MODELS FOR BLIND SPEECH DEREVERBERATION: A SUBBAND ALL−POLE FILTERED BLOCK STATIONARY AUTOREGRESSIVE PROCESS (WedAmOR6)

Author(s) : James Hopgood (The University of Edinburgh, United Kingdom)

Abstract : Single channel blind dereverberation of speech acquired in an acoustic environment is approached using parametric modelling and estimation theory to obtain channel estimates; inverse filtering is then applied to derevererate the speech. Models previously used in such an approach deal with relatively simple scenarios such as a gramophone horn modelled with 70 parameters; the weakness of those models, however, is in their attempt to simultaneously model the full channel spectrum by a single all−pole filter. Not only does this lead to a large computational load, it is not parsimonious, nor is it scalable such that the algorithm can be applied to higher dimensional channels. A better approach uses subbands; in this paper, a subband all−pole filter models the channel while the source is still represented by a single−band block stationary AR process. An example is given of blindly dereverberating a signal observed through the aforementioned gramophone horn, demonstrating an equally robust, but more flexible and scalable model.

Menu