In the Name of God

Signals and Systems

Project Phase 1

Spring Semester 1403-04
Department of Electrical Engineering

Sharif University of Technology
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Due Date: 1404/03/02

Important Notes

1. Feel free to reach out with any questions about the project via Telegram or by emailing me at Gmail.

2. Please submit your project in the form of a single ZIP file containing all the necessary files. The file should be named GroupXX.zip where XX is your group number.

3. Groups should be formed with a maximum of two students.

4. The project carries a total of 100 points. Bonus points, marked with (∗), will be added to your final score. The maximum score for each question is also provided.

1 Project Description and Theoretical Questions

1.1 Introduction to Singal Noise Cancelling

1.1.1 Noise Models

In many real-world applications, we encounter signals that are corrupted by noise. In such cases, we need to remove the noise to extract the original signal. This process is called **signal noise cancelling**.

Here is a simple model for signal and noise:

$$y(t) = s(t) + n(t) \tag{1}$$

where y(t) is the observed signal, s(t) is the original signal, and n(t) is the noise, which is assumed to be independent of the signal. Noices are usually modeled as random variables, such as:

- Gaussian noise: $n(t)$ where : $n(f) \sim N(0, \sigma 2)$
- Uniform noise: $n(t)$ Where : $n(f) \sim U(-A, A)$
- White noise: $n(t)$ Where : $n(f) \sim N(0, \sigma 2)$
- Impulse noise: $n(t) = \delta(t)$
- Power line noise: $n(t) = A \sin(2\pi f t)$

In Iran, power line noise is usually modeled as a 50 Hz sine wave.

1.1.2 Noise Cancelling Methods

There are several methods for noise cancelling. Here are some of them:

- **FIR Filtering:** In this method, we use a finite impulse response (FIR) filter to remove the noise. FIR filters are easy to design and implement.
- **IIR Filtering:** In this method, we use an infinite impulse response (IIR) filter to remove the noise. IIR filters are more efficient than FIR filters but are harder to design.
- **Adaptive Filtering:** In this method, we use an adaptive filter to remove the noise. The filter coefficients are updated based on the error between the observed signal and the estimated signal.
- **(∗) Wavelet Transform:** In this method, we use wavelet transform to remove the noise. Wavelet transform is a powerful tool for signal processing and noise cancelling.

1.1.3 Questions

1. Conduct research on the introduced noise models and describe them in your own words. Provide an explanation of these models in the context of discrete-time signals, and formulate them within the discrete-time domain. (5)

2. Research the introduced noise-canceling methods and explain them in your own words. Make sure to formulate these methods using mathematical equations. Additionally, discuss the advantages and disadvantages of each method. (Note: You do not need to explain the Adaptive Filtering method in this question.) (10)

1.2 Eigen Signals

1.2.1 Introduction

Eigen signals are a group of mutually orthogonal signals. They are utilized in signal processing to represent signals in a more compact and efficient manner. These signals have numerous applications in signal processing, including image compression, speech recognition, and noise cancellation.

1.2.2 Questions

1. Explain the concept of eigen signals in your own words. How are eigen signals used in signal processing? What are the advantages of using eigen signals? (5)

1.3 Adaptive Filtering

1.3.1 Introduction

Adaptive filtering is an effective technique for signal processing and noise cancellation. In this method, the filter coefficients are adjusted based on the error between the observed signal and the estimated signal. This dynamic adjustment enables the filter to adapt to variations in the signal and noise. Adaptive filtering is widely applied in various signal processing tasks, such as echo cancellation, noise reduction, and signal equalization.

1.3.2 Questions

1. Provide the mathematical formulation of the adaptive filtering algorithm. Clearly explain each step of the algorithm in your own words, ensuring specificity and clarity in every sentence. Using pictures and diagrams is strongly recommended for better understanding. Additionally, explain how eigen signals can be utilized in adaptive filtering. (5)
2. In adaptive filtering, selecting an appropriate loss function is essential for updating the filter coefficients. Describe the various loss functions that can be employed in adaptive filtering, with a particular emphasis on the LMS (Least Mean Squares) loss function. (5)
3. What is the primary advantage of adaptive filtering compared to simple filtering? Specifically, explain how adaptive filtering can be applied to remove power line noise from a signal and why it is more beneficial than using simple notch filtering. (5)

4. Research the convergence of the adaptive filtering algorithm. Explain the Gradient Descent and Stochastic Gradient Descent algorithms, and compare them based on their convergence speed. (5)
5. Research Active Noise Cancelling (ANC) and explain how it can be utilized to eliminate noise from a signal. Including pictures and diagrams is strongly recommended for better illustration. (5)

1.4 Decode a Hidden Text in an Audio File Using LSB Steganography

1.4.1 Introduction

In this task, you will decode a hidden message from an audio file that has been encoded using the Least Significant Bit (LSB) technique. The message was embedded by modifying the least significant bits of the audio samples, ensuring the audio quality remains mostly unaffected.

Your goal is to extract and retrieve the hidden text message from the given audio file.

1.4.2 Questions

1. Research the Least Significant Bit (LSB) method for hiding data in digital media, specifically in audio files. Why is this method useful for steganography? What are its main benefits and potential disadvantages? (5)

2 Implementation

In this subsection, you are required to implement the adaptive filtering algorithm to remove power line noise from a signal. You're ought to implement the following steps:

Noise cancelling :

1) Record your own voice for 10 up to 15 seconds. You can use your phone or any other recording device, but you have to turn off noise-cancelling features.
2) Add power line noise to the recorded signal. Write proper formulation for the power line noise and add it to the recorded signal. (10)
3) Implement the adaptive filtering algorithm to remove the power line noise from the recorded signal. Write the code in a clear and organized manner; use comments to explain the code and writing functions for building blocks is required. Use the LMS loss function and Gradient Descent algorithm. (10)
4) Implement FIR and IIR filters to remove the power line noise from the recorded signal. Compare the results of the adaptive filter with the FIR and IIR filters. (10)
5) Plot the original signal, the noisy signal, and the filtered signals. Write functions to play the audio signals. (5)

Decoding :

You have been provided with two encoded audio files. Your task is to uncover the hidden message within them. The message is the same in both files, except one contains noise while the other is noise-free. To decode the files, you need two pieces of information: the attenuation level of the hidden message and the index of the samples for extraction. These numbers are embedded in the filenames—the first indicates the message attenuation level, and the second specifies a **sampling multiple**. You are required to select one file and extract the hidden message.

- When amplifying the signal with a gain of 100, ensure the result is rounded to enable representation in binary format.
- Negative numbers are represented in two's complement format.

Noise-free :

1) Extract the hidden message. (15)

(*) Noisy audio :

It is recommended to first decode the hidden message from the noise-free file before attempting this part.

1) Plot the signal in the frequency domain. Determine the noise frequencies in the audio by performing an analysis of its frequency spectrum using FFT. (5)
2) Based on the previous section (noise cancelling), design a filter to remove the noise effectively. Note that you must adjust the order or quality factor of your filter in such a way that the samples remain unchanged. By comparing the FFT of the signal before and after filtering, you can design the filter effectively. After applying the filter, plot the signal in both the time and frequency domains to ensure that the noise has been effectively removed. If you are using a filter that was not mentioned in the previous section, be sure to first include an explanation of that filter in your report file. (10)
3) Extract the hidden message. (15)

You have to write a full report of your implementation. The report should include the following properties:

- Introduction: A brief introduction to the problem and the methods you used.
- Methodology: A detailed explanation of the methods you used, including the adaptive filtering algorithm, the FIR and IIR filters, the power line noise model and also all the filters and algorithms you have designed and used for the decoding section.
- Results: A detailed explanation of the results, including the original signal, the noisy signal, and the filtered signals for first section and also extracted message in the decoding section.
- Conclusion: A conclusion summarizing the results and discussing the advantages and disadvantages of the methods you used.

Put the code files, the recorded signal, and the report, screenshots of the plots in proper folders in the ZIP file. The report should be in PDF format and named Report.pdf. The code files should be in a folder named Code and etc. There is (5) bonus points for commenting the code properly and orderliness of the files as requested.