



Introduction to Telecommunications Network Engineering

Second Edition

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Second Edition

Tarmo Anttalainen



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Signals Carried over the Network

Services that the telecommunications networks provide have different characteristics. Required characteristics depend on the applications we use. To meet these different requirements, many different network technologies that are optimized for each type of service are in use. To understand the present structure of the telecommunications network, we have to understand what types of signals are transmitted through the telecommunications network and their requirements. In this chapter we look at the requirements of various applications, characteristics of analog voice channels, fundamental differences between analog and digital signals, analog-to-digital conversion, and a logarithmic measure of signal level, the decibel.

3.1 Types of Information and Their Requirements

Modern digital networks transmit digital information transparently; that is, the network does not necessarily need to know what kind of information the data contain. This information that is transmitted through the network may be any one of the following:

- Speech (telephony, fixed, or cellular);
- Moving images (television or video);
- Printed pages or still picture (facsimile or multimedia messaging);
- Text (electronic mail or short text messaging);

- Music;
- All types of computer information such as program files.

For digital transmission, analog signals such as speech are encoded into digital form and transmitted through the network as a sequence of bits in the same way computer files are transmitted. However, although all information is coded into digital form, the transmission requirements are highly dependent on the application; because of these different requirements, different networks and technologies are in use. Video and e-mail applications, for example, require different architectures. Network technologies have taken two main development paths: one for speech services and another for data services. The telephone network and ISDN have been developed for constant-bit-rate voice communication that is well suited to speech transmission. Data networks such as LANs and the Internet have been developed for bursty data transmission.

The constant-bit-rate requirement for speech follows from the principle that digitized voice signals have traditionally been transmitted in digital form as samples at regular intervals, as we will see in Section 3.6. Data transmission is bursty by nature. Sometimes we may copy a file across the network, whereas at other times we may work locally with our workstation.

When many different applications are integrated into multimedia communications, both basic types of service requirements of constant-bit-rate voice and bursty data have to be fulfilled and we need a concept that is able to meet both types of requirements.

In Table 3.1 different applications are compared from the communication requirements point of view. The applications are ordinary speech, *computer-aided design* (CAD) (a service in which high-resolution graphical information is transmitted), moving images (video), file transfer, and multimedia with integrated video, voice, and data. The importance of the transmission requirements for each application is explained next.

Data Rate or Bandwidth Requirement

Voice communication usually requires a constant data rate of 64 Kbps or less and high-resolution video a constant data rate of 2 Mbps or higher over the network. Characteristics of data communication are very different, for example, file transfer requires high-bit-rate transmission only during download, and high-resolution graphics on a Web page require high-data-rate transmission only when we download a new page. When we are reading a Web page we do not need transmission capacity at all. To define data transmission

Table 3.1 Communication Requirements of Different Applications

Transmission Characteristics	Voice	Video	File Transfer	Interactive Media
Bandwidth requirement	Low, fixed	Very high, fixed	High, variable	High, variable
Data loss tolerance	Tolerant	Tolerant	Nontolerant	Tolerant or nontolerant
Fixed delay tolerance	Low delay	Tolerant	Tolerant	Low delay
Variable delay tolerance	No	No	Tolerant	No
Peak information rate	Fixed	Fixed	High	Very high

capacity, we sometimes use the term *bandwidth* instead of *data rate* because these terms are closely related to each other, as we will see in Chapter 4.

Data Loss Tolerance

Noise and other disturbances in the network may cause errors in the transmitted data. If errors occur, some amount of data may be lost. Voice and video transmission services are used by human beings, and they can tolerate accidental short disturbances. In computer communications a single erroneous bit usually destroys a whole data frame, which may contain a large amount of data. The loss of one frame destroys the transmission of a large file that is transferred in multiple frames. Most of the data communication systems are able to retransmit data frames in error. Systems designed for voice or video transmission do not use retransmission schemes because temporary retransmission delay is even more disturbing for human users than the loss of some data.

Fixed Delay Tolerance

When communication is interactive, as voice communication usually is, the two-way transmission delay should be very short for good quality. In the case of voice it should be of the order of some tens of milliseconds. Otherwise, we feel that quality is degraded because the response from the other party is

delayed. We tolerate much longer delays in the case of ordinary data applications when we are waiting for a response to our "click" command.

Variable Delay Tolerance

Voice and video information is traditionally transmitted as samples at regular periods of time. The reconstruction of images and voice requires that all sample values be received sequentially and suffer the same delay. Conventional data networks recover from errors with the help of retransmission of the frames in error. This is a very efficient error recovery scheme, but it introduces some additional and variable delay. For voice applications this variable delay is often a worse solution than that of losing some data.

Peak Information Rate

Encoding of analog voice and video often produces a constant information data rate. Values of the samples with constant length contain voice or video information and they are transmitted at a constant rate. In data communication applications we usually work locally and every now and then a high data rate is needed to load graphical information or files. A peak load is typically of the order of 1,000 times higher than the average transmission capacity we use.

The different requirements just explained have supported development of the circuit-switched networks, such as PSTN and ISDN, for voice communications and packet-switched networks, such as LANs and the Internet, for data communications. *Asynchronous transfer mode* (ATM) technology was developed by ITU-T to be suitable and efficient for transferring all types of information. However, the expansion of the Internet has reduced its importance and Internet technology will be developed further to provide a platform for all kinds of communications.

3.2 Simplex, Half-Duplex, and Full-Duplex Communication

In telecommunications systems the transmission of information may be unidirectional or bidirectional. The unidirectional systems that transmit in one direction only are called *simplex*, and the bidirectional systems that are able to transmit in both directions are called *duplex* systems. We can implement bidirectional information transfer with *half-* or *full-duplex* transmission as shown in Figure 3.1.

In simplex operation the signal is transmitted in one direction only. An example of this principle is broadcast television, where TV signals are sent

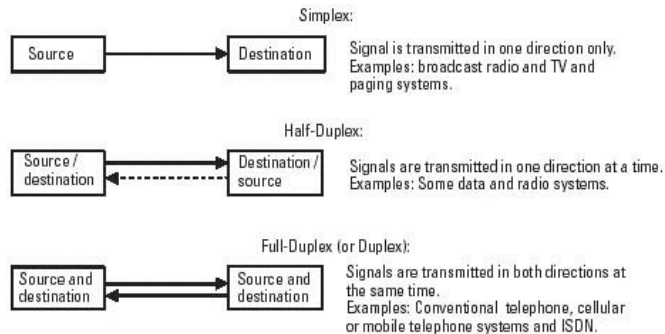


Figure 3.1 Simplex, half-duplex, and full-duplex transmission

from a transmitter to TV sets only and not in the other direction. Another example is a paging system that allows a user to receive only alphanumeric messages.

In half-duplex operation the signal is transmitted in both directions but only in one direction at a time. An example of this is a mobile radio system where the person speaking must indicate by saying the word *over* that she is done transmitting and the other person is allowed to transmit. LANs use a high-speed, half-duplex transmission over the cable even though users may feel that the communication is continuously bidirectional, that is, full duplex.

In full-duplex operation signals are transmitted in both directions at the same time. An example of this is an ordinary telephone conversation where it is possible for both people to speak simultaneously. Most modern telecommunications systems use the full-duplex principle, which we call *duplex operation* for short.

3.3 Frequency and Bandwidth

To understand the requirements of different applications for a telecommunications network, we must understand the fundamental concepts of frequency and bandwidth. The information that we transmit through a telecommunications network, whether it is analog or digital, is in the form of electrical voltage or current. The value of this voltage or current changes through time, and this alteration contains information.

The transmitted signal (the alteration of voltage or current) consists of multiple frequencies. The range of frequencies is called the *bandwidth* of the

signal. The bandwidth is one of the most important characteristics of analog information and it is also the most important limiting factor for the data rate of digital information transfer.

3.3.1 Frequency

We can see the telecommunications signal as a combination of many cosine or sine waves with different strengths and frequencies. The frequency refers to the number of cycles through with the wave oscillates in a second. As an example of the concept of frequency, we hear the oscillation of air pressure as sound. We are able to hear frequencies in the range of approximately 20 Hz to 15 kHz, where Hz (hertz) represents the number of cycles in a second. An example of the different frequencies is heard in the keys of a piano. The right-hand keys generate basic frequencies of the order of 1,000 Hz and the left-hand keys of the order of 100 Hz.

In electrical terms, an *alternating current* (ac) changes its direction of flow several times per second. This variation in direction is known as a *cycle*, and the term *frequency* refers to the number of cycles in a second that is measured in hertz. If a signal has 1,000 complete cycles in a second, then its frequency is 1,000 Hz or 1 kHz. A pure sine wave, like that shown in Figure 3.2, is generated with a loop of wire rotated in a magnetic field at a constant rate. This fundamental waveform can be seen as a cosine of the

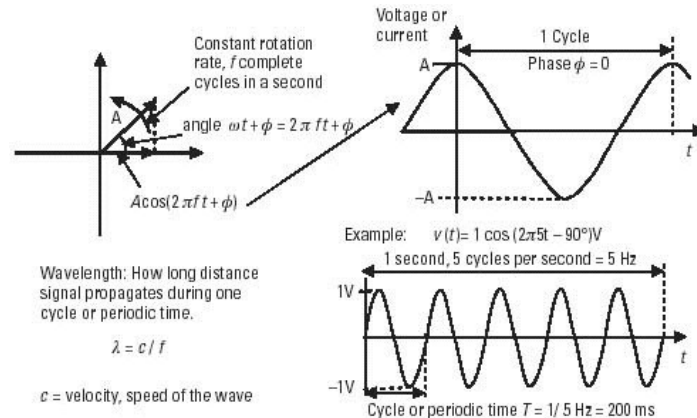


Figure 3.2 Cosine wave and frequency

angle of the phasor rotating at a constant rate. The strength of the voltage or current alters according to the cosine curve when time increases. The length of the phasor corresponds to the maximum value of the signal and it is called *amplitude*, shown as A in Figure 3.2.

We can see any telecommunications signal as a sum of these fundamental waveform cosine waves that are expressed as

$$v(t) = A \cos(\omega t + \phi) = A \cos(2\pi f t + \phi) \quad (3.1)$$

where f is frequency, the number of complete cycles in a second expressed in hertz, $1 \text{ Hz} = 1/\text{sec}$; t is time in seconds, and ϕ is the phase shift (phase of the cosine wave at time instant $t = 0$). The angular frequency ω in radians per second is $\omega = 2\pi f$, which comes from the fact that one complete cycle of a phasor makes up an angle of 2π radians.

The *periodic time* or *period* T in seconds represents the time of one complete cycle:

$$T = 1/f \text{ and } f = 1/T \quad (3.2)$$

Wavelength λ represents the propagation distance in one cycle time, thus,

$$\lambda = c/f = cT \quad (3.3)$$

where c is the velocity of the signal. For a sound wave, the velocity in the air is approximately 346 m/s; for light or radio waves, approximately, $c = 300,000 \text{ km/sec}$.

The example in Figure 3.2 shows a waveform with a frequency of 5 Hz and amplitude of 1 V. It corresponds to a phasor with length of $A = 1 \text{ V}$ making five complete cycles in a second. At time instant $t = 0$, the waveform has a value of 0 and the phase or angle of the phasor is -90° . As the time increases and the phasor rotates, its projection at the horizontal axis of the phasor diagram increases, corresponding to an increase in the value of the wave with time. The equation for this example waveform is then $v(t) = A \cos(\omega t + \phi) = 1 \cos(2\pi 5t - 90^\circ) \text{ V}$.

3.3.2 Bandwidth

The voice signal, which is the most common message in telecommunications network, does not look similar to a pure cosine wave in Figure 3.2. It

contains many cosine waves with different frequencies, amplitudes, and phases combined together. The range of frequencies that is needed for a good enough quality of voice, so that the speaker can be recognized, was defined to be the range from 300 to 3,400 Hz. This means that the bandwidth of the telephone channel through the network is $3,400 - 300 \text{ Hz} = 3.1 \text{ kHz}$, as shown in Figure 3.3. A human voice contains much higher frequencies, but this bandwidth was defined as a compromise between quality and cost. It is wide enough to recognize the speaker, which was one requirement for telephone channel.

Bandwidth is not strictly limited in practice, but signal attenuation increases heavily at the lower and upper cutoff frequencies. For speech, channel cutoff frequencies are 300 and 3.4 kHz, as shown in Figure 3.3. The bandwidth is normally measured from the points where the signal power drops to half from its maximum power. Attenuation or loss of channel is given as a logarithmic measure called a decibel (dB), and half power points correspond to a 3-dB loss. Decibels are discussed later in this chapter.

Bandwidth, together with noise, is the major factor that determines the information-carrying capacity of a telecommunications channel. The term *bandwidth* is often used instead of *data rate* because they are closely related, as we will see in Chapter 4.

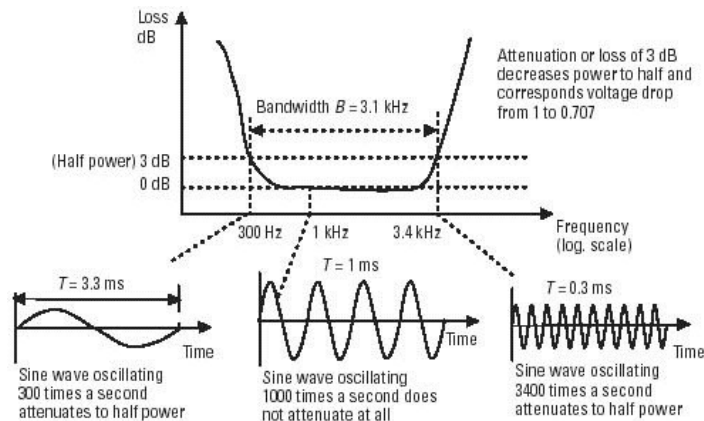


Figure 3.3 Bandwidth of the telephone speech channel