UNIT 1

Introduction

Formatted on September 4, 2000

Objectives

- To provide an overview to the subject
- To describe a number of fundamental concepts

Contents

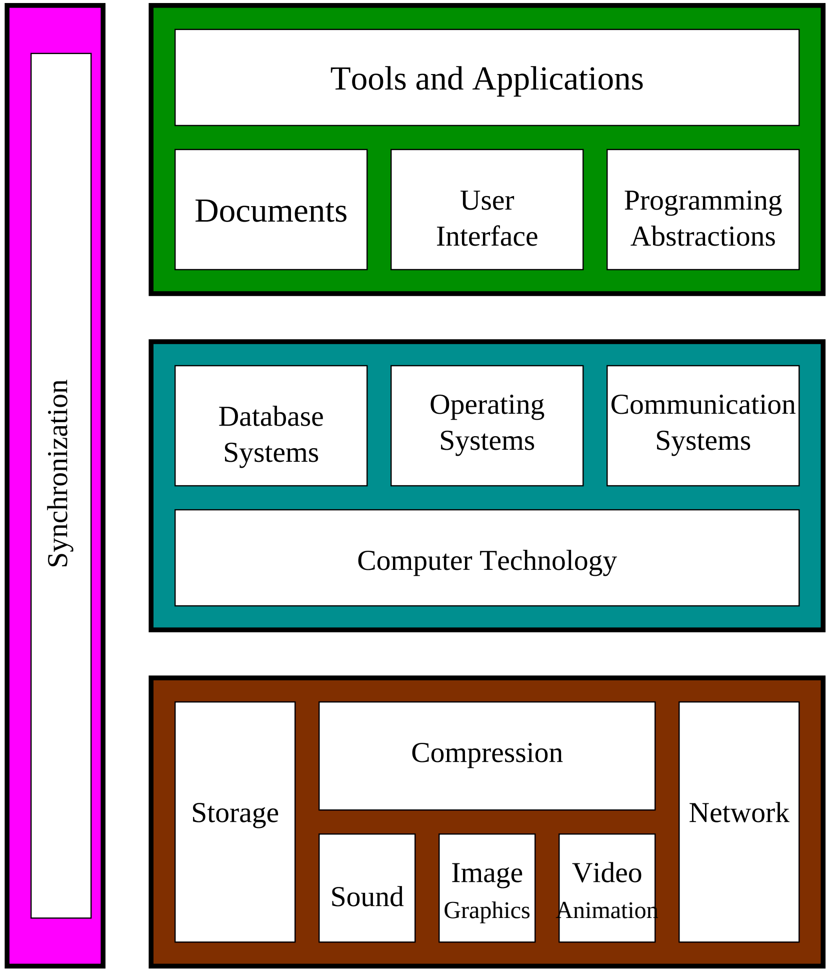
1 What is multimedia?

- *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

from American Heritage Electronic Dictionary 1991

- *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.

2 Global Structure



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.

We can roughly divide the lectures into the following parts:

Multimedia Elements which deals with the various properties of each kinds of elements:

- Sound, audio, voice and music (See Unit 2)
- Graphics, photographs and images(See Unit 3)
- Text, and layout(See Unit 4)
- Full-motion video and animation(See Unit 5)

Application domain which deals how to develop multimedia applications:

- Multimedia application development method(See Unit 6)
- Interface design and Software tools — element processing tools, authoring tools(See Unit 7)

Supporting technology which deals the technology that are needed to support multimedia applications, such as:

- Data Compression
- Data and file format
- Multimedia input, output and storage

Multimedia and the Internet (See Unit 12)

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.

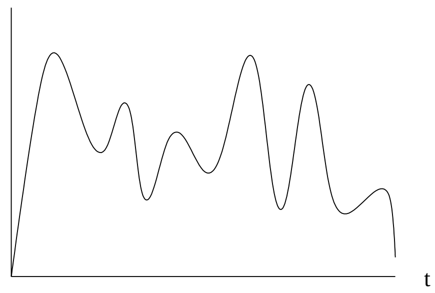
However, modern computers are built to deal with entities in completely different way. These are known as *digital* computers because they work with digits.

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.

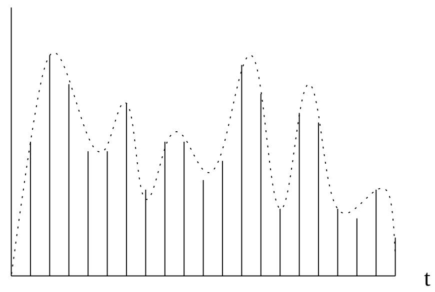
Then, we have to convert the continuous signal into the digital representation. This process is known as *digitisation*.

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*.

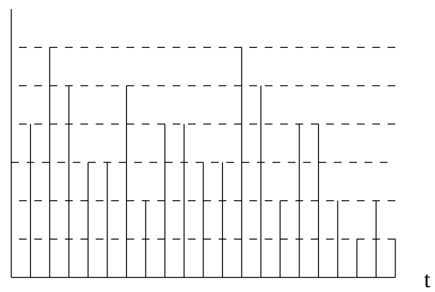
The second step is known as *quantisation* where we restrict the value of the samples to a fixed set of levels.



A Continuous Signal



Samples of the signal



Quantised Samples

Sampling Rate

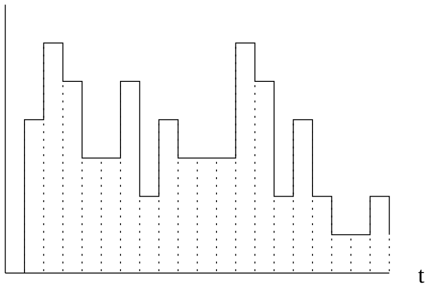
The unit of sampling rate is Hertz (Hz), i.e., 1Hz means taking one sample per second. For many signals, this is far too slow, therefore, we often use kHz, i.e., kiloHertz.

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.

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*.

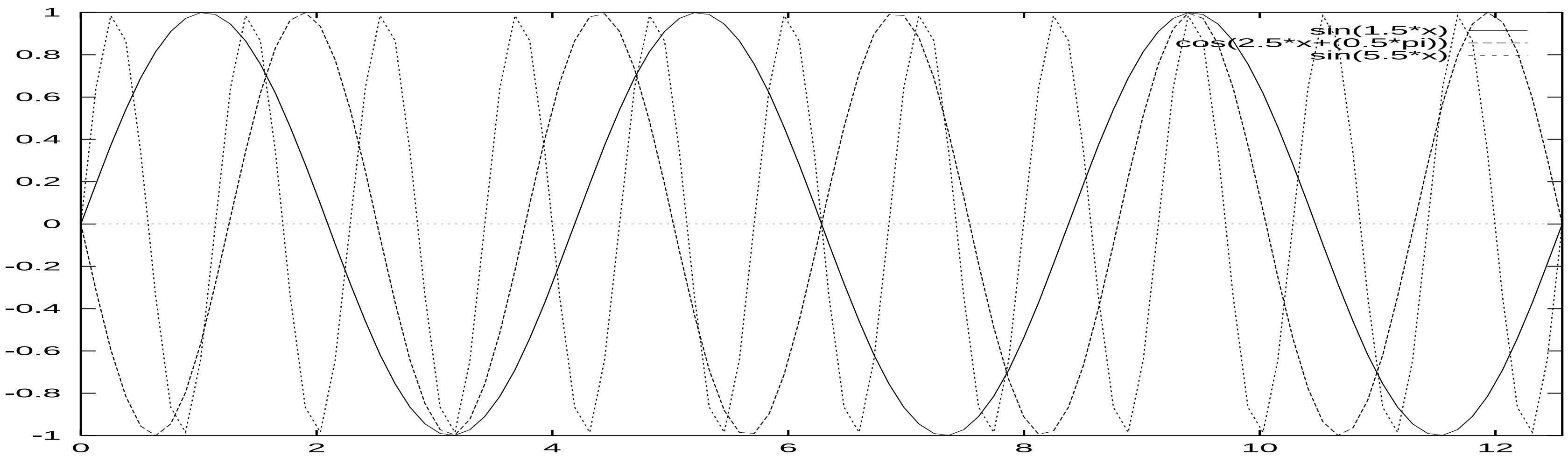
Clearly, if we want to reconstruct a signal that is as closed to the original signal as possible, we need to take sufficiently many samples, and we need to have as many levels to record the sample values in as possible.



Sample and hold reconstruction

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

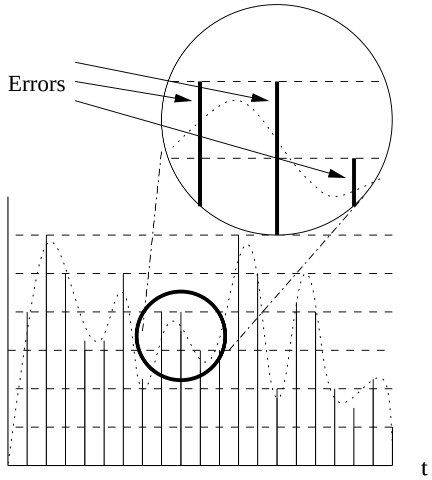
On the other hand, if we use too few levels to represent each sample value, there will be large amount of error for each sample. This is known as *quantisaton error*. These errors can be thought of as noise on the signal.

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