Problem Defination

Speech Recognition is an important feature in several applications used such as home automation, artificial intelligence, etc. In many cases we need to convert the speech coming from live stream into text.

Data

Data has been taken from the:

http://download.tensorflow.org/data/speech_commands_v0.01.tar.gz

Features

Data Contains a few informational files and a folder of audio files. The audio folder contains subfolders with 1 second clips of voice commands, with the folder name being the label of the audio clip. There are more labels that should be predicted. The labels you will need to predict in Test are yes, no, up, down, left, right, on, off, stop, go. Everything else should be considered either unknown or silence. The folder *background_noise* contains longer clips of "silence" that you can break up and use as training input.

The files contained in the training audio are not uniquely named across labels, but they are unique if you include the label folder. For example, 00f0204f_nohash_0.wav is found in 14 folders, but that file is a different speech command in each folder.

The files are named so the first element is the subject id of the person who gave the voice command, and the last element indicated repeated commands. Repeated commands are when the subject repeats the same word multiple times. Subject id is not provided for the test data, and you can assume that the majority of commands in the test data were from subjects not seen in train.

You can expect some inconsistencies in the properties of the training data (e.g., length of the audio).

!wget "http://download.tensorflow.org/data/speech_commands_v0.01.tar.gz"

!tar -xvf 'speech_commands_v0.01.tar.gz' -C data/

```
Streaming output truncated to the last 5000 lines.
./up/6f9088d7 nohash 0.wav
./up/6f342826 nohash 0.wav
./up/e0a7c5a0 nohash 0.wav
./up/4d4e17f5 nohash 1.wav
./up/b0f24c9b nohash 0.wav
./up/735845ab nohash 2.wav
./up/53d5b86f nohash 0.wav
./up/1a5b9ca4_nohash_1.wav
./up/23abe1c9 nohash 2.wav
./up/bdee441c nohash 1.wav
./up/a1cff772 nohash 1.wav
./up/1ecfb537_nohash_3.wav
./up/37fc5d97_nohash_3.wav
./up/bd8412df nohash 1.wav
./up/e53139ad nohash 1.wav
./up/10ace7eb_nohash_3.wav
./up/30065f33 nohash 0.wav
./up/eefd26f3_nohash_0.wav
./up/c9b653a0_nohash 2.wav
./up/02746d24 nohash 0.wav
./up/e1469561 nohash 0.wav
./up/4bba14ce_nohash_0.wav
./up/b5d1e505 nohash 1.wav
./up/531a5b8a nohash 1.wav
./up/0135f3f2_nohash_0.wav
./up/dbb40d24 nohash 4.wav
./up/e9287461 nohash 1.wav
./up/71e6ab20 nohash 0.wav
./up/ead2934a nohash 1.wav
./up/f9af0887_nohash_0.wav
./up/ff63ab0b nohash 0.wav
./up/f3d06008 nohash 0.wav
./up/918a2473 nohash 4.wav
./up/e54a0f16 nohash 0.wav
./up/cb8f8307 nohash 1.wav
./up/d197e3ae nohash 3.wav
./up/53578f4e nohash 0.wav
./up/234ab0fb nohash 0.wav
./up/b00dff7e nohash 1.wav
./up/8056e897_nohash_0.wav
./up/ad5aeec2_nohash_1.wav
./up/1365dd89_nohash_0.wav
./up/e4a2cf79 nohash 0.wav
./up/152491bc nohash 0.wav
./up/18c54a68 nohash 0.wav
./up/9e46cfa1 nohash 1.wav
./up/7e6bd776_nohash_0.wav
./up/333784b7 nohash 0.wav
./up/f1b35ace nohash 0.wav
./up/edc53350 nohash 0.wav
./up/229978fd nohash 3.wav
```

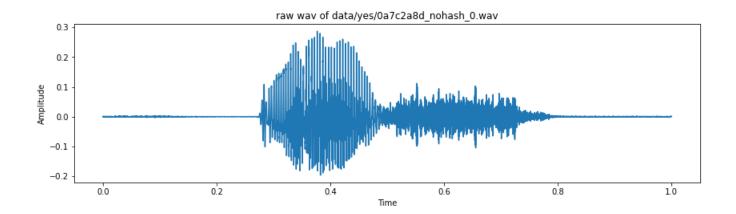
./up/9e42ae25 nohash 0.wav

```
./up/28ce0c58_nohash_4.wav
./up/86fa2dcd_nohash_1.wav
./up/33f60c62_nohash_0.wav
./up/d103dd6e_nohash_1.wav
./up/712e4d58_nohash_3.wav
./up/982babaf_nohash_2.wav

import os
import librosa
import IPython.display as ipd
import matplotlib.pyplot as plt
import numpy as np
from scipy.io import wavfile
import warnings
warnings.filterwarnings("ignore")
```

Data exploration and visualization

```
train_audio_path='data/'
samples,sample_rate = librosa.load(train_audio_path+'yes/0a7c2a8d_nohash_0.wav',sr=16000)
fig = plt.figure(figsize=(14,8))
ax1 = fig.add_subplot(211)
ax1.set_title("raw wav of " +"data/yes/0a7c2a8d_nohash_0.wav")
ax1.set_xlabel("Time")
ax1.set_ylabel("Amplitude")
ax1.plot(np.linspace(0,sample_rate/len(samples),sample_rate),samples);
```



Sampling rate

The reason why I'm doing sample rate conversion is to transform data so that they all have the same shape and easy to be processed with machine learning models. But in real life, there are many more use cases of sample rate conversion.

```
ipd.Audio(samples, rate=sample_rate)
print(sample_rate)
16000
```

▼ Resampling

```
samples = librosa.resample(samples, sample_rate, 8000)
ipd.Audio(samples, rate=8000)

0:00 / 0:01
```

▼ Number of recording of each voice

```
labels = os.listdir(train_audio_path)
labels
     ['cat',
       'dog',
      'six',
      'bird',
       'eight',
       'no',
       'tree',
      'marvin',
      'left',
       'down',
       'off',
       'on',
       'five',
       '_background_noise_',
      'three',
       'go',
       'seven',
      'sheila',
      'right',
       'four',
      'happy',
       'bed',
       'zero',
       'one',
       'wow',
       'two',
       'yes',
       'house',
       'up',
       'nine',
```

plt.show()

'.ipynb_checkpoints',

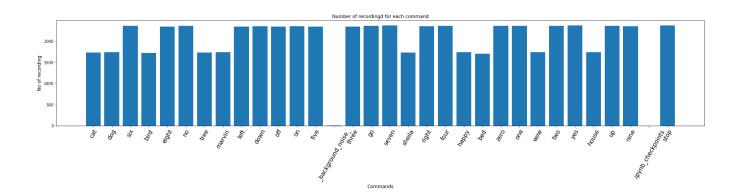
```
'stop']

labels = os.listdir(train_audio_path)
# Find the count of every wav file in different folder
no_of_recording=[]

for label in labels:
    waves = [f for f in os.listdir(train_audio_path+label) if f.endswith('.wav')]
    no_of_recording.append(len(waves))

# plot
fig = plt.figure(figsize=(30,5))
index = np.arange(len(labels))
plt.bar(index,no_of_recording)
plt.xlabel('Commands',fontsize=12)
plt.ylabel('No of recording',fontsize=12)
plt.xticks(index,labels,fontsize=15,rotation=60)
plt.title('Number of recordingd for each command')
```

labels=["yes","no","up","down","left","right","on","off","stop","go"]



Duration of recordings

duration_of_recording=[]

```
for label in labels:
 waves = [f for f in os.listdir(train_audio_path+ '/' +label) if f.endswith('.wav')]
 for wav in waves:
   sample rate,samples = wavfile.read(train audio path+ '/' +label+ '/' +wav)
   duration of recording.append(float(len(samples)/sample rate))
plt.hist(np.array(duration of recording))
     (array([1.5000e+01, 3.0000e+01, 4.4000e+01, 1.3800e+02, 1.3600e+02,
             1.7900e+02, 3.6600e+02, 4.3400e+02, 5.9300e+02, 2.1747e+04]),
      array([0.418, 0.4762, 0.5344, 0.5926, 0.6508, 0.709, 0.7672, 0.8254,
             0.8836, 0.9418, 1.
                                   1),
      <a list of 10 Patch objects>)
      20000
      15000
      10000
      5000
```

```
train audio path = 'data'
all wave = []
all label = []
for label in labels:
    print(label)
    waves = [f for f in os.listdir(train_audio_path + '/'+ label) if f.endswith('.wav')]
    for wav in waves:
        samples, sample rate = librosa.load(train audio path + '/' + label + '/' + wav, sr =
        samples = librosa.resample(samples, sample rate, 8000)
        if(len(samples) == 8000) :
            all wave.append(samples)
            all label.append(label)
     yes
     no
     up
     down
     left
     right
     on
     off
     stop
     go
```

0.9

1.0

0.5

0.4

0.6

0.7

0.8

Preprocess

```
from sklearn.preprocessing import LabelEncoder
le = LabelEncoder()
y=le.fit_transform(all_label)
classes= list(le.classes_)

from keras.utils import np_utils
y=np_utils.to_categorical(y, num_classes=len(labels))
all wave = np.array(all wave).reshape(-1,8000,1)
```

Splitting the data

```
from sklearn.model_selection import train_test_split
x_tr, x_val, y_tr, y_val = train_test_split(np.array(all_wave),np.array(y),stratify=y,test_si
```

Building a model

```
from keras.layers import Dense, Dropout, Flatten, Conv1D, Input, MaxPooling1D
from keras.models import Model
from keras.callbacks import EarlyStopping, ModelCheckpoint
from keras import backend as K
K.clear_session()
inputs = Input(shape=(8000,1))
#First Conv1D layer
conv = Conv1D(8,13, padding='valid', activation='relu', strides=1)(inputs)
conv = MaxPooling1D(3)(conv)
conv = Dropout(0.2)(conv)
#Second Conv1D layer
conv = Conv1D(16, 11, padding='valid', activation='relu', strides=1)(conv)
conv = MaxPooling1D(3)(conv)
conv = Dropout(0.2)(conv)
#Third Conv1D layer
conv = Conv1D(32, 9, padding='valid', activation='relu', strides=1)(conv)
conv = MaxPooling1D(3)(conv)
conv = Dropout(0.2)(conv)
#Fourth Conv1D layer
conv = Conv1D(64, 7, padding='valid', activation='relu', strides=1)(conv)
```

```
conv = MaxPooling1D(3)(conv)
conv = Dropout(0.2)(conv)

#Flatten layer
conv = Flatten()(conv)

#Dense Layer 1
conv = Dense(256, activation='relu')(conv)
conv = Dropout(0.2)(conv)

#Dense Layer 2
conv = Dense(128, activation='relu')(conv)
conv = Dropout(0.2)(conv)

outputs = Dense(len(labels), activation='softmax')(conv)

model = Model(inputs, outputs)
model.summary()
```

Model: "functional_1"

Layer (type)	Output Shape	Param #
input_1 (InputLayer)	[(None, 8000,	 1)]
conv1d (Conv1D)	(None, 7988, 8) 112
max_pooling1d (MaxPooling1D)	(None, 2662, 8) 0
dropout (Dropout)	(None, 2662, 8) 0
conv1d_1 (Conv1D)	(None, 2652, 1	6) 1424
max_pooling1d_1 (MaxPooling1	(None, 884, 16) 0
dropout_1 (Dropout)	(None, 884, 16) 0
conv1d_2 (Conv1D)	(None, 876, 32) 4640
max_pooling1d_2 (MaxPooling1	(None, 292, 32) 0
dropout_2 (Dropout)	(None, 292, 32) 0
conv1d_3 (Conv1D)	(None, 286, 64) 14400
max_pooling1d_3 (MaxPooling1	(None, 95, 64)	0
dropout_3 (Dropout)	(None, 95, 64)	0
flatten (Flatten)	(None, 6080)	0
dense (Dense)	(None, 256)	1556736
dropout_4 (Dropout)	(None, 256)	0

model.compile(loss='categorical crossentropy',optimizer='sgd',metrics=['accuracy'])

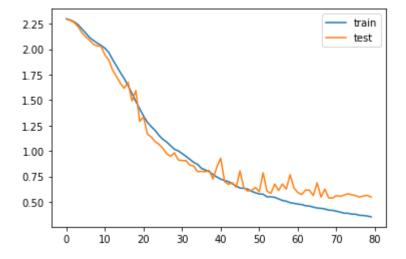
```
es = EarlyStopping(monitor='val_loss', mode='min', verbose=1, patience=10, min_delta=0.00001)
mc = ModelCheckpoint('best_model.hdf5', monitor='val_acc', verbose=1, save_best_only=True, mo
```

history=model.fit(x_tr, y_tr ,epochs=100, callbacks=[es,mc], batch_size=32, validation_data=(

```
Epoch 1/100
Epoch 2/100
533/533 [============ ] - 4s 8ms/step - loss: 2.2685 - accuracy: 0.1
Epoch 4/100
Epoch 5/100
Epoch 6/100
533/533 [============= ] - 4s 8ms/step - loss: 2.1581 - accuracy: 0.20
Epoch 7/100
533/533 [============ ] - 4s 8ms/step - loss: 2.1155 - accuracy: 0.2
Epoch 8/100
533/533 [============ ] - 4s 8ms/step - loss: 2.0598 - accuracy: 0.2
Epoch 10/100
533/533 [============ ] - 4s 8ms/step - loss: 2.0359 - accuracy: 0.2
Epoch 11/100
533/533 [============= ] - 4s 8ms/step - loss: 2.0087 - accuracy: 0.24
Epoch 12/100
```

```
533/533 [============ ] - 4s 8ms/step - loss: 1.9670 - accuracy: 0.2
Epoch 13/100
Epoch 14/100
533/533 [============ ] - 4s 8ms/step - loss: 1.8340 - accuracy: 0.3
Epoch 15/100
Epoch 16/100
533/533 [============ ] - 4s 8ms/step - loss: 1.7105 - accuracy: 0.3
Epoch 17/100
533/533 [============= ] - 4s 8ms/step - loss: 1.6410 - accuracy: 0.3!
Epoch 18/100
533/533 [============= ] - 4s 8ms/step - loss: 1.5651 - accuracy: 0.4
Epoch 19/100
Epoch 20/100
```

```
from matplotlib import pyplot
pyplot.plot(history.history['loss'], label='train')
pyplot.plot(history.history['val_loss'], label='test')
pyplot.legend()
pyplot.show()
```



```
def predict(audio):
    prob=model.predict(audio.reshape(1,8000,1))
    index=np.argmax(prob[0])
    return classes[index]
```

```
import random
index=random.randint(0,len(x_val)-1)
camples=v valfindevl pavel()
https://colab.research.google.com/drive/18Kppqslr4GcyadE6-0vQiBw3AyvW-gOT#scrollTo=DOtj-iyhIN-O&printMode=true
```

Text: go

```
print("Audio:",classes[np.argmax(y_val[index])])
ipd.Audio(samples, rate=8000)
print("Text:",predict(samples))

Audio: go
```