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短文

基于微分麦克风阵列的自适应语音增强算法研究及DSP实现

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摘要
自适应滤波是语音增强算法中的常用技术, 而算法复杂度与收敛速度是设计各种自适应算法需要首要考虑的问题. 本文提出一种用于片上的语音增强自适应滤波新算法. 该算法分两步实现, 首先, 利用一阶微分麦克风阵列, 获得噪声的实时估计; 其次, 对传统的仿射投影算法(Affine projection algorithm, APA)加以改进, 得到计算误差向量的快速算法, 并根据估计误差动态调整搜索步长以及仿射投影维数, 对带噪语音进行自适应滤波消噪. 在TMS320VC5509 DSP芯片上实现该算法. 实验表明, 算法的自适应滤波过程具有接近递推最小二乘算法(Recursive least squares, RLS)的快速收敛速度, 以及类似最小均方误差算法(Least mean squares, LMS)的低算法复杂度.

关键词 语音增强 微分麦克风阵列 仿射投影算法 自适应滤波 噪声消除

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Research on Adaptive Speech Enhancement Algorithm Based on Differential Microphone Array and Its Implementation with DSP

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Abstract
Adaptive filtering is in common use in most speech enhancement algorithms, while complexity and convergence speed should be considered first when an adaptive filtering method is designed. This paper presents a speech enhancement adaptive filtering method implemented on chip. This method is implemented in two steps. In the first step, first-order differential microphone array is utilized to obtain real-time noise estimation. In the second step, the traditional affine projection algorithm (APA) is modified so that a fast error vector calculation method is obtained, and the search step and projection dimension can be altered dynamically according to the estimation error. Thus, noise can be removed adaptively. This method is realized on a TMS320VC5509 DSP chip. Experiments show that the proposed method has fast convergence speed like RLS (Recursive least squares) and low computational complexity like LMS (Least mean squares).

Key words Speech enhancement differential microphone array affine projection algorithm (APA) adaptive filter noise reduction

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