# 2018 Fall
# CTP431: Music and Audio Computing

# Digital Audio

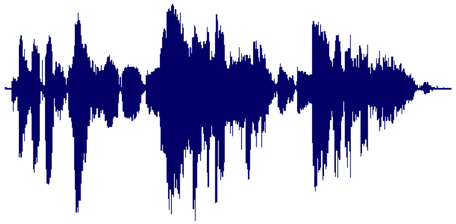
Graduate School of Culture Technology, KAIST
Juhan Nam

# Outlines

- Introduction
- Sampling
- Quantization
- Digital audio standards
- Playback Rate Conversion / Resampling

# Introduction

Sound



⟷ … 0 1 1 0 1 1 0 …

Image

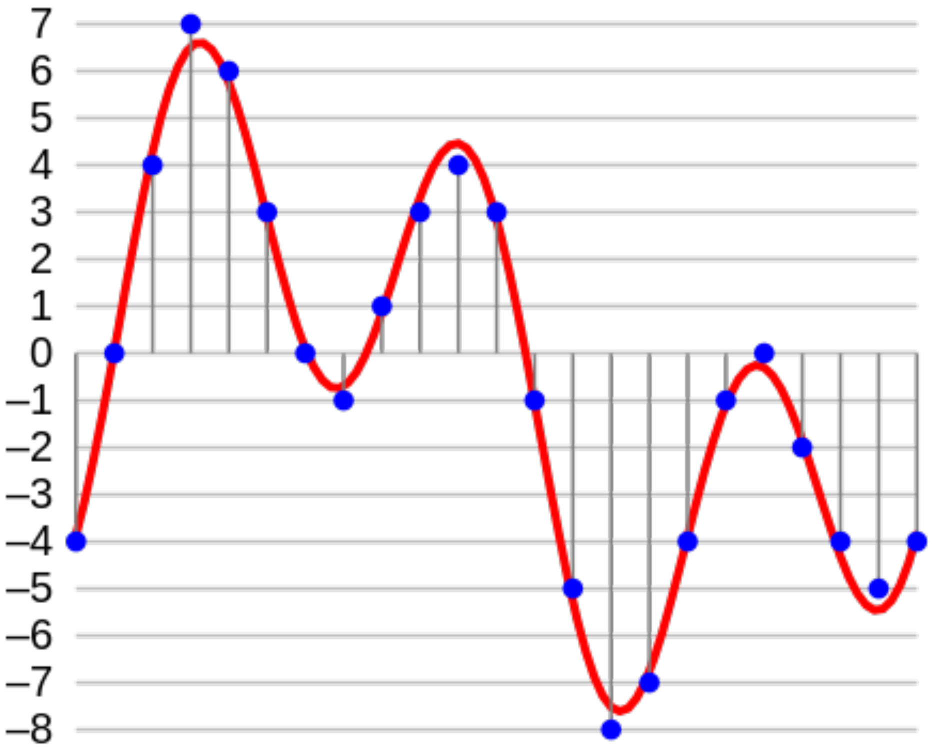
⟷ … 1 0 0 1 1 0 1 …

Text


⟷ … 0 0 1 1 0 1 1 …

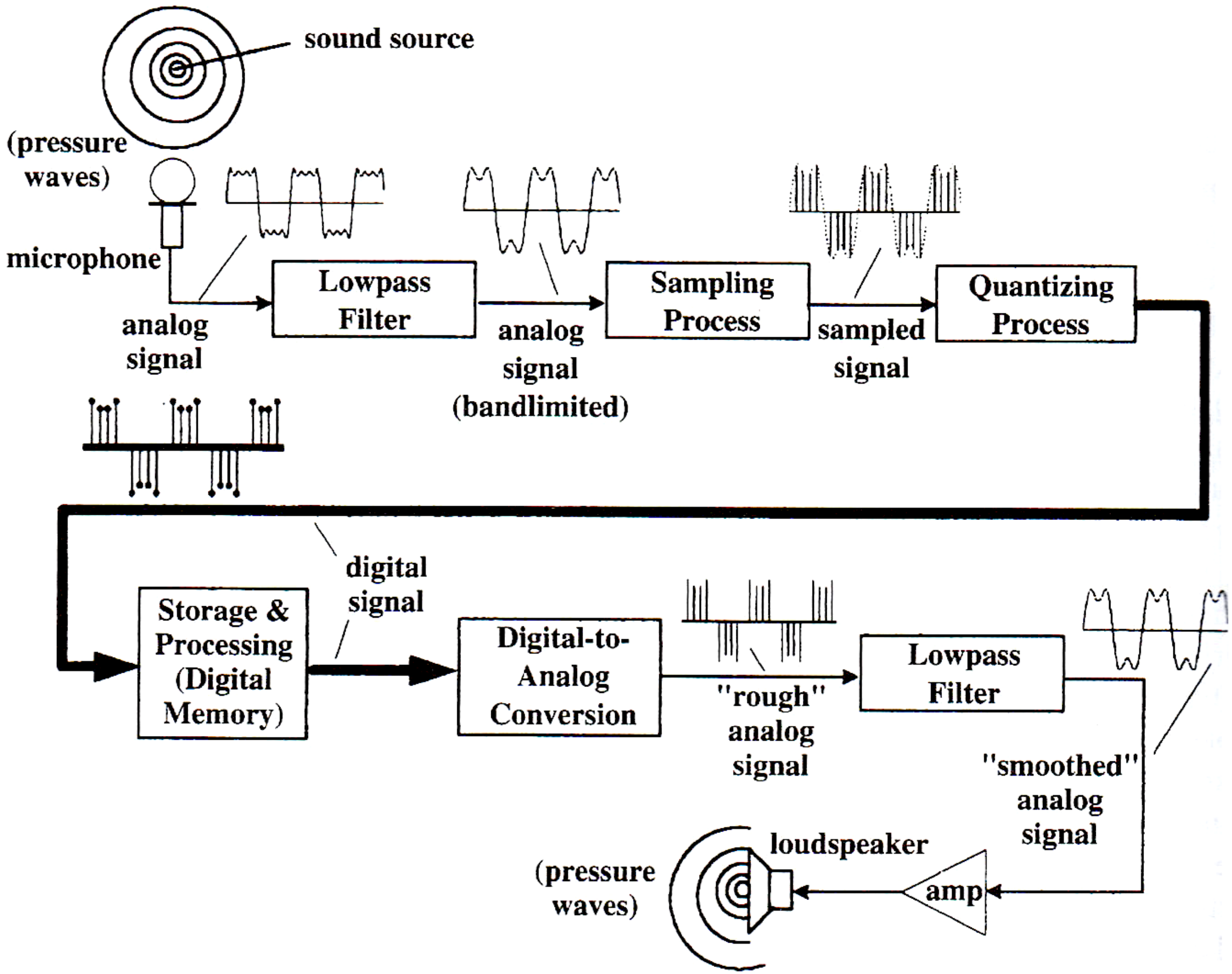
# Digital Representations

- Sampling and Quantization
  - Sound (samples)
  - Image (pixels)

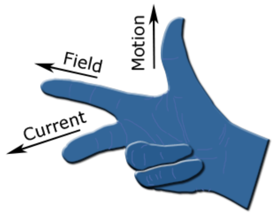



- Trade-off
  - Between quality and data size

# Digital Audio Chain

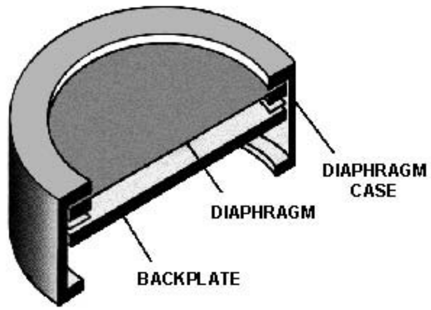


# Microphone and Speakers

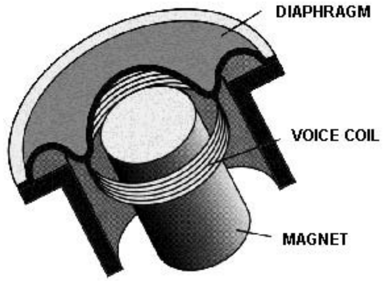



- Microphones
  - Sound to electrical signal
  - Dynamic mic: Fleming's right-hand rule
  - Condenser mic: Q = CV, C = A/d
  - Pre-amp

(A= area, D= distance)

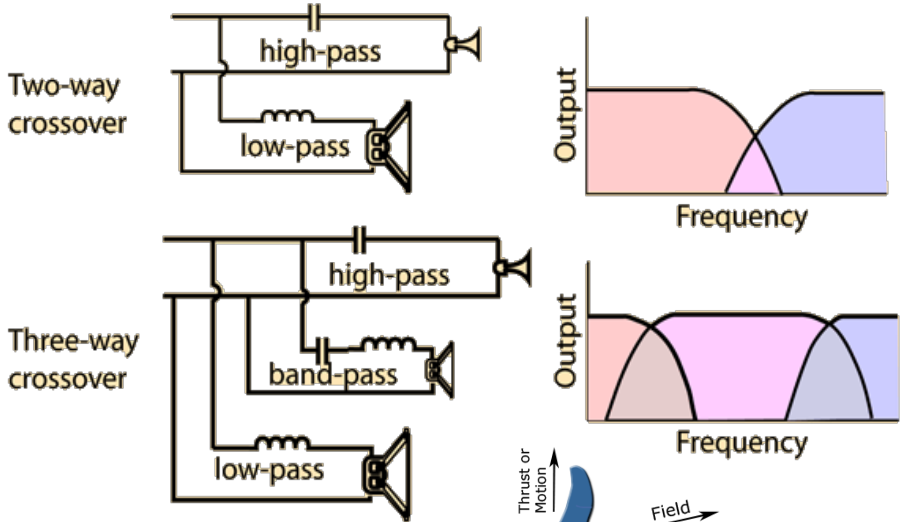


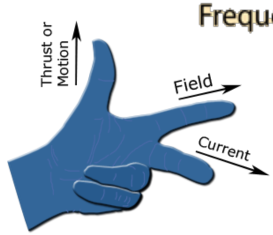

Condenser mic




Dynamic mic

- Loudspeakers
  - Electrical signal to sound
  - Similar to dynamic mic in principle
    - Fleming's left-hand rule
  - Crossover networks: woofer / tweeter
  - Power amp

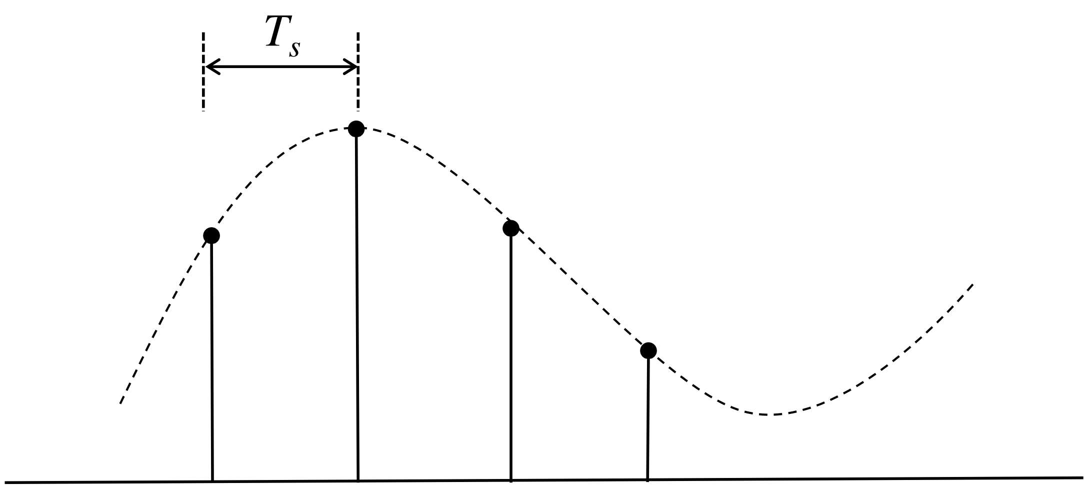



Source: http://www.shure.com/americas/support/find-an-answer/difference-between-a-dynamic-and-condenser-microphone
Source: http://hyperphysics.phy-astr.gsu.edu/hbase/Audio/cross.html

# Sampling

- Convert continuous-time signals to discrete-time signals by periodically picking up the instantaneous values
  - Represented as a sequence of numbers: pulse code modulation (PCM)
  - Sampling period ($T_s$): the amount of time between samples
  - Sampling rate ($f_s = 1/T_s$)




Signal notation

$$x(t) \rightarrow x(nT_s)$$

# Sampling Theorem

- What is an appropriate sampling rate?
  - Too high: increase data rate
  - Too low: become hard to reconstruct the original signal

- Sampling Theorem
  - In order for a band-limited signal to be reconstructed fully, the sampling rate must be greater than twice the maximum frequency in the signal

$$\boxed{f_s > 2 \cdot f_m}$$

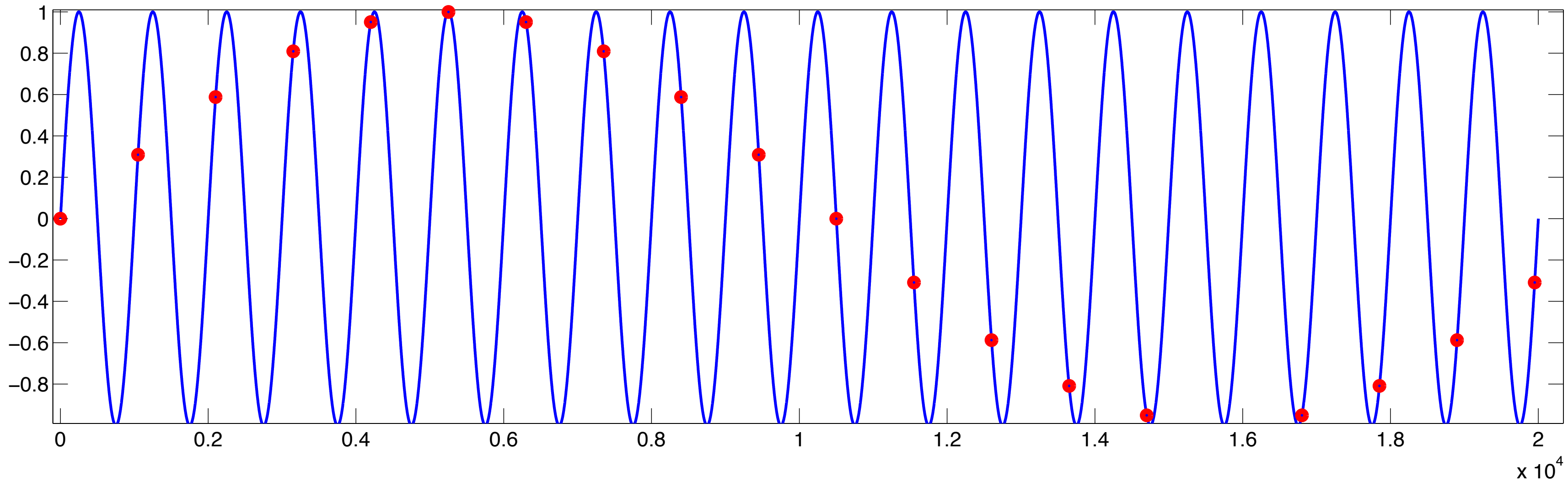
$f_s$: sampling rate

$f_m$: maximum frequency

  - Half the sampling rate is called Nyquist frequency ($f_s/2$)

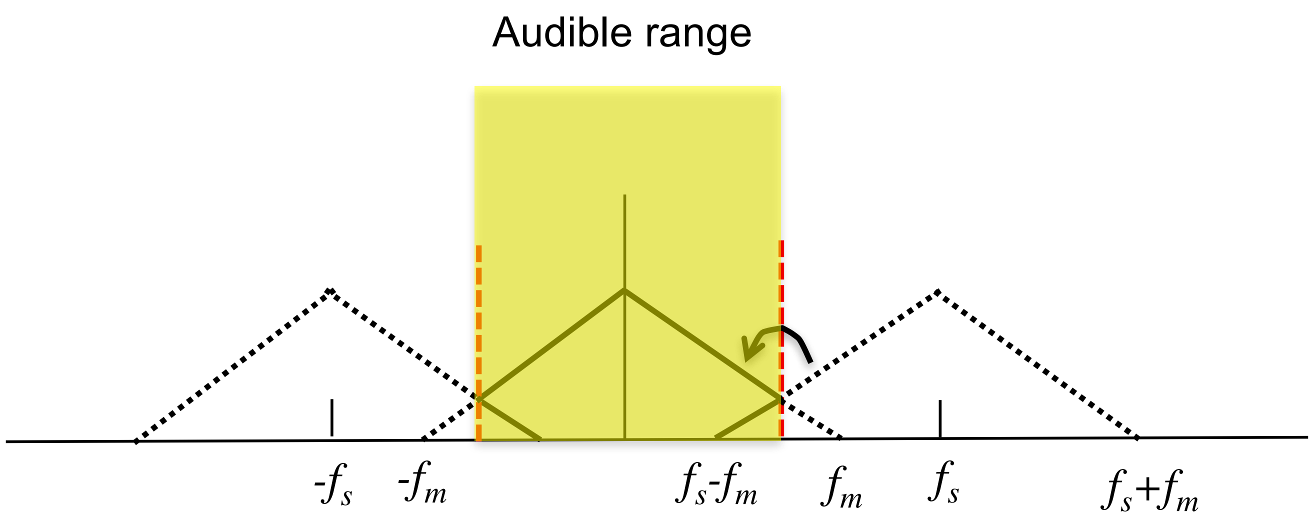
# Aliasing

- If the sampling rate is less than twice the maximum frequency, the high-frequency content is folded over to lower frequency range



# Aliasing in Frequency Domain

- For general signals, high-frequency content is folded over to lower frequency range

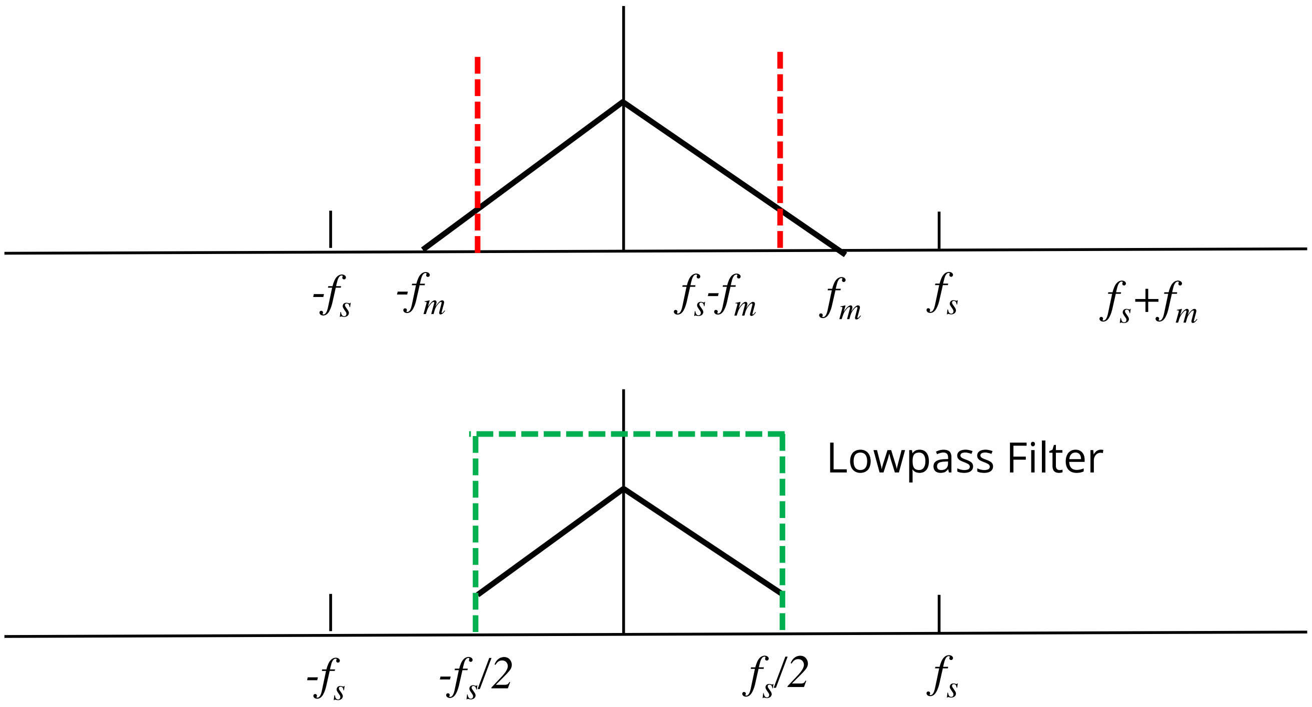


# To avoid Aliasing

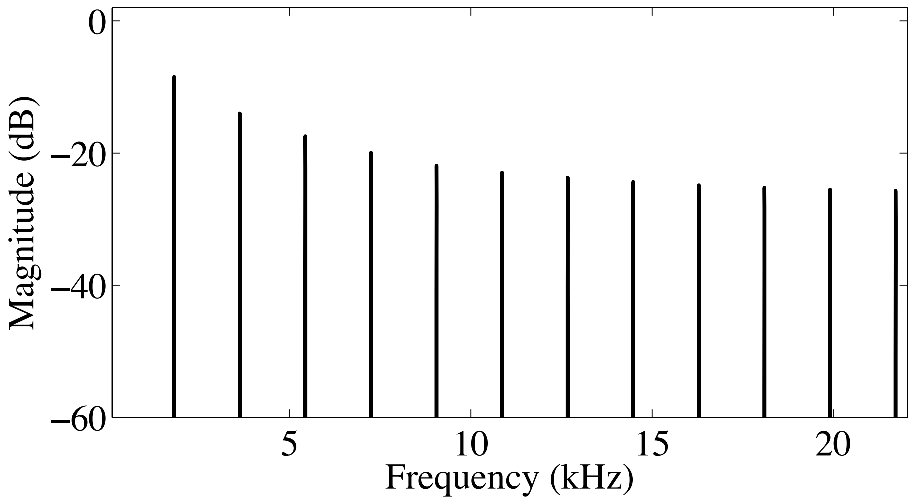
- Increase sampling rate

$$\boxed{f_s > 2 \cdot f_m}$$

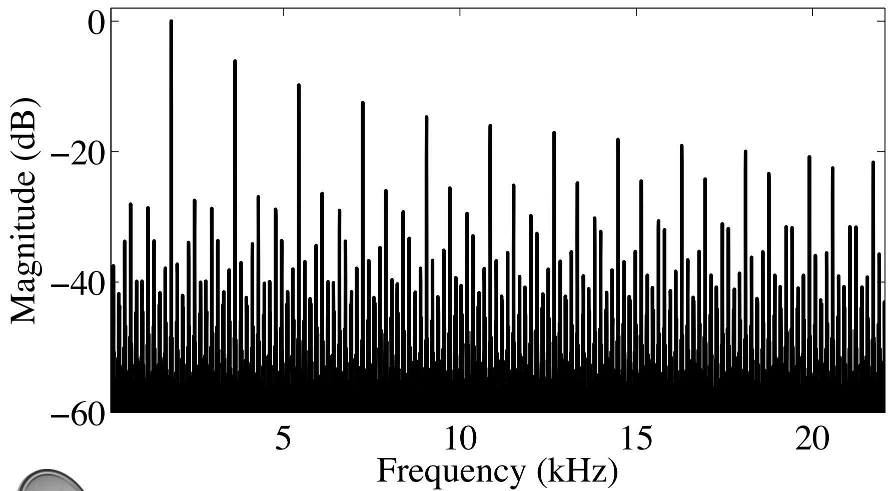
- Or use lowpass filters before sampling



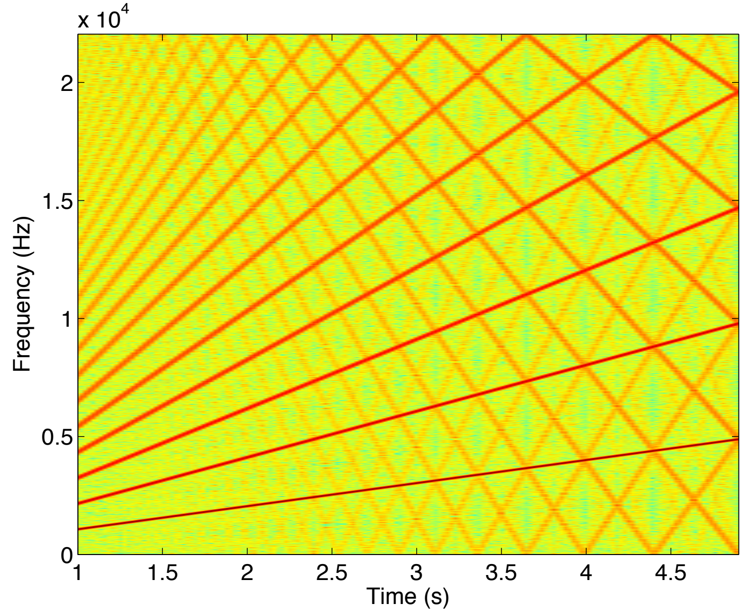
# Example of Aliasing




Bandlimited sawtooth wave spectrum




Trivial sawtooth wave spectrum



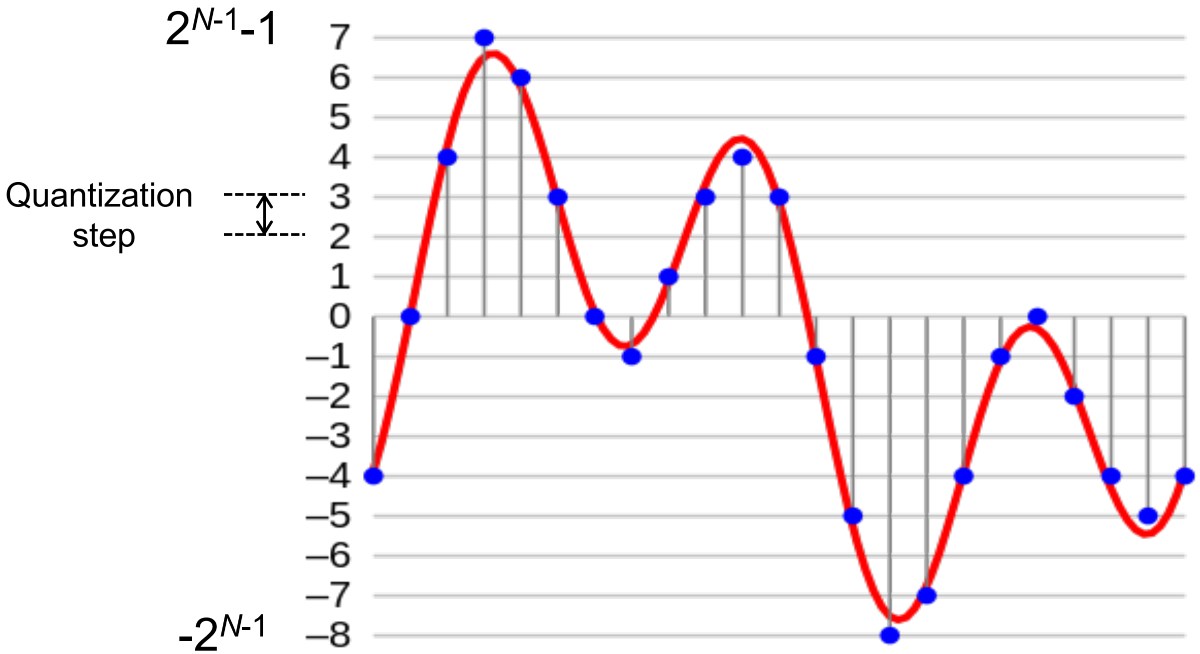

Frequency sweep of the trivial sawtooth wave

# Example of Aliasing in Video

https://www.youtube.com/watch?v=jHS9JGkEOmA

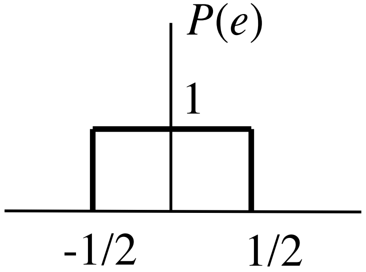
# Quantization

- Discretizing the amplitude of real-valued signals
  - Round the amplitude to the nearest discrete steps
  - The discrete steps are determined by the number of bit bits
    - $N$ bits can range from $-2^{N-1}$ to $2^{N-1}-1$



# Quantization Error

- Quantization causes noise
  - Average power of quantization noise: obtained from the probability density function (PDF) of the error




Root mean square (RMS) of noise

$$\sqrt{\int_{-1/2}^{1/2} x^2 p(e)\, dx} = \sqrt{1/12}$$

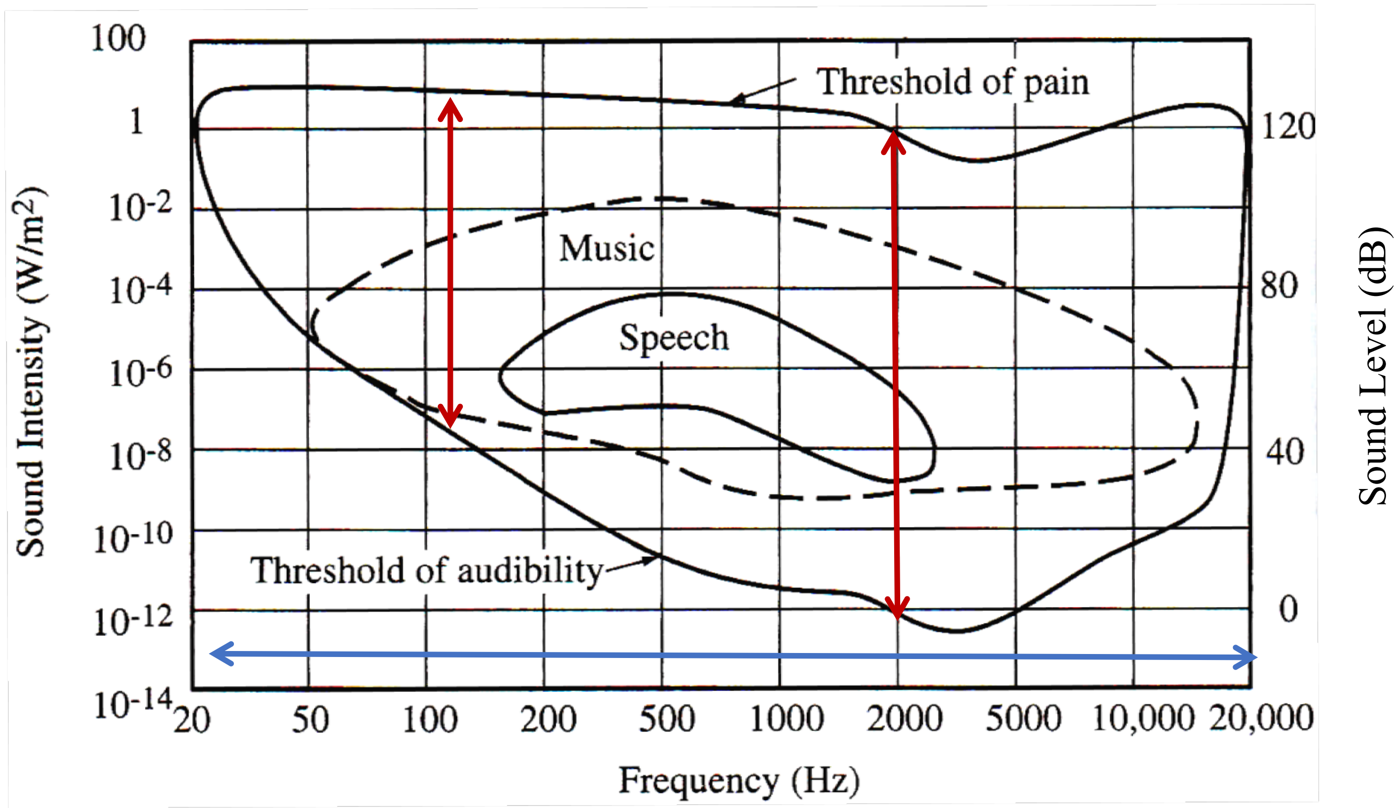
- Signal to Noise Ratio (SNR)
  - Based on average power

RMS of full-scale sine wave

$$20\log_{10}\frac{S_{\text{rms}}}{N_{\text{rms}}} = 20\log_{10}\frac{2^{B-1}/\sqrt{2}}{\sqrt{1/12}} = 6.02B + 1.76\ \text{dB}$$

# Digital Audio Standards

- Determined by the limit in human hearing
  - Maximum audible frequency (bandwidth): 20kHz
  - Dynamic range: depends on frequency (the maximum is about 120dB)




Human hearing range

# Digital Audio Standards

- Compact disc
  - Sampling rate: 44.1 kHz: > 2 x 20 kHz
  - Bit depth: 16 bits: SNR = 98.08dB
- Blu-ray disc / professional audio
  - Sampling rate: 48 / 96 /192kHz: > 2 x 20 kHz
  - Bit depth: 16 / 20 / 24 bits
- Telephone
  - Sampling rate: 8 / 16 kHz
  - Bit depth: 8 bits (with companding)

# Playback Rate Conversion

- Playback rate does not have to be the same as the recording rate

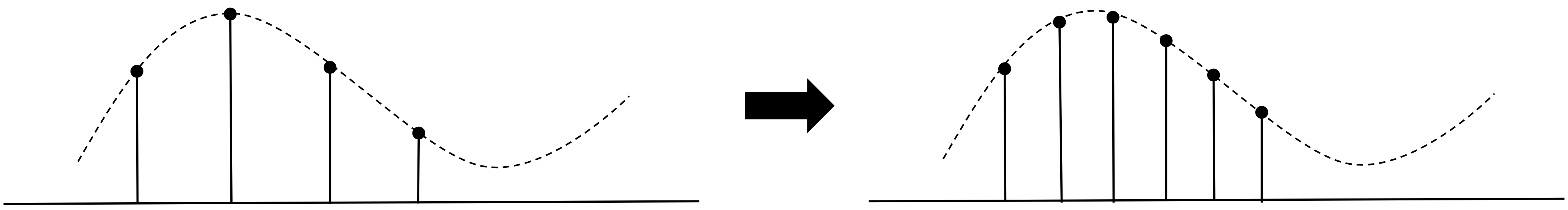
- Adjusting the playback rate given the recorded audio creates different tones
  - Sliding tapes on the magnetic header in a variable speed
  - Speeding down: “monster-like”
  - Speeding up: “chipmunk-like”
  - It can be even negative rate: reverse playback

# Demo: Playback Rate Conversion

- https://musiclab.chromeexperiments.com/Voice-Spinner

# Resampling

- Reconstruct the original signal and sample it with a new sampling rate



- For a digital system with a constant playback rate
  - **Up-sampling** makes the original have **slower speed and lower pitch**
  - **Down-sampling** makes the original have **faster speed and higher pitch**

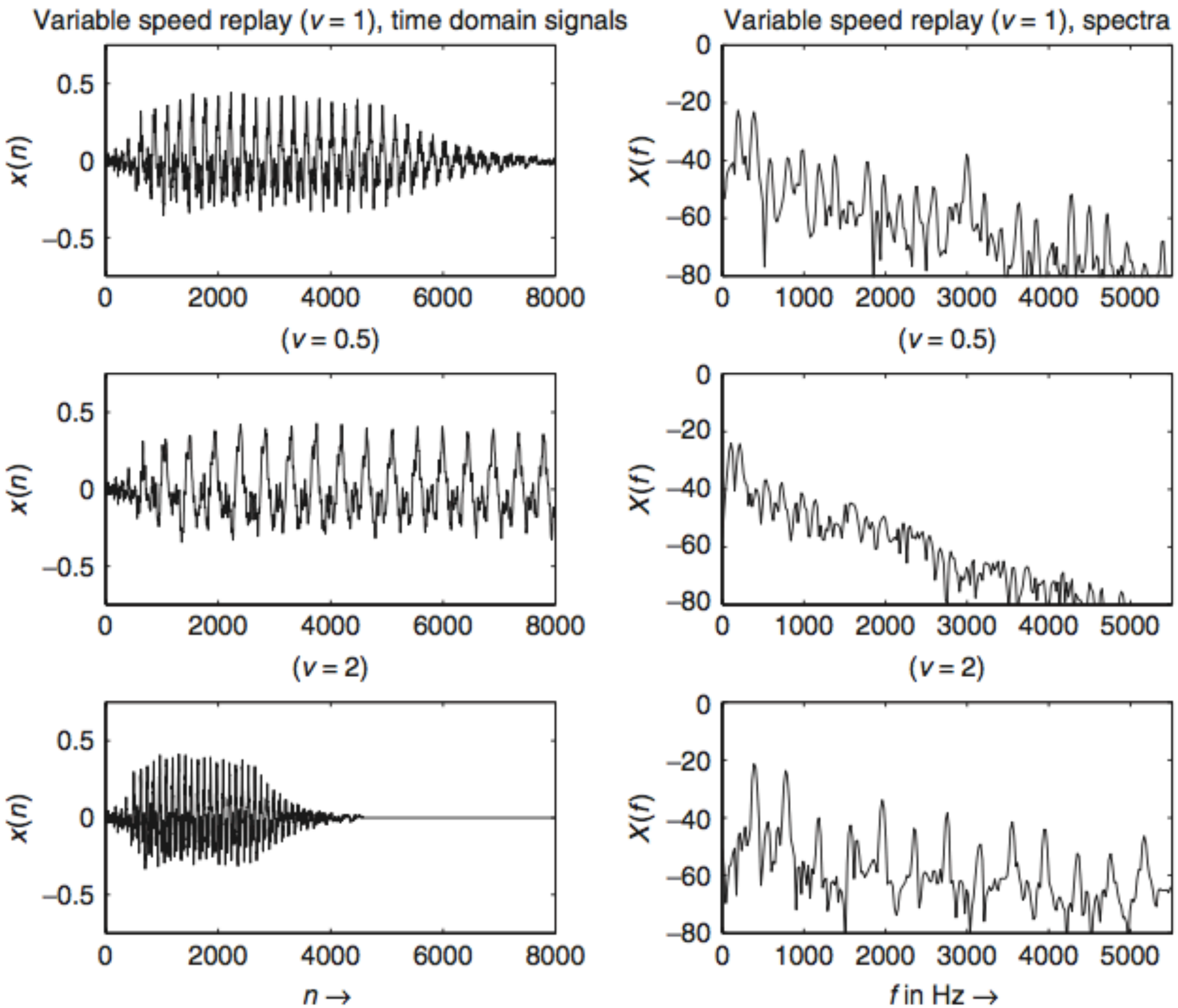
# Resampling

- Resampling changes pitch, length and timbre at the same time!

Original

Speed Down
(Up-sampling)

Speed Up
(Down-sampling)




[The DaFX book]

# Practice: Audacity

- Recording
- Editing
- Digital Audio Effects