

Speech Command Recognition

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- **OVERVIEW:**

Speech recognition is the process of converting human sound signals into words or instructions. It is based on speech. It is an important research direction of speech signal processing and a branch of pattern recognition. Speech recognition applications include voice user interfaces such as voice dialing (e.g. "call home"), call routing (e.g. "I would like to make a collect call"), domestic appliance control, search key words (e.g. find a podcast where particular words were spoken), simple data entry (e.g., entering a credit card number). In similar way Speech voice recognition model is based on concepts of Convolution, LSTM , Attention and recognise pretrained voice with accuracy of **99.6%**..

- **DATA:**

- 1: Set 16KHz as sampling rate.
- 2: Record 80 utterances of each command.
- 3: Save samples of each command in different folders.

- * Data/forward.
- * Data/back.
- * Data/left.
- * Data/right.
- * Data/stop.

- **Description:**

1: Using Audacity software we have generated the data set i.e voice commands Forward, Back, Left, Right, Stop with 80 utterances for each.

2: We have splitted the data into two categories i.e Train and Test. where 20% of the data set are used for testing and the rest for training. The shape of the data set which is used for testing will be **(1775, 49, 39, 1)** and for training set, shape will be **(7033, 49, 39, 1)**

3: We have also used Melspectrogram for extracting features from the data set and also normalization is used so that the model will achieve the converge point within few epochs.

4: LSTM neural network is also used so that our weights will get updated without showing any **vanishing gradient problem** during training of models.

5: Attention is used to get the required data even from a complex sentence, Also batch normalization is used in order to prevent unexpected behaviour of the weights.

6: Finally an audio is recorded from user and the model will predict this data based on the already given training and the desired result will be displayed on screen.

- **The model is successfully built and has achieved the highest accuracy of 99.6%**

- **Model Summary**

Model: "Attention"

Layer (type)	Output Shape	Param #	Connected to
Input (InputLayer)	[(None, 49, 39, 1)]	0	
Conv1 (Conv2D)	(None, 49, 39, 10)	60	Input[0][0]
BN1 (BatchNormalization)	(None, 49, 39, 10)	40	Conv1[0][0]
Conv2 (Conv2D)	(None, 49, 39, 1)	51	BN1[0][0]
BN2 (BatchNormalization)	(None, 49, 39, 1)	4	Conv2[0][0]
Squeeze (Reshape)	(None, 49, 39)	0	BN2[0][0]
LSTM_Sequences (LSTM)	(None, 49, 64)	26624	Squeeze[0][0]
FinalSequence (Lambda)	(None, 64)	0	LSTM_Sequences[0][0]
UnitImportance (Dense)	(None, 64)	4160	FinalSequence[0][0]
AttentionScores (Dot)	(None, 49)	0	UnitImportance[0][0] LSTM_Sequences[0][0]
AttentionSoftmax (Softmax)	(None, 49)	0	AttentionScores[0][0]
AttentionVector (Dot)	(None, 64)	0	AttentionSoftmax[0][0] LSTM_Sequences[0][0]
FC (Dense)	(None, 32)	2080	AttentionVector[0][0]
Output (Dense)	(None, 5)	165	FC[0][0]

- **RUN:**

The Code is written using Google Colab:

1. Open ColabNotebook.ipynb and change Runtime to GPU.
2. Upload Speech-Recognition/Speech to Colab.
3. Change data-dir in all cells to point to Speech-Recognition/speech.
4. Run the cells in the same order in Notebook Test.

- **TEST:**

- 1: Locate the folder where you save your model.h5 file.
- 2: Start speaking when you see mike in the bottom right pane of the task bar or see red blinking dot in the title bar.

- **Language Used:**

PYTHON

- **Libraries and Packages Used:**

KAPRE, SCIKIT LEARN, SOUND FILE, TENSORFLOW.