployment of telecommunication services for the purpose of electronic transmission of voice, fax, or data, between distant parties. The history of telephony is intimately linked to the invention and development of the telephone.

Telephony is commonly referred to as the construction or operation of telephones and telephonic systems and as a system of telecommunications in which telephonic equipment is employed in the transmission of speech or other sound between points, with or without the use of wires.

The term is also used frequently to refer to computer hardware, software, and computer network systems that perform functions traditionally performed by telephone equipment. In this context the technology is specifically referred to as Internet telephony or voice over Internet Protocol (VoIP).

The first telephones were connected directly in pairs. Each user had a separate telephone wired to the locations he might wish to reach. This quickly became inconvenient and unmanageable when people wanted to communicate with more than a few people. The inventions of the telephone exchange provided the solution for establishing telephone connections with any other telephone in service in the local area. Each telephone was connected to the exchange via one wire pair, the local loop. Nearby exchanges in other service areas were connected with trunk lines and long distance service could be established by relaying the calls through multiple exchanges.

Initially the switchboards were manually operated by an attendant, a switchboard operator. When a customer cranked a handle on the telephone, it turned on an indicator on the board in front of the operator who would plug the operator headset into that jack and offer service. The caller had to ask for the called party by name, later by number, and the operator connected one end of a circuit into the called party jack to alert them. If the called station answered the operator disconnected their headset and complete the station-to-station circuit. Trunk calls were made with the assistance of other operators at other exchangers in the network.

In modern times, most telephones are plugged into telephone jacks. The jacks are connected by inside wiring to a drop wire which connects the building to a cable. Cables usually bring a large number of drop wires from all over a district access network to one wire center or telephone exchange. When a telephone user wants to make a telephone call, equipment at the exchange examines the dialed telephone number and connects that telephone line to another in the same wire center, or to a trunk to a distant exchange. Most of the exchanges in the world are interconnected through a system of larger switching systems, forming the public switched telephone network (PSTN).

After the middle of the 20th century, fax and data became important secondary users of the network created to carry voices, and late in the century, parts of the network were upgraded with ISDN and DSL to improve handling of such traffic.

Today, telephony uses digital technology (digital telephony) in the provisioning of telephone services and systems. Telephone calls can be provided digitally, but may be restricted to cases in which the last mile is digital, or where the conversion between digital and analogue signals takes place inside the telephone. This advancement has reduced costs in communication, and improved the quality of voice services. The first implementation of this, ISDN, permitted all data transport from end-to-end speedily over telephone lines. This service was later made much less important due to the ability to provide digital services based on the IP protocol.

1.1.2 Telecommunication

The word telecommunication was adapted from the French. It is a compound of the Greek prefix tele-, meaning "distant", and the Latin communicare, meaning "to share" in 1904 by the French engineer and novelist Édouard Estaunié.

Defined as "Any transmission, emission or reception of signs, signals, writings, images and sounds or intelligence of any nature by wire, radio, optical or other electromagnetic systems."

Telecommunication occurs when the exchange of information between two or more entities (communication) includes the use of technology. Communication technology uses channels to transmit information (as electrical signals), either over a physical medium (such as signal cables), or in the form of electromagnetic waves. The word is often used in its plural form, telecommunications, because it involves many different technologies.

Early means of communicating over a distance included visual signals, such as beacons, smoke signals, semaphore telegraphs, signal flags, and optical heliographs. Other examples of pre-modern long-distance communication included audio messages such as coded drumbeats, lung-blown horns, and loud whistles. Modern technologies for long-distance communication usually involve electrical and electromagnetic technologies, such as telegraph, telephone, and teleprinter, networks, radio, microwave transmission, fibre optics, and communications satellites.

1.2 Recent Developments

The term's scope has been broadened with the advent of the different new communication technologies. In its broadest sense, the terms encompasses phone communication, Internet calling, mobile communication, faxing, voicemail and video conferencing. Telephony's initial idea returns to POTS, (plain old telephone service) technically called the PSTN (public-switched telephone network).

This system is being fiercely challenged by and to a great extent yielding to Voice over IP (VoIP) technology, which is also commonly referred to as IP Telephony and Internet Telephony. IP telephony is a modern form of telephony which uses the TCP/IP protocol popularized by the Internet to transmit digitized voice data. Also, unlike traditional phone service, IP telephony service is relatively unregulated by government. In the United States, the Federal Communications Commission (FCC) regulates phone-to-phone connections, but says they do not plan to regulate connections between a phone user and an IP telephony service provider. Using the Internet, calls travel as packets of data on shared lines, avoiding the tolls of the PSTN. The challenge in IP telephony is to deliver the voice, fax, or video packets in a dependable flow to the user. Much of IP telephony focuses on that challenge.

1.2.1 Digital Telephony

Starting with the introduction of the transistor, invented in 1947 by Bell Laboratories, to amplification and switching circuits in the 1950s, and through development of computer-based electronic switching systems, the public switched telephone network (PSTN) has gradually evolved towards automation and digitization of signaling and audio transmissions.

Digital telephony is the use of digital electronics in the operation and provisioning of telephony systems and services. Since the 1960s a digital core network has replaced the traditional analogue transmission and signaling systems, and much of the access network has also been digitized.

Digital telephony has dramatically improved the capacity, quality, and cost of the network. End-to-end analogue telephone networks were first modified in the early 1960s by upgrading transmission networks with Digital Signal 1 (DS1/T1) carrier systems, designed to support the basic 3kHz voice channel by sampling the bandwidth-limited analogue voice signal and encoding using Pulse code Modulation (PCM). While digitization allows wideband voice on the same channel, the improved quality of a wider analogue voice channel did not find a large market in the PSTN.

Later transmission methods such as fibre optic transmission further advanced digital transmission. Although analogue carrier systems existed that multiplexed multiple analogue voice channels onto a single transmission medium, digital transmission allowed lower cost and more channels multiplexed on the transmission medium. Today the end instrument often remains analogue but the analogue signals are typically converted to digital signals at the serving area interface (SAI), central office (CO), or other aggregation point. Digital loop carriers (DLC) place the digital network ever closer to the customer premises, relegating the analog local loop to legacy status.

With the advancement in telecommunications, packet traffic is rapidly becoming the mainstream of data traffic. The use and deployment of Synchronous Digital Hierarchy (SDH) networks for interconnection has gained traction worldwide due to its flexibility and standard for interconnecting multiple vendors, low operating cost and the high quality of service it provides. Plesiochronous Digital Hierarchy (PDH) on the other hand has been used before the introduction of the SDH standard and it also provides a means to transport large quantity of data via digital equipment such as radio wave systems, optic fibre and microwaves.

In modern telecommunication systems, the increasing demand for new services, like video and data, calls for more complicated transmission methods, higher communication speeds, and more complex network topologies. These requests, in turn, impose high design accuracy and perfect synchronization techniques of data signals. The term ‘Synchronization’ is nowadays broadly used in telecom to encompass the methods that enable oscillators at different locations to be set to the same frequency within specified limits. With the introduction of PCM for telephony in the late 1960s which allows a single line to be used by multiple signals; using a digital time - domain multiplexing where the analogue telephone signal is sampled, quantized and transmitted, network communications were being changed into digital technology and the demand for a bigger bit rate also increased. PDH was introduced by ITU - T G.702 to cope with the increasing demand for higher bit rates; it uses a basic multiplex of 2Mbps with other stages of 8, 34 and

140Mbps. Due to the fact that PDH wasn't quite synchronous, multiplexers uses a little overhead on their high speed trunks to help cater for the differences in the data rates of streams in ports with low speed.

Due to the varying developments adopted by different networks, interconnecting gateways between networks was expensive and difficult; also PDH was not flexible which made monitoring and management more difficult to realize. SDH was developed to fix some of the limitations experienced in PDH. As more people began to use SDH, management capabilities increased because of the comprehensive monitoring and the high capability management throughout the network.

The difference between Synchronous, Plesiochronous, Isochronous and Asynchronous.
These are all different ways of synchronising a data stream between the transmitter of the data stream and the receiver. They all refer to how a data stream is clocked.

- **Synchronous (Synchronised)**
  All of the clocks are synchronised to a master reference clock. They may be out of phase with each other but they will run at exactly the same frequency.

- **Plesiochronous (Almost Synchronised)**
  All of the clocks run at the same frequency to a defined precision. These clocks are not synchronised to each other so the data streams will run at slightly different rates.

- **Isochronous (Synchronised)**
  An Isochronous data stream has the timing information embedded in it (eg. a G.704 stream). These data streams can be carried over Synchronous or Plesiochronous networks.

- **Asynchronous (Not Synchronised)**
  The clocks are not synchronised. The transmitter and the receiver have independent clocks that have no relationship with each other.

Section Two - Communication Basics

2.1 Overview

Telecommunication Signal Feature:-
Amplitude is the maximum excursion from the zero value, and is generally measured in volts (V) or amps (A).

For periodic signals, the number of repetitions of the signal in one second is called the Frequency of the signal, measured in Hz and its multiples.

Frequency has an inverse relationship to the wavelength, λ (lambda). $f = c/\lambda$

- c = speed of light in vacuum (3×10^8 m/s)
- f = frequency (cycles per second, or Hz)
- λ = wavelength (meters)

The period, usually denoted by T, is the duration of one cycle, and is the reciprocal of the frequency f: $T = 1/f$

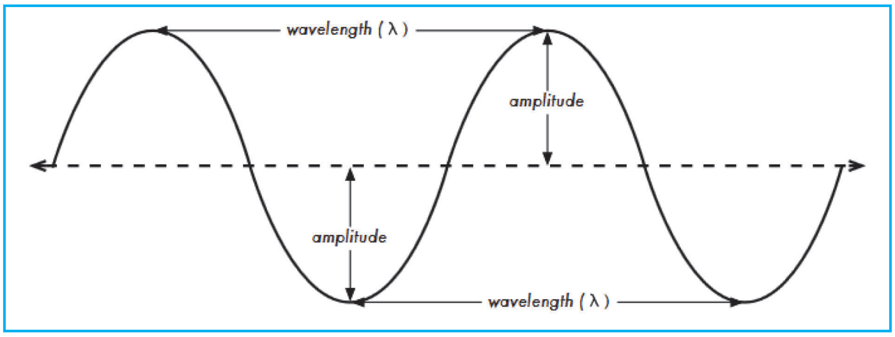

Figure 2.1: Wavelength and Amplitude

The Power of an electric signal is given by the product of its voltage and current and is measured in watts (W).

The Energy of the signal is give by the product power over the time considered and is measured in joules (J), and also in Wh, with its multiple, the kWh (kilo watt hour) most commonly used.

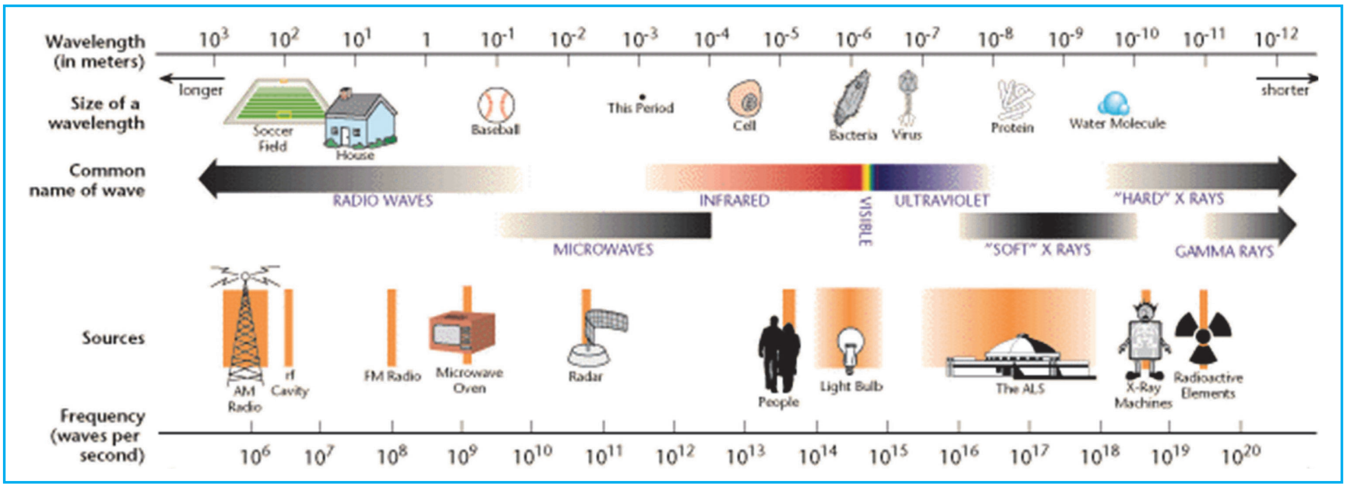

Figure 2.2: Electromagnetic Spectrum

Any analogue signal can be converted into a digital signal by appropriately sampling it. The sampling frequency must be at least twice the maximum frequency present in the signal in order to carry all the information contained in it (Nyquist Theory).

Random signal are the ones that are unpredictable and can be described only by statistical means. Noise is a typical random signal, described by its mean power and frequency distribution.

Behaviour of radio waves

The longer the wavelength, the further it goes.

The longer the wavelength, the better it travels through and around things.

The shorter the wavelength, the more data it can transport.

2.1.1 Analogue versus Digital Transmission

Table 2.1 summarizes the characteristics of analogue and digital networks.

Feature	Analogue Characteristics	Digital Characteristics
Signal	Continuously variable, in both amplitude and frequency	Discrete signal, represented as either changes in voltage or changes in light levels
Traffic measurement	Hz (for example, a telephone channel is 4KHz)	Bits per second (for example, a T-1 line carries 1.544Mbps, and an E-1 line transports 2.048Mbps)
Bandwidth	Low bandwidth (4KHz), which means low data transmission rates (up to 33.6Kbps) because of limited channel bandwidth	High bandwidth that can support high-speed data and emerging applications that involve video and multimedia
Network capacity	Low; one conversation per telephone channel	High; multiplexers enable multiple conversations to share a communications channel and hence to achieve greater transmission efficiencies
Network manageability	Poor; a lot of labor is needed for network maintenance and control because dumb analogue devices do not provide management information streams that allow the device to be remotely managed	Good; smart devices produce alerts, alarms, traffic statistics, and performance measurements, and technicians at a network control centre (NCC) or network operations centre (NOC) can remotely monitor and manage the various network elements
Power requirement	High because the signal contains a wide range of frequencies and amplitudes	Low because only two discrete signals—the one and the zero—need to be transmitted
Security	Poor; when you tap into an analogue circuit, you hear the voice stream in its native form, and it is difficult to detect an intrusion	Good; encryption can be used
Error rates	High; 10^{-5} bits (that is, 1 in 100,000 bits) is guaranteed to have an error	Low; with twisted-pair, 10^{-7} (that, is 1 in 10 million bits per second) will have an error, with satellite, 10^{-9} (that is, 1 in 1 billion per second) will have an error, and with fibre, 10^{-11} (that is only 1 in 10 trillion bits per second) will have an error

Table 2.1 summarizes the characteristics of analogue and digital networks.

2.1.2 Voice Sampling

Voice signal has a bandwidth of 4 kHz (300 Hz to 3300 Hz is transmitted on phone systems)

Nyquist sampling theorem: Sample at twice the highest signal frequency ⇒ Sample at 8 kHz ⇒ Sample every 125 μsec.

256 levels ⇒ 8 bits per sample × 8000 samples/sec = 64 Kbps

Natural Human Voice spans a frequency from 20Hz - 20KHz, however conventional telephone systems pass frequencies 400Hz - 3.5KHz. Therefore telephone conversations differ from face to face conversation.

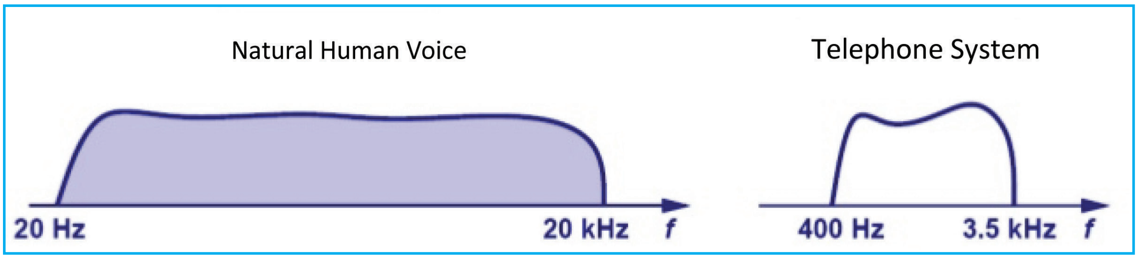

Figure 2.3: Voice communication

2.2 Technologies

2.2.1 Digitisation

Digitization is the representation of an object, image, sound, document or signal (usually an analogue signal) by generating a series of numbers that describe a discrete set of its points or samples. The result is called digital representation or, more specifically, a digital image, for the object, and digital form, for the signal. In modern practice, the digitized data is in the form of binary numbers, which facilitate computer processing and other operations, but strictly speaking, digitizing simply means the conversion of analogue source material into a numerical format; the decimal or any other number system can be used instead.

Digitization is of crucial importance to data processing, storage and transmission, because it “allows information of all kinds in all formats to be carried with the same efficiency and also intermingled. Unlike analogue data, which typically suffers some loss of quality each time it is copied or transmitted, digital data can, in theory, be propagated indefinitely with absolutely no degradation. This is why it is a favored way of preserving information for many organisations around the world.

Analogue signals are continuous electrical signals; digital signals are non-continuous. Analogue signals can be converted to digital signals by using analogue to digital converter (ADC). Digital signals can be converted to analogue signals by using a digital to analogue converter (DAC).

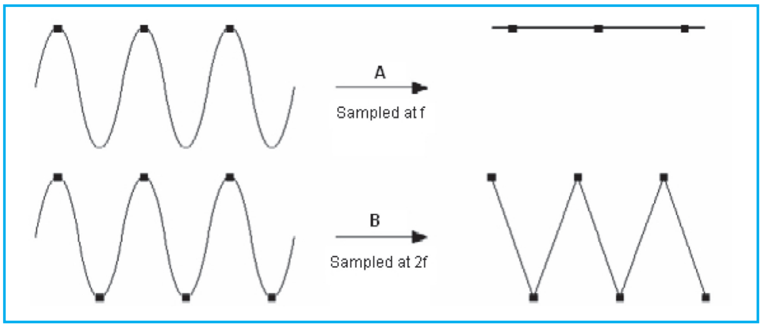

Figure 2.4: Digitising a signal

The sampling rate is defined as the number of samples acquired, per unit time, and is usually given in samples/sec or Hz. It is intuitively clear that at a higher sampling rate, the digital signal provides a better approximation to the analogue signal. It may furthermore be shown from theory that if the sampling rate is sufficiently high, the analogue signal can be reconstructed exactly from the samples, i.e. there is no loss of information in the process of sampling.

The sampling theorem (Nyquist Theory) states that this recovery of the analogue signal from its sampled version is possible, when the sampling rate is greater than twice the maximum frequency present in the signal. This provides the main criterion for selecting the sampling rate.

2.2.2 Multiplexing

Multiplexing is the sharing of a single communication channel among different users. The communication channel can be a copper wire, an optical fibre or the space between a transmitting and a receiving antenna. Different users can be distinguished by means of different frequencies, time slots, codes or regions of space.

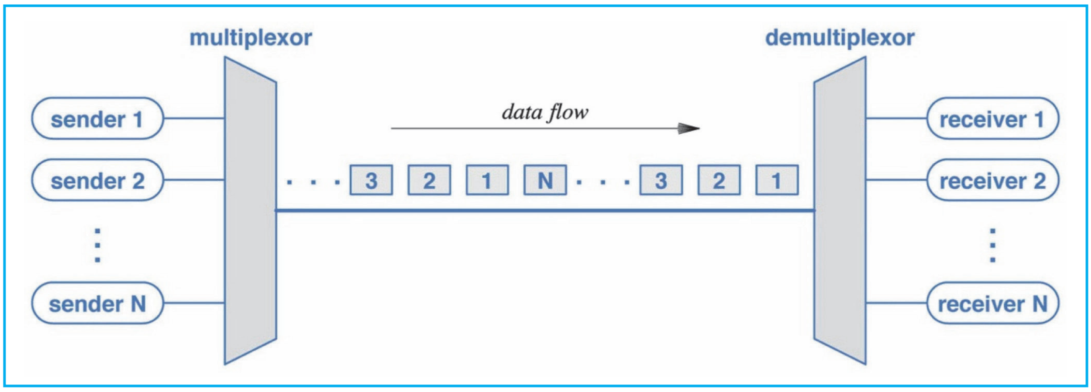

Figure 2.5: Digital Multiplexer

2.2.3 Duplex

Simplex:
One way only, example, TV Broadcasting.

Full-duplex:
In a full duplex system, both parties can communicate with each other simultaneously. An example of a full-duplex device is a telephone; the parties at both ends of a call can speak and be heard by the other party simultaneously. The earphone reproduces the speech of the remote party as the microphone transmits the speech of the local party, because there is a two-way communication channel between them, or more strictly speaking, because there are two communication paths/channels between them.

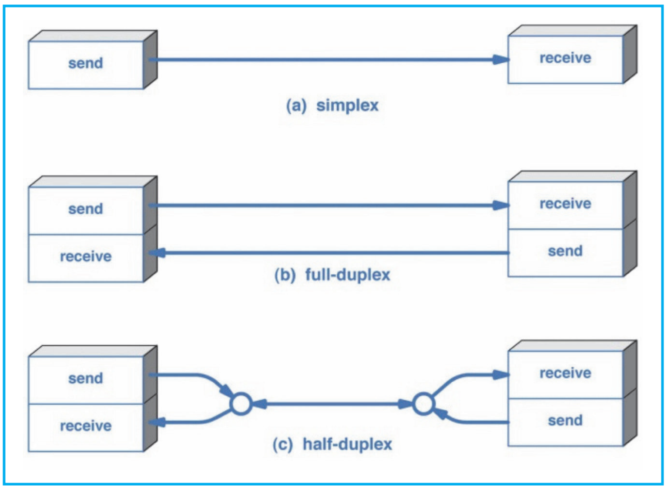

Figure 2.6: Types of Duplex

Half-duplex:
In a half-duplex system, there are still two clearly defined paths/channels, and each party can communicate with the other but not simultaneously; the communication is one direction at a time. An example of a half-duplex device is a walkie-talkie two-way radio that has a “push-to-talk” button; when the local user wants to speak to the remote person they push this button, which turns on the transmitter but turns off the receiver, so they cannot hear the remote person. To listen to the other person they release the button, which turns on the receiver but turns off the transmitter.

2.2.4 Switching

Circuit switching is a methodology of implementing a telecommunications network in which two network nodes establish a dedicated communications channel (circuit) through the network before the nodes may communicate. The circuit guarantees the full bandwidth of the channel and remains connected for the duration of the communication session. The circuit functions as if the nodes were physically connected as with an electrical circuit.

The defining example of a circuit-switched network is the early analogue telephone network. When a call is made from one telephone to another, switches within the telephone exchanges create a continuous wire circuit between the two telephones, for as long as the call lasts.

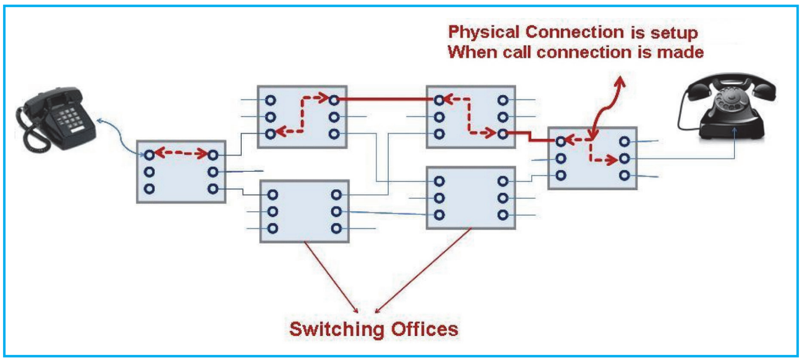

Figure 2.7: Public switched telephone network

The public switched telephone network (PSTN) is the aggregate of the world’s circuit-switched telephone networks that are operated by national, regional, or local telephony operators, providing infrastructure and services for public telecommunication. The PSTN consists of telephone lines, fibre optic cables, microwave transmission links, cellular networks, communications satellites, and undersea telephone cables, all interconnected by switching centers, thus allowing most telephones to communicate with each other. Originally a network of fixed-line analogue telephone systems, the PSTN is now almost entirely digital in its core network and includes mobile and other networks, as well as fixed telephones.

Message switching was the precursor of packet switching, where messages were routed in their entirety, one hop at a time. It was first built by Collins Radio Company, Newport Beach, California, during the period 1959–1963 for sale to large airlines, banks and railroads.

Message switching systems are nowadays mostly implemented over packet-switched or circuit-switched data networks. Each message is treated as a separate entity. Each message contains addressing information, and at each switch this information is read and the transfer path to the next switch is decided. Depending on network conditions, a conversation of several messages may not be transferred over the same path. Each message is stored (usually on hard drive due to RAM limitations) before being transmitted

to the next switch. Because of this it is also known as a ‘store-and-forward’ network. Email is a common application for message switching. A delay in delivering email is allowed, unlike real-time data transfer between two computers.

Packet switching is a digital networking communications method that groups all transmitted data into suitably sized blocks, called packets, which are transmitted via a medium that may be shared by multiple simultaneous communication sessions. Packet switching increases network efficiency, robustness and enables technological convergence of many applications operating on the same network.

Packets are composed of a header and payload. Information in the header is used by networking hardware to direct the packet to its destination where the payload is extracted and used by application software.

Packets are short (averaging a few hundred bytes) because networking devices handle short messages more efficiently.

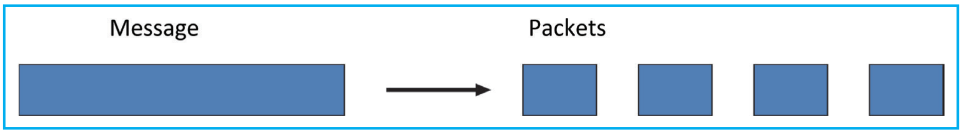

Figure 2.8: Packet Switching

Figure 2.9: Units of Telecommunication

2.2.5 Units of Telecommunication

1. Transmitter - takes the information and converts it into a signal.
2. Transmission Medium - also called the physical channel - carries the signal.
3. Receiver - Takes the signal and converts it into a usable information.

2.2.6 Basic Modulation

The digital sequence 1 0 1 0 is shown in Figure 2.10 is modulating a sinusoidal carrier in ASK (Amplitude Shifting Keying), FSK (Frequency Shifting Keying), PSK (Phase Shifting Keying) and QAM (Quadrature Amplitude Modulation). Quadrature modulation is another term used for binary phase modulation. There is a great number of modulation techniques derived from these basic schemes.

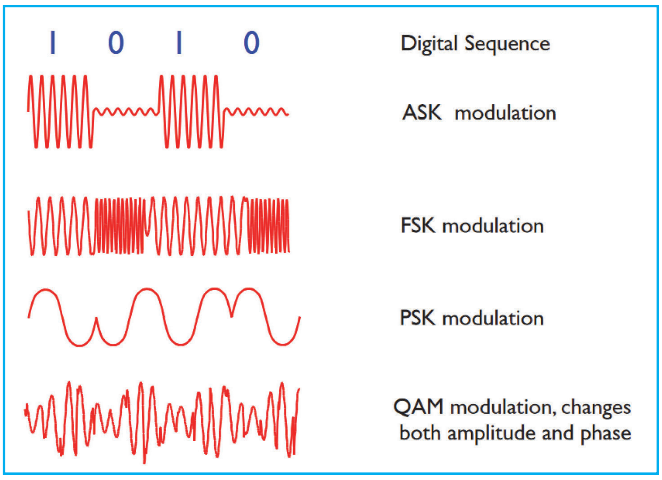

Figure 2.10: Types of Modulation

2.2.7 Pulse Code Modulation - PCM

The world’s telephone network as we know it today evolved as a mechanism for transporting voice conversations between telephone handsets. Until about 1970 this was achieved by carrying analogue signals over copper twisted pairs,with Frequency Division Multiplexing (FDM) used on long-haul routes to combine signals on coaxial cable. Such transmission

equipment was expensive in relation to the cost of a telephone exchange, so switching could be viewed as a means for rationing a scarce resource - transmission bandwidth.

In the early 1970s, digital transmission systems began to appear, utilising a method known as PCM which allowed analogue waveforms, such as the human voice, to be represented in binary form, and using this method it was possible to represent a standard 4 kHz analogue telephone signal as a 64 Kbps digital bit stream. Figure 2.11 shows the principles of PCM or Digitisation.

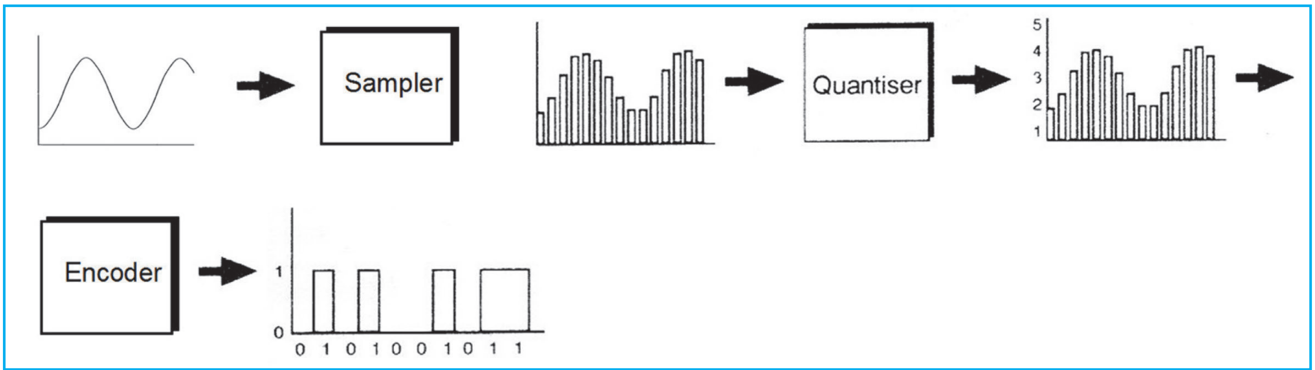

Figure 2.11: Pulse Code Modulation

2.2.8 Time Division Multiplexing - TDM

Engineers saw the potential to produce more cost effective transmission systems by combining several PCM channels and transmitting them down the same copper twisted pair as had previously been occupied by a single analogue signal. This phenomenon was termed ‘pair gain”. As the cost of digital electronics began to fall, major cost savings became possible through the use of these techniques.

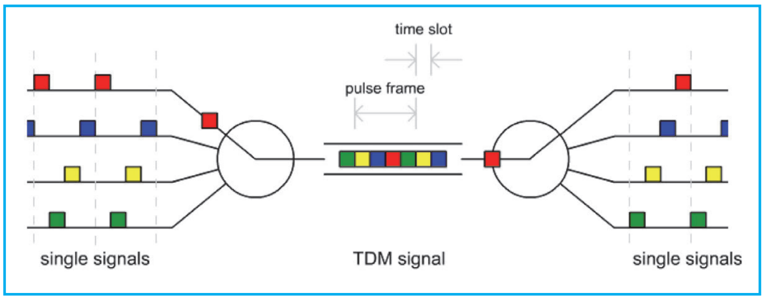

Figure 2.12: Time Division Multiplexing

The method used to combine multiple 64 Kbps channels into a single high speed bit stream is known as TDM, (see Figure 2.12). In simple terms, a byte from each incoming channel is transmitted in turn down the outgoing high speed channel. This process is sometimes referred to as ‘sequential byte interleaving”.

In Europe, and subsequently in many other parts of the world, a standard TDM scheme was adopted whereby thirty 64 Kbps channels were combined, together with two additional channels carrying control information, to produce a channel with a bit rate of 2.048 Mbps.

As demand for voice telephony increased, and levels of traffic in the network grew ever higher, it became clear that the standard 2 Mbps signal was not sufficient to cope with the traffic loads occurring in the trunk network. In order to avoid having to use excessively large numbers of 2 Mbps links, it was decided to create a further level of multiplexing. The standard adopted in Europe involved the combination of four

2 Mbps channels to produce a single 8 Mbps channel.

This level of multiplexing differed slightly from the previous in that the incoming signals were combined one bit at a time instead of one byte at a time i.e. bit interleaving was used as opposed to byte interleaving. As the need arose, further levels of multiplexing were added to the standard at 34 Mbps, 140 Mbps, and 565 Mbps to produce a full hierarchy of bit rates.

While the European transmission hierarchy was being developed, similar work was occurring in North America to develop their own hierarchy. While the same principles were used, a hierarchy evolved which differed slightly in that its lower bit rates were 1.5 Mbps, 6 Mbps, and 45 Mbps. These differences were to make interworking between the two hierarchies expensive to achieve.

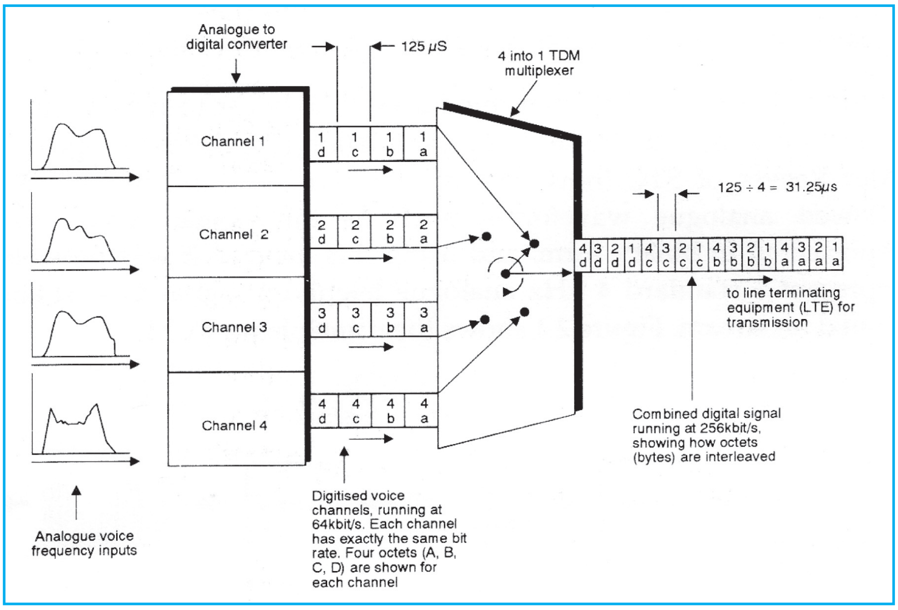

Figure 2.13: Time Division Multiplexing

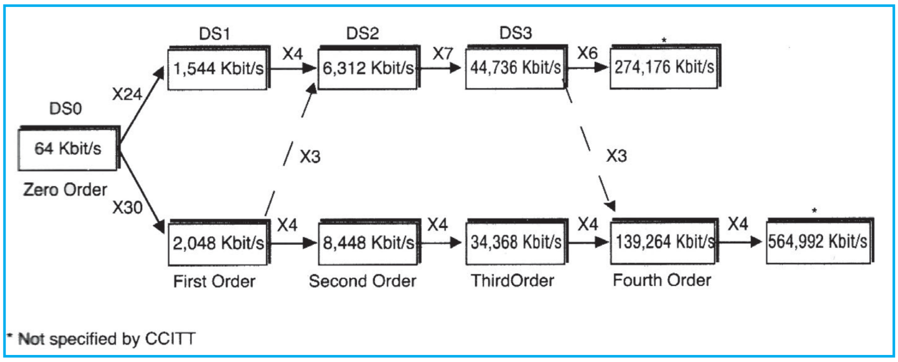

Figure 2.14: North American and European transmission hierarchies.

2.2.9 Frequency-Division Multiplexing - FDM

In telecommunications, FDM is a technique by which the total bandwidth available in a communication medium is divided into a series of non-overlapping frequency sub-bands, each of which is used to carry a separate signal. This allows a single transmission medium such as the radio spectrum, a cable or optical fibre to be shared by multiple independent signals. Another use is to carry separate serial bits or segments of a higher rate signal in parallel.

The most natural example of frequency-division multiplexing is radio and television broadcasting, in which multiple radio signals at different frequencies pass through the air at the same time. Another example is cable television, in which many television channels are carried simultaneously on a single cable. FDM is also used by telephone systems to transmit multiple telephone calls through high capacity trunk lines, communications satellites to transmit multiple channels of data on uplink and downlink radio beams, and

broadband DSL modems to transmit large amounts of computer data through twisted pair telephone lines, among many other uses.

An analogous technique called wavelength division multiplexing is used in fibre optic communication, in which multiple channels of data are transmitted over a single optical fibre using different wavelengths (frequencies) of light.

2.2.9.1 How It Works

The multiple separate information signals that are sent over an FDM system, such as the video signals of the television channels that are sent over a cable TV system, are called baseband signals. At the source end, for each frequency channel, an electronic oscillator generates a carrier signal, a steady oscillating waveform at a single frequency that serves to "carry" information. The carrier is much higher in frequency than the baseband signal. The carrier signal and the baseband signal are applied to a modulator circuit. The modulator alters some aspect of the carrier signal, such as its amplitude, frequency, or phase, with the baseband signal, "piggybacking" the data onto the carrier.

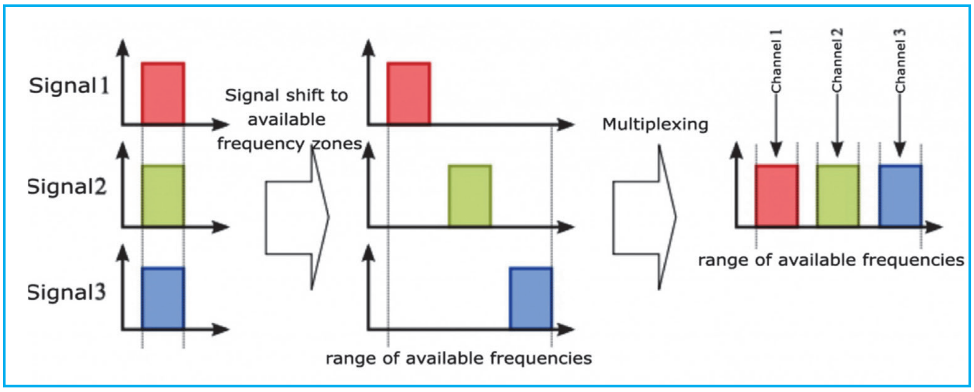

Figure 2.15: Frequency-division multiplexing

2.2.10 Asynchronous Transfer Mode

Asynchronous Transfer Mode (ATM) is, according to the ATM Forum, "a telecommunications concept defined by ANSI and ITU (formerly CCITT) standards for carriage of a complete range of user traffic, including voice, data, and video signals". ATM was developed to meet the needs of the Broadband Integrated Services Digital Network, as defined in the late 1980s, and designed to unify telecommunication and computer networks. It was designed for a network that must handle both traditional high-throughput data traffic (e.g., file transfers), and real-time, low-latency content such as voice and video. The reference model for ATM approximately maps to the three lowest layers of the ISO-OSI reference model: network layer, data link layer, and physical layer. ATM is a core protocol used over the SDH backbone of the PSTN and Integrated Services Digital Network (ISDN), but its use is declining in favour of all IP.

ATM provides functionality that is similar to both circuit switching and packet switching networks: ATM uses asynchronous time-division multiplexing, and encodes data into small, fixed-sized packets called cells. This differs from approaches such as the Internet Protocol or Ethernet that use variable sized packets

and frames. ATM uses a connection-oriented model in which a virtual circuit must be established between two endpoints before the actual data exchange begins. These virtual circuits may be “permanent”, i.e. dedicated connections that are usually preconfigured by the service provider, or “switched”, i.e. set up on a per-call basis using signalling and disconnected when the call is terminated.

ATM eventually became dominated by IP only technology (and Wireless or Mobile ATM never got any foothold).

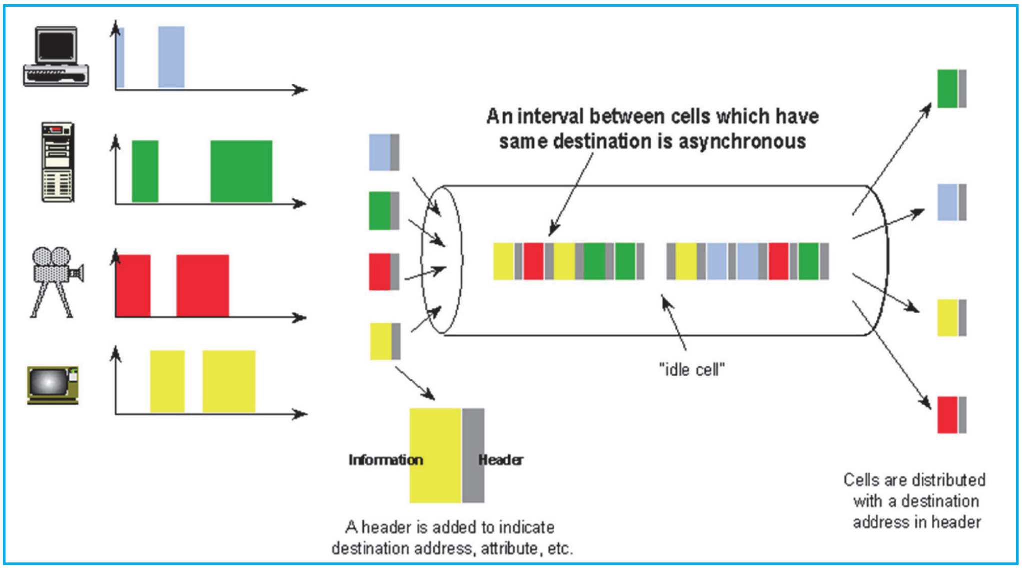

Figure 2.16: Asynchronous Transfer Mode