Multimedia System

Lecture 2 and 3: Digital Data Acquisition

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4. October 2017
11. October 2017

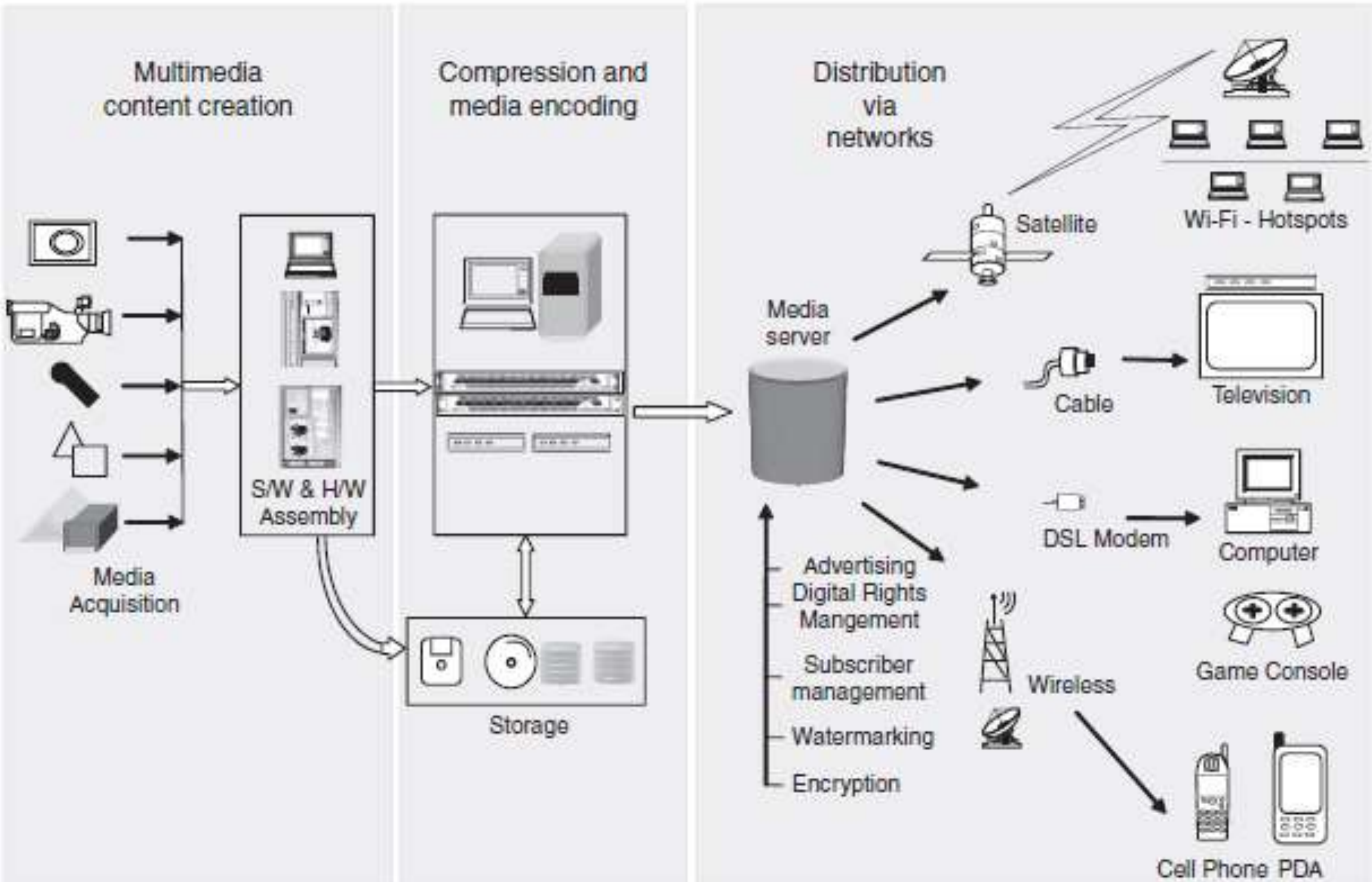
Outlines

- Analog and Digital Signals
- Analog to digital conversion
- Sampling theorem and aliasing
- Filtering

Overview

- Multimedia systems involve three major components:
  - multimedia content creation,
  - compression/storage of multimedia content
  - delivery or distribution of multimedia content.
- Multimedia information is
  - digital,
  - interactive
  - voluminous
- As depicted in the end-to-end multimedia system diagram Figure 1-1, one of the first tasks in creating multimedia content using text, audio, video, and images is to record these individual media types into a digital form, making it is easy to combine and assemble these heterogeneous entities

Overview


Multimedia content creation
Compression and media encoding
Distribution via networks
Media Acquisition
S/W & H/W Assembly
Storage
Media server
Advertising
Digital Rights Mangement
Subscriber management
Watermarking
Encryption
Satellite
Wi-Fi - Hotspots
Cable
Television
DSL Modem
Computer
Wireless
Game Console
Cell Phone
PDA

In this lecture

- This lecture describes the theoretical foundations underpinning the conversion and recording of information into a digital medium.
- It brings forth issues involved in digitizing
  - one-dimensional (such as audio),
  - two-dimensional (such as images),
  - three-dimensional (such as video) signals.
- Discusses the fundamental digitization process.
- Present common problems that occur during digitization and solutions to overcome them.

Introduction

- The physical world around us exists in a continuous form. We sense the environment by sensing light, sound energy, pressure, temperature, motion, and so on. All these properties are continuously changing.
- Recording instruments, such as cameras, camcorders, microphones, gauges, and so on, attempt to measure information in an electrical and digital form.
- Let us take the example of a digital camera. In the camera, there could be an image sensor CCD (charge coupled device) array. Each sensor releases an electric charge that is proportional to the amount of light energy falling on it; the more energy, the higher the charge (within a range).
  - The released charge is then converted into a digital representation in terms of bits, which are ultimately used to display the image information on a rendering device.

Introduction

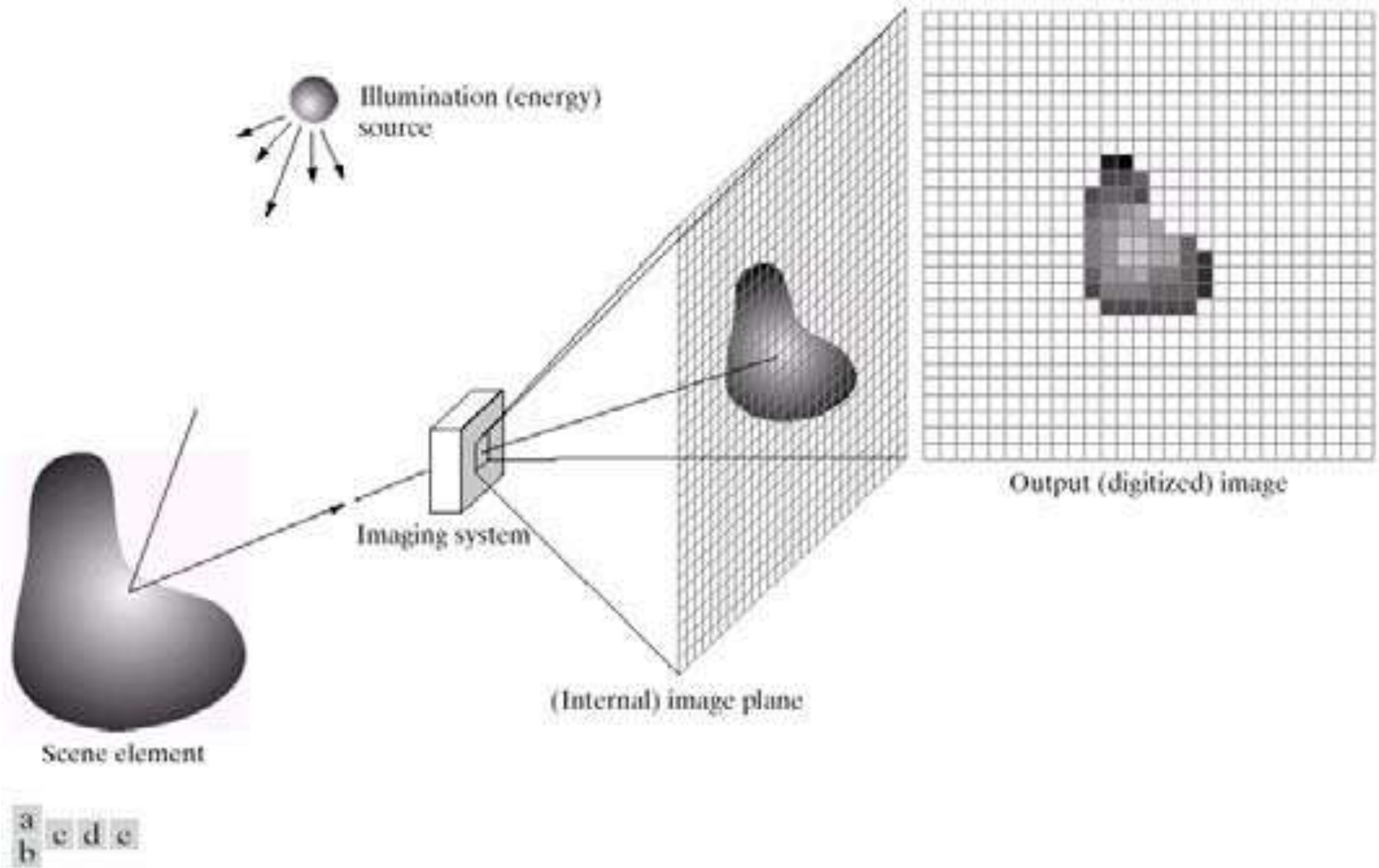


FIGURE 2.15 An example of the digital image acquisition process. (a) Energy ("illumination") source. (b) An element of a scene. (c) Imaging system. (d) Projection of the scene onto the image plane. (e) Digitized image.

Introduction

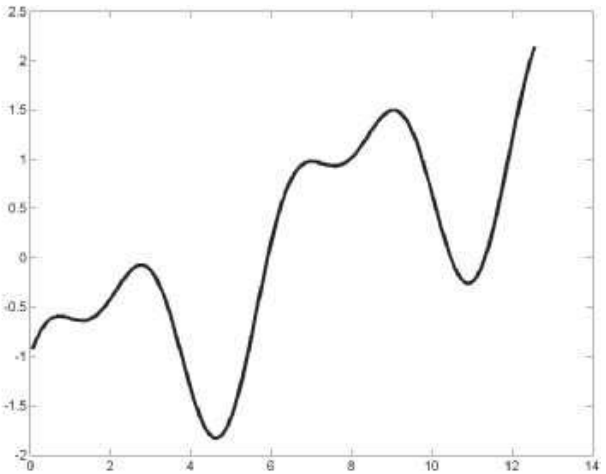
- It is natural to reason that because multimedia information and systems deal with digital data, we might as well assume that we start with digital data and bypass the understanding of conversions and processes necessary to obtain digital data.
- However, the conversion process, also known as analog-to-digital conversion, ultimately conditions the quality of digital data, as well as the quantity —both of which are important to the creation and distribution of multimedia.
- Understanding the conversion process helps in the design of end-to-end systems with the necessary digital data generation for the desired quality, at the same time keeping the generated quantity within the allowed bandwidth.

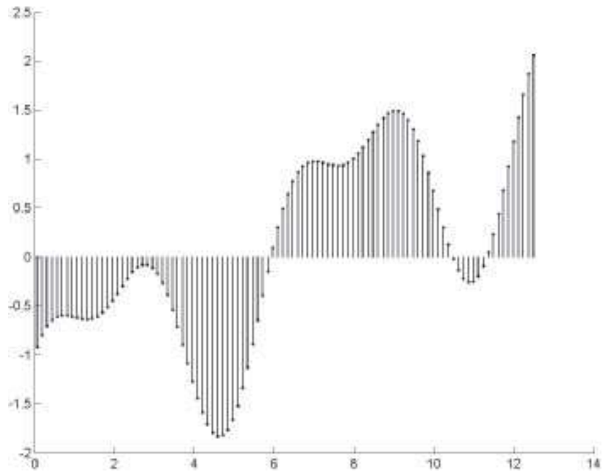
Outlines

- Analog and Digital Signals
- Analog to digital conversion
- Sampling theorem and aliasing
- Filtering

Analog and Digital Signals

- Analog signals are captured by a recording device, which attempts to record a physical signal.
  - A signal is analog if it can be represented by a continuous function. For instance, it might encode the changing amplitude with respect to an input dimension(s).
- Digital signals are represented by a discrete set of values defined at specific instances of the input domain, which might be time, space, or both.





Advantages of Digital Signals over Analog one

- When media is represented digitally, it is possible to create complex, interactive content. In the digital medium, we can access each unit of information for a media type, for example:
  - it is easy to access a pixel in an image, or a group of pixels in a region or even a section of a sound track.
  - Different digital operations can be applied to each, such as to enhance the image quality of a region, or to remove noise in a sound track.
  - Also, different digital media types can be combined or composited to create richer content, which is not easy in the analog medium.
- Stored digital signals do not degrade over time or distance as analog signals do.
  - One of the most common artifacts of broadcast VHS video is ghosting, as stored VHS tapes lose their image quality by repeated usage and degradation of the medium over time.
  - This is not the case with digital broadcasting or digitally stored media types.

Advantages of Digital Signals over Analog one (cont'd)

- Digital data can be efficiently compressed and transmitted across digital networks.
- It is easy to store digital data on magnetic media such as portable 3.5 inch, hard drives, or solid state memory devices, such as flash drives, memory cards, and so on.
  - This is because the representation of digital data, whether audio, image, or video, is a set of binary values, regardless of data type.

So, digital data is preferred because it offers better quality and higher fidelity, can be easily used to create compelling content, and can also be compressed, distributed, stored, and retrieved relatively easily.

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- Analog and Digital Signals
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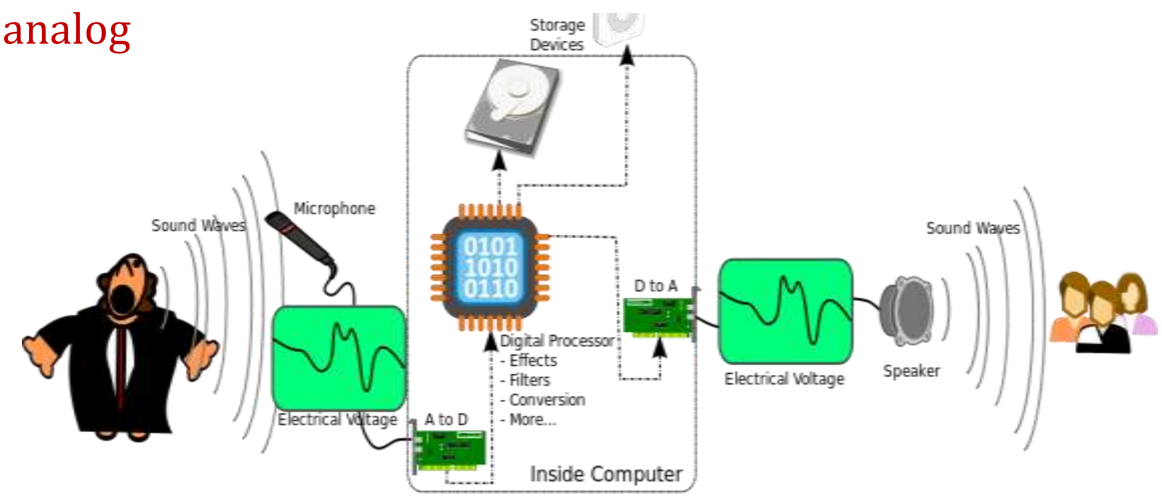
Analog to digital conversion (cont'd)

- Conversion Process
  1. A digital audio system starts with an ADC that converts an analog signal to a digital signal.
  2. A digital audio signal may be stored or transmitted. Digital audio can be stored on a CD, a **digital audio player**, a **hard drive**, a **USB flash drive**, or any other digital **data storage device**. The digital signal may then be altered through **digital signal processing**, where it may be **filtered** or have **effects** applied. **Audio data compression** techniques, such as **MP3**, or **Advanced Audio Coding**, are commonly employed to reduce the file size. Digital audio can be **streamed** to other devices.
  3. For playback, digital audio must be converted back to an analog signal with a DAC.



Analog to digital conversion

- Computers work with discrete pieces of information
- How do we digitize a continuous image?

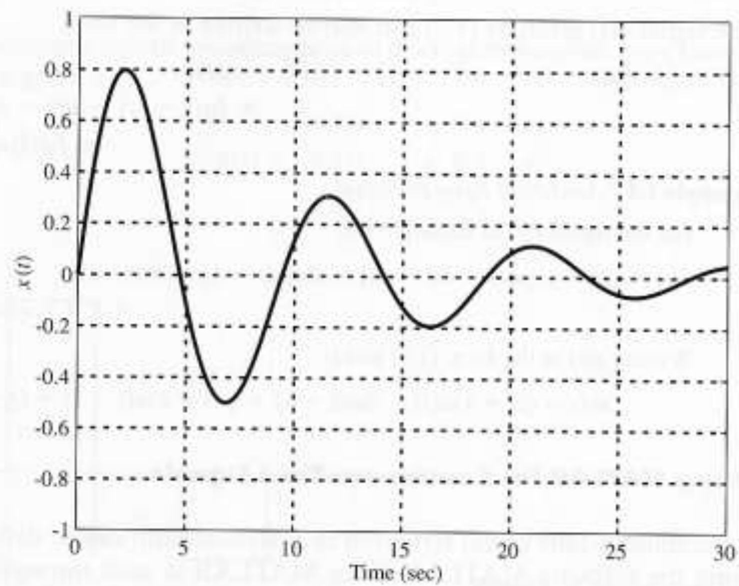


Figure 1.13 MATLAB plot of the signal $x(t) = e^{-0.1t} \sin \frac{2}{3}t$.

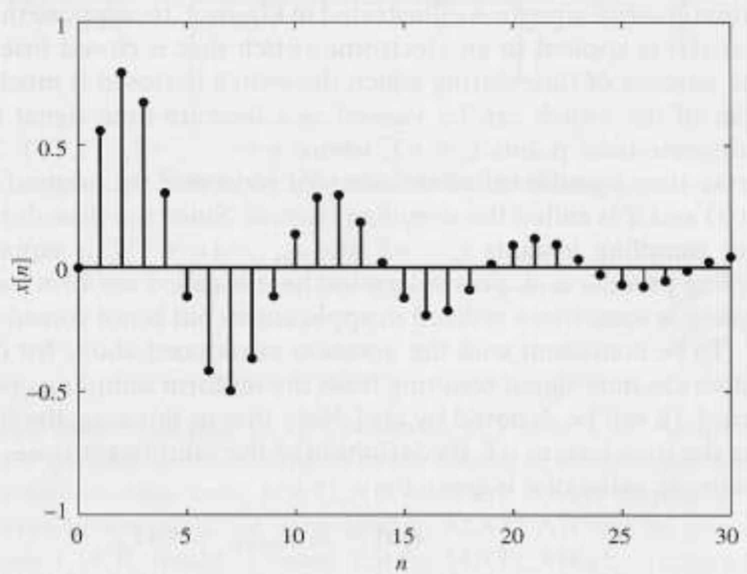


Figure 1.17 Sampled continuous-time signal.

Analog to digital conversion (cont'd)

- The conversion of signals from analog to digital occurs via two main processes:
  - *sampling*
  - *quantization*.
- The reverse process of converting digital signals to analog is known as ***interpolation***.
- One of the most desirable properties in the analog to digital conversion is to ensure that no artifacts are created in the digital data.
  - That way, when the signal is converted back to the analog domain, it will look the same as the original analog signal.

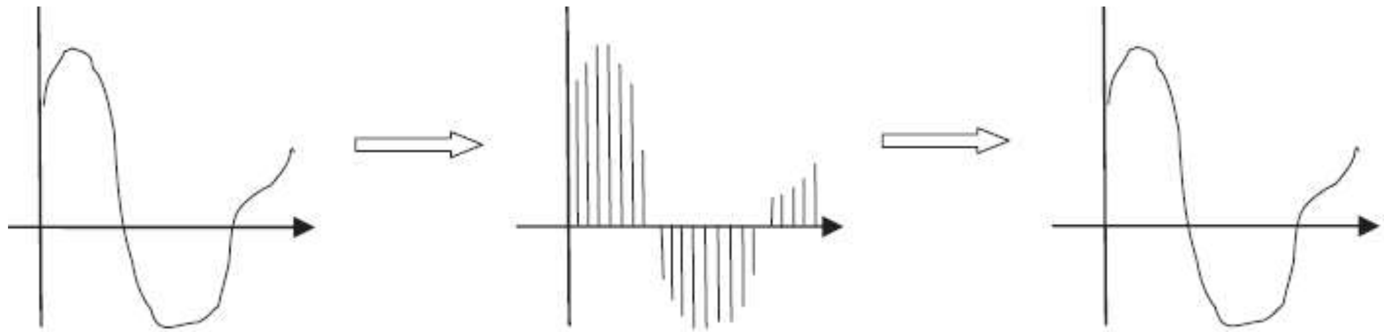
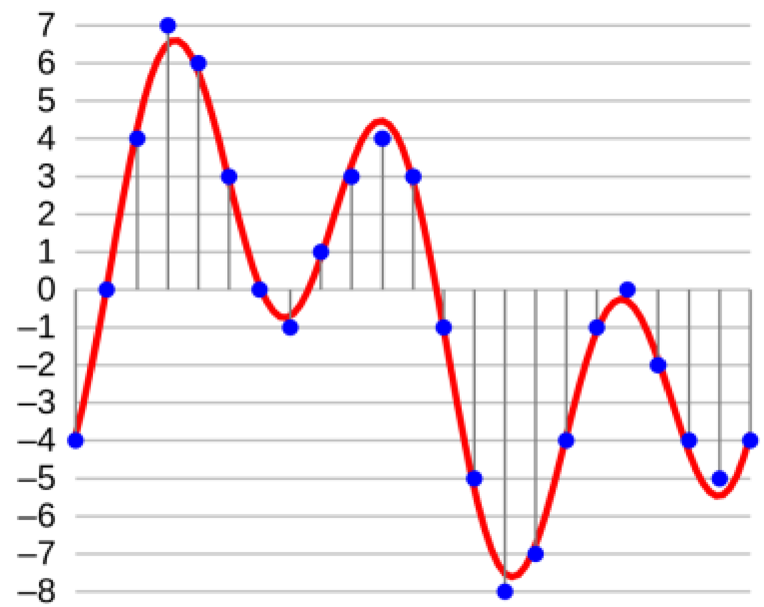


Figure 2-2 Analog-to-digital conversion and the corresponding interpolation from the digital-to-analog domain

Analog to digital conversion (cont'd)

- Basic Concepts in Sampling and Quantization
  - To convert an image to digital form, we have to sample the function in both coordinates and in amplitude.
  - Digitizing the coordinate values is called **sampling**.
  - Digitizing the amplitude values is called **quantization**.



Sampling

- For commonly used signals, sampling is done across one dimension (time, for sound signals), two dimensions (spatial x and y, for images), or three dimensions (*x, y*, time for video).
- Assume that we start with a one-dimensional analog signal in the time t domain, with an amplitude given by $x(t)$. The sampled signal is given by

  $x_s(n) = x(nT)$, where T is the sampling period

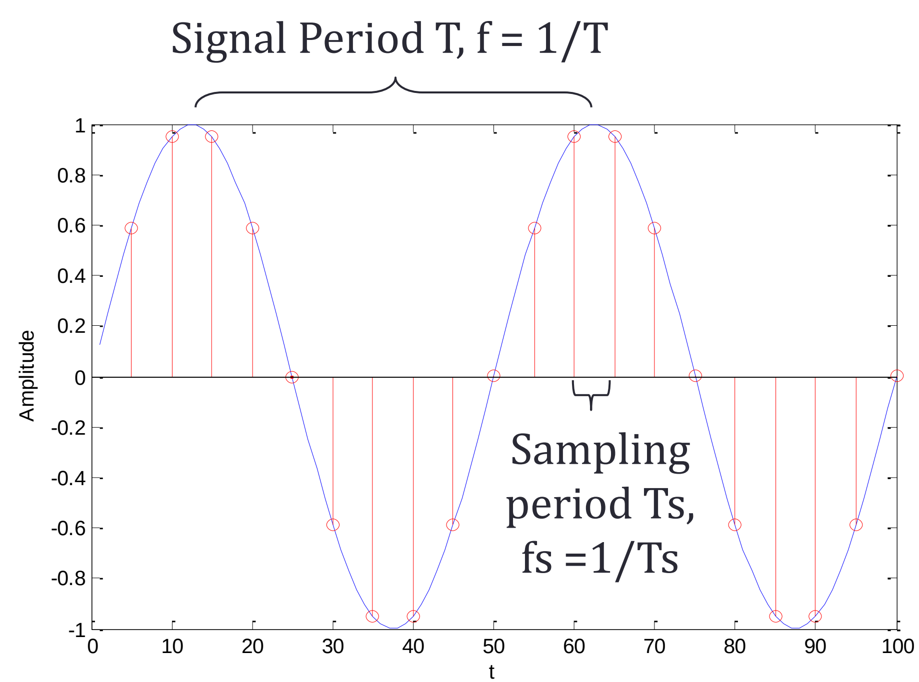
  $f = 1/T$ is the sampling frequency

  where T is the time interval between samples

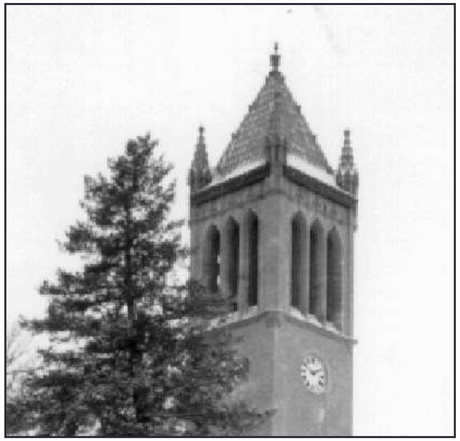
  f is sampling rate = no of samples per second

Sampling (cont'd)

- If you reduce *T* (increase *f*), the number of samples increases; and correspondingly, so does the storage requirement.
- Vice versa, if *T* increases (*f* decreases), the number of samples collected for the signal decrease and so does the storage requirement.
- *T* is clearly a critical parameter. Should it be the same for every signal?
  - If *T* is too large, the signal might be under sampled, leading to artifacts,
  - If *T* is too small, the signal requires large amounts of storage, which might be redundant.



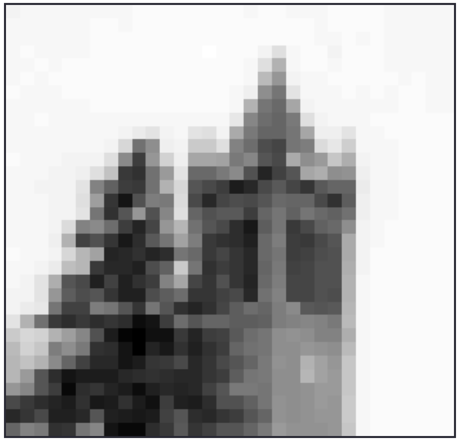
Sampling (cont'd)



256x256 pixels

128x128 pixels

64x64 pixels



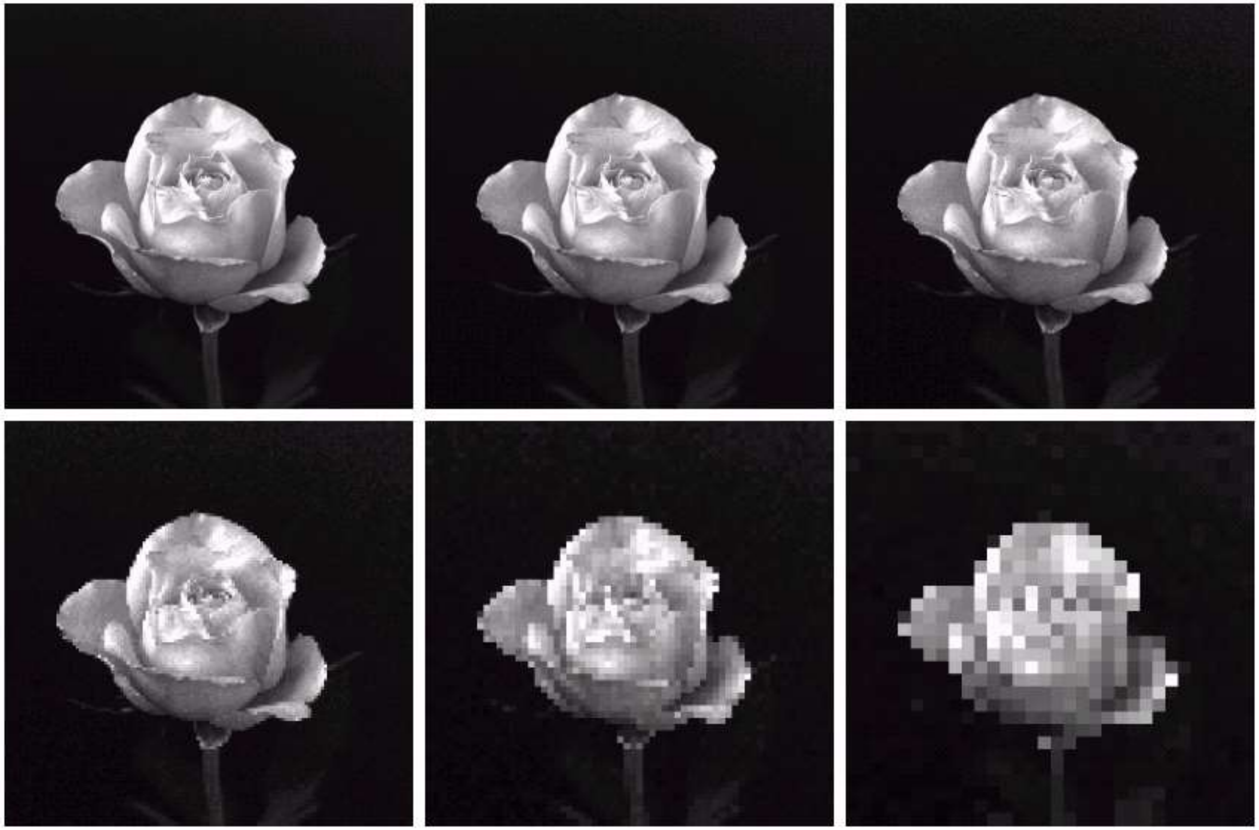
32x32 pixels

The effect of lowering the spatial resolution

(Example 1)

- **Higher sampling rates** allow the image to be more accurately represented and more storage.
- **Lower sampling rate**, allow less quality, less information and we will use less storage space and transmission will be faster.

Sampling (cont'd)



a b c
d e f

FIGURE 2.20 (a) 1024 × 1024, 8-bit image. (b) 512 × 512 image resampled into 1024 × 1024 pixels by row and column duplication. (c) through (f) 256 × 256, 128 × 128, 64 × 64, and 32 × 32 images resampled into 1024 × 1024 pixels.

The effect of lowering the spatial resolution

(Example 2)

Quantization

- Quantization deals with encoding the signal value at every sampled location with a predefined precision, defined by a number of levels.
- In other words, now that you have sampled a continuous signal at specific regular time instances, how many bits do you use to represent the value of signal at each instance?
- The entire range R of the signal is represented by a finite number of bits b. Formally,

$$x_q(n) = Q[x_s(n)]$$

- Q represents a rounding function that maps the continuous value $x_s(n)$ to the nearest digital value using b bits.
- Utilizing b bits corresponds to $N = 2^b$ levels, thus having a quantization step $delta = R/2^b$.
- Following figure shows an analog signal, which is sampled at a common frequency, but quantized using different number of bits, 4 or 2.

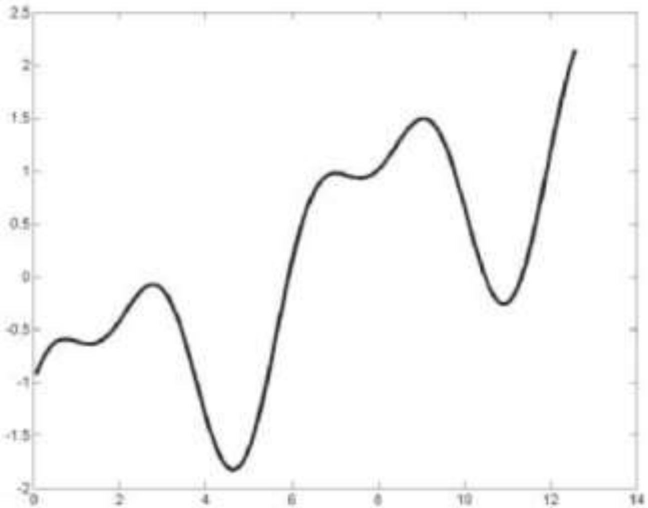
Quantization

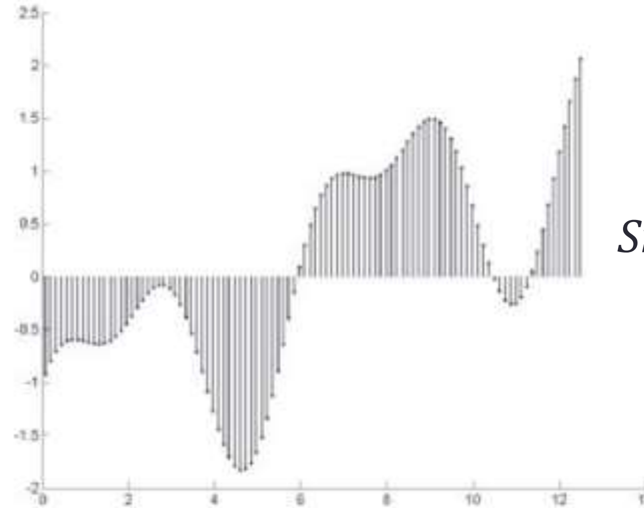
Notes

- The ***quantization*** of a sample value is dependent on the number of bits used to represent the amplitude.
- The greater the number of bits used, the better the resolution, but the more storage space is required.
- ***Quantization*** - divide the vertical axis (amplitude) into pieces.
  - 8 bit quantization divides the vertical axis into 256 levels.
  - 16 bit gives you 65536 levels.
- The value of each sample is rounded off to the nearest integer (**quantization**).

Quantization (cont'd)

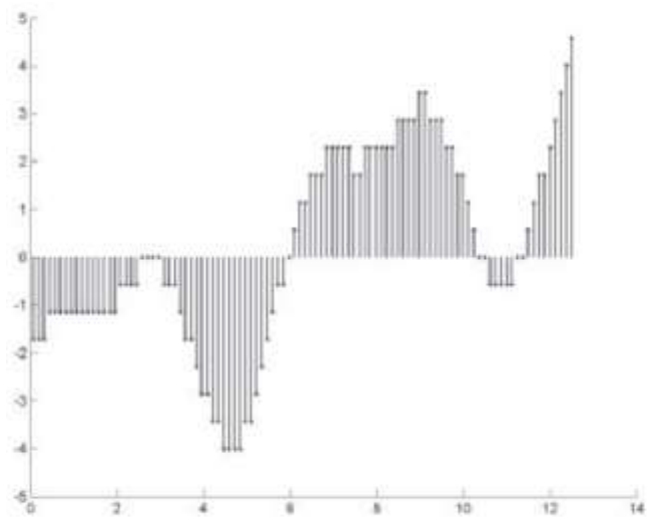
Original analog signal

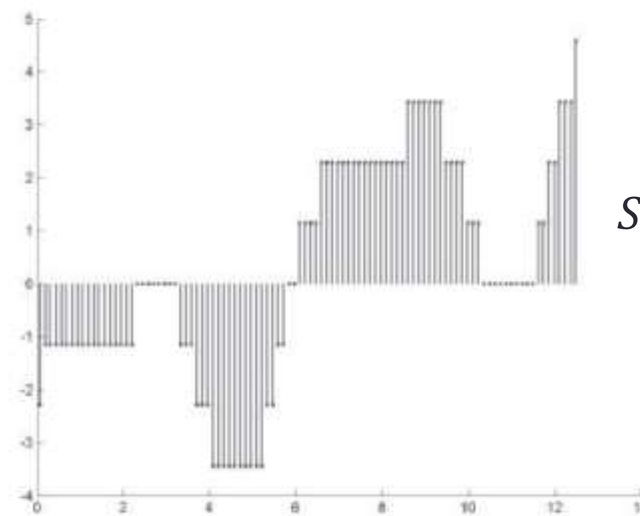




Quantized Signal at 8 bits (256 levels)

Quantized Signal at 4 bits (16 levels)

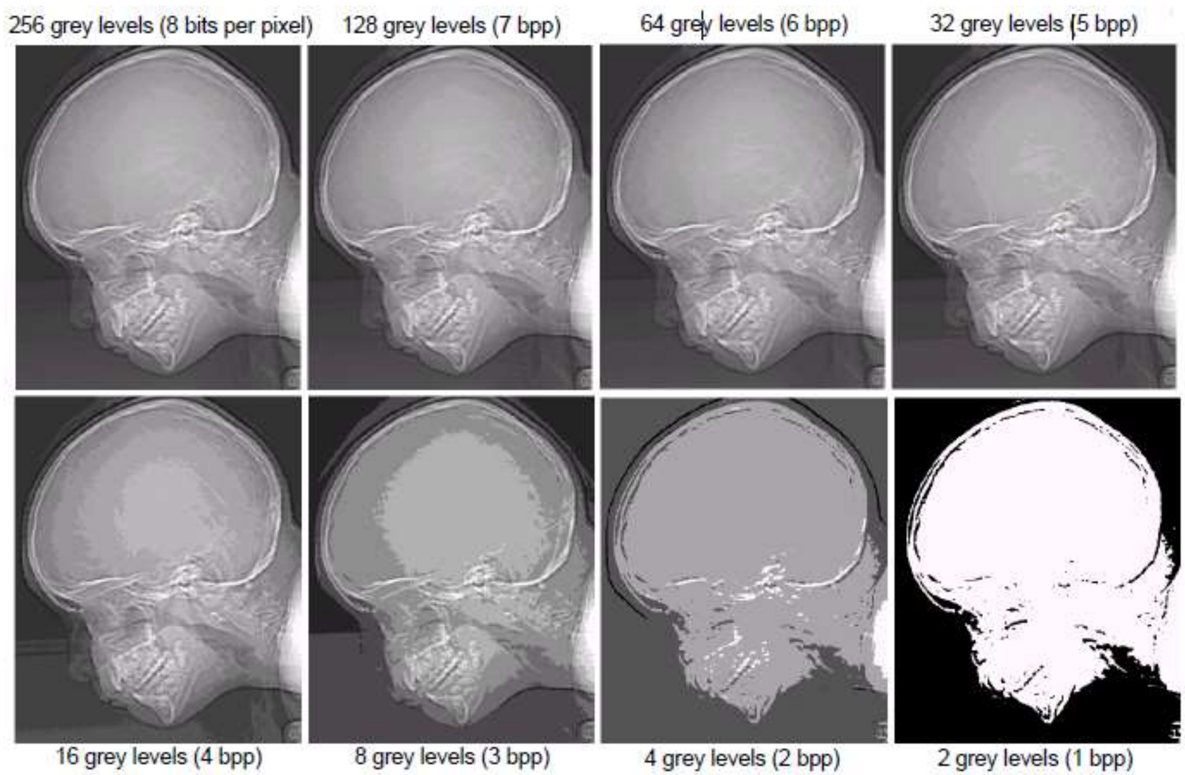




Quantized Signal at 3 bits (8 levels)

Quantization (cont'd)

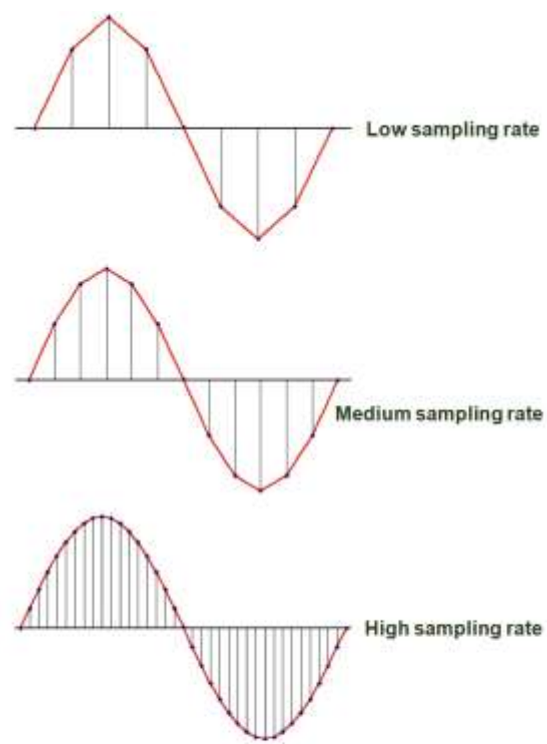
- Following Figures illustrates quantization effects in two dimensions for images.
- The results show that the error increases as the number of quantization bits used to represent the pixel samples decreases.



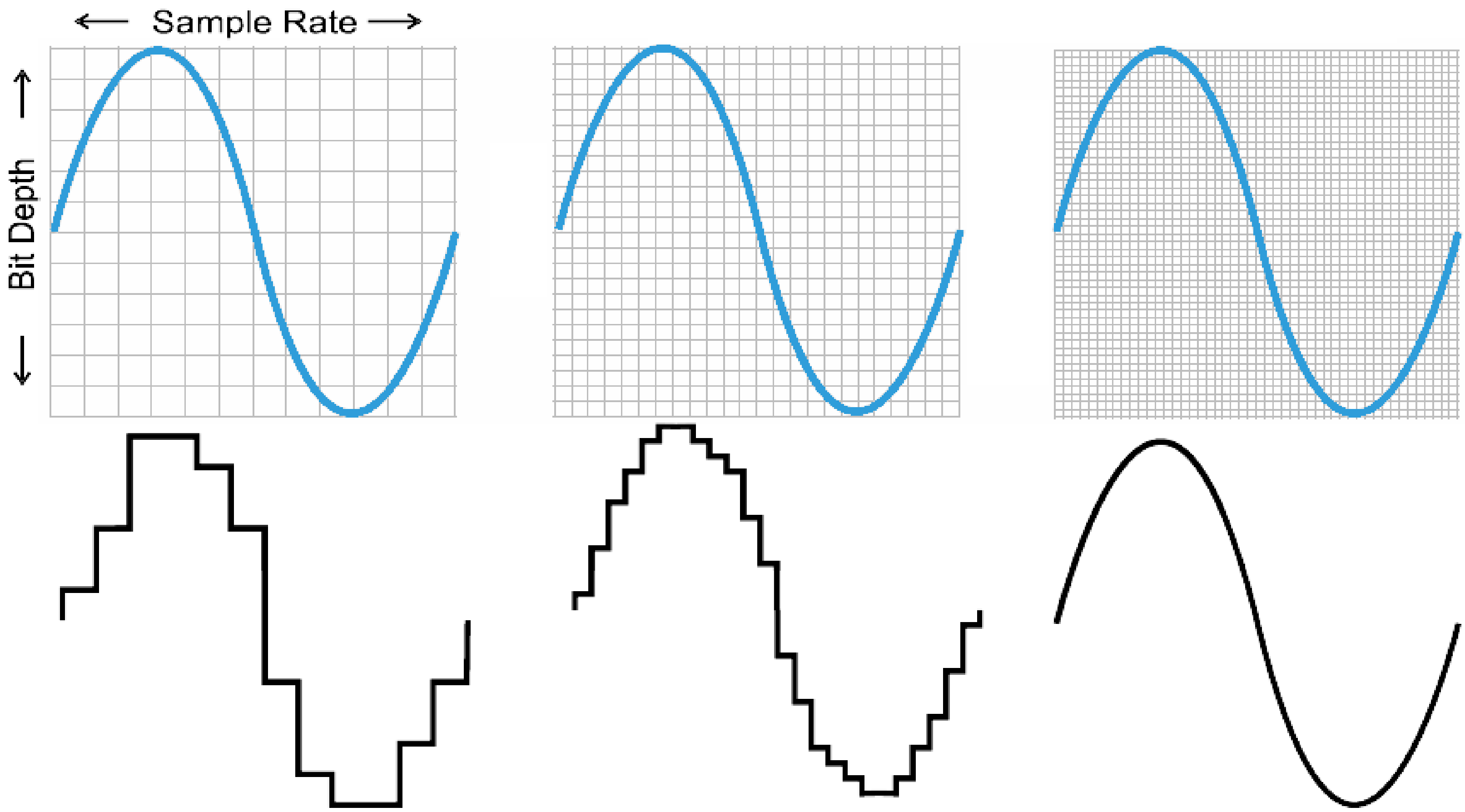
- False contouring effect is quite visible in images displayed using 16 or less gray levels

Quantization (cont'd)

- **There are two factors affects the quality of digital records:**
  - **Sampling rate**
    - Number of samples per second.
      - **Higher sampling rates** allow the signals to be more accurately represented and more storage.
      - **Lower sampling rate**, allow less quality, less information and we will use less storage space and transmission will be faster.
  - **Bit depth (sample size)**:
    - The number of bits required to represent the value of each sample.



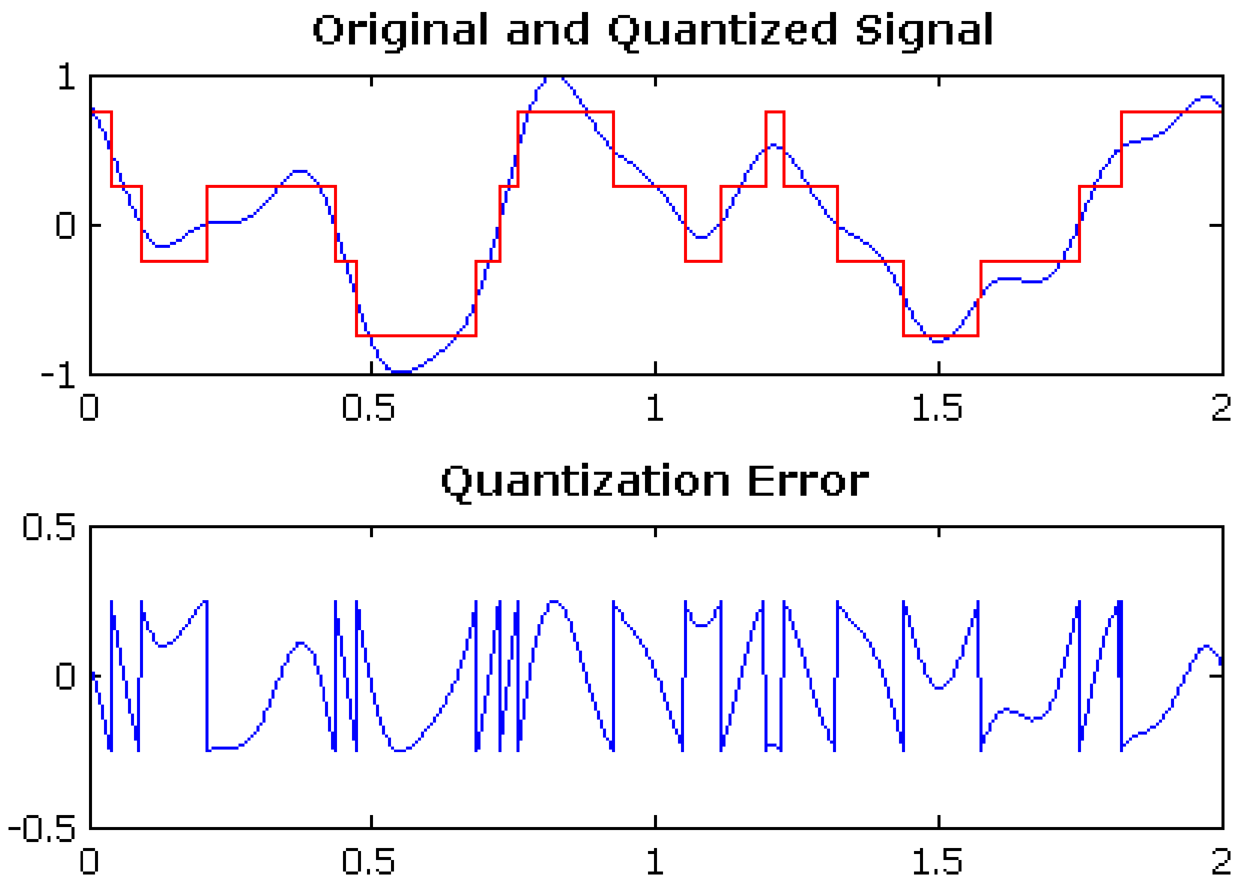
Quantization (cont'd)



Quantization (cont'd)

- Because each sample is represented by a finite number of bits, the quantized value will differ from the actual signal value, thus always introducing an **error**.
- The error decreases as the number of bits used to represent the sample increases.
  - This is an unavoidable and irreversible loss, as the sample would otherwise need to be represented with infinite precision, which requires an infinite number of bits.
- The question, then, is how many bits should be used to represent each sample? Is this number the same for all signals?
- This actually depends on the type of signal and what its intended use is.
  - Audio signals, which represent music, must be quantized on 16 bits
  - Speech only requires 8 bits.

Quantization (cont'd)



Types of quantization schemes

- The discussion so far depicts **uniform quantization** intervals in which the output range of the signal is divided into fixed and uniformly separated intervals depending on the number of bits used.
  - This works well when all the values in the range of the signal are equally likely and, thus, the quantization error is equally distributed.
- **Nonuniform quantization schemes** ➔ However, for some signals where the distribution of all output values is nonuniform, it is more correct to distribute the quantization intervals nonuniformly.
  - For instance, the output intensity values of many audio signals such as human speech are more likely to be concentrated at lower intensity levels, rather than at higher intensity levels in the dynamic audio range.
  - Because the distribution of output values in such signals is not uniform over the entire dynamic range, quantization errors should also be distributed nonuniformly.

Types of quantization schemes (cont'd)

- An illustration of this is shown in Figure 2-5, where the original signal on the left is shown digitized using eight uniform quantization intervals (center) and eight logarithmic quantization intervals (right).
  - The digitized signal to the right preserves the original signal characteristics better than the digitized signal in the center.

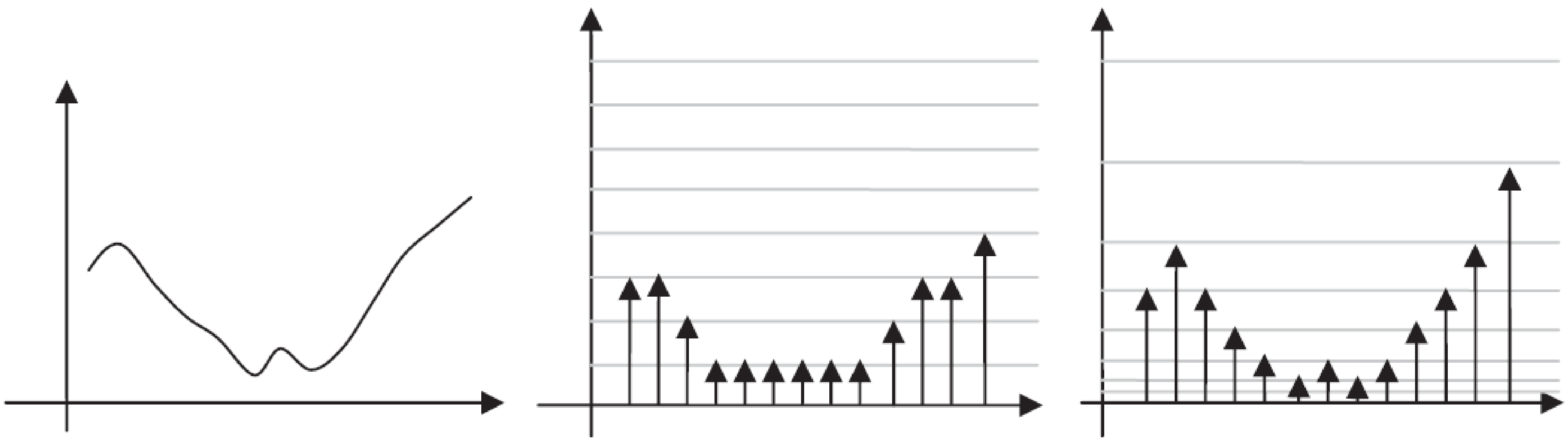


Figure 2-5 Nonlinear quantization scales. The left signal shows the original analog signal. The corresponding digitized signal using linear quantization is shown in the center. The right signal is obtained by a logarithmically quantized interval scale.

Bit Rate

- ***Bit rate*** describes the number of bits being produced per second.
- Bit rate is of critical importance when it comes to storing a digital signal, or transmitting it across networks, which might have high, low, or even varying bandwidths.
- Bit rate, which is measured in terms of bits per second, consists of the following:

$$\mathit{Bit\ rate} = \frac{\mathit{Bits}}{\mathit{Second}} = \left(\frac{\mathit{Samples\ produced}}{\mathit{Second}}\right) \times \left(\frac{\mathit{Bits}}{\mathit{Sample}}\right)$$
$$= \mathit{Sample\ rate} \times \mathit{Quantization\ bits\ per\ sample}$$

- Ideally, the bit rate should be just right to capture or convey the necessary information with minimal perceptual distortion, while also minimizing storage requirements.

Bit Rate (cont'd)

- Typical bit rates produced for a few widely used signals are shown in following table.

Signal	Sampling rate	Quantization	Bit rate
Speech	8 KHz	8 bits per sample	64 Kbps
Audio CD	44.1 KHz	16 bits per sample	706 Kbps (mono) 1.4 Mbps (stereo)
Teleconferencing	16 KHz	16 bits per sample	256 Kbps
AM Radio	11 KHz	8 bits per sample	88 Kbps
FM Radio	22 KHz	16 bits per sample	352 Kbps (mono) 704 Kbps (stereo)
NTSC TV image frame	Width – 486 Height – 720	16 bits per sample	5.6 Mbits per frame
HDTV (1080i)	Width – 1920 Height – 1080	12 bits per pixel on average	24.88 Mbits per frame

Figure 2-6 Table giving the sampling rate, quantization factor, and bit rates produced for typical signals

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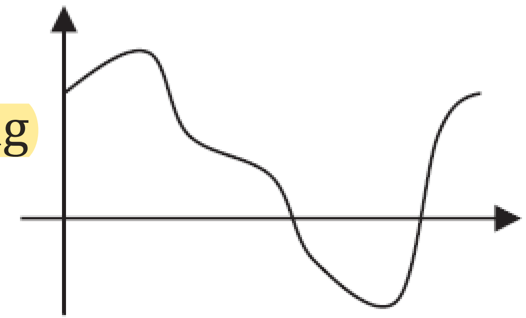
Sampling theorem and aliasing



- The rate at which sampling should occur.
  - The value of a nonstatic signal keeps changing depending on its frequency content.
- Theoretically,
  - if a signal is sampled at more than twice its highest frequency component, then it can be reconstructed exactly from its samples.
  - But, if it is sampled at less than that frequency (called undersampling), then aliasing will result.
    - This causes frequencies to appear in the sampled signal that were not in the original signal.
    - Aliasing is the term used to describe loss of information during digitization.

Sampling theorem and aliasing (cont'd)

- How frequently we must sample in time to be able to recover the original sound.

a) Figure 6.11(a) shows a single sinusoid: it is a single, pure, frequency

b) If sampling rate just equals the actual frequency, Figure 6.11(b) shows that a false signal is detected: it is simply a constant, with zero frequency.

c) Now if sample at 1.5 times the actual frequency, Figure 6.11(c) shows that we obtain an incorrect **(alias)** frequency that is lower than the correct one

d) Thus for correct sampling we must use a sampling rate equal to at least *twice the maximum frequency* content in the signal. This rate is called the **Nyquist rate**.

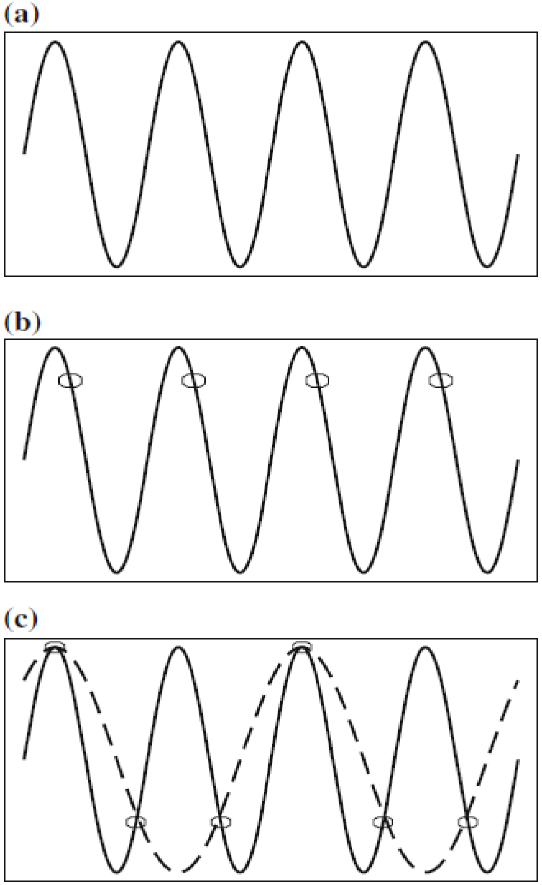


Figure 6.11: **(a)** a single frequency; **(b)** sampling at exactly the frequency produces a constant; **(c)** sampling at 1.5 times per cycle produces an *alias* frequency that is perceived

Example 1

Example 1: In a digital telephone system, the speech signal is sampled 8 kHz. What is the sampling period?

Solution:

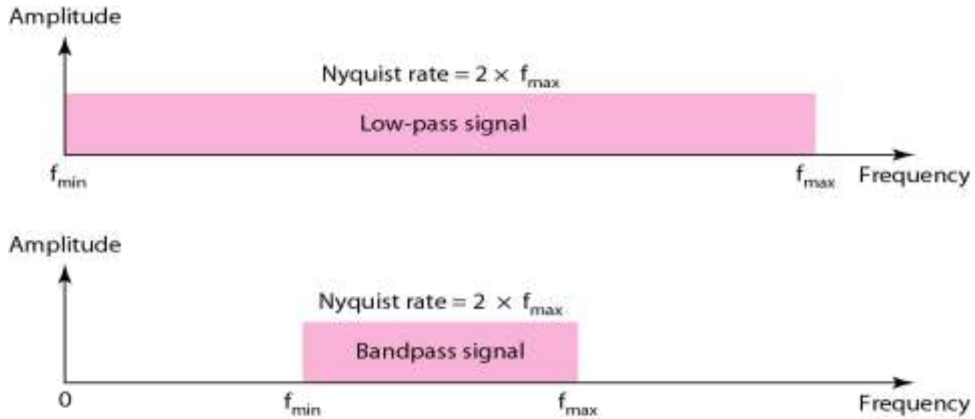
$$T_s = 1/f_s = 1/8000 = 0.000125\ s$$

Example 2: What would be the **minimum sampling rate** needed to accurately capture the human voice signal? (Highest voice component 3000 Hz)

Solution:

Minimum sampling rate: $f_s = 2 \times 3000\ Hz = 6000\ Hz$

Example 2



Example 3: A complex ***low-pass*** signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

Solution:

The bandwidth of a low-pass signal is between **0** and ***f***, **where *f* is the maximum frequency** in the signal. Therefore, we can sample this signal at 2 times the highest frequency

➔ The sampling rate is therefore 400,000 samples per second.

Example 4: If a signal is **band-limited** with a lower limit "5 KHZ" and an upper limit "12 KHZ" of frequency components in the signal, the sampling rate should be at least ....

Solution:

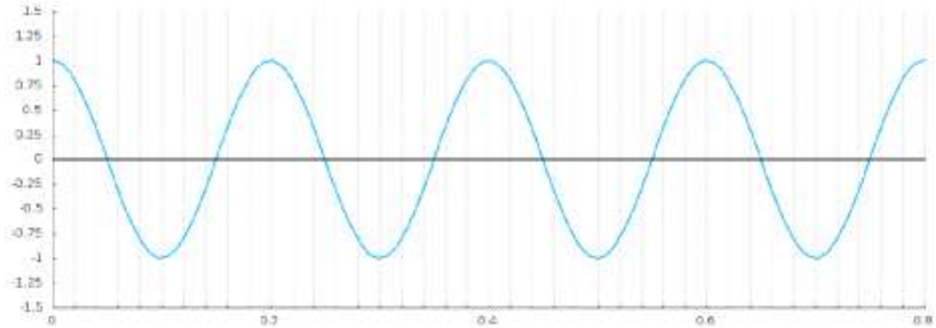
$$f_{max} = 12 - 5 = 7\ kHz$$

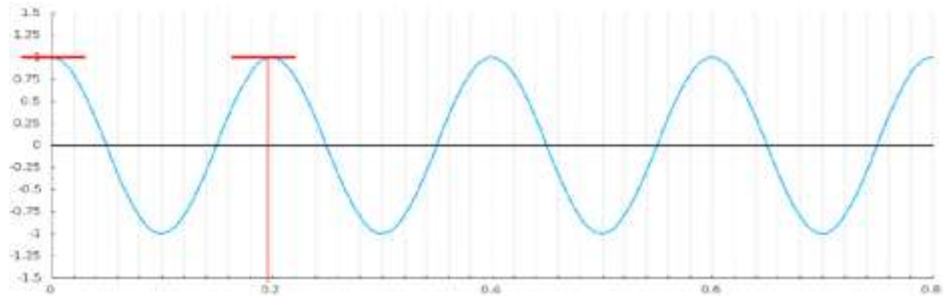
Minimum sampling rate: $f_s = 2 \times 7000\ Hz = 14000\ Hz$

Example 3

Example 6: Consider a pure sine wave, find the optimum sampling rate

Solution:





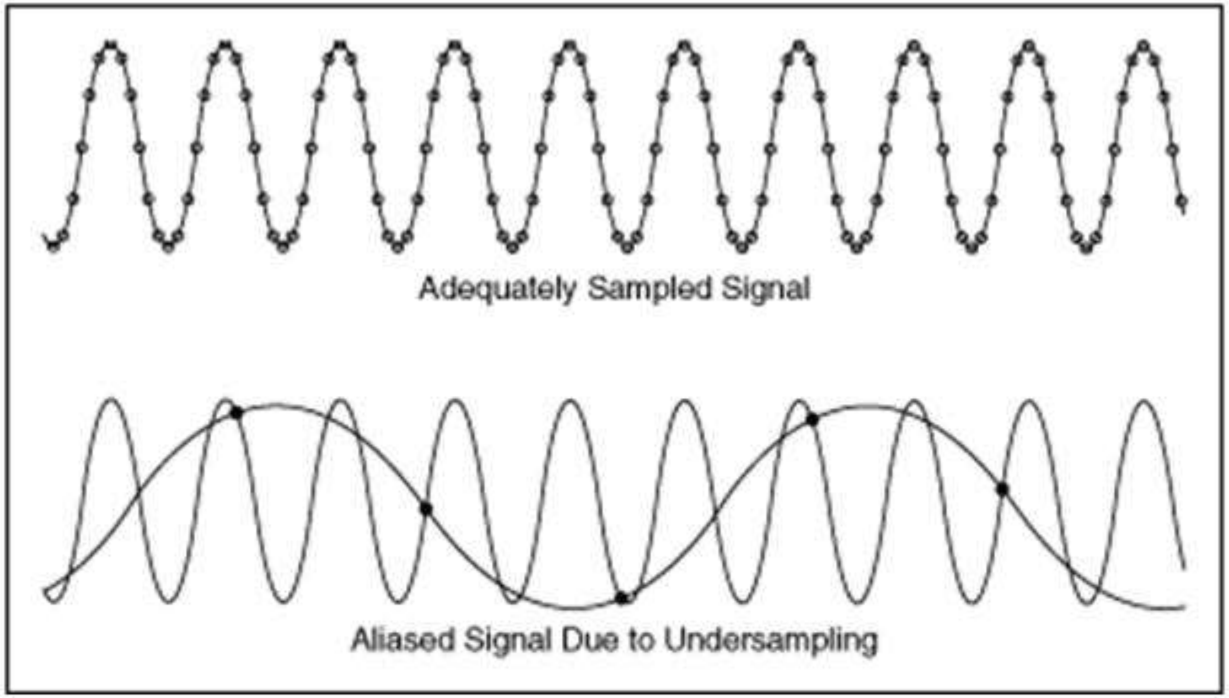
$$f_{max} = 1/T = 1/0.2 = 5\ Hz$$

Sampling rate: $f_s = 2 \times f_{max} = 10\ samples/second$

$$T_s = 1/f_s = 1/10 = 0.1 \text{ sec}$$

Aliasing in Spatial Domains

- Aliasing effects in the spatial domain are seen in all dimensions.
- In the one-dimensional case, the original sinusoid was sampled at a lesser frequency than the Nyquist rate.
  - When the samples are interpolated to reproduce the original, you can see that the two signals do not match, except at the exact sample points.



Aliasing in Spatial Domains (cont'd)

- In the two-dimensional example, the first image shows the original image consists of 750 samples in the x and 620 samples in the y direction.
- The succeeding figures show what the reconstructed signal looks like by reducing the sampling resolution in both directions.
- You can see that, as the number of spatial samples decreases, not all of the original frequencies are properly captured.
- Also, the versions with fewer samples display **increased effects of blur**.

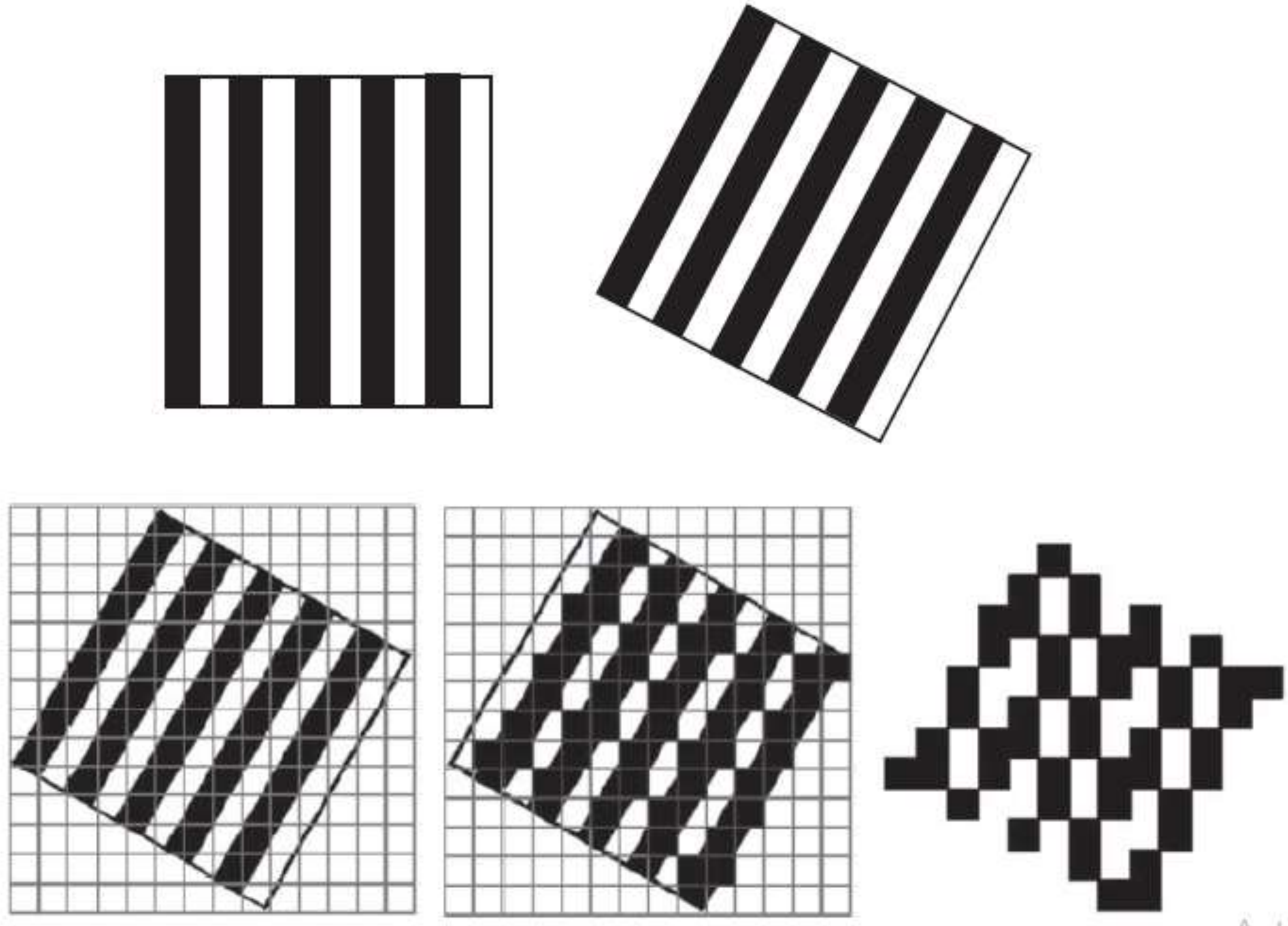
Aliasing in Spatial Domains (cont'd)

The top left shows the original signal and the remaining three show examples of the signal reconstruction at different sampling resolutions. In all cases, the output does not match the input because the sampling resolution falls below the Nyquist requirement.

Moiré Patterns and Aliasing

- Another interesting example of aliasing, called the *moiré effect*, can occur when there is a pattern in the image being photographed, and the sampling rate for the digital image is not high enough to capture the frequency of the pattern.
- If the pattern is not sampled at a rate that is at least twice the rate of repetition of the pattern, a different pattern will result in the reconstructed image.

Moiré Patterns and Aliasing



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Filtering

- In signal processing, the function of a **filter** is to
  - remove unwanted parts of the signal, such as random noise and undesired frequencies, and to extract useful parts of the signal, such as the components lying within a certain frequency range.
- There are two main kinds of filters: *analog* and *digital*.
  - An **analog filter** uses analog electronic circuits made up from components such as resistors, capacitors, and operational amplifiers (op-amps) to produce the required filtering effect.
    - Such filter circuits are widely used in applications such as noise reduction, video signal enhancement, and many other areas.
  - A **digital filter**, on the other hand, uses digital numerical computations on sampled, quantized values of the signal.
    - The processing might be done on a general-purpose computer such as a PC, or a specialized digital signal processor (DSP) chip.

Filtering (cont'd)

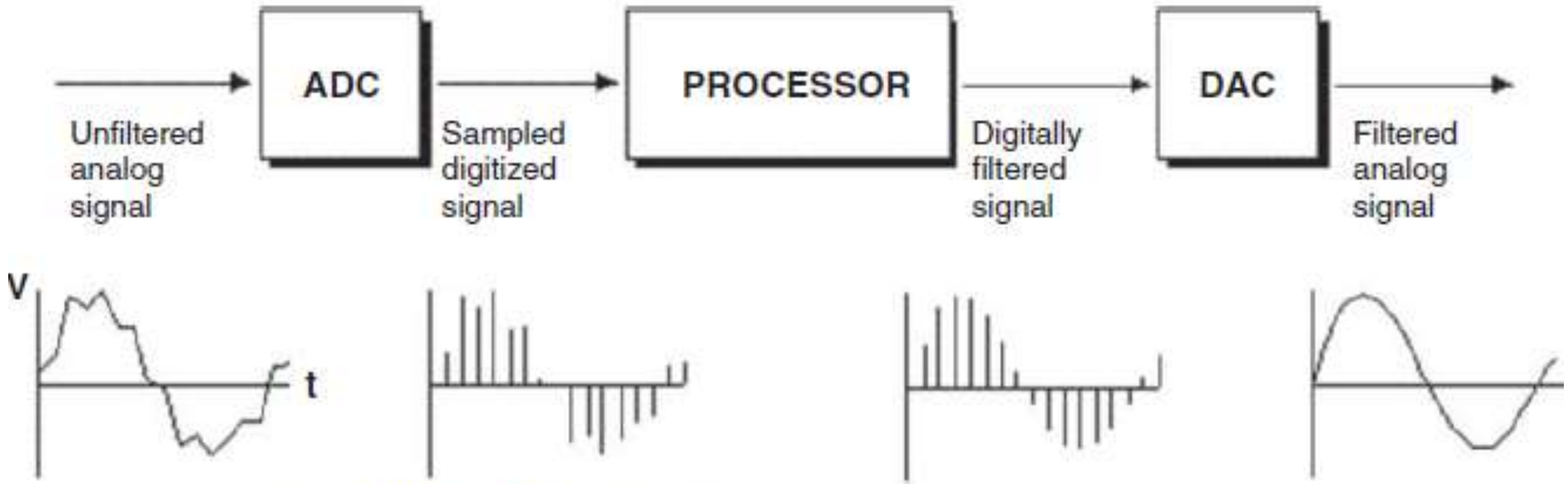


Figure 2-15 Digital filters—A digital filter takes a digital signal as input and produces another signal with certain characteristics removed. In the bottom row, the noisy analog signal is digitized and filtered.

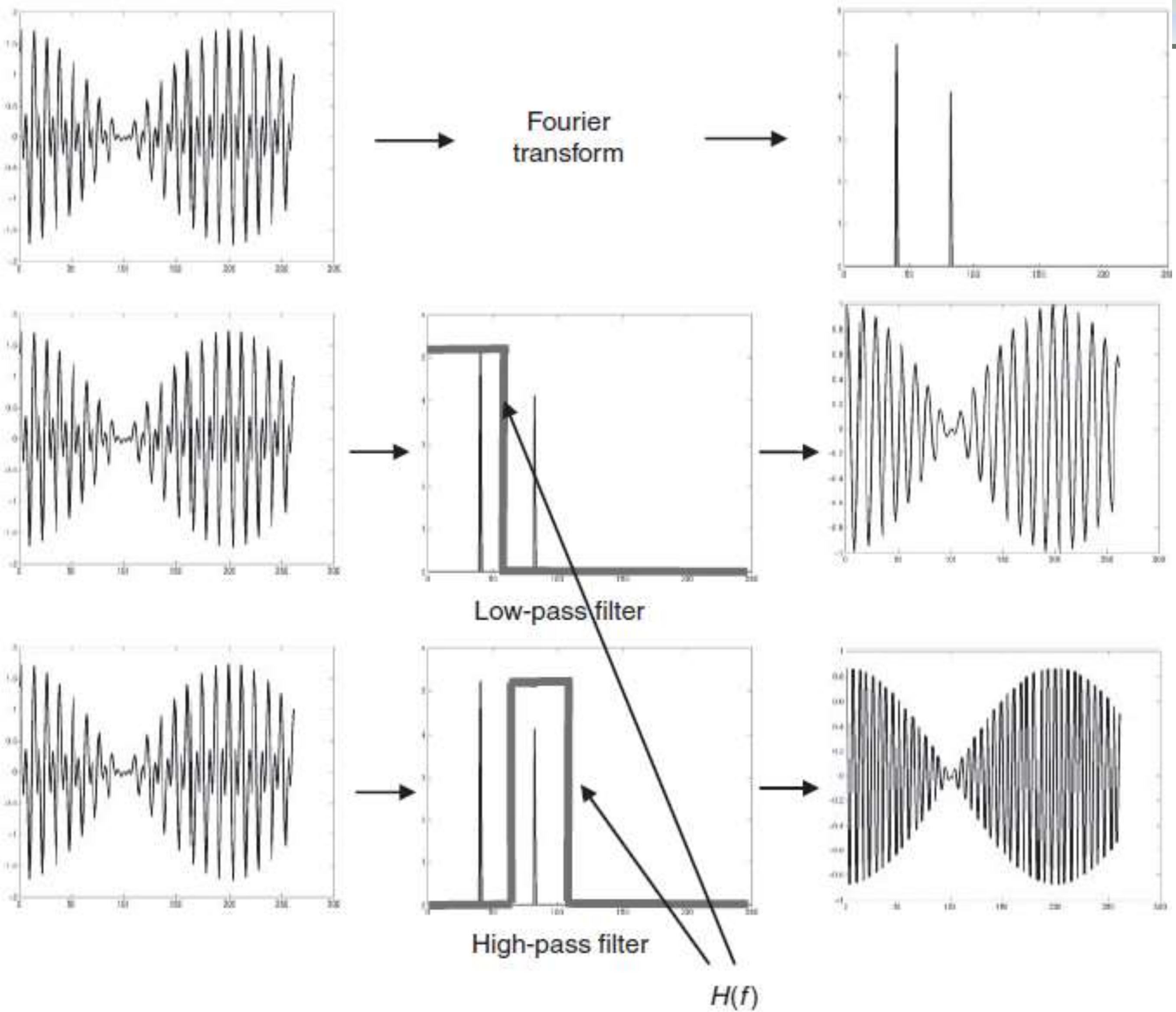
Filtering (cont'd)

- There are 2 type of frequency component in the signal:
  1. **High frequency component** are characterized by large changes in values over small distance.
     - Example: noise and edge
  2. **Low frequency component** are characterized by little changes in values over small distance.
     - Example: backgrounds, skin texture
- Both digital and analog filters can be classified into three categories: low-pass filters, band-pass filters, and high-pass filters.
  - **Low-pass filters** remove high frequency content from the input signal. Such filters are used to avoid aliasing artifacts while sampling.
  - **High-pass filters**, on the other hand, remove the low-frequency content and are used to enhance edges and sharpen an image.
  - **Band-pass filters** output signals containing the frequencies belonging to a defined band.

Filtering in 1D

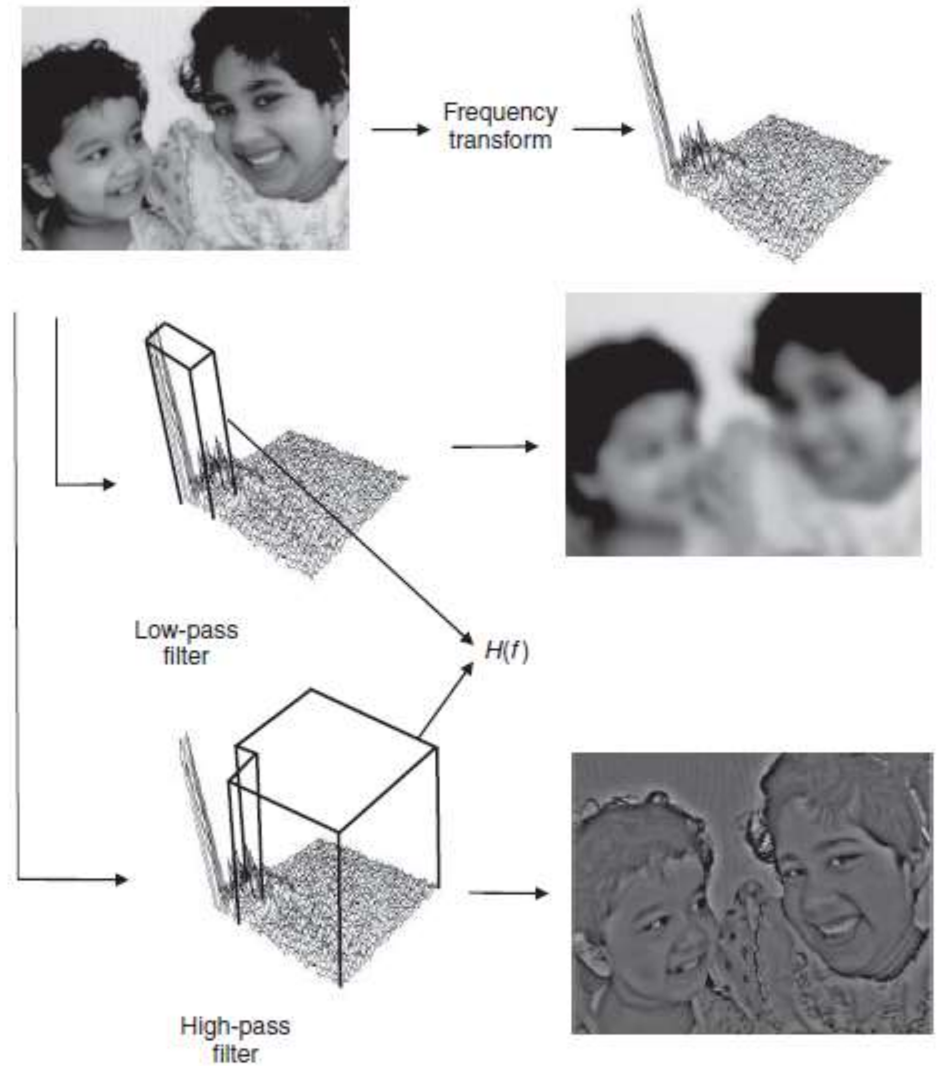
- A 1D signal is normally represented in the time domain with the *x*-axis showing sampled positions and the *y*-axis showing the amplitude values.
- Following figure show a signal, and The Fourier transform of this input signal shows that there are peaks at two frequencies, $X_1(f)$ and $X_2(f)$.
- The figure also shows the effects of low- and high-pass filters.
  - The middle row shows the output generated when passed through a low-pass filter. Here the filter is designed to allow lower frequencies, such as $X_1(f)$ to remain while the higher frequency $X_2(f)$ is removed.
    - In case of low-pass filters, all frequencies below a threshold or cutoff are allowed to remain, whereas frequencies above the cutoff frequency are removed.
  - A high-pass filter, on the other hand, removes all the lower frequencies in the signal, allowing only the higher frequencies above a threshold level to pass through the filter.

Filtering in 1D



Filtering in 2D

- Low-pass filters help remove aliasing in images, whereas high-pass filters are used in certain image processing operations such as image sharpening.
- Low pass filter is a filter that "passes over" the low frequency components and reduce or eliminates high frequency components
  - Used for blurring and noise reduction
- High pass filter is a filter that "passes over" the high frequency component and reduce or eliminates low frequency component.
  - Used for edge enhancement (sharpening)



Thank You