Master thesis

Subspace system identification using a data driven model

Introduction

For system identification, loudspeaker and microphone signals are observed to obtain an estimate of the loudspeaker-to-microphone impulse responses. This is necessary for various applications including acoustic echo cancellation (AEC) and acoustic feedback cancellation (AFC). In many real-world scenarios this task becomes very challenging because of interference in the microphone signals, mathematical ambiguities when multiple loudspeakers are used, or a mismatched optimization criterion for the adaptive filter. Many approaches to mitigate these problems have been presented [1,2,3,4] that are related to altering the loudspeaker signals, spatial filtering, step-size control, or physically motivated adaptive filtering structures.

A lesser-known alternative to these approaches is given by data-driven methods that exploit a large number of previously measured impulse responses to improve system identification. While this can be achieved by learning the statistical properties of the impulse responses online [5], there are scenarios where offline approaches are more suitable [6]. This hold especially for scenarios where loudspeakers and microphones assume known and fixed positions. The goal of this project is to evaluate under which conditions such an approach can be applied in a real-world scenario.

Project Description

In the proposed project, the approach presented in [6] shall first be evaluated in a given (AEC) scenario with provided signals and impulse responses. While this single-channel scenario is rather straightforward, the robustness of this approach can be challenged by increasing the interference in the microphone signals. In a second step, the performance of two approximations of the recursive least squares (RLS) algorithm used in [6] is to be evaluated in the same scenario. These approximations are closely related to the derivation presented in [7] and fully described in a document provided. Depending on the initial results, one or more of the following scenarios can be investigated: a multichannel (AEC) scenario, where nonuniqueness occurs, or a single-channel (AFC) scenario.

The algorithm descriptions, evaluation criteria, data, test scenarios, and control flow suggestions are provided. The implementation of algorithms and test framework is expected, where the code should be documented. The preferred programming language is Python, although Matlab is also accepted. The thesis should contain a comprehensive report on the results considering the given scenarios and evaluation criteria. Applicants should be fluent in the chosen programming language and have some background in audio signal processing. Knowledge of adaptive filters is beneficial but not required.

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References

[1] M. Valero and E. A. Habets, "Insight into a phase modulation technique for signal decorrelation in multi-channel acoustic echo cancellation," in Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP), Shanghai, China, Mar. 2016, pp. 519–523.

[2] W. Kellermann, "Strategies for combining acoustic echo cancellation and adaptive beamforming microphone arrays," in Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP), vol. 1, Munich, Germany, Apr. 1997, pp. 219–222.

[3] A. Mader, H. Puder, and G. Schmidt, "Step-size control for acoustic echo cancellation filters–An overview," Signal Process., vol. 80, no. 9, pp. 1697 – 1719, Sep. 2000.

[4] M. Schneider and W. Kellermann, "Multichannel acoustic echo cancellation in the wave domain with increased robustness to nonuniqueness problem," IEEE/ACM Trans. Audio, Speech, Language Process., vol. 24, no. 3, pp. 518–529, Mar. 2016.

[5] M. Fozunbal, T. Kalker, and R. Schafer, "Multi-channel echo control by model learning," in Proc. Int. Workshop on Acoustic Echo and Noise Control (IWAENC), Seattle, WA, USA, Sep. 2008, pp. 1–4.

[6] T. Koren, R. Talmon, and I. Cohen, "Supervised system identification based on local PCA models," in Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP), Kyoto, Japan, Mar. 2012, pp. 541–544.

[7] M. Schneider and W. Kellermann, "The generalized frequency-domain adaptive filtering algorithm as an approximation of the block recursive least-squares algorithm," EURASIP J. on Advances in Signal Process., vol. 2016, no. 1, pp. 1–15, Jan. 2016.