Chapter 4

Transmission systems

4.1 Introduction

This chapter describes the wide variety of transmission systems used in telecommunication networks today. In doing this, we will be drawing upon the concepts of multiplexing and analogue-to-digital (A/D) conversion introduced in the previous chapter. Earlier chapters have also introduced the important distinction between the Access Network, which serves the subscribers, and the Core Transmission Network, which provides links between network nodes only. In general, the transmission systems described in this chapter may be deployed in both the Access and the Core Transmission Networks – although they are usually more appropriate to one or the others, and this will be indicated. Chapter 2 introduced the model of the networks associated with the public switched telephone network (PSTN) (see Figure 2.11), in which the Access and Core Transmission Networks act as a common utility for providing circuits to the various specialised networks as well as the PSTN. However, we will leave the description of how the various transmission systems are deployed in the Access and Core Transmission Networks until Chapter 5.

4.2 Transmission bearers

In this section, we look at the general principles of transmission and the range of practical systems used in a telecommunication networks.

4.2.1 Transmission principles

The purpose of a transmission system is to provide a link between two distant points or nodes. The link may be unidirectional, as in the case of radio broadcast where transmission is only from the transmitter to the radio receivers, or bidirectional, as in the case of a telephone connection with its two-way conversation. In telecommunication networks, the links between nodes are either 2-wire or 4-wire circuits, as described in Chapter 1. With 4-wire circuits, the separate 'Go' and 'Return' paths are considered as transmission channels, as explained in the generalised view of a multiplexed transmission system in Figure 3.14 of Chapter 3.

In general, a transmission system takes as an input the signal to be conveyed (which may be a single channel or a multiplexed composite of channels) and converts that signal into a format suitable for the transmission medium, propagates the transmission signal to the far end, where after conversion from the transmission signal a reproduction of the input signal is produced (see Figure 3.13 of Chapter 3). The actual conveyance of the channel over the transmission system is achieved by 'modulating' an electrical bearer signal which travels over the transmission medium. It is the modulation or modification of the bearer signal that constitutes the transmitted information, not the bearer itself – indeed, the bearer signal is usually called a 'carrier' for this reason.

There are many forms of modulation [1]. The normal form of carrier is a sinusoidal waveform, as introduced in Chapter 1 (see Figure 1.2). Some of the common forms of modulating sinusoidal carrier waveforms are:

- Amplitude modulation (AM), in which the input signal directly varies the height or amplitude of the carrier, shown in Figure 4.1.
- Frequency modulation (FM), in which the input signal directly varies the frequency of the carrier, as shown in Figure 4.1.
- Phase modulation (PM), in which the input signal directly varies the phase (i.e. positioning of the start of the wave) of the carrier.

All three methods described above are used widely in radio and TV broadcasting, where a radio carrier is modulated, and most people would recognise the commonly used terms of 'FM' and 'AM'. They are also used in telecommunication transmission systems, as described later. However, these modulation systems are analogue because the carrier is continuously moderated by the input signal and the carrier can take any value within the working range of the system.

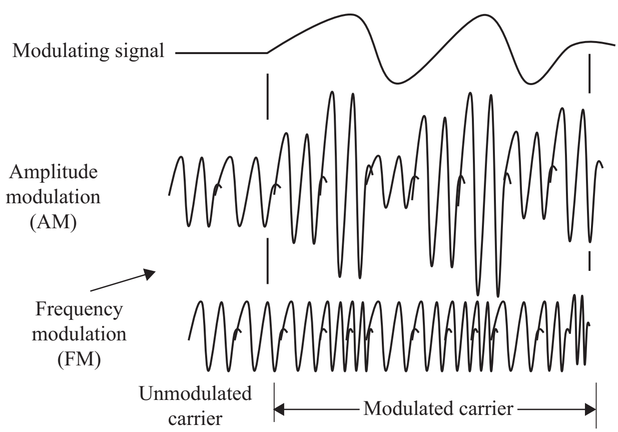


Figure 4.1 Carrier-wave modulation

A digital carrier, in the form of a stream of pulses can also be modulated, taking a fixed number of values, by an input signal. Examples of such modulation schemes include:

- Pulse amplitude modulation (PAM), in which the height of each carrier pulse is moderated by the corresponding bits in the digital input signal.
- Pulse position modulation (PPM), in which the position of each carrier pulse is moderated by the corresponding bits in the digital input signal.
- Pulse-code modulation (PCM), described in Chapter 3, whereby a binary stream is modified to convey the successive digital representations of a set of time-division multiplexing (TDM) (PAM) speech samples.

These digitally modulated streams may themselves also be used to modulate a radio or optical carrier on a transmission system. The input signal, in the form of pulses, is then used to modulate the carrier in a series of step changes between a fixed set of values. This is known as 'keying'. Examples of such systems include: phase-shift keying (PSK) – in which the output signal steps between two or more phases in response to the binary modulating signal; and frequency-shift keying (FSK) – in which the output shifts between two or more frequencies in response to the binary input. The advantage of these keying modulation systems is that the output signal can step between more than just two values, for example, 4, 8, or even 16 levels of phase can be used in what is known as 4-, 8-, or 16-phase PSK, respectively. In the example of the 4-phase PSK system, the input-modulating digital stream is taken 2 bits at a time (i.e. *00*, *01*, *10*, or *11*) to set the appropriate phase level in the output signal. Similarly, three bits at a time (*000*, *001*, *010*, *011*, *100*, *101*, *110*, or *111*) are taken for 8-phase PSK, four bits at a time for 16-phase PSK, and so on.

All transmission media introduce some impairment to the signal being conveyed due to the physical mechanisms involved. Examples of the most common forms of impairment are:

- Loss of power, due to the absorption, scattering and reflections of the signal within the transmission medium, known as 'attenuation'.
- Delay to the signal, due to the propagation through the medium. This will vary according to the characteristics of the medium (e.g. the purity of the glass in an optical fibre) and external factors such as temperature.
- Phase distortion, due to the different amount of delay introduced by the transmission medium to each of the component frequencies in a multi-frequency transmitted signal. The resulting spread in the arrival time of the waveform is known as 'dispersion'.
- Amplitude distortion, resulting from the different loss of power introduced by the transmission medium for each of the component frequencies in the transmitted signal.
- Noise, the pick-up of unwanted signal generated intrinsically by the medium (e.g. thermal noise), as well as induced noise from outside sources (e.g. crosstalk from adjacent transmission systems).

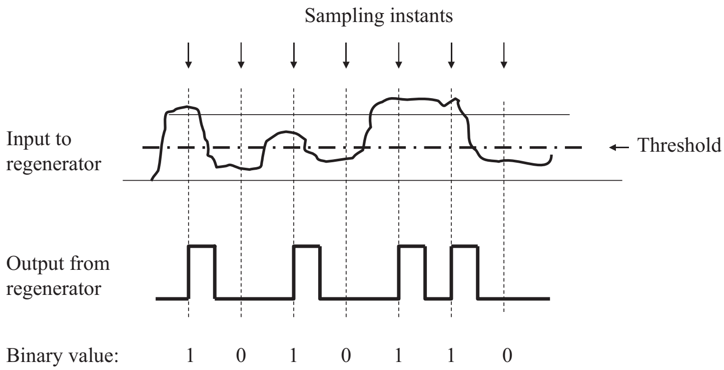


Figure 4.2 Digital regeneration

The design of transmission systems aims to minimise and compensate for the various impairments introduced by the medium. It should be noted that the extent of the above-mentioned impairments may change with the temperature of the transmission medium. There are methods for compensating for phase dispersion and amplitude distortion, known as equalisers, for example. However, these are effective only over a defined operating range. Thus, there will be limits on the length of a transmission system within a network if it is to operate within the accepted range of loss, delay, noise, and phase distortion. There will also be a limit to the bandwidth (range of frequencies carried) that may be used. Normally, the usable length of a transmission system is increased by introducing devices located appropriately along its route to boost the signal strength, compensating for the power loss.

In the case of an analogue transmission system, the signal accumulates the impairments along the length of the medium, so the introduction of an amplifier to compensate for power loss results in not only the wanted signal being boosted, but also that of the added noise. The ratio of the wanted signal to noise (signal-to-noise ratio) gets progressively worse at each point of amplification along the route of the transmission system – so setting a limit on the acceptable number of amplifiers and hence length of transmission link. By contrast, a digital transmission system can reconstitute a noise-free signal at each point of regeneration along the route. This is illustrated in Figure 4.2, where the impaired signal from the line is sampled by the digital regenerator and any signal above the threshold at the sampling instants will result in a clean digit '1' being generated and sent to line. (See also Chapter 3 and Figure 3.15 for a description of how this feature is exploited in an integrated digital network (IDN).)

Figure 4.3 illustrates the components of a generalised digital line transmission system. This is a development of the link functions as shown in Figure 3.13 of Chapter 3. The input digital signal in the 'Go' direction is shown in the left-hand side, this is either a multiplexed assembly of many channels (see Section 4.3.4 for synchronous digital hierarchy (SDH) or Section 4.3.3 for plesiochronous digital

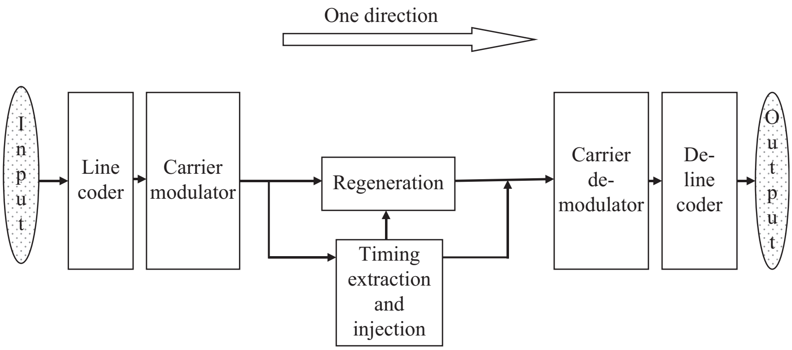


Figure 4.3 Digital line transmission system

hierarchy (PDH)) or one very large capacity single channel. The signal is then converted through the application of a line code to a format that suits the transmission medium. This will produce a signal that contains timing information that can be extracted by the regenerators along the line, to ensure that they stay in synchronism with the sending end. Line coding may also improve the efficiency of the transmission system by converting the two-level binary signal to a multilevel signal for actual transmission over the medium. An example of this was given with the use of four-phase PSK above, where the binary signal was converted to a four-level signal for transmission. This enables the transmission system line rate (known as the 'baud rate') to be half that of the binary input signal. In this way, a transmission system can carry higher capacities by using line codes of increasing numbers of levels. In practice, there is a trade-off between the increased capacity or efficiency of the transmission system and the cost and complexity of the line-coding detection mechanism, which generally increases with the number of levels used. There are many forms of line codes employed in transmission systems, each offering different levels of efficiency against complexity and equipment cost and, of course, suitability for the various transmission media [2,3].

Next, the line-coded signal is used to modulate the carrier for the transmission system. In the case of an optical fibre system, the signal modulates a laser sending a corresponding stream of light pulses down the fibre (i.e. the medium) – undertaking an electro-optical conversion. At each of the regenerators (one only is shown in Figure 4.3) the timing information (inserted by the line-coding process) is extracted and used to drive the sampling gates in order to regenerate the signal, as described above.

At the far end the appropriate demodulation occurs, for example a photo detector in an optical fibre system receives the transmitted light pulses and generates a stream of digital electrical pulses (i.e. opto-electrical conversion). This electrical signal is then transcoded from the line code back to the format of the binary input.

The principles described above apply to all the many forms of transmission systems carried over the different forms of media – namely: metallic cable, optical fibre cable, and radio over free air. Given this wide range of possible transmission options, the choice of system used in a network depends on the capacity to be carried, the existing transmission infrastructure, the distance involved, expected capacity growth potential, and the cost of the equipment. We can now look at these aspects as they affect a telecommunication network operator.

4.2.2 Transmission media

Figure 4.4 illustrates the physical characteristics of the transmission media used in modern networks.

4.2.2.1 Copper pair cable

As described in Chapter 1, copper pair cable has been adopted throughout the world as the means of providing the local loop between the subscriber and the serving telephone exchange. But, the humble copper pair cable can also be used to carry a variety of data high-speed services, and even video, by the addition of appropriate electronics (e.g. asymmetric digital subscriber line (ADSL)) at its ends, as described later in the chapter [4,5].

Copper pair cable is available as a single pair of wires, with plastic insulation, usually deployed between a subscriber's premises and the overhead or underground distribution point (DP), as described in Chapter 5. However, the copper pairs are also provided in multi-pair cable bundles of a variety of sizes (e.g. from 2-pairs up to 4,800 pairs). The thickness or gauge of the copper conductors also comes in a range of sizes, with corresponding electrical resistance values per km. Figure 4.4(a) shows the general construction of a multi-pair cable. The individual pairs have a thin insulating cover of polyethylene and these are grouped into quads in order to minimise adverse electrical interference problems. The coating of the pairs is

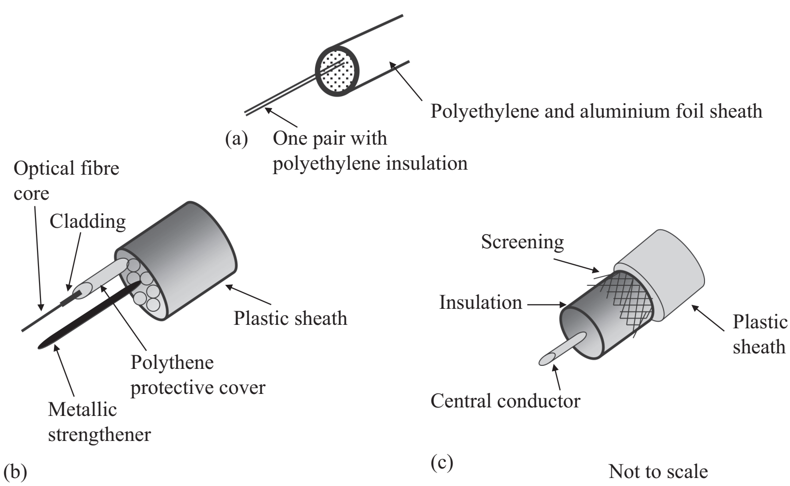


Figure 4.4 Cable transmission media. (a) Copper pair in a multi-pair cable. (b) Optical fibre in a multifibre cable. (c) Coaxial cable

usually colour-coded to assist their identification by the technician jointing the ends onto other cables and terminals – quite a task when there are tens, hundreds or even thousands of pairs in one sheath! The sheath is constructed from polyethylene for strength and durability, together with aluminium foil to provide a water barrier.

Although not shown in Figure 4.4, special forms of copper cable, arranged with one central conductor surrounded by a copper cylindrically shaped conductor – known as 'transverse screen' cable – were deployed in the United Kingdom and elsewhere during the 1970s and 1980s to provide high-speed digital circuits in the Access Network [5]. Now, optical fibre or digital transmission over standard copper pairs is used in preference.

4.2.2.2 Optical fibre cable

Optical fibres provide a transmission medium of huge potential capacity by constraining pulses of light within a thin core strand of highly pure glass or silica. The light stream is contained within the core by a cladding of glass which has a lower refractive index, causing the light to be totally reflected internally at the core-to-cladding boundary [2]. The light is pulsed at the sending end by solid state electro-optic transducers, such as light-emitting diodes (LED) and lasers. Detection at the distant end is by solid state opto-electric transducers, such as PIN and avalanche photodiodes.

The key requirement for an optical fibre transmission system is the use of a glass (silica) fibre material which offers very low loss to the light passing through. This is achieved through a rigorous manufacturing process in which the glass purity is strictly controlled. In addition, advantage is taken of the transmission characteristics of the glass which offers different attenuation to light of different frequencies, as illustrated in Figure 4.5.

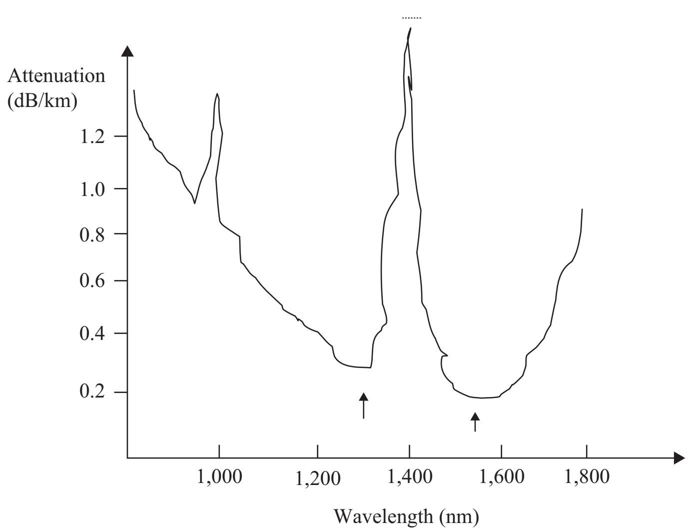


Figure 4.5 Optical fibre operation

Optical fibre transmission has the following characteristics compared to metallic cable transmission:

- Huge potential capacity due to the inherently higher operating frequencies.
- Low power loss (e.g. <0.5 dB/km), resulting in the need for few, if any, repeaters along the route. See Box 4.1 for an explanation of the term dB used as a measure power levels within telecommunication networks.
- The optical nature of the medium means that the transmission is unaffected by electrical inference in the vicinity.
- No escape of energy from the fibre, so that there is no transmission interference between fibres or other lines.
- Physically smaller than metallic cables of comparable capacity.
- Can be difficult to join optical fibres, special splicing techniques are required.

Figure 4.4(b) shows the construction of an optical fibre within a multifibre cable. The cables, which range in sizes (e.g. from 2 to 96 fibres), contain the fibres within a plastic sheath about a central metallic wire, which provides mechanical strength for the otherwise very fragile bundle of optical fibres. Typical diameters of the optical fibres are 50–200 μm core with 125–400 μm cladding for the earlier (multimode) types to 5–10 μm core with 125 μm cladding for the more recent (single mode) types. (A 'μm' is 1 millionth of a meter.) These fine glass threads, which are about the thickness of a human hair, are wrapped in a polythene protective cover for robustness.

Box 4.1 The decibel measure of power

Power levels in telecommunication networks are measured in units known as decibels (one-tenth of a 'Bel'). The important characteristic of dBs is that they represent a logarithmic comparison of the measured power level against some other power level. Thus, if the power output (P1) from a radio was turned up to twice the level (P2), the increase in power is calculated as $10 \log_{10} P2/P1 = 10 \log_{10} 2 = 3$ dB. Since the measure is a logarithmic ratio, it does not matter whether the two powers (P1 and P2) are large or small – a doubling of power is always 3 dB and, by the same logic, a halving of power is – 3 dB. The main advantage of the use of this logarithmic measure is that all the increases and decreases of power contributed by the elements of a circuit throughout the network are simply added or deducted from the total to give the overall power level or loudness rating.

However, where an absolute power level is to be measured, for example, the output from some electrical equipment, it is compared to a standard power level, typically 1 mW (milliWatt). Powers are then shown with units of 'dBm'. Thus, a power level of 6 dBm would be equal to 4 mW, that is, double times double (3 dB + 3 dB) the power of 1 mW.

4.2.2.3 Coaxial cable

As the name suggests, coaxial cable comprises of a metallic central core cable encased by a cylindrical conductor. The shielding effect of the outer conductor provides an interference-free transport medium for high-speed electrical signals. Coaxial cable, similar to that used for connecting aerials to a TV set, is used in the Cable TV networks to provide the video connection from the street electronics to an individual household, as described in Chapter 2 (see Figure 2.2). Early forms of long-distance transmission networks for the PSTN employing analogue FDM systems were provided over coaxial cables from the 1960s to the 1980s. Now, telecommunication networks use optical fibre in preference to coaxial cables for the Core Transmission Network.

Even though it is not now deployed in the external telecommunication networks, coaxial cable is still used extensively for interconnecting transmission and switching equipment within exchange and repeater station buildings. This interconnection is usually provided across digital distribution frames (DDFs), as described in the Section 5.3.2 on PDH transmission in Chapter 5.

The structure of a coaxial cable is shown in Figure 4.4(c). For physical flexibility, the shielding or screening conductor is made out of metallic braiding underneath a plastic sheath. The screening is separated from the central core conductor by a soft insulator so that the overall assembly of the cable is robust and flexible.

4.2.2.4 Atmosphere: radio

The medium of the atmosphere is unlike that of the confined environment of metallic cables or optical fibres, and its unbounded nature requires the transmission systems exploiting it to have a high degree of adaptability to the medium's changing characteristics. Radio transmission through the air is achieved in several ways, mainly depending on the frequency of the carrier wave used. The various types of radio paths are illustrated in Figure 4.6 and briefly described below [4].

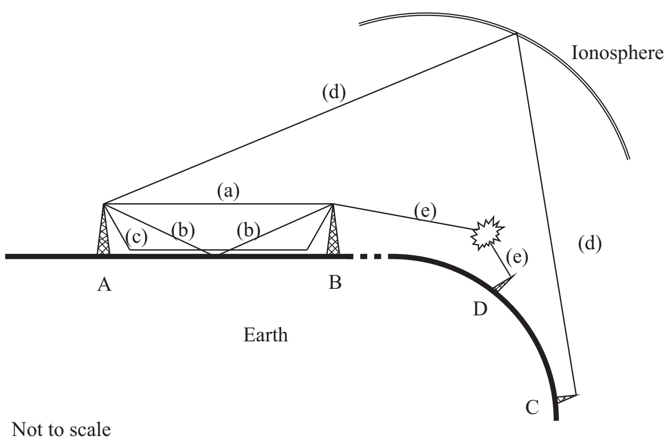


Figure 4.6 Different types of radio paths

Path type (a): direct wave (also known as 'free space wave')
This path uses free space propagation through the air, the radio signal following a straight line of sight between antennas. In order for a sufficient proportion of the transmitted power to be received the two antennas need to be as directional as possible. Since the directional characteristics of antennas increase with frequency, practical systems tend to be at microwave frequencies, i.e. above 1 GHz (1,000,000,000 Hz).

Path type (b): ground reflected wave
In addition to the direct wave there will inevitably be one or more reflected wave paths between the sending and receiving antennas. Normally, these reflected paths cause some degree of interference with the direct signal at the receiver because the reflected signal will experience greater delay, and hence be out of phase with the direct signal. Line-of-sight systems use a variety of methods for reducing the interference from the reflected waves.

Path type (c): ground or surface wave
Ground waves, which follow the surface of the Earth through the process of diffraction, are generated by electrical currents induced in the ground. Radio systems using surface waves are able to extend beyond the line of sight, following the Earth's curvature. The attenuation of surface waves increases with frequency, so practical systems operate in the low-frequency range (30 Hz to 30 kHz) and, thus, are of low capacity.

Path type (d): skywave
This path is created by the reflection of the radio waves off the Ionosphere, the ionised layers belting the Earth. The waves can carry over very long distances through multiple reflections off the Ionosphere and ground. Early long distance, that is, intercontinental, communications were achieved using high-frequency (HF) radio. However, the variable nature of the Ionosphere and the multipath nature of the propagation meant that the transmission was subject to fading and daily fluctuations. Thus, alternative transmission systems (e.g. optical fibre, microwave radio, and Earth satellite) are mainly used in preference to HF radio.

Path type (e): tropospheric scattering
This path, which is created by the scattering of the radio wave by perturbations in the atmosphere, enables communication to a receiver just over the horizon. Practical systems operate at microwave frequencies using very high-transmitting power and large high-gain receiving antennas to compensate for the very high attenuation of tropospheric scattering paths. Such systems are used by the military to provide communication in battle conditions. British Telecommunication (BT) uses tropospheric systems to connect all the oil rigs in the North Sea to mainland United Kingdom, thus extending the PSTN offshore beyond the horizon.

Figure 4.7 summarises the various radio propagation techniques described above and shows the radio spectrum frequency designations.

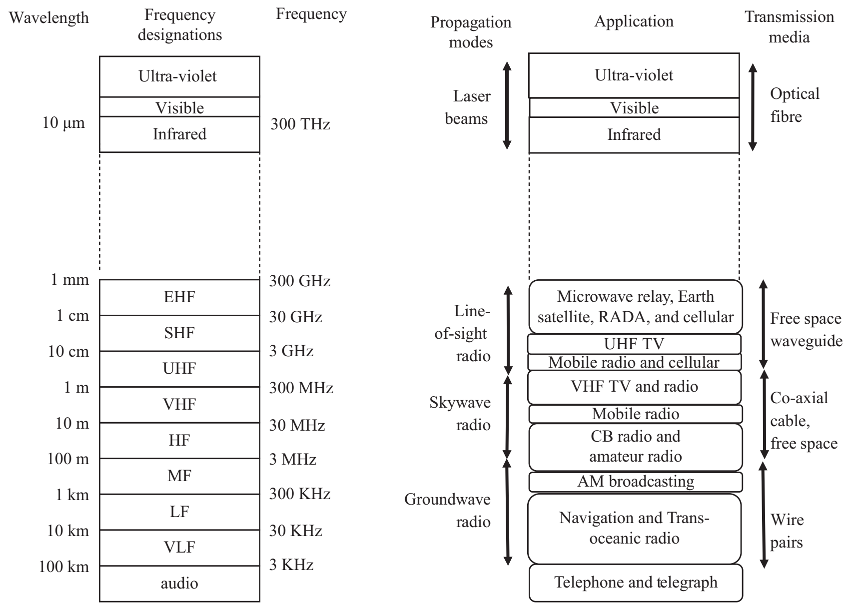


Figure 4.7 The ways that transmission systems use the electromagnetic spectrum

Telecommunication networks today use a mix of radio transmission systems. Invariably, these operate at microwave frequencies in order to exploit the higher bandwidth capacity and hence economy of scale that they provide. With the exception of the tropospheric (or 'troposcatter') systems, these systems work on a line-of-sight point-to-point basis to provide high-capacity links for the Core and Access Transmission Networks, or alternatively in a multipoint configuration in the fixed Access Network linking several customer locations to a central exchange.

Figure 4.8(a) illustrates the configuration for a microwave point-to point relay route, providing transmission capacity between two antennas via an intermediate relay stage (repeater or amplifier for digital or analogue transmission, respectively). The Go and Return channels are conveyed on separate carrier frequencies. Such transmission systems are able to provide routes across a country using many repeater antennas, typically spaced at between 20 km and 40 km apart.

4.2.2.5 Free space: Earth satellites

By locating the microwave radio relay in a communications satellite in space orbit around the World, the distance between a pair of antennas can be extended to intercontinental distances, as shown in Figure 4.8(b). Clearly, the free space loss of the radio signal is correspondingly large because of the long distances between Earth antennas and the satellite relay – typically this so-called 'path loss' up and down totals around 200 dB, depending of the frequency of the carrier wave and the path length, the latter depending on the latitude of the Earth stations. Satisfactory performance requires compensation for this loss by using high-gain repeaters and

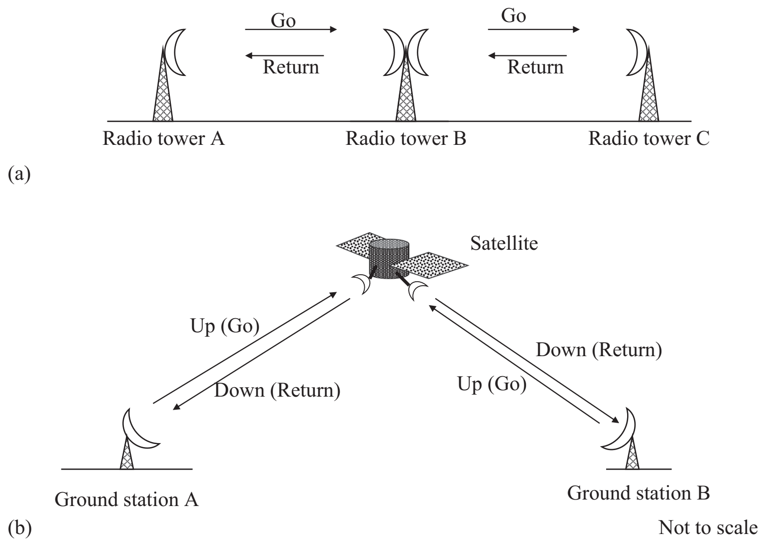


Figure 4.8 Free-space transmission media. (a) Microwave radio relay. (b) Microwave Earth satellite relay

antennas on board the satellite, as well as high-gain antennas on the ground. Since the gain of an antenna increases with size, the gain of the antennas on board the satellite is limited by the weight and diameter of the antennas that can be physically mounted on a space vehicle, so the gain of the antennas on the ground need to be very high – hence the use of the huge satellite dish aerials, which are a familiar sight at satellite Earth stations. Alternatively, smaller capacity transmission links can be achieved from a central large dish antenna at one end of the system and the use of small portable antennas at the other, for example, small-aperture satellite systems as used by news TV broadcasters, and handheld satellite mobile phones used in remote areas of the world.

4.3 Multiplexed payloads

The costs of a telecommunications network are minimised by exploiting the significant economies of scale offered by the above-described transmission systems. In general, the cost per circuit carried reduces as the capacity of the transmission system increases; thus, it pays to multiplex as many channels as possible on to each transmission link. In this section, we consider the standard ways in which channels carrying telephone calls or data communications are assembled into multiplexed composite signals – the payloads – for conveyance over digital transmission systems, that is, copper cable, coaxial cable, optical fibre, microwave radio or satellite, as appropriate. (It should be noted that the multiplexed payloads first used in telecommunication networks were analogue and based upon frequency-division

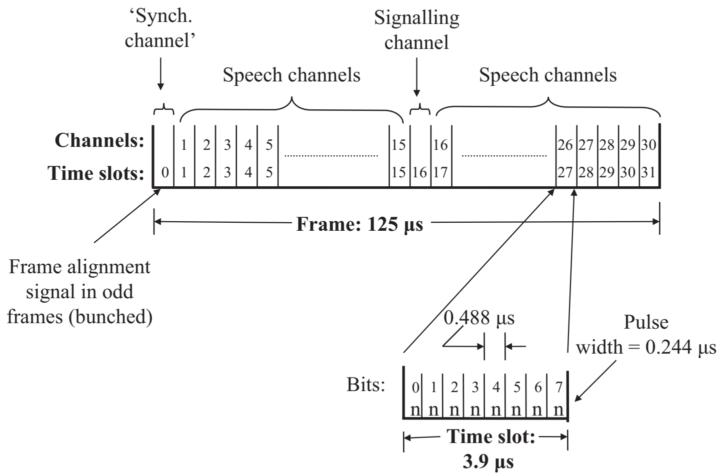


Figure 4.9 The 30-channel pulse-code modulation multiplex frame structure

multiplexing (FDM) – a technique described in Chapter 3.) However, these systems are now obsolete and so they are omitted from this book for brevity. A description of FDM transmission payload systems may be found in [6].

4.3.1 The pulse-code modulation multiplexed payload: the basic building block of digital networks

The primary multiplexed payload created by assembling a group of PCM channels forms the basic building block of all digital transmission and switching networks used throughout the world. Thus, it is an important entity to understand. There have been several versions of the PCM primary multiplex payload, but there are now just two versions standardised by the ITU (previously known as the 'CCITT'): the 30-channel standard, used in Europe, Asia, and elsewhere regionally and on all international links; and the 24-channel DS1 standard, used regionally, mainly in North and South America. Both versions are described below. Chapter 3 introduces the concept of TDM, analogue-to-digital conversion and PCM.

4.3.1.1 30-Channel pulse-code modulation multiplex

In this TDM digital standard, the primary multiplexed structure is based on a timeframe divided into 32 timeslots, as shown in Figure 4.9. The timeframe length is 125 µs, resulting from the sampling rate of 8,000 times a second – 8 kHz (the necessary rate to sample a speech waveform with its frequency content limited to 4 kHz), as explained in Chapter 3. Thus, each timeslot in this frame occupies 1/32 of 125 µs, that is, 3.9 µs. The 32 timeslots of the frame are permanently assigned to 30 speech channels, hence the name of the system, and two non-speech-carrying support channels, as shown in Figure 4.9. The first of these support channels is carried in timeslot 0 (written as 'TS0') and it is often erroneously known as the 'synchronisation' channel, which is used to indicate the start of the frame, as described later. The second support channel is carried in timeslot 16 (TS16) and is used to carry the call-control signalling between the exchanges at either end of the PCM route. (Chapter 7 describes the two sorts of signalling systems that can be carried in TS16.) Timeslots 1–15 are used to carry speech channels 1–15, respectively; timeslots 17–31 are used to carry speech channels 16–30, respectively.

The A/D conversion process (see Chapter 3) within the PCM system encodes each speech sample into an 8-bit binary word. Therefore, each of the 32 timeslots in the multiplex carries 8 bits, making a total of 256 bits per frame. Since there are 8,000 frames transmitted every second, the line rate of the 30-channel PCM multiplex is 8,000 bits/s × 256 bits, i.e. 2,048,000 bits/s. This rate is usually designated as '2,048 kbit/s' or '2.048 Mbit/s' – the latter is usually abbreviated to the more convenient and popular form of '2 Mbit/s'.

Another important figure to remember with the 30-channel multiplex is the line rate provided for each speech channel. This is given simply by the number of bits per sample, 8, times the number of samples per second, 8,000 – that is a line rate of 64,000 bits/s, or 64 kbit/s.

Figure 4.9 also shows how the 8 bits of an individual channel are represented by digital pulses every 0.488 μs (i.e. 3.9 μs/8), each pulse being 0.244 μs or 244 ns (nano equals 1/1,000,000,000) wide.

We are now able to describe the function of the synchronisation channel in TS0. Its role is best explained by considering the terminating end of the digital transmission system which is receiving a stream of digital pulses arriving at the rate of 2 Mbit/s. The timing of the pulses is extracted from the incoming stream and hence sets the receiver sampling rate. However, this stream of pulses is meaningless unless the start of the frame can be identified; thereafter, by counting the bits received the set of eight bits relating to each channel can be located. This frame-start identification is indicated by a special bit pattern, called the 'frame alignment pattern', which is inserted at the sending end into the TS0 of odd frames, as shown in Figure 4.9. At the receiving end the first 8 bits of two frames worth of bits are examined in a digital register. If the frame alignment pattern is not detected, the register shifts one bit and looks at the first 8 bits again. This continues until the pattern is found, the start of the frame has then been detected and the receiver is now in 'frame alignment' with the sender. The spare 8-bit capacity in the even frames of TS0 is available for special purposes – for example, BT used bit 5 to carry network synchronisation control signals for its UK digital network [7].

Digital telecommunication networks have been developed and built on the basis of the 2 Mbit/s building blocks – often referred as '2 Mbit/s digital blocks'. Not only are the blocks used over digital transmission networks, but they are also the entry level into digital exchanges (local, trunk, and international and mobile), as described in Chapter 6. In addition, data can be directly inserted into a 2 Mbit/s multiplex since it is already in digital format. Thus, the 2 Mbit/s digital block may be considered as a payload vehicle capable of carrying 30 speech (or voice) channels or 30 data channels, each of 64 kbit/s, over a digital transmission network.

The 2 Mbit/s block is also used by network operators as the primary rate for a digital stream which is delivered to customer's premises for a variety of business services, including digital leased lines, connections to digital ISDN PABXs, as well as ATM, Frame relay and SMDS data services, as described in Chapter 2. In all these cases the 2 Mbit/s block is carried through the network and presented to the customer as an unstructured stream of 2 Mbit/s, with no implied channel structure.

These 2 Mbit/s digital blocks, whether carry data or voice, may be carried directly over a transmission link or multiplexed together with other blocks to form higher-capacity payloads for PDH or SDH transmission systems, as described below.

4.3.1.2 24-Channel pulse-code modulation multiplex (USA DS1)

Figure 4.10 shows the frame structure for the 24-channel DS1 system. Although the speech is digitally encoded into 8 bits using 8 kHz sampling rate, as in the 30-channel system described above, there is an important difference between the two systems in the way that it is done. This is because the spacing of the graduations on the codec ruler (see Chapter 3) is based on the 'A-law' for the 30-ch. systems and the 'Mu-law' for the 24-ch. system. These two laws, being different ways of spreading the quantum steps using spacing that roughly approximate to a logarithmic scale, ensure a constant signal to quantisation-error ratio over the operating range. Thus, the same waveform would be encoded into a different set of 8 bits by the two systems and a transcoding is required when 30-ch. PCM is connected to 24-ch. PCM links. (E.g. this transcoding takes place at the international gateway exchanges into the United States for calls between the United Kingdom and the United States.) All the 24 timeslots within the 125 μs timeframe are normally used for speech channels (but see below), there being no equivalence of the TS16 or TS0 of the 30-ch. system. However, a single bit is contained at the front of

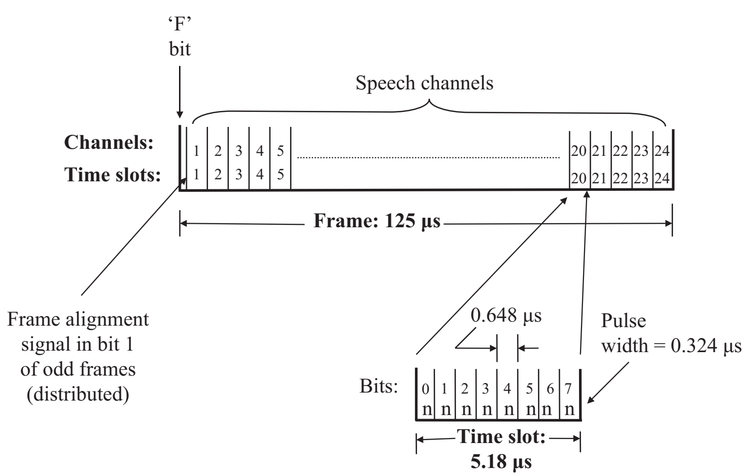


Figure 4.10 The 24-channel pulse-code modulation multiplex structure

the frame before TS1 which is used to carry the frame alignment pattern. The pattern is sent one-bit at a time over odd frames – that is dispersed over eight odd frames, rather than bunched into one frame, as in the 30-ch. system.

The frame comprises of 24 eight-bit time slots plus one bit, totalling 193 bits. The line rate is thus 193 bits per frame, with 8,000 frames/s, giving 1,544 kbit/s – which is usually written as '1.5 Mbit/s'. This rate is also known as the 'DS1' rate in the United States and each of the component 64 kbit/s channels are known as 'DS0'.

There are two ways that signalling can be conveyed over the 24-ch. system. The earlier systems used a method known as 'bit stealing' in which the last bit in each timeslot of every sixth frame was used to carry signalling related to that channel. This periodic reduction in the size of the PCM word from 8 to 7 bits introduced a slight, but acceptable, degradation to the quantisation noise. However, it also prevented the timeslots from being used to carry data at the full 64 kbit/s (i.e. 8 bits at 8,000 frames/s) and the reduced rate of 56 kbit/s (i.e. 7 bits at 8,000 frames/s) was the maximum rate that could be supported. The recently introduced alternative of common-channel signalling avoids the need for bit stealing, since the signalling messages are carried in a data stream carried over the bit 1 at the start of even frames – that is outside of the speech channels. However, this 4 kbit/s (i.e. 1 bit every other frame) of signalling capacity was considered too slow and now the DS1 system has been upgraded to carry common-channel signalling at 64 kbit/s in one of the time slots, leaving 23 channels for speech.

The 1.5 Mbit/s digital block is also offered by network operators as an unstructured primary rate of digital service delivery to customers' premises for business services (leased lines, ISDN, ATM, frame relay, and SMDS). This 1.5 Mbit/s digital block is frequently referred to as a 'T1' system in North America. (The term 'E1' is also used, particularly in North America, to describe the equivalent 2 Mbit/s digital block used in Europe and elsewhere.)

These 1.5 Mbit/s digital blocks, containing data or voice, may be carried directly over a transmission link or multiplexed together with other blocks to form higher-capacity payloads for PDH or SDH/SONET transmission systems, as described later in this chapter.

4.3.2 The time-division multiplexing of digital blocks

Larger payloads may be created by time-division multiplexing several 2 Mbit/s or 1.5 Mbit/s digital blocks. In Figure 4.11 the concept is illustrated using a rotating arm to sample each of four 2 Mbit/s digital block tributaries, each with timeslots 0–31. The 125 μs timeframes of the tributaries is assumed to be aligned so that they all start at the same instant. Since the frame size of the TDM highway is the same as that of the tributaries, the time spent by the wiper on each tributary can be no more a quarter of 125 μs in total and the line rate on the multiplexed highway is 4×2 Mbit/s (i.e. 8 Mbit/s). The wiper therefore must sample each tributary at the rate of 8 Mbit/s. The content of the TDM highway is therefore four times that of each single tributary, that is, 120 traffic channels in 128 time slots, with each time slot

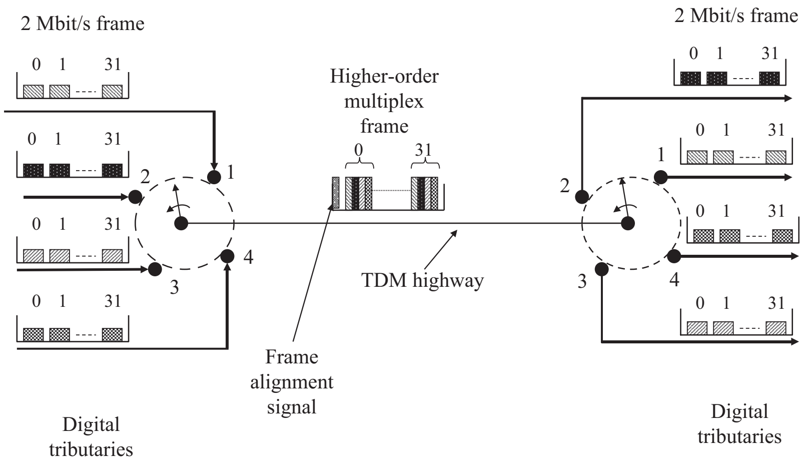


Figure 4.11 Interleaving of digital blocks

occupying 3.9 μs divided by 4, i.e. 0.975 μs. This gives each bit occupying a quarter of 0.488 μs, i.e. 0.122 μs. At the far end of the TDM highway the process is reversed and the appropriate contents are streamed to the corresponding output tributary.

Clearly, for this TDM system to operate accurately the two wipers have to rotate at the same speed and their starting points aligned so that corresponding tributaries at the input and output are sampled simultaneously. The speed of the distant wiper is set by the incoming bit stream, as described above. Whilst the alignment is achieved using a frame alignment pattern (known as a frame alignment signal (FAS)) to indicate the start of the TDM frame, in a similar way to that used to define the start of the PCM frame, described earlier. Figure 4.11 shows the FAS at the start of the TDM frame.

There are two basic ways of sampling the tributaries and interleaving their contents onto the TDM highway: 'bit' interleaving and 'word' or 'byte' interleaving. With the former, one bit at a time is sampled from each tributary in turn; with the latter the full PCM word of 8 bits (also called a 'byte') is sampled in one go from each tributary in turn. Bit interleaving is used in the PDH systems, the original TDM digital payload system, since the design is easier to construct and is readily compatible with digital-to-analogue conversion. Whilst the more recent TDM payload system of SDH, which is designed to carry both data and voice equally, and assumes full digital networking, uses the more complex technique of byte interleaving [8].

The TDM output from the multiplexer can itself be multiplexed again with other tributaries following either the PDH or SDH/SONET formats, and so on to create an appropriately-sized multiplexed payload, as described later in this chapter.

4.3.3 Plesiochronous digital hierarchy system

The plesiochronous digital hierarchy (PDH) was the first internationally standardised form of digital higher-order multiplexing and was deployed over a variety of cable and radio systems, as well as optical fibre cable around the World. Although now largely replaced by SDH systems, there is still some PDH capacity in most incumbent operators' transmission networks. The name 'plesiochronous', meaning nearly synchronous, relates to the situation where the individual 2 Mbit/s tributaries that are to be multiplexed over the PDH system are operating at close-but-slightly-varying rates. This has several consequences. The first consequence is that 'stuffing' or 'justification' bits need to be added to the TDM highway at each stage of multiplexing.

The concept of bit stuffing may usefully be explained by considering a cereal manufacturer who wishes to pack up 12 packets of cornflakes into a large cardboard box for transporting to a grocery shop. The cornflakes packets are nominally sized as 2 cm thick, which means that some boxes will be narrower (e.g. 1.98 cm), whilst others might be wider (e.g. 2.04 cm). Therefore, the size of the cardboard box will need to be greater than 12×2 cm, to allow for this variation in sizes. Once all 12 cornflakes packs are placed into the box the grocer will insert pieces of cardboard as packing to fill up any slack space so that the contents are held tightly. At the receiving shop the cardboard packing will be discarded and the individual 12 packets of cornflakes extracted. Different amounts of cardboard packing will be required in each subsequent box, depending on the mixture of cornflake packet sizes in each batch of 12. The boxes containing 12 cornflakes packets can themselves be packed into larger boxes, again using cardboard spacers to compensate for the variations in box dimensions, and this process can continue with further stages of packing. By analogy bit stuffing in a PDH system is applied to ensure that the TDM frame is filled tightly, irrespective of the actual number of bits sampled from each tributary. There are several ways in which the number and position of the stuffing bits can be indicated to the receiver of the PDH system, so that the appropriate bits can be discarded when the tributaries are extracted.

The European standard for PDH higher-order transmission payloads is shown in Figure 4.12. Four stages of TDM multiplexing are defined, each being a multiple of four. The first stage multiplexes four 2 Mbit/s digital blocks into an 8 Mbit/s stream. The actual line rate of the output of this 2/8 muldex (the name for the

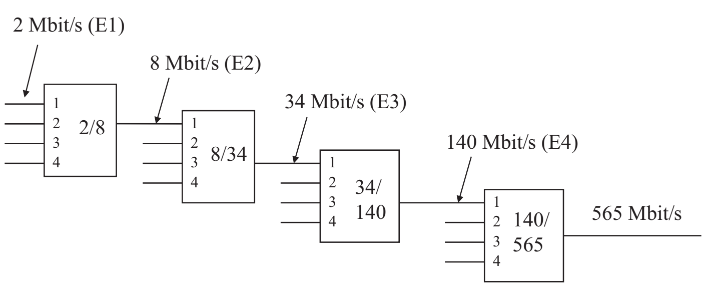


Figure 4.12 Plesiochronous digital hierarchy European standard

multiplexer-to-de-multiplexer assembly, allowing for both directions of transmission) is 8,448 kbit/s. This rate is usually referred to as 8 Mbit/s or E2 line rate. The difference of the E2 rate from four times 2,048 kbit/s results from the addition of an FAS and stuffing bits. The subsequent stages of multiplexing similarly include frame alignment and stuffing bits in the aggregate line rates. Table 4.1 shows all the stages of PDH multiplexing, their nominal and actual line rates, the designation used for line systems at that rate and the number of 64 kbit/s speech channels that could be carried.

The North American PDH standard is shown in Figure 4.13 and Table 4.2. In addition to the DS designations, the line rates when made available for customers' use have the 'T' designations; for example, a 45 Mbit/s leased line is referred to as a 'T3' line system.

Although PDH digital line systems have been successfully deployed since the 1970s around the World, they have proved to be inflexible and cumbersome to manage in large-scale transmission networks due to the so-called multiplexer mountain problem. Figure 4.14 shows that 42 multiplexers are required to provide a fully equipped 140 Mbit/s PDH multiplexed system (for clarity, the line

Table 4.1 European PDH standard

Nominal rate (Mbit/s)	**Actual line rate (kbit/s)**	**Designation**	**Number of channels (64 bit/s)**
2	2,048	E1	30
8	8,448	E2	120
34	34,368	E3	480
140	139,264	E4	1,920
565*	564,148	E5*	7,680

*Not standardised.

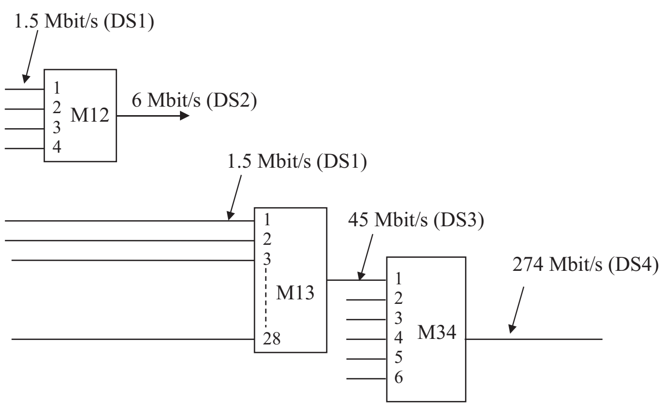


Figure 4.13 Plesiochronous digital hierarchy North American standard

Table 4.2 North American PDH standard

Nominal rate (Mbit/s)	Actual line rate (kbit/s)	Designation	Number of channels (64 or 56 kbit/s)
1.5	1,544	DS1, T1	24
6	6,132	DS2, T2	96
45	44,736	DS3, T3	672
274	274,176	DS4, T4	4,032

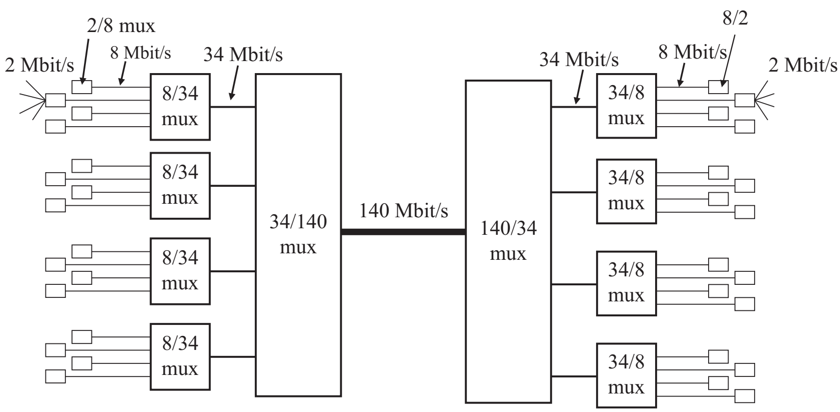


Figure 4.14 A 140 Mbit/s PDH line multiplexed system

transmission equipment is not shown). In order to extract a 2 Mbit/s digital block (e.g. a 30-channel module) from the 140 Mbit/s system, each stage of de-multiplexing is necessary in order to extract the stuffing bits from the 140 Mbit/s, 34 Mbit/s, and 8 Mbit/s frames, respectively. (This is analogous to extracting and discarding the cardboard packing sheets from each nested grocery box until the individual box of cornflakes is obtained.)

Not only does the multiplexer mountain of a PDH network incur many multiplexers, with the consequent cost and potential fault liability, but each item of equipment needs to be connected appropriately. Such connections are usually made manually at the time of setting up a transmission route, using coaxial jumper cables across a distribution frame in the Core Transmission Network station in an exchange or a standalone building, as described in Chapter 5.

4.3.4 SONET and synchronous digital hierarchy system

The PDH standard described above was defined at the start of the introduction of digital transmission networks during the 1970s. However, by the end of the 1980s most of the analogue transmission and switching equipment had been replaced by PDH digital and there was an increasing proportion of optical fibre in use.

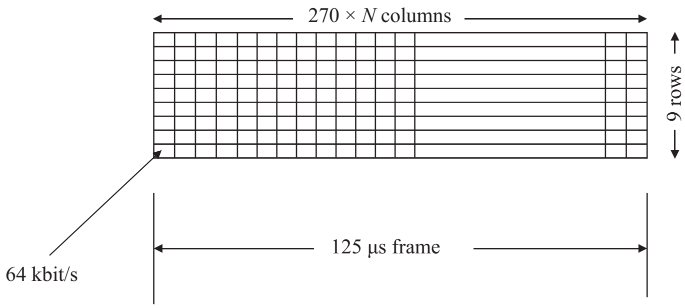
A new standard of digital transmission was defined, originally by American National Standards Institute (ANSI) in the United States (see Appendix 1), which was designed to overcome the multiplexer mountain and lack of flexibility of PDH and take advantage of the predominance of digital optical fibre working within the network and the advances made in very large-scale integrated circuits (VLSI) technology. Critically, the system is based upon optical rather than electrical interfaces between transmission and switching equipment in the network. The system is called SONET, derived from 'synchronous optical network'. Later the ITU-T defined a universal standard based upon SONET and extended to cover the European conditions called 'synchronous digital hierarchy' (SDH). The main features of SONET and SDH are:

- Direct access to tributaries through the add/drop multiplexing function (unlike the PDH multiplexer mountain);
- Standard optical interfaces to optical fibres carrying the aggregate signal;
- Compatible to both European and North American standards;
- Reduced operational costs compared to PDH because of the elimination of manual jumpering between the multiplexors;
- Network management features;
- End-to-end performance monitoring capability.

The overall effect of the above features is that SONET and SDH networks use less equipment than the PDH equivalent, with the consequent improvement in operational costs and quality due to the resulting lower fault rate. However, the biggest advantage is that the network management features allow the SONET-SDH transmission to be deployed as part of a coherent managed transmission network, as described in Chapter 5.

As the names SONET and SDH suggest, the system is based upon multiplexing synchronous tributaries, that is they are all operating exactly at one of a fixed set of standard bit rates [9]. This eliminates the need for the use of variable amounts of bit stuffing and the tributaries are time-division multiplexed by directly byte-interleaving onto an aggregate highway. The location of a required synchronous tributary within the frame can easily be found by counting the appropriate number of bytes. The system can also multiplex PDH tributaries without the need for bit stuffing by an ingenious use of pointers, which indicate where in the frame the start of the required PDH tributary is to be found.

The tributaries are packed into a basic container, which is known as the synchronous transport module (STM) in SDH or the optical carrier (OC) in SONET. Figure 4.15 illustrates the STM frame structure [10]. Although based on the 125 μs frame, the module actually stretches over nine such frames, creating a matrix of 270 columns and nine rows. Each element in the matrix is 64 kbit/s, that is the capacity of a single voice channel in the basic 2 Mbit/s block, and this is the smallest tributary that can be multiplexed into and extracted from the basic module. The overall rate of the basic STM is 155 Mbit/s. However, higher-speed STMs are created by increasing the number of rows that are contained within the 125 μs frame, each being a multiple 'N' of the basic STM – designated 'STM-N'.



STM–N = Synchronous Transport Module, N defined for 1, 4, 16 (8 and 12 for USA also)
Gross bit rate = N × 155,520 kbit/s

Figure 4.15 Basic STM-N frame structure

Table 4.3 Standard optical interfaces for SONET and SDH

Data rate (in Mbit/s)	**SONET optical carrier level (OC-N)**	**SDH synchronous transport module (STM-N)**
51.84 (52)	OC-1	–
155.52 (155)	OC-3	STM-1
622.08 (622)	OC-12	STM-4
1,244.16 (1.2 Gbit/s)	OC-24	–
2,488.32 (2.5 Gbit/s)	OC-48	STM-16
9,953.28 (10 Gbit/s)	OC-198	STM-64

The standard rates for the modules OC-N and STM-N for SONET and SDH, respectively, are shown in Table 4.3.

When packing the tributaries into the modules (i.e. OCs or STMs) other channels in the form of management information as well as the pointers, described above, are also included. This packing process takes place in two stages of multiplexing, as shown in Figure 4.16. In the first stage, tributaries are placed into a lower-order container which, together with an 'overhead' containing transmission-link network management information, is inserted into a virtual container (VC). Several of these VCs are then collected together to form a transmission unit (TU) using pointers to indicate their location. Several TUs form a group (TUG), which is the output of the first stage of multiplexing. In the second stage of multiplexing the TUGs are packed into higher-order containers, which together with an overhead carrying the network management information for the higher-order path – potentially comprising of several links – is inserted into the higher-order VC. Finally, several higher-order VCs may be grouped into an administrative unit (AU) which forms the

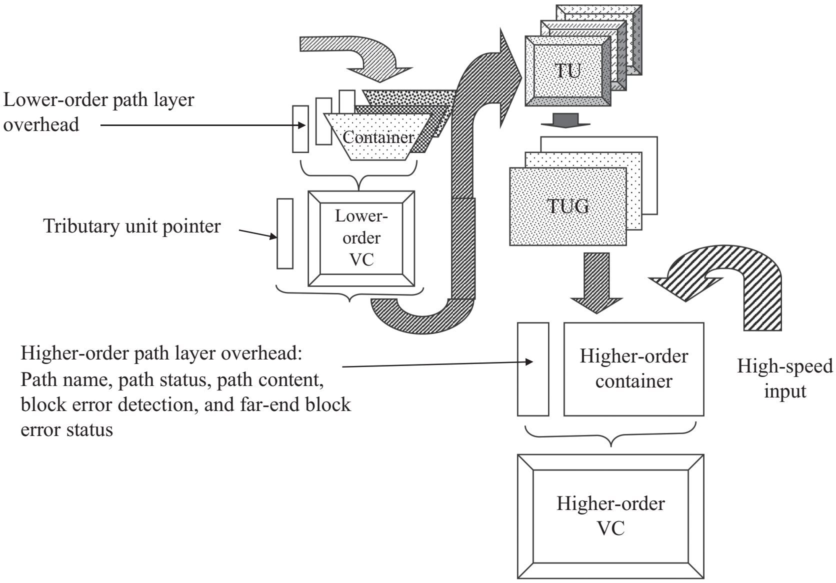


Figure 4.16 SDH packing mechanism

entire payload of the STM-1, again using pointers to indicate the location of the constituent higher-order VCs.

There are a variety of ways that the tributaries may be multiplexed up to the STM-N level, depending on the size of the tributaries and the configuration of the transmission links [10], and these are illustrated in Figure 4.17.

Whilst this multiplexing structure for SDH and SONET might appear complicated, the basic concept used is relatively straight forward. It can be simply explained by considering the synchronous transmission modules (STM) as containers on large transport lorries, which are trundling in a continuous stream down the (SDH/SONET) highways between warehouses (transmission network nodes). Figure 4.18 shows this simplified view of the concept with a lorry containing a single container (STM), in which two different freight companies each have a share – thus forming two AUs – separated by a net curtain inside the container. Within each AU are several large crates (transport unit groups) each containing several smaller crates (transport units). It is these transport-unit crates that contain the various sized parcels for delivery (i.e. virtual containers). The location of the parcels within the crates is indicated by information on written sheets (i.e. 'dockets' or 'inventory sheets'), corresponding to the pointers in the SDH multiplexing system. STMs-4, -16, and -48 are analogous to proportionately larger containers on the lorries or possibly lorries towing a series of container trailers.

As described above, one of the big advantages of equipment based on SONET or SDH is the ability to inject and extract a tributary from a multiplexed aggregate

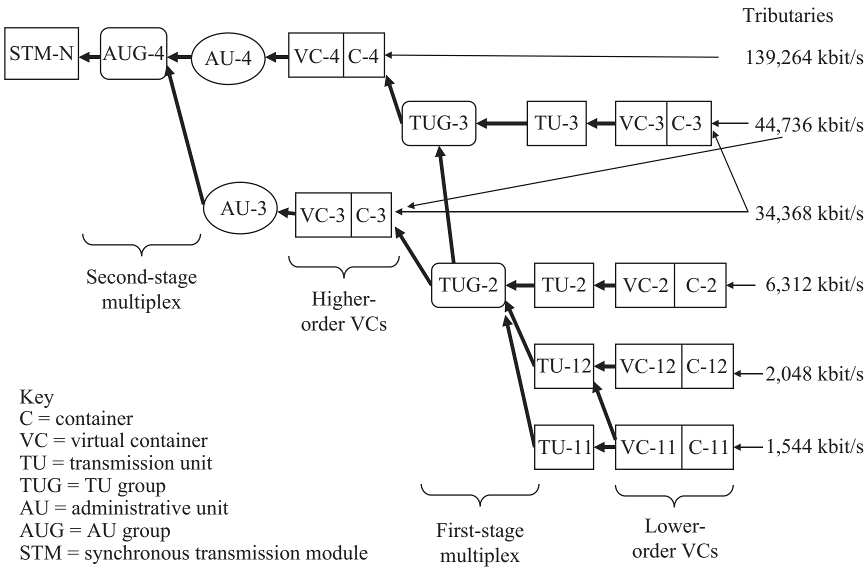


Figure 4.17 SDH multiplex structure

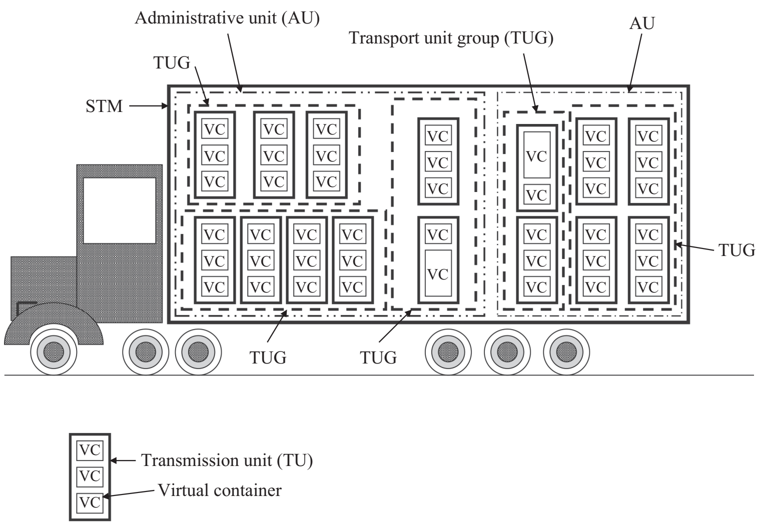


Figure 4.18 SDH/SONET concept

bearer using just one component. The functions of such a component, known as an ‘add-drop multiplexer’ (ADM) is shown in Figure 4.19, which lists the potential optical interfaces on the input and output ports to the optical fibre bearers, and the electrical interfaces to the set of tributaries.

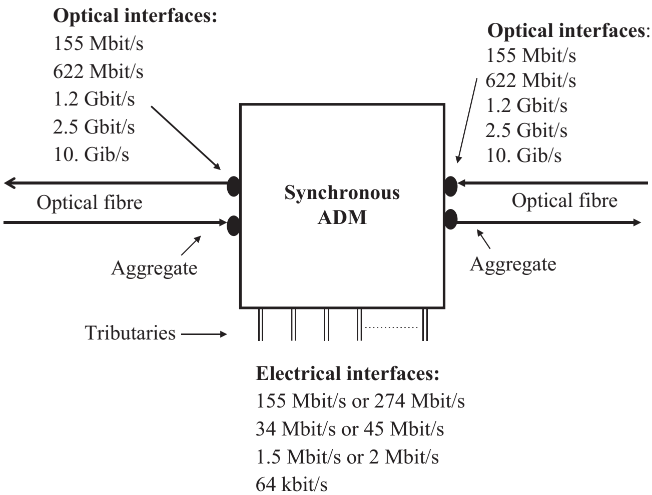


Figure 4.19 A synchronous add-drop multiplexor

The network aspects of a SONET or SDH transmission network are described in Chapter 5.

4.4 The range of transmission systems

In this section, we consider the range of transmission systems currently used in telecommunication networks. Each of these systems is carried over an appropriate medium with the required capacity provided by a direct or multiplexed payload, as described in Sections 4.2 and 4.3. The range of transmission systems is wide. Network operators need to choose which systems to deploy based on the costs, the extent of existing infrastructure, that is whether cables or ducts are already in place, the nature and capacity of transmission channels required, as well as future growth expectations, etc., as described in Chapter 5. In addition, of course, the technologies available for transmission systems are continually being improved, in terms of performance and costs, which can further complicate the choices for the network operator.

4.4.1 Metallic-line systems

Originally all transmission systems used in telecommunication networks were based on metallic – predominantly copper (but, also aluminium) – lines, carried as single pairs or bundled into cables, slung overhead between poles. These pairs carried just one telephone circuit as payload, either from the subscriber's premises (i.e. in the Access Network) or between exchanges providing 'trunk' and 'junction' routes. Today, single circuit payload copper pairs still constitute most of the subscriber lines in the PSTNs of the World. In addition, copper line systems – usually overhead – are still used in some networks, particularly in rural areas, between local exchanges and between a local exchange and its trunk exchange.

However, there has been a progressive use of, first analogue then digital, multiplexed payloads to provide extra capacity over the single copper pair. An example of the operational use of such extra capacity was the first generation of so-called 'pair gain' systems. This provided a two-channel analogue FDM system which enabled two subscriber's lines to carried over a single pair between the adjacent customer-premises and the serving local exchange [11]. This method of saving on the cost of copper pairs, which provides standard telephone service to the subscribers, is still used today by many network operators – although, the more recent deployments use digital pair-gain systems, based on DSL technology, described in Section 4.4.2.

The provision of multiplexed payloads over copper lines for customer-service reasons has until recently been directed at business subscribers, providing leased lines and multiple channels from their premises to the local exchange. Initially, FDM systems giving up to 12 channels (48 kHz) capacity were carried over the copper pairs. Since the 1980s, digital payloads (1.5 Mbit/s T1 in the United States and 2 Mbit/s E1 in Europe) have been deployed over copper pairs in the Access Network. Interference between pairs carrying such payloads in a single cable is minimised by using separate pairs for Go and Return directions. A further measure is the assembly of the pairs into 'balanced quads' within the cable sheath [4], such grouping reduces the mutual interference between pairs carrying the high-bit-rate signals. However, the distances the (quad) copper pairs can carry 2 Mbit/s payloads is restricted to just a few kilometres, and only a few pairs in each cable can use such signals because of the problems of interference. Thus, more robust copper systems (with longer reach and complete fills) such as transverse screen and coaxial cable are used to carry payloads up to 140 Mbit/s between the business subscriber premises and the serving exchange. To cope with long distances the transmission signal is boosted by digital repeaters located periodically along the length of the cable, housed in boxes submerged in the street – that is footway boxes – or mounted on the poles supporting the overhead or 'aerial' cable [12].

An extreme version of long-distance metallic cable transmission is that of submarine cable systems, used throughout the World to cross rivers, seas and oceans. The first generation of submarine cables used analogue FDM payloads using different sets of frequencies for the Go and Return direction of transmission over the single cable. These specialised systems are designed to withstand the rigours of being deployed under water over rough terrain (e.g. the sea bed) and generally being inaccessible for repairs. Apart from (shark resistant) armour plating on the cables the system's sub-sea analogue amplifiers use high levels of component redundancy to achieve the necessary reliability, since repairing amplifiers located on the ocean bed is a costly and difficult task! In addition to providing a low-loss transmission path for the payload, the submarine metallic cable also needs to carry electrical power from its two end stations (cable landing points) to all the amplifiers along its length [13,14]. Finally, of course, submarine transmission systems require specialist cable ships to lay the cables initially and later to recover and repair installed cables.

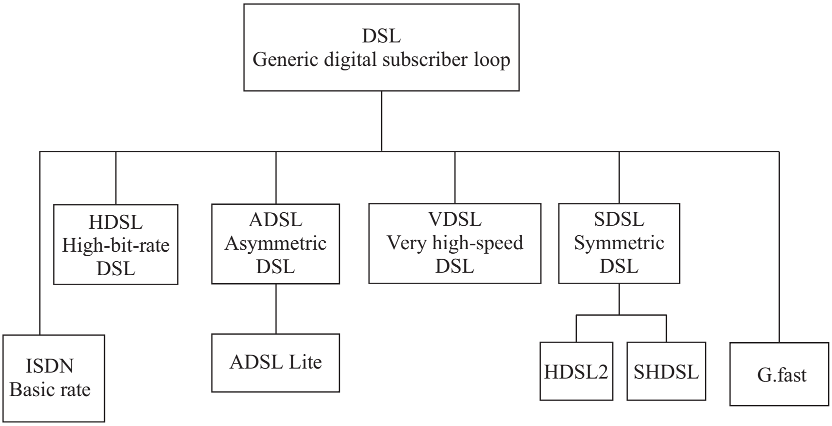


Figure 4.20 The DSL family

4.4.2 Digital subscriber line transmission systems

Advances in integrated circuit technology, particularly digital signal processors (DSPs), have enabled high-speed digital payloads to be carried over standard existing subscriber-line copper pairs originally provided for telephony only. There is a family of such DSL systems, known generically as ‘xDSL’, as shown in Figure 4.20. Each of the systems has the same basic architecture – that is digital send and receive transmission equipment located at each end of the copper local loop. However, there are important differences within the range. There are two main requirements for an xDSL system. The first requirement is to cope with the impairments introduced by the copper local network. The copper line is able to carry signals far in excess of those used for simple telephony – which are either constrained to 4 kHz by filters or, if unfiltered, extend to only 15 kHz – typically up to some 20 MHz over short distances. However, the attenuation introduced by the copper line rapidly increases with the higher frequencies. In addition, the use of HF signals over the copper Access Network causes interference between adjacent pairs within cables. The name for this interference is ‘crosstalk’, because of the way that speech from one pair is picked up and can be clearly heard by the listener on the other pair. The interference can occur at the near end to the transmitted signal – ‘near-end crosstalk’ (NEXT) or at the far end – ‘far-end crosstalk’ (FEXT). There are several techniques used by the xDSL systems to cope with the attenuation, as well as the FEXT and NEXT introduced by the copper lines.

The second requirement is for both Go and Return signals to be carried by the digital line system over the copper pair – that is ‘duplex’ working. Even though the copper pair can support both directions of transmission simultaneously, as described in Chapter 1, the xDSL electronic equipment at either end is inherently unidirectional because of its use of semiconductor technology. Generally, xDSL systems use a form of multiplexing to separate the Go and Return signals, for example, TDM (use of ‘ping pong’ alternate Go and Return bursts); FDM (use of separate frequency bands for the two directions); or use of separate Go and Return pairs (a form of space-division multiplexing, SDM). Other techniques, such as echo cancellation, are also used.

The first xDSL systems were associated with providing the so-called ‘basic rate’ ISDN over copper. These systems extend the IDN to the subscriber’s premises allowing digital data and telephony calls over the PSTN, as introduced in Chapter 3 and further described in Chapter 6. However, the xDSL family is now considered to refer mainly to those systems that enable ‘broadband’ access over the local loop. Such systems include (see Figure 4.20) the following [3,15,16].

4.4.2.1 Asymmetrical digital subscriber line system

This system was designed originally to provide ‘video on demand’ service over a copper pair, but is now widely deployed over many PSTNs in the world to provide broadband services (mainly high-speed Internet access) to consumers and small businesses, where the deployment of optical fibre to the premises is not warranted. ADSL uses a filter at the subscriber’s premises and at the exchange end of the local loop to extract the standard (4 kHz) telephony signal from the composite broadband signal; thus, the telephony service can be used simultaneously and separately from the broadband service. The latter comprises a medium speed upstream data channel and a high-speed downstream channel – hence the term ‘asymmetrical DSL’.

The system copes with crosstalk and other forms of interference by using a technique known as discrete multitone (DMT) modulation, or alternatively known as orthogonal frequency division multiplexing (OFDM) [17]. This approach conveys the data signal over a large set of individual carriers, each at a precise frequency in the available range – for ADSL the frequency separation is 4.3 kHz. At the receiving end of the link the composite waveform is processed so that each modulated carrier is sampled at the appropriate frequency. The use of OFDM enables the maximum possible rate of data through-put despite the limitations of the line due to crosstalk and interference across the whole range of frequencies used. This results in ADSL systems having variable data rates which continuously differ in real time from the nominal advertised rate, as conditions on the line vary. Therefore, DSL systems require a short period of automatic measurement at the DSLAM to set the optimum transmission rate for the conditions on the line. This action, known as ‘train up’, occurs when initialised or whenever there is a significant change to the induced noise or crosstalk on the line.

Duplex working – that is transmission in both directions over the single copper pair – is achieved using separate bands of frequencies for up and down stream, a technique known as frequency division duplex (FDD). Figure 4.21 shows the spectral allocation of ADSL in the United Kingdom [15], indicating the frequency spreads of the low-band splitter filter and the high-band data extraction filter. With perfect conditions, that is in the absence of interference, data rates of up to some 2 Mbit/s can be carried over about 5 km of copper line, with up to 8 Mbit/s over lengths of about 2.7 km. These figures will improve with progressive upgrades to the ADSL equipment, but they are in any case highly dependent on the

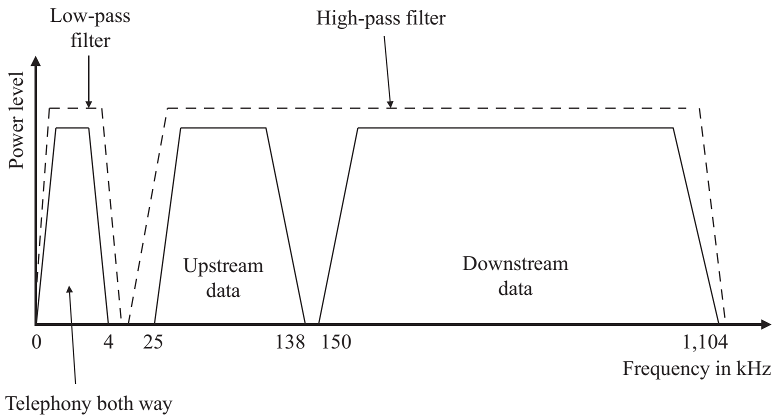


Figure 4.21 Spectral allocation for ADSL

quality of the local loop line plant within the PSTN. More details, including a block-schematic diagram showing the architecture of an ADSL line, is given in Chapter 5 and Figure 5.4.

4.4.2.2 High-bit-rate digital subscriber line system

The high-bit-rate digital subscriber line (HDSL) provides high-speed data transmission symmetrically over two or three pairs of copper cable. Rates of 2 Mbit/s symmetrical can be carried over three pairs up to distances of 4.6 km [3]. Network operators typically use HDSL to provide the local ends of digital private circuits (leased lines) for business customers, where optical fibre local ends cannot be justified or in advance of later deployment of optical fibre.

4.4.2.3 Symmetric digital subscriber line system

Symmetric digital subscriber line (SDSL) is a derivative of ADSL, but with equal (symmetric) transmission capacity upstream and downstream. This means that the NEXT interference dominates at each end and consequently the achievable speeds are much lower than for an ADSL system over a single pair in the same network.

4.4.2.4 Very high-bit-rate digital subscriber line system

The very high-bit-rate digital subscriber line (VDSL) is a derivative of ADSL which provides significantly higher data rates by restricting the use of copper line used for the broadband signal to just a few hundreds of metres. This requires the deployment of optical fibre from the exchange building to the VDSL DSLAM equipment housed in street cabinets – typically physically adjacent to the primary connection points of the Access Network (see Figure 2.8 in Chapter 2 and Chapter 5). This hybrid optical fibre-copper system provides data rates up to some 52 Mbit/s over 300 m of copper pair or some 12.96 Mbit/s over 1,350 m of copper line [16,18]. The spectrum of the VDSL signal over the copper pair goes as high as 30 MHz.

4.4.2.5 ADSL2 and VDSL2

The performance of high-speed digital signals over bundled pairs of copper wires, as in Access Network cables, is hindered by two main impairments, namely: crosstalk and power loss. Unfortunately, just increasing the signal power to compensate for the attenuation due to increasing copper length and signal frequency worsens the effect of crosstalk. Therefore, developments of the xDSL family have focussed on addressing these two impediments so as to increase the speed of broadband transmission. Both ADSL and VDSL have been enhanced – ADSL2 and VDSL2 – using the following techniques.

(a) Vectoring. Crosstalk in a copper pair is the induced electrical interference from adjacent pairs in the vicinity, particularly in the case of multi-pair cables. The technique of vectoring relies on being able to generate an inverse signal at the sending end which is applied at the receiving end to cancel the induced voltages due to far-end crosstalk. (Chapter 11 covers the phenomenon of crosstalk in more detail.) This requires a 'vectoring engine' at the DSLAM which computes the appropriate vectoring signal for all the pairs in a bundle terminating on that DSLAM. The effect of vectoring achieves the same bandwidth on a pair of copper wires in a bundled cable as on a single cable – which gives a significant improvement in achieved broadband performance in practical copper-pair networks [19].

(b) Forward error correction (FEC). In addition to crosstalk, induced noise along the line can causes errors to the received digital signal. The effect becomes more pronounced as the speed of transmission increases since the receiver has proportionally less time to detect whether a '1'or '0' was sent. One way of keeping the error rate acceptable is to employ FEC, in which the data to be sent is grouped into blocks which are treated with a suitable error-correcting code (e.g. Reed–Solomon or Hamming codes). At the subscriber's end, any received blocks with errors are detected and, if within limits, automatically corrected. Whilst this technique allows for higher transmission rates over noisy lines, it does introduce some additional delay and an overhead of additional bits (even if there are no errors incurred on the line) which itself reduces the potential gain in speed. An alternative used in some systems is to rely on retransmission of any errored blocks – which saves on the overhead (no correction code bits required) and the repeated content is carried only when errors are incurred [20].

4.4.2.6 G.fast

The latest development in the DSL family is known as G.fast. This system is based on taking the DSLAM very much closer to the subscriber's premises so that the length of copper wires is 500 m or less. In practice, the length of copper pair can be about 20–30 m. This means that transmission is substantially over optical fibre with just a short tail of copper pair delivering the broadband signal over the last few metres to the subscriber's premises. Broadband speeds up to about 1 Gigabit/s are therefore possible – perhaps justifying the name 'G.fast?' Actually, the name is an acronym of 'fast access to subscriber terminals' and the ‘G’ denotes the series of ITU-T Recommendations (G.9700 and G9701) specifying the system.

The specification has several variants, but the key features are as follows [21]:

- DMT is used to modulate data onto the carrier (as in ADSL and VDSL).
- A wider set of frequencies is used over the copper pair than in the rest of the DSL family. The spectrum of G.fast extends up to 108 MHz, which necessitates the use of special waveform shaping to avoid interference with FM radio broadcasts.
- Duplex working is achieved using TDD. There are several ratios of duplex working, ranging from 50:50 symmetry to 90:10.
- FEC is used to minimise effect of impulse noise.
- An improved form of vectoring is used to eliminate crosstalk.

4.4.3 Point-to-point optical fibre

Optical fibre systems offer the network operator the highest bandwidths of all transmission systems – potentially up into the Tbit/s (1,000,000,000,000 bits/s) range. They were initially introduced as replacements for the coaxial metallic cable systems carrying PDH payloads in the trunk network, but they are now used extensively throughout the national Trunk and International networks and increasingly in the Access Networks, carrying SDH/SONET payloads.

A generic picture showing the basic architecture of an optical fibre point-to-point digital transmission system is given in Figure 4.3. The carrier modulation system uses a LED or laser diode to pulse a single colour light beam at the send end and Avalanche photo diodes or PIN diodes are used to detect the light pulses at the receive end, as described earlier in this chapter. The operating characteristics of the send and receive transducers and the ‘windows’ in the attenuation curves of the single mode optical fibre now used in transmission systems – as shown in Figure 4.5 – give a wide range of potential wavelengths (colours) of light, centred on 850 nm, 1,300 nm, 1,500 nm. (Note: a nanometre, nm, is 1 billionth of a metre.) Special line codes, for example, ‘code-mark inversion’ are used in conjunction with the payload coding (e.g. scrambling) to ensure that there are sufficient on-off transitions in the pulsed light sent down the optical fibre for successful timing recovery by the regenerators along the line [14]. Actually, the very low rate of attenuation of modern optical fibre systems means that the regenerators can be spaced at least 50 km apart – a figure that is continually improving with equipment development. This means that on many optical fibre transmission links within medium-or-small-sized countries, such as the United Kingdom, no regenerators are required along the route, saving equipment, accommodation and operational costs.

Normally, separate optical fibres within a single cable are used to provide the Go and Return directions of transmission. The alternative configuration – that is the two directions of transmission carried as separate frequency bands (FDM) over a single optical fibre incurs extra electronic equipment costs and so is used only where the number of optical fibres in a single cable is unduly restricted.

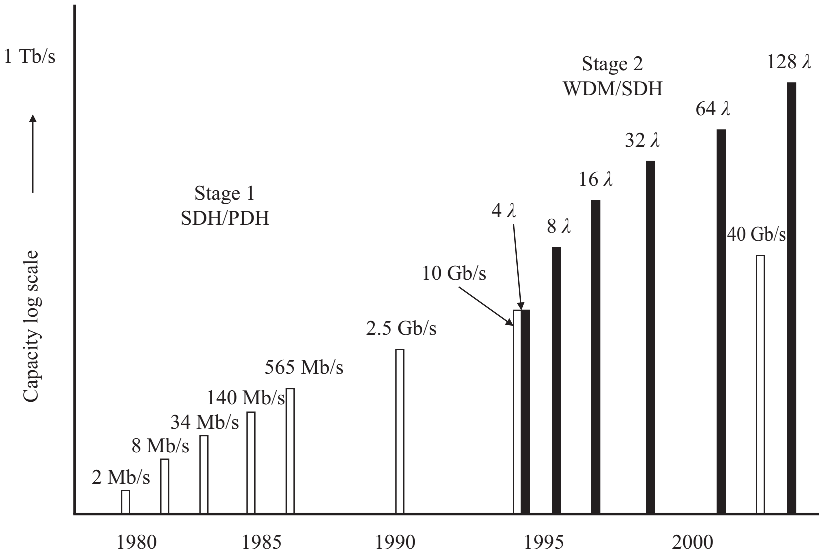


Figure 4.22 Transmission capacity per fibre

4.4.4 Dense wave-division multiplex system

Network operators have progressively introduced to their transmission networks digital line systems of increasing speed, to gain improvements in the cost-per-channel carried. Of course, the full economy of scale on transmission systems is only achieved if the transmission systems are fully loaded with traffic. Since the early 1980s the demand for transmission capacity has generally grown in line with the development of ever increasing line speeds. For example, between 1980 and 1990 the development of digital transmission systems over optical fibre has increased from 2 Mbit/s to 2.5 Gbit/s. However, this progression is based on increasing the speed of optical transmitters, receivers, and the repeater systems, all operating on a single wavelength of light over the optical fibre. Since the mid-1990s development work focused on the use of several different wavelengths of light – that is colours – as a way of increasing capacity on line systems beyond the 10 Gbit/s achieved on one wavelength in 1995. These systems, which are referred to as 'dense wave-division multiplexed' (DWDM), require very precise optical filters to separate the component colours. Figure 4.22 presents a historical view of the development of transmission capacity up to 10 Gbit/s over a single wavelength of light over an optical fibre, and the trend for increasingly higher numbers of wavelengths within DWDM systems.

Figure 4.23 illustrates the general arrangement for a DWDM system with *n* wavelengths. At the sending end *n* lasers are required, each generating a different colour (i.e. wavelength) of light. The optical fibre carries the composite DWDM signal containing the *n* wavelengths. At the receiving end of the system a filter is

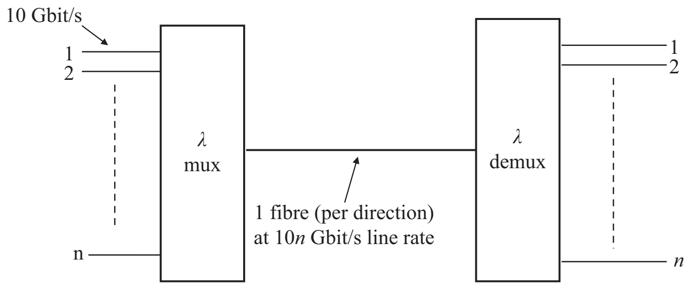


Figure 4.23 Dense wave-division multiplexing

required to separate the n wavelengths and these are passed to n individual photodiodes to detect the set of sent signals. For the 16-wavelength system described for TAT-14 above the potential capacity is 16×10 Gbit/s, i.e. 160 Gbit/s – which is the equivalent capacity of some 2.4 million voice channels!

4.4.5 Passive optical fibre network

So far, we have considered only point-to-point optical fibre transmission systems. Whilst this configuration is appropriate for delivering large capacities between network nodes, or to those business customer sites requiring many broadband and telephony circuits, it is not economical for delivering small payloads. Network operators have been seeking a cost-efficient method of deploying optical fibre systems in the Access Network to serve broadband services (e.g. Internet access) to residential and small business customer sites, as described in Chapter 5. There are two main cost factors that need to be addressed: the cost of deploying the many thousands of optical fibres from a local exchange to each subscriber premises, and the terminating equipment at each end needed to extract/insert the signal to/from each fibre. Passive optical fibre network (PON) systems have been developed to address these factors.

The basic premise of the PON system is that the cost of fibre and electronics is minimised by using a tree and branch configuration, whereby a single fibre from the local exchange splits several times in succession in order to serve many subscriber premises. The splitting is achieved by fusing bundles of optical fibres so that the light energy splits equally into each tributary. Since this is done along the fibre route without the use of electronics, it is known as a 'passive' system. The tree and branch configuration also minimises the cost of electronics since only one device per fibre spine (or 'tree trunk') is needed at the exchange end. Figure 4.24 shows the basic concept of a PON system. The optical fibre splitters are typically housed in footway boxes beneath the street. In the example shown, the first splitter is four-way, and the second set of splitters is eight-way; thus, 32 subscriber premises are served by a single optical fibre spine from the exchange.

At the exchange end, optical line termination (OLT) equipment associated with each optical fibre spine assembles the Go transmission for the 32 channels into

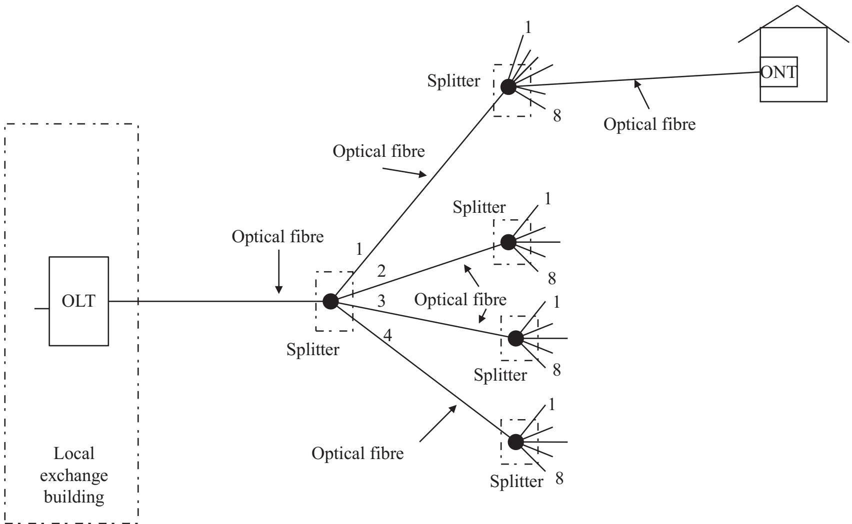


Figure 4.24 Passive optical fibre network

a time-division multiplexed signal that is broadcast over the PON fibre tree structure. Thus, at each subscriber premises a terminal unit, usually called an optical network termination (ONT) extracts the appropriate time slot from the received TDM broadcast signal. For the Return direction of transmission, each ONT is allocated a fixed time slot during which a burst of the subscriber's digital signal is sent. In effect, this is a dispersed version of a TDM multiplexed system, similar to that described in Chapter 3, but with each of the 32 tributaries located in different subscriber premises. An important variant on this configuration is the amalgamation of a set of subscriber ONTs in a single unit, known as an optical network unit (ONU), which is located at a business customer's premises where it serves the many terminations via internal wiring. With the business configuration of PON there is either just one or no stages of splitting in the network, with the necessary TDM channel extraction/injection performed by the ONU.

The PON was first developed in the BT Laboratories and trialled in 1987, with the first application designed for telephony subscribers – known as telephony over passive optical network (TPON). Since then there have been a series of PON developments defined by the full-service access network (FSAN) consortium and the ITU-T; more recently in the United States the IEEE 802.3ah standard has been defined. There are several types of PON, as follows [22,23].

APON: This broadband version uses ATM (asynchronous transfer mode – see Chapter 8) over the PON system to provide broadband service. The third-generation system provides a total capacity of 622 Mbit/s symmetrical for the maximum 32-split PON system.

BPON: Broadband PON is an expanded version of APON designed to carry video. Use is made of DWDM to add a separate wavelength for the video services

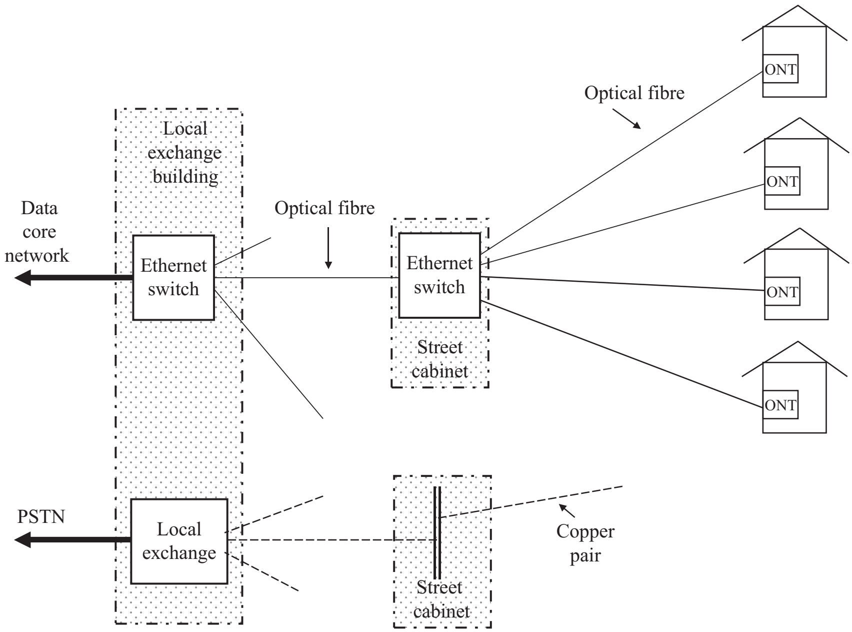


Figure 4.25 Ethernet fibre to the premises (home)

above the non-video traffic on the PON. For example, a BPON system may carry voice, data, and bidirectional video for 32 customers over a single fibre spine. Two wavelengths are used to carry voice and data at 622 Mbit/s downstream and 155 Mbit/s upstream, and a third wavelength is used to carry video up to 1 Gbit/s bidirectionally.

EPON: Ethernet PON offers the LAN packet-based protocol of Ethernet as an interface to the data networks on business customers' premises.

GPON: Gigabit/s PON is the latest high-speed PON design, offering up to 2.5 Gbit/s symmetrically.

4.4.6 Ethernet fibre to the premises

An alternative means of providing broadband delivery over optical fibre to residential or business customers' premises (i.e. FTTH, FTTP, FTTB) is through using Ethernet as the bearer system. This has the advantage of providing an exclusive fibre to each customer's premises from an Ethernet switch, which can deliver the full system bandwidth. As the schematic diagram of Figure 4.25 shows, the number of fibres carried through the Access duct network is kept to a manageable level by siting Ethernet switches in street cabinets (usually next to the copper-cable cabinets). This means that there is contention of the broadband capacity (see Chapter 8), which will restrict customers' data rates during busy periods. It should be noted that the telephony service to the customers continues to be delivered over the existing copper distribution network.

4.4.7 Dark fibre

The principle of sending a modulated light beam down optical fibre cables gave rise to the term ‘dark fibre’, which originally referred to unused fibres in an operator’s cable, that is spare capacity. However, the increasing desire for businesses to build their own data corporate networks has given rise to the now more usual meaning of the term as the leasing of unused fibres in a network operator’s cable. These can then be used by the business customers to construct their own data networks, say between two campus sites. Since the dark fibre is essentially raw and transparent, customers are free to use their own broadband transmission systems running any form of protocol. The network operators are, therefore, responsible only for the physical upkeep of the fibre.

Whilst network operators generally have several spare fibres in their cables, they may not always be able to meet customers’ demand. Also, there is an understandable reluctance by the operators to release their prime infrastructure – since their business models and mode of operations are designed to provide communication services. One approach to addressing both issues is for the operators to lease individual wavelengths within a DWDM system on optical fibres rather than the fibres themselves. Again, the leased wavelength can be used by the customers to carry any protocol data transmission stream generated by their private network equipment [24].

4.4.8 Submarine cable systems

One of the biggest challenges facing network operators in planning the deployment of cable systems is crossing over large expanses of water within the country or international links to other countries. If a bridge exists across the waterway the first preference is to make use of a cable duct way slung within the bridge. In the case of large waterways or seas, submarine cable systems are needed. These were originally introduced to carry telegraph signals in the late 1890s and the first telephone submarine cables were laid in the 1920s. A big step forward was achieved with the first trans-Atlantic telephone system, TAT-1, opened in 1956 based on a single coaxial cable. As is now normal practice, international submarine cables are owned and managed by a consortium of operators and service providers. In 1986 the first international optical fibre submarine cable system was laid between the United Kingdom and Belgium, and 2 years later the TAT-8, the first trans-Atlantic optical fibre system, was laid [25]. A network of submarine cables now criss-crosses the oceans, seas and channels linking islands and continents, carrying the vast majority of the World’s inter-landmass voice and data traffic. (Only a small proportion of such traffic is carried over radio or satellite systems.)

Figure 4.26 shows a simplified view of a typical multifibre submarine cable system. There are several important differences between submarine and terrestrial communication systems. The first is that once laid under the sea it is extremely difficult and expensive to gain access to the submerged cable system for repairs or upgrades. Therefore, the submerged parts of the system are designed to have an

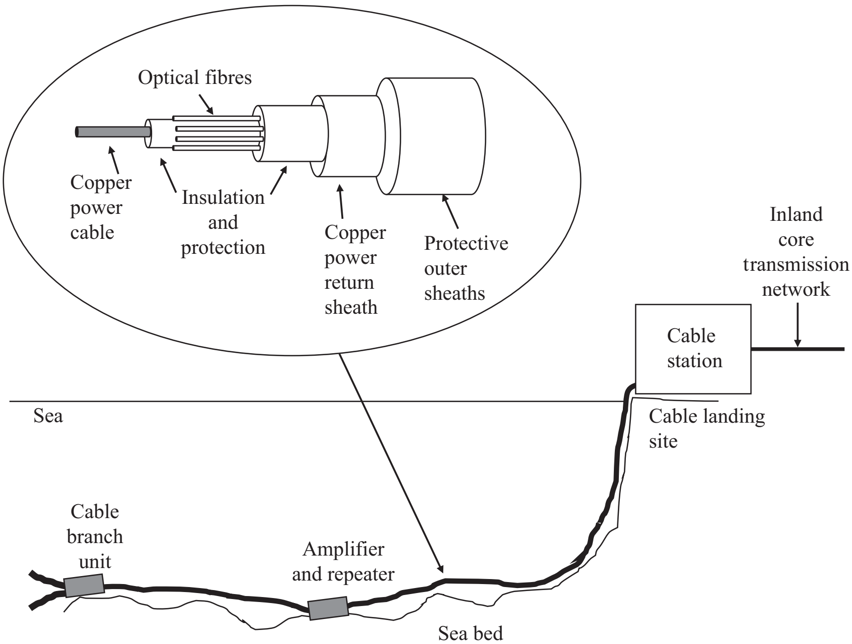


Figure 4.26 Submarine optical fibre cable system

operational life of 25 years. The second main difference is the harsh environment; the cable system and submerged electronics are designed for working at up to 8,000 m depth, experiencing some 800 atmospheres of pressure [26]. Thirdly, the underwater electronics in the optical amplifiers and repeaters need electric power, which means feeding electricity along the cable from both ends. The power is fed down the central copper cable with the return path provided by a copper sheath, as shown in Figure 4.26.

The submerged amplifier/repeaters are spaced about 50 km apart. Branching units are also used to give multiple accesses from landing sites in neighbouring countries to long-distance submarine cables (e.g. cables from the Netherlands, the United Kingdom, and France joining a trans-Atlantic cable at a branch in the English Channel). Finally, the outer layers of the cable are often armoured – to cope with shark attacks and fishing trawlers, etc., and contain various layers of protection wrappings against marine burrowing creatures. At the landing site, a cable station comprises of the submarine transmission termination, power-feeding termination, and a network-management centre for operations and maintenance of the system. The linkage to the terrestrial network is via the Core Transmission Network as shown in Figure 5.12 of Chapter 5.

The potential capacity of each new system laid continues to increase each year. For example, the trans-Atlantic telecommunications (TATs) optical fibre cable system laid in 2001, known as 'TAT-14', is capable of carrying 1,024 STM-1's across the Ocean (compared to TAT-1 which opened in 1956 with 36 telephone circuits). The system achieves this huge capacity by carrying payloads of STM-64 (10 Gbit/s) on each of 16 different wavelengths of light over the fibre pair using DWDM, as described in Section 4.4.4. It is configured as a pair of optical fibres (Go and Return) looping from Manasquan in New Jersey, the United States, to Denmark, then Holland, France, and the SW tip of the United Kingdom (Bude-Haven) and back to Tuckerton in New Jersey, the United States, totalling some 15,800 km. Transmission over current submarine cables is based on DWDM at 10 Gbit/s, 40 Gbit/s, and 100 Gbit/s, centred on wavelengths in the 1,552 nm range.

4.4.9 Line-of-sight microwave radio systems

The basic configuration of a line-of-sight (LOS) microwave radio system is shown in Figure 4.8(a). The key components are narrow-beam antennas, which can operate as precision point-to-point LOS systems, allowing reuse of the same carrier frequencies around the country without mutual interference. Frequency bands allocated by the Regulator and the ITI-R for this use by network operators reside at 4 GHz, 6 GHz, and 11 GHz – these are mainly used on systems providing trunk links between exchanges. In addition, frequency bands around 19 GHz, 23 GHz, and 29 GHz are also available, and are mainly used for shorter distance trunk links and subscriber access [27] and higher frequency bands are periodically being made available. In general, the size of the required antenna reduces with increasing carrier frequency, but the allowable spacing between antennas also reduces and the radio transmission become more susceptible to water absorption, that is rain and clouds.

The LOS microwave radio systems were initially introduced into operators' networks in the 1960s using analogue transmission, as an economical solution for the conveyance of TV programmes between the various studios, TV switching centres and the TV broadcast mast antennas around the country. Later, FDM payloads of up to 3,600 telephony channels were carried using FM over these radio systems between trunk exchanges.

Since the 1980s the microwave radio systems have been designed for digital transmission, initially carrying the standard PDH payloads of 34 Mbit/s and 140 Mbit/s (and multiples of 45 Mbit/s in the United States). With the advent of SDH/SONET, the microwave systems carry the standard 155 Mbit/s STM-1 digital block.

An alternative configuration for LOS microwave radio systems is point to multipoint, in which an omnidirectional central antenna radiates uniformly to any antenna/receiver in the vicinity. Such systems can be attractive in providing subscriber access links. Several broadband multipoint microwave radio systems have been developed to support the delivery of data services to business customer premises. One such example from the United States is the local multipoint distribution service (LMDS), which operates in the 28 GHz range and offers capacities up to 155 Mbit/s per customer. The deployment of point-to-point and multipoint microwave radio systems is covered further in Chapter 5.

4.4.10 Earth satellite systems

The established system of telecommunication global satellites uses a configuration of three satellites equally spaced on an equatorial plane at a distance of 36,000 km above the Earth. At this height the satellites spin around the Earth at exactly the same rate as the rotational speed of the Earth, and they therefore appear stationary to antennas on the Earth's surface [28,29]. Whilst this positioning enables all areas (except the far North and South) of the World to communicate with at least one of the satellites, the high orbit does incur a transmission propagation time for the microwave radio signal of some 250 ms, even travelling at the speed of light. This means a round trip delay of about half a second. Such a delay is noticeable to two people speaking over a satellite link, resulting in some difficulty in maintaining a natural conversation. Consequently, telephone call connections are never provided over more than one satellite hop; the remaining distance is always provided over terrestrial links with their lower delay.

However, the delay issues with geostationary satellites is not a serious problem for one-way communication, such as broadcast TV (e.g. Sky) or data communication. More recently, two new satellite configurations have been introduced: medium Earth-orbit (MEO) and low Earth-orbit (LEO) systems, both of which incur less transmission distance and hence delay [29]. MEO and LEO systems comprise of a constellation of satellites set in an elliptical orbit around the World, designed to give coverage to specific areas only of the Earth's surface. Since the satellites are not in a geostationary orbit they appear to move continuously across the sky; thus, the Earth-station antennas need to be able to track and move to receive the satellite's signal. Once a satellite has moved to the limit of visibility the Earth station antenna needs to switch to a following satellite, which should have just come into view. Typically, each satellite in a LEO system is visible for about 10 min, orbiting about every 120 min at some 780 km above the Earth. These LEO systems tend to be low capacity but can be tailored to give good coverage of polar regions and incur only 10 ms propagation delay. MEO systems use orbits around 10,000 km and incur some 70–80 ms propagation delay.

Unlike the three satellites necessary for full coverage with a geostationary system, MEO systems require some 12 satellites, whilst LEO systems may require up to 200 satellites. The complexity of the antenna steering control, the overall satellite system control and the large number of satellites required means that LEO and MEO systems are not currently economical for large-scale commercial application – although they are used for specialist applications, such as communications in the remote north of Russia and Canada.

4.4.11 Wireless local area networks

There is a family of short distance microwave radio systems designed for conveyance of data services between a user's terminal – for example, a laptop computer – which is normally static and some serving node in the network. Since the configuration is really an extension of the local area network (LAN) concept, originally used in offices but now also deployed in residences, the systems are known as ‘wireless LANs’. These systems have been specified in the United States within the ‘IEEE 802.11’ range, and have subsequently been widely adopted World-wide. They are also popularly referred to as ‘Wi-Fi’. These systems also use the OFDM techniques described earlier for ADSL to cope with the multipath distortions to the radio signal (see Chapter 9). The transmission network aspects of wireless LANs and their data network aspects are covered in Chapters 5 and 8, respectively.

4.4.12 Wireless metropolitan area networks

A new generation of LOS microwave access systems has been defined which provide broadband data links between LANs or between a LAN and a central server, creating so-called wireless metropolitan area networks (MAN). The system, defined by the IEEE 802.16 standard, is popularly known as ‘WiMax’ [30,31]. It uses 256- or 2048-carrier frequency OFDM systems to provide maximum data throughput over the multipath conditions experienced with radio links. However, the standard allows many variations in the way that manufacturer’s equipment is designed, thus giving a range of performance and cost options for users. Although WiMax is specified for use in the 10–66 GHz range of the radio spectrum, initial deployments use the unlicensed band of 5.725–5.850 GHz, and in the licensed bands of 2.5–2.69 GHz and 3.4–5.80 GHz. The data throughputs possible range from about 3–7 Mbit/s over a 5 MHz wide radio band; however, this capacity is shared among all the users in the WiMax cell. The use of WiMax to provide broadband access is considered in Chapter 5.

4.5 Summary

In this chapter, we looked at the all-important subject of transmission systems as used in telecommunications. In Section 4.2 we began with the basis mechanism of transmission and then reviewed the physical transmission media: copper cable, coaxial cable, optical fibre and radio. Section 4.3 addressed the various ways of building a digital multiplexed payload for carrying over the transmission media, covering:

- PCM and the basic 2 Mbit/s or 1.5 Mbit/s digital block – the basic building block of digital networks.
- PDH, the original digital transmission multiplex payload system;
- SONET and SDH, the synchronous digital payload system that facilitates a managed transmission network, as well as eliminating the multiplexer mountain problems of PDH.

  The wide range of transmission systems currently in use by network operators were then briefly reviewed in Section 4.4, including:
- Metallic cable
- The family of xDSL (i.e. ISDN, ADSL, VDSL, HDSL, and SDSL) and G.fast systems

- Optical fibre systems, primarily point-to-point (PDH, SDH, and Ethernet)
- Passive optical network systems, for cost-effective access applications
- DWDM, which by using many colours of light is providing ever increasing capacity over optical fibre
- Submarine cables
- Microwave radio systems
- Earth satellite systems (geostationary, LEO, and MEO)
- Wireless LANs and wireless MANs.

The deployment of digital multiplexed payloads over radio, metallic line or optical fibre systems in the Access, Core and International transmission networks is covered in the next chapter.

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