A Microphone-Array System for Speech Recognition Input

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Objective

This project is concerned with underlying mathematical algorithms, acoustics, hardware, and software to gain an understanding about, demonstrate the principles of, and, ultimately, to build an appropriate microphone-array system for speech-recognition input.

Approach

The approach taken might be called "recursive build-and-study". After investigating the layout problem and potential DSP architectures, we developed our first system. This allowed us to investigate real data and learn the real issues, begin to understand the difficult acoustics problems, and develop better DSP designs. This process is being repeated.

Recent Accomplishments

A new, nonlinear optimization algorithm called Stochastic Region Contraction (SRC), has been developed and has been applied to the microphone placement problem, talker location, and talker characterization. We have found that SRC is nearly two orders of magnitude faster than was simulated annealing. Our current research array system has been "hardened", and real-time, time-domain beamforming is operational.

Plans for the coming year

- Pefect one or more talker location algorithms and implement into real-time system
- Develop adjunct algorithm to work with locator to track a talker
- Demonstrate preliminary algorithms for talker characterization
- Complete a second research platform which can perform both tracking and beam forming in real time
- Develop accurate test methodology for understanding the acoustic properties of the sensor/room environment.