

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)**)



Back

Close

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*



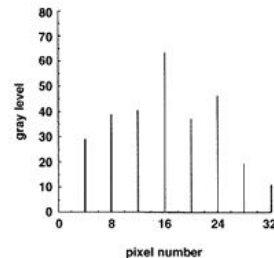
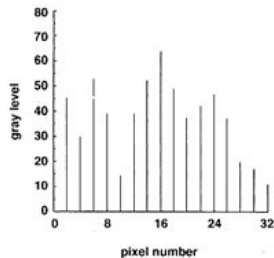
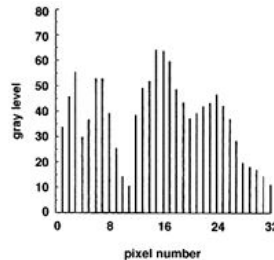
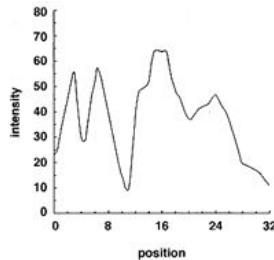
Back

Close

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



Back

Close

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.



Back

Close

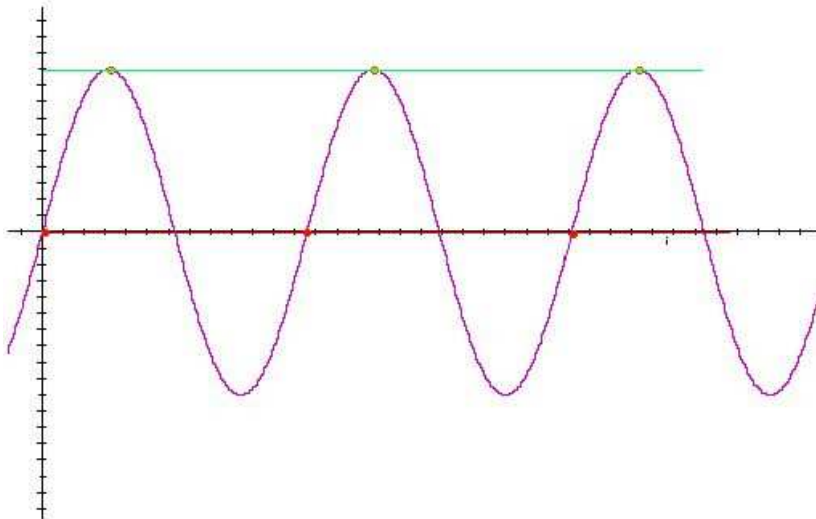


Figure 1: Sampling at Signal Frequency



Back

Close

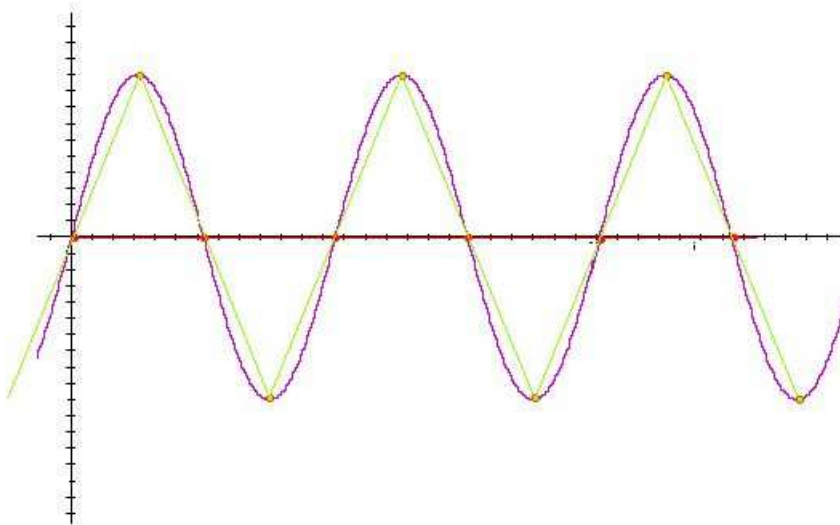


Figure 2: Sampling at Nyquist Frequency (Twice Signal Frequency)



Back

Close

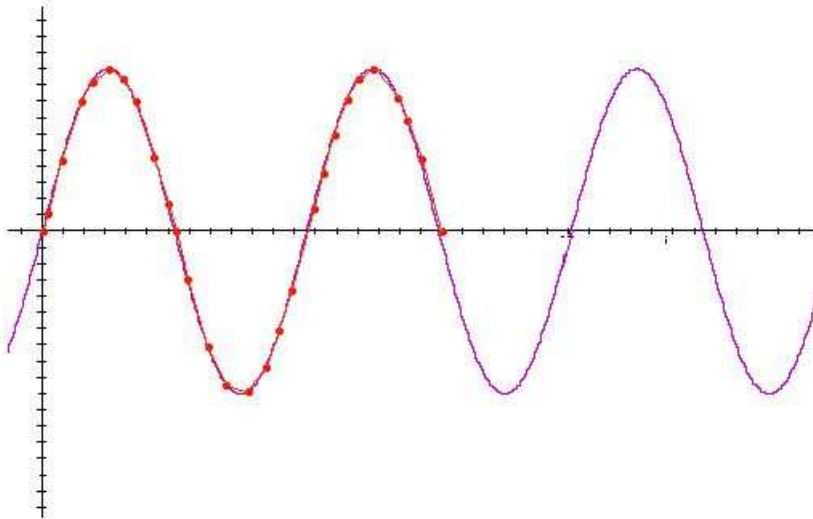


Figure 3: Sampling at above Nyquist Frequency



Back

Close

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

<i>File Type</i>	<i>44.1 KHz</i>	<i>22.05 KHz</i>	<i>11.025 KHz</i>
<i>16 Bit Stereo</i>	10.1 Mb	5.05 Mb	2.52 Mb
<i>16 Bit Mono</i>	5.05 Mb	2.52 Mb	1.26 Mb
<i>8 Bit Mono</i>	2.52 Mb	1.26 Mb	630 Kb

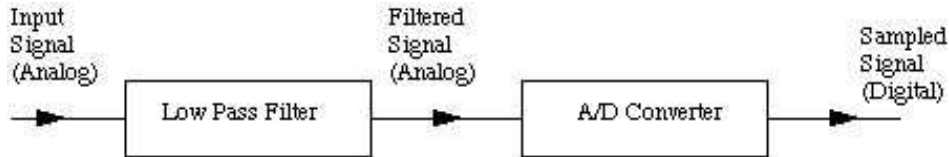


Back

Close

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 bother recording data above 22 KHz
- So filter them out — low pass filter.



Back

Close

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 \(Audio Compression\)](#)



Back

Close

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 \(MPEG-3 and MPEG-4\)](#)



Back

Close

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.



Back

Close

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

More on how synthesis works soon



Back

Close