1 INTRODUCTION

In essence, *electronic communications* is the transmission, reception, and processing of information with the use of electronic circuits. *Information* is defined as knowledge or intelligence that is communicated (i.e., transmitted or received) between two or more points. *Digital modulation* is the transmittal of digitally modulated analog signals (carriers) between two or more points in a communications system. Digital modulation is sometimes called *digital radio* because digitally modulated signals can be propagated through Earth's atmosphere and used in wireless communications systems. Traditional electronic communications systems that use conventional analog modulation, such as *amplitude modulation* (AM), *frequency modulation* (FM), and *phase modulation* (PM), are rapidly being replaced with more modern digital moduluation systems that offer several outstanding advantages over traditional analog systems, such as ease of processing, ease of multiplexing, and noise immunity.

Digital communications is a rather ambiguous term that could have entirely different meanings to different people. In the context of this text, digital communications include systems where relatively high-frequency analog carriers are modulated by relatively low-frequency digital information signals (*digital radio*) and systems involving the transmission of digital pulses (*digital transmission*). Digital transmission systems transport information in digital form and, therefore, require a physical facility between the transmitter and receiver, such as a metallic wire pair, a coaxial cable, or an optical fiber cable. In digital radio systems, the carrier facility could be a physical cable, or it could be free space.

The property that distinguishes digital radio systems from conventional analog-modulation communications systems is the nature of the modulating signal. Both analog and digital modulation systems use analog carriers to transport the information through the system. However, with analog modulation systems, the information signal is also analog, whereas with digital modulation, the information signal is digital, which could be computer-generated data or digitally encoded analog signals.

Referring to Equation 1, if the information signal is digital and the amplitude (V) of the carrier is varied proportional to the information signal, a digitally modulated signal called *amplitude shift keying* (ASK) is produced. If the frequency (f) is varied proportional to the information signal, *frequency shift keying* (FSK) is produced, and if the phase of the carrier (θ) is varied proportional to the information signal, *phase shift keying* (PSK) is produced. If both the amplitude and the phase are varied proportional to the information signal, *quadrature amplitude modulation* (QAM) results. ASK, FSK, PSK, and QAM are all forms of digital modulation:

$$v(t) = \underbrace{V}_{\text{ASK}} \sin(2\pi \cdot \underbrace{f}_{\text{FSK}} t + \underbrace{\boldsymbol{\theta}}_{\text{PSK}}) \qquad \text{ASK} + \text{PSK} \rightarrow \text{QAM} \tag{1}$$

Digital modulation is ideally suited to a multitude of communications applications, including both cable and wireless systems. Applications include the following: (1) relatively low-speed voice-band data communications modems, such as those found in most personal computers; (2) high-speed data transmission systems, such as broadband *digital subscriber lines* (DSL); (3) digital microwave and satellite communications systems; and (4) cellular telephone *Personal Communications Systems* (PCS).

Figure 1 shows a simplified block diagram for a digital modulation system. In the transmitter, the precoder performs level conversion and then encodes the incoming data into groups of bits that modulate an analog carrier. The modulated carrier is shaped (fil-

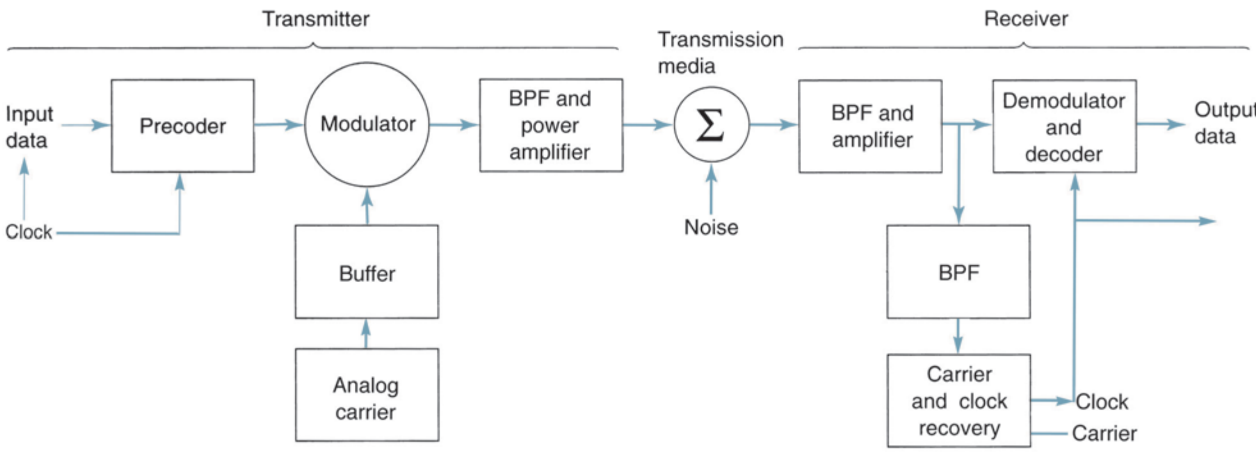

FIGURE 1 Simplified block diagram of a digital radio system

tered), amplified, and then transmitted through the transmission medium to the receiver. The transmission medium can be a metallic cable, optical fiber cable, Earth's atmosphere, or a combination of two or more types of transmission systems. In the receiver, the incoming signals are filtered, amplified, and then applied to the demodulator and decoder circuits, which extracts the original source information from the modulated carrier. The clock and carrier recovery circuits recover the analog carrier and digital timing (clock) signals from the incoming modulated wave since they are necessary to perform the demodulation process.

2 INFORMATION CAPACITY, BITS, BIT RATE, BAUD, AND *M*-ARY ENCODING

2-1 Information Capacity, Bits, and Bit Rate

Information theory is a highly theoretical study of the efficient use of bandwidth to propagate information through electronic communications systems. Information theory can be used to determine the *information capacity* of a data communications system. Information capacity is a measure of how much information can be propagated through a communications system and is a function of bandwidth and transmission time.

Information capacity represents the number of independent symbols that can be carried through a system in a given unit of time. The most basic digital symbol used to represent information is the *binary digit,* or *bit*. Therefore, it is often convenient to express the information capacity of a system as a *bit rate.* Bit rate is simply the number of bits transmitted during one second and is expressed in *bits per second* (bps).

In 1928, R. Hartley of Bell Telephone Laboratories developed a useful relationship among bandwidth, transmission time, and information capacity. Simply stated, Hartley's law is

$$I \propto B \times t \tag{2}$$

where I = information capacity (bits per second)
B = bandwidth (hertz)
t = transmission time (seconds)

From Equation 2, it can be seen that information capacity is a linear function of bandwidth and transmission time and is directly proportional to both. If either the bandwidth or the transmission time changes, a directly proportional change occurs in the information capacity.

In 1948, mathematician Claude E. Shannon (also of Bell Telephone Laboratories) published a paper in the *Bell System Technical Journal* relating the information capacity of a communications channel to bandwidth and *signal-to-noise ratio.* The higher the signal-to-noise ratio, the better the performance and the higher the information capacity. Mathematically stated, *the Shannon limit for information capacity* is

$$I = B \log_2\left(1 + \frac{S}{N}\right) \tag{3}$$

or

$$I = 3.32B \log_{10}\left(1 + \frac{S}{N}\right) \tag{4}$$

where I = information capacity (bps)
B = bandwidth (hertz)
$\frac{S}{N}$ = signal-to-noise power ratio (unitless)

For a standard telephone circuit with a signal-to-noise power ratio of 1000 (30 dB) and a bandwidth of 2.7 kHz, the Shannon limit for information capacity is

$$\begin{aligned} I &= (3.32)(2700) \log_{10} (1 + 1000) \\ &= 26.9 \text{ kbps} \end{aligned}$$

Shannon's formula is often misunderstood. The results of the preceding example indicate that 26.9 kbps can be propagated through a 2.7-kHz communications channel. This may be true, but it cannot be done with a binary system. To achieve an information transmission rate of 26.9 kbps through a 2.7-kHz channel, each symbol transmitted must contain more than one bit.

2-2 *M*-ary Encoding

M-ary is a term derived from the word *binary. M* simply represents a digit that corresponds to the number of conditions, levels, or combinations possible for a given number of binary variables. It is often advantageous to encode at a level higher than binary (sometimes referred to as *beyond binary* or *higher-than-binary encoding*) where there are more than two conditions possible. For example, a digital signal with four possible conditions (voltage levels, frequencies, phases, and so on) is an M-ary system where $M = 4$. If there are eight possible conditions, $M = 8$ and so forth. The number of bits necessary to produce a given number of conditions is expressed mathematically as

$$N = \log_2 M \tag{5}$$

where N = number of bits necessary
M = number of conditions, levels, or combinations possible with N bits

Equation 5 can be simplified and rearranged to express the number of conditions possible with N bits as

$$2^N = M \tag{6}$$

For example, with one bit, only $2^1 = 2$ conditions are possible. With two bits, $2^2 = 4$ conditions are possible, with three bits, $2^3 = 8$ conditions are possible, and so on.

2-3 Baud and Minimum Bandwidth

Baud is a term that is often misunderstood and commonly confused with bit rate (bps). Bit rate refers to the rate of change of a digital information signal, which is usually binary. Baud, like bit rate, is also a rate of change; however, baud refers to the rate of change of a signal on the transmission medium after encoding and modulation have occurred. Hence, baud is a unit of transmission rate, modulation rate, or symbol rate and, therefore, the terms *symbols per second* and *baud* are often used interchangeably. Mathematically, baud is the reciprocal of the time of one output *signaling element,* and a signaling element may represent several information bits. Baud is expressed as

$$\text{baud} = \frac{1}{t_s} \tag{7}$$

where baud = symbol rate (baud per second)
t_s = time of one signaling element (seconds)

A signaling element is sometimes called a *symbol* and could be encoded as a change in the amplitude, frequency, or phase. For example, binary signals are generally encoded and transmitted one bit at a time in the form of discrete voltage levels representing logic 1s (highs) and logic 0s (lows). A baud is also transmitted one at a time; however, a baud may represent more than one information bit. Thus, the baud of a data communications system may be considerably less than the bit rate. In binary systems (such as binary FSK and binary PSK), *baud* and *bits per second* are equal. However, in higher-level systems (such as QPSK and 8-PSK), bps is always greater than baud.

According to H. Nyquist, binary digital signals can be propagated through an ideal noiseless transmission medium at a rate equal to two times the bandwidth of the medium. The minimum theoretical bandwidth necessary to propagate a signal is called the minimum *Nyquist bandwidth* or sometimes the minimum *Nyquist frequency.* Thus, $f_b = 2B$, where f_b is the bit rate in bps and B is the *ideal Nyquist bandwidth.* The actual bandwidth necessary to propagate a given bit rate depends on several factors, including the type of encoding and modulation used, the types of filters used, system noise, and desired error performance. The ideal bandwidth is generally used for comparison purposes only.

The relationship between bandwidth and bit rate also applies to the opposite situation. For a given bandwidth (B), the highest theoretical bit rate is $2B$. For example, a standard telephone circuit has a bandwidth of approximately 2700 Hz, which has the capacity to propagate 5400 bps through it. However, if more than two levels are used for signaling (higher-than-binary encoding), more than one bit may be transmitted at a time, and it is possible to propagate a bit rate that exceeds $2B$. Using multilevel signaling, the Nyquist formulation for channel capacity is

$$f_b = 2B \log_2 M \tag{8}$$

where f_b = channel capacity (bps)
B = minimum Nyquist bandwidth (hertz)
M = number of discrete signal or voltage levels

Equation 8 can be rearranged to solve for the minimum bandwidth necessary to pass M-ary digitally modulated carriers

$$B = \left(\frac{f_b}{\log_2 M}\right) \tag{9}$$

If N is substituted for $\log_2 M$, Equation 9 reduces to

$$B = \frac{f_b}{N} \tag{10}$$

where N is the number of bits encoded into each signaling element.