ELEC5305 Project Proposal (Draft)

1. Project Title

Speech Denoising and Enhancement using Classical Signal Processing Methods

2. Student Information

Full Name: Tengfei Wang

Student ID (SID): 540542743

GitHub Username: Tengfei Wang5305

GitHub Project Link:

https://tengfeiwang5305.github.io/elec5305-project-TengfeiWang-540542743/

3. Project Overview

This project addresses the problem of speech degradation in noisy acoustic environments. Speech signals are frequently contaminated by environmental noise, leading to reduced intelligibility for human listeners and performance loss in automatic speech recognition (ASR). The goal is to implement and evaluate classical noise reduction methods that enhance noisy speech while preserving naturalness.

The proposed solution is to design and compare several time–frequency domain approaches, including spectral subtraction, Wiener filtering, and FIR low-pass filtering. The project emphasizes both quantitative evaluation (SNR improvement) and qualitative demonstrations (waveform, spectrum, and spectrogram visualizations).

4. Background and Motivation

Speech enhancement has been studied for decades, with classical approaches such as Boll's spectral subtraction (1979) and Ephraim and Malah's MMSE estimator (1984) forming the basis of modern research. While deep learning–based methods now dominate, traditional signal processing remains widely used in low-resource applications and provides valuable pedagogical insight.

I selected this topic because it balances feasibility and relevance: it can be implemented with MATLAB without requiring high computational resources, yet addresses a core challenge in speech technology. Furthermore, it allows both measurable outcomes (objective SNR gain) and perceptual demonstrations (listening tests, spectrogram comparisons), aligning with the project's emphasis on communication and demonstration.

5. Proposed Methodology

Tools and platforms: MATLAB (no special toolboxes required), GitHub for version control and documentation.

Signal processing sequence:

Input clean speech (from Archive.org sine/speech samples) and add synthetic noise at controlled SNR levels (5, 15, 30 dB).

Apply enhancement methods:

Spectral subtraction: estimate noise spectrum from silent segments, subtract from noisy magnitude spectrum.

Wiener filtering: estimate clean speech spectrum using SNR-dependent gain function.

Low-pass FIR filtering: baseline approach for noise suppression.

Reconstruct enhanced speech via inverse STFT.

Evaluate outputs using objective metrics (SNR, log-spectral distance) and visualizations (time-domain, FFT, spectrogram).

Data sources: Archive.org free speech/sine samples (public domain), supplemented by simple recordings if necessary.

6. Expected Outcomes

A MATLAB-based prototype that performs noise addition, speech enhancement, and signal reconstruction.

Clear visual demonstrations of noise suppression in time- and frequency-domain plots.Quantitative results showing SNR improvement for different algorithms.

A GitHub repository documenting code, plots, and explanations.

A video presentation that includes waveform and spectrogram comparisons and plays back clean vs. noisy vs. enhanced audio.

7. Timeline (Weeks 6–13)

Weeks 6–7: Literature review; dataset preparation; initial MATLAB STFT implementation.

Weeks 8–9: Implement spectral subtraction and Wiener filtering; debug and validate with toy signals.

Weeks 10–11: Evaluate algorithms under different noise levels; generate quantitative results and spectrograms.

Weeks 12–13: Optimize presentation of results; prepare written report and video demonstration; finalize GitHub documentation.

8. References

1.Boll, S. F. (1979). Suppression of acoustic noise in speech using spectral subtraction. IEEE Transactions on Acoustics, Speech, and Signal Processing, 27(2), 113–120.

2.Ephraim, Y., & Malah, D. (1984). Speech enhancement using a minimum mean-square error short-time spectral amplitude estimator. IEEE Transactions on Acoustics, Speech, and Signal Processing, 32(6), 1109–1121.