multimedia notes 2

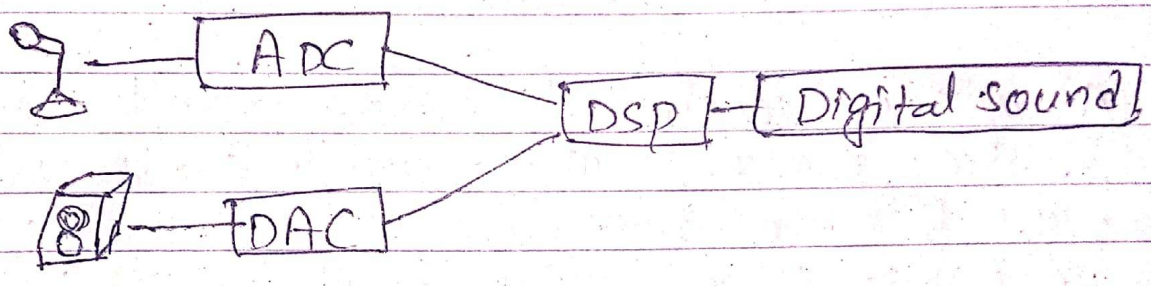
Multimedia Computing (Tribhuvan Vishwavidalaya)

(10 marks)

Chapter: 2: Sound / Audio System

2.1: Audio hardware and softwares

Audio hardware:



A) 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, eg: filtering, modulation, Compression and so on.

B) play back sound
- A digital signal processor processes the sample eg: decompression, demodulation.
- A 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. so, sound card with maximum sampling rate, stereo or mono and duplex or simplex should be choosen.

Audio Software

A) window device driver:- controls the hardware device. Many popular sound cards are plus and play. Windows has drivers for them and can recognize them automatically.

B) Device Manager: the user interface to the hardware for configuring the devices.

c) Mixer: It's functions are',
   a) to combine sound from different source.
   b) to adjust the play back volume of sound sources.
   c) to adjust the recording volume of sound sources.

d) Recording: Windows has a simple Sound recorder software.

e) Editing: There are many software programs for recording, editing and processing of audio.
   - windows also have a editing software.

2.2. Differences.

S.N.	MIDI	Digital Audio
1)	A MIDI file is a software for representing musical ~~instrume~~ information in a digital format.	A digital audio refers to the reproduction and transmission of sound stored in a digital format.
2)	Donot Contain a recording of sound.	Contain a recording of sound.
3)	No actual sound stored in MIDI file.	Actual sound stored in digital audio file.
4)	Advantages: a) Files are tiny often less than 10kb. b) Downloaded from a webpage in no time. c) Fit easily on floppy disk.	Advantages: a) They reproduce the exact sound files. b) It reproduce better than CD quality.
5)	Disadvantages: a) They sound little different from the original sounds.	Disadvantages: a) They takes high memory. b) Download time is high. c) When combined with video, the files can cause problems.

2.3. Differences.

MIDI	WAV formats.
1) A MIDI file is software for representing musical information in a digital format.	1) A WAV file is an audio file format, created by microsoft that has become a standard for pcs in everything from system and games to CD-quality audio.
1) It stands for Musical Instrument Digital Interface.	2) It stands for Waveform Audio file format.
3) File name exist in .MIDI or .MID format.	3) File name exist in .WAV format
4) File format type ~~is~~ is compressed.	4) File format type is uncompressed.
5) Size of file is very small in comparison to WAV.	5) Size of file is very large in campanison to MIDI
6)-Quality of sound totally depends on sound card's synthesizer so, ~~its~~ quality may be different on different devices.	6) Quality of sound depends on the sampling rate and will not be different on different devices.

2.4. Calculate the file size in bytes for a 20 seconds recording at 44.1 KHZ, 16 bits resolution stereo sound.

solution:

Formula:

Sound file size = Sample rate x sample size x channel x duration

Note:
- Sample size:
  - 8bit = 1
  - 16 bit = 2
- Channel:
  - Mono = 1
  - stereo = 2

given,

duration = 20 seconds

Sample rate = 44.1 KHZ
= 44.1 x 1000 HZ
= 44100 HZ

Sample size = 2 (for 16 bit)

channel = 2 (for stereo sound)

we have,

So file size = Sample rate x sample size x channel x duration
= 44100 x 2 x 2 x 20
= 3528000 bytes.

2.5. Computer representation of sound:

The smooth, continuous curve of a sound waveform is not directly represented in a Computer. A computer measures the amplitude of the waveform at regular time interval to produce a series of numbers. Each of these measurements are samples.

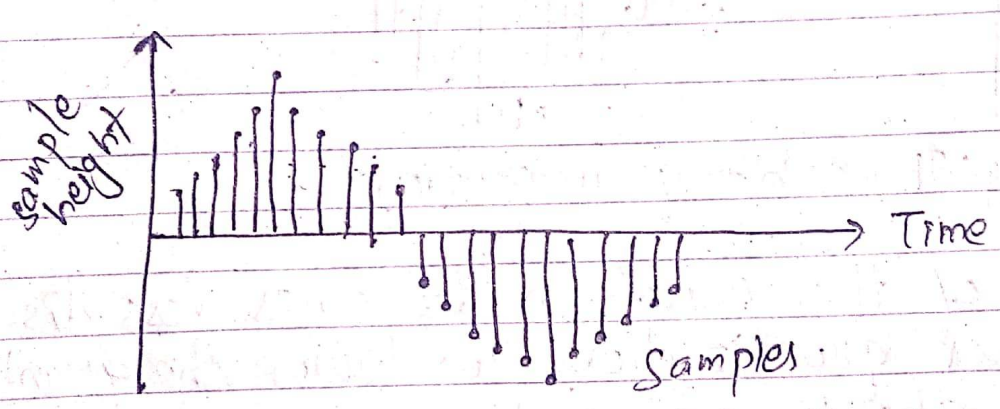


Fig:. Sampled waveform

This mechanism of converting audio signal into digital samples is called digitization.
The method of digitizing sound is known as pulse code modulation (PCM).

- Acc. to Nyquist Sampling theorem, in order to capture all audible frequency components of sound, ie. up to 20 kHz, we need to set the sampling to at least twice of this.
- Another aspect we need to consider is the resolution. often 16 bits are used for

each sample in high quality sound.
Resolution is also called the quantization of Sound.

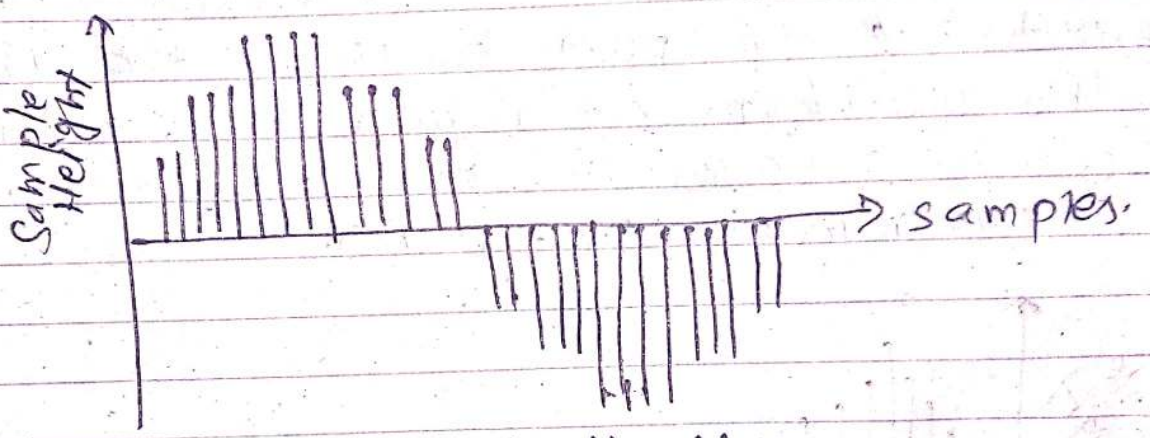


fig: Three-bit quantization

The shape of the waveform becomes less discernible with a lower quantization, i.e. lower the quantization lower the quality of sound.