TUTORIALS: SUNDAY, SEPTEMBER 4, 2005	
9:30 – 12:30	**14:00 – 17:00**
T1: MOBILE MULTIMEDIA SERVICES **S. Civanlar & A. Şimşek**	**T4:** COLOR IMAGE PROCESSING, **H. J. Trussell**
T2: ULTRA-WIDEBAND AND IMPULSE RADIO FOR WIRELESS COMMUNICATIONS **H. Arslan**	**T5:** AUDIO SIGNAL PROCESSING IN ACOUSTIC ENVIRONMENTS: REVERBERATION MODELLING AND DEREVERBERATION **J. R. Hopgood**
T3: ARCHITECTURES AND TOOLS FOR DIGITAL RIGHTS MANAGEMENT OF MULTIMEDIA CONTENT **A. M. Eskicioğlu**	**T6:** H.264/MPEG-4 Part 10 VIDEO CODING FOR NEXT GENERATION MULTIMEDIA **K. R. Rao**

AUDIO SIGNAL PROCESSING IN ACOUSTIC ENVIRONMENTS: REVERBERATION MODELLING AND DEREVERBERATION by Dr. James R. Hopgood

Abstract : Enhancement of audio signals acquired in acoustic environments presents a number of challenging signal processing problems that are of important practical, numerical, and theoretical interest to both the research community and also industrial applications, including the telecommunications industry. Usually, when audio signals are acquired in an acoustic environment, there is a physical separation between the sources and the microphone. This means that, in any confined acoustic environment, the effect of reverberation leads to a complicated acoustic impulse response (AIR) that distorts the original source. Reverberation causes problems in two major classes of signal processing applications. The first is in automatic speech recognition (ASR), and its variants, where it is found more difficult to identify reverberant natural speech, rather than anechoic or closely coupled speech. This prevents "hands-free" interaction with the computer without the undesirable constraint that the user must carry a microphone on their person near to their mouth. The second class of applications revolves around the desire to dramatically improve the speech quality and intelligibility from devices such as the mobile telephone, 'hands-free' teleconferencing systems and next generation digital hearing aids, by suppressing or reducing the presence of disturbances or distortions, such as those described above, to adequately low levels.

It is therefore of significant importance to investigate the application of signal processing techniques for the enhancement of the quality of speech distorted in an acoustic environment.

The techniques employed to achieve this consist of two stages: first, estimation of the properties of the acoustic environment from the observed distorted speech and, second, given both the observed speech and the acoustic properties of the room, estimation of the original speech. The first stage of this process is a parameter

estimation problem that is discussed elsewhere. In order to achieve the second stage involving enhancement, it is necessary to have an understanding of some basic theoretical acoustic properties that are important for understanding why particular signal models are used in audio restoration methods.

Main objectives: This tutorial provides an introduction to acoustic modelling, and discusses the following practical aspects:
1. An overview of the basic theoretical acoustic properties of rooms.
2. The suitability of well-known modelling techniques for the representation of room acoustics, their robustness to variations in the source and observer position, and the effect of parameter variation on the accuracy of the model. These include the pole-zero, all-zero, all-pole and common-acoustical pole and zero models.
3. Existing approaches for the enhancement of reverberant speech, notably the least-squares and homomorphic techniques.
4. The issue of the contribution of nonminimum-phase to the perception of reverberation.
5. Subband modelling of room acoustics.
6. The problems associated with dereverberating speech.

James Hopgood
James Hopgood received the M.A., M.Eng. degree in Electrical and Information Sciences in 1997 and a Ph.D. in July 2001 in Statistical Signal Processing, part of Information Engineering, both from the University of Cambridge . His thesis was entitled "Nonstationary Signal Processing with Application to Reverberation Cancellation in Acoustic Environments", and was concerned with the enhancement of audio in applications such as the mobile telephone, heads-free tele-conferencing systems, next generation digital hearing aids, and automatic speech recognition. James was a research associate for the year after his Ph.D, at which point he became a Research Fellow at Queens College continuing his research in the Signal Processing Laboratory in Cambridge. Since April 2004, James has been a lecturer in the Signals and Systems Group, in the Institute for Digital Communications, of the School of Engineering and Electronics, at the University of Edinburgh . His research interests include nonstationary signal processing, acoustic reverberation cancellation, single channel signal separation, ultrasound image restoration, and statistical image processing.

ARCHITECTURES AND TOOLS FOR DIGITAL RIGHTS MANAGEMENT OF MULTIMEDIA CONTENT by Dr. Ahmet M. Eskicioglu

Abstract: In recent years, advances in digital technologies have created significant changes in the way we reproduce, distribute and market intellectual property (IP). Digital media can now be exploited by the IP owners to develop new and innovative business models for their products and services. The lowered cost of reproduction, storage and distribution, however, also invites much motivation for large-scale commercial infringement. In a world where piracy is a growing potential threat, the rights of the IP owners can be protected using three complementary weapons: Technology, legislation, and business models. Because of the diversity of IP (ranging

from ebooks to songs and movies), no single solution is applicable to the protection of multimedia products in distribution networks.

IP is created as a result of intellectual activities in the industrial, scientific, literary and artistic fields. It is divided into two general categories: (1) Industrial property - includes inventions (patents), trademarks, industrial designs, and geographic indications of source, and (2) Copyright - includes literary and artistic works such as novels, poems and plays, films, musical works, artistic works such as drawings, paintings, photographs and sculptures, and architectural designs.

A digital home network is a cluster of digital audio/visual (A/V) devices including set-top boxes, TVs, VCRs, DVD players, and general-purpose computing devices such as personal computers. Copyrighted digital multimedia content may be delivered to the consumers from a number of sources including the Internet, and satellite, terrestrial or cable television systems. It may also be made available as prepackaged media (e.g., a digital tape or a digital video disc) at retail stores. Before releasing their content for distribution, the content owners may require protection by specifying certain access conditions and digital rights. Although legal institutions exist for protecting intellectual property, complimentary technical measures are needed to sustain financial returns and to ensure incentives for new creations. Recently, two fundamental groups of technologies, encryption and watermarking, have been identified for protecting copyrighted multimedia content in digital distribution networks. Encryption-based technologies transform content into unintelligible form. This transformation, being reversible in nature, allows perfect recovery of content before consumption. Technologies based on watermarking embed data directly into content, resulting in imperceptible degradation in visual quality.

End-to-end security is the most critical requirement for the creation of new digital markets where copyrighted multimedia content is a key product. Three major industries have a vital interest in this problem: The motion picture industry, the consumer electronics (CE) industry, and the information technology (IT) industry. This tutorial is an overview of the work done for protecting the content owners' investment in intellectual property. After an introduction to copyright and copyright industries, we examine how the technological, legal, and business solutions help maintain the incentive to supply the lifeblood of the markets.

Objectives of the tutorial: To understand the need and complexity of multimedia content protection, and to learn about the approaches and techniques used for protecting multimedia elements in digital distribution systems such as satellite, cable and terrestrial television networks, the Internet, and home networks.

Ahmet M. Eskicioglu

Ahmet M. Eskicioglu received the B.S. degree from the Middle East Technical University (METU), Ankara, Turkey, and the M.S. and Ph.D. degrees from the University of Manchester Institute of Science and Technology (UMIST), England . He was with the Computer Engineering Department, METU from 1983 to 1992, the Department of Computer Sciences, University of North Texas from 1992 to 1995, and Thomson Multimedia Corporate Research, Indianapolis from 1996 to 2001.

Dr. Eskicioglu is with the Department of Computer and Information Science, Brooklyn College of the City University of New York. He has actively participated in the development of several national and international standards for copy protection and conditional access in the US and Europe. These include the Content Scramble System (CSS) for DVD players, the Advanced Television Systems Committee (ATSC) conditional access system architecture, the Electronics Industries Alliance (EIA) National Renewable Security Standard (NRSS), and the European Union's Digital Video Broadcasting (DVB) Content Protection and Copy Management (CPCM) System. While in the industry, he was the chair of Consumer Electronics Association (CEA) Working Group on Copy Protection, a member of the Copy Protection Technical Working Group (CPTWG), Advanced Television Systems Committee (ATSC) T3/S8 Conditional Access Ad Hoc Working Group, EIA and National Cable Television Association (NCTA) Joint Engineering Committee National Renewable Security Standard (NRSS) Subcommittee, and Society of Cable Telecommunications Engineers (SCTE) Digital Video Subcommittee.

Dr. Eskicioglu's teaching and research interests include data security, conditional access, digital rights management, copy protection, digital watermarking, and multimedia applications. He has been a National Science Foundation panelist, and a guest lecturer at several universities and research organizations.

H.264/MPEG-4 Part 10 VIDEO CODING FOR NEXT GENERATION MULTIMEDIA by Dr. K. R. Rao

Abstract: The video coding standards developed to date by ISO/IEC and ITU-T have not been able to address all the needs required by varying bit rates of different applications and at the same time meeting the quality requirements. An emerging video coding standard named H.264/MPEG-4 part 10 (International standard by end of 2003) aims at coding video sequences at approximately half the bit rate compared to MPEG-2 at the same quality. It also aims at having significant improvements in coding efficiency, error robustness and network friendliness. It makes use of better prediction methods for Intra (I), Predictive (P) and Bi-predictive (B) frames. Arbitrary Block-size Transform (ABT) is used which is a simplified transform that avoids the mismatch error (DCT/IDCT) observed in the motion compensation hybrid coding adopted in MPEG-1 and MPEG-2. All these features along with others such as CABAC (Context Based Adaptive Binary Arithmetic Coding) have resulted in having a 2:1 coding gain over MPEG-2 at the cost of increased complexity. This emerging standard addresses various services/applications, transmission networks, enhanced efficiency, and diverse range of bit rates and spatial/temporal resolutions through profiles and layers. Parameter set concept, arbitrary slice ordering, flexible macroblock structure, redundant pictures, switched predictive and switched intra pictures have contributed to error resilience/robustness of this standard. Adaptive (directional) intra prediction, multiple reference pictures/frames for motion estimation and wei gh ted motion compensated (MC), variable block-size MC, deblocking filter, hierarchical block transform etc., have contributed to the hi gh coding efficiency of this standard developed jointly by the ITU-T Video coding Experts Group (VCEG) and the IS0/IEC Moving Picture Experts Group (MPEG). This combined group is called Joint Video Team (JVT). Other parts of this standard such as file format, verification testing, reference software, conformance bit streams, standardizing

example encoding description and potential extensions are being finalized soon. The tutorial highlights the various functionalities of the encoder, points out the differences between this new standard and the existing standards and describes the state-of-the-art development by the industry. ftp and web sites related to standard documents, software, databases, conformance bit streams, meeting schedules, vendors, file formats, research groups, faq etc are provided. This standard opens up several research areas based on software/hardware implementations, improvements etc.

The new standard addresses various applications such as video streaming over the internet, conversational services such as videophone/video conferencing over wired and wireless (mobile) networks, video-on-demand, near video-on-demand, multimedia messaging, pay-per-view, digital TV, HDTV, super HDTV, digital cinema, hi gh quality video transmission over cable, cable modem, DSL, satellite and terrestrial channels, high density optical storage media such as DVD, digital cameras, camcorders and related consumer electronics products. This emerging standard enables multimedia services/systems viable, feasible, practical, affordable and user friendly. Through various network layers/protocols, profiles and levels this standard is designed to meet the ever increasing needs of current and emerging multimedia applications. The innovative and ingenious approach adopted in this multidimensional signal processing has led to 2:1 bit rate reduction while maintaining the same high visual quality as MPEG-2. Several start-ups as well as established/reputed multinational companies, research institutes etc have embarked on ambitious projects for developing/marketing consumer oriented products based on this standard. It is projected that this will be a multibillion dollar market within the next few years. Latest developments under fidelity range extensions (FRExt, Amendment 1, approved in July 2004) are:

- Extension to 4:2:2 and 4:4:4 chroma formats
- 9 and 10 bit resolutions
- 10 and 11 bit resolutions
- Scalable coding, Lossless coding for digital cinema application
- High fidelity coding for the next generation optical discs
- 8x8 and 4x4 adaptive integer DCT
- Source editing functions such as alpha blending
- Encoder specified HVS weighted quantization
- Lossless coding of specific regions in video content
- Separate Cb and Cr QP control
- Residual color transform for 4:4:4 format

FRExt. Has led to four new profiles called High profiles. These are seriously being considered for application specifications such as:

- HD-DVD specification by the DVD Forum
- BD-ROM Video specification by the Blu-Ray Disc association
- DVB standards for European broadcast television
- ATSC (US)
- Various designs for satellite and cable TV

See papers:

• G. J. Sullivan, P. Topiwala and A. Luthra “The H.264/AVC advanced video coding standard: Overview and introduction to the fidelity range extensions”, SPIE Conf. on applications of digital image processing XXVII, vol. 5558, pp. 53-74, Aug. 2004,
• H. Schwartz, D. Marpe and T. Wiegand, “ SNR–scalable extension of H.264/AVC”, ICIP 2004, Singapore, Oct. 2004

Other extensions under development are:
• Standard systems and file format support specifications
• Standardizing reference software implementation
• Standardizing conformance bitstreams and specifications

K. R. Rao
K. R. Rao received the Ph. D. degree in electrical engineering from The University of New Mexico, Albuquerque in 1966. Since 1966, he has been with the University of Texas at Arlington where he is currently a professor of electrical engineering. He, along with two other researchers, introduced the Discrete Cosine Transform in 1975 which has since become very popular in digital signal processing. He is the co-author of the books “ Orthogonal Transforms for Digital Signal Processing” (Springer-Verlag, 1975), Also recorded for the blind in Braille by the Royal Institute for the blind. “Fast Transforms: Analyses and Applications” (Academic Press, 1982), “Discrete Cosine Transform-Algorithms, Advantages, Applications” (Academic Press, 1990). He has edited a benchmark volume, “Discrete Transforms and Their Applications” (Van Nostrand Reinhold, 1985). He has coedited a benchmark volume, “Teleconferencing” (Van Nostrand Reinhold, 1985). He is co-author of the books, “Techniques and standards for Image/Video/Audio Coding” (Prentice Hall) 1996 “Packet video communications over ATM networks” (Prentice Hall) 2000 and “Multimedia communication systems” (Prentice Hall) 2002. He has coedited a handbook “ The transform and data compression handbook,” (CRC Press, 2001). Some of his books have been translated into Japanese, Chinese, Korean and Russian and also published as Asian editions. He has been an external examiner for graduate students from Universities in Australia, Canada, Hong Kong, India, Singapore and Taiwan. He has conducted workshops/tutorials on video/audio coding/standards worldwide. He has supervised several students at the Masters and Doctoral levels. He has published extensively in refereed journals and has been a consultant to industry, research institutes and academia. He is a Fellow of the IEEE.

COLOR IMAGE PROCESSING : THEORY AND PRACTICE by Dr. H. Joel Trussell

Abstract: Color images are a fundamental part of daily life in academics, business, manufacturing and leisure. It is difficulty to find monochrome monitors for computers or black and white photographs in magazines. Yet, the actual definition, description and meaning of color remains only vaguely understood by most engineers. For example, most technical people know that color is often represented by a triplet of red, green and blue values, (RGB), but may not be clear about how those values have been obtained, transformed or their relationship to the perception of the displayed color.

This tutorial will acquaint the participant with the fundamentals of colorimetry, the science of measuring color, and its application in imaging. We will describe the color characteristics of common imaging input and output devices. The advantages and limitations of the devices and the various forms of color information will be discussed. An important topic is the practical communication of color information and its implementation in a multimedia environment, where various types of input devices need to interface with various types of output devices. The tutorial will indicate current problems, possible research areas and where advanced signal processing techniques can be used effectively.

Joel Trussell
Joel Trussell joined the faculty of N.C. State University in 1980 having worked for 11 years in image processing at Los Alamos National Laboratory. Joel's research interests include estimation theory, color imaging, signal and image restoration and reconstruction, and new mathematical techniques applied to signal processing. Specific applications include color measurement and reproduction, image restoration, system characterization, and improved signal measurement.

Joel's work in the area of estimation theory has resulted in applications and publications in color science, image and speech processing, seismic signal processing, adaptive filters, power line communications, computer tomography and magnetic resonance imaging.

MOBILE MULTIMEDIA SERVICES by Dr. Seyhan Civanlar & Altug Simsek and sponsored by Argela Technologies

Abstract : This tutorial covers the mobile operator's evolving new messaging services architecture and describes how multimedia components fit into the framework. It provides an overview of the evolving 3rd generation (3GPP) mobile infrastructure as the enabler of mobile video. It provides the key features of a video services infrastructure that overcomes the limitations of the current mobile networks and handsets. Finally, it details the technical challenges facing the ecosystem of a multitude of video and messaging content providers, aggregators and the mobile operator for service delivery. The tutorial will have several video related demo components as it pertains to mobile applications.

Seyhan Civanlar
Dr. Civanlar is the VP of Technology at Argela. Prior to Argela, she was the CEO and Founder of Lemur Networks, and Vice President of Technology at Coreon, both startups servicing the US operators. Dr. Civanlar worked in AT&T Bell Labs for 14 years leading several key networking and telecommunications services initiatives in data networking. She has over 20 patents and 80 publications. She has B.S. and M.S. degrees in Electrical Engineering from Middle East Technical University, Ankara, Turkey, and a Ph.D. degree in Electrical and Computer engineering from NCSU, USA. Seyhan is a Fullbright scholar, member of IEEE, Sigma Xi and Phi Kappa Phi societies. She was recently awarded America's Entrepreneur Honor Roll by the prestigious Harvard Business School. Her email address is : seyhan.civanlar@argela.com.tr.

Altug Simsek

Altug Simsek is the Director of Messaging and Video Technologies at Argela. Prior to Argela, Mr. Simsek was managing software projects for mobile operators in Oksijen Technologies. Mr. Simsek also worked at Nortel Networks in Turkey for 6 years on digital telecommunication systems. He holds both B.S. and M.S. degrees in Electrical Engineering from Bogazici University, Istanbul, Turkey. He is currently working towards his Ph.D. degree at the same university. His academic research interests include intelligent software systems for automation of analog microelectronic system design, and neural network based algorithms. He has two pending patents and several publications. His email address is : altug.simsek@argela.com.tr.

ULTRA-WIDEBAND AND IMPULSE RADIO FOR WIRELESS COMMUNICATIONS by Dr. Huseyin Arslan

Abstract: The high demand for communications anywhere and anytime has been the driving force for the development of wireless services and technologies. Wireless technologies and wireless services have evolved significantly over the last couple of decades, from simple paging to real-time voice communication and recently to very high rate data communications. With the number of users and services increasing along with the demand for high data rate, wireless communication systems need to deploy more efficient methods for communications, as the available spectrum is limited and very expensive.

Impulse radio (IR) based ultra-wideband (UWB) is becoming an attractive solution for wireless communications, particularly for short and medium range applications. The wide bandwidth of UWB offers a capacity much higher than the current narrowband systems. Conventional narrowband communication systems employ radio frequency (RF) carriers of much higher frequency than the information rate to transmit base-band signals. UWB is a carrierless (base-band) transmission. Therefore, it does not require the necessary up/down conversion of conventional communication systems. The result is much simpler and less costly circuitry compared to other systems. Other benefits of UWB include immunity to multi-path effects, high resolution (sub-decimeter range), robustness against eavesdropping, and easier material penetration.

Scope of the tutorial: In this presentation, ultra-wideband technology for low power and high data rate wireless communication systems will be discussed. First, an overview of the technology along with applications, advantages, and research issues will be covered. Then, more focus on UWB digital receiver design including synchronization, channel estimation, correlator and rake reception, narrowband and multi-user interference cancellation as well as multi-access code design will be discussed.

Importance of the tutorial: The proposed tutorial attracts both industry and academia. There are a lot of academically interesting research topics under the proposed tutorial. The importance of these topics is that they provide both practical aspects that are attractive to wireless industry, and they are open to further research and development which attracts researchers in the universities and other institutions.

Huseyin Arslan

Huseyin Arslan has received his PhD. degree in 1998 from Southern Methodist University (SMU), Dallas, Texas. From January 1998 to August 2002, he was with the research group of Ericsson Inc., NC, USA , where he was involved with several project related to 2G and 3G wireless cellular communication systems . Since August 2002, he has been with the Electrical Engineering Dept. of University of South Florida. His research interests are related to advanced signal processing techniques at the physical layer, with cross-layer design for networking adaptivity and Quality of Service (QoS) control. More specifically, he is interested in signal processing techniques for wireless communication systems including modulation and coding, interference cancellation and multi-user signal detection, channel estimation and tracking, equalization, soft information generation, adaptive receiver and transmission technologies etc. He is interested in many forms of wireless technologies including cellular, wireless PAN/LAN/MANs, fixed wireless access, and specialized wireless data networks like wireless sensors networks and wireless telemetry. He has served as technical program committee member, session and symposium organizer in several IEEE conferences. He is editorial board member for Wireless Communication and Mobile Computing journal, and was technical program co-chair of IEEE wireless and microwave conference 2004. Dr. Arslan is a senior member of IEEE.

Dr. Arslan has worked in UWB significantly. He has several publications and editing a book on UWB for Wiley publishing. He is also organizing a special issue on UWB for wireless communications and mobile computing journal.