Spoken Digit Recognition

In this notebook, You will do Spoken Digit Recognition.

Input - speech signal, output - digit number

It contains

- 1. Reading the dataset. and Preprocess the data set. Detailed instrctions are given below. You have to write the code in the same cell which contains the instrction.
- 2. Training the LSTM with RAW data
- 3. Converting to spectrogram and Training the LSTM network
- 4. Creating the augmented data and doing step 2 and 3 again.

Instructions:

- 1. Don't change any Grader Functions. Don't manipulate any Grader functions. If you manipulate any, i
- 2. Please read the instructions on the code cells and markdown cells. We will explain what to write.
- 3. Please return outputs in the same format what we asked. Eg. Don't return List of we are asking for
- 4. Please read the external links that we are given so that you will learn the concept behind the cod
- 5. We are giving instructions at each section if necessary, please follow them.

Every Grader function has to return True.

- 1. Use direct audio files to train the LSTM network
- 2. Create spectogram (visual representation of audio files) and use that to train the LSTM network
- 3. Perform augumentation on audio files and the size of the files will be much larger than original audio files and train the LSTM network
- 4. Convert above augumented audio files to spectogram and train the LSTM network

```
import numpy as np
import pandas as pd
```

```
import librosa
import os
from sklearn.model_selection import train_test_split
##if you need any imports you can do that here.
```

We shared recordings.zip, please unzip those.

```
#read the all file names in the recordings folder given by us
#(if you get entire path, it is very useful in future)
!gdown --id 1ZQFXDEQ4RomprWQLAGJUwg0jne 0b6Jl
!pip install patool
import patoolib
patoolib.extract archive("recordings.zip")
#https://thispointer.com/python-how-to-get-list-of-files-in-directory-and-sub-directories/
def getListOfFiles(dirName):
   # create a list of file and sub directories
   # names in the given directory
   listOfFile = os.listdir(dirName)
   allFiles = list()
   # Iterate over all the entries
   for entry in listOfFile:
       # Create full path
       fullPath = os.path.join(dirName, entry)
       # If entry is a directory then get the list of files in this directory
        if os.path.isdir(fullPath):
            allFiles = allFiles + getListOfFiles(fullPath)
        else:
            allFiles.append(fullPath)
   return allFiles
#save those files names as list in "all files"
all_files = getListOfFiles('/content/recordings')
Grader function 1
def grader_files():
   temp = len(all files)==2000
   temp1 = all([x[-3:]=="wav" for x in all_files])
   temp = temp and temp1
   return temp
grader_files()
```

True

Create a dataframe(name=df_audio) with two columns(path, label).

You can get the label from the first letter of name.

```
Eg: 0_jackson_0 --> 0
0_jackson_43 --> 0
```

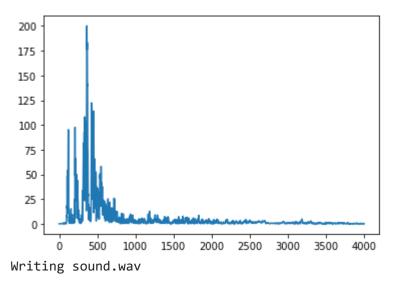
Exploring the sound dataset

#It is a good programming practise to explore the dataset that you are dealing with. This dat #https://colab.research.google.com/github/Tyler-Hilbert/AudioProcessingInPythonWorkshop/blob/#visualize the data and write code to play 2-3 sound samples in the notebook for better under #please go through the following reference video https://www.youtube.com/watch?v=37zCgCdV468

```
!git clone https://github.com/AllenDowney/ThinkDSP.git
     fatal: destination path 'ThinkDSP' already exists and is not an empty directory.
import sys
sys.path.insert(0, 'ThinkDSP/code/')
import thinkdsp
import matplotlib.pyplot as pyplot
import IPython
def visualize sound(file):
 # Read in audio file
 wave = thinkdsp.read wave(file)
 # Plot spectrum of audio file
 spectrum = wave.make spectrum()
 spectrum.plot()
 pyplot.show()
 # Play audio file
 wave.play()
visualize_sound('/content/recordings/0_yweweler_22.wav')
IPython.display.Audio('sound.wav')
```



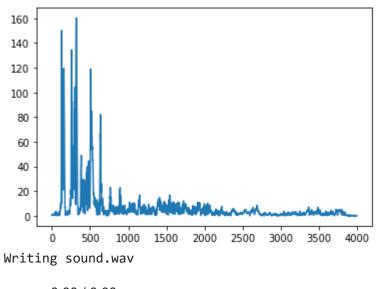
visualize_sound('/content/recordings/0_jackson_14.wav')
IPython.display.Audio('sound.wav')



0:00 / 0:00

visualize_sound('/content/recordings/0_nicolas_46.wav')
IPython.display.Audio('sound.wav')

```
visualize_sound('/content/recordings/0_theo_38.wav')
IPython.display.Audio('sound.wav')
```



0:00 / 0:00

Creating dataframe

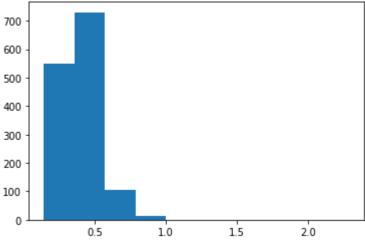
```
#Create a dataframe(name=df_audio) with two columns(path, label).
#You can get the label from the first letter of name.
#Eg: 0_jackson_0 --> 0
#0 jackson 43 --> 0
label = [i.split('/')[3].split('_')[0] for i in all_files]
df_audio = pd.DataFrame({'path':all_files,'label':label})
#info
df audio.info()
     <class 'pandas.core.frame.DataFrame'>
     RangeIndex: 2000 entries, 0 to 1999
     Data columns (total 2 columns):
          Column Non-Null Count Dtype
      0
          path
                  2000 non-null
                                  object
      1
          label
                  2000 non-null
                                  object
     dtypes: object(2)
     memory usage: 31.4+ KB
```

Grader function 2

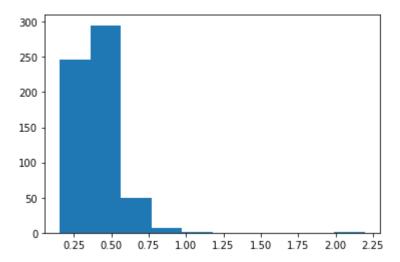
```
def grader df():
   flag shape = df audio.shape==(2000,2)
   flag columns = all(df audio.columns==['path', 'label'])
   list values = list(df audio.label.value counts())
   flag_label = len(list_values)==10
   flag label2 = all([i==200 for i in list values])
   final flag = flag shape and flag columns and flag label and flag label2
   return final flag
grader df()
     True
from sklearn.utils import shuffle
df_audio = shuffle(df_audio, random_state=33)#don't change the random state
 Train and Validation split
#split the data into train and validation and save in X train, X test, y train, y test
#use stratify sampling
#use random state of 45
#use test size of 30%
X_train, X_test, y_train, y_test = train_test_split(df_audio['path'],df_audio['label'],test_s
Grader function 3
def grader split():
   flag len = (len(X train)=1400) and (len(X test)==600) and (len(y train)==1400) and (len(x test)=1400)
   values_ytrain = list(y_train.value_counts())
   flag ytrain = (len(values ytrain)==10) and (all([i==140 for i in values ytrain]))
   values_ytest = list(y_test.value_counts())
   flag ytest = (len(values ytest)==10) and (all([i==60 for i in values ytest]))
   final flag = flag len and flag ytrain and flag ytest
   return final flag
grader split()
     True
 Preprocessing
```

All files are in the "WAV" format. We will read those raw data files using the librosa

```
sample rate = 22050
def load wav(x, get duration=True):
    '''This return the array values of audio with sampling rate of 22050 and Duration'''
    #loading the wav file with sampling rate of 22050
    samples, sample_rate = librosa.load(x, sr=22050)
    if get duration:
        duration = librosa.get duration(samples, sample rate)
        return [samples, duration]
    else:
        return samples
#use load wav function that was written above to get every wave.
#save it in X train processed and X test processed
# X_train_processed/X_test_processed should be dataframes with two columns(raw_data, duration
raw_data_tr = []
duration tr = []
raw data te = []
duration_te = []
for i in X train:
  x,y = load_wav(i)
  raw data tr.append(x)
  duration tr.append(y)
X_train_processed = pd.DataFrame({'raw_data':raw_data_tr,'duration':duration_tr})
for i in X test:
  x,y = load_wav(i)
  raw data te.append(x)
  duration te.append(y)
X_test_processed = pd.DataFrame({'raw_data':raw_data_te,'duration':duration_te})
#plot the histogram of the duration for trian
import matplotlib.pyplot as plt
plt.hist(X_train_processed.duration)
plt.show()
```



```
#plot the histogram of the duration for test
plt.hist(X_test_processed.duration)
plt.show()
```



```
#print 0 to 100 percentile values with step size of 10 for train data duration.
for i in range(0,100,10):
 print(f"{i}th percentile is {np.percentile(X train processed['duration'].values,i)}")
     Oth percentile is 0.1435374149659864
    10th percentile is 0.2606938775510204
    20th percentile is 0.30187755102040814
     30th percentile is 0.3324489795918368
    40th percentile is 0.35935600907029475
    50th percentile is 0.3905215419501134
    60th percentile is 0.41756009070294786
    70th percentile is 0.44897052154195016
    80th percentile is 0.48478911564625854
    90th percentile is 0.5549160997732426
##print 90 to 100 percentile values with step size of 1.
for i in range(0,100,10):
 print(f"{i}th percentile is {np.percentile(X test processed['duration'].values,i)}")
    Oth percentile is 0.15741496598639457
    10th percentile is 0.2595147392290249
    20th percentile is 0.29777777777775
    30th percentile is 0.3309115646258503
    40th percentile is 0.35822222222222
    50th percentile is 0.38834467120181404
    60th percentile is 0.41619047619047617
    70th percentile is 0.44286167800453513
    80th percentile is 0.4827573696145125
    90th percentile is 0.5569433106575965
```

Grader function 4

```
flag_columns = (all(X_train_processed.columns==['raw_data', 'duration'])) and (all(X_test_flag_shape = (X_train_processed.shape ==(1400, 2)) and (X_test_processed.shape==(600,2)) return flag_columns and flag_shape grader_processed()

True
```

Based on our analysis 99 percentile values are less than 0.8sec so we will limit maximum length of X_train_processed and X_test_processed to 0.8 sec. It is similar to pad_sequence for a text dataset.

While loading the audio files, we are using sampling rate of 22050 so one sec will give array of length 22050. so, our maximum length is 0.8*22050 = 17640 Pad with Zero if length of sequence is less than 17640 else Truncate the number.

Also create a masking vector for train and test.

masking vector value = 1 if it is real value, 0 if it is pad value. Masking vector data type must be bool.

```
max_length = 17640

## as discussed above, Pad with Zero if length of sequence is less than 17640 else Truncate t
## save in the X_train_pad_seq, X_test_pad_seq
## also Create masking vector X_train_mask, X_test_mask

## all the X_train_pad_seq, X_test_pad_seq, X_train_mask, X_test_mask will be numpy arrays ma
from tensorflow.keras.preprocessing.sequence import pad_sequences
X_train_pad_seq = pad_sequences(X_train_processed['raw_data'],maxlen=max_length,dtype='float3
X_test_pad_seq = pad_sequences(X_test_processed['raw_data'],maxlen=max_length,dtype='float32'
X_train_mask = np.array([i==0 for i in X_train_pad_seq])
X_test_mask = np.array([i==0 for i in X_test_pad_seq])
```

Grader function 5

```
def grader_padoutput():
    flag_padshape = (X_train_pad_seq.shape==(1400, 17640)) and (X_test_pad_seq.shape==(600, 1
    flag_maskshape = (X_train_mask.shape==(1400, 17640)) and (X_test_mask.shape==(600, 17640)
    flag_dtype = (X_train_mask.dtype==bool) and (X_test_mask.dtype==bool)
    return flag_padshape and flag_maskshape and flag_dtype
grader_padoutput()
```

True

▼ 1. Giving Raw data directly.

Now we have

Train data: X_train_pad_seq, X_train_mask and y_train Test data: X_test_pad_seq, X_test_mask and y_test

We will create a LSTM model which takes this input.

Task:

- Create an LSTM network which takes "X_train_pad_seq" as input, "X_train_mask" as mask input. You can use any number of LSTM cells. Please read LSTM documentation(https://www.tensorflow.org/api_docs/python/tf/keras/layers/LSTM) in tensorflow to know more about mask and also https://www.tensorflow.org/guide/keras/masking_and_padding
- 2. Get the final output of the LSTM and give it to Dense layer of any size and then give it to Dense layer of size 10(because we have 10 outputs) and then compile with the sparse categorical cross entropy(because we are not converting it to one hot vectors). Also check the datatype of class labels(y_values) and make sure that you convert your class labels to integer datatype before fitting in the model.
- 3. While defining your model make sure that you pass both the input layer and mask input layer as input to lstm layer as follows

```
lstm_output = self.lstm(input_layer, mask=masking_input_layer)
```

- 4. Use tensorboard to plot the graphs of loss and metric(use custom micro F1 score as metric) and histograms of gradients. You can write your code for computing F1 score using this <u>link</u>
- 5. make sure that it won't overfit.
- 6. You are free to include any regularization

```
y_train = np.array([int(i) for i in y_train])
y_test = np.array([int(i) for i in y_test])

from tensorflow.keras.layers import Input, LSTM, Dense
from tensorflow.keras.models import Model
import tensorflow as tf

#https://imgur.com/8YULUcu
from sklearn.metrics import f1_score
class Metrics(tf.keras.callbacks.Callback):
    def __init__(self, validation_data):
        super().__init__()
```

```
self.x_test = validation_data[0]
   self.y test = validation data[1]
 def on epoch end(self,epoch,logs={}):
   val_predict = (np.asarray(self.model.predict(self.x test)))
   val_label = np.argmax(val_predict,axis = 1)
   val targ = self.y test
   val_f1 = f1_score(val_targ, val_label, average='micro')
   print(" val f1 score", val f1)
## as discussed above, please write the architecture of the model.
## you will have two input layers in your model (data input layer and mask input layer)
## make sure that you have defined the data type of masking layer as bool
input = Input(shape=(X train pad seq.shape[1],1),name='Input Layer')
mask = Input(shape=(X train mask.shape[1]),dtype='bool',name='Mask')
lstm = LSTM(30)(input, mask = mask)
dense = Dense(32)(1stm)
output = Dense(10,activation='softmax')(dense)
model = Model(inputs=[input, mask],outputs=output)
model.summary()
```

Model: "model"

Layer (type)	Output Shape	Param #	Connected to
Input Layer (InputLayer)	[(None, 17640, 1)]	0	[]
Mask (InputLayer)	[(None, 17640)]	0	[]
lstm (LSTM)	(None, 30)	3840	['Input Layer[0][0]', 'Mask[0][0]']
dense (Dense)	(None, 32)	992	['lstm[0][0]']
dense_1 (Dense)	(None, 10)	330	['dense[0][0]']

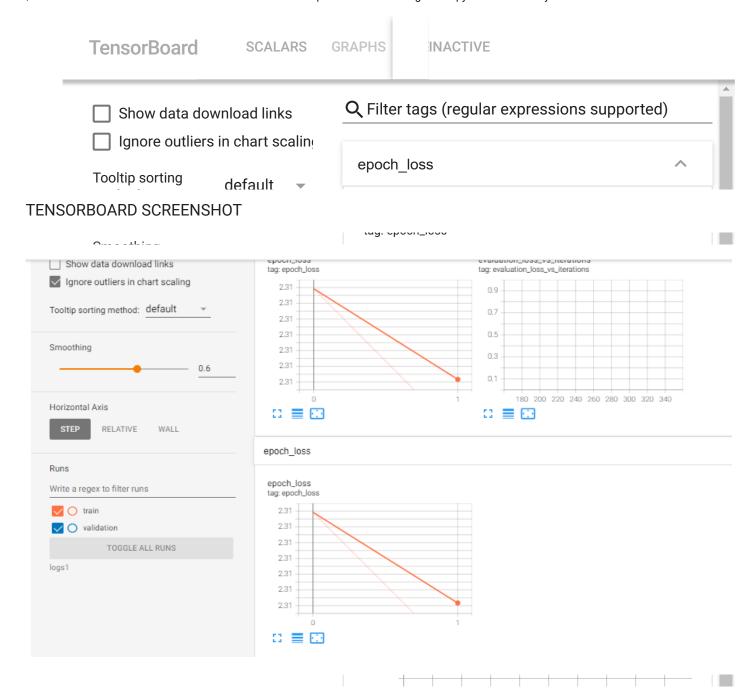
Total params: 5,162 Trainable params: 5,162 Non-trainable params: 0

```
from tensorflow.keras.callbacks import TensorBoard
```

```
tensorboard_callback = TensorBoard(log_dir='logs1',write_graph=True,histogram_freq=1)
metric = Metrics(([X test pad seq, X test mask], y test))
```

```
#train your model
model.fit([X_train_pad_seq, X_train_mask],y_train,validation_data=([X_test_pad_seq, X_test_ma
```

%reload_ext tensorboard
%tensorboard --logdir logs1



2. Converting into spectrogram and giving spectrogram data as input

We can use librosa to convert raw data into spectrogram. A spectrogram shows the features in a two-dimensional representation with the intensity of a frequency at a point in time i.e we are converting Time domain to frequency domain. you can read more about this in https://pnsn.org/spectrograms/what-is-a-spectrogram

```
def convert_to_spectrogram(raw_data):
    '''converting to spectrogram'''
    spectrum = librosa.feature.melspectrogram(y=raw_data, sr=sample_rate, n_mels=64)
    logmel_spectrum = librosa.power_to_db(S=spectrum, ref=np.max)
    return logmel spectrum
```

```
##use convert to spectrogram and convert every raw sequence in X train pad seq and X test pad
## save those all in the X_train_spectrogram and X_test_spectrogram ( These two arrays must b
X train spectrogram = []
X test spectrogram = []
for i in X train pad seq:
 X_train_spectrogram.append(convert_to_spectrogram(i))
for i in X test pad seq:
 X test spectrogram.append(convert to spectrogram(i))
X_train_spectrogram = np.array(X_train_spectrogram)
X_test_spectrogram = np.array(X_test_spectrogram)
X train spectrogram[0]
    array([[-32.064407, -29.277729, -30.074486, ..., -80. , -80.
           [-24.482014, -18.270372, -16.526098, ..., -80. , -80.
           [-19.260283, -14.424799, -15.12985, ..., -80.
                                                             , -80.
            -80.
                      1,
           . . . ,
           [-80.
                      , -80. , -80. , ..., -80.
                                                        , -80.
            -80.
                     ٦,
                     , -80.
                                 , -80. , ..., -80. , -80.
           [-80.
            -80.
                    1,
                                 , -80. , ..., -80.
                     , -80.
                                                             , -80.
           [-80.
                      ]], dtype=float32)
            -80.
```

Grader function 6

```
def grader_spectrogram():
    flag_shape = (X_train_spectrogram.shape==(1400,64, 35)) and (X_test_spectrogram.shape ==
    return flag_shape
grader_spectrogram()

True
```

Now we have

Train data: X_train_spectrogram and y_train Test data: X_test_spectrogram and y_test

We will create a LSTM model which takes this input.

Task:

- 1. Create an LSTM network which takes "X_train_spectrogram" as input and has to return output at every time step.
- 2. Average the output of every time step and give this to the Dense layer of any size. (ex: Output from LSTM will be (None, time_steps, features) average the output of every time step i.e, you should get (None,time_steps) and then pass to dense layer)
- 3. give the above output to Dense layer of size 10(output layer) and train the network with sparse categorical cross entropy.
- 4. Use tensorboard to plot the graphs of loss and metric(use custom micro F1 score as metric) and histograms of gradients. You can write your code for computing F1 score using this <u>link</u>
- 5. make sure that it won't overfit.
- 6. You are free to include any regularization

```
#https://stackoverflow.com/questions/50309488/how-to-get-the-average-of-a-time-series-lstm-ke
from tensorflow.keras.layers import GlobalAveragePooling1D
tf.keras.backend.clear session()
```

```
input = Input(shape=(64,35),name='Input Layer')
lstm = LSTM(200,return_sequences=True)(input)
avg = GlobalAveragePooling1D()(lstm)
dense = Dense(200, activation = 'relu')(avg)
output = Dense(10,activation='softmax')(dense)
model2 = Model(inputs=input ,outputs=output)
model2.summary()
```

Model: "model"

Layer (type)	Output Shape	Param #
Input Layer (InputLayer)	[(None, 64, 35)]	0
lstm (LSTM)	(None, 64, 200)	188800
<pre>global_average_pooling1d (G lobalAveragePooling1D)</pre>	(None, 200)	0
dense (Dense)	(None, 200)	40200
dense_1 (Dense)	(None, 10)	2010

Total params: 231,010 Trainable params: 231,010 Non-trainable params: 0

from tensorflow.keras.callbacks import TensorBoard

```
tensorboard_callback = TensorBoard(log_dir='logs2',write_graph=True,histogram_freq=1)
metric = Metrics((X_test_spectrogram, y_test))
```

model2.compile(loss='sparse_categorical_crossentropy',optimizer=tf.keras.optimizers.Adam(lear

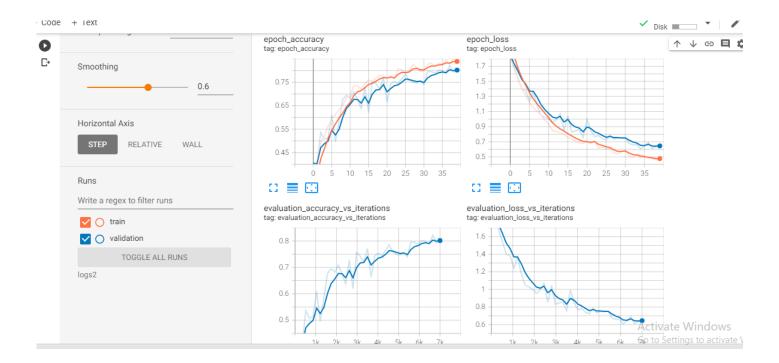
model2.fit(X train spectrogram, y train, validation data=(X test spectrogram, y test), batch siz

```
Epoch 1/40
1/175 [.....] - ETA: 57:47 - loss: 2.3538 - accuracy: 0.00
Epoch 2/40
Epoch 5/40
Epoch 6/40
Epoch 7/40
Epoch 8/40
175/175 [============= ] - 1s 8ms/step - loss: 1.1062 - accuracy: 0.6
Epoch 9/40
175/175 [============= ] - 1s 7ms/step - loss: 1.0504 - accuracy: 0.64
Epoch 10/40
175/175 [============== ] - 1s 8ms/step - loss: 0.9855 - accuracy: 0.6
Epoch 12/40
Epoch 13/40
Epoch 14/40
175/175 [============== ] - 1s 8ms/step - loss: 0.8165 - accuracy: 0.7
```

```
Epoch 15/40
Epoch 16/40
175/175 [============ ] - 1s 8ms/step - loss: 0.7561 - accuracy: 0.7
Epoch 17/40
175/175 [============ ] - 1s 8ms/step - loss: 0.7356 - accuracy: 0.7
Epoch 18/40
175/175 [============ ] - 1s 8ms/step - loss: 0.7073 - accuracy: 0.7
Epoch 19/40
_____1 ETA. Ac locc. A 696E
                          3661193611 0 7702
```

%reload_ext tensorboard
%tensorboard --logdir logs2





3. Data augmentation with raw features

Till now we have done with 2000 samples only. It is very less data. We are giving the process of generating augmented data below.

There are two types of augmentation:

- 1. time stretching Time stretching either increases or decreases the length of the file. For time stretching we move the file 30% faster or slower
- 2. pitch shifting pitch shifting moves the frequencies higher or lower. For pitch shifting we shift up or down one half-step.

```
## generating augmented data.
def generate_augmented_data(file_path):
    augmented_data = []
    samples = load_wav(file_path,get_duration=False)
    for time_value in [0.7, 1, 1.3]:
```

▼ Follow the steps

- 1. Split data 'df_audio' into train and test (80-20 split)
- 2. We have 2000 data points (1600 train points, 400 test points)

```
X_train, X_test, y_train, y_test=train_test_split(df_audio['path'],df_audio['label'],random_s
```

- 3. Do augmentation only on X_train,pass each point of X_train to generate_augmented_data function. After augmentation we will get 14400 train points. Make sure that you are augmenting the corresponding class labels (y_train) also.
- 4. Preprocess your X_test using load_way function.
- 5. Convert the augmented_train_data and test_data to numpy arrays.
- 6. Perform padding and masking on augmented_train_data and test_data.
- 7. After padding define the model similar to model 1 and fit the data

Note - While fitting your model on the augmented data for model 3 you might face Resource exhaust error. One simple hack to avoid that is save the augmented_train_data,augment_y_train,test_data and y_test to Drive or into your local system. Then restart the runtime so that now you can train your model with full RAM capacity. Upload these files again in the new runtime session perform padding and masking and then fit your model.

```
train_aug = []
for i in X_train:
    train_aug.extend(generate_augmented_data(i))
test = []
for i in X_test:
    x,y = load_wav(i)
```

```
test.append(x)
train_aug = np.array(train_aug)
test = np.array(test)
     /usr/local/lib/python3.7/dist-packages/ipykernel launcher.py:8: VisibleDeprecationWarnir
     /usr/local/lib/python3.7/dist-packages/ipykernel launcher.py:9: VisibleDeprecationWarnir
       if __name__ == '__main__':
y_aug = []
for i in y train:
  ele = [i] * 9
 y_aug.extend(ele)
X train pad seq = pad sequences(train aug, maxlen=max length, dtype='float32', padding='post', t
X_test_pad_seq = pad_sequences(test,maxlen=max_length,dtype='float32', padding='post',truncat
X train mask = np.array([i==0 for i in X train pad seq])
X test mask = np.array([i==0 for i in X test pad seq])
y_train = np.array([int(i) for i in y_aug])
y_test = np.array([int(i) for i in y_test])
X_train_pad_seq = np.array(X_train_pad_seq)
X test pad seq = np.array(X test pad seq)
input = Input(shape=(X train pad seq.shape[1],1),name='Input Layer')
mask = Input(shape=(X_train_mask.shape[1]),dtype='bool',name='Mask')
lstm = LSTM(25)(input, mask = mask)
dense = Dense(32)(1stm)
output = Dense(10,activation='softmax')(dense)
model3 = Model(inputs=[input, mask],outputs=output)
model3.summary()
```

Model: "model"

Layer (type)	Output Shape	Param #	Connected to
Input Layer (InputLayer)	[(None, 17640, 1)]	0	[]
Mask (InputLayer)	[(None, 17640)]	0	[]
lstm (LSTM)	(None, 25)	2700	['Input Layer[0][0]', 'Mask[0][0]']
dense (Dense)	(None, 32)	832	['lstm[0][0]']
dense_1 (Dense)	(None, 10)	330	['dense[0][0]']

```
Total params: 3,862
Trainable params: 3,862
Non-trainable params: 0
```

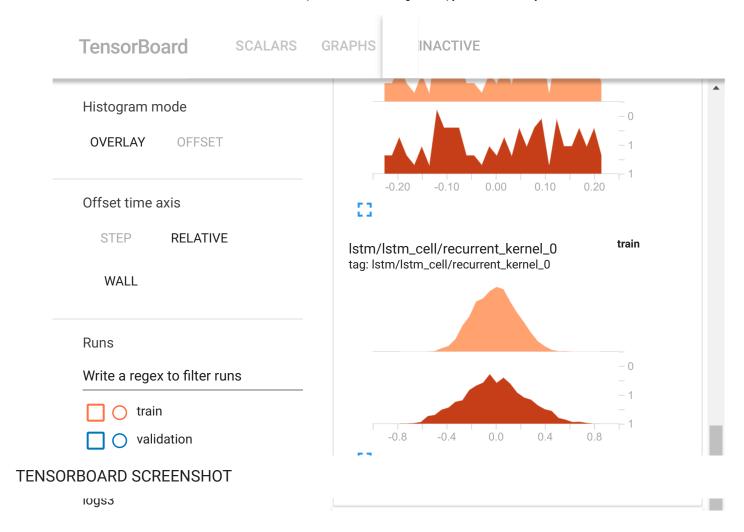
4

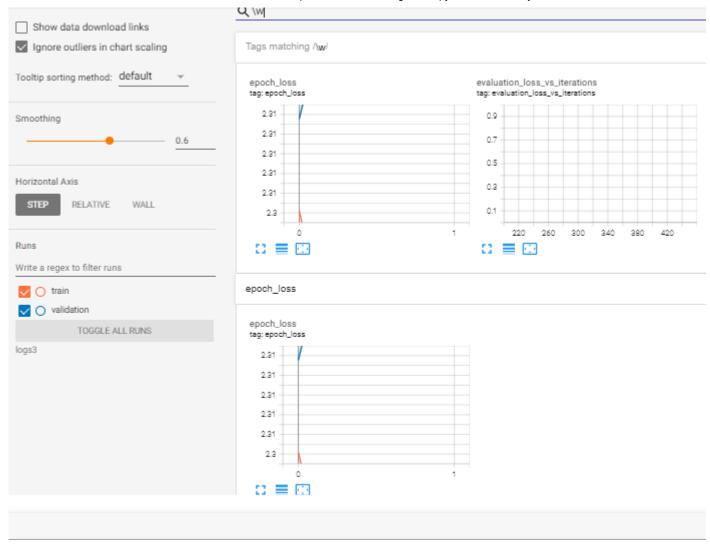
from tensorflow.keras.callbacks import TensorBoard

```
tensorboard_callback = TensorBoard(log_dir='logs3',write_graph=True,histogram_freq=1)
metric = Metrics(([X_test_pad_seq, X_test_mask], y_test))
```

```
epochs=2, steps_per_epoch=X_train_pad_seq.shape[0]//64
callbacks=[tensorboard_callback,metric])
```

%reload_ext tensorboard
%tensorboard --logdir logs3





▼ 4. Data augmentation with spectogram data

- 1. use convert_to_spectrogram and convert the padded data from train and test data to spectogram data.
- 2. The shape of train data will be 14400 x 64 x 35 and shape of test_data will be 400 x 64 x 35
- 3. Define the model similar to model 2 and fit the data

```
X_train_spectrogram = []
X_test_spectrogram = []
for i in X_train_pad_seq:
    X_train_spectrogram.append(convert_to_spectrogram(i))
for i in X_test_pad_seq:
    X_test_spectrogram.append(convert_to_spectrogram(i))
X_train_spectrogram = np.array(X_train_spectrogram)
X_test_spectrogram = np.array(X_test_spectrogram)
```

```
#https://stackoverflow.com/questions/50309488/how-to-get-the-average-of-a-time-series-lstm-ke
from tensorflow.keras.layers import GlobalAveragePooling1D

tf.keras.backend.clear_session()

input = Input(shape=(64,35),name='Input Layer')
lstm = LSTM(200,return_sequences=True)(input)
avg = GlobalAveragePooling1D()(lstm)
dense = Dense(200, activation = 'relu')(avg)
output = Dense(10,activation='softmax')(dense)
model4 = Model(inputs=input ,outputs=output)
model4.summary()
```

Model: "model"

Layer (type)	Output Shape	Param #
Input Layer (InputLayer)	[(None, 64, 35)]	0
lstm (LSTM)	(None, 64, 200)	188800
<pre>global_average_pooling1d (0 lobalAveragePooling1D)</pre>	G (None, 200)	0
dense (Dense)	(None, 200)	40200
dense_1 (Dense)	(None, 10)	2010

Total params: 231,010 Trainable params: 231,010 Non-trainable params: 0

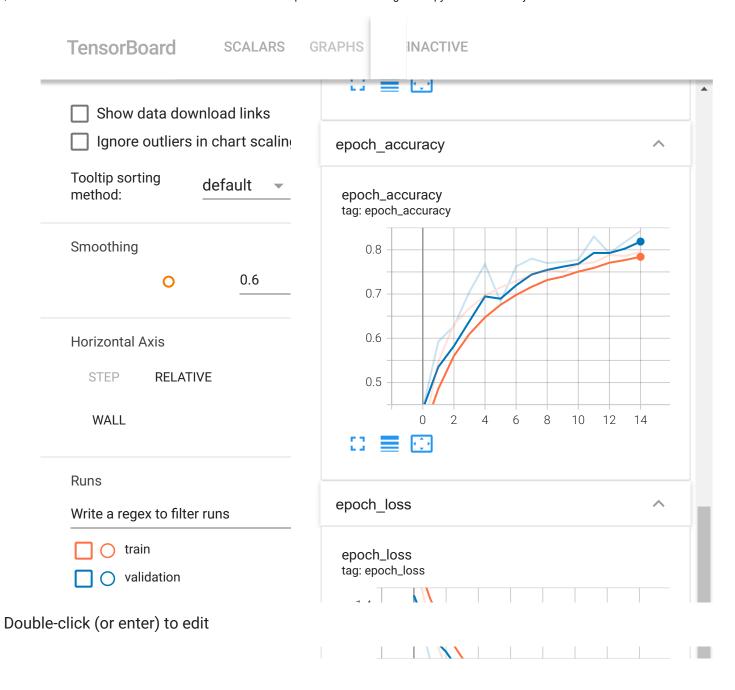
from tensorflow.keras.callbacks import TensorBoard

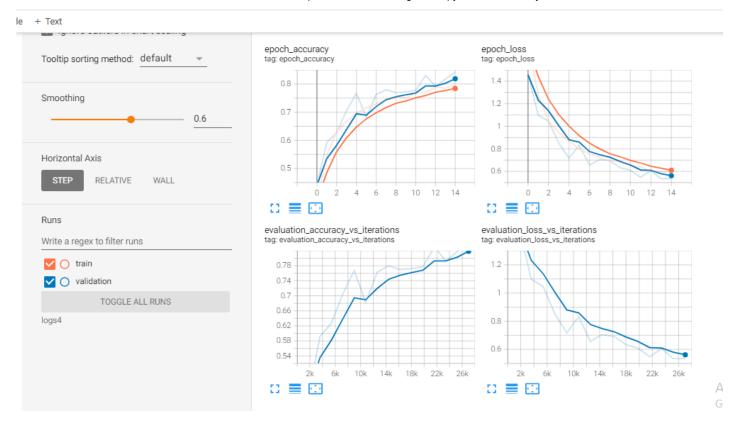
```
tensorboard_callback = TensorBoard(log_dir='logs4',write_graph=True,histogram_freq=1)
metric = Metrics((X_test_spectrogram, y_test))
```

model4.compile(loss='sparse_categorical_crossentropy',optimizer=tf.keras.optimizers.Adam(lear model4.fit(X_train_spectrogram,y_train,validation_data=(X_test_spectrogram, y_test),batch_siz

```
Epoch 4/15
Epoch 5/15
Epoch 6/15
Epoch 7/15
Epoch 8/15
Epoch 12/15
Epoch 13/15
Epoch 14/15
Epoch 15/15
<keras.callbacks.History at 0x7fa487570250>
```

%reload_ext tensorboard
%tensorboard --logdir logs4





1. The reason why accuracy and f1-score is almost the same because the formula to calculate the both is similar. In case multi-class classification where there is no true negative or false negative - it's just that a point is either classified correctly or not. Let a point being correctly classified be true positive and incorrectly classified as false positive, then

- 2. Accuracy = (tp+tn) / (tp+tn+fp+fn) = 2tp / (2tp + 2fp)
- 3. F1 score = 2tp / (2tp + fp + fn) = 2tp / (2tp + 2fp)

✓ 3s completed at 9:52 PM

×