Assignment 2

Digital Signal Analysis and Applications (DSAA) - IEC 239

Deadline: 8 March

February 29, 2016

1. Download the file corresponding to your Roll Number mod 8 from here and design your own filter to remove noise from the file, as much as possible. Write a brief description about the implementation of the filter.

2. Start Matlab. Load in the"train" signal with the command load ('train'). Recall that the audio signal is loaded into a variable "y" and the sampling rate into "Fs." The sampling rate is 8192 Hertz, and the signal contains 12,880 samples. If we consider this signal as sampled on an interval [0, T], then T = 12880/8192 1.5723 seconds. Now do the following:
a) Eliminate the insignificant frequencies by "thresholding", that is, zeroing out any Fourier coefficients below a given threshold. This becomes the compressed version of the signal. To recover an approximation to the signal, we use the IFFT to take the thresholded transform back to the time domain. Assuming Y be the FFT of the given signal, use threshold $T = a * max(abs(Y))$ and reconstruct the signal. Do the computations above for a values 0.001, 0.01, 0.1, and 0.5.
(b) The fraction of Fourier coefficient which survive denote the compression ratio (number of non zero coefficients after thresholding, divided by total number of unique coefficients). For each value of a above, compute the compression ratio, the distortion, and of course, play the audio signal and rate its quality. The distortion measures the variation between the original and the reconstructed signal (after thresholding + ifft), it can be computed by following formula: $100 * norm(y - ythresh)^2/norm(y)^2$, where $ythresh$ is the reconstructed signal.

3. Write a matlab script to compute the spectrogram of a given audio file. Use window size and the length of the stride as the input to your function. Test your code on the in built laughter and train audio files in matlab [load laughter; sound(y) or load train; sound(y)]. Compare your results with the inbuilt spectrogram function in matlab.

4. This link contains 5 files recorded in a concert. Stitch these files into one file based on cross correlation between them. There is at least 3 seconds of overlap between consecutive files.

NOTE : Numbering on files is not the exact order (file number three is the first file). There is also small amount of noise added due to recording (so you may consider denoising before any computation).