DSP FOR IN-VEHICLE AND MOBILE SYSTEMS

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Dedication

To Professor Fumitada Itakura

This book, “DSP for In-Vehicle and Mobile Systems”, contains a collection of research papers authored by prominent specialists in the field. It is dedicated to Professor Fumitada Itakura of Nagoya University. It is offered as a tribute to his sustained leadership in Digital Signal Processing during a professional career that spans both industry and academe. In many cases, the work reported in this volume has directly built upon or been influenced by the innovative genius of Professor Itakura.

While this outstanding book is a major contribution to our scientific literature, it represents but a small chapter in the anthology of technical contributions made by Professor Itakura. His purview has been broad. But always at the center has been digital signal theory, computational techniques, and human communication. In his early work, as a research scientist at the NTT Corporation, Itakura brought new thinking to bit-rate compression of speech signals. In partnership with Dr. S. Saito, he galvanized the attendees of the 1968 International Congress on Acoustics in Tokyo with his presentation of the Maximum Likelihood Method applied to analysis-synthesis telephony. The presentation included demonstration of speech transmission at 5400 bits/sec with quality higher than heretofore achieved. His concept of an all-pole recursive digital filter whose coefficients are constantly adapted to predict and match the short-time power spectrum of the speech signal caused many colleagues to hurry back to their labs and explore this new direction. From Itakura’s stimulation flowed much new research that led to significant advances in linear prediction, the application of autocorrelation, and eventually useful links between cepstral coefficients and linear prediction. Itakura was active all along this route, contributing among other ideas, new knowledge about the Line Spectral Pair (LSP) as a robust means for encoding predictor coefficients. A valuable by-product of his notion of adaptively matching the power spectrum with an all-pole digital filter gave rise to the Itakura-Saito distance measure, later employed in speech recognition as well as a criterion for low-bit-rate coding, and also used extensively in evaluating speech enhancement algorithms.

Itakura’s originality did not escape notice at Bell labs. After protracted legalities, a corporate arrangement was made for sustained exchange of research scientists between ATT and NTT. Fumitada Itakura was the first to initiate the program, which later encompassed such notables as Sadaoki Furui, Yoh`ichi Tohkura, Steve Levenson, David Roe, and subsequent others from

both organizations. At Bell Labs during 1974 and -75, Fumitada ventured into automatic speech recognition, implementing an airline reservation system on an early laboratory computer. Upon his return to his home company Dr. Itakura was given new responsibilities in research management, and his personal reputation attracted exceptional engineering talent to his vibrant organization.

Following fifteen years of service with NTT, the challenges of academe beckoned, and Dr. Itakura was appointed Professor of Electrical Engineering in Nagoya University – the university which originally awarded his PhD degree. Since this time he has led research and education in Electrical Engineering, and Acoustic Signal Processing, all the while building upon his expertise in communications and computing. Sophisticated microphone systems to combat noise and reverberation were logical research targets, as exemplified by his paper with colleagues presented in this volume. And, he has continued management responsibilities in contributing to the leadership of the Nagoya University Center for Integrated Acoustic Information Research (CIAIR).

Throughout his professional career Professor Itakura has steadily garnered major recognition and technical awards, both national and international. But perhaps none rivals the gratification brought by the recognition bestowed by his own country in 2003 -- when in formal ceremony at the Imperial Palace, with his wife Nobuko in attendance, Professor Itakura was awarded the coveted Shiju-hosko Prize, also known as the Purple Ribbon Medal.

To his stellar record of career-long achievement we now add the dedication of this modest technical volume. Its pages are few by comparison to his accomplishments, but the book amply reflects the enormous regard in which Professor Fumitada Itakura is held by his colleagues around the world.

Jim Flanagan
Rutgers University

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Preface

Over the past thirty years, much progress has been made in the field of automatic speech recognition (ASR). Research has progressed from basic recognition tasks involving digit strings in clean environments to more demanding and complex tasks involving large vocabulary continuous speech recognition. Yet, limits exist in the ability of these speech recognition systems to perform in real-world settings. Factors such as environmental noise, changes in acoustic or microphone conditions, variation in speaker and speaking style all significantly impact speech recognition performance for today systems. Yet, while speech recognition algorithm development has progressed, so has the need to transition these working platforms to real-world applications. It is expected that ASR will dominate the human-computer interface for the next generation in ubiquitous computing and information access. Mobile devices such as PDAs and cellular telephones are rapidly morphing into handheld communicators that provide universal access to information sources on the web, as well as supporting voice, image, and video communications. Voice and information portals on the WWW are rapidly expanding, and the need to provide user access to larger amounts of audio, speech, text, and image information is ever expanding. The vehicle represents one significant emerging domain where information access and integration is rapidly advancing. This textbook is focused on digital signal processing strategies for improving information access, command and control, and communications for in-vehicle environments. It is expected that the next generation of human-to-vehicle interfaces will incorporate speech, video/image, and wireless communication modalities to provide more efficient and safe operations within car environments. It is also expected that vehicles will become “smart” and provide a level of wireless information sharing of resources regarding road, weather, traffic, and other information that drivers may need immediately or request at a later time while driving on the road. It is also important to note that while human interface technology continues to evolve and expand, the demands placed on the vehicle operator must also be kept in mind to minimize task demands and increase safety.

The motivation for this textbook evolved from many high quality papers that were presented at the DSP in Mobile and Vehicular Systems Workshop, Nagoya, Japan, April 2003, with generous support from CIAIR, Nagoya University. From that workshop, a number of presentations were selected to be expanded for this textbook. The format of the textbook is centered about three themes: (i) in-vehicle corpora, (ii) speech recognition/dialog systems with emphasis on car environments, and (iii) DSP for mobile platforms

involving noise suppression, image/video processing, and alternative communication scenarios that can be employed for in-vehicle applications.

The textbook begins with a discussion of speech corpora and systems for in-vehicle applications. Chapter 1 discusses a multiple level audio/video/data corpus for in-car dialog applications. Chapter 2 presents the CU-Move in-vehicle corpus, and an overview of the CU-Move in-vehicle system that includes microphone array processing, environmental sniffing, speech features and robust recognition, and route dialog navigation information server. Chapter 3 also focuses on corpus development, with a study on dialog management involving traffic, tourist, and restaurant information. Chapter 4 considers in-vehicle dialog scenario where more than one user is involved in the dialog task. Chapter 5 considers distributed task management for car telematics with emphasis on VoiceXML. Chapter 6 develops an in-vehicle voice interaction systems for driver assistance with experiments on language modeling for streets, hotels, and cities. Chapter 7 concentrates more on high speech error corrective coding for mobile phone applications which are of interest for car information access. Chapter 8 considers a speech enhancement method for noise suppression in the car environment. Chapter 9 seeks to integrate prosodic structure into noisy speech recognition applications. Effective noise reduction strategies for mobile and vehicle applications are considered in Chapter 10, and also in Chapter 11. Chapter 12 considers a small vocabulary speech system for controlling car environments. Chapters 13 and 14 consider transmission and compression schemes respectively for image and video applications which will become more critical for wireless information access within car environments in the near future. Chapter 15 follows up with a work on adaptive techniques for wireless speech transmission in local area networks, an area which will be critical if vehicles are to share information regarding road and weather conditions while on the road. Chapter 16 considers the use of audio-video information processing to help identify a speaker. This will have useful applications for driver identification in high noise conditions for the car. Chapter 17 considers a rather interesting idea of characterizing driving behavior based on biometric information including gas and brake pedal usage in the car. Chapter 18 addresses convolutional noise using blind signal separation for in-car environments. Finally, Chapter 19 develops a novel approach using multiple regression of the log spectra to model the differences between a close talking microphone and far-field microphone for in-vehicle applications.

Collectively, the research advances presented in these chapters offers a unique perspective of the state of the art for in-vehicle systems. The treatment of corpora, dialog system development, environmental noise suppression, hands-free microphone and array processing, integration of audio-video technologies, and wireless communications all point to the rapidly advancing

field. From these studies, and others in the field from laboratories who were not able to participate in the DSP in Mobile and Vehicular Systems Workshop [<http://dspincars.sdsu.edu/>] in April 2003, it is clear that the domain of in-vehicle speech systems and information access is a rapidly advancing field with significant opportunities for advancement.

In closing, we would like to acknowledge the generous support from CIAIR for the DSP in Mobile and Vehicular Systems Workshop, and especially Professor Fumitada Itakura, who's vision and collaborative style in the field of speech processing has served as an example of how to bring together leading researchers in the field to share their ideas and work together for solutions to solve problems for in-vehicle speech and information systems.

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