應用適應性濾波演算法則於助聽器之聲學回授效應消除

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摘 要

助聽器於改善聽力上是非常有用且重要的裝置。本研究提出應用適應性濾波演算法則，於助聽器語音 接收訊號系統品質的改善。近年來，助聽器的收音與接收器因為要小型化，往往與耳道之間不夠密合 就會有聲學回授的問題，造成配戴者極大的困擾。科技的進步，為增加使用者的便利性、舒適性，消 除所產生聲學回授音與噪音干擾為主要的課題。數位式助聽器的訊號濾波處理器，可消除聲學回授效 應，將此回授音波峰濾除又不會影響其放大的功能。但有較於傳統的適應性濾波器有收斂速度較慢、 強健性差和運算量大的缺點，使得聲學回授消除性能不佳，影響語音接收品質。為解決這些問題，本研究提出一種包含消除回授音及適應性雜訊消的架構：適應性聲學回授音消除可消除在耳內所產生的 聲學回授音，雜訊消除可消除單頻噪音，線性增強可降低寬頻噪音。因此，本研究提出2種改善方案。 方案一:企圖使用卡耳曼濾波器來改善適應性濾波器的接收品質，希望能有效改善聲學回授音的消除。 為了驗證使用的卡耳曼濾波器性能，測試時利用不同的模擬噪音源來與傳統的LMS、NLMS和RLS三種不 同適應性濾波器作比較，由模擬後結果及特性的分析了解卡耳曼濾波器在音響回饋消除上有較佳性能。 方案二:企圖使用方向性麥克風,因其有不同角度接收特性，來減少回授音的接收強度，進而改善助聽 器的聲音品質。

關鍵詞 : 聲學回授音，適應性濾波器，助聽器，Kalman濾波器， 方向性麥克風

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