Multimedia Data Representation

Issues to be covered (Over next few lectures):

- 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 $\longrightarrow$ compression ideas

Building up to full Multimedia Compression Algorithms — following lectures

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 (**more later (MPEG Audio)**)

Digitising Sound (Recap from CM0268)

- **Microphone produces *analog* signal**
- Computer like discrete entities

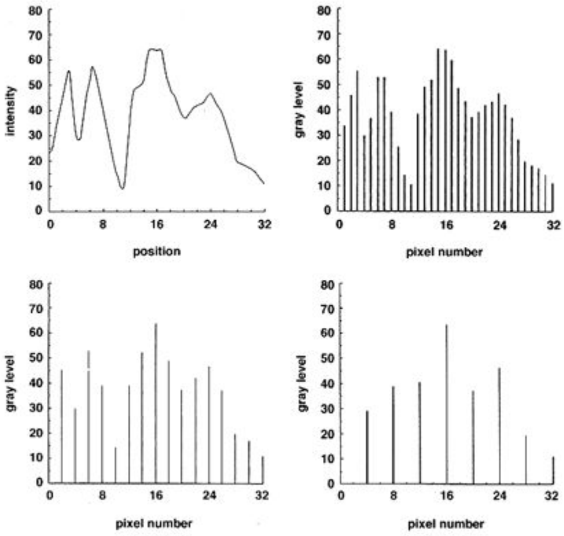
Need to convert Analog-to-Digital — Specialised Hardware

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.

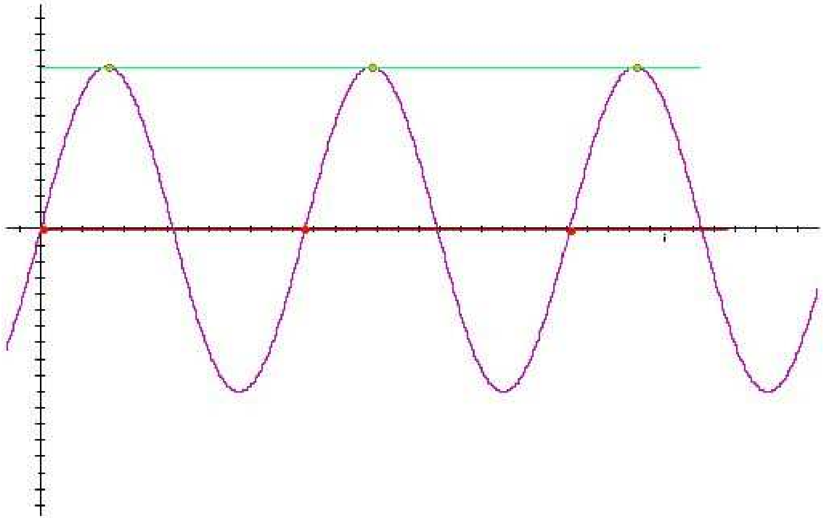


Figure 1: Sampling at Signal Frequency

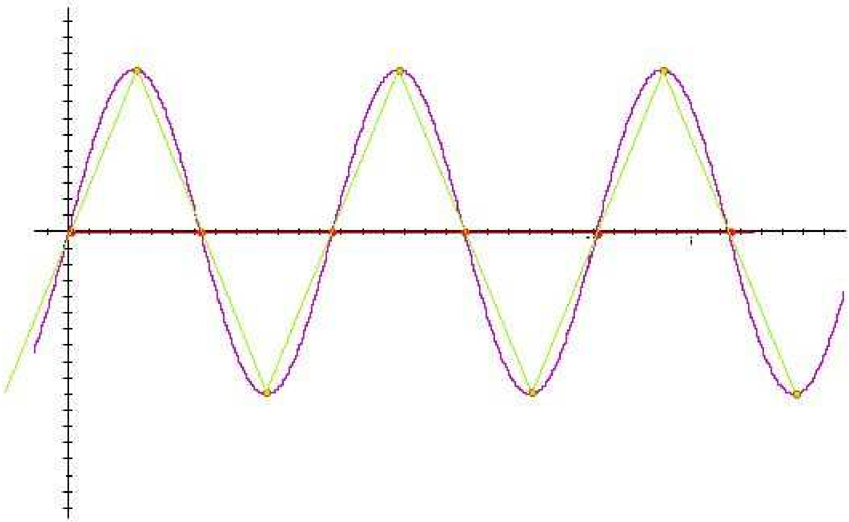


Figure 2: Sampling at Twice Nyquist Frequency

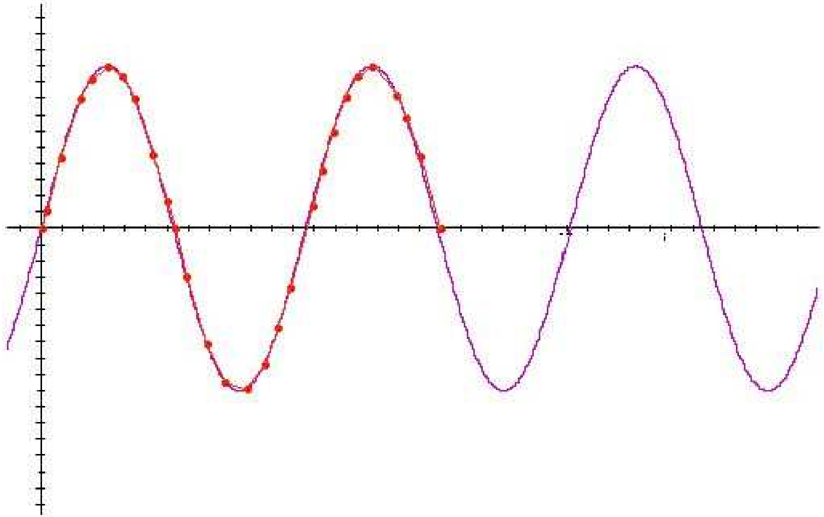


Figure 3: Sampling at above Nyquist Frequency

Implications of Sample Rate and Bit Size

Affects Quality of Audio

- Ears do not respond to sound in a linear fashion ((more later (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

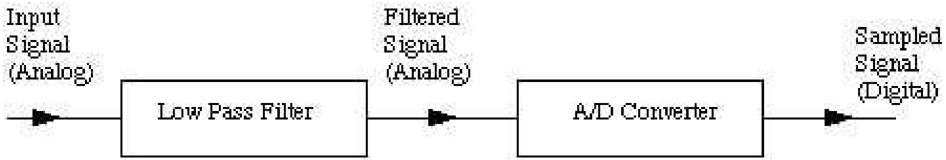
Implications of Sample Rate and Bit Size (cont)

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.
  - More on this later (Audio Compression)

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.

More on how synthesis works soon