1 Introduction

1 What is multimedia?

❖ *Multi-* means many; much; multiple

❖ *Medium* means:

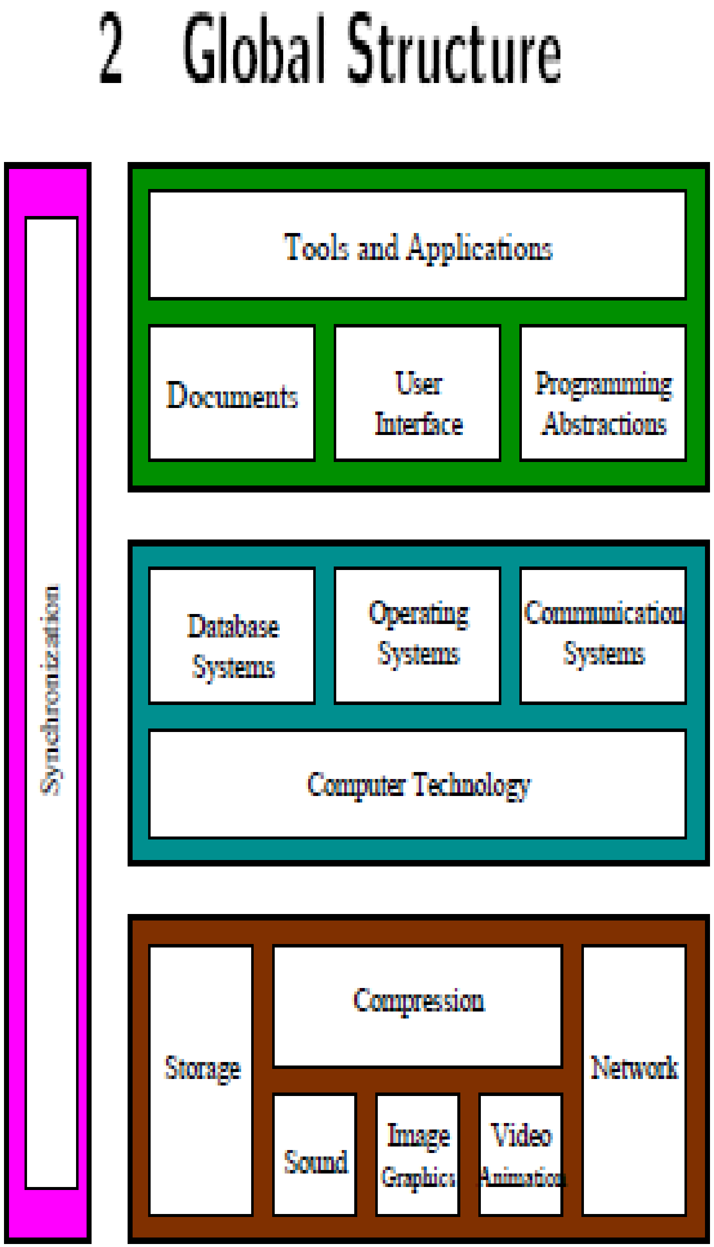
- An intervening substance through which something is transmitted or carried on
- A means of mass communication such as newspaper, magazine, or television

❖ *Multimedia* is woven combinations of text, graphic art, sound, animation, video and other kinds of elements.

❖ When a viewer of a multimedia presentation is allowed to control what elements are delivered and when, it is *interactive* multimedia.

❖Multimedia is an inter-disciplinary subject because it involves a variety of different theories and skills:

- these include computer technology, hardware and software;
- arts and design, literature, presentation skills;
- application domain knowledge



Application domain — provides functions to the user to develop and present multimedia projects. This includes *Software tools*, and multimedia projects *development methodology*.

System domain — including all supports for using the functions of the device domain, e.g., operating systems, communication systems (networking) and database systems.

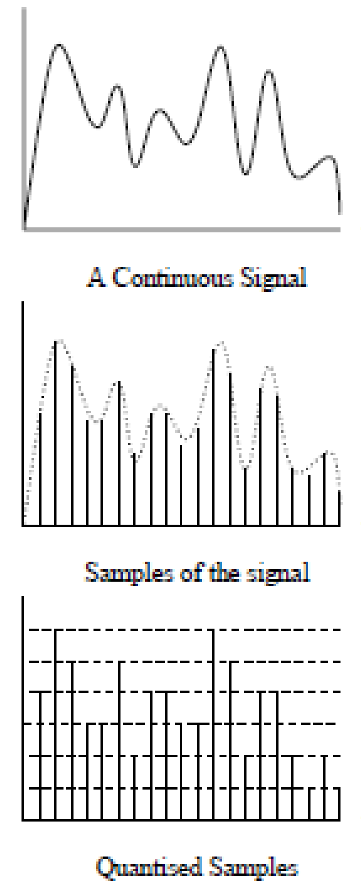
Device domain — basic concepts and skill for processing various multimedia elements and for handling physical device.

3 Digital Technology

Everyday, we encounter many values that change *continuously*, for example, the voltage of the electricity that lights up our room varies continuously over time. These are also known as *analogue* signals.

However, modern computers are built to deal with entities in completely different way. These are known as *digital* computers because they work with digits.

Because of this, when using a computer to process continuous signals,



we first need to find a way to represent them so that the computer is able to handle them. Usually, this is a *digital representation*, i.e., we use a series of numbers to denote the continuous signals.

Then, we have to convert the continuous signal into the digital representation. This process is known as *digitisation*.

The first step in the digitisation process is *sampling* which takes samples of the continuous signal. The number of samples taken during a time period is known as *sampling rate*.

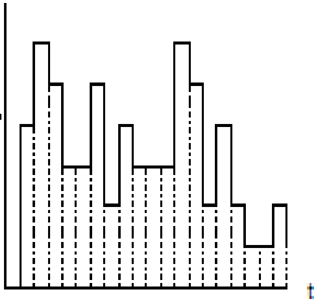
The second step is known as *quantisation* where we restrict the value of the samples to a fixed set of levels.

Sampling Rate

The unit of sampling rate is Hertz (Hz), i.e., 1Hz means taking one sample per second. For many signals, this is far too slow, therefore, we often use kHz, i.e., kiloHertz.

Because modern digital computers often store data in bytes, each sample is usually stored using either 8 bits (1 byte) or 16 bits (2 bytes). This corresponds to either 256 or 65536 levels for a digitised sample.

The device that we use to convert analogue signal to digital signal is known as an *analogue-to-digital* converter, ADC for short.



Sample and hold reconstruction

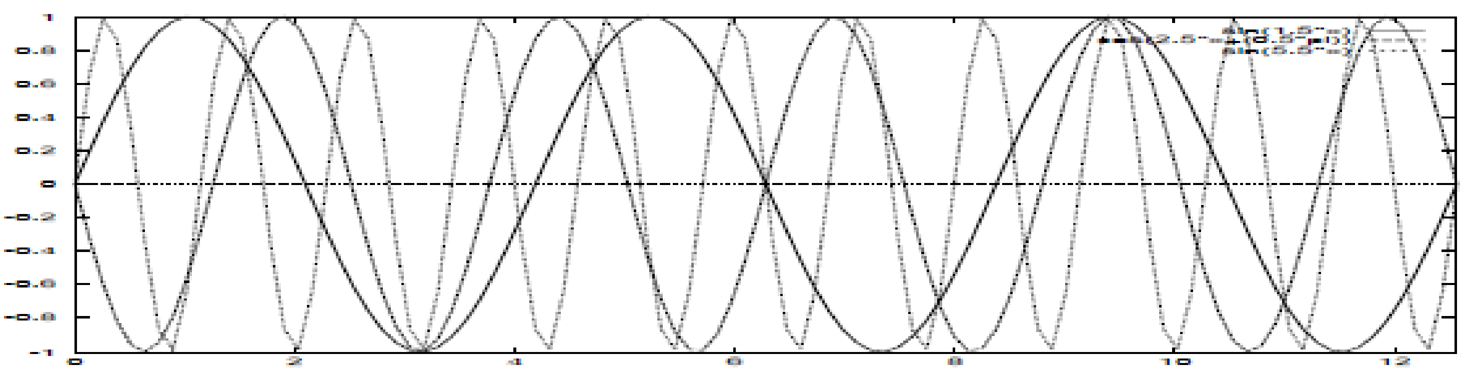
After being processed by the computer, the signal will be played back , i.e., we need to reconstruct the signal from the digital respresentation. One commonly used technique is known as *sample and hold*.

Clearly, if we want to reconstruct a signal that is as closed to the original signal as possible, we need to take sufficiently many samples, and we need to have as many levels to record the sample values in as possible.

Sampling Theorem

Nyquist sampling theorem tells us that, in order to reconstruct the signal, the sampling rate must not be less than twice the maximum frequency of the original signal. For example, if the maximum frequency is 3000Hz, the sampling rate must not be less than 6000Hz.

If we *undersample*, i.e., taking less samples than as required by Nyquist sampling theorem, some of the frequency components will be mistakenly converted into other frequencies. This is known as*aliasing*.



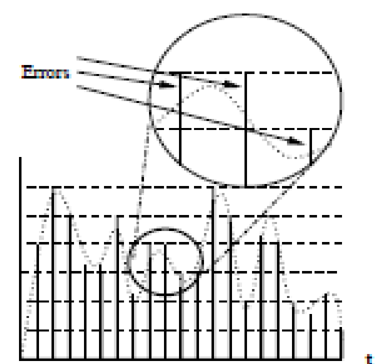
Quantisation Error

On the other hand, if we use too few levels to represent each sample value, there will be large amount of error for each sample.
This is known as *quantisaton error*. These errors can be thought of as noise on the signal.
We measure the quality of a sample by its *signal-to-noise ratio* (SNR). The higher the resolution, the smaller the noise, and the better the quality. The unit of SNR is dB (deci Bel). This is defined by

$$10 \log \frac{S}{N}$$

where S is the strength of the signal and N is the noise.

- For 8-bit samples, the SNR is $10 \log(256/0.5) \approx 48dB$.
- For 16-bit samples, the SNR is $10 \log(65536/0.5) \approx 96dB$.



Quatisation errors

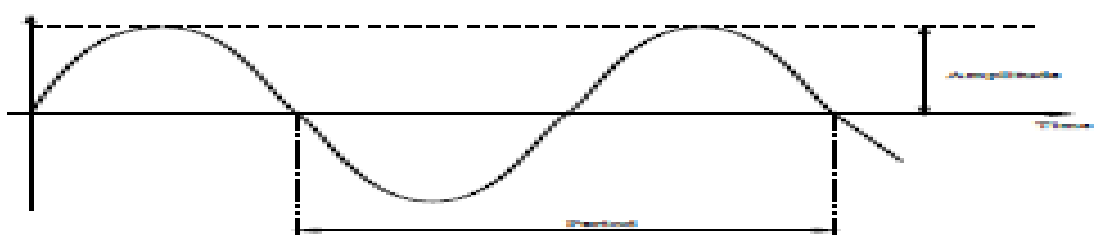
2 Sound/Audio

1 The Nature of Sound

Sound is a physical phenomenon produced by the vibration of matter and transmitted as waves.

However, the perception of sound by human beings is a very complex process. It involves three systems:

- the *source* which emits sound;
- the *medium* through which the sound propagates;
- the *detector* which receives and interprets the sound.



Sounds we heard everyday are very complex. Every sound is comprised of waves of many different frequencies and shapes. But the simplest sound we can hear is a sine wave.
Sound waves can be characterised by the following attributes:

Period	Frequency	Amplitude	Bandwidth
	Pitch	Loudness	Dynamic

1.1 Pitch and Frequency

Period is the interval at which a periodic signal repeats regularly.
Pitch is a perception of sound by human beings It measures how 'high' is the sound as it is perceived by a listener.
Frequency measures a physical property of a wave. It is the reciprocal value of period

$$f = \frac{1}{P}.$$

The unit is Herts (Hz) or kiloHertz (kHz).
Musical instruments are tuned to produce a set of fixed pitches.

Infra-sound	0 – 20 Hz
Human hearing range	20 – 20 kHz
Ultrasound	20 kHz – 1 GHz
Hypersound	1 GHz – 10 THz

Note	Ratio	Frequencies
C	1:1	264
D	9:8	297
E	5:4	330
F	4:3	352
G	3:2	396
A	5:3	**440**
B	15:8	495
C	2:1	528

1.2 Loudness and Amplitude

The other important perceptual quality is *loudness* or *volume*.

Amplitude is the measure of sound levels. For a digital sound, amplitude is the sample value.

The reason that sounds have different loudness is that they carry different amount of power.

The unit of power is watt. The intensity of sound is the amount of power transmitted through an area of $1m^2$ oriented perpendicular to the propagation direction of the sound.

If the intensity of a sound is $1watt/m^2$,

we may start feel the sound. T

damaged

This is known as the *threshold of feeling*. If the intensity is 10^{-12} watts/m^2 we may just be able to hear it. This is known as the *threshold of hearing*.

The relative intensity of two different sounds is measured using the unit *Bel* or more commonly *deciBel* (*dB*). It is defined by

$$\text{relative intensity in } dB = 10 \log \frac{I_2}{I_1}$$

Very often, we will compare a sound with the *threshold of hearing*.

Typical sound levels generated by various sources

160 dB	Jet engine
130 dB	Large orchestra at fortissimo
100 dB	Car on highway
70 dB	Voice conversation
50 dB	Quiet residential areas
30 dB	Very soft whisper
20 dB	Sound studio

Typical sound levels in music

Intensity ($watt/m^2$)	**Sound Level** dB	***Loudness***
1	120	*Threshold of feeling*
10^{-3}	90	*fff*
10^{-4}	80	*ff*
10^{-5}	70	*f*
10^{-6}	60	*mf*
10^{-7}	50	*p*
10^{-8}	40	*pp*
10^{-9}	30	*ppp*
10^{-12}	0	*Threshold of hearing*

1.3 Dynamic and Bandwidth

- *Dynamic range* means the change in sound levels. For example, a large orchestra can reach 130dB at its climax and drop to as low as 30dB at its softest, giving a range of 100dB.
- *Bandwidth* is the range of frequencies a device can produce or a human can hear.

FM radio	50Hz – 15kHz
AM radio	80Hz – 5kHz
CD player	20Hz – 20kHz
Sound Blaster 16 sound card	30Hz – 20kHz
Inexpensive microphone	80Hz – 12kHz
Telephone	300Hz – 3kHz
Children's ears	20Hz – 20kHz
Older ears	50Hz – 10kHz
Male voice	120Hz – 7kHz
Female voice	200Hz – 9kHz

2 Computer Representation of Sound

- Sound waves are continuous while computers are good at handling discrete numbers.
- In order to store a sound wave in a computer, samples of the wave are taken.
- Each sample is represented by a number, the ‘code’.
- This process is known as *digitisation*.
- This method of digitising sound is know as *pulse code modulation* (PCM).

Refer to Unit 1 for more information on digitisation.

- According to Nyquist sampling theorem, in order to capture all audible frequency components of a sound, i.e., up to 20KHZ, we need to set the sampling to at least twice of this. This is why one of the most popular sampling rate for high quality sound is 4410HZ.
- Another aspect we need to consider is the resolution, i.e., the number of bits used to represent a sample.

Often, 16 bits are used for each sample in high quality sound. This gives the SNR of 96dB.

2.1 Quality versus File Size

The size of a digital recording depends on the sampling rate, resolution and number of channels.

$$S = R \times (b/8) \times C \times D$$

S	file size	bytes
R	sampling rate	samples per second
b	resolution	bits
C	channels	1 - mono, 2 - stereo
D	recording duration	seconds

Higher sampling rate, higher resolution gives higher quality but bigger file size.

For example, if we record 10 seconds of stereo music at 44.1kHz, 16 bits, the size will be:

$$\begin{aligned} S &= 44100 \times (16/8) \times 2 \times 10 \\ &= 1,764,000\textbf{bytes} \\ &= 1722.7\textbf{Kbytes} \\ &= 1.68\textbf{Mbytes} \end{aligned}$$

Note: $\begin{aligned} 1\text{Kbytes} &= 1024\text{bytes} \\ 1\text{Mbytes} &= 1024\text{Kbytes} \end{aligned}$

High quality sound files are very big, however, the file size can be reduced by compression.

File size for some common sampling rates and resolutions

Sampling Rate	Resolution	Stereo /Mono	Size for for 1 Min.	Comments
44.1KHz	16-bit	Stereo	10.5MB	CD-quality recording
44.1KHz	16-bit	Mono	5.25MB	A good trade-off for high-quality recordings of mono sources such as voice-overs
44.1KHz	8-bit	Stereo	5.25MB	Achieves highest playback quality on low-end devices such as most of the sound cards
44.1KHz	8-bit	Mono	2.6MB	An appropriate trade-off for recording a mono source
22.05KHz	16-bit	Stereo	5.25MB	Darker sounding than CD-quality recording because of the lower sampling rate
22.05KHz	16-bit	Mono	2.5MB	Not a bad choice for speech, but better to trade some fidelity for a lot of disk space by dropping down to 8-bit
22.05KHz	8-bit	Stereo	2.6MB	A very popular choice for reasonable stereo recording where full bandwidth playback is not possible
22.05KHz	8-bit	Mono	1.3MB	A thinner sound than the choice just above, but very usable
11KHz	8-bit	Stereo	1.3MB	At this low a sampling rate, there are few advantages to using stereo
11KHz	8-bit	Mono	650K	In practice, probably as low as you can go and still get usable results
5.5KHz	8-bit	Stereo	650K	Stereo not effective
5.5KHz	8-bit	Mono	325K	About as good as a bad telephone connection

2.2 Audio File Formats

- The most commonly used digital sound format in Windows systems is
  .wav files.
- Sound is stored in .wav as digital samples known as *Pulse Code Modulation*(PCM).
- Each .wav file has a header containing information of the file.
  - type of format, e.g., PCM or other modulations
  - size of the data
  - number of channels
  - samples per second
  - bytes per sample
- There is usually no compression in .wav files.

Other format may use different compression technique to reduce file size.

- .vox use *Adaptive Delta Pulse Code Modulation* (ADPCM).
- .mp3 MPEG-1 layer 3 audio.
- RealAudio file is a proprietary format.

Some common audio files formats

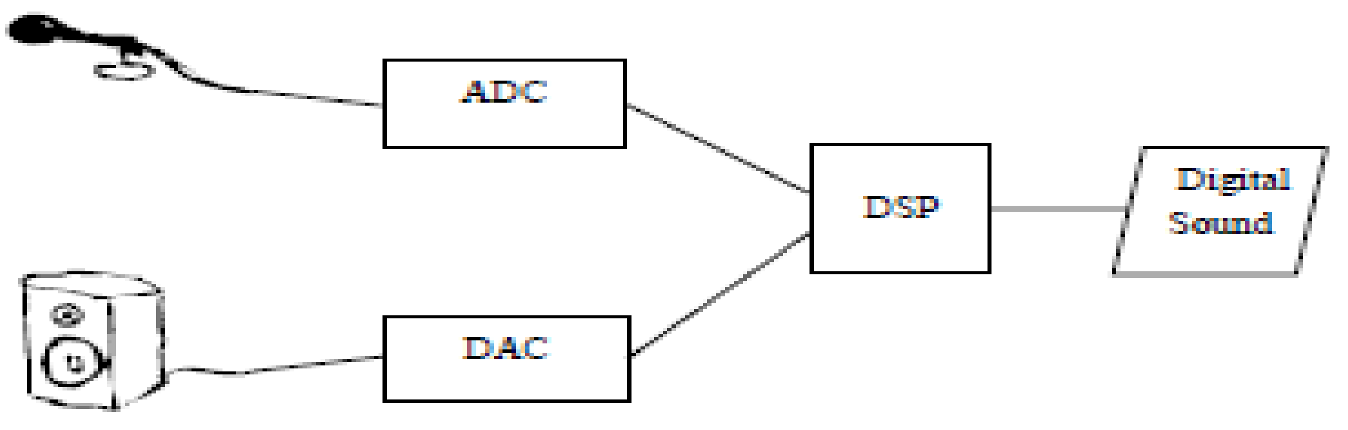
Extension	MIME Type	Platform	Use
aif	Audio/x-aiff	Mac, SGI	Audio
aifc	Audio/x-aiff	Mac, SGI	Audio (compressed)
AIFF	Audio/x-aiff	Mac, SGI	Audio
aiff	Audio/x-aiff	Mac, SGI	Audio
au	Audio/basic	Sun, NeXT	ULAW audio data
mov	Video/QuickTime	Mac, Win	QuickTime video
mpe	Video/mpeg	All	MPEG video
mpeg	Video/mpeg	All	MPEG video
mpg	Video/mpeg	All	MPEG video
mp3	Audio/x-mpeg	All	MPEG audio
qt	Video/QuickTime	Mac, Win	QuickTime video
ra,ram	Audio/x-pn-realaudio	All	RealAudio Sound
snd	Audio/basic	Sun, NeXT	ULAW Audio Data
vox	Audio/	All	VoxWare Voice
wav	Audio/x-wav	Win	WAV Audio

2.3 Audio Hardware

- Recording and Digitising sound:
  - An analog-to-digital converter(ADC) converts the analog sound signal into digital samples.
  - A digital signal processor(DSP) processes the sample, e.g. filtering, modulation, compression, and so on.
- Play back sound:
  - A digital signal processor processes the sample, e.g. decompression, demodulation, and so on
  - An digital-to-analog converter(DAC) converts the digital samples into sound signal
- All these hardware devices are integrated into a few chips on a sound card

•Different sound card have different capability of processing digital sounds. When buying a sound card, you should look at:

- maximum sampling rate
- stereo or mono
- duplex or simplex



2.4 Audio Software

•Windows device driver — controls the hardware device.
Many popular sound cards are Plus and Play. Windows has drivers for them and can recognise them automatically. For cards that Windows does not have drivers, you need to get the driver from the manufacturer and install it with the card.

•If you do not hear sound, you should check the settings, such as interrupt, DMA channels, and so on.

•Device manager — the user interface to the hardware for configuring the devices.

- •You can choose which audio device you want to use
- •You can set the audio volume

Mixer — its functions are:

- to combine sound from different sources
- to adjust the play back volume of sound sources
- to adjust the recording volume of sound sources

Recording — Windows has a simple Sound Recorder.
Editing — The Windows Sound Recorder has a limiting editing function, such as changing volume and speed, deleting part of the sound.

There are many freeware and shareware programs for sound recording, editing and processing.