CHAPTER 1

# Digital communications overview

## 1.1 Electronic communications

### History, present requirements and future demands

Communication can be defined as the imparting or exchange of information [Hanks]. Telecommunication, which is more narrowly the topic of this book, refers to communication over a distance greater than would normally be possible without artificial aids. In the present day such aids are invariably electrical, electronic or optical and communication takes place by passing signals over wires, through optical fibres or through space using electromagnetic waves.

Modern living demands that we have access to a reliable, economical and efficient means of communication. We use communication systems, particularly the public switched telephone network (PSTN), to contact people all around the world. Telephony is an example of point-to-point communication and normally involves a two-way flow of information. Another type of communication, which (traditionally) involves only one-way information flow, is broadcast radio and television. In these systems information is transmitted from one location but is received at many locations using many independent receivers. This is an example of point-to-multipoint communication.

Communication systems are now very widely applied. Navigation systems, for example, pass signals between a transmitter and a receiver in order to determine the location of a vehicle, or to guide and control its movement. Signalling systems for tracked vehicles, such as trains, are also simple communication systems.

All early forms of communication system (e.g. smoke signals, semaphore, etc.) used digital traffic. The earliest form of electronic communications, telegraphy, was developed in the 1830s, Table 1.1. It was also digital in that the signals, transmitted over wires, were restricted to four types: dots and dashes, representing the Morse coded letters of the alphabet, letter spaces and word spaces. In the 1870s Alexander Graham Bell made analogue communications possible by inventing acoustic transducers to convert speech directly into (analogue) electrical signals.

**Table 1.1** *Important events in the history of electronic communications.*

| *Year* | *Event* | *Originator* | *Information* |
|---|---|---|---|
| 1837 | Line telegraphy perfected | Morse | Digital |
| 1875 | Telephone invented | Bell | Analogue |
| 1897 | Automatic exchange step by step switch | Strowger | |
| 1901 | Wireless telegraphy | Marconi | Digital |
| 1905 | Wireless telephony demonstrated | Fessenden | Analogue |
| 1907 | First regular radio broadcasts | USA | Analogue |
| 1918 | Superheterodyne radio receiver invented | Armstrong | Analogue |
| 1921 | First use of land based PMR | Detroit police | Analogue |
| 1928 | All electronic television demonstrated | Farnsworth | Analogue |
| 1928 | Telegraphy signal transmission theory | Nyquist | Digital |
| 1928 | Information transmission | Hartley | Digital |
| 1931 | Teletype | | Digital |
| 1933 | FM demonstrated | Armstrong | Analogue |
| 1934 | Radar demonstrated | Kuhnold | |
| 1937 | PCM proposed | Reeves | Digital |
| 1939 | Commercial TV broadcasting | BBC | Analogue |
| 1943 | Matched filtering proposed | North | Digital |
| 1945 | Geostationary satellite proposed | Clarke | |
| 1946 | ARQ systems developed | Duuren | Digital |
| 1948 | Mathematical theory of communications | Shannon | |
| 1955 | Terrestrial microwave relay | RCA | Analogue |
| 1960 | First laser demonstrated | Maiman | |
| 1962 | Satellite communications implemented | TELSTAR 1 | Analogue |
| 1963 | Geostationary satellite communications | SYNCOM II | Analogue |
| 1966 | Optical fibres proposed | Kao & Hockman | |
| 1966 | Packet switching | | Digital |
| 1970 | Medium scale data networks | ARPA/TYMNET | Digital |
| 1970 | LANs, WANs and MANs | | Digital |
| 1971 | The term ISDN coined | CCITT | Digital |
| 1974 | Internet concept | Cerf & Kahn | Digital |
| 1978 | Cellular radio | | Analogue |
| 1978 | Navstar GPS launched | Global | Digital |
| 1980 | OSI 7 layer reference model adopted | ISO | Digital |
| 1981 | HDTV demonstrated | NHK, Japan | Digital |
| 1985 | ISDN basic rate access in UK | BT | Digital |
| 1986 | SONET/SDH introduced | USA | Digital |
| 1991 | GSM cellular system | Europe | Digital |
| 1993 | PCN concept launched | Worldwide | Digital |
| 1994 | IS-95 CDMA specification | Qualcom | Digital |

This led quickly to the development of conventional telephony. Radio communications started around the turn of the century when Marconi patented the first wireless telegraphy system. This was quickly followed by the first demonstration of wireless (or radio) telephony and in 1918 Armstrong invented the superheterodyne radio receiver which is still an important component of much modern day radio receiving equipment. In the 1930s Reeves proposed pulse code modulation (PCM) which laid the foundation for nearly all present-day digital communication systems.

Table 1.1 shows some of the principal events in the development of electronic communications over the last century and a half. The second world war saw rapid, forced, developments in nearly all areas of engineering and technology. Electronics and communications benefited greatly and the new, but associated, discipline of radar became properly established.

In 1945 Arthur C. Clarke wrote his famous article proposing geostationary satellite communications and 1963 saw the launch of the first successful satellite of this type. In 1966 optical fibre communication was proposed by Kao and Hockman and, around the same time, public telegraph and telephone (PTT) operators introduced digital carrier systems.

The first, general purpose, large scale data networks (ARPANET and TYMNET) were developed around 1970, provoking serious commercial interest in packet switching (as an alternative to circuit switching).

The 1970s saw significant improvements in the performance of, and large increases in the volume of traffic carried by, telecommunications systems of all types. Optical fibre losses were dramatically reduced and the capacity of satellites dramatically increased. In the 1980s first analogue, and then digital, cellular radio became an important part of the PSTN. Micro-cellular and personal communications using both terrestrial and satellite based radio technology are now being developed. It seems likely that before long wideband personal communications systems providing voice, data and perhaps even video services, will become possible. Video delivery will require a broadband rather than a narrow (speech) bandwidth connection, Table 1.2.

Increasing demand for traditional services (principally analogue voice communications) has been an important factor in the development of telecommunications technologies. Such developments, combined with more general advances in electronics and computing, have made possible the provision of entirely new (mainly digitally based) communications services. This in turn has stimulated demand still further. Figure 1.1 shows the past and predicted future growth of telecommunications traffic and Figure 1.2 shows the proliferation of services which have been, or are likely to be, offered over the same period.

In telecommunications there are various standards bodies which ensure interoperability of equipment. The International Telecommunications Union (ITU) is an important international communications standards body which only has the power to make recommendations for specifications. Within the ITU are the PTTs (post, telephone and telegraph organisations) from individual nations, e.g. British Telecom and Deutsche Bundespost. In Europe there was, until recently, the Confederation of European PTTs (CEPT), responsible for overseeing the actual implementation of technical standards. CEPT has now been replaced with the European Telecommunications Standards Institute (ETSI) [Temple].

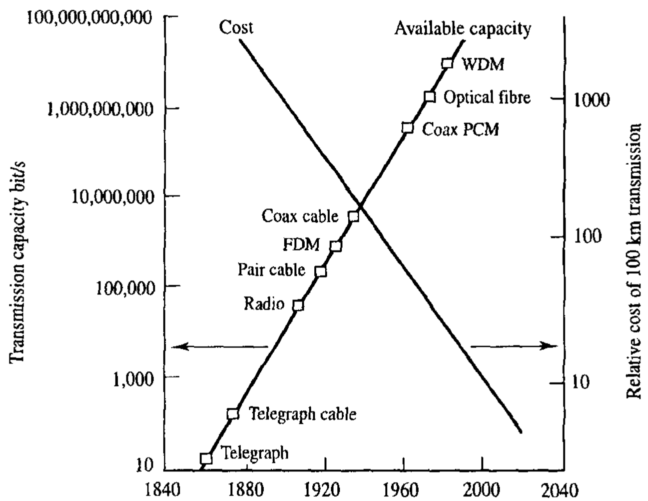


**Figure 1.1** *Past and predicted growth of telecommunications traffic (source: Technical demographics, 1995, reproduced with permission of the IEE).*

## 1.2 Sources and sinks of information

Sources of information can be either natural or man-made. An example of the former might be the air temperature at a given location. An example of the latter might be a set of company accounts. (A third example, speech, falls, in some sense, into both categories.) Digital communication systems represent information, irrespective of its type or origin, by a discrete set of allowed symbols. It is this alphabet of symbols and the device or mechanism which selects them for transmission that, in this context, is usually regarded as the information source. The amount of information conveyed by each symbol, as it is selected and transmitted, is closely related to its selection probability. Symbols likely to be selected more often convey less information than those which are less likely to be selected. Information content (measured in bits) is thus related to symbol rarity.

Sinks of information are, ultimately, people although various types of information storage and display devices (computer disks, magnetic tapes, loudspeakers, VDUs etc.) are usually involved as a penultimate destination.

Transmitters are the devices that impress source information onto an electrical wave (or carrier) appropriate to a particular transmission medium (e.g. optical fibre, cable, free space). Receivers are the devices which extract information from such carriers. They often also reproduce this information in the same form as it was originally generated (e.g. as speech).

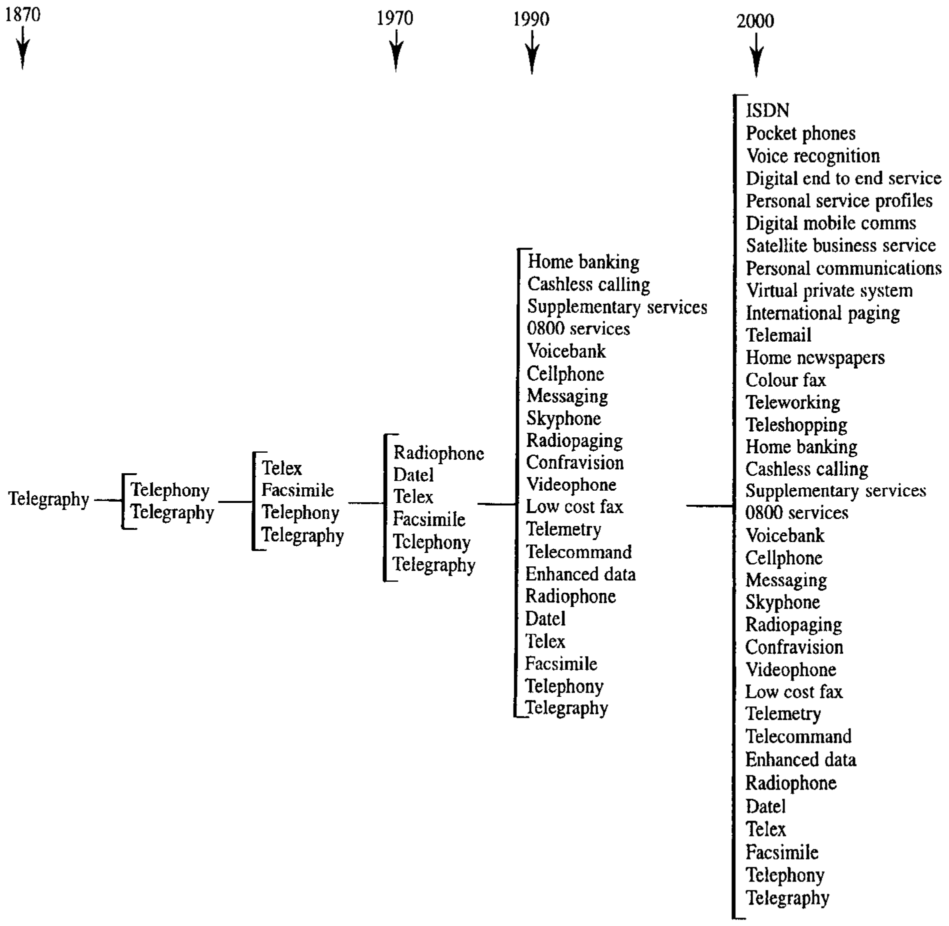


**Figure 1.2** *Service proliferation in telecommunications (source: Earnshaw, 1991, reproduced with permission of Peter Peregrinus).*

## 1.3 Digital communications equipment

An important objective in the design of a communications system is often to minimise equipment cost, complexity and power consumption whilst also minimising the bandwidth occupied by the signal and/or transmission time. (Bandwidth is a measure of how rapidly the information-bearing part of a signal can change and is therefore an important parameter for communication system design. Table 1.2 compares the nominal bandwidth of three common types of information signal.) Efficient use of bandwidth and transmission time ensures that as many subscribers as possible can be accommodated within the constraints of these limited, and therefore valuable, resources.

**Table 1.2** *Comparison of nominal bandwidths for several information signals.*

| *Information signal* | *Bandwidth* |
|---|---|
| Speech telephony | 4 kHz |
| High quality sound broadcast | 15 kHz |
| TV broadcast (video) | 6 MHz |

The component parts of a hypothetical digital communications transceiver (transmitter/receiver) are shown in Figure 1.3. Much of the rest of this book is concerned with the operating principles, performance and limitations of a communication system formed by a transmitter/receiver pair linked by a communications channel. Here, however, we give a qualitative overview of such a system, incorporating a brief account of what each block in Figure 1.3 does and why it might be required. (The transceiver in this figure has been chosen to include all the elements commonly encountered in digital communications systems. Not all transceivers will employ all of these elements of course.)

### 1.3.1 CODECs

At its simplest a transceiver CODEC (coder/decoder) consists of an analogue to digital converter (ADC) in the transmitter, which converts a continuous, analogue, signal into a sequence of code words represented by binary voltage pulses, and a digital to analogue converter (DAC) in the receiver, which converts these voltage pulses back into a continuous, analogue, signal.

The ADC consists of a sampling circuit, a quantiser and a pulse code modulator (Figure 1.3). The sampling circuit provides discrete voltage samples taken, at regular intervals of time, from the analogue signal. The quantiser approximates these voltages by the nearest level from an allowed set of voltage levels. (It is the quantisation process

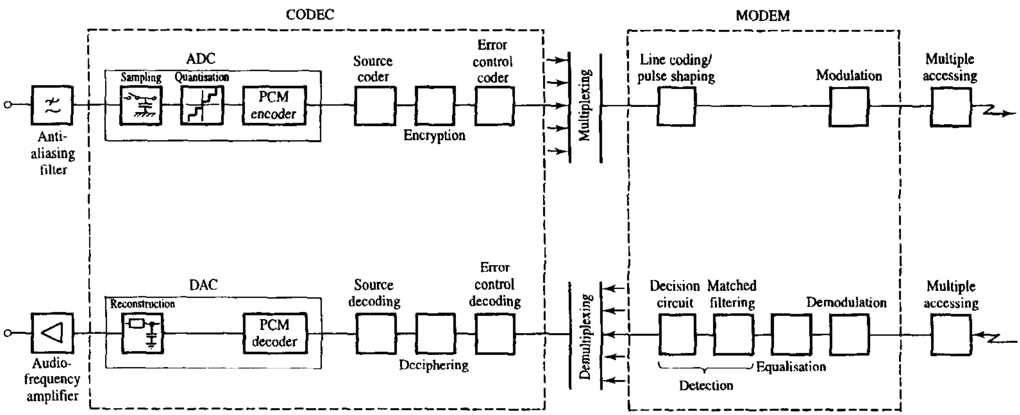


**Figure 1.3** *Hypothetical digital communications transceiver.*

which converts the analogue signal to a digital one.) The PCM encoder converts each quantised level to a binary code word, digital ones and zeros each being represented by one of two voltages. An anti-aliasing filter is sometimes included prior to sampling in order to reduce distortion which can occur as a result of the sampling process.

In the receiver's DAC received binary voltage pulses are converted to quantised voltage levels by a PCM decoder which is then smoothed by a low-pass filter to reconstruct (at least a good approximation to) the original, analogue, signal.

Digitisation of analogue signals usually increases the signal's transmission bandwidth but it permits reception at a lower signal-to-noise ratio than would otherwise be the case. This is an example of how one resource (bandwidth) can be traded off against another resource (transmitter power).

CODECs make widespread use of sophisticated digital signal processing techniques to encode efficiently the signal prior to transmission and also to decode the received signals when they are corrupted by noise, distortion and interference. This increases transceiver complexity, but allows higher fidelity, repeatable, almost error-free transmission to be achieved.

### 1.3.2 Source, security and error control coding

In addition to PCM encoding and decoding a CODEC may have up to three additional functions. Firstly (in the transmitter) it may reduce the number of binary digits (called bits, or sometimes binits) required to convey a given message. This is source coding and can be thought of as effectively removing redundant (i.e. unrequired or surplus) digits. Secondly it may encrypt the source coded digits using a cipher for security. This can yield both privacy (which assures the sender that only those entitled to the information being transmitted can receive it) and authentication (which assures the receiver that the sender is who he/she claims to be). Finally the CODEC may add extra digits to the (possibly source coded and/or encrypted) PCM signal which can be used at the receiver to detect, and possibly correct, errors made during symbol detection. This is error control coding and has the effect of incorporating binary digits at the transmitter which, from an information point of view, are redundant.

In some ways error control coding, which adds redundancy to the bit stream, is the opposite of source coding, which removes redundancy. Both processes may be employed in the same system, however, since the type of redundancy which occurs naturally in the information being transmitted is not necessarily the type best suited to detecting and correcting errors at the receiver.

The source, security and error control, decoding operations in the receiver, Figure 1.3, are the inverse of those in the transmitter.

### 1.3.3 Multiplexers

In digital communications, multiplexing, to accommodate several simultaneous transmissions usually means, more specifically, time division multiplexing (TDM). Time division multiplexers interleave either PCM code words, or individual PCM binary digits, to allow more than one information link to share the same physical transmission medium (e.g. cable, optical fibre, radio frequency channel). If communication is to occur in real time this implies that the bit rate of the multiplexed signal is at least $N$ times that of each of the $N$ tributary PCM signals and this in turn implies an increased bandwidth requirement.

Demultiplexers split the received composite bit stream back into its component PCM signals.

### 1.3.4 MODEMs

MODEMs (modulators/demodulators) condition binary pulse streams so that the information they contain can be transmitted over a given physical medium, at a given rate, with an acceptable degree of distortion, in a specified or allocated frequency band. The modulator in the transmitter may change the voltage levels representing individual, or groups of, binary digits. Typically the modulator also shapes, or otherwise filters, the resulting pulses to restrict their bandwidth, and shifts the entire transmission to a convenient, allowed, frequency band. The input to a modulator is thus a baseband digital signal whilst the output is often a bandpass waveform.

The demodulator, in a receiver, reconverts the received waveform into a baseband signal. Equalisation corrects (as far as possible) signal distortion which may have occurred during transmission. Detection converts the demodulated baseband signal into a binary symbol stream. The matched filter, shown as one component of the detector in Figure 1.3, represents one type of signal processing which can be employed, prior to the final digital decision process, in order to improve error performance.

### 1.3.5 Multiple accessing

Multiple accessing refers to those techniques, and/or rules, which allow more than one transceiver pair to share a common transmission medium (e.g. one optical fibre, one satellite transponder or one piece of coaxial cable). Several different types of multiple accessing are currently in use, each type having its own advantages and disadvantages. The multiple accessing problem is essentially one of efficient and (in some sense) equitable sharing of the limited resource represented by the transmission medium.

## 1.4 Radio receivers

Many radio receivers (both digital and analogue) incorporate superheterodyning as part of their demodulation process. In these receivers (Figure 1.4) the incoming radio frequency (RF) signal, with carrier frequency $f_{RF}$, is mixed (i.e. multiplied) with the signal from a local oscillator (LO) of frequency $f_{LO}$. The sum $(f_{RF} + f_{LO})$ and difference $(f_{RF} - f_{LO})$ frequency products which appear at the mixer output are then filtered to select only the latter which is called the intermediate frequency or IF. The LO frequency is, therefore, always altered or tuned to ensure that the receiver operates with a fixed value of IF (i.e. $f_{RF} - f_{LO}$) irrespective of which RF channel is being received. This allows a considerable effort to be invested in the design of the receiver beyond this point, consisting typically of high gain (fixed frequency) IF amplifiers and high selectivity filters followed by an appropriate IF signal demodulator and/or detector. The superheterodyne receiver can be made more sophisticated by using double frequency conversion in which there are two mixing stages. This enables higher gain and greater selectivity to be achieved in order to increase rejection of unwanted, interfering, signals.

The principal problem with the superheterodyne design is that the receiver is equally sensitive to radio frequency bands centred on $f_{LO} + f_{IF}$ (which is the wanted band) and $f_{LO} - f_{IF}$ (which is an unwanted band). The unwanted 'image' band of frequencies, separated from the wanted RF band by twice the IF frequency, represents a potentially serious source of RF interference and additional noise. A tunable image rejection filter (needing only modest selectivity) can be placed before the mixer in the RF amplifier of Figure 1.4 to attenuate or remove this unwanted band of frequencies.

## 1.5 Signal transmission

The communications path from transmitter to receiver may use lines or free space. Examples of the former are wire pairs, coaxial cables and optical fibres. The most important use of the latter is radio, although in some situations infrared and optical free space links are also possible (e.g. remote controls for TV, video and hi-fi equipment and also some security systems). Whatever the transmission medium, it is at this point that much of the attenuation, distortion, interference and noise is encountered. Attenuation can be compensated for by introducing amplifiers or signal repeaters at intermediate points along the multiple hop link, Figure 1.5. Distortion may be compensated by equalisers and interference and noise can be minimised by using appropriate predetection signal processing (e.g. matched filters). The nature and severity of transmission medium effects is one of the major influences on the design of transmitters, receivers and repeaters.

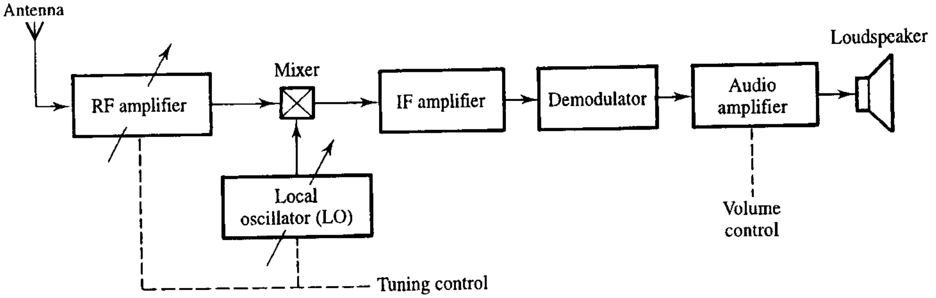


**Figure 1.4** *Superheterodyne receiver.*

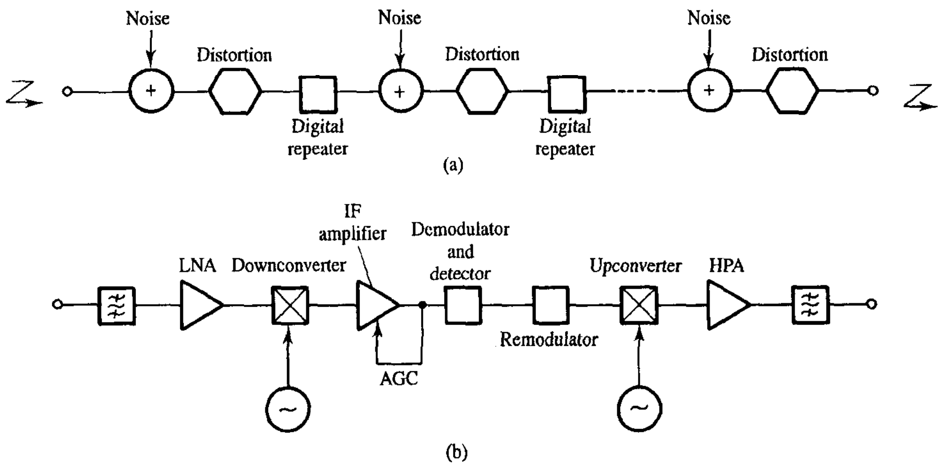


**Figure 1.5** *(a) Digital communications (multi-hop) channel; (b) digital repeater (as typically used in terrestrial microwave relay applications).*

## 1.5.1 Line transmission

The essential advantages of line transmission are:

1. Path loss is usually modest.
2. Signal energy is essentially confined and interference between different systems is seldom severe and often negligible.

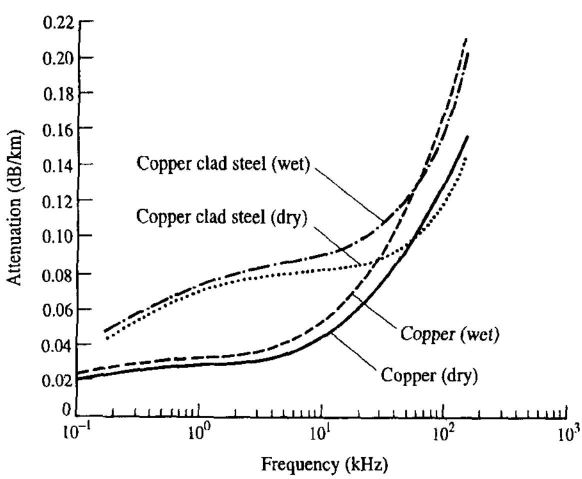


**Figure 1.6(a)** *Typical attenuation/frequency characteristics for aerial open wire pair lines.*

3. Path characteristics (e.g. attenuation and distortion) are usually stable and relatively easy to compensate for.
4. Capacity is unlimited in that bandwidth can always be reused by laying another line.

The disadvantages of line transmission are:

1. Laying cables in the ground or constructing overhead lines is generally expensive.
2. Extensive wayleaves and planning permission may be needed for underground cables and overhead wires.
3. Broadcasting requires a physical connection to a complex network for each subscriber.
4. Mobile communications services cannot be provided.
5. Networks cannot easily be added to, subtracted from, or otherwise reconfigured.

The degree to which a signal is attenuated by a transmission line depends on the material from which the line is made, its physical construction, and the signal's frequency. Figures 1.6(a) to (e) show some typical attenuation/frequency characteristics for the most common types of line. Open wire has particularly low loss but it is expensive to maintain and susceptible to interference. Loaded cable, Figure 1.6(c) and Table 1.3, is only effective for speech bandwidth signals. Twisted pairs, as used underground, have higher installation cost but lower maintenance costs. (Low loss, circular, waveguides can also be used as a transmission medium but advances in optical fibre technology have, at least for the present, made this technology essentially redundant.) Optical fibre cables have an enormous information carrying capacity with typical bandwidth-distance products of 0.5 GHz-km.

**Table 1.3** *Nominal properties of selected transmission lines.*

| | *Frequency range* | *Typical attenuation* | *Typical delay* | *Repeater spacing* |
|---|---|---|---|---|
| Open wire (overhead line) | 0 – 160 kHz | 0.03 dB/km @ 1 kHz | 3.5 $\mu$s/km | 40 km |
| Twisted pairs (multi-pair cables) | 0 – 1 MHz | 0.7 dB/km @ 1 kHz | 5 $\mu$s/km | 2 km |
| Twisted pairs (with $L$ loading) | 0 – 3.5 kHz | 0.2 dB/km @ 1 kHz | 50 $\mu$s/km | 2 km ($L$ spacing) |
| Coaxial cables | 0 – 500 MHz | 7 dB/km @ 10 MHz | 4 $\mu$s/km | 1 – 9 km |
| Optical fibres | 1610 – 810 nm | 0.2 to 0.5 dB/km | 5 $\mu$s/km | 40 km |

Table 1.3 summarises the nominal frequency range of each type of line, their typical attenuations and transmission delays, and typical repeater spacings. The useful bandwidths of the lines, which determine the maximum information transmission rate they can carry, are often, but not always, determined by their attenuation characteristics. Twisted wire pairs, for example, are normally limited to (line coded PCM) data rates of 2 Mbit/s. Coaxial cables, Figure 1.6(d), routinely carry 140 Mbit/s PCM signals but can handle symbol rates several times greater. Optical fibres have very large bandwidth

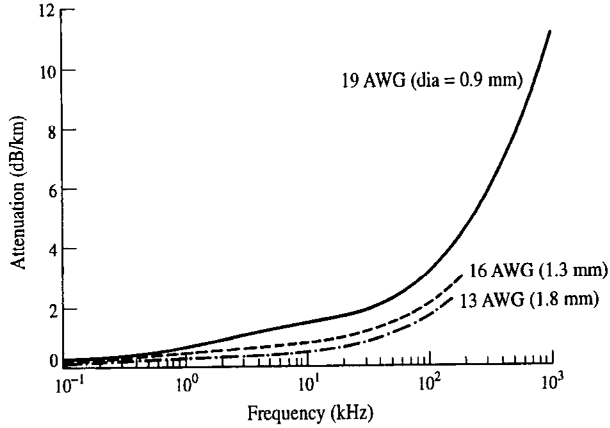


**Figure 1.6(b)** *Typical characteristics for twisted pair cable transmission lines.*

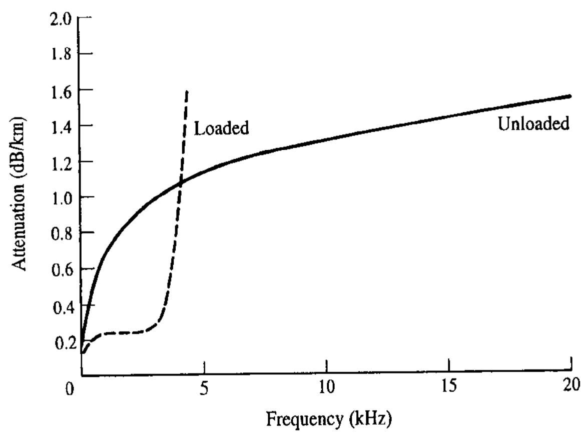


**Figure 1.6(c)** *Comparison between inductively loaded and unloaded twisted wire pairs.*

handle symbol rates several times greater. Optical fibres have very large bandwidth potential but may be limited to a fraction of this by factors such as the spectral characteristics of optical sources and dispersion effects. Nevertheless, optical fibre PCM bit rates of Gbit/s are possible.

### 1.5.2 Radio transmission

The advantages of radio transmission are:

1. It is relatively cheap and quick to implement.

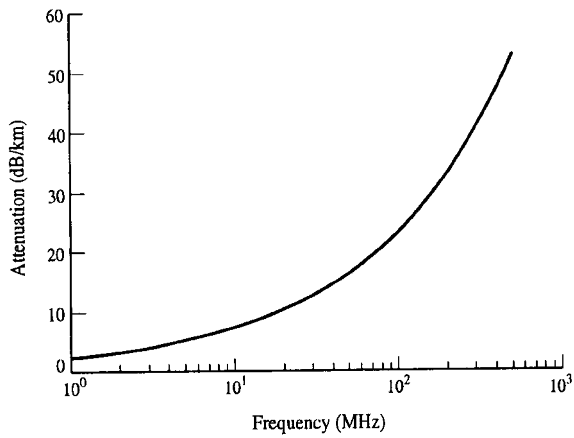


**Figure 1.6(d)** *Typical attenuation/frequency characteristic for coaxial cable.*

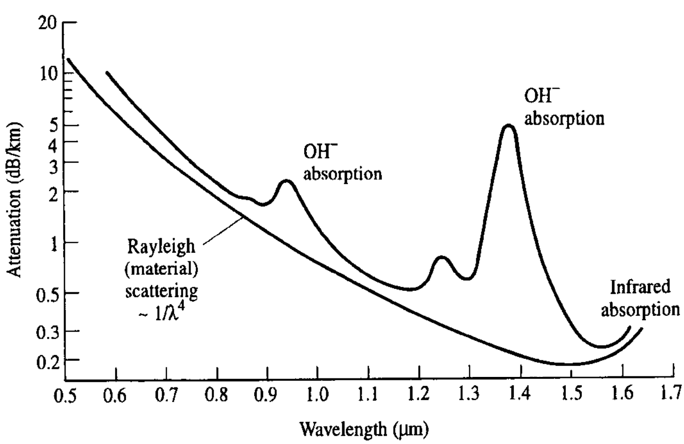


**Figure 1.6(e)** *Typical attenuation/wavelength characteristics for optical fibres (source: Young, 1994, reproduced with permission of Prentice Hall).*

2. Wayleaves and planning permission are often only needed for the erection of towers to support repeaters and terminal stations.
3. It has an inherent broadcast potential.
4. It has an inherent mobile communications potential.
5. Communications networks can be quickly reconfigured and extra terminals or nodes easily introduced or removed.

The principal disadvantages of radio are:

1. Path loss is generally large due to the tendency of the transmitted signal energy to spread out, most of this energy effectively missing the receive antenna.

2. The spreading of signal energy makes interference between different systems a potentially serious problem.
3. Capacity in a given locality is limited since bandwidth cannot be reused easily.
4. Path characteristics (i.e. attenuation and distortion) tend to vary with time, often in an unpredictable way, making equalisation more difficult and limiting reliability and availability.
5. The time varying nature of the channel can result in anomalous propagation of signals to locations well outside their normal range. This may cause unexpected interference between widely spaced systems.
6. Points 2 and 5 mean that frequency coordination is generally required when planning radio systems. Such coordination is difficult to achieve comprehensively and is expensive.

**Table 1.4** *Frequency bands commonly used for radio communication.*

| *Band* | *Frequency* | *Wavelength* | *Propagation mechanism* | *Fading process* | *Noise process* | *Range* | *Applications* |
|---|---|---|---|---|---|---|---|
| ELF | 30 – 300 Hz | $10^4 - 10^3$ km | Waveguide modes | Diurnal variations due to D-layer | Man-made and atmospherics (lightning discharges) | Worldwide | Submarine |
| VF | 300 – 3000 Hz | 1000 – 100 km | | | | | |
| VLF | 3 – 30 kHz | 100 – 10 km | Surface waves | None | | | Standards/navigation |
| LF | 30 – 300 kHz | 10 – 1 km | | | | 1000s km | Maritime mobile, LW b'cast |
| MF | 300 – 3000 kHz | 1000 – 100 m | Sky waves | Surface/sky wave intf. | | 100s km | MW broadcast |
| HF | 3 – 30 MHz | 100 – 10 m | | Complex ionospherics | | 4000 km/hop | Amateur |
| VHF | 30 – 300 MHz | 10 – 1 m | Line of sight | None | Galactic (synchrotron radiation) | Line of sight | FM broadcast |
| UHF | 300 – 3000 MHz | 100 – 10 cm | | Ray bending and multipath | Cosmic background | | TV broadcast, mobile, LOS |
| SHF | 3 – 30 GHz | 10 – 1 cm | | | | | Microwave LOS, satellite |
| EHF | 30 – 300 GHz | 10 – 1 mm | | Rain attenuation | Thermal noise from ground & atmosphere | Short | |

The appropriate radio-propagation model for a communication system, the dominant fading and noise processes, and typical system range, all depend on frequency. Table 1.4 shows the electromagnetic spectrum used for radio transmissions and summarises these models and processes. At the lowest frequencies propagation is best modelled by oscillating electromagnetic modes which exist in the cavity between the concentric conducting spheres formed by the earth and its ionosphere. From a few kilohertz up to a few hundred kilohertz vertically polarised radio energy will propagate (by diffraction) around the curved surface of the earth for thousands of kilometers. This is called surface wave propagation and is the mechanism by which long wave radio broadcasts are received.

At slightly higher frequencies in the medium frequency (MF) band some radio energy propagates as a surface wave and some is reflected from the conducting ionosphere as a sky wave. The relative path lengths and phasing of these two signals may result in destructive interference causing fading of the received signal which will vary in severity as the relative strengths and phases of the sky wave and surface wave change. The exact condition of the ionosphere may be critical in this respect, making the quality of signal reception vary, for example, with the time of day or night.

In the high frequency (HF) band the sky wave is usually dominant and ranges of thousands of kilometers are possible, sometimes involving multiple reflections between the ionosphere and ground. At very high frequencies (VHF), and above, signals propagate essentially along line-of-sight paths although reflection, refraction and, at the lower frequencies, diffraction can play an important role in the overall characteristics of the channel. At ultra high frequencies (UHF) currently used both for TV transmissions and cellular radio communications, multipath (i.e. multiple path) propagation caused by reflections from, and diffraction around, buildings and other obstacles in urban areas is the principal cause of signal fading. In the super high frequency (SHF) band (usually called the microwave or centimetric wave band) applications tend to be point-to-point (or fixed-point) communications and the first order fading problem is often due to rain induced attenuation. Extra high frequencies (EHF) and higher are not yet widely used for communication systems, partly due to the significant gaseous background attenuation and large fades which occur in rain. As the electromagnetic spectrum becomes more congested, however, and as the demand for communications becomes yet greater (in terms of both traffic volume and service sophistication) the use of these higher frequency bands will almost certainly become both necessary and economic.

## 1.6 Switching and networks

Many modern communication systems are concerned exclusively with data traffic. One example is the Internet, over which users can transmit e-mail messages or browse distant information sources and transfer large data and text files. The data networks themselves, and the configuration of computer terminals on a user site, can be organised in many different ways using ring, star or bus connections. In order to ensure interoperability, standards have developed for these topologies and also for the signalling and switching protocols which control the assembly and routing of traffic.

The seven layer ISO model is used throughout data communications networks as the standard hierarchical structure for organising data traffic. The data itself is usually sent as fixed length packets with associated overhead bits which provide addresses, timing or ordering information, and assist in error detection. The physical data interfaces follow evolved standards, e.g. X.25, IEEE 802, which develop progressively to higher data rates as the new high speed (wideband optical) transmission systems are introduced.

With packet data traffic there are inevitable delays while the packets are queued for access to the transmission system. These queues are not serious problems in simple mail networks but, if attempting to transmit speech or video traffic in real time, queue delays, and lost packets due to queue overflow in finite length buffers, can seriously degrade the operation of the communications link.

## 1.7 Advantages of digital communications

Digital communication systems usually represent an increase in complexity over the equivalent analogue systems. We therefore list here some of the reasons why digital communication has become the preferred option for most new systems and, in many instances, has replaced existing analogue systems.

1. Increased demand for data transmission.
2. Increased scale of integration, sophistication and reliability of digital electronics for signal processing, combined with decreased cost.
3. Facility to source code for data compression.
4. Possibility of channel coding (line, and error control, coding) to minimise the effects of noise and interference.
5. Ease with which bandwidth, power and time can be traded off in order to optimise the use of these limited resources.
6. Standardisation of signals, irrespective of their type, origin, or the services they support, leading to an integrated services digital network (ISDN).

The increase in demand, for voice and data connections, is the principal driving force behind the growth in telecommunications. The traffic, in the backbone network, expressed as equivalent voice circuits, is shown in Figure 1.7. This figure does not merely reflect the explosive growth in mobile communications for the final customer connections but shows world capacity for transmission.

## 1.8 Summary

The history of electronic communications over the last century and a half has demonstrated an essentially exponential growth in traffic and a continuously increasing demand for greater access to ever more sophisticated services. This trend shows no sign, at present, of changing.

Most modern telecommunications systems are digital and use some form of PCM irrespective of the origin of the information they convey. PCM signals are often coded themselves to improve system performance and/or provide security. Many PCM signals can be combined as a single (time division) multiplex to allow their simultaneous transmission over a single physical medium. Line coding and/or modulation can then be used to match the characteristics of the resulting multiplex to the transmission line or radio channel being used. Multiple accessing techniques allow many transceiver pairs to share a given transmission resource (e.g. cable, fibre, satellite transponder). Switching allows telecommunications networks to be designed which, at reasonable cost, can emulate a fully interconnected set of transceivers.

It is the purpose of this book to describe the operating principles and performance of modern digital communications systems. The description is presented at a systems, rather than a circuit, level and, in view of this, Part One of the book (Chapters 2 to 4) reviews some pertinent mathematical models and properties of signals, noise and

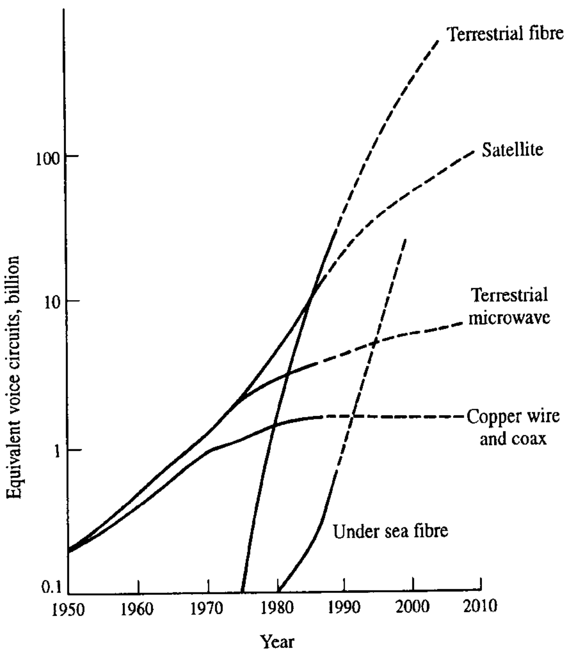


**Figure 1.7** *Growth in world transmission capacity (source: Cochrane, 1990, reproduced with permission of British Telecommunications plc.).*

systems. Part Two (Chapters 5 to 13) describes the analogue to digital conversion process, coding, and modulation techniques used to ensure adequate performance of a wide range of digital communications systems (Chapters 5 to 11); Chapter 12 is concerned with physical aspects of noise and the prediction of CNR at the end of a single or multi-hop transmission link; Chapter 13 discusses the computer simulation of communications systems. Part Three (Chapters 14 to 16) discusses modern digital telephony, terrestrial and satellite microwave systems, mobile cellular radio and video coding systems. Part Four (Chapters 17 to 19) describes switching and telecommunications networks including queuing theory and packet data transmission. It also includes a discussion of the current plesiochronous digital telephone network and the evolving synchronous digital hierarchy (SDH) for both telephony and data traffic.

# Part One

# Signals and systems theory

Signals and systems theory is the body of knowledge related to the definition and description of signals, and the behaviour of systems. In electrical engineering the study of signals is central to telecommunications, whilst the study of systems is probably more closely identified with control. It is obvious, however, that control engineers must be concerned with the signals which form the inputs and outputs of their systems, and conversely, communications engineers must be concerned with the systems which transmit, receive and otherwise process their signals. Nevertheless the closeness of the relationship between signal theory and communications means that the material presented, in Part 1 is biased in favour of signals.

Chapter 2 presents the principal mathematical tools (Fourier series and Fourier transforms) normally used to describe, analyse, and synthesise, waveforms and transient signals. A unifying theme, here, is that of determinism, i.e. the waveforms and signals addressed all allow descriptions which permit their values to be determined, precisely, at any point in time. The choice of Fourier analysis as a technique for splitting complicated signals into their simpler (sinusoidal) component parts leads to the important concepts of spectrum and bandwidth. Much of the communication engineer's effort is directed at conserving spectrum and utilising available signal bandwidth efficiently.

Chapter 2 also introduces ideas of signal orthogonality and correlation which relate to common sense notions of similarity. These concepts are important in communications since only signals which are in some way dissimilar can be assigned different meanings. In digital communications, especially, it is usually a requirement to generate signals which are easily distinguishable.

Chapter 3 deals with random signals (i.e. those which are not deterministic and are thus excluded from Chapter 2). Random signals are important, partly because information cannot be communicated by deterministic signals, and partly because unwanted random signals (constituting noise) always exist in a communications receiver. Such noise has the potential to modify, or obscure, wanted, information bearing signals. Due to their unpredictable nature random signals must be described in terms of their statistical properties. Chapter 3 therefore reviews probability theory and defines the mean, variance, covariance and other statistics, which are used to summarise the behaviour of random signals and noise. The similarity of a signal with a time shifted version of itself determines how rapidly the signal can change with time and provides information about the signal's spectrum. Chapter 3 makes the precise connection between self similarity (or autocorrelation) functions and the Fourier based spectral descriptions presented in Chapter 2.

Chapter 4 is concerned with systems, and in particular the effect that linear systems have on the spectral and autocorrelation properties of signals. The importance, and defining characteristics, of linear systems are discussed and use is made of the Fourier transform to link their equivalent time domain (impulse response) and frequency domain (frequency response) descriptions. The ways in which impulse and frequency responses are used to predict the effect of a system on both deterministic and random signals are thus developed.

The importance of systems to the communications engineer lies in the fact that signals conveying information must be processed many times by subsystems (filters, modulators, amplifiers, equalisers etc.) before they reach their final destination. It is only through a thorough understanding of the modifying effect of these subsystems that one can ensure, in the presence of noise, that signals will remain adequately distinguished to achieve message reception without error.