Department of Engineering and Aviation Sciences
ENCE 460 – Digital Signal Processing
Project 2

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Please select one of the projects for your project 2.

- **Signal denoising**. Signals get corrupted by noise all kinds of ways - by electrical interference, by mechanical damage of recording media, or simply because there were unwanted sounds present during the original recording. I will provide a number of speech examples that have been artificially corrupted by additive noise or by reverberation. Some approaches to noise reduction involve estimating the steady-state noise spectrum (by looking during 'gaps' in the speech), then designing a *Wiener filter* (see a signals and systems text) to optimize the signal-to-noise ratio. More advanced techniques can use the noise floor estimate to apply dynamic time-frequency gain that attempts to boost signal while cutting noise. The 'noise gate' found in a recording studio is one example; a more sophisticated frequency-dependent approach is the widely-used technique of spectral subtraction, which is described in the book *Discrete-time processing of speech signals* by Deller, Proakis and Hansen (Macmillan, 1993).
- **Speech endpointing** - find the beginning and end of each speech phrase or utterance in a recording (which may include background noise). The speech examples with added noise that provided by me can be used here, as well as the speech-over-music examples. The basic idea is to design a filter that will do the best job of identifying speech energy as compared to the nonspeech noise, then somehow setting a threshold to mark the beginning and end of the speech when the filtered energy exceeds that threshold. The following recent paper, chosen somewhat arbitrarily, describes a more sophisticated approach as well as including references to more classic papers: "Robust Entropy-based Endpoint Detection for Speech Recognition in Noisy Environments," Shen, Hung and Lee, Proc. Int. Conf. on Spoken Lang. Processing, Sydney, 1998.

Here are the suggestions of the project report. Your report must have the following structure.

1. Introduction: A general description of the area of your project and why you're doing it.
2. Problem Specification: A clear and succinct technical description of the problem you're addressing. Formulating a general problem (e.g., transcribing music) into a well-defined technical goal (e.g., reporting a list of estimated fundamental periods at each time frame) is often the most important part of a project.
3. Data: What are the real-world and/or synthetic signals you are going to use to develop and evaluate your work?
4. Evaluation Criteria: How are you going to measure how well your project performs? The best criteria are objective, quantitative, and discriminatory. You want to be able to demonstrate and measure improvements in your system.
5. Approach: A description of how you went about trying to solve the problem. Sometimes you can make a nice project by contrasting two or more different approaches.
6. Results and Analysis: What happened when you evaluated your system using the data and criteria introduced above? What were the principal shorfalls? (This may require you to choose or synthesize data that will reveal these shortcomings.) Your analysis of what happened is one of the most important opportunities to display your command of signal processing concepts.
7. Conclusions: What did you learn from doing the project? What did you demonstrate about how to solve your problem?
8. References: Complete list of sources you used in completing your project, with explanations of what you got from each.