

Eigenvalue Decomposition based Acoustic Source Localization

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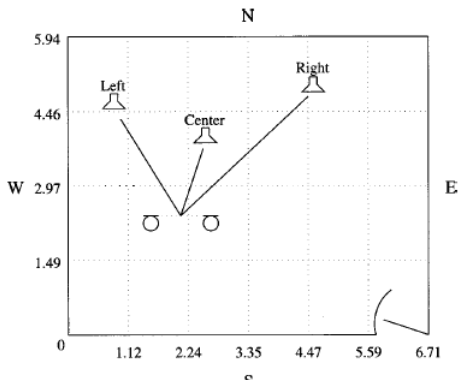
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Objective and Setup

- Objective: To determine the location of acoustic signal source, i.e. TDE(Time Delay Estimation) given the observations and positions of microphones.
- Setup: Number of microphones depends on the dimension of the intended localization.



Introduction

- Models

- ① Ideal Model

$$x_i(n) = \alpha_i s(n - \tau_i) + b_i(n)$$

- ② Real Model

$$x_i(n) = g_i * s(n) + b_i(n)$$

- GCC(Generalized Cross-Correlation)

Adaptive Algorithm

- Define \mathbf{u} as $[\mathbf{g}_2, -\mathbf{g}_1]^T$, and \mathbf{R} as covariance matrix of microphone signals, x_1 and x_2
- Minimize $\mathbf{u}^T \mathbf{R} \mathbf{u}$ to obtain minimum eigenvalue with corresponding eigenvector as \mathbf{u} using LMS algorithm,

$$\mathbf{u}(n+1) = \mathbf{u}(n) - \mu e(n) \nabla e(n)$$

where,

$$e(n) = \frac{\mathbf{u}^T(n) \mathbf{x}(n)}{\|\mathbf{u}(n)\|}$$

- Estimate Time Delay from eigen vector or impulse responses vector \mathbf{u}

Results

Results: Between signals captured simultaneously at two different microphones

Impulse response and Error plot: Between signals captured simultaneously at two different microphones

Negative peak in g_1 observed at index: 81 corresponding to
 $81/32000 \times 330 = 83.53\text{cm}$

Actual direct-path distance between microphones = 80cm

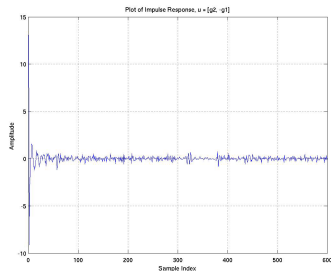


Figure: Impulse Response for a delay of 2.5ms between microphones.

Results

Results: Between signals captured simultaneously at two different microphones

Impulse response and Error plot (contd.)

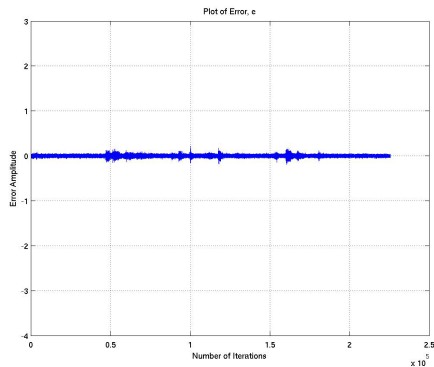


Figure: Error plot

Results

Results: Between signals captured simultaneously at two different microphones

Impulse response and Error plot (contd.)

Negative peak in g_1 observed at index: 161 corresponding to $161/32000 \times 330 = 166.03\text{cm}$

Actual direct-path distance between microphones = 160cm

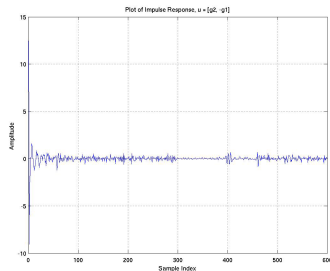


Figure: Impulse Response for a delay of 5ms between microphones.

Results

Results: Between signals captured simultaneously at two different microphones

Impulse response and Error plot (contd.)

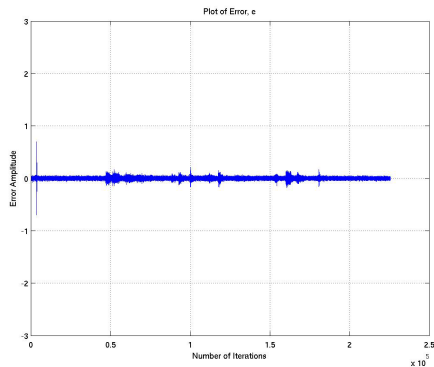


Figure: Error plot

Results

Results: Between signal captured at one microphone and its own delayed version

Impulse response and Error plot: Between signal captured at one microphone and its own delayed version

Negative peak in g_1 observed at sample index: 76
Shift provided between two signals = 75 samples

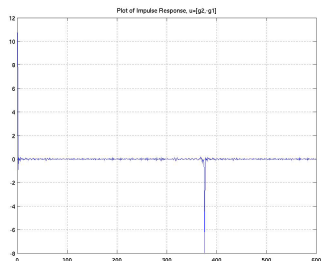


Figure: Impulse Response for a sample shift of 75 samples

Results

Results: Between signal captured at one microphone and its own delayed version

Impulse response and Error plot (contd.)

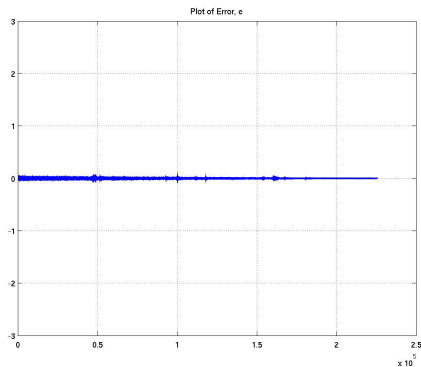


Figure: Error plot

Observations

Selection Criteria:

- Filter length: corresponds to the order of time delay between microphones
- μ : lesser values do not give dominant peaks
- Amplitude: position of peak does not change with change in amplitude of any of the signals

Direction of echoes (if observed) can also be resolved, based on time delay estimate for the delay in less-dominant peak in impulse response corresponding to echoes.

- ① J. Benesty, 'Adaptive Eigenvalue Decomposition Algorithm for Passive Acoustic Source Localization', Bell Labs Tech. Memo., 1998.
- ② C. H. Knapp and G. C. Carter, 'The generalized correlation method for estimation of time delay', IEEE Trans. Acoust., Speech, Signal Processing, vol. ASSP-24, no. 4, pp. 320-327, Aug. 1976.
- ③ Yiteng Huang, J. Benesty and Gary W. Elko, 'Adaptive Eigenvalue Decomposition Algorithm for Realtime Acoustic Source Localization System', Proceedings of the Acoustics, Speech and Signal Processing, IEEE International Conference, vol. 2, pp. 937-940, 1999.

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