CSc 461/561
Multimedia Systems
Audio representation

Jianping Pan

Spring 2006

NSERC USRA

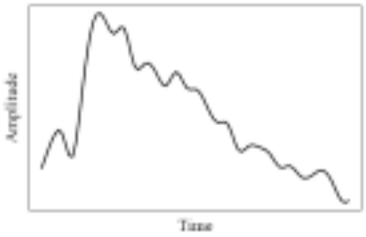
- Undergraduate Student Research Awards
  - UVic CSc awarded 8+ for May'06 to Apr'07
  - work with NSERC research awardees
  - $4,500/term plus 25+% top-up
  - deadline: Feb 20 for May'06; June 16 for Sep...
- For more details
  - go to: google.com; key in: nserc usra
  - click: "I'm Feeling Lucky"

IBM Extreme Blue

- Summer 2006 internship
  - www.ibm.com/extremeblue
  - IBM Toronto or Ottawa Software Lab
  - as well as location in US and elsewhere
  - working in a team of 3 technical students and 1 business student
  - working with technical and business mentors
  - application deadline: Jan 31, 2006

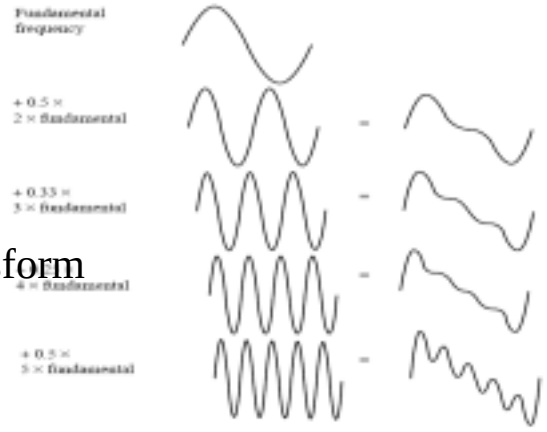
Sound is a wave

- How we hear
  - mouth / transmitter
  - air / cord
  - ear / receiver
- Continuous in both
  - time
  - amplitude



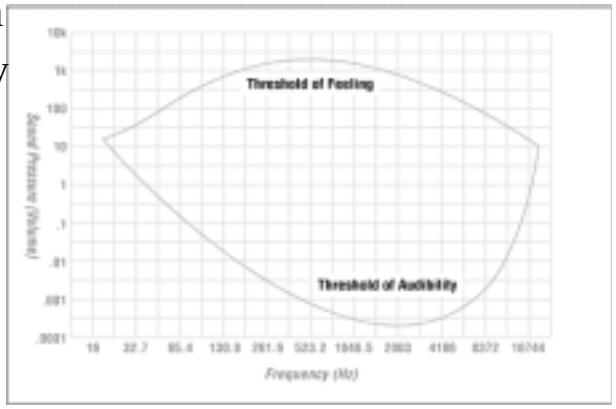
A wave of many *wave*s

- Frequency
  - pitch
- Amplitude
  - loudness
- Fourier Transform

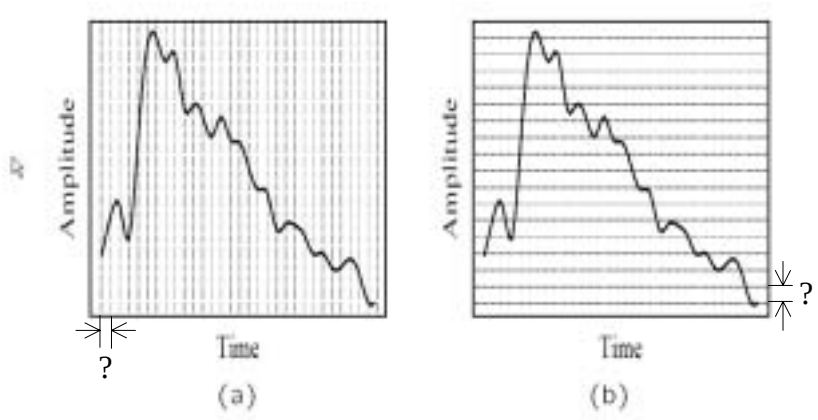


Not all waves are equal

- For human
  - frequency
  - volume

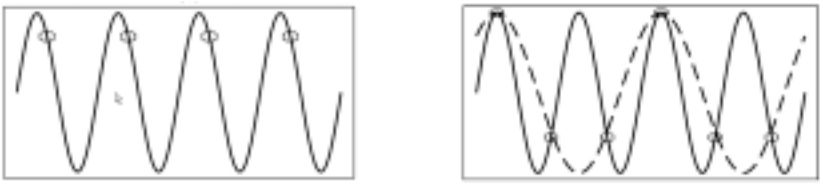


Sampling and quantization



Sampling rate

- If sampling_freq = true_freq
  - in this example, a constant signal
- If sampling_freq = 1.5x true_freq
  - an alias_freq of 0.5x true_freq



Nyquist Theorem

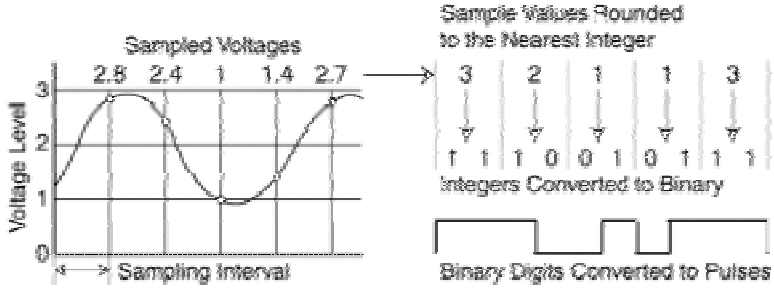
- If a signal is bounded by frequency (f1,f2), then the sampling rate should be *at least* 2(f2-f1) to reconstruct the signal
- Human can hear 20Hz - 20KHz
  - CD sampling rate: 44.1KHz
- Human voice 300Hz - 4KHz
  - telephone systems sampling rate: 8KHz

Alias frequency

- If sampling_freq >= 2x true_freq
  - true_freq is recovered
- If sampling_freq in [true_freq, 2x true_freq)
  - alias_freq = sampling_freq - true_freq
- If sampling_freq < true_freq
  - true_freq' = true_freq mod sampling_freq
  - follow the first two approaches

Quantization depth

- How many bits per sample
  - In this example: 2 bits
  - telephone systems: 8 bits; CD: 16 bits



Quantization noise

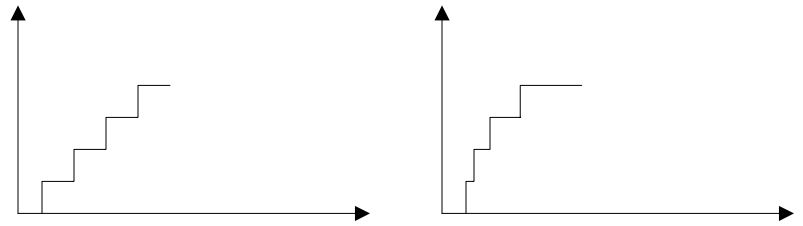
- Quantization is lossy
- More bits, less information loss
- Signal-to-quantization-noise ratio
  - defined as a (log) ratio of signal/noise power
  - one more bit can add 6 db to SQNR

$$SQNR = 20\log_{10}\frac{V_{signal}}{V_{quan\_noise}} = 20\log_{10}\frac{2^{N-1}}{\frac{1}{2}}$$

$$= 20 \times N \times \log 2 = 6.02N(\text{dB}) \qquad (6.3)$$

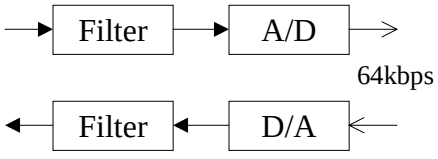
Not all quantizer are linear

- Nonlinear/non-uniform quantization
  - humans are more sensitive to *small* changes from *small* values
- u-law or A-law (more bits for small values)



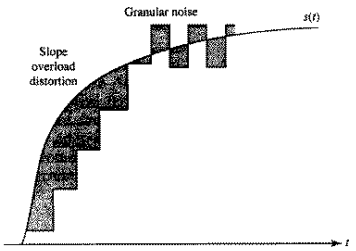
Pulse Code Modulation (PCM)

- Band-limit filter
  - sampling rate: 8KHz
- Quantizer
  - sample size: 8-bit (output) with u-law or A-law
- Data rate: 64Kbps (output)
- Reconstruction: low-pass filter
- G.711



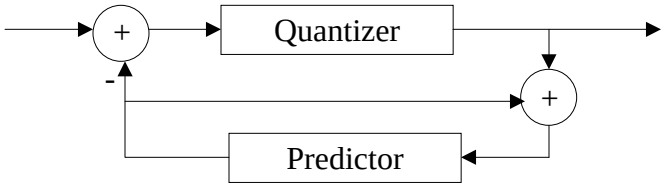
Delta Modulation (DM)

- Prediction
  - If current > last, output 1 (to increase)
  - if current < last, output 0 (to decrease)
- slope overload
  - change too fast
- granular noise
  - change too slow



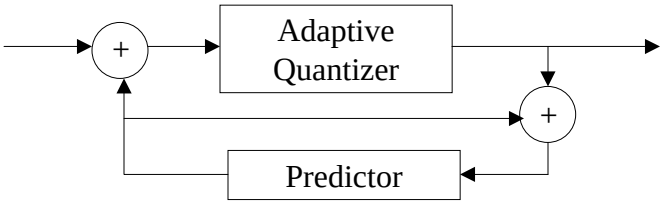
Differential PCM (DPCM)

- Predictor
  - Predict the current based on the history
- Quantizer
  - quantize the difference



Adaptive DPCM (ADPCM)

- Adaptive quantizer
  - adapt quantization step to the difference



- G.723: 8KHz, 4-bit, 32Kbps

This lecture

- Audio presentation
  - sampling rate, alias frequency
  - quantization depth, SQNR
  - PCM, DPCM, ADPCM
- Explore further
  - MIDI [Li&Drew 6.2]

Next lecture

- Multimedia representation
  - image [Ref: Li&Drew Chap 3-4]
    - bitmap vs vector
    - gray-scale vs color [3.1.1/2, 3.1.4-6]
    - picture resolution, pixel depth
    - color space (RGB, CMY, YUV) [4.2.1-3, 4.3.2]
    - gamma correction [4.1.6]