Signal Processing and System Design Laboratory

Experiment 3: Denoising Noisy Audio Speech Signal using Recursive Least Squares (RLS) Method

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Objective

The objective of this experiment is to understand and implement the Recursive Least Squares (RLS) method for denoising noisy speech signals. Students will learn to apply the RLS adaptive filter on a noisy speech corpus and evaluate the performance of the algorithm by calculating the error (Signal-to-Distortion Ratio, SDR) for different levels of SNR (0dB, 5dB, 10dB, and 15dB).

Introduction

Speech signals often get corrupted by noise, which significantly degrades the quality and intelligibility of the speech. Denoising these signals is crucial in many applications such as telecommunication, hearing aids, and speech recognition systems. The Recursive Least Squares (RLS) algorithm is an adaptive filter technique widely used for denoising because of its fast convergence and low sensitivity to noise.

In this experiment, we will work with a noisy speech corpus (NOIZEUS) designed to evaluate speech enhancement algorithms. This corpus contains IEEE sentences corrupted by real-world noise at various Signal-to-Noise Ratios (SNRs). The noises include suburban train noise, babble, car, exhibition hall, restaurant, street, airport, and train station noise.

RLS Method Overview

The Recursive Least Squares (RLS) algorithm seeks to minimize the least squares error between the desired and estimated signals by recursively updating the filter coefficients. Unlike the Least Mean Squares (LMS) algorithm, RLS provides faster convergence and better performance in the presence of noise.

Algorithm for RLS

Below is the algorithm for implementing the RLS method:

Mathematical Formulation

The RLS algorithm is mathematically represented by the following equations:

1. **Kalman Gain:**

$$K[n] = \frac{R[n-1] \cdot X[n]}{\lambda + X[n]^T \cdot R[n-1] \cdot X[n]}$$

Algorithm 1 Recursive Least Squares (RLS) Algorithm for Denoising Speech Signals

Initialize filter coefficients $\mathbf{W}[0] = \mathbf{0}$
Initialize inverse correlation matrix $\mathbf{R}[0] = \mathbf{I}$ (Identity matrix)
Set forgetting factor λ
for each sample n from M to N **do**
    Extract input vector $\mathbf{X}[n] = [x[n-1], x[n-2], \ldots, x[n-M]]^T$
    Compute estimated output $\hat{d}[n] = \mathbf{W}[n-1]^T \cdot \mathbf{X}[n]$
    Compute estimation error $e[n] = d[n] - \hat{d}[n]$
    Compute Kalman gain $\mathbf{K}[n] = \frac{\mathbf{R}[n-1] \cdot \mathbf{X}[n]}{\lambda + \mathbf{X}[n]^T \cdot \mathbf{R}[n-1] \cdot \mathbf{X}[n]}$
    Update filter coefficients $\mathbf{W}[n] = \mathbf{W}[n-1] + \mathbf{K}[n] \cdot e[n]$
    Update inverse correlation matrix $\mathbf{R}[n] = \frac{1}{\lambda}\left(\mathbf{R}[n-1] - \mathbf{K}[n] \cdot \mathbf{X}[n]^T \cdot \mathbf{R}[n-1]\right)$
    Return denoised signal $\hat{d}[n]$

2. **Update of filter coefficients:**

$$W[n] = W[n-1] + K[n] \cdot e[n]$$

3. **Estimation Error:**

$$e[n] = d[n] - W[n]^T \cdot X[n]$$

4. **Update of inverse correlation matrix:**

$$R[n] = \frac{1}{\lambda}\left(R[n-1] - K[n] \cdot X[n]^T \cdot R[n-1]\right)$$

Steps to Perform the Experiment

Follow the steps below to implement the RLS algorithm for denoising the noisy speech signal:

1. **Load the Clean and Noisy Signals:**
    - Load the clean speech signal and the noisy speech signals from the AURORA Database, link mentioned in the reference section. Noisy signals will be pre-corrupted with different SNR levels (0dB, 5dB, 10dB, and 15dB).
2. **Initialize RLS Parameters:**
    - Define the filter order M and the forgetting factor λ.
    - Initialize the filter coefficients and the identity matrix R.
3. **Apply RLS Algorithm for Each SNR Level:**
    - Perform the RLS algorithm on the noisy speech signal at 0dB, 5dB, 10dB, and 15dB SNR levels.
    - For each SNR level, process the noisy signal using the RLS algorithm to generate the denoised signal.
4. **Plot the Results for Each SNR Level:**
    - Plot the clean speech signal, noisy speech signal, and the denoised speech signal for each SNR level (0dB, 5dB, 10dB, and 15dB).
    - Calculate the error for the following pairs at each SNR level:
        (a) Clean speech signal and noisy speech signal.
        (b) Clean speech signal and denoised speech signal.
    - Compare the performance of the RLS algorithm across different SNR levels.

Conclusion

After completing this experiment, students should be able to implement the RLS algorithm for denoising noisy speech signals across different SNR levels, understand its performance through SDR calculations, and compare the results visually and quantitatively.

References

[1] IEEE Transactions on Audio, Speech, and Language Processing, various issues.

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[3] S. Haykin, *Adaptive Filter Theory*, 5th ed., Pearson, 2014.

[4] P. Vary and R. Martin, *Digital Speech Transmission: Enhancement, Coding and Error Concealment*, 1st ed., Wiley, 2006.