EEC-201 Final Project

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1 Speech Data Files

Down load the ZIP file of the speech database from canvas. After unzipping the file, you will 11 speech files, named: S1.WAV, S2.WAV, ...; each is labeled after the ID of the speaker. These files were recorded in WAV format.
Our goal is to train a voice model (e.g., a VQ codebook in the MFCC vector space) for each speaker using the corresponding sound file. After this training step, the system would have knowledge of the voice characteristic of each (known) speaker. Next, in the testing phase, you should add noises to distort the existing training signals to generate a test set. The amount of noises would vary to test the robustness of your system.

1.1 Test 1

Play each sound file in the TRAIN folder. Can you distinguish the voices of the 11 speakers in the database? Next play each sound in the TEST folder in a random order without looking at the groundtruth and try to identify the speaker manually. Record what is your (human performance) recognition rate. Use this result as a later benchmark.

Human Performance		
TEST Audio	Jonathan	Igor
s1	s1	s
s2	s2	s
s3	s3	s
s4	s4	s
s5	s5	s
s6	s6	s
s7	s7	s
s8	s8	s

2 Speech Processing

2.1 Test 2

In Matlab one can play the sound file using "sound". Record the sampling rate and compute how many milliseconds of speech are contained in a block of 256 samples? **Now plot the signal to view it in the time domain. It should be obvious that the raw data are long and may need to be normalized because of different strengths.**

Use STFT to generate periodogram. Locate the region in the plot that contains most of the energy, in time (msec) and frequency (in Hz) of the input speech signal. Try different frame size: for example N = 128, 256 and 512. In each case, set the frame increment M to be about N/3.

2.2 Test 3

Plot the mel-spaced filter bank responses. Compare them with theoretical responses. Compute and plot the spectrum of a speech file before and after the mel-frequency wrapping step. Describe and explain the impact of the melfb.m or melfbown.m program.

2.3 Test 4

Complete the "Cepstrum" step and put all pieces together into a single Matlab function, e.g., mfcc.m

3 Vector Quantization

Now apply VQ-based pattern recognition technique to build speaker reference models from those vectors in the training set before identifying any sequences of acoustic vectors from unmarked speakers in the test set.

3.1 Test 5

To check whether the program is working, inspect the acoustic space (MFCC vectors) in any two dimensions in a 2D plane to observe the results from different speakers. Are they in clusters?

Now write a function that trains a VQ codebook using the LGB algorithm.

3.2 Test 6

Plot the resulting VQ codewords using the same two dimensions over the plot of in TEST 5. You should get a figure like Figure 4.

4 Full Test and Demonstration

Using the programs to train and test speaker recognition on the data sets.

4.1 Test 7

Record the results. What is recognition rate our system can perform? Compare this with human performance. Experiment and find the reason if high error rate persists. Record more voices of yourself and your teammates/friend. Each new speaker can provide one speech file for training and one for testing. **Record the results.**

4.2 Test 8

Use notch filters on the voice signals to generate another test set. Test your system on the accuracy after voice signals have passed different notch filters that may have suppressed distinct features of the original voice signal. Report the robustness of your system.

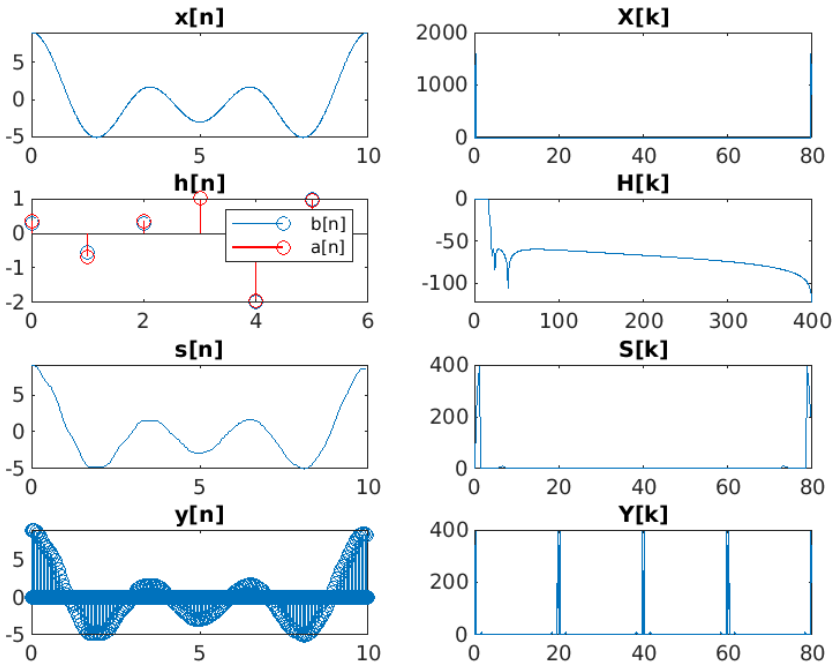
4.3 Test 9

Test the system with other speech files you may find online. E.g. `https://lionbridge.ai/datasets/best-speech-recognition-datasets-for-machine-learning/`

5 LaTex Reference (temp)

```
%
%EEC-201, Winter Quarter 2021, Final Project
%
%Title: Speaker Recognition
%
%Description: This is main function of the final project for
EEC-201.
%             This program will store features in the recorded
audio of
%             different speakers in order to recognize which
speaker is
%             talking on further recordings.
%
%Authors: Igor Sheremet and Jonathan Tivald
%
%Date: 2/7/2021

clear all;
close all;
clc;
```



(a) DC gain of g[n] is 0.2

(b) The zeros of G(z) = -1/(zeros of H(z))

(c) $abs(G(w)) = abs(H(w - pi))$