DATA 3464: Fundamentals of Data Processing

Signals and Audio

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March 17, 2026

Topic overview

- Introduction to signals
- Audio as a 1D signal
- File formats
- A brief intro to signal processing

Resources used:

- Various textbooks from my undergrad
- DSPguide.com seems like a pretty good resource

What is a signal?

> *"A [continuous/discrete] signal is a function of independent variables that range over [a continuum/discrete] values" - Jerry L. Prince, Medical Imaging Signals and Systems*

- Common notation: $x(t)$ for continuous, $x[n]$ for discrete
- Signals are **discretized** by **sampling** at some fixed interval dt
- The **sampling rate** is informed by the frequency content of the data:

$$f_s \geq 2f_{max}$$

(but in practice is much higher)

Frequency content of a signal

- A discrete time domain signal can be represented as:

$$x[n] = \sum_{k=0}^{N-1} \left[a_k \cos\left(\frac{2\pi kn}{N}\right) + b_k \sin\left(\frac{2\pi kn}{N}\right) \right]$$

- Or, using Euler's formula $e^{j\theta} = \cos\theta + j\sin\theta$:

$$x[n] = \sum_{k=0}^{N-1} c_k e^{j\frac{2\pi kn}{N}}$$

where the complex coefficients $c_k = a_k + jb_k$ and $j = \sqrt{-1}$

Fourier Transform

- To figure out what the coefficients c_k are, we can use the **Discrete Fourier Transform** (DFT):

$$X[k] = \sum_{n=0}^{N-1} x[n] e^{-j2\pi \frac{k}{N} n}$$

  where each element of $X[k]$ is the coefficient c_k for frequency k

- This can also be inverted to get back the original signal:

$$x[n] = \frac{1}{N} \sum_{k=0}^{N-1} X[k] e^{j2\pi \frac{k}{N} n}$$

This is skipping over several entire math courses

Where we left off on March 17

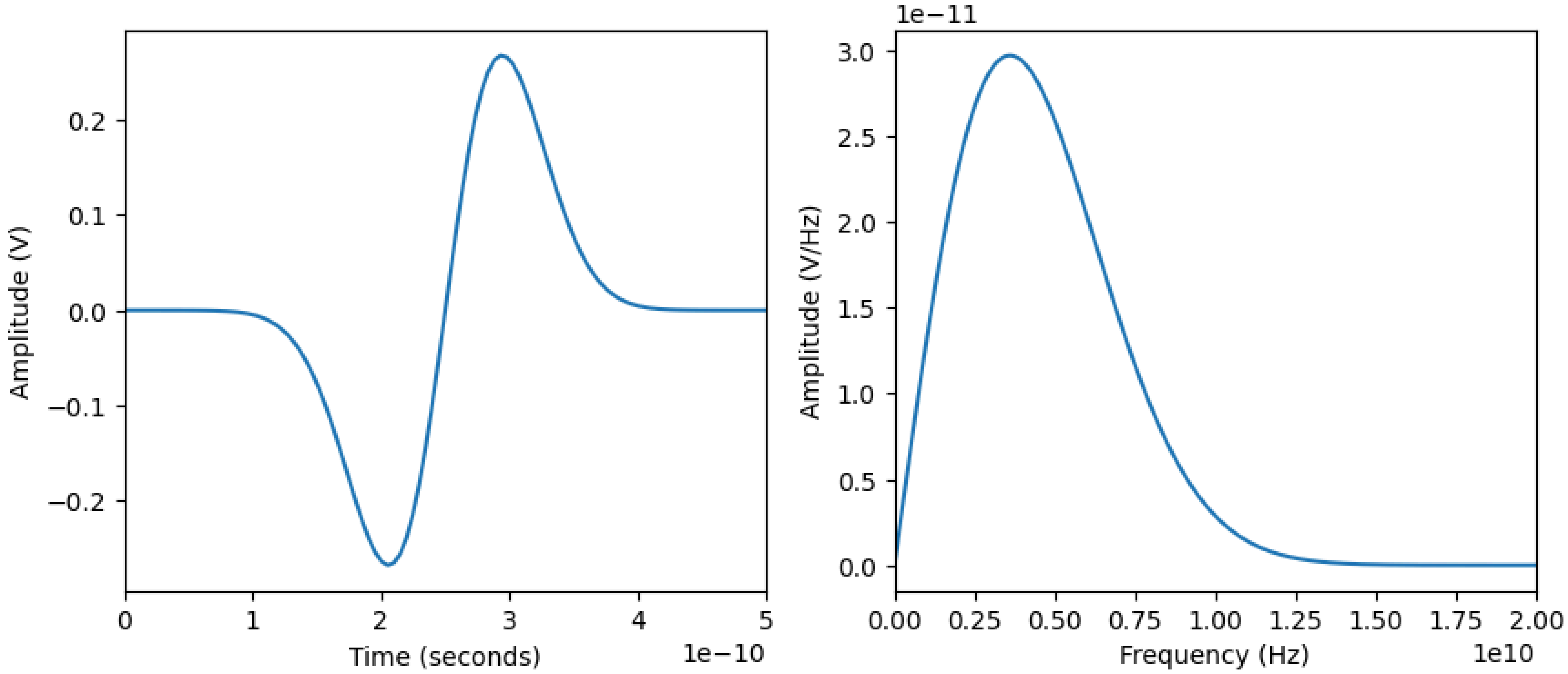
Symmetry in the frequency domain

- Since a real-valued signal in time is composed of both sine and cosine components, its DFT has **conjugate symmetry**

  $$X[N-k] = X[k]^*$$

  where * denotes the complex conjugate
- This means the **negative-frequency half** of the spectrum is redundant
- In practice, for real-valued data, we often only inspect:
  - **magnitude**: $|X[k]|$ to see "how much" of each frequency is present
  - **phase**: $\angle X[k]$ to see alignment/shift information

Frequency vs Time Domains



- $f = \frac{1}{t} \implies$ short time = high frequency, small frequency = long time

Example signal: Audio

- Once you think of a signal as being a weighted sum of frequency components, you can do some fun things with it
- We can extract information, downsample, remove noise, etc
- Example: a typical .wav file
  - Uncompressed
  - 16 bits per sample (bit depth)
  - 48 kHz sampling rate
  - mono (1 channel) or stereo (2 channels)

> *What about .mp3? .ogg? I would use ffmpeg to convert to .wav*

Preparing data

- Assuming we're starting with a collection of audio files, we can either:
    - Extract features and save as tabular data
    - Use the raw audio signal as input
- We can preprocess and store the data, or preprocess on the fly

> *What considerations might go into this decision?*
>
> *What should always be stored regardless of the approach?*

Preparing audio data

- Data for learning tasks is easiest to work with if it is **consistent**
- For audio signals, this could include:
  - Decompressing and converting to .wav
  - Downsampling
  - Aligning and cropping primary signal
  - Converting to mono/stereo
  - Extracting features
- librosa can help with this (and can apparently handle mp3 too!)

Coming up next

- 2D signals (aka images)
- Strategies and software for labelling data

> *By next week you should have some idea of what kind of dataset you want to curate and label for Assignment 3*