# Multimedia System

## Lecture 6: Media Representation and Media Formats (Part 3)

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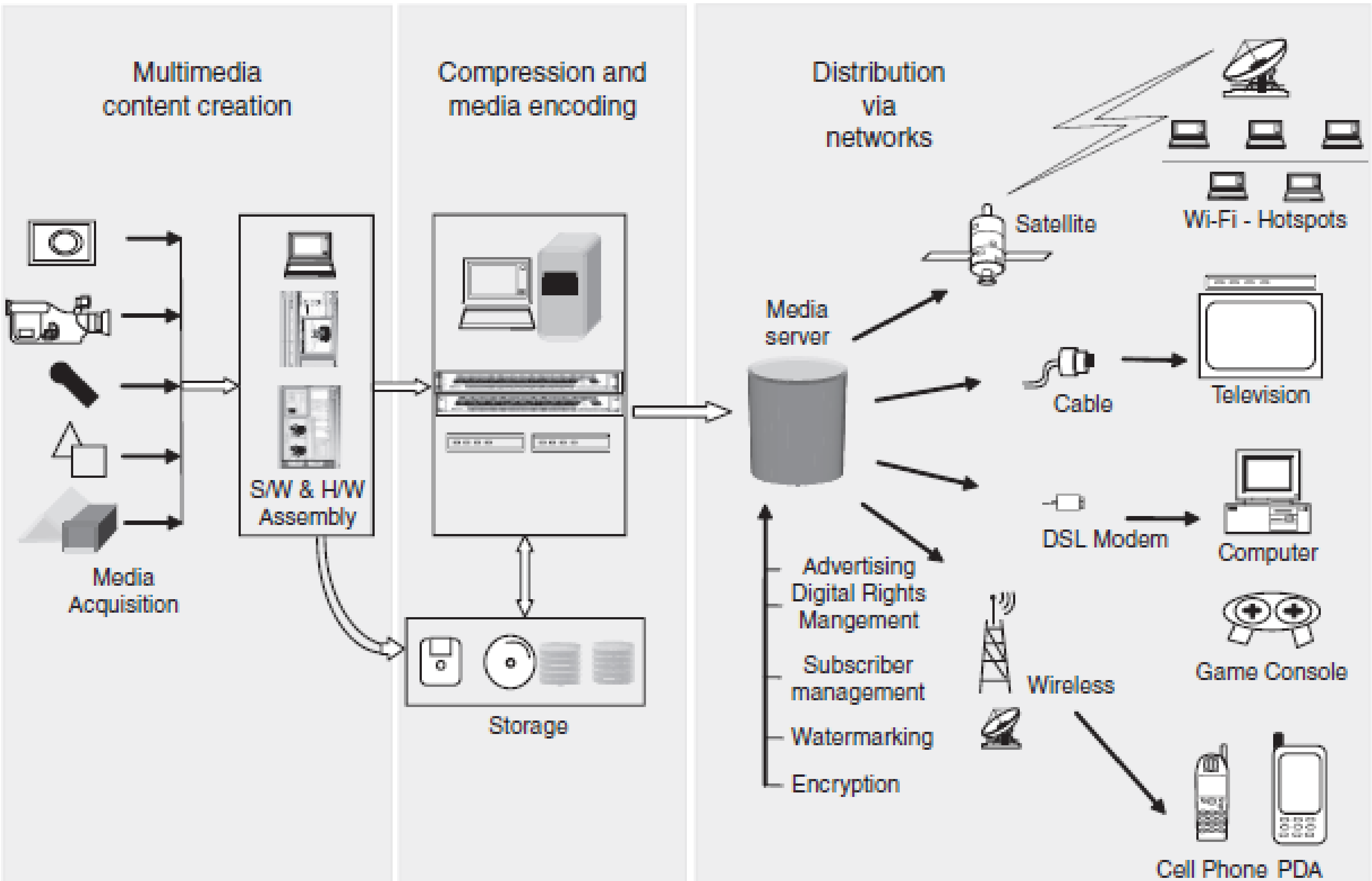
# Overview

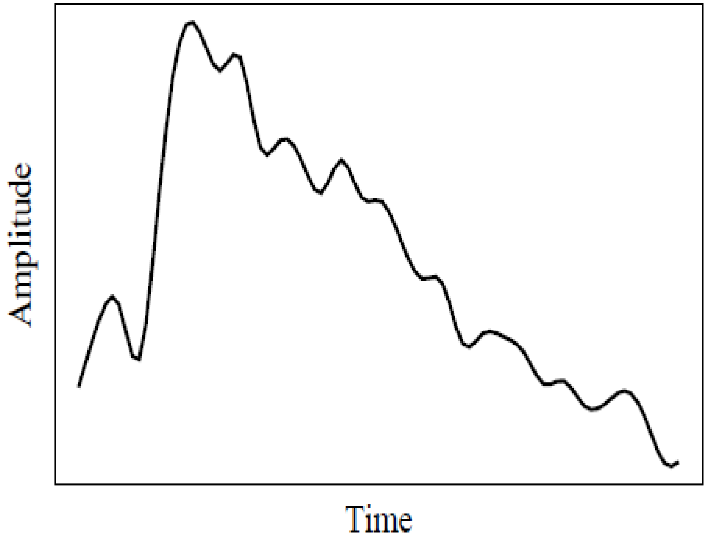
- Multimedia systems involve three major components:
  - multimedia content creation,
  - compression/storage of multimedia content
  - delivery or distribution of multimedia content.
- Multimedia information is
  - digital,
  - interactive
  - voluminous
- As depicted in the end-to-end multimedia system 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

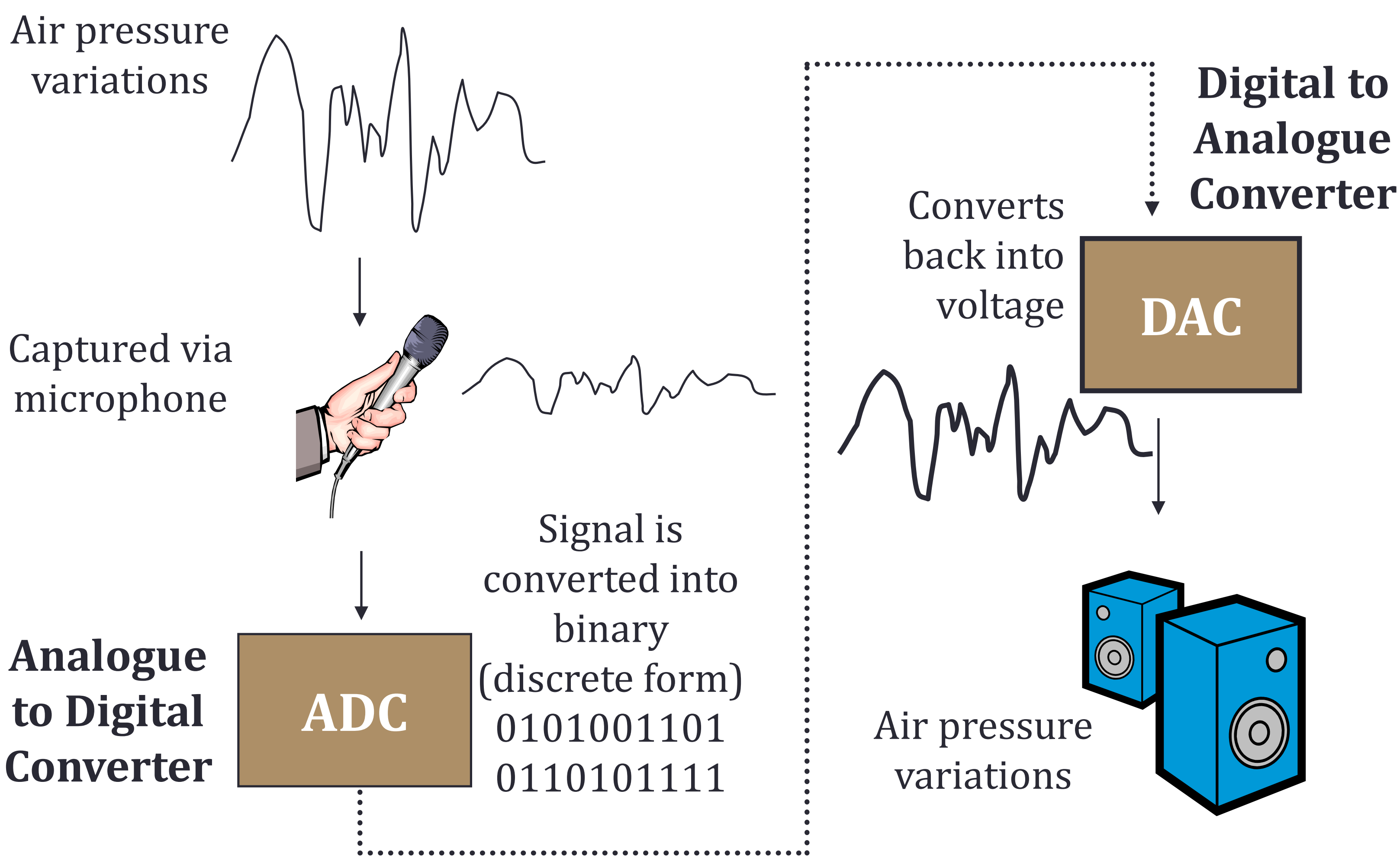

Multimedia content creation
Compression and media encoding
Distribution via networks
Media Acquisition
S/W & H/W Assembly
Storage
Media server
Satellite
Wi-Fi - Hotspots
Cable
Television
DSL Modem
Computer
Wireless
Game Console
Cell Phone
PDA
Advertising
Digital Rights Mangement
Subscriber management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

# Overview

- Physics describes **sound** as a form of energy that is transmitted through a medium as pressure waves making refractions and compressions.
- These pressure changes can be captured by an electromechanical device, such as a microphone, and converted into an analog signal, which can be further stored on an analog recording device such as a magnetic tape.
- If we wish to use a digital version of sound waves we must form digitized representations of audio information.

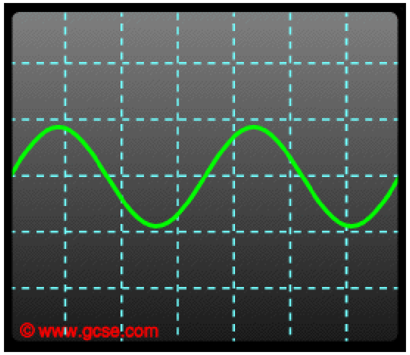


# Overview


Air pressure variations
Captured via microphone
Analogue to Digital Converter
ADC
Signal is converted into binary (discrete form)
0101001101
0110101111
Digital to Analogue Converter
Converts back into voltage
DAC
Air pressure variations

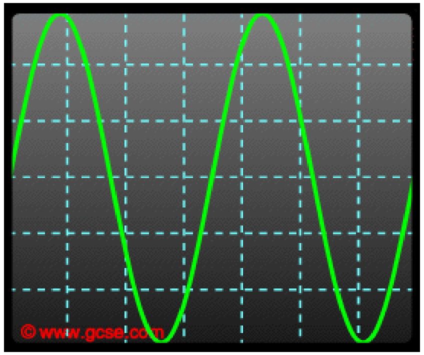
# Overview (cont'd)

- Any **signal** can be characterized by **Amplitude**, **Frequency**, and **Phase (period).**
  - **Amplitude (A)** is the magnitude (displacement) of the signal at a given instant in time (t).
    - High volume implies high amplitude and vice versa
    - The louder a sound, the more energy it has. This means loud sounds have a large amplitude.



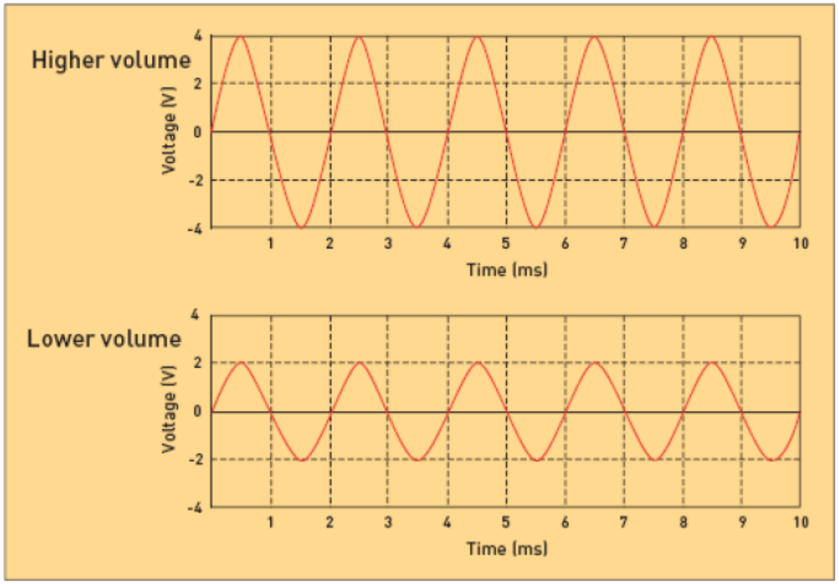

Low amplitude

Quiet




High Amplitude

Loud

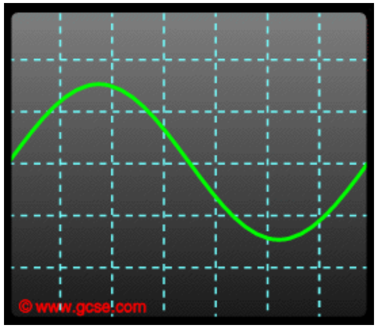


# Overview (cont'd)

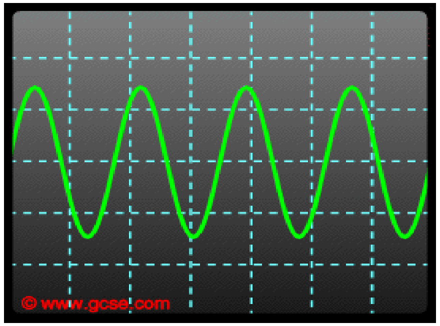
- Any **signal** can be characterized by **Amplitude**, **Frequency**, and **Phase (period).**
  - **Period (T)** is the time a wave requires to complete a single cycle
    - Measured in seconds (s)
  - **Frequency (f)** is the number of cycles a wave completes in one second (s)
    - Measured in hertz (Hz)

$$f = \frac{1}{T}$$

    - corresponds to the ***pitch*** of a sound.
      - The more frequent vibration occurs the higher the pitch of the sound.
    - Optimally, people can hear from **20 Hz to 20,000** Hz (20 kHz)
      - Sounds below 20 Hz are infrasonic
      - sounds above 20 kHz are ultrasonic.



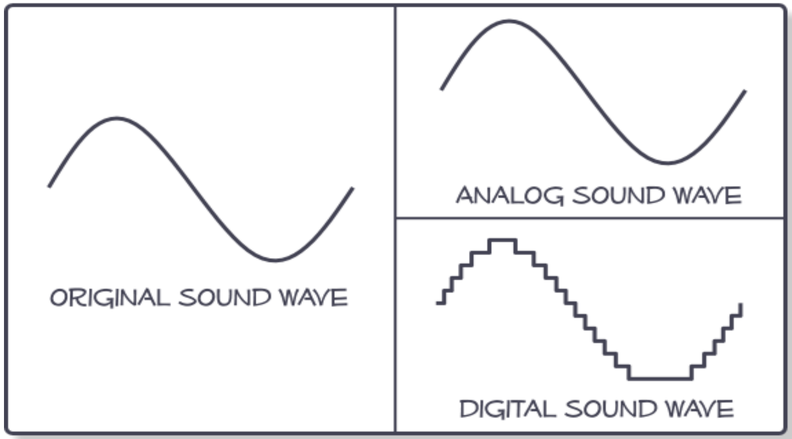

Low pitch




High pitch

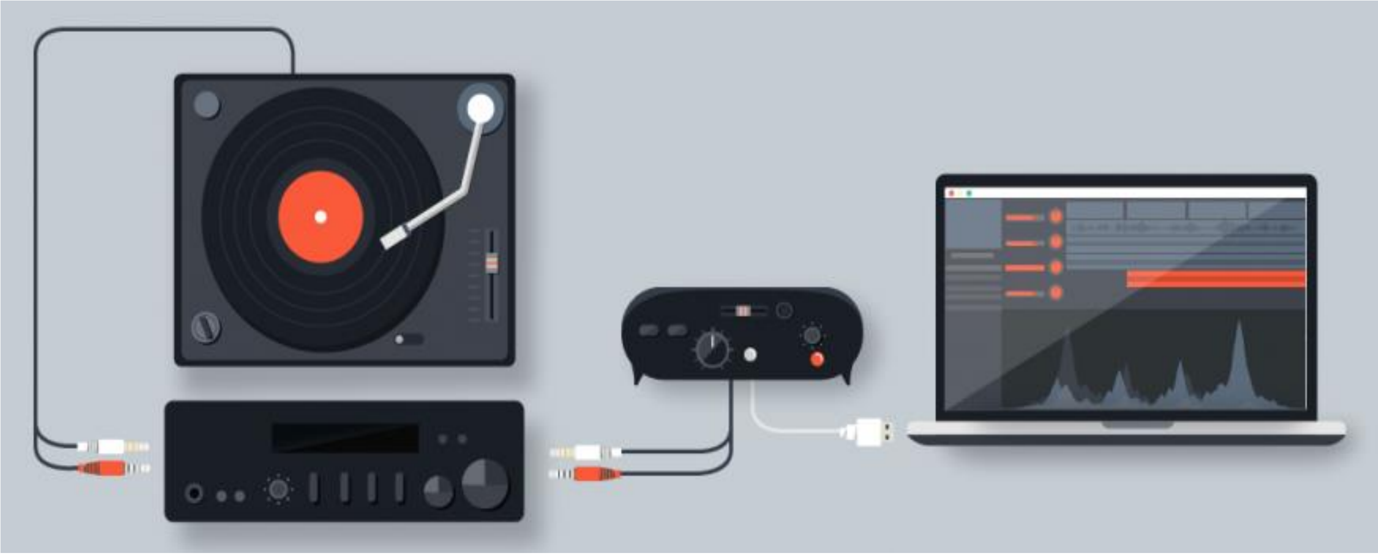
# Overview (cont'd)

- Audio recordings come in two basic types; **analog** and **digital**.
  - **Analog** refers to audio recorded using methods that replicate the original sound waves.
    - Vinyl records and cassette tapes are examples of analog mediums.
  - **Digital** audio is recorded by taking samples of the original sound wave at a specified rate.
    - CDs and Mp3 files are examples of digital mediums.
- As you can see from the following figure,
  - the analog sound wave replicates the original sound wave, whereas
  - the digital sound wave only replicates the sampled sections of the original sound wave.



# Overview

- The potential fidelity of an analog recording depends on:
  - —the sensitivity of the equipment and
  - —medium used to record and playback the recording.
- digital audio fidelity heavily depends on the
  - —rate at which the recording equipment sampled the original sound wave over a specified increment of time

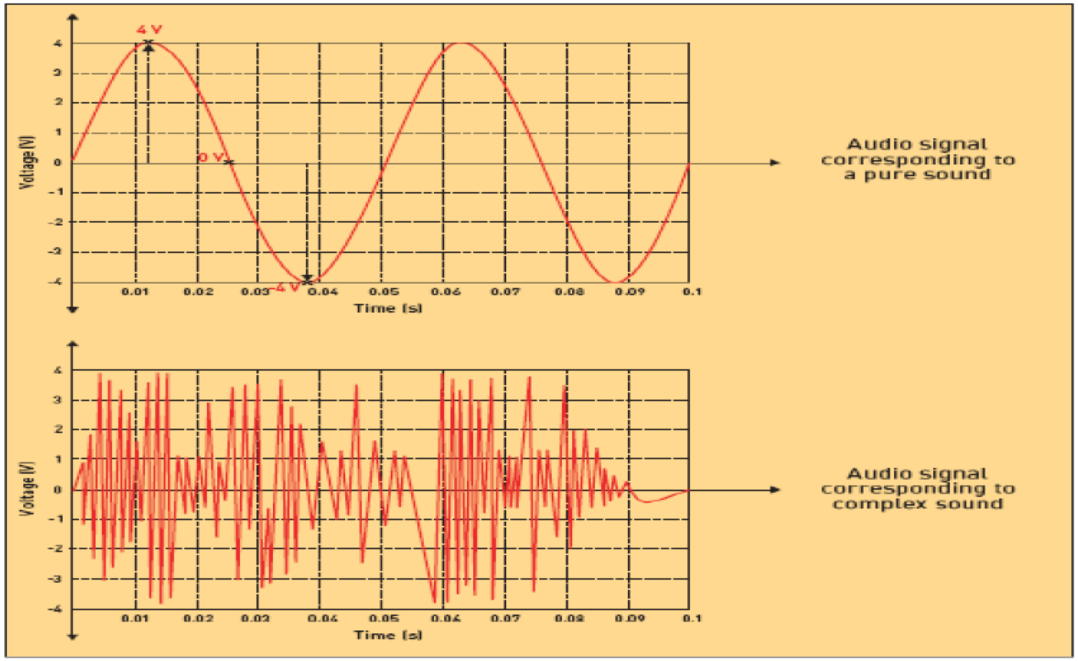


# Outlines

- Digital Audio
  - Digital Representation of audio
  - Signal-to-Noise Ratio (SNR)
  - Signal-to-Quantization-Noise Ratio (SQNR)
  - Filtering
  - Audio file size
  - Commonly Used Audio Formats
- Graphics

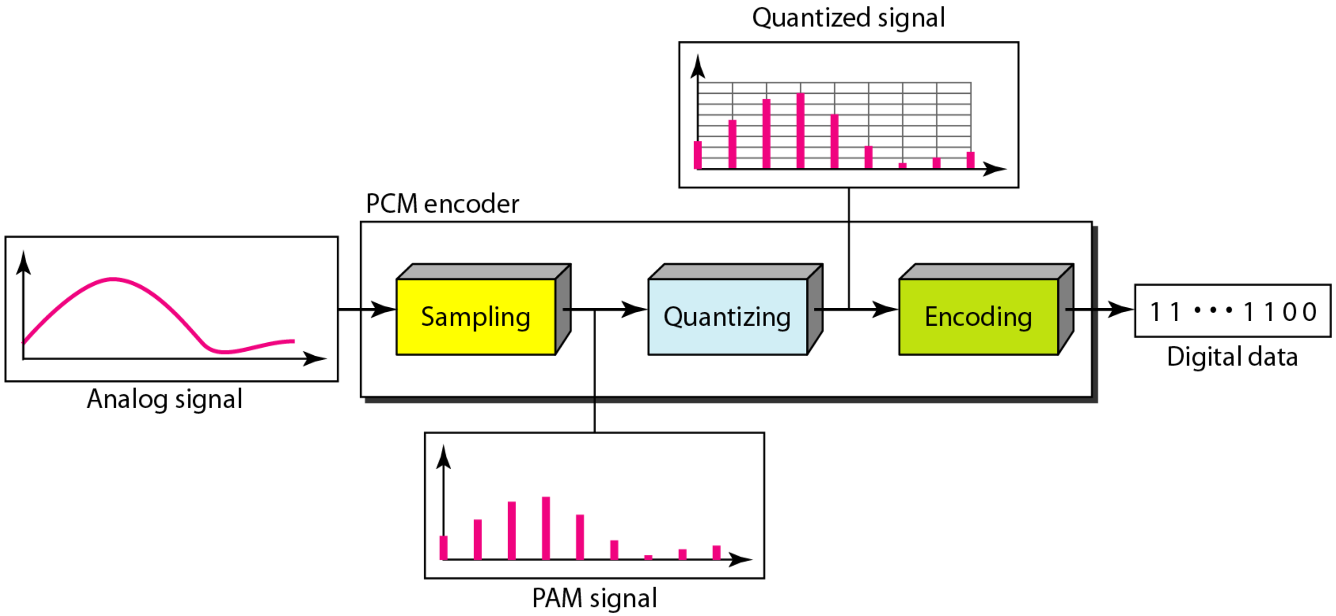
# Representation of digital Audio

- Analog audio signals are typically represented as waveforms, either **simple** or **complex**.
- A simple sinusoidal wave corresponds to a pure tone at a single frequency, or pitch.
- The amplitude of the wave gives the strength of the sinusoid at that time.
- A complex wave, on the other hand, consists of multiple frequencies or sinusoidal waves combined together.



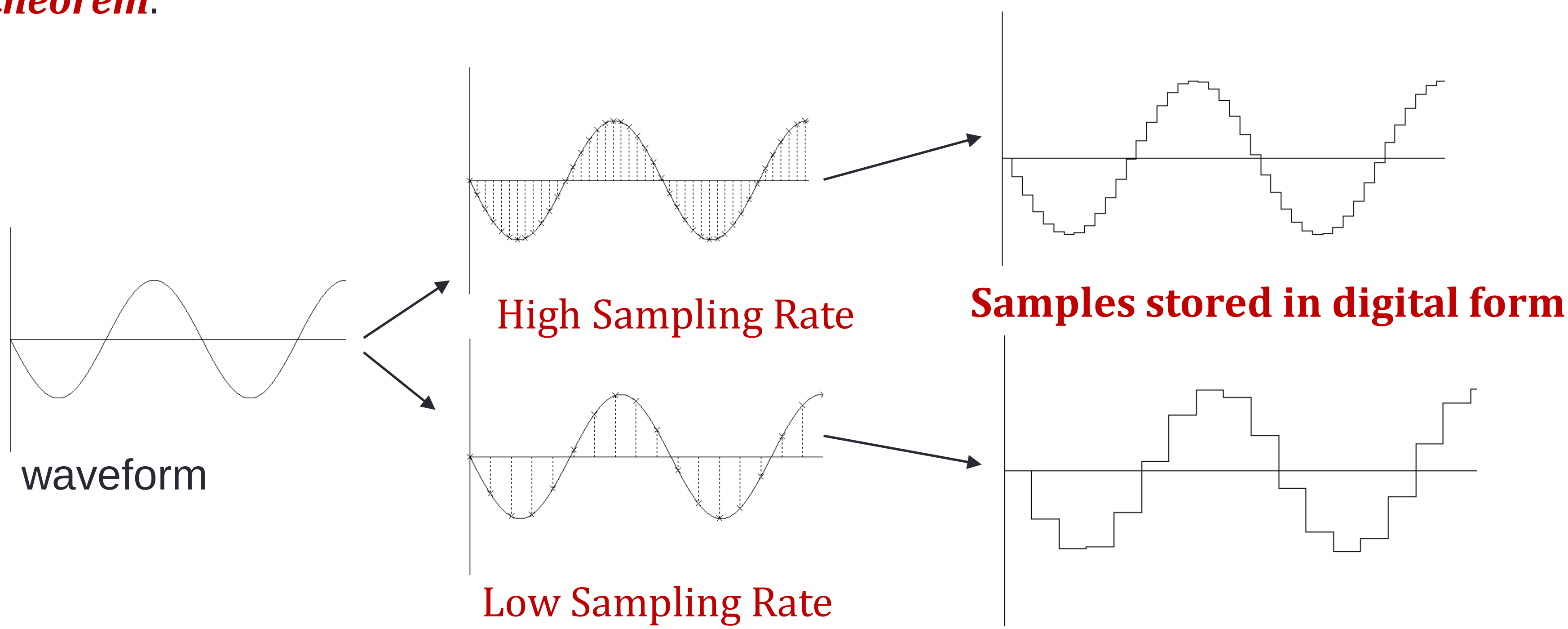
# Representation of digital Audio

- **Digitizing** an analog audio signal requires sampling and quantization.
- The process of conversion to digital sound is known as pulse code modulation (PCM).
- PCM consists of three steps to digitize an analog signal:
  1. Sampling
  2. Quantization
  3. Binary encoding



# Sampling

- The analog sound is sensed at spaced time intervals, producing digital audio samples.
- Quality of digital recording depends on the ***sampling rate***, the number of samples point taken per second (Hz).
  - The **higher the sampling rate**, the **more the measurements** are taken (better quality).
  - The **lower the sampling rate**, the **lesser the measurements** are taken (low quality).
- The minimum sampling rate that is needed is given by the ***Nyquist sampling theorem***.



# Sampling (cont'd)

- 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 at a higher sampling rate (usually twice or more) than necessary to prevent aliasing is called oversampling.

# Sampling (cont'd)

- Undersampling:
  - — There are not enough samples to capture all the peaks and troughs in the signal
  - — We are getting the wrong information about the signal.
  - — results in **aliasing** (signal now takes on a different or a false presentation)
- Oversampling:
  - — There are more than enough samples to capture the variations in the signal.
  - — This is called **oversampling** (more values to digitize and process)
  - — As sampling rate increases, sampled signal looks more and more like the original.
  - — More samples means more storage required.
- 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.**

$$Nyquist\ rate = sampling\ rate\ (f_s) = 2f_{max}$$

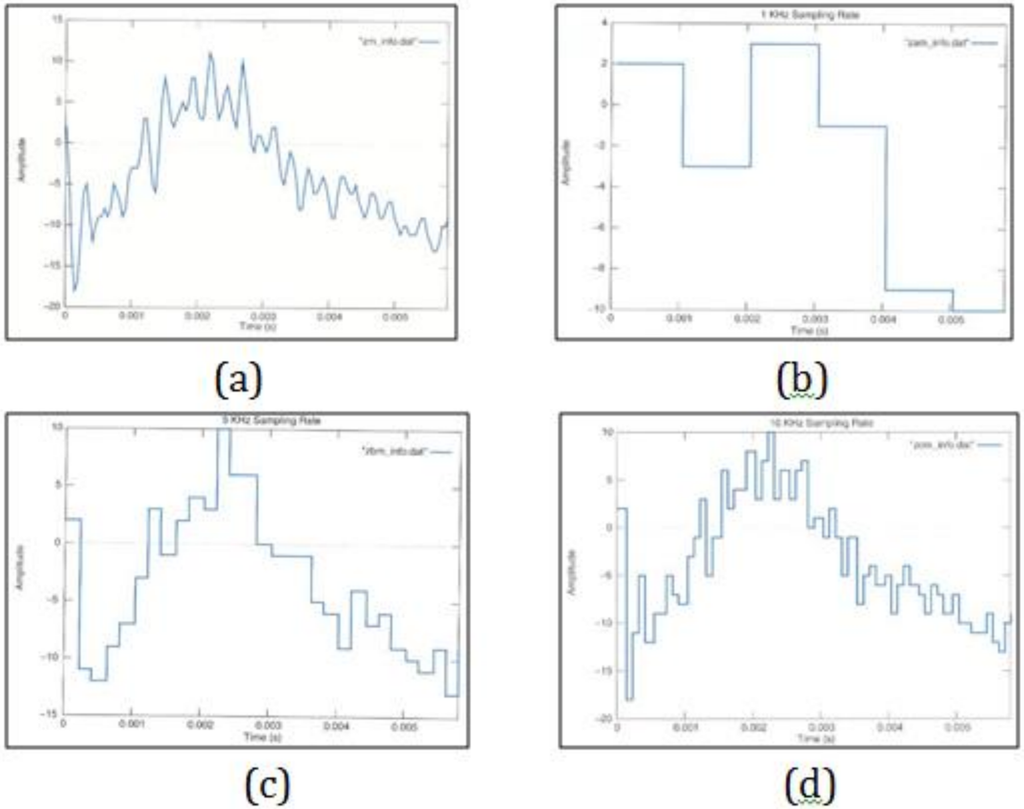
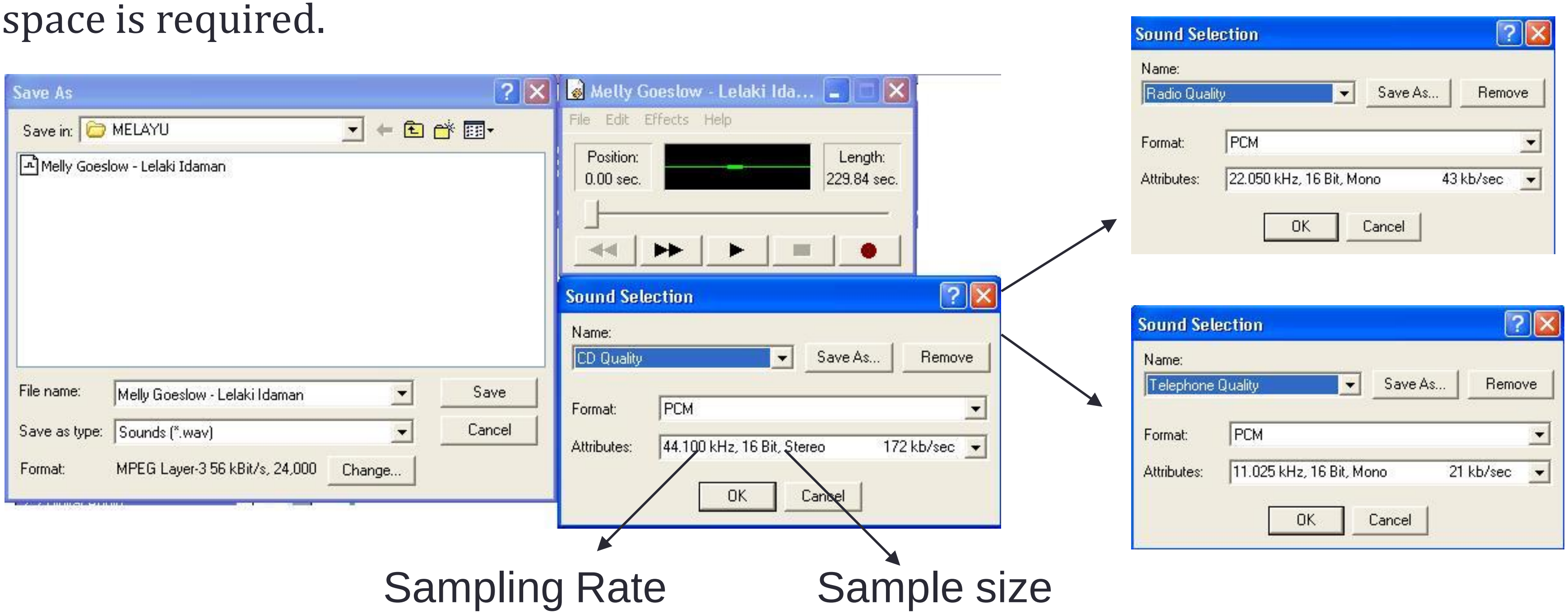



Figure: (a) A short section of a signal (highest frequency component is 3 KHZ) (b) Reconstructed signal with 1 KHz sampling rate (c) Reconstructed signal with 5 KHz sampling rate (d) Reconstructed signal with 10 KHz sampling rate

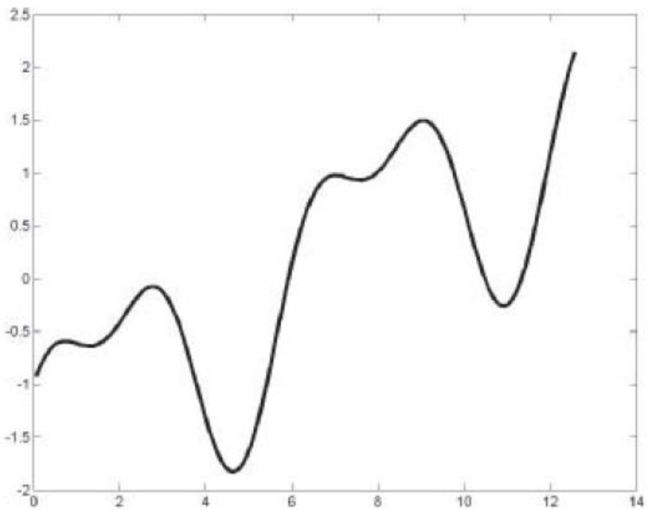
# Quantization

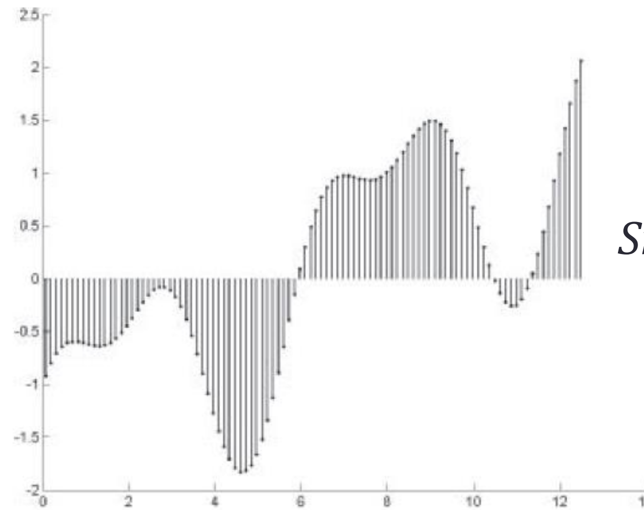
- Given a sample, the signal amplitude at that position is encoded on a fixed number of bits.
  - All samples are represented by the same number of quantization bits.
- The ***quantization*** of a sample value is dependent on the number of bits used to represent the amplitude.
- The greater the number of bits used, the better the resolution, but the more storage space is required.



# Quantization (cont'd)

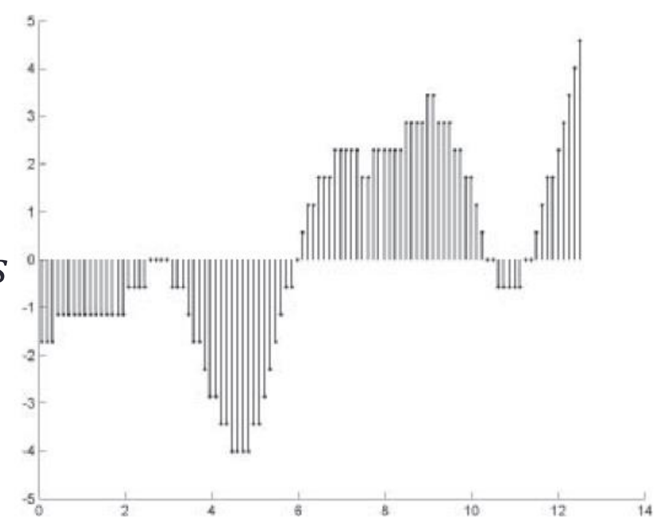
*Original analog signal*

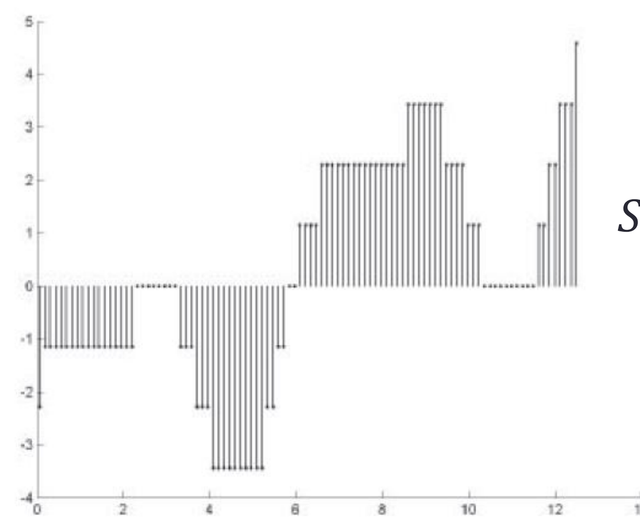




*Quantized Signal at 8 bits (256 levels)*

*Quantized Signal at 4 bits (16 levels)*

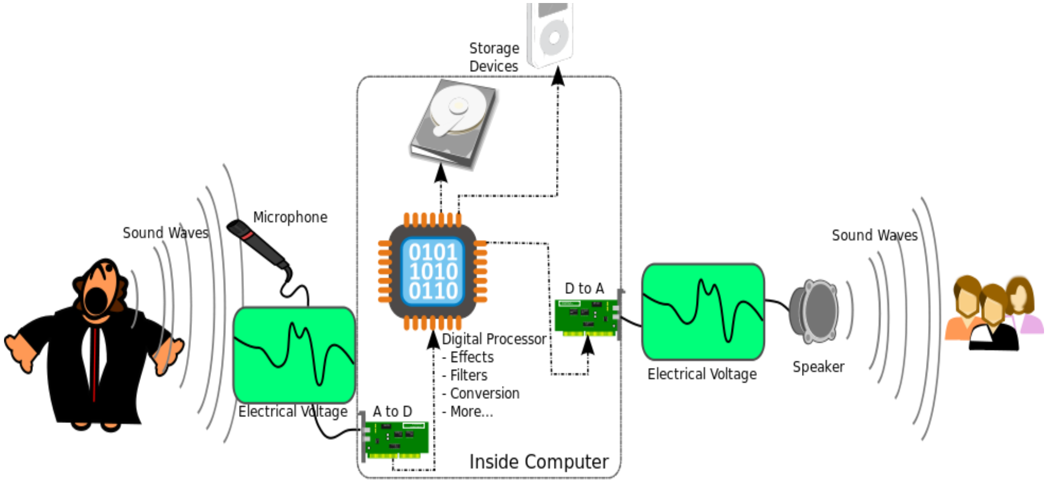




*Quantized Signal at 3 bits (8 levels)*

# Representation of digital Audio

- The sampling rate and the quantization bits per sample are the main properties of the PCM signal and need to be carefully chosen so that it is possible to reconstruct the analog equivalent.
- The digital audio signal is finally rendered by converting it to the analog domain and so the choice of sampling rate and sample size need to be chosen appropriately in order to faithfully re-create the original sound.
- In addition to **sampling rate** and **quantization**, another characteristic commonly used to describe audio signals is the **number of channels**, which may be one (mono), two (stereo), or multichannel (surround sound).



# Outlines

- Digital Audio
  - Digital Representation of audio
  - Signal-to-Noise Ratio (SNR)
  - Signal-to-Quantization-Noise Ratio (SQNR)
  - Filtering
  - Audio file size
  - Commonly Used Audio Formats
- Graphics

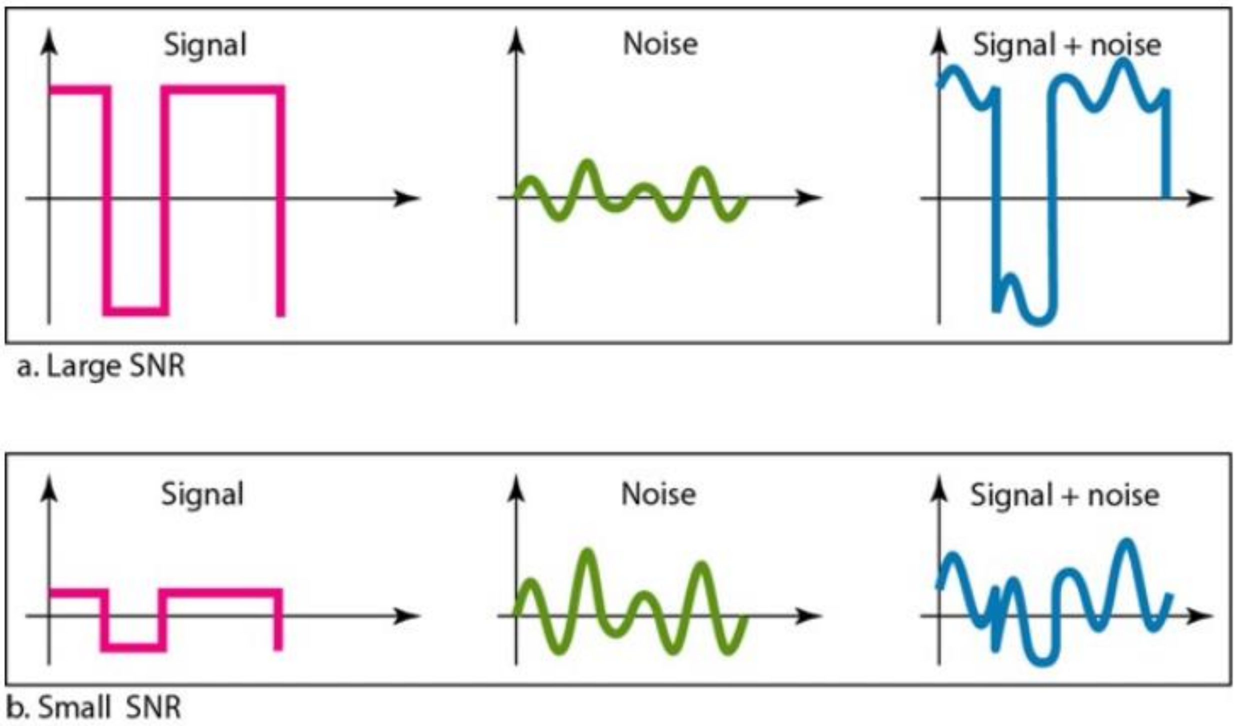
# Signal-to-Noise Ratio (SNR)

- In any analog system, random fluctuations produce *noise* added to the signal (figure 6.17), and the measured voltage is thus incorrect.
- The ratio of the power of the correct signal to the noise is called the *signal-to-noise ratio* (*SNR*). Therefore, the SNR is a measure of the quality of the signal.
- **Signal-to-noise ratio** is defined as the ratio of the power of a signal (meaningful information) to the power of background noise (unwanted signal):
- The SNR is usually measured in *decibels* (*dB*), where 1 dB is a tenth of a *bel*. The SNR value, in units of dB, is defined as follow:

$$SNR = \frac{P_{signal}}{P_{noise}} = 10\ log_{10}\frac{V_{signal}^2}{V_{noise}^2} = 20\ log_{10}\frac{V_{signal}}{V_{noise}}$$

- The power in a signal is proportional to the square of the voltage.

# Signal-to-Noise Ratio (SNR)



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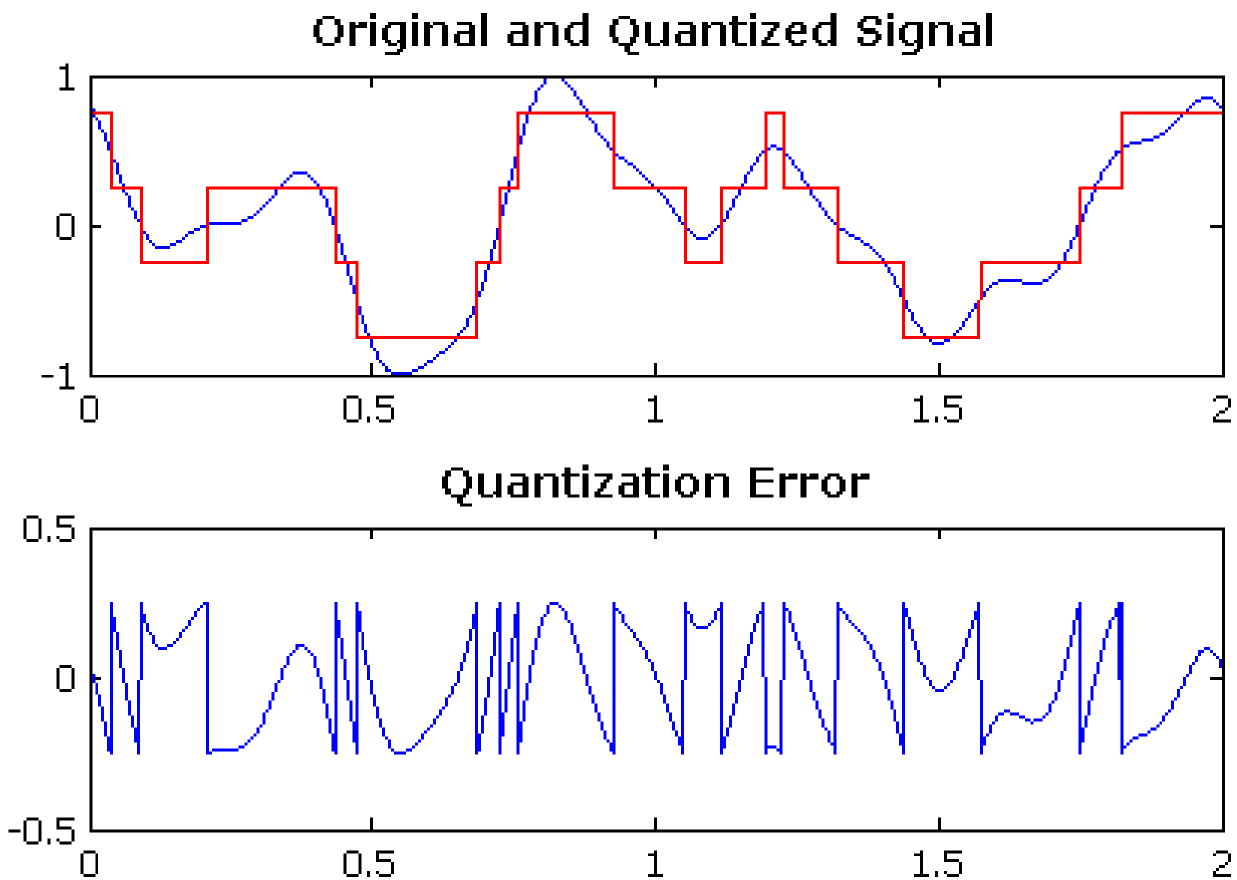
# Signal-to-Quantization-Noise Ratio (SQNR)

- For digital signals, we must take into account the fact that only quantized values are stored.
- For a digital audio signal, the precision of each sample is determined by the number of bits per sample, typically 8 or 16.
- Aside from any noise that may have been present in the original analog signal, there is also an additional error that results from quantization.
  a) If voltages are actually in 0 to 1 but we have only 8 bits in which to store values, then effectively we force all continuous values of voltage into only 256 different values.
  b) This introduces a roundoff error. It is not really "noise". Nevertheless it is called **quantization noise** (or quantization error).

# Signal-to-Quantization-Noise Ratio (SQNR)



# Signal-to-Quantization-Noise Ratio (SQNR)

- The quality of the quantization is characterized by the Signal to Quantization Noise Ratio (**SQNR**).
  a) Quantization noise: the difference between the actual value of the analog signal, for the particular sampling time, and the nearest quantization interval value.
  b) At most, this error can be as much as half of the interval.
  c) For a quantization accuracy of $N$ bits per sample, the SQNR can be simply expressed:

$$SQNR = 20\, log_{10} \frac{V_{signal}}{V_{quan_{noise}}}$$

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# Filtering

- Before we sample, we have to filter the signal to limit the maximum frequency of the signal as it affects the sampling rate.
- Filtering should ensure that we do not distort the signal, i.e. remove high frequency components that affect the signal shape.
- 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

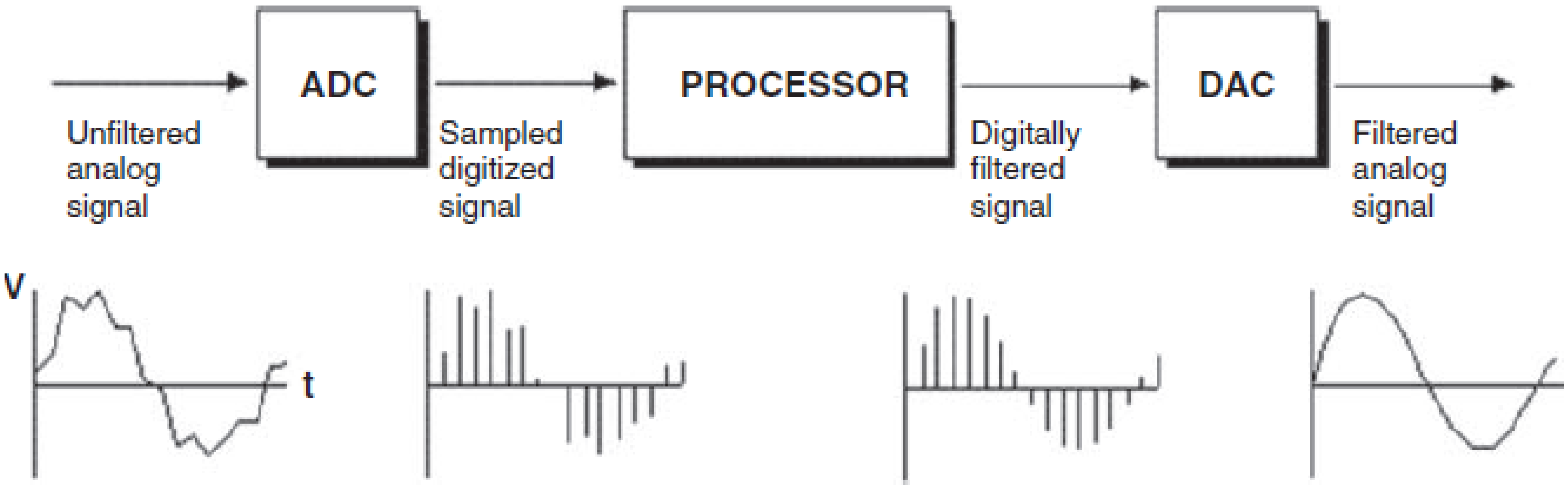



*Figure 2-15 Digital filters—A digital filter takes a digital signal as input and produces another signal with certain characteristics removed. In the bottom row, the noisy analog signal is digitized and filtered.*

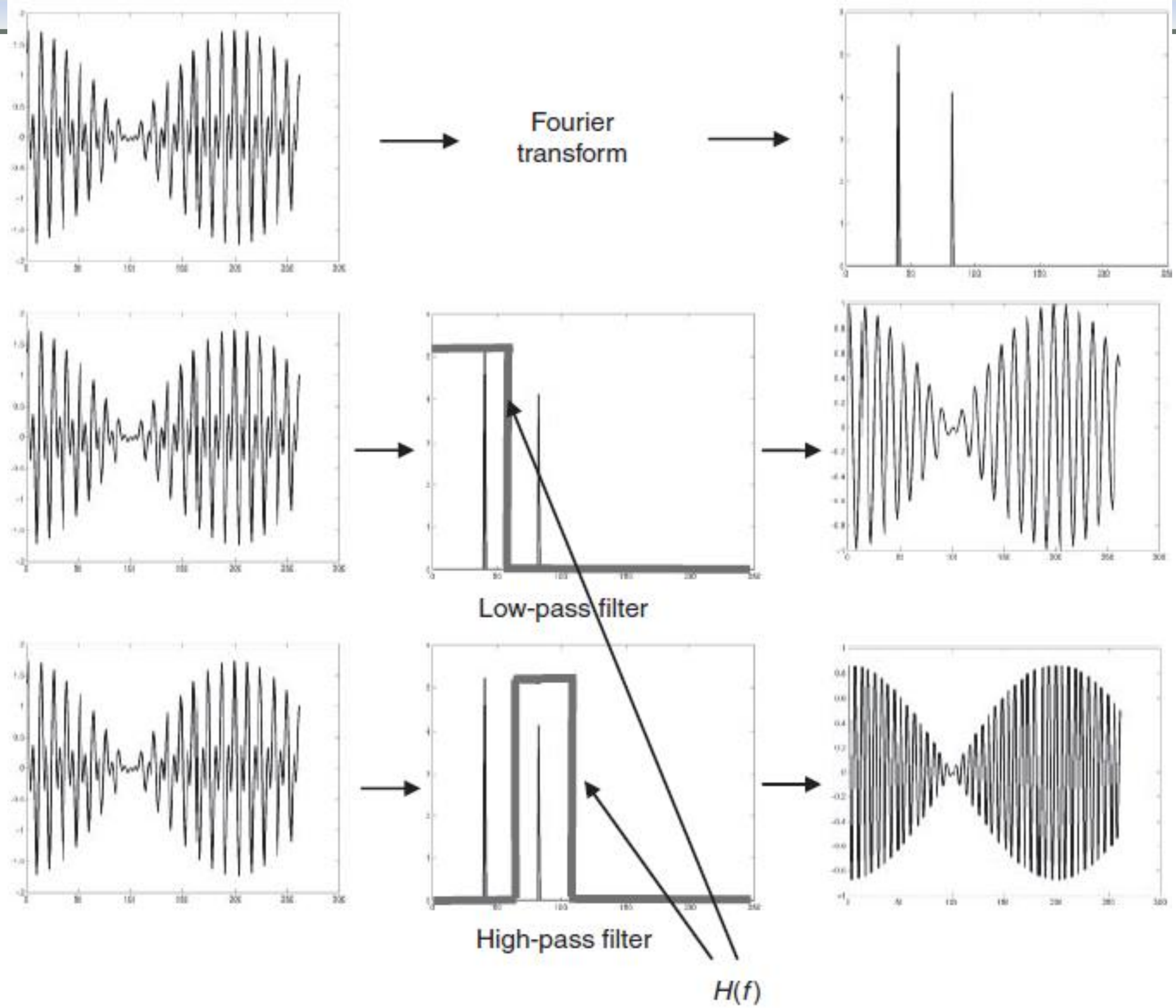
# Filtering

- 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

- 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



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# Audio file size

- **There are two types of recording such as:-**
  1. **Stereo** recordings are made by recording on two channels, and are lifelike and realistic.
  2. **Mono** sounds are less realistic, flat, and not as dramatic, but they have a smaller file size.
- Stereo sounds require twice the space as compared to mono recordings.
- **Audio resolution** (such as 8- or 16-bit) determines the accuracy with which a sound can be digitized. Using more bits for the sample size yields a recording that sounds more like its original.
- Here are the formulas for determining the size (in bytes) of a digital recording.
  - For a **monophonic recording**:

    sampling rate * duration of recording in seconds * (bit resolution / 8) * 1
  - For a **stereo recording**:

    sampling rate * duration of recording in seconds * (bit resolution / 8) * 2

# Audio file size

- **Example1:** Thus the formula for a 10-second mono recording at 22.05 kHz, 8-bit resolution would be

$$22050 * 10 * 8 / 8 * 1 = 220{,}500\ bytes$$

- **Example 2:** A 10-second stereo recording at 44.1 kHz, 16-bit resolution would be

$$44100 * 10 * 16 / 8 * 2 = 1{,}764{,}000\ bytes$$

- **Example 3:** A 40-second mono recording at 11 kHz, 8-bit resolution would be

$$11000 * 40 * 8 / 8 * 1 = 440{,}000\ bytes.$$

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# Commonly Used Audio Formats

Sounds, like other computer files, can be saved in different formats.

1. **WAV**
   - WAV is a sound file developed by Microsoft for use on windows based machines.
   - WAV is also the file format for standard music CDs.
   - The WAV file uses interesting algorithms to compress raw sound without a loss in quality. (Lossless compression)
2. **AIFF**
   - Developed by Apple, the "Audio Interchange File Format" is mostly used by Macintosh machines.
   - AIFF files are easily converted to other file formats, but can be quite large. One minute of 16-bit stereo audio sampled at 44.1kHz usually takes up about 10 megabytes.
   - AIFF is often used in high end applications where storage space is not a consideration.

# Commonly Used Audio Formats (cont'd)

Sounds, like other computer files, can be saved in different formats.

3. **MP3**
   - MP3 is a type of compression that can dramatically reduce file size without drastically reducing sound quality. (Lossy compression)
   - MP3 works by chopping off all sounds that are outside of the normal human range of hearing.
4. **RealAudio**
   - Developed by Progressive Networks, RealAudio was the first format to allow for real time streaming of music and sound over the web.
   - Listeners are required to download the Realplayer to enjoy sound in RealAudio Format.
   - The Realplayer can also stream video and is currently in use by millions of Internet users worldwide.

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# Graphics

- Computer graphics has evolved into an advanced field having applications in high end markets such as visual effects movies, interactive and multiplayer games, scientific visualization, and computer aided design, to name a few.
- Most of today's graphics-related applications tend to be three dimensional, interactive, and available on a variety of platforms from computers, to game consoles, to handheld devices.
- Graphics objects can be represented as vectors or rasters.
  - Vector graphics are geometric entities saved in a vector format having attributes such as color.
    - For example, a shape can be represented as a sequence of points to be connected in a specific order and filled with a color.
    - The advantage of vector representations is that they provide infinite resolution; however, vector graphics need to be converted to raster images to be displayed.
  - Raster images are represented as a grid of pixels, each pixel having *x,y* coordinates and a value that corresponds to a color. Sample operations performed on raster images include painting, compositing, and filtering effects.

# Graphics (cont'd)

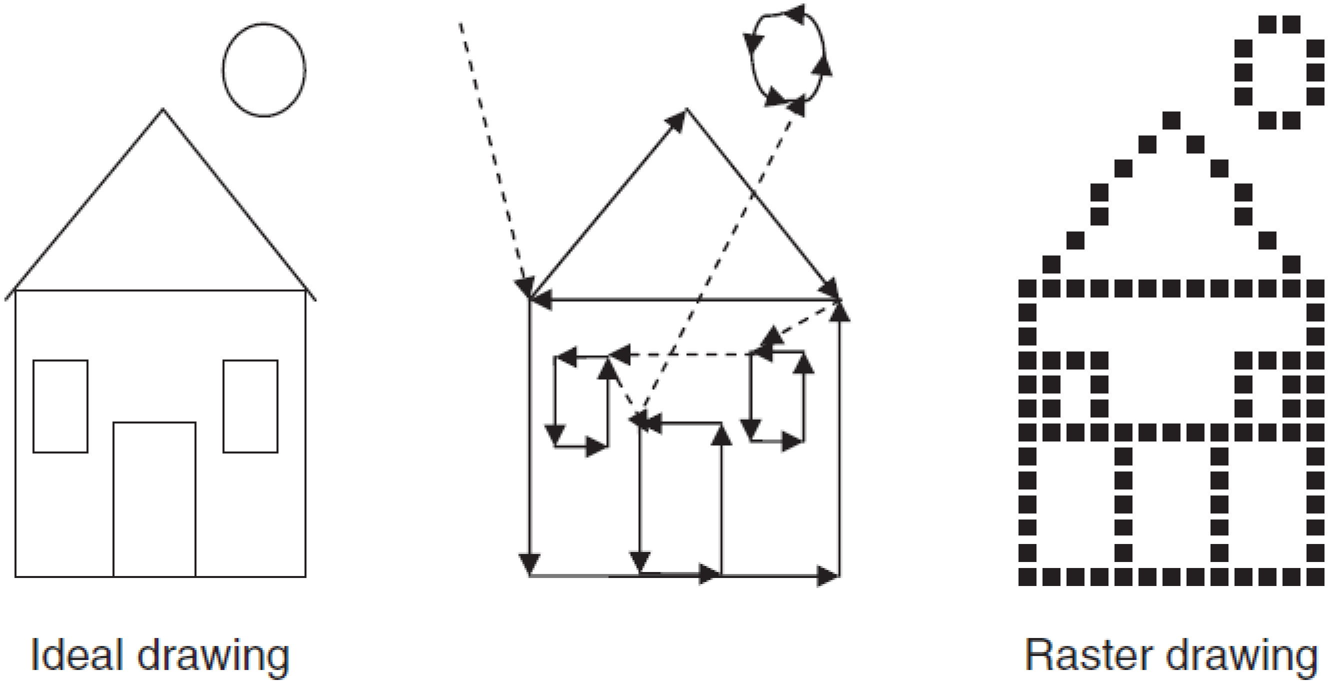


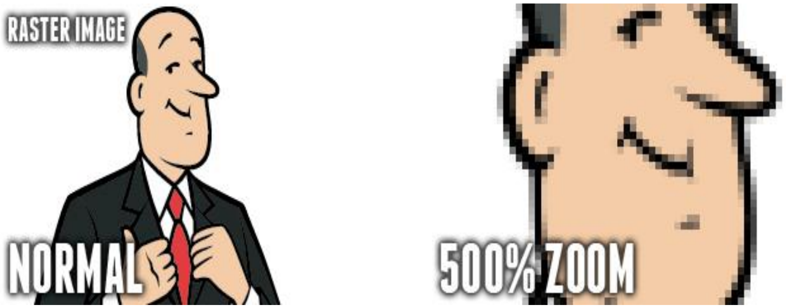

*Figure 3-17 The left image shows a drawing that needs to be represented graphically. The middle image shows the corresponding representation using vectors. This is shown as a sequence of points joined by lines. The dotted lines show cursor movements. The right image shows a raster representation of the drawing.*

# Raster images

- **Raster images (Bitmap images)** are made up of a set grid of dots called pixels where each pixel is assigned a color value.
- Unlike a vector image, raster images are **resolution dependent**.
- A **raster file** is usually larger than a **vector graphics** image file.
- A raster file is usually difficult to modify without loss of information
- Examples of raster image file types are: BMP, TIFF, **GIF**, and **JPEG** files.

# Raster images (cont'd)

- **There different sources of raster image:**
  - Photographs from digital cameras
  - Screenshots
  - Scanned images
  - Captured video frames are bitmaps




- **Advantages of raster image:**
  - Bitmap imaging is suitable for continuous tone pictures like photographs

- **Disadvantages of raster image:**
  - When we resize a bitmap image it tends to lose quality. As you **shrink** or **stretch** the pixels **themselves** which results in a significant loss of clarity and very **blurry image.**
  - The file size of a bitmap image is large. Computer has to **store information** about **every single pixel** in the image.

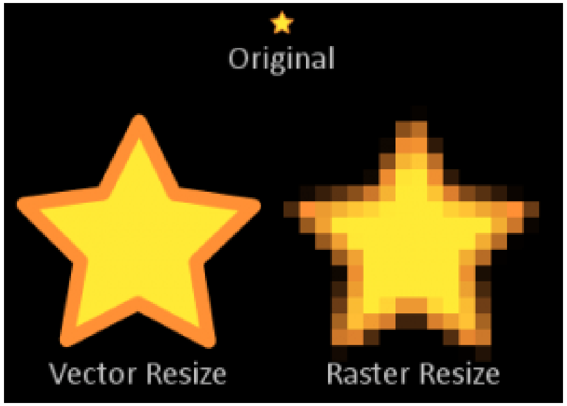
# Vector images

- ***Vector graphics*** is an image file format suitable for pictures with areas of solid, clearly separated colors—cartoon images, logos, and the like.
- Instead of being painted pixel by pixel, a vector graphic image is **drawn object by object in terms of each object's geometric shape**.
- Many file formats for vector graphics—*.fh*, *.ai*, *.wmf*, *.eps*, etc.—they contain the parameters to mathematical formulas defining how shapes are drawn.
  - A line can be specified by its endpoints,
  - a square by the length of a side,
  - a rectangle by the length of two sides,
  - a circle by radius
- **Sources of vector image:**
  - Charts
  - Diagrams
  - Data visualizations.

# Vector images (cont'd)

- **Advantages of vector images:**
  - Vector graphics imaging is suitable for pictures that have solid colors and well-defined edges and shapes
  - It **does not** take up so much space (small size).
  - Only **contain data** about the **points**, **mathematical formulas**, **lines** and **curves** which form the object.
  - Easily converted to bitmapped.
  - Can be scaled (resized) keeping original quality and **file size**.

- **Disadvantages of vector images:**
  - Only **individual objects** can be edited not pixel by pixel.
  - **Difficult** to recreate **realistic images**.



# Conversion between vector Image and raster image

- **Converting vector graphics into raster image is allowable.**
- **In another hand, converting raster (bitmaps) image is** difficult since it is based on computing **the bounds** of a bitmapped image or **the shapes** of colors within an image and then **derive the polygon** object that describe the **image**.


**Raster to vector conversion**

# Thank You