Abstract

Currently, about 5% of people around the world suffer from one or another form of hearing loss. Hearing aids have emerged as the most common intervention to compensate for hearing loss. A basic digital hearing aid consists of one or more microphones, a loudspeaker, digital signal processing unit and a battery. In addition to amplifying the sounds based on the hearing profile of the user, the signal processing unit implements several algorithms including ones for feedback cancellation, noise reduction and dynamic range compression. The close proximity of the loudspeaker and the microphone(s) in a hearing aid leads to acoustic feedback, which deteriorates the quality of the processed sound produced by the hearing aid. The acoustic feedback path in a hearing aid usually has a sparse impulse response. In order to take advantage of this property of the acoustic path, this thesis proposes a class of feedback cancellation schemes which are sparse aware. The signal processing algorithms must adapt quickly to the changing environment of the hearing aid user to provide enhanced usability and hearing comfort. With this objective in mind, this thesis also proposes a set of adaptive algorithms to improve convergence and tracking performance. To further enhance the quality of feedback cancellation, a two-microphone feedback cancellation scheme based on a delayless multiband-structured subband adaptive filter has also been developed. Further, to reduce the deterioration of signal to noise ratio caused by leakage of unprocessed noise into the ear canal in an open fitting hearing aid, a low complexity integrated noise reduction and active noise cancellation algorithm has been proposed. Also, a set of nonlinear adaptive filtering schemes have been developed to deal with nonlinearities that may arise in a hearing aid.