Multimedia Data Representation

Issues to be covered:

- Digital Audio
  - Sampling Theorem
  - Digital Audio Signal Processing/Audio Effects
  - Digital Audio Synthesis
  - MIDI — Synthesis and Compression Control
- Graphics/Image Formats
  - Colour Representation/Human Colour Perception
- Digital Video
  - Chroma Subsampling

General Themes across all above

- Sampling/Digitisation
– Sampling Artifacts — Aliasing
- Compression requirements
– Data formats especially size
– Human Perception!compression ideas
- Compression Algorithms

Digital Audio

What is Sound?

Source — Generates Sound

- Air Pressure changes
- Electrical — Loud Speaker
- Acoustic — Direct Pressure Variations

Destination — Receives Sound

- Electrical — Microphone produces electric signal
- Ears — Responds to pressure hear sound (MPEG Audio)

Digitising Sound

- Microphone produces analog signal
- Computer like discrete entities

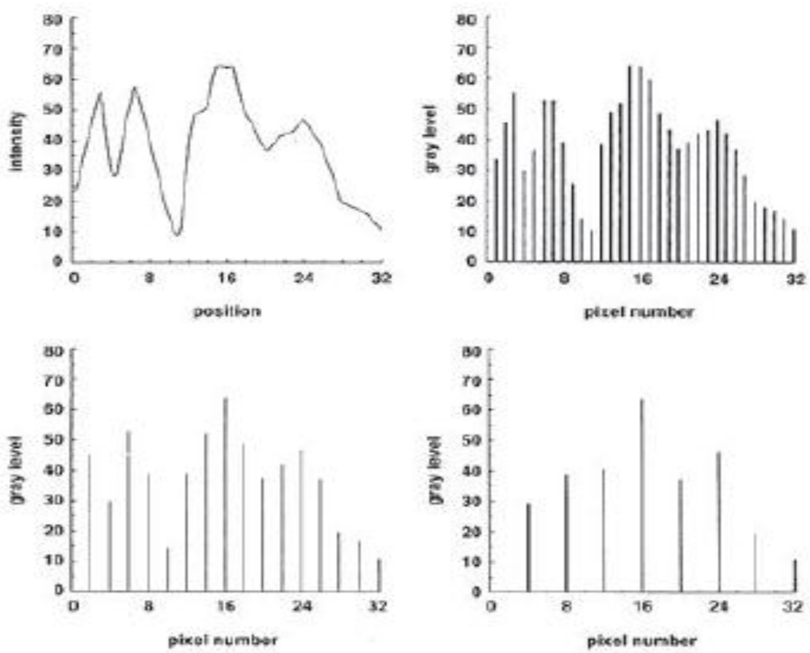
Need to convert Analog-to-Digital — Specialised Hardware

Also known as Sampling

Digital Sampling

Sampling basically involves:

- Measuring the analog signal at regular discrete intervals
- Recording the value at these points



Sample Rates and Bit Size

•How do we store each sample value (Quantisation)?
8 Bit Value (0-255)
16 Bit Value (Integer) (0-65535)

•How many Samples to take?
11.025 KHz — Speech (Telephone 8 KHz)
22.05 KHz — Low Grade Audio
(WWW Audio, AM Radio)
44.1 KHz — CD Quality

Nyquist’s Sampling Theorem
Sampling Frequency is very important in order to accurately reproduce a digital version of an Analog Waveform.

Nyquist’s Theorem:
The Sampling frequency for a signal must be at least twice the highest frequency component in the signal.

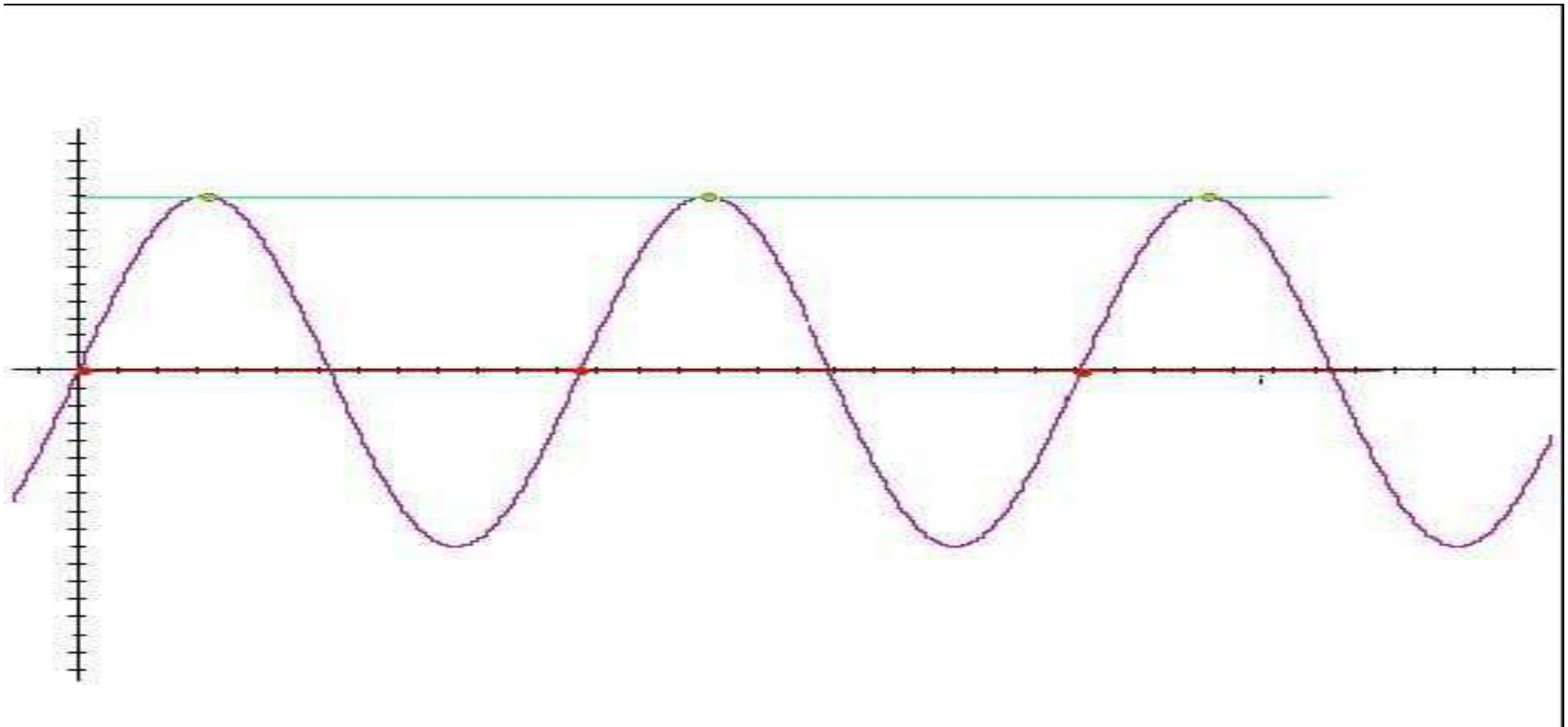


Figure 1: Sampling at Signal Frequency

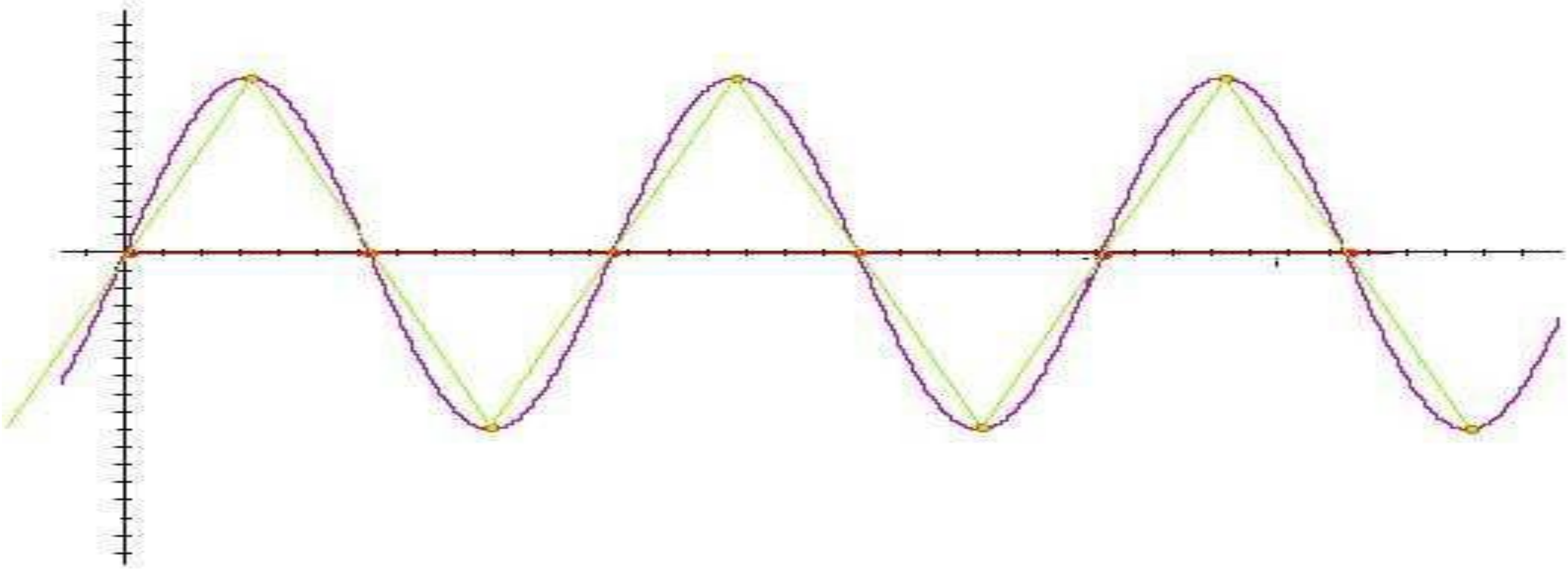


Figure 2: Sampling at Twice Nyquist Frequency

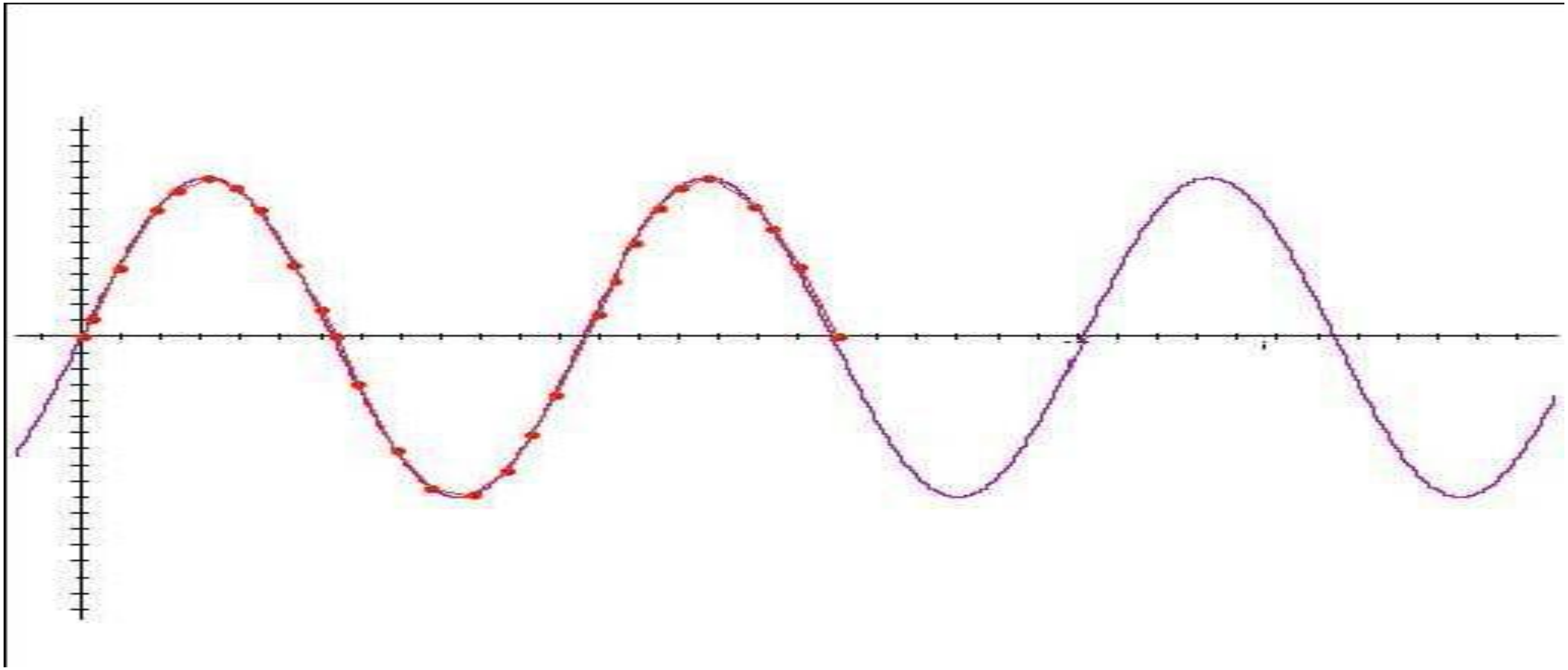


Figure 3: Sampling at above Nyquist Frequency

Implications of Sample Rate and Bit Size

Affects Quality of Audio

- Ears do not respond to sound in a linear fashion (MPEG Audio)
- Decibel (**dB) a logarithmic measurement of sound**
- 16-Bit has a signal-to-noise ratio of 98 dB — virtually inaudible
- 8-bit has a signal-to-noise ratio of 50 dB
- Therefore, 8-bit is roughly 8 times as noisy

– 6 dB increment is twice as loud

- Click Here to Hear Sound Examples

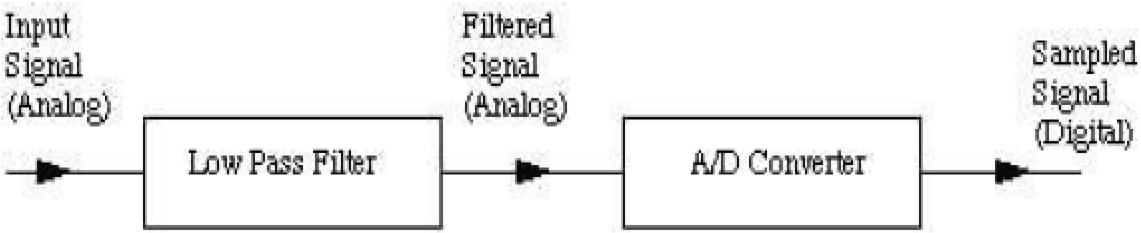
Implications of Sample Rate and Bit Size (cont)

Affects Size of Data

File Type	*44.1 KHz*	*22.05 KHz*	*11.025 KHz*
16 Bit Stereo	10.1 Mb	5.05 Mb	2.52 Mb
16 Bit Mono	5.05 Mb	2.52 Mb	1.26 Mb
8 Bit Mono	2.52 Mb	1.26 Mb	630 Kb

Practical Implications of Nyquist Sampling Theory

Must low pass filter signal before sampling:



Analog low pass filter used as signal is not yet digitised. Otherwise strange artifacts from high frequency signals would appear — Aliasing

Why are CD Sample Rates 44.1 KHz?

Upper range of human hearing is around 20-22 KHz — Apply Nyquist Theorem

Emerging Theme throughout this course:

- Perceptual traits of Human Auditory system.
- Don’t both recording data above 22 KHz
- So filter them out — low pass filter.

Common Audio Formats

- Popular audio file formats include

– .au (Origin: Unix, Sun),

– .aiff (MAC, SGI),

– .wav (PC, DEC)

- Compression can be utilised in some of the above but is not Mandatory
- A simple and widely used (by above) audio compression method is Adaptive Delta Pulse Code Modulation (ADPCM).

– Based on past samples, it predicts the next sample and encodes the difference between the actual value and the predicted value.

Common Audio Formats (Cont.)

- Many formats linked to audio applications
- Most use some compression
- Common ones:

– Sounblaster — .voc (Can use Silence Deletion (More on this later (Audio Compression))

– Protools/Sound Designer – .sd2

– Realaudio — .ra.

– Ogg Vorbis — .ogg

– AAC , Apple, mp4 — More Later

– Flac — .flac, More Later

– Dolby AC coding — More Later

- MPEG AUDIO— More Later (MPEG-3 and MPEG-4)

Synthetic Sounds —reducing bandwidth?

- Synthesise sounds — hardware or software
- Client produces sound — only send parameters to control sound (MIDI/MP4 later)
- Many synthesis techniques could be used, For example:

– FM (Frequency Modulation) Synthesis – used in low-end Sound Blaster cards, OPL-4 chip, Yamaha DX Synthesiser range popular in Early 1980’s.

– Wavetable synthesis – wavetable generated from sampled sound waves of real instruments

– Additive synthesis — make up signal from smaller simpler waveforms

– Subtractive synthesis—modify a (complex) waveform but taking out (Filtering) elements
– Granular Synthesis — use small fragments of existing samples to make new sounds
– Physical Modelling — model how acoustic sound in generated in software
– Sample-based synthesis—record and play back recorded audio, often small fragments and audion processed.
• Most modern Synthesisers use a mixture of samples and synthesis.