Tutorial

Session: Tutorial 2
Time: Wednesday December 16, 09:00-12:20
Place: Room Y302
Speakers: Woon-Seng Gan, Digital Signal Processing Lab, School of EEE, Nanyang Technological University, (Singapore)
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Assisted Listening for headphones and hearing aids: Signal Processing Techniques

Abstract
With the strong growth of the mobile devices and emerging virtual reality (VR) and augmented reality (AR) applications, headsets are becoming more and more referable in personal listening due to its convenience and portability. Assistive listening (AL) devices like hearing aids have seen much advancement. Creating a natural and authentic listening experience is the common objective of these VR, AR, and AL applications. In this tutorial, we will present state-of-the-art audio and acoustic signal processing techniques to enhance the sound reproduction in headsets and hearing aids. This tutorial starts with an introduction of the recent examples of audio applications in VR, AR, and AL. To ensure the tutorial is understandable to the novice audience, some background on spatial hearing fundamentals and different classes of spatial audio reproduction techniques will be briefly introduced. This is followed by an outline of the three key parts of this tutorial that focuses on binaural techniques, especially their connections. In part I, we will address recent advances in rendering natural sound in headphones. Based on a source-medium-receiver model, we analyze the differences between headphone sound reproduction and natural listening, which lead to five categories of signal processing approaches that could be employed to reduce the gap between the two. The five categories are virtualization, sound scene decomposition, individualization, equalization, and head-tracking. At last, an integration of these techniques are discussed and illustrated with an exemplar system (a.k.a., 3D audio headphones) developed at our lab.

In part II, we will discuss natural augmented reality audio. Natural listening in augmented reality requires listener to be aware of surrounding acoustic scene. In augmented reality, virtual sound sources are superimposed with the real world such that listeners are able to connect with the augmented sound sources seamlessly. Three typical headset systems for augmented reality audio will be presented, including a natural augmented reality (NAR) headset developed at our lab. The NAR headset employs adaptive filtering techniques to adapt to the listener's specific responses, environmental characteristics, and compensate for the headphone response to achieve natural listening in real-time.

In part III, other aspects to augment human listening, i.e., reducing unwanted noise and enhance speech perception, will be discussed. We will present active noise control (ANC) techniques for headsets and discuss how to integrate ANC with sound playback. Moreover, noise reduction and speech enhancement in hearing aids will be presented, with a focus on the spatial information. Furthermore, ANC can also be

incorporated in hearing aids to further reduce the ambient noise.

In the concluding part of the tutorial, we will provide some demonstrations (video and apps) to illustrate some of the advancements in assisted listening and natural sound rendering in headphones, and highlight new trends of signal processing approaches for natural and augmented listening in headsets.

This tutorial is an extension of the APSIPA 2014 Plenary Talk and also includes new work reported in recent publications published in the IEEE Signal Processing Magazine, March 2015 issue on Signal Processing Techniques for Assisted Listening.

Biographies

Woon-Seng Gan received his BEng (1st Class Hons) and PhD degrees, both in Electrical and Electronic Engineering from the University of Strathclyde, UK in 1989 and 1993, respectively. He is currently an Associate Professor and the Head of Information Engineering Division, School of Electrical and Electronic Engineering in Nanyang Technological University. His research interests span a wide and related areas of active noise control, adaptive signal processing, directional sound system, spatial sound processing, and real-time embedded systems. Professor Gan won the Institute of Engineer Singapore (IES) Prestigious Engineering Achievement Award in 2001 for his work on Audio Beam System. He has published more than 230 international refereed journals and conferences, and has granted four US patents. He had co-authored a book on Digital Signal Processors: Architectures, Implementations, and Applications Prentice Hall, 2005). This book had since been translated to chinese for adoption by universities in China. He was also the leading author of a new book on Embedded Signal Processing with the Micro Signal Architecture, (Wiley-IEEE, 2007). A book on Subband Adaptive Filtering: Theory and Implementation was also published by John Wiley in August 2009. His had also co-authored a book chapter in Rick Lyon's latest book on Streamlining Digital Signal Processing: A Trick of the Trade Guidebook, 2nd Edition, published by Wiley-IEEE press, 2012.

Professor Gan is currently a Fellow of the Audio Engineering Society (AES), a Fellow of the Institute of Engineering and Technology (IET), a Senior Member of the IEEE, and a Professional Engineer of Singapore. In 2012, he has been the Series Editor of the new SpringerBriefs in Signal Processing. He is also an Associate Technical Editor of the Journal of Audio Engineering Society (JAES); Associate Editor of the IEEE Transactions on Audio, Speech, and Language Processing (ASLP); Editorial member of the Asia Pacific Signal and Information Processing Association (APSIPA) Transactions on Signal and Information Processing; and Associate Editor of the EURASIP Journal on Audio, Speech and Music Processing. He is a technical committee member of the Design and Implementation of Signal Processing Systems (DiSPS), and the Industry DSP Technology (IDSP) standing committee of the IEEE Signal Processing Society. Professor Gan is currently a member of the Board of Governor of the APSIPA (2013-2014) and also an APSIPA Distinguished Lecturer (2014-15).

Jianjun He received his B.ENG. degree in automation from Nanjing University of Posts and Telecommunications, China in 2011 and is currently pursuing his Ph.D. degree in electrical and electronic engineering at Nanyang Technological University (NTU), Singapore. In 2011, he was working as a general assistant in Nanjing International Center of Entrepreneurs (NICE), building platforms for start-ups from oversea Chinese scholars in Jiangning District, Nanjing, China. Since 2015, he has been a project officer with School of Electrical and Electronic Engineering in NTU. His Ph.D. work has been published in IEEE Signal Processing Magazine, IEEE/ACM Transactions on Audio, Speech, and Language Processing (TASLP), IEEE Signal Processing Letters, and ICASSP, etc. He has been an active reviewer for IEEE TASLP, Journal of Audio Engineering Society, IEICE Transactions on Fundamentals of Electronics, Communications and Computer Sciences, and IET Signal Processing, etc. Aiming at improving humans' listening, his research interests include audio and acoustic signal processing, 3D audio (spatial audio), psychoacoustics, active noise control, source separation, and emerging audio and speech applications. Currently, He is a student member of IEEE and Signal Processing Society (SPS), a member of APSIPA, and an affiliate member of IEEE SPS audio and acoustic technical committee.