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Book of Abstracts

Computing an FIR controller using an efficient iterative scheme

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1 Introduction

Acoustic anechoic chambers are used to conduct measurements and experiments under free-field conditions. In the ideal case, all acoustic waves are absorbed at the walls, such that no reflections exist. In reality low frequency sound will be reflected by the walls, due to limitations of passive absorption measures [1]. Active noise control is effective at lower frequencies, which makes this a promising addition to the passive wall absorption. Because of the typical large dimensions of the chamber, a large number of sources and sensors is required for satisfactory performance. The computation of a controller for such a large-scale system is computationally expensive. This work shows a scheme to efficiently obtain a finite impulse response (FIR) controller, demonstrated by a single-channel simulation in a duct.

2 Approach

Following the approach in [2], both a preconditioner/decoupler of the secondary path ($\hat{G}_{mi}(z)$ and $\hat{G}_{ai}(z)$) and a prewhitening/decorrelation filter ($\hat{F}_w(z)$) are derived via the frequency-domain. The filters are delayed to ensure causality. The delay to be added is found by minimizing errors due to wrap-around and truncation effects. The adaptive part of the algorithm used in the simulation incorporates normalization of the input by its power.

3 Simulation

A duct with a single-channel control setup is used to simulate the control performance. The setup is schematically shown in Fig. 1, where $d_1 = 0.8575m$, $d_2 = 0.343m$, $d_3 = 0.1715m$, $d_4 = 1.3720m$ and $R_0 = 0.90$, $R_L = -0.95$ are the reflection coefficients at $x = 0$ and $x = L$ respectively.

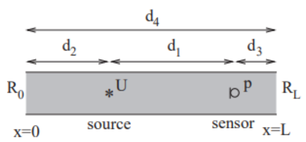
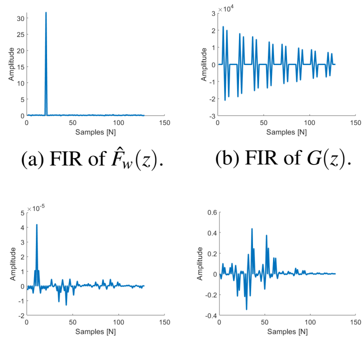


Figure 1: Simulation setup: single-channel control in a duct.

The FIR filters corresponding to the decomposition are shown in Fig. 2. All FIR filters are truncated after $J = 128$ samples. A delay of 20, 10 and 50 samples is added to $\hat{F}_w(z)$, $\hat{G}_{\mathrm{mi}}(z)$ and $\hat{G}_{\mathrm{ai}}(z)$ respectively. A normalized tolerance of 0.0137 is used to truncate 68 of 513 singular values to compute $\hat{G}_{\mathrm{mi}}(z)$.



(a) FIR of $\hat{F}_w(z)$. (b) FIR of $G(z)$.

(c) FIR of $\hat{G}_{mi}(z)$. (d) FIR of $\hat{G}_{ai}(z)$.

Figure 2: FIR filters used in the iterative algorithm.

The algorithm is compared to a normalized filtered-error LMS algorithm. The results are shown in Fig. 3, from which can be concluded that the preconditioned algorithm has improved convergence speed.

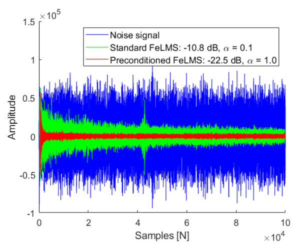


Figure 3: Convergence of the algorithms.

4 Acknowledgements

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