4

Transmission

Transmission is the process of transporting information between end points of a system or network. As we have seen in previous chapters, the end-to-end communication distance is often very long and there are many electrical systems on the line. These systems, network elements such as exchanges, are connected to the other elements with connections provided by the transmission systems. In this chapter we discuss the basic restrictions and requirements for transmission and the characteristics of various transmission media and equipment used in the telecommunications core network. The transmission systems for access networks for high-data-rate customer access to the Internet are discussed in Chapter 6.

4.1 Basic Concept of a Transmission System

In this first section we look at the basic elements present in all transmission systems. We introduce the basic functions of these elements and discuss their roles for the successful transmission of information.

4.1.1 Elements of a Transmission System

The main elements of a communication system are shown in Figure 4.1. The transducers, such as a microphone or a TV camera, that we need to convert an original signal to an electrical form are omitted; unwanted disturbances such as electromagnetic interference and noise are included. Note that

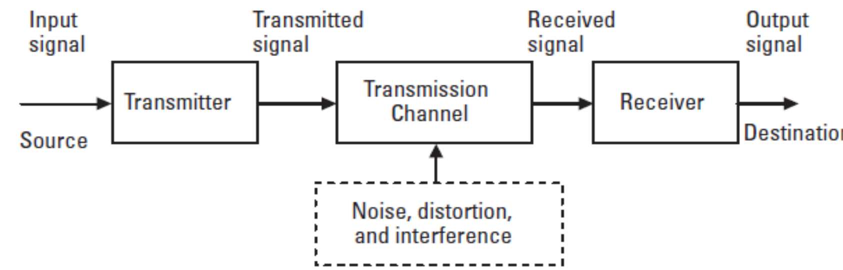


Figure 4.1 Basic concept of transmission system.

bidirectional communication requires another system for simultaneous transmission in the opposite direction.

4.1.1.1 Transmitter

The transmitter processes the input signal and produces a transmitted signal suitable to the characteristics of a transmission channel. The signal processing for transmission often involves encoding and modulation. In the case of optical transmission, the conversion from an electrical signal format to an optical one is carried out in the transmitter.

4.1.1.2 Transmission Channel

The transmission channel is an electrical medium that bridges the distance from the source to the destination. It may be a pair of wires, a coaxial cable, a radio path, or an optical fiber. Every channel introduces some amount of transmission loss or attenuation and, therefore, the transmitted power progressively decreases with increasing distance. The signal is also distorted in the transmission channel because of different attenuation at different frequencies. Signals usually contain components at many frequencies and if some are attenuated and some are not, the shape of the signal changes. This change is known as *distortion*. Note that a transmission channel often includes many speech or data channels that are multiplexed into the same cable pair or fiber. The principle of multiplexing is explained later in this chapter.

4.1.1.3 Receiver

The receiver operates on an output signal from the channel in preparation for delivery to the transducer at the destination. Receiver operations include filtering to take away out-of-band noise, amplification to compensate for transmission loss, equalizing to compensate for distortion (different attenuation of frequency components), and demodulation and decoding to reverse the signal processing performed at the transmitter.

4.1.1.4 Noise, Distortion, and Interference

Various unwanted factors impact the transmission of a signal. Attenuation is undesirable because it reduces signal strength at the receiver. Even more serious problems are distortion, interference, and noise, the last of which appears as alterations of the signal shape. To decrease the influence of noise, the receiver always includes a filter that passes through only the frequency band of message frequencies and disables the propagation of out-of-band noise.

4.1.2 Signals and Spectra

Electrical communication signals are time-varying quantities such as voltage or current. Although a signal physically exists in the *time domain*, we can also represent it in the *frequency domain* where we view the signal as consisting of sinusoidal components at various frequencies. This frequency-domain description is called the *spectrum*.

Any physical signal can be expressed in both domains. In the time domain we draw the amplitude along the time axis and in the frequency domain we draw the amplitude (and phase) along the frequency axis. Although both of them give a perfect description of the signal, both presentations are needed for easier understanding of the different phenomena. The time-domain signal is the sum of the spectral sinusoidal components. Fourier analysis gives the mathematical connection between the time- and frequency-domain descriptions. Here we merely introduce the connection between the time-and frequency-domain descriptions with a couple of examples. The reader is referred to [1] for mathematical treatment of the transformation between the time and frequency domains.

In Figure 4.2, two examples of time-domain signals and corresponding spectrums are presented. In the first example we see an ordinary rectangular digital pulse with duration of T seconds and its corresponding spectrum. If, for example, the pulse a duration $T = 1$ ms, the strongest spectral content lies below 1 kHz ($1/T = 1/1$ ms = 1,000 1/s = 1 kHz), as shown in Figure 4.2. From this we get a thumb rule that we can send 1,000 pulses of this kind in a second through a channel with a bandwidth of 1 kHz, which corresponds to a 1-Kbps binary data rate.

To increase the data rate, we should decrease T and the spectral width and the required bandwidth is increased correspondingly. For example, for a

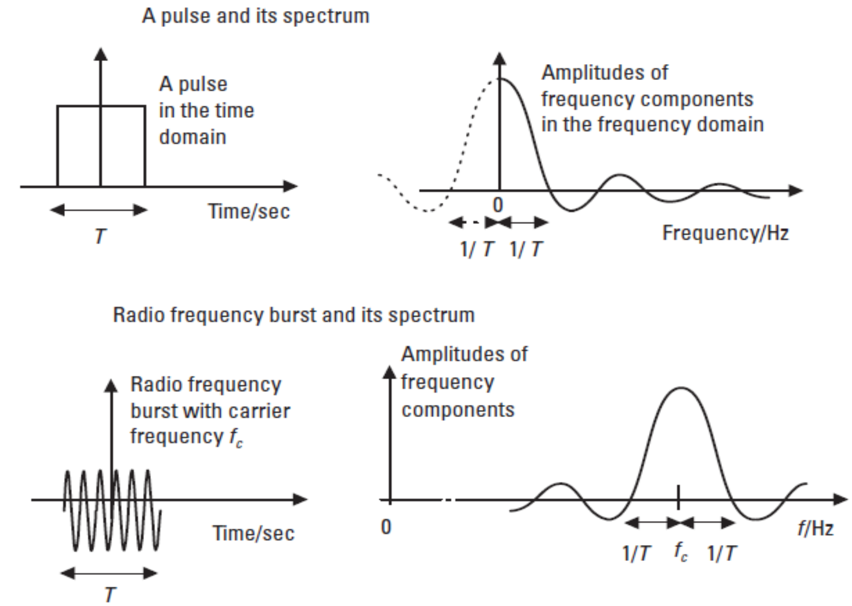


Figure 4.2 Signals in the time domain and the spectrum.

10 times higher data rate, we must use a 10 times shorter pulse, which would require a 10 times wider bandwidth.

In the other example in Figure 4.2, a digital pulse is sent as a radio-frequency burst. This is one example of digital *amplitude modulation* (AM) known as *amplitude shift keying* (ASK). Now the spectrum is concentrated around the carrier frequency, f_c, instead of zero frequency. Note that the spectral width around carrier frequency depends only on the pulse duration T, as in the previous example. If we now increase the data rate (decrease pulse duration), we make the spectrum wider, and a wider radio-frequency band is required. Note that if we let T increase without limit, the spectral width decreases and we finally have only one component in the spectrum, the carrier frequency.

Bandwidth is one of the main restricting factors for transmission. The goal of the two preceding examples was to help us understand the connection between the data rate and the required bandwidth. By understanding this we can understand, for example, why efficient speech-coding schemes are required in cellular systems. By reducing the data rate we increase the network capacity in terms of maximum number of simultaneous calls via a radio-frequency band available for the system.

4.2 Radio Transmission

In radio transmission we have to transfer the spectrum of the message into the radio-frequency band for transmission. For this we use *continuous* or *carrier wave* (CW) modulation.

4.2.1 CW Modulation Methods

The primary purpose of CW modulation in a communication system is to generate a modulated signal suited to the characteristics of a transmission channel. Modulation is needed in the transmission systems to transfer the message spectrum into high radio frequencies that propagate over radio channels. CW modulation is also used in voice-band modems where digital data modulate the carrier frequencies inside the voice frequency band.

In CW modulation the message alters the amplitude, frequency, or phase of the high-frequency carrier (Figure 4.3). This alteration is detected in the demodulator of the receiver and the original message is reproduced.

We saw in Section 3.3 that a cosine wave such as a carrier is defined by three characteristics: amplitude, frequency, and phase. In the CW modulation that we use in radio systems, we insert the message into the carrier wave by altering these three factors of the carrier wave according to the message to be transmitted.

4.2.2 AM

The original carrier wave has a constant peak value (amplitude) and it has a much higher frequency than the modulating signal, the message. In AM the peak value of the carrier varies in accordance with the instantaneous value of the modulating signal and the outline wave shape, or envelope, of the modulated wave follows the shape of the original modulating signal as shown in Figure 4.4. Thus, the unique property of AM is that the envelope of the modulated carrier has the same shape as the message.

We can show with the help of a simple mathematical analysis that when a sinusoidal wave at carrier frequency f_c Hz is amplitude modulated by a sinusoidal modulating signal at message frequency f_m Hz, the modulated wave contains the following three frequencies, as shown in Figure 4.4:

- The original *carrier frequency*, f_c Hz;
- The *sum* of the carrier and modulating signal frequencies, $(f_c + f_m)$ Hz;
- The *difference* between the carrier and modulating signal frequencies, $(f_c - f_m)$ Hz.

These sum and difference frequencies are new, produced by the AM process and they are called *sideband* frequencies. In this case the bandwidth of the modulated signal is

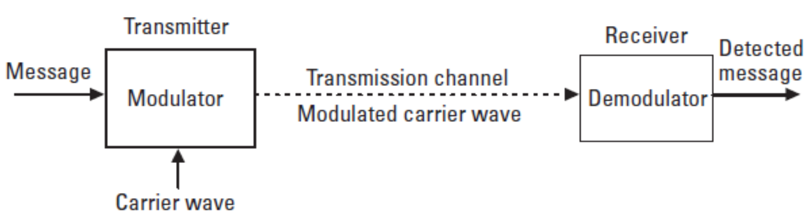


Figure 4.3 CW modulation.

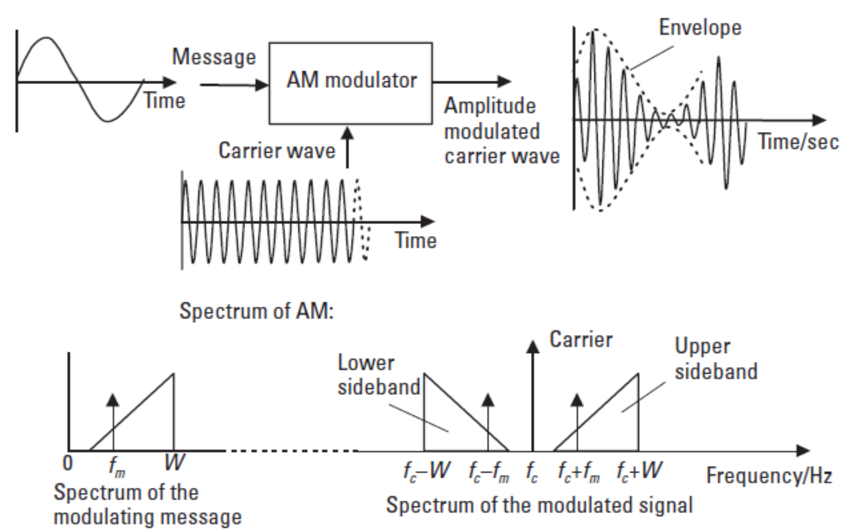


Figure 4.4 Amplitude modulation and its spectrum.

$$(f_c + f_m) - (f_c - f_m) = 2f_m \qquad (4.1)$$

If the modulating signal contains multiple frequency components, a band of frequencies such as those in speech or music, the AM process transfers the message spectrum with the carrier. The message spectrum appears after the modulation on both sides of the carrier and the required bandwidth is doubled. Figure 4.4 shows an example in which the original message with baseband bandwidth W modulates a carrier at the frequency f_c. Each individual frequency component that the message contains produces upper and lower sideband frequencies around the carrier frequency, and complete upper and lower sidebands that contain all frequencies of the message are obtained.

If the message is in digital format, the amplitude of the carrier is changed rapidly from one value to another. This is called "keying" because in early wireless telegraph systems the carrier was switched on and off with each keystroke by an operator. This type of digital AM is called *amplitude shift keying* and its spectrum was presented previously in Figure 4.2.

AM is the oldest modulation method but it is still used in radio broadcasting. The original AM has further developed into the *suppressed carrier double-sideband* (SCDSB), *single-sideband* (SSB), and *vestigal-sideband* (VSB) versions, which are briefly introduced next. These principles are explained in the frequency domain, because they are more difficult to understand in the time domain (Figure 4.5).

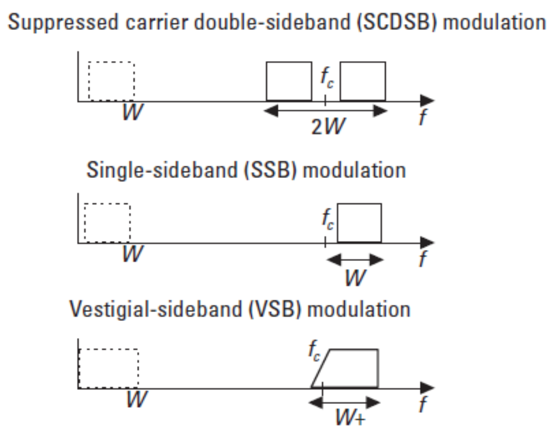


Figure 4.5 Modulation methods SCDSB, SSB, and VSB.

4.2.2.1 SCDSB

In the case of AM, the carrier is in the air even when there is no information to be transmitted. It can be shown that even with the maximum information amplitude, at least 50% of the total transmission power is spent on the carrier wave in AM. Constant amplitude, frequency, and phase carrier wave do not carry any information and transmission of the carrier wave is a waste of power from a performance point of view. In the SCDSB, or DSB for short, modulation scheme, the carrier wave is suppressed and all the power is used for sidebands that carry the information as shown in Figure 4.5.

The cost incurred to save power with the help of SCDSB is that more complex transmitters and receivers are required, but this is no longer important with current technology. The detector in the receiver cannot find the message by following the envelope only. The received carrier wave reverses phase every time the message crosses zero and, in addition to the amplitude, the phase also has to be detected. SCDSB is used, for example, for stereo information processing in analog FM radio broadcasting systems, and together with phase modulation it is used in many modern systems, such as digital radio and TV broadcast systems.

4.2.2.2 SSB Modulation

Conventional AM doubles the bandwidth of the message wasting bandwidth in addition to power. Suppressing one of the sidebands reduces the transmission bandwidth and leads to SSB modulation, as shown in Figure 4.5.

The bandwidth of a transmission channel is an especially important restriction of the carrier systems in the telecommunications networks. SSB modulation is used in the analog carrier systems that are designed to transmit as many telephone channels as possible through a bandwidth-limited channel such as a cable. SSB modulation doubles the capacity (the number of speech channels) compared with AM and SCDSB.

4.2.2.3 VSB Modulation

Consider a modulating signal, for example, the video portion of a television signal, that has a very wide bandwidth and significant low-frequency content. The bandwidth conservation principle argues in favor of SSB modulation, but practical SSB systems have a poor low-frequency response because of the filtering of the other sideband. The SCDSB would be better for this kind of application but it requires a double bandwidth. Clearly, a modulation scheme that negotiates a compromise between SSB and SCDSB is required and this is called VSB modulation.

VSB modulation is derived by filtering SCDSB (or AM; VSB is often used with a carrier) in such a fashion that one sideband is passed on almost completely while just a trace, or vestige, of the other sideband is included. In the receiver detection circuitry the vestige of the lower sideband is added to the upper sideband. This improves the quality, making the frequency response flat to very low frequencies of the message. This method is used in analog TV video transmission.

All of the modulation methods described in this section belong to the class of linear CW modulation method. Consider their common properties:

- The modulated bandwidth never exceeds twice the message spectrum.
- The transmission spectrum is basically the transferred message spectrum.
- The destination S/N is never better than if the baseband transmission was used (no modulation at all). This means that the noise power added to the transmitted signal on the line is detected in the receiver together with the desired modulating signal and the S/N is not improved in detection.

The exponential modulation methods of *frequency modulation* (FM) and *phase modulation* (PM) that we will discuss next differ on all of these counts.

4.2.3 FM

In contrast to linear modulation, exponential modulation is a nonlinear process and therefore the modulated spectrum is not related to the message spectrum in a simple fashion.

The modulated waveform after exponential modulation can be expressed by the following equation:

$$x_c(t) = A_c \cos[\omega_c t + \phi(t)] = A_c \operatorname{Re}\left\{e^{j[\omega_c t + \phi(t)]}\right\} \tag{4.2}$$

where $\phi(t)$ represents the varying phase or the frequency containing the message, A_c is the constant amplitude, $\omega_c = 2\pi f_c$ is the angular frequency of the carrier wave, and Re means that we take the real part of the exponential function in brackets. As we can see, the message is inserted into the angle of the carrier wave or in the exponent of the function describing a cosine wave. This is why these modulation methods are called either *angle* or *exponential modulations.*

In FM the instantaneous frequency of the carrier is varied according to the message and its amplitude is kept constant. Figure 4.6 shows an example in which the frequency of the carrier is increased when the value of the modulating message is increased and vice versa. We can assume that FM has good noise performance, because if the amplitude is distorted we can cut it back to the constant value in the receiver, thus eliminating most of the external disturbances. In the detector of the receiver only the instants when the signal curve crosses zero voltage need to be detected. The disturbances are highly attenuated because a large amplitude change has only a slight impact on the position of the crossing points. This helps us understand that the noise added to the transmitted signal on the line does not reduce the postdetection S/N as much as in the case of linear modulation. Actually the S/N can be improved in detection. This advantage is paid for by a wider transmission bandwidth. For example, commercial FM broadcasting uses more than 200 kHz of bandwidth for the transmission of a 15-kHz message band.

The characteristics of the spectrum of an FM signal are not as simple as those for linear modulation methods. However, in digital FM we use one carrier frequency for each digital symbol value. In the binary case we may transmit 0 for a lower frequency and 1 for the higher frequency, and each

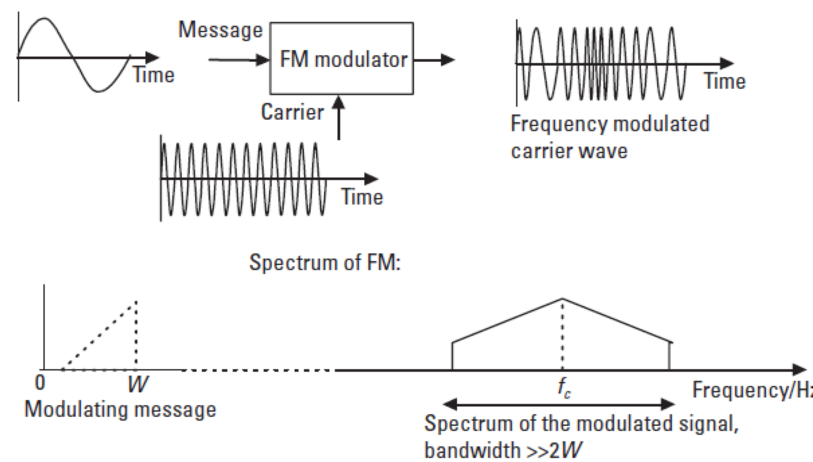


Figure 4.6 FM.

transmitted bit generates spectrum similar to the radio-frequency burst shown in Figure 4.2 around its center frequency. Now we see that width of the spectrum also depends here on the data rate, which defines length of the burst in Figure 4.2, and the distance between higher and lower frequencies used.

As an example of digital FM, some older generation voice-band modems use the digital form of FM called *frequency shift keying* (FSK). For example, a 1,200-bps V.23 modem uses two frequencies, 1,300 Hz for binary 0 and the 2,100 Hz for binary 1. Another example is digital frequency modulation of GSM in which two frequencies, 67.7 kHz above and below the nominal carrier frequency, are used for binary transmission [2].

4.2.4 PM

PM is another method in the class of exponential modulations. In PM the instantaneous phase, instead of frequency, is varied linearly according to the message. Therefore, if the message has discontinuities, there will be discontinuities in the modulated carrier wave as well (Figure 4.7). The spectral characteristics are nearly the same as in the case of FM. Figure 4.7 shows an example where the phase of the carrier is increased with the strength of the message. When message returns to zero there is a sudden phase change when the carrier returns to its nominal phase.

In digital binary PM, which is called *binary phase shift keying* (BPSK), the phase of the carrier is varied according to whether the digital signal is 1 or 0. Figure 4.8 shows an example of BPSK where the digital sequence of 0011011... is transmitted. In binary phase modulation we need only two carrier phases, which are chosen to be 0° for binary 0 and 180° for binary 1 in Figure 4.8.

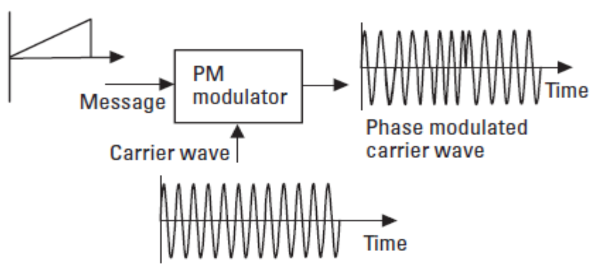


Figure 4.7 Principle of PM.

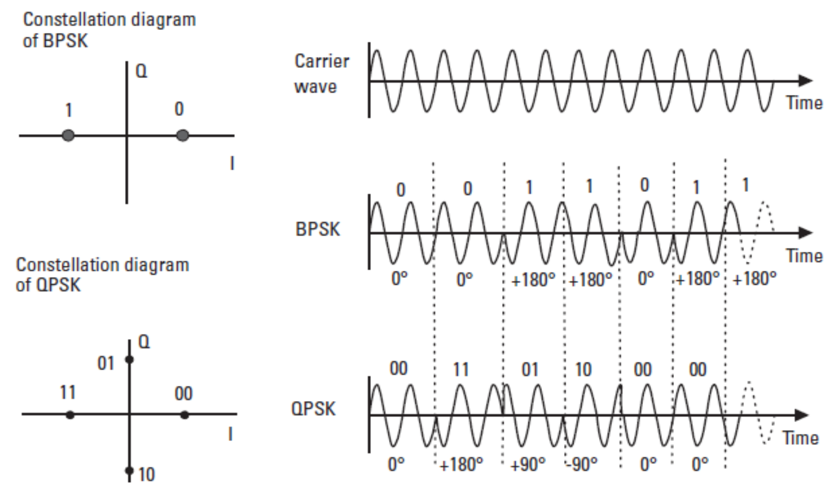


Figure 4.8 Digital PM.

Often we use more than these two phases of the carrier in digital modulation. When four carrier phases are used, each phase transmits the value of two binary bits and we talk about *quadrature phase shift keying* (QPSK). Figure 4.8 illustrates an example of QPSK. An original carrier wave and the modulated one are drawn in the figure. At a point in time a pair of bits is taken from the incoming bit stream (110001101111...) of the modulator and the carrier phase is shifted according to the value of these two bits until the next two bits are received.

One easily understandable way to describe digital phase modulation is by means of a constellation diagram, as shown in Figure 4.8. In the constellation diagram, the I axis refers to the in-phase carrier wave and Q stands for the quadrature carrier, that is, the carrier in 90° phase shift. Each signal point in the diagram represents one possible transmitted "symbol" or waveform that represents binary values of one or two bits in the examples of Figure 4.8. We can see easily from the constellation diagram for QPSK that, for example, the bit combination 01 is sent as a carrier with a +90° phase shift. The distance of the signal point from the origin tells the carrier amplitude that is the same for all symbols in our examples in Figure 4.8.

To get an idea about the spectral requirements of digital phase modulation, we can consider a single BPSK carrier burst representing for example a single 0-bit. Its spectral width depends on the duration of the symbol, which equals T in Figure 4.2. Symbol rate or modulation rate $1/T$ is expressed in bauds. Then most of the spectrum resides in the frequency range from $f_c - 1/T$ to $f_c + 1/T$ as shown in Figure 4.2. Binary 1 differs only by the carrier phase and the amplitude spectrum is the same. If we double the data rate of BPSK we have to cut symbol duration T to half, which doubles the required bandwidth. On the other hand, we can double the bit rate without increasing the bandwidth by using QPSK, in which each symbol carries two bits instead of one as shown in Figure 4.8. If symbol duration remains the same, the spectral width remains the same as well.

We could increase the data rate further by using eight different carrier phases as in 8-PSK in Figure 4.9. If the modulation rate is the same for BPSK in Figure 4.8 and for 8-PSK in Figure 4.9, both methods occupy the same frequency band but the bit rate of 8-PSK is three times that of BPSK. The cost we pay for this increased data rate is lower noise tolerance. If the transmission power of both systems is the same, the distance of signal points from origin is the same in Figures 4.8 and 4.9. Then the 8-PSK signals are much closer to other signals than are those in BPSK, and much lower noise or interference can cause errors in the receiver. The 8-PSK is used in cellular networks to increase the data rate in low-interference environments. If interference increases, modulation is changed to binary modulation, which tolerates higher interference.

Use of more phases than in 8-PSK is usually not feasible because of reduced noise tolerance. Instead we can combine AM and PM as shown in Figure 4.9 to become 16-QAM. This combination of phase and amplitude

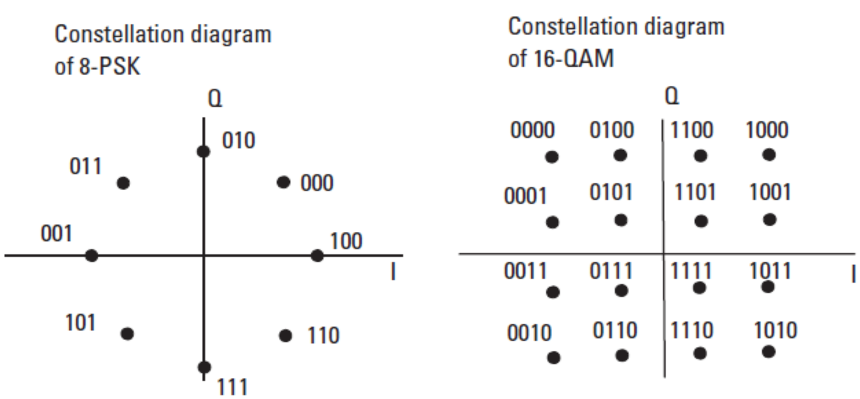


Figure 4.9 8-PSK and 16-QAM.

modulations is called *quadrature amplitude modulation* (QAM). In Figure 4.9, 16-QAM uses three amplitudes and 12 different phases to create 16 different carrier waveforms, each representing one combination of four bits. If the symbol or modulation rate is the same in 16-QAM as in BPSK, the spectral width of the radio signal remains the same but the bit rate of 16-QAM is four times that of BPSK. If we prefer to save spectrum instead of increasing the data rate, 16-QAM could use four times longer radio bursts than BPSK for the same bit rate. This would reduce the radio-frequency band to one-fourth of BPSK as we can see from Figure 4.2.

The optimum modulation method for a particular system depends on the quality of the transmission channel. In voice-band modems, which use low-noise speech channels, very large constellations with hundreds of different combinations of phases and amplitudes are feasible. In bad quality channels, such as in cellular networks, binary modulation may be the best choice.

Phase modulation together with amplitude modulation is used in many modern digital transmission systems, such as in digital radio relay systems, voice-band modems, and *digital video broadcasting* (DVB) systems, which use 64-QAM.

4.2.5 Allocation of the Electromagnetic Spectrum

Signal transmission over an appreciable distance always involves the traveling of an electromagnetic wave, with or without a guiding medium. The efficiency of any particular transmission method depends on the frequency of the signal being transmitted. With the help of CW modulation, the spectrum of the message is transferred to the suitable frequency band of the medium.

The use of frequency bands is controlled internationally by the ITU-R and nationally by national telecommunications authorities. Radio systems are often the most economical solution when new connections are required and there are no free cables or fibers between the end points of the connection. Figure 4.10 illustrates the frequency range that is used in telecommunications and it also shows some examples of the usage of different frequencies.

In Figure 4.10 the electromagnetic spectrum used in telecommunications is shown together with typical transmission media, the propagation modes, and some application examples.

However, radio systemss have one important problem that restricts the use of radio communication, namely, lack of frequency bands. The most suitable bands are overcrowded, and new technical inventions are needed in order to overcome this problem. Among these are, for example, cellular

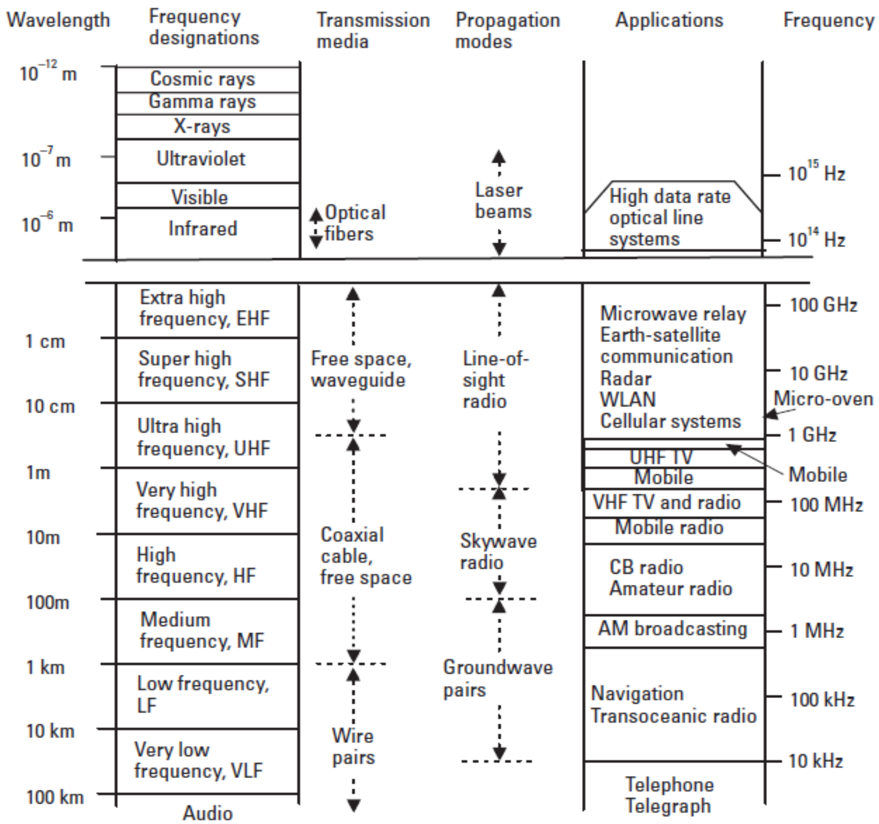


Figure 4.10 Allocation and applications of electromagnetic spectrum.

mobile systems and WLANs with small cell areas that enable them to use frequencies again in other cells of the same network, narrow beam radio relay systems, sophisticated modulation schemes for radio relays, and digital broadcasting systems. We saw in Section 4.2.4 that we can decrease the modulation rate and, correspondingly, the required bandwidth with the help of more complicated modulation schemes.

4.2.5.1 Wavelength and Frequency

The wavelength shown on the left-hand side of Figure 4.10 indicates the propagation distance during one cycle of the radio wave. It is related to the frequency and speed of light and electromagnetic wave according to $\lambda = c/f$, where λ is the wavelength in meters; c is the propagation speed of light or radio wave in meters per second, approximately 300,000 km/sec; and f is the frequency in Hz = 1/sec.

4.2.5.2 Propagation Modes

Radio waves at different frequency bands propagate in different propagation modes. They are very briefly explained as follows:

- *Ground wave:* The radio wave follows the surface of the Earth, and thus communication over the horizon is possible.
- *Skywave:* The radio wave is reflected from the ionosphere back to Earth. The wave is reflected back from the Earth's surface and back to the Earth again making long-distance communication possible. The communication quality is not stable because the characteristics of the ionosphere vary with time.
- *Line of sight:* The radio wave propagates along the straight line from the transmitter to the receiver. A general requirement for good performance is that the receiving antenna be visible from the transmitter. The radio frequencies above 100 MHz that propagate in line-of-sight mode are used in most modern communication systems.

As the demand for radio communications has increased, higher and higher frequencies have been put into use. However, as we will see in the next section, the attenuation of the radio wave increases with frequency and at extremely high frequencies, beyond 10–100 GHz, even weather conditions affect attenuation. This is why there are no applications at frequencies higher than the *extra high frequency* (EHF) band (Figure 4.10).

4.2.5.3 Optical Communications

At the infrared light frequencies just below visible light (wavelength 400–700 nm) a controlled transmission medium, optical fiber, provides very low attenuation. Optical fiber is the most important media for high-capacity long-distance transmission. It is used in national long-distance networks as well as in international and intercontinental submarine systems.

The commercial optical communication systems of today use binary light pulses for transmission. The transmitted information is usually in binary form, which means that the receiver either detects light or does not. The present optical systems are not able to use transmitted light as a carrier wave in the same way that radio systems do. Radio systems are able to change phase and frequency of the carrier wave, not just intensity. Traditionally, one optical signal occupies the whole fiber although a small portion of its very wide frequency would be feasible. Characteristics of optical fibers are introduced in Section 4.7.

However, development of narrowband optical transmitters and optical filters has made it possible to increase the data transmission capacity by inserting multiple optical channels into the same fiber with the help of the *dense wavelength division multiplexing* (DWDM) system, which is introduced later in this chapter.

As technology is developed, we will be able use light at a certain frequency as a carrier wave. Then we can increase fiber capacity further by utilizing the CW modulation methods discussed previously. The utilization of this so-called coherent optical technology will increase the transmission capacities of optical fibers dramatically in the future.

4.2.6 Free-Space Loss of Radio Waves

Most radio systems of today operate well above 100 MHz where the radio wave travels a direct path from the transmitting antenna to the receiving antenna. This propagation mode is called *line-of-sight propagation.*

The power of the radio wave is reduced with distance just as a cable attenuates propagating electrical signals. The attenuation of a radio wave, free-space loss, on the line-of-sight path is due to the spherical dispersion of the radio wave. Here we assume that both antennas are isotropic antennas, which radiate to and receive from all directions equally. The transmitted power from isotropic antennas is distributed over a spherical surface and the radiated power per unit area decreases in proportion to the square of the radius because the area of the spherical surface increases in proportion to the square of the radius. The area of the spherical surface follows the equation $A = 4\pi l^2$, where l is the radius. The power density flow F through the surface of a sphere at distance l from isotropic antenna becomes $F = P_T/(4\pi l^2)$ [W/m^2] as shown in Figure 4.11 [3].

The receiving antenna is able to receive the power that passes through its *effective aperture area* or *capture area* [4]. The effective aperture area of the receiving isotropic antenna is proportional to the square of the wavelength according to $A_{ei} = \lambda^2/(4\pi)$, and received power becomes $P_R = A_{ei} F$ [W]. From these two facts we can easily derive that the free-space loss, that is, the ratio of transmitted power and the received power in the case of isotropic antennas, where antenna gains $g_T = g_R = 1$, is

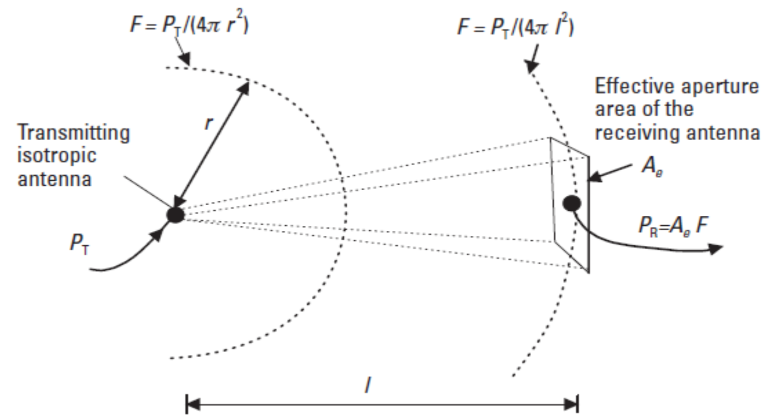


Figure 4.11 Radio wave loss with isotropic transmitting antenna.

$$L = \frac{P_T}{P_R} = \left(\frac{4\pi l}{\lambda}\right)^2 = \left(\frac{4\pi f l}{c}\right)^2 \tag{4.3}$$

where λ is the wavelength, f is the frequency of the signal, c is the speed of light, and l is the transmission distance (Figure 4.11).

We usually prefer to describe attenuation or loss in decibels instead of by the absolute value as given in the previous equations. We obtain the formula that gives decibel values by taking $L_{dB} = 10 \log_{10} L$. Now if we express the frequency f in gigahertz ($f = f_{GHz} \cdot 10^9$) and l in kilometers ($l = l_{km} \cdot 10^3$), we get the free-space loss of a radio wave in decibels as follows:

$$L_{dB} = 92.4 + 20 \log_{10} f_{GHz} + 20 \log_{10} l_{km} \text{ dB} \tag{4.4}$$

We see that the loss or attenuation is proportional to 20 times the logarithm of frequency and distance. So if the distance or frequency is doubled, the attenuation increases by 6 dB. If we want to maintain the received power, we have to increase the transmitted power by 6 dB, which requires a four times higher transmission power. This comes from the fact that the power ratio in decibels is $10 \log_{10}(P_n/P_o)$ dB, as we saw in Chapter 3.

The free-space loss shown in (4.4) may give results that are too optimistic by as much as 30 dB in actual conditions. Additional attenuation is introduced if there is a hill, a building, or a wall on or close to the straight line between the transmitting and receiving antennas. This is most often the case in mobile radio communication where actual attenuation may be of the order of 30 dB higher than free-space loss. To estimate the impact of the environment, several propagation models have been developed for cellular network planning. However, free-space loss in (4.4) clearly explains the impact of frequency and distance on radio wave attenuation.

4.2.7 Antennas

Link loss was calculated assuming that antennas are isotropic, which means that they transmit and receive equally to and from all directions. This assumption keeps the attenuation independent of the antennas in use. However, practical antennas have a focusing effect that acts like amplification, compensating for some of the propagation loss. This focusing effect can be expressed as a gain of an antenna, although a passive antenna cannot actually amplify the signal. The maximum transmitting and receiving gain (to direction of maximum radiation or sensitivity) of an antenna with effective aperture area A_e is [1]

$$g = \frac{A_e}{A_{ei}} = \frac{4\pi A_e}{\lambda^2} = \frac{4\pi A_e f^2}{c^2} \tag{4.5}$$

The value of A_e for a dish or horn antenna approximately equals its physical area and large parabolic dishes may provide gains in excess of 60 dBi, where dBi stands for gain in decibels compared with an isotropic antenna. The received power and overall radio link loss when antenna gains are considered becomes (Figure 4.12)

$$P_R = \frac{g_T g_R}{L} P_T; \quad L_{Tot} = \frac{P_T}{P_R} = \frac{L}{g_T g_R} \tag{4.6}$$

In decibel format received power levels and link loss become

$$\begin{aligned} P_{R,dBm} &= P_{T,dBm} + g_{T,dBi} + g_{R,dBi} - L_{dB} \\ L_{Tot,dB} &= L_{dB} - g_{T,dBi} - g_{R,dBi} \end{aligned} \tag{4.7}$$

Note that both antennas have equal impact on the received power level and use of a directional receiving antenna at, for example, the base station site of a cellular network reduces link loss and required transmission power of the mobile station.

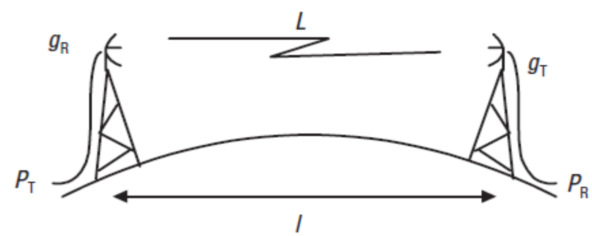


Figure 4.12 Attenuation of the radio wave.

In this section we have reviewed radio transmission at different frequencies and modulation methods that are used to transfer a message to the radio-frequency band for transmission. We have also examined the propagation loss of radio waves. Many other things must considered in radio system engineering but they are beyond the scope of our brief introduction to radio transmission.

In the following section we look at the general characteristics of transmission channels and how the maximum transmission data rate depends on the bandwidth and noise of the channel.

4.3 Maximum Data Rate of a Transmission Channel

A fundamental limit exists for the data rate through any transmission channels, as we will see later in this section. The main restricting factors are the bandwidth and the noise of the channel.

4.3.1 Symbol Rate (Baud Rate) and Bandwidth

Communication requires a sufficient transmission bandwidth to accommodate the signal spectrum; otherwise, severe distortion will result. For example, a bandwidth of several megahertz is needed for an analog television video signal, whereas the much slower variations of a telephone speech signal fit into a 4-kHz frequency band.

Every communication channel has a finite bandwidth. The higher the data rate to be transmitted, the shorter the digital pulses that can be used, as we saw in Section 4.1. The shorter the pulses used for transmission, the wider the bandwidth required, as we saw in Figure 4.2. When a signal changes rapidly in time, its frequency content or spectrum extends over a wide frequency range and we say that the signal has a wide bandwidth.

Figure 4.13 shows the shape of a rectangular pulse with duration T before and after it passed through an ideal lowpass channel of bandwidth B. For example, if the duration of the pulse $T = 1$ ms, distorted pulses are shown in the figure for the channel with bandwidths $B = 2 \cdot 1/T = 2$ kHz, $B = 1/T = 1$ kHz, $B = 1/2 \cdot 1/T = 500$ Hz, and $B = 250$ Hz. If the next pulse is sent immediately after the one in the figure, the detection of the pulse value will be impossible if the bandwidth is too narrow. The spread of pulses over other pulses, which disturbs detection of other pulses in the sequence, is called *intersymbol interference.*

In baseband transmission, a digital signal with r symbols per second, bauds, requires the transmission bandwidth B to be in hertz:

$$B \geq r/2 \tag{4.8}$$

Thus the available bandwidth in hertz determines the maximum symbol rate in bauds. Note that the symbol is not necessarily the same as the bit, but it can carry a set of bits if it is allowed to get many different values.

We can find the theoretical maximum of the symbol or baud rate with the help of a special pulse called the *sinc pulse.* The shape of the sinc pulse is drawn in Figure 4.13 and it has zero crossings at regular intervals $1/(2W)$. With the help of Fourier analysis, we can show that this kind of pulse has no spectral components at frequencies higher than W. If the channel is an ideal

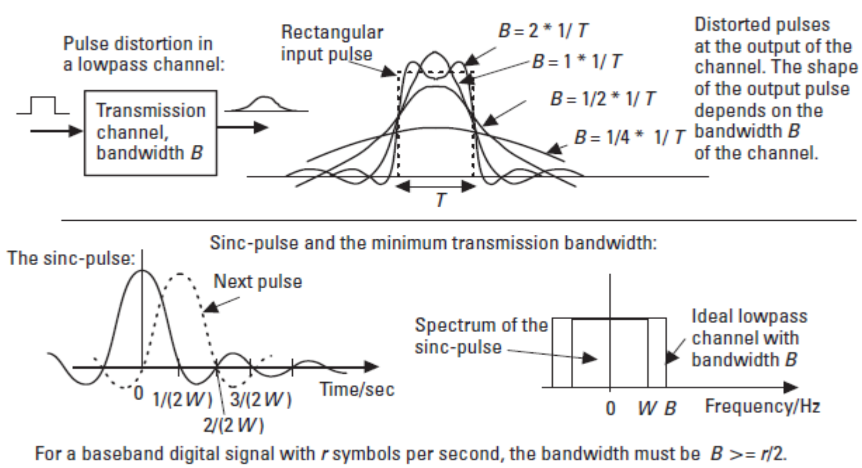


Figure 4.13 Symbol rate (baud rate) and bandwidth.

lowpass channel with a bandwidth higher than W, it is suitable for transmitting sinc pulses that have their first zero crossing at $t = 1/2W$ without distortion. The shape of the pulse remains the same because all frequency components are the same at the output as at the input of the channel.

The sinc pulses have zero crossings at regular periods in time. These periods are $1/(2W)$ seconds for a sinc pulse with a spectrum up to frequency W as shown in Figure 4.13. We can transmit the next pulse at the time instant $1/(2W)$ so that the previous pulse has no influence on the reception because it crosses zero at that time instant. The decision for the value of the pulse is made in the receiver exactly at time instants $n \cdot 1/(2W)$, where $n = \pm 1, \pm 2, \pm 3, \ldots$. The time between pulses $T = 1/(2W)$, which makes data rate $r = 1/T = 2W$. If we now increase the data rate so that $W \rightarrow B$, the time between pulses becomes $T \rightarrow 1/(2B)$; $r \rightarrow 1/T = 2B$, which gives the theoretical maximum rate for transmission of symbols and we can say that the symbol rate and bandwidth are related according to $r \leq 2B$ or $B \geq r/2$.

This kind of pulse does not exist in reality, but the result gives the theoretical maximum symbol rate, which we can never exceed, through a lowpass channel. In real-life systems quite similar pulse shapes are in use and typically a 1.5 to 2 times wider bandwidth is needed.

4.3.2 Symbol Rate and Bit Rate

In digital communications a set of discrete symbols is employed. Binary systems have only two values represented by binary digits 1 and 0. In the previous section we found that the fundamental limit of the symbol rate is twice the bandwidth of the channel. With the help of the symbols with multiple values the data rate, in bits per second, can be increased. As an example, with four pulse values we could transmit the equivalent of 2-bit binary words 00, 01, 10, and 11. Thus each pulse would carry the information of 2 bits and one symbol per second (1 baud) would correspond to 2 bps.

If we use a sinc pulse as in Figure 4.13, the preceding and following pulses do not influence the detection of a transmitted pulse, because each received pulse is measured at a zero crossing point $n \cdot 1/(2W)$ of the other pulses. We may increase the number of peak values of sinc pulses from two to four, from four to eight, for example, in order to increase the bit rate while keeping the symbol rate unchanged. Figure 4.14 shows a simple example where symbols are rectangular pulses with four symbol values and each symbol carries two bits ($k = 2$) of information. Generally, the bit rate depends on modulation rate according to

$$r_b = k \cdot r \text{ bps} \tag{4.9}$$

where k represents the number of bits encoded into each symbol. Then the number of symbol values is $M = 2^k$ and the bit rate is given as $r_b = r \log_2 M$ bps. In the example of Figure 4.14, the number of symbol values is $M = 2^k = 2^2 = 4$, and the bit rate $r_b = k \cdot r$ bps $= 2r$ bps. Then the symbol rate of 1 kbaud makes the bit rate 2 Kbps.

The unit of symbol rate, sometimes called the modulation rate, is bauds (symbols per second). Note that the transmission rate in bauds may represent a much higher transmission rate in bits per second. Table 4.1 shows how the bit rate of a system depends on the number of symbol values.

Figure 4.14 also shows a data sequence of sinc pulses with four amplitude values. The required bandwidth for pulses in this sequence is the minimum bandwidth $B = r/2 = 1/(2T)$ according to (4.8) and Figure 4.13. When pulses are detected by sampling as shown in Figure 4.14 each pulse can be detected independently because values of all other pulses are equal to zero.

In the preceding examples, the amplitudes of the pulses contain the information. This is the principle of PAM, as discussed earlier. This is not the only alternative. We can use other characteristics of the signal as well to create multiple symbol values, for example, the phase of a carrier, as we did in the case of QPSK and 8-PSK in Figures 4.8 and 4.9. There we used a certain modulation rate r in bauds (how many times the phase can change in a second), which defines a required bandwidth. For QPSK 2 bits ($k = 2$) are encoded into each symbol and the bit rate is two times the modulation rate. For 8-PSK, $k = 3$ and $r_b = 3r$. The 16-QAM example in Figure 4.9 used 16 combinations of carrier amplitude and phase amplitude values and the bit rate is four times the modulation rate.

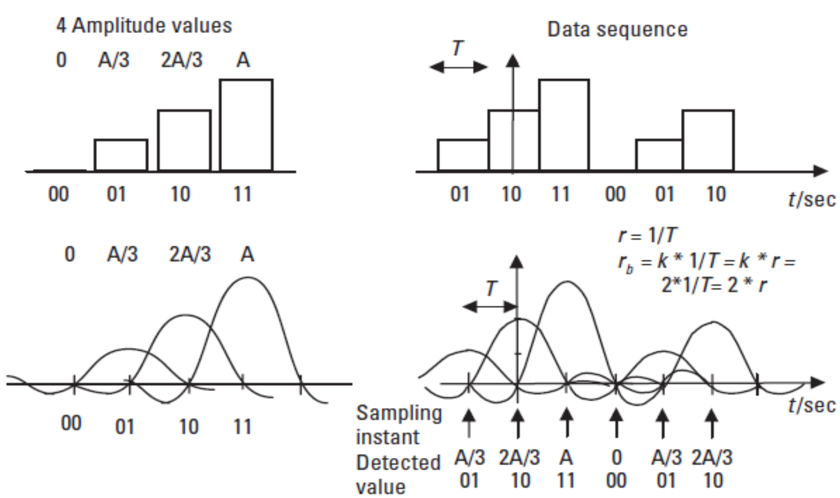


Figure 4.14 Symbol rate and bit rate.

Table 4.1
Bit Rate of a System Using Multiple Symbol Values

Number of Bits, *k*, Encoded into Each Symbol	Number of Symbol Values, *M*	Bit Rate Compared with Symbol Rate
1	2	Same as symbol rate
2	4	2 × symbol rate
3	8	3 × symbol rate
4	16	4 × symbol rate
5	32	5 × symbol rate
…		
8	256	8 × symbol rate
…		

As we can see from Table 4.1, by increasing the number of different symbols used in the system the data rate could be increased without a limit if there were no other limitations than bandwidth. This is not possible in practice because of the noise. The influence of noise is discussed next.

4.3.3 Maximum Capacity of a Transmission Channel

We saw previously that the bandwidth of a channel sets the limit to the symbol rate in bauds but not to the information data rate. In 1948, Claude Shannon published a study of the theoretical maximum data rate in the case of a channel subject to random (thermal) noise.

We measure a noise relative to a signal in terms of the S/N. Noise degrades fidelity in analog communication and produces errors in digital communication. The S/N is usually expressed in decibels as

$$S/N_{\text{dB}} = 10 \log_{10}(S/N) \text{ dB} \tag{4.10}$$

Taking both bandwidth and noise into account, Shannon stated that the error-free bit rate through any transmission channel cannot exceed the maximum capacity C of the channel given by:

$$C = B \log_2 (1 + S/N) \tag{4.11}$$

where C is the maximum information data rate in bits per second; B, the bandwidth in hertz; S, the signal power; N, the noise power, and S/N, the S/N power ratio (absolute power ratio, not in decibels).

Equation (4.11) gives a theoretical limit for the data rate with an arbitrarily low error rate when an arbitrarily long error correction code is used. It also assumes that the signal has a Gaussian distribution as does the noise, which is not the case in practice. The influence of bandwidth and noise in the case of binary and multiple value signaling is summarized in Figure 4.15.

The signal power and, thus, the highest value of the signal are always restricted to a certain maximum value. Then, the more symbol values we use, the closer they are to each other, and the lower noise level can cause errors. Thus, a higher bit rate requires a wider bandwidth that allows a higher symbol rate. Alternatively, a better S/N is required to allow for more symbol values.

The example in Figure 4.15 shows what happens to the distance between symbol values when the maximum amplitude is A and four symbol values are used instead of binary symbols that have only two values. In our examples we have used symbols with different amplitudes. This transmission scheme is called PAM, as discussed earlier. Transmission of this type of pulses without CW modulation is called baseband transmission.

In radio systems or modems that use CW modulation, different phases of a carrier wave represent different symbol values. In the Figure 4.8 and 4.9 digital phase modulation methods, BPSK, QPSK, and 8-PSK all require the same bandwidth if symbol rate is the same, but the information data rate for QPSK is double and for 8-PSK triple compared with BPSK. The cost we have to pay is reduced noise tolerance because signals become closer to each other as more symbol values or different signals are used. It is not usually reasonable to use more than eight phases; instead, we use different amplitudes as in 16-QAM in Figure 4.9. The 16-QAM tolerates more noise than 16-PSK because with the same average signal power distances between signals can be increased. However, if we would analyze noise tolerance in more detail, we could form a general rule stating that the increase in the number of signals in use reduces noise tolerance. In low-noise channels, such as telephone voice channels, many different signals can be used but in high-interference channels, such as radio channels for cellular systems, binary symbols are often the best choice.

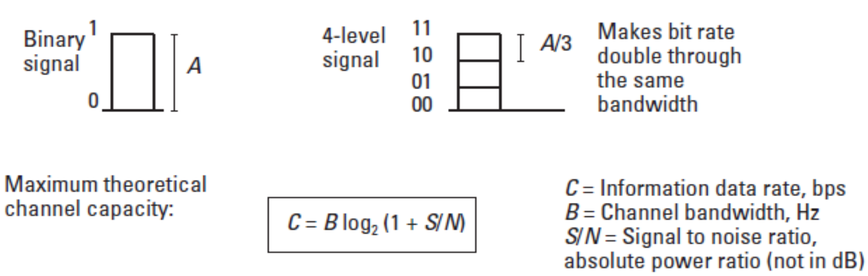


Figure 4.15 The maximum capacity of a transmission channel.

However, modulation moves the spectrum of the pulse from low frequencies to carrier frequencies, and the bandwidth is typically doubled when compared with baseband systems as was shown in Figure 4.2. This is why the symbol rate in radio systems is less than or equal to the transmission bandwidth, that is:

$$r \leq B_T \tag{4.12}$$

where r is the symbol rate in bauds and B_T is the transmission bandwidth in hertz.

The accurate requirement of bandwidth depends on the modulation scheme in use, the study of which is beyond the scope of this book.

Example 4.1

Assume that the transmission channel is an ideal lowpass channel with a bandwidth of 4 kHz. The maximum symbol rate via this channel is $r \leq 2 \cdot B$ = 8 kbauds; that is, we can transmit up to 8,000 independent signals, symbols, in a second. [To transmit the same symbol rate through a bandpass channel, we would need a bandwidth of 8 kHz according to (4.12); see also Figure 4.2.]

Example 4.2

Assume that the S/N of a lowpass channel is 28 dB and its bandwidth is 4 kHz. Then $S/N_{dB} = 10 \log_{10} S/N$, $S/N = 10^{2.8} \approx 631$. The maximum bit rate according to (4.11) is $C = B \log_2(1 + S/N) = 4{,}000 \log_2(432) = 4{,}000 (\log_{10} 632)/\log_{10} 2 = 37.2$ Kbps. In Example 4.1 we learned that the maximum symbol rate is 4 kbauds, which depends only on the bandwidth. To achieve the maximum bit rate, we transmit 4,000 symbols in a second and each of them carries 3 bits (with 4 bits, the maximum bit rate would be exceeded). The number of different symbols that can be used is $2^3 = 8$ and this depends only on the S/N maximum, not on bandwidth.

4.4 Coding

We have described modulation as the processing of a signal for efficient transmission in a different frequency band than where the information originally exists. Coding is a digital symbol processing operation in which the digital form of the information is changed for improved communication. In general, coding contains many different processes, such as ciphering, compression, and error control coding. For ciphering, the transmitter and the receiver may simply perform an exclusive-or operation with data and a ciphering sequence known only by the transmitter and receiver. An eavesdropper is not able to detect information content without knowing the ciphering sequence.

Most modern systems use error control codes for handling of transmission errors. By appending extra check digits to the transmitted data, we can detect or even correct errors that occur on the line. Error control coding increases both the required bandwidth (data rate increases) and the hardware and software complexity, but it pays off in terms of nearly error-free digital communication even when the S/N is low.

Still another purpose for coding is for compressing information. By using data compression we can reduce the disk space needed to store data in a computer. In the same way we can decrease the required data rate on the line to a small fraction of the original information data rate. We could, for example, use very short codes for the most common characters instead of the full seven-bit ASCII code. Rarely needed characters would use long codes and the total data rate would be reduced. Some compression schemes for voice and video information were introduced in Chapter 3. The study of compression methods is a complex matter and will not be covered in any detail here.

From now on we concentrate only on line coding, which changes source symbols into another form for transmission. The operation of line *encoding* transforms a digital message into a new sequence of symbols. *Decoding* is the opposite process that converts the encoded sequence back into the original message (Figure 4.16).

Consider a computer terminal with a keypad. Each key represents a discrete digital symbol. Uncoded transmission would require as many different waveforms as there are keys, one for each key (or more, one for shift, one for Alt, and one for Alt Gr). Alternatively, each symbol can be an encoder into a binary code word consisting of a number of binary digits for binary transmission.

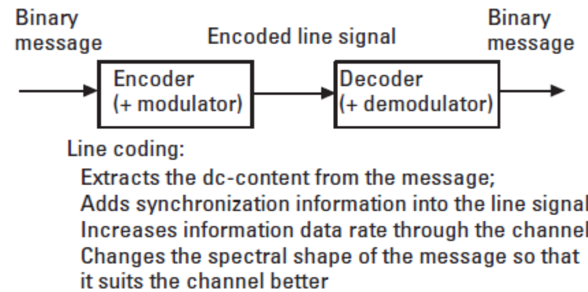


Figure 4.16 Line coding.

4.4.1 Purpose of Line Coding

One purpose of line coding is to make the form of the spectrum of a digital signal suitable for a certain communication media. The line codes usually have no dc content (direct current, frequency component at 0 Hz). We want to get rid of the dc that does not transmit any information but wastes power.

Another reason for line encoding is to help to synchronize the receiver. In digital transmissions the receiver must be synchronized with the transmitter in order to receive the information when each new symbol arrives. For this the data should be transmitted in a form that contains synchronization information so that there is no need to transmit additional clock or timing signals.

The systems that use only line coding, but not modulation, are called baseband transmission systems. The spectrum of the line signal is still in the frequency range of the original message's "baseband." In radio systems both coding and modulation are used.

Line coding can be used to increase the data rate as shown in, for example, Figure 4.17, where each sequence of 2 data bits is encoded into four-level pulses for transmission. At the receiving end decoding is carried out and the original bits, 2 for each received symbol, are regenerated. Note that the symbol rate on the line is half of the bit rate seen by the data source and the destination and thus the required bandwidth of the channel is reduced to half compared to binary transmission. The line code in Figure 4.17 also cancels dc and similar code is used in ISDN subscriber lines. Note that Gray coding, in which neighbor symbols differ by only one bit, is used in Figure 4.17. The symbol in error is typically a neighboring symbol of the transmitted symbol and with the help of Gray coding only one information data bit in error is generated.

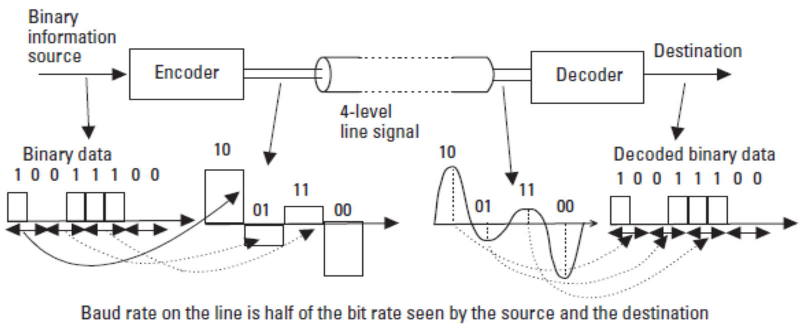


Figure 4.17 An example of the line coding.

We often combine coding and modulation and instead of four or more pulse amplitude values we may transmit four symbol values in carrier waveforms with, for example, four different phases. This so called QPSK was discussed in Section 4.2.4 and it can be seen to be a combination of four-level line coding followed by ordinary phase modulation.

4.4.2 Spectrum of Common Line Codes

To determine what kind of impact line encoding has on the spectrum, we look at the characteristics of some common line codes. Figure 4.18 presents their power density spectrums showing how the signal power of random data is distributed over the frequencies.

4.4.2.1 Nonreturn to Zero (NRZ)

NRZ is the most common form of digital signal used internally in digital systems. Each symbol has a constant value corresponding to binary symbol values 1 and 0. The spectrum has a high dc component, and there are no discrete spectral components at the harmonic frequencies of the data rate. The harmonic frequencies are multiples of the data rate. An external clock signal is always needed for the timing of the receiver.

4.4.2.2 Return to Zero (RZ)

RZ each symbol is cut into two parts. The first half of the symbol represents the binary value and the rest of the symbol is always set to zero. Because pulses are shorter than in the case of NRZ the spectrum is wider, as we saw in Figure 4.2, and the spectrum of a random data has strong discrete frequency components at the harmonic frequencies of the data rate. With the help of

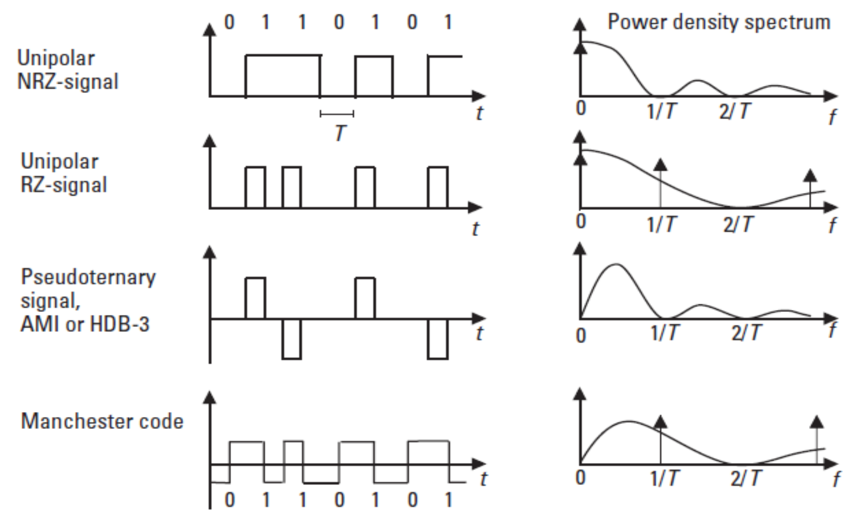


Figure 4.18 Common line codes and their power spectrums.

these components, timing information can be extracted from the signal spectrum and an external clock is not necessarily needed. However, because RZ code has high low-frequency content and a wide spectrum (see Figure 4.18), it is never used in long-distance transmission. Another problem is that synchronization is lost if the data content is all zero for a long period of time.

4.4.2.3 Alternate Mark Inversion (AMI)

If every other mark or 1 of the NRZ or RZ symbols is transmitted as an inverted voltage polarity, an AMI signal is produced. The advantage of this is that no dc component is present on the transmission line. The dc component is unwanted because it does not carry any information; it merely wastes power. With the help of this kind of code we can avoid the problem caused by transformers on the line. Transformers are needed on copper cable lines for matching impedance, for overvoltage or surge protection, and for other purposes. Direct current does not propagate through transformers.

AMI code is used in American telecommunications network in primary rate 1.5-Mbps transmission systems. We may extract the timing information by rectifying the AMI signal into an RZ signal in the receiver and then the discrete spectral components appear as in the spectrum of RZ code in Figure 4.18.

4.4.2.4 High-Density Bipolar 3

HDB-3 was developed from AMI and standardized for European primary rate 2-Mbps systems. HDB-3 overcomes the problem of the original AMI code that occurs in the timing when a data message contains long periods of subsequent zeroes. In this coding scheme, a pulse with the same polarity as the previous one is added in such a way that no more than three sequential zeroes are allowed. In the decoder these pulses are taken away according to the AMI coding rule that they violate.

4.4.2.5 Manchester Coding

Manchester coding is used in LANs. Binary digit 1 is coded as a "+ to –" transition and binary 0 as a "– to +" transition. The most important advantage of the Manchester code is that each symbol contains the timing information and the receiver needs only to detect the transition in the middle of each received symbol to extract the clock signal. Its main disadvantage is a wide spectrum because of short pulses and this is why it is suitable for LANs but not for long-distance transmission.

4.5 Regeneration

In long-haul transmissions the transmitted signal is attenuated and amplifiers or repeaters are needed. *Analog amplifiers* amplify the signal at the input, and the signal contains both the desired message and channel noise. In every amplifier and cable section some noise is added and the S/N decreases with distance.

Unlike analog amplifiers, digital repeaters are regenerative. A regenerative repeater station consists of an equalizing amplifier that compensates the distortion and filters out the out-of-band noise and a comparator as shown in Figure 4.19. Output of the comparator is high if the input signal is above the threshold voltage V_{ref}, and low if the input is below the threshold value. The regenerator also contains timing circuitry, which extracts the clock signal from the received data, and a D-type flip-flop which decides if a digit is high (1) or low (0) at the instant of the rising edge of the clock signal (see Figure 4.19). At the rising edge of the clock signal the input value is read into the output by the D-type flip-flop. The output value remains the same until the next rising edge of the clock signal. The operating principle of a binary regenerative repeater is presented in Figure 4.19. The regenerated digits that contain no noise are delivered to the destination or via a cable to the next repeater station (in the case of an intermediate repeater).

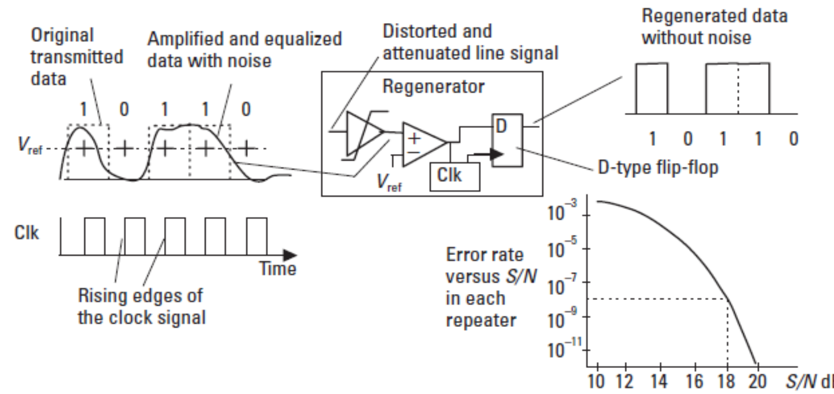


Figure 4.19 Operating principle of a regenerative repeater.

If the equalized signal is below threshold V_{ref} at the input of the comparator, the output is low and a zero is regenerated at the rising edge of the clock signal. If noise is too high, the input of the comparator may be above threshold even though a zero is transmitted. If this occurs at the rising edge of the clock signal, the value 1 is regenerated and an error has occurred. In the same way, high values may be in error if noise reduces the high-amplitude value below the threshold level at an instant of the rising edge of the clock signal. Then 0 is regenerated and an error has occurred.

How frequently errors occur depends on the noise level or S/N. If noise is assumed to have a *Gaussian amplitude distribution* (as thermal noise does), the error rate (bit error probability) follows the shape of the curve, error rate versus S/N, in Figure 4.19.

As an example let us assume that we have a channel, for instance a cable, that attenuates a signal so much that the resulting S/N in the repeater is 15 dB. The error rate would then be around 1×10^{-5} according to the curve in Figure 4.19. If we place a new repeater in the middle of the repeater section (in the middle of the cable), attenuation of the signal is 3 dB less, giving an S/N value of 18 dB in both repeaters and the error rate at both repeaters would be 1×1^{-8}. This means that one error occurs on average after 100,000,000 correct bits. Now we have two repeaters and we have an overall error rate of 2×10^{-8} because each of them creates on average one error in each sequence of 1×10^{8} bits. We can see that the improvement of 3 dB in the S/N that we achieved with the help of the new repeater reduces the number of errors by a factor of 0.001.

In practice, the error rate of an operational transmission system is often much better and we have close to error-free transmission and the exact equivalent of the original signal is received in the end regardless of the distance (the number of repeaters).

The error rate decreases rapidly with noise as shown in Figure 4.19 because of the Gaussian nature of thermal noise. Not only thermal, but many other types of noise in real-life systems are assumed to follow a Gaussian distribution. With this model the reduction of noise by 1 dB improves the error rate by factor of 10 or more, as seen in Figure 4.19. The digital transmission systems installed in telecommunications networks are designed in such a way that noise is low enough in all regenerators and the error rate is extremely low. For example, optical line systems usually have a design practice of worst-case lifetime error rate of 1×10^{-10}. In normal operational conditions the error rate is several orders of magnitude better and they operate nearly error free.

From the error rate curve in Figure 4.19 we see how the error rate depends on the S/N. From the error rate we can easily calculate the mean time between errors when the data rate is known. Table 4.2 gives examples of error rates and mean times between errors for a 64-Kbps (ISDN B-channel) data channel.

Digital systems have a certain threshold value for the S/N. From the curve in Figure 4.19 and Table 4.2, we find that if the S/N is worse than 18 dB, errors occur quite frequently. At a few decibels better value for the S/N, the transmission is almost error free. The S/N values in the

Table 4.2
Examples of Error Rates and Mean Times Between Errors for a 64-Kbps Channel

S/N (dB)	Error Rate	Mean Time Between Errors
10.3	10^{-2}	1.5 ms
14.4	10^{-4}	150 ms
16.6	10^{-6}	15 seconds
18	10^{-8}	26 minutes
19	10^{-10}	2 days
20	10^{-12}	6 months
21	10^{-14}	50 years

Figure 4.19 curve and Table 4.2 are examples and are based on certain assumptions. The actual S/N value in decibels at a certain error rate of a specific system depends on the system characteristics and how the S/N is defined and measured. However, the shape of the error curve is the same as in Figure 4.19 and the threshold value is usually between 8 and 20 dB.

When the S/N of a digital system decreases, errors occur more and more frequently and when the error rate becomes too high, information is lost. An error rate of 1×10^{-3} is standardized to be the worst allowed communication quality for PCM speech in the telecommunications network. If the error rate becomes worse, ongoing calls are cut off. Data are transmitted in large packets and if a packet contains one or more errors it needs to be retransmitted. As a rule of thumb, we can say that data transmission requires an error rate of 1×10^{-5} or better, otherwise retransmissions slow down the end-to-end transmission data rate.

4.6 Multiplexing

Multiplexing is a process that combines several signals for simultaneous transmission on one transmission channel. Most of the transmission systems in the telecommunications network contain more capacity than is required by a single user. It is economically feasible to utilize the available bandwidth of optical fiber or coaxial cable or a radio system in a single high-capacity system shared by multiple users. The main principles of multiplexing are described in the following sections.

4.6.1 Frequency-Division Multiplexing (FDM) and TDM

FDM modulates each message to a different carrier frequency. The modulated messages are transmitted through the same channel and a bank of filters separates the messages at the destination (Figure 4.20). The frequency band of the system is divided into several narrowband channels, one for each user. Each narrowband channel is reserved for one user all the time. FDM has been used in analog carrier systems in the telephone network. The same principle is also used in analog cellular systems in which each user occupies one FDM channel for the duration of the call. In such a case, we call the process *frequency-division multiple access* (FDMA) because the frequency-division method is now used to allow multiple users to access the network at the same time.

A more modern method of multiplexing is TDM, which puts different messages, for example, PCM words from different users, in nonoverlapping

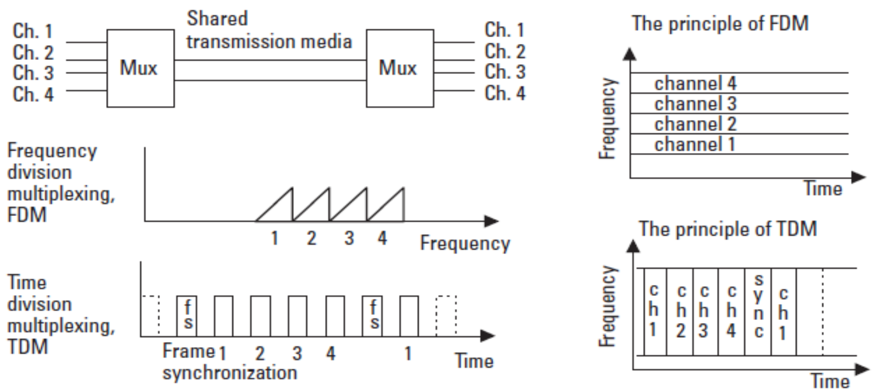


Figure 4.20 Multiplexing methods FDM and TDM.

time slots. Each user channel uses a wider frequency band but only a small fraction of time, one time slot in each frame as shown Figure 4.20. In addition to the user channels, framing information is needed for the switching circuit at the receiver that separates the user channels (time slots) in the demultiplexer. When the demultiplexer detects the frame synchronization word, it knows that this is the start of a new frame and the next time slot contains the information of user channel 1.

This method of TDM is used in high-capacity transmission systems such as optical line systems but also in digital cellular networks where we call it *time-division multiple access* (TDMA). One user occupies one time slot of a frame, and the time-division principle allows multiple users to access the network at the same time using the same carrier frequency.

4.6.2 PCM Frame Structure

We introduced the principle of TDM in the previous section. As an example of TDM and to get a clear view of TDM, we now look at the most common frame structure in telecommunications networks, namely, the primary rate 2,048-Kbps frame used in the European standard areas. This is the basic data stream that carries speech channels and ISDN-B channels through the network and it is called E-1. The corresponding North American primary rate is 1.544 Mbps, which carries 24 speech channels and it is known as DS1 or T1. It is also introduced in this section.

In the European scheme, the primary rate frame is built up in digital local exchanges that multiplex 30 speech or data channels at bit rate of 64 Kbps into the 2,048-Kbps data rate. ITU-T defines this frame structure in Recommendation G.704.

4.6.2.1 The 2-Mbps Frame Structure

PCM-coded speech is transmitted as 8-bit samples 8,000 times a second, which makes up a 64-Kbps data rate. These eight-bit words from different users are interleaved into a frame at a higher data rate.

The 2,048-Kbps frame in Figure 4.21 is used in the countries implementing European standards for telecommunications. It contains 32 time slots, and 30 of them are used for speech or 64-Kbps data. The frame is repeated 8,000 times a second, which is the same as the PCM sampling rate. Each time slot contains an eight-bit sample value and the data rate of each channel is 64 Kbps. These voice channels or data channels are synchronously multiplexed into a 2-Mbps data stream, which is often called E1 (first level in European hierarchy). For error-free operation the tributaries (64-Kbps data streams of the users) have to be synchronized with the clock signal of the 2-Mbps multiplexer. The data rate of 2,048 Kbps for the multiplexer is allowed to vary by 50 *parts per million* (ppm), and as a consequence each user of the network has to take timing from the multiplexer in the network and generate data exactly at the data rate of the multiplexer divided by 32.

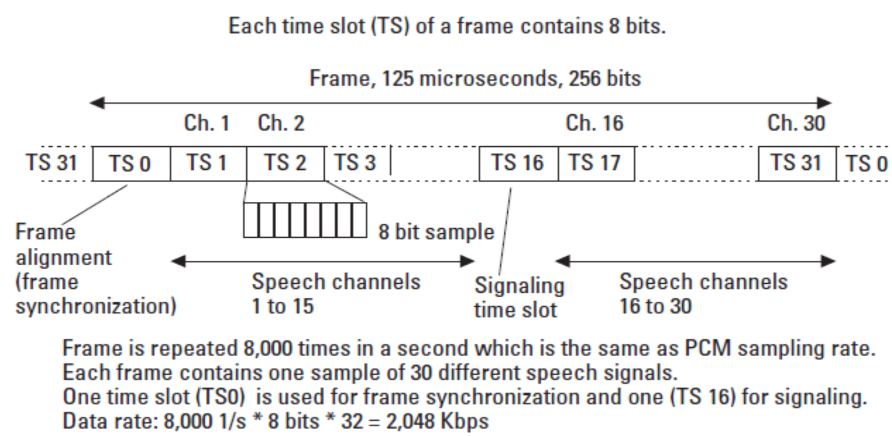
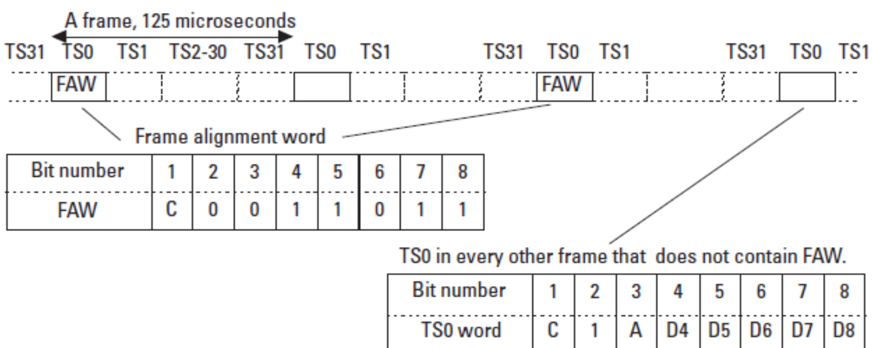


Figure 4.21 The 2,048-Kbps frame structure from Recommendation G.704.

Frame Synchronization Time Slot

The frame alignment word is needed to inform the demultiplexer where the words of the channels are located in the received 2-Mbps data stream. The frame synchronization time slot (TS0) includes frame alignment information and it has two different contents that are alternated in subsequent frames (Figure 4.22). The demultiplexer looks for this time slot in the received data stream and, when it is found, locks onto it and starts picking up bytes from the time slots for each receiving user. Each user receives 8 bits in 125-μs periods, which makes 64 Kbps. A fixed alignment word is not reliable enough for frame synchronization because it may happen that a user's data from one channel simulates the synchronization word and the demultiplexer might lock to this user time slot instead of TS0. This is why there is one alternating bit (D2) in time slot 0 (see Figure 4.22) and due to this the demultiplexer is able to detect the situation where one channel constantly transmits a word that is equal to the *frame alignment word* (FAW).

To make frame alignment even more reliable, the *cyclic redundancy check 4* (CRC-4) procedure was added in the mid-1980s. C-bits are allocated to carry a four-bit error check code that is calculated over all bits of a few frames. The receiver performs error check calculations over all bits of the frames and it is able to detect false frame alignment even if the frame alignment word is simulated by one user that alters bit two.



A = Far end alarm, alarm condition "1".
D = Spare bits that can be used for specific point-to-point low data rate applications (for example, network management information).
Bit 2 alternates from frame to frame to prevent accidental simulations of the frame alignment signal.
C = CRC-4 procedure for protection against simulation of frame alignment and enhanced error monitoring. If not in use, C-bit is set to "1".

Figure 4.22 The 2,048-Kbps frame alignment word in TS0.

Each receiver of the 2,048-Kbps data stream detects errors in order to monitor the quality of the received signal. Error monitoring is mainly based on the detection of errors in the frame alignment word. The receiver compares the received word in every other TS0 with the error-free frame alignment word. In addition to the frame alignment word, the CRC-4 code is used to detect low error rates. Errors in the frame alignment word do not give reliable results when the error rate is very low. It may take a long time before an error is detected in TS0 although many errors may have occurred in other time slots of the frame. The C-bit in Figure 4.22 is set to 1 if CRC is not used [5].

The TS0 in every other frame also contains a far-end alarm information bit A as shown in Figure 4.22. This is used (set to 1) to tell the transmitting multiplexer that there is a severe problem in the transmission connection and reception is not successful at the other end of the system. This can be caused by, for example, a high error rate, loss of frame alignment, or loss of signal. With the help of the far-end alarm, consequent actions can take place. These actions include rerouting user channels to another operational system.

D-bits can be used for transmission of network management information. At international borders they are usually set to 1.

Multiframe Structure of the Signaling Time Slot

Time slot 16 (TS16) is defined to be used for the channel associated signaling to carry separate signaling information to all user channels of the frame. TS16 is a transparent 64-Kbps data channel like any other time slot in the frame. Thirty channels share the signaling capacity of TS16. A frame structure is needed to allocate the bits of this time slot to each of the 30 speech channels. The information about the location of the signaling data of each speech channel is given to the signaling demultiplexer with the help of the multiframe structure containing a multiframe alignment word for multiframe synchronization. The data rate available for each speech channel is 2 Kbps. Because the CAS signaling systems are or will in the near future be replaced by common channel signaling we do not cover multiframe structures in detail here.

For CSS, multiframe is not needed and the signaling information of all users is carried in data packets and any time slot can be used for this. Each packet carries information about the call to which it is related and signaling information. CCS packets can in some cases, for example, in the short message service of GSM, also carry user data.

4.6.2.2 The 1.544-Mbps Frame Structure

The primary data rate in the United States and Japan is 1.544 Mbps instead of the 2.048 Mbps used in areas that go by European standards. Both

European PCM frame and the 1.544-Mbps frame are repeated at PCM sampling rate that is 8,000 times in a second. The frame structure shown in Figure 4.23 is used in North America and known as T1 or DS1 frame [5].

The North American PCM system accomplishes frame alignment differently than does the European 2-Mbps system. Like its European counterpart, it uses eight-bit time slots, but each frame contains 24 channels. To each frame, one-bit frame, a frame alignment, or synchronization bit (S-bit) is added, and we get a 1.544-Mbps data rate as shown in Figure 4.23. A multiframe is constructed from 12 subsequent frames and their 12 S-bits make up the 6-bit frame and 6-bit multiframe synchronization words [5].

In T1 there is no reserved time slot for CAS information as we have in the 2-Mbps frame structure. Instead of that, the least significant bit of each channel in every sixth frame is used for signaling. As a consequence, only seven bits in each time slot are transparently carried through the network and the basic user data rate is 56 Kbps instead of the 64 Kbps in the European systems.

For frame synchronization and for demultiplexing of signaling information, frames make up a multiframe structure with two alternative lengths, a superframe containing 12 frames or an *extended superframe* (ESF) containing 24 frames. The framing bits of ESF, one in each frame, carry frame synchronization information including CRC code and data channel for network management messages. The detailed structure of the multiframe is explained, for example, in [5].

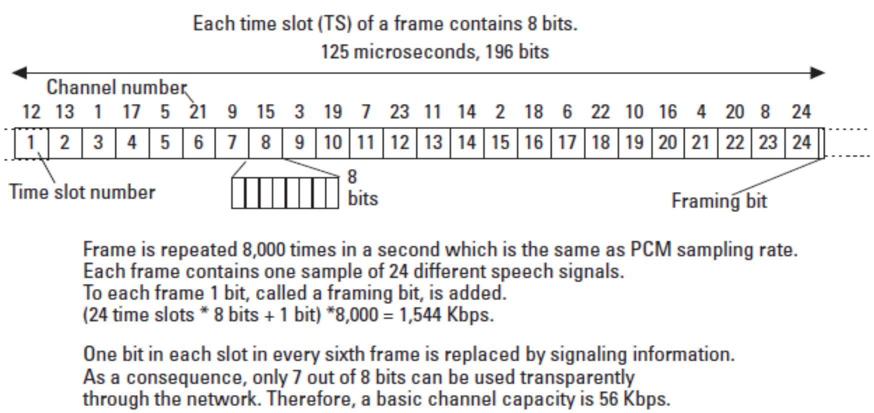


Figure 4.23 The 1.544-Mbps PCM frame.

In transatlantic connections, E1 frames are adapted to the T1 frame structure and transcoding between μ-law and A-law PCM is carried out. Each time slot in E1 is transmitted further in one time slot of T1.

4.6.3 Plesiochronous Transmission Hierarchy

A primary rate of 1.5 or 2 Mbps is usually too slow for transmission in trunk or even in local networks. This was noticed in the early 1970s and the ITU-T, at that time CCITT, standardized the higher data rate systems for transmission in the latter half of the 1970s. The digital systems of those days carried primarily analog information and end-to-end synchronization was rarely required. The first standardized digital higher-order transmission hierarchy is known as *plesiochronous digital hierarchy* (PDH). We review first the European hierarchy of higher-order multiplexing.

4.6.3.1 European PDH for Higher-Order Multiplexing

The higher-order multiplexers of PDH are allowed to operate according to their own independent clock frequencies. These standards are based on plesiochronous operation ("almost the same data rate"), which allows a small frequency difference between tributary signals that are multiplexed into a higher aggregate rate. For example, at 2,048 Kbps the frequency tolerance was standardized at 50 ppm, and at 8,448 Kbps the allowed tolerance is 20 ppm. This means that, for example, the data rate of a 2,048-Kbps system may deviate by 100 bps.

The basic principle of the European standard for higher-order multiplexers is that each multiplexer stage takes four signals of a lower data rate and packs them together into a signal at a data rate that is a little bit over four times as high, as shown in Figure 4.24. In addition to tributaries, aggregate frames contain frame alignment information and justification information.

The tributary frequencies may differ slightly and their frequencies must be justified to the higher-order frame. This process, called *justification* or *stuffing*, adds a number of justification bits to each tributary in order to make the average tributary data rates exactly the same. In the demultiplexer these justification bits are extracted and the original data rate for each tributary is generated.

At each hierarchy level the tributary signals are *bit interleaved* to the aggregate data stream, which means that the aggregate data stream contains one bit from tributary 1, one bit from tributaries 2, 3, and 4, and then again from tributary 1, and so on. Additional bits are needed in the frame for frame synchronization (frame alignment) and justification, and therefore the next

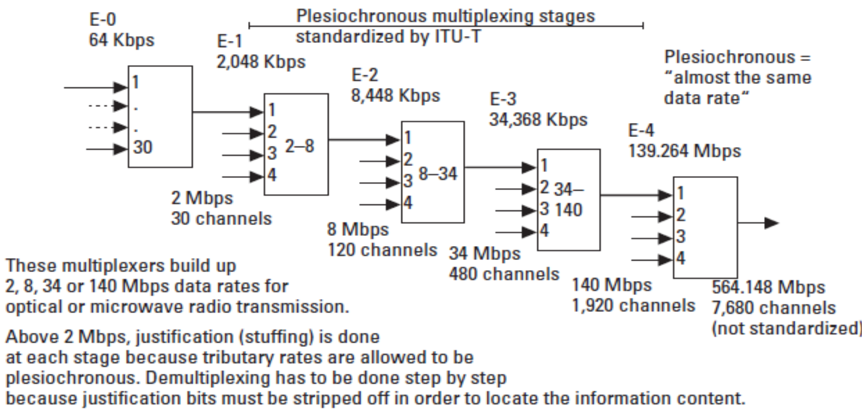


Figure 4.24 The PDH (European standard).

level has a slightly higher rate than four times the nominal tributary rate. Justification bits are added to tributaries so make their data rates equal for framing. The frame also contains some spare bits that can be used, for example, for management data transmission for a network management system. Bits for far-end alarms are included in the frames just as in the 2,048-Kbps frame discussed previously.

The standards for PDH ensure compatibility in multiplexing between systems from different manufacturers. The management functions are not standardized and they differ from manufacturer to manufacturer.

Only the local interfaces and the multiplexing scheme are standardized in PDH. The multiplexers are connected for transmission via standard interfaces at 2, 8, 34, or 140 Mbps to separate line terminal equipment or to a higher-order multiplexer as shown in Figure 4.24. The line interfaces of the line terminals for copper cable, optical fiber, and radio transmission are manufacturer specific so the vendor has to be the same at both ends.

4.6.3.2 North American PDH for Higher-Order Multiplexing

The North American PDH is shown in Figure 4.25. Higher-order rates are DS1C (3.152 Mbps), DS2 (6.132 Mbps), DS3 (44.736 Mbps), and DS4 (274.176 Mbps) [5]. The higher-level multiplexers are named in such a way that we know the DS levels, which are being combined. For example, M13 in Figure 4.25 has inputs from level DS1 and outputs at level DS3.

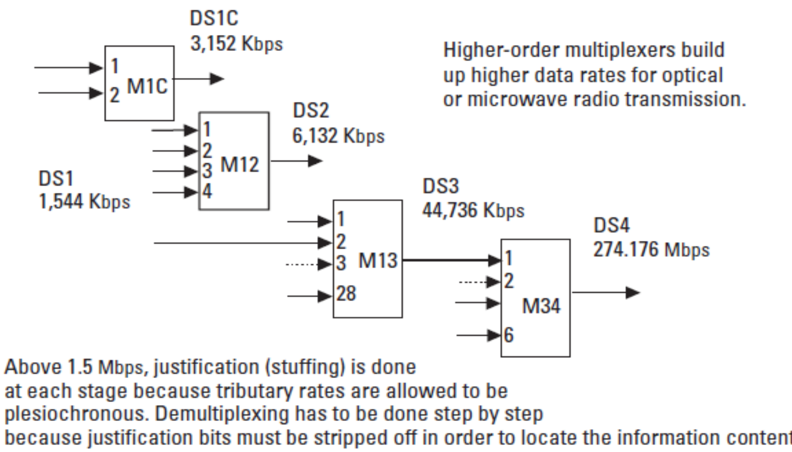


Figure 4.25 North American PDH.

As we can see in Figure 4.25 the higher-order bit rate for each multiplexer is a little bit higher than the sum of the tributary data rates. The aggregate data stream at each level contains, in addition to tributary signals, framing information and the stuffing bits that are used to justify tributary data rates, which may have slightly different data rates, into the higher-order frame. In the demultiplexer these stuffing bits are stripped off and the original tributary rate is produced.

4.6.4 SDH and SONET

The PDH higher-order systems were standardized more than 20 years ago. By the end of the 1980s, a lot of optical fiber cable had been installed and analog networks upgraded into digital networks. Then researchers realized that new standards were required to meet future requirements.

Problems with the PDH standards include the following:

- Access to a tributary rate requires step-by-step demultiplexing because of stuffing (justification).
- Optical interfaces are not standardized but vendor specific.
- To use optical cables, a separate multiplexer for each level (e.g., multiplexing from 2 to 140 Mbps in European PDH requires 21 pieces of multiplexing equipment) and separate line terminals are needed.

- American and European standards are not compatible.
- Network management features and interfaces are vendor dependent.
- High data rates (above 140 or 274 Mbps) are not standardized.

ANSI started to study a new transmission method in the middle of the 1970s to utilize optical networks and modern digital technology more efficiently. This system is called the *synchronous optical network* (SONET) and it is used in the United States.

ITU-T made its own worldwide standard, called SDH, by the end of the 1980s. SDH is actually an international extension of SONET and it was based on SONET but adapted to European networks. A subset of SDH recommendations from the ITU-T was selected as a standard for the European SDH by ETSI. You might say that there are two different synchronous optical systems: SONET in the United States and SDH in areas of Europe where European standards have been adapted. The operating principles of SONET and European SDH are quite similar and they use the same data rate at some levels, as shown in Table 4.3.

Figure 4.26 shows data rates for European SDH as well as an example of SDH equipment. SDH is a standardized multiplexing system for both plesiochronous tributaries, for example, 1.5, 2, or 34 Mbps, and synchronous tributaries.

The main advantages of SDH over PDH standards are as follows:

- The data rates for optical transmission are standardized (i.e., vendor independent).

Table 4.3
Data Rates of SONET (United States) and Corresponding SDH Data Streams (Europe)

OC-N Optical Carrier Level	STS-N Electrical Level	Data Rate (Mbps)	SDH STM-N
OC-1	STS-1	51.84	
OC-3	STS-3	155.52	STM-1
OC-12	STS-12	622.08	STM-4
OC-24	STS-24	1244.16	
OC-48	STS-48	2488.32	STM-16
OC-192	STS-192	9953.28	STM-64

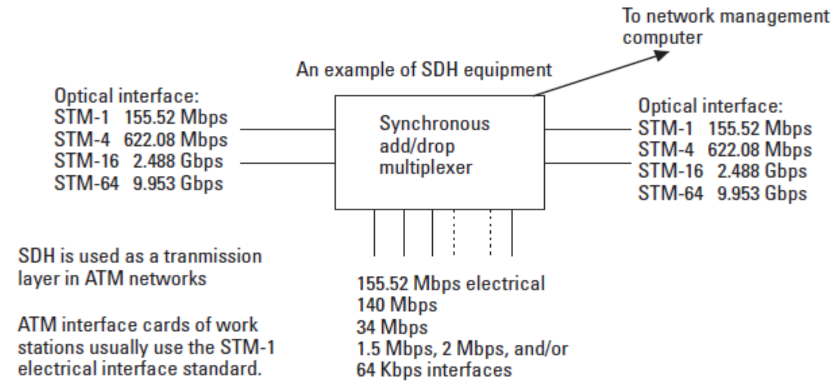


Figure 4.26 The synchronous digital hierarchy of ETSI.

- Different systems are included in standards, for example, terminal, add/drop, and cross-connection systems. These systems are discussed in Section 4.7 and they make SDH networks more flexible than PDH systems, which include only terminal multiplexer functionality.
- Access to the tributary data rates is efficient (no step-by-step multiplexing is required).
- The system is tolerant against synchronization and other system faults. Standardized redundancy functions allow operators to switch from a faulty line to an operational line.
- In the future, network management is slated to become vendor independent, with sophisticated management functions.

SDH is replacing PDH systems in the transport network. By *transport network* we mean the flexible high-capacity transmission network that is used to carry all types of information. By *flexible* we mean that telecommunications operators are able to easily modify the structure of the transport network from the centralized management system. This makes the delivery times for leased lines shorter. Leased lines are needed, for example, for LAN interconnections between the offices of a corporation.

4.6.4.1 Multiplexing Scheme in SDH

The transmission data streams of SDH are called *synchronous transport modules* (STMs) and they are exact multiples of STM-1 at the 155.52-Mbps data rate, as we can see in Table 4.3. STM-1 data are simply byte interleaved with other STM-1 data streams to make up a higher transmission data rate; no additional framing information is added. Byte interleaving means that, for example, an STM-4 signal contains a byte (8 bits) from the first STM-1 tributary, then from the second, third, and fourth tributaries, and then again from the first one. The demultiplexer receives all STM-1 frames independently.

The STM-1 frame is repeated 8,000 times a second, a rate equal to the PCM sampling rate. This makes each 8-bit speech sample visible in a 155.52-Mbps data stream. When PCM coding is synchronized to the same source as SDH systems, we can demultiplex one speech channel just by picking up 1 byte from each STM-1 frame. The frame contains frame alignment information and other information such as management data channels and pointers that tell the location of tributaries in the frame.

If tributaries are not synchronous with the STM-1 frame, a pointer (a binary number) in a fixed location in the STM-1 frame tells the location of each tributary. By looking at the value of this pointer, we can easily find the desired tributary signal. This is a great advantage over PDH systems, which require step-by-step demultiplexing (to separate information and stuffing bits) to the level of the tributary that we want to take out from the high-data-rate stream.

Multiplexing in SDH is quite a complicated matter because the multiplexing supports many different PDH and SDH streams to be multiplexed into an STM-1 stream. For example, a single STM-1 may carry 63 E-1 signals or alternatively one E-4 signal. The STM-1 frame structure and how ATM cells are inserted in it are demonstrated in Chapter 6 as an example of SDH framing. A more detailed treatment of the framing subject is not included here.

4.6.4.2 Data Rates of North American SONET

The *synchronous transport signal level 1* (STS-1) is the basic SONET module that corresponds to STM-1 of SDH. These modules have a bit rate of 51.840 Mbps and they are multiplexed synchronously into higher-order signals STS-N. Each STS-N signal has a corresponding optical signal called an *optical carrier* (OC-N) for optical transmission. Table 4.3 presents data rates for SONET and corresponding signal levels for European SDH.

An STS-1 signal consists of frames and the frame duration is 125 μs (8,000 times a second, that is, equal to the PCM sampling rate) just as in SDH. Each frame contains 810 bytes that makes up a bit rate of 51.840 Mbps. Transport overhead information such as frame synchronization and pointers uses 27 bytes in each frame and the rest of it is used for payload; for example, for 1.544-Mbps signals that contain PCM speech channels. The detailed multiplexing scheme of either SONET or SDH is not presented here; for more detailed information the reader may refer to, for example, [5].

SONET and SDH were originally designed for transmission of 64-Kbps PCM channels. In Chapter 6 we will see how they are used when data consist of IP packets or ATM cells.

4.9 Problems and Review Questions

Problem 4.1

How wide a bandwidth does a pulse with duration of (a) 1 ms and (b) 1 μs require if only the strongest part of the spectrum needs to be transmitted? What is the bandwidth of a carrier wave with a duration of (a) 1 ms and (b) 1 μs?

Problem 4.2

What is continuous wave modulation and why is it often used in transmission systems?

Problem 4.3

(a) Draw the spectrum of a cosine wave at a frequency of 1 kHz. (b) Draw the spectrum of an AM signal when the carrier frequency is 100 kHz and the modulating message is a cosine wave at 1 kHz. (c) Draw the spectrum when the modulation method is SCDSB. (d) Draw the corresponding spectrum of SSB modulation.

Problem 4.4

(a) Draw the constellation diagram (or signal space diagram) for an 8-PSK signal so that the in-phase carrier waveform represents bit combination 000. Write in the diagram which bit combination each signal could represent. Take care that you minimize the bit error rate. (b) Draw the constellation diagram of a 16-QAM signal where a carrier with a 45° phase shift and a high amplitude corresponds to a bit combination of 1100. Write bit combinations for each signal so that the bit error rate is minimized. [*Hint:* Use Gray code for two bits for columns (I component) and two bits for rows (Q component) and combine them for each signal in the constellation.]

Problem 4.5

Explain how the radio wave propagation modes differ at (a) low-frequency, (b) medium frequency, and (c) and ultra high frequency bands.

Problem 4.6

Estimate the transmission capacity of an optical fiber that operates over the 0.9- to 1.6-m wavelength range if coherent optical transmission is used. Assume that the speed of light is the same as in space (300,000 km/sec) and the following modulation methods are in use: (a) Voice signal bandwidth is 4 kHz and it is SSB modulated into the fiber. (b) Voice signal is PCM coded and transmitted in a binary form through the cable. Assume that the modulation scheme in use is capable of transmitting 1 bps/Hz.

Problem 4.7

Derive on your own the formula, $L = [4\pi f l / c]^2$, step by step for the free-space loss (see Section 4.2.6). Use the formula for the effective aperture area of isotropic antenna, $A_{ei} = \lambda^2/(4\pi)$, and a spherical surface area $A = 4\pi l^2$ over which transmitted power is distributed.

Problem 4.8

Show that the equation for radio wave attenuation in decibels, $L_{dB} = 92.4 + 20 \log_{10} f/\mathrm{GHz} + 20 \log_{10} l/\mathrm{km}$ dB, follows from the equation of attenuation $L = [4 f \pi l / c]^2$. Note that, for example, $f = f/\mathrm{GHz} \times 10^9$.

Problem 4.9

The approximate distance between an Earth station and a geostationary satellite is 40,000 km. (a) What is the attenuation of the uplink radio section at the 6-GHz frequency? (b) What is the attenuation in the downlink direction at 4 GHz?

Problem 4.10

Consider a cell in a GSM cellular network operating at 900 MHz and a cell in a DCS-1800 network operating at 1.8 GHz. The DCS-1800 base station is installed in the same site as the GSM base station. Assume that all system parameters except frequency are equal and use the free-space loss formula. What would be the radius of the DCS-1800 cell if the radius of the GSM cell is 1 km?

Problem 4.11

How much higher transmission power is needed, according to the free-space loss formula, if the radio transmission distance is doubled (for the same performance)?

Problem 4.12

A telecommunications network operator is aiming to update a GSM network with DCS-1800 base stations. The cells of GSM (900 MHz) are designed for a maximum transmission power of 1W. What should be the maximum transmission power of DCS-1800 (1.8-GHz) base stations with the same cell structure? Assume here a free-space environment and that the only difference between systems is the frequency.

Problem 4.13

What is the approximate gain of the satellite TV antenna when the diameter of the dish is 0.6m and the frequency is 10 GHz? How much better is the S/N ratio if the antenna is changed to a larger one with diameter of 1m?

Problem 4.14

What is the received power level (dBm) and power (W) when transmitted power is 1W, frequency 1 GHz, distance 1 km, and transmitter and receiver antenna gains are 14 and 2 dB, respectively? Assume a free-space loss approximation for link loss.

Problem 4.15

What are the theoretical maximum symbol rate r and the maximum binary bit rate C through the following baseband channels: (a) bandwidth $B = 3$ kHz and S/N = 20 dB (degraded speech channel); and (b) bandwidth $B = 5$ MHz and S/N = 48 dB (typical video channel)?

Problem 4.16

How many bits can be encoded into each symbol in the case of baseband systems (a) and (b) in Problem 4.15? How much higher is the data rate in case (b) in Problem 4.15 because of the wider bandwidth and how much higher is the bit rate because of the improved S/N compared with the channel in case (a) of Problem 4.15?

Problem 4.17

Estimate how many symbol values (signals in the constellation diagram) there should be in the case of a 28.8-Kbps modem using QAM if the symbol rate is 3,200 bauds.

Problem 4.18

Why do we perform line encoding before data are transmitted to the transmission channel?

Problem 4.19

Explain how binary values 1 and 0 are represented in the following codes: (a) NRZ, (b) RZ, (c) AMI, and (d) Manchester.

Problem 4.20

Explain the operating principle of a regenerator (regenerative repeater).

Problem 4.21

What are the main two multiplexing methods and how do they operate?

Problem 4.22

Explain the structure of a 2-Mbps PCM frame.

Problem 4.23

Explain the structure of a 1.5-Mbps PCM frame.

Problem 4.24

Explain what is PDH?

Problem 4.25

What is SDH and what advantages does it provide over PDH?

Problem 4.26

The measured attenuation at 1 MHz of a 1-km copper cable pair is 18 dB. What is the approximate attenuation at (a) 250 kHz, (b) 500 kHz, (c) 2 MHz, and (d) 4 MHz?

Problem 4.27

What are the advantages of (a) optical transmission, (b) microwave radio transmission, and (c) satellite transmission? Compare their characteristics.

Problem 4.28

What do we mean by *dispersion* in optical fibers?

Problem 4.29

What do we mean by *dense wavelength-division multiplexing*?