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<<数字语音处理理论与应用>>

图书基本信息

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内容概要

本书是作者继1978年版经典教材digital processing of speech signals之后的又一著作，本书除有简练精辟的基础知识介绍外，系统介绍了近30年来语音信号处理的新理论、新方法和在应用上的新进展。本书共14章，分四部分：第一部分介绍语音信号处理基础知识，主要包括数字信号处理基础、语音产生机理、(人的)听觉和听感知机理和声道中的声传播原理；第二部分介绍语音信号的时频域表示和分析；第三部分介绍语音参数估计算法；第四部分介绍语音信号处理的应用，主要包括语音编码、语音和音频信号的频域编?、语音合成、语音识别和自然语言理解。

本书可供高等院校通信、电子、信息、计算机等专业作为研究生和本科生教材，也可以供有关科研和工程技术人员参考，是一本既有系统的基础理论讲解、又有最新研究前沿介绍并密切结合应用发展的教材。

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Lawrence R. Rabiner，美国工程院和美国科学院院士，美国声学学会、IEEE、Bell实验室、AT&T会士，以及Eta Kappa Nu、Sigma Xi、Tau Beta Pi等荣誉学会会员。
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其主要研究方向包括：通信、控制与信号处理、数字信号处理、数字语音处理、多媒体通信、多模态处理等。
Rabiner教授于2002年从AT&T退休，随后担任Rutgers大学和加州大学圣巴巴拉分校的教授，以及Rutgers大学先进信息处理中心副主任。

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