New Mexico State University
Klipsch School of Electrical Engineering

EE589 Digital Speech Processing
Spring 2011 – Project #5
Due: 8.55am Thu. Apr. 7

Name: ______________________

Grade: ______________________

Project

The goal of this project is to develop codes for speech enhancement methods, namely spectral subtraction and iterative Wiener filtering. Codes for these methods will be developed by students in MATLAB based on the *DSP Software Toolkit* which can be found at

`http://www.ece.nmsu.edu/~pdeleon/Teaching/EE589/DSP_Toolkit.pdf`

(Due to a recent revision, please download a new copy of *DSP Software Toolkit*)

Report

Please submit a printed report with your results including commentary and plots. Please `mailto: pdeleon@nmsu.edu` a zip file containing your report and all MATLAB code to recreate your results.

Notes

Unless otherwise noted, in your simulations, use speech signals from the TIMIT-sample corpus

`http://www.ece.nmsu.edu/~pdeleon/Teaching/EE589/TIMIT-sample.zip`

and noise signals from the NOISEX-92 corpus

`http://www.ece.nmsu.edu/~pdeleon/Teaching/EE589/NOISEX-92.zip`

Students are encouraged to discuss detailed, technical aspects with each other and Prof. De Leon. Students are encouraged to utilize existing MATLAB functions in this project. However, students must write all other required codes on an *individual* basis.

Some problems ask for comments. As is always the case, your analysis, comparison to expected theory or model, and resulting comments are far more important than the code or plot. Therefore, please pay special attention to your comments.

1 Software Tools

Please write the following functions from the *DSP Software Toolkit*: **add_noise.m**, **overspecsub.m**, and **wiener.m**. Be sure to use the appropriate function header described on p. 5 and available at

`http://www.ece.nmsu.edu/~pdeleon/Teaching/EE589/funchead.m`

Note that code for spectral subtraction and iterative Wiener filtering is available in the Voicebox toolbox and other sources on the Internet. However, students are expected to develop their own codes based on the *DSP Software Toolkit*.

2 Problems

Spectral Subtraction

For the following problems, assume a 30 ms Hamming window with 50% overlap.

1. Let $x(n)$ be the speech signal, dr1-fvmh0.wav from TIMIT-sample/test. Let $d(n)$ be the noise, white16.wav from NOISEX-92. Let $y(n) = x(n) + d(n)$ be the noisy speech signal where the signal and noise are mixed at 15 dB SNR using add_noise.m. Window-by-window, attach the noisy speech phase spectrum to the clean speech magnitude spectrum, i.e. $\hat{X}(\omega) = |X(\omega)|e^{j\phi_y}$ where $e^{j\phi_y} = Y(\omega)/|Y(\omega)|$. Invert the windows and overlap-add to obtain $\hat{x}(n)$. Listen to $x(n)$ and $\hat{x}(n)$ and comment on the impact of attaching a noisy phase spectrum to a clean magnitude spectrum. You may wish to repeat this experiment for a variety of SNRs to determine at what SNR the noisy phase creates an audible distortion.

2. Prepend 100 ms of zero-valued samples to dr1-fvmh0.wav. Mix dr1-fvmh0.wav with white16.wav at a 15 dB SNR using add_noise.m. Plot the spectrograms of the clean and noisy speech signal.

3. Enhance the signal from Prob. 2 using overspecsub.m. Use olaistft.m to invert the enhanced spectrum back to the time-domain. Listen to the signal and comment.

4. On a single figure, plot the PESQ scores of noisy and enhanced (spectral subtraction) speech signals as function of SNR in dB for $0 \leq \text{SNR} \leq 40$ in steps of 5 dB. In order to obtain a more accurate evaluation, you should use the first ten TIMIT-sample speech signals for each SNR and average the PESQ scores.

Wiener Filter

For the following problems, assume a 30 ms Hamming window with 50% overlap.

5. Repeat Prob. 3 using the iterative Wiener filter.

6. Repeat Prob. 4 using the iterative Wiener filter. If you prefer you can combine the noisy speech signal and spectral subtraction curves from Prob. 4 with the Wiener filter curve and submit a single figure similar to that in Lecture 15.

Bonus (+10 points)

Repeat Probs. 4 and 6 using babble16.wav and f16.wav from NOISEX-92. For each noise signal, combine all three curves (noisy speech, spectral subtraction, Wiener filter) and submit a figure for babble16.wav and a separate figure for f16.wav.