**METHODOLOGY**

1. ANALYZING APIs FOR SPEECH RECOGNITION:

Up till now we found that speech recognition can be performed by speech recognition API and it supports multiple engine here is the list of some

* Google Speech Recognition
* Microsoft Bing Voice recognition
* CMU Sphinx (works offline)

And many more, we initiate by using Google Speech Recognition which works online and efficiency is so good but our goal is to move towards offline library and currently we are working on it by start using that Sphinx engine but accuracy is low and we are trying to improve it.

1. NOISE REMOVAL MODULE:
2. REAL TIME SOUND DETECTION MODULE:
3. DESIGN NAME PROVIDER MODULE:

For this purpose we are writing our own algorithm actually we are trying to train a model initially we train a model which can detect this is the voice of a female or a male.

* First of all for speech/speaker recognition, the most commonly used acoustic features are Mel-scale frequency cepstral coefficient (MFCC for short). MFCC takes human perception sensitivity with respect to frequencies into consideration, and therefore are best for speech/speaker recognition.
* Secondly, trying to extract features from each sample input of data set with the label(e.g. female or male) and training it on that this voice is female and this is male with large amount of sample
* Now in the end we input a file and it returns with its predict method whether it is male or female

This is our initial step towards this module.

1. CREATING A CONVERSATION FILE:

* In last we are creating a text file which will store whole conversation like this.

Male: How are you?

Female: Fine what about you?

Male: I am good