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论文与报告

基于多统计模型和人耳听觉特性的麦克风阵列后滤波语音增强算法

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摘要
针对麦克风阵列后滤波语音增强算法的不足, 结合人耳的听觉掩蔽效应, 提出了改进的后滤波语音增强算法. 提出了最大化目标语音存在概率来确定信号子空间维度的方法, 在噪声子空间上, 利用条件概率估计出噪声功率谱. 基于人耳的听觉掩蔽效应, 提出了后滤波器的一种合理的设计方法. 实验证明, 所提的噪声估计方法比传统方法更加准确, 所提的后滤波算法比传统的后滤波算法更好, 在多项语音评价指标上, 都取得了更好的实验效果.

关键词 麦克风阵列 基于听觉特性的后滤波器 语音增强 多统计模型

分类号

Microphone Array Post-filter Based on Multi-statistical Models and Perceptual Properties of Human Ears

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Abstract
To overcome the drawbacks of the conventional microphone array post-filter speech enhancement method, some improvements are proposed using the masking properties of human ears. A subspace selection method is proposed by maximizing the present probability of the target speech. In the noise subspace, the conditional probability is used to estimate the noise power spectrum. A novel post-filter is proposed based on the masking properties of human ears. Experiments prove that the proposed noise estimation method and post-filter are much better than the conventional ones. The proposed speech enhancement technique has shown to produce impressive results in terms of quality measures of the enhanced speech.

Key words Microphone array auditory properties based post-filter speech enhancement multi-statistic models

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