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ICA CLASSIFIER AND CROSS CORRELATION BASED SOUND SOURCE LOCALIZATION DESIGN WITH HIGH THROUGHPUT

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Abstract: Proposed paper utilize a general source separation technique, Independent Component Analysis. Particularly, basic ICA was applied to separate mixtures of low frequency, narrow band, non-Gaussian signals by using closely spaced uni-directional microphones. The localization routine worked with an average condition number of test cases. The routine was tested on data collected form MATLAB standard audio files. Localizing sounds with different frequency and time domain characteristics in a dynamic listening environment is a challenging task that has not been explored in the field of robotics as much as other perceptual tasks. This thesis presents an integrated auditory system localization method which can be used in humanoid robot, Sounds with different frequency components and time domain characteristics have to be localized
Keywords: Independent Component Analysis, Linear Discriminate Analysis, Discrete Wave Transform

I-INTRODUCTION

Sound localization is a listener's ability to identify the location or origin of a detected sound in direction and distance. It may also refer to the methods in acoustical engineering to simulate the placement of an auditory cue in a virtual 3D space (see binaural recording, wave field synthesis).

The sound localization mechanisms of the mammalian auditory system have been extensively studied. The auditory system uses several cues for sound source localization, including time- and level-differences (or intensity-difference) between both ears, spectral information, timing analysis, correlation analysis, and pattern matching.

II-PROPOSED DESIGN

Figure 1 below shows the proposed localization scenario here it can be observed that there are total 16 different sound source objects and one sound receiver in center, this situation clearly shows receiver has accuracy of observation at 360/16 = 22.5 degree, hence if source objects move by 22.5 degree proposed work will clearly recognition that.

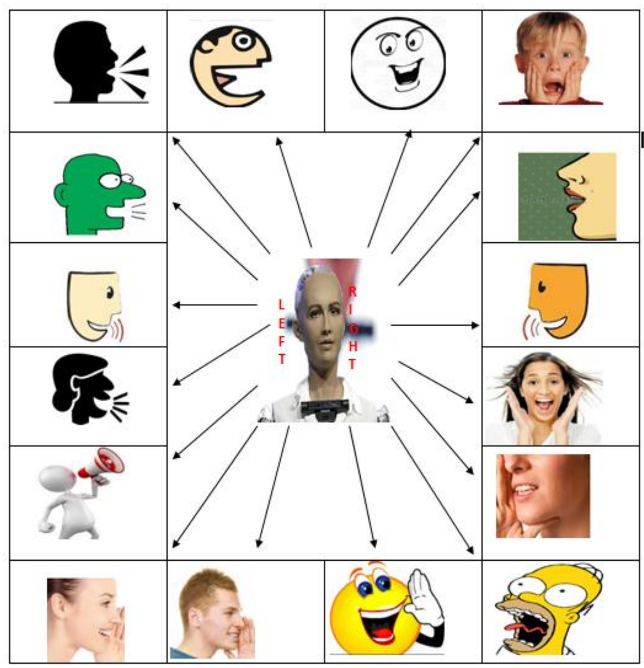

Figure 1 Audio localization Precision in Proposed Method

The position of objects can be monitor with the time difference between right and left sensor and the distance of object can be monitor with amplitude of the source object.

In figure 2 below T1 is the is the distance from the object and T1 can be computed by cross correlation between the average amplitude of different signals it tells the object location distance. T2 and T3 time difference can be computed for locating the object after its distance observation, the values of T2 and T3 helps us to identify whether the object is in left or right and more precise observation can be computed with accurate rectification of difference between T2 and T3 exact. In figure 2 the sound source 'C' has bigger difference (time taken form source to receiver T1) from the receiver as compare with sound source 'A' and Sound source 'B', hence the receiver will let know that source 'C' is far as compare with Source 'A' and source 'B'. In the same example the source 'A' and source 'B' taking same T1 time (source to receiver) in this case now receiver need to identify the location of sources and because the receiver is a small object and there is very small difference between its left and right audio sensor a accurate method required to compute the difference between the T2 (time taken form source to receiver Right sensor) and T3 (time taken from source to receiver left sensor), hence Cross correlation cannot be used and ICA is been used instead of cross correlation for more accurate differentiate

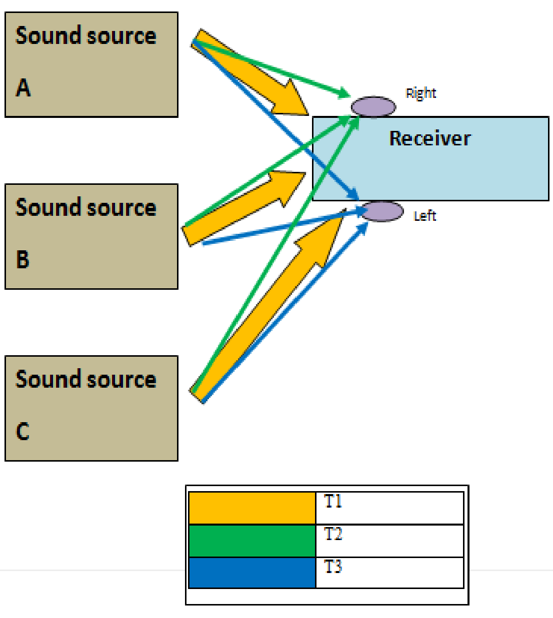

Figure 2 Propose techniques for identification

If T2 > T3 and T1 is same for both the sources means the source is in left side

If T3 > T2 and T1 is same for both the sources means the source is in right side

The localization device measure time on behalf of the average amplitude of the sound source at each 1 ms. Let say x(n) is sound signal receiver from any source placed at any location in that case average amplitude can be computed from below formula

$$A = \frac{\sum_{n=0}^{\frac{Fs}{1000}} x(n)}{Fs/1000}$$

The receiver will get the same amplitude information from many different sound sources it keeps compute the average amplitude in each 1 ms, and then it finds maximum A out of all receiver sound signals and Minimum A from all sound signals, the values of A gives the exact idea about the distance.

Maximum A = minimum distance

Minimum A = Maximum Distance

The block diagram below shows how to know the position of the sound source after computing the distance.

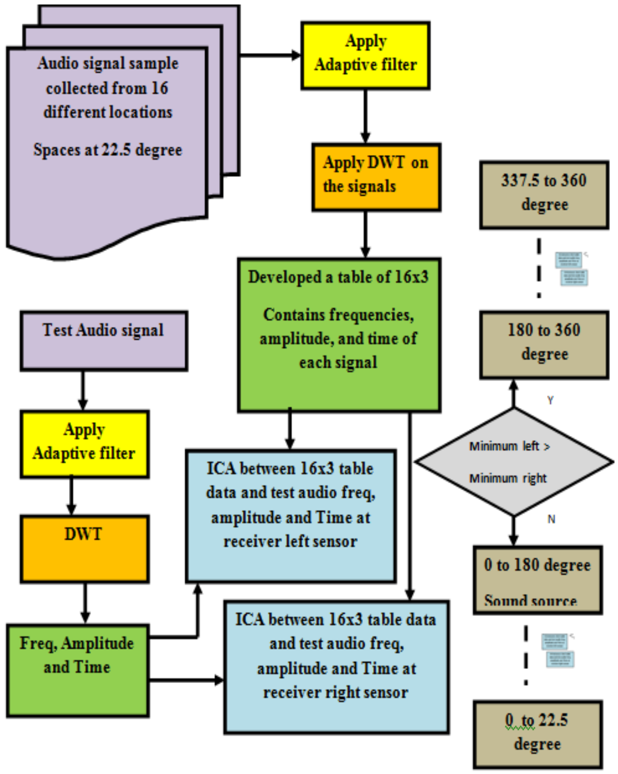

Figure 3 block diagram

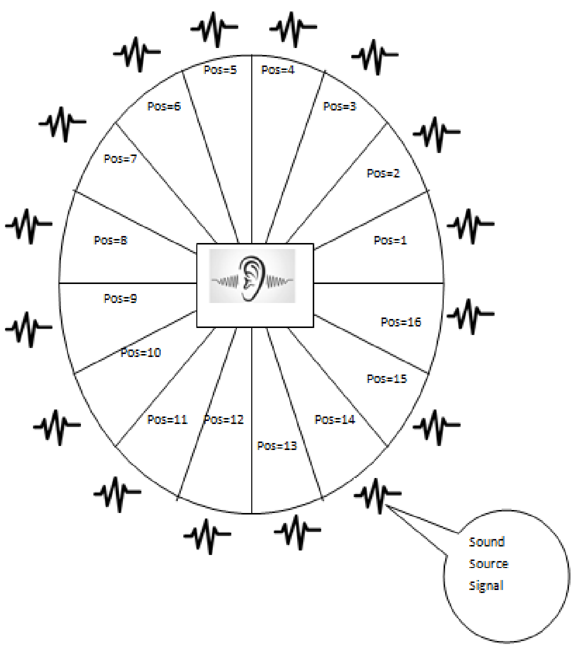

Figure 4: observe sound source angular precision

Proposed work It will give us the exact audio source angle with precision of 22.5 degree from where the test sound source came. Pos be can be anyone out of 1, 2, 3….16.

Pos	**Localization (Angle position of sourcefromthereceiver)**
1	0-22.5
2	22.5-45
3	45-67.5
4	67.5-90
5	90-112.5
6	112.5-135
7	135-157.5
8	157.5-180
9	180-202.5
10	202.5-225
11	225-247.5
12	247.5-270
13	270-292.5
14	292.5-315
15	315-337.5
16	337.5-360

Table 1 The sound source Localization accuracy

IV-RESULTS

Results of Source Localization: The source localization routine was tested on signals simulated for a geometry that consists of microphones placed at tips of a tetrahedral and a source that could be placed anywhere in 3D. The source localization routine turned out being robust with a condition number in order of magnitude of 1, for sources that are close to the center of the tetrahedral. This means a percent error introduced into the locations of the microphones gets amplified by a factor with order of magnitude of 1 before manifesting itself as a percent error in the location of the source determined.

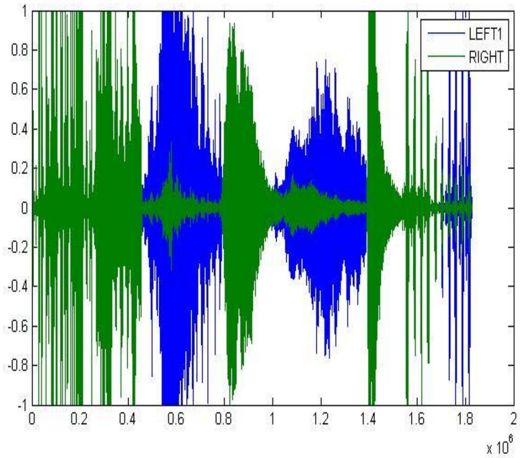

Figure 5 Audio signal having sound samples coming from 16 different angels

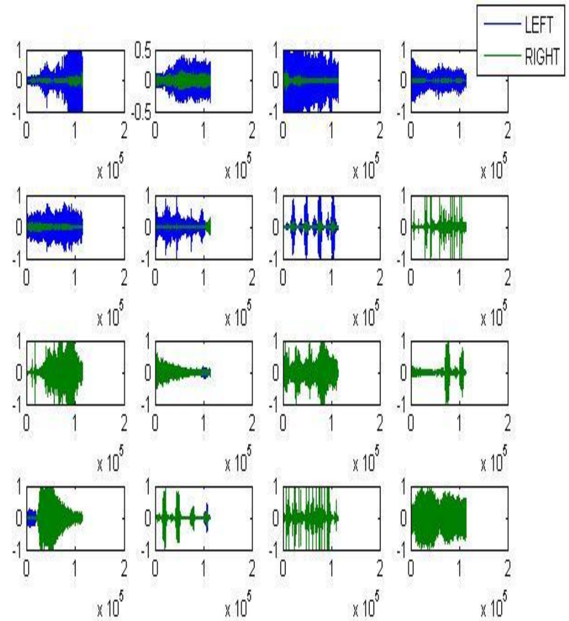

Figure 6: 16 Audio samples from 16 different locations spaced at 22.5 degree each

Proposed work is supporting 16 audio channels for 360 degree rotation
Hence precision of
360/16=22.5 degree

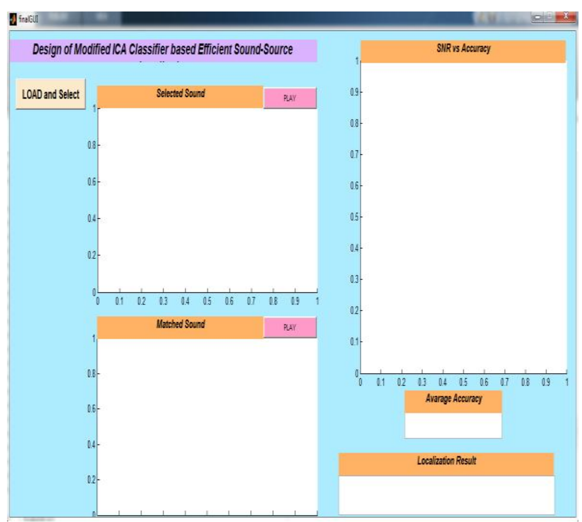

Figure 7: Proposed GUI

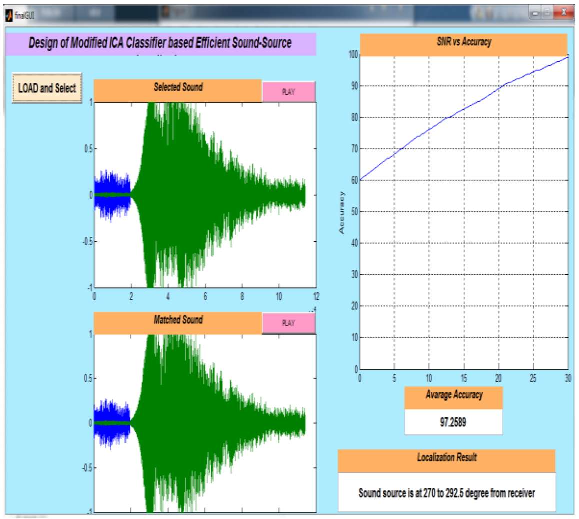

Figure 8 Results after sound localization in GUI

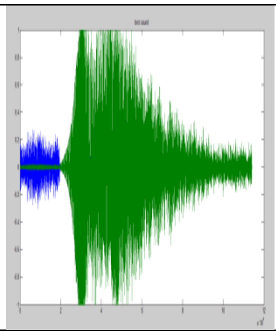	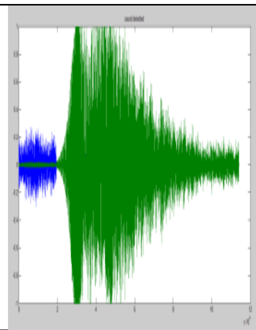
Test Sound source which need to detect	Sound signal detected by the proposed method

Figure 9 Sound Signal recognition

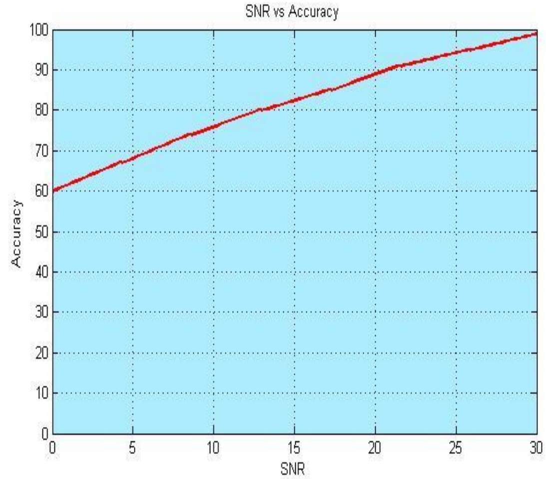

Figure 10: SNR vs Accuracy for proposed wok

V-CONCLUSION

This paper has put forward a sound source localization method based on ICA classifier. Using GCC-PHAT function with further processing as the feature vector, then compare the performance of ICA classifier and Naive-Bayes classifier. According to the simulation result, with the increasing of the reverberant time, the localization approach based on ICA classifier shows its higher localization performance than the Naive-Bayes classifier. Therefore, in such hash environment, it is better to use ICA classifier to locate sound source. In addition, in this simulation, we just use one frame test data and two microphones. So it is obvious that multiple microphones and multiple test frames to be used in practice will improve the performance greatly

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