

# Assignment 1: Time Domain Features Extraction

CS 4347: Sound and Music Computing

due Wednesday 8 February 2017, 11:59 pm

0. This assignment will make use of the “Music & Speech” dataset of Marsyas:

- [http://opihi.cs.uvic.ca/sound/music\\_speech.tar.gz](http://opihi.cs.uvic.ca/sound/music_speech.tar.gz)
- This dataset has two copies of each song; delete the `music/` and `speech/` directories and use the files in `music-wav/` and `speech-wav/` directories. There are 64 music and 64 speech files; each file is 30 seconds of audio stored as 16-bit signed integers at 22050 Hz.

- Ground truth data for this dataset:

[http://www.comp.nus.edu.sg/~duanzhy/music\\_speech.mf](http://www.comp.nus.edu.sg/~duanzhy/music_speech.mf)

Format of the file is `filename \t (tab) label \n (newline)`, one song per line:

```
filename1\tlabel1\n
filename2\tlabel2\n
...
filename128\tlabel128\n
```

The `label` will be `music` or `speech`.

1. Write a python program that will:

- Read the ground-truth `music_speech.mf` file
- Load each wav file and convert the data to floats by dividing the samples by 32768.0. Hint: use `scipy.io.wavfile.read()`
- Calculate 5 features for each file according to the given formulae. Use only one vector per file (don't use multiple buffers for each file).

Given  $X = \{x_0, x_1, x_2, \dots, x_{N-1}\}$ ,

(a) Root-mean-squared (RMS)

$$X_{\text{RMS}} = \sqrt{\frac{1}{N} \sum_{i=0}^{N-1} x_i^2}$$

(b) Peak-to-average-ratio (PAR)

$$X_{\text{PAR}} = \frac{\max |x_i|}{X_{\text{RMS}}}$$

(c) Zero crossings (ZCR)

$$X_{\text{ZCR}} = \frac{1}{N-1} \sum_{i=1}^{N-1} \begin{cases} 1 & \text{if } (x_i \cdot x_{i-1}) < 0 \\ 0 & \text{else} \end{cases}$$

(d) Median absolute deviation (MAD)

$$X_{\text{MAD}} = \text{median}_i \left( \left| x_i - \text{median}_j(x_j) \right| \right)$$

Hint: numpy has a built-in `numpy.median()` function!

(e) Mean Absolute Deviation (MEAN-AD):

$$X_{\text{MEAN\_AD}} = \frac{1}{N} \sum_{i=0}^{N-1} \left( \left| x_i - \frac{1}{N} \sum_{j=0}^{N-1} x_j \right| \right)$$

- Output the data to a comma separated value (CSV) text file in the format:

```
filename1,RMS1,PAR1,ZCR1,MAD1,MEAN_AD1\n
filename2,RMS2,PAR2,ZCR2,MAD2,MEAN_AD2\n
...
filename128,RMS128,PAR128,AC128,MAD128,MEAN_AD128\n
```

Concretely, the beginning and ending of the file should be:

```
music_wav/bagpipe.wav,0.063492,8.149929,0.191660,0.031769,0.044588
music_wav/ballad.wav,0.029699,7.320233,0.039395,0.012695,0.018975
...
speech_wav/voice.wav,0.070688,4.163124,0.082435,0.031982,0.049338
```

To pass our automated grading system, the format of your file must match this exactly. The order of filenames must match the order in the `music_speech.mf` file.

2. Upload your CSV file, and source code to **Assignment 1a** at:

<http://cs4347.smcnus.org>

This will automatically grade the values you calculated. If any mistake is found, please check your program and resubmit – you are welcome to submit as many versions as you wish before the submission deadline.

Submit a zip file containing your program’s source code files (as a `.py` file), the CSV file, and an optional `README.txt` file to the same website.

3. Make a copy of the previous program. Modify it such that it will:

- Read the ground-truth `music_speech.mf` file
- Load each wav file (divide by 32768.0) and split the data into buffers of length 1024 with 50% overlap (or a hopsize of 512). Only include complete buffers; if the final buffer has 1020 samples, omit that buffer.

**Hint:** the starting and ending indices for the first few buffers are:

Buffer number	start index	end index (not included in array)
0	0	1024
1	512	1536
2	1024	2048
...		

I recommend that you use the numpy “array slice” feature:

```
for i in range(num_buffers):
    start = ...
    end = ...
    buffer_data = whole_file_data[start:end]
```

This should give you 1290 buffers of length 1024. If you are comfortable with linear algebra and numpy, the program will run significantly faster if you store this as a single 1290x1024 matrix and avoid any loops during the feature calculations.

- Calculate 5 features for each buffer: RMS, PAR, ZCR, MAD, MEAN\_AD. This means that  $N = 1024$  in the equations, instead of the previous  $N = 661500$ . You should end up with 1290 feature vectors of length 5, or a single 1290x5 matrix.

- After calculating the features for each buffer, calculate the mean and uncorrected sample standard deviation for each feature over all buffers for each file. You should end up with a single feature vector of length 10, or a 1x10 matrix for each file.
- For a later assignment we will be using a tool which reads ARFF files. Output the data to a new ARFF file with the header:

```
@RELATION music_speech
@ATTRIBUTE RMS_MEAN NUMERIC
@ATTRIBUTE PAR_MEAN NUMERIC
@ATTRIBUTE ZCR_MEAN NUMERIC
@ATTRIBUTE MAD_MEAN NUMERIC
@ATTRIBUTE MEAN_AD_MEAN NUMERIC
@ATTRIBUTE RMS_STD NUMERIC
@ATTRIBUTE PAR_STD NUMERIC
@ATTRIBUTE ZCR_STD NUMERIC
@ATTRIBUTE MAD_STD NUMERIC
@ATTRIBUTE MEAN_AD_STD NUMERIC
@ATTRIBUTE class {music,speech}
```

The data lines should be:

```
@DATA
RMS_MEAN1,PAR_MEAN1,ZCR_MEAN1,MAD_MEAN1,MEAN_AD_MEAN1,RMS_STD1,PAR_STD1,ZCR_STD1,MAD_STD1,MEAN_AD_STD1,music
...
```

Concretely, the @DATA section should be:

```
@DATA
0.057447,3.234547,0.191595,0.037308,0.044607,0.027113,0.397827,0.036597,0.018317,0.021898,music
0.026583,2.590328,0.039416,0.016488,0.018977,0.013284,0.384978,0.015575,0.011143,0.012273,music
...
0.062831,2.822102,0.082504,0.042271,0.049270,0.032323,0.799688,0.070962,0.027540,0.029103,speech
```

4. Upload your ARFF files, and source code to **Assignment 1b** at:

<http://cs4347.smcnus.org>

This will automatically grade the values you calculated. If any mistake is found, please check your program and resubmit – you are welcome to submit as many versions as you wish before the submission deadline.

Submit a zip file containing your program's source code files (as a .py file), the ARFF file, and an optional README.txt file to the same website.

- You may use anything in the python standard library, numpy (including pylab / matplotlib), and scipy libraries. No other libraries are permitted.

If you are familiar with python and understood the lecture, this should take about 2 hours. Grading scheme:

- **2/6 marks:** correct CSV file (automatically graded by computer).
- **2/6 marks:** correct ARFF file (automatically graded by computer).
- **2/6 marks:** readable source code (good variable names, clean functions, comments when needed).