Speech Intelligibility Enhancement using Microphone Array via Intra-Vehicular Beamforming

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Problem Statement

According to the National Safety Council [1], there are approximately 1.6 million crashes each year due to distracted driving involving mobile phones. Drivers often hold their phone while making or taking a call which causes their eyes to leave the road. In an attempt to discourage the handheld use of mobile phones while driving, hands-free bluetooth calling connectivity has become the auto-industry standard. This hasn't entirely solved the problem, however.

The level of near-end speech intelligibility being sent is reduced due to multiple sources of noise. Some noises occur outside the car cabin such as engine noise, wind noise, conductive vibration, and road noise such as tires against pavement. Others occur inside the cabin including talking passengers, air conditioning, and music. Regardless of their source location, all of these noise sources and others combine to reduce the intelligibility of phone conversations. This causes frustration and often affects the driver's concentration. They simply pick up the cellphone and use it as normal.

Signal processing techniques are commonly used to improve signal quality and combat the noise. In audio signal processing applications, beamforming can be applied to selectively emphasize audio signals based on the direction-of-arrival (DOA) in the relationship to an array of microphones. Acoustic beamforming is a process by which multiple signals from a microphone array are filtered and combined in order to increase the amplitude of a target source's signal at a static DOA without increasing the amplitude of signals with differing DOAs.

This project aims to enhance speech intelligibility using microphone array via intra-vehicular beamforming. In particular, a uniform linear array (ULA) of microphones is used for data acquisition. The beamforming technique is used to combat near-end noise. The processed signal is sent to a far-end user over a hands-free bluetooth system with in increased near-end speech intelligibility.

Functional Requirements

The system functional requirements are described as follows.

- ❏ The system is tested and demonstrated in intra-vehicular or similar environment.
- ❏ The system includes a ULA microphone array. Each microphone is routed to a system (such as Matlab) for data acquisition.
- ❏ The beamforming is implemented in real-time.

Specifications of Requirements

1. Microphone Array

   An array of microphones is designed and constructed. To determine the number of microphones and the array dimensions, the general physical limitation of intra-vehicular environment, the processing ability of beamforming algorithm, as well as the computation power of the system are considered. The microphone array provides a sufficient spatial resolution for the beamforming algorithm. The array design is validated through simulation and testings.

2. Microphone Data Acquisition

   A data acquisition system is included to interface the microphone array. The system is able to acquire the signal from all microphones in real-time. The received data are processed by the beamforming algorithm. A predefined sine wave is used as the stimulus of microphones for the purpose of testing. Functional testing such as intra-vehicle are also important to testing the final performance.

3. Beamforming Algorithm

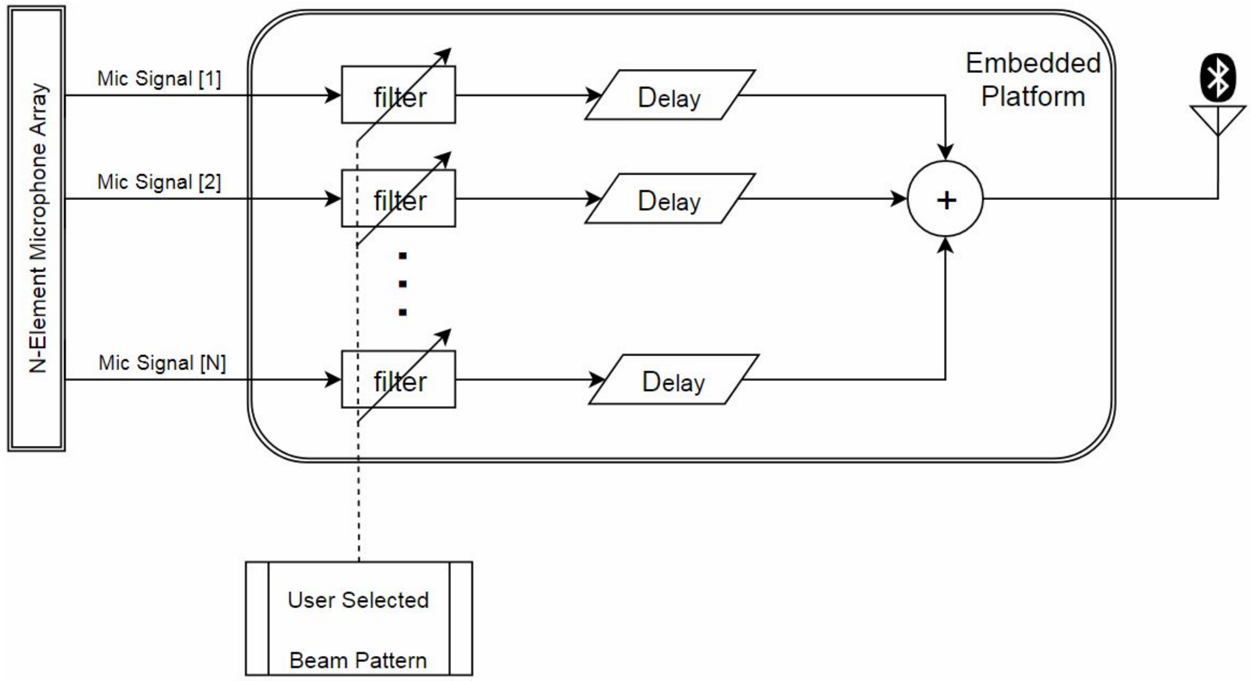
   A minimum variance distortionless response (MVDR) beamforming algorithm is investigated. It minimizes the variance/power of noise and interference at the output of beamformer while maintaining the output power of the microphone array while maintaining a unity gain at the desired response direction (i.e., the speaker). At the early stage of the project, it is developed and tested in MATLAB. The performance of beamforming algorithm is assessed. In general, beamforming demands high computation. For hardware implementation, delay-and-sum beamforming may be considered. It is a classic beamforming technique, useful because of its straightforward, simplified implementation. The performance of beamforming techniques are compared in terms of signal processing quality and computation cost.

4. Other Design Considerations

   The platform must have available I/O for multiple microphones as well as the processing power to run the algorithm. Matlab will be the system for its ability to connect to an audio interface. This allows for easy implementation of the algorithm.

   In order to further increase the performance of signal processing, echo cancellation could be considered to work together with beamforming. In particular, adaptive filtering can be used to reduce noise from engine, wind and road.

Functional Block Diagram



References

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[2] "Phased Array System Toolbox - mvdrweights" (R2017b). MathWorks.com. Accessed July 14, 2017. https://www.mathworks.com/help/phased/ref/mvdrweights.html