CM3106 Chapter 4: Introduction to Digital Audio

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What is Sound? (**Recap from CM2202**)

Sound Generation

Source — Generates Sound

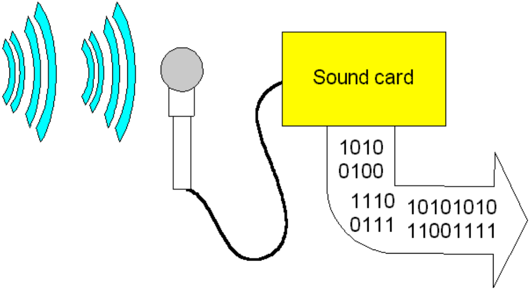
- Air Pressure changes
- *Electrical* — Loud Speaker
- *Acoustic* — Direct Pressure Variations

Sound Reception

Destination — Receives Sound

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

Digitising Sound



- Microphone:
  - Receives sound
  - Converts to **analog signal**.
- Computer like **discrete entities**

Need to convert Analog-to-Digital — Dedicated Hardware (*e.g.* Soundcard)

Also known as Digital Sampling

Sample Rates and Bit Size

Bit Size — Quantisation

How do we store each sample value (**Quantisation**)?

8 Bit Value (0-255)

16 Bit Value (Integer) (0-65535)

Sample Rate

How many Samples to take?

11.025 KHz — Speech (Telephone 8 KHz)

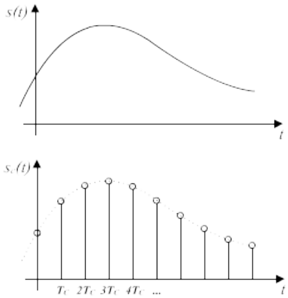
22.05 KHz — Low Grade Audio
(WWW Audio, AM Radio)

44.1 KHz — CD Quality

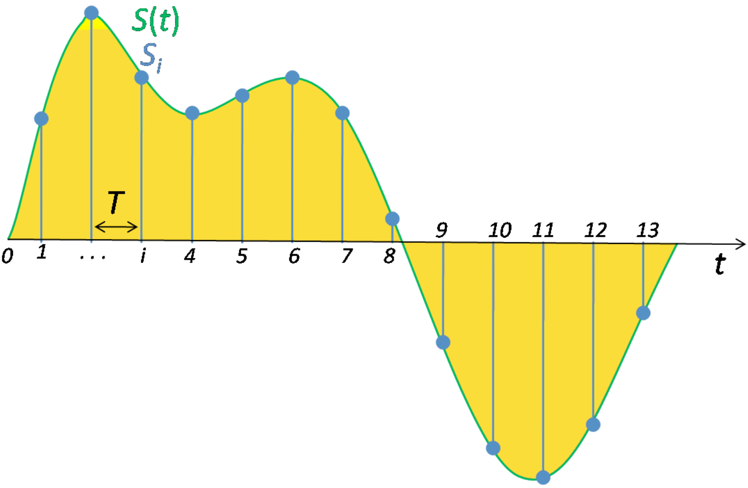
Digital Sampling (1)

Sampling process basically involves:

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



Digital Sampling (2)



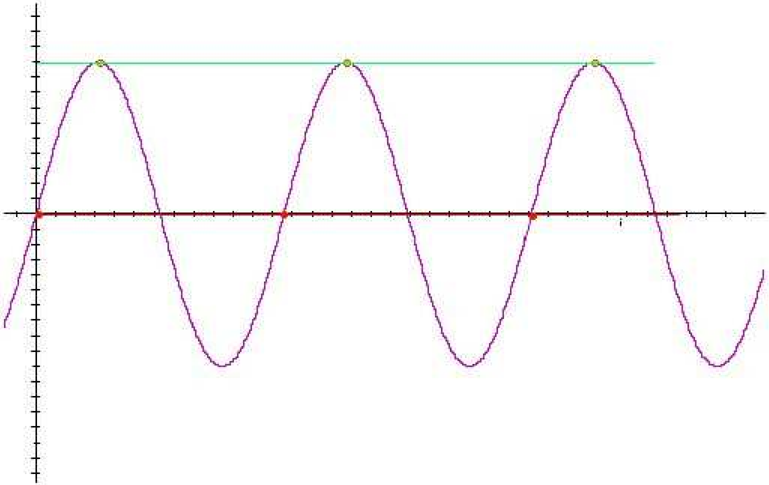
Nyquist's Sampling Theorem

The **Sampling Frequency** is **critical** to the **accurate reproduction** of a **digital version** of an analog waveform

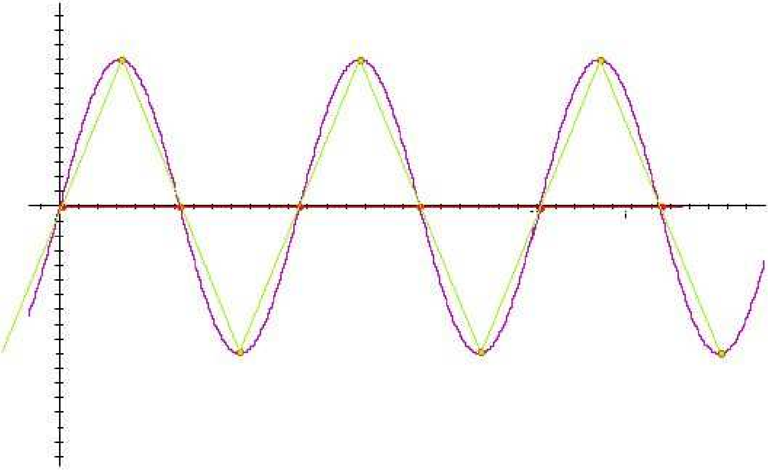
Nyquist's Sampling Theorem

The Sampling frequency for a signal must be **at least twice** the **highest frequency component** in the signal.

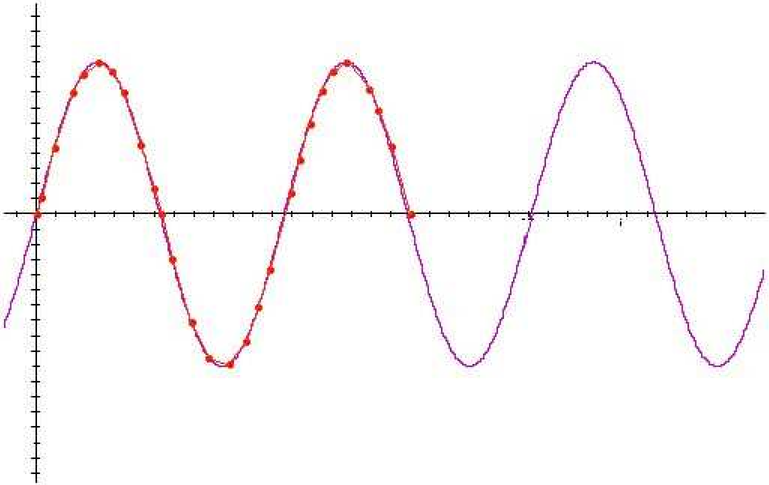
Sampling at Signal Frequency



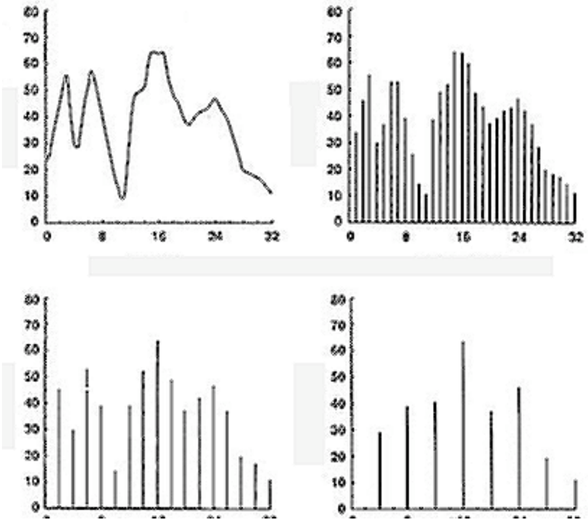
Sampling at Twice Nyquist Frequency



Sampling at above Nyquist Frequency



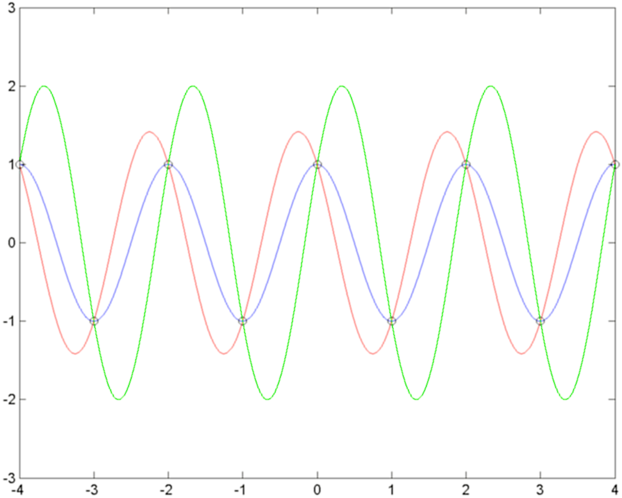
If you get Nyquist Sampling Wrong? (1)



Digital Sampling Artefacts Arise — Effect known as **Aliasing**

Affects Audio, Imagery and Video

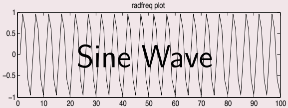
If you get Nyquist Sampling Wrong? (2)

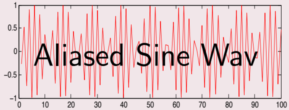


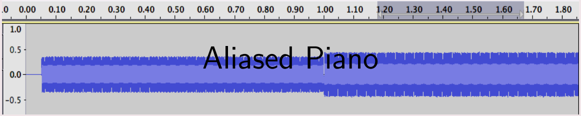
If you get Nyquist Sampling Wrong? (3)

What does aliasing sound like?

(**Click on Images to play sounds**)







MATLAB Code for Sine Demos above: Plot Version , Audio Version

More on image and video sampling artefacts later.

Implications of Sample Rate and Bit Size (1)

Affects Quality of Audio

- Ears do not respond to sound in a linear fashion
- 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

Implications of Sample Rate and Bit Size (2)

Audio Sample Rate and Bit Size Examples

File Type	Audio File (all mono)
44Hz 16 bit	
44KHz 8-bit	
22 KHz 16-bit	
22KHz 8-Bit	
11KHz 8-bit	

Web Link:
Click Here to Hear Sound Examples

Implications of Sample Rate and Bit Size (2)

Affects Size of Data

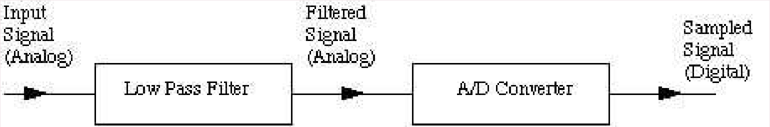
File Type	*44.1 KHz*	*22.05 KHz*	*11.025 KHz*
16 Bit Stereo	10.1 Mb	5.05 Mb	2.52 Mb
16 Bit Mono	5.05 Mb	2.52 Mb	1.26 Mb
8 Bit Mono	2.52 Mb	1.26 Mb	630 Kb

Memory Required for **1 Minute** of Digital Audio

Practical Implications of Nyquist Sampling Theory

Filtering of Signal

- Must (low pass) filter signal before sampling:



- Otherwise **strange artefacts** from high frequency (**above** Nyquist Limit) signals would appear in the sampled signal.

Why are CD Sample Rates 44.1 KHz?

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Why are CD Sample Rates **44.1 KHz**?

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

Common Digital 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 (MP3 and MPEG-4)**

Synthetic Sounds — reducing bandwidth?

Synthesis Pipeline

- Synthesise sounds — hardware or software (**more later**)
- Client produces sound — only send parameters to control sound (**MIDI/MP4/HTML5 later**)

Synthesis Methods (More Later)

- **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 audio processed.

Most modern Synthesisers use a mixture of sample and synthesis

Synthetic Sounds — Analogies with Vector Graphics

- Use more high-level descriptions to represent signals.
- Recorded sounds and digital images: regular sampling; large data size; difficult to modify
- Synthetic sounds and vector graphics: high level descriptions; small data size; easier to edit. **Conversion is needed before display – synthesis or rasterisation**
- Difference: 1D vs 2D

MULTIMEDIA

Module No: CM3106

More on how sound synthesis works soon