EEM602 Internet of Things

Lecture # 8
(IOT Course: Digital communications Basics)

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Electronic Signals

Electricity Equations

General Purpose Input / Output (GPIO)

Pulse Width Modulation (PWM)

Analog to Digital Converters (ADC)

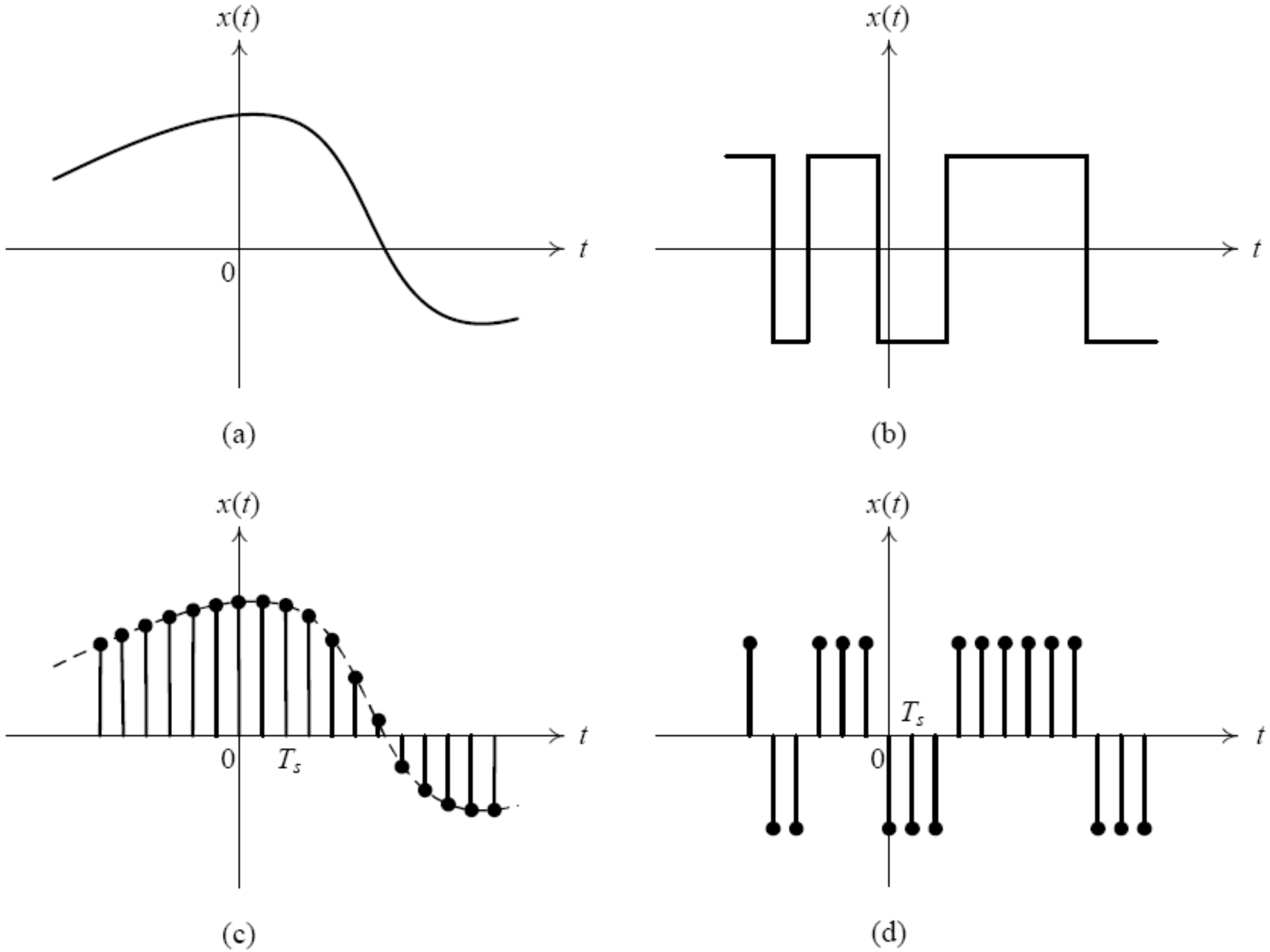
Microcontrollers and Computers

Questions

Analog or Digital

- Common Misunderstanding: Any transmitted signals are ANALOG. NO DIGITAL SIGNAL CAN BE TRANSMITTED
- Analog Message: continuous in amplitude and over time
  - AM, FM for voice sound
  - Traditional TV for analog video
  - First generation cellular phone (analog mode)
  - Record player
- Digital message: 0 or 1, or discrete value
  - VCD, DVD
  - 2G/3G cellular phone
  - Data on your disk
  - Your grade
- Digital age: why digital communication will prevail

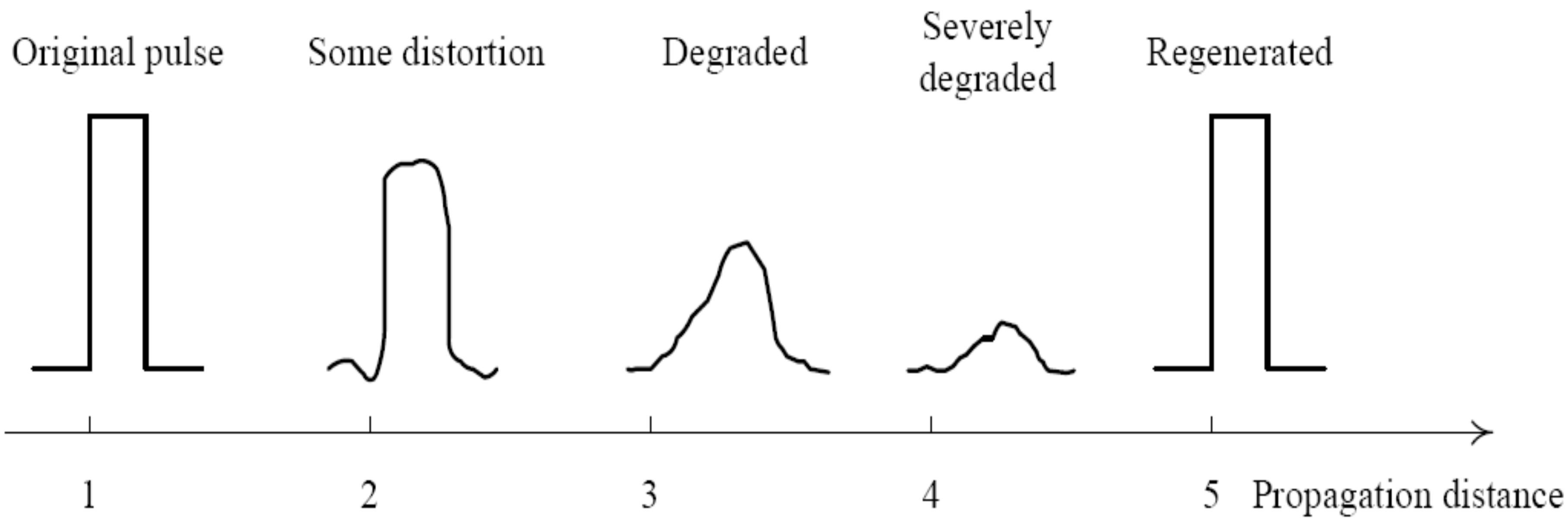
What is Digital Communication?



Digital vs. Analog

- Advantages:
  - Digital signals are much easier to be regenerated.
  - Digital circuits are less subject to distortion and interference.
  - Digital circuits are more reliable and can be produced at a lower cost than analog circuits.
  - It is more flexible to implement digital hardware than analog hardware.
  - Digital signals are beneficial from digital signal processing (DSP) techniques.
- Disadvantages:
  - Heavy signal processing.
  - Synchronization is crucial.
  - Larger transmission bandwidth.
  - Non-graceful degradation.

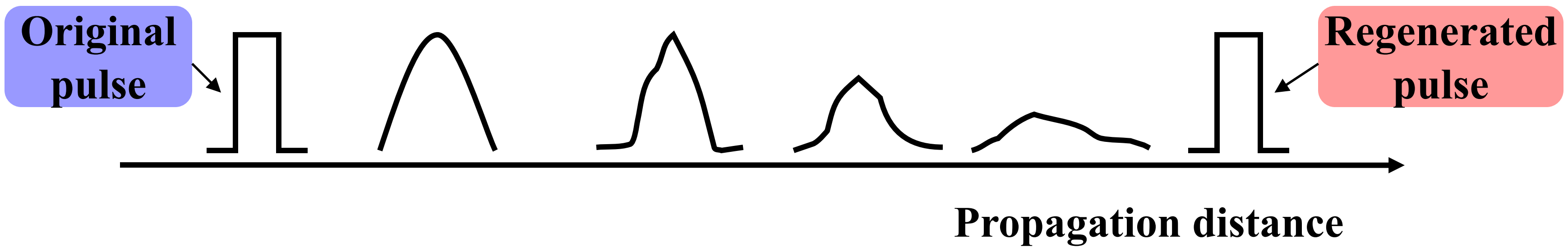
Regenerative Repeater in Digital Communications



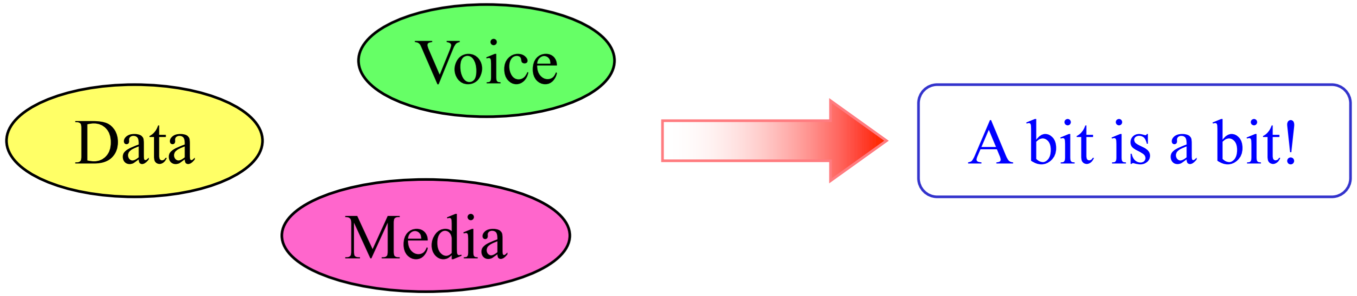
- Digital communications: Transmitted signals belong to a finite set of waveforms → The distorted signal can be recovered to its ideal shape, hence removing all the noise.
- Analog communications: Transmitted signals are analog waveforms, which can take infinite variety of shapes → Once the analog signal is distorted, the distortion cannot be removed.

Digital versus analog

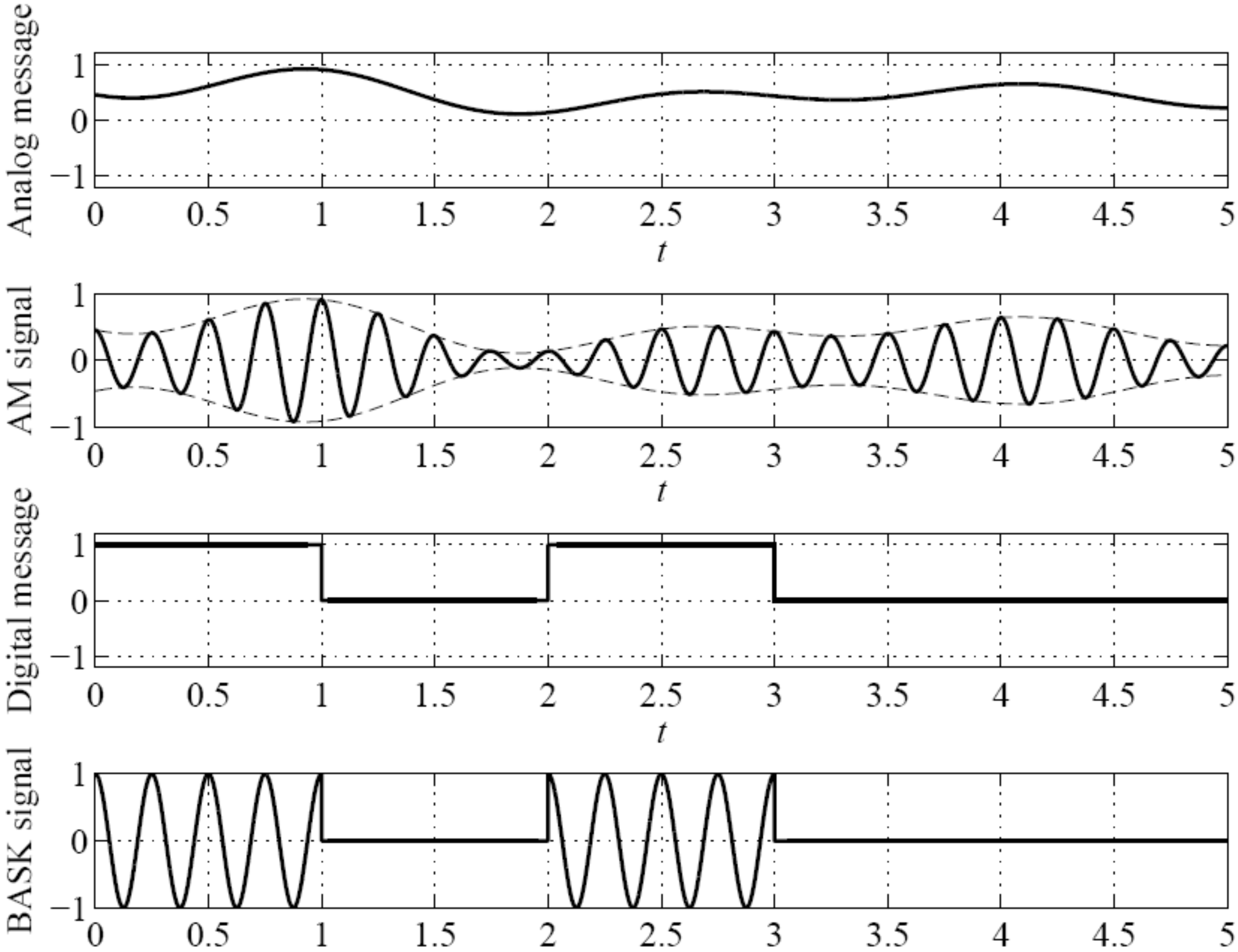
- Advantages of digital communications:
  - Regenerator receiver



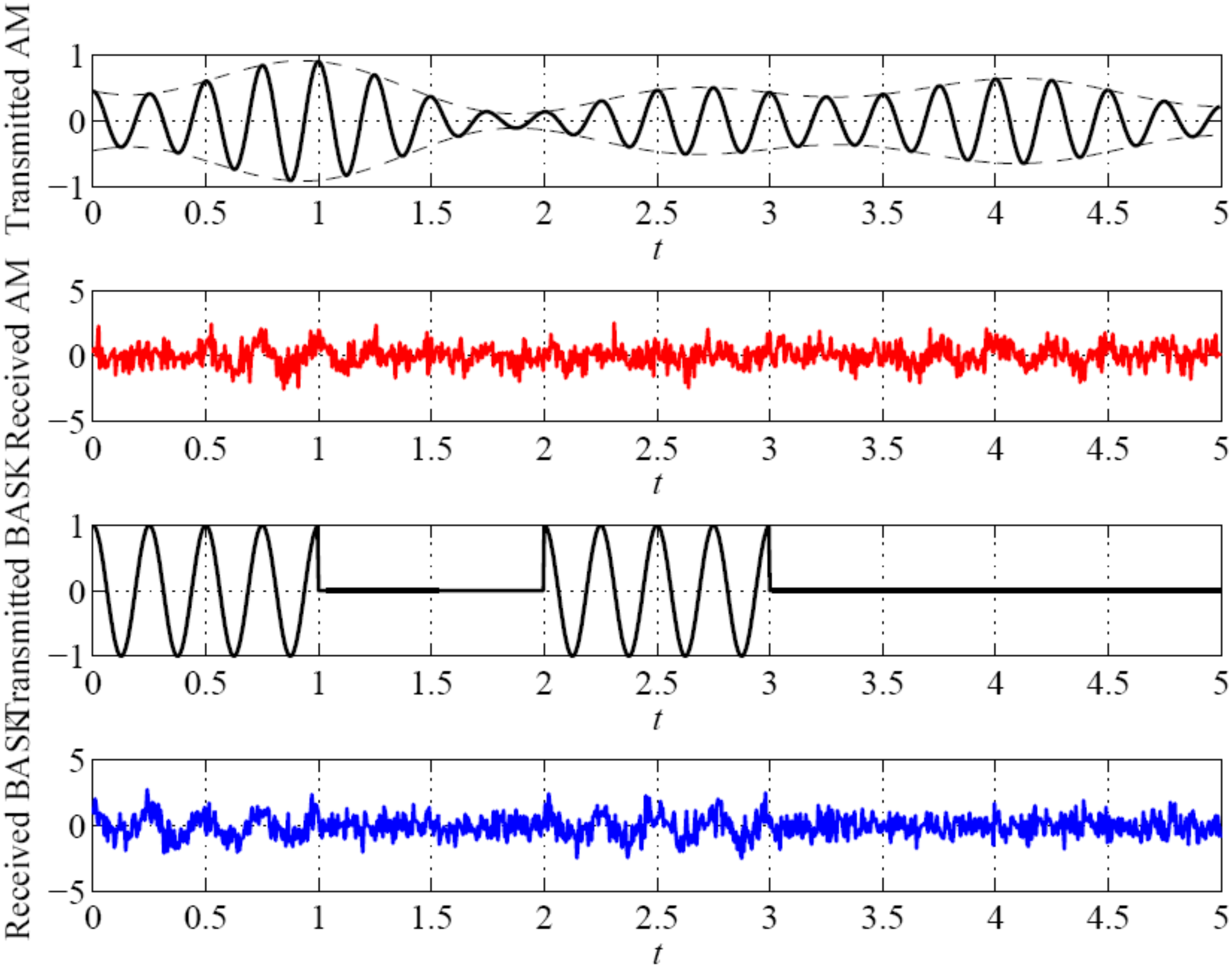
  - Different kinds of digital signal are treated identically.



Analog and Digital Amplitude Modulations



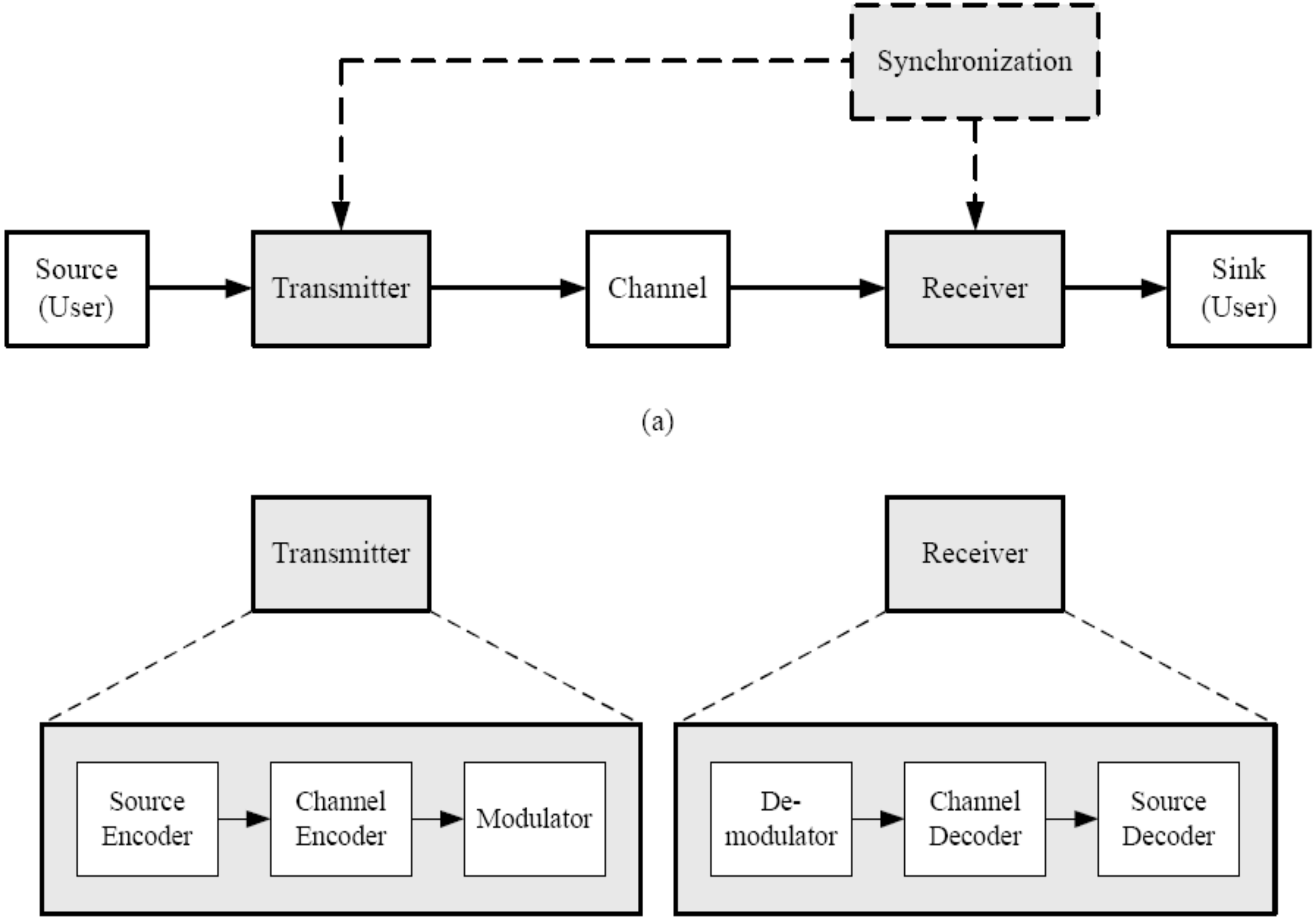
Why Digital Communications?



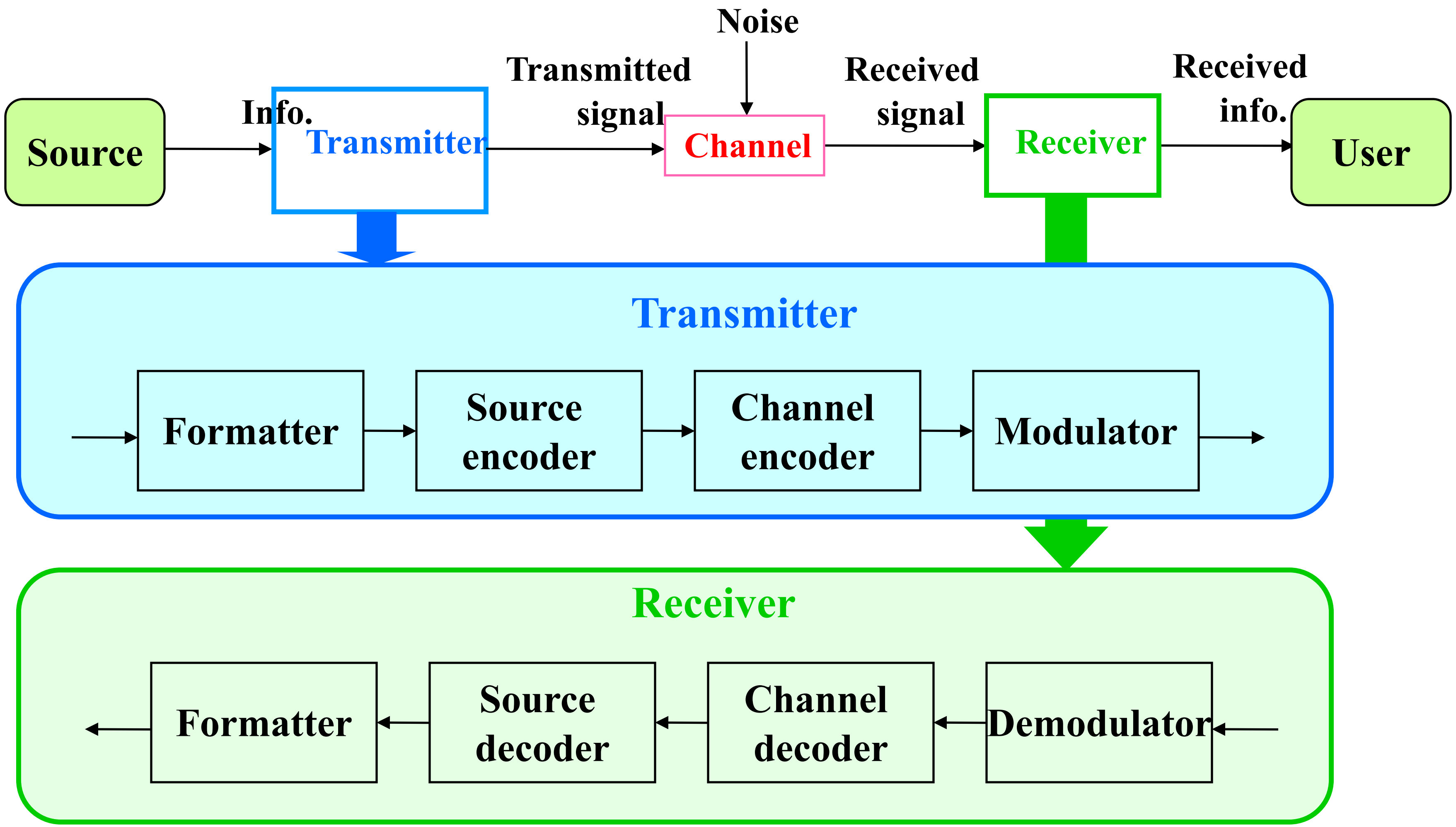
Why digital communications?

- Any noise introduces distortion to an analog signal. Since a digital receiver need only distinguish between two waveforms it is possible to exactly recover digital information.
- Many signal processing techniques are available to improve system performance: source coding, channel (error-correction) coding, equalization, encryption
- Digital ICs are inexpensive to manufacture. A single chip can be mass produced at low cost, no mater how complex
- Digital communications allows integration of voice, video, and data on a single system (ISDN)
- Digital communications systems provide a better tradeoff of bandwidth efficiency and energy efficiency than analog

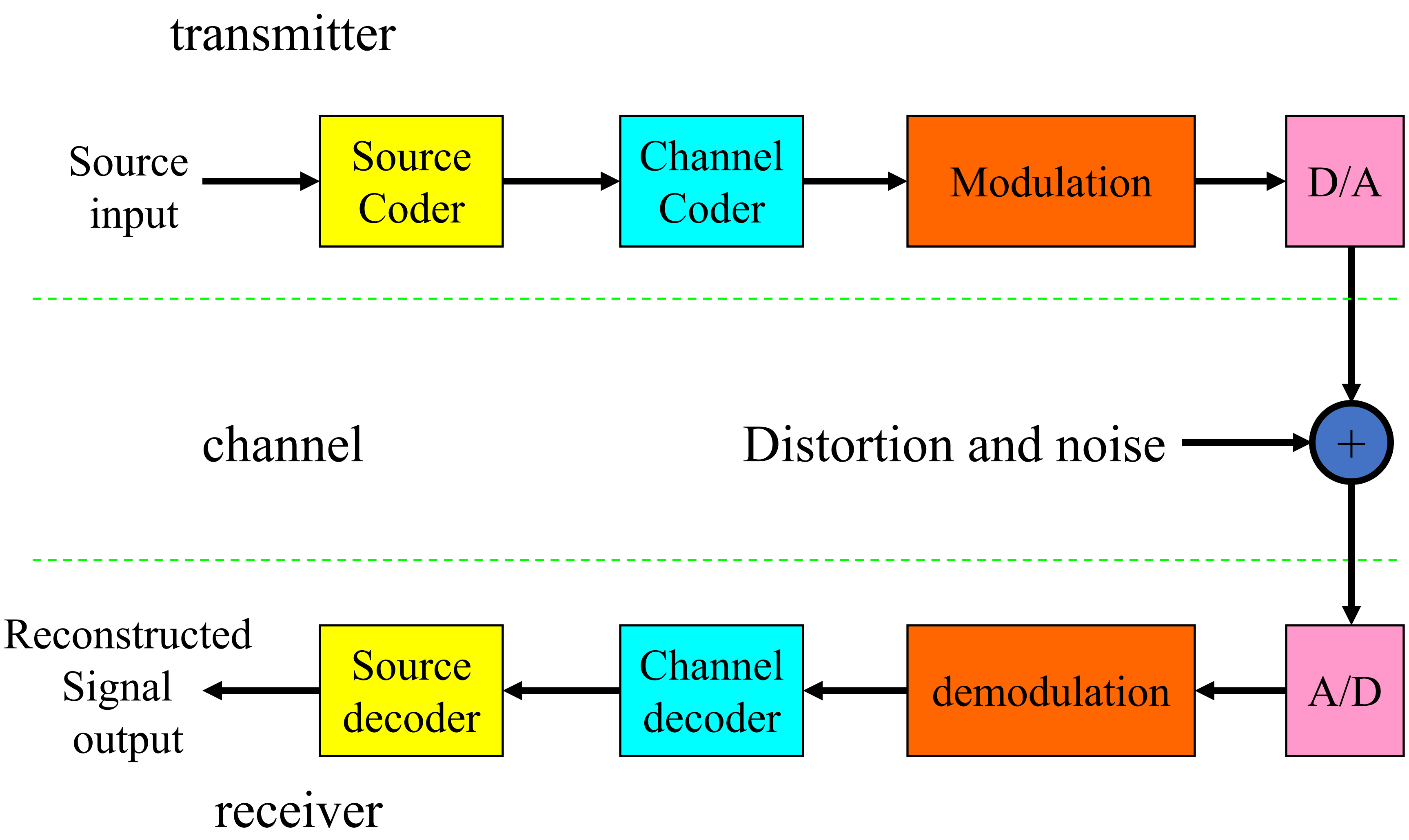
Block Diagram of a Communication System



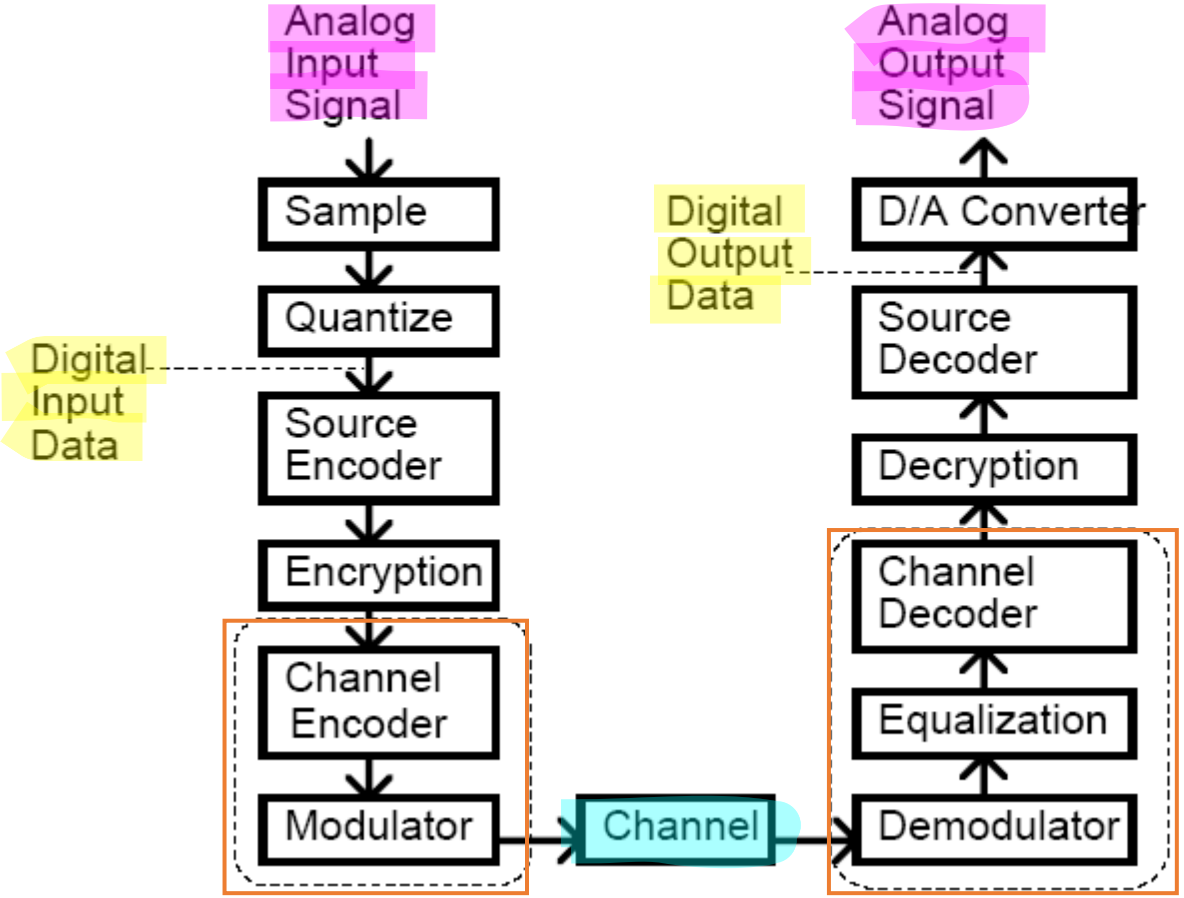
General structure of a communication systems

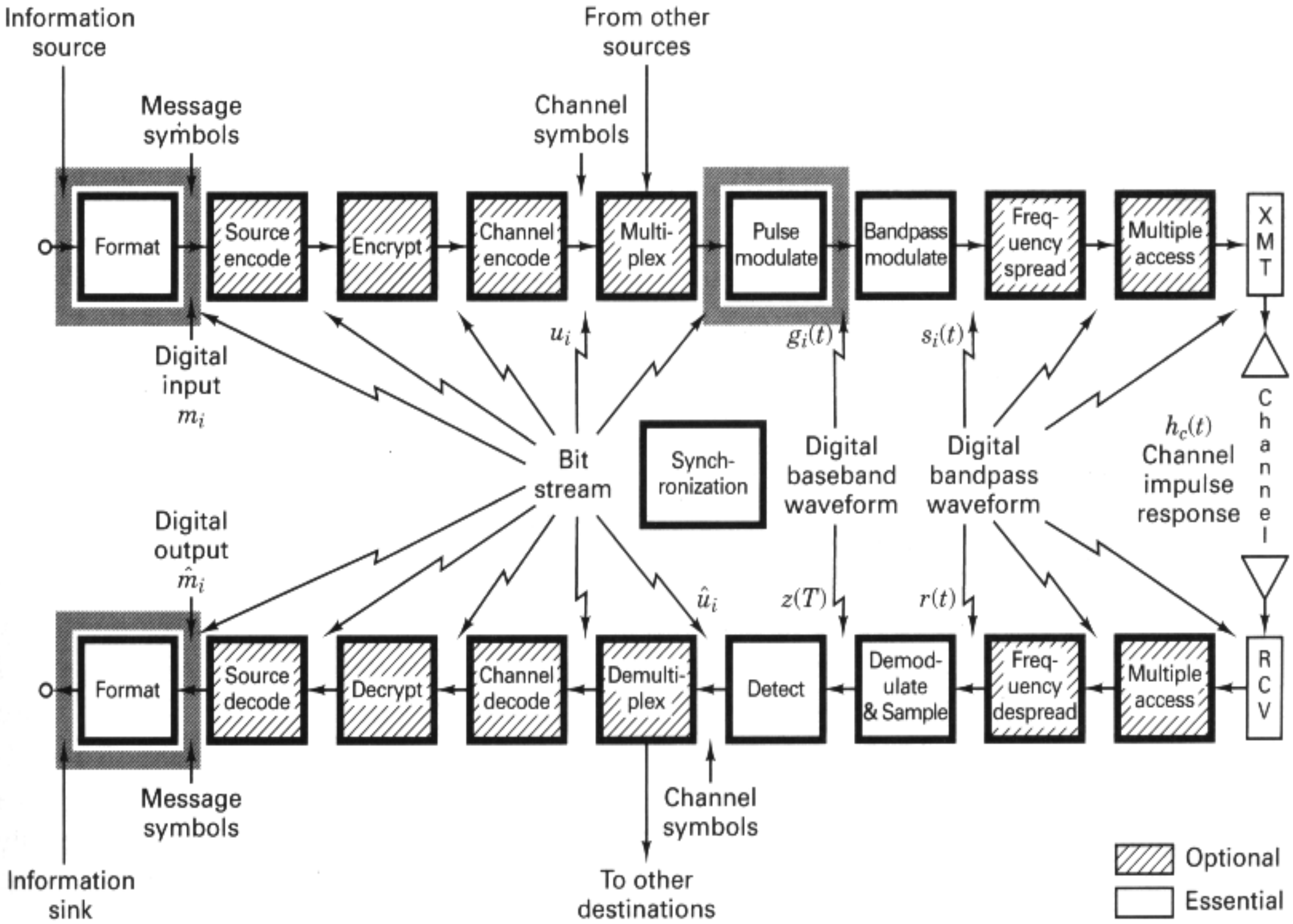


Communication System Components



Block Diagram of Typical Digital Communications System




Information source
Message symbols
Format
Source encode
Encrypt
Channel encode
Channel symbols
From other sources
Multi-plex
Pulse modulate
Bandpass modulate
Freq-uency spread
Multiple access
XMT
Digital input
m_i
u_i
$g_i(t)$
$s_i(t)$
Bit stream
Synch-ronization
Digital baseband waveform
Digital bandpass waveform
$h_c(t)$
Channel impulse response
Channel
Digital output
$\hat{m}_i$
$\hat{u}_i$
$z(T)$
$r(t)$
Format
Source decode
Decrypt
Channel decode
Demulti-plex
Detect
Demod-ulate & Sample
Freq-uency despread
Multiple access
RCV
Message symbols
Information sink
Channel symbols
To other destinations
Optional
Essential

Digital Communication

Q: What is Digital Communication ?

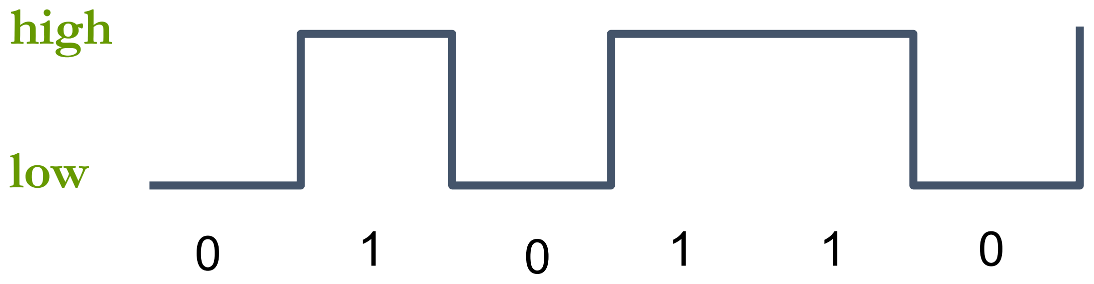
The transfer of information (analog or digital) using digital signals and techniques.

Q: What are digital signals?

Digital signals are binary pulse that have two distinct states, each represented by a voltage level.

The two levels are:

1. Binary 0 or low.
2. Binary 1 or high.



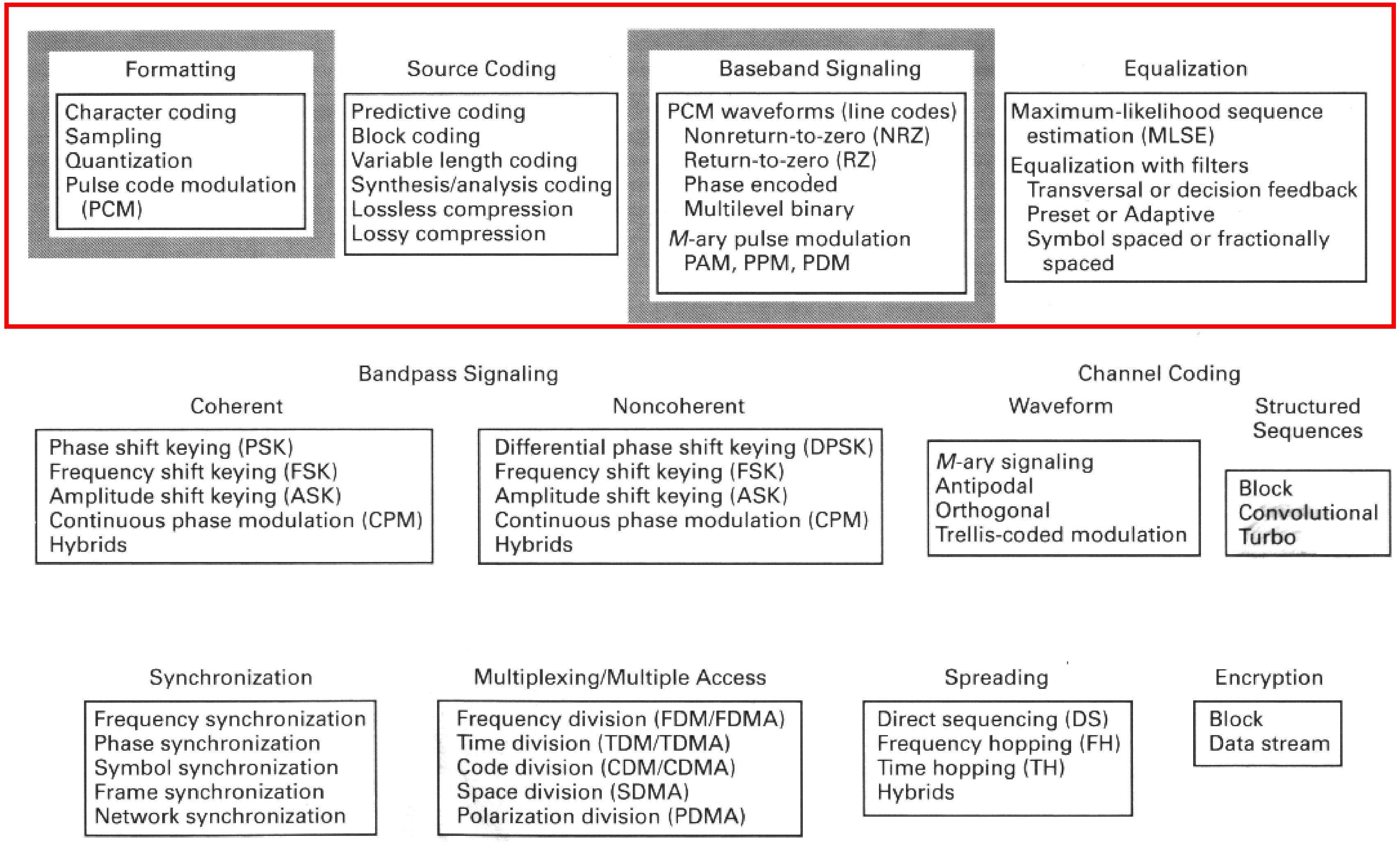
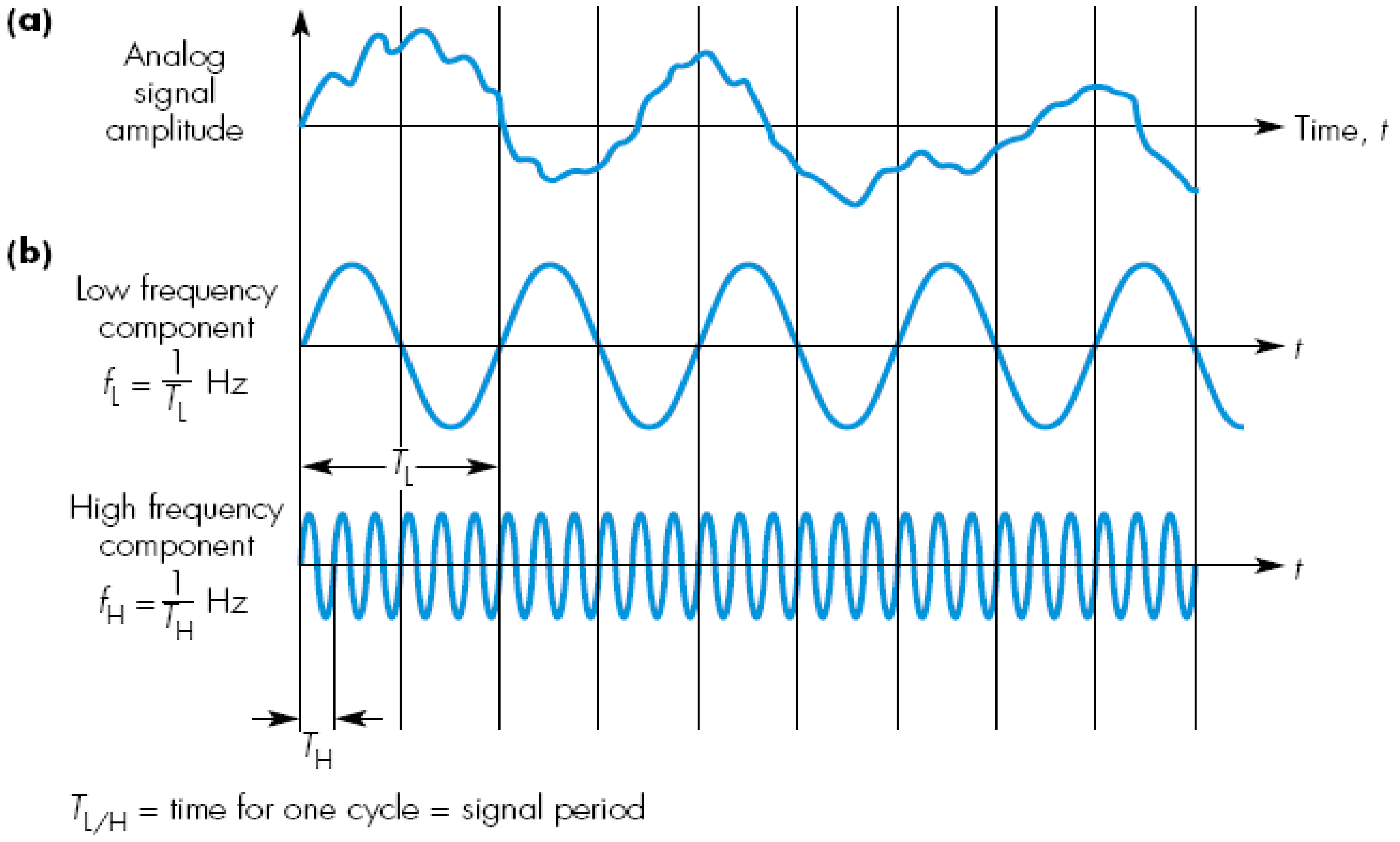


Figure 2.1 Basic digital communication transformations

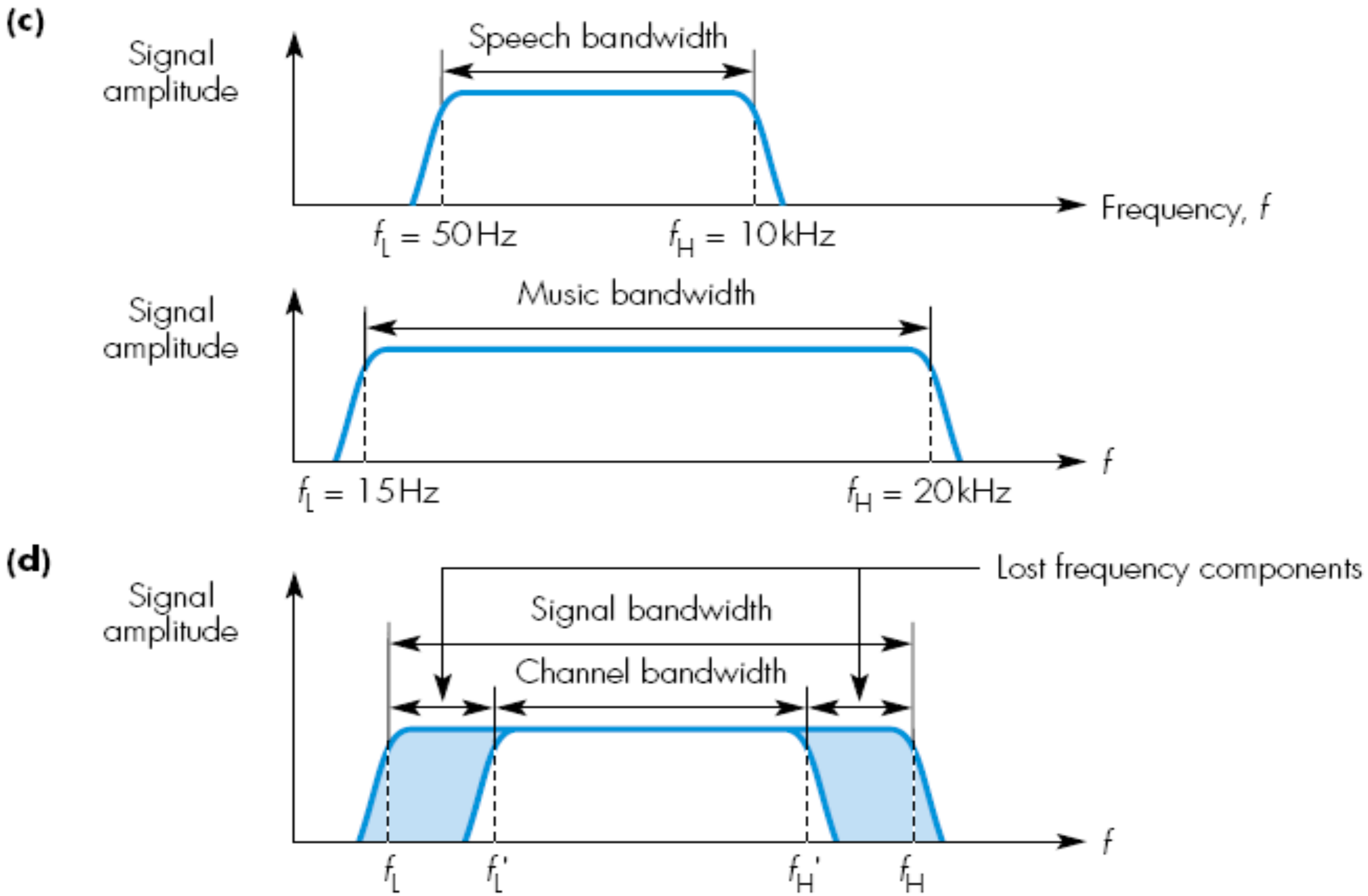
Digitization principles

- The conversion of an analog signal into a digital form
- signal encoder.
- signal decoder.

Signal properties: (a) time-varying analog signal; (b) sinusoidal frequency components;

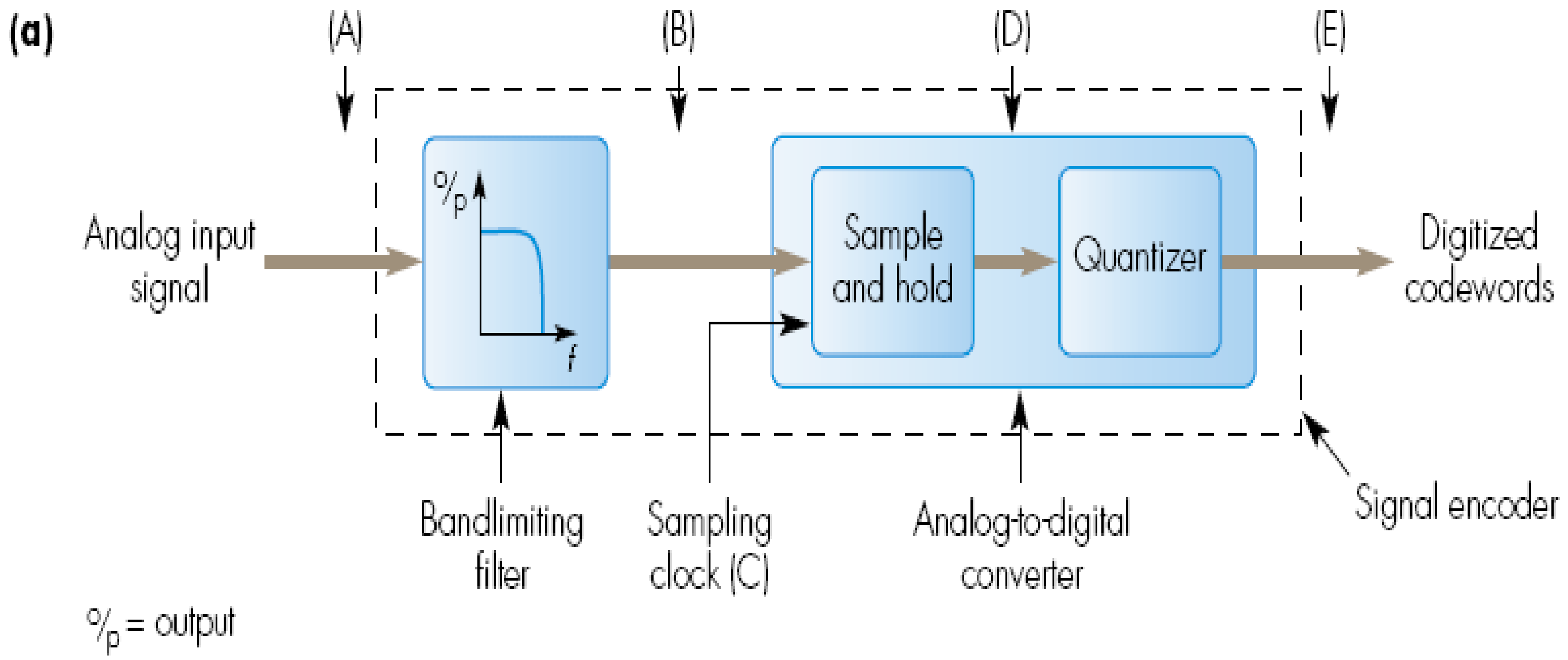


Signal properties: (c) signal bandwidth examples;
(d) effect of a limited bandwidth transmission channel.

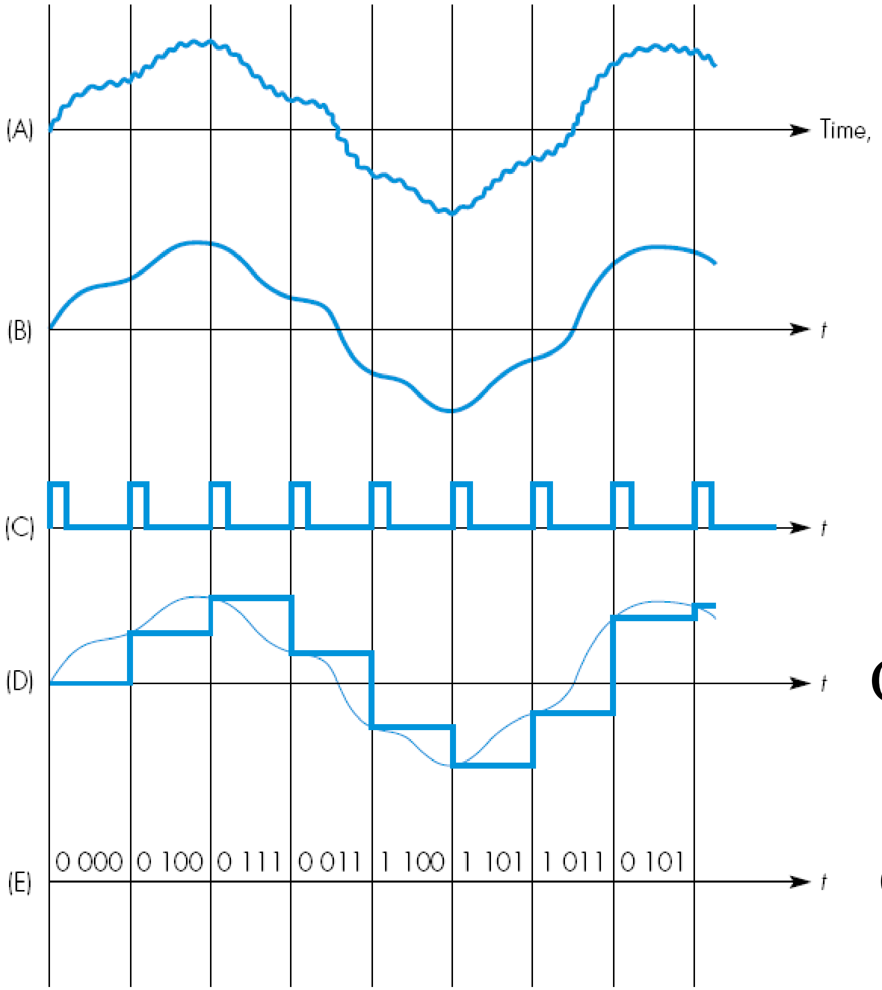


F_L' and F_H' are known as the cut off frequencies of the channel

Signal Encoder Design: Circuit Components



Signal Encoder Design: Associated Waveform Set



Analog Signal

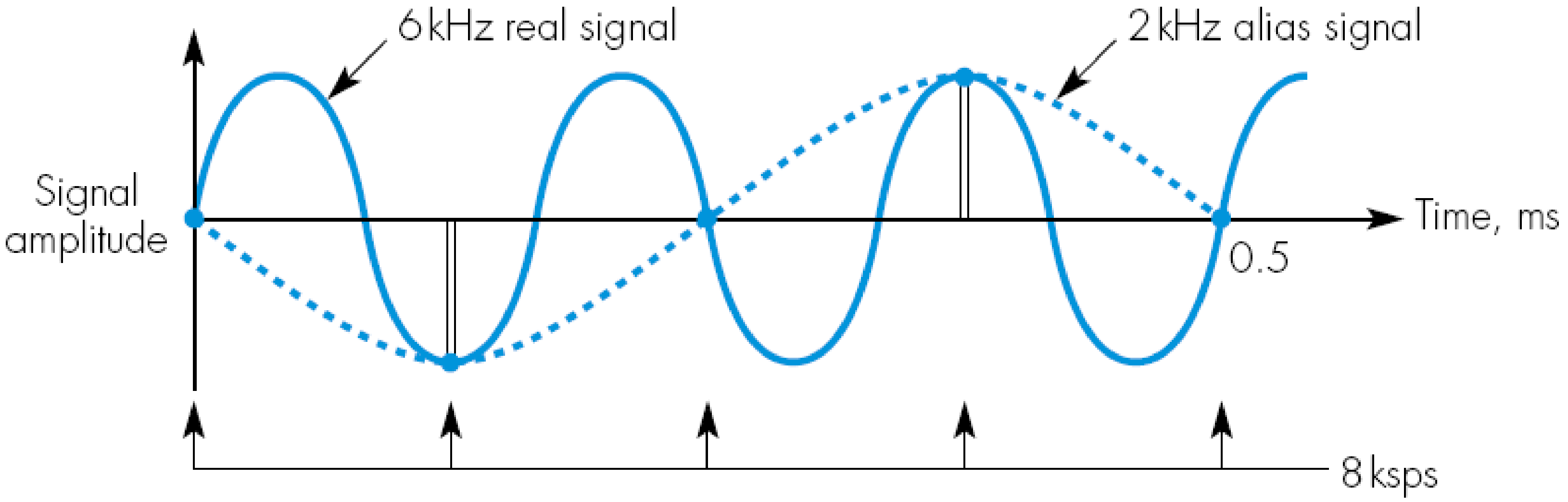
Low Pass Filtered Version

Clock

Output from Sample & Hold

Quantizer Output

Alias Signal Generation Due to Under Sampling



Exa

Determine the rate of the sampler and the bandwidth of the bandlimiting filter in an encoder which is to be used for the digitization of an analog signal which has a bandwidth from 15 Hz through to 10 kHz assuming the digitized signal:

(i) is to be stored within the memory of a computer,

(ii) is to be transmitted over a channel which has a bandwidth from 200 Hz through to 3.4 kHz.

Answer:

The Nyquist sampling rate must be at least twice the highest frequency component of the signal or transmission channel. Hence:

(i) The sampling rate must be at least $2 \times 10\,\text{kHz} = 20\,\text{kHz}$ or 20 ksps and the bandwidth of the bandlimiting filter is from 0 Hz through to 10 kHz.

(ii) The sampling rate must be at least $2 \times 3.4\,\text{kHz} = 6.8\,\text{kHz}$ or 6.8 ksps and the bandwidth of the bandlimiting filter is from 0 Hz through to 3.4 kHz.

In practice, it should be noted that, because of imperfections in filters, some higher frequency components above the filter cut-off frequency may be passed and hence the sampling rate is normally higher than the two derived values. In the case of (ii), for example, it is common to assume that frequency components of up to 4 kHz may be passed by the bandlimiting filter and hence a sampling rate of 8 ksps is normally used.

2.4 FORMATTING ANALOG INFORMATION

If the information is analog, it cannot be character encoded as in the case of textual data; the information must first be transformed into a digital format. The process of transforming an analog waveform into a form that is compatible with a digital communication system starts with sampling the waveform to produce a discrete pulse-amplitude-modulated waveform, as described below.

2.4.1 The Sampling Theorem

The link between an analog waveform and its sampled version is provided by what is known as the *sampling process*. This process can be implemented in several ways, the most popular being the *sample-and-hold* operation. In this operation, a switch and storage mechanism (such as a transistor and a capacitor, or a shutter and a filmstrip) form a sequence of samples of the continuous input waveform. The output of the sampling process is called *pulse amplitude modulation* (PAM) because the successive output intervals can be described as a sequence of pulses with amplitudes derived from the input waveform samples. The analog waveform can be approximately retrieved from a PAM waveform by simple low-pass filtering. An important question: how closely can a filtered PAM waveform approximate the original input waveform? This question can be answered by reviewing the *sampling theorem*, which states the following [1]: A bandlimited signal having no spectral components above f_m hertz can be determined uniquely by values sampled at uniform intervals of

$$T_s \leq \frac{1}{2f_m} \text{ sec} \tag{2.1}$$

This particular statement is also known as the *uniform sampling theorem.* Stated another way, the upper limit on T_s can be expressed in terms of the sampling rate, denoted $f_s = 1/T_s$. The restriction, stated in terms of the sampling rate, is known as the *Nyquist criterion*. The statement is

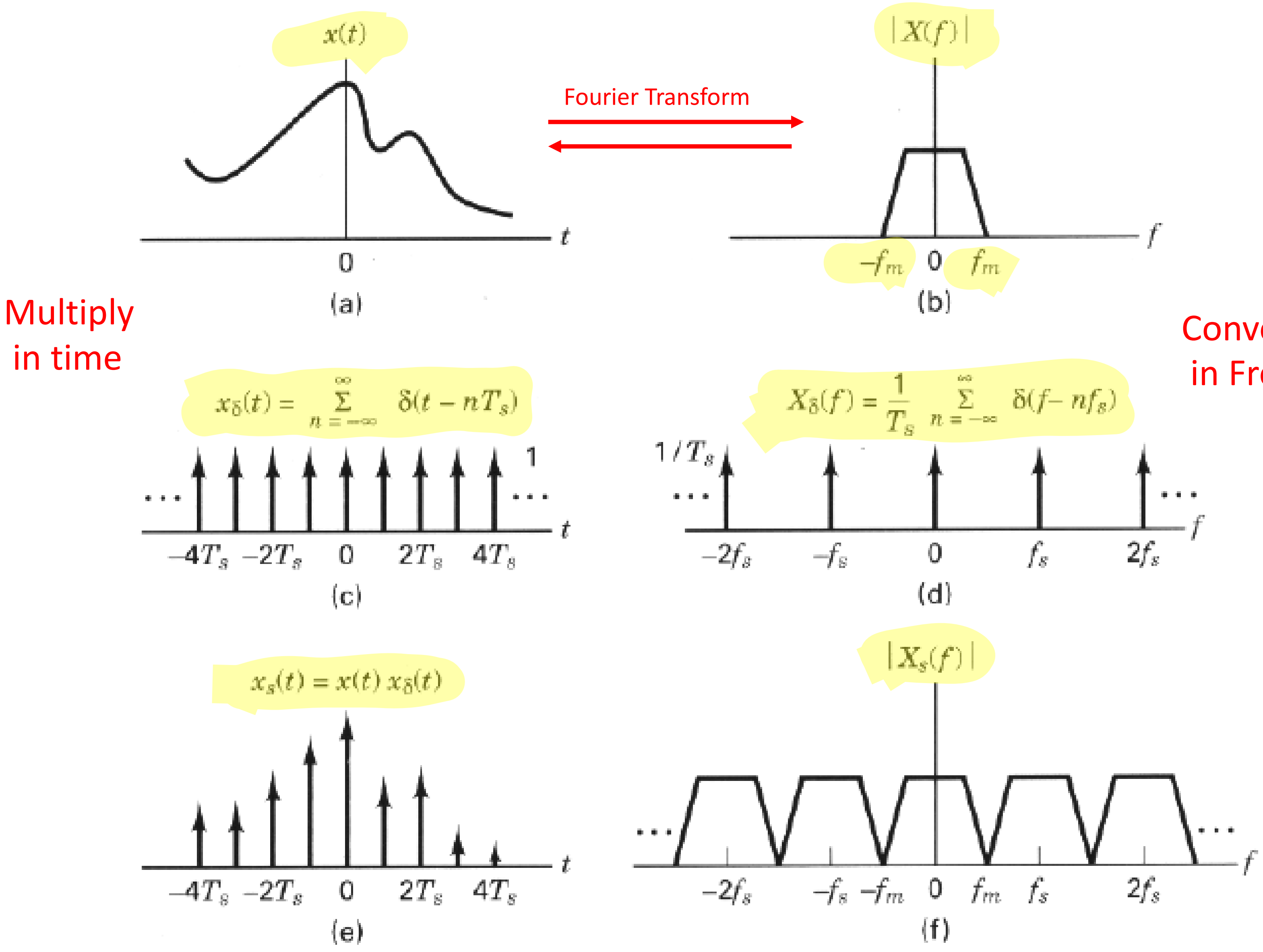
$$f_s \geq 2f_m \tag{2.2}$$

The sampling rate $f_s = 2f_m$ is also called the *Nyquist rate.* The Nyquist criterion is a theoretically sufficient condition to allow an analog signal to be *reconstructed completely* from a set of uniformly spaced discrete-time samples. In the sections that follow, the validity of the sampling theorem is demonstrated using different sampling approaches.

2.4.1.1 Impulse Sampling

Here we demonstrate the validity of the sampling theorem using the frequency convolution property of the Fourier transform. Let us first examine the case of *ideal sampling* with a sequence of unit impulse functions. Assume an analog waveform, $x(t)$, as shown in Figure 2.6a, with a Fourier transform, $X(f)$, which is zero outside the interval $(-f_m < f < f_m)$, as shown in Figure 2.6b. The sampling of $x(t)$ can be viewed as the product of $x(t)$ with a periodic train of unit impulse functions $x_\delta(t)$, shown in Figure 2.6c and defined as

$$x_\delta(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s) \tag{2.3}$$


x(t)
0
t
(a)
Fourier Transform
|X(f)|
−fm 0 fm
f
(b)
Multiply
in time
Convolution
in Frequency
xδ(t) = Σ δ(t − nTs)
1
−4Ts −2Ts 0 2Ts 4Ts
(c)
Xδ(f) = 1/Ts Σ δ(f − nfs)
1/Ts
−2fs −fs 0 fs 2fs
(d)
xs(t) = x(t) xδ(t)
−4Ts −2Ts 0 2Ts 4Ts
(e)
|Xs(f)|
−2fs −fs −fm 0 fm fs 2fs
(f)

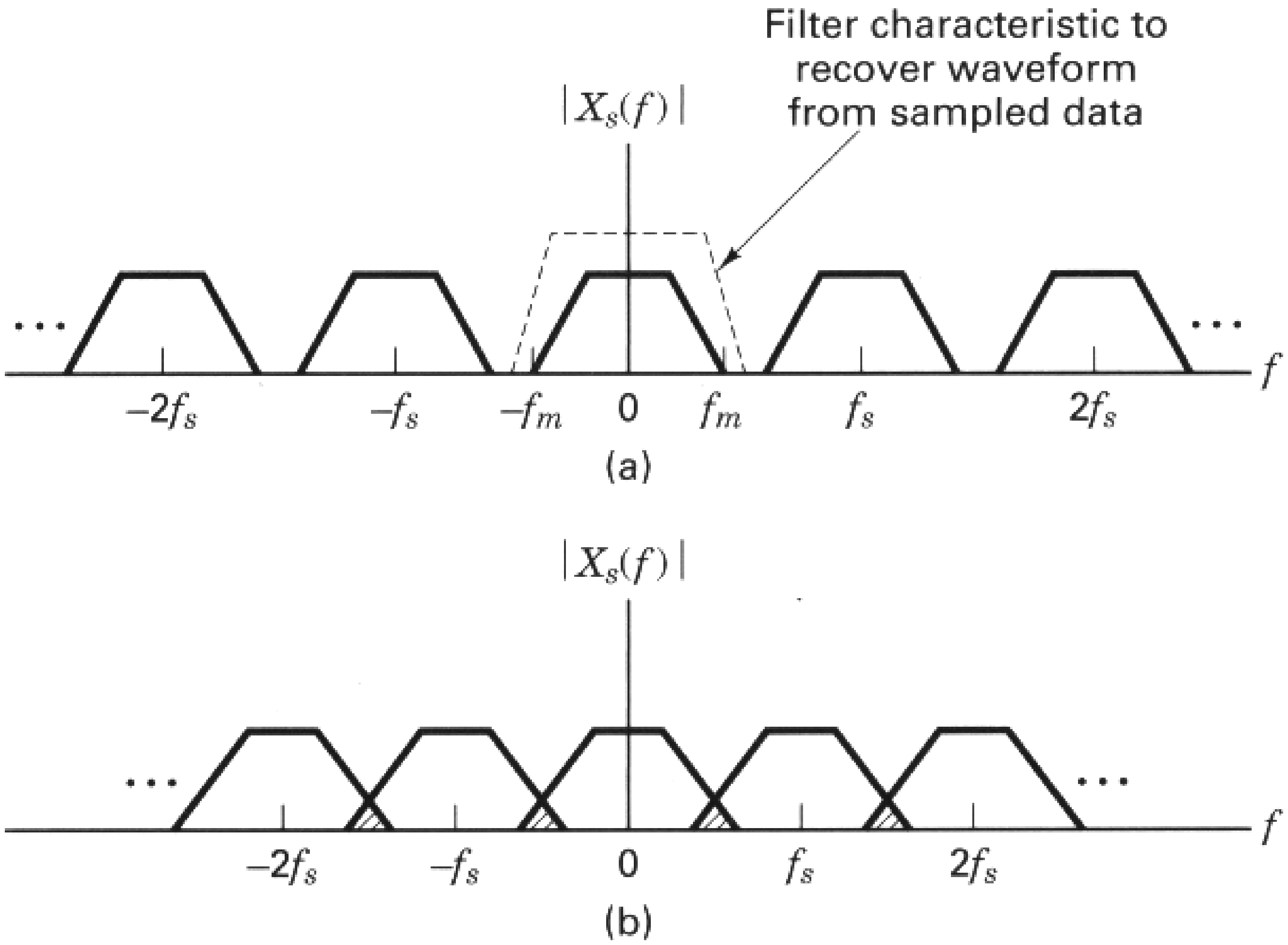


Figure 2.7 Spectra for various sampling rates. (a) Sampled spectrum ($f_s > 2f_m$). (b) Sampled spectrum ($f_s < 2f_m$).

this we mean that a bandwidth can be determined beyond which the spectral components are attenuated to a level that is considered negligible.

2.4.1.2 Natural Sampling

Here we demonstrate the validity of the sampling theorem using the frequency shifting property of the Fourier transform. Although instantaneous sampling is a convenient model, a more practical way of accomplishing the sampling of a bandlimited analog signal $x(t)$ is to multiply $x(t)$, shown in Figure 2.8a, by the pulse train or switching waveform $x_p(t)$, shown in Figure 2.8c. Each pulse in $x_p(t)$ has width T and amplitude $1/T$. Multiplication by $x_p(t)$ can be viewed as the opening and closing of a switch. As before, the sampling frequency is designated f_s, and its reciprocal, the time period between samples, is designated T_s. The resulting sampled-data sequence, $x_s(t)$, is illustrated in Figure 2.8e and is expressed as

$$x_s(t) = x(t)x_p(t) \tag{2.9}$$

The sampling here is termed *natural sampling,* since the top of each pulse in the $x_s(t)$ sequence retains the shape of its corresponding analog segment during the pulse interval. Using Equation (A.13), we can express the periodic pulse train as a Fourier series in the form

$$x_p(t) = \sum_{n=-\infty}^{\infty} c_n e^{j2\pi n f_s t} \tag{2.10}$$

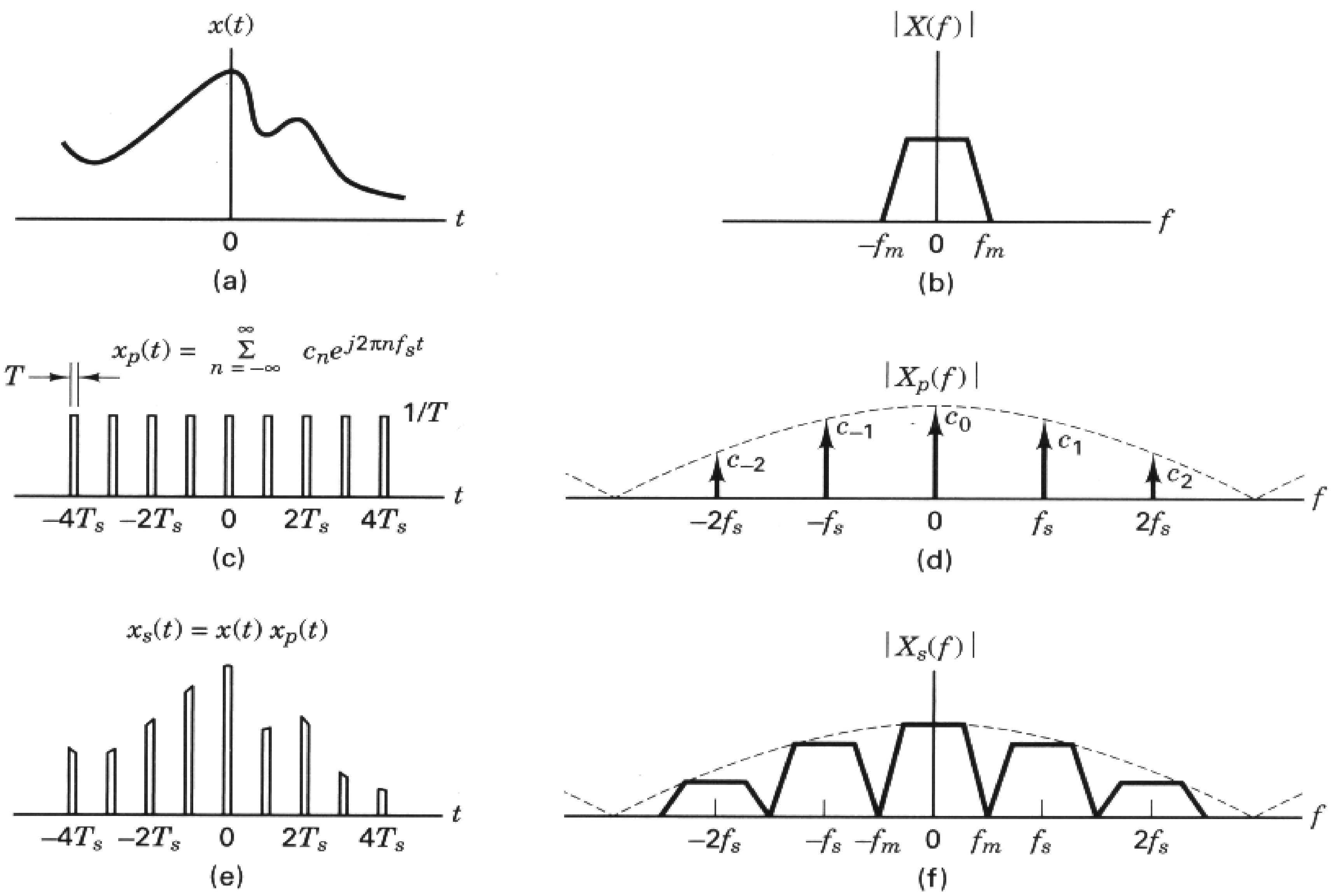


Figure 2.8 Sampling theorem using the frequency shifting property of the Fourier transform.

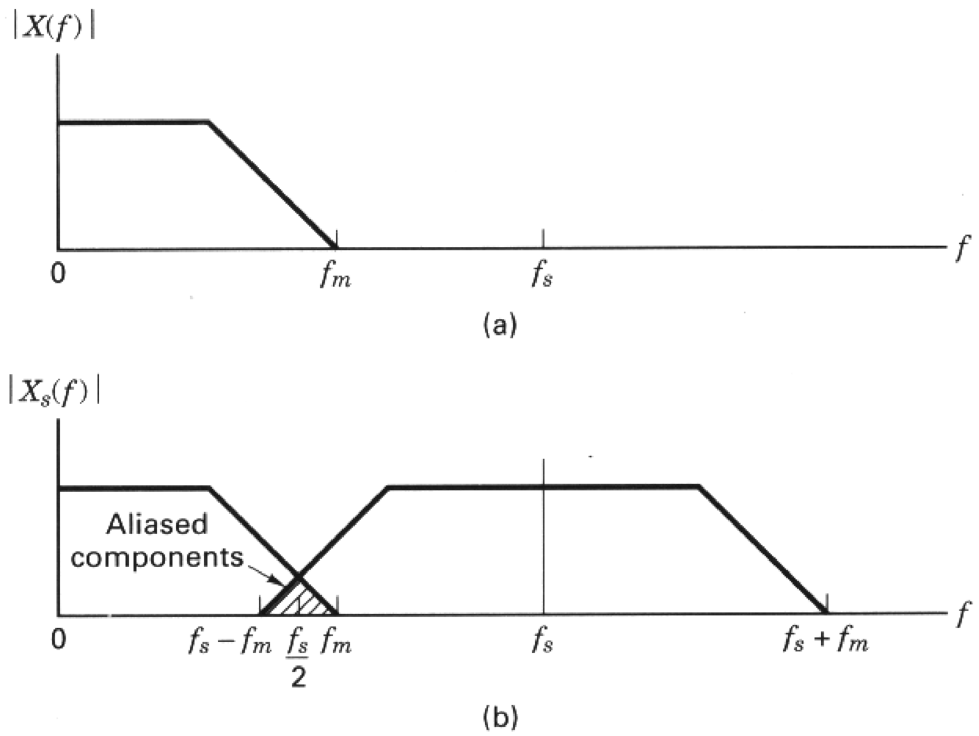


Figure 2.9 Aliasing in the frequency domain. (a) Continuous signal spectrum. (b) Sampled signal spectrum.

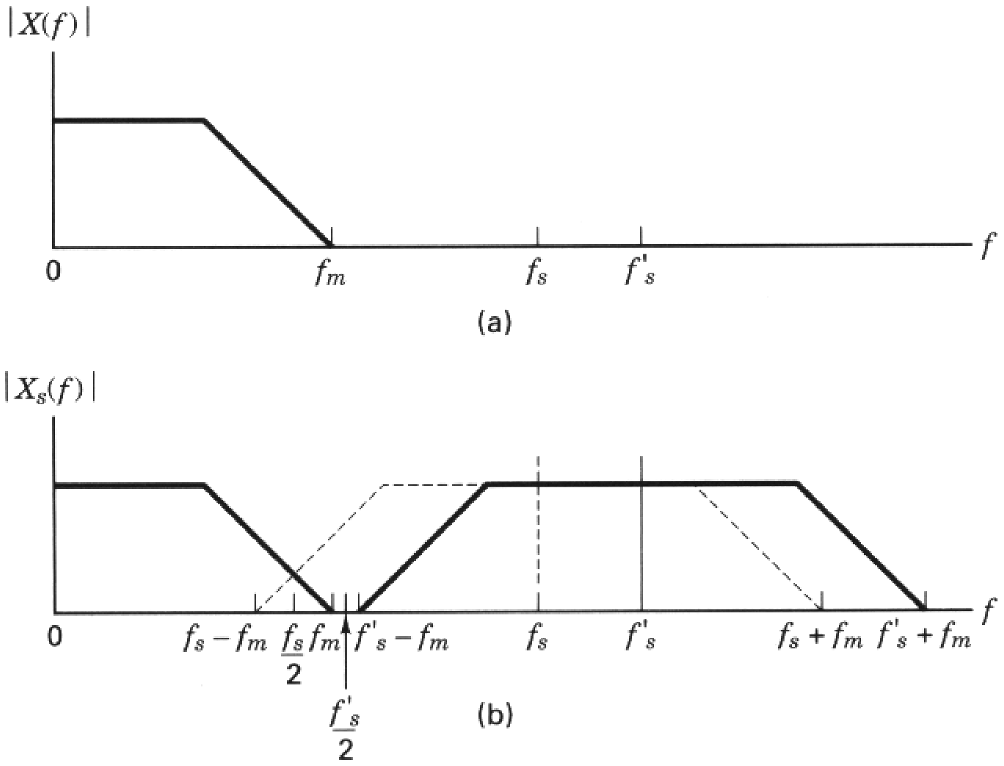


Figure 2.10 Higher sampling rate eliminates aliasing. (a) Continuous signal spectrum. (b) Sampled signal spectrum.

2.4.4 Signal Interface for a Digital System

Let us examine four ways in which analog source information can be described. Figure 2.14 illustrates the choices. Let us refer to the waveform in Figure 2.14a as the *original analog waveform.* Figure 2.14b represents a sampled version of the original waveform, typically referred to as *natural-sampled data* or PAM (pulse amplitude modulation). Do you suppose that the sampled data in Figure 2.14b are compatible with a digital system? No, they are not, because the amplitude of each natural sample still has an infinite number of possible values; a digital system deals with a finite number of values. Even if the sampling is flat-top sampling, the possible pulse values form an infinite set, since they reflect all the possible values of the continuous analog waveform. Figure 2.14c illustrates the original waveform represented by discrete pulses. Here the pulses have flat tops *and* the pulse amplitude values are limited to a finite set. Each pulse is expressed as a level from a finite number of predetermined levels; each such level can be represented by a symbol from a finite alphabet. The pulses in Figure 2.14c are referred to as *quantized samples;* such a format is the obvious choice for interfacing with a digital system. The format in Figure 2.14d may be construed as the output of a sample-and-hold circuit. When the sample values are quantized to a finite set, this format can also interface with a digital system. After quantization, the analog waveform can still be recovered, but not precisely; improved reconstruction fidelity of the analog waveform can be achieved by increasing the number of quantization levels (requiring increased system bandwidth). Signal distortion due to quantization is treated in the following sections (and later in

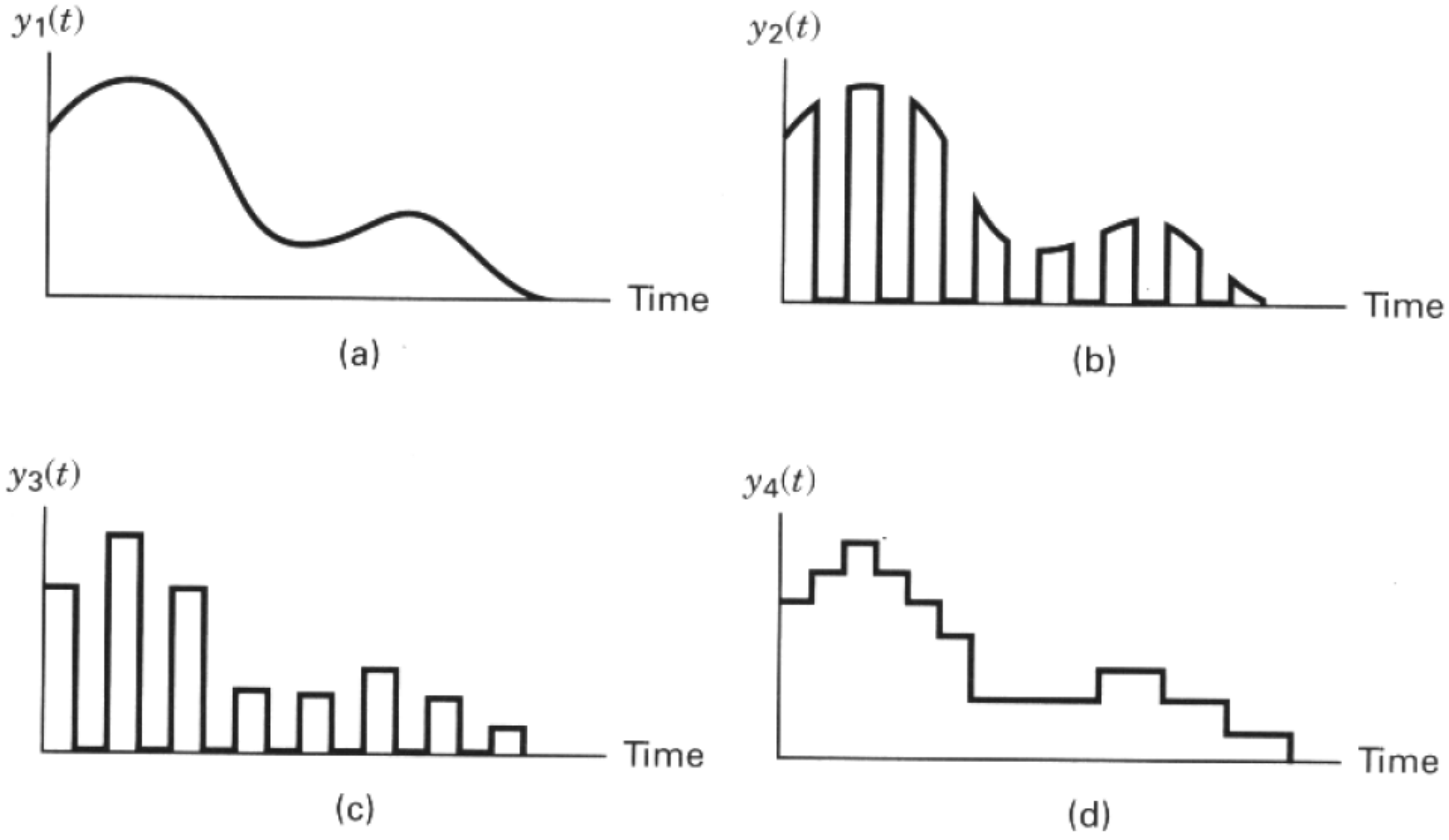
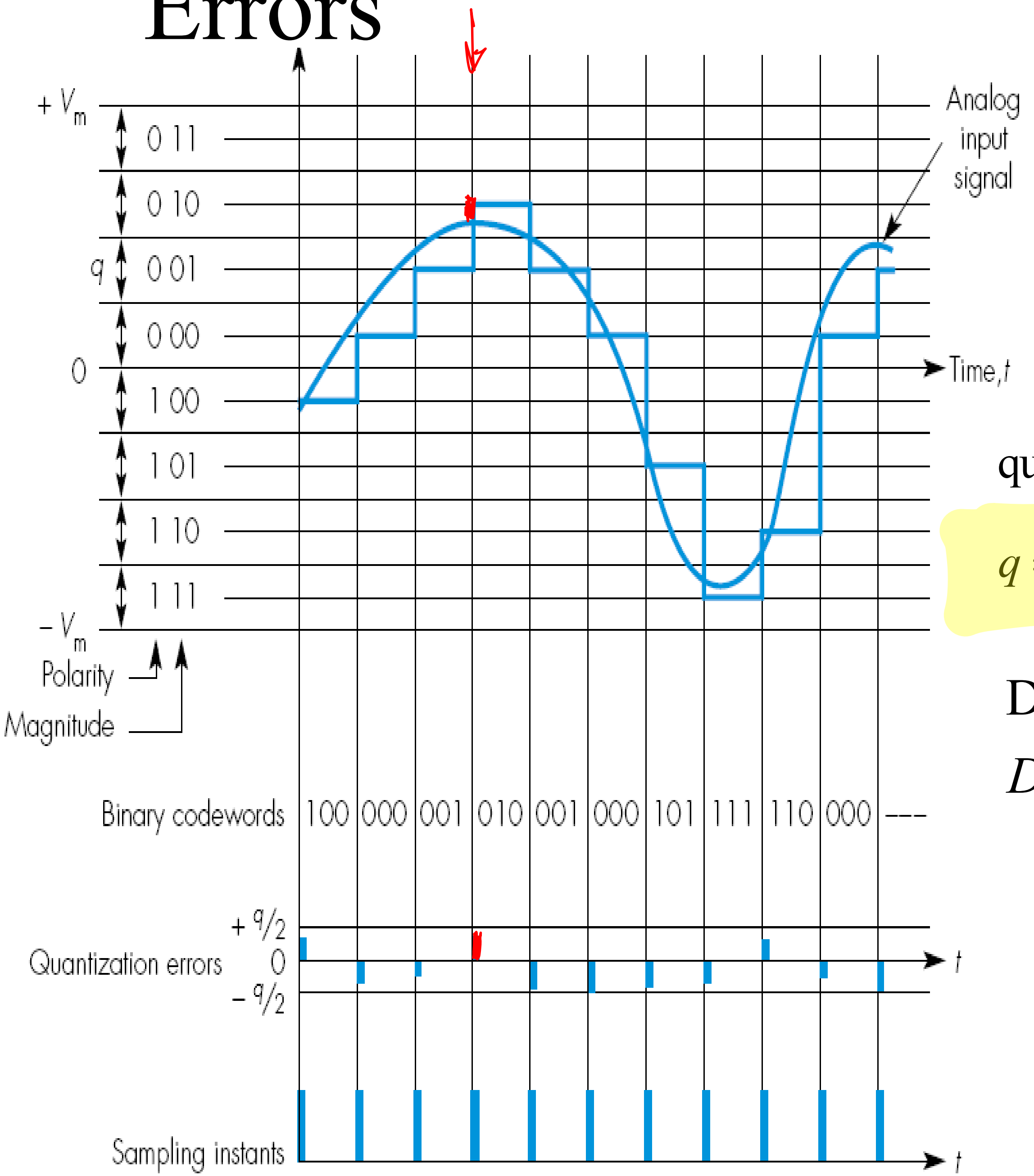


Figure 2.14 Amplitude and time coordinates of source data. (a) Original analog waveform. (b) Natural-sampled data. (c) Quantized samples. (d) Sample and hold.

Quantization Procedure: Source of Errors



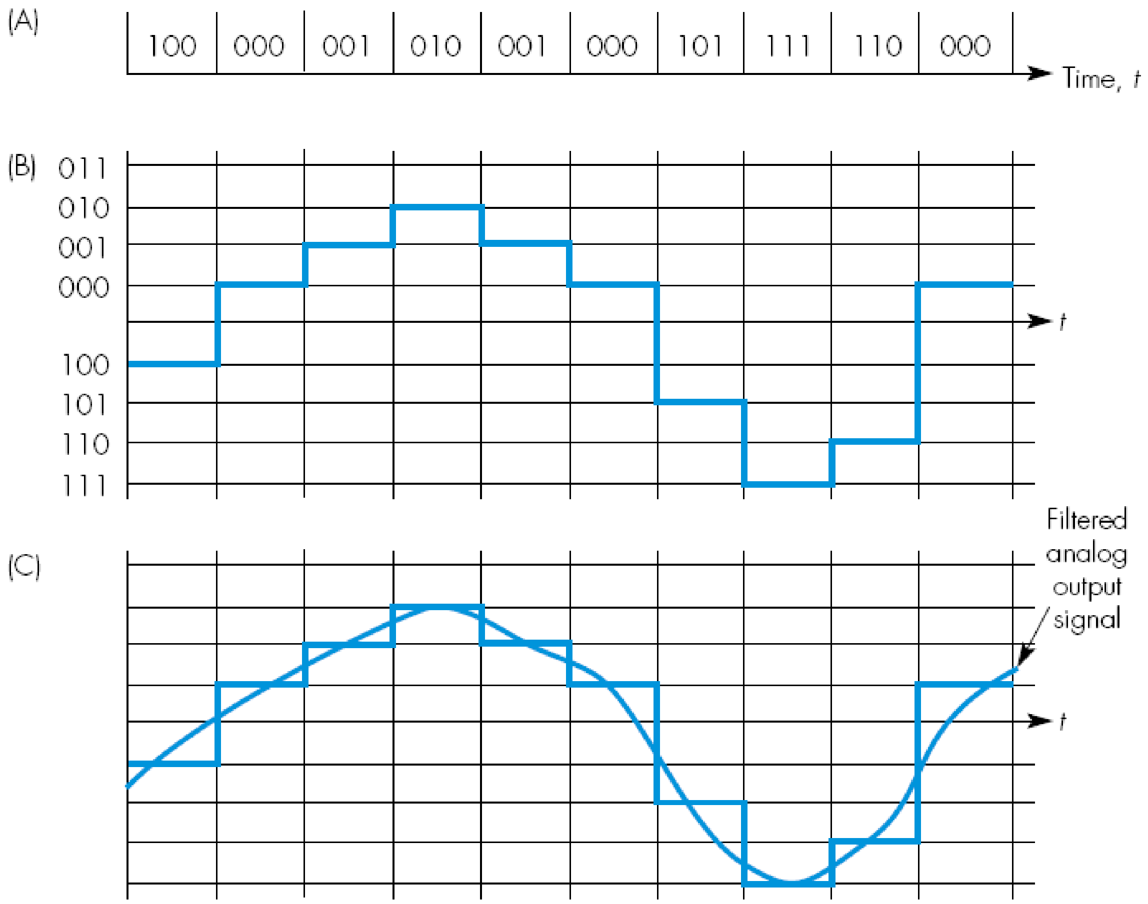
quantization interval q

$$q = \frac{2V_{max}}{2^n} \text{ where } n = \text{number of bits}$$

Dynamic Range D in Decibels

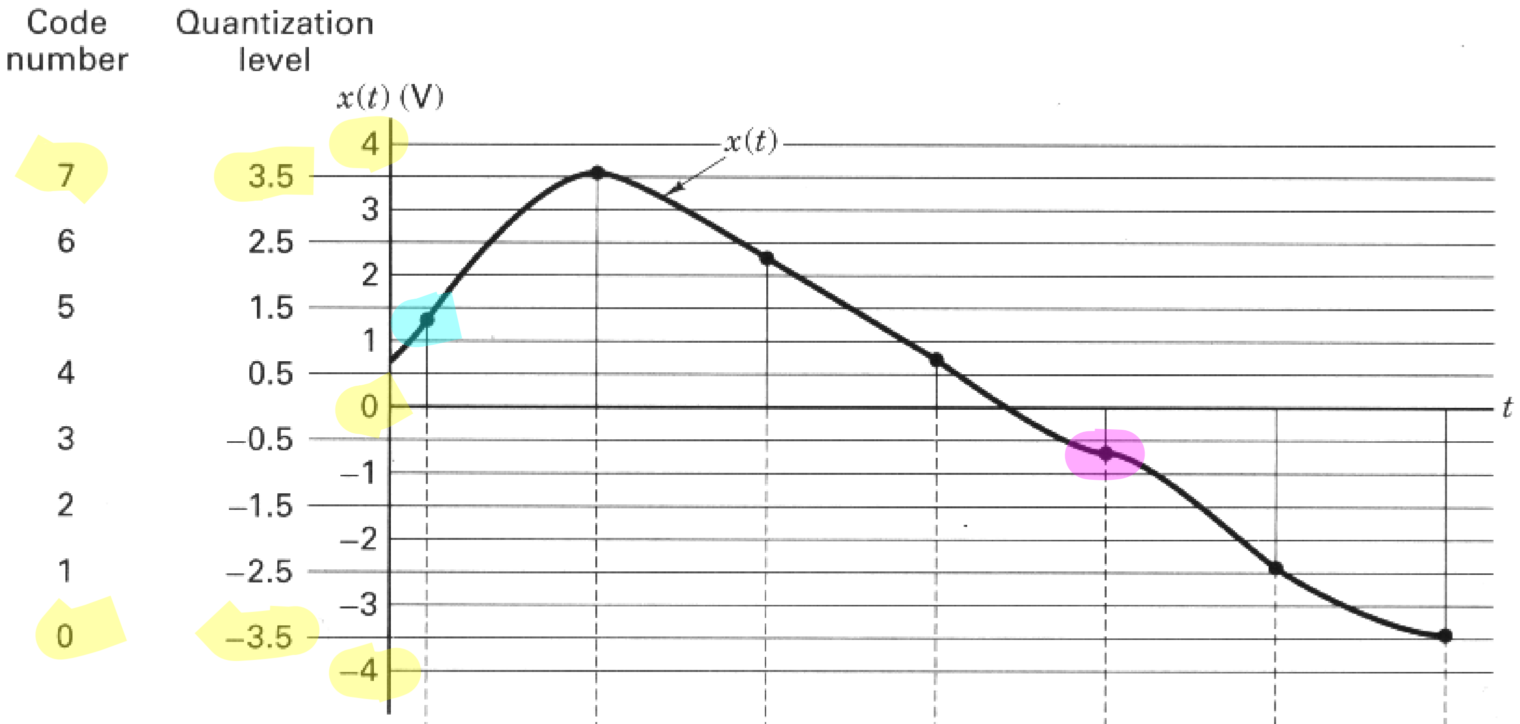
$$D = 20\log_{10}\left(V_{max}/V_{min}\right)\text{dB}$$

Signal Decoder Design: Associated Waveform Set



2.6 PULSE CODE MODULATION

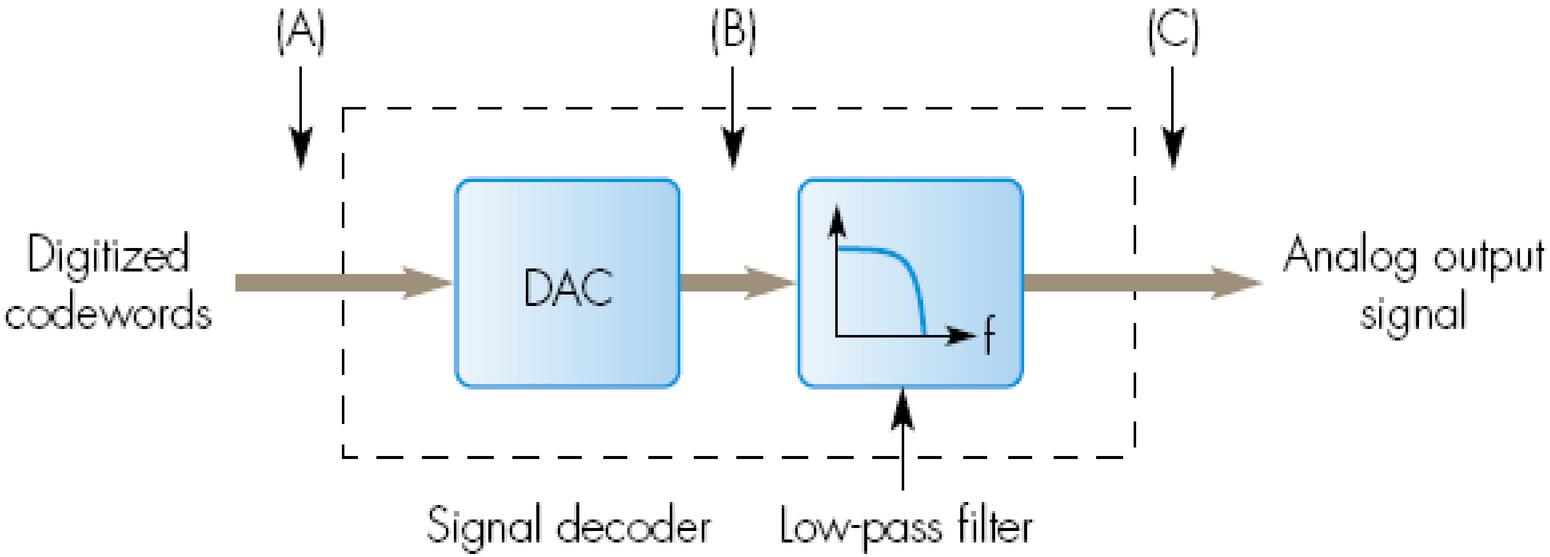
Pulse code modulation (PCM) is the name given to the class of baseband signals obtained from the quantized PAM signals by encoding each quantized sample into a *digital word* [3]. The source information is sampled and quantized to one of L levels; then each quantized sample is digitally encoded into an ℓ-bit ($\ell = \log_2 L$) codeword. For baseband transmission, the codeword bits will then be transformed to pulse waveforms. The essential features of binary PCM are shown in Figure 2.16. Assume that an analog signal $x(t)$ is limited in its excursions to the range −4 to +4 V. The step size between quantization levels has been set at 1 V. Thus, eight quantization levels are employed; these are located at −3.5, −2.5, . . . , +3.5 V. We assign the code number 0 to the level at −3.5 V, the code number 1 to the level at −2.5 V, and so on, until the level at 3.5 V, which is assigned the code number 7. Each code number has its representation in binary arithmetic, ranging from 000 for code number 0 to 111 for code number 7. Why have the voltage levels been chosen in this manner, compared with using a sequence of consecutive integers, 1, 2, 3, . . . ? The choice of voltage levels is guided by two constraints. First, the quantile intervals between the levels should be equal; and second, it is convenient for the levels to be symmetrical about zero.



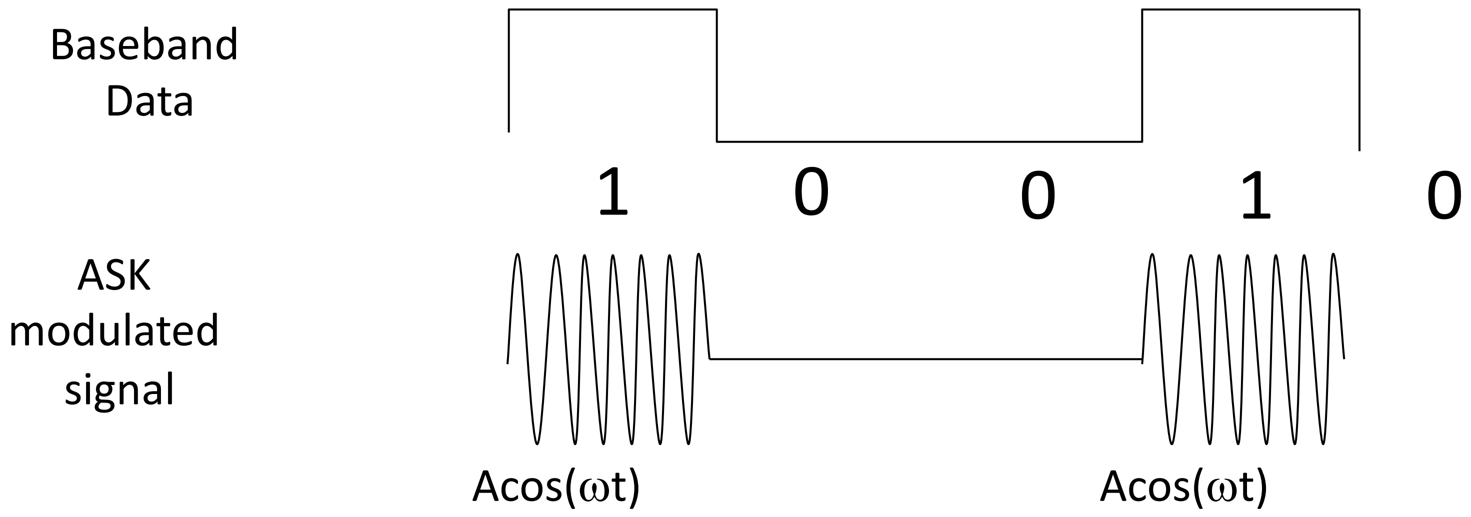
Natural sample value	1.3	3.6	2.3	0.7	–0.7	–2.4	–3.4
Quantized sample value	1.5	3.5	2.5	0.5	–0.5	–2.5	–3.5
Code number	5	7	6	4	3	1	0
PCM sequence	101	111	110	100	011	001	000

Figure 2.16 Natural samples, quantized samples, and pulse code modulation. (Reprinted with permission from Taub and Schilling, *Principles of Communications Systems,* McGraw-Hill Book Company, New York, 1971, Fig. 6.5-1, p. 205.)

Signal Decoder Design:
Circuit Components

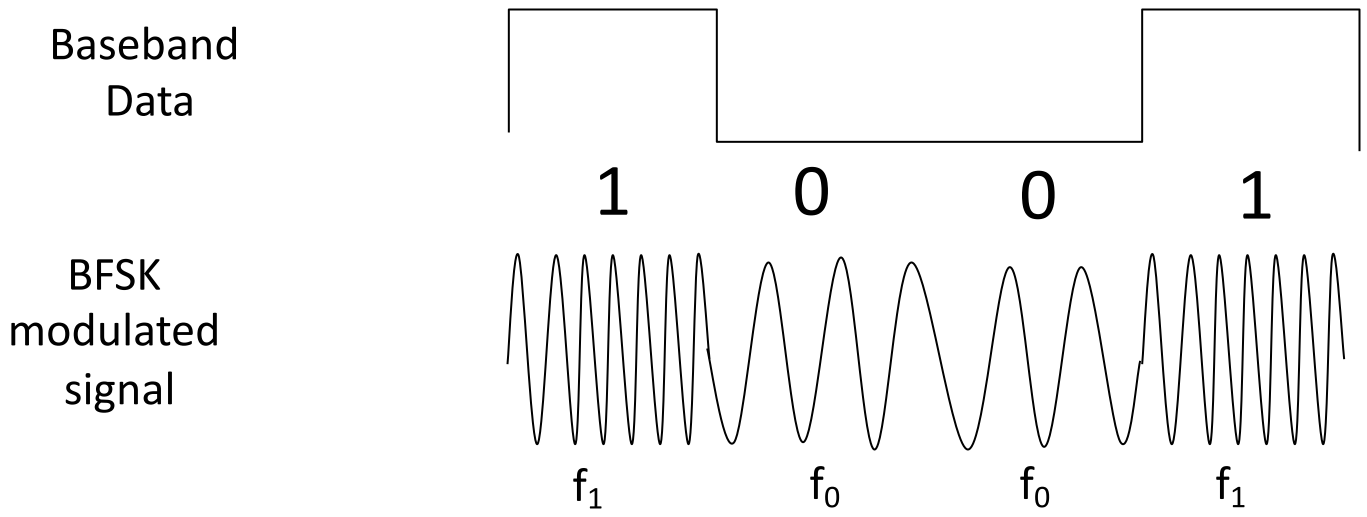


Amplitude Shift Keying (ASK)



- Pulse shaping can be employed to remove spectral spreading
- ASK demonstrates poor performance, as it is heavily affected by noise, fading, and interference

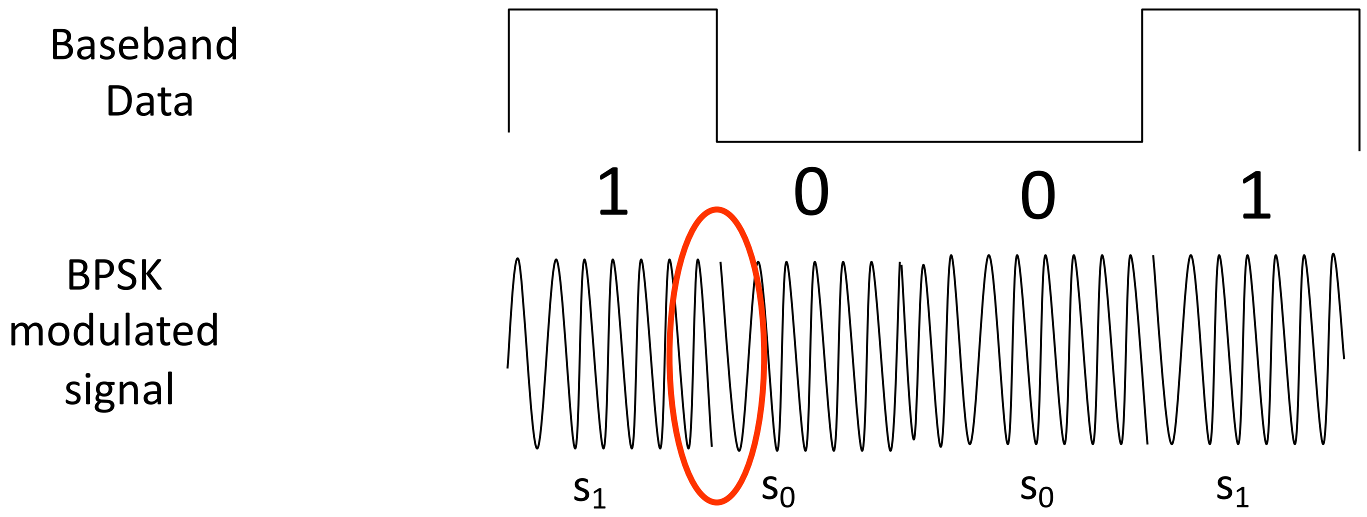
Frequency Shift Keying (FSK)



where $f_0 = A\cos(\omega_c - \Delta\omega)t$ and $f_1 = A\cos(\omega_c + \Delta\omega)t$

- Example: The ITU-T V.21 modem standard uses FSK
- FSK can be expanded to a M-ary scheme, employing multiple frequencies as different states

Phase Shift Keying (PSK)



where $s_0 = -A\cos(\omega_c t)$ and $s_1 = A\cos(\omega_c t)$

- Major drawback – rapid amplitude change between symbols due to phase discontinuity, which requires infinite bandwidth. Binary Phase Shift Keying (BPSK) demonstrates better performance than ASK and BFSK
- BPSK can be expanded to a M-ary scheme, employing multiple phases and amplitudes as different states

Examples of Modulation

- Amplitude Shift Keying (ASK) or On/Off Keying (OOK):

$$1 \Rightarrow A\cos(2\pi f_c t)$$
$$0 \Rightarrow 0$$

- Frequency Shift Keying (FSK):

$$1 \Rightarrow A\cos(2\pi f_1 t)$$
$$0 \Rightarrow A\cos(2\pi f_0 t)$$

- Phase Shift Keying (PSK):

$$1 \Rightarrow A\cos(2\pi f_c t)$$
$$0 \Rightarrow A\cos(2\pi f_c t + \pi) = -A\cos(2\pi f_c t)$$

What Makes a Good Communication System?

- Large data rate (measured in bits/sec)
- Small bandwidth (measured in Hertz)
- Small signal power (measured in Watts or dBW)
- Low distortion (measured in S/N or bit error rate)
- Low cost - with digital communications, large complexity does not always result in large cost
- In practice, there must be tradeoffs made in achieving these goals

Tradeoffs in System Design: Data Rate vs. Bandwidth

- Increased data rate leads to shorter data pulses which leads to larger bandwidth.
- This tradeoff cannot be avoided - however, some systems use bandwidth more efficiently than others.
- We will define Bandwidth Efficiency as the ratio of data rate R_b to bandwidth W: $\eta_B = R_b / W$
- We want large bandwidth efficiency η_B

Bits/s/Hz

Tradeoffs in System Design: Fidelity vs. Signal Power

- One way to get an error free signal would be to use huge amounts of power to blast over the noise.
- Some types of modulation achieve relative error free transmission at lower powers than others.
- We define Energy Efficiency: $\eta_E = E_b / N_o \big|_{P_b = \text{target error rate}}$
- We desire small η_E

Tradeoffs in System Design: Bandwidth Efficiency vs. Energy Efficiency

- It is possible for a system design to trade between bandwidth efficiency and energy efficiency.
- Examples:
  - Binary modulation sends only one bit per use of the channel. M-ary modulation can send multiple bits, but is more vulnerable to errors.
  - Error correction coding: inserting redundant bits improves bit error rate, but increases bandwidth.
- This is a fundamental tradeoff in digital communications.