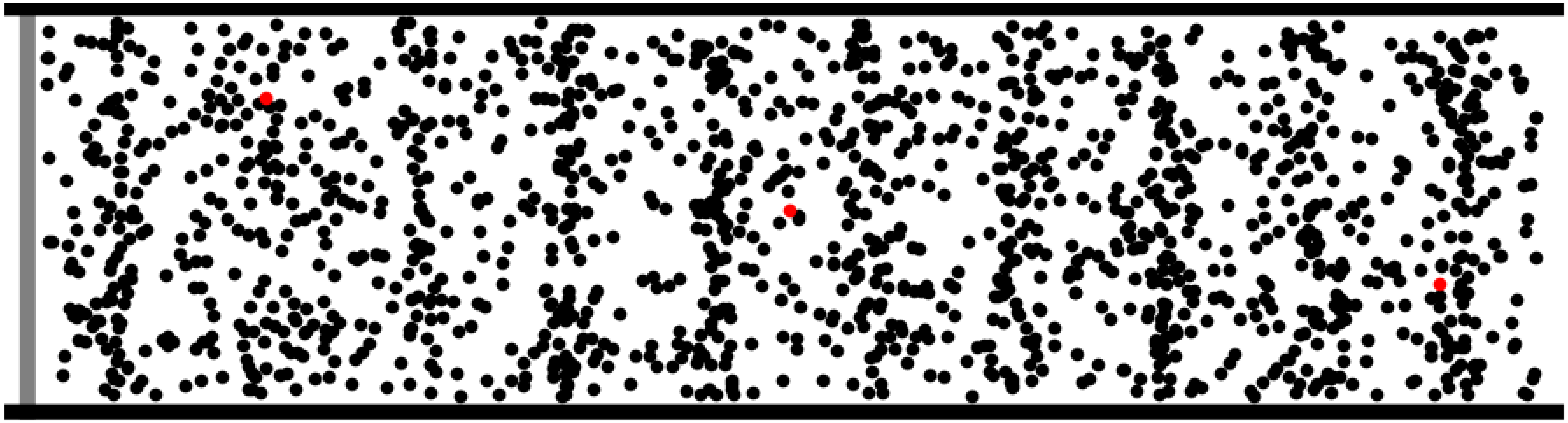
Digital Audio

The mystery behind sampling and reconstruction

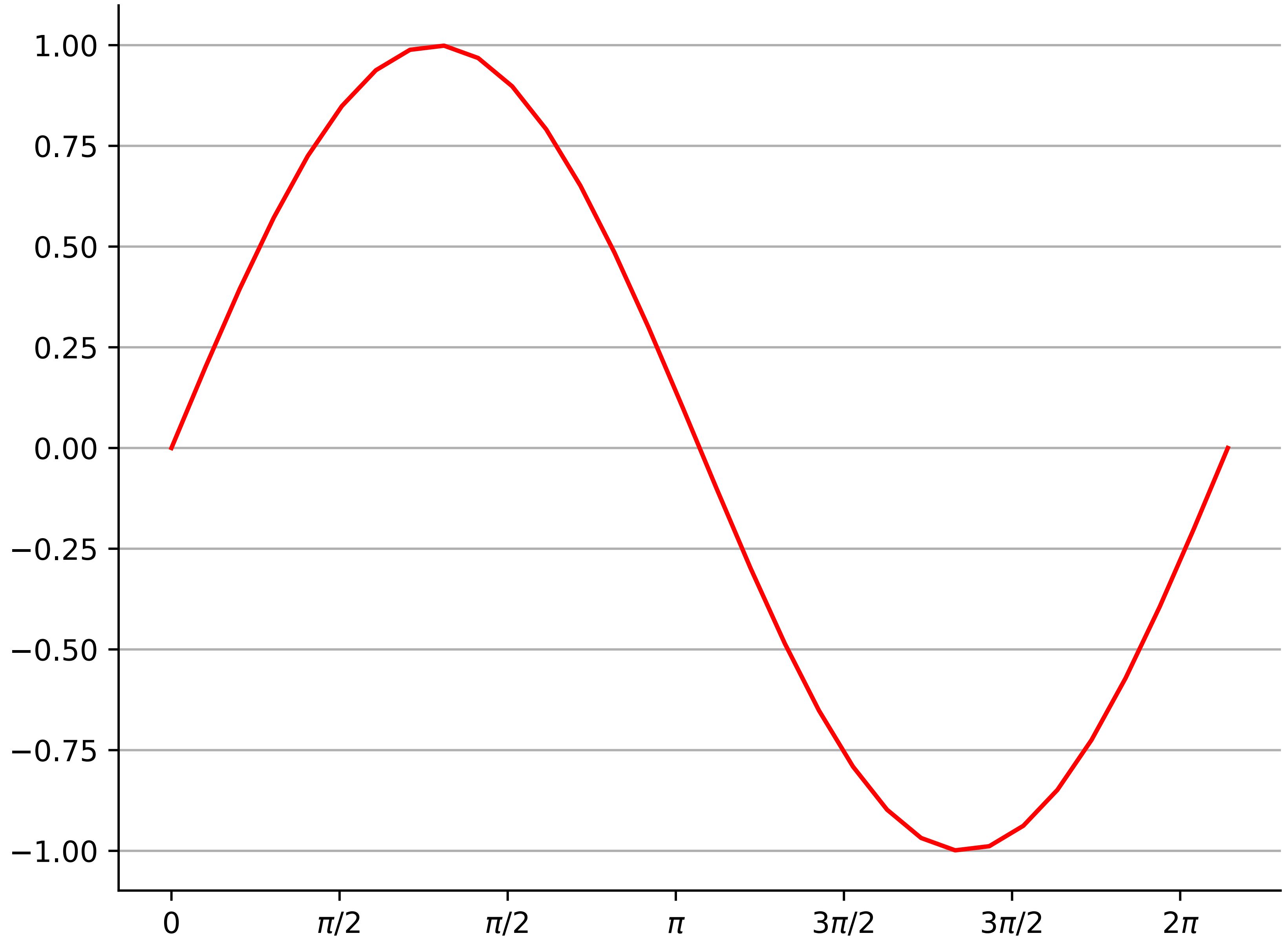
Spring 2019 - Audio Tech Talk Series

January 22, 2019

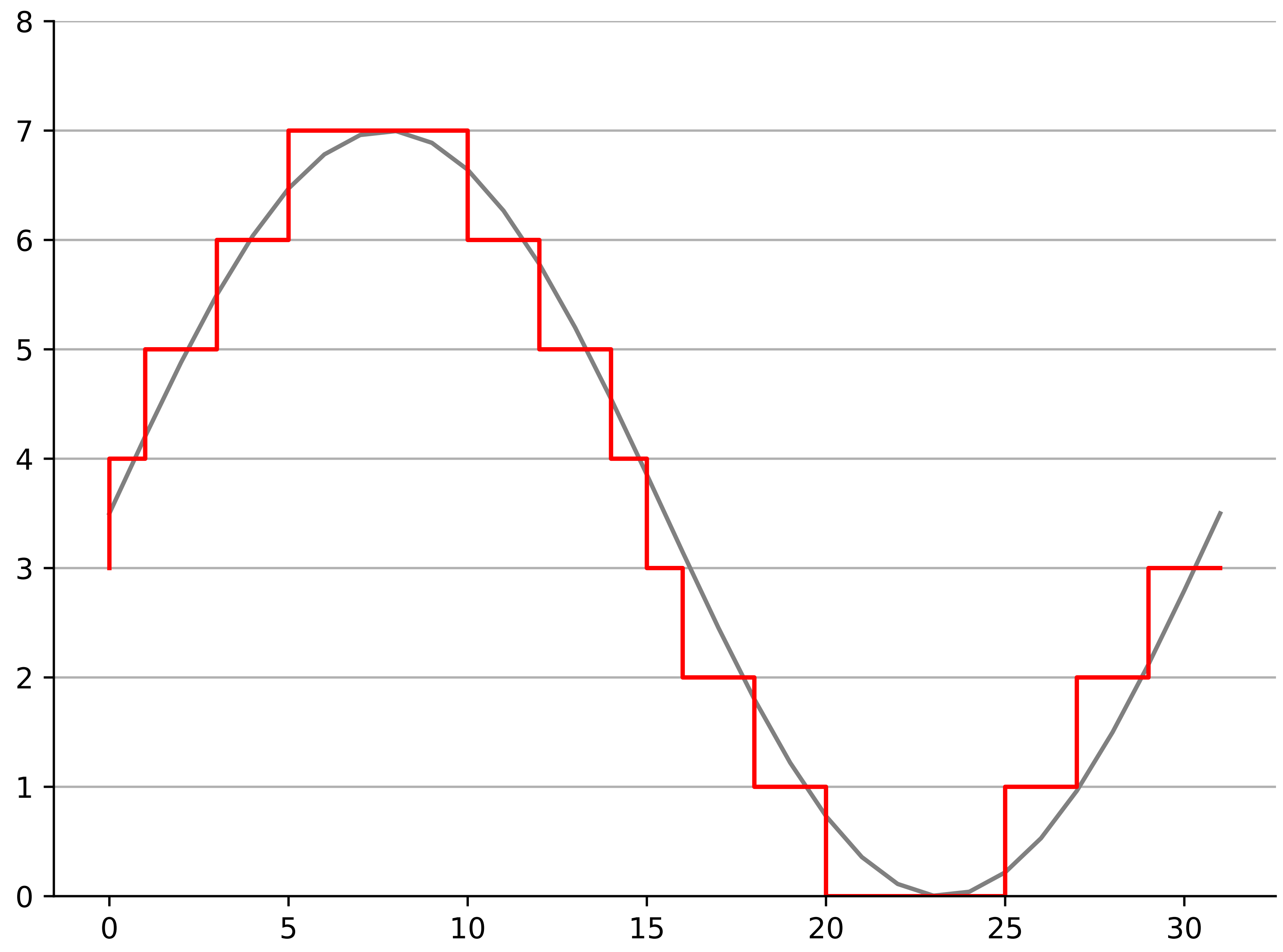
What is audio?

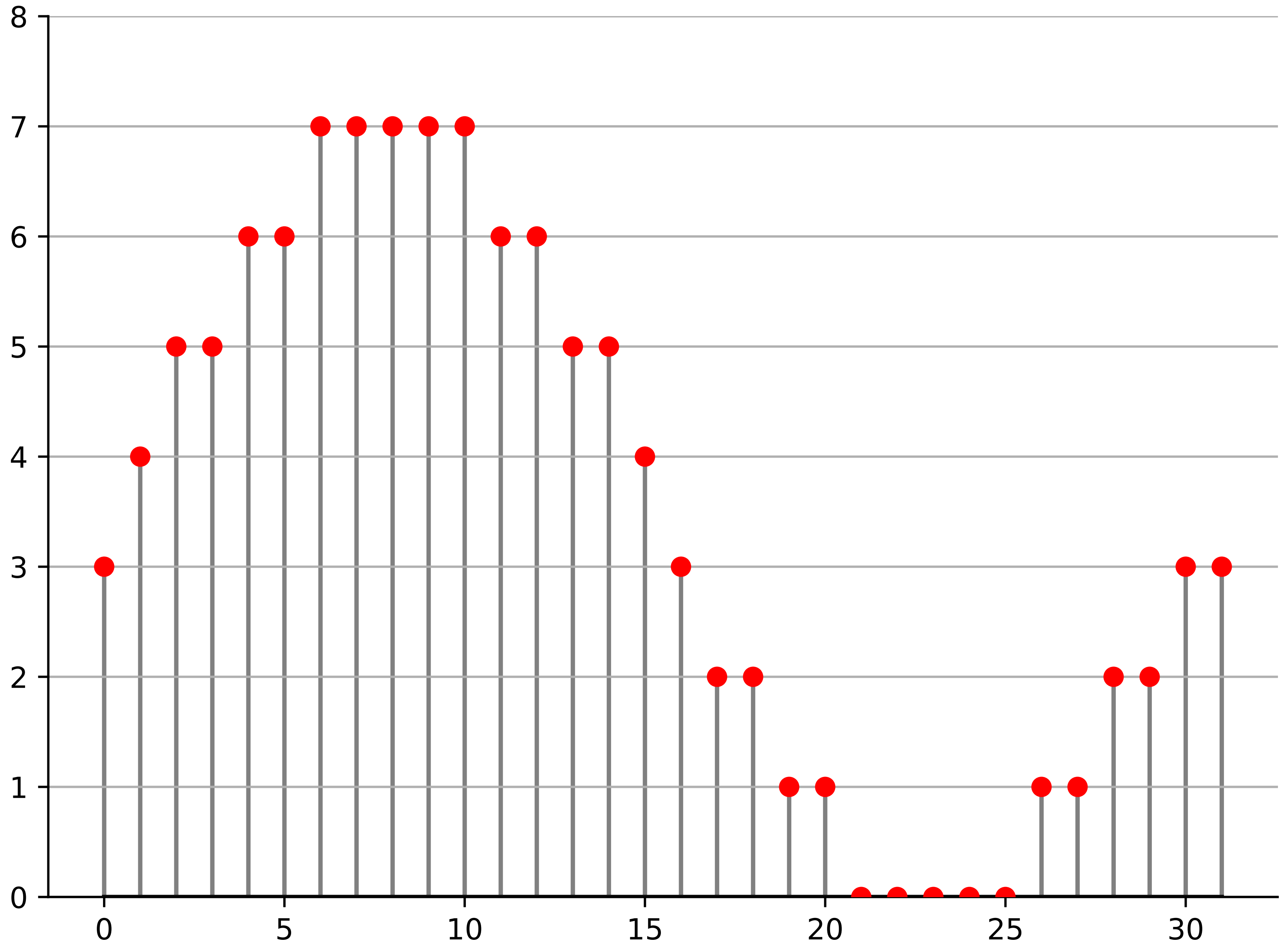


Analog Audio



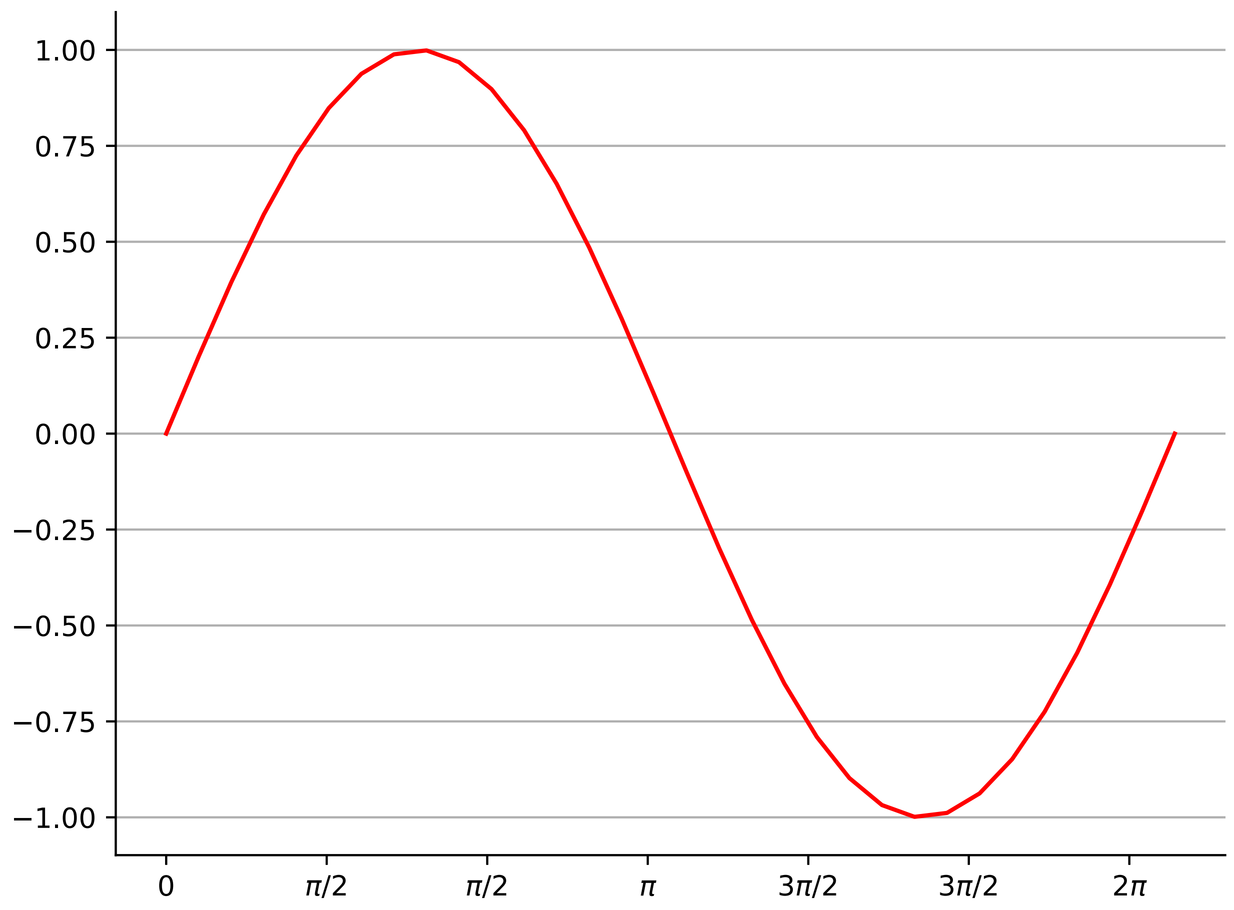
Digital Audio

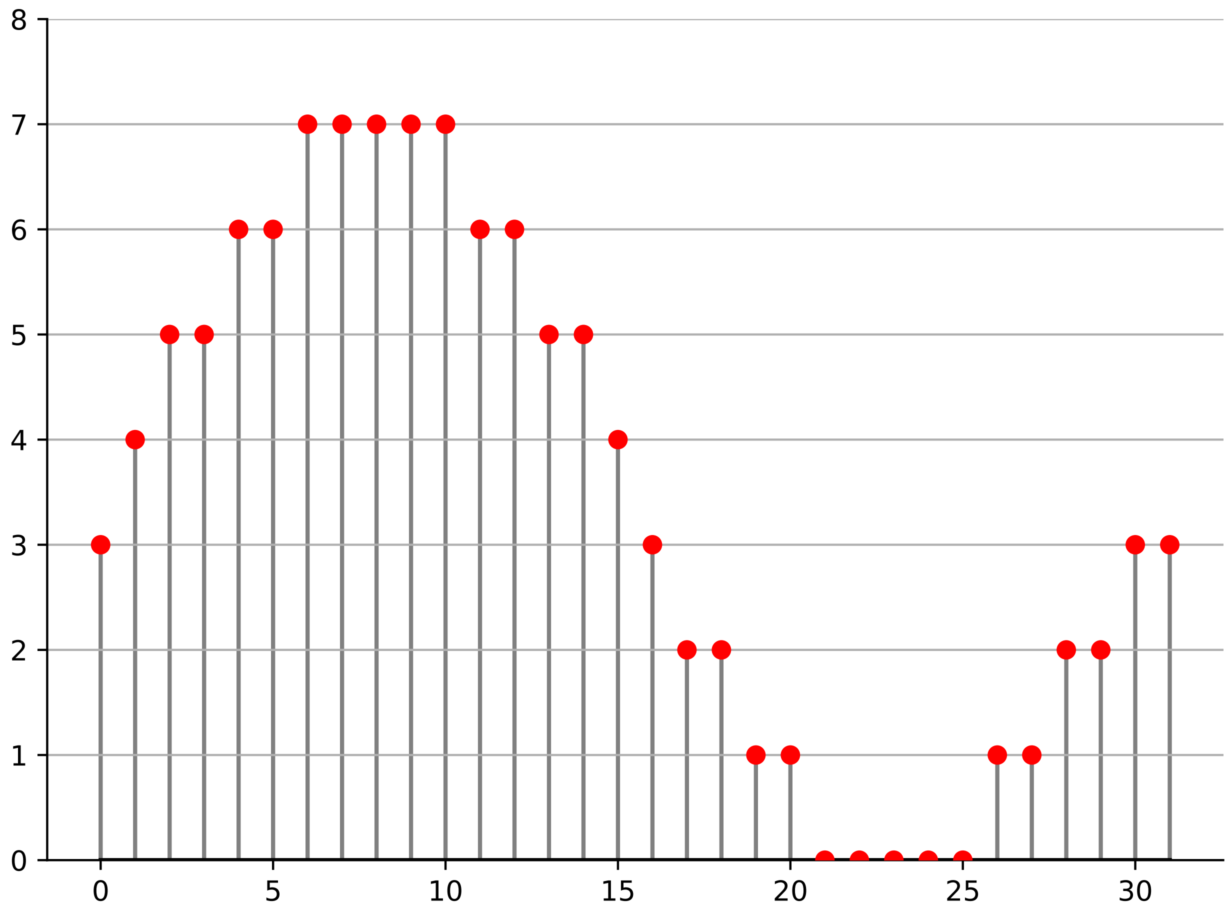




0
1
2
3
4
5
6
7
8
0
5
10
15
20
25
30

Sampling Theorem





Sampling Theorem

A continuous time signal (analog) can be completely represented by its samples and can be recovered back when the sampling frequency $\mathbf{f_s}$ is greater than or equal to the twice the highest frequency component $\mathbf{f_m}$ of the signal. i.e.

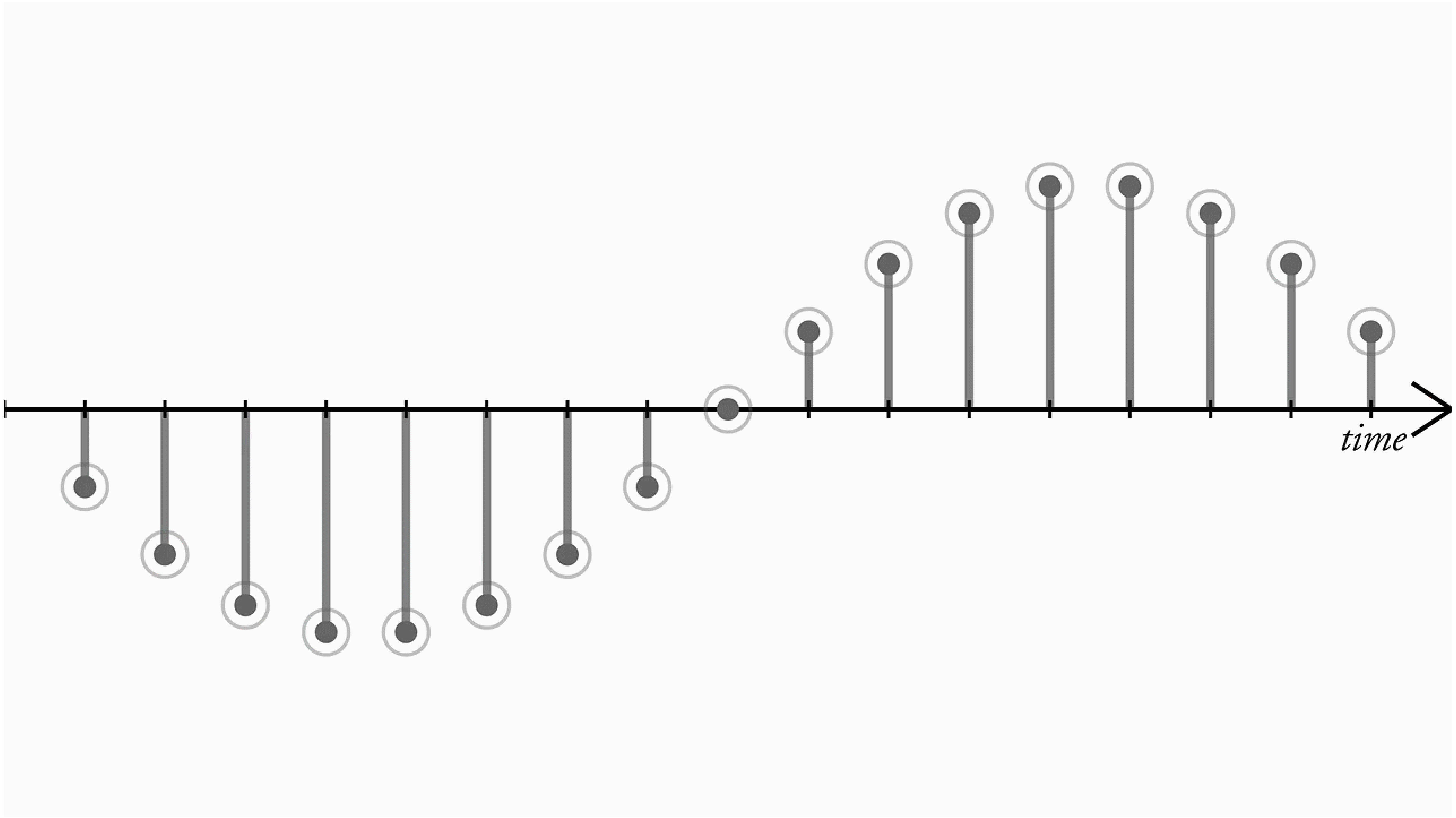
$$\mathbf{f_s \geq 2 * f_m}$$

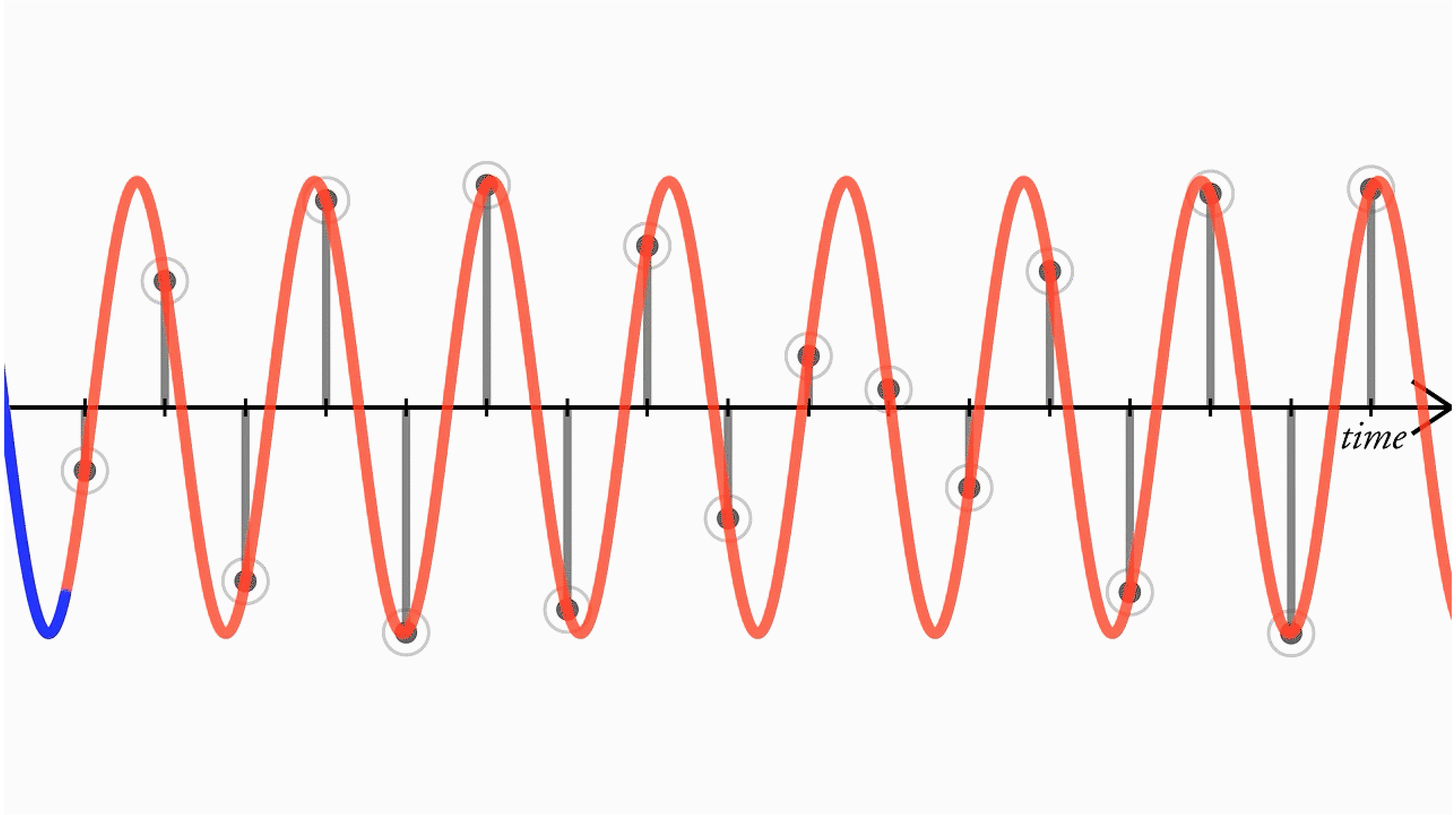
Examples

- Human speech (100 Hz - 4,000 Hz)
- Music (20 Hz - 20,000 Hz)
- Temperatue sensor (~0 Hz - 0.25 Hz)

Reconstruction

There is only ONE bandlimited signal that fits the sampled points.



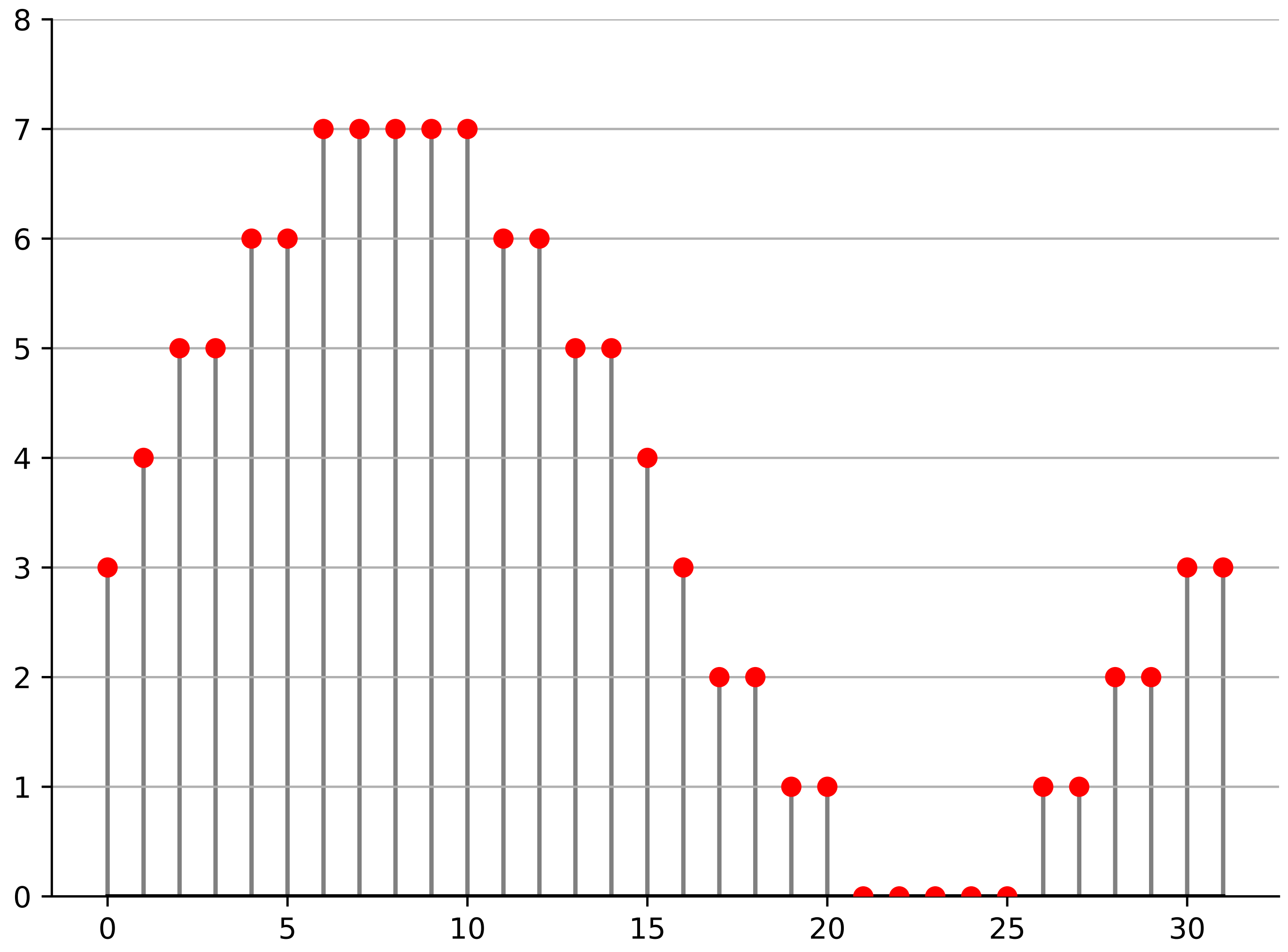

time

Caveats

- Sampled signal MUST be bandlimited -> **Aliasing**
- Finite bit depth (e.g. 16 bits) -> **Quantization noise**

Pulse Code Modulation (PCM)

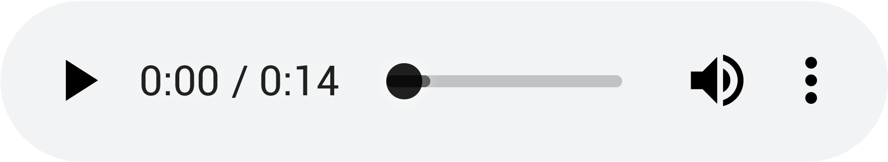
Sampling Rate and Bit Depth



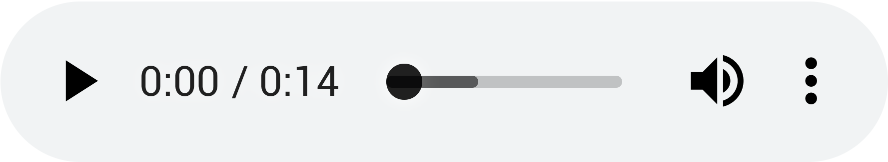
Bit depth = number of discrete levels -> **Dynamic range**

Sampling rate = number samples per second -> **Bandwidth**

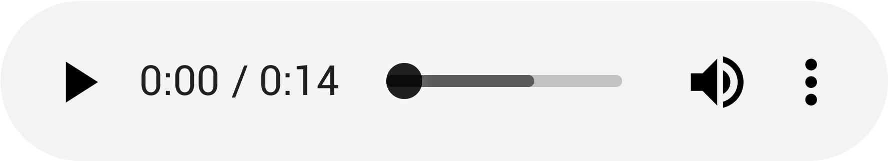
Sampling Rate



f_s=44.1 kHz -> f_m=22.05 kHz

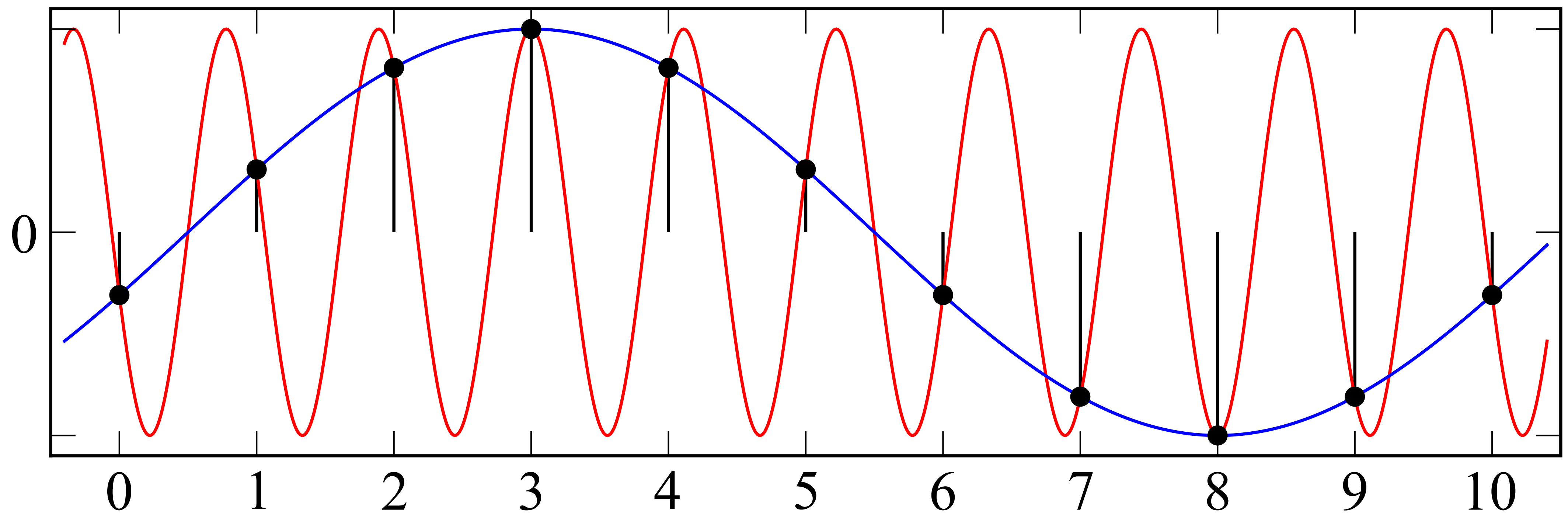


f_s=22.05 kHz -> f_m=11.025 kHz

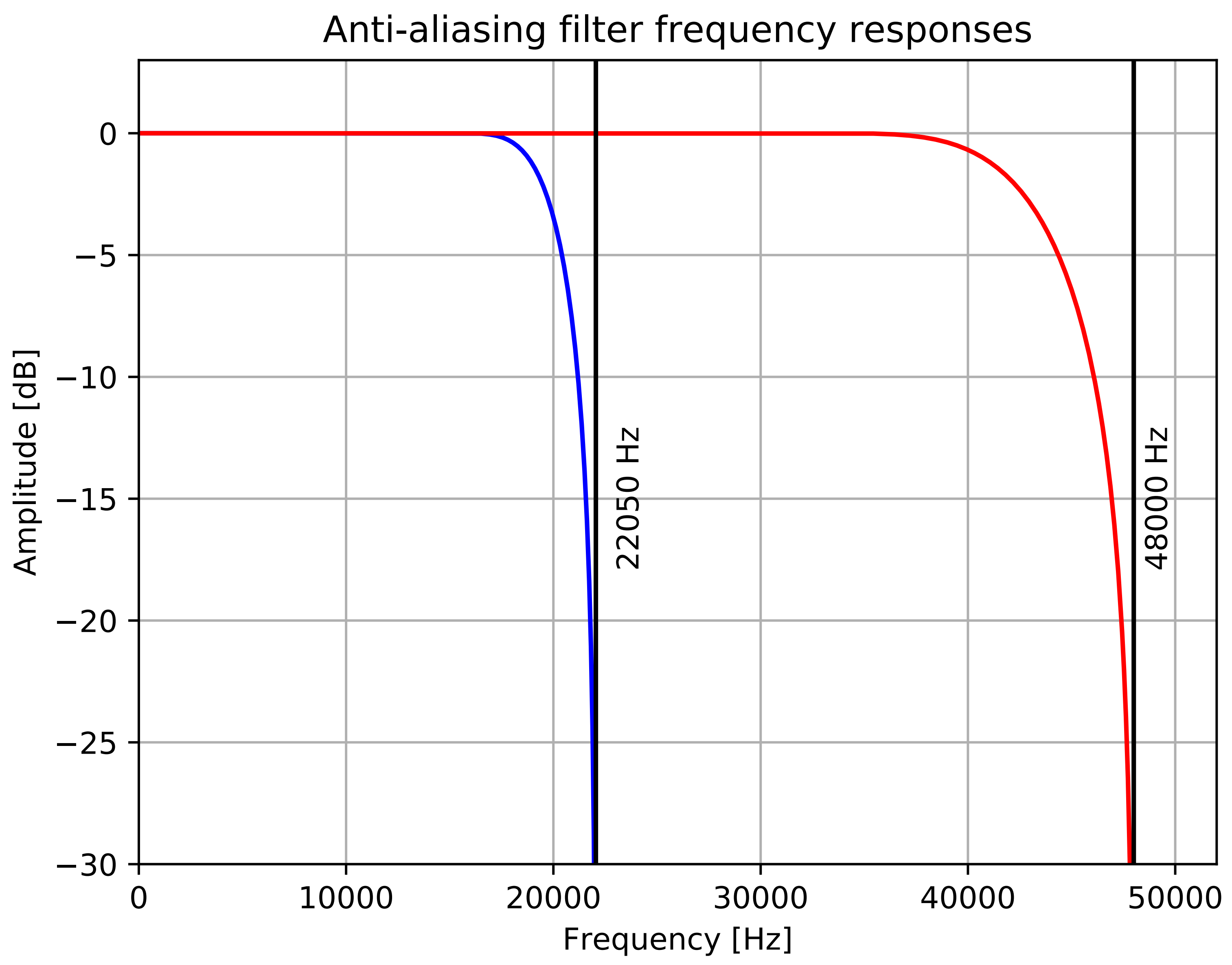


f_s=11.025 kHz -> f_m=5.5125 kHz

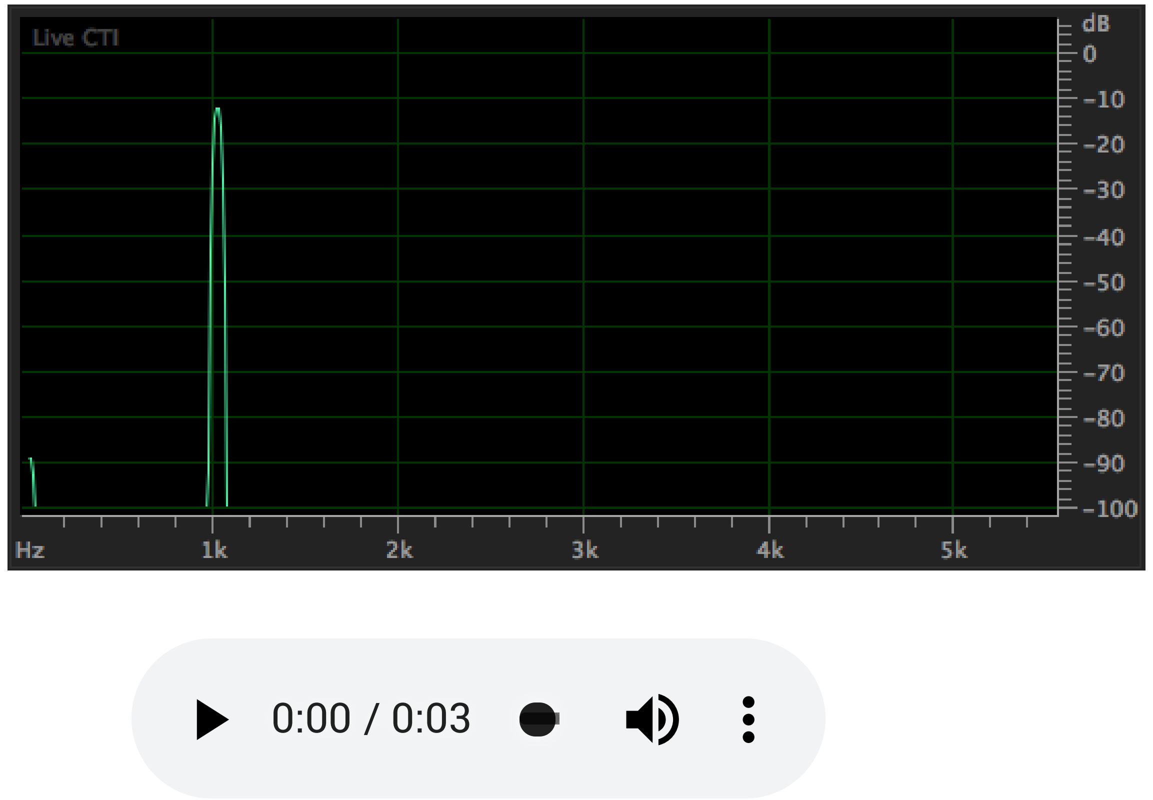
Aliasing



Anti-Aliasing Filters

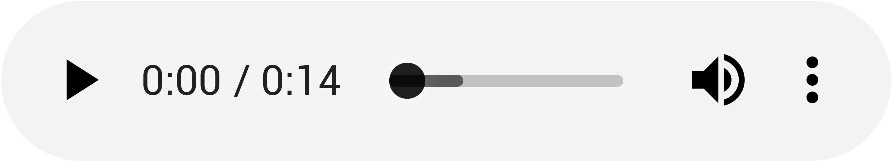


1 kHz + 8 kHz @ 16bit 44.1 kHz

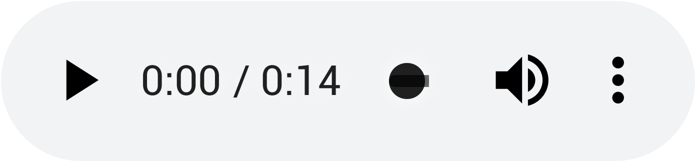


1 kHz + 8 kHz @ 16bit 11.025 kHz

1 kHz + 8 kHz @ 16bit 11.025 kHz (No Anti-Aliasing filter)



f_s=22.05 kHz -> f_m=11.025 kHz



f_s=22.05 kHz -> f_m=11.025 kHz (no anti-aliasing filter)

Bit Depth

0:00 / 0:14

16 bit -> 96 dB

0:00 / 0:14

8 bit -> 48 dB

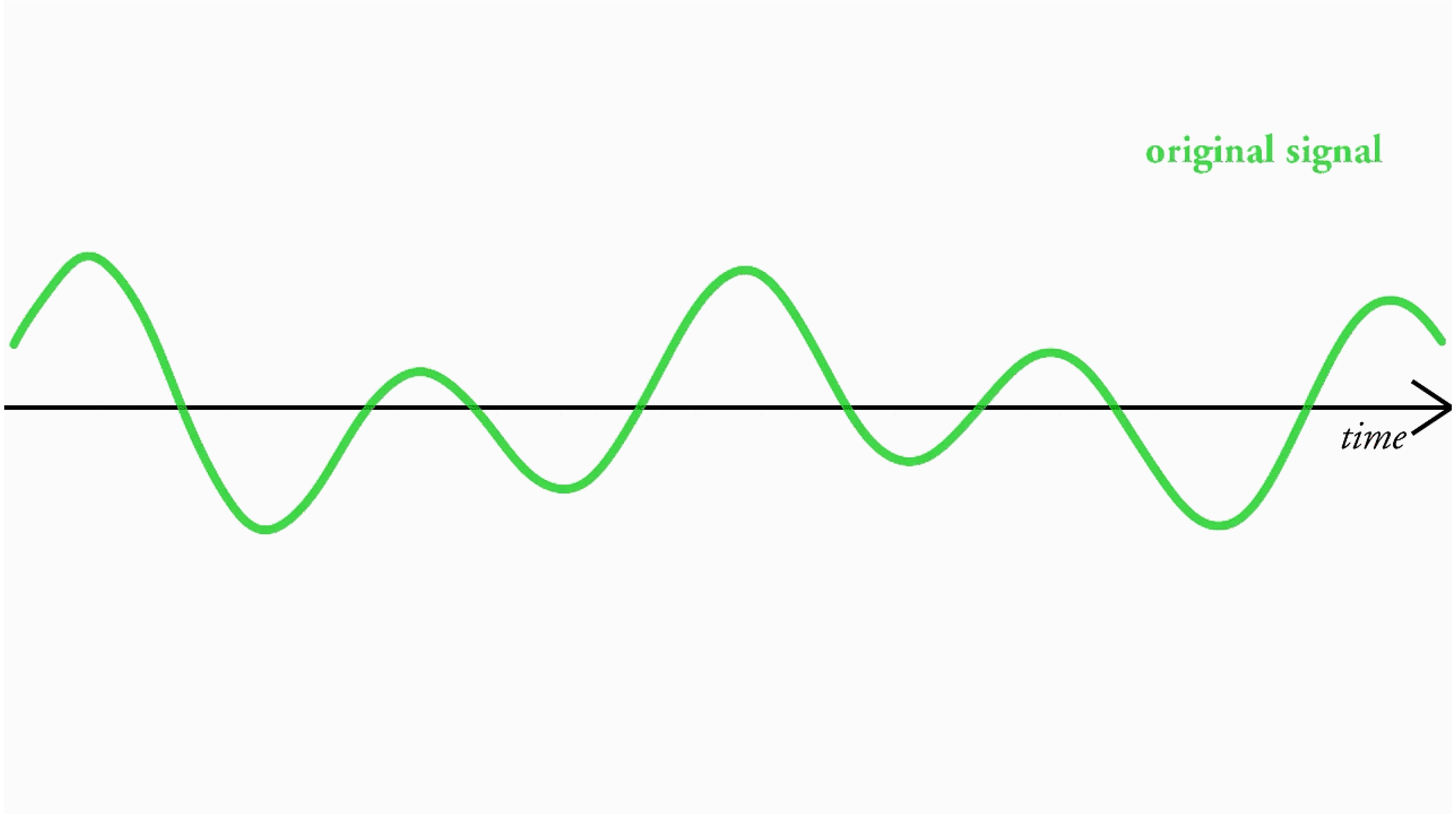
0:00 / 0:14

Difference between 16 bit and 8 bit (with dither)

0:00 / 0:14

Difference between 16 bit and 8 bit (no dither)

Quantization Noise



Dither

0:00 / 0:03

1 kHz sine wave @ 16 bit 44.1 kHz

0:00 / 0:03

1 kHz sine wave @ 8 bit 44.1 kHz (with dither)

0:00 / 0:03

1 kHz sine wave @ 8 bit 44.1 kHz (no dither)

Analog comparison

Format	**Dynamic range**	**Effective Bit Depth**
Cassette	40 dB	6 bits
Vinyl	60 dB	10 bits
Reel-to-Reel	80 dB	13 bits
CD	96 dB	16 bits
HD Audio	144 dB	24 bits

* These are estimates -> analog hardware performance varies

Further Reading

- Sigma-Delta converter (PDM)
- Error-correction codes (EFM, Reed-Solomon, etc.)
- Perceptual Audio Coding (MP3, AAC, etc.)
- Relevant ECE Courses
  - ECE 3300: Signals & Systems
  - ECE 4270: Communcation Systems
  - ECE 3170: Random Signal Analysis
  - ECE 4670: Digital Signal Processing

Next Talk - February 5

Spectral Analysis

Decomposing audio with algorithms