BENC 3443
Multimedia Technology & Application

Chapter 4: Multimedia Data
(Part 1)

Outline

- Analog & Discrete Phenomena
- Image & Sound Data Represented as Functions & Waveforms
- Sampling & Aliasing
- Quantization, Quantization Error and Signal-to-Noise Ratio
- Data Storage
- Bandwidth & Data Rate

Analog & Discrete Phenomena

Multimedia Data Basics:

Multimedia systems/applications have to deal with:

- Generation of data
- Manipulation of data
- Storage of data
- Presentation of data
- Communication of information/data

A majority of this data is large and the different media may need synchronisation:

- The data will usually have temporal relationships as an integral property.

Analog & Discrete Phenomena

Static & Continuous Media

Static or Discrete Media :

- Some media is time independent:
  - Normal data, text, single images, graphics are examples.

Continuous Media :

- Time dependent Media:
  - Video, animation and audio are examples.

Analog & Discrete Phenomena

Analog & Digital Signals

Analog signal - A signal that has a continuous nature rather than a pulsed or discrete nature. @
continuous signals of sometime varying quantities; can't be processed directly by computers.

Example:
Electrical or physical analogies, such as continuously varying voltages, frequencies, or phases, may be used as analog signals

Advantages of Analog Signal:

- has the potential of infinite resolution of the signal (high density)
- processing is simple

Disadvantages of Analog Signal :

- noise – as the signal is copied and re-copied or transmitted over long distances random variations occur
- impossible to recover from noise/distortion

Analog & Discrete Phenomena

Analog & Digital Signals

Digital signal - a signal which is represented as a sequence of numbers (usually in binary numbers) @

digital samples of the signals at regular interval; can be readily processed by computers

Example:

digital image – matrix of pixels, digital sound – vector of sound amplitudes

Advantages of Digital Signal :

- as opposed to analog signals, degradation of the signal (i.e. noise) can not only be detected but corrected as well
- scales well with the increased complexity of the system

Disadvantages of Digital Signal :

- it is error prone (due to quantization and sampling)
- it has lower resolution than analog signals

Analog & Discrete Phenomena

Analog & Digital Signal Conversion

The world we sense is full of analog signals:

Electrical sensors convert the medium they sense into electrical signals

- Example:
  - transducers, thermocouples: temperature sensor,
  - microphones: acoustic sensor
  - Cameras (Still and Video): light sensor.

Analog or continuous signals must be converted or digitized for computer processing.

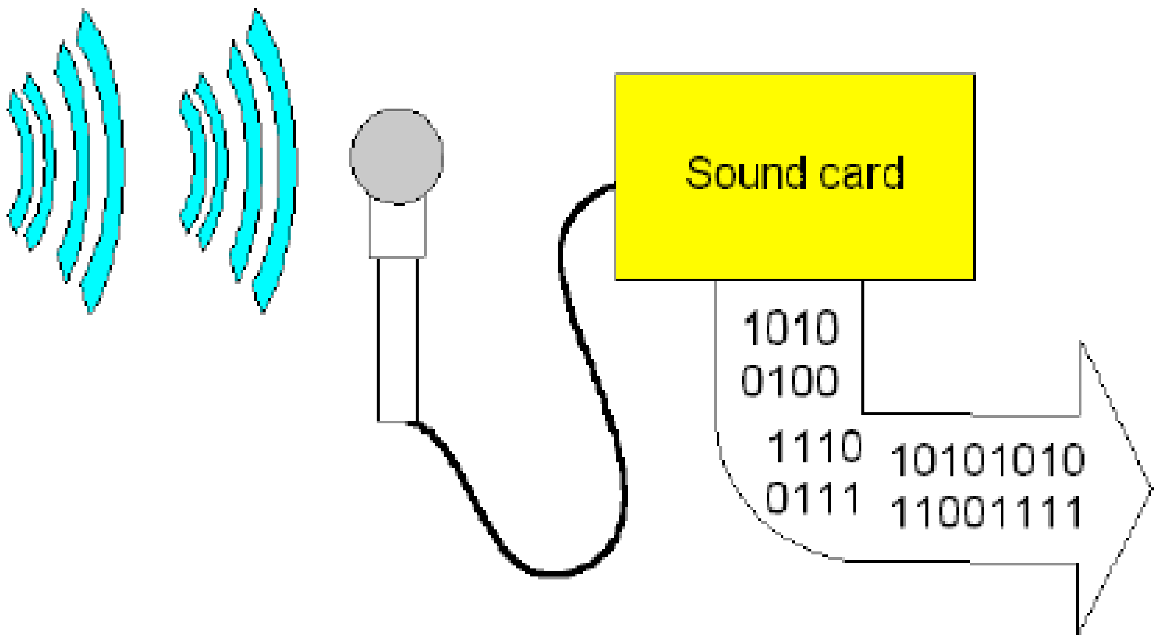
Digital or discrete signals are signals that computer can readily deal with.

Analog & Discrete Phenomena

Analog to Digital Signal Converter (ADC)

- Special hardware devices : Analog-to-Digital converters.
- E.g. Audio:

  Take analog signals from analog sensor (e.g. microphone) and digitally sample data

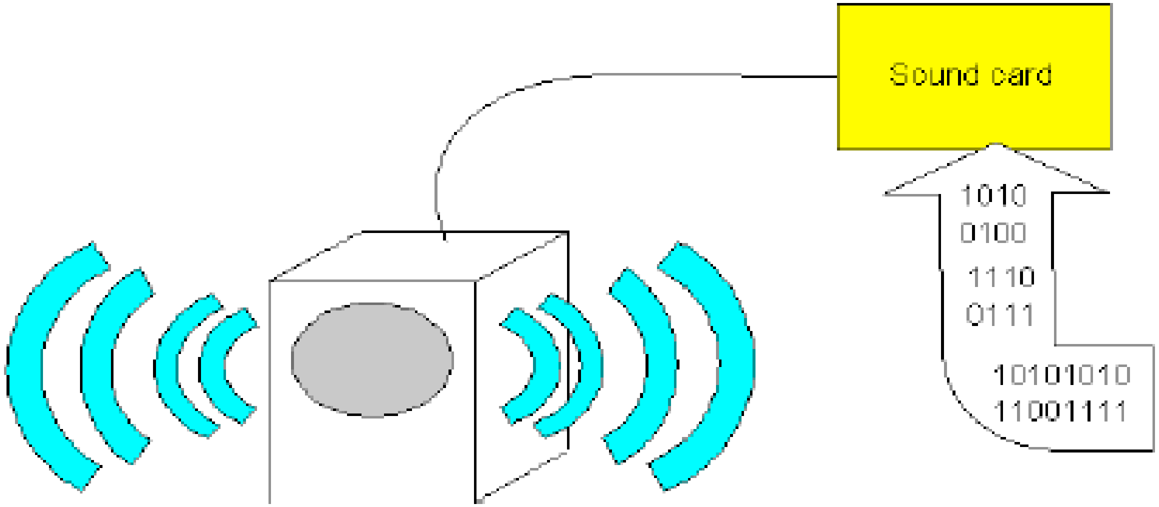


Analog & Discrete Phenomena

Digital to Analog Signal Converter (DAC)

Playback - a converse operation to Analog-to-Digital

- Takes digital signal, possible after modification by computer (e.g. volume change, equalization) and outputs an analog signal that may be played by analog output device (e.g. loudspeaker, RGB monitor/display)



Analog & Discrete Phenomena

Analog to Digital Signal Conversion:

Converting a continuous analog signal into a discrete digital signal has TWO (2) sub processes:

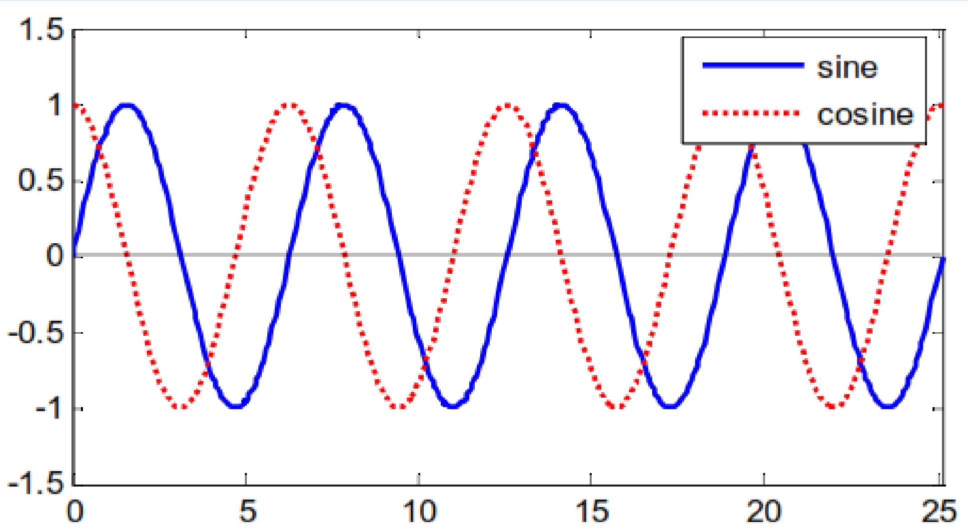
1) sampling - conversion of a continuous-space/time (audio, video) signal into a discrete-space/time (audio, video) signal
2) quantization - converting a continuous-valued (audio, video) signal that has a continuous range (set of values that it can take) of intensities and/or colors into a discrete-valued (audio, video) signal that has a discrete range of intensities and/or colors; this is usually done by rounding, truncation or other irreversible non-linear process of information destruction

Sound & Image Representation

- two primary media in digital media are images and sound
- in the sense that the two are combined to produce video
- can be represented as functions, and we can visualize these functions by means of their corresponding graphs
- Sound, is a one-dimensional function – i.e., a function with one variable as input

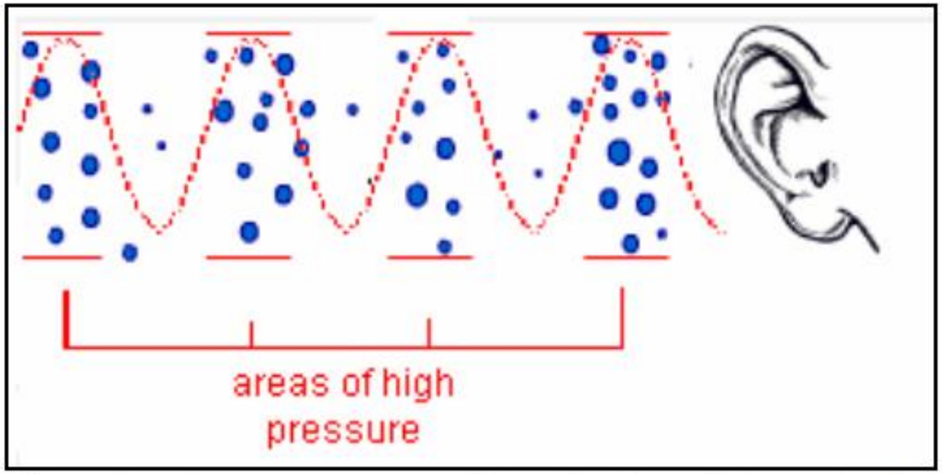
Sound & Image Representation

- If we think of sound as a continuous phenomenon, then we can model it as a continuous function where *x* is time and *y* is the air pressure amplitude.
- the essential form of the function representing sound is sinusoidal - the shape of a sine wave
- sines and cosines - ***sinusoidal functions.***

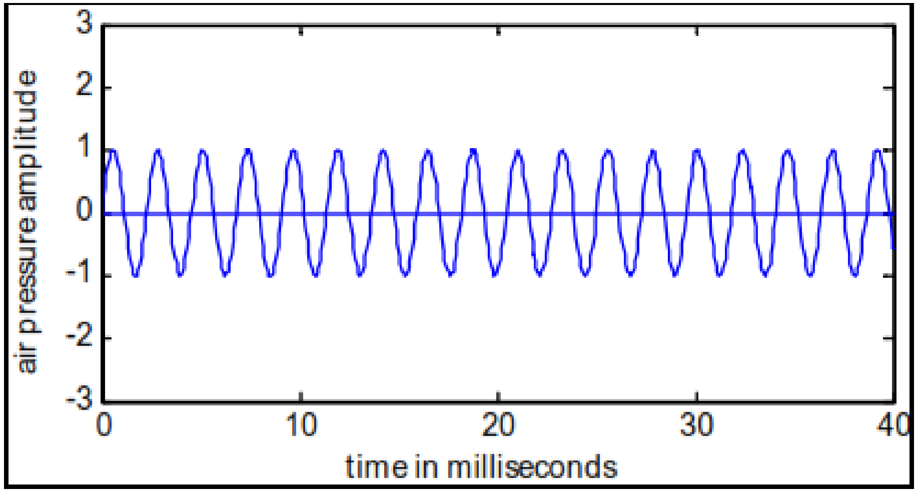


Graphs of sine and cosine waves

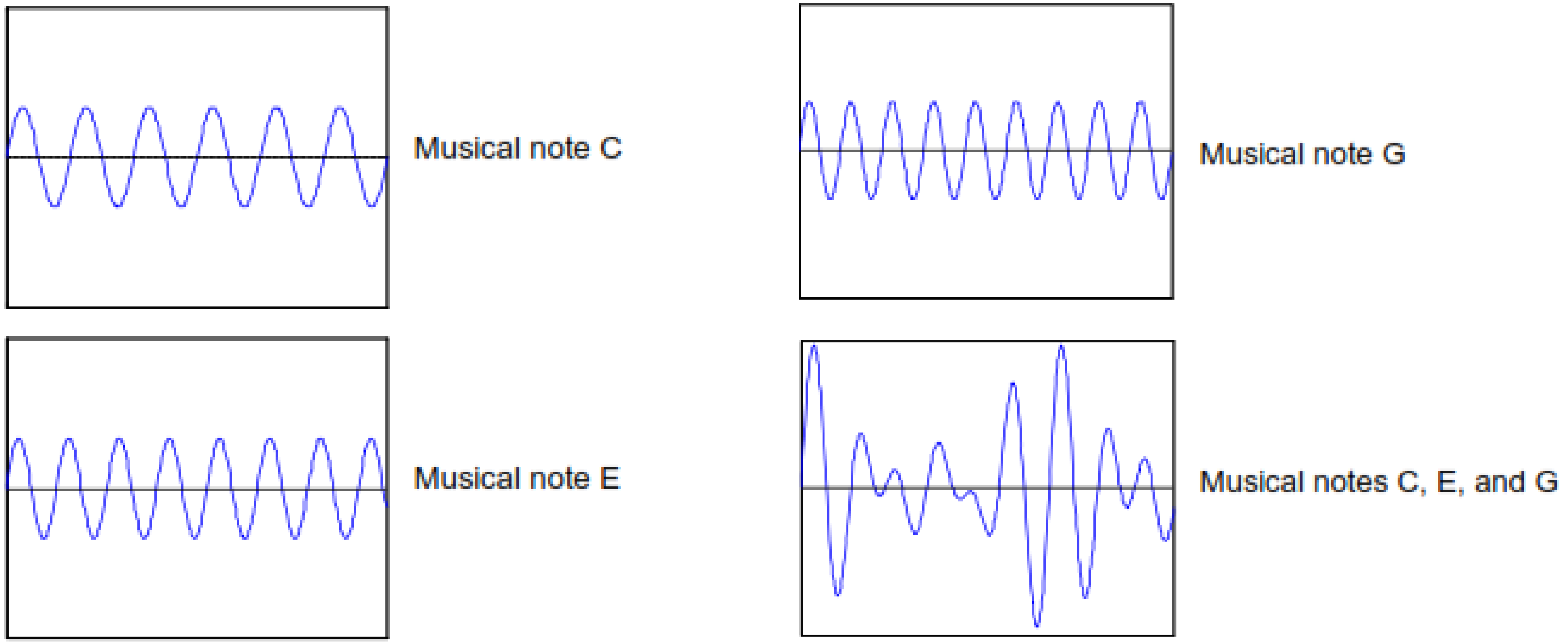
Sound is a *mechanical wave*, which means that it results from the motion of particles through a transmission medium – for example, the motion of molecules in air.



Changing air pressure caused by vibration of air molecules



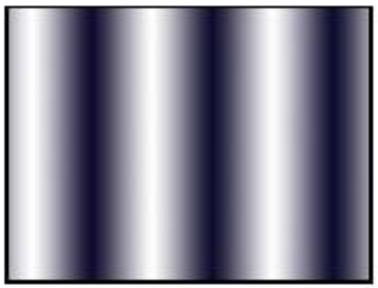
A single-frequency (440 Hz) tone with no overtones, represented as a waveform



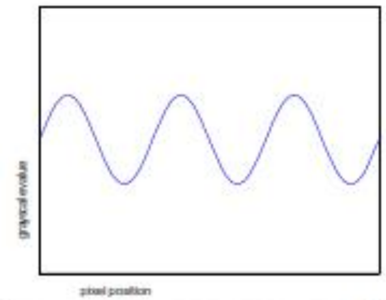
Adding sound waves

Image Representation

Sinusoidal waveforms can also be used to represent changing color amplitudes in a digital image.

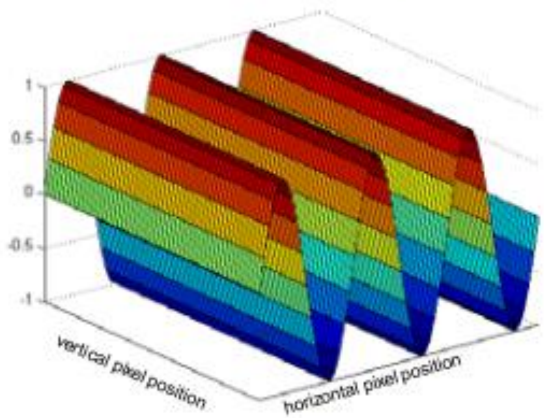


assume that the image is continuous, with an infinite number of points in each row and column of the image



Grayscale image, smooth gradient

Sine wave corresponding to one line of grayscale values across the image to the left



Two-dimensional waveform corresponding to the grayscale image above

Sampling Rate & Aliasing

Analog-to-digital conversion: **sampling** and **quantization**

Sampling - chooses discrete points at which to measure a continuous phenomenon (in general a signal).
Images - the sample points are evenly separated in space.
Sound -the sample points are evenly separated in time.

The number of samples taken per unit time or unit space is called the **sampling rate or, alternatively, the resolution.**

Quantization - each sample be represented in a fixed number of bits, called the **sample size, or, equivalently, the bit depth.** The bit depth limits the precision with which each sample can be represented.

- Sampling rate
  - Number of sample taken of a signal in a given time (usually one second)
- Bit depth
  - Describes the accuracy of the audio data
  - Also called “sampling resolution” or “word length”.
  - The more bits, the better is the quality of the audio (and a larger file of course).
  - Common bit depths are 8-bit (telephone like), 16-bit (CD quality), and 20, 24, 32, 48-bit depths.
  - How many signal can a 8-bit and a 16-bit data represent?

    0000 0000 → 1111 1111

    0000 0000 0000 0000 → 1111 1111 1111 1111

Sampling Rate & Aliasing

Aliasing - is an effect that causes different signals to become indistinguishable (or aliases of one another) when sampled.

It also refers to the distortion or artifact that results when the signal reconstructed from samples is different from the original continuous signal

The Nyquist theorem

To guarantee accurate reconstruction of a sampled analog signal, we must use a sampling rate that is greater than twice the frequency of the highest frequency component in the signal.

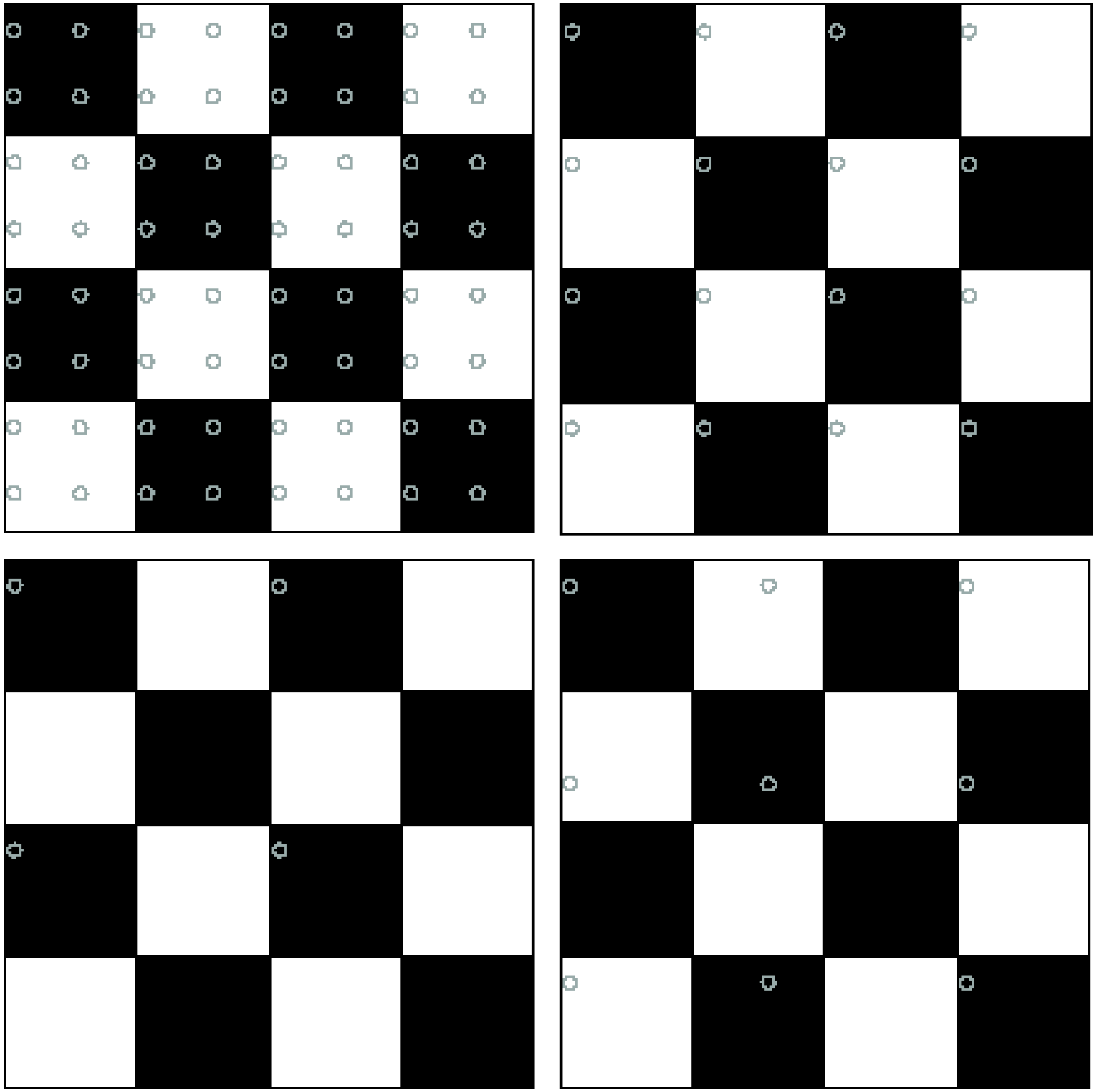
Nyquist frequency:

- the highest actual frequency component that can be sampled at the given rate without aliasing

- half the given sampling rate

e.g., samples at a rate of 8000 Hz (i.e., 8000 samples/sec), then the Nyquist frequency is **?**

Image Sampling & Aliasing

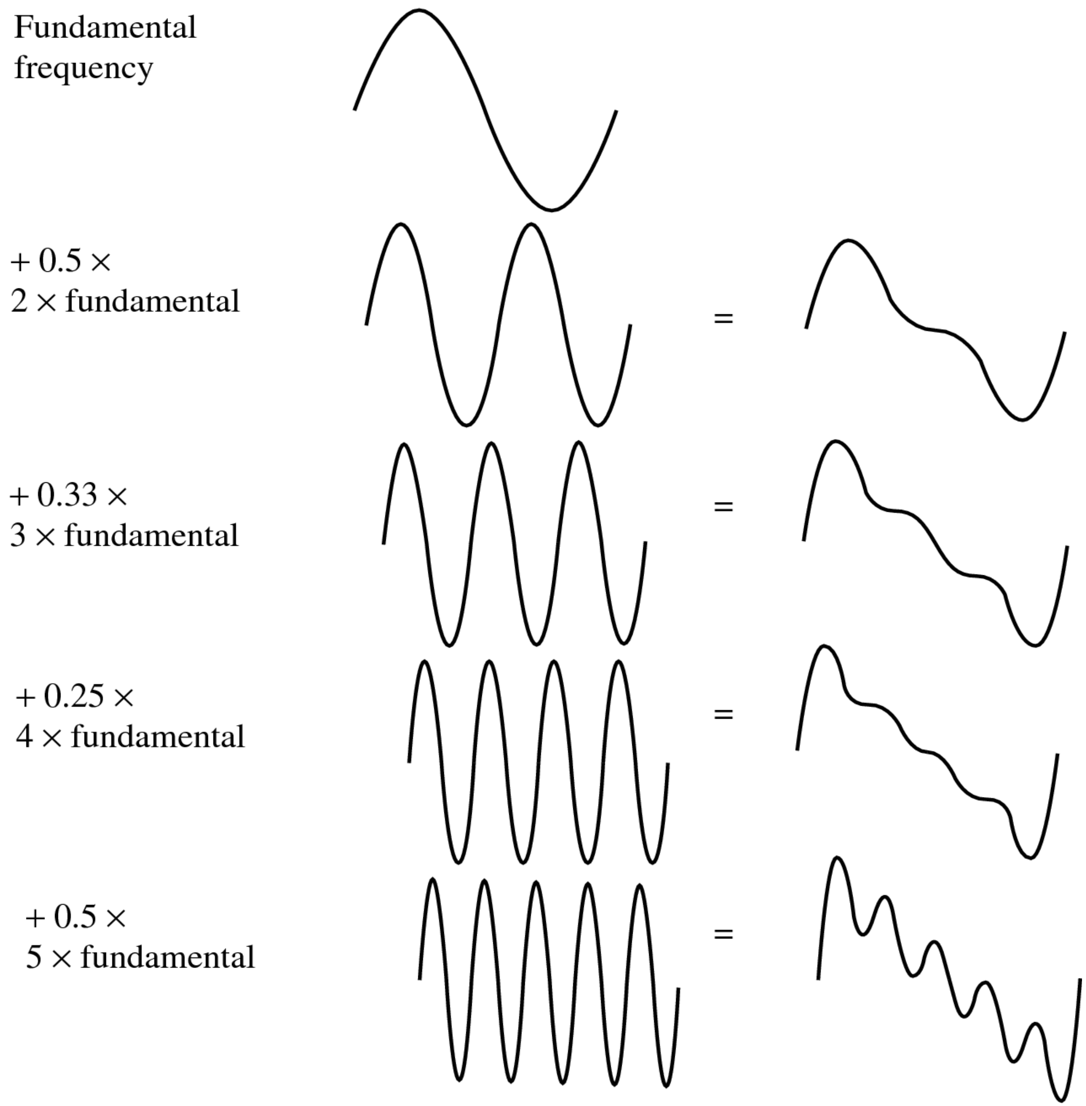


Good sampling

Bad sampling

Aliasing - undersampling and results in an image that does not match the original source – it may be blurred or have a false pattern.

Sound Sampling & Aliasing



Harmonics – any series of signals whose frequencies are integer multiples of the frequency of the fundamental signal.

Building up a complex signal by superposing weighted sinusoids

UNIVERSITI TEKNIKAL MALAYSIA MELAKA

Nyquist rate:

- the lowest sampling rate that will permit accurate reconstruction of an analog digital signal

- the Nyquist rate is twice the frequency of the highest frequency component in the signal being sampled

e.g., if you want to sample an audio wave that has a frequency component of 10000 Hz, then the sampling rate must be **?**

Given $f_{\max}$, the frequency of the highest-frequency component in an audio signal to be sampled, then the ***Nyquist rate***, f_{nr}, is defined as

$$f_{nr} = 2f_{\max}$$

Given a sampling frequency f_{samp} to be used to sample an audio signal, then the ***Nyquist frequency***, f_{nf}, is defined as

$$f_{nf} = \frac{1}{2} f_{samp}$$

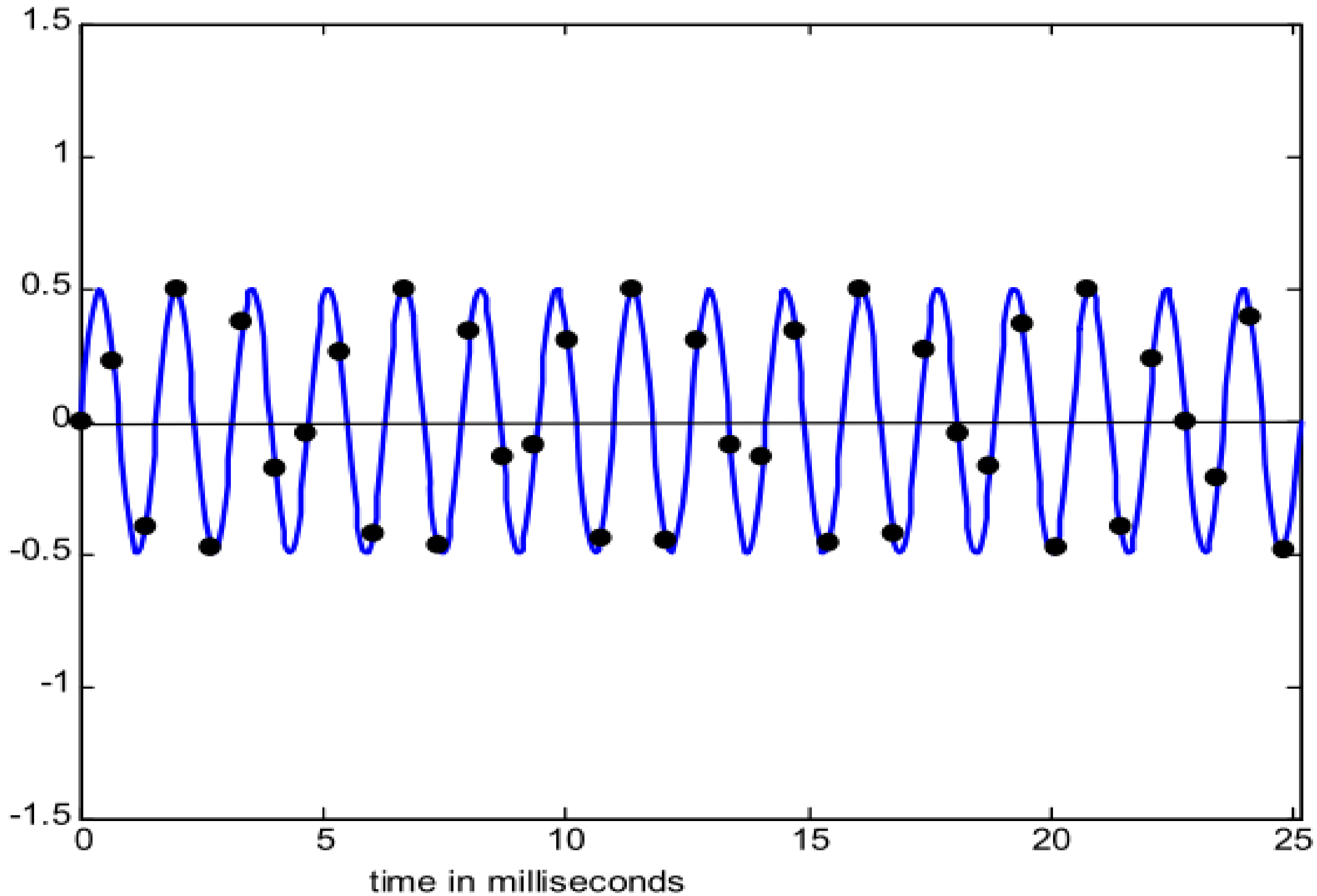
Nyquist Theorem for **band-limited signal**:

If a signal have components of lower limit f_1 and an upper limit f_2 of frequency the *sampling rate* should be at least $\mathbf{2(f_2 - f_1)}$.

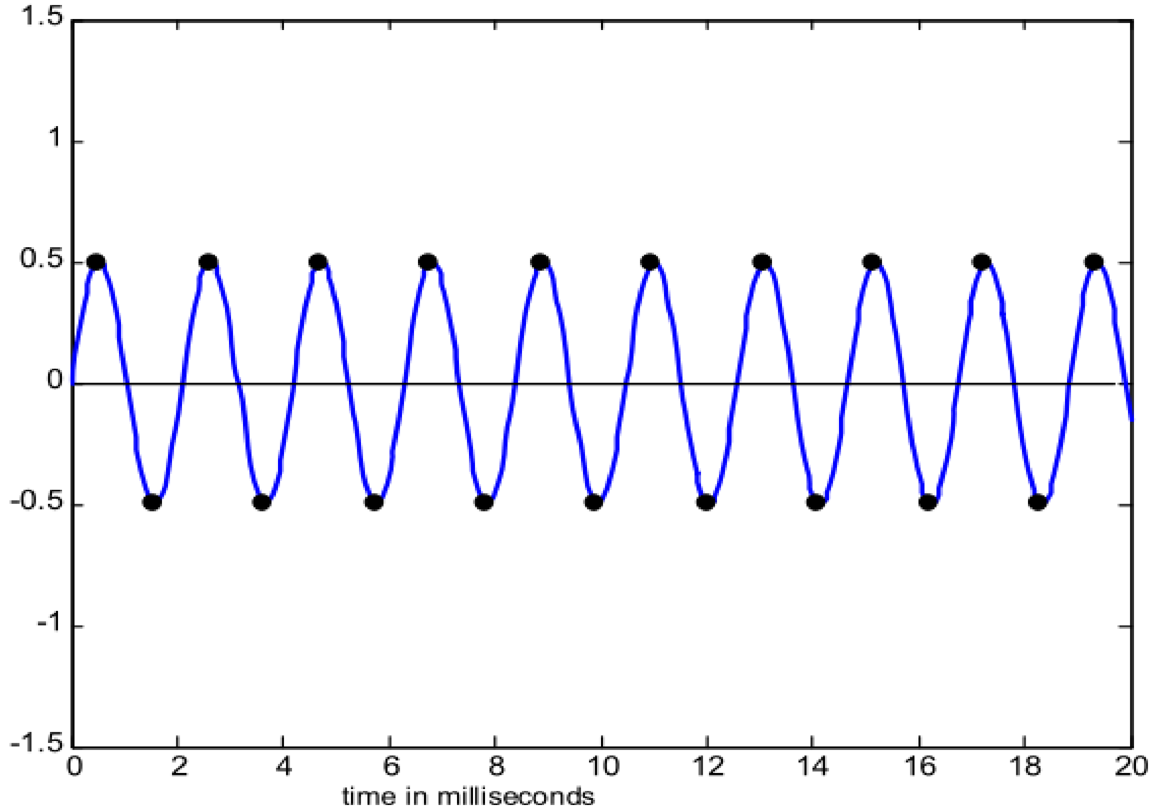
Audio Aliasing

When the digitized sound is played, the frequency from the original sound will be translated to a different frequency, so the digitized sound doesn’t sound exactly like the original.

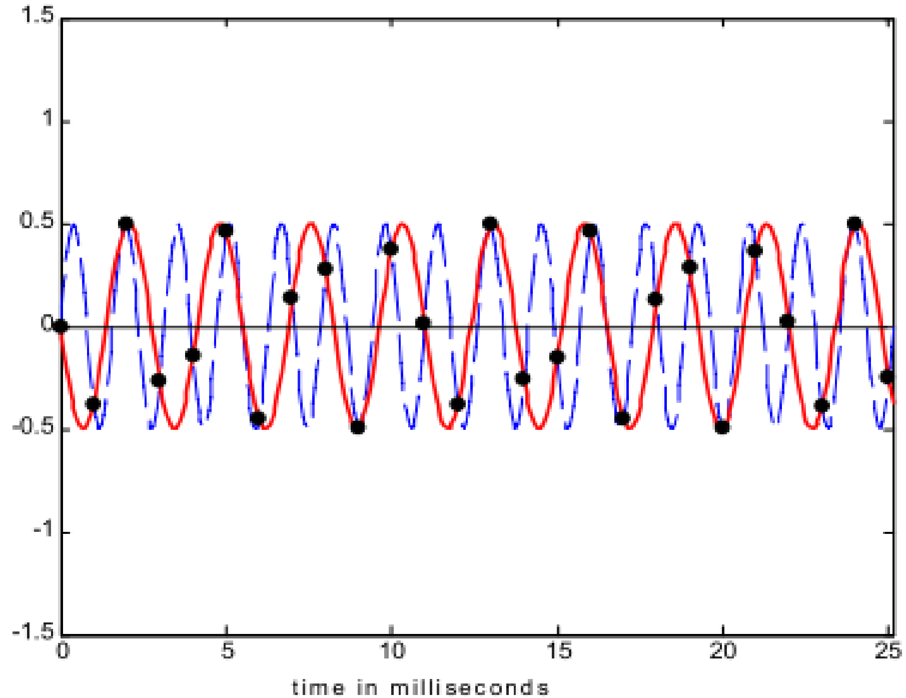
- the reason a too-low sampling rate results in aliasing is that there aren't enough sample points from which to accurately interpolate the sinusoidal form of the original wave
- take *more than two samples per cycle* on an analog wave, the wave can be precisely reconstructed from the samples



The signal is 637Hz, which is 637 cycles/s.
What is the appropriate sampling frequency?



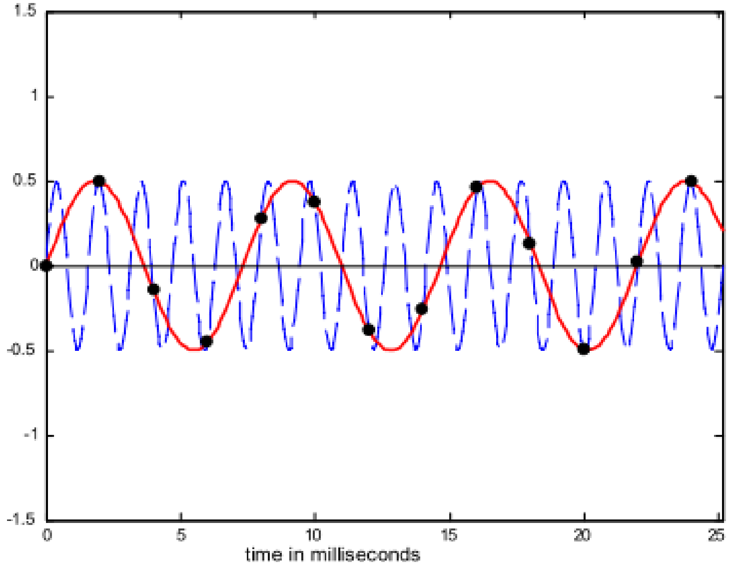
If we have exactly two samples per cycle and the samples are taken at precisely the maximum and minimum values of the sine wave, once again the digitized wave can be reconstructed. However, if the samples are taken at locations other than peaks and troughs, the frequency may be correct but the amplitude incorrect.



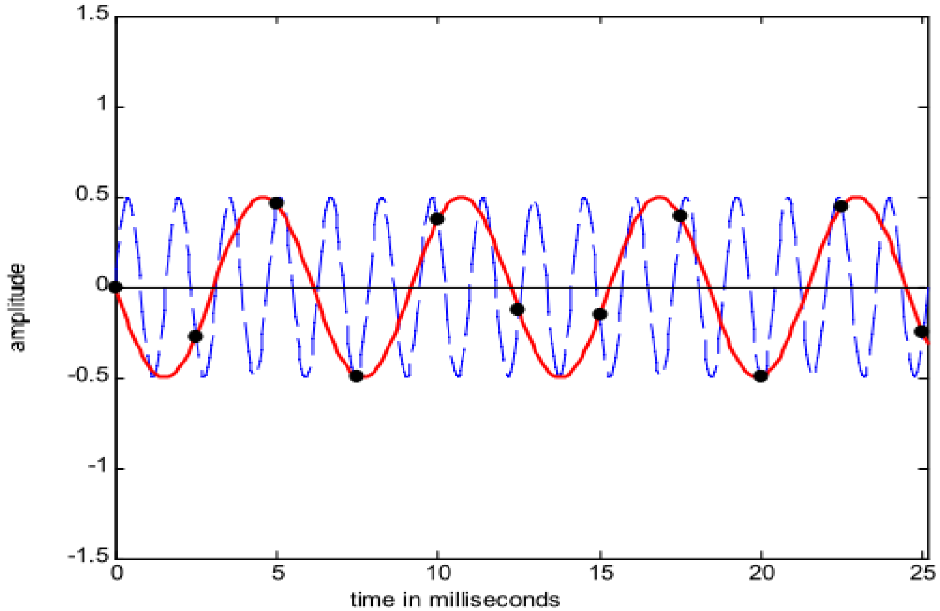
A 637 Hz wave sampled at 1000 Hz aliases to 363 Hz

The inadequate sampling rate "skips over" some of the cycles, making it appear that the frequency of the actual wave is **lower than it really** is.

The actual frequency is the sine wave drawn with the dashed line in the background.



A 637 Hz wave sampled at 500 Hz aliases to 137 Hz



A 637 Hz wave sampled at 400 Hz aliases to 163 Hz

Exercise

1. What is the cause of aliasing in digital media?

2. Say that you want to digitally record a sound that has a highest frequency component of 650 Hz. According to the Nyquist theorem, what sampling rate do you have to use to be sure that you don't get sound aliasing?

3. If you are recording an audio file and expect that the highest frequency in the file will be 10kHz,what is the minimum sampling rate you can use to ensure you won't get audio aliasing?

Quantization

Digital images - each sample (pixel) represents a color at a discrete point in a two dimensional image

How many colors can we possibly represent in a digital image?

- determined by the number of bits used to represent each sample – i.e., the sample size or bit depth (which, in the case of an image file, can also be called the *color depth*)

Let n be the number of bits used to quantize a digital sample. Then the maximum number of different values that can be represented, m, is

$$m = 2^n$$

Quantization

If one bit is used per sample, only 2 colors are possible - 1 bit can take 2 values, 0 or 1.

How about if 8 bits and 24 bits are used?

Quantization

The "different values" in the definition above correspond to color levels or sound amplitudes.

Image

- 2-bit: binary image
- 8-bit: grayscale image
- 24-bit: color image

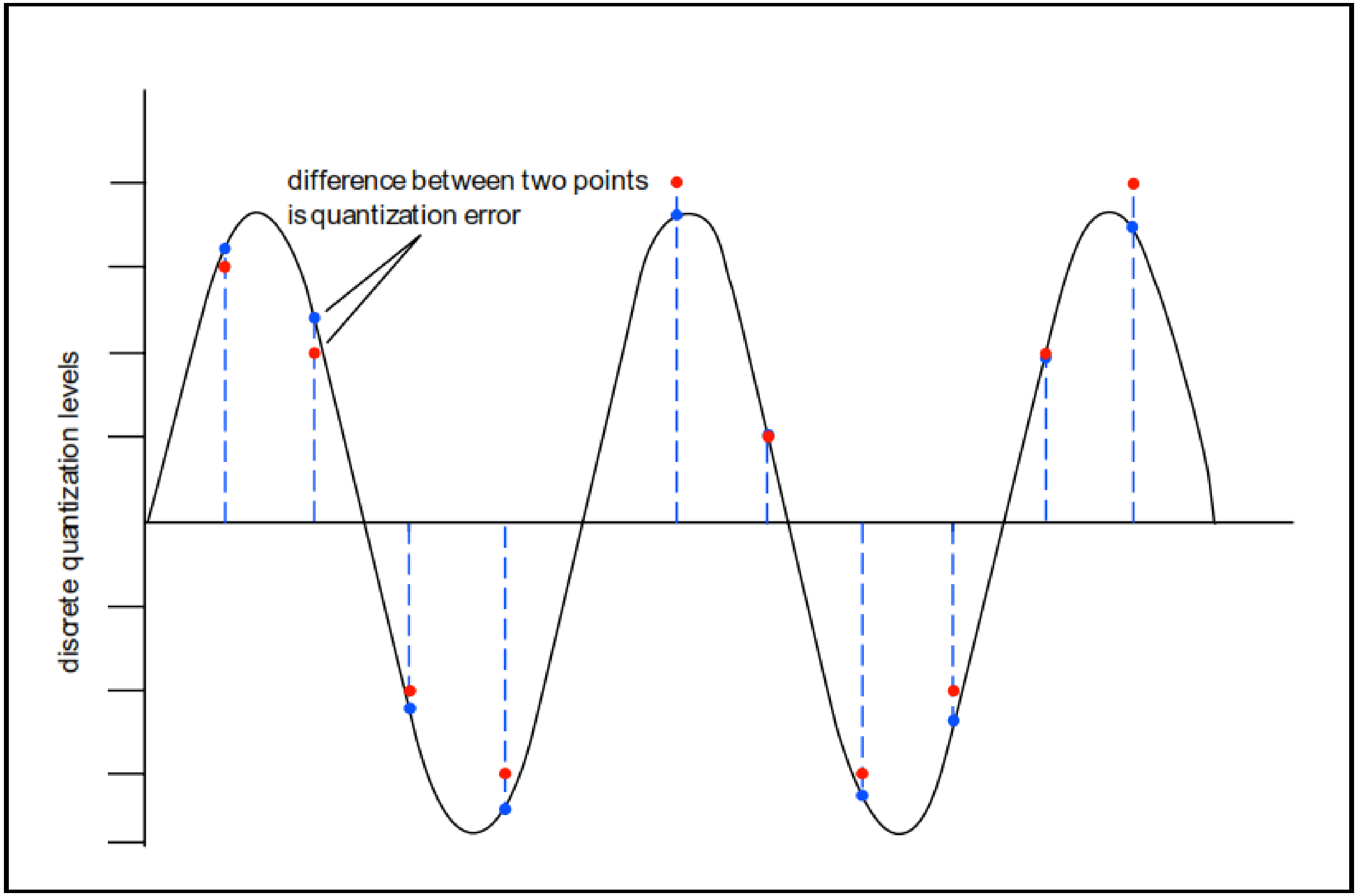
Sound

Stereo CD-quality digital audio - 16 bits per sample in each channel, for a total of 32 bits per sample

Sample size affects how precisely the value of a sample can be represented.

Quantization Error

A sample must be rounded to the closest discrete level. The difference between its actual value and its rounded value is the quantization error.



Quantization error

Exercise

1. You have a digital image in 8-bit color that has blocky areas of color, lacking the subtle gradations from one color to the next. Is this a matter of aliasing or quantization error? Explain.

2. Give two (2) examples that may results in an image with low sampling rate. Give reason.

Signal-to-quantization error (SQNR)

With n bits for quantization, the samples values range from -2^{n-1} to $2^{n-1}-1$.

The maximum sample value is 2^{n-1}. On the scale of sample values, the maximum quantization error is half a quantization level (the denominator $\frac{1}{2}$, assuming rounding is used in quantization).

$$\boldsymbol{SQNR = 20log_{10}\left(\frac{\max(quantization\ value)}{\max(quantization\ error)}\right) = 20log_{10}\left(\frac{2^{n-1}}{1/2}\right)}$$

Let n be the bit depth of a digitized media file – e.g., digital audio. Then the signal-to-quantization noise ratio, ***SQNR*** is

$$SQNR = 20\log_{10}(2^n)$$

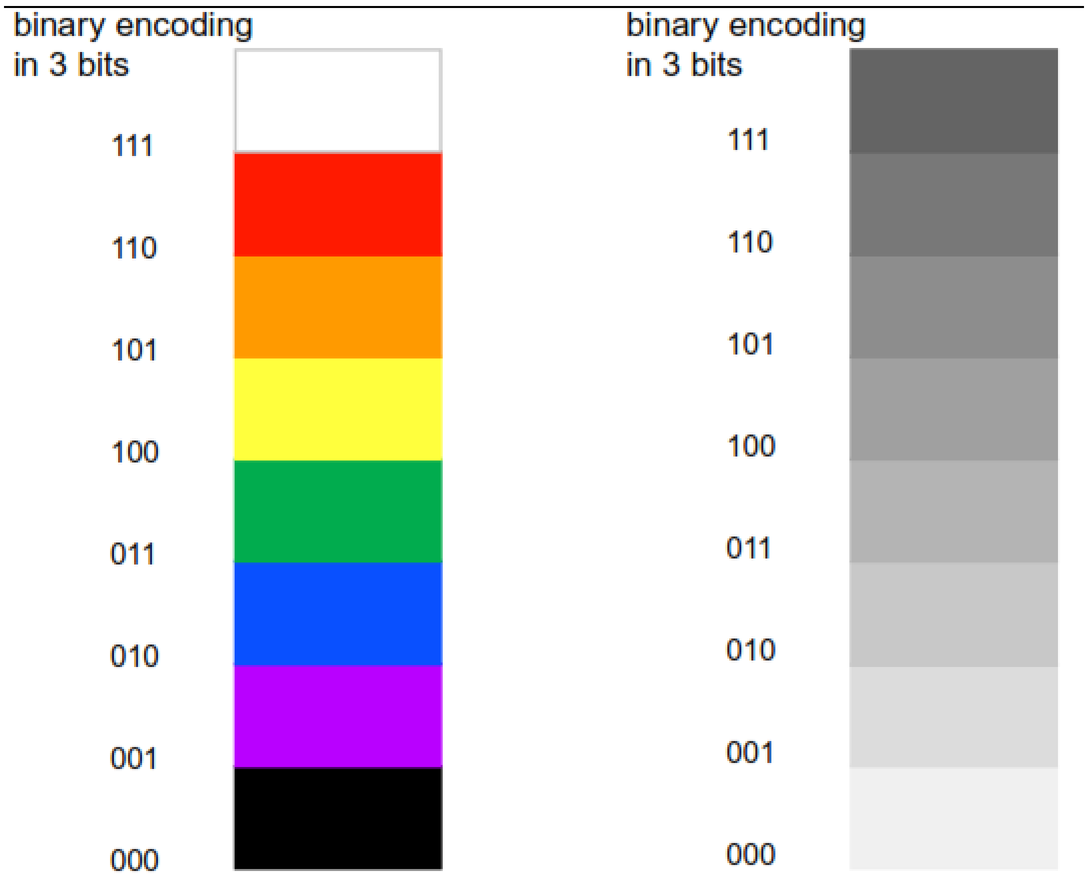
Dynamic Range

Dynamic range gives you a measure of the range of amplitudes that can be captured relative to the loss of fidelity compared to the original sound/image.

Dynamic range in digital image- the ratio of the largest amplitude of colors and the smallest that can be represented with a given bit depth.

Dynamic range in digital audio- the relative difference between the loudest and softest parts representable in a digital audio file, as a function of the bit depth.

Dynamic Range in Image



With 3 bits, you have eight colors. You can spread these colors out over a wide range, with big differences between one and the next, or spread them out over a narrow range, with small differences between one and the next. In either case, the dynamic range is the same, dictated by the bit depth, which determines the maximum error possible (resulting from rounding to available colors) relative to the range of colors represented.

Dynamic Range in Audio

Let n be the bit depth of a digital audio file. Then the ***dynamic range of the audio file***, d, in decibels, is defined as

$$d = 20n \log_{10}(2) \approx 6n$$

For example, a 16-bit digital audio file has a dynamic range of **?**

A 8-bit digital audio file has a range of **?**

- the **sampling rate** and **bit depth (related to quantization)** are needed for creating a new digital audio file
- Audio sampling rates are measured cycles per second, designated in units of **Hertz (Hz)**

  - **8000 Hz** mono for telephone quality voice
  - **44.1 kHz** two-channel stereo with 16 bits per channel for CD-quality sound.
  - Digital audio tape (DAT) format uses a sampling rate of **48 kHz**
  - sampling rates of **96 or 192 kHz** for two-channel stereo DVD with **24 bits** per channel

Data Storage

kilobyte	kB
megabyte	MB
gigabyte	GB
kilobit	kb
megabit	Mb
gigabit	Gb
terabit	Tb
terabyte	TB

For memory and file sizes, assume the following equivalences:

1 byte = 8 bits

1 kB = 2^{10} bytes = 1024 bytes

1 MB = 2^{20} bytes = 1,048,576 bytes

1 GB = 2^{30} bytes = 1,073,741,824 bytes

1 TB = 2^{40} bytes = 1,099,511,627,776 bytes

Example digital image file size (without compression):

Resolution: 1024 X 768 pixels

Total number of pixels: 786,432

Color mode: RGB

Bits per pixel: 24 (3 bytes)

Total number of bits: 18,874,368 (2,359,296 bytes)

File size: 2.25 MB

UNIVERSITI TEKNIKAL MALAYSIA MELAKA

kilobyte	kB
megabyte	MB
gigabyte	GB
kilobit	kb
megabit	Mb
gigabit	Gb
terabit	Tb
terabyte	TB

For memory and file sizes, assume the following equivalences:

1 byte = 8 bits

1 kB = 2^{10} bytes = 1024 bytes

1 MB = 2^{20} bytes = 1,048,576 bytes

1 GB = 2^{30} bytes = 1,073,741,824 bytes

1 TB = 2^{40} bytes = 1,099,511,627,776 bytes

Example digital audio file size (without compression):

Sampling rate: 44.1 kHz

Bit depth: 32 (16 for each channel)

Number of minutes: one

Total number of bits: 84,672,000 (10,584,000 bytes)

File size: 10.09 MB for 1 minute

Data rate of the file: 1.35 Mb/s

Example digital Video file size (without compression):

Frame size: 720 X 480 pixels
Frame rate: ~30 frames/s
Number of minutes: one

Total image requirement:
14,929,920,000 bits
Audio requirement: 84,672,000
Total number of bits: 15,014,592,000 (=1,876,824,000 bytes)
File size: > 1.7 GB
Data rate of the file: 238.65 Mb/s

Storage Medium	Maximum Capacity
Portable Media	
CD (Compact Disk)	700 MB
DVD (Digital Versatile Disc or Digital Video Disk), standard one sided	4.7 GB standard; 8.5 GB dual-layered
DVD video or high capacity	17 - 27 GB
Memory stick or card	8 GB
HD-DVD (High Definition DVD), standard one sided	15 GB standard; 30 GB dual-layered
Blu-ray Disk	25 GB standard; 50 GB dual-layered
Flash drive	64 GB
Permanent Media	
Hard disk drive	1 terabyte (1000 GB)
Note: These values are approximate. Assume 1 GB = 1,000,000,000 bytes, as this is the assumption in the storage capacities reported by manufacturers.	

Storage media and their capacity

Bandwidth & Data Rate

- Sound and video are time-based media that require large amounts of data.
- Both capturing and transmitting sound and video in real-time require that the data transmission keep up with the rate at which the data is played.

Thus, issues of bandwidth and data rate are crucial in the capturing and transmitting of digital audio and video.

Bandwidth as Maximum Rate of Change in Digital Data Communication

- transmission of discrete 0s and 1s
- discrete pulses – discrete changes of voltages in baseband data transmission

The question with regard to bandwidth in this context is this: **How fast can the signal be changed (from a voltage of V to a voltage of –V and back again)?**

A baseband transmission system with a bandwidth of 5000 Hz: Every 1/5000th of a second this system can communicate two things. If one voltage represents a 0 and another represents a 1, then the system can transmit a 0 and a 1 every 1/5000th of a second. This means that it can transmit 10,000 bits every second.

Assume that a signal is sent with two possible signal levels and a bandwidth of b Hz. Then the data rate, d, in bits/s is

$$d = 2b$$

Instead of having one voltage represent 0 and the other represent 1, we could have one voltage represent 00, the second represent 01, the third represent 10, and the fourth represent 11.

Allowing more than two signal levels such that more than one bit can be communicated at a time is called ***multilevel coding.***

Assume that a signal is sent with k possible signal levels and a bandwidth of b Hz. Then the data rate, d, in bits/s is

$$d = 2b\log_2(k)$$

Bandwidth of a Signal in Terms of Frequency

- the frequency range for a particular signal, assuming that the signal is sent as a waveform
- the term ***width of a signal*** to avoid confusion with the other use of the term bandwidth

For a signal that can be represented as a periodic waveform, let f_{max} be the frequency of the highest-frequency component and let f_{min} be the frequency of the lowest-frequency component. Then the width of the signal, w, is

$$w = f_{max} - f_{min}$$

Bandwidth of a Communication Channel in Terms of Frequency

- When data is communicated across the airwaves, it is sent along some particular channel, which is a band of frequencies.
- The range of frequencies allocated to a band constitutes the ***bandwidth of a channel.***

- For example, each AM radio station is allocated a bandwidth of 10 kHz. FM radio stations have bandwidths of 200 kHz. Analog television has a bandwidth of about 6 MHz. Digital high definition television (HDTV) requires a bandwidth of approximately 20 MHz.

Data Rate/Bit Rate

- Bandwidth is measured in cycles per second – Hz

- Data rate is measured in bits per second –
  - kilobits per second (kb/s),
  - kilobytes per second (kB/s),
  - megabits per second (Mb/s),
  - megabytes per second (MB/s),
  - gigabits per second (Gb/s),
  - gigabytes per second (GB/s).

If measured in bits per second, *data rate* is synonymous with *bit rate.*

Data Rate/Bit Rate

Recall this:

Assume that a signal is sent with k possible signal levels and a bandwidth of b Hz. Then the data rate, d, in bits/s is $$d = 2b\log_2(k)$$

- ***Shannon's theorem***

$$c = b\log_2(1 + s/p)$$

s is a measure of the signal power, and p is a measure of the noise power.

Data transfer rates for common communication links

Wide Area Network	
Type of Data Connection	**Data Rate**
telephone modem	28.8-56 kb/s
ISDN (Integrated Services Digital Network)	64-128 kb/s
ADSL (Asymmetric Digital Subscriber Line)	1.544-8.448 Mb/s (downstream) 16-640 kb/s (upstream)
ADSL2	0.8-3.5 Mb/s up, 5-12 Mb/s down
ADSL2+	1-3.5 Mb/s up, 24 Mb/s down
VDSL (Very High Bit DSL)	12.96-55.2 Mb/s (~12 Mb/s down and 52 Mb/s up, or ~26 Mb/s symmetrical at 1000 feet, 10 Mb/s at 4000 feet)
Cable modem	20-40 Mb/s
VDSL2	50-250 Mb/s

Data transfer rates for common communication links

Local Area Network	
Type of Data Connection	**Data Rate**
Token ring	16 Mb/s
Ethernet (10base-X)	10 Mb/s
Fast ethernet (100base-X)	100 Mb/s
FDDI	100 Mb/s
Gigabit ethernet	1 Gb/s
Wireless 802.11b	11 Mb/s
Wireless 802.11g	54 Mb/s

Data transfer rates for common communication links

Computer Interfaces	
Type of Data Connection	**Data Rate**
Serial	10-230 kb/s
Parallel	8 Mb/s
SCSI 1	12 Mb/s
SCSI 2	80 Mb/s
Fast wide SCSI	160 Mb/s
SCSI (various ultra versions)	320-2560 Mb/s
USB, USB2	12-480 Mb/s
SDI (serial digital interface)	143-360 Mb/s
Firewire (IEEE 1394)	400-800 Mb/s
DMA ATA	264-1064 Mb/s

Audio Data Size

Digital Audio Sample Precision

- Typically 8 bits or 16 bits.
- Each bit will added about 6 dB of resolution, so 16 bits => 96 dB.
- Samples are typically stored as raw numbers (*linear format*), or as logarithms (µ-law (or *A-law* in Europe)).

Data rate & bandwidth in sample audio applications

Quality	Sample Rate (KHz)	Bits per Sample	Mono/ Stereo	Data Rate (Uncompressed)	Frequency Band
Telephone	8	8	Mono	8 KBytes/sec	200-3,400 Hz
AM Radio	11.025	8	Mono	11.0 KBytes/sec	
FM Radio	22.050	16	Stereo	88.2 KBytes/sec	
CD	44.1	16	Stereo	176.4 KBytes/sec	20-20,000 Hz
DAT	48	16	Stereo	192.0 KBytes/sec	20-20,000 Hz

How do we calculate uncompressed data rate based on given sample rate, quantization size and mono/stereo type?

kilobyte	kB
megabyte	MB
gigabyte	GB
kilobit	kb
megabit	Mb
gigabit	Gb
terabit	Tb
terabyte	TB

For memory and file sizes, assume the following equivalences:

1 byte = 8 bits

1 kB = 2^{10} bytes = 1024 bytes

1 MB = 2^{20} bytes = 1,048,576 bytes

1 GB = 2^{30} bytes = 1,073,741,824 bytes

1 TB = 2^{40} bytes = 1,099,511,627,776 bytes

Example digital audio file size (without compression):

Sampling rate: 44.1 kHz

Bit depth: 32 (16 for each channel)

Number of minutes: one

Total number of bits: 84,672,000 (10,584,000 bytes)

File size: 10.09 MB for 1 minute

Data rate of the file: 1.35 Mb/s

Exercise

a. Based on the information below, calculate the memory space require for 3 minutes and 2 second of song?

b. If the song file is compressed with the compression ratio of 2.5, calculate the size of compressed song file?

Quality	Sample Rate (KHz)	Bits per Sample	Mono/ Stereo	Data Rate (Uncompressed)	Frequency Band
CD	44.1	16	Stereo	176.4 KBytes/sec	20-20,000 Hz