ABSTRACT

Control of loudspeaker arrays have been studied in the past few years for a wide variety of applications. One of such application is to control a pre-defined listening environment to achieve audible and inaudible zones so as to gain a personalized and immersive sound experience. One existing feasible approach to achieve soundfield control is to compute time- or frequency-domain coefficients for each loudspeaker such that the generated sound signals at the pre-defined listening locations can achieve the desired sound energy level for the superimposed sound signals. Several methodologies have recently been proposed to achieve a desired soundfield within a listening zone of interest. These methods include maximizing the energy ratio between energies of audible and inaudible zones, maximizing energy difference between two zones and maximizing the energy ratio while maintaining a desired sound energy level in the zone of interest. Although the above methods provide directional sound to the zone of interest, they cannot guarantee precise control of the soundfield within the audible/inaudible regions. As a result, undesired fluctuations of the soundfield or even sound distortion may occur. To address the issue, least squares (LS) methods along with regularisation techniques such as ordinary cross validation (OCV), generalized cross validation (GCV), the L-curve and the truncation method have also been proposed to achieve a desired sound energy level in a pre-defined zone of interest. Although these LS methods can improve the performance of soundfield control system, direct application of these methods will however introduce regularisation error (RE).

In this thesis an iterative LS based algorithm is proposed. This algorithm utilizes the round-off error (ROE) as a regularisation parameter for the ill-conditioned system matrix over the zone of interest. Unlike existing LS methods, the proposed approach estimates the system matrix inverse iteratively and identifies the best estimate of the matrix inverse. In addition, the proposed algorithm does not accumulate significant ROE when the system matrix is ill-conditioned. Instead, the accumulated ROE is taken as a regularisation parameter which is evaluated iteratively to achieve a good trade-off between matrix inverse accuracy and the amount of RE. To achieve the above, a matrix sparseness criterion is proposed to evaluate the accuracy of the inverse estimate. This measure is subsequently used to terminate the iteration process. Furthermore, incorporating this criterion to the proposed method of estimating the loudspeaker coefficients, one can take into account the trade-off between the estimation error (during the convergence of the matrix inverse estimate) and the ROE (caused by mathematical computation when the system is ill-condition). The proposed algorithm is further extended to the scenario when the misplacement error (MPE) is present in the sound control system. It is found that the proposed algorithm can compensate and minimize the effect of MPE on the generated soundfield by exploiting the characteristics of the MPE when computing the regularisation parameter. Simulation results show that the proposed algorithm can outperform other LS-based regularisation methods in terms of normalized output error (NOE) in the presence or absence of MPE.