

Audio Signal

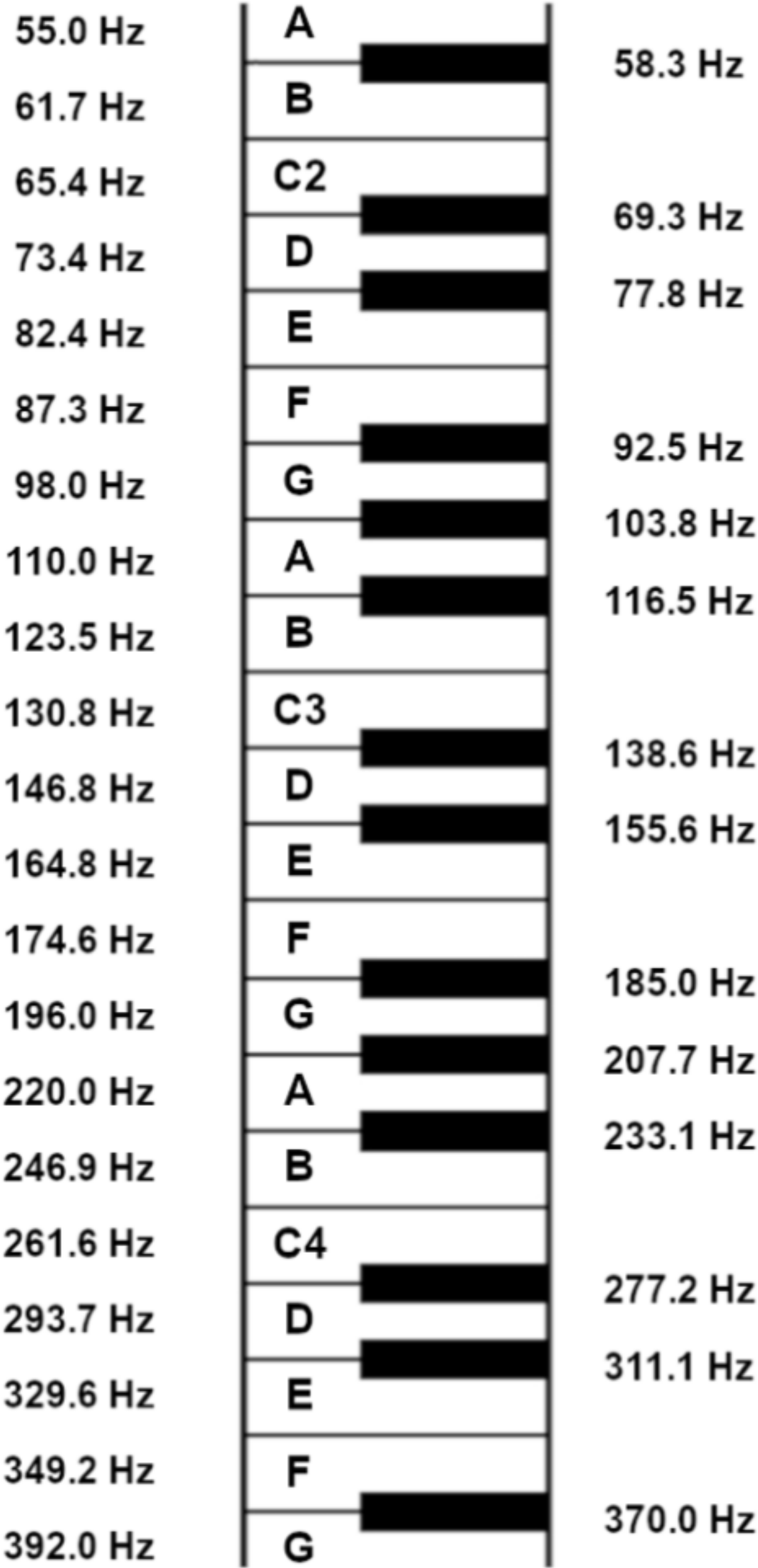
An introduction

Seanghay Yath

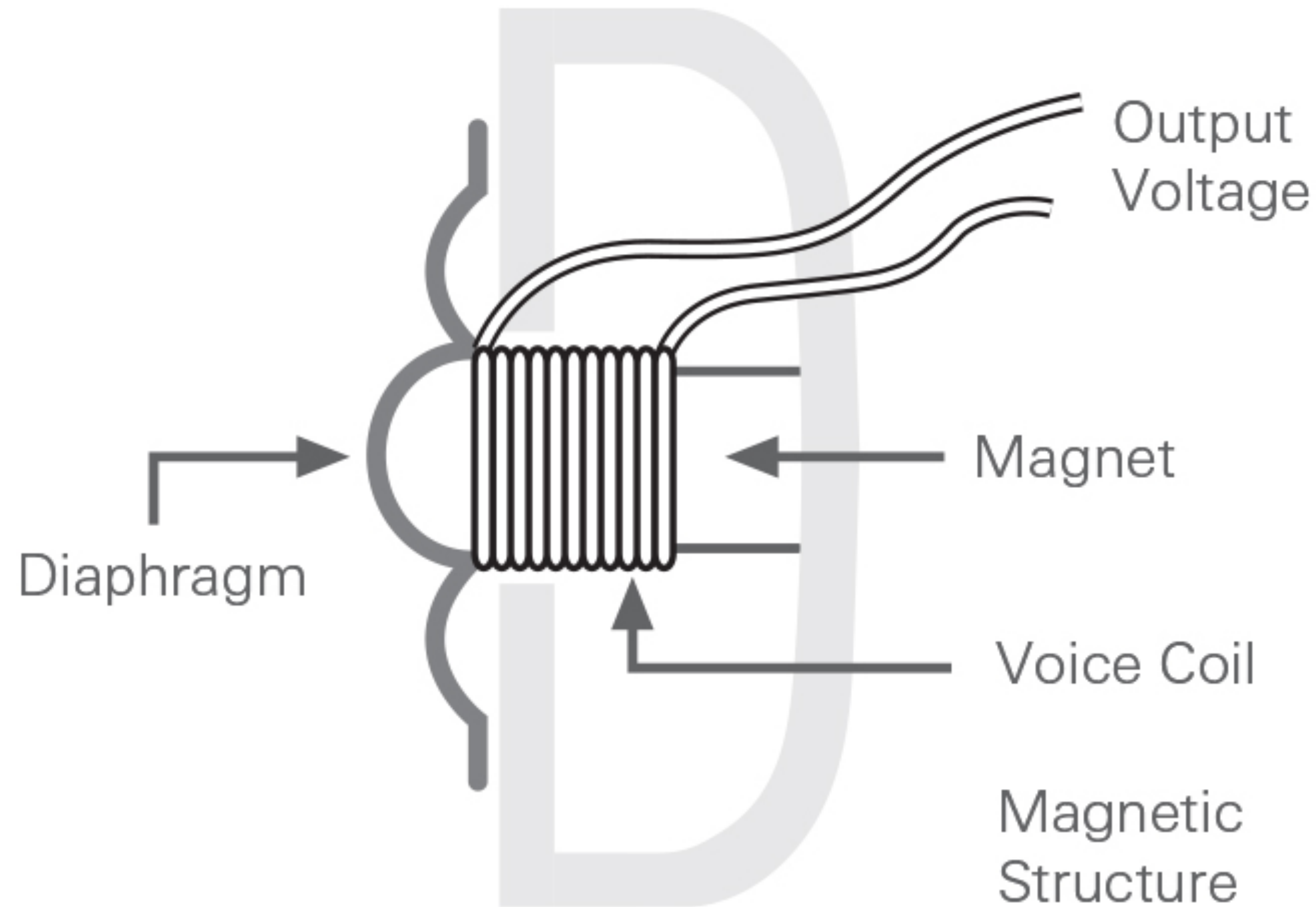
What is sound?

In physics, sound is a vibration that propagates as an acoustic wave through a transmission medium such as a gas, liquid or solid. Sound is measured in Hertz.

<https://ciechanow.ski/sound>



Digital Sound Recording



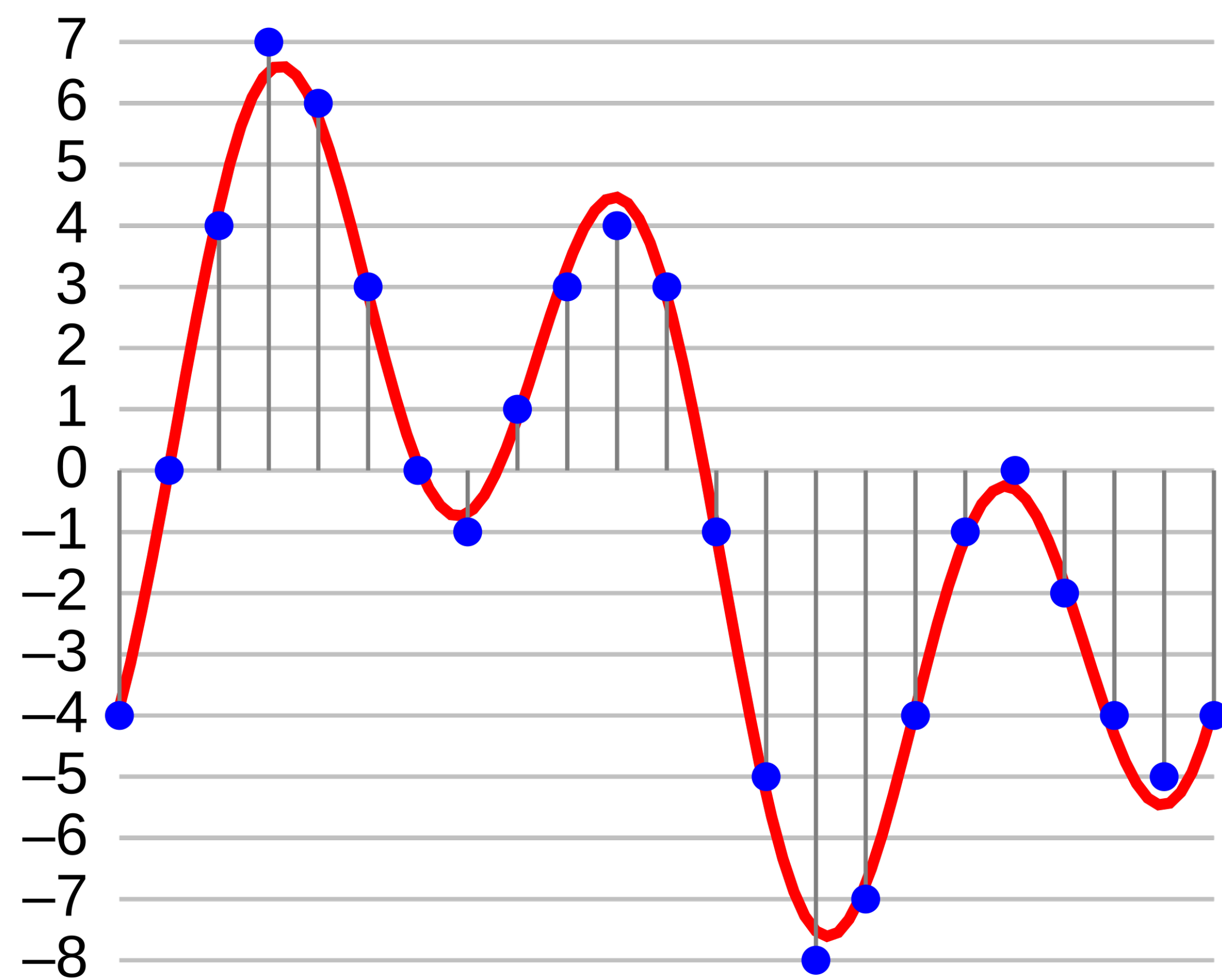
Output voltage from microphone is AC (Analog Signal)

It's not possible to store analog signal digitally. So we need to get creative.

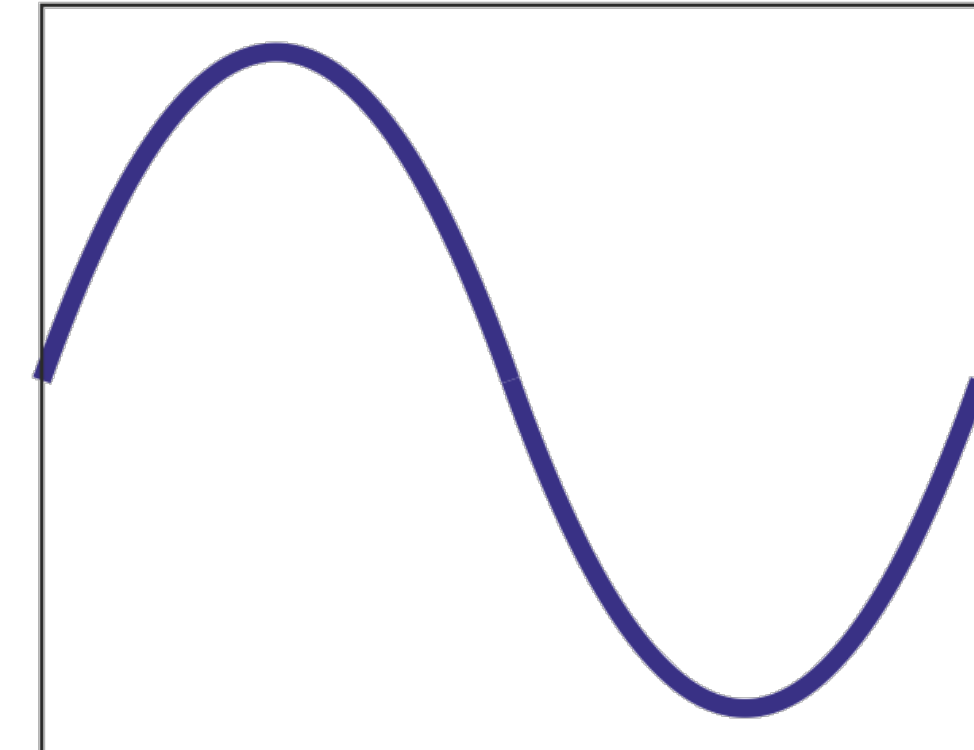
Digital Sound Recording



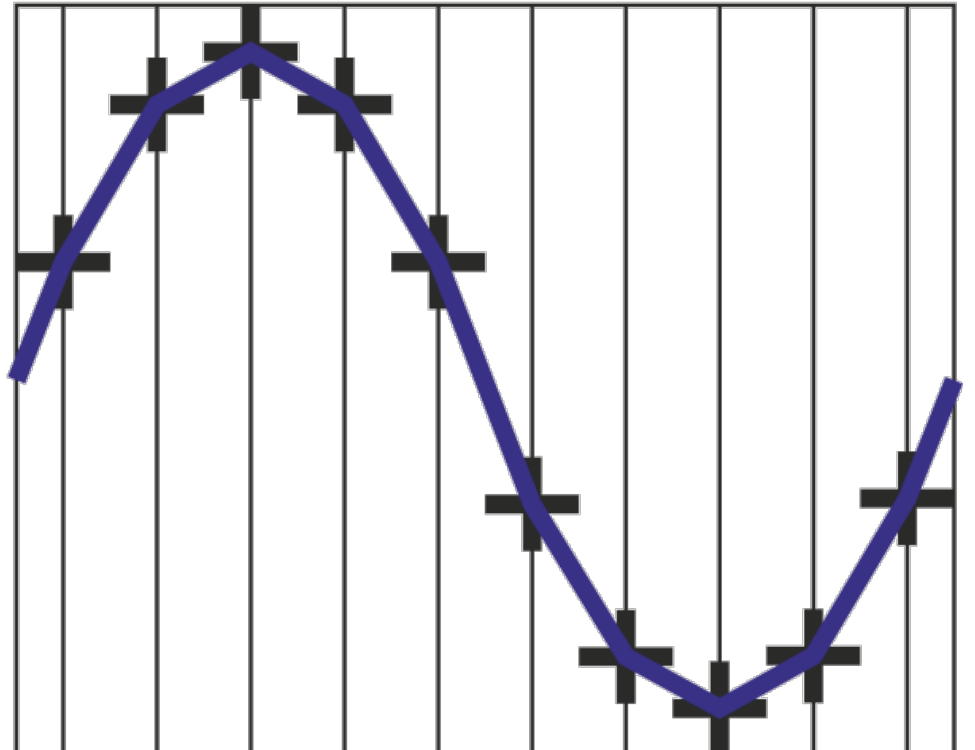
Digital Sound Recording



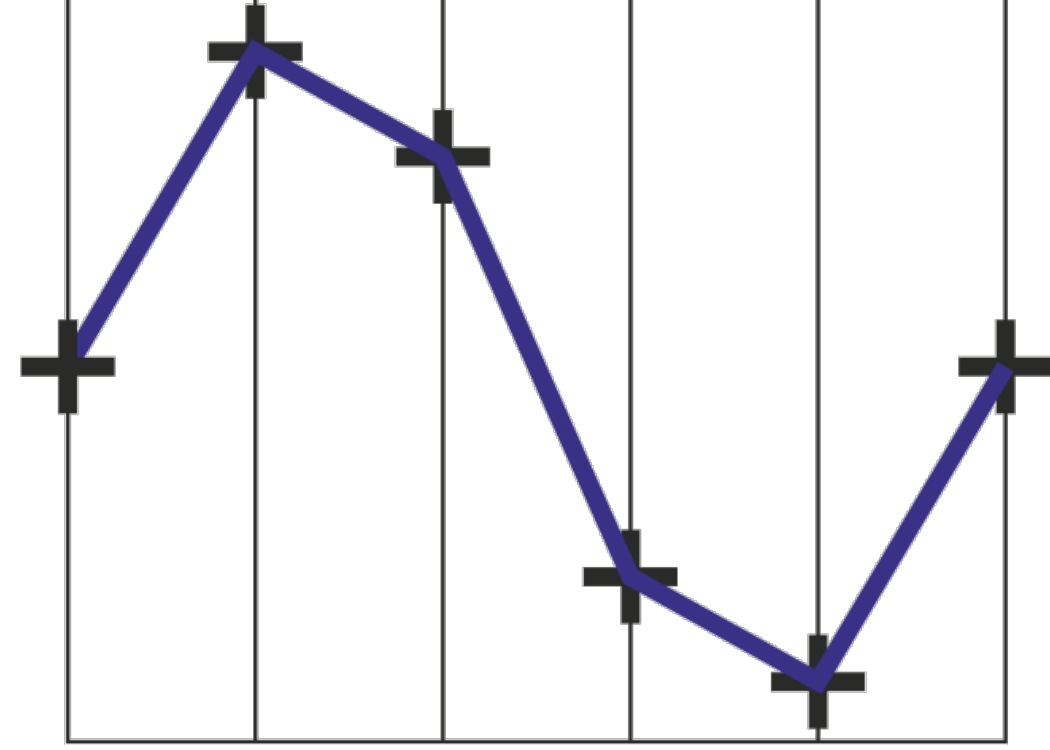
Original Waveform



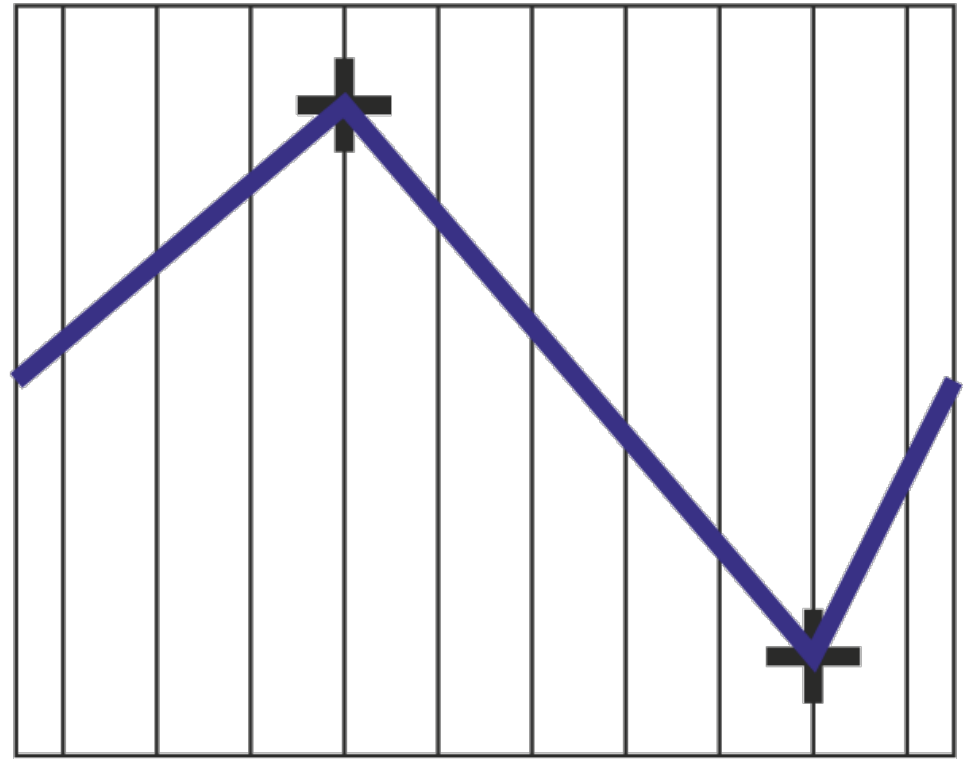
Sampled at 10 points



Sampled at 6 points



Sampled at 2 points

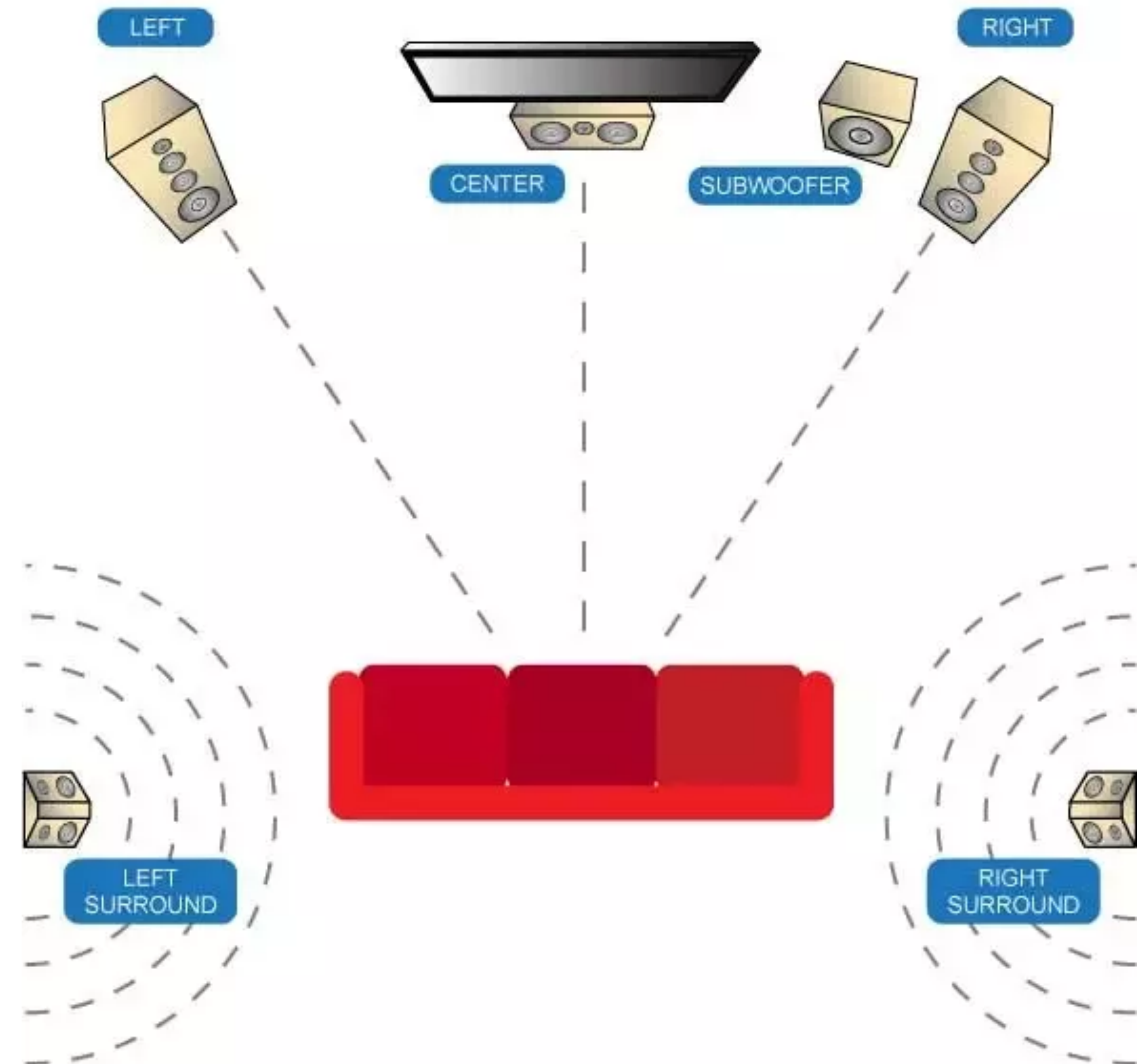


Audio Channels

1-Channel = Mono

2-Channel = Stereo

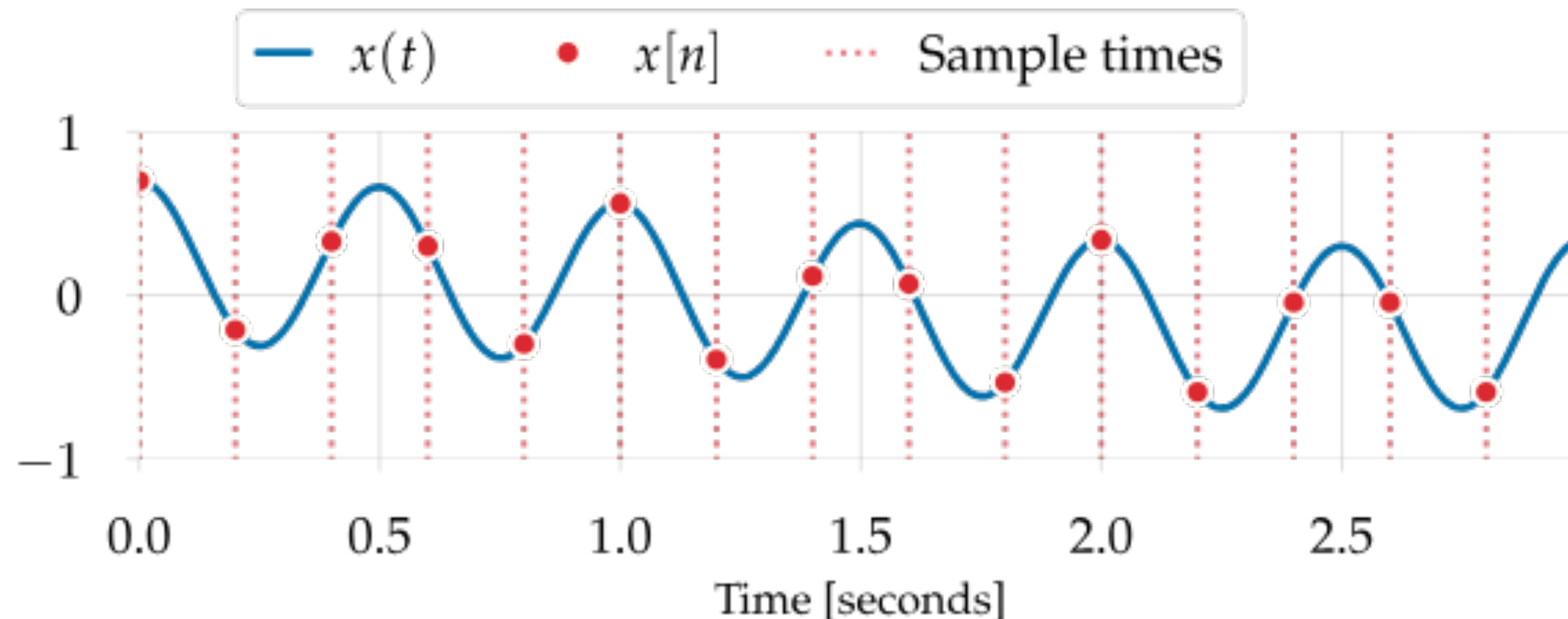
5.1 Surround



Sample Rate

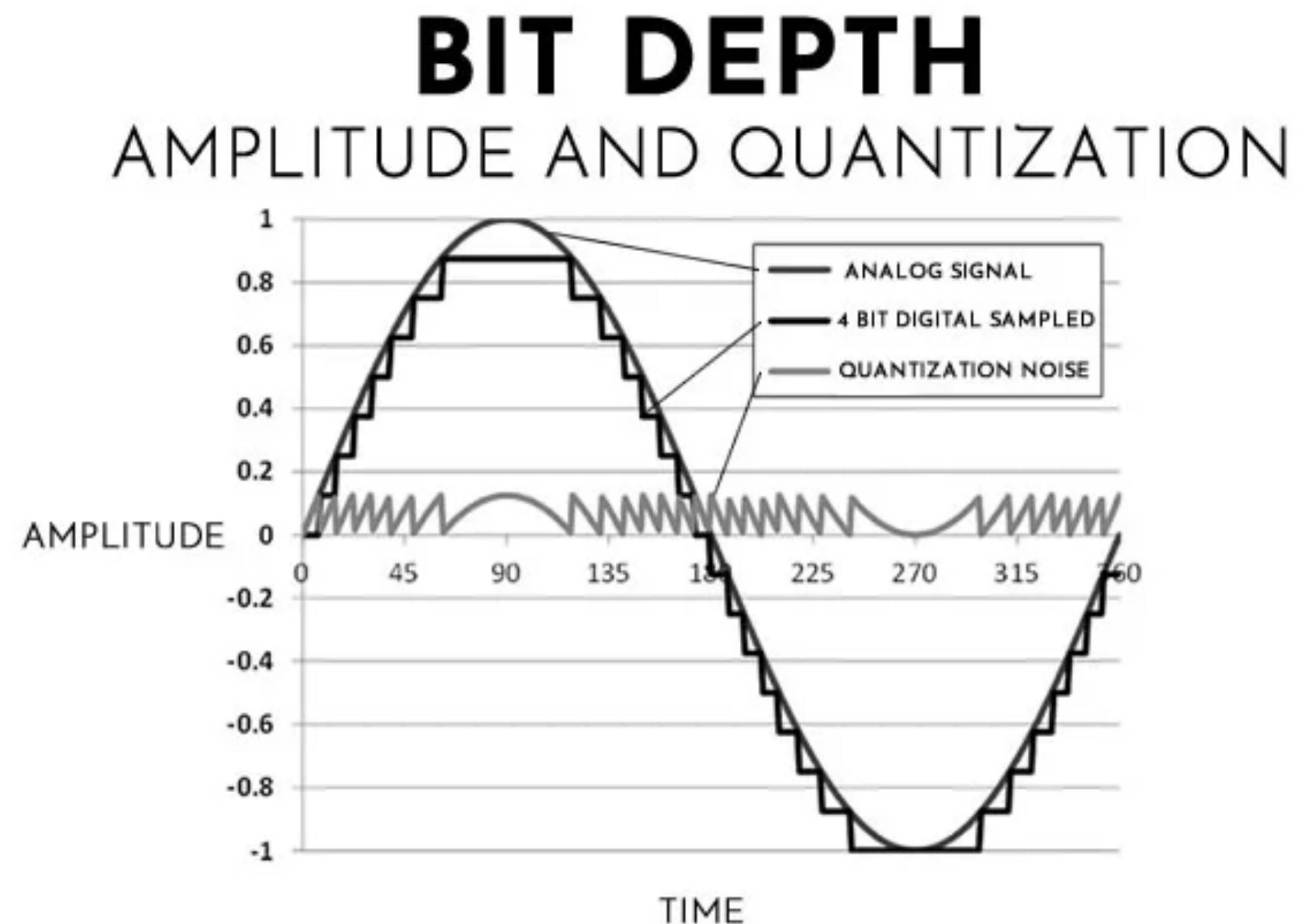
Sampling rate or sampling frequency defines the number of samples per second (or per other unit) taken from a continuous signal to make a discrete or digital signal. e.g. 16kHz, 22.05kHz, 44.1kHz 48kHz

Audio Duration In Seconds = Number of Samples / Sample Rate



Bit Depth / Sample Width

Bit depth is the number of bits of information in each sample, and it directly corresponds to the resolution of each sample. e.g. 32-bit 16-bit, 8-bit,



Commonly used

32-Bit float32

16-Bit int16

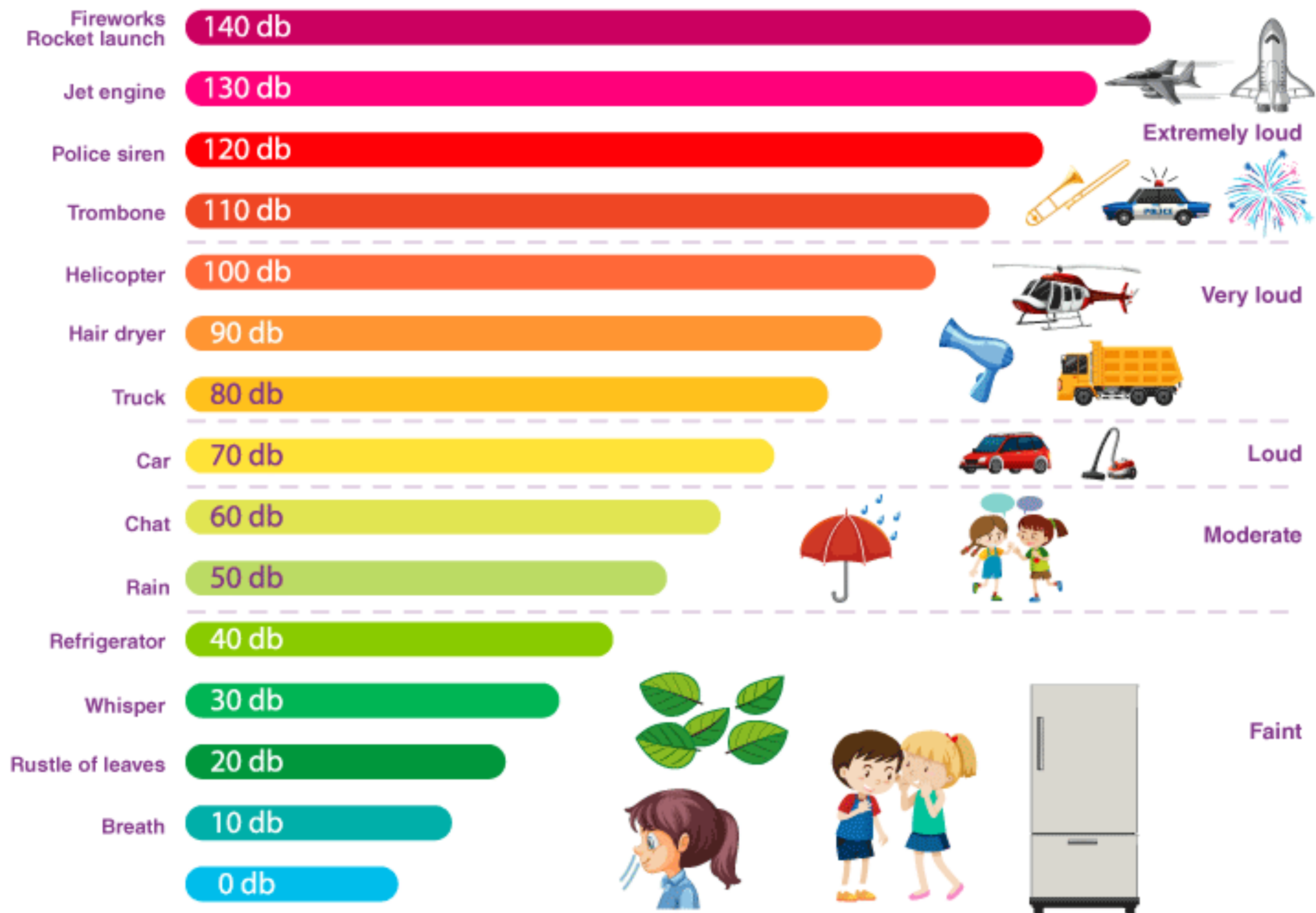
How does it look like?

```
audio = [  
    [0.3, 0.1, ..., 0.1, .8, .75],  
    [0.45, 0.2, ..., 0.3, .23, .1],  
]
```

Pulse-code modulation (PCM)

Pulse-code modulation is a method used to digitally represent analog signals. It is the standard form of digital audio in computers, compact discs, digital telephony and other digital audio applications.

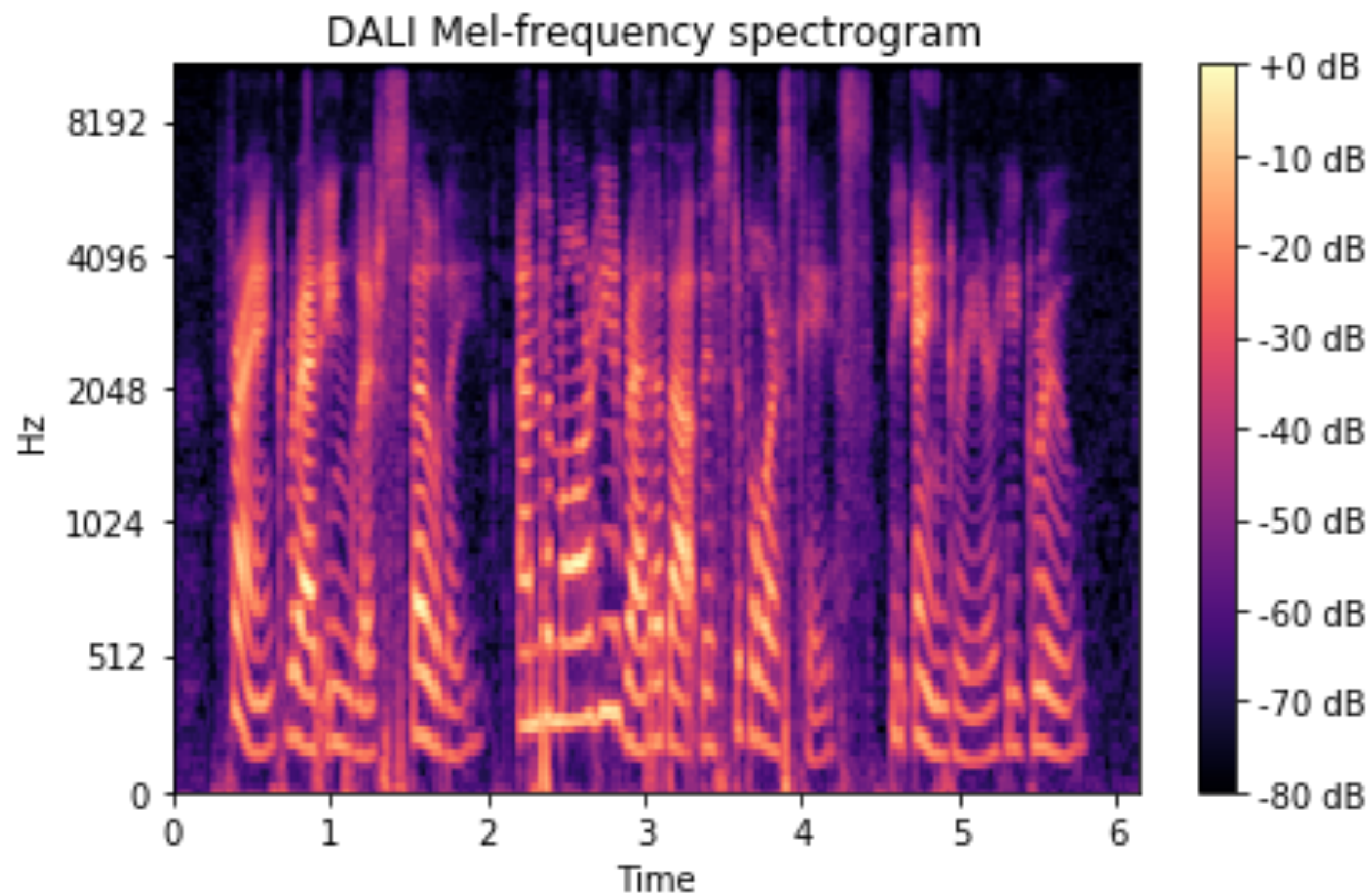
Decibel Scale (dB)



Mel-scale (Melody Scale)

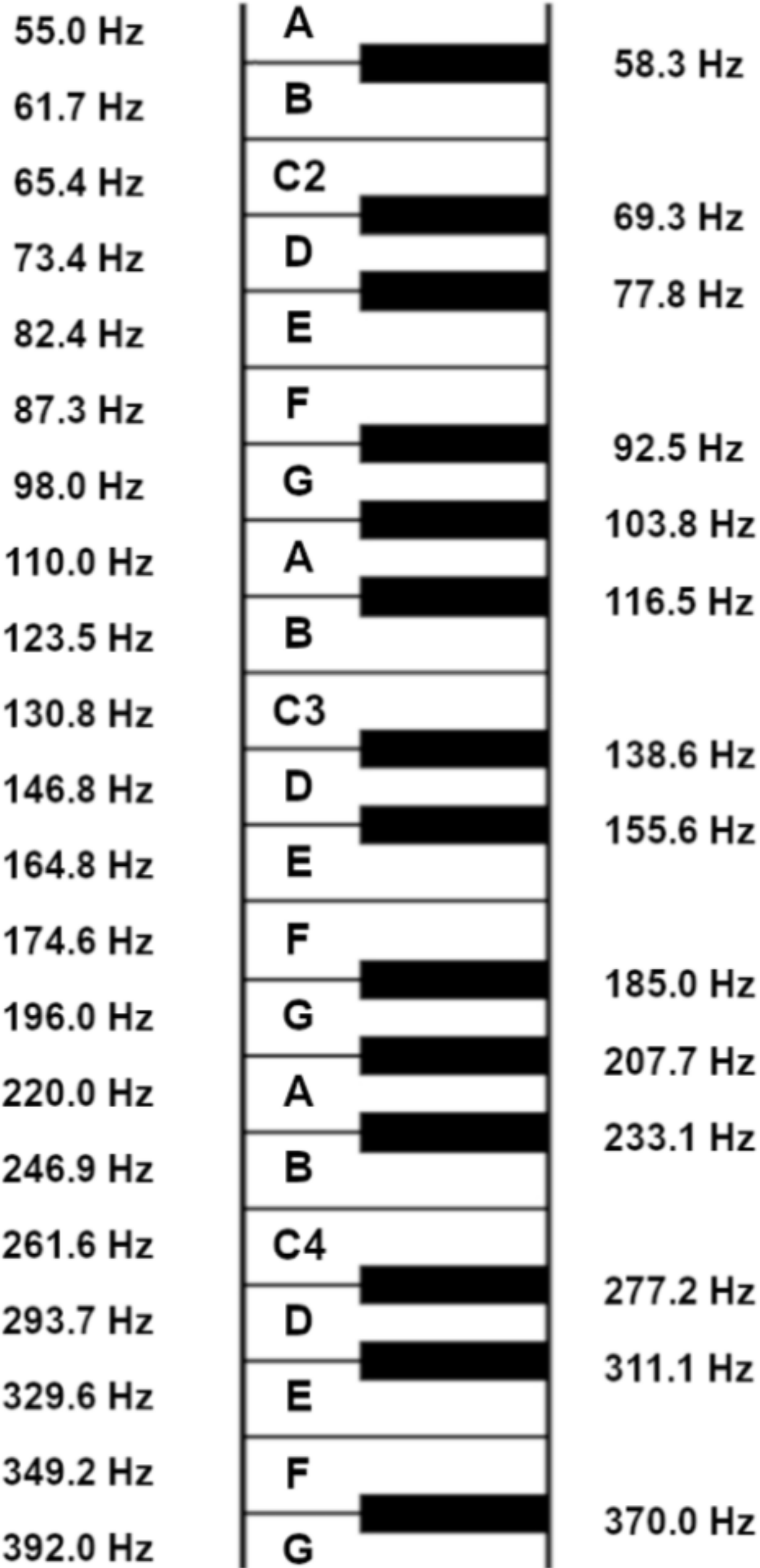
$$m = 1127 \cdot \log \left(1 + \frac{f}{700} \right)$$

The mel scale is a (human) perceptual scale of pitches judged by listeners to be equal in distance from one another. The reference point between this scale and normal frequency measurement is defined by assigning a perceptual pitch of 1000 mels to a 1000 Hz tone, 40 dB above the listener's threshold.



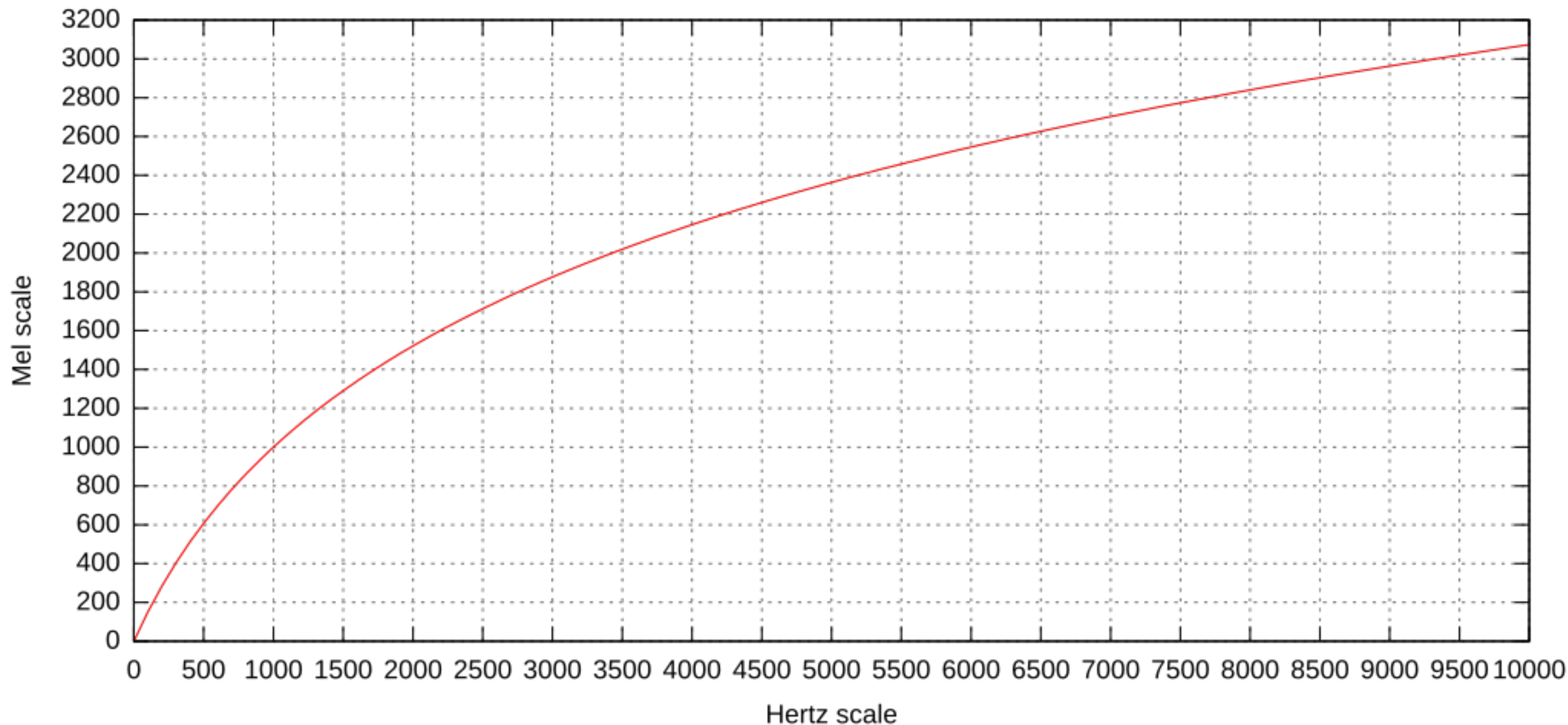
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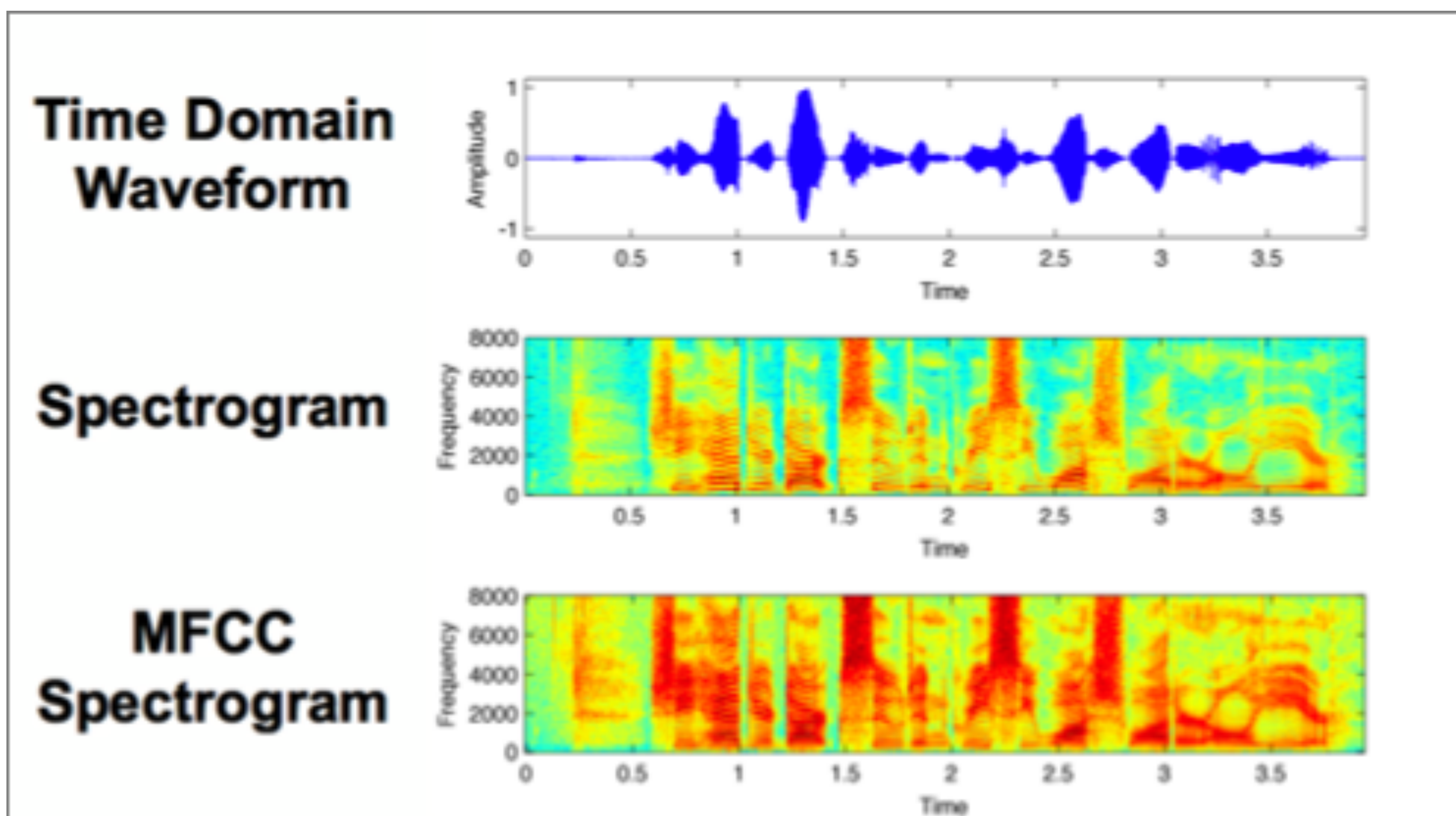


Mel-scale (Melody Scale)

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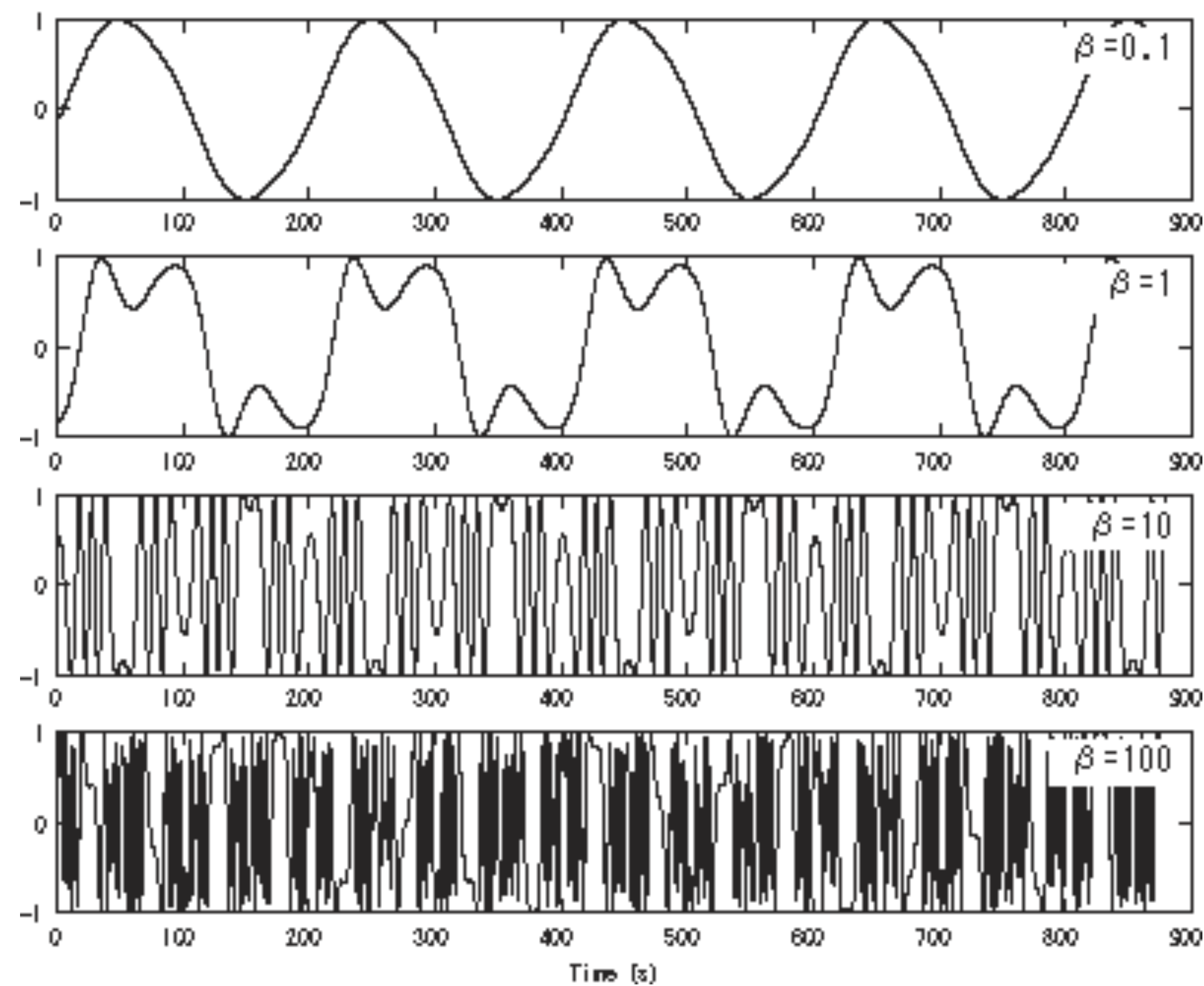
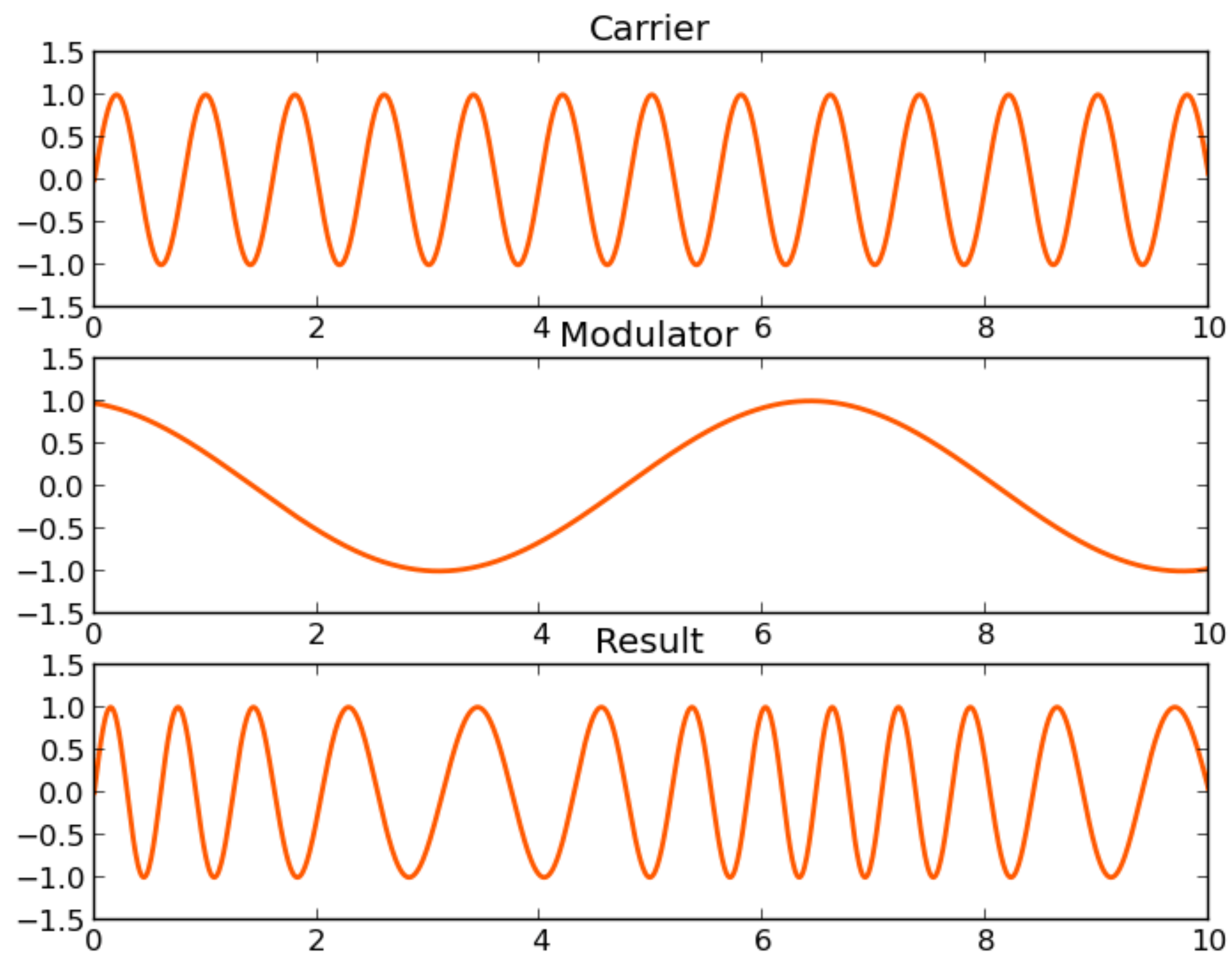
Mel-frequency cepstral coefficients (MFCCs)



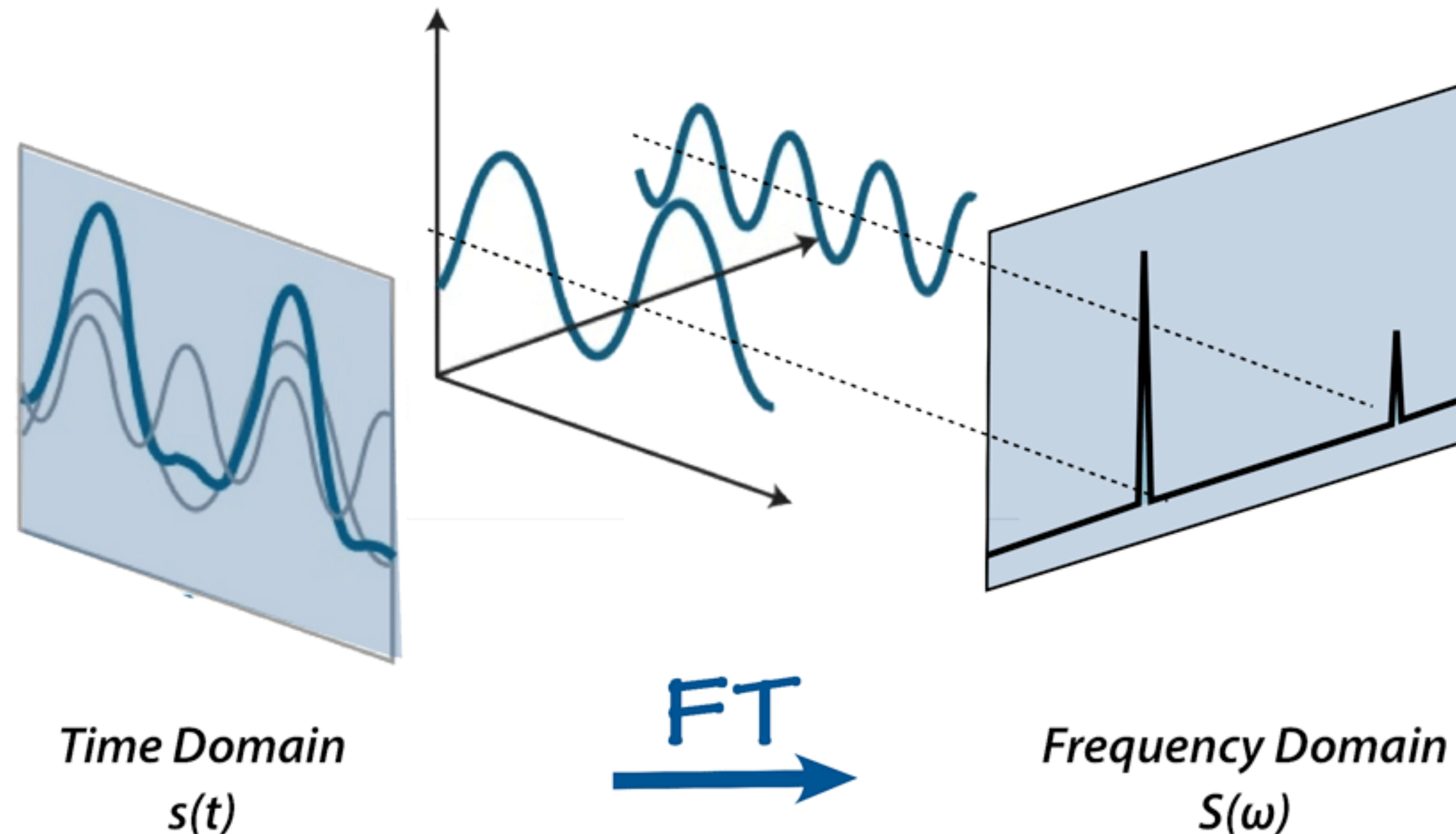
Mel-spectrograms are often used as inputs to deep learning models, which can learn relevant features directly. MFCCs are typically used in more traditional machine learning models where dimensionality reduction is beneficial.

<https://maelfabien.github.io/machinelearning/Speech9/>

Modulation



Fourier Transform (FFT, STFT, InverseSTFT)



<https://www.youtube.com/watch?v=spUNpyF58BY>

Sound Storage

Choose the Right!

Waveform Audio File Format (WAV, WAVE)

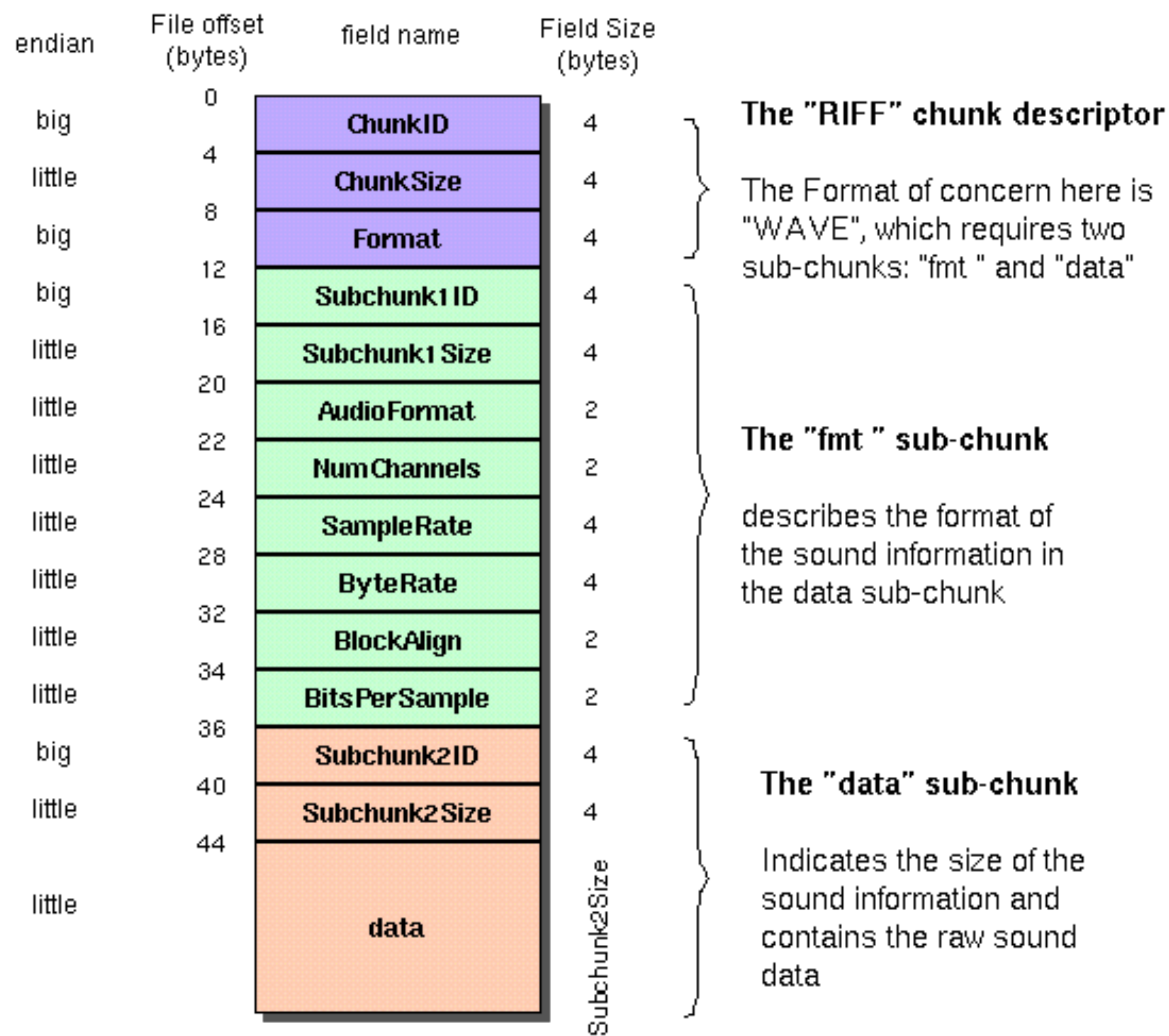
Offset	0	1	2	3	4	5	6	7	8	9	A	B	C	D	E	F	ASCII
00000000	52	49	46	46	BA	10	00	00	57	41	56	45	66	6D	74	20	<u>RIFF</u> e... <u>WAVE</u> fmt
00000010	10	00	00	00	01	00	07	00	11	2B	00	00	11	2B	00	00	+...+...
00000020					00	64			4B	10	00		78	00	80	80	dataK... FFB
00000030					7C	80			7C	8			7C	7C	7C	7C	B FFB FFB
00000040					7C	80	80	80	7C	7C	7C	7C	80	80	80	80	FFFF FFFFFFF

RIFF
Signature

Data Size:
4,282

WAVE Signature

The Canonical WAVE file format



FLAC - Free Lossless Audio Codec

FLAC is an audio coding format for lossless compression of digital audio, developed by the Xiph.Org Foundation.

FLAC compresses files at a ratio of approximately 2:1, reducing the size of the original by **50%**.

MP3 - MPEG-1/2 Audio Layer III

Audio coding format for digital audio that offers **Lossy Compression**.
Think of it as a zip file.

Requires LAME MP3 Audio Encoder / Decoder

Bit Rate is the amount of data processed per second in an audio file, measured in kilobits per second (kbps). It directly affects the file size and the audio quality. Higher is better. Higher results in larger file size.

Advanced Audio Coding (AAC)

an audio coding standard for lossy digital audio compression.

OGG

Ogg Vorbis is a fully open, non-proprietary, patent-and-royalty-free, general-purpose compressed audio format for mid to high quality (8kHz-48.0kHz, 16+ bit, polyphonic) audio and music at fixed and variable bitrates from 16 to 128 kbps/channel. This places Vorbis in the same competitive class as audio representations such as MPEG-4 (AAC), and similar to, but higher performance than MPEG-1/2 audio layer 3, MPEG-4 audio (TwinVQ), WMA and PAC.

Audio Library for ML

- librosa
- torchaudio
- audiomentations
- scipy
- numpy
- pydub
- sox
- soundfile
- pedalboard

Audio Signal Processing Tools

- FFmpeg
- Audacity
- Praat

<https://github.com/Yuan-ManX/audio-development-tools>

End.