



Speech Emotion Recognition

Why Speech to Emotion directly?

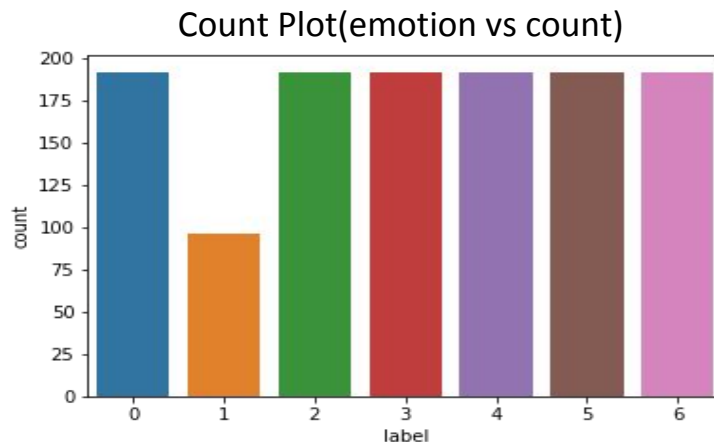
- The main problem with traditional systems is that the **errors** occurred while **transcribing audio** is **propagated** and that affects the emotion recognition task.
- We **lose** important **acoustic features** like pitch, loudness etc.
- **Confusing emotional contexts** in some cases.
For example, the phrase, 'What a man!' can indicate surprise or even disgust.

Motivation

- A human lacking essential brain function or having a malfunction on the neural subsystem responsible for handling emotions cannot efficiently perform decision-making.
- Computers are no longer just logical computing machines. With the advent of human machine interaction, apart of '**what is said**' and '**who said it**', '**how it is said**' plays a key role for effective human-machine communication and for the machine to react properly.
- Other applications include the **psychiatric diagnosis**, **intelligent toys** and **lie detection** in call centers as well as **evaluation of mental state** of the driver for the start of his/her day.

Datasets Used

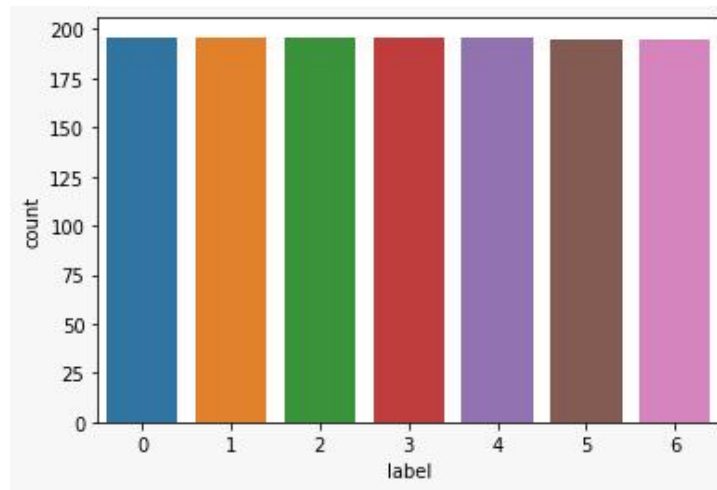
- Name: The Ryerson Audio-Visual Database of Emotional Speech and Song (RAVDESS)
- For our project, only the **audio tracks** have been used.
- Description about the audio dataset:
 - Gender balanced consisting of **24** professional actors.
 - 7 emotions in total: **calm, happy, sad, angry, fearful, surprise, and disgust expressions.**
 - Each expression is produced at two levels of emotional intensity, with an additional neutral expression.
 - **1250** samples in total.



Datasets Used

- Name: The Toronto Emotional Speech Set (TESS)
- Description about the audio dataset:
 - Gender balanced consisting of **2** professional actresses (aged 24 and 64 years).
 - 7 emotions in total: **anger, disgust, fear, happiness, surprise, sadness, neutral**.
 - Audiometric testing indicated that both actresses have thresholds within the normal range.
 - **1370** samples in total.

Count Plot(emotion vs count)



MFCC Feature Extraction

- The signal has a sampling frequency of **16KHz**.
- Used Scipy function to load the audio file and extract the features.

Steps:

- Apply **pre-emphasis** on the signal to amplify the high frequencies.
 - Help in **balancing the magnitude**
 - **Improve signal-to-noise(SNR) ratio**.
- After pre-emphasis, split the signal into short time frames and apply a **Fourier transform over the short time frame**.
 - Frame size: **25ms**, stride: **10ms**
- After slicing the signal into frames, we apply a window function(Window function) to each frame.

MFCC Feature Extraction (Contd.)

Steps:

- We apply **N-point FFT**(Fast Fourier Transform) on each frame to calculate the **frequency spectrum** and then compute the **power spectrum**.
- To computer Filter banks, **Triangular filters** are applied, typically **40** filters, to extract the frequency bands.
- Apply **Discrete Cosine Transform**(DCT)
 - To decorrelate the filter bank coefficients
 - Yield a compressed representation of the filter banks .
- The resulting coefficients are the 40 dimensional MFCC features.

Data Augmentations

- **Noise Injection:** It simply adds some random value or random background music into data.
- **Changing pitch:** Pitch is the fundamental period of the speech signal. We used **librosa** library which changes the pitch randomly.
- **Adding Reverb:** It is the reflection of sound waves created by the superposition of echoes. Used **pysndfx** library for this.
- **Changing Speed:** Used **cv2 resize** function which slightly increases or decreases the audio speed.

Model used

- We split the dataset in the ratio **80:20** for train and test sets respectively.
- We trained two **Random Forest Classifiers** with **100** estimators each.
 - One was trained on the **raw dataset** examples. The other was trained on **augmented** samples.
 - Each of the augmentations was applied on the original signal itself, one at a time.
- After augmenting the data, the size of the dataset increased from **1250** samples to **6250** samples.

Results on RAVDESS

- The confusion matrices, the classification reports and accuracies for the **test sets**(using Random Forest with and without augmentation)

Raw Data

```
Raw test data
Classification Report
precision    recall  f1-score   support
```

```

0          0.47        0.66        0.55         32
1          0.67        0.43        0.52         14
2          0.84        0.75        0.79         48
3          0.57        0.51        0.54         41
4          0.48        0.67        0.56         33
5          0.76        0.66        0.70         38
6          0.59        0.50        0.54         44
```

```

accuracy          0.61         250
macro avg         0.62         250
weighted avg      0.63         250
```

```
Confusion Report
[[21  1  0  3  1  4  2]
 [ 2  6  2  0  4  0  0]
 [ 2  0 36  3  4  1  2]
 [ 7  1  1 21  6  1  4]
 [ 3  1  1  4 22  0  2]
 [ 3  0  1  4  0 25  5]
 [ 7  0  2  2  9  2 22]]
```

Test accuracy: 0.612

Augmented Data

```
Augmented test data
Classification Report
precision    recall  f1-score   support
```

```

0          0.64        0.73        0.69        187
1          0.91        0.53        0.67        109
2          0.77        0.81        0.79        193
3          0.71        0.71        0.71        198
4          0.67        0.75        0.71        186
5          0.77        0.82        0.79        180
6          0.82        0.71        0.76        195
```

```

accuracy          0.74       1248
macro avg         0.76       1248
weighted avg      0.75       1248
```

```
Confusion Report
[[137  0  6  8 20  7  9]
 [ 14 58 11  9 14  2  1]
 [  7  1 157 10  4 12  2]
 [ 13  1  8 141 13 18  4]
 [ 18  3  5  6 140  2 12]
 [  9  0  8 10  3 147  3]
 [ 15  1 10 14 14  3 138]]
```

Test accuracy: 0.7355769230769231

Results on TESS

- The confusion matrices, the classification reports and accuracies for the **test sets**(using Random Forest with and without augmentation)

Raw Data

Raw test data					
Classification	Report				
	precision	recall	f1-score	support	
0	0.97	0.95	0.96	37	
1	0.97	1.00	0.99	39	
2	0.96	0.90	0.93	52	
3	0.87	0.97	0.92	35	
4	1.00	0.97	0.99	38	
5	1.00	1.00	1.00	28	
6	1.00	1.00	1.00	45	
accuracy			0.97	274	
macro avg	0.97	0.97	0.97	274	
weighted avg	0.97	0.97	0.97	274	
Confusion Report					
[[35 1 1 0 0 0 0]					
[0 39 0 0 0 0 0]					
[0 0 47 5 0 0 0]					
[0 0 1 34 0 0 0]					
[1 0 0 0 37 0 0]					
[0 0 0 0 0 28 0]					
[0 0 0 0 0 0 45]]					
Test accuracy: 0.9671532846715328					

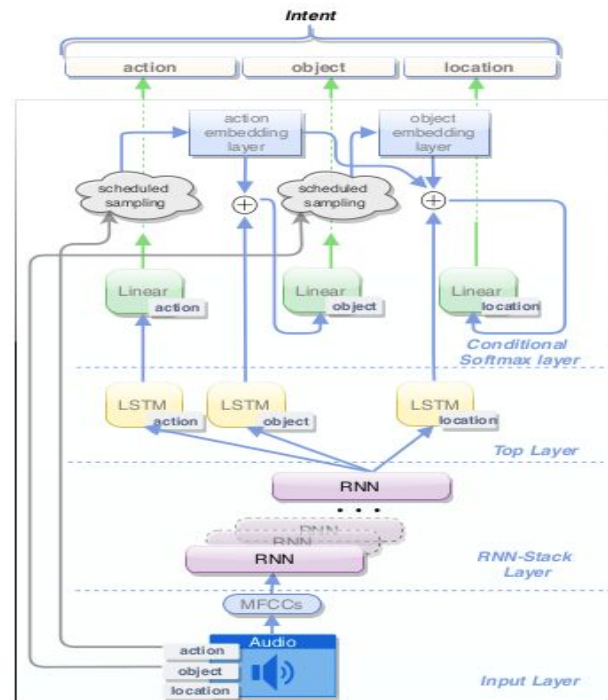
Augmented Data

Augmented test data					
Classification	Report				
	precision	recall	f1-score	support	
0	0.98	0.99	0.98	194	
1	1.00	0.98	0.99	189	
2	0.97	0.97	0.97	208	
3	0.97	0.97	0.97	190	
4	0.99	1.00	0.99	191	
5	0.99	0.99	0.99	196	
6	1.00	0.99	0.99	202	
accuracy			0.99	1370	
macro avg	0.99	0.99	0.99	1370	
weighted avg	0.99	0.99	0.99	1370	
Confusion Report					
[[192 0 2 0 0 0 0]					
[1 186 0 0 2 0 0]					
[2 0 202 4 0 0 0]					
[0 0 4 185 0 1 0]					
[0 0 0 0 191 0 0]					
[0 0 0 0 0 195 1]					
[1 0 0 1 0 0 200]]					
Test accuracy: 0.9861313868613139					

Emotion Recognition using Bidirectional LSTM^[3]

(**Ref.** E. Palogiannidi, I. Gkinis, G. Mastrapas, P. Mizera and T. Stafylakis, "End-to-End Architectures for ASR-Free Spoken Language Understanding," ICASSP 2020 - 2020 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), Barcelona, Spain, 2020, pp. 7974-7978, doi: 10.1109/ICASSP40776.2020.9054314.)

- Uses 40 dimensional **MFCC features** extracted every **10ms**.
- The model performs very well on **Spoken Language Understanding** task, achieving an accuracy of **98.85%**.
- The model comprises of three parts:
 - RNN stack
 - Representation layer
 - Classifier layer



References

1. Breiman, L. Random Forests. *Machine Learning* 45, 5–32 (2001).
<https://doi.org/10.1023/A:1010933404324>
2. M. A. Hossan, S. Memon and M. A. Gregory, "A novel approach for MFCC feature extraction," 2010 4th International Conference on Signal Processing and Communication Systems, Gold Coast, QLD, 2010, pp. 1-5, doi: 10.1109/ICSPCS.2010.5709752.
3. E. Palogiannidi, I. Gkinis, G. Mastrapas, P. Mizera and T. Stafylakis, "End-to-End Architectures for ASR-Free Spoken Language Understanding," ICASSP 2020 - 2020 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), Barcelona, Spain, 2020, pp. 7974-7978, doi: 10.1109/ICASSP40776.2020.9054314.



Thank You