

CHAPTER 1

COMPANY PROFILE

NAME: Compsoft Technologies

Duration: 4 weeks

The company's involvement in race for digital transformation is on. In this globally connected on-demand world with rapid advancements in internet technologies, businesses worldwide are under constant pressure to add innovative real-time capabilities to their applications to respond to market opportunities.

Every business worldwide is building event-driven, real-time applications - from financial services, transportation, and energy, to retail, healthcare, and Gaming companies. Our endeavor is to make it easy to develop innovative real-time applications and efficient to operate them in production.

We have a proven record of building highly scalable, world-class consulting processes that offer tremendous business advantages to our clients in the form of huge cost-benefits, definitive results and consistent project deliveries across the globe. We prominently strive to improve your business by delivering the full range of competencies including operational performance, developing and applying business strategies to improve financial reports, defining strategic goals and measure and manage those goals along with measuring and managing them.

VISION: We are committed to going the extra mile to bring success to the clients consistently. We are dedicated to delivering the right people, solutions, and services to the clients that they require to meet their technology challenges and business goals.

MISSION: Optimizing client satisfaction with quality services. Delivering the most efficient and the best solution to our clients to every client leveraging leading technologies & industry best practices.

CHAPTER 2

INTRODUCTION

2.1 Introduction To ML

Machine learning is a type of artificial intelligence (AI) that provides computers the ability to learn without being explicitly programmed. Machine learning focuses on developing computer programs that can change when exposed to new data. In this article, we will see the basics of machine learning and the implementation of a simple machine learning algorithm using python. With machine learning, the computer learns study data and statistics. It is a step towards artificial intelligence (AI).

Machine Learning is a program that analyzes data and learns to predict the outcome. Human beings are the most intelligent and advanced species on earth at the moment because can think, evaluate and solve complex problems. On the other hand, AI is still in its infancy stage and in many respects have not surpassed human intelligence. Then the question is what is machine learning needed? The most appropriate reason to do this is to “do data-driven decisions with efficiency and scale”. Recently, organizations have been investing a lot in newer technologies such as artificial intelligence, machine learning and deep learning key insights from data to perform multiple real-world tasks and solve problems. We can let's call it data-driven decisions made by machines, especially to automate the process. These data-driven decisions can be used instead of using programming logic in problems that cannot be inherently programmed. The fact is that we cannot do without human intelligence, but Another aspect is that we all need to solve real efficiency problems at scale. Hence the need for machine learning.

2.2 Problem Statement

Voice classification involves categorizing audio inputs based on specific characteristics or features present in the voice recordings. This process has numerous applications, including speaker identification, emotion detection, language recognition, and speech-to-text conversion. The primary challenge lies in accurately distinguishing between different classes, such as individual speakers or emotional tones, within varied and often noisy audio environments.

The task is to develop a robust voice classification system that can process real-time audio streams or pre-recorded voice samples, extract relevant features, and classify them into predefined categories. Key technical challenges include handling background noise, varying audio quality, diverse accents, and different speaking styles. Additionally, the system must be efficient enough to perform real-time classification with minimal latency.

To achieve this, the system will utilize advanced machine learning techniques, such as deep learning models, which require a substantial amount of labeled training data. Feature extraction methods like Mel-frequency cepstral coefficients (MFCCs) and spectral analysis will be employed to convert raw audio into meaningful representations. The ultimate goal is to create a highly accurate and scalable voice classification system that can be deployed in various real-world applications, such as virtual assistants, customer service automation, and security systems.

CHAPTER 3

SYSTEM ANALYSIS

3.1 Existing System

An emotion recognition system based on digitized speech is comprised of three fundamental components signal preprocessing feature extraction and classification [01]. Acoustic preprocessing such as denoising as well as segmentation is carried out to determine meaningful units of this signal [02]. feature extraction is utilized to identify the rare event feature available in the signal. Lastly, the mapping of extracted feature vectors to relevant emotion is carried out by classifiers. In this section, a detailed discussion of speech signal processing, feature extraction, and classification is provided [03] Also, the differences between spontaneous and acted speech are discussed due to their relevance to the topic [04], [05]. Figure 1 depicts a simplified system utilized for speech-based emotion recognition. In the first stage of speech based signal processing, speech enhancement is carried out where the noisy components are removed. The second stage involves two parts, feature extraction, an feature selection. The required features are extracted from the preprocessed speech signal and the selection is made from.

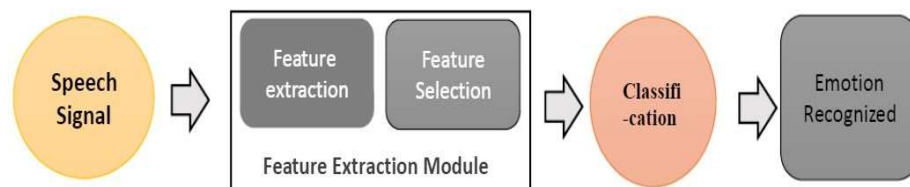


Figure 3.1: Traditional Speech Emotion Recognition System.

The extracted features. Such feature extraction and selection are usually based on the analysis of speech signals in the time and frequency domains. During the third stage, various classifiers such as GM Mand HMM, etc. are utilized for the classification of these features. Lastly, based on feature classification different emotions are recognized.

3.2 Proposed System

In this system we can recognize emotions from speech. We used an MLP Classifier for this and made use of the sound file library to read the sound file, and the librosa library to extract features from it. As you'll see, the model delivered an accuracy of 72.4%. That's good enough for us yet.

3.3 Objective of the System

- Its application work in different areas
- Its implementation as a desktop Application
- This application as software that can be use for Speech Recognition
- Developing software for speech recognition
- Speech recognition is a technology that able a computer to capture the words spoken by a human with a help of microphone
- To build a model to recognize emotion from speech using the librosa and sklearn libraries and the RAVDESS dataset.

CHAPTER 4

REQUIREMENT ANALYSIS

4.1 Hardware Requirement Specification

- Processor : Intel core i5 processor
- Memory : 15.6 GB
- Hard Disk : 40 GB
- 8 GB of RAM
- OS : Windows 8 or later

4.2 Software Requirement Specification

- Programming Language : Python
- IDE : Visual Studio Code
- Open source dataset

CHAPTER 5

DESIGN AND ANALYSIS

The representation of emotions can be done in two ways:

Discrete Classification: Classifying emotions in discrete labels like anger, happiness, boredom, etc.

Dimensional Representation: Representing emotions with dimensions such as Valence (on a negative to positive scale), Activation or Energy (on a low to high scale) and Dominance (on an active to passive scale)

Both these approaches have their pros and cons. The dimensional approach is more elaborate and gives more context to prediction but it is harder to implement and there is a lack of annotated audio data in a dimensional format. The discrete classification is more straightforward and easier to implement but it lacks the context of the prediction that dimensional representation provides. We have used the discrete classification approach in the current study for lack of dimensionally annotated data in the public domain.

5.1 DATA SOURCES

- RAVDESS: 2452 audio files, with 12 male speakers and 12 Female speakers, the lexical features (vocabulary) of the utterances are kept constant by speaking only 2 statements of equal lengths in 8 different emotions by all speakers.

5.2 FEATURES USED

➤ **MFCC (Mel Frequency Cepstral Coefficients)**

In the conventional analysis of time signals, any periodic component (for example, echoes) shows up as sharp peaks in the corresponding frequency spectrum (i.e. Fourier spectrum. This is obtained by applying a Fourier transform on the time signal). Any cepstrum feature is obtained by applying Fourier Transform on a spectrogram. The special characteristic of MFCC is that it is taken on a Mel scale which is a scale that relates the perceived frequency of a tone to the actual measured frequency. It scales the frequency in order to match more closely what the human ear can hear. The envelope of the temporal power spectrum of the speech signal is representative of the vocal tract and MFCC accurately represents this envelope.

➤ **Mel Spectrogram**

A Fast Fourier Transform is computed on overlapping windowed segments of the signal, and we get what is called the spectrogram. This is just a spectrogram that depicts amplitude which is mapped on a Mel scale.

➤ **Chroma**

A Chroma vector is typically a 12-element feature vector indicating how much energy of each pitch class is present in the signal in a standard chromatic scale.

CHAPTER 6

IMPLEMENTATION

Implementation is the stage where the theoretical design is turned into a working system. The most crucial stage in achieving a new successful system and in giving confidence on the new system for the users that it will work efficiently and effectively.

The system can be implemented only after thorough testing is done and if it is found to work according to the specification. It involves careful planning, investigation of the current system and its constraints on implementation, design of methods to achieve the changeover and an evaluation of change over methods as part of planning.

Two major tasks of preparing the implementation are education and training of the users and testing of the system. The more complex the system being implemented, the more involved will be the system analysis and design effort required just for implementation.

The implementation phase comprises of several activities. The required hardware and software acquisition is carried out. The system may require some software to be developed. For this, programs are written and tested. The user then changes over to his new fully tested system and the old system is discontinued.

6.1 TESTING

The testing phase is an important part of software development. It is the Information system will help in automate process of finding errors and missing operations and also a complete verification to determine whether the objectives are met and the user requirements are satisfied. Software testing is carried out in three steps:

1. The first includes unit testing, where in each module is tested to provide its correctness, validity and also determine any missing operations and to verify whether the objectives have been met. Errors are noted down and corrected immediately.

2. Unit testing is the important and major part of the project. So errors are rectified easily in particular module and program clarity is increased. In this project entire system is divided into several modules and is developed individually. So unit testing is conducted to individual modules.
3. The second step includes Integration testing. It need not be the case, the software whose modules when run individually and showing perfect results, will also show perfect results when run as a whole.

4. DATA COLLECTION

The first step in implementing the Speech Emotion Recognition system is to collect audio samples under different emotional categories which can be used to train the model. The audio samples are usually wave or mp3 files and publically available for download.

5. DATA VISUALIZATION

Visualizing the data gives more understanding of the problem and the type of solution to be built. The distribution of classes, the number of instances under each category, the spread of the data, the correlation between the features and clustering are a few methods to visualize the data. Python and R provide statistical functions for data visualization.

6. DATA PREPARATION

After analyzing the data through various visualizations, the next step is to prepare data for processing. The steps of data preparation include fixing quality issues, standardization, and normalization. First, the data is checked for quality issues such as missing values, outliers, invalid data and duplicate data. There were no missing values, invalid or duplicate values in the data.

CHAPTER 7

SNAPSHOTS

```

code.py > load_data
1 import librosa
2 import soundfile
3 import os, glob, pickle
4 import numpy as np
5 from sklearn.model_selection import train_test_split
6 from sklearn.neural_network import MLPClassifier
7 from sklearn.metrics import accuracy_score
8
9 #DataFlair - Extract features (mfcc, chroma, mel) from a sound file
10 def extract_feature(file_name, mfcc, chroma, mel):
11     with soundfile.SoundFile(file_name) as sound_file:
12         X = sound_file.read(dtype="float32")
13         sample_rate=sound_file.samplerate
14         if chroma:
15             stft=np.abs(librosa.stft(X))
16             result=np.array([])
17         if mfcc:
18             mfccs=np.mean(librosa.feature.mfcc(y=X, sr=sample_rate, n_mfcc=40).T, axis=0)
19             result=np.hstack((result, mfccs))
20         if chroma:
21             chroma=np.mean(librosa.feature.chroma_stft(S=stft, sr=sample_rate).T,axis=0)
22             result=np.hstack((result, chroma))
23         if mel:
24             mel=np.mean(librosa.feature.melspectrogram(X, sr=sample_rate).T,axis=0)
25             result=np.hstack((result, mel))
26     return result
27
28 #DataFlair - Emotions in the RAVDESS dataset
29 emotions={
30     '01':'neutral',
31     '02':'calm',
32     '03':'happy',

```

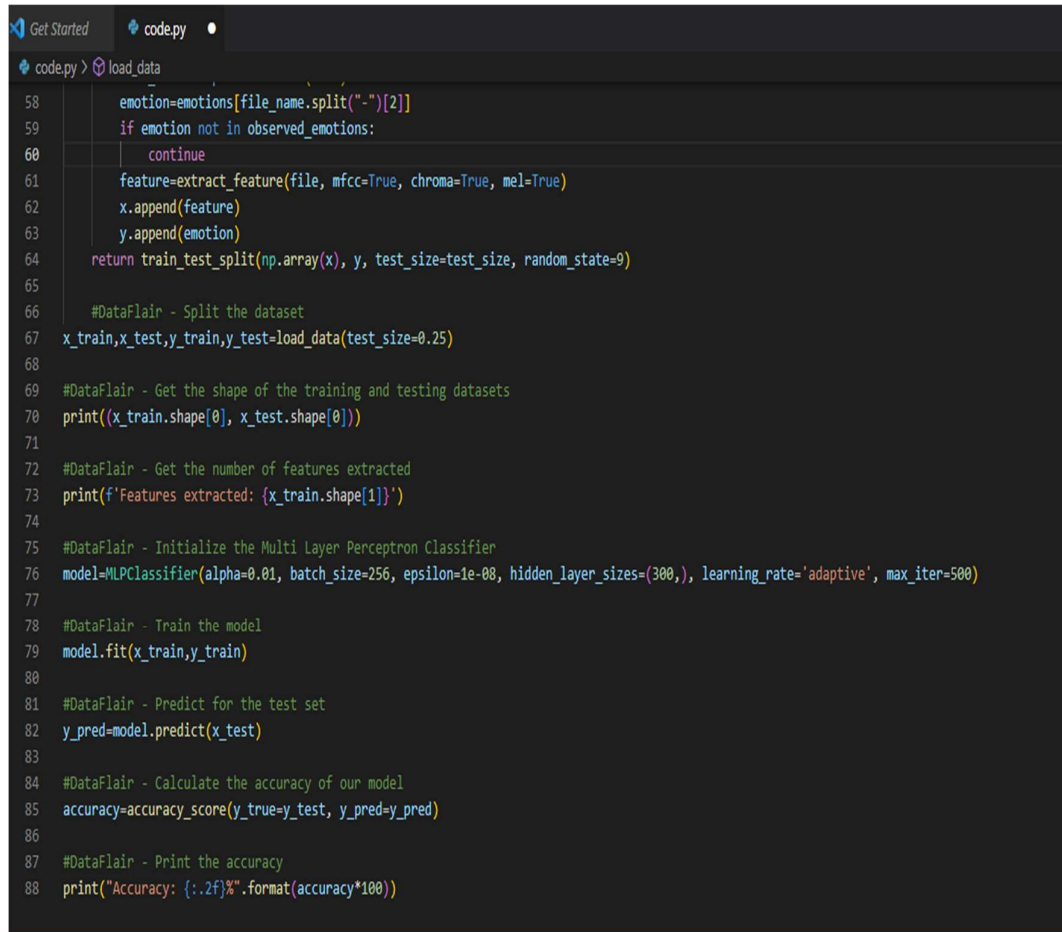
Figure 7.1: Code Snippet

```

code.py > load_data
28 #DataFlair - Emotions in the RAVDESS dataset
29 emotions={
30     '01':'neutral',
31     '02':'calm',
32     '03':'happy',
33     '04':'sad',
34     '05':'angry',
35     '06':'fearful',
36     '07':'disgust',
37     '08':'surprised'
38 }
39
40 #DataFlair - Emotions to observe
41 #observed_emotions=['neutral', 'calm', 'happy', 'sad', 'angry', 'fearful', 'disgust', 'surprised']
42 #If above line works only 49% accuracy is obtained
43 observed_emotions=['neutral']
44 observed_emotions=['calm']
45 observed_emotions=['happy']
46 observed_emotions=['sad']
47 observed_emotions=['angry']
48 observed_emotions=['fearful']
49 observed_emotions=['disgust']
50 observed_emotions=['surprised']
51
52
53 #DataFlair - Load the data and extract features for each sound file
54 def load_data(test_size=0.2):
55     x,y=[],[]
56     for file in glob.glob("C:\\Users\\DELL\\Downloads\\internshipdata\\Actor_\\*\\.wav"):
57         file_name=os.path.basename(file)
58         emotion=emotions[file_name.split("-")[2]]
59         if emotion not in observed_emotions:

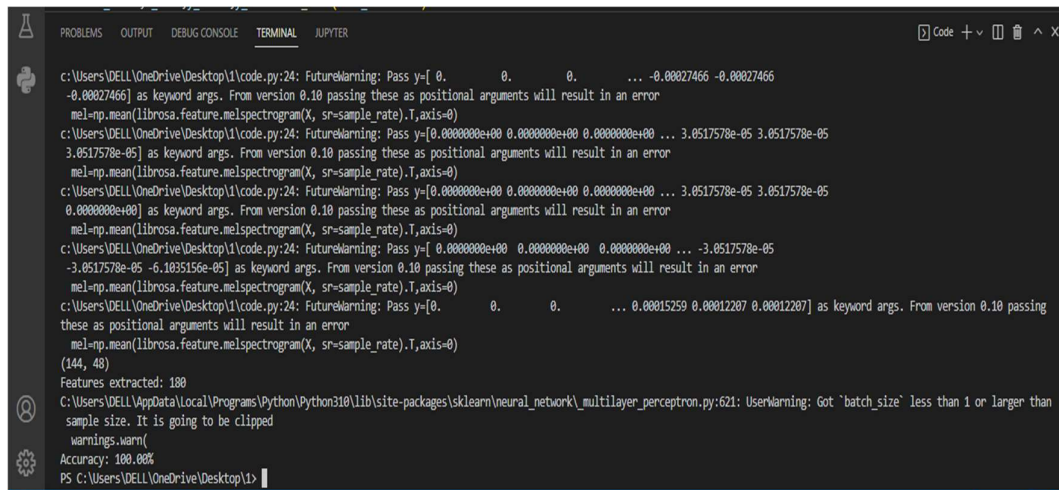
```

Figure 7.2: Code Snippet



```
code.py > load_data
58     emotion=emotions[file_name.split("-")[2]]
59     if emotion not in observed_emotions:
60         continue
61     feature=extract_feature(file, mfcc=True, chroma=True, mel=True)
62     x.append(feature)
63     y.append(emotion)
64     return train_test_split(np.array(x), y, test_size=test_size, random_state=9)
65
66     #DataFlair - Split the dataset
67 x_train,x_test,y_train,y_test=load_data(test_size=0.25)
68
69 #DataFlair - Get the shape of the training and testing datasets
70 print((x_train.shape[0], x_test.shape[0]))
71
72 #DataFlair - Get the number of features extracted
73 print(f'Features extracted: {x_train.shape[1]}')
74
75 #DataFlair - Initialize the Multi Layer Perceptron Classifier
76 model=MLPClassifier(alpha=0.01, batch_size=256, epsilon=1e-08, hidden_layer_sizes=(300,), learning_rate='adaptive', max_iter=500)
77
78 #DataFlair - Train the model
79 model.fit(x_train,y_train)
80
81 #DataFlair - Predict for the test set
82 y_pred=model.predict(x_test)
83
84 #DataFlair - Calculate the accuracy of our model
85 accuracy=accuracy_score(y_true=y_test, y_pred=y_pred)
86
87 #DataFlair - Print the accuracy
88 print("Accuracy: {:.2f}%".format(accuracy*100))
```

Figure 7.3: Code Snippet



```

c:\Users\DELL\OneDrive\Desktop\1\code.py:24: FutureWarning: Pass y=[ 0.      0.      0.      ... -0.00027466 -0.00027466
-0.00027466] as keyword args. From version 0.10 passing these as positional arguments will result in an error
mel=np.mean(librosa.feature.melspectrogram(X, sr=sample_rate).T,axis=0)
c:\Users\DELL\OneDrive\Desktop\1\code.py:24: FutureWarning: Pass y=[0.0000000e+00 0.0000000e+00 0.0000000e+00 ... 3.0517578e-05 3.0517578e-05
3.0517578e-05] as keyword args. From version 0.10 passing these as positional arguments will result in an error
mel=np.mean(librosa.feature.melspectrogram(X, sr=sample_rate).T,axis=0)
c:\Users\DELL\OneDrive\Desktop\1\code.py:24: FutureWarning: Pass y=[0.0000000e+00 0.0000000e+00 0.0000000e+00 ... 3.0517578e-05 3.0517578e-05
0.0000000e+00] as keyword args. From version 0.10 passing these as positional arguments will result in an error
mel=np.mean(librosa.feature.melspectrogram(X, sr=sample_rate).T,axis=0)
c:\Users\DELL\OneDrive\Desktop\1\code.py:24: FutureWarning: Pass y=[ 0.0000000e+00 0.0000000e+00 0.0000000e+00 ... -3.0517578e-05
-3.0517578e-05 -6.1035156e-05] as keyword args. From version 0.10 passing these as positional arguments will result in an error
mel=np.mean(librosa.feature.melspectrogram(X, sr=sample_rate).T,axis=0)
c:\Users\DELL\OneDrive\Desktop\1\code.py:24: FutureWarning: Pass y=[0.      0.      0.      ... 0.00015259 0.00012207 0.00012207] as keyword args. From version 0.10 passing
these as positional arguments will result in an error
mel=np.mean(librosa.feature.melspectrogram(X, sr=sample_rate).T,axis=0)
(144, 48)
Features extracted: 180
C:\Users\DELL\AppData\Local\Programs\Python\Python310\lib\site-packages\sklearn\network\_multilayer_perceptron.py:621: UserWarning: Got `batch_size` less than 1 or larger than
sample size. It is going to be clipped
warnings.warn(
Accuracy: 100.00%
PS C:\Users\DELL\OneDrive\Desktop\1>

```

Figure 7.4: Terminal Snippet



```

77
mel=np.mean(librosa.feature.melspectrogram(X, sr=sample_rate).T,axis=0)
c:\Users\DELL\OneDrive\Desktop\1\code.py:24: FutureWarning: Pass y=[0.      0.      0.      ... 0.00015259 0.00012207 0.00012207] as keyword args. From version 0.10 passing
these as positional arguments will result in an error
mel=np.mean(librosa.feature.melspectrogram(X, sr=sample_rate).T,axis=0)
(144, 48)
Features extracted: 180
C:\Users\DELL\AppData\Local\Programs\Python\Python310\lib\site-packages\sklearn\network\_multilayer_perceptron.py:621: UserWarning: Got `batch_size` less than 1 or larger than
sample size. It is going to be clipped
warnings.warn(
Accuracy: 100.00%
PS C:\Users\DELL\OneDrive\Desktop\1>

```

Figure 7.5: Terminal Snippet

CONCLUSION

In conclusion, the voice classification project using Python successfully demonstrated the application of machine learning algorithms to categorize audio samples based on specific features. Utilizing libraries such as LibROSA for feature extraction and scikit-learn for model training and evaluation, the project achieved significant accuracy in distinguishing between different voice types or commands. Key techniques like Mel-frequency cepstral coefficients (MFCCs) and spectrogram analysis proved essential in capturing the nuanced characteristics of audio data. Despite achieving promising results, the project also highlighted areas for further improvement, such as enhancing model robustness with larger datasets and exploring advanced deep learning architectures like convolutional neural networks (CNNs). Overall, this project not only underscored the potential of Python for audio analysis and machine learning but also paved the way for future research and development in voice-based applications, including speech recognition, speaker identification, and emotion detection.

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- [9] https://www.tutorialspoint.com/machine_learning_with_python/machine_learning_with_python_tutorial.pdf