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# Digital Communication Systems

## Session 1

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# Information Representation

Communication systems convert information into a form suitable for transmission

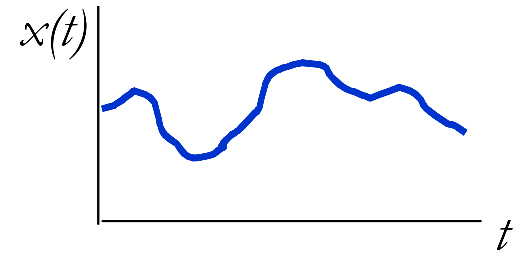
Analog systems→Analog signals are modulated (AM, FM radio)

Digital system generate bits and transmit digital signals (Computers)

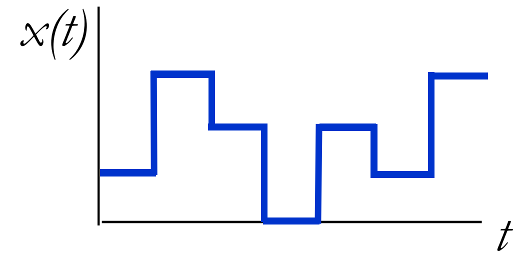
Analog signals can be converted to digital signals.

# Analog vs. Digital

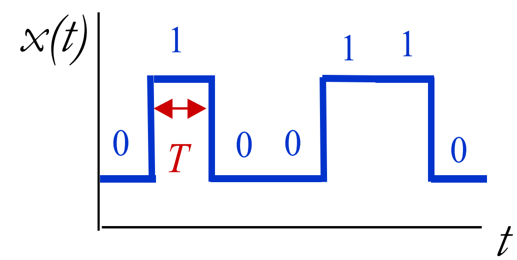
- Analog signals
  - Value varies continuously




- Digital signals
  - Value limited to a finite set




- Binary signals
  - Has at most 2 values
  - Used to represent bit values
  - Bit time $T$ needed to send 1 bit
  - Data rate $R=1/T$ bits per second



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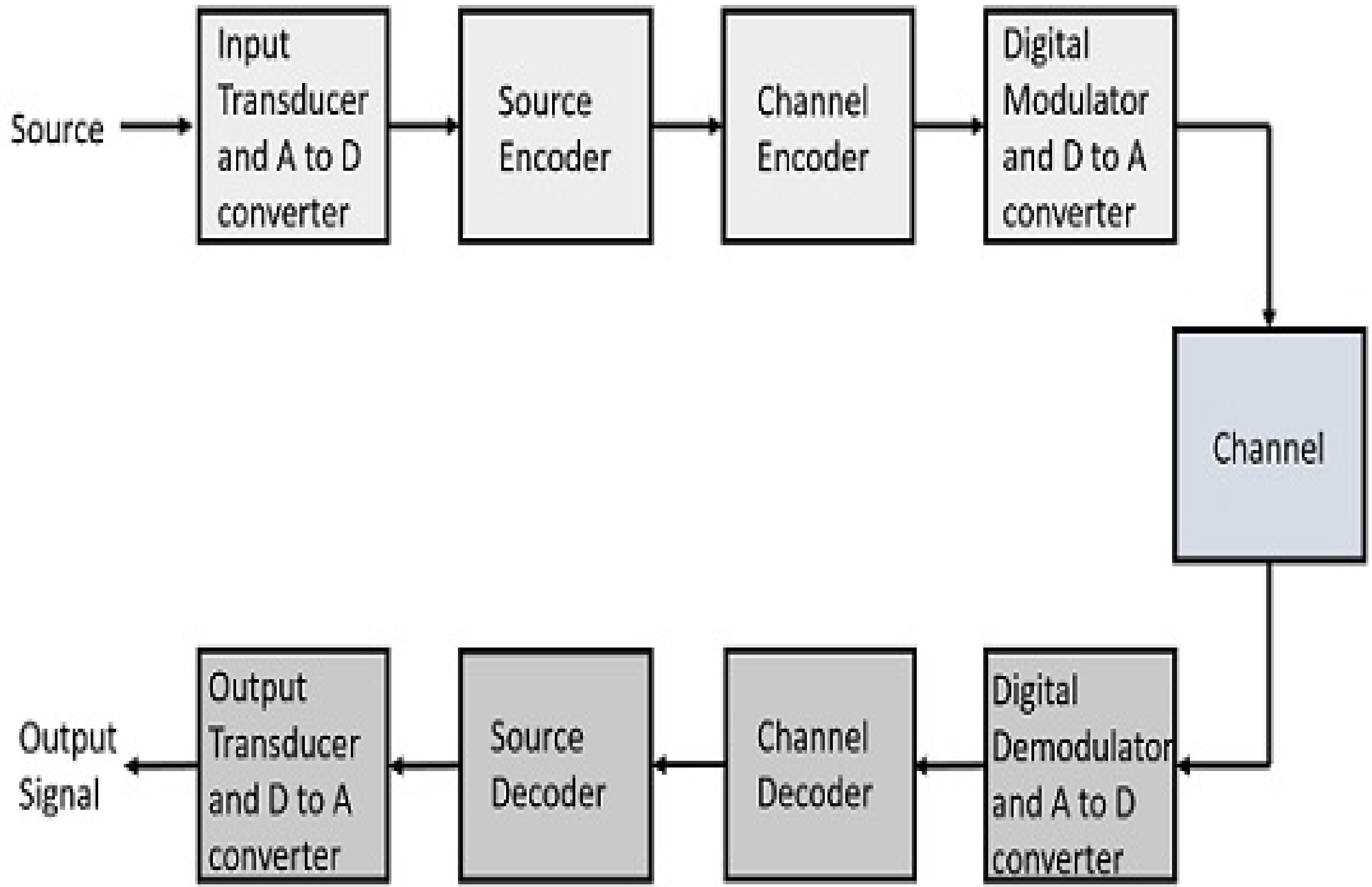
# Block Diagram



Basic Elements of a Digital Communication System

# Digital Data System

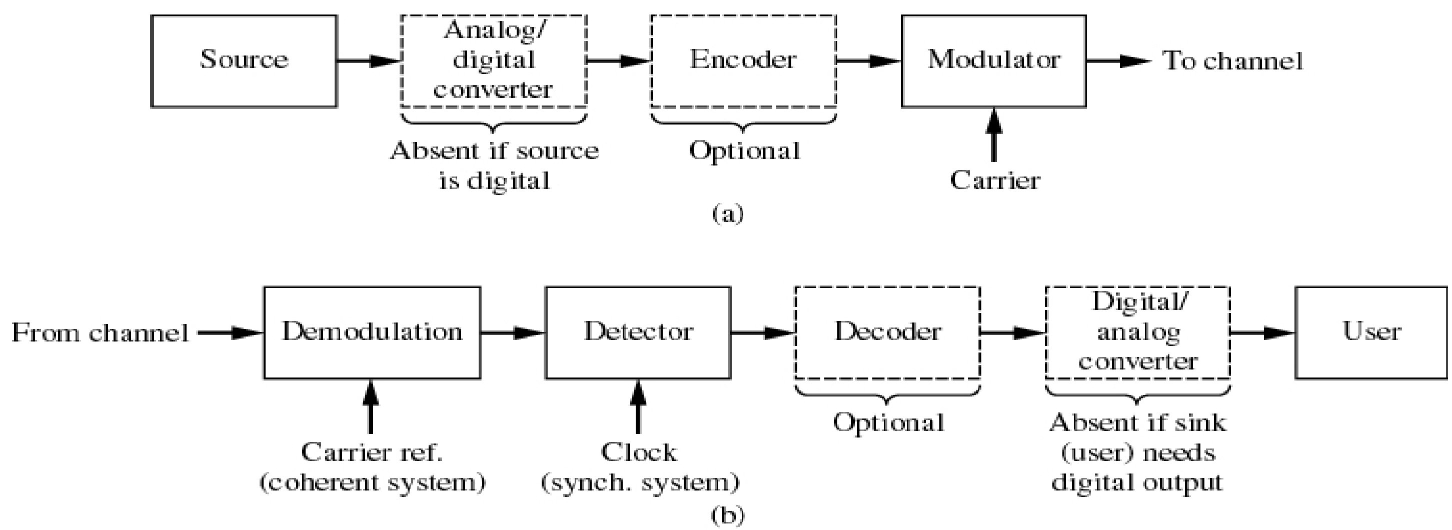



**Figure 7-1** Block diagram of a digital data system. (a) Transmitter. (b) Receiver.

# Components of Digital Communication

**Sampling:** If the message is analog, it's converted to discrete time by sampling.

(What should the sampling rate be ?)

**Quantization:** Quantized in amplitude.

Discrete in time and amplitude

**Encoder:**

Convert message or signals in accordance with a set of rules

Translate the discrete set of sample values to a signal.

**Decoder**: Decodes received signals back into original message

# Advantages

- Stability of components: Analog hardware change due to component aging, heat, etc.
- Flexibility:
  - Perform encryption
  - Compression
  - Error correction/detection
- Reliable reproduction

# Advantages of Digital communication

- The greatest advantage of digital communication over analog communication is the viability of regenerative repeaters in the former.
- Digital hardware implementation is flexible and permits the use of microprocessors, mini-processors, digital switching, and large scale integrated circuits.
- Digital Signals can be coded to yield extremely low error rates and high fidelity as well as privacy.
- It is easier and more efficient to multiplex several digital signals.
- Digital communication is inherently more efficient than analog in realizing the exchange of SNR for bandwidth.
- Digital signal storage is relatively easy and inexpensive.

# Applications

Digital Audio Transmission

Telephone channels

Lowpass filter, sample, quantize

32kbps-64kbps (depending on the encoder)

Digital Audio Recording

LP vs. CD

Improve fidelity (How?)

More durable and don't deteriorate with time

# Classification of Digital Communication

- Digital communication is classified into two types.

(1) Baseband data transmission

(2) Bandpass data transmission

- Baseband data transmissions are of two types
  - a) Pulse analog (PAM, PWM, PPM)
  - b) Pulse digital (PCM, DPCM, DM, ADM)
- The fundamental difference between baseband and bandpass data transmission is with respect to channel.
- For bandpass, the channel is free space and for baseband, the channel is coaxial cable or fibre optic cable or twisted pair.

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# Sampling: Time Domain

- Many signals originate as continuous-time signals, e.g. conventional music or voice
- By sampling a continuous-time signal at isolated, equally-spaced points in time, we obtain a sequence of numbers

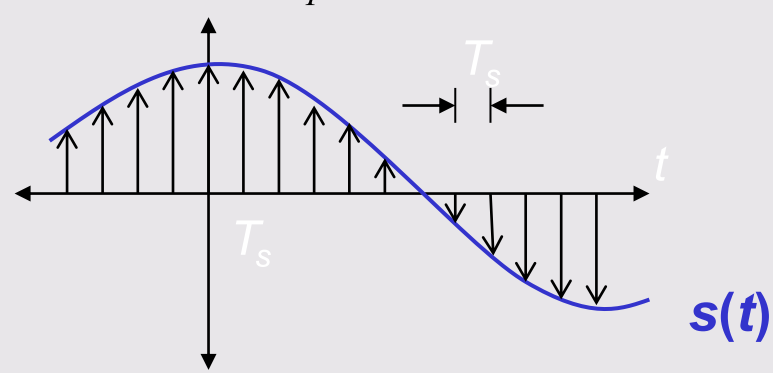
$n \in \{\ldots, -2, -1, 0, 1, 2, \ldots\}$

$T_s$ is the sampling period.

$$s[n] = s(n\,T_s)$$

$$s_{sampled}(t) = s(t)\underbrace{\sum_{n=-\infty}^{\infty}\delta(t - n\,T_s)}_{\text{impulse train}}$$

$s_{sampled}(t)$




***Sampled analog waveform***

# Sampling: Frequency Domain

Replicates spectrum of continuous-time signal

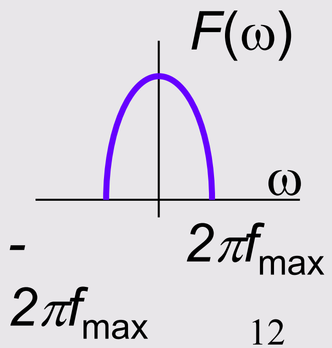
At offsets that are integer multiples of sampling frequency

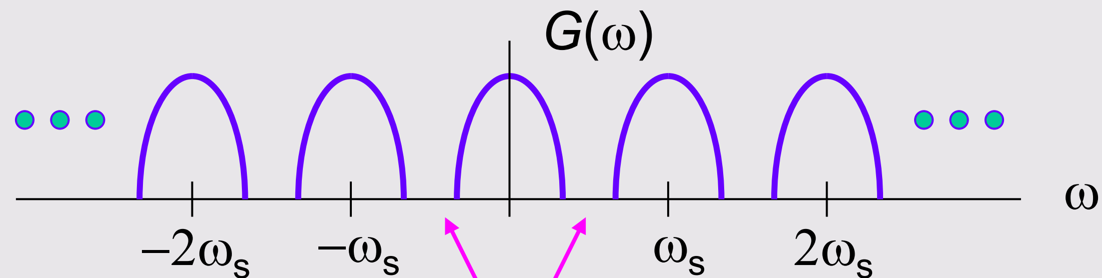
Fourier series of impulse train where $w_s = 2\ \mathrm{p}\ f_s$

## Example

$$g(t) = f(t)\,\delta_{T_s}(t) = \frac{1}{T_s}\left( f(t) + 2 f(t)\cos(\omega_s t) + 2 f(t)\cos(2\,\omega_s t) + \ldots \right)$$

$$\delta_{T_s}(t) = \sum_{n=-\infty}^{\infty} \delta(t - n\,T_s) = \frac{1}{T_s} + \underbrace{\frac{2}{T_s}\cos(\omega_s t)}_{\text{Modulation by }\cos(\omega_s t)} + \underbrace{\frac{2}{T_s}\cos(2\,\omega_s t)}_{\text{Modulation by }\cos(2\omega_s t)} + \ldots$$




gap if and only if $2\pi f_{max} < 2\pi f_s - 2\pi f_{max} \Leftrightarrow f_s > 2 f_{max}$

# Shannon Sampling Theorem

A continuous-time signal $x(t)$ with frequencies no higher than $f_{max}$ can be reconstructed from its samples $x[n] = x(n\ T_s)$ if the samples are taken at a rate $f_s$ which is greater than 2 $f_{max}$.

Nyquist rate = 2 $f_{max}$

Nyquist frequency = $f_s/2$.

What happens if $f_s = 2f_{max}$?

Consider a sinusoid sin(2 p $f_{max}$ $t$)

Use a sampling period of $T_s = 1/f_s = 1/2f_{max}$.

Sketch: sinusoid with zeros at $t = 0$, $1/2f_{max}$, $1/f_{max}$, …



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# Shannon Sampling Theorem

## Assumption

Continuous-time signal has no frequency content above $f_{max}$

Sampling time is exactly the same between any two samples

Sequence of numbers obtained by sampling is represented in exact precision

Conversion of sequence to continuous time is ideal

## In Practice

# Why 44.1 kHz for Audio CDs?

Sound is audible in 20 Hz to 20 kHz range:

- $f_{max}$ = 20 kHz and the Nyquist rate 2 $f_{max}$ = 40 kHz

What is the extra 10% of the bandwidth used?

- Rolloff from passband to stopband in the magnitude response of the anti-aliasing filter

Okay, 44 kHz makes sense. Why 44.1 kHz?

- At the time the choice was made, only recorders capable of storing such high rates were VCRs.
- NTSC: 490 lines/frame, 3 samples/line, 30 frames/s = 44100 samples/s
- PAL: 588 lines/frame, 3 samples/line, 25 frames/s = 44100 samples/s



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# Sampling

As sampling rate increases, sampled waveform looks more and more like the original

Many applications (e.g. communication systems) care more about frequency content in the waveform and not its shape

Zero crossings: frequency content of a sinusoid

- Distance between two zero crossings: one half period.
- With the sampling theorem satisfied, sampled sinusoid crosses zero at the right times even though its waveform shape may be difficult to recognize

# Aliasing

Analog sinusoid

$x(t) = A\cos(2\pi f_0 t + \phi)$

Sample at $T_s = 1/f_s$

$x[n] = x(T_s\, n) =$
$A\cos(2\pi\, f_0\, T_s\, n + \phi)$

Keeping the sampling period same, sample

$y(t) = A\cos(2\pi(f_0 + l f_s)t + \phi)$

where $l$ is an integer

$$\begin{aligned} y[n] &= y(T_s\, n) \\ &= A\cos(2\pi(f_0 + l f_s)T_s n + \phi) \\ &= A\cos(2\pi f_0 T_s n + 2\pi\, l f_s T_s n + \phi) \\ &= A\cos(2\pi f_0 T_s n + 2\pi\, l\, n + \phi) \\ &= A\cos(2\pi f_0 T_s n + \phi) \\ &= x[n] \end{aligned}$$

Here, $f_s T_s = 1$

Since $l$ is an integer, $\cos(x + 2\pi l) = \cos(x)$

$y[n]$ indistinguishable from $x[n]$

Frequencies $f_0 + l\, f_s$ for $l \neq 0$ are aliases of frequency $f_0$