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EE 679: Computation Assignment 3

The functions required and overall pipeline for GMM-HMM is provided here.

Code for connecting to drive folder (ignore/do not run if not on google colab)

In [2]:

```
from google.colab import drive
drive.mount('/content/gdrive/', force_remount=True)
import os
root_dir = "/content/gdrive/MyDrive/EE 679 Speech Processing Assignments/"
project_folder = "3"

def create_and_set_working_directory(project_folder):
  if os.path.isdir(root_dir + project_folder) == False:
    os.mkdir(root_dir + project_folder)
    print(root_dir + project_folder + ' did not exist but was created.')
  os.chdir(root_dir + project_folder)

create_and_set_working_directory(project_folder)
! pwd
```

```
Mounted at /content/gdrive/
/content/gdrive/MyDrive/EE 679 Speech Processing Assignments/3
```

Importing packages that are required in later sections

In [3]:

```
! pip install hmmlearn
#!cp -r /root/.local/lib/python3.7/site-packages/hmmlearn /usr/local/lib/python3.7/dist-p
ackages/
import hmmlearn
from hmmlearn import hmm
import numpy as np
import matplotlib.pyplot as plt
from IPython.display import Audio
import scipy.signal as sp
import librosa
import librosa.display
import soundfile as sf
from scipy.signal import find_peaks

def save_as_wav(y_input,file_name,F_samp):
  y_norm = ((y_input-np.min(y_input))/(np.max(y_input)-np.min(y_input))) - 0.5 #making m
ean = 0 and swing = 1
  sf.write(file_name+'.wav',y_norm,F_samp,'PCM_24')

def play_sound(file_name):
  """file_name: the name of the audio file along with the extension"""
  audio = Audio(filename='./'+file_name)
  display(audio)
```

```
Requirement already satisfied: hmmlearn in /usr/local/lib/python3.7/dist-packages (0.2.6)
Requirement already satisfied: numpy>=1.10 in /usr/local/lib/python3.7/dist-packages (fro
```

```
m hmmlearn) (1.19.5)
Requirement already satisfied: scikit-learn>=0.16 in /usr/local/lib/python3.7/dist-packag
es (from hmmlearn) (1.0.1)
Requirement already satisfied: scipy>=0.19 in /usr/local/lib/python3.7/dist-packages (fro
m hmmlearn) (1.4.1)
Requirement already satisfied: threadpoolctl>=2.0.0 in /usr/local/lib/python3.7/dist-pack
ages (from scikit-learn>=0.16->hmmlearn) (3.0.0)
Requirement already satisfied: joblib>=0.11 in /usr/local/lib/python3.7/dist-packages (fr
om scikit-learn>=0.16->hmmlearn) (1.1.0)
```

Code for obtaining the MFCC Coefficients

In [4]:

```
def window_mult_and_padding(input_signal,F_s,t_start,window_type,t_window,N):
  """input_signal: the speech waveform segment input to the system
  window: the windowing function that will be used for STFT analysis, it has support = L
points
  window_start_index: the index of the input signal from which the window's starting poin
t will be multiplied
  Output: windowed input signal for sample points [window_start_index,window_start_index+
L-1], zero padded to total N points
  Thus final output: x[n]w[n]: [window_start_index,window_start_index+L-1] and zeros: [wi
ndow_start_index+L,window_start_index+N-1]"""

  L = int(t_window*F_s) #support of the window
  window_start_index = int(t_start*F_s) #starting point of the window placement

  if window_type == "rectangle":
    window = np.ones(L) #rectangular window function
  else: #by default hamming window is selected(better)
    window = sp.windows.hamming(L, sym=True) #hamming window function

  windowed_signal = np.zeros(N)
  windowed_signal[0:L] = input_signal[window_start_index:window_start_index+L]*window

  return windowed_signal

def calc_feature_vector(in_signal,n_hop,n_window):

  in_signal = pre_emphasize(in_signal,0.99); #comment out in case pre-emphasis is not req
uired
  mel_spect = librosa.feature.melspectrogram(y=in_signal,sr=F_s,n_fft=1024,hop_length=n_
hop,window="hamming",win_length=n_window)
  mfcc = librosa.feature.mfcc(S=librosa.power_to_db(mel_spect),n_mfcc=13)
  delta1 = np.zeros(mfcc.shape); delta2 = np.zeros(mfcc.shape);
  delta1[:,0] = mfcc[:,0]; delta2[:,0] = mfcc[:,0];
  delta1[:,mfcc.shape[1]-1] = mfcc[:,mfcc.shape[1]-1]; delta2[:,mfcc.shape[1]-1] = mfcc[
:,mfcc.shape[1]-1];
  for n in range(1,mfcc.shape[1]-1):
    delta1[:,n] = (mfcc[:,n+1]-mfcc[:,n-1])/2
  for n in range(1,mfcc.shape[1]-1):
    delta2[:,n] = (delta1[:,n+1]-delta1[:,n-1])/2

  feat_vec = np.concatenate((mfcc,delta1,delta2), axis=0)
  return np.transpose(feat_vec) #each row is a observation and each column a feature

def generating_codebook(feat_vecs,num_centroids,plot_distortions=False):
  """feat_vecs: contains all the 39-length feature vectors(as rows) in for a particular u
tterance
  num_centroids: choose between 6 to 64, check the distortion for best match
  Returned values:
  centroids: a num_centroids*39 matrix containing all the estimated centroids from the KM
C algorithm
  sd_vec: a 39 length, capturing the original SD value for all of the features, will be h
elpful in testing part
  """
  from scipy.cluster.vq import vq, kmeans, whiten
  white_feat_vecs = whiten(feat_vecs)
```

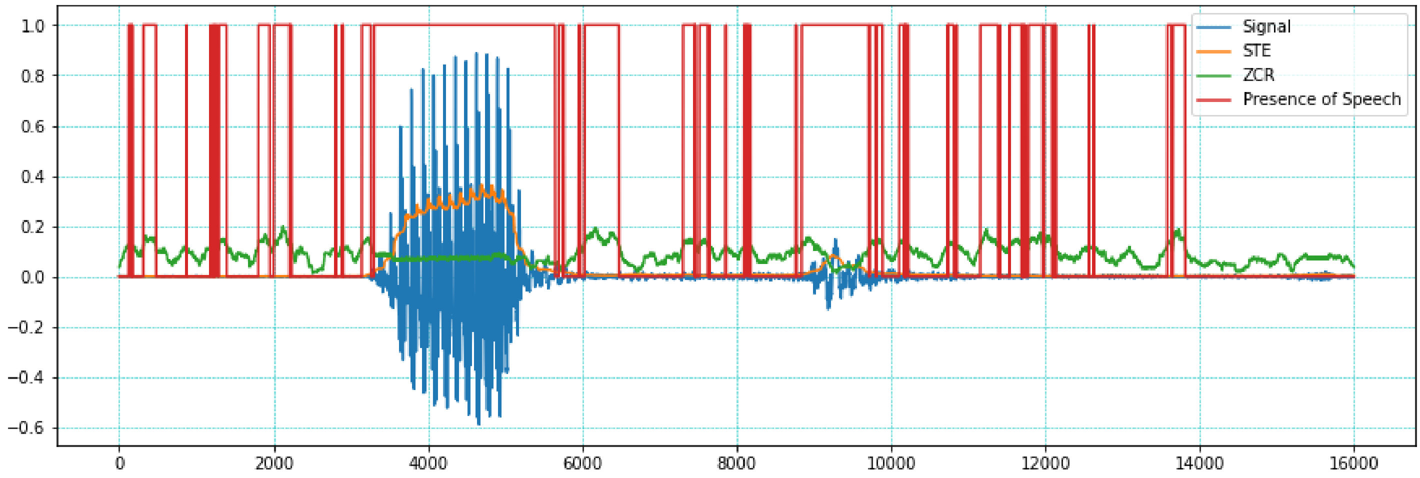
```
  if plot_distortions:
    distortion = np.zeros(num_centroids)
    for j in range(1,num_centroids+1):
      codebook, distortion_this = kmeans(white_feat_vecs, j)
      distortion[j-1] = distortion_this
    plt.plot(distortion); plt.show()

  codebook, distortion = kmeans(white_feat_vecs, num_centroids)
  sd_vec = np.std(feat_vecs, axis=0) #calculates and returns the sd of each column of dat
a
  #note that, the whitened matrix, each of the column entries feat_vecs is divided by the
sd of same feat_vecs column
  return sd_vec, codebook, distortion #the codebook has the required num_centroids vector
s as rows

def normalize_audio(x):
  """The range of output audio will be -1 to 1 for any input given, a DC shift and consta
nt multiplication doesn't changes the utterance"""
  high = np.max(x); low = np.min(x)
  y = (2*x-low-high)/(high-low)
  return y

def end_pointing_using_STE(x,F_s,t_window,th_frac_ste,th_frac_zcr,ZCR=False):
  """x: input signal
  th_frac_ste: threshold for STE, th_frac_zcr: threshold for ZCR
  output: modified version of signal x, s.t. silences are removed"""
  n_per_frame = int(t_window*F_s);
  STE = np.transpose(librosa.feature.rms(x,frame_length=n_per_frame,hop_length=1)); ste_
th = th_frac_ste*np.max(STE);
  speech_present = (STE>ste_th)

  if ZCR:
    ZCR = np.transpose(librosa.feature.zero_crossing_rate(x,frame_length=n_per_frame,hop
_length=1)); zcr_th = th_frac_zcr*np.max(ZCR);
    speech_present = np.logical_or(STE>ste_th,ZCR>zcr_th)
  speech_present = speech_present[1:,0] #making the size correct, now select out the rele
vant entries of x

  return x[speech_present]

def pre_emphasize(x,alpha):
  y = librosa.effects.preemphasis(x,coef=alpha)

  return y
```

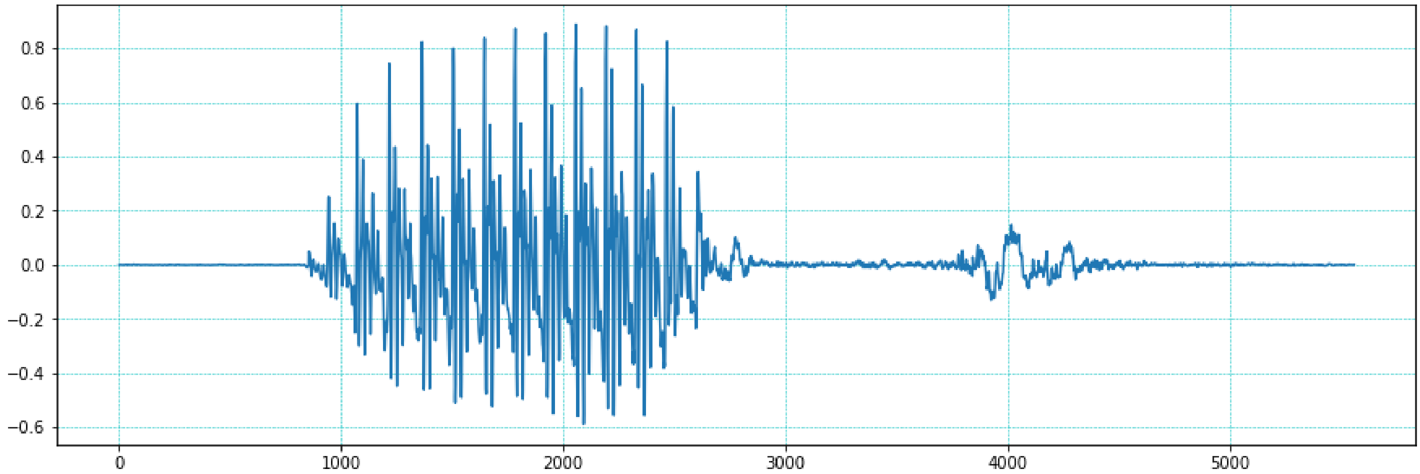
In [5]:

```
#Listening to the sound wave first
file_name = "sound_10.wav"
x, F_s = sf.read(file_name); #x = normalize_audio(x)
play_sound(file_name)
t = np.arange(0,x.shape[0])/F_s

#testing of preprocessing algorithms
t_window = 10e-3; n_per_frame = int(t_window*F_s);
th_frac_ste = 0.05; th_frac_zcr = 0.6;
STE = np.transpose(librosa.feature.rms(x,frame_length=n_per_frame,hop_length=1)); ste_th
= th_frac_ste*np.max(STE);
ZCR = np.transpose(librosa.feature.zero_crossing_rate(x,frame_length=n_per_frame,hop_len
gth=1)); zcr_th = th_frac_zcr*np.max(ZCR);

#speech_present = (STE>ste_th)
speech_present = np.logical_or(STE>ste_th,ZCR>zcr_th)

speech_present = speech_present[1:,0]
#plotting signal with RMSE and ZCR
plt.figure(figsize = (15,5))
plt.plot(x,label='Signal'); plt.plot(STE,label='STE'); plt.plot(ZCR,label='ZCR'); plt.plo
t(speech_present,label='Presence of Speech');
plt.grid(color = 'c', linestyle = '--', linewidth = 0.5)
```

```
plt.legend()
plt.show()

#showing signal after end-pointing
x_end_point = end_pointing_using_STE(x,F_s,10e-3,0.1,0.6,True)
plt.figure(figsize = (15,5))
plt.plot(x_end_point);
plt.grid(color = 'c', linestyle = '--', linewidth = 0.5); plt.show()

Audio(data=x_end_point,rate=F_s)
```

Your browser does not support the audio element.





Out[5]:

Your browser does not support the audio element.

Reading from files and making feature vectors

In [6]:

```
permanent_dir = '/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3'
os.chdir(permanent_dir)

train_adress = permanent_dir + "/DATASET" + "/train"
path = train_adress
train_list = os.listdir(path)
print("Files and directories in '", path, "' :")
print(train_list)

#required order = ['go', 'left', 'down', 'yes', 'stop', 'on', 'off', 'up', 'no', 'right']
```

```
Files and directories in ' /content/gdrive/My Drive/EE 679 Speech Processing Assignments/
3/DATASET/train ' :
['go', 'left', 'down', 'yes', 'stop', 'on', 'off', 'up', 'no', 'right']
```

In [ ]:

```
t_window = 20e-3; n_window = int(t_window*F_s)
t_hop = 10e-3; n_hop = int(t_hop*F_s)

num_train = 2000 #keep it less than the min number of utterances available for any class value
all_words_feats_list = []; lengths_each_sample = [];

for folders in range(len(train_list)):
  feat_vec_this_word = np.zeros((1,39))
  lengths_each_sample_this_word = []
  word_path = path + "/" + train_list[folders]
  word_list = os.listdir(word_path)
  #print(word_list);
  print(word_path);
  os.chdir(word_path)
  #num_utterances = len(word_list)
  num_utterances = num_train #using a little less than all training samples
  for files in range(num_utterances):
    file_name = word_list[files]
    x,_ = sf.read(file_name);
    #x = end_pointing_using_STE(x,F_s,10e-3,0.1,0.6,True) #for end-pointing, set the thresholds appropriately, comment out if end-pointing not required
    x = normalize_audio(x); #comment out if normalization is not required
    #print(x.shape)
    feat_this = calc_feature_vector(x,n_hop,n_window)
    lengths_each_sample_this_word.append(feat_this.shape[0])
    #print(feat_this.shape)
    feat_vec_this_word = np.concatenate((feat_vec_this_word,feat_this),axis=0)
    #print(file_name); print("******")
    #play_sound(file_name)

  feat_vec_this_word = feat_vec_this_word[1:,:] #to remove the first all zero row
  print(feat_vec_this_word.shape)
  lengths_each_sample.append(np.array(lengths_each_sample_this_word))
  all_words_feats_list.append(feat_vec_this_word)

#returning back to the original directory
%cd '/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/'
```

```
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/train/go
(197934, 39)
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/train/left
(199571, 39)
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/train/down
(199144, 39)
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/train/yes
(199103, 39)
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/train/stop
(199051, 39)
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/train/on
(198335, 39)
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/train/off
(199108, 39)
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/train/up
(197614, 39)
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/train/no
(198367, 39)
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/train/right
(198938, 39)
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3
```

In [9]:

```
#all_words_feats_list = np.load("hmm_feats_list.npy",allow_pickle=True)
#lengths_each_sample = np.load("hmm_feats_lengths.npy",allow_pickle=True)

all_words_feats_list = np.load("hmm_feats_list.npy",allow_pickle=True)
lengths_each_sample = np.load("hmm_feats_lengths.npy",allow_pickle=True)
```

In [ ]:

```
feats_list = np.array(all_words_feats_list)
lengths_list = np.array(lengths_each_sample)

np.save("hmm_feats_list.npy", feats_list)
np.save("hmm_feats_lengths.npy", lengths_list)
```

```
/usr/local/lib/python3.7/dist-packages/ipykernel_launcher.py:1: VisibleDeprecationWarning
: Creating an ndarray from ragged nested sequences (which is a list-or-tuple of lists-or-
tuples-or ndarrays with different lengths or shapes) is deprecated. If you meant to do th
is, you must specify 'dtype=object' when creating the ndarray
  """Entry point for launching an IPython kernel.
```

Train the GMM-HMM models using each of the vectors

In [7]:

```
all_words_feats_list_noisy = np.load("aug_feats_list.npy",allow_pickle=True)
lengths_each_sample_noisy = np.load("aug_feats_lengths.npy",allow_pickle=True)

all_words_feats_list_clean = np.load("hmm_feats_list.npy",allow_pickle=True)
lengths_each_sample_clean = np.load("hmm_feats_lengths.npy",allow_pickle=True)
```

In [8]:

```
#order of words = ['go', 'left', 'down', 'yes', 'stop', 'on', 'off', 'up', 'no', 'right']
---> stored in train_list
#vectors in all_words_feats_list are as per the above order
#full_codebook = entries of the list all_words_feats_list
#shape of each of these full_codebooks = (n_samples, n_features), for our case n_features
= 39 always
#the length of the individual sequences arranged in a matrix form is given by (lengths_ea
ch_sample), an array of n_sequences,
#sum of all entries(there are total n_sequences of them) in (lengths_each_sample) = n_sam
ples

trained_model = {} # the trained GMM-HMM for each of the words will be stored in a dict,
the key being the word itself
hmms = 16; gmms = 12;
batch_size = 2000; num_batches = int(2000/batch_size);

for word in range(len(train_list)): #this loop will train the GMM-HMM model for a particu
lar word, suppose 'go'
  this_word_model = hmm.GMMHMM(n_components=hmms, n_mix=gmms, covariance_type='diag', n_
iter=10, init_params='s' , verbose=False ) #initialize the GMM-HMM to be trained

  #training_X = np.concatenate((all_words_feats_list_noisy[word],all_words_feats_list_cle
an[word]),axis=0) #choosing all the MFCC vectors for the word(the full_codebook), shape =
(n_samples, n_features)
  #lengths_this_word = np.concatenate((lengths_each_sample_noisy[word],lengths_each_sampl
e_clean[word]),axis=0)

  training_X = all_words_feats_list_noisy[word] #choosing all the MFCC vectors for the wo
rd(the full_codebook), shape = (n_samples, n_features)
  lengths_this_word = lengths_each_sample_noisy[word]

  #print(training_X.shape)
  print("Training HMM of word ",train_list[word])
  for batch_index in range(num_batches):
    print("Training for batch number: ",batch_index+1)

    lengths_this_batch = lengths_this_word[(batch_index*batch_size):((batch_index+1)*bat
ch_size)]
    training_this_batch = training_X[(lengths_this_word[0:(batch_index*batch_size)].sum(
axis=0)):(lengths_this_word[0:((batch_index+1)*batch_size)].sum(axis=0)),:]

    print(lengths_this_batch.sum()); print(training_this_batch.shape)
    this_word_model.fit(training_this_batch,lengths=lengths_this_batch) #this trains the
GMM-HMM model, the lengths_each_sample is an array of lenghth n_sequences(2000 each in ou
r case)
```

```
    #print(this_word_model)
    print("Done with HMM of word ",train_list[word])
    trained_model[train_list[word]] = this_word_model
```

```
Training HMM of word  go
Training for batch number:  1
96441
(96441, 39)
Done with HMM of word  go
Training HMM of word  left
Training for batch number:  1
87970
(87970, 39)
Done with HMM of word  left
Training HMM of word  down
Training for batch number:  1
110275
(110275, 39)
Done with HMM of word  down
Training HMM of word  yes
Training for batch number:  1
106825
(106825, 39)
Done with HMM of word  yes
Training HMM of word  stop
Training for batch number:  1
84176
(84176, 39)
Done with HMM of word  stop
Training HMM of word  on
Training for batch number:  1
108993
(108993, 39)
Done with HMM of word  on
Training HMM of word  off
Training for batch number:  1
88041
(88041, 39)
Done with HMM of word  off
Training HMM of word  up
Training for batch number:  1
71383
(71383, 39)
Done with HMM of word  up
Training HMM of word  no
Training for batch number:  1
111429
(111429, 39)
Done with HMM of word  no
Training HMM of word  right
Training for batch number:  1
90721
(90721, 39)
Done with HMM of word  right
```

Testing using the test audio files on the trained GMM-HMMs

In [9]:

```
#Code to find the best class given the HMMs

def predict_best_word(test_vec_sequences,HMM_list,train_list):
    """"test_vec: each of the test_vec_sequences is of shape (n_samples x 39)
    HMM_list: list of length 10 HMMs that are trained on the respective words
    output: vector(length = 10) which has the score calculated for each of the HMMs"""

    score = np.zeros(len(HMM_list))
    for hmm_model_index in range(len(HMM_list)):
```

```
    hmm_this = HMM_list[train_list[hmm_model_index]] #can be a 16x39
    #print(test_vec_sequences.shape)
    score[hmm_model_index] = hmm_this.score(test_vec_sequences,lengths=None)

  return score
```

In [13]:

```
test_adress = permanent_dir + "/DATASET" + "/test_noisy"
path = test_adress
test_list = os.listdir(path)
print("Files and directories in '", path, "' :")
print(test_list)
```

```
Files and directories in ' /content/gdrive/My Drive/EE 679 Speech Processing Assignments/
3/DATASET/test_noisy ' :
['up', 'left', 'no', 'stop', 'yes', 'go', 'off', 'down', 'on', 'right']
```

In [14]:

```
t_window = 20e-3; t_hop = 10e-3;
n_window = int(F_s*t_window); n_hop = int(F_s*t_hop);

num_test = 240 #keep it less than the min number of utterances available for any class va
lue
y_actual = []; y_pred = []; #this will be used to make the confusion matrices

for folders in range(len(test_list)):
  word_path = path + "/" + test_list[folders]
  word_list = os.listdir(word_path)
  #print(word_list);
  print(word_path);
  os.chdir(word_path)
  num_utterances = len(word_list)
  #num_utterances = num_test #using a little less than all test samples
  for files in range(num_utterances):
    file_name = word_list[files]
    x,_ = sf.read(file_name);
    x = end_pointing_using_STE(x,F_s,10e-3,0.1,0.6,True) #for end-pointing, set the thres
holds appropriately, comment out if end-pointing not required
    x = normalize_audio(x); #comment out if normalization is not required
    test_vec = calc_feature_vector(x,n_hop,n_window)
    #print(test_vec.shape)
    score = predict_best_word(test_vec,trained_model,train_list)
    arg_max = np.argmax(score)
    #print(train_list[arg_max])

    y_actual.append(test_list[folders]); y_pred.append(train_list[arg_max])

#returning back to the original directory
%cd '/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/'
```

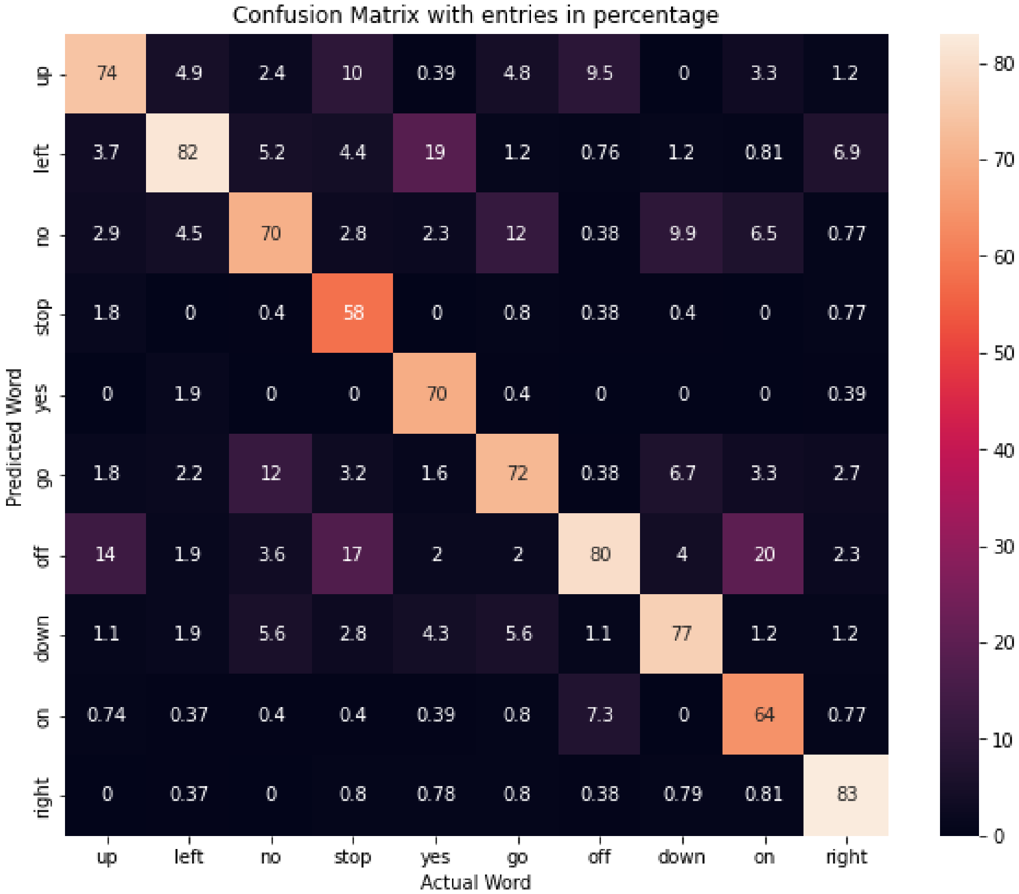
```
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/test_noisy/up
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/test_noisy/left
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/test_noisy/no
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/test_noisy/stop
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/test_noisy/yes
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/test_noisy/go
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/test_noisy/off
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/test_noisy/down
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/test_noisy/on
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3/DATASET/test_noisy/right
/content/gdrive/My Drive/EE 679 Speech Processing Assignments/3
```

Displaying confusion matrix and summary of recognition

In [15]:

```
from sklearn import metrics
conf_matrix = metrics.confusion_matrix(y_pred, y_actual, labels=test_list)
```

```
conf_matrix = 100*conf_matrix/conf_matrix.astype(np.float).sum(axis=0)

import seaborn as sns
plt.figure(figsize = (10,8))
sns.heatmap(conf_matrix, annot=True, xticklabels= test_list,yticklabels= test_list)
plt.title("Confusion Matrix with entries in percentage")
plt.xlabel("Actual Word"); plt.ylabel("Predicted Word");
plt.show()

print("SUMMARY: ")
print(metrics.classification_report(y_pred,y_actual,labels=test_list))
```




```
SUMMARY:
              precision    recall  f1-score   support

          up       0.74      0.68      0.71       295
        left       0.82      0.67      0.73       329
          no       0.70      0.62      0.66       284
        stop       0.58      0.92      0.71       157
         yes       0.70      0.96      0.81       185
          go       0.72      0.67      0.69       267
         off       0.80      0.55      0.65       378
        down       0.77      0.76      0.76       258
          on       0.64      0.84      0.73       187
       right       0.83      0.95      0.88       227

    accuracy                           0.73      2567
   macro avg       0.73      0.76      0.74      2567
weighted avg       0.74      0.73      0.73      2567
```

Save the final important matrices such as models and features

In [16]:

```
import pickle

a_file = open("HMM_16_12_acc_85_73.pkl", "wb")
```

```
pickle.dump(trained_model, a_file)
a_file.close()

a_file = open("HMM_16_12_acc_85_73.pkl", "rb")
output = pickle.load(a_file)
```