Robust And Most Efficient Hearing Aid Using Microphone Array

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ABSTRACT

An estimated 63 million people suffer from significant hearing loss in India According to a study by the World Health Organization, five million Indian children were suffering from hearing and speech impairment in 2016. In India, “hearing handicapped” as defined by the Rehabilitation Council of India Act., 1992, is – hearing impairment of 70 dB and above, in better ear or total loss of hearing in both ears. This law is applicable to only those persons with severe hearing impairment whose hearing loss is 70 dB and above.Hearing aids have advanced significantly over the past decade, primarily due to the maturing of digital technology. The next decade should see an even greater number of innovations to hearing aid technology, and this article attempts to predict in which areas the new developments will occur. Both incremental and radical innovations in digital hearing aids will be driven by research advances in the following fields: (1) wireless technology, (2) digital chip technology, (3) hearing science, and (4) cognitive science. The automatic estimation of DOA is very important for many practical applications such as in Automatic Speech Recognition (ASR), Speaker Tracking, Teleconferencing, Human Computer Interaction (HCI) in particular and Human Machine Interface (HCI) in general, Robotic Audition or Active Audition, Blind Signal Separation (BSS) etc. For a human being it is very easy to infer position of active sound source due to a pair of spaced ear, but when it comes to design a system for the performance of similar task by a computer or a machine it becomes very difficult. The two ears, eyes in biological system like us represent a natural array of sensors. Following the same, concept of using array of sensors, in place of single sensors, developed in the engineering applications to exploit spatial diversities of the signal being captured. The poor performance of the conventional hearing aids in background noise motivated the use of microphone array to create directional sensitive hearing aids that amplify the signal arriving in a particular direction. Microphone array is used to improve desired speech signal when the interference arises from different directions. The microphone array is considered as a preprocessor, followed by conventional hearing aid processing [2]. It is used to improve the SNR value and speech intelligibility. Microphone array is used in various applications such as audio, teleconference, voice recognition applications.

Common Methods of Hearing Enhancement

Hearing aid manufacturers typically improve microphone directivity in one of two ways. The first strategy uses a single microphone with two openings (pressure-gradient microphones) designed to enhance sounds coming from a particular direction. The second strategy is to electronically combine the outputs of two separate microphones, to get the same effect as with the two openings. Each approach has advantages and disadvantages. In laboratory situations, hearing aids with these simple directional microphones improve directivity gains by factors of two to four (or 3-6 dB ("decibels")), greatly improving speech intelligibility in noisy surroundings. By combining the outputs of even more microphones, further improvements are in principle possible. However, practical issues have largely confined three-microphone systems to the laboratory. A standard multi-microphone system works well only when the microphones are exquisitely matched; even a few percent difference in the volume level from each microphone (such as naturally occurs due to changes in temperature, humidity, or accumulation of dirt or ear wax) can wipe out most of the directivity benefit. A pressure-gradient microphone shares the same hardware between the two openings, and is therefore always naturally matched. However, it is limited to two openings and it cannot be adjusted to the head's effects on the incoming sounds.

Proposed Algorithm

Research work on DOA estimation has also been done from the separation matrix learned by the BSS algorithms. ICA based sound signal separation is an unsupervised statistical learning process. The learned separation matrix does not form null direction in the exact direction of the sources to be jammed.

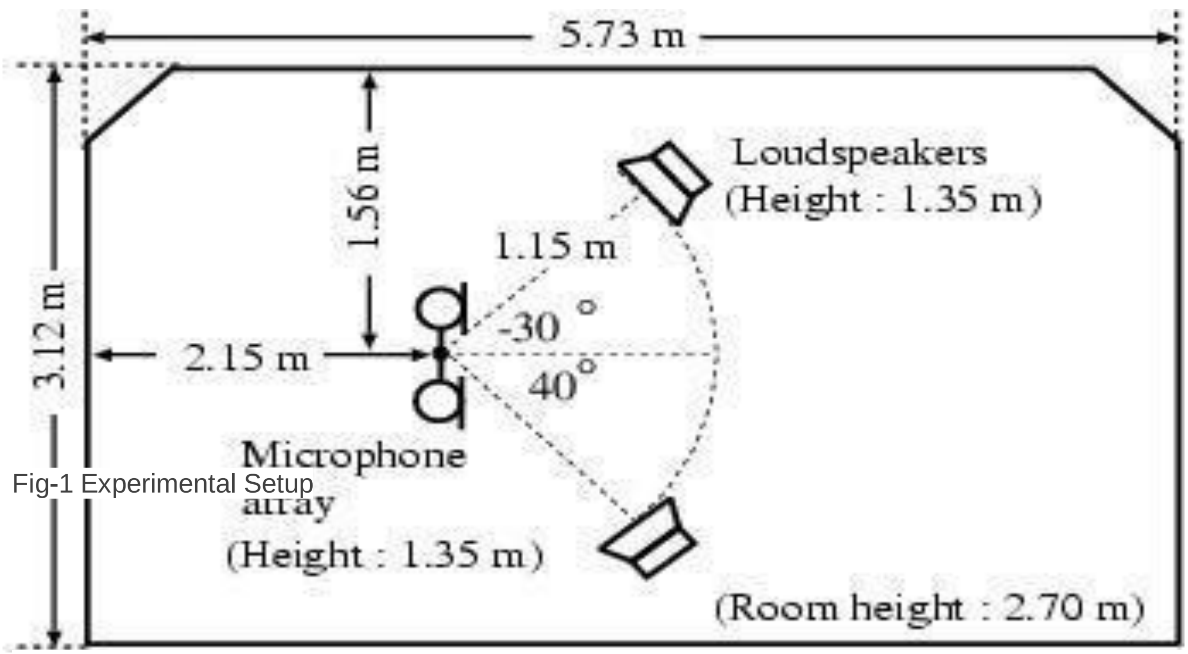

Fig-1 Experimental Setup

This trend becomes more prominent in the case of increasing noise and reverberation time. BSS algorithms reject the unwanted sound source by forming directional null, one can help BSS algorithms using DOA information to learn better sound separation system, however, this will be not purely blind setup but can be done under semi-blind BSS process. Thus under the semi-blind setup will try to develop new BSS algorithm by using DOA information. This algorithm will be of great functional use in mobile communication (smart antenna array processing), radar application. This will enable to design super efficient hearing aid devices and will prove to be boon for millions hearing impaired peoples.

BSS Algorithms for Speech Separation

The class of DOA algorithms based on use of BSS are recent development in the area of array processing BSS refers to estimation of hidden sources from the observed mixed signal at microphone of the array .The topic of BSS in the area of signal processing is itself result of development in the last two decades. In the BSS process nothing is known about the mixing process. The BSS algorithm is inputted with only mixed signal and produced outputs are individual signals. The application area of BSS is very wide due to which researches from almost areas such as such signal processing, image processing, machine vision, neuro-informatics, bioinformatics etc. have contributed [50] . The blind separation process of speech signal is shown in Figure 2 .

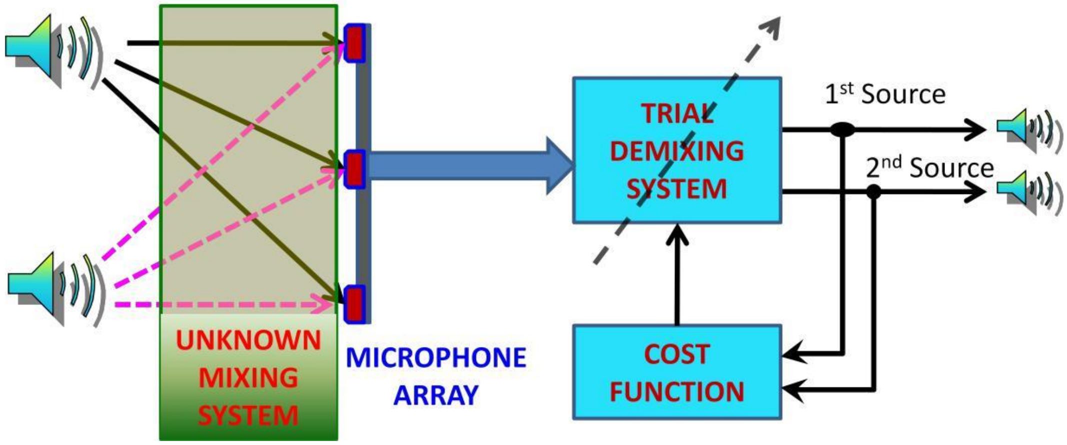

Figure 2. Functioning of ICA based BSS algorithm for speech signal separation.

In Figure 2 the blind separation of acoustic signal has been demonstrated. The microphone array, which is a ULA of 3-elements, captures mixture of speech signals from two speakers. The captured three channel mixed signals are fed into BSS algorithms which fitter out individual signal. In recovery of individual signal the mixing process is not utilized, as it is unknown, hence the process is known as blind.

The problem of BSS in general has been tackled from many angles and accordingly one can find many algorithms for general and specific purposes, developed using different principles ,Similarly for the speech signal separation too, many algorithms such as based on beamforming CASA, ICA etc. have been purposed. However, ICA based algorithms are better and numerous. ICA is a statistical technique which assumes each source of acoustic signal as independent and during mixing process independency of each such sound signal is lost. Therefore, any methods that can restore/ extract independent sources from mixed signal by restoring statistical independency can be termed as Independent Component Analysis in the loose sense.

Conclusion

Since the separation matrix obtained by ICA algorithms are permuted. The DONs so obtained in different frequency bins are also permuted and DONs are not exactly same for the same source in each frequency bin. Thus, here again some estimation technique is required to estimated DOA from collection of DONs

In the present research paper, only DP based methods have been considered. As stated above the DONs are permuted, hence, some methods are required to estimate true DOA from the set of DONs. Accordingly, there have been proposed many methods such as histogram and distance based techniques for DON classification. Modern digital hearing aids include sophisticated circuitry to avoid acoustic feedback (that occurs when the amplified sound is picked up by the microphone), provide compression (to reduce the gain at high amplitude to compensate for the reduced dynamic range of an impaired ear), and so on. Modern hearing aids are highly miniaturized, can talk to smart phones, include noise-reduction algorithms, and so on.

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