**UNIVERSITY OF MIAMI**
**DEPARTMENT OF ELECTRICAL AND COMPUTER ENGINEERING**
**EEN 436 - INTRODUCTION TO DIGITAL SIGNAL PROCESSING**

**Project No.1**

# EXPERIMENTING WITH DOPPLER SOUND EFFECTS

## Framework

In this project we will deal with some of the early topics of DSP, such the reconstruction of sampled signals, under sampling and time-domain filtering using the overlap-add (OLA) method. You are required to present your report in a **web page** by the due date. Include all figures and all audio samples used or generated. Summarize your results and conclusions. It is required that you use Matlab for your simulations.

1. An analog signal $x_a(t)$ of low pass characteristics has been appropriately sampled resulting in sequence $x[nT] \rightarrow x[n]$. The Nyquist theorem states that if sampling has satisfied certain criteria, as it is the case here, the continuous time signal can be perfectly reconstructed by ideally low pass filtering its sampled version, or, equivalently, by using an ideal interpolator in the time domain. Mat file guitar_8k.mat is located on the course site includes 4000 samples of x[nT] (fs = 8000 Hz). Write a program that generates a new version of the samples signal by increasing its temporal resolution by a factor of 100 using an ideal interpolator of the form sin(x)/x.

2. In touch-tone telephones dual-tone multi-frequency (DTMF) signaling generates two tones for each key pressed, as shown in the table below. In speed dialing every digit is 100 ms and it is followed by 20 ms of silence. Note in this table the standard frequencies have increased by 303 Hz.

| (Hz) | 1512 | 1639 | 1780 |
|---|---|---|---|
| 1000 | **1** | **2** | **3** |
| 1073 | **4** | **5** | **6** |
| 1155 | **7** | **8** | **9** |
| 1244 | ***** | **0** | **#** |

   a. Generate a sampled version of a 10-digit number using the Nyquist rate. Plot a 100ms digit.
   b. Generate the same 10-digit sequence using the smallest possible sampling rate. Plot a 100ms digit.
   c. Recover and upsample each version to 8 kHz and confirm that they sound identical.

3. Violin music was recorded in an anechoic room and it is provided on the course site. The impulse response of a reverberant room is also provided in file impulse_response.mat. Implement the filtering process of the music with the room response by using the overlap and add (OLA) method. Plot the impulse response and provide the original and enhanced music pieces.

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Date Due: Tuesday, October 21, 2008