車輛免持聽筒行動通訊系統音質改善

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摘 要

本研究提出一種新的適應性整合架構與演算法則，應用在車輛免持聽筒通訊系統通話品質的改善。 由於科技的進步使得個人通訊系統隨時隨地都可與人溝通，免持通訊系統被廣泛的應用於汽車行動 電話上，增加使用上的便利性、舒適性及安全性，但卻產生聲學回音與噪音干擾的問題。噪音干擾 可分為兩種，一種是窄頻的引擎噪音，另一種為寬頻噪音，如輪胎和風切產生的噪音。為解決這些 問題，提出一種包含適應性雜訊消除、線性增強和回音消除系統的架構：適應性回音消除可消除車 廂內聲學的回音，雜訊消除可消除單頻噪音，線性增強可降低寬頻噪音。為了提高性能，改良傳統 的演算法而提出可變收斂因子仿射投影演算法(VSS APA)。此演算法結合仿射投影演算法(APA)和可 變收斂因子最小均方演算法(NVSS LMS)的優點，因為仿射投影演算法輸入訊號為矩陣，比可變收斂 因子最小均方演算法輸入為向量的資料更完整，使預測的權重向量更準確又快速。但由於仿射投影 演算法的收斂因子為固定的，本研究利用發展完整的可變收斂因子最小均方演算法來調整收斂因子 ，使收斂因子可根據系統得到最理想的值。為了驗證本研究所提出之可變收斂因子仿射投影演算法的效果，則將可變收斂因子仿射投影演算法與傳統演算法模擬比較，結果顯示可變收斂因子仿射投 影演算法性能較優越並且適合用於本研究的系統中。

關鍵詞 : 回音消除，適應性濾波器，免持通訊系統，數位信號處理器，可變收斂因子仿射投影演算法

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