Immersive Audio Rendering Algorithms

1. Research Team

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2. Statement of Project Goals

Additive background noise and convolutive noise in the form of reverberation are two common types of noises in audio conferencing and hands-free telecommunication environments. In this project, we developed an efficient two-step adaptive noise reduction scheme to suppress these two types of noise. In particular, an Affine Projection Algorithm (APA) based Adaptive Noise Canceller is employed to reduce the additive noise in the first step, while a Constant Modulus Algorithm (CMA) based blind deconvolution technique is used to dereverberate the "wet" signal in the second step. To reduce the computational burden and accelerate the convergence speed, both causal and non-causal real-valued Delayless Subband Adaptive Filters (DSAF) were studied and integrated with the APA and CMA separately. The utilization of this real-valued Delayless SAF architecture can eliminate the substantial transmission delay existed in the conventional subband processing methods, while the real-valued subband signals produced by Single Sideband modulated filterbanks can make the implementation more efficient. Our results show that this method can achieve better performance with lower computational complexity and avoid transmission delay, thus it is very suitable for real time applications.

3. Project Role in Support of IMSC Strategic Plan

Immersive audio is one of the critical elements in IMSC's vision of Immersipresence. The algorithms that we are developing will enable multiple listeners to be immersed in a remote environment simultaneously. We are looking to go beyond the individual in front of their PC, but rather to focus on the group experience. Using our algorithms, people can be transported from the middle of a presidential press conference to a sports event or a concert, to a distant classroom exploring the human brain. All of these experiences require very high quality, noise-free signals for rendering. This method takes a significant step in that direction.

4. Discussion of Methodology Used

In this project, we have developed an efficient two-step adaptive noise reduction scheme to suppress the additive noise and convolutive noise successively. In the first step, a simple and effective Adaptive Noise Canceller (ANC) based on Affine Projection Algorithm (APA) [13] is employed to eliminate the additive noise. The resulting signal is then fed into the second phase –

the blind deconvolution processor. This processor tries to dereverberate the "wet" signal using the Higher Order Statistics (HOS) based blind deconvolution technique – Constant Modulus Algorithm (CMA) [14].[16].

5. Short Description of Achievements in Previous Years

Work in previous years focused on the development of a novel method for multiple listener room equalization:

- Development of fuzzy c-means clustering method for room acoustics problem
- Software implementation for room response measurement and filter design
- Best paper award in 2003

5a. Detail of Accomplishments During the Past Year

- Development of basic concepts behind the Adaptive Noise Canceller for additive noise
- Development of Constant Modulus Algorithm for convolutive noise
- Publication submitted to IEEE Transactions

6. Other Relevant Work Being Conducted and How this Project is Different

Existing methods for additive background noise reduction include Spectral Subtraction [1], Adaptive Noise Cancellation (ANC) [2], and Kalman Filtering [3]. Alternatively, Computational Auditory Scene Analysis (CASA) methods are also utilized to separate the distinct sound sources from the background noise [4][5]. Among the above approaches, ANC is one of the most popular additive noise reduction techniques due to its effectiveness and easy implementation. As for the solutions for convolutive noise suppression, previous work has concentrated on beam-forming techniques based on Microphone Array Processing [6], Complex-Cepstrum Based Dereverberation [7] and Inversion Methods by finding inverse filters for the room impulse response [8]. Recently, with the rapid development of blind deconvolution techniques [9][10], more research has been conducted in the application of blinding deconvolution techniques in audio signal dereverberation, which is also known as blind dereverberation [11][12].

The utilization of adaptive techniques in our approach makes it possible to track the real-time variations of the surrounding environment. Furthermore, our method requires significantly lower computational resources, thus making it appropriate for real-time implementation.

7. Plan for the Next Year

Improve the performance of convolutive noise suppression with further investigation of blind deconvolution techniques.

8. Expected Milestones and Deliverables

Software implementation of the basic algorithm.

9. Member Company Benefits

N/A

10. References

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