Chapter 1

Introduction to Communications Systems

1.1 An Introduction to Communications Systems

One of the first communications networks known was built by Mediterranean cultures more than 1,000 years ago and consisted of a series of successive towers with a distance of about 5 to 12 km between them. A message could be coded and transmitted from the first tower to the second one by using optical signals, and then be passed on along the line until it reached its final destination.

In this primitive system we can already identify all the elements of a genuine communications network (see Figure 1.1):

- *Information* consists of the messages interchanged between final users. In order to be introduced into the network, information needs to be coded into signals.
- *Signals* are a physical magnitude, specific for each transmission medium, that change with respect to time.
- The *transmission medium* consists of the links that connect distant nodes.
- *Nodes* are those network elements that receive the signals and retransmit them further along until reaching the final users.

In other words, in a telecommunications network, user information is distributed as signals from one point to another through the transmission medium that connects the nodes in the system.

1.1.1 Signals and Information

The messages to be transmitted are meaningful for the users and are structured hierarchically in lexical, syntactic, and semantic layers, in line with the grammar of the natural language used, whereas signals, by comparison, are only meaningful inside the telecommunications network. The signals used in telecommunications systems can be of two types (see Table 1.1):

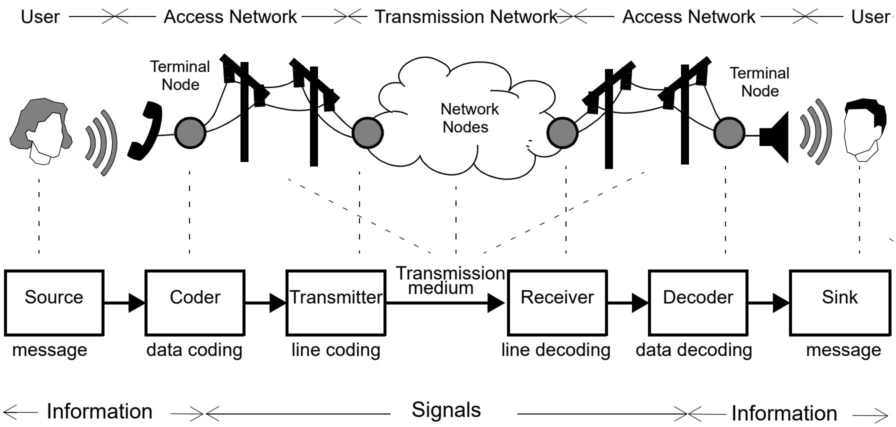

Figure 1.1 Elements of a telecommunications network.

1. *Analog or continuous*: They can take any of an unlimited number of values within a given range.
2. *Digital or discrete*: They can only take a limited number of values. In a binary system, the only valid values are 0 and 1.

Table 1.1
Combinations of signals and information.

Signal	*Analog Information*	*Digital Information*
Analog	Modulation (e.g., AM/FM radio and TV)	Digital modulation (e.g., ADSL)
Digital	Digitalization (e.g., audio CD, GSM)	Coding (e.g., frame relay)

1.1.2 Transmission Medium

The transmission medium can be defined as the environment where a signal is transmitted, be it material (electrical wires, optical fiber, open air, etc.) and nonmaterial, or vacuum, through which only electromagnetic waves are propagated.

The material transmission medium can be divided into two main groups:

1. A *conductive medium*, in which the information is transmitted in the form of electrical impulses. Typical examples of this medium are twisted-pair and coaxial cables.
2. A *dielectric medium*, in which the information is transmitted in the form of radioelectrical or optical signals; for example, the atmosphere and optical fiber.

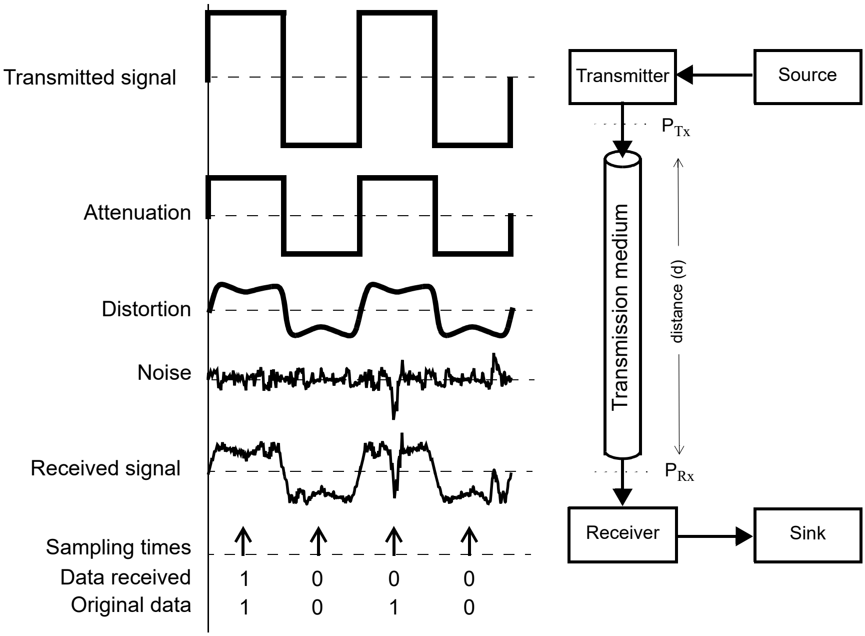

Figure 1.2 Effects of attenuation, distortion, and noise on transmission.

The propagation of signals over one of these media is what we call transmission. The success of transmission of information in telecommunications networks depends basically on two factors: the quality of the signal transmitted, and the quality of the transmission medium used. In addition, there are natural forces that can resist transmission and modify the original characteristics of the signals, which may end up being degraded by the time they reach their destination.

The most significant impairments are attenuation, noise, and distortion. We look at these below in respect to a communications channel, which is defined as a means of unidirectional transmission of signals between two points.

1.1.2.1 Attenuation

Attenuation weakens the power of the signal proportionally to the transmission medium length. It is expressed in decibels (A_{dB}) through the logarithmic ratio of the transmitted power (P_{Tx}) and received power (P_{Rx}), measured at both ends of the distance (d) being examined (see Figure 1.2). Transmission media can usually be characterized by their attenuation per unit of length ($A_{dB\ /\ Km}$):

$$10log(P_{Tx}/P_{Rx}) = d \cdot A_{dB/Km}$$
$$A_{dB} = d \cdot A_{dB/Km}$$

Example: Thus for a transmission medium with A=0.2 dB/Km, after 15 Km, the attenuation is A_{dB}=3 dB. If the transmitted power is P_{Tx}=1W. After 10 Km received power is P_{Rx}= 0.5W, because 10 log (1/P_{Rx}) = 3 dB (see Figure 1.3).

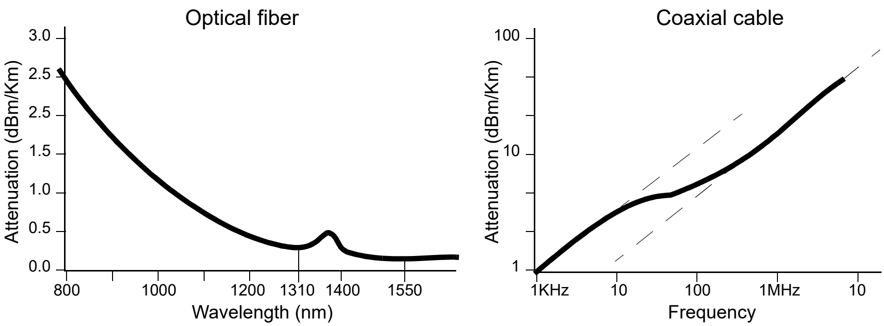

Figure 1.3 Typical attenuation values for single mode optical fiber and coaxial cable.

At the far end the received signal must have enough power (P_{Rx}) to be interpreted, otherwise amplifiers (also known as repeaters or regenerators in digital transmission) must be inserted along the transmission medium to improve the power of the received signal.

1.1.2.2 Distortion

Distortion produces a change in the original shape of the signal at the receiver end. There are two types: amplitude distortion and delay distortion.

- When the impairments affect the amplitudes of the frequency components of the signal differently, this is said to produce *amplitude distortion* (sometimes called absorption). Amplitude distortion is caused because the transmission channel is limited to certain frequencies (see Figure 1.4). To overcome this problem amplifiers must equalize the signal, separately amplifying each band of frequencies.[1]

1. Note that attenuation is a specific case of amplitude distortion that equally affects all frequencies of the signal.

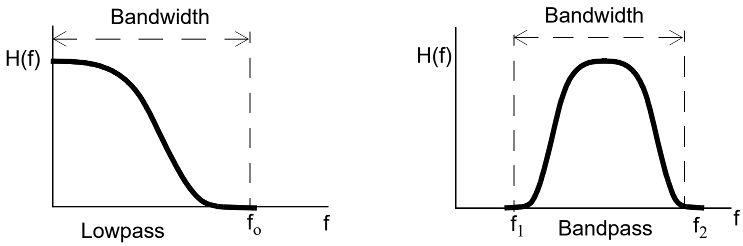

Figure 1.4 The two basic transmission channels. In the frequency domain the channel transfer function H(f) determines the attenuation of each frequency and consequently the amplitude distortion.

- When the velocity of propagation of a signal varies with the frequency, there is said to be *delay distortion* (sometimes called dispersion). Delay distortion is particularly disturbing in the digital transmission producing *intersymbol interference* (ISI), where a component of the signal of one bit is misplaced in the time slot reserved for another bit. ISI limits the capacity to extract digital information from the received signal.

Harry Nyquist showed that the maximum transmission capacity (C) is limited by ISI and depends on the channel bandwidth (B) and the number of signal elements (M) coding the information.

$$C_{bps} = 2B\log_2 M$$

Example: For a modem using 16 signal elements and a channel bandwidth (B) of 4,000 hertz (Hz), the maximum data transfer rate (C) is 32,000 bits per second (bps).

1.1.2.3 Noise

Noise refers to any undesired and spurious signal that is added to an information signal. It is usually divided into five categories:

1. *Thermal noise:* This is caused by the agitation of electrons in any conductor in a temperature different than absolute zero. The noise (N) is independent of the frequency and proportional to the bandwidth (B) and the temperature (T) in degrees Kelvin:

   $$N = k \cdot T \cdot B$$

   (k is the Boltzmann's constant in joules/kelvin, k = 1.3803 x 10^{-23})

2. *Intermodulation noise:* This is caused when two or more signals of frequencies f_1 and f_2, transmitted in the same medium, produce a spurious signal at frequencies that are a linear combination of the previous ones.
3. *Atmospheric noise:* This is caused by the static discharge of clouds, or ionized gas from the sun, or high frequency signals radiated by the stars.
4. *Impulse noise:* Of short duration but high amplitude, these energy bursts are caused by sources such as electrical machinery, a drop in voltage, atmospheric interference, and so on. These do not tend to be a problem for analog signals, but are a prime cause of errors in digital transmission.
5. *Crosstalk:* Whenever a current flows through a conductor a magnetic field is set up around it that can induct a current into a second conductor collocated in a short distance.

Noise is always present in transmission channels, even when no signal is being transmitted. A key parameter at the receiver end to distinguish between information and spurious power is the signal-to-noise ratio (S/N):

$$(S/N)_{dB} = 10log(Power_{Signal}/Power_{Noise})$$

Claude Shannon proved that the signal-to-noise ratio (S/N) determines the theoretical maximum transmission capacity (C) in bits per second of channel with a limited bandwidth (B):

$$C_{bps} = Blog_2(1 + S/N)$$

Example: A typical value of S/N for a voice grade line is 30 dB (equivalent to a power ratio of 1,000:1). Thus for a bandwidth of 3,100 Hz the maximum data transfer rate (C) should be 30,894 bps.

If we pay attention only to the Nyquist formula (see Section 1.1.2.2) we could inaccurately conclude that for a given bandwidth (B) the data rate can be increased endlessly, by increasing the number of signal elements. However in reality, the signal-to-noise ratio sets up the theoretical limit of the channel capacity.

The Shannon theorem makes no statement as to how the channel capacity is achieved. In fact, channels only approach this limit. The task of providing high channel efficiency is the goal of coding techniques.

1.1.2.4 The transmission channel

A digital channel is a communication subsystem with capacity to send and receive information between two points: a source and a sink. Related concepts are:

- *Bandwidth*, expressed in hertz (Hz). This is the difference between the highest and the lowest frequency that can be transmitted across a line or a network.
- *Data rate*, expressed in bits per second (bps). This is a measure of the speed with which information is transferred. It depends on the bandwidth, transmission medium impairments, and the technological capacity to efficiently use the available bandwidth.
- *Performance*, expressed in bit error rate (BER). This is the probability of a single bit being corrupted in a defined interval. Performance is on indication of the quality of the channel.

Channel capacity is the data rate that can be transmitted over a communication path under specific conditions.When two channels define a two-way communication, it is more usual to talk about a circuit.

1.1.3 Channel Coding

Channel coding is the process that transforms binary data bits into signal elements that can cross the transmission medium. In the simplest case, in a metallic wire a binary 0 is represented by a lower voltage, and a binary 1 by a higher voltage. However, before selecting a coding scheme it is necessary to identify some of the strengths and weaknesses of line codes:

- *High-frequency* components are not desirable because they require more channel bandwidth, suffer more attenuation, and generate crosstalk in electrical links.
- *Direct current* (dc) components should be avoided because they require physical coupling of transmission elements. Since the earth/ground potential usually varies between remote communication ends, dc provokes unwanted earth-return loops.
- The use of *alternating current* (ac) signals permits a desirable physical isolation using condensers and transformers.
- *Timing control* permits the receiver to correctly identify each bit in the transmitted message. In synchronous transmission, the timing is referenced to the transmitter clock, which can be sent as a separate clock signal, or embedded into the line code. If the second option is used, then the receiver can extract its clock from the incoming data stream thereby avoiding the installation of an additional line.

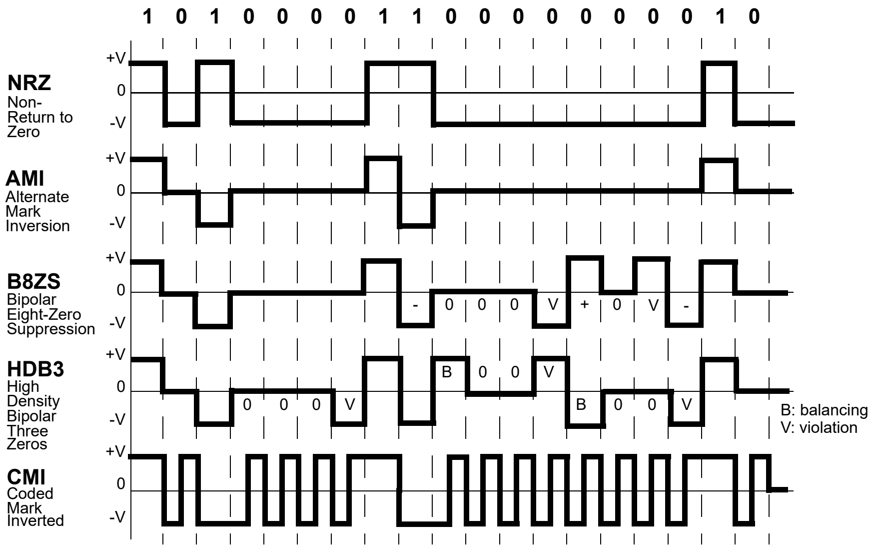

Figure 1.5 Line encoding technologies. AMI and HDB3 are usual in electrical signals, while CMI is often used in optical signals.

In order to meet these requirements, line coding is needed before the signal is transmitted, along with the corresponding decoding process at the receiving end. There are a number of different line codes that apply to digital transmission, the most widely used ones are alternate mark inversion (AMI), high-density bipolar three zeros (HDB3), and coded mark inverted (CMI).

1.1.3.1 Non-return to zero

Non-return to zero (NRZ) is a simple method consisting of assigning the bit "1" to the positive value of the signal amplitude (voltage), and the bit "0" to the negative value (see Figure 1.5). There are two serious disadvantages to this:

1. No timing information is included in the signal, which means that synchronism can easily be lost if, for instance, a long sequence of zeros is being received.
2. The spectrum of the signal includes a dc component.

1.1.3.2 Alternate mark inversion

Alternate mark inversion (AMI) is a transmission code, also known as pseudoternary, in which a "0" bit is transmitted as a null voltage and the "1" bits are represented alternately as positive and negative voltage. The digital signal coded in AMI is characterized as follows (see Figure 1.5):

- The dc component of its spectrum is null.
- It does not solve the problem of loss of synchronization with long sequences of zeros.

1.1.3.3 Bit eight-zero suppression

Bit eight-zero suppression (B8ZS) is a line code in which bipolar violations are deliberately inserted if the user data contains a string of eight or more consecutive zeros. The objective is to ensure a sufficient number of transitions to maintain the synchronization when the user data stream contains a large number of consecutive zeros (see Figure 1.5 and Figure 1.6).

The coding has the following characteristics:

- The timing information is preserved by embedding it in the line signal, even when long sequences of zeros are transmitted, which allows the clock to be recovered properly on reception
- The dc component of a signal that is coded in B8Z3 is null.

B8ZS

		Substitution
Last pulse polarity	+	$000V^{+}{-}0V_{-}{+}$
	–	$000V_{-}{+}0V^{+}{-}$

HDB3

		Number of ones	
		Odd	Even
Last '1' polarity	+	$B_{-}00V_{-}$	$000V^{+}$
	–	$000V_{-}$	$B^{+}00V^{+}$

Figure 1.6 B8ZS and HDB3 coding. Bipolar violations are: V^{+} a positive level and V_{-} negative.

1.1.3.4 High-density bipolar three zeroes

High-density bipolar three zeroes (HDB3) is similar to B8ZS, but limits the maximum number of transmitted consecutive zeros to three (see Figure 1.5). The basic idea consists of replacing a series of four bits that are equal to "0" with a code word "000V" or "B00V," where "V" is a pulse that violates the AMI law of alternate polarity, and B it is for balancing the polarity.

- "B00V" is used when, until the previous pulse, the coded signal presents a dc component that is not null (the number of positive pulses is not compensated by the number of negative pulses).
- "000V" is used under the same conditions as above, when, until the previous pulse, the dc component is null (see Figure 1.6).
- The pulse "B" (for balancing), which respects the AMI alternation rule and has positive or negative polarity, ensuring that two consecutive "V" pulses will have different polarity.

1.1.3.5 Coded mark inverted

The c*oded mark inverted* (CMI) code, also based on AMI, is used instead of HDB3 at high transmission rates, because of the greater simplicity of CMI coding and decoding circuits compared to the HDB3 for these rates. In this case, a "1" is transmitted according to the AMI rule of alternate polarity, with a negative level of voltage during the first half of the period of the pulse, and a positive level in the second half. The CMI code has the following characteristics (see Figure 1.5):

- The spectrum of a CMI signal cancels out the components at very low frequencies.
- It allows for the clock to be recovered properly, like the HDB3 code.
- The bandwidth is greater than that of the spectrum of the same signal coded in AMI.

1.1.4 Multiplexing and Multiple Access

Multiplexing is defined as the process by which several signals from different channels share a channel with greater capacity (see Figure 1.7). Basically, a number of channels share a common transmission medium with the aim of reducing costs and complexity in the network. When the sharing is carried out with respect to a remote resource, such as a satellite, this is referred to as multiple access rather than multiplexing.

Some of the most common multiplexing technologies are:

1. *Frequency division multiplexing/frequency division multiple access* (FDM/FDMA): Assigns a portion of the total bandwidth to each of the channels.
2. *Time-division multiplexing/time division multiple access* (TDM/TDMA): Assigns all the transport capacity sequentially to each of the channels.
3. *Code-division multiplexing access* (CDMA): In certain circumstances, it is possible to transmit multiple signals in the same frequency, with the receiver being responsible for separating them. This technique has been used for years

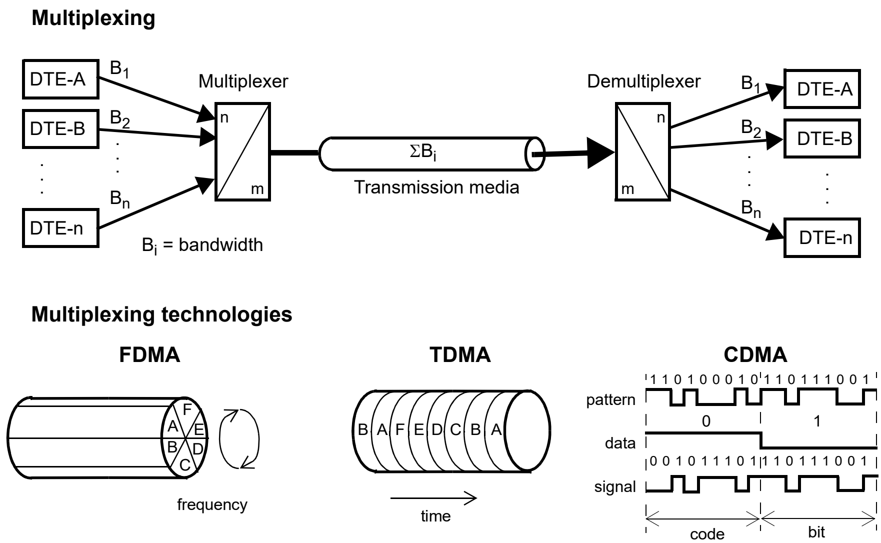

Figure 1.7 Multiplexing consolidates lower capacity channels into a higher capacity channel. Frequency division multiplexing access (FMDA) is used by radio, TV, and global system mobile (GSM). Time division multiplexing access (TDMA) is used by the integrated services digital network (ISDN), frame relay (FRL), and GSM. Code division multiplexing access (CDMA) is used by the third generation networks (3G) of mobiles.

in military technology, and is based on artificially increasing the bandwidth of the signal according to a predefined pattern.

4. *Polarization division multiple access* (PDMA): Given that polarization can be maintained, the polarization direction can be used as a multiple access technique, although when there are many obstacles, noise can make it unsuitable, which is why it is not generally used in indoor installations. Outside, however, it is widely exploited to increase transmission rates in installations that use microwaves.
5. *Space division multiple access* (SDMA): With directional antennas, the same frequency can be reused, provided the antennas are correctly adjusted. There is a great deal of interference, but this system lets frequencies obtain a high degree of reusability.

1.2 Pulse Code Modulation

The *pulse code modulation* (PCM) technology (see Figure 1.8) was patented and developed in France in 1938, but could not be used because suitable technology was not available until World War II. This came about with the arrival of digital

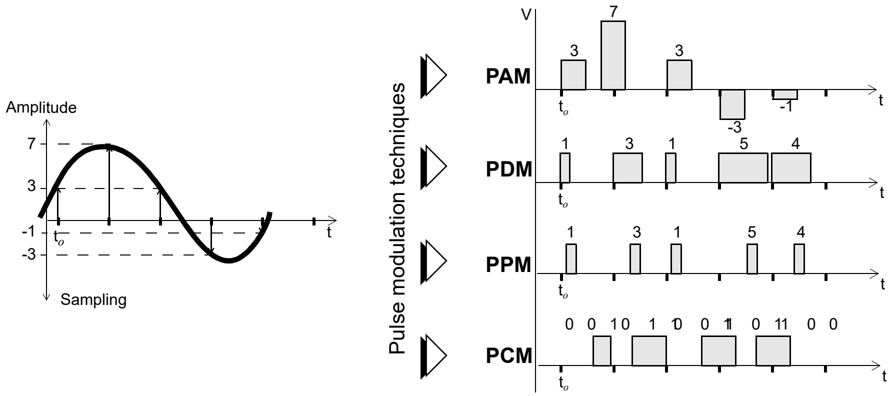

Figure 1.8 Pulse code modulation (PCM) was the technology selected to digitalize the voice in telephone networks. Other pulse techniques are pulse amplitude modulation (PAM), pulse duration modulation (PDM), and pulse position modulation (PPM).

systems in the 1960s, when improving the performance of communications networks became a real possibility. However, this technology was not completely adopted until the mid-1970s, due to the large amount of analog systems already in place and the high cost of digital systems, as semiconductors were very expensive. PCM's initial goal was that of converting an analog voice telephone channel into a digital one based on the sampling theorem (see Figure 1.9):

The sampling theorem states that for digitalization without information loss, the sampling frequency *(f_s)* should be *at least twice* the maximum frequency component *(f_{max})* of the analog information:

$$f_s > 2 \cdot f_{max}$$

The frequency $2 \cdot f_{max}$ is called the Nyquist sampling rate. The sampling theorem is considered to have been articulated by Nyquist in 1928, and mathematically proven by Shannon in 1949. Some books use the term *Nyquist sampling theorem*, and others use *Shannon sampling theorem*. They are in fact the same theorem.

PCM involves three phases: sampling, encoding, and quantization:

1. In sampling, values are taken from the analog signal every $1/f_s$ seconds (the sampling period).

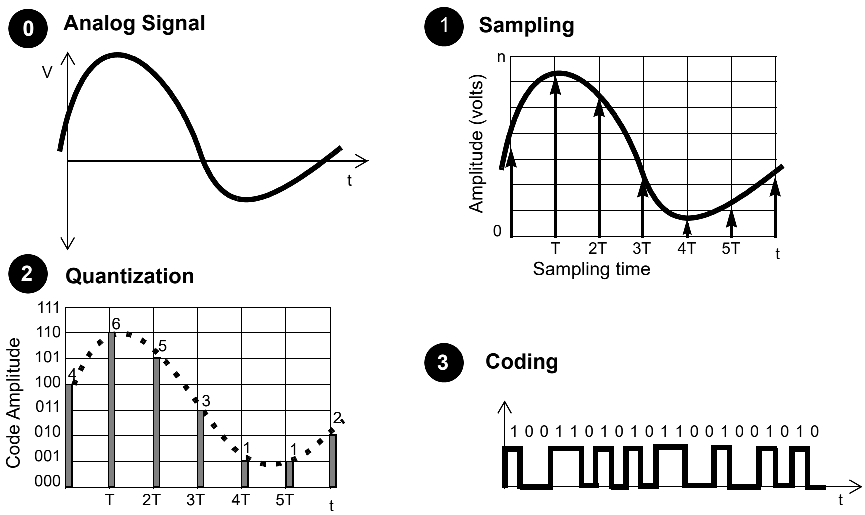

Figure 1.9 The three steps of digitalization of a signal: sampling of the signal, quantization of the amplitude, and binary encoding.

2. Quantization assigns these samples a value by approximation, and in accordance with a quantization curve (i.e., A-law of ITU-T [2]).
3. Encoding provides the binary value of each quantified sample.

A telephone channel admits frequencies of between 300 Hz and 3,400 Hz. Because margins must be established in the channel, the bandwidth is set at 4 kHz. Then the sampling frequency must be $f_s \geq 2 \cdot 4{,}000 = 8{,}000\ Hz$; equivalent to a sample period of $T = 1/8{,}000 = 125\mu s$.

In order to codify 256 levels, 8 bits are needed, where the PCM bit rate (v) is:

$$v = 8{,}000_{\ samples/s} \times 8_{bits/sample} = 64Kbps$$

This bit rate is the subprimary level of transmission networks.

2. This is a International Telecommunication Union (ITU-T) ratified audio encoding and compression technique (Rec. G.711). Among other implementations, A-law was originally intended as a phone-communications standard.

1.3 PDH and T-Carrier

At the beginning of the 1960s, the proliferation of analog telephone lines, based on copper wires, together with the lack of space for new installations, led the transmission experts to look at the real application of PCM digitalization techniques and TDM multiplexing. The first digital communications system was set up by Bell Labs in 1962, and consisted of 24 digital channels running at what is known as T1.

1.3.1 Basic Rates: T1 and E1

In 1965, a standard appeared in the U.S. that permitted the TDM multiplexing of 24 digital telephone channels of 64 Kbps into a 1.544-Mbps signal with a format called T1 (see Figure 1.10). For the T1 signal, a synchronization bit is added to the 24 TDM time slots, in such a way that the aggregate transmission rate is:

$$(24_{channels} \times 8_{bit/channel} + 1_{bit})/125\mu s = 1{,}544Mbps$$

125 μs is the sampling period

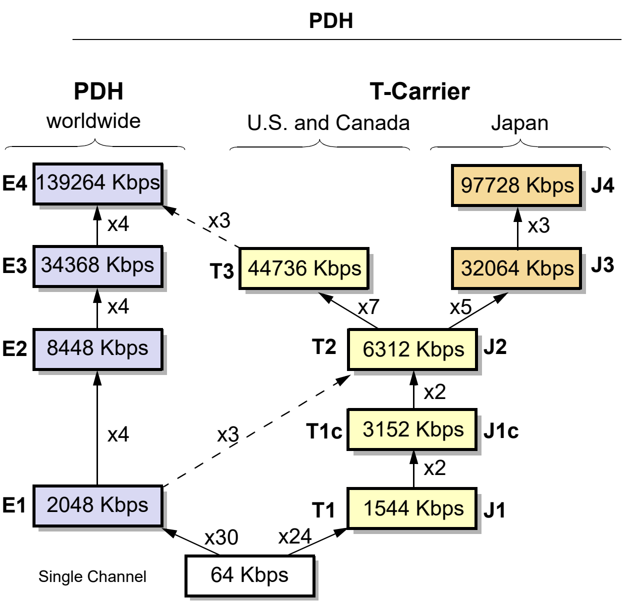

Figure 1.10 The PDH and T-carrier hierarchies, starting at the common 64-Kbps channel and the multiplexing levels. Most of the narrowband networks are built on these standards: POTS, FRL, GSM, ISDN, ATM (asynchronous transfer mode), and leased lines to transmit voice, data, and video.

Europe developed its own TDM multiplexing scheme a little later (1968), although it had a different capacity: 32 digital channels of 64 Kbps (see Figure 1.10). The resulting signal was transmitted at 2.048 Mbps, and its format was called E1 which was standardized by the ITU-T and adopted worldwide except in the U.S., Canada, and Japan. For an E1 signal, the aggregate transmission rate can be obtained from the following equation:

$$(30_{channels} \times 8_{bit/channel})/125\mu s = 2{,}048Mbps$$

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